a RIDER publication

GUIDE TO

AUDIO REPRODUCTION

by

David Fidelman

IKEN

Cat. No. 148

$3.50

GUIDE TO

AUDIO REPRODUCTION

By
David Fidelman

JOHN F. RIDER PUBLISHER, INC.
480 Canal Street • New York 13, New York

Courtesy: National Broadcasting Co.

One goal of good audio reproduction is to make available to wider audiences the beauty of great orchestras, such as the NBC Symphony.

First Edition

Library of Congress Catalog Card Number: 53-9954

Printed in the United States of America

PREFACE

The reproduction of sound has become one of the most active fields of electronics at the present time. It is of particular interest to the amateur as well as the professional experimenter, and a constantly increasing number of experimenters, engineers, and technicians are becoming interested in the design and construction of equipment and systems for sound reproduction. Many radio repairmen are also being called upon to assemble such systems for the general public, and to keep them in repair.

The purpose of this book is to present a complete and basic introduction to the principles and techniques of sound reproduction, so that those interested in it can acquire the necessary background to pursue their interest in this field. The book is intended for those who have some familiarity with the basic principles and components of electronic circuits, but who are not necessarily specialists in electronics or in audio. Radio experimenters, amateurs, servicemen, and engineers in other fields will find this book a complete presentation of the information required to understand, design, and construct sound reproducing systems, and a guide to the selection of items which are to be purchased.

An effort has been made to present the material in such a way that readers with different backgrounds will find it useful. The book starts with the fundamental principles of sound production and reproduction and of audio amplifiers, then progresses to the application of these principles to form complete units and audio systems. All phases of the subject are covered, from the basic theory of sound and musical instruments, to the design and construction of amplifiers and loudspeaker enclosures and their placement in the room. The amount of mathematics has been kept to a minimum, so that the book will be useful to readers without an extensive mathematical background. (Note: For those with somewhat less technical backgrounds and for those who are interested in a complementary book on high fidelity reproduction, the author recommends *"High Fidelity Simplified"* by Harold D. Weiler, published by John F. Rider Publisher, Inc.)

Portions of the text in this book are an expansion of the author's articles which appeared in *Radio and Television News magazine*. The author is grateful to the publisher and editors of that magazine for permission to reprint this material along with some of the illustrations which appeared in the series.

The author also wishes to express his gratitude to Mr. Milton S. Snitzer, managing editor of John F. Rider Publisher, Inc., for his help in reading and correcting the manuscript, and for his many fine suggestions which were incorporated into the text.

November, 1953 D.F.

CONTENTS

1. INTRODUCTION 1

2. HIGH FIDELITY REPRODUCTION OF SOUND 18

3. SOUND REPRODUCING SYSTEMS 25

4. INPUT AND PICKUP UNITS 41

5. AUDIO AMPLIFIER THEORY 68

6. PRACTICAL AUDIO AMPLIFIER CIRCUITS 86

7. A-F NETWORKS AND CORRECTIVE CIRCUITS 106

8. LOUDSPEAKERS AND LOUDSPEAKER ENCLOSURES 134

9. COMPLETE HIGH FIDELITY SYSTEMS 157

10. MAGNETIC RECORDING 188

11. MEASUREMENT OF QUALITY OF AUDIO REPRODUCTION 206

INDEX 229

Chapter 1

INTRODUCTION

Production and Reproduction of Sound

The production of sound is an art which originated at a very early stage in man's progress toward civilization. Speech probably originated shortly after men began to utter sounds, and the first production of music by singing and musical instruments took place not long afterwards. Historical records and studies of contemporary primitive societies show that the earliest instruments were usually drums and simple reed wind instruments. The progress of civilization improved the structure and increased the types of musical instruments until the present advanced stage of development was reached. At the present time not only do we have the highly developed manual and breath instruments, but also new classes of instruments in which the production of sound is aided by modern electronic techniques.

For several thousand years "reproduction of sound" meant the production, by voice or musical instrument, from memory or sketchy instruction, of sounds which had previously been produced by others. With the introduction of adequate systems of musical notation such reproduction of sound and music could be done much more accurately. Since the development of modern broadcasting and recording methods, the reproduction of sound means the exact replaying, at a different place and time, of sound which has previously been produced. The sound is stored on some type of record, or transmitted by some type of transmission system, and when reproduced is almost identical with the original.

With the constant improvement in methods of sound reproduction the demands of the listener have gradually become more critical. The earliest phonographs gave very poor reproduction of recorded sound, but this poor quality was accepted by the listener because of the many cultural and musical advantages offered by even this quality of reproduction. However, because of the obvious need for improvement in quality, there has been a constant technical improvement to the present science of sound reproduction where the differences between the original and the reproduced sound are almost negligible.

Until a few years ago most listeners were satisfied with the quality of the sound being produced by their standard commercial radios, phonographs, and amplifiers; only a relatively small group was interested in true high-fidelity reproduction. This group was composed mainly of audio engineers and technicians whose specialization in the field of sound reproduction had taught them the advantages and increased enjoyment to be gained from better reproduction. The great majority of listeners were quite satisified with the reproduction which could be obtained from small table model receivers with no appreciable low-frequency reproduction, or from large consoles with very little output above about 4,000 or 5,000 cycles per second.

During the past few years many more people have begun to realize that their enjoyment of speech and musical reproduction can be greatly increased when the program material is faithfully reproduced without appreciable audible distortion. There has been a great increase in public interest in good reproduction of sound, and today every music lover and record collector is anxious to obtain the best reproduction he can. At the present time the average layman can, with relatively little difficulty and expense, have a really good high fidelity system which will reproduce sound that is practically identical with the original.

High Fidelity Reproduction

For a number of years there has been considerable discussion concerning what is most to be desired in the reproduction of sound, and what basis to use in judging the results. Many experiments have been performed in attempts to obtain quantitative data that would represent the best type of sound reproduction. Essentially the question is: Should the reproduction system attempt to improve the final output sound or should it reproduce the original exactly?

The only way to answer this question is to find out the overwhelming preferences of large numbers of listeners, since public acceptance is always the ultimate test of any type of endeavor relating to artistic creation. Therefore, all tests which are performed to obtain the answer to this question depend upon statistics obtained from large numbers of listeners, with the preference of the greatest number taken to represent the most desirable type of reproduction.

The earliest tests performed on this subject tended to indicate that the listener preferred reproduction having a restricted frequency range rather than wide-range reproduction. This result seemed to explain why most radio receivers in the home are found to have the tone control knob set for minimum high-frequency response, and to justify the design of radio receivers and audio amplifiers with practically no response above 5,000 cps. However, these tests were inconclusive and their results did not agree with other known facts. Experience shows that when the tonal quality of a musical instrument is not fully pleasing to the public ear, that instrument is gradually changed to make it more acceptable. Modern instruments have reached their present form and have remained as they are for many years because the public ear is satisfied with them. If this were not so, they could readily have been changed to another form.

Further investigations showed that these early tests were not given quite properly in that some important factors were not carefully controlled. Some of the important factors being measured were masked by other spurious factors. Improved tests were performed in which more accurate control was maintained over the conditions of the tests, and the results of these tests were more reasonable. They show that the best sound reproduction is that in which the listener hears as nearly as possible an exact reproduction of the original sound, and that this should not be "improved" upon by the reproduction system. Any differences between the original and the reproduced sound are distortions. These distortions may take various forms, all of which should be avoided (or kept to negligible proportions) in order for the reproduction to be good.

The term "high fidelity" refers to sound reproduction in which the various distortions are kept below the limits which are audible to the great majority of listeners. Considerable time, effort, and expense have been put into studying different methods of accomplishing such reproduction, and at the present time there are numerous ways of accomplishing it. Methods and techniques have been developed until the degree of fidelity with which sound can be reproduced depends only upon the time, effort, and expense which are devoted to the task.

The Nature of Sound

Sound is the sensation produced through the ear when certain vibrations are set up in the surrounding air (or other elastic medium) by a vibrating body. Sound may also be considered to be the vibrations themselves or the vibrational energy that produces this sensation. It has two important characteristics: (1) frequency — the number of vibrations per second, representing pitch; and (2) amplitude or intensity — which determines the loudness.

The sound wave in air consists of periodic changes in pressure in the direction in which the sound is traveling. The front of the wave may start as a compression of the air molecules at the point where the sound is being produced. This compression causes a region of high pressure which pushes the adjacent air particles in an outward direction against the neighboring particles, thus causing the compression to move away from the sound source. While this compression is moving away, the vibrating source is moving in the opposite direction, causing a

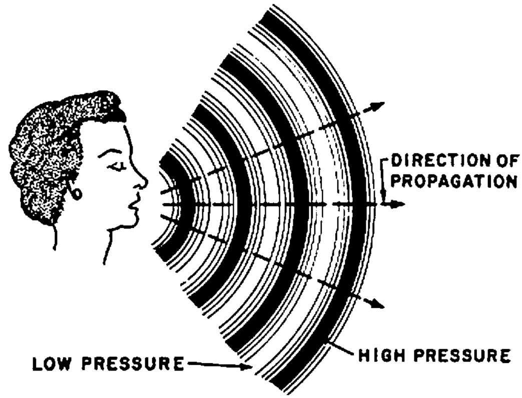


Fig. 1-1. Pictorial representation of the production of sound waves, illustrating the compressions and rarefactions of air.

lowering of air pressure, known as a rarefaction, to follow the compression wave. Another compression follows, then another rarefaction, and this process is repeated at the rate of vibration of the source as long as this vibration continues.

A pictorial representation of the production of sound waves is shown in Fig. 1-1, which illustrates how a sound wave consists of compressions and rarefactions of the air. If a graph of the sound pressure along the direction of propagation is drawn, its variations will be as shown in Fig. 1-2. The variations of pressure moving away from the source of sound in the direction of propagation will have the shape of a sine wave, for a pure tone, with pressures varying above and below normal atmospheric pressure. The air particles themselves do not move with the wave, but merely oscillate back and forth to transmit the pressure variations. Since the particles move back and forth in the direction of sound energy travel, the sound wave is known as a *longitudinal* vibration. (In some other types of wave motions, such as ocean waves and vibrations of a string, the particles which transmit the wave move in a direction at right angles to the direction of propagation; waves of this type are

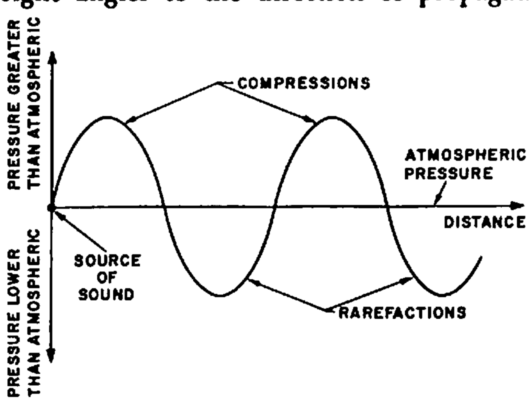


Fig. 1-2. Graph showing the sound pressure variations along the direction of propagation of a typical sound wave.

called *transverse.*) Sound waves can be transmitted in other media besides air; they can travel in any elastic medium in which vibrations occur and the particles return to their original position after the sound has been transmitted.

One complete set of variations starting at one condition and returning once to the same condition comprises one complete cycle. The number of such cycles which occur in one second is the *frequency* of the sound. The greater the number of cycles per second the higher the frequency. The number of cycles produced per second multiplied by the actual physical length of the wave, as measured between corresponding points on two adjacent compressions or rarefactions, gives the velocity of the sound. This is stated mathematically as:

$$v = f\lambda$$

where v is velocity in feet per second, f is frequency in cycles per second, and λ is the wavelength in feet. The velocity of sound in air is approximately 1,100 feet per second.

The difference between the maximum pressure and the normal atmospheric pressure (which is also equal to the difference between atmospheric pressure and the pressure of greatest rarefaction) is the *peak amplitude* of the wave and is a measure of the loudness of the sound. If any point is taken along the wave shown in Fig. 1-2, its distance along the axis represents the *phase* at that point. Since the sound pressure variation is a sine wave in this case, a complete cycle is 360 degrees with the phase measured from zero at atmospheric pressure at the beginning of any positive half-cycle.

Audible sound is produced by any object vibrating at a frequency within the limits audible to the human ear. The exact limits vary widely depending on age and sex, with the greatest frequency ranges generally belonging to women and children. For 50 percent of the average population the audible frequency limits are 20 and 14,000 cycle per second; for about 5 percent of the average population, these limits are about 16 and over 20,000 cycles per second. Frequencies below the lower limit of audibility are called *subsonic,* while frequencies above the upper limit of audibility are called *supersonic* or *ultrasonic.*

A number of familiar vibrating objects which can produce sound are shown in Fig. 1-3. If a string supported at both ends is plucked the impulse travels along the string to the point of support, where its phase is reversed. It then travels back to the other point of support where it is again reversed in phase. Because of the addition of the waves traveling back and forth along the string between the two points of support, the strongest vibration of the string will occur at the frequency whose corresponding half-wavelength is equal to the length of the string. This

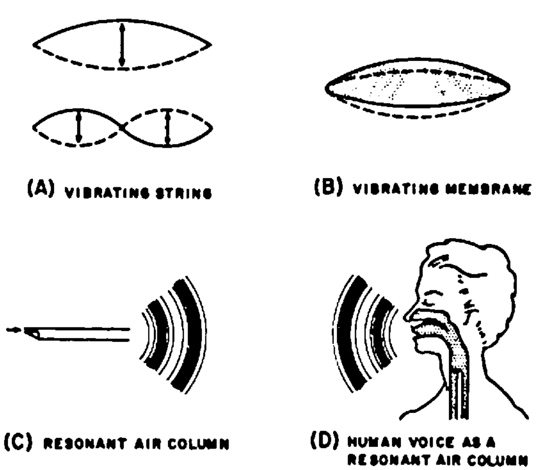


Fig. 1-3. Familiar vibrating objects which are capable of producing sound.

type of vibrating motion of a string is the basis of the sound production of all musical stringed instruments, whether they be plucked, bowed, or struck by a hammer. Similar vibrations can also be produced by a stretched membrane where the vibrations can travel in any direction along the membrane. This type of sound production is represented by the various types of drums. Another common type of vibrating object is the reed, which generally excites a resonant column of air that produces the actual sound which is heard. A column of air in a tube acts effectively for sound waves in the way a string does for transverse vibrations, and the frequency of resonance is changed by changing the length of the air column. The wind instruments produce their sound by exciting such a resonant column of air and by changing the length of this column either by holes in the tube or by using telescoping tubes of variable length.

Any resonant vibrating element like a string or open air column will support a strong vibration whose half-wavelength is equal to the length of the string, and will also support any higher frequency vibrations of which an integral number of half-wavelengths equals the length of the string. The lowest frequency vibration is the one for which the resonant structure is a half-wavelength, and is called the *fundamental* vibration. The higher frequency vibrations, in which more than one half-wavelength is contained in the same length as the fundamental, are called *harmonics* or *overtones*. The frequency of the second harmonic is twice that of the fundamental, the third harmonic three times that of the fundamental, etc.

If a string, a membrane, or an air column is caused to vibrate, the waves which are set up along its length will not be purely the fundamental, but will also contain appreciable amounts of the various harmonics. The harmonic content of the sound which is produced depends upon a number of factors, such as the construction of the musical instrument, the type of sounding board, the point at which a string is set into vibration, or the manner in which an air column is caused to vibrate. This harmonic content determines the *quality* of the sound, and is the characteristic which distinguishes the tone of one musical instrument from another.

If a graph of the sound pressure variations is drawn for a sound having an appreciable quantity of harmonics, the curve will no longer be a pure sine wave as shown in Fig. 1-2. Instead it will consist of the sum of a number of sine waves of different frequencies, amplitudes, and phases which add together to produce a complex wave shape. The difference in a wave shape of different sounds may best be understood from

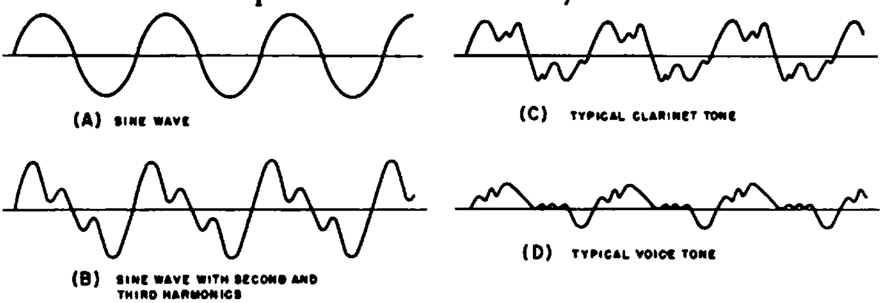


Fig. 1-4. Typical wave shapes containing different amounts of harmonics.

the curves in Fig. 1-4, which show the type of pressure waves obtained from the same fundamental frequency with different harmonic contents. The important differences between these curves, and the differences in the tonal quality which they represent, are due to their harmonic content, which may be of much higher frequency than the fundamental. In the reproduction of such sounds, the amplitude and phase relations of all the higher frequency harmonics must be maintained in order to reproduce accurately the tonal quality as well as the pitch and amplitude.

The Human Ear

The mechanism of hearing is of fundamental importance in the reproduction of sound, since the capabilities and limitations of the ear determine the requirements of the reproduction.

The ear picks up the sound vibrating in the air and transmits nerve impulses to the brain, which analyzes the sound into its various components. The ear is divided into three chambers as shown in Fig. 1-5. The outer ear is a short, narrow tube closed at the inner end by an elastic membrane, the ear drum. Any pressure wave in the air which reaches the ear is transmitted through the outer ear to the ear drum. Behind the ear drum is a small cavity containing three small bones (hammer, anvil,

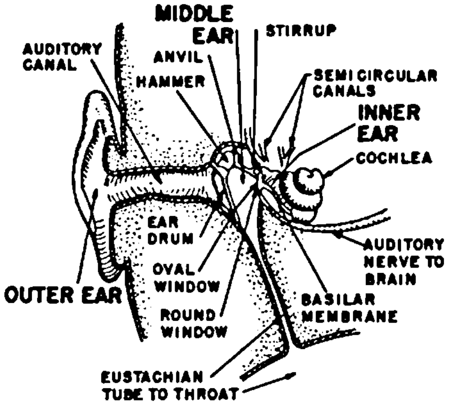


Fig. 1-5. Internal construction of the human ear.

and stirrup) linked together to form a system for transmitting vibrations of the ear drum to the oval window of the inner ear. The inner ear consists of a coiled spiral (cochlea) with bony walls containing a number of small hairs which are the ends of the auditory nerves which transmit the vibrations to the brain.

If the spiral of the inner ear were straightened out its appearance would be as shown in Fig. 1-6. It is essentially an acoustical transmission line divided along its length by a membrane along which lie the auditory nerve endings. The sound vibrations are transmitted along its length by the fluid with which it is filled. Each of the nerves along the membrane is resonant at a different frequency, and responds to sound of that particular frequency; therefore the brain knows the frequency by knowing which particular nerve is excited by the sound. The nerves which respond to high-frequency sound are located at the end of the membrane closest to the oval window where the sound enters, while those which respond to the low frequencies are located at the far end away from the oval window. In Fig. 1-6 the approximate frequencies of response at various distances along the membrane are indicated by the scale along the top; the actual physical dimensions of the inner ear are shown by the scale drawn along the bottom. Since the various sections along the

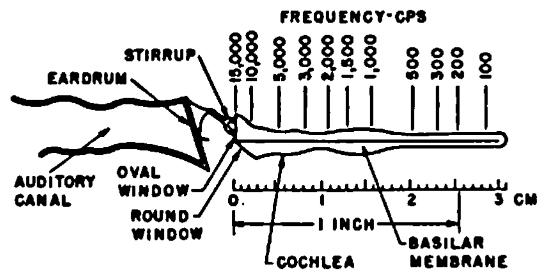
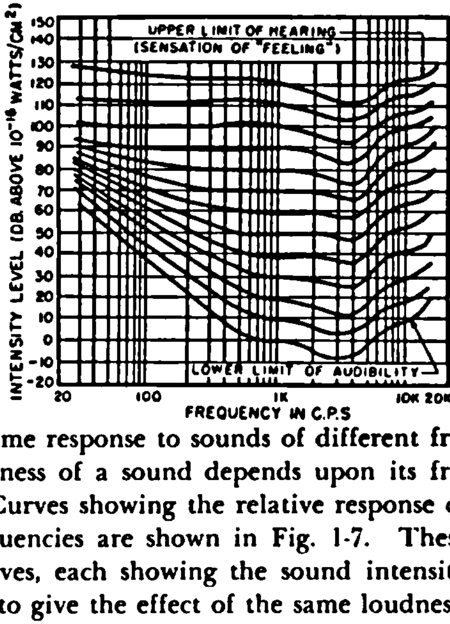


Fig. 1-6. The structure of the ear, with the inner ear straightened out and simplified. Actual physical dimensions are indicated by the scale of centimeters underneath, and the frequency scale indicates the distances along the basilar membrane which respond to the various frequencies.

membrane are resonant structures, small differences in frequency are distinguished from one another by the phase of the response.

The loudness of the sound determines the amplitude of the vibration in the fluid of the inner ear, and therefore the degree of response of the auditory nerves. The response to sounds of different intensities is proportional to the *logarithm* of the intensity, rather than directly to the intensity. Thus, if there are three sounds with relative intensities of 1, 10, and 100 units, the ear will perceive the same relative difference in intensity between 1- and 10-unit sounds as between 10- and 100-unit sounds, even though on an absolute intensity basis the difference of 90 between 10 and 100 is much greater than the difference of 9 between 1 and 10. The ear, however, is interested only in the fact that there is a ratio of 10 to 1 between the two sounds in each case.

Fig. 1-7. Relative response of the ear to different frequencies at various loudness levels. Each curve shows the variation with the frequency of sound intensity required to give the effect of the same loudness.



The ear does not have the same response to sounds of different frequencies, and the apparent loudness of a sound depends upon its frequency as well as its intensity. Curves showing the relative response of the average ear to different frequencies are shown in Fig. 1-7. These are experimentally measured curves, each showing the sound intensity required at different frequencies to give the effect of the same loudness. They show that fairly high loudness levels of 80 to 100 db are heard almost uniformly over the entire audio range, but at lower levels the ear is much less sensitive to the low and high frequencies than it is to the middle frequencies. Thus, a barely audible sound at a frequency of 80 cycles per second will have to be more than 100 times (20 db) as intense as a barely audible sound at 1,000 cps.

The Decibel and the Logarithmic Frequency Scale

The property of the ear to respond to ratios of sound intensity is expressed mathematically by the use of the logarithmic ratio called the *decibel.* The decibel is defined as a ratio between two powers, and this definition can also be used to express ratios of voltages, currents, and intensities:

$$\text{Decibels (db)} = 10 \quad \log_{10} \quad \frac{P2}{P1}$$

$$= 20 \quad \log_{10} \quad \frac{E2}{E1}$$

The first equation expresses the difference in decibels between two different powers (P2 and P1), while the second expresses the difference in decibels between two different voltages (E2 and E1) or intensities. A value of 0 db means that the two powers or intensities are equal.

The decibel is the most widely used unit in audio engineering because it represents so accurately the response of the ear to different intensities, and because it can be used to express a wide range of intensities on a very useful scale. However, it should be clearly understood that the decibel is only a unit comparing any two levels; it does not give any absolute value for either. Thus, 20 db means a 10 to 1 ratio of intensities, but does not tell whether the intensities are 1, 10, 100, 1000, or any other specific values. When graphs are plotted in which different intensities are represented in decibels, some arbitrary intensity is taken as the reference value of 0 db, and the others are plotted relative to this reference. To relate the graph to absolute intensities, the actual measured value of 0 db must be included. For example, when sound intensities are being discussed, a common reference level is taken as 10^{-16} watt/cm^2, and for power levels in audio circuits a common reference level for 0 db is 1 milliwatt across a 600-ohm impedance. The abbreviation "dbm" is frequently used in audio work to indicate a power expressed in decibels with a reference level of 1 milliwatt.

When curves or graphs of audio frequency response are drawn, such as those in Fig. 1-7, a logarithmically spaced frequency scale is used along the horizontal axis rather than a linear one. The reasons for this type of scale are that the ratios of the frequencies are important, and a very wide range of frequencies must be shown on the graph. When the logarithmic frequency scale is used, the response at frequencies as low as 20 cps can be indicated as clearly as the response at frequencies as high as 20,000 cps. This is not true of a linear frequency scale. In addition, a constant frequency ratio is represented by the same distance between any points on the curve, so that harmonic frequency relations can be indicated as clearly at low frequencies as at high frequencies.

Since the decibel and the logarithmic frequency scale represent the response of the ear to different sounds, audio-frequency graphs are generally drawn in a way which takes these factors into account. They are drawn on semi-logarithmic graph paper, on which the frequency scale is logarithmic and the amplitude scale is linear. Frequencies are then indicated directly, and amplitudes are shown in decibels.

Characteristics of Musical Sounds and Speech

The three important characteristics of sound are *amplitude, frequency,* and *quality*. If these three characteristics of a sound are known, its nature can be described. Sounds which have no regular frequency of vibration are called *noise,* and are disagreeable to the ear because of their lack of definite frequency or pitch. Musical sounds are more pleasing to the ear because of the regularity of their vibration.

The vibrations of musical sounds are produced by setting into vibration a resonant structure, such as a stretched string or membrane, or a resonant column of air. The vibrating element generates the sound, supplies the energy, and also generally determines the frequency of the sound (although other parts of the instrument may serve to keep this frequency constant). The vibrations originating in the generator are usually amplified in musical instruments, either by resonance or by forced vibrations in other parts of the instrument.

Stringed Instruments. In stringed instruments the sound is generated by causing a stretched string to vibrate by either plucking, bowing, or striking. The frequency of vibration is determined by the *length, tension,* and *mass* of the string. Frequency is inversely proportional to the length, directly proportional to the square root of the tension, and inversely proportional to the square root of the mass, according to the formula:

$$f = \frac{1}{2L}\sqrt{\frac{T}{m}}$$

where f=frequency, L=length, T=tension, and m=mass.
Therefore the low-frequency notes are produced by long, loose, heavy strings, while the high-frequency notes are produced by short, tightly stretched, light strings. The strings are tuned to exactly the desired frequency by adjusting the tension. In instruments such as the violin, guitar, and cello, a wide range of frequencies is produced with a few strings by pressing down with the fingers at the appropriate points on the string to shorten the vibrating length and therefore increase the frequency. In the harp, pedals are used to change the tension of the strings to produce different frequencies.

The manner in which the string is set into vibration is the main factor determining the quality of the sound produced by stringed instruments. The strings of a piano are struck at one end by felt covered hammers to give a damped oscillation. The violin string is bowed to give a steadily sustained tone as long as the bow is being drawn across the string. The strings of the harp are plucked at the center to give a damped oscillation whose sounds differ from that of the piano.

The manner in which the sound of the string is amplified depends upon the construction of the individual instrument. In instruments such as the violin, cello, and guitar the vibrations of the strings cause the entire body of the instrument to vibrate and amplify the sound. The piano has a special large thin sheet of wood, called the sounding board, located underneath the strings to amplify their sound. The sound which is heard from a stringed instrument comes from the large amplifying surface, and not directly from the string; without this amplification the instrument would be barely audible. The amplifying surface itself has no resonances within the frequency range of the instrument, since it must be equally responsive over the entire frequency range. Therefore, it amplifies the sound only by being set into forced vibration by the strings.

Wind Instruments. Wind instruments produce sound by setting into vibration a resonant column of air. They fall into two separate groups: those in which nothing vibrates but the air, and those which have a mechanical vibrator such as a reed or the player's lips. The sound is generated either by a thin stream of air or by the motion of the reed; although the generator supplies the sound energy, it usually has no definite resonant frequency of its own. The air column has a fundamental resonant frequency and a number of higher harmonic resonances, therefore it vibrates very strongly at any of those frequencies which may be present in the generator's vibrations. The large volume of air inside a tube is thus set into vibration, and sends out a good volume of sound into the outer air. The air column, therefore, determines the frequency of the note and also acts as the amplifier. Different notes are obtained by selecting different harmonics of the air column or by changing its length. Changing the length is accomplished either by opening holes in the side of the tube (as with flutes and clarinets), by inserting additional lengths of tubing with valves (as with cornets and French horns), or with sliding tubes (as with slide trombones).

The resonant frequency of the air column is determined by its length and by whether the ends of the tube are open to the air or closed. When the tube is open to the air at both ends the lowest frequency of resonance is that at which a half-wavelength is equal to the length of the tube. When the tube is open at one end and closed at the other the lowest frequency of resonance is that at which a quarter-wavelength is equal to the length of the tube. The lowest frequency of the open-ended tube is thus twice as high as that of the tube with one end closed. Harmonics of the fundamental resonant frequency are, of course, also resonant frequencies of the tube.

The human voice is a wind instrument of the reed type. The sound is generated by the vocal chords, whose frequency can be varied by changes in their tension, length, and thickness. The energy in the sound is supplied by the steady flow of air from the lungs over the vocal chords and through the mouth and nose cavities. The quality of the tones is determined by the position of the lips and the shape of the mouth and nose cavities. The vocal mechanism can produce an extremely great number of sounds differing in frequency, quality, loudness, duration, growth, and decay.

Musical Scales. Musical sounds are generally considered to represent a pleasing combination of sounds. A combination of two frequencies will sound pleasing to the ear if the ratio of their frequencies can be expressed by small numbers such as 1:2, or 2:3, or 3:4. The sound wave caused by these combinations of frequencies has a simple waveshape, as can be seen from Fig. 1-4, and the ear seems to prefer simple waveshapes. The ear recognizes a marked similarity between tones related in frequency by 2:1. Such notes are said to be an octave apart, and are denoted by the same letter or name in musical notation. The reason they sound so much alike separately, and so well together, is probably because they occur together so commonly in the overtones produced by musical instruments that the ear has become accustomed to them.

The frequency range in an octave is divided into smaller intervals to form the musical scales. The musical scale which has the most musically pleasing combination of notes is the *diatonic scale* in which the frequencies of the various notes are related according to the sequence:

$$\begin{array}{ccccccccccccccc} C & & D & & E & & F & & G & & A & & B & & C \\ 1 & : & \frac{9}{8} & : & \frac{5}{4} & : & \frac{4}{3} & : & \frac{3}{2} & : & \frac{5}{3} & : & \frac{15}{8} & : & 2 \end{array}$$

The *major chords* in this scale are composed of three tones whose frequencies are in the ratio of 4:5:6, and the *minor chords* are tones whose frequencies are in the ratio of 10:12:15. Most of the notes of this scale form exact and simple ratios with one another.

A typical scale based on these frequency ratios is shown in Table 1-I, which shows the key of C Major and the frequencies of the various notes based on the diatonic scale. Suppose the musical scale starts on the second note of this scale – to form the key of D. The frequencies of this key are seen to require the addition of two new notes – one between F and G, the other between C and D. If musical scales are based on

	FREQUENCY – Cycles Per Second										
	C		D		E	F		G		A	B	C′
Scale based on C	256		288		320	341		384		426	480	512
Scale based on D		270	288		324		360	384		432	480	
Scale based on E		268		300	320		360		400	426	480	

Table 1-I. Frequencies of the notes in the diatonic scale in the keys of C, D, and E Major based on C of 256 cps, showing the different frequencies required by the use of the diatonic scale.

each note in succession, altogether five new notes are needed. The complete scale containing these five additional notes is called the *chromatic scale.* However, even if these five new notes are added to the scale, the frequency ratios would still be incorrect with changes in key, as can be seen by comparison of the required frequencies in the key of C and D shown in Table 1-I. To obtain correct frequency ratios for all the changes in key would require about 40 or more notes in each octave.

Since it is obviously impractical to have the extremely large number of notes required by the diatonic scale, especially for instruments with notes fixed in frequency, a compromise has been adopted. From Table 1-I it can be seen that for changes in key the musical scale consists of twelve intervals. If each of these intervals is approximated by the twelfth root of 2, an approximation of the correct intervals results, and the octave is then made up of 13 notes, each of which is $\sqrt[12]{2}$ or 1.06 times the frequency of the next lower one. This scale based on equal intervals (i.e., a constant factor of multiplication) is known as the *equally tempered scale,* and is the scale actually used in music.

A comparison of the equally tempered and the diatonic scales is shown in Table 1-II. (The reference frequency in Table 1-II is A=440 cps, which is the American standard pitch.) This scale gives no perfect consonances, but the departures from the ideal frequencies are very small. It has the advantages that the scale can start at any frequency and maintain the same frequency ratios, and that all the keys can be played on a musical instrument using the same notes without requiring

	FREQUENCY – Cycles Per Second							
	C	D	E	F	G	A	B	C′
Freq. on equally-tempered scale	261.6	293.7	329.6	349.2	392	440	493.9	523.2
Freq. on diatonic scale	261.6	294.3	327	348.7	392.4	436.1	490.5	523.2

Table 1-II. Comparison of frequencies of notes in equally tempered and diatonic scale in key of C Major. (Frequencies shown for equally tempered scale are present American standard of 440 cps as frequency for tone A.)

retuning. The 12 intervals are called half-tones and, as can be seen from Table 1-I, there is a half-tone interval between the third and fourth notes of the scale and between the seventh and eighth notes, all the other intervals are whole tones. This relationship is true in all major scales, regardless of the note upon which the scale begins. In minor keys there are several different distributions of half-tone and whole-tone intervals which changes the character of the music played in these keys.

Frequency and Amplitude Ranges of Sounds

When the musician speaks of the *range* of a musical instrument or a voice, he refers to the frequency range of the fundamental frequencies which can be produced. Musical scales and notation also refer to the fundamental frequency, since the fundamental frequency determines the note which is produced, while the overtones add the color and quality to it. The fundamental frequency ranges of voices and various musical instruments are shown in Fig. 1-8. Although there may be some individual variations, the ranges shown here are typical of average voices and instruments.

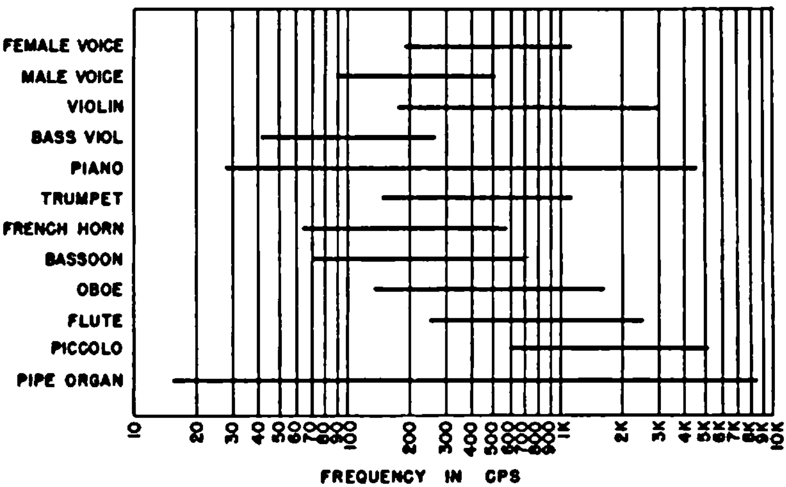


Fig. 1-8. Frequency ranges of the fundamental frequencies of voices and various musical instruments.

The sounds produced by the different musical instruments are distinguished by the overtone structure. If the fundamental were produced with no overtones, all instruments would be producing a pure sine wave identical with that produced by an oscillator, and there would be no differences between them. The range of frequencies present in a single note of a musical instrument therefore extends from the fundamental frequency (which is the lowest frequency present) to the fre-

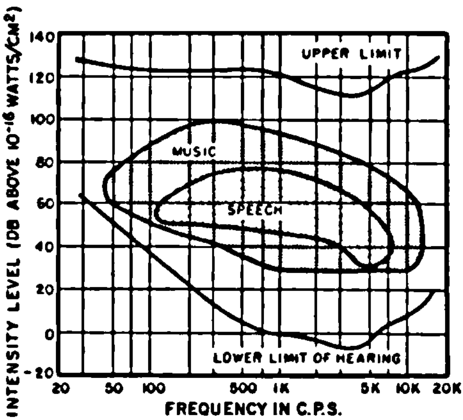


Fig. 1-9. Frequency and volume ranges of speech and orchestral music.

quency of the highest overtone. The complete frequency range of the instrument extends from the fundamental frequency of the lowest note to the highest overtone of the highest note which it can produce. In addition to these musical notes, many instruments produce characteristic noises accompanying the music (such as the scraping of the bow, or breath noise) which must be considered as part of the sound produced by the instrument, and which may often extend to higher frequencies than the overtones of the musical notes. The overtones and the accompanying noise range extend the upper limit of the spectrum by a factor of two or more octaves in many cases.

The frequency and volume ranges of speech and ochestral music are shown in Fig. 1-9. The frequency range of music is from about 40 to

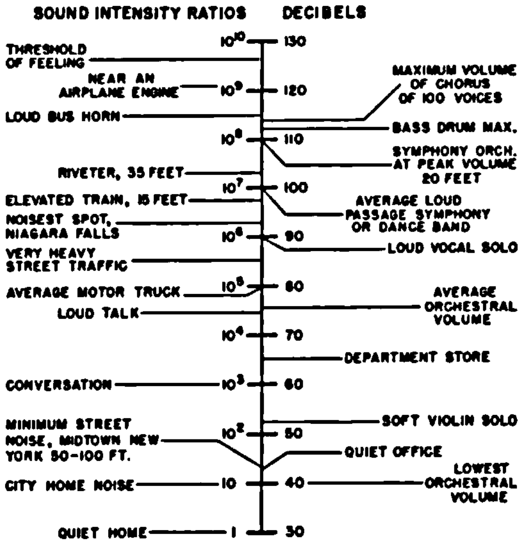


Fig. 1-10. Relative intensities of different types of sounds and noise levels in various locations.

14,000 cps and the range of volumes is about 70 db. Speech covers a much narrower frequency and volume range. The two curves for the limits of hearing are included in Fig. 1-9 for reference. It can be seen from comparison of these curves that the reproduction of speech and music does not require the full frequency and amplitude range of the ear.

The relative amplitude range of sounds which are normally encountered is shown in Fig. 1-10. This chart shows the measured intensity of the different types of sounds and the noise level at various locations. This comparison serves the function of relating the different sound levels to well known and familiar types of sounds, and compares the different types of sounds with the better known ones. It shows also, for example, that the loudness levels in orchestral music cannot be adequately reproduced in a city apartment, because the room noise is as loud as the lowest sounds, and adequate reproduction of the loudest levels would cause equally loud complaints from the neighbors and the management.

Chapter 2

HIGH FIDELITY REPRODUCTION OF SOUND

Limitations and Preferences of the Human Ear

Comparison of the amplitude and frequency ranges of speech, orchestral music, and various other types of sounds with the limits of hearing, as shown in Figs. 1-9 and 1-10, shows that the range of the ear is wider than is needed for the perception of these sounds. Moreover, the ear has a number of other abilities which are not represented by these curves. It is able to analyze complex sounds into their various components, to detect the wave envelope and transient nature, changes in amplitude and frequency, and many other important qualities of the sound to which it is listening. A change in amplitude of about 1 decibel, and changes in frequency of about 3 or 4 cps (over a frequency range from about 100 to 2,000 cps) can be detected by the critical listener. Transient tones can be heard if they last 1/20 of a second or longer. By comparing the difference in intensity and phase of the sound at the two ears, the brain is able to localize the source of the sound with great accuracy.

Because of the abilities of the ear to determine these important characteristics of complex sounds, many factors are important in determining the accuracy and fidelity with which such sounds must be reproduced. The accepted criterion of sound reproduction is that the listener should hear as nearly as possible an exact reproduction of the original sound. Any differences between the reproduced and the original sound are *distortions.* Restricted frequency range in reproduction is one form of distortion, but there are also several others which may occur. The following list includes all the types of distortion which are at present known to have an important effect on the quality of the reproduced sound:

(1) Frequency-amplitude distortion.
(2) Reproduction noise.
(3) Harmonic distortion.
(4) Intermodulation distortion.
(5) Transient distortion.
(6) Phase distortion.
(7) Frequency-modulation distortion (wow and fluttter).

These distortions occur within the reproducing system, and are capable of measurement and analysis. However, there are certain other fundamental limitations which are present in any attempt to reproduce sound. These might properly be called *acoustic* distortions:

(8) Differences in acoustics between the room in which the sound originates and the room in which it is reproduced.

(9) Spatial distribution effects.

(10) Limited dynamic range in reproduction.

The acoustic distortions are not strictly defects of the audio system, since they are inherent in any attempt to reproduce sound from one place to another.

Types of Distortions

Frequency Range. At one time the main criterion of good reproduction was thought to be the width of the frequency range. The "high-fidelity" enthusiast would, for example, judge the excellence of a phonograph system by the amount of high-frequency needle scratch he could hear. Thus, the more high-frequency noise and record scratch, the better the quality of reproduction. This type of judgment has by now become more or less obsolete, since it is now realized that for true high-fidelity reproduction all forms of distortion must be eliminated and the original sound reproduced as accurately as possible in all respects. Some of the other distortions are much more distasteful to the ear than loss of high and low frequencies, and when they are present the listener usually prefers the restricted range which reduces their effects. (This explains the results of the earliest tests in which listeners were found to prefer narrow-band reproduction, since in these tests no attempts were made to remove the various distortions which are related to the frequency range.)

Reproduction Noise. The noise introduced by the reproducing system is generally the most obvious form of distortion, and must be kept as low as possible, both because it limits the dynamic range and because of its disagreeable quality. The highest noise level is usually found in the reproduction of phonograph records, due to the record scratch introduced by the surface of the disc. Other parts of a high-quality system should introduce practically no noise.

Nonlinear Distortions. Harmonic and intermodulation distortion are extremely important factors in most of the equipment in current use. These distortions occur whenever the amplifier is not perfectly linear (that is, the output is not perfectly proportional to the input signal). This nonlinearity gives rise to spurious harmonics which are not present in the original signal, and causes modulation of the high frequencies by the low-frequency components. It is usually the intermodulation rather than the harmonic distortion which is responsible for the disagreeable quality when a system is overloaded, since the intermodulation products generally contain frequencies which are discordant with the original sounds. At the present time these are perhaps the most important and troublesome forms of distortion, since when they are high they tend to mask other distortions because their effects are more unpleasant.

Transient Distortion. When the noise and distortion of the system are within the limits required for good reproduction, then the various other forms of distortion are not masked and become important. It has been found that two systems with identical frequency and non-linear distortion characteristics may often sound quite different when they reproduce speech or music. This difference generally occurs when one of the systems has good transient response, and the other has poor transient response. Very few natural sounds (including speech and music) are sustained for any long period of time — the various components generally occurring in bursts of short duration. This can easily be observed by watching an oscilloscope trace (swept at about 20 to 50 times per second) and noticing how the various components are constantly being generated and disappear. It is, therefore, extremely important that for good reproduction the entire system must have good transient as well as steady-state characteristics, and should have no undamped vibrations which will appear for a dynamic signal and produce undesired hang-over that distorts the effect of the sound. This important factor is generally overlooked by many experimenters and technicians who are not sufficiently familiar with the requirements of good reproduction.

Phase Distortion. This type of distortion takes place when the relative phases of the various components are not the same in the output as in the input of a reproducing system. As a result, a change in waveshape occurs even though the same harmonics are present and in the same magnitude. This has some effect on the sound quality and on the transient response.

Frequency Modulation Distortion. When sound is reproduced from disc or magnetic recordings, the frequency of the output sound is determined by the relative speeds of the recording and reproducing drive motors. When these motors do not run at absolutely constant speed, there is heard a *flutter* or *wow* in the frequency of the reproduced sound which is quite annoying and serious. These effects are apparent in the reproduction of sustained steady tones, where variations in pitch can be most easily detected. Such speed variations can also be heard in the reproduction of percussive tones in disc recordings, where the sharp transients impose very heavy louds upon the driving motor and may cause it to change speed due to this variation in load.

The results of the listener preference tests show that the various electronic distortions must be kept down to inaudible or barely audible levels for good reproduction. However, the situation is different in the case of the acoustic distortions, since many psychological factors enter into the question of listening to sound in a location different from that where it originates. Since listening to the sound of a full orchestra in the average living room is completely different from listening to it in the concert hall, the listener may feel free to go in for considerable experimentation to decide which type of reproduction he prefers for himself. Dynamic range may be increased by the use of volume expanders. Experiments to correct for the difference in spatial distribution of the sound may be performed by several methods which have been tried for obtaining stereophonic reproduction. The reverberation and absorption of sounds of different frequencies can also be controlled by changing the absorptive and reverberant properties of the room, and by means of artificial reverberation systems.

Requirements for High-Fidelity Reproduction

Although the psychological factors involved in the transfer of sound to a new location are often a matter of individual preference, the most logical procedure is to try to make the final effect upon the ear as much as possible like that obtained from the same sound in the location where it originates. Normally, the sound heard from an orchestra in the concert hall is considerably louder than is practical for the average listener to obtain from his loudspeaker in the home, especially if he lives close to his neighbors or in an apartment house. However, the ear does not have the same response for different acoustic levels, therefore some compensation must be made for this difference. Figure 1-7 shows the effect of the intensity level at various frequencies on the ear. Each curve shows the intensity at different frequencies to give the effect of a certain loudness. A loudness level of 80 to 100 db is heard almost uniformly over the entire audio range, but a frequency of 60 cps has to be heard at a 60 db level to have the same loudness as 1,000 cycles at a 20 db level. Thus, there should be considerable bass boost, when listening at normal home levels, to give the same effect as the actual orchestra's performance.

The systems which exist at the present time for the reproduction of sound are, unfortunately, not capable of producing without distortion all the sounds which are required from them. This may be better understood when it is realized that vacuum-tube amplifiers are not strictly linear in their response, sometimes cannot be used over the complete frequency range, and can be overloaded. In addition, the electro-mechanical components of the system, particularly the loudspeaker, are required to duplicate at one time all the sounds which can be produced by every instrument in a large orchestra – which is by no means a simple achievement. Therefore some compromise must be made between the requirement of perfect fidelity of reproduction and the limitations of the reproducing system.

The electronically generated distortions must always be kept to an absolute minimum (except in the case of the intentionally introduced frequency-amplitude distortions, such as by "tone controls"). There are practically no occasions where it would be desirable to accept such distortions as intermodulation, for example, and such distortions are permitted only to a minimum degree since it is a physical impossibility to eliminate them entirely.

By proper design and setup of the system these distortions can be kept below the level at which their effects can be detected by the average listener. Tests and experience have indicated certain distortion levels which may be accepted as the requirements for good reproduction, and the extent to which a system meets these requirements may be considered a measure of the quality of reproduction.

Complete reproduction of all the fundamentals and harmonics which are present in the music produced by a symphony orchestra would require a frequency range of approximately 20-14,000 cycles. However, for almost all practical purposes a frequency range of 40-10,000 cycles, with satisfactory distortion characteristics, has been found quite acceptable to about 90 percent of the listeners for all types of speech and music. While such a reduction of bandwidth will remove some of the highest frequencies present in the original sound, this cannot be detected by large numbers of listeners, and is in general felt to be not objectionable.

For a system of good quality the noise and hum level should be of the order of 50 to 60 db below full output, although the noise level from a phonograph record will generally not be better than about 35 to 40 db below peak signal.

At one time it was believed that 5 percent total harmonic distortion was acceptable in a high quality system, but today it is felt that this figure is too high (especially in amplifiers using beam power tubes or pentodes, since 5 percent total harmonic distortion from them appears to be more objectionable than 5 percent from triodes). Present thinking is that the total harmonic distortion (exclusive of the loudspeaker) should not exceed 1 or 2 percent. From general experience it seems reasonable to rate an amplifier having 10 percent intermodulation distortion with 2 percent harmonic distortion as fair quality, and one having 4 percent intermodulation with 1 percent harmonic distortion as good.

With the addition of a good loudspeaker total harmonic distortion is increased by only 1 or 2 percent, provided the speaker is not overdriven. No general conclusions have yet been derived concerning intermodulation distortion for loudspeakers, although it should be as low as possible.

Although no standards have as yet been determined, it is known that a good system should have good transient as well as steady-state characteristics, and should have no undamped vibrations when a square wave is applied to the input of the system. In general the system should have a smooth response with a minimum number of peaks and dips in the response curve for good reproduction of transients. This factor is especially important in the electromechanical components of the system (particularly in the loudspeaker).

Flutter and wow in the reproduction of disc and magnetic recordings can be serious and annoying; the ear is quite sensitive to this type of distortion since it seldom occurs in nature. Under some conditions the ear can detect the presence of as little as 0.001 percent flutter. The best systems which are being used in broadcast and recording studios at the present time maintain constant speed to about 0.1 percent or better. About 1 percent flutter or wow is acceptable for a home reproducing system.

RESPONSE OR DISTORTION BEING MEASURED	TYPE OF SIGNAL WITH WHICH MEASURED	LIMITS	
		Good Reproduction	Acceptable Reproduction
Frequency response	Steady sine wave	20-14,000 cps	40-10,000 cps
Noise level	Zero	−60 db below full output	−50 db below full output
Harmonic distortion	Steady sine wave	2% total harmonics	3-5% total harmonics
Intermodulation distortion (amplifier alone)	Sum of high freq. and low freq. steady sine waves	4%	10%
Transient response	Square wave or tone burst	No set standards	
Wow and flutter	Steady sine wave	0.1%	1%

Table 2-1. Summary list of the various electronic distortions and their limits for the entire high-fidelity system.

The above distortion information is summarized in Table 2-I, which gives a list of the various electronic distortions and their acceptable limits, and also the type of signal with which the amount of this distortion may be detected. When these factors are taken into account in the design and setup of the reproducing system and the various distortion limits carefully considered, the best sound reproduction available with modern techniques can be attained.

General Considerations

The reproduction of music and speech should be as nearly perfect as possible with the present state of electronic engineering. The setup for installation of a truly good reproducing system requires a basic understanding of the factors involved in the reproduction of sound, in the basic requirements for good reproduction, in the capabilities and physical limitations of sound reproducing systems and their components, and in the practical applications of these various considerations to specific conditions. The attainment of the best performance from the reproducing system, with the most efficient use of equipment at the least expense, also requires an understanding of the operation of the various components and their interrelationships.

The basic considerations which are important in the reproduction of sound have already been discussed. The factors which affect the setup of the reproducing system, the manner in which they must be taken into account, and their application to the actual design and construction of the various phases and components of the system will be described in the next chapter. The overall setup and design of the complete sound reproduction system according to the listener's specific requirements are based upon the fundamental principles and requirements of good reproduction.

Chapter 3

SOUND REPRODUCING SYSTEMS

General

Because conditions vary so widely in individual application, each high-fidelity installation should be set up with individual requirements in mind. Usually a single complete unit cannot be purchased which will give results as satisfactory as can be obtained by a system set up from an intelligent consideration of the various individual factors. The best approach for the audio experimenter to follow is to study the conditions which exist in his own case and compare them with the results he wants to achieve as outlined in the earlier chapters of this book. (Of course, economic as well as physical factors must be given important consideration.) Next consider the various component parts of the system which are capable of giving the desired results and performing the necessary functions. Certain of these units may be constructed and others purchased, according to the individual circumstances. The interrelations between the various components must then be considered in combining them to form the complete system.

Before even starting to set up the reproducing system first consideration should be given to the acoustic conditions in the sound pickup and listening rooms. The electronic and electromechanical components of the system all may be physically perfect, yet bad room acoustics can completely destroy the quality of the reproduced sound. The techniques of controlling acoustic conditions in rooms are quite well known and are widely used in the design of theaters and broadcast studios, but they have not been very widely applied in the home even when great care and considerable expense have been involved in setting up a high quality sound reproduction system. The quality of reproduction will be considerably improved if proper consideration is given to room acoustics.

The important factors which affect the quality of sound in a room are (1) the size, (2) the acoustical reflecting quality, and (3) the noise level. Normally, the size and the noise level are factors which cannot be controlled without considerable expense and must, therefore, be compensated for. The sound reflecting characteristics, on the other hand, can be controlled and cannot be compensated for if they are not good. Two important effects depend upon the sound reflection in a room: one is the reverberation, the other is the spatial distribution of sound in the room.

Reverberation and Diffusion of Sound

When any sound starts, its intensity does not immediately reach maximum because it takes an appreciable time for some of the sound to reach the walls and undergo one or more reflections before reaching the listener. The intensity reaches its maximum when the steady-state condition is attained – the listener then hearing both the direct and the reflected sounds at the same time. After the sound source stops, it also takes an appreciable time for the various reflections to be completely absorbed so that they can no longer be heard. This persistence of sound due to multiple reflections is caller *reverberation*. When sounds are heard with too little reverberation they appear unnatural, while too much reverberation causes them to lose in intelligibility due to overlapping of the various reflections.

The space characteristics of the sound reflections are also important in determining the acoustic quality of a room. The behaviour of sound presents a very complex problem, since a room is actually an acoustic resonant cavity of fairly large dimensions with many resonant frequencies. At the frequencies of resonance the sound is over-accentuated, while at other frequencies the sound may be suppressed. With parallel walls, transient vibrations known as "flutter echo" occur due to the reflections between the walls. Concave surfaces tend to focus sounds toward their center of curvature, giving a greater sound intensity at that point than at other points in the room and creating the impression that the sound originates at the concave surface. For a room to have good acoustic properties, the spatial sound pattern should be as diffuse as possible at all frequencies, with no standing-wave patterns and no points of excessive sound concentration. If acoustic frequency response measurements are taken in rooms with good and with poor sound diffusion characteristics, it will generally be found that those rooms with diffuse sound patterns show response curves which are fairly smooth with not too many irregularities, while rooms with poor diffusion have a great many irregularities in their response curves.

Sound diffusion and reverberation are best controlled by using the proper quantity and quality of absorbing material, correctly placed so as to eliminate sound concentrations and resonances. Diffusion without absorption can be obtained by adding irregularities (such as convex projections on a wall, and panels which are not parallel to opposite walls) to give more diffuse rather than direct sound reflections. Use of the proper proportion of absorbing material and diffusion techniques serves the dual purpose of diffusing the sound pattern and preventing an excessive amount of room reverberation. One type of wall treatment which gives good results where it can be employed, is to arrange some decorative pattern of serrated or convexly curved reflective surfaces alternated with absorbing areas. Another method is to have absorbing surfaces, such as heavy draperies, rugs, or large openings, opposite the large flat reflecting surfaces.

Typical Room Arrangements

These various techniques can also be combined in many ways, and generally the actual room layout and design will be a combination of the various sound diffusion techniques tailored to fit the particular requirements. For example, a typical living room having good acoustic properties might be one laid out as shown in Fig. 3-1. Note how the facing walls are handled so as to prevent excessive reflections and the setting up of standing-wave patterns within the room. The placement of furniture is also arranged to produce good sound diffusion. The larger wall adjacent to the loudspeaker is partially covered with heavy draperies and curtains, and that portion of it which is bare is facing a large archway that opens into an adjacent room. The smaller adjacent wall is completely covered with draperies and curtains. Note the use of upholstered chairs and a sofa in front of the bare walls in the listening room. Thus, any tendency toward resonances between the walls is

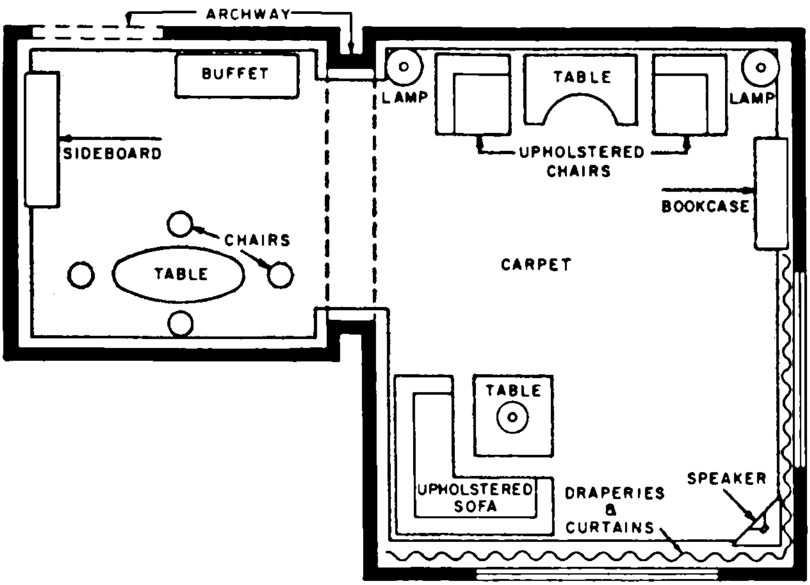


Fig. 3-1. Layout of a typical living room which has been arranged to give good acoustic properties for the acceptable reproduction of music and speech.

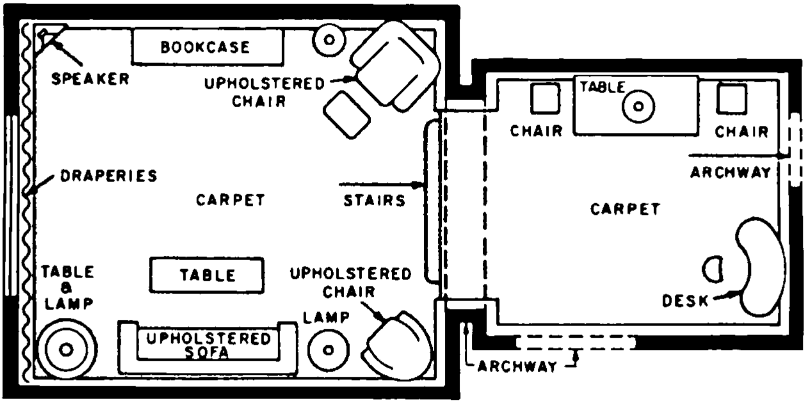


Fig. 3-2. Layout of a living room which displays satisfactory acoustic properties, although not as good as that shown in Fig. 3-1.

considerably reduced. A rug on the floor reduces reflections between the floor and ceiling.

The room shown in Fig. 3-2 is another layout which has demonstrated satisfactory acoustic properties, although the layout is not quite as good as that of Fig. 3-1. One wall is covered with curtains and draperies, opposite this is a large archway leading into another smaller room, the floor is covered with a fairly thick rug under which is a soft pad, while the ceiling and two remaining walls are reflective. There would seem to be some tendency toward resonance between the two opposite reflecting walls in this room, but the furniture layout seems to provide sufficient absorption and diffusion to eliminate any marked resonances. If any wall treatment had been found necessary, a panel with either an absorbing or a diffusing effect in one of the walls would have been enough to correct the difficulty.

Reverberation Times in Good Reproduction

Numerous listening tests have shown that there is an optimum reverberation time for rooms of various sizes. It is not desirable to make the walls of a room too absorbent, since when there is too little reverberation the sound has a dull, lifeless quality; however, too much rever-

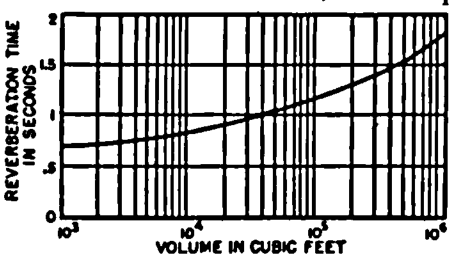
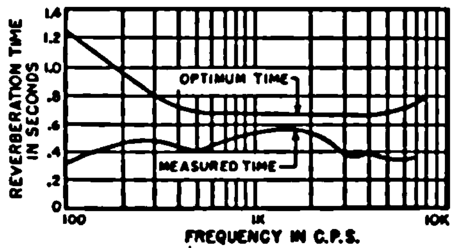


Fig. 3-3. Optimum reverberation time, in rooms of various sizes, for a frequency of 1000 cycles per second.

beration results in a loss of intelligibility. The most desirable reverberation times of various size rooms for a frequency of 1,000 cps have been found to be those shown in Fig. 3-3. The reverberation times for reproduced music should be less than for the same live music, since the reproduction already contains the reverberation of the production studio. The desirable amount of reverberation as a function of frequency, relative to the 1,000 cycle value, is shown as the "optimum time" curve in Fig. 3-4. Note the optimum reverberation time at 100 cps is almost twice the value required at 1,000 cps. At frequencies over about 5,000 cps, the optimum reverberation time is slightly more than is required at 1,000 cps.

Some reverberation time measurements have been made in rooms like those shown in Figs. 3-1 and 3-2, with results as shown in the "measured time" curve of Fig. 3-4. The most important factor which can be observed as a result of this measurement is the considerable lack of low-frequency reverberation. The reason for this effect is that the wall and room resonant frequencies occur in just this frequency range, so that sound energy which should be reflected is, instead, dissipated in friction due to the wall and room structure vibrations.

Fig. 3-4. Reverberation time as a function of frequency in a small room, similar to those shown in Figs. 3-1 and 3-2, showing optimum reverberation time compared with the measured reverberation.



The net result of this reverberation-frequency effect is to make reproduced music in small rooms sound deficient in bass. However, this deficiency cannot be completely compensated by simple bass boost, because reverberation adds color as well as volume to the sound. Listening tests readily demonstrate these conclusions. Another effect which occurs in small rooms arises from the fact that a small room has more reflection than a large room or auditorium. The main impression of sound quality is formed in the first 250 milliseconds of reflection; therefore, because of the greater number of reflections during these first 250 milliseconds the smaller room has a greater opportunity to impose its characteristics on the reproduced sound. A listening comparison of the same material reproduced in the small room whose measurement is given in Fig. 3-4 and in a motion picture theater showed it to sound definitely better in the theater.

The lack of low-frequency reverberation in small rooms represents a serious problem in setting up a high quality sound reproducing system. Years of listening to sound reproduction have, to some extent, made us accustomed to this factor in listening to sound reproduction in the home; however, this is certainly no solution to the problem since it merely means accepting inferior quality of reproduction. One compromise is to use a certain amount of bass boost to overcome the apparent deficiency in bass. Although this does not give the complete effect of the desired reverberation, with a good loudspeaker system fairly good sound will be obtained. The best method of compensating for the lack of low-frequency reverberation is by means of a frequency selective, synthetic reverberation system which can give complete compensation for the room defects.

Basic Components of Sound Reproduction Systems

Once the room acoustics are considered satisfactory, attention can be given to the details of the electronic reproducing system. Any system for reproducing sound consists essentially of: (1) a microphone for picking up the sound in the air and converting it into an electrical signal, (2) a means of transferring this signal, either in time by recording, or in space by transmission, and (3) a loudspeaker for converting this vibration back into sound.

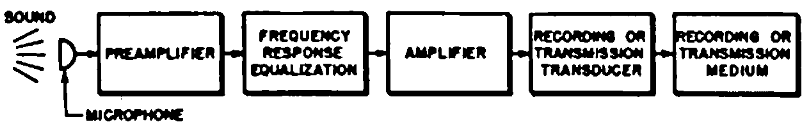


Fig. 3-5. Block diagram showing the basic system for the recording or transmission of sound.

At the present time sound reproduction systems operate almost exclusively by means of electrical signals. The basic system for recording or transmitting sound is essentially that shown in the block diagram of Fig. 3-5. A microphone converts the sound vibrations in the air into electrical signals which are then amplified by a sensitive preamplifier. This signal may then have its frequency-amplitude characteristics changed in any manner which may be desirable for the reproduction process. Further amplification supplies the power to operate whatever type of transducer or converter is required for the particular type of re-

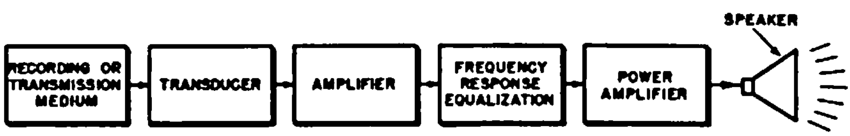


Fig. 3-6. Block diagram showing the components of the basic system for reproduction of sound from a recording or transmission.

cording or transmission system being used. The final step in the system is the recording or transmission itself.

The basic components of any system for reproducing sound are shown in the block diagram of Fig. 3-6. The reproduction process starts with the recording or transmission, which is converted into an audio-frequency electrical signal by the appropriate type of transducer (such as a phonograph pick-up or radio receiver). The electrical signal is then amplified, and passed through an equalizer to give the required frequency response. A power amplifier supplies the energy for the loud-speaker to convert the electrical audio signal into sound again. (The various units and components of these systems will be discussed in greater detail in later chapters.)

The availability of simple and inexpensive disc recording equipment and the development of magnetic recording have made the recording process as simple as the reproduction process, therefore many serious listeners and experimenters are now interested in recording as well as reproduction. The program material may originate either in the experimenter's home or studio or, more often, may be a broadcast from a radio station. A versatile system which can be used for recording from

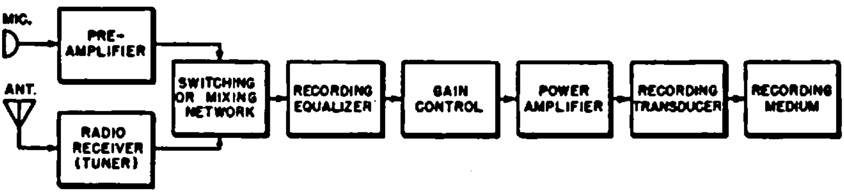


Fig. 3-7. Block diagram showing the setup of a system for recording from either a direct pickup or from a radio broadcast.

either an actual performance or from a radio broadcast is shown in Fig. 3-7. The audio signal may originate either in a radio receiver or from a microphone, with the specific signal to be recorded at any specific time being selected by means of an appropriate switching or mixing unit. The microphone output would be amplified sufficiently so that the two signals are switched at the same level, and the rest of the recording channel can be the same as shown in Fig. 3-5.

If a completely new system is being set up, it is generally desirable to consider both the recording and the reproducing channels at the same time. In this way both economy of equipment and simplicity of operation are obtained, since the entire system is better integrated as a whole. Usually the signals which are to be recorded will be either directly picked up by a microphone or from some remote point of origin by a radio receiver, whereas the program material which is listened to in repro-

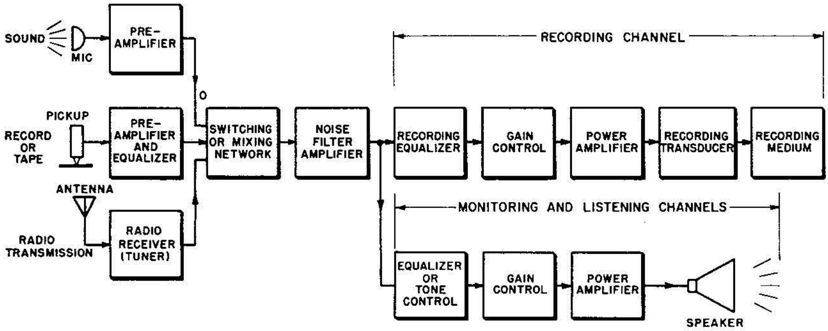


Fig. 3-8. Block diagram showing the setup of a system which can be used for simultaneous recording and reproduction from either reproduced sound or direct pickup.

duction will be obtained from records or from radio broadcasts. However, there are occasions when duplicates are to be made from other records, and where it is desirable to listen to certain original sounds in order to hear how they sound when reproduced through a loudspeaker. The complete system can be set up to permit simultaneous recording and reproduction from any of the three types of sound signal sources in the manner shown in Fig. 3-8. This particular setup is quite simple and flexible, while at the same time combining all the essential functions for both recording and reproduction. A photograph showing the components in a complete system appears in Fig. 3-9.

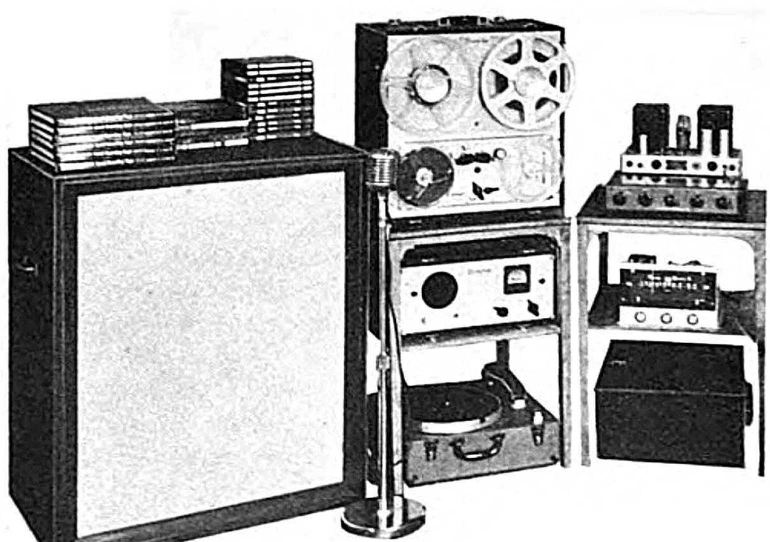


Fig. 3-9. Components in a complete recording and reproduction system.

The precise form which the various components will take, and their specific characteristics, will depend upon the particular conditions which are to be met.

Setup of the System

The setup of the channel itself will be one of the four basic setups shown in Figs. 3-5, 3-6, 3-7 and 3-8, depending upon the functions which are to be performed. The basic characteristics of the various units will always be more or less the same, but certain essential characteristics (such as power output, voltage gain, sensitivity, etc.) must be chosen specifically for the particular application. In setting up the system, considerable care must be taken in matching the various units correctly. Among the major points which must be considered are:

(1) Type of signal.
(2) Input and output impedances.
(3) Signal levels and d-c voltages.
(4) Power requirements.
(5) Physical placement of the components.

Good reproduction will be obtained only if all of these factors are properly considered.

Type of Signal. The type of signal at any point in the system may take any one of several different forms. The original signal, of course, is the sound in the air which is a mechanical vibration. Signals on discs are in the form of groove variations, and on magnetic tape or wire they are in the form of variations in a magnetized medium. Radio signals consist of an amplitude-modulated or frequency-modulated high-frequency electromagnetic field. The correct type of unit must be used to pick up each of these signals, as indicated in Fig. 3-8, and to convert them to an audio-frequency electrical voltage or current. This audio-frequency voltage or current is then the signal which is present throughout the rest of the reproducing system until it is again converted into sound by the loudspeaker.

Input and Output Impedances. Every electronic unit is designed to operate best when the unit to which its output is connected has some specific impedance. These input and output impedances will depend entirely upon the design of the unit, and when a commercial unit is purchased this information must be obtained from the manufacturer. The impedances may be as low as 2 to 4 ohms for a loudspeaker, or as high as 1 to 2 megohms for the input of a voltage amplifier. Mismatching the input and output impedances will usually introduce distortion or result in a loss of signal level, and should, therefore, be avoided.

Signal Levels and D-C Voltages. Careful consideration must also be given to matching the input and output voltages of the various components of the system. If the output voltage of one unit is too high for the input required of the unit to which it is applied, serious overloading and distortion will result, and often the equipment itself may be damaged. If too little signal voltage is supplied, a high noise-to-signal level may result, and it may not be possible to obtain a sufficient sound output from the system. Attention must also be paid to the d-c voltage levels at various points in the system, since some units can be operated with high voltages applied to their inputs, while others would be damaged by such voltage.

An examination of the manufacturer's literature furnished with commercial units will indicate proper operating conditions.

COMPONENT OR FUNCTION	INPUT				OUTPUT				GAIN OR INSERTION LOSS (in db)
	Type of signal	Impedance	Signal level (approx.) Volts	DBM*	Type of signal	Impedance	Signal level (approx.) Volts	DBM*	
Microphone	Sound	—	—	—	Electrical	High/low	—	-55	—
Pickup lead	Record	—	—	—	Electrical	High low	1.0 0.01	-30 -30	—
Radio receiver (Tuner)	Radio transmission	—	10^{-6} to 0.01	—	Electrical	High Low	2.0 —	— 0	—
Preamplifier	Electrical	High/low	—	-60 to -30	Electrical	High/low	—	0	+30 to +60
Switching and mixing networks	Electrical	High/low	—	0	Electrical	High/low	—	-12 to 0	0 to -12
Noise filter amplifier	Electrical	High/low	—	-15 to 0	Electrical	High/low	—	0	0 to +15
Equalizer or tone control	Electrical	High/low	—	0	Electrical	High/low	—	-15	-15
Gain control	Electrical	High/low	Any		Electrical	High/low	Any		0 to −∞ Usually set at -15
Power amplifier	Electrical	High/low	—	+30 or higher	Electrical	Low	1 to 100w	+30 to +50	Approx. +70
Recording transducer	Electrical	Usually low	1 to 50w	+30 to +47	Mechanical or magnetic	—	—	—	—
Loudspeaker	Electrical	Low	1 to 100w	+30 to +50	Sound	—	—	—	—

*Reference level: 0 dbm = 0.001 watt

Table 3-1. Input and output characteristics of the various components in a sound reproducing system.

More specific details and information about these various considerations are summarized in Table 3-1. This table lists the various components of the system, and gives typical values for input and output signal levels (in voltage or decibels), type of input and output signals, input and output impedances, and signal amplification or attenuation of each component. This data can serve as a guide in setting up a complete system, and may be compared with the signal levels and gains of the various units in the block diagram of Fig. 3-8 to show its application in a specific system.

Power Requirements. The required power output of the amplifier will depend primarily upon the size of the listening room. The amplifier power requirements can be determined from the chart given in Fig. 3-10, which shows the actual acoustic output power needed for

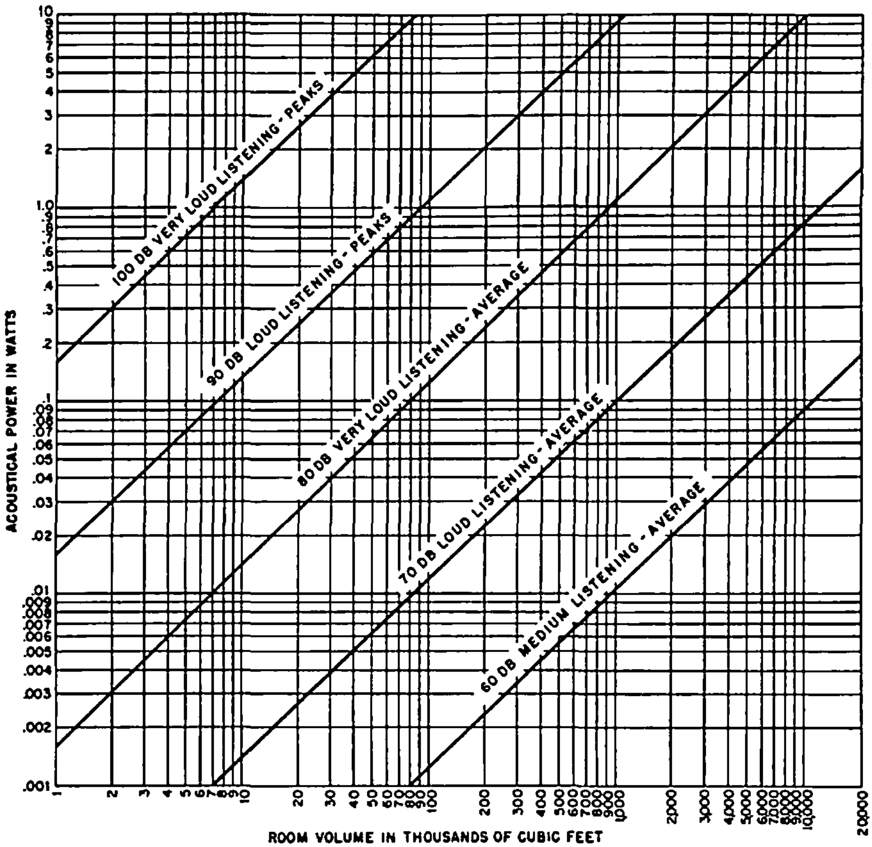


Fig. 3-10. Approximate acoustic power requirements for rooms of various sizes.

various listening levels. The lines marked "60 db," "70 db," and "80 db" are for medium (as a background) listening, loud (serious) listening, and very loud listening. These lines represent the acoustic power required for reproduction of the average level of the sound. To provide a reserve of power that can accommodate the very high amplitude transient peaks that occur in music reproduction, it is common to require an additional 20 db of listening level. Therefore, for loud and very loud listening levels for these peaks, the 90-db and 100-db curves are provided.

Note that these curves indicate the *acoustic* power needed and *not* the amplifier electrical power. To determine the amplifier power it is necessary to know the approximate efficiency of the loudspeaker used. Loudspeakers, in common with musical instruments, have very low efficiencies. Typical values are 2 to 3 percent, with very good quality speakers having efficiencies of about 5 percent. When such speakers are mounted in properly designed enclosures, higher efficiencies of 10 percent to as much as 50 percent in certain special cases are possible.

Assume we are interested in knowing the *amplifier electrical power* needed for possible very loud listening that will handle the transient peaks adequately. The listening is to be done in a living room of average size, say 2,000 cubic feet. Referring to the 100-db line in Fig. 3-10, it is found that an acoustic power of 0.3 watt is needed. Assuming a loudspeaker efficiency of 5 percent, this means that an amplifier output power of 6 watts is required. Consequently, in an average-sized room in the home, an amplifier and loudspeaker capable of handling 8 to 10 watts of electrical power will have an adequate reserve power for any purpose. For unusually large rooms in a home, somewhat more power is required.

Physical Placement. The physical placement of the various units of the system and the permissible distances between them depend almost entirely upon the impedance of the particular circuits involved. In a circuit whose impedance is on the order of 0.5 to 1 megohm, a lead with a capacitance to ground of as little as 30 $\mu\mu$f will appreciably affect the high-frequency response. In the output of a crystal phonograph pickup, which appears as a capacitance, somewhat longer leads can be tolerated. Thus, the leads from the radio receiver should be as short as possible if the receiver has a high-impedance output; while the leads from a crystal phonograph pickup can be as long as 10 to 15 feet, but should not be any longer for good high-frequency response. Lines at low impedance can be as long as desired for mounting certain units remotely. Low-level lines should be shielded to avoid pickup of 60-cycle hum.

Once the complete system has been set up, an overall test should be performed to determine whether there are any impedance mismatches or overloading at any point in the system. The simplest and most direct test of this type is to apply to the input an audio signal which is known to have good quality, and to listen to its reproduction from the loudspeaker. If the system has been properly set up, there will not be any appreciable distortion introduced by the reproduction system, and the reproduced sound will be the same as the original input signal.

Economic Factors Affecting Choice of System Components

The basic sound reproducing system will be one of the basic setups described in the following sections, however the actual components and the separate units which make up the system must be chosen to suit individual requirements. The economic factors entering into this choice will vary with each individual installation, therefore there are no set rules for setting up systems without complete information concerning these factors. Some of this information may be obtained from the following considerations:

(1) How much money can reasonably be spent for the system, and how much time can be spent in construction of equipment and installation?

(2) Will the system be used intensively enough to warrant a substantial investment of time and money, or will it be used mostly to provide pleasant background music?

(3) In what type of room will the reproduced sound be heard? The size of the room, its reverberation and sound absorption characteristics will have an important effect on the system requirements.

(4) What type of music will generally be reproduced, and what will be the loudness level at which the reproduced sound is listened to?

The answers to these questions are extremely important in determining the exact components and the specific setup of the reproducing system to meet individual requirements.

Other important factors which determine the specific nature of the installation are concerned with the esthetics of furniture design and with the convenience of installation and placement of the various components of the system. Because the answers to the different questions which determine the system can vary so widely in different individual cases, very little can be said about the resulting reproducing system without knowing some of them.

However, certain very general estimates can be made of the possible cost of the system. The minimum cost for an acceptable reproduction system composed of commercially available units, to be used in an average-size living room and built into existing furniture, would be of the order of about $200. The maximum cost for a reproducing system composed of the highest quality commercial units currently available, mounted in good quality furniture units, might be as high as $1,500 or $2,000. The cost of systems designed to meet specific individual requirements might be anywhere within this price range. Of course, if any of the units are built instead of purchased, the costs will be reduced accordingly, and better reproduction will be obtained at lower cost than if only commercial units are used.

In most systems the record-player is a commercially purchased one, although an experimenter with adequate shop facilities can build or assemble his own turntable. In most cases the commercial unit is preferable, and may be either a record-changer or a single-play turntable, according to the user's choice of better reproduction or ease and convenience of operation. Many engineers prefer to experiment with the construction of the pickup cartridge, but the availability of many fine commercial cartridges makes it unnecessary for the average experimenter to use any but a standard purchased unit. The electronic units in the the system may be purchased commercially or constructed, the choice depending only upon the preference and ability of the experimenter. Electronic circuits can frequently be assembled and wired just as well by the qualified amateur as by the commercial manufacturer, and the greatest saving in money can be effected by home construction of these units. Home construction has the further advantage that the circuits can be designed specifically to meet the exact system requirements, whereas commercial units must be integrated into the system. The loudspeaker is almost always a purchased unit, although occasionally the cone or the suspension may be modified. The choice of the loudspeaker enclosure is almost entirely a matter of the installer's ingenuity and good taste. It may be either a cabinet especially designed to house the loudspeaker, or the loudspeaker may be mounted in a convenient door, partition or wall leading to another room, or a large closet. Excellent results can be obtained with any correctly designed type of loudspeaker enclosure.

Integration of Commercial Units Into the System

When the units of a sound reproduction system are specifically designed and constructed for a particular installation, it can meet all the individual requirements. On the other hand, if the system is composed mainly of commercial units, they must be properly integrated into a system which meets the requirements of high-fidelity production of sound, ease and convenience of operation and adjustment, and the space and mounting requirements of the room arrangement. One typical well-planned, high-quality home sound reproduction system appears in Fig. 3-11.

The actual integration of the system depends primarily upon the specific units which are to be used, but a number of statements may be made which are true of sound reproduction systems in general. The major difficulty in integrating a number of commercial units into a complete reproduction system is the physical location of the different units in regard to their electrical interconnection and the location of control panels. For example, when the f-m tuner, the a-m tuner and the amplifier are three separate units, then there will be three separate control panels which may each contain volume and tone controls, and two of which contain tuning controls. Furthermore, because of the physical size of the units, the various sets of controls cannot be grouped close to one another on a single panel, but are mounted some distance from one another. Some users may not mind this multiplicity of con-

Courtesy: Electronic Workshop Sales Corp.

Fig. 3-11. One typical well planned, high-quality home sound reproduction system.

trol knobs and the minor inconvenience of their location. Those who do object to this inconvenience should either build or purchase a control panel which contains all controls except the two tuning adjustments, or else use an a-m/f-m tuner which contains a complete set of controls. All controls on other units in the system may then be left at some convenient setting, and all control functions accomplished from the front panel of the tuner or the control panel. Many such combination tuners also contain preamplifiers so that the utmost in control convenience is possible.

Another problem in the integration of a sound reproduction system using commercial units is the electrical interconnection of the various units. The input and output impedances must be properly matched, and the signal levels should be correct at each point in the system. The input and output impedances and signal levels of the various components given in Table 3-I can be used as a guide to their electrical integration into a complete system which meets the requirements of the installation. If the record-player is to be mounted at a considerable distance from the amplifier, a low-impedance line must be used from the output of the record-player to the input of the amplifier. This means that if a crystal pickup is used, some type of impedance-transforming circuit must be located close to the record-player. Such a circuit may be a cathode-follower, or a low-level amplifier having a transformer-coupled output to a low-impedance line. The need for such an impedance-transforming circuit is avoided by the use of a low-impedance magnetic pickup. However, a magnetic pickup has a lower output level than a crystal; the amplifier must therefore have an additional preamplifier section to bring the phonograph output up to the level of the radio tuner output.

Very often it may be desirable to locate the control panel at some distance from the other units in the system. In such installations, it is preferable to have all audio interconnections at as low an impedance as possible. In radio broadcast practice, 600 ohms and 150 ohms have generally been chosen as the standard audio line impedances — 600 ohms being the older standard, and 150 ohms being a more recent standard in many new studio constructions. This impedance can be obtained by either a transformer or a cathode-follower.

When the various components are to be located close to each other, the integration of the complete sound reproduction system is not too difficult and does not require the construction or purchase of special units. The use of a commercial a-m/f-m tuner which can be used as the main control panel will generally accomplish the major system integration function. If separate a-m and f-m tuners are used and a separate control panel is desired, one of the several commercially available units may be used.

Since the integration of the complete sound reproduction system is an important problem for both laymen and technicians alike, the construction of special units for accomplishing this function will be described in detail in later chapters. This information will simplify the setup of a variety of very flexible sound reproduction systems from commercial units with the purchase or construction of one or two simple interconnecting circuits.

Chapter 4

INPUT AND PICKUP UNITS

General

All systems for the recording or reproduction of sound must start with some device for picking up the signal which represents the sound to be reproduced. The basic input signal in any reproduction of sound is, of course, the original sound in air which must be picked up and converted into an electrical signal by a microphone. In home reproduction systems the input signal is most commonly either (1) a recording on a disc or a magnetic medium, which must be converted into an electrical signal by a phonograph or tape pickup, or (2) a high-frequency electromagnetic signal representing a radio broadcast of a sound signal, which must be picked up by a radio tuner or receiver. A proper understanding of the functioning of the sound reproducing system must begin with these various pickup devices.

No sound reproducing system of any sort would be possible without the use of microphones, since all sound reproduction starts with the sound pressure vibrations in the air. Since the energy that is available to the microphone is extremely small, its electrical output is almost always the lowest level in the entire audio system. The energy which can be supplied from a record groove without damage to the groove or loss of reproduction quality is much higher, therefore the output of the phonograph pickup is generally 10 db to 20 db greater than that of the microphone.

Microphones and phonographs may best be considered in two groups, since the principles of their operation, their construction, and their connection into the audio system are related to the output impedance: (1) low-impedance microphones and pickups, and (2) high-impedance microphones and pickups. This grouping determines the type of preamplifier and circuit techniques which must be used with them. The low-impedance devices include the various types of carbon, velocity (or ribbon), dynamic microphones, and the various types of magnetic and strain guage phonograph pickups. The high impedance units are the crystal and condenser microphones and pickups.

The important characteristics of the various types of microphones – frequency range, output voltage, output impedance, directional characteristics, and other important features – are summarized in Table 4-I for convenient reference. The output characteristics of the various types of phonograph pickups – those which are in widespread use, plus a number of different types of pickups which are not widely used but which also have features which are of merit – are summarized in Table 4-II. Frequency range of these pickups varies widely depending on the construction used, the stylus and arm employed, and the load into which the pickup works. When properly equalized, some of these units provide flat frequency response from 20-50 cps to 10,000-15,000 cps.

Radio tuners for the reception of a-m and f-m broadcasts are electronic units which all have essentially the same principles of operation except, of course, for the difference in principle between the a-m and the f-m systems.

Their characteristics do not differ in many important respects, and the differences between the various units in respect to sensitivity, stability, output voltage, and output impedance are due to differences in circuit details rather than to fundamental circuit differences.

Low-Impedance Microphones

Carbon Microphone. The carbon microphone (see Fig. 4-1) depends for its operation upon the resistance of the contacts between carbon granules. It consists essentially of a carbon button, which is a cylindrical cavity filled with carbon granules, and a diaphragm coupled to this carbon button in such a manner that pressure variations in the air cause proportional variations in pressure between the carbon granules. This pressure variation causes a change in the resistance between the plates which are in contact with opposite sides of the carbon button. An audio-frequency electrical signal is obtained by using the microphone resistance as one arm of a series voltage divider and measuring either the voltage variation across the microphone due to its resistance variation, or the current variation in the circuit by means of a transformer. The resistance of this type of microphone is generally of the order of about 100 ohms. It has high electrical output and good enough fre-

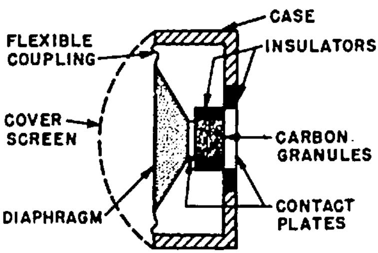


Fig. 4-1. Internal construction of a typical single-button carbon microphone.

TYPE OF MICROPHONE	DIRECTIONAL CHARACTERISTICS	OUTPUT IMPEDANCE	OUTPUT LEVEL	FREQUENCY RANGE	MAJOR APPLICATIONS
Velocity (ribbon)	Bidirectional	50/250 ohms	–55 dbm	30-15,000 cps	Hi-qual. pickup
Dynamic (moving coil)	Nondirectional	50/250 ohms	–55 dbm	40-10,000 cps	Studio and remote
Crystal	Nondirectional	1 megohm (approx.)	–51 dbm	40-10,000 cps at best	Home and p. a.
Condenser	Nondirectional	About 10 megohms (requires special preamp)	Approx. –55 to –50 dbm after preamp	30-15,000 cps or better	Hi-qual. pickup and measurement standard
Carbon	Nondirectional	100 ohms	Approx. –45 dbm	100-4,000 cps	Voice reproduction

Table 4-I. Performance characteristics of the various types of microphones.

TYPE OF PICKUP	OUTPUT LOAD IMPEDANCE	OUTPUT LEVEL
Crystal	Approx. 1 megohm	Approx. 1 volt
Variable Reluctance (G. E.)	Approx. 6,800 ohms*	.01 volt
(Pickering)	Approx. 3,000 ohms*	.055 volts
Dynamic (moving coil)	50-100 ohms	Approx. .01 volt
Strain gauge	100 ohms to ¼ meg.	Approx. .005 volt
Eddy current (Zeneth Cobra)	Uses special preamp	Approx. 1 volt after preamp
Frequency modulation	About 10 megohms, requires special preamp	Approx. 1 volt after preamp

*AES curve

Table 4-II. Performance characteristics of the various types of commercial phonograph pickups.

quency response and distortion characteristics for voice reproduction, but is not suitable for high-quality reproduction of music.

Velocity Microphone. In the velocity (or ribbon) microphone, (see Fig. 4-2), a thin, corrugated aluminum ribbon is suspended in the air gap between the pole pieces of two parallel sections of a permanent magnet. The difference in air pressure between the front and back of the ribbon, due to the soundwaves in the air, causes it to move in the magnetic field and generate a voltage proportional to the air-particle velocity in the sound wave. The ribbon, which is a thin strip, can move forward and backward but not sideways or up and down; therefore, it is most sensitive to sounds which approach it perpendicularly, and has no response to sound waves which move in a direction parallel to its

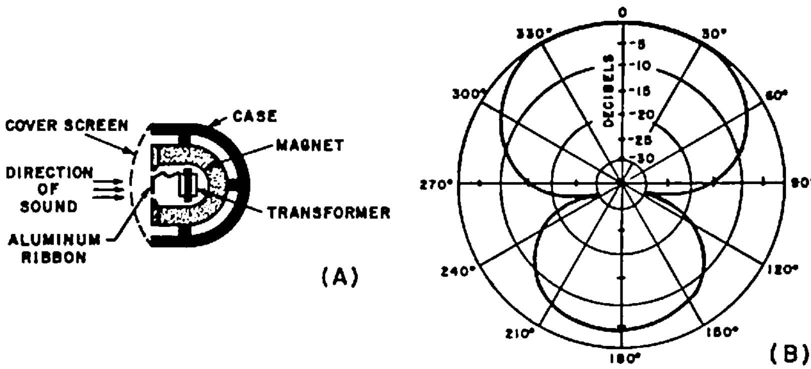


Fig. 4-2. (A) Internal construction of a typical ribbon velocity microphone. (B) Typical figure-eight directional response of a velocity microphone.

plane. Its directional response is, therefore, that of the figure-eight shown in Fig. 4-2 (B). The ribbon is extremely light and is loosely stretched between its supports so that its natural resonant frequency of vibration is below the audible range, therefore its response is uniform over the entire audio-frequency range. Good velocity microphones are capable of high electrical output level and uniform frequency response from 40 to over 10,000 cps; they are probably the most widely used studio microphones in high-quality commercial radio broadcasting and sound reproduction.

Dynamic Microphone. The dynamic (or moving-coil) microphone consists of a coil which is placed in a magnetic field and coupled to the air by means of a diaphragm. The sound pressure vibrations in the air cause the coil to move in the magnetic field and cut the magnetic lines of force, thus generating a voltage in the coil. The details of construction, given in Fig. 4-3, show how the circular coil is mounted in the

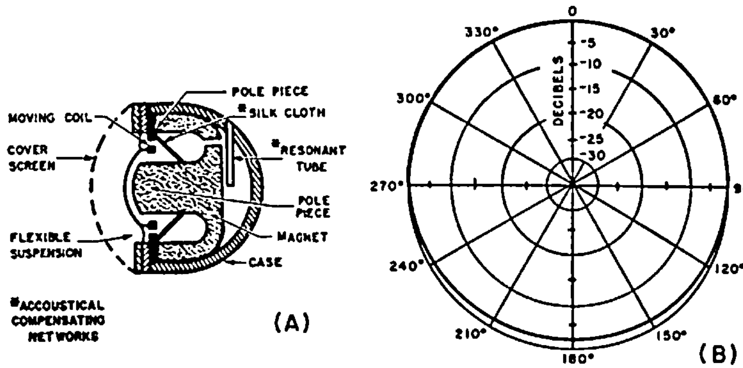


Fig. 4-3. (A) Internal construction of a typical dynamic or moving coil microphone. (B) Typical nondirectional response of a dynamic microphone.

movable diaphragm so that it is suspended in the radial field between the two poles of a cylindrical magnet, and can move into and out of the field according to the sound pressure variations in the air. The basic construction is extremely simple, and the principle of operation is essentially that of a loudspeaker in reverse. (This is why loudspeakers also can be used as microphones in low-cost intercommunication systems.) In microphones for high quality sound reproduction, acoustic and mechanical compensating networks are added to the basic microphone design to improve the frequency response and directional characteristics. Moving-coil microphones with such compensating systems are capable of giving a frequency response which is flat from about 40 to 10,000 cps with a non-directional spatial response. Because of its construction, the moving-coil microphone is more rugged than the velocity microphone, is less easily damaged by rough handling, less sensitive to overloading, and not adversely affected by wind; therefore, it is well

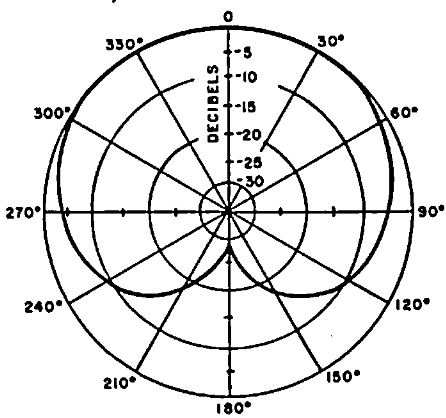
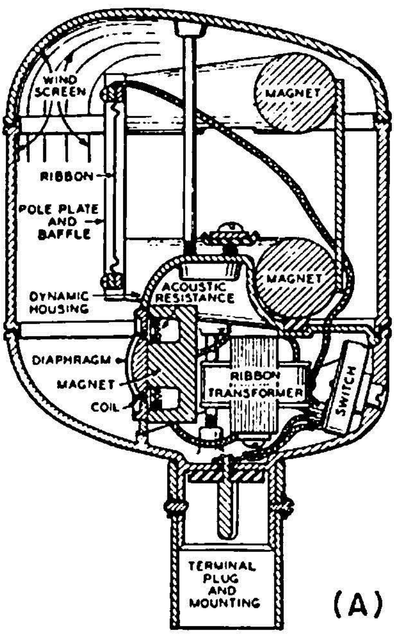


Fig. 4-4. Cardioid type of directional response obtained by combining the bidirectional velocity response of Fig. 4-2(B) with the nondirectional pressure response of Fig. 4-3(B).

suited to public-address work and broadcasting of outdoor events as well as for studio and indoor sound pickup.

A cardioid (heart-shaped) type of uni-directional response illustrated in Fig. 4-4 is obtained by combining the bi-directional velocity response of Fig. 4-2 (B) with the non-directional pressure response of Fig. 4-3 (B). In one microphone of this type, shown in Fig. 4-5, the figure-eight response of a ribbon velocity section is added to the pressure response of a moving-coil section. When the sound originates behind the microphone the velocity section is moving out-of-phase with the pressure section, therefore the addition and cancellation of the signals from the two sections results in the cardioid directional pattern.



Part (B) Courtesy: *Altec Lansing Corp.*

Fig. 4-5. The Western Electric-Altec Lansing type 639 microphone. Cross-sectional drawing shows how ribbon element (above) and dynamic element (below) are assembled in one compact unit to operate as a nondirectional, bidirectional, or cardioid directional microphone. Any of these three effects may be selected by changing the setting of the flush-type switch (lower right).

High-Impedance Microphones

Crystal Microphone. Crystal microphones depend for their operation upon the fact that a piezoelectric crystal generates an electrical voltage when it is distorted by a mechanical force. Therefore, if a crystal is arranged so that the sound pressure variations in the air cause it to be mechanically distorted, it can be used as a microphone to give an

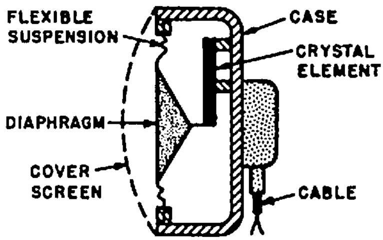


Fig. 4-6. Internal construction of a typical diaphragm-actuated crystal microphone.

electrical voltage proportional to the sound pressure. There are two different types of mechanical arrangements which are used for crystal microphones: the *diaphragm-actuated crystal,* and the *sound cell.*

The *diaphragm-actuated crystal* is set up as shown in Fig. 4-6. The diaphragm moves back and forth according to the sound pressure in the air, and is attached to one or two corners of a bender or a twister bimorph Rochelle salt crystal. The crystal is thus distorted according to the pressure of the sound waves, and produces an audio-frequency voltage of the same wave pattern as the incident sound wave.

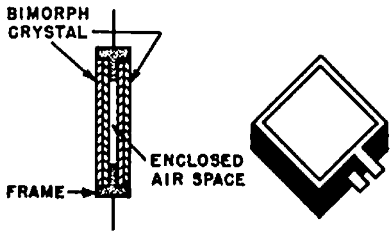


Fig. 4-7. Basic construction of a crystal sound cell.

The *sound cell* consists of two bimorph crystals assembled as shown in Fig. 4-7, with a completely enclosed cavity between them. If the instaneous external air pressure is greater than the pressure in the enclosed space, the crystals will tend to be bent inwards, whereas a lower external-than-internal air pressure will cause the crystals to bend outwards. In one case a positive voltage will be generated, in the other a

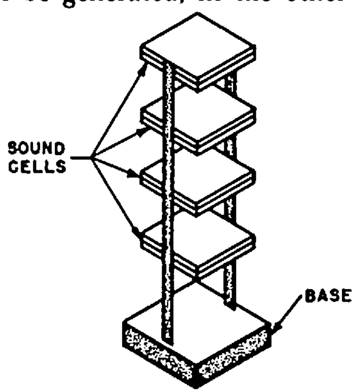


Fig. 4-8. A four-element sound cell consisting of four single units in parallel.

negative. If sound pressure variations occur in the air outside the sound cell, a proportional audio-frequency voltage will, therefore, be produced by the cell. No diaphragm is required since sound pressure acts on the crystal elements directly. The output impedance of a single sound cell is quite high, therefore a lower output impedance is often obtained by the use of multiple sound cells stacked in parallel as shown in Fig. 4-8.

Crystal microphones give fairly good frequency response to over 10,000 cps and are suited to high quality sound reproduction. They must operate into a high-impedance load, but with special coupling transformers can be operated into low impedance lines. Sound cells discriminate against mechanical shock and vibration, since the cell responds only to changes in air pressure and generates no voltage when it moves as a unit.

Condenser Microphone. The condenser microphone is one whose operation depends upon variations in electrical capacitance between two electrodes. The basic construction and principle of operation of the condenser microphone is shown in the diagram in Fig. 4-9. It con-

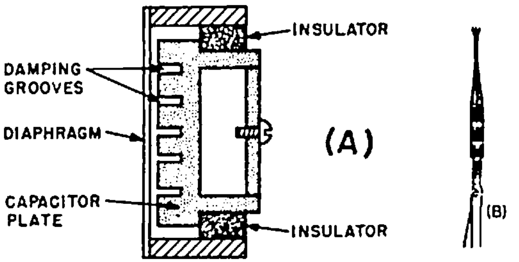


Fig. 4-9. (A) Basic construction of the condenser microphone. (B) Typical high-quality condenser microphone.

Part (B) Courtesy: Altec Lansing Corp.

sists essentially of a thin stretched diaphragm which is separated by a small distance from a parallel rigid plate. Variations in the pressure of the air on the diaphragm cause it to move toward or away from the plate, changing the electrical capacitance between them. Thus an electrical capacitance variation is produced which is proportional to the sound pressure variations in the air.

Good condenser microphones have uniform frequency response over an extremely wide frequency range (from 0 cps to well over 10,000 cps) with very stable characteristics, and at the same time are small and compact physically. For this reason they are often used as a standard in acoustic measurements. However, they do not give very large capacitance variations for variations in air pressure, resulting in a very low output signal. When used in a conventional series dropping resistor circuit they give a very low output voltage at high impedance, but can be used in frequency-modulated oscillator circuits to overcome this limitation. Because of the inconvenience involved in their use, condenser microphones have not been widely used in sound reproduction and broadcasting work in this country. However, recently small, compact units have appeared which are quite popular.

The Phonograph Record

Since it is not always desirable or practical to listen to music or other sound programs at the time they are being produced, one of the most common input signal sources for sound reproduction systems is the recording upon which program material has been stored for convenient reproduction at some other time or location. There are many ways of accomplishing this recording process, but at the present time the most widely used method is the recording on the phonograph disc.

The phonograph record is a thin disc less than 1/8 inch in thickness and up to 10 or 12 inches in diameter, made either of a vinylite or shellac composition. Upon the surface of this disc is impressed a long spiral groove which is modulated to contain the sound pressure variations. These records are produced in commercial quantity by an injection molding process which stamps them at high temperature, with the grooves pressed into the surfaces by a stamper made from the original studio master recording.

When the master recording is made, a disc of soft acetate composition coated upon an aluminum base is mounted upon a rotating turntable (see Fig. 4-10). A sharp stylus presses into the soft surface layer so that it cuts a groove when the disc turns. The cutter is mounted upon a lead screw which moves it toward the center as the record turns, so that the groove forms a spiral. (The groove may spiral either from the outside in, or from the inside out, but in present commercial records the standard is a groove spiralling inward.) The stylus is caused to move in a direction perpendicular to the direction of the spiral, by an amount equal to the instantaneous sound pressure being recorded, so

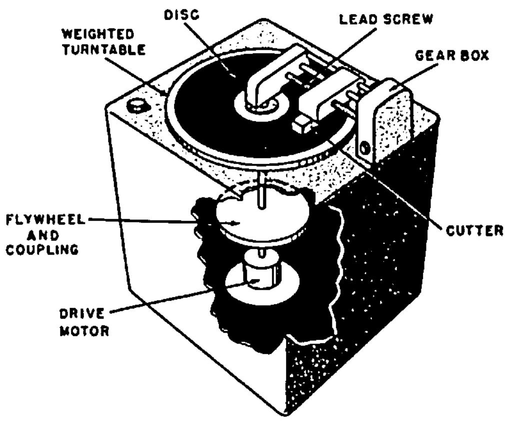


Fig. 4-10. Construction of a typical disc recording turntable.

that the groove is modulated by this sound pressure. The modulation may be either vertical or lateral, but lateral modulation is in almost exclusive use at the present time. The stylus can be made to move by either of two methods: it may be mounted upon a coil in a magnetic field so that current in the coil causes it to move by an amount proportional to the current, or it may be mounted upon piezoelectric crystal so that voltage applied across the crystal causes it to deform and move the stylus proportionally to the voltage. From this master recording is made the stamper which presses these grooves into the commercial record.

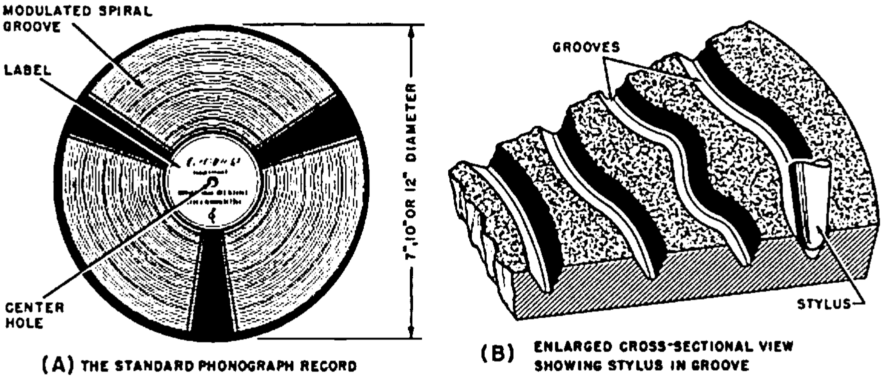


Fig. 4-11. (A) The standard phonograph record. (B) Enlarged cross-sectional view showing stylus in groove.

To reproduce the sound the record is played back by rotating it on a turntable at the same speed at which the sound was recorded, with the reproducing stylus placed in the groove so that the modulation can cause it to move laterally according to the instantaneous sound pressure represented by the displacement of the groove (see Fig. 4-11). The reproducing stylus is part of a transducer which produces an output voltage proportional to the position of the stylus. Thus the voltage from the phonograph pickup is proportional to the instantaneous sound pressure in the original program material being reproduced.

If a graph of the instantaneous sound pressure is drawn as shown in Fig 4-12 (A), a laterally modulated groove representing this same sound will appear as shown in Fig. 4-12 (B), where the lateral displacement relative to the position of the unmodulated groove has approximately the same curve as the graph of instantaneous sound pressure. (Since the record rotates in a clockwise direction, those sections of the groove at the left reach the stylus at a later time than those at the right, therefore the groove will actually be a left-right inverted representation of the sound pressure graph in which later times are shown at the right.)

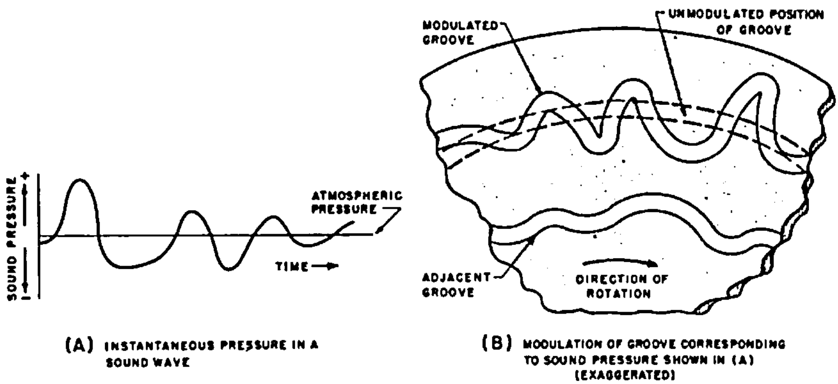


Fig. 4-12. (B) Modulation of groove corresponding to sound pressure shown in (A).

Record Standards. The present standard speeds of rotation of the record are 78.26 rpm, 33⅓ rpm, and 45 rpm. The 78-rpm rotation is the older standard which has been in use for many years, but is gradually being replaced by the newer 33⅓-rpm and 45-rpm standards for commercial pressings. Since the velocity of the groove past the stylus is much lower in the slower rotations, a smaller stylus and narrower groove must be used to give the same high-frequency response. Since they are narrower, the grooves can also be spaced closer together without interference between adjacent grooves. In 78-rpm recordings, a 2.5 or 3 mil (0.0025" or 0.003") stylus is used with approximately 100 grooves to the inch, while in 33⅓-rpm and 45-rpm commercial records a 1 mil (0.001") stylus is used with approximately 200 or more grooves to the inch. (The shape and dimensions of the grooves are chosen specifically for these stylus diameters, therefore use of different styli will give less satisfactory quality of reproduction.)

Stylus Wear. Because of the small amount of contacting surface the pressure between the stylus and the walls of the groove is very great, and the resulting friction causes the stylus to wear down after a relatively few playings. Since this can cause distortion and damage to records the stylus must be made of a very hard material if it is to be used for any appreciable number of records. The most common long-life styli are tipped with osmium, sapphire, or diamond.

The useful life of a record stylus depends on the type of equipment with which the pickup is used (equipment which reproduces the high frequencies fully will require a more nearly perfect stylus), the degree to which the listener is critical, the material from which the record is made, and the weight of the pickup. An ordinary listener with ordinary equipment will probably be able to use a sapphire stylus (3 mils) for about 1,000 12-inch, 78-rpm sides. A 1-mil sapphire stylus should be satisfactory for about 100 to 300 12-inch 33⅓-rpm sides. A more critical listener with wider range equipment will begin to notice distortion due to stylus wear with as few as one-quarter these numbers of playings. The life of an osmium is usually given as about one-sixth to two-fifths that of the sapphire, while the diamond stylus should last for about 10 to 20 times longer. Another authority has given the following figures for stylus life for the critical listener with wide-range equipment: an osmium-tip stylus can generally be used for about 35 playings of 12-inch, 33⅓ recordings, and about 100 playings of standard 12-inch 78-rpm recordings; a sapphire stylus is good for about 75 playings of 33⅓-rpm recordings, and about 250 playings of standard 78-rpm recordings; while a diamond stylus is good for over 1,000 33⅓-rpm playings, and over 2,000 standard-groove records.

Playing Time. A standard 78-rpm recording is capable of reproducing up to 4¾ minutes of program material on one side of a 12-inch disc. The slower speed of rotation and the closer groove spacing make it possible to record for a much longer time on 33⅓-rpm and 45-rpm discs of the same size. A commercial 33⅓-rpm recording is capable of reproducing from 20 to 25 minutes of program material on one side of a 12-inch disc. Because of the narrower groove and the longer playing time, the commercial 33⅓-rpm records are generally known as Microgroove or LP (Long-Playing) Records. The maximum playing time for the regular 45-rpm record is 5¼ minutes, while the extended-play 45-rpm record will provide almost 8 minutes of program material. All the various types of pickups may be used for both long-playing and standard 78-rpm record reproduction. The difference in pickups for these two types of recording are in the size of the stylus, the downward pressure of the stylus, the fact that long-playing record grooves have half the modulation, and slight differences in frequency equalization characteristic.

The limitations imposed by the dimensions of the groove and the physical characteristics of the recording and reproducing transducers make it necessary to use frequency response equalization in cutting and reproducing disc recordings. If a flat amplitude-frequency characteristic were used in the recording, the maximum groove displacement would be available at any frequency, but with present cutters this maximum amplitude cannot be attained at high frequencies and cannot be reproduced by the pickup without considerable distortion. To avoid this difficulty, a flat amplitude-frequency characteristic is used at the low frequencies below a *crossover frequency,* which is generally somewhere in the range between 300 cps and 800 cps. The characteristic curve drops off at a rate of 6 db for each octave above this crossover frequency. This type of high-frequency attenuation is one in which the peak velocity of the stylus is constant regardless of frequency, thus the recording characteristic is *constant amplitude* below the crossover frequency and *constant velocity* above this crossover frequency.

Frequency response curves of this recording characteristic are shown in Fig. 4-13. If the response is drawn on an amplitude basis it will appear as shown in (A) with constant amplitude at the low frequencies and dropping off at the high frequencies; if the record is played back by a pickup having an amplitude characteristic (crystal type), it will have this type of frequency response. If the response is drawn on a velocity basis it will appear as shown in (B), with constant velocity at the high frequencies and dropping off at the low frequencies; if the record is played back by a pickup having a velocity (magnetic type) characteristic, it will have this type of frequency response. When the record is played back, the output signal from the pickup must be properly corrected to give the original flat frequency characteristic which existed before the sound was recorded. This correction must be made for both the type of pickup and the exact crossover frequency, in order not to overemphasize either the low or the high frequencies.

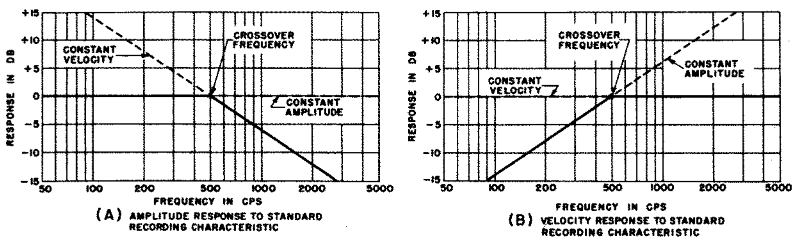


Fig. 4-13. Comparison of amplitude and velocity response to a standard 500 cps crossover recording characteristic (with no high-frequency pre-emphasis).

In practice, the standard recording frequency characteristic as shown in Fig. 4-13 is not used in exactly the form which has been described. To reduce the effects of record scratch and noise, a certain amount of pre-emphasis is used at the high frequencies. Under these conditions, a certain amount of high-frequency de-emphasis is required during playback. The use of high-frequency pre-emphasis to reduce reproduction noise will be discussed in detail in a later chapter.

Phonograph Reproducers

The program material on the phonograph record is reproduced by rotating the disc at the proper speed on a turntable, with the reproducing stylus in the groove so that the modulation causes a proportional output signal from the pickup. Therefore a phonograph reproducer for disc recordings consists essentially of (1) a turntable caused to rotate at the proper speed by a motor, (2) a pickup on which is mounted the stylus that rides in the groove to produce an audio output voltage from the pickup, and (3) an arm (the "tone arm") on which the pickup is mounted so that it can follow the spiral of the groove from the outside to the center of the disc.

The basic arrangement of these essential components into a phonograph record player is shown in Fig. 4-14. The record rests on the turntable, centered by a pin through its center. This turntable is driven by a small motor, either from the center or through a rubber wheel friction-

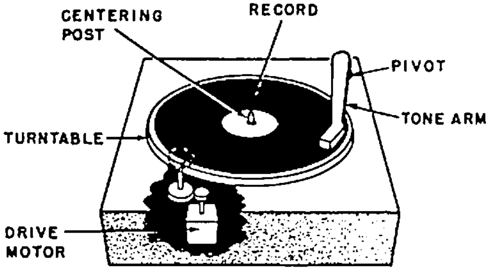
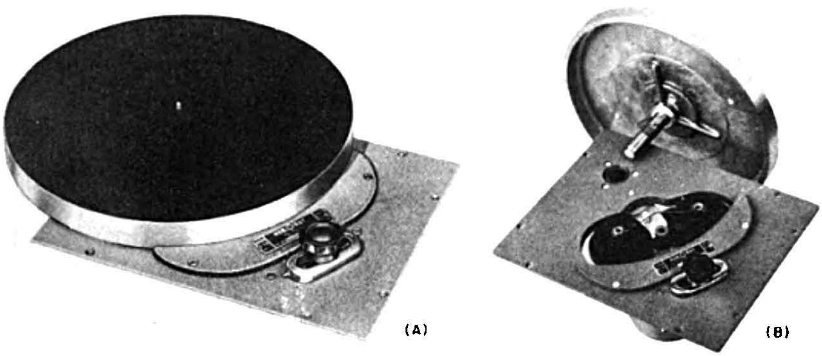


Fig. 4-14. Basic elements of the phonograph record player.

drive at the rim. The pickup is mounted on the tone arm, which is supported beyond the edge of the record on a pivot, so that the stylus can follow the slow lateral spiral of the groove toward the center without any motion along the length of the groove (since such a lengthwise movement would result in frequency modulation of the audio signal). The motor should be an induction or a synchronous motor sufficiently strong to drive the turntable at the correct speed even if there are variations in line voltage, and the turntable should be well balanced and sufficiently heavy to give a flywheel action so that any variations in the drive motor are not reflected in the record reproduction. The coupling system between the motor and the turntable should not cause any irregularities in the motion of the turntable (as, for example, caused by a flat spot on the rim of a rubber drive wheel), since such irregularities would cause unpleasant flutter, wows and low-frequency rumble to appear in the reproduced sound. The turntable speed should be very close to the correct speed, since otherwise the pitch of the reproduced sound would be undesirably different from that of the original. The tone arm should be rigid and have no resonances in the audible frequency range.

There can be a number of different types of phonograph record reproducers built according to these basic principles. The expensive broadcast-quality units are made with very large motors and very heavy turntables; considerable effort is spent to attain much closer speed regulation and much more accurate speed control than is practical for home reproduction. Good phonograph playback machines for home reproduction may be made for operation at a single speed or selection of different speeds, and may play a single record or be record-changers which will play a number of records automatically.

Single-Record Turntables. A high-quality, dual-speed turntable for the playing of single records up to 12 inches in diameter is shown in Fig. 4-15. It consists of a heavy cast-aluminum turntable, rim-driven by a hysteresis synchronous motor for extremely good speed regulation and stability. The turntable is machined and well balanced to eliminate, as far as possible, any variations in loading on the drive system with turntable rotation. As shown in part (B) of the figure, the turntable is driven from inside the rim by a neoprene idler wheel which is friction-driven by the motor. Either one of two idlers may be selected by a knob to give a choice of speed without stopping the turntable or removing the record. The two idlers may be selected to give any two of the three common playback speeds, and also may be replaced by an adapter



Courtesy: Rek-O-Kut Co.

Fig. 4-15. Commercial high-quality, dual-speed phonograph turntable.

to give a third speed whenever desired. When the table is not turning, both idlers can be disengaged from contact with either the motor shaft or the turntable, so that no flat spots on the neoprene can result from pressure during periods of idleness. The entire unit is mounted on a rigid cast-aluminum mounting plate, and the motor is shock-mounted to prevent any motor vibration from being transmitted to either the turntable or the tone arm. The turntable is mounted by this aluminum plate in the record-player cabinet, and the pickup tone arm is mounted at the appropriate position relative to the center of the turntable. A large number of high-quality tone arms are commercially available for use with this type of turntable.

This particular turntable is a unit which may be used in the highest quality home sound reproduction systems. Although it is not appropriate for radio broadcast applications, where a 16-inch turntable is generally required, it has been designed and constructed according to the same standards.

This turntable is also available with a less expensive four-pole induction motor which, while not capable of the same performance as the synchronous motor, will still give very good results.

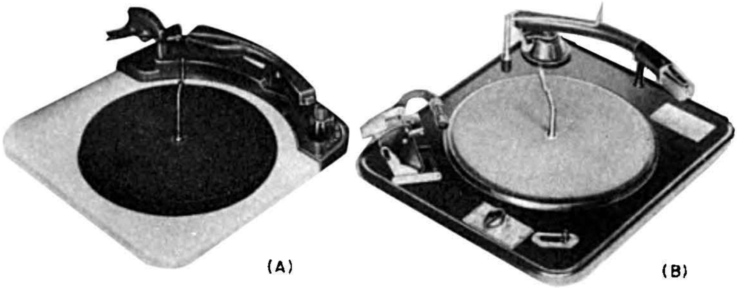
Fig. 4-16. Three-speed phonograph turntable.

Courtesy: Garrard Sales Corp.

Other single-record turntables are available in three-speed (see Fig. 4-16) and continuously variable speed models. The choice of the particular unit depends primarily upon the requirements of the specific installation.

Record Changers. Record changers are playback machines which are capable of playing a number of records in succession automatically. Generally the records to be played are supported above the turntable, and when one record has been finished the lowest record of the stack drops on top of the one which has been played, and is reproduced in this position. The records are usually not turned over, therefore only the "upper" sides can be played in succession and program material must be recorded in the proper sequence for continuity.

Photographs of typical record changers for home reproduction are shown in Fig. 4-17. These units can be used for the reproduction of 33⅓-rpm, 45-rpm, and 78-rpm records, the desired speed being selected by a knob. The turntable is weighted to give flywheel action, and is driven by a four-pole motor which maintains constant speed throughout a wide variation in line voltage, does not change speed appreciably regardless of the number of records on the turntable, and induces a minimum of hum pickup into low-level pickup output leads. The turntable



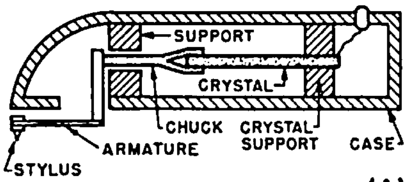
Courtesy: (A) Webster-Chicago Corp. and (B) Garrard Sales Corp.
Fig. 4-17. Typical record changers for home reproduction.

is friction-driven by a rubber idler wheel which is pulled away when the turntable is not running, to avoid flat spots in the rubber which would otherwise be caused by the pressure against the rubber during periods of idleness.

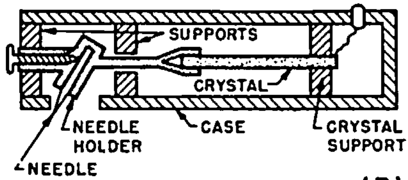
The changer mechanism operates by supporting the stack of records on a platform at the edge, and by a spindle through the center. When the record on the turntable has been finished, an eccentric groove close to the center of the record initiates the record-changing cycle. The platform pushes on the edge of the bottom record of the supported stack, to push it off the supporting ledges of the platform and the center spindle, then the record drops on top of whatever records may already be on the turntable. A muting switch disconnects the pickup from the audio input line while the changer is operating on the run-in or run-off grooves, so that there is no noise while records are being changed. The proper stylus is chosen either by means of a small knob in the pickup head (which either turns over the cartridge so that the proper stylus is exposed or places the proper stylus into playing position), by means of turning over the entire head (with its two styli), or by plugging another head with the proper stylus into the arm.

High-Impedance Phonograph Pickups

Crystal Pickup. The high-impedance crystal pickup is, at the present time, the most widely used of any type of pickup for the reproduction of phonograph records. The basic construction of a low-noise pickup is shown in the diagram of Fig. 4-18 (A). It consists of a torsional Rochelle salt crystal element which is caused to twist by the modulation of the groove in the record, and thus produces an electrical signal whose amplitude and frequency are proportional to the groove modulation. The crystal is supported in the tone arm so that one end is held securely in place, while the other end is coupled to the needle and is supported flexibly so that it can be twisted by the modulation of the groove.



(A)



(B)

Fig. 4-18. Internal construction of (A) a typical low-noise crystal pickup, and (B) a typical removable-needle crystal pickup.

Another type of construction of crystal pickup is that shown in part (B) of the figure, in this type the needle is mounted in a chuck from which it is removable. The needle chuck may be connected to the crystal either by a rigid, direct mechanical coupling, or through a rubber coupling which acts as a mechanical transformer.

The response of a crystal pickup is proportional to the amplitude of the groove modulation, therefore its response to a 500 cps crossover frequency characteristic would be somewhat as shown by the typical curve at *A* of Fig. 4-19. When high-frequency pre-emphasis is used for noise reduction, the high frequencies must be attenuated in playback, therefore the constant-amplitude characteristic of the crystal can be used to give this required attenuation. The resulting response of curve *A* to the standard NAB pre-emphasized recording frequency response characteristic is given in curve *B* of the figure.

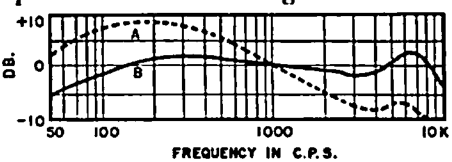


Fig. 4-19. Typical frequency response of a low-noise crystal pickup.

In some of these pickups the active element is barium titanate in the form of a ceramic. Such a pickup is practically unaffected by extremes of temperature or humidity which may affect ordinary unprotected Rochelle salt.

Condenser Pickup. Another high-impedance pickup is the condenser pickup, whose operation is similar in principle to that of the condenser microphone. One capacitor plate is fixed on the tone arm, while the other capacitor plate is movable and coupled to the stylus in the groove. The modulation of the groove therefore causes a variation in capacitance by variation of the spacing between the two plates. Good condenser pickups are capable of very good reproduction from records, but at the present time they are not widely used because of the circuit complications which are involved in their use.

Low-Impedance Phonograph Pickups

Most of the low-impedance pickups which are in current use are magnetic pickups, in which variations in the magnetic field, or in the magnetic induction, generate an electrical signal proportional in magnitude to the amount of motion of the needle in the groove. The two most important types of magnetic pickups at the present time are the *variable-reluctance* pickup (see Fig. 4-20) and the *moving-coil* pickup.

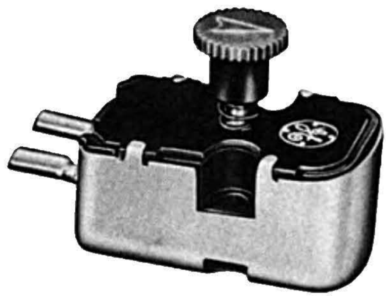


Fig. 4-20. One type of variable-reluctance phonograph pickup. The knob shown is used to position either one of two styli tips for the reproduction of standard or long-playing records.

Courtesy: General Electric Co.

Variable-Reluctance Pickup. The basic principle of operation of the variable-reluctance pickup can be understood by reference to Fig. 4-21. The magnetic lines of force are concentrated in the iron and in the air gap as shown in the diagram. For the magnetic field, this may be thought of as a "magnetic circuit" analogous to an electrical circuit in which the iron or steel is a conductor and the air gap a resistance. If the length of the air gap changes, the reluctance of the magnetic circuit changes, resulting in an increase or a decrease in the number of magnetic lines of force. If a coil is wound around the iron in the magnetic circuit, a

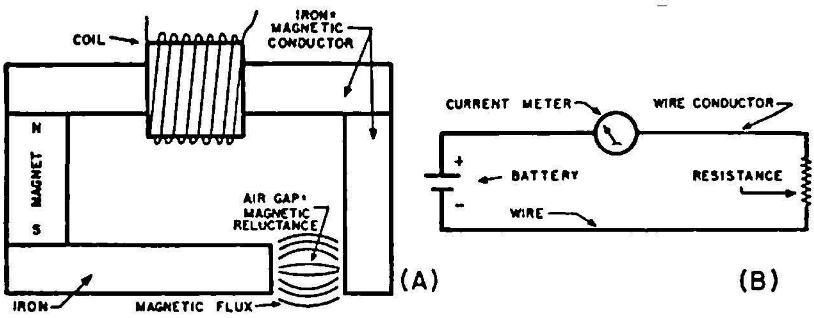


Fig. 4-21. (A) Simple magnetic circuit. (B) Analogous electrical circuit for this simple magnetic circuit.

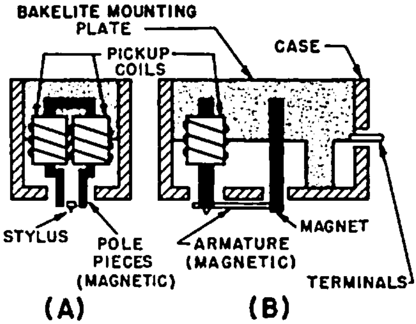


Fig. 4-22. Simplified diagram showing basic internal construction of a variable-reluctance magnetic pickup.

voltage is generated in the coil due to the change in the number of magnetic lines of force through the coil.

The construction of a variable-reluctance pickup is shown in detail in Fig. 4-22. The stylus is mounted on an armature of magnetic material which is fixed at one end and can be moved at the stylus end toward either of two coils by the modulation of the record groove. The fixed end of the armature is close to a small bar magnet, while a Mumetal pole piece through the two coils is solid at the top and split at the lower end near the stylus armature. Thus the magnetic circuit is from the bar magnet through the air into the coil yoke at the top, through the two sections of the yoke and into the armature equally from each pole piece when the stylus is in the undeflected position, then back through the armature to the bar magnet. When the armature is deflected by the stylus toward either one of the sections of the yoke, the magnetic field through that coil is increased and the field through the other coil decreased. The windings are in series, so that a push-pull action results. Since vertical motion of the armature does not cause the magnetic field

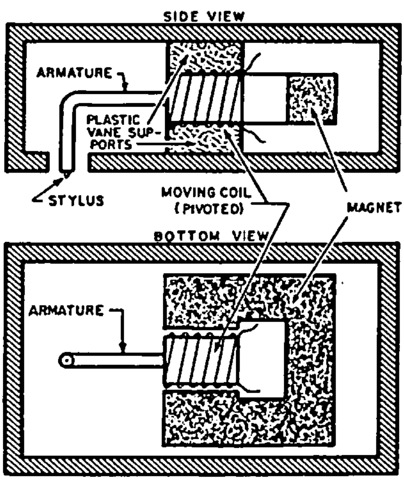


Fig. 4-23. Simplified diagram showing the internal construction of a typical moving-coil dynamic pickup.

in one coil to be greater than in the other, no voltage is generated due to vertical components of the motion, and the output voltage due to record scratch is, therefore, reduced. The armature is the only moving component in this pickup, and since this mass is extremely small the high frequency response is excellent.

Dynamic Pickup. In the dynamic pickup, the voltage is generated by the motion of a coil in a constant magnetic field. (The principle of operation of this type of pickup is the same as that of the dynamic microphone, although the physical construction is, of course, different.) The construction of a typical moving-coil pickup is shown in the diagram of Fig. 4-23. The stylus is mounted in one end of an armature,

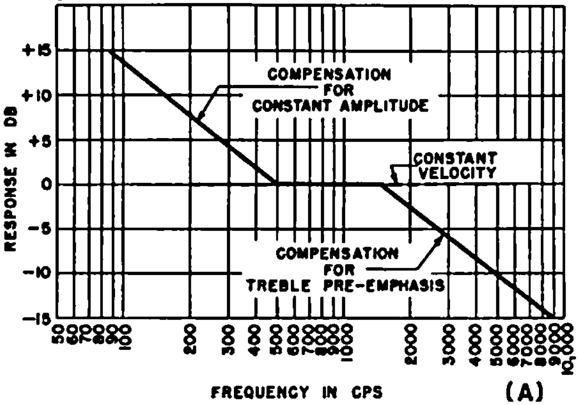
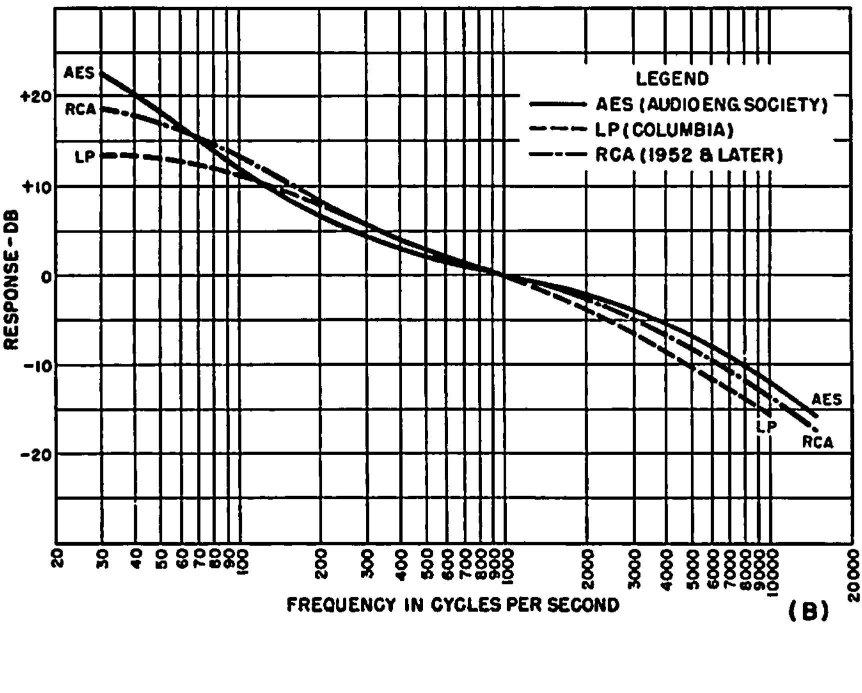


Fig. 4-24. (A) Generalized equalization curve for a magnetic pickup. (B) Specific equalization curves required for playback of commonly used recording characteristics.



around whose other end is wound a small coil on a thin sleeve of silicon steel. This coil is supported in the magnetic field between the two poles of a magnet, and is mounted so that lateral motion of the stylus causes it to pivot in the magnetic field about its own center. This motion in the magnetic field generates a voltage in the coil proportional to the velocity of motion. Because the moving system is very light and pivots about a small radius, good high-frequency response is obtained; and because of the design of the coil in respect to the magnetic field, vertical motion produces no voltage in the coil. The results obtained with the dynamic pickup are approximately the same as can be obtained with the various forms of variable-reluctance pickups.

The output voltage of a magnetic pickup is proportional to the lateral velocity of the stylus in its deflection by the groove modulation. The basic recording frequency response characteristic is constant amplitude at low frequencies and pre-emphasized above constant velocity at the high frequencies, therefore the pickup output must be equalized to give the correct output frequency response, as shown in generalized form in Fig. 4-24 (A). (Note: The exact shape of this curve depends on the exact nature of the recording characteristic used, as shown in part (B) of the figure, and should be the complement of the recording curve.) Since the output voltage is low (approximately 0.01 volt rms at 1,000 cps for a standard 78-rpm record), an additional preamplifier with about 40 db gain is required to raise the signal level to compare with the average crystal pickup. Generally this preamplifier includes the equalization so that its output need just be plugged into a flat-frequency-response amplifier.

A-M and F-M Radio Tuners

The two major sources of program material in home sound reproduction are phonograph records and radio tuners. The radio tuner is a means of reproducing sound originating at some other location and being transmitted in the form of a radio broadcast. Radio-frequency carriers may be modulated in any of a number of different ways, but at the present time the most widely used methods of modulation are by changing either the amplitude or the frequency of the carrier. Amplitude modulation of low-frequency carriers in the standard broadcast band of approximately 0.55 mc to 1.6 mc, and frequency modulation of high-frequency carriers from 88 mc to 108 mc are used almost exclusively for commercial broadcasting of audio signals.

Amplitude Modulation. The earliest and most widely used method of radio transmission is by amplitude modulation in which a radio-frequency carrier is varied in amplitude according to the waveshape of the audio signal being transmitted, as shown in the diagram of Fig. 4-25 (B).

A more recently developed method of transmission is by frequency modulation, in which the frequency of a constant-amplitude carrier is varied by an amount proportional to the instantaneous sound pressure of the audio signal being transmitted, as shown in Fig. 4-25 (C).

In present commercial practice, the quality of reception of amplitude-modulated signals is not generally good enough for high quality reproduction of sound. The transmitted signal from the broadcast station is generally quite good in quality, but most commercial a-m receivers attenuate the frequencies above about 5,000 cps, so that the reproduced sound is lacking in high frequencies. A good receiver with sufficient bandwidth will give extremely good reproduction of amplitude-modu-

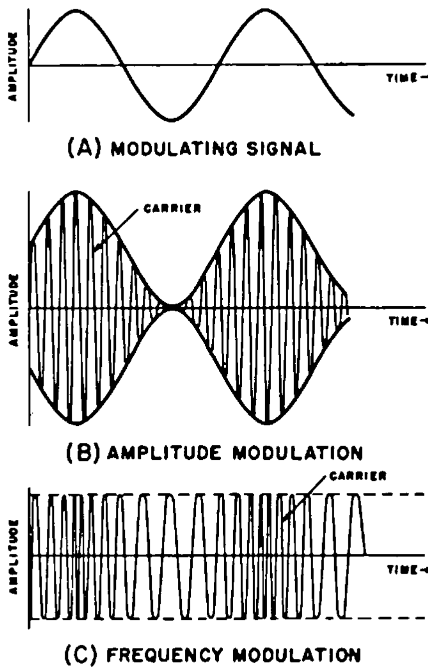


Fig. 4-25. Comparison of amplitude and frequency modulation for radio transmission.

lated signals; however, when the signal strength is weak the noise picked up by the receiver is of the same nature as the modulation, therefore any such noise is quite obvious in the reproduced sound. The purpose of the restricted bandwidth in commercial receivers is primarily to decrease the amount of noise in the reproduced sound, since this noise is more objectionable to the listener than the loss of high-frequency response.

Frequency Modulation. When frequency modulation is used for the transmission of sound, much higher fidelity of reproduction is obtained than with amplitude modulation. There are two reasons for this:

(1) Frequency-modulated signals are constant in amplitude, therefore in the receiver the received signals are amplified to a level at which the peak amplitudes can all be clipped at a constant level by a limiter, thus eliminating the effects of any amplitude-modulation noise.

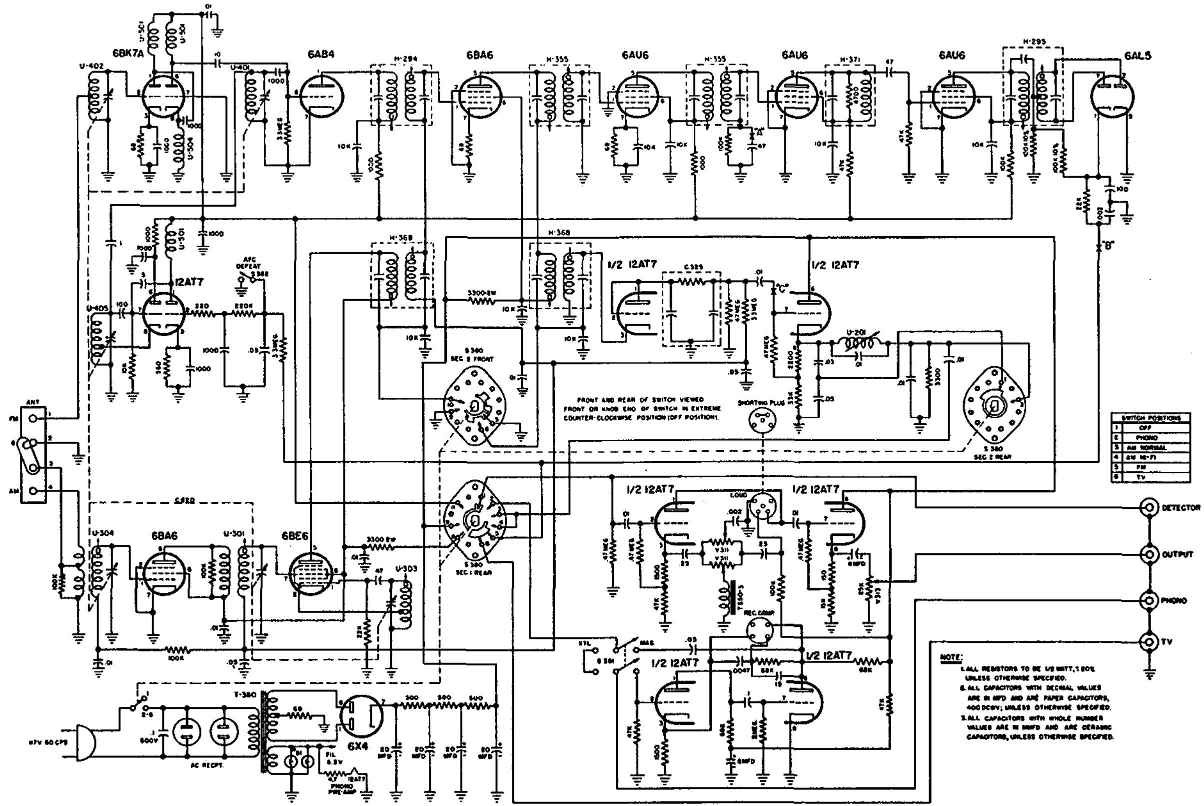
(2) Because any noise which is picked up by the receiver is proportional to the bandwidth and is therefore worse at high frequencies, a system of high-frequency pre-emphasis in transmission and de-emphasis in reception has been made standard in f-m broadcasting. Thus, high frequencies are boosted in the audio signal before modulation, so that when the high frequencies are attenuated in the receiver to restore a flat-freqquency-response characteristic, the noise will also be attenuated by this amount. This process of pre-emphasis for noise reduction will be discussed in further detail in a later chapter.

The use of high-frequency pre-emphasis would also improve the quality of reproduction obtainable with a-m broadcasting, but since the standards have already been fixed and so much equipment is already in use, there is very little likelihood of any change in the present broadcast standards. Therefore, at the present time, the best a-m receivers are capable of giving good frequency response to about 10,000 cps with low noise level for strong signals, while f-m receivers give response to 15,000 cps with better signal-to-noise ratio and dynamic range than can be obtained from a-m broadcasting.

Radio Tuner. The radio tuner is actually a complete receiver, except that it does not contain an audio power amplifier. It includes the complete r-f and i-f sections of the receiver, and generally at least one stage of audio amplification. The audio output level from a tuner is generally of the order of 2 volts or more into a high-impedance circuit, although some tuners have an additional low-impedance output at about a 0 dbm level into 600 ohms. Commercial tuners may be obtained for either a-m or f-m reception alone, or for both a-m and f-m reception; most of them give very good quality of reproduction. The simplest units may contain only a single stage of audio amplification and receive

Fig. 4-26. Photograph of a typical a-m/f-m tuner.

Courtesy: David Bogen Co.



Courtesy: *David Bogen Co.*

Fig. 4-27. Circuit diagram of a typical a-m/f-m tuner.

only one type of transmission. The more expensive and versatile units receive both a-m and f-m broadcasts, and contain audio amplifiers with tone controls, switching circuits, and preamplifiers for low-level magnetic phonograph pickups.

The circuit diagram of a typical a-m/f-m tuner (whose photograph appears in Fig. 4-26) is shown in Fig. 4-27, and is seen to be quite straightforward in design. The f-m section consists of an r-f amplifier stage, a mixer, two stages of i-f amplification, two stages of limiting, and the discriminator. The oscillator section includes an automatic-frequency control feature, by means of which a reactance tube in the tuned circuit is controlled by the d-c component of the discriminator output, so that the oscillator always remains on the correct frequency. This afc feature eliminates any oscillator drift and simplifies tuning, and may be disabled by a switch when it is desired to tune in a weak signal close to a strong one. The a-m section is conventional in design, containing an r-f amplifier, mixer, i-f amplifier, and a detector. In this particular unit, the proper networks and tube sections are switched into the circuit by the a-m/f-m selector switch. The circuit also includes a two-stage audio amplifier and a low-level preamplifier-equalizer for use with magnetic pickups.

When this tuner is used in a sound reproduction system, all of the different input signals to be reproduced are connected into the tuner, and the selector switch is used to choose the particular signal which is to be reproduced. The two-stage amplifier, through which the selected signal is amplified, contains complete tone controls and volume control, and has a low output impedance. The output signal from the tuner can, therefore, be fed directly to a remotely located high-level amplifier, with all the operating controls on the panel of the tuner.

The performance characteristics of this particular tuner may be summarized as follows:

Sensitivity: FM: 3 microvolts, for 30 db quieting
AM: 5 microvolts
Audio output: 3 volts at 6,000 ohms
Distortion: 3 volts at 0.2 percent harmonic distortion
Hum and noise: FM, AM: –65 db below 100 percent modulation TV, PHONO: –65 db below 2 volts
Phono preamplifier: 35 db gain, plus 21 db equalization at 30 cps
Tone control: Bass: 17 db boost to 19 db attenuation at 10,000 cps
Frequency response: FM: 20-20,000 cps, within ±0.5 db.
AM: 20-7,500 cps, within ±0.5 db (Hi-Fi pos.)

These characteristics are extremely good and are more or less typical of a large number of commercial a-m/f-m tuners which have been designed for use in high-quality sound reproducing systems. Most of these units are basically similar in circuit design, but may be different in many details and have different features and characteristics.

The integration of different types of radio tuners into sound reproduction systems presents individual problems which will be discussed in detail in a later chapter.

Chapter 5

AUDIO AMPLIFIER THEORY

General

An amplifier consists of a number of stages of amplification combined in such a manner as to meet the requirements of the system. In general these requirements, and the important characteristics of the amplifier, will be:

(1) Overall gain.
(2) Input and output impedance.
(3) Input and output voltages.
(4) Frequency response.
(5) Distortion.

The gain, terminal impedances, and signal voltage levels are the amplifier design specifications, while the distortion and frequency response characteristics should always be the best that can be reasonably attained.

The required gain and output voltage can be obtained by using a sufficient number of amplifier stages, while the correct impedances can be obtained by resistance or transformer matching. However, it is more difficult to obtain the required good frequency response and distortion characteristics simply by use of basic, straightforward amplifier techniques. Bad frequency response can be corrected to a certain extent by equalizer circuits, but once distortion is present in the audio signal there is no way of removing it. The best method of obtaining good frequency response and low distortion is by the use of negative feedback. From the specific requirements of the system which is under consideration, it is possible to decide upon a good general tube lineup, choose the specific tube types to be used, the voltage gain of each stage, and the specific values of the circuit components to be used.

The Voltage Amplifier Stage: Graphical Construction

The most important single component of any complete amplifier is the single vacuum-tube amplifier stage (since the amplifier is essentially a combination of single stages) ; the correct design and operation of each stage is necessary for proper operation of the system. For any experimenting in sound and audio reproduction, it is essential for the experimenter to understand the basic operation and the fundamental principles of design of amplifiers so that he can better understand the circuits with which he is working. He can then design and construct equipment more intelligently, and will find troubleshooting easier and quicker in case of circuit failure. The design of the amplifier is done graphically by use of the curves of tube operation published by the tube manufacturer, and does not require the use of complicated mathematical formulas or extensive calculation.

The use of these curves in the design makes it possible to predict in advance what the performance will be, without the necessity of first building the circuit in order to find out whether it meets the requirements. The most important curves of vacuum-tube operation are called the *plate characteristics*, which consist of a number of curves in which the plate current is plotted against the plate voltage, each curve being for a different constant grid-cathode voltage. There are the tube characteristic curves which are most generally given by the manufacturer in the tube manuals, and are most often used in the amplifier design procedure. The plate characteristics of a typical widely used triode (one section of the type 6SL7 dual triode) are shown in the diagram of Fig. 5-1. These curves show the variation of plate current with plate voltage for different constant values of grid voltage from 0 to −7 volts. The various tube factors (plate resistance, amplification factor, transconductance) are determined from these curves.

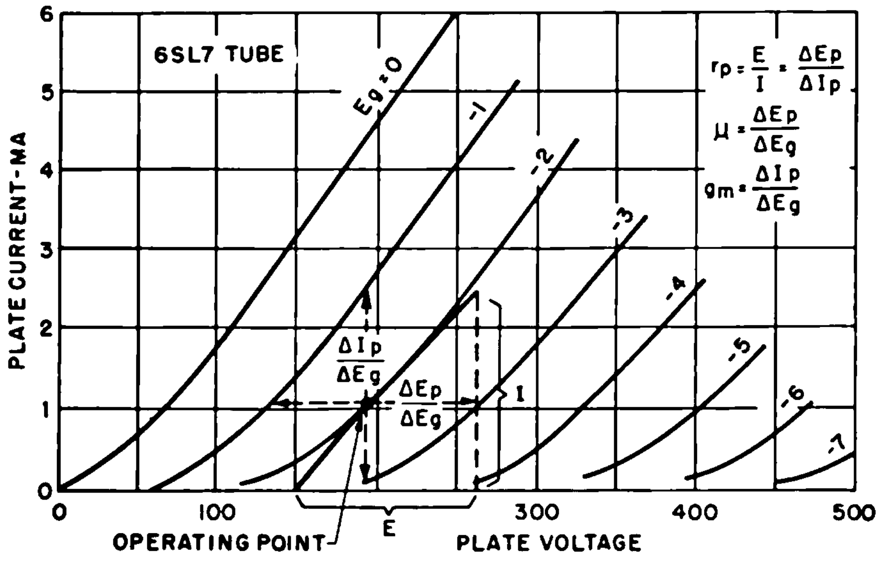


Fig. 5-1. Use of plate-current characteristics to determine plate resistance, amplification factor, and transconductance of a tube.

The plate resistance r_p of the tube for a given grid voltage is immediately obtained from Ohm's law if it is correctly applied. If a straight line is drawn tangent to the curve at the operating point in question, the inverse slope of this line represents the plate resistance. From the voltage E and the current I shown in the diagram, Ohm's law gives this resistance as E/I. The amplification factor μ is the change in plate voltage with change in grid voltage, for constant plate current. This is found simply by measuring along a horizontal line of constant plate current, and dividing the change in plate voltage by the change in grid voltage required to produce this voltage change. The transconductance g_m is the change in plate current with the change in grid voltage, for constant plate voltage. This is found by measuring along a vertical line of constant plate voltage, and dividing the change in plate current by the change in grid voltage required to produce this current change.

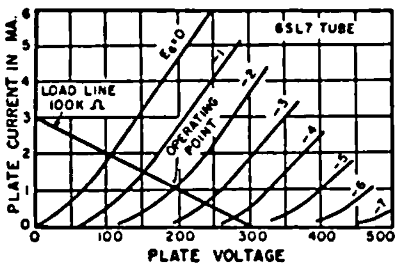


Fig. 5-2. Use of plate current characteristics and load line to design a voltage amplifier stage. The load line which is superimposed on the tube characteristics represents the voltage at the plate of any tube connected to a 300 volt B supply through a 100,000-ohm resistor.

In a graph of current plotted against voltage, consider the straight line which is drawn as shown in Fig. 5-2. This line represents the effect of a resistance in series with the plate of the tube, and the line will be the same regardless of what tube characteristics are drawn on the graph. Any point on this line shows the voltage from plate to ground for the particular current which is indicated. For example, when the tube draws no current the full supply voltage is on the plate since no voltage is developed across the resistor; when the tube draws maximum current, the entire voltage is developed across the resistor and the voltage from plate to ground is zero. This line is known as the *load line,* since it represents the operation voltage of the tube for this specific value of load resistance.

If a load line is drawn over a set of plate characteristics of a specific tube, the resulting curves will give the operating characteristics of the tube for the particular power supply voltage and plate resistance which have been selected. Consider, for example, the set of plate characteristics and the load line which are drawn together in the graph of Fig. 5-2. These particular curves represent a typical triode amplifier stage and practical circuit values which are widely used in audio amplifier design.

The tube characteristics are those of the 6SL7 tube, and the load line represents a 100,000-ohm plate resistance for a power supply voltage of 300 volts d.c. The load line is drawn from the facts that (1) when there is no current the voltage from plate to ground is 300 volts, and (2) when the voltage from plate to ground is zero the voltage across the resistor is 300 volts, resulting in a current of 3 ma through the resistor.

These two points are connected with a straight line. All the points along this line then show the operation of the tube under these conditions. For example, if the grid voltage of the tube is selected as −2 volts, then the current through the tube is given by the intersection of the load line with the −2 volt grid-voltage line, showing that the current is approximately 1 ma and the voltage from plate to ground is approximately 200 volts.

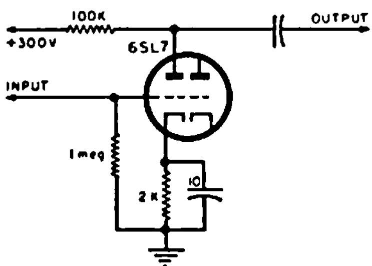


Fig. 5-3. Voltage amplifier stage designed from the curves shown previously.

This information can be used to design the amplifier stage shown in Fig. 5-3. The grid-cathode bias voltage is obtained by means of a bypassed 2,000-ohm resistor between cathode and ground. The effect upon the plate circuit of a signal voltage applied to the grid can be seen by taking the different points along the load line and observing the plate voltages and currents. Thus, a peak grid swing of +1 volt to −1 volt will cause the grid-cathode voltage to swing between −1 and −3 volts, and the voltage at the plate will swing from about 155 to about 245 volts, which is 45 times the grid signal voltage.

The Voltage Amplifier Stage: Equivalent Circuit

The circuit of Fig. 5-3 can be redrawn in another way which makes it possible to predict the frequency response and output impedance of the amplifier stage without the necessity of building the circuit in order to measure it. This method of redrawing the tube circuit is shown in Fig. 5-4, drawn also to include the grid input circuit of the following tube. The amplifier circuit has the same characteristics for the a-c signal as if the voltage $-\mu e_g$ were applied in series through a resistor (R_p) equal to the plate resistance of the tube to the load circuit, which consists of the plate load resistor R_b to ground and the coupling capacitor (C_c) to the grid and grid resistor (R_g) of the next stage. Also in the

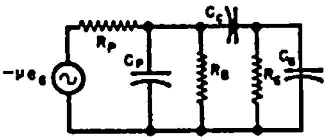


Fig. 5-4. Equivalent plate circuit of the amplifier stage shown previously, including the grid input circuit of the following tube.

circuit are the plate-cathode capacitance and the plate circuit wiring capacitance (C_p), as well as the next tube input capacitance and grid circuit wiring capacitance (C_g) to ground. The circuit of Fig. 5-4 is called the *equivalent plate circuit* of the amplifier.

The manner in which the equivalent plate circuit can be used to predict the amplifier performance can be seen from the three circuits shown in Fig. 5-5, which are derived from the circuit of Fig. 5-4. These circuits show the components which are important at the low, the middle, and the high frequencies. At middle frequencies the series coupling capacitor can be replaced by a short circuit and the shunt capacitors by open circuits, leaving only the resistances in the circuit. The gain is then determined by the voltage divider composed of the R_b and R_g parallel combination in series with R_p. If the resistance of R_b in parallel with R_g is calculated and called R_l, then the gain of the stage from grid input signal to signal applied to the next grid is equal to $-\mu e_g R_l\ (R_p+R_2)$. This value is negative because the signal in the plate circuit of a vacuum tube is opposite in phase to the grid input signal.

At low frequencies, the impedance of the coupling capacitor must be considered in series with the following grid resistor. This impedance determines the low-frequency response of the amplifier circuit, since it forms one arm of a voltage divider whose output decreases as the capacitor reactance increases for lower frequencies. The response is 3 db down (71 percent of maximum voltage) at the frequency where the reactance of the coupling capacitor is equal to the grid resistor, and approaches a falling off of 6 db for every octave below this frequency.

At high frequencies the shunt capacitances must be considered. The shunting capacitance is the total capacitance to ground on both the plate and grid sides of the coupling capacitor. This capacitance determines the high-frequency response of the amplifier stage since it

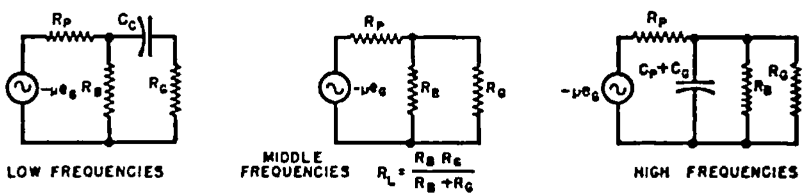


Fig. 5-5. Simplified forms of the equivalent circuit, showing the components which are important at the low, middle, and high frequencies.

is in parallel with the shunt arm of the plate-resistance/load-resistance voltage divider and causes the output voltage to decrease as the capacitive reactance decreases for higher frequencies. The response is 3 db down at the frequency where the reactance of the total shunt capacitance is equal to the combined resistance R_1, and approaches a falling off of 6 db for every octave above this frequency.

The effects of these capacitances account for the frequency range limitations of resistance-coupled voltage amplifiers. The resulting frequency response due to these effects is of the type shown in Fig 5-6.

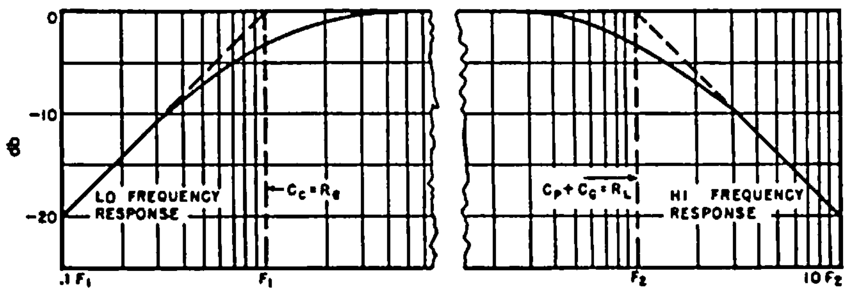


Fig. 5-6. Frequency response of an amplifier stage at low and high frequencies due to the shunt and coupling capacitances as shown by the equivalent circuits.

Amplifier Design Procedures

From the information contained in the plate characteristic curves and the equivalent circuit of the amplifier stage, its performance can be quite accurately predicted. The actual details and arithmetic of these procedures, that is, the basic practical steps in selecting the circuit values for an amplifier design, together with a brief summary of the most essential points which have been described in the previous sections, are presented in the following paragraphs:

(1) The *gain* of the stage is determined from the equivalent circuit of Fig. 5-5. The circuit at middle frequencies is used for this calculation, and the gain at high and low frequencies are obtained from the frequency response curve. The formula used for this calculation is:

$$\text{gain} = -\mu\frac{R_l}{R_p + R_l}$$

where R_l equals $(R_b\, R_g)/(R_b+R_g)$ and is the total resistive load in the plate circuit, and the negative sign indicates that there is a 180-degree change of phase in a single-tube amplifier. For pentodes a more convenient simplified formula is:

$$\text{gain} = -G_m R_l$$

which is approximate but fairly accurate, because of the high plate resistance of pentodes.

(2) The *output impedance* of the tube is important when matching to attenuators, equalizers, transmission lines, and various other types of networks. It can be determined from the equivalent circuit of Fig. 5-7. The grid resistor generally does not exist in such circuits, and the tube circuit is considered to consist of the circuit elements up to this point, as shown in Fig. 5-7 (A). The output impedance at middle frequencies therefore appears as a resistance tube plate resistance $(R_p\ R_b)/(R_p+R_b)$ (in series through the coupling capacitor), as shown in Fig. 5-7 (B).

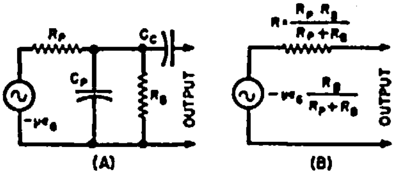


Fig. 5-7. Method of calculating the output impedance of an amplifier: (A) equivalent plate output circuit of a typical triode amplifier stage; (B) the effective output circuit at the middle frequencies.

(3) *Frequency response* can be predicted from the equivalent circuits at high and low frequencies shown in Fig. 5-5, together with the curves of Fig. 5-6. The coupling capacitor and the following grid resistor give the low frequency response, while the total shunt capacitance and the load resistance give the high frequency response. In determining the high frequency response, it is essential to take the input capacitance of the following tube into account.

(4) *Harmonic distortion* can be measured from the plate current characteristics of the tube as given in tube handbooks and from the load line. Considering the set of curves shown in Fig. 5-8, it can be seen that if +1 volt is added to the grid voltage (–2 volts) to change the bias to –1 volt, the voltage at the plate decreases by about 35 volts, while if –1 volt is added to the grid voltage to change the bias to –3 volts, the voltage at the plate increases by about 35 volts. Therefore for a +1 volt peak grid swing this amplifier is very linear and shows little distortion. However, when +2 volts is added to the grid voltage to change the bias to 0 volts, the plate voltage decreases by 70 volts, while it only increases about 50 volts when –2 volts is added to the grid voltage to change the bias to –4 volts. Therefore, for a 2-volt peak grid swing the amplifier is not linear, and harmonic distortion is introduced into the output signal. This is illustrated graphically in the figure. The maximum signal which can be applied to the grid of the amplifier is that voltage which will still produce linear changes in plate voltage as measured on the plate characteristic curves.

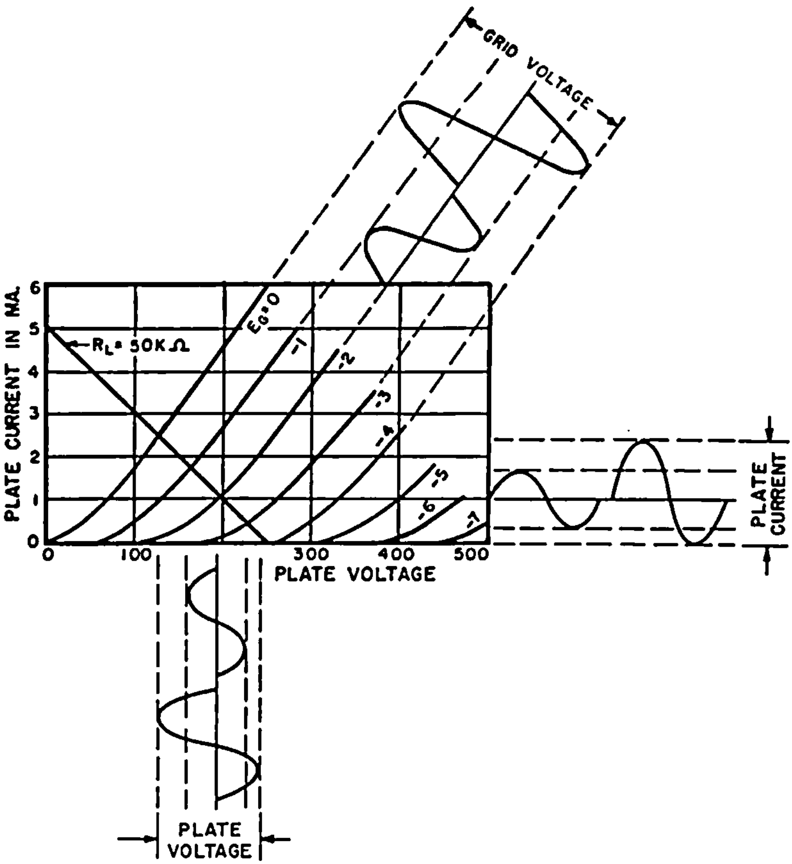


Fig. 5-8. Curves showing how large input signals cause distortion in a vacuum-tube amplifier.

These formulas and curves contain sufficient information for the design of the voltage amplifier stage and for predicting accurately what will be its performance under practical operating conditions.

Miller Effect

An extremely important factor which imposes certain limitations on the practical design and choice of tubes in audio amplifiers is the *Miller effect*. In the equivalent circuit of the amplifier, shown previously in Fig. 5-5, it can be seen how the capacitance in the grid circuit of the following stage affects the response at high frequencies. Thus, if the input capacitance of the following stage is high, the effect on the high-frequency response may be considerable. This input capacitance is not a constant of the tube and is different when there is a load in the plate

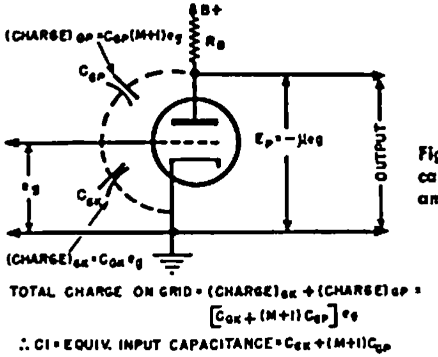


Fig. 5-9. Miller effect increase in input capacitance of an amplifier stage due to amplification by the tube with a resistive plate load.

circuit and amplification occurs than when there is no load. This effect is known as the *Miller effect.*

When there is a resistive load in the plate circuit of a tube, as shown in Fig. 5-9, the voltage on the plate is $-M$ times the voltage on the grid (M being the amplification of the stage). The difference in voltage between the grid and plate is $e_g-(-Me_g)$ or $e_g\ (M+1)$. Since the charge on any capacitor is equal to the product of the capacitance and the voltage applied, the charge on the grid-cathode capacitance (C_{gk}) is $C_{gk}e_g$. Likewise the charge on the grid due to the grid-plate capacitance and the voltage $e_g\ (M+1)$ is $C_{gp}\ (M+1)\,e_g$. Hence the total charge on the grid is $(C_{gk} + (M+1)\ C_{gp})\,e_g$. The total input capacitance corresponding to this total charge is $C_{gk} + (M+1)\,C_{gp}$. In high-gain triodes this effect is quite large, whereas the Miller effect is negligible for pentodes because they have such a low grid-plate capacitance.

As an example of the importance of the Miller effect, consider one section of a 6SL7 as a voltage amplifier:

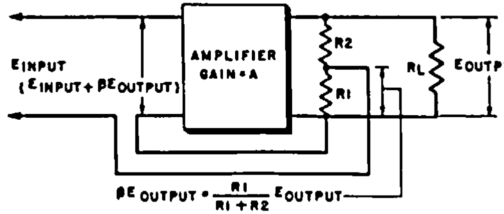
$$C_{gk} = 3.4\ \mu\mu\text{f},\ C_{gp} = 3.2\ \mu\mu\text{f}$$
$$M = 41$$
$$C_{\text{total input}} = 3.4 + (41 + 1)\ 3.2 = 137.8\ \mu\mu\text{f}$$

This value of total input capacitance is 21 times the input capacitance of 6.6 $\mu\mu$f with no load in the plate circuit. This is quite a high capacitance, and if this input capacitance is located in a high-impedance circuit (such as the plate circuit of a pentode amplifier stage, or a 1-megohm volume control), it will cause a serious loss of high frequencies. In the worst cases, this high-frequency loss cannot even be compensated for by equalization. For example, in a 1-megohm volume control the attenuation at 10,000 cps may vary between 0 and 10 db, depending upon the volume control setting, and no one equalization curve will effectively compensate for all volume control settings. The only practical method of eliminating the problems of the Miller effect is to avoid the use of high-gain triode stages in high-impedance circuits whenever good high-frequency response is desired, and to design the circuit and choose tube types with this limitation in mind.

Feedback

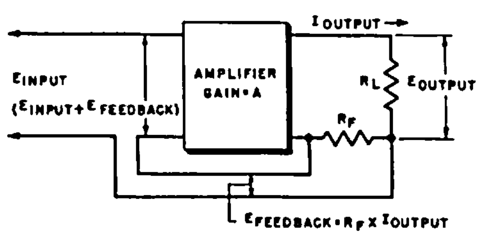
A feedback amplifier is one in which a certain amount of voltage or current from the output is introduced back into a previous stage or circuit of the amplifier. The basic circuits for feedback in an amplifier are shown in the diagrams of Figs. 5-10 and 5-11. The arrangement in Fig. 5-10 shows the circuit for voltage feedback, in which the amount of signal that is fed back is proportional to the voltage across the output load by means of the high-resistance voltage divider. The arrangement in Fig. 5-11 shows the circuit for current feedback, in which the amount of signal that is fed back is proportional to the current through the output load by means of the resistor placed in series with the output load resistance.

Fig. 5-10. Block diagram of the basic circuit for the application of voltage feedback to an amplifier by feeding a fraction of the output voltage back to the input.



The diagrams show that the voltage fed back from the output will add or subtract from the input signal, depending upon whether the two voltages are in-phase or out-of-phase. Suppose, for example, that the voltage fed back is in-phase so that it adds to the input signal to increases the gain of the system. However, if the voltage which is fed of the amplifier and, by increasing the output voltage, effectively increases the gain of the system. However, if the voltage which is fed back is out-of-phase so that it decreases the input signal, the output voltage is decreased and the gain of the system is effectively reduced. A feedback connection of the type in which the voltages add together is known as *positive feedback,* while the type of connection in which the voltages subtract is known as *negative* (or *degenerative*) *feedback.*

Fig. 5-11. Block diagram of the basic circuit for the application of current feedback to an amplifier, by feeding back to the input a voltage proportional to the output current.



The actual gain of an amplifier with feedback can be calculated if the feedback factor and the gain without feedback are known. If the fraction of the output voltage which is fed back is $\pm\beta$ (the plus and minus sign indicates positive or negative feedback), and the amplifier gain without feedback is A, then the output voltage with feedback is $A\,(E_{input} \pm\beta\, E_{output})$. The gain of the amplifier (with the input signal increased sufficiently to give the same output voltage as was obtained prior to feedback) with degenerative feedback is:

$$\text{Gain feedback} = \text{A}' = \frac{E_{output}}{E_{input}} = \frac{\text{A}}{1 - \beta A}$$

The action of positive and negative feedback can be seen in more detail from the above formula by noting that when β is negative the gain with feedback is less than without feedback, and when β is positive the gain is greater than without feedback. The feedback equation also shows that if the amplification and feedback are such that the value of $\text{A}\beta$ is equal to $+1$, then the gain will be infinite and the circuit will oscillate. This condition is extremely important in the construction of oscillators, since a stable oscillator is obtained by the application of positive feedback at the desired frequency.

In a previous section it was shown that the gain of a single stage of amplification is negative, that is, the voltage in the plate circuit is 180 degrees out-of-phase with the voltage in the grid circuit. Therefore, for any odd number of stages the output is out-of-phase with the input and can give negative feedback, while for an even number of stages the output is in-phase with the input and will give positive feedback.

Negative feedback reduces the amplification and also reduces the noise and distortion introduced within the feedback loop. The percentage of distortion is decreased, and the frequency response curve is made more flat. In general, the following desirable effects are obtained by the use of negative feedback:

(1) Greater stability. The circuit characteristics will remain constant for wide changes in tube characteristics and applied voltages.

(2) Reduction of harmonic distortion and intermodulation distortion.

(3) Reduction of phase distortion.

(4) Improvement in the frequency response characteristics.

(5) Reduction of noise.

(6) Modification of the input and output impedances.

In applying negative feedback to audio amplifiers, great care must be taken to apply the feedback in such a manner that phase shift does not cause it to become positive feedback at any frequency, because such positive feedback will tend to produce instability and increase distortion at these frequencies, and may even cause oscillation.

Transformer Coupling

In setting up audio systems it is often necessary to perform the functions of impedance transformation and phase inversion. Impedance transformation is required whenever it is necessary to couple between circuits of different impedances, such as the plate of an amplifier stage to a transmission line or other low-impedance load. Some type of circuit must be used which will match the two impedances properly. Phase inversion is required whenever it is necessary to couple from a single-ended to a push-pull circuit.

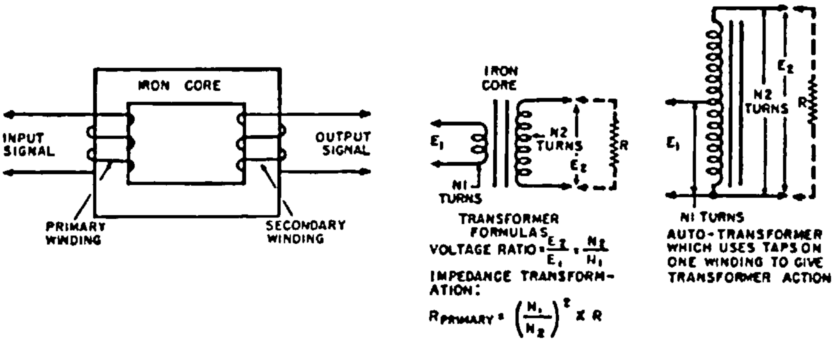


Fig. 5-12. Basic characteristics of transformers: (A) basic physical construction of a transformer; (B) electrical representation.

The simplest method of performing these functions is by the use of transformers, as shown in Fig. 5-12. A transformer consists of one or more windings placed on a single iron core, in such a manner that the electromagnetic field of the primary winding induces a signal in the secondary winding. If the two windings have different numbers of turns, then there will be an increase or a decrease in the voltage from one winding to the other, proportional to the ratio of the number of turns in each winding. When the secondary winding is loaded with a resistance, then this resistance appears in the primary circuit multiplied by the square of the turns ratio between the two windings. These basic characteristics of transformers are indicated by the formulas included in the diagram of Fig. 5-12.

For increased flexibility, the windings in a single transformer will often have taps brought out at various numbers of turns in order to give a wide variety of turns ratios. Different numbers of turns of the same winding can also be used as a transformer, as shown in Fig. 5-12 (B); such a transformer is called an autotransformer.

Transformer coupling is used mainly between impedances ranging from a few ohms up to about 50,000 ohms, and is most useful in applications such as:

(1) Input coupling from low-impedance sources to high impedance grid circuits.

(2) Interstage coupling from plate to grid circuits.

(3) Output coupling from plate circuits to low-impedance lines or loudspeaker voice coils.

(4) Impedance matching between low-impedance lines or from line to loudspeaker voice coil.

As well as numerous other applications which will become evident upon further consideration of various audio systems.

One of the other important characteristics of transformers (other than autotransformers) is that the various windings are isolated from each other, since there is no direct connection between them; therefore, transformers can be used for d.c. and ground isolation between circuits. In coupling from the plate circuits of amplifiers, transformers are used mainly with general-purpose triodes and pentode power amplifiers, since other types of tubes require too high an impedance.

Although they have a number of desirable features which make them quite convenient for use in audio circuits, transformers also have certain disadvantages which often make their use undesirable. One of the main disadvantages is their cost. A good-quality transformer will cost considerably more than a single amplifier stage. Thus, if a tube can be made to perform the same function, it is obviously less expensive to use a tube instead of a transformer. The frequency range of transformers is limited by the practical considerations of maximum obtainable primary inductance and minimum distributed capacitance across the windings, so that when extremely wide frequency bands are required transformers cannot always be used. The frequency response is very important when the feedback loop includes the transformer, since phase shift may cause the feedback to become positive at some frequency and cause instability. Care must also be taken that too large an unbalanced direct current does not flow through the windings, since such currents may saturate the iron core and cause distortion in the audio signal. One other factor which may make transformers undesirable for certain applications is their size, since the shielding or the size of the core may often cause high-quality transformers to be too large for applications where small size and compactness are required.

Considering the relative advantages and disadvantages of using transformers in audio circuits, it is evident that they should be used mainly in applications where their particular advantages are of value. A few typical examples are:

(1) To obtain high output voltage for limited supply voltage.

(2) To match to low-impedance lines (and voice coils), particularly where isolation is required between primary and secondary circuits.

(3) When a low d-c resistance is essential in the grid circuit of the following stage.

As well as other cases which may occur in the design and setup of specific audio systems.

Cathode Followers and Phase Inverters

In many applications where it is not convenient or practical to use transformers, the functions of impedance transformation and of phase inversion can be performed by vacuum tubes. By using current feedback of the type shown in Fig. 5-11, these functions can be performed by the *cathode-follower* amplifier.

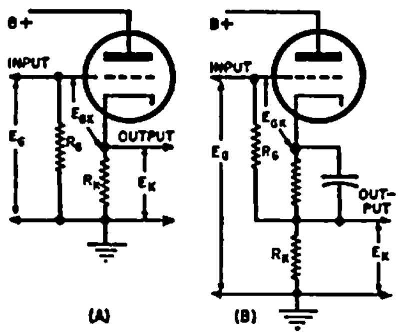


Fig. 5-13. A typical cathode-follower circuit. The output voltage is developed across the resistor in the cathode circuit.

The cathode follower is a single-stage negative-current feedback amplifier in which the output voltage is taken from across a load resistance in the cathode circuit. Typical circuits of cathode-follower amplifiers are shown in Fig. 5-13. In these circuits the signal plate current flows through the load resistor as in the conventional amplifier stage, but this resistor (R_k) is placed between cathode and ground instead of between plate and B+. Since the input grid voltage is applied between grid and ground, while the control signal is the grid-cathode voltage, the grid-cathode signal is the output voltage subtracted from the input voltage. The circuit shown in part (B) is useful in providing more flexibility in the choice of operating bias for the tube.

The result of this type of connection is a considerable amount of negative feedback over the one stage, and the output signal across the cathode resistor must always be less than the input signal between grid and ground, otherwise the tube would have to have an infinite amplification factor. The gain of the stage from the input grid-signal voltage to the output cathode-signal voltage is:

$$A' = \frac{E_k}{E_g} = \frac{\mu R_k}{(\mu+1)\ R_k+R_p} = \frac{1}{1+\frac{R_k+R_p}{\mu R_k}}$$

which is always less than one.

The cathode-follower amplifier connected in this manner has certain characteristics which are often very useful. The output impedance across the cathode resistor is very low because of the large amount of feedback, and is approximately equal to $1/g_m$ in parallel with R_k, which will almost always be well under 1,000 ohms. The low output impedance is one of the most important characteristics of the cathode-follower, since it can be used to give a transformation from a high to a low impedance with no loss in signal level, over a very wide frequency band, without the use of a transformer. This circuit is particularly useful in feeding a low-impedance line from a high-impedance amplifier circuit. In addition, the input impedance is higher than for the same tube used in a conventional amplifier stage. This high impedance is important in many cases where it is desirable to have the minimum amount of loading across some high-impedance source as, for example, in the preamplifier of a condenser microphone to obtain adequate bass response.

Because the cathode follower has so much negative feedback, it introduces very little amplitude distortion, and is capable of handling a high input voltage without overloading.

Consideration of the basic cathode-follower circuit in Fig. 5-13 shows another extremely important characteristic. Suppose a positive voltage is applied to the grid: then the plate current increases and the cathode becomes more positive. This means that the output taken

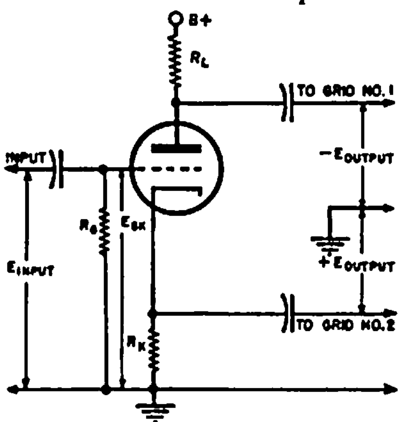


Fig. 5-14. Use of equal resistances in the cathode and plate circuits of a cathode-follower "phase-splitter" to give a push-pull output from a single-ended input signal.

across the cathode load is in phase with the input signal applied to the grid. However, in a conventional amplifier stage the output signal in the plate circuit is exactly opposite in phase with the grid input signal. Therefore, the cathode output signal in a cathode-follower amplifier is exactly opposite in phase with the signal in the plate circuit, and this factor can be used to construct a "phase splitter". The manner in which this can be done is shown in the circuit in Fig. 5-14. Equal resistances are placed in the cathode and in the plate circuit, one output is taken from the cathode circuit and the other from the plate circuit. This gives two equal signals opposite in phase, which can then be applied to the two grids of a push-pull amplifier.

Driver Amplifiers

The driver amplifier couples the output of the voltage-amplifier section to the input of the power amplifier. It must be able to supply enough voltage and power to the power-amplifier grid to drive it to full output without overloading. When a push-pull power amplifier is used, the driver section should also contain circuit arrangement to couple the output of a single-ended voltage amplifier to the push-pull grids of the power amplifier. Special techniques may also be used in the driver to minimize the effects of grid current in the power amplifier.

Because the amount of voltage required from the driver is determined by the power-amplifier requirements, different circuits may often be required for different power amplifier tubes. Triodes, as power amplifiers, in general are less sensitive and require higher grid-signal voltages than pentodes and beam-power tubes. Normally a general-purpose triode will furnish sufficient output voltage to drive most of the power tubes used in all except the very high power sound reproducing systems.

The circuits of several different driver amplifiers which are suitable for coupling the output of a single-ended voltage amplifier to a push-pull power amplifier are shown in Figs. 5-14, 5-15, and 5-16. The circuit of

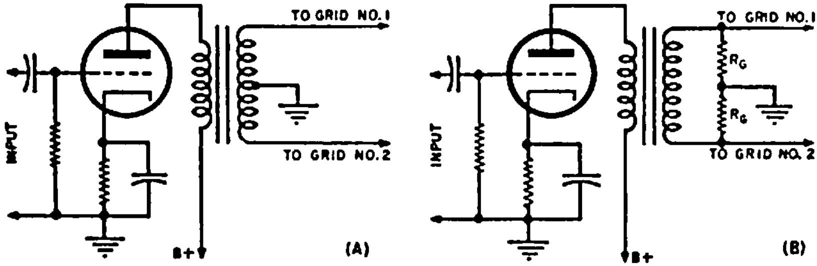


Fig. 5-15. Use of a transformer to give a push-pull output from a single-ended input signal: (A) by means of a center-tapped secondary winding; (B) by means of a resistive voltage divider with its center grounded.

Fig. 5-15 (A) shows the simplest and best known method of performing this function, i.e. by use of a transformer having a center-tapped secondary winding. Since this provides two equal voltages 180 degrees out-of-phase, perfectly satisfactory results are obtained with properly designed transformers. If the transformer secondary is not center-tapped, it can still be used for push-pull operation by connecting a resistance voltage divider with its center grounded, as shown in Fig. 5-15 (B). The center-tapped winding is preferable because it introduces low resistance into the power-amplifier grid circuit so that there is less distortion when grid current is drawn, and because it does not reflect any resistive load into the driver circuit on the primary side. However, because good transformers are expensive, they are not too widely used in most of the average-size sound reproducing systems.

For most applications, good results are obtained with the use of resistance-coupled amplifier stages. The simplest circuit which can be used for this purpose is the cathode-follower phase-splitter circuit shown in Fig. 5-14. With equal load resistors in the plate and cathode circuits two equal output voltages, opposite in phase, are obtained which are then applied to the two power amplifier grids. Since the gain to each side is approximately that of the cathode-follower with a maximum of 1, the total grid-to-grid output voltage gain of the phase inverter has a maximum value of 2. With this low gain, the tube does very little amplification, but takes the place of the transformer in coupling from the single-ended to the push-pull stages.

Another type of circuit which can be used for phase inversion is shown in Fig. 5-16. In this circuit the voltage amplifier ($V1_a$) drives one of the push-pull grids directly, while an additional amplifier stage ($V1_b$) is used to amplify a small part of this voltage with a 180-degree

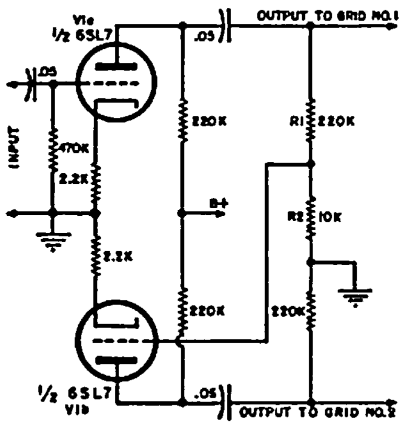


Fig. 5-16. Phase-inverter amplifier which gives a push-pull output from a single-ended input signal.

phase reversal to drive the second push-pull grid with the proper voltage and phase. The driving voltage for this phase-inverter amplifier tube is obtained from the voltage divider formed by the two resistors in the first push-pull grid (R_1, R_2). The resistance should be chosen so that:

$$A = \frac{R_1 + R_2}{R_2}$$

where A is the gain of the phase inverter tube. The exact value of the resistor (now shown as 10,000 ohms) must be selected very carefully for satisfactory balance of the push-pull amplifier. A disadvantage of this circuit is that as the tubes age, the gain of the phase inverter stage may change and cause unbalance in the push-pull amplifier. Cathode degeneration by use of an unbypassed cathode resistor can help stabilize the gain to reduce this effect, or the use of a variable instead of a fixed resistor for the 10,000-ohm unit permits periodic readjustment to maintain accurate balance.

Chapter 6

PRACTICAL AUDIO AMPLIFIER CIRCUITS

General

In any system for sound reproduction, amplifiers are needed in order to increase the electrical power from the pickup unit or transducer to a high enough level for operation of a loudspeaker or recording transducer. Microphones and pickups derive their energy from extremely low power sources – such as the sound pressure vibrations in the air, the modulation of the groove in a record, or the magnetization of an iron wire or an oxide coated tape. The energy which is available to generate an electrical signal is generally a fraction of a microwatt, therefore the resulting electrical voltage is extremely low. However, a considerable amount of audio-frequency power (on the order of 5 to 10 watts or more on occasions) is required to generate a sufficient amount of sound energy from the loudspeaker. Therefore the amplifier is required to raise the signal level from the low output of the pickup to the output required to drive the loudspeaker, both for amplification and to make up the loss in any mixing and frequency-response equalization systems which are used.

Functional Requirements of the Amplifier

The amplifier may be on a single chassis or it may be mounted as several units, but in general it will consist of four different sections:

(1) The preamplifier.
(2) The voltage amplifier.
(3) The driver section.
(4) The power amplifier.

These four sections perform different functions in the system, therefore different design and construction techniques must be used for each of them.

Preamplifier. The most important feature of the preamplifier is that it must amplify extremely low voltages without introducing any noise or other spurious signals into the audio signal which is to be amplified. Its function is to increase the signal power: either by delivering the same signal voltage at a lower impedance, or a higher voltage at the same impedance, or by a combination of both impedance transformation and voltage gain. Since the signal level in the preamplifier is the lowest in the entire system, the greatest care must be taken here to keep the possibilities of extraneous hum and noise pickup to the absolute minimum, and all preamplifier design and construction techniques are set up for this specific purpose. The preamplifier may also be used to provide equalization.

Voltage Amplifier. The function of the voltage amplifier section is to give voltage gain. The output signal from a preamplifier, a mixer, or an equalizer network is generally at somewhat higher than microphone level, therefore the introduction of noise or other extraneous signals into such a signal is not the major factor that it is in the preamplifier. However, this signal is at much too low a level to be used for generating sound or driving a recording transducer directly. It is therefore fed through a voltage amplifier, which raises the voltage to a level at which it can be used for such operations as further mixing, equalization, transmission over telephone lines, etc. The voltage amplifier is designed to have the best possible frequency and phase response characteristics, and since the voltage gain is usually fairly high, care must be taken to avoid oscillations.

Driver Section. The output of the voltage amplifier is coupled to the power amplifier by the driver section. The driver must supply sufficient voltage to the grid of the power amplifier to obtain full power output to the load without overloading. When a push-pull power amplifier is used, the driver section usually contains a phase inverter to couple the output of a single-ended voltage amplifier to the push-pull grids of the power amplifier. The most important consideration in the design of the driver section is that it must be able to supply enough power to the grids of the power amplifier to drive it to full output, and it may also contain features to minimize the effects of grid current in the power amplifier grids.

Power Amplifier. The audio-frequency electrical output of the entire system is supplied by the power amplifier. The most important feature of the power amplifier is that it must be able to supply as much power as required by the loudspeaker or the recording transducer to perform its function properly. Generally, the major harmonic distortions in the amplifier system arise in the power amplifier stage, and this must be designed to introduce the minimum amount of distortion into the electrical signal. Since the transducers into which the power amplifier operates are mostly electromechanical in nature, it should also preferably present a low electrical impedance to the load, in order to introduce the proper amount of damping into the mechanical system.

Requirements of the Preamplifier

The primary requirement of the amplifier is the amplification of low-level signals, therefore this is the main consideration in the design and construction techniques used for preamplifiers. The major efforts are aimed at achieving a unit which introduces the minimum amount of noise and vibration pickup while giving the required voltage gain or impedance transformation. Since the signal from the transducer is the lowest level in the entire reproducing channel, any noise which is picked up in this portion of the circuit will have the most serious effect upon the quality of reproduction, and will be most audible in the reproduced sound.

In general, the most critical applications of preamplifiers are in the amplification of signals originating in microphones, since these are almost always the lowest level signals in any audio system. The electrical power output of most microphones, whether high impedance or low impedance, is of the order of 0.01 microwatt, i.e., 10^{-8} watt or less. For example, a microphone with an output impedance of 250 ohms will deliver an electrical output in the neighborhood of 1 millivolt, while a high-impedance crystal microphone will deliver 0.01 to 0.1 volt peak output signal into a 1-megohm load resistance. Low-level signals are also obtained from the various magnetic phonograph pickups which have an output impedance of about 100 to several thousand ohms and deliver approximately 10 millivolts of electrical signal. When it is realized that a good sound reproducing system should have a signal-to-noise ratio of 50 to 60 db or better, the care that must be taken in the construction of preamplifiers becomes very obvious. Thus the noise pickup should be less than 1 microvolt at the input of a low-impedance microphone preamplifier, and less than 10 microvolts at the input of a preamplifier used for crystal microphones and magnetic phonograph pickups.

Noise problems almost always originate in the first stages of the preamplifier, where the signal level is lowest. The most serious types of noise problems are hum pickup, thermal noise in resistors, and vacuum tube noise. These can be kept to a minimum by the use of proper techniques of design and construction. However, even with the use of the best noise-reducing techniques, there are limits to the extent to which the noise in any amplifier can be reduced. Noise considerations impose very important practical limitations on how low a level of signal can be useful, and on signal-to-noise ratios at low levels. Transducers whose output level is too low cannot be used because too much noise will be introduced in the input stages of the preamplifier. Low-level signals from microphones and pickups should be amplified before mixing or attenuation, otherwise the signal level would be too low for satisfactory signal-to-noise ratio.

Another important consideration in the design of preamplifiers is that they must not overload at high signal levels. Since preamplifiers are designed for low-level inputs, they can overload if the input signal is too great. Therefore a high-level and a low-level signal can be amplified by the same preamplifier only if the high-level signal is attenuated so that both are at the same level.

The output impedances and signal level obtainable from the various types of pickup devices and transducers used in the reproduction of sound have been listed in reference form in Tables 4-I and 4-II. Consideration of the information contained in these tables shows the types of preamplifiers which should be used with the various units.

Noise in Preamplifiers

The most serious types of noises are: (1) a-c hum pickup, (2) thermal noise in resistors, and (3) vacuum tube noise. Although these cannot be completely eliminated, there are a number of design and construction techniques which will reduce these types of extraneous noises to a minimum. A more detailed consideration of the origin and effects of these types of noise will give a better indication of the importance of keeping them to a minimum, and will show what techniques should be used.

Alternating-Current Hum Pickup. This is caused by the presence of any 60-cps or 120-cps field and may be picked up either capacitively, inductively, or by direct conduction. When the input lead from the microphone or phonograph pickup (where the signal level is at its lowest) it at a high impedance-to-ground, 60-cps voltages may be picked up by this lead because of its electrostatic capacitance to some part of the circuit which is at a relatively high a-c potential. If the input lead is low-impedance, 60-cps voltages can be picked up by electromagnetic induction from any part of the circuit which carries relatively heavy alternating currents. Inductive pickup of this type can occur both in the signal leads and in any input transformer that may be used. If the heaters of the preamplifier tubes are operated from a.c., hum pickup can be caused by the alternating voltage drop in the heater, magnetic effect of the heater current on space current, and by temperature variations of the cathode. The first of these causes introduces a 60-cps hum component (although in some cases 120-cps component may be introduced), while the last two causes introduce a 120-cps hum component. It is also possible for 120-cps hum to be introduced into the signal through the power supply if there is any ripple present in the B+ supplying the tube, but this effect is not a major one since the remedy is merely to add more filtering to the power supply for the preamplifier.

Capacitive hum pickup in high-impedance input leads can be reduced by covering the signal lead with a grounded shield to decrease its capacitance to other parts of the circuit. Either a single-conductor or a double-conductor shielded lead may be used, depending upon whether the signal is balanced or unbalanced with respect to ground. When a shielded lead is used, high-impedance transducers (such as crystal microphones and phonograph pickups) can then be situated at some distance from the preamplifier: the main consideration governing the length of lead being the capacitance-to-ground that can be tolerated by the circuit to maintain good high-frequency response and signal level.

Inductive pickup in low-impedance input leads is kept to a minimum by keeping the signal lead and its return lead as close together as possible (i.e., they should be run alongside one another in the same cable), so that there is very little loop for induction pickup. In addition, the circuit impedance should not be too low, so that the induction currents will tend to be limited by the circuit impedance. Long leads, even at low impedance, should be shielded to minimize the possibility of capacitive hum pickup. When an input transformer is used, it must be one with good magnetic shielding and a humbucking type of winding; it should be placed as far away as possible from any other transformer or motor carrying a.c. and oriented in such direction that its hum pickup is at a minimum.

Hum pickup from heater to cathode can be kept to a minimum by taking the heater return from the tap of a potentiometer, connected across the heater terminals, either to ground or to an adjustable positive d-c voltage, and adjusting the potentiometer and the voltage to give minimum hum pickup. If the hum is not sufficiently reduced, the preamplifier heaters should be operated with d.c.

Thermal Noise in Resistors. This is caused by the random motion of free electrons. This electron motion causes small potentials to be developed across the resistor. These are called thermal voltages, and the noise associated with them increases with temperature, frequency range, and the size of the resistance. As an example of the effects of thermal noise, the rms thermal voltage developed across a 0.5-megohm resistor at room temperature for a frequency band of 10,000 cps is 9 microvolts. Ordinary carbon resistors also generate considerably more noise than the normal thermal noise when current is passed through them, due to fluctuations in the contact resistance between adjacent carbon granules. Since resistors of different makes often vary considerably, resistors having the minimum amount of noise should be selected as the grid and plate resistors of low-level stages.

Vacuum Tube Noise. This consists of noise which is generated inside the preamplifier tubes. The generation of hum voltages due to heater alternating currents has already been described, but there are also other types of noise which can originate in the tube. *Microphonics* are caused by variations in the spacings of the different elements inside the tube due to mechanical vibration and shocks. These can be kept to a minimum by using tubes which have been specifically designed to have a minimum of microphonics, and by mounting the first stages on soft rubber mountings which will absorb much of the vibration from the chassis to the tubes. Noise is also generated in the tube due to the fact that the current emitted from the cathode is not a perfectly smooth, uniform stream, but is emitted as a large number of discrete electron charges. The variations in current due to this effect will result in fluctuations in the plate current of the tube, and have the same effect as current variations due to noise in the signal circuit. Examples of tubes which have been designed specifically for a low amount of microphonics and noise are the type 1620 and 1603 pentodes, the 5879 miniature pentode, and the 12AY7 miniature dual triode.

Preamplifier Circuits

Basically the circuit of the preamplifier has to be one which provides a certain amount of gain or an impedance transformation, and which introduces very little noise into the signal in the process. The specific form which the circuit will take depends primarily upon the impedance of the transducer whose voltage is being amplified, and also upon its output voltage. The lowest voltages which are encountered in sound reproducing systems are generally obtained from low-impedance magnetic microphones and phonograph pickups. Preamplifiers are mainly used to amplify these signals to a voltage comparable to the output of the average radio tuner or crystal pickup, so that mixing or switching of signals can be done at this level.

Microphone Preamplifier. A schematic diagram of a typical preamplifier to be used with a magnetic microphone is shown in Fig. 6-1. An important feature of this circuit is that preamplifiers for low-impedance microphones invariably make use of an input transformer to match the low impedance of the microphone to the high impedance of the grid circuit and to obtain a voltage step-up in this process. For example, in matching a 50-ohm microphone to a 50,000-ohm grid circuit, the impedance ratio is 1,000:1, and a voltage step-up of about 30:1 is obtained. This increase in the level of the voltage applied to the grid is extremely important, because it results in a considerable improvement in signal-to-noise ratio over what would be obtained without the input transformer. However, the transformer itself, must be magnetically

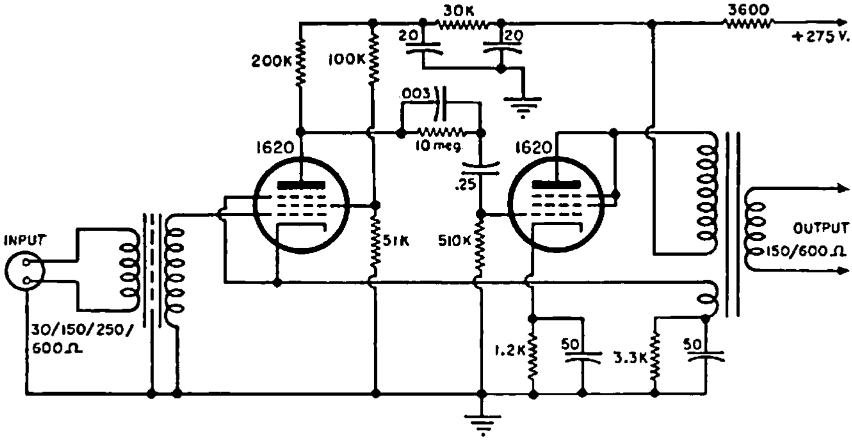


Fig. 6-1. Schematic circuit diagram of a high-quality magnetic microphone preamplifier for broadcast use.

shielded and constructed for minimum a-c hum pickup if the lowest noise level is to be attained.

The secondary voltage of the transformer is applied to the grid of the preamplifier tube, either with or without a terminating resistor, depending upon the design of the transformer. The first preamplifier tube is one of the low-level audio amplifier tubes which have been designed and especially selected for a low amount of microphonics and noise. Additional protection against noise due to vibration and shocks is obtained by mounting this tube on rubber, which absorbs mechanical vibrations that otherwise would be transmitted to the tube from the chassis.

The required gain and output impedance of a preamplifier are usually determined by the requirements of the complete sound reproducing system with which it is to be used. In broadcasting and many other applications it is desirable to have long leads running from the preamplifier to other components of the system, and these leads must, therefore, be run at low impedance, usually anywhere between 150 and 600 ohms. This generally means that a transformer should be used to couple the preamplifier output tube to the line, so that the line may have either one side or center-tap grounded. The output tube is, therefore, connected as a triode, since the plate impedance of a pentode is too high to permit a transformer to be used as the plate load.

This particular unit, which is a high-quality preamplifier suitable for broadcast use, has a multiple-impedance input transformer to match various microphone impedances to the first grid, and an output transformer to match a 150- or 600-ohm line. It has an absolute noise level equivalent to a −120 db input signal, a gain of 40 db, and a maximum output level of +18 dbm.

Phonograph Pickup Preamplifier. Magnetic phonograph pickups have a higher signal voltage than microphones, therefore they can be used with a preamplifier either with or without an input transformer. A preamplifier with an input transformer for magnetic pickups is basically the same as used for a magnetic microphone, such as the circuit of Fig. 6-1. Otherwise, without the input transformer, the output of the pickup may be applied directly to the grid of the first tube. However, there is one important difference between the preamplifier requirements for magnetic microphones and for magnetic phonograph pickups. This is due to the frequency-response equalization with which a recording is made. A magnetic pickup has an output voltage proportional to the velocity with which the stylus moves, and since the recording is made with constant amplitude below the crossover frequency, the output of a magnetic pickup will be deficient in low frequencies. This deficiency increases gradually from the crossover frequency until it is 15 db at 90 cps (with 500-cps crossover frequency), and therefore must be compensated by an equalizer. This equalization is usually incorporated into the preamplifier as a bass-boost section.

A typical preamplifier which has been designed for use with a magnetic phonograph pickup is shown in Figs. 6-2 and 6-3. The unit is the original equalizer-preamplifier which was designed for use with the General Electric variable-reluctance pickup. No input transformer is used in this unit, the pickup being terminated in the 6,800-ohm input resistance, and its signal applied directly to the grid of the first tube. This first stage is one-half of a 6SC7 dual triode with a voltage gain of about 26 to 30 db. The three resistors, 200,000 ohms, 27,000 ohms and

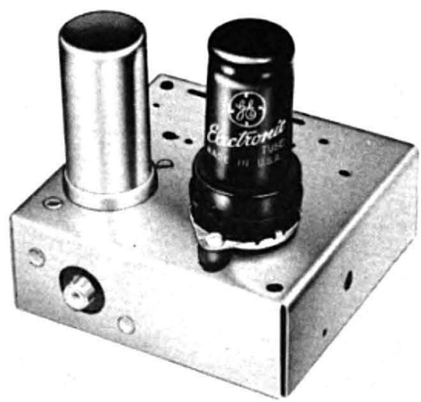


Fig. 6-2. Photograph of equalizer-preamplifier for use with G. E. variable-reluctance pickup.

Courtesy: *General Electric Co.*

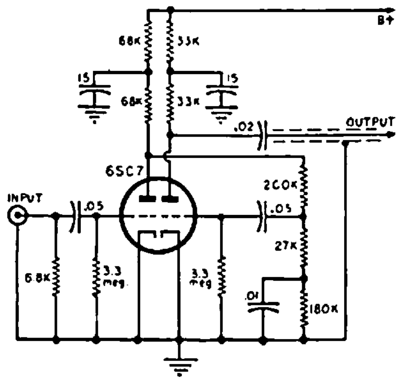


Fig. 6-3. Schematic circuit diagram of the equalizer preamplifier designed for use with the G. E. variable-reluctance pickup.

180,000 ohms, and the 0.01-μf capacitor form a bass-boost circuit which gives a total boost of approximately 14 db at the low frequencies, with an insertion loss of 19 db. Further amplification is provided by the second triode section of the tube, which also provides a fairly low output impedance so that a shielded lead several feet long may be used for connecting to the main amplifier or to whatever other units follow in the system. This preamplifier has an overall voltage gain of 35 to 40 db at 1,000 cps, an output impedance of approximately 25,000 ohms, and when used with the variable-reluctance pickup will deliver a maximum signal up to 1 volt into a high impedance.

The preamplifier for the magnetic pickup may also be designed for an input transformer, such as the circuit shown in Fig. 6-4. In this unit the pickup is matched through the input transformer to the grid of the 6J7 preamplifier pentode. Since the voltage delivered by a magnetic pickup is considerably higher than the signal from a microphone, the 6J7 can be used instead of the 1620 tube which is preferred for microphone preamplifiers. At the output of the plate circuit of this amplifier tube, coupling this tube to the grid of the second amplifier

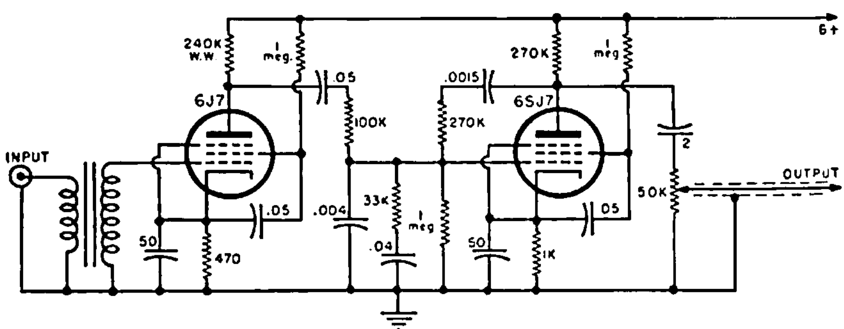


Fig. 6-4. Schematic circuit diagram of a preamplifier for magnetic pickup which uses an input transformer for best signal-to-noise ratio.

tube, is the frequency-response equalizer section which compensates for the recording frequency characteristic. This particular unit is designed to have a high-impedance output, therefore the second tube is also operated as a pentode. The plate circuit of this tube is operated into a 50,000 ohm potentiometer which acts as the volume control. The output of either of the above preamplifiers can be applied directly to the input of the main voltage amplifier.

Voltage Amplifier Circuits

The use of preamplifiers with the proper gain brings the signal level of magnetic microphones and phonograph pickups to the same level as the output of the average radio tuner or crystal pickup (about 1 to 2 volts into a high impedance). Mixing or switching of the various signals for reproduction is generally done at this level, and the resulting signal must then be further amplified by a fairly high-gain voltage amplifier. This additional amplification serves two functions: (1) it increases the signal to that voltage necessary to obtain full power output from the power amplifier through the driver amplifier, or for further mixing, equalization, transmission over telephone lines, etc.; and (2) makes up for any insertion loss introduced by the use of any mixing, equalization, or transmission units in the sound reproducing system.

The reproducing system may be set up in a number of different ways, according to the specific requirements of the individual application. Mixing or equalization may take place ahead of the voltage amplifier directly after the preamplifier (or even in the preamplifier unit), or after one section of the voltage amplifier. In either case, the input signal to the voltage amplifier is at a higher level than the input signal to the preamplifier. In the voltage amplifier, the major requirement is high gain without distortion or instability.

Essentially, the voltage amplifier consists of a number of amplifier stages whose total gain and output voltage meet the systems requirements. The individual amplifier stages may be either triodes or pentodes, the choice depending upon both the requirements of the circuit and the individual preference of the designer (since there is still considerable discussion concerning the relative merits of triodes and pentodes). The procedure followed in the design of the amplifier is to start from the knowledge of the input and output voltages, the required input impedance, and the impedance of the load which the output of the amplifier sees. Then the various amplifier stages and impedance-matching circuits are designed for the required voltage gain and impedance.

Example of Design. As the simplest example of a voltage amplifier design consider the requirements of a voltage amplifier to be used with a standard type of crystal phonograph pickup or a radio tuner. The

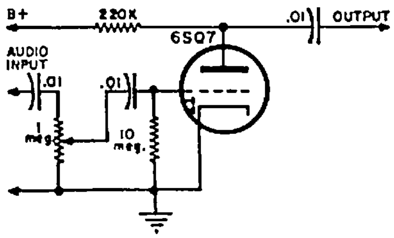


Fig. 6-5. Schematic diagram of the typical high-gain triode amplifier used as the voltage amplifier in many radio receivers.

input voltage to this amplifier will be in the neighborhood of 1 volt, and the output voltage should be at least 10 to 15 volts. Allowing for a reserve amplification of two or three times this amount so that the volume control will not have to be set full up, the required amplification has to be of the order of 30 to 50 times. The volume control is generally placed at the input of this amplifier to prevent overloading with high-level signals. This type of amplifier is used in most radio receivers, therefore a number of important points are illustrated in considering its design features.

The voltage amplifier circuit used in most radio receivers usually consists of a high-gain triode, such as the 6SQ7, in a circuit similar to that shown in Fig. 6-5. The amplifier meets the requirements of gain, distortion, and output voltage, but cannot have good response at the higher audio frequencies. The reason for this can be readily understood by considering the Miller effect of the tube, especially when the volume control is set near the middle of its range. Specifically, for a tube gain of 40 the input capacitance of the tube is at least 70 $\mu\mu$f or higher; therefore, with a 1.0-megohm volume control set halfway up, the response can be as much as 7 to 10 db down at 10,000 cps. The manufacturers of commercial radio receivers may consider this frequency response satisfactory for a-m reception, but it is certainly not acceptable for high-quality sound reproduction.

A good frequency response in this voltage amplifier can be achieved by using a pentode instead of the high-gain triode stage. A typical

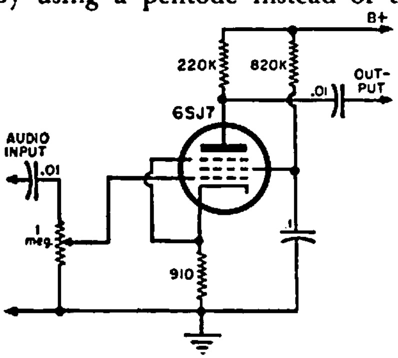


Fig. 6-6. Schematic diagram of a typical pentode amplifier stage which can be used in place of the triode in Fig. 6-5 to provide an improved frequency response.

pentode voltage amplifier which can be used for this purpose is shown in Fig. 6-6. It is a standard pentode amplifier, designed from the tube plate-current characteristics as described in Chapter 5, and has quite satisfactory gain, output voltage, distortion, and frequency response characteristics.

In many applications, a more elaborate voltage amplifier than this is required. Often there may be additional gain and impedance matching requirements which must be met. The schematic of such an amplifier, which illustrates the methods of design to meet specific gain and impedance requirements, is shown in Fig. 6-7. This particular ampli-

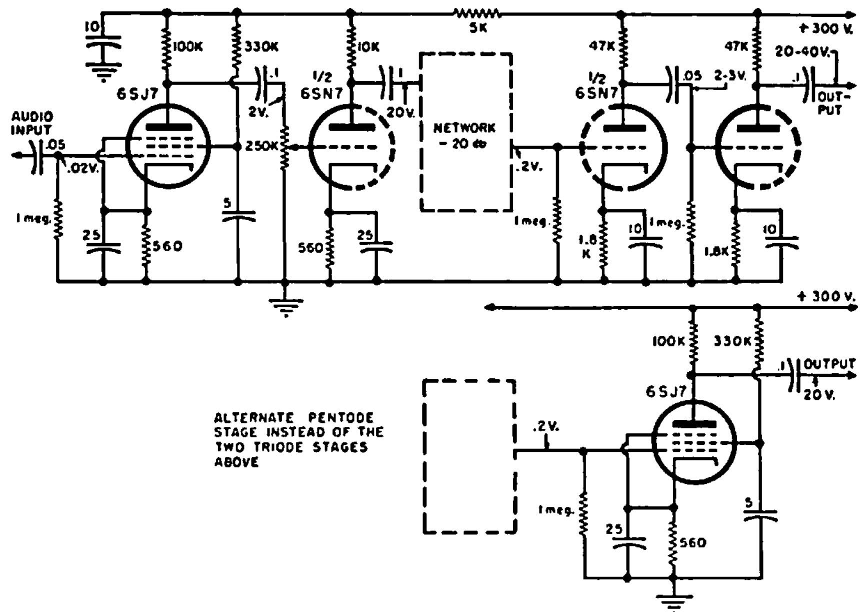


Fig. 6-7. Schematic circuit diagram of an audio amplifier which operates from a low-level, high-impedance input signal and includes a 5,000-ohm output impedance to a network having a 20 db insertion loss.

fier is designed to give full voltage output to the driver with an input of 0.02 volt at high impedance, and includes sufficient gain to compensate for a 20 db insertion loss network (such as a tone control or mixer circuit), which is fed from a 5,000-ohm impedance.

The first stage is a pentode, which has an amplification of 100 and whose output feeds into a 0.25-megohm volume control. Because of the Miller effect, the tube after the volume control is a low-gain triode. With the 6J5 (or one-half 6SN7) and the circuit constants as shown, the input capacitance of the tube is about 45 $\mu\mu$f, which does not greatly affect the frequency response at any setting of the volume control. From the equivalent circuit, since the plate resistance of the tube is about 7,000 ohms, the source impedance which is presented to the network is about 5,000 ohms through the 1-μf coupling capacitor. The output of this network can then be amplified again either by a two-stage triode amplifier or by a single pentode, as shown in the diagram. The two triodes will have up to 6 db more gain, but either arrangement will have enough gain and supply adequate voltage to the driver. The approximate signal voltage levels at the various points in the circuit are indicated on the schematic diagram.

Transformer-Coupled Voltage Amplifiers. Some reproducing systems may require voltage amplifiers which are coupled through input and output transformers. The schematic in Fig. 6-8 shows the circuit of an amplifier of this type which has extremely good frequency response, noise and distortion characteristics, and which has been widely used for broadcast applications. This particular unit is a two-stage, push-pull amplifier with a fixed gain of 50 db with various input and output impedances available. The amplifier stages are designed accord-

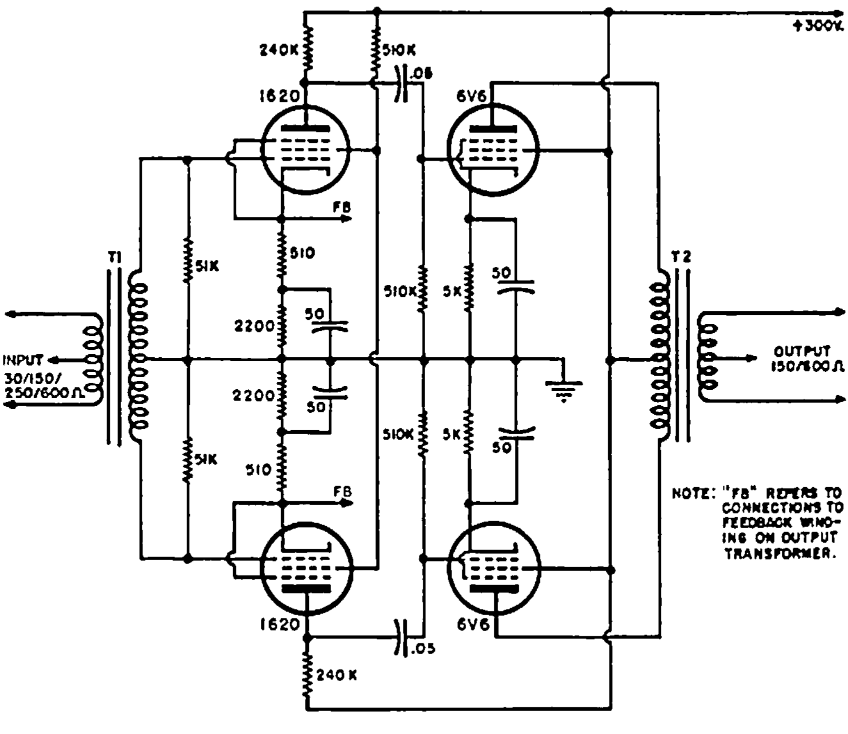


Fig. 6-8. Circuit of a push-pull voltage amplifier, with input and output transformers, which has extremely good characteristics suitable for broadcast applications.

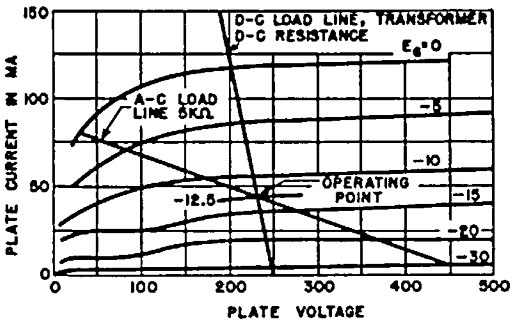


Fig. 6-9. Curves showing different load lines for d-c and a-c signal when using a transformer in the plate circuit, and method for determining operating conditions.

ing to the principles described in Chapter 5, and illustrate an important point in the design of transformer-coupled stages.

It should be noted that the impedance of a transformer is different for direct-current and for alternating-current signals, therefore the static operating point is determined by the d-c resistance of the winding, while the signal gain is determined by the a-c impedance reflected into the transformer primary. This is illustrated in the set of curves of Fig. 6-9. The amplifier shown in Fig. 6-8 has a frequency response of ± 1 db from 30-15,000 cycles, and has a 1-watt output at less than 0.5 percent distortion and up to 8 watts with slightly higher distortion. An amplifier with these characteristics can be extremely useful in setting up a sound reproducing system.

The Power Amplifier

The power amplifier is the final stage of amplification in the electronic channel of the recording or reproducing system; its function is to supply driving power to the output electromechanical transducer. The transducer, which converts this electrical energy into the appropriate mechanical motion, may be either a recording head or a loudspeaker. The output signal of the power amplifier must be free from distortion and must satisfy power and output impedance requirements.

Since electromagnetically operated transducers, such as recording heads and loudspeakers, are generally very low impedance devices, the power amplifier must feed into a load impedance on the order of 2 to 15 ohms. The output impedance of the amplifier appears in the electromechanical equivalent circuit of the loudspeaker and affects the transient response, therefore it should be less than one-half the load impedance for best mechanical transient response.

The output power requirements for sound reproduction are determined by the specific application, particularly upon the size of the room in which the sound is being reproduced. The power required for rooms of different sizes can be determined from the curves of Fig. 3-10, and from the discussion involving power requirements in Chapter 3. It has been shown that in an average-sized room in the home, an amplifier and loudspeaker capable of handling 8 to 10 watts of electrical power will have an adequate reserve.

The power amplifier consists of a single amplifier stage using a tube or tubes that will supply the necessary output power, and a transformer to match the relatively high plate impedance of the tube to the low impedance of the load. However, it is not simple to obtain the required power from the tubes and pass it through the output transformer to the load, while still maintaining the required distortion and frequency characteristics. For this reason, most of the difficulties in the reproducer circuits center about the power amplifier, and a wide variety of solutions has been attempted.

For a number of reasons, high-quality power amplifiers should be push-pull rather than single-ended:

(1) Push-pull amplifiers have less distortion, because even-harmonic distortions are cancelled leaving only the odd-harmonic distortion.

(2) Since the effects of the plate currents of the two tubes cancel one another in the transformer core, there is no d-c saturation of a well balanced output transformer, and the low frequency response is better.

(3) The effects of power supply hum are greatly reduced, since this hum is cancelled out in the transformer.

(4) The push-pull stage is less likely to cause motorboating in the amplifier.

These advantages are so important that a push-pull circuit using two smaller tubes is definitely preferable to a single larger tube capable of delivering the same total power output.

There is still considerable question as to whether it is preferable to use triodes or beam-power tetrodes in audio power amplifiers. It is generally agreed, however, that best results are obtained from push-pull amplifiers with overall negative feedback from output to input, including the output transformer in the feedback loop. The use of negative feedback in this application has the following advantages:

(1) The linearity of the output/input amplitude response curve is considerably improved, resulting in a decrease in harmonic distortion and intermodulation.

(2) The frequency response is improved, becoming flatter over a wider range of frequencies.

(3) The output impedance is reduced, thus improving the transient response of the loudspeaker.

(4) The effects of changes in tube characteristics, of random changes of the parameters of the amplifier, and of power-supply voltage changes are reduced.

Various circuits in use at the present time make use of triodes with and without negative feedback, and of beam-power tetrodes generally with feedback. Triode amplifiers have an advantage in that their distortion components contain a smaller distribution among the higher order harmonics than beam-power tetrodes, therefore they have a less unpleasant type of intermodulation distortion. However, beam-power tubes with feedback give better results than triodes without feedback, although perhaps not quite as good as triodes with feedback, and in general give better power efficiency and require lower drive voltages.

A considerable amount of engineering effort at the present time is being put into the development of new types of amplifiers and circuits which have superior performance characteristics. The main features of most of these circuits are improvements in the power amplifier and driver circuit, and in the method of application of negative feedback, based upon the fundamental principles which have been described in the preceding sections of Chapters 5 and 6.

The choice of tubes for the specific amplifier being designed depends upon the power required. Basic information concerning the operation of the various tubes – such as power output, plate load impedance, grid input, grid bias voltage, plate voltage, etc., are given in the receiving tube handbooks and can be used as a guide in the selection of tubes. However, the final design of the power stage must be done with the use of the plate-current characteristics.

Design of Power Amplifiers

In designing the push-pull amplifier from the plate-current characteristic curves which are given for the tubes, a *composite* set of curves must be constructed and used instead of those given for the single tube. The reason this is necessary is that the steady-state current in each tube has no effect on the audio-output signal because of the transformer coupling, and it is only the dynamic difference in the plate currents of the two tubes which appears in the secondary winding of the transformer. The composite curves are constructed essentially by subtracting the currents through the two tubes to approximate the effect of the transformer. This is done by placing the plate voltage-current curves of the individual tubes back-to-back, with the common operating voltage superimposed, and then averaging the plate current for grid-potential curves corresponding to the same applied signal, as shown in Fig. 6-10.

The precise manner in which the composite curves are constructed may best be understood by a more detailed study of Fig. 6-10, which

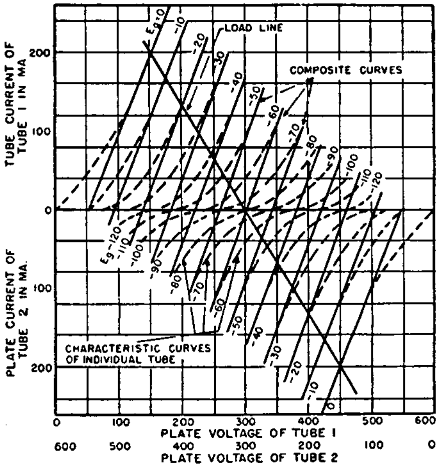


Fig. 6-10. Construction of the composite plate-current characteristics used in designing a push-pull amplifier. (These particular curves have been drawn for push-pull 2A3's.)

illustrates this procedure for a push-pull 2A3 amplifier. Two sets of curves are redrawn from a tube handbook, and each set of curves is taken to represent one of the tubes. Assuming a plate voltage of 300 volts, place the two sets of curves back-to-back with the 300-volt points coinciding as shown. Then, assuming some value of grid bias voltage (for example, –60 volts as shown), draw a line which represents the difference in the currents of the two tubes. Next, draw similar plate-current-difference lines for the various other values of grid voltage. (It should be noted that the curve for –70 volts for one tube is matched with –50 volts for the other tube, since the signal voltages on the two grids are opposite in phase, and the voltage on one increases while the voltage on the other decreases.) These lines will be practically straight, and represent the current-voltage curves taking into account the transformer, and the push-pull method of operation of the circuit.

The load line is then drawn over the composite characteristics in the same manner as for an ordinary set of tube characteristics. The load line in Fig. 6-10 intersects the zero-curent axis at 300 volts (which was initially selected by superimposing the two sets of curves at this voltage), which is the quiescent or zero-signal operating point. When extended, the load line intersects the zero-voltage axis at 400 milliamperes, therefore it represents a resistance of 750 ohms. Multiplying by 4 gives the total plate-to-plate load resistance, which, in this case is 3,000 ohms. When signals are applied to the grids, the values of the plate currents of the tubes lie along this line. Desirable operating conditions for the two 2A3 tubes in a push-pull amplifier are therefore: plate voltage, 300 volts; grid bias voltage, –60 volts; and load resistance (plate-to-plate), 3,000 ohms. However, the composite curves represent the signal currents through the plate load, and not the actual tube currents. Each tube will still draw 40 ma of plate current for zero signal. The total plate current for each value of grid voltage is found by adding the two plate currents, instead of subtracting them as for the composite curves, and the average plate current is found by averaging the sum of the two plate currents for a complete signal cycle.

Power Amplifier Circuits

This method of graphical construction for the push-pull power amplifier is the basis for the data given in the handbooks, and whenever such information is given it can be assumed to give the same values as would be obtained by an independent calculation of this type. This forms the basis of the various types of push-pull amplifiers which are in general use at the present time. The schematic diagram of a basic push-pull power-amplifier circuit based upon these design principles is shown in Fig. 6-11. The driver stage is a typical phase-inverter amplifier circuit which has already been described in Chapter 5. The input signal from the voltage amplifier is amplified by the first triode and drives one of the push-pull tubes, while the second triode section amplifies a small part of this drive voltage with a 180-degree phase reversal to drive the second push-pull grid with the proper voltage and phase.

The power amplifier consists of two 2A3's in the circuit designed from the curves in Fig. 6-10, with +300 volts on the plates, –60 volts grid bias, and a 3,000 ohm plate-to-plate output transformer matched

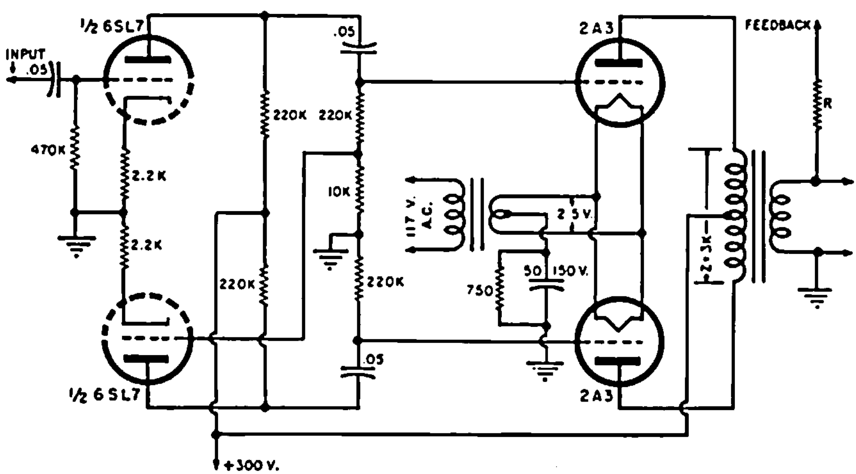


Fig. 6-11. Circuit diagram of the basic push-pull amplifier designed from the composite curves of Fig. 6-10.

to the loudspeaker impedance. (The feedback connection which is indicated will go to an appropriate point in the voltage amplifier. The value of *R* is determined by the amount of feedback required.) The grid-drive signal voltage required by each 2A3 for full output is approximately 40 volts rms, and is easily obtained from the 6SL7 driver tubes. The amplifier, as shown, is capable of delivering 10 watts into the loudspeaker, and will give quite good results in the average home sound reproducing system. This circuit is shown using triodes, but can also be designed for pentodes or beam-power tetrodes such as the 6L6 and 6V6.

Another power amplifier circuit which has recently been designed in England and has achieved wide popularity because of its excellent performance is the *Williamson amplifier*. The schematic circuit diagram of this amplifier is shown in Fig. 6-12. The input signal from the voltage

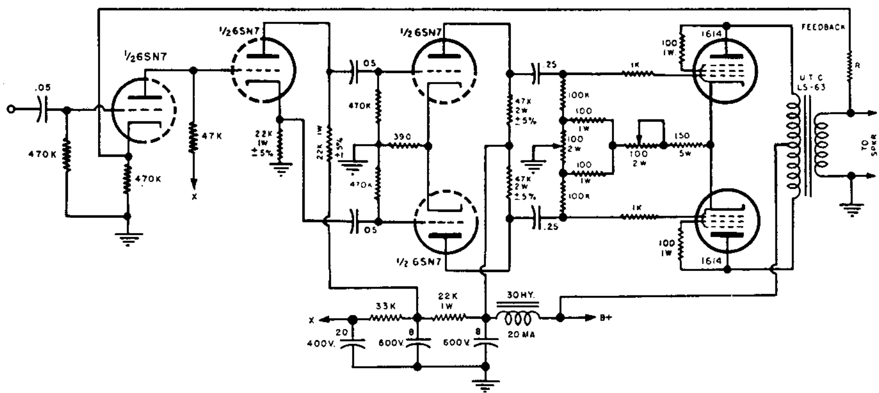


Fig. 6-12. Modified schematic of the Williamson power amplifier circuit.

amplifier is applied (through direct coupling) to a cathode-follower phase inverter, which supplies the two equal signals, opposite in phase, required by the push-pull amplifier. These signals, however, do not drive the power amplifier directly, but are instead amplified in a push-pull driver stage that supplies the signal voltage for the power amplifier grids. The power tubes are connected as triodes, and feedback is taken from the secondary of the output transformer to the cathode of the first voltage amplifier. Other common output tubes for this circuit besides the ones shown are the 807, 5881, and the British KT-66. The value of the feedback resistor *R* is usually made equal to 1,200 times the square root of the voice-coil impedance.

In this circuit, the output transformer is extremely important, and must very closely meet the original specifications for satisfactory performance. Too low a primary inductance in the transformer tends to produce instability at the low frequencies, while variations in the leakage inductance and capacitances between the windings will affect the high frequency response. At the present time a number of manufacturers are producing transformers for use with this circuit. It is also possible to purchase commercial kits which contain all the components necessary for construction of this amplifier. Commercially constructed amplifiers based upon this design are also available. An indication of the performance of this circuit may be obtained from the following typical specifications: frequency response, ± 0.1 db, 20-20,000 cps; frequency response, ± 2 db, 5-100,000 cps; harmonic distortion, less than 0.1 percent at 10 watts output at mid-frequencies; intermodulation, less than 0.5 percent at 10 watts output. These characteristics are extremely good, and are in fact considerably better than those considered to be the minimum requirements for good reproduction.

A recent modification of the above circuit involves the use of an output transformer with a tapped primary winding (taps at about 20 percent of the total primary turns). The screen grids of the power output tubes are connected to the taps. As a result, partial triode and partial pentode operation occurs. With this arrangement (often referred to as "ultra-linear"), an increase in power output results which is accompanied by a fidelity characteristic which is even somewhat better than the figures given above.

Chapter 7

A-F NETWORKS AND CORRECTIVE CIRCUITS

General

The most widely accepted criterion of performance in the design and setup of sound reproducing systems is that the reproduction should sound exactly the same as the original program material. However, this does not mean that the signal at all points in the system must correspond exactly with the original sound — *it means only that the sound reaching the listener's ear from the loudspeaker should reproduce accurately the sound reaching the microphone from the original source.* The proper application of this principle has caused considerable confusion in the field of sound reproduction, and its meaning should be clearly understood by anyone who is designing or setting up any sound reproducing system.

There is no necessity for making the signal at all points in the system reproduce exactly the original sound, provided that any alterations which are made in the signal are corrected before they reach the ear from the loudspeaker. In fact, the limitations of practical recording, transmission, and reproduction systems make it almost imperative that certain changes be made in the signal in order to obtain the best quality of reproduction. These limitations are related primarily to questions of relative sound level, noise level, and dynamic range. The actual dynamic range of orchestral music is approximately 75 db, which cannot ordinarily be reproduced by modern equipment, especially when phonograph records are included in the system. The major factor which prevents the reproducing system from attaining the required dynamic range is the inherent noise level of the system, which is considered to be good if it is more than –60 db below full output, while –50 db is considered acceptable. Therefore, as might be expected, most of the changes which are made in the signal are intended to reduce the noise level.

This chapter will discuss the various methods which can be used to reduce the noise level and increase the dynamic range of the reproduced sound, and to reduce the effects of differences in level between the reproduced and the original sound. These methods include the use of tone controls and equalizers, loudness controls, volume compressors and expanders, and various types of noise suppressors. The fundamental principles of operation will be described, with a discussion of the important factors which must be kept in mind for their proper use and to prevent their misuse, and a number of basic practical circuits which can be included in the reproducing system.

High-Frequency Pre-Emphasis for Noise Reduction

Probably the most obvious and objectionable form of distortion introduced in the reproduction of sound is noise, which can be extremely disagreeable to the listener. The highest noise levels are generally found in the reproduction of sound from phonograph records due to the record scratch introduced by the surface of the disc, and in the reproduction of sound by radio transmission due to static noise in the reception of weak signals.

The nature of the noise which is introduced in sound reproduction is such that it is not related to frequency in quite the same way that the ear perceives speech and music as a function of frequency. Sound is heard by the ear in a logarithmic manner, with respect to frequency as well as to volume, as illustrated by the use of the octave in musical scales and notation to denote a frequency range of 2:1 regardless of the absolute frequency. The noise, on the other hand, is generally a direct function of frequency. Therefore, since an octave at higher frequencies covers a greater absolute frequency range than an octave at lower frequencies, the effects of noise are relatively more important at high frequencies than at low frequencies.

This property of noise is realized instinctively by those people who listen to their phonograph records with the tone controls set to decrease the high frequency response and thereby reduce the record scratch. This method of reducing the record noise is not a desirable one, since the higher frequencies are lost from the reproduced sound. However, to these listeners the effects of the noise are more objectionable than the loss of high frequencies.

Because of the different frequency characteristics of the noise and of the reproduced sound, it is possible to reduce the reproduced noise level without reducing the high frequency range of the system, by the use of *high-frequency pre-emphasis*. The relative frequency distribution of sound energy in orchestral music is shown in curve *A* of Fig. 7-1. This curve shows that there is considerably less sound energy at the higher frequencies than at the lower. Curve *B* shows the sound spectrum of a random noise plotted to the same frequency scale (and at an arbitrary 0 db level), and indicates its relatively greater effect at the higher frequencies. It must be noted that such noise is generally introduced into the signal *after* the sound has been transmitted or recorded.

Therefore, the most basic and simplest method of reducing the reproduced noise level is to increase the amplitude of the high frequencies in the channel before recording (that is, before the introduction of the noise), and then to decrease the high frequency level by the corresponding amount in playback (which is after the introduction of the noise). The net effect is to decrease the noise level by the amount the high frequencies have been pre-emphasized.

This system of pre-emphasis is used in f-m broadcasting and accounts for a considerable amount of the noise superiority of f.m. over a.m. If pre-emphasis were used in a-m broadcasting, the received signal would contain a much lower noise component than it does with the present method. The technique of high frequency pre-emphasis is also used in disc and tape recording to reduce the effects of playback noise.

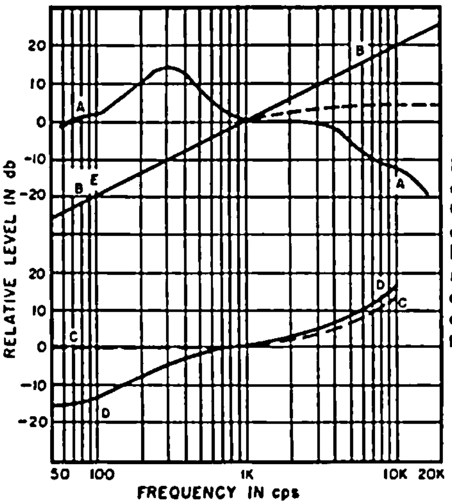


Fig. 7-1. Use of high-frequency pre-emphasis to reduce noise in reproduction. Curve *A* is the most probable distribution of frequency components in music at high levels; *B* is the frequency spectrum of random noise; *C* is the f-m pre-emphasis characteristic; *D* is NAB recording characteristic; and *E* is reproduced noise when the proper de-emphasis characteristic is used in playback.

The frequency response curves which are used for pre-emphasis in recording and f-m broadcasting are also shown in Fig. 7-1. The frequency response curve used in f-m broadcasting, shown in curve *C*, is flat at the low frequencies up to 1,000 cps and rises to about 14 db boost at 10 kc. This curve is said to be a 75 *microsecond* pre-emphasis characteristic, since this is the response which would result when the voltage is measured across the inductance in a series RL circuit with a 75 microsecond time constant, driven by a constant-current signal generator. The standard NAB recording characteristic is shown as curve *D*. This curve represents a constant-amplitude response at the low frequencies with a 500 cps crossover frequency, and a high-frequency pre-emphasis slightly more than that used for f-m broadcasting. (It should be noted that the NAB recording characteristic is not the only one in current use, and that many record producers use different amounts of high frequency pre-emphasis and different crossover frequencies.) In reproduction, the response should be the exact inverse of these curves to give a flat overall frequency response. The effective decrease in reproduced noise level by use of this pre-emphasis technique is shown by curve *E* in the figure.

The frequency response curves required for pre-emphasis and de-emphasis are obtained by the use of RC, RL, and RLC circuits. Such circuits are called *equalizer networks,* and are widely used not only to obtain the proper recording and playback frequency response, but where-ever else it is necessary or desirable to change the frequency response characteristic in any audio system. For example, they may also be used to correct for different crossover frequencies between recording and playback, and as tone controls to adjust the overall channel frequency response to correct any defects or to suit the hearing preferences of the individual listener.

Equalizer and Tone Control Circuits

An equalizer circuit is any network whose response varies more or less gradually in some desired manner over a given frequency range. Therefore, if a signal containing components of different frequencies is passed through the network, the relative amplitudes of the different components will have been altered in the desired manner when the signal is delivered to the load circuit. All equalizer circuits depend for their operation upon the fact that the reactances of capacitors and inductors change with frequency, while a resistance is fairly constant, and independent of frequency.

Basic RL and RC Circuits. The basic RL and RC equalizer circuits are shown in Fig. 7-2. In the simple series attenuator shown in (A) where the two impedances are pure resistances, the output level does not change with frequency because the impedances do not change with frequency. However, when either of the resistances is replaced by an impedance which changes with frequency, the output level will change with frequency accordingly. The response increases at high frequencies when either the series impedance is a capacitor or the shunt impedance is an inductor, as shown in (B). Therefore, this is a treble boost or bass attenuation network. The response decreases at high frequencies when either the series impedance is an inductor or the shunt impedance is a capacitor, as shown in (C). Therefore, this is a treble attenuation or bass boost network. Combinations of resistance, inductance, and capacitance in various arrangements (for example, in series-resonant and parallel-resonant networks) can also be used to give a different slope to the attenuation curve, or to put a peak or a dip at a specific frequency in the response.

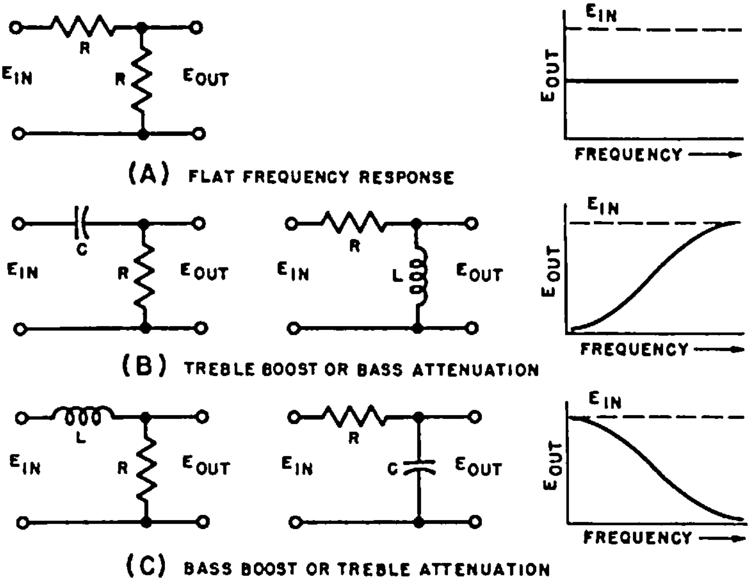
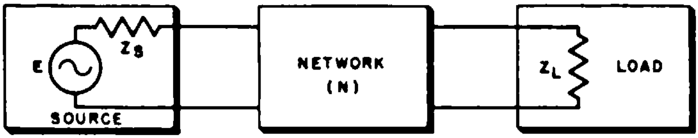


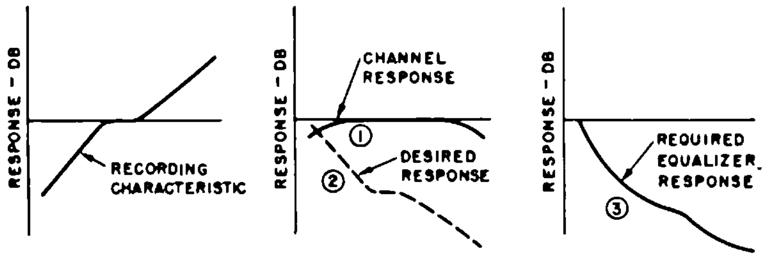
Fig. 7-2. The basic RL and RC equalizer circuits.

Equalizer networks which consist entirely of resistances, inductances, and capacitances as circuit elements, and do not contain any amplifiers or sources of voltage, are *passive* circuits. Most of the commonly used equalizers are passive circuits of this type. All such equalizers operate on the principle of attenuating certain frequencies to give their frequency response, since they obviously cannot deliver a greater voltage than is applied to their input. Therefore, for example, an equalizer which gives a 15 db treble boost actually attenuates all frequencies but the high frequencies by 15 db, and leaves the high frequency level unchanged.

Use of Equalizer. A typical equalizer or tone control network would be used in an audio system approximately as shown in the diagram of Fig. 7-3. This schematic represents the insertion of the network at any point in the audio channel, where the network represented by N operates between a signal source (of output impedance Z_s) and a load circuit of impedance Z_L). This network is to be used to compensate for the recording characteristic shown in part (B) of the figure. Then, if the audio circuit without the network has the frequency response characteristic *1* in Fig. 7-3 (B), and the desired response curve is *2*, the response of the equalizer should be the difference between the two curves as represented by *3*. Since the equalizer is a passive network and can only attenuate the unboosted frequencies, the response will be an attenuation curve as shown in the graph. The network may have an appreciable insertion loss, therefore it must be connected into the circuit at a point where the



(A) GENERALIZED DIAGRAM SHOWING THE CONNECTION OF A NETWORK IN A CIRCUIT.



(B) DETERMINATION OF EQUALIZER RESPONSE CHARACTERISTICS FROM DESIRED RESPONSE AND CHANNEL RESPONSE.

Fig. 7-3. Method of using equalizer networks and determination of frequency response requirements.

loss can be handled properly. The signal level should be sufficiently high that noise will not be a problem after attenuation by the network, but not so high that the tubes which are employed overload.

High- and Low-Impedance Networks. The most widely useful equalizer circuits for general home reproduction applications consist primarily of RC rather than RL networks, since inductances are generally too expensive and bulky for home use. The most general RC frequency correction circuit, which can be used as a basis for the practical design of all types of equalizers with a constant voltage input, is shown in Fig. 7-4. A number of useful equalizer and tone control circuits based upon this general circuit will be described in the next section, together with typical frequency response curves and formulas for design in order to

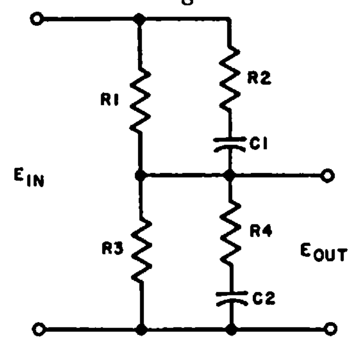


Fig. 7-4. General RC equalizer circuit which can be used to obtain a wide range of frequency response characteristics by proper choice of component values.

give response characteristics as might be needed for individual applications. Circuits very similar to these are used wherever equalization is required in the majority of sound reproduction systems.

These circuits are high-impedance networks and should have a constant voltage input, therefore the method of their connection into the playback circuit is somewhat critical. For best results, they should be isolated by a triode amplifier stage (such as the 6J5, ½6SN7, ½6SL7, ½12AT7, etc.) and should feed directly into the high-impedance grid of the following stage. The method of isolating the tone control network by a triode amplifier stage is shown in the circuit of Fig. 7-5.

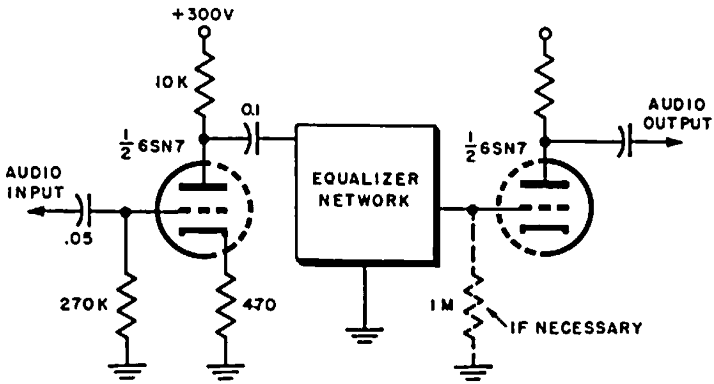


Fig. 7-5. Typical circuit for connecting a high-impedance equalizer into an amplifier.

In some applications (particularly in low-impedance circuits) it is desirable to have equalizer networks which present relatively constant impedance at their input and output circuits over the entire frequency band. Such networks are known as *constant-impedance* networks and, when connected into a circuit as indicated in Fig. 7-3, must be designed to match the impedance of the circuit, with an input impedance equal so the impedance of the signal source Z_s, and an output impedance equal to that of the load Z_L. To obtain the desired impedance, they contain inductive as well as capacitive and resistive elements. Networks of this type form the basis for the design of a wide variety of equalizers and sharp cutoff filters. They are widely used in studio recording and broadcast applications, but do not usually have wide applications in home sound reproducing equipment where their major uses are in loudspeaker frequency dividing networks and in dynamic noise suppressors. Their specific applications to sound reproducing systems will be described in later sections in connection with these applications.

Design of Equalizer and Tone Control Circuits

Different equalization and frequency response characteristics are often required in different audio systems; therefore, it is necessary to be able to design equalizers to meet specific requirements. Unfortunately, the frequency response and impedance equations which are involved in the exact calculations of equalizer performance include complex functions representing capacitive reactances with resistances, and are not capable of simple solutions. However, by means of a number of fairly simple approximations and observations, it is possible to design such equalizers and predict their performance without great difficulty.

To examine the response of any equalizer it is necessary to establish certain reference frequencies from which to measure the amount of boost or attenuation. A convenient reference frequency generally used is 1,000 cps. Generally, for bass boost or attenuation equalizers, any high frequency may be used since the response will be constant above about 1,000 cps, while for treble boost or attenuation any low frequency may be used since the response will be constant below about 1,000 cps.

The basic equalizer circuits using RC networks are shown in Fig. 7-6. All the circuits are simple a-c voltage dividers. For low-frequency attenuation the circuit consists of a capacitor in the series arm with a

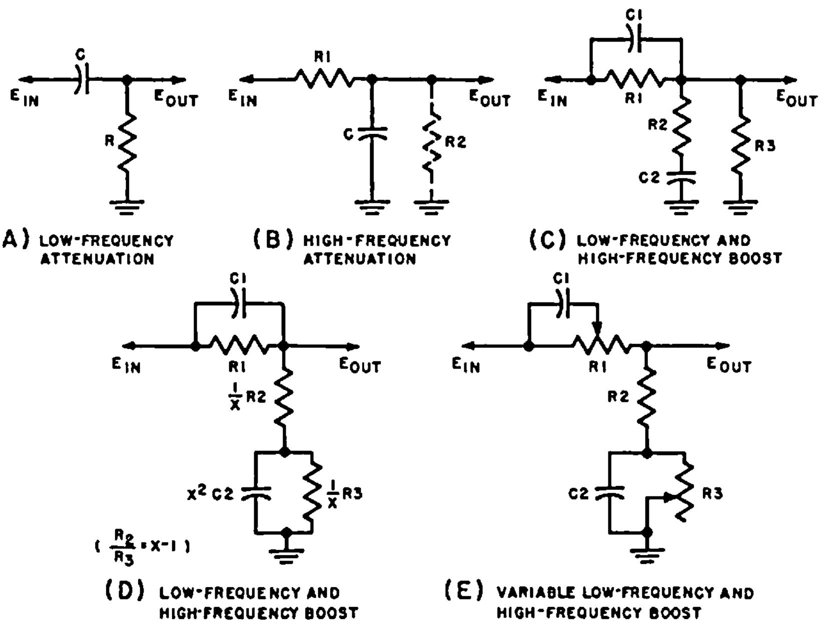


Fig. 7-6. Basic RC equalizer circuits.

resistor as the shunt arm: for high-frequency attenuation the circuit is a resistor in the series arm, with a capacitor (in parallel with a resistor) as the shunt arm. For high- and low-frequency boost, the network consists of a resistor and capacitor in parallel as the series arm, and two resistors and a capacitor to form the shunt arm. The input and output impedances vary with frequency, and these networks are most practical for use in high-impedance circuits such as interstage coupling and feedback in resistance-capacitance coupled amplifiers.

Low- and High-Frequency Attenuation. Low-frequency and high-frequency attenuations are obtained from the circuits in Fig. 7-6 (A) and (B) in the following manner: In the network of part (A), the output is determined by the voltage divider consisting of the series capacitor and the shunt resistor. At high frequencies, the reactance of the capacitor is small compared to the resistance, and essentially the entire input voltage is delivered to the load. Below a certain crossover frequency, determined by the relative values of the resistance and capacitance, the reactance of the capacitor becomes large compared to the resistance, as the frequency becomes lower the voltage delivered to the output becomes progressively less and less as the reactance of the capacitor increases still more.

In the network of part (B), the output is determined by the voltage divider consisting of the series resistance (*R1*) and the parallel resistance (*R2*) and capacitor (C) in shunt. At low frequencies the capacitor has essentially infinite reactance and the output is determined by the ratio of the resistances. Above some crossover frequency, the reactance of the capacitor begins to short-circuit the shunt resistance, and as the frequency increases more and more, the output voltage becomes progressively less and less as the reactance of the capacitor decreases.

Low- and High-Frequency Boost. Low-frequency and high-frequency boost are obtained from the circuits in Fig. 7-6 (C) and (D) in the following manner: At the middle frequencies the circuit is essentially a resistive voltage divider consisting of *R1* in series with *R* (where *R* is the combined resistance of *R2* and *R3* in the shunt arm when the capacitor is short circuited), since at these frequencies capacitor *C1* is chosen to be effectively a very high reactance and *C2* is chosen to form practically short-circuit. Thus, the insertion loss of the network is determined by the output of this voltage divider, and this also determines the maximum amount of frequency correction, since the maximum voltage output cannot be more than the input voltage.

At higher frequencies the reactance of capacitor *C1* becomes smaller until, at sufficiently high frequencies, resistor *R1* is effectively short-circuited and the entire input voltage appears across the output terminals.

At low frequencies, the series arm *R1* remains constant, but the reactance of *C2* in the shunt arm of the network increases as the frequency decreases. Therefore, at low frequencies the shunt arm of the voltage divider becomes relatively a higher impedance compared to *R1*, and a greater proportion of the input voltage appears across the output terminals. The maximum amount of low-frequency boost or lift is determined by the relative values of resistors *R1* and *R2*.

In this circuit, the specific amounts of high- and low-frequency equalization are determined by the relative values of resistors *R1*, *R2* and *R3*. The frequencies at which the boost occurs are determined by the value of *C1* relative to *R1*, and of *C2* relative to *R2*.

Design Curves. The equations for the frequency response characteristics of these circuits, together with a chart and a set of curves by which practical RC corrective networks may be designed for specified characteristics, are given in Fig. 7-7. The design curves are plotted in terms of frequency ratio and in decibels of equalization. The formulae by which the values of resistance and capacitance are determined from the curves are given in the figure. When they are drawn in this manner the curves and the figure are thus universal in application to the design of equalizers of this type.

In designing an equalizer in a practical problem the desired curve is compared with the curves in Fig. 7-7, and the most suitable curve is selected or interpolated from the given curves. This comparison gives the values of high- and low-frequency equalization, the middle frequency insertion loss, and the frequencies. From these, values of resistance and capacitance are found by means of the formulae.

One additional piece of information is required before the high- or low-frequency boost equalizer can be designed; one of the resistors *R2* or *R3* must be selected as the reference impedance for the network. These values are generally dictated by the requirements of the circuit in which the network is to be used. The network has its minimum input impedance (equal to *R*) at high frequencies, and its highest output impedance (equal to *R3*) at low frequencies. In most cases at least one of these values is determined by the circuit in which the network is used, and this furnishes complete data for the design of the RC equalizer.

The basic circuit for high- and low-frequency boost has been given in two different forms in Figs. 7-6 and 7-7. Usually the circuit in part (C) of Fig. 7-6 is more convenient to use when the amount of low-frequency equalization required is constant, as in interstage coupling in an amplifier to correct for deficiencies elsewhere in the system; while the circuit in part (D) is more convenient to use when a variable amount of equalization is required, as in the tone control of a radio

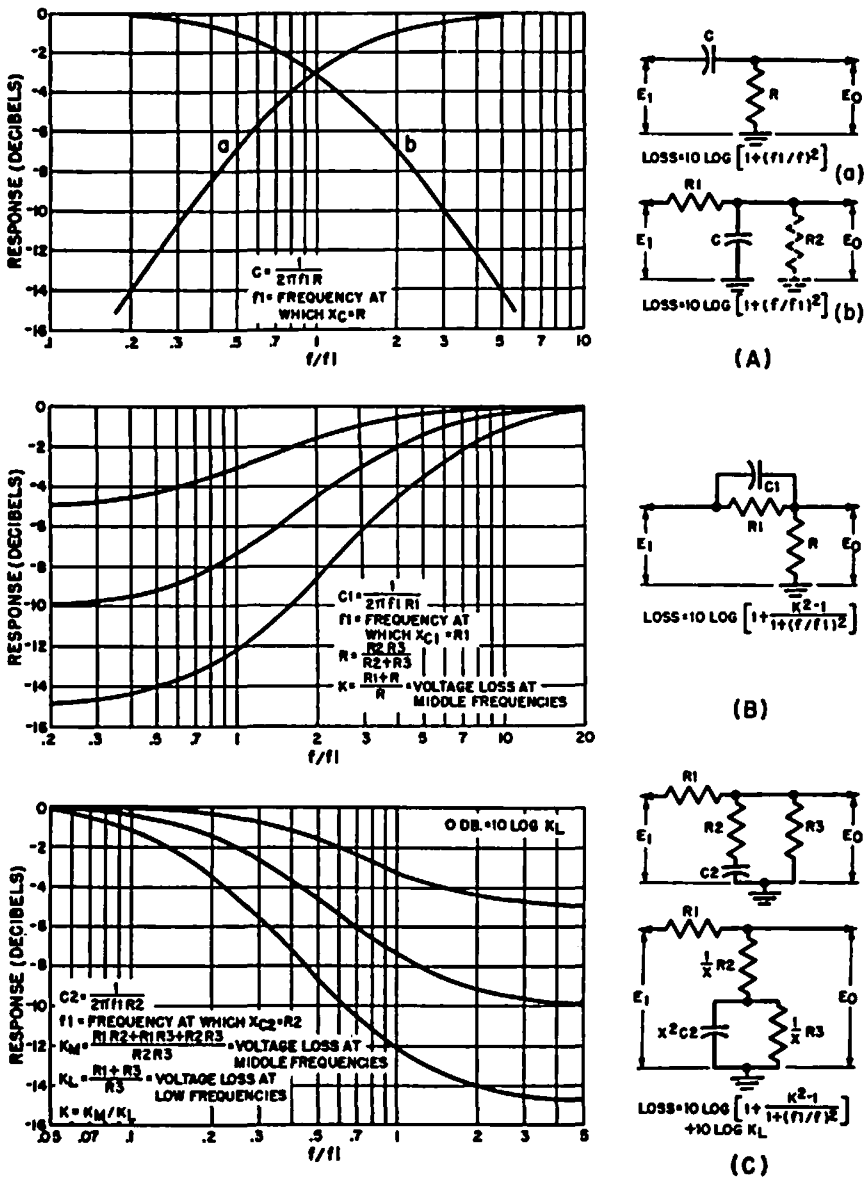
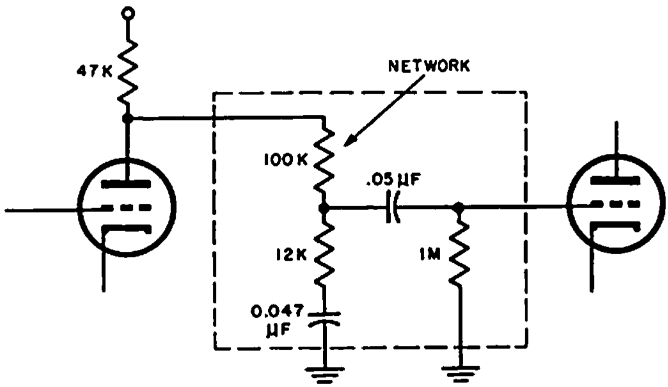


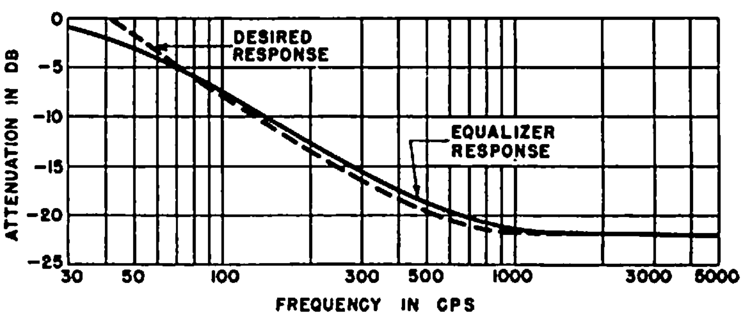
Fig. 7-7. Design data and charts for the design of various types of RC equalizers.

receiver or a phonograph amplifier. An example of how this network may be used as a variable equalizer is given in Fig. 7-6 (E). This variable equalizer has the feature that the high- and low-frequency equalization may be varied independently while the middle-frequency level remains constant regardless of the amount of equalization. The basic circuit which has been described may also be used for either high- or low-frequency correction alone. For low-frequency boost alone capacitor *C1* is omitted. For high-frequency boost alone, capacitor *C2* is removed and the two resistors *R2* and *R3* are replaced by a single resistor *R* having the appropriate value.

Example of Chart Use. An example of the use of these charts in the design of practical equalizer circuits to meet the specific requirements of audio systems may be illustrated by the procedure used in the design of a typical bass boost circuit for equalizing the low frequency response of a magnetic phonograph pickup.



(A) BASS BOOST CIRCUIT



(B) FREQUENCY RESPONSE CHARACTERISTIC OBTAINED FROM THE BASS BOOST CIRCUIT SHOWN IN (A)

Fig. 7-8. Design of a typical bass-boost equalizer circuit for magnetic pickups.

Since the response of a magnetic pickup is proportional to velocity, its frequency response to the standard NAB recording characteristic will be the same as the curve shown in (D) of Fig. 7-1. The required equalization to a flat response is accomplished in two sections: one to roll off the high frequencies, and the other to boost the low frequencies. Since the pickup has appreciable inductance, the high frequencies can be attenuated by using a load resistance of the proper value, to form a high-frequency attenuation equalizer of the type shown in Fig. 7-2 (C). The value of this inductance, and therefore the correct resistance value, must generally be obtained from the manufacturer of the pickup.

The low-frequency equalization is generally accomplished by a bass boost circuit inserted after the first stage of the preamplifier. This first stage will have a gain of about 30 db, therefore the insertion loss of the equalizer network can be tolerated, and the signal level will not be high enough to cause any overloading or distortion. The bass boost circuit is shown in Fig. 7-8 (A), and is seen to be of the type shown in part (C) of Fig. 7-7, except for the coupling capacitor whose reactance is so low that it may be ignored. Assume that the required response is as shown by the dashed curve in Fig. 7-8 (B). The required response is compared with the curves in the chart (Fig. 7-7) to determine the required resistance values to be used. Since none of the curves in the chart meet the requirements, a new curve can be drawn from the formula given in the chart, or a new curve may be extrapolated. The response for a maximum loss of 21 db is shown as the solid curve of Fig. 7-8 (B), and is seen to match the desired response within 1 db over most of the required frequency range. The resistance values of 12K and 125K (including the output impedance of the amplifier) give a loss at middle frequencies of:

$$K_m = \frac{125k + 12k}{12k} = 11.4 = 21 \text{ db}$$

The response is matched to the 500 cps crossover frequency by choosing the capacitor value to be 0.047μf so that *f1* is equal to 500 cps. The use of different capacitance values will give equalization for other crossover frequencies. The response curves show that this equalizer design gives very satisfactory equalization of the recording characteristic for good reproduction.

Variable Tone Control Circuits

In many cases the pre-emphasis curves in recording and the de-emphasis curves in playback may not always be properly matched to one another. In the resulting reproduced sound either the high frequencies or the low frequencies may be overemphasized or underemphasized. An adjustable equalizer or *tone control* network should be included in the

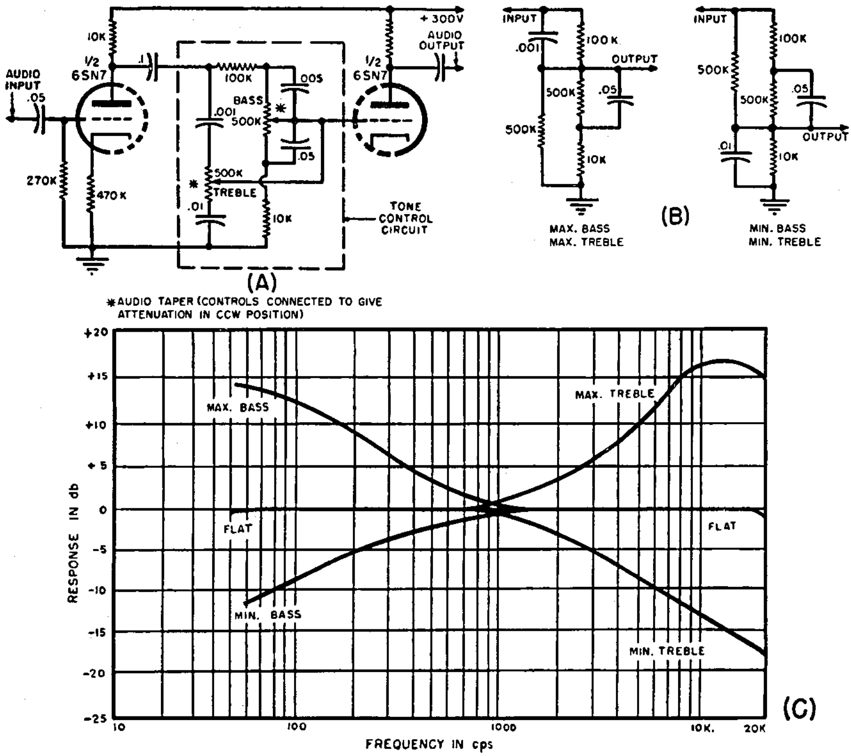


Fig. 7-9. (A) Circuit of a simple network which can be used as a variable tone control, showing method of matching into amplifier circuit. (B) Simplified circuits showing operation of the circuit at extreme settings of maximum bass and treble (left), and minimum bass and treble (right). (C) Range of frequency response curves which can be obtained with this circuit.

reproduction channel to provide for frequency response correction when the reproduced sound does not have the proper balance between high frequencies and low frequencies. A flexible tone control circuit should have a high-frequency control which could be set for either boost or attenuation of the highs, and a low-frequency control which is capable of either boost or attenuation of the lows.

The circuit of a simple network which serves as a variable tone control is shown in Fig. 7-9. This network has independent bass and treble controls capable of giving the range of frequency response curves shown in (C) of Fig. 7-8. The treble control can be adjusted over the complete range of settings from 15 db boost to over 15 db attenuation at 20,000 cps, while the bass control may be set to give from about 13 db accentuation to about 13 db attenuation at 50 cps. Since the controls are continuously adjustable, any intermediate setting between these two ex-

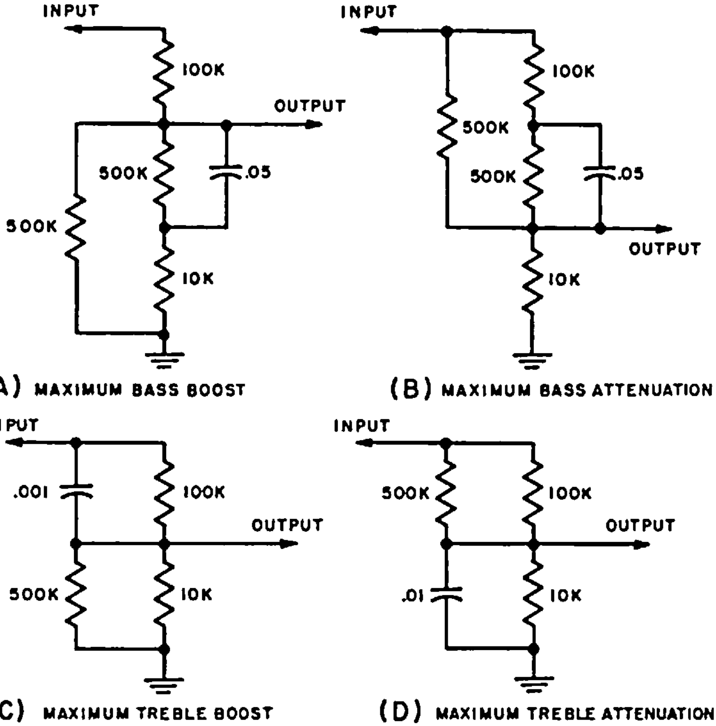


Fig. 7-10. Simplified circuits showing the operation of the variable tone control equalizer of Fig. 7-9 at the extreme control settings.

tremes may be obtained, including a flat frequency response position. The principle of operation of this circuit may be more clearly understood from the simplified circuits shown in Fig. 7-8 (B), which show the equivalent circuits in the maximum and minimum bass and treble positions, and by comparing them with the basic tone control circuits of Fig. 7-6.

When the bass control is set at the *maximum bass* position, if the high-frequency section is ignored, the bass boost circuit can be considered to be that shown in Fig. 7-10 (A). This circuit is seen to be similar to those described in (C) of Fig. 7-7, and has the frequency response shown. At low frequencies the insertion loss of this network is approximately 3 db, while at middle and high frequencies the insertion loss is approximately 21 db, therefore this network would have a bass boost of about 18 db.

When the bass control is set at the *minimum bass* position, ignoring the high-frequency section, the bass attenuation circuit can be considered to be that shown in Fig. 7-10 (B), and is seen to be similar to that described in (B) of Fig. 7-7. At middle and high frequencies the insertion loss is approximately 19 db, while at low frequencies the insertion loss is approximately 29 db. Therefore this network would have a bass attenuation of about 10 db.

When the treble control is set at the *maximum treble* position, ignoring low-frequency section, the treble boost circuit can be considered to be that shown in Fig. 7-10 (C), and is seen to be the circuit described in (B) of Fig. 7-7. At middle and low frequencies the insertion loss is 20 db, while at high frequencies there is no insertion loss. Therefore this network would have a treble boost of about 20 db. However, it can be seen that at high frequencies the impedance of the network is about 10k ohms, which would load the preceding amplifier stage slightly, and the maximum treble boost is closer to 16 db or 17 db, depending upon the specific circuit of the preceding stage.

When the treble control is set at the *minimum treble* position, ignoring the low-frequency section, the treble attenuation circuit can be considered to be that shown in Fig 7-10 (D), and is seen to be the circuit described in circuit (*b*) in part (A) of Fig. 7-7. At middle and low frequencies the insertion loss is approximately 19 db, and the high frequencies are attenuated at a rate of 6 db per octave above 1,500 cps.

When the various simplified circuits are combined into the complete able, any desired intermediate response characteristic between the ex- frequency and high-frequency sections which modify the response curves slightly from the calculated values. The insertion loss at 1,000 cps with the controls set for *flat response* is approximately 20 db; the frequency response curves which can be obtained are shown in Fig. 7-9 (C). Since the bass and treble controls are continuously and independently adjustable, any desired intermediate response characteristic between the extremes can be obtained. A typical circuit diagram showing the method of inserting this tone control into an audio channel is shown in Fig. 7-8 (A).

Bass Compensation and Loudness Controls

The experimental curves for the frequency response of the ear at different loudness levels given in Fig. 1-7 show that the response is different at different sound levels. From these curves it can be seen that the effect is most important at low frequencies, and its consequences are obvious in the apparent lack of bass in sound reproduced at low intensities. Therefore, when sound is reproduced at a level different from that at which it was originally produced, the tonal quality and balance is changed because of the different frequency response of the ear. Since it is not usually practical to listen to reproduced sound (particularly, for example, orchestral music) at the same level at which it was produced, this effect leads to generally undesirable results unless it is corrected.

Aside from increasing the level of the reproduced sound, the most direct method of correcting for this effect is by introducing frequency response equalization to compensate for the difference in the ear's sensitivity corresponding to the difference between the original and the reproduced levels.

The amount of bass correction required at different levels is shown in Fig. 7-11. Of course it would be possible to use a variable tone control setting to make the correction, however this method has the disadvantage that different settings would be required for each setting of the volume control and for each different loudness condition. Because of the difficulty of calibration, the proper settings would have to be obtained from the listener's judgment of the tonal quality and balance in the reproduced sound. This system is not too satisfactory, and methods have been devised for obtaining the required correction in a simpler and more reliable manner.

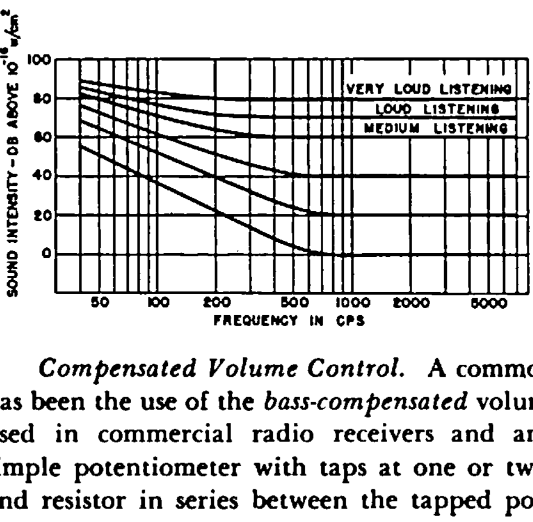


Fig. 7-11. Amount of bass-boost correction required to give the same tonal balance when level of reproduced sound is reduced below 100 db level.

Compensated Volume Control. A common and very simple method has been the use of the *bass-compensated* volume control, which is widely used in commercial radio receivers and amplifiers. It consists of a simple potentiometer with taps at one or two points, with a capacitor and resistor in series between the tapped point and ground, as shown in Fig. 7-12. The component values used with single-tapped and double-tapped volume controls in typical commercial radio receivers and amplifiers are shown in the figure.

The basic principle of operation of the circuit is quite simple. Comparison of the single-tap control with the circuits of Figs. 7-6 (B) and 7-7 (C) shows it to be essentially a modified bass boost circuit. If the signal is supplied from a relatively low-impedance source (such as a triode amplifier stage), then there will be practically no bass boost obtained when the arm is set at the high terminal of the potentiometer. As the arm is moved down, the amount of bass boost increases to a maximum at the tap, and remains constant for any settings below the tap.

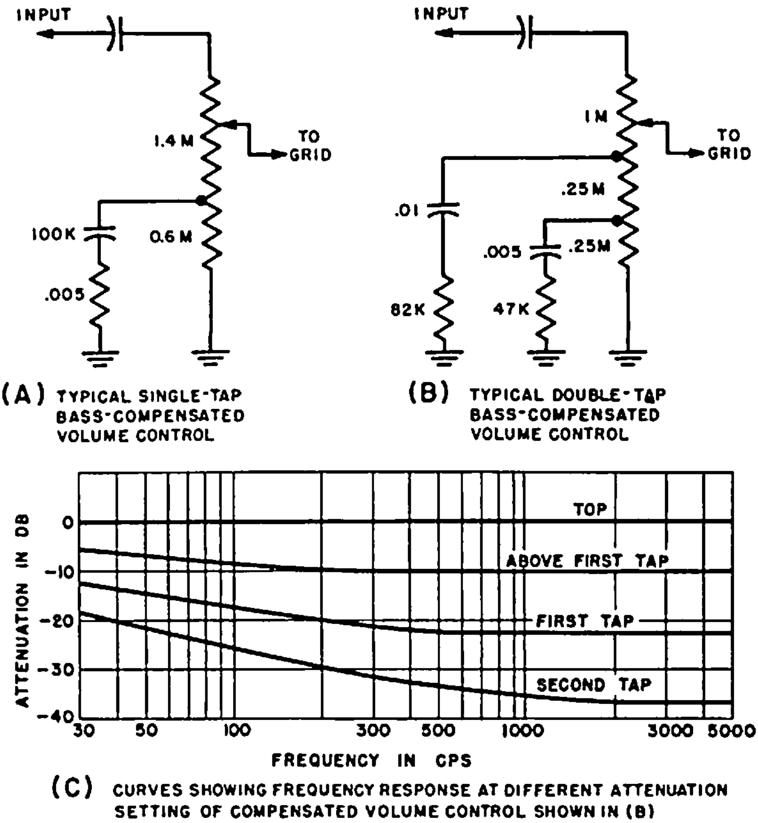


Fig. 7-12. Typical bass-compensated volume control circuits.

In this manner an increased amount of bass boost is obtained at low output sound levels, with the maximum boost at the lowest levels.

Typical frequency response curves of this type of compensated volume control are shown in Fig. 7-12 (C), which shows the response of the two-tap control at different settings of the arm. These curves approximate the amount of bass boost required for different output sound levels as shown by the curves in Fig. 7-11, and have given very good results in commercial radio receivers and audio amplifiers. However, when more accurate response curves are required for high fidelity reproduction, a more elaborate circuit is required to obtain this better response.

Loudness Control. Units for obtaining more accurate variations of frequency response with output sound level are called *loudness controls*, and are essentially elaborations of the bass-compensated volume control with a much greater number of taps shunted to ground. The circuit of a typical loudness control is shown in Fig. 7-13 (A). This unit can be built on a standard commercial 23 position selector switch, giving a con-

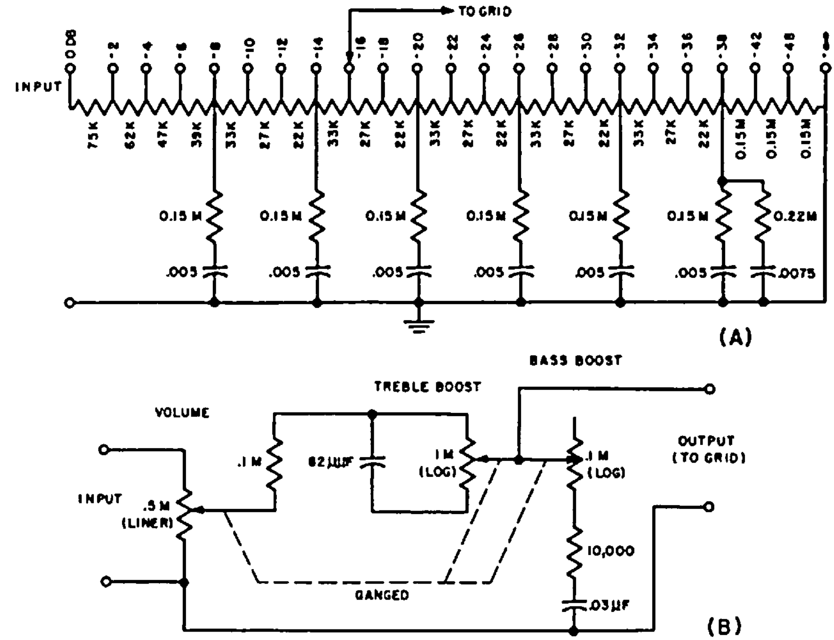


Fig. 7-13. Typical loudness control circuits.

trol with an attenuation range of about 50 db with an average change in level of about 2 db per step. It may be installed in the reproducing system in place of the volume control, or may be added to the circuit by inserting it between the plate and the following grid of some appropriate amplifier stage.

Another form of the loudness control is continuously variable instead of being a step type. In the continuously variable unit, three potentiometers are used which are ganged together as shown in the diagram in part (B) of Fig. 7-13. One of these controls is the volume control while the other two provide variable bass and treble boost. The degree of boost increases at low volume settings and, as required, the amount of bass boost is considerably greater than the treble boost.

Because it is intended as a correction for changes in level between the original and the reproduced sound, a certain amount of care must be exerted in the use of the loudness control. All input signals must be adjusted to the same level into the amplifier, so that when they are switched into the system the same volume setting will always correspond to the same output sound level regardless of what input signal is being reproduced. Otherwise, the correct compensation would not always be used, depending upon the particular input device from which the sound is being reproduced. Also, the output of the system at a given setting of the loudness control may not be at such a level that the proper amount of tonal correction is applied.

To overcome these effects, an auxiliary resistive volume control should be included in the system ahead of the loudness control. This is used for overall gain adjustment so that, with the loudness control in the top position, the output sound has the same level as the sound when it was originally produced. The loudness control can then be used to produce changes in the reproduced level, with the flat response for equal intensities being the reference characteristic. Whether the adjustment is made in this manner or by using the tone control as an auxiliary adjustment, the use of the loudness control for adjusting the output sound level can be quite valuable and useful in audio systems, since it eliminates any apparent change in the quality and tonal balance of the reproduced sound even at low intensity levels.

Volume Compression and Expansion

Even with the use of pre-emphasis techniques, an overall noise level of better than –60 db below full output is fairly difficult to attain. When reproduction from records or a-m broadcasting is included in the channel, the noise level will generally be considerably higher, as high as –40 db below peak signal or even more for bad records. Since the actual dynamic range to be reproduced may be as high as 75 db, the noise level (even assuming the –60 db figure) prevents the reproduction of this entire dynamic range of signals. Thus, if the output is set for maximum at the highest signal level, the low-level signals will be lost in the noise. Furthermore, the maximum output of the system may be too loud for comfortable listening.

To improve this condition, the sound signal is usually monitored during recording or transmission, and the gain in the channel is adjusted to reduce the dynamic range. This function may be performed either manually, by an operator who watches a signal-level meter and adjusts a gain control to keep the signal peaks within specified limits, or automatically by electronic gain adjusting circuits. The automatic electronic units need just be set up properly for the particular signal being transmitted or recorded, and then observed periodically to insure proper operation.

The functioning of the electronic units for automatic gain adjustment is based upon two different principles:

(1) *Volume compression*: in which the channel gain is reduced in proportion as the signal level increases, so that the resulting dynamic range of the signal is the same as that of the reproducing system. The operation of this system is based upon approximately the same principle

as that of manual gain adjustment, in which the operator decreases the gain for loud signals to keep them below some specified maximum, and increases the gain for low levels to keep them above some specified minimum.

(2) *Peak limiting*: in which the channel gain is adjusted to keep all low-level signals above the noise level, and gain control is applied only to the loud peaks. The high levels are then transmitted at their normal level as long as they are not greater than some fixed maximum which can be handled by the system without excessive distortion. Whenever the signal has a greater amplitude the gain is reduced enough to bring it down to this fixed maximum amplitude.

The volume compression method decreases the dynamic range gradually for signals of all levels, whereas the peak limiting method does not affect signals below the channel maximum and limits high-level peaks to this maximum. In playback, a *volume expander* circuit which performs the inverse function, may be used to restore the dynamic range of the original signal. The curves in Fig. 7-14 illustrate the basic func-

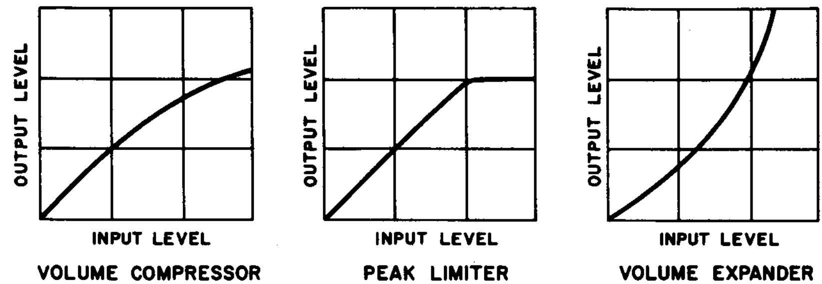


Fig. 7-14. Typical amplitude response curves of volume compressor, peak limiter, and volume expander.

tions of these units by showing typical curves of output signal levels for various input levels. If a volume expander circuit is used with a signal which has been compressed, the original dynamic range is restored. If a peak limiter has been used in transmitting the signal, the reproduced dynamic range is increased by the expander, but the relative signal levels are altered.

Circuits for Volume Expansion, Compression, and Limiting

The basic principle of operation of the volume compressor or expander is shown in the block diagram of Fig. 7-15. The audio signal is amplified to a sufficiently high level, and then rectified to give a d-c voltage proportional to the signal level. This voltage controls the gain of a variable-gain stage in such a manner that it may be either reduced or increased as the signal voltage increases, depending upon whether the circuit is to be a compressor or an expander. A gain control in the

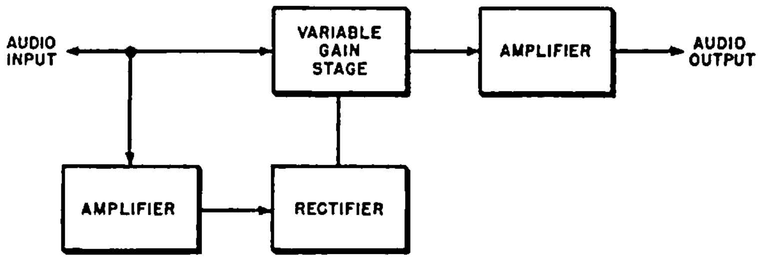


Fig. 7-15. Block diagram illustrating the basic principle of operation of the volume compressor, peak limiter, and volume expander.

amplifier-rectifier circuit can be used to control the amount of compressor or expander action, and a delay voltage can be introduced into the rectifier circuit so that the action occurs only above a certain level and peak limiting occurs.

The circuit of a simple compressor or expander is shown in Fig. 7-16 (A), and the principle of its operation may be seen more clearly from the simplified diagram and equivalent circuit shown in Fig. 7-16 (B). The input signal is rectified and the developed voltage applied to the grid of a triode whose plate resistance forms the shunt resistor in an audio-frequency voltage-divider network. The equivalent circuit shows this principle of operation; it can be seen that with proper selection of the resistor values, if the plate resistance of the tube is made to vary properly, the output signal level can be controlled over a wide range of input levels.

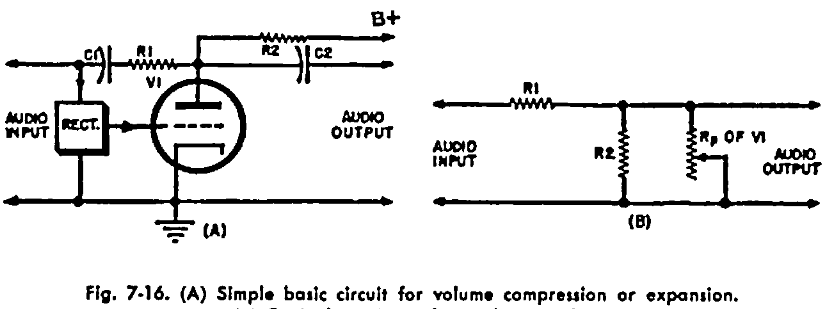


Fig. 7-16. (A) Simple basic circuit for volume compression or expansion. (B) Equivalent circuit for audio signals.

Compressor Circuit. A simple practical circuit based on this principle is shown in Fig. 7-17. The input signal is amplified by one-half of the 6SL7 dual triode, whose output is rectified by the second half of the tube connected as a diode. The voltage from the diode rectifier is then a measure of the signal level. This voltage is properly filtered and applied to the grid of one-half of a 6SN7 tube to control its plate resistance, which is used as the variable resistance in the audio voltage-divider

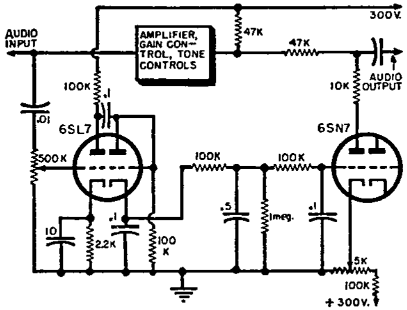


Fig. 7-17. Simple practical circuit for a volume compressor.

circuit. The voltages are chosen so that for high signal levels a positive voltage is applied to the grid of the variable-resistor tube, which lowers its plate resistance and therefore decreases the gain of the channel. This compression circuit is capable of giving a compression up to 10 to 15 db with 0.5 volt rms applied to the input.

Peak Limiter Circuit. The operation of a peak limiter may be compared to that of the compressor, and may be considered as having a very high degree of compression which takes place only when the input signal becomes greater than some specified level. The circuit of a peak limiter whose operation is based upon the same audio voltage divider principle of Fig. 7-16 is shown in Fig. 7-18 (A). The input signal is rectified by a high-impedance cathode detector which is the first section of the 6SL7, and the developed voltage is applied between the grid and cathode of the second section of the 6SL7. This acts as the variable shunt resistance of the voltage divider. The values of the components have been selected to give the best voltage signal-level characteristic to result in a linear compression curve. This circuit gives a peak-limiting curve similar to that shown in Fig. 7-18 (B), and is capable of 15 to 20 db limiting with an output level change of less than 1 db. The limiting action is initially adjusted by means of the 25,000-ohm cathode resistor, then the degree of

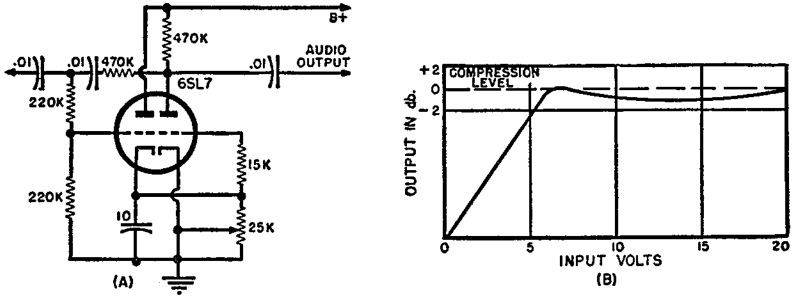


Fig. 7-18. (A) Simple practical peak limiter circuit. (B) Typical amplitude response of this circuit.

limiting for any transmitted audio signal is adjusted by controlling the level of the signal applied to the input of the limiter circuit.

Expander Circuit. A volume expander is essentially the reverse of the compressor circuit which has already been described. The major difference between the two is that the diode rectifier must be reversed and the voltage levels reset, so that for high signal levels the channel gain is increased. Thus the circuit of Fig. 7-19 is essentially the compressor circuit of Fig. 7-17, however, the rectifier connections are reversed to give a negative voltage when the audio level increases, and the cathode of the variable-resistance tube is connected directly to ground, so that the plate resistance is low for low-level signals and high for high-level signals. The gain, therefore, increases as the audio level increases. This circuit is capable of giving up to 10 db expansion; component values may be selected to give greater expansion if desired.

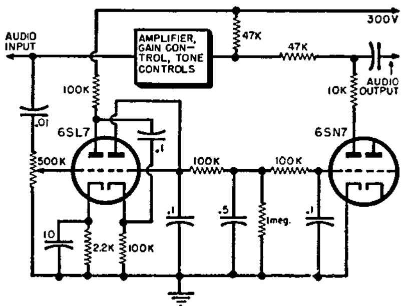


Fig. 7-19. Simple practical volume expander circuit.

In the use of volume compression and expansion circuits, the time constants of operation are extremely important, and generally represent a compromise of several factors. The time of initial operation should be rapid to act properly on the initial peaks, but the release time presents a more difficult problem. If the release time is too short, a sort of "pumping" effect is obtained in which the gain is constantly adjusting to rapid changes in level. If the release time is too long, sharp dynamic effects in the program material will suffer. Optimum conditions are a very rapid initial operation time, and a release time of about one-half to one second.

The units which have just been described can either be included directly in the amplifier or constructed on a separate chassis with the gains and levels adjusted for unity gain at some specified level (which will require the inclusion of additional amplification), and switched into the channel whenever they are required.

Dynamic Noise Suppressors

Another approach to the problem of background noise has been an actual attempt to remove the noise which has been introduced into the reproduced sound. A number of different methods have been developed for this purpose. One of the most widely accepted of these methods is the *dynamic noise suppressor;* its operation is based essentially upon the fact that the frequency response of the ear changes with sound level. This method is especially useful in the reproduction of sound from records, where the high noise level which is often encountered can be extremely objectionable to the ear.

Principles of Operation. The experimental curves of the frequency response of the ear at different loudness levels show that not only does the ear have different frequency responses to different levels, but that the upper and lower frequency limits of hearing also depend upon sound intensity. At low levels the audible frequency range is considerably narrower than at high levels. The basic data from the curves is reproduced in Fig. 7-20, showing the curve for the lowest levels which can be heard by the ear at each frequency. In order for a tone to be audible, its amplitude must be greater than the threshold of audibility curve at the frequency of that tone.

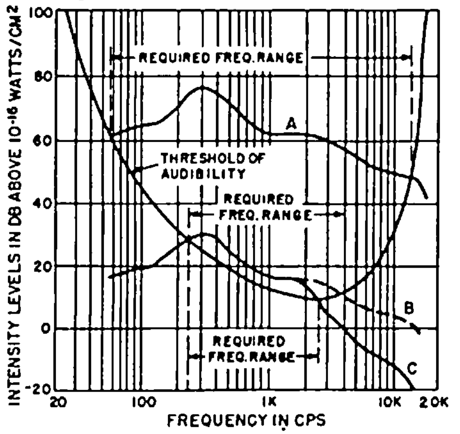


Fig. 7-20. Required frequency range for reproduction of music at different levels. Curve A is the most probable frequency distribution in music at high levels. Curve B is the same as A at lower volume. Curve C is the most probable frequency distribution with the orchestra playing soft passages.

On the same graph (Fig. 7-20) the most probable frequency distribution in orchestral music at fairly high levels (curve *A*) is also shown. If this orchestra is reproduced at a very low level the curve *B* applies. The frequency distribution when the orchestra is playing very softly (curve *C*) shows that the production of harmonics in soft playing is less than in loud. All components which are outside the intersections of these curves with the hearing threshold curve are not perceived by the ear. Note how the required frequency range is reduced at low levels of reproduction and with soft orchestral playing. Now, if a filter which cuts off at these frequencies is inserted in the channel it will have no effect on the music. However, it will result in a tremendous decrease in the high-frequency and low-frequency noise, whose levels are well above

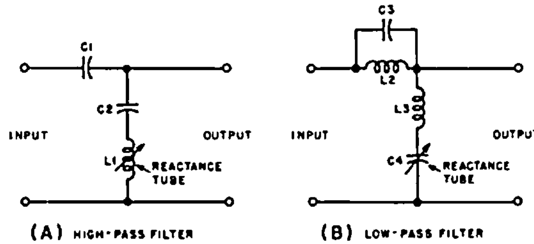


Fig. 7-21. Basic filter circuits of a dynamic noise suppressor.

the threshold of hearing. Of course, as the sound levels change, the cutoff frequencies will change correspondingly. A filter of this sort, whose cutoff frequencies are controlled by the level of the reproduced sound, forms the basis of the dynamic noise suppressor. Although it is extremely difficult to evaluate the degree of noise reduction resulting from such a circuit, it may be estimated to be in the neighborhood of approximately 20 db.

Basically, the dynamic noise suppressor consists of two filters in cascade: a variable low-pass filter to cut out the high-frequency noise components *above* the range of the reproduced sound, and a variable high-pass filter to cut out the low-frequency noise components *below* the range of the reproduced sound. This is shown in Fig. 7-21. These are simple filter sections in which the shunt capacitance in the high-frequency cutoff section, and the shunt inductance in the low-frequency cutoff section are replaced by reactance tubes. The frequency response is made variable by controlling the reactances of the tubes with the amplitude of the reproduced audio signal. Typical frequency response characteristics which can be obtained with these filter circuits are shown in Fig. 7-22.

These curves show the response for conditions of maximum frequency range, intermediate suppression, and maximum suppression. In normal operation the frequency response characteristics vary rapidly between the two extremes, according to the instantaneous requirements of the reproduced audio signal. Sharp cutoff characteristics are required in both sections, but particularly in the high-frequency section. The high-frequency section is required to attenuate noise such as record scratch and other random background noise which is proportional to

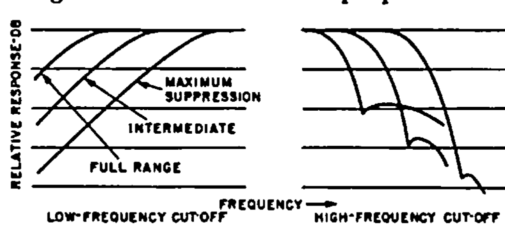


Fig. 7-22. Typical frequency response characteristics obtained with the variable filters shown in Fig. 7-21.

the absolute bandwidth in cycles per second. The low-frequency section attenuates noise such as hum and rumble in phonograph turntables. The circuits which have been selected for this purpose give this sharp cutoff characteristic with fairly simple arrangements.

Characteristics. This system of noise suppression has the following characteristics:

(1) The reproduced frequency range is continuously and automatically adjusted to produce optimum signal-to-noise ratios without loss of important high-frequency or low-frequency components in the program material.

(2) The bandwidth is controlled only by the audio signal level, and is not broadened by strong noise signals.

(3) The high-frequency and low-frequency cutoffs are controlled independently according to the requirements of the reproduced signal, and can be adjusted to provide the proper aural balance.

(4) Time constants in the control circuit can be adjusted to give proper speed of operation of the variable frequency range to reproduce transients, and to filter noise without damaging reverberation which may be present in the sound.

(5) Negligible harmonic and intermodulation distortion are introduced into the reproduced signals when they are passed through the variable-cutoff filters, and the circuit parameters can readily be selected so that the operation of the dynamic control introduces no audible thumps or other transients into the reproduced signal.

Because it possesses these characteristics, and can be built into a
Circuit Diagram. The circuit diagram of a simple dynamic noise
suppressor has found wide application in sound reproducing systems where noise reduction is an important problem.

Circuit Diagram. The circuit diagram of a simple dynamic noise suppressor of this type is shown in Fig. 7-23. It consists essentially of a low-pass filter and a high-pass filter of the type shown in Fig. 7-21, with two 6SG7 tubes as the variable reactance tubes in the filters. The first section (high-frequency gate) is the high-frequency cutoff filter, with the tube acting as the variable capacitive reactance to control the high-frequency cutoff. The second section (low-frequency gate) is the low-frequency cutoff filter, with the tube acting as the variable inductive reactance to control the low-frequency cutoff. The signal is also amplified in an auxiliary amplifier by the triode section of a 6SQ7. The output of this amplifier is then passed through two separate control circuits and rectified in the two diode detectors of the 6SQ7 to give d-c voltages proportional to the signal level in each channel. The low-frequency con-

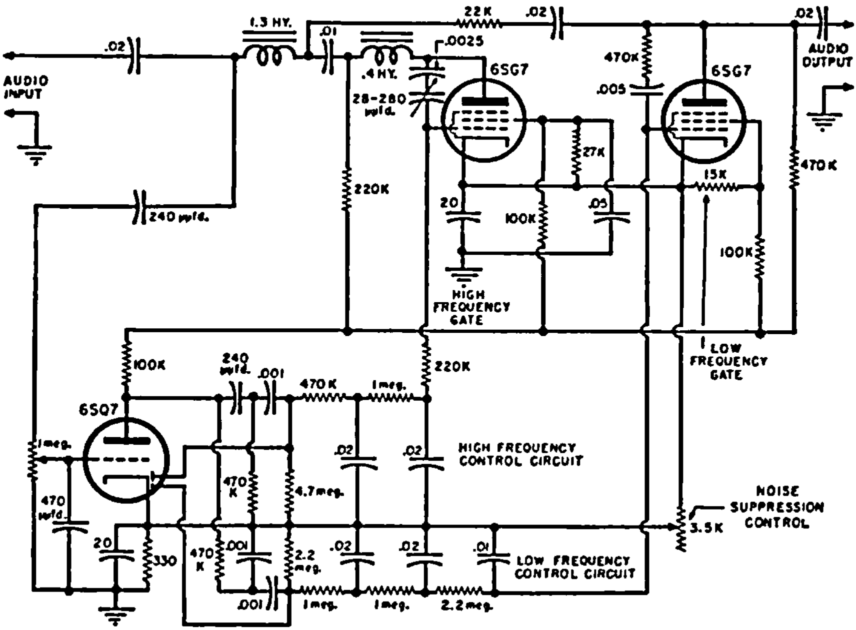


Fig. 7-23. Circuit diagram of a simple and practical dynamic noise suppressor.

trol channel contains a low-pass filter so that the rectified control voltage is proportional only to the amplitude of the low-frequency components: the high-frequency control channel contains a high-pass filter so that the rectified control voltage is proportional only to the amplitude of the high-frequency components. The rectified control voltages are then filtered and applied to the control grids of the respective reactance tubes to control the cutoff frequencies as required by the reproduced signal. The noise suppression control alters the bias voltages on the gate tubes so that the degree of suppression may be varied.

This noise suppressor circuit may be used either alone in the audio channel as a separate unit which is switched into the channel as desired, or may be constructed as an integral part of a more elaborate preamplifier and tone control unit. Generally a switch is desirable for removing the noise suppression from the audio channel if the program material being reproduced has a very low noise level, since in such a case the restriction in frequency range introduced by the noise suppressor would very likely be more undesirable than the small amount of noise present in the reproduced sound. However, in the reproduction of sound from transmissions or recordings in which a moderate to a great amount of background noise is present, the use of the dynamic noise suppressor can be very useful in improving the quality of the reproduced sound.

Chapter 8

LOUDSPEAKERS AND LOUDSPEAKER ENCLOSURES

General

The most difficult question which faces the high-fidelity enthusiast or the audio experimenter in setting up a high quality sound reproduction system is what loudspeaker arrangement to use. Unfortunately there is no simple and easy answer to this question. The choice of a loudspeaker system depends upon both the amount of money and the effort which can be invested in it.

In designing and setting up any sound reproduction system, it should always be kept in mind that the quality of the reproduced sound can be no better than that produced by the poorest component in the system. Generally this "poorest component in the system" is the loudspeaker. The loudspeaker is required to produce the same sound which is produced by all the instruments of a large orchestra, over the entire audible frequency range. It is required to project into the air of the listening room low-frequency vibrations identical with those of the large instruments such as the bass viol and the pipe organ, and the high-frequency vibration of the triangle and the piccolo. The difficulty of accomplishing this function is obvious, and most of the improvement which has taken place in the quality of sound reproduction has resulted directly from the improvement in loudspeaker design and manufacture.

The loudspeaker performs its functions as an electromechanical system; its performance is limited by the wavelength and the amount of air it can move at the low frequencies, and by the mass of the moving parts at the high frequencies. Because of these fundamental difficulties, the quality of the loudspeaker system usually is the major factor which determines the overall fidelity of any sound reproduction system. The other components of the system can be made to give good performance at a fairly reasonable cost, however, really good loudspeakers are quite expensive. In the design and construction of any sound reproduction system, the loudspeaker which is selected should be the best one which can be afforded, and the necessary expense or effort should be put into the choice of the proper enclosure for the loudspeaker. If the proper attention is thus paid to the loudspeaker system, the effort and expense will be justified by the improvement in overall sound reproduction quality.

Requirements. The loudspeaker must be of comparable quality and should meet the same requirements as the rest of the system. That is, when an electrical signal of the proper characteristics is applied to the input terminals, the output sound should be free of frequency, amplitude, transient, and other distortions at all rated input power levels. Ideally, the distortions introduced by the loudspeaker should be within the limits specified for the rest of the system, but in practice there are no speakers available at the present time which meet these requirements. (However, this does not mean that the quality of the rest of the system should be made worse, since this would only further lower the overall quality of the system.) It is difficult even to measure the characteristics of loudspeakers since such a measurement requires a sound-standard microphone and must be done in a room with no resonances and whose walls do not reflect any sound to interfere with the measurement. The information which is supplied by the manufacturer must be accepted by the experimenter who does not have extensive testing facilities. Generally this information is quite reliable.

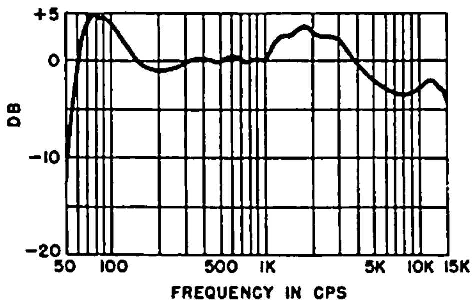


Fig. 8-1. Frequency response of a typical high-quality loudspeaker mounted in a good bass-reflex cabinet.

The selection of the loudspeaker itself is primarily a matter of listening to a number of different units in appropriate enclosures, studying the performance specifications supplied by the manufacturer, and making a choice which almost always represents a compromise between individual preferences and expense.

The frequency response curve of a typical high quality commercial loudspeaker is shown in Fig. 8-1. This response illustrates some of the important factors which should be found in a good loudspeaker.

(1) The response should be reasonably flat over a frequency range of 50 to 10,000 cps.

(2) The frequency response curve should be fairly smooth, with as few sharp peaks and dips as possible, since these discontinuities in the response represent mechanical resonances which result in bad transient response.

(3) The power rating of the speaker or speakers used should correspond to the rest of the system and to the requirements of the listening room, so that there will be no distortion at high sound levels.

Even these requirements can only be a guide in the selection of appropriate loudspeakers for sound reproduction systems, since the loudspeaker is a complex electromechanical system whose properties in relation to the ear are not yet completely understood. The loudspeaker particularly, of all the components of the sound reproducing system, should be chosen by a listening test, because the ear is the best judge of the overall integration of the many complex factors which are involved in loudspeaker design. However, a loudspeaker which has the above properties will generally be capable of giving very good sound reproduction when properly baffled and used with a good electronic system.

Loudspeakers for Sound Reproduction

The basic dynamic loudspeaker is the simple cone type in which the paper cone is caused to vibrate by the current in a small coil mounted between the poles of a magnet. In most modern loudspeakers, this magnet is a permanent type using highly magnetic material; earlier units employed an electromagnet. The construction is shown in Fig. 8-2. Other types of loudspeakers are essentially variations of this basic design.

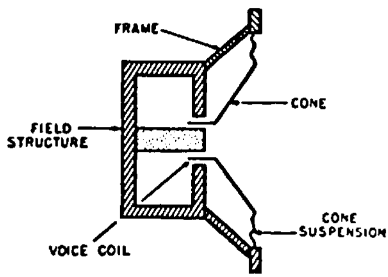


Fig. 8-2. Basic construction of the dynamic loudspeaker.

Equivalent Circuit. The electromechanical properties of the loudspeaker are best described by the use of the *elctromechanical equivalent circuit* as shown in Fig. 8-3. It must be understood that this is not the electrical circuit of the loudspeaker itself, but is just an analogy by which the mechanical and acoustical properties of the loudspeaker are represented by electrical quantities which are better understood by electronic technicians. In this equivalent circuit, the mechanical inertia of the loudspeaker cone behaves in a manner similar to an inductance in an electrical circuit, and is therefore represented by the inductance M_0. The compliance of the suspension system (which is inversely proportional to its stiffness) has the same mechanical properties that a capacitor has electrically, therefore it is represented by the capacitance C_s. The

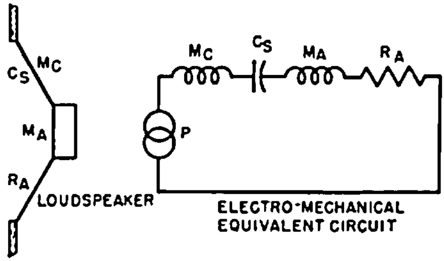
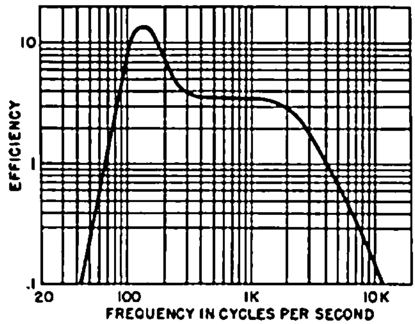


Fig. 8-3. Equivalent electrical circuit representation of loudspeaker performance.

C_S = ACOUSTIC CAPACITANCE OF SUSPENSION SYSTEM
M_A = INERTNESS OF AIR LOAD UPON CONE
R_A = ACOUSTIC RESISTANCE OF AIR LOAD
P = FORCE GENERATED IN VOICE COIL/AREA OF CONE
M_C = INERTNESS OF CONE

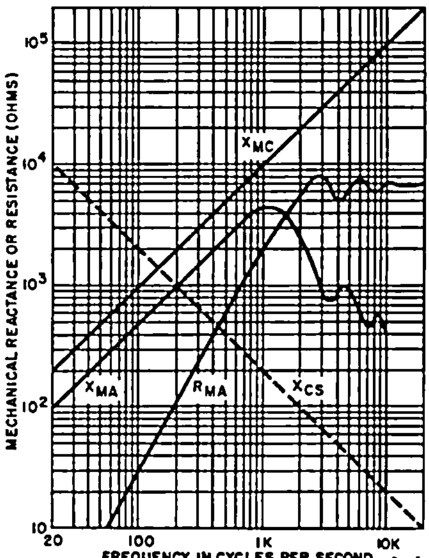
volume of air which is in contact with the surface of the cone is the load to which the mechanical energy is being delivered by the speaker. This air load has a mass which can be represented by an inductance M_a, and a resistive component which can be represented by a resistor R_a. The useful sound energy delivered by the loudspeaker into the air is represented by the power absorbed by the load resistor R_a. The pressure P of the generator in the acoustic system is the force generated in the voice coil divided by the area of the cone, and the current in the circuit represents the motion of the components.



(A)

CONE DIAMETER	4 in.
MASS OF CONE	1 gm.
MASS OF VOICE COIL	9.35 gm.
VOICE COIL MATERIAL	copper
AIR GAP FLUX	10,000 gauss

(B)



(C)

Fig. 8-4. Practical application of the electromechanical equivalent circuit representation to the design of a loudspeaker. (A) Efficiency characteristics of the loudspeaker. (B) Mechanical characteristics. (C) Various mechanical impedance components as a function of frequency.

From the electromechanical equivalent circuit it can be seen that the loudspeaker is basically a band-pass circuit, giving good response at the center frequencies and dropping off at the high and low frequencies. The manner in which this circuit applies to a practical loudspeaker design is shown in the graphs of Fig. 8-4. The particular loudspeaker for which these curves are drawn has a 4-inch cone diameter and the mechanical characteristics shown in Fig. 8-4 (B). Its various impedance components as functions of frequency are shown in Fig. 8-4 (C). (Here, X_{mc} is the mechanical reactance due to the inertness of the cone, X_{ma} is the mechanical reactance due to the air load, X_{cs} is the mechanical reactance due to the capacitance of the suspension system, and R_{ma} is the mechanical resistance due to the air load.) The efficiency, which is the ratio of the sound power output to the electrical input, is shown in Fig. 8-4 (A).

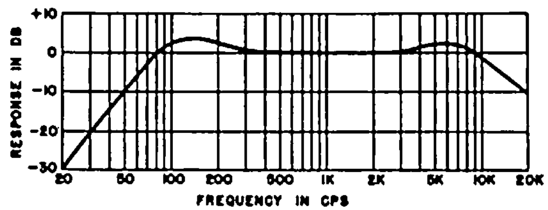


Fig. 8-5. Frequency response on the axis of the loudspeaker shown in Fig. 8-4.

If the loudspeaker were nondirectional, the efficiency characteristic would be the frequency response characteristic. However, the response is measured on the axis of the speaker, and the directional effects give a proportionately greater sound pressure on the axis at the higher frequencies. If this is taken into account, calculations show the sound pressure response on the axis to be as shown in Fig. 8-5, and this is very close to the actual measured frequency response characteristic.

Speaker Size and Frequency Response. The curves in Fig. 8-4 and 8-5 show that a loudspeaker with a small and relatively lightweight cone and voice coil is capable of giving good response and efficiency over a wide frequency range. However, a loudspeaker with a small cone is not able to deliver much acoustic power at low frequencies because the required amplitude of vibration would be too great. If the cone is made to move with the required amplitude considerable distortion would result, and such reproduction would not be acceptable. A number of different methods have been used in avoiding this difficulty, and to obtain sufficient sound power output and good response over the entire audio frequency range. The simplest method is to use a single loudspeaker with a cone diameter large enough to deliver sufficient acoustic power at the low frequencies, but not too large to reproduce the high frequencies well. Loudspeakers with 8-inch, 10-inch, and 12-inch diameters have been widely used for this purpose with a reasonable amount

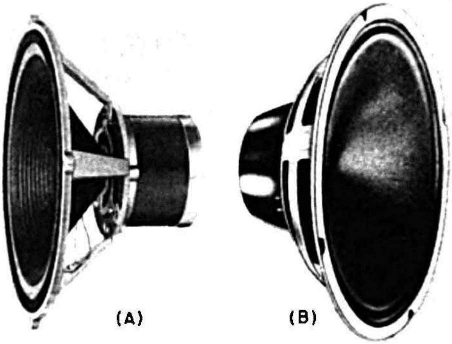


Fig. 8-6. Typical 10-inch (A) and 12-inch (B) loudspeakers suitable for use in sound reproduction systems.

Courtesy:
(A) British Industries Corp.,
(B) General Electric Co.

of success (see Fig. 8-6). Frequently the cone is specially treated (usually at the apex) to enhance high-frequency response. However, the use of a single loudspeaker generally represents a compromise solution. Since the response of the 12-inch loudspeaker is not completely satisfactory at the high frequencies (and begins to drop off at 10,000 cps or below), and the response of the 8-inch loudspeaker is not sufficient at the low frequencies (below about 100 cps), special techniques or multiple speakers are used.

One example of such a special technique is shown in the loudspeaker in Fig. 8-7. In this unit, called an "accordian-type loudspeaker," the outer edge of the cone does not touch the speaker chassis. Instead, the cone is supported from the rear by a folded structure of cone material. With this arrangement, cone resonance may be reduced by as much as a full octave below that obtainable with a speaker of similar size and ordinary construction.

Use of Multiple Speakers. Another method of obtaining sufficient response over the entire range involves the use of multiple speakers. A number of small loudspeakers may be used, so that each individual unit

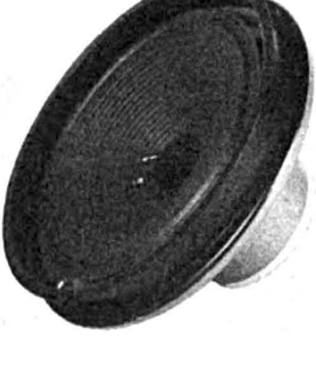


Fig. 8-7. Seven-inch accordian-type loudspeaker with low-frequency resonance at about 45 cps.

Courtesy: RCA

is operating within its power ratings, while at the same time a sufficient amount of sound power is delivered into the air by the total number of speakers. The voice coils may, of course, be connected either in parallel, series, or series-parallel to give the proper impedance to the amplifier. However, they must be properly phased so that the sound outputs do not cancel. When a multiple speaker arrangement of this type is used, the units may also be inclined at slight angles to each other to improve the high-frequency spatial distribution.

A more usual technique is the use of two loudspeaker cones: a large-diameter cone for reproduction of the low frequencies, and a small-diameter cone for reproduction of the high frequencies. Most of the best loudspeaker systems at the present time use this method, in one form or another, to obtain good frequency response. When two speaker cones are used in this manner to operate over the entire frequency range, it is necessary to separate the audio signal into two frequency ranges, so that the low frequencies alone are applied to the large cone while only the high frequencies are applied to the small cone. This function is accomplished by filters, which may either be mechanical ones in the loudspeaker, or electrical ones in the electrical circuit before the loudspeaker.

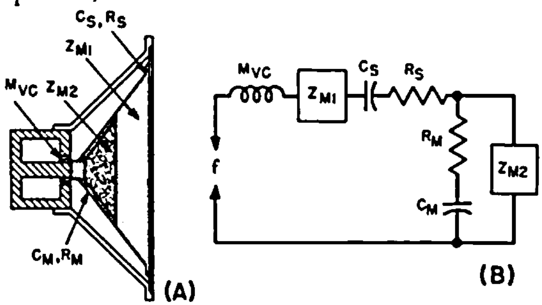


Fig. 8-8. (A) Basic system for use of a dual-cone loudspeaker to obtain wider frequency range. (B) Equivalent electrical circuit for the dual-cone loudspeaker.

The simplest and most basic method of mechanical filtering of the signal into the two frequency ranges is by use of the dual-cone principle, which is illustrated in Fig. 8-8. It consists of a single voice coil coupled to a two-section cone as shown in (A), with the two cones coupled together by a compliance. The electromechanical equivalent circuit in (B) describes the operation of this system. At low frequencies the reactance of the compliance is large compared to the mechanical impedance Z_{M2} of the large cone, therefore the entire current flows through both Z_{M1} and Z_{M2}, and the entire system moves as a whole. At high frequencies the reactance of C_M is small compared to Z_{M2} and bypasses it, therefore the small cone moves while the large one remains stationary. A system of this type makes it possible to extend the frequency range of the loudspeaker by almost a full octave, depending upon the mass and electrical characteristics of the voice coil. This type of loudspeaker gives extremely good

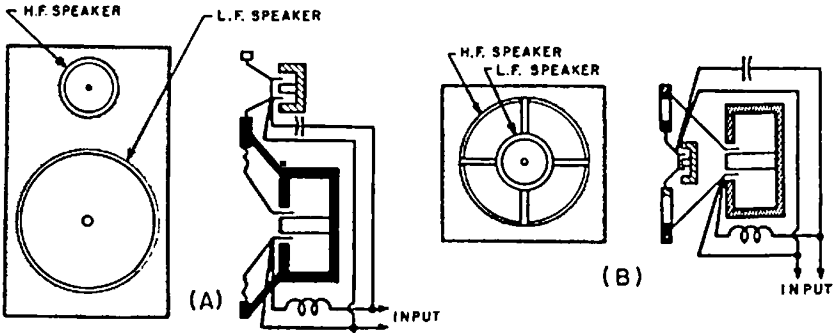


Fig. 8-9. Use of separate high-frequency and low-frequency loudspeakers to obtain a wide frequency range. (A) Two separate speakers mounted separately. (B) Coaxial structure with the high-frequency unit mounted in the center of the low-frequency loudspeaker.

results and has a very simple construction, with all the sound originating in the one loudspeaker. It is also quite simple to use in the audio system, since the output of the amplifier is applied only to two loudspeaker terminals, and no special networks are required.

The other method of using two loudspeaker cones to cover the audio frequency range is to use two separate loudspeakers: one for low frequencies (commonly known as the "woofer"), and another for the high frequencies (commonly known as the "tweeter"). Two types of dual units are commonly used, as illustrated in Fig. 8-9. The system shown in (A) consists of two separate speakers which may be mounted separately, with the high-frequency unit physically separated from the low-frequency unit. The two units may be purchased separately, and need not be mounted in the same cabinet, as long as they are kept close to one another so that the sound does not seem to come from two separate sources. In the system shown in Fig. 8-9 (B) the two loudspeakers are integrally mounted in a single coaxial structure, with the high-frequency unit at the center of the low-frequency unit. The use of two separate individual loudspeakers to cover the entire frequency range also has the advantage of flexibility, since any two suitable units may be used, and the experimenter may select any units which meet his requirements.

Loudspeaker Crossover Networks

When two loudspeakers are used in the above manner, the frequency below which the low-frequency speaker receives the electrical signal, and above which the high-frequency speaker receives the signal, is called the *crossover frequency*. This crossover frequency is generally in the region between 500 and 2,500 cps. When the signal is to be separated electrically into the two frequency ranges, a *crossover network* must be used to perform this function. The crossover network should present a constant impedance to the amplifier, and deliver power to the speakers from the proper impedance. The simplest network for separating the signal into two frequency ranges consists merely of a capacitor in series with the high-frequency unit, and an inductor in series with the low-frequency unit. A more elaborate crossover network, which presents better impedance matching and greater attenuation away from the crossover frequency, is shown in Fig. 8-10. This network gives an attenuation of approximately 15 db for the first octave away from the crossover frequency, as shown in the curves of Fig. 8-10 (B). The design formulas for this network are included with the circuit diagram in Fig. 8-10 (A). Since the loudspeaker frequency dividing networks carry the full output power of the amplifier, they must be designed for low insertion loss. When high-Q coils having low resistance are used, the loss may be kept down to the order of 0.5 db.

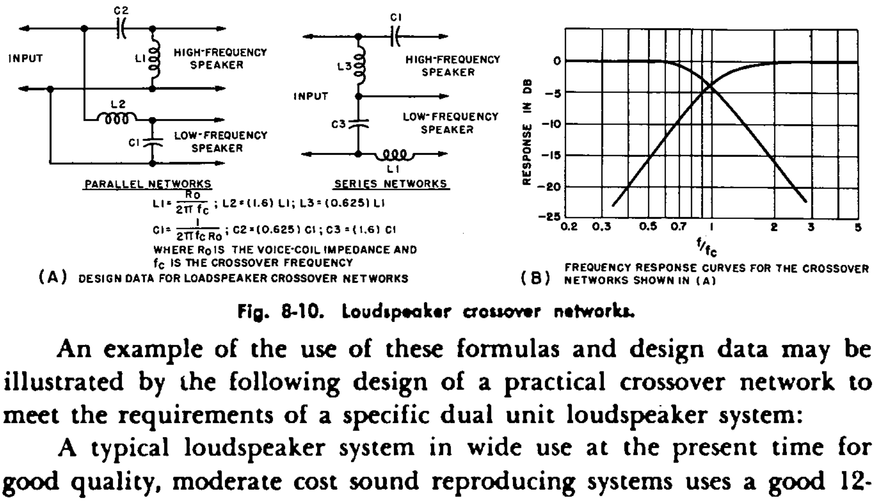


Fig. 8-10. Loudspeaker crossover networks.

An example of the use of these formulas and design data may be illustrated by the following design of a practical crossover network to meet the requirements of a specific dual unit loudspeaker system:

A typical loudspeaker system in wide use at the present time for good quality, moderate cost sound reproducing systems uses a good 12-inch unit for the production of the low frequencies, and a separate high-frequency unit for the production of the high frequencies. A crossover frequency of 2,500 cps is required and a separate network is used to divide the audio signal into the two frequency ranges above and below this frequency. The two loudspeakers have 8-ohm impedances; therefore, with the 2,500 cps crossover frequency, the application of the formulas in Fig. 8-10 (A) results in the following values for the components of the parallel-type network:

$$L1 = \frac{8}{2\pi \times 2{,}500} = 0.51 \text{ millihenry}$$

$$L2 = 1.6 \times L1 = 0.82 \text{ millihenry}$$

$$C1 = \frac{1}{2\pi \times 2{,}500 \times 8} = 8 \text{ microfarads}$$

$$C2 = 0.625 \times C1 = 5 \text{ microfarads}$$

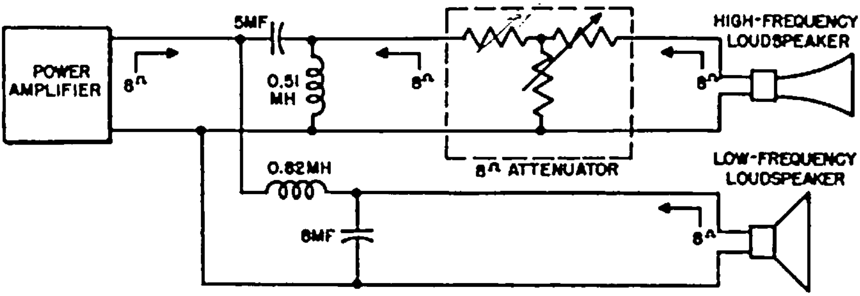


Fig. 8-11. Design of a practical frequency dividing network.

The final circuit becomes that shown in Fig. 8-11. Because the two speakers are not necessarily made by the same manufacturer and have not been designed specifically to be used together, they will have different efficiencies and will not give equal sound outputs over the two frequency ranges. A constant-resistance attenuator is inserted in the line to the more efficient speaker, and can be used to balance the responses over the two ranges, as well as for a certain degree of tone control. This attenuator may be any one of a number of commercial units, designed to have constant input impedance for any attenuation setting, and to dissipate the power which is not applied to the speaker.

Loudspeaker Enclosures and Baffles

Once the loudspeaker has been chosen the question arises as to the type of enclosure in which to mount it. The type of enclosure to be used must be given careful consideration because good results can be obtained from loudspeakers only when they are properly baffled. The rest of this chapter will describe the most widely used acceptable types of loudspeaker enclosures, and will give dimensions and constructional information that will permit the audio experimenter to construct his own baffle for whatever speaker has been selected. Home constructed rather than commercially purchased baffles have two advantages for the experimenter: (1) he can select the dimensions and size to suit any special space requirements he may have, and (2) the amount of money saved can make possible the purchase of a better loudspeaker and result in better reproduction quality.

Need for Baffle. Loudspeakers are designed for a wide frequency response, with a large cone for good low frequency reproduction and a small cone for good high frequency reproduction. However, the large size of the loudspeaker cone alone will not insure adequate low frequency reproduction, and for good low frequency performance, the loudspeaker must be mounted in a proper type of cabinet.

The reasons for the difficulty in obtaining proper baffling for loudspeakers may not be immediately apparent until it is realized that the primary purpose of the baffle is to prevent sound from the back of the speaker cone (which is 180 degrees out-of-phase with that from the front) from cancelling the sound transmitted from the front of the cone. The smaller the difference in air path compared to the wavelength of the sound, the more complete is the cancellation. If the loudspeaker is mounted along in free air, the sound from the back of the cone is 180 degrees out-of-phase with the sound from the front. At low frequencies where the wavelength of the sound is much greater than the dimensions of the loudspeaker, the sounds from the two sides of the cone tend to cancel each other. Therefore if the loudspeaker is not mounted in a baffle, or is mounted in a very small one, there will be cancellation up to a relatively high frequency and the reproduced sound will be deficient in low frequencies.

To reduce this effect, it is necessary to mount the loudspeaker in a baffle which prevents this interference. Many different types of baffles have been developed to give better low frequency reproduction from a loudspeaker, at the same time being reasonably economical and not requiring an excessive amount of space.

Plane Surface Baffle. The simplest type of baffle is obtained by mounting the loudspeaker in a very large plane surface or wall, so that the sound from the back has to travel a great distance to reach the front, and cancellation will take place only at very low frequencies which are normally not heard. For good reproduction of frequencies below 100 cps, the baffle should be approximately 8 feet square, with the loudspeaker mounted off the center. When loudspeaker response curves are given by the manufacturer they are generally measured in a large baffle of this type. This method of loudspeaker baffling is quite popular for built-in home reproduction systems, where the speakers are often mounted in a wall or a closet. It has, however, the disadvantage that it requires a considerable amount of space or a convenient wall.

Because of the large size required if flat boards are used for loudspeaker mountings, a number of different types of baffles have been developed which do not require as much space. Some types perform the additional function of increasing the low-frequency response by increasing the coupling between the loudspeaker cone and the air into which the sound is radiated. At low frequencies the area of the loudspeaker becomes insufficient for proper coupling to the air; this is one reason why small loudspeakers are not capable of the same low-frequency response as larger loudspeakers. The enclosures which increase the low-frequency response do so by increasing the area of radiation into the air at low frequencies.

Open-Back Cabinet. The most common type of mounting for loudspeakers, found in almost all commercial radio receivers and radio-phonograph combinations sold at the present time, is the unsatisfactory conventional open-back cabinet which also contains the receiver-amplifier chassis and the phonograph mechanism. When the sound path from the back of the cone is sufficiently long (as in the case of the large console cabinets) the low frequencies are reproduced, while in the midget radio cabinets the sound path from the back to the front is very short and the low frequencies are not reproduced because of the out-of-phase cancellation. However, the most objectionable acoustical feature of such cabinets is that the back of the cabinet behind the loudspeaker acts as a resonant enclosure. It is an open-ended resonant tube which accentuates the loudspeaker response at the frequency of resonance due to the increased efficiency of the acoustical system. A diagram showing the typical physical layout of such a system is shown in Fig. 8-12, together with the type of frequency response obtained. This cabinet resonance

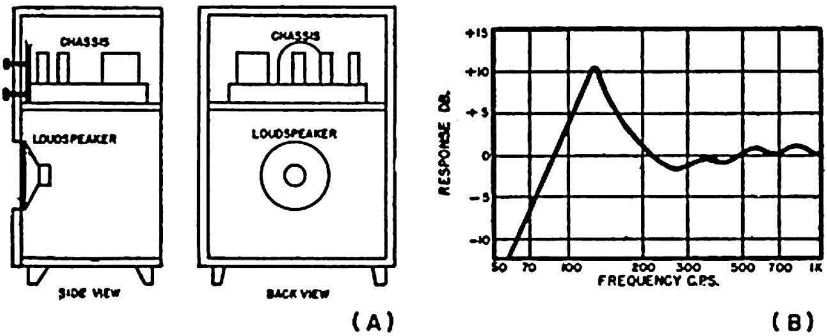


Fig. 8-12. (A) Method of mounting loudspeaker in conventional open-back cabinet. (B) Typical frequency response characteristic obtained from this type of loudspeaker mounting.

causes a sharp peak in the response, generally in the range between 100 and 200 cps, which very unfavorably affects the intelligibility and naturalness of the reproduced sound, and is especially noticeable in the reproduction of music and male speech. This is the "boomy" quality which is so characteristic of almost all commercial radio receivers.

In addition to the "boomy" quality of the sound reproduction, the open-back loudspeaker cabinet has the following further disadvantages:

(1) The loudspeaker has poor low-frequency response due to the inadequate baffle area afforded by the cabinet.

(2) There is insufficient acoustic damping of the loudspeaker diaphragm, resulting in overshooting of the moving system and consequent distortion.

(3) Because of inadequate damping of the mechanical system, there is a large variation in the electrical impedance of the loudspeaker at its primary resonant frequency, which causes poor impedance matching and additional distortion in the output stage ot some types of amplifiers.

Electrical compensation in the amplifier frequency response to correct for the cabinet resonance is not entirely satisfactory, since it cannot damp the overshoot of the mechanical system on loud signals, and does not improve the poor transient response which is characteristic of a system having such a peak in the response at one frequency. In addition, the peak in the response due to the cabinet resonance will change according to how close the back of the cabinet is placed against a wall, whether it is standing upon a bare hardwood floor or upon a soft rug, and with other such conditions of cabinet placement.

This open-back construction of the loudspeaker enclosure is used in mass-produced receivers because of its low cost and simplicity of construction, but it should be avoided in any system being set up for high-quality reproduction. The faults of the open-back loudspeaker enclosure can be eliminated by use of a properly designed loudspeaker housing which will give wide range reproduction of sound with good frequency response and without undesirable peaks. In general, proper design of a housing for best loudspeaker performance consists of incorporating acoustical networks into the cabinet to eliminate the faults of open-back cabinets and to improve the loudspeaker characteristics.

Good results are obtained by mounting the loudspeaker either in a back-enclosed cabinet, in a bass-reflex cabinet, in a labyrinth, or in a folded-horn cabinet.

The Back-Enclosed or "Infinite Baffle" Cabinet

One of the simplest types of loudspeaker cabinets is one with a completely enclosed back. By making the cabinet as rigid as possible and padding the inside with absorbent material, the sound from the back of the loudspeaker cone is completely absorbed and prevented from reaching the front. Such a cabinet is sometimes known as an "infinite baffle" cabinet, since its effect is similar to mounting the loudspeaker on an infinitely large flat board. However, the volume inside the box must be sufficiently large, or else the low-frequency response will be reduced.

The best way to understand the effect of the enclosure upon the performance of the loudspeaker is to consider the electromechanical equivalent circuits. The electromechanical equivalent circuit of the loudspeaker mounted on an infinitely large flat board is given in Fig. 8-3, and shows that the loudspeaker has the same properties as the

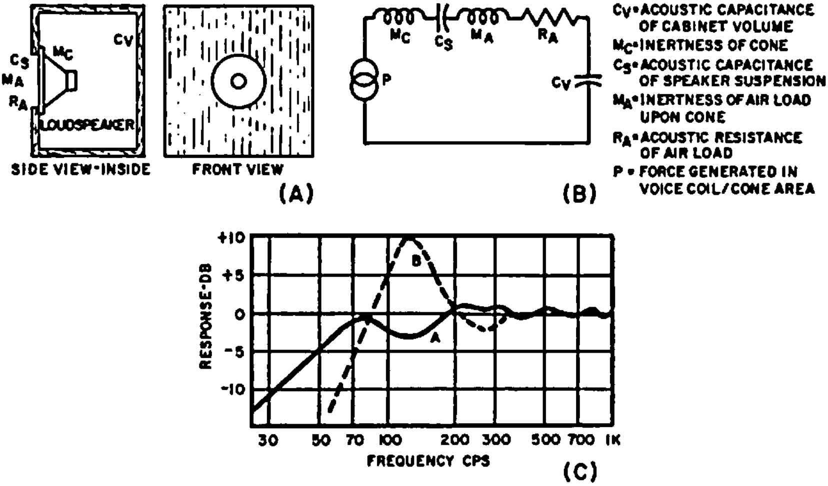
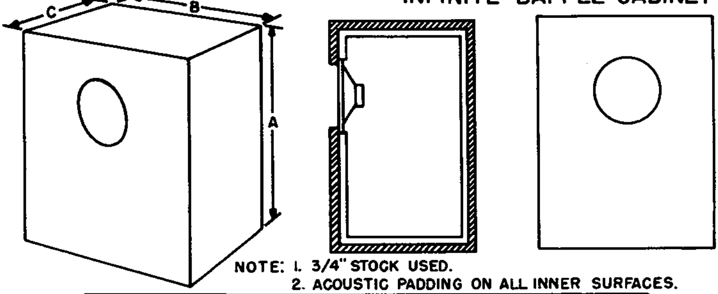


Fig. 8-13. (A) Loudspeaker mounted in totally enclosed (or infinite baffle) cabinet. (B) Equivalent electrical circuit representation of loudspeaker mounted in this type of cabinet. (C) Typical frequency response characteristic of loudspeaker in infinite baffle cabinet; curve B shows response when the back of the cabinet is removed.

series LCR circuit. It has a resonant frequency above which the sound output is independent of frequency, and falls off rapidly below resonance. When the loudspeaker is mounted in an infinite baffle closed box, the volume of the box has the effect of capacitance added in series in the circuit, as shown in Fig. 8-13. A capacitance added in series in such a circuit lowers the effective capacitance and raises the resonant frequency. The larger the capacitance added, the less will be this effect of raising the resonant frequency. Since the effect of the cabinet volume coupled to the speaker is to raise its resonant frequency, the volume should be as large as is conveniently possible. This type of cabinet is most suitable for loudspeakers whose resonant frequency is quite low.

The constructional details of infinite baffle cabinets for the various sizes of loudspeakers in common use at the present time are given in Fig. 8-14. Typical dimensions are indicated which show minimum volumes for the particular loudspeakers in question. With these dimensions, the increase in resonant frequency over that which occurs when the speaker is mounted in a large flat baffle is no more than about 10 percent. If the space requirements make other dimensions preferable, the same minimum volume should be maintained, although the volume may be decreased slightly if the consequent decrease in low frequency is not objectionable. Acoustic padding should be used on all inside surfaces as shown.

INFINITE BAFFLE CABINET



NOTE: 1. 3/4" STOCK USED.
2. ACOUSTIC PADDING ON ALL INNER SURFACES.

LOUDSPEAKER SIZE	OPENING	OVER-ALL CABINET DIMENSIONS		
		A	B	C
8"	6-1/2" diam.	31"	23-1/2"	13-1/2"
		Enclosed volume-7800 cu. in.		
10"	8-1/2" diam.	35-1/2"	26-1/2"	15"
		Enclosed volume-11500 cu. in.		
12"	10-1/2" diam.	39-1/2"	29-1/2"	16-1/2"
		Enclosed volume-16000 cu. in.		
15"	13-3/4" diam.	44-1/2"	33-1/2"	19-1/2"
		Enclosed volume-24800 cu. in.		

Fig. 8-14. Constructional details and design data for infinite baffle cabinets for various sizes of loudspeakers.

The size of the cabinet may, of course, be made as great as desired, subject only to practical limitations. The limiting case of this type of cabinet occurs when the loudspeaker is mounted in a door or wall between two different rooms, or between a room and a closet. This method has been used extensively in many installations where the room arrangement permits it. If the room or closet at the rear is large, the mounting approaches the properties of an infinite plane and no treatment of either room is needed. When the room or closet at the rear is so small that it approaches the dimensions of the cabinets as listed in the chart (i.e., where the maximum dimensions of the enclosure is less than a quarter-wavelength at the low frequencies) the walls must be lined with absorptive material, as shown for the smaller cabinets. In such cases it might be better to construct a suitable cabinet which would then be built into the room or closet.

The Bass-Reflex Cabinet

At low frequencies, the coupling between the loudspeaker and the air depends upon the size of the cone. If the cone is made large enough for effective low-frequency response then high-frequency response is reduced, and considerable expense is involved in purchasing the large low-frequency unit and an additional high-frequency unit. The bass-reflex cabinet (sometimes called an acoustic phase inverter or a vented enclosure) is a simple and effective method of increasing the coupling to the air by acting as an acoustic phase inverter, and adds the sound from the back of the cone in-phase with the sound from the front (at low frequencies).

The bass-reflex cabinet is probably the most popular and widely used of all the different loudspeaker cabinet designs. It is very simple to construct and, when properly designed, gives excellent acoustic results. Many manufacturers provide such cabinets for use with their loudspeakers, and such cabinets have been used commercially for loudspeakers ranging in size from 8-inch to the large 18-inch low-frequency units of dual systems used in theaters and auditoriums. It consists of a closed cabinet with an opening through which the volume is coupled to the air, and has the same effect as a resonant LC circuit. Best results are obtained when this resonant frequency is the same as that of the loudspeaker, and when the area of the opening is approximately that of the cone.

The basic principle of operation of the bass-reflex cabinet is the use of acoustical networks to increase the low-frequency response of the loudspeaker. The construction is shown in Fig. 8-15, and can be seen to consist of a closed cabinet with an opening in the front close to the

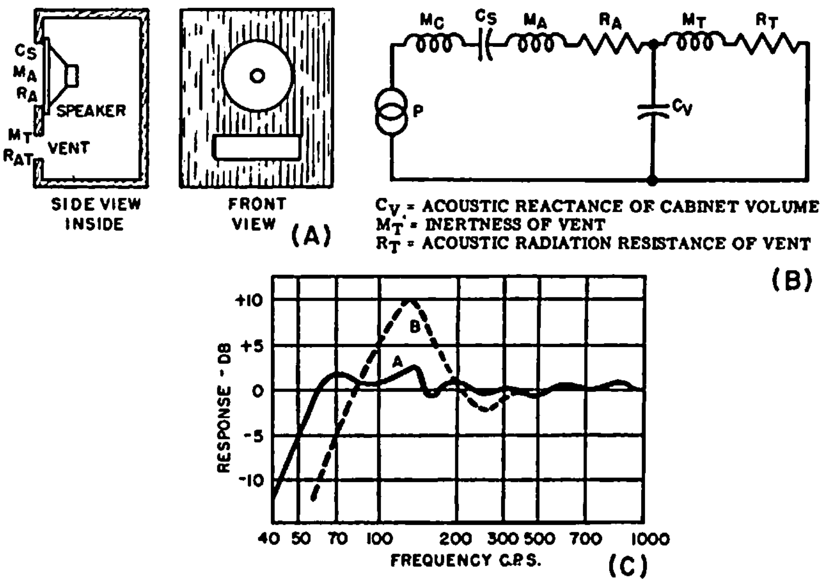


Fig. 8-15. (A) Loudspeaker mounted in bass-reflex cabinet. (B) Equivalent electrical circuit representation of loudspeaker mounted in this type of cabinet. (C) Curve A is a typical frequency response of a bass-reflex loudspeaker system; curve B is the frequency response of an open-back cabinet having the same volume.

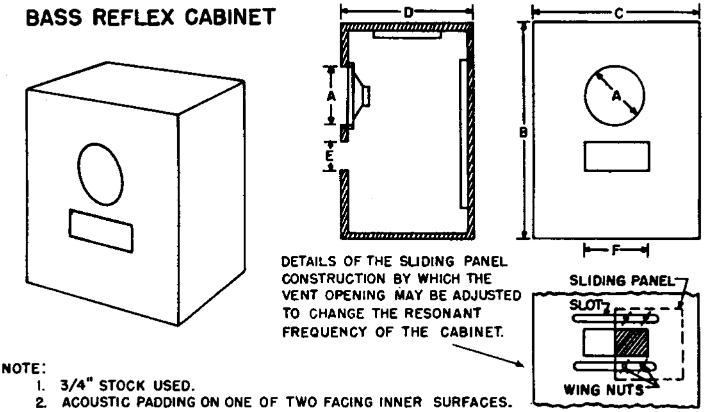
loudspeaker. The volume of the cabinet has the properties of an acoustical capacitance, while the opening in the front has the properties of an inductance in series with the acoustic resistance of the air. The effect of the bass-reflex cabinet on the response of the loudspeaker is, therefore, the same as if the system were replaced by the circuit of Fig 8-15 (B). The response at low frequencies is that of two tuned circuits coupled together, with the currents in the two resistors representing the sound radiated into the air. The low frequency response is increased by the coupling of the two tuned circuits, because the currents in the two resistors are in phase, therefore the sound from both the front and the back of the cone is useful.

However, good results are obtained from the bass-reflex cabinet only when it is properly designed to match the size and resonant frequency of the loudspeaker with which it is to be used. Improperly designed cabinets will produce undesirably boomy and resonant bass; therefore, the experimenter who constructs his own bass-reflex cabinet should be careful to use proper dimensions in his construction. The design conditions which have been found to give satisfactory results are:

(1) The resonant frequency of the vented enclosure should be approximately the same as that of the unenclosed loudspeaker.

(2) The aperture or area of the vent should approximate the effective radiating surface of the loudspeaker.

The table shown in connection with the bass-reflex information in Fig. 8-16 gives the various physical values and dimensions for the design of bass-reflex cabinets for any of the good standard high-fidelity 8-inch, 10-inch, 12-inch and 15-inch loudspeakers in general use at the present time. The data given should be suitable for 8-inch speakers with cone resonances from about 90 to 100 cps, for 10-inch speakers with cone resonances from about 70 to 80 cps, for 12-inch speakers with cone resonances from about 60 to 70 cps, and for 15-inch speakers with cone resonances from about 50 to 60 cps. Acoustical absorbing material is placed on the inside of the back wall opposite the speaker and on one of each of two opposing walls. This is done to absorb the middle and higher frequencies in the cabinet, prevent any destructive interference with the radiation from the front of the speaker, and eliminate any resonant vibrations inside the cabinet. The vent should be placed close to the loudspeaker, since mutual coupling between the two results in better sound radiation into the air. Because of normal variations in different commercial loudspeakers, the dimensions given in the chart may not be exactly optimum for the particular loudspeaker used in any individual application, and certain adjustments of these dimensions may be necessary.



LOUDSPEAKER SIZE	OPENING A	OVER-ALL CABINET DIMENSIONS B	C	D	VENT DIMENSIONS E	F
8"	6-1/2" diam.	23"	17-3/4"	10-1/4"	3"	9-1/4"
		Enclosed volume-2800 cu. in.			Area-28 sq. in.	
10"	8-1/2" diam.	28-1/2"	22"	12-1/2"	4"	12-1/2"
		Enclosed volume-5700 cu. in.			Area-50 sq. in.	
12"	10-1/2" diam.	34"	26"	14-3/4"	5-1/4"	16-1/2"
		Enclosed volume-10,000 cu. in.			Area-86 sq. in.	
15"	13-3/4" diam.	41"	30-1/2"	17-1/4"	7-1/4"	21"
		Enclosed volume-17,000 cu. in.			Area-150 sq. in.	

Fig. 8-16. Constructional details and design data for bass-reflex cabinets for various sizes of loudspeakers.

Tuning the Enclosure. The simplest method of adjusting the characteristics of the cabinet is by making the area of the vent adjustable. The constructional details of such an adjustable vent arrangement are shown in Fig. 8-16. A sliding panel held in place by wing nuts is used to tune the cabinet by changing the port opening, and the initial opening is made larger than required to permit tuning above and below the optimum frequency. The adjustment of the opening can then be done by ear until the cabinet has been matched to the loudspeaker and to the room to give the best overall reproduction quality.

A more exact method of tuning can be accomplished as follows: an audio oscillator is connected through a 100-ohm resistor to the mounted loudspeaker. A low-range a-c voltmeter (about 2½ volts) is then connected across the voice coil of the speaker. As the audio oscillator is adjusted through a range of low frequencies (say 20 to 200 cps), the voltage reading will vary in direct relation to the impedance of the mounted loudspeaker and its enclosure. If the volume and port area of the enclosure are correct for the particular speaker used, then *two equal-amplitude voltage peaks* (also impedance peaks) will occur which are equally spaced above and below the resonance peak of the loudspeaker when it is unmounted. If the enclosure frequency is much too high or too low, then two large peaks also occur, but one of these is at the resonant frequency of the unmounted speaker while the other is at the frequency of the enclosure. If the enclosure frequency is only slightly too high or too low, two peaks occur, equally spaced above and below the speaker's resonant frequency, but one peak is considerably larger than the other. The enclosure frequency should then be adjusted to equalize the peaks. Finally, when the enclosure has been properly tuned, it is advisable to damp the part by stretching one or more thicknesses of burlap or heavy grill cloth across the opening so that the amplitude of the resonant peaks are reduced. When this has been done, a properly tuned and damped bass-reflex enclosure is obtained.

The bass-reflex cabinet is one of the most satisfactory types of loudspeaker enclosures for home construction because it is quite simple to construct and lends itself easily to modification to compensate for variations in speakers and in listening conditions.

The Labyrinth Cabinet

Another type of resonant phase-inverter cabinet which makes the radiated sound from the back of the loudspeaker useful at low frequencies is the acoustical labyrinth.[1] In this type of enclosure the acoustic tuned

[1] "Acoustical Labyrinth" is a registered trade-mark of Stromberg-Carlson Co.

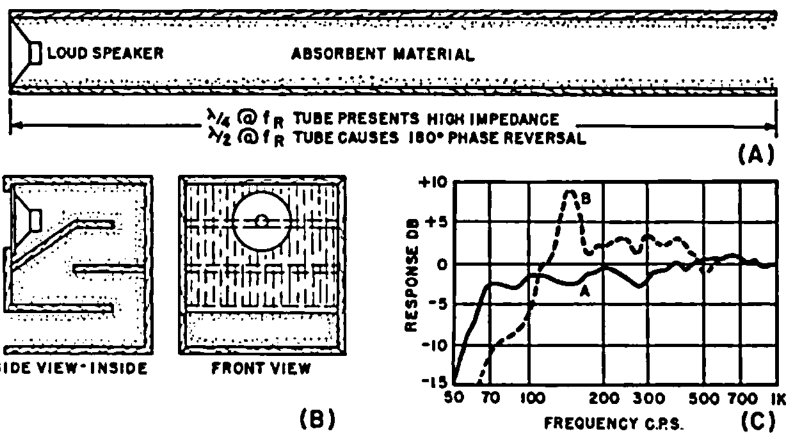
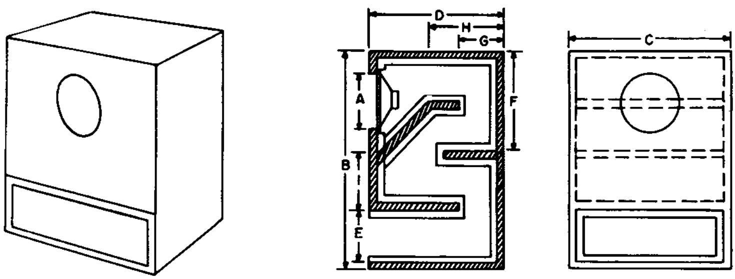


Fig. 8-17. (A) Method of coupling the back of the loudspeaker cone to the air through a resonant tube. (B) Labyrinth cabinet formed by folding the resonant tube. (C) Curve A is typical frequency response of a labyrinth loudspeaker system; curve B is the frequency response of the corresponding open-back cabinet system for comparison.

circuit of the bass-reflex cabinet is replaced by a resonant line. An absorbent-walled tube is coupled to the back of the loudspeaker at one end, and is open to the air at the other end, as shown in Fig. 8-17. At the frequency for which it is one-quarter wavelength long, this tube sees a low acoustic impedance at the open end, and therefore presents a high impedance to the back of the loudspeaker cone. Thus, by choosing the length of the tube so that it is a quarter-wavelength at the resonant frequency of the loudspeaker suspension, the resonance of the speaker is damped. At double the resonant frequency, the tube is a half-wavelength long and the phase is reversed, therefore the sound through the tube is in phase with that from the front of the loudspeaker and the response is increased. The tube lining absorbs almost all of the sound above 150 cps, therefore the higher resonances have no effect.

In the labyrinth cabinet, this resonant tube is folded so that the total outside dimensions are practical for use in the home or studio. The practical values and physical dimensions for construction of typical labyrinth cabinets for the various commercial loudspeakers are given in Fig. 8-18. For the same loudspeaker, the labyrinth occupies less space than the bass-reflex cabinet, while the bass-reflex has the advantages of being

LABYRINTH CABINET



LOUDSPEAKER SIZE	SPEAKER OPENING	OVER-ALL CABINET DIMENSIONS			PARTITION POSITIONING					MATERIAL
	A	B	C	D	E	F	G	H	I	
8"	6-1/2" diam.	17"	14"	11-1/2"	3"	9-1/2"	2-1/2"	5"	3-3/4"	1/2" plywood 1/2" felt padding
10"	8-1/2" diam.	21-3/4"	17"	14"	4-1/2"	10-1/2"	4"	5"	3"	3/4" plywood 3/4" felt padding
12"	10-1/2" diam.	27-3/4"	21"	16-3/4"	6"	13-1/2"	5-3/4"	6-1/4"	5"	3/4" plywood 1" felt padding
15"	13-3/4" diam.	35"	25"	21"	7"	18"	6-1/2"	7-1/2"	4"	3/4" plywood 1" felt padding

NOTE: ACOUSTIC PADDING ON ALL INNER SURFACES

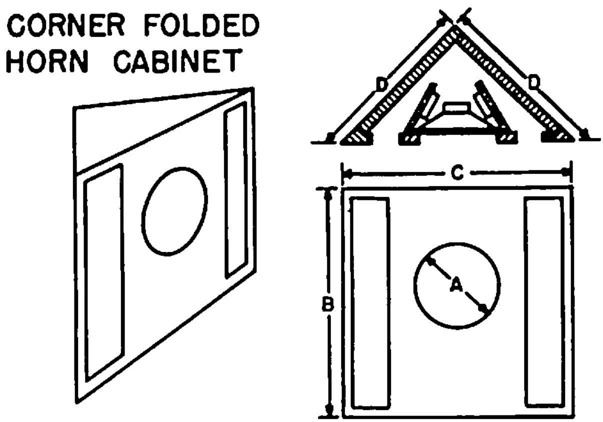
Fig. 8-18. Constructional details and design data for labyrinth cabinets for various sizes of loudspeakers.

simpler and less expensive to construct, and can be tuned over a certain range by adjusting the vent opening.

The Folded-Horn Cabinet

A type of loudspeaker cabinet which is becoming widely used because of its good low-frequency response is the folded-horn cabinet. In this type of cabinet the sound is radiated from the front of the speaker cone at high frequencies, and through a horn coupled to the back of the cone at low frequencies. For home use, it is generally designed to be placed in a corner of the room so that the walls and floor form part of the horn. In some well designed horns, frequencies as low as 20 to 30 cps can be reproduced using standard commercial speakers in cabinets of practical sizes.

The horn is used in loudspeaker applications because it is the acoustical equivalent of the electrical transformer. Since at low frequencies, the air represents too low an impedance for proper coupling to the loudspeaker cone, the horn can be used to transform this low impedance to a higher impedance which permits more efficient energy transfer. Generally a volume of air is maintained between the loudspeaker and the entrance to the horn, to act as an acoustic capacitance which bypasses the horn at higher frequencies. Thus all the high-frequency sound radiation is from the front of the speaker.



LOUDSPEAKER SIZE	SPEAKER OPENING	CABINET DIMENSIONS		
	A	B	C	D
12"	10-1/2" diam.	32"	32"	22-1/2"
15"	13-3/4" diam.	32"	36"	24-1/2"

Fig. 8-19. Constructional details and design data for simple corner folded horn cabinet for various sizes of loudspeakers.

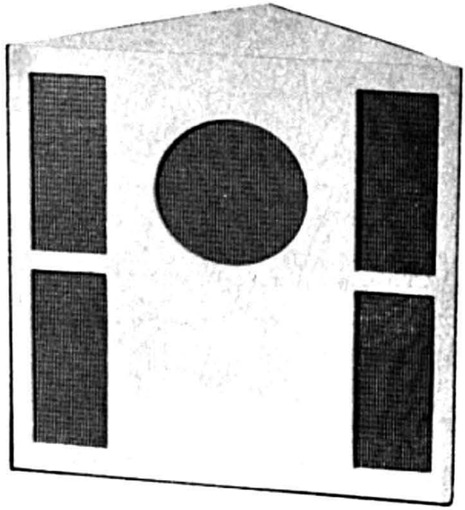


Fig. 8-20. Corner folded horn loudspeaker cabinet assembled from commercial home-contruction kit.

Courtesy: G. and H. Wood Products Co.

The details of construction of one of the very simplest corner folded-horn cabinets are shown in Fig. 8-19, while a photograph of such a cabinet assembled from a home-construction kit appears in Fig. 8-20. Practical dimensions are given for the construction of such cabinets for use with commercial 12-inch and 15-inch loudspeakers. No dimensions are given for use with smaller speakers because speakers capable of producing the very low frequencies give the horn its major usefulness.

General and Constructional Details

Generally, the quality of the loudspeaker and the type of cabinet used with it in any specific system will be determined by economic considerations, since a loudspeaker of the best quality can be the most expensive single component in the system. When a back-enclosed or a bass-reflex cabinet are to be used, these may either be purchased commercially or, for home construction, information given previously in this chapter may be used. In addition, information is generally available from the loudspeaker manufacturers for the best cabinet dimensions to be used with their particular speakers.

In the construction of the various loudspeaker cabinets which have been described, a number of precautions must be taken to obtain proper performance:

(1) In assembling cabinets, all mating joints should be securely glued and screwed together. Cracks or holes should be filled with plastic wood. All attempts should be made to make air-tight joints.

(2) Large surfaces of the cabinet should be stiffened on the inside to prevent low-frequency vibrations. Stiffening braces should be fastened to such surfaces to eliminate any low-frequency resonances. When surfaces are tapped, only highly damped high-frequency vibrations should result, and various sections should have different resonant frequencies.

(3) Interior of the cabinet should be well padded to prevent standing waves from being set up. Absorbing material should be placed directly behind the loudspeaker, and at least one of each two opposing surfaces should be covered over most of its area with absorbing material.

(4) Grille cloth should be as light weight and porous as possible for minimum loss of high frequencies.

Some typical sound insulating materials which may be used inside loudspeaker cabinets for sound absorption are: rock wool, Kimsul insulation, Cellufoam, Fiberglass padding or tile, Acousti-Celotex, Fiberglass, type FP-OC9, and Tufflex.

For the experimenter who does not have extensive woodworking facilities at his disposal, some of the cabinet types which have been described are available in commercial kits whose dimensions are very similar to those given in the tables.

The cabinet dimensions which are given in the tables may be changed to suit individual space requirements, provided certain precautions are taken: in the infinite baffle cabinet, the total internal volume should not be decreased. In the bass-reflex cabinet the overall dimensions may be changed, but the internal volume and the vent area must be kept the same. In the labyrinth, the cross-section and length of the resonant tube must be kept the same. In the corner folded horn, the cross-section of the horn must be maintained.

If the loudspeaker is placed in the proper cabinet and constructed according to the information given in this chapter, then the good sound quality which the loudspeaker is capable of producing will be obtained. There will be no resonances in the frequency response and the cone will be properly damped for best transient response. With a good electronic system, such a loudspeaker system will be capable of a naturalness and clarity of reproduction which could otherwise be attained only by expensive commercial systems.

Chapter 9

COMPLETE HIGH FIDELITY SYSTEMS

General

The basic requirements and typical setups of sound reproduction systems have been discussed in general terms in Chapter 3. Other previous chapters have discussed in detail the various components and different sections of the sound reproduction system, and described their design principles and method of operation. This chapter will describe in detail the manner in which they are combined into complete systems for recording and reproducing sound.

Complete Audio Amplifiers

In the previous chapters, the various circuits and individual units which make up the complete audio amplifier and electronic channel have been described in detail, and many practical circuits have been described. The next few sections of this chapter will show how these circuits are designed for practical sound reproduction systems, how they are combined to form the various major component units in the system, how they may be designed and constructed to meet the requirements of an individual installation and integrated into the complete system, and their relationship with the other components of the system. The circuits of a number of complete audio amplifiers will be described in detail in order to illustrate the application of the principles which have already been described to the design of the complete system. These will include single-amplifier units as well as systems integrated by separate control panels. The experimenter may build his own system from these designs or may be guided by them in the purchase of commercial units.

The major component of the audio reproduction system which permits the widest versatility in setup and choice of components, is the electronic channel. The electronic channel does each of the following: (1) accepts the electrical audio-frequency signal from whatever type of transducer or input device is being used for the reproduction or transmission, (2) amplifies this sufficiently, (3) selects and applies the desired amount of frequency correction or equalization, (4) selects the desired input signal for reproduction, (5) operates on the signal in whatever manner is desired to improve it by reducing noise or by increasing or decreasing the dynamic range and, (6) supplies sufficient power to produce the required sound intensity from the output of the loudspeaker. The electronic channel in this sense does not include other electronic units which may be used in the system as transducers, such as radio tuners and tape playback machines which may have electronic speed controls. It may be either a single unit which performs all the required functions, or a number of different units which may be in several different locations and integrated by one or more control units. In general, all of the various types of electronic channels which are in current use at the present time are made up of circuits which have been described in the previous chapters.

Typical Basic Unit

A typical unit which contains in one chassis all the functions required in a basic sound reproduction system is shown in the schematic circuit diagram of Fig. 9-1. It includes a preamplifier for magnetic pickups, an input signal selector switch, volume and tone controls, and a

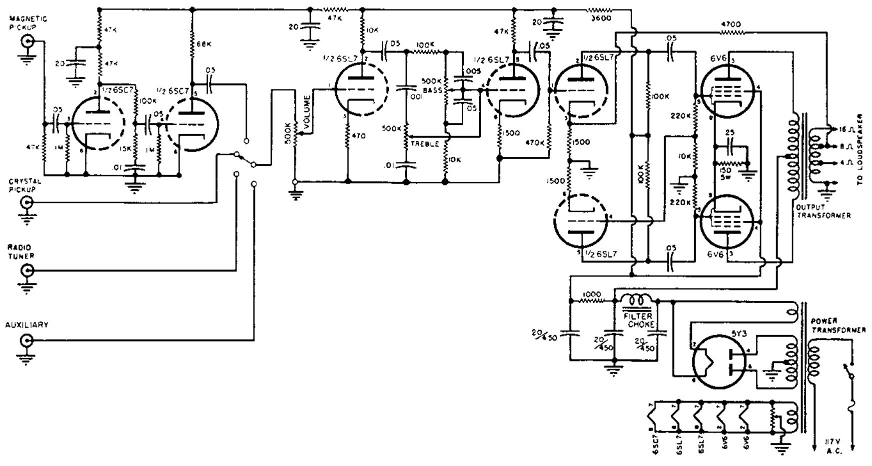


Fig. 9-1. Schematic of a typical single-unit amplifier which contains, in one chassis, all the electronic functions required in a basic sound reproducing system.

voltage and power amplifier. The various individual circuits used in this unit have all been described in previous chapters; this amplifier unit offers an illustration of practical audio design based on fundamental circuits.

Circuit Description. The circuit of the low-level preamplifier for magnetic phonograph pickups is similar to the basic equalizer preamplifier described in Chapter 6. It consists of a two-stage dual-triode amplifier which provides the necessary low-frequency equalization and amplification for the pickup output. Other input signals may be obtained from a crystal pickup instead of a magnetic pickup, or from a radio tuner.

The desired input signal to the amplifier is selected from either the radio tuner or the phonograph pickup signal by means of a four-position selector switch. This selected input signal is amplified by a single triode stage with a gain of approximately 18 db, an output impedance of 9,000 ohms, and 4 db of negative current feedback due to the unbypassed cathode resistor. The level of the output signal from this first voltage amplifier stage is sufficiently high so that it can be applied to an adjustable tone-control circuit which has a 20 db insertion loss. This point in the circuit is the most desirable point for the location of the tone control because the output level from the preceding stage is adequate, and its output impedance is low for best matching to the tone control circuit.

The tone control circuit is the one described in Fig. 7-8, capable of giving up to about 15 db bass and treble boost or attenuation. This network has a 20 db insertion loss, therefore another triode amplifier stage is needed to amplify the signal to feed the driver stage.

The driver stage is the same as the circuit described in Chapter 5, consisting of a simple phase-inverter amplifier circuit which supplies the necessary balanced push-pull signal with a voltage gain of 29 db. The power amplifier consists of two 6V6 tubes in a push-pull circuit which is capable of delivering approximately 10 watts maximum output power. The output transformer, which is used to couple the plate of the power amplifier tubes to the loudspeaker voice coil, may be any one of a number of good commercial transformers designed to be used with these tubes.

Negative feedback is used from the secondary of the output transformer to the cathode of the driver amplifier, thus it includes the entire driver section and power amplifier in the feedback loop. The selected values give approximately 10 db of negative feedback, thereby reducing the voltage gain from the driver grid to the speaker voice coil from about 8 without feedback to about 3 with feedback. The use of this feedback decreases the distortion in the power amplifier, improves the frequency response, reduces the effects of tube changes and aging, and lowers the output impedance to improve the transient response of the loudspeaker.

It should be noted that the negative feedback does not reduce the maximum output power of the amplifier, but merely reduces the voltage gain of the section included in the feedback loop. It also cannot increase the maximum power of the amplifier, since when the tubes are overloaded, thus introducing serious distortion, no amount of feedback can affect the overload limit of the tubes. However, below the overload point the characteristics are considerably improved.

The feedback loop is taken around only the driver and power amplifier stages; it is not necessary nor particularly desirable to include the earlier stages in the loop. A feedback loop certainly cannot include a tone control network of the type shown, since the use of the feedback would tend to reduce the desired frequency-response equalization introduced by the tone control. (In some circuits the feedback itself is used to give the equalization, but this type of feedback equalization is a different application from that being described.) The low-level stages are not as subject to distortion as the high-level stages because the tube characteristics are more linear for small signals than for large. However, cathode degeneration is used in each stage to give about 5 db to 7 db of negative feedback to improve distortion and frequency response characteristics. The negative feedback in the cathode of the phase-inverter amplifier also serves to maintain proper balance in the push-pull signal and reduces the effects of tube changes and aging on the balanced output to the power amplifier grids.

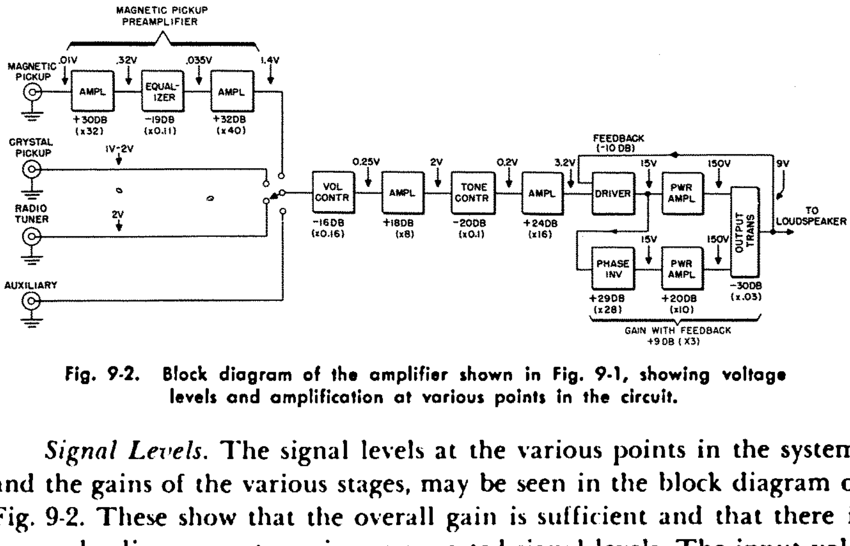


Fig. 9-2. Block diagram of the amplifier shown in Fig. 9-1, showing voltage levels and amplification at various points in the circuit.

Signal Levels. The signal levels at the various points in the system, and the gains of the various stages, may be seen in the block diagram of Fig. 9-2. These show that the overall gain is sufficient and that there is no overloading even at maximum expected signal levels. The input voltage to the magnetic pickup preamplifier is generally of the order of 0.01 volt (rms) at maximum signal. The first preamplifier stage has a gain of 30 db, the equalizer network a loss at middle and high frequencies of 19 db, and the second stage a gain of 32 db. Therefore, the total gain of the preamplifier is 43 db. The output voltage of the preamplifier is thus about 1.4 volts (rms) at maximum signal level. Since the signal levels from crystal pickups and radio tuners are also from 1 volt to 2 volts at maximum signal, all the inputs to the mixer are at about the same level, therefore the selector switch followed by an 0.5-meg volume control may be used for selecting the desired input signal to be amplified.

The volume control should be set so that there will be a considerable amount of reserve amplification for exceptionally weak signals, and can be set for as much as 16 db to 20 db attenuation or more, depending upon the desired sound level. The input signal to the voltage amplifier section will be of the order of 0.25 volt (with an input of a little over 1.5 volts), and is amplified by the first stage (gain of 18 db) so that 2 volts is applied to the input of the tone control circuit. The 20 db insertion loss in the tone control circuit reduces this signal level to 0.2 volt, and the following stage amplifies this to approximately 3 volts input to the phase inverter driver amplifier. The negative feedback around the driver and power amplifier results in an overall voltage amplification of 3 from this point to the loudspeaker voice coil, therefore the 3-volt signal at this point results in a 9-volt signal across the 8-ohm loudspeaker voice coil — or a power output of 10 watts. At this output level, which is the maximum that can be obtained from the output tubes, all the lower level tubes are well below their overload voltages. Because of this, overloading in the amplifier is determined only by the output of the power amplifier stage.

Fig. 9-3. Typical commercial amplifier whose circuit is fairly similar to that shown in Fig. 9-1.

Courtesy: Mark Simpson Mfg. Co., Inc.

This amplifier gives quite good sound reproduction quality, and its measured characteristics are well within the generally recognized limits for acceptable reproduction. Frequency response, harmonic and intermodulation distortion, noise level, transient response, etc., are all better than the minimum requirements as listed in Table 2-I.

The unit which has just been described is a basic standard design illustrating, in a practical unit, many of the principles and procedures which have been described in earlier chapters. Many of the present commercially available medium cost audio amplifiers (see Fig. 9-3) are similar in basic design to this circuit; it is a very practical one well suited to home construction.

Basic System with Separate Preamplifier/Control Unit

The unit which has just been described is a basic standard unit which will give excellent performance in a system where there are no problems of physical location and no further requirements of flexibility. However, in many sound reproduction system installations there are problems of space and physical location which cannot be met by a single-unit electronic system of this type. In many installations there is little space available for the electronic units; in others the convenience of remotely located control panels may be desired. These requirements can be met by an electronic system in which the control panel and switching functions are performed in a small self-contained unit, which can be placed in any convenient location and connected by long wires to the other components in the system.

A properly designed unit for this purpose can serve as a preamplifier and control unit for the integration of the complete sound reproducing system, whether it is completely custom-built or assembled from commercially purchased units. The amplifier system which will be described in this section consists of a preamplifier/control unit of this type, and a high-level amplifier which will supply sufficient amplification and output power from the output of this unit to drive the loudspeaker to the required sound level.

The Preamplifier/Control Unit. The circuit of a completely self-contained preamplifier and control unit designed for this system is shown in Fig. 9-4. This circuit is essentially based upon the circuit of Fig. 9-1, with a number of improvements and modifications to make it more flexible and useful in a wider variety of installations. It contains provision for input signals from one or more of the following: (1) magnetic phonograph pickup, (2) radio receiver or crystal phonograph pickup, and (3) crystal microphone. The level of each signal is adjusted by an individual volume control (if desired), and the desired signal is selected by a switch.

The control and mixing unit includes the preamplifier for magnetic pickups, this can be done because the input impedance from the pickup is sufficiently low so that even if the record player is at a considerable distance from the unit there will be very little hum pickup in the long lead. (Generally, for convenience of operation, the record player and radio tuner will be placed close to the control panel for convenient adjustment of controls when records are being played or the tuner adjusted.) The preamplifier circuit is similar to the one used in the single-chassis unit, except that there is additional switching of different capacitors into the record crossover equalizer to compensate for different crossover frequencies of 300 cps, 500 cps, and 800 cps.

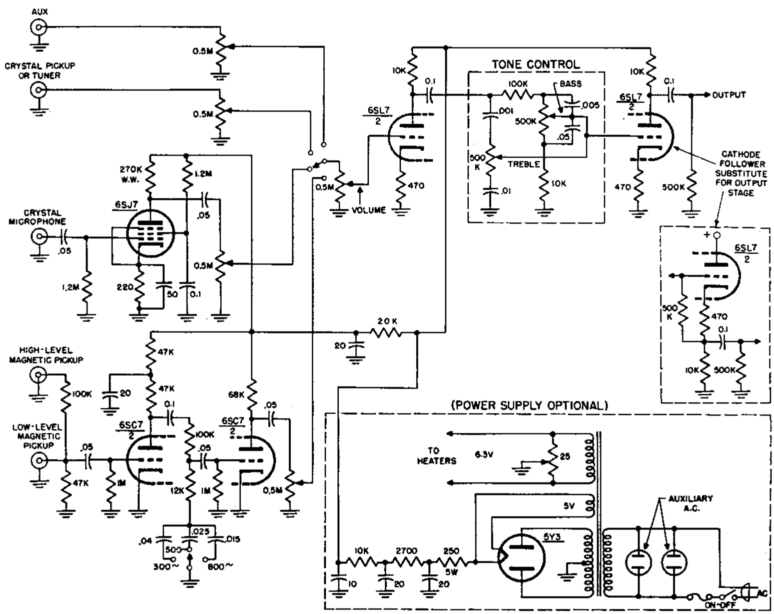


Fig. 9-4. Schematic of a typical preamplifier/control unit.

The input signal from radio tuner or crystal phonograph pickup is applied directly, without amplification, to the input selector switch since this signal level is relatively high – in the neighborhood of 1 volt peak. The microphone input channel makes provision for a high-impedance crystal microphone (with an output of approximately 0.02 volt peak), since this is a common and inexpensive type of microphone for amateur sound reproduction which gives good results. The preamplifier for this high-impedance microphone channel is a 6SJ7 pentode (although a 1620 or 6J7 may be substituted to give a little better signal-to-noise ratio) giving a voltage gain of approximately 40 db, to match the signal level in this channel to the level of the crystal phonograph and radio tuner channel.

As shown in the circuit (Fig. 9-4) the different input signals are selected by a switch to which the inputs are connected directly, since they are all at approximately the same level. If there is any appreciable difference in level, then the inputs may be taken to individual volume controls (as shown) before switching, and all levels set to match the lowest so that there will be no change in level when the different inputs are switched. If it is desirable to reproduce different inputs simultaneously, then the switch cannot be used and the various inputs must be combined in a resistive mixer so that the selection of the desired reproduced signals are made by settings of the individual volume controls.

The first voltage amplifier is one section of a 6SL7. This gives sufficient gain and output voltage to compensate for the 20 db insertion loss of the tone control circuit which follows it, and has a sufficiently low output impedance. The tone control circuit is the same one shown in Chapter 7 and used in the circuit of Fig. 9-1. Its output is at a high impedance, therefore it cannot be connected by a long lead to the main amplifier. This difficulty can be overcome by using the second section of the 6SL7 as a cathode-follower, to furnish a low-impedance output to the line so that there will be negligible noise pickup or reduction of frequency range. The cathode-follower circuit shown in Fig. 9-4 has an output impedance of 375 ohms, which is quite appropriate for this application. If more amplification is required, then the output stage may be connected as an amplifier instead of a cathode-follower. This output amplifier stage gives an additional gain of about 20 db with an output impedance of the order of about 6,000 ohms; therefore it would permit the connection of 20 to 30 feet of shielded lead across its output terminals without appreciable loss of high frequency response.

If the preamplifier unit is to be mounted in a cabinet or bookcase which also contains the main amplifier, so that the interconnecting wires are sufficiently protected and inaccessible, the plate voltage and other power may be taken from the power supply of the main amplifier. However, if it is to be mounted in some exposed location, such as an end table or a cocktail table, safety considerations would make it undesirable to have long, exposed leads carrying plate voltage from the main amplifier to the preamplifier. In such applications it is preferable to include a self-contained power supply in the preamplifier unit. The circuit diagram in Fig. 9-4 shows a self-contained power supply which may be omitted if it is not needed, depending upon the particular application. If power is taken from the main amplifier, the amount of current required by the preamplifier is so small that there will be very little effect on the power supply and no allowances need be made for it. However, filtering should be used in the B+ voltage supply, as shown in the circuit, to provide isolation between the various plate circuits, and for additional hum filtering in the power supply. (If a reasonably well filtered voltage is available, only one additional section of filtering is necessary, otherwise two may be used.) In this unit, filaments are returned to ground through the variable hum potentiometer, which is adjusted for minimum hum in the output.

The High-Level Amplifier. The preamplifier/control unit may be used with any amplifier which is capable of delivering full output power from the appropriate input. When the preamplifier is used with the cathode-follower output, the amplifier must deliver full output with approximately 0.1 to 0.2 volt input levels. If the amplifier requires inputs of the order of 1 volt, then the preamplifier unit must have an amplifier output stage in place of the cathode-follower; it will then be capable of delivering an output signal of the order of 2 volts or more.

The amplifier with which this preamplifier unit may be used is essentially the same as the high-level sections of the basic unit described in the previous section. The first stage is a 6J5 triode amplifier which has a gain of 19 db (including 4 db of cathode degeneration) to compensate for the insertion loss of the tone control circuit. The rest of the circuit, shown in Fig. 9-5, is exactly the same as in the circuit of Fig. 9-2. Since the circuits are essentially the same, the performance of this system will be approximately the same as that of the basic unit described in the previous section. Any improvements in performance would be a result of the changes in the preamplifier unit (giving the greater convenience re-

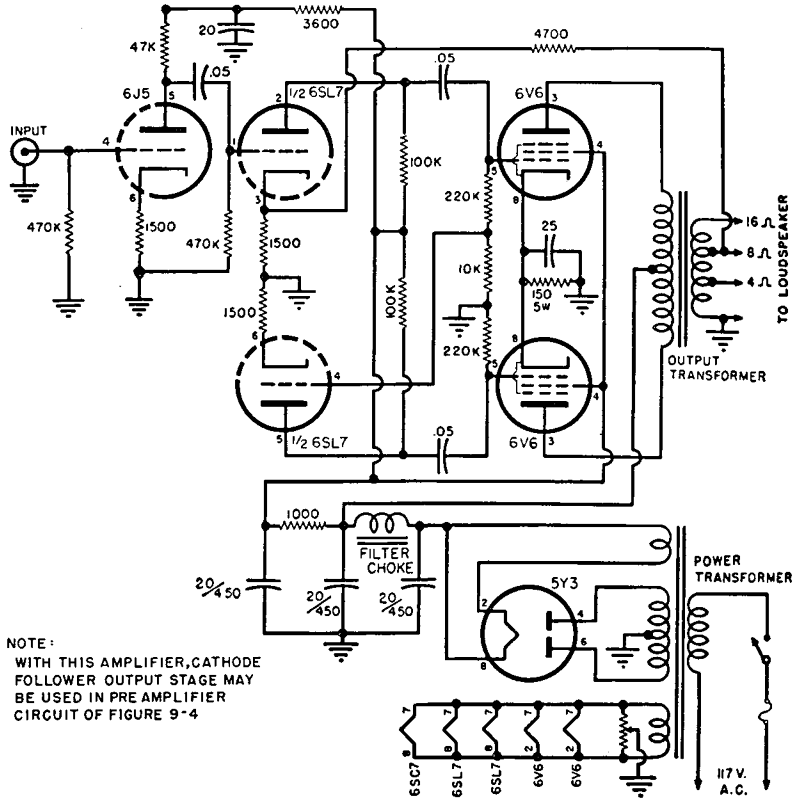


Fig. 9-5. Circuit diagram of a typical high-level amplifier which may be used with the preamplifier/control unit of Fig. 9-4.

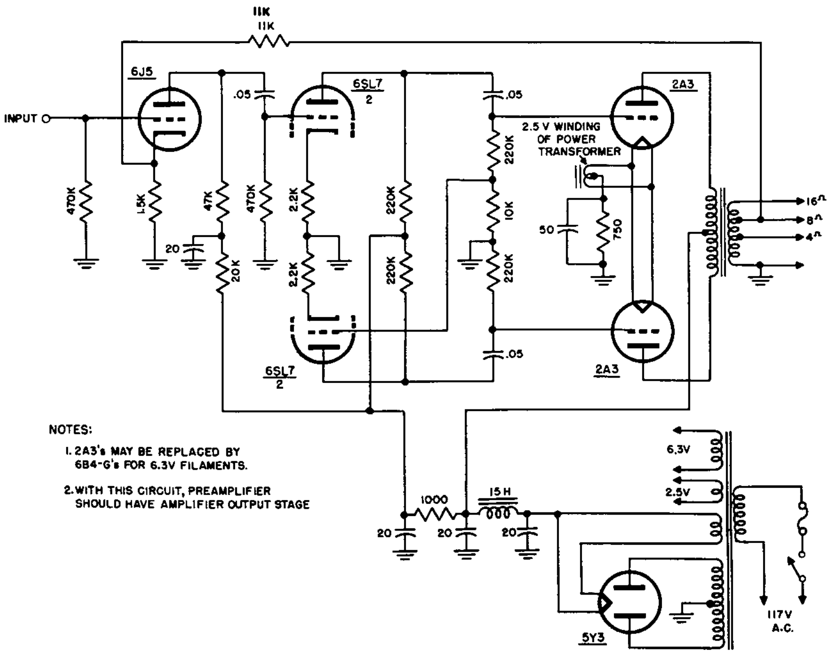


Fig. 9-6. Triode amplifier which may be used with the preamplifier/control unit of Fig. 9-4.

sulting from remote control of the system), and the addition of the adjustable crossover frequency for more accurate low-frequency compensation in record reproduction.

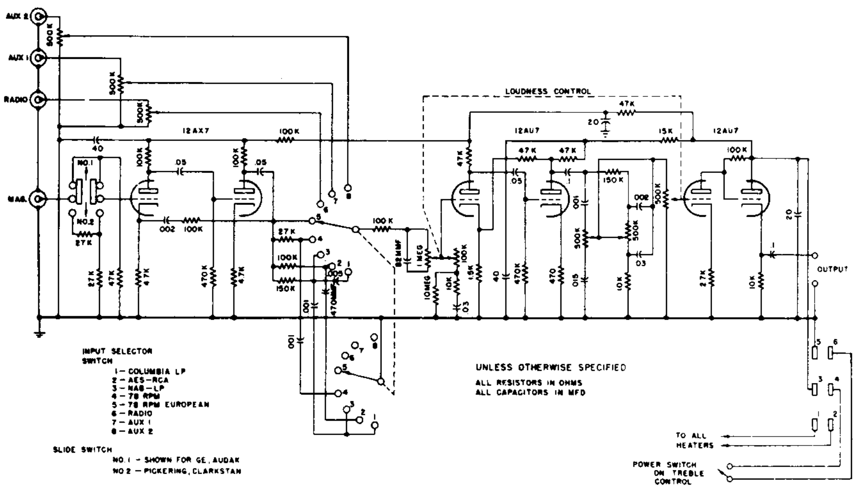
Another high-level amplifier circuit which may be used with this preamplifier/control unit is the all-triode amplifier shown in Fig. 9-6. This is essentially the power amplifier circuit shown in Chapter 6. It consists of one stage of voltage amplification, a phase-inverter amplifier, and 2A3's in push-pull as the power amplifier, with approximately 20 db of negative feedback taken from the secondary of the output transformer to the first amplifier stage. This circuit delivers a maximum of 10 watts of output power, and requires an input voltage of approximately 1.2 volts (rms) for full output, therefore it can easily be used with the preamplifier unit when the amplifier output stage is used instead of the cathode-follower.

High Quality High Cost System

The two systems which have been described are basic systems which will give very satisfactory reproduction for average home installations. The measured characteristics will be within the recognized limits for acceptable reproduction, but there will still be present certain amounts of the various forms of distortion. The finest sound reproduction systems are those in which the distortions have been reduced to the lowest amounts possible with the present development of electronic engineering, until distortions are, to all practical purposes, negligible. Such systems are, of course, more expensive because a considerable amount of care and effort must go into the manufacture of high quality components, and considerable attention must be paid to the design and setup of the electronic channel as well as to the selection of the other components of the system. This section will describe the type of circuit design which is used for such a high quality system, and will indicate the various units and functions which may be included.

This system is based on the Williamson driver and power amplifier circuit, which a separate preamplifier/control unit which may also, if desired, contain a dynamic noise suppressor for use in phonograph reproduction.

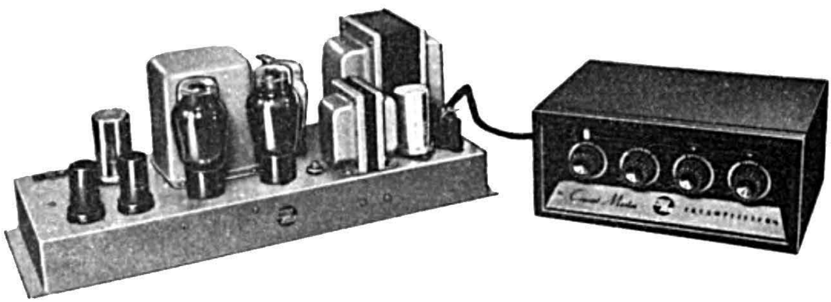
The preamplifier may be made as flexible as desired, and may contain a number of different equalization and compensation characteristics to be switched into the circuit whenever they are needed. Since the use of many of these compensations is a matter of individual preference, and can readily be included into any circuit, the preamplifier unit is shown with a certain number of basic controls so that any additional characteristics may be added at the choice of the individual experimenter.



Courtesy: Mark Simpson Mfg. Co., Inc.

Fig. 9-7. Circuit diagram of a typical preamplifier/control unit which is suitable for use in high-quality systems.

Preamplifier/Control Units. A basic high quality preamplifier would include the same functions as in the unit of Fig. 9-4, but with the required circuit modifications which are made necessary by the more flexible characteristics and greater number of functions required of the unit. The phonograph input is assumed to be from a magnetic pickup, and is amplified in a two-stage preamplifier, as shown in Fig. 9-7. Several different record equalization curves are selected by a switch to match the characteristics of the various types of records which may be played. (The preamplifier also includes volume controls so that the output level for the particular type of pickup used may be adjusted to match the input level of the various other inputs to the system.) Any one of four input positions is selected by an input selector switch so that a record player, an a-m/f-m or separate a-m and f-m tuners, and a television sound channel, may be applied to the input of the system. This unit also contains a loudness control, in place of the standard potentiometer volume control, to avoid changes in tonal balance at low listening levels. A photograph of this unit connected to a high-level amplifier is shown in Fig. 9-8.



Courtesy: Mark Simpson Mfg. Co., Inc.
Fig. 9-8. Commercial high-quality preamplifier/control unit whose circuit is shown in Fig. 9-7 connected to a high-level amplifier.

A further modification of the preamplifier unit may include a dynamic noise suppressor for use with the magnetic phonograph pickup. The circuit diagram of a dynamic noise-suppressor preamplifier which may be used in place of the standard preamplifier for magnetic pickups is shown in Fig. 9-9. This is based upon the circuit of a standard commercial unit which may be purchased and included into the system. (See Fig. 9-10.) The input signal is amplified in a two-stage 6SC7 preamplifier which contains a volume control and a crossover frequency selector switch to select correct equalization for either 300 cps, 500 cps, or 800 cps. From the output of this preamplifier section, the signal is then passed through the variable, dynamic noise-suppressor low-pass and high-pass filters of the type described in Chapter 7. The output of this noise-suppressor pre-

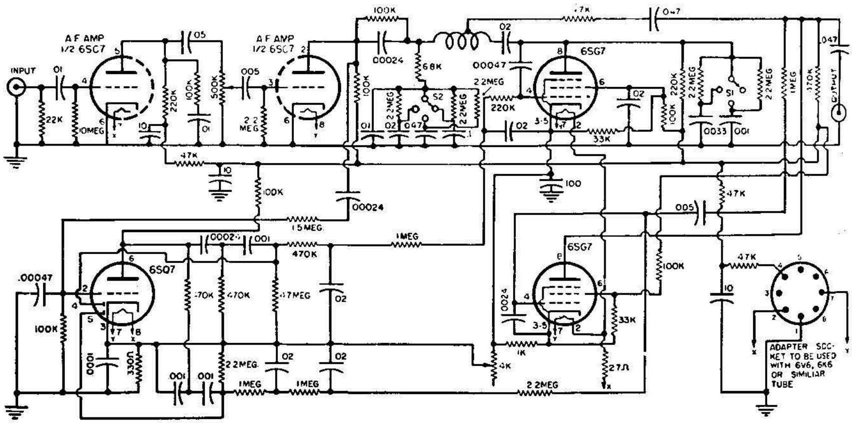
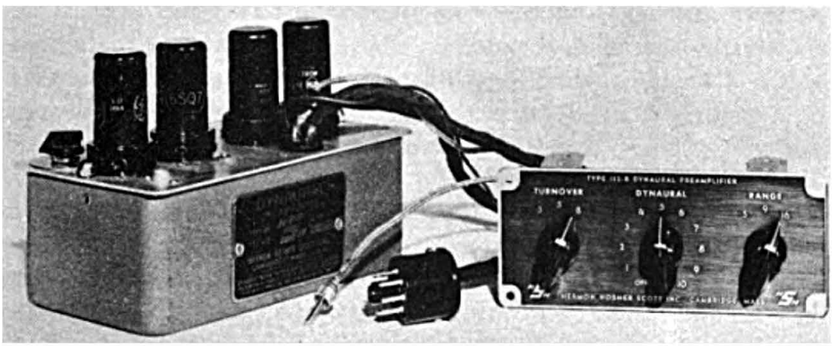


Fig. 9-9. Circuit diagram of a dynamic noise suppressor for use with magnetic pickups.

amplifier unit is at approximately the same level as the various other inputs, so that it may be connected to one position on the input selector switch. The rest of the circuit of the preamplifier unit is essentially the same as for the unit shown in Fig. 9-7. It may obtain its power either from a self-contained power supply or from the main amplifier power supply according to the requirements of the individual installation.



Courtesy: H. H. Scott, Inc.

Fig. 9-10. Commercial dynamic noise suppressor preamplifier for use with magnetic pickups.

High-Level Amplifier. The circuit of a Williamson driver and power amplifier which is suitable for use in the high quality sound reproduction system, based on the circuit discussed in Chapter 6, is shown in the schematic of Fig. 9-11. The first voltage amplifier stage is one section of a 6SN7, and is direct-coupled to a cathode-follower phase inverter using the second section of the 6SN7. The use of direct coupling in this type

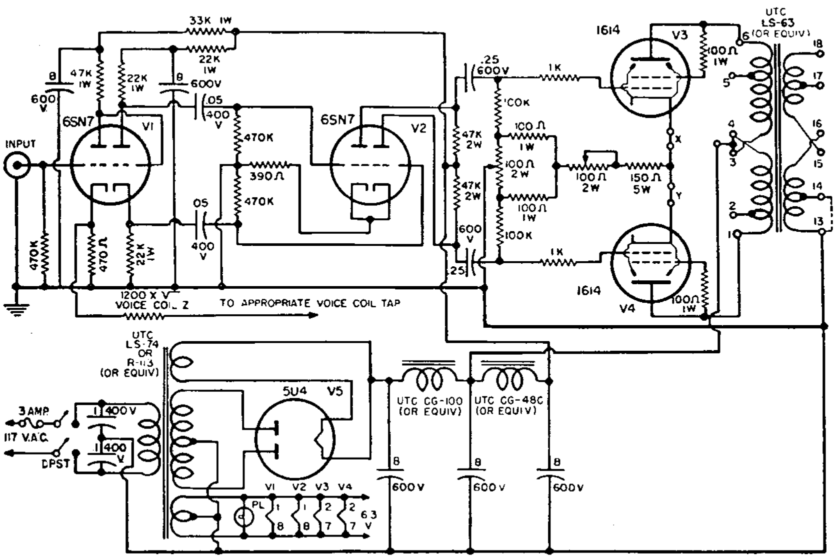


Fig. 9-11. Circuit diagram of a Williamson driver and power amplifier which is suitable for use in the high-quality sound reproducing system.

of circuit has a number of distinct advantages. Since no coupling capacitors are needed, there can be no problem of inadequate low-frequency response in this stage, and no phase shifts which might cause instability when negative feedback is applied. The cathode resistor in the phase splitter is 22k ohms; when the voltage from the plate (about 50 to 60 volts) is connected directly to the grid, the cathode current increases until the cathode voltage is slightly greater than this voltage and automatically adjusts itself so that the proper bias voltage and plate current are obtained. With the values shown in the schematic, the cathode voltage will be about 2 volts higher than the grid, and the plate current will be about 2.5 to 3 ma. The use of equal values of resistance in plate and cathode circuits results in balanced output voltages of opposite phase (as previously described in Chapter 5) which can be used for the push-pull circuit. In the Williamson amplifier, a stage of push-pull amplification is used to drive the power amplifier grids, with a common unbypassed cathode resistor to help maintain accurate balance in the output signal. Adjustable resistors are used in the power amplifier grid and cathode circuits to permit accurate balancing of the plate currents of the two tubes (a current meter having a range of 100 ma may be inserted at links *X* and *Y*) and for accurate bias setting.

The power tubes are connected as triodes and coupled to the loudspeaker load through an output transformer which has been specially designed for this circuit. Negative feedback is taken from the secondary of the output transformer to the cathode of the first voltage amplifier tube. The values shown result in 20 db of feedback; therefore, an input of 1.2 volts results in 10 wats of output power. A 20-watt amplifier may be built using the same circuit design, where higher power is desirable, by using four power amplifier tubes in push-pull parallel and making the necessary changes in the output transformer and feedback resistance value.

The electronic system which has been described in this section gives excellent performance suitable for use in a high quality sound reproduction system. Typical performance specifications of an electronic system built according to the circuits given in this section are:

Frequency response (tone controls in flat position):
- ±0.1 db 20-20,000 cps
- ± 3 db 5-100,000 cps

Tone control range:
- Bass: 17 db boost to 13 db attenuation at 50 cps
- Treble: 21 db boost to 22 db droop at 20,000 cps

Harmonic distortion (at mid-frequencies):
- 8 watts output: less than 0.1%
- 10 watts output: less than 0.3%

Intermodulation distortion:
- 10 watts output: less than 0.5%

These characteristics are considerably better than what are considered to be the minimum requirements for good reproduction.

Selection of Electronic Components

In the previous sections a number of circuits have been described which will satisfactorily perform the functions required in the various types of sound reproduction systems. The circuits have been described for home or custom construction, and may be built exactly as shown, or any desired modifications or changes may be made according to the individual requirements. However, these units need not be built by the individual experimenter if the necessary home construction facilities are not available. A number of units with very similar characteristics to those described can be purchased commercially.

Custom construction has two main advantages: the circuits can be selected and designed specifically to meet the exact requirements of the individual system, and an appreciable saving in cost can be effected. This saving in cost, of the electronic system and possibly in the loudspeaker cabinet, can be applied to the other components of the system which are not practical to build and which must be purchased, thus better components can be purchased and the overall quality of the reproduction system improved. However, if the saving in cost of home constructed units compared to comercially purchased units and the satisfaction of personal construction are not a major factor, many excellent units are readily available which have a wide variety of different features, so that a comparison of the characteristics of the various commercial units will very likely indicate one with the desired features.

In many cases a number of different units may be combined to form the complete system and, according to the individual needs and preferences, all may be commercial units or some may be home constructed. In either case, careful consideration must be given to the matching of the individual units so that impedances and voltage levels are consistent. The output voltages from low-level magnetic pickups are of the order of 0.01 volt, while the output voltages from high-level magnetic pickups are of the order of 0.1 volt. This represents a difference in level of 20 db. Therefore, if a preamplifier is to be used with both types of magnetic pickups, it should be one with different inputs or with a volume control to adjust the output level to be the same as the other inputs to the system. If a number of different inputs are to be selected by a switch, they all should be at approximately the same level and require the same input impedance. Therefore, if a tuner, a crystal pickup, a television sound channel, an output from a magnetic pickup amplifier, and possibly other auxiliary input signals are to be reproduced by the system, they should all deliver signals of about the same level into the same impedance – generally in the range of 1 to 2 volts into approximately 0.5 to 1 meg with the standard units available at the present time.

If a remote preamplifier is to be used, remember that the high-level amplifier will generally require 1 to 2 volts for full power output, therefore the preamplifier should deliver at least this level with an adequate reserve of at least 6 db to 10 db attenuation in the volume control. Since the output lead connecting the preamplifier may extend for a considerable distance, the output impedance should be less than 10,000 ohms. The amplifier should be able to deliver full output power from the input voltage delivered by the preamplifier unit. If a single-unit amplifier is used, the amplification in each input position should be sufficient to deliver full power output with the signal voltages to be expected from the input transducers, and should be able to deliver the required power to the loudspeaker. When an amplifier with its own controls is used in a system which also has a preamplifier control unit, the output from the preamplifier should be connected to one of the high-impedance, high-level inputs, and the amplifier used with its tone controls set for flat response and the volume control left at some convenient setting.

Generally the complete system will be set up with the electronic unit, either the single-unit amplifier or the preamplifier/control unit, if one is used, as the central control of the entire system. Careful attention must also be paid to the a-c and d-c interconnections between the different units. If the preamplifier does not have its own power supply, then plate and filament power must be available from the amplifier power supply. A-c power to all components in the system will generally be available from auxiliary a-c outlets mounted on the control unit, so that all power is turned on by the single switch.

Typical High Fidelity Sound Reproducing Systems

The complete system consists essentially of a combination of a number of electromechanical and electrical transducers with an electronic system. It may include many or all of the following components:

(1) Microphone.
(2) Phonograph turntable and pickup.
(3) Radio tuner.
(4) Preamplifier/control unit.
(5) Voltage amplifier and power amplifier.
(6) Tape recording and playback system.
(7) Loudspeaker.
(8) Loudspeaker enclosure.

Not all of these units are used in every system, but most of them are necessary for any sound reproduction. The systems that will be described in the following sections will contain all except (1) and (6), which can be included with very little effort.

Basic High Fidelity System. The simplest and most basic system which will give acceptable high fidelity at about the minimum cost for good reproduction is set up as shown in the block diagram of Fig. 9-12 (A). The specific units which are used in setting up this system are the following: three-speed record changer, high quality crystal or magnetic pickup, f-m tuner, a-m tuner (if desired), single-unit amplifier, 12-inch loudspeaker (with tweeter if desired), and loudspeaker enclosure (bass-reflex, infinite baffle, or corner folded horn).

The record-player is one of the good three-speed record changers which can be used for playing records at 78, 45, or 33⅓ rpm; it can use either a good lightweight crystal pickup or a good magnetic pickup. The pickup may have two styli for playing microgroove or standard 78 rpm records with the same cartridge, or may be two single-stylus units which are changed for the different types of records. Separate a-m and f-m tuners are used for radio reception, and one or the other may be omitted according to the individual needs and preferences. The amplifier is a single-chassis unit with an input selector switch, complete volume

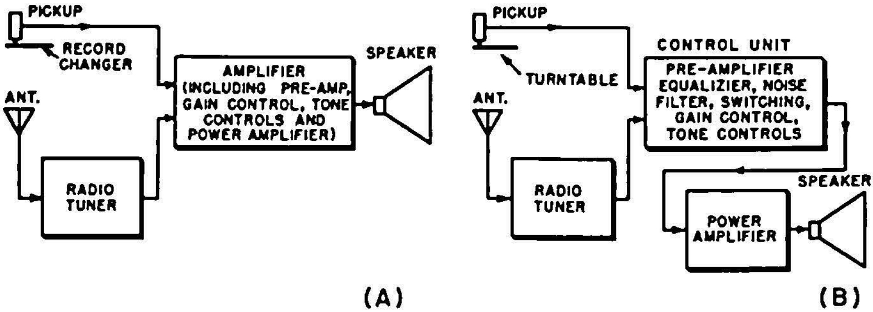


Fig. 9-12. Block diagrams showing basic setups of sound reproducing systems. (A) System using single-unit amplifier. (B) System using preamplifier/control unit and high-level amplifier.

and tone controls, and an equalized preamplifier for magnetic pickups. Its maximum output level is approximately 10 to 12 watts to the loudspeaker. The loudspeaker should be capable of handling this power, and may be any one of the fairly good economical ones available, mounted in an appropriate cabinet or baffle. A good 12-inch single-unit loudspeaker or a woofer-tweeter combination would be quite adequate. It may be mounted in a bass-reflex cabinet, a folded-horn corner cabinet, an infinite baffle, or any other of the acceptable baffles or cabinets which provide proper loading on the loudspeaker cone for good low-frequency response. The various units may be mounted in any cabinets or other furniture which is available for the purpose and the loudspeaker mounted in a special compartment built for this purpose or in a separate commercial cabinet. A number of cabinets are available for the construction of such systems. This sound reproduction system is a good, economical one for an average home installation. Its measured performance is within

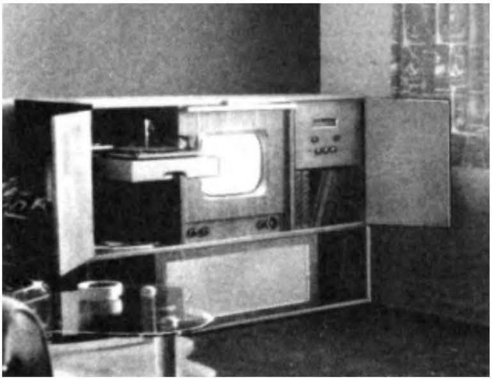


Fig. 9-13. Complete sound reproducing system mounted in a single cabinet containing: three-speed record changer with magnetic reluctance cartridge, f-m tuner, amplifier, loudspeaker and enclosure, and tv receiver.

Courtesy: Vector Laboratories

the recognized limits for acceptable reproduction, and it will perform very satisfactorily.

A photograph of a system installation of this type is shown in Fig. 9-13. The entire system is mounted in a single cabinet which also contains a television receiver whose sound channel it reproduces. The record player uses a good commercial three-speed record changer with a magnetic reluctance cartridge, and the radio tuner inputs are obtained from a commercial f-m tuner and from the television receiver, with no provision for a-m reception. A single-unit commercial amplifier is used as the electronic system. The loudspeaker consists of a good 12-inch loudspeaker in a bass-reflex cabinet for the low frequencies, and two separate tweeters for the high frequencies.

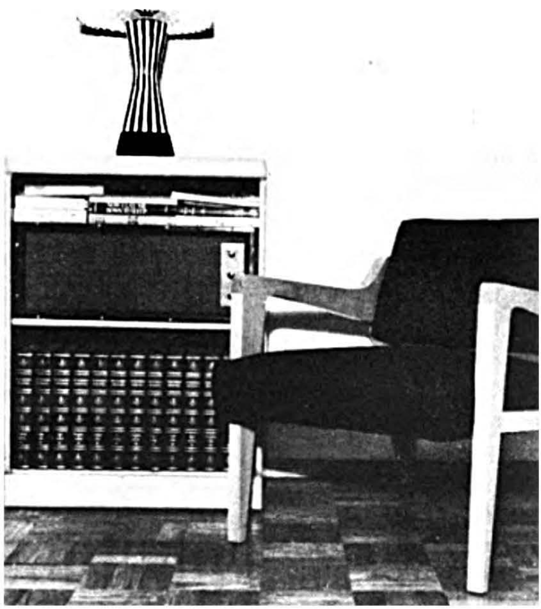


Fig. 9-14. Complete amplifier-loudspeaker combination unit mounted in a bookcase to illustrate the compactness and convenience of this type of unit.

A compact unit which provides low cost high-fidelity reproduction when used in conjunction with a record-player or a tuner is shown in Fig. 9-14. It contains a complete amplifier similar to the one shown in Fig. 9-1, with a high quality loudspeaker in a suitable enclosure, and includes an equalized preamplifier for magnetic pickups and a four-position input selector. It was designed as a complete low-cost compact system which may be used with any desired type of input units, and may also be used as a high-level amplifier and loudspeaker system with separate preamplifier/control units, or as an auxiliary reproducer or monitor in existing systems. Its convenient overall dimensions are such that it may be placed either horizontally or vertically in any standard bookcase shelf or in any other location where space is at a premium.

High Quality System Using Tuner-Preamplifier Unit. Another typical sound reproduction system which uses more elaborate equipment to attain greater convenience and versatility in installation and operation uses the following components: high quality 3-speed record changer, magnetic pickup, a-m/f-m tuner with preamplifier/control unit, high-level amplifier, 12-inch or 15-inch dual-unit or duocone loudspeaker, and loudspeaker enclosure. The record-player is one of the better three-speed record changers, and uses a magnetic cartridge which has a low-impedance output, so that it can be mounted at considerable distances from the amplifier. The radio tuner is an a-m/f-m unit which also includes a preamplifier for the magnetic pickup, a switch to select a-m/f-m/phono reproduction, as well as the complete system controls such as the volume control, bass and treble tone controls, the on-off switch, and the radio tuning control. The amplifier used in this system needs no controls of any sort, and may be similar in design to the ones shown in Figs. 9-5 and 9-6, which have sufficient gain and power output to deliver 10 to 12 watts with low distortion to the loudspeaker. The loudspeaker may be a 12- or a 15-inch dual or duocone unit mounted in an enclosure which is appropriate to both the furniture arrangement and to good sound reproduction.

A photograph of a typical installation of such a system is shown in Fig. 9-15. The complete system is mounted in a cabinet which was already in the room. This system is very similar to the one described, but with a modification in the loudspeaker system. The record-player is one of the better three-speed record changers available, with a dual-stylus magnetic reluctance pickup. The tuner is a high quality a-m/f-m tuner

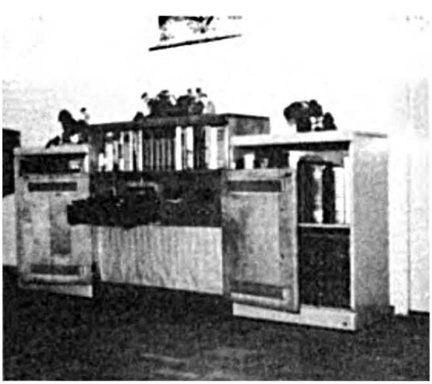


Fig. 9-15. Typical installation of a complete sound reproducing system in an already existing cabinet. This system includes a three-speed record changer with a magnetic reluctance cartridge, a-m/f-m tuner, high-level amplifier, and two accordion-type loudspeakers in a labyrinth baffle.

with a complete control system and an equalizer preamplifier for the magnetic pickup. The high-level amplifier is a home constructed unit which uses a circuit very similar to the one shown in Fig. 9-6. Because of space and furniture requirements, this system does not use the 12-inch or 15-inch loudspeaker which is recommended above. The loudspeaker arrangement consists of two 7-inch accordion type loudspeakers mounted in a labyrinth enclosure, with an enclosed volume of approximately 7,000 cubic inches. The two speakers in parallel will handle the output power of the amplifier and have sufficient cone area to deliver considerable sound power into the air at low frequencies, while each separate speaker is small enough to have good high-frequency response.

Highest Quality System with Separate Units. A more expensive and elaborate system which gives the highest quality sound reproduction that is practical for a home installation may use the following components: high quality single-record turntable, or high quality three-speed record changer, magnetic cartridge with diamond stylus, dynamic noise-suppressor preamplifier (if desired), a-m/f-m tuner (with preamplifier/control unit optional), preamplifier/control unit (if not in tuner or in noise suppressor preamplifier), Williamson type high-level amplifier or equivalent, 15-inch dual-unit or duocone loudspeaker, and bass-reflex or folded-horn loudspeaker enclosure. All the components which are used in a system of this type should be of the highest quality which are available at the present time for use in home sound reproduction systems. The record player should be either a high quality single-record turntable or one of the best three-speed record changers available, which have very constant speed with very little wow and flutter. The pickup should be a high quality magnetic unit, with a diamond stylus for long wearing properties, mounted in a good tone arm. The output of the pickup may be taken through a dynamic noise-suppressor preamplifier, which may be used when old or especially noisy records are being played, or switched out of the circuit when the noise level is sufficiently low. The radio tuner is one of the commercial high quality a-m/f-m units whose performance is suitable for use in this system. If the system is to be used only for reproduction of records, with no provision for radio reception, then a separate preamplifier/control unit should be used. Such units can either be custom constructed with circuits similar to those shown previously in this chapter, or commercially purchased. The high-level amplifier uses a Williamson type circuit similar to the one shown in Fig. 9-11, or an amplifier of equivalent performance, which delivers a maximum output power of 10 watts to the loudspeaker with extremely low distortion, and gives excellent reproduction. It should be noted that, in some installations, the physical requirements will be such that a single-unit preamplifier and amplifier will be desirable; for such installations the single-chassis unit should meet the same high performance standards as the separate units which have been described.

The fine reproduction of which this system is capable can only be realized fully when a sufficiently good loudspeaker is used to produce sound from the electrical energy delivered by the amplifier, therefore, the loudspeaker should be the best that can be afforded. It should, of course, be one which has good frequency and transient response, good efficiency, and is capable of handling the maximum electrical power without distortion. One of the extremely good commercial loudspeakers should be used: most likely a 15-inch dual unit in either a bass-reflex or a folded-horn cabinet. The best loudspeakers are very expensive, but because of their extreme importance in determining the overall quality of reproduction which can be obtained from a high quality sound reproduction system, an effort should be made to use the best loudspeaker and enclosure that is economically practical.

Construction and Assembly of Sound Reproduction Systems

For satisfactory performance of a sound reproducing system, proper attention must be paid to the details of installation and wiring between the various units, and to the construction of any of the units which are home or custom built. The individual units must be properly constructed; once built they must be tested and any mistakes corrected so that all units are in proper operating condition. The entire system must then also be tested to insure that all units are properly matched to one another as to input and output impedances and signal levels. The tests required are not performance tests, which will be described in Chapter 11, but are operational tests to insure that the equipment is in operating condition as designed. It is assumed, of course, that good workmanship will be maintained in all construction and need not be discussed in detail.

Layout and Orientation of Components. One of the most important considerations in the design and construction of audio electronic units is chassis layout and orientation to avoid the introduction of noise and a-c hum into the audio signal. One typical layout of a standard single-unit audio amplifier is shown in Fig. 9-16, and indicates how these various factors have been worked out in a practical unit:

(1) All low-level tubes and components have been placed as far as possible from the power supply and the high-level section, with the lowest level (the preamplifier) the farthest away. This separation reduces the amount of capacitive and inductive coupling from the a-c in the power supply, and from the high-level signal in the power amplifier, into the most sensitive part of the system.

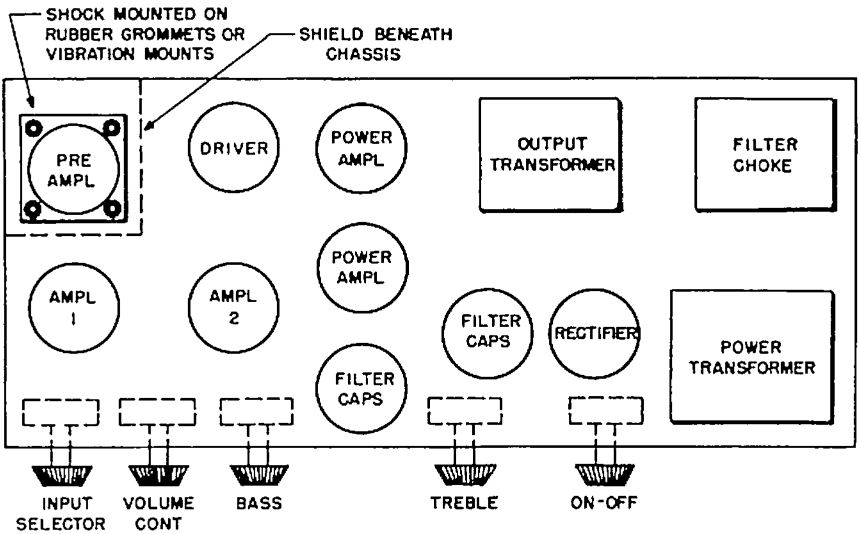


Fig. 9-16. Typical chassis layout of a complete single-unit amplifier.

(2) If the a-c fields are great due to power supply requirements or chassis size, it may be necessary to place a grounded metallic shield around the entire low-level circuit, especially if a microphone preamplifier is included in the system. Since the shells of metallic tubes are normally grounded it is generally necessary only to shield the circuit components beneath the chassis.

(3) The low-level preamplifier tube (or tubes if there is more than one low-level input) are shock-mounted to prevent microphonics, due to shocks and vibration, from being introduced into the audio signal.

(4) Transformers and chokes are located and oriented to reduce to a minimum the amount of hum and noise introduced into the signal through magnetic induction pickup between transformers and chokes. The filter chokes must generally be located close to the power transformer, therefore they should be oriented so that their induction and pickup fields are at right angles to one another, with the second choke farthest from the transformer so that maximum filtering is obtained with the least amount of a-c hum pickup. Since the output transformer will generally also be located close to the power supply, it must be oriented with its induction field at right angles to that of the power transformer for minimum hum pickup.

(5) If an input transformer is used, it must be a well-shielded and low-hum-pickup unit. It should be mounted as far as possible from the power supply, output transformers, and filter chokes. It should not be

mounted in permanent position until the amplifier has been turned on, and should then be oriented for minimum hum pickup and mounted permanently in this optimum position.

A number of other factors which are not obvious in the diagram and which must be taken into account are:

(6) All low-level leads must be shielded to avoid hum pickup due to electrostatic coupling from leads carrying high a-c voltages, and to avoid inductive pickup in low-impedance leads. When balanced low-level circuits are used, both leads should be twisted and placed inside the same shielding. To further minimize the amount of pickup, the low-level lines should be located in the chassis so that they do not run close to any of the high-level or hum-inducing components. In high-impedance, low-level circuits, loss of high-frequency response might result from too high a capacitance in the shielded lead, therefore the chassis layout must be such that these leads will be short enough to avoid loss of high frequencies.

Assembly of Complete Systems. The above considerations are also important in the assembly of complete systems, since it is conceivable that in some installations undesirable component location might result even with the use of separate units and chassis, or that low-level leads might be made longer than is desirable. Other considerations which arise in the assembly of complete systems are:

(7) Systems should be laid out so that magnetic phonograph pickups are not located in high a-c fields, since they are susceptible to hum pickup. Because of their construction, the principles of their operation, and the requirement that they must be very light in weight, it is not possible to shield them completely against such pickup. The only method of avoiding such hum and noise pickup is by avoiding the presence of noise-inducing electromagnetic fields, and since the signal level in the cartridge is approximately the same as in input transformers, this consideration is extremely important. In phonograph construction, the drive motor and the pickup are located for minimum pickup hum; this same care must be taken in the complete system setup. For example, the cartridge should never be located so close to the power transformer that, in such orientation, hum is induced in it.

(8) Acoustic feedback should be avoided between the loudspeaker and the low-level components which might be susceptible to the acoustic vibrations – such as the low-level and preamplifier tubes, and the phonograph pickup. This feedback might result either from direct conduction of the loudspeaker cabinet vibrations through the cabinet and chassis to the critical component, or through acoustic vibrations in the air. Cabinet vibrations are isolated by vibration isolation in mounting of the low-

level tubes and the entire phonograph mechanism (thus also preventing vibrations in the phonograph mechanism from causing noise in the low-level tubes), and by mounting the loudspeaker in a separate cabinet whenever practical. By locating the sensitive components far enough away from the loudspeaker so that they are not in a strong sound field, acoustic vibrations in the air are prevented from introducing noise back into the reproducing system.

If the above factors are taken into account in the initial layout, construction, and setup of the system, the major factors which cause difficulty in any sound reproduction system will have been eliminated.

In the assembly and installation of the complete system, proper attention must be paid to the electrical and physical requirements of the individual units and their integration into the system. The electrical requirements have already been described. Assuming that the various units have been properly selected, and that their input and output impedances are matched, a number of additional factors must be remembered in the actual installation. A crystal pickup should not be located more than 10 or 15 feet from the amplifier – while a tuner or preamplifier which does not have a low-impedance output should be located as close to the amplifier as possible. Low-impedance leads can be quite long without affecting frequency response, but should be well shielded to avoid pickup of 60-cps hum. The line from the amplifier to the loudspeaker is a high-level, low-impedance line, therefore it may be as much as 100 feet long and need not be shielded. If power is taken from one unit for another (for example, from the amplifier to supply power to a preamplifier), one of the leads will carry B+ voltage, therefore if this lead is exposed it should be kept as short as possible for safety.

Physical Installation Conditions. The physical installation conditions are determined largely by room layout and furniture considerations. The most important unit in the system from the viewpoint of installation and physical location is the loudspeaker. For good reproduction it must be mounted in a proper enclosure; acoustic considerations must be most important in deciding the type and location of the loudspeaker enclosure. When the room arrangement permits a separate cabinet for this purpose, any of a large number of appropriate commercial cabinets may be used, the particular choice depending upon the size and furniture style of the cabinet as well as upon the loudspeaker's characteristics. If a convenient wall or large closet is available, it may be used as an infinite baffle for the loudspeaker with excellent results. Cabinets which are already present in the room may also be adapted to act as a loudspeaker enclosure when the proper dimensions and construction are used. The other units in the system may be mounted in any convenient location, subject to the electrical considerations which have already been mentioned. The record player, radio tuner, and amplifier may be mounted in any convenient bookcase, table, or cabinet, with care taken to allow sufficient ventilation for the tuner and amplifier. If all units are mounted in the same cabinet the electrical wiring problems are considerably simplified, however the loudspeaker section should be an independent compartment which is a correctly dimensioned and constructed baffle for the speaker.

If these precautions are taken in the installation and setup of the system, high fidelity reproduction will be obtained from a system which can at the same time meet the functional and decorative requirements of a room.

Troubleshooting and Servicing Procedures

After any construction or assembly is done on any phase or component of the audio system it is necessary to troubleshoot and check all of the work that has been performed. This is done to locate any errors that may have been made and to insure that they are corrected. The use of proper troubleshooting procedures greatly simplifies and reduces the amount of work and effort required, and prevents possible damage to the equipment.

Checking for Wiring Errors and Proper Operation. When individual units of the electronic system have been assembled and wired they must be tested for wiring errors and proper operation. The following procedure may be used:

(1) The entire circuit should first be checked against the schematic and wiring diagram to eliminate obvious errors. A continuity check with an ohmmeter will also disclose possible errors before the equipment is turned on.

(2) The ohmmeter, connected between B+ and ground, should show a very high resistance if the circuit is wired correctly.

(3) Remove the rectifiers from their sockets before power is turned on, so that the a-c and heater circuits can be checked without the application of plate voltage.

(4) If all the tube filaments light up and the tubes become warm. and no wiring errors show up, the rectifiers should then be plugged into their sockets to obtain plate voltage.

(5) With no signal applied to the input, but with a resistor across the input equal to the appropriate output impedance of the unit which will be supplying signal, and the correct load or load resistance connected to the output, the output signal from this unit should be listened to or observed across the load. For no input signal there should be no output.

If there is excessive hum there may be a wiring error in the filament circuit or in the plate voltage filtering, or there may be an open grid circuit. If oscillations are present they indicate undesired in-phase feedback which must be eliminated. In amplifiers designed without negative feedback, this may be due to improper wiring and component layout which causes a high amount of capacitive coupling between high-level and low-level parts of a circuit which has considerable amplification. In amplifiers where negative feedback is taken from the secondary winding of the output transformer back to an earlier stage, the presence of oscillations may mean that the feedback connection has been made to the wrong side of the secondary; reversing the connections removes the oscillation by correcting the phase of the feedback from positive to negative. In some feedback power amplifiers there may be a high-frequency oscillation which is not eliminated by reversing the connections to the output-transformer secondary winding. This oscillation is due to high-frequency phase shift, in the circuit or transformer, which causes the feedback to become positive instead of negative at high frequencies — and is corrected by connecting a small capacitor (say 100 $\mu\mu$f) and a high resistance (about 1 or 2 megohms) in series to ground in one of the high-impedance plate circuits of the amplifier within the feedback loop. This capacitor-resistor combination to ground corrects the gain-phase characteristic to reduce the high-frequency positive feedback and eliminates the oscillation. The highest impedance combination which is effective should be used.

Signal-Tracing Techniques. After the initial procedure has been followed to prevent damage to the unit, an audio signal should then be applied to the input and the resulting output signal will indicate whether the remaining wiring in the signal circuit is correct. If the proper output signal is obtained when the appropriate input signal is applied, then the unit is operating properly and may be used in the system. If the proper output signal is not obtained, then the difficulty can best be located by the use of signal-tracing techniques:

(1) A single-frequency sine wave (a frequency of 1,000 cps is quite suitable) of the proper voltage is applied to the input of the unit and the output terminated in the correct load resistance. The signal generator should have the correct output impedance required by the unit, or should be matched to it.

(2) Then an oscilloscope or a vacuum-tube voltmeter is used to observe the signal at progressive points in the circuit, starting from the input and progressing to the output.

(3) If the unit is not operating properly then, at some point in the circuit, the improper operation will be found. The signal may either disappear entirely between two successive test points, one stage may not have the correct gain, the distortion may become excessive, or any of a

number of other difficulties may be encountered depending upon the exact nature of the circuit failure or wiring error.

(4) When an incorrect signal is located in the plate circuit of a specific tube or the grid circuit to which it is coupled, the following tube in the circuit should be removed from its socket to eliminate any effects reflected into its grid circuit.

(5) If removing this tube corrects the difficulty the trouble may be an open connection in the plate circuit causing the grid to conduct as a diode and heavily load the preceding plate circuit.

(6) Otherwise the error is in the circuit between the two points where the improper operation shows itself.

(7) Once the difficulty is localized in this manner, inspection of the circuit visually and with an ohmmeter and a voltmeter will usually indicate the wiring error.

(8) If the correction of the error causes the proper signal to be obtained at the output of the unit, then it is operating properly.

(9) If not, then the signal tracing process should be continued until all errors have been located and corrected, so that application of the appropriate input signal produces the correct output signal.

The unit will then be operating properly within the limitations and capabilities of the basic circuit design. Methods and principles of further testing of the specific characteristics and limitations of the basic ciruit will be discussed in detail in Chapter 11.

Procedure with Systems Assembled from Commercial Components. The same procedures should be used when a complete sound reproducing system is turned on for the first time after it has been assembled. Before assembling the system, each individual unit should be checked for proper operation. Although this may seem to be an unnecessary precaution in the case of purchased units, they occasionally are received from the manufacturer in defective condition and should be returned for repair or replacement. After each unit is known to operate properly all the units in the system should be checked again to insure that they can be used together. Input and output impedances and voltage levels should be determined. D-c resistances are easily measured by an ohmmeter, while operating and output voltages are measured by d-c voltmeters, output meters, and vacuum-tube voltmeters. Other information may best be obtained from the design data and the manufacturer's literature. When it is determined that all voltages, signal levels and impedances are consistent with one another then the system interconnections can be made and the overall system operation tested. If it is operating properly so that the desired inputs produce the required output sound from the loudspeaker without excessive distortion, and all the controls and switches operate properly, then the system is ready for use and for any further tests of reproduction quality that may be desired. If it is not operating properly, then the signal tracing technique should be used at the major points of unit connections to locate and correct the troubles until correct system operation is obtained.

Installations That Have Developed a Breakdown. The same procedures of troubleshooting that are used for the testing of new constructions and installations also apply to the service of equipment and installations that have had a failure after a period of correct operation. However, since the equipment was at one time in operating condition, the procedure can be made simpler than for new equipment. Instead of looking for wiring and component errors it is necessary to look for failures of components. After an initial visual observation to detect any obvious failures, and a measurement of resistance from B+ to ground, the equipment should be turned on and the power supply voltages measured. If the plate voltage or the heater voltage are not correct, a failure should be looked for in the power supply and in those sections of the circuit which could cause unnatural loading conditions due to component failures. Measurement of resistance from B+ to ground (with power off) indicates any direct breakdown in the d-c load circuit. If there is any such breakdown it may be necessary to unsolder certain key connections, remove certain tubes from their sockets, and perform further measurements both with the ohmmeter and voltmeter in order to locate it. If the power supply voltages are correct (or any breakdowns have been corrected) and all the tubes become warm when filament voltages are turned on, but the equipment is still not operating properly, then signal-tracing techniques should be used to localize the failure in a particular section of the circuit, after which visual inspection and meter readings should quickly locate the component which has failed and should be replaced.

This troubleshooting and servicing technique is used in locating and correcting both system and unit failures. A system failure must first be localized by signal tracing to a particular unit, then more intensive servicing locates the particular component failure in the unit so that it may be corrected. After equipment is serviced in this manner it should be tested again for proper operation, however it is generally not necessary to perform the intensive set of tests that would be performed on new equipment because after the repairs have been made performance should be essentially the same as before the failure.

Servicing Electromechanical Components. It is generally simpler to troubleshoot and service the electronic components and sections of the system than the electromechanical transducers and other mechanical components. These units are usually completely assembled at the factory, and only minor repairs and adjustments can be made without seriously affecting their operation and performance. However, it is not usually necessary to make such repairs to any greater extent than to replace a worn pickup stylus or to cement a damaged loudspeaker cone.

The major electromechanical component in any sound reproducing system which may require any extensive amount of service is the record player – particularly the record changer mechanisms. Fortunately, the work involved is not very difficult when it is performed properly. Since the various units of different manufacture are generally considerably different from one another, it is wise to have available a set of the manufacturer's service instructions for the particular unit on which any work is to be done. There are, however, a number of rules and procedures which can be followed for all types of units.

When first installed, such units should be tested thoroughly for proper operation and performance. Turntables should be level and the units should float freely when they are mounted on springs. Record changers should be tested through their complete cycle to see that they are operating properly. Turntable speeds should be checked with a stroboscopic disc under standard operating conditions with the appropriate number of records on the turntable and the stylus on the record. The output sound should be listened to for quality to insure that no excessive distortion or noise is introduced through improper performance of the turntable mechanism or the pickup cartridge. Once the unit is in operation the mechanism should be periodically cleaned and lubricated, with care being taken to avoid excessive lubrication and to prevent any of the lubricant from coming in contact with the motor drive pully, the idler wheel rubber tire, or the turntable drive rim.

Record Player Mechanism Failures. The failures which may occur in record player mechanisms fall into a number of definite categories, and the service procedures depend upon the specific type of failure (but whenever possible reference should be made to the specific manufacturer's service instructions for the particular unit):

(1) Turntable rotation failure. The motor should be checked to see that there is power to the motor. If there is voltage at the motor its operation should be checked to see that it is not defective, and that it is not binding anywhere due to damaged or frozen bearings, or to gummed oil and foreign material between its armature and pole-piece. The idler wheel should be turning properly and making contact with the turntable drive rim. The turntable should be checked to see that it is not binding and that its bearings are not defective.

(2) Improper operation during record change cycle. The records should be visually inspected for correct size, evidence of warping, damage, size of center-hole, and correct run-in and cutoff grooves. Spring tensions should be checked, as well as the adjustments of clutches, cams, and trip levers. The center post should be inspected for evidence of damage or bending.

(3) Incorrect tone arm indexing and needle pressure. Spring tensions and the adjustment of lever and guide pin assemblies should be checked. The tone arm counterbalance spring should be adjusted for the correct needle pressure, and the tone arm bearing should be checked and any binding eliminated.

(4) Rumble, wow, or slow speed of rotation. Check for damaged or worn rubber rim or flat spots on idler wheel, damaged or poorly lubricated motor or turntable bearings, slipping of idler wheel due to weak spring tension or to oil on rubber rim, record slipping on record below (due to warping), records not properly aligned on turntable.

(5) Noises and squeaks. Rubber idler wheel may be bumpy or out of round. If a squeak is due to the records rubbing against the center-post the squeak may be removed by lightly coating the center-post with wax or vaseline to eliminate friction. If a portion of the label is in the center hole, it should be removed by reaming.

(6) Hum, distortion, lack of output. Check for shorted or open pickup leads and shielded cable, defective pickup cartridge or stylus, or leakage through output plug.

If the procedures, which have been outlined above, are followed in the installation and servicing of the electronic and mechanical components, then the sound reproduction of the system will be maintained at the standards to which it was designed.

Chapter 10

MAGNETIC RECORDING

General

The most widely used method of recording sound at the present time is by means of phonograph discs. However, there are a number of disadvantages to disc recording which have led to constant study and research for better and more convenient methods of recording sound. The most important of these disadvantages are:

(1) The noise level in reproduction from discs is the highest of any section of a high fidelity reproducing system.

(2) Due to the fundamental physical nature of the system it is extremely difficult to obtain good reproduction of the high frequencies.

(3) The recording process is difficult; a degree of proficiency and skill are required in making good disc recordings.

(4) The discs themselves, and the grooves, are fairly fragile and cannot be played a large number of times without appreciable loss of quality.

Although remarkable progress has been made in the use of low-noise surfaces, improved recording techniques for better high-frequency response, and the use of difficult-to-break plastic materials for the disc, the above limitations still exist. The greatest advantage of the method of disc recording is that large numbers of acceptable-quality discs can be reproduced from the same master recording quickly, easily, and at low cost. At the present time no other system of recording compares in this respect with the phonograph disc.

For many applications this advantage of the ease of producing phonograph discs is not important compared to the disadvantages. For example, in commercial rebroadcasting of recorded program material low noise level and good quality of reproduction are probably the most important requirements. For recording of difficult material or under difficult conditions simplicity and reliability of the recording process are important considerations. The amateur experimenter is interested in a system capable of good quality of reproduction without requiring a great degree of skill in the operator. Last, it is generally desirable for the recording not to be fragile and not to deteriorate with age or with repeated playings.

The recording system which most nearly overcomes the disadvantages of disc recording is the magnetic recording on wire or tape. The most widely used and highly developed of the various magnetic recording systems generally make use of a paper or plastic tape upon which is deposited a magnetic layer as the recording medium. Although such recordings cannot be produced as easily and cheaply as commercial disc pressings, they have the following advantages:

(1) The reproduced noise level is very low, and an extremely wide frequency range can be reproduced, depending primarily upon the tape speed.

(2) The recording process is extremely simple, once the equipment is properly set up it can be used simply by turning it on with no operator present to monitor it.

(3) The recordings themselves are quite rugged and can be used indefinitely without loss of quality; they can be cut and edited by splicing the cut ends together with ordinary adhesives.

(4) In making the recording there is very little danger of spoiling the record because of difficulties in the process, therefore good recordings can usually be obtained the first time.

(5) The recording equipment is relatively light and easily transportable.

Because of these various advantages and characteristics the system of magnetic tape recording is replacing disc methods in a very large number of applications at the present time, and is being used in many applications where disc recording is not practical.

Principles of Magnetic Recording

Magnetic recording consists of magnetizing a ferromagnetic material in proportion to the instantaneous amplitude of an audio signal. In recording, the magnetic tape (or wire) is moved past a recording head which impresses upon it a magnetization proportional to the current through a recording coil. In playback, the magnetized medium is moved past a pickup coil and causes a current to be induced which is proportional to the magnetization.

Magnetic Poles and Fields. In order to understand the process of magnetic recording a knowledge of certain basic properties of magnetized materials is required. The recording medium is magnetized in essentially the same way as a simple bar magnet, but with the intensity of magnetization proportional to the amplitude of the recorded signal. The simplest and most basic magnet is seen in Fig. 10-1 (A), showing a magnetized bar with a north pole at one end and a south pole at the other. This basic magnet illustrates one of the most important fundamental principles of physics: that for every magnetic pole there must exist a corresponding pole of opposite polarity. If the bar magnet is cut in half, two new poles come into existence at the broken ends, as shown in Fig. 10-1 (B).

There is a force of attraction between two unlike magnetic poles, and a repelling force between two similar poles. This force is proportional to the strengths of the magnetization of the poles and inversely proportional to the square of the distance between them.

A magnetic field can best be visualized by drawing imaginary "lines of force" between the north and south poles. The intensity of the field at any point is measured by the number of lines of force per square centimeter at that point. If an isolated single pole could be placed at any point in the field the force on it would cause it to move along a line of force from one pole to the other; the entire field can be considered to consist of such lines of force. Although it is not possible to isolate a

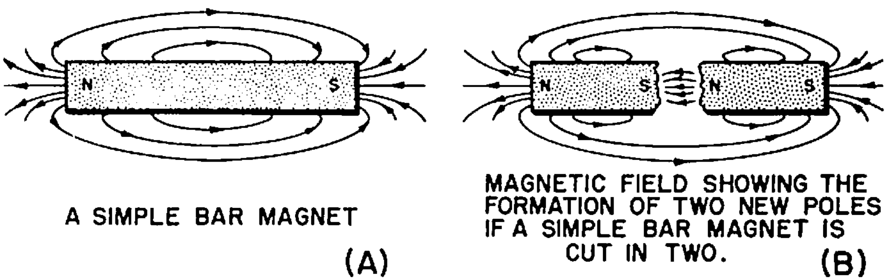


Fig. 10-1. The simple basic bar magnet, showing the lines of flux in the magnetic field.

single magnetic pole in the same manner that an electrical charge can be isolated, the magnetic lines of force can be mapped in practice by use of small compass needles to indicate the direction of the specific line of force at any point in the field and moving them along this line of force, or else iron filings scattered in the field will align themselves along the lines of force. Typical lines of force in a magnetic field are illustrated in Fig. 10-1 for the simple bar magnet.

The bar magnet shown in Fig. 10-1 need not be magnetized in such a manner as to have only the two poles at the ends. It may also be magnetized to have additional poles along its length, as illustrated in Fig. 10-2. For example, the bar magnet in (A) has north poles at each end, and a south pole at the center. The magnet in (B) is magnetized with several poles along its length, unequally spaced and of different magnetic strengths. This process of different magnetizations along the length of

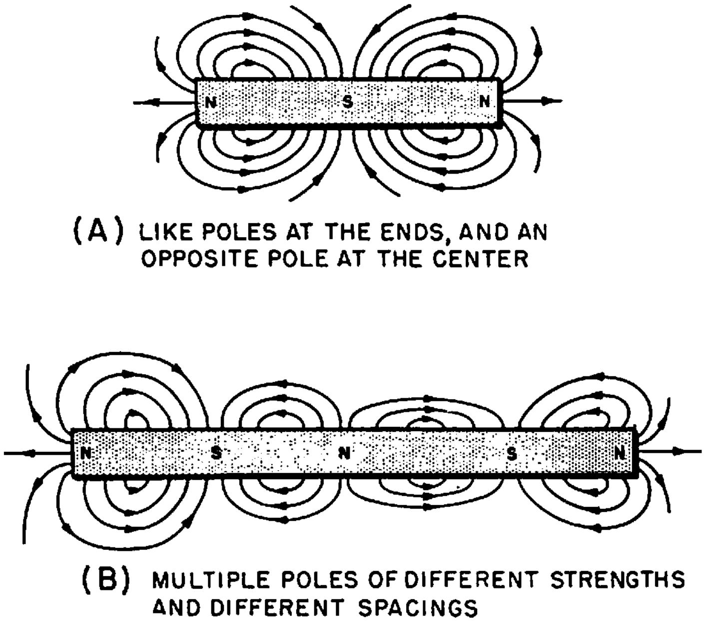
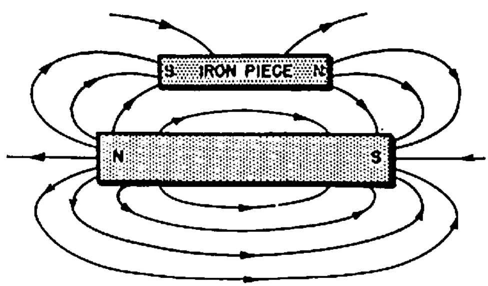


Fig. 10-2. Different methods of magnetizing bars.

the bar can be extended to form magnetic patterns corresponding to almost any desired wave shape.

Induced Magnetism. When a piece of iron is placed in a magnetic field the distribution of the magnetic lines of force is altered in such a way that more of them pass through the iron than the space if the iron were not there. The magnetic field inside the iron is therefore greater than in air before the iron was placed in the field. This effect is illustrated in Fig. 10-3, which shows a bar magnet with its field disturbed by

Fig. 10-3. Disturbance of the magnetic field by the presence of a piece of iron.



the presence of a soft iron piece. The field on the side of the magnet away from the iron piece is undisturbed. If the field near the iron piece is observed alone it looks very similar to that of a bar magnet. The iron piece is said to be *magnetized by induction,* with its south pole near the north pole of the inducing magnet and its north pole near the south pole of the inducing magnet. If the piece is made of very soft iron it will not retain its magnetization very long after it has been removed from the magnetic field, in fact only a very small amount of residual magnetization will remain. However, if the piece is made of hard steel it will remain a permanent magnet after it has been removed from the inducing field.

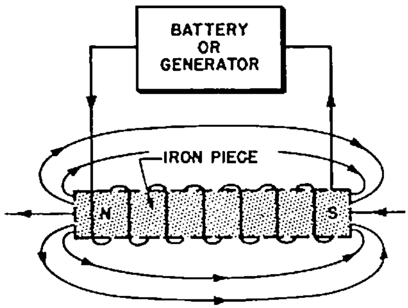


Fig. 10-4. Magnetic field due to a current flowing through a coil.

Magnetization can also be induced in a magnetic material by means of an electric current in a coil, as shown in Fig. 10-4. When a current flows through a coil shaped in the form of a helix, a magnetic field is set up which is almost exactly like the field of a bar magnet. In the inside of the coil there will be a certain field strength H, which depends upon the current, the number of turns, and the geometry of the coil. If a bar of iron is inserted into the coil the strength of the field is greatly increased, and an electromagnet is formed. If the bar is soft iron it will lose most of its magnetization when the current through the coil is stopped; if it is a steel bar it will retain most of its magnetism.

The converse of the above can also occur; that is, if a magnetized bar, shown in Figs. 10-1 and 10-2, is moved relative to a coil, a voltage is induced across the terminals of the coil. This voltage is proportional to the number of turns in the coil, the strength of the magnetic field, and the speed with which the magnet is moved relative to the coil:

$$E = N\frac{d\phi}{dt} \quad \text{abvolts}$$

where E is the induced voltage in abvolts, N is the number of turns in the coil, and $\frac{d\phi}{dt}$ is the time rate of change of flux.

This process of magnetization in iron by a current in a coil, and the induction of voltages in coils by magnetized iron, are the basis of magnetic recording. A magnetic wire, or a tape with a magnetic coating, is moved past a coil carrying a current proportional to the instantaneous intensity of the sound which is being recorded. This current produces a magnetic field which is proportional to the sound intensity, and as the magnetic recording medium is moved past the coil it is magnetized in proportion to the audio wave shape. The sound is reproduced by moving this magnetized material past a coil, in which a voltage is induced which is proportional to the intensity of magnetization, therefore representing the original audio current in the recording coil.

The basic principle of magnetic recording was developed over fifty years ago, but only recent advances in the science and technology of electronics and sound reproduction have made it possible to use this system for the high fidelity reproduction of which it is capable at the present time.

Recording Bias

Unfortunately, the magnetic properties of ferromagnetic materials are not linear, so that the recording performed in the simple manner described will not give undistorted reproduction. If a bar of iron is placed in a magnetic field with a magnetizing force H, such as that inside a coil, the number of lines of force emerging from the iron is much greater than would be there without the iron. The number of lines of force per square centimeter is called the *flux density* and is indicated by the letter B. The magnetic properties of the iron are described by its *permeability*, designated by the symbol μ and defined by the ratio:

$$\mu = \frac{B}{H}$$

The permeability is not a simple property of the iron; it also depends considerably on the intensity of the magnetizing field.

The magnetization characteristics of a typical magnetic material are shown in the curve of Fig. 10-5. Starting with the material unmagnetized, if the magnetic field is increased from zero the magnetization curve is *ABD*. However, at any point (say B) on this curve when the magnetic field is reduced to zero, the material retains some of the magnetization and the flux density decreases only to the value C. If the magnetization is increased to saturation as represented by point D, and then the field

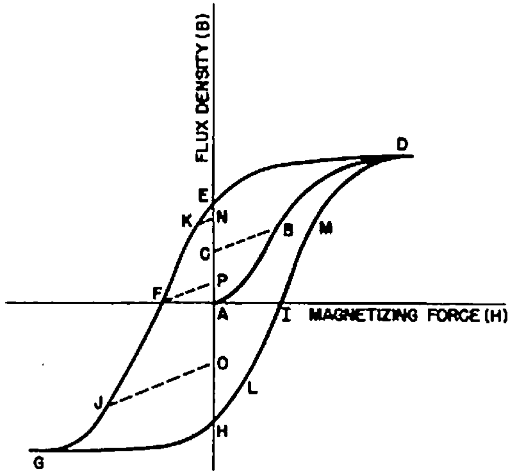


Fig. 10-5. Magnetization characteristics of a typical magnetic material.

intensity reduced through zero to saturation intensity of the opposite polarity, the magnetization curve will be *DEFG*. If the field is then increased again to point *D* the magnetization curve will be *GHID*. This magnetization cycle is known as the *hysteresis* curve. Most of this curve is non-linear and would lead to considerable distortion if it were to be used directly in recording, therefore special techniques must be used to eliminate any distortion due to this non-linearity.

D-C Bias Method. One method of eliminating distortion is by the use of d-c bias in recording. It can be seen from observation that the hysteresis curve of Fig. 10-5 is fairly linear over the ranges *JFK* and *LIM*. Therefore, a d-c bias can be added to the recording signal to bring the magnetizing field to the linear range. With no recording signal the field will have the value *F*: when the recording material is taken out of the field, the residual magnetization has the value *P*. When the recording signal is added to the fixed field *F*, the field varies along the line *JFK* and the residual magnetization has the values *OPN*. Since the magnetization curve and the residual magnetization are fairly linear in this range, the reproduction process will be reasonably free of distortion.

However, there are a number of drawbacks to this method of attaining linearity in magnetic recording. The exact value of the fixed bias is very important: if it is either too great or too small there will be distortion. The amplitude of the recording signal is also limited since serious distortion will result if it is too great. There is also an inherent noise in the recording medium when the recording is played back.

High-Frequency Bias. A more satisfactory method of obtaining linearity in magnetic recording is by the use of high-frequency bias. This type of bias consists of a supersonic signal (whose frequency should be at least five times the highest recorded audio frequency, and may be anywhere in the range from 30 kc to about 80 kc) which is added to the audio signal being recorded. The method of operation of this type of bias may be better understood by reference to Fig. 10-6, which shows the effects of the bias and audio frequency magnetizing currents upon the residual flux density of the recording medium. If the only magnetic field acting on the medium is that due to the supersonic bias signal, with the audio signal zero, the magnetization will then vary at a high-frequency rate symmetrically in both polarities about the point of zero flux density.

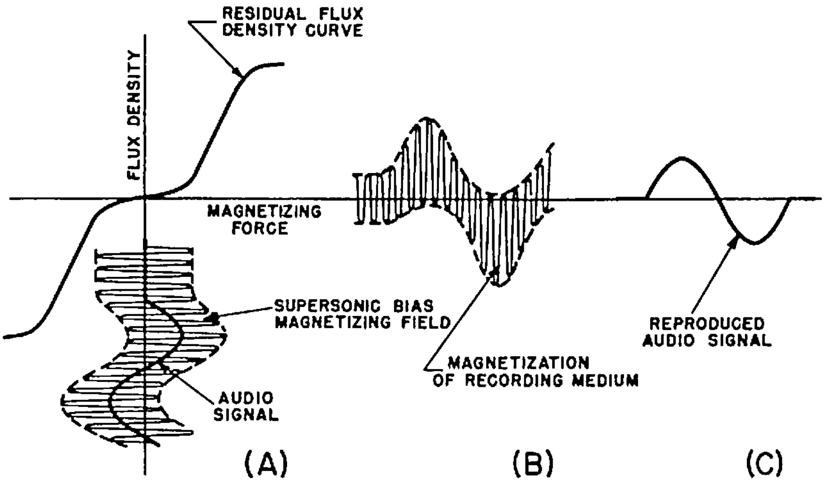


Fig. 10-6. Use of a supersonic biasing field to reduce distortion in magnetic recording.

When the medium leaves the magnetic field it retains the value of residual magnetization corresponding to the particular value at the instant it left the field. The residual flux density which the recording medium will have for a typical audio signal superimposed on the high-frequency bias is shown in Fig. 10-6 (B).

When the recording is reproduced the playback channel responds to the average value, rather than to the actual high-frequency magnetization of the medium. For example take the recorded signals shown in Fig. 10-6 (A) and (B), when the recording is reproduced by a system which averages curve (B), the resulting audio output signal is then shown by (C). This reproduced signal is seen to contain a slight amount of distortion, but proper choice of the magnetic field and bias currents reduces the distortion to a very low value.

Magnetic Recording Systems

The basic magnetic recording system consists of: (1) the electronic circuits for amplifying the audio signal to a sufficient level to supply the magnetizing currents, and for generating the bias signal, (2) the recording and reproducing transducers which convert the electrical signal to a magnetic field and the magnetic field back to an electrical signal, and (3) the transport mechanism for moving the recording medium past the recording and reproducing transducers.

The block diagram in Fig. 10-7 shows the basic operation of the various component parts, and the manner in which they are combined into a typical magnetic recording system, with switching circuits for most economical use of the same components for both recording and reproduction. The signal to be recorded may originate in either a microphone,

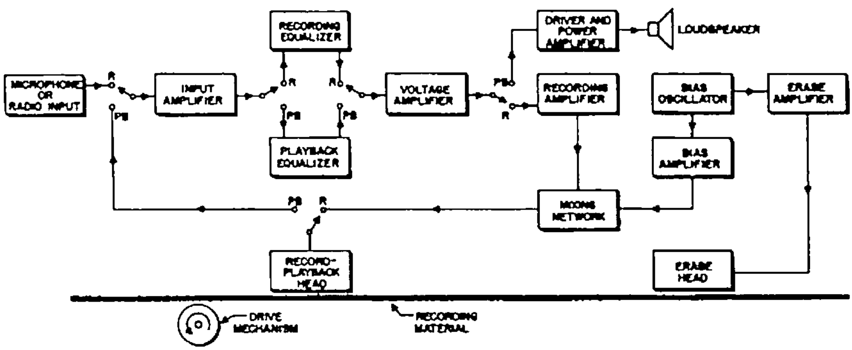


Fig. 10-7. Block diagram showing the functional operation of a typical magnetic recording system.

a radio tuner, or a disc recording pickup. This input signal is amplified by a low-level preamplifier and a voltage amplifier, then applied to the recording head by the recording amplifier stage. Any recording and playback equalizers which may be needed in the system are included in the voltage amplifiers. The bias signal is generated in a separate bias oscillator and applied to the recording head, is mixed with the audio signal, and the composite signal is recorded.

Since the recording material may not always be completely demagnetized when it is used, a demagnetizing field is included to insure that the material will be completely unmagnetized when it reaches the recording head. To accomplish this the material is first moved past a special *erase head* which generates a demagnetizing field to remove any previous magnetization or recording. The material to be erased is first subjected to a very strong alternating field which is sufficient to saturate it in both positive and negative polarities. The material then goes through a complete hysteresis cycle for each cycle of this alternating field, this completely removes any previous residual magnetization. It is then gradually removed from the field; therefore, the hysteresis cycles gradually become smaller in amplitude so that when it is completely removed from the field the material is left completely unmagnetized. The alternating current which supplies this erasing field is generally obtained from the supersonic bias oscillator through a power amplifier.

When the recording is played back both the erase and bias fields are turned off; the connections in the various circuits are switched for playback amplification. A separate head may be used for playback, or the same head may be used for both recording and reproduction. The output of the head is amplified by the low-level preamplifier, then by the voltage and power amplifiers to supply the required output level to the loudspeaker or other load.

A mechanical drive system moves the recording material from the storage reel past the erase and record/playback heads to the takeup reel. A number of light pressure-actuated switches are generally in contact with the material, to stop the drive motors at the end of the reel or in case of a break in the material.

The performance requirements of the drive system for high quality recording and reproduction are very rigid. Since variations of speed produce wow and flutter, accuracy of speed must be kept within limits at least as close as for disc recordings, and should preferably be better. Because of these rigid requirements the material cannot be driven from the reels. (Also, unless special speed controls are used, the speed would change with diameter as the material is taken from one spool to the other.) A type of drive which is capable of meeting the various requirements is the *capstan drive.* The speed of motion is controlled by wrapping the material a half-turn around a capstan which drives it through friction. This capstan is driven at constant speed, with a flywheel attached to it to maintain constant speed and avoid variations in speed. The two reels are driven by motors: one on the takeup reel to keep tension on the material while recording, and another on the storage reel for rewinding after the record has been made, but these motors have no part in controlling the speed of the material in recording or playback. This type of drive system is capable of maintaining the correct speed with variations of less than ± 0.1 percent.

Basic Considerations in Magnetic Recording

A number of basic factors determine the performance of a magnetic recording system and the quality of reproduction of which it is capable.

The design of the record/playback head is extremely important; it must be a practical design for obtaining the required magnetic field intensity and frequency response with practical electronic circuits. It must also be able to withstand the considerable amount of friction and wear which result from the continuous contact with the rapidly moving recording material, and its performance should not be too badly affected by the dirt which will be accumulated from the recording material.

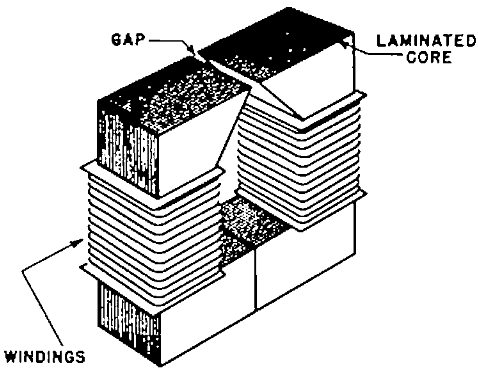


Fig. 10-8. Construction of a typical record/playback head for magnetic tape recording.

Magnetic heads generally consist of coils wound on laminated cores having a construction of the type shown in Fig. 10-8. This is an open type of construction which does not require the material to be threaded through the coil area. It is constructed from two pieces by winding parts of the coil on each piece, with the windings properly tapped to present the correct impedances and ampere-turns to the amplifier when used as a recording head, and a sufficiently high output voltage when used as a playback head. The two parts of the core are then clamped together to form a continuous core with the one recording/playback gap. The two pieces are made with smoothly ground surfaces to make extremely good magnetic contact at the one side opposite the gap; a non-magnetic spacer is generally inserted in the gap to give an accurately held gap of about 0.0005 inch to 0.002 inch. The entire head must be very carefully made, with exact balance between the two legs in order to keep to a minimum the amount of hum pickup due to the motors and other strong magnetic fields, and with very close tolerances on the width of the gap for the desired frequency response.

Other factors which are important in determining the performance of a magnetic recording system are the speed of motion of the recording material past the head, the characteristics of the material itself, and the amplifier frequency compensation.

When recorded signals of equal amplitudes and different frequencies are reproduced by a magnetic pickup head the outputs at the different frequencies will not be equal, but will depend upon the individual frequencies. The reason for this difference in response is that the voltage induced by a changing magnetic field is proportional to the rate of change of the field. Since the field changes faster in direct proportion to the frequency, the output voltage will be directly proportional to the frequency. For example, if the frequency is doubled the output voltage will be doubled, thus giving a 6 db/octave boost at the frequency increases. (It should be noted that this type of response is similar to the response in disc recording of a velocity type pickup to a constant amplitude recording.) However, this characteristic is obtained only at the lower frequencies of recording. At higher frequencies the response depends to a greater extent upon the length of the gap. The response begins to drop off when the wavelength of the recorded signal approaches the length of the gap. This effect depends upon both the length of the gap and the speed of the material, since the same frequency will have a longer wavelength if the frequency is increased.

Recording speeds in general use at the present time are 7½ inches/second and 15 inches/second for high fidelity reproduction, and 3¾ inches/second for economy with a corresponding loss in high-frequency response. At a recording speed of 7½ inches/second, one wavelength at 10,000 cps is equal to 0.00075 inch, while at 15 inches/second the same 10,000 cps signal will have a wavelength of 0.0015 inch. The gap in commercial heads for use with these recording speeds is usually somewhere between 0.001 inch and 0.002 inch. A typical unequalized frequency response, for constant current recording played back through a flat response amplifier, is shown in Fig. 10-9 to illustrate the importance of these various effects which have been described.

Because of the frequency response shown in Fig. 10-9, equalization is required in the amplifiers to give an overall flat frequency response for the program material reproduced by magnetic recording. Because most of the reproduction noise is at the higher frequencies it is not desirable to have any high-frequency boost in the playback channel, therefore the high-frequency equalization is included as pre-emphasis in the recording amplifier. The dotted curve *B* in Fig. 10-9 shows the pickup output when an input signal of constant amplitude is applied to the

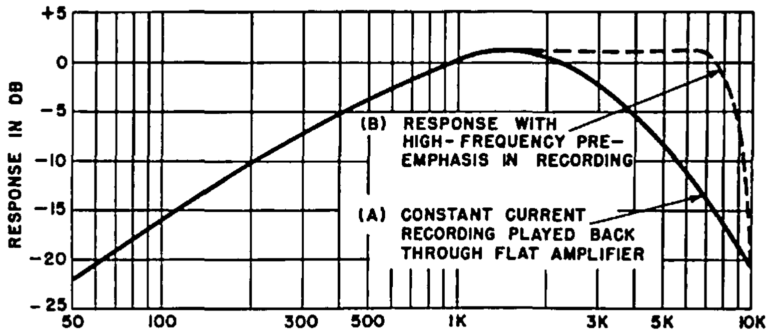


Fig. 10-9. Typical frequency response in magnetic tape recording at a tape speed of 7½ inches per second.

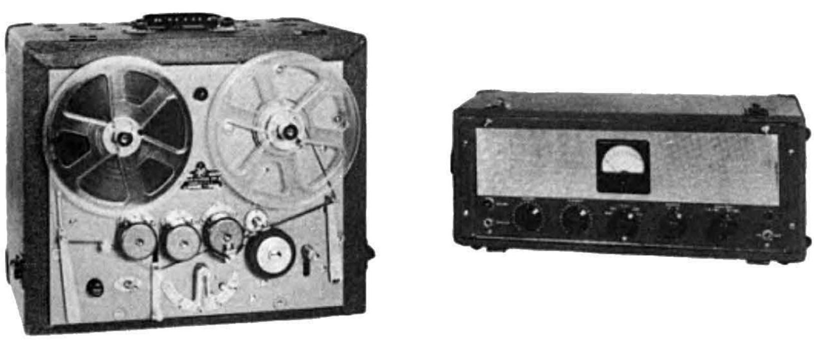
input of an equalized recording amplifier. The low-frequency equalization is generally included in the playback amplifier, and is designed to boost the low frequencies to give a flat output response for a pickup output characteristic such as that shown in the curve *B*. The required equalizations in both the recording and the playback amplifiers are accomplished by standard equalizer networks of the types that have already been described in previous chapters.

By proper choice of the various magnetic field strengths, careful construction and design of the equipment, and selection of a good recording material, the reproduction noise in a magnetic recording system can be kept to a very low amount. The main criterion in the selection of the magnetic field intensities is the amount of distortion in the reproduction: the bias signal must be chosen so that the maximum undistorted audio signal can be recorded. The mechanical components and the transport mechanism must be designed to introduce a minimum variation in speed, maintain proper contact between the recording material and the heads, and should always be kept clean and in proper adjustment to obtain maximum performance. Noise due to the recording medium is due to irregularities in the medium which cause spurious flux changes that cause noise in the reproduction; this type of noise can be reduced only by use of the best available material. When proper attention has been paid to the design and operation of the magnetic recording system, audio signals can be reproduced with a noise level of better than −50 db below maximum signal level with a harmonic distortion of better than 2 percent at this maximum level. The quality of reproduction obtained with the best magnetic recording systems is so good that, when reproduced through the best home reproduction sound systems, program material which has been recorded magnetically can hardly be distinguished from reproductions from direct sound pickup.

Magnetic Recording Circuits

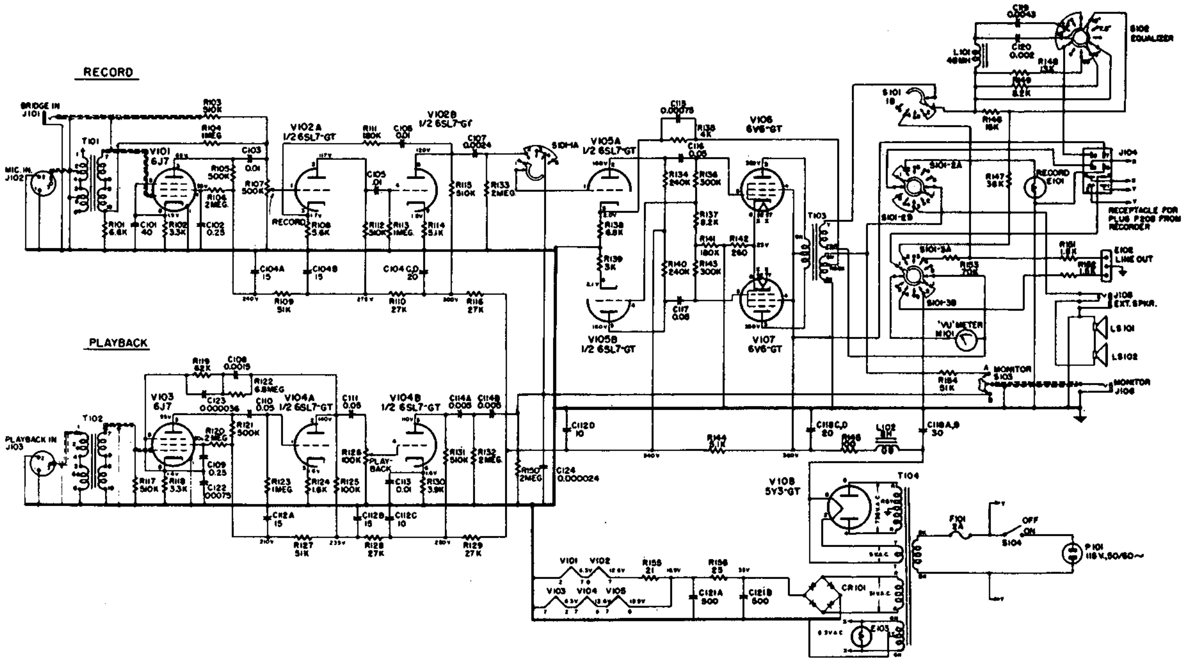
Commercially available magnetic recorders range in quality and price from the relatively inexpensive units for office dictation, where quality of reproduction is a very minor consideration, to the most expensive and carefully designed units for commercial radio broadcast and recording applications where fidelity is the most important consideration. A number of units are available for the amateur experimenter who is interested in good quality of reproduction at moderate cost. These units may be used for overall reproduction from sound input at the microphone to sound output from a loudspeaker, or they may be incorporated as part of an existing system to record and reproduce the audio electrical signals obtained from the system.

A typical high quality magnetic tape recorder suitable for use in sound reproducing systems is shown in Fig. 10-10. This unit is similar in operation to the basic block diagram which is shown in Fig. 10-7. The schematic circuit diagram in Fig. 10-11 shows the details of the circuits used in the recorder of Fig. 10-10. The signals to be recorded may be obtained either from direct sound pickup by a microphone or as an electrical signal from some other reproducing system. The microphone signal is amplified by a low-level preamplifier to a high enough level that it can be mixed with electrical signals from other sources such as radio tuners and disc reproducers. This signal is then amplified by a voltage amplifier and by the high-level recording amplifier. The high-frequency pre-emphasis is accomplished by an LC treble boost network in the output



Courtesy: Presto Recording Corp.

Fig. 10-10. Photographs of typical high-quality magnetic tape recorder.



SCHEMATIC DIAGRAM OF AMPLIFIER UNIT

S101 LEGEND	
POS. NO.	FUNCTION
1	BIAS
2	RECORD
3	PLAY
4	LINE
5	P.A.

NOTES

1-ALL RESISTORS ARE IN OHMS. K=1,000 OHMS; MEG=MEGOHMS.

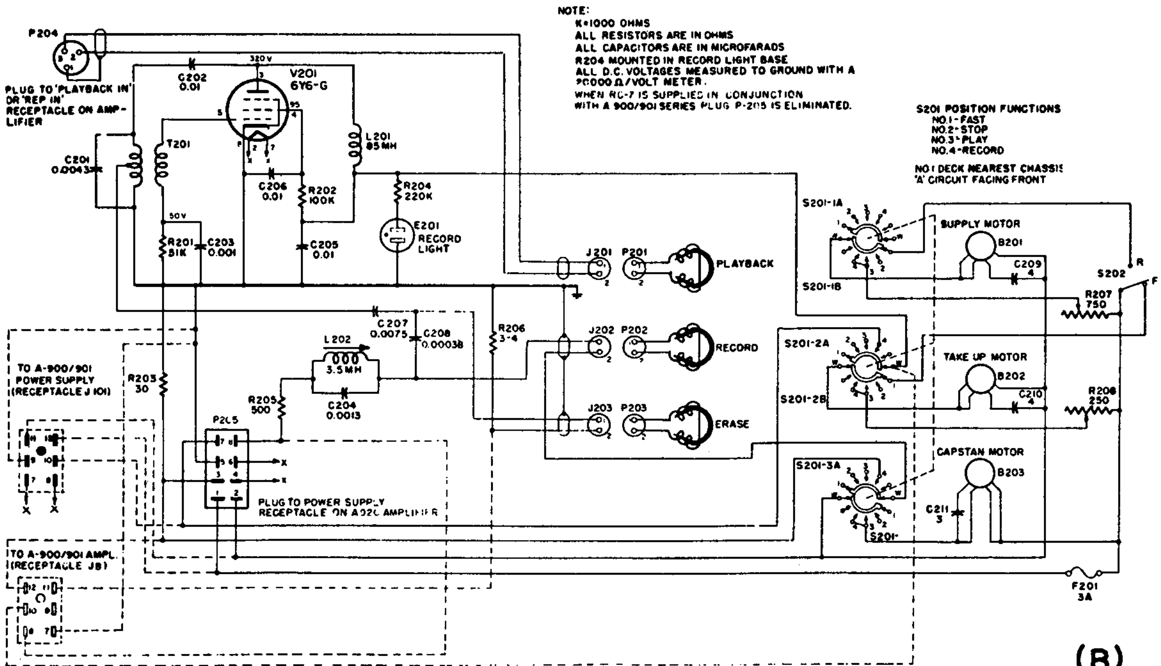
2-ALL CAPACITORS ARE IN MICROFARADS.

3-ALL DC VOLTAGES MEASURED TO GROUND WITH A 20,000 Ω/VOLT METER SELECTOR SWITCH S101 SET IN "PLAY" POSITION.

(A)

Courtesy: Presto Recording Corp.

Fig. 10-11. (A) Schematic diagram of amplifier portion of magnetic tape recorder shown in previous figure.



SCHEMATIC DIAGRAM OF THE RECORDER UNIT

(B)

Courtesy: Presto Recording Corp.

Fig. 10-11. (B) Schematic diagram of recorder units of magnetic tape recorder.

line to the recording head. The high-frequency bias is generated by an oscillator in the recorder unit, where it is applied to the erase head, mixed with the audio signal, and applied to the recording head.

Separate playback head and voltage amplifier circuits are used so that the actual recording on the tape may be monitored if a power amplifier is plugged into the monitor jack. The playback head and amplifier are connected to the recording power amplifier and loudspeakers in the normal playback position. The playback amplifier consists of a low-level preamplifier and a voltage amplifier in which the required bass boost is obtained by an RC feedback equalizer. In playback, the bias/erase oscillator is turned off by disconnecting its plate voltage. A meter is included in the unit to monitor the recording level to permit maximum intensity without overloading, and is switched with the different functions of the circuit to act as a test meter to indicate proper operation.

The various functions of the unit are selected by switches that control the different functions of the circuit and the motors. The motor on the takeup reel has two speeds: one speed for normal recording and playback, and a fast forward speed when it is desired to skip part of the reel rather than play it through from the beginning. A fast rewind motor is connected to the storage reel for rewinding the tape after it has been recorded and reproduced. The tape speed is controlled by a capstan drive to provide accurate speed control and very little flutter and wow. The tape speeds are 7½ inches/second and 15 inches/second, selected by a switch. The frequency response at 7½ inches/second is flat from 50 to 7,500 cps at less than 2 percent total harmonic distortion, at 15 inches/second it is flat from 50 to 15,000 cps at less than 2 percent total harmonic distortion. The noise level is better than 50 db below full output. At 7½ inches/second a standard 7-inch diameter reel can contain 32 minutes of program material, and at 15 inches/second can contain 16 minutes of program.

The tape recorder which has been described is typical of a number of units which are intended for high quality service. The designs are essentially the same, although there may be a number of differences in various circuit and operational details. Some units may have switching arrangements which use the same circuits and heads for both recording and playback. Other differences may be the inclusion of a 3¾ inch/second speed to permit 64 minutes of program material on a standard 7-inch reel of tape. Some units record on half of the tape in one direction, then have an automatic reversing of direction and record on the other half of the tape in the reverse direction to double the recording time (this gives a slight increase in noise level). Units of this type are called "twin-track" recorders. The individual characteristics of the various commercial units can best be obtained from the manufacturer's literature, which should be consulted before the selection of a specific unit for any application.

Chapter 11

MEASUREMENT OF QUALITY OF AUDIO REPRODUCTION

General

Once the complete sound reproduction system has been set up its performance should be evaluated according to the ultimate criterion that the reproduced sound should be exactly the same as the original sound. This condition is unattainable in actual practice, but the degree to which it is attained is a measure of the quality of reproduction. Since hearing is done by the ear, the best method of judging performance is a listening test, and the ear must always be the final judge of audio quality.

The most basic listening test applies a sound to the input of the system and, using the ear as the measuring instrument, compares the reproduction with the original. This test is actually the basic criterion against which all tests of audio quality must eventually be made. However, it is quite difficult to perform, because the original sound is generally not available for direct comparison. Furthermore, such a test is completely subjective and is not capable of accurate quantitative measurements for comparison and evaluation under standardized conditions.

Any type of measurement consists essentially of causing the system under test to perform its function under controlled conditions, and of measuring the success with which it performs this function. The accuracy of the measurement is determined by the degree to which the input test signal represents or simulates the true operating condition, and by the accuracy with which the operation of the system and the relevant variables can be measured.

It is necessary to find a quantitative measurement of the quality of sound reproduction which can be used as a basis for comparison and evaluation. However, for accurate measurements the average sound is much too complex a function to permit a ready determination of its characteristics. The sound must be simulated by simpler types of signals which are capable of direct measurement and which simulate the characteristics of the sounds which are of interest.

The complete testing of an audio system requires measurements of several different types of input signals in order to represent accurately the various factors known to be important in determining the quality of a complex sound. By using simplified signals which can be generated and measured fairly easily the more complex relationships which occur in sound are separated and simplified into a form capable of exact measurement. Some of these measurements require the use of a single steady sine wave, others require two such sine waves of different frequencies, and still other tests require the use of more complex signals having certain transient characteristics similar to those occurring in natural sounds. By using such simplified signals which can be generated and measured fairly easily, the more complex relationships which occur in sound are separated and simplified into a form capable of exact measurement. Proper correlation of these simplified measurements will then give an indication of the extent to which the system under measurement can reproduce complex sounds.

The individual factors which represent the overall sound reproduction quality of any system or component unit of the system have already been discussed in Chapter 2; the experimentally determined limits for good reproduction have also been listed. The techniques and test setups for measuring these factors in practical sound reproduction systems will be described in this chapter. A complete set of measurements of any type of audio system would include measurement of all the following factors:

(1) Frequency response.
(2) Noise level.
(3) Maximum power output.
(4) Harmonic distortion at different power levels.
(5) Intermodulation distortion at different power levels.
(6) Transient response.
(7) Phase response.
(8) Wow and flutter (in disc, film, or magnetic reproduction).

At the present time methods and equipment exist for measurement of all these quantities in all types of sound reproduction systems. A number of basic scientific investigations have been carried out with great numbers of people to determine the proper correlation of these simplified measurements with listening preferences. The results of these tests indicate that the various distortions should not exceed the limits shown in Table 11-I, which is expanded from a table given in Chapter 2.

When carrying out such measurements, the tests can, in general, be performed either over the complete system or for any individual component of the system. For example, overall measurements may be performed from microphone to loudspeaker, or over any individual phase of the system such as the amplifier, the phonograph pickup, the loud-

Response or Distortion Being Measured	Input Signal	Output Signal	Limits	
			Good Reproduction	Acceptable Reproduction
Frequency response	Steady sine wave	Steady sine wave	20-14,000 cps	40-10,000 cps
Power output	Steady sine wave	Steady sine wave	Depends upon size of listening room (typical requirement is 5-10 watts)	
Noise level	Zero	Random noise	−60 db (below full output)	−50 db (below full output)
Harmonic distortion	Steady sine wave	Fundamental plus harmonics	2% total harmonics	3-5% total harmonics
Intermodulation distortion (amplifier alone)	Sum of high-frequency and low-frequency steady sine waves	Amplitude-modulated sine wave	4%	10%
Transient response	Step voltage or tone burst	Step voltage or tone burst	No set standards	
Phase response	Steady sine wave	Steady sine wave	No set standards	
Wow and Flutter	Steady sine wave	Frequency-modulated sine wave	0.1%	1.0%

Table 11-1. Test signals used for performing the various response and distortion tests, and acceptable limits for these distortions.

speaker, etc. The overall method is advantageous in that it measures the interactions as well as the characteristics of the individual components and gives an indication of the complete reproduction quality. However, the method of individual measurements has the advantage of being generally simpler and more convenient to perform, as well as serving to localize any deficiencies which may exist. With this method each component should be carefully matched into the proper input and output impedances to approximate as closely as possible its interactions with the overall system. Then the overall quality can be obtained by correlation of the individual measurements.

General Techniques and Instruments for Audio Measurements

The basic setup for any type of measurement is shown in Fig. 11-1. A known input of the proper form is applied through a generator of the desired impedance to the input of the system, and the resulting output is measured across the proper load impedance. Each different type of audio unit will, of course, require the correct type of input signal and the output must be measured by the proper type of measuring instrument. A listing of all the various types of audio units which it may be necessary to test at times is given in Table 11-II. This table represents an overall picture of the general techniques of audio measurements.

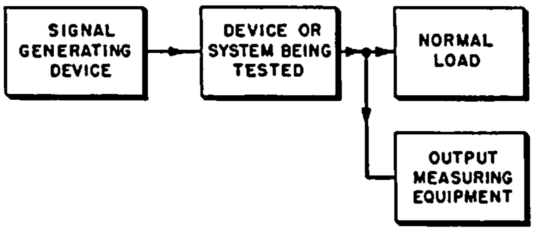


Fig. 11-1. Basic setup for making measurements.

In performing any measurements whatever, careful attention must also be paid to the selection and characteristics of the test equipment itself. All equipment and instruments used for the measurements must be sufficiently better than the system under test so that their defects can reasonably be neglected. This factor should always be given careful consideration, and if any auxiliary equipment must be used in making a particular measurement or set of tests, precautions should be taken that the equipment used should not introduce errors which might be large enough to make the readings unreliable or meaningless.

Since measurements consist basically of applying input signals and measuring output signals, the test instruments which must be used for audio measurements will be instruments for performing these two functions. In addition, d-c test instruments are necessary for troubleshooting

Type of Audio Reproduction Unit	Input			Output			
	Type of Signal	Source of Signal	Terminal Impedance	Type of Signal	Type of Measuring Equipment	Terminal Impedance	Comments and precautions
Microphone	Sound	Calibrated loud-speaker	Air in room	Voltage or current	Vacuum-tube voltmeter	Resistance or trans-former	Acoustics of measuring room must be planned to avoid acoustical resonances. Signal source and measuring meter must not introduce inaccuracies. Calibration of loudspeaker must be accurately known.
Amplifier or other electrical transmission circuit	Voltage or current	Electrical signal generator or amplifier	Resistance or trans-former	Voltage or current	Vacuum-tube voltmeter	Resistance or trans-former	Input and output impedance must be properly matched to correspond to actual operating conditions. If auxiliary amplifiers are used in measurement, they must not introduce inaccuracies.
Recording head Disc Film Magnetic	Voltage or current	Electrical signal generator or amplifier	Resistance or trans-former	Mechanical optical or magnetic	Calibrated pickup	Record material	Mechanical drive should be free of flutter. Effects of record noise and flutter must be taken into account. Characteristics of calibrated pickup must be known.
Record Disc Film Magnetic	Mechanical optical or magnetic	Recording head		Mechanical optical or magnetic	Calibrated pickup		Mechanical drive should be free of flutter. Effects of record noise and flutter must be taken into account. Characteristics of calibrated record must be accurately known.
Pickup Disc Film Magnetic	Mechanical optical or magnetic	Calibrated record		Voltage or current	Vacuum-tube voltmeter	Resistance or trans-former	
Loudspeaker	Voltage or current	Electrical signal generator or amplifier	Resistance or trans-former	Sound	Calibrated micro-phone	Air in room	Acoustics of measuring room must be planned to avoid acoustical resonances. If auxiliary amplifier is used in measurement, it must not introduce inaccuracies. Characteristics of calibrated microphone must be accurately known.

Table 11-II. Summary of the various types of input and output signals and measuring equipment which must be used in testing the various units of an audio reproducing system.

of newly constructed equipment, and for test and servicing of equipment which may have suffered a failure in operation. These instruments do not necessarily include only electronic instruments for measuring the characteristics of electrical signals, but also instruments which may be needed for the testing of the electromechanical units in the system, therefore certain acoustic and electromechanical signal generators and measuring instruments will be required. The most useful and basic instruments in the testing of audio systems are:

(1) Electronic test equipment
- a. Variable-frequency sine-wave oscillators
- b. Variable-frequency square-wave generators
- c. Vacuum-tube audio-frequency voltmeters
- d. Oscilloscopes
- e. D-c voltmeters

(2) Acoustic and electromechanical test equipment
- a. Standard calibrated microphones
- b. Standard test records

These are the most basic instruments needed for testing the performance of sound reproducing systems. Instruments for performing these various functions are all commercially available, or may be built from home construction kits which are also available.

These instruments alone will not permit the determination of all the characteristics which have been listed in the previous section as being a measure of the quality of reproduction. However, the more complex instruments which are needed for performing these measurements are generally combinations of various of the instruments on this list, or may be built by use of these instruments. The most important of these more complex instruments are:

(3) Instruments for measurement of characteristics of audio signals
- a. Distortion and noise meters
- b. Harmonic and wave analyzers
- c. Intermodulation analyzers
- d. Wow and flutter meters

Generally many of the instruments listed in category (1) will be a part of the equipment of the experimenter who does much work in the construction and assembly of audio equipment. They should be available to anyone who undertakes the construction of any chassis from the circuit diagram or from home construction kits, because it is extremely difficult to troubleshoot or service any equipment without them. (They are not strictly necessary in the assembly of audio systems entirely from commercially purchased units as these units may be tested and found to be operating satisfactorily before they are accepted from the vendor.) The test instruments in category (2) may also be available to the audio experimenter (although much less frequently), since they are useful in performing overall measurements of the system's performance from acoustic or electromechanical input to sound output. The instruments in category (3) are generally a part of the equipment only of the experimenter who does extensive work with audio systems; they are fairly complex and expensive to buy. However, these instruments are commercially available or may be assembled using the instruments listed in (1).

The basic principles of the various types of audio measurements, the specific techniques for performing the different tests, and the necessary equipment and test setups, will be described in detail in the following sections.

Measurement of Specific Factors Affecting Reproduction Quality

Certain of the basic tests have been in general use for many years — namely, frequency response, power output and noise level — and are fairly well known to experimenters and technicians so that they do not require any greatly detailed discussion. However, other measurements which have long been standard procedure among audio engineers (for example, measurement of harmonic distortion) are not very well known to those who are not audio specialists. The remaining measurements which have been described (particularly those whose importance has only recently become clearly understood) are certainly not very well known to the average technician, in many cases they are not sufficiently understood even by audio specialists. In fact, techniques and equipment for measuring some of these factors are at the present time still in the development stage. However, wider knowledge and recognition of the methods of measuring and evaluating the various distortion factors will be of considerable importance in helping to improve the general overall level of quality in all types of audio reproducing systems.

Frequency Response, Power Output, and Noise Level. Frequency response is generally measured by applying a signal of constant amplitude to the input and measuring the output signal amplitude as the frequency of the input test signal is varied. Maximum power output is measured by increasing the input signal level (with the system set for full gain), and observing the output signal (either aurally, or visually by means of an oscilloscope or a meter) to determine the output power level at which the system overloads or becomes excessively distorted. The noise level is determined by measuring the output signal with zero input signal and the gain control of the amplifier set for full gain.

Harmonic Distortion. Harmonic distortion has long been known to be a measure of amplitude non-linearity. The general method of measuring the total harmonic distortion introduced by the reproducing system is shown in Fig. 11-2. It consists essentially of applying to the input a steady single-frequency pure sine wave (known to be relatively free of distortion), and measuring the harmonic content of the output signal.

Total harmonic content is measured by filtering out the fundamental component and measuring the remaining signal as a percentage of the total. The fundamental may be filtered out either by a high-pass

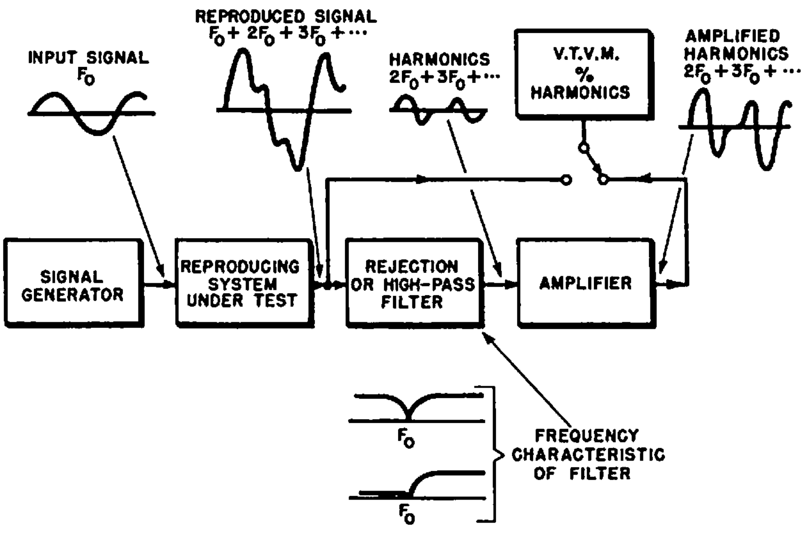


Fig. 11-2. General method of measuring the total harmonic distortion, introduced by the reproducing system.

filter (which greatly attenuates the fundamental but passes all its harmonics), or by a signle-frequency rejection filter (such as the RC parallel-T type) tuned to it. The high-pass filter has the advantage of eliminating the effects of a-c hum and other low-frequency noise, but the rejection filter is generally easier and more convenient to use.

The magnitude of the individual harmonic components may be measured with the wave analyzer, as indicated in Fig. 11-3. The wave analyzer contains an accurately calibrated variable tuned circuit so that it measures only one harmonic component at a time. Such a measurement is capable of giving information concerning the order as well as the amount of harmonics, and therefore can also be used for estimating the

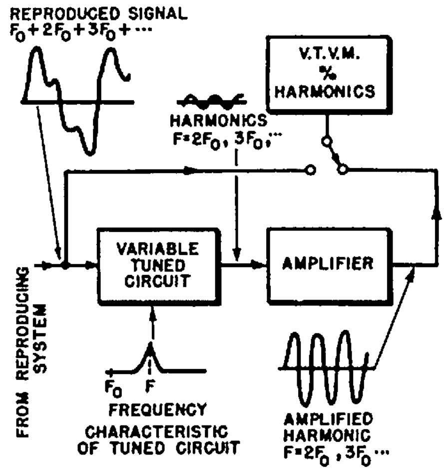


Fig. 11-3. Wave analyzer method of measuring each individual harmonic distortion frequency by using a variable tuned circuit instead of the rejection filter of Fig. 11-2.

amount of intermodulation distortion which may be expected from the system.

When either of these two methods is used the general procedure is first to measure the total signal (including the fundamental) with the vacuum-tube voltmeter, then switch the signal through the filter and measure the amplitude passed by the filter. This, then, gives the harmonic amplitude as a percentage of the fundamental.

Intermodulation Distortion. The measurement of intermodulation distortion is a relatively new technique in audio measurements. Intermodulation is caused by the same amplitude non-linearity which causes harmonic distortion, but neither one can be readily calculated from the other.

The intermodulation characteristics of a system are measured by applying two known frequencies simultaneously to the input, and determining the degree of interaction and distortion of these two frequencies by measuring the magnitude of the new frequencies generated in the system. The general method of performing this type of measurement is shown in Fig. 11-4. Two units are required: a signal generator which supplies the composite input signal, and the analyzer which determines the amount of cross-modulation generated in the reproducing system.

For the purpose of determining intermodulation, the composite signal effectively simulates those characteristics of a normal audio signal that are important in generating the intermodulation products which unpleasantly affect quality of the reproduction. It consists of a low-frequency component between 40 and 150 cps, and a high frequency component which may be either about 2,000 cps or between 7,000 and 12,000 cps. The amplitude of the low frequency is about 12 db higher than the high-frequency component. These two signals are generated separately and combined in a mixing and attenuator circuit in such a manner that there is no appreciable interaction or intermodulation between them. By means of the attenuator, the composite signal can be applied at any desired level to the system under test. This choice of low and high frequencies, and of their relative amplitude of four-to-one, gives a fairly accurate representation of the sounds which are most importantly affected by intermodulation distortion.

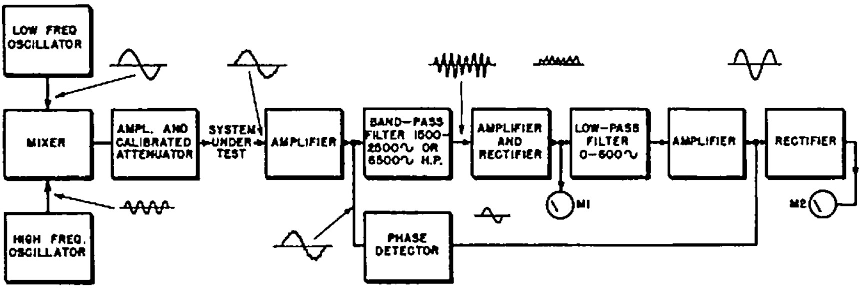


Fig. 11-4. Block diagram showing general method of measuring intermodulation distortion.

The extent to which the reproduction system produces cross-modulation between these two components is a measure of the amount of intermodulation that will be introduced into the more complex sounds of speech and music. The amount of intermodulation which the system introduces into the composite signal is measured by the analyzer unit. This distortion consists of amplitude modulation of the high-frequency component at the low-frequency rate (or at some multiple of it). It is not necessarily sinusoidal, but has a wave shape which depends upon the characteristics of the system under test. The fundamental basis of intermodulation distortion measurements is to measure the amount of this amplitude modulation as a function of the amplitudes of the two input frequencies. The percentage of intermodulation is generally defined as:

Percentage intermodulation = Percentage amplitude modulation of the high frequency signal for the composite signal (as described above).

The measurement is accomplished by passing the reproduced signal through suitable filters to separate the desired frequency components, then measuring their relative levels. The low-frequency component is removed first by a band-pass or a high-pass filter which passes only the high frequency signal and whatever modulation products may be present. The high-frequency signal is then demodulated and passed through a low-pass filter to determine the amount of low frequency modulation present in the reproduced high-frequency component. The relative amplitudes are measured by vacuum-tube voltmeters as shown in the block diagram.

In certain audio applications (and particularly in the design of new equipment) is is often desirable also to have a distortion phase detector for measuring the relative phase of the intermodulation. This would indicate whether the intermodulation occurs on the positive or negative swing of the low-frequency signal, or whether it is symmetrical. However, this measurement need not be performed when the primary purpose of the test is to determine the quality of the reproduction.

A number of commercial units are available at the present time which incorporate these various features for intermodulation measurements.

Transient Response. It is only recently that the importance of transient response has been fully realized, and there are as yet no standardized methods of equipment for measuring the transient response of audio reproducing systems. However, by application of certain basic principles and proper analysis of the results, good measurements can be obtained by using equipment which is generally available.

The measurement of transient response must, obviously, consist of measuring the response of the system to some standard transient signal. This transient test signal does not necessarily have to have the same form for all types of systems under test, and may in general depend upon the system being tested and the type of measurements. However, it must in all cases be possible to interpret the response to the test signal in terms of the response of the system to audio-frequency transients.

In the transient analysis of any type of physical system, the basic input test signal is the unit step function. In electrical measurements this is a voltage which is zero until some given reference time, and then rises with a square wave front to unit voltage and remains at that voltage. This voltage is illustrated in Fig. 11-5 (A). The transient response of the system to all other waveforms can be completely determined by observing its response to this unit step voltage.

In practice, the step voltage can generally be approximated by a square wave to facilitate observation upon the screen of an oscilloscope.

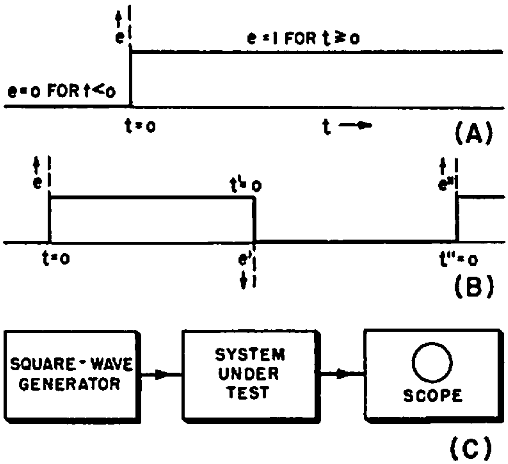


Fig. 11-5. General method of measuring transient response. (A) Graph of unit step voltage used in transient analysis of electrical systems. (B) Representation of unit step voltage by means of square wave of sufficiently long period. (C) Step-function transient analysis of a system by means of a square-wave generator and an oscilloscope.

However, when this approximation is made care must be taken to keep the period of the square wave long enough so that the system has responded completely before the end of the cycle. Th transient response of the system may then be determined qualitatively by visual observation of the oscilloscope trace.

The response is determined by comparison of the reproduced wave shape with the applied square wave. Differences between the two wave

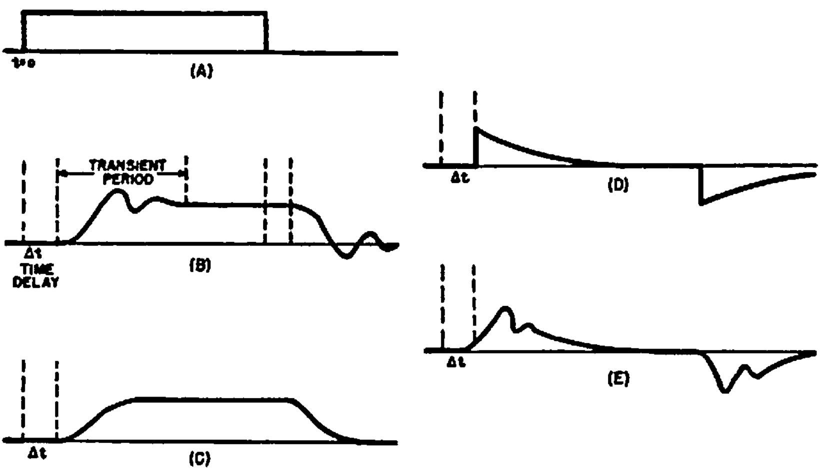


Fig. 11-6. Transient response of a number of common types of audio response characteristics to a unit step voltage. (A) Input step voltage. (B) Response of low-pass circuit, underdamped. (C) Same, only highly damped. (D) Response of high-pass circuit. (E) Response of band-pass circuit, underdamped.

shapes are readily recognized and identified; if the repetition frequency has been chosen properly they will be a measure of the true transient response of the system. The method of analysis of the response to this step-function signal can best be understood by reference to a number of typical responses as indicated in Fig. 11-6. The waveforms shown in (B) and (C) of Fig. 11-6 are of particular interest because they show that a reproducing system which is insufficiently damped can give rise to a spurious damped oscillation that depends only upon the characteristics of the system and has no relation to the reproduced signal. In a system with negative damping (positive feedback) this oscillation would tend to increase with time rather than to decrease. In general, the response of most audio systems can be derived from the responses shown in Fig. 11-6, with variations depending mainly upon the repetition rate of the square wave.

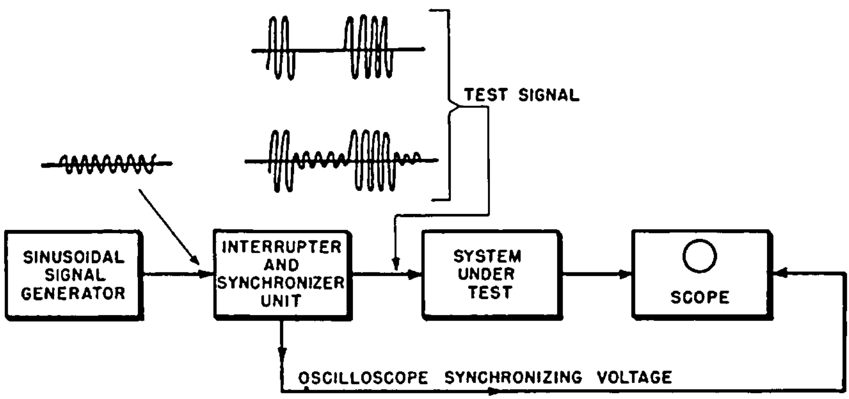


Fig. 11-7. Block diagram of method for measuring the transient response of a system by means of an interrupted sine wave.

In some cases it is not convenient to use the step-function or square-wave method of transient analysis, either because the proper repetition rate cannot be attained conveniently or because the results do not appear in a sufficiently convenient form. In such cases a method may be used which takes closest account of the actual physical form of the sounds occurring in speech and music. Many of the transient signals which are encountered often have rapid decay as well as build-up times. The transient response of a system to such signals, and the residual vibrations of the system, may therefore be measured by applying short bursts of signal to the system and observing the decay after the signal has been removed.

A method of using this technique to measure the rate of decay of vibration at all parts of the frequency spectrum is illustrated in Fig. 11-7. The system under test is supplied with a variable test tone through an interrupter, giving make and break period on the order of 1/100 to 1/20 second duration. The output of the system is applied to an oscilloscope whose horizontal sweep is synchronized with the interrupter. The trace on the oscilloscope screen then represents the decay envelope at any frequency. The amplitude at any time after the signal has been removed can be measured by moving a vertical slit mask across the face of the tube and observing the height of the trace in the slit at the point corresponding to the desired time. The results obtained by this method are an accurate indication of the transient response and residual vibrations in the reproducing system.

Phase Response. The phase response of a system can readily be determined by a number of different methods. In general they involve vector addition and subtraction of the applied and reproduced signals, and are quite simple and convenient to use.

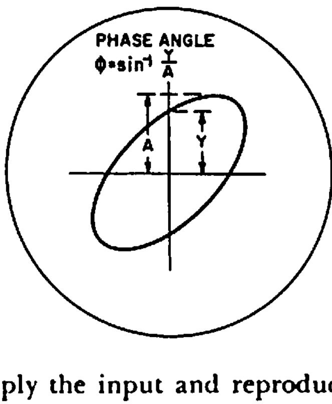


Fig. 11-8. Determination of a phase angle from Lissajous pattern on an oscilloscope.

One of the simplest methods is to apply the input and reproduced signals to the horizontal and vertical plates, respectively, of an oscilloscope and observe the resulting Lissajous pattern. The scope amplifiers should be adjusted to produce equal horizontal and vertical deflection. When the two are in phase (or 180 degrees out of phase) the figure is a straight diagonal line. If there is a phase difference between them, the angle can be obtained by measuring the point of intersection with the Y-axis, as indicated in Fig. 11-8.

Wow and Flutter. Measurement of the wow or flutter introduced by a mechanical recording system (such as phonograph turntables and magnetic recorders) consists essentially of measuring the amount of frequency modulation of a steady tone when it is reproduced by the system. The technique for performing this measurement is shown in the block diagram of Fig. 11-9.

A signal of constant frequency is applied to the input of the reproducing system, and the reproduced output obtained in the form of an electrical signal. (In motion picture work, a frequency of 3,000 cps has been chosen as standard for this test signal. This might be a reasonable standard to use for other types of sound reproduction as well.) The reproduced signal is then passed through a bandpass filter which passes only the test frequency and its possible variations, removing any noise and hum which may be present. This signal may then be amplified if necessary, and passed through a limiter. The amount of frequency variation is then measured directly by a discriminator, which gives the frequency change as a voltage which may be measured by a low-frequency voltmeter calibrated in percentage frequency change. The output of the discriminator may also be amplified so that a direct-inking recorder may be used to record the actual frequency changes to permit a more complete analysis of speed variations.

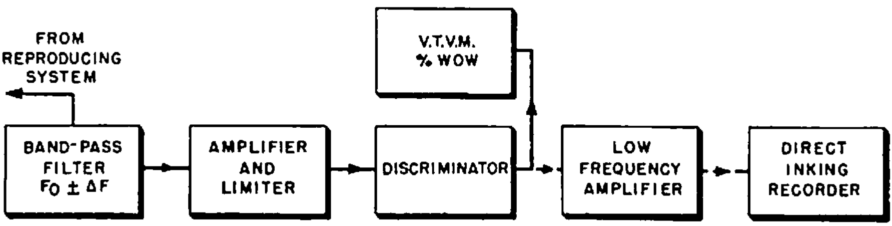


Fig. 11-9. Block diagram of a wow meter for measuring frequency variation in the electromechanical reproduction of a steady tone.

The amount of flutter can be evaluated quantitatively from the information obtained by a measurement performed in this way, and certain quantities can be defined which will permit a quantitative comparison and representation of the flutter present in the reproduction. The *percent flutter* is the ratio (in percent) of the rms frequency deviation to the average frequency. The *flutter rate* is the number of complete cycles of frequency deviation per second. The *flutter index, I,* may be expressed as:

$$I = (fx/r)\,\Delta f$$

where Δf is the rms frequency deviation in cycles, f is the tone frequency and r the flutter rate. For flutter rates greater than 5 per second $x = 1$, for rates from 1 to 5 per second $x = r/5$ and for rates less than 1 per second $x = r^2/5$.

Measurements in Audio Systems

When sound reproduction systems are tested for quality in actual practice, it is extremely important that such measurements be performed properly. Otherwise, the factors which are to be measured may be completely masked by errors due to the methods of measurement.

In all measurements the basic requirements are that the input signals have the correct form and be relatively free of distortion, that their characteristics be accurately known, and that the measuring equipment be sufficiently free of errors to permit measurement of the desired quantities. These considerations must be taken carefully into account in measuring quality in audio reproduction systems.

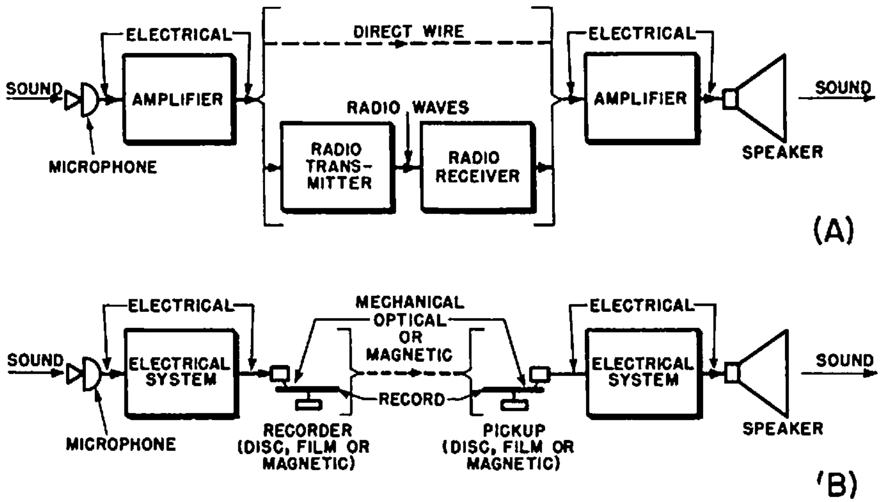


Fig. 11-10. The various types of audio reproduction systems. (A) Purely electrical systems. (B) Reproduction systems involving recordings.

The complete system reproduces sound to sound, but sections of this system may convert sound to electrical signals, or mechanical signals on phonograph records, or reproduce records or electrical signals as sound (see Fig. 11-10). Therefore, it may be necessary to produce standard sounds, electrical voltages, and calibrated records as test signals; and it must also be possible to perform accurate measurements upon these sounds, electrical voltages, and records.

The electrical signals can be tested most conveniently and with the greatest amount of precision, because instruments for generating and measuring electrical voltages have reached a high state of development. To measure the characteristics of components whose function is to reproduce electrical signals, the input voltage is supplied by an electrical signal generator, and the output measured by a voltmeter. The main precaution which must be taken in performing such measurements is that the input and output impedances should represent as closely as possible the impedances that the component will see in the system in which it will be used. The best method of accomplishing this is to terminate the unit in the actual output system with which it will be used, while applying the input signal from a generator of the proper impedance. The output may then be measured with a voltmeter of sufficiently high impedance so that it will not appreciably affect the output.

Greater difficulties are encountered when it is desired to test systems which include microphones or loudspeakers. Precise measurements of sound and the production of standard sound signals are more difficult than for electrical signals, and a more careful experimental technique is required. All such measurements must be performed in rooms or spaces which have been carefully planned to avoid acoustical resonances, or performed in such a manner as to avoid the production of resonances; considerable attention must also be given to the correct calibration and measurement of a standard of sound intensity and quality.

The most practical approach to acoustical measurements in the average laboratory is to use a calibrated standard microphone as the standard for all sound measurements. Such a microphone is one which

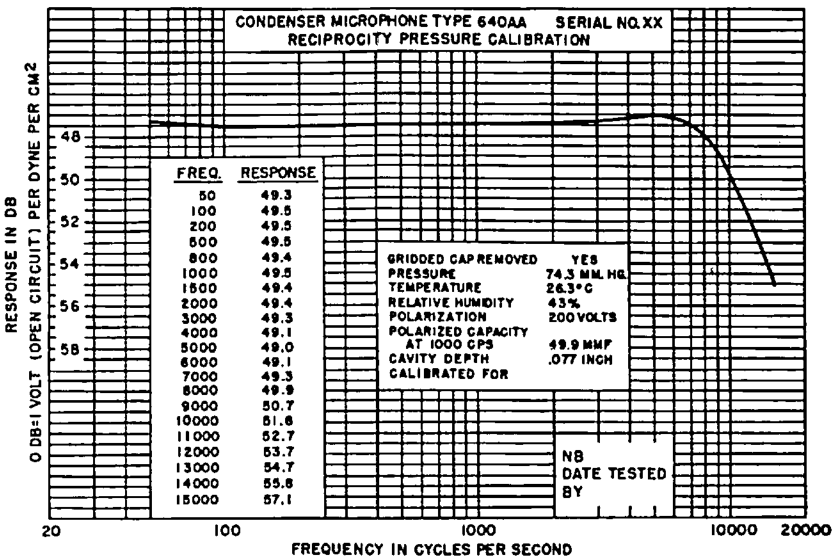


Fig. 11-11. Typical frequency response of calibrated condenser microphone.

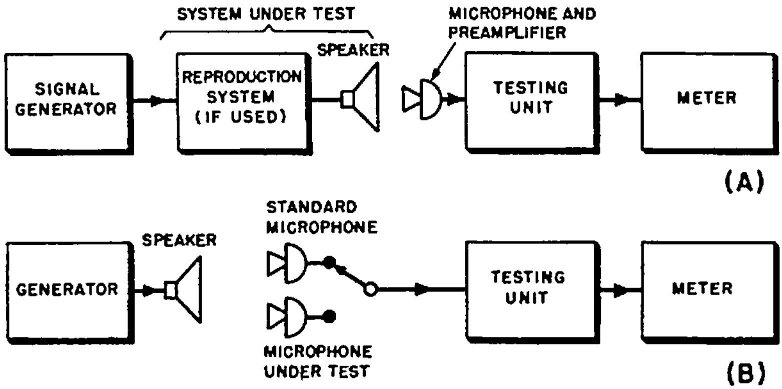


Fig. 11-12. Testing of reproduction systems involving direct sound pickup or reproduction. (A) Testing of a loudspeaker with standard calibrated microphone. (B) Testing sound pickup system by comparison with standard calibrated microphone.

has been calibrated against a primary standard sound source, and may be used as a secondary measurement standard. A calibrated microphone which has been widely used for this type of service is the condenser microphone. This microphone is effectively a "point pickup", therefore does not appreciably disturb the sound field, and has a typical frequency response (in combination with its companion preamplifier) as shown in Fig. 11-11.

The methods of measurement of acoustic devices with the aid of a calibrated microphone are illustrated diagrammatically in Fig. 11-12. Systems including a loudspeaker are tested by applying the input signal from the appropriate type of generator, and picking up the sound with the calibrated microphone and preamplifier (see Fig. 11-13). The electrical output from the microphone preamplifier is then tested for the desired characteristics in the normal manner by use of the measuring equipment which has already been described. Since the characteristics of the microphone are known, the characteristics of the reproducing system are readily determined.

When the system under test includes sound pickup by a microphone, it must be tested as shown in Fig. 10-13 (B). The test sound is produced by feeding the signal generator into a loudspeaker capable of reproducing the signal without excessive distortion. This sound is then picked up by both the microphone under test and by the standard microphone. Comparison of the output of the two microphones then immediately gives the characteristics of the unit under test.

Courtesy: Western Electric Co.

Fig. 11-13. Measurement of performance characteristics of a loudspeaker in an acoustically "dead" room which has been treated to minimize all reverberation to the greatest possible degree.

Systems which include mechanical and electromechanical methods of recording and reproduction, such as disc, film, and magnetic recording, also require special methods of measurement. (Of course, it is always possible merely to make a record from an applied electrical voltage, reproduce it and measure the resulting electrical voltage; but this procedure only gives information concerning the specific setup and does not tell anything about the individual units and their performance in more general systems.) To test the recorder and the reproducer individually it is necessary to have a standard of some sort. This may be either a standard record, recorder, or pickup, since any one may be used to calibrate the other two.

In certain measurements it may be necessary to use additional equipment (such as amplifiers and filters) which are not part of the reproducing system or of the measuring instruments. Any such equipment should always be tested first itself since errors in the test equipment necessarily set the limit of accuracy which can be attained in any measurement.

General Comments and Summary

If the various factors which affect reproduction quality are measured accurately and evaluated properly, a very good indication will be obtained of how well the system will reproduce any physical sounds. As the techniques of sound reproduction and measurement improved, it was found that the relative importance of many of the distortions had been misjudged and needed revision. As a result, the present trend is toward wide-range, low-distortion equipment rather than restricted-range, high-distortion equipment. Sound reproduction systems tested and rated according to this principle will correspond closely with the preferences of the human ear, which is, after all, the final judge and has up to now been the determining factor in acoustical progress.

The application of the tests described in this chapter has already changed some previous ideas concerning audio quality, particularly those concerning transient response. Some idea of their importance in actual practice may be obtained from consideration of their application in testing some specific audio systems. The results of measurements of a typical audio amplifier are shown in Figs. 11-14 and 11-15. These measurements were taken with and without inverse feedback, to test the quality of the amplifier and the difference with the feedback. The steady-state curves

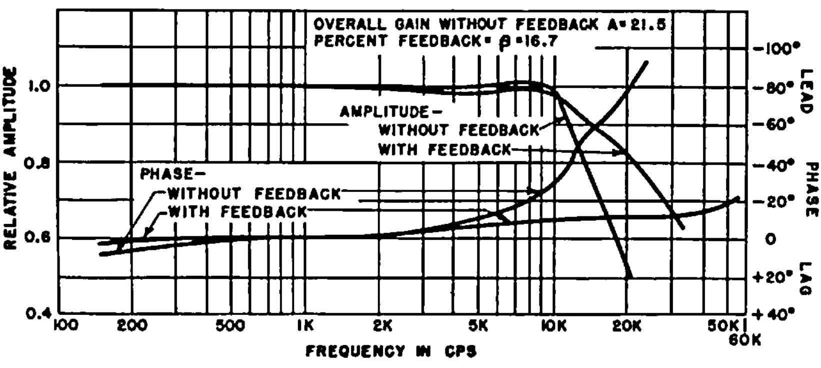


Fig. 11-14. Steady-state response measurements of an amplifier with and without negative feedback.

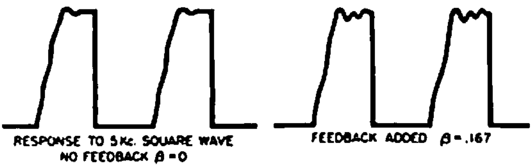


Fig. 11-15. Transient response measurements of the amplifier of Fig. 11-14 with and without negative feedback.

in Fig. 11-14 show that both the frequency and the phase response have been improved by the addition of the feedback. However, the square-wave response shown in Fig. 11-15 shows that the transient response has not been improved, and has actually become worse due to the increase in the damped high-frequency transient oscillation.

Such measurements are also of considerable importance in the testing of loudspeakers, and have considerably increased present knowledge of the factors which determine loudspeaker quality. For a long time, steady-state response and distortion measurements were taken as the criterion of loudspeaker performance. However, although these measurements are valuable in determining the bad resonances of inferior loudspeakers, they do not give the complete picture of the quality of reproduction to be expected from the better grades. Loudspeakers with similar steady-state distortion characteristics and substantially flat frequency response often sound quite different to the ear in listening tests. Since the steady-state measurements take no account of the transient nature of natural sound, the tests are incomplete unless the transient response of the speakers has also been determined. This may be done by the method previously illustrated.

The results obtained by measurement of the transient response in this manner are shown in Fig. 11-16, which shows the measured charac-

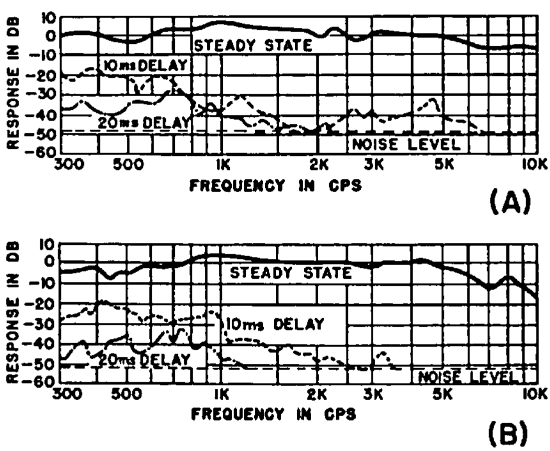


Fig. 11-16. Transient response measurements of two loudspeakers by the method of Fig. 11-7.

teristics of two similar loudspeakers. Under steady-state conditions the two speakers seemed very much alike, but they sounded quite different to the ear. The high-frequency response of the loudspeaker whose characteristics appear at (A) was found to be a little irritating after long periods of listening (exhibiting a roughness normally associated with intermodulation, but the speaker was known to be free from this type of distortion); with the loudspeaker whose characteristics appear at (B), this effect was not present. The transient response curves show that the loudspeaker in (A) has a longer decay time of residual vibrations than the loudspeaker in (B), and that at some points the output actually rises with time (suggesting the transfer of energy from one vibrating element to another during the decay period). Comparison between the aural effects and the results of the tests shows that the transient response gives a measure of quality of the system which cannot be obtained by steady-state measurements, and that the aural impressions were more related to the transient curve.

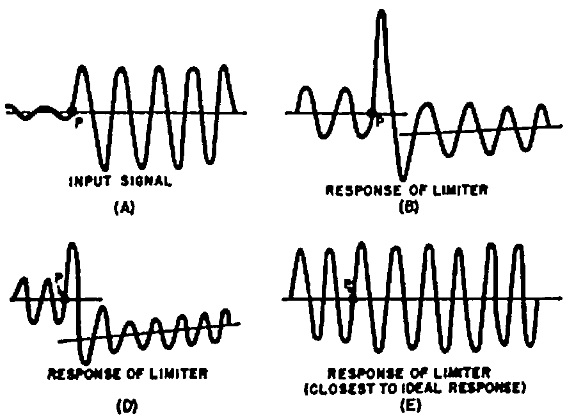


Fig. 11-17. Results of testing four different commercial peak-limiting amplifiers by the transient method of Fig. 11-7. Response E is nearly ideal.

In radio broadcasting and in sound recording, peak-limiting amplifiers are often used to prevent very loud signals from affecting the operation of the system by causing overmodulation or overcutting. Until very recently the characteristics of peak-limiting amplifiers were specified in terms of steady-state measurements. However, most users of such equipment know from their own experience that the performance of limiting amplifiers under actual operating conditions frequently has little correlation with that indicated by steady-state measurements, and is much more dependent upon the transient characteristics. Limiting amplifiers which have similar steady-state characteristics are often found to perform quite differently for speech and music. Transient measurements, such as those described in Fig. 11-7, must be included in the testing of such equipment to specify their performance adequately.

This is the recommended method of testing the transient operation of peak-limiting amplifiers, and a few results of such measurements upon various commercial units are given in Fig. 11-17. The input signal consists of a sine wave whose amplitude is periodically changed (at point *P* in the figure) between a lower and a higher level, the resulting output of the amplifier is then observed upon the screen of an oscilloscope. The response of a number of different types of peak-limiting amplifiers to this input signal is shown in Fig. 11-17. The results of this measurement are in agreement with the aural impressions obtained with these amplifiers, and indicate the value of this method of testing.

These few examples have been described to illustrate the importance of proper measurement of quality in audio reproduction systems. If the factors which determine the quality of reproduction are taken into account properly, and measured according to the methods described in this chapter, then the performance of the system can be completely described and its specification will have attained increased precision and accuracy.

INDEX

Acoustics,
- acoustic feedback, 180-181
- of rooms, 25-30
- (*see also*: Sound)

AES, playback response, 61
Amplification factor, 70
Amplifier,
- basic unit, 157-161
- circuits, 95-99, 157-171
- complete units, 157-171
- design procedures, 73-75, 95-98
- distortion in, 74
- driver amplifiers, 83-85, 87, 161, 165-166, 169-170
- equivalent circuit, 71-73
- feedback in, 77-78, 81, 100-101, 159-160, 166
- frequency response, 74, 96, 161, 171
- gain, 73, 159
- high-level, 164-166, 169
- high-quality high-cost, 166-171
- output impedance, 74, 99, 164
- power amplifier, 99-101
- power amplifier circuits, 103-105, 161, 165-166, 169-171
- preamplifier/control units, 162-164, 168-169, 177
- signal levels in, 95, 160-161
- testing and measurement of, 206-209, 222, 226
  - (*see also*: Measurements)
- transformer-coupled, 98-99
- voltage amplifier, 87, 95-99
- voltage amplifier circuits, 95-99, 164-171
- voltage amplifier stage, 68
- Williamson, 104-105, 167, 169-171, 177

Amplitude,
- of sound vibrations, 3-5
- range in speech and music, 16-17

Amplitude modulation (AM), 62-63

Baffles, 143, 144
- (*see also*: Enclosures)

Bass boost, 109, 115-118, 120
- (*see also*: Equalizers)

Bass compensation, 122-123
Bass-reflex cabinet, 148-152, 175, 178
Bias, magnetic, 193-196
- d-c, 194
- high-frequency, 194-196

Cathode follower, 81-83
Components, selection of, 171-178
Composite curves, 101-103
Crossover frequency,
- in disc recording, 52-53
- in loudspeaker systems, 141-143

Crossover networks, for loudspeakers, 141-143

Decibel, 10
Diffusion of sound, 26
Distortion,
- limits, 23
- measurement of, 207-220
- types of, 18-21

Dynamic range, 16-17, 19, 105, 125-129

Ear,
- response of, 9, 18
- structure of, 8-9

Enclosures, for loudspeakers, 143-156
- back-enclosed, 146-148
- bass-reflex, 148-152, 175, 178
- construction, 147, 150, 153, 155-156
- folded-horn, 154-155, 178
- "infinite-baffle," (*see*: back-enclosed, plane surface)
- labyrinth, 152-154
- open-back cabinet, 145-146
- plane surface, 143
- requirements, 143-144

Equalizers, 109-118
- constant-impedance, 112
- design curves, 115-118
- design of, 113-118
- impedance of, 111-112

matching into circuit, 110-112, 120, 159-160
tone controls, 118-121, 159
Equivalent circuit,
of amplifier, 71-73
of back-enclosed ("infinite-baffle") cabinet, 146-147
of bass-reflex cabinet, 149-150
of loudspeaker, 136-138

Feedback, 77, 159-160, 182-183
acoustic, 180-181
in practical amplifiers, 100-101, 159-160, 166
negative, 77,78
positive, 77,78
with tone controls, 160
Flutter, 20
limits of, 23
testing, 207, 219-220
Flutter index, 220
Folded-horn enclosure, 154-155, 178
Frequency correction (*see*: Tone Controls, Equalizers)
Frequency modulation (FM), 63-64
Frequency range,
distortion of, 19
of voice and instrument fundamentals, 15-16
requirements in reproduction, 22-23
Frequency response,
requirements, 22-23
testing, 207, 212
Harmonic distortion, 19
limits, 22-23
testing, 207, 213-214

Harmonics, 6-7
High-fidelity reproduction, 2-5, 18-24
definition of, 18-19
requirements for, 21-24
Hum,
in amplifier construction, 179-180
in preamplifiers, 89-90, 178
system layout, 178-181

Impedance,
matching in systems, 33
of equalizers, 111-112
of microphones, 41, 43
of pickups, 43, 57, 59
Infinite baffle, 143, 146-148
Instruments, musical,
stringed, 11-12
wind, 12-13
Intermodulation distortion, 19
limits, 22, 23
testing of, 207, 214-216

Labyrinth, 152-154, 177
Limiters, peak, 126
circuits, 128
testing of, 227-228
Load line, in amplifier stage, 70
Logarithmic frequency scale, 10
Loudness controls, 123-125, 168
Loudspeakers, 134-141
acoustic efficiency, 35-36, 137-138
coaxial, 141
construction, 136
dual-cone, 140
dynamic, 136
equivalent circuit, 136-138
frequency response of, 136, 138-139
limitations of, 134
measurement and test of, 222-224, 226-227
(*see also*: Measurements)
multiple loudspeakers, 139-141
power output, 34-36, 137-138
requirements for good reproduction, 135-136
selection of, 38, 134-136, 174-175, 176-177, 178
size, 138-140

Magnetic fields, 189-190
Magnetism,
induced, 191-193
principles of, 189-193
Measurements and test,
applications and results of, 225-228
equipment, 210-212
general, 182-187, 206-209

of sound and acoustic signals, 221-223
techniques, 209-210, 221-225
Microphones, 41-49
carbon microphone, 42-44
condenser microphone, 48-49
crystal microphone, 46-48
dynamic microphone, 44-46
measurements with, 222-223
sound cell microphone, 47-48
velocity microphone, 44
Miller effect, 75-77, 96-97
Musical scales, 13-15
chromatic, 14
diatonic, 13, 14
equally tempered, 14-15

NAB playback response, 61
Networks, 106-125
Noise, 19
definition of, 11
in preamplifiers, 89-91
in sound reproduction, 107
limits, 22, 23
reduction of, 107-109, 129-132
Noise level,
in tape recording, 204
testing, 207, 212
Noise suppressors, 129-132
circuits, 132-133, 168-169

Octave, 13
Overtones, 6-7

Phase, 5
Phase distortion, 20
Phase inverter, 81, 83-84
Phase response testing, 207, 219
Phase splitter, 82-83, 170
Phonograph records, 49-53
equalization, 53
grooves, 50
playing time, 52
processing, 49
reproduction of, 50
standards and characteristics, 51-53
Phonographs, 53-57
motor requirements, 54
record changers, 56-57
single-record turntables, 55-56
Pickups, 57-62
condenser, 58-59
crystal, 57-58
dynamic, 61-62
moving-coil, (*see*: Dynamic)
stylus wear, 51-52
variable magnetic reluctance, 59-61
Plate current characteristics, of tubes, 69
Plate resistance, of tube, 70
Power amplifiers, (*see*: Amplifiers)
Power output,
requirements, 34-36
testing, 207, 212
Preamplifier, 86, 87-95
circuits, 91-95, 162-164, 168-169
microphone, 91-92
phonograph pickup, 93-95, 157
Preamplifier/control units, 162, 168-169
Pre-emphasis, high-frequency, 107-109
Push-pull amplifiers,
advantages of, for power amplifiers, 100
composite curves, 101-103
design, 99-103
power amplifiers, 100-105
power amplifier circuits, 103-105

Record, (*see*: Phonograph records)
Record changers, 56-57
service of, 186-187
Record player mechanisms, service of, 186-187
Recording, disc, 49-53
constant amplitude, 52-53
constant velocity, 52-53
equalization and pre-emphasis, 61
recording process, 49-50
recording speeds, 51
Recording, magnetic, 188-205
circuits, 201-205
equalization and pre-emphasis, 199-201
heads, recording and playback, 198
recording speeds, 199
systems, 196-197
twin-track, 204
typical unit, 201-205
Reference level, decibel, 10
Reflections of sound in rooms, 25-26, 27

Reproduction systems, (*see*: Sound reproduction systems)
Resonances, room, 25-27
Reverberation, 26
Reverberation times in rooms, 28-30
Room acoustics, 25-30
  absorption of sound, 25-27
  loudspeaker placement, 27-28
  reflection of sound, 25-27
  resonances, 25-27
  reverberation, 26, 28-30
  sound power requirements, 34-36
  typical layouts, 27-28, 180-181

Servicing, 182-187
  equipment for, 210-212
  of complete systems, 184-186
  of new constructions, 182-183
  of record player mechanisms, 186-187
  of transducers, 183-184
Signal levels,
  in amplifiers, 160-161
  in systems, 33, 171-172
Signal-tracing, 183-184
Sound,
  characteristics of, 3-7, 11
  frequency and amplitude ranges, 15-17
  limits of audibility, 5, 18
  measurement of, 221-223
  reflections in rooms, 25-26, 27
  vibrations in air, 4-7
Sound reproduction systems, 30-40, 171-187
  assembly of, 180-182
  basic system, 30, 173-176
  construction, 178-180
  costs and economic factors, 36-38
  highest quality, 177-178
  impedance matching in, 33, 171-172
  integration of commercial units, 39-40, 171-172
  matching of components, 33, 171-172
  physical placement, 36, 38-40, 171-172
  power output requirements, 34-36
  recoid ng and reproduction, 32
  reproduction from recording or transmissions, 30
  selection of components, 171-173
  signal levels in, 33, 171-172
  testing of, 206-209
  troubleshooting and servicing of, 182-187
  using tuner/preamplifier unit, 176-177
Stylus, 51-52
  characteristics and dimensions of, 51
  materials, 51
  wear, 51-52

Tape recording, (*see*: Recording, magnetic)
Test equipment, 210-212
Threshold of audibility, 9, 16, 18
Threshold of feeling, 9, 16
Tone controls, 118-121, 159,
  (*see also*: Equalizers)
Transformer, 79-81, 83-84, 98-99
  transformer-coupled amplifier, 98-99
Transient distortion, 20, 23
  limits, 23
Transient response,
  sine-wave method, 218, 219
  square-wave method, 216-218
  step-function method, 216-219
  testing of, 207, 216-219, 226-228
Treble boost, 109, 115-117
  (*see also*: Equalizers)
Troubleshooting, 182-187
Tuners, 62-67, 176
Turntables, (*see*: Phonograph:)
"Tweeter," 141
  (*see also*: Loudspeakers)

Volume compression, 125, 126-128
Volume expansion, 126, 129

Williamson amplifier, 104-105, 167, 169-171, 177
"Woofer," 141
  (*see also*: Loudspeaker)
  v, 20
  limits, 23
  testing of, 207, 219-220