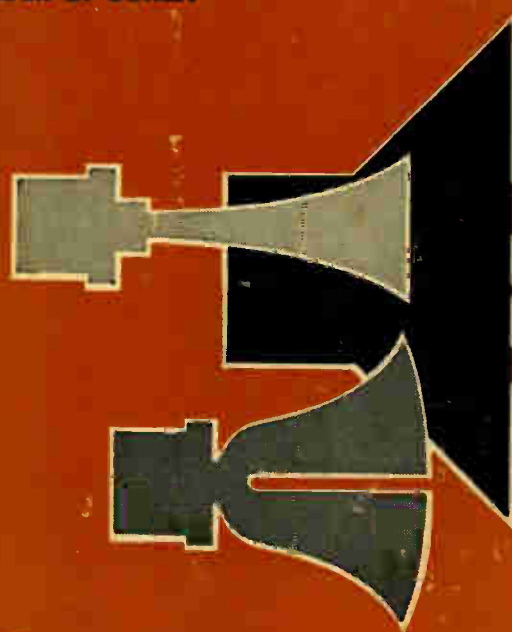
HI-FI LOUDSPEAKERS AND ENCLOSURES

by ABRAHAM B. COHEN



complete, authoritative, brilliantly written book that is estined to become a classic in the field of HI-FI literature!

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hi-fi Loudspeakers & Enclosures

Abraham B. Cohen, ENGINEERING MANAGER
UNIVERSITY LOUDSPEAKERS, INC.

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Preface

The many attempts to arrive at a rigid definition of high fidelity have all failed. They have failed for the simple reason that high fidelity is an art, and art defies formulation. The fact that nearly all high-fidelity program material is musical explains to a great extent why this art, once limited to the technician's domain, is now accepted by people in all walks of life.

No one is immune to music. We march to it, relax with it, are inspired by it, even rock and roll to it. It is this universal appeal that made success inevitable for any method of increasing the enjoyment of music in the home.

Because the writer considers high fidelity an art, he hesitates at proposing any all-embracing definition. It is our opinion that the use of such terms as "faithful," "realistic," and "lifelike" (and there is little agreement in defining these) can only be sanctioned on a relative basis. True, compared to the quality of reproduction of twenty years ago, that of today *is* lifelike and realistic. But this is not enough. We must delve beyond technical excellence, beyond the concept of realism, to primarily artistic considerations. We must ask, for instance, how well the composer's intentions are served.

We have perhaps been so overawed by the relative realism of high-fidelity sound reproduction that we have lost sight of the fact that the music undergoes many modifications, technical as well as artistic, before emerging as a finished recording. These modifications, along with those made by the listener through the adjustment of controls on the reproducing equipment, are the subject of the closing chapters, with the emphasis on the personal factor, which ultimately dictates the choice of components. This personal factor is an essential element in our consideration of loudspeakers, enclosures, room acoustics, and hearing characteristics as interdependent factors in an integrated whole, the sound circuit. The elements in this circuit are described briefly in the opening chapter; then, after the reader has had a chance to acquire a working knowledge of the components, they are integrated in the final chapter.

The reader is taken in gradual steps from basic loudspeaker principles toward an understanding of the variations used in specialized high quality reproducers. In our discussion of enclosures, we apply the same method, proceeding from the simplest types to the most elaborate structures in use today.

We have attempted to provide a comprehensive treatment that should be useful both to the "do-it-yourself" enthusiast and to the purchaser of a packaged system. The information gathered here is intended as an aid to the former in designing and coordinating the system that best suits his needs; to the latter we offer a better understanding and appreciation of the functioning of his equipment.

We acknowledge our debt of gratitude to scores of people, specifically to the staff of University Loudspeakers, Inc. A decade or more of close association with them has been fruitful in developing what we hope is a well balanced approach to acoustic problems. To the worthy competitors we owe thanks for their kind cooperation in providing much of the material presented herein. Finally, we wish to thank Mr. Milton S. Snitzer, Managing Editor of John F. Rider Publisher, Inc., for his appreciative reading of the text and his clarifying comments and advice.

White Plains, N. Y.
March 1956

ABRAHAM B. COHEN

Contents

To Ruth, who heroically kept the ambient noise level somewhere below the threshold of distraction,

to Paul, who nevertheless often broke through the parental sound barrier, and

to Michael, whose acoustic output was perhaps too severely damped, I gratefully dedicate this book.

PART 1

THE LOUDSPEAKER

CHAPTER 1: *The Sound Circuit*

The Loudspeaker: the Start of the Acoustic Circuit

Although the loudspeaker is the prime source of sound in any reproducing system, the sound we actually hear when we listen to a radio or phonograph is not entirely the result of the loudspeaker performance. We hear the result of many interacting factors, which constitute the subject matter of applied and practical acoustics. Not until the electrical signal (which is the counterpart of the original sound) acts upon the loudspeaker mechanism is the signal transformed into sound waves.

Contrary to popular belief, this transformation is not the end of the road to fidelity. The loudspeaker is only the beginning of one chain among others in the high fidelity circuit, but this chain, consisting of the acoustic circuit extending from the vibrating system of the speaker diaphragm to the nerve endings in the listener's brain, is obviously of more than passing interest. This is in fact the circuit most intimately and critically involved in the subjective or personal factor in the listener's high fidelity equation. The personal factor ultimately sets the standards that make a system sound good or bad *in his opinion.* This factor is an important element; it has made an art of the science of high fidelity.

We have jumped the gun somewhat in discussing one of the more personal elements of the acoustic circuit in order to emphasize the fact that there is more to the sound than just the original vibrations of the loudspeaker. We shall discuss these more personal factors in greater detail after we have examined the entire acoustic chain in orderly sequence. An understanding of this acoustic chain will be the first step toward a full appreciation of the scope that high fidelity sound reproduction must cover; this must be understood before an honest and practical approach can be taken to the *personal* high fidelity system problem.

Sound to "Signal" Transformation

The loudspeaker is the device that produces sound from a pure electrical signal. This signal has itself gone through many subsidiary chains before reaching the terminals of the loudspeaker. In these subsidiary chains there have been many different kinds of link (acoustical and mechanical, electrical, electronic, and electromagnetic, as illustrated in Fig. 1-1).

In even the simplest of broadcast linkages between the original performer in the broadcast studio and your radio antenna, there are involved at least these five distinctly different transformations of the original program. The voice of the singer has vibrated the air particles, setting up a wave motion in the air in the studio; this wave motion in air is acoustical in nature. The traveling acoustic wave impinges on the sensitive microphone, whose ribbon or diaphragm in turn is forced to vibrate in synchronism with the acoustic wave flowing past it. The delicately suspended diaphragm will move back and forth, riding the sound wave just as a cork will bob up and down in a water wave. Here is the first transformation. Acoustical energy has been changed to mechanical energy. The vibrating diaphragm with a small coil attached to it, vibrating in a magnetic field, generates an electrical current in its coil, just as the rotating armature in an automobile generator produces a charging current. This then is the second transformation, the change from vibrating mechanical energy to electrical "vibrations." The electrical current or voltage thus generated (depending upon the type of microphone being used) operates upon a whole chain of amplifiers utilizing vacuum tubes, which convert the electrical signal to an electronic signal (transformation number three). Other vacuum tubes in the transmitter impress these signals onto high-frequency electric (radio) currents (transformation number four). At the final stages of the radio transmitter the fifth transformation takes place: the electronic

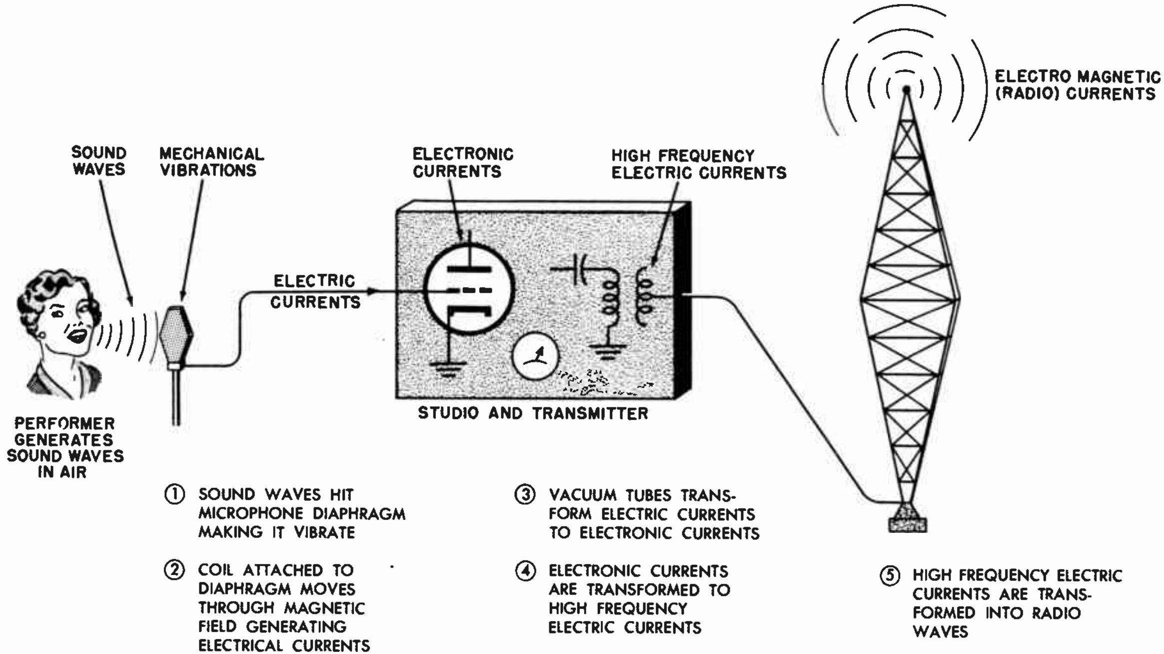


Fig. 1-1. The chain of transformations from sound waves to radio waves.

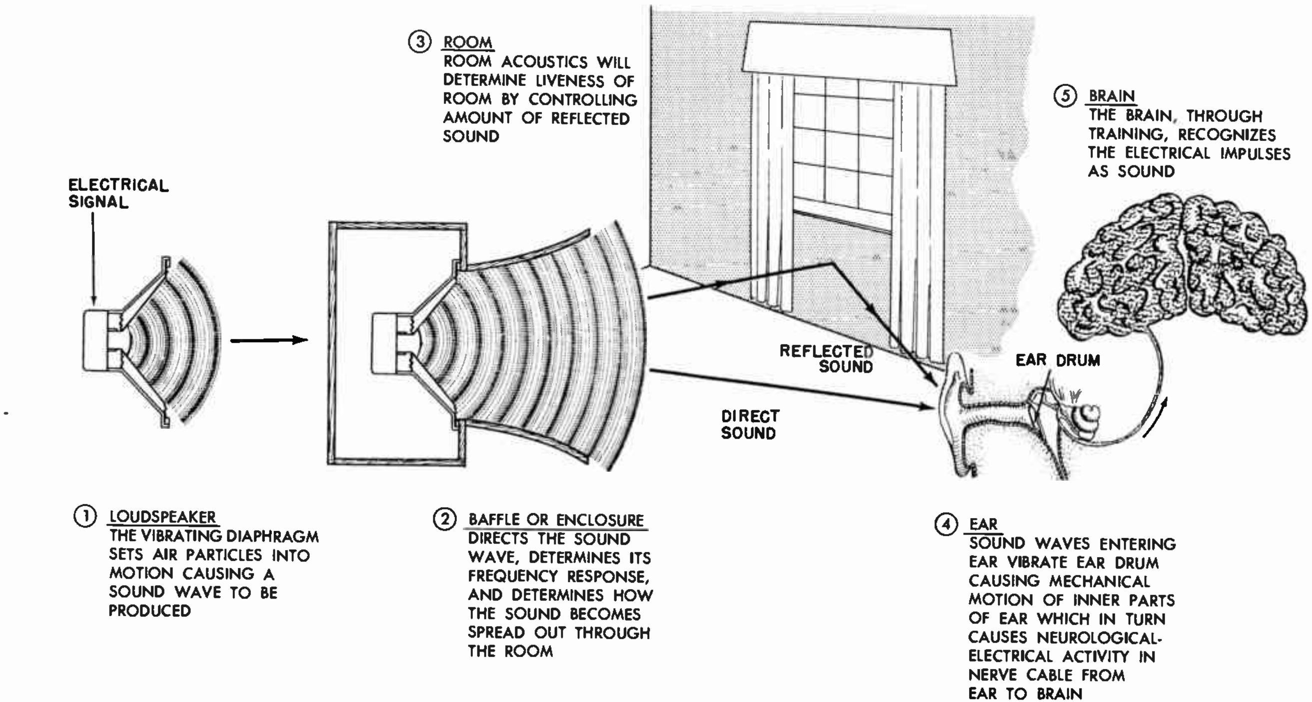


Fig. 1-2. The acoustic circuit from loudspeaker to "hearing."

signal is converted to an even different type of signal, an electro-magnetic wave motion, which then travels to your antenna with the speed of light.

As brief as this resume has been, it should be apparent that between the performer and the antenna of the home radio there are many technological bridges. In any event this is the chain that must exist before your radio "hears" anything. This same chain of events must be completed in reverse after your radio has received the space-borne signal before you finally recognize the signal as sound.

"Signal" to Sound Transformation

Rather than belaboring the point by describing the manner of reversal of this chain in your home apparatus, we shall proceed on the premise that an electrical signal finally finds its way to the terminal of your loudspeaker, and that this signal is in some general way representative of the original sound in the studio. The signal the loudspeaker sees is purely electrical in nature. This signal, "vibrating" electrically in accordance with the original signal, in turn causes the loudspeaker "motor" to vibrate in synchronism with it.

The Loudspeaker: Prime Source of Sound

Thus the loudspeaker mechanism, being energized to provide mechanical vibrations of its own, vibrates the air in contact with it and becomes the *prime source* of sound in the reproducing system. *The degree of fidelity of the reconversion of these electrical impulses to sound depends not only upon the loudspeaker itself and the preceding electronic circuits, however, but also upon the rest of the sound circuit, the other links in the chain, which impose other restrictive operating conditions upon the speaker.* This acoustic chain is illustrated in Fig. 1-2.

Nothing in nature works by itself without affecting something else; action and reaction are synonymous in the world of physical reality. The loudspeaker is a real physical object and it reproduces very real airborne sound waves. You cannot feel these sound waves under ordinary conditions, but you can feel the source that gives them life. One can readily feel the vibrations of the diaphragm (the paper cone of a loudspeaker) by placing the fingers lightly upon the diaphragm while it is reproducing sound. This is factual evidence that the diaphragm is mechanically in motion while it is making sound. It is often very easy to see the loudspeaker vibrations as well as feel them. When the loudspeaker is reproducing low frequencies of moderate intensities, the diaphragm may be clearly seen to be moving back and forth in its housing.

However, whether the eye can see or the hands feel the diaphragm of the loudspeaker move, move it does, and as it moves it reacts with something else. You will notice that when you put your finger even lightly upon the vibrating diaphragm of a loudspeaker the sound changes; your touch has modified the sound. The loudspeaker has reacted to your touch to change its mode of vibration. Suppose your finger does not touch the vibrating diaphragm. What is there for the diaphragm to react with? Quite simply, the air. It is always in contact with the loudspeaker. It always presents a definite physical *load,* as real as the weight of your finger, which the diaphragm must cause to vibrate.

Loudspeaker Reacts with Surrounding Air

Air has weight, as well as other acoustic properties (which we will discuss later) that affect the operation of a loudspeaker. The very fact that the loudspeaker makes the air in contact with it vibrate introduces an air reaction upon itself. The vibrating air acts upon the diaphragm to modify its original motion, and there perhaps is the key to the entire acoustic circuit concept — modification of one element of the chain by another, and modification of that element by the next, until in the end there is a complete chain. This is the acoustic circuit, in which the series of modifications finally comes into a state of equilibrum.

Let us take one more brief look at our chain before narrowing our focus to examine the links more closely. The vibrating diaphragm, because it is contact with the air around it, makes that air vibrate more or less in accordance with the original vibration of the loudspeaker diaphragm. But how well does this air vibrate? In an ordinary sense we might, with reasonable justice, say that if the sound is loud the air has been vibrated rather well; if the sound is weak, the air has been vibrated inefficiently. As general as this statement seems to be, it is nevertheless quite true.

"Baffling" Controls the Sound Intensity and Distribution

Is there anything that can be done to the speaker or the air to make an originally weak sound appear louder without altering the loudspeaker itself? Can it pull itself up by its bootstraps, as it were? The answer is an unequivocal yes. Without increasing the power input, a mere whisper (especially a low frequency whisper) may be transformed into a roar if we modify the surrounding elements with which the loudspeaker has to react. For example, if you are playing baseball, and you want to make yourself heard in left field, you do a very natural thing, something you learned long before you knew anything about acoustics. You cup your hands around your mouth and shout into a "megaphone" made up of your two hands, which shape the sound as it comes out of your mouth.

The fact is that you have "horn loaded" your mouth. You have produced certain modifications of the physical environment through which the sound had to propagate. In making these modifications, you have altered the degree of intensity (and directivity) of the sound. Not only have you made it louder, but you know intuitively that you have also directed it into a location where you want it most to be heard. You have "baffled" the sound, modified it, caused it do do your bidding.

Here then is the next link in the acoustic circuit. First the vibrating diaphragm and the air that is vibrated by the diaphragm, and now the baffle, which modifies the manner in which the air is vibrated, both in *intensity* and in *directivity*. The size of the baffle, its shape, and its construction will (along with other factors) determine how the air vibrations are modified. Our hands cupped around the mouth modify a shout to a degree; a cheerleader's long megaphone will modify it to a much greater extent. The baffle then is the first physical item outside of the loudspeaker itself over which we have some physical control. A good baffle is a battle half won in the struggle to perfect a high fidelity system.

The loudspeaker baffle or enclosure is the one determining factor for the final performance of the loudspeaker itself. The size of the baffle, its construction, and its actual shape will determine how well the loudspeaker will reproduce the low frequencies, how well high frequencies will be dispersed in the room, and even where in the room it should be placed for optimum performance.

The Sound is Modified by the Room

However, the sound has yet to reach our ear, and our consciousness. The diaphragm is vibrating, as is the air, with the baffle molding these vibrations in intensity and in direction. Some of the sound so transformed breaks away from the speaker-baffle combination, traveling straight to the listener's ear. More of the sound, however, travels as quickly to other parts of the room. In fact, only a small part of the total sound produced actually moves directly to the listener's ear, for his ear occupies only a tiny portion of the total physical volume of the room. The overall sound fills the room, every nook and cranny of it, although in different degrees.

It traverses the room, bouncing from wall to wall, and sooner or later some of it reaches the listener's ears. When it finally gets to the ear some of the sounds that were generated at the same time have already arrived.

Thus the ear receives an "echo" of the original sound. To put it more scientifically, the reverberant nature of the listening room will greatly affect the total make-up of the sound waves reaching the listener's ears. The "singer in the bathroom" knows the truth of this statement. The hard shiny walls reflect sound well, causing part of it to bounce around a good deal before it comes to the listener's ear. Let the same singer close himself up in a well-stocked wardrobe closet, and shout though he may at the top of his voice, he will sound "dead" because the room is dead. It has no life, no "reverberation."

The room condition then is the next link in the acoustic chain. The "liveness" of the room determines not only the apparent spaciousness of the sound but also what sounds will be re-enforced and what sounds will be absorbed. The amount and texture of draperies in the room, the wall textures, the amount of large untreated plaster surfaces, the thickness of pile of the rugs, and other structural and decorative factors affect the sound before the ear hears it. The listening room modifies the sound in intensity, in tonal characteristics, and in directivity.

Different Ears "Personalize" the Sound

Now the sound has actually entered the ear, but not the consciousness. Hearing is a very personal matter, and must be dealt with on a personal basis. You may have been born with large bone structures; I with small ones. You were born with certain anatomical features in your ear. I was born with the same general features, but in different mechanical proportions. The same sound that I hear, as it strikes your ear drum, may move that sensitively stretched membrane in a slightly different manner. Why? We may be of different ages; perhaps my ear mechanism has become more ossified than yours. Perhaps you have had a sinus condition that has affected your hearing mechanism, or perhaps you are a woman and have more sensititive physiological auditory reactions. Our hearing processes grow older, just as our hair grows grayer, and our hearing processes change accordingly. All these factors combine to produce different reactions within the inner ears of different individuals.

Not only are our ears different, but we have different neurological reaction times. We react differently to different stimuli. You might shout "ouch" sooner than your neighbor in response to the same unexpected pin prick, simply because your nervous system reacted faster to the stimulus. The pin jab actually caused minute nervous electrical currents to flow between your wounded finger and your brain. In a factually similar manner, the sound waves acting upon the auditory mechanism cause a transformation to be made from sound waves back to minute electrical (neural) impulses, which the brain then recognizes as "sound." Even if the actual ears of different listeners were identical, the listener's neurological activity would in the end determine what the brain heard; and that is what counts. It is the brain, the mind, the consciousness that determines whether one likes what his ears "hear." What a remarkable process this is! We started with an electrical signal in the loudspeaker and we end with an electrical signal in the nervous system.

CHAPTER 2: *Basic Loudspeaker Types*

The Dynamic Loudspeaker

Many different types of loudspeakers go into the making of high fidelity reproducing systems. However, the differences between the characteristics of various speakers are mostly a matter of degree. The actual functional differences between loudspeakers will be treated in Chap. 3. In the present chapter we shall deal with the basic theories of loudspeakers in general, in terms that will help later in judging their qualifications for specific hi-fi applications.

Perhaps the most popular type of loudspeaker today is the permanent magnet dynamic type. Because of its comparative simplicity of construction and design, the precision that may be built into it, the ease with which it is coordinated with other equipment, its easy adaptability to many different applications, and its comparative freedom from electrical trouble, the dynamic loudspeaker has found acceptance in all kinds of reproducing systems. It is found in the smallest pocket radios and is a major component of the most elaborate theater systems. Figure 2-1 is illustrative of the scope of applications of the dynamic loudspeaker.

One of the major factors contributing to the popularity of the permanent magnet dynamic loudspeaker is the fact that it has its own powerhouse. It requires no external source of power other than the signal power to make it operate. This powerhouse (the magnet) has a virtually eternal life. The permanent magnet of the dynamic loudspeaker, once it has been charged to full capacity by the manufacturer of the loudspeaker, retains its magnetomotive power almost unaltered forever, and unless it is subjected to severe mechanical shock, it cannot be drained or run down. It is always ready for action.

The first elements seen by the eye in the usual dynamic loudspeaker are the *basket,* or housing (with some sort of rear structure attached to it, which encloses the magnet circuit) and the paper *diaphragm* (or cone) supported on the front of the basket. But there is much more to a loudspeaker than this. Let us dissect such a loudspeaker assembly and lay bare the various components to get a working knowledge of the roles they play in the operation of the loudspeaker.

Basically the dynamic loudspeaker is made up of the following components:

a. The voice coil
b. The voice coil former
c. The centering spider
d. The magnet
e. The magnetic circuit
f. The diaphragm (cone)

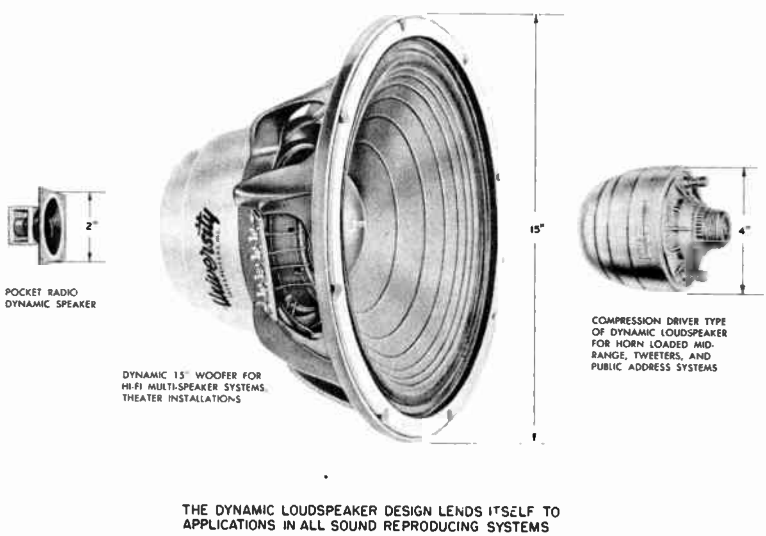


Fig. 2-1. Typical dynamic loudspeakers. (Courtesy University)

g. The apex radiator (and dust cap)

h. The basket (or housing)

Although many dynamic loudspeakers have other important components as integral parts of their design, these are the *basic* elements of the simplest type. Yet when properly designed this same group of elements, with no additions whatsoever, may constitute a high quality loudspeaker of special application. Figure 2-2 shows a full cutaway of a typical loudspeaker in which all these elements are clearly marked and arranged in a somewhat exploded view.

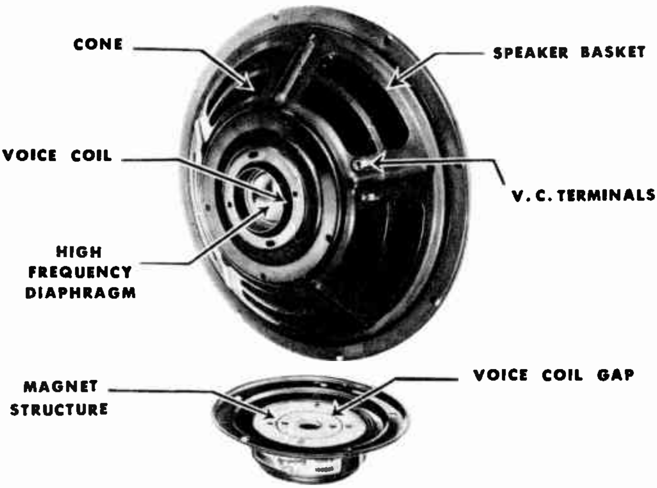
The dynamic loudspeaker is a device that functions with other electronic equipment, but in itself it has little that is of an electrical nature. The loudspeaker is simply an assembly of mechanical parts — a cloth spider, a paper cone, a metal basket, a magnetic piece of iron — to which is added a few turns of copper or aluminum wire (the voice coil).

The Voice Coil: Sole Electrical Element in the Dynamic Speaker

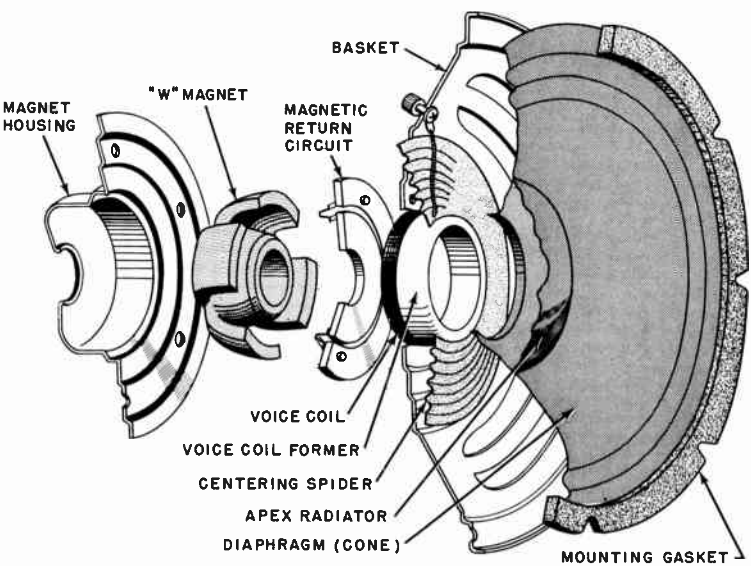
This voice coil is the only thing within the speaker that carries any electrical current or signal. It is energized directly from the amplifier, as illustrated in Fig. 2-3. The voice coil, as its name implies, is the part of the loudspeaker that does the "talking" by virtue of the fact that it is energized by the signal or "speech" currents fed to it by the amplifier. The voice coil consists of several turns of wire wound on a supporting bobbin. Depending upon the functional design of the loudspeaker, the coil itself may be either copper or aluminum wire, although insulated aluminum ribbon is also used. (In the case of the latter, the ribbon is wound on edge with the flat surfaces of neighboring turns adjacent to each other and all the turns held together by a binding cement.) The bobbin upon which the wire is wound (the voice coil form, or *former* as it sometimes called) may be made of a strong grade of thin paper, wound around on itself several times to provide a rigid cylinder. Sometimes the voice coil is wound on aluminum or duraluminum forms, and in some designs the forms are made of rigid paper, reinforced by an aluminum ring around the outer edge.

Voice Coil Must be Immersed in Fixed Magnetic Field

The voice coil doesn't do the amplifier's bidding all by itself. If this voice coil were isolated in free space it wouldn't make the slight-



(A)



(B)

Fig. 2-2. An exploded and cut-away view of a dynamic loudspeaker. (Courtesy University)

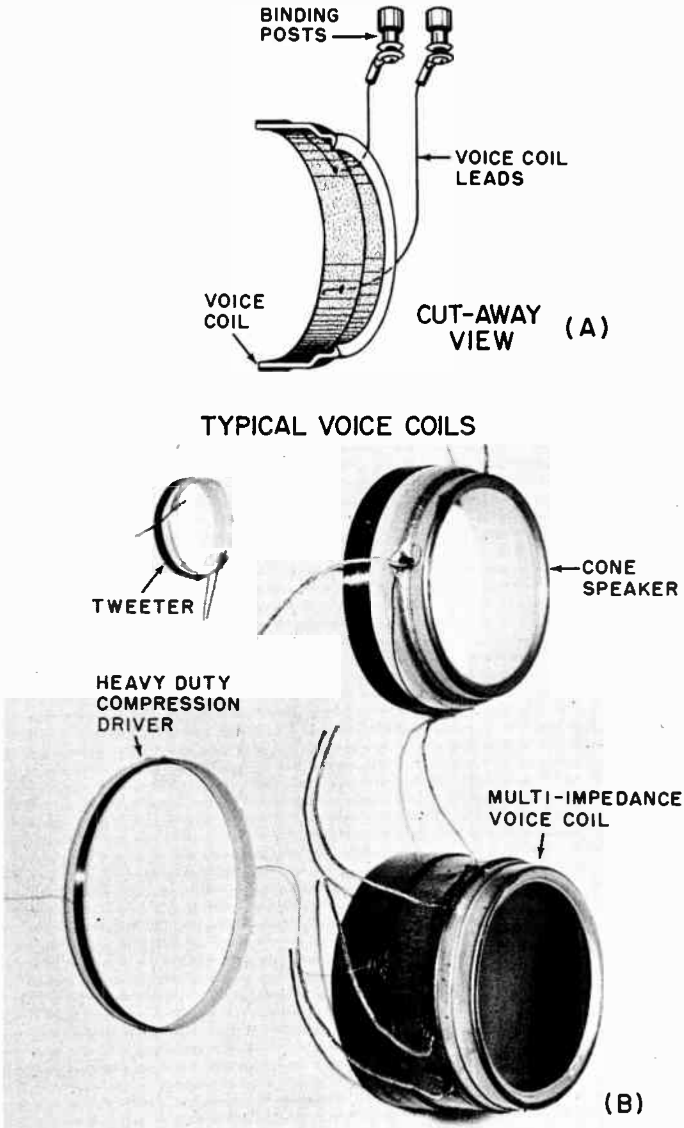


Fig. 2-3. The only electrical element in the basic dynamic loudspeaker is the voice coil, with its connecting leads to the basket terminal posts.

est sound, even if the signal current flowing through it represented the full power output of a 50-watt amplifier. In fact, not only would the coil be speechless; it would probably turn red and burn to a crisp in utter silence. It is not the input power alone that makes the loudspeaker work, nor is it the voice coil, or even the combination of

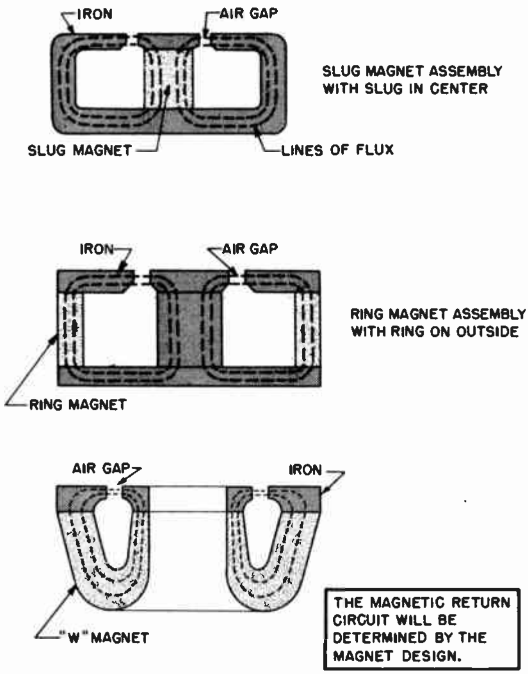


Fig. 2-4. The complete magnetic circuit consists of the magnet, the return keeper circuit, and the air gap in which the voice coil rides. The illustration shows cross-sections of three common arrangements.

the two. We have to add our "powerhouse" to the system, for it is the magnetic field of the permanent magnet that makes the voice coil speak. The voice coil of a dynamic loudspeaker can "speak" only when it is immersed in a magnetic field. Such a magnetic field may be produced by constructing an iron "loop" with the magnet in one section and an air gap in another, as illustrated in Fig. 2-4.

Once the magnet structure has been charged there is immediately formed in the magnetic circuit a *magnetomotive force* which drives *lines of force* across the air gap. The strength of the magnetic field set up by the magnet is naturally dependent upon the quality and size of the magnet, the size of the air gap, and the amount and quality of iron disposed about the magnet. These items are all factors that determine the figure of merit of the magnetic circuit, which plays a great part in the efficiency of operation of the hi-fi loudspeaker. Accordingly, the more specific and specialized details concerning such magnetic circuitry will be reserved for Chap. 3. However, we cannot continue without discussing magnetic circuitry in general, because this factor is fundamental to basic dynamic loudspeaker design.

The effect of the permanent magnet upon the performance of the loudspeaker depends upon the magnet composition, weight, and physical shape. There are several grades of magnet material, all possessing different magnetic properties. Magnet weight, without giving the grade of magnet specified, is a meaningless figure. Nearly all loudspeakers use magnets made of "Alnico." There are many grades of Alnico, each quite different from the others in magnetic properties. Very often a one-pound magnet of "Alnico V" may make a much better speaker than a five-pound magnet of "Alnico II." One must be careful, in examining the specifications of a loudspeaker, to consider the *combination* of grade of magnetic material and weight of the magnet, rather than either factor by itself.

Magnetic Circuit Plus Moving System Determine Performance

The truth of the matter is that not even these two factors give the complete picture as far as the entire function of the magnetic circuit is concerned. Our British cousins make an effort to give a figure representative of the "gap flux" — the strength of the magnetic field in the air gap — which signifies how well the magnet is working in its completed iron circuit. This figure, indicative of the strength of the magnetic field in which the voice coil rides, is intended to convey some idea of the loudspeaker performance. In a sense it does, but it does not describe *total* performance. As we shall see in Chap. 3, the actual level of performance of the loudspeaker depends on a complete interrelationship of magnetic circuit with diaphragm weight, composition, and shape; the size of the voice coil and whether it is made of copper or aluminum; and other mechanical features of the assembly. Therefore, rather than give some figure of magnet worth that is only partially descriptive of speaker performance, American manufacturers have limited themselves to the statement of the basic facts of magnet weight and type.

The various types of magnets classified as "Alnico" all contain iron mixed with various amounts of aluminum, nickel, and cobalt. (It will readily be seen that the term "Alnico" is made up of the first two letters of each of these elements.) The proportions of these elements, in conjunction with the other basic magnetic iron with which they are compounded, form the Alnico series. In general terms, the heavier the magnet and the higher it is in the Alnico series, the better the loudspeaker will function.

Magnets may be cast in many shapes, depending upon the design of the equipment in which they are to work. There is no standard or best shape of magnet, other than one that fits best what the design engineer is trying to build into his item as far as performance and cost are concerned. Figure 2-5 shows some representative shapes of magnets commonly found in today's loudspeakers. The simplest of these shapes is the slug magnet. The cored slug and the ring are a little more difficult to manufacture, and the W magnet is the most intricate of the lot. It is apparent that for a given weight of magnet material the cored slug, ring, and W types will be more expensive than the plain slug (and also larger in overall dimensions). The choice of a magnet for a particular speaker depends upon the use for which the speaker is designed and the particular properties required in the speaker. Figure 2-4 illustrates the typical magnetic circuits for the types of magnets shown in Fig. 2-5.

The magnetic circuit, comprising the magnet, the return circuit, and the gap, is completely dead magnetically until the magnet is charged. In practice it is never charged unless it is in its completed mechanical structure. As soon as the magnet is charged, it sets up a complete magnetic field, which reaches out from the top surface of the magnet, goes through the iron and through the air gap interposed in the iron circuit, and then returns to the other side of the magnet. The magnetic field (the lines of "flux" the magnet has been able to push out) is the "force field" of the magnet, for in this air gap rides the voice coil, ready to interact with the magnet field. For a fixed amount of magnet, the amount of flux that actually gets into the gap will depend upon the size of the gap and the quality of the iron from which the magnetic return circuit is designed. With high grades of irons, like "Armco," high values of magnetic flux may be obtained. Cheap grades

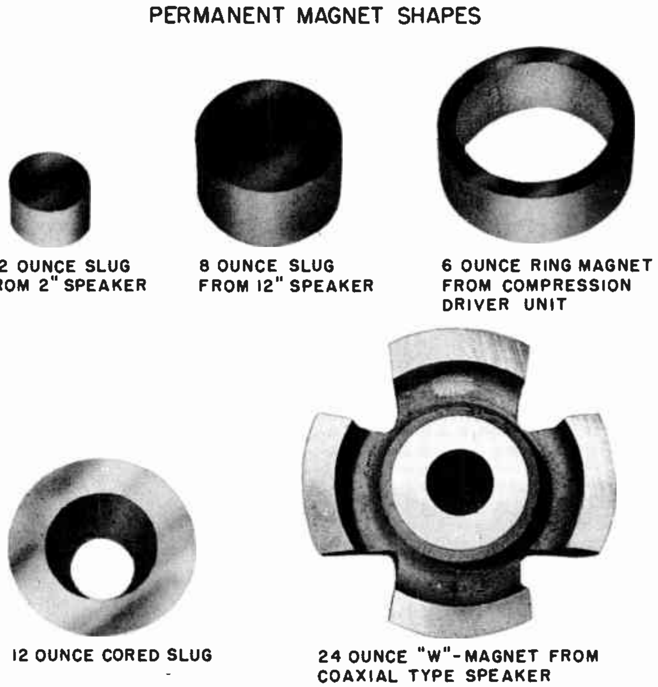


Fig. 2-5. The shape of the permanent magnet is determined by the performance and cost objectives of the loudspeaker.

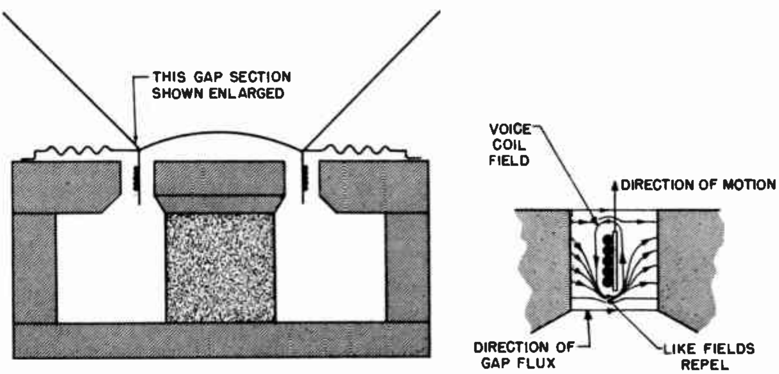


Fig. 2-6. Current flowing through voice coil sets up magnetic field around coil. Interaction is set up between this field and the permanent magnetic field, forcing it to move.

of iron are easily saturated; that is, they are not capable of carrying the full charge of the stronger magnets, thus the air gap flux is not as strong as in Armco circuits. Once the flux gets into the air gap, the magnetic circuit is completed and ready to make the voice coil do some useful work as soon as it gets an electrical "go-ahead" from the amplifier.

Voice Coil Field Reacts with Fixed Magnetic Field

This "go-ahead" is the electrical signal current, flowing through the voice coil from the amplifier. With current flowing through the voice coil, what we actually have achieved is the formation of a varying magnetic field, caused by the signal current flowing through the voice coil, in close proximity to the magnetic field of the permanent magnet, as illustrated in Fig. 2-6.

In the region where the fields are alike, there will be a strong concentration of magnetic flux, which tends to exert a repelling force. In the region where the lines of force are in opposite directions, there will be a cancellation and reduction in total magnet flux, which tends to exert an attracting force. *Motor action* will thus be developed between the field of the magnet, which is stationary, and the field of the voice coil. Because of the interaction of these two fields, one fixed and the other movable, the voice coil tends to move one way or the other (up or down in the figure), depending upon the direction of the current through the voice coil. (The direction of the current determines the direction of the voice coil field.)

The direction in which the voice coil travels is, in general, parallel to the length of the gap in which it is balanced. This problem of balancing the voice coil in the magnetic gap is an important one in loudspeaker design. The coil must be balanced both magnetically and physically within the gap. The importance and the manner of these balancing techniques will be discussed in subsequent chapters. For our present purposes, however, we may examine these balancing means with an eye to how, rather than to how well.

Voice Coil is Aligned in Gap by "Spider"

The voice coil is balanced in the gap by means of a device usually referred to as the spider. The spider, or centering device, as it is more accurately called today, holds the voice coil centered in the gap radially as shown in Fig. 2-7. The voice coil must be centered radially so that in vibrating in and out of the gap it does not strike the metal walls of the gap. To obtain high magnet efficiency the gap clearance between the coil and the iron is usually very small. (In the case of tweeters it

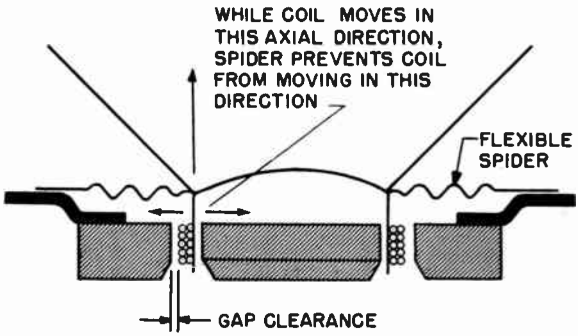


Fig. 2-7. The centering device (the spider) must keep the voice coil travelling in an unswerving axial direction in and out of the gap. It must keep the voice coil radially centered so that it does not hit the iron side walls of the magnetic circuit air gap.

may be only 3 to 4 thousandths of an inch; in the case of larger speakers, it may be 10 to 15 thousandths of an inch.) Thus the spider or centering device plays a very precise role in keeping things lined up as the voice coil vibrates in and out of the gap.

The centering device made up of the long twisted "legs" that attach to the voice coil (from which the name "spider" was derived) was one of the earliest types. Figure 2-8 shows some spiders commonly in use today. The open spider shown has a shortcoming in that the open area between the legs permits foreign particles to get into the small gap spaces. This may seriously hamper loudspeaker operation, and often causes considerable damage to the coil, especially if these particles are magnetic in nature, or large, rough, and coarse. In order to mitigate this defect of the open spider, fibrous material (commonly cotton) is frequently placed behind the spider, between it and the all-important gap area.

Another means of overcoming these problems in the cloth spider. This is made of a loosely woven material, impregnated in a thermosetting resin, such as a phenolic, and then formed under heat and pressure into the flexible structure shown in Fig. 2-8. The outer edge of this spider is permanently affixed to the basket, and the inner diameter suspends the voice coil in the gap. There is remarkable radial stability to this type of spider, and the built-in springiness of the waves molded into it allow for precision of motion of the voice coil in the

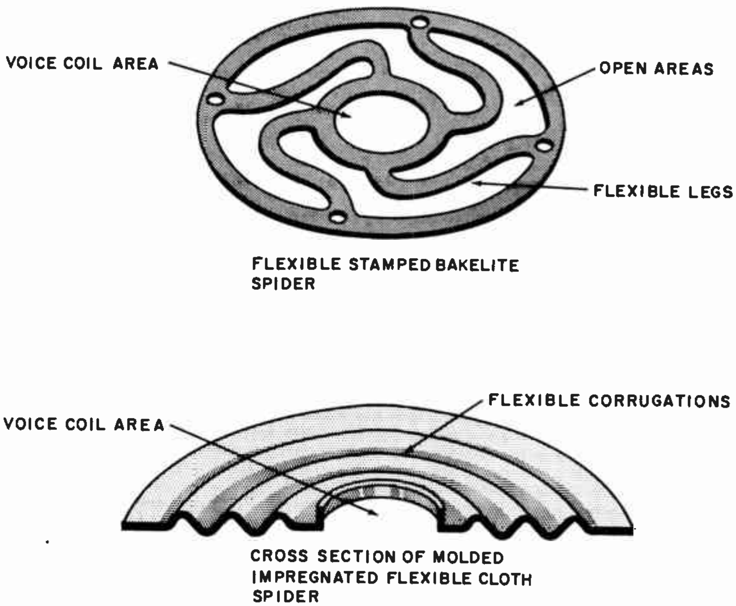


Fig. 2-8. Some typical voice coil centering devices (spiders).

axial direction. By means of devices like these spiders, the voice coil is kept to a true and rigidly aligned direction of motion in and out of the magnetic gap.

Voice Coil Moves Diaphragm

To the voice coil is attached the diaphragm, which actually "fans" the air into motion. In Fig. 2-9 are shown some typical loudspeaker diaphragms. These are usually made of special paper or impregnated cloth, although metal has also been used. Because the diaphragm is directly fastened to the voice coil, it must follow in a rather exact fashion the vibrational pattern of the voice coil itself. A great deal of modification of this original vibratory motion may actually be introduced between the voice coil and the diaphragm, for reasons we will discuss later. These modifications fortunately give rise to designs for specialized speakers for hi-fi reproducing systems. We may nevertheless accept the proposition that in general the diaphragm (or cone) of the speaker moves back and forth, impelled by the voice coil to do

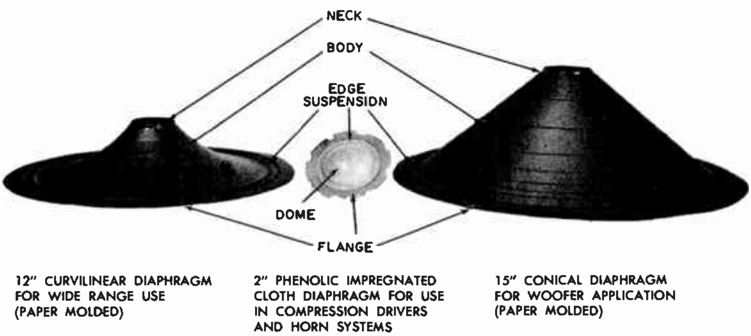


Fig. 2-9. Typical diaphragms with specialized shapes for different applications.

so, in a manner representative of the actual signal current. As the diaphragm vibrates back and forth, it causes the air with which it is in contact to vibrate in exactly the same manner, and sound waves are then created.

In order to allow the diaphragm to vibrate back and forth freely, it is necessary to provide it with some sort of flexible support that will allow it to have motion and yet keep it vibrating in a true axial direction. The diaphragm is provided with a flexible area at its outer edge sufficiently compliant to allow the diaphragm to flex in and out. In the great majority of present-day speaker structures, this flexible area is actually part of the diaphragm itself, and it acts as a spring that suspends the diaphragm in a state of equilibrium. This area of the diaphragm is usually referred to in acoustic terminology as the rim compliance of the loudspeaker; it plays an important part in controlling the characteristic performance of the loudspeaker. In Fig. 2-10, this

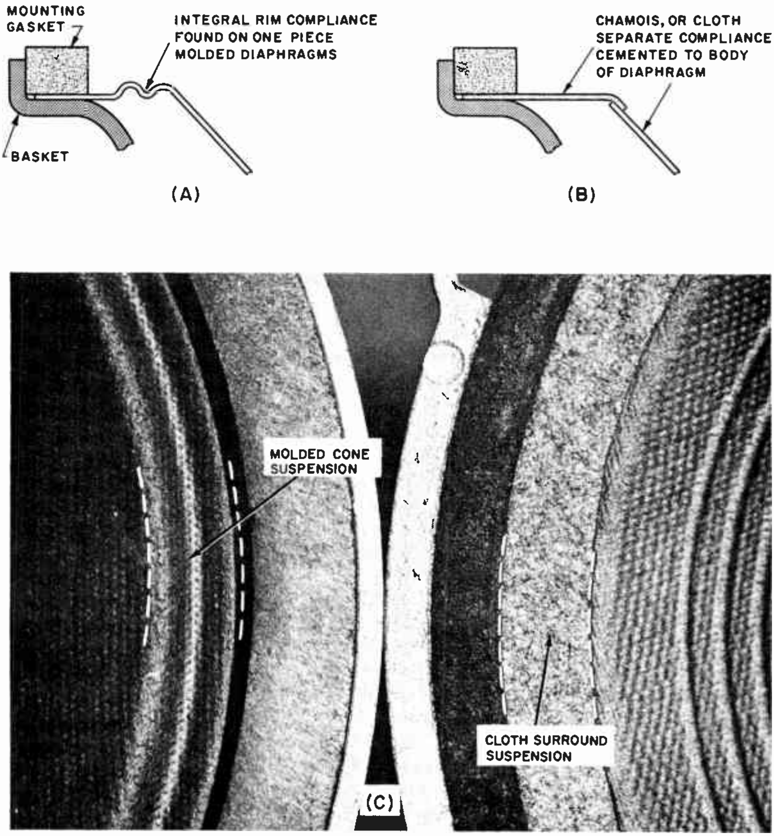


Fig. 2-10. Forms of diaphragm suspension.

integral rim compliance is illustrated along with other means of accomplishing the diaphragm suspension.

Sometimes the rim compliance is accomplished by providing an annular ring of soft chamois leather, which is cemented both to the basket edge and to the paper diaphragm. When made in this way, it is sometimes called a skiver, since skiver, according to Webster, means "a piece of sheepskin." Sometimes the term skiver is improperly used to signify any type of rim compliant material cemented to the cone, such as a loose cloth (which, although as compliant as the chamois, may have undesirable side effects). A general term sometimes used for this edge compliance is the "surround," which is quite obvious in its meaning. It literally surrounds the speaker.

Many types of diaphragm go into the making of the dynamic loudspeaker. The diaphragm may be made of paper, wool, and other fabric components. This type of diaphragm is usually found in the "cone" type speaker, a type of speaker in which the large radiating diaphragm is clearly visible to the naked eye. This is the type most commonly used as a general purpose speaker. However, for the type of speaker in which specialized performance may be required, such as the compression type driver unit, the diaphragm may be made of molded phenolic, or even duraluminum. This type of diaphragm is seldom seen by the user, because it is usually sealed into a comparatively small head structure that contains other important acoustic elements requiring it to be sealed off acoustically except for its throat. Such a unit may usually be recognized by the fact that it is connected at its throat to some type of horn. However, despite the fact that in external appearance there may seem to be no similarity between the open cone type speaker (the direct radiator) and the closed compression type (to be used with horn radiator), the mode of operation is exactly the same for both. They differ in degree of performance, in frequency coverage, and in several other ways; but the manner in which they set the air in motion (through their respective magnet structures, voice coils, and diaphragms, along with auxiliary devices such as centering means) is functionally the same for both types.

Diaphragm Apex Important for High Frequencies

Due to manufacturing contingencies peculiar to the means of assembling the various parts of the speaker, the apex area of the cone is usually open. Before the speaker may be considered completed, this open area must be closed off for several good reasons. From this apex area of the cone comes the greater proportion of the high frequencies of the cone. Every possible surface in this section should be utilized as an active part of the diaphragm; every square inch in this area should "push air."

If this apex were left open, there would be less active diaphragm area in the apex, and less air would be "pushed." Therefore, in the general type of loudspeaker, there is usually added some sort of apex "dome" to the cone. However, in some instances, this apex "dome" consists simply of a loose piece of felt across the apex opening placed there to keep dust particles out of the sensitive gap area, and yet allow the speaker to "breathe." Free flow of air in and out of the rear of the apex of the cone is necessary in some speaker designs to release the captured air cushioned in these rear areas, which might restrict the free rearward motion of the diaphragm. This situation is usually found to be the case in small general purpose utility type speakers, and usually a piece of felt is placed across the apex to provide the breather. In the more specialized speakers, where large magnet assemblies are the rule and large internal volumes exist behind the voice coil, this air cushion in the rear sections is greatly minimized, so that "breathers" as such are relatively unnecessary. More functional devices may then be designed into the apex areas to provide better high fidelity performance. These specific details will be covered in Chap. 3.

Basket Supports Entire Structure

The various elements of the loudspeaker are all supported by the housing, or basket. Depending upon the size of the speaker and its weight, this basket may be made either of cast metal or of stamped sheet metal. Whatever it is made of, however, it must be rigid in construction so that it may adequately support all the parts of the speaker without deforming under the weight of the magnet or by being screwed to irregular or warped baffle boards. Any mechanical weakness of the basket structure will in time result in damaged voice coils, distorted acoustic performance, and shortened life. These factors will be treated in more detail in Chap. 4.

Dynamic Speaker is Current Sensitive: Low Impedance

Because the dynamic loudspeaker is actuated by current flowing through the voice coil, this type of loudspeaker might be called *current sensitive.* For an electrical device to be current sensitive it must present very little resistance to the flow of current through it. More specifically, it must have low impedance. Therefore, the dynamic loudspeaker falls in the low impedance class. Typical impedance values range from 2 to 16 ohms.

The Electrodynamic Loudspeaker

Although we discussed the permanent magnet dynamic loudspeaker first, it did not come first in technological development. The reason for this is that magnet materials available in the early days of audio devices were not very efficient in comparison with other means of obtaining a magnetic field within the gap. With the advent of better alloys of permanent magnet materials, these deficiencies in magnetic

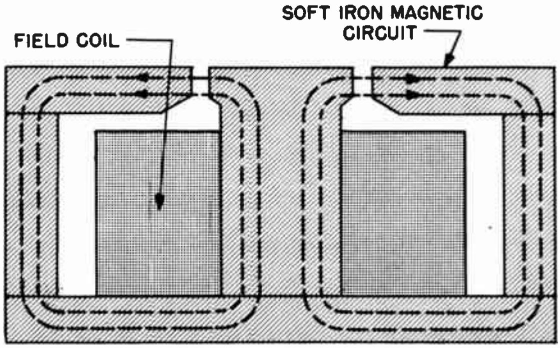


Fig. 2-11. Field coil, carrying current, sets up magnetic field, which magnetizes iron circuit and produces flux in the air gap only while current flows through coil.

circuits were gradually overcome. But until they were, the type of loudspeaker that had widespread popularity was the "elecrodynamic," a type of speaker that even today enjoys widespread use in certain applications. The electrodynamic speaker and the permanent-magnet (p-m) dynamic speaker function in exactly the same manner. The parts are identical with one very important exception.

Magnetic Field is Electrically Energized

In place of the permanent magnet found in the p-m dynamic speaker, there is instead a completely "soft" iron circuit. Around the center leg of this iron circuit is wound a large multi-layer coil, as shown in Fig. 2-11. When a direct current is made to flow through this "field" coil, it sets up a magnetic field about itself, which magnetizes the iron circuit within the coil. A field of magnetic flux is thus set up across the air gap of the return keeper. The strength of this field is a direct function of the strength of the current that flows through the coil, and the design of the iron circuit. With proper magnetic circuit design it is possible to get large values of gap flux, which means a more powerful speaker. The current through the coil does not permanently charge the magnetic circuit, however, since the circuit has no permanently magnetizable material in it. The "soft" iron of the magnetic circuit becomes magnetized and stays magnetized only while there is

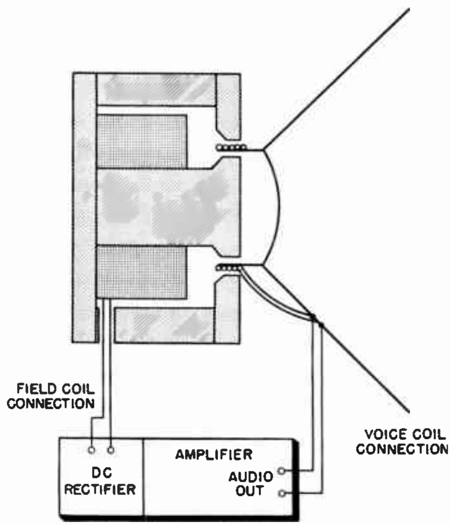


Fig. 2-12. The electrodynamic speaker must have its field coil connected to a source of direct current, such as the power rectifier that feeds the amplifier, or to a separate power supply.

magnetizing current present. Once current is shut off, there is no magnetism. The energizing current is obtained from the same power supply that powers the associated amplifier or radio equipment.

The manner of connecting an electrodynamic loudspeaker is shown in Fig. 2-12. This type of speaker is in widespread use in auto radios even today because of some economies it effects in actual power supply equipment for the dashboard radio. In cases where it is desirable to eliminate bulky filtering elements from large power supplies, such speakers may find usefulness, because of the filtering action of their field coil upon the power supply.

Electrodynamic Loudspeaker Requires External Power Supply

The speaker does have disadvantages, obviously. It must be rather close to the amplifier with which it is working so that it may be efficiently interconnected with the power supply. An even more serious fault is that only one such speaker may usually be connected to one combined power supply and amplifier. The continued efficiency of the loudspeaker is naturally dependent upon the efficiency of the power supply itself. As power rectifier tubes lose their output, the energizing current drops, and the speaker output falls off. Furthermore, special hum bucking coils are required in the structure to eliminate residual hum in the speaker that cannot be completely filtered out by the field coil. And finally, the field coils themselves may burn up or short out due to various operating malfunctions of the system, or because of the drying out of the coil insulation due to heat and age.

It should be realized that in the electrodynamic loudspeaker the field coil has absolutely nothing to do with the voice coil. The two are distinctly different both in function and in design. The field coil produces the steady magnetic field, the voice coil produces the varying signal field. Connections from the field coil are made directly to the system power supply, while the voice coil connects to the amplifier signal output terminals exactly as in the permanent magnet dynamic loudspeaker. Because of these problems, because of the dependency upon the efficiency of the power supply, and because of the possible inherent failures of additional electrical components, the electrodynamic speaker is seldom found in hi-fi installations.

The Electrostatic Loudspeaker

Of recent interest but ancient vintage is the "electrostatic" speaker, also called the condenser (capacitor) loudspeaker. Actually, the condenser loudspeaker was first pioneered after it was noticed that poorly made glass plate condensers had a tendency to "sing"; to actually make sound in some likeness to the signal they were carrying (in the same manner as transformers, which have a tendency to "sing" when their laminations are loose). Very recently, the electrostatic loudspeaker has attracted some attention. In the form in which it has been popularly offered, it affords a means of providing a rather economical tweeter (high-frequency speaker). This puts the smaller, more economical "ready-made" systems into the "two-speaker" class.[1] The word system is used quite advisedly, because as we shall soon see, not only is the electrostatic loudspeaker integrally married to the power supply of its driving amplifier, but it must be connected to the amplifier output tubes through specific coupling circuits that best match the electrostatic speaker to the particular tubes in the output stages of the amplifier; furthermore, it must be matched with auxiliary speakers of comparable levels of efficiency.

In the case of the dynamic loudspeaker, electrical energy is transformed to mechanical motion by the forces of magnetic attraction and repulsion. In the case of the electrostatic speaker the electrical signal is

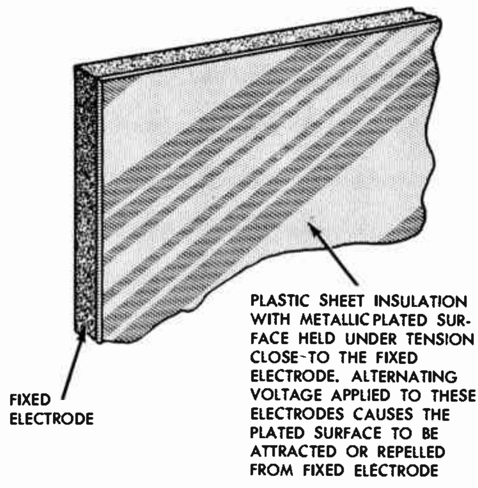


Fig. 2-13. Basic structure of the electrostatic speaker.

transformed into mechanical motion through the forces of electrostatic attraction or repulsion. Basically, the electrostatic loudspeaker consists of a movable electrode held very close to a fixed electrode, but insulated from it. A simplified structure of such a speaker is shown in Fig 2-13. A unidirectional "polarizing" voltage, fixed in magnitude, is applied

[1]Although it has been used most widely in low-cost commercial units, the electrostatic speaker is emerging as a high-quality "custom" component in several versions. One of these, the Janszen tweeter, uses a push-pull arrangement (one movable electrode between two fixed electrodes) to overcome the nonlinearity inherent in the two-element type.

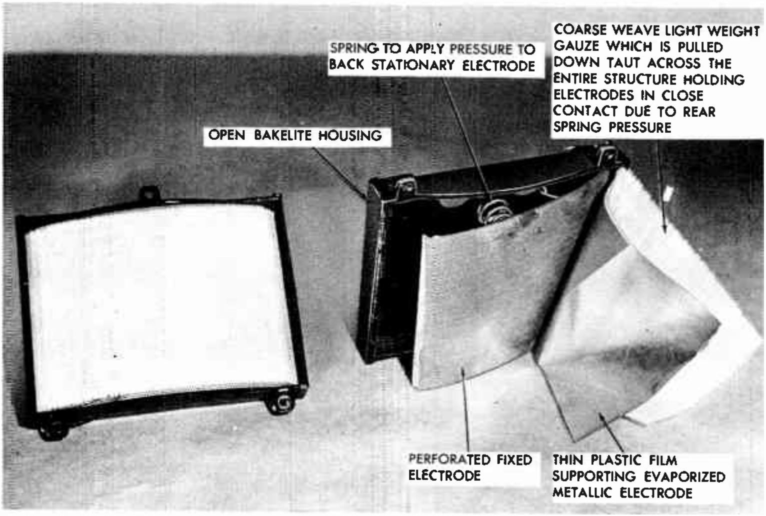


Fig. 2-14. One form of electrostatic speaker.

to the two plates, and the signal voltage is superimposed upon this polarizing voltage. The purpose of the polarizing voltage is to prevent "frequency doubling" by the loudspeaker. Frequency doubling is what occurs when a note is reproduced an octave higher by the loudspeaker, which would hardly be hi-fi! Because of the necessity of providing this high biasing voltage, it becomes mandatory to tie the speaker into some already existing power supply, or to provide it with its own power supply.

In Figs. 2-14 and 2-15 are shown the actual internal construction of two typical electrostatic tweeters. In the former (Fig. 2-14) the movable electrode (the thin piece of plastic coated on one side with a conductive material) is held in direct contact with an arched and perforated screen. Against the back of this screen is applied the pressure of two springs. This spring pressure, applied to the perforated screen, which is the fixed electrode of the speaker, keeps the electrode taut against the plastic sheet, which carries the movable electrode. Thus the tautness of the perforated screen is imparted to the plastic film and its superimposed electrode. Stretched across the whole assembly is a fine piece of gauze, which covers the movable electrode and holds the whole assembly in place.

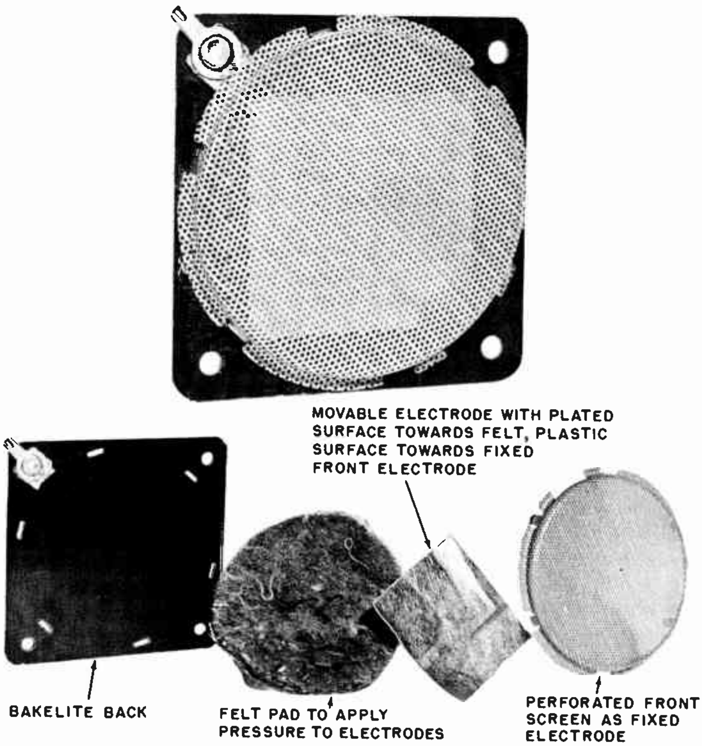


Fig. 2-15. Another form of electrostatic speaker.

The speaker shown in Fig. 2-15 is, in concept, essentially the same device shown in Fig. 2-14, but with a different mechanical arrangement. In this case, the plastic film is forced against the perforated fixed screen by the pressure of a somewhat resilient pad of felt, which is in turn backed up by a bakelite wafer supporting plate. On the felt side of the plastic sheet, we again find the thin metallic plated surface, which is the movable diaphragm.

Variable Electrostatic Attraction Between Electrodes Causes Diaphragm Motion

The basic principle of operation of the electrostatic loudspeaker is as simple as that of the dynamic type. If we let the matter of the polarizing voltage go for the moment, we may examine the way the speaker operates in its simplest form. Consider the two plates of the capacitor of Fig. 2-16 uncharged at the start. Now let us apply a slowly varying alternating potential to the two plates. As the difference of potential between the two plates increases from an initial value of zero, there will be a force of static attraction between the two

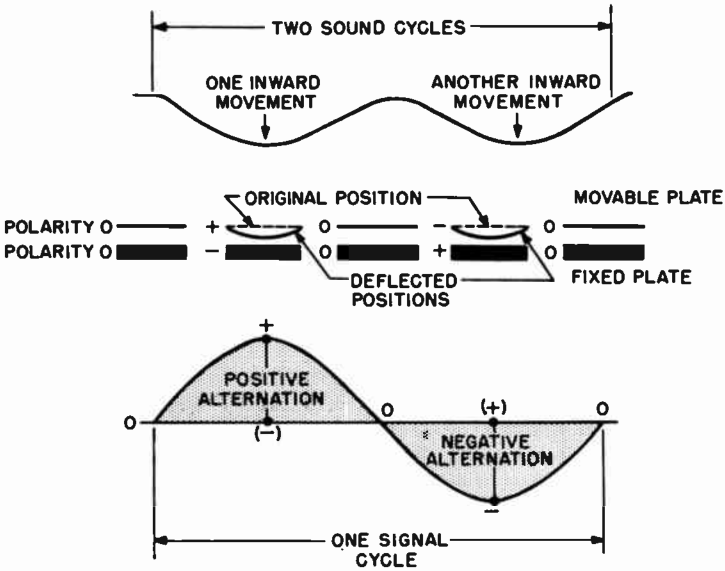


Fig. 2-16. When electrostatic speaker is not polarized the movable electrode is attracted to the fixed electrode twice for each cycle of signal voltage. This produces "doubling" of the signal.

plates, growing larger in value as the voltage difference between the two plates grows larger. The movable plate will consequently be attracted to the fixed plate and tend to move toward it. When the voltage reaches its crest and starts to decrease in value, the electrostatic force between the two plates decreases, and the movable plate returns to its original position as the voltage reduces to zero. Thus one pulse of air motion has been produced in an inward direction for a positive alternation of the signal.

However, the audio signal is an alternating voltage that has a negative half cycle along with the positive half cycle. Thus, after the applied signal voltage has returned to zero following its positive half cycle, it goes into its negative half cycle, again imparting its charges to the two plates but in opposite polarity. The plates, however, will *again be attracted to each other* because, despite the fact that the charges on them have been reversed, they are still opposite in polarity to *each other* and the force between them is still one of attraction. Therefore, the movable plate will again be pulled in toward the fixed plate, and will subsequently return to its equilibrium condition when the signal voltage has again reached zero. Thus, for *one* complete alternating cycle of electrical signal voltage consisting of a positive and negative alternation between the two plates, *two* identical pulses or cycles of air motion have been produced, with the air pulses moving in the same direction. This means that if a signal of 1000 cps were applied to the unpolarized electrostatic speaker as shown, there would be a sound output at 2000 cps. This phenomenon is called *frequency doubling.*

Polarizing Voltage Needed to Prevent Frequency Doubling

Frequency doubling is of course highly undesirable in sound producing devices. Naturally, we want to hear middle C if the piano is playing that note and not a note one octave above middle C. Frequency doubling also may exist in other types of reproducing systems. However, in such cases, it is usually corrected through proper initial design of the components and is not an inherent characteristic of the speaker. The electrostatic speaker system is theoretically and essentially a voltage doubler device, and no amount of proper design of the speaker itself can remedy it. What must be done is to provide some *external* factor that will prevent the doubling effect; this is accomplished by the polarizing voltage. The polarizing voltage (in the order of several hundred volts) is a fixed unidirectional voltage applied to the two plates of the capacitor to provide a sort of mechanical bias, as shown in Fig. 2-17. This fixed high voltage between the two plates exerts a steady electrostatic force of attraction between the two plates so that, even with no signal voltage applied to them, the movable plate is already attracted to the fixed plate to a considerable degree.

As the positive alternation of the signal is applied to the two plates, the movable electrode is drawn in to the fixed plate beyond that position where it was held simply by the polarizing voltage. As the signal voltage returns from its positive value toward zero, the attraction between the two plates is reduced to the original pre-strained position. As the signal voltage reverses its direction, an entirely different situation comes into being. This time the polarity of the signal voltage, which assumes a direction opposite to that of the polarizing voltage, has the effect of *reducing* the polarizing voltage between the two plates, and the force of attraction between the two is now decreased below the value produced

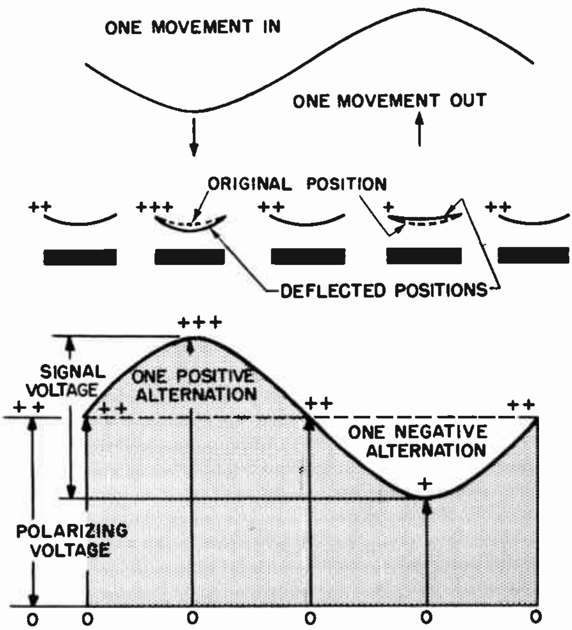


Fig. 2-17. When polarizing voltage is applied between plates of electrostatic speaker, the movable plate takes an initial set. As the signal voltage is applied this set is increased or released depending upon the signal polarity fluctuation, and doubling is eliminated.

by the polarizing voltage. The amount of release from the pre-strained condition that the movable diaphragm experiences follows precisely the swing of the signal voltage. The end effect of such action (positive signal alternation working in the same direction as the polarizing pull and negative alternation working in a direction opposite to the polarizing pull) is to pull the movable plate in beyond its fixed pre-strained position and then allow it to swing out away from that position. Thus *one* *alternating* cycle of air pulses is produced by the movable electrode, following the original single electrical cycle — voltage doubling has been eliminated by the polarized voltage.

Electrostatic Speaker Not Suitable for Low Frequencies

Because of the inherent mechanical limitations of the movable diaphragm of the electrostatic speaker, this type of speaker is not adaptable to wide range reproduction by itself. A self-sufficient speaker must produce both low and high frequencies. For the reproduction of reasonable low frequencies, the vibrating diaphragm must experience large motions, and must be large in size. Such large diaphragm motions are not possible in the typical structures illustrated.

Close spacing between the electrodes is necessary to obtain maximum signal sensitivity from the device and to get reasonable polarizing effects from the voltages found in the average power supply of the amplifier or radio. These conditions of intimate contact and close spacing between the electrodes place natural restraints upon the motion of the movable diaphragm, and low frequency response is entirely eliminated. Consequently, the electrostatic speaker cannot be used as a single all-purpose speaker, but only as an upper range speaker to supplement a speaker capable of adequate bass reproduction.

There is another fundamental difference between the electrostatic speaker and the dynamic type of speaker. The dynamic loudspeaker has its own "powerhouse" (the magnet), which gives the weak voice coil signal a powerful ally in getting its message across. The electrostatic speaker has no such ally. It must broadcast its message entirely by the strength of the signal applied to it. Having no reservoir of auxiliary "push" power upon which to draw, the output level of the electrostatic speaker seldom reaches that of the dynamic type. However, there are features that may be built into the electrostatic speaker that in some measure overcome some of its limitations, and make it suitable for hi-fi systems.

Electrostatic Speaker is Voltage Sensitive: High Impedance

Since the electrostatic speaker is a true capacitor and is energized by voltage rather than current, it is a "high impedance device." This means simply that for optimum performance it must be connected to a source that is also of high impedance. Amplifiers, however, have output transformers that are low impedance devices designed to match the low impedance dynamic loudspeaker, a current fed device. This output transformer must still be employed for the woofer (low-frequency) speaker in the system employing the electrostatic speaker as a tweeter, as shown in Fig. 2-18. However, the electrostatic speaker cannot be connected to the low impedance output transformer. It must be connected to the high impedance circuit of the output tubes of the

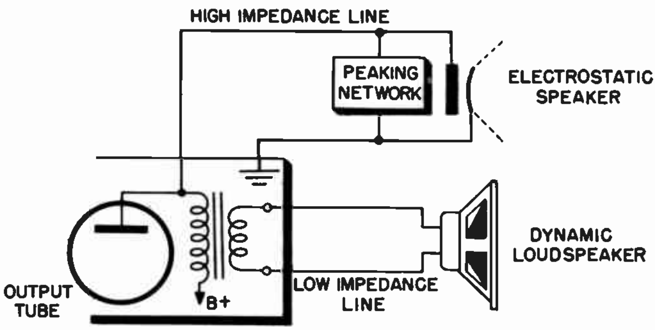


Fig. 2-18. Since the electrostatic is a high impedance, voltage operated device, it must be connected to the source of high signal voltage at the plate of the output tube, from which it may also obtain its polarizing voltage.

amplifier. Since some amplifiers use pentode and some triode output, the impedance the electrostatic speaker sees is different, from amplifier to amplifier, depending upon what tubes are used in the output stages. Its output efficiency and frequency response will accordingly change from amplifier to amplifier. This factor makes it difficult to properly match the electrostatic speaker to an amplifier, unless the manufacturer of the speaker gives the user the circuit coupling information necessary to enable him to properly couple the speaker into the high impedance side of the set, while the woofer is still connected to the usual low impedance side.

Because of these "matching" problems (the dependence of the electrostatic speaker upon the existing power supply and the necessity of coupling it with a woofer low enough in efficiency to correspond to that of the electrostatic speaker), the electrostatic speaker has found widest acceptance as a tweeter in ready made small radio, phonograph, and television equipment, where it may be integrally designed into the system by the manufacturer.

The Crystal Loudspeaker

There is another type of loudspeaking device, which, although voltage actuated, is not of the electrostatic type. This is the piezoelectric, or crystal loudspeaker. The principle behind the crystal loudspeaker is one of contraction and expansion of a certain crystal material (usually Rochelle salt) under the influence of an alternating electric field applied to the surfaces of the crystal. No polarizing voltages are necessary for the crystal loudspeaker as the flexural motion of the crystal follows directly in step with the polarity of the applied voltage. However, crystals are rather fragile devices, and it is not possible to drive them sufficiently hard to obtain useful output power for loudspeaker performance, especially on the low frequency end. Therefore, they are at present used only for earphones and for pillow speakers. In this latter application the listener lies with his ear pressed down upon the pillow to hear the loudspeaker.

The low frequency response of this type of loudspeaker is limited because any reasonable excursion of the crystal is not possible without fracture. For general communication or restricted range music it is perfectly satisfactory, however. A bank of crystal loudspeakers could be used as a tweeter to supplement a woofer, but the woofer would have to be low in efficiency to properly balance the output of the crystal bank. Like the electrostatic loudspeaker, the crystal unit must be tied in directly to the high impedance tube circuit ahead of the output transformer; however, no polarizing voltage is necessary for the crystal speaker. It encounters the same complications, however, of proper matching to the output tubes and of affecting the loading of these tubes, and the same integral amplifier connection problems that confront the electrostatic speaker.

The Ionic Loudspeaker

The newest working addition to the loudspeaker field is the Ionophone. However, a study of patent history will show that loudspeakers working on the principle of *ionized air particles* are not at all new. In fact this type of loudspeaker is an offshoot of another phenomenon, just as the electrostatic loudspeaker was an offshoot of the singing capacitor. Actually, the first "ionic" loudspeaker was the "singing arc." In the early days of wireless transmission, radio waves were generated by electric arcs jumping across gaps in high voltage circuits. It was soon noticed that these arcs would "sing" as the voltage producing them

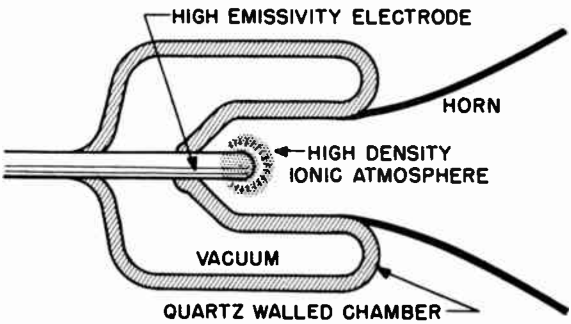


Fig. 2-19. A dielectric field placed around the quartz cylinder produces intense heat in the quartz, which in turn heats the high emissivity electrode. The ionic atmosphere boiled off from the electrode is modulated by a surrounding modulated radio frequency field and pulses accordingly to put the air into motion.

changed. Thus was born the "Ionophone" — a loudspeaker utilizing the principle of an ionized (electrically charged) gas. Today, however, the Ionophone has come far from the days of Nernst, Duddel, DeForest, and the "singing arc."

Charged Air Particles Vibrate as a Diaphragm

In its present form it consists of a mechanism by which *air particles* are given an electric charge. After the air molecules have been charged, they form an "atmosphere" of ions, the ions in this case being, of course, charged air particles. A modulating voltage is then applied to this atmosphere, making it pulsate in accordance with the signal-modulated voltage. This pulsing gives rise to an actual sound wave (in the air), which is then propagated in the usual manner. This mode of operation makes the Ionophone truly a "diaphragmless" loudspeaker.

The air molecules are actually agitated electrically without anything physical pushing them.

As shown in Fig. 2-19, the heart of the system is the small, narrow, slowly expanding quartz tube located within another quartz tube, with the space between them evacuated. There is no electrical function served by this evacuated area. It is a heat barrier to keep the inner section of the quartz cone reasonably protected from outside temperature changes that might work their way into the apex area of the inner section of the ionizing chamber through wall conduction. In the apex of the quartz tube is fixed an electrode of very high ionic emissivity character. This ion-producing electrode is made of a special platinum compound that, when heated, throws off large amounts of charged particles. The ions bubble away from the electrode surface and impart their charge to the air particles in the immediate vicinity of the heated electrode. Thus is formed an "atmosphere" of charged air particles around the emissive electrode.

The method of heating the electrode is dielectric heating, as used in diathermy equipment. A high voltage of very high frequency is applied to the quartz capsule that supports the electrode. By diathermy principles, the high frequency high voltage induces heat in the outer and inner quartz tube, which closely surrounds the electrode. This pocket of heat, in turn, causes the electrode surface to be heated, and the charged particles "boil off" forming the "ionic" atmosphere.

The voltage required to produce the necessary dielectric heating effect is quite high, in the order of several thousand volts. It is obtained from an external power supply and high frequency radio modulating device. This power supply also serves as the modulator unit, which receives the audio signal and converts it into the radio frequency field that surrounds the charged atmosphere.

Under the influence of this field, the atmosphere of charged particles is pulsed back and forth to form the genesis of the sound wave. A horn system must be coupled to the ionic generating element to provide an acoustic load for the small sound source.

Ionic Loudspeaker Requires External Power and Modulator

The Ionophone requires its own booster amplifier to feed the modulator device so that the performance of the system may be balanced with the auxiliary woofer that must be used with it. The Ionophone is not a complete system by itself, but forms a high frequency adjunct to existing systems. The life of the system is limited by the life of the ion emissive element, which (like a vacuum tube) may be replaced when deteriorated. The high frequency range of the Ionophone is theoretically unlimited, because there is no motion of anything physical in the system. The low frequency range is specifically limited by the nature of the horn load applied to it. In its present prototype form, the low frequency limit is about 500 cps.

Summary

In this chapter we have covered four basic types of loudspeaker: the dynamic, functioning by interaction of magnetic fields; the electrostatic, functioning by means of electrostatic fields; the crystal, functioning by stresses produced within crystal formations; and the ionic, functioning by the electromagnetic pulsing of a gaseous atmosphere.

We have attempted to carry this discussion only far enough to indicate the general modes of operation of the various types, so that we may, in the next chapter, use these introductory principles to determine what modifications must be made for the hi-fi variations within the basic types.

There are several schools of thought concerning high fidelity — the "play it exactly as performed" school, the "play it so it sounds real" school, and the "play it the way I like to hear it" school. There is logic in all three approaches. Someday, perhaps, they will all be synthesized into one, but until such time as the controversy is resolved we must accept the proposition that there are as many types of high fidelity as there are listeners. In order to satisfy these many concepts, it is necessary to provide a sufficient variety of component parts to make possible many different kinds of system. Accordingly, we shall discuss speakers from the point of view of their general application to the hi-fi field and then concentrate on the more specific and specialized units that become component parts in the more expanded hi-fi reproducer systems.

CHAPTER 3: *Hi-Fi Variations of Basic Loudspeaker Types*

Dynamic Loudspeaker Most Versatile for Hi-Fi

The speaker that is most common in hi-fi systems is the dynamic type. It is simple to install, simple to connect, may have a wide or a narrowly restricted frequency response as desired, may be used as a direct radiator or converted into a horn loaded system of high efficiency, and may be obtained in a variety of sizes to fit any equipment layout or budget. For this reason, the primary loudspeaker for hi-fi is the "general purpose" dynamic speaker.

The general purpose speaker is one that, when used with other high grade electronic equipment, provides reasonably good musical performance over the complete musical spectrum. This means that the same speaker must reproduce the low notes of the bass violin as well as the high treble notes of the piccolo, and these notes must be reproduced with adequate cleanness, clarity, and precision, permitting no unpleasantness to the listener other than to create in him the desire to own an even more elaborate reproducer system.

When one realizes that the single loudspeaker is called upon to reproduce the whole gamut of notes and tones originally produced by some 75 to 100 instruments ranging in size from the diminutive piccolo to the ponderous tuba and the double bass violin, it becomes nothing short of miraculous that such good quality is obtainable from one comparatively small speaker. One must look with respect upon even the single general purpose speaker.

In any loudspeaker the diaphragm is perhaps the most important factor in the actual *sound* reproduction because it is the prime vibrating element in contact with the air to be set in motion. The magnetic circuit is important in determining how well the diaphragm is vibrated. Consequently, we find that the intensity of the sound produced and the frequency range of the sound produced will in a large measure be linked to the magnet circuit design and the diaphragm shape, size, and construction.

The General Purpose Loudspeaker Produces Lows and Highs

Good reproduction of low frequencies is made possible by large diaphragm size; reproduction of good high frequencies by small diaphragm size. In the general purpose speaker, these performance characteristics must be combined into one structure, one "cone" size; consequently, somewhere along the line compromises must be made between low frequency and high frequency extension of range. The eight-inch speaker is perhaps the smallest size that will reproduce low frequencies with reasonable efficiency; its high frequency reproduction is in general excellent. The twelve-inch speaker gives better lows but is usually somewhat less effective in the high frequency region. The fifteen-inch speaker provides the best lows of the three, but in its high end is the most limited. Figure 3-1 shows in a general way the frequency overlapping areas of these three popular sizes of loudspeakers.

Where space for housing the speaker is at a minimum, the eight-inch size does a reasonably good job; where space is ample the larger speakers give considerably better low frequency performance. It should be realized that, even though the small speaker may actually reproduce the same low tones as the larger ones, it does so with much less efficiency. A violin may struggle to make a low note sound loud, but its big brother, the cello, produces that same note with ease and with greater loudness. This is not meant to infer that the eight-inch speaker cannot make good sound, even in a small enclosure. Although there are low notes that this speaker may not reproduce very well, they are not entirely absent. They are reproduced, but with less emphasis. Over 90 percent of the musical spectrum may fall easily within the range of the eight-inch speaker, and the other 10 percent is only partly reproducible at reasonable volumes. This is the compromise (not a severe one at that) that must be accepted when the smaller speaker systems are used.

The response of any speaker is, of course, greatly dependent upon the enclosure with which it works. Therefore, compensations may be introduced into the speaker-enclosure *combination* to extend the low end of the smaller systems. These applications are discussed in Part 2.

Dual Cone Features Built into Single Diaphragm for Lows and Highs

Because the general purpose loudspeaker must reproduce both lows and highs, it must be designed so that the diaphragm is stable enough to be moved as a *single whole* piston, for good low frequencies depend on the amount of active diaphragm area available to push the air into vibration. At the same time the diaphragm must have built into it at its apex "uncoupling" or disconnecting devices, so that smaller segments may be effectively isolated from the main body of the diaphragm for efficient high frequency reproduction. (See Fig. 3-2.)

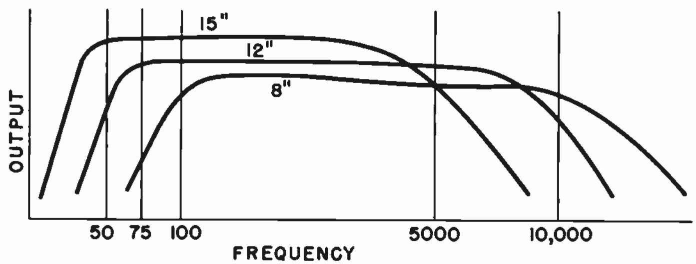


Fig. 3-1. The popular sizes of speakers have overlapping characteristics of response. The large speakers have more extended lows, the smaller ones more extended highs.

It may be helpful to point out how a cone may vibrate at low frequencies and at high frequencies at the same time. The uncouplers that make such a dual cone action possible are very much like shock absorbers on an automobile. When you go over a long low swell on the road, the car follows the contour of the road; the action of the shock absorber is at a minimum. If, however, the car meets a section of the road that has many little bumps in it, the car will ride smoothly over it while the shock absorbers quickly take up the small bumps. Thus

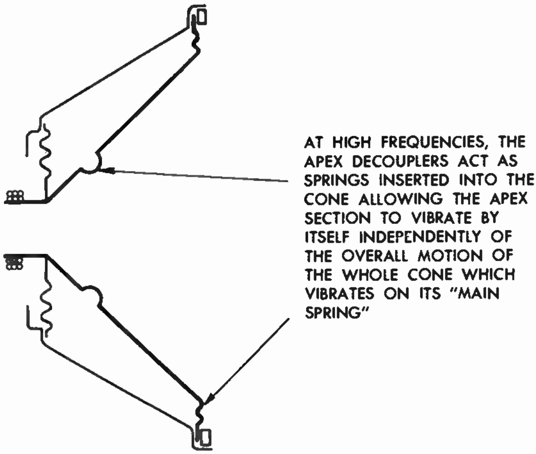


Fig. 3-2. In the general purpose speaker the diaphragm reproduces both high and low frequencies. In many instances the cone is designed with apex area "decouplers" to allow more efficient high frequency response.

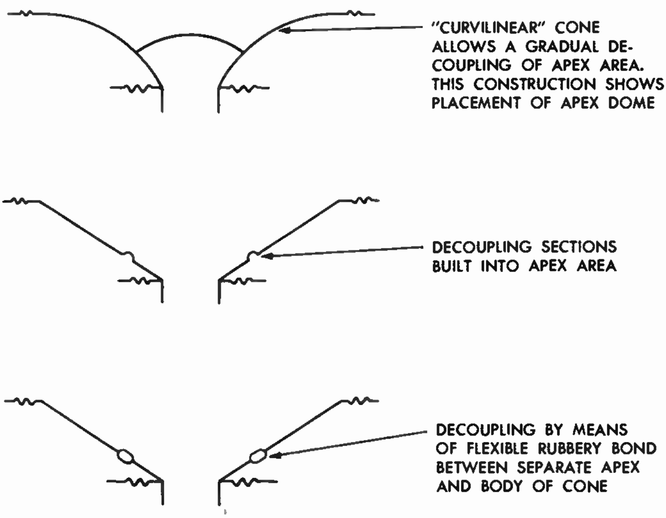


Fig. 3-3. Various means of uncoupling the apex area of a cone for better high frequency performance.

the wheels of the car vibrate at a high rate while the car body itself vibrates at a slow rate. The shock absorbers are the "uncoupling" devices between the lower and upper sections of the car.

The actual shape of the cone of the loudspeaker itself may produce the necessary uncoupling effects; or uncoupling devices may be incorporated in the cone; or the material of the cone may produce the uncoupling effect. The "curvilinear" cone is a popular shape for the general purpose wide range speaker because its slowly curving surface near the apex allows the apex area to "unhinge" itself gradually, as shown in Fig. 3-3. This illustration also shows the built-in type of uncoupler, which is like a small spring built into the cone. (Actually, the uncoupler takes the form of a ridge or corrugation molded into the paper cone.) This spring serves to disconnect the small apex area from the main cone, but does so more abruptly than the curvilinear shape. Finally, there is the segmented cone, in which the central area is completely separated from the main cone except for a soft rubbery bond at the place where they meet.

Diaphragm Apex Area Originates High Frequencies

When the proper uncoupling point has been determined it is usually to the advantage of the loudspeaker performance to provide an auxiliary apex dome to the area where the uncoupling takes place. This apex dome is frequently made of a very rigid but lightweight duraluminum or stiff paper cap. This dome cap serves two purposes. It closes up the voice coil area, protecting that very vital section from foreign particles. This function is, of course, important for longevity of the loudspeaker. Of more importance, however, is the effect of the dome cap upon the acoustic performance of the loudspeaker. It serves to provide better "polar response" (wide angle response) than the sharp angular section of the cone itself. The apex area of the cone is, in effect, a small megaphone, and like a megaphone will tend to direct its sound into a very sharp directional pattern. In order to minimize this effect, the dome is placed in such a position that it is vibrated by the uncoupled apex area of the cone. Being curved outwardly the radiation it provides is more uniformly circular in pattern than is the conical apex radiation. Thus better wide angle dispersion, in some degree, is obtained by transferring the sound radiation to the more "forward-looking" dome.

High Gap Flux Improves Efficiency and Transient Response

Outside of diaphragm considerations of the general purpose loudspeaker, the other important factor affecting performance is the magnetic circuit. The magnetic circuit, which is made up of the magnet itself and the iron around it that completes the path in which the flux exists, should provide as high a gap flux as possible. It should be recalled, however, that gap flux by itself does not determine total performance, because a very heavy diaphragm of improper design can be inferior in performance to a lighter diaphragm of good design integrated with a weaker magnetic gap flux. For fuller treatment of this problem, the reader is referred to Chaps. 4 and 5. However, as a rule, optimum performance may be expected from good magnetic circuit design.

The heavier the magnet, the higher in the Alnico series the magnet is, and the better the grade of the return iron in the circuit, the higher will be the gap flux. The Alnico V, or "Gold Dot" magnet is in general use today for the better type of speaker. The return iron circuits are usually made of high grade electrical iron such as "Armco", which is steel subjected to special heat treatment in its manufacture so that it may carry heavy values of flux without saturating. High gap flux, which is dependent on magnet type and weight plus good magnetic circuit design results in high conversion efficiency and good transient response. High conversion efficiency simply means getting more out for what is put in. The magnet is the powerhouse; it drives the speaker. The more there is of it, the more it makes itself felt when it interacts with the field set up by the voice coil current.

Transient response of the loudspeaker is the ability of the loudspeaker to follow faithfully sharp attacks of notes and to let go of a note once the signal has stopped. The staccato strike of the piano key hammer on the string is a sharp attack; the fading away of the tone is the decay. If the loudspeaker cannot follow the sharp attack of the tone, it will reproduce softer, less staccato tones; and if it doesn't let go of the tone (if there is "hangover") it will introduce a muddiness into the sound, because it is still playing one note when the next one has already started. (See Fig. 3-4.) A heavy magnetic field provides a strong magnetic impulse to start the speaker moving when the attack comes, and it acts as a strong magnetic "brake" when the signal stops.

Thus the overall magnet conditions of the general purpose speaker determine its tonal fidelity as well as it power efficiency.

Addition of Separate High Frequency Diaphragm to Cone Improves Highs

Although the general purpose loudspeaker may be built to produce acceptable musical reproduction, it may be varied greatly to provide specialized designs for improved reproduction. One of the

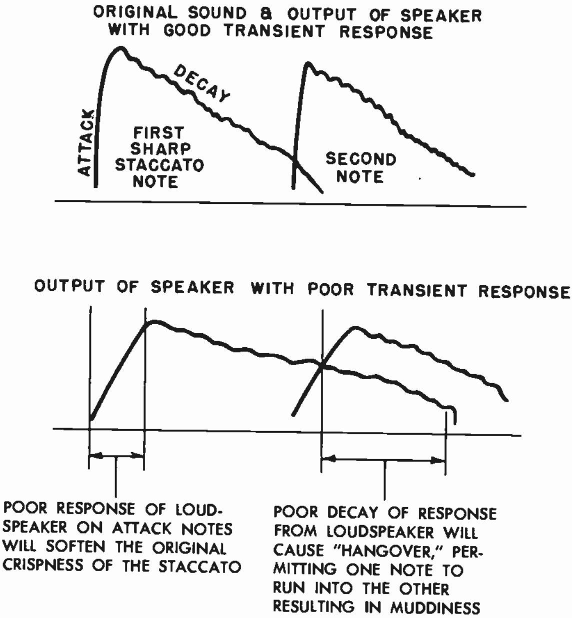


Fig. 3-4. Transient response of a loudspeaker is the ability of a loudspeaker to follow sharp sudden signals, and to let go of the sound as soon as the signal stops.

first variations that may be made upon the general purpose speaker is to convert it into a "two-way" speaker. Multi-speaker *systems* will be discussed in detail in Chap. 6, where their advantages and means of compounding the systems will be detailed. For the present we may easily accept the fact that two speakers are usually better than one, and that a single speaker converted to "two-way" action is better than a single one unconverted.

The purpose behind this conversion is to provide more effective high frequency response from the speaker than is possible by the simple decoupling means previously discussed. Through "mechanical crossover" networks it is possible literally to drive two *separate* diaphragms, a large one for the low frequencies and a small one for the

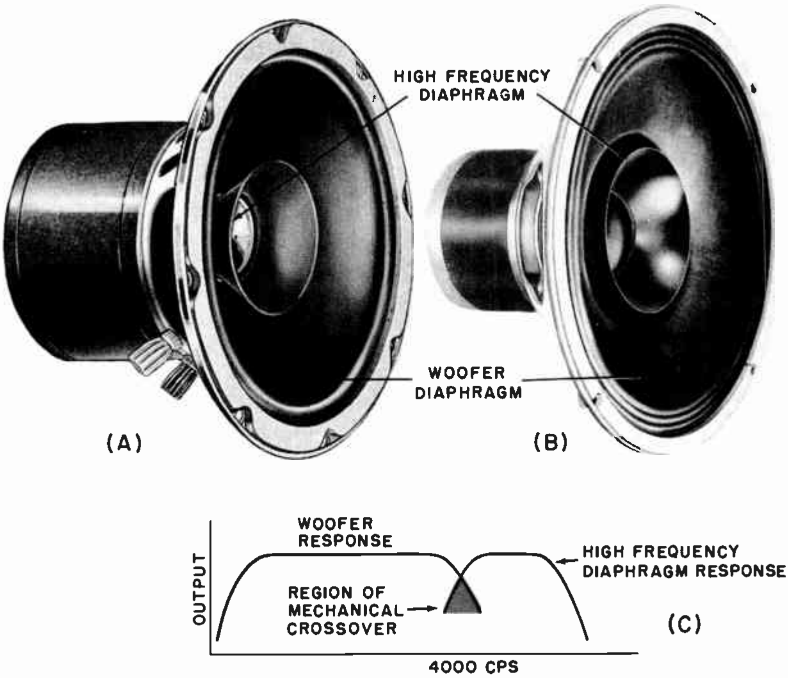


Fig. 3-5. Two-way speakers employing mechanical crossover between woofer diaphragm and auxiliary high frequency diaphragm. ((A) Courtesy Electro-Voice. (B) Courtesy Goodmans)

high frequencies, from one voice coil. Speakers of this type are easily recognized by the fact that there is a separate diaphragm affixed to the apex area of the main diaphragm. Two versions of this type of speaker are illustrated in Fig. 3-5. A crossover network (See Chap. 7) is a device, either electrical, acoustical, or mechanical, for slicing a frequency spectrum into separate parts or "bands" for purposes of providing better performance characteristics within the individual bands by separate components than is possible with a single component covering the entire band. In the case of the general purpose speaker previously discussed, the built-in decoupler in the diaphragm actually performed the job of a mechanical crossover; it separated, by means of a mechanical "shock absorber," the high frequency vibrations from the low frequency vibrations, and directed them into separate sections of one diaphragm. In a very general sense, then, even the general purpose speaker is a two-way speaker of sorts.

High Frequency Diaphragm Separated from Main Diaphragm by Mechanical Crossover

But now we are going to approach the problem of mechanical crossover from another angle. Providing two separate diaphragms, rather than making two out of one, is of advantage in that the high frequency diaphragm may be more precisely controlled in dimension, weight, shape, and size, for improved high frequency response than if it were physically integral with the main diaphragm. If the reader were to examine the constructions shown in Fig. 3-5 "in the flesh," he would see that not only are the two diaphragms of one speaker completely different in geometric shape, but the auxiliary high frequency diaphragms are, in addition, made of a paper stock different from the woofer diaphragms.

The woofer diaphram is made entirely of the paper stock best suited to low frequency performance, and the smaller high frequency diaphragm is made of the paper stock best suited to high frequency reproduction. If the woofer diaphragm is fabricated of soft stock, no special decoupling devices are necessary to isolate the high frequency vibrations of the voice coil from the woofer. Its own softness will prevent the high frequency vibrational energy from getting any appreciable distance up the main cone. The cone material in this case is in its own decoupling agent; it provides the mechanical crossover that admits low frequencies into the woofer but discriminates against highs.

What then prevents the low frequency vibrations from being transmitted to the small diaphragm? Actually, nothing prevents the small diaphragm from being *driven* mechanically by the low frequency vibrations. But as far as contributing to the production of low frequency sound energy, the small diaphragm plays a very small part.

It is too small physically to reproduce low frequency sound adequately. The mechanical crossover thus works in two ways. High frequency vibrations are kept out of the woofer cone by its own decoupling characteristics, and thus it does not radiate them. Low frequencies do tend to get into the high frequency diaphragm, but are not radiated because of the small size of the diaphragm. Thus the mechanical

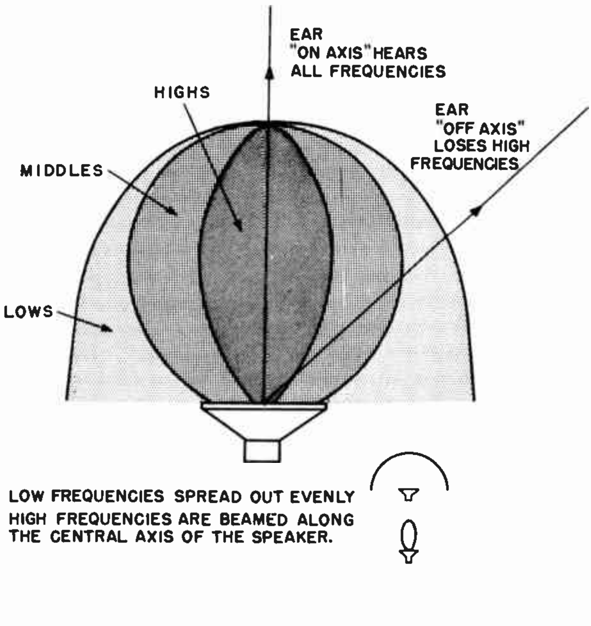


Fig. 3-6. A cone speaker tends to concentrate the high frequencies on the central axis of the speaker, causing loss of fidelity off the axis.

crossover is completed. Figure 3-5(c) gives the typical response of the woofer and tweeter sections of assemblies such as those illustrated.

The overall frequency response of this type of structure is usually considerably better than that obtained from the general purpose cone. The low frequency response of he two-way mechanical crossover system is improved because the large diaphragm does not have to be compromised in design to handle high frequencies. In addition, the high frequency radiator may be designed for optimum high frequency radiation without being compromised by the woofer diaphragm proper.

Separate High Frequency Apex Diaphragm Aids Dispersion of Highs

The reproduction of high frequencies poses another problem besides that of the level of acoustic output. High frequencies do not spread out as easily as do low frequencies; they tend to beam straight ahead. The higher the frequency, the tighter the beam. This means that if the listener were to be seated to one side of the speaker ("off axis"), he would hear much less high frequency sound from his loudspeaker than if he were sitting directly in front of it. This effect, illustrated in Fig. 3-6, may be minimized in multi-speaker systems that employ special tweeters

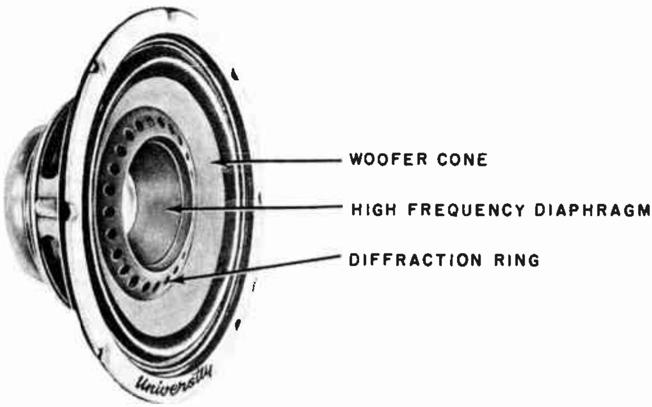


Fig. 3-7. A two-way speaker with mechanical crossover between woofer diaphragm and high frequency diaphragm, using a diffraction ring to aid the diffusion of the high frequencies over a wider angle. (Courtesy University)

providing "wide angle coverage." However, in the single speaker structure such as the extended range speakers just discussed, high frequency beaming may occur unless special additional elements are built into the speaker design.

One of the high frequency spreading devices added to an auxiliary high frequency diaphragm with a mechanical crossover system is illustrated in Fig. 3-7. This is the "Diffusicone" construction. The auxiliary diaphragm is driven by the voice coil, with the main diaphragm effectively uncoupled from it for these high frequencies by its shape and material construction. The treble diaphragm now radiates high frequency sound from its forward surface. However, as in all loudspeakers, there is also sound coming from the back of the treble diaphragm. Before this rear sound can emerge, it passes through a diffraction ring containing many relatively small holes. When the sound from the back of the auxiliary high frequency diaphragm tries to squeeze through these holes, it "spurts out" in many directions, giving the overall high frequency response a wider angular coverage.

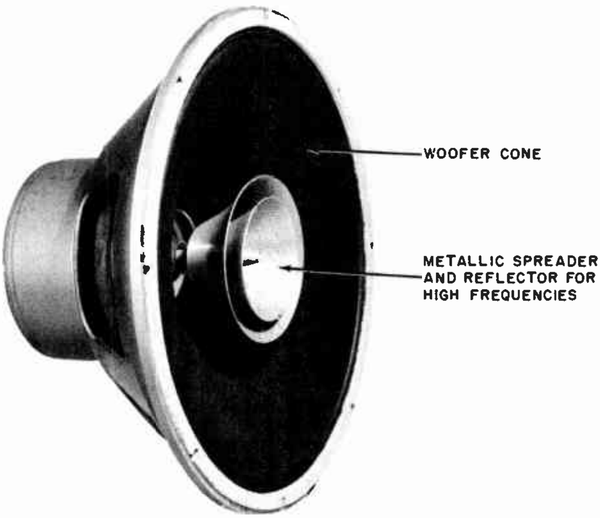


Fig. 3-8. Spreading of the high frequencies from the apex area of this cone is accomplished by the stationary metal co-spiral structure placed in the center of the cone, acting as a scattering reflector for the highs. (Courtesy Stephens)

Another diffusion scheme is the spiral deflector shown in Fig. 3-8. This structure consists of spiral metal deflectors within the apex area of the cone proper. Sound waves that hit this deflector are spread out over a wider angle than if the deflector were not used. Being placed at the apex area, this deflector will be most effective for those frequencies that are generated at the apex, i.e., the high frequencies. It will be observed, however, that there is no auxiliary diaphragm in this structure, hence it falls in the category of the general purpose speaker rather than in the mechanical crossover class.

It is hoped that what has been accomplished so far is an appreciation of the general purpose hi-fi speaker and the extension of its principles into the wider range two-way systems with mechanical crossover features. Accordingly, we are now ready to discuss in detail such components as woofer, mid-range, and tweeter speakers which go into the making of such multi-speaker systems.

Specialized Limited Band Speakers Improve System Performance

Although the wide extended range speaker has certain qualities that make it a fine speaker for simple hi-fi systems, there are definite limitations to its performance. There is a wastefulness of precious high frequency audio power (see Chap. 7) despite the fact that there is definite mechanical crossover. Furthermore, there may be mechanical intermodulation between the two portions of the two-diaphragm speaker. The answer to better utilization of audio power obviously lies in the use of more highly efficient units.

Increased efficiency may be obtained by making the reproducer units more specialized, *within limited bands,* rather than spreading their power out over a wide band. Thus "woofers" are narrow band speakers that are highly efficient for low frequencies; similarly, "tweeters" are narrow band speakers that are highly efficient for high frequencies. A combination of a specialized woofer and tweeter thus makes a multi-speaker system of higher efficiency than is generally obtainable from a wide range single speaker. In addition, the use of two separate speakers considerably reduces the possibilities of intermodulation distortion (the interaction between two different notes, one high, the other low, both emanating from the same moving structure when it is not completely linear in its motion). Problems such as wide-angle response, presence, brilliance, and balance, may also be more effectively treated by means of multi-speaker systems. Accordingly, we are now ready to discuss the elements of multi-speaker systems — the woofer, midrange, and tweeter reproducers.

Woofer Designed for Maximum Efficiency at Low Frequencies

Although a woofer is an integral part of a good high fidelity wide range speaker system, it is not in itself a wide range unit. It is, on the contrary, a loudspeaker that is very restricted in range; it is limited to the reproduction of the lower part of the acoustic spectrum — the bass notes. But, because it is restricted, it may be designed to provide much more efficient and clean bass reproduction than would be possible from a single wide range speaker in which compromises must be made between highs and lows to obtain reasonably good all-around reproduction. The fact of the matter is that when a speaker is designed for specialized work, we may give special consideration in the design to those factors which are specific and peculiar to the range in which the speaker is to operate.

Whereas a single wide range speaker must be compromised in size to obtain both good lows and good highs, the woofer is concerned only with the reproduction of the low frequencies. Therefore, it is characterized generally by the same features that characterize musical instruments designed to create the bass notes. The woofer is large in size, rugged in construction and attuned to (or has a resonant frequency falling in) the very low end of the audible range. It is also capable of producing large amounts of sound. All these factors are integrally related to one another in the final design of the speaker. It is possible, however, to dissect then separately to understand the general bearing they have upon the performance of the speaker.

Woofer Diaphragm is Large, Has Low Resonance

Perhaps the cornerstone upon which all the other design parameters are built is that of the natural resonance frequency of the woofer. Woofers have low resonance frequencies; that is, they are "tuned" to resonate and to be most efficient in the low frequencies. Low resonance frequency is obtain by heavy diaphragms lightly suspended as shown in Fig. 3-9. The heavier the diaphragm, the lower the resonant frequency. The looser the edge compliance that holds it in place, the lower the resonance.

There are specific limits to weight of diaphragm and looseness of suspension that are determined by the mechanical problems involved in good hi-fi speaker design. Even though a diaphragm may have low resonance and be attuned to the low frequencies, it has to have a large vibrating surface for the low notes to be efficiently radiated and adequately projected into the listening area. The bass violin, for instance, has a large vibrating body. The bass drum has a large vibrating membrane. The tuba pushes out a large volume of air. Similarly, the woofer must actuate a large volume of air. Its diaphragm is therefore made as large as possible, consistent with the other factors of the weight of the diaphragm and the actual structural mechanics of its design. The larger the diaphragm, the more efficient the reproduction of the low frequencies. If it were possible to build a speaker as large as a bass violin, it would reproduce the original tones of that instrument with comparative ease. However, such a speaker would pose other problems in design that would make it impractical. We must be content with a reproducer somewhat smaller.

It will be realized, then, that in accepting a vibrating body that is smaller than the original instrument, either a compromise must be

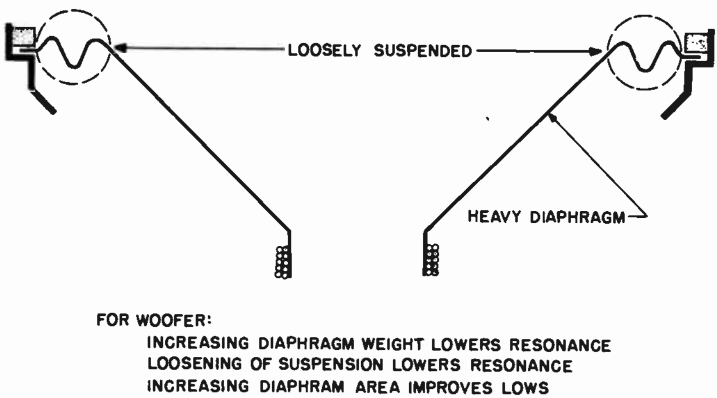


Fig. 3-9. To achieve low natural resonance, woofers have heavy diaphragms loosely suspended. To achieve efficient low frequency radiation, woofer must be large.

made in performance or other characteristics must be built into the design of the reproducer to compensate for the necessary reduction in diaphragm size of the woofer as compared to the original instrument. The acoustic power output of any vibrating instrument is, first, directly proportional to the area of the vibrating surface. The larger the vibrating membrane, the more air is set into motion. Secondly, the power output is proportional to the displacement of the diaphragm in a given short period of time, for the more the diaphragm moves and the faster it moves, the more air will be set in motion. If, then, we cannot have a vibrating body as large as a bass violin as a reproducer, and we have to be satisfied with a smaller vibrating instrument, such as the diaphragm of a 15-inch woofer, we may compensate for the small size of the diaphragm by making it move over larger distances than did the panels of the body of the original bass violin. Thus the

DIAPHRAGM SIZE (IN.)	RESONANT FREQUENCY (CPS)	DIAPHRAGM MASS (GRAMS)	DIAPHRAGM COMPLIANCE (CM/DYNE)	PEAK EXCURSION FOR 1 WATT AT 100 CPS (IN.)
15	45 - 55	18-20	1.8×10^{-7}	.055
12	60 - 70	9-11	2.0×10^{-7}	.10
8	90- 110	4-6	2.5×10^{-7}	.19

Fig. 3-10. Physical characteristics of typical loudspeakers of popular sizes showing relationship of size to resonant frequency, diaphragm weight, and excursion necessary to generate one acoustic watt.

woofer diaphragm is characterized by large available vibrational excursion, in addition to as large a physical size as it practical, plus a low resonant frequency obtained from a heavy diaphragm loosely suspended in relation to its mass, so that large amounts of low frequency acoustic power may be generated. Figure 3-10 indicates the general physical characteristics of some typical loudspeakers in terms of these factors.

Woofer Voice Coil Carries Most of Audio Power

Acoustic measurements made on live orchestral performances show that by far the largest amount of sound power produced by a symphony orchestra resides in the lower bass registers. We would naturally expect, therefore, that the woofer would be constructed to handle the largest proportion of the amplifier power representative of the original program source. Accordingly, the voice coil of the woofer is generally large and heavy.

The woofer voice coil must be able to dissipate the heat generated in it by the large proportion of electrical power fed to it; otherwise, the cement binding it to the coil form would soften or overbake to a crumbling crispness. The form on which the coil is wound may become overheated, charred, and therefore weak and brittle. All of these effects would rapidly shorten the life of the speaker. However, the larger the coil, the more radiating surface is available for the generated heat to escape and the cooler the coil.

There is another more important reason, acoustically speaking, for the voice coil of a woofer to be large. In order to drive the comparatively heavy diaphragm of the woofer, a considerable driving force on the diaphragm must be developed by the reaction of the voice coil current with the permanent magnet gap field. This reaction is obtained by providing a voice coil with many ampere-turns. This means that we want to get as much current (amperes) as possible through as many turns (of coil) as possible in order to develop a large voice coil magnetic field, which will then interact with the gap field. Thus the more turns we can put on the voice coil the more ampere-turns we can have with a given current. Also, the larger we make the wire size, the more current can be forced through the wire; again we have more ampere-turns. Thus more wire turns of larger size wire means more driving force developed by the voice coil, and it also means a larger coil.

However, there is a practical limit beyond which it does not pay to increase the voice coil mass, because, after all, the voice coil has to move itself as well as the diaphragm. If most of its developed driving force is used up in moving itself, its usefulness has been wasted as far as developing acoustic power is concerned. Therefore, in practice there is an optimum weight of voice coil dictated both by the necessary power handling capacity and the mass of the moving diaphragm.

Dense Gap Flux Needed to Drive Heavy Woofer Moving System

In view of the relatively heavy diaphragm and voice coil of the woofer, it is important that there be sufficient magnet power available to drive adequately this heavy mass. The large low frequency currents flowing through the woofer voice coil will produce only feeble results if the field it produces cannot reach out and react with a dense gap flux. Dense gap flux means a high degree of reaction between the gap flux and the voice coil field, and large resultant motions will occur. Consequently, woofer magnetic structure are heavy and massive.

The density of the gap flux not only imparts drive to the system, but also serves a very important function as far as cleanness of response is concerned; namely, the damping of the loudspeaker (see Chap. 5). A heavy magnetic field provides a good sturdy propulsive force for the attack of the notes, and also provides a good magnetic brake for the speaker when the signal note has stopped. The heavier the gap flux, the better the electromagnetic propulsive force and damping effects. In woofers where the inertia of a heavy moving system is relatively great, these effects of dense magnetic gap fields in overcoming the inertia of the heavy moving system both in starting and in stopping its motion are of great importance.

In summary then, the general characteristics of woofers are low resonant frequencies achieved by large heavy diaphragms and loose suspensions, large acoustic outputs achieved by large diaphragm excursions, high power handling capacity through large voice coils necessary to drive the heavy diaphragms, and large magnet structures and gap flux for adequate driving force and damping control.

Tweeter Designed for Maximum Efficiency at High Frequencies

For the high end of the acoustic spectrum there is the tweeter. Again, because the tweeter is restricted in range, it may be more specialized in design in favor of improved high frequency efficiency, with sacrifices being made in areas where good performance is not expected of the speaker. Since the tweeter is to reproduce the high frequencies, or the treble notes, it must be attuned, or resonated, in the high frequency region. High resonant frequency is obtained with structures that are light in weight and stiffly supported, as indicated in Fig. 3-11. Consider, for instance, the very light, tightly stretched E string of the violin, which produces the very high treble notes, compared to the long, loose, heavy G string of the bass violin, which produces the low bass notes. Thus in the tweeter the diaphragm is very light, and relatively stiffly supported.

Tweeter Diaphragm is Small, Has High Resonance

Naturally, if we wish to make the diaphragm light in weight, it must become small in size. Therefore, in contrast to the large 15-inch woofers, we usually find cone type tweeters in the 3-inch class. By reducing the diaphragm both in size and in weight, it becomes necessary to proportionately reduce the voice coil to get optimum drive of the diaphragm by the coil. As a result of the entire moving structure of the cone type tweeter being so diminutive, it becomes almost a scaled down model of the larger woofer, but with a high resonance frequency.

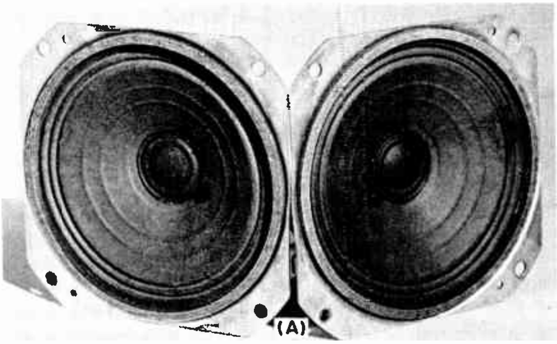
It is fortunate for the design of the tweeter voice coil that only a comparatively small amount of electrical power resides in the high frequencies. Therefore, the small voice coil found to be mandatory for acoustic reasons will not be subjected to excessive electrical power beyond the capacity of the scaled down voice coil structure. Invariably, the tweeter voice coil is wound with aluminum wire or ribbon because of the lighter mass of such a coil as compared to a copper coil. Inasmuch as lightness of moving system is the prime requisite of tweeters (along with stiffness of suspension), the lightweight aluminum wire affords specific high frequency advantages over copper voice coils.

Cone Type Tweeters Produce High Frequency Beams

In our discussion of the general purpose of wide range speakers, we touched upon the problem of the beaming effect of high frequencies from cone type speakers, which tend to concentrate those high fre-

Fig. 3-11. To achieve high natural resonance, tweeters have small, light diaphragms, stiffly supported.

quencies along the axis of the speaker. This means that a single cone type tweeter does not produce even distribution of high frequency sounds into the listening area it serves. It "lobes" the higher frequencies out directly in front of it and discriminates against high frequencies at the sides. This condition may be overcome by arranging two or more cone type tweeters in an arced array (Fig. 3-12) so that several individual lobes coming from the separate speakers cover the listening area. This provides some measure of desirable wide-angle response,



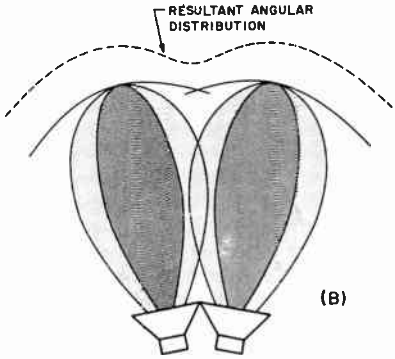


Fig. 3-12. Cone type tweeters may be arranged in an arc array to improve the angular radiation by providing multiple beams.

although there are irregularities in distribution because of the spacing of these lobes, as shown.

Horn Loading Permits Diaphragm Size Reduction and Increases Efficiency

In order to get reasonable output from an acoustic radiator, we must vibrate large volumes of air; in order to vibrate large volumes of air, we must usually have large vibrating surfaces. However, this is true only of the direct radiator, the type of loudspeaker that faces the listener directly as it plays. In the direct radiator, the larger the diaphragm surface the greater the output, if the excursion is maintained.

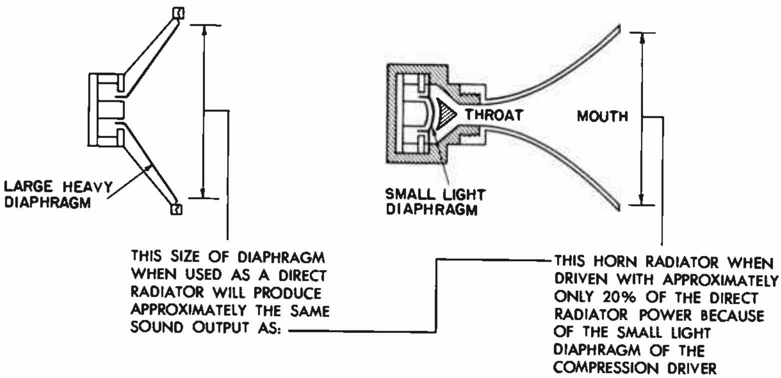


Fig. 3-13. The compression type driver has a small, light diaphragm, which is easier to drive than a large cone diaphragm. The pulsations produced by the diaphragm at the narrow throat are transmitted to the horn mouth, which becomes the radiating surface.

However, if the diaphragm must be made small to have a high resonant frequency, only a small volume of air is moved and reduced acoustic output occurs.

However, increased acoustic output from any type of diaphragm may be obtained if it is coupled directly to a horn, converting the system to a horn loaded system. Furthermore, by the addition of a horn, the size and mass of the diaphragm of the driver unit may be reduced even further.

Figure 3-13 shows in general how the size of the diaphragm may be diminished, when driving a horn, in contrast to the size of a direct cone radiator. The driving force of the voice coil of a horn loaded driver is imparted to a small diaphragm and the expanding air column of the horn. The mouth of the horn may actually be considered to be the final vibrating diaphragm. The driving force in such a system may thus be considered to be distributed between the mass of the diaphragm, which has been greatly decreased, and the mass of the air in the horn. However, since air is so much lighter than paper, the overall load of the horn column on the voice coil, for the *same acoustic output* as a direct radiator cone, may be greatly reduced and the efficiency of the system correspondingly increased. Conversely, of course, for

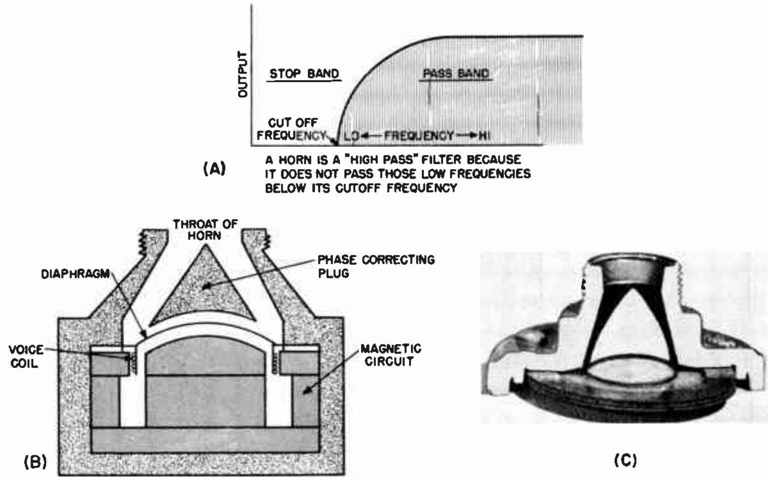


Fig. 3-14. In compression driver, back of diaphragm is sealed off, creating an acoustic stiffness, which helps raise the resonance of the unit into the pass band of the horn. (C) Shows a cutaway photo of part of the structure illustrated at (B).

the *same electrical input,* the horn loaded system will produce more output than the direct radiator type.

As will be shown in Part 2, a horn is essentially a baffle. It provides an acoustic load for the actual mechanical driving system. Horns are "high-pass" filters; i.e., their high frequency range is generally unlimited, but the low frequency range is determined by the acoustic design of the horn. (See Fig. 3-14.) Therefore it is necessary to make sure the driver unit is designed to be most efficient within the range in which the horn is to operate, and not spread out in efficiency over a non-usable range.

Accordingly, we find that driver units for horn systems are of the compression type, with the back of the unit completely sealed to provide acoustic stiffness (a stiff air cushion for the back of the diaphragm). This added stiffness raises the resonance of the unit and makes it more efficient in those high frequency ranges where the horn is most efficient. Thus the combination of the compression driver horn loaded system becomes an extremely efficient high frequency reproducer. Figure 3-15 illustrates the use of the important phase correcting plug that is found in high frequency driver units. This plug serves to equalize the path lengths of the sound "rays" from the diaphragm so

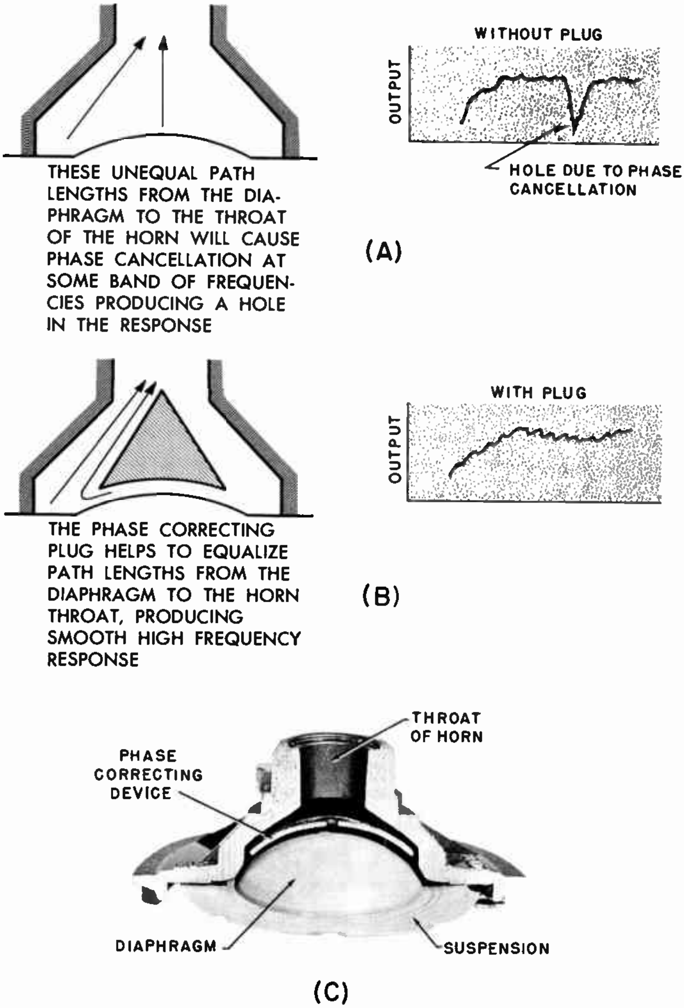


Fig. 3-15. The phasing plug in the acoustic circuit of the compression type driver unit eliminates holes in the high frequency response due to path length differences from the diaphragm to the throat. Another type of phasing device is shown in the cutaway photo at (C).

that they all meet in-phase at the throat of the horn, giving maximum high frequency efficiency, without frequency cancellation effects.

Tweeter Horn Design Provides Controlled Angular Dispersion of Highs

However, there is more to acoustic reproduction than just efficiency. There is the all-important matter of angular dispersion. The characteristic lobing effect of the cone type tweeter in the high frequency region may be overcome by the use of horns designed for wide angle distribution. In fact, the horns may be designed so that they may actually mold the sound field into various patterns as desired, giving a controlled wide angle distribution. Horns for such application are generally "cobra" shaped. They are not round, but rectangular in mouth configuration. Several of these horns are shown in Figs. 3-16 through 3-20, and it will be observed that they obviously function on different principles. The four general types are the *multicellular, diffraction, acoustic lens,* and *reciprocating flare* horns.

The Multicellular Horn

The multicellular horn as shown in Fig. 3-16 is essentially a group of horns all radiating by themselves as separate and distinct little horns, but energized by a common driver unit. The method by which this type of horn distributes the high frequencies depends on the manner in which the individual horns are stacked, and not at all upon the design of the individual horns. As will be observed from the illustration, the multicellular cluster is made up of small square-mouthed horns. Since these individual component horns have symmetrical mouths and symmetrical expansions in both the horizontal and vertical directions, the single horn element itself does not determine the direction in which the sound propagates. That is, the sound from the individual horn spreads out equally into the vertical and horizontal directions. The elemental horn of the cluster does not by itself produce wide angle dispersion of the high frequencies. However, since these component horns are stacked in any array, which is broader than it is high, as shown in Fig. 3-16, more horns throw sound into the horizontal direction than in the vertical. Thus wide angle dispersion of the sound is

THE MULTICELLULAR HORN

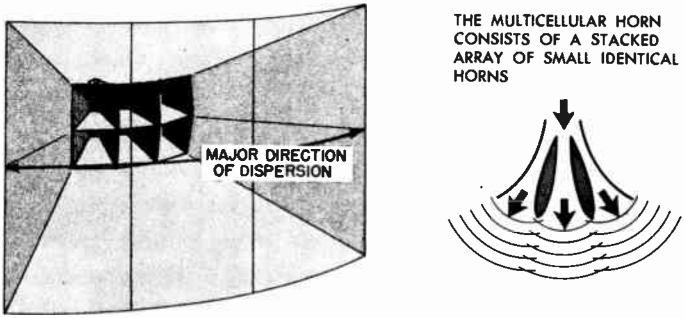


Fig. 3-16. The multicellular horn field pattern is made of separate sound beams radiated by small individual horns arranged in an arc.

effected in the horizontal plane. The effect is materially aided by the fact that the horns are swung around in an angular array.

Since the multicellular horn is made up of individual radiators, it is to be expected that the radiated field pattern of each individual horn will overlap those of its adjacent horns. The result of the interactions of the individual field patterns is an overall radiation pattern characterized by lobes and valleys not too different from those which occur when cone speakers are ganged into an array. Each little horn mouth sends out a more or less directional lobe similar to that from a cone. Arrays of this type are usually described by the number of horns in one direction as against the number in the opposite direction. Thus the one shown in the illustration is a "three by two" horn.

There are some types of tweeter horns used in coaxial assemblies that give the appearance of being multicellular but are actually composed of a simple flared horn with fins inserted in the mouth of the horn. These fins sometimes serve to reflect the sound into a wider pattern, but here again, the fins cast acoustic shadows just as easily as they cast acoustic reflections; consequently, the field pattern of the latter type of "pseudo-cellular" horn is also characterized by lobar irregularities.

The Diffraction Horn

The diffraction type of horn, illustrated in Fig. 3-17, works on the principle that sound coming out of a narrow slit that is small compared to the wavelength acts somewhat as a "point source," and the sound coming through it will consequently emerge in a cyclindrical wavefront pattern from the slit, as shown in the illustration. Especially will this be true if there is no baffling on the sides of the narrow slit. The absence of the baffling allows the cylindrically formed wavefront to flow around the side walls of the narrow mouth of the horn. In more precise terminology, the wavefront diffracts out of the narrow slit around the mouth edges; thus the term "diffraction" horn.

THE DIFFRACTION HORN

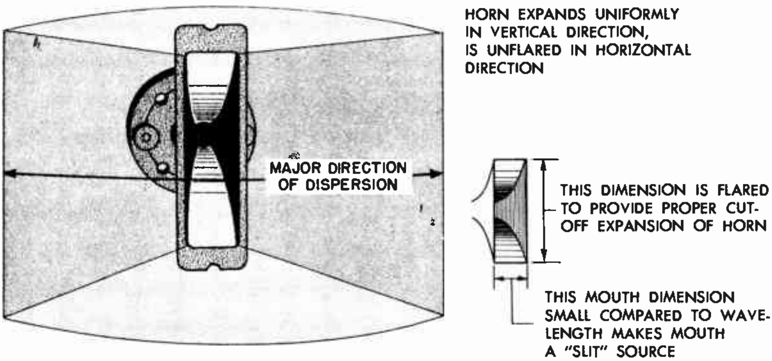


Fig. 3-17. The narrow mouth behaves as a "slit" and wavelengths that are large compared to the slit emerge as from a point source, diffracting into a cylindrical wave.

Because the mouth diameter is so small in the horizontal direction (with the horn mounted as shown in the illustration), it must flare out rapidly in the vertical direction to provide the proper overall horn expansion for the cutoff frequency for which the horn is designed. Thus in appearance it may seem that the greatest horn dispersion will be in the vertical plane because the mouth contour seems to greatly favor this direction. Actually, however, the reverse is true, and for the diffraction horn of this variety, the major energy distribution is in the horizontal plane. However, since the slit effect exists for conditions in which the width of the slit is small compared with the wavelength of the sound, the effective dispersion will be variable with frequency. As the frequency goes up (smaller wavelength) the slit "opens up" and there is comparatively less diffraction. However, despite this reduction of the dispersion of the higher frequencies, in this type of horn there is smooth angular response devoid of lobes or valleys because of the absence of beaming cells.

The Acoustic Lens

The acoustic lens is a means of controlling the directional spread of sound by placing in the sound beam arrays of obstacles arranged in specific geometric pattern. As a point of interest, the acoustic lens was an outgrowth of research on microwave radio transmission. In the course of studying the behavior of electromagnetic lenses for the purpose of focusing and beaming these very high frequency waves, certain difficulties were encountered in constructing suitable laboratory models and equipment with which to carry on the work in an expeditious manner. Since the actual wavelength of the electromagnetic microwaves was very similar to that of ordinary high frequency sound waves, it was found that valuable laboratory time could be saved and the study accelerated if acoustic models of focusing devices were used along with beams of acoustic energy to replace the electromagnetic focusing devices and the electromagnetic wave. From these studies emerged the acoustic lens as a means of aiding the distribution of high audio frequencies over a wide horizontal plane.

One form of the acoustic lens is shown in Fig. 3-18. It consists of stacked layers of perforated screening with discretely enlarged open areas in the center of the disc. The spacing between the screens, the size and separation of the individual small holes in the screen, and the progressively changing large opening in the center of the screens all determine the exact manner in which sound dispersion takes place. Although the axact analysis of the manner in which this dispersion takes place is founded on optical theory, we may readily understand the function of the lens by examining what happens to a sound wave as it hits an obstacle array. It will be realized of course that a screen with many holes in it is as much an array of obstacles as would be the case if a lot of little discs were placed in the beam. In the case of the perforated screen, the holes are "free space," and the material between the holes constitutes the "obstacle" of the obstacle array.

What then happens when a sound wave hits an array of obstacles in its path? Naturally, where there are openings through the obstacle array, the sound wave will try to squeeze through these openings. Where there are obstacles, the sound wave will not get through, but will be stopped and reflected back in the general direction from which it came. Thus, we may say that if only part of the sound wave got through the obstacle array, and the other part were stopped *within* the array, the density of the air in the array would increase, due to part of the sound

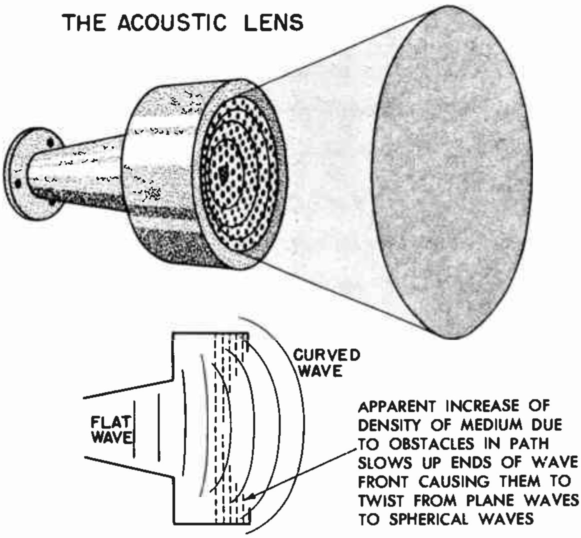


Fig. 3-18. The acoustic lens converts a plane wavefront to a curved wavefront.

wave being retained momentarily while the next one came in on top of it. There is thus an apparent increase in the density of the medium (in the vicinity of the obstacle array) through which the sound wave must travel. Since the velocity of sound decreases as the medium through which it flows increases in density, the speed of sound when passing through the array becomes less than that of sound in free space. If now the obstacle array is fashioned so that it slows up only a portion of the wavefront, the wavefront will be bent and the direction of the propagation will be changed. Inspection of Fig. 3-18 indicates how this differential slowing up of the wavefront at its edges causes the overall direction of the front to become more circular and consequently more spread out.

Since the scattering effect of the acoustic lens is a function of many critical factors concerning the spacing of the obstacles, the size of the obstacles, and the frequency involved, it will be found that the dispersion produced will have large side lobes of energy, which in some cases exceed the amplitude of the on-axis response. The acoustic lens thus furnishes wide angle response, but at the expense of some smoothness of response in the desired plane of dispersion.

The Reciprocating Flare Horn

The reciprocating flare horn accomplishes wide angle distribution of sound through the expedient of reversing the direction in which the pressure builds up within the horn and maintaining a mouth configuration that minimizes undesirable diffraction in the vertical plane. All horns are pressure controlling devices. At the throat of the horn the pressures are greatest; at the mouth they are the smallest. At any point along the horn between the mouth and the throat, the sound pressure will be of some value intermediate between these two extremes. The manner in which the sound pressure throughout the horn will vary depends upon the growth of the cross-sectional area of the horn. Thus, by controlling the manner in which the horn expands, it is possible to "contour" the sound pressure into any direction. To perform useful acoustic work, however, the sound pressures built up within the horn must emerge into space through the mouth of the horn. The mouth of the horn thus becomes the actual sound radiator. Thus the size and configuration of the horn mouth *in conjunction* with the horn flare, and together with the frequency involved determine the degree of dispersion.

The reciprocating flare horn is one in which the rate of expansion of the horn remains constant, but the direction in which it expands changes from one plane to another for purposes of contouring the pressure fronts within the horn. Reference to Fig. 3-19 will indicate that the reciprocating flare horn first expands rapidly in the vertical direction with practically no expansion in the horizontal direction. A cut taken across this first horn section would show it to be a fairly well elongated rectangle oriented vertically. The sound pressure traveling down a channel of this type of cross-section finds it relatively easy to expand into the vertical direction, for the walls flare away in that direction. In trying to expand into the horizontal direction, however, the sound pressure meets the side walls of the horn, which restrict the wave from expanding, and consequently the sound pressure finds itself building up at these side walls rather than being allowed to freely

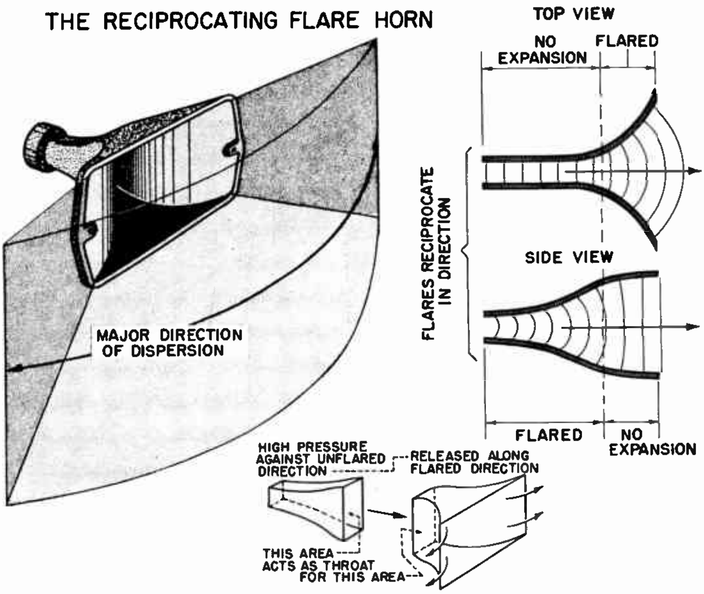


Fig. 3-19. The reciprocating flare horn builds up high pressure against the walls in unflared direction; pressure is then released into the front flared section to produce uniform wide angle dispersion.

expand. We might therefore say that looking into this rectangular section of the horn, there are high pressures against the unflared vertical walls, and low pressure against the flared horizontal walls of the horn.

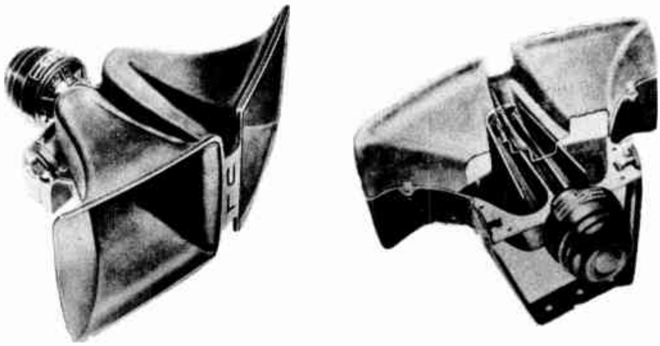
Now, any one cross-section of a horn may be considered to be the "throat" area of the horn section in front of it; that is, one section feeds the next, and so forth. Thus we may look upon one horn section as the "driver" for the next section, with the very important attribute that it is a driver that is pushing greater sound pressure against the side walls than against the top and bottom walls. If, at some discrete point along the horn, we change the direction of the flares so that there is little expansion into the vertical direction and more into the horizontal direction, we have in effect a new horn with a flared directional advantage for horizontal dispersion, energized by a "driver" with already existing high sound pressure against the vertical walls anxious to be relieved into the horizontal direction. The overall effect of this reversal of the flare direction and the differential pressures at the throat of the reversed section contributes to exceptionally wide horizontal dispersion.

In the design of the horn, care is taken to so proportion the horn expansion that the final mouth of the horn is large compared with the wavelength of the frequency to be radiated. By this precaution, minimum diffraction of energy into the vertical plane takes place; it is all concentrated into the horizontal plane. In consequence of the energy being restricted from radiating into the vertical direction in free space and being concentrated in the horizontal plane, the overall efficiency of the horn is increased. In consequence of the combined design of the reciprocating flares of the horn, its mouth configuration, and total lack of cells or fins, the reciprocating flare horn delivers high efficiency wide angle response of the entire high frequency spectrum without acoustical losses due to diffraction or irregularities due to beaming.

In summary, we see that because of the restricted range in which they are designed to work, tweeters may be made quite specialized in their design. They may be made more efficient, and features highly desirable for high frequency applications may be incorporated. These features are obtained from small light moving diaphragms with tight suspensions producing high natural resonance. They usually are of the sealed compression unit type, operating with horn loaded systems designed for high efficiency and wide angle response.

The Midrange Speaker is a "Band Pass" Speaker

There are many good high fidelity systems that consist only of a woofer and a tweeter. The more elaborate systems utilize in addition a midrange speaker for even more controlled acoustic performance. This midrange unit may be either a cone speaker or a horn driven system. Its design is again covered by the principles that control loudspeakers in general, but in this instance, they are adapted to more specialized designs, because of the restricted range over which the mid-frequency speaker is to perform. Those factors governing the choice of the type of mid-frequency loudspeakers are integrally tied



A WIDE ANGLE MID RANGE PROJECTOR FOR USE WITH A COMPRESSION TYPE DRIVER UNIT, SHOWING HALF SECTION VIEW

Fig. 3-20. For low frequency extension of the midrange horn projector, the reflex type of structure gives good horn length in small space for high efficiency operation with compression type driver. (Courtesy University)

up with the overall design of the system, because such speakers must bridge the gap between the woofer and the tweeter and must be designed to be compatible with them. Therefore, the more detail consideration of the midrange speaker will be presented in Chap. 6, which covers multi-speaker system design. We may treat the midrange speaker more generally at this point so that it may be fitted into the complete picture of these hi-fi variations from the basic types.

Midrange Speaker Range Determined by Woofer and Tweeter Limits

The midrange speaker begins to operate in that frequency region where the upper range of the woofer section becomes attenuated either by itself or through the intermediate devices of networks. Thus the expected low frequency response of the midrange speaker is usually fairly high in the acoustic spectrum. The upper range of the midrange speaker is governed by the lower range coverage of the tweeter, for the midrange unit generally need go up in frequency only far enough to provide a smooth overlap with the tweeter range.

The midrange speaker may be either a direct radiator cone speaker or a compression driver horn loaded system. The choice will be governed by the design of the system as a whole. If the woofer and tweeter sections of the system are direct radiator cone types, a midrange speaker of the cone type will match them in efficiency and provide a balanced system. If the woofer and the tweeter are high efficiency horn loaded sections, the midrange unit should preferably be a horn type, although as we will see in Chap. 6, there are other important considerations governing the choice of speaker components. If it is a cone type, the midrange speaker may preferably be of the wide range type so that it may be operated low enough in frequency to match the woofer and at the same time give good high frequency reproduction up to the area where the tweeter takes over.

In the case of the compression driver horn type of midrange projector, the low frequency extension is limited by its natural cutoff frequency, and its efficiency is governed by the length of the horn. Figure 3-20 shows a typical midrange horn projector designed to extend down to 350 cps. This horn, of the reflex type, provides a long acoustic column in a short compact space which results in a high efficiency device for use even in restricted sizes of enclosures. The high end performance of a horn system is determined, in general, by the number and type of bends of the horn, and by the driver unit performance. A driver unit for operation down to the low frequencies required of a typical midrange reproducer will be of such size as to compromise the extreme high end efficiency. However, since this upper end may be taken over by the separate tweeter, this compromise is perfectly acceptable. In fact, the compromise breeds specialization. In order to get optimum performance over the restricted *middle* range, the horn driver system is designed for maximum efficiency within that range.

Summary

Woofers are low frequency reproducers, large in size and low in resonant frequency; tweeters are treble reproducers, light in size and high in resonant frequency, and usually of the compression driver horn loaded type; and in between the two are the midrange speakers, chosen on the basis of low end and high end response compatible with the woofer and tweeter portions of the reproducing system.

CHAPTER 4: *The Mechanics of Good Hi-Fi Loudspeaker Design*

Speaker Performance Depends Mostly Upon Mechanical Design

The acoustic designs of the specialized hi-fi loudspeakers discussed in the previous chapter stem directly from the principles of good mechanical design. In that chapter we touched briefly on diaphragm weight, diaphragm stiffness, and related matters simply to lay out the broad boundary conditions for the design of the various types of loudspeakers. In the present chapter, we will show how these general conditions are specifically met through the proper mechanical design of the component parts. We will see that even though the loudspeaker is a device to be used with electronic equipment, it has very little about it that is specifically electronic in nature. If we discount the few turns of wire on the voice coil, the entire loudspeaker is an integrated system of mechanical parts in motion relative to one another. The quality of performance of the speaker, the range of its frequency response, its sensitivity, its angular dispersion will, in large measure, be dependent upon the design and mechanical structure of its parts, and the precision with which these parts move.

It will come as a revelation to the newcomer to acoustics to learn that when the loudspeaker is working it may undergo stresses and strains far in excess of the stresses and strains that airplanes undergo in coming out of very steep fast dives. In analyzing these stresses and strains, we shall start with the diaphragm because the diaphragm is the major element; it is (more specifically) the one and only element that actually provides the sound we hear. The operating conditions prevailing about the diaphragm will naturally be determined by whether the speaker is a large woofer structure or a small tweeter structure. As we have seen, the woofer diaphragm is comparatively heavy and large and undergoes high degrees of excursion, whereas the tweeter diaphragm is small and light and undergoes restricted degrees of excursion. It will be only natural, therefore, to find a different set of mechanical properties for the woofer moving systems than for the tweeter.

Diaphragm Withstands Tremendous Mechanical Stresses

The electrical power fed to the voice coil of a speaker is transformed into mechanical power, which moves the diaphragm, and the mechanical power of the moving diaphragm, in putting the air into vibration, is converted to acoustic power. For instance, it may be shown by acoustic calculations that a 12-inch diaphragm has to move to a peak excursion of ¼-inch when vibrating at 60 cps in order to radiate one acoustic watt of power. When it reaches its peak excursion, it has to come to a complete standstill for an instant, and then start going backward to complete its vibration cycle. In other words, it has come out of its dive as the airplane does, and reverses its direction. By the laws of physics, it may be calculated that when the diaphragm comes out of its "dive," under the above power conditions, it experiences a maximum acceleration equivalent to 93 "g's." More astounding yet, if similar calculations were to be made for this speaker when reproducing 10,000 cps, we would come up with the fantastic figure of 2060 g's for its acceleration. When we consider that ordinary airplanes may virtually come apart when subjected to more than 10 g's, we must look with deep respect upon the vibrating paper diaphragm of a loudspeaker.

Diaphragm Stability Determined by Size, Shape, and Paper Stock

In order to ensure that diaphragms withstand these tremendous stresses and strains, great care must be exercised both in the design of their physical shapes and the selection of material out of which they are made. As shown in Fig. 4-1, a flat piece of paper supported at its edges will be rather weak at its center. It will readily give under the gentle press of one's finger. On the other hand, if that same piece of paper is rolled up into a conical shape and placed with its wide base down on the table, it is possible to exert considerable pressure upon the point of the cone before it collapses. If we apply this concept to loudspeaker diaphragm design, we find that, in general, the deeper the loudspeaker diaphragm, the more rigid it becomes, and the more it will move as one whole piston. This is actually the condition we seek

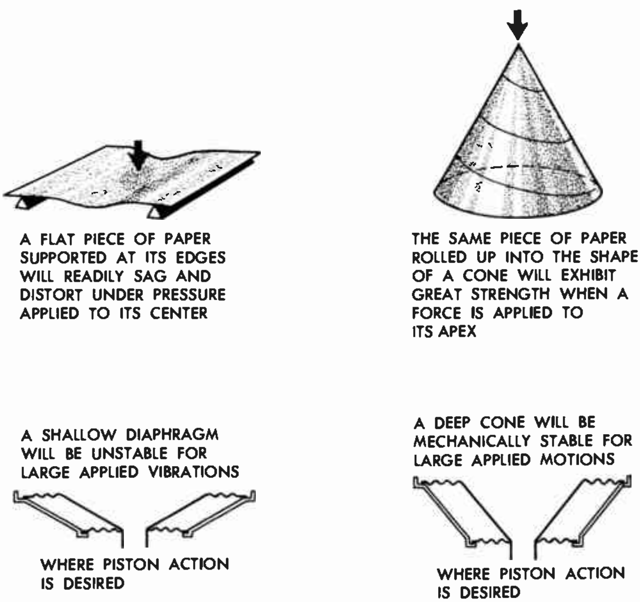


Fig. 4-1. Where complete piston motion of a diaphragm is required, maximum mechanical stability may be obtained by a deep conical structure.

in a woofer. We want the diaphragm to move as a whole without flexing of the vibrating surface itself so that we may get as much active radiating surface as possible. Therefore, we find that woofers, which have to move large volumes of air, obtain great mechanical stability by means of deep cone structures.

The stability of the cone is also greatly determined by the paper stock from which the cone is made. Almost all of today's hi-fi speakers use diaphragms compounded by a molding process or a flotation process by which it is possible to blend in the mixing vats many ingredients to get a paper stock that will perform in the acoustic manner desired. Many kinds of fibers, cloth, wool, and paper go into the making of the diaphragm. By choosing the proper compound, we may get a soft-stock cone, or a hard-stock cone, or any variation between the two. Obviously, the type of stock used will also determine both the mechanical stability of the cone and its acoustical performance.

A soft material will not transmit a force as well as a hard material. This is especially true where vibratory motion is concerned. The soft material acts as a cushioning factor for the force applied to it, and the higher the frequency the greater is this cushioning effect. Thus a loudspeaker cone made from a soft material will vibrate at its apex area when driven at its apex by a high frequency signal, but the rest of its diaphragm area will stand relatively still, due to the cushioning effect of the soft material along its body length. This cushioning prevents the driving force from being transmitted to the body of the diaphragm as a whole. The soft material has been instrumental as a decoupling factor for the high frequencies. Actually, if the diaphragm material is of a very soft nature so that the high frequencies are lost even before they activate the apex area of the cone, the speaker will not reproduce the highs at all and the cone will be of the woofer class.

If the cone is made from a hard material, which transmits the vibrating forces throughout its body, there will be less uncoupling of the body of the cone, and the cone will tend to move as a piston for all frequencies. However, this also results in rather inefficient high frequency reproduction, because the diaphragm is too heavy to be adequately moved as a whole by the weak high frequency driving forces. Thus a compromise must be made by the designer of the loudspeaker in the selection of paper stock for the loudspeaker on the basis of the expected balance between lows and highs and on the basis of whether the speaker is to be a woofer or a tweeter. (See Fig. 4-2.) Often the compromises between hard and soft material are quite visible to the eye. Some cones appear to be treated in their apex area with a stiffening impregnation. This is usually referred to as an "apex dip." It is an application to the apex of the cone of a substance that flows into the fibers and stiffens them, thus producing a relatively hard but small apex area for good high frequency transmission. Beyond the treated area, the cone retains its soft state, and uncouples itself as far as the high frequencies are concerned, but remains fully active along with the apex for the reproduction of the low frequencies. Hence we see that between cone shape and cone material are many of the design paramenters of the loudspeaker performance characteristic.

The loudspeaker manufacturer must compound a design (as far as shape and stock of cone are concerned) that will not only produce the proper acoustic response with maximum mechanical stability, but will also prevent cone "break-up." Break-up is a term applied to the tendency of a weak membrane to separate into separate vibrating sections

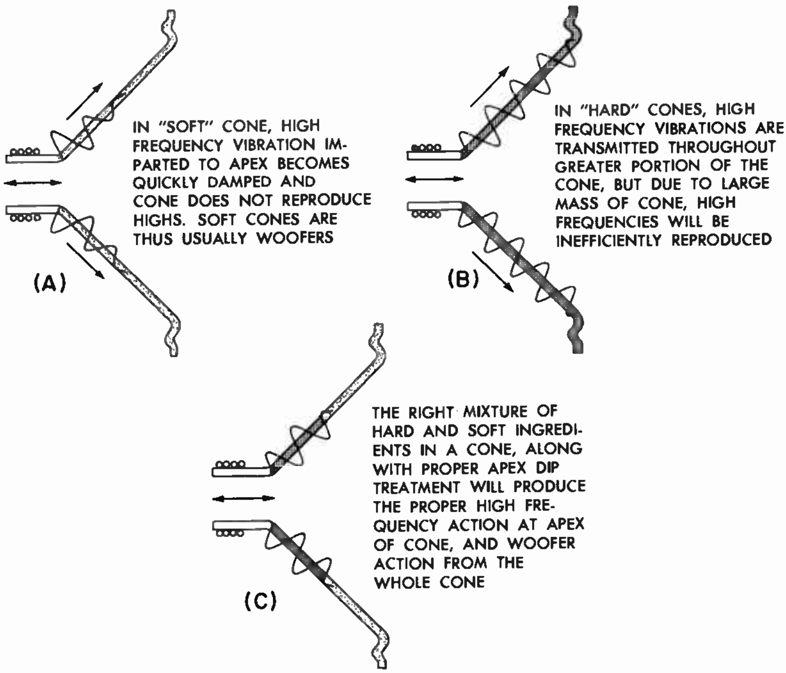


Fig. 4-2. The type of pulp formulation of the cone will determine its acoustical performance, as well as its mechanical stability.

(Fig. 4-3) where the frequencies of these induced vibrations are not harmonically related to the main driving frequency. These spurious vibrations, because they are not harmonically related, result in a "cry" of a very unmusical nature, usually falling somewhere in the middle region of the musical spectrum. Such cone break-up may be partly controlled by cone material, cone shape, and cone reinforcement. This reinforcement may consist of structural ribs placed circumferentially about the midsections of the cone to apply a stiffening factor running over those sections which would normally tend to vibrate independently. (See Fig. 4-4.) These strengthening ribs are put in not for appearance but for performance, and go hand in hand with diaphragm weight, shape, and stock in determining mechanical and acoustic stability.

Woofers Need Large Deep Diaphragms, Tweeters Small Shallow Ones

Thus in woofers, where the diaphragm is large and moves as a whole with large excursions, we find deep cones for strength and rigidity during these large motions. However, in the case of cone type tweeters, where lightness of moving system is highly important, but

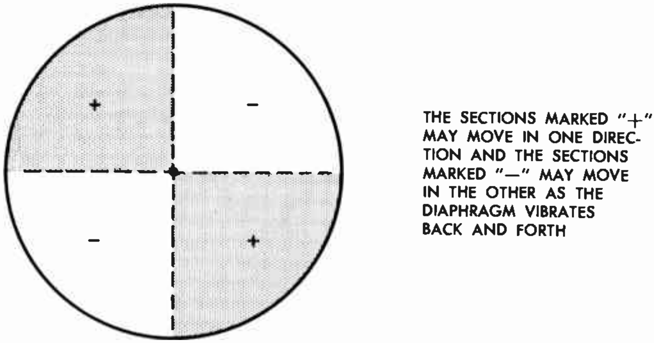


Fig. 4-3. Diaphragm "break-up" may occur in a weak vibrating membrane, causing large sections of the membrane to vibrate at frequencies not harmonically related to the fundamental. This results in cone "cry" at the middle frequencies.

where, fortunately, diaphragm size is small and motion greatly reduced, the diaphragm may be made considerably shallower and of thinner stock. In contrast to the deep mechanically rugged and massive appearance of woofers, cone tweeters have a flat-faced and frail appearance.

In the case of the compression type driver units, the diaphragms are usually made of molded, laminated phenolic impregnated woven cloths. Diaphragms of this nature are molded under pressure and, in addition, are considerably smaller than direct radiator cones. Hence these may be designed to give extreme mechanical rigidity with light-

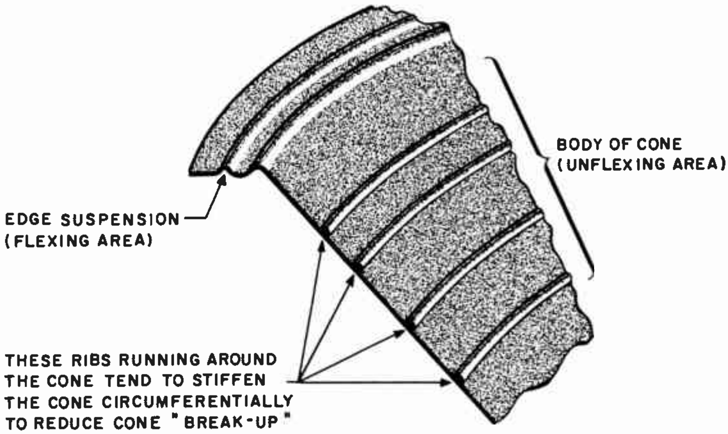


Fig. 4-4. Cone break-up may be minimized by the application of stiffening ribs as well as proper diaphragm weight, shape, and stock.

ness of weight. This makes for overall increased efficiency and cleanness of response.

Diaphragm Suspension Must be Linear

Thus far we have been discussing the mechanics of the diaphragm itself. However, the diaphragm must be supported in its basket by means of its "rim compliance," or edge suspension, as shown in Fig. 4-4. It will be recalled that the heavy woofer cone is comparatively lightly suspended in order to realize the necessary low resonance frequency. "Lightly suspended" means that the paper spring or compliant rim around the edge of the cone, which holds it to the basket, is very loose. One very important characteristic that every rim compliance must have is that of linearity; it must stretch evenly. If it were to pull out too tautly while the cone tried to continue its motion, the motion of the cone would become "flat-topped" and nonlinear motion would result, introducing undesirable harmonic distortion. This is illustrated in Fig. 4-5. Thus we see that a term normally applied to electronic devices is in this case the result of mechanical design of a speaker.

By providing many folds in the paper (Fig. 4-6A), linear spring action may be obtained, so that the diaphragm may move out fully to

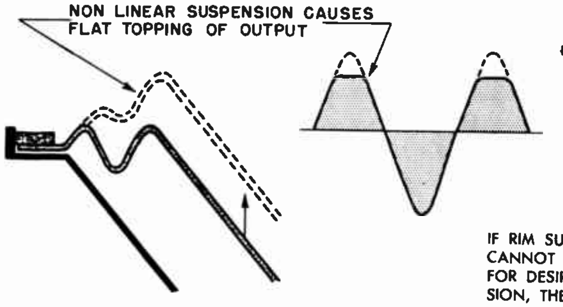


Fig. 4-5. For clean peak signal reproduction, edge suspension must have mechanically linear motion for full diaphragm excursion.

its maximum position without stretching the spring beyond its linearity limits. It might appear that the larger the spring the better. Not so, however. If the spring is made up of many folds, we immediately remove considerable active area from the overall diaphragm, leaving less *active* piston area. Secondly, the large spring begins to exhibit a resonance of its own because it also has its own mass, with the result that somewhere in the middle frequencies the larger rim compliance and the diaphragm piston may actually vibrate out-of-phase with one another. Thus a depressed area in the frequency response, usually referred to as edge resonance effect, will occur (Fig. 4-6C). In order to minimize this effect, the rim compliance may be designed to be small in area, but with large spring loop as shown in Fig. 4-6(B). Although this expedient will eliminate much of the edge resonance effect, it has the disadvantage that, because the edge is restricted, its motion cannot be maintained as linear as with the larger spring. Here again, compromise must be made, and the acoustic conditions of operation are once more determined by the mechanics of how well a piece of fibrous material may be flexed.

Minimizing Rim Resonance Effect

This edge resonance may be minimized by treating the rim compliance with a mechanically resistive material that will effectively dampen the self resonance of the rim area. One very successful means of accomplishing this damping is to coat the edge compliance with special compounds that are "dead" and have no resilience, compounds that tend to stay where they are put. When such a material is applied to the edge of the cone, it imparts its characteristic lack of resilience to the rim, so that even though the rim may still want to vibrate at its own natural resonance, it does so very sluggishly, with the result that edge resonance effects are generally reduced. This edge treatment is easily recognized by the shiny surface that covers the rolled section of the diaphragm. Another means of accomplishing this damping effect is

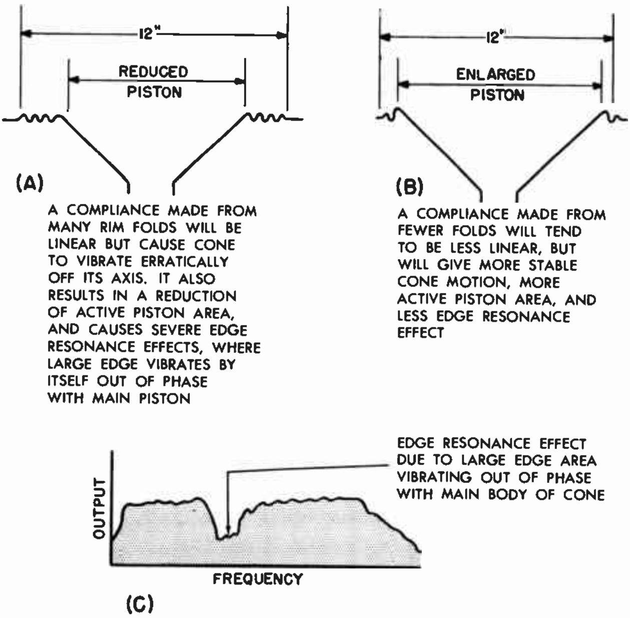


Fig. 4-6. The design of the rim suspension must give good mechanical linearity, but not at the expense of smoothness of response or vibrational equillibrium of cone.

to cement sections of resistive material (such as rubber) over the compliant area sections to provide mechanical damping of these areas. Thus a judicious choice of the edge roll configuration, with proper damping added to it, will give maximum linearity of motion with minimum edge resonance effect.

Woofer Voice Coil is Large, to Dissipate Heavy Audio Power

Due to the fact that most acoustic power resides in the low frequencies, woofers must be designed to produce large amounts of acoustic power. However, they must accordingly be able to take large amounts of electrical power. No piece of machinery is 100 percent efficient. There are losses of power in all physical systems, and loudspeakers are no exception. A great proportion of the electrical power fed to a loudspeaker is not converted into acoustic energy. It is lost as heat or in other ways. The voice coil structure of a woofer must be designed to dissipate large amounts of heat, or it may suffer serious damage and cause failure of the loudspeaker. Accordingly, we find that the voice coils of woofers are usually large in diameter, at least 2 inches, and that the coil is wound of comparatively heavy wire so that maximum dissipation of heat may be accomplished. However, there are limits to the size of coil that may be designed. Obviously, we could design a large massive coil that would easily accept and dissipate all the heat we required. However, when we realize that in addition to dissipating heat, the coil has to move itself under its own force, it will be realized that sooner or later we would have good heat radiation but, due to the massiveness of the coil, no motion and no sound. There is a definite relationship between the weight of the coil and the weight of the diaphragm for optimum acoustic performance, which determines the overall size of the voice coil for a given weight of diaphragm.

Within this general size, the loudspeaker manufacturer must design his voice coil structure to give optimum acoustic output with most efficient electrical power handling capacity. Since metals are good conductors of heat, voice coils wound on dural coil forms add to the effective heat dissipation of the coil by absorbing the heat from the coil through conduction. The large dural surface of the voice coil form then radiates the heat into the surrounding iron parts of the speaker structure.

Power Capabilities Depend Also Upon Diaphragm Stability

The power handling capacity of a loudspeaker is not a function only of the heat its voice coil can dissipate electrically. Its power rating is also governed by the ability of the vibrating system to withstand continuous cyclic motion without physical change. Thus, even though a loudspeaker voice coil may be able to take 25 watts of electrical power, the diaphragm itself may move so violently that its compliant edge quickly cracks or tears away, or the voice coil may ride erratically in the air gap of the magnet and scrape itself to destruction against the walls. Thus power handling capacity is an overall figure that must cover not only the electrical characteristics but the physical characteristics of the speaker as well. The mechanical power handling capacity of the cone itself is usually determined by the design of the rim compliance and the ability of the cone to vibrate without acoustic break-up. If the rim is designed to be able to withstand constant and excessive flexing without breakage, the diaphragm as a whole will last for the life of the speaker.

However, the mechanical precision with which the suspension holds the whole assembly together determines the mechanical longevity of the voice coil itself. The voice coil must be kept perfectly aligned mechanically in its magnetic gap at all times during its vibration cycle. If, during its vibration, it sways or twists sideways, it may scrape itself against the walls of the gap, causing rubbing of the voice coil, which in turn results in distortion, in shorted turns, and sooner or later, in an open voice coil. Although the spider is normally the device that keeps the coil centered in the gap when no motion is involved, it is the combination of the spider and the outer cone suspension that dictates how the coil behaves in the gap under motion. Obviously, from Fig. 4-7 (A) it will be seen that, even though the spider may keep the coil centered, if the cone tilts, it will tilt the coil and cause rubbing. Where small motions are involved, as in tweeters, this condition does not present a problem. However, in the case of woofers, where voice coil motions of the order of 3/8 inch or more are prevalent, added insurance against voice coil misalignment may be had by the double spider suspension shown in Fig. 4-7 (B). The addition of the second stabilizing spider part way up the cone of the speaker acts as a second guide to the linear in-and-out motion of the voice coil, keeping the voice coil not only rigidly centered, but also moving in a rigidly proscribed axis.

In loudspeakers in which the voice coil travels a straight and narrow path, and in which the side play of the coil is rigidly controlled, the clearances between the overall gap walls and the voice coil may be reduced, with resultant improved magnetic circuit efficiency, because of

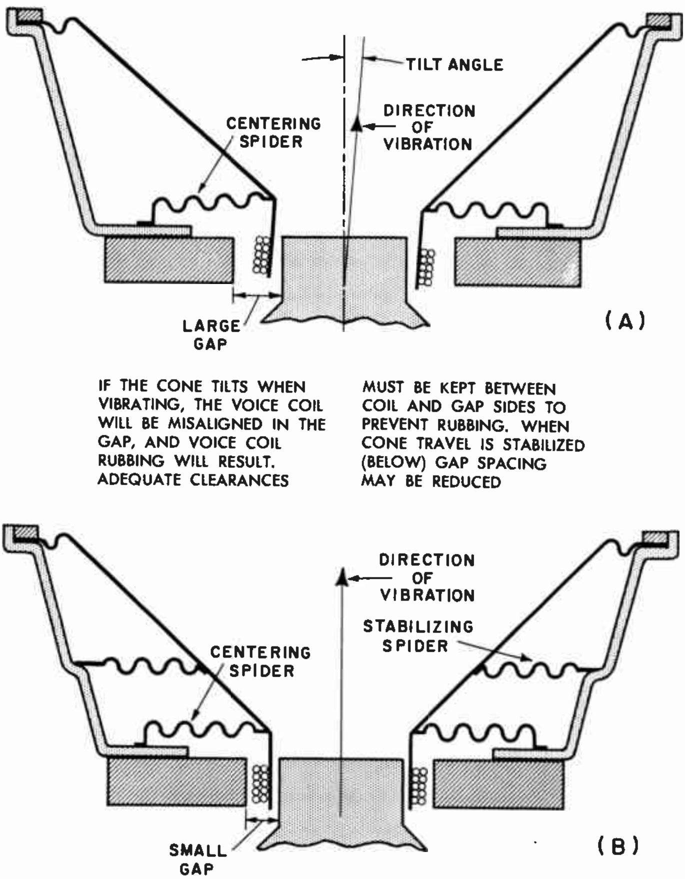


Fig. 4-7. Where mechanical vibration is precisely controlled, greater magnetic gap efficiency may be obtained.

the reduced gap cross-section. Thus we see again that the final acoustic performance of the loudspeaker is a function of the mechanical tolerances permissible between controlled moving parts.

Magnetic Circuit Construction Determines Gap Flux

In connection with magnetic circuits and their dependence upon mechanical design, a type of magnet structure has been developed that is designed for high magnetic efficiency, and which at the same time provides means of forcibly cooling the voice coil that rides in its magnetic gap. This is the "W" magnet illustrated in Fig. 4-8. In any magnetic circuit, the more breaks there are in the circuit, the lower is the efficiency. No matter how well two adjoining surfaces of a magnetic circuit are in contact, such contact is made only through a large number of small areas, because no surface is absolutely smooth. Thus, two

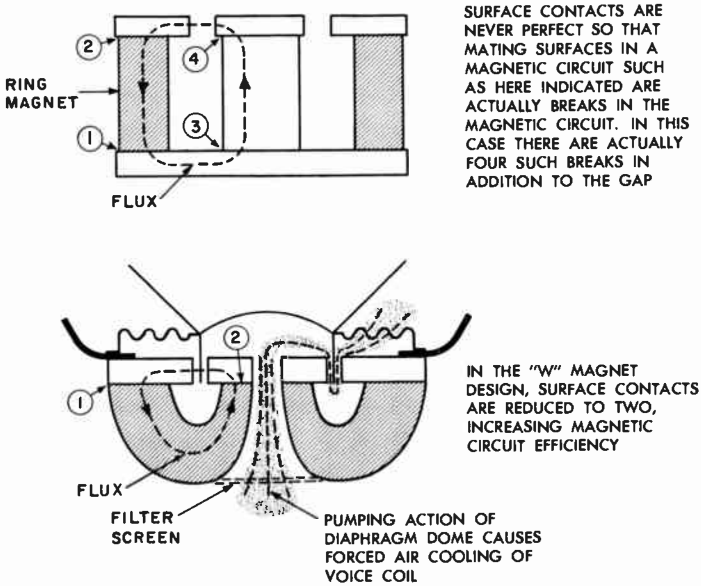


Fig. 4-8. Magnetic circuit efficiencies may be increased by reducing number of surfaces.

facing surfaces are actually a break in the circuit, and even though it is small, this represents a loss of magnetic flux. There are usually four such contact breaks in the magnetic circuit, but in the "W" magnet assembly there are only two, and greater magnet efficiency is therefore made available.

The open center of the "W" magnet, working in conjunction with the apex area of the diaphragm, and closed by means of the dural cap, provides a forced cooling system for the voice coil. As the diaphragm pulses back and forth, the air is sucked in and out of the voice coil area through the rear filter screen.

Heavy Magnetic Circuits and Voice Coil Alignment Require Sturdy Baskets

The size of the magnet of a loudspeaker is, in a large measure, dependent upon the size of the speaker. Large woofers are driven by heavy magnet circuits of perhaps 7 to 10 pounds. These magnet circuits, which are invariably located at the very rear of the speaker, represents a heavy dead weight on the structure holding the entire speaker together. If this speaker basket, or housing, is not properly designed, the weight of the magnetic circuit may cause the basket structure to be deformed to the extent that the entire alignment between diaphragm, voice coil,

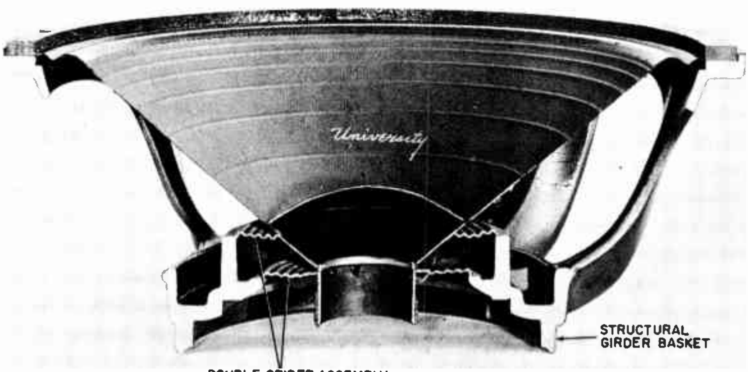


Fig. 4-9. Girder basket construction and double spider construction ensure minimum basket deformation and precision voice coil alignment. (Courtesy Audio Engineering Society, July 1954, "Mechanics of Good Loudspeaker Design")

and gap is upset. Furthermore, improperly designed housings may also become warped out of round when they are screwed down to irregular baffles or plates. This stressed warping of the basket causes misalignment of the voice coil in the gap. Today's loudspeakers, designed for good high acoustic output at low frequencies, call for more and more pounds of magnet. More pounds of magnet means greater strain on the supporting structure. Where the design calls for high conversion efficiency at low frequencies, it is desirable to build the basket as structurally strong as possible, so that it may carry the weight of the magnet necessary to power the speaker. Figure 4-9 illustrates in a cutaway view the structural girder design that a modern loudspeaker should use to insure optimum resistance to mechanical stress on the basket structure.

Summary

We have now examined all the component parts of a loudspeaker, from the voice coil, which first receives the electrical impulse, down to the basket, which supports the whole assembly. Also, we have seen that those elements of design that are necessary to provide structural and motional stability for the loudspeaker also bring about improved acoustical performance. We must conclude that good loudspeaker performance can only stem from good mechanical design.

CHAPTER 5: *The Meaning of Loudspeaker Resonance, Impedance, & Damping*

Loudspeaker Impedance Varies with Frequency

The performance of a loudspeaker is greatly affected by its impedance characteristic. There is far more to the matter of loudspeaker impedance than simply that of its 8- or 16-ohm nominal specification. The impedance of a loudspeaker is not a constant. It varies considerably with frequency. The impedance of a speaker at its low frequency resonant point may be four to five times its rated impedance At high frequencies it may again be more than four to five times the rated impedance. Since the power a loudspeaker can accept from an amplifier is dependent upon the speaker impedance, it is apparent that the output of the loudspeaker will be greatly affected by the manner in which its impedance varies from point to point throughout its operating range.

Rated Impedance Usually Taken in 400-cps Area

The rated impedance of a loudspeaker is actually the value taken at one small narrow band of the overall impedance curve. Figure 5-1 illustrates an impedance curve of a typical 12-inch speaker as taken in free air; the speaker is unrestricted by any baffled conditions. The standard method of rating the impedance of a loudspeaker, according to RETMA standards, is to choose the value of the impedance at the bottom of the trough immediately following the first peak of the impedance characteristic. It so happens that in the conventional 15-, 12-, and 8-inch speakers, this trough falls in the general neighborhood of 400 cps. Therefore, the rated impedance of the speaker usually holds at this general mid-low frequency. However, the impedance characteristic may convey a great quantity of information, in addition to the rated impedance, that will be extremely helpful in understanding the performance characteristics of a loudspeaker.

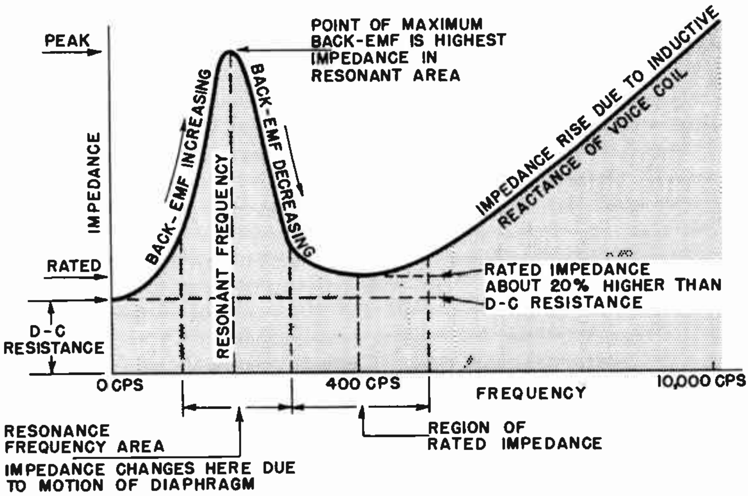


Fig. 5-1. The impedance characteristic of a loudspeaker will determine the power that the speaker will accept at low and at high frequencies in relation to the power at its rated frequency and the damping required.

Actually, the rated impedance does tell us what the matching impedance of the amplifier should be to get the best power match to the speaker at approximately 400 cps. At the low-mid frequency range, an 8-ohm speaker should be connected to the 8-ohm output tap of an amplifier in order to obtain the best performance from the combination. The rated impedance of the loudspeaker is also a means of determining what power is fed to the loudspeaker. Thus, when 8 volts is applied across an 8-ohm speaker, 8 watts will be delivered to the speaker. These are some of the simple facts conveyed by the rated impedance of the loudspeaker. However, we must approach these facts with considerable caution, realizing that they apply only for a relatively narrow frequency band (specifically, the low-middle and middle frequency ranges).

Impedance Varies Greatly at Resonance

The greatest variation in impedance of a speaker is in the vicinity of the resonant frequency. All mechanical vibrating systems have a natural resonance. At this natural resonance, they vibrate most freely. The degree to which any mechanical body vibrates at its natural frequency is dependent upon the mass of the body, the springiness by which the mass is held, and the mechanical resistance to motion within the system itself or to resistance externally applied.

If a weight were hung by a spring suspended from the ceiling, and the weight were pulled down and then released, it would bob up and down at some rate, or frequency, which would be determined by the actual weight of the mass and the stiffness of the spring. If the spring were a very loose one, the weight would bob up and down *slowly* over *large distances*. The system would then have a low natural resonance, and a large "excursion," or amplitude of motion. If the spring were a very stiff one, the weight would bob up and down *rapidly* over *small distances*. The system would now have a high natural resonance and small excursion. On the other hand, if the spring were kept unchanged and the weight were reduced, the resonance would rise and the excursion would decrease. Therefore, for mechanically vibrating bodies, the heavier the mass, the lower the resonant frequency; the stiffer the suspension, the higher the resonant frequency. In general, then, if we have a *large ratio* of stiffness to mass, we will have a high resonant frequency. If we have a *low ratio* of stiffness to mass, we will have a low resonant system.

These factors are determining elements in the natural resonance of the loudspeakers, and they help to explain why larger speakers have lower resonant frequencies and smaller speakers higher resonant frequencies. In a 15-inch woofer, the cone is relatively heavy because of its size. In order for it to be able to move over the considerable distances necessary for good low frequency reproduction, the cone must be rather loosely suspended in relation to its weight. The woofer diaphragm system, therefore, has a small ratio of stiffness to mass, and has a low natural resonant frequency. Smaller speakers intended for higher frequency operation have smaller and lighter diaphragms, but these are more tightly held in relation to their mass. There is a high ratio of stiffness to mass, and consequently small speakers and tweeters have a high resonant frequency. The important consideration of this resonance characteristic is how it affects the impedance and operation of the loudspeaker.

Impedance is a highly descriptive term. It obviously refers to the property of blocking or stopping something. In electrical terminology, impedance of an item is essentially a measure of how the flow of alternating or changing current is restricted through it. A high impedance permits only small values of current to flow and, conversely, a low impedance allows large values of current to flow. Therefore, let us examine the highly variable impedance characteristic of the loudspeaker as it pertains both to the electrical circuit with which the loudspeaker has to cooperate and to the way in which the performance of the loudspeaker itself is affected by its mechanical resonance.

High Back-emf at Resonance Blocks Current: High Impedance

The diaphragm of a loudspeaker is attached to a voice coil balanced in the gap of the magnetic circuit of the loudspeaker. Now let us apply a signal to the loudspeaker and examine how it behaves as we change the frequency of the signal. (See Fig. 5-1.) If we start at one extreme, say "zero" frequency, we are applying direct current to the loudspeaker voice coil. Under the influence of this zero frequency (or direct) current, the voice coil is displaced from its equilibrium position and remains displaced without subsequent motion. Under this condition the voice coil appears as a simple resistance, and its "impedance" is therefore only the actual direct current resistance of the voice coil. Now, as we leave the point of zero frequency, and begin to cycle the current, vibration of the diaphragm will take place. As soon as this happens, the voice coil, which now moves in the magnetic gap, acts as a generator of electricity and produces the well known "back-emf." This is the electromotive force generated by the voice coil while in motion in its surrounding magnetic field. According to basic electrical laws, this electromotive force is generated in opposition to the applied voltage. If this internally induced voltage bucks the actual external voltage applied to the voice coil, it is obvious that less current can get to the speaker. Its impedance is thus higher than at the rest condition.

As the voice coil vibrates more energetically over larger and larger distances, and at a faster and faster rate, the back-emf induced in the voice coil rises higher and higher, effectively causing a further increase in the impedance of the loudspeaker. The maximum excursion of the vibrating diaphragm occurs at the frequency of its natural resonance, i. e., where it is most free to vibrate. At this resonant frequency, where the diaphragm is under-going maximum excursion, the back-emf is largest and the impedance highest. As the applied frequency continues to increase, the point of resonance is passed and the diaphragm excursion begins to grow smaller. There is less back-emf produced in the coil, and the impedance begins to drop.

Due to the fact that in the area of resonance the change in apparent electrical impedance of the speaker is almost wholly due to its motion, this impedance is usually referred to as its motional impedance. Since the effective motion of the voice coil is dependent upon the interaction of the field due to the voice coil current with the strength of the fixed magnetic field, we may expect two results if the weight of the loudspeaker magnet is increased. First, the driving force on the voice coil is increased, causing greater motion of the voice coil. Secondly, the back-emf is greater because of the more violent voice coil motion in a denser magnetic field. We should therefore expect the resonant frequency *impedance* of the loudspeaker with a heavier magnetic structure to be higher in value than for the same loudspeaker with a light magnet. The effect is illustrated in Fig. 5-2. This impedance variation with magnet weight is very important as it affects damping, efficiency, and load regulation upon the amplifier, and will be covered in later sections of this chapter after we have finished analyzing the impedance curve as a whole.

Trough (Rated) Impedance is Lowest A-C Impedance

As the input frequency to the loudspeaker is now further increased the resonance area is left behind and the motion of the coil in the magnetic gap becomes gradually limited in excursion while the back-emf is correspondingly reduced. The reduction of the back-emf results in a drop from the high resonance impedance value down to the trough, or rated impedance. (Refer back to Fig. 5-1). This trough impedance is higher than the original d-c resistance of the coil, for now there is added to the coil's d-c resistance the inductive reactance of the coil carrying the alternating current. The total impedance of the voice coil to the flow of alternating current through it is the (vector) sum of its d-c resistance and its inductive reactance, which is a function of the frequency.

In the trough area then the impedance drops to a value close to the d-c resistance of the coil, but somewhat raised by the inductive reactance of the coil at this trough area frequency. In this area the actual

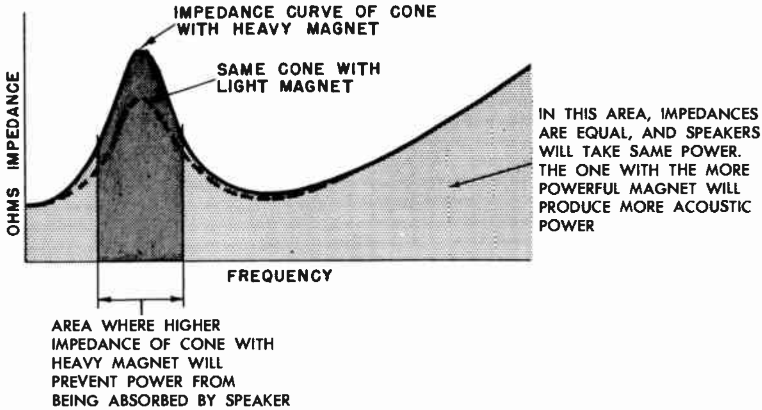


Fig. 5-2. A heavier magnet raises the resonant frequency impedance causing power input discrimination in this area unless the combined speaker enclosure and amplifier system are properly damped.

impedance may be between 15 and 20 percent higher than the d-c resistance. It is therefore easy to ascertain the *rated* impedance of a loudspeaker by measuring its d-c resistance with a simple ohmmeter and then increasing this measured value by 15 to 20 percent. Thus a speaker whose d-c resistance measures 6½ ohms probably has a rated impedance of 8 ohms.

Voice Coil Inductance Raises High Frequency Impedance

After this trough area has been passed, the increasing frequency of the voltage applied to the voice coil causes it to exhibit higher and higher *inductive reactance,* which effectively causes the impedance characteristic of the loudspeaker to rise at a fairly steady rate. Thus, at low frequencies, the effective impedance of the speaker (and its resultant electrical performance) is greatly dependent upon its physical motion; while, at the higher frequencies, its effective impedance (and its resultant electrical performance) is greatly dependent upon the voice coil winding characteristics.

Speaker Most Efficient at Resonance

However, before we are in a position to analyze the true meaning of the loudspeaker impedance characteristic and to see how the overall acoustic response of the loudspeaker is affected by its preponderantly motional impedance at the resonant area and its preponderantly inductive impedance at the higher frequencies, we must examine one further property of vibrating systems that affects the freedom with which the system vibrates. This factor is the *mechanical resistance* of the system.

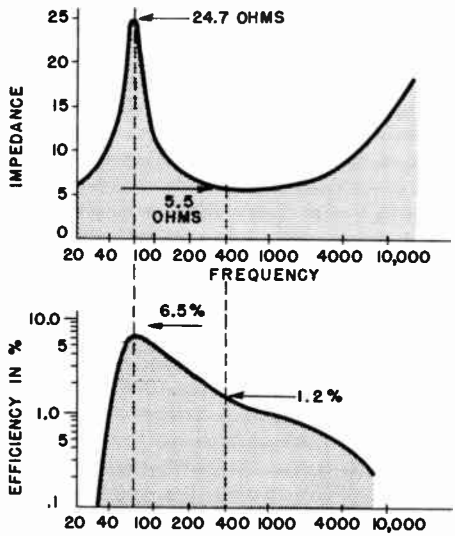
When a system goes into resonant vibration, the frequency at which it vibrates is such that the inertia effects of the dead weight of the mass and liveness of the spring are balanced out and the system wants to continue to vibrate indefinitely. If there were no internal resistance in the system, and no external resistance applied to it, the system *would* vibrate forever even after the driving force had been removed. However, all mechanical vibrating systems have inherent mechanical resistances which limit their motion. Yet, no matter what the magnitude of the restraining resistance may be, the vibrating system still works most efficiently at the resonant point, because of the self cancellation between the mass of the system and the springiness of its suspension. At resonance the driving force has to overcome only the internal resistance of the system. Therefore, at the resonant frequency the vibrating system is working at top efficiency. Consequently, in the case of a loudspeaker, where we look for optimum *low frequency efficiency,* we look for *low resonant frequency.*

Power Input is Minimum at Resonance

We must not me misled by the fact that because the loudspeaker is most efficient at its resonant frequency it is being *operated upon* most efficiently by the amplifier. In fact, because of its high impedance at resonance, the loudspeaker may suffer severe power input discrimination unless proper baffling and damping considerations are followed. If the loudspeaker is connected to an amplifier of a constant-voltage type (such as one having much negative feedback), no matter what the amplifier sees as a load impedance, it will always deliver the same voltage to the load, regardless of its impedance. If the *resonant* impedance of a loudspeaker is 32 ohms, while its trough (or rated impedance) at 400 cycles is 8 ohms, the loudspeaker *will receive* only one-quarter of the power at the resonant point that it would receive at 400 cycles from a constant voltage amplifier. Thus, despite the fact that the loudspeaker may be most efficient at its resonant point, in combination with its amplifier its power *input* efficiency is low.

Now the question may be asked whether maximum acoustic efficiency and minimum power input at the resonant point do not cancel each other. In a general manner, the increased efficiency of the speaker at resonance is balanced by the decreased power getting into the circuit of the speaker. The exact manner in which this balance would take place would be a function of the manner in which the amplifier output voltage changed with the changing load presented to it by the changing impedance of the speaker, and also of the efficiency characteristic contour of the speaker.

Figure 5-3 shows the efficiency characteristic of a certain direct radiator loudspeaker and its corresponding impedance curve. Let us make some sample calculations of expected power output for this loudspeaker when connected to an amplifier that maintains perfectly constant voltage. The impedance at resonance is 24.7 ohms, its efficiency at resonance is 6.5 percent. If we assume that the amplifier voltage is 10 volts, the power input to the speaker is (E^2/Z) or 100/24.7, or 4.05 watts. If the speaker efficiency (at resonance) is 6.5 percent the acoustic power output will be 4.05 × .065, which equals .263 watt. At 400 cps, where the impedance is 5.5 ohms, the input power will be 100/5.5, which is 18.2 watts input. Note that this power input of 18.2 watts at 400 cps is considerably greater than the 4.05 watts input at the resonant point. If we now apply the 400-cps efficiency factor (which is 1.2 percent) to this value, we get 18.2 × .012, which equals .218 watt. Thus, although the power into the speaker at 400 cps is almost 4½ times the power input at resonance, their different efficiencies



	FREQUENCY	IMPEDANCE	POWER INPUT	POWER OUTPUT
	(MEASURED)	(MEASURED)	$P_{IN} = E^2/Z$	$P_{OUT} = P_{IN} \times Eff$
FOR CONSTANT VOLTS INPUT (10 VOLTS)	RESONANT	24.7Ω	100/24.7 = 4.05 W	4.05 X .065 = .263 W
	400 ~	5.5Ω	100/5.5 = 18.2 W	18.2 X .012 = .218 W
12.5 VOLTS IN → (NON-CONSTANT)	RESONANT	24.7Ω	156/24.7 = 6.3 W	6.3 X .065 = .410 W
10 VOLTS IN →	400 ~	5.5Ω	100/5.5 = 18.2 W	18.2 X .012 = .218 W

Fig. 5-3. Although the high impedance at resonance discriminates against power getting into the speaker, the higher efficiency at resonance may overcome this input power deficiency.

virtually balanced out the system to produce nearly equal acoustic output — .263 watt compared to .218 watt.

Now let us apply the same problem to an amplifier whose voltage regulation is not perfect. As an example, let us allow the voltage of the amplifier to go up by 25 percent as we go from the rated low load impedance to the much less heavily loaded condition (24.7 ohms). Thus, if the voltage across the speaker impedance of 5.5 ohms at 400 cps were 10 volts, the voltage across the speaker at the resonant point would be 25 percent higher, or 12.5 volts. Now the power into the speaker at resonance would be $12.5^2/24.7$, which is 6.3 watts. Applying the efficiency factor of 6.5 percent for the resonance frequency, we get $6.3 \times .065$, which is .410 watt. In this case the different power input characteristics of the speaker are *not* balanced out by the respective efficiencies — .410 watt output at resonance compared to .218 watt at 400 cps.

Damping Equalizes Power Input to Speaker

From the illustrated numerical example, it can be readily seen that more electrical power would be delivered to the voice coil of the speaker if the resonant impedance peak were subdued, and that the extent of this increase in input power would be dependent upon the characteristic regulation of the amplifier and the actual degree to which the resonant impedance peak were to be lowered. However, whether this increase in power input would result in a subsequent increase in acoustic output depends upon the manner in which the resonant peak is minimized and upon the baffling of the speaker. Thus, for instance, a speaker may be designed with a large value of mechanical resistance in the moving system, which will tend to restrict both the excursion and the impedance peak. However, much of the electrical power delivered to the speaker under these conditions of high mechanical resistance is wasted in overcoming this resistance. There is thus no essential gain in acoustic output. However, if the speaker is damped through proper enclosure *loading,* in which useful acoustic output is obtained, as in a true horn baffle, the diaphragm moves very little at resonance, though it still delivers full power at high efficiency.

The reason why the diaphragm moves only small distances under a horn load stems from the fact that the large horn mouth is actually the vibrating surface that communicates with the air, and it is quite large in comparison to the actual diaphragm. Hence, in a horn load, the mouth of the horn may be many times greater than the actual piston diameter. For instance, in a typical horn loaded enclosure, the mouth may be at least 14 square feet in area. By horn transformer action, which is discussed in more detail in Part 2, it will be seen that this large "diaphragm of air" at the mouth of the horn is transformed into a high acoustic impedance at the throat of the horn by the "squeezing-in" action of the horn as one looks down the horn toward the throat area. The diaphragm sees this high impedance of the air load of the horn and is consequently restricted from moving over any great distances. It is, as we might say, highly damped by the *acoustic* load of the horn. But because this acoustic load is one that actually is effective in *radiating* acoustic power into the surrounding air, the subdued peak is then representative of useful loading upon the diaphragm; that is, the resonance of the diaphragm is subdued due to the load, and represents useful power consumption. Similarly, but to a lesser degree, in the case of the bass-reflex enclosure and speaker, the final impedance characteristic peaks are considerably lower than the original free air impedance peak because of the acoustic load imparted to the speaker by the baffle and the air which is driven by the baffle conditions. The impedance peaks are not only lowered but actually distributed and spread out over a larger area. This condition is again representative of, and due to, the radiation load as seen by the system, and thus represents good acoustic efficiency, although not of the same order as that of the horn-loaded speaker.

In both of these cases, then, the damping of the impedance peak is due to useful acoustic power being radiated without power being wasted in mechanical resistance. Thus it will be seen that the increased power into the loudspeaker, due to resonance damping, is converted into useful acoustic power to a degree governed by the nature of the damping. Acoustic radiation damping will convert input power to acoustic output. Mechanical or acoustic *loss* methods of damping will only convert input power to heat.

In the case of the horn, additional power gets into the speaker (either for constant power or for constant voltage) at the resonant frequency, due to the extreme damping of the diaphragm motion because of the horn, and this increased input power to the voice coil results in smooth power input to the speaker at resonance, with increased acoustic output at high levels. However, this does *not* represent a peaked acoustic output but rather a smooth acoustic output because the input power to the speaker is smoothed out and the horn itself represents a smooth acoustic load. Over the broad resonance area, there will be an increase in acoustic output due to the acoustic loading and improved power input in contrast to the one specific peak of efficient acoustic output in the case of the unloaded speaker. Similarly, in the case of the bass-reflex enclosure, the speaker diaphragm, although not as beneficially loaded, still exhibits lowered resonance peaks on either side of the normal free-air resonance. The power input to the loudspeaker is thus smoothed out over a considerably broader area than that originally accepted by the free-air resonance peak of the speaker.

One cannot categorically state, however, that there will necessarily be improved acoustic output because of the damping of the speaker, unless one has the complete data concerning the extent of acoustic damping and the manner in which it is applied, and the regulation of the amplifier. Hence one might find that, in a horn-loaded system operated from a perfectly *constant voltage* amplifier, the power input to the speaker might not necessarily be as high in the resonant areas as the power input to a speaker in a bass-reflex enclosure fed from a *constant power* source.

To summarize this problem, then, in order to obtain high efficiency it is beneficial to damp the impedance peak of the speaker for purposes of allowing more power to be admitted to the loudspeaker. The means of damping should be that of good acoustic loading and not resistive mechanical loading or acoustic loss loading. By damping the impedance peak of the speaker more uniform power input and more uniform power output are obtained in sharp contrast to the peaked output of the undamped speaker; and the regulation of the amplifier is markedly effective in determining the smoothness of power input to the speaker.

Damping Improves Transient Response

Efficient damping of the resonant peak, however, does more than equalize the power input to the speaker. It also improves the transient response of the loudspeaker. The moving system of a loudspeaker has a certain amount of inertia. It takes time for a force to overcome the inertia of the moving system when the latter is standing still. When a suddenly applied electrical signal is fed to the voice coil, the force of the interaction between the field it produces and the gap flux tends to move the voice coil. It takes a certain time for the inertia of the voice coil and its diaphragm to be overcome before the system can begin to move. Accordingly, there is some delay between the instant when the signal is applied and the time when the diaphragm responds to the signal. This time delay between the action and the reaction constitutes a form of transient distortion.

The delay time may be reduced by increasing the magnetic field, for then the driving force on the voice coil becomes greater. The inertia of the moving system will be overcome faster, and consequently there will be a shorter delay between the originating signal and the resultant motion of the diaphragm. It is much like the case of two or three people pushing a stalled car into motion slowly compared to a sudden jar applied by a tow truck. Thus a strong magnetic field improves the "attack" time of a note. However, once started, some notes stop sharply, some slowly. Whatever the rate at which the tone decays, the loudspeaker should stop producing that note when the signal stops, or the loudspeaker will continue to "ring" or "hang over" after the signal has stopped.

Here again, for the decaying note there will be faster reaction to the signal if the magnetic field is stronger. Once a system is moving, it will continue to move unless it is stopped. Inertia is just as effective for a body in motion as it is for a body at rest. In the loudspeaker from which the signal has suddenly been removed, there is no voltage driving the voice coil. It immediately stops acting as a motor. However, due to the inertia of the moving diaphragm, the system continues to vibrate. The loudspeaker voice coil now acts as a pure generator, whereas before it had generated a "back-emf." The current generated in the voice coil now flows back *into* the low impedance output transformer of the amplifier. This constitutes a closed circuit for the loudspeaker voice coil and interaction is now set up between the voice coil generated current and the magnetic field, bringing the coil to a stop by means of magnetic braking effects. Thus, if the magnet is weak, there will be a small degree of magnetic braking and the diaphragm will continue to vibrate for some time after the signal has been removed. This condition is usually termed "hangover." If the magnetic field is strong, there will be heavy braking action and the diaphragm will come to an immediate stop, reducing hangover. Thus it is seen that by the use of a heavy magnet field, both attack and decay time response of a loudspeaker are considerably improved, which means that the loudspeaker transient response is improved.

Inspection of the two impedance curves in Fig. 5-2 will show that, despite the fact that the resonant impedance peak of the heavy magnet is higher in peak amplitude than that of the weaker magnet, the two impedance curves come together at the middle and at the high frequencies. This means that for a given voltage applied to both speakers the heavy magnet speaker will allow less current to enter the voice coil in the resonant area than will the weaker magnet structure. Therefore, in this resonant area, the speaker with the heavy magnet will receive less power input than the speaker with the lighter magnet (although the former will be more efficient than the latter).

However, at the middle and the high frequency ranges, where the impedance curves are governed more by the electrical characteristics of the voice coil than by the motion of the voice coil, these impedance curves, being alike, will result in equal power getting into both voice coils. If equal power gets into both speakers, the one with the more powerful magnet will deliver more acoustic power output. Therefore, we want to maintain all the benefits of a powerful magnet at those middle and upper frequency areas, but in addition find some means to overcome the high impedance peak at the resonance area so that the same power input efficiency may be obtained for the heavy-magnet speaker at resonance as is obtained at other frequencies.

Critical Damping Provides Optimum Transient Response

There are several means of reducing this high impedance peak. A great deal may be accomplished with the aid of the enclosure into which the loudspeaker is to work. Also, much can be done through the amplifier that drives the speaker. Whatever the means employed, however, the end effect of any compensation is optimum when the combined circuit of the loudspeaker system and the amplifier has a Q of unity. This means that the *effective* resistance in the total electroacoustical circuit as related to the mass reactance is such that there is just enough resistance in the circuit to prevent the speaker from running wild and overshooting on a signal pulse. (See Fig. 5-4.) When the system is underdamped (running wild and overshooting), the Q is high; when the circuit is overdamped (held way down), the Q is low; when the circuit is operating just right (optimum output without overshoot), the circuit is "critically damped" and the Q is unity.

The amount of damping necessary will, in part, be determined by the extent to which the resonant impedance peak of the speaker rises above the rated impedance of the speaker. This is just one of the important facts evident from the impedance curve. That curve also presents an overall picture of the efficiency of the loudspeaker over the entire frequency spectrum. It tells us the degree to which the input power at low frequencies would be restricted because of the *high motional impedance;* it also tells us how the input power to the loudspeaker is restricted at the high frequencies due to the rising *inductive reactance* characteristic. We might, therefore, say that the impedance curve of a loudspeaker is the clue to its overall performance.

Amplifier Regulation Also Determines Power Input

In the light of the above discussions, it would be worthwhile to discuss in a little more detail the combined effect of this varying impedance curve and the amplifier performance. It was previously seen

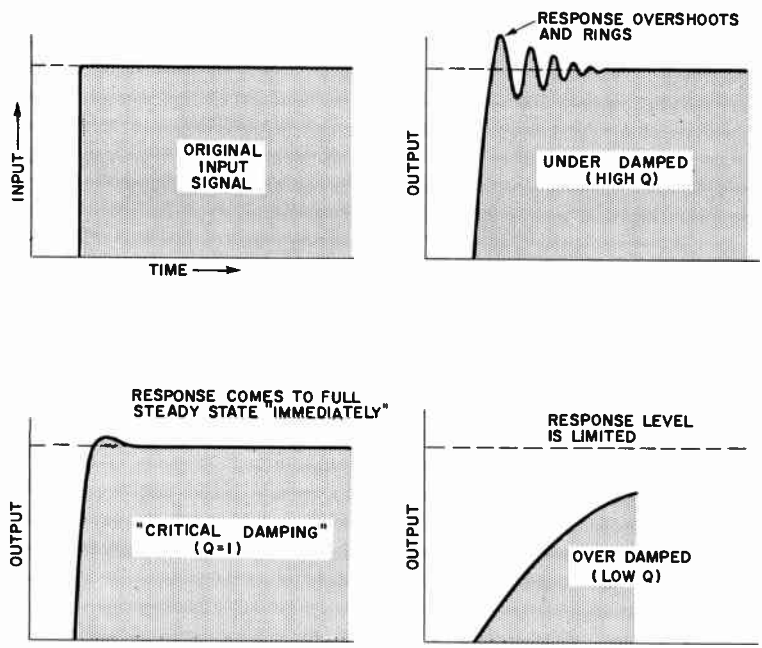


Fig. 5-4. Proper damping will make the loudspeaker respond to a signal without excessive overshoot, and without loss of output.

that constant voltage amplifiers operate in a manner to push the least power into the speaker where the impedance curve is high, as it is at resonance and at the high frequencies. Consequently, there will be comparatively poorer response in these areas. On the other hand, constant voltage amplifiers obtain that particular characteristic by means of large amounts of feedback, which effectively reduces the internal impedance of the amplifier. Reduced internal impedance of the amplifier simply means that there will be improved damping of the loudspeaker. The overall resistive component of the combined circuit of the constant voltage amplifier output and the loudspeaker itself will not be so high that it will exceed the value needed for critical damping of the loudspeaker-amplifier *combination.*

On the other hand, there are amplifiers that may be adjusted to give constant power instead of constant voltage. These amplifiers are sensitive to changes in impedance of the loudspeaker. This means that regardless of how the impedance of the loudspeakers changes, the loudspeaker always receives the same amount of power for a given setting of the amplifier volume control. In amplifiers of this sort, however, the internal impedance of the amplifier is not always kept low, and optimum damping is not always obtained.

Amplifier Compensation Only Partly Overcomes Poorer Speaker Efficiency

The question of damping has recently attracted attention from amplifier manufacturers because it is fairly easy to accomplish good damping by electrical means. The introduction of these various damping means into the amplifier may make the combined amplifier-speaker combination sound better. However, we must not let this approach make us lose sight of the fact that, if the loudspeaker itself is inefficient, the system as a whole will not operate to optimum satisfaction. For instance, it is true that there are amplifiers that may take a speaker with a very weak magnet and, through the amplifier's greatly extended variable damping capabilities, produce just the right amount of damping on the speaker to give it good transient response and input power regulation. However, the overall efficiency of the speaker with the weak magnet is so low that the amplifier may have to be turned way up in gain in order to obtain reasonable output. This may result in overdriving the amplifier on high sudden program peaks with resulting distortion. Therefore, despite the fact that the amplifier can take care of certain deficiencies in loudspeakers, it is better policy to start with the best we can obtain in the way of loudspeaker performance and then use the amplifier compensation to supplement the loudspeaker characteristics rather than to replace them.

CHAPTER 6: *Multi-Speaker System Design*

Multi-Speaker System Provides Flexibility

Multi-speaker systems have much to offer for good high fidelity reproduction, in the way of characteristics that are virtually impossible to obtain from a single wide range speaker. Advantages derived from the multi-speaker system are due to the fact that with two, three, or four speakers in the reproducing system, we have better control of the overall performance characteristic of the system through control of the individual component speakers. The situation may be likened to the difference between having only one ceiling lighting fixture in a room to provide overall illumination, and using several lamps in corners and on tables to provide more adequate light coverage in specific areas where light is most needed and in degrees best suited to those areas.

Special Purpose High Efficiency Components Conserve Audio Power

One important point concerning multi-speaker systems, which we have already covered in the treatment of the specialized speaker, is the greater degree of efficiency that may be obtained from a special purpose speaker. Since multi-speaker systems are composed of combinations of these special purpose speakers, the multi-speaker system as a whole is a very efficient system.

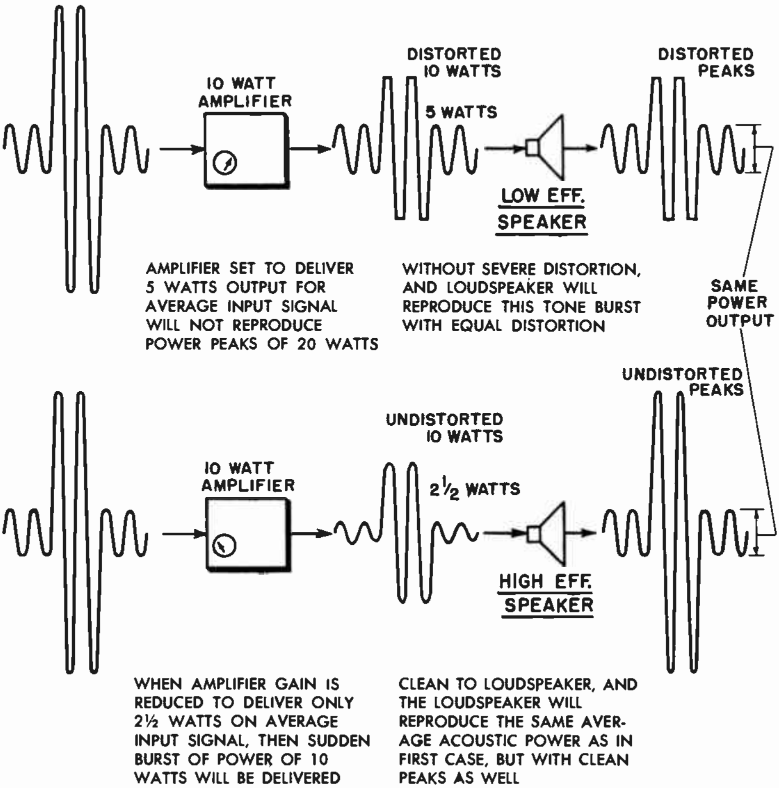


Fig. 6-1. A low efficiency speaker requires a high amplifier gain setting producing distortion on peaks. A high efficiency speaker permits low amplifier gain setting resulting in less overdriving of amplifier on peaks.

But just what does efficiency mean in this case? Primarily, it means greater utilization of the electrical power in transforming it into sound. It means more conservative use of available amplifier audio power, resulting in more adequate reserve in the driving amplifier to take care of sudden bursts of signal. This point should be clearly understood, for it may make the difference between good listening and poor listening. Consider the case of a system with only one low efficiency speaker as a reproducer connected to a 10-watt amplifier. Speaking broadly, 10 audio watts is more than enough to fill the average listening room. In fact, only a small portion of this available power is actually used under normal conditions. The actual setting of the volume control, however, depends upon how loudly one wants to hear the music. The ultimate loudness that the ear hears, however, depends upon more than the amplifier capabilities. If the speaker used is low in efficiency due perhaps to small magnet weight, the sound reproduction may be too low in power, making it necessary to turn up the amplifier volume control. If the room is full of overstuffed furniture and drapes, the high frequencies will be absorbed and the listener will probably compensate for this condition *at the amplifier* by boosting the treble gain control. Thus, despite the fact that it doesn't take many acoustic watts for comfortable listening, the overall listening conditions, plus the speaker sensitivity, may dictate rather more than moderate audio wattages to be fed to the loudspeaker.

Let us, for the sake of discussion, say that the volume control of the 10-watt amplifier has been turned up so that 5 watts of audio power are fed to the single low efficiency speaker system for a given set of listening conditions, and that the music is playing at an even level. Suddenly there is a burst of signal voltage into the amplifier. Perhaps the cymbals of the orchestra have clashed loudly, or maybe the trumpets have blared forth with a loud fanfare. Or suppose the whole orchestra has erupted into sudden fortissimo chords. This sudden signal voltage may in its peak value be four or more times higher than the average signal level before the sudden and momentary onslaught of power. What happens to the amplifier? Having been originally set to deliver 5 watts of audio power, it is now overdriven far beyond its rated power; its output will be severely distorted. The natural consequence is that the speaker will reproduce distorted sound on this momentary high burst of power. (See Fig. 6-1.)

Now let us change the conditions and replace the low efficiency speaker with a much more efficient speaker. What shall we consider a much more efficient speaker? We shall be conservative, and apply the common rule of thumb. If you can just barely detect a difference in the general loudness of the sound, the output from the more efficient speaker is 3 db higher. In actual power measurement, this 3 db represents twice as much acoustic power developed. If we now turn down the volume control of this 10-watt amplifier (which was previously delivering 5 watts) to deliver 2½ watts (half its previous power), we get the same sound output from the high efficiency speaker that we previously obtained with the low efficiency speaker for the 5-watt setting.

Now, with this reduced gain setting, let a sudden tone burst of four times the average power be applied to the amplifier. It will be driven just about to its peak power ($2\frac{1}{2} \times 4 = 10$) and its output distortion under these conditions will be far less than when it is more than 100 percent overdriven, as it was in the first case.

Note that this comparison was made between two loudspeakers in which one could barely hear the difference in sensitivity. It is obvious that an amplifier feeding an exceptionally efficient speaker could operate at a tiny fraction of its rated output, with virtually no danger of being overdriven on even the most severe peaks.

Reserve Power Handling Capacity Necessary for High Program Bursts

Thus we see that no part of the electro-acoustic circuit may stand alone. All parts must be considered as one integrated system. It is interesting to note that there are on the market many hi-fi amplifiers intended for the home that have power ratings of 20 to 50 watts. When used with high efficiency systems, the actual average power drawn from these amplifiers for home consumption is in the order of only 2 to 3 watts or less. The reserve power of as much as twenty-five times the average lies in reserve for those sudden impluses and bursts so that they may be transferred to the loudspeaker with a minimum of distortion. However, these benefits of reserve power in the amplifier may not be reaped without the aid of such high efficiency loudspeakers as the woofer, midrange, and tweeter units designed for multi-speaker systems.

Multi-Speaker System Reduces Intermodulation Distortion

In addition to their ability to conserve valuable audio power, multi-speaker systems also give cleaner reproduction than single speaker systems. There is less intermodulation distortion in a multi-speaker system. This type of distortion occurs when a single vibrating diaphragm tries to reproduce a high frequency note while straining itself to the limit to reproduce a low note. Any irregularity of performance of the diaphragm in reproducing the low note will be imparted to the high note, because the high note, in a sense rides "piggy-back" on the low note, as shown in Fig. 6-2. The high note is thus modulated by

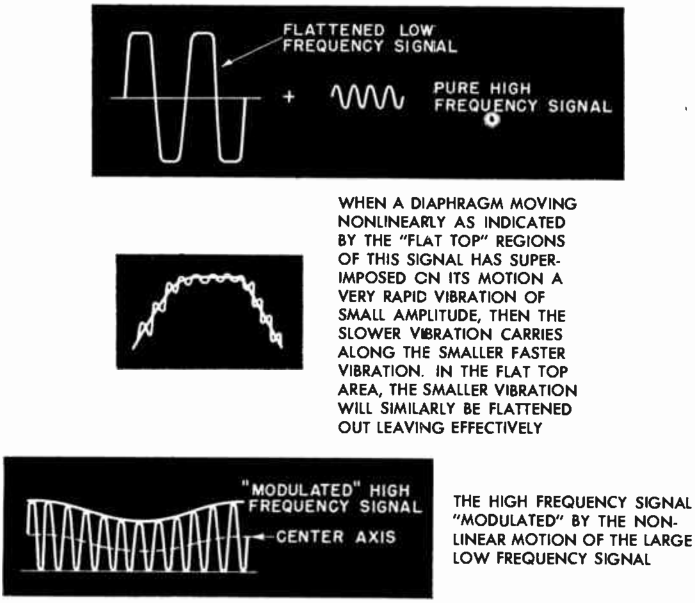


Fig. 6-2. Intermodulation distortion may occur where high frequencies are carried by a low frequency diaphragm into regions of non-linearity of motion.

the low note. If, however, the high note has its own private loudspeaker, it becomes completely free, mechanically, of any low frequency effects, and come what may out of the woofer, the tweeter sounds clean; that is, there is no intermodulation between the lows and the highs. Obviously, then, the more separate speakers there are in a system, each with its own restricted range, the cleaner will be the overall sound.

Multi-Unit Speakers May be Balanced Against Each Other

There is one more general advantage to the multi-speaker system, and that is the ease with which the acoustical picture may be balanced to suit the ear of the individual. Speakers may be balanced, one against the other, by means of volume controls to give that particular feeling of concert hall reality that most pleases the listener. The "presence" of the midrange speaker and the brilliance of the tweeter may be readily adjusted to one's own particular musical taste.

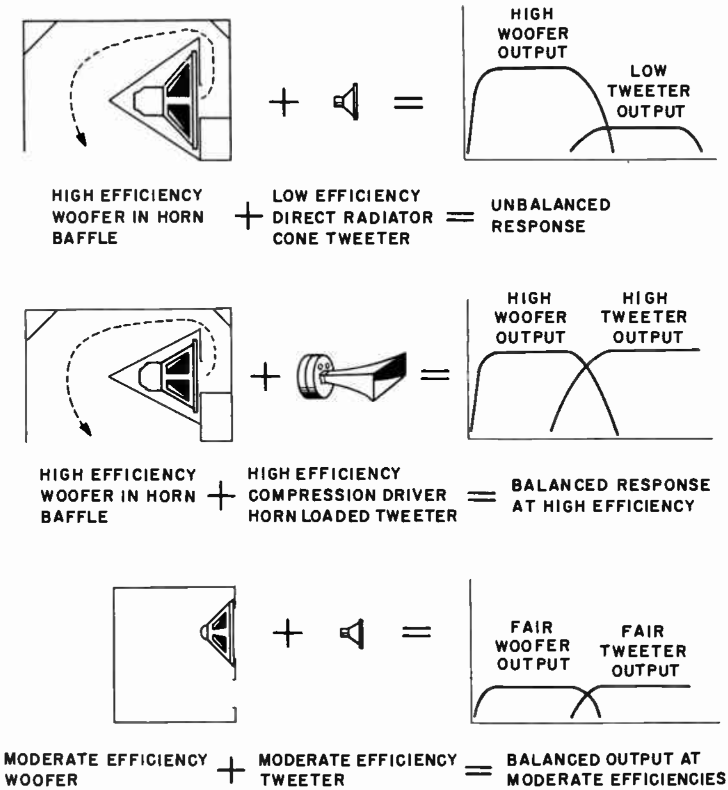


Fig. 6-3. For maximum smoothness of response, speakers should be matched in efficiency.

Speakers Must be Matched in Efficiency

The discussion thus far has been concerned with the benefits that may be obtained in listening pleasure from multi-speaker systems. There are, however, prerequisites that must be met by the components of the multi-speaker system in order to produce these advantages. The speakers must all be matched in efficiency so that a smooth overall response may be obtained, without holes or bands of silence in the audible spectrum. Thus, for instance, if a high efficiency 15-inch woofer were to be installed in a folded horn enclosure (which would further raise its operating efficiency), it would be necessary to balance these heavy lows with a tweeter equivalently high in efficiency (for example, a compression type horn loaded system, such as that illustrated in Fig. 6-3). A small cone type tweeter of the direct radiator type would be completely lost in listening perspective. There would be a complete absence of brilliance and life in the music. On the other hand, a cone type tweeter may be a perfectly good adjunct to a 12-inch moderate efficiency woofer when the latter is used in a bass-reflex enclosure. If the system is a three-way reproducer, all three units should be matched in efficiency so that the listener may have a flat system to adjust in accord with his individual preferences.

Speaker Ranges Must Overlap

Not only must the speakers of a multi-speaker system be matched in efficiency; they must also have compatible "roll-off" or "cutoff" characteristics; if they don't there will be holes in the response curve. Every loudspeaker, no matter how well designed, has limits to its performance. It has a low frequency limit, below which it loses output rapidly, and it has a high frequency limit, above which output also falls rapidly. The more specialized the loudspeaker the narrower the range within which it works, and consequently, the closer in frequency are the low frequency cutoff and high frequency roll-off points. We must be careful, in choosing the components of multi-speaker system, that one speaker's response doesn't fall short of meeting that of the other speaker. Thus, as illustrated in Fig. 6-4, if the woofer horn circuit response fell off rapidly above 500 cps, we could not employ a midrange speaker that started to operate at 1000 cps. If we did there would be a whole octave of depressed response between 500 and 1000 cps. On the other hand, a *minimum* overlapping of the areas of operation of the components of a multi-speaker system is necessary for smooth response, although more extensive overlapping may be permissible under certain conditions. There is, of course, no hole in the response where such overlapping exists; thus smoothness is maintained.

However, other effects not necessarily either desirable or undesirable enter the picture. If the overlapping area of response between the two speakers is very great, the audio power in this region is divided between the two speakers in equal proportions. If both speakers are equally efficient in this common overlapping area, and further, if each speaker is able to handle the additional power at frequencies that may be slightly outside its normal operating range, perhaps nothing has been lost, except that some degree of intermodulation may have crept in due to the increased bandwidth now covered by the overlapping speakers. However, if the overlapping area is not desired, it may be

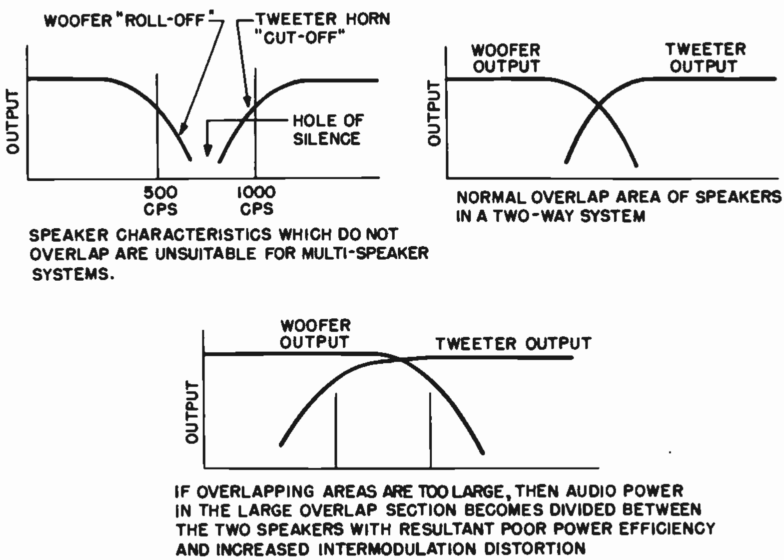


Fig. 6-4. For maximum smoothness of response, speakers should have compatible roll-off and cutoff slopes.

controlled by means of a crossover network (which should be part of every multi-speaker system).

Crossover Networks Determine Speaker Overlapping

The crossover network (Chap. 7) functions very much like a traffic policeman, channeling the various bands of frequencies into speakers specifically designed to handle them. Between the speaker roll-off characteristics and the crossover network attenuation characteristic, the response of the speakers in the overlap areas may be controlled to provide the proper degree of overlap. Too broad an overlap and too slow a roll-off may result in a lack of definiteness of response from the individual speakers. If the listener wishes clear-cut demarcation between the sounds of the various speakers, short overlap and fast roll-off are prescribed. This may produce a feeling of separation of the instruments, producing an illusion of a close-up view of the orchestra. If, however, the listener desires an overall blending of sound, as if he were seated well back in the concert hall, greater overlapping and slower roll-off are required so that there will be no sharp demarcation of sound between the individual loudspeakers.

Those factors which actually determine the roll-off and cutoff characteristics of the speakers are, in general, those of enclosure and baffling conditions, and will be treated in the chapter on the baffle as a crossover element. (See Part 2.) Of immediate importance is the necessity of choosing the individual speakers so that they definitely overlap sufficiently in their respective performance ranges to be subsequently controlled to proper advantage to provide the specific amount of overlap desired by the individual.

Speakers Must Overlap in Angular Distribution

Thus far we have discussed the necessity of matching the speakers of a multi-speaker system in terms of efficiency and compatible overlapping areas of response. In addition to these, there is a third condition of performance that speakers of a multi-speaker system must meet — that of polar response, or angular distribution of the sound, especially at the upper frequencies. This criterion applies to the overlapping area of the midrange speaker and the tweeter of a three-way system, or of the upper end of the woofer and tweeter of a two-way system. Angular distribution is most likely to be a problem when a cone type speaker is used, as is often the case for the midrange unit.

We have seen (Chap. 3) that as the frequency of reproduction of a cone speaker goes up the energy becomes more tightly beamed, with the result that the overall angular performance of the cone drops off. In order to maintain effective smoothness of response projected into all parts of the listening area without holes, it is desirable to bring into play a tweeter of the wide angle type to provide uniform angular dispersion of those frequencies at which the cone distribution begins to deteriorate. Thus, for instance, if we find that at 8000 cps the midrange cone is 10 db down in level for an angular dispersion of 45 degrees off axis, as shown in Fig. 6-5, it may be desirable to roll off the speaker output well below this frequency, say at 5000 cps (where the polar response is fairly good), and bring into play at that frequency the wide angle tweeter, which will carry the high frequency response into the angular position with far less loss than the cone.

By satisfying these three conditions of compatability for the multi-speaker system (equal efficiency, overlapping frequency ranges, and compatible polar response crossover points), we obtain an overall

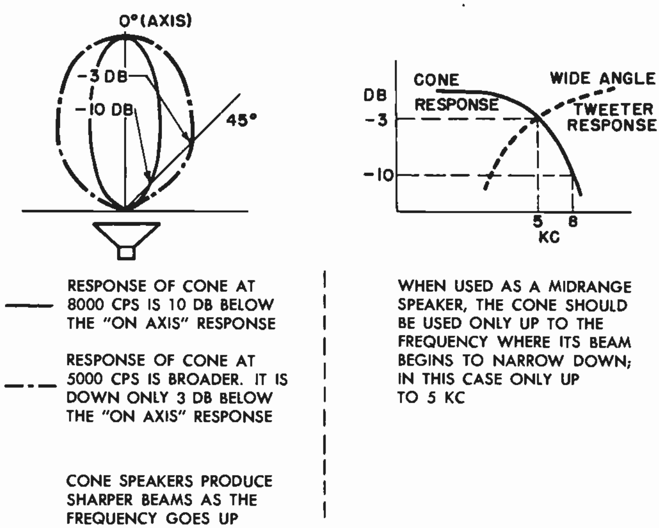


Fig. 6-5. When used as a midrange speaker, the useful upper range of a cone is determined by its high frequency beaming effect. At the frequency where the beam narrows down, a wide angle tweeter should be used.

response that is smooth both in level and in angular distribution with clean performance and a feeling of concert hall reality for the system as a whole.

"Presence" is Determined by Middle Frequency Performance

In describing concert hall realism, a term commonly used in audio terminology is *presence.* This is a neat and specific word signifying a feeling that the performer is actually present in the room with the listener. This feeling is created by the middle frequencies of the reproducing system, a phenomenon that may be easily demonstrated with a three-way speaker system in which the midrange speaker volume is adjustable. When a vocal selection is being played and the midrange speaker level is turned down, producing some appreciable lowering of sound from this speaker, the vocalist appears to have receded into the background, while the rest of the music remains predominantly unchanged. Turn the midrange level back up a little and the performer seems to step right back into the room. The same is true of instrumental selections. When the middle frequencies are lowered in level, the music seems to have lost its reality, its "presence." Presence is an important quality in any single loudspeaker or multi-speaker system. It is more readily obtainable with the more efficient horn type projectors. The very fact that the midrange speaker is a real horn causes the middle frequencies to be projected right into the listener's lap. A stage full of violins, for instance, doesn't have nearly the sound projecting power as a handful of trumpets and trombones.

However, as we explained in the opening chapters, high fidelity is a personal thing governed by one's own personal makeup. There is nothing essentially right or wrong about strong presence in music. If one likes a more remote effect, the cone speaker will provide the proper middle. It must, of course, be realized that this remoteness will be the result of the comparatively low efficiency of the cone speaker as compared to a horn system. There will also be a more subtle difference than that of efficiency. There may be a difference in quality as well. There are those who like horn sounds, and there are those who like cone sounds. If we could provide a set of horn reproducers to play back only the music originally produced by the actual horn type instruments, and cone type reproducers to play back only the music of instruments whose sound is produced by vibrating panels (like the violins), perhaps we would have better fidelity. However, we have to choose one type or the other to play back sounds of *both* horn and membrane instruments. The choice is specifically a personal one, but the source of presence is indisputable. The middle frequencies have it. When they are reduced, presence is reduced. When they are increased, presence is increased.

Multi-Speaker Systems may Grow from Single Wide Range Speaker

Although it is true that multi-speaker systems are made up of special purpose speakers, it is nevertheless possible to develop a multi-speaker system from a single wide range speaker, adding other components as means permit, or as one becomes more familiar with the art. Hence it is possible for the beginner in high fidelity to start out on a modest basis with a single speaker and enjoy it without feeling forever bound to that system, or fearing its obsolescence should he want a more extensive system. Recently there has been a trend to the "progressive speaker expansion" system, which permits the hi-fi fan full enjoyment of that initial choice until he is ready to add further components. It enables him to grow in the art in stages and learn exactly what he wants out of his high fidelity system. We are now in a position to discuss examples of such systems, having already covered the elements of multi-speaker systems.

Added Tweeter Gives Improved and Better Controlled High Frequencies

As an example, let us assume that one's finances have dictated that he start his adventures in hi-fi with a single 12-inch wide range speaker, such as the one shown in Fig. 6-6. Although this speaker by itself may give commendable performance, we know from our listening experience that there are certain limitations in its performance. Perhaps the most fertile area of improvement in a reasonably good 12-inch cone is the extension of high frequency response and of high frequency angular distribution. In order to accomplish the improvement in these directions, we obviously must look to the use of a tweeter. Therefore, to the already existing wide range speaker, we now add a wide angle compression driver horn loaded tweeter that will accomplish both of these objectives. More high frequencies will now be available in the region in which the cone speaker was beginning to roll off; and these newly added high frequencies will be dispersed more uniformly throughout the room.

The addition of such a tweeter is accomplished through the use of a network or a high-pass filter for optimum utilization of the high frequencies of audio power. By adding a tweeter to the wide range speaker, we have expanded our system in frequency, and we have literally added a new listening dimension. Albeit the original speaker may have given adequate and pleasing high frequency response, the addition of the tweeter adds a tingle and a sparkle; a new brilliance possible only with a speaker designed specifically for such reproduction.

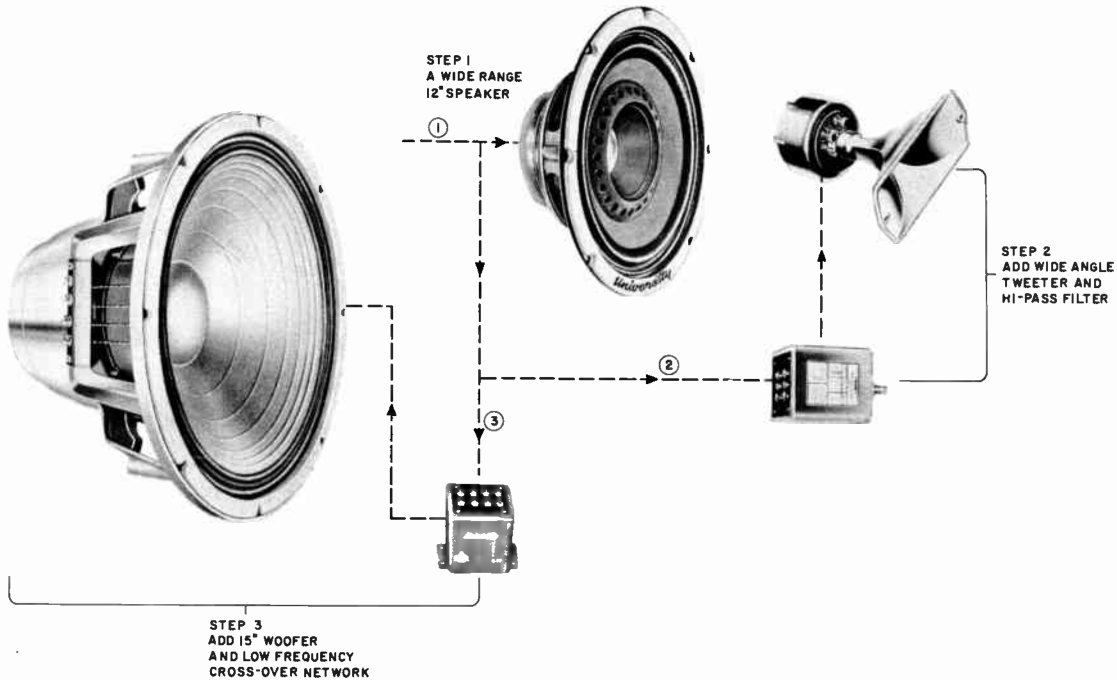


Fig. 6-6. A typical progressive speaker expansion system starting from one basic wide range speaker, expanding first the treble end and then the bass end. (Courtesy University)

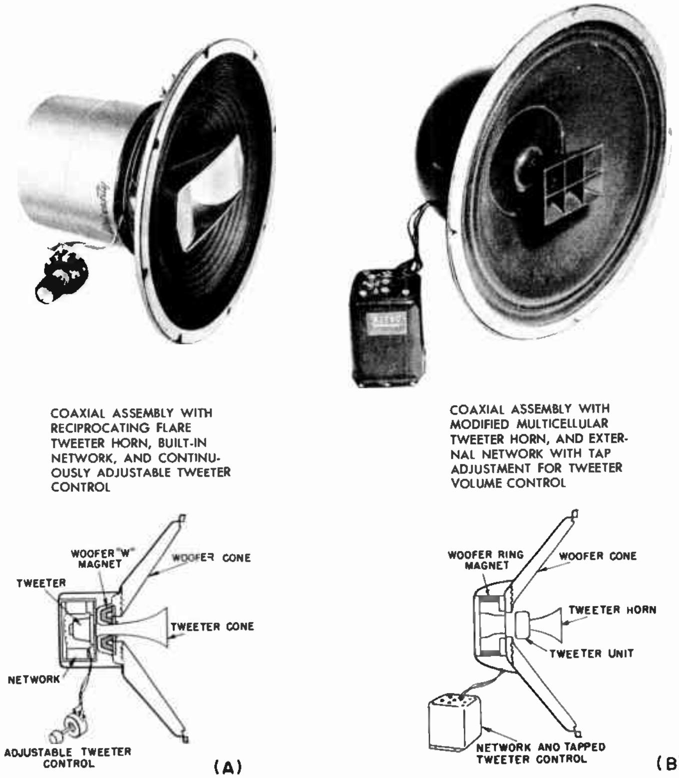


Fig. 6-7. Two typical 12-inch coaxial speakers showing construction, network placement, means of tweeter level control. {(A) Courtesy University; (B) Courtesy Altec}

Added Woofer Gives Improved Low Frequencies

Having now added to the high range and spread of our system, we may stop there for a while, and enjoy the improved performance for weeks or months. The time may come, however, when awareness of the improvement in the high frequency performance of the system fosters a desire for low frequency improvement. Knowing that better low frequencies may be obtained with a 15-inch woofer speaker, it becomes a simple matter to provide a broader base (or bass, if you prefer) for the original system by the addition of such a specialized low frequency speaker as that shown in Fig. 6-6. Again it is desirable that a network be used between the new adjunct and the original speaker to provide the necessary frequency separation between the two.

Original Wide Range Speaker Now Becomes Improved Midrange Unit

In fact, with networks tied to both sides of the original speaker (one at the high end, relieving it of the high frequencies, and one at the low end, relieving it of the low frequencies), the original wide range speaker has now become a specialized midrange speaker. It now reproduces only a restricted middle band of frequencies and does so more cleanly than if it were carrying the whole spectrum. Furthermore, by being relieved of the necessity of reproducing the low frequency range, the speaker may be put into a much smaller enclosure. This smaller midrange enclosure will boost the middle range efficiency of the original general purpose wide range speaker above what it was when matched to the large enclosure. The smaller midrange enclosure acts as an acoustic stiffener to the speaker, raising its resonance frequency and giving it increased efficiency in the middle range. Once again, by restricting its range, we have added more specialization to a speaker, and as a consequence we have obtained better performance within that range. Now we have progressively expanded our system from a single wide range speaker system to a multi-speaker system of specialized components without obsolescence of the original equipment.

Coaxial and Triaxial Speakers are "Packaged" Multi-Speaker Systems

Up to this point we have been treating the matter of multi-speaker systems from the standpoint of separate components. Multi-speaker systems may of course be packaged, or integrated into a single structure, as in a coaxial or a triaxial speaker. There are advantages and disadvantages to both systems; we shall deal with these in due time. Essentially, the coaxial and triaxial multi-speaker units are a combination of specialized speakers, all integrated into one design by the manufacturer of the loudspeaker. If the speaker is a coaxial unit, it has two units, a woofer and a tweeter. If the speaker is a triaxial unit, it has a woofer, a midrange section, and a tweeter. Figure 6-7 shows two typical 12-inch coaxial speakers, with cross-sectional views of both. These structures have their own matched networks to provide the necessary frequency separation between the sections. In one, it is packaged directly with the assembly and there is a continously adjustable tweeter level control. In the other the network is mounted externally, and tweeter volume is adjusted by choosing the proper tap on the network. Thus the manufacturer of each of these two packaged speakers has integrated the components in the manner he deems to give the best balance, for the particular size and price of the speaker.

Multi-unit Speakers are Systems "Balanced" by the Manufacturer

In Fig. 6-8 are shown two triaxial assemblies. In one structure there is a 15-inch woofer section, a midrange diffusion section coupled to the woofer and using the mechanical crossover principles described in Chap. 3, and a tweeter concentrically arranged through the woofer and midrange sections. The other consists of a woofer, a separate midrange speaker, also concentrically mounted with the woofer, and a tweeter unit mounted at the front, off center, so as not to impede the sound issuing from the mouth of the midrange horn. In both structures the manufacturer has provided specifically coordinated three-way systems in which the components and the integrated crossover network are matched to each other in efficiency, in overlapping frequency areas, and in angular distribution. These three factors have been integrated by the manufacturer to provide what appears to him to be the optimum in listening balance, smoothness, and cleanness of response.

However, it is recognized that, despite the fact that the manufacturer has built into his system the ultimate in listening performance as he sees it, the listener should be provided with a means of adjusting the speaker system to suit *his* particular needs and desires. Therefore, controls have been added to the upper frequency portions. It is seldom that the bass end needs any compensation. The actual fact is that we always seem to want all the bass we can get out of a speaker system, and we adjust the treble to match. Volume controls are usually provided as integral parts of the whole assembly of the coaxial or triaxial units to allow the listener to strike his own balance between the highs and the lows.

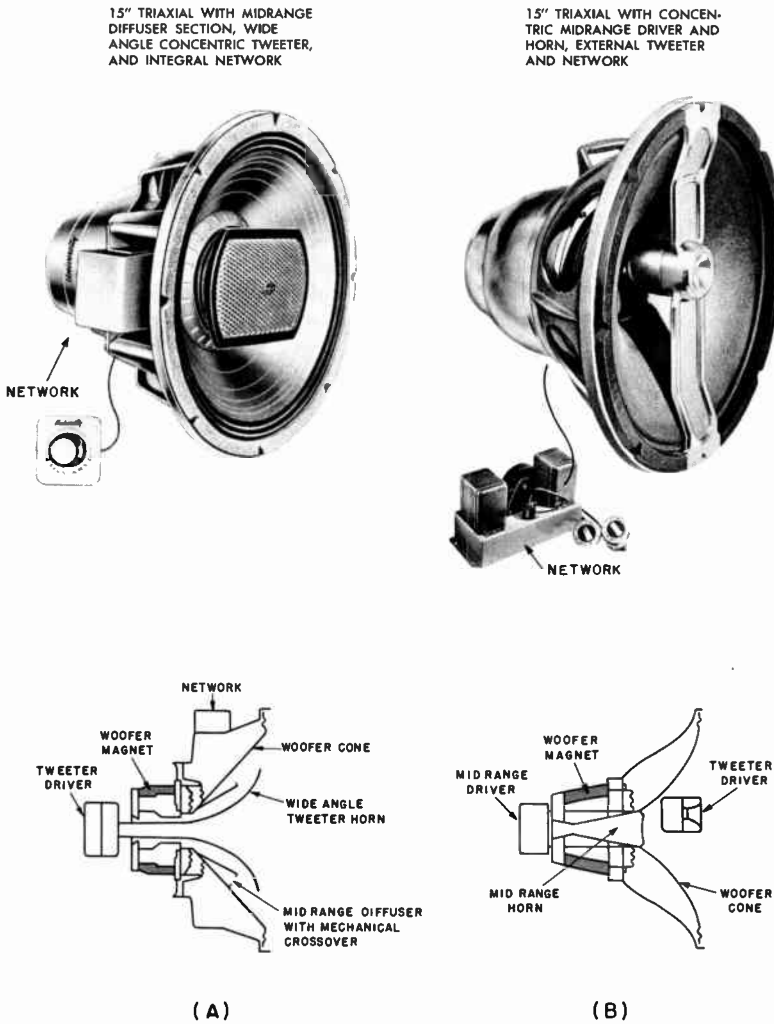


Fig. 6-8. Two typical 15-inch triaxial speakers showing construction and network placement. {(A) Courtesy University; (B) Courtesy Jensen}

Tweeter "Brilliance" Control Affects Level, Not Range, of High Frequencies

It is interesting to note that these tweeter controls do not function as do the treble controls on an amplifier. The tweeter control does not roll-off the high frequencies progressively as it is turned down; it maintains the entire plateau of the frequency range of the tweeter, but subdues the overall level of this plateau, as indicated in Fig. 6-9. Thus we still maintain full frequency coverage but balance the overall level of the various parts of the spectrum.

Packaged System Has Compatible Units, and Needs Direct Radiator Enclosure

Obviously, the complete packaged speaker system leaves nothing to chance as far as the ultimate user is concerned. The system is completely engineered as far as compatibility of the components goes. These

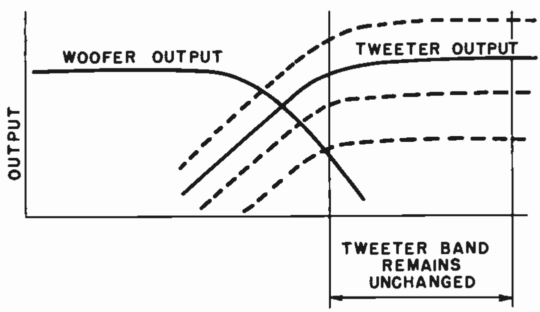


Fig. 6-9. Volume controls of tweeters are usually arranged to raise or lower the entire band of response, thus keeping the tweeter frequency band *unchanged but simply altered in level. This retains full fidelity coverage at any level of operation.*

components are matched in all their important parameters. All the user has to do is to connect two wires between the packaged speaker and the amplifier. For the man who wants a complete integrally engineered and packaged multi-speaker assembly, the coaxial or triaxial speaker is the answer. In addition, simplicity of enclosure design lends itself to speakers of this type. Due to the fact that these integrated assemblies contain their treble reproducers on their front faces, it is absolutely necessary that this type of speaker be mounted in a type of enclosure that permits full unobstructed front radiation from the speaker at all times, so that the high frequencies will flow out directly into the room unobstructed. For this type of radiator, the convenient and efficient bass-reflex cabinet and all its variations make very compatible enclosures. The bass-reflex enclosure affords good balanced bass response, and permits direct forward radiation from the speakers. The bass-reflex enclosure is easy to build and relatively inexpensive to buy. The combination of an engineered package and simplicity of operation makes life easy for the man who wants balanced listening with a minimum of "do-it-yourself."

Multi-Speaker Systems Are Flexible

Although it is not as simple in construction or in choice of components, the separate multi-speaker system has its own definite and specific advantages. A major advantage, of course, stems from the fact that the system may be built progressively from small beginnings. More important, however, the units may be selected on a personal basis. The system may be engineered or put together in ways that will afford the hobbist the widest opportunity to play with the various components until he has nurtured the system to the conditions in which he thinks it sounds best. Such matters as choice of crossover points, attenuation rates of networks, and physical placement or displacement of speakers for different acoustic perspectives make the high fidelity field what it is.

Fortunately, the constructor of a multi-speaker system need not be entirely on his own in selecting the proper components. Many manufacturers have selected combinations of speakers as components for a multi-speaker system on the three point compatibility system (efficiency, overlapped area, and angular spread). The systems shown in Figs. 6-10, 6-11, and 6-12 are considered by their manufacturers to be balanced and compatible three-way systems.

Comparative Packaged Systems

It is of interest to note the approaches of different manufacturers to system design. In the University system, shown in Fig. 6-10, the components responsible for the middle and high frequency ranges are

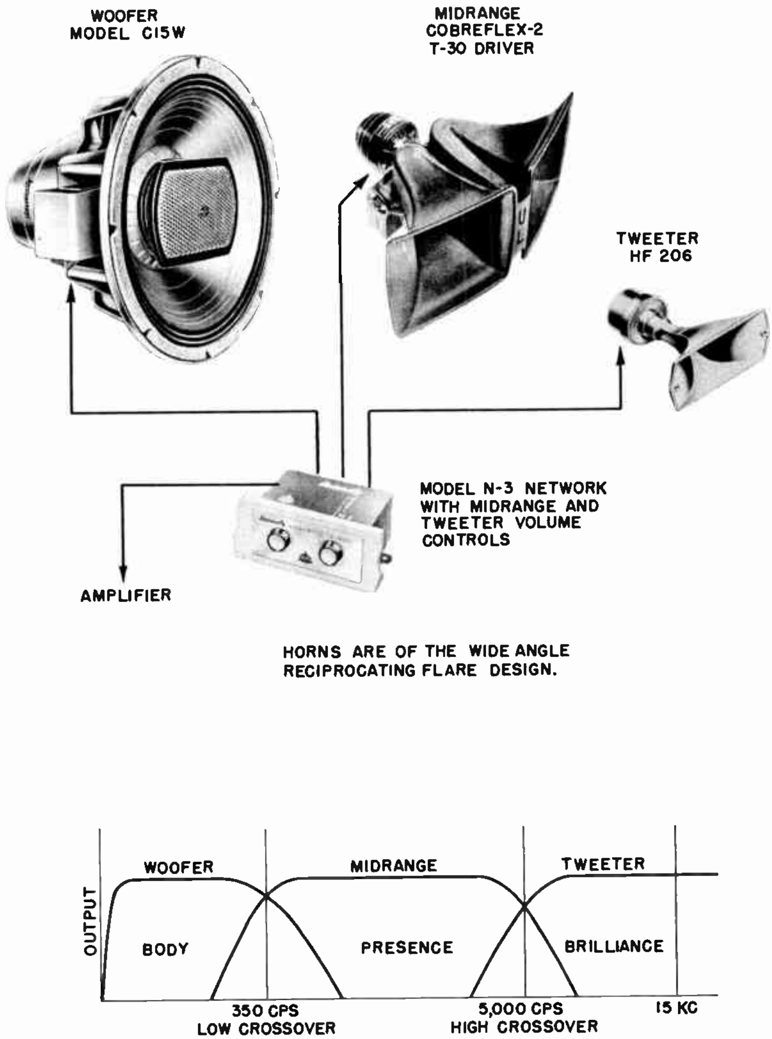


Fig. 6-10. A three-way multi-speaker system comprising components selected by the manufacturer on the basis of compatible efficiencies, overlapping areas, and angular dispersion, with matched network. System uses wide angle reciprocating flare horns and crossovers at 350 and 5000 cps. (Courtesy University)

members of the family of horns, which give wide angle dispersion through the principles of their reciprocating flares. In the Electro-Voice system (Fig. 6-11) horizontal dispersion is accomplished through the diffraction type horn. Figure 6-12 shows a Jensen system in which the middle frequency band is dispersed by a horizontally flared horn and the high frequencies are permitted to radiate in all directions by the symmetry of the high frequency horn. In the University system the crossover points are at 350 cps and 5000 cps, in the Electro-Voice system they are at 800 cps and 3500 cps, and in the Jensen system they are at 600 cps and 4000 cps. In all these systems, crossover points were chosen to be compatible with the cutoff characteristics of the horns.

In choosing components such as these, which have been engineered by the manufacturers to be the equivalent of a package, the home constructor is faced only with the choice of the enclosure; the fact that the units are separate components gives the listener a rather wide choice. He may use a bass-reflex enclosure, and employ direct radiation from the woofer, or he may use a horn system to load down the woofer. As far as the midrange speaker and the tweeter are concerned, they require no baffling, because they are horn loaded. They may be mounted on any convenient panel such as the front baffle board of the bass-reflex enclosure, or they may be boxed in a simple open-front enclosure covered by a grill cloth to be mounted directly on top of the bass enclosure. They may be hung in the mouth of the low frequency horn if a horn loaded woofer is used.

Separate Speakers Provide Acoustic Breadth of Sound

Some measure of increased acoustic perspective may be obtained by actually separating the three speakers from each other by two or three feet, either within the same enclosure or in adjacent equipment cabinets. The physical separation of the speakers, added to the frequency response distribution between the speakers, provides greater apparent acoustic breadth for the reproducing system. The different sounds coming out of the expanded area of, say, 5 feet in width, become much more real than the same sounds struggling to emerge from a single hole in the baffle. If we may speak of this matter rather lightly, the physical separation of the woofer, midrange unit, and tweeter enable the reproduction of violin and trombone music without the violinist's bow getting tangled up with the trombonist's slide.

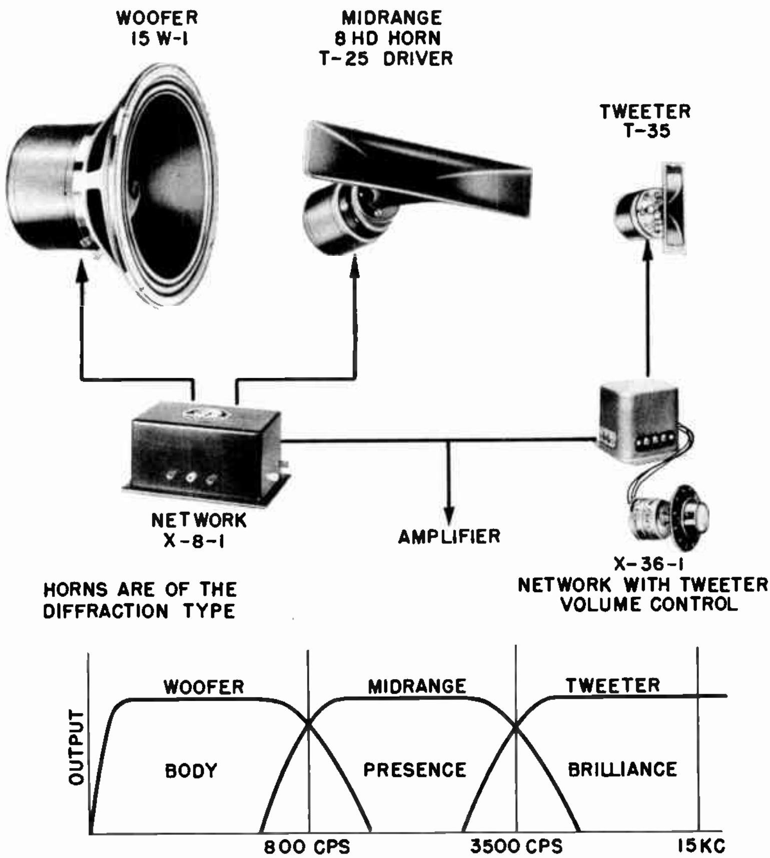


Fig. 6-11. A three-way multi-speaker system in which compatibility of components is accomplished by properly designed diffraction type horns, with the crossover points falling at 800 and 3500 cps. (Courtesy Electro-Voice)

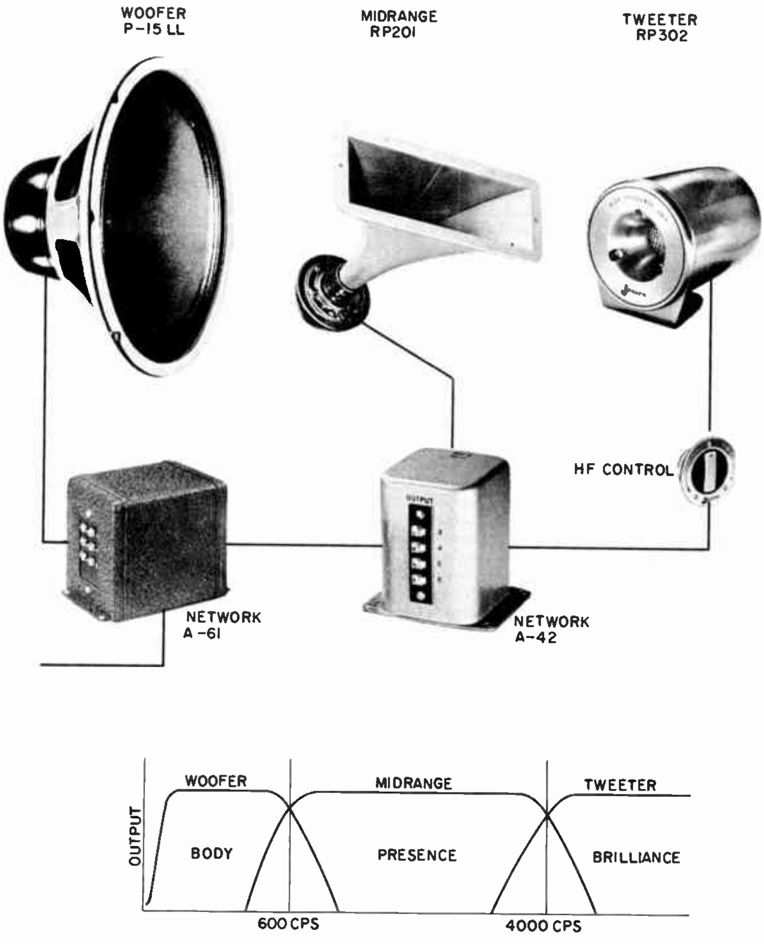


Fig. 6-12. A three-way multi-speaker system using components that are compatible for crossovers chosen by the manufacturer at 600 and 4000 cps. (Courtesy Jensen)

Adjustable Units Aid Hobbyist in Balancing System

If the hobbyist wishes to make his own selection of component parts, he must be guided by the manufacturer's specifications for frequency range, angular dispersion, and efficiency. Often the hi-fi listener has a tweeter with which he is throughly satisfied, but to which he would like to add a new and improved woofer. He must then be sure, for instance, that the cutoff frequency of the tweeter in the low frequency overlapping area falls below the point where the woofer begins to lose output. One way of meeting this objective is to choose the woofer so that it considerably overlaps the tweeter and then to interconnect them with a network that makes available a choice of crossover points within that overlapping area. This will ensure a system having smooth transition of response between the two components at the frequency chosen by the crossover networks. Then, as shown in Fig. 6-13, there are two specialized components for multi-speaker systems, an adjustable response woofer, and a variable-crossover adjustable high-pass filter. The woofer has built into it the low-pass elements of a crossover network that inserts attenuation into the woofer circuit at the different frequencies noted, so that it may be rolled off in output at the frequency most compatible with the tweeter to be used. Now if a tweeter is chosen whose cutoff is at least as low as the lowest tap of the adjustable woofer, and if a high-pass filter is used to connect the tweeter to the woofer, the listener has a wide adjustable overlapping area of both speakers from which he may pick at will the response he prefers. If he wants good forceful middles and strong horn tones, he will crossover at about 700 cps, and let the tweeter horn reproduce these middles and trebles. If he wants subdued presence and mellow middles, he will crossover at some higher frequency such as 5000 cps, and let the cone reproduce the lows and middles and the tweeter only the trebles. Moreover, along with the high-pass filter, the level control allows the listener to balance the spectrum in efficiency, while the taps balance it in crossover. Thus we see that, because of the multiplicity of combinations of specialized speakers for multi-speaker systems, it becomes possible for the hi-fi devotee to "play" the instrument, to make tonal changes, to "voice" in much the same manner that an organist plays

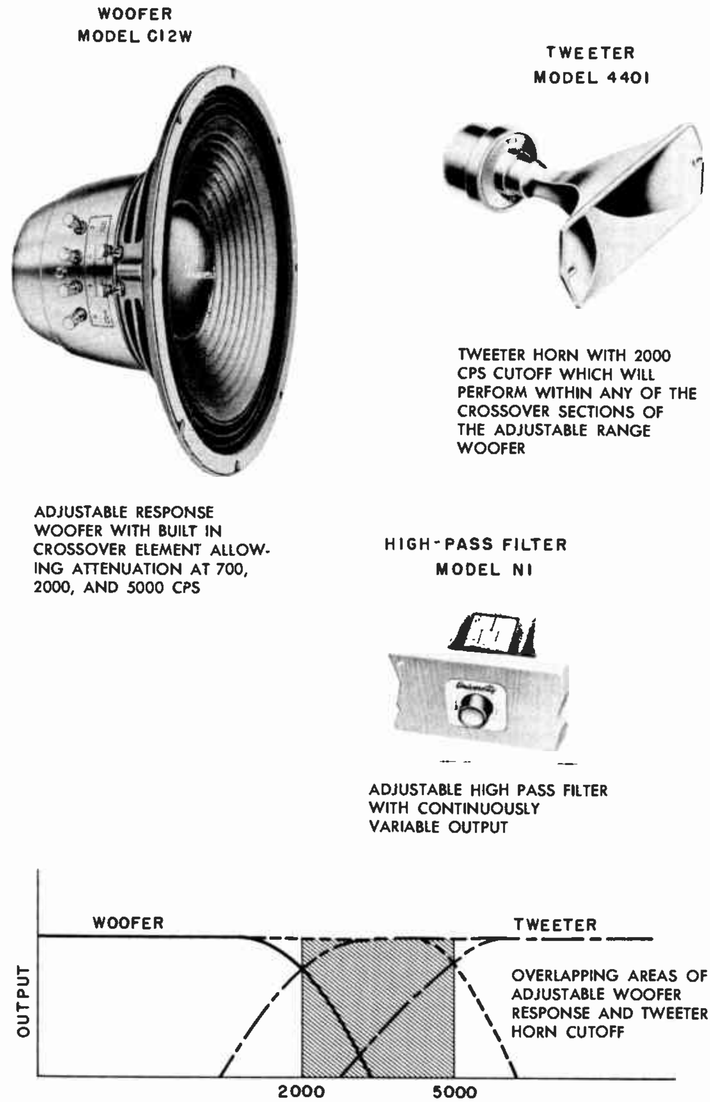


Fig. 6-13. The adjustable response woofer plus the high-pass filter enables the crossover between the woofer and tweeter to be chosen either at 2000 or at 5000 cps. (Courtesy University)

by setting the stops in the console and swelling the various voices. However, if the listener is of the type who would rather buy a ticket to the concert and let the experts do the playing for him, he will buy the manufacturer's complete integrated speaker system and sit back and enjoy his home "concert hall."

CHAPTER 7: *Networks in Multi-Speaker Systems*

Networks Separate Frequency Bands

In multi-speaker systems in which specialized speakers are used for different frequency bands it is necessary and desirable to segregate these different bands of frequencies into the respective speakers designed to handle them. As discussed in the previous chapter, this segregation of the various bands of acoustic energy insures optimum utilization of audio power, resulting in better overall performance of the system.

Networks are composed of elements which allow to pass, or pervent from passing, certain bands of frequencies. A capacitor will pass high frequencies and block low frequencies. Therefore, it is to be found in high-pass filter sections (Fig. 7-1). A choke or inductor will block high frequencies and pass low frequencies. It is therefore found in low-pass filter sections. If both a low-pass element and a high-pass element are in series in a single branch of a circuit, a band of frequencies is passed. The band has its low frequencies restricted by the high-pass element and its high frequencies restricted by its low-pass element. Those frequencies in between these two restricted ends become the "pass band." From these simple concepts we may derive almost all the popular networks.

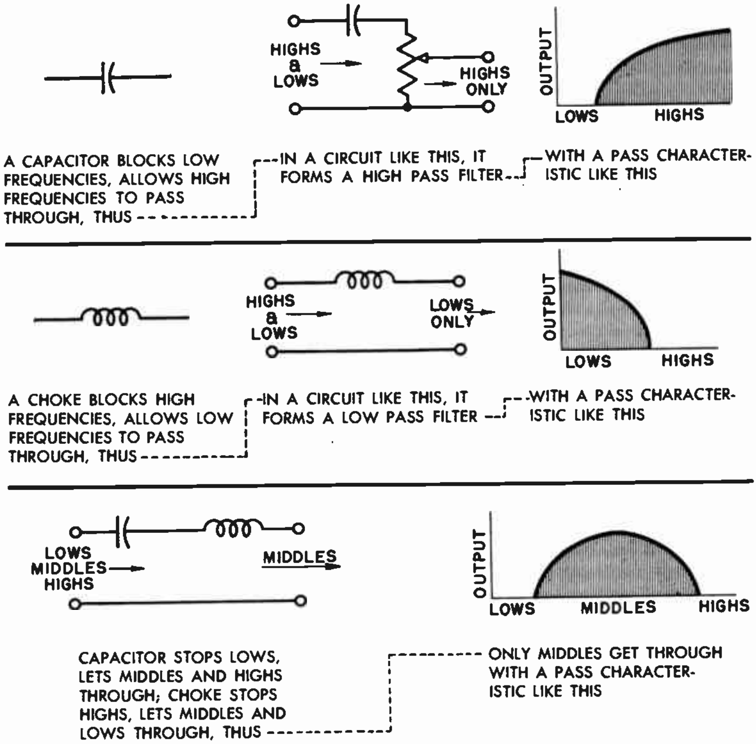


Fig. 7-1. Networks are composed of elements which allow to pass, or prevent from passing, certain bands of frequencies.

High-Pass Filter Keeps Lows out of Tweeter

The frequencies at which the filter elements become active depend upon their particular electrical values, and the effectiveness of the network as a whole depends upon the number of elements in the network. Applying the above concepts, let us lay out the various types of networks, and apply each to a particular situation. The simplest popular network is the high-pass filter. This type of network is employed where there is an existing speaker or speaker system, to which a tweeter is to be added. The high-pass filter is connected to the tweeter as shown in Fig. 7-2. The high-pass element blocks the low frequencies from getting into the tweeter, allowing only a certain range of high

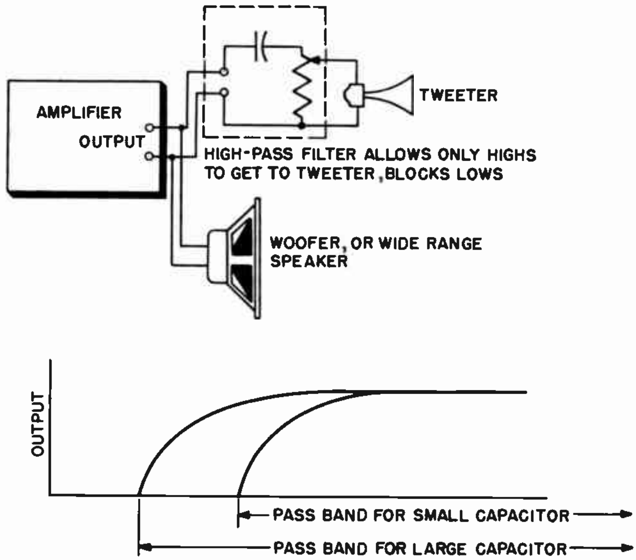


Fig. 7-2. Method of connecting high-pass filter to amplifier and tweeter.

frequencies to pass through, the lower limit depending on the size of the capacitor. A smaller capacitor will pass the highs, with some middles. A large capacitor will pass the highs with more middles.

Low-Pass Filter Keeps Highs out of Woofer

If we desire to keep the high frequencies out of the woofer (and pass them to the tweeter, where they will be more efficiently reproduced), we put an inductor into the woofer circuit, as shown in Fig. 7-3. Such an inductor (choke) will be a low-pass filter. Now the woofer will reproduce only a certain range of lows, as determined by the size of the choke. A large choke will pass fewer highs, a small choke will pass more highs. The choice of the choke will therefore determine the limit of the low pass. By the addition of the choke to the woofer circuit we have converted the simple high-pass arrangement shown in Fig. 7-2 to a two-way system with a crossover network, because now the high-pass band and the low-pass band cross over at some particular frequency.

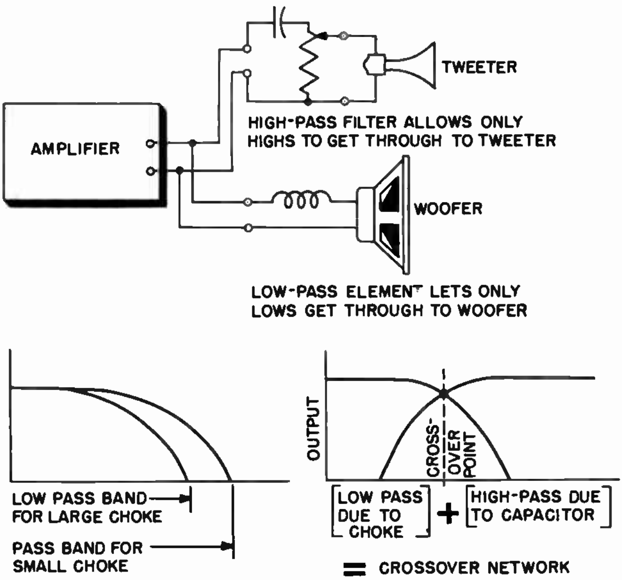


Fig. 7-3. Low-pass filter plus high-pass filter form a crossover network. (Network shown has a 6 db per octave attenuation.)

Obviously, the frequency at which they cross over will be determined by the electrical values of the filter elements involved.

Crossover Networks are Composed of High-Pass, Low-Pass, and Band-Pass Filters

The problem of constructing a three-way crossover network is only slightly more involved. Figure 7-4 illustrates a simple three-way network. As before, the response of the woofer is cut off in the high end by the low-pass choke. The middle frequency band has its lows limited by the high-pass capacitor, and its highs limited by the low-pass choke, thus allowing only the middle frequency band to be passed. The tweeter reproduces only the highs passed on to it by its capacitor. The characteristic bands of output of the three sections cross each other in succession so that we have in this three-way network a low crossover point between the woofer and midrange sections, and an

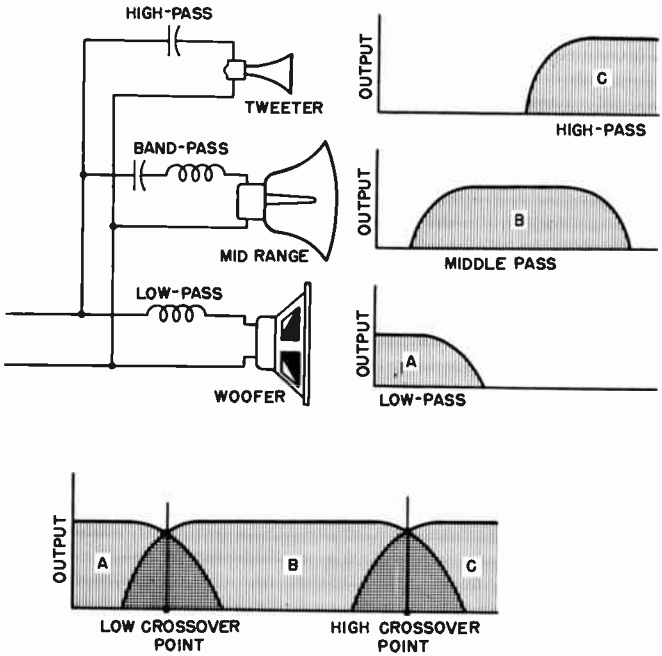


Fig. 7-4. A three-way crossover network is made by combining separate pass band elements that individually cover different sections of the band.

upper crossover frequency between the middle and high treble frequencies. In this case, as in the previous one, the crossover points will be determined by the electrical values of the filter crossover elements involved.

Number of Filter Elements in One Circuit Determines Attenuation Rate

Crossover networks have another characteristic in addition to their low-, high-, or pass-band features. They have an "attenuation rate" at the crossover point. The attenuation rate simply signifies how decisive the filter action is. Figure 7-5 shows the high-pass filter response with two different attenuation rates, a slow one and a fast one. A sharp cutoff filter may be made by combining two filter elements of opposite types (a high-pass element and a low-pass element) in the same branch circuit with one element in series with the speaker, the other in parallel with the speaker. In Fig. 7-6 is shown a sharp cutoff high-pass filter. The series capacitor will allow high frequencies to get into the tweeter, but the choke *across* the tweeter, being a low-pass element, will detour around the tweeter the lower frequency end

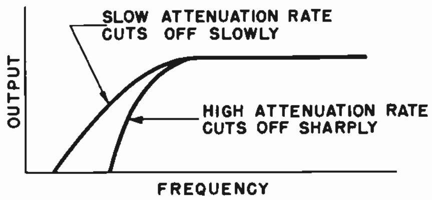


Fig. 7-5. High-pass characteristic with two different attenuation rates.

of the high frequency band passed on by the capacitor. The highs, however, will not be bypassed by the choke, so they will continue to go into the tweeter. Accordingly, by adding a low-pass element across the tweeter, we have effectively sharpened the high frequency pass band that the tweeter sees. We have increased the attenuation rate of the network. In a similar fashion the attenuation rate of a low-pass filter

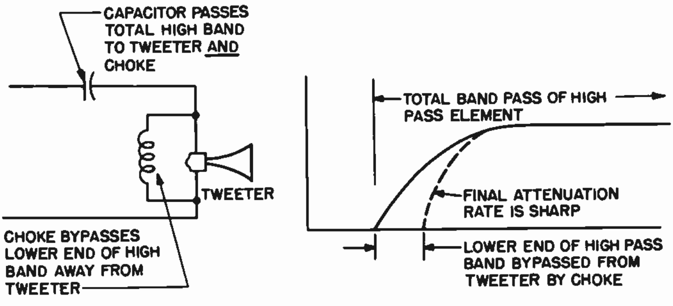


Fig. 7-6. A sharp cutoff high-pass filter is made by adding a low-pass element across the tweeter circuit.

may be sharpened by shunting the load with a high-pass element as shown in Fig. 7-7. By combining these sharper cutoff high-pass and low-pass elements into the previously discussed filter circuits we obtain sharp cutoff two-way networks, as shown in Fig. 7-8, or three-way networks, as shown in Fig. 7-9.

Sharpness, or attenuation, of a crossover network is usually given in terms of db's per octave in the crossover region. The term *db* is an abbreviation for the word *decibel*. The decibel scale provides a convenient way of compressing large changes into small changes so that they may be easily handled and so that they bear a closer relationship to the manner in which the ear responds to the intensities of sounds. Precisely, it is a "logarithmic ratio." (See Fig. 7-10.) Every time power is doubled, we say there is a 3-db gain. If the power has *doubled* itself 10 times, then there has been a 30-db increase. If the voltage has

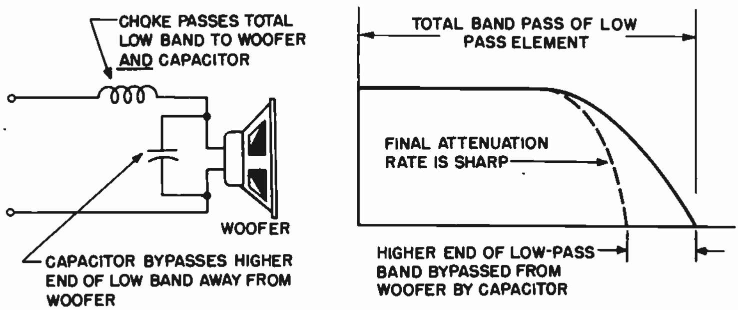


Fig. 7-7. A sharp cutoff low-pass filter is made by shunting a high-pass element across the woofer circuit.

doubled itself, we say there is a 6-db gain. Thus, if the voltage has *doubled* itself 10 times, there has been a 60-db gain. Reference to Fig. 7-10 will show how conservative these db values are in relation to the actual growth of the power or voltage figures.

Thus, getting back to the network, if the characteristic of the circuit is such that the voltage across the speaker branch of the network drops to one-half that of the total voltage input across the network in a space of one octave, causing a power drop to one-quarter of the original power, the attenuation rate is 6 db per octave. If the voltage drops to one-quarter in the same octave span, causing a power drop to one-sixteenth of the original power, there is a 12 db per octave roll-off. (Refer to Fig. 7-11.) Obviously, the 12 db per octave roll-off is the sharper, and the 6 db the slower. The former is produced by the type of network shown in Fig. 7-8, and the latter by the network in Fig. 7-3.

The actual design conditions for a particular network are governed by the choice of crossover frequencies and the attenuation rate desired.

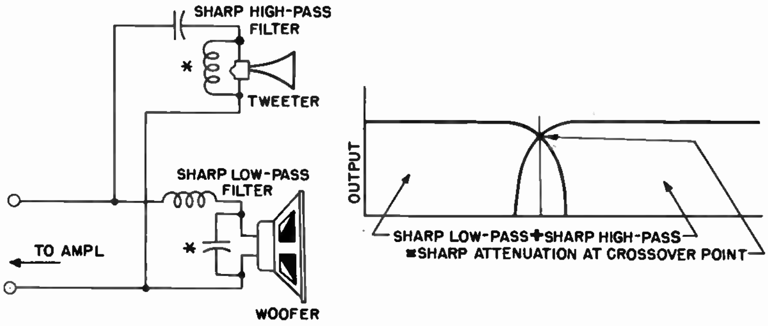


Fig. 7-8. Sharp cutoff filter sections as shown combine to produce a sharp attenuation rate (12 db per octave) crossover network. Asterisks show added bypass elements that convert network from 6-db to 12-db type.

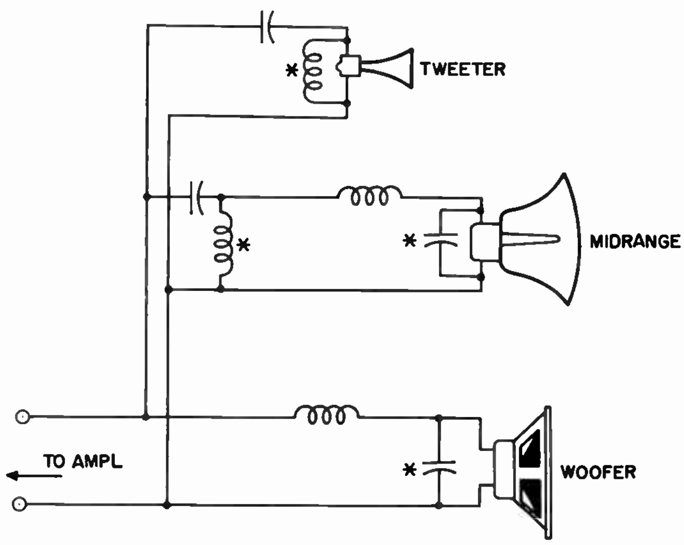


Fig. 7-9. A three-way network arranged for 12 db per octave attenuation at the crossover points. Asterisks indicate the added bypass elements that convert the network from 6-db to 12-db type.

POWER	DB GAIN	VOLTS
.1 WATTS	REFERENCE 0	.1 VOLTS
.2	3	
.4	6	.2
.8	9	
1.6	12	.4
3.2	15	
6.4	18	.8
12.8	21	
25.6	24	1.6
51.2	27	
102.4	30	3.2
	33	
	36	6.4
	39	
	42	12.8
	45	
	48	25.6
	51	
	54	51.2
	57	
	60	102.4

A POWER RATIO OF 2 TO 1 REPRESENTS A 3 DB CHANGE

A VOLTAGE RATIO OF 2 TO 1 REPRESENTS A 6 DB CHANGE

A POWER INCREASE OF 1000 TO 1 IS EQUIVALENT TO A 30 DB GAIN

A VOLTAGE INCREASE OF 1000 TO 1 IS EQUIVALENT TO A 60 DB GAIN

Fig. 7-10. The decibel scale is a means of compressing large changes in physical quantities into small numberical changes.

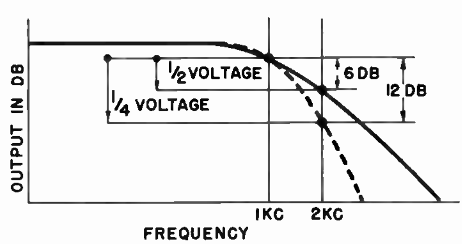


Fig. 7-11. When voltage drops to ½ the value of the voltage at the crossover frequency, the attenuation is 6 db; if it drops to ¼, the attenuation is 12 db. If this drop takes place in a span of one octave, there is either a 6 or a 12 db per octave attenuation.

These factors, in turn, depend upon more subtle considerations concerning the type of speakers that make up the system, and what is actually desired from the system. Therefore, we shall now give full consideration to those factors which govern the choice of network, and then we shall present some practical design data.

Networks Conserve Audio Power

In breaking up the total reproducible spectrum into various bands and then directing these bands into separate specialized speakers, we may realize a great degree of conservation of audio power. Let us refer back to the simplest system of Fig. 7-2, where the tweeter is connected across the audio signal line through a simple high-pass filter.

Let us assume, however, for the sake of illustration, that the capacitor in this high-pass filter has been eliminated. This condition of direct parallel operation of the woofer and the tweeter means that the amplifier power now will be *equally* divided between woofer and tweeter (if they have the same impedance). If a total of 10 watts of audio power is available from the amplifier, the woofer will take 5 watts of *full frequency range* power and the tweeter will take 5 watts of the same *full frequency range power.* Obviously, the woofer, which is a low frequency device, will not be able to do justice to the high frequencies; nor will the tweeter, which is a high frequency device, be able to do justice to the low frequencies. Since the total power is equally divided between the two units, half the high frequency power is *lost* in the woofer, and half the low frequency power is lost in the tweeter. What is more, the omission of the high-pass filter may result in overloading the tweeter if this loudspeaker is not able to handle the additional power that would now be applied to it.

Now let us put in the high-pass filter. Immediately, the low frequency power formerly lost in (or possibly overloading) the tweeter has been stopped from going into this blind alley. Instead, the low frequency power blocked from entering the tweeter must look for another route. It must now join the rest of the low frequency energy going directly to the woofer. Essentially, then, the woofer now gets twice the audio power it was receiving before the insertion of the high-pass filter in the tweeter circuit. As a result, the woofer now plays louder for the same power output from the amplifier. If, however, one wished to maintain the previous listening level, the amplifier gain control must be turned down, resulting in more conservative operation of the amplifier and consequent reduction of distortion.

We may go one step further, and prevent the highs from being lost in the woofer, by inserting the low-pass choke into the woofer circuit, as in Fig. 7-3. This blocks the highs from the woofer but lets the lows through. Accordingly, we now realize a saving in high frequency power, for the highs, blocked from entering the woofer, naturally take the path of least resistance and go to the tweeter. The tweeter now receives twice as much treble energy for the same setting of the amplifier. Thus we see that a well-designed crossover network, usually hidden away somewhere where it cannot be seen, in its own unobtrusive way goes far toward increasing the available power from the amplifier, by simply routing the energy where it will do the most good.

Networks Reduce Intermodulation Distortion

Of equal importance to the matter of power utilization, however, is the fact that a well-designed crossover network will improve the cleanness of the audio reproduction by minimizing intermodulation distortion. Let us go back to the previous example and briefly examine the tweeter operation with and without the low frequency isolating element. With the isolating capacitor left out of the tweeter circuit, electrical currents of both low and high frequencies are sent through the tweeter voice coil. Even though the tweeter diaphragm may have a high resonant frequency, it still has a tendency to vibrate below its resonant point, especially if these low frequencies are large in magnitude. Thus, the small tweeter diaphragm tends to bounce back and forth in its magnetic circuit under the strong low frequency impulses. However, because the diaphragm is small and is coupled to the horn (which will not transmit low frequencies) it will accomplish no useful acoustic purpose. On the contrary, it deteriorates the high frequency performance of the tweeter by causing the small high frequency vibrations of the diaphragm to be carried along on these spurious and excessive low frequency vibrations of the diaphragm. Hence, the irregular and nonlinear low frequency vibrations of the diaphragm modulate the high frequencies, causing intermodulation distortion. However, if the high-pass element is now inserted in the tweeter circuit, the low frequencies do not enter the tweeter; the diaphragm moves only under the insistence of the high frequencies and they radiate cleanly, and free of low frequency effects. Thus, in addition to their audio power saving characteristics, crossover network serve to improve the overall cleanness of reproduction of the segregated bands of frequencies.

Crossover Characteristics Determined by Network Plus Speakers

The success with which a crossover network accomplishes its purpose is dependent not only upon the network design itself, but also upon the speaker components with which it must operate. Those factors which determine how well the network operates are the crossover points, the attenuation rates at the crossover points, and the compatibility of these two factors with the corresponding ones for the speakers of the system.

A 6-db attenuation rate crossover network will separate the frequencies at the crossing bands at a moderate rate but with considerable overlapping areas between the two different bands. On the other hand, the 12-db attenuation rate network will separate the channels more sharply and with less overlapping area between the two. One system is not necessarily better than the other. What determines the choice of system, as far as crossover points and attenuation rate are concerned, is the natural characteristics of the speakers that constitute the system. All speakers have their own natural high frequency roll-off characteristics and low frequency cutoff characteristics. These speaker characteristics have to be integrated into the network system for compatibility of their attenuation and roll-off characteristics.

The choice of a crossover frequency is not determined by any hard and fast rule. The fact of the matter is that, for a given set of speakers with wide overlapping areas, there may be a wide choice of crossover frequencies. The example given at the end of the previous chapter (Fig. 6-13), where there was an adjustable crossover high-pass filter with a tweeter capable of covering a wide band, was a case in point; the actual crossover point could be chosen on a listening basis rather than on a purely electrical basis. Of course, once the listening objective has been established, the electrical crossover characteristics may fall right into place if the networks are of the adjustable type. If we let the matter of listening preference go for the moment, however, we may examine the more rigorous acoustic basis upon which the crossover point may be determined.

At Crossover Point, Speakers Should Have Equal Output

The first criterion of choosing the crossover point of a network is that at the crossover frequency chosen, the speakers have equal acoustic output. This simply means that the speakers must have a reasonable area of overlapping response within which the network may be applied without causing holes in response. Thus at the crossover point the speakers will both have equal acoustic output, but will then be attenuated on either side of the crossover point as dictated by the network design. The sharpness of the attenuation rate of the network, plus the natural attenuation of the speakers themselves, will naturally determine the sharpness of separation of the various acoustic ranges of operation of the speakers. This acoustic separation raises the question of the *auditory* sharpness of the separation of the individual speakers of the system as far as the listener is concerned. The present acceptance of stereophonic sound in professional entertainment houses is a clear indication that this new dimension is an entirely desirable one for modern sound reproduction. Any approach to such panoramic sound in the home may be valued by the listener who wants some measure of concert hall separation of instruments in his reproducing system.

While it is true that separation of the music in sharp frequency bands is not the same as separating the music of one instrument from the other, there is some feeling, synthetic though it may be, that at least some of the instruments have been spatially stretched out into a greater listening area, giving a substantially greater acoustic perspective to their reproduction. For people who prefer this separation, sharp separation of the speaker ranges is desired and the network should have a steep attenuation rate to provide this sharp degree of separation. On the other hand, where the listener prefers more blended sound, more unity to the overall reproduction, he will require that the speakers of the system do not separate sharply but rather that they overlap to some degree. Accordingly, the network characteristic for this application will be one of moderate attenuation rate.

Attenuation Rate Determines Sharpness of Acoustic Separation Between Speakers

In general, then, the problem of slow attenuation rate versus fast attenuation rate in networks is similar to the problem of the in-

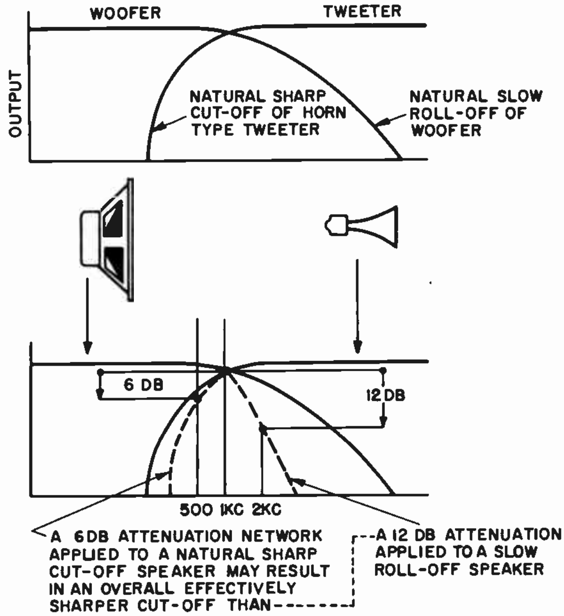


Fig. 7-12. The sharpness of the crossover characteristic is determined both by the network and by the speakers of the system.

dividual multi-speaker system types compared to the integrated packaged units. The listener who wants to be "on stage" and hear close-ups will choose the multi-speaker system made up of individual units, for then he will have clean separation of instruments. He will also choose steep attenuation rates for his networks. The listener who wants to sit way back in the family circle and hear overall sound pictures will choose the integrated coaxial and triaxial type speakers and slow attenuation rates for the network.

As previously stated, the system crossover characteristic attenuation rate is a *combined function of the network attenuation rate plus the attenuation rate of the speakers themselves.* This interdependence of the network and speaker attenuation characteristics is, however, only partly related to the problem of overlapping areas. Thus, as is shown in Fig. 7-12, two speakers may have the same overlapping area of equal level, but one may have a much steeper cutoff characteristic than the other. It is therefore probable that a fast cutoff speaker with a 6 db per octave network would provide a sharper crossover characteristic than a slow cutoff speaker with a 12 db per octave network. A horn type reproducer, such as a compression tweeter, is essentially a sharp cutoff radiator with virtually no usable output below its theoretical cutoff frequency. On the other hand, a cone reproducer, such as a woofer, may have useful energy output extending into the middle high frequencies. It is quite apparent, then, that even without the benefit of the crossover networks, the horn to some extent provides its own crossover networks, while the cone remains relatively unaltered. Therefore, if the tweeter horn system is now integrated with a 6 db per octave crossover network, the combined attenuation will be greater than the cone type system operating with a 12 db per octave network. Accordingly, the attenuation rate of the network must be integrated with the speaker characteristic not only for equal ouput level at the crossover point, but also for compatible attenuation rates of the network and speaker combination.

If relatively long overlapping areas of performance between adjacent speakers are desired, the combined network-speaker attenuation characteristic will be chosen on a slow basis. If sharp distinct separation of adjacent speaker performance is desired, then the combined network-speaker attenuation characteristic will be comparatively steep.

Speaker Enclosures Are Acoustic Crossover Elements

It will be realized that baffle arrangements are actually acoustic crossover elements, as is, for instance, the sharp cutoff horn of the tweeter. Actually, *all* baffles are acoustic crossover elements, and as such will be treated more specifically in the section on baffles and enclosures. For the present, however, it is important to keep in mind the interdependence of the baffle as an acoustic crossover element, the natural roll-off or cutoff of the speakers themselves, along with the network characteristics, determining the sharpness of the crossover system as a whole.

As for the crossover network itself, sharp network attenuation specifically means sharper separation of the chosen bands of frequencies. Accordingly, there will be less interaction of the lower band of frequencies upon the speaker operating on the upper adjacent band (as illustrated in Fig. 7-13). With less interaction between the bands, there will be cleaner reproduction of the upper frequencies, because the disturbingly large lower frequencies have been eliminated from the treble speaker.

Networks Affect Transient Response

There is another type of distortion, called transient distortion, which is a measure of the ability or lack of ability of the loudspeaker to faithfully follow the suddenly applied impulse type of tones, and to let go of these tones as dictated by the signal itself. The ability of

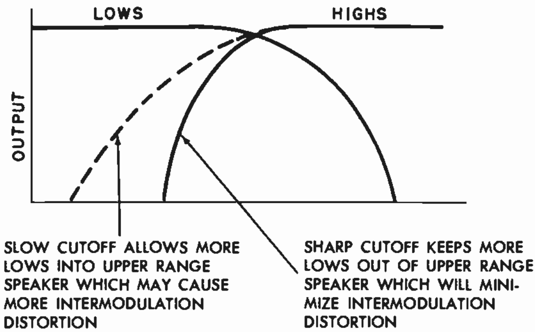


Fig. 7-13. The sharpness of the attenuation rate of the system will determine the degree of intermodulation distortion which may occur if the speakers are not perfectly linear.

the loudspeaker to function in this manner was treated in the previous chapter, where it was pointed out that a heavy magnetic gap flux would tend to minimize such transient distortion. In the present considerations, however, we are going to analyze the part that the crossover network plays in introducing transient distortion into the reproducing system. Transient distortion in networks is introduced at the crossover point region due to too sharp an attenuation rate. As we have seen, a sharp attenuation rate is obtained by the use of several sections of filter elements in a speaker circuit, whereas low attenuation rate is the result of a minimum of filter elements. Where a multiplicity of such elements exists, two effects take place.

First there may be phase shift of the various component frequencies that make up a complex sound. In any electrical network, phase shift of the signal always occurs as the signal passes through a single filter component. In general, the more components the more drastic the phase shift. If these frequencies thus become electrically too far displaced from their original relationships, they may not recombine acoustically into the same pattern. As a result, the final tone may not have the same overall sharp form as the original; transient distortion will thereby have been introduced.

The second effect of steep cutoff networks is their tendency to cause "ringing." Complex networks are made up of components that form resonant circuits. Being resonant they may be "shock excited" at certain frequencies, causing a momentary self-oscillation or ringing for some short, but appreciable, time after the original signal has been removed. On the other hand, the more simple networks, although not producing as clean a demarcation between the various bands, are less subject to these types of distortion. For this reason, present practice is to avoid networks that have an attenuation rate in excess of 12 db per octave. For modern loudspeaker practice, 6 db and 12 db per octave attenuation rates provide adequate control, and offer ample choice to match the particular characteristics desired for most reproducing systems. Before we take up the matter of the actual construction of networks by the hobbist, it will be of benefit to examine commercially available networks designed for multi-speaker systems.

Packaged Networks for Packaged Systems

There are available commercial networks of many designs and for many applications. Many manufacturers recommend specific networks for use in conjunction with their engineered and matched combinations of speakers for a multi-speaker system. Such recommended "system" networks are shown in Figs. 6-10, 6-11, and 6-12. Of great value in any network application to multi-speaker systems is the incorporation of controls for adjusting the presence and brilliance of the system. As will be recalled from Chap. 6, the "presence" of a system resides in the middle frequencies, and "brilliance" in the upper treble region. For adjustment of these factors, controls are inserted into the electrical output of the network that will ultimately feed the midrange and the tweeter reproducer. Thus, where these controls are readily available to the listener, there is a sort of "acoustical baton," which permits the listener to balance his "home orchestra" to best suit his tastes.

Adjustable Networks Permit Hobbyist to "Tune" System to his Preference

For the hi-fi enthusiast who chooses his own components, however, there is a choice of crossover networks or network components to enable him to get the most out of his system, or to enable him to experiment with his system at will. Kits of network elements, such

SOME POSSIBLE COMBINATIONS

IMPEDANCE	CROSS-OVER FREQUENCY
16	2200 1100 700 350 175
8	1100 550 350 175 85
4	550 275 175 85

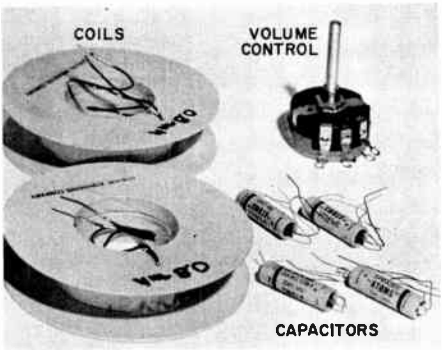


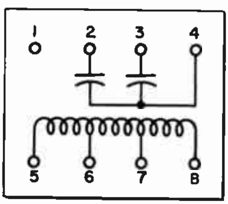
Fig. 7-14. Network kit that may be obtained for constructing network. Individual coils or kits are available for the above tabulated requirements. (Courtesy General Apparatus)

as that shown in Fig. 7-14, are made available for the constructor who knows exactly what his requirements are for impedance of the system speakers, crossover points, and attenuation rates. The assembly of these components is fairly straightforward.

There is also available a very convenient already built multi-purpose network design that offers the hi-fi constructor a wide choice of network parameters with which to work. This network package is shown in Fig. 7-15. It consists of two independently complete but also complimentary network systems by means of which it is possible to select any of a number of crossover points at any of the common speaker impedances, plus a choice between 6 db or 12 db per octave attenuations. By selecting the proper designated terminals on the terminal board of the network, he may readily make his choice of the various conditions of operation. The short table accompanying the photograph

MODEL N2A
(LOW BAND)

MODEL N2B
(HIGH BAND)



TYPICAL INTERNAL WIRING AND COMPONENTS OF MODEL N2A NETWORK

	FUNCTION	OHMS	FREQUENCY
N2A	12 DB HI, LO PASS FILTER	8	350, 700, 1400
		16	350, 700
		4	700
	2 WAY 6 DB CROSSOVER	8	350, 700
		16	350, 700
		4	700
N2B	12 DB HI, LO PASS FILTER	8	1250, 2500, 5000
		16	2500, 5000
		4	2500
	2 WAY 6 DB CROSSOVER	8	1250, 2500, 5000
		16	2500, 5000
		4	2500
N2A N2A	2 WAY 12 DB CROSSOVER	8	350, 700, 1400
		16	350, 700
		4	700
N2B N2B	2 WAY 12 DB CROSSOVER	8	1250, 2500, 5000
		16	2500, 5000
		4	2500
N2A N2B	3 WAY 6 DB CROSSOVER	8	350/1250, 2500, 5000
			700/2500, 5000
		16	350/2500, 5000
			700/2500, 5000
		4	700/2500
N2A-N2A N2B-N2B	3 WAY 12 DB CROSSOVER	8	350/1250, 2500, 5000
			700/2500, 5000
		16	350/2500, 5000
			700/2500, 5000
		4	700/2500

Fig. 7-15. Crossover units which permit the wide selection of crossovers, attenuation rates, and impedances shown from the two single models. (Courtesy University)

of these multi-purpose networks shows the versatility that has been built into the networks to accommodate any selection of speakers.

Adjustable Networks Eliminate Obsolescence of Components

However, there is a more important asset in such diversified networks, and that is the elimination of the obsolescence factor. If the constructor has begun a progressive speaker expansion program, a single network may meet his immediate need. As his system grows, he may simply add the complementary network to achieve a three-way system, or to realize steeper crossover characteristics at the crossover points if he desires it. Let us assume that one's original system was a 15-inch woofer in a bass-reflex enclosure, with a tweeter crossing over at 750 cps, this crossover being accomplished by the type N2A network shown. After a couple of years of indoctrination into the hi-fi family, the listener may decide to expand his two-way system to a three-way system, using the same woofer and tweeter, but adding a cone speaker as a midrange reproducer. (See Fig. 7-16.) He may now use the N2A as the low crossover network and drop the crossover point from 700 to 350 cps by simply connecting the proper terminals on the network. Then he may add the type N2B network to provide the upper crossover between the midrange cone speaker and the tweeter. Since the tweeter originally went down to 700 cps, and since the cone speaker will probably go well up into the treble region, there will be a large overlapping area, in which the N2B network may be used to find the crossover point best suited to the system. Universal networks of this type are a sort of master key to multi-speaker system design, allowing for obsolescence-free system expansion planning and for a choice of personal operating conditions.

The Home-Constructed Network

There are, however, many hi-fi hobbyists who find pleasure in building as much of their own equipment as they possibly can. For these, we will now offer a simple but concise method of building a workable crossover network.

The design of a crossover network is based on three factors: (a) speaker impedance; (b) crossover frequency; and (c) attenuation at the crossover point. These factors determine the electrical size of the capacitors and inductors that make up the various branch circuits. The chart in Fig. 7-17 provides a simple means of selecting the proper value of components for a chosen crossover frequency between any two speakers whose frequency ranges overlap. The chart also presents values for either a 6 db or a 12 db per octave attenuation rate. It is not limited in any way to the number of branch circuits that may be used in the system. It may be applied to a two-, three-, four-, or even five-way system, if desired. This versatility is accomplished by taking the speakers one by one, starting from the lowest frequency range, and selecting the proper constants to cross that speaker with the next adjacent speaker in the frequency range. Then, in a like manner, the constants are chosen to cross this second speaker with the third speaker. An example will illustrate its application.

Sample Network Design

Let us design a network for a three-way system, such as that shown in Fig. 7-4, with crossovers at 350 and at 5000 cps, for speakers of 8-ohm impedance, and to have a 6 db per octave roll-off. (See Fig. 7-17.) We are first concerned with the woofer and the speaker adjacent to it, the midrange speaker. The crossover frequency is to be 350 cps. Locate 350 on the frequency scale. Extend a horizontal line to the right until it intersects the proper *C/L* line, which gives the electrical constants necessary for the low frequency speaker circuit — 3.7 millihenries for the coil and 58 microfarads for the capacitor in

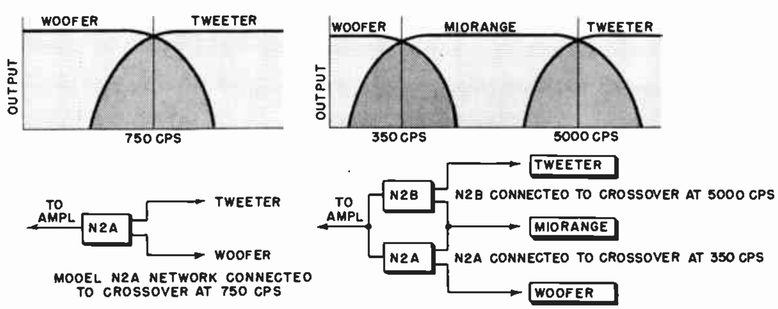


Fig. 7-16. The multiple crossover frequencies available from the commercially available networks permit system expansions without obsolecence of equipment.

the lead to the midrange speaker. Incidentally, if we were concerned only with a two-way system (as shown in Fig. 7-3) to cross over at 350 cps, these two values would give us the complete design of the network. However, since we are building a three-way network, we must now deal with the second crossover point.

To select the constants for this upper crossover frequency of 5000 cps, we regard the midrange speaker as the lower speaker. Entering the chart from the 5000-cps mark, we again proceed to the 8-ohm *C/L*

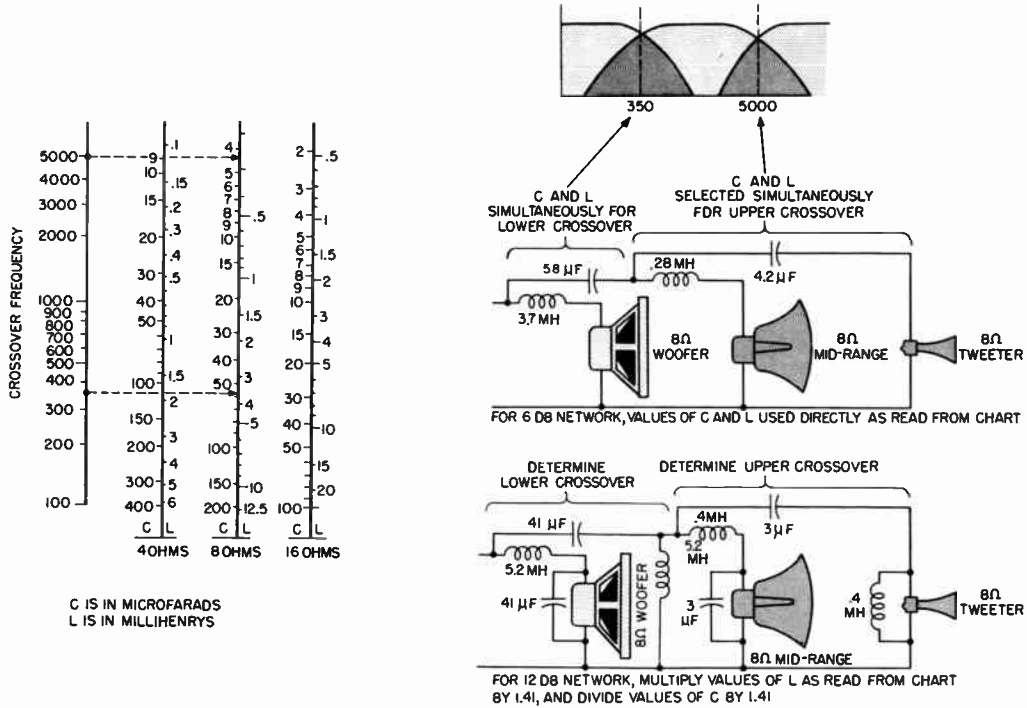


Fig. 7-17. Universal chart for finding component values for crossover networks with either 6-db or 12-db attenuation, for systems of 4, 8, or 16 ohms impedance.

line and find the value of the choke (.28 millihenry) to be put into the lower speaker circuit, and the value of the capacitor (4.2 microfarads) to be put into the upper speaker circuit. The network design for a system to cross over at 350 cps and 5000 cps with 6-db attenuation is now complete.

To design a 12 db per octave network for a three-way system we proceed exactly as above to select the necessary capacitor and coils, but this time we apply the factor of 1.41 as specified by the chart. The value of L is multiplied by 1.41 and the value of C is divided by 1.41. Thus we have obtained all the necessary electrical values for the making of the crossover network.

Choosing the Type of Capacitor

The capacitors for the network must be bought; they cannot be made by the home constructor. The smaller capacitors, those of 4 µf and under, may be of the paper variety. However, they should be a good grade, or they may become leaky (low in leakage resistance across their terminals.) This will result in poor network response. Preferably, these capacitors should be of the oil-filled paper type, usually obtainable in the bathtub variety. These are self-healing in case of internal rupture and do not age or deteriorate. The alternating voltage rating of the capacitors need not exceed 50 volts if the network is designed for the common speaker impedances of 16 ohms and under. Where larger values of capacitance are required, the cost of the unit may be rather high and its size large if the oil-filled type is specified. Fortunately, it is possible to obtain good performance and high capacitance by using "non-polarized" electrolytic capacitors.

Normally, an electrolytic capacitor is marked "minus" and "plus" on its terminals, and this is how it is intended to be connected when used as a filter in d-c power circuits. This is the polarized type. The non-polarized type is actually two of these polarized capacitors in one case, internally connected in a series circuit ("back to back") with their similarly polarized terminals connected together. This results in a capacitor that may be put across a line carrying audio voltage. Such non-polarized electrolytic capacitors, rated in very large capacitances, are readily obtainable from radio supply houses at reasonable cost.

If the constructor wishes, he may make his own non-polarized electrolytic combination by simply connecting two equal electrolytic filter capacitors back to back. In making this combination it must be remembered that the effective capacitance of two capacitors in series is equal to only one-half that of a single unit. Thus, to obtain an 8 μf non-polarized capacitor, two 16 μf polarized capacitors may be connected in series as shown in Fig. 7-18.

However, there is a precaution that should be exercised when a small electrolytic capacitor is used for a particular crossover point and this crossover element is to function out to the very end of the tweeter

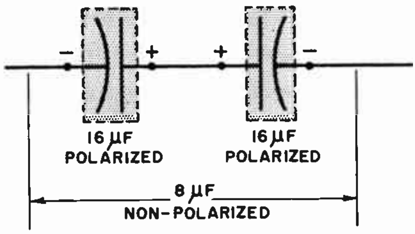


Fig. 7-18. Polarized (electrolytic) capacitors may be connected back-to-back to form a non-polarized capacitor. It should be noted that the total capacitance of the combination is equal to one-half of a single unit.

spectrum. In this case one-half the total capacitance in the circuit should be provided by the electrolytic type and one-half by the paper type, where the total capacitance of these two is equal to the value required to obtain the desired crossover frequency. The paper capacitor is wired in parallel with the electrolytic. The reason for this precaution is that the electrolytic capacitor becomes less effective as a by-pass element at the higher frequencies. This is a result of its higher power factor and increased impedance (it may actually become inductive) at these higher frequencies.

Making the Coils

The inductor may be made by winding onto the bobbin shown in Fig. 7-19 the number of turns required for the specific inductance. The bobbin may be made of wood or any other non-magnetic form rigid enough to support the necessary turns of wire. It is not necessary that the wire be accurately layer-wound. The coil may be random-wound, provided the winding space is evenly filled up.

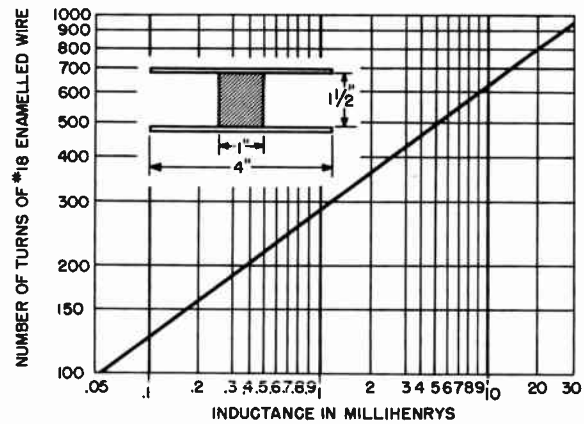


Fig. 7-19. Chart for winding inductors for crossover components as read from universal nomograph of Fig. 7-17.

Volume Controls may be Inserted in Speaker Lines

Volume controls may be inserted between the terminal of the network and the speakers for the purpose of controlling the presence and brilliance of the system. (See Fig. 7-20.) These controls may be either the simple potentiometer type, or the more elaborate constant-impedance pad types. The potentiometer has the advantages of simplicity in installation and of economy. The value of the potentiometer should be at least five time the value of the impedance of the speaker to which it is to be connected. Thus, if the speaker is 8 ohms, a 50-ohm potentiometer will be suitable. *L*- or *T*-pads may also be used as volume controls. They have the advantage of maintaining more constant impedance on the amplifier as the level of operation is changed. They are considerably more expensive than the potentiometer type of volume control and somewhat more involved to install. However, concerning the matter of mismatching of impedance when simple potentiometers are used, we should not worry too much about it in connection with crossover circuits. There are several factors affecting crossover network operation that are not too widely appreciated and are far more important than impedance matching.

Take first the case of the simple potentiometer volume control. When the potentiometer is turned on for full volume, the network is essentially looking into the rated impedance of the speaker, and the volume control, which is over five times the value of the speaker impedance, then becomes only a negligible shunt. In this case, there is no impedance matching problem. Now, when we turn down the volume control, it is true that the impedance matching between the network and its load has been upset, but this is now relatively unimportant. If

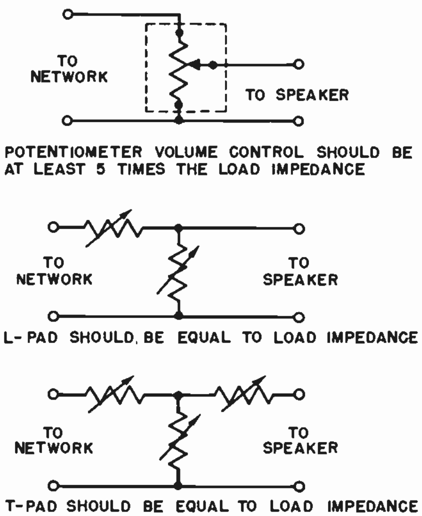


Fig. 7-20. Means of controlling the volume of loudspeakers.

we are operating the tweeter at a low level, we are obviously not seeking to get the most highs out of the system. It becomes unnecessary to match to the theoretically correct impedance.

Crossover Frequency of Network Affected by Variable Speaker Impedance

A word of advice is in order in connection with the values of the components chosen. Although the desired values are selected on a very rigorous basis of crossover frequency and speaker impedance, they may be varied considerably without upsetting the overall performance of the system. Speakers are not constant in impedance. (Refer to Chap. 5) A speaker rated at 8 ohms may have that impedance only in a small range of frequencies at about 400 cps; at 5000 cps its impedance may actually be 15 ohms. Therefore, if a crossover network is being designed for this speaker to crossover at 5000 cps, where its impedance may actually be twice the rated value, the values of the crossover elements may be 100 percent in error. It is therefore apparent that even the best networks may at time be severely compromised in performance by the impedance characteristics of the speakers themselves. Moderate variance in the values of the filter elements themselves will not upset the crossover frequencies nearly as much as will the varying impedance of the speaker.

Therefore, in choosing capacitors, and in winding coils, the constructor may be up to 20 percent away from the exact values and still realize substantially good network performance.

Crossover Frequency Affected by Acoustic Efficiency of Speakers

Of equal significance in the matter of network performance is the accuracy of the crossover characteristic in connection with the problems of level adjustment and compatible speaker combinations. If the speakers chosen for a multi-speaker system are widely different in efficiency, their actual *acoustical crossover points* may be considerably different from the *electrical crossover points* of the networks feeding them. Properly designed networks will deliver equal voltage outputs for the various bands, as shown in Fig. 7-4. If these equal voltages are then fed to loudspeakers of equal sensitivity, their acoustical crossovers will follow the general pattern of the electrical network crossover characteristic. However, if one of the speakers, say the tweeter, is only 50 percent as efficient as the midrange speaker, the actual acoustical output of the tweeter will be reduced by 3 db in relation to the midrange speaker, and the *acoustical* crossover will have been moved up one whole octave (as illustrated in Fig. 7-21), if the network has a 6 db per octave attenuation. If the network is more sharply attenuated at the crossover point, the shift in crossover frequencies will be less severe. In brief, for speakers of equal efficiency the acoustical crossover characteristic is similar to the electrical crossover characteristic. Where speakers are of unequal efficiency, the acoustic crossover may shift considerably from the electrical crossover point, and this shift may be minimized if desired by the use of sharper rates of attenuation of the crossover circuit.

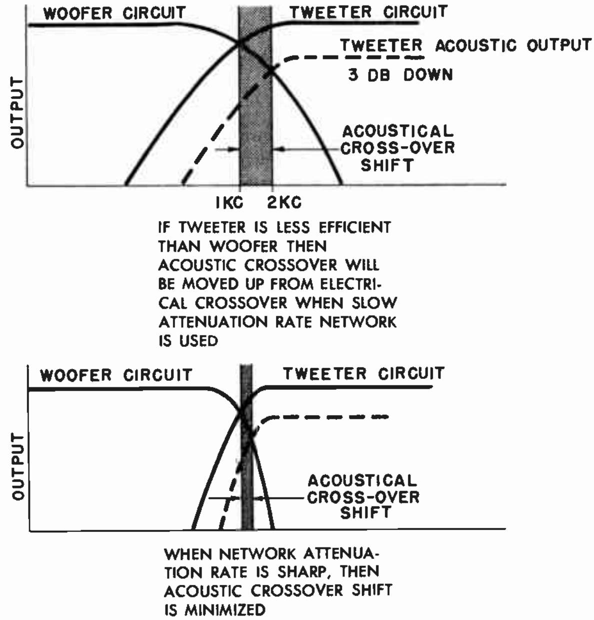


Fig. 7-21. Where speakers are not of equal efficiency, the actual acoustic crossover point may be considerably different from the applied electrical crossover.

Crossover Frequency Affected by Level Setting of Speakers

It will be realized that these shifts in crossover points may occur even for speakers of equal efficiencies if they are arranged with volume controls to vary their acoustic output. As soon as the level of output of the speaker is dropped, a change in acoustical output is produced, and the acoustic crossover moves around as the level is changed. Thus, for variable output networks, it may be desirable to use sharp rates of attenuation.

Crossover Frequency Affected by Level of Program Material

There is still another aspect of network performance that is beyond the control of the user, and that is the actual level of the program recorded or transmitted over the various bands. If the instruments of the orchestra are all playing at full loudness over the entire musical scale, the network-speaker-crossover combination obeys a general pattern similar to the crossover characteristic designed into the system. However, if only one portion of the original sound, say the middle frequencies, drops to pianissimo, the signal to the midrange speaker will drop, the acoustic output of that speaker will accordingly drop, and there will be a shift in the acoustical crossover.

Overall Crossover Characteristic Depends on Many Factors

It is for these reasons (commercial tolerances of components available for use in network, constantly varying speaker impedance, different efficiency levels of speakers, setting of volume controls, and signal distribution of the program) that the combined network-speaker performance may not be pinpointed microscopically. Inasmuch as it is impossible to take all these factors into account in the design of a network, the logical approach is to design the network to be electrically correct for the desired crossover points, at the attenuation rate desired, for the rated speaker impedance of the system, and to use commercially available components in its construction if the network is being built at home. If the ability to compensate for these changes in crossover points with frequencies or levels of speakers is desired, the constructor may use multi-impedance, multi-crossover adjustable networks.

PART 2

THE ENCLOSURE

CHAPTER 8: *Basic Enclosure Types*

The Enclosure is an Acoustic Circuit for the Loudspeaker

An essential part of any loudspeaker system is the enclosure in which the loudspeaker is to be used. An enclosure for a loudspeaker is often called a *baffle,* and perhaps technically speaking, *baffle* is a more descriptive word than enclosure, as the reader will soon see. The prime purpose of any enclosure is to provide the proper acoustic circuit for the loudspeaker to work with, so that maximum efficiency and best performance may be obtained from the *combination.* In the effort to provide this acoustic circuit, the sound coming from the loudspeaker is routed into certain paths and prevented from going into other paths by blank walls put in its way.

Thus the term "baffle" as used in a technical sense connotes a means of routing the sound energy. Since most of today's baffles are built into more or less complex box-like structures (even though the box may contain a horn), the words enclosure and baffle are used interchangeably. However, in certain instances, we shall specifically use the word *baffle* because in no sense will the word *enclosure* describe the function of the unit. This situation will arise in the discussion of horn tweeters as baffle crossover elements, for the horn is directly and precisely a baffle rather than an enclosure.

The choice of the proper enclosure or baffle for a desired loudspeaker system is governed by several factors. These factors are the speaker size, the performance range expected from the speaker-enclosure combination, and the manner in which the speaker is to be operated. The speaker-enclosure *combination* may be of the direct radiator type (such as the bass-reflex enclosure), of the indirect radiator type (such as the horn), or a combination of the two. Variations of these basic types are illustrated in Figs. 8-1, 8-2, and 8-3 respectively. We shall examine here the general differences between these basic types, and subsequent chapters will provide a more detailed study of each.

Direct Radiator: Speaker Works Directly Into Listening Space

As will readily be seen from Fig. 8-1, the direct radiator enclosure allows the loudspeaker diaphragm to be directly exposed to the surrounding atmosphere in which the sound is to be heard. *The loudspeaker radiates directly into the listening area.* There are no intervening acoustic elements between the loudspeaker and the ear other than the air itself. Despite the fact that there are no restraining elements upon the forward propagation of the sound, there may however, be some drastic modification made upon the overall performance of the system by the degree to which the rear of the speaker is enclosed. Here again, the word "enclosed" is used rather loosely. For instance, the loudspeakers shown in (A) and (B) of Fig. 8-1, are by no means enclosed. They are *baffled*, however, to the extent that the sound from the back of the speaker is prevented from spreading around to the front of the speaker by the extent to which the open baffle walls prevent it from spreading.

Enclosures Improve Low Frequency Response

Although open baffles of this nature are not usually encountered in present day high-fidelity practice, a brief discussion of them will illustrate two important points concerning enclosures. The first consideration, and the one more commonly appreciated, is that the purpose of the baffle or enclosure is to improve the low frequency response of the loudspeaker. The baffle accomplishes this by minimizing rear-to-front cancellation of the energy coming from the loudspeaker. If there were no obstruction in the way of such roundabout circulation of the acoustic energy, the loudspeaker would be virtually "short circuited" upon itself. As soon as the diaphragm moved forward, pushing out

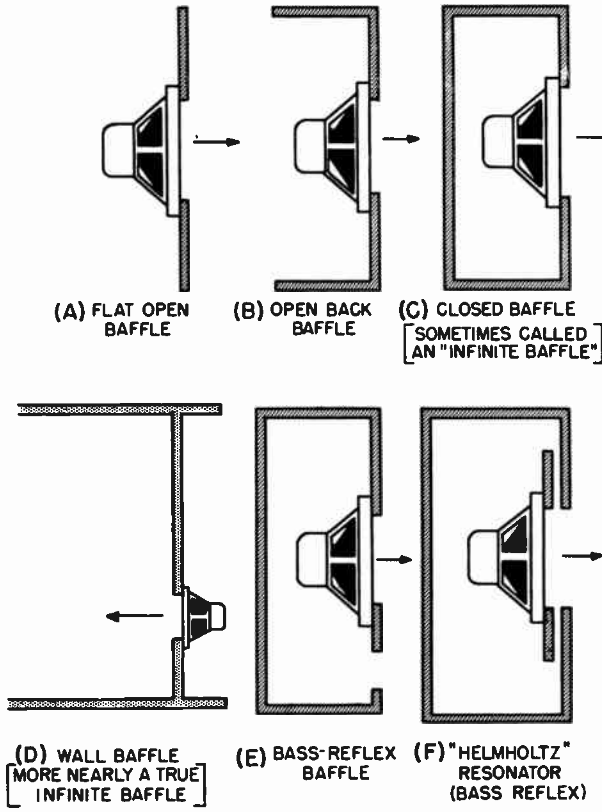


Fig. 8-1. Forms of direct radiator baffles. The face of the diaphragm feeds directly into open space.

a forward pulse of sound, the rear of the diaphragm would create a momentary vacuum in the back, thereby sucking in the sound the front of the diaphragm was trying to push out. Such short circuiting of the sound from the front to the back would simply result in no sound being *propagated*. This effect is most pronounced at the low frequencies because it takes a comparatively long time for a low frequency wave to become established in the air, and the longer this takes the more easily the wave is destroyed by the rear "pulling-in" process.

Obviously, if the low frequency wave is given an opportunity to build itself up *in space* before it is sucked into the rear, there will be an improvement in low frequency sound propagation. Any wave, radio or sound, once it has been produced in open space will tend to radiate outward from its source. One way of encouraging the buildup of the low frequency wave from a loudspeaker is to put some obstacle in its path so it cannot get back to the rear of the speaker. This is precisely

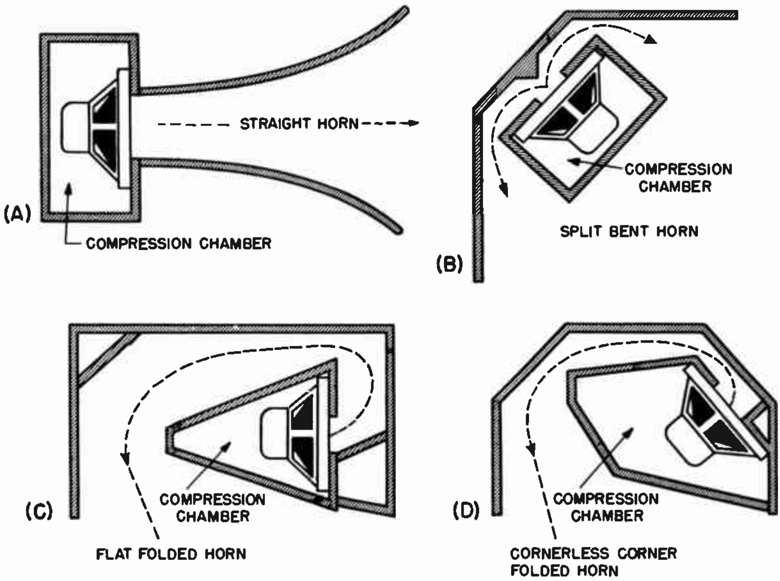


Fig. 8-2. Forms of indirect radiator baffles. The horn element is interposed between listening area and diaphragm. Compression chamber seals off rear of diaphragm. (Parts (C) and (D) after University)

what the flat baffle and the folded open baffle of Fig. 8-1 accomplish. These baffles elongate the path from front to rear of the speaker and thereby give the sound a chance to develop into some sort of wave motion before being pulled in around the back. Since low frequencies are long wavelengths, it follows that the longer the baffle length the lower will be the minimum frequency the loudspeaker can radiate before destructive front-to-rear cancellation takes place. This explanation of the baffle effect is considerably simplified, however. In the next chapter we will discuss the "dipole" effect, which explains baffles in a more technically accurate manner.

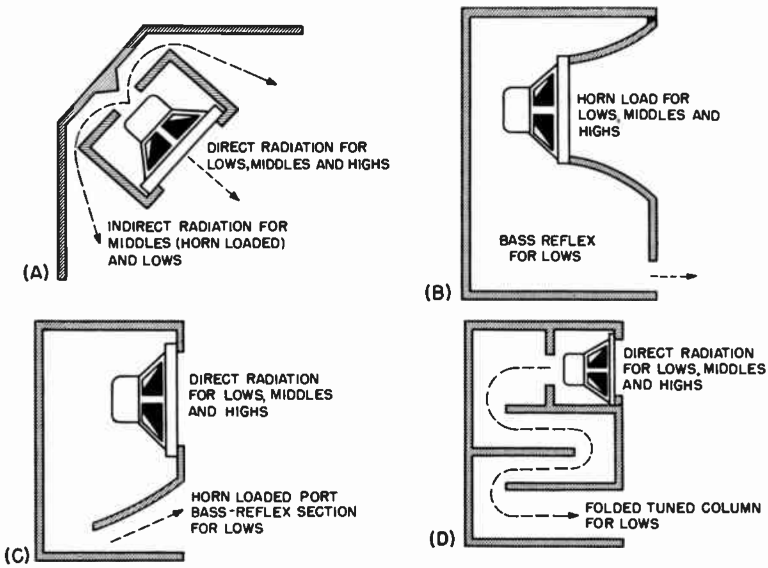


Fig. 8-3. Combinations of direct and indirect radiators.

Enclosure Shape Affects Performance

The second property of the baffle is to act as a *radiating surface itself in addition to the loudspeaker.* This statement should absolutely not be interpreted to mean that the baffle should vibrate. It should, on the contrary, be as rigid and non-vibratable as possible, so that it will not wastefully absorb sound power. As the sound comes out of the loudspeaker and spreads out, hitting the outside surface of the baffle, it literally should meet a hard immovable wall that will immediately bounce the sound energy into the listening area where it is needed. Thus the disposition of the baffle surfaces is as important as the length of the baffle.

Figure 8-4 shows two baffles, a conventional flat one and an unconventionally open folded one. Although the length of the baffle walls of both these systems may be identical, the extremely folded one will have considerably poorer low frequency response than the flat open one. The flat baffle will act as an auxiliary "springboard" for the sound wave being built up in front of it, and give the sound wave an added send-off into space. On the other hand, the extremely folded baffle will permit the sound wave being developed to fall in immediately on the sides of the baffle. Now, although the wave energy will not be

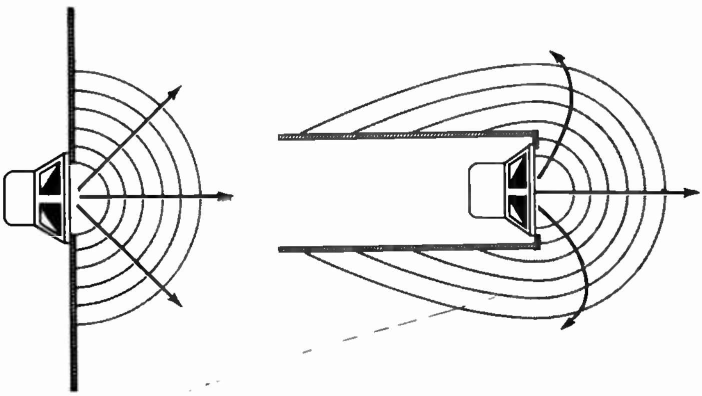


Fig. 8-4. Baffle shape is as important as baffle size. Baffle (A) will provide proper "sounding board" action and develop good radiation. Baffle (B) will allow wave motion to diffract around the sides and wave formation will be deteriorated. This baffle may also become tuned to one frequency depending upon its length.

absorbed into the back of the speaker in this baffle any more than it was in the flat baffle, it is difficult for that wave motion to be established, because of this "falling back" process. In more technical terms, the *small front* surface of the above open folded baffle permits "diffraction" to take place around the baffle *for long wavelengths,* which deteriorates the setting up of the wave motion.

There is still another peculiarity of such an extreme baffle shape. In the shape in which it is shown, this baffle actually represents a long tube, which may become resonant (like an organ pipe) to a very small band of frequencies whose wavelengths correspond to a multiple of the length of the baffle "tube." Thus, instead of giving smoothed-out low frequency response, as should be expected from a good baffle, a baffle such as this may give a very peaked narrow band response. However, when properly matched to the resonance of the speaker and enclosed in a structure that has a reasonably large front surface as well, this tuned tube may give quite good low frequency response. Such an enclosure is the folded tuned column type shown in Fig. 8-3 (D). It will thus be seen that the shape of the baffle is as important as its size.

The shape of the baffle is an important factor not only in the low frequency region, but in the high frequency area as well. The shape of a high frequency horn, which is fundamentally a baffle, has a great deal to do with the manner in which the high frequencies are distributed throughout the listening area. The shape of the horn and its flares will greatly determine the degree to which the necessary high frequency wide angle coverage is obtainable.

These factors (rigid non-vibratable panels, size of enclosure, and shape of enclosure) are thus important aspects of baffle design and will be treated in detail in Chap. 14.

Completely Closed Enclosure Stops Front to Rear Cancellation

Let us return to the concept of the elongated baffle path making it difficult for the front propagating wave to reach the back of the loudspeaker. It is obvious that by completely enclosing the back of the speaker we get an infinite degree of blocking of the front wave. Such an enclosure is therefore often called an *infinite baffle,* although in a strict technical sense, the term is not accurate. An infinite baffle is one that presents an infinitely large *baffle surface and rear volume* to the speaker. Referring to (C) of Fig. 8-1, it will be immediately apparent that a closed box, if small in size, falls far short of presenting to the loudspeaker either an infinitely large baffle surface or an infinitely large rear volume. In fact, if this box is made small enough, it may literally function in a manner opposite to that of a true infinite baffle.

It is of course a physical impossibility to obtain an infinitely large baffle surface and rear volume. These are only theoretical terms; but, for practical purposes, they may be effectively approximated within reasonable dimensions to give performance close to what a true infinite baffle would produce. If our closed box were made quite large [if it were, for example, a large closet or another room with the speaker mounted in the intervening wall, as in (D) of Fig. 8-1], it would very nearly constitute an infinite baffle. There would be no rear-to-front wave cancellation. Furthermore, the baffle surface would be so large that it would provide more than enough active surface to allow the lowest frequency of which the speaker is capable to develop fully into a well established wavefront.

One thing will immediately be obvious from the above discussion. It is necessary to "enclose" the speaker in order to get good low frequency response. However, when taken to the extreme, a completely enclosed box structure or a wall structure will completely prevent the use of the sound from the rear of the speaker. This rear energy is lost as far as useful acoustic output is concerned. One way of overcoming this, among other deficiencies in the infinite baffle, is to provide a vent in the closed box so that sound from the rear of the speaker may be given a chance to emerge from its container and do useful work.

Vented Box Makes Rear Wave Aid Front Wave

This type of structure, shown in (E) of Fig. 8-1, is commonly referred to as a bass-reflex enclosure. The actual operation of this enclosure is not as simple as it might seem. There is a very definite relationship between the size of the enclosure, the size of the opening (usually called the port), and the resonance of the loudspeaker. When these three factors are properly integrated, the rear wave from the loudspeaker is delayed just enough so that when it emerges from the port it is in phase with the wave motion from the front of the loudspeaker. Thus, the rear wave reenforces the front wave and more acoustic output is obtained than if the rear of the enclosure were completely closed; also, through action that will be described in later chapters this structure broadens and extends the low frequency response.

The wave delaying action is actually accomplished by reversing the phase of the rear wave, as will be explained in more detail in the next chapter. The bass-reflex enclosure is therefore often called a "phase inverter." The principles upon which this type of enclosure were developed all stem from the work of a well-known pioneer in acoustics, Dr. H. Helmholtz, who outlined the relationships that determine the resonance of ported enclosures. Although his work was done in the late nineteenth century as part of his researches on the sensations of hearing, his results became extremely useful to the early designers of loudspeaker enclosures.

In Fig. 8-1(F) is shown another form of Helmholtz resonator, which operates in essentially the same way as the bass-reflex enclosure. The port in this case will be seen to be a channel that surrounds the speaker and is rather narrow in cross section. In addition to providing the necessary porting action, this narrow channel provides an acoustic resistance by means of the slot, as part of the phase reversal system. The *proper* amount of acoustic resistance in any enclosure, in addition to the damping produced by the radiation resistance of the device, will improve the low frequency damping characteristic of the system. The R-J enclosure is an example of this type of structure.

The six structures of Fig. 8-1 exemplify some of the speaker enclosures that are adaptable to *direct radiator use.* In all instances, the acoustic load that the loudspeaker sees on its front side is the whole atmosphere directly. These systems may be used with any type of loudspeaker, be it a woofer or a complex triaxial type. The fact that the loudspeaker radiates directly into the listening area means there will be nothing to impede the forward propagation of the sound from the loudspeaker. Thus, in the case of a wide range, coaxial, or triaxial speaker some type of direct radiator enclosure is a must, for these units with their integral treble and high frequency sections must play directly to the ear with no intervening walls in the enclosure to obstruct the direct flow of high-frequency sound.

In contrast to the direct radiator enclosure there is the indirect radiator enclosure. Here one side of the vibrating diaphragm of the loudspeaker works into an acoustic load presented to it by a *horn, which acts as an intermediary coupling agent between the diaphragm of the loudspeaker and the surrounding atmosphere.* A few of such indirect radiator combinations are indicated in Fig. 8-2. It will be apparent from these illustrations that the horn portion of the system may take a variety of shapes.

It may be a straight horn, a split bent horn, or a folded horn. The horn acts as an acoustic transformer that matches the high mechanical impedance of the vibrating diaphragm to the relatively low acoustical impedance of the air at the large mouth of the horn. By virtue of this transformer action, the comparatively small area of the diaphragm finds it relatively easy to "grab hold" of a large quantity of air by easy small steps through the gradually expanding cross-sectional area of the horn through which the sound has to travel outward. Because of this improved interaction between the diaphragm of the loudspeaker and the air, we might say that the horn transformer provides more efficient coupling between the speaker and the air. The efficiency of the system may thus be greatly increased. A well designed horn reproducer may have an efficiency of between 40 and 50 percent compared to a direct radiator efficiency of only about 10 percent.

Another important aspect of the completely indirect radiator system is the complete back enclosed volume of the loudspeaker. This back volume is quite small, so small, in fact, that the rear radiation from the speaker is considerably cramped. Stating the situation somewhat differently, the rear of the loudspeaker is under acoustic *compression.* In general, the purpose of providing the compression chamber behind the speaker is to adjust the acoustic stiffness of the vibrating diaphragm so that it operates with optimum efficiency in the area where the horn becomes effective, and to balance the horn air load mass. This matter will be treated in more detail in Chap. 10. For the present, it is important to realize that the *compression chamber* is an essential part of the indirect radiator. When the cone speaker is enclosed in this manner, it becomes what is commonly called a "compression type driver."

It should be realized, however, that the same loudspeaker in some other kind of enclosure, such as the bass-reflex type, is not a compression type unit. *It is simply the application in which the speaker is used that makes it one or the other.* Thus, in the present instance of the indirect radiators illustrated, we have true compression driver, horn loaded systems.

Horns (Indirect Radiators) May Take Many Shapes

It will be apparent that the horn may take several different shapes, depending upon the application desired, or the space available. The straight horn of (A) in Fig. 8-2 takes up the most space in its construction, but is perhaps the easiest type of horn to build *with complete theoretical accuracy* because of the absence of bends. Because of the space requirements demanded of this straight horn when designed for low frequency reproduction, it is seldom found in home installations. However, we may still use this horn type of enclosure in the home if we rearrange its shape. For instance, it may be put into the corner of the room if it is of the type shown in (B) of Fig. 8-2, which illustrates

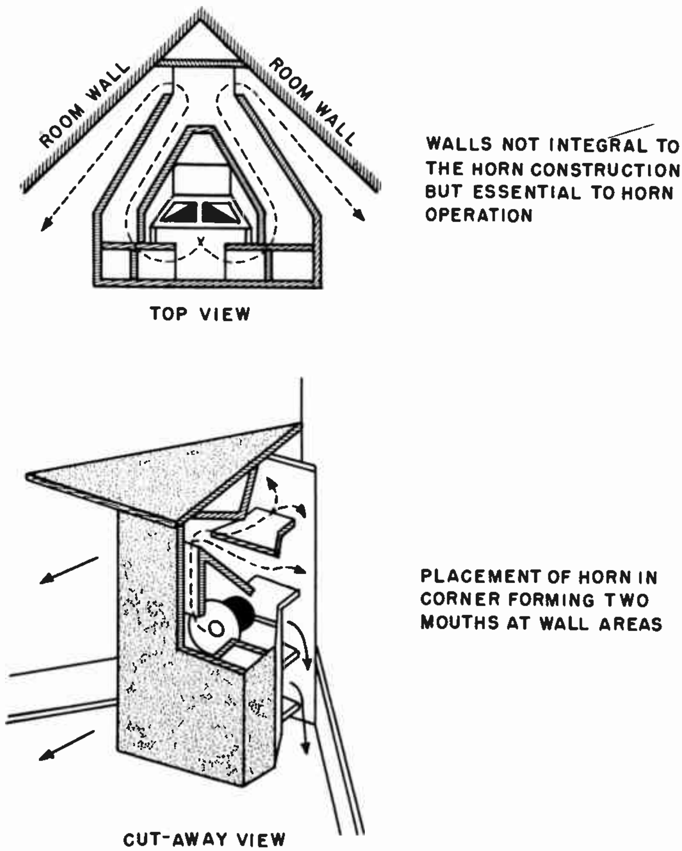


Fig. 8-5. The folded horn may be designed to be completely corner loaded by the walls of the corner of the room. (Klipschorn)

a form of split bent horn. Although the figure shown here is quite simple in concept, it may be elaborated upon in such corner horns as the University Classic and Dean systems of Fig. 8-2(C) and (D), the Klipschorn type of construction (Fig. 8-5), and the Lee Catenoid system (Fig. 8-6). In these structures the proper length of horn is obtained (by means of multiple bending) in reasonable enclosure sizes. Although there are essential differences between these horn structures, such as the use of walls of the corner of the room to complete the horn element of the Klipsch and the Lee systems and the self contained horn walls of the University systems, they fall into the same category; they are all indirect radiators (specifically, compression driver, horn loaded systems), operating on more or less the same basic principles. In application, there are differences between these systems, which will be discussed in Part 3.

Indirect Radiator is Anti-resonant

One very important characteristic of such horn systems differentiating them from the "enclosure" type is that, while the "enclosure" types are resonating devices, the horn systems are anti-resonating de-

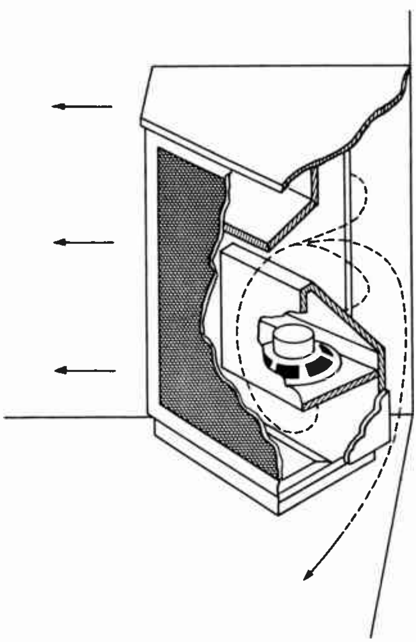


Fig. 8-6. Another form of folded horn that utilizes the corner walls of the room for completion of the horn load. (Lee Catenoid Horn)

vices. In the bass-reflex enclosure we try to tune the enclosure to work in conjunction with the natural resonance of the loudspeaker driving it; in the horn loaded system we try to eliminate any resonances in the horn system itself.

It will be realized that the horn type of baffle does not readily lend itself to use with extended range types of loudspeaker. Note in Fig.

8-2 that the low frequencies flow readily out of the loudspeaker, through the horn, and out into space, because low frequencies readily diffract or bend around corners. Middle and high frequencies, however, will not travel around bends as easily, and therefore are absorbed or lost in the maze of the horn. Consequently, folded horn systems using cone type loudspeakers are suitable almost solely for the reproduction of low frequencies. To complete the reproducible spectrum it will, of course, be necessary to use supplementary speakers, such as mid-range speakers and tweeters mounted in the front of the enclosure so that they radiate directly into free space, as described in the section on multi-speaker systems in Part 1.

Direct and Indirect Radiator Characteristics May be Combined

There are several types of enclosure in which the principles of the direct radiator and the indirect radiator are combined to obtain some of the advantages of both. These are illustrated in Fig. 8-3. Item (A) shows the previously-described split horn, now arranged so that the speaker in its compression chamber may simultaneously radiate directly to the listening area and feed the rear horn. This speaker may now be an extended range unit, or a coaxial or triaxial speaker. The high frequency components have an unobstructed path of operation into the room, while the low end of the spectrum emerges from the rear of the speaker through the horn load.

Item (B) is a combination of a bass-reflex enclosure, in which the low end of the spectrum is re-enforced by the phase inversion action, and the comparatively small horn that loads the front of the diaphragm, re-enforcing the middle range of the speaker. (The size of the horn is compatible with these middle frequency wavelengths.)

Item (C) is a bass-reflex enclosure in which the port is horn loaded. In this structure, the action of the enclosure is essentially that of the bass-reflex device in which the cabinet is tuned to the loudspeaker in the usual manner by providing a port that, together with the cabinet volume, resonates with the speaker. The port in this structure, however, does not radiate directly into the listening area, but works into the horn section that intervenes between it and the listener. The *mouth* of this port horn provides a larger radiating area than the port itself has. Because of this enlarged mouth, or radiating area, the low frequencies that radiate from this section are more efficiently reproduced than they would be if they were reproduced from the smaller port area.

Item (D) is a folded tuned column structure, which is a combination of a direct radiator for the front of the speaker and a tuned column for the reproduction of low frequencies from the back of the diaphragm. This structure differs from the rear horn in item (A) in that while the horn is a high-pass device passing all frequencies above its theoretical cutoff frequency, the folded column is a tube of uniform cross-section cut to resonate at a particular frequency as dictated by the resonant frequency of the speaker used in the system. When the tube is made the proper length, the resonant frequency peak of the speaker is severely damped and a new lower resonant peak is introduced, which extends the low frequency response of the system.

These short introductory paragraphs have served to acquaint the reader with the various basic types of enclosure prevalent in hi-fi practice. More intensive consideration of these various types will be presented in the following chapters.

CHAPTER 9: *The Enclosure Type Baffle*

Radiation Resistance of Loudspeaker Determines Power Output

The term *circuitry* is just as applicable to acoustic devices as it is to electrical components; a loudspeaker requires an acoustic load, just as an electrical circuit requires an electrical load, if power is to be developed. The purpose of any baffle is to provide the speaker with the proper load, into which it operates to produce acoustic power output.

However, acoustic circuitry is somewhat more complex than electrical circuitry because the sound waves developed are not as easily confined and channeled as are electrical currents flowing through wired circuits. On the other hand, there is a very close analogy between sound wave propagation and radio wave propagation. The student of physics will find the relationships that describe the propagation of these two types of energy to be very nearly identical, and one term common to both fields is "radiation resistance." The radiation resistance of an antenna determines how much radio power will be transmitted (or received). Similarly, the radiation resistance of a loudspeaker determines its power transmitting capabilities.

Radiation resistance is not a constant factor. It changes with the frequency of the power being transmitted and with the dimension of the device that does the transmitting. Common everyday illustrations of this fact are seen on almost every rooftop today. Television antennas are cut to size for a particular band of frequencies. Where optimum reception for all channels is desired, we may find a separate antenna cut to exact size to match the frequency of transmission of each channel. However, where economy of installation requires that only one antenna be used for all channels, it is probable that some channels will not come in as well as others. This occurs because they are not matched in impedance (mainly radiation resistance) for the frequencies of those channels which are not well received. This mismatch reduces the signal sensitivity, and may set up energy reflection, which the layman commonly recognizes as "ghosts" on his television screen. To briefly summarize, all wave transmitting devices, electrical *and acoustical,* have a definite *radiation resistance, which determines the power output, and which is a function of the frequency being transmitted, the method of coupling, and the size of the transmitting "antenna."*

In the case of the acoustic "antenna" (the loudspeaker), the general curve that describes its radiation resistance is shown in Fig. 9-1. Note that the radiation resistance of the loudspeaker increases from some very low value to a value of approximately 42 acoustic ohms (per square centimeter). Now, the specific acoustic impedance of air is also approximately 42 ohms. This means that maximum power will be transferred to the air from the loudspeaker when the loudspeaker approaches an impedance of 42 ohms per square centimeter, for then the "generator" impedance will be equal to the load impedance. This principle is as true in acoustics as it is in electrical systems.

Radiation Resistance Varies with Size of Speaker and Frequency

Inspection of Fig. 9-1 will show that the radiation resistance of the loudspeaker drops quite rapidly as the ratio of the diameter of the loudspeaker to the wavelength of the sound decreases. Although this may appear somewhat complex, let us examine it in terms of simple principles, for once it is understood the entire subject of baffles and enclosures will be more readily appreciated. Briefly, the curve means that if we keep the wavelength (or frequency) unchanged and let the diameter of the loudspeaker decrease, the ratio of the diameter to the wavelength decreases, and the radiation resistance per unit area of the loudspeaker drops. The power transmitted for that particular frequency drops correspondingly. If the speaker size remained unchanged, however, and instead the wavelength increased (frequency decreased), the ratio of the diameter of the speaker to the wavelength would also de-

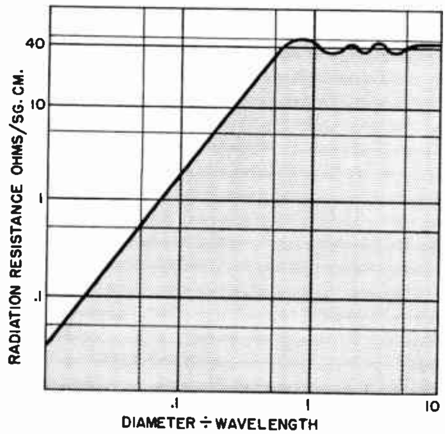


Fig. 9-1. The acoustic power developed by a loudspeaker is a function of its radiation resistance, which increases as the ratio of the diameter of the piston divided by the wavelength of operation.

crease, again there would be a drop in the radiation resistance of the speaker, and less power would be radiated.

This explains the two commonly experienced phenomena occurring in low frequency reproduction from loudspeakers. First, for a given low frequency, the smaller the loudspeaker the poorer will be the power output. Secondly, for a given size of loudspeaker, the low frequency output will drop off as the frequency goes down. Now let us apply these principles to actual baffle configurations to illustrate how a baffle provides the proper "load" for a loudspeaker.

In the previous chapter, it was stated that in the absence of a baffle the sound transmitted from the front of the loudspeaker is sucked back to the rear of the loudspeaker as the diaphragm moves forward. This pulling back of the sound was referred to as a "short circuiting" effect on the loudspeaker. We may now re-examine this "short circuiting" effect in more exact terms of baffle size and of another phenomenon called "doublet operation."

The Acoustic Doublet: Radiation from Both Sides of Diaphragm into Space

A loudspeaker actually radiates sound from both sides of its diaphragm (Fig. 9-2A). However, these two sound fields are out of phase with one another. As the diaphragm moves forward, there is a pressure area directly at its front surface. At its rear surface, there is an area of rarefaction. In other words, two distinct wave radiations start traveling away from the respective sides of the loudspeaker diaphragm in an out of phase condition as if they had come from two adjacent bodies pulsing out of phase. Such a system is referred to as an "acoustic doublet" — an obvious name — and is illustrated in Fig. 9-2(B) in which the two sides of the diaphragm are represented by a negative and a positive

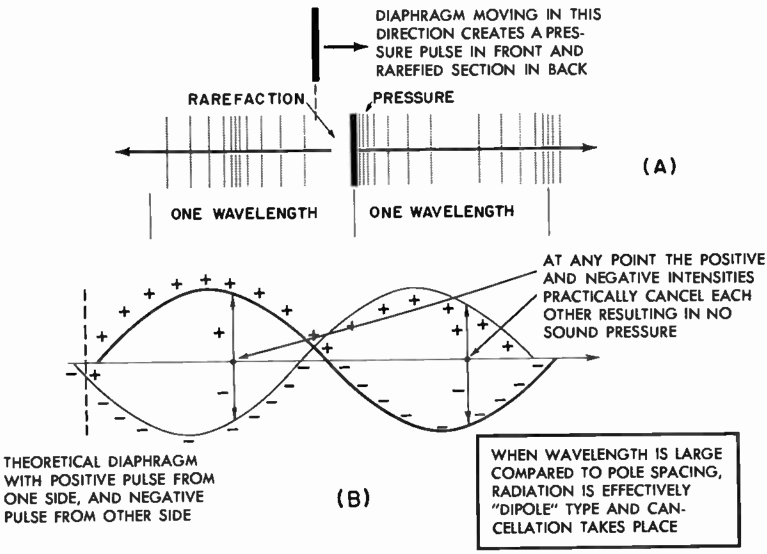


Fig. 9-2. Doublet operation of unbaffled diaphragm.

pole radiating respectively a "negative" and a "positive" wave. If these doublet sources are close together, it is obvious that the "negative" wave from one will be directly canceled by the "positive" wave from the other at almost all points in space. This is true, however, if the poles of the doublet are so close that their spacing is only a small fraction of the length of the radiated wave as shown in Fig. 9-2(B). This means that the longer the wavelength (low frequency), the closer the poles comparatively speaking, and the greater the low frequency cancellation. This condition we readily recognize in any loudspeaker when tested on a work bench without benefit of any baffle at all. Under this condition, the doublet mode of operation is most effectively working, with the result that there is a complete absence of low frequency response.

It is an obvious conclusion that to prevent the low frequency cancellation resulting from doublet operation, we should place some barrier between the front and the back, hence the baffle. Let us now examine the baffle size in terms of the wavelength being transmitted to see its effect upon doublet operation and its value in the elimination of such operation. Figure 9-3 shows a loudspeaker mounted on a small baffle. We will examine what happens at some point (P) in space without a baffle, and then with the baffle. Without a baffle, let us assume that the point (P) is at a given instant receiving a positive pulse from the front side of the loudspeaker. If the frequency of operation

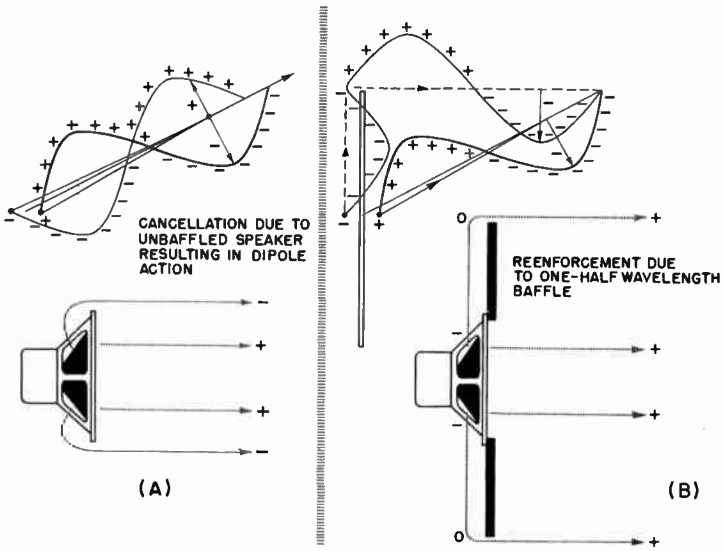


Fig. 9-3. Baffle eliminates doublet action by increasing back wave path length by one-half wave length.

of the loudspeaker is low, there will be effective doublet operation. The wave from the rear of the loudspeaker will reach around and approach point (P) with a polarity directly opposite to that of the front wave. The acoustic pressure at this point will be effectively canceled out, giving no response for this frequency.

Now let us put the small baffle in place, and assume the baffle size to be such that the wave from the rear has to travel an extra length of time equal to that taken up by one-half of the wavelength of the sound being transmitted. This means that now the out-of-phase wave from the rear of the speaker, having been delayed by the necessary amount, will reach point (P) *in phase* with the front wave, and the doublet action will have been eliminated. The result will be re-enforced lows rather than canceled lows.

Baffle Minimizes Doublet Operation

It is obvious that to eliminate this doublet action, the dimensions of the baffle should add an extra half wavelength of the desired low frequency sound to the total front to rear path. For frequencies above the dimensions at which doublet operation is minimized by the proper size baffle, there will be relatively smooth output. However, where the baffle size becomes too small to prevent doublet operation, the response curve begins to fall abruptly. It follows, conversely, that the larger the baffle the more effective will be the elimination of the doublet action for lower frequencies and the better will be the low frequency response. Figure 9-4 illustrates the effect of increased baffle size on low frequency output. (Note: There will be improved smoothness of response above the frequency at which doublet operation occurs if the speaker is not mounted directly at the center of the flat baffle board or if an irregular baffle shape is used.)

Although the open type baffle does assist in the improvement of low frequency response, it also has its shortcomings. In order to obtain acceptable low frequency extension from the open baffle type, the dimentions become rather unwieldy for normal home use. If we assume that for elimination of the doublet operation we need a baffle at least one-half the wavelength of the sound to be transmitted, for a frequency of 50 cps, which corresponds to a wavelength of approximately 22 feet, the distance from the speaker to the edge of the baffle would have to be 11 feet. Of course, the baffle may be folded back on itself as in the open back variety but such a baffle would still be an ungainly structure.

The open type baffle has another serious shortcoming in that there is no acoustic damping control of the free resonance of the loudspeaker. It is true, however, that the air load the loudspeaker sees adds some "mass" to the total moving system. This air mass lowers the resonance point slightly, and also reduces the excursion of the cone at resonance to some degree. In the main, however, there is no other type of acoustic constraint upon the motion of the diaphragm. It is virtually as free to move at its resonant condition as if the baffle weren't there at

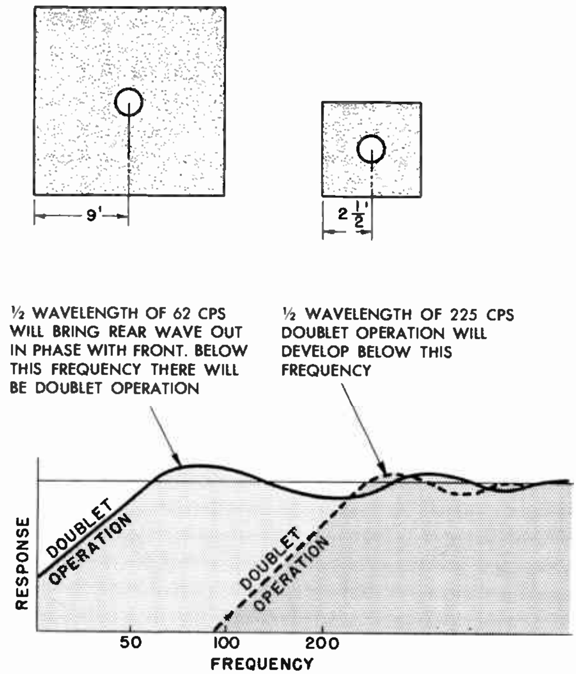


Fig. 9-4. Larger baffle lowers frequency at which doublet operation starts.

all. There will, therefore, be a tendency toward boominess in the response of the system at the resonant frequency. This boominess should not be interpreted as "bass." It is in truth a "one note" bottom that lends some feeling of bass, but is entirely out of proportion to the original signal.

Because of the low frequency limitations of the open baffle and the raggedness of its response, this type of enclosure is practically never used in modern high fidelity practice. However, an understanding of its principles is necessary since most other enclosure structures are in essence attempts at eliminating the defects of the open baffle type.

A True Infinite Baffle Completely Eliminates Doublet Action

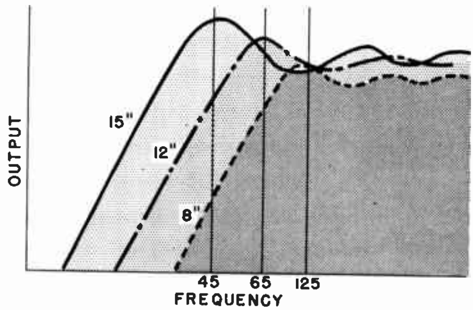
A natural extension of the open baffle is the *true* infinite baffle. In this type of baffle there is absolutely no front-to-rear cancellation of sound energy at *any* frequency. This is accomplished by putting the speaker in a partition that completely seals off the front from the rear, but which at the same time allows perfect freedom of sound radiation from both sides of the loudspeaker into the areas they face. Such an infinite baffle might be the wall between two rooms as shown in Fig. 8-1(D). If the loudspeaker were mounted in such a wall, one side of the speaker would have no effect upon the other. At the same time, the rooms on either side of the wall would enjoy virtually equal acoustic output.

The complete absence of doublet operation in this type of installation will ensure that the *low frequency response obtained will be limited not by the baffle, but by the low frequency capabilities of the particular loudspeaker used in the installation.* Thus, if an 8-inch loudspeaker with a resonance of 125 cps were used in such a wall bafflle, its response would be fairly smooth down almost to this frequency, then there would be a rise at 125 cps due to the resonance of the speaker. Progressing downward in frequency from this resonant peak at 125 cps, there would be a rapid drop in response at the rate of 12 db per octave. The same general condition would hold for a larger speaker, but with a change in resonant frequency as illustrated in Fig. 9-5.

This type of operation is characteristic of the wall type infinite baffle. It works down to the resonant frequency of the speaker being used, but fails rapidly below its resonant point. Its other shortcoming, as in the case of the open baffle, is the lack of control that may be exercised acoustically over the resonant frequency excursions of the loudspeaker. With the exception of the air mass and radiation resistance loading, which will affect the resonance and damping of the speaker, for all practical purposes both sides of the loudspeaker are perfectly free unconstrained radiators into free space. The resonance is materially the same as in the unbaffled condition. There is a consequent one-note boominess to true infinite wall baffles (although not as pronounced as in open baffles) with rapidly dropping response below this area of resonance. This effect becomes more aggravated as the size of the speaker is increased, for the larger the speaker the lower the resonant frequency under normal circumstances; and consequently, the lower will be the undamped resonant frequency of the diaphragm excursion and the greater the output at the resonant point.

There are variations of the wall baffle that still constitute, in essence, true infinite baffles. One such variation is the closet installation, in which the speaker is mounted in the door of a closet, the rear of the speaker facing into the cavity of the closet. If the closet is large, its

Fig. 9-5. The performance of the true infinite baffle (wall type) is a function completely of the loudspeaker itself with the sole purpose of the baffle being the complete elimination of the dipole effect. After this is accomplished the response capabilities of the speaker itself determine the response of the system.



volume will for all practical acoustic purposes be equivalent to an open room, and the loudspeaker will perform as it would in a pure wall baffle, with the important exception, of course, that only one side of the loudspeaker will be doing useful work.

It is obvious that the installation of a speaker in a wall baffle intended to feed two rooms must be made on the basis of a loudspeaker of the all-purpose wide range variety, in order to have equivalent performance from both sides of the loudspeaker. If a two or three-way speaker were used, only one room would benefit from the full operation of the speaker, while the other room (on the rear side of the speaker) would receive only the woofer portion of the response. Infinite baffles of the closet type exhibit good low frequency response down to the natural resonance of the speaker by virtue of the high order of radiation resistance of the speaker when mounted in such baffles; below this resonant point, the low frequency output drops at the rate of 12 db per octave.

Closed Box Raises Resonance of Speaker

There is another type of enclosure called an infinite baffle that is not strictly in that category. This is the completely closed box, with no openings other than the one in which the loudspeaker itself is mounted (Fig. 8-1C). The reason for calling this an infinite enclosure is quite obvious, considering that the back of the loudspeaker is completely sealed off from the front. In this sense *only,* the enclosure acts as does the true infinite wall baffle. However, there is one important difference between the wall baffle and the closed box, specially when the closed box takes on the small proportions necessary to make it suitable for placement in the usual living room. The relatively small and unrelieved volume of space that encloses the back of the loud-

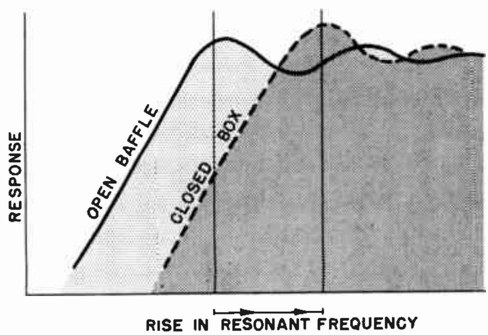


Fig. 9-6. Air captured in a sealed enclosure acts as an acoustic stiffener upon the back of the loudspeaker and raises the resonant frequency of the speaker.

speaker becomes an acoustic restraint upon the loudspeaker. The small volume of air captured by the enclosure has no way of escaping and acts as an acoustic spring upon the loudspeaker. It *stiffens* the loudspeaker and actually may raise the resonant frequency of the system (see Fig. 9-6) to some slightly higher value than the speaker had when it was unbaffled.

We must be careful, however, in interpreting what actually happens to the acoustic performance of the system. It would be wrong to state categorically that the low frequency response would suffer because of this rise in resonance frequency. The matter must be considered in the light of the original resonance and the rise in the resonance frequency *in relation to the size of the box in which the loudspeaker is to be operated.* Let us illustrate this with a practical example, as indicated in Fig. 9-7. At (A) we have a loudspeaker with a natural resonance of 100 cps; mounted in an open baffle, it still has the same general resonant point and produces the response indicated. Now, if this same enclosure were to be sealed off at the back, making it a closed

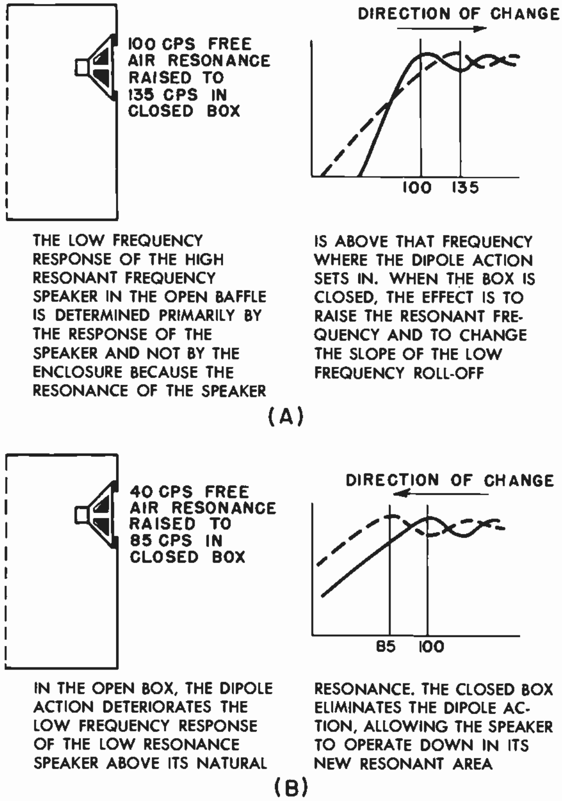


Fig. 9-7. Low frequency response is function of speaker resonance plus enclosure size.

box, the cone would be stiffened by the unrelieved air cushion at its back and the speaker resonance might go up to 135 cps. However, despite this increase in the resonant frequency, the actual low frequency output of the enclosed box system rolls off more slowly than the open back type, because of the elimination of the doublet action.

Suppose we substitute a speaker of much lower resonance, say 40 cps (Fig. 9-7B). In the same open back cabinet used before, the output of the 40 cps speaker will suffer from the same doublet action of the enclosure as did the higher frequency speaker, because it is the baffle and not the loudspeaker that determines where doublet action begins. Therefore, the low frequency output of this open system with the 40 cps speaker also begins to fall off at about 100 cps, despite the fact that it has a lower resonance. However, *because* it has a lower resonance, its output does not drop quite as rapidly as did that of the 100 cps speaker, as shown in the illustration. Now let this same enclosure be sealed again at its back, providing a stiffening air cushion at the back of the loudspeaker. Again the resonance of the loudspeaker is raised because of this added stiffness. In the present instance, it may go up from 40 cps to perhaps 85 cps. Note that the rise in resonance of the low resonance speaker (from 40 to 85 cps) is much greater proportionally than the rise in resonance of the high resonance speaker (from 100 to 135 cps). This occurs because the same box volume presents a comparatively stiffer cushioning effect to a loose system (40 cps resonance) than to a system that is already comparatively tight (100 cps resonance), acoustically speaking.

The response of the same speaker (40 cps) in the closed box shows a low frequency peak at a point much lower in the band than in the open baffle, despite the fact that it has been stiffened by being enclosed. The doublet action has been entirely removed, and the low frequency response is now limited only by the resultant resonance of the speaker-enclosure combination. To summarize: The closed box raises the resonant frequency of the speaker. For a given size box the rise in resonant frequency will be much greater for a low resonant speaker than for a high resonant speaker. The overall performance is a function of both box size and speaker characteristic resonance.

Closed Box and Infinite Baffle Systems Have Single Resonant Peak

Despite the fact that the closed box makes possible better response than the open baffle and also provides some "infinite" baffle action without recourse to breaking through walls, it does suffer from the same defects as true wall baffles and open baffles, but to a smaller degree. All three types may develop a resonant condition that produces a single note boominess. This condition is most prevalent when a low resonance speaker is used in an enclosure of moderate size that does not stiffen the speaker sufficiently. Invariably, boxed enclosures are thoroughly treated internally with acoustic damping material, which materially reduces this resonant boominess. All in all, however, the infinite baffle does represent a means of obtaining acoustic performance decidedly better than that obtainable from the open baffle, particularly if a fairly large box is used. (Note: To prevent the stiffness of the enclosure from raising the loudspeaker's resonant frequency by more than 10 percent, the enclosure volume should be at least 3½ cubic feet for a typical 8-inch speaker, 8½ cubic feet for a typical 12-inch speaker, and 15 cubic feet for a typical 15-inch speaker.)

Vented Enclosure Acts on Phase of Rear Wave Before It Emerges

One important disadvantage of the infinite baffle enclosure is the fact that radiation from the rear of the speaker is completely lost in the closed box. It cannot be made to do useful work because it has no way of getting out of the box. This condition is rectified in the "bass-reflex" (or "phase-inverter") enclosure. This is in all respects like the closed box, except that there is an opening, usually on the face of the enclosure. It would be simplifying the situation considerably to say that this opening allows the sound from the rear of the speaker to come around to the front. True, it does just that, but not before something important has happened to that sound. To provide an understanding of the action that takes place within the cabinet, we shall present briefly one or two simple concepts dealing with purely acoustic elements.

A volume of space of roughly symmetrical proportions represents an acoustical capacitance. If the reader refers to Fig. 9-8(A), he will see a structural configuration that represents a box with an inlet and an outlet. Let us now send sound into one opening. Nothing will come out of the other opening until the box portion has been "filled up," because this box is directly across the "line" — this structure has "capacitance." It has the capacity to store the flow of sound. In Fig. 9-8(B) is a long hollow tube wound into the form of a coil. Now, when one attempts to send sound into this "coil," the pulsations of air (which has actual mass) will exhibit a true inertia effect. They have difficulty negotiating the turns of the tube because of the inertia of the pulses, and the tube acts as an acoustic inductor. It will have the property of preventing the flow of the sound pulsations. Even if this tube were straightened out, it would still exhibit acoustic inductance, or "inertance," which would impede the passage of sound through it just as a straight wire may exhibit a self-inductance that offers considerable opposition to the flow of very high frequency currents. Even if the

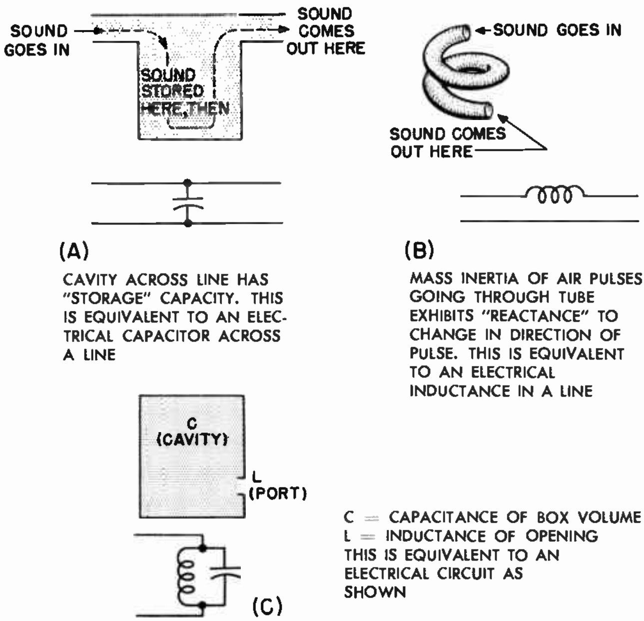


Fig. 9-8. Simple electrical analogies of acoustic elements in a vented enclosure.

tube were cut into small sections, each of these small sections would still produce the inertance effect. In fact, taking it to extremes, if a very thin section were cut and inserted in a partitioning wall to support it (in essence just a "hole in the wall"), it would exhibit its due measure of inertance.

Bass-Reflex Enclosure has Tunable Acoustic Properties

Thus we come to the two basic acoustic properties that must be dealt with in the simple bass-reflex enclosure. We have the box itself, which has *acoustic capacitance,* and we have the hole (the port), which has *acoustic inertance.* This combination in a single structure of both capacitance and inductance (inertance), as shown in Fig. 9-8(C), represents a resonant circuit, the resonant frequency of which is a function of the actual values of the capacitance and the inertance. If either of these two elements or both are made variable, the combination of the two becomes a tunable circuit.

This is the important feature of the bass-reflex enclosure; it can be tuned to resonate with the loudspeaker with which it is to work. The simplest type of bass-reflex enclosure is shown in Fig. 8-1(E), where the loudspeaker is usually mounted in the top section of the front panel and the opening or port placed near the bottom edge of the panel. By adjusting the port opening, the resonance of the enclosure is altered so that it matches the natural resonance frequency of the loudspeaker itself.

Now, why all this emphasis on resonating the enclosure to the speaker? Simply because this gives us a direct means of *matching* the enclosure to the speaker, by means of which we may greatly improve the performance of the *system* over that of a closed box. The speaker and the enclosure become a *system* rather than just two components put together. Because the loudspeaker is located in intimate contact with the interior of the bass-reflex enclosure, we may say that the loudspeaker and the enclosure are closely *coupled* together. This means that the performance of the enclosure is dependent upon the speaker, and vice-versa.

Tuning Bass-Reflex Enclosure Converts Single Resonance to Dual Resonance

A very interesting phenomenon occurs when two circuits of the same resonant frequency are closely coupled together. The system no longer exhibits a resonance at the original resonant frequency of the individual sections. The original resonant frequency has disappeared completely, leaving in its place two other resonant frequency points, straddling the original as shown in Fig. 9-9. We have seen from Part 1 that at its resonant point of operation the speaker is working at its greatest efficiency. If we now carry over this principle of maximum efficiency at resonance points to the present situation, we see immediately that the area of high efficiency (in the low frequency region) has been increased by the extent of the spread between the two new resonant points. This extension of the resonance area to about an octave above and below the original resonance point produces an additional effect upon the performance of the system. The resonance peaks are considerably reduced in amplitude partly by the close coupling condition and partly by the radiation resistance. The system peaks thus become considerably damped. Resonant boominess all but disappears and is replaced by fairly smooth unpeaked response over a broad low frequency area.

Phase of Rear Wave is Reversed Before it Emerges

There is one other important characteristic of bass-reflex operation that adds to its high level of performance. This is the degree to which the rear radiation is made to do useful work. Speaking very generally, if we can make use of this rear radiation, we have available

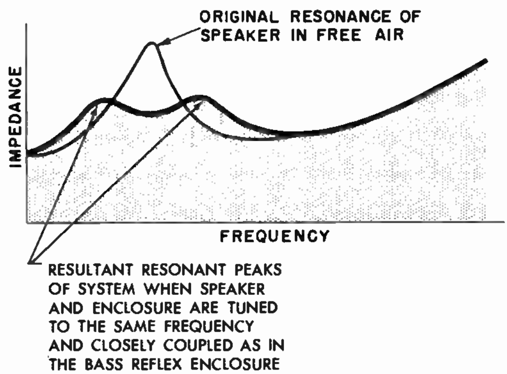


Fig. 9-9. Area of high acoustic efficiency is spread out over a broader band by the two separated resonant peaks of the bass-reflex tuned system.

about twice as much acoustic power. Now, this is not a matter of simply letting the sound come out of the box, for we know that if this sound emerges out of phase with the forward radiation, there will be cancellation rather than reinforcement. The trick is to bring this rear radiation out *in phase* with the front radiation.

A rigorous analysis of the mesh circuit phase relation that produces this reversal is beyond the scope of this treatment. It may be said, however, that the action of the acoustic elements of the enclosure is to reverse the phase of the sound coming out of the port for certain frequencies in the low frequency area. In a very general fashion we might explain this action on the basis of the flow of acoustic energy from the back of the speaker. It will be recalled that in the brief discussion of the acoustic elements of an enclosure a volume of space was referred to as an acoustic capacitance that has to be "filled" up before it can pass on any energy; that it *stores* acoustic energy like an electrical capacitor stores an electrical charge. So we might now come back to our bass-reflex enclosure and carry the analogy further by saying that the sound from the back of the speaker must fill the cavity of the enclosure before it reaches the port. The sound from the back of the speaker is therefore delayed enough in time before it comes out of the port (without traversing any large baffle distance) so that when it finally does emerge, it does so in-phase with the front-radiated sound. This, in a very general fashion, is the "phase-inverter" action of the bass-reflex cabinet, which makes the sound power from the back of the diaphragm as usable as that from the front of the diaphragm for low frequencies.

It must be realized that the phase inverting characteristic of the bass-reflex cabinet is effective only at the low frequencies. The "inertia" reactance of the port increases as the frequency of the air pulses is raised, just as the inductive reactance of a coil goes up for an increasing frequency of electrical current. The higher the frequency of the sound pulses trying to emerge from the port, the more inertia is exhibited by the air and the more difficult it is for these pulses to get through the port. Therefore, for the middle and upper frequencies, the bass-reflex enclosure reverts essentially to a closed box because all the radiation is from the front of the speaker. As the sound goes down in frequency, however, the port reactance also goes down. The sound can make its way out of the port, and the enclosure radiates both from the front of the speaker and from the port. This is the bass-reflex principle.

Port of Bass-Reflex Enclosure Has Radiation Resistance

The sound that eventually comes out of the port would not be radiated if the port were a *pure inertance. For sound power to be radiated, there must be an acoustic radiation resistance for the sound pulsations to work with.* We have seen from Fig. 9-1 that the radiation resistance of a vibrating diaphragm is a function of the size of the diaphragm and the frequency to be radiated. The pulsating air at the port opening is in reality a "mass-less" diaphragm. Its radiation resistance determines the efficiency of power transmission through the port. If the port is too small in relation to the wavelength, there will be low radiation resistance and low power output for the low frequencies.

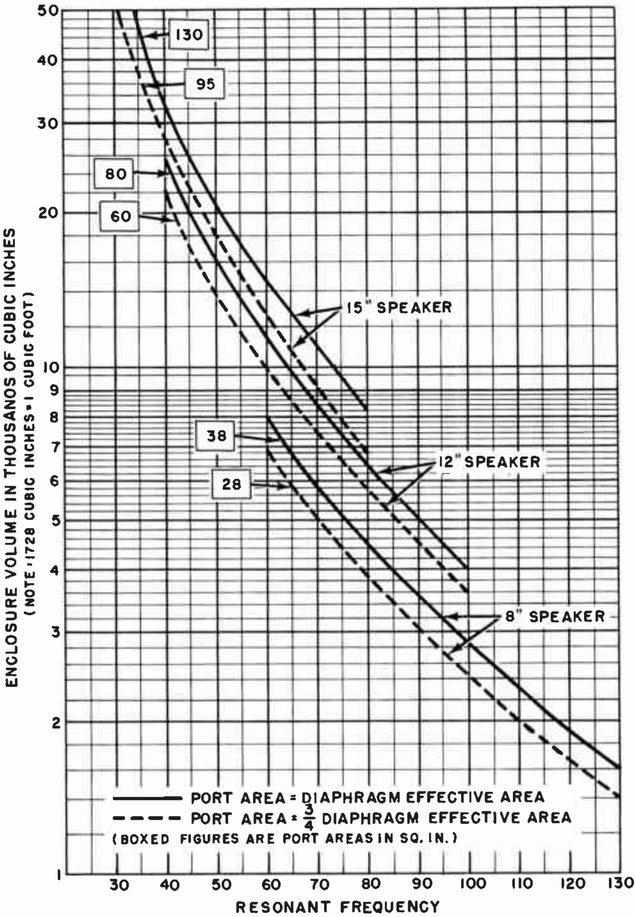


Fig. 9-10. Chart relating speaker resonance, speaker size, enclosure volume, and port area for a bass-reflex enclosure.

Designing the Bass-Reflex Enclosure

For this reason, best performance of the bass-reflex system is obtained when the port area is between three-quarters and one times the actual effective area of the loudspeaker diaphragm. (This effective area is πr^2, where r is the radius of the piston equivalent to the speaker cone at low frequencies.) Thus, if an enclosure is to be made for an 8-inch speaker, one should provide a port opening of from 28 to 38 square inches; for a 12-inch speaker, the port opening should be from 60 to 80 square inches; and for a 15-inch speaker, the port opening should be from 95 to 130 square inches.[1]

Once this port size has been chosen, the enclosure volume will be determined, for the two go hand in hand with the resonance of the speaker. Figure 9-10 is a chart for selecting the proper enclosure size for a speaker of a given resonance. This chart is made up for two conditions; in one the port size is equal to three-quarters the effective diaphragm area, and in the other the port size is equal to the effective diaphragm area. In the case of the larger port area (equal to the total active diaphragm area) the enclosure will necessarily be larger, but improved low frequency performance will result.

In order to use this chart, it is of course necessary to know the resonant frequency of the loudspeaker. If this information is not available from the manufacturer of the loudspeaker or if one wants to de-

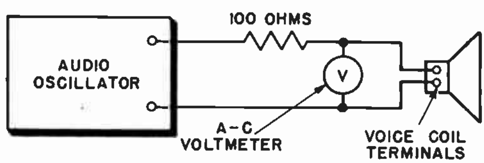


Fig. 9-11. Apparatus for determining the resonant frequency of a loudspeaker. At the resonant frequency the voltmeter will read maximum.

termine the exact resonant frequency for himself, the procedure is not too complicated. It is necessary, however, to have at one's disposal some elementary test equipment, such as a properly calibrated audio oscillator and a simple a-c voltmeter, in addition to a resistor of about 100 ohms. (This value is not at all critical.) Figure 9-11 shows how these elements are connected in order to find the resonance frequency of the unmounted speaker. As the audio oscillator is slowly swept through the low frequency end of the audio spectrum, it passes through the resonance frequency of the loudspeaker. The voltmeter, which is connected directly across the voice coil terminals, will show a definite rise in voltage at this frequency. The point at which the maximum reading of the voltmeter is obtained in the low frequency area is the resonant frequency of the loudspeaker. (Note: In the above procedure

[1]Refer to Fig. 12-6 for discussion of effective piston area.

do not place the speaker face down on a table or bench as its resonance will be completely damped out. The cone should point into free space as much as possible.)

Once this resonant frequency has been determined, and it has been decided whether the larger port area (for optimum port performance) or the smaller port size (to reduce the volume of the enclosure in general), will be chosen, the actual volume of the enclosure may be selected from the chart. In any event, it is desirable to make the port somewhat larger than necessary so that it may subsequently be tuned to the exact resonance where precision is desired.

Note that the chart gives the interior volume of the enclosure. For exact results, this volume should be increased by the space occupied by the loudspeaker itself, the internal bracing, and the sound absorption material used. Typical approximate volumes displaced by loudspeakers are 250 cubic inches for an 8-inch speaker, 650 cubic inches for a 12-inch speaker, and 1400 cubic inches for a 15-inch speaker. The volume occupied by the internal sound absorption material should be calculated from the area covered and the thickness of the material when held temporarily tightly compressed.

Tuning the Bass-Reflex Enclosure

This tuning of the finished system may be accomplished by closing off a portion of the opening with a movable panel of wood until the system is exactly tuned. Critical tuning may be accomplished by using the same equipment as before, but progressing in a somewhat more detailed fashion, as follows. Start with the port in its most open condition. Sweep the oscillator over the low frequency area of the spectrum *very slowly,* noting the frequencies at which the voltmeter connected across the voice coil shows a voltage rise. Two peaks of resonance will usually be found, one higher in frequency than the original resonant frequency of the unmounted speaker, and the other lower. However, the *amplitude* of these peaks will not necessarily be equal, nor will the peaks straddle the original resonance equally on either side, unless the enclosure should happen to be properly adjusted. Correct tuning exists when the two resonance peaks are of equal amplitude as shown by the voltmeter readings and when these peaks are equally displaced from the free air resonance. It will also be observed that when these two peaks are balanced in amplitude, their common amplitude is considerably smaller than the original free air resonance peak voltage. This is an indication of the damping effect on the speaker of the closely coupled, properly tuned bass-reflex enclosure. The voltmeter reading is actually a measure of the excursion of the diaphragm of the loudspeaker.

Figure 9-12 shows a set of experimentally determined tuning curves for a bass-reflex enclosure for various conditions of port opening. Note how the amplitudes of the two peaks tend to equalize and how their frequencies shift upward as the port is opened and the correct adjustment is approached.

Although proper tuning of the system may be made by the methods described above and considerable damping accomplished, use of this procedure does not necessarily mean that *optimum* damping has been accomplished. Optimum damping in the combined speaker enclosure system occurs when the Q of the system is equal to unity. This means that the ratio of resistance in the circuit to inductive reactance is unity. It will be realized that since we have not as yet made any resistive compensations in the above system, we have obviously neglected to examine the damping factor critically. Fortunately, it is not at all difficult to make the necessary adjustments to the Q of the system to ensure that no transient "ringing" or "hangover" remain in the system. We have seen in Part 1 that ringing in the speaker system is the condition in which the speaker overshoots and continues to oscillate for a short period of time after a single pulse has been applied to it. Hangover is the tendency of the speaker to continue to vibrate around its equilibrium no-signal position after the pulse has been removed. In order to make the necessary adjustment to eliminate these undesirable effects from the enclosure system, all the constructor needs is a fresh flashlight battery and some silk cloth of the thickness and weave of ordinary handkerchief material.

When tightly stretched across an opening through which sound is to flow, like the port, a layer of such cloth exhibits an acoustic resistance. The many small holes in the cloth and the fibrous filaments of the material over the opening provide a resistive block to the sound waves. This works only when the cloth is tightly stretched. When it is not, it may try to vibrate under the influence of the sound. If this happens, the cloth acts like a membrane and adds inductance to the circuit as well as resistance.

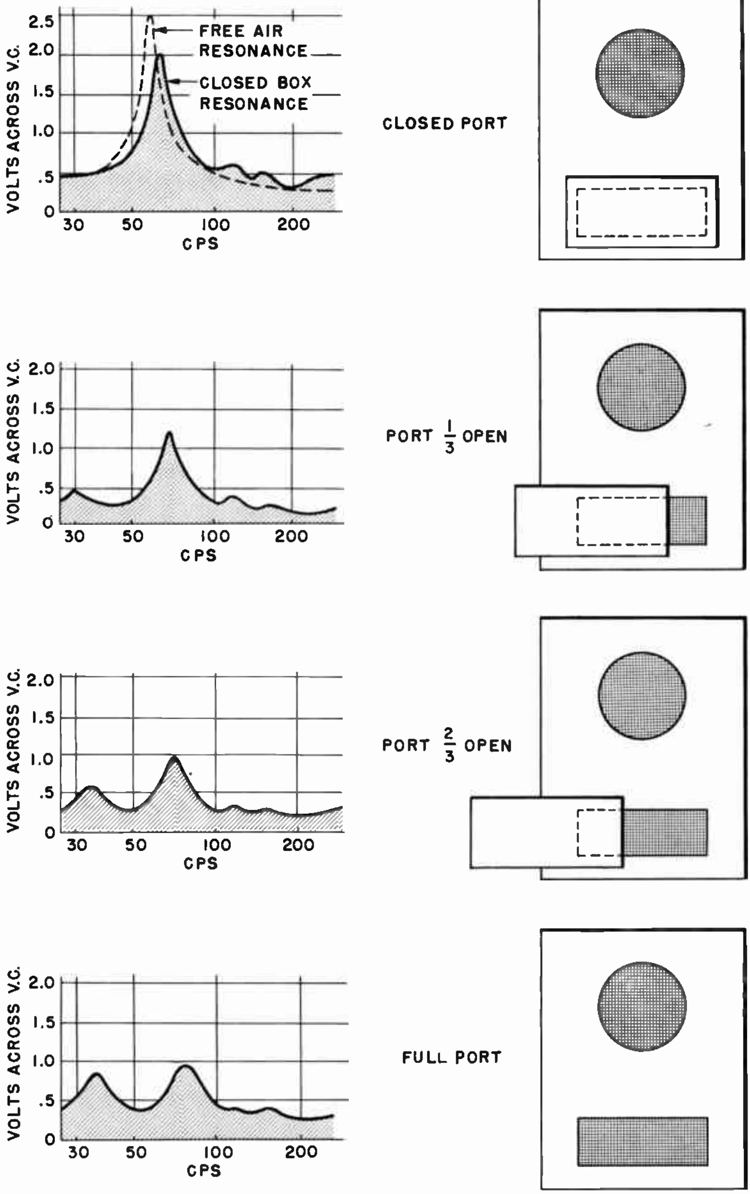


Fig. 9-12. Experimentally determined impedance curves of a bass-reflex enclosure as its port area is changed from zero to full port. (Milton S. Snitzer, "Adventures With a Bass Reflex," Audio Engineering, *January 1954.)*

The number of layers of resistive cloth to be stretched over the port is determined by the following test. Connect one side of the dry cell to one of the voice coil terminals, then make provisions so that the other terminal of the loudspeaker may be readily connected and disconnected from the battery. Since the dry cell has a constant d-c voltage, the application of this fixed voltage to the loudspeaker will constitute a "unit pulse," for once it is applied to the speaker, it persists unchanged until it is removed. As the battery connection of the loudspeaker is completed, the loudspeaker will make a sound that can best be described as a "bong" *(if the system is not properly damped).* When the battery is disconnected, the loudspeaker will make a similar sound. The "bong" represents a ringing condition of the loudspeaker, brought about by the application of a non-ringing unit pulse. The objective of this test is to reduce the sound of the "bong" to that of a "click," for the click is truly representative of a steady pulse signal. If the "bong" continues, additional layers of cloth should be applied to the port opening, one at a time, until the "bong" has been reduced to a "click." When this condition has barely been reached, the system is critically damped.

Application of more material than necessary does not constitute adding more of a "good thing." Too much damping material will overdamp the system, causing it to operate inefficiently.

In converting the "bong" to a "click," it is important to make sure that the conversion is complete on the *breaking* of the battery circuit as well as the making, for often the "make" portion of the test will arrive at the click condition before the "break." One should not stop, therefore, when the click is heard on the completion of the circuit; he should continue with the damping procedure until the click is also barely heard on the breaking of the circuit. When this damping operation is successfully completed, not only will the system be tuned properly for optimum smooth low frequency response, but it will also be free of undesired transient generation. This damping material in the port circuit should not be confused with damping material used to line the interior of the enclosure. The port damping material is used primarily to provide the proper resistive component in the radiating "circuit" so that the correct ratio of resistance to reactance in this circuit may be obtained. The damping material on the inside of the enclosure is used to prevent standing waves within the enclosure (which would cause the enclosure to deviate from true bass-reflex action in the

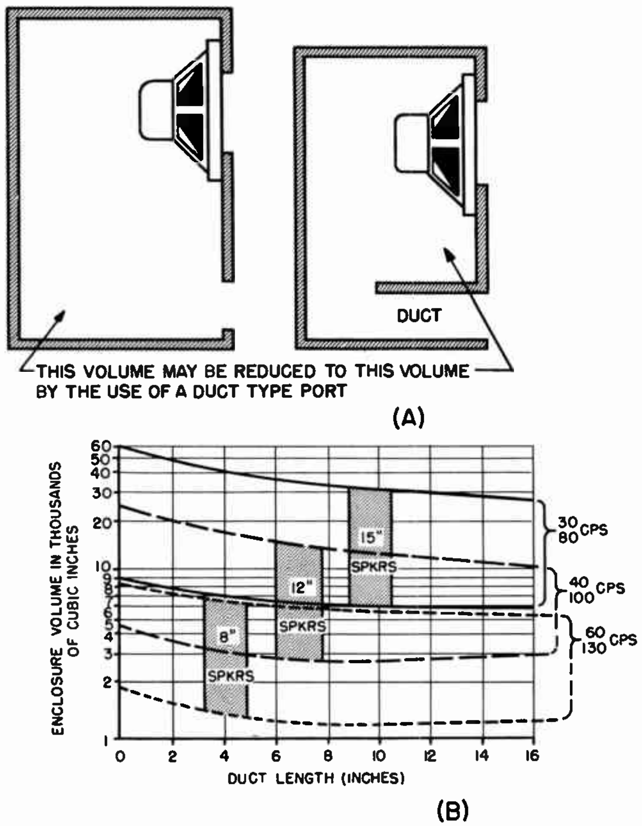


Fig. 9-13. Reduction in size of enclosure that may be realized through use of a duct, which creates a larger port inertance, permitting smaller cavity capacitance.

low frequency end) and to minimize mid-frequency irregularities. These matters will be discussed in Chap. 14.

Enclosure Volume May be Reduced by Venting With a Duct

In developing the concept of acoustic inertance for purposes of explaining the action of the bass-reflex enclosure, mention was made of the inertance effect of a long tube. In the development of the bass-reflex enclosure, we reduced this tubular inertance in size so that it became a mere hole in a partition. This explanation was given purely for illustrative purposes, since most of today's bass-reflex enclosures are constructed with only an opening as a port (for ease in construction, especially by the home hobbyist). It is possible, however, to use a tube or a duct advantageously as the inertance element of the bass-reflex enclosure with beneficial results. The duct, as shown in Fig. 9-13(A), will exhibit considerably more inertance than the port of the same cross sectional dimensions. *Because of this increased inertance, the volume of the cabinet may be reduced to get the same resonance frequency condition.* The chart in Fig. 9-13(B) graphically indicates the reduction in size of the cabinet that may be realized for ducts of various sizes. Thus, where construction costs or problems are not considered of importance and reduced size of enclosure is, the ductal vent enclosure shown will prove beneficial. The length of the duct need not be more than necessary to reduce the cabinet volume to its minimum as shown by the graph. In fact, a length about one-half or three-quarters of that indicated by the graph will be perfectly acceptable, because the longer duct takes up more space in the enclosure, and that space must be made up in other dimensions. It is wise, therefore, to use a duct that is moderately short, but of sufficient length to bring the cabinet volume down to the desired size.

Bass-Reflex Enclosure is Flexible and Adaptable to Many Speaker Types

The bass-reflex enclosure enjoys a wide popularity because of the relative ease with which it may be constructed, its adaptability to a wide variety of speaker types, and the ease with which adjustments may be made to make it a matched system with the speaker. Because of the fact that in this enclosure the speaker faces directly out from the enclosure into the listening area, any type of speaker may be used. A speaker (such as a coaxial or triaxial unit) that must face directly toward the listener must be used with a direct radiator type of enclosure (such as the bass-reflex) in which there is free forward radiation.

Yet the bass-reflex enclosure is not limited to such speakers. If a multispeaker system is desired with separate woofer, midrange, and tweeter units, the woofer may be mounted in the bass-reflex system and the other components in a small auxiliary cabinet on top or on the front face of the enclosure, without in any way upsetting the performance of the bass-reflex system.

Because of the adaptability of this type of enclosure to single wide range speakers or two or three-way multispeaker systems, it lends itself admirably to the program of progressive speaker expansion described in Part 1. One may use the cabinet to house his single wide range speaker for general purpose reproduction, and then (by the addition of the proper crossover network and midrange and tweeter units) convert the system *in the same cabinet* to a multi-speaker system, still utilizing the original wide range speaker, converted by the network to function as a woofer.

CHAPTER 10: *Horn Type Enclosures*

Historical Note

The horn is perhaps the oldest type of baffle in the art of acoustic reproduction. Edison's early phonograph had a short conical horn to "amplify" the sound coming from the mechanically energized diaphragm. Later versions of this instrument sprouted the "morning glory" type of horn; and just before the advent of the electro-acoustic reproducer the console-type acoustic phonographs so well known in the living room and parlor of the 1920's boasted well-designed folded horns for "concert-hall reproduction." This type of horn gave good reproduction for the recorded material with which it had to work, and for the acoustic "tone box" feeding it.

When the day of electro-acoustic reproduction dawned upon us, it was only natural that the horn be used again. The most remembered types were offshoots of the "morning glory" variety, coupled to a large heavy driver unit on the table beside the radio. These driver units were actually overgrown models of telephone receivers. In fact, the unit was commonly referred to as a "loud speaking telephone." From that derivation we were left with the term "loudspeaker" for any device that makes sound loud enough so that the reproducer does not have to be held up to the ear for the sound to be heard. Despite the fact that the horn is of ancient vintage, it is a highly efficient device, acoustically speaking. With the modern studies that have been made of horn theory and design and the improvements in their driver units, the acoustic horn has taken a firm foothold in the high fidelity art.

Horn Provides High Acoustic Efficiency

The acoustic horn is an acoustic transformer, *not an amplifier,* despite the fact that a source of sound appears louder when a horn is applied to it. The reason for the increase in sound output from a driver when it is coupled to a horn is that the horn, through its transformer action, creates a better impedance match between the driver and the

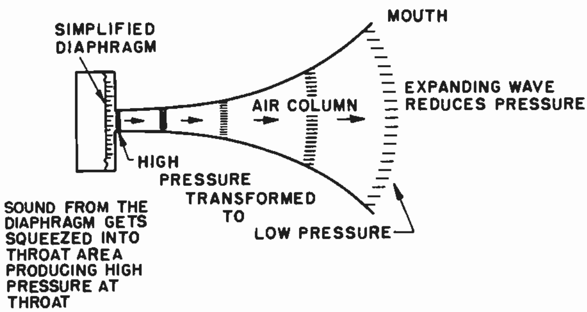


Fig. 10-1. Simplified horn structure illustrating transformation of high pressure at throat to low pressure at mouth.

air. With improved impedance matching of the driver unit to the air comes improved power output and efficiency of operation.

Horns may take many shapes, but they have one characteristic in common. They expand from a narrow opening called the throat to a larger opening called the mouth. (See Fig. 10-1.) The intervening space is called the "air column." When the horn is coupled to a driver unit at its throat, all the sound from the driver unit must travel into the throat area. This means that if the diaphragm of the driver unit located at the throat of the horn is five times larger than the area of the throat, the sound coming from the surface of the diaphragm must be compressed in a five-to-one ratio. As a result, the sound pressure at the throat of the horn is high.

As the sound wave progresses along the horn toward the mouth, it finds its confining walls continually expanding. This allows the high pressure wave to spread out over a larger and larger surface. As it spreads out, the pressure per unit area decreases. Finally, at the mouth of the horn, the sound wave breaks away from the mouth of the horn and is propagated into space. Thus the horn has acted as a transformer;

it has changed the acoustic flow of energy from a high pressure condition at the throat to a low pressure condition at the mouth.

Horn Couples High Throat Impedance to Low Mouth Impedance

This statement should now be interpreted in terms of impedance because it is the impedance into which something works that determines the power that is developed. We know that in electrical transmission systems or electrical coupling systems carrying a given power.

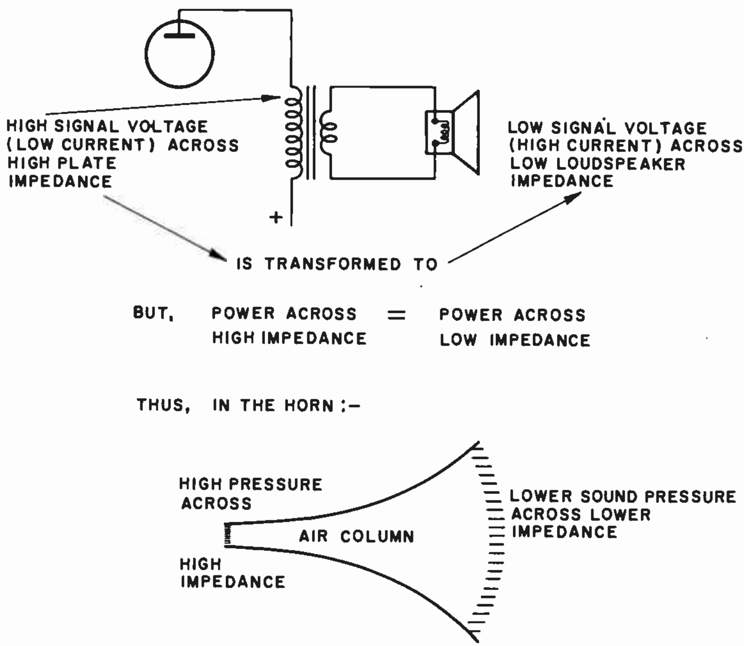


Fig. 10-2. Analogy showing the impedance transformation function of a horn.

if we have high voltage at one portion of the circuit and low voltage at another the *high voltage exists across the high impedance portion of the circuit and the low voltage appears across the low impedance portion.* Take for example the output transformer in Fig. 10-2. The primary side of the transformer, which is connected to the high signal output plate voltage (at low current) is of high impedance, while the secondary, which connects to the loudspeaker, is of low impedance, across which there is low signal voltage (at high current).

Sound pressure is analogous to voltage. Voltage drives the current through the circuit; sound pressure imparts the "volume velocity" to the sound wave. Thus, in an *acoustic transformer,* high sound pressure exists across high acoustic impedance, and low sound pressure exists across low acoustic impedance. What happens in the acoustic horn is that the gradually tapering column of air is a means of matching the relatively low loading impedance of the nebulous outside air to the relatively high mechanical impedance of the comparatively massive vibrating piston. When this impedance match is properly made, the combined driver and horn system may operate at considerably higher efficiency.

Putting it in simple terms, the direct radiator type of loudspeaker used in a simple baffle must "grab hold" of the entire air merely through the air with which the surface of the diaphragm itself comes in direct contact. Because the diaphragm size is always relatively small compared to "all the air," the impedance match is a poor one and the efficiency of the direct radiator system is correspondingly low. In the case of the horn, however, the diaphragm is allowed to contact a "surface" of air much larger than its own area by means of the much larger horn mouth, which is *directly coupled to the diaphragm through the horn.* The larger the mouth of the horn, the better the impedance match between "all the air" and the diaphragm, and the more efficiently the diaphragm can grab hold of the air. As a consequence, the horn system has high efficiency, somewhat in the order of 40 to 50 percent, whereas direct radiator efficiency is in the order of 10 percent or less.

The mouth of the horn may be considered to be an actual vibrating piston by itself. If reference is made to Fig. 9-1, it will readily be seen that by making the mouth of the horn large for a given low frequency a much larger radiation resistance may be presented to the relatively small diaphragm driving the large mouthed horn. This is illustrated in Fig. 10-3.

High System Efficiency Reflects Itself in Cleaner Reproduction

Because of the high order of efficiency of the horn-loaded system, it is possible to obtain much more linear low frequency response than with a direct radiator system driven to produce the same amount of acoustic output. The power output of a vibrating diaphragm is a function of the *size of the diaphragm and its displacement.* The more the diaphragm moves away from its equilibrium position, the more air it is pulsing, and the higher the acoustic output. Thus, if two identical diaphragms were installed, one in a horn and the other in a direct radiator enclosure, the latter diaphragm would have to vibrate much more violently than the one in the horn system to produce the same

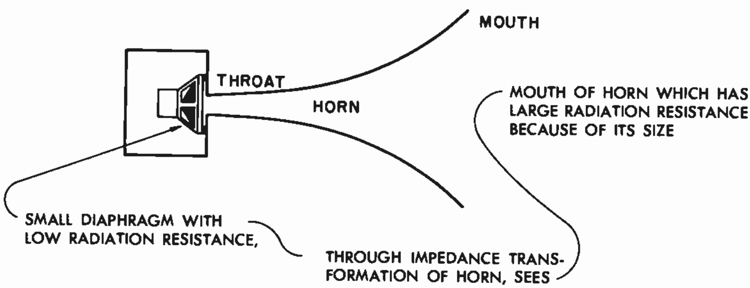


Fig. 10-3. Small diaphragm is better match to "all the air" through the horn as an impedance transformer.

amount of acoustic power output. When a diaphragm vibrates too energetically, it may leave the linear portion of the magnetic field in which its voice coil is balanced, with the result that it will move into areas of magnetic nonlinearity for heavy signals, resulting in distortion. Also, large excursion of the diaphragm may cause the diaphragm to pull the centering spider out beyond its elastic limits and cause further nonlinear distortion of the waveform. It should, of course, be realized that we are here painting a rather dark picture of the direct radiator system as compared to the horn system. Actually, these deleterious effects appear only when very large audio outputs are demanded from the relatively inefficient direct radiator system. Hence, in large auditoriums and theaters, where high sound power with minimum distortion is required, we invariably find horn loaded systems.

Shape of Horn is Determined by Frequency Range Reproduced

The shape of the horn is all important in determining the function and efficiency of the horn. There are horns for low frequency reproduction and horns for high frequency reproduction, and their designs are radically different. Horns for high frequency work, in addition to being small in size, have many forms of compensation built into them for better wide angle dispersion of the high frequencies. Horns for low frequency reproduction are not affected by dispersion problems,

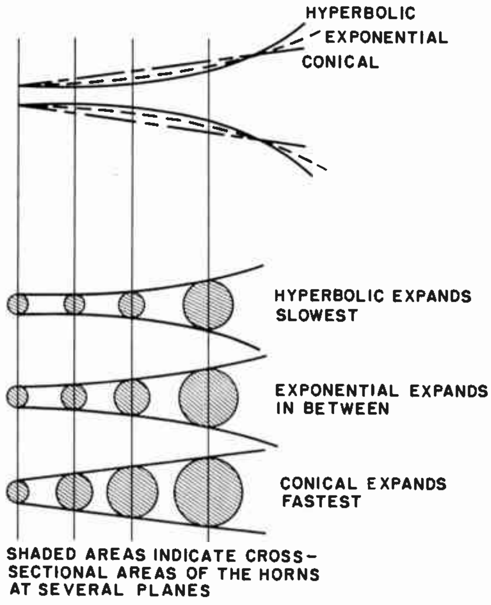


Fig. 10-4. A comparison of the rates of expansion of hyperbolic, exponential, and conical horns all having the same theoretical cutoff frequency.

because low frequencies spread out rather evenly without any auxiliary devices. Secondly, low frequency horns are usually large and massive, high frequency horns small and compact. The difference in their relative sizes is not simply an arbitrary matter. Horn design must employ precise, mathematically determined expansion rates for proper performance in the desired frequency band.

Horns are essentially high-pass filters. They pass a band of frequencies above the cutoff frequency. Once this theoretical cutoff frequency has been chosen, the development of the horn is a straightforward geometrical layout. Horns may flare out rapidly or they may flare out slowly. The rate at which they "grow" is determined by the family of horns to which they belong, and by the low frequency cutoff desired from the horn.

There are several types of horn in use as acoustic reproducing elements. However, despite the fact that they are all high-pass filters, they do not necessarily follow the same laws of geometric expansion (even though they may all have the same theoretical cutoff frequency). The two most prevalent types are the "exponential" and the "hyperbolic" (actually, hyperbolic exponential or "Hypex"). These terms are the more popular designations for them; actually, the mathematician will readily recognize that exponential and hyperbolic devices belong to the same family of curves.

For the layman, however, the two general classifications noted specifically refer to two types of horns, which expand to different degrees for the same cutoff frequencies. The fact that two different types of horn accomplish the same end result, as far as pass characteristics are concerned, leads to the conclusion that the differences in operation are not a matter of frequency. Before we discuss these differences in opration, let us examine the manner in which they differ geometrically.

Comparison of Conical, Exponential, and Hyperbolic Horns

Figure 10-4 illustrates the manner in which the cross-sectional area of the exponential and hyperbolic horns differ for a given increase in the length of the horn, where both horns have the same cutoff frequency. Along with these is shown another horn, the conical, which will come into later discussions of the characteristics of these horns. From this figure it will be observed that, for a given length of horn up to the point where the mouths of the three horns are all equal, the hyperbolic horn expands at the slowest rate, the exponential at a faster rate, and the conical at the most rapid rate. This matter has significance in relation to the level of reproduction at the area very near the theoretical cutoff point, and to the distortions to be expected from various types of horn as well as the amount of space taken up by them.

Figure 10-5 illustrates the acoustic radiation resistance of these three types. It will be noted that the horn with the slowest rise in radiation resistance in the pass band (which also means poorest low frequency response) is the conical. Next comes the exponential, which rises quite sharply and attains the maximum value quite soon. Last is the hyperbolic, which rises immediately at the cutoff frequency to a value of radiation resistance above the optimum and then drops down to the optimum at the same frequency as the exponential type. It is apparent from these curves of radiation resistance that the hyperbolic horn would be the best reproducer of low frequencies close to the cutoff point, the exponential second best, and the conical last. For this reason, the conical is never used as an acoustic reproducing device, except in the form of a simple handheld megaphone, which may be made cheaply by simply rolling up a piece of heavy fibreboard.

What then determines which of the two remaining horns — the hyperbolic or the exponential — is to be used? The fact that they are both used is evidence that there is an honest difference of opinion about

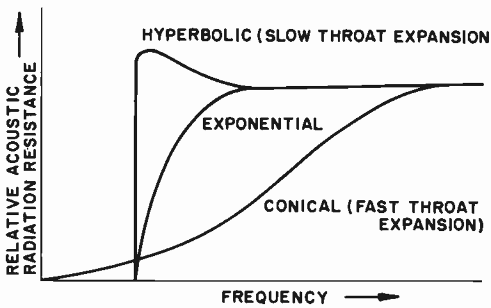


Fig. 10-5. Relative acoustic radiation resistance of conical, exponential, and hyperbolic horns, indicating that the rise of the radiation resistance of the horn varies inversely with the rate at which the throat area expands.

the two. Let us examine a seldom appreciated fact concerning horns. Although it is true that with horn loading a system may be designed to give low overall distortion because of the higher efficiency of the system (which in turn allows the energizing diaphragm to move much more conservatively), there is *distortion introduced by the horn itself*. The horn, because of the manner in which it functions, introduces distortions of its own, depending upon the degree of constriction over the narrow first part of the horn, and the sound pressures that are being forced through this narrow section.

Horns Introduce Distortion at High Pressure Throat Sections

Nonlinear distortion is produced in this first constricted section of the horn because of the unequal expansion and compression of a gas (air) volume for a given change in pressure. A simple analogy illustrating this principle is shown in Fig. 10-6. Here we have shown a simple piston type of air pump, similar to the automobile tire pump once included as standard equipment with every car. In section (A) of the illustration, the piston is shown near the top of the pump cylinder with the air inside the pump near atmospheric pressure. Now let us move the piston back and forth just a very slight amount. Under these conditions the amount of compression of the air within the pump will be small, and little resistance to the in-and-out motion of the piston over this small distance of travel will be presented to the piston by the slightly compressed gas. Now, however, let us move the piston way down, severely compressing the air to a small proportion of its original volume. It will then become progressively harder and harder to push the piston *down* because of the high internal pressure of the compressed

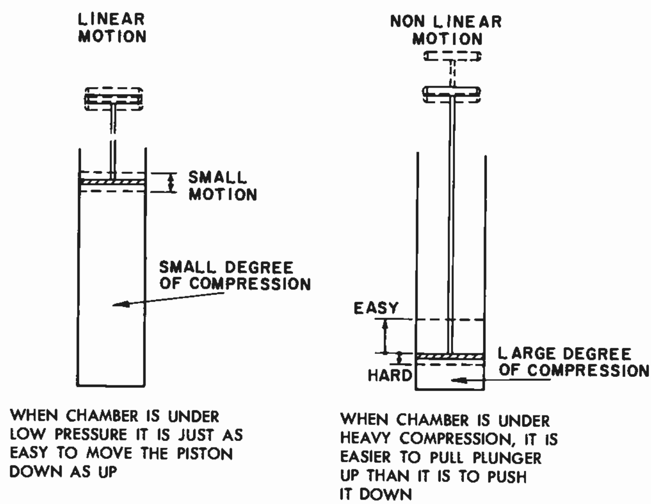


Fig. 10-6. An analogy indicating the non-linearity of piston motion when piston works against a high pressure on one side and a low pressure on the other.

air pushing back on it. On the other hand, it will be relatively easy to pull the piston back *up* because of the comparative absence of pressure on the outside of the piston. Thus there is nonlinearity of motion of the piston in opposite directions for equal applied forces in those directions. It is obvious that the smaller the cavity into which the piston works, the faster will the pressure in that cavity increase for a given motion of the piston, and the greater will be the resultant nonlinearity of motion of the piston.

Now how does this apply to the horn loudspeaker? Examination of the method in which the horns expand (Fig. 10-4) will show that the horn with the largest volume (hence lowest pressure) near the piston location is the conical. The horn with the smallest volume (hence highest pressure) in the throat section is the hyperbolic. Between these two extremes is the exponential. It would be expected that the conical horn would exhibit the least distortion due to built-up pressure, the hyperbolic the most distortion, and the exponential a degree of distortion between the two. It must be noted, of course, that this is distortion contributed *solely by the horn itself* and not by the driver unit.

One must, therefore, make a compromise between horns on the basis of their response *and* their distortions. These air nonlinear distortions become larger with greater pressure in the constricted areas. Thus we may generalize as follows: the tighter the throat section and the higher the pressure in this throat section, the higher will be the resultant horn distortion. Hence, if a horn system is to be working at low level with equivalent low internal horn pressures, we may utilize the tighter horn, the hyperbolic, to get the most efficiency at the low end of the pass band with small horn distortion. If, however, heavy momentary bursts of high level sound (high power) are fed to the horn, the distortion for these bursts will be higher.

Now let us consider the exponential horn under the same conditions. If the system is working at low level, it may not have quite the extreme low end efficiency of the hyperbolic, but its output will have lower distortion. Sudden bursts of low frequency power will be reproduced with considerably less distortion than in the case of the hyperbolic horn. In view of the fact that horns in general are high efficiency units, and that considerable acoustic power is developed within them, it seems that some slight loss of the very extreme lows may be accepted at the advantage of lower overall horn-induced distortion. Consequently, the exponential horn is a happy compromise between efficiency and distortion. However, where distortion is not the determining factor, and considerable low frequency output is demanded from the system, the hyperbolic horn may be used.

In view of the fact that the hyperbolic horn throat section is under greater compression than the exponential type, it becomes necessary that much greater accuracies be adhered to in its construction.

Horns are Aided by Room Corner Placement

There are differences in woofer horns other than the manner in which they expand. There are specific advantages to the placement of an enclosure in the corner of a room. Corner operation is especially beneficial if the enclosure is of the horn type. Consequently, there are corner horns designed to work specifically and only in corners, such as the Klipschorn and the Lee Catenoid horn. These structures are actually folded horns built so that the outer side walls of the horn are the walls of the room. The horn intended only for corner use is not a complete horn when removed from its shipping carton. The side pieces appear to be missing. Actually, the horn is incomplete *as received,* but not as used. In operation, the unit is placed in the corner of the room so that the walls of the room act as the side panels of the unit, thus completing the horn. In the "cornerless" corner horn, as typified in the University "Dean," the unit as unpacked is a complete horn, all side panels being a permanent part of the structure. Thus, even when placed in the center of a wall, the unit acts as a complete horn without the benefit of corner loading. Naturally, when put into the corner, this horn is still complete in itself, but in addition it uses the radiation characteristics of the corner to enhance the low frequency reproduction.

Another difference between the corner-dependent and self-contained horns is that the corner-dependent variety consists of two air columns folded side by side, finally terminating in two mouths lying along the wall areas, while the self-contained type is a single folded air column that terminates in one mouth coupling to the floor and the walls. This matter of coupling to the walls of the room is important in connection with room acoustics. The manner in which these horns work in conjunction with the room will be treated in Part 3.

Horn Design Data

Although the exact relationships by which the actual expansion rates for these various types of horn may be found are beyond the scope of this treatment of the subject, it is easy enough to present design data in tabulated form to guide the man who desires to build his own woofer horn. The table in Fig. 10-7, which will be discussed in more detail shortly, gives the percentage increase in cross-sectional area per unit length for a given cutoff frequency for the exponential horn, and the suggested minimum mouth dimensions for that frequency. It is suggested that the home constructor apply his skill to the type of horn that is easiest to build — the single-air-column folded horn terminating in a single mouth. Construction details for this type of horn are given in the appendix.

Low Frequency Horns Expand Slowly, are Long, and Have Large Mouths

It will be observed from Fig. 10-7 that the lower frequency horns "grow" at a much slower rate than the higher frequency horns. A simple physical explanation may be given for this condition to provide a basis for the understanding of what goes on within the horn structure.

In a very general sense, we may consider the horn somewhat as a guide for the development of the acoustic wave. With this waveguide concept in mind, let us start with an extreme horn such as that shown in Fig. 10-8(A), which has such a sudden "flare" that it is almost a flat baffle with a hole in the center rather than a horn. Perhaps it may seem surprising to find this structure being called a horn. It may be considered to be just that — a horn with an exceedingly rapid rate of expansion — it grows so fast it virtually flattens out before it has had a chance to go anywhere. It may be parenthetically pointed out that this apparent indecision as to whether to call this a baffle or a horn is really no indecision at all. We must recognize the fact that a *horn*

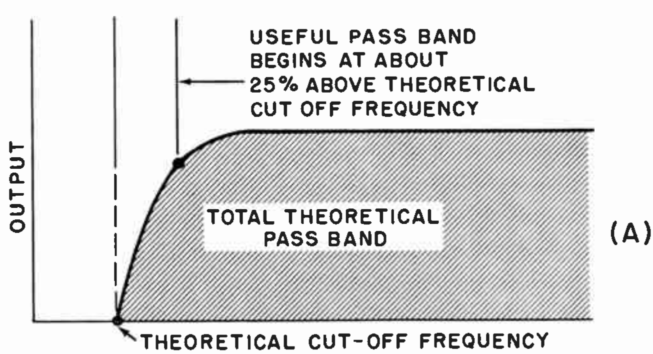
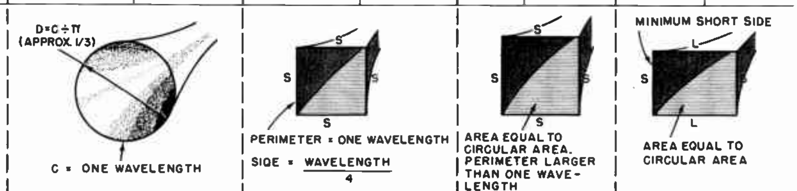


Fig. 10-7A. Response of exponential horn.

A	B	C		D	E	F	G	H	I		J	
					MINIMUM CIRCULAR MOUTH DIMENSIONS		MINIMUM SQUARE MOUTH DIMENSIONS		SQUARE MOUTH DIMENSIONS TO PROVIDE MOUTH AREA EQUAL TO THAT OF CIRCULAR MOUTH (COLUMN F)		RECTANGULAR MOUTH DIMENSIONS	
DESIRED LOW FREQUENCY RESPONSE	THEORETICAL HORN CUT-OFF FREQUENCY BASED ON COLUMN A ÷ 1.25	PERCENT INCREASE IN HORN CROSS-SECTION AREA, BASED ON COLUMN B		WAVELENGTH BASED ON 1129 ÷ (COLUMN A)	DIAMETER BASED ON MOUTH CIRCUMFERENCE OF ONE WAVELENGTH (COLUMN D ÷ π)	AREA BASED ON DIAMETER, (COLUMN E)$^2 \frac{\pi}{4}$	SIDE OF SQUARE BASED ON MOUTH PERIMETER = ONE WAVELENGTH (COLUMN D) ÷ 4	AREA BASED ON SIDE DIMENSION. (COLUMN G)^2			MINIMUM SHORT SIDE BASED ON COLUMN G	MINIMUM LONG SIDE BASED ON AREA OF COLUMN F ÷ COLUMN J
		PER INCH	PER FOOT						SIDE	PERIMETER = 4S		
30 CPS	24 CPS	2.3%	31%	37.6 FT.	12.0 FT.	113 SQ.FT.	9.4 FT.	88.3 SQ.FT.	10.6 FT.	42.4 FT.	9.4 FT.	12.0 FT.
40	32	3.1	44	28.2	9.0	63.7	7.0	49.0	8.0	32.0	7.0	9.1
50	40	3.9	58	22.6	7.2	40.8	5.7	32.5	6.4	25.6	5.7	7.2
60	48	4.7	73	18.8	6.0	28.3	4.7	22.1	5.3	21.2	4.7	6.0
70	56	5.5	89	16.1	5.1	20.5	4.0	16.0	4.5	18.0	4.0	5.1
80	64	6.3	107	14.1	4.5	15.9	3.5	12.2	4.0	16.0	3.5	4.5
90	72	7.1	128	12.5	4.0	12.6	3.1	9.6	3.5	14.0	3.1	4.1
100	80	7.9	149	11.3	3.6	10.2	2.8	7.8	3.2	12.8	2.8	3.7

(B)



DECIMAL PARTS OF FOOT	0.1	0.2	0.3	0.4	0.5	0.6	0.7	0.8	0.9	1.0
APPROX. EQUIVALENT INCHES	$1\frac{3}{16}$	$2\frac{3}{8}$	$3\frac{5}{8}$	$4\frac{13}{16}$	6	$7\frac{3}{16}$	$8\frac{3}{8}$	$9\frac{5}{8}$	$10\frac{13}{16}$	12

Fig. 10-7B. Exponential horn expansion rates and mouth dimensions for various cutoff frequencies.

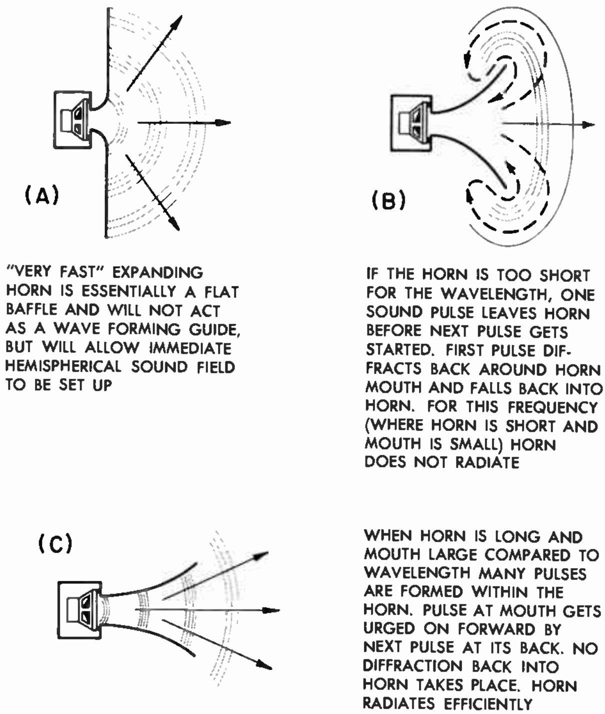


Fig. 10-8. The radiating efficiency of a horn depends upon its length and mouth diameter as compared to the wavelength of the sound.

is a baffle. Once this fact is accepted, much confusion concerning baffles and enclosures will be eliminated.

However, let us get back to our suddenly expanding "flat" horn. How will this flat horn perform as a guide for acoustic energy? Obviously, it is no guide at all. As soon as the wavefront of acoustic energy is thrown off from the diaphragm, it will immediately tend to spread out from the hole (throat) area in a hemispherical pattern that will, generally speaking, be sharpened as the frequency goes up.

Now let us close the horn down so that it looks more like a horn, by reducing its rate of flare (Fig. 10-8B). In this case, as soon as the wavefront leaves the diaphragm and is developed at the throat of this less rapidly expanding horn, it tends to travel down whatever length of horn has been presented to it. If the wavelength of this sound is such that before the next wavefront from the diaphragm reaches the throat of the horn the first pulse has found its freedom on the outside of the horn, this first pulse will begin to diffract back around the edges of the horn and "fall back" into the horn. For *this* frequency then, the horn does not exist. But now, without changing the horn, let us increase the frequency of the sound. Again, the first pulse will start traveling down the horn as shown in Fig. 10-8(C). Now if the frequency is made high enough, a second wavefront pulse will come into the horn before the first has had a chance to get out of its way. There may thus be two (or more) pulses within this same horn for this higher frequency before a pulse reaches the mouth of the horn. The forward pulse will then be forced outward, as it were, by the succeeding pulses (which prevent it from falling back into the horn). A waveguide action has been set up within the horn for this frequency. Thus for this particular horn, with an arbitrary rate of flare, the higher frequencies see the structure as a horn, but low frequencies do not. This is the high-pass filter action of the horn.

Since low frequencies are longer in wavelength, for reasonable waveguide action to be set up within the horn it must be comparatively long, and must expand slowly so that the pulse train may be set up before the horn mouth is reached. Thus low frequency horns are long, and flare slowly, while high frequency horns are short, and flare rapidly. There is one other important requirement of horns in respect to the frequencies they pass. For a circular horn, the mouth diameter must be at least one-third the wavelength of the sound to be transmitted from the horn. This is predictated upon the circumference of the horn being a full wavelength. Regardless of whether the taper is mathematically correct for a particular frequency, if the mouth is too small, that frequency will not be transmitted properly. Thus in low frequency horns we find large mouths, and in high frequency horns we find small mouth. If we think in terms of the horn allowing a set of pulses to be developed, we will see that a quickly flaring short horn with small mouth will transmit only higher frequencies, while a slowly flaring long horn with a large mouth will pass lower frequencies. The exact frequency where the transition between pass and no-pass occurs is the theoretical cutoff point of the horn.

In choosing a horn for a particular application, attention must be given to the chosen cutoff frequency. It must be remembered that every horn has its design cutoff frequency. However, the horn is not

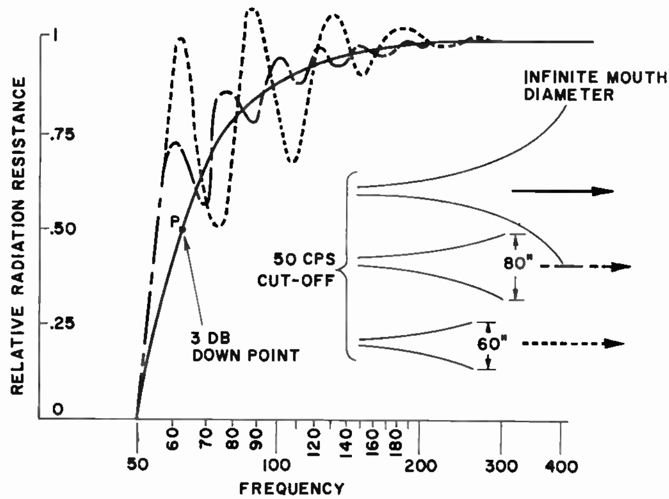


Fig. 10-9. Relative throat impedances of finite horns compared with a perfect (infinite) horn. As the mouth gets smaller, the radiation resistance becomes increasingly more irregular but with some response peaks above that of the perfect horn.

usable down to this frequency, because the output of the horn drops rapidly as the theoretical cutoff point is approached, as has already been shown in Fig. 10-7.

Thus far we have discussed the general differences in shapes of horns in terms of flare, length, and mouth size, and have seen that the low frequency horn is long, with a slow flare and a relatively large mouth compatible in dimension with the wavelength of the low frequency sound. On the other hand, the high frequency horn is short, with a fast flare and a relatively small mouth, also compatible with the transmitted wavelength. There is now the question of the dispersion characteristics that may be expected from different types of horns. This we know to be an important question inasmuch as the proper dispersion of *all* the frequencies throughout the listening room is of vital importance to optimum high fidelity listening.

Low Frequencies are Easily Dispersed

Fortunately, the problem of dispersion of low frequencies is not one that causes us much concern. Low frequencies readily diffract around obstacles and go around corners with relative ease. This means that once a sound wave of low frequency leaves the horn, it spreads out rather evenly. This ease in diffraction for low frequencies also means that the long horn, which is necessary for low frequency reproduction, may be folded, bent, or "reflexed" to reduce the overall size of the structure without affecting the low frequency response. The structures in Figs. 8-2, 8-5, and 8-6 illustrate the degree of bending that may be utilized in low frequency horns.

In the design of low frequency horns, we are somewhat bound by the size of the enclosure. It would be relatively simple to construct a good low frequency horn that would have smoothly uniform level of low frequency reproduction if we could make the horn as long and as big as necessary. The designer or builder of the cabinet for home use must, however, make some compromises. We have seen that the theoretical low end cutoff of the horn is determined by the rate of flare of the horn in conjunction with the mouth diameter. Once the flare rate has been chosen, and the mouth and throat sizes postulated, the length of the horn automatically falls in place. Inasmuch as we could not make a horn perfect in all respects for a given real low frequency because of the monstrous proportions it would attain, let us briefly examine these parameters of flare rate, length, mouth diameter, and throat diameter to see where we may compromise in size and still get reasonably good low frequency performance.

Comparison Between "Perfect" Horn and Finite Horn

Figure 10-9 shows a set of curves depicting the throat radiation resistance (which is a measure of the power developed) of a horn with a taper designed to cut off at 50 cps. The solid line is indicative of a perfect horn; that is, one of infinite length and consequently of infinite mouth size. The pass characteristic of this horn is perfectly smooth. Note that at 50 cps the radiation resistance is zero. This means that *at the design cutoff frequency, the horn will not radiate any energy.*

Superimposed over this theoretically perfect curve are two other curves of horns that have the same theoretical taper, but which have some reasonably finite mouth size. Note that as the mouth size grows smaller, and the length correspondingly shorter, the response curve of the horn becomes more irregular. However, it is important to realize that despite the increasing irregularities of the smaller horns, they have the same zero output at the theoretical cutoff frequency.

Now let us see what these irregularities actually mean in performance. Obviously, we would not want to operate even the perfect horn down to its cutoff point, because at *that* point we would get no output. It is not desirable to operate the horn below the frequency at which its output drops below a level fairly comparable in listening efficiency to the main body of the response. If we accept the proposition that a drop of 3 db from full level response to the low frequency limit of response is acceptable, this will put us at point "P" on the theoretical curve of performance, which represents a throat resistance of one-half normal. We will now use this frequency as a point of reference and compare the two curves of the usable finite horns with the perfect curve of response.

It will be seen that, while the two irregular curves in general oscillate above and below the reference curve, there are greater variations in the curve for the horn with the smaller mouth. This means that a variety of horns may be built, all with the same theoretical low frequency cutoff, and all generally tapering down into that region at the same rate; and that irregularities of response in the low frequency region will be experienced in all as the horn gets smaller (waveguide action is decreasing and mouth becoming too small). However, note that these irregularities are comparatively small in relation to the average impedance value. It will, therefore, be realized, that despite the irregularities of the finite horns, if they are moderate in nature (as in the horn with the larger of the two mouths), smooth low frequency reproduction will be obtained.

Horn Cutoff, Mouth Size, and Throat Considerations

The plan then, in designing a low frequency horn, is to decide on a theoretical cutoff rate of flare somewhat *lower* than the lowest frequency to be reproduced with good output, and then to cut the horn at some point where the mouth diameter (if the horn is circular) or the mouth height (if the horn is rectangular) is compatible with the *actual* low frequency desired, and not the theoretical cutoff low frequency. In Fig. 10-7 we have shown the average compatible mouth dimensions of circular, square, and rectangular horns for given desired low frequency performance.

The table shown in this figure starts with column *A,* which gives the *actual* low frequency performance desired from the horn. This does not mean that this chosen frequency is the theoretical cutoff frequency for which the horn is designed. The cutoff frequency that will determine the rate of expansion of the horn is given in column *B,* which is calculated on the basis that the desired low frequency response is 25 percent higher than the theoretical cutoff frequency, so that the usable response will be fairly linear in the area desired. Thus, considering the horn that has a 40-cps *theoretical cutoff frequency*, the actual low frequency response we might expect (at reasonable comparative levels with the rest of the pass band of the horn), will be about 25 percent higher than 40 cps, or 50 cps. This is, of course, the desired low frequency response as indicated in column *A*.

Column *C* indicates the percent increase in cross-sectional area of the exponential horn for the *theoretical cutoff frequency* of the horn as chosen. Note that this increase in cross-sectional area has been indicated for increments of 1-inch lengths along the horn axis, and also for increments of 1-foot lengths. The reason for this is that, when the horn is bent, it is quite important that the cross-sectional area going around the bend be maintained at a fairly accurate rate, especially if these areas exist near the throat of the horn, where bends are usually found. Therefore, in order to make the horn more theoretically correct, it would be wise to use the expressions for the increase in area per inch when working in small areas, and then use the expansion per foot when working in large areas. Once we have chosen the desired frequency for which the horn is to be designed, it is a simple matter to tabulate the various cross-sectional areas of the horn for increasing distances along the horn. These tabulated areas should be extended out to that length where the horn mouth has the necessary geometrical configuration to provide proper low frequency radiation.

The conditions for good radiation from a circular mouth and a rectangular mouth are a little different, and the chart in Fig. 10-7 includes these variations. The primary characteristic of the mouth of any horn is that its perimeter (the dimension that circumscribes the outside of the horn) be at least equal to one wavelength of the sound to be transmitted. In this connection, the length of the horn determines the mouth configuration, for the longer the horn the larger the mouth.

Column *D* of the chart indicates the various wavelengths for the chosen frequencies from column *A*. Knowing the circumference of the circular horn, we can readily derive from simple geometry the diameter of the circular mouth for radiation of these frequencies. Thus the circumference divided by the value of π gives the diameter of the *minimum mouth* of a circular horn that should be used for the radiation of a particular frequency. This is sometimes arbitrarily referred to as a diameter equal to one-third of the wavelength, and when reference is made to one-third the wavelength, it is usually intended that the mouth of the horn be circular.

If, then, we have determined the diameter of the mouth of the circular horn, the area at this point is readily obtainable, and this area is given in column *F*. These figures in column *F* will then determine the first practical termination of the horn as the tabulated areas of cross section are laid out per increase in length of horn. Thus, in the case again of the 50-cps horn (40-cps theoretical cutoff) the cross-sectional areas for a given length of horn, whether it be in inches or in feet, may be tabulated until we arrive at an area of 38½ square feet at the mouth. The length along the axis of the horn at which this mouth area occurs will determine the *minimum* length of horn for good reproduction at 50 cps.

If the horn we are building is not circular, but square or rectangular, as is more usually the case, the mouth dimensions are somewhat different from those for a circular mouth. The same wavelength determines the periphery of the square horn, and the individual side of this square horn will obviously be the total length (one wavelength) around the mouth of the horn divided by four. This information is given in column *G*, which takes the same wavelengths for the mouth perimeter from column *D* and divides them by four. Corresponding to these mouth dimensions are the areas of the mouth of the horn, when the horn sides are given by the figures of column *G*. The minimum areas for the square horn are then shown in column *H*.

It will be observed, by comparing column *H* with column *F*, that the areas for the square horn are somewhat smaller than the areas for the circular horn; even though both the square and the circular mouth have the same periphery based on the same wavelength, the circular mouth has a greater area. Because of this fact, the square horn will be a little shorter than the circular horn for the same mouth radiation conditions. This means that the square horn will have a little less efficiency then the circular horn, where both have the same perimeter length. However, if we wished to design a square-mouth horn with the same area as the circular horn (which means that the length of the horns would be the same) we would end up with a square mouth whose dimensions are larger than that of one wavelength. Accordingly, in column *I* are given the side dimensions of a square mouth horn, for which it will be noted that the perimeter of the enlarged square mouth is now longer than one wavelength. This is a step in the right direction.

It is common practice to design the horn for rectangular configurations, in which case the smallest dimension should not be any less than the minimum side dimension that was obtained for the smaller of the square-mouth horns. Although the perimeter of the horn determines the radiation characteristics, one cannot design a horn that is extremely narrow in one direction and long in another direction, for then there would be undesirable diffraction from the slit mouth thus formed, with resultant poor radiation efficiency. It is therefore desirable to keep the minimum dimensions of the rectangular horn at least equal to that of the smaller square-mouth horn. Correspondingly, column *J* of the chart in Fig 10-7 indicates the minimum short side based on the same factors that determine the side length of the square horn (as in column *G*). And, along with these minimum sides for the rectangular horn are given the minimum length of the longest side of the rectangle in order to achieve an area of mouth equal to that of the circular horn. Either the square horn of Column *I*, or the rectangular horn of column *J*, will give results comparable to the circular horn of the same mouth area.

Although these minimum dimensions are given to determine the smallest horn that will function reasonably well, they do not represent an upper limit. It is, of course, possible to enlarge the horn as far as one's physical space allows. The larger the horn the more efficient it is, especially in the lower frequency regions.

The remaining factor to be determined for the horn design is the throat size of the horn. The proper throat size for the horn is determined by a relationship of several factors in the driver design not commonly known to the hobbyist. Among these are the flux density in the gap, the length of wire in the voice coil, and the mass of the moving system. These factors are frequently of direct value only to the designer of the loudspeaker, and are not usually published. Their use in making the necessary throat area calculation is a somewhat involved process, inasmuch as it entails making a well considered compromise between optimum power desired from the system and the upper frequency at which the horn is to be used. It is recommended that for maximum efficiency of the system over the range in which it is to operate, the ratio of the area of the diaphragm to that of the throat be approximately 2:1. In other words, the diaphragm area should be twice as large as the throat area. However, it should be remembered that, regardless of the throat size, the mouth size must still be compatible with the lowest wavelength to be radiated. With throat size selected, all the important parameters of the horn have been established and the horn dimensions may be laid out.

The horn may then be folded or bent into sections that lie adjacent to one another, or within one another, in order to conserve space. Care should be exercised in making these modifications so that reasonable accuracy of cross-sectional area be maintained as one goes around the bend. Also, very sharp irregularities should be avoided. Inaccuracies of cross-sectional areas and sharp irregularities in the horn will result in erratic response due to impedance reflections at these points of discontinuity. One other precaution that must be exercised in designing the reflex type of horn is that the acoustic area of cross-section of any section of the horn should be only that area which passes the sound. Thus, if one section is placed within another, and area from the larger one is used up by the smaller section that goes through it, then the *total* area of the larger section will be made up of the theoretically necessary acoustically open area for sound to flow through, *plus* the area taken up by the smaller section running through it. This is an obvious but sometimes necessary reminder for the home constructor.

Cone Speaker Energizing Horn Becomes a "Compression Driver"

Horn reproducers of the above type intended for low frequency reproduction in the home are invariably energized by cone type loudspeakers, rather than the self-contained compression driver unit. The reason for this is that only with the larger varieties of cone type speakers is it possible to get the low resonances needed to energize the low frequency horn. However, despite the fact that a cone type speaker is used, in its application in the horn it becomes a compression type driver unit, because the back of the speaker is completely sealed off from the air. The entire system then constitutes a "compression driven horn-loaded system," even though it is energized by a type of speaker that under "normal" conditions is referred to as a direct radiator.

The choice of the loudspeaker must be made on the basis of compatibility with the horn it is to drive. The resonance of the loudspeaker must bear some definite relationship to the design goal of the horn. To take an extreme illustration, let us suppose that the horn is designed to be operated down to 50 cps, but the resonance of the loudspeaker is 200 cps. Because the output of the loudspeaker drops off rapidly below its resonance point, the horn will not be driven below about 150 cps. Therefore, too high a frequency of resonance of the driver in relation to the horn cutoff frequency results in poor usage of the low frequency capabilities of the horn.

Now let us go to the other extreme, and put the resonance of the loudspeaker considerably below that of the low cutoff of a 50-cps horn, say at 30 cps. Now, although it is true that the entire passable spectrum area of the horn will be energized by the loudspeaker, the area in which the speaker is most efficient (at its resonant point) will be completely out of the range of the horn. Therefore, the horn will not be driven as efficiently as it would if the resonance point of the speaker were *within* the operating range of the horn. We see then that too low a resonance for the driver is also undesirable. Somewhere between these two extremes lies the proper resonance for the cone.

Perhaps the proper resonance point for the loudspeaker driving the woofer horn is a frequency somewhat above the theoretical cutoff frequency of the horn (the point 3 to 5 db down from the flat portion of the horn characteristic), and close to the actual lowest frequency to be radiated, as determined by the mouth size diameter. The choice of resonant frequency on this basis will ensure that the driver will be working *most efficiently in the low frequency range where most is desired from the system.*

Compression Chamber Improves Linearity of Cone Excursion

There is also another important factor to be considered in the horn design, and that is the size of the compression chamber in back of the driver unit. The rear chamber actually performs two functions. It preserves the linearity of motion of the moving system and provides a load for the speaker to work against. Let us examine the matter of linearity. Linearity exists in any system if that system is as free to move in one direction as it is to move in the opposite direction. In the case of the horn-loaded system, the long horn actually places an air load (the air column) upon the front face of the diaphragm. This air load has actual mass, so that the diaphragm, in trying to move forward, must work against a fairly appreciable air mass load. Now let us assume that the back of the speaker unit is completely open to free space, as shown in Fig. 10-10(A). When the diaphragm now reverses itself and moves backward, it will not find the "heavy" air column loading it down but will instead find a very compliable open space. It will therefore move backward much more easily than forward. For the same driving force, consequently, the diaphragm will tend to move more in the rear direction than it will in the forward direction. When this occurs, there is an undesirable nonlinearity of response.

Obviously, the way to overcome this imbalance is to restrict the rearward motion of the diaphragm. This may be accomplished by enclosing the diaphragm within a stiff air cushion, as in Fig. 10-10(B), so that the acoustic stiffness applied to its back will restrict its backward motion to the same extent that the air mass load in front restricts its forward motion. When this has been properly accomplished, most of the nonlinearity due to the uneven loading on both sides of the diaphragm will be removed.

Compression Also Improves Power Conversion

The second effect of this rear compression chamber is that it presents to the diaphragm a much better resistive load. Figure 10-11 shows the acoustic resistance and the acoustic reactance (inertance) char-

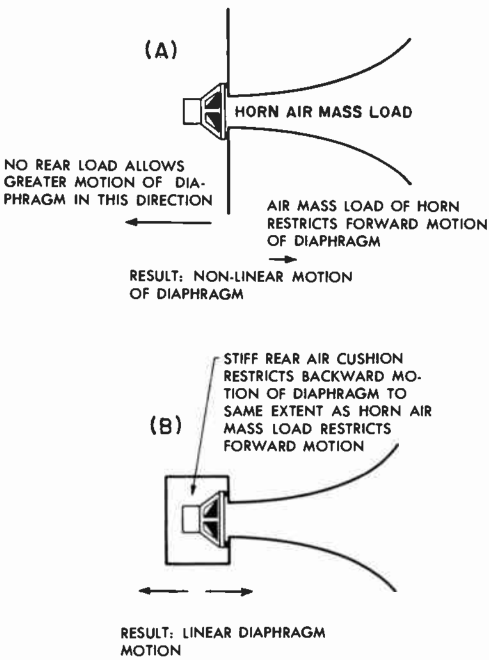
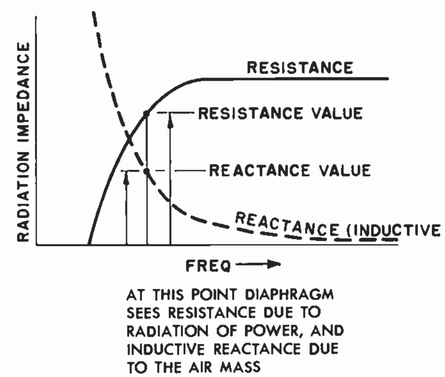


Fig. 10-10. One effect of the compression chamber is to produce linearity of diaphragm motion by balancing horn air mass load with compression chamber stiffness.

acteristic of a horn operating in its pass band. Note that as the resistance factor begins to fall off the reactance factor rises. Although sound pressure may be developed across a reactance as well as across a resistance, no useful acoustic *power* will be developed in the reactance any more than useful electrical power can be developed in a choke coil. However, just as a choke coil, even though it doesn't absorb power, does produce a drop in voltage in a circuit, so the reactance element of the loudspeaker, though not developing power, does reduce the acoustic pressure available for *useful power production across the resistive component of the horn characteristic.* If we can balance out this reactive element, more *usable* power will be obtained from the system. Fortunately, the acoustic stiffness in the air chamber also has reactance, but it is opposite in effect to the "mass" reactance of the air load. This is analagous to capacitive reactance, the effects of which are opposite to those produced by inductive reactance in electrical circuits. The stiffness and the mass balance each other out so that the overall transmission characteristic of the horn is more nearly resistive, with the result that more useful acoustic power may be developed by the driver-horn combination. For the information of the home constructor, the size of this rear air chamber should be equal to approxi-



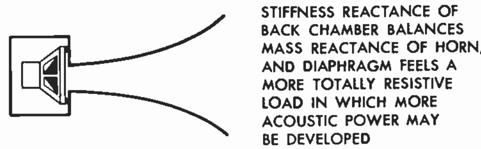


Fig. 10-11. Another effect of the compression chamber is to make the load seen by the diaphragm more resistive by balancing the air mass inertance with the chamber stiffness reactance.

mately three times the throat area of the horn times the length of a section of the horn in which the area doubles; thus volume = 3 × (throat area) × (horn length for area to double).

With the parameters of horn throat, flare, and mouth size established, the woofer horn may be built. The project of building a compression driver horn-loaded system is an ambitious one and should be undertaken with the intention that it will be both acoustically correct and of the best workmanship possible. The success of a horn depends even more upon the quality of the workmanship than does that of a more simple enclosure-type system such as the bass-reflex cabinet. The importance of the manner in which these systems are constructed will be dealt with in detail in Chap. 14.

Advantages of Horn-Loaded Systems

The advantages of horn-loaded systems lie in the fact that it is possible in such systems to obtain relatively distortion-free output at the low frequencies because of the small motions of the diaphragm even when large amounts of acoustic power are realized. Secondly, the high efficiency of the horn-loaded system means, of course, that for a given power output the system does not have to be driven as hard electrically as the direct radiator baffle. This naturally results in more conservative use of amplifier power with consequently reduced amplifier distortion and better linearity of response during peak bursts of power.

Restrictions Imposed by Woofer Horn Enclosures

The disadvantages of home construction of the horn enclosure are the difficulty of construction and the fact that it cannot be adjusted after it is built to match it to its driver. Furthermore, in true horn systems of the compression type, separate speakers are necessary to cover the entire audible spectrum. The compression driver horn system itself is suitable only for woofer use because of its necessarily folded configuration, which prevents higher frequencies from negotiating the tortuous low frequency path. The fact that separate units are necessary for a horn installation is not necessarily a disadvantage, of course, for we know that such multi-speaker systems may be very fine systems. The disadvantageous aspect is in the *necessity* of using such systems where one's budget will only permit starting with a single wide range speaker, or where one has decided to use a coaxial or triaxial assembly. Because of the wide range features of these speakers, it is necessary for them to be mounted in enclosures from which direct radiation is possible.

Horns are Not Limited to Woofer Applications

Horn type baffles are, of course, not limited to woofer application. In fact, they are perhaps more popular as high frequency baffles, and when so used may differ somewhat from the low frequency type. In *basic d*esign they are, however, identical; that is, they expand at some very definite rate determined by the theoretical cutoff required for the horn. Also, the mouth of the horn must bear some definite relationship to the lowest frequency to be transmitted. However, in one important aspect the high frequency horns do differ radically from the low frequency horns. This is in the variety of means used to control the diffusion of the high frequencies as they emerge from the horn so that they will be spread out over the room.

Every loudspeaker has a directional radiation characteristic, and the horn type is no exception. A round horn projector allows the low frequencies to spread out and beams the high frequencies, as does a cone speaker. However, with the horn several factors may be manipulated to obtain control over directivity. The manner in which these manipulations are made is discussed in Chap. 3, which deals specifically with tweeters and the horns that match them. It is seldom that the home constructor builds his own tweeter horn; the specialized driver unit invariably comes with its own horn attached. The horn is of the design recommended by that manufacturer to match the driver unit and the auxiliary equipment with which it is to work. However, one should not lose sight of the fact that the high frequency horn is just as true a baffle as the bass-reflex or any other kind of enclosure, but with the added features of *control* of the dispersion of the sound coming from the horn.

CHAPTER 11: *The Combined Enclosure-Horn System*

Enclosures and Horns May be Combined

The previous two chapters have discussed in detail the manner of operation of the pure enclosure type baffle and the pure horn type baffle, and it was seen that there were specific advantages and disadvantages to both types. There are many designs of loudspeaker enclosures that attempt to combine the features of both these systems into a single structure. This chapter will deal with the most prevalent examples of these combinations.

Horn Loading of Bass-Reflex Port Improves Port Radiation

The most straightforward (and the newest) variation of the bass-reflex enclosure is that in which the port itself is loaded by a small section of horn, as illustrated in Fig. 8-3(C). The addition of this "port horn" accomplishes two beneficial results that improve the overall performance of the bass-reflex enclosure. From the chapter on the enclosure type baffle, the reader will recall that specific mention was made of the effect of the introduction of a duct type port. Because this duct introduced more inertance into the resonant circuit, it was thereby possible to make the enclosure volume smaller and still maintain the same resonant condition. This situation is, of course, highly desirable where it is the objective to make a cabinet of unobtrusive size. Inspection of the structure shown in Fig. 8-3(C) will reveal that the horn-loaded port is in essence an extension of this principle. Thus it is apparent that the horn-loaded port enables the formulation of a structure of smaller proportions for equivalent low frequency output.

Besides this function of shrinking cabinet size, the port horn plays an important acoustical part in improving the low frequency radiation that emanates from the port. We have seen from the chart of radiation resistance (Fig. 9-1) that the radiation resistance of a piston for a particular frequency goes up as the size of the piston increases. This characteristic does not necessarily apply only to a physically solid piston. Actually, the mouth of a horn may be considered as a piston, for it is the virtual source of the pulsations of sound. We might call any opening from which sound emerges an infinitely thin and infinitely light membrane, and we may apply to it in all theoretical justice the same laws of radiation physics that apply to an actual physical diaphragm. Thus, the larger we can make the opening from which the sound is to emerge, the better its radiation characteristic will be for a given frequency, especially in the low end.

This is the second advantage of the port horn. It provides better radiation efficiency for the low frequencies coming out of its comparatively large mouth than does the simple port. It is recognized that the horn in this section seems almost too short to be theoretically efficient. However, as often happens, shortcomings in theory may turn out to be blessings in disguise, if properly utilized. Figure 11-1 shows the radiation characteristic of a very short horn (in relation to the theoretical cutoff wavelength of the horn). Because of this shortness and the corresponding comparatively small size of the mouth (even though it is larger than the original port), the radiation characteristic of the horn is quite ragged, as illustrated. If we were to try to utilize this horn for *broad band* reproduction, it would be entirely useless because of its ragged and irregular nature. However, we are not interested in broad band response from the port horn. We are interested primarily in its output at one very narrow band of frequencies, at the lower resonant peak of the impedance characteristic of the system. Consequently, we can design this port horn short in length in relation to this low frequency, thereby getting a peaked radiation from the short horn just where we want to reinforce the radiation characteristic. We have, as it were, turned the tables on the nonresonant horn, and made it resonant for the specific purpose of improving the low frequency radiation over a restricted range, and from an otherwise smaller opening.

Port Improved When Placed in Corner

This sort of horn-loading of the port may be considerably improved if the enclosure is placed so that it is in the corner of the room, as illustrated in Fig. 11-2, Such placement will improve the overall efficiency of the system for low frequencies and provide an actual

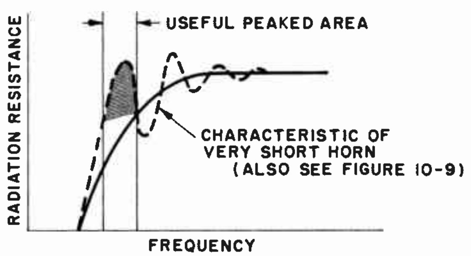


Fig. 11-1. Definite peaked characteristic of short horn and small mouth compared to wavelength make the horn useful as a high efficiency radiator over a small desired frequency range as in the horn loading of a port of a bass-reflex enclosure.

extension of the port horn by the corner of the room to make the radiation from the port horn even more effective. A more thorough discussion of this placement problem will be given in Part 3, but for the present it is sufficient to say that, when properly designed and properly used, the horn-loaded port may greatly improve the performance of the bass-reflex enclosure. The overall characteristic of this system is a high level of bass performance within moderately small structures with speakers of the direct radiator type that must face directly into the listening space, and for high efficiency woofer enclosures in which may be mounted auxiliary tweeter components.

Front Horn on Bass Reflex Improves Middle Frequencies

The variation in the combination type enclosure system opposite to the one just described is shown in Fig. 8-3(B). In this case, the rear section of the enclosure is an altered bass-reflex unit, while the forward part of the speaker is horn-loaded. Although diagrammatically the horn applied to the front of the speaker may appear to be a short one, it is actually one of fairly decent size *acoustically*. A horn should not be measured in absolute dimensions, but should be considered *in relation to the frequencies it is to transmit.* Thus, although a tweeter

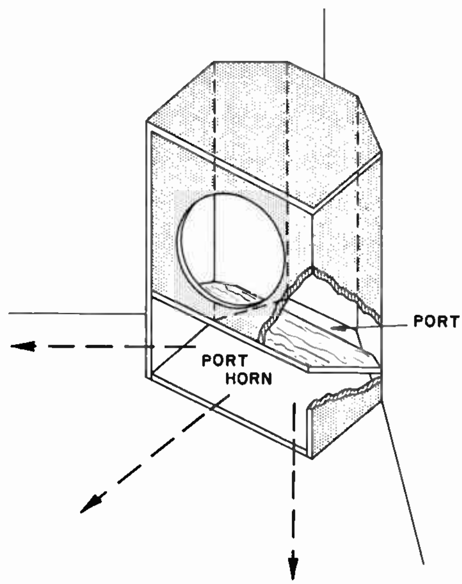


Fig. 11-2. Corner placement for port horn makes horn more effectively loaded. (After University)

horn may be small in physical size, in relationship to the actual very short wavelengths of the high frequencies involved, the size of the tweeter horn is considerable. (For example, the wavelength of a 13,500 cps sound signal is only about an inch.) We must, therefore, consider the presently indicated horn in reference to its intended pass band of frequencies.

On this basis, we see that a horn 10 inches or more in length would represent sizable dimensions for frequencies of approximately 700 cps and up. Seven hundred cps is equivalent to a wavelength of 20 inches; and the half wavelength horn of 10 inches in length is long enough to support the formation of the wave train down this horn for that frequency and all higher frequencies. By thus loading the forward face of the direct radiator with a horn whose dimensions

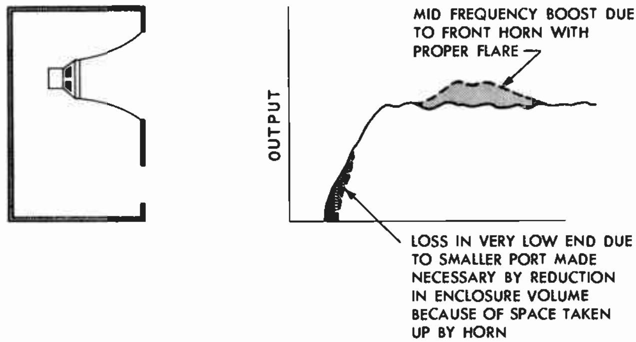


Fig. 11-3. Providing a front midrange horn makes it possible to tilt the performance curve of a given size of enclosure to enhance the midrange "presence" of the system where necessary.

are comparable to the mid-frequency wavelengths, we increase the middle range efficiency of the system.

As far as low frequencies are concerned, the system acts as a conventional bass-reflex enclosure with the rear of the speaker facing directly into the resonant cavity of the system. For proper low frequency response from the bass-reflex section, the rear of the speaker must utilize the same effective volume as if there were no front horn attached to it. Consequently, the overall size of the enclosure must contain the volume originally required for the desired bass response from the speaker, plus the space taken up by the front horn.

Inasmuch as the front horn adds to the midrange efficiency of reproduction of the speaker, it will be found that this sort of structure is of value where the particular speaker to be used has insufficient middles, or where more middles are desired for increasing the "presence" of the system. It is, therefore, possible to "tilt" the performance curve of the speaker (as shown in Fig. 11-3) by maintaining the same overall volume for the structure as for an ordinary bass-reflex enclosure, and using some of that space for the midrange horn. The reduction of the bass-reflex volume necessitates a smaller port with decreased port radiation efficiency. This results in a small drop of the low frequency response, while the horn raises the mid-frequency response. The characteristic performance is thus made more "projective" with a slight loss at the low end. If, however, space is not a factor to be considered, the bass-reflex section of the cabinet may be enlarged to the optimum volume and port size necessary to give it the proper bass reproduction, and overall performance of the system will then be improved.

It will be apparent that this system is, of course, adaptable to direct radiator loudspeakers. However, there may be some loss of high frequencies at the extreme angles because of the shading effect of the sides of the front horn on the wide angle radiation from the tweeter units that are integral with the recessed main speaker.

Rear Horn on Direct Radiator Provides Additional Low-Frequency Loading

In the next variation of the combination type of enclosure the loudspeaker performs as a simple direct radiator from its forward side with nothing interposed between it and its audience, but the rear of the speaker works into a full low-frequency horn. In this structure, as shown in Fig. 8-3(A), the entire volume of the cabinet is used for the woofer horn, which must be folded to be contained within reaonable proportions. It is obvious that the intent of this type of enclosure is to obtain some of the benefits of woofer horn-loading and retain direct radiation action from the forward side.

These benefits of direct radiator action are completely realizable from the illustrated structure, but there are some compromises in woofer horn performance in comparison with the pure horn system. Without the compression chamber, we have no way of compensating the speaker for the relatively heavy air load mass that the horn focuses onto the rear of the diaphragm. Thus, the diaphragm is working into a light load on its front side and a relatively heavy load on its rear side. (See previous chapter.) Because of this, undesirable nonlinearity may be introduced into the low frequency reproduction. Furthermore, the absence of the "stiffness reactance" (which the compression chamber would normally ensure) means that the horn "mass reactance" at the low frequency is not balanced out, and therefore, optimum utilization of acoustic pressure is not made by this open horn in the low frequency regions.

However, despite the shortcomings of this rear horn, it does provide good low frequency response with reasonably good efficiency. Note that in this sort of construction there is actually an acoustical crossover between the woofer horn and the speaker that allows only the low frequencies to be transmitted from the horn and allows the middles and high frequencies to radiate from the front of the speaker (in addition to the lows). This acoustical crossover action is an important part of the characteristic performance of these combination systems, and is treated in detail in Chap. 13.

Acoustic Labyrinth is Tuned Rear Column

A combination type of enclosure, which enjoys some popularity, is the folded tuned column enclosure or Acoustical Labyrinth[1] as illustrated in Figs. 8-3(D) and 11-4. Althongh it may look like the rear horn-loaded structure, the rear section is not a horn but a long tube of fairly uniform cross-section. The term labyrinth arises from the fact that the tube, being folded to preserve space, presents the appearance of a maze or labyrinth. Its convolute twisted appearance, however, has no bearing upon the manner in which it works. Essentially, the purpose of the tubular column that loads the back of the speaker is to present to the speaker a definitely tuned "pipe" much like an organ pipe.

All tubes are resonant devices. The frequency of their resonance depends upon their length. Blow across the neck of a bottle and it will respond with a certain tone. Partly fill the bottle with water and the pitch will rise because the air column has been shortened. The trombonist lowers the pitch of his instrument by pulling out the slide and making the instrument longer, and the trumpeter changes the pitch of his instrument by closing off different sections of the length of his instrument with the valves. In the pipe organ the different tones are elicited from tubes of different length corresponding to the frequency desired. Thus, if we want to reinforce a particular frequency, we must couple to the source of sound a pipe cut to the right length to correspond to this frequency.

Tuned Column Impedance Varies with Frequency

This is essentially the action of the folded tube in the acoustic labyrinth. The length of the resonating tube is chosen to reduce the resonant frequency impedance peak of the speaker and reinforce other frequencies close to an octave below and above the resonant frequency of the speaker. This is similar in effect to the bass-reflex action. Figure 11-4 illustrates how this action is accomplished. If the length of the tube is such that it corresponds to a half wavelength of the sound transmitted through it, at the instant the sound appears at the open end of the tube, it is already a half wavelength behind the speaker pulse of that instant. However, since the back of the cone radiation started out-of-phase with the forward radiation of the cone, the net result

[1]Stromberg-Carlson trade-mark.

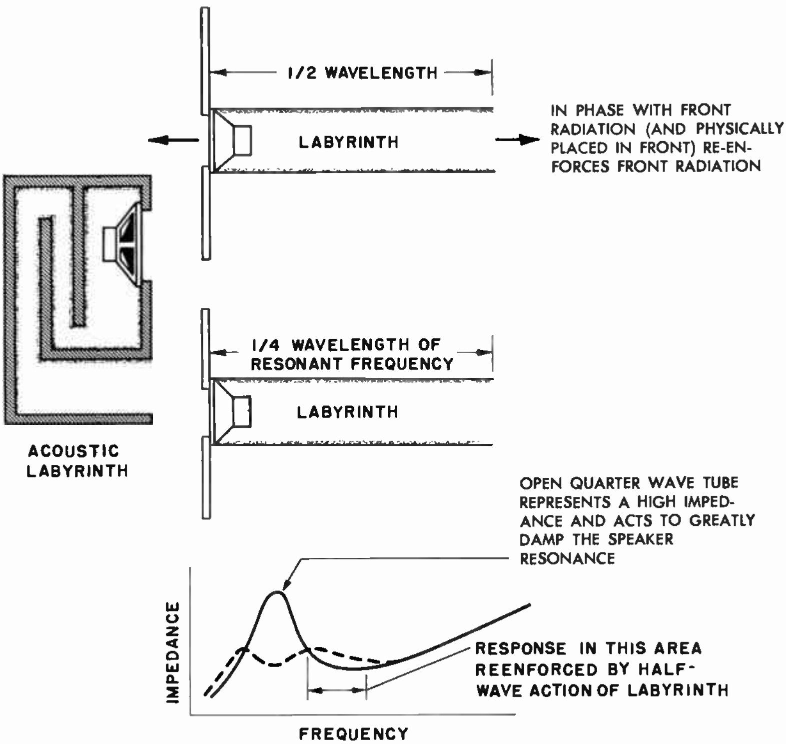


Fig. 11-4. The acoustic labyrinth. Rear tube highly damps the resonant peak of speaker, re-enforces area above resonance, producing smooth low frequency response. (After Stromberg-Carlson)

is that the sound coming out of the end of the tube is now completely in phase with the front and reinforcement of the front radiation takes place. Thus, at the particular frequency where the tube is one-half wavelength long the effect of the tube is to cause reinforcement.

Now, as we go down in frequency (longer wavelength), we come to a frequency (an octave lower) where the tube is only one-quarter the length of the sound wavelength. When this happens, the open one-quarter wavelength tube presents a high impedance to the diaphragm, and the motion of the diaphragm becomes severely damped. If this "quarter wavelength" is made to correspond to the resonance frequency of the speaker, the tube acts as an anti-resonant device at the resonance of the speaker. Thus the acoustic labyrinth damps the resonant peak of the speaker while reinforcing those frequencies an octave above the resonant frequency. The fact that the tube is tuned to the resonant frequency and puts a severe damping restraint upon the diaphragm motion produces a severe depression in the resonance characteristic of the speaker directly at the point of its maximum value. This leaves two humps of response on either side of the frequency where the anti-resonance effect of the tube is most effective. The end result is a resonance characteristic as shown in Fig. 11-4 that looks very similar to the curve obtained from a bass-reflex structure. The effect of the tuned tube of the labyrinth is then to provide good damping characteristics and extended flat low frequency response. Due to the large cross-section of the mouth of the tube, a greater value of radiation resistance is seen by the labyrinth, reflected in improved damping of the system.

One feature of the acoustic labyrinth is that its resonant tube is lined with highly absorbent material, which adds a resistive characteristic to the selectivity of the tube, causing smoother resonant and anti-resonant action than if the tube were not so treated. The sound absorbing material within the tube also serves to decrease the high frequency transmission of sound through it.

The acoustic labyrinth structure is obviously suitable for use with loudspeakers that are required to act as direct radiators from the front, and where, at the same time, bass reinforcement is required at the rear of the speaker. An important factor that must be considered, as far as such a system is concerned, is that in the normal course of usage, and over a period of time, the self-resonant point of a speaker may gradually go down because of the gradual loosening of the suspension of the speaker. When this happens, the labyrinth goes out of tune with the speaker and its effectiveness is somewhat decreased (unless this factor was originally taken into account).

This same action is, of course, true of all speaker systems to a varying degree. In the bass-reflex enclosure for instance, the cabinet is also tuned to the resonance of the speaker, and should the resonance change with constant use the cabinet adjustment will also suffer. However, in this case, the reflex action covers quite a broad band at the low end and the detuning effect is not seriously notable. In any event, one may readily change the port area to compensate for this resonance change.

In the case of the horn loudspeaker system, changes in the resonant frequency of the speaker may also result in a shift in the low frequency performance because of the point upon the horn transmission characteristic where the speaker is placed. If the speaker resonance drops too far, its point of maximum efficiency has left the most favorable area of the radiation characteristic of the horn and the system suffers.

The Combination Enclosure Performance Depends Upon the Combined Features

It is important to point out that these structures, being combinations, may naturally be combined to varying degrees to get different performance characteristics. Thus, in one case we had a front horn and a rear reflex chamber; in another case we had a full rear horn and direct forward radiation, and the choice determined the performance. It meant increased presence with moderate lows in the former case and increased lows with moderate presence in the latter case. It stands to reason that there is an almost infinite variety of such combinations between one extreme and the other. However, whatever the design, it will represent a system in which compromises are made in one area to obtain benefits in other areas. The design must be closely integrated with the particular speaker to be used, for obviously we would not want to increase the presence of a speaker already rich in middles and deficient in lows. Nor would we want to extract the absolute maximum of low frequency output from a speaker by subjecting it to a legitimate horn load, if our system were already unbalanced in favor of the low end. These are the facts that make it difficult to prescribe the "best" of these various combinations.

CHAPTER 12: *Enclosure Size, Speaker Resonance, and System Response*

Enclosure Must Be Adaptable to Size of Listening Area

A requisite for any loudspeaker system is that it be able to fit into the area where the listener intends to use the system. Despite the fact that this requisite *apparently* does not concern itself with performance, and as trite as this statement may seem to be, acceptance of it may clear the air as far as system size *and quality* are concerned. As an example, if the user wants a hi-fi loudspeaker system for his rather small den, in which he has a desk in one corner, a stack of bookshelves along one wall, an easy chair in another corner, and a closet and a window on another wall, there is not much room left for a really massive system. However, he may find that he can weed out a couple of dozen books from one of the shelves, and in their place install a "bookshelf" type enclosure such as those illustrated in Figs. 12-1 and 12-2. Now, many of these smaller enclosures are legitimate hi-fi systems. They are designed around high efficiency speakers; they may be full two- or three-way systems employing well engineered crossover networks; and great pains may have been spent in designing and building the enclosures so that the utmost would be obtained from the speakers these enclosures will accommodate. In many ways, the bookshelf system is a legitimate member of the high fidelity family. It will provide the den in question with remarkably good reproduction, far superior to that of the garden variety of speaker-console assemblies so often found in "big" radios that are frequently simple open boxes with inexpensive speakers.

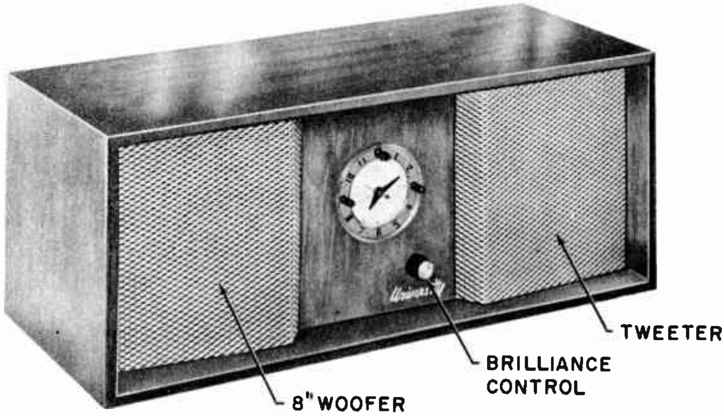


Fig. 12-1. Although the application for an enclosure may limit its size, its components may constitute a true hi-fi system as shown here, using multispeaker elements, crossover network, and brilliance control. (Courtesy University)

However, despite the remarkable performance of such small systems *in this application,* we would certainly be foolhardy to suggest they could therefore be used to provide the proper sound reproduction for a large open reception room. Large spacious rooms have the physical space to accommodate the more elaborate enclosures, such as the folded horn variety, from which considerably superior bass reproduction is possible. Yet, in spite of this, we could not categorically say it is *the* best system. It certainly would not fit the needs of the man with the small den. Thus it is a question not of superiority but of adaptability, both to environment and to the budgetary means of the user. All enclosures have some usefulness, depending upon the circumstances in which they are to be used.

Speaker Size Generally Determines Enclosure Size

There is one basic fact, however, that determines the size of the enclosure, and that is the size of the largest speaker of the system, for there is a definite relationship between speaker size and enclosure volume due to the general correlation between a speaker's size and its resonance frequency. This then is our first concern — the matter of enclosure size and speaker resonance. In Chap. 3 the question of the general performance characteristics of loudspeakers of different sizes was discussed. Figure 3-10 lists some average resonant frequencies of

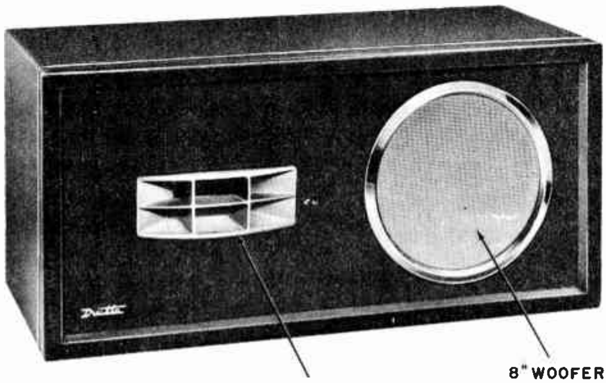


Fig. 12-2. Another form of a "bookshelf" enclosure, utilizing multi-speaker system components to provide a two-way system. (Courtesy Jensen)

the three popular size speakers (8-inch, 12-inch and 15-inch); note that the larger the speaker the lower the resonant frequency. This results, of course, from the generally heavier diaphragms found in these speakers, for resonant frequency is inversely proportional to the mass of the moving system (other factors remaining unchanged). It is therefore, to be expected that the larger speakers will have better low frequency response than the smaller speakers; this condition is illustrated in Fig. 3-1. It will be observed that not only does the frequency extension of the response go lower for the larger speakers, but the level of output is also higher. This will in general be the case if the speakers have magnetic circuits of increasing strength to match the increasing weight of the moving system to be driven.

The larger speakers, being larger in diameter, present a higher order of radiation resistance (Fig. 9-1) for a given low frequency than do the smaller speakers. Therefore, for equivalent driving force, the larger speaker will produce better low frequencies. What happens, for instance, if two loudspeakers of different size have the *same* resonant frequency? If this be the case, the larger speaker will have a higher level of response at this frequency, because of its greater radiation resistance.

Another point of comparison between the low frequency characteristics of two speakers of different size but of the same resonant frequency is cleanness of response. If the same acoustic output is derived from both (that is, if they both sound equally loud) we may be reasonably sure that the large speaker is vibrating less severely than the smaller one. Acoustic power output of any vibrating diaphragm is proportional to its area and to its excursion; that is, the more air is pushed, the higher is the power output. If two pistons of different size are both pumping *equal* quantities of air (equal power output), the larger area piston will naturally have the smaller excursion. If this is the case, we may be reasonably sure that the larger piston is experiencing less nonlinearity in its motion, that diaphragm break-up is reduced (if any existed), and that less amplifier power is required. All in all, then, the larger speaker has a better low frequency response because of its lower resonance capabilities and its greater radiation resistance due to its size.

Enclosure Design Determined by Both High and Low Frequencies

Because larger speakers have better low frequency response, they generally are deficient in high frequency response. Their heavy moving systems preclude the same efficiency of reproduction of the high frequencies that smaller speakers enjoy. The smaller speakers present less of a moving mass to the small amplitude high frequency signal, hence these high frequencies are more efficiently reproduced. In general, the smaller speakers excel in the high frequency range. The enclosure size and design will be limited to a great extent by both low frequency and high frequency performance of the speaker. The *size* will be determined by the low frequency capabilities of the speaker. The *design* will be determined by the high frequency performance of the speaker, depending on whether it is a woofer or a full range direct radiator.

There are two ways of attacking the problem of the speaker-enclosure combination. One method is to put a small speaker into as large an enclosure as possible in an attempt to get "large enclosure performance" from as economical a speaker as possible. The antithesis of this is to squeeze as large a speaker as possible into as small a space as possible in the attempt to get "large speaker performance" from enclosures of reasonable size. In both of these extremes there must be compromises, and somewhere in between is the middle ground of rational design.

Large Enclosure Response Determined by Speaker Size

In discussing baffles in general, it will be recalled that the wall baffle serves as an excellent means of preventing dipole action of the speaker; that is, it completely blocks the rear radiation and thus prevents back-to-front low frequency cancellation. Under these circumstances, whatever low frequencies the speaker is capable of radiating are actually transformed into useful acoustic power. However, the speaker receives no assistance from the wall baffle other than to prevent the dipole action. Once this has been accomplished, the speaker behaves essentially as if it were in free space; its resonance frequency remains virtually unaltered, its peak impedance remains virtually unchanged (except for the free air loading). The wall baffle then is one

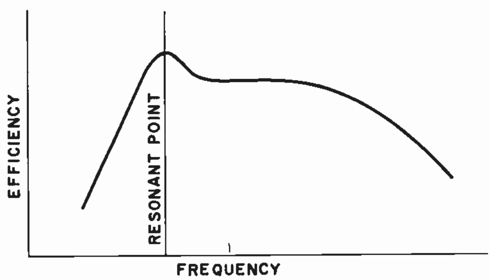


Fig. 12-3. The efficiency of a loudspeaker is highest at its resonant point and then gradually drops off as the frequency rises.

in which speaker operation is completely independent of baffle dimensions. In extremely large enclosures, therefore, differences in the overall speaker systems for different types, sizes, and resonances of speakers are thus wholly dependent upon the speakers themselves, and not upon the combination of the speaker and the wall "enclosure." The same is true of the closet enclosure if the closet is big enough, or even of the closed box type of "infinite" baffle if the box is big enough. We have merely extended our thinking from the wall baffle to actual enclosures. If these are so large that they are essentially "unseen" by the speaker (except for the elimination of the dipole action), total performance will be a function only of the loudspeaker. If we put a 15-inch speaker in the closet door, we get more low frequency reproduction than if we were to put an 8-inch speaker in the same door, and the same difference in speakers will occur if we put them into box enclosures if the volume of the box is large enough.

Small Enclosure Response Determined by Both Enclosure and Speaker

As we decrease the size of the enclosure, however, the reduced volume becomes an important factor in the overall performance of the system. As the volume of the closed box gets smaller, the captured air within the box, against which the diaphragm has to work, exhibits an acoustic stiffness proportional to the size of the diaphragm, the motion of the diaphragm, and the volume of the box. If the diaphragm is large, and has a large degree of motion at the low frequencies, it will tend to compress the air within the box as it moves inward. This compression pushes back upon the diaphragm; thus the air appears "stiff." Since the cabinet and the loudspeaker are connected into one integral acoustic circuit, this stiffness reacts upon the overall resonant condition of the circuit.

Low Resonance Speaker in Small Enclosure

It is now a question of proportioning the enclosure size to the resonance characteristic of the speaker for the desired response. Although it is ordinarily impossible to realize this in practice, let us assume that we have an 8-inch loudspeaker with *zero* resonance. This is a practical impossibility because if the speaker has any substance at all, it must have some weight of diaphragm and some compliance in the suspension of the moving elements; therefore, from the inexorable laws of physics, it must have a specific natural resonance. Let us, therefore, attribute to the speaker some resonance figure below the audible range, say 5 cps. Like any speaker, it is most efficient at its resonant point, and its efficiency drops gradually but constantly as the frequency goes up from its resonant point. This is indicated in Fig. 12-3.

Small Enclosure Raises Speaker Response

If this speaker is now put into a reasonably small box, it will be stiffened in proportion to the decreased volume of the box, and the resonant frequency of the combination will be increased. We can continue to make the box smaller and smaller, continually causing the resonance of the combination to go up in frequency. The point of maximum efficiency will move up with the resonant frequency. At first glance this may seem like a very simple way of getting good low frequency response from a small enclosure with a small speaker. It would seem that all we have to do is to get a small loudspeaker with a very low resonance (below audibility) and put it into a box that will stiffen it to the extent that its point of maximum efficiency will come within the range of usable sound output, say above 50 cps as shown in Fig. 12-4, and we have the panacea for all the problems of small speaker size and large performance. However, it is not quite that simple.

While it is true that the point of maximum efficiency may be moved up from the non-usable frequency to the usable frequency range, it is nevertheless true that the final efficiency may still be far below what might have been obtainable if we started with a speaker of the same size, but of a higher resonance (in the audible and usable area and with correspondingly high initial efficiency in that area). As indicated in Fig. 12-5, the initial efficiency of the higher resonance speaker for the same driving force will be considerably higher than that of the low resonant speaker even after it has been stiffened. This occurs because the overall operating range of the higher resonant speaker is more restricted, and its overall power capabilities are distributed over a narrower range, giving higher overall efficiency in that range.

It will also be realized that if a very low resonant system, such as our theoretical one, were to be constructed with the thought in mind of "creeping up" into the musical spectrum by making the box smaller and smaller, the size of the box would become quite small. Now *too* small a box will not provide the proper diffraction characteristics for these low frequencies. Furthermore, it is not very beneficial to the reduction of internal reflection of low frequencies, because of the close walls. It may become necessary in such a system to utilize a great deal of internal damping material to free the small enclosure of these internal reflections.

Efficiency Considerations in Small Damped Systems

A large amount of such resistive damping will cause the system efficiency to drop even lower. This secondary drop in efficiency, due to heavy internal damping, will reflect itself in poor amplifier power utilization. The amplifier will have to be overdriven to push enough power into this low resonance small enclosure system to make it produce useful sound. If the amplifier, after being turned up high enough to satisfy normal listening requirements, still has enough power reserve to handle the sudden heavy peaks and sustained loud passages without distorting, the system will be usable. If, however, the amplifier used with the system is one of low power output, only low level reproduction will be realizable from this system. It will thus be seen that, although this system may have certain advantages as far as frequency extension is concerned, there are other factors in its design that are disadvantageous.

In making an *A-B* listening test of this type of enclosure in comparison with similar small types, the listener should be aware of this question of overall efficiency. If the test is made simply by switching the amplifier from one speaker to the other without compensating for

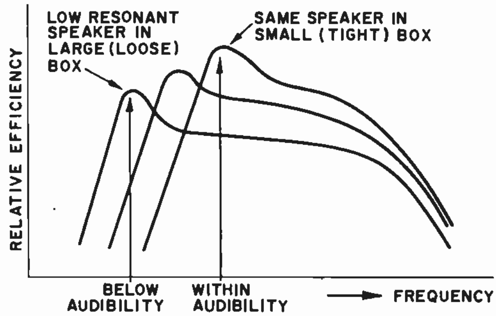


Fig. 12-4. When a very low resonance speaker is put into tight acoustic volume, the resonance of the system is raised and the region of maximum efficiency is raised to a higher frequency and the efficiency to a higher value.

the differences in efficiency between the two systems, he may get a feeling that the lower efficiency system is *deficient* because it does not sound nearly as loud as the alternate systems. His ears will, in fact, be telling him the technical truth. However, he should also listen to each of the

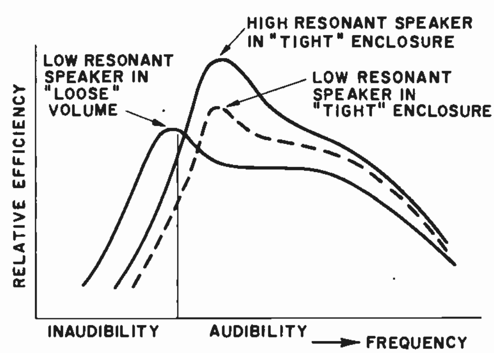


Fig. 12-5. Initial efficiency of high resonance speaker in tight enclosure will be higher than ultimate efficiency of the low resonant speaker stiffened by the tight enclosure.

systems individually for some time, at the loudness that will be most comfortable to him, as well as comparing them at equal input, before he comes to a conclusion as to the merits of one small system over the other.

Higher Resonant Speaker in Small Enclosure

What can we do now with the same relatively small sized enclosure utilizing an 8-inch speaker of more conventional resonance characteristic, say in the 90-cps region? The first thing that becomes obvious is that we certainly do not want to put this speaker into any sort of enclosure that would raise its resonance. If possible, we would like to lower the resonance characteristic of the system. It follows then that for this higher resonance speaker, we should steer clear of the completely closed box. It would seem logical, therefore, to use some sort of enclosure that does not stiffen the speaker and that, if anything, will bring it down to a lower resonance point.

This, of course, we can do by means of the simple bass reflex enclosure. In the bookshelf size enclosure, it may well be possible to convert this 90 cps peak of the speaker into the two other straddling peaks on either side of it, one at 110 cps, and the other at 70 cps. Now, while it is true that 70 cps is not as low as we would like to go in the best system, there are particular advantages to this mode of operation for smaller speakers, despite the fact that the low frequency performance does not reach down to the lowest note on the piano. Of great importance is the fact that this sort of speaker-enclosure system is of the high efficiency type. In consequence, greater linearity of overall response will be obtained because of the conservative demands made upon the amplifier by the speaker, and because of the moderate excursion of the cone for good listening level. When the cone has to move large distances in order to produce the desired listening level in low efficiency enclosure systems, these large excursions may run into regions of nonlinearity of motion of the suspension system. In the higher efficiency systems, the excursion of the cone need not be as severe, and more linear motion will result. Thus, "power-wise" and "efficiency-wise," the higher resonance system outperforms the lower resonance system.

Now let us discuss for a moment the matter of those very low frequencies. It was with good reason that we used the phrase "reach down to the lowest note on the piano." The fact of the matter is that very seldom indeed do we hear music played that contains a note this low. Notes this low are ready and waiting to be sounded on the piano, the bass violin, the harp, and one or two other instruments, but they are elicited only when called for by the composer. Fortunately for the acoustic hobbyist, composers have shown their good taste in that they realized there was a whole gamut of notes to work with, and not just low frequencies. We find therefore, if we take the trouble to do some statistical research that perhaps for 90 percent of the time, music falls in the range above 60 to 70 cps. Perhaps the one big exception to this statement is the pipe organ that can produce a note low enough and heavy enough to shake the timbers of the ordinary auditorium. It seems that the composers for this instrument delight in showing off these low massive notes, and it is unfortunate if the acoustic system cannot reproduce them. However, even with this shortcoming, the smaller systems can give quite satisfactory reproduction over an exceedingly wide useful range for their size and application.

Tunable Enclosures are Adaptable to Either Large or Small Systems

We have examined the effect of an infinitely large (wall) baffle and the infinite (closed box) baffle upon the response of a loudspeaker. We are ready to deal with the middle ground between the two — the vented box or bass-reflex cabinet. We have seen from Chap. 9 that one characteristic of the bass-reflex is that the low frequency resonance impedance of the system is spread out over an area considerably broader than the original single response peak. This serves to produce a wider band of low frequency reproduction. In the case of the bass-reflex enclosure, the question of the size of the structure is not as critical in affecting response as in the case for the completely closed box-type enclosure. The reason for this is that the bass-reflex system is a *tunable* one in which the two elements of internal volume and port opening may be juggled into many combinations and the desired resonance condition still maintained. The general conditions that govern this tuning procedure are based on the principles of the Helmholtz resonator. They are as follows. For a given volume of box, the resonance frequency increases as the opening in the box increases; for a given size of opening the resonance frequency increases as the volume decreases. Thus we can have a large volume with a large opening, or a small volume with a small opening, both with the same resonant frequency. These conditions are illustrated in Fig. 12-6. The various combinations in which volume and port opening may be arranged for a selected number of resonant frequencies have been given in Fig. 9-10.

This relationship may bear some analysis. A small volume is a stiff volume. It has small "capacity" to absorb acoustic vibration, and its acoustic capacitance is therefore small. To cause it to resonate at a low frequency, we must provide it with a large value of acoustic

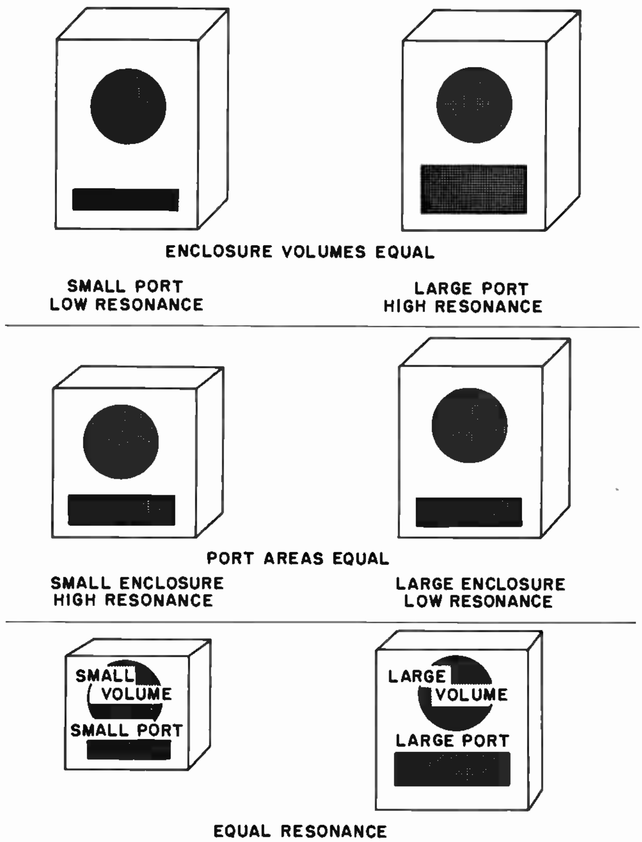


Fig. 12-6. Graphic representation of relationship between bass-reflex volume, port area, and resonance.

inductance, for in the resonant circuit the frequency of resonance is inversely proportional to the square root of the product of capacitance and inductance. In order to obtain this small acoustic capacitance with a necessarily large inertance, we must provide an opening that will make it difficult for air pulses to get through. Obviously, the smaller the hole the more difficult it is for the pulses to get through. Therefore, a small hole represents a high inertance. Thus, the combination of a small volume and a small opening resonates at a low frequency.

Conversely, a large volume, which can absorb large values of acoustic pulsations, has a large acoustic capacitance, and requires a small inertance to resonate it at a low frequency. A large hole allows sound pulses to get through quite readily, and represents a small acoustic inertance. Consequently, a large volume with a large opening will resonate at a low frequency just as well as a small volume with a small opening.

It now becomes a matter of evaluating the two different structures for qualities other than their resonance capabilities. One very easy way to examine a set of conditions is to go to the extreme values of the set, for at the extremes some of the component factors drop out and analysis becomes easier. Let us examine the bass-reflex enclosure in this manner, taking it first to an extremely small size. As the enclosure is made smaller and smaller, the port necessarily becomes smaller and smaller, until theoretically we are left with no port at all. We have a small tight closed box that is not a reflex enclosure at all. Therefore, there must be a lower limit for the size of a reflex enclosure. Looking at the other extreme, as the enclosure gets larger and larger the port size becomes larger and larger until we have the large room type of enclosure (wall baffle) with a very large auxiliary hole somewhere in the wall. Under these conditions of operation, the bass-reflex principle completely deteriorates, because the large dimensions allow standing waves to be set up in the structure, and the capacitance as such ceases to exist. Anything large enough to permit the setting up of standing waves belongs to the family of transmitting and receiving devices and not simple circuit elements. The room will act as a "sink," absorbing the rear radiation rather than behaving as an active circuit element in a resonant circuit. The end result of this "king sized" bass-reflex enclosure will be operation as a simple wall baffle with some back-to-front cancellation at some frequency where the distance from the speaker to the hole in the baffle is equal to a wavelength of the sound being radiated. Between these two extremes lies the proper size for the bass-reflex enclosure for a given size of speaker.

Enclosure Port Must be Compatible With Diaphragm Size

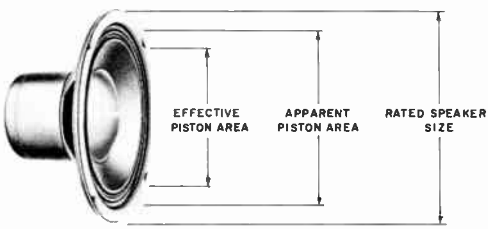
We must start with the premise that the purpose of the bass-reflex speaker is to allow radiation from the port at some low frequency region. The port must then have a radiation resistance characteristic compatible with the frequency it is to transmit. It would therefore be logical to assume that the port area should be at least as large as the speaker itself. This would give the port radiation resistance the same value as the speaker radiation resistance. If the port were made very small, there would be poor low frequency radiation resistance for the port, and the low frequency reproduction of the system would suffer. It is therefore general practice to make the port area equal to at least three-quarters of the effective area of the speaker itself.

The word *effective* is important here in that the area of the diaphragm should *not* be calculated on the basis of the advertised diameter of the loudspeaker. This is the measurement that spans the very outside of the speaker housing. From that must be subtracted that section of the diaphragm that is taken up by the rim compliance of the speaker. In a 12-inch speaker, this reduces the diameter of the speaker by at least another 1-½ inches. Then there is a somewhat more subtle effective reduction of the diaphragm because of inability of the diaphragm to vibrate completely as a piston. All in all, if we consider the effective area of a 12-inch loudspeaker to be that of a 10-inch *piston,* we have a fairly good conservative approximation of its active diameter. Using this figure for the diameter of the speaker, we arrive at an area of approximately 75 to 80 square inches for the area of the port. If we want to go down to three-quarters of the effective area of the diaphragm, the port area should be at least 60 square inches. This figure places the average limit to the lowest value of port area for a bass-reflex enclosure for a 12-inch speaker. In like manner, the smallest port suitable for an 8-inch speaker (effective diameter, about 7 inches) is 28 square inches, and for a 15-inch speaker (effective diameter, about 13 inches) about 95 square inches. These figures (tabulated in Fig. 12-7) give the smallest port size practical, from which the enclosure may then be designed.

With this port area figure available and the resonant frequency of the speaker either known or determined, the enclosure volume is readily selected from the chart of Fig. 9-10. The set of conditions thus determined will provide the optimum resonance impedance compensation for the chosen speaker by being matched to it in frequency, and by acting as a good radiating source itself (through the port).

If the constructor is building his enclosure from the ground up, it is possible for him to follow these precepts quite literally. However, the user often finds it necessary to use an already existing piece of furniture. He will then find it necessary to work backwards from the fixed enclosure size to the port size. If, because of a fixed small cabinet size, the port turns out to be too small to produce the proper radiating characteristic for the desired low frequencies, there is not too much that can be done except to make a compromise. The best compromise that can be made in this case is to provide a port of at least one-third to one-half the effective area of the diaphragm so that reasonable low frequency efficiency of radiation from the port will be obtained. This will result in an enclosure volume that will resonate somewhat higher than the speaker requires and the area of bass compensation will be moved up in the frequency scale.

In the case of enclosures that are originally large, like built-in wall cabinets, it is possible and desirable to build the full-size port into the structure as called for by the chart. However, the port should not exceed twice the effective diaphragm area, even though theoretically the enclosed volume calls for a larger port than this. By this last expedient of making the port larger than the diaphragm requires, but still smaller than the resonance condition calls for, we effectively reduce



RATED SPEAKER SIZE	EFFECTIVE DIAPHRAGM AREA	MINIMUM PORT AREA
8"	36-38 SQ. IN	28 SQ. IN.
12"	75-80	60
15"	125-130	95

Fig. 12-7. The effective diaphragm area of a speaker is smaller than its rated size. This table gives the effective areas of speakers for use in designing bass-reflex enclosures. (See also Fig. 9-10.)

the resonance frequency of the enclosure. The overall effect of this low resonance cabinet is to permit the cabinet to function at a lower frequency than the original speaker resonance frequency, with consequent boosting of the output in those areas below resonance. The effect of the larger port is to produce better radiation characteristics from the opening for the low frequencies. However, in this mode of operation, the section of the response *immediately at* and above the speaker resonance area will not benefit by the reflex action to the optimum degree, and the response in that area will be compromised. Furthermore, too large an enclosure may cause actual deterioration of the bass-reflex function as described above. In large fixed enclosures it is of course possible to make changes to the internal volume by simply closing off as much space as is necessary to bring the port area down to a respectable size, so that proper tuning may be realized.

In laying out the volume of the enclosure, some thought must be given to the proportions of height to width to depth. It is not generally desirable to proportion the enclosure to a symmetrical cubic form. The uniformly equal lengths of such a structure make it easy for standing waves of one particular frequency to be set up. The box will tend to exhibit a "normal mode" of vibration because of its symmetry, and this normal mode of vibration will not be related to the bass-reflex resonance condition. Where volumes of cubical form must be used because of already available structures, it is necessary to provide a lot of internal wall damping material so that the internal waves

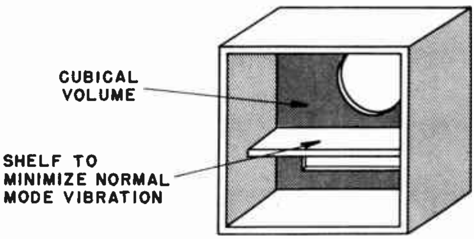


Fig. 12-8. Where nearly cubical volumes are used, the interior should be well padded and broken up by a shelf placed midway in the structure. This will minimize the strong normal resonance modes characteristic of cubical structures.

are well absorbed by the reflecting layers. This will greatly reduce the normal mode resonance. In such structures, it is also common to place a well-padded shelf along the middle of the cabinet, reaching about halfway to the rear wall. Such a structure is shown in Fig. 12-8. This will further reduce the internal standing wave condition and minimize the normal resonances of the box. Practice has shown that good results will be obtained if the height-to-width-to-depth ratio is 4 to 3 to 2.

CHAPTER 13: *The Enclosure as a Crossover Element*

Enclosures Provide Acoustical Crossover Characteristics

Crossover networks are vital components of multi-speaker systems. They perform the important function of segregating specific bands of energy into specific channels. The electrical means of accomplishing this result were fully treated in Part I. In addition to electrical networks, there are mechanical networks that separate mechanical vibrations of different frequencies and channel them to the vibrating members best able to handle those bands. This was treated when we discussed two-way speakers with mechanical crossover elements. Then there is the matter of *acoustical crossover networks,* which play as important a part in the overall picture of frequency band segregation as do the electrical and mechanical elements.

In the discussion on multi-speaker systems, stress was placed on the compatibility of the component parts. To catalog them briefly: (a) the component speakers must be of equal efficiency for overall uniformity of level; (b) their ranges of operation should overlap so that there will be no depressions of the response characteristic in the areas joining the adjacent bands of the speakers; and (c) there must be compatibility of the greater angular dispersion of the high frequencies from the tweeter, as the high frequency beam from the lower range speaker sharpens. This is essentially a problem of acoustical crossover for the treble and tweeter units, and the reader may refer back to that section for a review of the question of compatibility in general. Our present concern is specifically with the action of woofer enclosures as part of the acoustic network of the system. Ae we examine them we find that they play as important a part in the function of the crossover characteristics of the system as do the smaller treble horns previously discussed.

Bass Reflex Channels Lows Only Through Port

Acoustical crossover principles are not specifically limited to multi-speaker systems. In fact, a single speaker in a simple bass-reflex enclosure is subject to the inexorable laws of acoustic crossover whether we like or not. Let us once again look at the bass-reflex enclosure, but this time a little more critically and just a bit more technically. We have said that the bass-reflex enclosure is a resonant acoustic device, being made up of an acoustic capacitance and an acoustic inductance, usually referred to as an inertance. There is an easy way of illustrating this combination if we wish to use electrical symbols; it is shown in Fig. 13-1. The section shown within the dotted square represents the bass-reflex enclosure. The electrical symbol of the capacitor is representative of the volume (capacity) of the enclosure. The electrical symbol of the coil is representative of the inertance of the air in the port of the enclosure, and the electrical symbol of resistance represents the radiation resistance of the air load that the port sees.

We will assume the speaker to be a generator of electrical signals of *all* frequencies that are to be transmitted through the enclosure network connected to it. We know that a choke coil will pass low frequencies and block high frequencies; that is, its reactance goes up with frequency. Thus, when the generator (loudspeaker) is producing low frequency signals, these signals will pass through the choke (the port) into the resistive load (radiation resistance of port), and power will be developed in the resistance. Transposing this statement into acoustic terms, we can say that when the loudspeaker is reproducing low notes, these notes will come out of the port and into the acoustic radiation resistance of the port, and low frequency power will flow out through the port.

Port "Closes Up" as Frequency Goes Up

Now let us return to the generator. If we raise the frequency of the signal being generated, the choke will begin to offer more and more opposition to the passage of the signals of increasing frequency, and fewer will get to the resistance. At the same time, however, the

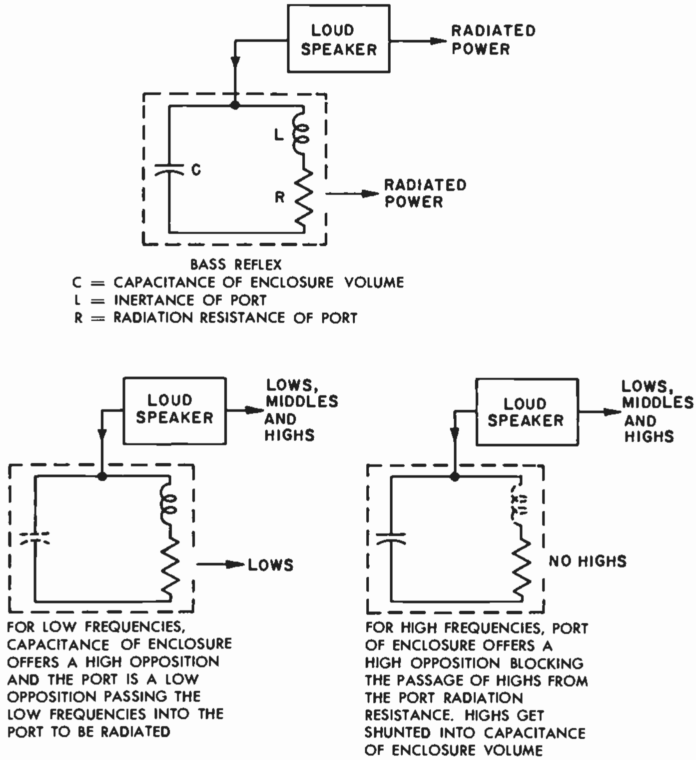


Fig. 13-1. The capacitance of the bass-reflex enclosure and the inertance of its port act as L-C *elements of a crossover network, allowing lows from the port and restricting middles and highs to the speaker.*

capacitor will begin to offer *less* opposition to those same signals. The higher frequency currents will thus begin to flow through the capacitor as they cease to flow in the other branch of the network. Again, translating this statement into acoustic terms, we can say that as the frequency of reproduction of the loudspeaker goes up, the port "closes up" acoustically, and those higher frequencies find their way into the acoustic capacitance volume of the chamber, instead of into a useful radiation resistance. As these higher frequencies enter the capacitance, no acoustic work is produced. Instead, as we have seen, these higher frequencies bounce back and forth from wall to wall within the cavity unless proper soundproofing is applied to the interior walls. Hence, the bass-reflex enclosure acts as a true crossover element, working entirely on acoustic principles. At low frequencies, it transmits sound from the port and there is both forward and rear transmission of power from the loudspeaker to the outside. At higher frequencies, the box closes up acoustically, and only direct forward radiation takes place from the diaphragm.

A Small Enclosure Provides Low Frequency Roll-off For Midrange Cone

This effect of the bass-reflex enclosure is, of course, not unique among enclosures. In fact, definite use is made of the simpler box enclosure and its crossover characteristic in the design of multi-speaker systems using cone speakers as midrange units. We have seen that as we decrease the size of the enclosure, the resonance frequency of the system increases. A small tight volume in back of a speaker acts like a tight air spring, preventing the speaker from vibrating as freely at low frequencies as it would in a larger enclosure. This enforced rise in resonance of the speaker, due to the smaller enclosure, has two effects. It raises the area of most efficiency of the system to a higher frequency, where it will be more useful, since this is to be strictly a midrange system. (See Fig. 13-2.) Secondly, the output falls off more rapidly below the new higher resonance point that results from the tightened enclosure. This transposition of the resonance point of high efficiency from some normally low frequency to a higher frequency within the midrange is, of course, very desirable. Just as desirable is the enforced cutoff of response below the new resonant frequency, for this may now be used as an acoustic component of the electrical crossover network whose job it will be to limit the low end response of the midrange speaker. Thus, in addition to boosting the midrange efficiency, the addition of a small closed box as a baffle for the midrange system affects the attenuation rate at the crossover point. Consequently, the final crossover characteristic may be made sharper than that obtainable from the electrical network alone.

Midrange Cone Enclosure Isolates it from Woofer Enclosure Pressures

The rear enclosure of the midrange speaker serves another useful purpose. It isolates the midrange cone speaker and prevents it from being affected acoustically by the woofer speaker. If the back of the midrange cone speaker were left open, the strong internal pressure developed within the main enclosure by the large excursions of the woofer diaphragm would tend to put the midrange diaphragm into sympathetic vibration with the woofer itself. This would result in cross-modulation of the midrange speaker by the woofer. The heavy low frequency pressures within the cabinet would tend to bounce the midrange diaphragm around. Boxing the midrange speaker removes these back low frequency pulsating forces from its diaphragm, leaving it free to perform the function for which it was installed. This condition of cross-modulation of the smaller by the larger speaker is not

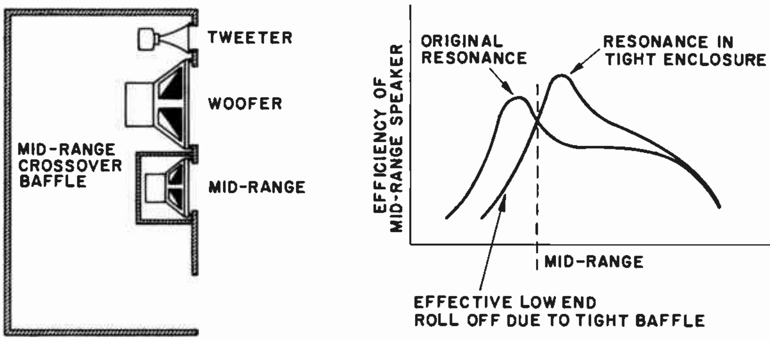


Fig. 13-2. Tight midrange speaker enclosure boosts high efficiency resonance range of speaker into desired midrange band, and rolls off the low end of the speaker. The tight box thus prevents the midrange speaker from producing lows in the same manner as an inductive element of an L-C *crossover network.*

so important when looked at from the front, because on this side, where there is free unconstrained radiation of the pressure, the pressures are not as intense as they are in the back of the cabinet, where they are confined into a relatively small space. Thus, the small midrange enclosure acts as an acoustical crossover element, and works hand in hand with the electrical network to produce the desired end result.

The Horn is a High-Pass Filter

Perhaps the type of enclosure in which acoustical crossover comes most into play is the horn. The important crossover factor in the horn is, of course, its own theoretical cutoff frequency. We have seen that the choice of the proper place to crossover electrically, in the case of a two-way system employing a horn for the upper range, is determined in part by the frequency at which the horn cuts off. Above that point (theoretically), the horn, operating essentially as a high-pass filter, passes all frequencies. This, in practice, however, is not necessarily true of the horn and driver unit *combination.* Even though the driver unit itself may have perfect response and the horn may be a perfect

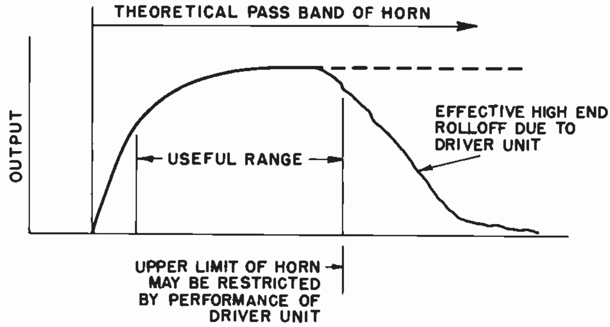


Fig. 13-3. In addition to its natural low frequency cut which acts as an acoustic crossover characteristic, the upper limit of the horn will be determined by the driver unit feeding it.

horn, in combination an acoustical phenomenon may take place that could make it necessary to consider another crossover point *within* the pass band of the horn in addition to that determined by the cutoff point, as suggested in Fig. 13-3.

In this treatment, we shall deal first with the woofer horn and then discuss the same problem briefly in connection with the higher frequency horns. Figure 13-4 illustrates the typical construction of the compression chamber and the throat area of the woofer horn, with the throat diameter smaller than the diaphragm area, as is normally the case. We are now to examine the way the sound progresses from the diaphragm to the throat area. For the sake of clarity of exposition, the diaphragm has been shown to be flat. It is assumed, of course, that the diaphragm is one rigid piston, and that it moves as a whole. This means that all parts of the diaphragm move forward simultaneously by the same amount, and that they all move backward together by the same amount. We are going to consider what happens to the sound coming from the front of the diaphragm into the acoustic chamber between it and the throat of the horn.

Upper Band Pass of Horn Controlled by Driver Unit Head Design

Since all parts of the diaphragm are moving in synchronism, the sound coming from the diaphragm will constitute a flat or plane wavefront. This may be considered as equivalent to saying that every section of the diaphragm is producing its own ray of sound. As these rays all move out together, they form a flat wavefront, as would a line of soldiers in company front formation. Now, if this flat wavefront had nothing to do but travel forward unimpeded, there would be no problem. However, in order to get into the throat of the horn, those rays coming from outer sections of the diaphragm will have to be bent from their line of march. They will consequently arrive at the throat of the horn at some later time than those rays coming directly from the center of the diaphragm. Acoustically speaking, there will be a phase delay between the center sound rays and the outer sound rays as they reach the throat.

If this difference in path length becomes large enough so that the longer paths are actually as much as a half wavelength of a particular sound to be produced, the delayed rays and the center rays will be com-

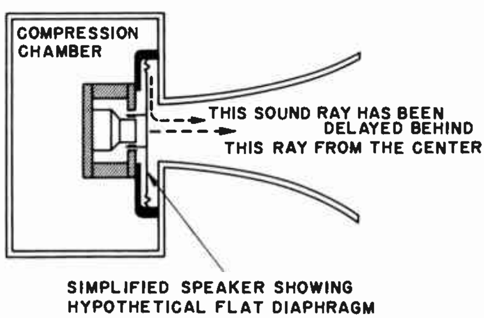


Fig. 13-4. Due to unequal path lengths from the different parts of the diaphragm to the throat, these rays may arrive out of phase at the throat, causing cancellation of sound.

pletely out-of-phase at the throat, and severe cancellation for that frequency will occur at the throat. Consequently, for this frequency, there will be no sound transmission through the horn.

The response of the horn for the lower frequencies will not be so affected because these frequencies are of such long wavelength, and the path lengths in the chamber are so small compared to the long wavelengths of the low frequencies, that internal phase cancellation does not take place. Cancellation will take place for those wavelengths for which the path lengths from the center of the diaphragm and

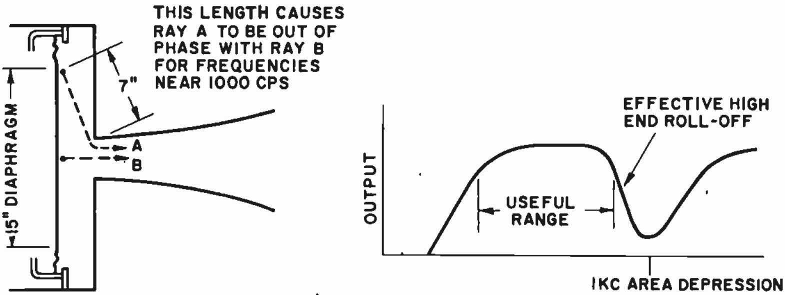


Fig. 13-5. Cancellation of energy at the throat due to path length differences makes it necessary to cross over below area of cancellation.

the outer areas differ by a half wavelength. As shown in Fig. 13-5, if the path length from the outer effective section of the diaphragm to the central part of the throat is 7 inches (as with a 15-inch speaker), this represents a half wavelength at approximately 1000 cps. Therefore, with the throat size shown, and for the large speaker, the *combination of horn and driver* unit will fall off sharply at this critical frequency.

This characteristic of driver-horn combinations may be readily corrected at the driver unit. It will be found that in driver units made for high frequency operation, the heads are designed with phase correcting plugs or devices that help to equalize the various path lengths from all parts of the diaphragm to the throat, so that this destructive internal cancellation does not take place within the desired operating range of the horn. Such phase correcting units are shown in Fig. 3-14.

In the case of the folded woofer horn, however, it would obviously be of no practical value to employ such high frequency correcting devices. Therefore, we allow this higher frequency deterioration to take place and employ it as a natural acoustic crossover effect of the horn, when the horn is employed as a woofer. In this sense, then, the woofer-horn combination is a bandpass filter rather than just a high-pass filter, as illustrated in Fig. 13-6. It passes frequencies above that of the theoretical cutoff point of the horn, and below the frequencies at which destructive throat air chamber cancellation takes place.

Chamber Between Diaphragm and Throat is a Low-Pass Filter

There is another acoustical crossover characteristic of the throat chamber of the horn, where only the back of the speaker is horn loaded and the front of the speaker radiates directly. In this structure, shown

Fig. 13-6. Horn is changed from high-pass element to band-pass element by allowing throat cancellation to take place. This is permissible where horn is applied to woofer. Horn thus acts as high end rolloff for woofer section.

in Fig. 8-3(A), there is a low-pass acoustic filter between the rear of the diaphragm and the throat of the horn. This cavity represents a capacitance across the line, as illustrated in Fig. 13-7. The narrow throat represents an inductance in series with the line. That is, the sound from the back of the diaphragm can disperse in this cavity but, at the same time, must get through the throat of the horn to be radiated. This is essentially an acoustical counterpart of the familiar electrical low-pass filter. Highs are shunted out of the line by the capacitor. Accordingly, those lower frequencies within the band, which are not

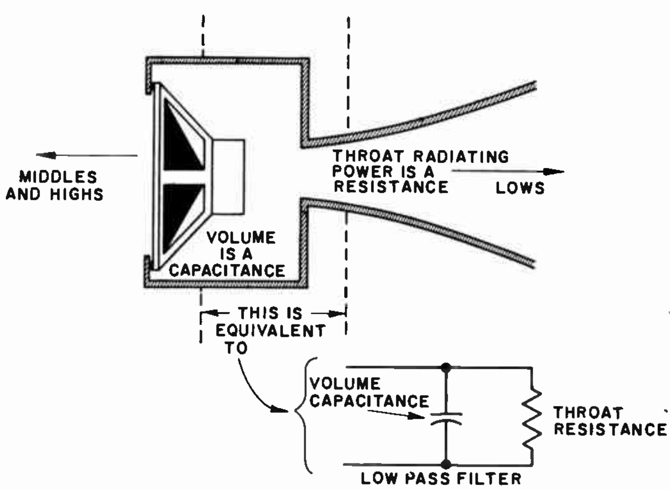


Fig. 13-7. In compound horn, rear volume is a low-pass element. It "stores" highs and middles, stiffening the rear of the speaker for these frequencies and improving the middle and high frequency loading of the diaphragm for forward radiation. Acoustic crossover action is thus accomplished by the enclosure.

materially affected by the cavity capacitance, will be passed on by the acoustic filter and will get into the horn to be radiated as sound power, *provided* the theoretical cutoff of the horn itself is such as to allow them to pass.

This air chamber, by preventing the passage of middles and highs into the horn, becomes a "storage chamber" for these frequencies. The middles and high frequencies from the back of the cone will, therefore, see an acoustically "stiffer" enclosure at their back, due to this filter imposed between the cone and the throat. Consequently, those upper frequencies which are front radiated will now issue from an enclosure that has been tightened up, and as a result there will be somewhat better rear-loading of the speaker *for these higher frequencies.* Better middle and high frequency efficiency will be obtained. For the lower frequencies, the rear enclosure chamber "opens up," due to the low-pass characteristic of the filter, and the proper rear loading of the horn to the speaker takes place for the low frequencies.

Horn Bends are Low-Pass Filters

There is another acoustical crossover phenomenon that takes place in folded horns, and this is brought about by the necessity for the sound wave to travel around the bends of the horn. Destructive cancellation of high frequencies may take place because of the difference in the path lengths that sound rays have to travel in negotiating a corner. Secondly, a loss of high frequencies will also occur because of the greater difficulty that high frequency waves experience in diffracting around corners. These two effects are not necessarily the same.

Taking first the more familiar effect, that of diffraction, we know that diffraction of a sound wave is accomplished much more easily for those frequencies at which the wavelength is large compared with the opening from which they emerge. Thus, as shown in Fig. 13-8, if the wavelength of the sound is such that the pulse leaves the opening of the duct before the succeeding pulse has been thoroughly established behind it, this first pulse is free to expand in all directions once it is out in the open. It will, therefore, spread out in all directions, and "diffract" around the sides of the duct. However, in the case of frequencies at which the sound pulses are literally on top of one another as they emerge from the duct, one pulse will keep urging the one in front of it in the direction in which it started — straight ahead. Therefore, there will be less diffraction of the higher frequencies from this same duct. If this duct happens to be the end of one section of a folded

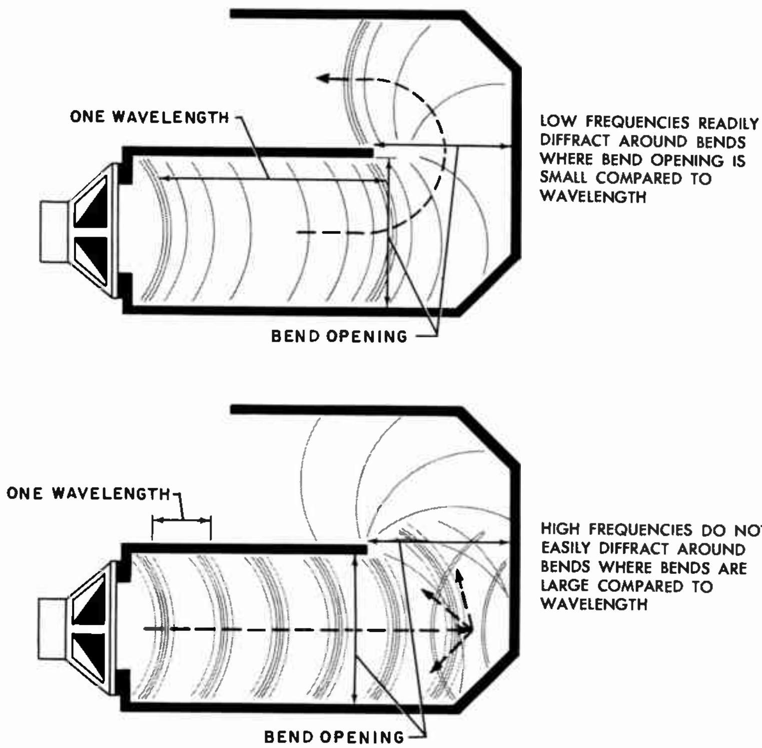


Fig. 13-8. High frequency attenuation will occur in a folded horn due to poor diffraction around bends, unless horn is specifically designed for high frequency radiation. (See Part 1, Fig. 3-20.)

horn, the higher frequencies will not get around the bend as easily as the lower frequencies. The bent horn, therefore, presents an acoustic high frequency filtering action; that is, it acts as a crossover element in the total acoustic circuit. In addition to the above effect, multiple reflections of the higher frequency sounds occur at the bend. This causes some of the energy to be bounced back toward the speaker and to be scattered in directions that may not contribute toward getting the high frequency sound waves around the bend.

Figure 13-9 illustrates the further cancellation of high frequencies that actually do manage to get around a bend. If the bend is quite large in cross-section compared to the radius of the bend, the path lengths close to the inner wall will be much shorter than the path lengths close to the outer walls, as can easily be seen from the figure.

This means that, if the wavefront for a particular high frequency started out straight at the beginning of the bend, by the time it got around to the end of the bend, a good part of it would be delayed. As the bends continue to occur, or as their sharpness increases, parts of this advancing wavefront of the high frequencies become more and more delayed until soon half of it is out of phase with the other half. The end result is a sharp drop in acoustic output for the higher frequencies from the folded woofer horn. This deterioration of the high

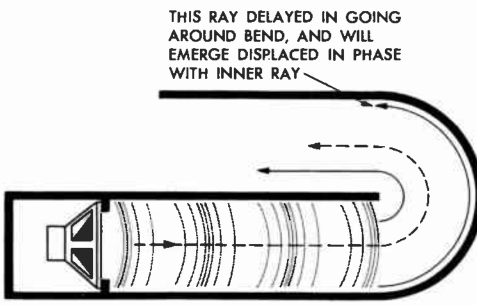


Fig. 13-9. Where bend cross-section is large, there will occur phase displacement of the rays from different areas of the bend resulting in eventual cancellation of high frequencies where these phase shifts amount to 180°. In bent woofer horns this constitutes an effective high frequency rolloff characteristic.

frequencies may be minimized in folded or reflexed horns where it is desired to operate them in their high frequency region. In such a reflexed horn, as shown in Fig. 3-20, the expansion of the horn is accomplished so that the width of the horn at the bends is very small compared to the radius of the bends, and very little high frequency attenuation occurs. Thus it is seen that the folded woofer horn becomes an element in the crossover system by attenuating the higher frequencies through the combined effect of poor diffraction around the bends of the horn and the increasing phase differences in the wavefront of the high frequencies passing around the bends of the horn.

It will be recognized that almost the same set of conditions of bending and diffraction exists in the acoustic labyrinth, causing the acoustic labyrinth to serve the same acoustical crossover function. There is one difference between the two enclosures, however. In the folded horn, only the horn itself has a low frequency cutoff and it will pass *all* frequencies above that point that reach its throat in fairly uniform phase. In the case of the labyrinth, however, the rear duct is tuned to one particular low frequency, and above that frequency its radiation drops off at a gradual, steady rate. Therefore, the low-pass filter at the back of the diaphragm in the labyrinth, in conjunction with the labyrinth itself, usually cuts off at a considerably lower frequency than the filter for the folded horn, since the horn extends out further in its transmission band.

Final Crossover Characteristic is Function of Electrical, Mechanical, Acoustical Elements

Although many of the factors discussed in this chapter are often beyond the control of the home constructor, it is still of value to realize that they exist in baffles and enclosures, just as they exist in the speakers themselves (over which he likewise has no control). The value of being cognizant of these phenomena, however, lies in the realization that one is dealing with a *system* of components whose performance is a function of the teamwork between all the components throughout the entire electrical-mechanical-acoustical circuit. It is not correct to state categorically that because the electrical network someone happens to buy or make is a 6- or 12-db per octave type, his system is going to operate at that rate of attenuation. The final performance will be almost equally tempered by the response characteristic of the speaker units used *and* the crossover effects of the auxiliary acoustic enclosures in which the loudspeakers are to be installed.

CHAPTER 14: *The Mechanics of Good Enclosure Design*

Enclosures Should Transmit, not Absorb Energy

Enclosures and baffles in general are means of transferring acoustic energy from the loudspeaker to the surrounding air. Whether the enclosure is a simple closed box or the most complex of horn structures, its main purpose is to cause the sound from the loudspeaker to be most efficiently radiated into free space so that it may be properly heard and enjoyed. Enclosures and baffles obviously do not by themselves produce any acoustic power; neither should they absorb any power, for if they do the total acoustic efficiency of the system will suffer.

Enclosure Internal Pressures are High

The key to loudspeaker enclosure construction is rigidity, or resistance to vibration. The wooden panels must be rigid so that they will not be vibrated by the sound pressures within the enclosure. If they are vibrated, acoustic work is done on them and that acoustic power is wasted.

The internal sound pressures in enclosures can be quite sizable. Take the case of the closed box (the "infinite" baffle). It will immediately be realized that as much sound as is allowed to fill the entire room on the front side of the enclosure is also present on the inside of the enclosure, but here it is captured with no place to go. Because the sound is captured in the back, large internal sound pressures are built up, especially on low frequencies, where most of the acoustic power resides anyway. Even in the bass-reflex enclosure, in which the sound does finally come out from the port, that sound must still encounter the comparatively cramped quarters inside the enclosure before emerging.

These internal pressures can become quite high and may cause deleterious effects to the total acoustic performance if the enclosure is not properly constructed. It is only necessary to place one's hand on the side walls of a poorly constructed enclosure when it is reproducing a full throbbing bass note to feel the vibrations of thin or poorly braced wood panels. Practice has shown that good results may be obtained when the lumber used in the construction of the enclosure is at least ¾-inch thick and of a good grade of plywood or cored wood.

Back of Bass-Reflex Enclosure Must be Rigidly Made and Secured

In the case of the bass-reflex enclosure, just as much care must be shown in constructing the removable back as is given to the enclosure as a whole. Too often, the constructor gives this section the minimum of attention because it is not seen. However, if it is neglected, it may not be seen but it certainly will make itself heard in undesired ways. Figure 14-1 shows the response of a bass-reflex enclosure with a properly constructed back, and again with a thin ¼-inch back held on loosely by four screws in the corners of the panel. While the sound output waveform of the enclosure with the good back is clean and undistorted, the output of the same cabinet with the poor back is rough, distorted, and lower in level than the former. The loss in output is, of course, due to the fact that much of the back radiation from the cone was used up in making the rear panel vibrate, and so was lost as far as useful acoustic output was concerned. The waveform has deteriorated because of the irregular and comparatively uncontrolled vibrations of the rear panel, which are heard directly by the ear, and which also are reflected back internally upon the speaker to affect its initial performance.

Because the back of the bass-reflex cabinet must be removable, it is often made to be held on with just one screw in each corner, as in the test just described. This is a serious mistake. Even though the rear panel itself may be of rigid stock, if it is not securely held down around its entire contact area, it will be subject to vibration. It is good practice to have at least four good sized screws along *each* edge of the rear panel securing it to the main structure. In the construction of the

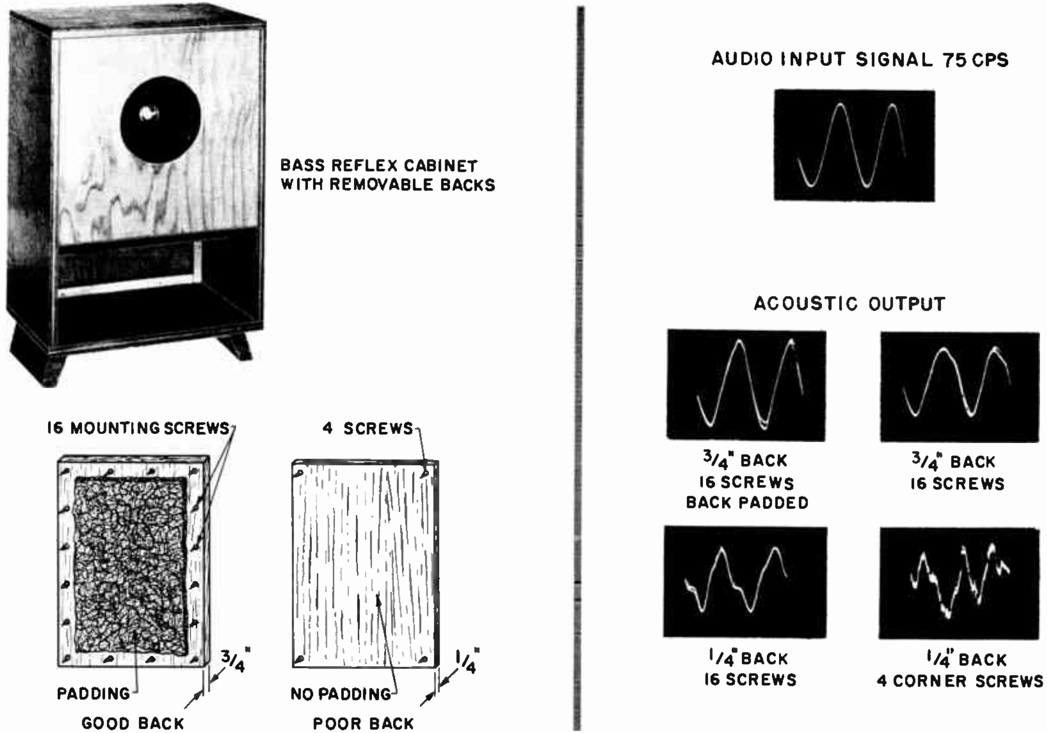


Fig. 14-1. The back of the bass-reflex cabinet will cause waveform deterioration and loss of level at low frequencies if it is weak and poorly mounted.

enclosure, the use of nails should be avoided as the sole means of holding the sections together. Nails will not hold them together properly for any length of time nor will the mating sections be adequately joined.

Acoustic Enclosures Deserve Fine Cabinetmaking Techniques

One should pay the same attention to the making of the acoustic enclosure as a cabinetmaker pays to the construction of a fine piece of furniture. The use of plenty of good furniture glue and screws where needed is highly recommended. The home constructor does not always have available cabinetmaker clamps to hold the sections together while the glue is setting. In this case it is advisable to make use of large screws to hold the sections together immediately after the glue has been applied to the mating edges of the panels. Not only will the screws be an aid in the construction of the cabinet, they will act as reinforcement to the finished cabinet after the glue has set, and will prevent the panels from warping away from the glued seams.

Adequate Glue Necessary for Strength and Freedom from Buzz

Adequate use of glue is more than a construction must, it is an acoustic must as well. Not only does glue provide a good mechanical bond between the sections, it also prevents buzzing of joints between those panels that are subject to high internal acoustic pressure. There is nothing as annoying as an intermittent buzz somewhere in the system that cannot be traced down. More often than not, such a buzz may occur at the areas where the panels meet, but where they are not joined together rigidly. Buzzing may be set up at these unsupported edges when the panels experience excessive internal acoustic pressures. Insurance may be taken against this sort of troublesome buzz by making certain that all these joined edges are well filled with glue before they are put under pressure. After the glue has been applied, and the sections clamped together or screwed together, there should be evidence of glue seeping out from the entire length of the joint. The surplus glue thus squeezed out should be wiped away before it hardens, to facilitate finishing of the cabinet.

It is also common practice to use glue blocks on the interior of the cabinet to add further rigidity to the entire structure and to reinforce the joining edges of the panels. A typical construction illustrating the use of the glue block is shown in Fig. 14-2.

Where it is desired to keep the rear panel of the bass-reflex cabinet removable (or for that matter any section of an enclosure, such as the compression chamber of a horn in which the driver unit must be inserted), felt stripping may be applied to the edges of the panel so that when it is screwed down it seals itself and inter-panel buzzing is eliminated at the same time.

Bracing May be Necessary on Large Panels

Sometimes, even heavy panels will vibrate if they are large in size, and additional means must be employed to prevent them from being energized into vibration by the internal pressures. Such mini-

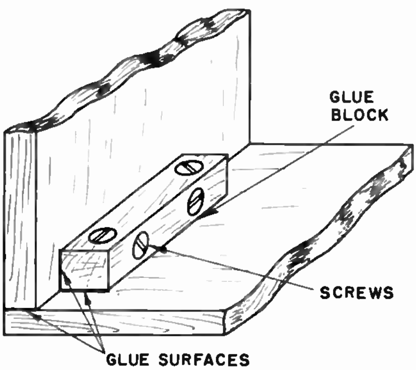


Fig. 14-2. The use of glue blocks in cabinet construction insures squarely braced surfaces, a means of screwing the panels together to hold them while the glue is setting, and a means of construction whereby screws do not show on the finished surfaces.

mizing of the vibrations of large panels may be accomplished by the use of cross braces applied diagonally across the panel area. This diagonal position is the longest dimension of the panel, and so needs the most support. Often a sturdy cross-strut, reaching across from one panel to the opposite one, may be sufficient to break up the vibrating condition of large panels.

Speaker Mounting Panel Should be Very Flat

Of great mechanical importance to the performance of the baffle is the sturdiness and the *flatness* of the speaker mounting panel itself. It goes without saying that the panel that is most subject to vibration is the one on which the vibrating loudspeaker itself is mounted. This front panel should, therefore, be chosen as the sturdiest one of the lot. If possible, it should even be thicker than the other panels, even though this may mean paying a slight premium for one piece of wood from the local lumber yard.

Of equal importance is the flatness of the panel in the area in which the speaker is mounted. A warped section of panel in this area will prevent the proper seating of the loudspeaker in its appointed place, creating two undesirable effects. First, it may be impossible to screw the speaker down *flat* to a warped baffle. If the speaker has a hard unpliable rim gasket, or simply has a hard cast ring on its mounting rim to give the speaker structural rigidity, there will be gaps between the speaker and the panel to which it is mounted. An acoustic leak will therefore exist between the back of the speaker and the front, which, if large enough, may cause some deterioration of the low frequency response in the bass-reflex enclosure. More important, however, if such a leak occurs in the compression chamber of a horn system, the action of the horn may be quite seriously upset. It is for these reasons of adaptibility to baffle panels that are not completely flat, that many of today's speakers are designed with comparatively soft edge gaskets, made of a cork composition or laminated newsboard. When a speaker with such an edge-sealing gasket is used, the mounting panel irregularities are somewhat taken up by the "give" in the gasket.

Sometimes actual damage may be done to a loudspeaker if the mounting baffle is too badly warped. In the effort to screw down the edge of the basket at all points, the basket rim itself may be warped out of its normally flat condition if the basket design is not well thought out. If such warping of the basket occurs, the alignment of the entire moving system may be thrown out of its true balanced position, especially insofar as the voice coil itself is concerned. If the voice coil is thus forced into a misaligned position, it may eventually start to rub against the narrow gap faces, which will result in fuzziness of reproduction, distortion, and ultimate rupture of the coil.

Speaker Should be Securely Fastened to Mounting Panel

There is often difficulty in screwing the speaker down to the mounting panel. Tragic results sometimes occur when ordinary wood screws are used. Often the wood screw is not started correctly. It begins to tilt, and the screwdriver slips from the head of the screw and goes through the fragile paper cone of the speaker. This is especially heart-

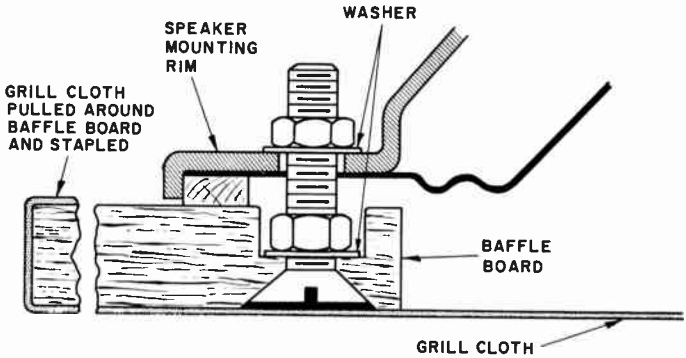


Fig. 14-3. A means of preparing the baffle panel of an enclosure to insure positive seating of the speaker to the baffle board.

breaking when you have just bought the speaker and haven't even heard it woof or tweet. Such damage may be prevented by drilling small holes in the spots where the wood screws are to be driven into the panel. The holes should be small enough so that the screw does not lose its grip on the wood when driven into the hole, but on the other hand, the hole should be large enough to provide a proper guide for the screw so that it may be easily driven in.

There are better means of mounting the loudspeaker than with wood screws. If the mounting panel is readily accessible from both sides, machine screws and bolts may be used to secure the speaker. If, however, the construction is such that the front panel is to be directly covered with grill cloth, there is another expedient that may be used. Machine bolts may be permanently inserted into the mounting panel and held tightly in place by nuts countersunk into the wood. This will leave the mounting bolts sticking out from the back of the mounting panel ready to accept the speaker, as shown in Fig. 14-3. A word of caution is in order, however, if the speaker is so mounted. When placing the speaker over the bolts, be very careful not to place the speaker down in such a position that the cone itself is over one of the bolts; it might easily puncture the cone. However, with reasonable care, the speaker may be slipped over the mounting bolts and then readily secured by means of lock washers and nuts.

A means of overcoming this bolt problem is available through the use of "T" nuts. These are simple and relatively inexpensive devices readily available from a local hardware merchant or cabinetmakers' supply house. They are illustrated in Fig. 14-4. Holes are drilled where the speaker mounting bolts are to be inserted. Into these holes are

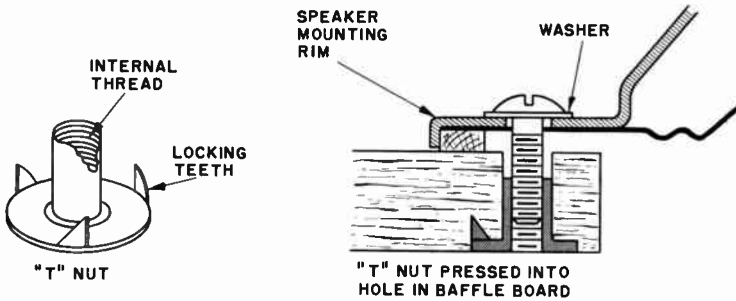


Fig. 14-4. A preferred method for preparing the baffle panel with "T" nuts. This method provides optimum ease and safety of the speaker installation with positive seating of the speaker to the board.

forced these "T" nuts from the side of the baffle opposite to that on which the speaker is to be mounted. They seat into the tight hole drilled for them and stay in place until the speaker is to be mounted. When ready for mounting, the speaker is placed over the holes, and bolts are then driven through the speaker rim into the "T" nut. This is perhaps the safest, easiest, and surest way of mounting the speaker to its panel without danger of going through the cone with either wood screws or protruding bolts.

Panel Vibrations Deteriorate Horn Action

In the case of the horn type of enclosure, the matter of the rigidity and resistance to vibration is of even more fundamental importance than in the simpler bass-reflex type. The reason for this is twofold. First, as we have seen, all the sound the diaphragm produces is squeezed into the comparatively small space in the throat of the horn. If we compare the volume at the throat of the woofer horn with the volume behind the diaphragm in the bass-reflex enclosure, we have an idea of the ratio of pressures between these two conditions. If the internal pressure in the bass-reflex enclosure is considerable, how much more intense must the sound pressure be when the equivalent of this amount of sound is compressed into the relatively small throat area of a horn. It is, therefore, of utmost importance in horn construction that no pains be spared in making the throat section as strong, as rigid, and as resistant to vibration as possible. If there is any "give" to these panels at the throat, the horn will suffer severe acoustic losses.

Special attention must also be paid to the compression chamber, for the internal pressures built up in this small volume may also be of considerable magniture. One should make sure that the panel in this compression chamber, which is usually made removable for purposes of mounting the loudspeaker on the inside, is provided with a sealing

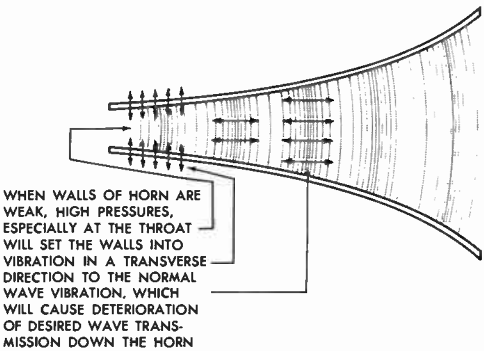


Fig. 14-5. Horn action may be deteriorated by weak horn construction, especially at the high pressure areas.

gasket around its rim and that it is secured to the housing with enough screws so that with the felt sealing gasket in place there will be no pressure leaks in this vital area.

The second reason for making sure that there are no sections in the horn with a tendency to vibrate is that if these critical area members do vibrate, the horn action is partially destroyed. The sound pressure developed by the driver will be partly dissipated in setting up transverse waves across the horn through putting the panels into vibration, rather than setting up longitudinal waves down the horn where they belong. This is illustrated in Fig. 14-5.

Sound Absorbing Material Stops Internal Reflections in Bass-Reflex Enclosure

Enclosure type baffles, such as the bass-reflex, require treatment with sound absorbing material on the inside of the structure. The main purpose of this treatment is to minimize standing waves within the enclosure. These waves would cause ragged mid-frequency response and boominess in the low frequency region. Let us examine the case of the bass-reflex enclosure. As the loudspeaker diaphragm vibrates back and forth, sound waves travel out from the diaphragm in two directions, forward and backward. The forward radiation naturally goes out into free space. The rearward wave motion travels backward, hits the rear panel and is then bounced back, with the greater proportion of the reflected sound coming directly back upon the speaker diaphragm, the rest being randomly scattered throughout the enclosure, and hitting the other walls of the structure.

We are concerned first with the reflected wave motion coming back directly upon the speaker. Let us suppose that the loudspeaker is playing a note of 900 cps. This frequency corresponds to a wavelength of approximately 15 inches. Now, it so happens that 15 inches is rather a common depth for an enclosure of this type, and this leads to some interesting situations. For the sake of illustration, let us assume that the loudspeaker diaphragm is flat. When the 900-cps wave leaves the back of the diaphragm, it travels across the depth of the cabinet and reaches the back panel at exactly the same time the loudspeaker is ready to generate its next wave. Upon hitting the back panel, the wave motion is reflected back toward its source. In the process of being reflected from the panel, the wave motion is turned out of phase with respect to its original pulsation. Now, in its reversed phase, the reflected sound reaches the back of the diaphragm again just as the diaphragm is about ready to start its next cycle of motion. This produces interference at the diaphragm between the reflected out-of-phase wave and the new wave. The net result is that a *severe dip* will be introduced into the output of the speaker in this frequency region. Similar cancellation effects would occur for those frequencies at which the distance between the speaker diaphragm and the back reflecting panel is a half wavelength, one and a half wavelengths, two wavelengths, or any *even* number of quarter wavelengths. When the distance is such that it is an *odd* number of quarter wavelengths of the sound being radiated, reinforcement takes place and an accoustic *peak* occurs.

However, the diaphragm of the speaker is not flat. The diaphragm is generally of a conical shape. This means that the wavefront traveling toward the rear is not a plane wave, but is somewhat spherical. All parts of this spherical wave do not hit the back panel at the same time. The forward bulging area of the wave hits first, then the other zones of the sphere make contact. Consequently, the reflected wave is sent back not in one plane but in a correspondingly varied spherical pattern, and accordingly, it reaches the back of the diaphragm over a period of time rather than all at once. In addition, the back of the diaphragm, being generally conical, presents different path lengths between its various sections and the back of the cabinet. As a result of (a) these path differences from the back of the cone to the cabinet back and (b) the spherically delayed wavefronts, the effect of the interaction of the reflected wave on the diaphragm is such as to produce a succession of fairly closely spaced *dips* and *peaks* in the mid frequencies, where the wavelengths are comparable to the various distances from the back of the cone to the back of the cabinet.

In enclosures with large flat backs that do not have sound absorbent material applied to them, the mid-frequency region of response of the system suffers from these irregularities. There are, of course, reflections set up within the cabinet between the other panels of the enclosure. Because the sound glances off obliquely from one panel and then subsequently goes from panel to panel like a billiard ball bouncing from one cushion to the other, the path length, before the reflection finally comes back upon itself, may be quite large. This means that low frequencies will "roll" around the enclosure if they are not damped, and the enclosure will sound like a rain barrel. It is thus advisable to use sound absorbent material freely within the cabinet to eliminate this wall-to-wall bouncing of the sound energy.

Although it is theoretically best to treat all the panels with sound absorbent material, satisfactory results may be obtained by treating only three non-parallel sides. Such treatment will obviously do away with the major cause of repeated reflection, the bouncing back and forth of sound between the same two walls. In any event, the back of the cabinet should definitely be treated, because this panel is the worst offender as far as reflections upon the loudspeaker itself are concerned. Next in importance is the top, because in the bass-reflex enclosure, this is usually the panel closest to the loudspeaker. Any reflection from a surface close to the source of the sound will be correspondingly more intense if the surface is not treated. After the back and top have been treated, either of the sides may be treated.

The application of sound absorbent material within the enclosure does more than minimize reflection internally. It also prevents the panels from being vibrated by the sound pressures impinging upon the panels. The sound absorbing material, being resistive in nature, reduces the intensity of the sound wave as the wave goes through it.

Corner Type Bass-Reflex Enclosure Reduces Internal Reflections

Internal reflection within the bass-reflex type of enclosure may also be minimized by the proper geometrical design of the cabinet and the vertical angle of the speaker mounting baffle. If the area of the back panel is reduced, obviously there will be less direct reflection

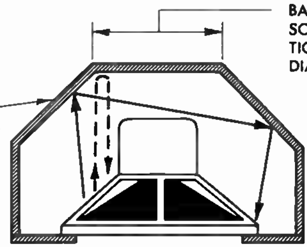


Fig. 14-6. The "cornerless corner" enclosure minimizes internal normal mode vibration and reduces reflections upon back of speaker through its non-parallel wall type of construction.

from the rear of the cabinet back upon the speaker. This effect may be accomplished in enclosures of the "cornerless-corner" type, as illustrated in Fig. 14-6.

This type of construction decreases the overall area of parallel surface both from back to front and from side to side, and inserts instead oblique reflection conditions. The combined effect of this structure is to reduce direct back-and-forth and side-to-side reflection, and to elongate the path of reflection, which tends to reduce the amplitude of the reflection. Fortunately, this type of cornerless-corner construction also makes the cabinet adaptable for placement either against a flat wall or in the corner of a room. This matter of placing the enclosure in the room is an important one and will be discussed in more detail in Part 3.

In addition to breaking up the internal reflections by this corner construction, additional reduction in front-to-back reflection may be obtained by tilting the front panel so that it faces in a direction about ten degrees above the horizontal, as shown in Fig. 14-7. This eliminates the parallel condition between the back and the front and causes oblique reflections to be set up instead, hence the rear of the speaker suffers much less from sound pressures impinging back upon it.

Horns do not Need Absorption Treatment

In the horn enclosure, the problem of sound absorption is somewhat different. Because of the continually expanding sections of the horn, there are seldom any parallel walls between which reflections may be set up, and therefore, sound absorbent material on the walls of the horn is not necessary. In fact, it is possible that the placement of true sound absorbing material within the horn (on the interior walls) may actually be detrimental to the operation of the horn. If the material is truly sound absorbent, it is highly resistive, acoustically.

Since the wave motion that is propagated down the horn is intimately connected with the way the walls of the horn expand, it is possible that the wavefront progressing down the horn would be retarded where the outer sections of the wavefront come in contact with the absorbent resistive material. If this happens, the smooth expansion of the wavefront will be altered and the operation of the horn will

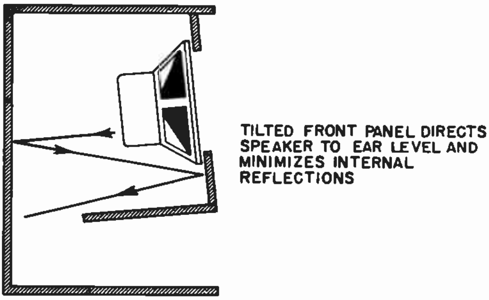


Fig. 14-7. By tilting front panel, internal reflections are again randomized, which reduces normal mode vibration of enclosure and reduces reflections back to rear of speaker.

not be strictly as designed. Soundproofing material, however, may be applied within the compression chamber that houses the driver unit, because in this section the sound pressure is high, confined, and in very close proximity to the back of the speaker.

Suitable Absorbent Materials

There are many types of material that may be used for soundproofing enclosures. Among these materials are Celotex panels, Fibreglas panels, Ozite-type rug padding, Kimsul insulation material, or even furniture stuffing. The Fibreglas and Celotex sections may be easily tacked onto the panel to be treated, using roofing nails with large washers underneath the heads to provide a firm grip on the soft fibrous structure of the material. Old rug padding may be successfully used if it is sufficiently thick and not too tightly packed down. Kimsul is sold at local lumber yards and builders' supply houses for purposes of heat insulation. It is a soft fluffy matted paper material which may be fluffed up into usable thickness and then stapled directly to the panel. The material is sewn together in layers by the manufacturer, and does not come apart when used. Upholstering material may be padded over the surfaces, and held down by porous cheesecloth or loosely woven muslin cloth, and the entire section secured by means of ordinary staples or long carpet tacks with washers beneath the heads. Whatever material is used, as long as it is absorbent, at least ½-inch thick, and applied to at least three non-parallel walls, there will be adequate attenuation of internal reflections within the enclosure.

Grill Cloth Should be Acoustically Transparent

One matter that does not always receive adequate attention in enclosure design is the grill cloth that covers the cabinet. Too often this material is selected on the basis of appearance rather than performance. The grill cloth is just as much a part of the acoustic circuit as any other section of the system. All the sound produced by expensive speakers installed in the enclosure will go for naught if the grill cloth put in front of them destroys the sound before it has had a chance to get out. Especially severe are the high frequency losses that may be caused if the cloth is opaque to these high frequencies. Many a high-fidelity dollar spent on a good tweeter has been wasted by covering the tweeter with an acoustically opaque material. If the grill cloth is too dense, too heavily woven, or too closely woven, high frequency losses will occur. One should also avoid using a woven type of plastic material that looks like narrow seat caning strips. These smooth flat strips may cause reflections back upon the tweeter source if placed close to it, with consequent irregularities in high frequency response.

There is a wide variety of commercially available woven plastic grill cloths made from fibrous plastic filaments that are quite suitable for closing off the face of the enclosure. If, however, the home constructor desires to select a grill cloth material that will be more in harmony with the decor of his home, he must select this material on the basis of its acoustic porosity. It should be relatively thin to the touch. Its acoustic opacity may be roughly checked by inspecting it optically. When held up to the light, the open spaces should be equal to at least 50 percent of the area of the cloth, and the cloth should be loosely woven. In applying the grill cloth to the front panel, it should be tightly stretched and securely stapled or tacked down to the *back side* of the panel, after being pulled around the edges. This ensures that the grill cloth will not loosen up in the course of time. Should the cloth become loose or flabby, it may flutter against the flat panel upon which it rests, and cause unpleasant buzzing on loud signals, especially if it is a hard plastic cloth.

PART 3

THE ROOM

CHAPTER 15: *The Room as Part of the Acoustic Circuit*

The Listening Room Deserves Acoustic Consideration

Unfortunately, too little attention has been accorded the home listening room in the general question of hi-fi reproduction in the home. Probably the main reason for this is that the hobbyist must first be introduced to, and learn to understand, the newest tools of the art — the loudspeaker and the enclosure. Secondly, of course, there doesn't *seem* to be much a man can do with his living room unless he is willing to undertake a major rebuilding and refurbishing project. Fortunately, however, if one has a basic understanding of what it takes to make a good listening room, it will be found that there is indeed much that can be done to make sure that the many dollars spent on the reproducing equipment itself are not blown out of the nearest window or lost in the nearest drapes. The vast store of knowledge that has been accumulated on the acoustic problems of concert halls may readily be applied to the listening quarters in one's home. If we are engaged in making "miniature music halls" in our own homes, as all manufacturers of hi-fi equipment claim, we must extract what we can from this background of knowledge and apply it in home-size doses to get the best possible performance out of our hi-fi system.

Rooms are Basically Enclosures

It will be recognized that a room is an enclosure. There is no basic difference between the room in which we listen and the enclosure in which the loudspeaker is mounted. There are, of course, differences in degree. We will, therefore, find that the study of the room in which we listen is virtually an extension of our previous analysis of baffles and enclosures in general.

It will be recalled that in the wall infinite baffle, the baffle and the room were actually one and the same. Although the other baffles and enclosures described in Part 2 may not seem to be as closely connected with the room problem, our further analysis in the next chapter will show that a great interdependence does exist between these two elements. It will therefore help in understanding the problem of room acoustics to consider a room in the same category as an actual loudspeaker enclosure.

It will be recalled that enclosures fell into two general classifications. The "enclosure" type (such as the bass reflex) is a *resonant* device. It has its own resonating characteristic determined by its size and the opening into it. Then there is the anti-resonant enclosure (such as the horn baffle), which theoretically attempts to attain a truly unresonant condition by approaching as large a horn mouth as possible, consistent with the design flare of the horn. In a similar vein, we may have resonant listening rooms and listening rooms which are anti-resonant. The degree to which they are one or the other is dependent upon their shape and the quality and quantity of sound absorbent material (or lack of it) that is deployed about the room. In general, "live" rooms, rooms that are resounding and reverberant, are more resonant than rooms that are "dead" and muffled.

We must not mistakenly use the word *resonant* as if it were a desirable quality. True, we speak of the fine resonance of a stringed instrument, or the full bodied resonance of a baritone's voice. These are resonances that are deemed desirable because they aid in the projection of the instrument or the voice. However, if a violin had a single peaked resonant spot in the middle of the D-string tones, or if the baritone could ellicit good projection at only one or two notes in his alloted vocal scale, their music would be erratic in volume, unsmooth in general projection, and difficult to listen to. Under these conditions, resonance is undesirable. Just as we controlled the resonance peak of the loudspeaker in order to get smooth operation, so we must temper the resonance of the room so that the room will perform properly.

Rooms May Have Many Resonances

Unlike the loudspeaker with its one major resonance, rooms and enclosures may in general have several resonances or "modes" of vibration. Acoustically, there is no great difference between a rectangular room and a long tube. There is a difference in dimension, of course. In the long tube, the major dimension is its length, and the frequency to which it responds most easily is closely related in wavelength to the actual length of the tube itself. A tube whose length is a half wavelength of the sound being transmitted into it will reproduce that frequency very efficiently from its open end. However, this is not the only frequency that the tube will radiate. It will transmit all the harmonics (overtones) of this one note, if these overtones existed in the original signal, because the tube is still an integral number of half wavelengths of these overtones. In like manner, rooms that are roughly rectangular in shape may have very many natural resonances or modes of operation, depending not only upon the length of the room, but upon the other dimensions as well. These other dimensions can develop their own resonance modes. It is therefore to be expected that the normal living room will exhibit very many natural resonances that are completely a function of the size of the room and its relative dimensions, and partly a function of the liveness of the room.

Live Rooms Have Small Degree of Sound Absorption

Very resonant rooms are characteristically "live" rooms. The best example of a live room in the home is a tile lined bathroom. These hard smooth surfaces reflect sound almost as efficiently as mirrors reflect light. We are all good singers when it comes to vocalizing in the bathroom, because the walls reflect our feeble voices back to our ears and magnify our vocal powers many-fold. Our ears receive the same magnified impression of the sound from the walls of the bathroom that our eyes receive from the large concave mirror used for shaving or cosmetic purposes. The constant and repeated reflection of the sound from one hard wall to the other, with little absorption by these hard smooth surfaces builds up the sound around the ear, giving it life, or "liveness." The fact that this quality of liveness in any room is a function of the normal mode of vibration of the room and the reflection properties of the room leads to a situation that may be detrimental to smooth audio reproduction.

Live Rooms Introduce Sound Pressure Irregularities

The reflection of sound from one wall to the other and the normal mode of vibration of the room set up "hot" spots and null points of

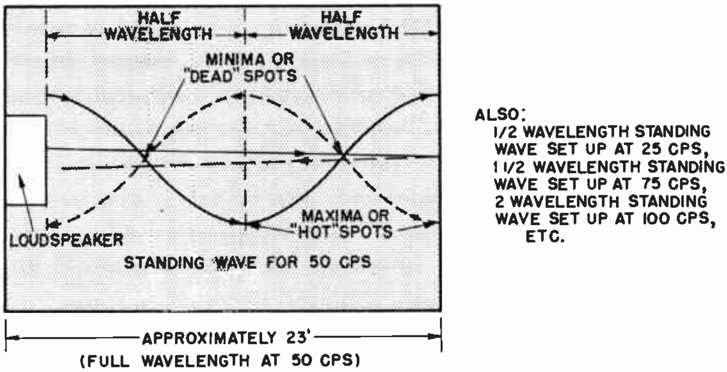


Fig. 15-1. Standing waves for low frequencies may be set up in a room leading to "hot spots" or "dead spots" for different locations along the direction of the standing wave.

sound energy for different frequencies and for different areas of the room. Because rooms are large "enclosures," they vibrate and respond most readily to long wavelengths (low frequencies). Thus, as shown in Fig. 15-1, a standing wave may be produced for a particular frequency. Under these conditions there will be in certain spots a peak condition of sound pressure and in other spots there may be a complete cancellation or reduction of sound pressure. This condition often makes for disagreement as to what one person hears from the system and what someone else hears. For instance, if the listener were at a spot where there is re-enforcement due to the standing wave, he would hear "plenty of bass." If his companion were standing at a point of cancellation, he would hear much less bass.

This is not such a completely hypothetical condition. It is a serious situation that must be recognized in listening to systems. The reader may prove this to himself if he has an oscillator, or even a test frequency record. Select some note down in the 100-cps range and reproduce it at fairly good volume. As the note is being sounded, move slowly across the room, and you will clearly hear the great differences in sound pressure for that note as you change position throughout the room.

As mentioned above, this state of affairs may cause differences of opinion between two listeners located in different parts of the room as to whether or not a certain portion of the low frequency response is missing from the system. In making a truly objective listening test, it is therefore mandatory that a system be heard from many different positions of the room. Obviously, if the listener's easy chair happens

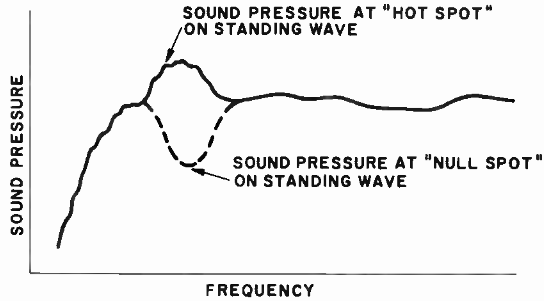


Fig. 15-2. Standing waves may change the manner in which listeners in different parts of the room hear the speaker output. Irregularities and abnormal high and low level pressures are introduced at those frequencies for which standing waves are set up.

to be located at a hot spot, or a null point, for a particular small band of frequencies, he will consistently be subjected to frequency irregularities in the response of the system. The matter of the location of the listener is thus of considerable importance not only for low frequencies but in connection with dispersion of the high frequency beam from the loudspeaker system as well. Secondly, the actual placement of the loudspeaker in the room, irrespective of the listening ear, will greatly affect the final performance. These two latter matters will be discussed in the succeeding chapters. It is of importance at this point, however, to realize that because of the resonance condition of the room due to its normal mode of operation and due to reflection from the walls, a live room may produce many irregularities of response and materially affect the smoothness of the system, as illustrated in Fig. 15-2.

Sound Decays Slowly in Live Room: Long Reverberation Time

A "live room," in which there are excessive reflections from the walls, will have a high "reverberation time." The reverberation time of a room is a measure of the way in which a sound decays after the source of sound has stopped. This factor is defined as the time (in seconds) required for the sound energy to decay 60 db, which is equivalent to one millionth of its original power value, as shown in Fig. 15-3. The actual value of the reverberation time of a room is a function of the volume of the room as well as the reflective properties of the walls for a particular frequency in question. Obviously, if the walls are highly reflective, the sound will continue to bounce back and forth with little absorption at the reflective surfaces, and it will take a long time for the sound to reach the low level of −60 db in relation to its original intensity. Similarly, if the room is very big and the walls are correspondingly further away, the sound will not be attenuated as quickly by the absorbing characteristics of the walls, and the sound energy will be sustained longer within the room. Thus, in general, a large room has a high reverberation time constant, and reflecting walls produce a high reverberation time constant.

High Reverberation Time Produces Inarticulation

Now what does this reverberation time mean in terms of listening effects? Let us take the case of the live room, the one that is highly reverberant. As soon as the first note leaves the loudspeaker, it will

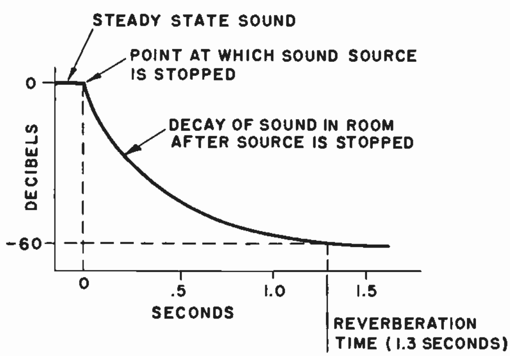


Fig. 15-3. After the sound source has been stopped, the sound in the room begins to decay. The rate at which it decays is a function of the reverberation of the room. Reverberation time is defined as the time in seconds for the sound to decay 60 db.

begin to travel into the room, hitting the walls, ceiling, floor, doors, and windows, and will be reflected back and forth for a long period of time before it is reduced by the necessary 60 db. However, while this first note is bouncing around, other notes are coming out of the loudspeaker. As a result, the first note may still be audible when the second note is produced, and there is overlapping of the notes, as indicated in Fig. 15-4. This leads to indistinct reproduction and poor articulation of the program. Therefore, *too* live a room is detrimental to good musical reproduction, on the basis of cleanness of reproduction of the individual notes.

It would also appear that the degree of overlapping of these notes would be a function of the *type* of music being reproduced. If, as an extreme example, a composition were being played in which there were large gaps between notes as well as a slow tempo, a larger reverberation time might be acceptable, for then one note would not run into the other. However, in a fast staccato movement there would be a severe loss of definition. There is no "optimum reverberation time" for all applications. Therefore, in addition to the factors of room size and reflective conditions (which determine the reverberation time of a

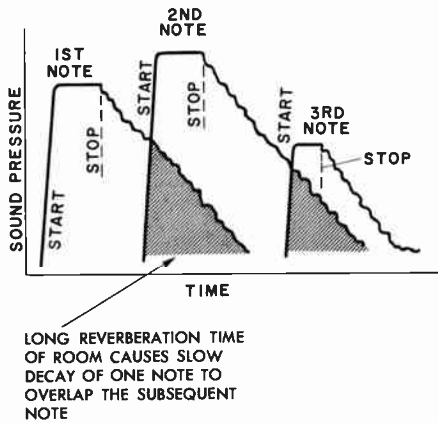


Fig. 15-4. When a room is very "live" its reverberation time is high and the decaying sound overlaps subsequently heard notes. This leads to inarticulation for fast speech and to indistinct musical reproduction.

particular room) we must remember that the actual musical composition will determine whether the reverberation time given by the previous factors is suitable. Although the constructor cannot do anything about the type of music that comes out of his loudspeaker (other than select it, of course) he can do something about his listening room. These matters will be discussed in Chap. 17.

Live Rooms Sound Louder

Another characteristic of the highly reverberant room is that the reproduction seems louder than normal for moderate audio powers. This condition exists both objectively and subjectively. Since a high reverberation time means that the sound is not being absorbed but is moving around in "space," the reflected sound is there waiting for the ear to hear *it* as well as the direct sound coming from the loudspeaker. In other words, the ear receives additional acoustic power because of the reflections. This increases the overall acoustic pressure at the ear, which makes the sound louder to the ear.

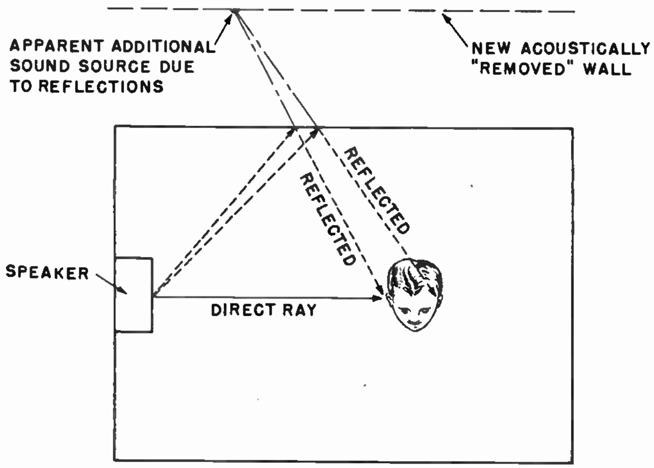


Fig. 15-5. A very live room gives the listener a feeling of largeness of the room due partly to the many acoustic images that appear to come from areas beyond the wall boundaries. This provides an artificial enlargement to the original source of sound increasing the "acoustic perspective" of the music.

Live Rooms Sound Bigger

Subjectively, a very live room also gives the illusion of largeness. A small room with highly reflective walls may produce a feeling of psychoacoustic spaciousness because of the moving around of the sound within the enclosure. We may consider the reflecting walls as creating new "acoustic sources" for the original sound, as shown in Fig. 15-5. The reader will recognize the optical counterpart of this demonstration. The ear in the location shown will hear the direct ray from the loudspeaker and also the reflected ray, which will appear to be emanating from a spot well in back of the wall itself. Thus the highly reverberant room will be psychoacoustically enlarged.

Live Rooms Reduce Sound Directivity

It is also a characteristic of live rooms that the directivity characteristic of the reproducing unit is considerably altered by the reflective properties of the room. Where the walls are highly reverberant, the several radiating beams from the loudspeaker become greatly diffused about the room by successive reflections, effectively increasing the uniformity of the sound reproduction throughout the room in respect to the higher frequencies. (See Fig. 15-6.) In fact, some constructors have made use of this principle in building their tweeter units into small enclosures in the upper ceiling corner of the room so that multiple reflections from the smooth walls in the ceiling area and from the ceiling itself will help diffuse the high frequency radiation.

In the room that it too live, we may say then that deteriorated articulation of the program will occur, that irregularities of low frequency response will be produced within the room, that the system will sound louder, that the sound will appear to come from a larger area, and that the sound in general will be more diffused throughout the room.

Dead Room Has High Absorption of Sound

Let us now examine the other extreme, the dead room. This is a room characterized by a high degree of absorption of the sound at the walls of the room. It is necessary to mention an important difference between the performance characteristics of the dead room and the live room other than that of reflective properties. We discussed above the matter of the normal modes of operation of the rectangular

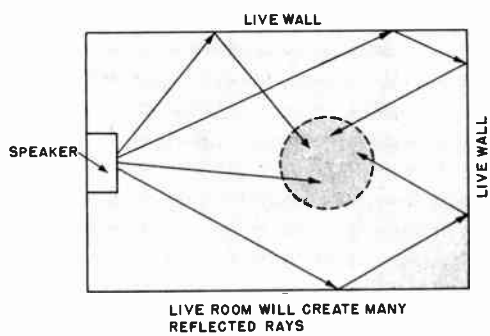


Fig. 15-6. Due to the many reflected rays in a live room the directivity of the sound as it reaches the listener is a function of the room liveness plus the speaker directivity characteristic itself.

enclosure (the natural resonance of the room itself). However, resonances can be set up only where there are reflective surfaces. (For the sake of technical accuracy, it should be stated that an open end of a tube may also be characterized as a reflective surface, inasmuch as its abrupt termination causes impedance discontinuities and wave reflections back into the tube.) In our present instance, where we are dealing with total enclosures, we shall proceed on the assumption that to obtain normal resonances of the room we must have reflective walls. It follows that although the frequencies of the normal vibrations within the room are governed by the dimensions of the room, the intensities of the resonances are governed by the reflectivity of the walls themselves. Suppose the walls were totally absorbing? The entire room would act as a perfect "sink" for the sound energy leaving the loadspeaker. The sound would all be lost in the completely absorbing surface, no waves would be reflected, and there would be no normal mode of vibration of the room. This, it will be realized, is the reason for treating the insides of the bass-reflex enclosure with damping material; it is thus prevented from setting up its own mode of resonance apart from that dictated by its volume and the port opening size. Accordingly, then, a total absorbing room will not only reduce the wall-to-wall reflections but will also reduce the normal mode of resonance of the enclosure.

Dead Rooms Sound Softer

Because of the great lack of reverberance in the room with a very short reverberation time constant, the room sounds dead. In relation to a live room, the sound seems lower in intensity, because the ear does not benefit from any reflected sound. In fact, the sound hitting the highly absorbent walls will be completely lost in the absorbing material, and will be lost to the ear. In effect, the overdamped room is wasteful of acoustic power delivered to it, just as the overdamped loudspeaker enclosure is wasteful of acoustic power delivered to its interior. The analogy between the loudspeaker enclosure and the listening room carries over completely.

Dead Room Sounds Smaller

In the dead room, there is also the psychoacoustic effect upon the ear of the lack of sound coming to it from sources other than the loudspeaker directly. In the case of the very live room, the reflected acoustic images create an apparent source of sound that seems to be behind the walls. This gives a synthetic depth to the source of the sound; it seems to come from a large area. In the dead room, no such reflection exists. As a consequence, the sound all appears to emanate from the one point source of the loudspeaker itself, without benefit of sound images from the wall. The end effect is to produce a feeling of a small sound source playing directly to the listener in cramped quarters. There is no acoustic "spread" to aid the acoustic imagination. The loss of ambient sound around the listener's head thus shrinks the sound source in perspective as well as in amplitude (Fig. 15-7).

Dead Room Makes Sound More Directive

In the matter of the dispersion characteristic within a very dead room, the total characteristic is controlled almost completely by the speaker characteristic itself, and by the placement of the speaker in the room. Where there are no reflections from the wall, there is no diffusive assistance to the various rays of sound coming out of the loudspeaker. In such a room then, the high frequency distribution of the system depends solely upon the loudspeaker and baffle combination. If the speaker high frequency beam is sharp, the position of the

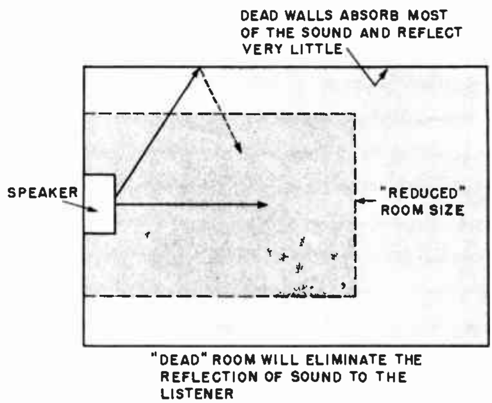


Fig. 15-7. Dead room eliminates sound reflections to the observer, and the sound appears to come more directly from the loudspeaker. The acoustic spread of the sound source is thus narrowed down and the room shrinks psychologically.

listener in the room will determine how much and how many highs he will hear. As far as the low frequencies are concerned, however, the dead room exhibits fairly uniform distribution characteristics free of standing wave effects. There are few low frequency hot spots or null points throughout the room, due to the absence of standing waves. This means that the dead room has *uniform* low frequency response characteristics without hills or valleys, but at the expense of reduced apparent power.

Reverberation Time Depends Upon Room Size

It was pointed out above that the reverberation time of a room is a function of the volume of the room and the absorption characteristics of the wall surfaces. The room size is rather closely related to the degree of reverberation desired in the system. In general, small rooms *should* have less reverberation time than large rooms. When the listener is in a small room, he naturally finds himself closer to the loudspeaker enclosure and closer to the sound reflecting surfaces. Reflection will therefore reach the listener sooner from these close walls than from the walls of a larger room. Consequently, if it is desired to maintain the same clarity of separation between notes in the small room as compared to the larger room, the time taken for the reflected sound to reach the ear in the small room will have to be reduced. Accordingly, the smaller room should have a reverberation time smaller than that of a large room, for the same type of program material. In the large room, where the ear is far from the source of sound and far from the reflected sound, time must be allowed for the reflected sound to reach the ear from the greater distances of the comparatively far away walls, and so the reverberation time may be longer.

CHAPTER 16: *Placing the Enclosure in the Room*

Enclosure Placement in Room Affects Overall Performance

In the previous chapter we discussed the degree of the reflection of sound by the walls of the room in connection with the reverberation characteristics of the room. It was pointed out that the quality of these reflections made the room a very integral part of the acoustic circuit. However, there is one other important manner in which the room affects the acoustical performance of the system, and that is in actually changing the "baffle" characteristics of the enclosure by the placement of the enclosure in certain specific locations of the room. It is with this aspect of the problem that we are presently concerned.

Closely Located Image of Speaker "Enlarges" Speaker Size

It was seen that in a live room the reflected ray of sound energy coming to the listener's ear gave rise to a virtual new source of sound apparently on the other side of the wall. For all practical purposes, the wall created a new source of sound, which, in the discussion of reverberation properties, was credited with lending increased acoustic perspective to the system. However, there is more to the question of reflected sources of sound than their accumulated "perspective" qualities. It is actually possible to increase the low-frequency efficiency of a loudspeaker-enclosure combination by judiciously choosing the proper place in the room for the system. If the sound source produces an image of itself *close* to the original source of sound, the reflected image, which is a real source of sound that works back upon the primary source (the loudspeaker), changes the loading characteristics upon the speaker, *effectively increasing the "size" of the speaker by an amount proportional to the size of the "image."* The acoustic coupling that exists between the primary source (the loudspeaker) and the image is such as to increase the radiation resistance for the original vibrating system, because of the apparent increase in size of the source. The more sources of reflection there are close to the original source, the better the radiation characteristic.

Mid-wall Placement: No Image

For purposes of developing this thought, let us place a loudspeaker in the center of a wall and well above the floor. In this case, the fact that the speaker is removed far from all reflecting walls means that any images formed on the walls are comparatively weak and far from the speaker, and are relatively ineffective in changing the original performance of the speaker. Thus, in this mid-wall position, the operation of the speaker is completely a function of the capabilities of the speaker itself in its wall baffle.

Baseboard Center Wall Placement: One Image, Plus "Two-Sided Horn"

Now let us move the loudspeaker down to the bottom of the wall so it is near the floor, as shown in Fig. 16-1(A). Because low frequencies diffract so readily, the low frequencies from the loudspeaker in the present position will immediately bounce over to the adjacent floor plane, which lies in such close proximity to the speaker. Because the floor is so close, the low frequency beam from the loudspeaker will create a sharp image of itself in the floor and bounce away from the floor. In effect, we now have two sound sources *close to one another,* one original and one reflected. The fact that the floor area where the image is formed does not vibrate while the image is radiating does not in any way detract from its action as a piston source roughly equivalent to that of the original source. Accordingly, we ought to be able to apply the same radiation resistance relationships for this created image as for the original source — the radiation resistance the piston sees is proportional to the frequency times the diameter of the piston. It would seem then that we now have *two* piston sources of a total area almost twice the original area of the

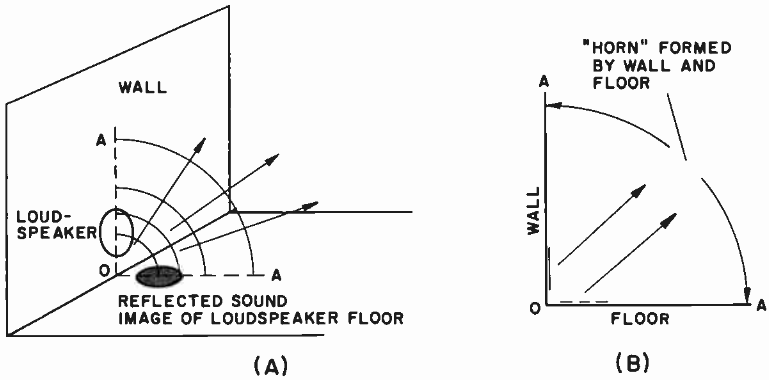


Fig. 16-1. The speaker close to the floor creates an acoustic image of itself. These two sound sources, close together, form a larger single source located at the apex of a "horn" formed by the wall and the floor.

speaker. Consequently, we should expect considerably better low frequency response for this system because it has effectively a product of frequency and area that is twice as large as the original. Although the mutual radiation coupling is not quite as large as this approximation shows, there is sufficiently greater radiation resistance in this position to warrant placing the speaker close to the floor.

This artificially increased piston size is, however, not the only action that comes into play when the speaker is located in this position. If we were to take a cross-section down through the wall and the floor as shown in part (B) of Fig. 16-1, we would readily recognize that the corner formed by the wall and the floor presents a form of horn that feeds the room. The horn thus formed is driven by the original source of sound and the secondary source close to it at the apex of the horn. Although this is not a true horn, it certainly represents a good approach to a horn, much better in fact than the simple speaker placement in the center of the wall. It is fortunate, of course, that almost all of today's loudspeaker enclosures are placed on the floor, and this baseboard "horn" coupling does exist. Those few individuals who use the wall infinite baffle necessarily deprive themselves of the benefits of such auxiliary floor-to-wall loading.

Corner Placement: Two Images Plus "Three-Sided Horn"

We may now extend this concept to the adjacent wall of the room by placing the loudspeaker in the actual corner of the room (but

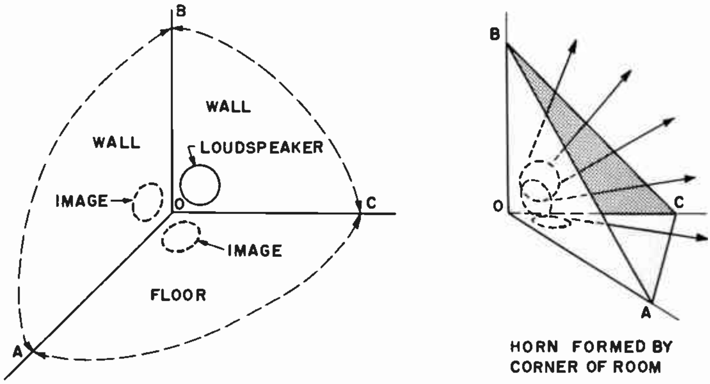


Fig. 16-2. When the speaker is placed down in the corner, two acoustic images are formed on the wall and floor. These multiple sound sources, located in the corner, act as a single larger radiator at the apex of a three sided horn.

still on one of the walls) as shown in Fig. 16-2. Incidentally, this should not be interpreted to mean that we are talking about room corner horn enclosures as such. They are a separate application, which we will discuss in later paragraphs. Our present comments have to do with *any* kind of enclosure or loudspeaker placed in the corner of the room. In this case, there will be two images formed from the original speaker, one on the floor, and one on the other wall. In combination, then, these images and the original speaker all close to one another will result in a total radiation resistance far greater than that of the original speaker, because of the great increase in the effective area of the combined "diaphragms" (the original and the images). Equally important, however, is the improved "horn" that results with the speaker placed in this position. The corner of the room now constitutes a large three-sided megaphone working directly into a room closely coupled to it. Thus the corner of the room causes the speaker-enclosure combination to be better loaded for the low frequencies, because of the coupling of the speaker to the walls in one concentrated location, creating multiple images of itself, which are then horn-loaded by the corner of the room to be projected into the room with considerably improved efficiency.

In Bass-Reflex Enclosure, Port is Aided by Corner

Although in general these conditions hold if the speaker is close to the reflective surfaces, it is often a practical impossibility to achieve such close coupling because of other extenuating circumstances. If the speaker-enclosure system is of the bass-reflex variety, it is often the custom to put the loudspeaker near the top of the cabinet and the port near the bottom. This is done so that the high frequencies that come directly from the face of the loudspeaker may be more nearly at ear level. With the speaker at ear level, the listener obtains the maximum benefits of the high frequency response of the system. This placing of the speaker at ear level substantially removes the speaker from *floor* coupling, although the wall coupling still exists. However, if the port of the enclosure is designed to couple close to the floor, the benefits of wall and corner coupling are obtained through the radiation of the port rather than of the speaker itself into the floor area. If the enclosure is of the cornerless-corner type, or even if a rectangular type is placed in the corner, the port will still engage the corner of the floor, and the speaker itself will engage the two adjacent walls, and very good corner coupling of the enclosure to the room will result, with commendable improvement in low frequency response.

There may, of course, be some cases in which it is impossible to place the enclosure in the corner, and placing it against the flat wall area is a necessity. This may, of course, be done only with the type of enclosure that does not employ the walls of the room as part of the enclosure. With the self-contained enclosure, which has all its own wall sections, whether it is the bass-reflex type *or* the horn type as shown in Part 2, Fig. 8-2(C), placement at the center of the wall is possible. The question that arises here is the choice of the wall against which to place the enclosure. Shall it be against the long wall or the short wall?

This question may be resolved by again considering the wall and the floor as constituting a two-sided horn, feeding the room. If the enclosure is put against the long wall, the horn is more "open-sided" than if the short wall is used. That is, the side walls of the horn are widely separated. If the enclosure is mounted against the short wall, the horn is not quite so open, for the other side walls are closer to the enclosure and produce more of a horn effect feeding the room. It is, therefore, advisable to place the enclosure against the *shorter* wall of the room, if it cannot be placed in the corner.

Inasmuch as the corner of the room appears to offer such favorable conditions for the reproduction of low frequencies, it is only natural that enclosures have been developed specifically for the corner. In fact, these enclosures, designed to operate *only* from the corner, actually use the walls of the corner of the room as part of the horn

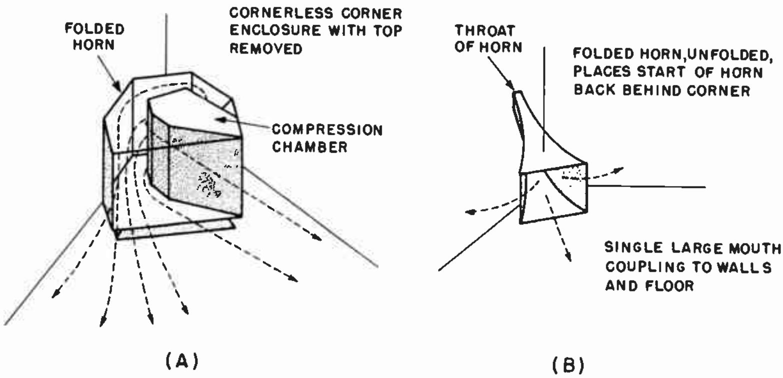


Fig. 16-3. A folded horn when placed in a corner actually puts the start of the horn behind the corner, which effectively elongates the corner of the room into a better horn.

construction. They should not be confused with the construction of the cornerless-corner horn, which may be put in the corner if desired, with all the benefits of corner loading, or in the middle of the room, and still function as a pure horn, but without the benefits of corner loading. (See Fig. 8-2D.) The horn intended to operate directly with the corner wall is of the type shown in Figs. 8-5 and 8-6 (the Klipsch and the Lee). What does placing a folded horn of either of these two types in the corner of a room produce in contrast to placing an ordinary box type enclosure in the corner?

Horn Enclosures are Extended by Room Corner

The box type enclosure will, of course, benefit considerably as outlined above. The horn enclosure will benefit much more in proportion. Since these horns are all of the folded variety (in order to obtain the proper length of horn in a reasonable living room size), the actual *acoustic* column length within the horn may be somewhere between five and seven feet. If we could now unfold the horn, leaving the mouth area where it is in the corner of the room and extend the straightened horn towards the back, the horn would obviously extend into the area behind the corner. In other words, if we want to consider the source of sound in the horn, we find its virtual location removed several feet behind the corner with the horn protruding into the corner as shown in Fig. 16-3. By this expedient we have not only elongated the horn effect of the room corner but we have obtained even better

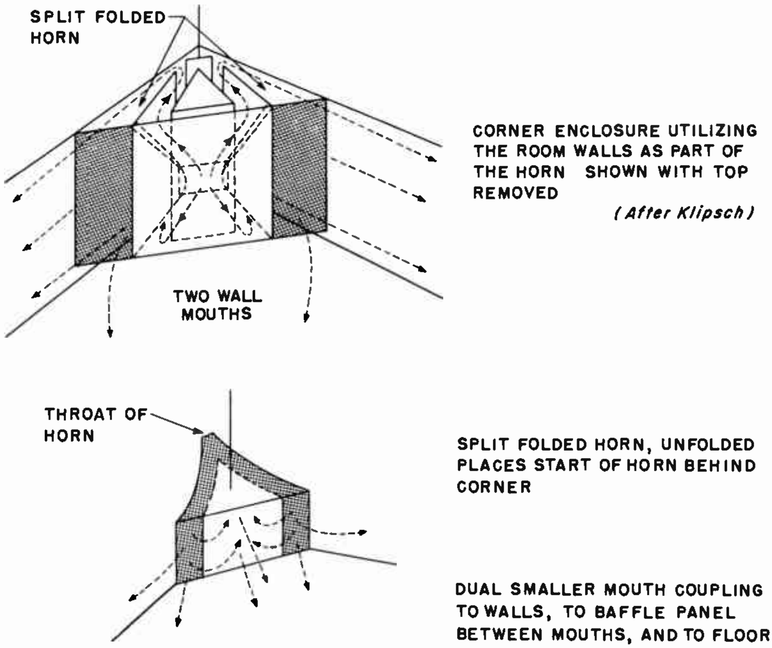


Fig. 16-4. The split folded horn couples into the corner of the room through two mouths, with the horn sections effectively removed behind the corner.

coupling into the room corner. The overall effect of a horn enclosure in a corner of the room is to get high efficiency, low frequency sound well projected into the room.

Inspection of the horn intended solely for corner use (Figs. 8-5 and 8-6) will show that its final configuration is not quite as simple as shown in Fig. 16-3(B). It will be observed that this corner dependent type of horn actually has two mouths on the sides separated by a panel board between them, which actually is the housing for part of the horn. Hence, the horn is split into two symmetrical horns starting from a single throat, folded side by side and coming out of separate mouths. Accordingly, if we were to represent this structure in its unfolded state as it couples into the room, we would have the structure shown in Fig. 16-4, in which are shown two horns of comparatively smaller mouth dimensions coupling into the side walls with a common baffle between.

Comparing these two structures, we see that there is a difference in the manner in which the self-contained horn, Fig. 16-3, and the

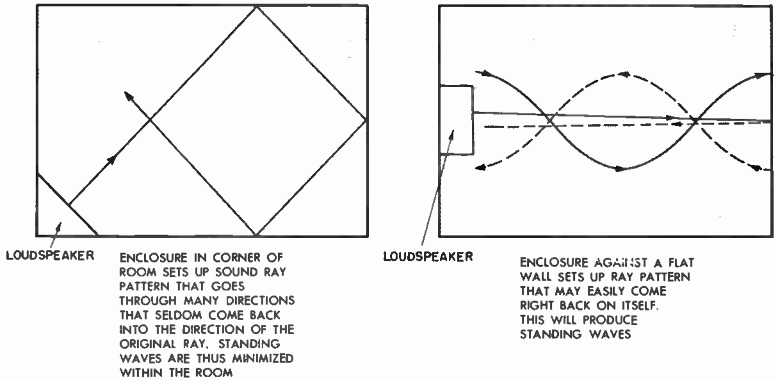


Fig. 16-5. Placing the enclosure in the corner of the room will minimize standing wave pattern and will thus produce more even distribution of low frequencies throughout the room with absence of hot spots or null points, in addition, of course, to better low frequency response in general.

corner integral horn, Fig. 16-4, couple into the room. In the self-contained horn with its large one-piece mouth, the horn is coupled to the two walls and the floor with the sound from the mouth distributing itself rather evenly between the three planes because the mouth is symmetrically placed with respect to them. In the case of the integral corner horn, the sound is more closely coupled to the walls of the corner rather than the floor, with additional coupling from the two mouths to the flat panel between them, which then in turn couples into the corner. In the former, there is direct horn coupling to the corner; in the latter there is combined horn coupling to the walls and "baffle" coupling into the corner. In both cases, however, once the coupling has been established, the corner radiation takes effect and the horn is well coupled to the room.

Corner Placement Minimizes Room Standing Waves

Placing the enclosure in the corner of the room has another beneficial effect upon low frequency reproduction. Corner placement minimizes standing wave effects within the room. Figure 16-5 compares the two paths that may be taken for a particular low frequency sound ray from a corner placement and a center wall placement. The corner placement sends off the rays at an oblique angle to all the wall surfaces, hence the sound must bounce around from wall to wall many times. In this case, the sound energy would seldom find itself travelling back

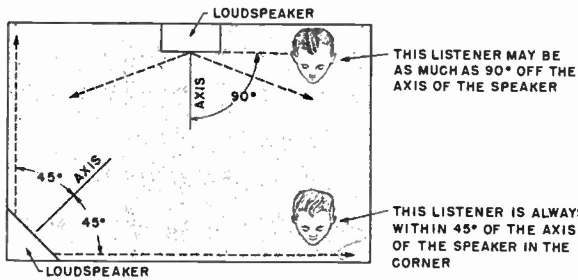


Fig. 16-6. Corner placement of enclosure not only helps lows, but also insures that the listener is always in a very favorable position for the reception of the high frequencies.

toward the loudspeaker along the same line as the original radiated sound. As it is essential for the setting up of standing waves to have the incident sound energy and the reflected sound energy travelling in opposite directions but along the same line of travel, standing waves are not ordinarily set up when corner placement is used. In the case of the flat wall setup, however, the reflection from the wall falls back upon the incident or transmitted energy and standing waves are readily set up.

Corner Placement Improves High Frequency Coverage

Although not specifically affecting the high frequency dispersion from the loudspeaker, placing the enclosure in the corner of the room may provide better high frequency coverage over the room as a whole. There are, of course, high frequency reproducers designed to give wide angle coverage through the design of the radiating horn. The various types of tweeter horns, such as the reciprocating flare type and the diffraction type, do generate considerable high frequency sound energy at a wide angle from their horn axis. However, despite the spread that may be obtained from the horns designed for this application, there is still much to be gained for the listener if he is placed closer to the axis of the horn. Figure 16-6 illustrates the fact that when the speaker is in the corner of the room, the listener cannot possibly be more than 45 degrees off the axis of the horn, because the corner of the room subtends at the most an angle of 90 degrees. On the other hand, in the case of the wall placement, the listener may find himself as much as 90 degrees off of the axis of the high frequency horn. Consequently, from the corner placement, more uniform high frequency coverage will be obtained throughout the room.

Thus, for high efficiency low frequency reproduction and for more uniform high frequency coverage, it is of specific advantage to place the enclosure in the corner of the room. Whether it is a simple bass-reflex enclosure or a horn-loaded type, it will work better in the corner of the room.

There Should be No Obstacle Between Ear and Sound

Since the ear is just as much an element in the acoustic circuit as the loudspeaker and its enclosure, we must consider the problem of placing the ear in the correct position in the room as well as placing the enclosure in the right position. It should not be inferred that there is only one best place for the listener to sit to enjoy his system. What is important is to realize that there is a certain relationship that should obtain between the loudspeaker, the room, and the ear if all the benefits of a hi-fi system are to be enjoyed.

Obviously, we desire the same effect in the listening room as in the concert hall. It would be completely distracting to a listener in an auditorium if he were to be placed behind a post. Of course, his vision would be hindered, but he would suffer auditory discrimination as well. Essential parts of the musical spectrum would be blocked from his ear and he would not get the full benefit of the music. However, such a situation seldom arises in the concert hall. Such rooms are designed to give a completely unobstructed line of hearing from the stage to the listener, no matter where he sits. This then should be one of the prime objectives in arranging one's high-fidelity music room. As obvious as this precept seems, it is too often violated, unintentionally, of course, to the detriment of the system as a whole.

Now there is a distinction between making sure that there is an unobstructed path of sound between the loudspeaker and the ear, and the principle of point sources of sound. The two are by no means the same. Point sources of sound are to be avoided if realistic concert hall reproduction is desired. By no stretch of the imagination does a point source of sound exist in the concert hall. In the case of the full orchestra, naturally, the sound comes to our ears from a very wide acoustic aspect. Even in the case of a soloist with a small instrument, or a singer, despite the fact that the sound they produce comes virtually from a point source, our ears hear that original sound *plus all the reverberant sound of the hall.* So we hear not a point source of sound, but a diffuse source of sound added to the point source, tending to give an illusion of breadth. Point sources of sound are to be avoided for the sake of acoustic realism. However, direct line of "sight" to all the sources that provide the music should be maintained. If the source is a simple enclosure, there should not be any "pillars or posts" between the listener and the enclosure. An obstruction will simply cause a reduction of the high frequencies because they will not diffract around the obstacle as easily as will the low frequencies. Consequently, the program material will be unbalanced.

There Should be No Absorption Surfaces Close to the Ear

The closer the obstruction to the listener, the worse will be the masking effect upon the high frequencies. As an example that is not too far from typical, consider the listener to be seated in a wing back chair with his head comfortably resting in the corner of the wing. Several things may now happen. Even if his chair is placed so that it faces directly out into the room, the wing section may completely block the sound from coming to one of his ears, and the overall effect will be one of muffled highs. Moreover, if the chair is of the deeply overstuffed variety, the listener's ears will be closely surrounded by highly absorbent material, which will have the same deadening effect as the obstructing wing itself. Perhaps this raises the general question of hi-fi "furniture." Well, to be truthful about the matter, the acoustic properties of furniture have been of considerable concern to the designers of the music halls and concert rooms. They figure quite prominently in the acoustic equations that determine how well the room is going to function. Tables of absorption coefficients for many types of upholstery are available, showing the relative absorption of the fittings, and such factors are closely integrated into the calculation of the reverberation time of the room itself. This matter will be illustrated in the next chapter, where it will be shown how the room may be adjusted acoustically to provide the proper liveness for good all around musical reproduction.

Getting back to the listener's chair, even if it is a hard back slender occasional chair, we may still run into difficulty if the chair is placed against a heavily draped wall. The highly absorbent draped area will cause deadening of the program brilliance because of the elimination of the apparent "wall source sound" near the head. With the absence of the reflective surface close to the ear, all the ear hears is the original sound from the speaker, without benefit of any room diffusion. This results in pinpointing the sound back at the speaker area, a condition we try to avoid. Consequently, despite the overall liveness of the room, if the listener's ears are close to heavily absorbent material, the program will be deadened.

Perhaps the simplest example with which to illustrate the importance of the listener position in relation to the enclosure is to suggest that the reader leave the room in which his system is playing and go into some adjacent room. Immediately, he will sense the loss in brilliance and in the "presence" of the music because he is out of direct line of hearing from the system as a whole. It is thus seen that the combined effect of the listener's position and the loudspeaker-enclosure position in the listening room is one that may very greatly affect the overall performance of the system.

Enclosures and Their Placement for Binaural Listening

Before we leave this specific subject and go on to the matter of adjusting the room to match the system, it is necessary to spend a few moments on enclosure placement in connection with "binaural" or "stereophonic" program material and sources. The question of binaural or stereophonic reproduction is befitting a study all by itself, and is beyond the scope of this book. However, despite the pros and cons of the argument, there is no question that such reproduction is considerably *different* from the more simple conventional program sources. Despite the fact that there is no unanimity of opinion as to the best form of such binaural reproduction, or whether it is really "*n*"-dimensional, the fact remains that added acoustic perspective and spread is produced by these systems, resulting in increased realism of reproduction. There are today several "binaural" broadcast programs which may be enjoyed by individuals who have separate a-m and f-m receivers; also there are binaural discs and tapes for those who have the reproducing equipment to handle them.

The placement of the loudspeaker enclosures for binaural reproduction is somewhat more critical than for conventional monaural reproduction, and the placement of the listener is also of greater importance. If the enclosures are of similar kind (as they should be for optimum performance) they should be placed in parts of the room that are acoustically similar. Thus, if two corner systems are used, they should be placed in two similar corners. If the enclosures are of the non-corner type, they should be placed along a wall that is symmetrically decorated. The objective of this precaution is, of course, to ensure that the acoustic properties of the walls are the same for both enclosure systems so that they will have the same acoustic compensation. For instance, it would be undesirable to have one-half of the wall heavily draped, and the other bare, with the enclosures located in front of these vastly different acoustical fronts, which would change the acoustic performance of the systems.

The enclosures should not be too close together, for their reproduction would then merge into a point source, and the binaural effect would be lost. On the other hand, the enclosures should not be placed too far apart; the various musical sources would then become too "disembodied" from one another and there would be a loss of integration of the musical group. The room should not be too live or the resultant excess diffusion would defeat the stereophonic positioning of the sound sources. The listener should be seated equidistant from both systems. This insures that he will get the optimum balance of sound between the two systems without electrically overdriving one system over and above the other. This is to say that both systems should be played at equal loudness, and that for equal reception from both sources, the listener should place himself roughly equidistant between the two.

CHAPTER 17: *Adjusting the Room to Match the System*

Irregular Rooms Provide Desired Characteristics

Unfortunately, in many instances the room in which the hi-fi system is installed is not determined by how well the room is suited to such equipment. In most instances it is the living room. Although the living room of a particular house may be an ideal place for social gatherings, it might lack some important prerequisites for good acoustic performance. If the loudspeaker enclosure *must* be located in the living room, even though it is originally unsuitable, adjustments may be made to the room to bring it into a suitable condition to do justice to the hi-fi system.

It is seldom, however, that the living room is so poor that the loudspeaker enclosure cannot be installed in it. Perhaps it is more than simple coincidence that the general optimum shape of a good listening room at home is very closely related to those dimensions which also make it easy to look at or to live in. One seldom finds perfectly square rooms in a house because they lack eye appeal; square rooms are also poor acoustically, for they easily set up undesired natural modes of vibration. Long narrow rooms are also seldom found in homes because they cause much difficulty in furniture arrangement and are equally "unpretty" to the eye; acoustically, long narrow rooms introduce tubular resonances that are a function of the length of the room, which is also undesirable. Somewhere in between the perfectly square room and the long narrow room, there is a compromise that is of the right size esthetically and of suitable proportions acoustically. Statistical studies have shown that the room dimensions that provide the best results are those that fall in the ratio of approximately 1 : 1.3 : 1.6 (height to width to length). Naturally, these figures may be juggled with fairly good results. One would hesitate to completely discard a room for hi-fi purposes if its dimensions were 20 to 30 percent away from this optimum, for it should be remembered that this figure holds for a rectangular room only. Many of today's modern living rooms are combined living rooms and dining alcoves built in the shape of an ell, and the above figure of the overall ratio of the dimensions of a good room would not hold for the latter combination. In fact, an ell shape may be better than a perfectly rectangular room, for irregular shaped volumes tend to minimize room resonances. Room shape then is as important as room condition.

The first step in adjusting the room to the system is to *select* a room that is not square but roughly of the above proportions. If, perforce, one finds it necessary to use a room that deviates considerably from this ideal, compensations may be made through the treatment of the room that will considerably ameliorate the otherwise adverse acoustic conditions. However, more often than not, the listener's hi-fi room is close enough to the optimum. Most of his concern will be with the treatment of the room rather than with its shape.

Room Must Have Acceptable Reverberation Time

The prime concern in adjusting a room for satisfactory acoustic performance is to provide it with the right amount of liveness so that the musical reproduction will be real and vibrant, or "lifelike." The qualities that determine the liveness of a room are the physical volume of the room and the degree of reflection and absorption by the walls of the room *in addition to the effects of furniture and people in the room.* The combined effect of all these factors is summed up in what is called the reverberation time of the room. This was previously defined as the time it takes for a given sound to drop 60 db from its original level after the sound producing element has stopped operating.

That the reverberation time of a room may be greatly changed simply by its decorative treatment is certainly well known to everybody. When one walks into an empty apartment devoid of furniture, rugs, draperies, and the like, the least little whisper resounds through the place. However, as soon as the same apartment is made livable with the usual apertenances of home decorators, the "resounding" nature of the room, its reverberance, has been reduced. Speaking more technically, its reverberation time has been reduced.

Reverberation Time Dependent upon Room Size and Program Content

What then constitutes the proper reverberation time for the average listening room at home? Just as there are many factors that determine what the reverberation time *may actually be,* there are many factors that determine what it *should be.* The optimum reverberation time will vary with the frequency of the sound being radiated, with the type of program, and with the size of the room. Perhaps the most interesting of these factors is the matter of the program material. The articulation of a man's speech may be completely obliterated if he talks at a rapid-fire rate in an auditorium that is highly reverberant. His echoes come back in such profusion that his original delivery is completely lost. On the other hand, if slow organ music were to be played in the same auditorium the room would properly re-enforce the musical characteristic by allowing the pipes to "resound" throughout the edifice. Because of this question of the program material, we see at once that the proper reverberation time for a room must be selected by judicious compromise. In fact, researchers in the problem of room acoustics have actually classified the proper reverberation time of a room in terms of what composer's work is being performed.

Of course, even the acoustical engineer cannot be that precise in designing his buildings, even though they may be devoted entirely to the performance of a particular type of music. He may, however, use a sliding scale of optimum reverberation time for particular types of reproduction such as small classrooms intended for simple tutorial purpose, through medium sized auditoriums for solo work or small ensemble groups, to large concert halls and opera houses for the production of massive works. These sliding scales have been worked out with statistical methods after very many researches on existing structures of known acoustic quality of performance, in addition to laboratory analysis of the factors circumscribing the problem. From all these studies there has evolved a set of figures that are applicable to the home as well as to commercial buildings.

For the home, the reverberation time that will produce optimum listening conditions ranges from 0.75 second to 1.25 seconds, depending upon the room size and the type of music. These factors are tabulated in Fig. 17-1, from which is will be noted that for a given

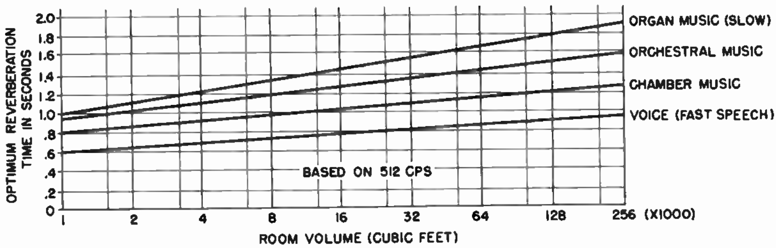


Fig. 17-1. The optimum reverberation for a given room depends upon the size of the room and the type of sound being produced. A large room requires longer reverberation time. "Fast" music requires short reverberation time.

size room, organ music requires the longest reverberation time, while speech material requires the shortest reverberation time. How then do we go about achieving this desired room characteristic?

Liveness of Room may be Adjusted by Absorption Materials

As complex as the art may be that enables us to make the necessary adjustment to the room for optimum listening pleasure, the application of the principles is exceedingly straightforward and simple. In order to apply these principles intelligently, however, it is necessary to understand them in their simple technical aspects. We know that we can make a room live by making the walls reflective, as are the tile walls of the bathroom, or we can make a room dead by making the walls absorbent, as in a heavily draped foyer. The reverberation time depends then upon the amount of absorption of sound by the walls. In the acoustic laboratory there are many ways in which the reflection or absorption characteristics may be measured. As in all measurements, there must be a standard unit against which other quantities are measured. Dimensionally we use the inch; electrically we use the ohm. In acoustics, we use the term "sabin" as the unit in which absorption of materials is measured. This term stems from the name of the man who did much of the pioneer work in the field of room acoustics, Wallace Sabine.

One of the problems with which Sabine concerned himself was the matter of the absorption characteristics of materials. As in all measurements there had to be a standard "yardstick" of absorption. Sabine decided to use literally "nothing" as his yardstick against which to measure absorption. His "nothing" was simply an open window.

Obviously, if the window is completely open it can reflect no sound at all; all the sound approaching it must go through it. The open window must then act as a perfect absorbing device, for it reflects no sound at all. Here then is the unit against which other materials may be compared. If a slab of material placed in the open window reflects half the sound that hits it, the other half must have entered the material to be completely absorbed by it and/or subsequently transmitted through it. Then we may say that this piece of material has an absorption coefficient of 0.5 as compared to the open window. The higher the value of the absorption coefficient, the more absorbent is the material (maximum value of 1.0 for the open window).

Absorption Varies with Type and Amount of Material

In this manner the absorption of many materials may be measured and tabulated for use where a knowledge of the sound absorbent properties of the materials is necessary for the computation of other acoustical data. Such a table is given in Fig. 17-2, listing the absorption coefficients of various types of material that may be found in the average living room, or for that matter in any room of the house. In addition to these we have listed the total absorption units of objects and individuals in the room; after all, not only do their ears absorb the sound, but their bodies absorb sound energy. Note that *absorption units* differ from *absorption coefficients;* the latter are simply numerical ratios with a maximum value of unity, showing what *proportion* of the sound is absorbed by the exposed surface; the former units show the *total* absorption effect of an object on the sound energy. One unit of sound absorption is produced by a material with a surface area of one square foot and having an absorption coefficient of 1.0. We will have more to say about this audience problem in later paragraphs. Our present concern is how to make use of the above tabulated data in adjusting the room acoustically.

A Simplified Approach to the Room Adjustment Problem

Stating the problem directly, we wish to determine how much and what kind of material we need to apply to a room of a particular size to achieve the optimum reverberation time as called for by the type of music we are going to reproduce in the room. Before going into the method of solving this problem, it is necessary to comment on the approach to the problem. It is possible to make a very accurate

COEFFICIENTS OF ABSORPTION FOR USE IN APPROXIMATING THE TOTAL ABSORPTION OF A ROOM FOR PURPOSES OF ADJUSTING IT TO THE PROPER REVERBERATION TIME (SEE TEXT)

	MATERIAL	128 CPS	512 CPS	4096 CPS
ABSORBANT MATERIALS	FIBREGLASS (1" THICK)	.17	.91	.77
	ACOUSTIC CELOTEX (1")	.25	.99	.50
	WOOL CARPET ON PAD	.20	.35	.40
	DRAPERIES			
	HUNG FLAT — LIGHT WEIGHT	.04	.11	.38
	HUNG FLAT — MEDIUM WEIGHT	.05	.13	.37
	HUNG FLAT — HEAVY WEIGHT	.05	.35	.37
	DRAPED TO 1/2 WIDTH — LIGHT WEIGHT	.06	.40	.58
	DRAPED TO 1/2 WIDTH — MEDIUM WEIGHT	.07	49	.62
	DRAPED TO 1/2 WIDTH — HEAVY WEIGHT	.14	.55	.62
REFLECTIVE MATERIALS	CONCRETE UNPAINTED	.01	.02	.03
	BRICK WALL UNPAINTED	.02	.03	.06
	WOOD FLOOR	.04	.03	.02
	LINOLEUM ON SOLID FLOOR	.04	.04	.02
	PLASTER ON METAL LATH AND WOOD STUDS	.04	.04	.04
	GLASS	.04	.03	.02

INDIVIDUAL OBJECTS	TOTAL ABSORPTION UNITS		
MOHAIR TYPE UPHOLSTERED CHAIR	2.5	4.5	4.8
ALL WOOD CHAIR	.18	.24	.37
ADULT PERSON, STANDING	2.5	4.2	5.00
ADULT PERSON IN UPHOLSTERED CHAIR	3.0	4.5	5.2
CHILD	1.8	2.8	3.5

Fig. 17-2. Absorption properties of material in the home, including individual objects. Although the absorption varies with frequency, the value given for 512 cps may be used satisfactorily because of variations prevalent even in similar materials.

analysis of the treatment necessary for a room if all the pertinent statistics are available. It is not merely enough to know that a wall is plastered in order to know what its absorption coefficient is. There are several different types of plaster which may have different absorption characteristics. Even the way the plaster is supported will make a difference in its acoustical properties. Plaster on hollow tiles will have one characteristic absorption. When applied over metal lath and wooden studs, its absorption may be twice as high. Moreover, the actual factors of absorption of the various materials must be obtained from the manufacturers' published data concerning them, which often are not consistent. For reasons such as these, an exact paper analysis of a room and its treatment is at best a judicious approximation. It may be said, of course, that the professional man has available instruments for measuring the sound decay characteristics of rooms, enabling him to determine accurately the reverberation time of the room before and after treatment. With such instruments, the room may be very precisely adjusted.

However, where the layman must make use of published data concerning material and make educated guesses as to the type of walls he has in his room, or the quality of the wood on his floor, or the grade of the drapes that hang from the moldings, precision analysis is out of the question. However, this is not to infer that he cannot arrive at a close and workable adjustment to his room by making these educated guesses. Furthermore, because of the necessary approximations that must be made in the selection of the numbers that go into the solution of the problem, it is of advantage also to simplify the theoretical approach to the problem, so that the layman may proceed with a straightforward solution to the problem of adjusting the room for optimum listening conditions. The results obtained from the following simplified treatment of the subject will be within 10 to 15 percent of the more rigorous solution, but inasmuch as the numbers with which the layman will have to work are in many instances no more accurate than this, the simplified attack is fully justified. The problem to be worked out will demonstrate the workability of this rule-of-thumb method of adjusting the room.

Illustrative Problem on Room Adjustment

Stating the problem directly then, what we wish to determine is how much and what kind of material do we need to apply to a room of a particular size to achieve the optimum reverberation time, as called for by the type of music we are going to reproduce in the room. Figure 17-3 contains a chart that enables us to determine the necessary total absorption units of a room of a given size for the particular type of music for which optimum results are desired. Let us choose the condition that will probably be the one most often selected — the average music condition. (It may be stated parenthetically, however, that by incorporating the proper variable adjustments in the room, we may convert it from an organ recital room to a speech room. This we shall discuss in due time.) Let us choose a room that is meant for average music, and which has dimensions of 10 feet high, 15 feet wide, and 20 feet long. What must the absorption of the room be? From the above dimensions, the volume of the room is 10 feet × 15 feet × 20 feet, which is 3000 cubic feet. For this volume, the chart shows

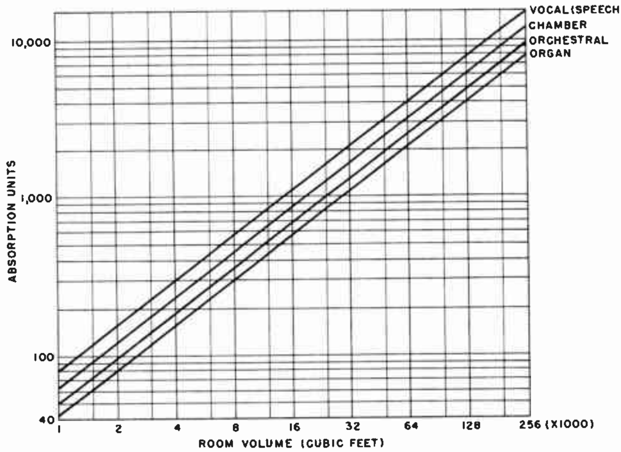


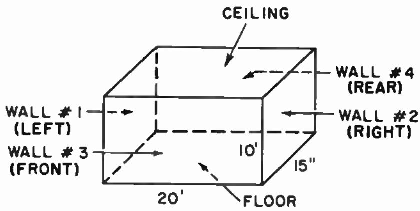
Fig. 17-3. Total absorption units required for a given room volume for different types of program material.

that the room will have to have a total absorption of approximately 150 units for average orchestral music.

We have, now, a quantitative measure of the room that will produce the desired results. Our objective is to adjust the room to this value. The next step is to use the data concerning the absorption characteristics of various types of material to realize this objective. The *total* absorption characteristic of a piece of material depends not only upon its absorption coefficient (how well it absorbs in reference to an open window) but on the expanse of the piece of material as well. Thus, for a slab of gypsum plaster, which has an absorption coefficient of .04 and an area of 50 square feet, the total absorption of the section is $50 \times .04$, which is 2 absorption units. Now all we have to do is total up the absorption characteristics of all the surfaces of the room and change them, in part, so they equal the value we need (in this problem) of 150 units.

Completely Bare Room has too Little Absorption

For simplicity of illustration, let us assume that we start with a completely bare room, consisting of a wooden floor, four plaster walls and a plastered ceiling for the area in question of 3000 cubic feet. Figure 17-4(A) illustrates this room, and at the same time tabulates the areas of the walls, ceiling, and floors along with the absorption coefficients for these sections. Notice that the individual area is mul-



(A) LIVE ROOM

SAMPLE RECTANGULAR ROOM IN COMPLETELY UNTREATED STATE

	MATERIAL	AREA	ABSORPTION COEFFICIENT	TOTAL ABSORPTION
CEILING	PLASTER	15 X 20	.04	12
WALL #1	PLASTER	15 X 10	.04	6
WALL #2	PLASTER	15 X 10	.04	6
WALL #3	PLASTER	20 X 10	.04	8
WALL #4	PLASTER	20 X 10	.04	8
FLOOR	WOOD	20 X 15	.03	9
		TOTAL ABSORPTION UNITS = 49		

(B) DEAD ROOM

SAMPLE RECTANGULAR ROOM IN COMPLETELY TREATED STATE

	MATERIAL	AREA	ABSORPTION COEFFICIENT	TOTAL ABSORPTION
CEILING	PLASTER	15 X 20	.04	12
WALL #1	DRAPES IN FOLDS	15 X 10	.55	82.5
WALL #2	DRAPES IN FOLDS	15 X 10	.55	82.5
WALL #3	DRAPES IN FOLDS	20 X 10	.55	110
WALL #4	DRAPES IN FOLDS	20 X 10	.55	110
FLOOR	CARPET	20 X 15	.35	105
		TOTAL ABSORPTION UNITS = 502.0		

Fig. 17-4. Sample calculations showing a given room in a completely untreated "live" state has few absorption units. Same room when completely overtreated has a high number of sound absorption units and is very dead.

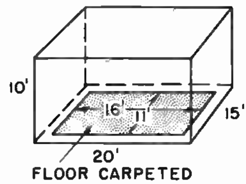
tiplied by the absorption coefficient for the material of which that area is made to obtain the total absorption units for that area. After each area is calculated, they are all added together to give the final absorption of the whole enclosure. For the present instance, this figure turns out to be approximately 49 units, which is far less than the required 150 units. This means that there is not enough absorption, that there is too much reflection, that the room is too live. This is what we would expect from the postulated conditions of a bare room.

Completely Treated Room has too Much Absorption

Now let us go to the other extreme for this room and apply full treatment to the floor, walls, and ceilings. Our new conditions now will be those in which the floor is heavily carpeted from wall to wall, the walls heavily draped with velour trappings gathered in folds (draped to half their total area), and the ceiling treated with conventional sound absorbing acoustic tiles. The tabulation for this set of conditions is given in Fig. 17-4(B). Again all the various sections have been computed individually and then added up to give a final absorption of 502 units. This is far in excess of the required 150 units, and the room will be completely dead, as we probably expected from the statement of the problem. Somewhere between the totally bare room and the totally treated room is the final answer. It should be fairly apparent at this stage how the final treatment of the room is resolved. We select the right amount of decorator's material not only to match the decor of the room but also to provide the right number of total absorption units so that these materials in conjunction with the untreated areas of the room approach the correct value.

Selecting the Absorbent Material

To complete the problem, let us assume that the ceiling will remain in its original plastered condition, and that the floor will have 16 foot $\times$ 11 foot carpeting. How much drapery treatment will have to be added to the walls to arrive at the figure of 150 absorption units? These primary conditions of the problem are set forth in Fig. 17-5(A), where it will be seen that the total absorption of the room in this state is 106 units. At some place in the room we have to add 44 units of absorption. Looking at the chart of Fig. 17-2, we see that a heavy weight velour drapery cloth has an absorption coefficient of 0.35. If we divide this factor into the total additional units needed, which in



(A) ROOM BEFORE ADJUSTMENT

ROOM OF FIGURE 17.4 WITH FLOOR CARPETED, OTHER WALLS UNTREATED

	MATERIAL	AREA	ABSORPTION COEFFICIENT	TOTAL ABSORPTION
CEILING	PLASTER	15 X 20	.04	12
WALL # 1	PLASTER	15 X 10	.04	6
WALL # 2	PLASTER	15 X 10	.04	6
WALL # 3	PLASTER	20 X 10	.04	8
WALL # 4	PLASTER	20 X 10	.04	8
FLOOR	CARPET,	16 X 11	.35	62
	WOOD BORDER	124 SQ. FT.	.03	4
		TOTAL ABSORPTION UNITS = 106		

(B) DESIRED ABSORPTION UNITS = 150 FOR THIS ROOM (10 X 15 X 20 = 3000 CUBIC FEET) FOR AVERAGE MUSIC AS OBTAINED FROM CHART IN FIG. 17-3

(C) TOTAL SQUARE FEET OF ADDITIONAL TREATMENT

$$= \frac{\text{ABSORPTION UNITS NEEDED}}{\text{ABSORPTION COEFFICIENT}}$$

(FOR HEAVY VELOUR DRAPES FROM FIG. 17-2) $= \frac{150 - 106}{.35} = \frac{44}{.35} =$ 126 SQUARE FEET

(D)

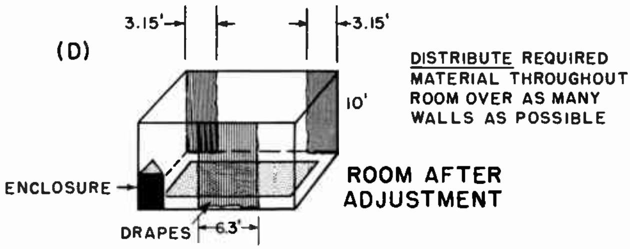


Fig. 17-5. Simplified method of adjusting the absorption of a room to produce the optimum reverberation for its size.

this case is 44, we will get the total square footage of this material that will provide the necessary additional absorption. Thus 44 divided by .35 equals 126 square feet, the total area of the drapery necessary to bring the room to 150 absorption units.

Since the room is 10 feet high, we need 12.6 running feet of this material. This amount of material may then be divided into several sections, say into four 3.15-foot sections, and these sections may then be hung on the several walls as befits the arrangement of the room with its doors and windows, as shown in Fig. 17-5(D). Of course, in the hanging of the drapes, we have removed (covered up) some of the active surface of the bare plaster walls so there will be some reduction in the overall absorption of the room from the calculated value. However, in relation to the absorption characteristics of the covering material, the amount lost from the bare walls themselves is rather insignificant, and may be neglected. One may, of course, if he desires to be very precise, calculate the amount of bare wall absorption that has been done away with because of the drapes (in this case 126 square feet) and add to the original figure for the drapes an additional amount to compensate for this loss.

Windows and Doors are Important

The reader will notice that we have left out some very important features of a room, namely the doors and windows. If the opening into the room is a simple archway, the room into which the archway leads becomes part of the acoustic problem, and its own characteristics must be added to the original room for the final evaluation of the necessary treatment of the combination. If, however, the opening into the room is a conventional door, the survey of the absorption units must be made on the basis of the absorption characteristic of the door material added to the other absorption factors, provided the door of the room is to remain closed when the hi-fi system is being used. On the other hand, if the door is open, the room beyond naturally becomes an acoustic element in the problem, as in the case of the archway.

Windows are treated in the same fashion. When the windows are closed, the room sees panels of glass, which have a particular absorption characteristic that must be added to the calculations. If it is summer and the windows are open, they constitute exceptionally good "absorbing" panels, for all the sound goes through them and none is reflected. It is remarkable how an open window will change the listening characteristic of a room. If, on a warm summer night, four living room windows are opened, and the open area is 9 square feet, the total open window area will be 36 square feet, with an absorption power of 36 units. If we originally designed the room for 150 units, it is thrown off by approximately 25 percent. For all practical purposes, the windows are devices that allow the room acoustics to be changed at will, although this was naturally not the original purpose of the window. There are more specific devices that may be used as manually adjustable elements within the room, but before we consider these, it is necessary to discuss the one or two more items that have been left out of our reckoning.

Furniture may be Highly Absorbent

In the main, the furniture of the room remains fairly well fixed in quantity. There are the usual easy chairs, sofas, side chairs, breakfronts, and similar pieces in the average living room. These act as absorbing devices in exactly the same manner as do the walls of the room, and their characteristics may be approximated from the type of material from which they are made. Evidently, a large, well upholstered chair will be a good sound absorber, while a straight-backed wooden chair will be a poor absorber. Absorption figures for typical chairs are given in Fig. 17-2.

Liveness is Affected by People in the Room

Now we get down to the matter of the people themselves. The absorption of the average human body will, of course, vary with its covering material as well as its size. An adult presents more of an absorption area to sound than does a child, and the clothes he wears similarly affect the amount of absorption. It will be realized from this simple statement how very approximate many of these calculations must be, for it is impossible to list all combinations of sizes and modes of dress in terms of sound absorbing constants. We must accept a figure that is representative of the average. Thus the adult represents a total of 4.2 absorption units when standing. When seated in a chair, he merges with the chair; if the chair is an upholstered one, the total absorption figure is only a little more than the chair itself represents.

Of importance, however, is the matter of the number of people in a room at any one time. If the room has been adjusted to the proper reverberation time for the immediate members of the family as the listening audience, an influx of ten party visitors will immediately add

40 to 50 absorption units to the room. This is a condition we often recognize when the guests pile in, for the sound becomes deadened and lifeless, and we invariably have to turn up the volume if we want to continue to have usable music coverage. The reverberation time has gone down because of the absorption added to the room.

Having learned how to permanently adjust the listening room, what can we do to provide a measure of adjustibility to take care of contingencies that arise due to seasonal changes, for instance, with open doors and windows, or social affairs with their many guests, or even to different types of music. To have such adjustable devices is to add to the versatility of the high-fidelity system as a whole. A room liveness control is perhaps just as important as a treble boost control on an amplifier, or a presence control on a multi-speaker network. The liveness of the room is the final closing link in the acoustic chain between the loudspeaker and the ear.

Drawstring Drapes Provide Controllable Absorption

One of the easiest ways of providing variable acoustic adjustment in the home is to use drawstring drapes. There is a difference in absorption effect between a drape that is pulled out flat and one that is pulled together into folds. As shown in Fig. 17-2, a medium weight velour drape may have close to four times the absorption coefficient when it is bunched into half its area than when it is simply hung without folds over its entire area. Thus, although we cut down by half the total absorbing area when we bunch up the drape, we increase its absorption by a factor of 4; or we have a total gain of 2 in the absorption of the drape *in folds.* If the flat drape has an area of 160 square feet with an absorption coefficient of 0.13, its total absorption will be 160 × 0.13, which equals 21 units. Now, if we bunch up the drape to half its area and apply its new absorption figure, we get 80 × 0.49 = 39 absorption units. If we add to this figure the original absorption characteristics of the wall that has now been exposed, 80 × .04 = 3 units, we have effected a total increase of absorption from 21 to 42 units simply by bunching the drapes. We have attained considerable control over the room by a simple means.

Venetian Blinds may be Used as Absorption Control

There is another simple expedient that the man at home may use to adjust his room if he has any reasonable area of venetian blinds over his windows. Some of today's homes utilize large picture windows in their living rooms, and frequently they are equipped with full width venetian blinds covering what amounts to a large expanse of wall. The blind may be used as an adjustable acoustic control by treating one side of the slats with an absorbent material and leaving the other side untreated. Suppose we had a window 8 feet long and 5 feet high. This amounts to an expanse of glass reflecting surface of 40 square feet. If the blind were pulled all the way up, or if it were left in the maximum open condition (slats horizontal), the room would feel the full absorption (or lack of it) of the glass. The figure for this would be $40 \times .03 = 1.2$ units, which provides very little absorption. Now suppose we left one side of the (wooden) blind untreated, and treated the other side with some common highly absorbent acoustic tile material cemented to the slats. We thus obtain a wide adjustment of acoustic absorption by varying the attitude of the setting of the slats of the blind. When the slats are all closed, with the smooth wood surfaces facing the inside of the room, there will be a probable absorption of approximately $40 \times .2 = 8$ units (taking .2 as the absorption coefficient of wood paneling). With the slats oriented so that the tiled sides are facing the room, the absorption would be $40 \times .85 = 34$ units (taking an average absorption coefficient of .85 for the tiles). Thus we can go from an open blind condition of 1.2 units gradually through a rising scale to 34 units.

It will be appreciated that this is simply an adaptation of the commercial means of controlling studio reverberation constants through the devices of rotating panels or sliding wall sections. When properly applied in the home, the method can be just as effective as in the studio. It is conceivable that many more such variable devices may occur to the reader that may fit directly into his home decorating motif. It will be realized, of course, that in a room that is made to be *adjustable* to different conditions, the room must be left live enough at one end of the adjustment so that it will never be overdamped under the most adverse conditions. Starting with this amount of liveness, the adjustable devices may then be designed to provide as much additional sound damping material as necessary to bring the reverberation to the other limit.

Enclosure Corner Should be Very Live

In treating the room for acoustic adjustment, one should be cautious about the manner in which the corners of the room are manipulated. When the corner of the room is to be used as a sound source for the placement of the enclosure, it should be completely devoid of any kind of acoustic treatment. The corner should be considered in the same light as the horn that is to go into the corner, namely that it be strong, rigid, and *non-absorbing* of acoustic energy. Thus, if the corner of the room is draped, much of the effectiveness of the acoustic radiation of the corner will be lost, especially if the horn is one that actually depends upon the walls of the room to complete the horn. These walls must be totally reflective, and so must be the floor for the horn action to be properly established. In the case of the self contained horn, in which the side walls are integral with the enclosure structure, the matter of the corner treatment is not quite as critical, although even in this case better results will be obtained if the corner retains its non-absorbing qualities.

Absorption Material Should be Divided into Several Areas

In making the necessary acoustic adjustment to the room, it is advisable to break up the sound absorbent material into several smaller areas rather than one large area of treatment, where other committments, such as interior decorating, will permit. Such a procedure will scatter the reflective and absorptive properties throughout the room, making the room more uniform from wall to wall. This will produce smoother acoustical performance of the room by the tendency to minimize excessive reflections from what might otherwise be a large bare surface. Reflections of this sort might still exist from such a wall, even though the room as a whole had been compensated by the treatment of some other wall. In general, broken up surfaces, differently treated, and with non-parallel walls (if possible) provide the optimum conditions for smooth room performance.

CHAPTER 18: *Adjusting the System to Match the Ear*

The Ear is a Variable Element

Having gone through all the problems of the loudspeaker, its enclosure, and the environment in which the sound is to be heard, we come to the last element in the acoustic chain. This is the all-important ear, which does the listening. The ear is a variable device. This is not to say that we can vary our acuity of hearing at will; it is not variable in that sense. Perhaps it would be more accurate to call the ear the component with the most variants built into it.

No two people have the same hearing characteristic, any more than they have the same fingerprints. An individual's hearing capabilities may even differ from one ear to the other. There are many factors that affect one's sense of hearing. These factors are the subject of the present chapter, along with means that may be utilized to adjust the performance of a system to an individual's ear so that he may obtain optimum satisfaction from it.

The ear is not a linear device. What it hears, and how well it hears it, depends upon not only the ear itself but in a great measure upon the sound. One of the most important characteristics of hearing is the change in sensitivity of the ear for different frequencies as the sound grows softer or louder. When music is loud, the ear has one response characteristic. When the music is soft, the ear has an entirely different hearing pattern. Therefore, a system that may be balanced for the ear under one condition of loudness may be entirely out of balance for another condition. This is shown in Fig. 18-1. This par-

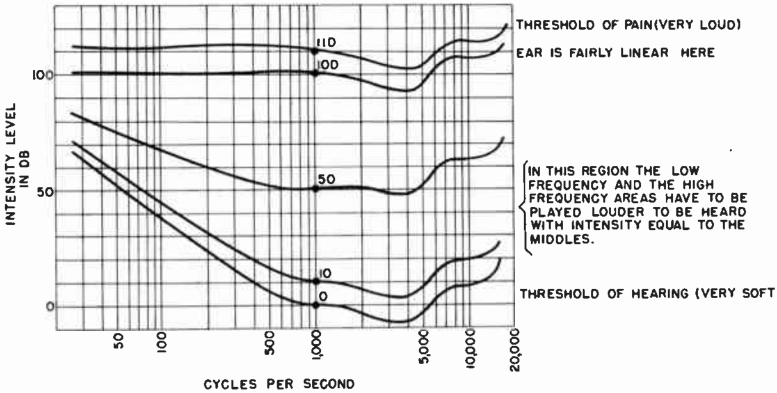


Fig. 18-1. The Fletcher-Munson contour lines of equal loudness for normal ears. The curves show how much louder than the 1000 cps note other frequencies have to be produced to be heard equally as loud as the 1000 cps note.

ticular characteristic is called the Fletcher-Munson curve, after the two researchers who formulated it. This family of curves so well known to the art is indicative of the *average* ear. Any individual's ear may, and probably does, differ considerably from it.

To what practical application may we put this piece of clinical information? Certainly, it shows that what the ear hears under one condition it may not hear under other conditions, and that a system adjusted for one condition may not be properly adjusted for another condition. Let us explore this in a little more detail.

Sensitivity to Frequency Varies with Loudness

Examination of these hearing acuity curves tells us that for very loud sounds the ear hears all frequencies at almost the same intensity; for loud program material, the ear may be considered fairly "flat." It hears the sound exactly as reproduced. As the loudness of the sound decreases, the ear begins to lose sensitivity to the lowest frequencies and the highest frequencies. Therefore, when the level of the music is very low, the ear does not hear the program exactly as reproduced. It hears the middle frequencies fairly well, and loses both ends of the spectrum. Putting it more bluntly, the ear decides to concentrate on a small portion of the sound, as it were, so it may "understand" it better. As lightly put as this thought is, perhaps there may be considerable evolutionary truth to the statement. We do know that most of the intelligibility of sound exists in the middle spectrum area of the frequency curve. We receive perfectly good communicative intelligence from the telephone with a reproduction of frequencies ranging from about 400 cps to 5000 cps. The lower and higher ends are not at all necessary for the ear to make sense out of what it hears. Thus perhaps the ear has learned to adapt itself to very low level sounds in such a fashion that it closes its "consciousness" to those parts of the spectrum that it does not need, and concentrates entirely on the small middle band, from which it may extract the desired message.

Whatever the reason, the fact exists that the ear does have this differential frequency sensitivity with a change in signal level. In high-fidelity practice. however, we are concerned with more than just communicative intelligence. We want the lowest throb from the bass violin and the highest tweet of the piccolo. These low and high limits belong to the musical intelligibility of the sound, and they must be present in the individual's hearing mechanism for the full program to be received by his consciousness.

Adjusting the Volume Changes Frequency Sensitivity of Ear

Suppose we are listening to our high-fidelity system at a good comfortable concert hall listening level (one at which our ears are "flat"). Let us assume that the amplifier response is flat too, with the bass and treble controls set at midpoint. We now hear the program to our ear's psychoacoustical satisfaction. Suppose now that the hour is late, the children are in bed, the neighbors do not like our selection of music, and we want light background music by which to relax. We turn down the volume control to where the music can barely be heard. What do we actually hear? By simply turning down the *volume control* of the amplifier, we have destroyed the wide range high-fidelity characteristic of the *chain* of the system. The ear no longer hears the even rhythmic beat of the bass at the low end, or the tinkling of the triangle at the high end. It hears only the middle frequencies of the music. That this need not be so we will soon show.

First, however, it will be of interest to the reader to try a little experiment for himself. Select a vocal composition, preferably a female vocal rendition backed by a full orchestra. Play the recording at a fairly loud level and listen carefully to the low notes and the high notes that "surround" the soloist. Play one section over several times so that you are familiar with the musical accompaniment as well as the vocal part. Now turn down the volume so that you can *barely hear* the vocalist. Invariably, the ear will completely miss the accompanying low notes and high notes that it heard before. It hears only the voice and the notes immediately surrounding it. The fidelity of the system has been watered down by the poor frequency response of the ear at low listening levels. This situation does not, fortunately, present us with an insoluble dilemma. We can adjust the system either manually or automatically to take care of this characteristic of the ear.

Loudness Controls on Amplifiels Compensate for Ear Deficiencies

Many amplifiers are equipped with "loudness controls" in addition to or in place of ordinary volume controls. There is a definite difference between these two types of controls. The loudness control, sometimes called a contour control, changes the characteristic of the amplifier so that as it is turned down to lower levels of loudness, the low end response and the high end response are automatically boosted in relation to the middle frequencies by an amount necessary to compensate the ear for its deficiencies at the ends of the band. Once the loudness control has been set, the *volume* control may then override it varying the overall level of the intentionally deformed curve of the amplifier as affected by the contour or loudness control.

Amplifiers that do not have such loudness controls may be adjusted manually through the bass and treble controls. When the volume control is turned way down, the bass and treble controls may be turned up to boost the ends of the spectrum to compensate for the hearing loss at these points.

Ear Compensation in Multi-Speaker Systems

There is still another method for compensating the ear for its deficiency and that is through the use of controls on the loudspeaker systems themselves. For instance, in the case of a three-way system, with its usual crossover network and volume controls, aural compensation may be made without touching the amplifier, thus avoiding certain disadvantages inherent in the other method. If the amplifier has been turned way down in level and the bass control turned way up, intermodulation distortion may be severely increased within the amplifier. Intermodulation distortion exists between low frequencies and high frequencies if there are nonlinearities in the circuit. This is true acoustically, mechanically, and electrically. Thus, if we greatly increase the level of the low frequency signal in the amplifier in relation to the middle frequencies, the probability of intermodulation distortion will be increased. This particular situation may be especially aggravated under conditions of extreme bass boost. The amplifier may be forced to work near the limits of its power handling capacity in this lowest part of the frequency spectrum, which will introduce non-linearity into the amplifier.

Obviously, there is a definite advantage to control of the loudspeaker characteristic at the speaker system itself without touching the amplifier. The control usually found in the midrange speaker of the three-way system may be used to turn down the level of the middle range of the reproducible spectrum, so that by comparison the low end response and the high end response are boosted far above the midrange response. Then the *volume* control of the amplifier may be turned *down* to produce the necessary listening level, and the total sound emanating from the system will be properly compensated for the listening deficiencies of the ear at the low levels.

How Much Adjustment Should be Made to the System?

The degree of compensation that will be necessary will depend on the individual's particular ear characteristics and upon average level of the program material. The number of controls on today's high fidelity equipment is indicative of the realization that high fidelity is a personal thing as well as a technical art. The many controls on pre-amplifiers, amplifiers, record compensating devices, and loudspeakers are all means by which the listener may alter the colors in his musical picture. It is a moot question whether the listener should have the privilege of editing (or shall we say "auditing") the music presented to him. According to one school of thought, the listener should not corrupt the conductor's interpretation of the work; he should leave the controls flat. This is the concept of realism. Then there are the knob and gadget fanciers, who prefer to "set the stage" before listening. The "music lovers" and the "gadgeteers" are not necessarily on opposite sides of the fence. Together they constitute the portion of the listening public that appreciates good reproduction.

Perhaps we should change the word "reproduced" to "recreated." To the purist, reproduction means an exact duplication of the original. On the other hand, the word "recreated" implies something basically different. To recreate is to make over. We have all at many times *recreated* the music to which we have listened. For example, when Beethoven wrote the thunder into his Pastorale Symphony, he intended it to sound near at hand. Yet in hearing his work *at a subdued level* we have moved the storm away to some far hilltop. We did not "reproduce" the music; rather we recreated it to confrom to standards different from those Beethoven had in mind.

Moreover, we have seen that the mere act of turning down the volume may entirely change the *tonal* value to the ear, changing completely the effect the composer wanted to produce upon us. Thus, both in *dynamic range and in tonal nuances,* we have "recreated" the music. Whether we like it or not, we become our own impressarios simply by turning down the volume control.

Nature also takes a hand in this. As we grow older our sight grows weak and our ears grow insensitive. The aging process that turns our hair gray causes changes to our ear drums and auditory processes as well. These changes destroy to varying extents the tonal ranges of our hearing and there is nothing (physiologically) that we can do about it. Nature limits not only the sensitivity but also the frequency response of our hearing. Of course, the deterioration of hearing with age is very gradual. Constant changes are produced in our ears, and in our auditory processes. Also, childhood maladies may leave us with ear impairment, and even though they may be small they may cause one to hear things differently.

Even sex enters into the picture insofar as women as a rule are more sensitive to high frequencies than men. Often the woman of the household hears the high pitched scanning frequency from the television set (15,750 cps), while the mere man is totally oblivious to this spurious sound. When the high-fidelity system is playing, her ears, being more sensitive to highs, find piercing and shrill the sounds that to the man of the house are rich and brilliant. Because of these differences in the hearing characteristics of individuals, there can be no best balance, no best range, no optimum loudness. These are matters of personal physiological inheritance; health; and musical, educational, and listening environment. In order to enable the average music enthusiast to recreate the music so that it is satisfying to his ear, he is given the many controls on the pre-amplifier, amplifier, and speaker.

For that matter, long before the music reaches the listener, whether it be over the radio or on disc or tape, many different "editors" have had a hand in altering the music to a far greater extent than the composer ever thought possible. Today, all music reproduction is mechanical. It is mechanical if we hear it through any means other than the original *composer-performer.* We do not hear the Philharmonic at home in the same way the audience in Carnegie Hall hears it. Between us and the conductor is that long acoustic and electronic chain discussed in the first chapter of the book. The number of microphones and their placement, the "hearing" abilities of these microphones, the technical excellence (or lack of it) in the long series of speech input amplifiers, the technical manipulations of the telephone lines that bring the program to the various broadcast stations, the quality of the broadcast transmitter itself, and the characteristic performance of the tuner, amplifier, and loudspeaker all intervene between the orchestra and the listener.

At each of these bridges the human factor is introduced. An individual or group of individuals is constantly at watch at these crucial points to see that everything is "right." But "right" for whom and for what? For *the technological equipment,* of course. Is there a tremendous crescendo coming? Quick, crank down the gain or we'll blow the transmitter off the air. Is there a long sustained pianissimo passage coming? Crank up the gain, or we won't modulate the transmitter properly. It is, of course, true that all these precautions are taken to insure that the listening audience gets optimum service from the technical apparatus involved, but in the accomplishment of this deed, it is the technical apparatus that gets the babying, and the technical apparatus that determines what we are going to hear.

In the days before the invention of mechanical reproducing systems, there was only one person between the performer and the listener, and that was the conductor. Today there are dozens of people and dozens of technical processes, through which the music has to run the gauntlet. *These* are the editors of the music that we hear. These processes and the people who supervise them are not always perfect. In fact, we cannot even be sure that the conductor of the orchestra is perfect. We need only read the various concert reviews and record reviews in the daily newspapers, the music magazines, and the high-fidelity publications to learn that critics often express diametrically opposite views of a particular conductor's efforts. They discuss at length whether he interpreted a phrase right, whether his dynamics were good, whether he lacked lyricism here or forcefulness there. Certainly, the conductor thinks he is doing a good job, yet here are critics whose privilege and duty it is to criticize the work. It all boils down to the fact that in art there is no right or wrong. It is largely a matter of taste. To satisfy this taste, high fidelity brings to the listener all the necessary ingredients to flavor the music to suit his own artistic palate, and perhaps most vital to these individual recipes for musical *recreation* at home are the important acoustic phenomena associated with loudspeakers, their enclosures, and the rooms in which they perform.

APPENDIX

ENCLOSURE CONSTRUCTION DETAILS

(These illustrations are reproduced through the courtesy of the manufacturers and publications designated in the captions.)

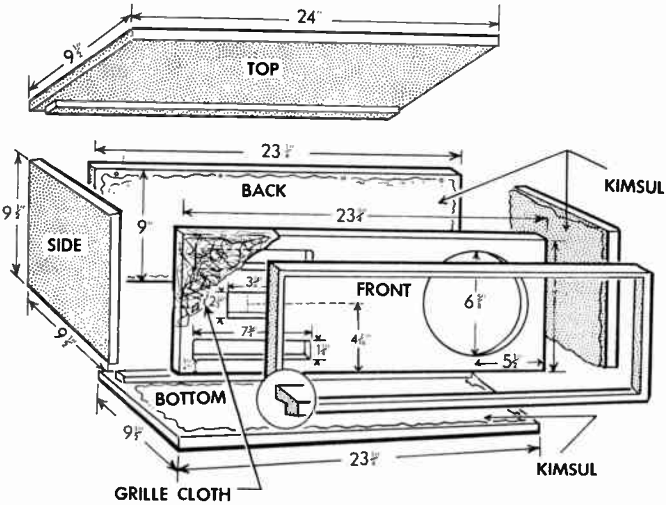


Fig. A-1. Bookshelf bass reflex (University Companionette).

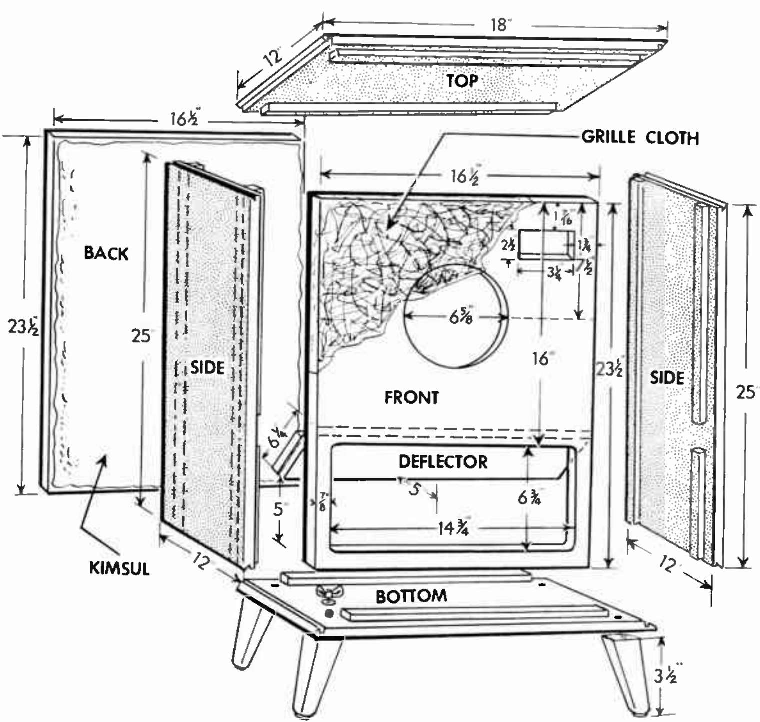


Fig. A-2. Wall type horn loaded port bass reflex (University Triumph).

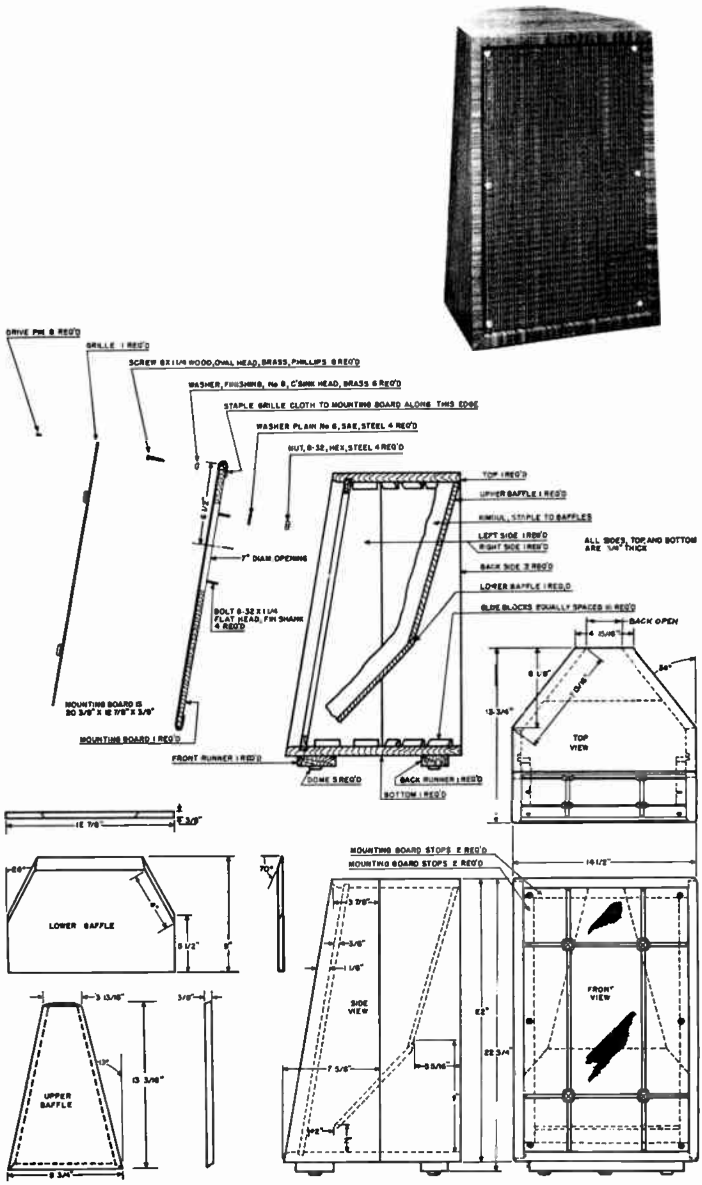


Fig. A-3. Corner folded horn (Electro-Voice Baronette — Klipsch design, courtesy Radio & Television News).

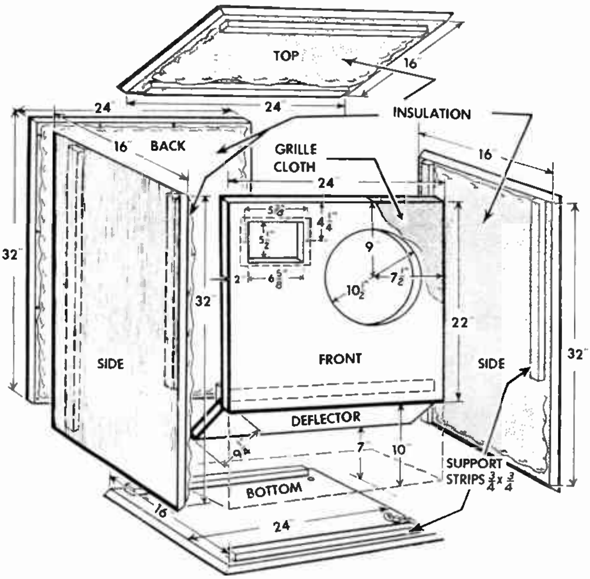


Fig. A-4. Wall type horn loaded port bass reflex (University).

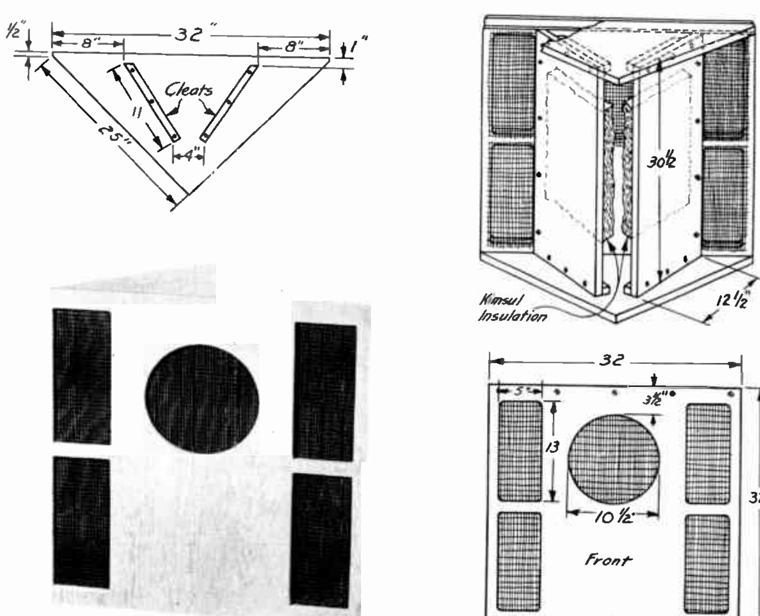


Fig. A-5. Corner folded horn (Cabinart Model 61).

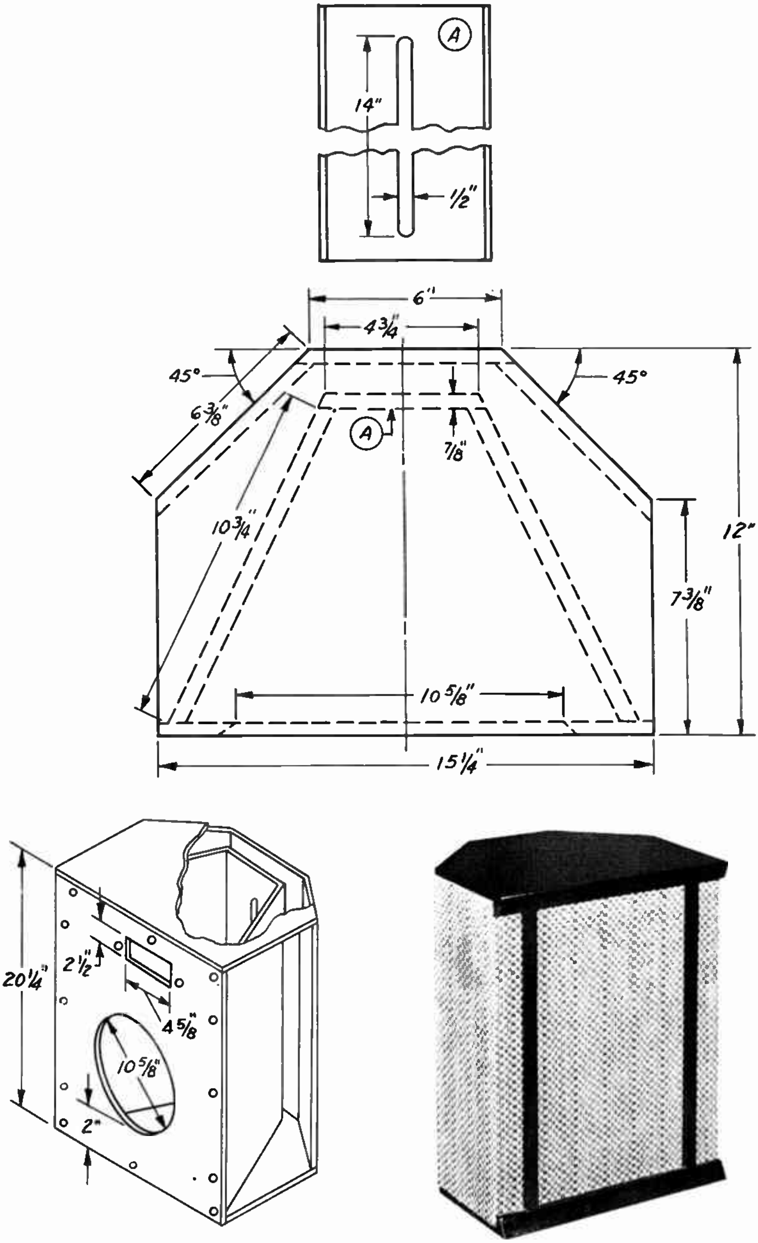


Fig. A-6. Corner folded horn (Cabinart KR-5 — Klipsch design).

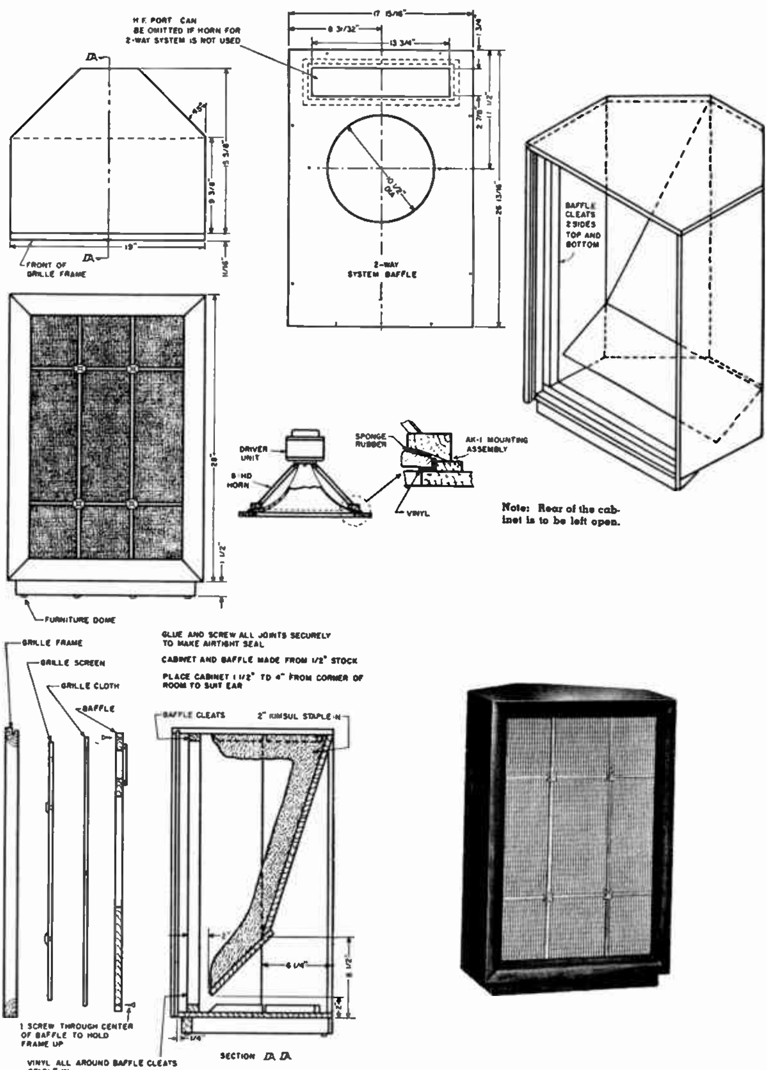


Fig. A-7. Corner folded horn (Electro-Voice Aristocrat — Klipsch design, courtesy Radio & Television News).

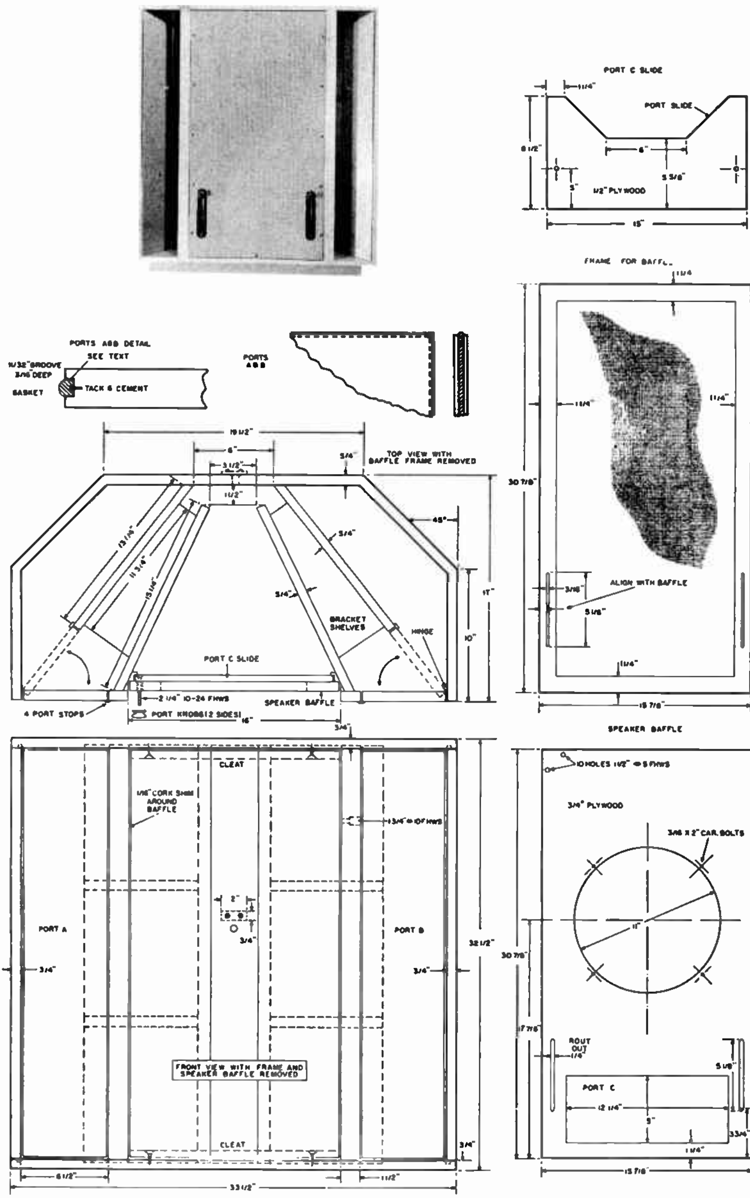
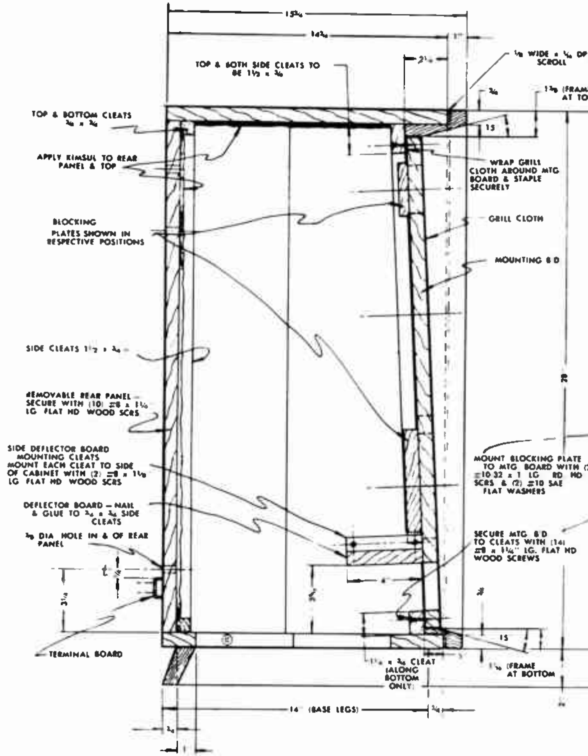
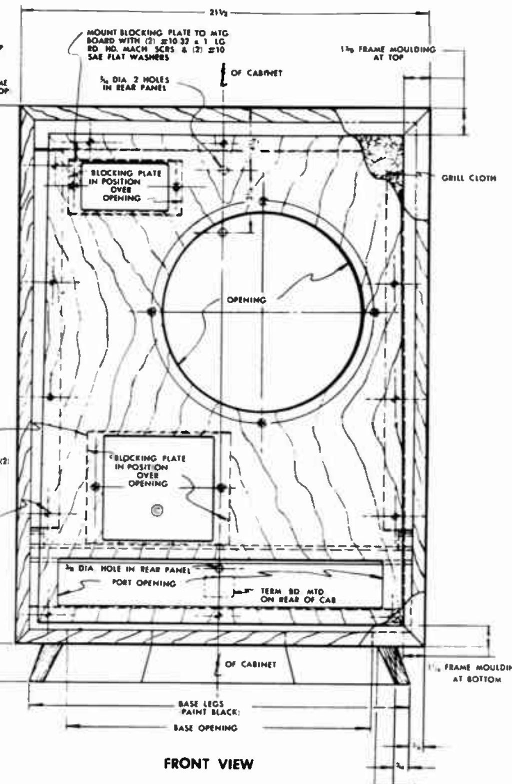


Fig. A-8. Foldaflex bass reflex or folded horn (Oliver Read design, courtesy Radio & Television News).



SECTION



FRONT VIEW

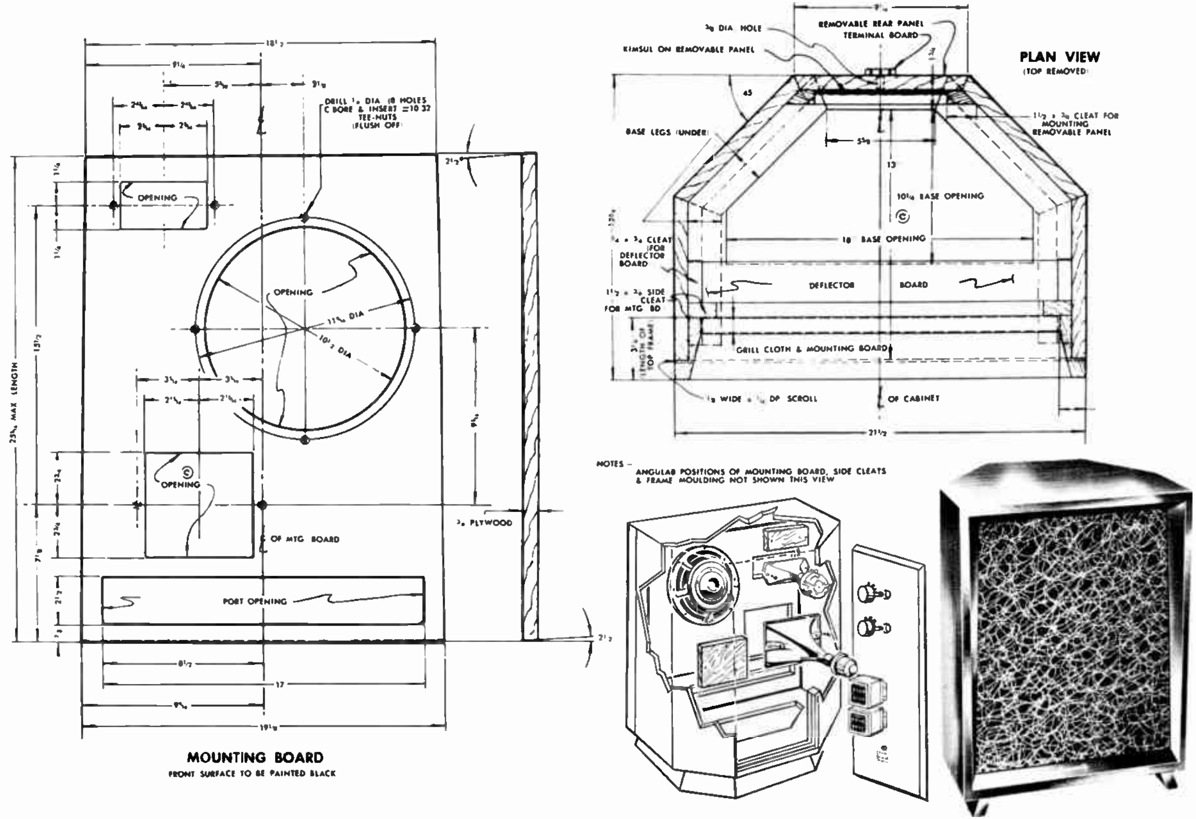


Fig. A-9. Cornerless corner horn loaded port bass reflex (University EN-12).

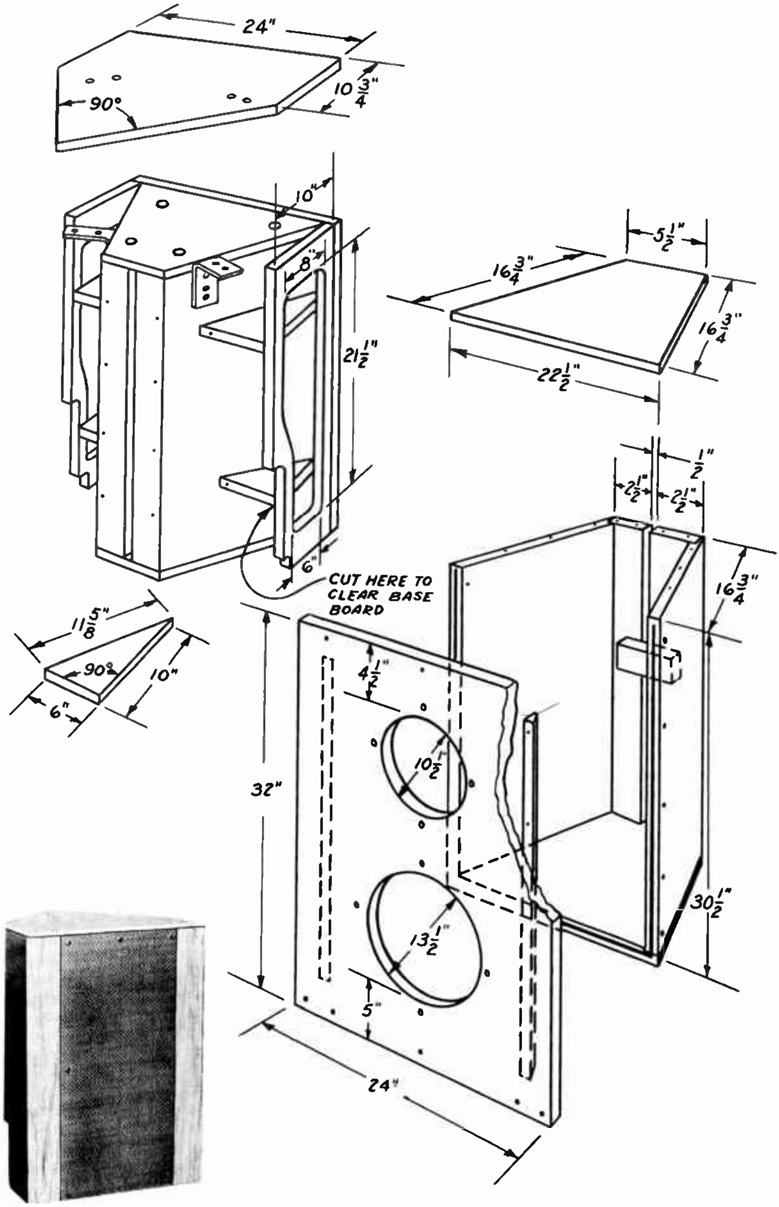


Fig. A-10. Corner folded horn (Cabinart KR-3 — Klipsch design).

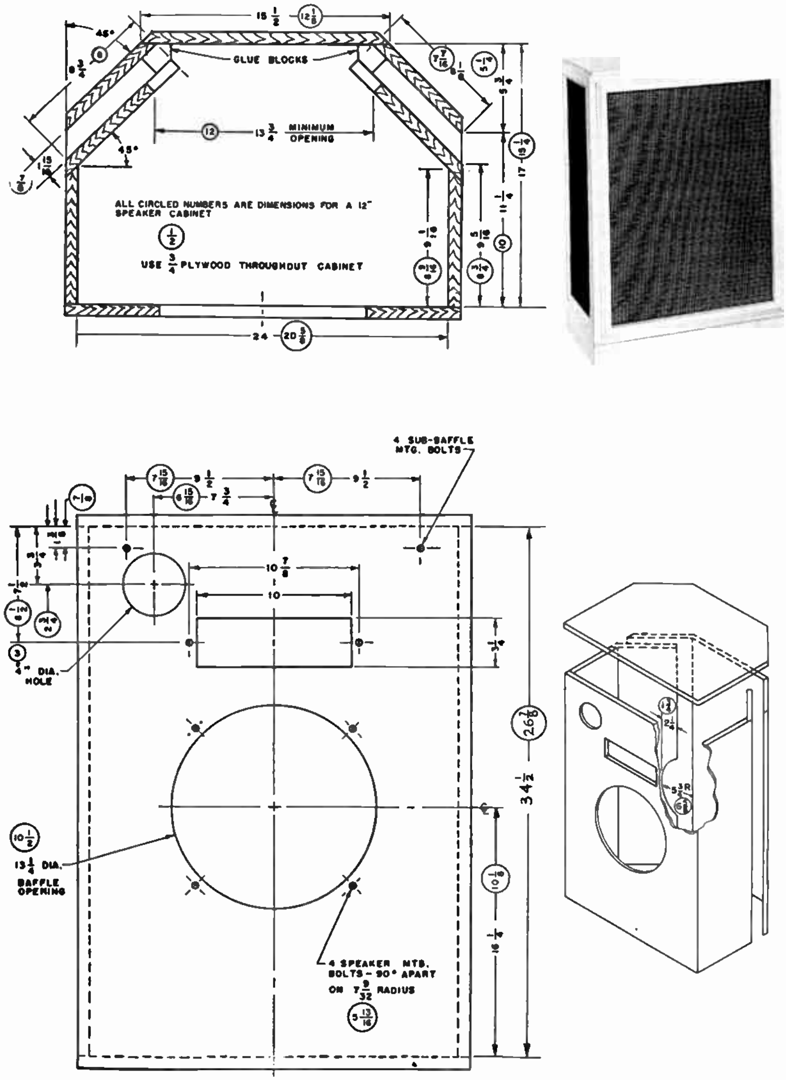
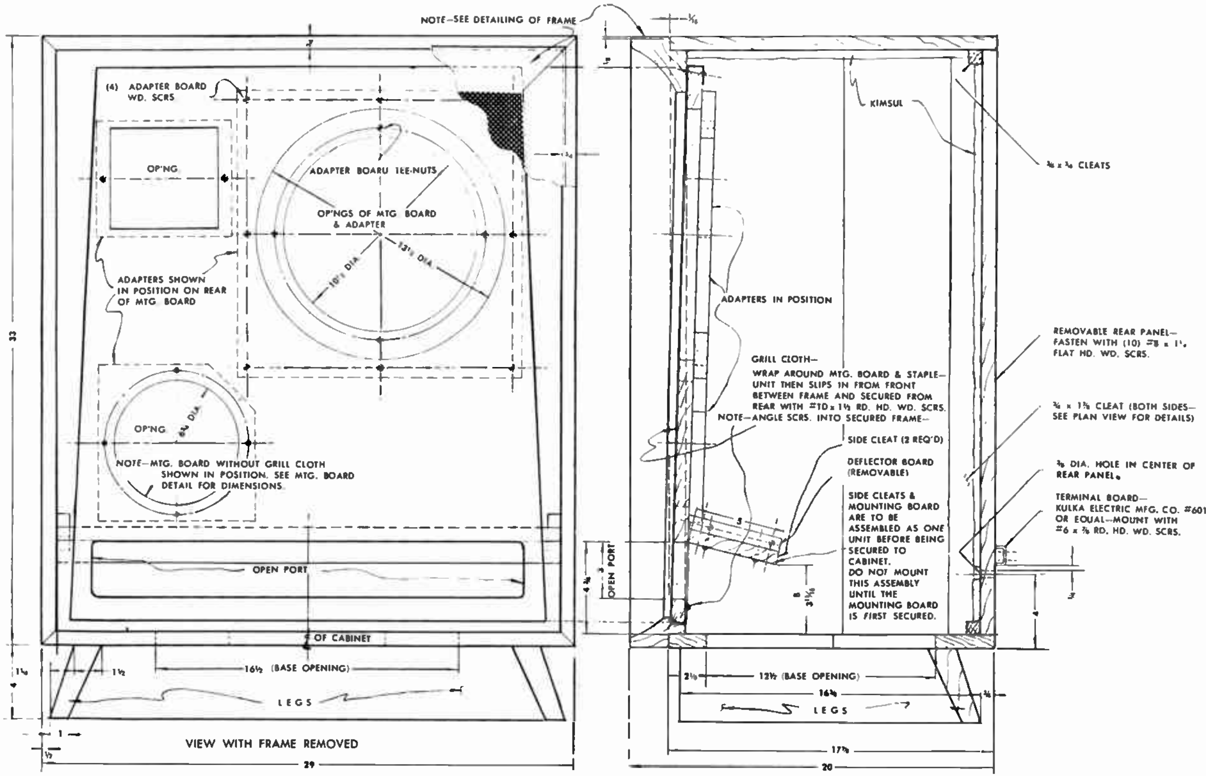


Fig. A-11. Back loaded bass reflex (Jensen Ultraflex).


NOTE—SEE DETAILING OF FRAME
(4) ADAPTER BOARD WD. SCRS
OP'NG
ADAPTER BOARD TEE-NUTS
OP'NGS OF MTG. BOARD & ADAPTER
ADAPTERS SHOWN IN POSITION ON REAR OF MTG. BOARD
OP'NG
NOTE—MTG. BOARD WITHOUT GRILL CLOTH SHOWN IN POSITION. SEE MTG. BOARD DETAIL FOR DIMENSIONS.
OPEN PORT
℄ OF CABINET
16½ (BASE OPENING)
LEGS
VIEW WITH FRAME REMOVED
29
33
KIMSUL
¾ x ¾ CLEATS
ADAPTERS IN POSITION
GRILL CLOTH—
WRAP AROUND MTG. BOARD & STAPLE—UNIT THEN SLIPS IN FROM FRONT BETWEEN FRAME AND SECURED FROM REAR WITH #10 x 1½ RD. HD. WD. SCRS.
NOTE—ANGLE SCRS. INTO SECURED FRAME—
SIDE CLEAT (2 REQ'D)
DEFLECTOR BOARD (REMOVABLE)
SIDE CLEATS & MOUNTING BOARD ARE TO BE ASSEMBLED AS ONE UNIT BEFORE BEING SECURED TO CABINET.
DO NOT MOUNT THIS ASSEMBLY UNTIL THE MOUNTING BOARD IS FIRST SECURED.
REMOVABLE REAR PANEL—FASTEN WITH (10) #8 x 1¼ FLAT HD. WD. SCRS.
¾ x 1⅝ CLEAT (BOTH SIDES—SEE PLAN VIEW FOR DETAILS)
⅜ DIA. HOLE IN CENTER OF REAR PANEL.
TERMINAL BOARD—KULKA ELECTRIC MFG. CO. #601 OR EQUAL—MOUNT WITH #6 x ⅝ RD. HD. WD. SCRS.
OPEN PORT
12½ (BASE OPENING)
LEGS
17⅞
20

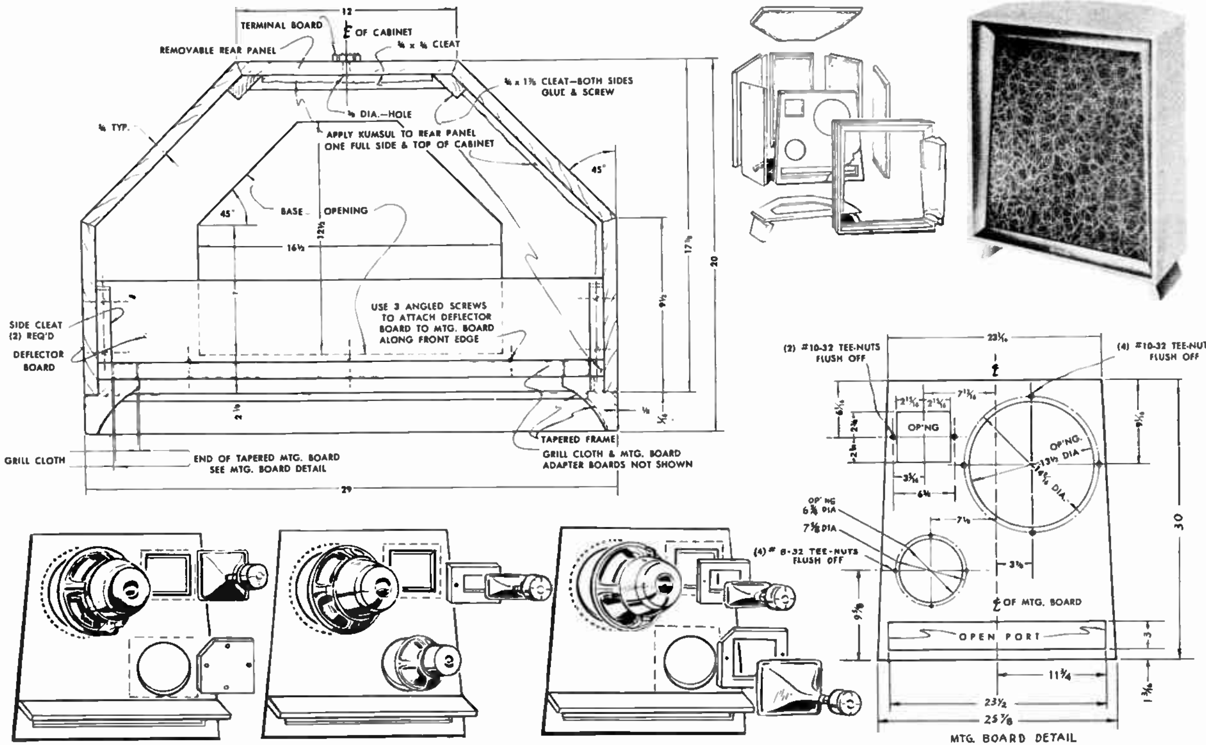


Fig. A-12. Cornerless corner horn loaded port bass reflex (University EN-15).

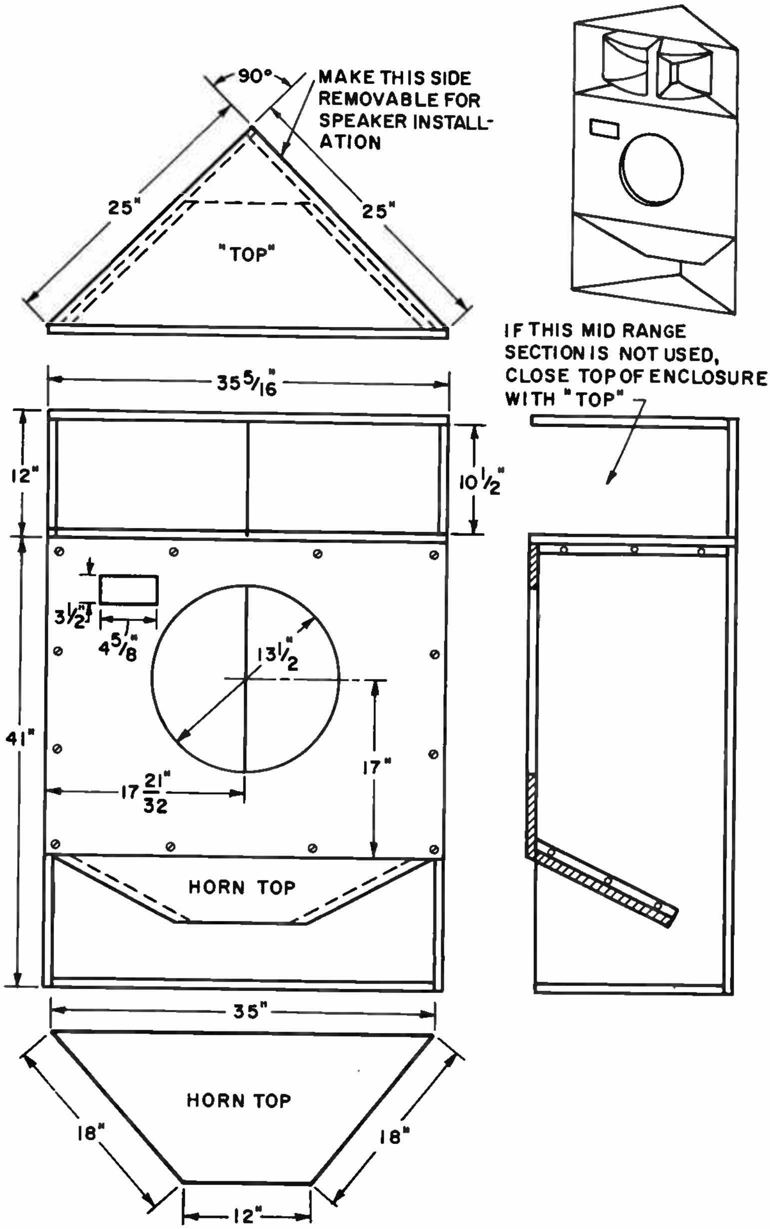


Fig. A-13. Corner horn loaded port bass reflex (University).

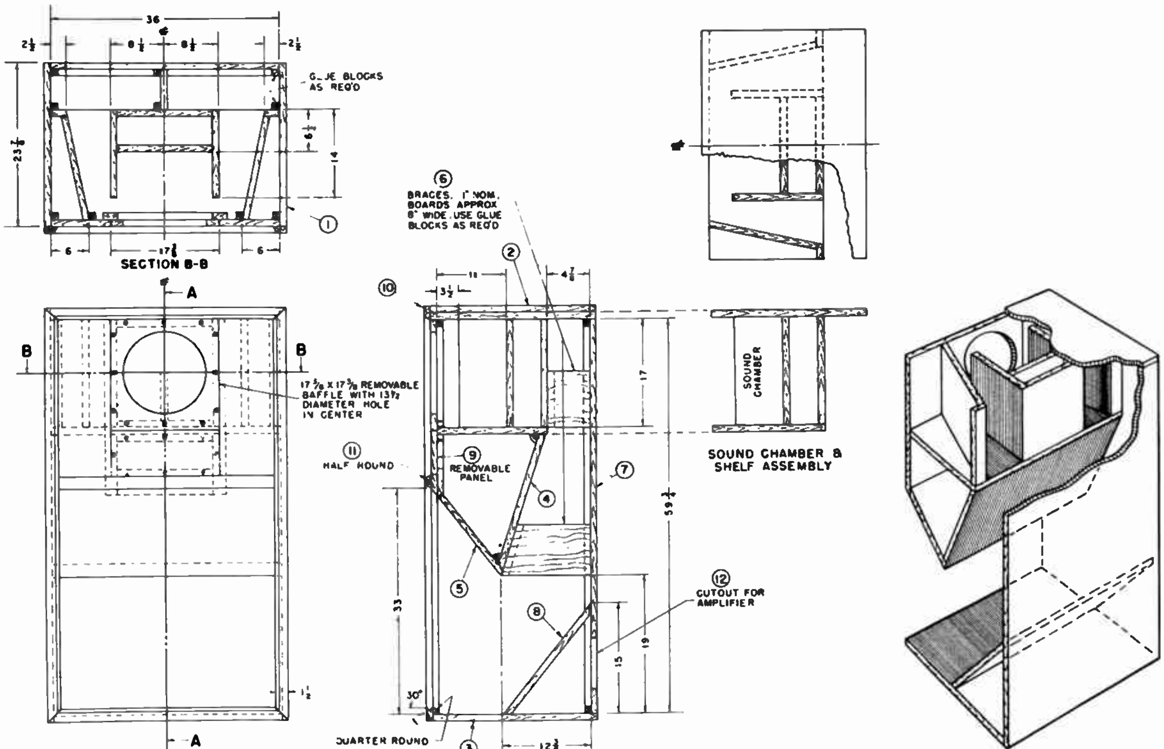
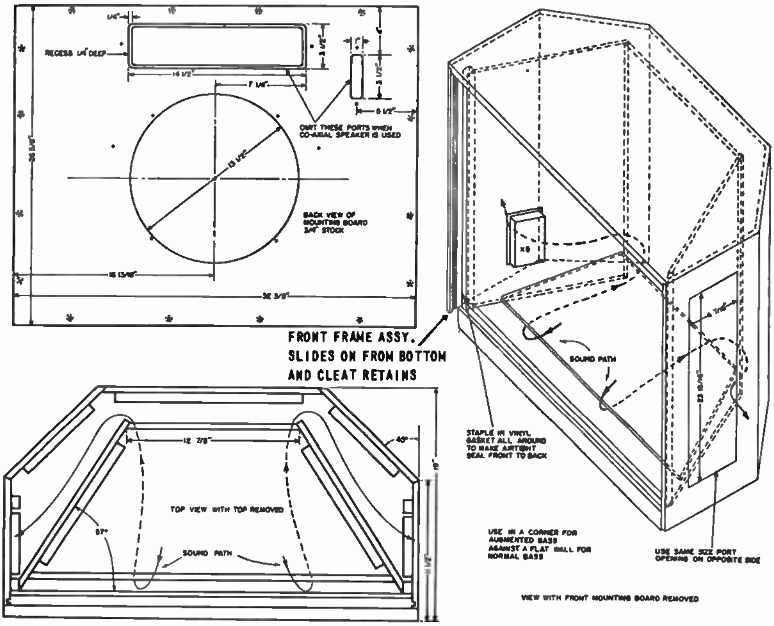
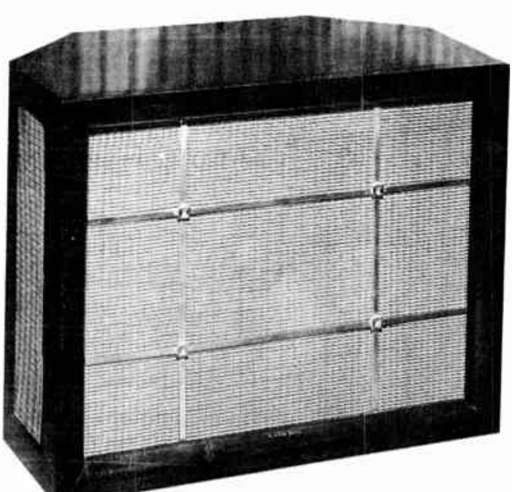


Fig. A-14. Back loaded folded horn (Jensen).


RECESS 1/4" DEEP
14 1/2"
7 1/4"
OMIT THESE PORTS WHEN CO-AXIAL SPEAKER IS USED
13 1/2"
BACK VIEW OF MOUNTING BOARD 3/4" STOCK
32 3/8"
FRONT FRAME ASSY. SLIDES ON FROM BOTTOM AND CLEAT RETAINS
12 7/8"
45°
TOP VIEW WITH TOP REMOVED
SOUND PATH
STAPLE IN VINYL GASKET ALL AROUND TO MAKE AIRTIGHT SEAL FRONT TO BACK
SOUND PATH
USE IN A CORNER FOR AUGMENTED BASS AGAINST A FLAT WALL FOR NORMAL BASS
USE SAME SIZE PORT OPENING ON OPPOSITE SIDE
VIEW WITH FRONT MOUNTING BOARD REMOVED



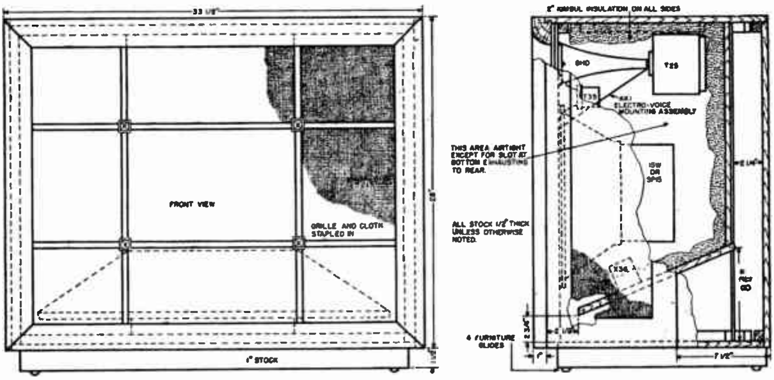


Fig. A-15. Back loaded folded horn (Electro-Voice Regency — Klipsch design, courtesy Radio & Television News).

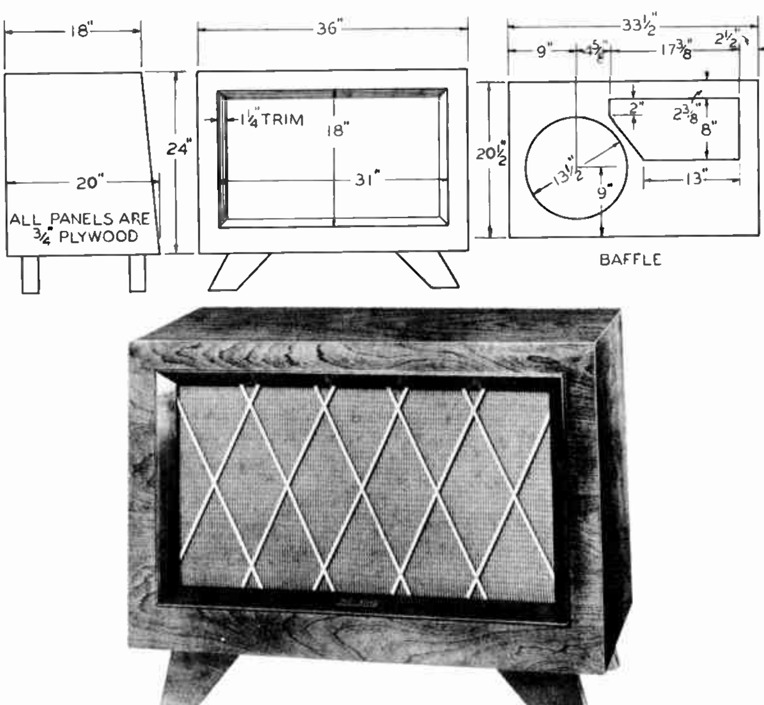
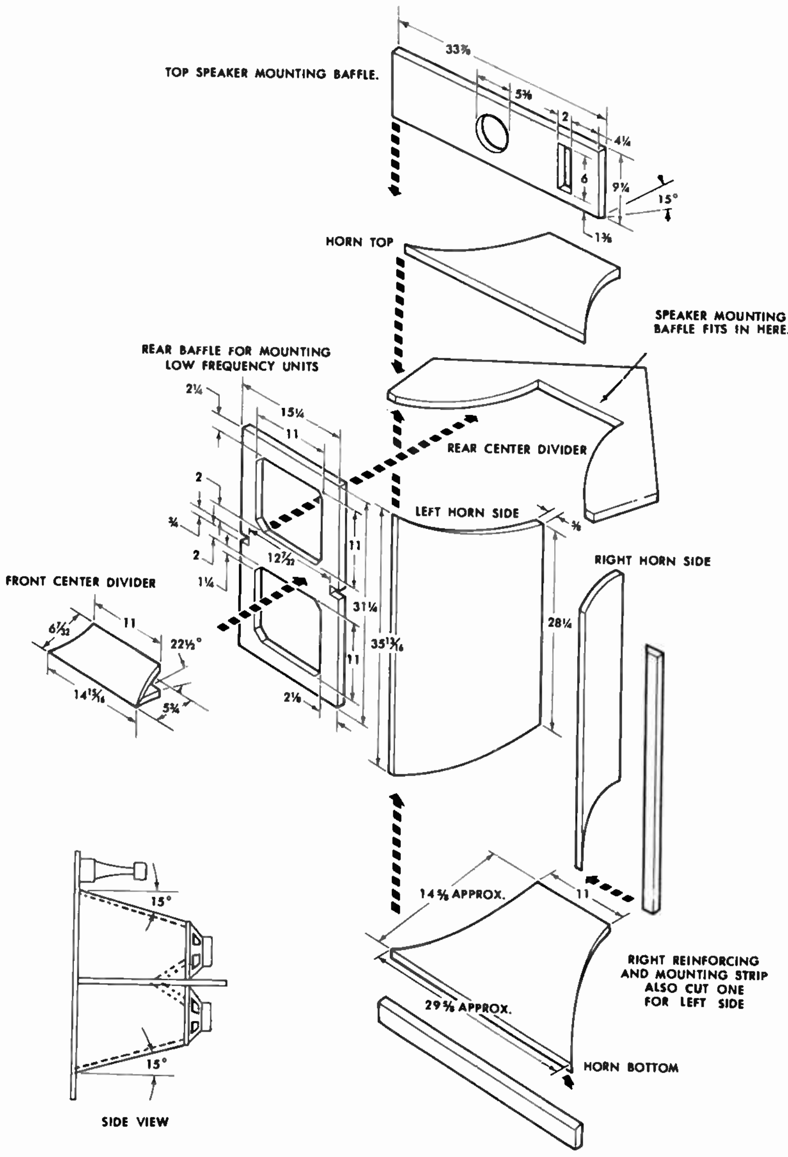
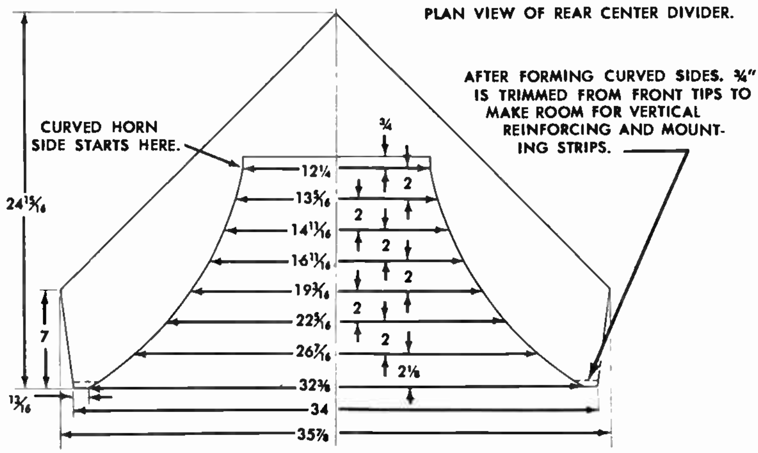
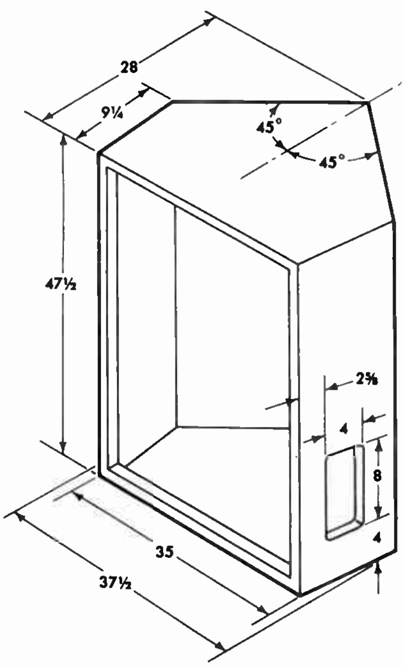


Fig. A-16. Wall type bass reflex (Stephens Low-Boy, Model 610).


TOP SPEAKER MOUNTING BAFFLE.
33⅜
5⅜
2
4¼
6
9¼
15°
1⅜
HORN TOP
SPEAKER MOUNTING BAFFLE FITS IN HERE.
REAR BAFFLE FOR MOUNTING LOW FREQUENCY UNITS
2¼
15¼
11
REAR CENTER DIVIDER
2
¾
LEFT HORN SIDE
⅜
11
2
12⅞
1¼
RIGHT HORN SIDE
FRONT CENTER DIVIDER
31¼
28¼
6⅞
11
22½°
35¹³⁄₁₆
11
14¹⁵⁄₁₆
5¾
2⅛
15°
14⅝ APPROX.
11
RIGHT REINFORCING AND MOUNTING STRIP ALSO CUT ONE FOR LEFT SIDE
29⅝ APPROX.
15°
HORN BOTTOM
SIDE VIEW
BOTTOM MOUNTING STRIP





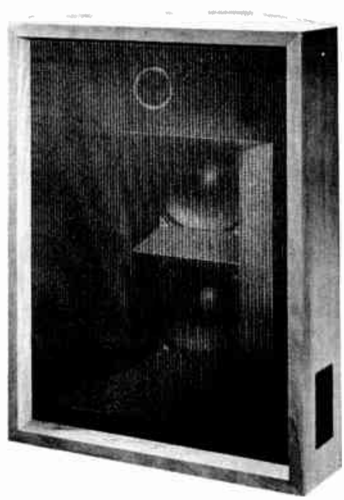
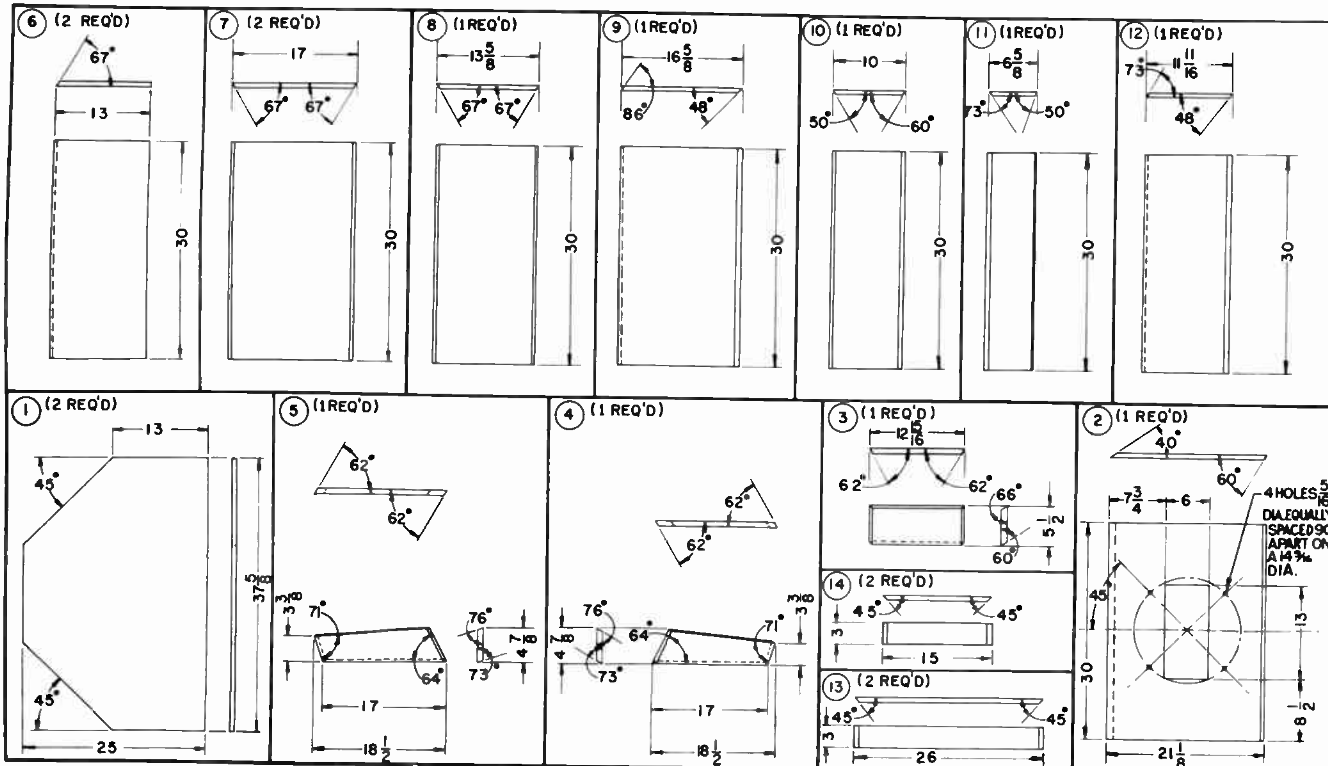


Fig. A-17. Front loaded, rear compression loaded, rear vented dual 15 horn (Jim Lansing Signature, Model D31050).



6 (2 REQ'D)
67°
13
30
7 (2 REQ'D)
17
67° 67°
30
8 (1 REQ'D)
13 5/8
67° 67°
30
9 (1 REQ'D)
16 5/8
86° 48°
30
10 (1 REQ'D)
10
50° 60°
30
11 (1 REQ'D)
6 5/8
73° 50°
30
12 (1 REQ'D)
73° 11 11/16
48°
30
1 (2 REQ'D)
13
45°
37 5/8
45°
25
5 (1 REQ'D)
62°
62°
3 3/8
71°
76°
7/8
4
64°
73°
17
18 1/2
4 (1 REQ'D)
62°
62°
76°
7/8
4
64
73°
71°
3 3/8
17
18 1/2
3 (1 REQ'D)
12 15/16
62° 62° 66°
5 1/2
60°
14 (2 REQ'D)
45° 45°
3
15
13 (2 REQ'D)
45° 45°
3
26
2 (1 REQ'D)
40°
60°
7 3/4
6
4 HOLES 5/16 DIA. EQUALLY SPACED 90° APART ON A 14 3/16 DIA.
45°
13
30
8 1/2
21 1/8

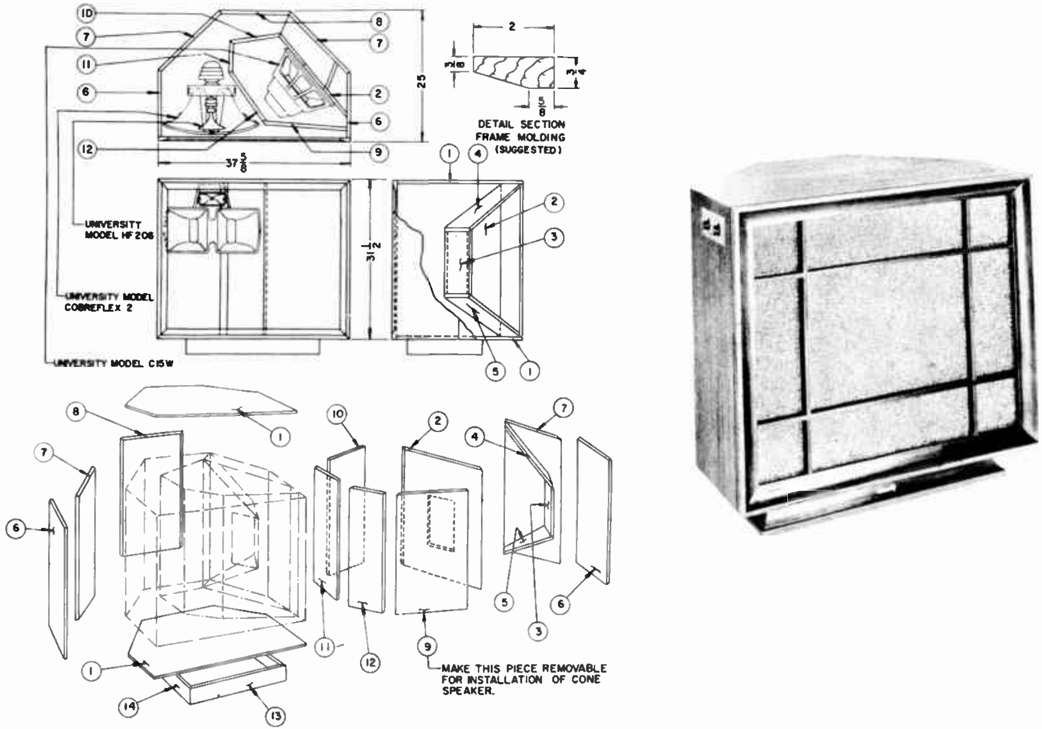


Fig. A-18. Wall type rear folded horn (University Classic Hi-Boy Lo-Boy).

Index

$4.60

ABRAHAM B. COHEN

Audio Engineer

Abraham B. Cohen has been active in music and music reproduction since his teens. During his high school years he was Concertmaster of the Boston School Symphony Orchestra and first violinist in the Boston Civic Symphony Orchestra. Although his extra curricular interests lay in musical activties, his professional goal was electrical engineering and in 1932 he received his BEE degree, cum laude, from Northeastern University and his BS in electrical engineering in 1933.

Upon graduation he became a member of radio station WCAU in Philadelphia where from 1934 through 1936 he served as studio control engineer. From 1936 through 1942, Mr. Cohen was a member of the Engineering Department of Westinghouse Broadcast Stations serving with radio station KYW in Philadelphia. He came to New York in 1942 as Chief Engineer for Metropolitan Television Inc. Following this, Mr. Cohen took the positions of Senior Acoustic Engineer with Best Manufacturing Co., Industrial Electronics Field Engineer with Radio Receptor Co., and then to University Loudspeakers in 1946 as Laboratory Supervisor where in 1953 he became Engineering Manager. In 1958 Mr. Cohen organized his own company, Advanced Acoustics Inc. He then formed a new company, Acoustic Developments Inc.

Mr. Cohen is a charter member of the Audio Engineering Society, a member of the Acoustical Society of America, and a member of the Institute of Radio Engineers. Mr. Cohen has twice served as Chairman of the Loudspeakers Sections of the annual conventions of the Audio Engineering Society and is a member of the Electro-acoustical Committee of the Institute of Radio Engineers.

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