REALISTIC

5/-

HIGH FIDELITY

by H.A. HARTLEY

BERNARDS RADIO MANUALS

no. 159

Realistic High Fidelity

by H. A. Hartley

BERNARDS (Publishers) LTD.

THE GRAMPIANS
WESTERN GATE
LONDON W.6

General Editor:
CLIVE SINCLAIR

First published March 1960

We invite all authors, whether new or well established, to submit manuscripts for publication. The manuscripts may deal with any facet of electronics but should always be practical. Any circuit diagrams that may be included should have been thoroughly checked by the author. If you are considering trying your hand at writing this type of book we suggest that you let us have a short summary of the subject you intend to cover. We will then be able to let you know the size of book required and perhaps give you some advice on presentation.

C.M.S.

Printed by
V. Cooper & Partners Ltd., Flitcroft Street, London, W.C.2
for Bernards (Publishers) Ltd., The Grampians, Western Gate,
London, W.6

CONTENTS

Material in this book was originally prepared for "Radio and Television News" (now "Electronics World"). It proved of such interest that it was reprinted in the "Hi Fi Annual". The Author and Publishers of this British edition express their acknowledgments to "Electronics World" and the Ziff Davis Publishing Company for permission to reprint.

CHAPTER ONE

SOUND AND SOUND WAVES

The object of this book is to show you how to achieve very good musical reproduction in your home without putting you to the task of learning mathematics, electronic theory and acoustics, while saving you from the snags of making unsuitable purchases in your equipment. Many people with a cultivated taste in music have spent large amounts of currency putting together a music system and some have not bothered about the music very much, but have paid a great deal of attention to reproducing very high treble and very low bass. Both types are called "high fidelity fans" and I am going to suggest that except in rare cases the results they get are not music. Perhaps no compact phrase has ever been so overworked as "high fidelity", so before I show you how it may be achieved (and it can be achieved at quite modest cost) it would be a good idea to let us work out a definition of what it really means. In case you wonder what qualifications I might have for such a discussion I can only say that as I was the inventor of the phrase, way back in 1927, I know, at least, what was in my mind when I first used the words.

I do appreciate that there are many people who are quite seriously interested in what can only be described as audio stunts, and a number of record manufacturers have produced special hi-fi demonstration records which show that the technique of recording can produce quite amazing results for people who want amazing results. Basically, however, we are most interested in deriving the utmost pleasure from the works of the musical masters, and that is what I shall try to see that you get. You might think that you have enough power of discrimination to choose what you like yourself, and I do not deny that you may have; but are you sure that your comparison tests are going to be fair to you? Let me give you two examples of where they might not be.

Serious-minded dealers have gone to considerable trouble and expense to install A-B test demonstration rooms, so that you can judge for yourself which speaker you like best and which amplifier gives the best results on the speaker of your choice. Assuming the dealer has no axe to grind, that he has not loaded the dice in favour of the product which gives him the best discount, you are left with the inescapable fact that you are listening in an auditorium which probably has no acoustic properties resembling those of your own private room in which you do your listening. Other manufacturers have given public demonstrations of their equipment, where a live performance has been repeated as a recording and reproduction and you are invited to make the comparison.

Assuming that the demonstration has been so good that you cannot tell the difference, what does it prove beyond the fact that the demonstrator has so arranged matters that that is the impression he wished you to form. Technically speaking, it is comparatively easy to stage such a demonstration (I have done it myself many times) but it does not prove that this is the equipment you want in your home, for once again the acoustics of the demonstration auditorium do not resemble those of your private room. It might even be that equipment of less perfect performance would give better results in your own conditions.

Now you may well ask where do we go from here? And that is what I want to show you. Some of what I say you may have to take on trust until you can prove it independently. My technical facts will be beyond dispute, but when, as is inevitable, I have to wander a little into the intangibles, you will have to judge for yourself. But before you go wandering there is a well-marked technical route which cannot be left without disaster, so the technical side must come first, and I shall try to make the technology as easy to follow as possible. When you have got that far, then comes the final test by which your efforts and my arguments stand or fall. Go to concerts just to get accustomed to what real live music sounds like; then go home and play your records. If you get the same pleasure at home as when you listened to the real thing, then you have achieved what you intended, and I assure you that you can.

It is usual to liken sound waves to the ripples set up in the surface of a pond by dropping a stone into it. Except for the appearance of radiating circles which suggest that the sound waves radiate in a similar manner, there is nothing else in common. If the pond is a rectangular glass tank then disturbance of the surface at the one end will enable you to

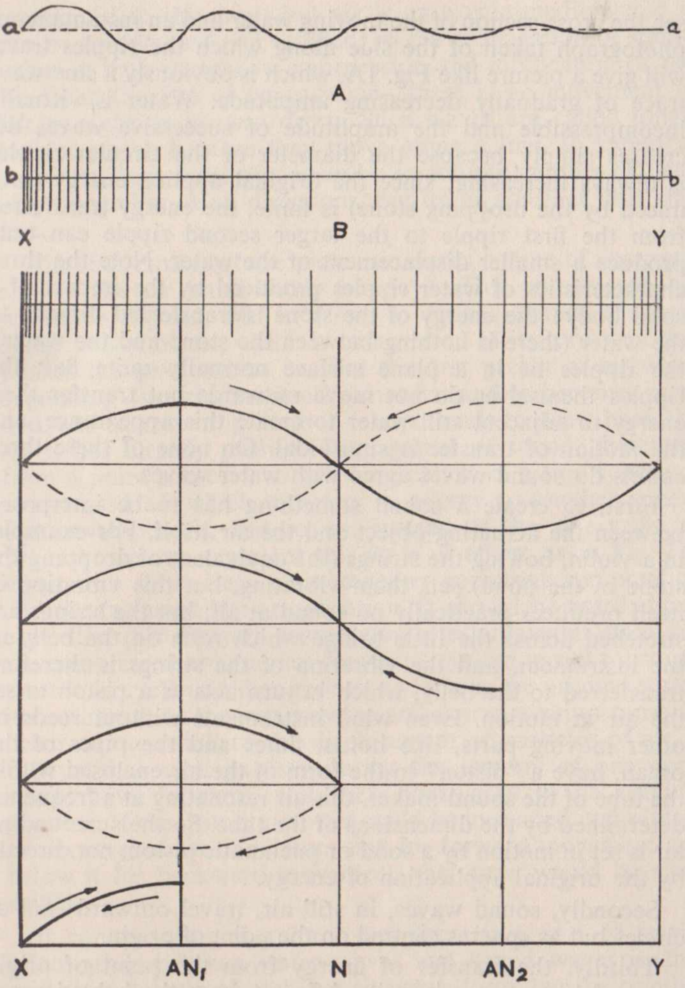


Fig. 1

A—Cross-section of transverse waves of water ripple.

B—Compression of longitudinal sound wave.

C—Effects of a reflector on sound waves at nodes and antinodes.

see the cross-section of the moving water and an instantaneous photograph taken of the side along which the ripples travel will give a picture like Fig. 1A, which is obviously a sine-wave trace of gradually decreasing amplitude. Water is virtually incompressible and the amplitude of successive waves decreases simply because the diameter of the circular ripples is always increasing; since the original applied energy (produced by the dropping stone) is finite, the energy transferred from the first ripple to the larger second ripple can only produce a smaller displacement of the water. Note the three characteristics of water ripples produced by the impact of a solid body: the energy of the stone is transferred directly to the water (there is nothing between the stone and the water); the ripples lie in a plane surface normally quite flat; the ripples themselves do not move outwards but transfer their energy to adjacent still water to create this appearance, and the motion of transfer is sinusoidal. On none of these three counts do sound waves agree with water waves.

First, to create a sound something has to be interposed between the actuating object and the air itself. For example, in a violin, bowing the strings (the equivalent of dropping the stone in the pond) sets them vibrating, but this vibration in itself produces practically no sound at all; but the strings are stretched across the little bridge which rests on the belly of the instrument, and the vibration of the strings is therefore transferred to the belly, which in turn acts as a piston to set the air in motion. Even wind instruments without reeds or other moving parts, like horns, flutes and the pipes of the organ, have a "piston" in the form of the air enclosed within the tube of the sound-maker, this air resonating at a frequency determined by the dimensions of the tube. So the surrounding air is set in motion by a solid or pneumatic piston, not directly by the original application of energy.

Secondly, sound waves, in still air, travel outwards not as circles but as spheres centred on the point of origin.

Thirdly, the transfer of energy from the point of origin outwards into space is quite different from the behaviour of ripples. The diagram of Fig. 1A shows that the ripples move sinusoidally across the datum line represented by the surface of the still water. Therefore, they are transverse waves since they are continually crossing the line of propagation. Since water cannot be compressed the radiating energy must be transferred in this way; but air is an elastic medium and it can be compressed and rarefied, so the propagation of sound waves is by a successive compression and rarefication of the air along the line of propagation—there is no movement to left or right or up and down. Such waves are called longitudinal waves because they move along a line. In spherical radiation there must obviously be an infinite number of lines of propagation in all directions, but let us consider only one line.

The first impact of the piston produces a state of compression in the air immediately beside it. This compressed air wishes to expand, and in doing so pushes against the next small packet of air, which is compressed in its turn, and this pushes the next and so on. But the first packet of air when expanding over-reaches itself somewhat and so becomes rarefied, and in resuming normal volume tends to draw back the air it has already pushed. Propagation of sound, therefore, from a point source involves the creation of a tiny sphere of compressed air which transfers its energy to another sphere just enveloping it, and so on. Instead of the sine wave of Fig. 1A, we can represent this state of affairs as in Fig. 1B, where the short lines close together represent compression and the far apart lines rarefication. It should be noted, however, that this diagram represents an instantaneous state, for the compressed area moves forward from left to right through the whole cycle and then repeats as long as the original sound is continued. As the compression and decompression can only occur as the result of the displacement of particles of air, it follows that each particle during the interval of one cycle must move forwards and then backwards to its original position. If the distance moved could be measured and plotted on a curve, above the datum line for forward movement and below it for backward movement, the curve would be sinusoidal.

It can be seen, therefore, that there is a sort of family relationship between water waves and sound waves in that one characteristic of each is a sinewave form, but the peculiar characteristic of a sound wave is that it is created by little packets of compression travelling along a straight line, and when multiplied by infinity create spheres of compression travelling outwards. Each compressed packet is charged with energy which impinges on your ear drum. If the sound is transient then there is only one impact on the ear; if a steady tone, then the ear is successively hit with packets of air as frequently as the originating "piston" moves the air. If X and Y in Fig. 1B are the points of maximum compression, then the distance XY is called the wavelength of the sound, and the wavelength is a function of the frequency.

The discussion thus far deals only with a simple wave having indirect sinusoidal motion of the type described. The behaviour of the air can be analysed by strict mathematical methods but there seems little point in giving the mathematical proof if you are prepared to accept what I have written as correct. The discussion, and its mathematical treatment, can be developed for complex waves, which consist of a fundamental frequency and one or more harmonics, each harmonic having a frequency which is 2, 3, 4, 5 . . . . times the fundamental frequency. The movement of each particle of air is more complex, but follows the same general principles, as long as it is not confined in a closed space. But the room in which you are listening is an enclosed space, for it has walls, and the walls not only arrest the sound wave but reflect it back along its path.

Now you have seen that the wave assumes the form of an expanding sphere, and if the room in which it was generated was a sphere also then it requires little thought to imagine that the reflection would be constant throughout the room. Rooms being rectangular and not spherical, it follows that different sorts of reflection take place.

Let us return to a single ray of sound, one isolated wave travelling along a line of propagation. Let this ray continue until it meets a wall which is 100% reflective and perpendicular to it. Clearly, the sound will be reflected back along its original path. In Fig. 1B, the particles are moving to create compression and rarefication and move from a condition of maximum forward movement through zero to maximum backward movement (which is the same as the greatest negative forward movement). The maxima and minima of compression are called nodes and in Fig. 1B one node is exactly halfway between X and Y. Those points exactly halfway between the nodes are called antinodes. As the linearly-increased density of the particles moves along the line of propagation there is no change of position and the amplitude is zero, but there is a change of density at the nodes; at the

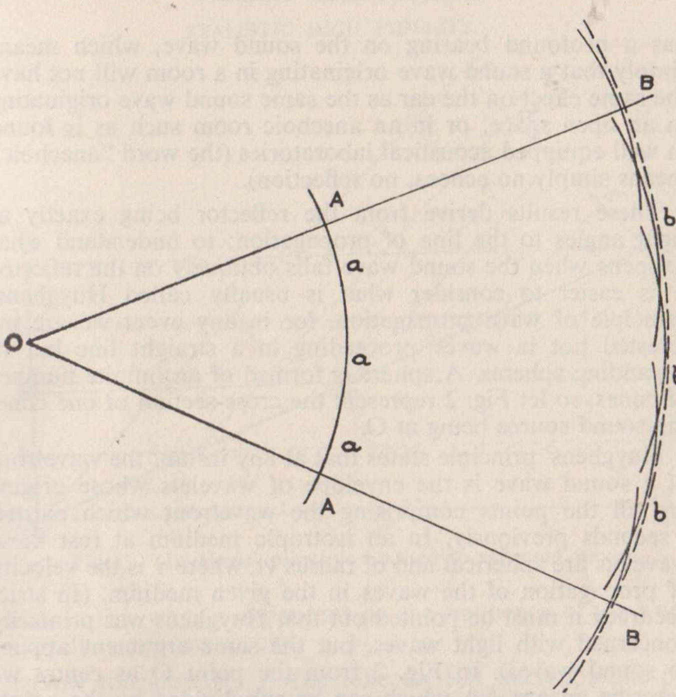


Fig. 2 Huyghens' principle of propagation of sound waves.

antinodes there is maximum amplitude but no change of density.

Now consider what happens when there is reflection. Assume a reflector, such as a hard polished wall, with 100% reflective power. In Fig. 1C the first wavelength XY of Fig. 1B is shown at the top, and below it a pictorial representation of rate of change in density, which is of sine-wave form. The nodes X and Y are lettered as before and the intermediate node at half wavelength is lettered N; the antinodes are AN_1 and AN_2. The outgoing wave is shown as a solid line and when reflected by the wall at Y it is dotted; the arrows show the direction of travel. With the reflector at a node it is seen that the resultant of the two waves is zero, but when the reflector is at an antinode the reflected wave takes the same course (of compression and decompression) as the original wave. It is obvious, therefore, that the position of the reflector has a profound bearing on the sound wave, which means simply that a sound wave originating in a room will not have the same effect on the ear as the same sound wave originating in an open space, or in an anechoic room such as is found in well equipped acoustical laboratories (the word "anechoic" means simply no echoes, no reflection).

These results derive from the reflector being exactly at right angles to the line of propagation; to understand what happens when the sound wave falls obliquely on the reflector it is easier to consider what is usually called Huyghens' principle of wave propagation, for in any event we are interested not in waves proceeding in a straight line but in expanding spheres. A sphere is formed of an infinite number of cones, so let Fig. 2 represent the cross-section of one cone, the sound source being at O.

Huyghens' principle states that at any instant the wavefront of a sound wave is the envelope of wavelets whose origins are all the points comprising the wavefront which existed t seconds previously. In an isotropic medium at rest these wavelets are spherical and of radius vt, where v is the velocity of propagation of the waves in the given medium. (In strict accuracy it must be pointed out that Huyghens was primarily concerned with light waves, but the same argument applies to sound waves). In Fig. 2 from the point O as centre we describe an arc AA which can be sub-divided by the points a, a, a . . . . ; these points can be considered air particles affected by the emergence of the original particle from O. In practice, of course, the distance OA would be extremely small, for we assume that only one particle from O affected several particles a.

From AaaaA we now describe a series of arcs of radius AA to produce the form shown at BbbbB. The envelope, that is the line enclosing this form and shown dotted, is the new wavefront. From this new wavefront a further series of arcs can be described, and so indefinitely. The distance from A to B is vt. This principle of Huyghens was stated as long ago as 1678 and there is no proof that it is correct; yet it is generally accepted because it is a reasonable explanation of what happens, and experiment has not contradicted it. Moreover, it does give an understandable picture of how a sound wave progresses, and since the factor t is involved it can be understood that the scale of the diagram, if one may use the

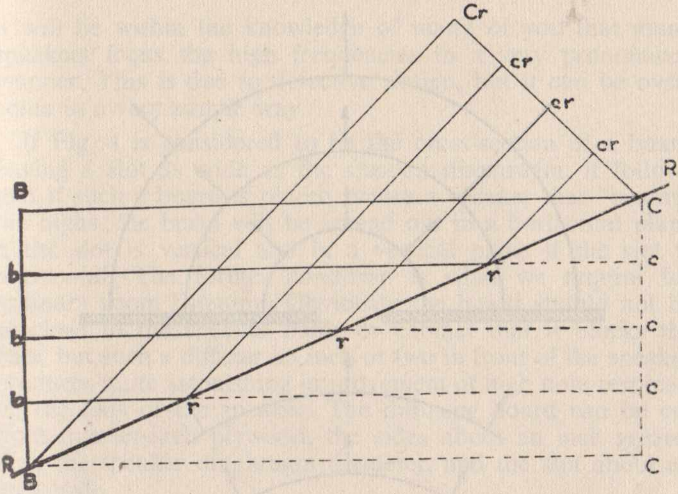


Fig. 3 Huyghens' principle applied to reflected waves.

term in this way, is dependent on the frequency of the sound wave in cycles-per-second.

Now consider Fig. 3. The reflector RR interrupts the passage of the sound wave whose wavefront is BbbbB. If it were not there the track of the sound wave would obviously be within the rectangle BBCC, but that part of the rectangle shown dotted is the part reflected by RR. Using the Huyghens' idea we can consider the approaching wavefront as BB with wavelets starting from the points b, b, b. The point of incidence of the lower B on the reflector indicates that at the instant this wavelet hits the reflector the wavelet from the upper B has still to travel the distance BC, and the intermediate wavelets the distance br. The dotted line BC represents the path of wavelet lower B if it were not reflected, but as it is reflected by a 100% efficient sound mirror it must have the same magnitude, so we describe an arc with centre lower B and radius BC. Similarly, the wavelets emanating from b, b, b are reflected at r, r, r and are reflected onto the wavefront CrC at positions cr, the distances rcr being equal to the distance rc. So, then, at a given instant, part of the wavefront is wholly reflected, part is not reflected at all, and the intermediate wavelets are partially reflected. In the whole

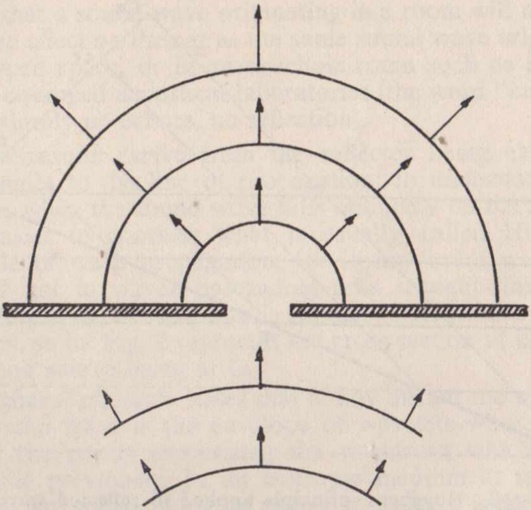


Fig. 4 Dispersion of sound wave by a slot or hole.

process it will be noticed that the wavefront is reversed with respect to the plane of the reflector.

By a similar argument it can be shown that where the reflector is only a poor reflector, so that it is transparent to sound waves, refraction of sound waves takes place in a manner similar to that of the refraction of light waves; this is of importance when considering the effect of hanging "diffusing" materials over the sound source, or, for that matter, the use of fabrics over the front opening of a speaker cabinet.

One further characteristic of the behaviour of sound waves should be noted before we apply these generalisations to the consideration of room acoustics. In Fig. 4 is shown the approach of a sound wave to a hole in a sound-insulating partition. Most of the wave is blocked, but that part passing through the hole takes on the characteristic spherical form. In other words, the sound passing through the hole is diffused throughout the space on the forward side of the partition. This may not seem to be a very exciting thing to illustrate but it happens to be of considerable value in improving listening conditions with unsatisfactory speakers. We have not yet reached the stage when we can criticise speaker design but it will be within the knowledge of many of you that many speakers focus the high frequencies in a very pronounced manner. This is due to defective design, but it can be overcome in a very simple way.

If Fig. 4 is considered to be the cross-section of a board having a slot as wide as the speaker diaphragm, it follows that if such a board is placed before a speaker that "beams" the highs, the beam will be spread out in a horizontal plane if the slot is vertical and in a vertical plane if the slot is horizontal. The former condition is what we require for ordinary room listening. Obviously the board should not be so close to the speaker baffle or cabinet that it blocks the bass, but such a diffuser an inch or two in front of the speaker produces quite astonishing improvement of high note response off the axis of the speaker. The diffusing board can be cut from quarter-inch plywood, the sides about an inch greater than the speaker diaphragm diameter, and the slot about an inch wide.

CHAPTER TWO

THE EFFECT OF ROOM ACOUSTICS

Our rooms reflect sound in varying degrees and the amount of reflection is determined not only by the furnishings but by the frequency of the sound waves being reflected. The size of the room as well as its shape, has a bearing on what is actually heard; the position of the speaker can alter everything; the very nature of the music being reproduced has some bearing on the way it is heard in the auditorium. Given unlimited wealth and resources the way to solve the problems is to hire an architect who is an expert in acoustics and get him to build a music-listening auditorium somewhere on the grounds of your estate, with enough seats in the thing to accommodate the many people who will come to hear the nearly perfect. But most of us are not like that. We are ordinary people and have to use what we have, for better or worse. Let us try to work out how to do it for the better.

First of all, the size of the room. No doubt you have read over and over again that you will get loss of bass if the room is not big enough, because to reproduce a low-frequency sound the room must be at least as long as the wavelength you wish to reproduce. Fig. 5 is a chart showing the wave lengths of various frequencies of sound waves, and from this you will see that, according to the text books, a room which will reproduce a 50-cycle note must be at least 24 feet long and one to reproduce 30 cycles would have to be at least 38 feet long. Sometimes you are told that if the speaker is in one corner it will sound better, for the diagonal of the room is obviously longer than one side, so all you have to do is to sit in the opposite corner and there you have it! But if you do not want to put the speaker in one corner and do not wish to sit in the opposite corner, what are you to do? My suggestion is that you put the speaker where you want to put it and sit where you want to sit. And you will still hear the bass, in spite of the textbooks!

I do not want to decry the efforts of my fellow writers, but it is a fact that a lot of textbooks are just a rehash of material that has appeared in print before, and if somebody many years ago came out with a "law" or a "principle" or an "axiom", it is likely enough that it will be repeated over and over again, without its alleged validity being questioned. Being a difficult and unbelieving person myself, I very often do not accept these laid-down principles, and as I can hear exceedingly well reproduced low notes in my own room which is nowhere near as large as the minimum size laid down by the experts, it follows that there must be some other explanation of what is going on.

As has been explained, a sound wave progresses in ever-growing spherical zones of compression followed by zones of rarefication; a human ear in the path of the sound wave will be acted on by the compressed and rarefied air. If there is only one sound wave the ear drum will be affected only once, but if the sound is continuous then the eardrum will be affected every time a zone of compression and decompression passes it, at a frequency determined by the frequency of the original sound. If it is a 50-cycle note the ear will be affected 50 times a second, and the fact that the wavelength, the distance between the successive spheres of compression, happens to be 24 feet has nothing to do with your hearing the sound in any way at all. You could hear the 50-cycle sound in the open air or in a pair of headphones (if these are capable of reproducing a 50-cps note) or in any room between these extremes. But,

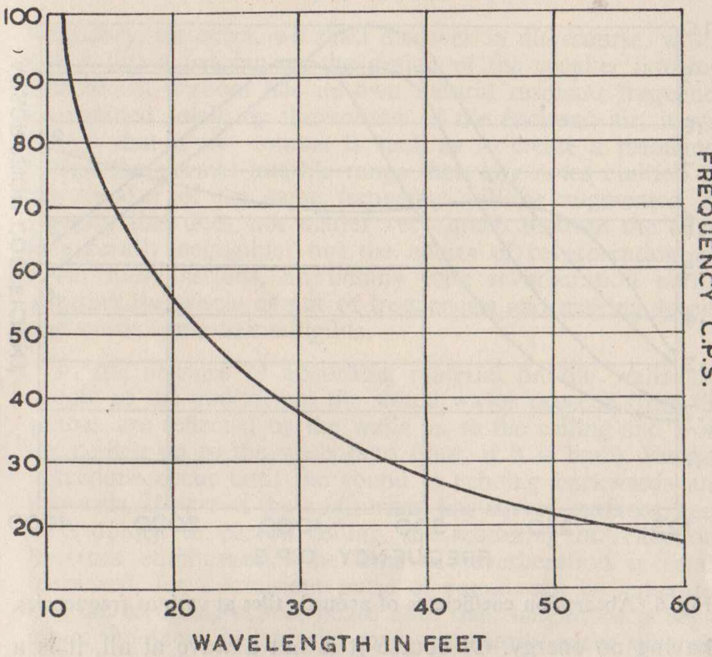


Fig. 5 Wavelengths of sound waves at frequencies up to 100 c.p.s.

and it is a big but, reflections from the walls of the room have a great deal to do with what happens.

Without considering any factors other than reflection let a 50-cycle sound be sent out from a speaker in a room which is 30 feet long. According to the textbooks this room is large enough for you to hear the sound properly because it is big enough to contain a whole wavelength. But if you walk about the room while the sound is emerging from the speaker you will find that there are points where you hear no sound at all. This is due to the reflections from the walls. If you refer back to Fig. 1 you will see that reflectors on the nodes produce cancellation of the sound and those on the antinodes do not. If a whole wavelength and a bit have merged from the speaker and the bit is reflected from a wall in such a way that a zone of rarefication meets an equal and opposite zone of compression the result will be nothing at all. Such a condition is called a standing wave, because it is a "wave"

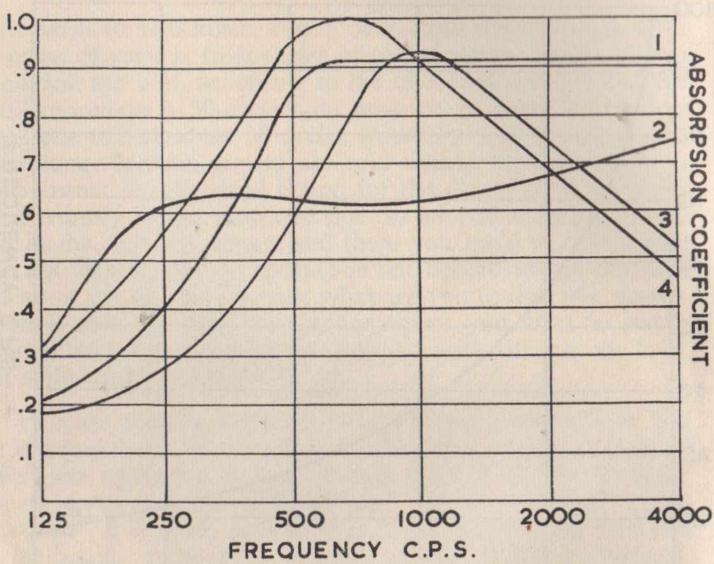


Fig. 6 Absorption coefficients of acoustic tiles at various frequencies.

having no energy. Of course it is not a wave at all, it is a zone of no wave, but the term conveniently describes the condition. Standing waves exist in terms of frequency, room dimensions, and the nature of the reflecting surfaces, and it is an instructive experiment to feed an amplifier with the output of an audio oscillator and listen to the speaker in various parts of the listening room. There are acoustically blind spots all over the room, and just outside of these acoustical blind spots the sound can be heard at full strength.

With great patience a map of the room could be drawn for each frequency showing the location of the blind spots, but as the sounds used for the map are pure and sustained notes, which very rarely occur in real music, the value of a set of such maps seems to be very doubtful.

Independent of the frequency of the sound wave there are two factors which determine the behaviour of the sound wave once it is injected into the room—resonance and reverberation. Sometimes these terms are used interchangeably but they are two quite distinct effects. As in an open or closed organ pipe any enclosed body of air resonates at its natural frequency, an effect, we shall discover in due course, which has a direct bearing on the design of the speaker housing. The air in a room has its own natural resonant frequency determined solely by the volume of the enclosed air; it will follow that if the volume is such as to create a resonance within the normal audible range then any notes emitted by the speaker of the same frequency will be augmented. In practice this does not matter very much because the effect is generally negligible; but the effects of reverberation are much more serious, an unduly long reverberation period affecting the whole gamut of frequencies and making speech and music quite unintelligible.

In the absence of absorbing material on the walls and ceiling of the auditorium the sound waves proceed from the source, are reflected by the walls on to the ceiling and from the ceiling on to the walls (and floor, if it is bare). Further reflections occur until the sound is echoing backwards and forwards. If part of the auditorium has curved surfaces, such as a domed or curved ceiling, the scattering of reflections becomes emphasised. The time of reverberation is easily measured, for a transient pulse of sound can be generated and the recurring echoes heard until their magnitude is negligible; the decay in intensity must necessarily occur since the walls and ceiling are not perfect reflectors, and a little is lost with each reflection. The time in seconds required to reach practical audibility is called the reverberation period.

Before it was treated, a lecture room at Harvard University which was used in some of the earlier experiments in architectural acoustics had a 5.5 second period for an ordinary human voice; obviously even a slow speaker could utter several syllables in this time, so the result was simply a jumble of sounds if the room was fairly empty. Adding cushions to the seats improved matters, the more cushions the greater the reduction in reverberation time, and the further addition of a packed audience brought the time down to a period when a speaker could be heard very well indeed. These somewhat primitive and nowadays obvious results did at least start proper investigations into room acoustics, and it can be taken as a simple generalisation which always works that if you stand in the middle of your listening room, clap your hands, and hear that "the melody lingers on", then conditions are not right for high fidelity reproduction and sound absorbing materials must be introduced.

These properties of various materials can be arranged in a simple table. If the co-efficient of absorption is unity, representing complete absence of reflection, then the co-efficients for various substances are as given in Table 1.

As far as the usual room accessories are concerned, unglazed book cases fairly full of books are good sound absorbers, but if glazed the co-efficient becomes that of glass; unupholstered furniture can be taken as equal to pine boards. Wallpaper is more effective than paint on plaster but there is not much improvement by using it. Thickly upholstered furniture is much more absorbent than modern functional designs; a fitted carpet provides a more manageable listening room than one with a polished wooden floor and rugs. Picture windows without curtains are almost impossible to correct or compensate; if you have one of these quite admirable features in your music room, your listening will have to be done after dark, and the curtains must be substantial. Uncased radiators and wall heating panels can be very troublesome, as can be a piano, either upright or grand.

Controlled absorption can be set up by the use of acoustical tiles. These are usually recognised by their perforated appearance. Usually the front portion is of compressed asbestos pierced with a regular pattern of small holes; this is backed with a layer of rock-wool from a half-inch to one inch thick. The tiles are not fastened to the wall but to battens fastened to the wall; alternatively the rock wool can be obtained as separate cushions to be laid between the battens, the front tiles being fastened to the battens. Other tiles, cheaper and much lighter in weight, consist of compressed sugar cane fibre; others, again, are made up of exploded mica granules cemented to shape. Typical absorption curves for tiles of these various types are shown in Fig. 6, and it will be seen that maximum absorption usually occurs in the frequency band 700-1500 cps. Manufacturers of these tiles will supply the absorption curves on request.

The acoustical treatment of rooms intended for music listening has to be considered from two aspects, insulation from external sounds which would interfere with enjoyment of the music and removal of reflections in the room, particularly in the reduction of reverberation. Acoustical tiles will help in both ways, although sound insulation between rooms and from external noise should have been incorporated when the building was erected. Subsequent treatment for reducing reverberation can be calculated for tiles by using this formula:

$$t = \frac{.049\ V}{(-2.3\ \log_{10}(1-a))S}$$

where t is the reverberation time in seconds, V is the volume of the room in cubic feet, a is the absorption co-efficient of the tiles, S is the surface area of the walls and ceiling (and the floor if not completely carpeted). V being fixed, since the room exists, the formula gives the area of tiles required to reduce the reverberation time to any desired figure.

As the absorption varies with frequency and as the reverberation period varies with room volume, the determination of absorbing area must be a matter for your personal taste. As I mentioned before, a completely dead room lacks the life needed for pleasant musical reproduction, so if you have overdamped, then the sound from the speaker must be diffused by suitable reflectors. This is regularly done in recording and broadcasting studios, for with the varying types of sound to be recorded or broadcast, conditions must be varied to suit the requirements of the control engineers. Suitable reflectors can be either flat or convex but not concave, since these focus sound, just like the concave reflector of a car headlamp focuses the light from the bulb. In studios the flat or convex reflectors are frequently mounted on pivots so that reflection can be controlled as desired. This elaboration is not needed for home listening; all you should consider is the avoidance of concave surfaces. If your listening room is L-shaped much better diffusion will be secured if the outer corner of the "L" is faced off with a diagonal panel set across it. If your room is long and narrow, a similar panel set across the angle between the ceiling and the wall farthest from the speaker will help to ensure a better mean sound distribution. All corners filled with convex mouldings will help; the normal concave plaster coves on ceilings detract.

Naturally the position and direction of the speaker will determine to a great extent how the sound is ultimately distributed. Most speakers focus the highs, and speakers which focus severely may sound best by having the front covered with a slotted board, as mentioned previously. This particularly applies to speakers which have an unduly large output at 2,000 to 3,000 cps (a common fault), for frequencies of this order do not seem to fit the absorption curves of tiles.

CHAPTER THREE

REVERBERATION AND SPEAKER CURVES

You now know how sound waves behave, you have been introduced to room resonances, reverberation, standing waves and absorption co-efficients. You realise that something should be done about the room in which you will listen to your radio, your records and your tape. You may even, as I, have gone to the libraries and read book after book to find the answer, and come away, as I did, knowing no more about it than when you went in. The treatment of large auditoriums has been studied intensively, but what is the good of consulting tables and examining curves if they start off with a smallest room of 10,000 cubic feet? You and I have to make do with something very much nearer 1,000 cubic feet, and then it is cluttered up with all sorts of domestic bric-a-brac. What is worse, if we design a perfect auditorium then we are faced with the fact that our speakers are not perfect, and some of them are a very long way from being even near perfect. It seems sensible to arrange matters in the room to compensate some of the shortcomings of the speaker that is going to be used in it.

Well, we have to make a start somewhere, so let us start with reverberation. If the reverberation period is too long then good reproduction is impossible, so have this reduced to not more than $1\frac{1}{2}$ seconds; I prefer it to be not more than 1 second, otherwise the "attack" of the reproduction is spoiled, to my ears.

If the floor is covered with a fitted carpet, so much the better. If not, and bare wood or linoleum forms an appreciable part of the floor, this added to bare walls and ceiling will result in too long a reverberation period. The clapped hands test can be used, or a very good instantaneous sound source is the old school boy trick of inflating a paper bag and bursting it between the hands. Have all the normal furnishings in

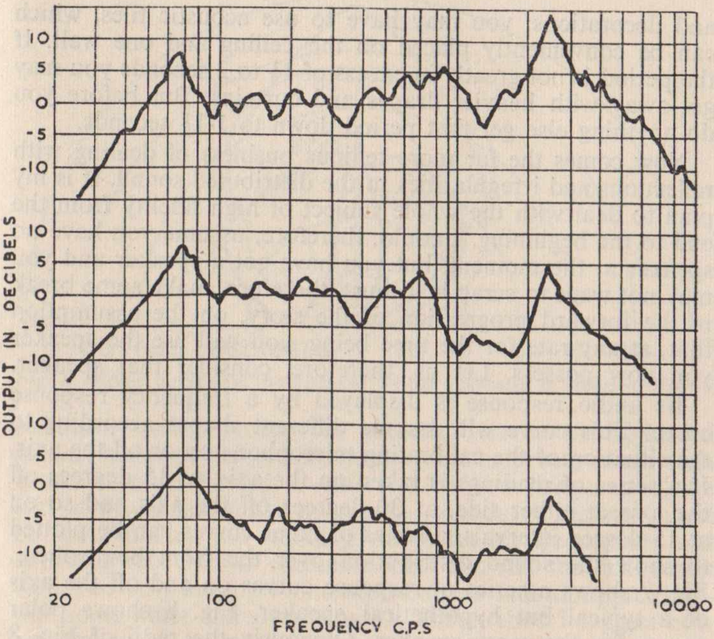


Fig. 7 Response curve of a speaker (from top to bottom) on the axis and at 30 and 60 degrees off the axis.

the room; have the average number of people in the room who will be listening with you; let them sit in the chairs as they would normally do; have a helper with a stopwatch and a keen pair of ears.

Explode the paper bag, your helper starting the watch at that instant. When the echoing sound has died away to negligible proportions the watch is stopped. Repeat this measurement with other helpers and other ears, in fact for as long as your supply of paper bags holds out, then take the average of all the figures, thus averaging out errors too. If the time for the sound to die away is greater than $1\frac{1}{2}$ seconds (I still advise 1 second) more absorbing material must be used in the room, this being quite independent of the type of speaker, type of housing or location of the speaker in its housing; it is a fundamental property of the room itself. If your room is sparsely furnished, or very modern, with reflective furniture and decorations, you may have to use acoustic tiles, which can be conveniently placed on the ceiling and one wall. If the period is not greatly in excess of 1½ to 2 seconds you may get away with heavier drapes and curtains. But before you do anything else get that period down to 1-1½ seconds.

Now comes the far more tedious business of dealing with reflections and irregularities in the distributed sound. It is my plan to deal with the whole subject of high fidelity from the end to the beginning. I could, therefore, assume you have no speaker at the moment, but you have got a speaker and you may not want to scrap it. I must, therefore, make some break in the forward progression of the story, on the assumption that, at any rate for the time being, you will use the speaker you now possess. Let us, therefore, consider that speaker.

Its audio response is displayed by a frequency response curve. This curve will assume different shapes according to the situation of the calibrating microphone on or off the axis. If a series of readings is taken on the axis, at 15 degrees off the axis at either side, at 30 degrees off the axis and so on at 15 degree intervals, a series of polar curves can be plotted to show the sound distribution over the front hemisphere. Fig. 7 shows a series of response curves on and off the axis of a typical but hypothetical speaker. Fig. 8 shows polar curves for the same speaker. Obviously the radii of Fig. 8 are a sort of ground floor plan of the "vertical" curves of Fig. 7 so the whole response of a speaker could be shown by a solid model, whose shape is determined by a long series of response curves taken at intervals of a few degrees; the curves of Fig. 8 are contours of this solid model taken at specific intervals.

These response characteristics of speakers are measured either in the open air or in anechoic chambers so that the surroundings do not influence the readings; yet the speaker will not be so used in real life. It will be obvious that whereas the frequency response determines the nature of the emitted sound, the room itself will decide what happens afterwards, since reflection is differential, both as to direction, determined by the angle at which the sound waves strike the walls and ceiling, and to magnitude, determined by the frequency absorption characteristics of the reflecting surfaces.

It would be possible to find out what happens to each frequency by feeding the amplifier with the output of an audio

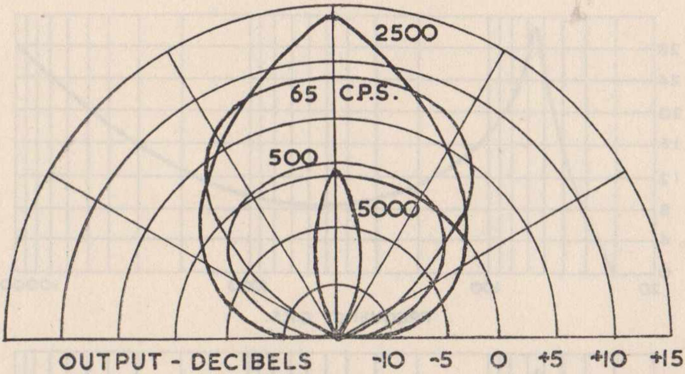


Fig. 8 Polar curves of the speaker whose response is given in Fig. 7.

oscillator and listening for standing waves as explained earlier, but this, unfortunately, does not help very much with complex waves having several simultaneous frequencies, the sort of waves that make up musical sounds. A lifetime might be spent finding the standing waves for all frequencies and adjusting reflectors to eliminate them and still the final result would be only an approximation. Can we find an approximation some other and simpler way? The method I suggest now has not, to my knowledge, ever been made public before.

Another characteristic of a speaker, is its impedance curve. In free air, and on an infinite baffle of negligible interference, the ordinary dynamic speaker has an impedance curve something like that of the curve of Fig. 9A. The peak at the bass end of the frequency scale is caused by the natural resonant frequency of the cone-coil assembly and its associated suspension. Speakers having paper cones with moulded corrugated surrounds resonate somewhere between 35-80 cps, but this resonant frequency is also added to by the resonant frequency of the suspension washer at the apex of the cone, the device which holds the voice coil central in the gap. If the resonant frequency of the cone surround coincides with that of the spider washer, the impedance curve will have a very pronounced peak indeed, but usually the spider washer resonates at a higher frequency than the cone surround, owing to its smaller physical dimensions. The curve would then have two peaks, but the amplitude of the lower peak will be masked by that of the higher simply because if the speaker

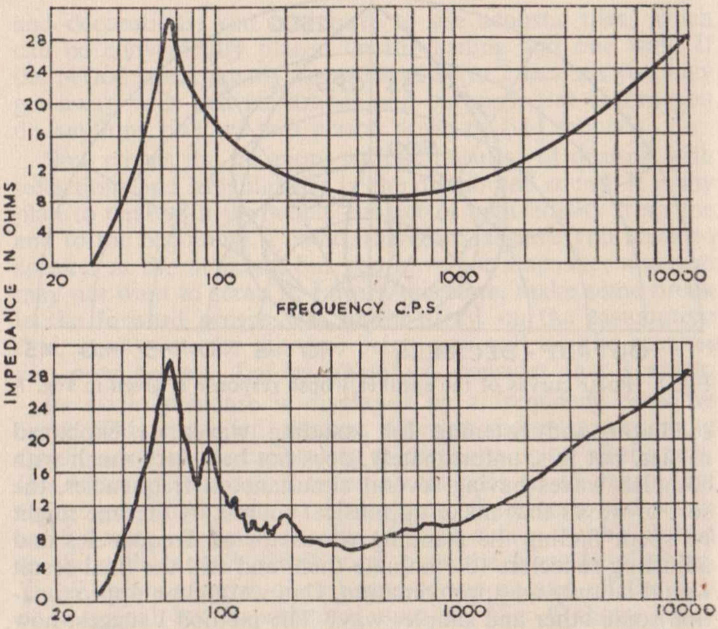


Fig. 9 Impedance curves of a typical speaker. The smoothed curve shows the bass resonance impedance peak and rising impedance with frequency. The actual impedance curve also shows minor resonant peaks due to cone deformation and chassis resonances.

is unable to reproduce a frequency lower than that of the spider washer, the cone surround peak will not show on the curve.

The peak at the treble end is due to the inductance of the voice-coil, apart from certain other subtle mechanical causes; for a given inductance the impedance must increase with frequency, but the increase only becomes appreciable at frequencies over 1,000 cps.; below this the mechanical design of the speaker is more important. The curve of Fig. 9A, then, shows how the impedance varies with frequency, but it is a smoothed curve. If the curve is taken very carefully indeed it will be more like the curve of Fig. 9B, for such phenomena as noding of the cone, radial "break-up" of the cone, even resonances in the metallic structure of the speaker chassis, will be revealed by irregularities in the curve. If the curve is taken again with the speaker in a cabinet of some sort, instead of being mounted on a rigid infinite baffle, the curve will be of a vastly different shape.

The reason for this is that the speaker will only have output when it is doing work. The output of a car engine is measured in a brake-horsepower test; that is, its power output is measured in terms of the work required to stop it. If you race your car engine in neutral it is not doing any work, and has no output to speak of. Similarly, a speaker working into a vacuum has no work to do so it has no output. The impedance curve is therefore, a picture of the work the speaker has to do, and at the highest points the speaker has the greatest output. Any speaker with a fairly high bass resonant frequency, say 60-80 cps., has a very audible bass thump of one note, and the treble resonance is noticeable as a shrieky edge to the music. If the speaker is working into a horn it has a higher efficiency because it is better loaded—instead of dissipating its energy in all directions it is concentrated in a column of air, and the output is also more linear and the impedance curve flatter. We can, therefore, associate speaker efficiency and capacity for work with its impedance curve. I suggest that this simply-determined characteristic can be used as an index of what is happening outside the speaker. I have used the method with great success.

The method is really very simple; it involves setting up a circuit to determine the impedance at all frequencies, and then making adjustments to the room furnishings and arrangements to reduce individual peaks. Of course, it is necessary to get a datum, which involves taking an impedance curve of the speaker in its housing in the open air. Then with this curve before you, you have a basis from which you can compare the performance of the speaker in the room. The open air curve may not strike you as being very good, in which case you would make adjustments in the room to absorb or reflect the sound on a trial and error basis to flatten the curve. If the original curve looks pretty good, you would take care to see that it is not made worse by the room. Even a simple adjustment like moving the speaker about the room will make an appreciable difference in the impedance curve. In this way you may find where it will work best, and where it works

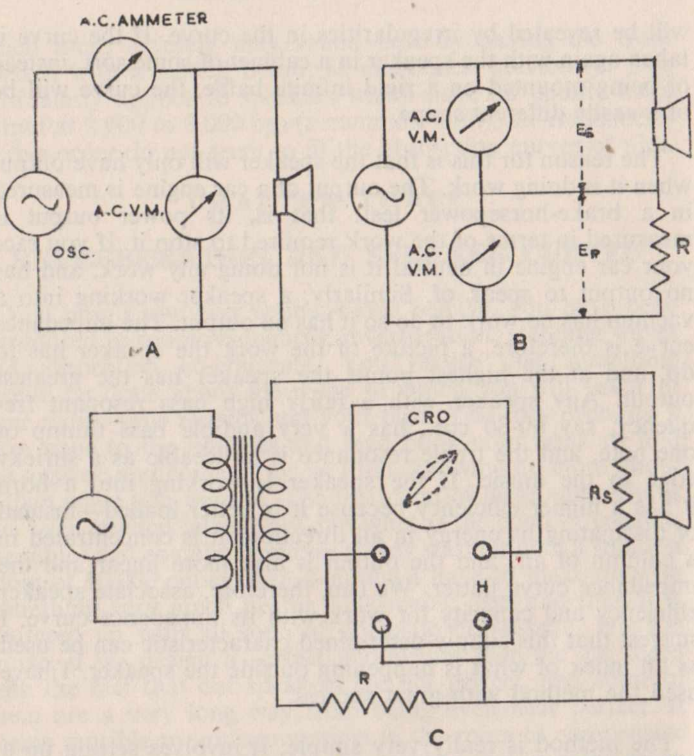


Fig. 10 Three methods of taking impedance measurements of speakers.
A—simple ammeter-voltmeter method.
B—comparison with a standard resistance using two voltmeters.
C—oscilloscope used for direct comparison of resistance and speaker impedance.

best you may be sure is the place where it sounds best. Do not forget that your human "guinea pigs" must be there when you are taking your measurements.

There are three different methods of taking impedance curves easily. For all three an audio oscillator is required, but the rest of the equipment varies. Fig. 10A shows how to measure the current through the speaker and the voltage dropped across it; this requires an a.c. ammeter and an a.c. voltmeter. The impedance at any frequency Z is simply E/I, where E is the voltage reading on the voltmeter across the voice coil and I is the current in amperes through it. In taking the measurements, advance the oscillator in steps of 10 cycles from say, 30 up to 100 cps, then in steps of 100 cycles up to 1,000 cps, and thereafter steps of 1,000 cycles up to the limit. Note particularly the exact frequency at which the voltage rises and the current falls momentarily, which marks resonant peaks.

Since a.c. ammeters are not always easy to come by, another method using two voltmeters is described. This is shown in Fig. 10B. The method is simply to compare the voltage drop across two resistances in series. Select R to be exactly the same as the d.c. resistance of the voice-coil of the speaker. With d.c. passing through the voice-coil and R in series, the voltage drop across each will be equal. Now apply an a.c. source, your audio oscillator. R must be a non-inductive resistor, otherwise its impedance will change with frequency, and if you use a moulded composition resistor, but sure that it will dissipate enough watts. Now apply various frequencies as indicated previously, when the impedance of the speaker can be calculated from the simple formula:

$$Z_s = R(E_s/E_r)$$

Both these methods have the disadvantage that a certain amount of observing meters and simple calculating has to be done. A more elegant and much simpler way is to use an oscilloscope, which has the further advantage that for continuous observation you do not have to observe two separate meters. The hookup is shown in Fig. 10C. Here again R is a non-inductive resistor having the same resistance as the d.c. resistance of the voice-coil. For setting up purposes you will require two non-inductive resistors of the same value as the voice-coil d.c. resistance, since the oscilloscope must be set on a.c. Connect a resistor across each pair of plate terminals, apply a signal from the oscillator and adjust the sensitivity controls of the oscilloscope internal amplifiers so that the trace is a straight line at 45° inclination. If the trace is adjusted so that it passes through a convenient point on the lower left-hand corner of the graticule, then the graticule can be used as a scale in the subsequent measurements.

Now replace the resistor across the vertical plates by the speaker. When the speaker acts as a pure resistance the trace will remain a straight line, but when the inductive and capacitative components take effect the straight line will become a narrow ellipse. It is the major axis of the ellipse in which you are interested.

As the frequency from the oscillator varies and as the ratio between the volts across the speaker and the volts across R vary, so the trace will move from the 45° position, and the relative magnitudes of the two voltages can be measured by counting graticule divisions. This is not absolutely necessary if you are mainly concerned in making the impedance constant. What is useful in this method is that divergence from the normal 45° position can be checked against adjustments to room furnishings. For example, putting your hand in front of the speaker is enough to cause a shift. The effect of putting a diffusing slot in front of the speaker can be instantly observed. Modifications to the housing can be checked instantly.

You will never get the impedance curve flat, but the methods just detailed will indicate the effect of the adjustments you make to your listening room. It may sound very tedious, and I am prepared to admit that it can be a trial on one's patience, but if your desire is honest-to-goodness

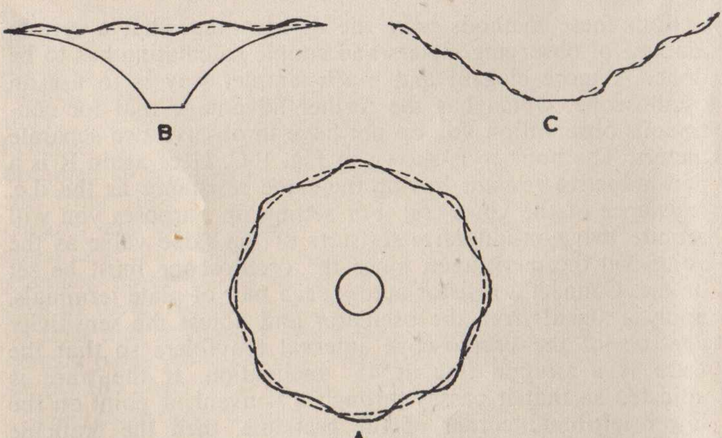


Fig. 11 Deformation of speaker diaphragm at low frequencies.
A—nodes on a straight-sided cone.
B—nodes on an exponentially curved cone.
C—wave motion along the sides of a cone.

high fidelity, then the room must be right. When all is said and done it only has to be done once (unless, of course, you change your speaker and housing) but that is the way to do it. Get the background right and all that you do afterwards can be planned with some degree of certainty. If you do not, then you are inevitably working in the dark, and all the twisting of control knobs on the most elaborate preamplifier will not get matters right.

CHAPTER FOUR

SPEAKER DESIGN

The home constructor, and the enthusiastic amateur, are at a serious disadvantage in the matter of speaker design, although many volumes have appeared dealing with the subject. Making a speaker is not easy, for in order to secure reasonable sensitivity, tolerances have to be close in the voice-coil, yet it is made from materials which do not lend themselves to fine limit construction. I shall assume, therefore, that in the matter of speakers you will buy something ready made. The problem is, therefore, which speaker to buy and how to use it.

Taking direct radiators first, loudspeakers that are mounted on flat or folded baffles, the first thing you see is the cone. This has four properties having bearing on its performance: diameter, included angle, shape of cross-section on the axis and material.

I have said that my experiments have shown that the optimum size of cone is about 8 or 9 inches. Why should this be so? Since a diaphragm is not infinitely rigid it must distort when force is applied to it from the voice coil. Major distortions occur in three different ways, particularly at low frequencies. Assuming a free-edge cone, a straight-sided cone develops flower patterns as a result of nodes when viewed from the front under a stroboscopic light; an exponential cone develops nodes in an axial direction when viewed from the side; any cone develops transverse wave motion along the cone. These three phenomena are illustrated in Fig. 11.

Now for a given material of specified thickness it does not require a great deal of imagination to see that the larger the cone in Fig. 11A, the more likely will there be an inherent tendency to develop nodes. Make up two cones of the same included angle with ordinary writing paper, one having a diameter of 3 inches and the other of 6 inches. You will find that the smaller cone is less easily deformed by pushing the free edge. This lesser rigidity at the edge can be counteracted by making the larger cone of thicker or stronger material or by making the included angle narrower (thus effectively redusing the diameter of the cone). Unfortunately, in a practical speaker, this has a detrimental effect on the performance because the heavier the cone the less response in the treble, and the narrower the cone the more intense is the focusing of the high notes. Even a flat diaphragm will not give uniform spherical radiation at all frequencies and a narrow angle cone produces a highly concentrated beam for all frequencies over about 1,500 cps.

You may well ask, therefore, why not let the nodes form and stop worrying? The answer to this is that energy transmitted to the cone through the medium of the voice-coil is being used up to produce the nodes in the cone instead of pushing the air in front of the speaker and so the response at the low frequencies will be reduced. A loudspeaker with linear response converts all the applied electrical energy into air (sound) waves; none is wasted in deforming parts of the speaker. The formation of nodes must be prevented by making the diaphragm as rigid as possible.

The first widely adopted method was to make the cone with an exponential cross-section as shown in Fig. 11B. Such a diaphragm is very rigid across a diameter, but now, as I have shown, the nodes develop in the direction of the axis of the cone. The flatter shape of the exponential diaphragm gives less focusing of the highs, but to stiffen it circumferentially, concentric corrugations are moulded in the cone. Almost every speaker you examine will be found to have such corrugations incorporated in the diaphragm. But when we consider the case of Fig. 11C, a defect which has only recently been noted by some loudspeaker designers, the circumferential corrugations are no help at all, for they give no stiffness along the material of the diaphragm.

B. F. Miessner has developed an ingenious solution to this difficulty by cementing soda straws to the cone, like spokes in a wheel. If too many are used, the mass of the diaphragm is unduly increased, and the method, from a commercial point

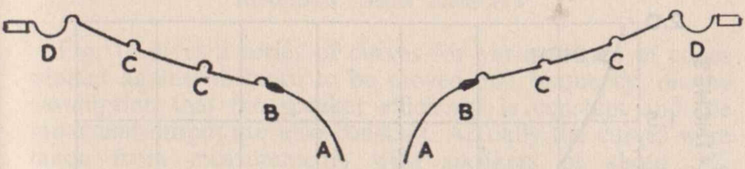


Fig. 12 Cross-section of the Hartley 215 speaker diaphragm.
AA—exponentially curved cone apex.
BB—isolating flexible compliance.
CC—circumferential ridges to give radial stiffness.
DD—fully flexible flannel surround to give free edge suspension.
The curvature of the diaphragm from B to D is the reverse of the ordinary exponential diaphragm.

of view, would be very costly. The deformation in a straight-sided cone is not as great as in the flat area of an exponential cone, as you can imagine by thinking of the plane rigidity, if I may call it that, of a sheet of paper as compared with a cone; it occurred to me that if the flat part of the exponential cone could be abolished, a substantial improvement could be brought about. Experiment proved this to be the case. One solution to this problem (see Fig. 12) shows a cross-section of the diaphragm of my 215 speaker, in which the outer part of an ordinary exponential diaphragm has imparted to it a reverse curvature, so that the outer zone is itself reasonably rigid axially. Just before the flattest part the wave motion has been interrupted by the presence of points "B". The compliance (points "B") was not introduced specifically for this reason (its real purpose will be described later), but its presence does act as a barrier for the wave motion originating in the apex of the cone. What is transmitted beyond this point is neutralised by the curvature of the outer part of the diaphragm.

So far, then, it would seem that there are many snags attending the use of a large cone; why do so many loud-speaker manufacturers use them? Let us summarise these drawbacks; the large cone is heavier than a small one, so restricting the response at high frequencies; its mass is such that transient response is impaired because it is more difficult to start a heavy object moving rapidly than a light one; and its size makes it too flexible in various directions, thus causing loss of bass through energy being wasted in deforming the cone. Everything points to the use of a small cone, but the

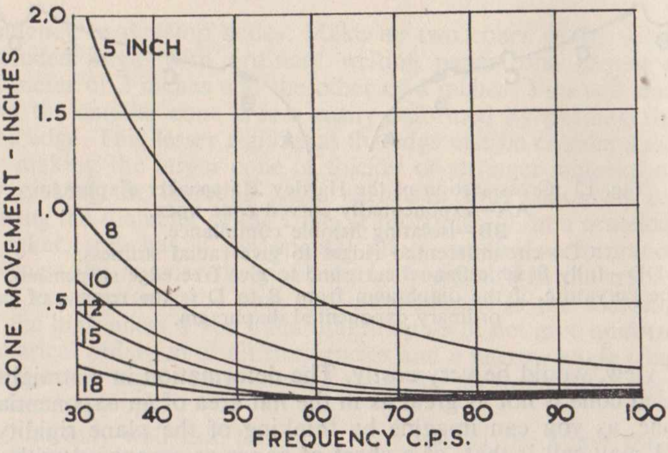


Fig. 13 Movement required of various diaphragm sizes at low frequencies to avoid bass distortion. Speaker efficiency about 5%. Power input 5 watts.

small cone has one fatal drawback — its power-handling capacity is very seriously limited.

Apart from the resistance to movement offered by the rear suspension spider and the front surround of the cone, that of the air in front of the diaphragm is substantial, as indeed it must be, since the function of the loudspeaker is to move air to create sound waves. It will be obvious that a small cone will move less air than a large one, and the air resistance to the movement of a small cone is less than that of a large one. For a given input, therefore, the small cone moves forward more easily and has less output, and because it moves more easily it reaches its limit of movement, determined by the suspension system, sooner than in the case of a large cone. This is of importance only at the low frequencies, for the amount of movement for a given input depends on the frequency of the current applied to the voice-coil. This is why a large cone is said to be better for bass reproduction than a small one, but this only holds good for a given amount of displacement of the diaphragm. A small cone can move as much air as a large one provided it has greater freedom of movement.

Fig. 13 gives a series of curves for various sizes of cones plotted against distance to be moved and frequency, on the assumption that the speaker efficiency is constant and the input and output are also constant. Actually the curves were taken from measurements with speakers of about 5% efficiency (a not unusual figure for ordinary dynamic speakers) with an input of 5 watts; this would give an acoustic output of approximately 0.25 watts. It will be seen that to maintain constant output the movement required from a 5 inch cone rises very rapidly as the frequency approaches 30 cps. whereas with an 18 inch cone the increased movement required is very small. It will also be noticed that as soon as the cone size has increased beyond 8 inches, the advantages of increased power-handling and acoustic output is proportionately much less, for the curves crowd together as the cone size increases.

Despite the fact that this book is concerned only with high fidelity reproduction, I maintain that what happens below 40 cps. does not matter very much. It is almost impossible to hear a 32 cycle note but it can be felt, and I believe that to attempt to create this "feeling" is a waste of time, money and effort. Even 50 cps. is a very low note and quite a high proportion of high fidelity installations cannot reproduce it without some sort of distortion; I am certainly content to have the lower limit of my frequency range at 40 cps. but it must be free from distortion. I would rather have a limit of 50 cycles without distortion than one of 40 with some distortion.

At this lower limit, therefore, a study of the curves of Fig. 13 suggests that there is not much to be gained by having a cone larger than 10 inches, for you must remember the serious disadvantages of large cones from the point of view of treble reproduction, noding and wave transmission along the cone itself. But you will still want to know why, in the face of this, speakers with cones from 12-15 inches are readily obtainable in any Hi-Fi shop. There can be no definite answer to this question. We do know that there are many individuals and speaker manufacturers who believe that a large speaker gives "better" bass reproduction than a small unit, so the bigger and more expensive the speaker the "better" the bass. Hand in hand with this argument is the one which states that it is well known that large cones have no treble, which is why the best systems are multi-channel jobs, where a tweeter looks after the highs while the woofer looks after the lows.

Both these arguments are completely specious. The large speaker will certainly give more bass than the small one, but its larger output has more distortion owing to noding and wave transmission. There is the further disadvantage that the air partially enclosed by the large cone has a resonant frequency at a point which can seriously impair the reproduction by imposing a one note hoot on the whole sound coming from the loudspeaker. This you can test for yourself. Place one ear right inside the cone of the speaker and tap the cone with your fingernail. You will hear at least one low sound which is caused by the resonant frequency of the suspension. Now grip the cone-coil joint with two fingers while you tap the cone with a fingernail of the other hand and you will hear another note of higher frequency than the previous one. This is caused by the reaction of the paper of the cone on the air within it and causes the hoot I have mentioned. It is avoided by taking care that the air within the cone is not even partially enclosed, best achieved by making the cone as flat as possible but not so flat that it allows axial nodes to develop easily, and obviously still more certainly achieved by making the cone small.

If you have a pair of musically trained ears and are listening for distortionless musical reproduction, if your ears have not been pre-conditioned by long bouts of listening to sound reproducers, hi-fi or otherwise, you will hear this hooting effect with any large diaphragm speaker. It has nothing whatever to do with the bass resonant frequency, it is a necessary acoustic accompaniment of a large cone. The nearest equivalent to it in the instruments of the orchestra is found in the drums, which emit sounds at the resonant frequency of the stretched skins but have an accompaniment on the resonance of the air inside. This effect I sardonically christened many years ago as "the characteristic sound of a loudspeaker" and there seems to be absolutely no cure but that of using smaller cones of the correct design. The question of power handling is answered by using two smallish speakers instead of one large one, perhaps a superficially clumsy way of doing it but there is great merit in using two speakers widely spaced for they give an extremely good imitation of binaural reproduction.

Finally, we come to the question of the cone. On the face of it nothing need be said on this point as all diaphragms seem to be made of paper pulp treated with some sort of dope or varnish; but as in the following chapter we shall meet tweeters with metallic diaphragms, it seems desirable to point out that the material of which the diaphragm is made has a bearing on the sound emitted by the speaker, independent of frequency response. This must happen with speakers as it happens with musical instruments, the woodwinds of the orchestra sound different from the brass, the wooden pipes of the organ different from the metal pipes. These musical tubes are not diaphragms but they are part of a vibrating system which includes air, as is the diaphragm of a loudspeaker. Speaking very crudely it might be said that wooden pipes sound "tubby" and metal pipes sound "tinny" or shrill; this seems so obvious that it is hardly worth saying, yet the obvious is sometimes overlooked in designing loudspeakers.

The cone should be acoustically inert, it should impose no colouration of its own. It is my considered opinion that the cone should be made of a very high grade Bakelite resin, containing not more than 10% rag tissue as a binder. To make such a cone in quantities by moulding is an almost impossible manufacturing proposition, owing to the danger of the moulds sticking; but cones can be fabricated out of flat sheets of this material. Unfortunately only straight-sided cones can be made in this way and, as I have shown, a straight-sided cone is not very good from the point of view of high note diffusion. Bakelite of this type is strong, resists wave motion along the material and gives extremely good treble response. I made many thousands of speakers in the 1930's with cones of this material, but finally abandoned it because of the manufacturing difficulties and the focusing of the highs.

Moulded paper pulp is now used almost universally because it is cheap, light, comparatively strong and can be given almost any shape. Moreover the outer surround can be incorporated with the cone and such intermediate corrugations as the designer may call for are easily provided. Inspection of such a cone must, however, be particularly directed towards any apparent varnish applied to the paper. Without varnish the material is hygroscopic and the cone will become limp in a humid atmosphere, spoiling the response at all frequencies; too much varnish, intended to make the cone stiffer, will only add to the weight, reduce transient response, and do nothing to improve the output at high frequencies. A simple test for varnishing is to wet a fingertip and press the surface of the cone; if the moisture seems to be absorbed as it would be with blotting paper it can be assumed the speaker must be kept very dry to give its best performance.

Some cones will, however, show a glazed hard surface for a few inches near the apex; this is due to hot pressing between dies after painting with Bakelite varnish and is intended to help the treble response. Only the apex of the cone propagates the highs and obviously the cone "breaks up" in the process; the purpose of the Bakelising is to restrict the break up to the zone beyond the apex, so that the extreme treble response is under control.

The flexibility of suspension can be checked by grasping the cone between the thumb and index finger of each hand across a diameter, the thumbs in front of the cone, the fingers inserted through the cone basket. The cone is then gently pulled forward and pushed backward and an estimate made of the amount of permissible movement. When this has been determined reference back to Fig. 13 will show what performance can be expected at the low frequencies. If, for example, a loudspeaker with a 12 inch cone has less than a quarter-inch free movement it cannot reproduce a 40-cycle note output with 5 watts input because it will be overloaded. A little at 80, and most at 120 cps. will be reproduced. This refers only to the freedom of suspension; other factors involving the magnetic system and dimensions of the voice-coil will be discussed later.

CHAPTER FIVE

DUAL CONE SPEAKERS

Reproduction of frequencies up to about 2,000 cps. is a matter of getting the cone size, shape, and material right, as has been explained previously. Provided sufficient freedom of movement is incorporated in the design of the cone suspension to give the necessary bass output at low frequencies and the cone is stiff enough to counteract the tendency to develop nodes, no further attention need be paid to the material of the cone and the method of making the voice-coil. Cone break up will give an output above 2,000 cps. but the response will be uneven and certainly deficient above 5,000 cps. The difference between an ordinary speaker such as is produced by the millions for commercial radios and television receivers, and an alleged high fidelity speaker is in the attention that has been paid to the matter of reproducing frequencies above this figure and at the extreme low end.

Some designers maintain it is impossible to get a wide smooth response up to 10,000 or 15,000 cps. from a single speaker unit, and confine their attention to tweeter-woofer combinations; others, including your present author, do not accept this. They maintain that the disadvantages of the multi-channel speaker outweigh the advantages and that the desired results can be secured with only one magnet system, although it is generally agreed that something special has to be done to the cone, and I insist that something special has to be done to the voice-coil as well. Assessing the merits of a complex diaphragm speaker as compared with a tweeter-woofer combination by looking at it involves some knowledge of the merits of the various methods used by designers.

The first step forward in extending the response of a diaphragm was made by P. G. A. H. Voigt in, I think, 1934, when he patented and produced a composite diaphragm consisting of a main cone to which was firmly cemented a tweeter cone of smaller diameter and narrower angle. Voigt's idea was that the main cone was too massive to reproduce the extreme highs; it was better that these should emanate from a smaller lighter cone. It was of little consequence that the two were driven by the same voice-coil, since cones break up in any case; using a subsidiary treble cone simply meant that the break up came within controlled limits.

This invention has been widely copied both in the U.S. and Britain. One application of Voigt's invention resulted in a commercial version which incorporated a twin cone. Admittedly, this improved the treble response considerably, but it should be remembered that at the time Voigt's speakers were introduced they were intended for use with logarithmic or exponential curved horns. A speaker working into a properly designed horn has a much greater electro-acoustic efficiency than a speaker working in a flat or box baffle owing to the better loading of the diaphragm. For a given acoustic output, a horn-loaded speaker requires much less electrical input and the Voigt speakers produced a very healthy noise with inputs of only 1 or 2 watts. Even more was achieved in the celebrated Western Electric 555 theatre speaker whose efficiency was considered by some to be phenomenal.

I was so impressed with the results Voigt obtained with horn loaded speakers that I supposed, incorrectly as it turned out, it would work equally well with baffle speakers, and I obtained permission, under licence, to use the idea in my own loudspeakers. I then discovered the limitations of the idea. The outer edge of the tweeter cone is quite undamped, except for the stiffness of the material of which it is made. If you flip the edge of the small cone of any double cone speaker with your finger nail, in a direction towards the centre of the speaker, you will hear the same sort of noise as if you flipped the edge of an ordinary sheet of paper, but probably sharper, because these small subsidiary cones are usually treated with Bakelite varnish and pressed in dies (this being shown by the much smoother surface of the paper as compared with the main cone's surface). This same buzzy sound is created when the whole assembly is driven hard with big inputs, the frequency of the emitted note being a function of the size of the small cone and the material of which it is made. This spurious note has nothing to do with the frequency of the applied signal, it is merely a resonance generated by sudden impacts of energy from the voice-coil. If an attempt is made to kill this resonance by applying damping material to the edge of the small cone, the mass of the cone is so increased that the extra treble response is lost. The only way to avoid this distortion is to limit the input to the speaker.

Unless the speaker is sufficiently sensitive, that is, efficient, the output may not be adequate; as I have pointed out, loading the diaphragm by using a horn is the certain way of achieving high efficiency, but for a baffle speaker improvement can be gained by using small magnetic gaps and high flux sensitivity in the gap. This, however, introduces trouble at the bass end, for freedom from bass resonance can only result from freedom of cone movement, and this implies generous gap clearances and special attention to the design of the magnetic field so that at no point of its excursion does the voice-coil pass into a less intense field. These considerations impose a limit to what can be done by stepping up sensitivity in a baffle speaker. If the speaker is sufficiently sensitive to produce a respectable acoustic output without generating buzzes from an improperly designed tweeter cone, it is almost certain that the magnetic circuit will be of a type that will introduce bass distortion unless the input is limited. The design engineer is faced with a serious problem. The ultimate in treble performance results in poor bass and vice versa. As a result, most careful design is needed to effect a suitable compromise.

With this in mind, therefore, it is possible to formulate a few simple rules when assessing twin-cone speakers. Measurement in a laboratory with small constant inputs of varying frequency show that the use of a subsidiary tweeter cone gives more treble than a speaker with a simple diaphragm. When such a speaker is used in a cabinet or flat baffle, the power necessary to produce adequate acoustic output will set up a spurious resonance in the free edge of the small cone unless special steps are taken to prevent this. Twin-cone speakers are, accordingly, better with horn loading as this gives better electro-acoustic efficiency. I would estimate that an efficiency of 15% would be necessary to put the tweeter cone beyond suspicion, and this figure of efficiency is not easily reached except with a large and very carefully designed horn.

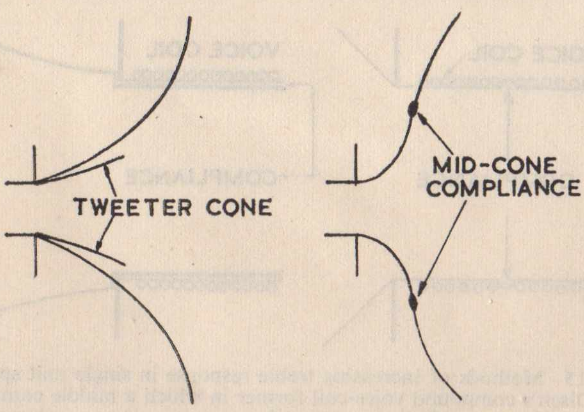


Fig. 14 Cross-section of a typical twin cone speaker (A). And of the Hartley 215 in which the cone is divided into two sections and rejoined by a flexible compliance, as shown in Fig. 12 (B).

The failure of the Voigt twin-cone idea when applied to baffle speakers kept nagging at me for some years. I felt that the basic principle was right—that a small cone should be used for the treble, but there seemed no way of stopping the buzzing of the free edge when driven hard. As I have said, loading this edge to kill the resonance was no remedy, for the mass of the loading neutralised the lightness of the small cone. It finally occurred to me that if the loading weight could be taken away from the tweeter cone the desired results would be obtained, and then I had my brainwave. Fig. 14A shows a section of the typical twin-cone speaker; while my solution of the problem is shown in Fig. 14B, where the apex of the large cone has virtually been removed and the small cone put in its place. The two parts of the cone are joined by a small zone of flexible material to form a compliance. The idea behind this innovation was that the weight of the flexible material damping the erstwhile free edge of the tweeter cone was supported by the main cone, but if sufficient flexibility existed in the compliance, the small cone would still be free to oscillate at the higher frequencies. At the same time it would have to be sufficiently stiff to transmit substantial movements set up at lower frequencies so that the whole cone moved at those frequencies. It called for a considerable amount of experimenting to find just the right

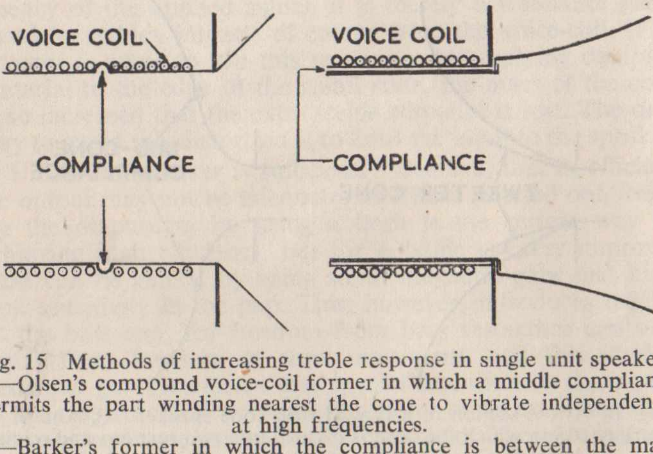


Fig. 15 Methods of increasing treble response in single unit speakers.
A—Olsen's compound voice-coil former in which a middle compliance permits the part winding nearest the cone to vibrate independently at high frequencies.
B—Barker's former in which the compliance is between the main former (metal) and the subsidiary former carrying the winding.

degree of flexibility for the compliance, but tabulated results finally enabled a schedule to be compiled for any frequency response desired, within the limits of a two-cone speaker.

The other part of the diaphragm assembly that naturally restricts extreme treble output because of its mass is the voice-coil itself. It is not possible to reduce the weight of this component beyond the point where it loses rigidity; the coil is subjected to heavy a.c. current impulses and must, therefore, be quite strong. The coil and its former (coil form) usually consist of a paper tube on which is wound two layers of copper wire, the whole firmly cemented together. The method by which the voice-coil former is cemented to the apex of the cone is vitally important, for any weakness here will result in a peak in the response curve, usually at about 3,000 cps. This effect is made use of in cheap mass-produced speakers to provide a spurious treble output to compensate for the top cut-off in cheap radios, but is quite out of place in high fidelity work. The weight of the coil can be reduced by using aluminium wire and this provides some small increase of response at the higher frequencies. The problem is to reduce the weight of the coil-former assembly as much as possible.

The first original approach to solving this problem was that of H. F. Olson, who described a voice-coil in two parts, connected by a compliance. Fig. 15A shows the arrangement of two voice-coils in series but separated by a flexible compliance in the former; the bass coil is heavier than the treble and is by-passed by a capacitor of such size as to act as a short-circuit at high frequencies. At low frequencies the whole moves together; at high frequencies the flexibility of the compliance permits the treble coil to move independently (the required movement is really very small). The idea works all right, but the compliant former is a troublesome thing to make with any consistency and the separate leads for the by-passing capacitor difficult to provide; but, like so many things in loudspeaker design, it seemed to set up trains of thought in two minds in Britain.

Of my own case I can speak with authority. It seemed to me that the logical thing to do was to have the treble coil inside the bass coil, concentric with it and separated by a plastic film. In the other case I have no justification in linking the Olson idea with the "Duode" idea of A. C. Barker. It was a case of two independent workers hitting on much the same idea at the same time, as has happened often enough in the history of human thought. Barker's invention, patented in Britain and described in the "Wireless World" in about 1938, described a composite voice-coil consisting of the regular winding separated from a secondary "winding" by a flexible compliance, but the secondary winding was of only one turn, being an aluminium tube carrying the plastic film and the primary winding; the aluminium tube was the only part of the assembly fastened to the cone. This arrangement is shown in Fig. 15B.

Electromagnetic connection to the secondary of the "transformer" thus formed is purely inductive. The coupling is negligible at low frequencies and the whole assembly moves as a solid entity; at high frequencies currents are induced in the tube which moves independently of the winding proper, so response in the upper register is very good. I have not had a chance to dissect a Barker speaker so I am unacquainted with the details of his design; but in the case of my own speakers I can say that the degree of compression of the compliance has a decided influence on the response. I have found that winding the coil straight on to the plastic is useless because the tension of the wire cannot be maintained with great accuracy. I wind the voice-coil on to a very thin paper former, slip the wound former over the plastic which is already fitted to the aluminium tube, and then expand the tube by a predetermined amount. The combination of this compliance already described gives a response of not more than 4db down at 20,000 cps. over an approximate cone of radiation of 120°.

It seems fairly obvious that the greater the magnetic flux in the gap in which the voice-coil works the more sensitive the speaker. The equally obvious way to get more flux is to use a big magnet. In the case of high fidelity speakers it is not as simple as that since flux alone is not all that matters; it is just as important that the field of flux should have certain characteristics. Cheap mass-produced speakers have small magnets because permanent magnet steel is very costly; the efficiency of such speakers is secured by having the smallest possible clearance between the voice-coil and the walls of the gap. Such small clearances are only practical when the permissible movement of the voice-coil is small, since it is almost

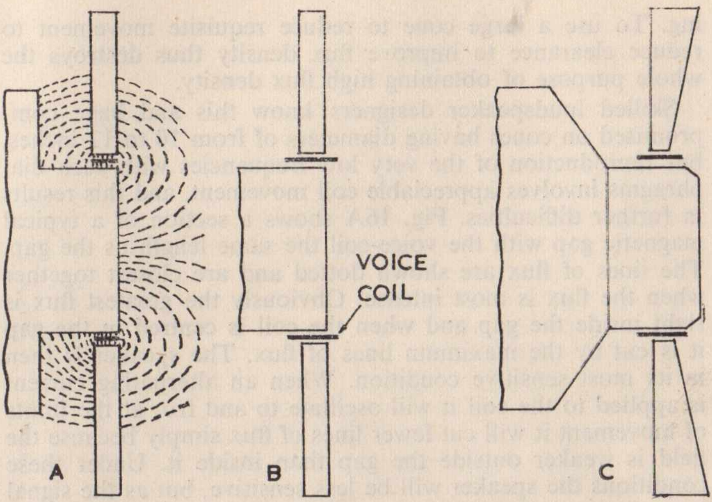


Fig. 16
A—voice-coil same length as gap, also showing the flux distribution in front of and behind the gap. In such a magnet design the field of flux is not symmetrical about the gap.
B—long coil in short gap to maintain constant cutting of the flux in the gap.
C—Hartley design of symmetrical flux field gap.

impossible to preserve absolutely true axial movement with the materials of which the coil, cone and suspension system are made. Limitation of the axial movement of the voice coil, through tight suspension, results in a bass resonance of comparatively high frequency. Some such speakers are incapable of reproducing any frequency below 120 cps.

The high fidelity speaker is required to reproduce very low frequencies and this demands much greater freedom of movement; as I explained earlier, the amount of movement required to reproduce a certain bass frequency depends on the size of the cone, but large cones have certain acoustical disadvantages; they also have the physical disadvantage that, being heavy, they are difficult to start moving and difficult to stop moving. The former property takes the sharp edge off transients; the latter spoils the damping. The attributes of high flux density apart from improved sensitivity, are good "attack" (immediate response to transients) and good damping. To use a large cone to reduce requisite movement to reduce clearance to improve flux density thus destroys the whole purpose of obtaining high flux density.

Skilled loudspeaker designers know this and have compromised on cones having diameters of from 10 to 12 inches, but reproduction of the very low frequencies with such diaphragms involves appreciable coil movement, and this results in further difficulties. Fig. 16A shows a section of a typical magnetic gap with the voice-coil the same length as the gap. The lines of flux are shown dotted and are closest together when the flux is most intense. Obviously the greatest flux is right inside the gap and when the coil is centred in the gap it is cut by the maximum lines of flux. The speaker is then in its most sensitive condition. When an alternating current is applied to the coil it will oscillate to and fro; at the limits of movement it will cut fewer lines of flux simply because the field is weaker outside the gap than inside it. Under these conditions the speaker will be less sensitive, but as the signal input is constant the acoustic output will be less when the coil is partly outside the gap; the result will be a wobble of twice the frequency of the applied signal.

Someone once called this the "Döppler" effect in speakers, apparently under the impression that the wobble tone was due to the diaphragm approaching and receding from the listener's ear. It is nothing of the sort and despite the audio pundits I maintain there is no Döppler effect with speakers, a fact I have demonstrated to many electronic societies by the simple experiment of demonstrating one of my own speakers moved to the limit of the cone excursion by an applied 50-cycle signal with the addition of a 1,000 cps. signal. The two frequencies are heard separate and distinct, with no variation in the pitch of the 1,000 cycle note.

In Fig. 16B the voice-coil is seen to be twice as long as the gap. Provided either end of the voice-coil winding does not at any point of its excursion pass within the gap itself, then, to a great extent, the number of lines of flux cut will be equal and the phenomenon of the bass modulating the treble will not occur. Even then, however, the magnetic field is not symmetrical about the gap, because of the natural cussedness of things. There is no need to embark on an exposition of magnetic theory; I need only explain that there are magnetic characteristics of materials resembling the units of electricity.

An electrical conductor can carry just so much current and if this is exceeded the conductor gets hot and finally melts, as when you blow a fuse. A magnetic conductor has permeability which represents its flux carrying capacity, just like an electrical conductor, depending on the area of cross-section and the nature of the metal. But in a magnetic circuit you cannot blow a fuse, the conductor simply refuses to pass any more flux; it is said to be saturated. An electrical conductor has resistance; similarly a magnetic conductor has reluctance. A current passing through a conductor does not spray the moving electrons outside the limits of the conductor, but it creates an external magnetic field. The magnet in a speaker not only creates flux in the pole-pieces but it also creates an external magnetic field (as you can demonstrate with the old school boy experiment of sprinkling iron filings on a sheet of paper placed on a horseshoe magnet). These lines of flux outside the magnetic circuit proper are called leakage flux.

A given size and design of electro-magnet or permanent magnet has a magnetomotive force which provides the flux in the magnetic circuit. The permeability of the centre pole of the magnet system determines the maximum flux that can be created in the gap, but the reluctance of the pole-piece tries to stop it. In addition some of the flux is lost as leakage flux between the centre-pole and various parts of the whole magnet system. Reluctance is reduced by increasing the diameter of most of the centre pole, as shown in Fig. 16, for by doing so an improvement of something like 50% in useful flux in the gap can be obtained as compared with a pole-piece having the same diameter throughout. But the presence of this extra mass of metal near the front plate, which is the other pole-piece, results in an increase of leakage flux, for the flux naturally takes the line of least resistance. My sketch (Fig. 16A) therefore shows flux lines which have avoided the actual gap completely and it will be obvious that the leakage flux behind the front plate is greater than in front of it, simply because of the unavoidable presence of the centre-pole. The field must therefore be asymmetrical about the gap with such an arrangement as that shown in Figs. 16A and 16B.

This sets up a condition of strain in the loudspeaker. Let us assume that the first half cycle of an applied signal drives the voice-coil inwards. The repulsion of the magnetic field in and behind the gap drives the coil forward on the second half cycle, and into a magnetic field which is weaker. On the next half cycle this weaker field has not the same repulsive effect as that behind the front plate, so the tendency of the voice-coil is to stay outside the gap. If the cone is aperiodically suspended on threads, it will be driven out of the gap and stay there. This effect I originally christened electro-mechanical rectification (in 1926 when I first noticed it). In practice of course, the cone is returned to its normal position by the action of the outer surround and the rear suspension spider, but this only tends to neutralise the effect, the basic cause is still there. Hence the condition of strain, which I have observed, helps to create the phenomenon of cross-modulation.

In my search for a method of producing a symmetrical field, I had no alternative but to make up a large number of experimental magnets and exploring their fields with a very shallow search coil connected to a fluxmeter. I do not think there is any other way of doing it, and it is extremely tedious. Owing to the bulk of the centre-pole behind the front plate some extension of the pole is necessary and this must be supplemented by chamfering the front plate itself. Fig. 16C shows the magnet system I finally determined as a result of many experiments and it does give a truly symmetrical field, but has some loss of sensitivity. If the centre-pole is saturated (as it would be in the most economical design) the total flux behind the front plate has to be transferred to the front and this lessens the actual flux in the gap. Since I insisted on a freely suspended diaphragm, I had to choose between good sensitivity with bass cross-modulation or lower sensitivity with distortionless bass—one more instance of loudspeaker design always being a compromise.

If the speaker is not fitted with a dustcover and has a suspension spider of the open type, you can see if the voice-coil winding sticks out of the gap. Then you can grasp the cone between forefingers and thumbs and pull it towards you to the limit of its movement taking great care not to overstrain the suspension. If there still seems to be plenty of winding in the gap, then the coil is longer than the gap and the risk of cross-modulation is reduced. However, many speakers are fitted with dustcaps and closed rear suspensions, making visual examination of the coil and magnet systems almost impossible. The properties of the speaker must then be tested electrically (if the dealer will allow you to do so!)

Apply, from an audio oscillator or from the a.c. power line through a "Variac" or other variable transformer, an alternating current (anything between 40 and 60 cps.) of such magnitude as to move the cone to its limits. This point is determined by gradually increasing the input to the speaker until there is a suggestion of the voice-coil former or the rear suspension hitting on the centre-pole or front plate, then reducing the input slightly. Now, from another signal source, apply a 1,000 cycle signal and listen carefully. If the 1,000 cycle note is modulated in strength by the low frequency note usually having a sort of burbling effect, cross-modulation is present. You may think that this would not be heard on ordinary music, but it will. One bang on the drum will affect the sound from the rest of the orchestra and one held pedal note of the organ will make all the higher frequencies sound dreadful.

CHAPTER SIX

MULTIPLE SPEAKER SYSTEMS

Any speaker has overall colouration of the reproduced music. A musically trained ear can tell at once if what he is hearing is the original performance or a reproduction of it. If there is no measurable distortion in the reproducing equipment, there is still the colouration by the diaphragm material. This need not be a matter of great concern for the human ear is an adaptable sort of device, and within a few minutes will accommodate itself to this subtle distortion and ignore it, but only if it is constant. A two-way speaker system consisting of different diaphragms cannot maintain constant colouration over the whole frequency range.

Every musical instrument emits a fundamental frequency and a series of harmonics, ranging from the comparatively simple waveform of the flute to the highly complex acoustic output of the oboe (to take only the woodwind section of the orchestra). Depending on the frequency of the original instrumental note and the crossover frequency of the two-way speaker, the fundamental and the first two or three harmonics will be reproduced by the woofer and the rest of the harmonic range by the tweeter. It follows that the characteristics of the reproduced note will vary with frequency, simply due to the different colourations of the woofer and tweeter diaphragms, and a sensitive ear will not like it.

I venture to suggest, therefore, that in a multi-channel speaker system, it is logical to assume that the diaphragms should be of similar material and in any event not metallic, which everybody knows has a ringing quality; our diaphragms should be as inert as possible. What is just as important is the spatial relationship of the two or more diaphragms. If you set up two identical speakers some distance apart and drive them with the same signal you will get quite an impressive imitation of stereophonic reproduction. It is not true stereophony since only one channel is used, but the effect is noticeable as long as you are not equidistant from the two speakers. This effect is most noticeable when, if the two speakers are in the two corners at the ends of one wall, you sit near an adjacent wall.

The effect is due to the sound from one speaker being out-of-phase to some extent, with respect to the other, since the sound takes a little longer to travel from the more distant speaker. The two outputs are combined in the human hearing mechanism to create an illusion of depth; but the effect will only be obtained if the two speakers each reproduce the whole frequency range. If one of the speakers is a tweeter and the other a woofer, all you hear are the two separate outputs, the treble coming from one corner, the bass from the other, no matter where you are sitting in the room.

It follows, therefore, that a tweeter-woofer combination must be so disposed that the two sound sources are as close together as possible. Ideally they should coincide, which accounts for the development of co-axial speakers. But, as happens over and over again in speaker design, one problem solved leads to another requiring solution, in this case the reaction of one unit on the other. I can illustrate this by referring to some of my early work on two-channel systems.

In 1927 I was fully aware of the difficulty of making one dynamic unit cover the whole frequency spectrum and considered methods of propagating the extreme highs from a separate unit; as I also wanted some measure of co-axiality I conceived the possibility of putting the woofer inside the tweeter! This thought was not quite so crazy as it seems, for I had done quite a lot of work on electrostatic speakers and they had very good treble and very poor bass. The larger an electrostatic the better it is equipped for radiating sound over a large front, so I mounted my 10 inch dynamic unit on a baffle, and fixed the electrostatic unit on the front of the baffle, with, of course, a hole cut in the centre to avoid masking the woofer. At other than small inputs to the speaker I found that the pressure of the sound waves from the woofer deflected the foil of the electrostatic, causing modulation of the highs by the lows from the woofer. This suggests some thought should be given to the relative placing of two speakers of these types now that electrostatics are being re-introduced after a lapse of 30 years.

I believe that the best way of laying out a dual-range speaker is to have each unit horn loaded, so as to avoid interaction between the two units (horns being much more directional at the sound source), and if the tweeter horn can be curved into the mouth of the woofer horn, co-axiality is achieved. Discussion of this, however, is best left until I deal with horn-loaded speakers in general in a later chapter, so we can resume our discussion of existing co-axial speakers.

An original and ingenious attempt to resolve the problem of maintaining similarity of cone material with co-axiality is found in the "Duo-Cone" principle of H. F. Olson. Fig. 17 shows a section of the cone assembly and the magnet system. This is a true two unit assembly, for each cone has its own voice-coil, but the outer suspension of the tweeter cone is cemented to the diaphragm of the bass unit, this providing some measure of independence of movement. It will be obvious, of course, that the movement of the bass cone must be transmitted through the tweeter cone suspension at low frequencies, even if movement of the tweeter cone is not transmitted to the bass cone at high frequencies (the relative mass of the two cones has considerable bearing on this), unless something is done to prevent it. The inventor claims that adequate venting of the air space behind the small cone can reduce this transfer of movement to negligible proportions.

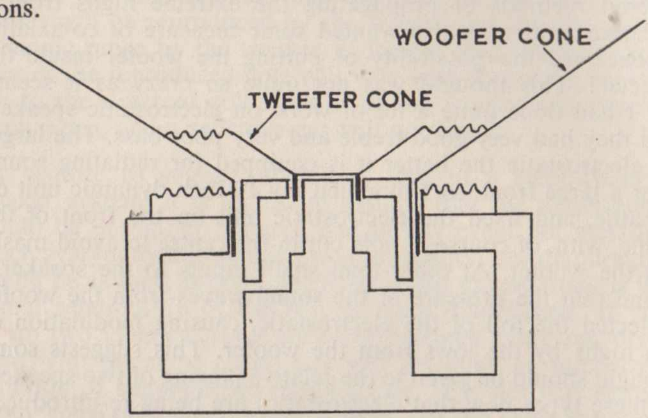


Fig. 17 The Olsen "Duo-cone" co-axial speaker. Movement of the woofer cone must necessarily have some effect on the tweeter cone's movement.

The more popular type of co-axial loudspeaker consists of a small horn loaded tweeter built into a normal woofer. The Jensen ingeniously uses a bored out centre pole of the woofer unit as the tweeter horn, the tweeter field magnet being located behind the woofer magnet. The voice-coil of the woofer must necessarily be of fairly large diameter to provide enough magnetic material in the centre-pole to avoid saturation. The University co-axial avoids this difficulty by using a special magnet for the woofer which completely surrounds the tweeter unit. This magnet is an annular casting of U-section, the tips of the "U" being in the same plane; the inner tip applied to its face plate constitutes the centre pole; the outer tip with its face plate represents the normal magnet outer pole. It should be realised that it is not of any consequence where the mass of magnet casting is located; the outer casing can be an unmagnetised casting or pressing and the magnet forms part of the centre-pole fitted with a separate tip machined to size (since high permeability magnets tend to be so hard that they can only be ground; they are also so brittle that they could not be turned even with a diamond tool).

Alternatively, the magnetic material can be cast in the form of a tubular ring the magnet circuit being completed by an iron or steel centre pole and round plates back and front. The former type of magnet is usually called a slug magnet, the latter a ring magnet; and there is no performance difference between the two types. Since the slug type magnet is virtually screened by the exterior pot, waste of flux through stray fields is less than with the ring type. The University magnet is a combination of the slug and ring types. The Jensen speaker assumes that the woofer cone forms part of the tweeter horn, since the curvature of the two sections is continuous; the University uses a separate horn for the tweeter, and this is recognised as a projection within the woofer cone. Some makes of speakers have this tweeter horn divided into cells to achieve dispersion of the high frequencies.

The two speakers just mentioned show evidence of careful design and manufacture, but it cannot be assumed that any speaker with a small trumpet sticking out in the middle is necessarily a good reproducer. My earlier suggestion that a speaker's performance can be assessed by looking at it does not apply to a co-axial of this type, since there are unseen factors that modify the performance. The woofer can be examined by the methods I have given but not the horn-loaded tweeter.

I have explained that a large cone can reproduce quite high frequencies by "break-up". This term has various usages, so I had better explain what I mean by it. By "break-up" I mean deformation of the cone at various applied frequencies. A cone is not an infinitely rigid piston, and to put it crudely, it bends in places when actuated by the voice-coil impulses. The cone can node radially and axially, and there is wave transmission along the material of the diaphragm itself. If a light powder, such as lycopodium is sprinkled on the cone (face up) and the speaker driven by an oscillator feeding an amplifier, patterns will be developed by the powder. These nodal patterns are controlled by the material of the cone, its size and shape, and the applied frequency. The patterns indicate that the cone is bending in varying degrees in different parts, and the three-dimensional shape of the diaphragm at a specific frequency is the "piston" moving the air. For linear response it is obvious that the efficiency of the piston must be constant, but if part of the energy from the voice-coil is dissipated in bending the cone that part is not available for pushing the air. At the same time, small parts of the cone are in motion when other parts are not, hence the propagation of higher frequencies than one would suppose possible. The actual movements of the whole diaphragm are very complex and no hard-and-fast rule can be laid down, but it can be assumed that, in general, it is quite a difficult matter to control the break-up at frequencies higher than about 1,500-2,000 cps.

If a tweeter is not used to get the extreme highs, very special care in design is essential for high fidelity results. If a tweeter is used, then there is no point in trying to get even medium highs from the large diaphragm. With a tweeter available, the woofer can have its cone size increased to avoid the need for very free suspension at very low frequencies when considerable power is fed into the speaker. This usually calls for a 15 inch cone in a high grade unit. Such a cone will give a very good output up to about 1,000 cps. but beyond this figure cone deformation—break up—is the determining factor and in a large cone this cannot easily be controlled. It was generally agreed in the days before "widespread high fidelity" (and you can put any construction you like on that phrase) that the optimum crossover frequency was in the region of 800-1,000 cps. This opinion was not based only on cone properties but took into account the impedance characteristics of the dividing network.

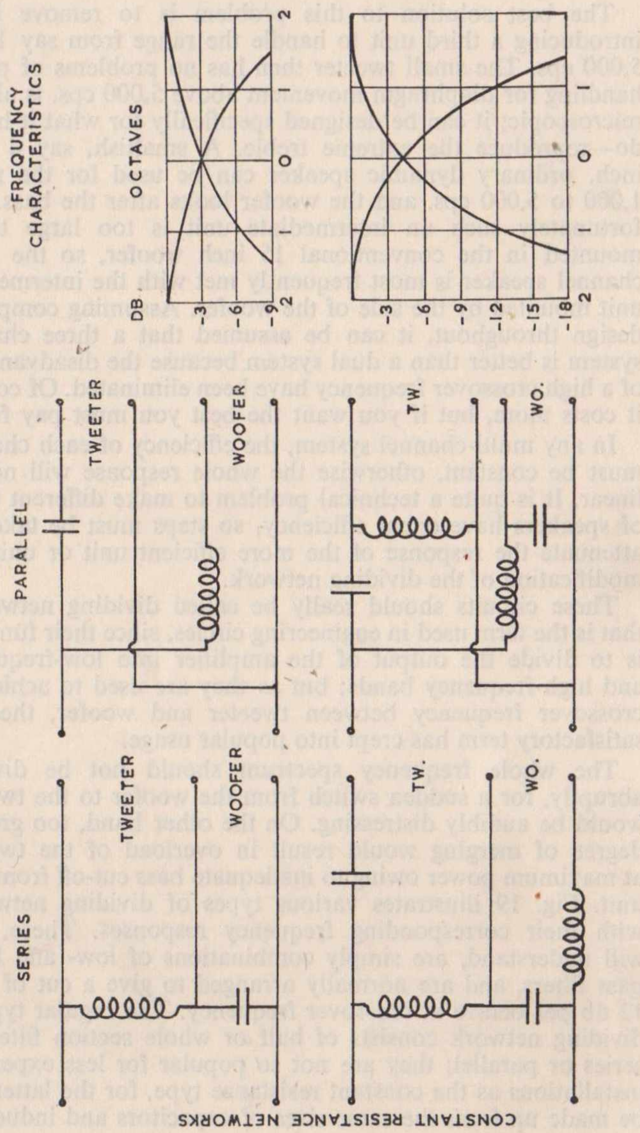
The two frequency bands of the individual speakers should overlap to avoid an abrupt change, and with a crossover frequency of 1,000 cps. the tweeter must handle the band from about 800 cps. to the upper limit. If the tweeter is small, its power handling capacity at even 1,000 cps. is quite limited, even when horn-loaded, so the power handling capacity of the woofer cannot be used because of the limitations of the tweeter. This undesirable state of affairs has led designers to put the crossover frequency much higher, even as high as 5,000 cps. I think it would be fair to say that some designers know quite well that this is not good practice, but are forced by the state of the market to put a limit on what the whole system will cost. If the market demands a dual concentric speaker, the designer can produce it, but it is no criticism of the designer to say that in the opinion of quite a number of qualified engineers the high crossover frequency is not the way to produce the best possible speaker.

The best solution to this problem is to remove it by introducing a third unit to handle the range from say 1,000-5,000 cps. The small tweeter then has no problems of power handling for diaphragm movement above 5,000 cps. is almost microscopic; it can be designed specifically for what it has to do—reproduce the extreme treble. A smallish, say 4 or 5 inch, ordinary dynamic speaker can be used for the range 1,000 to 5,000 cps. and the woofer looks after the bass. Unfortunately such an intermediate unit is too large to be mounted in the conventional 15 inch woofer, so the three channel speaker is most frequently met with the intermediate unit mounted by the side of the woofer. Assuming competent design throughout, it can be assumed that a three channel system is better than a dual system because the disadvantages of a high crossover frequency have been eliminated. Of course, it costs more, but if you want the best you must pay for it.

In any multi-channel system, the efficiency of each channel must be constant, otherwise the whole response will not be linear. It is quite a technical problem to make different types of speakers have equal efficiency, so steps must be taken to attenuate the response of the more efficient unit or units by modification of the dividing network.

These circuits should really be called dividing networks; that is the term used in engineering circles, since their function is to divide the output of the amplifier into low-frequency and high-frequency bands; but as they are used to achieve a crossover frequency between tweeter and woofer, the less satisfactory term has crept into popular usage.

The whole frequency spectrum should not be divided abruptly, for a sudden switch from the woofer to the tweeter would be audibly distressing. On the other hand, too great a degree of merging would result in overload of the tweeter at maximum power owing to inadequate bass cut-off from that unit. Fig. 19 illustrates various types of dividing networks with their corresponding frequency responses. These, you will understand, are simply combinations of low- and high-pass filters, and are normally arranged to give a cut of 6 or 12 db per octave at crossover frequency. The regular type of dividing network consists of half or whole section filters in series or parallel; they are not so popular for less expensive installations as the constant resistance type, for the latter can be made up from the same sizes of capacitors and inductors,


FREQUENCY CHARACTERISTICS
DB.
OCTAVES
-3
-6
-9
2
1
0
1
2
-3
-6
-9
-12
-15
-18
2
1
0
1
PARALLEL
TWEETER
WOOFER
TW.
WO.
SERIES
TWEETER
WOOFER
TW.
WO.
CONSTANT RESISTANCE NETWORKS

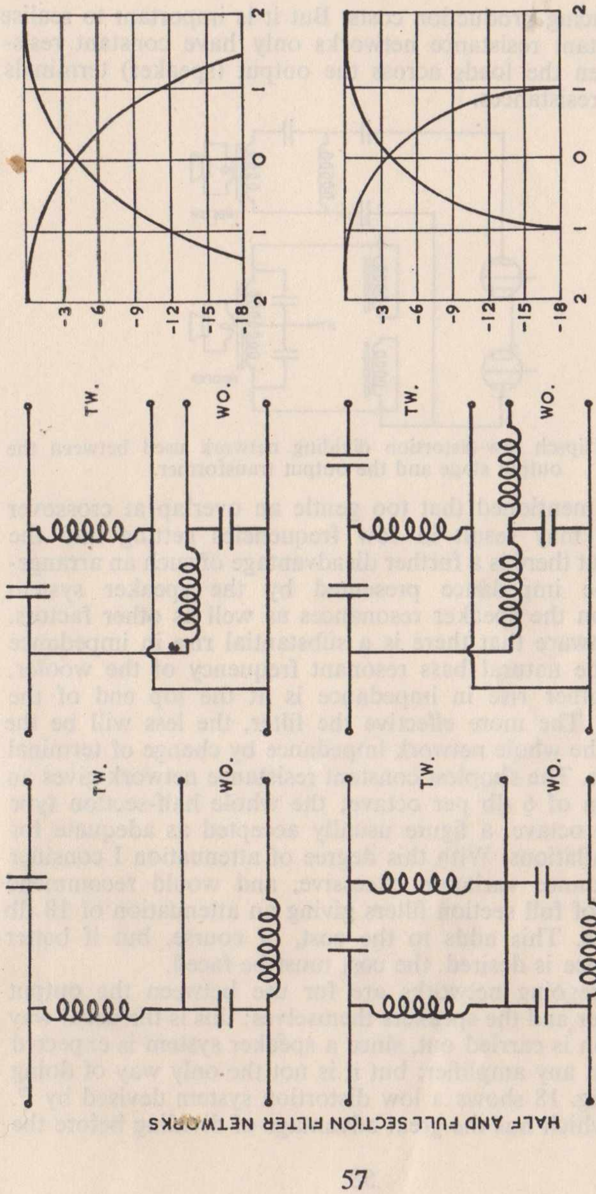


Fig. 19 The various types of dividing networks, having attenuations of 6, 12 and 18 dB. per octave. At crossover frequency the dip below linearity should be between 3 and 4 dB.

thus reducing production costs. But it is important to realise that constant resistance networks only have constant resistance when the loads across the output (speaker) terminals are pure resistances.

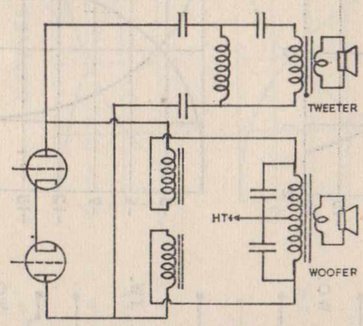


Fig. 18 Klipsch low-distortion dividing network used between the output stage and the output transformer.

I have mentioned that too gentle an overlap at crossover frequency may result in low frequencies getting into the tweeter, but there is a further disadvantage of such an arrangement. The impedance presented by the speaker system depends on the speaker resonances as well as other factors. You are aware that there is a substantial rise in impedance through the natural bass resonant frequency of the woofer, and the other rise in impedance is at the top end of the spectrum. The more effective the filter, the less will be the effect on the whole network impedance by change of terminal impedance. The simplest constant resistance network gives an attenuation of 6 db per octave; the whole half-section type 12 db per octave, a figure usually accepted as adequate for good installations. With this degree of attenuation I consider the impedance variation excessive, and would recommend adoption of full section filters giving an attenuation of 18 db per octave. This adds to the cost, of course, but if better performance is desired, the cost must be faced.

The foregoing networks are for use between the output transformer and the speakers themselves; this is the usual way the division is carried out, since a speaker system is expected to work on any amplifier; but it is not the only way of doing the job. Fig. 18 shows a low distortion system devised by P. Klipsch, which has the great advantage of dividing before the output transformer. One of the limiting factors in any audio installation is the output transformer, for it is quite an expensive matter to build an audio transformer which has low distortion and a wide frequency range; in the Klipsch circuit there are two transformers each handling a restricted frequency range, and the cost of the two can be appreciably less than that of one wide-range transformer of equivalent performance.

The frequency division can be carried out between stages in the amplifier itself. The driver of the output stage is used to feed a high-pass and a low-pass filter, each of which leads to its own output transformer and speaker. There is a great deal in favour of such an arrangement, for the cost of the filters is substantially reduced, since their terminals are high impedance instead of the very low impedance existing between the output transformer and speaker; the input to the two output stages can be controlled to a very substantial degree, both as to frequency and amplification, so very careful control can be applied to the respective speakers to balance them for acoustic output. Unfortunately these technical advantages are not likely to be received on the open market with any degree of enthusiasm, since the average high fidelity enthusiast prefers to select his amplifier for one reason and his speaker or speakers for some other reason. For myself I would always consider the power output stage of the amplifier as an inseparable part of the loudspeaker design.

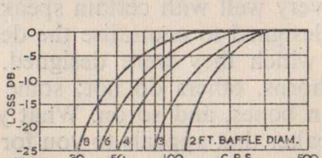


Fig. 20 Bass loss of finite circular flat baffles.

CHAPTER SEVEN

SPEAKER BAFFLES

Some aspects of audio engineering, such as amplifier design, can be done with precision. Speakers as I have indicated, are not such an exact science; room acoustics is partly exact, partly guesswork; the design of speaker mountings and enclosures can be undertaken on a strict mathematical basis.

But when you come to the practical usage of speakers in their enclosures in your listening room, there is a combination of unforeseen factors that makes the final decision a matter of quite exceptional difficulty. Here, more than anywhere else, the decision comes from the sort of reproduction you like, and it could be that what I like is not what you like. Yet throughout this book the aim is to guide you into achieving realism, that is, freedom from distortion.

The response of a speaker measured in free air is directly associated with its design. If it were a perfect speaker with a linear response it might not sound so good in a room with non-linear characteristics as another with a less perfect performance. The defects of the speaker might neutralise the defects of the room. But a perfect speaker's response is modified by the way it is mounted or housed, and no mounting is perfect; every type, flat baffle, cabinet or horn, has its own acoustic properties, for none is acoustically inert. When the speaker is used in your nonlinear listening room, four sets of data affect the final performance—the speaker in free air, the behaviour of the mounting or housing, the performance of the speaker when mounted, and the room acoustics. These four factors cannot be merged in any precise and scientific manner, but I can lay down for you a number of guiding principles to bring some sort of order out of apparent chaos.

There are as many different ways of mounting or housing speakers as there are designers who had a brainwave. Some enclosures work very well with certain speakers, because the enclosures were designed to neutralise the defects of the particular units for which they were designed. Some speakers work well with horns, others do not; some work better on flat baffles than in boxes, and so on. What you must not do is to choose a speaker that appeals to you for certain reasons, an enclosure that may make an entirely different appeal, and bring them together in a room without regard to any other consideration than where the combination looks best or is most convenient. Of course, you can do just that if you want to, but the odds against your getting realistic reproduction are pretty high. What, then, is the best way of setting about the problem?

At low frequencies, the sound waves from the front and back of a speaker diaphragm must be separated, because the sound from the front is 180° out-of-phase with respect to that from the back. At medium and high frequencies the wavelength is too short for cancellation to occur, but bass frequencies have wavelengths up to several feet. Without some form of baffle, reproduction of low frequencies is impossible, and a large baffle is needed for very low frequencies. Fig. 20 shows the bass attenuation resulting from the use of various sizes of finite circular baffles. You will notice that an 8-foot diameter baffle causes a loss of 5 db at 70 cps., so the problem is a very real one, for where can we place even an 8-foot baffle in a room without it being an eyesore? Note particularly a point not always realised: You cannot make good this loss by giving the amplifier a bass boost, for the loss is inherent in the mechanics of the sound waves themselves in relation to the baffle; if you try bass boosting you are, in effect, pouring your audio watts down the drain.

From time to time, audio enthusiasts have decided to put up with the inconvenience of a large baffle in the interests of high fidelity, but there are two reasons why they did not get it. Of course, they got the bass, but they got other things they did not bargain for. It is not difficult to appreciate that a large baffle is likely to be less rigid than a small one unless it is very thick and heavy.

Since all baffles are flexible to some degree, they will bend when activated by a speaker. Every baffle has its own natural resonant frequency, which you can prove for yourself by hitting it with your closed fist. But it also produces harmonic frequencies, for the thud you hear when hitting it does not sound like the note of a pure sine wave. The harmonics are the result of the baffle noding; second harmonic nodes occur by bending across a diagonal or a midway axis; third harmonics by bending across two axes dividing the baffle into three equal zones and so on. Harmonics up to the seventh can be perceived, and in this way a speaker may reproduce a sine wave input as a pure note, but the activated baffle becomes a producer of complex notes. The cure for this is to make the baffle as stiff as possible by strong bracing, particularly along the outer edges.

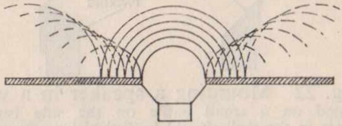


Fig. 21 Waves radiating from speaker are reflected from baffle, causing interference.

Another form of distortion occurs even with an infinitely rigid baffle. Fig. 21 shows a section of a speaker mounted on a flat baffle. The emergent sound waves have a hemispherical form and partially impinge on the front of the baffle. From this they are reflected in the way I have described in an earlier chapter. In the diagram the emergent waves are shown by solid lines, the reflected waves by dashed lines. These waves mutually interfere and cause uneven response.

Finally, the placement of the speaker on the baffle has a bearing on the response. The effective size of the baffle is the shortest distance from the centre of the front of the speaker, round the baffle, and on to the centre of the back. This distance is equal to the diameters of the circular baffles covered in the graph of Fig. 20. In a square baffle, those parts outside the circle can have no baffling effect. If the speaker is exactly in the centre of the baffle and a response curve is taken of the speaker so mounted, there will be found a characteristic narrow dip in the lower register. This can be avoided by placing the speaker off centre or making the baffle of irregular shape, but the effective baffle size is reduced, for still the shortest path from front-to-back determines the bass cut-off.

The only truly satisfactory flat baffle is the time-honoured one of mounting the speaker in a wall between two rooms. Such a baffle is virtually infinite, it is rigid, and with suitable draperies, is non-reflecting. The short tunnel in the wall should be flared at not less than a 45° angle outwards from the speaker when the speaker is mounted in the far side of the wall from the listening room. If the front of the speaker is flush with the wall, then the hole in the wall must be about twice as large as the speaker, the front sealed with a small thick baffle, and the empty space filled with Fibreglass or similar sound absorbing material. See Fig. 22.

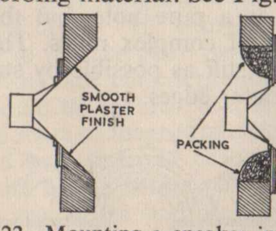


Fig. 22 Mounting a speaker in a wall

A—speaker is mounted on a small baffle on the side furthest away from the listening room; the hole in the wall should be finished off funnel-shaped with plaster. B—speaker set into the thickness of the wall; the edges of the hole should be rounded off with sound absorbing material, such as fibre-glass, to avoid tunnel effect.

From what has been said it will be fairly obvious that closet and cupboard doors do not form very good baffles, unless they are strong and thick. Moreover the space behind the door may cause undesirable reverberation effects, which will be considered more fully later.

The flat baffle can be made more compact by being folded into a box. As before, the effective baffle size is the distance from the front of the speaker, across the front, along a side and on to the back of the speaker. As far as pure baffling is concerned it is of no consequence whether the box type cabinet is large and shallow or small and deep, but there is a great difference in the acoustic properties of two such enclosures.

A box baffle partially encloses a column of air; this air will resonate at a frequency determined by its volume, as happens with organ pipes. For a given size of pipe the frequency of the emitted sound depends on whether the pipe is open or closed ("stopped"). An open back box baffle resembles a short wide pipe; a closed box baffle is like a stopped pipe. A shallow box baffle has very little air column resonance, and so needs no treatment except bracing to prevent cabinet resonance. A deep box baffle (of cubical shape, for example) has a pronounced air column resonance, and lining it with sound absorbing material has no effect on this resonance. The lining will help damp out cabinet resonance and standing waves caused by reflection from the interior sides of the box, but the contained air is still in the box.

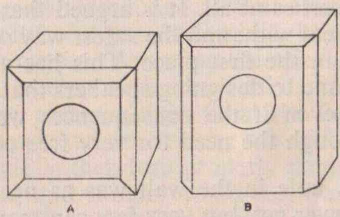


Fig. 23 Deeply chamfered cabinet edges give smoother bass response

Whether the back is closed or open, distortion caused by reflected waves (Fig. 21) will occur.

The distortion is revealed by an irregular response particularly at the lower end of the frequency spectrum. It has been proved that these irregularities can be smoothed out by chamfering the edges of the cabinet, as shown in Fig. 23. If the box is quite spherical there are no irregularities at all, as could be imagined from a consideration of the disposition of the reflected waves; but a spherical enclosure is an extremely inconvenient thing to make. Chamfered corners are, therefore, the best compromise, and if the floor-type enclosure of Fig. 23B is adopted, the speaker should be located so that it is not equidistant from the top and sides, nor in the centre between top and bottom. An intermediate position will be the best way of obtaining an asymmetrical baffle to avoid the dip mentioned previously.

This type of enclosure, since nothing can be done to eliminate air column resonance, should be rather large and shallow, which suggests the floor type; but if such an enclosure is placed against the wall, the air is trapped and the air resonance will be pronounced. With a definite closed back, new problems are encountered, and these will be discussed later. For the moment I just point out that cabinets of the Fig. 23B type should be placed across the corner of the room and the top should not be of triangular form to close the gap between cabinet and walls.

I am going to describe the Hartley "Boffle", not for any commercial reasons but because it is a unique design of some general interest. There is quite a strong feeling among many acoustic engineers that the reproducing system should have no resonant properties at all. It is argued that resonances are tricky things to deal with, and the safest way out is to prevent them happening in the first place. This has always been my belief, which led me to designing speakers that had no audible bass resonance, even if the consequences were a reduction in sensitivity through the need for very free cone suspension.

I knew that a hole in the wall was as near a perfect flat baffle as we can ever get, but very few can manage this happy state of affairs. Putting the speaker in a closed lined box simulates an infinite baffle, but the enclosed air resonates. If the whole of the space inside is filled with sound absorbing material to eliminate the air resonance, the freedom of suspension of the speaker will be impaired owing to the stiffness of the air compliance.

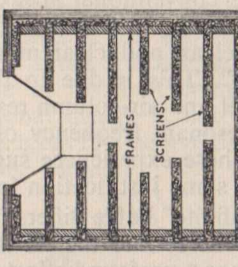


Fig. 24 Cross-section of Hartley "Boffle" non-resonant enclosure.

The inert non-resonant device I finally produced I called a "Boffle" an abbreviation of box-baffle. A cross-section is given in Fig. 24. It is quite unlike any other form of enclosure, for it is an acoustic filter. In electrical filters we have inductance, capacity and resistance; in mechanical filters (and acoustics is a form of mechanics) the elements are masses, springs and friction. In the "Boffle" the sound waves from the back of the speaker hit the second screen (the first is merely an anti-reflection device); if it were not perforated the screen would be unduly stressed, so part of the pressure passes through to the third screen, and so on. The diagram shows two graded filter stages, but except in deep cabinets, one filter with up to 8 screens is all that is necessary. The semi-porous screens of carpet felt act as masses, their slight elasticity and the air pockets between the screens as springs and their acoustical semitransparency as friction. The back must not be rigidly closed, and all that emerges from the rear is a very low-pitched "grumble" which has no harmful effect on the speaker output. Wrapping the felt around the wooden frames of the screens is an essential feature of the device. The screens are rather a tight fit in the box and the felt is slightly compressed as the screens are slid into place. Every part of each side is therefore properly damped against nodes and resonances, and thinner wood can be used for the box than is necessary for any other form of enclosure.

The "Boffle" has been described for home constructors with interesting consequences. Designed for my own speakers, I did not suppose it would be much favoured for housing speakers that normally require a reflex enclosure for neutralising the bass resonance of the speaker. It turns out, however, that owners of more conventional speakers than mine have made it up and like it very much indeed. They say that the "Boffle" gives very clean and clearcut reproduction having noticeable "presence". This is due to the almost complete suppression of cabinet and air-column resonances. With these removed, the bass resonant frequency of the speaker is not unduly noticeable. These experiences suggest that the "non-resonant school" has some justification for thinking that way.

The closed box "infinite" baffle differs from the hole-in-wall infinite baffle, for in the former the air is trapped and in the latter it is free. This has a profound effect on the reproduction. The closed box is a resonator, frequently called a type of Helmholtz resonator, although the distinguished physicist did not invent it, but he did analyse its properties. The air within the box resonates at a frequency determined by the volume, and the sharpness of the resonant peak depends on the reflective power of the internal surfaces of the box. Moreover, about 25% of the third harmonic of the fundamental resonant frequency is generated when the air is activated by the speaker diaphragm. In addition, for reasons given in an earlier part of the book, reflections from the sides of the box create standing waves. All three phenomena cause distortion and must be eliminated as far as possible. The sharpness of resonance is flattened by lining the box with sound absorbing material; if there is sufficient thickness of lining the third harmonic will be suppressed, as will standing waves at all but low frequencies. The basic physical properties of a closed box are, therefore, that it must be very strongly constructed (since the bass is not wholly absorbed) and lined with a substantial thickness of acoustically absorbent material.

Under these conditions it will be found that interaction of the bass resonance of the speaker and the air resonance of the closed space results in an effective raising of the bass resonant frequency of the speaker. The larger the speaker cone the larger must be the box, and as a rough working guide it can be taken that an 8 inch speaker requires 5 cubic feet of cabinet volume, a 12 inch 14 cubic feet and a 15 inch 20 cubic feet. The normal speaker bass resonance should be as low as possible, which implies free suspension, and, contrary to what might be expected, the larger the cone the "freer" must be the suspension. A large cone moves more air than a small one, and the air trapped behind the cone is what causes the rise in frequency, hence, the need for larger boxes with larger speakers. If, therefore, you wish to use a small closed box, a small speaker must be used with it to avoid an undue rise in resonant frequency, but the small speaker of conventional design is not very effective at low frequencies. This dilemma can be avoided by providing some form of air leak in the cabinet.

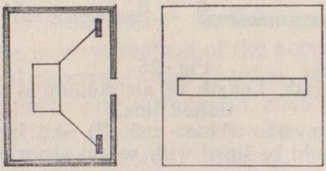


Fig. 25 Section and front of R-J box baffle

Robbins and Joseph have devised an enclosure which is stated to be a modified Helmholtz resonator, and is shown simplified in Fig. 25. The box is not substantially larger than the speaker itself, but a form of air duct is provided by the space between the small baffle carrying the speaker and the front of the box. It is claimed that this reduces the sharpness of the resonance of the air within the box, producing a two-peak curve comparable to that of a reflex housing. If the speaker is intended to reproduce the whole frequency range, then, as pointed out previously, the slot should be vertical to secure horizontal dispersion of the high frequencies. Some models of the enclosure have a slot at the bottom, suggesting that no attempt is made to reproduce the highs, which makes the unit simply a woofer. Obviously the performance of the whole assembly must depend on the size of the box, the size and bass resonant frequency of the speaker, the thickness of the air duct and the size of the slot. A comprehensive mathematical analysis of these critical dimensions has not been published.

A different type of vented enclosure is the acoustic labyrinth, shown in section in Fig. 26A. This is virtually an air column loading the back of the speaker diaphragm, in contrast to a horn which loads the front. The operation differs from

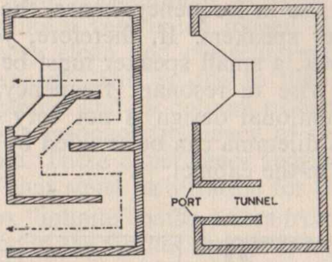


Fig. 26
A—acoustic labyrinth. Length of air column is taken along the dotted line.
B—acoustic phase inverter ("bass reflex"). All interior surfaces of both types should be lined with sound absorbing material.

that of the horn, for whereas a horn gives correct loading over the whole frequency range (if big enough) the labyrinth can only act as an efficient load at its resonant frequency. The dimensions should be such that the effective length of the air column (taken along the centre line) is one quarter of the wavelength of the bass resonant frequency of the speaker. The wavelength, of course, equals the speed of sound (1,129 feet per second) divided by the frequency in cycles per second. Under these conditions the air column resonance will neutralise the bass resonance of the speaker. The whole of the interior surfaces must be covered with sound absorbing material to prevent reflections as far as possible. This will cause considerable attenuation of the high frequencies from the rear of the speaker. A tweeter may be necessary to maintain overall balance.

The most popular vented enclosure is that usually called the bass reflex, due originally to A. L. Thuras. Since the patent expired the design has appeared in many forms, but in some cases there is evidence that the basic principles have not been clearly understood, with unsatisfactory results in the quality of reproduction. When properly designed and correctly applied, the acoustic phase inverter (a title which explains its function exactly) improves the bass response and increases the power handling capacity of the speaker at low frequencies. With this goes a decided flattening of the bass resonant peak in the impedance curve. These advantages result from reduction of the travel of the voice coil at resonant frequency by accurate loading of the diaphragm at that frequency; in other words, the resonant frequency of the air in the enclosure must be the same as that of the speaker. It follows that the enclosure must be carefully tuned to the frequency of the speaker resonance.

I should explain that what follows refers to acoustic phase inverters. This type of enclosure must be accurately matched to the speaker. Other enclosures which are not so matched resemble the genuine bass reflex but their effect is different. Some notes will be added later on this type.

Fig. 26B shows a cross-section of the acoustic phase inverter with the essential elements—the speaker, the enclosed air, the tunnel and the port. The volume of enclosed air equals the total internal volume of the cabinet less the volume of the speaker unit and any internal bracing, but not the sound-absorbing lining since this latter is virtually part of the air space. For a given volume of air the frequency of resonance in the port is modified by the size of the speaker diaphragm and the length of the tunnel. The larger the speaker the greater must be the volume of air; the longer the tunnel the smaller the volume. Bass reflex enclosures can be found with and without tunnels; the purpose of the tunnel is to reduce the size of the cabinet for a given resonant frequency. As a result of this you can assume that any enclosure offered to you of compact size, housing a large speaker and having no tunnel, will not perform as an acoustic phase inverter unless the normal bass resonant frequency of the speaker is so high as to make it unsuitable for high grade reproduction.

Herein is the fallacy of buying a speaker which you fancy and fitting it into a reflex enclosure which also appeals to you. The two may not be compatible. The information required by an engineer to enable him to design an acoustic phase inverter for any particular speakers includes the equivalent piston diameter of the speaker cone, the bass resonant frequency of the speaker and its total volume. There is an optimum length of tunnel for any given enclosure volume, neither too long nor too short. The end of the tunnel should not be nearer the back of the cabinet than the radius of the speaker diaphragm. The area of the port should equal the area of the speaker opening (or more accurately the area of a circle whose diameter equals the diameter of the equivalent piston). If there are errors in design the system can be tuned by altering the port area, but doing so conflicts with the "equal area" condition.

For the speaker of your choice you can assess the merits of the enclosure by visual inspection and electrical measurement. The cabinet should be strongly made and free from drumming when hit with the fist. The interior should be well lined with sound absorbing material to prevent the formation of standing waves. The port area should equal the area of the speaker opening. The rear end of the tunnel should not be nearer the back of the cabinet than half the diameter of the speaker opening. These points checked, the speaker is then mounted in the cabinet and its impedance measured at low frequencies by one of the methods given earlier.

If you had previously taken a curve of the unenclosed and unmounted speaker from 1,000 down to about 20 cps. you would get something like the solid curve in Fig. 27, with the characteristic single peak produced by the bass resonance of the speaker. Below this peak the response falls off rapidly.

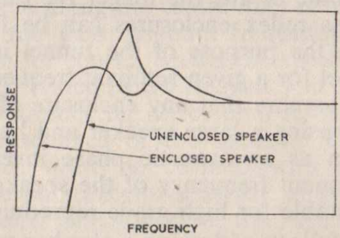


Fig. 27 An acoustic phase inverter properly calculated and designed for a given speaker will convert the single resonant peak of the unmounted speaker into a two-peak response of greater linearity, as shown above.

Now with the speaker properly mounted in the enclosure, take another curve. This should look like the dashed curve in Fig. 27, with two peaks, one on either side of the original peak. It is obvious that the response is much closer to linearity and the bass cut-off is lower. This is the advantage of the matched acoustic phase inverter and mismatching will not give the desired results.

I can almost hear you say "Why should I go to all this trouble?" There is no "must" about it. You are quite free to do the job properly or incorrectly, but if you want the undoubted merits of this type of enclosure to improve your audio reproduction, then you cannot expect to get them by hit and miss methods, otherwise there would be no need for engineers at all. If you do not get the proper double hump curve then the dimensions of the enclosure must be altered until you do; but it is just possible that you would get the desired results by placing the cabinet in another part of the room. The room acoustics influence the impedance of the speaker, as I have already explained.

Unmatched vented baffles can be home-constructed or bought ready made, and are an easy way of dodging the technical requirements of the true acoustic phase inverter. They do not work with the precision or efficiency of the genuine article, but they are better than a casually constructed box baffle. As before, they must be strongly constructed and properly lined. No tunnel is used as this introduces difficulties in adjustment. The port area should be greater than the speaker opening, so that tuning can be carried out over a fairly wide band.

The speaker should have as low a bass resonance as possible and the volume of air within the cabinet should not be less than 6 cubic feet. With alteration of the port area, the air resonant frequency is changed, in general, raised as the area is increased. This affects the impedance curve of the speaker and some simulation of the characteristics of the true phase inverter is possible, but the transient reproduction will not be so good as the condition of optimum loading is never reached.

If a small cabinet is insisted on, then the speaker should have the more conventional value of the bass resonance, but if adjustment of the port to give a reasonable flat impedance curve involves raising the resonance of the system to something in the order of 80 to 100 cps. is necessary, the reproduction will not be satisfactory as a whole, even allowing for the loss of bass.

In short, the properly designed vented baffle can neutralise some of the defects of the ordinary sort of speaker, but the best results are only obtained when the enclosure is properly designed to do the job. It is curious that keen audio fans often give their speakers a very raw deal.

CHAPTER EIGHT

STRAIGHT, FOLDED AND CORNER HORNS

A horn is fitted to a loudspeaker driver unit simply and solely to increase its electro-acoustic efficiency. A properly designed horn increases the acoustic loading on the diaphragm and this is bound to improve the efficiency since the diaphragm has something to work against. From this follows the obvious conclusion that a horn-loaded speaker requires less input than one using a flat baffle for a given sound output, and for a given size of diaphragm the horn-loaded speaker calls for less movement of the suspended system. From what you have learned in this book you can see, therefore, that the disadvantage of a small diaphragm for reproducing low frequencies, the large amount of free movement required, can be overcome to some extent, while retaining the advantages of the small cone for good high note response. Since the driver unit is subjected to smaller stresses, it would seem that fitting a horn instead of a flat baffle or box type enclosure is a great step forward. This supposition is correct. A properly designed horn-loaded speaker will give a wider and more linear response than any other type of loading, and when perfectly designed and without regard to "contingent liabilities" does not require the use of a multi-channel system. One unit will do the job. Yet almost every horn type speaker system you see has a tweeter; am I therefore talking nonsense? I mentioned contingent liabilities, and the inate cussedness of all loudspeaker problems is well to the fore in designing loudspeaker horns.

In this article I cannot possibly even attempt to classify the multitude of designs on the market. The good ones are the result of technical know-how and intensive development work. The bad ones are non-scientific copies of good designs but without the knowledge necessary for modifying basically good designs. Some have resulted from the efforts of writers who profess to provide hi-fi for a few pounds. But it so happens that designing a good horn is not all that easy and making it can be even more difficult.

Here I shall explain the fundamental rules of the game, so that you can make your choice in an intelligent manner.

But whereas there may be two schools of thought in speaker design there can only be one in horn design, for the matter is simple enough—does the horn enclosure add distortion to the speaker unit's performance? If it does then it is a bad enclosure and that is all there is to it.

The worst snag in adopting the horn as a speaker loading device is the size required for fidelity of reproduction. The diameter of the mouth of the horn, for perfection, should equal the wavelength of the lowest frequency it is desired to reproduce. The wavelength of a 50-cycle note is 22½ feet! Moreover, the rate of expansion from the throat (the narrow end) to the mouth, called the flaring constant, must conform to certain laws, so the length of our perfect horn for no cut-off at 50 cps. would be about 70 feet. In this imperfect world we can afford to make some compromise, but you can take it that a straight horn of proper design to reproduce down to 50 cps. calls for a length of about 22 feet and a flare circumference of 24 feet, and that is not a thing you can get into an ordinary room. Not only is the mouth as large as the sort of flat baffle you ought to have, but where are you going to put those 22 feet of length? As you can fold a baffle, so you can fold a horn, but with this added complication—that the highs do not like being pushed round sharp corners or along rough surfaces, and the lows, as in box baffles, set up vibration in the various parts of the assembly. Whereas the folded and curved horns of the brass section of the orchestra are resonant, to give the instrument its peculiar timbre, the horn of the reproducer must be inert and unable to impart colouration.

Probably the first superbly designed and engineered folded-horn speaker was the celebrated Western Electric 555. The speaker unit itself was made with very close tolerances to avoid loss of useful flux in the gap. The voice coil was wound with aluminium ribbon on edge, so that gap space was not wasted by a comparatively thick and rigid former, and the small aluminium diaphragm was properly ribbed to ensure stiffness (for it is important to a horn speaker that diaphragm breakup should not occur). This specialised unit then fed into a long folded horn with the right flare constant, made of smoothly finished, non-resonant material (at least down to the lower middle frequencies!), which terminated in a large rectangular mouth. The result was a fine speaker, but it was so big it could only be used in cinemas. Since that time, we engineers have not increased our basic knowledge of horn design; we have made no discoveries that enable us to do things that could not be done 30 years ago. The mechanics of horns are perfectly straightforward and we cannot do the impossible "even if it takes a little longer". Our efforts have been directed towards producing speakers that fit conveniently into an ordinary sitting room, while retaining as many of the characteristics of the perfect horn as possible. In other words, compromises have had to be made, and some compromises are very good and others are not.

As no folded horn can be as good as a perfectly designed straight horn, it is necessary to determine the characteristics of the straight form to have some standard of reference. There are three main types: conical, exponential, and hyperbolic exponential. The only merit of the first is that it can be constructed out of flat sheets of material, and in case you wonder how a cone can be made out of flat material I should explain that what really matters is that the area of cross-section has to expand in a certain way. To all intents and purposes a square horn of pyramidal form is just as satisfactory as a truly conical one. By a conical horn, I mean, therefore, one whose sides are a straight line, and by analogy I call an exponential horn one whose sides follow an exponential curve, whether the area of cross section is a square or a circle. The name hyperbolic exponential is usually shortened to "Hypex". Fig. 28 gives cross sections of the three types.

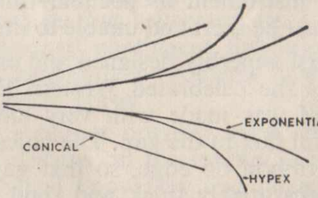


Fig. 28 The three main types of straight horns—
A: conical. B: exponential. C: hypex.

The conical horn is easy to design and easy to build. All that matters is that the narrow end should more or less fit the driver unit and that the length and mouth dimensions should be great enough to handle the lowest bass frequency it is desired to reproduce. The serious drawback of the conical horn is that its cut-off characteristic is not good.

In any high fidelity system it is desirable that the wide frequency response should terminate with sharp cut-offs at bass and treble. A linear frequency response from 50 to 12,000 cps. with very sharp cut-offs at each end will give truer reproduction than one linear from 60 to 11,000 with gradual roll-offs even if there is appreciable response at 40 and 15,000 cps. You may not believe this, but it is so. Now if you refer to Fig. 29 you will see that the conical horn has a roll-off whereas the exponential and Hypex horns have a cut-off, and the Hypex has the sharper.

It is not difficult to understand why this should be so. As I have explained, a sound wave progresses through the air by setting up zones of compression followed by zones of rarefication. The distance between successive zones of compression is the wavelength of the sound wave of that particular frequency. Now imagine such a sound wave passing through the horn. Obviously the horn must be as long as one wavelength otherwise part of the wave will be inside the horn and the rest outside and the only part to load the diaphragm with "horn efficiency" is the part inside. That is why inadequate length and mouth size give a bass cut-off.

In free air the speaker diaphragm produces a hemispherical propagation in front and when a conical horn is used this whole hemisphere has been collected into a cone but the general distribution throughout the horn is unaltered. When

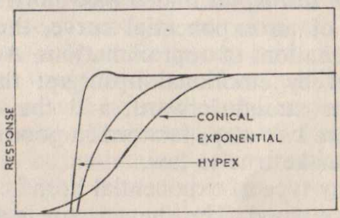


Fig. 29 Cut-off characteristics of the three types of horns of Fig. 28.

the wavelength is a substantial part of the horn length there will be an instant when a zone of compression is inside the horn and a zone of rarefication is at the mouth of the horn. Nature abhors a vacuum so air at normal pressure around the circumference of the mouth rushes in and hinders the progress of the next pocket of compressed air. If there are several "cycles" inside the horn this does not matter; but at low frequencies the effect is very pronounced, and the interference pattern comes out like the curve in Fig. 29.

To take the Hypex as a contrast, the sound wave emerges from the mouth and not being confined by the straight line trend of the conical type progresses in a hemispherical manner. The air inside is protected by the shape of the horn mouth and by a hemispherical barrier of compressed air beyond the mouth. The description I have given is admittedly crude but it does account for the cut-off characteristic of the Hypex horn. In the Hypex, and to a lesser degree in the exponential, the cut-off is "pure" and determined solely by the horn dimensions.

Having vowed to keep higher mathematics out of this manual I cannot give you the design data for these horns. Being exponential curves they involve mathematical exponentials which are reckoned highbrow; but the omission of this data is not a matter of great importance. Fig. 28 shows that the types cannot be confused, for the exponential increases quite gradually in a curved sort of way whereas the Hypex flares out quite suddenly near the mouth. May I add a note about other wonder-working curves announced from time to time? We get paraboloids and catenoids all heralded as new achievements. Do not believe it. These other "curves" are so near exponential that it could not matter less, and except for moulded or cast horns, no folded bass horn is other than an approximation of an exponential curve, these fancy curves are just approximations of approximations. Acoustic engineers are not swayed by emotional upsurges; the laws of horn design are quite straightforward, and the exponential and Hypex curves are two steps forward in good design. But they are difficult to make true to law.

No part of any type of exponential horn is flat, so it cannot be made of thick wood. The shaped panels are usually made of laminated or reconstructed wood and should be strongly braced with frames at fairly short intervals; the intervening areas should be covered with sound and vibration absorbing material or cement. This must be applied outside the horn, for the inner surfaces should be as smooth as possible to avoid air friction. The whole horn could be made of reinforced concrete with a smooth cement finish inside, and super-enthusiastic high fidelity fans have made such concrete

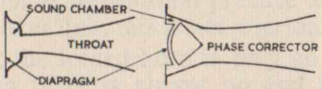


Fig. 30 Two designs of horn throat sound chambers—without and with phase correcting plugs.

monsters with most impressive results. Of course, the horn has to be built outside the house, so it is not very convenient for multi-storey apartments. But it does show what has to be done to carry the horn to its logical conclusion.

It would seem a simple enough matter to match a horn to any loudspeaker by making the throat (the narrow end) the same size as the speaker diaphragm but this does not give the highest efficiency. Better acoustic loading is obtained by having the throat smaller than the diaphragm and including a sound chamber, as in Fig. 30A. At high frequencies this scheme does not work very well because the distance between the various parts of the diaphragm and the centre of the throat can differ by several wavelengths, causing phase distortion. It is usually corrected by making the diaphragm concave and inserting a convex plug in the horn throat, as shown in Fig. 30B. This phase-correcting plug as it is usually called, should be a feature in any well-designed horn-loaded tweeter.

The throat itself causes second harmonic distortion, varying directly with acoustic watts per unit area of throat and with the ratio between emitted frequency and cut-off frequency. For a given power input to the speaker, it follows that second harmonic distortion will be smaller the larger the throat and the smaller the emitted frequency/cut-off frequency fraction. As the bass must be maintained, this fraction is kept small by removing the highs from the large throat speaker. This suits the general design very well since a large throat calls for a large diaphragm and a large diaphragm (subject to the special cases mentioned previously) is not efficient for high frequencies. Then, since the first section of the horn has to be removed to provide the large throat, the removed part becomes the horn of the tweeter, so we quite logically arrive at the conclusion that, as far as horn speakers are concerned the tweeter-woofer combination is best. Whether my theory that baffle-loaded speakers are best as single-channel systems

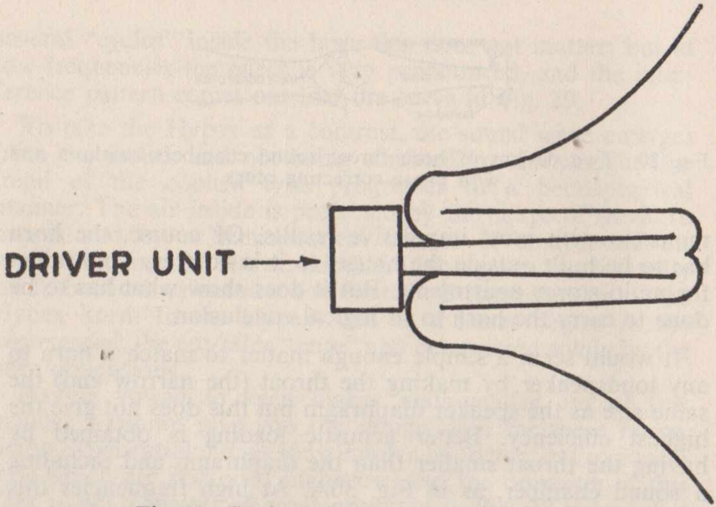


Fig. 31 Section of concentric folded horn.

is right or not, I cannot be accused of undue partisanship if I say that multi-channel systems are best with horn speakers.

In theory, as I have already pointed out, there should be no loss by folding a properly designed exponential horn. As far as the high frequencies are concerned there is very little loss due to reflections and interference in the concentric folded type shown in Fig. 31. A horn of this design is usually made up from metal spinnings, although it can be moulded from non-metallic materials. The size required for adequate reproduction of low frequencies makes this type of horn very costly for wide range reproduction, but it is an efficient horn for the frequency range 200 to 8,000 cps.

In practice, a folded horn is usually made up as an assembly of flat wooden panels which can only be an approximation to the true exponential flare, so losses are inevitable (and "losses" includes distortion). As both sides of each panel usually form part of the horn acoustical lining and reinforcing battens cannot be used, so there must be some reverberation and cabinet resonance. To reduce this as far as possible the material used must be thick and rigid. The rate of flare does not conform to any law since the horn consists of a series of truncated pyramids, with the consequent disadvantages mentioned earlier. The shape, too, is bad for the transmission of high frequencies, but as it is normal practice to use a separate high frequency speaker unit, this is not a serious consideration.

Since, therefore, the folded horn is only an approximation to the ideal design there is almost no end to the ways in which this approach to perfection can be achieved. Reputable manufacturers of speaker units have been forced to produce horn designs which are suitable for their products, and it can be supposed that some research has been carried out to evolve a good design. Other manufacturers of cabinets are equally interested in selling their wares, but in all this activity one thing can be emphasised—since no speaker has a perfectly linear response and since no cabinet imposes a constant load at all frequencies—the cabinet must be designed for the speaker selected.

Despite all this, the curious fact remains that some combinations of units and horns that were not specifically designed for each other do sound extremely good, and there can be only one reason for this—luck. Good luck is not to be despised in the hunt for perfection. It is quite possible for a defective speaker to be housed in a defective cabinet so that the defects more or less cancel out, and it does not matter if such results came about by blind chance. What really matters is that the results are there. I have pointed out that speakers cannot be designed by mathematicians alone, nor, for that matter, can cabinets. Marrying the two is best done by practical experiment.

There is some dispute as to who first thought of the corner horn by which I mean a device which uses the adjacent walls as part of the horn system. Sandeman refers to a sound generator working into the literal corner of a room formed by the meeting of two walls and the ceiling. There is a later device, the small Ephraim corner horn, extended by the same three plane surfaces. But the first high fidelity job I met was the Voigt in the early thirties. A section is given in Fig. 32, and the section line is from the middle of the front of the housing (it is not an enclosure) into the actual corner of the room. The loading on the front of the diaphragm is not effective below about 50-60 cps., so a tuned air column is used as a supplementary resonator for lower frequencies, driven by the back of the diaphragm.

The whole device, while it works very well, is rather ugly and clumsy and has been superceded, at least in the U.S.

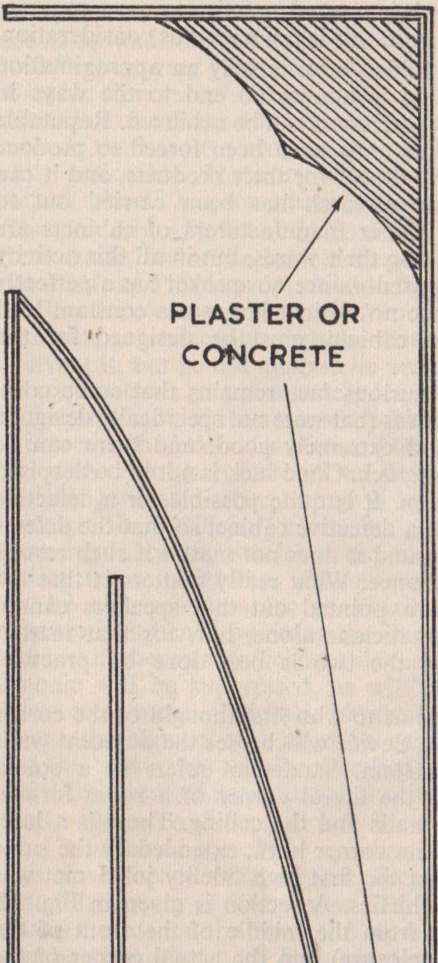
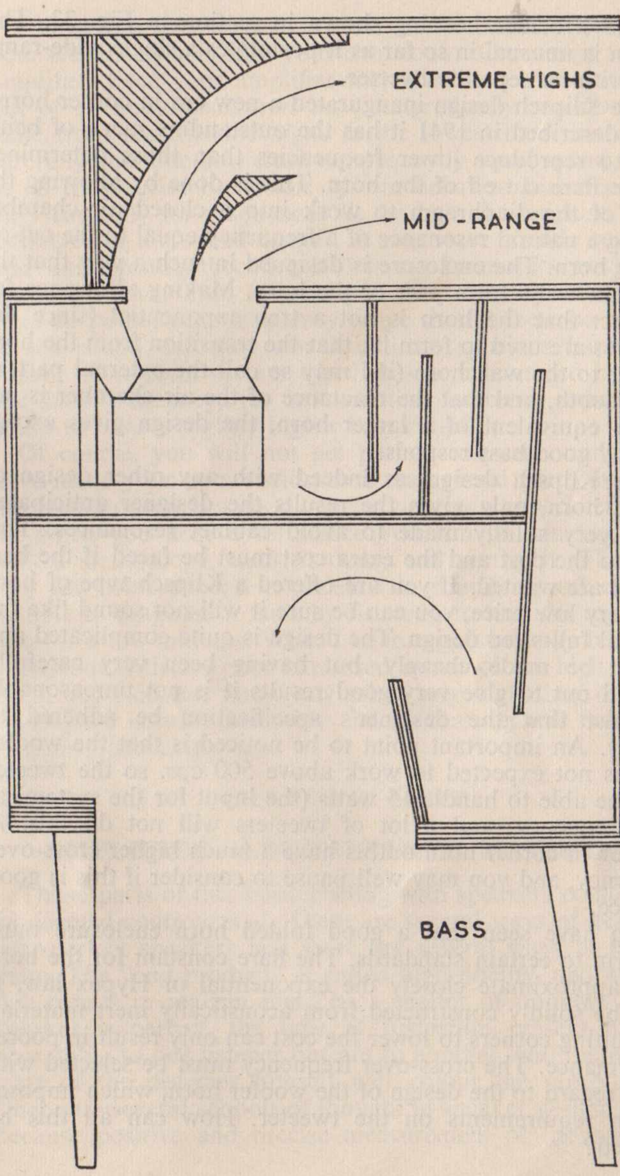


Fig. 32 The original Voigt corner horn. A tuned air column driven by the rear of the speaker cone is used as a supplementary resonator for lower frequencies, since the loading on the front of the cone is not effective below about 60 c.p.s.

Fig. 33 Brociner cabinet for Voigt or Lowther unit, showing curved front horn for highs and mid-range and long folded rear horn for the bass.


EXTREME HIGHS
MID-RANGE
BASS

by the Brociner housing shown in section in Fig. 33. This design is unusual in so far as it postulates a single wide-range unit with no separate tweeter.

The Klipsch design inaugurated a new era in corner horns. First described in 1941 it has the outstanding merit of being able to reproduce lower frequencies than those determined by the flare cut-off of the horn. This is done by allowing the back of the diaphragm to work into a closed air chamber having a natural resonance of a frequency equal to the cut-off of the horn. The enclosure is designed in such a way that the adjacent walls form part of the horn. Making allowance for the fact that the horn is not a true exponential (since flat surfaces are used to form it), that the transition from the horn proper to the wall horn (if I may so call the external part) is not smooth, and that the reactance of the air chamber is not a true equivalent of a larger horn, the design gives exceptionally good bass response.

The Klipsch design, as indeed with any other design of folded horn, only gives the results the designer anticipated when very solidly made to avoid cabinet resonances. This adds to the cost and the extra cost must be faced if the best results are wanted. If you are offered a Klipsch type of horn at a very low price, you can be sure it will not sound like the original full-sized design. The design is quite complicated and cannot be made cheaply, but having been very carefully worked out to give very good results it is not unreasonable to insist that the designer's specification be adhered to exactly. An important point to be noticed is that the woofer horn is not expected to work above 500 cps. so the tweeter must be able to handle 15 watts (the input for the system) at that frequency, and a lot of tweeters will not do this. A number of corner horn outfits have a much higher cross-over frequency, and you may well pause to consider if this is good practice.

You have seen that a good folded horn enclosure must conform to certain standards. The flare constant for the horn must approximate closely the exponential or Hypex law. It must be solidly constructed from acoustically inert material, and cutting corners to lower the cost can only result in poorer performance. The cross-over frequency must be selected with a due regard to the design of the woofer horn, which imposes certain requirements on the tweeter. How can all this be checked?

I think the only answer is that you test what is offered to you. Get the system into your own room. Connect it to your amplifier. Feed your amplifier with the linear sine-wave output of an audio-oscillator. As you gradually run down the scale from the extreme highs to the lowest bass, listen very carefully and note how close to apparently equal sound output at all frequencies the whole system behaves. Listen particularly carefully for resonances in the bass. Listen also to the character of the sound output. A sine wave sounds very dull and uninteresting because it has no harmonics to give it musical colour. That is what you want from your speaker, so at no point in the frequency range should there be an edge to the sound, for that would indicate spurious harmonics. Above all reject a system which has a pronounced boom at one bass frequency, for in time that becomes unbearable; better to have a slightly higher bass cut-off.

Of course, you will not get perfect response, and if you have made your own enclosure it may sound pretty bad; but the oscillator test is a good one for your own experimentation, for when you hear a resonance you can go hunting for it with a stethoscope, track it down and rectify it. There are very few with enough experience in sound reproduction to be able to diagnose a fault by listening to musical reproduction for a few minutes. What I have suggested may be highly unpopular with the poor harrassed owner of the high fidelity shop, but I do not know any other way of finding out how a complete speaker assembly and its housing will behave in your own listening room.

CHAPTER NINE

AMPLIFIER—LOUDSPEAKER MATCHING

Those parts of this book dealing with speakers could justly be deemed controversial. There are several ways of designing near-perfect speakers and there are several ways of judging them. The "end product" is called high fidelity, and whether this results in realism must be a matter of opinion. As no speaker is perfect, the type of distortion present may be acceptable to one listener but not to another, and the musical taste of the designer himself will colour the reproduction. There are several schools of thought in speaker design, simply because positive and precise measurement of the sound of music is not possible. Moreover, however conscientiously I strove to give an impartial account of the important features of speaker design, I suppose inevitably I should feel that my way was the best, otherwise I would not have done it that way.

When it comes to considering the power required to drive the speaker, there can be (or perhaps, should be) no argument at all. There should be no conflicting schools of thought. Our requirements can be stated precisely—there must be no distortion in the amplifier output within the audible limits, and this can be achieved at reasonable cost. Further, the amplifier performance can be measured with precision so an absolute and objective standard of performance can not only be postulated but achieved and proved. In addition, I, as a writer, have no financial or business interests in any amplifier extant or projected. All I want is undistorted power for the speakers of my choice, and I assume that that is what you want too.

I had hoped to give the answer very briefly but I needed an answer to the basic question—what is the best output stage? As a result of much experience I know what I prefer, but when I recalled that in this presumably exact field of amplifier design there is a strong body of opinion in favour of triode output stages and another equally insistent on tetrodes or pentodes, something more was needed than just another resumé of the various types of output stages. And the high fidelity enthusiast must have heard of or tried dozens of different circuits, each of which was supposed to be the last thing in perfection. Writing about amplifiers is the easiest form of technical journalism; the demand is insatiable, for everybody wants something better, and most amateurs can build an amplifier if they cannot build a speaker.

This is my thirtieth year in speaker design. All that time I have wanted better and still better amplifiers; being something of a specialist I have gone through the process, year after year, of hooking up every circuit that has come along, in the belief that others knew more about it than I did. I do not know any more about amplifiers than others, but I have found out where most of these did not match up to my requirements, and it is that knowledge I shall try to give you. This chapter, therefore, will deal with the approach to the problem; the next will constructively criticise the various types of hi-fi output stages so that you can make your own selection.

Any exhibitor at an audio fair knows quite well that if he stages a demonstration with artistic restraint, with a genuine desire to display his equipment as it should be heard in a civilised home, he will lose business. It is not a case of one exhibitor trying to shout the next man down; it is what draws the crowds that matters. Every show has a large proportion of acoustic hypocrites who do not know much about music, but think they know a lot about high fidelity. They dash from one room to another, listen for a minute and off to the next, rather like the traditional American tourist "doing" Europe in three days. If nothing very much seems to be happening in room A and room B is raising hell, then the crowd will be in room B, whatever the real quality of the reproduction. In due course these people will report to their friends that the Company in room A does not know how to put on a show. A manufacturer hires a room at an audio show for the sole purpose of selling his equipment, and whether he likes the noise he creates in that room or not, his main interest is in the order book. If he gets the orders, he is doing the right thing; that seems to be all there is to it.

But there is more to it, for this unfortunate state of affairs has pre-conditioned the audio fan into assuming that high fidelity and high volume go hand in hand, and that is not only bad for your neighbour, but bad for yourself. If you have never been to a first class symphony concert, I suggest you go to one. You will get the shock of your life, for the first thing that will strike you is the fragility (the only word I think fits the case) of the orchestra. I am assuming you normally run your equipment at a fairly high "realistic" volume, and what you will find is the conductor working quite hard, egging on the instrumentalists to do something grand, and all that comes out is a thin strain of music which, if it is a Mozart or Haydn programme, may be so quiet that any noise from the audience will ruin the whole thing. If, however, it is the Dies Irae from Berlioz's Requiem, with full orchestra, 16 timpani, 4 brass bands and a choir of 300, then it does not matter very much what the audience does; it will be something like Haydn being played at an audio fair. The great "trick" record of the 1952 New York Fair was the fine Westminster recording of the Haydn Military Symphony (No. 100). Haydn composed far finer symphonies but could I demonstrate these? No! Over and over again I was asked to "put on The Military and give it all you have. I want to hear that big bass drum". I had to do it or out they went! Music is more than big bass drums and that was no way to demonstrate realistic sound reproduction.

If you are a regular concert goer you are accustomed to the refinement of good music beautifully played and conducted. If you can get the same pleasure from a record of a work you love as you got in the concert hall, you have a good reproducer, and the volume will be adjusted to suit.

The amount of power required to produce that volume depends on the size of the room, the way it is furnished and the sensitivity of the speaker. As I have explained, a horn loaded speaker is more efficient than a direct radiator, and the sound output of the latter depends on whether it is enclosed in a housing which projects the sound from the back of the diaphragm or absorbs it. Order of sensitivity is, therefore, horn loaded, direct radiator in acoustic phase inverter, direct radiator in infinite baffle or closed box. For these three types of speaker systems the output power required for an average living room of about 2,500 cubic feet is about 3, 6 and 11 watts undistorted peak. As the smallest high fidelity amplifier generally available is a 10 watt job, and others are available with claimed undistorted outputs up to 60 watts, there seems to be something wrong with my figures. Which brings us to the situation that there is more in assembling a hi-fi system than buying an amplifier whose looks and price appeal to you and using it to drive the speaker of your choice.

The apparently simple process of connecting a speaker to an output stage by means of an audio transformer is, in reality, an extremely complicated business indeed. The problem is usually avoided by adopting what might be called technical cliches. Given the optimum load of the output stage, as revealed in the tube catalogues, and the nominal impedance of the speaker, the ratio of primary to secondary turns in the output transformer is obtained from the formula:

$$\text{Ratio} = \sqrt{\frac{\text{Optimum load of output stage}}{\text{Speaker Impedance}}}$$

It is common knowledge that a reserve of power will guard against distortion through overload on peaks, and if the amplifier tends to distort, either through poor design or because of the critical load of tetrodes and pentodes, put in some negative feedback which will reduce distortion and lower the plate impedance of the output stage. It seems so easy. Now let us consider what really does happen.

To conform to the foregoing ratio formula it is obvious that the transformation ratio must be constant for all frequencies if the load (i.e. the speaker) has constant impedance. A transformer is an impedance matching device, and the load reflected on to the output tubes is that of the impedance of the secondary circuit multiplied by the turns ratio squared. This is with an ideal transformer, but practical transformers are not ideal. At low frequencies the ratio is less by a factor which includes the plate resistance of the output tubes, the resistance and inductance of the primary winding. At high frequencies loss of ratio results from leakage inductance (through imperfect coupling between the two windings), self-capacity of the windings (acting as a short circuit at high frequencies). To make things more difficult, the transformer will peak at a high frequency through resonance of a low—"Q" circuit formed by the primary reactance and resistance and the self-capacity of the windings; beyond this peak the response falls rapidly.

The design of audio transformers is a perfectly straightforward matter for a competent technician, but is too complex to be included in this book. The reader will probably buy his output transformer from a specialist manufacturer, but the best results will not be obtained by using a so-called universal transformer. As you can see, even a well designed transformer will not have a constant transformation ratio unless the actual output tubes are specified as well as the speaker impedance. A tapped secondary may not have equal coupling for all frequencies, and although the primary inductance may be adequate to give good bass, the actual value of the primary inductance depends not only on the lowest frequency to be reproduced, but the relationship between the optimum load and the a.c. resistance of the output tubes. Different tubes may have the same load resistance yet differ in their plate resistance. This, in turn, determines the damping factor and accounts for the triode-pentode controversy. The a.c. resistance of triodes is about a quarter of the optimum load; tetrodes and pentodes have an a.c. resistance about five times the load resistance. A good deal of the prejudice against the latter is due to the fact that they are not properly used.

Apart from the acoustic performance of a speaker, it has two properties which are directly associated with the output stage—power handling capacity and impedance. Advertisements and catalogues frequently state that some particular model, is say, a 15 watt speaker, but this bald statement means nothing beyond an implication that it is suitable for use with a 15 watt amplifier. It may not be.

As far as frequency is concerned, the power-handling capacity of a speaker depends on the flux density in the gap, the freedom of suspension and the size of the cone. Fig. 13 gives some information on this: it indicates that for speakers of 5% efficiency with a free movement of cone and coil of $\frac{1}{4}''$ (a fairly usual state of affairs) a 5 watt input produces maximum deflection at 30 cps. in a 15 inch speaker; at 45 cps. in a 10 inch speaker; and at 80 cps. in a 5 inch speaker. Any greater power can only result in gross distortion and mechanical damage. It follows that the application of any power greater than 5 watts is restricted to those frequencies higher than those just listed at which the cone movement does not exceed $\frac{1}{4}''$. In any case the lower limit of non-distorted reproduction is the bass resonant frequency, for below that the output is mainly third harmonic. A speaker has, therefore, virtually no power handling capacity below the bass resonant frequency, and above that is limited by the cone size—free movement factor. (Certain types of enclosures can modify the bass response, as described previously in this book, but acoustic output of a speaker and its enclosure should not be confused with the fundamental power handling capacity of the speaker itself).

At higher frequencies, where cone movement is of no consequence, the limiting factor is dissipation of heat generated in the voice coil. If watts go into the coil, the inductive component is wattless, but the resistive component must create heat, and if the temperature rise is too great the coil assembly will be destroyed. Some readers may have had the unhappy experience of burning out a speaker when no signal was fed into the amplifier, simply because there was enough supersonic oscillation in the output stage to do the damage. It has happened to me. At middle and high frequencies, therefore, the power handling capacity of the speaker is a function of the actual size of the voice coil and the heat radiating abilities of the adjacent metal parts.

Finally, what is the impedance of the speaker? It is not the figure quoted by the manufacturer, for it varies widely with frequency. Quoted speaker impedance follow on from an old rule-of-thumb concept that the impedance of a speaker is approximately twice the d.c. resistance of the voice coil. For design purposes of cheap equipment this is near enough not to matter, but it is not near enough for the best results. The speaker manufacturers quote as usual impedances 4, 8 and 16 ohms, and the output transformer manufacturers obligingly tap their secondaries at these figures.

There are dozens of versions of the so called "equivalent loudspeaker circuit", which consist of more or less complicated networks of resistance, inductance and capacitance; the variations derive from different opinions of how the various parts of the speaker's construction and behaviour will be interpreted in terms of inductance and capacitance. Pure resistance does not vary with frequency but the inductive and capacitive reactances do, so the impedance of the speaker must vary with frequency. In general, there is a sharp rise in impedance at bass resonant frequency, than the normally quoted impedance at about 500 to 1,000 cps.; after this the impedance rises at an increasing rate owing to the inductance of the voice coil. How, then, if you cannot get a guaranteed impedance curve from the maker of your speaker, can you determine its impedance? The simple answer is to measure it, and this is almost obligatory in the case of multi-channel systems with dividing networks, for a very complicated total network involved.

Fig. 34 shows the output transformer of an amplifier which is fed from an audio oscillator. Across the secondary a known resistance R and the speaker under test are connected in series. An a.c. peak voltmeter can be connected across either R or the speaker. R must be either a non-conductive wirewound resistor or a bank of composition resistors of a wattage as high as the audio power from the amplifier. If R were not used, the speaker might be burned out with steady high inputs. Signals of various frequencies are injected into the amplifier and readings at each frequency taken across R and then across the speaker. Call the voltages across these E_r and E_s respectively, then:—

$$\text{Impedance of Speaker} = \frac{R \times E_s}{E_r}$$

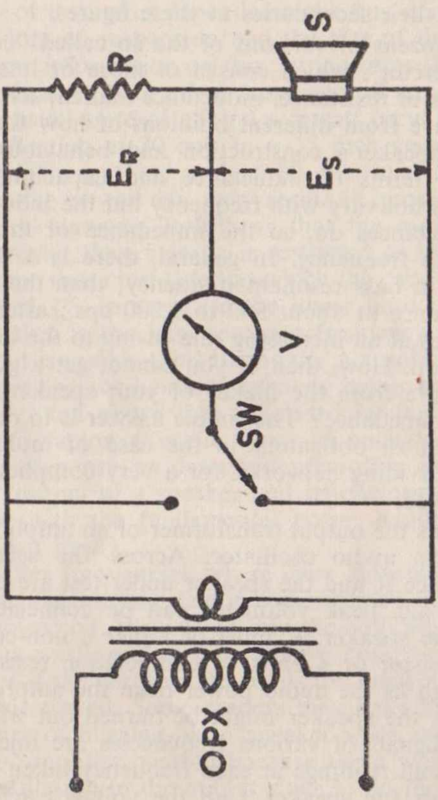


Fig. 34 Hookup for impedance measurement of a speaker.

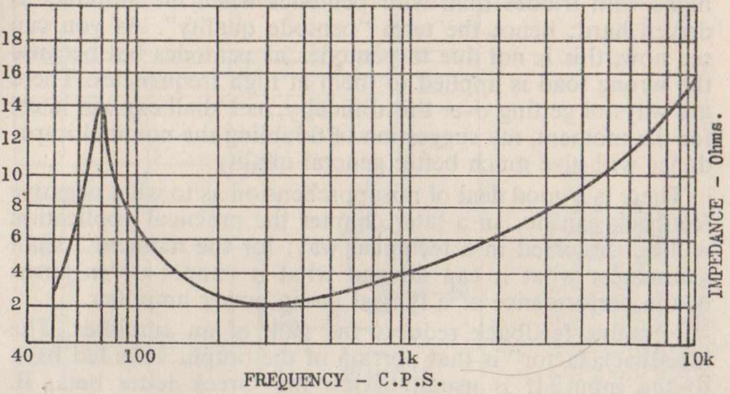


Fig. 35 Smoothed impedance curve of a typical 4 ohm speaker with a bass resonance at 70 c.p.s.; 8 and 16 ohm speakers would have proportionate variations of impedance.

It is important to take a careful reading exactly on the bass resonant frequency, indicated by a sharp rise in the voltage reading across the speaker. When all the readings are taken, a curve is drawn, which will look like Fig. 35, which is a curve of a typical 4 ohm speaker with a bass resonance at 70 cps.

If the output transformer has been chosen to give the optimum load with a secondary impedance of 4 ohms, then there will be serious mismatching at the bass resonant frequency. This has an effect on the reproduction when the speaker is coupled up for its nominal impedance.

If you study the figures for triodes and pentodes or tetrodes in the tube manuals you will see that the latter give more power and less distortion than triodes for a given plate supply, but this is only when the load is reasonably correct. The optimum load gives the optimum power without distortion, but if that amount of output power is required and the load is wrong, distortion is excessive. Triodes are not as critical as to optimum load, and unless the amplifier is driven hard, the distortion from this mismatching will not be enough to worry about. As it is apparently impossible to produce a speaker with constant impedance, a speaker assessed at its nominal impedance will give less distortion in the extreme highs with triodes than with pentodes when the amplifier is driven hard; hence the term "pentode quality". As you can see now, this is not due to pentodes as pentodes but because the wrong load is applied to them at high frequencies. There are ways of getting over this difficulty, as I shall explain later; for the moment, my suggestion of doubling the nominal impedance will give much better general quality.

There is a good deal of misapprehension as to what negative feedback can do. In a later chapter the practical application will be discussed in a technical way; for the moment I shall summarise what it can do and what it cannot do in terms of the performance of a typical audio power amplifier.

Negative feedback reduces the gain of an amplifier. The "feedback factor" is that portion of the output volts fed back to the input. It is usually given the Greek letter beta, B. Obviously B cannot be greater than unity, and if there is no feedback then it equals 0. If the gain of the amplifier is expressed in db, then B can be expressed in db. If we call the amplification of the amplifier A, without feedback, then the amplication after feedback is $A/(1+BA)$. You may find this formula in textbooks with the sign in the denominator negative, but if it is negative feedback then B carries a negative sign itself, so my formula is finally correct. If in this formula you call A distortion or output tube plate resistance, these parameters are reduced by the same amount. So negative feedback reduces gain, distortion and the effective plate resistance of the output tubes.

Reducing gain seems a futile sort of thing to do but it is quite important. In the absence of feedback the amplifier will have a certain frequency response, and it will tail off in the bass and treble. If, now, negative feedback is applied, it will be clear that less voltage will be fed back in the bass and treble simply because the output voltage is less, so there will be loss of gain at each end of the frequency response and the feedback amplifier will show a wider flat response than the original. It sounds wonderful, which is why is it used so frequently, but now creeps in a very serious liability.

Change of phase occurs in every valve and every RC coupling. With triode output stages more amplification is needed than with tetrodes or pentodes, perhaps even to the extent of having to provide an extra stage to do it. At any rate the phase change in a multi-stage amplifier can become

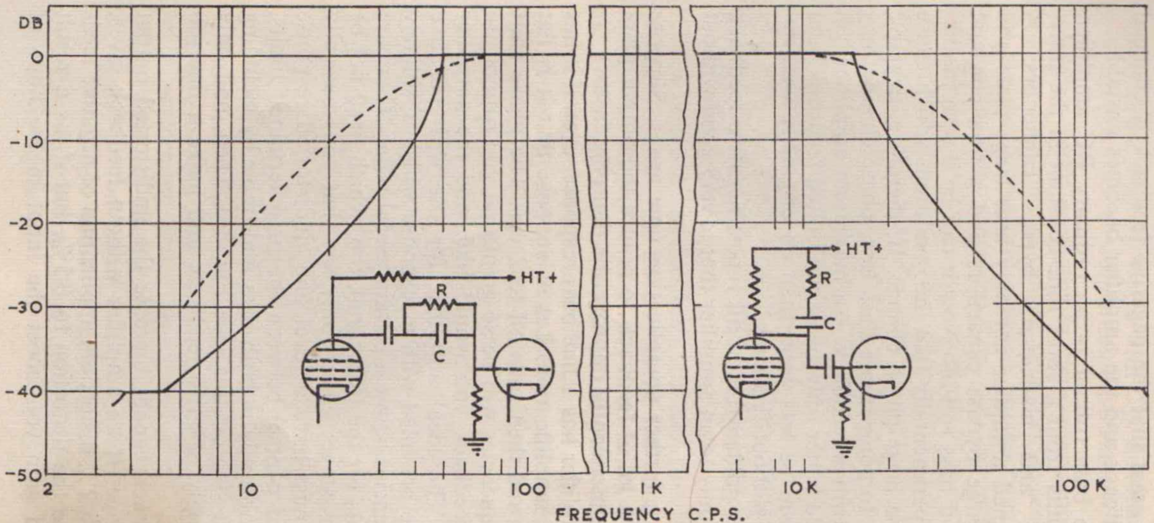


Fig. 36 The ideal curve of a multistage amplifier to give a level response from 50 to 15,000 c.p.s. with 30 dB. of negative feedback and a safety margin of 10 dB. (to guard against instability after heavy transients) is shown by the solid line. The dashed line shows the response of a well-designed amplifier without step circuits. The step circuits for the desired bass and treble attenuation are shown below their respective portions of the frequency scale. These are the networks marked R and C in the intervalve circuits shown.

so progressively great that the negative feedback is changed into positive feedback and the amplifier becomes unstable.

Take only the bass roll-off in an ordinary RC amplifier. This usually results from a short time-constant in the inter-stage couplings and inadequate bypass capacitors. A generously designed amplifier not only uses large plate-grid capacitors but large bypass capacitors and a time-constant is chosen to avoid bass roll-off. Now it can be shown mathematically and experimentally that the conditions which cause bass roll-off cause large phase change. If, therefore, negative feedback is used to compensate bass loss, phase change may convert it into positive feedback. An indifferent amplifier can therefore be made better only by using limited feedback, and the final result will be less good than an originally well designed amplifier without feedback.

Remember this golden rule at all times; negative feedback is of real service only to an amplifier that is very good without feedback.

Unfortunately a further complication now arises. Suppose you want to use a lot of feedback to reduce the output plate resistance (and consequently improve the damping factor).

Your good amplifier has a fine bass response, even if it has several stages of amplification. Let us suppose that it is flat down to 20 cps. and then rolls off to 2 cps. The phase shift at the lowest frequencies will be so great that you cannot use the amount of feedback you would like, and the amplifier will motorboat. The ideal amplifier for use with negative feedback must be provided with a response which absolutely cuts off all frequencies below a certain useful point. There are parallel arguments for the treble end, too, which need not be considered at this stage; I need only say that your basically good amplifier must have a level and undistorted response (within reason, of course) between the predetermined limits, and then cut off abruptly in both bass and treble by including suitably designed step circuits (see Fig. 36). Then, and only then, you can apply negative feedback and make a fine job better.

Negative feedback cannot increase the undistorted output of any power stage. If an amplifier without feedback gives, say, 10 watts with 2% distortion, application of negative feedback may reduce the distortion to 0.5%, but if, as a result of the decreased gain you boost the input in the hope of getting more than 10 watts with 2% distortion you will find it is not possible. A simple demonstration will prove this.

Use an audio oscillator to drive your amplifier and with a resistive load on the output transformer secondary to equal the speaker impedance marked on the transformer, connect an oscilloscope across the load. Set the oscillator frequency to anything you like, but 1,000 cps. is a very safe one. Any amplifier ought to be able to handle that frequency without distortion. Disconnect the feedback circuit. If there is variable feedback, so much the better. Using a sine wave input, adjust the volume control until the tube picture is just not flat-topping, and not the height of the trace above the datum line. If you increase the input or turn up the volume control, the sine wave will now take on a flat top, getting a wider flat as you increase the signal to the output stage but the trace will not get any higher.

Now connect the feedback circuit. The flat top will disappear because you have reduced the gain of the amplifier and so the output stage is not overloaded. You can increase the input until the flat top is on the point of appearing again, and if you have variable feedback you can increase the input still more and cut out the flat top by increasing feedback. But you cannot heighten the trace. In other words, you cannot get more power out of the tubes.

It is sometimes rather difficult to spot the divergence from a pure sine wave of the trace on a small tube. The feed-back distortion cleans up the wave shape but when maximum undistorted power is reached and further input r volume (gain) is applied there will be a very slight increase of height with feedback and the wave form will look sinusoidal actually, however, the sides of the wave will be slightly straighter, implying some distortion.

Without feedback the change in waveform is rather gentle until flat-topping starts and it may be thought that the amplifier is performing better than it really is. With feedback the shape is seen to change quite suddenly. Feedback, therefore, reduces distortion before overloading starts, but the overload point is reached suddenly, from a practical point of view, with no greater output than that obtainable from an amplifier without feedback.

getting more than 10 watts with 2% distortion you will find it is not possible. A simple demonstration will prove this.

Use an audio oscillator to drive your amplifier and, with a resistive load on the output transformer secondary to stand for the speaker impedance, connect an oscilloscope across it. An oscillator [illegible] the load. Set the oscillator frequency to anything you like, but 1,000 c.p.s. is a very easy one to use. The amplifier should be able to handle that frequency without distortion. Disconnect the feedback circuit. If there is a variable feedback control, set it to minimum. Using a sine wave input, adjust the volume control until the tube picture is just flat. Note the output and see the height of the trace where the distortion starts. If you increase the input a little and turn the volume control, the sine wave will now have a flat top, getting wider the more you increase the signal to the output stage, but the trace will not get any higher.

Now connect the feedback circuit. The flat top will disappear because you have reduced the gain of the amplifier and so the output stage is not overloaded. You can increase the input until the flat top is on the point of appearing, and if you have variable feedback you can turn up the feedback still more and look at the flat top by increasing the input. But you cannot [illegible] times. In other words, you cannot get more power out of the tubes.

It is sometimes rather difficult to spot the divergence from a pure sine wave on the trace on a small tube. The feed-back distortion [illegible] the wave shape but when the maximum undistorted power is reached and further input volume is applied, there will be a very slight increase of height with feedback, and the wave will look sinusoidal. Actually, however, the peaks of the wave will be slightly flattened, [illegible].

Without feedback the change in waveform is rather gentle until the [illegible] and it [illegible] that the amplifier is performing better than it really is. With feedback the change is seen to [illegible] suddenly. Feedback, therefore, reduces distortion before overloading starts, but the overload point is reached suddenly. From a practical point of view, with no greater output than that obtainable from an amplifier without feedback.

BERNARDS RADIO BOOKS

No.		Price
56.	Radio Aerial Handbook	2/6
57.	Ultra-Shortwave Handbook	2/6
58.	Radio Hints Manual	2/6
64.	Sound Equipment Manual	2/6
68.	Frequency Modulation Receivers' Manual	2/6
73.	Radio Test Equipment Manual	2/6
83.	Radio Instruments and their Construction	2/6
96.	Crystal Set Construction	1/-
99.	One Valve Receivers	1/6
100.	A Comprehensive Radio Valve Guide, Book 1	5/-
103.	"Radiofolder" A. The Master Colour Code Index for Radio and T.V.	1/6
104.	Three Valve Receivers	1/6
107.	Four Valve Circuits	1/6
108.	Five Valve Circuits	2/6
121.	A Comprehensive Radio Valve Guide, Book 2	5/-
123.	"Radiofolder" F. The Beginners' Push-Pull Amplifier	1/6
126.	Boys' Book of Crystal Sets and Simple Circuits	2/6
129.	Universal Gram Motor Speed Indicator	1/-
134.	F.M. Tuner Construction	2/6
135.	All Dry Battery Portable Construction	2/6
138.	How to Make F.M. and T.V. Aerials, Bands 1, 2 and 3	2/6
141.	Radio Servicing for Amateurs	3/6
143.	A Comprehensive Radio Valve Guide, Book 3	5/-
145.	Handbook of AM/FM Circuits and Components	2/-
146.	High Fidelity Loudspeaker Enclosures	5/-
147.	Practical Tape Recording Handbook	5/-
148.	Practical Transistor Receivers, Book 1	5/-
149.	Practical Stereo Handbook	3/6
150.	Practical Radio Inside Out	3/6
151.	Transistor Superhet Receivers	7/6
155.	Portable Transistor Radio and Radiogram	2/6
156.	Transistor Circuits Manual, No. 1	2/6
157.	A Comprehensive Radio Valve Guide, Book 4	5/-
158.	Radio, Television, Industrial Tubes, Semiconductor and Diodes Equivalents Handbook (208 pages)	9/6
159.	Realistic High Fidelity	5/-
160.	Coil Design and Construction Manual	5/-
161.	Radio, Television and Electronics Data Manual	3/6
162.	High Fidelity Stereo Gramophone	5/-
163.	Transistor Circuits Manual, No. 2	2/6
164.	High Fidelity Tape Recorder for the Home Constructor	2/6
165.	Hartley on Hi-Fi, Book 1, Radio Tuners	5/-
166.	Public Address Systems	2/6
167.	Transistor Circuits Manual, No. 3	2/6
168.	Transistor Circuits Manual, No. 4	2/6
169.	Hi-Fi Transistor F.M. Tuner for the Home Constructor	3/6
170.	Transistor Circuits for Radio Controlled Models	7/6
171.	Super Sensitive Transistorised Pocket Set Construction	3/6
172.	International Radio Stations List	2/6
173.	Practical Transistor Audio Amplifiers...	3/6
174.	Transistor Subminiature Receivers Handbook	5/-
175.	Transistorised Test Equipment and Servicing Manual	3/6
176.	Transistor Audio Amplifiers Manual	6/-
177.	Modern Transistor Circuits for Beginners	7/6
178.	Comprehensive Radio Valve Guide, Book 5	6/-
179.	Transistor Circuits Manual No. 5	5/-
180.	British Semiconductor Survey	15/-
181.	22 Tested Circuits Using Micro Alloy Transistors	5/-
182.	"At A Glance" Radio Valve & T.V. Tube Equivalents...	3/6
183.	How to Receive Foreign T.V. Programmes on your Television Set by simple Modifications	5/-
	Resistor Colour Code Disc Calculator	1/6
	Engineers' Reference Tables	1/6
	International Radio Tube Encyclopaedia—3rd Edition	63/-