1958 EDITION

hi·fi ANNUAL
& AUDIO HANDBOOK

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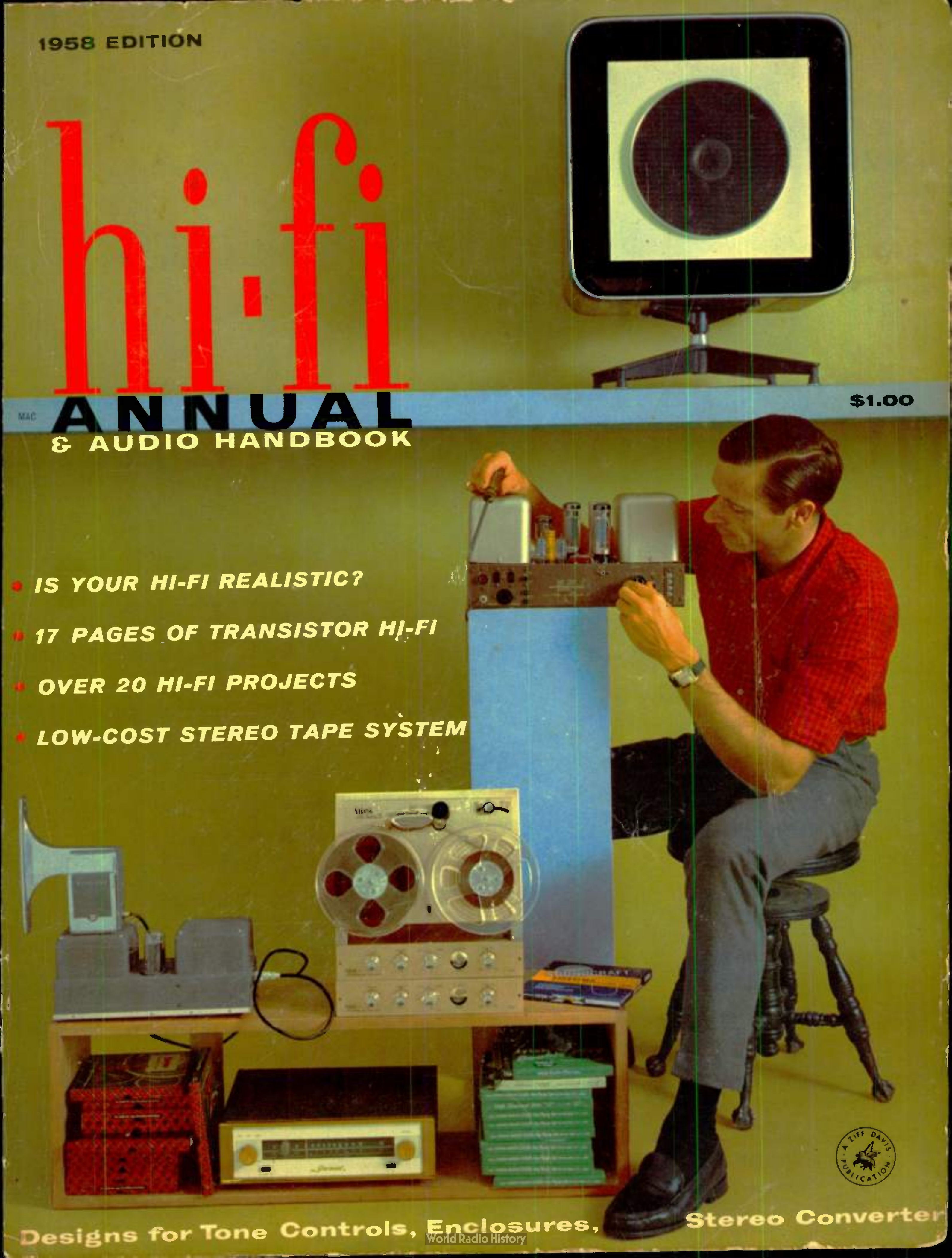

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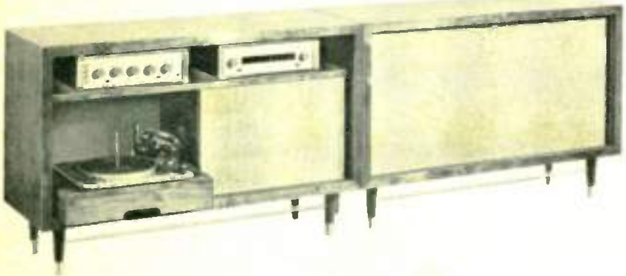

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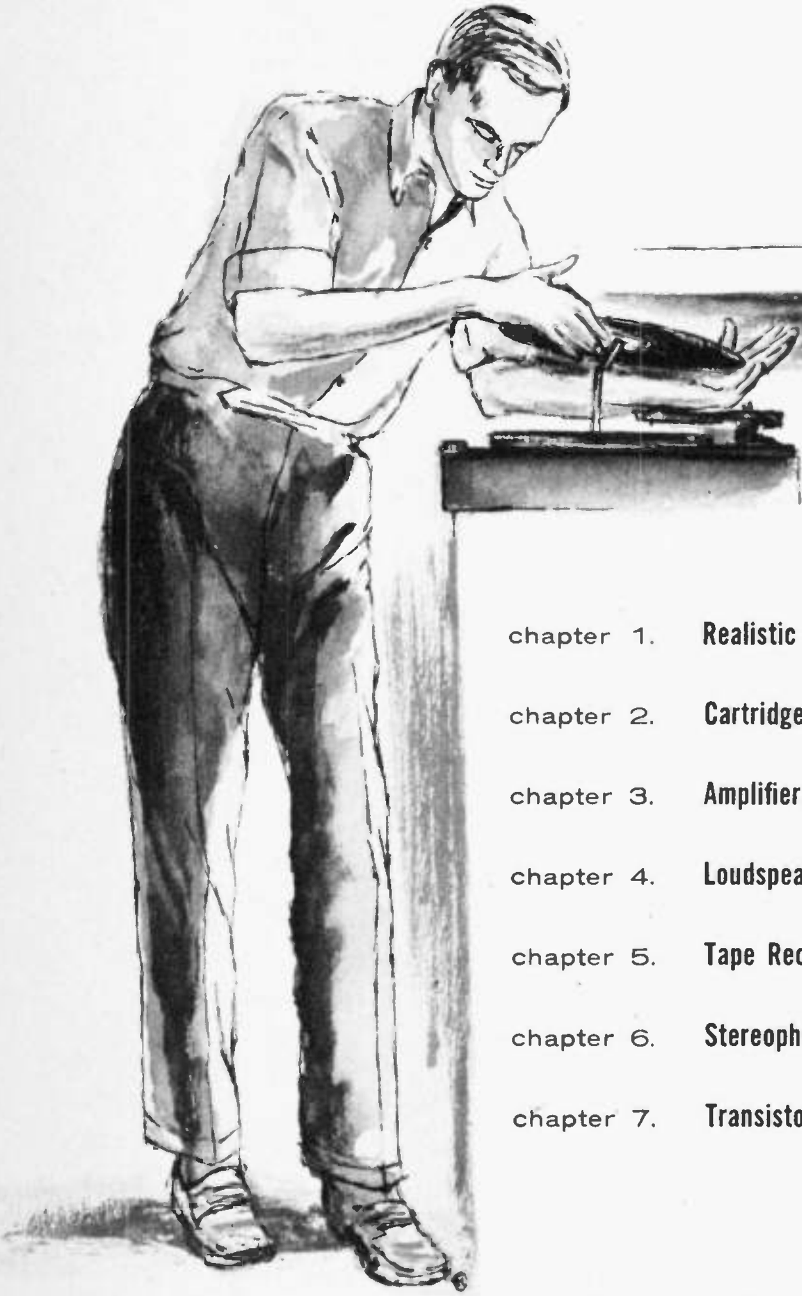

CONTENTS

a selection of articles from **RADIO AND TV NEWS**

ZIFF-DAVIS PUBLISHING CO., 366 Madison Ave., New York 17, N.Y. William Ziff, PRESIDENT; H. J. Morganroth, VICE PRESIDENT; Michael H. Froelich, VICE PRESIDENT; Michael Michaelson, VICE PRESIDENT and CIRCULATION DIRECTOR; George Carney, SECRETARY - TREASURER; Albert Gruen, ART DIRECTOR.

ON THE COVER: Some of the most interesting new high-fidelity components, performance and style wise, are the "Ionovac" of Electro-Voice, the stereo tape setup of Viking, the AM-FM tuner of Sherwood, the amplifier kit of Acrosound, and the Eames-designed speaker of Stephens.
Ektachrome by Gerald Hochman

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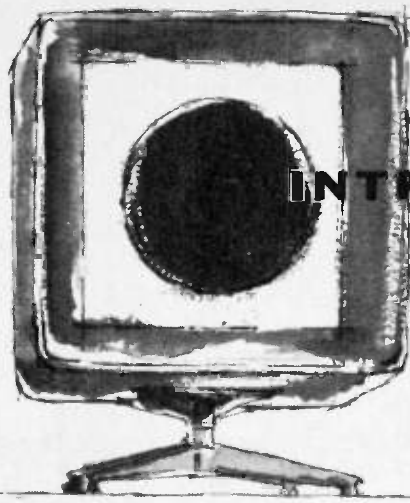

INTRODUCTION

The search for an improved method of presenting stereophonic sound, the application of transistors to high fidelity equipment and the improvement in piezoelectric phono cartridges all made exciting news in 1957. These advances are detailed in Chapters 6, 7 and 2, respectively. But, there was a great deal more done in the improvement of conventional equipment and in the challenging of long-accepted notions that did not make big news but may be of more lasting importance. Some of this material is also presented in this second edition of the HI-FI ANNUAL & AUDIO HANDBOOK.

As high fidelity has become more and more popular, and as it spreads to great numbers of technically uninformed consumers, opportunistic manufacturers have not hesitated in applying the label "high fidelity" to their products which bear no relation to the carefully-engineered and quality-controlled components made by the hi-fi specialists. Also, many hi-fi cliches relative to speakers and amplifiers are being used and pushed in advertising claims without serious thought being given to their effect on the future of hi-fi. Mr. H. A. Hartley, the well-known British designer and manufacturer of hi-fi components analyzes and attacks many of these notions and makes suggestions for selecting speakers, amplifiers, etc., in Chapter 1.

Many new ceramic and crystal cartridges claim to have excellent response characteristics. Are these claims substantiated by objective tests? The article on ceramic cartridges in Chapter 2 answers this question and gives information on equalization besides. In addition, there are build-it-yourself articles on tone controls, preamps, and, in Chapter 3, amplifiers. Chapter 4 offers information on enclosure design and construction as well as details on some really new speakers.

Complete tape recording systems are now available in "building block" form for custom installations. How to expand a typical installation in easy-on-the-budget steps from playback only to stereo record and playback is described in Chapter 5.

The articles contained herein are those which received the greatest response from the most knowledgeable and critical hi-fi fans in the world, the readers of RADIO & TV NEWS magazine. Those of our readers who wish to keep up with the monthly advances in hi-fi, and also desire to build their own equipment, should read RADIO & TV NEWS.

Charles Tepfer

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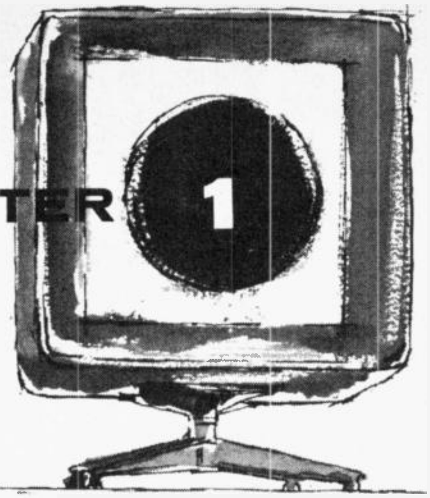

CHAPTER 1

REALISTIC HIGH FIDELITY

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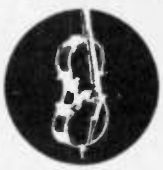
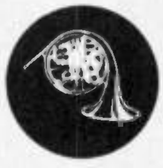
Realistic High Fidelity





By **H. A. HARTLEY***
Audio Consultant

Part I. "Is today's high-fidelity really realistic?" Subsequent articles cover facts of room acoustics and the design and operation of all the various units in a home sound reproducing system setup.

THE object of this series of articles 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 may 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 had been in radio and audio research ever since the "wireless" industry came into existence. I was also interested in music. In those days the reproduction obtainable from the usual commercial equipment may have satisfied the ordinary listener, but no one with any knowledge of what live music sounded like could pretend that it was anything better than a very thin ghost of the real thing. With a few co-workers of similar outlook I came to the conclusion that something could be done about it, and we developed amplifiers and speakers that were immeasurably better than anything on the commercial market. We tried out our equipment on the sales department of the company for which we worked, and it was turned down flat because it sounded uncomfortable. It was criticized because the reproduction was edgy and thin, and had no nicely rounded "tone" (whatever that meant). Actually what was wrong with the idea was that in reproducing all that was best in the whole chain of sound reproduction it also reproduced all the defects, and the salesmen said that although we might have gotten something if we could make the regular equipment sound better, they couldn't sell something that made it sound worse.

This very primitive argument contains the whole schedule of drawbacks of what we christened high-fidelity reproduction, for when we said something was high-fidelity reproduction we meant simply that the equipment itself did not add distortion to the signal going into it. Of course, we needn't have used the word "high" at all, for fidelity without any qualification is just fidelity, no more, no less; but those were the days when the cheapest and most loathsome radios were advertised as having "perfect reproduction," so the addition of "high" was just a gimmick.

However, although the commercially-minded gentlemen wouldn't have anything to do with our new sound, the idea clung to a number of enthusiasts and tempted your present contributor to throw up his job as a scientific worker in a commercial organization and start in the hard way, by becoming his own boss. What I have to tell you, therefore, is the cumulative experience of over twenty-five years of specializing, commercially, in high-fidelity sound reproduction; and particularly during the past two or three years I have noted that advertising agents, with their customary ingenuity, have found that "hi-fi" today is nearly as effective as the "perfect reproduction" of twenty years ago, and means as little.

I am not going to suggest that this misuse of the term is always intended to be misleading. Certainly some quite ordinary equipment is labeled "high fidelity" when the manufacturer knows perfectly well that it isn't; but if the market has become conditioned to the broad principle that high-fidelity equipment is something better than ordinary equipment (which it is, since the

* Mr. Hartley is one of the "old timers" in the electronics field having served his apprenticeship in British broadcasting during its experimental phases at Writtle, Essex. With P. K. Turner, he founded Hartley-Turner Radio Ltd. to manufacture hi-fi equipment. When the plant was destroyed during the war, Mr. Hartley switched to the design of airborne radar. In 1946, the Hartley-Turner firm was reactivated as H. A. Hartley Co. Ltd. and is still operating in London. Mr. Hartley sold his British interests in 1953 and is now scientific and technical adviser to Hartley Products Co., New York.

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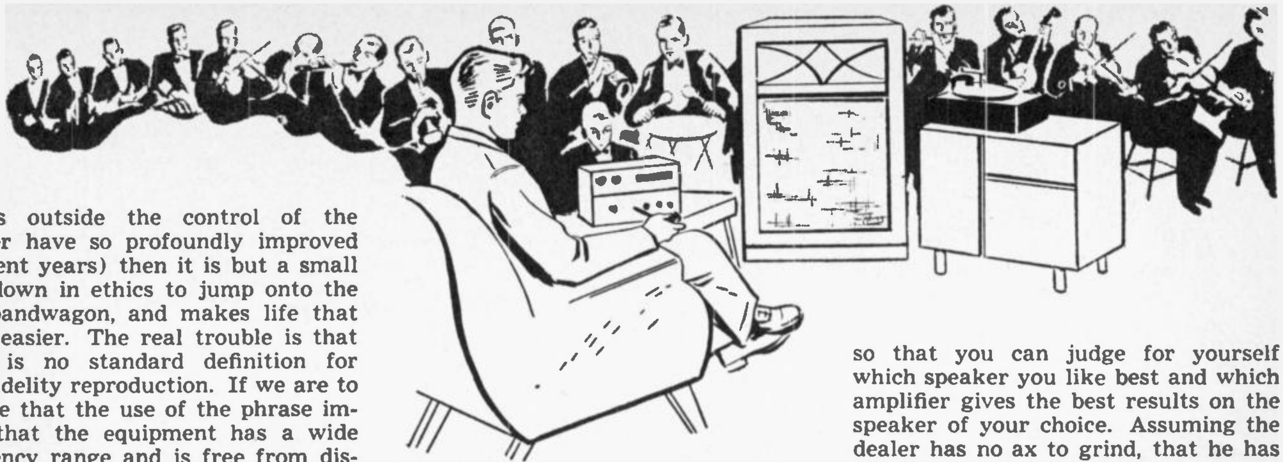

factors outside the control of the listener have so profoundly improved in recent years) then it is but a small slide down in ethics to jump onto the hi-fi bandwagon, and makes life that much easier. The real trouble is that there is no standard definition for high-fidelity reproduction. If we are to assume that the use of the phrase implies that the equipment has a wide frequency range and is free from distortion, then we might very well ask "how wide a frequency range?" and "how much distortion?", for it is quite obvious that mere mortals like ourselves cannot produce perfect equipment, nor can we expect perfect phonograph records, microphones, and transmitters. There has got to be a compromise between perfection and possibility at a price the people can pay, and where that compromise is established is a matter of considerable dispute.

The engineer can lay down what he considers the correct specification. Sounds can be analyzed, auditorium acoustics studied, pickups, tuners, amplifiers, speakers, and speaker housings can be designed to deal with the frequency range in a way which produces no more distortion than the ear has been measured to accept as fidelity. By scientific method it could probably be shown that a musical score could be translated into reproduced sound in such a way that the composer's intentions had been realized. And I assert, in the teeth of objections by my fellow engineers, that the result would *not* necessarily be high fidelity.

The reason for my being apparently "difficult" is that I insist that what I originally meant by high-fidelity, and what I should think the average music-lover supposes he wants when he buys high-fidelity equipment; is sound emerging from the speaker which is a very close approximation of the sort of thing I hear when I go to a concert. The whole idea of music is to create certain effects in the mind of the listener and the composer of the music intends, and has always intended, that the musical sounds he has invented or created shall be heard "live" at first hand. I agree that there is a school of thought which argues that there is a new field of enterprise for musical composers to think in terms of electronic reproduction, and I have no fault to find with such a philosophy; but this sort of musical composition must not be allowed to impinge on the music that has been written for direct hearing.

Mozart, Beethoven, Brahms, Berlioz, Elgar, Walton, Copland, to mention only a few composers who are "hi-figenic" (if I may coin a terrible word) intended their music to be heard directly by the human ear and not through a loudspeaker. The coming of electrical reproduction made it possible for a much wider audience to participate in the delights of listening to their compositions, but the interposition of the electrical reproducer must not permit any coloration to act as a sort of filter between the live music and the listener's ear. If a musically trained listener, fully aware of what a particular symphony sounds like in a properly designed concert hall when played by a competent orchestra, can hear the same work through a reproducing system in an ordinary living room and get practically the same aesthetic enjoyment from it, then he will be listening to what I call high-fidelity reproduction, and, from the artistic point of view, it doesn't matter two hoots what the frequency response is or what electrical and acoustical tricks the designer has played on him. What matters is the end product. For the purposes of these articles I am calling the system of reproduction *realistic high fidelity* simply because there are quite a number of excellently designed audio devices, technically correctly labeled high fidelity, because they do give wide frequency response without appreciable distortion, but which do not give that aesthetic satisfaction the realist demands for adequate enjoyment.

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 possible 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 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 ax to grind, that he has not loaded the dice in favor 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 the demonstration has been so good that you can't 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 home.

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 really 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 you can.

It is usual to liken sound waves to the ripples set up on 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,

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there is nothing else in common. If the pond is a rectanguar glass tank then disturbance of the surface at one end will enable you to 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 diapason 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 centered 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 still water. They are, therefore, *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 rarefaction 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 rarefaction. 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 of sine-wave form, but the peculiar characteristic of a sound wave is that it is created by little packets of compression traveling along a straight line, and when multiplied by infinity create spheres of compression traveling 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.

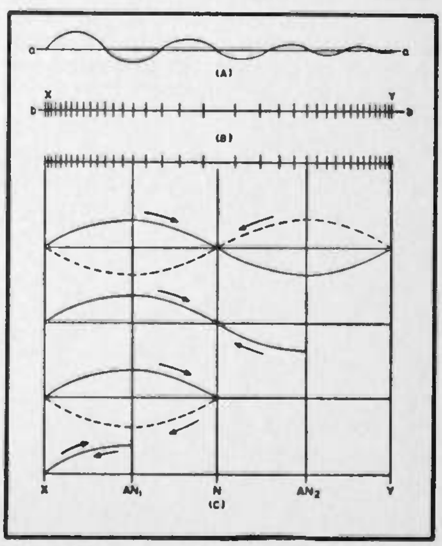
The discussion thus far deals only with a simple wave having indirect sinusoidal motion of the type described. The behaviour of the air can be analyzed by strict mathematical methods, but there seems little point in giving the mathematical proof if you are prepared to accept what I have written as correct. 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 will do your 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 traveling 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 rarefaction and move from a condition of maximum forward movement through zero to maximum backward movement (which is the same as 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 node and X and Y 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 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

Fig. 1. (A) Cross-section of transverse waves of water ripples. (B) Compression of longitudinal sound waves. (C) Effects of a reflector on sound waves at nodes and antinodes. Refer to article for details.



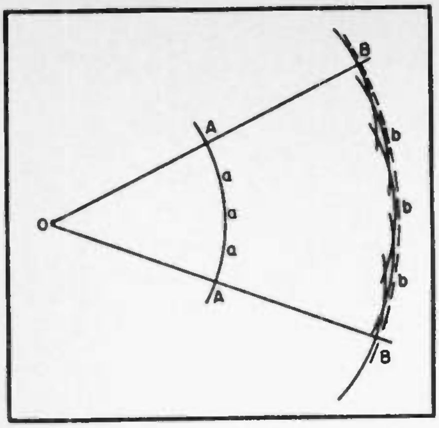


Fig. 2. Huyghens' principle of propagation of sound waves. This is discussed in text.

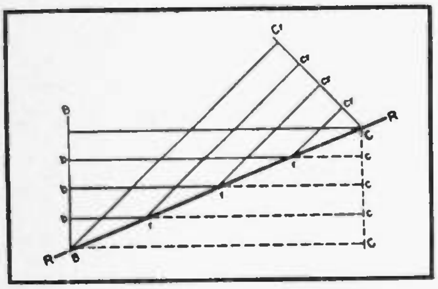


Fig. 3. Huyghens' principle, demonstrated in Fig. 2, applied to reflected sound waves.

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).

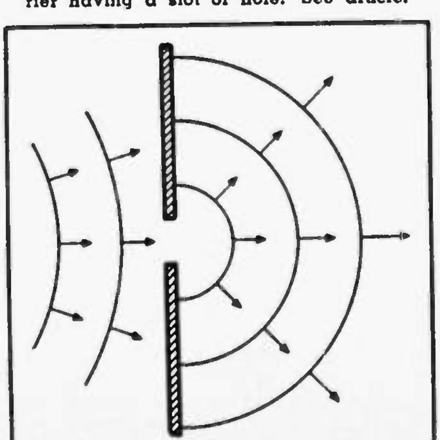
These results derive from the reflector being exactly at right angles to the line of propagation; to understand what happens when the sound wave falls obliquely on the reflector it is easier to consider what is usually called Huyghens' principle of wave propagation, for in any event we are interested 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 center we describe an arc AA, which can be subdivided 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 on 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 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 distances *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 center 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 distances *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 process it will be noticed that the wavefront is reversed with respect to the plane of the reflector.

Fig. 4. Dispersion effect of a sound barrier having a slot or hole. See article.



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 the 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 generalizations 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 criticize 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.

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Part 2. One important factor that is overlooked by many hi-fi enthusiasts is the effect of room acoustics on the quality of sound obtained from their home hi-fi systems. Here are some of the facts to know to obtain realistic audio reproduction.

IN PART 1 of this series I explained that A-B testing of audio products in a dealer's store might enable you to assess the differences between various competing products as heard in that store, but did very little in guiding you as to which would sound best in your own home. You have also seen how sound waves behave in unenclosed air and in chambers having reflective containing walls. Reflections seem to be undesirable, so it would seem a simple matter to hang draperies all over the walls and ceiling, put a thick carpet on the floor, and conceal the naked wood of the furniture with the hideous tablecloths, tasseled drapes, and even leg frills of the worst of the Victorian period of stuffiness; then get a good speaker and housing from the dealer and set it up in this acoustic morgue. Unfortunately it isn't as easy as that, for experiment will show that without reflection the musical reproduction has no life. But as soon as we do introduce reflections and echoes we also introduce complications.

Our rooms do 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 problem 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 aren't like that. We are ordinary folk and have to use what we have, for better or worse. Let us try and work out how to do it for the better.

First of all is the size of the room. No doubt you have read over and over again that you will get loss of bass if the room isn't 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 textbooks, 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 don't want to put the speaker in one corner and don't want 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 don't 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 someone 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 don't accept these laid-down principles, and as I can hear exceedingly well reproduced low notes in my own room which isn't anywhere near as big 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 rarefaction; a human ear in the path of the sound wave will be acted on by the compressed and rarified air. If there is only one sound wave the eardrum 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 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,* 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 (Part 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 emerged from the speaker and the bit is reflected from a wall in such a way that a zone of rarefaction 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" 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 tones, 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 behavior 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 a 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 doesn't 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 emphasized. 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

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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 generalization 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 material must be introduced.

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

As far as the usual room accessories are concerned, unglazed bookcases fairly full of books are good sound absorbers, but if glazed the coefficient 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 recognized 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 directly 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 fiber; 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 to 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 coefficient 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 an automobile 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 moldings will help; the normal concave plaster coves on ceilings detract.

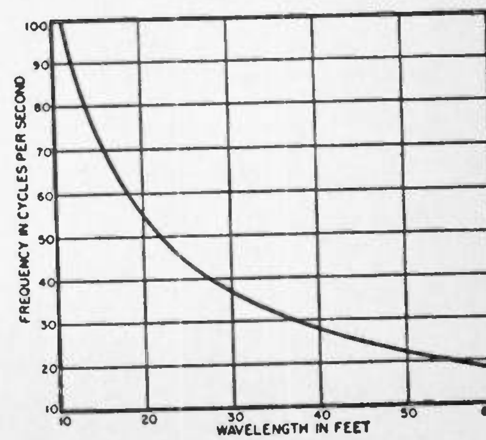
Naturally the position and direction of the speaker will determine to a great extent how the sound is ultimately distributed. Most speakers focus the highs, and speakers which focus severely may sound best by having the front covered with a slotted board, as mentioned in Part 1. This particularly applies to speakers which have an unduly large output at 2000 to 3000 cps (a common fault), for frequencies of this order don't seem to fit the absorption curves of tiles.

Open window	1.00
Audiences	0.96 to 0.44
Felt 1 inch thick (varying density)	0.80 to 0.50
Cork 1½ inch thick	0.32
Insulating panels	0.70 to 0.25
Rugs and carpets (varying thickness)	0.30 to 0.20
Unglazed oil paintings	0.28
Velvet curtains	0.25 to 0.20
Cretonne curtains	0.15
Linoleum	0.12
Pine boards	0.06
Plaster on laths	0.034
Glass	0.027
Bricks and hard wall plaster	0.025
Cheesecloth and similar material	0.02 and less*

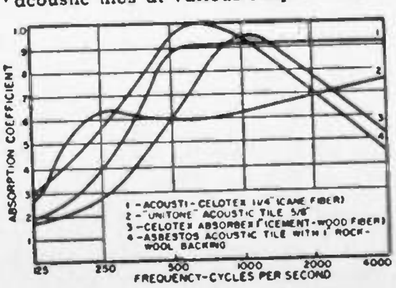
*Lace and nylon curtains are almost completely transparent to sound, so their presence in front of various wall materials has no effect on the absorption coefficients of those materials; drapes to be effective must be of velvet, cretonne, art jute (burlap), or similar substantial material.

Table 1. Absorption properties of various materials found in the average living room.

→Fig. 5. Wavelengths of sound waves, in feet, at frequencies up to 100 cps.



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



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Part 3. Reverberation is an important factor in obtaining good sound reproduction. Here are simple tests you can make in your own listening room and methods for correcting faulty acoustics.

YOU now know how sound waves behave, you have been introduced to room resonances, reverberation, standing waves, and absorption coefficients. You realize 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 1000 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's start with reverberation. If the reverberation period is too long then good reproduction is impossible, so have this reduced to not more than 1½ 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 schoolboy trick of inflating a paper bag and bursting it between the hands. Have all the normal furnishings in 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 1½ 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 conveniently be 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 to 1½ seconds.

Now comes the far more tedious business of dealing with reflections and irregularities in the distributed sound. It is the plan of this series 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 on 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 oscillator and listening for standing waves, as explained in an earlier part, but this, unfortunately, doesn't 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 molded corrugated surrounds resonate somewhere between 35 and 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 will be masked by that of the higher simply because if the speaker is unable to reproduce a frequency lower than that of the spider washer, the cone surround peak will not show on the curve.

In case this is not quite clear, I should explain that the output of the speaker, particularly at low frequencies, depends on the electrical input to it. This is determined by the freedom of movement of the cone-coil assembly. As long as the limit of movement has not been reached either in the outer surround or the spider washer, a sine-wave input to the speaker will produce a pure tone output. When the input is so great that the cone movement limit has been reached, what emerges from the speaker is not a pure tone of the same frequency as the input signal but a mixture of harmonics of the fundamental frequency; no fundamental (because the cone can't move enough), a very large proportion of second and third harmonics, and some higher harmonics. In other words, the speaker distorts because it is working beyond its power-handling capacity. Therefore, to measure the impedance of the speaker at any frequency it must not be made to move beyond a proportion

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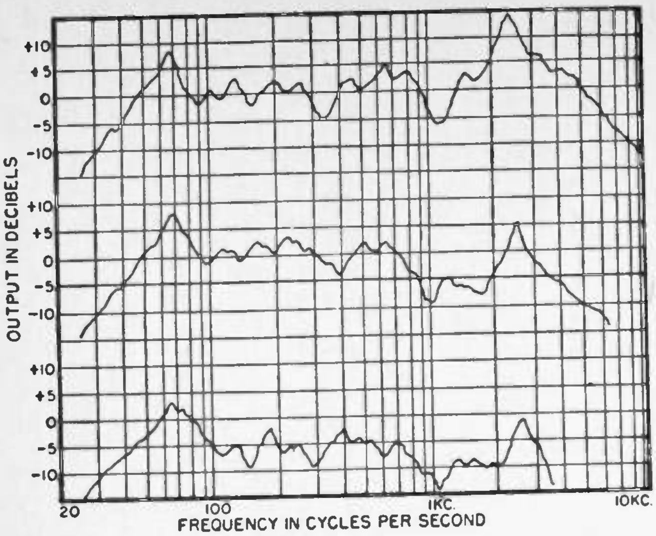


Fig. 7. Response curve of a speaker (top to bottom) on the axis and at 30 and 60 degrees off the axis. See text.

of the limit of freedom of movement; under these conditions only is it possible to observe, in impedance measurements, the effects of the cone surround and spider resonances.

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 1000 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 an automobile motor is measured on 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 isn't doing any work, and has no output to speak of. Similarly, a speaker working into a vacuum hasn't 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 to 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 best you may be sure is the place where it sounds best. Don't 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 to 100 cps, then in steps of 100 cycles up to 1000 cps and thereafter steps of 1000 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 molded composition resistor, be 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 don't 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

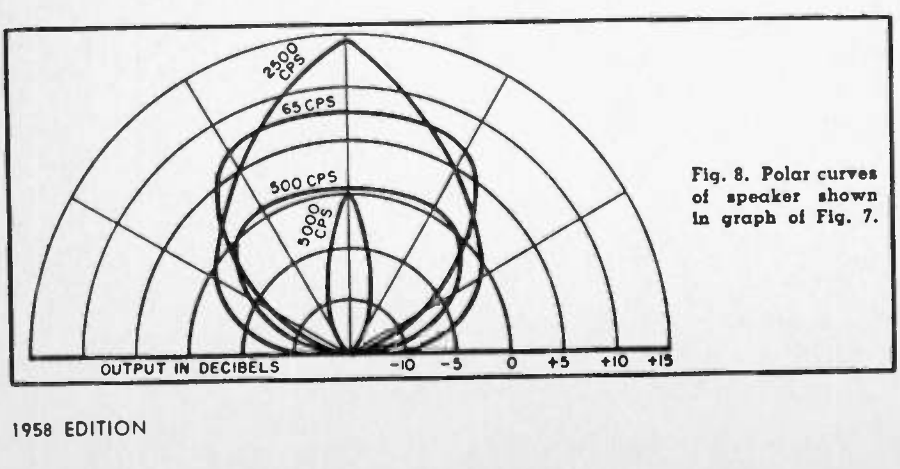
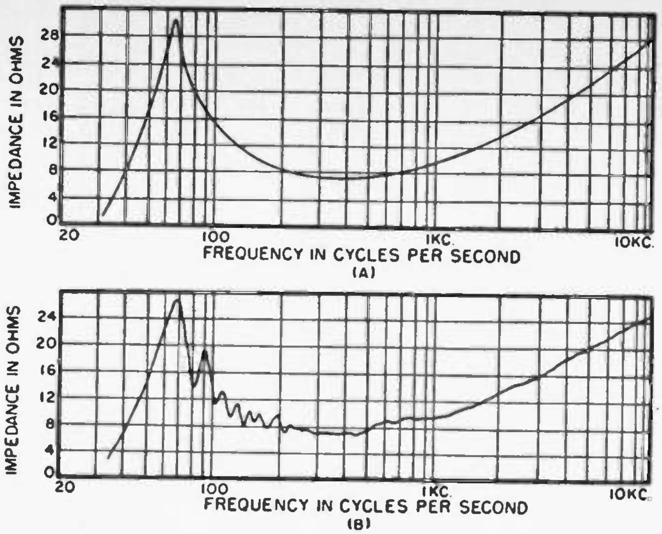


Fig. 8. Polar curves of speaker shown in graph of Fig. 7.

Fig. 9. Impedance curves of a typical speaker: (A) The customary smoothed curve showing bass resonant impedance peak and impedance rising with frequency; (B) Actual impedance curve showing major and minor bass resonant peaks and irregularities due to cone distortion and the chassis resonances. See text.



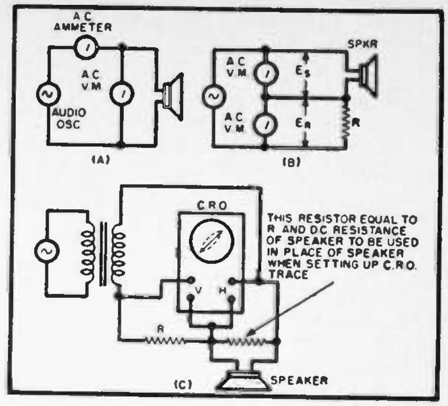


Fig. 10. Three methods of taking impedance curves of speakers. (A) Simple ammeter-voltmeter method. (B) Comparison with a standard resistor using two voltmeters, and (C) Oscilloscope method of direct comparison of resistor and speaker impedance.

internal amplifiers so that the trace is a straight line at 45 degrees 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 varies, so the trace will move from the 45 degree 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 degree position can be checked against adjustments of the 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 all sound very tedious, and I am prepared to admit that it can be a trial to one's patience, but if your desire is honest-to-goodness 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 its 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 don't, 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.

Part 4. Design factors influencing loudspeaker performance and a discussion of the simple, diaphragm type reproducers.

DESIGNING high-fidelity loudspeakers is partly a science but very much a fine art. The requisite "know-how" has to be acquired the hard way—by making mistakes; in other words, the engineers who have made their mistakes and have profited from their experiences are in a position to contribute more to the industry than a newcomer. This seemingly untechnical approach is not an attack on science or straight-thinking, it is straight-thinking since a speaker cannot be designed by mathematics. It is a known fact that only the simplest actions of a speaker can be analyzed mathematically, such as an infinitely rigid diaphragm driven by a sine wave; but since musical reproduction, in all cases, results from an extremely flexible (speaking comparatively) diaphragm actuated by very complex waves, the motion of it and the sound emanating from it cannot be computed very readily. Putting it crudely, speaker design is a sort of inspired guesswork. Certain parts of the speaker, particularly the magnetic system, can be designed with great precision, but the thing that makes the noise, the voice-coil-cone assembly, cannot be treated in this tidy manner.

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.

Strange as it may seem, the performance of a speaker can be fairly well estimated merely by looking at it, if you know what to look for. This is due to the fact that experienced designers know that such and such an assembly of materials produces cer-

EDITOR'S NOTE: *As is quite obvious, the opinions presented in this article are those of the author. Actually, there are quite a few individuals who, like the author, believe that the use of multiple small-cone speakers is preferable, for high-fidelity reproduction, to a large, single-cone unit. There are several facts that validate this thought. On the other hand, you will find many other individuals, including most U. S. speaker manufacturers, who believe the larger cone unit is the ultimate answer. They, too, can present reasonable arguments for its use. Some of these facts are obvious. Although the author has shown that, theoretically, the bass response of a small-cone speaker can be made as good as a large-cone speaker, the large voice-coil and cone excursions required present major manufacturing and performance difficulties. As a result, we do know that, considering present-day techniques, the bass response extends farther down (frequency-wise) as the cone size increases. On the other hand, the smaller speakers provide better coverage of the high end when compared with the larger single-cone speakers. For greatest frequency range, it is usually necessary to use a combination of one large-cone woofer and a high-frequency tweeter.*

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tain sounds and has certain defects (for no speaker is perfect). The wider the experience the more certain is the work of the designer, but it does not follow that a given manufacturer produces some specific model because the designer wanted it that way; the man who usually fixes the design is the sales manager, whether he is selling loudspeakers or automobiles. Fashion trends are created by advertising experts and the public trained to accept these fashions as the thing they ought to have. New ideas are created by original thinkers and competitors rush in as soon as the new feature has proved to be a seller and copy it, without having any clear notion of what they are doing. Others know full well what they are doing, but for the sake of novelty christen an accepted design feature with a new name.

I speak from direct experience of this sort of thing. Many years ago I came to the conclusion, after innumerable experiments, that the optimum cone size for an honest-to-goodness high-fidelity speaker was between 8 and 9 inches in diameter, according to the material used for the cone. That conclusion was forced on me by purely objective research. I have produced speakers having cones of this size for 25 years and they are well liked by those who have them; but my dealers tell me they could sell many more if it were a 12-inch speaker. What am I to do? Tell them, as I have been telling them for years, that if they want more power-handling capacity and more sound output they should use two or three of them or sacrifice performance to meet the requirements of a whimsical market? Actually my task now is to find out how to increase the size of the cone without spoiling the performance of the speaker, and that involves a complete redesign. At any rate, the 12-inch speaker remains a project and nothing else until I can find a way of making a 12-inch cone behave as I want it to.

The following discussion, like my past experiments, is objective. Many will disagree with what I postulate as to the requirements of good design, and it may be that I am biased because I am a loudspeaker designer. But I said in my introduction that this series is devoted to high fidelity considered as a means of reproducing music with as little equipment coloration and distortion added as may be possible; the speaker is the worst offender, and my thesis is based on genuine research into creating speakers that have no "personality," speakers that are not better nor worse than other speakers, but don't sound like speakers at all.

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.

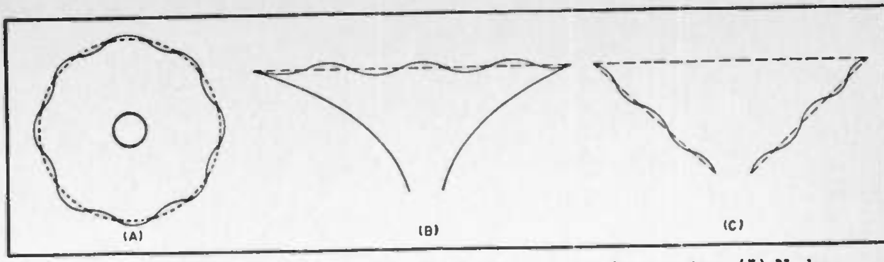


Fig. 11. The distortion of speaker diaphragms at low frequencies. (A) Nodes on a straight-sided cone. (B) Nodes on an exponentially-curved cone, and (C) wave motion along the sides of a cone. Refer to text for details on this.

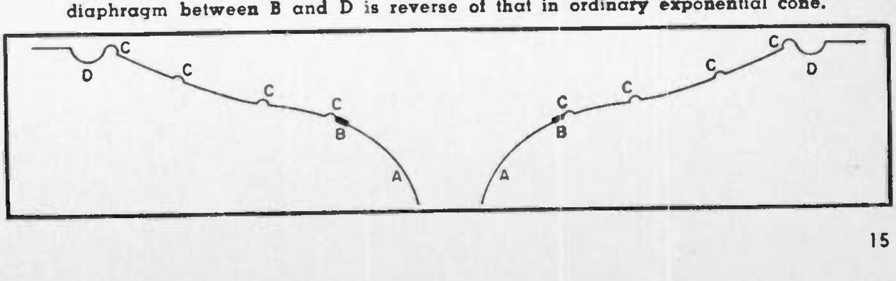
Now for a given material of a specified thickness it does not require a great deal of imagination to see that the larger the cone in Fig. 11A, the more likely will there be an inherent tendency to develop nodes. Make up two cones of the same included angle with ordinary writing paper, one having a diameter 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 reducing 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 1500 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 molded 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 (in a letter published in *Radio-Electronics,* May 1954, pp. 134 *et seq.*) explains an ingenious solution for 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 of view, would be very costly. The deformation in a straightsided 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

Fig. 12. Cross-section of the Hartley 215 diaphragm. "AA" is the exponentially curved cone apex; "BB" the isolator consisting of a plastic compliance-junction. "CCCC" are the circumferentially molded ridges on cone. "DD" is a loose flannel surround to provide completely free edge. The curvature of the diaphragm between B and D is reverse of that in ordinary exponential cone.



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 neutralized 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 loudspeaker manufacturers use them? Let us summarize 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 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 watt. 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 these articles are concerned only with high-fidelity reproduction, I maintain that what happens below 40 cps doesn't 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 inches to 15 inches are readily obtainable in any audio store. 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 your 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 preconditioned 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 in 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 next article 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 coloration 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% of rag tissue as a binder. To make such a cone in quantities by molding is an almost impos-

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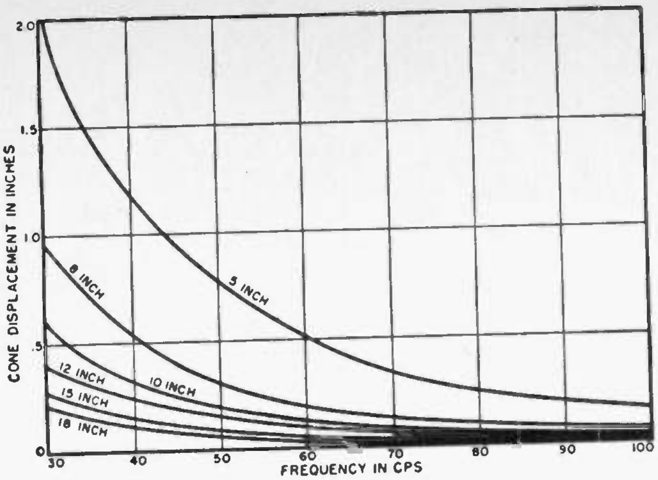

Fig. 13. Movement required of the various sizes of diaphragms at low frequencies to avoid bass distortion. Speaker efficiency is about 5%. Audio input is 5 watts.

sible manufacturing proposition, owing to the danger of the molds 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 nineteen-thirties with cones of this material, but finally abandoned it because of the manufacturing difficulties and the focusing of the highs.

Molded 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 corrugation 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 like it would 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 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 with 5 watts input because it will be overloaded. There will be no 40-cycle output, a little at 80, and most at 120 cps. This refers only to the freedom of suspension; other factors involving the magnetic system and dimensions of the voice-coil will be discussed later.

Part 5. The personal opinions of a noted speaker designer on the advantages and disadvantages of the dual-cone unit.

REPRODUCTION of frequencies up to about 2000 cps is a matter of getting the cone size, shape, and material right, as has been explained in Part 4 of this series. 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 2000 cps, but the response will be uneven and certainly deficient above 5000 cps. The difference between an ordinary speaker, such as is produced by the millions for commercial radios and TV 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 its 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

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more was achieved in the celebrated *Western Electric* 555 theater speaker, whose efficiency was considered by some to be phenomenal.

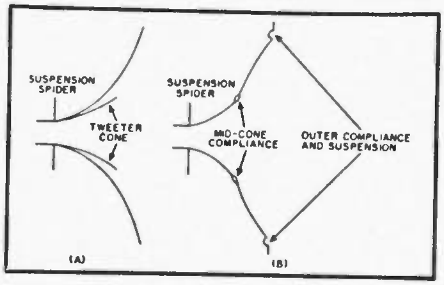
I was so impressed with the results Voigt obtained with horn-loaded speakers that I supposed, incorrectly as it turned out, it would work equally well with baffle speakers, and I obtained permission under license to use the idea in my own loudspeakers. I then discovered the limitations of the idea. The 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 fingernail, in a direction towards the center 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 density 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.

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

Fig. 14. (A) Cross-sectional view of a typical twin-cone speaker. (B) The Hartley design whereby the apex of the large cone has virtually been removed and replaced by a small cone. The two parts are joined by a small zone of flexible material.



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 neutralized 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 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 3000 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 aluminum 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 ("A New Cone Loud Speaker for High Fidelity Sound Reproduction," *Proceedings of I.R.E.*, January, 1934) 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 bypassed 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 bypassing 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 Olsen 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

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regular winding separated from a secondary "winding" by a flexible compliance, but the secondary winding was of only one turn, being an aluminum tube carrying the plastic film and the primary winding; the aluminum 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 haven't had a chance to dissect a Barker speaker so I am unacquainted with the minutiae 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 onto the plastic is useless, because the tension of the wire cannot be maintained with great accuracy. I wind the voice-coil onto a very thin paper former, slip the wound former over the plastic which is already fitted to the aluminum tube, and then expand the tube to a predetermined amount. The combination of this compliance with the mid-cone compliance already described gives a response not more than 4 db down at 20,000 cps over an approximate cone of radiation of 120°.

Field Magnet Design

It seems fairly obvious that the greater the magnetic flux in the gap in which the voice-coil works the more sensitive the speaker and the equally obvious way to get more flux is to use a big magnet. In the case of high-fidelity speakers it isn't 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 gap and this involves the smallest possible clearances between the voice-coil and the walls of the gap. Such small clearances are only practicable when the permissible movement of the voice-coil is small, since it is almost 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.

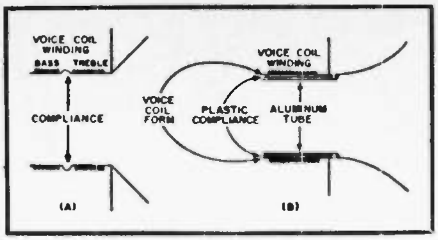
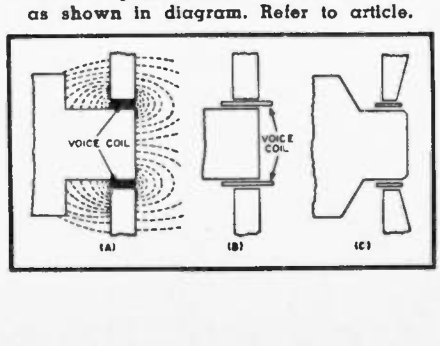


Fig. 15. (A) How two voice coils are connected in series, separated by a flexible compliance. (B) A composite voice coil, developed by A. C. Barker, which consists of a regular winding separated from a secondary "winding" by a flexible compliance. The secondary is an aluminum tube. See text.

Skilled loudspeaker designers know this and have compromised on cones having diameters of from 10 to 12 inches, but reproduction of 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 centered 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.

Fig. 16. (A) A section of a magnetic gap with the voice-coil the same length as the gap. (B) The voice-coil in this case is twice as long as the gap. (C) Method for obtaining field symmetry by extending the center pole to stick out beyond the front plate which must be modified as shown in diagram. Refer to article.



Someone once called this the "Doppler" 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 Doppler effect with speakers, a fact I have demonstrated to many electronics 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 1000 cps signal. The two frequencies are heard separate and distinct, with no variation in the pitch of the 1000-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 can't 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 schoolboy 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 electromagnet or permanent magnet has a *magnetomotive force* which provides the flux in the magnetic circuit. The permeability of the center 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 center-pole and various parts of the whole magnet system. Reluctance is reduced by increasing the diameter of most of the center-pole, as shown in Fig. 16, for by doing so an improvement of something like 50% in

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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 center-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 neutralize 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.

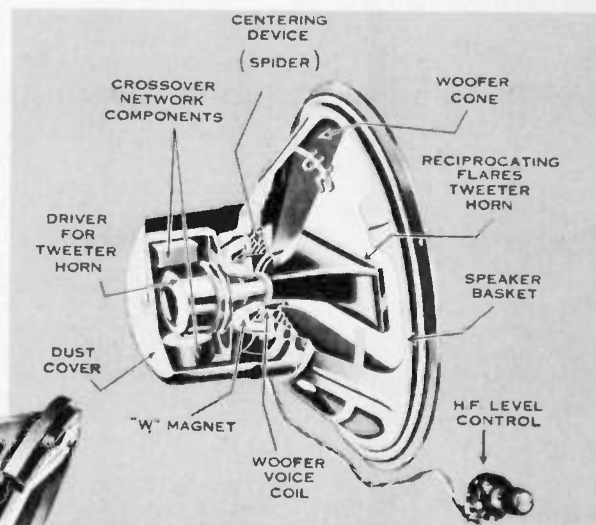
In my search for a method of producing a symmetrical field, I had no alternative but that of making up a large number of experimental magnets and exploring their fields with a very shallow search coil connected to a fluxmeter. I do not think there is any other way of doing it, and it is extremely tedious. Owing to the bulk of the center-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 at some loss of sensitivity. If the center-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.

Application

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 system 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 center pole or front plate, then reducing the input slightly. Now, from another signal source, apply a 1000-cycle signal and listen carefully. If the 1000-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 wouldn't 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.

Part 6. A leading advocate of the single-cone, wide-range loudspeaker discusses the pros and cons of coaxial units and other multi-channel systems using crossover networks.



The University Model 6201

Cutaway of Jensen's G-610 Triaxial unit.

THE term "multi-channel speakers" as I am using it in this article means loudspeaker systems consisting of two or more units whose electrical performances are separated by some sort of filter circuit; the separate units may be entirely separate speakers or mounted in a true or partial coaxial manner. They may be baffle mounted or horn loaded or a mixture of these general types. In an earlier part I have mentioned that one school of thought is opposed to this principle of speakers dealing with only part of the audio spectrum, and it is important that the validity of these objections be considered in an unbiased manner.

First is the case of systems wherein the component speakers are constructed from different materials, such as a woofer with a paper cone and a tweeter with an aluminum cone. It is, of course, well known that materials capable of emitting sounds impart their own particular coloration to the music, such as wooden and metallic wind instruments and organ pipes. It is undesirable for a loudspeaker to impart any coloration to the reproduction at all, since it is a transducer, not an originator of sounds. In the absolute sense, therefore, it is impossible for a speaker with a metallic diaphragm to have exactly the same sound as a speaker with a paper cone

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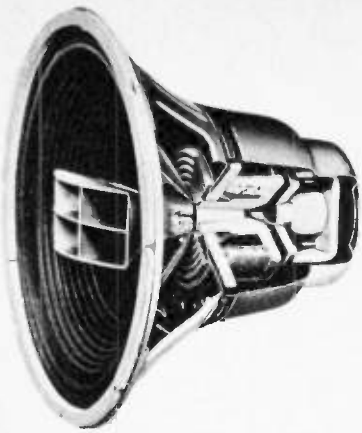


Cutaway of Jensen H-222 coaxial.

and, in practice, it is quite difficult to make two such disparate speakers sound alike at any frequency. The use of the dividing network, which keeps the bass in the woofer and the treble in the tweeter, tends to mask this fundamental difference between the two sorts of coloration, but it certainly does not get rid of it.

Now, any speaker has over-all coloration 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 coloration 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 disparate diaphragms cannot maintain constant coloration 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. At a higher frequency perhaps the whole gamut of fundamental and harmonics will be reproduced only by the tweeter. It follows that the characteristics of the reproduced note will vary with frequency, simply due to the different colorations of the woofer and tweeter diaphragms, and a sensitive ear will not like it.

What I have just written cannot be proved in a scientific sense and a convincing argument could, no doubt, be made for a quite different point of view. Why should paper be a better diaphragm material than aluminum? The instruments of the orchestra do not contain either paper or aluminum in their sound producing parts, so a transducer using either of these materials will not reproduce with absolute precision. That I have already admitted. But is there any reason why both units should not be fitted with aluminum diaphragms?

The designer's decision can only be based on what he hears or thinks he hears. If we admit that a speaker cannot be *entirely* designed by mathematics and scientific know-how, then the designer must either be a musical expert himself or get the opinion of some musical expert to listen to his various prototypes and state which one is the nearest approach to the original sound, the original sound being there also for immediate changeover comparison. When I did my basic research many years ago, the considered opinion of the experts I hired was that paper seemed to impart less coloration than other materials. And, as I have pointed out in part 4 on page 16, the method of imparting a finish to the paper has an appreciable effect on the quality of reproduction. It must be quite obvious to discriminating listeners that some speakers now on the market must have been designed by, at best, purely technical methods. I have listened to demonstrations when the salesman has stood seemingly lost in rapt wonderment at the sound produced by his wares, when the sound, in a musical sense, was excruciating. When I have ventured to point out that it didn't sound very good, I would be assailed by an avalanche of technical data supposed to prove to me that I didn't know good reproduction when I heard it. On the other hand, a study of the designs of the more esteemed manufacturers shows a certain similarity in method and in this consistency of thought is a reluctance to use metallic diaphragms.

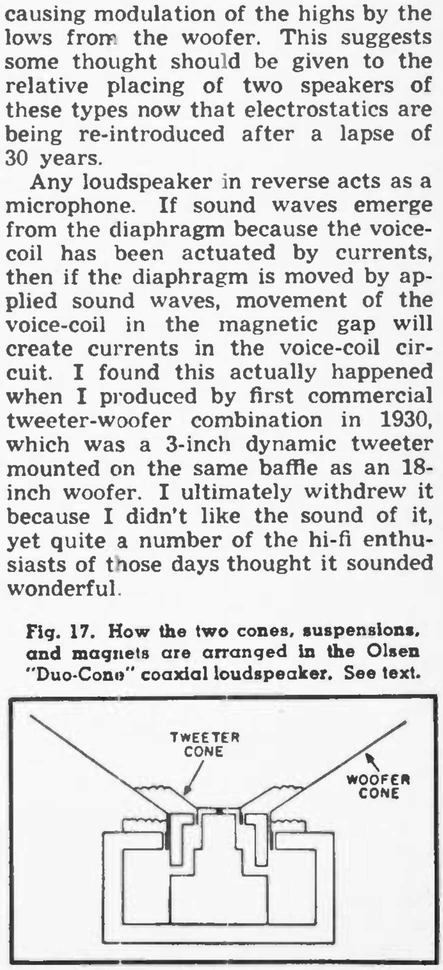
I venture to suggest, therefore, that in a multi-channel speaker system it is logical to assume that the diaphragms should be of similar material and in any event not metallic, which everybody knows has a ringing quality; our diaphragms should be as inert as possible. What is just as important is the 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 coaxial 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 coaxiality 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 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 center 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.

Any loudspeaker in reverse acts as a microphone. If sound waves emerge from the diaphragm because the voice-coil has been actuated by currents, then if the diaphragm is moved by applied sound waves, movement of the voice-coil in the magnetic gap will create currents in the voice-coil circuit. I found this actually happened when I produced by first commercial tweeter-woofer combination in 1930, which was a 3-inch dynamic tweeter mounted on the same baffle as an 18-inch woofer. I ultimately withdrew it because I didn't like the sound of it, yet quite a number of the hi-fi enthusiasts of those days thought it sounded wonderful.

Fig. 17. How the two cones, suspensions, and magnets are arranged in the Olsen "Duo-Cone" coaxial loudspeaker. See text.



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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, coaxiality is achieved. Discussion of this, however, is best left until I deal with horn-loaded speakers in general in a later article, so we can resume our discussion of existing coaxial speakers.

An original and ingenious attempt to resolve the problem of maintaining similarity of cone material with coaxiality 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.

The more popular type of coaxial loudspeaker consists of a small horn-loaded tweeter built into a normal woofer. The *Jensen* ingeniously uses a bored out center 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 center pole to avoid saturation. The *University* coaxial 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 center pole; the outer tip with its face plate represents the normal magnet outer pole. It should be realized that it is not of any consequence where the mass of magnet casting is located; the outer casing can be an unmagnetized casting or pressing and the magnet forms part of the center pole, fitted with a separate tip machined to size (since high permeability magnets are 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 magnetic circuit being completed by an iron or steel center 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 recognized 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.

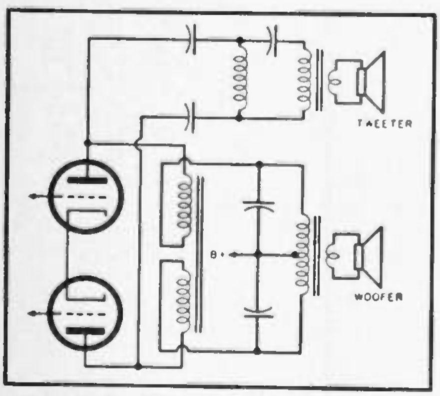

Fig. 18. The Klipsch low-distortion dividing network for use between the output stage and the output transformer. Refer to article for a discussion on network.

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 coaxial 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 in part 4 on page 15 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 1500 to 2000 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 1000 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 800 to 1000 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 1000 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 1000 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 5000 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, 1000 to 5000 cps. The small tweeter then has no problems of power-handling, for diaphragm movement above 5000 cps is almost microscopic; it can be designed specifically for what it has to do—reproduce the extreme treble.

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A smallish, say 4 or 5 inch, ordinary dynamic speaker can be used for the range 1000 to 5000 cps, and the woofer looks after the bass. Unfortunately such an intermediate unit is too large to be mounted in a 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.

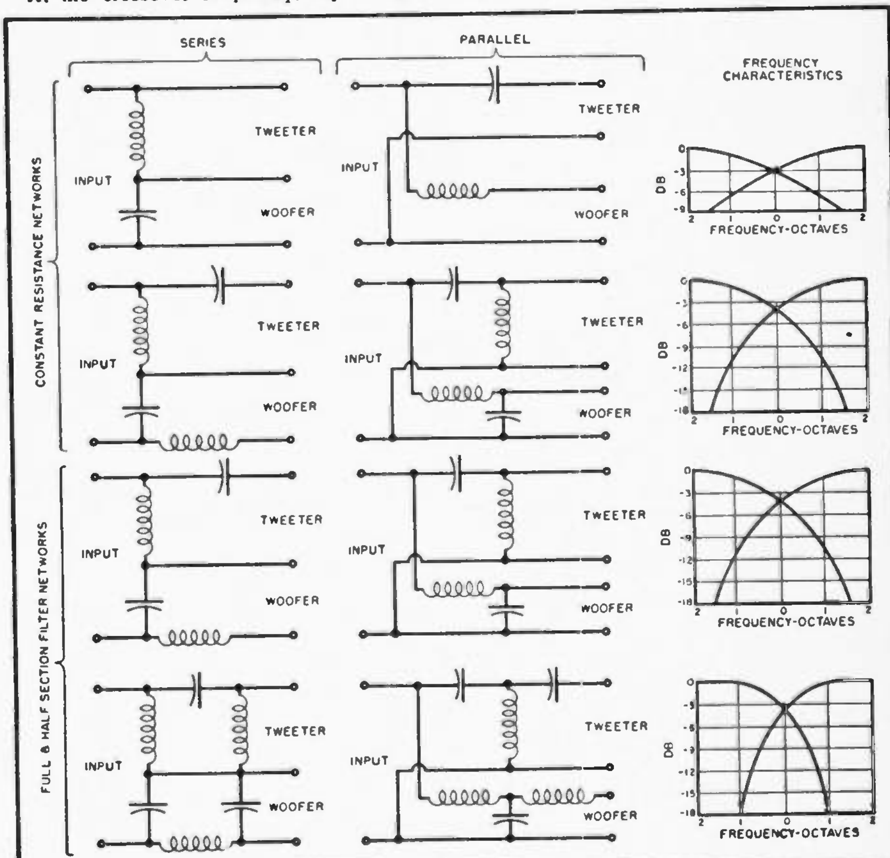
Crossover Networks

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, thus reducing production costs. But it is important to realize that constant resistance networks only have constant resistance when the loads across the output (speaker) terminals are pure resistances, and loudspeakers are not pure resistances; they have inductance as well.

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.

Fig. 19. Various dividing networks having attenuations of 6, 12, and 18 db per octave. At the crossover frequency, dip in linear response should be between 3 and 4 db.



Performance

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 stage, output transformer, and speaker. There is a great deal in favor 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. But who am I to tell my readers what is good for them? At any rate I am telling you, dear reader, now; but I wonder how long it will be before there is a big enough market for us to revert to the sensible methods of Rice and Kellogg in 1926, who marketed their speaker complete with its own suitably designed power stage.

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Part 7. A discussion of loudspeaker baffles and enclosures and the important role they play in realistic reproduction.

SOME aspects of audio engineering, such as amplifier design, can be done with precision. Speakers, as the last few articles in this series 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 series 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 nonlinear characteristics as another with a less perfect performance. The defects of the speaker might neutralize 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 behavior 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 the seeming 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 neutralize 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?

Flat Baffles

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 realized: you can't 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 didn't get it. Of course, they got the bass, but they got other things they didn't 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 harmonic by bending across two axes dividing the baffle into three equal zones, and so on. Harmonics up to the 7th 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.

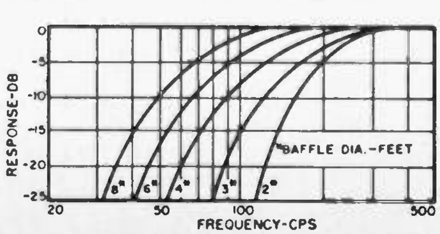

Fig. 20. Base loss of finite circular baffles.

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 article. 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 center of the front of the speaker, round the baffle, and on to the center 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 center 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 center 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.

Folded Baffles

The only truly satisfactory flat baffle is the time-honored 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 on 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 Fiberglas or similar sound absorbing material. See Fig. 22.

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

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door may cause undesirable reverberation effects, which will be considered more fully in the next section.

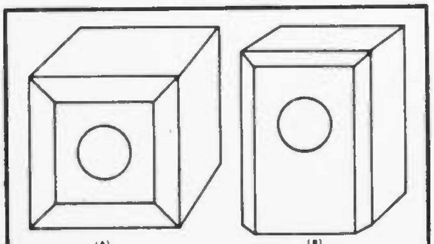
The flat baffle can be made more compact by being folded into a box. As before, the effective baffle size is the distance from the front of the speaker, across the front, along a side and on to the back of the speaker. As far as pure baffling is concerned it is of no consequence whether the box type 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.

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 center 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.

Fig. 23. Deeply chamfered cabinet edges give smoother bass response. See text.



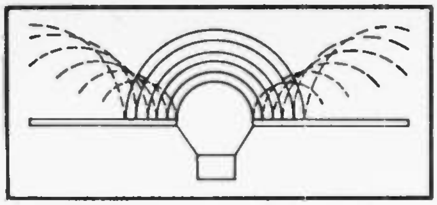


Fig. 21. Waves radiating from speaker are reflected from baffle, causing interference.

Hartley "Boffle"

I describe this specialized enclosure 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.

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 semi-transparency 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 (*"Radio-Electronics,"* February, 1956) with interesting consequences. Designed for my own speakers, I did not suppose it would be much favored for housing speakers that normally require a reflex enclosure for neutralizing 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.

Closed Box Baffles

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

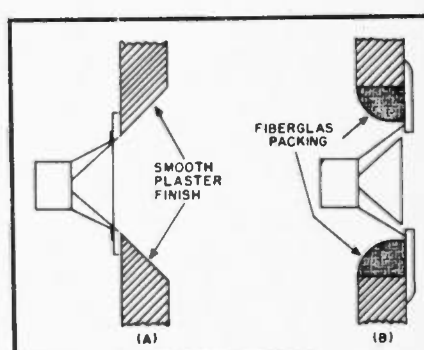


Fig. 22. Mounting a speaker in a wall. (A) Speaker on small baffle on the far side of the wall from the listening room. The hole in the wall should be funnel-shaped, smoothed off with plaster. (B) Speaker mounted in the wall. Front baffle covers a hole twice the diameter of the speaker. The hole can be circular or square but must be "radius-ed" off with Fiberglas or similar material. In each case the treatment is needed to avoid "tunnel effect."

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physicist did not invent it, but he did analyze 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 3rd 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 this series, 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 3rd 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 "free-er" 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 large 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.

Vented Box Baffles

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 in Part 1 of this series, 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 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 center line) is one quarter of the wavelength of the bass resonant frequency of the speaker. The wavelength, of course, equals the speed of sound (1129 ft. per sec.) divided by the frequency in cycles-per-second. Under these conditions the air column resonance will neutralize 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 over-all balance.

Acoustic Phase Inverter

The most popular vented enclosure is that usually called the bass reflex, due originally to A. L. Thuras (U. S. Patent No. 1869178, July 1932). 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 describes 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 resonance 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.

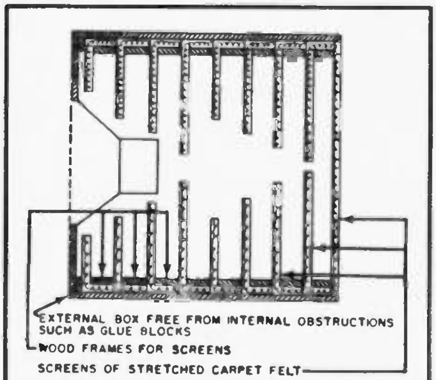

Fig. 24. Cross-section of Hartley "Boffle."

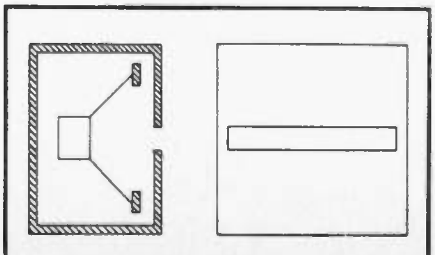

Fig. 25. Section and front of R-J box baffle.

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 speaker 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

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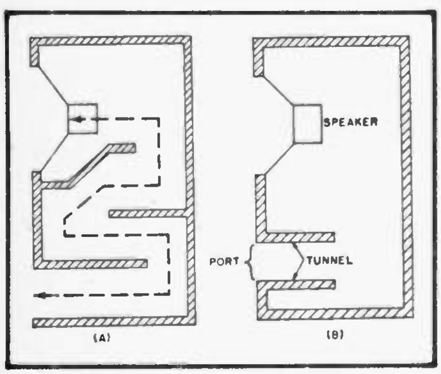
area of the port should equal the area of the speaker opening (or more accurately the area of a circle whose diameter equals the diameter of the equivalent piston). If there are errors in design the system can be "tuned" by altering the port area, but doing so conflicts with the "equal area" condition.

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 in Part 3 of this series of articles.

If you had previously taken a curve of the unenclosed and unmounted speaker from 1000 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. 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.

Fig. 26. (A) Section of an acoustic labyrinth. Length of air column is taken along dotted line. (B) Cross-section of acoustic phase Inverter. All interior surfaces must be lined with a sound absorbing material.



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 improperly, but if you want the undoubted merits of this type of enclosure to improve your audio reproduction, then you can't expect to get them by hit and miss methods, otherwise there would be no need for engineers at all. If you don't 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 explained in an earlier part. You might be lucky!

Unmatched Vented Baffles

Such 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

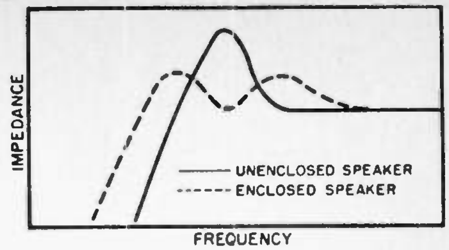


Fig. 27. Impedance curve of an ideal acoustic phase inverter compared with curve of an unenclosed speaker. Refer to text.

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 bass resonance, but if adjustment of the port to give a reasonably flat impedance curve involves raising the resonance of the system to something on 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 neutralize 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.

Part 8. A discussion of straight, folded, and corner horns used to load tweeters and mid-range drivers as well as load and enclose woofers.

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 previous articles 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 multi-channel systems. 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 innate 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 dollars. But it so happens that designing a good horn is not all that easy, and making it can be even more difficult.

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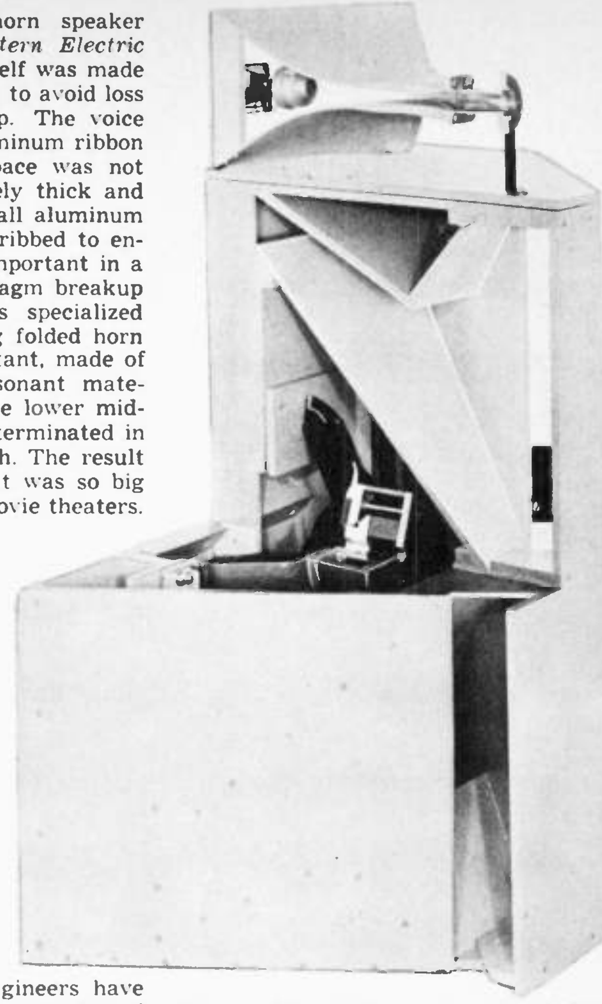
Here I shall explain the fundamental rules of the game, so that you can make your choice in an intelligent manner. But whereas there may be two schools of thought in speaker design, there can only be one in horn design, for the matter is simple enough —does the horn enclosure add distortion to the speaker 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 living 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 don't 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 coloration.

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 aluminum ribbon on edge, so that gap space was not wasted by a comparatively thick and rigid former, and the small aluminum diaphragm was properly ribbed to ensure stiffness (for it is important in a horn speaker that diaphragm breakup should not occur). This specialized unit then fed into a long folded horn with the right flare constant, made of smoothly finished nonresonant 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 movie theaters.



Cutaway model Klipschorn corner loudspeaker system showing a folded horn used for the woofer and straight horns used for mid-range and tweeter.

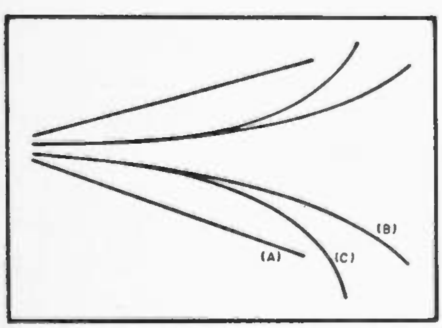
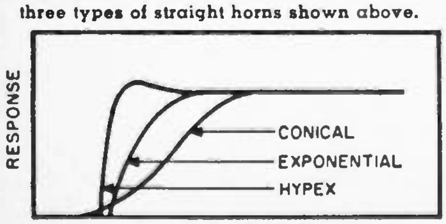


Fig. 28. The three types of straight horns: (A) conical, (B) exponential, and (C) Hypex.

Fig. 29. Cut-off characteristics of the three types of straight horns shown above.



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 couldn't be done 30 years ago. The mechanics of horns are perfectly straightforward and we can't do the impossible "even if it takes a little longer." Our efforts have been directed towards producing speakers that can be gotten into an ordinary living 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.

Design of Straight Horns

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.

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 sharpest.

It is not difficult to understand why this should be so. In Part 1 I explained that a sound wave progresses through the air by setting up zones of compression followed by zones of rarifaction.

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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 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 rarifaction 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 doesn't 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 series 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. And may I add a note about other wonder-working curves announced from time to time? We get paraboloids, catenoids (and some day we may get adenoids) all heralded as new achievements. Don't you believe it. These others "curves" are so near exponential that it couldn't matter less, and except for molded 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 monsters, with most impressive results. Of course the horn has to be built outside the house, so it is not very convenient for multi-story apartments. But it does show what has to be done to carry the horn to its logical conclusion.

The Horn Throat

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 center 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 2nd 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 2nd 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 in Part 4) 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 thesis that baffle-loaded speakers are best as single-channel systems is right or not, I cannot be accused of undue partisanship if I say that multi-channel systems are best with horn speakers.

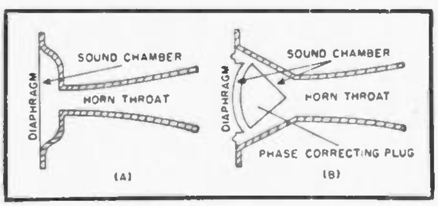


Fig. 30. Two designs of throat sound chambers without and with phase correcting plug.

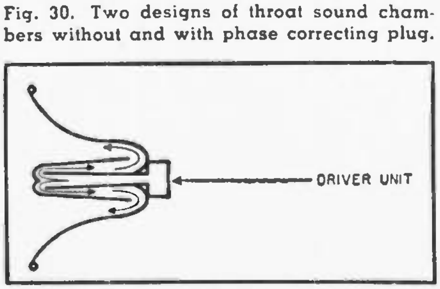


Fig. 31. Section of concentric folded horn.

Folded Horns

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 molded 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 8000 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 pos-

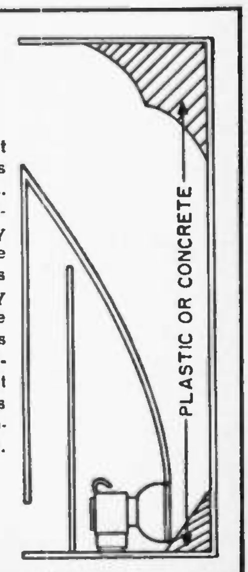


Fig. 32. Voigt corner horn as used in early 30's. A tuned air column, driven by the rear of the cone, is used as a supplementary resonator for the lower frequencies since the loading on the front of the cone is not effective below 50 to 60 cps.

sible 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.

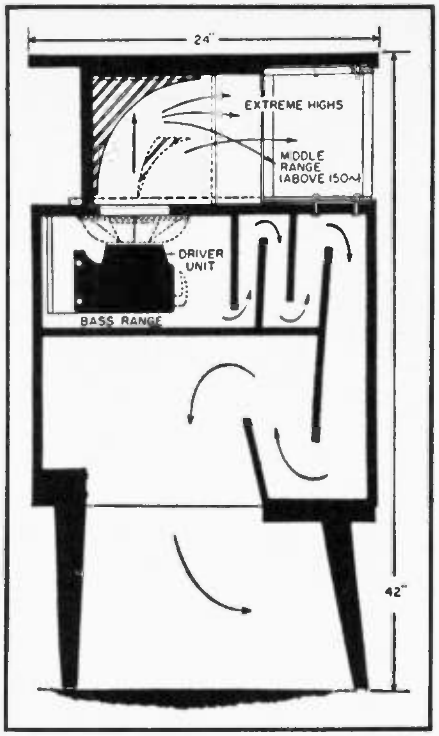
Since, therefore, the folded horn is only an approximation to the ideal design there is almost no end to the ways in which this approach to perfection can be achieved. Reputable manufacturers of speaker units have been forced to produce horn designs which are suitable for their products, and it can be 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 emphasized—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 of 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. And 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 doesn't matter if such results came about through blind chance. What really matters is that the results are there. I have pointed out in an earlier part that speakers can't be designed by mathematics alone, nor, for that matter, can cabinets. Marrying the two is best done by practical experiment.

Corner Horns

There is some dispute as to who first thought of a corner horn, by which I mean a device which uses the adjacent walls as part of the horn system. Sandeman, in U. S. Patent No. 1,984,550 of 1929, 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 to 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

Fig. 33. Cross-section of Brociner Model 4 horn showing curved front horn for highs and mid-range and long folded rear horn for lows.



been superseded, at least in the U. S., by the *Brociner* housing shown in section in Fig. 33. This design is unusual in so far as it postulates a single wide-range speaker unit with no separate tweeter.

The Klipsch Horn

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 won't 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 won't do that. A number of corner horn outfits have a much higher crossover frequency, and you may well pause to consider if this is good practice.

What Is the Answer?

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 crossover 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 color. That is what you want from your speaker, so at no point in the frequency range should there be any 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 won't 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 harassed owner of an audio store, but I don't know any other way of finding out how a complete speaker assembly and its housing will behave in your own listening room.

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Part 9. Output stage characteristics — including power requirements and loudspeaker matching.

THOSE parts of this series 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 literally color the reproduction. As your editor pointed out in a note, 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 it was inevitable that I should feel that my way was the best way, otherwise I wouldn't 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 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 in a single article, and when I wrote this part and read it through I had to tear it up because it did not answer 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 favor of triode output stages and another equally insistent on tetrodes or pentodes, something more was needed than just another resume 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 word in perfection. Writing an article on an amplifier 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 can't 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 don't know any more about amplifiers than others, but I have found out where most of these didn't match up to my requirements, and it is that knowledge I shall try to give you. This article, therefore, will deal with the approach to the problem; the next will constructively criticize the various types of hi-fi output stages so that you can make your own selection.

How Much Output Power?

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 civilized home, he will lose business. It isn't 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 rubbernecks, wise guys who don't know much about music, but reckon they know a lot about hi-fi. They dash from one room to another, listen for a minute and off to the next, rather like the traditional Yankee doing a three-day tour of Europe. 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 reproduction. In due course these people will report to their friends that the company in room A doesn't 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 the order book. If he gets the orders, he is doing the right thing; if he doesn't, he isn't. And that seems to be all there is to it.

But there is more to it, for this unfortunate state of affairs has preconditioned the audio fan into assuming that hi-fi and hi-volume go hand-in-hand, and that is not only bad for your neighbor 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 program, 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 tympani, 4 brass bands and a choir of 300, then it doesn't 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." And 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.

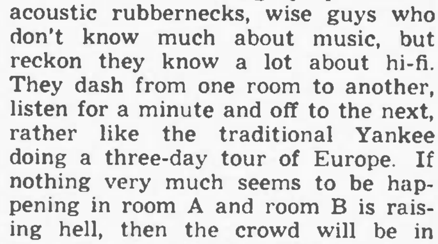

Fig. 34. Hookup for impedance measurement of loudspeaker. See text for details.

If you are a regular concertgoer 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 in an earlier article, 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 2500 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

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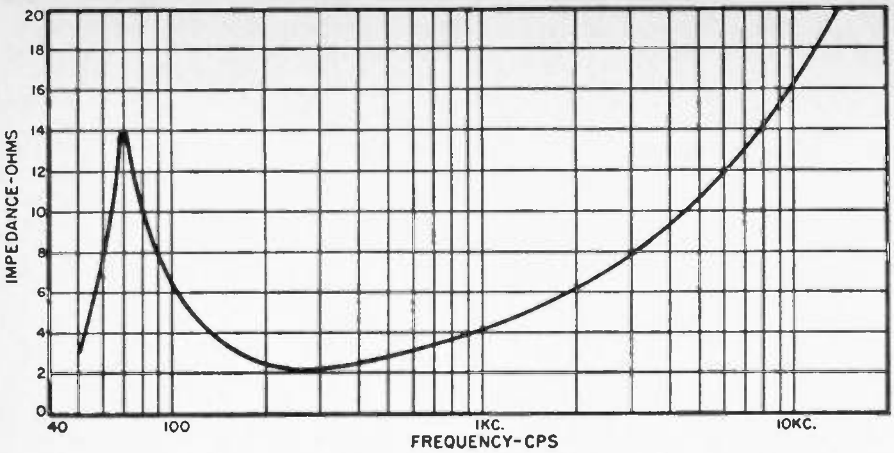

Fig. 35. Impedance curve of a typical 4 ohm loudspeaker with a bass resonance at 70 cps. Eight and 16 ohm speakers would have proportionate variations of impedance.

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 clichés. 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:

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

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.

Output Transformer Characteristics

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 onto 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 series. 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.

Loudspeaker Characteristics

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 (Part 4) gives some information on this; it indicates that for speakers of 5% efficiency with a free movement of cone and coil of ¼ inch (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 ¼ inch. 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 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 series, 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 is the question—what is the impedance of the speaker? It is not the figure quoted by the manufacturer, for it varies widely with frequency. Quoted speaker impedances 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 in 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 a speaker's construction and behavior shall be interpreted in terms of inductance and capacitance. Pure resistance does not

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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, then the normally quoted impedance at about 500 to 1000 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 is 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 noninductive 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:

$$\textit{Impedance of speaker} = \frac{R \times E_S}{E_R}$$

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 and in the extreme treble. It has been my experience that, since all frequencies are equally important, such a speaker should be considered to have an average impedance of 8 ohms. You will notice that the rise in impedance is much greater in the extreme treble than 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 tetrodes or pentodes 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 seemingly 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." But as you can now see, 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 in the next article; for the moment, my suggestion of doubling the nominal impedance will give much better general quality.

Negative Feedback

There is a good deal of misapprehension as to what negative feedback can do. In later articles the practical application will be discussed in a technical way; for the moment I shall summarize 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*, β. Obviously β 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 β can be expressed in db. If we call the amplification of the amplifier A, without feedback, then the amplification after feedback is $A/(1+\beta A)$. You may find this formula in textbooks with the sign in the denominator negative, but if it is negative feedback then β 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 less 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 it is used so frequently, but now creeps in a very serious liability.

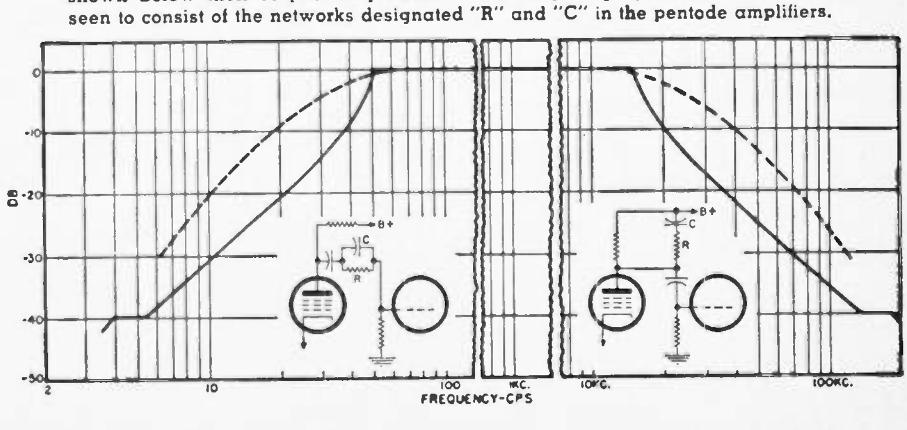
Change of phase occurs in every tube and every RC coupling. With triode output stages more amplification is needed than with tetrodes or pentodes, perhaps even to the extent of having to provide an extra stage to do it. At any rate the phase change in a multi-stage amplifier can become so progressively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

Fig. 36. The ideal curve of a multi-stage amplifier to give a level response from 50 to 15,000 cps with 30 db of negative feedback and a safety margin of 10 db (to guard against instability following heavy transients) is shown by the solid line below. 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 spectrum. These are seen to consist of the networks designated "R" and "C" in the pentode amplifiers.



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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 basic 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 to any frequency you like, but 1000 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 note 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 won't 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 can't heighten the trace. In other words, you can't 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 fed-back distortion cleans up the wave shape but when maximum undistorted power is reached and further input or volume (gain) is applied there will be a *very slight* increase of height with feedback and the waveform 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.

The characteristics of the power output stage will be discussed more fully in the next article. For the moment it need just be pointed out that negative feedback, by reducing the effective plate resistance of the output tubes, provides a more favorable load impedance/plate resistance factor, thus improving the speaker damping factor and making tetrodes and pentodes more like triodes without impairing their better power efficiency.

The other point to be mentioned here is that the power output figures given in the tube manuals presuppose the use of a pure resistive load, and the load lines drawn on the characteristic curves are straight lines. The distortion can, therefore, be computed by simple graphical methods. But the load formed by an output transformer and speaker is not a pure resistive load but a reactive one. This has to be shown on the curves as an ellipse. The implications of this may not be quite clear to you for the moment, but it does mean that a reactive load reduces the undistorted output from the stage as compared with that obtained with a pure resistive load

Performance Requirements

All these factors I have described have to be taken into account when considering the accuracy of my 3, 6, and 11-watt postulates. These are peak undistorted outputs and the mean output will be considerably less than that. But we have no room for distortion on peaks when providing realistic sound reproduction; it must be undistorted all the time. To provide a safety margin to take care of all the extraneous sources of distortion the simplest way is to use a bigger output stage and the margin I would suggest is 100%. So my figures for the three types of speakers first mentioned are, roughly 7, 15, and 25 watts.

The danger is that with the extra power you may be tempted to run the volume a little louder than necessary and get distortion on peaks. Many equipments sound bad because they are overdriven, not because they are inherently bad.

Part 10. Practical recommendations on selection of tubes and circuits to be used.

Before discussing the sorts of tubes that will give undistorted outputs of from 7 to 25 watts, I should like to make a "heretical" observation. We all know that negative feedback greatly improves an amplifier, both as to damping factor and distortion, and so feedback, used wisely or otherwise, is written into the constitution. Associated with this is usually a specification of fantastically low intermodulation distortion for the output the tubes are supposed to give, according to the tube manuals. These claims may be quite right, and I will agree that they can be proved right, but it calls for pretty good engineering, and tubes right on the top of their form. Now comes my heresy, for speaking as an engineer I ask "is this pretty good engineering absolutely necessary?" The reply can be that as a matter of satisfaction to the engineer it is necessary, and if a manufacturer has to produce the maximum results at the minimum cost it is also necessary and that is why he hires engineers to do the design work. But quite a lot of people like to build their own amplifiers, and I should like to emphasize that the semi-technical amateur is really up against it when he tries to get the maximum possible from a given output stage.

It is much easier to design roughly for twice or three times the ouput you would normally require, use just a little feedback (to avoid any possibility of low-frequency instability) and keep the volume down, for your roughly designed amplifier will sound pretty bad if going all out. Put it this way—design for a 50-watt amplifier and call it a 25 and don't try and get more from it. Under these conditions it would have to be a fairly bad amplifier to sound other than good at ordinary room level.

Later on in this part I shall give the circuit of a 20-watt output stage of an amplifier I made and sold for some years. I don't make it any more, for I have enough to do with speakers alone, but that amplifier was produced in 1948 in response to an insistent demand for something better than usual. It was most carefully designed, and

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its 20 watts fed into one of my ordinary 215 speakers were enough to fill lecture theaters with an audience of 200. But . . . it was designed for home use, and I suppose in the average home its peak output was on the order of 5 watts. Then it sounded very good indeed, and I am told even today that it sounds cleaner than many highly respected amplifiers incorporating all the latest developments. There is nothing mystically wonderful in the design; it is just working well within its possible maximum output, even when the output tubes are getting tired after many years' use.

This, however, is by the way, and no fit subject for engineers and would-be engineers. Let us seriously consider the selection of tubes for output powers of 7 to 25 watts. Seven watts can be obtained from a single tube, but the advantages of push-pull operation are so manifold that it is invariably used in high-fidelity amplifiers.

Selection of Tubes

The first decision is, therefore, that two tubes will be used in class A or AB push-pull, but there is no general agreement as to which tubes should be used, or how they should be used. It is an easy matter to skim through the data sheets in the tube manuals and note that some particular pair of tubes gives x watts at y% distortion, but to depend on this is rather like visiting a foreign country with no knowledge of the language beyond what you will find in a phrase-book. That will enable you to get around in a limited sort of way, but you won't get the utmost pleasure from your vacation abroad. So, if you want the best amplifier, you must learn a little of the language of electronics. Only then can you judge critically the various circuits in the textbooks and magazines and the amplifiers shown to you in the dealer's store.

Some of what follows has formed the basis of many articles and textbooks over the past 30 years. I am sorry to have to put it in, for diagrams of tube characteristics have been repeated *ad nauseum,* but you must know how to get the right answer. I have cut out all the padding and got down to the skeleton; but the skeleton may have more bones than you thought. I think the comparative tables will enable you to determine the best tubes for your particular requirements without having to refer to other texts, and as some tubes *are* better than others, you should know wherein they are better.

The tubes considered are of two main types, those with plate or plate plus screen dissipation of 12 or 25 watts. Note the distinction between plate dissipation and output power; dissipation indicates the power consumed (d.c.) within the tube itself to produce audio power (a.c.) to drive the speaker. The ratio between the audio output power and the total d.c. input power, including the power dissipated within the tube, is the efficiency of the stage. This is rarely greater than 30-50%, so two tubes each of 12 watts dissipation are usually needed for 10 to 12 output watts and two 25-watt tubes for powers up to 25 watts. Higher efficiencies can be obtained by special techniques, which will be described in some detail later in this article.

Most people seem to want greater outputs than 10 watts, so my treatment of characteristics curves will be restricted to 25 watt tubes, but the same basic principles apply to the smaller ones.

As to tubes that are available, it is a surprising thing that very few *are* available. In the old days British manufacturers had a wide range of triodes ranging from 12 to 250 watts plate dissipation, and some of them were very good, consistent, and reliable. Most of these have disappeared since the advent of the beam power tetrode. In the U. S. the 2A3 (now available in standard octal base as the 6B4G) is hardly ever used, and it is a sardonic commentary on audio usage that whereas the beam tube was developed to give greater stage efficiency, it is now *mis-used* by being triode connected to overcome the disadvantages of beam tubes, real or alleged, simply because the beam power tube has supplanted the triode. This development is not only illogical, it is also technically unsound, for the highly competent designers of tubes know how to design tubes for a specific purpose. The beam power tube is designed to be used as a beam power tube, a triode as a triode.

One real advance in output tube design is found in the Dutch *Philips-Mullard* EL34 and the American *Tung-Sol* 6550 pentodes. These are intended for "distributed load" or "Ultra-Linear" operation, but I am quite sure it won't be very long before some writer comes out with an amplifier design using these very same pentodes as triodes.

In Table 1 I have displayed side-by-side the characteristics of the more popular tubes. The parameters of single tubes are given so that you may assess the basic attributes of each type, while the power output and distortion of push-pull pairs gives you the data you require for various plate supply voltages. The table shows some interesting facts.

The KT66 made its appearance in the U. S. as a result of publicity given to the Williamson amplifier circuit. This circuit did not have any novel features, and the elaborate balancing controls included only contributed to getting the maximum undistorted output from an output stage of somewhat limited capabilities. There are other, and simpler, ways of getting the desired results. I have had my own opinion of the circuit—lack of stability—confirmed by many competent engineers, but it "caught on" because at the time it was described it seemed to behave better than many contemporary American amplifiers. In the process the tube specified by the author came to be endowed with special properties, but reference to the table (which has been compiled from the data issued by the manufacturers) shows that the 6L6 is a rather good

Table 1. Data on 19-35 watt tubes, used singly (A) and in push-pull pair (B). For triode operation, screen grids are tied to plates through 100-ohm resistors.

	6L6		KT66			EL34		6550	
Connected as:	Pentode	Triode	Pentode	Triode		Pentode	Triode	Pentode	Triode
Plate volts	250	250	250	250	400	250	375	250	....
Screen grid volts	250	....	250	....	....	250	....	250	....
Cathode bias res. (ohms)	170	490	160	315	600	120	370	....	....
Peak a.f. grid volts	14	20	15	19	38	13.5	18.9	14	....
Transconductance (μmhos)	6000	4700	6300	....	5500	11,000	8000	11,000	....
Load res. (ohms)	2500	6000	2200	2750	4500	2000	3000	1500	....
Plate res. (ohms)	22,500	1700	22,500	1450	....	15,000	....	12,000	....
Distortion (%)	10	6	9	6	7	10	8	7	....
Power output (w.)	6.5	1.3	7.25	2.2	5.8	10	6	12.5	

(A)

	6L6		807	KT66				EL34					6550			
Connected as:	Tetrode		Triode	Tetrode		Triode		Pentode	"Ultra-Linear"		Triode		Pentode (self-bias)	Pentode (fixed bias)	"Ultra-Linear"	Triode
Plate volts	270	360	400	250	390	250	400	375	430	430	400	430	400	400	450	450
Screen grid volts	270	270	...	250	275	....	....	375	425	425	....	....	300	275	450	....
Peak a.f. volts (grid to grid)	28	41	90	36	70	40	80	59.5	45	77.5	58	64	53	46	96	92
Load res. (ohms, plate-to-plate)	5000	9000	3000	4000	8000	2500	4000	3400	6600	6600	5000	10,000	4500	3500	3300	4000
Distortion (%)	2	4	3	4	6	2	3.5	5	.8	1.3	3	<1	4	3	2.5	2.5
Power output (w.)	18.5	24.5	15	17	30	4.5	14.5	35	20	37	16	14	41	55	70	28

(B)

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match. It should be remembered that the 6L6 was not designed to work on plate voltages as high as 400, so its power output is somewhat limited; but the 807 has the same characteristics and can be used safely up to 600 volts. If you *must* build a Williamson amplifier then the tube to use is an 807. As an amplifier manufacturer in Britain before World War II I had experimented with precursors of the Williamson circuit and decided then the tube for me was the 807. In regular production, however, I used 6L6's as pentodes, for even 20 years ago it was well known by engineers that feedback over several stages was liable to introduce very-low-frequency instability through phase-shift. High performance amplifiers were best arrived at by using as few stages as possible, and this implied the utilization of the beam power pentodes. Feedback took care of the high output impedance.

Another suitable tube that should be mentioned at this point is the 5881. This can be considered to be a single-ended 6L6 with about 20 per-cent more plate dissipation. Plate voltages as high as 400 volts may be employed.

Table 1 also shows that as compared with the 6L6 or the KT66 the new power pentodes are a great step forward, either as pentodes or triodes. The transconductance is about twice as good as the prewar beam tubes, which means higher efficiency as an input voltage—power output converter. It also means greater power output for a given plate voltage. It would be unfair to say that the EL34 is better or worse than the 6550, for everythings depends on the way it is used. The EL34 usage is described by the *Mullard Co.* as of the "distributed load" or "Ultra-Linear" type. To get the best out of these newer tubes involves special techniques, which will shortly be described.

The power output figures of Table 1 are without negative feedback. As I pointed out in Part 9, feedback cannot increase the power output of a tube but it can reduce the distortion. The figures in Table 1 for distortion can be reduced with feedback, but you cannot get more power output. If you use feedback, you will find that up to the figure given in the table, the power output for a pair of tubes contains less distortion with feedback, but the moment you go past that figure, the distortion with feedback will be greater than without. It is a characteristic of negative feedback that up to the optimum point of power output the gain in quality is substantial, but the moment you cross the threshold the results are terrible. The overload point is reached very suddenly.

If you fancy KT66's in push-pull triode operation you can't get more than 14½ watts at the maximum rated plate voltage advised for the tube. You can get more with 807's as triodes because the tube will just about stand 600 volts as a triode. Or you can use 6550's. But if you really want substantially higher powers than those shown in Table 1, then you must use four tubes in push-pull parallel. If you don't want to do that, then you must give up the idea of having a triode amplifier.

Output tubes with 12 watts plate dissipation are detailed in Table 2. The 6V6 you all know, and a pair is used in almost every 10-watt amplifier on the market. The KT61 is quoted because it is used in an imported British amplifier, although the tube was never designed for hi-fi amplifiers. Its very high transconductance is explained by its intended use immediately after the detector diode of a mass-produced radio set. It gives reasonable power output of a quality associated with such equipment. It does this very successfully, but is surpassed by the *Mullard* EL84, as can be seen at once by looking at the data for push-pull pairs. What Table 2 does show is that the triode addict is very badly served, for he cannot get more than 6 watts from a pair of any of them. The only 12-watt solution I know is the 25-year-old British PX4 triode, for which there is no American equivalent, but the tube is fitted with a British 4-pin base, has a 4-volt filament, and so calls for non-standard tube sockets and power transformers. A pair gives 13.5 watts with 2.5% distortion.

The two tables list nearly every output tube that can interest us (I did not include the 5881, a tube which I like, for its characteristics are similar, if a little better, than the 6L6). But we must also face the fact that these figures are for average tubes with a resistive load. In practice, the inductive load of a speaker system will result in less output for a given distortion or the same output for more distortion, although the discrepancy will not be as great with triodes as with tetrodes. But for a clearer idea of what really goes on inside a tube recourse must usually be made to the tube's characteristic curves as published in many tube manuals.

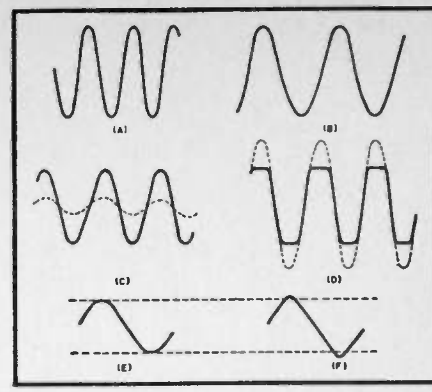


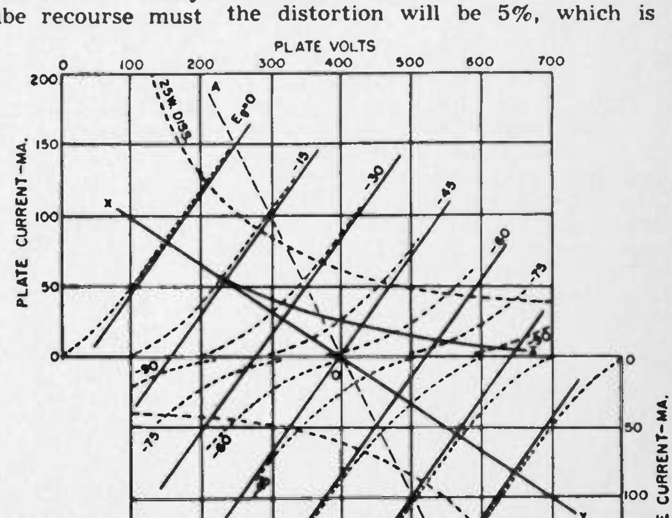
Fig. 37. Oscillograms illustrating the effects of negative feedback.

Characteristic Curves

I have no wish to use up a lot of space on this matter, for the material has been printed over and over again, but a few notes will not come amiss. Fig. 38 shows the plate volts—plate current curves of a 6L6 connected as a triode, the sort of curves you see in the tube manuals. To these I have added a 25-watt dissipation curve to indicate the limit of the tube's handling capacity. The load line *XOY* has a gradient representing 2500 ohms, and to prove this I have extended it dotted to cut the voltage and current axes; you will notice it cuts the horizontal axis at 500 volts and the vertical at 200 ma., and 500 volts/0.2 amp =2500 ohms.

If the distance *OX*, representing a signal swing from —15 to 0 volts on the grid, equals the distance *OY*, which represents a signal swing from —15 to —30 volts, there will be no second harmonic distortion. If these two distances are not equal, there will be. If the ratio of *OX* to *OY* is 11 to 9, the distortion will be 5%, which is

Fig. 39. Composite characteristics of a pair of 6L6's in class A push-pull. The point "O" is the operating point, through which passes the basic composite characteristic labeled "—45 volts" and marked as a solid line. Other composite characteristics are shown as solid lines. The load line XOY represents a load of 3000 ohms as recommended in Table 1. For the meaning of the load line AOB, representing a 10,000-ohm load, see text.



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usually reckoned a suitable figure for second harmonic distortion (but 5% of third harmonic would sound very bad indeed, our ears being what they are!). You can mark off a rule of stiff paper, starting 1,2,3 etc. from a center point O, eleven millimeters apart on the left and nine millimeters apart on the right, which will enable you to pick the right load line position for 5% distortion, by putting the center zero of the rule on the mean grid voltage curve. Swinging the rule about point O on the 6L6 curve will indicate that you can't dodge that 5% distortion anyhow. If you moved the operating point O from the —15 volt curve to the —30 volt curve you only need to look at Fig. 38 to see that the distortion would become excessive owing to the crowding of the characteristic curves. And before I go on to the next step, remember that precisely the same rules apply to the curves of the tube when used as a tetrode. The curves have a different shape, but the "family" of curves is subject to the same considerations.

But, of course, you have heard that push-pull operation cancels out second harmonic distortion. Why should it do this? If you imagine the load line *XOY* bent upwards in its lower half so that it runs roughly parallel to the dissipation curve, you can also see that it would cut the grid voltage curves in a much more regular manner, and so reduce distortion. This is what, in effect, happens with push-pull operation, and is illustrated by Fig. 39, which shows two 6L6's in class A triode push-pull. This diagram is simply two Fig. 38's joined back to back, with the 400-plate-volt ordinates (the operating point) coinciding. The dotted families of curves are those of Fig. 38, but new solid lines (composite "curves") take up positions exactly between the corresponding pairs of curves of the two single tubes. These solid lines are the characteristics of the combined pair, and are cut by the load line *XOY*. This is the equivalent of the load line *XZ* shown in the upper half of the diagram, being that imaginary curved load line I mentioned before. I have also drawn the load line *AOB*, which will give a greater power output than *XOY*, but this particular value was chosen to show a considerable trespass into the region greater than 25 watts dissipation, so the tubes would be overloaded on large signal inputs.

In practice the position of the load line affects the distortion at third and fourth harmonics as well as second. The amount of distortion of each harmonic can be calculated from data on the diagram, but the easiest way to check this point is to refer to the distortion charts given in many tube manuals, where distortion harmonic and total is frequently given for any value of load line with related power output.

When the load is a speaker, as it must be if we are to get any sound from the audio power generated, the load is partly resistive, partly inductive, and partly capacitive. In practice, the capacitive component is not usually considered important (although capacitances are shown in so-called equivalent circuits of loudspeakers). The reactance of the inductive component depends on the frequency and combining the resistive component with the reactance involves consideration of the phase of the reactance. The inductive load line is elliptical, and the major axis can be considered to coincide with the zero current ordinate of Fig. 39. Combining this with the pure resistive load line will produce an elliptical load resultant whose major axis is at an angle with respect to the original major axis, but not at the same angle as the resistive load line. To explain how this comes about would involve a fairly long mathematical discussion, but a little reflection will indicate that at certain frequencies the load on the output stage will be seriously out even if the slope of the major axis has been set for minimum distortion. As tetrodes and pentodes are more critical as to exact loading for maximum power output than triodes, the bias against the former has resulted in the fairly widespread belief that only triodes can be considered for really true fidelity.

Triodes, Tetrodes and Pentodes

This belief is founded on misapprehensions as to what is involved. The accuracy of load for tetrodes and pentodes is only of importance when the maximum possible "undistorted" power is needed, and early in this part I have suggested that for normal listening fewer watts than many people say they want is all they really need. On the other hand it is undeniable that the high plate impedance of tetrodes and pentodes does not give a good damping factor, and if feedback hadn't been invented, then the triode would be absolutely essential for the best results. But feedback can reduce the plate impedance of a pentode to a figure comparable with that of a triode —provided you remember my warning that feedback is not a cure-all for every amplifier ailment.

A triode amplifier can sound very good indeed without any feedback at all. It can be made better with feedback, but triode amplifiers as well as those using beam power tubes can be badly designed and made to sound respectable by adding as much feedback as possible. In the process low-frequency instability can be set up, due to the excessive phase shift caused by several voltage amplifier stages, for the basic disadvantage of triode output tubes is that they need plenty of volts applied to the grid to get the maximum output power. The performance of a well designed triode amplifier without feedback can be paralleled by using a pentode output stage with feedback across the output stage only. I made such amplifiers more than 20 years ago, when the beam power tube was still a novelty, and they sounded pretty good.

Also, using output tetrode or pentodes, the smaller amount of voltage amplification required usually saves one stage in the whole amplifier, so if feedback is applied across the whole amplifier, more can be used with pentodes than with triodes because of the smaller phase shift. I state in all seriousness that it is possible to design an amplifier with pentode output tubes with at least as refined a performance as one with triodes and, provided certain steps are taken to deal with the pentodes in a special way, the results can be better than any triode amplifier, for with at least one stage fewer the distortion over the whole amplifier can be less and the stability greater.

Practical Amplifier Circuit

Fig. 40 gives the circuit layout of my own amplifier referred to earlier in this part. It can be shown that if the screen voltage on a beam power output tube is held constant, whatever the input, the distortion is much less. If you examine the data sheets for the 6L6 you will find that curves are given for various values of screen voltage. The plate characteristics alter as screen volts are changed. At the same time you will find, from the tube manual data, that values are given for plate and screen current for zero signal and maximum signal. The plate and screen supplies originate in the rectifier tube and have to pass through the resistances of the smoothing choke and the output transformer primary winding. There is a voltage drop across these resistances, the drop depending on the current. Since both plate and screen currents change as the input grid volts change, the voltages on the plates and screens of the output tubes also change. Change of plate voltage is by no means as critical as change of screen voltage, and if the latter can be held constant, the almost chaotic conditions under which the output tubes have to work can be reduced to regulated order.

There are various ways by which the supply voltage can be regulated, a usual method involving a gas tube

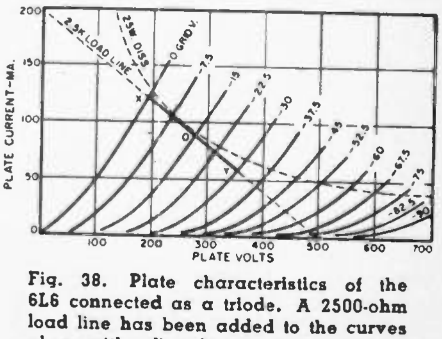

Fig. 38. Plate characteristics of the 6L6 connected as a triode. A 2500-ohm load line has been added to the curves along with a line denoting the maximum plate dissipation power of 25 watts.

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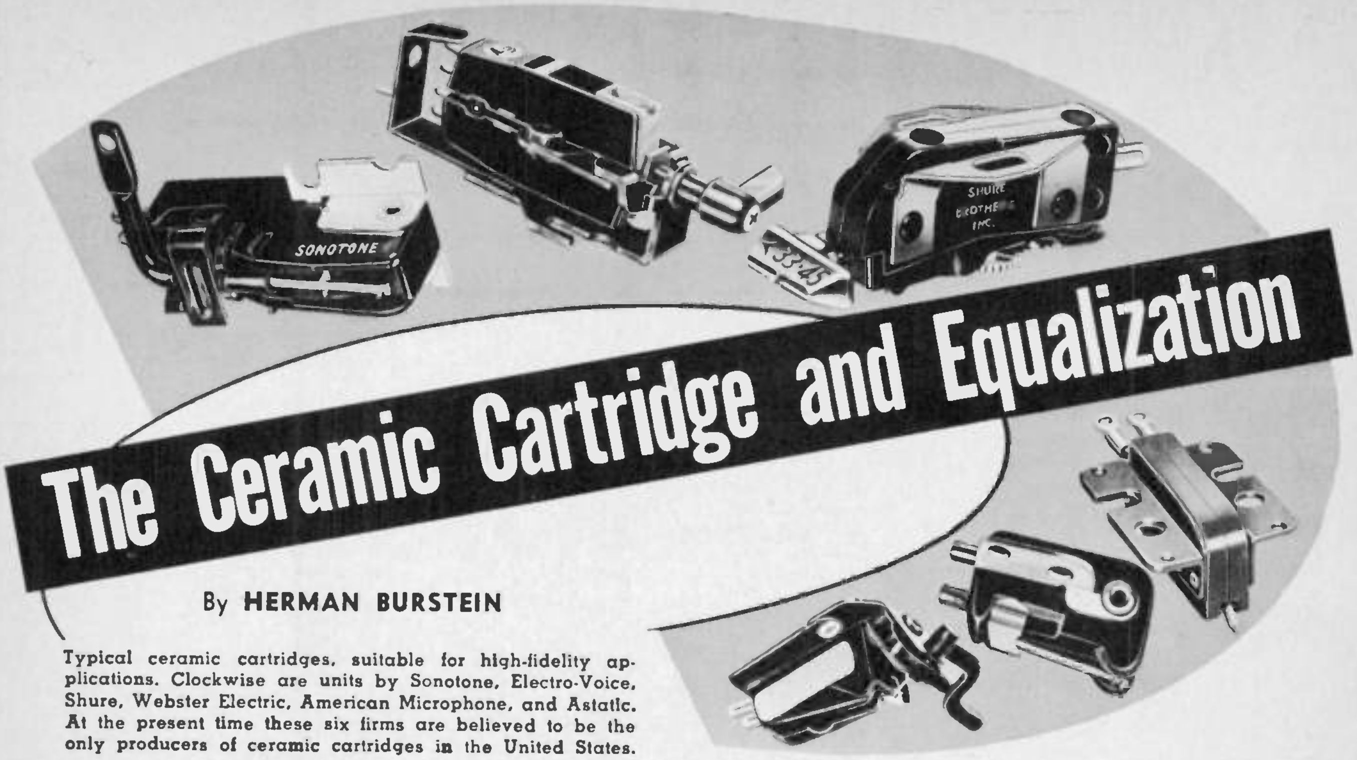

The Ceramic Cartridge and Equalization

By **HERMAN BURSTEIN**

Typical ceramic cartridges, suitable for high-fidelity applications. Clockwise are units by Sonotone, Electro-Voice, Shure, Webster Electric, American Microphone, and Astatic. At the present time these six firms are believed to be the only producers of ceramic cartridges in the United States.

Hi-fi performance is now possible with piezoelectric cartridges. Although the author's discussion is more or less confined to ceramic units, all data presented applies equally as well to crystal-type cartridges.

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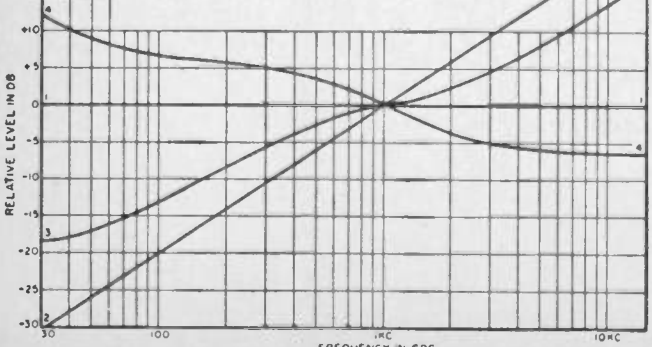
Fig. 1. The RIAA recording characteristics widely used today.



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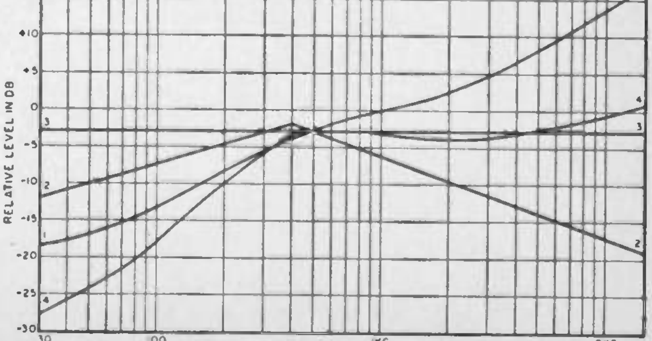

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usually reckoned a suitable figure for second harmonic distortion (but 5% of third harmonic would sound very bad indeed, our ears being what they are!). You can mark off a rule of stiff paper, starting 1,2,3 etc. from a center point O, eleven millimeters apart on the left and nine millimeters apart on the right, which will enable you to pick the right load line position for 5% distortion, by putting the center zero of the rule on the mean grid voltage curve. Swinging the rule about point O on the 6L6 curve will indicate that you can't dodge that 5% distortion anyhow. If you moved the operating point O from the —15 volt curve to the —30 volt curve you only need to look at Fig. 38 to see that the distortion would become excessive owing to the crowding of the characteristic curves. And before I go on to the next step, remember that precisely the same rules apply to the curves of the tube when used as a tetrode. The curves have a different shape, but the "family" of curves is subject to the same considerations.

But, of course, you have heard that push-pull operation cancels out second harmonic distortion. Why should it do this? If you imagine the load line *XOY* bent upwards in its lower half so that it runs roughly parallel to the dissipation curve, you can also see that it would cut the grid voltage curves in a much more regular manner, and so reduce distortion. This is what, in effect, happens with push-pull operation, and is illustrated by Fig. 39, which shows two 6L6's in class A triode push-pull. This diagram is simply two Fig. 38's joined back to back, with the 400-plate-volt ordinates (the operating point) coinciding. The dotted families of curves are those of Fig. 38, but new solid lines (composite "curves") take up positions exactly between the corresponding pairs of curves of the two single tubes. These solid lines are the characteristics of the combined pair, and are cut by the load line *XOY*. This is the equivalent of the load line *XZ* shown in the upper half of the diagram, being that imaginary curved load line I mentioned before. I have also drawn the load line *AOB*, which will give a greater power output than *XOY*, but this particular value was chosen to show a considerable trespass into the region greater than 25 watts dissipation, so the tubes would be overloaded on large signal inputs.

In practice the position of the load line affects the distortion at third and fourth harmonics as well as second. The amount of distortion of each harmonic can be calculated from data on the diagram, but the easiest way to check this point is to refer to the distortion charts given in many tube manuals, where distortion harmonic and total is frequently given for any value of load line with related power output.

When the load is a speaker, as it must be if we are to get any sound from the audio power generated, the load is partly resistive, partly inductive, and partly capacitive. In practice, the capacitive component is not usually considered important (although capacitances are shown in so-called equivalent circuits of loudspeakers). The reactance of the inductive component depends on the frequency and combining the resistive component with the reactance involves consideration of the phase of the reactance. The inductive load line is elliptical, and the major axis can be considered to coincide with the zero current ordinate of Fig. 39. Combining this with the pure resistive load line will produce an elliptical load resultant whose major axis is at an angle with respect to the original major axis, but not at the same angle as the resistive load line. To explain how this comes about would involve a fairly long mathematical discussion, but a little reflection will indicate that at certain frequencies the load on the output stage will be seriously out even if the slope of the major axis has been set for minimum distortion. As tetrodes and pentodes are more critical as to exact loading for maximum power output than triodes, the bias against the former has resulted in the fairly widespread belief that only triodes can be considered for really true fidelity.

Triodes, Tetrodes and Pentodes

This belief is founded on misapprehensions as to what is involved. The accuracy of load for tetrodes and pentodes is only of importance when the maximum possible "undistorted" power is needed, and early in this part I have suggested that for normal listening fewer watts than many people say they want is all they really need. On the other hand it is undeniable that the high plate impedance of tetrodes and pentodes does not give a good damping factor, and if feedback hadn't been invented, then the triode would be absolutely essential for the best results. But feedback can reduce the plate impedance of a pentode to a figure comparable with that of a triode —provided you remember my warning that feedback is not a cure-all for every amplifier ailment.

A triode amplifier can sound very good indeed without any feedback at all. It can be made better with feedback, but triode amplifiers as well as those using beam power tubes can be badly designed and made to sound respectable by adding as much feedback as possible. In the process low-frequency instability can be set up, due to the excessive phase shift caused by several voltage amplifier stages, for the basic disadvantage of triode output tubes is that they need plenty of volts applied to the grid to get the maximum output power. The performance of a well designed triode amplifier without feedback can be paralleled by using a pentode output stage with feedback across the output stage only. I made such amplifiers more than 20 years ago, when the beam power tube was still a novelty, and they sounded pretty good.

Also, using output tetrode or pentodes, the smaller amount of voltage amplification required usually saves one stage in the whole amplifier, so if feedback is applied across the whole amplifier, more can be used with pentodes than with triodes because of the smaller phase shift. I state in all seriousness that it is possible to design an amplifier with pentode output tubes with at least as refined a performance as one with triodes and, provided certain steps are taken to deal with the pentodes in a special way, the results can be better than any triode amplifier, for with at least one stage fewer the distortion over the whole amplifier can be less and the stability greater.

Practical Amplifier Circuit

Fig. 40 gives the circuit layout of my own amplifier referred to earlier in this part. It can be shown that if the screen voltage on a beam power output tube is held constant, whatever the input, the distortion is much less. If you examine the data sheets for the 6L6 you will find that curves are given for various values of screen voltage. The plate characteristics alter as screen volts are changed. At the same time you will find, from the tube manual data, that values are given for plate and screen current for zero signal and maximum signal. The plate and screen supplies originate in the rectifier tube and have to pass through the resistances of the smoothing choke and the output transformer primary winding. There is a voltage drop across these resistances, the drop depending on the current. Since both plate and screen currents change as the input grid volts change, the voltages on the plates and screens of the output tubes also change. Change of plate voltage is by no means as critical as change of screen voltage, and if the latter can be held constant, the almost chaotic conditions under which the output tubes have to work can be reduced to regulated order.

There are various ways by which the supply voltage can be regulated, a usual method involving a gas tube

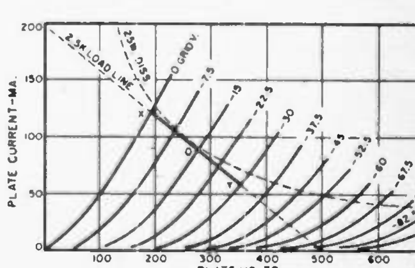

Fig. 38. Plate characteristics of the 6L6 connected as a triode. A 2500-ohm load line has been added to the curves along with a line denoting the maximum plate dissipation power of 25 watts.

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Connected as:	6V6	KT61		EL34	
	Pentode	Pentode	Triode	Pentode	Triode
Plate volts	250	250	250	250	250
Screen volts	250	250	....	250	....
Peak a.f. grid volts	12.5	4.3	5.5	6	8.3
Transconductance (μmhos)	4100	10,500	9800	11,300	....
Load resistance (ohms)	5000	6000	5000	5200	3500
Distortion (%)	8	8	5	10	9
Power output (watts)	4.5	4.3	.7	5.7	1.9
		(A)			
Plate volts	285	275	350	300	300
Screen volts	285	275	....	300	....
Peak a.f. volts (grid-to-grid)	38	16	23	28	28
Load resistance (ohms)	8000	10,000	6000	8000	10,000
Distortion (%)	3.5	6.5	2	4	2.5
Power output (watts)	14	11.5	6	17	5.2
		(B)			

Table 2. Comparative data pertaining to beam power pentodes with plate dissipations of 12 watts. Part (A) is for a single tube, part (B) is for a push-pull pair.

shunted across the supply. This operates by providing a variable leak so as to keep the total current drawn from the supply constant; but it is not necessary to regulate the plate voltage; only the screen voltage is important. In Fig. 40 I use a 6J5 tube connected in a special way so that it acts as an automatic variable resistance in series with the screen supply, and has the effect of maintaining the screen voltage absolutely constant whatever the variation in plate or screen current. A separate 6.3 volt winding on the power transformer must be provided for the heater of this tube, since the whole tube is "hot" with respect to ground.

Other refinements will be noticed in the circuit. The screen supply is taken to a potentiometer, the ends of which are connected directly to the screens. By this means the output tubes can be balanced perfectly by slight compensating changes of screen voltage. In practice all that is needed for balancing is to connect a voltmeter across the whole of the output transformer primary winding. The potentiometer is adjusted until the meter shows zero reading indicating that there is no out-of-balance current in the transformer. This adjustment takes care of almost any pair of tubes; there is no need for buying them in matched pairs, and if only one tube has to be replaced at any time, the "set zero" adjustment is all that is needed.

Each output tube is provided with screen and plate suppressor resistors, connected directly to the tube sockets, and an r.f. bypass capacitor to each plate. These form supersonic oscillation filters, for the output stage is very efficient and could oscillate at r.f. if precautions such as these were not taken. Such oscillation is not audible (although sometimes visible when smoke comes from the speaker) but it can ruin the response of an otherwise fine amplifier. Associated with these two filters are series grid resistors. Again these should be connected directly to the tube sockets. The phase splitter is the conventional split load device, frowned on by some engineers, but I have always found it to be very reliable. The plate load is shown as two resistors. These need not be separate (in production this was found convenient) but what is more important is that the total plate resistance must equal the total cathode resistance. Similarly the output coupling capacitors and following grid resistors must be equal. Their exact value is not important; they must *match*.

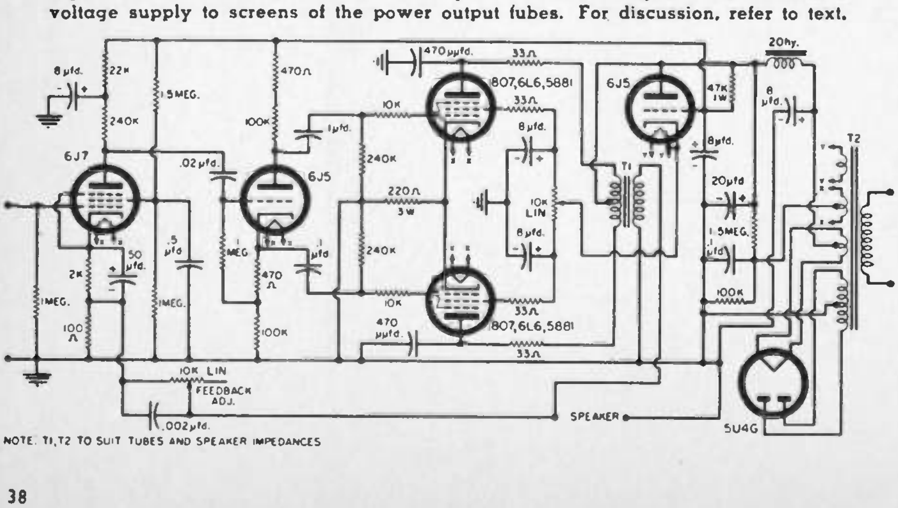
If 807's are used to get maximum power, then, owing to the heat from the tubes (which have the plate connection on the top of the bulb) and the operating conditions, the 470 $\mu\mu$fd. plate bypass capacitors have a very rough time of it and should be at least 2000 volt ceramics. Alternative tubes are 6L6 and 5881, and the r.f. capacitors can be the ordinary molded mica type.

Apart from the voltage control complication in the power supply, the circuit is very simple indeed, with the minimum number of stages. Because of this I have been able to apply as much as 28 db of feedback without running into trouble with low-frequency instability. The response is +1 db at 20 cps, −1 db at 50,000 cps and dead flat in between. Intermodulation distortion is negligible (almost unmeasurable) up to 15 watts, and only 1% at 20 watts. These figures may appear to be unimpressive, compared with some claims, but they are factual. Twenty watts is developed with an input of 1 volt r.m.s.

The now famous "Ultra-Linear" output stage was first described by Hafler and Keroes (*Audio Engineering*, Nov. 1951) and used conventional pentodes with taps on the output transformer connected to the screens of the tubes, thus distributing the load between plates and screens. This forms a hybrid sort of output stage, for the devisers of the circuit found that in altering the ratio of screen load to plate load from zero (pentode operation) to unity (triode operation) a point was reached when most of the disabilities of the pentode disappeared and the output stage took on the characteristics of triodes, without losing the high efficiency of the pentode. In a practical sense, therefore, the high power output and stage efficiency of pentodes is retained, but the tubes, from a low plate impedance and damping factor point of view, behave like triodes, and with a carefully designed amplifier, the distortion is less than can otherwise be obtained. The latest development of this circuit has been described by Hafler in RADIO & TELEVISION NEWS (June 1956) and need not be repeated here.

The design cannot be applied indiscriminately, for the amplifier as a whole has to be considered carefully. The design of the output transformer is critical, and only components specified for use with particular tubes should be used. The authors' original article suggested that a new tube was really wanted to get the best from the scheme, and such tubes are now available in the *Mullard* EL34 and the *Tung-Sol* 6550. —30—

Fig. 40. Schematic diagram of the Hartley 20-watt audio amplifier with stabilized voltage supply to screens of the power output tubes. For discussion, refer to text.



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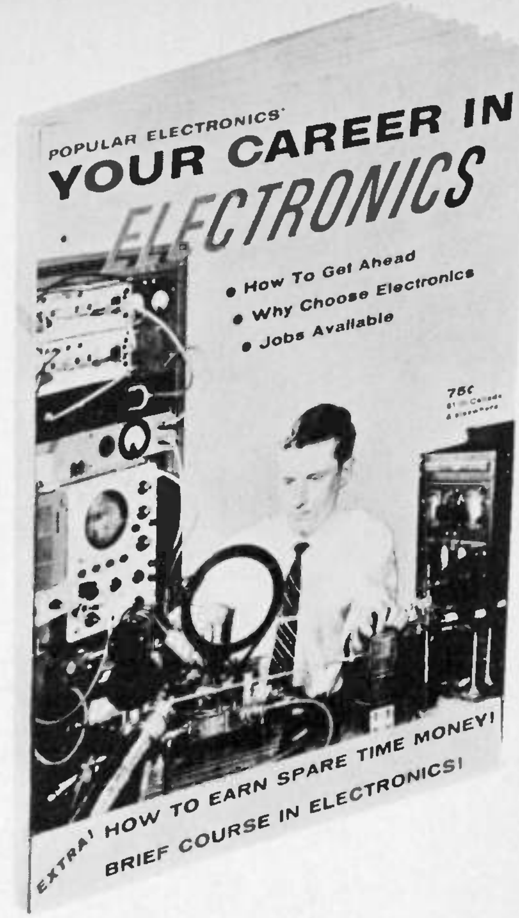

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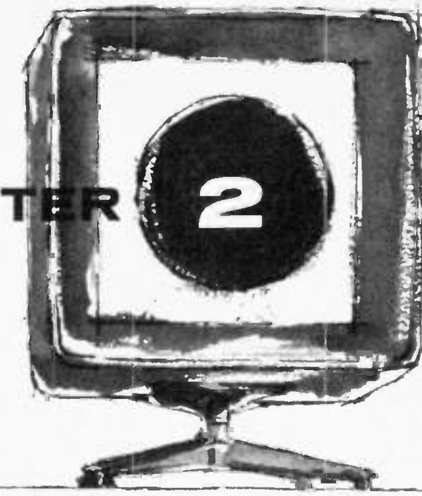

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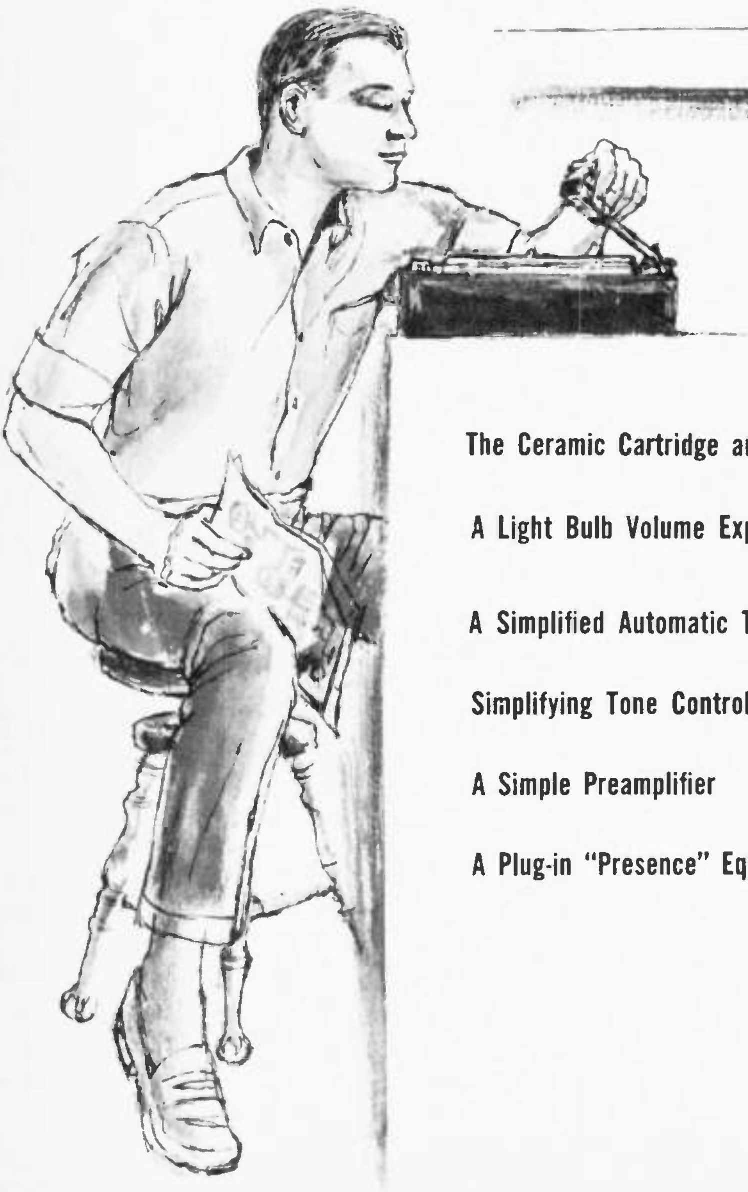

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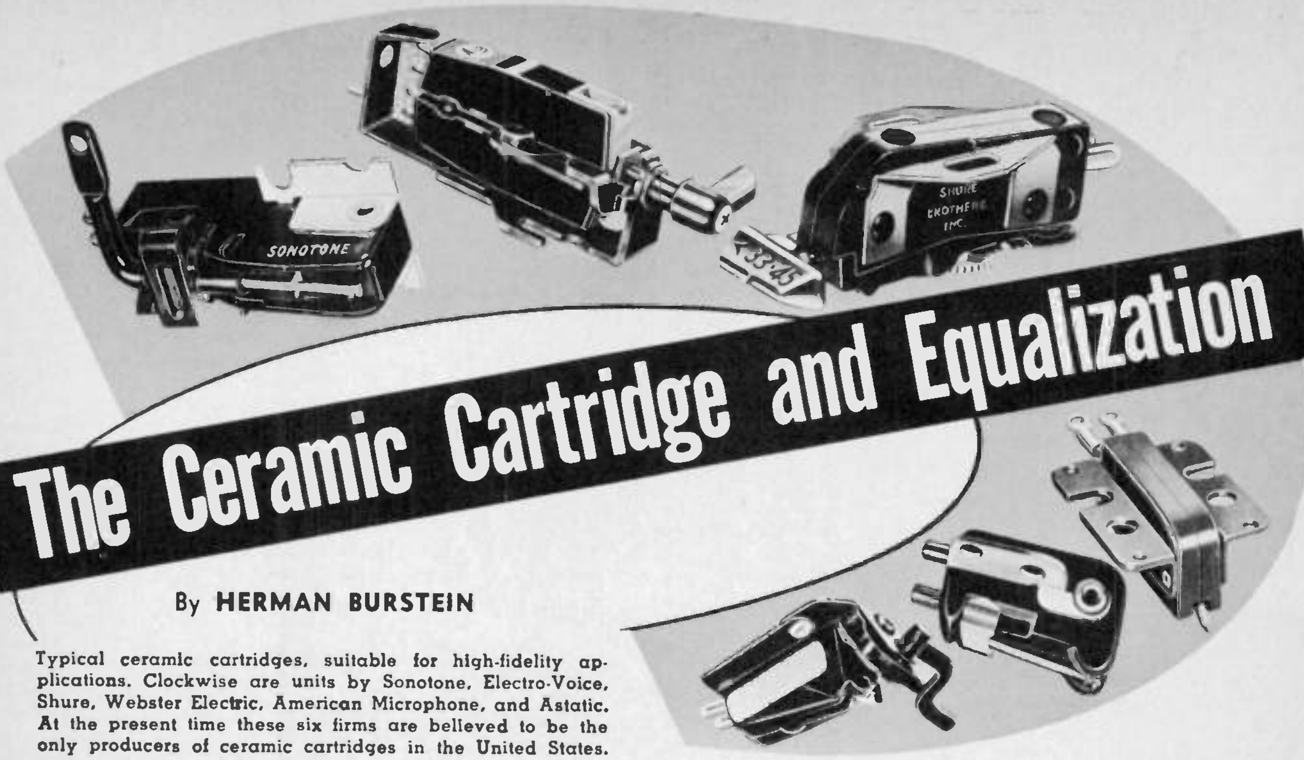

The Ceramic Cartridge and Equalization

By **HERMAN BURSTEIN**

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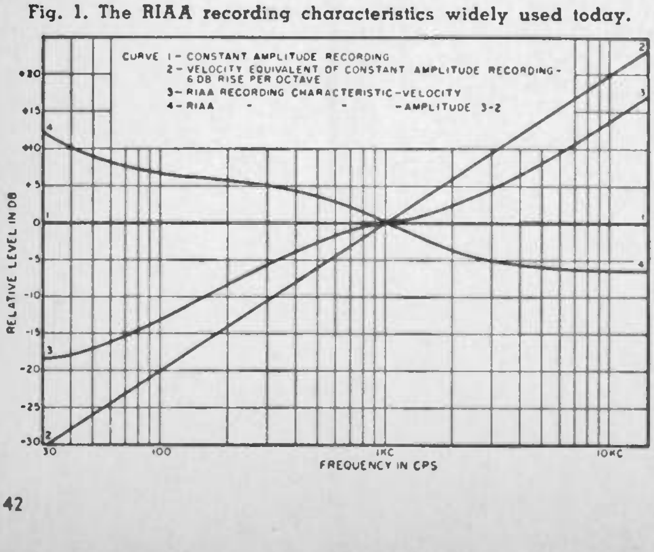
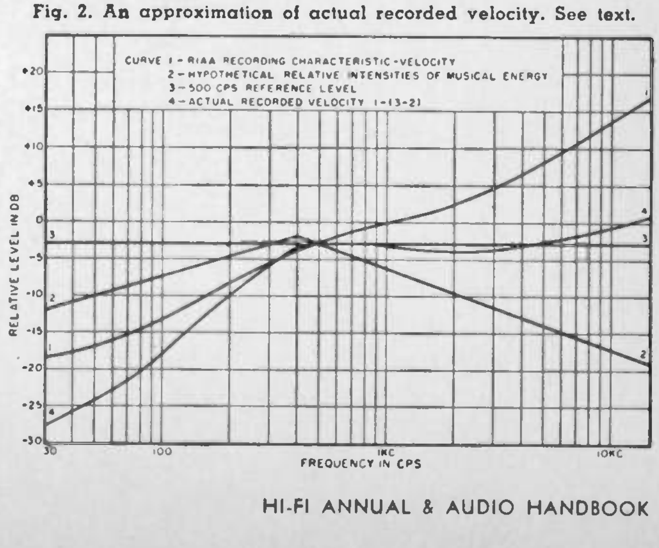


Fig. 2. An approximation of actual recorded velocity. See text.



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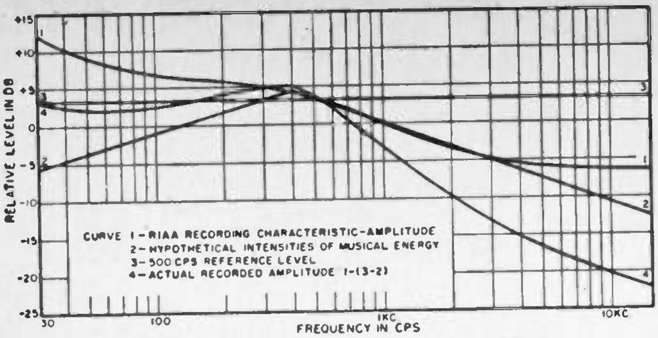


Fig. 3. Approximation of actual recorded amplitude. Curve 1 represents RIAA recording characteristic in terms of amplitude.

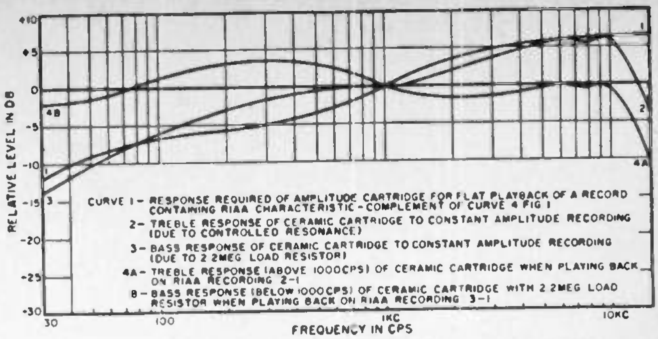


Fig. 4. Response of ceramic cartridge to RIAA recording. Curves 2, 3, and 4 were made using the Sonotone 3T-S ceramic unit.

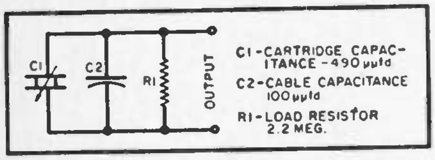


Fig. 5. Electrical values for ceramic cartridges having the response characteristics shown in Fig. 4. A Sonotone 3T-S is used here. Cartridges from other manufacturers may vary somewhat in capacitance and may thus require different loading.

netic ones, they contain no inductance and therefore are not sensitive to hum fields.

Although preamplification is no longer necessary, low-frequency equalization is still required, as in the case of magnetic cartridges. The type of equalization used, however, is entirely different than for magnetics, as well as being simpler on the whole. The output of a high-fidelity piezoelectric cartridge can be fed directly to an amplifier input intended for a high level source such as a tuner, TV set, or tape recorder. While the low-frequency response obtained in this way may be tolerable, it may be made more uniform by proper loading.

The purpose of this article is to explain why these pickups require low-frequency equalization and how this is achieved; and to discuss high-frequency equalization in connection with such cartridges. The discussion is based on the premise that the pickup is used on discs recorded according to RIAA standards, now quite generally employed in the United States. For clarity and complete understanding, it is first necessary to review the nature of the RIAA curve and the significance of the terms "velocity" and "amplitude" in disc recording.

For constant voltage input, the magnetic-type cutter of a disc recording machine, without equalization, operates at the same velocity for all frequencies. Inasmuch as velocity (V) is proportional to groove amplitude (A) times frequency (F), that is, $V = 2\pi AF$, amplitude declines as frequency goes up. However, this is wasteful of groove space at high frequencies. Also, at high frequencies the small signal excursions become comparable in amplitude to random imperfections in the record groove, which constitute noise. Therefore, as an approximation, the recording process should use equalization which produces constant amplitude over the audio range for constant signal input, as depicted by Curve 1 in Fig. 1.

From the relationship $V = 2\pi AF$, it is apparent that velocity varies directly with frequency when amplitude is held constant. Therefore the constant amplitude characteristic, Curve 1, corresponds to a velocity characteristic rising 6 db-per-octave, as shown by Curve 2.

Curve 3 shows the RIAA velocity characteristic, which, for reasons that appear in the last two paragraphs of this section, rises less than 6 db-per-octave. In other words, this is the frequency equalization generally used in the recording process. When a magnetic cartridge is employed in playback, the required playback equalization is the complement of Curve 3, providing bass boost and treble droop. This is so because the magnetic pickup, being an inductive device, produces a voltage proportional to velocity.

The piezoelectric pickup, on the other hand, responds not to velocity but to recorded amplitude. The amplitude of the RIAA recording curve is shown by Curve 4 and is obtained by subtracting Curve 2 from Curve 3. Thus Curve 4 is the amplitude characteristic corresponding to the RIAA velocity characteristic.

It is sometimes stated that modern discs have constant velocity above 500 cps and constant amplitude below 500 cps. This statement initially seems at odds with Fig. 1, which shows a *rising* velocity above 500 cps and a *rising* amplitude as frequency goes below 500 cps. The explanation lies in the fact that Fig. 1 depicts the *equalization* used in recording, while the statement referred to concerns the *actual recorded level* at various frequencies, which depends upon the relative intensities of sound energy over the frequency spectrum as well as upon the equalization used. This can be made clear by reference to Figs. 2 and 3.

The RIAA recording characteristic in terms of velocity is repeated as Curve 1 in Fig. 2. Curve 2 shows the hypothetical intensity of sound energy at various frequencies produced by a musical source. It is known that, generally speaking, audio energy is greatest around 400 cps and declines below and above this frequency. For a specific source, the actual curve would doubtless have different slopes than those of Curve 2 and would be marked by peaks and valleys. Nevertheless, it would bear a family resemblance to Curve 2, which is adequate for explanation purposes.

Curve 2 has been drawn so that the treble slope goes through 500 cps, which is taken as a reference level, depicted by Curve 3. The net result of frequency equalization per Curve 1 and varying intensity of sound at different frequencies is Curve 4. This is obtained by subtracting from Curve 1 the difference between Curves 3 and 2. It may readily be seen that actual recorded velocity, as illustrated by Curve

Fig. 6. The effect of load resistor upon response of a ceramic cartridge to RIAA disc. Curves 1, 2, 3, 4 are with Sonotone unit.

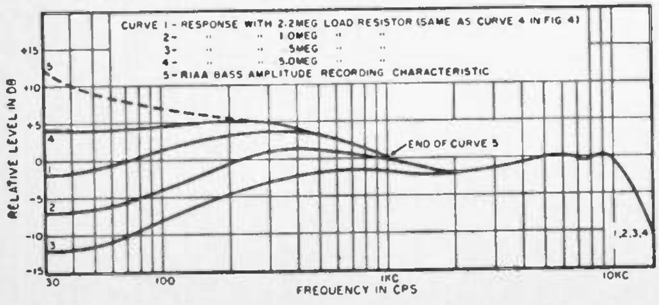
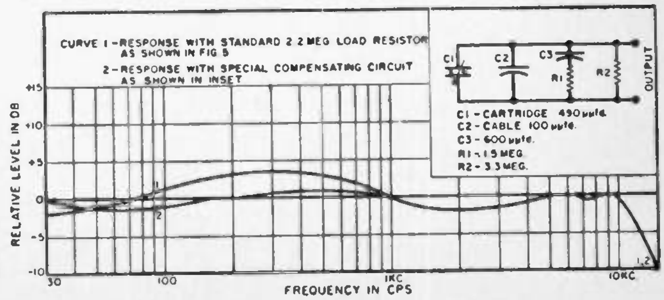


Fig. 7. Effect of a compensating circuit upon response of a ceramic cartridge to RIAA recording. All with Sonotone unit.



4, tends to be fairly constant above 500 cps. Hence the designation "constant velocity recording" for frequencies above 500 cps.

The RIAA recording characteristic in terms of amplitude is repeated as Curve 1 in Fig. 3. Curve 2, intensity of audio energy, is the same as in Fig. 2 and is drawn to pass through the reference level of 500 cps. The net result of Curves 1 and 2 is Curve 4, which represents actual recorded amplitude. Below 500 cps, Curve 4 substantially lives up to the description "constant amplitude recording."

RIAA recording equalization was designed with a careful view to the relative intensity of audio energy at various frequencies, as produced by typical sources of music and other sound. Although it would be advantageous from the viewpoint of signal-to-noise ratio to have rising recorded velocity in the treble range (note that the reference here is to recorded level, not equalization) an increase in velocity is limited by the ability of the playback stylus to track properly. Excessive velocity produces excessive acceleration forces (proportional to frequency times velocity) which would cause the stylus to lose contact with the groove walls when the tone arm is adjusted for reasonable vertical pressure, resulting in distortion.

Below 500 cps, it is necessary to impose a limit on recorded amplitude to prevent the disc cutter from cutting excessively wide grooves, which would result in cross-modulation between adjacent grooves and produce distortion. On the other hand, it is necessary to consider the playback problems that arise for a magnetic pickup if velocity drops as frequency declines. A magnetic pickup requires bass boost in playback which is complementary to the RIAA velocity recording equalization shown by Curve 3 in Fig. 1. To the extent of this boost, the reproduction of hum and rumble is increased. It is therefore desirable to limit the velocity droop due to recording equalization as much as possible so as to reduce the corresponding need for playback boost when using a magnetic cartridge. Translating this requirement into terms of amplitude, it is desirable that recorded amplitude be kept as high as feasible below 500 cps. The RIAA curve, taking into account the variation of sound energy with frequency, is therefore a compromise between the need for high recording amplitude to minimize magnetic cartridge playback problems and the need for restricted amplitude to avoid excessive groove excursions. This compromise results in approximately constant *recorded* amplitude below 500 cps, and is effected by bass recording *equalization* that falls moderately in terms of velocity and rises moderately in terms of amplitude with declining frequency.

Playback Equalization

In the following discussion it should be noted that all references to recording characteristics pertain to the RIAA (or other) characteristic; they do not pertain to *actual* recorded velocity or amplitude as determined by the distribution of audio energy.

Curve 1 in Fig. 4 shows the playback equalization required to achieve flat response when playing an RIAA recording with a hypothetical pickup that produces a voltage directly proportional to amplitude. This curve is the complement of the RIAA amplitude characteristic, shown as Curve 4 in Fig. 1.

In the case of a piezoelectric cartridge, the required treble boost can be achieved physically, by proper design, in the cartridge itself through resonance. The required bass droop can be achieved electrically by means of a suitable load resistor, which acts as part of a frequency-discriminating voltage divider.

Curve 2 in Fig. 4 shows the treble response of an actual high-fidelity piezoelectric pickup, with a ceramic element, when playing a constant amplitude recording. (The pickup used to obtain the data of Figs. 4 through 9 is the *Sonotone* 3T-S ceramic unit.) The rising characteristic is achieved by means of several damped mechanical resonances built into the vibratory system of the cartridge. High-frequency response of *all* types of cartridges is limited by resonance of the effective mass of the stylus and the compliance of the disc material. Response falls rapidly at frequencies above resonance. Furthermore, the high-frequency droop of Curve 2 is partly due to "playback losses" that normally occur when a typical 33⅓ rpm 12" vinyl record is played. These losses affect all types of cartridges.

In the case of magnetic cartridges, the manufacturer tries to prevent a rise in response at the resonant frequency by means of damping. The manufacturer of the high-fidelity piezoelectric pickup, however, uses resonance to achieve the required treble boost, with reference to constant amplitude recording, as indicated by Curve 1. By proper physical design and damping of his cartridge, he locates resonances in the portion of the audio spectrum where they will do the most good and regulates their amplitude and spread. Curve 4A, obtained by subtracting Curve 2 from Curve 1, shows the resulting treble response to an RIAA recording obtained through controlled resonance.

Curve 3 in Fig. 4 pictures the bass droop of the ceramic pickup, with reference to a constant amplitude recording, when loaded by a 2.2 megohm resistor. Fig. 5 indicates the electrical values of the ceramic cartridge in question, together with its cable and load resistor. This particular cartridge has a capacitance of 490 μμfd., while cable capacitance is assumed to be 100 μμfd., a reasonable average value. The time constant of R_1, C_1, and C_2 determines the shape of Curve 3 in Fig. 4. The difference between required and actual bass droop, respectively represented by Curves 1 and 3, determines bass response, as shown by Curve 4B.

It is important that for a given piezoelectric cartridge the correct load resistor be used. Too low a value produces inadequate bass response; too high a value, excessive bass. This is illustrated in Fig. 6 for the cartridge discussed here. It may be seen that a 2.2 megohm resistor provides the closest approach to flat response. If the pickup in question is fed into a .5-megohm load such as commonly found at amplifier inputs intended for radio, TV, etc., bass is attenuated more

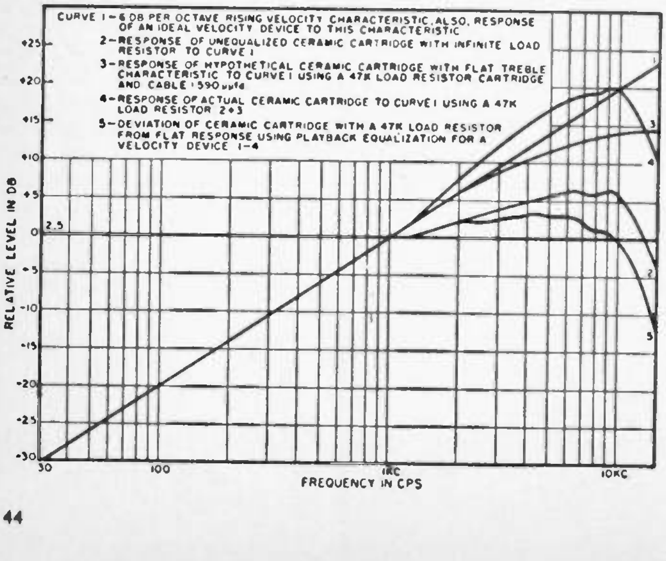
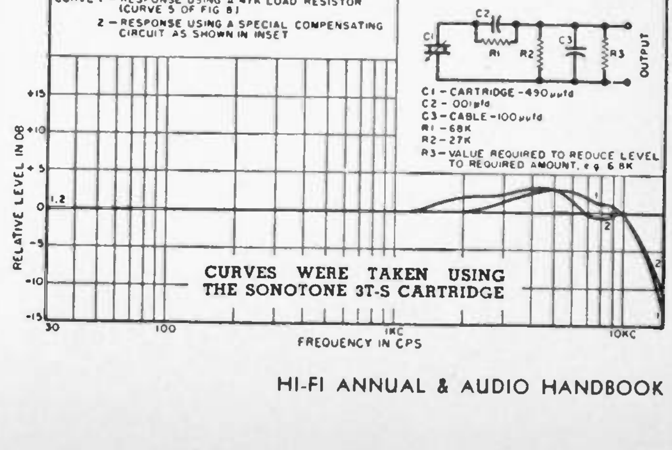


Fig. 8. Response of a ceramic pickup when equalized to produce an output proportional to velocity. Curves 2, 5 Sonotone unit.

Fig. 9. Effect of compensating network on response of a ceramic cartridge when treated as a velocity device.



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than 11 db at 50 cps. Even if the load resistor is 1 megohm, attenuation at 50 cps exceeds 6 db. On the other hand, a 5-megohm resistor would result in 4 to 5 db boost over most of the bass range.

Curve 5 in Fig. 6, shown for reference purposes, is the bass portion of the RIAA amplitude recording curve. A sufficiently high load resistor would provide bass response closely following this curve all the way inasmuch as at low frequencies the ceramic pickup represented in Fig. 6 is quite flat in its response to a constant amplitude recording if the load resistor is large enough.

Occasionally, considerable effort may be involved in getting at and changing the existing load resistor, typically .5 megohm, in a sound reproducing system. To the user, the cost of having this change made by a technician can approach or exceed the cost of the pickup itself. In order to obtain good bass response without changing the input load resistor, one may instead follow the expedient of increasing C_2, shown in Fig. 5. C_2 can be increased by wiring a miniature capacitor directly across the cartridge. Provided that the time constant $R_1(C_1 + C_2)$ remains the same (1300 microseconds for the pickup in question), frequency response as shown by Curve 4 in Fig. 4 remains unaffected.

However, in following this course one gets lower output. For example, in matching the ceramic cartridge of Fig. 5 to a .5-megohm load, it would be necessary to use a shunt capacitance (including cable) of approximately .002 μfd., resulting in nearly 13 db attenuation. A 1-megohm load would require a shunt of about .0008 μfd., resulting in about 7 db loss. Some reproducing systems with a good deal of gain can easily endure, or may require, such sacrifices, while other systems cannot tolerate these losses. In the latter cases, if good bass response is desired, the load resistor must be changed to the correct value.

Returning to Curve 4 in Fig. 4, it may be observed that over-all response is quite creditable, staying within ± 3 db between 30 and 11,000 cps. Useful response is maintained to 15,000 cps. The chief criticism concerns the gradual hump that occurs between about 100 and 1000 cps. However, this is easy to remove by means of a simple compensating circuit, suited to the particular pickup in question, which appears in the inset of Fig. 7. Curve 2 in Fig. 7 is the resulting response, and may be compared with Curve 1, which is the response obtained when the load consists of a 2.2 megohm resistor. Curve 2 is very little short of excellent, remaining within ± 2 db over the range of 30 to 11,000 cps, with useful response extending to 15,000 cps.

The discussion thus far has been based on the premise that the piezoelectric cartridge is used as an amplitude-responsive device to play recordings made with the RIAA characteristic. Naturally the question arises as to how the pickup can be used to play recordings having one of the other equalization characteristics that were in widespread use until about two years ago and are encountered today to a limited extent.

The writer inclines to the view that generally satisfactory adjustment for the difference between the RIAA curve and other curves can be effected by means of bass and treble controls. (See the article by Burt Hines, "Do You Need a Preamp-Control Unit?" RADIO & TELEVISION NEWS, January 1956.) However, the fact remains that many listeners prefer to obtain nominally precise equalization by turning one or two switches on a preamplifier to a setting which is marked as providing LP, Old AES, 78, European, or other equalization.

To enable the user to take advantage of the multiplicity of equalization characteristics afforded by many preamplifiers, it is generally necessary to convert the amplitude-responsive piezoelectric pickup into a velocity-responsive device. The reason is that the phono-input of such preamplifiers is usually designed to accommodate magnetic cartridges, which respond to velocity.

Converting the piezoelectric pickup into a velocity device is simple. It is merely necessary to load the cartridge with a suitably small resistor, so that the time constant of the resistor, cartridge capacitance, and cable capacitance (see Fig. 5) results in rising response with frequency over all or most of the audio range. This is illustrated in Fig. 8.

Curve 1 represents a 6 db-per-octave rising velocity characteristic, as well as the response of an ideal velocity device, magnetic or otherwise, to this characteristic. Curve 2 is the response of a specific ceramic pickup to Curve 1 when an infinite load resistor is used. The hump in the treble range is due to physical resonances, as previously discussed.

Assume that we have a hypothetical ceramic pickup with the same capacitance as in Fig. 5 but without resonance in the audio range. Its response to Curve 1 would be flat. Next, assume that this hypothetical pickup is loaded by a 47,000-ohm resistor. The resulting response would be that of Curve 3, which is a 6 db-per-octave rising velocity characteristic except at the upper end. Now, by adding Curves 2 and 3, one can determine the response of the ceramic pickup in question to Curve 1 when it is loaded by a 47,000-ohm resistor. Curve 4 shows the resulting response. When the ceramic pickup is fed into a preamplifier input intended for a magnetic cartridge, and equalized accordingly, its deviation from flat response is as delineated by Curve 5. Curve 5 is the difference between the response of a perfect velocity device (Curve 1) and the response of the ceramic cartridge in question when loaded by a 47,000-ohm resistor (Curve 4).

If the load resistor were appreciably lower than 47,000 ohms, say 10,000 ohms, the ceramic pickup in question would have a response proportional to velocity throughout the audio range plus treble emphasis due to resonance. Its resulting response, when equalized by a preamplifier as a velocity device, would be that of Curve 2. The reason for selection of a 47,000-ohm load resistor instead of a much higher or lower value is to provide a rising characteristic, such as Curve 3, that tapers off at the high end and thereby compensates the resonance hump as much as possible without unduly impairing response in the vicinity of 10,000 cps.

Curve 5 in Fig. 8 provides quite good response. The treble hump gradually reaches a peak of only slightly more than 3 db and response is maintained out to 11,500 cps before it is 3 db down.

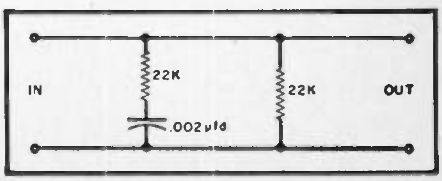
Somewhat smoother response, that is, a treble hump with about as great a peak as in Curve 5 but extending over a smaller portion of the spectrum, can be obtained by using the compensating circuit shown in Fig. 9 with the ceramic cartridge in question. Curves 1 and 2 respectively show response when a 47,000-ohm load resistor is used and when the compensating network is used. The network can be mounted at the magnetic phono input of the preamplifier.

It is quite possible that a 47,000-ohm load resistor for the ceramic pickup under discussion may give more pleasing results than the network of Fig. 9, because the broader treble hump may compensate more satisfactorily for treble deficiencies in speaker response, room acoustics, hearing acuity, etc. Only a listening test can decide which is the better method of loading a ceramic cartridge.

Figure 10 shows the circuit recommended by *Electro-Voice* for converting its ceramic cartridge into a velocity device.

A correctly chosen load resistor, or a network such as shown in Figs. 9 and 10, is the best way of converting a piezoelectric cartridge into a velocity device. Another method sometimes employed is that of Fig. 11, where the cartridge output is in series with a capacitance small enough to produce a rising response characteristic over the entire audio range. The reactance of C in Fig. 11 is large compared to that of R at all audio frequencies. This method is used in several amplifiers on the market that have an input marked "ceramic or crystal phono input." Unfortunately, this method

Fig. 10. Recommended circuit for converting the Electro-Voice ceramic cartridge into a velocity device. Refer to article.



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produces a departure from flat response to the full extent of the cartridge's resonant characteristic in the treble range. Thus in the case of the cartridge used for illustrative purposes in this article, response would be that of Curve 2 in Fig. 8.

When R_3 is omitted from the compensating circuit in Fig. 9, output available from the cartridge is on the order of 36 millivolts, referred to the standard reference level, which will overload many preamplifiers unless they have an input designed to accommodate high output magnetic cartridges. Preamplifiers often cannot accommodate more than 10-15 millivolts. To prevent overloading, a suitably low value of R_3 should be used; the value selected does not appreciably affect response of the compensating network.

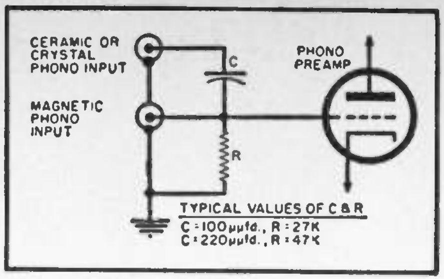


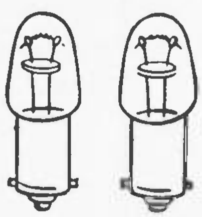
Fig. 11. Method used in some amplifiers to convert a piezoelectric cartridge into a velocity device. This is discussed in text.

For example, a 6800 ohm resistor produces an output of about 7 millivolts from the pickup depicted in Fig. 8. This is about right for many preamplifiers.

If a simple load resistor is used instead of a more complex compensating network, it is still necessary to guard against overloading. To reduce output, the single load resistor may be replaced by a voltage divider consisting of a potentiometer or two resistors.

The author wishes to thank *Astatic Corp., Electro-Voice, Inc., Ronette Acoustical Corp.,* and *Sonotone* for their courtesy in supplying data on their high-fidelity piezoelectric cartridges. —30—

A Light Bulb Volume Expander-Compressor



An inexpensive, easy-to-build gadget which provides added enjoyment when listening to your hi-fi system.

By **W. H. CALDWELL**

THE volume compressor is useful for reducing the dynamic range of speech or program material. In recording, too large a signal will cut into an adjacent groove, while too small a signal will be lost in the surface noise of the record. With tape, too strong a signal will saturate the tape and may cause "print-through" of a strong passage on an adjacent layer of tape, while too weak a signal will be lost in the background hiss.

In communication radio transmitters, voice compression helps to override noise. In high-fidelity broadcasting and recording the trend today is away from volume compression, to insure full dynamic range.

The average listener, however, may not want his music too loud, to avoid offending the neighbors; nor can he have it too soft since ordinary room or street noises will drown out the music.

The volume expander, by the same token, is useful for restoring full dynamic range to an old record or to reception over a channel where compression has previously been employed.

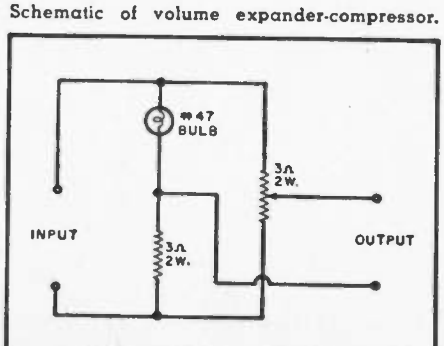
A good volume expander-compressor is a useful addition to any audio system, provided it doesn't introduce unnecessary distortion and is easy to operate.

The complete circuit of such a device is shown in the diagram. It uses only three components, and connects directly between the output transformer and the speaker voice-coil. It is designed for speaker impedances from one to eight ohms and for amplifiers in the 1-10 watt class.

The operation depends on the characteristics of the #47 pilot bulb. The resistance of the bulb increases as more power is applied to the circuit; however, the thermal heat capacity of the filament is sufficient so that its resistance remains constant over several cycles of audio. This latter characteristic preserves the waveform of the audio. Some may find that the attack and decay times are far too long due to the thermal lag of the bulb's filament, but the simple gadget is worth trying if this characteristic is not too objectionable.

There is only one control to the compressor-expander: the potentiometer. This should be mounted near the volume control for greater convenience, since a change in the setting of the expander will affect the volume as well as the degree of expansion.

Schematic of volume expander-compressor.



In operation, the lamp will glow and also serve as a visual volume indicator. When testing the circuit for the first time, set the volume so that the bulb glows moderately brightly. Now by adjusting the potentiometer, you will find there will be a null point near the center of its range, where the sound will be weakest. This is the balance point of the resistance bridge, and the greatest degree of compression will be found just below this point, and the greatest degree of expansion just above this point.

For moderate compression, turn all the way down; for moderate expansion, turn all the way up.

For use as a squelch circuit, set the potentiometer to balance on noise; the signal will then unbalance the circuit and come through, when present, while in the absence of signal the noise is blocked-out. This is especially useful for playing old scratchy records since it not only squelches the scratch, but also expands the dynamic range of the record.

In home recording, the compressor setting can be used when recording and the expander used on playback.

A few words about the losses of the expander. You will need to set the volume control higher than usual to overcome the losses in this circuit. And these losses are greater near balance where the most effective action takes place. If more power is needed, it should be added after the expander in the form of a power amplifier. —30—

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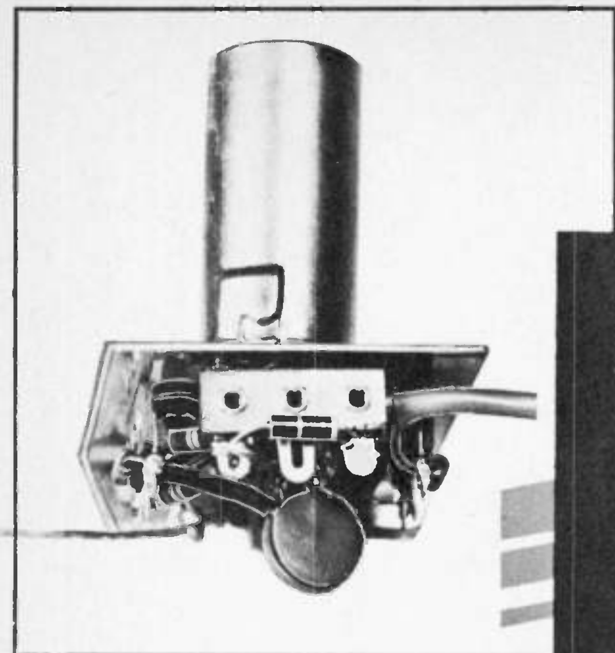
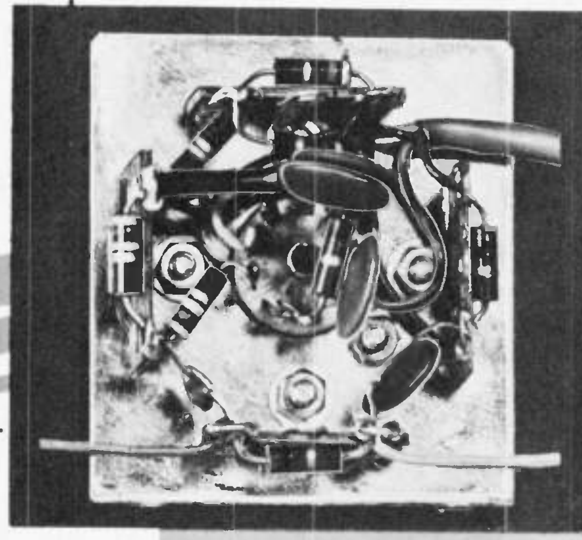
A Simplified Automatic Tone Compensator

By
ED. C. MILLER
Tech. Dir., Inland Broadcast Co.





Two views of the automatic tone compensator. It is built on a small sub-chassis which can be put into amplifier chassis.

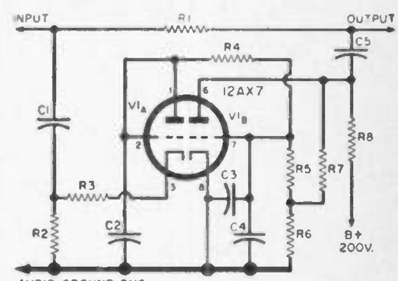
Details on a one-tube device which automatically boosts bass response at low audio levels. It is easy to build.

SINCE the appearance of the author's previous article on "The Auto-Tone" (RADIO & TELEVISION NEWS, April, 1954, page 54), many letters have been received asking for information on a less critical and more easily built device to provide automatic bass boost at low volume levels. This article covers such a circuit, reduced to its simplest practical form.

The unit shown in the photographs and schematically in Fig. 1 differs from the previous automatic tone control in many ways besides being of much simpler construction. Most significant, probably, is that because the control tube uses bias increasing in a negative direction with the application of signal, the control capacitor is in series with the plate impedance of the control tube, rather than in parallel. Fig. 2 is the equivalent circuit, with R_b inserted in place of the plate resistance of the control tube. If capacitor C in Fig. 2 is selected so that its reactance is large in relation to R_a at some frequency under consideration, changing the value of R_b will have little or no effect on the output. But if the reactance of C is small in relation to R_a, and R_b is at all times less than R_a, then changing the value of R_b will have a marked effect on the output at that frequency. As the reactance of a capacitor increases with a decrease in frequency, maximum control will take place at the higher frequencies.

In effect, what we have is a volume expander that for a given range of input levels provides more expansion at the higher frequencies than at the bass frequencies. By proper selection of component values, a satisfactory expansion rate can be had that will be the same rate at all frequencies above about 500 cycles, with as much as a 2 or 3 to 1 reduction in rate at 50 cycles.

At first it may seem improper to use volume expansion as a type of control. Actually, it is one of two choices: 1. linear reproduction of the middle and high range of audio, with compression of the bass frequencies at higher volume levels; or 2. linear reproduction of a particular bass frequency with expansion of the middle and high frequencies. The use of the latter method adds about 15 db to the dynamic range of recording, if it is used with commercial pressings, and a similar amount of dynamic range increase in radio reception also (to a somewhat lesser extent, due to the sharp knee of the compression curve on broadcast



R_1—2.2 megohm, 1/2 w. res.
R_2, R_3—470,000 ohm, 1/2 w. res.
R_4, R_8—1 megohm, 1/2 w. res.
R_5, R_7—4.7 megohm, 1/2 w. res.
R_6—15,000 ohm, 1/2 w. res.
C_1—500 μμfd., 400 v. capacitor
C_2, C_3, C_4—.02 μfd., 400 v. capacitor
C_5—.005 μfd., 400 v. capacitor
V_1—12AX7 tube

Fig. 1. Complete schematic of the automatic tone compensator. Parts are standard.

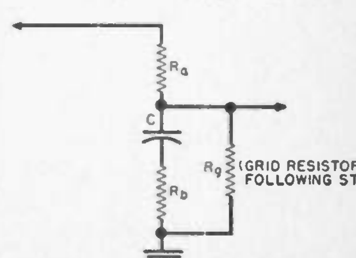


Fig. 2. Equivalent circuit to that of Fig. 1.

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transmitters).

In Fig. 1, C_5 is the capacitor that determines the amount of effective "bass-boost" at low volume levels. R_6 controls the amount of d.c. feedback

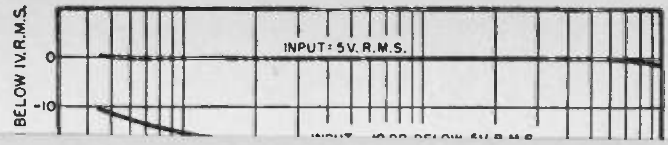


Fig. 3. Measured performance of compensator over a 20 decibel range.

surface noise of the best LP discs. The r.m.s. sum IM distortion of the last two stages is, under any circumstances, entirely negligible, being on the order of 0.2% at three volts output with the tone controls in flat position, the volume control wide open, and a 27,000-ohm resistor shunted across the output terminals. The IM distortion of the entire preamp on phono position is difficult to measure because of the impossibility of leveling the response without affecting other factors, but it appears to be well under 0.6% with an input of about 30 mv.

For some purposes a somewhat higher gain on the phono inputs may be desirable, and this may be had by substituting a 12AX7 for the 12AY7. The substitution does not involve any compromise whatever, and the only result is to approximately double the gain of the unit on the high-gain input position. A very careful selection must, however, be made since many 12AX7's are quite microphonic. During the experimental work one case was observed in which mechanical feedback from the speaker to the 12AX7 was sufficient to cause singing at 500 cps even though the speaker cabinet was ten feet from the preamplifier! It is probably wise to obtain premium tubes (6136 for 6AU6, and 5751 for 12AX7) for particularly critical applications. Shock-mounting the tubes may also prove helpful.

A Plug-in "Presence" Equalizer

By ALLAN M. FERRES

THE mid-range, or "presence" equalizer is an interesting addition to any home high-fidelity installation. It furnishes a boost of up to 6 db at 3000 cycles without affecting the response at the extreme ends of the frequency range. This type of equalization is often used in film and disc recording to emphasize solo voices and instruments. When used on speech reproduction, the articulation factor, or "understandability," is increased without producing the annoyance of excessive sibilance.

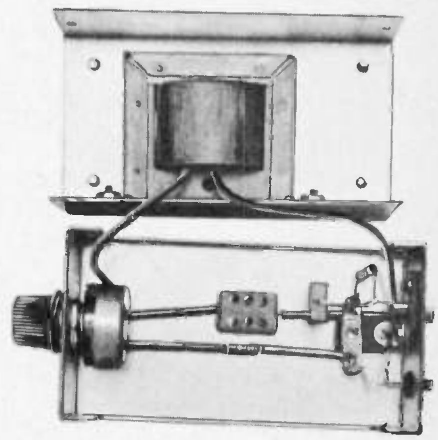
In a home reproducing system the presence equalizer will not produce as marked an effect as the bass or treble controls, but it will tend to highlight most solo instruments and for this reason it is a worthwhile attachment.

The presence equalizer described here is a simple device which can be attached to the existing equipment between the equalizing preamp and the power amplifier.

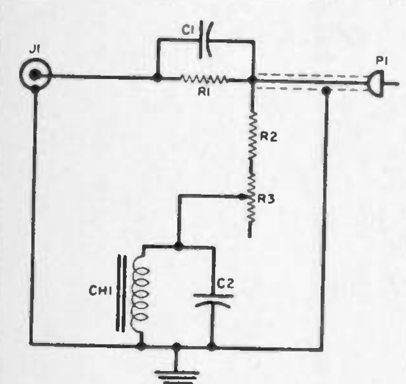
The circuit is simple and uses only a few standard, readily available parts. As can be seen in the schematic, Fig. 1, the equalizer is basically a voltage divider, the series leg consisting of R_1, shunted by C_1, and the shunt leg, made up of a parallel tuned circuit, CH_1-C_2, in series with R_2 and R_3. As CH_1-C_2 is resonant at 3000 cycles, the impedance of the shunt leg is highest at this frequency and the loss in the voltage divider is minimum, furnishing the mid-range boost. The amount of the boost is controlled by the total value of R_2 and R_3 in series. The boost is 6 db when R_3 is set at minimum resistance and zero when set at maximum. C_1 is used to compensate for high-frequency losses caused by stray and cable capacities in the divider circuit.

The frequency response curves of Fig. 2 are obtained when the input resistor of the power amplifier is about 500,000 ohms, the usual value. If the input resistor is less than 470,000 ohms, it should be changed to this value. If the resistor is 1.0 megohm, a 1.0 megohm resistor can be connected across the output cable of the equalizer and no wiring change will be necessary in the power amplifier.

The length of the shielded leads to the preamp and the main amplifier should be kept as short as possible to reduce the chance of hum pickup and uncompensated high-frequency loss.



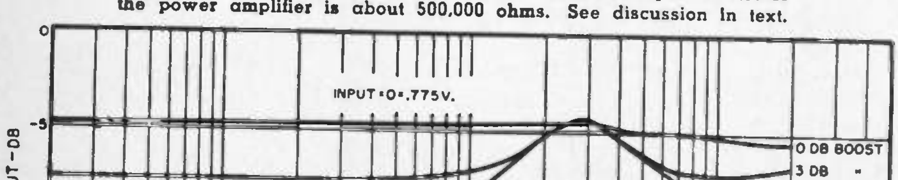
Two views of author's home-built equalizer. It is housed in a small box and uses standard, readily available parts.



R_1—220,000 ohm, 1/2 w. res.
R_2—100,000 ohm, 1/2 w. res.
R_3—1 megohm reverse log taper or linear pot
C_1—75 μμfd. mica capacitor
C_2—100 μμfd. mica capacitor
CH_1—20 hy., 15 ma. choke (Stancor C-1515 or equiv.)
J_1—RCA-type phono jack
P_1—RCA-type phono plug

Fig. 1. Schematic of the equalizer unit.

Fig. 2. Frequency response curve of equalizer when input resistor of the power amplifier is about 500,000 ohms. See discussion in text.



The construction of the presence equalizer can be seen in the photographs. CH_1 is mounted on the bottom section of the case and wired into the circuit with its flexible leads. The pot should be wired so that minimum resistance is obtained when the control shaft is in the full clockwise position. The winding-to-core capacitance of the *Stancor* C-1515 choke used in this equalizer causes the circuit to resonate at 3000 cycles when the choke is shunted by the 100 μμfd. capacitor. If another make of choke is used, the value of C_2 may have to be changed to produce the required resonant frequency.

The insertion loss varies between 4.8 db and 10.5 db, depending upon the amount of boost. The usual type of equipment used in home high-fidelity installations has enough spare gain to compensate for this loss.

As the choke is subject to hum pickup from the magnetic fields of power transformers and tape machine and record player motors, the equalizer should not be mounted permanently until it has been determined that its position in relation to such hum-producing fields is correct for minimum hum. —30—

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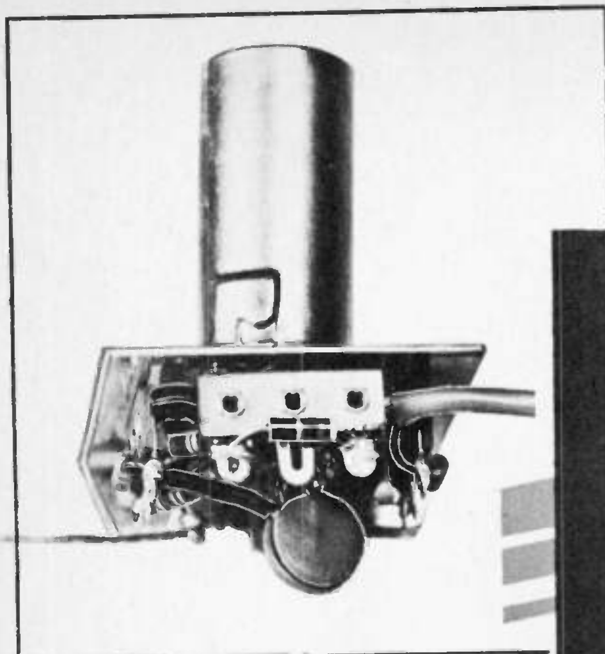
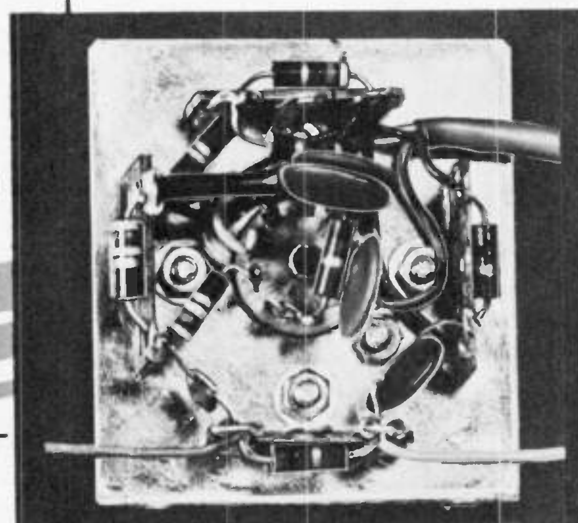
A Simplified Automatic Tone Compensator

By
ED. C. MILLER
Tech. Dir., Inland Broadcast Co.





Two views of the automatic tone compensator. It is built on a small sub-chassis which can be put into amplifier chassis.

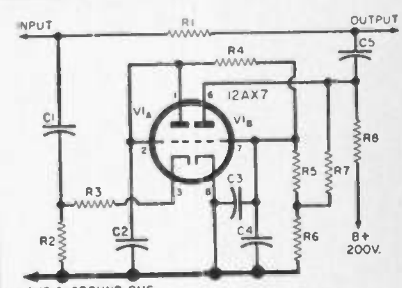
Details on a one-tube device which automatically boosts bass response at low audio levels. It is easy to build.

SINCE the appearance of the author's previous article on "The Auto-Tone" (RADIO & TELEVISION NEWS, April, 1954, page 54), many letters have been received asking for information on a less critical and more easily built device to provide automatic bass boost at low volume levels. This article covers such a circuit, reduced to its simplest practical form.

The unit shown in the photographs and schematically in Fig. 1 differs from the previous automatic tone control in many ways besides being of much simpler construction. Most significant, probably, is that because the control tube uses bias increasing in a negative direction with the application of signal, the control capacitor is in series with the plate impedance of the control tube, rather than in parallel. Fig. 2 is the equivalent circuit, with R_b inserted in place of the plate resistance of the control tube. If capacitor C in Fig. 2 is selected so that its reactance is large in relation to R_a at some frequency under consideration, changing the value of R_b will have little or no effect on the output. But if the reactance of C is small in relation to R_a, and R_b is at all times less than R_a, then changing the value of R_b will have a marked effect on the output at that frequency. As the reactance of a capacitor increases with a decrease in frequency, maximum control will take place at the higher frequencies.

In effect, what we have is a volume expander that for a given range of input levels provides more expansion at the higher frequencies than at the bass frequencies. By proper selection of component values, a satisfactory expansion rate can be had that will be the same rate at all frequencies above about 500 cycles, with as much as a 2 or 3 to 1 reduction in rate at 50 cycles.

At first it may seem improper to use volume expansion as a type of control. Actually, it is one of two choices: 1. linear reproduction of the middle and high range of audio, with compression of the bass frequencies at higher volume levels; or 2. linear reproduction of a particular bass frequency with expansion of the middle and high frequencies. The use of the latter method adds about 15 db to the dynamic range of recording, if it is used with commercial pressings, and a similar amount of dynamic range increase in radio reception also (to a somewhat lesser extent, due to the sharp knee of the compression curve on broadcast



R_1—2.2 megohm, ½ w. res.
R_2, R_3—470,000 ohm, ½ w. res.
R_4, R_8—1 megohm, ½ w. res.
R_5, R_7—4.7 megohm, ½ w. res.
R_6—15,000 ohm, ½ w. res.
C_1—500 μμfd., 400 v. capacitor
C_2, C_3, C_4—.02 μfd., 400 v. capacitor
C_5—.005 μfd., 400 v. capacitor
V_1—12AX7 tube

Fig. 1. Complete schematic of the automatic tone compensator. Parts are standard.

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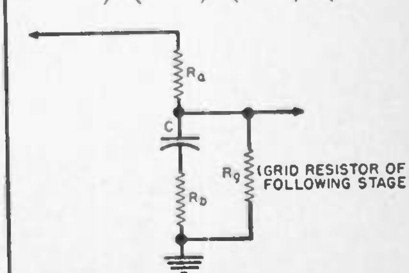

Fig. 2. Equivalent circuit to that of Fig. 1.

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transmitters).

In Fig. 1, C_5 is the capacitor that determines the amount of effective "bass-boost" at low volume levels. R_6 controls the amount of d.c. feedback from the plate to grid of the control tube and, hence, the rate of expansion or control. The values shown will give an operating range in excess of 20 db, with a 1 kc. to 50 cps ratio of about 2 to 1.

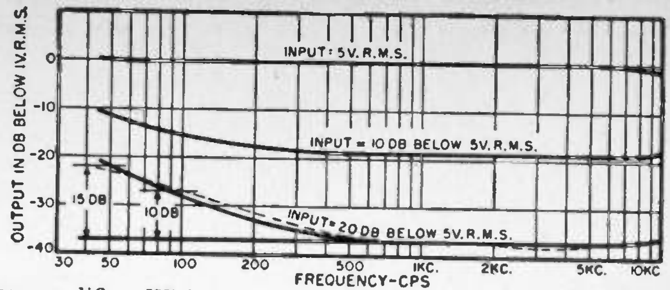

Fig. 3. Measured performance of compensator over a 20 decibel range. The dotted line is change in human ear response over a similar range of intensity. See text.

The principle of operation is as follows: V_{1A} functions as a diode rectifier, providing a negative, filtered d.c. voltage on the grid of V_{1B} proportional to the amount of applied audio signal. As this voltage increases, less current flows in V_{1B}, resulting in a higher plate impedance. As this impedance is in series with C_5, the increasing impedance reduces the effectiveness of C_5, and, hence, provides a more nearly flat response to all frequencies at the output. A portion of the voltage present at the plate of V_{1B} is applied through a voltage divider network to its grid return, giving a sort of d.c. negative feedback, to control the sensitivity of the circuit and, to a certain extent, its linearity.

In order that the level that determines the amount of control is primarily from the middle and higher frequencies, C_1 is made small so the amount of signal reaching the rectifier (V_{1A}) is reduced at the lower frequencies. R_3 is a limiting resistor, whose purpose is to prevent undue loading of the input with sudden dynamic increases. C_2, C_3, and R_4 form an *RC* filter network. R_5 determines the recovery rate (in conjunction with C_2, C_3, C_4, and R_4; R_6 and R_7, the d.c. feedback, or expansion rate. Because R_7 must be kept large (because it is effectively in parallel with R_8 and the plate impedance of V_{1B}), R_6 is the determining component.

The unit pictured was built on a small subchassis to be mounted under the main chassis of an amplifier. With the layout shown, all but three of the necessary connections are made by the component leads themselves. This simplifies the wiring and also reduces the problems of hum and noise. There is no reason why this unit could be built only in this manner, though. Any method where good audio construction practices are employed, should be satisfactory.

It should be remembered that there is a minimum loss of 15 db, also, that the unit should be inserted in an amplifier at a point where there is about 5.0 volts r.m.s. of audio present during normal programming. This circuit must follow any equalizers and precede any tone controls. The volume control should be prior to this compensating device, so that manual adjustment of the volume level will have the same effect as will changes in the intensity of particular passages of sound from the record or radio.

The curves of Fig. 3 show the measured performance of the unit over a 20 db range. It can be seen how closely it follows the change in the human ear response over the same range (obtained from information supplied by the Acoustical Society of America). In taking these measurements, C_1 is shorted out, so that the *control* frequency and the *controlled* frequency can be the same. In actual operation, the controlling frequencies are primarily above 500 cycles because of the *RC* circuit used in the input of the rectifier. Using different tube types and various changes in C_5 and/or R_6, various amounts of control, over a wide range of frequencies, can be obtained.

A switch can be inserted in the cathode of V_{1B}, if desired, to disable the circuit's operation. In this manner, the effectiveness of automatic tone compensation can be more easily realized by comparison. If the cathode of V_{1B} is opened, the plate impedance will rise to its maximum, and the response of the circuit will be nearly flat over the audio range, with an insertion loss of 15 db. Closing the switch (grounding the cathode of V_{1B}) will leave the response and insertion loss unchanged at high volume levels, but will increase the insertion loss at the middle and high frequencies on low volume levels. This, in effect, provides an increase in the bass boost at lower volume levels, compensating for the non-linear intensity response of the human ear to low-frequency sounds.

This circuit should not be considered the solution to all audio problems, but rather as a building block, in the continuing search for the reproduction of sound in a small, acoustically imperfect room in our home, that will closely resemble the original sound as recorded in an acoustically treated room, at a much louder sound intensity. When considered in this light, it performs its function simply and inexpensively. —30—

SIMPLIFYING TONE CONTROL CIRCUITS

By **HAROLD REED**

Research and Engineering, U. S. Recording Company

Details for constructing a bass boost-bass cut, treble boost-treble cut, dual-type tone compensating circuit.

IN THE schematic diagrams of audio amplifiers, tone control circuits often appear complex and difficult to understand. Like any other electronic network, the circuit can be broken down from the seemingly complex configuration to simple equivalent circuits and then analyzed and the function of each component part clearly visualized.

Actually the simple circuit of the coupling capacitor from the plate of one stage of a voltage amplifier to the grid of the following stage, in conjunction with the grid resistor of the driven stage, affects the tone or frequency response of the amplifier. This circuit is shown in Fig. 1A and redrawn for analysis in Fig. 1B.

It can be seen that *C* and *R* constitute a voltage divider network and that the voltage available to the grid of V_2 is that which appears across *R*. Neglecting the internal capacitances presented by the tubes and wiring capacitances, the voltage across *R* which is available to V_2, can be found by solving the equation $E_o = E_i\,[R/(R+X_c)]$, where X_c is equal to $1/2\pi fc$ and f is the frequency in the audio range being considered. This gives the approximate value of E_o. To find the absolute value, still ignoring circuit capacities, we cannot combine *R* and X_c by simple addition. The absolute value is equal to the equation, $E_o = E_i\,(R/\sqrt{R^2+X_c^2})$. If the values of *C* and *R* are properly chosen there will be little attenuation at the selected reference frequency in the audio band, say between 400 and 1000 cycles. It can be seen from the equation of Fig 1B that this is a low-frequency, or bass-attenuation, network. As the frequency of the signal from V_1 increases, the reactance of *C* decreases and a greater portion of the total value of E_i appears across *R* and the grid of V_2. With decreasing frequency of the signal from V_1, the reactance of *C* increases, causing less of the signal voltage, E_i, to appear across *R* and the V_2 grid. In other words,

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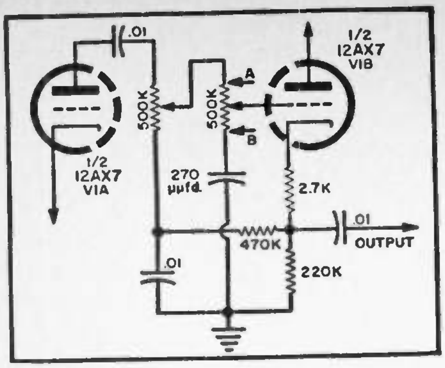


Fig. 2. Practical application of the circuit discussed theoretically in Fig. 1C.

there is a greater voltage drop across C at the lower frequencies. The attenuation in decibels is equal to 20 *log* E_o/E_i or may be found by using the equation $db = 10\ log\ [1 + (f_1/f)^2]$, where f_1 is the reference frequency at which R and X_c are equal and f is the operating frequency or frequency at which the test is being made, referred to f_1.

Suppose we now reverse the positions of R and C as shown in Figs. 1C and 1D. C_1 is the usual interstage coupling capacitor and R_1 the grid resistor of the following stage. This is the familiar type of tone control for high-frequency, or treble, attenuation. The 500,000 ohm potentiometer, R_2, is the volume control.

The equation $E_o = E_i\ [X_c/(R + X_c)]$ applies and R and C still constitute a voltage divider network, but E_o appears across C instead of across R as occurred in the previous arrangement. Now, at low frequencies, the reactance of C becomes high and the greater portion of E_i appears across C and the grid of V_2. The signal voltage to V_2 is then determined primarily by R and the grid resistor R_1. As the frequency of the signal from V_1 is increased, the reactance of C becomes smaller and smaller and less of the signal voltage appears across C and the grid of V_2. The resistance, R, is usually made variable so that attenuation at the higher frequencies can be varied. For instance, if R consists of a 500,000 ohm potentiometer and C is a 270 μμfd. capacitor, the attenuation from the reference point is variable, depending upon the setting of R which varies the ratio of X_c to R. Loss in db is again equal to 20 *log* (E_o/E_i) or to 10 *log* $[1+(f/f_1)^2]$.

As an example, this network as used between the triode sections of a 12AX7 with the output section functioning as a cathode follower, as shown in Fig. 2, the actual measured attenuation with the component part values shown was 2 db at 1000 cycles, 16 db at 8000 cycles, and 24 db at 20,000 cycles when the 500,000 ohm potentiometer arm was at position B. With the potentiometer arm at position A, the amplifier response is essentially flat over the audio frequency band.

High-fidelity amplifiers are usually equipped with controls for boosting or attenuating both the high and low frequencies. A circuit satisfying these operating conditions is given in Fig. 3. It consists of four capacitors, a single 3 megohm potentiometer, and a 500,-000/50,000 ohm potentiometer.

It can be seen that this tone compensating network, working between the triode sections of a 12AX7 tube, provided a treble cut of 17 db and a treble boost of 16 db at 20,000 cycles. The bass control effected a bass cut of 15 db and a bass boost of 16 db at 30 cycles. Of course, many variations of these control settings may be employed so that the listener may alter the frequency response of the amplifier to suit his particular taste. A curve of the compensating effect of this network is shown in Fig. 4.

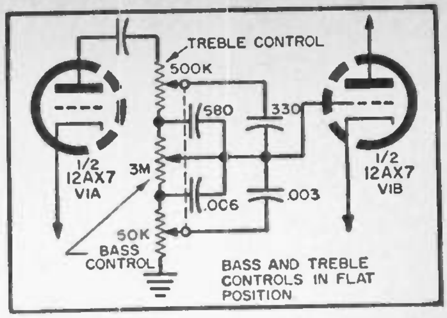


Fig. 3. A dual-type tone compensating circuit. Parts specifications are included.

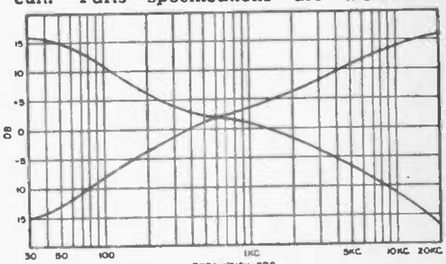


Fig. 4. Curve of the compensating effect of the network shown in diagram, Fig. 3.

It is beyond the scope of this article to mathematically analyze each possible configuration of Fig. 3. Those interested may break the circuit down into simpler arrangements for study and analysis as explained previously. For instance, Fig. 3, assuming that the arms of the dual controls are exactly at center position of the total resistance of the controls, might be redrawn to give the equivalent circuit of Fig. 5 and so on for each circuit condition.

For the reader interested in constructing the tone compensating network just discussed, parts values are shown. Although dual controls for commercial sound reproducing systems are usually designed especially for such applications, the home constructor may build a satisfactory network using the multi-section controls currently listed by *International Resistance Co.* These multi-section controls can be combined with the *IRC* "Q" controls for numerous dual-control combinations. They are attached to the single control in the same manner as an "on-off" power switch is added to a variable control.

Fig. 1. (A) Frequency-sensitive circuit found in most commercial amplifiers. (B) Equivalent circuit redrawn for analysis. (C and D) Variations of circuit of (A).

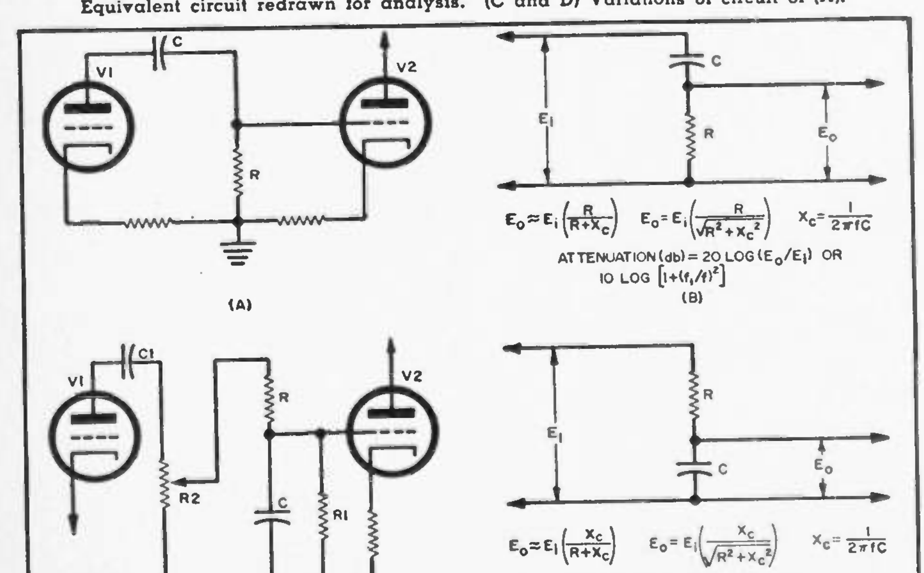
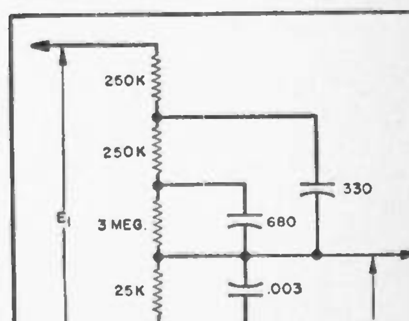


Fig. 5. Example of how the various circuits can be redrawn for easy analysis.



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A Simple Preamplifier

By **CHARLES P. BOEGLI**
Cincinnati Research Company
and
RAYMOND H. WILLIS

for the Home Builder

Fig. 1. Front panel view of the preamplifier. It is housed in a wood case to permit its use on a table top but could, just as easily, be used in its chassis form.

Construction details on a compact, versatile unit that provides an equalized low-impedance output of 0.5 volt and includes a highly effective tone-control circuit.

A NUMBER of power amplifiers are now available which require considerably less than one volt r.m.s. to drive them to full output. One such amplifier was recently described in this magazine. The majority of preamplifiers, however, are usually designed to provide an output of one to two volts r.m.s., which would overload such amplifiers and lead to excessive distortion by driving them into the clipping region.

It is possible, of course, to adjust the output level of a preamplifier by means of a volume control or voltage divider so that it will not introduce an overloaded condition. Such an expedient, though, is wasteful of gain and may lead to a poorer signal-to-noise ratio. Also, designing a preamplifier for high output may lead to more distortion than would be the case in a low-output unit. Although such distortion could be reduced by means of inverse feedback, it seems silly to design a preamp capable of delivering two or three volts or more when less than one volt is required.

Shown in Fig. 1 is a preamplifier designed specifically for those amplifiers requiring about .5 volt of drive for full output. It can, of course, be used with lower sensitivity amplifiers, but cannot drive them to full output. It is ideally suited for the 13-watt "infinite-feedback" amplifier referred to previously.

The output impedance of this preamplifier is quite low—comparable, in fact, to that of a cathode-follower. However, it should not be used with a power amplifier having a very low input impedance (less than 1000 ohms) such as the 35-watt "infinite-feedback" amplifier described some time ago in this magazine. It would be better to revamp the input circuit of such an amplifier so that it can be used with conventional preamps having a high output impedance.

Circuit Design

Earlier preamplifiers designed by one of the authors have always used a dual triode with a passive equalizer between the two sections, and have had a gain (at 1000 cps) of about 14. If a 10-millivolt cartridge is connected to such a preamplifier, the output signal will be around 0.14 volt. This is a rather inconvenient figure; it is too low to be used directly, and if it is passed through even a low-mu triode with a gain of 15, it attains the value of 2.1 volts, which is too high for the present purpose.

On the other hand, a 5879 low-noise pentode has, when properly connected, a gain of about 150, and when it is followed by an equalizer of the type to be shown, the over-all gain is 2.4. The output from a 10-millivolt cartridge is 0.024 volt, which can be passed through a medium-mu triode like a 12AY7 (gain around 25) to obtain a signal of 0.6 volt.

With input signals as small as those delivered by a magnetic cartridge, there is little to choose between two triodes and one pentode as far as distortion is concerned; both will be found negligibly small. The pentode has the advantage of low input capacity, an important consideration where the cartridge to be used has a high inductance. The pentode also permits simplification of the power-supply filtering requirements, since the signal from the cartridge is raised to a level high enough to be non-critical before anything in the way of equalizing is done to it.

Fig. 2 shows the circuit diagram of the complete preamplifier. The first stage is a 5879; the second, one-half of a 12AY7. Between these two stages is an equalizer switch essentially the same as that previously developed and considerably improved by Howell . The high output resistance of the first stage, however, permits the elimination of the large equalizer input resistor. The first two stages are used only when the input-selector switch is turned to one of the two available phono positions. When the switch is turned to a low-gain input position the grid of the first stage is grounded.

Tuners and other sources of signal to be connected to the low-gain inputs must be able to deliver at least 0.6 volt. The last two stages of the preamp have no gain; the first is a cathode-fol-

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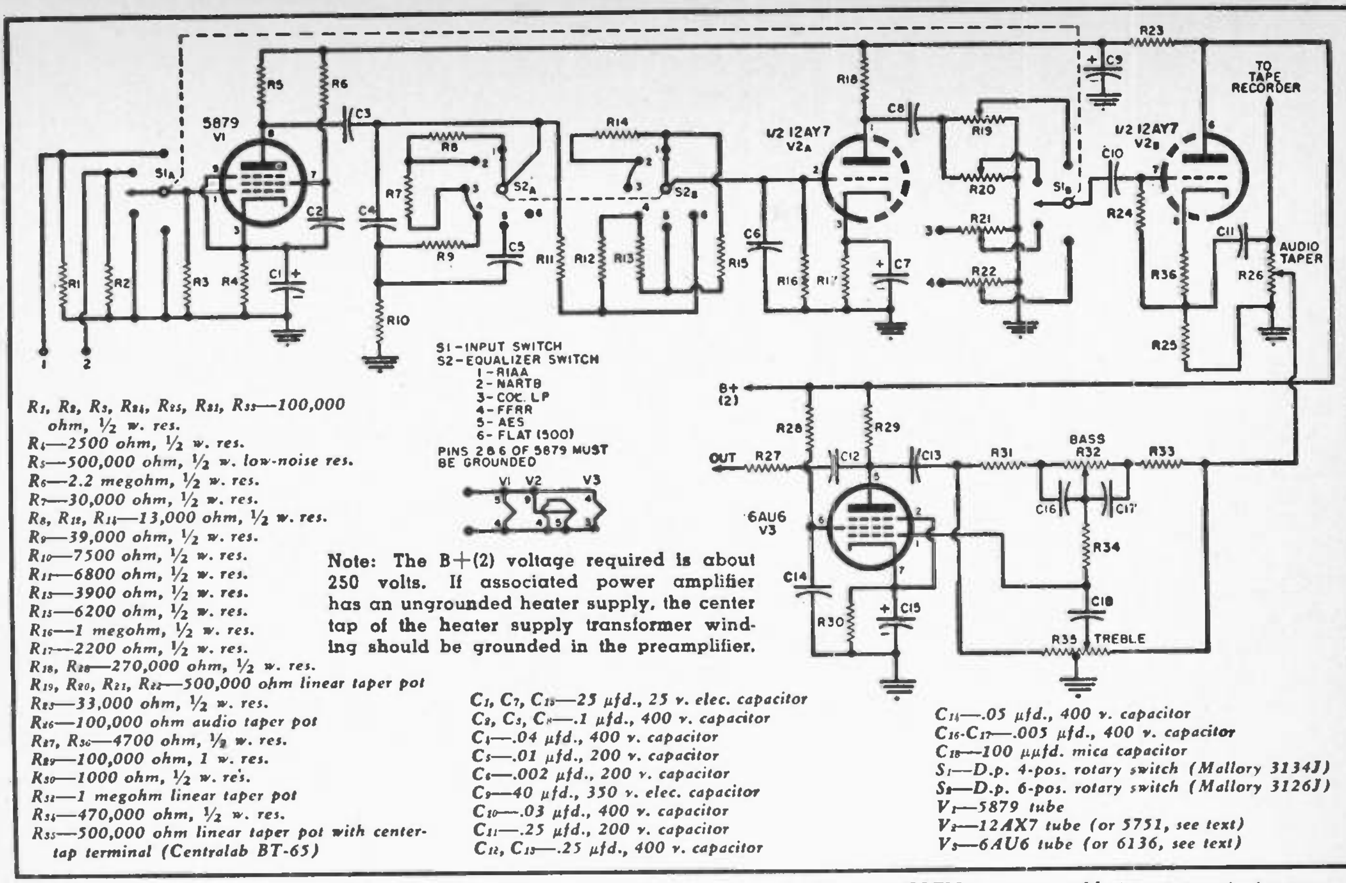


Fig. 2. Schematic of preamp. It can be used with one or two record players, an AM-FM tuner, or with two separate tuners.

lower to which the tape-recorder output is directly connected. A volume control of low resistance also follows this stage, and the slider of the control is attached directly to the tone-control circuit. It has been found that with a low-resistance volume control like the one used here, there is no interaction of tone- and volume-control functions.

The excellent tone-control circuit devised by Baxandall is used in exactly the form he recommended, including the high-gain pentode. The original article details the advantages of a pentode over a triode in this circuit. The gain of the tone-control stage is about 1.0 irrespective of the type of tube used in it but the distortion with the pentode is less than that resulting from the employment of a triode. It should be noted that the center tap of the treble control is connected directly to ground. Inserting a resistor here causes considerable loss of performance and is not recommended.

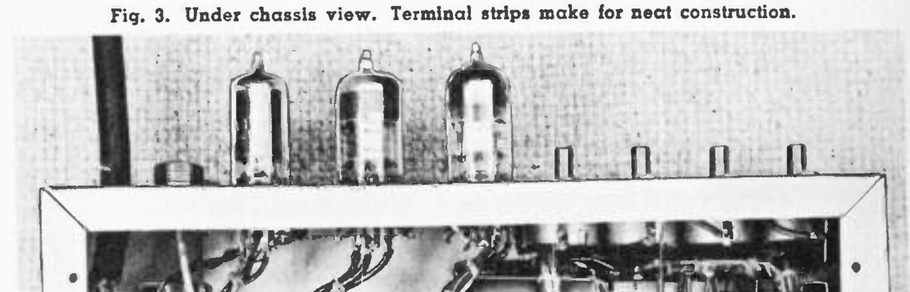
The output resistance of the preamplifier with tone controls in flat position is about 5000 ohms. The frequency response consequently remains unaffected even though very long cables are used to connect preamplifier to amplifier. As a matter of fact, a capacity of 800 μμfd. may be shunted across the output terminals before the response drops as much as 1 db at 20,000 cps.

Construction and Performance

The preamplifier was constructed on a 5x10x3-inch chassis in such a manner that it could be slipped into a finished wood case and used on a table top. Wiring was done mainly on 13-point *Cinch-Jones* terminal strips. Fig. 3 is an under-chassis view of the finished preamplifier. Except for the equalizer, parts tolerances are not critical. For the equalizer, resistors should be of 5% tolerance and capacitors should be checked with a bridge if at all possible.

Rotating either the input or equalizer switch causes no audible clicks in the output even with the volume control fully open and with one phono input unterminated. With the input selector on a low-gain position, the hum level of the preamp-amplifier combined is no greater than that of the amplifier alone. On a high-gain position, tube noise becomes audible when the volume control is wide open, but at normal settings it cannot be heard, and it is at any setting far less than the

Fig. 3. Under chassis view. Terminal strips make for neat construction.



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surface noise of the best LP discs. The r.m.s. sum IM distortion of the last two stages is, under any circumstances, entirely negligible, being on the order of 0.2% at three volts output with the tone controls in flat position, the volume control wide open, and a 27,000-ohm resistor shunted across the output terminals. The IM distortion of the entire preamp on phono position is difficult to measure because of the impossibility of leveling the response without affecting other factors, but it appears to be well under 0.6% with an input of about 30 mv.

For some purposes a somewhat higher gain on the phono inputs may be desirable, and this may be had by substituting a 12AX7 for the 12AY7. The substitution does not involve any compromise whatever, and the only result is to approximately double the gain of the unit on the high-gain input position. A very careful selection must, however, be made since many 12AX7's are quite microphonic. During the experimental work one case was observed in which mechanical feedback from the speaker to the 12AX7 was sufficient to cause singing at 500 cps even though the speaker cabinet was ten feet from the preamplifier! It is probably wise to obtain premium tubes (6136 for 6AU6, and 5751 for 12AX7) for particularly critical applications. Shock-mounting the tubes may also prove helpful.

A Plug-in "Presence" Equalizer

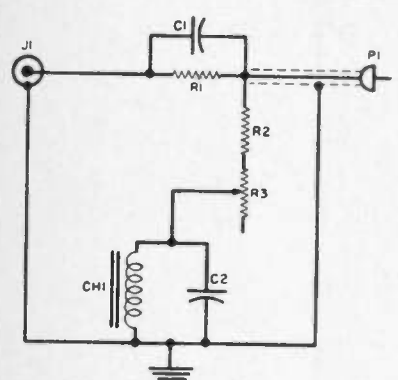
By ALLAN M. FERRES

THE mid-range, or "presence," equalizer is an interesting addition to any home high-fidelity installation. It furnishes a boost of up to 6 db at 3000 cycles without affecting the response at the extreme ends of the frequency range. This type of equalization is often used in film and disc recording to emphasize solo voices and instruments. When used on speech reproduction, the articulation factor, or "understandability," is increased without producing the annoyance of excessive sibilance.

In a home reproducing system the presence equalizer will not produce as marked an effect as the bass or treble controls, but it will tend to highlight most solo instruments and for this reason it is a worthwhile attachment.

The presence equalizer described here is a simple device which can be attached to the existing equipment between the equalizing preamp and the power amplifier.

The circuit is simple and uses only a few standard, readily available parts. As can be seen in the schematic, Fig. 1, the equalizer is basically a voltage divider, the series leg consisting of R_1, shunted by C_1, and the shunt leg, made up of a parallel tuned circuit, CH_1-C_2, in series with R_2 and R_3. As CH_1-C_2 is resonant at 3000 cycles, the impedance of the shunt leg is highest at this frequency and the loss in the voltage divider is minimum, furnishing the mid-range boost. The amount of the boost is controlled by the total value of R_2 and R_3 in series. The boost is 6 db when R_3 is set at minimum resistance and zero when set at maximum. C_1 is used to compensate for high-frequency losses caused by stray and cable capacities in the divider circuit.

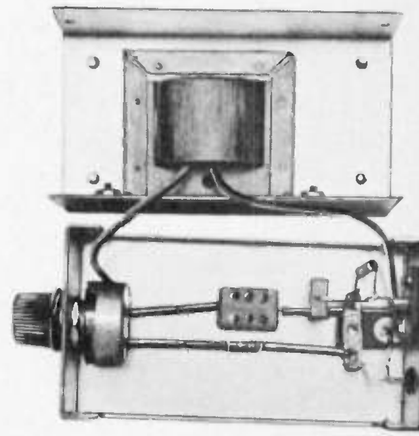


R_1—220,000 ohm, 1/2 w. res.
R_2—100,000 ohm, 1/2 w. res.
R_3—1 megohm reverse log taper or linear pot
C_1—75 μμfd. mica capacitor
C_2—100 μμfd. mica capacitor
CH_1—20 hy., 15 ma. choke (Stancor C-1515 or equiv.)
J_1—RCA-type phono jack
P_1—RCA-type phono plug

Fig. 1. Schematic of the equalizer unit.

The frequency response curves of Fig. 2 are obtained when the input resistor of the power amplifier is about 500,000 ohms, the usual value. If the input resistor is less than 470,000 ohms, it should be changed to this value. If the resistor is 1.0 megohm, a 1.0 megohm resistor can be connected across the output cable of the equalizer and no wiring change will be necessary in the power amplifier.

The length of the shielded leads to the preamp and the main amplifier should be kept as short as possible to reduce the chance of hum pickup and uncompensated high-frequency loss.



Two views of author's home-built equalizer. It is housed in a small box and uses standard, readily available parts.

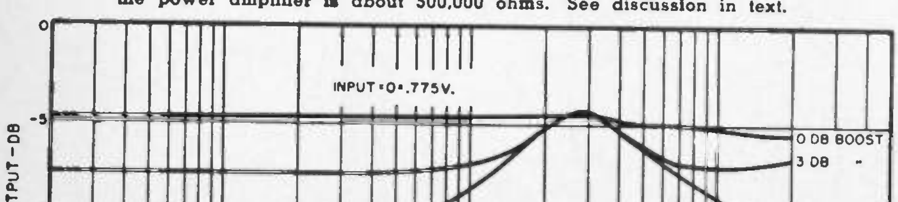
The construction of the presence equalizer can be seen in the photographs. CH_1 is mounted on the bottom section of the case and wired into the circuit with its flexible leads. The pot should be wired so that minimum resistance is obtained when the control shaft is in the full clockwise position. The winding-to-core capacitance of the *Stancor* C-1515 choke used in this equalizer causes the circuit to resonate at 3000 cycles when the choke is shunted by the 100 μμfd. capacitor. If another make of choke is used, the value of C_2 may have to be changed to produce the required resonant frequency.

The insertion loss varies between 4.8 db and 10.5 db, depending upon the amount of boost. The usual type of equipment used in home high-fidelity installations has enough spare gain to compensate for this loss.

As the choke is subject to hum pickup from the magnetic fields of power transformers and tape machine and record player motors, the equalizer should not be mounted permanently until it has been determined that its position in relation to such hum-producing fields is correct for minimum hum.

—30—

Fig. 2. Frequency response curve of equalizer when input resistor of the power amplifier is about 500,000 ohms. See discussion in text.



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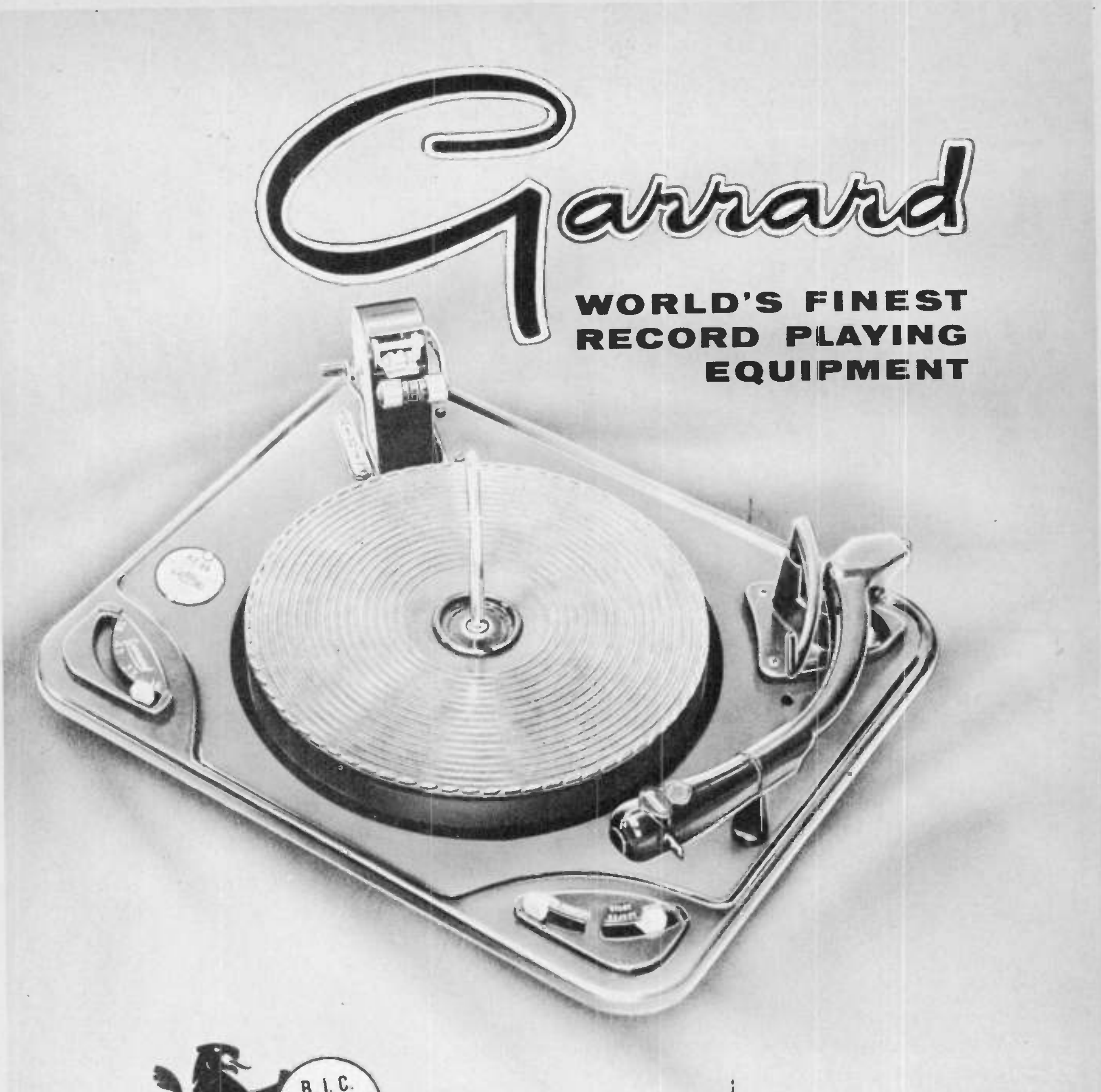

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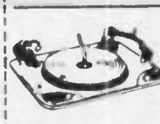

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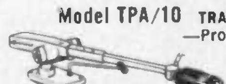

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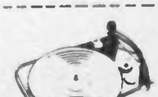

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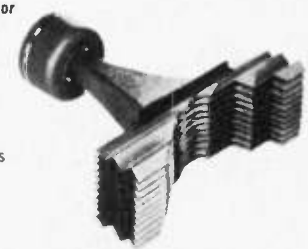

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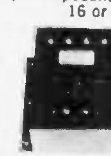
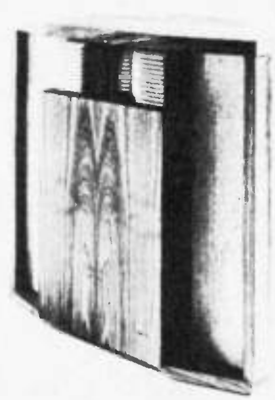


N1200 Impedance Compensated Network
Crossover . . . . 1200 cps
Impedance . . . . 16 ohms
Attenuation 12 db/octave

ENCLOSURES



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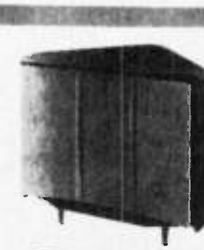

The Harlan / C39
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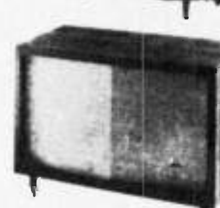

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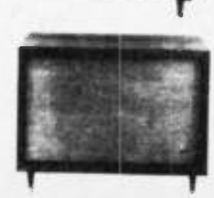

The C35 Console Reflex
Rigidly constructed, pleasing in appearance, conforming to optimum acoustical design. The C35 has long served as a standard of comparison. Will accommodate 12 or 15 inch **Signature** systems.

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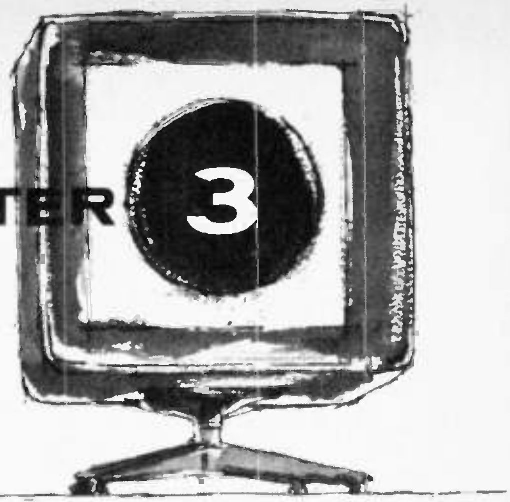
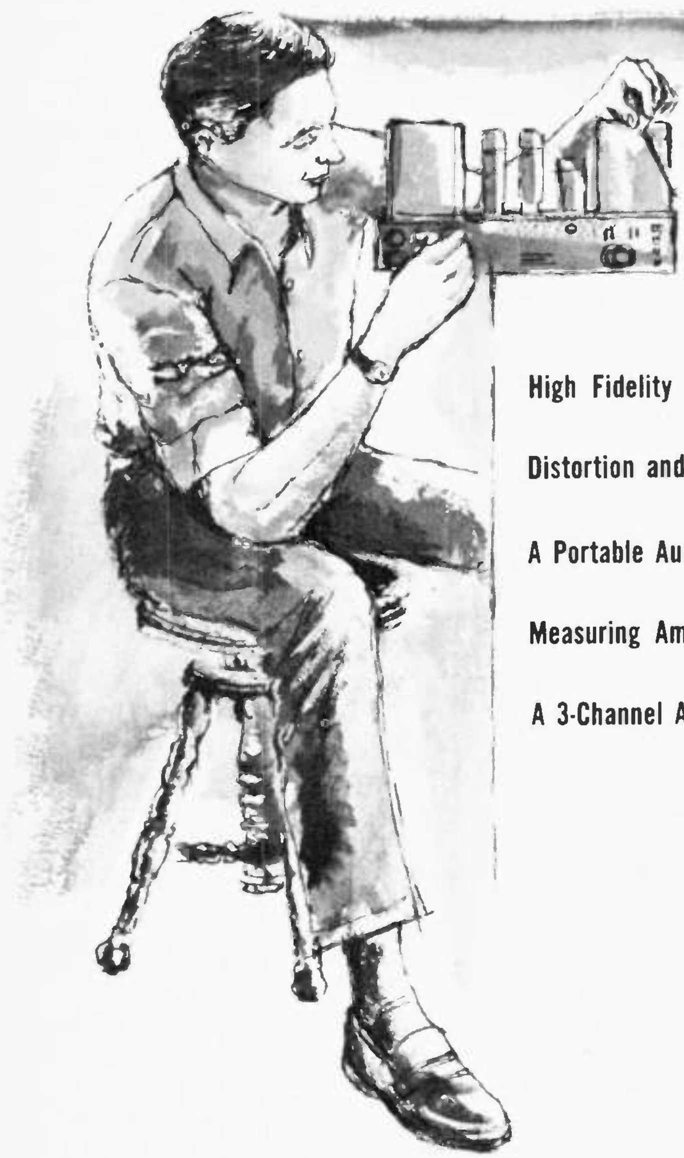


CHAPTER 3



AMPLIFIERS

World Radio History

High Fidelity Performance with Mullard's 520 Circuit

Over-all view of power amplifier. The three fuses are mounted on the left flange while the "on-off" switch appears on the right flange. The large housing (which is vented at top and bottom) directly behind the tubes covers the output and power transformers and filter choke. The other components are mounted below the chassis.

By E. J. PORTO*

The original British-designed power amplifier was conservatively rated at 20 watts. Actual tests showed an output of 36 watts with .2% harmonic distortion. Maximum output is with .3 volt input.

CONSIDERABLE international attention has been focused on a new audio output pentode, the EL34, recently introduced in England by *Mullard Ltd.*, in view of the many American power amplifier designs based around this new tube. The circuit described in this article is basically an "Ultra-Linear" design that was originally worked up by *Mullard Ltd.* and published in *"Wireless World."* According to published data, it was rated at 20 watts with a total harmonic distortion of .05% at rated output. Actual tests by American standards resulted in a rather surprising performance in that, instead of 20 watts output, we were able to obtain up to 36 watts with a total harmonic distortion of .2%.

Before going into the actual details of this power amplifier, let us review the basic requirements of amplifier designs that are important considerations for high-fidelity reproduction. Briefly, they are as follows:

1. Low harmonic and intermodulation distortion
2. Linear frequency response throughout the audible range
3. Good response to transient signals
4. Low phase shift
5. Low hum and noise level
6. Enough power output to allow peaks to be reproduced without overloading
7. Low output resistance to provide electrical damping for the loudspeaker system
8. Stability under feedback conditions

Amplifier Designs

A low level of inherent distortion can be obtained in a push-pull triode output stage operating under virtually Class A conditions. It is found that with 25-watt pentodes or tetrodes, wired as triodes, a power output of from 12-15 watts can be easily obtained with harmonic distortion levels below 1%, using a supply voltage of from 430-450 volts.

The maximum power output and the corresponding distortion vary appreciably with the value of load impedance. Fig. 2 illustrates typical performance of the *Mullard* EL34 high slope output pentode, triode-connected, in a push-pull stage operating slightly below its rated plate dissipation of 25 watts.

Increasing interest is being shown in circuits employing distributed loading ("Ultra-Linear" operation) of the output stage (Fig. 1). These circuits apply negative feedback in the output stage itself. In the simplest form, the screen grids of the output tubes are fed from taps on the primary of the output transformer. The stage can be considered as one in which negative feedback is applied in a non-linear manner *via* the screen grids. The characteristics of the distributed load stage are intermediate between those for pentode and triode operation, approaching triode operation as the per-

TABLE 1.
THE MULLARD EL34 TUBE UNDER VARIOUS OPERATING CONDITIONS

MODE OF OPERATION	OPERATING CONDITIONS					IM DISTORTION (in per-cent at)		
	E_p (volts)	E_{g2} (volts)	R_k (ohms)	Impedance (P-P, ohms)	R_{g2} (ohms)	10 w.	14 w.	36 w.
Triode-connected	400		470 (each)	10K		.4	.6	....
Distributed-load ("Ultra-Linear")	400	400	470 (each)	6.6K	1000 (each)	.5	.6	.8
Pentode-connected (push-pull)	330	330	130 (common)	3.4K	470 (common)	1.5	2.0	4.0

*International Electronics Corporation, 81 Spring Street, New York, New York.

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centage of the primary winding common to plate and screen-grid circuits increases. It is found that under optimum conditions about two-thirds of the power-handling capacity of the corresponding pentode stage can be used with greatly reduced distortion, while at power levels corresponding to triode operation, a similar order of distortion is obtained. At the same time, the output impedance is reduced to a level approaching that obtained when a conventional push-pull triode stage is used. Such a stage can, therefore, be used with pentodes of the 25-watt class in high-quality amplifiers designed for power outputs well in excess of 30 watts, the over-all power efficiency being much greater than with triode operation.

Table 1 is a comparison of triode, pentode, and distributed-load operation for the EL34. For tubes of the EL34 type, comparison with triode operation is of most interest. It will be seen that distributed-load operation enables the power-handling capacity to be more than double that possible with triode operation while, at the same time, distortion in the stage can be held to a very low level. Although with a common winding ratio of 0.2 to 1 the distortion level is comparable to triode conditions, it has been found that appreciable improvement is obtained at higher power outputs if the common winding ratio is further increased.

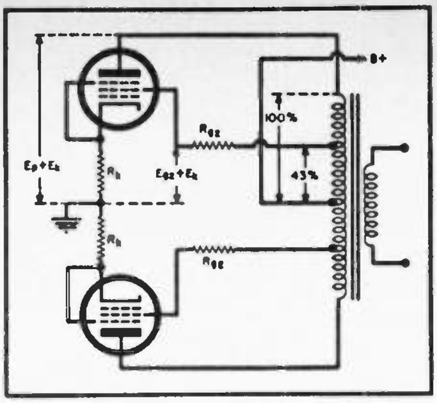


Fig. 1. Theoretical design for "Ultra-Linear" circuitry used in amplifier. The percentage figure is turns ratio.

From the figures of Table 1, little advantage would appear to be gained by further approaching triode conditions. There are, however, at least two advantages in using a tap at about 40% of primary turns, particularly with the EL34 where a high power output is still available. In the first place, almost identical performance is obtained under cathode and fixed bias conditions since with a closer approach to Class A triode operation, variations in plate and screen grid currents are reduced when the stage is driven. Secondly, as with normal triode operation, power output and distortion are less dependent on the precise value of the load impedance. With a primary tap

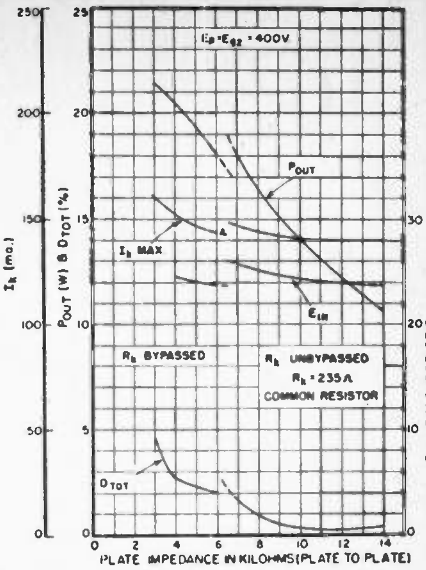


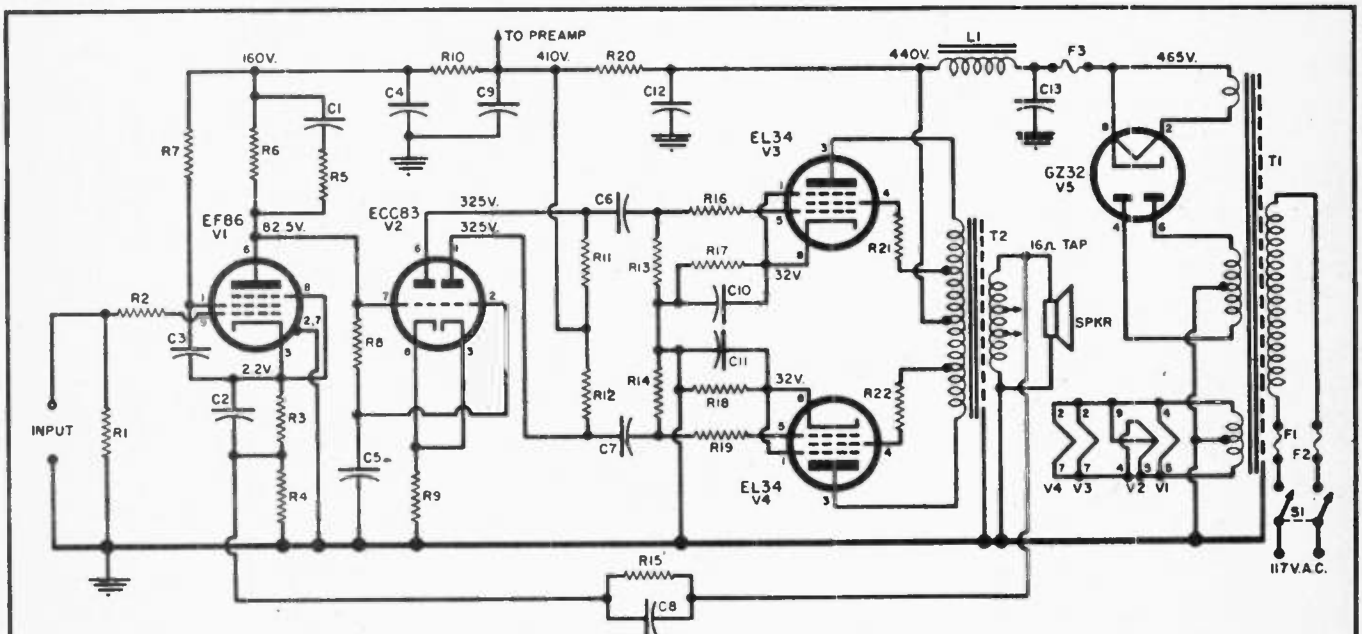
Fig. 2. Performance curves of two triode-connected EL34 tubes in push-pull. Refer to text for complete discussion.

of about 40% of the turns, little change in performance is produced by a change in the plate-to-plate load impedance of 6000 to 9000 ohms. In addition the output impedance of the stage is still further reduced by the use of the larger common winding ratio.

Circuit Arrangements

The next-to-the-last-stage of the amplifier must be capable of providing a

Fig. 3. Schematic of Mullard 520 amplifier. All parts are available at local parts jobbers. Maximum current drain for preamp is 40 ma.



R_1, R_8—1 megohm, ½ w. res.
R_2, R_5—4700 ohm, ½ w. res.
R_3, R_{16}, R_{19}—2200 ohm, ½ w. res.
R_4—100 ohm, ½ w. res.
R_6—100,000 ohm, ½ w. res.
R_7—390,000 ohm, ½ w. res.
R_9—8200 ohm, ½ w. res.
R_{10}—270,000 ohm, ½ w. res.
R_{11}, R_{12}—180,000 ohm, ½ w. res. (Matched ±5%, R_{12} to be greater than R_{11})
R_{13}, R_{14}—470,000 ohm, ½ w. res. (Matched ±5%)
R_{15}—8200 ohm, ½ w. res. ±5% (for 16-ohm load)
R_{17}, R_{18}—470 ohm, 5 w. wirewound res. ±5% (critical value)
R_{20}—15,000 ohm, ½ w. res.
R_{21}, R_{22}—1000 ohm, ½ w. res.
C_1—47 μμfd. disc ceramic capacitor
C_2—50 μfd., 12 v. elec. capacitor
C_3—.05 μfd., 400 v. capacitor
C_4, C_9—8 μfd., 450 v. elec. capacitor
C_5—.25 μfd., 400 v. capacitor
C_6, C_7—.05 μfd., 400 v. capacitor
C_8—220 μμfd. ceramic capacitor (for 16-ohm load)
C_{10}, C_{11}—50 μfd., 50 v. elec. capacitor
C_{12}, C_{13}—8 μfd., 500 v. elec. capacitor
L_1—8 hy, 180 ma., 145 ohm filter choke (Thordarson T-20-C-54 or UTC H-62)
F_1, F_2—2 amp fuse
F_3—250 ma. fuse
S_1—D.p.s.t. switch (Power "on-off")
T_1—Power trans. 400-0-400 v. @ 200 ma.; 5 v. @ 3 amps.; 6.3 v. c.t. @ 5 amps. (Stancor PM-8412 or equiv.)
T_2—Output trans. (Any of the following may be used: Acro Products Company TO-300, 6600 ohms p-to-p, 4-8-16 ohms; Chicago-Standard Transformer Corp. A-8072, 7600 ohms p-to-p, 4-8-16 ohms; Dyna Company A-420, 6600 ohms p-to-p, 8-16 ohms; Partridge P3878, 6600 ohms p-to-p, 4-16 ohms)
V_1—EF86 tube (Mullard)
V_2—ECC83 tube (Mullard)
V_3, V_4—EL34 tube (Mullard)
V_5—GZ32 or GZ34 tube (Mullard)

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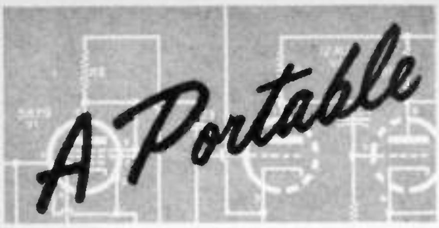
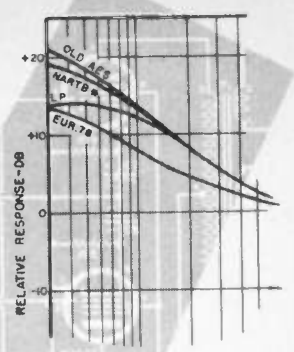
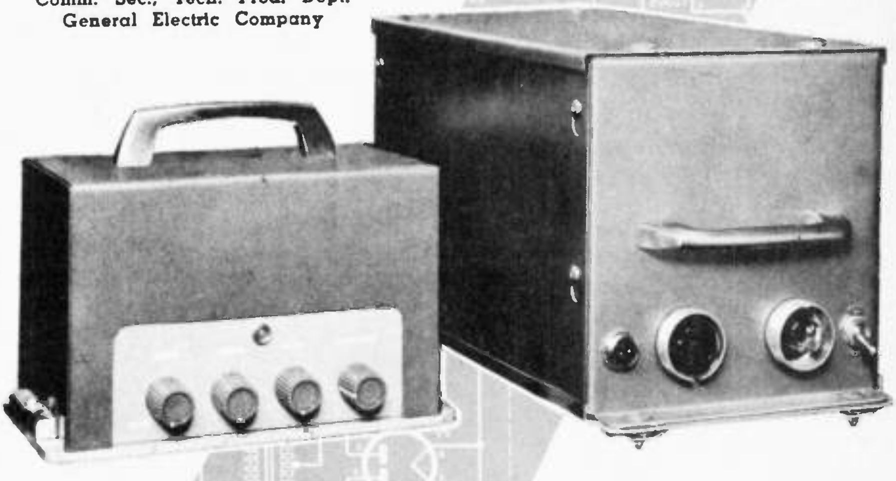
loop feedback from output/input, instability will occur if the loop gain ex-



A Portable Audio Amplifier System



By
RICHARD T. BUESING
Comm. Sec., Tech. Prod. Dept.
General Electric Company



Matching preamp with operating controls is shown at the left, with the associated power amplifier unit at the right.

Can be operated from 6- or 12-volt battery supply or from a.c. line. Power amplifier and matching preamp provide a 10-watt audio output with low distortion.

THE literature is replete with designs for amplifiers and preamplifiers whose main application is for living room use. However, there are many instances when the audio enthusiast is called upon to furnish an amplifier for parties, mobile public-address work, and special events. He seldom relishes the task of dismantling the living room amplifier, providing suitable enclosures, and rigging up the requisite power supplies. It is for such applications that the author has designed a "universal" system that is convenient for such uses besides serving as a high quality entertainment or public-address amplifier for the home.

Design Specifications

The first consideration was the rated-power output of the amplifier. If the required acoustic power is known, the electrical power may be found from the relationship:

Electrical Power = Acoustic Power/ Speaker Efficiency.

It is unfortunate that the two expressions on the right side of the equation are not easily established. Let us, therefore, make certain preliminary assumptions which will lead to expected power requirements. The average orchestral intensity level has been found[1] to be approximately 75 db above 10^{-16} watt per square centimeter where 0 db is taken as the level of the weakest 1000 cps tone that can be heard by a normal ear in a silent room. Peak orchestral intensities up to 110 db have been measured with bass-drum-solo intensities reaching approximately 113 db.

The acoustic power required to produce a level of 75 db depends upon the volume and reverberation time of the room in which it is to be heard. Mitchell[2] has prepared data from which acoustic power may be determined if the room dimensions and characteristics of the reflecting surfaces within the room are known. For a room 30′ x 20′ x 10′ (such as a game room), the acoustic power needed is on the order of 10^{-5} watts. Assuming a speaker efficiency of 5%, the electrical power is found to be $10^{-5}/(5 \times 10^{-2}) =$ 0.0002 watt. If the peak intensity of 115 db is to be reproduced, 2 watts of electrical power will be needed; and, with a 6 db margin, an amplifier with 8 watts of electrical power will be sufficient. Since it is customary to rate amplifiers in this power region in multiples of 5, the amplifier to be described was designed for 10 watts output.

It is recognized that outdoor public-address work means that the intensity level will vary inversely as the square of the distance from the source, thus demanding more power for large areas. With re-entrant horns or compound-diffraction projectors, conversion efficiencies of 30% to 40% may be expected as compared to the 5% more typical of cone speakers, thus helping to compensate for the added power required.

It was decided, at the outset, that no compromise in performance would be tolerated in order to achieve a compact and economical design. Since the best ears can perceive frequencies from 16 to 22,000 cps at intensities near the threshold of pain, there is no reason to exceed this bandwidth and very little reason to attempt to equal it. It was decided to hold to broadcast practice and design the main amplifier with a frequency response of 30 to 15,000 cps ±1 db from a 1000 cps reference.

Bandwidth and power output specifications, in themselves, are relatively easy to meet until a distortion specification at rated power output is set. With a total harmonic distortion on the order of 1%, few critical listeners are offended, thus 1% over the 30 to 15,000 cps band was chosen as the design specification. The intermodulation distortion compared to 1% harmonic distortion is approximately 4%. It is extremely important, in any amplifier design, not to be content with measuring the single-frequency distortion at the output. Internal feedback and non-linearities may cause cancellations that yield meaningless results when only the output is measured. It can be shown philosophically and demonstrated experimentally that two stages whose harmonic distortions are 3% may result in a measured distortion of, say, 1% at the output. The r.m.s. of the individual stage distortion, however, indicates the true performance of the amplifier. This is the reason for measuring the intermodulation, which takes into account the sum and difference products of each stage.

Finally, it was desired that the hum and noise level at the output of the main amplifier be at least 70 db below rated output.

In the preamplifier, it was deemed desirable to include four controls: "Loudness/Volume," "Treble," "Bass,"

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centage of the primary winding common to plate and screen-grid circuits increases. It is found that under optimum conditions about two-thirds of the power-handling capacity of the corresponding pentode stage can be used with greatly reduced distortion, while at power levels corresponding to triode operation, a similar order of distortion is obtained. At the same time, the output impedance is reduced to a level approaching that obtained when a conventional push-pull triode stage is used. Such a stage can, therefore, be used with pentodes of the 25-watt class in high-quality amplifiers designed for power outputs well in excess of 30 watts, the over-all power efficiency being much greater than with triode operation.

Table 1 is a comparison of triode, pentode, and distributed-load operation for the EL34. For tubes of the EL34 type, comparison with triode operation is of most interest. It will be seen that distributed-load operation enables the power-handling capacity to be more than double that possible with triode operation while, at the same time, distortion in the stage can be held to a very low level. Although with a common winding ratio of 0.2 to 1 the distortion level is comparable to triode conditions, it has been found that appreciable improvement is obtained at higher power outputs if the common winding ratio is further increased.

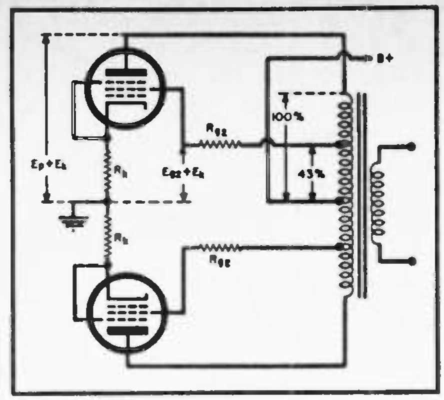

Fig. 1. Theoretical design for "Ultra-Linear" circuitry used in amplifier. The percentage figure is turns ratio.

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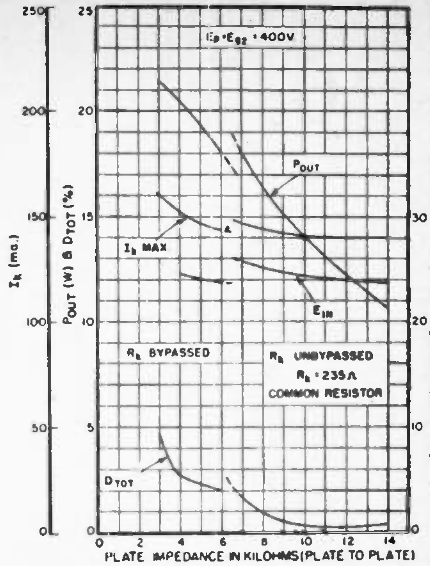

Fig. 2. Performance curves of two triode-connected EL34 tubes in push-pull. Refer to text for complete discussion.

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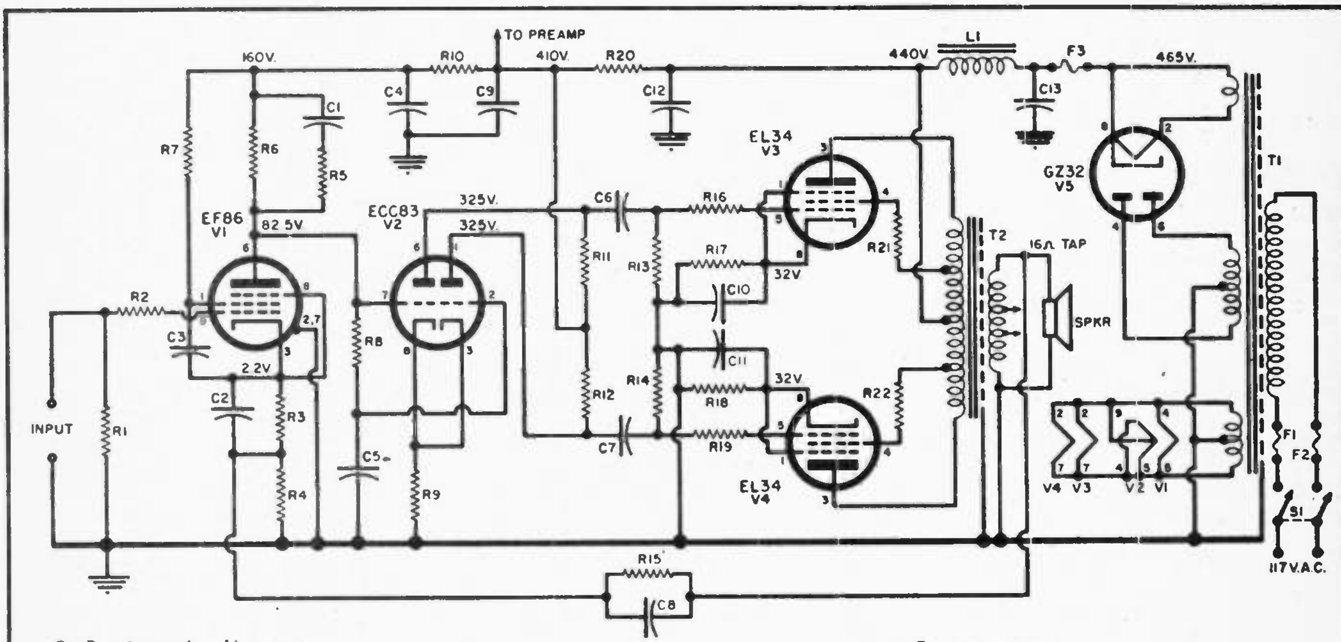
Fig. 3. Schematic of Mullard 520 amplifier. All parts are available at local parts jobbers. Maximum current drain for preamp is 40 ma.



R_1, R_9—1 megohm, ½ w. res.
R_2, R_5—4700 ohm, ½ w. res.
R_3, R_{16}, R_{19}—2200 ohm, ½ w. res.
R_4—100 ohm, ½ w. res.
R_6—100,000 ohm, ½ w. res.
R_7—390,000 ohm, ½ w. res.
R_8—8200 ohm, ½ w. res.
R_{10}—270,000 ohm, ½ w. res.
R_{11}, R_{12}—180,000 ohm, ½ w. res. (Matched ±5%, R_{12} to be greater than R_{11})
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R_{15}—8200 ohm, ½ w. res. ±5% (for 16-ohm load)
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C_1—47 μμfd. disc ceramic capacitor
C_2—50 μfd., 12 v. elec. capacitor
C_3—.05 μfd., 400 v. capacitor
C_4, C_9—8 μfd., 450 v. elec. capacitor
C_5—.25 μfd., 400 v. capacitor
C_6, C_7—.05 μfd., 400 v. capacitor
C_8—220 μμfd. ceramic capacitor (for 16-ohm load)
C_{10}, C_{11}—50 μfd., 50 v. elec. capacitor
C_{12}, C_{13}—8 μfd., 500 v. elec. capacitor
L_1—8 hy, 180 ma., 145 ohm filter choke (Thordarson T-20-C-54 or UTC H-62)
F_1, F_2—2 amp fuse
F_3—250 ma. fuse
S_1—D.p.s.t. switch (Power "on-off")
T_1—Power trans. 400-0-400 v. @ 200 ma.; 5 v. @ 3 amps.; 6.3 v. c.t. @ 5 amps. (Stancor PM-8412 or equiv.)
T_2—Output trans. (Any of the following may be used: Acro Products Company TO-300, 6600 ohms p-to-p, 4-8-16 ohms; Chicago-Standard Transformer Corp. A-8072, 7600 ohms p-to-p, 4-8-16 ohms; Dyna Company A-420, 6600 ohms p-to-p, 8-16 ohms; Partridge P3878, 6600 ohms p-to-p, 4-16 ohms)
V_1—EF86 tube (Mullard)
V_2—ECC83 tube (Mullard)
V_3, V_4—EL34 tube (Mullard)
V_5—GZ32 or GZ34 tube (Mullard)

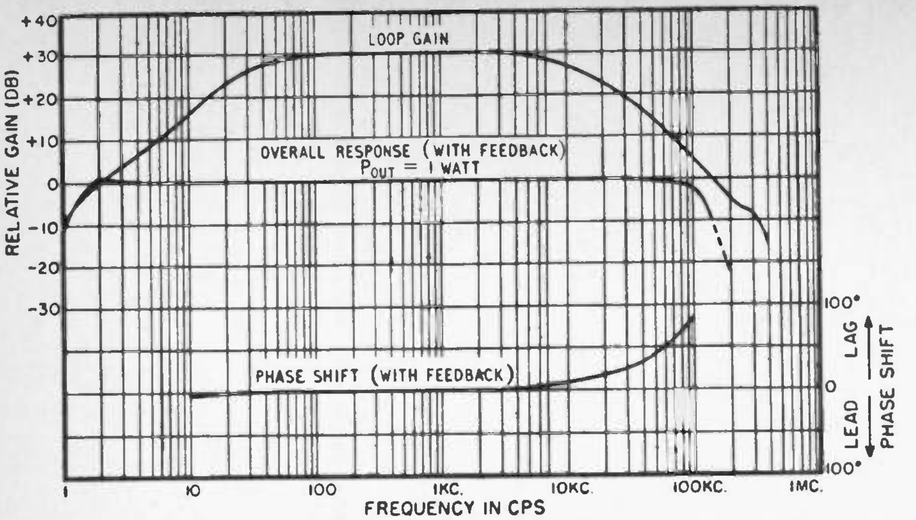


Fig. 4. Loop gain, frequency response, and phase characteristics. Over-all frequency response at 36 watt operation is flat, 30 to 20,000 cycles-per-second.

TABLE 2.
SUMMARY OF PERFORMANCE CHARACTERISTICS OF THE POWER AMPLIFIER

Power output:	36 watts; 30 to 20,000 cps
Frequency response: (at 36 watts)	within 1 db from 20 to 20,000 cps
Harmonic distortion: (at 400 cps)	.05% at 20 watts, .2% at 36 watts
IM distortion: (40 to 10,000 cps, 4:1 ratio)	.8% with peak corresponding to 36 watts sine-wave power
Hum and noise:	−89 db (relative to 36 watts)
Sensitivity:	.3 volt for 36 watt output
Damping factor:	50

well balanced push-pull drive of adequate amplitude and low distortion content. With the EL34 the maximum drive voltage required is approximately 2 x 25 volts r.m.s. Input voltage requirements are similar for triode, pentode, or distributed-load operation.

Bearing in mind the need to insure stability when feedback is applied over the whole amplifier, the circuit should contain the minimum number of stages in order to reduce phase shifts. If the function of phase splitting and amplification can be combined in the next-to-the-last-stage, so much the better. This can be conveniently achieved by using a cathode-coupled form of phase splitter. A high degree of balance is possible with this circuit, combined with a low distortion level at maximum drive to the output stage. By using a high-impedance double triode, an effective gain of about 25 times can be obtained simultaneously. This, combined with a preceding high-gain stage, enables a high over-all sensitivity to be obtained, even when a large amount of negative feedback is used. A high sensitivity in the main amplifier enables the output voltage requirements of preamplifier and tone control circuits to be reduced, thereby enabling low distortion to be more easily achieved in these circuits. It should be remembered that circuits preceding the main amplifier must be capable of handling, without appreciable distortion, voltages which are much greater than those necessary to load the amplifier fully.

With the use of such tubes as the EF86, which is particularly suited for use in a high-sensitivity input stage due to its low hum and noise levels, it is found that when feedback is applied, input sensitivities of 100 to 300 millivolts for 36-watt output can be achieved while keeping hum and noise levels extremely low.

In an amplifier employing single-loop feedback from output/input, instability will occur if the loop gain exceeds unity at frequencies for which the total phase shift around the loop becomes either 0° or 360° and so renders the feedback signal in-phase with the input. The conditions for negative feedback imply a phase change of 180° so that instability is approached as the additional phase shift in the amplifier and feedback network approaches 180°.

It is, therefore, necessary to control the amplifier characteristics over a frequency range greatly in excess of the designed working band. As the degree of feedback increases, control becomes more difficult and is usually limited by the leakage inductance, self-capacitance, and primary inductance of the output transformer. It is quite difficult in practice to provide a constant and high level of feedback over the whole audible range in a 3- or 4-stage amplifier where the main feedback loop includes the whole circuit and the output transformer. An adequate margin of stability in such circumstances is very difficult to obtain. Thus it is more usual to find that the effective feedback decreases towards the upper and lower audible frequencies. The *Mullard* 520 circuit is especially designed and engineered to maintain a constant degree of feedback throughout the audible range.

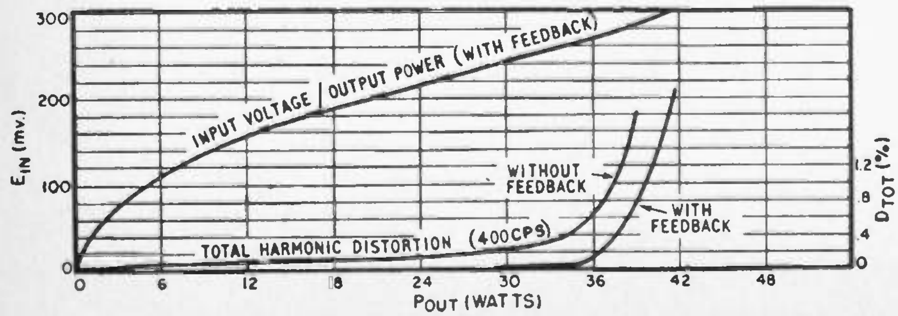
The performance of any high-quality amplifier is, ultimately, dependent on the quality of the output transformer. The use of distributed-load conditions does not modify the essential requirement of a first-class component; on the contrary, the output transformer may be a more critical component since precise balance of primary windings must be maintained.

We can summarize by stating that with the introduction of distributed-load operation using the *Mullard* EL34 output pentode we can design efficient high-quality amplifiers with very high power handling capacities to reproduce the widest dynamic range of modern program sources.

Construction Details

The plate-to-plate loading of the output stage is 6600 ohms and with a feed voltage of approximately 440 at the center-tap of the output transformer primary the combined anode and screen-grid dissipation of the output tubes is 28 watts per tube. With the particular screen-grid-to-plate-turns ratio used, it has been found that improved linearity is obtained at power levels above 15 watts when resistors on the order of 1000 ohms are inserted in the screen-grid feeds. The slight reduction in peak power-handling capacity which results is not significant in practice. Separate cathode-bias resistors are used to limit the out-of-balance d.c. current in the output transformer primary; the use of further d.c. balancing arrangements in the output stage has not been considered necessary primarily because of the uniform characteristics of the EL34. It is necessary, in this type of

Fig. 5. Harmonic distortion and input/output characteristics of the amplifier.



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output stage, that the cathodes be bypassed to ground even when a common cathode resistor is used.

The power supply is conventional and uses a *Mullard* GZ32 or GZ34 indirectly-heated, full-wave rectifier with capacitor input filter.

The driver stage uses a *Mullard* ECC83 twin-triode and fulfills the combined function of phase splitter and driver amplifier. It is of the cathode-coupled type and enables a high degree of push-pull balance to be obtained.

The first stage is a high-gain pentode voltage amplifier using an EF86 low-hum pentode. High-stability carbon resistors are used in plate, screen-grid, and cathode circuits and give appreciable improvement in measured background noise level as compared with ordinary carbon resistors. This stage is d.c.-coupled to the input grid of the phase splitter in order to minimize low-frequency phase shift in the amplifier and improve low-frequency stability when feedback is applied.

Despite the high degree of negative feedback used in the present design, an adequate margin of stability has been achieved. Complete stability is maintained under open-circuit conditions in this circuit. An increase in feedback of at least 10 db, obtained by reducing the value of R_{15}, should be possible before signs of high-frequency instability occur. The loop gain, overall frequency response, and phase shift characteristics of the whole amplifier are shown in Fig. 4.

The harmonic distortion of this amplifier at 400 cps, measured without feedback under resistive load conditions, is shown in Fig. 5. The distortion curve towards the overload point is also shown for feedback conditions. At the 20-watt level the distortion level without feedback is well below 1% and with feedback applied falls to below 0.05%. Harmonic distortion at 400 cps reaches 0.2% at approximately 36 watts output. The loop gain characteristics are such that at least 20 db feedback is maintained from 15 to 25,000 cps.

Measurement of intermodulation products has been made, using a carrier frequency of 10,000 cps, and a modulating frequency of 40 cps, with a ratio of 40 to 10,000 cps amplitudes of 4:1. With the combined peak amplitude of the mixed output at a level corresponding to the peak sine wave amplitude at 36 watts r.m.s. power, intermodulation products expressed in r.m.s. terms totaled 0.8% of the 10,000 cps carrier amplitude.

The sensitivity of the amplifier is approximately 0.3 volt for 36 watts output. The background level in this amplifier was 89 db below at 36 watts, measured with a source resistance of 10,000 ohms.

—30—

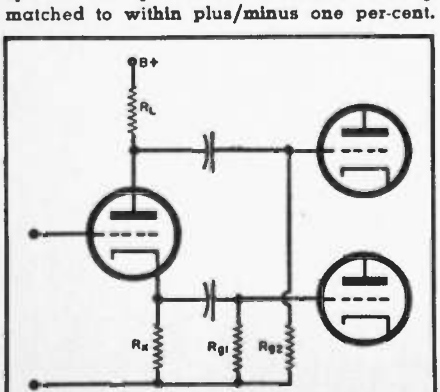
DISTORTION & PHASE-SPLITTER UNBALANCE IN PUSH-PULL AMPLIFIERS

By **NATHAN O. SOKAL**
Massachusetts Institute of Technology

ALMOST all of the published material on high-fidelity audio amplifiers stresses the need for precise balancing of the phase splitter used in push-pull amplifiers. Various authors have their pet phase-splitters; the one most commonly used is the split-load circuit shown in Fig. 1. The cathode and plate resistors, R_K and R_L, are usually matched within 1 per-cent, as are also the following grid resistors, R_{g1} and R_{g2}. The purpose of the matching is to get exactly equal a.c. voltages of opposite phase from the two outputs of the phase splitter. This precise balancing is quite useless and unnecessary—and even tends to defeat its own purpose. Why? Let us examine the reasons behind the balancing.

Push-pull connection of the output tubes cancels even-harmonic distortion generated by the tubes if their grid drives, plate loadings, and d.c. operating conditions are the same. This, then, is the reason behind the desire for equal drive on the tubes. However, the reasoning is based on the assumption that *the output tubes have identical transfer characteristics.* In actual practice, this is likely to be far from true. An idea of the range of characteristics from one tube to another of the same type can be had from the JAN specifications for the 6L6, a popular audio-output tube. A brand-new 6L6 can have a g_m between 5200 and 6800 micromhos (under standard test conditions) and the "end of life" is considered to be when the g_m drops to 4500 micromhos. There can be a difference in g_m of as much as 40 per-cent between two "good" 6L6's. Of course, they are not *all* that bad, but you get the idea.

Fig. 1. A widely used split-load phase splitter. The pairs of resistors are usually matched to within plus/minus one per-cent.



It is true that d.c. balancing of the output tubes tends to equalize their characteristics, but this is only a partial correction. The result of the remaining difference between the tubes is simply that the even harmonics are not completely cancelled. (It should also be noted that for exactly the same reason, the driver tubes can add still more noncancellation to that caused by the output tubes.) The net effect of all this is that the "push" and "pull" halves of the amplifier can easily have gains differing by 50 per-cent. What, then, is the sense of equalizing the phase-splitter outputs to within 1 percent?

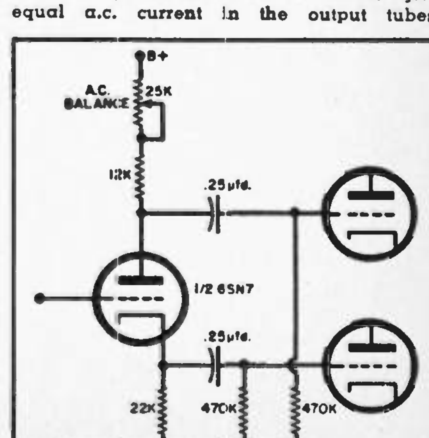
Now, if you want more nearly perfect cancellation of distortion (and it could be a moot point), what you can do is make the phase-splitter outputs enough unequal to compensate for the unequal gains of the "push" and "pull" halves of the amplifier. This can be done by making one of the phase-splitter loads adjustable, as shown in Fig. 2. The "a.c. balance" potentiometer is then set to give equal a.c. currents in the output tubes. (This is approximately the condition for best cancellation of distortion.) The results of this setup on an "Ultra-Linear Williamson" amplifier are shown in Fig. 3, where harmonic distortion of a 1 kc. input is plotted against power output for two conditions:

1. Equal resistors in the phase-splitter (dotted curve)
2. Potentiometer set for a.c. current balance in the output tubes (solid curve)

In both cases the direct currents were

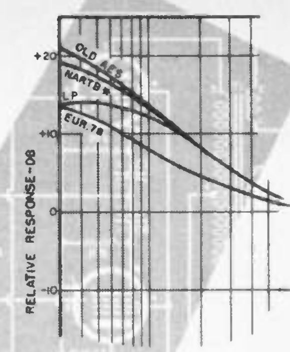
(*Continued on page* 65)

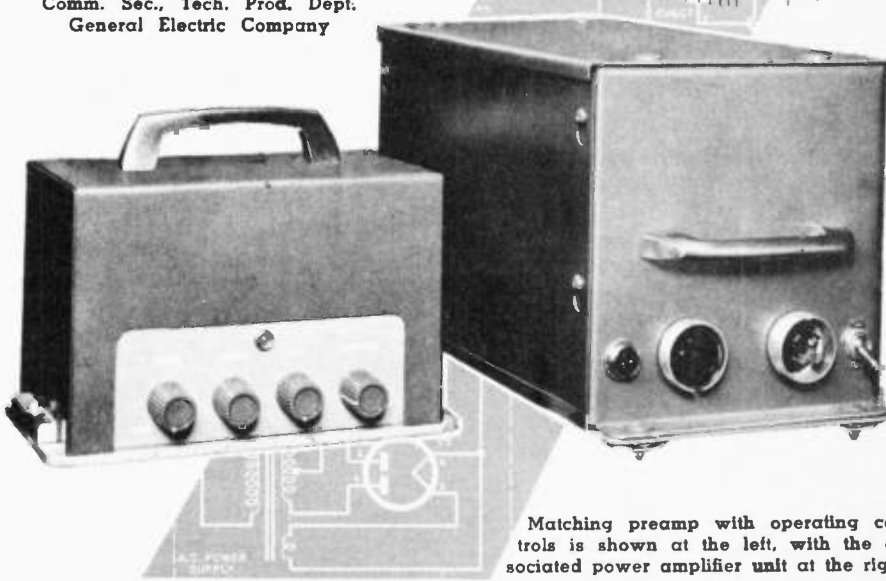
Fig. 2. Adjustable phase splitter. The "a.c. balance" potentiometer is set to give equal a.c. current in the output tubes.



A Portable Audio Amplifier System

By
RICHARD T. BUESING
Comm. Sec., Tech. Prod. Dept.
General Electric Company





Matching preamp with operating controls is shown at the left, with the associated power amplifier unit at the right.

Can be operated from 6- or 12-volt battery supply or from a.c. line. Power amplifier and matching preamp provide a 10-watt audio output with low distortion.

THE literature is replete with designs for amplifiers and preamplifiers whose main application is for living room use. However, there are many instances when the audio enthusiast is called upon to furnish an amplifier for parties, mobile public-address work, and special events. He seldom relishes the task of dismantling the living room amplifier, providing suitable enclosures, and rigging up the requisite power supplies. It is for such applications that the author has designed a "universal" system that is convenient for such uses besides serving as a high quality entertainment or public-address amplifier for the home.

Design Specifications

The first consideration was the rated-power output of the amplifier. If the required acoustic power is known, the electrical power may be found from the relationship:

Electrical Power = Acoustic Power/ Speaker Efficiency.

It is unfortunate that the two expressions on the right side of the equation are not easily established. Let us, therefore, make certain preliminary assumptions which will lead to expected power requirements. The average orchestral intensity level has been found[1] to be approximately 75 db above 10^{-16} watt per square centimeter where 0 db is taken as the level of the weakest 1000 cps tone that can be heard by a normal ear in a silent room. Peak orchestral intensities up to 110 db have been measured with bass-drum-solo intensities reaching approximately 113 db.

The acoustic power required to produce a level of 75 db depends upon the volume and reverberation time of the room in which it is to be heard. Mitchell[2] has prepared data from which acoustic power may be determined if the room dimensions and characteristics of the reflecting surfaces within the room are known. For a room 30′ x 20′ x 10′ (such as a game room), the acoustic power needed is on the order of 10^{-5} watts. Assuming a speaker efficiency of 5%, the electrical power is found to be $10^{-5}/(5 \times 10^{-2}) =$ 0.0002 watt. If the peak intensity of 115 db is to be reproduced, 2 watts of electrical power will be needed; and, with a 6 db margin, an amplifier with 8 watts of electrical power will be sufficient. Since it is customary to rate amplifiers in this power region in multiples of 5, the amplifier to be described was designed for 10 watts output.

It is recognized that outdoor public-address work means that the intensity level will vary inversely as the square of the distance from the source, thus demanding more power for large areas. With re-entrant horns or compound-diffraction projectors, conversion efficiencies of 30% to 40% may be expected as compared to the 5% more typical of cone speakers, thus helping to compensate for the added power required.

It was decided, at the outset, that no compromise in performance would be tolerated in order to achieve a compact and economical design. Since the best ears can perceive frequencies from 16 to 22,000 cps at intensities near the threshold of pain, there is no reason to exceed this bandwidth and very little reason to attempt to equal it. It was decided to hold to broadcast practice and design the main amplifier with a frequency response of 30 to 15,000 cps ±1 db from a 1000 cps reference.

Bandwidth and power output specifications, in themselves, are relatively easy to meet until a distortion specification at rated power output is set. With a total harmonic distortion on the order of 1%, few critical listeners are offended, thus 1% over the 30 to 15,000 cps band was chosen as the design specification. The intermodulation distortion compared to 1% harmonic distortion is approximately 4%. It is extremely important, in any amplifier design, not to be content with measuring the single-frequency distortion at the output. Internal feedback and non-linearities may cause cancellations that yield meaningless results when only the output is measured. It can be shown philosophically and demonstrated experimentally that two stages whose harmonic distortions are 3% may result in a measured distortion of, say, 1% at the output. The r.m.s. of the individual stage distortion, however, indicates the true performance of the amplifier. This is the reason for measuring the intermodulation, which takes into account the sum and difference products of each stage.

Finally, it was desired that the hum and noise level at the output of the main amplifier be at least 70 db below rated output.

In the preamplifier, it was deemed desirable to include four controls: "Loudness/Volume," "Treble," "Bass,"

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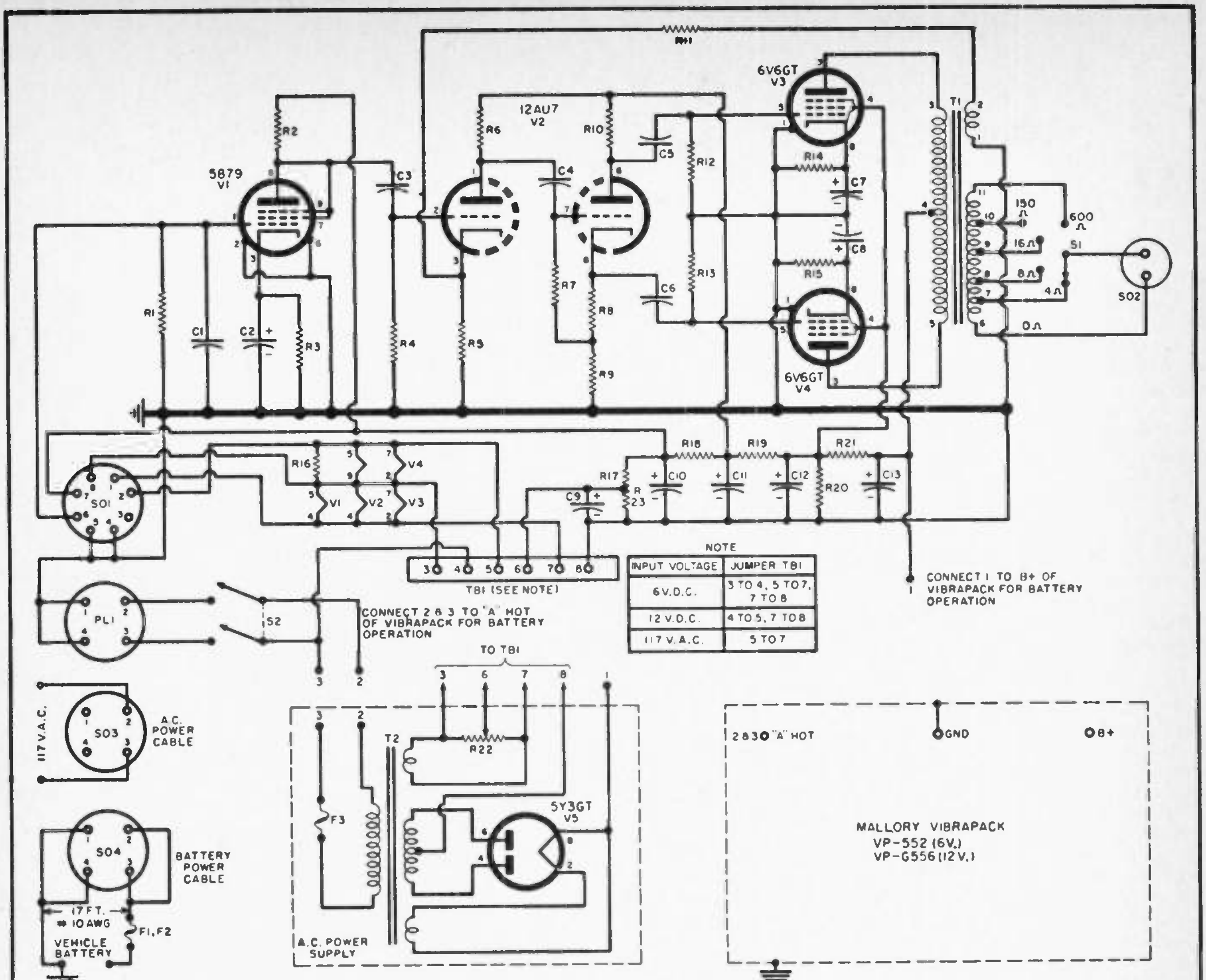

R_1, R_{17}—100,000 ohm, ½ w. res.
R_2—68,000 ohm, ½ w. res.
R_3—2200 ohm. ½ w. res.
R_4, R_{12}, R_{13}—470,000 ohm, ½ w. res.
R_5—3600 ohm, ½ w. res.
R_6—56,000 ohm, ½ w. res.
R_7—1 megohm, ½ w. res.
R_8—2000 ohm, ½ w. res.
R_9, R_{11}—22.000 ohm, ½ w. res.
R_{10}—24,000 ohm, ½ w. res.
R_{14}, R_{15}—390 ohm, 2 w. res.
R_{16}—43 ohm, 2 w. res.
R_{18}—10,000 ohm, ½ w. res.
R_{19}—2000 ohm, 1 w. res.
R_{20}—20,000 ohm, 20 w. wirewound res.
R_{21}—2000 ohm, 2 w. res.
R_{22}—100 ohm, 2 w. pot
R_{23}—30,000 ohm, ½ w. res.

Note. All resistors are to be ±10% carbon units except as indicated.
C_1—100 μμfd. mica capacitor
C_2—40 μfd., 25 v. elec. capacitor
C_3, C_4, C_5, C_6—.047 μfd., 200 v. capacitor
C_7, C_8—40/40 μfd., 25 v. elec. capacitor
C_9—50 μfd., 50 v. elec. capacitor
C_{10}, C_{11}, C_{12}, C_{13}—20/20/20/20 μfd., 450 v. elec. capacitor
SO_1—8-contact female receptacle (Amphenol Type 79-PO8F)
SO_2—2-contact microphone connector (Amphenol Type 80MC2F1)
SO_3, SO_4—4-contact female receptacle (Amphenol Type 79-PO4F)
PL_1—4-contact male receptacle (Amphenol Type 79-PO4M)
TB_1—Terminal board (Jones ¾W-164-5)

F_1—10 amp cartridge fuse (General Electric "Silverend," for 6-volt operation)
F_2—7 amp cartridge fuse (General Electric "Silverend," for 12-volt operation)
F_3—2 amp fuse (3AG type)
S_1—S.p. 4-pos. rotary switch
S_2—D.p.s.t. 10 amp switch
T_1—Output trans. 10,000 ohms c.t. to 600/150/16/8/4 ohm (Chicago PSC-150 or PCO-150, same specs except for case)
T_2—Power trans. 350-0-350 v. @ 200 ma.; 6.3 v. c.t. @ 6 amps.; 5 v. @ 3 amps. (Thordarson T-22R07 or equiv.)
Vibrapack—Vibrator (Mallory VP-552 for 6-volt operation and VP-G556 for 12-volt operation)
V_1—5879 tube
V_2—12AU7 tube
V_3, V_4—6V6GT tube
V_5—5Y3GT tube

Complete schematic diagram of the power amplifier and rectifier circuit portions of the portable high-quality audio system.

and microphone "Mixer." Provision for equalization of the "New Orthophonic," AES, LP, and European 78 recording curves was made, but only the "New Orthophonic" equalization was incorporated.

With the current controversy over loudness controls, it was decided that a "universal" amplifier should have provision for operating the front panel control either as a loudness or volume control, hence the circuit to be described.

The treble control was designed for either boost or droop, the mid-position being flat. This allows "shading" the equalization more closely for differences in room acoustics, playback level, and other recording curves. The bass control was designed for boost only, since bass droop is used so seldom. When bass droop is desired, the operation of the loudness control as a volume control removes the Fletcher-Munson compensation and gives the illusion of bass droop.

Microphone mixer controls are generally of multi-element or multi-stage design and were not felt to be sufficiently economical of space to warrant their use in this application. A new and novel mixer-fader circuit (Patent Applied For) was developed for this use.

Again in the preamplifier, it was the designer's purpose to provide a circuit with negligible distortion, 1.0% harmonic distortion being considered tolerable. Hum and noise 70 db below full output was the accepted goal.

In order to meet the requirements set forth in the first paragraph, it was mandatory that both amplifier and preamplifier be capable of being operated from primary sources of 117 volts a.c., 6 volts d.c., or 12 volts d.c.

The following step was to determine an acceptable packaging arrange-

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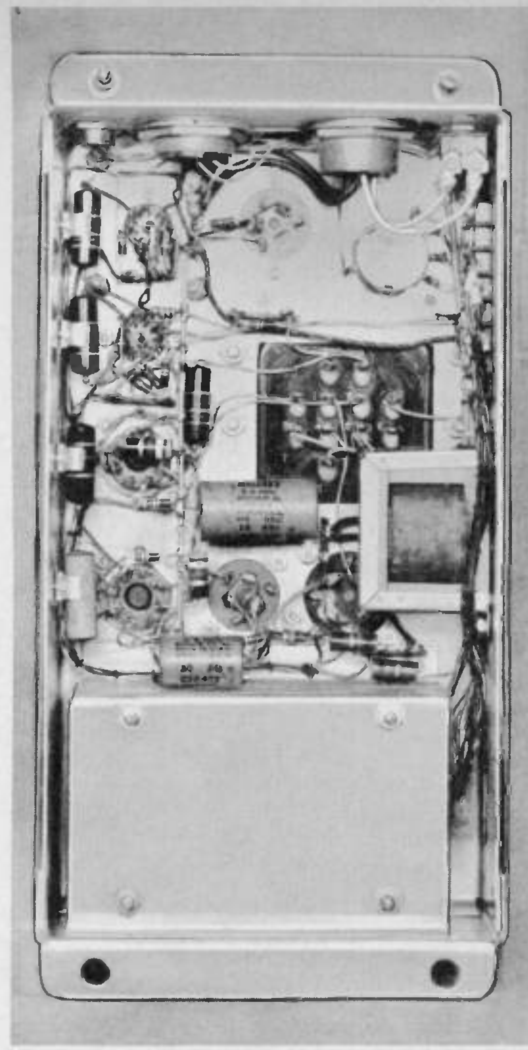

ment. Ease of service and portability were of paramount importance. For minimum chassis currents and induced hum into the low-level stages, two separate chassis for the main amplifier and preamplifier were chosen, thus also giving flexibility in mounting and operating the units. For vehicular use, the main amplifier could easily be trunk mounted; and, for home or party use it would be advantageous to have only the preamplifier at the operating position.

↑ Top view of the power amplifier with top and side panels removed. The audio output transformer can be clearly seen near the center of the chassis, while the power transformer is at the left. The four amplifier tubes, the electrolytic filter capacitors, and impedance selector switch are also visible on the chassis.

Underside of the power amplifier unit. Note the use of the ground bus immediately to the right of the line of four tube sockets for the amplifier tubes. All components are rigidly mounted so that the vibration to which a portable unit of this type may be subjected does not cause any serious problems. →

The Output Stage

Once the power output of an amplifier has been determined, the next step in the design is to select the output stage. A push-pull arrangement was chosen so that even-order harmonics would be cancelled, there would be no d.c. saturation in the core of the output transformer, and two small tubes of modest power requirements could be used rather than one large tube. Among the tubes that could have been used are the 6V6, 6L6, 807, 6146, etc. The available data for the 6V6 in the tube manuals show that 14 watts at 3.5% harmonic distortion may be expected with plate and screen voltages of 285 volts and zero signal plate current of 70 ma. It is true that each of the other tubes could produce the same power output at less distortion but with greater input power requirements. Since it is seldom wise to send a man to do a boy's job (he may be expensive to feed), 6V6GT's were selected. Bear in mind that the 14 watts are referred to the primary, thus allowing an output transformer whose efficiency is 70% or greater.

The circuitry which evolved with the use of the push-pull stage is relatively simple and straightforward; certain fundamentals being important and having been adhered to in this design. Since high G_m tubes operating at high currents always draw some grid current and since this current is generally not a linear function of grid voltage, hence causing distortion and limiting power, the grid resistors were kept to as low a value as possible, consistent with minimum loading on the phase inverter. The cathodes were biased with separate resistors rather than a single resistor of half the value. This arrangement tends to equalize the plate currents when two tubes, whose characteristics are slightly different, are used. These resistor values were carefully chosen to yield minimum distortion when the output stage was loaded *via* the output transformer to be described and when driven by the phase inverter—*before the feedback loop was closed.* This latter point is an important one which is often overlooked. In designing a feedback amplifier, the safest avenue to follow is to design the best possible amplifier without feedback and then apply just enough feedback to meet the design specifications. It is generally much wiser to apply 10 db of feedback to reduce 3% distortion to 1% than to apply 20 db to reduce 10% to 1%.

A bleeder in the screen circuit was used to improve regulation. (Note that the constant plate current curves shown for beam power tubes assume constant screen voltage.)

The Output Transformer

After the output stage is selected, a device to couple it to the load must be selected. With a push-pull stage, a transformer is the logical choice, but a decision about the method of applying feedback must be made before the transformer can be specified. Among the more practical and popular methods of forming the feedback loop are: feedback from the plate of the output stage, from the load winding, and from a tertiary winding.

If feedback is taken from the plate of the output stage, the points to which the feedback can be applied are fixed by the number of stages. Also, the load presented by the feedback loop to the output stage must be very large with respect to the load resistance referred to the primary. Furthermore, the output stage "B+" must be very well filtered since it can be shown that as the feedback increases, the hum in the output will approximate the power-supply ripple.

Feedback, in any case, follows the relationship: $e_{out}/e_{in} = A/1\text{-}AB$

where: e_{out} = output voltage

e_{in} = input voltage

A = forward gain of the amplifier without feedback

B = portion of the output signal applied as feedback

The term "A B" is a complex variable and must never equal $1 + j0$ if the amplifier is to be stable. Thus, the limiting conditions for the design of the feedback amplifier are (*a*) all of the coupling networks in the "A B"

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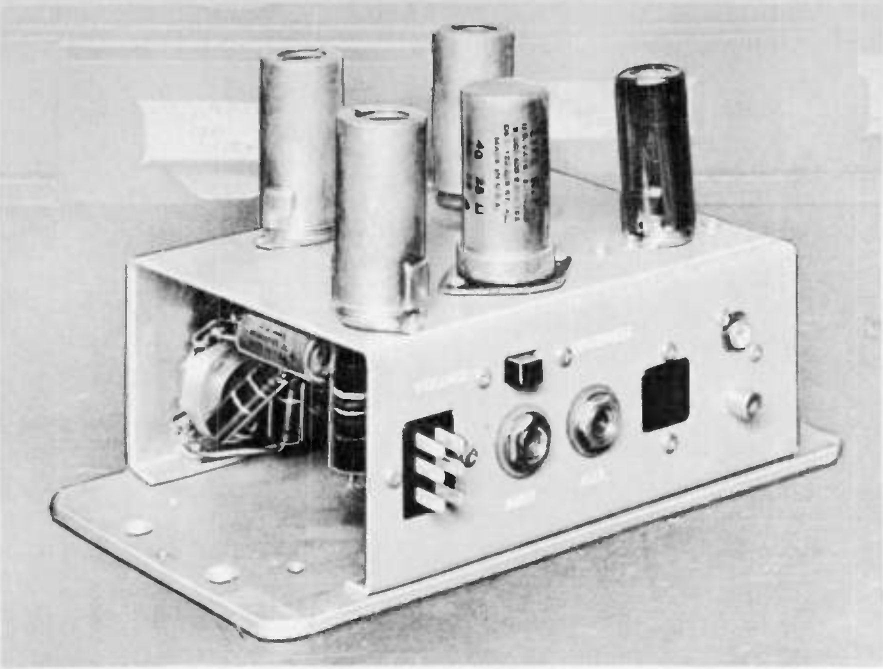


Photo of preamplifier unit with top cover removed. Rear panel is toward reader.

circuit are of the minimum phase-shift type, (*b*) all of the amplifying elements in themselves unilateral, and (*c*) all of the circuit parameters possess a value independent of time. Condition (*a*) implies a formal relationship between the amplitude transmission characteristic and the phase-shift of output relative to input of the network. Practically, the phase-amplitude characteristic must be controlled one octave above and below the signal-channel passband for every 10 db of feedback, plus an octave for margin.

When the feedback is taken from the load-winding of the output transformer, rather stringent requirements are placed on the transformer. If the passband of the forward amplifier is 30 to 15,000 cps, the phase-amplitude characteristic of the feedback loop must be controlled from 4 cps to 120 kc.

In order that the load winding be required to pass only the signal frequencies, tertiary feedback was selected. Thus the signal circuit is sampled and the distortion products appearing in the transformer are fed back. Also, the hum voltage appearing at the output is reduced by a constant factor as the feedback is increased.

Transformer Design

The output transformer specifications must be quite rigid since it must be considered as another circuit parameter. The following are the more important items to be specified.

1. Since the transformer can deliver only about two-thirds of the available volt-amperes to the load, the reactance of the primary inductance at the lowest frequency to be amplified must be approximately twice the primary load impedance.

2. The leakage inductance of the primary to tertiary must be low in order not to load the primary; and, the tertiary surge impedance must be much smaller than the tertiary load. The first resonance of the tertiary must be at or above the highest frequency in the feedback loop.

3. The turns ratio of tertiary to primary must be as close to "B" as possible.

4. The electrostatic coupling between windings must be kept to a minimum. Operating one side of the tertiary at ground potential is an important step in this regard.

The *Chicago Transformer* PSC-150 or PCO-150 (see parts list), although designed for p.a. work, has been found very satisfactory for use in this "universal" amplifier.

Phase Inverter-Voltage Amps

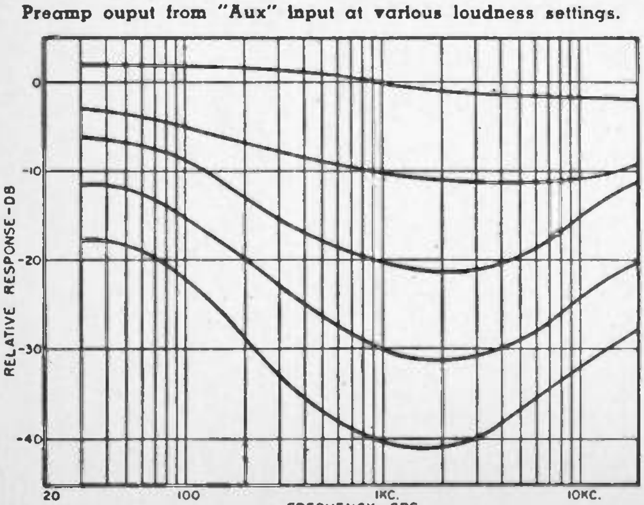
The output stage was fed from a split-load phase inverter despite the fact that the plate and cathode circuits are at different impedance levels, thus offering the possibility of different amplitude-frequency characteristics between plate and cathode circuits. These differences are distinct and measurable but are, for all practical purposes, undistinguishable even to critical listeners.

Feedback was applied to the cathode of the second voltage amplifier. A 5879 was triode-connected to serve as a low-noise first voltage amplifier and completed the tube complement for the main amplifier. The input required for 10 watts output was chosen as 0.5 volt r.m.s.

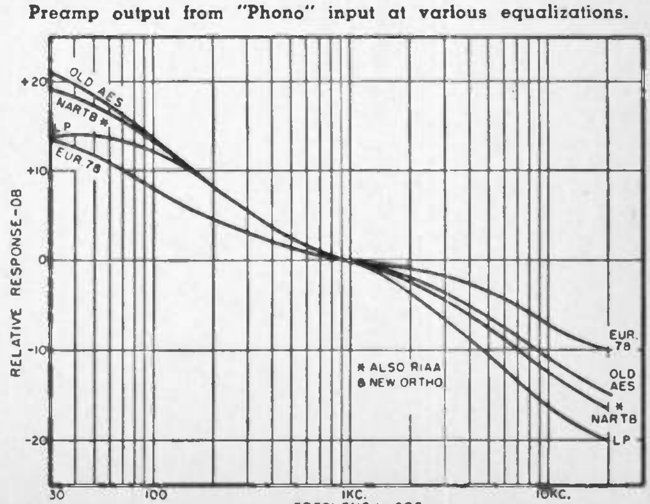
All ground connections were made to a ground bus connected to chassis ground only at the input. As a further precaution against hum, the heaters were biased at +50 volts d.c. to minimize heater-to-cathode leakage. The bias was applied to the arm of a 100-ohm potentiometer across the twisted pair used to feed the heaters. Adjusting this potentiometer for minimum hum enables one to reduce the output from 10 to 20 db over an unbiased arrangement.

Both a.c. and d.c. power supplies were needed as indicated on the amplifier schematic. A d.p.s.t. "on-off" switch energizes either type of power supply, depending upon the input cable used. Other connections to adapt the amplifier for a.c. or d.c. use are made to terminal board TB_1. It is important

Preamp ouput from "Aux" input at various loudness settings.



Preamp output from "Phono" input at various equalizations.



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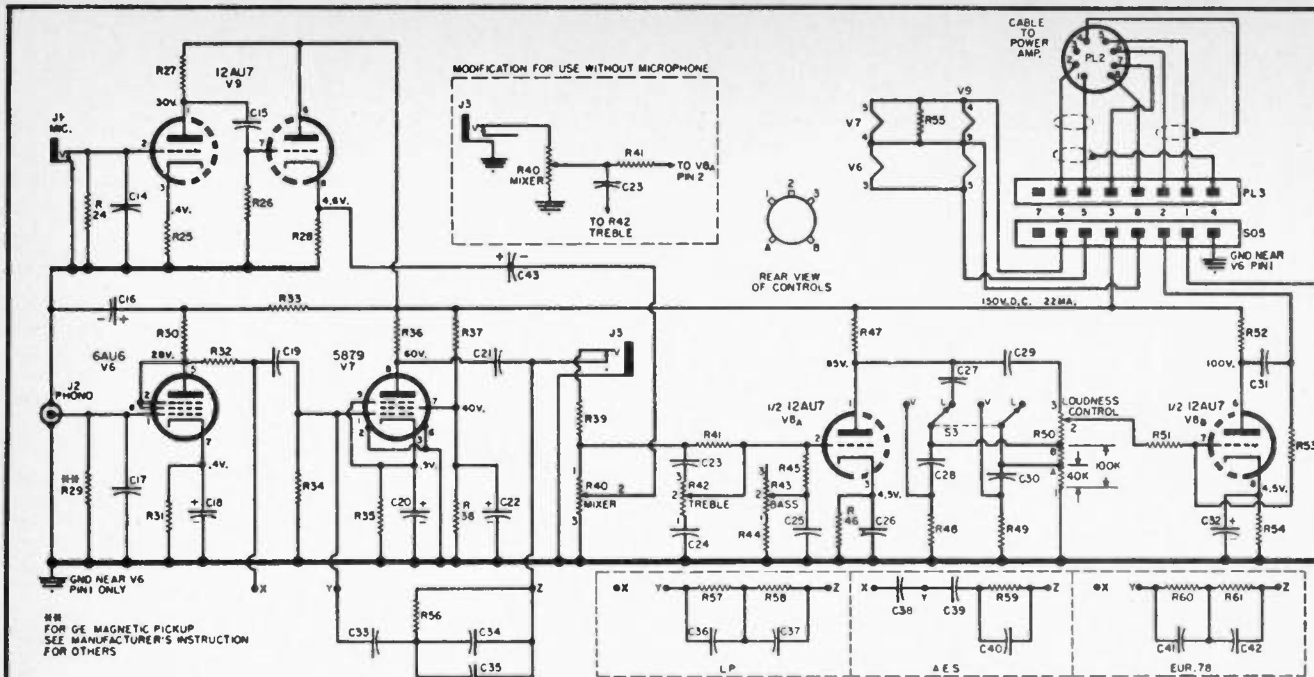


R_{24}, R_{27}, R_{30}, R_{36}, R_{37}, R_{47}, R_{56}, R_{58}, R_{59}, R_{61}—100,000 ohm, ½ w. res.
R_{25}—470 ohm, ½ w. res.
R_{26}, R_{39}, R_{41}, R_{51}—220,000 ohm, ½ w. res.
R_{28}, R_{33}—1000 ohm, ½ w. res.
R_{29}, R_{38}, R_{45}, R_{52}—47,000 ohm, ½ w. res.
R_{31}—330 ohm, ½ w. res.
R_{32}—27,000 ohm, ½ w. res.
R_{34}, R_{53}—470,000 ohm, ½ w. res.
R_{35}—680 ohm, ½ w. res.
R_{40}, R_{42}—500,000 ohm log taper pot
R_{43}—1 megohm log taper pot
R_{44}, R_{48}, R_{49}—15,000 ohm, ½ w. res.
R_{46}—6200 ohm, ½ w. res.
R_{50}—250,000 ohm log taper pot
R_{54}—3600 ohm, ½ w. res.
R_{55}—43 ohm, 2 w. res.
R_{57}, R_{60}—1 megohm, ½ w. res.

Note: All resistors are to be ±10% carbon units except as indicated

C_{14}, C_{17}—100 µµfd. mica capacitor
C_{15}, C_{41}—.01 µfd., 200 v. capacitor
C_{16}—20 µfd., 250 v. elec. capacitor
C_{18}, C_{20}—50/50 µfd., 25 v. elec. capacitor
C_{19}—.22 µfd., 200 v. capacitor
C_{21}, C_{31}—.047 µfd., 200 v. capacitor
C_{22}—5 µfd., 150 v. elec. capacitor
C_{23}—470 µµfd. mica capacitor
C_{24}, C_{26}—.0022 µfd., 200 v. capacitor
C_{25}—.01 µfd. ceramic capacitor
C_{27}—270 µµfd. mica capacitor
C_{28}, C_{30}—.022 µfd. mica capacitor
C_{29}—.15 µfd., 200 v. capacitor
C_{32}, C_{43}—25 µfd., 25 v. elec. capacitor
C_{33}, C_{36}—.0033 µfd., 200 v. capacitor
C_{34}—510 µµfd. mica capacitor
C_{35}—150 µµfd. mica capacitor
C_{37}—.001 µfd., 200 v. capacitor
C_{38}—.1 µfd., 200 v. capacitor
C_{39}—.0047 µfd., 200 v. capacitor
C_{40}—560 µµfd. mica capacitor
C_{42}—330 µµfd. mica capacitor
J_1, J_3—Microphone jack
J_2—Phono connector
PL_2—8-contact male receptacle (Amphenol Type 79-PO8M)
PL_3—Female 8-point plug (Jones type)
SO_5—Male 8-point connector (Jones type)
S_3—D.p.d.t. slide switch
V_6—6AU6 tube
V_7—5879 tube
V_8, V_9—12AU7 tube

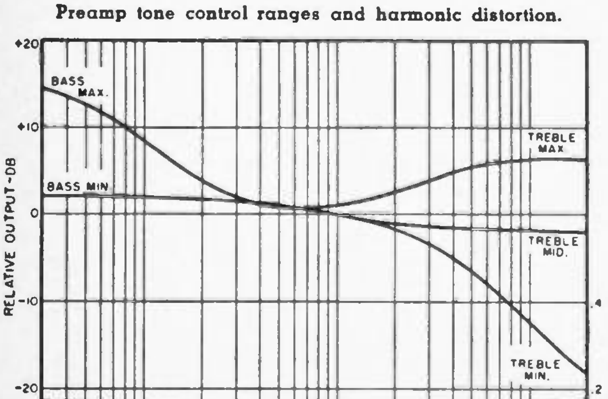
Complete schematic diagram of the preamplifier-equalizer unit of the portable high-quality audio system.

to note that the cables for mobile use are AWG No. 10 for both the "A" hot and ground returns. The fuse block and fuse must have low d.c. resistance to eliminate excessive voltage drops between the battery and power supply.

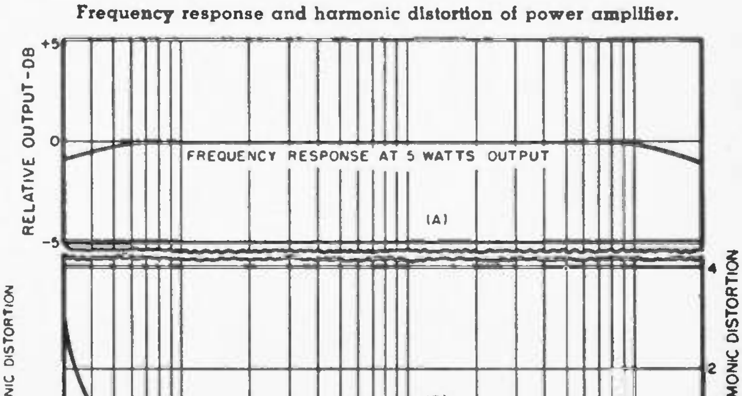
Preamp-Equalizer Stages

The input stage (V_6) for the preamplifier was designed around a triode-connected 6AU6. Several tubes are suitable as low-level amplifiers, but the 6AU6 shows little variation from tube to tube and the noise level referred to the first grid is consistently in the range from 5 to 7 microvolts. The second stage, V_7, was designed as a feedback-type equalizer. It is common to provide high-frequency equalization by varying the load on the magnetic pickup; and the low frequencies are often equalized by placing combinations of *RC* networks in shunt with one of the low-level stages. The author prefers to take advantage of the distortion-reducing and gain-stabilizing properties of feedback equalization, hence the plate-to-grid feedback shown around V_7. Resistor R_{32} from the plate of V_6 was included to isolate the feedback loop from variation in the plate resistance of V_6. The compensating network comprising C_{33}, C_{34}, C_{35}, and R_{56} is for the "New Orthophonic" curve and provides equalization within 1.2 db of the standard

Preamp tone control ranges and harmonic distortion.



Frequency response and harmonic distortion of power amplifier.



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curve from 30 cps to 15 kc. The chart shown on the schematic diagram for the preamplifier lists the parameters for equalization of the LP, AES, and European 78 curves. It is important, in wiring V_7, that pins 2, 6, and the center post on the tube socket be grounded if low hum is to result.

Inasmuch as a "universal" preamp would be universal only if a microphone input were provided, the circuit to be described was incorporated. The first half of V_9 is used as a conventional voltage amplifier feeding the second half which is a cathode follower. The low impedance output is coupled by C_{43} to the arm of the mixer control R_{40}. The output of the equalizer (or from the auxiliary source) is high impedance and is further isolated by R_{39}. With the arm of R_{40} at the high end (counterclockwise), the low impedance output from V_9 is in shunt with R_{40} and the high impedance of the equalizer stage is shorted out, applying only the microphone signal to the tone control stage. If R_{40} is turned fully clockwise, the output from V_9 is connected to ground, allowing the full equalizer output to be coupled to the tone control stage. Any intermediate position of the arm of R_{40} results in mixing of the two signals in proportion to the position of the arm. Hence, the signal from the microphone may not only be mixed with the auxiliary and equalizer stages, but may be faded from one to the other with only a single control. Greater microphone sensitivity may be had by substituting a 12AX7 for V_9 and replacing R_{27} by a 3300-ohm resistor. Should a microphone input not be desired, R_{10} may be wired as shown in the inset and R_{39} may be shorted out. Transfer contacts on J_3 are used to disconnect the equalizer when an auxiliary input, such as a tuner or TV, is used. If space is no problem, a switch may be used instead.

The tone controls were designed around standard circuitry and the first half of V_8 was used to compensate for the insertion loss of the tone controls.

The loudness control was selected so that it could also be used as a volume control should that function be desired. S_3 is a d.p.d.t. slide switch which disconnects the frequency-sensitive components C_{27}, C_{28}, and C_{30}. The loudness control is coupled to the second half of V_8 through R_{31} which isolates the plate-to-grid feedback around V_8. The output stage is a conventional *RC* coupled amplifier, whose output impedance is reduced by feedback, rather than a cathode follower since greater voltage output at lower distortion is then possible.

The interconnecting cable between the preamplifier and main amplifier was made as shown on the preamplifier schematic. Twisted-shielded pairs are used both for the heater and signal circuits. The heater-lead shield is grounded to the preamplifier ground bus and the signal-circuit shield is grounded at the main amplifier. Only one ground lead connects the units. A ground jack was provided on the preamplifier for the purpose of connecting an external earth ground (seldom necessary) or to the phonograph motor frame (frequently necessary).

Both units were housed in steel enclosures to minimize the effects of stray magnetic fields. The preamplifier was constructed on an aluminum chassis to minimize chassis currents.

The frequency response and harmonic distortion measurements were made with a *Hewlett Packard* Model 206A audio oscillator and Model 330B distortion analyzer.

The frequency response of the main amplifier is flat within 0.7 db from 30 to 15,000 cps. The harmonic distortion is 1% or less above 45 cps. Since the 1% specification selected originally is exceeded only below 45 cps and since few recordings carry these frequencies, the design was not altered to further improve the low end. The hum and noise is 72 db below 10 watts output.

The intermodulation distortion was measured with an *Altec Lansing* TI401 generator and TI402 intermodulation analyzer using frequencies of 60 and 2000 cps mixed 4:1. The preamplifier IM is 1% while the main amplifier IM is 5% at 10 watts output.

The preamplifier characteristics are shown in the accompanying curves. The harmonic distortion is less than 1% at all frequencies. The level required at the auxiliary input for the rated 1 volt r.m.s. output is 2.2 volts r.m.s. when the mixer circuit is used or 1 volt r.m.s. when modified for use without the microphone. The hum and noise level is 72 db below 1 volt at the output. At the phono input, .022 volt at 1000 cps is required for 1 volt output with the mixer while .010 volt will drive the preamplifier to 1 volt output when modified for use without the microphone.

Listening tests, using a *G-E* magnetic cartridge and a *G-E* A1-400 speaker in a distributed port enclosure, were extremely gratifying. The violinist's fingers sliding on the strings; the clear, vibrant ring of the chimes; and the staccato roll of the snare drum all point out that a simple, straightforward, and universal approach can result in a pleasing design.

REFERENCES

1. *Sivian, Dunn & White: "Absolute Amplitudes and Spectra of Certain Musical Instruments and Orchestras," Journal of Acoustical Society of America, Vol. 2., pp. 330-371, January 1931.*
2. *Mitchell, James A.: "Loudness and Power in Audio Systems,"* RADIO & TELEVISION NEWS, *February 1951.*

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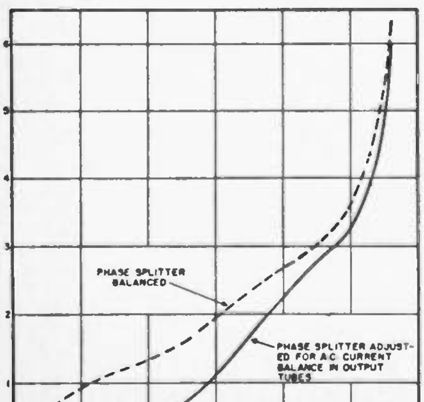
Phase Splitter Unbalance

(Continued from page 59)

Fig. 3. Distortion vs power output for an "Ultra-Linear Williamson" with balanced and adjusted phase splitter. The feedback loop was opened in order to obtain data.

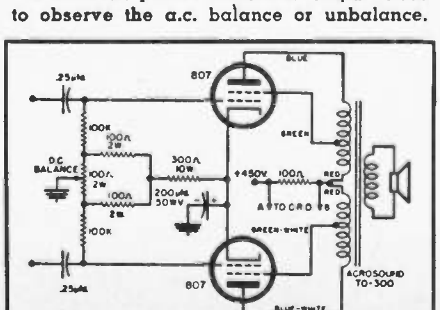


balanced. It can be seen that adjusting the phase-splitter reduced the distortion by as much as 700 per-cent.

The 807 tubes used in the amplifier might be considered a "matched pair" They both passed a tube-checker test, and indicated g_m's were within 8 per-cent of each other. The feedback loop was opened to take the data, and a regulated power supply was used.

The "a.c. balanced" condition corresponds to a minimum alternating current in the common plate lead or common cathode lead of the output tubes. The easiest way to observe the condition is with an oscilloscope connected across a small resistor (say 100 ohms) in the common plate lead as shown in Fig. 4, and a sine-wave input to the amplifier. (Caution—the scope chassis will be "hot".)

Fig. 4. A 100-ohm resistor is inserted in the common plate lead of the output tubes to observe the a.c. balance or unbalance.



Another way to observe the balance condition is to connect an a.c. meter or a headphone in the common plate lead through a step-up transformer, instead of the 100-ohm resistor shown in Fig. 4. Balance is indicated by minimum meter indication or minimum sound in the headphone.

In summary we can say: 1. If you are satisfied to accept the distortion likely to occur if the phase-splitter uses matched resistors—then you don't have to bother matching them, and 2. If you seriously want the lower distortion implied by the effort made to match the phase-splitter resistors—then you *should not* match them, but should make their ratio adjustable, as in Fig. 2.

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Measuring Amplifier Damping Factor

Described here are two methods that may be used to measure the damping factor of an audio amplifier.

THE KNOWLEDGE of what your amplifier's damping factor is has become more important recently. With high-efficiency speakers it was usually felt that the higher an amplifier's damper factor the better this would be for the speaker in damping out undesired resonances. However, with the advent of high-quality, low-efficiency speakers, and especially with such speakers mounted in infinite baffles or other such enclosures that do not add to the bass response, the use of an amplifier with too high a damping factor is undesirable. This would result in over-damping the speaker and reducing its output, especially at the bass end. Perhaps the day is not too far away when speaker manufacturers will specify optimum damping factors for their speakers when used in specific enclosures. Several values might be given for different types of listening tastes. The amplifier manufacturers have already started the ball rolling by providing, in some cases, a variable damping control that allows the user to set the damping factor at whatever value he requires.

At present most speaker manufacturers are quite reluctant to quote any optimum damping factors for amplifiers to be used with their loudspeakers. This is partly due to the fact that not enough experiments have been done along these lines and also that the results obtained at various damping factors is so subjective. Some listeners might like the way their speakers sound with one certain damping factor, others might prefer another value.

Typical values of amplifier damping factor depend, in the main, on the output circuit used. For example, with push-pull triodes without feedback the damping factors are in the range of 2 to 4. With push-pull beam power tubes the damping factor is apt to be less than this unless negative feedback is used. With large amounts of negative feedback, tetrode damping factors may run as high as 10. Recent designs using triodes with feedback or "Ultra-Linear" stages with feedback may have damping factors from 10 to 30.*

The measurement of an amplifier's damping factor is quite simple. This article will describe two methods of measurement.

Variable Resistance Method

To use this method a signal voltage is introduced into the input of the amplifier to be checked. This signal may come from an audio generator or even from the a.c. heater supply of the amplifier itself. Next the load is removed from the output terminals and the output voltage is measured with a suitable audio voltmeter. The input signal should be kept well below the overload point of the amplifier and below that point that might cause arc-over in the unloaded output transformer. Then a low value variable resistor (the author uses a 15-ohm wire-wound unit) is connected across the output terminals and this is adjusted until the voltmeter reading falls to one-half the unloaded value. Under these conditions the voltage across the variable resistor is equal to the voltage across the actual output impedance of the amplifier. See (A) of diagram. Now the variable resistor is removed and its resistance is carefully measured. This value of resistance is equal to the output impedance (technically, the effective source impedance) of the amplifier. If this resistance value is simply divided into the *nominal* output impedance of the amplifier (the value frequently marked on the amplifier itself), the result is the damping factor. For example, in the diagram shown assume that the value of the variable resistor is 1 ohm and that the measurement is being made on the 8-ohm tap of the amplifier, the damping factor is 8/1, or 8.

If it is found that the output voltage of the amplifier should rise when the load is applied, then this indicates that the amplifier has a negative damping factor. The resistance value that causes the loaded output voltage to be doubled is equal to the negative impedance of the amplifier.

Several problems may be encountered in using this method. One is that with amplifiers having very high damping factors, it might be difficult to adjust and to measure the very small values of resistance required to make the voltage drop to one-half. Another problem occurs when very low resistances are shunted across the output terminals of an amplifier; the primary impedance of the output transformer falls and the plate current through the output tubes may be excessive. In one case where this method was used, the plates of the output tubes became dangerously red. Another method that will yield the same answer but with none of the drawbacks is described below.

Voltage Regulation Method

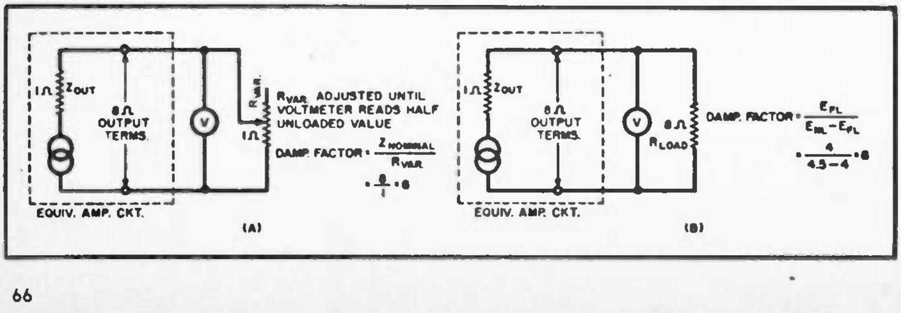
To use this method a signal is applied to the amplifier and an output voltage measurement is taken with no load connected. Let us call this no-load voltage E_{NL}. Next, a resistor with a value equal to the nominal output impedance is connected to the output terminals of the amplifier. The amplifier is now properly matched. Then, a second voltage reading is taken under these fully loaded conditions. Call this full-load voltage E_{FL}. The damping factor of the amplifier is simply equal to the full-load voltage divided by the difference between the no-load and the full-load voltage, or $E_{FL}/(E_{NL} - E_{FL})$. This may be recognized as the inverse of the regulation formula. As a matter of fact the damping factor of an amplifier is a measure of its regulation. If, for example, the output voltage falls only a very small amount when the circuit is loaded, its regulation is good (low) and its damping factor is high.

To show that this method gives the same results as the variable resistance method, consider (B) of the diagram. Assume that the same amplifier with its 8-ohm output tap and its 1-ohm output impedance mentioned before is used. Assume further that the no-load voltage measures 4.5 volts. There is no drop across the 1-ohm output impedance with no current flow. With the 8-ohm load connected, the full-load voltage will fall to 8/9 the no-load voltage, or 4 volts, in the simple series circuit. The damping factor then is 4/(4.5 — 4), or 8. This is the same value that was obtained with the first method described above.

Obtaining the damping factor, using either of the methods described, may help you to evaluate the performance of your loudspeaker system and it will certainly give you one more important characteristic of your amplifier. —30—

* See "Control of Amplifier Damping Factor" by David Hafler in the July, 1955 issue.

Equivalent circuits for two methods described above for measuring damping factor.



World Radio History

A 3-Channel Amplifier

By
KENNETH W. BETSH

Fig. 1. Over-all view of the three-channel high-fidelity power amplifier.

Designed for three-way speaker systems, this triple channel high-fidelity amplifier has variable electronic crossovers and provisions for electrostatic loudspeakers.

HERE is an amplifier, designed particularly for three-way speaker systems, that incorporates two popular innovations in the high-fidelity field. They are the use of "electronic" crossovers and electrostatic tweeters. The former amounts to resistance-capacitance crossover networks preceding individual amplifier channels for each speaker. Two crossovers are involved. The upper is fixed at 5000 cps while a choice of four—188, 375, 550, and 800 cps—is available for the lower crossover. The crossover switch also permits two-channel operation using the 5000 cps crossover and single-channel operation for conventional amplifier use.

The high-frequency channel of the amplifier was designed for two series-connected *Isophon* electrostatic speakers. Selling at a low price, information regarding these speakers may be had from *Arnhold Ceramics, Inc.*, 1 E. 57th St., New York 22, N. Y., the U. S. distributor.

Electrostatic speakers are not as easy to connect to a regular amplifier as one might wish. They are high-impedance devices, require d.c. polarizing voltage, and are limited to frequencies above 5000 cps. If a separate amplifier, limited to the high-frequency range, is used to drive these speakers, however, they may merely be connected between plate and ground. The output transformer, as a coupling device, is eliminated. This easily justifies an "electronic" crossover for separating electrostatic speaker output.

Reasons for using separate amplifiers for the low- and mid-range speakers in a three-way speaker system are quite different. First of all, there is the advantage of eliminating the conventional crossover network between the amplifier and speaker. Such networks are physically large and usually expensive, especially when the crossover frequency is quite low. They interfere with speaker damping and transient response in the region of the crossover frequency. At best they are only approximate since the builder or purchaser seldom knows the exact speaker impedance at the crossover frequency. The resistance of the crossover coils constitutes a small power loss. Second, with separate amplifier channels, the problems of speakers with different impedances or efficiencies are eliminated. Since attenuation in the speaker or crossover circuit, such as with L-pads is not required, amplifiers for the more efficient speakers operate at a lower power and, therefore, lower distortion level. Last of all, there is the advantage for electronic crossovers in the case of changing the crossover frequency. It is only necessary to change four ½ watt resistors.

The added cost of a separate amplifier channel to drive the mid-range speaker need not be great. With such a limited frequency range, an inexpensive output transformer and moderate output power will suffice. A common power supply further reduces the cost.

Individual Amplifier Channels

The high-frequency channel is designed especially for the operation of electrostatic speakers. Two of the speakers in series have an impedance in the range of 7000 to 10,000 ohms. This is a good load for a 6AK6. The combination of a pentode with cathode degeneration plus some voltage feedback results in a fairly constant power output for varying load. This is desirable in that the speakers have a capacitive element whose impedance will vary with frequency. The primary winding of a small output transformer is used to supply plate voltage to the 6AK6. Since the signal is not coupled through the transformer, an inexpensive transformer of any impedance will be suitable.

Referring to the schematic, Fig. 2, the network of C_2, C_3, R_4, and R_5 determines the frequency range over which the channel operates. The value of C_5 attenuates the low-frequency response but has negligible effect on the crossover characteristic. The cathode-follower is used to provide a low source-impedance to the crossover network.

In the range of 200 cps to 10 kc., the mid-range channel has a low distortion output of 8 watts and a maximum of 15 watts. A split plate-cathode phase inverter is used. Since balance of the push-pull stage is necessary for low distortion, a pot is placed in the cathode side of the inverter. Using an intermodulation analyzer, the author cut the low-level distortion in half by the use of this pot over that with fixed matched resistors. About 50,000 ohms of the pot is used. Either a linear 100,000 ohm or a 500,000 ohm

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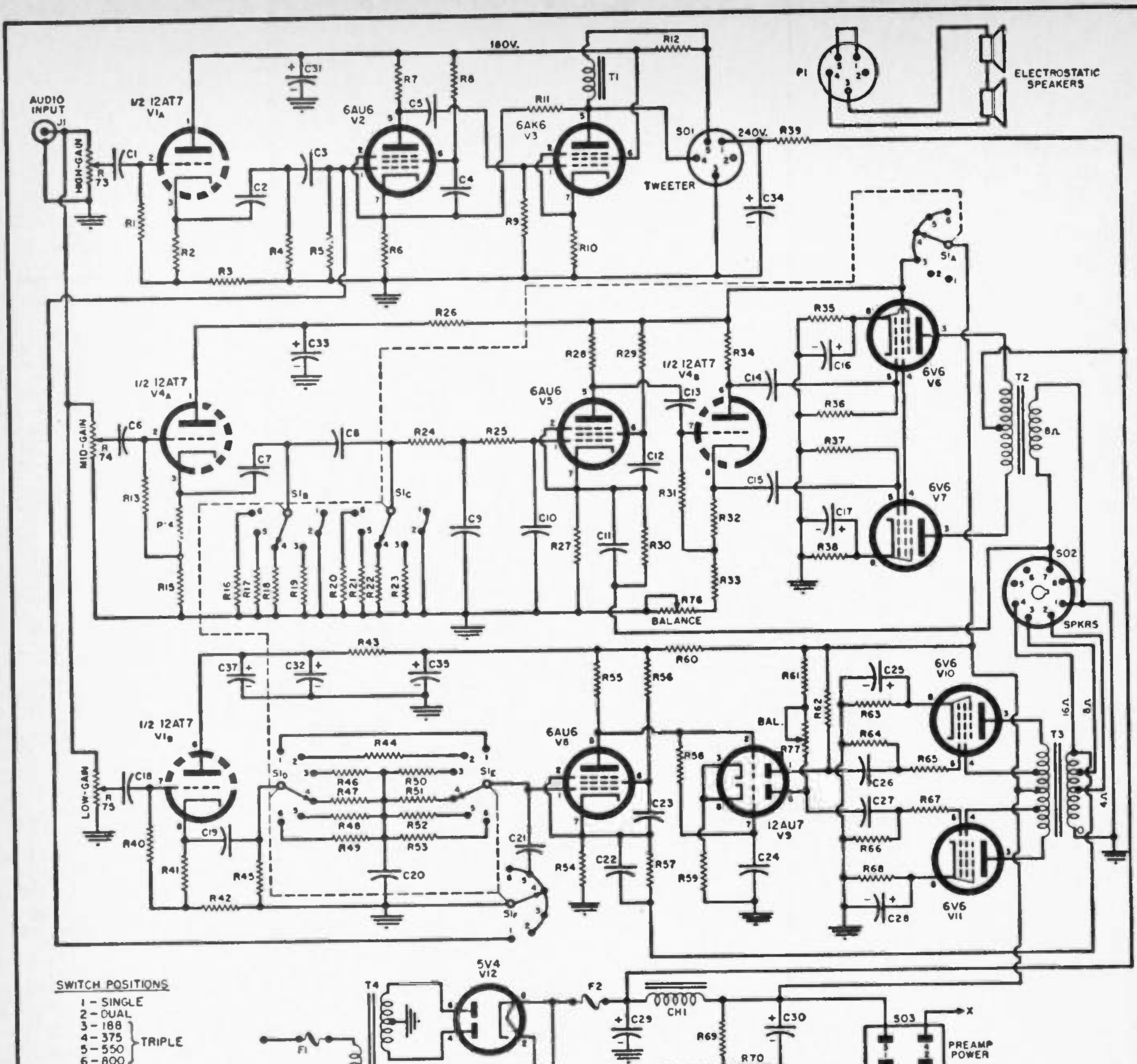
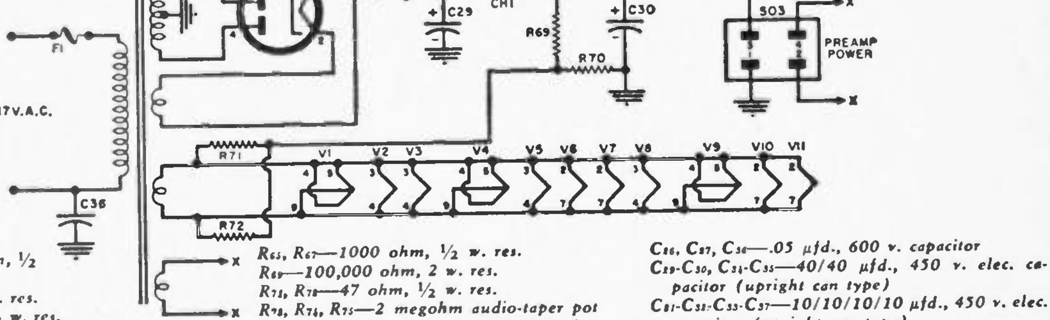


SWITCH POSITIONS
1 - SINGLE
2 - DUAL
3 - 188
4 - 375
5 - 550
6 - 800
(3–6) TRIPLE



R_1, R_9, R_{13}, R_{31}, R_{40}, R_{45}, R_{64}, R_{66}—470,000 ohm, ½ w. res.
R_2, R_{14}, R_{41}—820 ohm, ½ w. res.
R_3, R_{15}, R_{42}—56,000 ohm, 1 w. res.
R_4, R_{18}, R_{24}, R_{47}, R_{57}—100,000 ohm, ½ w. res. 5%
R_5, R_{22}, R_{25}, R_{51}—330,000 ohm, ½ w. res. ± 5%
R_6, R_{27}, R_{54}—1500 ohm, ½ w. res.
R_7, R_{28}, R_{55}—220,000 ohm, ½ w. res.
R_8, R_{29}, R_{56}—560,000 ohm, ½ w. res.
R_{10}—510 ohm, ½ w. res.
R_{11}—8.2 megohm, ½ w. res.
R_{12}—12,000 ohm, ½ w. res.
R_{16}, R_{49}—47,000 ohm, ½ w. res. ± 5%
R_{17}, R_{30}, R_{48}—68,000 ohm, ½ w. res. ± 5%
R_{19}, R_{46}—200,000 ohm, ½ w. res. ± 5%
R_{20}, R_{53}—150,000 ohm, ½ w. res. ± 5%
R_{21}, R_{52}—220,000 ohm, ½ w. res. ± 5%
R_{23}, R_{50}—680,000 ohm, ½ w. res. ± 5%
R_{26}, R_{43}, R_{59}—22,000 ohm, ½ w. res.
R_{32}—1200 ohm, ½ w. res.
R_{33}, R_{61}—47,000 ohm, ½ w. res.
R_{34}—100,000 ohm, ½ w. res.
R_{35}, R_{38}, R_{63}, R_{68}—510 ohm, 2 w. res. ± 5%
R_{36}, R_{37}—270,000 ohm, ½ w. res.
R_{39}—2700 ohm, 2 w. res.
R_{44}—43,000 ohm, ½ w. res. ± 5%
R_{58}—1 megohm, ½ w. res.
R_{60}, R_{70}—10,000 ohm, ½ w. res.
R_{62}—33,000 ohm, ½ w. res.
R_{65}, R_{67}—1000 ohm, ½ w. res.
R_{69}—100,000 ohm, 2 w. res.
R_{71}, R_{72}—47 ohm, ½ w. res.
R_{73}, R_{74}, R_{75}—2 megohm audio-taper pot
R_{76}, R_{77}—100,000 ohm linear or 500,000 ohm audio-taper pots (see text)
C_1, C_6—.001 μfd., 200 v. capacitor
C_2—600 μμfd. mica capacitor ± 5%
C_3—180 μμfd. mica capacitor ± 5%
C_4, C_{13}, C_{14}, C_{15}—.01 μfd., 400 v. capacitor
C_5—.001 μfd., 400 v. capacitor
C_7—8600 μμfd. mica capacitor ± 5%
C_8—2600 μμfd. mica capacitor ± 5%
C_9—170 μμfd. mica capacitor ± 5%
C_{10}—51 μμfd. mica capacitor ± 5%
C_{11}—100 μμfd. mica capacitor ± 5%
C_{12}—.05 μfd., 400 v. capacitor
C_{16}, C_{17}—10 μfd., 50 v. elec. capacitor
C_{18}—.02 μfd., 200 v. capacitor
C_{19}—.1 μfd., 200 v. capacitor
C_{20}—2400 μμfd. mica capacitor ± 5%
C_{21}—730 μμfd. mica capacitor ± 5%
C_{22}—47 μμfd. mica capacitor ± 10%
C_{23}, C_{24}—.22 μfd., 400 v. capacitor
C_{25}, C_{28}—50 μfd., 50 v. elec. capacitor
C_{26}, C_{27}, C_{36}—.05 μfd., 600 v. capacitor
C_{29}-C_{30}, C_{34}-C_{35}—40/40 μfd., 450 v. elec. capacitor (upright can type)
C_{31}-C_{32}-C_{33}-C_{37}—10/10/10/10 μfd., 450 v. elec. capacitor (upright can type)
CH_1—100 ma. choke
J_1—Phono jack
S_1—6-pos., 6-pole shorting-type rotary switch
F_1—3 amp. fuse
F_2—½ amp. fuse
SO_1—5-pin tube socket
SO_2—Octal tube socket
SO_3—4-pin socket
P_1—5-pin plug
T_1—Any single-ended output trans. (see text)
T_2—Output trans., 8 watts, 10,000 ohms to 8 ohms
T_3—Output trans. "Ultra-Linear" type for 6V6 tubes (Acro or Dyna)
T_4—Power trans. 600 or 650 v. @ 175 ma.; 5 v. @ 2 amps.; 6.3 v. @ 4 amps. (min.)
V_1, V_4—12AT7 tube
V_2, V_5, V_8—6AU6 tube
V_3—6AK6 tube
V_6, V_7, V_{10}, V_{11}—6V6 tube
V_9—12AU7 tube
V_{12}—5V4 tube

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audio taper pot gives this at half rotation (providing the counterclockwise end of the 500,000 ohm pot is used). Separate cathode resistors for the output tubes reduce unbalanced d. c. flow in the output transformer when tubes of unequal emission are used. Here again, the values of C_6, C_{12}, C_{13}, C_{14}, and C_{15} restrict low-frequency response. If this channel is to be used much below 150 cps, the values of these capacitors should be increased. Two dual *RC* crossover networks, one each for lower and upper limits, are placed between a cathode-follower and a 6AU6 amplifier. The mid-range channel has 8 db of feedback coupled through R_{50}. This resistor is bypassed by C_{11} to control ringing. The intermodulation was measured using test frequencies of 250 and 6000 cps and found to be 1% at 1 watt and 5.3% at 5 watts.

The low-frequency channel is, other than for the crossover network, capable of full audio range operation. For added gain, a 12AU7 amplifier-phase inverter is used. It offers low distortion and a stage gain of about 5. A balancing pot is also used in this channel. The output stage is of the "Ultra-Linear" type using 6V6's. Again, separate cathode resistors are used. About 12 db of feedback is employed. Using test frequencies of 60 and 6000 cps, mixed 4 to 1, the intermodulation measured .6% at 5 watts and 4.5% at 8 watts.

Crossover Networks

Crossover networks with a 12 db per octave slope are widely used and accepted. With "electronic" crossovers a network of two resistors and two capacitors, such as shown in Fig. 3B, will produce a characteristic that in a couple of octaves reaches a 12 db per octave slope.

The characteristic of a single *RC* network, such as shown in Fig. 3A, is curve A in Fig. 5. Note that there is 3 db attenuation at the frequency *F*. This is the frequency where the reactance of the capacitor equals the resistance of *R*. At each crossover frequency of the amplifier, the response of each channel should be 3 db down. This is because, by definition, the 3 db point is the half-power or 70 per-cent of maximum voltage point.

This characteristic curve is based on the use of constant input voltage to the network and no load. With networks using resistances in the 25,000-ohm to one-megohm region, cathode followers can meet these conditions satisfactorily. If a cathode follower were connected between two identical *RC* networks, a characteristic as shown by curve B in Fig. 5 is obtained. This curve shows twice the attenuation of curve A, and is, therefore, 6 db down at *F*. The 3 db point, by calculation, occurs at .63*F*. If the values of the capacitors are multiplied by .63, the curve is shifted to the right, as in

← **Fig. 2. Complete schematic diagram of three-channel audio power amplifier.**

CROSSOVER FREQ.	C_7	C_9	C_{20}	C_{21}
		(Capacity in $\mu\mu$fd.)		
200	15.015	4550	4211	1276
350	8580	2600	2400	729
400	7510	2275	2110	638
500	6000	1820	1650	510
600	5000	1520	1400	425
800	3750	1140	1050	320
1000	3000	910	825	255

Table 1. Calculated values of capacitors for various fixed crossover frequencies.

curve C, so that the 3 db point is at *F*. This allows accurate comparison of the two characteristics.

If two *RC* sections are directly joined without the cathode-follower between, the characteristic curve (if moved to the right to place the 3 db point at *F*) will fall between curve A and curve C, depending on the ratio of the impedance of the second section to the first. Calculations were made for the ratios of 1, 2, 3.3, 5, and 10. The characteristic for the ratio of 1, where R_{46} equals R_{50} and C_{20} equals C_{21}, is shown as curve D. The ratio of 3.3 is shown as curve E. Since it is only about 2 db away from the ideal and since higher ratios introduce other problems, this ratio is used in the calculations for the crossover networks of this amplifier. For this ratio, the 3 db point is at .53 of the frequency where X_c equals *R*. For a low-pass network, calculations must be made on the basis of X_c equals *R* at the crossover frequency divided by .53, or in other words, at 1.89 times the crossover frequency.

The low-pass network of Fig. 3B is the crossover network of the low-frequency channel. Ideally, the resistors should be fixed and the capacitors varied to change the crossover frequency. Over the limited range involved in this amplifier, 188 to 800 cps, the capacitor may be held fixed and the resistance varied. A frequency of 375 cps is taken as a median point and values for the capacitors are deter-

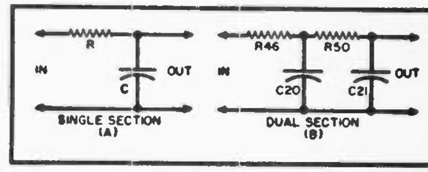


Fig. 3. RC low-pass networks. See text.

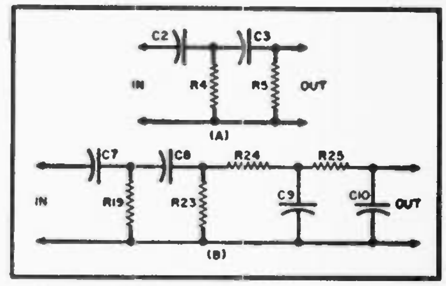
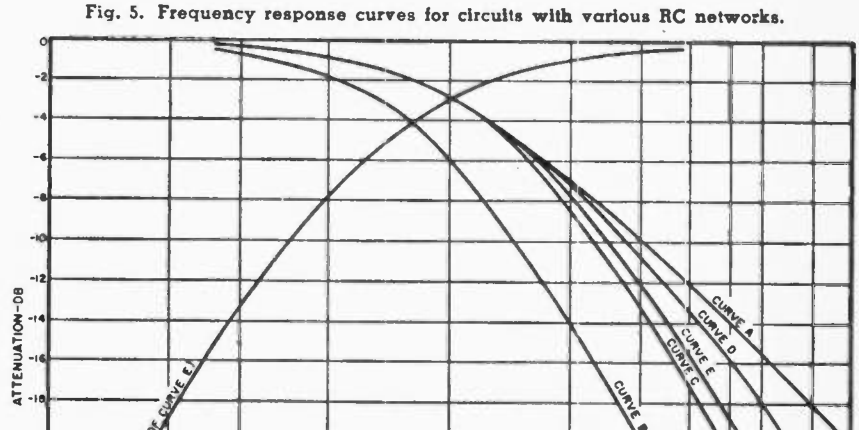


Fig. 4. (A) RC high-pass network. (B) Combined high- and low-pass networks.

mined for the best resistance values, which are set at 100,000 and 330,000 ohms for R_{46} and R_{50}, respectively. To change the crossover frequency, other resistors ranging from about one-half to twice these values are switched in. In the schematic, C_{19} and R_{45}, which precede the low-pass network, are used to provide d.c. isolation and a grid return.

For the high-frequency channel, a high-pass network, as shown in Fig. 4A, is necessary. Its characteristic is the reverse of curve E and is shown as curve F in Fig. 5. The 3 db point

Fig. 5. Frequency response curves for circuits with various RC networks.



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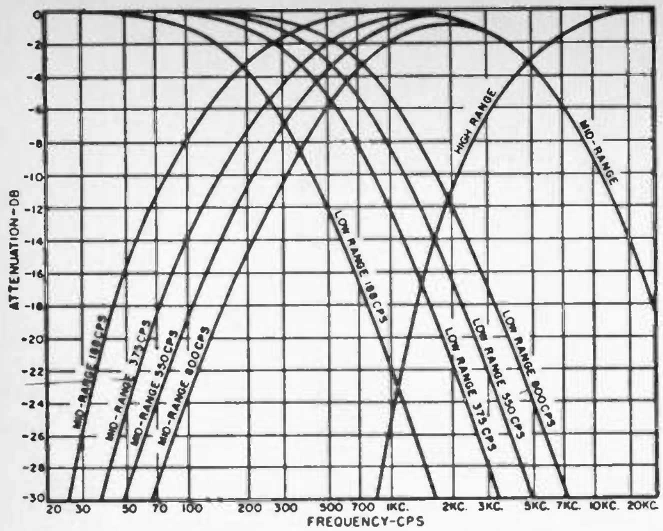

Fig. 6. Measured frequency response curves of three-channel amplifier.

occurs at 1.89 times the frequency where X_c equals R. Therefore, for a crossover at 5000 cps, calculations must be made on the basis of X_c equals R at 5000/1.89 or 2646 cps. Using resistances of 100,000 and 330,000 ohms for R_4 and R_5, the capacitors C_2 and C_3 are 600 $\mu\mu$fd. and 180 $\mu\mu$fd.

For the mid-range channel both low-pass and high-pass networks are necessary. To prevent a fixed loss the high-pass network precedes the low-pass network. The resultant network is shown in Fig. 4B. The resistors are the same as those in the corresponding networks in the other channels. From the foregoing discussion showing the frequency where X_c equals R at either 1.89 or .53 times the crossover frequency, depending on whether it is a low-pass or high-pass network, one finds the capacitor values will be quite different for corresponding networks even though equal resistances are used. The second half of Fig. 4B is a low-pass network for the 5000 cps crossover. Using resistances of 100,000 and 330,000 ohms, capacitor values are 170 and 51 $\mu\mu$fd. For the first half, the high-pass network of the lower crossover, identical resistor values to those used in the low-frequency channel are used. Capacitor values for C_7 and C_8 are 8600 and 2600 $\mu\mu$fd.

For two-channel operation the selector switch disables the mid-range channel and allows the low-frequency channel to operate up to 5000 cps. This circuit becomes a single-section *RC* network. This is for the sake of simplicity and cannot be considered serious since the efficiency of cone speakers at high-frequencies drops off when operated from a nearly constant voltage source such as the output of the low-frequency channel. For single-channel use, the low-pass network is disconnected and the other two channels disabled. The "B-plus" drain of the high-frequency channel can be removed by disconnecting the tweeter plug. Pins 1 and 5 are shorted on the plug.

Naturally, for exact crossover frequencies, the resistors and capacitors used in the networks must be accurate. It is not so important that the crossover frequency be exactly as desired. Rather, the component accuracy is important so that corresponding high- and low-pass networks will have 3 db points at the same frequency. Resistors with a 5% tolerance are satisfactory, particularly if all are of one brand and obtained at one time and place. A batch of resistors manufactured at the same time are almost always very close to each other in resistance. For maximum stability, mica capacitors should be used. Hand picking and measuring is most desirable. Otherwise, 5% tolerance capacitors should be used. In some cases it will be necessary to parallel standard values to obtain the proper value.

If the amplifier is to be used with a speaker system that is not subject to change, the selector switch can be eliminated and fixed values used. The resistances shown for a 375 cps crossover, namely, 100,000 and 330,000 ohms, can be wired in directly and capacitors, as listed in Table 1, used for other frequencies. These are calculated values. For best results the capacitors should be within five percent of these values. For a higher crossover, the use of fixed frequencies has the advantage of permitting the use of physically smaller capacitors.

The frequency response of each channel was measured and is shown in the response curves of Fig. 6. The low- and mid-range channels were measured with resistive loads at a level of 5 watts. The high-frequency channel output was measured across a 100-ohm resistor connected between the low side of the speakers and ground. Thus, the measurement was with the electrostatic speakers connected.

The amplifier is constructed on a 10" x 12" aluminum chassis. Small brackets were made to hold the balancing pots beneath the chassis.

If an intermodulation distortion meter is available, adjust the balance pots for minimum distortion at about a one watt level. Those without distortion meters may adjust the pots for equal a.c. signals at the plates of the 6V6's. If no measuring equipment is available, the balancing pots may be eliminated with these two changes: R_{33} and R_{31} both 100,000 ohms plus or minus 5%, and R_{61} is 56,000 ohms.

No power switch or master gain control are included since these functions can be provided by a preamplifier. The four-prong *Jones* plug permits heater and "B-plus" connections for a preamplifier. It can be eliminated or changed to an octal socket, which is quite popular for this purpose.

Many of the surplus TV power transformers on the market are ideal for this amplifier. If an extra 6.3 volt winding is available, wire it to the preamp power plug. If the regular 6.3 volt winding has a center tap, use it and omit R_{71} and R_{72}.

Because the individual channel amplifiers have considerable gain, care must be taken in the wiring to avoid hum and noise. Use good quality resistors and wire all ground points together.

The two electrostatic speakers are wired in series. Single-conductor shielded cable is good for connecting the speakers to the amplifier. Unless coaxial cable is used, this interconnecting wire should be less than 15 feet. While no enclosure is needed for the electrostatic speakers for acoustical reasons, something is necessary to hold them in position. The author built a small thin cabinet that positions the two speakers with a horizontal angle of approximately 45° between them.

It should be possible to set the channel gain controls once and not alter them again. Adjust the mid-range gain first to get a suitable volume level, the low-range next, and finally the high-range. The control for the weakest channel should be at maximum with the others adjusted as required.

One problem with the mid-range amplifier showed up when an efficient horn-type mid-range unit was tried, namely, an audible hiss introduced in the 6AU6 (V_5) stage. Therefore, if the builder plans to use an efficient mid-range unit, he may want to make the following circuit changes: Replace the 6AU6 with a 6AV6 triode, eliminate R_{29} and C_{12}; change R_{27} to 2700 ohms and change R_{30} to 39,000 ohms. These changes are optional and have no effect on over-all performance other than to reduce the small hiss that can be heard in the horn unit during periods of no signal.

The author has found the two electrostatic speakers to be highly satisfactory. With switches on the amplifier and speaker cabinet to switch from a conventional to an "electronic" crossover, a small but immediately noticeable improvement with separate amplifiers for the woofer and mid-range speaker is noted. The construction of this amplifier, which originally was undertaken as an experiment to determine the usefulness of these two innovations, has turned out to be a worthwhile improvement to the author's home music system. —30—

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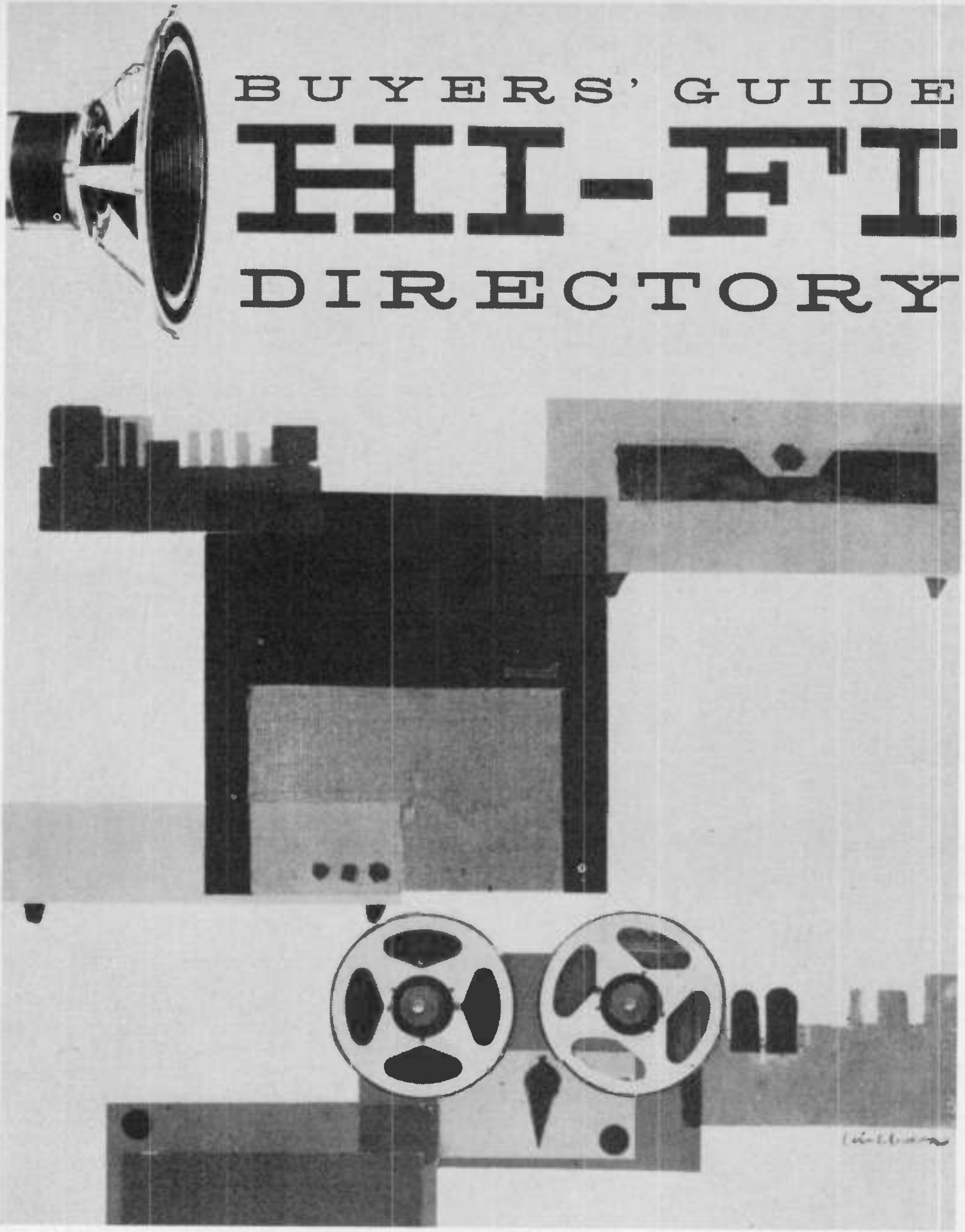

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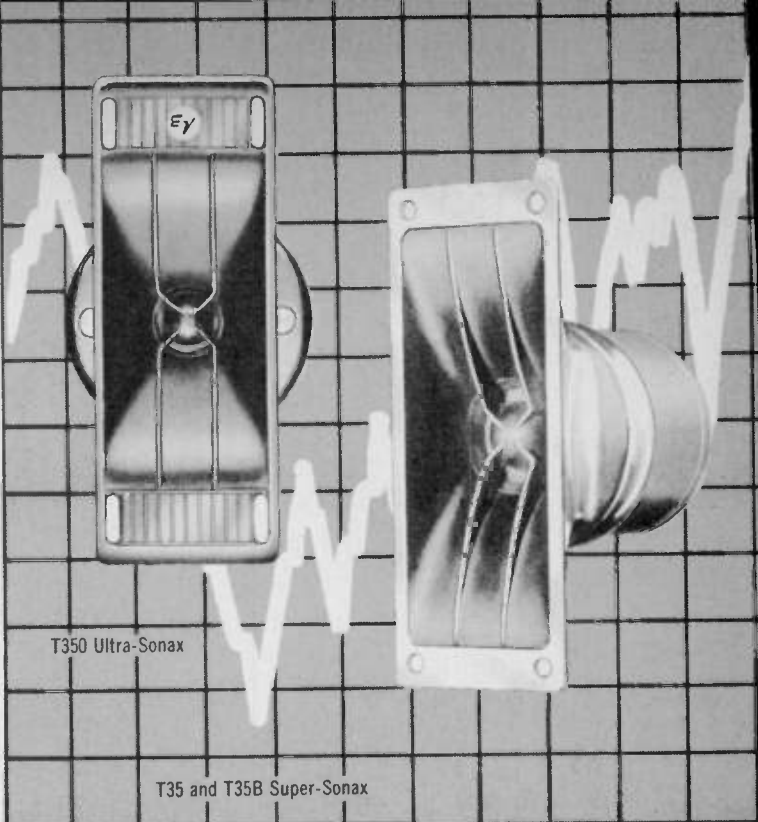
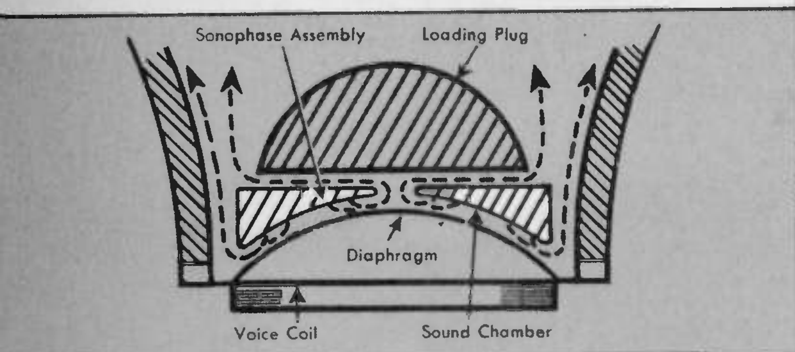
No other manufacturer gives you very high frequency drivers combining all the customer benefits of these unique new Electro-Voice models. Today's folded horn and phase loaded speaker systems with their low first-octave response *require* flat, extended high range response beyond the very limit of audibility if essential musical balance is to be achieved. These very high frequency drivers, employing the time-tested diffraction principle and the new Avedon throat design, overcome range and sensitivity limitations, function without distortion at the highest ranges.

All three models—T35, T35B and T350—have 180° dispersion patterns, program capacities of 50 watts, peak 100 watts, voice coils one inch in diameter and 16 ohms impedance. Chart shows other characteristics of each model.





Specifications	T35	T35B	T350
Frequency Response:	± 2 db 2 kc—19 kc	± 2 db 2 kc—18 kc	± 2 db 2 kc—21kc
RETMA Sensitivity Rating:	57 db	54 db	60 db
Magnet Weight:	7 oz.	4 oz.	1 lb.
Gauss	13,500	9,000	20,000
Size:			
Horn:	5¼ in. long x 2 in. wide		7½ in. long x 2⅞ in. wide
Pot Diameter:	2¼ in. maximum		3½ in. maximum
Depth:	3¼ in. overall	3 in. overall	4½ in. overall
Shipping Weight:	3 lbs.	3½ lbs.	9½ lbs.
Net Price:	$35.00	$22.00	$60.00

And These are the Reasons Why

The Avedon Throat Design

The unique throat design illustrated here overcomes a problem common in conventional high frequency drivers. This is diaphragm deformation at high velocities, occurring at frequencies above 5 kilocycles. Piston action is destroyed, the phase is shifted and the result is destructive interference.

These Electro-Voice VHF drivers solve the diaphragm distortion problem with a longer sound path from the center of the diaphragm. This restores proper phase relationship. This is important above 12 kilocycles, where sound must be taken from the center of the diaphragm *and* from the outer edge simultaneously. The diagram shows E-V's Avedon construction.

The Hoodwin Diffraction Horn

This is the Electro-Voice development which is used in all E-V horns to disperse sound *equally in all lateral directions* from a single point source. This is especially important in stereophonic reproduction to preserve the undistorted depth and width of the original sound. Diffraction horns insure balanced levels of both right and left stereo speakers. These drawings tell the diffraction horn story:

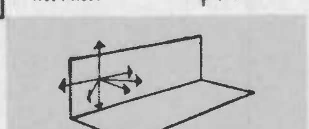


Figure A

Figure A—This shows how sound disperses equally in all directions from a single point source.

Figure B

In Figure B two sound sources are shown. On the axis, at point "x," double the sound power results as the resultant pressures are in phase.

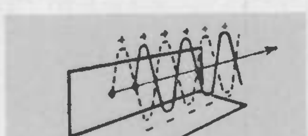


Figure C

But in Figure C, if the distance between the two sources is ½ wavelength or greater, the sound from the two sources will be considerably out of phase for points off the axis, resulting in decreased sound pressure.

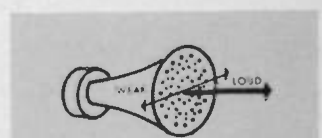


Figure D

Figure D will show the deficiencies in horns of wide lateral dimensions compared to the wavelength being emitted. Any horn mouth can be considered as a group of small point sources of sound. They must beam the sound down the axis by their very nature.

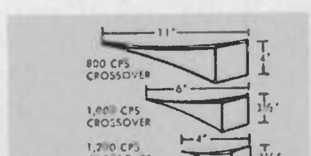


Figure E

Figure E shows representative horns, illustrating that horns must have a certain length, as well as cross sectional area along this length and at the mouth to load the driver diaphragm down to the lowest frequencies to be reproduced. The lower we go, the longer must be the horn and the greater the mouth area.

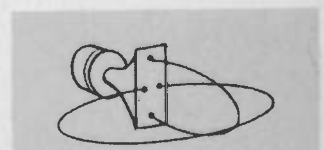


Figure F

Figure F shows that narrowing the horizontal area and extending the vertical dimension of the horn mouth preserves the loading area necessary for good low end response, disperses the sound perfectly in the horizontal direction where it is so necessary, and keeps interfering reflections off the floor and ceiling.

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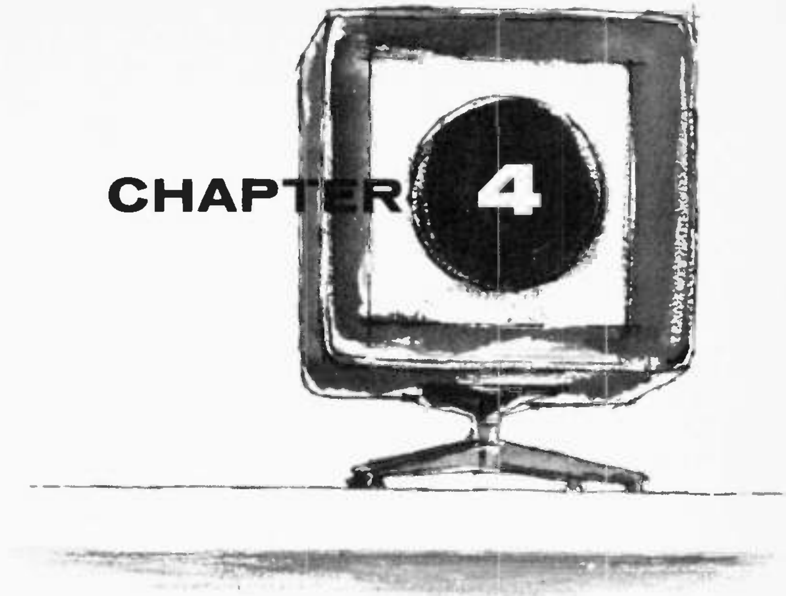

CHAPTER 4

LOUDSPEAKERS AND ENCLOSURES

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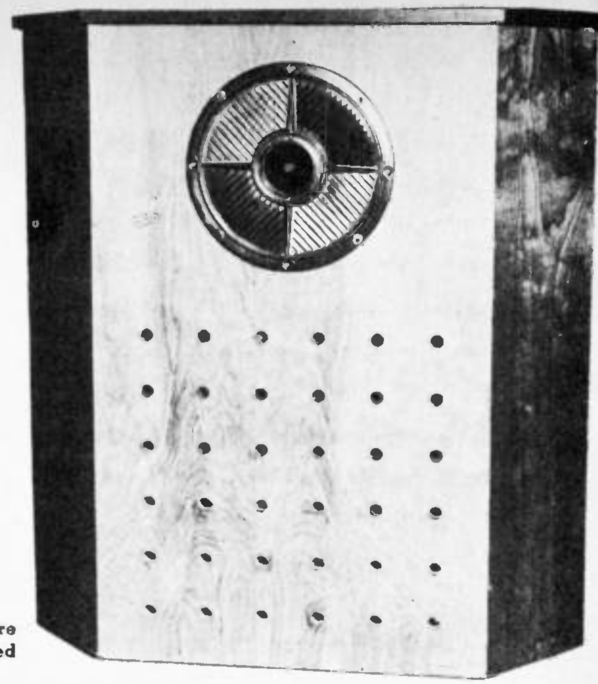
The "Distributed Port" LOUDSPEAKER ENCLOSURE



Laboratory model of enclosure with holes forming "distributed port" clearly visible.

By **ADELORE F. PETRIE**

Audio Products Engineering
General Electric Company

The performance of a "distributed port" enclosure is evaluated, along with useful general information on the low-frequency performance of quality loudspeakers.

THERE are three main requirements that a loudspeaker enclosure should fulfill: (1) it should prevent interference between the front and back radiation from the loudspeaker, (2) it should improve low-frequency response, and (3) it should improve low-frequency power output capability.

A loudspeaker acts as a dipole radiator. The sound coming from the back of the loudspeaker is 180° out-of-phase with the sound coming from the front. This means that there will be cancellations and additions at various frequencies when a loudspeaker is operated without some sort of baffle. The amount of cancellation or addition depends on the difference in distance which the sound has to travel when coming from the back and front of the loudspeaker and the frequency involved.

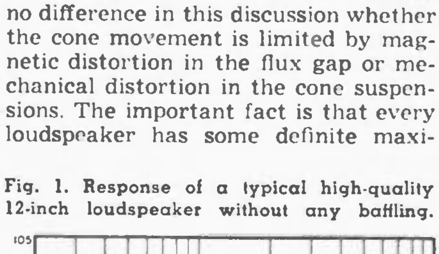
In other words, a loudspeaker acts like the piston in an air pump. Without some sort of baffle around the piston (loudspeaker), the air displaced in front of the piston will merely circulate around to the back without building up any useful pressure. Fig. 1 shows what happens to the low-frequency response of a good 12" loudspeaker when it is operated without any baffle.

It is desirable that the low-frequency response of a loudspeaker system extend down to the lowest frequency that is contained in the program material normally fed into it. It is also necessary that the response be smooth so that the tonal balance of the program is maintained. A gentle slope in bass response is permissible since this may be offset by additional bass boost in the driving amplifier. However, this boosting wastes power and may severely limit power output.

It is not enough for a loudspeaker to have smooth response and wide range. Due to the fact that the ear is insensitive at low frequencies, a loudspeaker system must also have sufficient power output capability to make the frequencies it reproduces audible to the listener. For example, sounds at 50 cycles-per-second must be 52 db louder than those at 1000 cycles-per-second in order to be heard. Therefore, it is necessary to check the low-frequency acoustic power output capability (sound pressure or intensity—not power input) of a loudspeaker system at a tolerable level of distortion to see if it is sufficient for application.

The acoustic power output capability of a loudspeaker is limited by the maximum sound pressure that the loudspeaker can produce without running into distortion. At low frequencies this pressure is determined by the maximum distance that the loudspeaker cone can move before it is limited mechanically or magnetically and by the enclosure used. It makes no difference in this discussion whether the cone movement is limited by magnetic distortion in the flux gap or mechanical distortion in the cone suspensions. The important fact is that every loudspeaker has some definite maximum cone movement. The amount of sound pressure this cone movement produces at low frequencies is determined by the characteristics of the loudspeaker enclosure. Because the acoustic power output capability in a given enclosure depends on cone excursion, it is interesting to note that the stiffness of the cone suspensions or the fundamental cone resonance of the loudspeaker has little effect on the power output capability of the loudspeaker system (providing the linearity of the stiffness is not changed).

Fig. 1. Response of a typical high-quality 12-inch loudspeaker without any baffling.



"Infinite Wall" Baffle

The previously mentioned interference problems may be completely eliminated simply by preventing the sound from the back of the loudspeaker reaching the front. One means of doing this would be to mount the loudspeaker in a hole in a large wall, see Fig. 2. This is known as an "infinite wall" type of enclosure. Because this is a simple enclosure which is comparatively easy to analyze and actually may be realized in practice, it can be used as a standard reference for comparison when evaluating other types of enclosures.

With this type of baffle, obviously there is no interference between the front and back radiation—they are separated by a wall "infinite" in size. The low-frequency response of a loudspeaker in this type of baffle depends on the size of the speaker, its efficiency, the damping factor of the amplifier driving the speaker, and the low-frequency resonance of the loudspeaker in the baffle. Fig. 2 shows the response of a typical high-quality 12" loudspeaker in this type of enclosure. (Note: In order to eliminate the amplifier from this discussion, the *frequency response* curves shown in this article are plotted with a constant voltage input to the loudspeaker.)

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This particular loudspeaker had a fundamental resonance of 50 cycles-per-second when mounted to the wall baffle. Note that this fundamental resonance was not apparent in the response curve. Instead of a resonant rise, the response was gently sloping off in the region around resonance. This is characteristic of all high-efficiency loudspeakers when driven from a feedback amplifier or other low impedance source. The electromagnetic coupling is so great that the amplifier shunts the resonant circuit, causing it to be overdamped (low "*Q*"). This makes for good transient response, but requires some sort of compensation to bring the low-frequency response up to where it should be.

In order to show the effect of efficiency on response, the efficiency of this loudspeaker, in an "infinite wall" type of baffle, was reduced by demagnetizing it in steps and thus plotting a family of response curves (Fig. 3). Curve A shows a high efficiency loudspeaker, curve E, shows a low-efficiency loudspeaker. With similar construction, each time the 200-cycle response was reduced by 3 db, it was equivalent to reducing the magnet weight by one-half. This is the reason why loudspeakers of high efficiency often seem to lack bass response unless the bass end is compensated in some way (either in the amplifier or the enclosure).

Fig. 4 shows the acoustic power output capability (P_a) of this loudspeaker in the "infinite wall" baffle for 5 and 10 per-cent harmonic distortion. (Note: The average sound pressures that the loudspeaker systems shown in this article can produce throughout a typical living room are approximately 3 db below the recorded levels, which are measured 18" in front of the loudspeakers in a "dead" room.)

Fig. 4 also shows the corresponding electrical input to the voice coil (E_{vc}). Note that the power at higher frequencies is limited, not by distortion, but by the maximum heat dissipation rating of the voice coil.

Note also that the electrical input to the loudspeaker must be greatly reduced at very low frequencies to prevent driving the loudspeaker into severe distortion. Rumble coming from the turntable and/or the record will often drive a loudspeaker into distortion, even though the rumble itself is too low in frequency and amplitude to be heard by the listener. To eliminate this difficulty it is usually desirable to have a sharp cut-off low-frequency filter somewhere in the system.

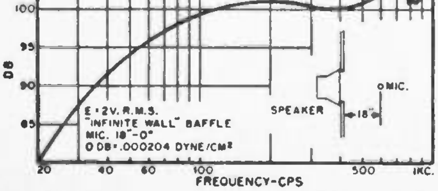


Fig. 2. Response of a typical high-quality 12" speaker in an "infinite wall" baffle.

If the loudspeaker in Fig. 4 were less efficient, it would handle more *electrical* input at low frequencies; however the maximum acoustic output at low frequencies would be the same and, due to lower efficiency, the maximum acoustic output would be less in the region where the power is limited by the voice coil dissipation.

"Distributed Port" Enclosure

The "distributed port" enclosure is a reflex type of enclosure in which the back radiation is added to the front radiation at low frequencies. By this means it improves both the low-frequency response and the power output capability. It differs from most reflex enclosures in that the response and impedance characteristics are controlled by the addition of a specific amount of acoustic resistance.

There is no interference between the front and back radiation at high frequencies due to the fact that the reflex action has an inherent high-frequency cut-off of back radiation. In the useful low-frequency range, a controlled amount of back radiation is added to the front radiation, increasing the output and power handling ability.

At very low frequencies (usually sub-audio) there is cancellation between the front and back radiation and air is merely pumped from the front to the back of the loudspeaker. This characteristic is common to all loudspeaker enclosures which utilize both front and back radiation, and makes it even more desirable to use a "rumble" filter, as mentioned previously.

Fig. 5 shows the low-frequency response of a typical high-quality 12" loudspeaker in three "distributed port" enclosures. Also shown, is the "infinite wall" response for comparison. Note the large increase in output and the smoothness of response at low frequencies, where the reflex action occurs. Adding the correct amount of damping in this design matches the impedance of the enclosure to that of the loudspeaker. This means maximum power output from the enclosure and broadband response (low "*Q*"). Putting this damping in the form of a "distributed port"—a fixed number of holes spread over a definite area—assures the permanent accuracy of the damping.

Fig. 6 shows the acoustic power handling ability (P_a) of these same "distributed port" enclosures. Note the great increase in the undistorted sound pressure that these enclosures are capable of delivering at low frequencies. Note that, at some frequencies, the larger enclosures will handle less voltage input (E_{vc}) than the smaller ones, although the sound pressure (P_a) produced by the larger enclosures is greater. This is due to the fact that they are more efficient than the smaller enclosures at these frequencies.

Dimensions for constructing three "distributed port" enclosures are given in Fig. 7. (The 6-cubic foot unit is available commercially as the *G-E* A1-406 loudspeaker enclosure.)

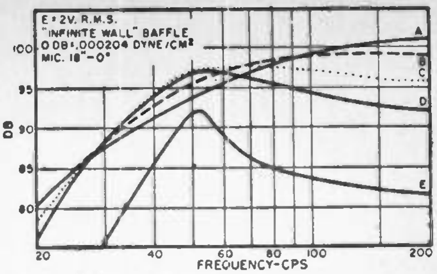


Fig. 3. Effect of magnet size on the response of a typical high-quality 12" loudspeaker (made by demagnetizing in steps).

The *G-E* A1-400 loudspeaker, used for all curves in this article, features a styled protective front plate, making the use of a grille cloth unnecessary in many installations. To take full advantage of this feature, the speaker should be mounted on the front surface of the speaker mounting board. If a grille cloth is required for styling purposes, the material used must not impair the transmission of high frequencies. Suitable materials are woven plastic or fabric having a light porous weave. The grille cloth should be mounted in a manner which will not allow vibration of the cloth against the cabinet. When grille cloth is used, the speaker is attached to the rear surface of the speaker mounting board.

Use plywood at least ½" thick for the 3- and 6-cubic foot sizes and at least ⅝" thick for the 10-cubic foot size. Preferred plywood thickness is ¾". Line the back, bottom, and one side of the 3-cubic foot enclosure with 1" of Fiberglas or similar soft acoustic material. Line the bottom and two back sides of the 6- and 10-cubic foot enclosures with 2" of Fiberglas or equivalent. Glue all joints. Make front or back removable, if the speaker is to

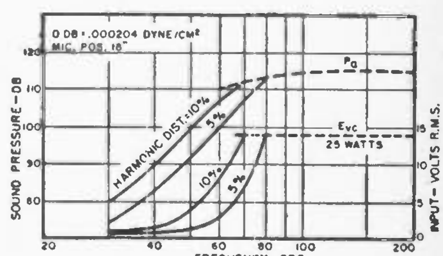
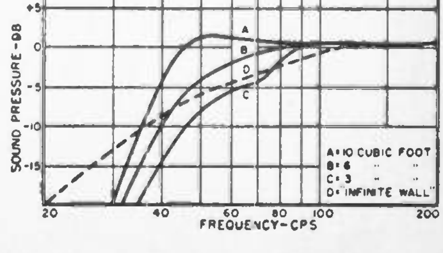


Fig. 4. Power output capability of a typical high-quality 12" loudspeaker in an "infinite wall" type speaker enclosure.

Fig. 5. Frequency response of a 12" speaker in several "distributed port" enclosures. The "infinite wall" response under same condition is shown for comparison.



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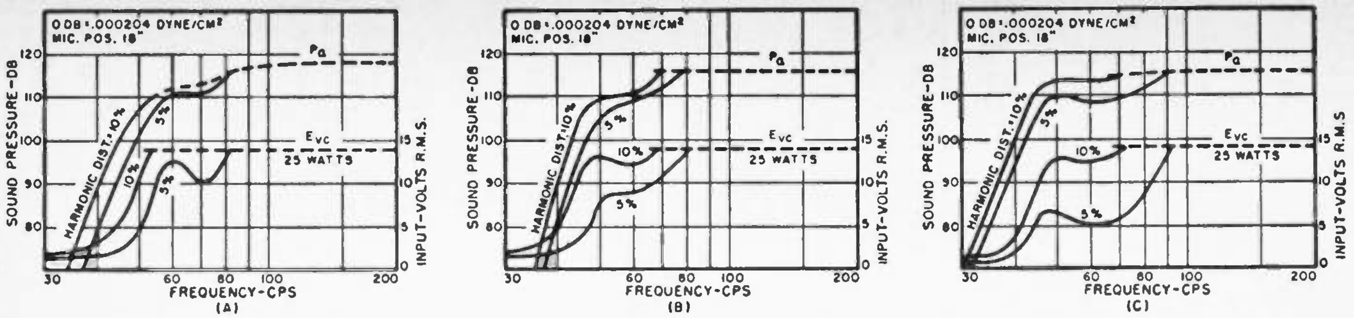


Fig. 6. Power output capability of a high-quality 12-inch loudspeaker in "distributed port" enclosures of various sizes. (A) 3-cubic feet, (B) 6-cubic feet, and (C) 10-cubic feet.

be mounted on the inside surface of the mounting board. The 2" x 4" brace is to keep the speaker from setting up vibrations in the front panel, which will subtract from the low-frequency output.

The shape of the enclosures may be altered to suit the needs of the user as long as the internal volume and the configuration of the front panel are maintained.

Loudspeaker Characteristics

The curves presented here are accurate only when the loudspeaker used has characteristics which are similar to those of the *G-E* A1-400 loudspeaker. However, the general characteristics indicated are typical of all loudspeakers and must be kept in mind when choosing or designing a loudspeaker, or enclosure, or associated equipment.

The important characteristics of the A1-400 "woofer" for enclosure design include: Nominal diameter—12 inches; Effective cone diameter—10¼ inches; Free air resonance—60 cycles/second; Mass of moving system (exclusive of air load)—25 grams; BL (force constant)—12,000,000 gauss cm; and Magnet weight—14 oz. *Alnico* V.

Fairly similar types are the *G-E* 1201-A and 1203-A loudspeakers.

Using loudspeakers of larger diameter than 12" in these "distributed port" enclosures will, in general, improve the power handling ability but result in a poorer low-frequency response. A smaller loudspeaker will give correspondingly better low-frequency response and poorer power handling ability. A larger loudspeaker should not be used with the 3-cubic foot enclosure and a smaller loudspeaker should not be used with the 10-cubic foot enclosure.

Room Effects

It should be noted that the curves presented in this article were taken under conditions which did not include reflections and resonances usually found in the listening environment. This is desirable because the effect of a room on the performance of a loudspeaker system is usually independent of the system used. These same effects also color the music as the listener hears it when live performers are in the room and therefore are not necessarily harmful.

Every hard object of appreciable size in a room sets up reflections. The most important objects to be considered at low frequencies are the walls, ceiling, and floor. Standing waves are set up in a room at frequencies where the distances between parallel surfaces are equal to ½, 1, 1½, 2, etc. wavelengths. These standing waves always build up high pressure points at the walls which cause them. The impedance match between the loudspeaker and the room is best at these high pressure points. For this reason, the best place for a loudspeaker system is the corner of the room where efficient coupling is made to all room resonances.

In most rooms, the only resonance which might be objectionable is the standing wave between the floor and ceiling, which usually occurs between 60 and 80 cycles-per-second. Coupling to this resonance may be reduced by placing the loudspeaker enclosure in the corner, but off the floor (half way up the wall is best). This is usually objectionable appearance-wise. An equally good solution is to damp the resonance by applying sound absorbing material to the ceiling. To absorb these frequencies may require padding up to 8" thick, or hung 8" below the true ceiling.

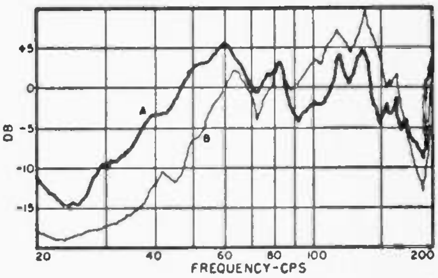
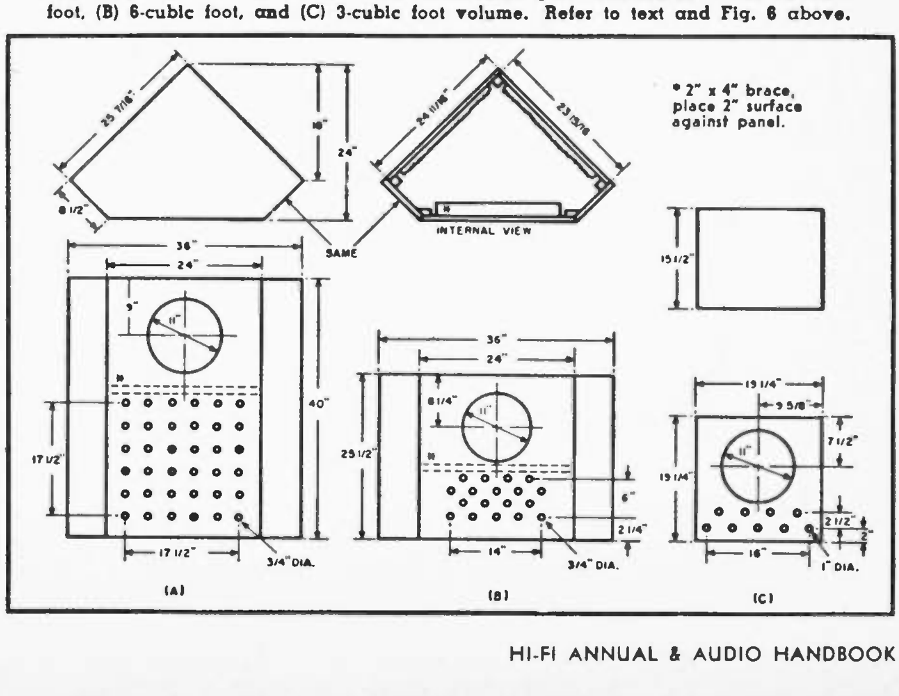


Fig. 8. (A) Response of a typical high-quality 12" loudspeaker in a 6-cubic foot "distributed port" enclosure in the corner of a "live" room, illustrating the effect of standing waves and reflections on response. (B) Response of a typical back-loaded folded corner horn enclosure of equal size measured under the same conditions as (A).

Fig. 8A shows the response of a 6-cubic foot "distributed port" enclosure placed in a corner of a live room. Curve B shows the response of a typical back-loaded folded corner horn enclosure of equal size under the same conditions. These curves were originally run to answer those who feel that horn enclosures have a "corner" on response in the corner of a room. However, it shows very nicely the effects of room resonances on two different loudspeakers.

Fig. 7. Mechanical details on three "distributed port" enclosures of (A) 10-cubic foot, (B) 6-cubic foot, and (C) 3-cubic foot volume. Refer to text and Fig. 6 above.



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Bass Reflex Design or Accident

Over-all view of the author's home "entertainment center" which includes a TV set and bookshelves in addition to the amplifier and speaker enclosure. A grille cloth covers front of enclosure and entire unit was painted to match living room decor.

By
ROBERT C. SANFORD
Asst. Editor, Bell Laboratories Record

Practical design data on such enclosures is rare—the author offers workable information for various sizes of speakers.

WHETHER you are a hi-fi addict or just happen to like good clean bass reproduction, the least expensive and certainly the simplest type of speaker enclosure for home construction is a bass-reflex or "phase-inverter" type; all it amounts to is a box with a hole in it. Having a fairly good 15-inch speaker on hand, and being in the process of rebuilding a radio-TV-phono combination and putting it into a new cabinet, the author decided to build a bass-reflex.

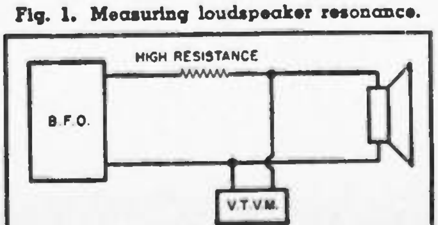
With access to one of the best technical libraries, it was just a matter of looking up the details as to the size, shape, and location of the port, and the size and shape of the enclosure. This beautiful day-dream was quickly shattered because, instead, some authorities treat a bass-reflex as a Helmholtz resonator, some as a special type of resonator, and others, not wishing to commit themselves, simply ignore it!

Reputable authorities in the acoustic field as well as several writers of magazine articles disagree widely on the principles involved, the descriptive equations, the equivalent diagrams, and even the symbols. Nowhere could the author find a simple down-to-earth explanation with a workable equation. Some of the equations advanced by various writers are not even dimensionally correct; others, while dimensionally correct, have no logical basis. Most important, they don't work. After pages of "design" information, each explanation ends up with, "for best performance, tune the port." Why bother? You might as well build a box, cut a hole in it, measure the response, and try again until it's right.

This is no way to build a speaker enclosure. So, with absolutely no qualms, the author worked out his own explanation, equivalent diagram, and equation using his own symbols. Undoubtedly they will not agree with those of certain authorities but they are bound to agree with some of the others! The important thing is, the equation works. For those who just want the facts, Table 1 gives all the information; for those who want more—read on.

A loudspeaker, with or without a baffle, is a mechanism that converts electrical energy into mechanical motion, which in turn is converted into acoustic energy. A speaker has quite a low impedance; coupled to an output amplifier tube through a matching transformer, it sees the high plate impedance of the tube. Thus, a speaker is fed from a high-impedance source and is essentially a constant-current device. To measure the impedance, feed the speaker from an audio signal generator through a high series resistance, Fig. 1. With a constant current through it, the voltage measured across the speaker will go up or down with the impedance. Fig. 2 was plotted for the author's speaker, using a v.t.v.m. and a b.f.o. output of 10 volts through 200 ohms.

Fig. 1. Measuring loudspeaker resonance.



The electrical resistance, inductance, and capacitance are of little importance at low frequencies; the mass of the cone, voice coil, and their supports and the compliance of the supports determine the resonant frequency. One other quantity is viscous friction set up by the cone moving through the air. It is low at low frequencies, increasing with frequency until the diaphragm circumference approximates the wavelength, and then stays nearly constant.

The current is the same throughout a series electric circuit and inductance is the controlling factor because it tends to keep the current constant. Current through a capacitance can be changed at will, so capacitance offers

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a "compliance" to current. In a series mechanical circuit, mass offers resistance to change in position or motion while a restoring element or spring supplies compliance. Mass, then, is analogous to inductance and compliance is analogous to capacitance. Viscous friction is the mechanical analogue of resistance.

Fig. 3 shows how these quantities are coupled to the electric circuit. The voice coil is part of the electric circuit and is *mechanically* coupled to the cone and its supports. *There is no electromagnetic coupling* as is shown in some equivalent diagrams. At mechanical resonance, the cone excursions become quite large, indicating minimum mechanical impedance, yet at the same time the measured electrical impedance becomes a maximum. Why the discrepancy? The answer is the voice coil moving in a magnetic field. As it moves, it induces a counter-e.m.f. in itself, reducing the effective applied voltage. This is equivalent to raising the voice-coil impedance. Minimum mechanical impedance produces maximum voice-coil motion and therefore maximum electrical impedance. When mechanical impedances are translated into their electrical equivalent, they are inverted and result in a parallel electrical circuit, Fig. 4. The effect of mass is now analogous to that of capacitance while compliance acts as inductance.

Another way of looking at it is: with a constant current, a higher impedance produces a higher voltage and therefore more power is used. More electric power means more movement of the cone. In either case, higher impedance goes with more cone movement so the impedance is a measure of the sound pressure output. Whether or not a given sound pressure at a given frequency is audible is determined by your ear. For example, the sound pressures at 30 and 120 cycles represented by Fig. 2 are about the same, yet the 30-cycle note sounds much weaker. The idea of a bass-reflex is to damp the speaker resonant peak and boost the frequencies below the peak.

If the speaker is placed in a completely enclosed box, the air inside the box will act as a spring, tending to reduce the cone movement. This is shown in Fig. 3 as an additional series compliance. Translated into the electrical circuit of Fig. 4, it acts as another inductance in parallel, raising the amplitude and frequency of resonance. See Fig. 5.

If we now cut a hole or port in the box, the springiness of the enclosed air will be reduced. The mass of air in the port acts as a virtual diaphragm; this, together with the compliance of the enclosure, forms a second tuned circuit. The enclosure compliance is common to both the speaker and port tuned circuits. This time, however, the compliance instead of the mass is the driving element. There is a time lag between motion of the air near the speaker and of the air mass in the port because of the compliance, or "squeezeability," of the enclosed air. At the port resonant frequency the time lag is that of a half wave and the virtual diaphragm moves in the same direction as the speaker cone. This "phase inversion" results in sound from the port being in-phase with that from the speaker while at the same time the speaker is heavily damped to reduce its output.

The effect of the port is equivalent to adding mass and viscous friction in parallel with C in Fig. 3 or inductance and resistance in series with $1/C$ in Fig. 4. At the frequency where the port and compliance resonate, the impedance across the circuit in Fig. 4 will be very low. If this is the frequency at which the speaker resonates, the low impedance of the series resonant circuit will shunt that of the speaker, holding down its normal resonant impedance rise.

The resulting curve, Fig. 5, shows two smaller peaks, one above and one below the speaker resonant frequency. The frequency of the dip between the two peaks is that of the enclosure and port; for a given volume and port area, this frequency will be the same *irrespective of the speaker used.* The speaker does, however, affect the amplitude at the dip and the amplitude and frequency of the two peaks. Above resonance the speaker-tuned circuit is capacitive, Fig. 4, while the port-tuned circuit is inductive. As the frequency is lowered, the reactances approach each other in value and eventually become equal; this produces the upper peak. Below this, the shunting effect of the port-tuned circuit predominates and causes the dip. At still lower frequencies, the reactances of the two tuned circuits are reversed, producing the lower peak. It is this lower peak that extends the low-frequency range. The small peak at the bottom of the dip is the result of placing the speaker and port as far apart as possible. It is a form of overcoupling of the tuned circuits.

Apparently, a complete mathematical description of the impedance of a speaker system involves Bessel functions of the first and second kind. However, over the frequency range to be considered, the impedance approximates that of a simple resonant circuit. No matter how it is produced, resonance always occurs when two elements provide equal amounts of opposing types of reactances. For a

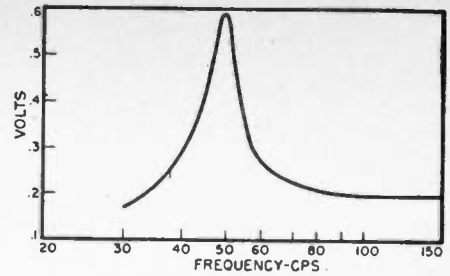

Fig. 2. Voltage measured across voice coil when the speaker is fed as shown in Fig. 1.

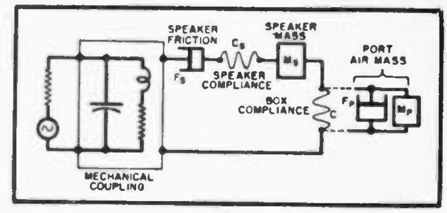

Fig. 3. Coupling of the mechanical elements.

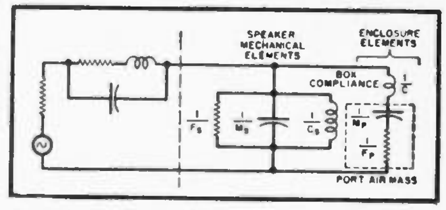

Fig. 4. The equivalent electrical circuit.

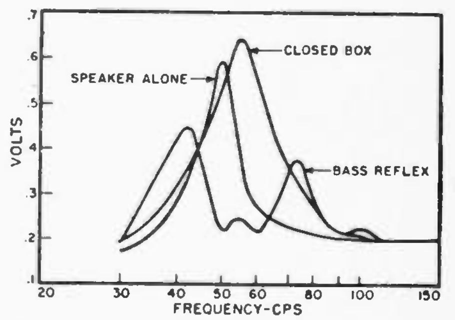

Fig. 5. Composite curves of the system.

Table 1. Basic formulas for determining cabinet dimensions and sizes of port areas.

$F_s = A\rho c$ above $2\pi R/\lambda \approx 1$

$X_{ms} = A\, d\rho c\omega$

$X_{cs} = A^2 \rho\, c^2/V\omega$

All values in mechanical ohms where A is the port area and R is the equivalent radius $\sqrt{A/\pi}$, both in square meters. ρ is the air density in kilograms-per-cubic meter; c is the velocity of sound in meters-per-second; V is the enclosure volume in cubic meters; ω is $2\pi f$; and d is the thickness of the enclosure wall at the port in meters.

$$f = \frac{c}{2\pi}\sqrt{\frac{A}{d'V}} = 54.7\sqrt{\frac{A}{V(d + 1.13\sqrt{A})}}$$

$$V = \frac{2992A}{f^2(d + 1.13\sqrt{A})}$$

and:

$$A = \frac{.64V^2f^4}{k^4} + \frac{f^2Vd}{k^2} \pm \frac{f^3V}{2k^4}\sqrt{1.63V^2f^2 + 5.12k^2Vd}$$

where: f is in cycles-per-second; k = 54.7, and all dimensions are in meters

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speaker system, the reactances can be expressed in three different units: electrical ohms, mechanical ohms, or acoustic ohms. The author chose to work with mechanical ohms because dimensionally they are easiest to handle.

Using dimensional analysis and common sense, the author came up with values for the viscous friction, mass reactance, and springiness (reciprocal of compliance) reactance in mechanical ohms that agreed with most of the texts. Setting the reactances equal, the equation for resonance was derived:

$$f = (c/2\pi)\sqrt{A/dV}$$

or, in a more familiar form:

$$f = 1/(2\pi\sqrt{Ac^2/dV})$$

where A is the port area, V is the enclosure volume, d is the thickness of the enclosure wall at the port, and c is the velocity of sound.

But, this is the equation most authorities give for a Helmholtz resonator! Such a resonator is usually spherical so that no two surfaces are parallel, the inside surfaces are smooth, and the largest dimension of the port is considerably less than the smallest dimension of the enclosure. The dimension d usually refers to the length of a small tube extending from the resonator and forming the port. All these criteria are disregarded in a conventional bass-reflex, so it cannot be a typical Helmholtz resonator. Besides, the equation doesn't work for practical enclosures.

What to do? Further reading brought to light the fact that a resonant column such as an open-ended organ pipe is slightly shorter than the sound wave produced. The length must be modified by an "end correction." If the pipe is open on both ends, the end correction must be doubled. The value of the end correction varies according to how the pipe is terminated, but for a pipe ending in an infinite baffle (flanged pipe) it is considered by most authorities to be 0.85 times the radius; for both ends, this becomes 1.7 times the radius. Since the wall thickness of a bass-reflex enclosure corresponds to the length of the exit tube of a Helmholtz resonator, it was assumed that a "finagle factor" was needed.

The thickness now becomes $d' = d + 1.7R$ where R is the equivalent radius of the port area, or $\sqrt{A/\pi}$. Unfortunately, the calculated frequency does not agree with that measured for an actual enclosure. Some port areas were tried as were the values of areas and resulting frequencies from published articles. Something was still wrong with the equation.

Considerable thought went into the problem and it was finally decided that the end correction was too small. The value for a pipe presupposes that the radius will be small in comparison with the length. If the pipe is shortened almost to nothing, certainly to less than the radius, the end correction becomes the major factor. What would be the limiting value? Assuming the thickness to be negligible, it seemed logical that the maximum value of the end correction should be 1.0, or equal to the radius. Since both sides of the port have to be considered, the "finagle factor" should be 2.0. The enclosure that resulted is shown in the photograph, page 77. The port area is 50 square inches; checking back from the measured curve, this gives a "finagle factor" of 1.96, approximating the value of 2.0, well within the accuracy of the measuring equipment. The final equation is:

$$f = 54.7\sqrt{A/V(d + 1.13\sqrt{A})}$$

with all dimensions in meters.

Since the cabinet had already been constructed, finding the proper port area was next. As can be seen from the equation, the lower the frequency of resonance, the larger the volume or the smaller the port. The port size affects the "Q" of the port-tuned circuit but not in any easily determined way. The "Q," in turn, affects the amplitude at the dip and the amplitude and frequency of the peaks. A generally accepted rule is that the port area should be nearly the same as that of the speaker. This is OK for small speakers but the enclosure gets unwieldy for large speakers because as the port area goes up, the enclosure volume must also go up to maintain the same resonant frequency. A good rule to follow is to make the port area A, in square inches, equal to the speaker resonant frequency when the speaker area, in square inches, is greater than the resonant frequency.

Ordinarily, the best procedure is to buy the speaker you want, decide on the value of A, and then find the volume required. Once the volume is known, you can decide on the shape of the enclosure to fit your needs. In finding A, use the *active* speaker area. This is the area out to, but not including, the first corrugation at the cone edge. The author's 15-inch speaker has an active diameter of only 12.5 inches. Once the port area is determined, the required volume can be found by:

$$V = 2992A/f^2(d + 1.13\sqrt{A})$$

again with all dimensions in meters.

Just a note at this point. The volumes of the enclosures discussed have been calculated from chosen values of port areas. In order that the port areas would be the same as speaker areas, typical values of active cone diameters were selected for several sizes of speakers and the active cone areas calculated.

Since speakers vary widely in construction, the active diameters and hence the calculations are only approximate. Moreover, all the data in Table 2 was calculated using a value of zero for the diameter, the thickness of the cabinet wall. From the equation for volume it is obvious that the thicker the cabinet wall, the smaller will be the volume required.

While the data given in the table is a close approximation of the correct volumes, it is not a substitute for individual calculations. Each particular speaker and enclosure must be calculated individually.

Because the author's enclosure is part of a TV set and bookcase combination, ½-inch plywood was used to keep the weight down. Three-quarter-inch or one-inch plywood is preferable. The builder's enclosure should be rectangular with at least one long front dimension. Don't make the depth too shallow, but try to avoid having any two dimensions the same. The author's is 20 x 22 x 30 inches inside. Use *both* glue and screws to put it together. Internal braces should be placed diagonally across each surface to reduce vibration.

The speaker and port should be at the ends of the long side, and it doesn't matter whether the enclosure is upright or on its side. If it is upright, move the port up from the floor a few inches. Don't use a shelf inside because this divides the volume into two acoustically-coupled resonators and will necessitate a return to cut-and-try methods to determine the shelf and port size. To avoid unwanted box resonance from the parallel surfaces, line the inside with some soft sound-absorbing material. The author used cotton because it is cheap and light, but better materials are available. Finally, don't forget to subtract the volume occupied by the speaker, braces, and soundproofing. —30—

Table 2. Approximate dimensions calculated for a few of the more popular speaker sizes. As mentioned in text, no allowance was made for thickness of cabinet wall.

SPEAKER	A (square inches)
6"	19.65
8"	33.25
10"	51.50
12"	78.50
15"	123.00

FREQ. (cps)	VOLUME (in cubic feet) 6-INCH	8-INCH	10-INCH	12-INCH	15-INCH
50	4.201	5.365	6.800	8.380	10.520
60	2.911	3.805	4.720	5.815	7.160
70	2.140	2.796	3.468	4.272	5.360
80	1.640	2.140	2.660	3.270	4.110
90	1.290	1.690	2.097	2.585	3.245
100	1.050	1.370	1.702	2.095	2.630

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Loudspeaker Damping and Tonal Response

By
GEORGE L. AUGSPURGER
Audio Research Laboratories

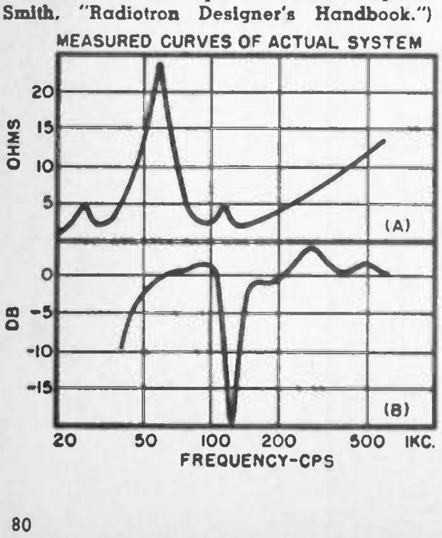
Fig. 1. Photograph showing the AR-1 Acoustic Suspension System speaker enclosure.

Recommendations for the proper setting of variable damping control for various types of speaker systems.

A GREAT DEAL of material has been written about loudspeaker damping—the significance of critical damping, the relationship of speaker efficiency to optimum damping, the use of variable damping to reduce distortion, and so on. Somehow in the excitement over this new technique a general impression has arisen that the only audible effect of juggling damping factors is a shift in apparent bass efficiency.

This is generally true in practice because most amplifiers are designed so that the damping control regulates only the range below 400 cps or so. Some circuits even include corrective filters to help compensate for the change in bass efficiency as damping is varied. This change, however, is not necessarily the smooth sort of variation one gets with tone controls. It is related to the impedance curve of the speaker system used and by looking at some simplified curves we can get a general idea of what happens.

Fig. 2. (A) Measured impedance and (B) axial response of actual system. (From Langford-Smith, "Radiotron Designer's Handbook.")



Knowing the response curve of a given speaker system with constant voltage drive and its impedance curve as well, it is easy enough to get a rough idea of what will happen with various damping factors by setting up two other general cases. The first is the constant current (I_c) system where driving voltage follows impedance, and the second is the negative impedance ($-Z$) system in which driving voltage is inversely proportional to impedance variations.

Fig. 3 is a common family of curves for a loudspeaker in an infinite baffle. Fig. 3A is the impedance curve and 3C might be the axial response of the same speaker driven by a constant voltage source. This is the same thing as saying that the amplifier has good regulation, or low internal impedance, or a high damping factor. Factors from 5 to infinity fall in this general classification.

Since impedance below 500 cycles is almost purely motional, the curve of Fig. 3A is also representative of relative cone travel. With a constant voltage source, the increased cone movement at resonance tends to offset the loss due to poor acoustic coupling and the response is reasonably smooth.

But if the speaker is driven from a constant current source, curve Fig. 3B is the acoustic response. A constant current source is an amplifier with very high internal impedance, poor regulation, and a damping factor of about 0.1 or lower. The output voltage rises with impedance, the cone is driven harder at resonance, and a ubiquitous boom is heard throughout the listening room. Notice that even in these simplified examples, the peak of Fig. 3B is not the sort of thing one can flatten out with tone controls.

What happens if the output voltage follows an inverse relationship with impedance changes? This is the situation of Fig. 3D where damping has been shifted to the negative region and amplifier internal impedance is less than zero. In the case of extremely inefficient speakers, the amplifier impedance may be made negative without even reaching the curve of Fig. 3C, much less the 6 db-per-octave slope of Fig. 3D.

Although most enclosed cabinet speakers are designed to work with damping factors from 9 to 20, some will not reach the point of smoothest bass until damping becomes quite low. The AR-1 Acoustic Suspension System (Fig. 1) has an optimum factor of one. What figure is best for a given system depends upon acoustic damping, maximum impedance variation, magnetic flux density, and all sorts of things. The *Hartley* speaker, for example, has an insignificant impedance peak in its "Boffle." Obviously, wide variations in damping factors will have little effect on its response.

Things become more interesting when we investigate the effects of variable damping on bass-reflex type

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systems such as the *Jensen* "Bass Ultraflex" cabinet of Fig. 6. The family of curves for vented systems is shown in Fig. 4. The fly in the ointment here is that while impedance is still related to cone movement, the acoustic output of the system is not. Too low a damping factor will peak the response of the system about a half octave above vent resonance, giving a nice "thunky" effect. Running the source impedance into the $-Z$ region, however, has consequences even more weird. The upper impedance peak is "damped out" all right, but the main system resonance is completely out of control.

An interesting point in connection with these bass-reflex curves is that the $-Z$ system gives *more* audible bass than the constant current circuit. While for an enclosed cabinet speaker, a negative damping factor may effectively swamp out resonances, when used with a reflex system the chamber resonance is accentuated instead.

A horn-type system would be expected to have high acoustic damping and small variations in impedance. When this holds true, electrical damping has little to do with frequency response, as can be demonstrated experimentally. Commercial horns, however, employ acoustic resonance to extend their bass response below the horn cut-off frequency and this resonance is reflected in the impedance characteristics of the system. Fig. 5 is the *Jensen* "Imperial" which uses a "reactance annulling" principle to deliver usable bass below 35 cycles.

The *Klipsch*-type horns with enclosed back chambers have impedance curves similar to that of an infinite baffle and react somewhat the same way to variations in damping. Most of these large units are designed to work under conditions approximating constant voltage input. Paul Klipsch suggests a damping factor of 4 or more for his speakers, and reports that any attempt to use constant current feed, or negative damping factors, invariably deteriorates the performance of the system.

Smaller rear-loaded horns such as the *Klipsch* "Rebel" and *Electro-Voice* "Aristocrat" have impedance curves more akin to reflex cabinets. Unless the response of Figs. 4B and 4D are what you want, damping should be set at the figure recommended by the manufacturer.

In spite of the instructions furnished, many users try to adjust the damping control to the kind of bass response they happen to like. The illustrations given should explain why an audiophile with this philosophy is unlikely to find satisfaction. Moreover, the curves shown are merely representative and greatly simplified; the relationship between impedance and acoustic output for any given speaker may vary rather sickeningly from these graphs. Fig. 2 shows the impedance and frequency response of an actual system.

It must be remembered that the ordinary listener has no idea of the impedance curve of his speaker system and has probably never looked closely at the instructions to see what damping factor is recommended. In this situation, it is no wonder he is confused by a damping control. Things haven't been helped by some salesmen who twist the control to minimum damping—"Hear that bass come out now?"—and mumble something about the amplifier automatically adjusting to speaker impedance.

A variable damping control is valuable because different speakers do work better with different damping factors, but once the proper source impedance has been established, the damping control should be hidden and forgotten until the speaker system is changed. —30—

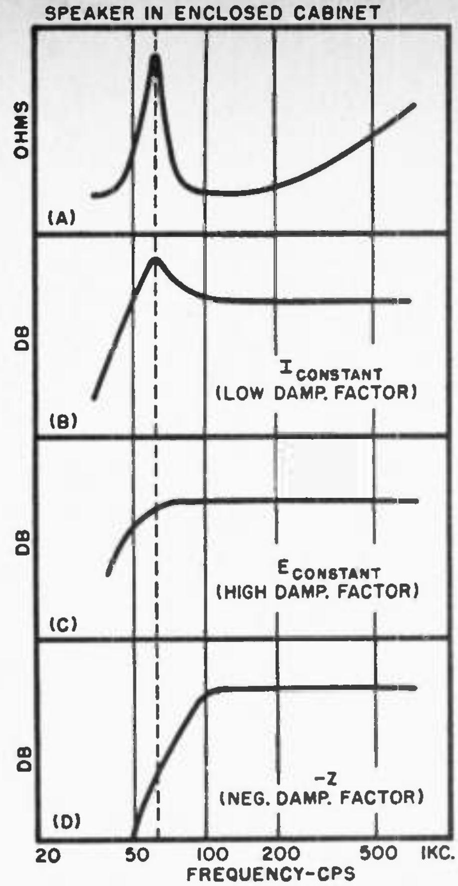


Fig. 3. (A) Impedance, (B) response with constant current, (C) constant voltage, and (D) negative impedance sources of signals.

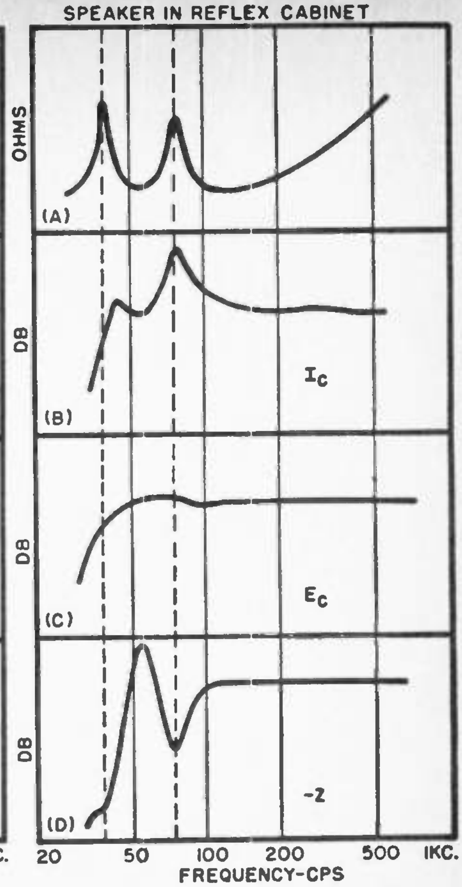
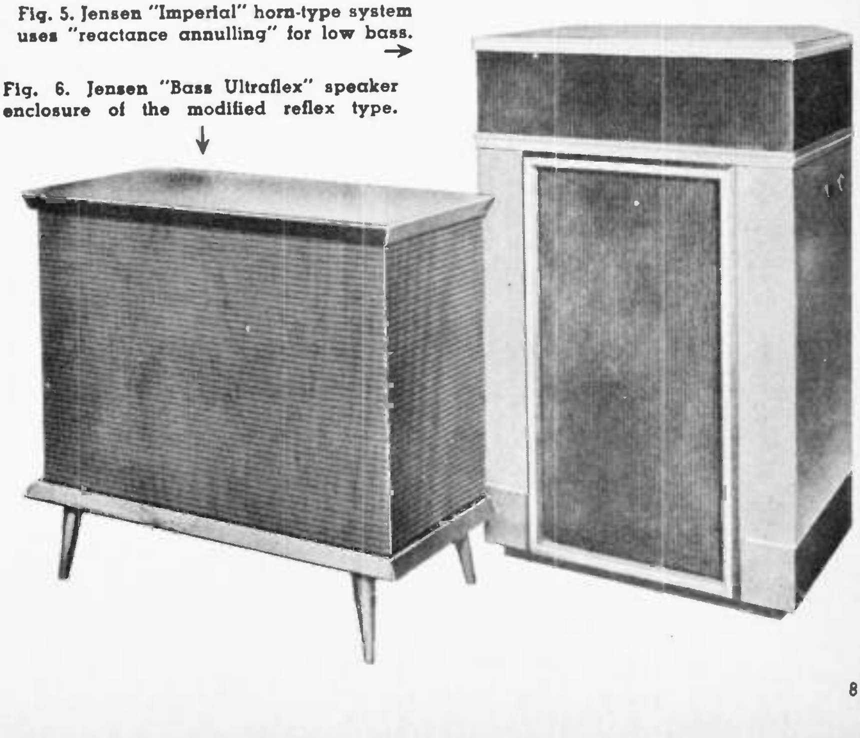


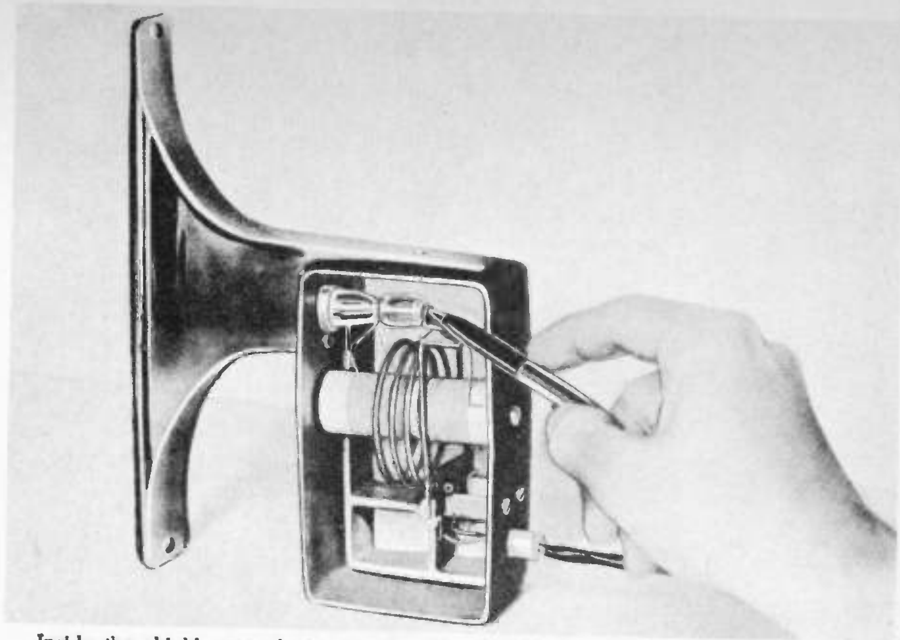
Fig. 4. Curves corresponding to Fig. 3 pertaining to speaker installed in a bass-reflex rather than completely closed cabinet.

Fig. 5. Jensen "Imperial" horn-type system uses "reactance annulling" for low bass. →

Fig. 6. Jensen "Bass Ultraflex" speaker enclosure of the modified reflex type. ↓



Ionic Cloud Tweeter



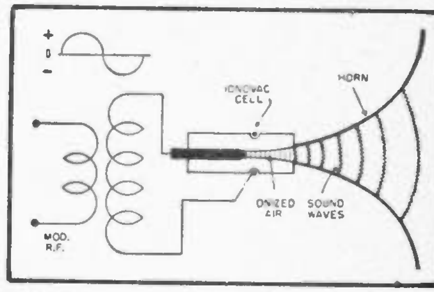
Inside the shield case of new tweeter. Pencil points to quartz cell, heart of unit.

RECENTLY, Electro-Voice, Inc., introduced a radically new high-frequency tweeter, the "Ionovac". This is an ionic cloud transducer which uses a small quartz cell in which corona discharge is set up. As a result of the changing ionization of air in the cell, sound waves are set up directly in the air without the need for the conventional diaphragm used in ordinary tweeters.

At a recent demonstration of this unit, originally shown in this country by the *DuKane Corp.* of St. Charles, Illinois, the tweeter appeared to be remarkably clean without noticeable peaks or unnatural coloration. This unit could be very favorably compared with some of the better electrostatic tweeters. With the tweeter horn mounted with its long dimension vertical, good horizontal dispersion of highs occurred.

The response curve of this unit is within 2db from 2000 to 40,000 cycles. The low frequency limit is set only by the horn dimensions. The recommended crossover point is 3500 cycles. The tweeter is capable of handling 50 watts of program material, 100 watts on peaks.

Shown inside the open housing above are the r.f. link coils to which are fed 17 watts of 20 mc. amplitude modulated r.f. from a companion modulated oscillator and power unit. It is this unit, with its 2 per-cent distortion, that limits quality of the output. Extensive shielding prevents interference. Since there is quite a bit of heat produced at the electrodes within the cell, it is expected that these would have a life "only as long as a permanent phono stylus." Replacement is quite simple, though, since the little cell may be changed as a unit. —30—



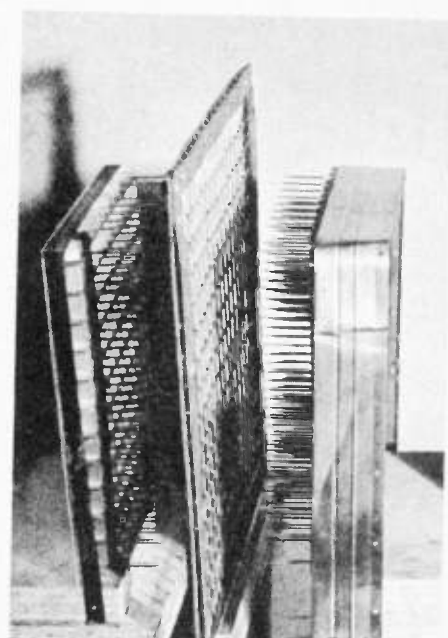
Principle of operation of the new tweeter.

Corona Loudspeaker

IT HAS been known for many years that corona, a byproduct of high voltage, is accompanied by a movement of air away from the discharge point. TV service technicians and experimenters know this from their work with high voltage. Reasoning that any source of wind might also be a potential source of sound, since sound is a sort of a.c. wind, Dr. David Tombs, a New Zealand born engineer, constructed a special apparatus to study the phenomenon more closely and to find a method of modulating the corona winds.

Starting with a pair of electrodes, one of which was sharply pointed and the other blunt, it was found that a corona wind moves away from the sharp point only. The next step was to mount a smooth ring coaxially around the sharp electrode, in the expectation that it might be made to act as a grid to control the corona current, and hence the corona wind. The inventor found that with a grid potential of "the same order of voltage as the sharp electrode" there is a position of the grid with respect to the tip of the sharp electrode "at which comparatively small changes of the grid potential produce a maximum effect on the intensity of the corona wind . . ."

The amount of sound produced by a single corona triode is very faint, so that it becomes necessary to parallel



or stack a number of them. The first small model was constructed with a needle spacing of one-half inch, and with an area of a quarter of a square foot, as shown in the photograph.

The frequency response of the "Corona Loudspeaker" is much smoother than that obtained with cone type speakers, and it is expected that in its perfected form it will be essentially flat from zero to or beyond the upper limit of audibility. Even in its present form there is *no audible resonant frequency* at the low end.

Patent applications have been filed on the "Corona Loudspeaker," and exclusive rights for North America have been assigned to the *Televex Co.*

Audio engineers who have seen and heard the "Corona Loudspeaker" even in its present elementary form are considerably impressed with its possibilities, and some have expressed the opinion that it may well be the "loudspeaker of the future." However, you will not be able to buy one next week, and possibly not for another year or so, since further research and development are required to determine all the optimum parameters, both mechanical and electronic, before it can be produced commercially. But when it does reach the market, the "Corona Loudspeaker" should prove to be a contender in the hi-fi sweepstakes. —30—

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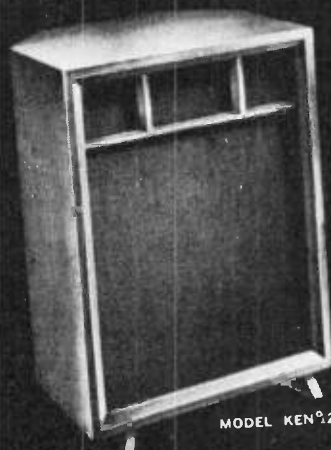

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Adapter for mounting 12" speakers in KEN-15 $1.50 User Net

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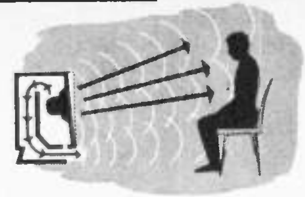

KwiKit acoustic design and tilted baffle combine direct speaker radiation and compensated rear horn loading in a way that blends bass, middle and treble ranges perfectly . . . for *uniform* response throughout the listening areas of a room.

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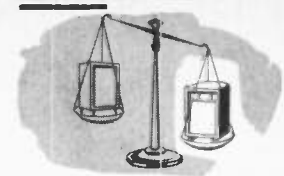

Heavy ¾" first grade, fully cured lumber for top, bottom, sides and back . . . *not* flimsy ½" wood commonly used in "kits." Bigger, sturdier — *as much as 30% heavier* than others in the same price class. Eliminates spurious resonances so detrimental to achieving richer, cleaner bass reproduction.

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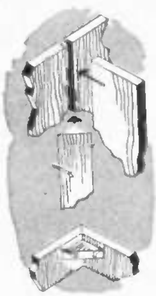

Exterior and interior elements, even the cleats, fit snugly within close tolerance "rabbeted" grooves. Gluing and screwing of each piece results in reliably air-tight, permanent joints. No nails used. No pencil markings necessary. Mitering and plenty of glue blocks and bracing for truly rigid construction.

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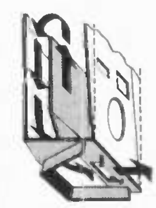

Underside view shows how advanced design, self-contained folded horn extends to the *front* of the cabinet, projecting low frequencies out into the room . . . not back into a corner, splashed against the walls. Small slot in base is resistively controlled vent which equalizes woofer diaphragm excursions in compression chamber. KwiKits are therefore independent of room furnishings, shape or placement and can be used against a flat wall, in a corner . . . even up in the air!

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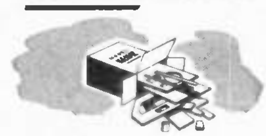

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Your KwiKit includes all required hardware, plastic wood, glue, sandpaper, Tufflex insulation, easy-to-follow instructions *and* . . . special attention is given to packaging of KwiKits to insure safe, intact delivery to your door.

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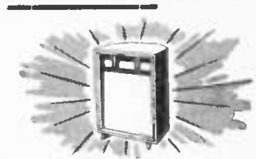

There's no end of decorative treatment you can give your KwiKit enclosure. Genuine Korina veneer is same as used in fine furniture, and provides a beautiful finish. Decorative front mouldings have been designed to complement and enhance your present decor. Exquisite, textured grille fabric is equally at home in settings of any period and is acoustically correct to prevent high frequency attenuation.

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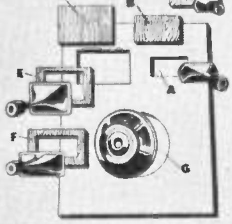
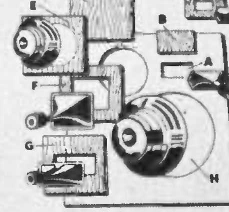


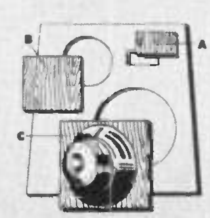
A—Hole cut out for HF-206.
B—Blank plug supplied when tweeter isn't used.
C—Adapter supplied cut out for UXT-5.
D—Blank plug supplied.
E—Adapter supplied cut out for 4409.
F—Adapter supplied cut out for H-600 horn.
G—Takes 312, UXC-123, Diffusicone-12, UXC-122, Diffaxials, 6200, 6201 wide-range speakers and C-12W woofer.

MODEL KEN-12
takes any 12" wide-range or woofer cone speaker and any tweeter or mid-range speakers.



A—Hole cut out for HF-206.
B—Blank plug supplied when tweeter isn't used.
C—Adapter supplied cut out for UXT-5.
D—Blank plug supplied.
E—Adapter supplied cut out for C-8W or Diffusicone-8.
F—Adapter supplied cut out for 4409.
G—Adapter supplied cut out for H-600 horn.
H—Takes 315-C, 6303, Diffusicone-15 Diffaxials, and C-15W, C-63W woofers.

MODEL KEN-15
takes any 15" wide-range or woofer cone speaker and any tweeter or mid-range speakers.



A and B—Blank plugs supplied.
C—Takes 312, UXC-123, Diffusicone-12, UXC-122 Diffaxials, 6200, 6201 wide-range speakers and C-12W woofer.

MODEL KEN-15
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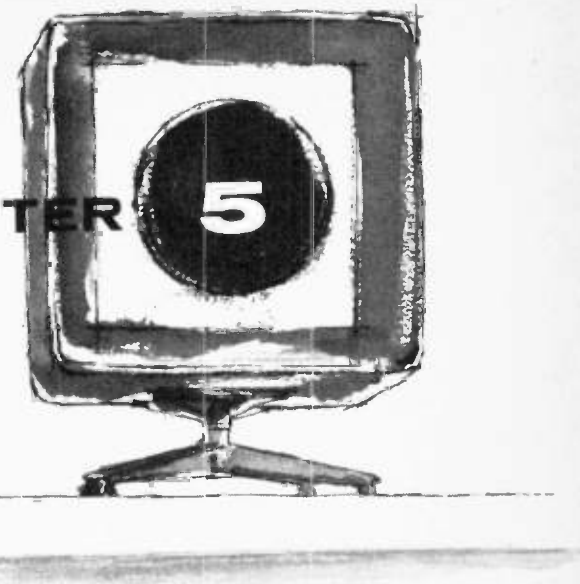
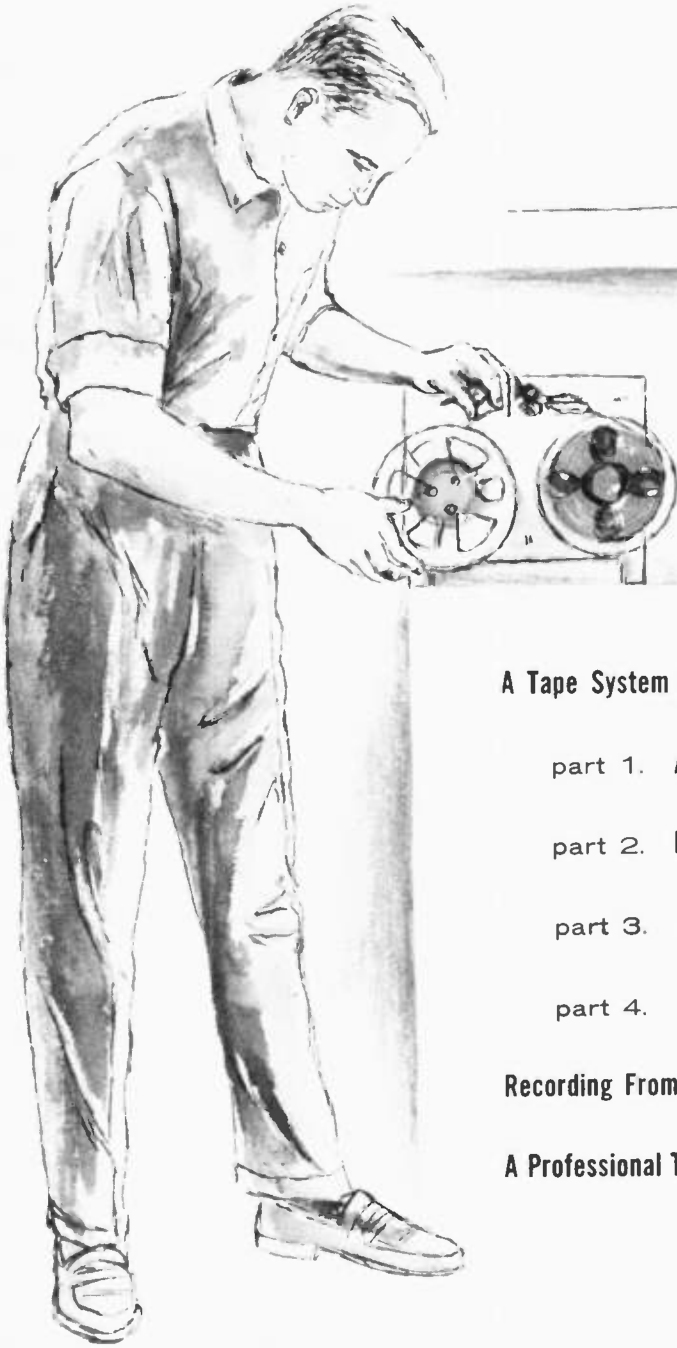
CHAPTER 5

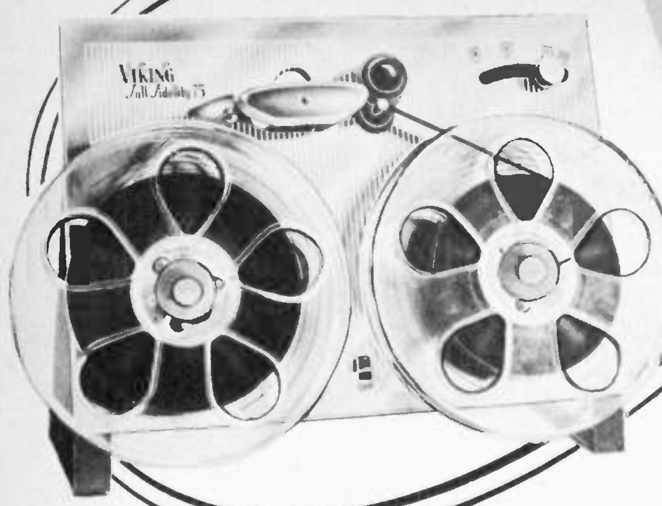




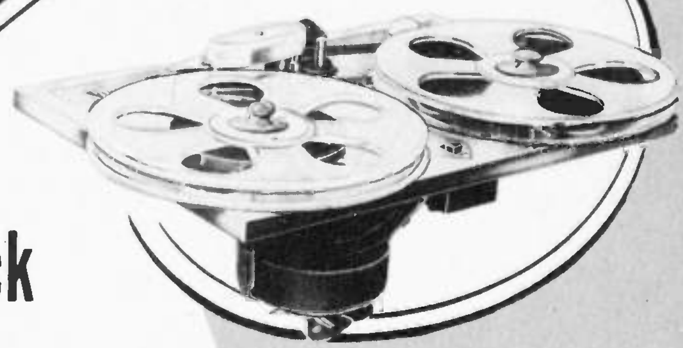
TAPE RECORDING

World Radio History

A Tape System You Can Build



Two views of the Viking FF75, a basic tape deck which can be used in all home systems.



A Compatible Tape Deck

Part I. This basic tape deck can be used by the hi-fi builder for a monaural or stereo record-playback installation.

By JOHN L. MacALLISTER

Viking of Minneapolis

IN A MATTER of years tape equipment and, specifically, "tape recorders" have graduated from the realm of mystery into the category of "assemble-it-yourself" components with which the audiophile feels completely at home.

Tape decks and their associated electronic components are now available from many manufacturers. The audiophile can purchase one of these basic decks and add the necessary preamplifiers and recording amplifiers *ad lib* or hook the deck up to the preamps and amplifiers he already has in operation.

One of the most flexible of these component tape decks is the *Viking* FF75P which retails at $71.50. It is basically a tape transport for playback (both full- and half-track) only and can be used with most present-day high-fidelity preamps and power amplifiers without the need for additional accessory equipment.

Many of the high-fidelity preamps now being marketed provide a preamplifier input equalized expressly for tape. Any of these units may be used with this tape deck. Similarly, some amplifiers provide a means of adjusting roll-off and attenuation to provide almost any desired characteristic. These may be adjusted to provide the equalization characteristic indicated in Fig. 1. Preamplifiers with fixed equalization to compensate for the various disc recording characteristics require modification for use with NARTB recorded tapes.

For the user who does not have a preamplifier, the company's PB60 unit is available. It is specifically designed to be used for this purpose and provides exact equalization for NARTB recorded tapes. Its output can be introduced at the input of a power amplifier.

Considering the cost of this new tape deck, it is obvious that it was not intended to compete with the more expensive units available today. It does, however, provide many of the features of better quality units. It has negligible flutter and wow; provides a frequency response of 40 to 14,000 cps (±3 db) at a tape speed of 7.5 ips; and is easily adaptable to many varied modes of operation.

It can be used for monaural playback, monaural record and playback, stereophonic playback, and stereophonic record and playback by simply adding the necessary heads and the required erase and recording bias. In this respect the deck may be considered to be "compatible."

This and succeeding articles will cover in detail the design and construction of the various electronic units required to convert this tape deck to its many possible modes of operation. In addition, the mechanical problems of installing the additional heads, for these various operations, will be discussed.

"Dynamu" heads, manufactured by the *Maico Company,* a Minneapolis concern, are used exclusively on all of the FF75 series decks. This head, one of the best in the industry, is characterized by a very short flux gap. This short gap, .00015" (.15 mil), provides for full playback response at frequencies to 14,000 cps (±3 db) at a tape speed of 7.5 ips. The electrical specifications for the record-playback and erase heads are given in Table 1.

The Tape Transport

In the interest of economy and for the purpose of providing maximum over-all rigidity of the assembly, the motorboard on which the deck is mounted consists of a single stamping of 16 gauge steel. The thickness of the steel prevents local distortion which would affect performance. The deck itself measures only 12⅛" x 8⅝" and requires little mounting or shelf space. The 7" tape reels extend approximately an inch beyond the sides and bottom of the deck.

The motor is ordinarily a very critical component of a tape drive system. On this item alone a manufacturer can spend almost as much as he chooses. The ultimate in linearity is the hysteresis synchronous motor found in transports designed for professional use. Such a motor is desirable in that the output is completely linear, *i.e.,* it suffers no interpole variations in rotor speed as does an

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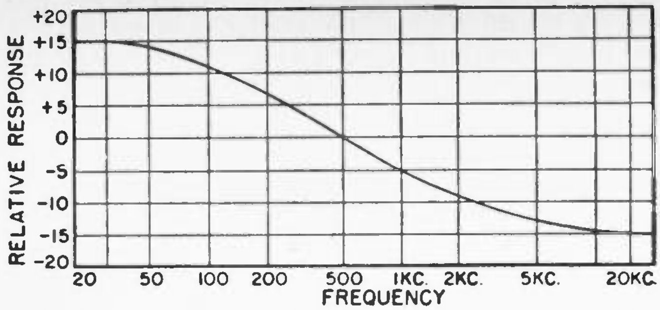
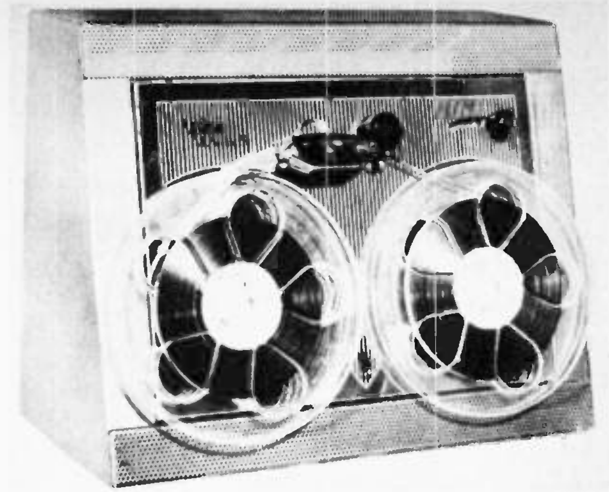


Fig. 1. Playback equalization required for the FF75 tape deck.



The consolette shown here consists of the standard Viking tape deck mounted in a special cabinet obtainable from Viking. It is the model 400 and nets for $14.

ordinary induction motor. Also its speed is absolutely constant when operating within its load limits. However, the cost of such a motor alone would make the compatible deck much too rich for our average man's high-fidelity budget. So, like other manufacturers of popularly priced decks, we must compromise on the universally used four-pole induction motor—proven in turntables and tape transports for many years—but characterized, unfortunately, by certain inherent problems of flutter and speed regulation.

At this point, however, *Viking* departs from heretofore accepted practice. The motor is mounted on a separate motor platform and this entire assembly is floated on resilient, vibration-absorbing, rubber shock-mounts. This is the first important point of departure. The second is in the capstan drive itself. Instead of a roller or conventional belt drive, the capstan is driven through a slender, resilient synthetic belt, leaving the motor free to move within the restricted orbit of the shock-mounts. Actually, any other type of drive would be difficult under this circumstance; however, the advantage is two-fold. The resilience of the belt in this case is matched to the inertia constant of a very heavy capstan flywheel (approximately one and a half pounds). The effect is to provide a filter for smoothing out the interpole variations in motor speed, the most common source of flutter. It is equally effective in the elimination of irregularities due to motor or fan unbalance.

The capstan, as in any tape equipment, is a very critical component. The capstan on the *Viking* FF75 is ground to a tolerance of 1/10,000th of an inch. An eccentricity of as little as 5/10,000th of an inch here would result in noticeable flutter. The same critical limitations apply to the capstan bearings and to the capstan roller.

Long Term Accuracy

One disadvantage of the induction motor that is not easily eliminated is that of long-term accuracy of tape speed. Through rather careful check and selection of motors, where necessary, the accuracy in this respect is held within 2 per-cent c~ about 1.2 seconds per minute. While not accurate enough for cueing or synchronizing effectively, this is not sufficient to appreciably affect pitch. It should be pointed out that the error, whatever it may be, is a positive or negative error consistent for a given deck.

Shift Mechanism

The shift mechanism involves only a few moving parts—all flat steel stampings, reinforced when necessary. In the center position, both the feed reel and takeup reel brakes are engaged and the capstan pressure roller is released. The capstan continues to turn but the tape does not feed.

In the "forward" position, the supply reel brake is permitted to drag slightly, the takeup reel brake is released, and the pressure roller engages the capstan, causing the tape to feed. The takeup reel is belt-driven from the capstan flywheel but provides sufficient slippage so that only a moderate and uniform tension is exerted on the tape. In the "rewind" position, the supply reel is shifted a few thousandths of an inch engaging a drive ring on the motor pulley, providing a brisk rewind. Actual rewind time is 1¼ minutes for a standard 7" reel.

The "fast forward" mode on the FF75 deck is a comparatively recent development. Actually this mode, which has been characterized as "gentle fast forward," is not really a "fast forward" in the general sense of the word. It provides for tape transport at four to five times normal speed yet offers maximum protection for the new thin-base recording tapes currently enjoying widespread popularity.

In order to adapt the basic tape deck to the other modes of operation mentioned earlier, the deck is modified merely by the addition of the appropriate head or heads and bracket components. The heads are supplied complete with output connector jacks and can be installed with simple hand tools. Only minor disassembly of the deck is involved and adjustments entail nothing more, in most instances, than careful azimuth alignment.

The over-all performance of the deck is given in Table. 2.

Table 1. Specifications on the "Dynamu" heads used with deck.

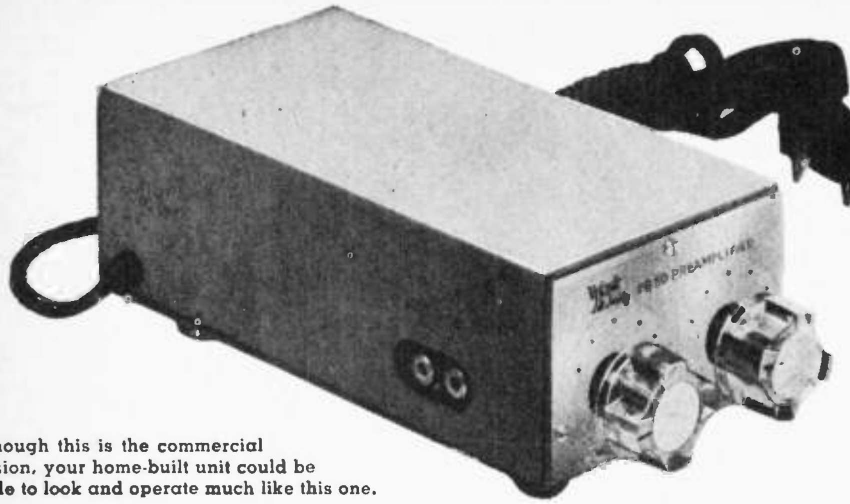
RECORD-PLAYBACK HEAD	
Track width	.090"
Gap length (effective gap can be considered as .0002")	.00015"
Impedance	2000 ohms @ 1000 cps
Frequency response	30 to 15,000 cps @ 7.5 ips
Head output (average recorded music tape)	2.5 mv.
Recommended preamplifier gain	55 db
Driving current (recording)	0.06 ma. @ 6000 cps
Recommended bias current (recording)	0.6 ma.
ERASE HEAD	
Track width	.125"
Gap length	.008"
Inductance	30 mhy.
Recommended erase bias frequency	60 to 70 kc.
Recommended erase current	12 to 14 ma.

Table 2. Over-all performance of the Viking FF75 tape deck.

Frequency response	40 to 14,000 cps ± 3 db, 7½ ips 40 to 7000 cps at 3¾ ips
Signal-to-noise ratio	45 db or better (This characteristic can be adversely affected by location of transformers or other sources of strong magnetic fields within 12 inches of the head or heads)
Flutter (average)	0.2 per-cent or less
Total harmonic distortion	Less than 1.0 per-cent with NARTB recorded tape and NARTB compensated preamplifier
Tape speed	7½ ips (3¾ ips available by changing belt to smaller groove on motor pulley)
Maximum reel size	7"

World Radio History

Playback Preamplifier



Although this is the commercial version, your home-built unit could be made to look and operate much like this one.

Part 2. To obtain quality reproduction from your tape unit you will need a playback preamp. There are three choices—either buy or build your own unit or add proper equalizing network to your present preamp. Here are all the details.

THIS PART covers the playback preamplifier required for a custom-type home tape recorder designed for playback only. Such a preamplifier must incorporate the proper equalization circuits for commonly available prerecorded tape.

If stereophonic playback is the final goal, two preamplifiers, one for each channel, will be required. If record and playback are desired at the outset, it would not be necessary to use a separate preamplifier as described herein. Instead, the user should build the combined record-playback preamp and bias oscillator that will be described in next article.

The Preamplifier

Two, or possibly three, alternatives are provided in the matter of a preamplifier. The first, and possibly most attractive, alternative would be to use an existing phono-preamplifier providing it has an unused equalization position that can be altered to give the required equalization setting.

Let's take, as an example, the man who owns an all-purpose phono-preamplifier which provides adequate gain for tape, but doesn't include correct equalization for NARTB recorded tapes. The prospect of modifying this amplifier for use with tape would certainly be the most attractive of the three alternatives to be suggested.

First, however, let's delineate the requirements for a good tape preamplifier to make sure this phono-preamplifier will be suitable. In playback of recorded music tapes the head output will average approximately 1 to 3 mv. (A head manufacturer is apt to rate his product at 5, or even 10 mv., but this, it must be realized, is with tapes recorded to saturation.) A one-volt output will be required from the preamplifier in order to drive the average music system amplifier to normal output. This, in itself, represents a gain of approximately 50 db. It will be shown also, that at least an additional 25 db of gain will be dissipated in a properly designed equalizing network. Thus, the net requirement is for approximately 75 db of gain before equalization. This gain can be provided by a pentode voltage amplifier, followed by a single triode, or by three triode stages. Two triodes usually fall just short of the mark.

The second possibility would be to construct a suitable preamplifier using the circuitry suggested in the article. All of the components except the transformer (T_1) are readily available at electronic supply houses.

The third, and possibly simplest way, would be to purchase a tailor-made unit. Excellent general purpose amplifiers featuring NARTB tape equalization, in addition to the various phono equalization characteristics, are now available. A preamplifier designed expressly for a tape system, such as the *Viking* PB60 preamplifier described here, may be purchased for $29.50 as a plug-in-and-play accessory.

Next, let's look at the equalization characteristic which is required.

The NARTB Standard

Most manufacturers of tape heads provide a response curve which, except for slight departures in frequency range or linearity, has the general conformation shown in Fig. 1. This, it should be pointed out, is the over-all record and playback characteristic of the head, based on *constant current recording on tape, and subsequent unequalized playback of the tape.*

To obtain this curve a sine wave is applied to a given head at constant current and at selected chromatic test frequencies. The output is recorded on tape. The same head is then used without equalization to play back the recorded track. The head output at these several frequencies serves for plotting the curve shown.

This curve at 7.5 ips, it will be seen, can be divided into three segments. It shows a rising characteristic from approximately 30 cycles, a short plateau of approximately flat response, and a descending response characteristic thereafter. Secondary curves (shown dotted) indicate the performance of the same head at 3.75 ips and 15 ips, respectively.

The 6 db-per-octave rise displayed over the first segment is typical of all record-playback heads. The increase results from the fact that with the current held constant, doubling the frequency results in doubling the voltage at the head. This, therefore, is merely a uniformly rising characteristic. The point of departure between representative heads is the plateau at which the response levels off. This can be described as the region of maximum response and is determined almost entirely by the length of the head gap and the tape speed.

The head output then decreases very sharply as the frequency becomes such that the length of the head gap approaches the length of a wavelength on the recorded tape. There are some lesser contributing factors such as losses due to eddy currents, I^2R losses, etc., but, in general, this falling off in response can be termed a "wavelength" loss.

One other irregularity or loss appears on this curve, *i.e.*, the falling off below approximately 50 cycles. This is customarily referred to as a "wrap-around" loss. It begins at the frequency where the length of a wavelength on the recorded track exceeds the length of the head itself. This too can be compensated for, but may well be ignored, since it represents only a negligible loss at the commonly usable audio frequencies above 30 cycles.

Prior to the adoption of the NARTB standard, tape recorder manufacturers

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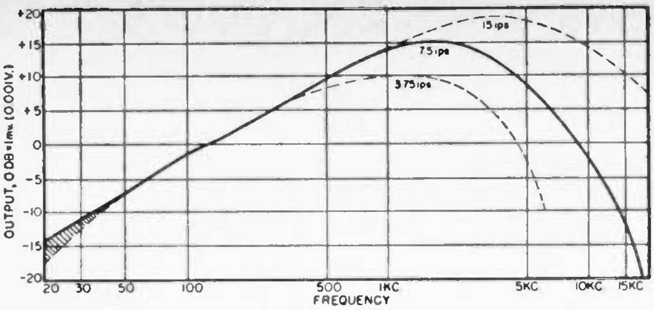


Fig. 1. Record-playback response characteristic of Dynamu 75-1 tape head. (Constant current recording, unequalized playback.)

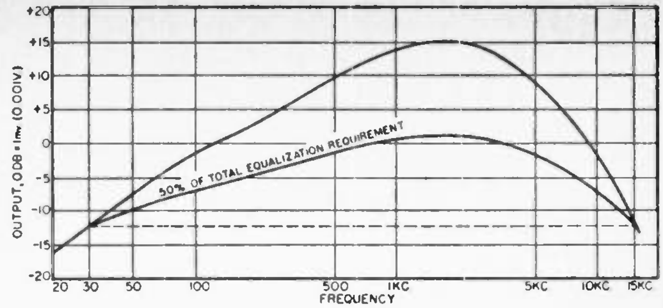


Fig. 2. A typical pre-NARTB equalization characteristic. Since there was no standardization, this varied widely in industry.

solved the problem of equalization in accordance with their own individual preferences. The most common practice was to halve (approximately) the equalization requirement, as shown in Fig. 2, attenuating the middle range by a factor of fifty per-cent in recording and the balance during playback. This entails a minimum of circuitry; the same equalization network suffices for both recording and playback. The disadvantage, however, is that such a system provides a smaller dynamic range, since it does not provide for pre-emphasis at the higher frequencies to compensate for the decreasing retention characteristic of the tape itself at these frequencies. As a result, the tape is likely to be overloaded at the lower frequencies in an effort to obtain sufficient high-frequency response.

The NARTB standard has been thoroughly covered in the article "Why the NARTB Curve for Magnetic Tape?" (in Radio & TV News, June 1955 issue) and is too involved for explanation here, but briefly it amounts to this. It was decided that the optimum means would be to compensate for the "wavelength loss," as well as the high-frequency deficiencies of the tape itself during the recording process, and to compensate for the 6 db-per-octave rise to the region of maximum response during the playback half of the cycle. This provides flat response for the over-all record and playback cycle, and at the same time provides maximum dynamic range.

Actually the current NARTB standard, as established, applies only to tapes recorded at a speed of 15 ips; however, at 7.5 ips it is only necessary to increase the pre-emphasis slightly at frequencies above 3000 cycles.

In the preamplifier, therefore, it is necessary only to compensate for the 6 db-per-octave slope from 30 to 2500 cycles. The preamplifier response characteristic which accomplishes this function is shown in Fig. 4.

Equalization for Existing Units

The actual circuitry employed in various commercial preamplifiers is too varied to permit individual treatment, but if schematics are available for reference it is not difficult to trace the circuit for a given switch setting, remove existing equalization components, and replace them with an *RC* network having the proper values.

Practically all commercial preamplifiers of recent vintage fall into one of two types; amplifiers employing triodes (or double triodes) such as 12AX7, 6F5, 12AY7, 6AV6, or 6AT6 tubes, or amplifiers employing pentodes such as the 5879, 6J7, or 6AU6 tubes. Fig. 3A shows the *RC* network required to provide a 6 db-per-octave equalization characteristic using triode amplifiers, and Fig. 3B shows the *RC* components required for pentodes. The tables provided beneath these diagrams indicate the correct component values for given values of plate load resistors commonly used.

It should be pointed out that if an inverse feedback loop is employed, the effect of an equalizing network inserted within that loop is not practical unless the feedback loop itself is redesigned or removed. Similarly, in some preamplifiers equalization is derived by means of frequency discriminating feedback loops. In such cases it is more practical to build the preamplifier described here than to attempt redesign of the equalization circuit.

Building a Preamplifier

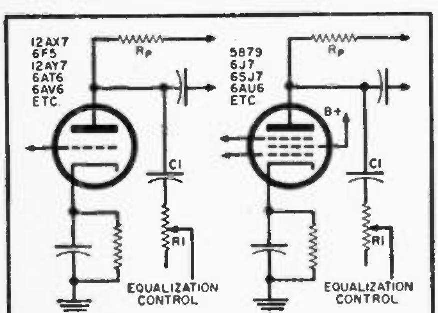
The preamplifier circuit, shown in Fig. 6 and described here, is equivalent to that of the *Viking* PB60, designed for use with the *Viking* decks and, consequently, the "Dynamu" heads. However, since the equalization is variable over approximately a 14 db range at 10,000 cps, the same circuit can be used satisfactorily for any of the currently-available heads.

A type 5879 pentode amplifier was selected as most suitable for the first amplifier stage because of its inherent low noise level and freedom from microphonics. Following this pentode amplifier stage, one section of a 12AX7 double triode is employed as a voltage amplifier. The other section of the 12AX7 tube is used as a cathode-follower output stage.

Construction

Since the preamplifier must operate from a very low signal level and provide high gain, good design and careful construction are essential. The input circuitry must be completely isolated

Fig. 3. (A) Equalization for a triode amplifier and (B) for a pentode amplifier.



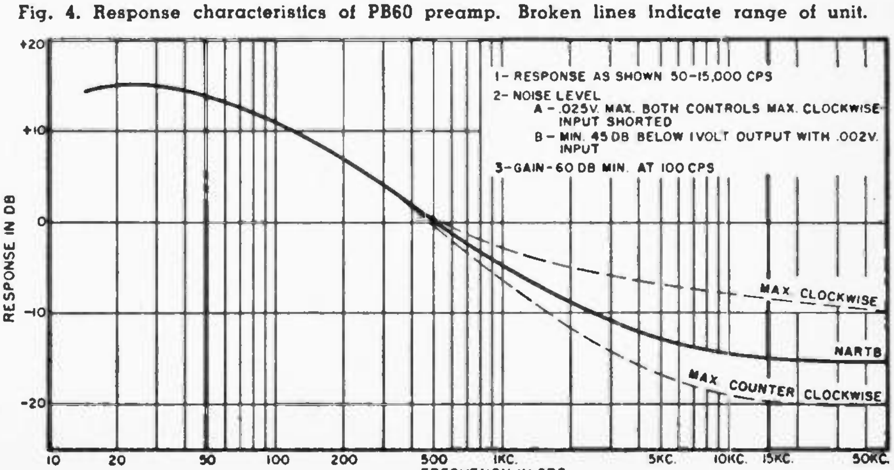
Rp	C1	R1
100K	0.05μfd	10K
330K	0.047μfd	15K
560K	0.033μfd	20K

(A)

Rp	C1	R1
220K	0.01μfd	25K
470K	0.008μfd	30K
680K	0.005μfd	50K

(B)

Fig. 4. Response characteristics of PB60 preamp. Broken lines indicate range of unit.



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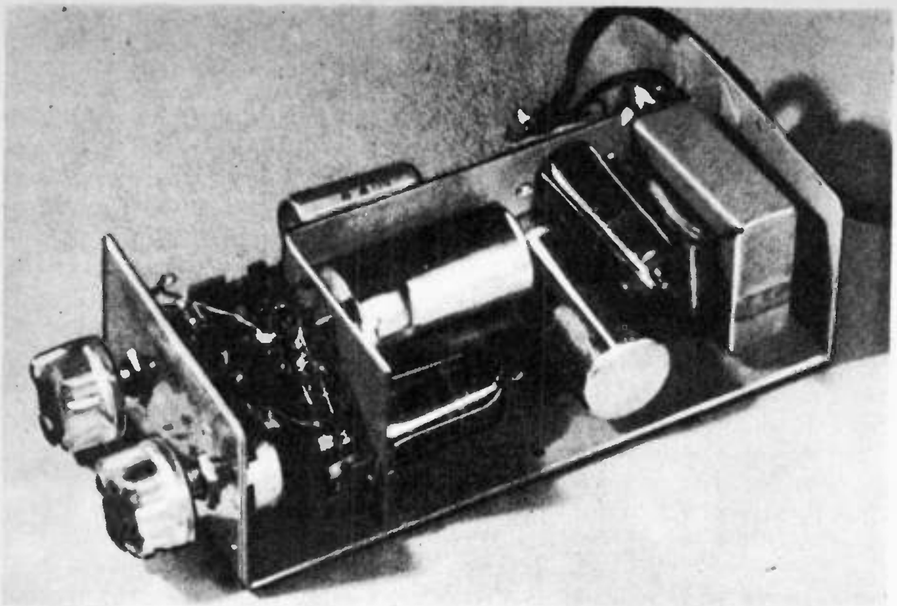


Fig. 5. Internal view of the commercial version. Your home-built unit can duplicate this design but, as suggested in article, the housing should be made larger.

and shielded from the a.c. components. One hundred per-cent shielding must be employed if the preamplifier is to operate satisfactorily in proximity to the tape deck.

A photograph of a *Viking* PB60 preamplifier with shield cover removed is shown in Fig. 5. Individual shield and chassis elements are shown in Fig. 7. A conventional stock-chassis and panel layout could be used as well in duplicating the circuit; however, the metal work shown here is not extensive and the design should be given consideration by the home-builder. The design was dictated primarily by the requirement for a minimum size package, total shielding, and, of course, adaptability to economical production. The home-builder using this same design would do well to "let-out" the dimensions somewhat, increasing the height, width, and length of the preamp by as much as an inch. The over-all construction and wiring would be somewhat simplified as a result. This would also eliminate the need for selecting components such as the power transformer and filter capacitors on the basis of physical size as well as electrical characteristics. The same general arrangement should be used, however. It will be noticed that in this design the chassis itself provides a part of the shielding, isolating the power supply components from the input and interstage amplifier circuitry.

Operating convenience would dictate the use of an a.c. switch on either the equalization control or the gain control. To do so, however, would necessitate bringing a.c. leads into proximity with the input stage and would increase the hum level by several db. The amplifier should therefore be operated from a switched outlet, such as is usually provided on commercial high-fidelity amplifiers.

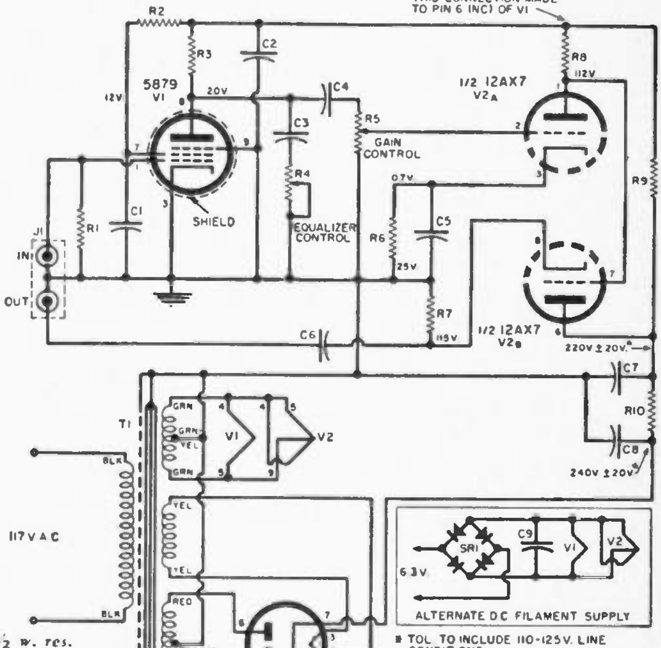
A number of wiring precautions are prerequisite. All filament leads must be tightly twisted and run along the non-critical side of the chassis, away from the tube sockets, resistors, and capacitors. It should be noted that the circulating a.c. currents in the shielding have a pronounced effect if capacitors or signal circuit wiring are so dressed as to lie close to a gap in the shielding. Such a gap occurs, for example, along the sides of the internal chassis where a gap is formed between the chassis and the case.

Only one ground is permissible and this must be at the cathode of 5879 amplifier tube. All other grounds within the amplifier should terminate at this same point. For example, if the outside shield of the input cord is grounded where it enters the case, the resultant ground loop increases the hum voltage 6 to 10 db. Be sure that rubber covered shielded wire is used between the input connector and the grid of the 5879 stage. The input connector can be eliminated entirely, of course, by bringing the rubber-covered, shielded lead directly to the grid, grounding the shield at the cathode of this stage.

A Mumetal shield may be provided for the 5879 as well as the standard metal shield. The precaution is worth 2 to 3 db of added signal-to-noise ratio at the least. Remember that wide dynamic range requires high signal-to-noise ratio. The time and effort spent in these little niceties of construction will be well compensated for by improved performance.

The required 6 db-per-octave compensation is affected by the plate im-

Fig. 6. Schematic diagram of preamplifier. Like all preamps, hum is always a problem. In this design an "on-off" line switch has been purposely omitted to reduce possibility of hum pickup. If such a switch is required, it must be mounted externally. Further reduction of hum is possible by using d.c. on the heaters of V_1 and V_2. The wiring details for such a supply are shown in the inset. In view of the limited space, it would be necessary to mount the required components for this supply on outside of the cabinet.



* TOL. TO INCLUDE 110-125V. LINE CONDITIONS

NOTE: CRITICAL TEST VOLTAGES ARE SHOWN AT PERTINENT TEST POINTS

R_1, R_2—2.2 megohm, 1/2 w. res.
R_3—330,000 ohm, 1/2 w. res.
R_4—25,000 ohm linear taper pot
R_5—500,000 ohm "C" taper pot
R_6—1200 ohm, 1 w. res.
R_7, R_8—100,000 ohm, 1/2 w. res.
R_9—22,000 ohm, 1/2 w. res.
R_{10}—6800 ohm, 2 w. res.
C_1—50 μfd., 50 v. elec. capacitor
C_2-C_7—20/20 μfd., 250 v. elec. capacitor
C_3, C_4—.01 μfd. paper or ceramic capacitor
C_5—25 μfd., 25 v. elec. capacitor
C_6—.047 μfd. paper capacitor
C_8—30 μfd., 350 v. elec. capacitor
C_9—1000 μfd., 15 v. elec. capacitor*
J_1—Double phono jack
SR_1—500 ma. full-wave rectifier (Sarkes Tarzian No. 304 may be used)*
T_1—Trans. 180-0-180 v. @ 8 ma.; 6.3 v. @ .5 amp.; 6.3 v. @ .6 amp. (Available from Viking of Minneapolis, 3520 E 43rd. St., Minneapolis 6, Minn. for $3.75 each. No commercial equivalent available)
V_1—5879 tube
V_2—12AX7 tube
V_3—6X4 tube
Note: Viking will supply escutcheon plates like that shown on the commercial model for a nominal fee. See address under T_1.
** Required only if builder wishes to put d.c. on the heaters in order to obtain greater signal-to-noise ratio. See text.*

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pedance, the 0.01 μfd. capacitor C_3 and the 25,000 ohm potentiometer R_1. The use of a potentiometer permits variation of the actual compensation characteristic within limits of approximately 8 db above and below the NARTB response curve at 10 kc.

The blocking capacitor C_6 in the output of the cathode-follower section of the 12AX7 can be a critical component under certain conditions. The cathode follower has an effective output impedance of 400 ohms, but with the 0.047 μfd. capacitor shown, is intended for operation into a relatively higher impedance, such as the input to the typical high-fidelity amplifier. Under this condition, it provides all the advantages of a low-impedance line, yet the instantaneous voltage appearing across this line appears merely as a voltage to a high-impedance input. Thus, there is no problem of matching impedances. If, however, this 400-ohm cathode follower is to be fed to a low-impedance input, such as a coupling transformer in a broadcast console, it is necessary to short out C_6 entirely, or replace it with a capacitor having a value of at least 25 μfd. An electrolytic capacitor of such value installed at this point would be inserted with the positive side of the capacitor at the cathode level.

Feeding into a high-impedance amplifier input, C_6 can be shorted out or omitted entirely if a blocking capacitor is provided in the amplifier input circuitry. If such a capacitor were not provided, however, the d.c. voltage appearing across the cathode-follower stage would be applied to the grid of the amplifier.

The power supply is of typical design and requires no specific consideration. The transformer chosen, however, should provide very nearly the exact voltages indicated.

Performance Check

A listening test using a recorded music tape should suffice to show whether the preamplifier is functioning properly. Connect the output of the preamplifier to an unequalized (flat response) input on the power amplifier. Preferably, use a shielded lead no longer than 36" to connect the input of the preamplifier to the tape playback head. If the preamplifier is operating properly, the gain control should permit adjustment of the gain from zero to full amplifier output playing standard recorded music tapes. The compensation control should permit adjustment above and below the desired high-frequency response.

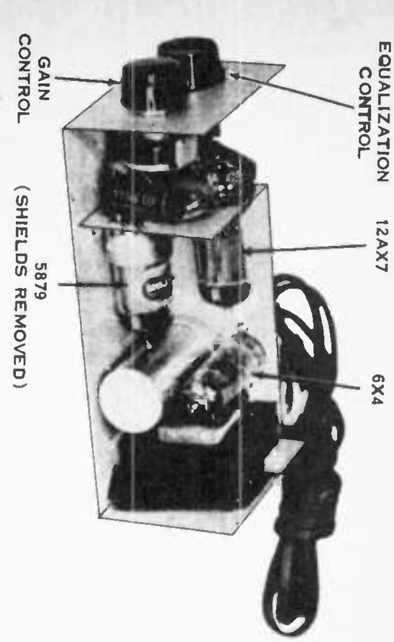
If a signal generator and vacuum-tube voltmeter are available, the following test procedure will result in conclusive performance data. Using a 100-cycle signal (the region of maximum preamplifier gain), adjust the signal input level to 0.002 volt. The preamplifier output should read approximately 2 volts at a maximum setting of the gain control.

Insert a shorting plug in the preamplifier input connector and again measure the output. This reading, representative of the residual hum, should not be greater than .025 volt.

To determine the NARTB equalization setting for the compensation control, apply a .002-volt, 100-cycle signal at the input of the preamplifier and adjust the gain control to provide an output of 15 db. Apply a 10 kc. signal of the same value and adjust the equalization control to provide a reading 15 db below zero reference. Plot a curve, if desired, using frequencies of 100, 500, 1000, 5000, 10,000, and 20,000 cycles-per-second. The curve obtained should correspond closely to that indicated in Fig. 3.

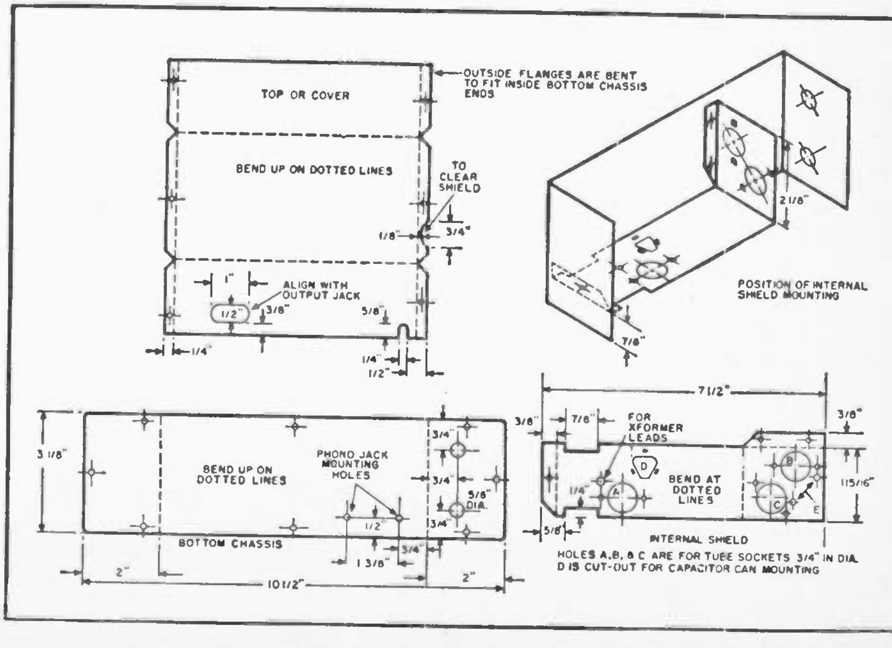
One of the most persistent questions asked by new enthusiasts is, "If I add a bias oscillator can my playback preamplifier be used for recording as well as playback?"

This is not practical because of the almost diametrically opposite equalization characteristics required. The switching circuitry necessary to make a dual purpose record-playback amplifier is so extensive that the construction of a complete, integrated, record-playback and high-frequency bias unit is infinitely more practical. Those who intend ultimately to build such a unit will still find the preamplifier described here to be a useful adjunct to their tape systems, however, since it may then be used with a third head in an erase-record-monitor head configuration for simultaneous monitoring directly from the recorded track.



Basic playback preamplifier with shield removed and various parts identified.

Fig. 7. Mechanical details of metal work. There are three parts involved; shield, chassis, and a cover. Since these three parts must fit together many of the dimensions on the diagram have been purposely omitted. Variations in sizes of components would also change specific locations of mounting holes. Make sure that all parts will fit before proceeding too far. The method of mounting the transformer is for the Viking unit. The bracket flanges are simply bent around the shield. The rectifier tube socket is so mounted that the base of the tube itself sticks through the shield to the opposite side. Use ¼" metal spacers to mount socket away from the shield. Both phono jacks are mounted on a single Bakelite base. Metal brackets are then used to mount to the chassis. To avoid possible hum pickup the filament leads between the two amplifier tubes are run through the two holes (E) to the opposite side of the shield. These two holes should be placed as close to the heater pins of their respective tubes as possible. As mentioned in the article, the mechanical details shown below are for the commercial unit but can be duplicated at home. Increasing over-all size would make construction easier.



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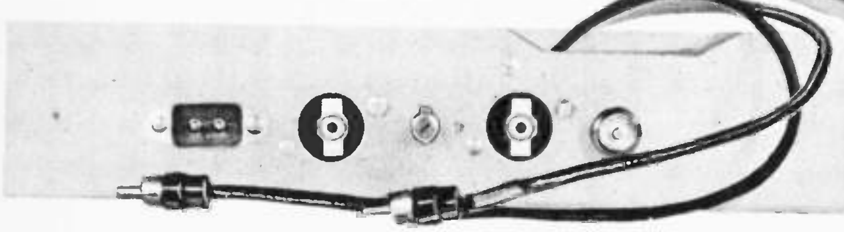
The Record-Playback Preamp



Front and rear views of commercial version of circuit described. It can easily be duplicated by the home constructor.

Part 3. Complete construction details on a record-playback preamp including a bias-erase oscillator which can be used with any tape deck that employs "Dynamu" type heads.

IT IS SAFE to assume that sooner or later, anyone who has a playback only system will wish to convert it to one which permits recording as well as playback. Such a system will utilize the same deck, but now equipped with an erase head as well as the original record-playback head. It also requires a record-playback preamp, consisting of an amplifier channel which may be switched to provide either a recording or playback function, and an erase-bias oscillator. The specific head requirements and the construction of a suitable record-playback preamplifier are covered in this article.

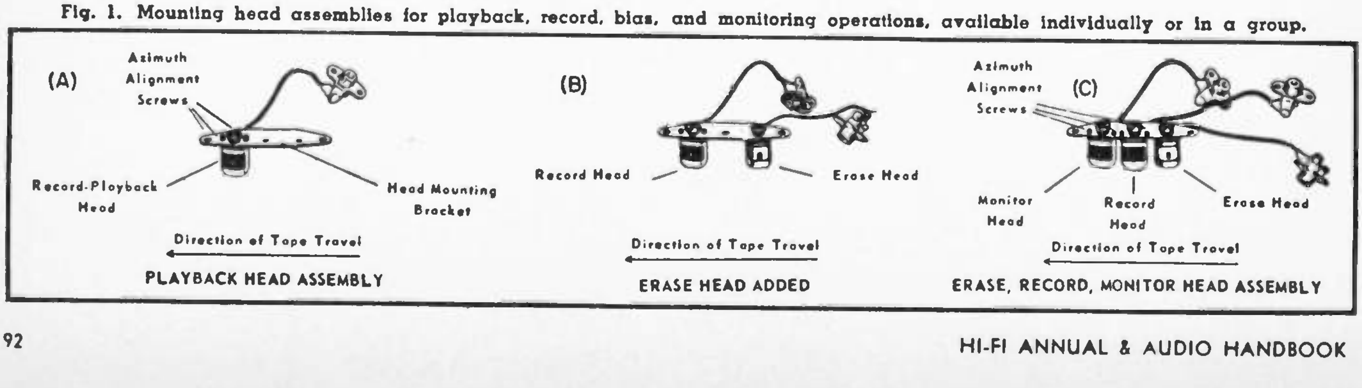
The *Viking* FF75 tape deck, like any other presently available tape decks for this purpose, may be obtained as a complete, pretested unit equipped with heads of a desired configuration. The FF75R and FF75RM models are typical. However, one of the purposes of this series was to begin with the basic single-head deck and explain the evolution of the various modes of operation, and the components required. The system, as described, can be used with other tape decks providing the head equalization is properly adjusted and that it is compatible as far as adding the extra heads is concerned.

The simple, single-head assembly provided on the FF75 playback deck is shown in Fig. 1A. It consists of a head mounting bracket and a single head. This head, incidentally, serves with equal effectiveness for both recording and playback. Fig. 1B shows the same head and bracket assembly, but with an erase head added. The FF75 deck so equipped becomes an FF75R. The addition of the erase head is the only modification necessary to convert the basic playback deck for recording. The three-head assembly, shown in Fig. 1C, differs only in that the bracket permits addition of a second record-playback head for simultaneous monitoring from the recorded track on the tape. If a monitoring head is to be added, a playback preamplifier will be required.

The purpose of the erase head is merely to remove any previously-recorded track from the tape before the tape passes over the record-playback head. In some recorders the erase function is contributed by an ordinary *Alnico* bar magnet over which the tape passes. This device effectively removes the previously recorded track from the tape, but the uniform polarization of the tape results in noticeable hiss upon playback. The more usual practice, therefore, is to use a head quite similar to a record-playback head, but with a considerably longer gap, and to drive this head from an ultrasonic alternating current generator, *i.e.*, an erase oscillator.

The erase head used with this deck has heavier pole pieces than the record-playback head, and a gap length of 0.007" as compared to the 0.00015" gap length of the *Dynamu* record-playback head. Also, the pole pieces are wider, covering a track width of 0.125" as compared to the 0.090" recording track. Thus it can be seen that the erase head can be expected to fully cover the previously recorded track, erasing the tape before it reaches the record head. To accomplish this, the erase head must be energized with a driving current of approximately 13 ma. from the ultrasonic erase oscillator.

Fig. 1. Mounting head assemblies for playback, record, bias, and monitoring operations, available individually or in a group.



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The Erase-Bias Oscillator

One of the first considerations in the design of the erase-bias oscillator is that of selecting a bias frequency. An immediate prerequisite is that the bias frequency must be at least four to five times that of the highest frequency which is to be recorded. This is due to the fact that the erase oscillator has one function other than supplying current to the erase head. During recording, a small portion of this same high frequency current is added to the signal applied to the record head. This is called a *bias current,* hence the name, erase-bias oscillator. The function of this current, added to the signal current, is to promote linearity of the flux pattern on the tape. It establishes the need, however, for a four or five to one bias-to-signal frequency ratio in order to avoid interaction of harmonics of the highest recorded frequencies with this bias frequency.

The immediate inference is that a frequency of 100 kc. or more should be selected, to permit recording of frequencies to 15,000 cycles, or more. This is entirely practical in commercial duplicating equipment or for exacting professional applications. At the home recordist's level, however, an erase bias frequency of this order is impractical for two reasons. First, a 100 kc. frequency takes on many of the characteristics of radio frequency. The shortest of leads between the oscillator and the heads are essential. Second, presently available erase heads are not easily driven at this frequency. Excepting such luxuries as double-gap erase heads, complete erasure becomes increasingly difficult as the erase frequency is increased.

Therefore, it can be stated that unless bulk-erasure is employed to bring tapes down to their virgin noise level prior to recording, a bias frequency of 60 kc. to 70 kc. is as high as is practical. This bias frequency permits recording audio frequencies as high as 12,000 cps without exceeding 2% distortion.

Equalization

Fig. 2, curve A, shows the response curve required for recording. This, as is the response characteristic for playback, curve *B*, is that required for over-all equalization to the commonly accepted NARTB standard.

It is immediately apparent that the two equalization characteristics (playback and record) are practically diametrically opposite. Thus, the switching circuitry employed must select equalizing circuit components as well as switch the input and output circuits for record and playback.

As stated in the previous article, the NARTB equalization characteristic requires pre-emphasis of the high frequencies during the recording process. This is necessary to compensate both for the recording characteristics of the tape itself and the losses which occur in playback.

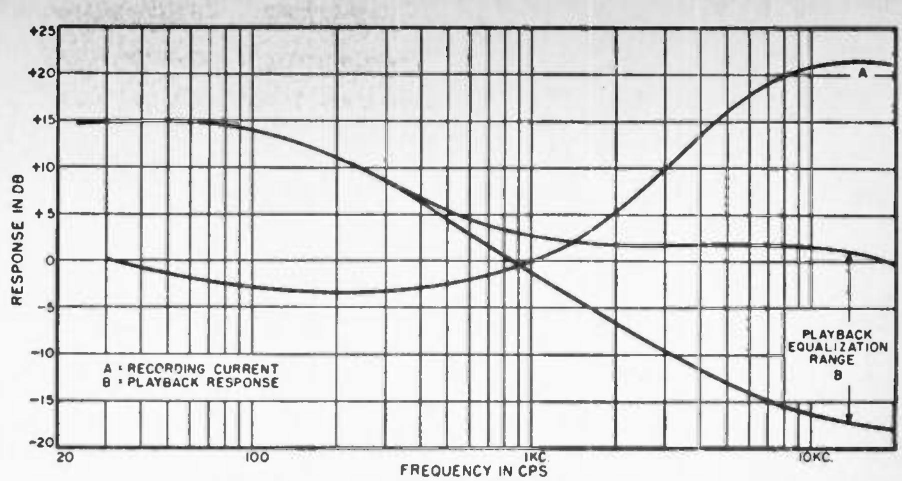


Fig. 2. Approximate record and playback equalization characteristics of unit.

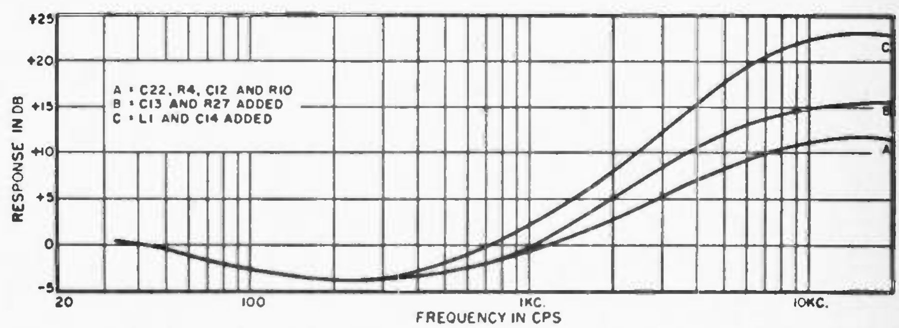
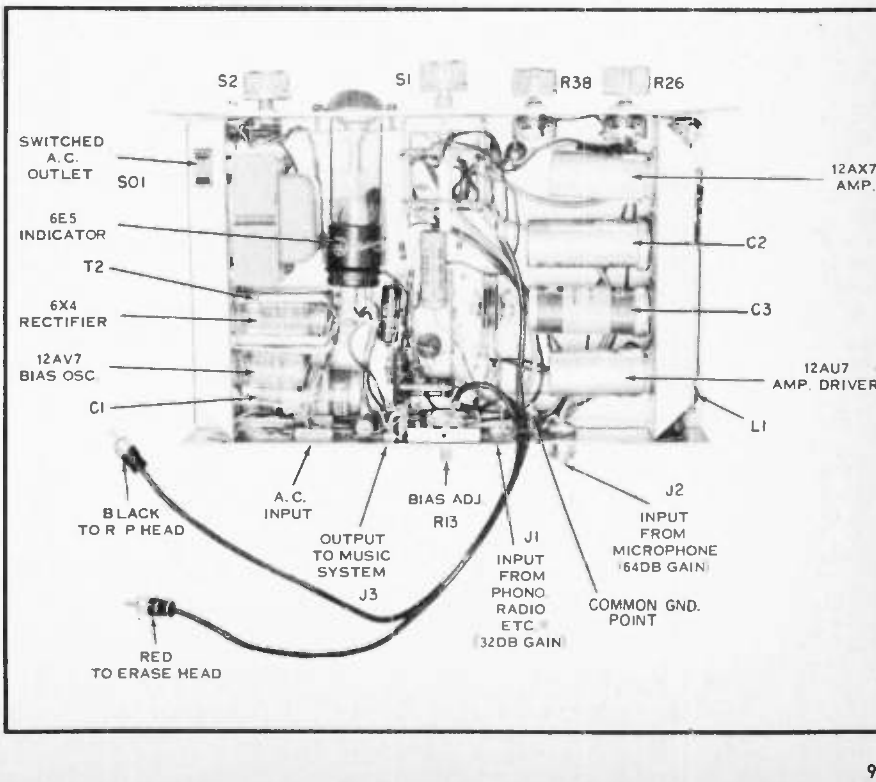
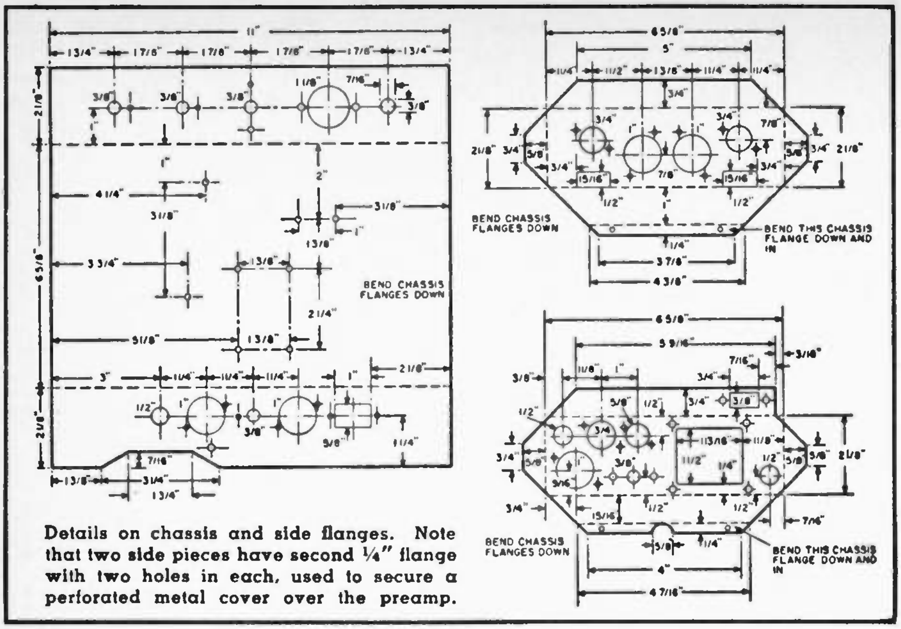


Fig. 3. Effect of equalizing networks, record operation. See discussion in text.

Fig. 4. Top chassis view of preamp with cover removed. All of the major components are identified. Of particular interest is the position of the side flanges. Both flanges are mounted 1½" in from the edge of the chassis. These flanges are welded in the commercial version. For home construction, they can be soldered, riveted, or bolted. The cover, which is not described in detail, completely encloses the top and both sides of the chassis. It is perforated for ventilation. It is shaped to fit the contour of the rear flange as shown in the over-all view on page 92 and in the mechanical diagram of the metal components. The only reason for this indentation is to permit the assembly of this unit as close to the tape deck as possible, without interfering with the fan in the tape deck. This indentation is not necessary if compactness is not required. The a.c. input, shown as a chassis receptacle, could be omitted if the a.c. power-line cord is connected directly into the preamp's power circuit.





Details on chassis and side flanges. Note that two side pieces have second ¼" flange with two holes in each, used to secure a perforated metal cover over the preamp.

An equalization factor of 6 db-per-octave is possible using an ordinary single-section *RC* network. It will be recalled that one such network sufficed in the design of the preamplifier described last month. A similar network, R_{38} and C_6, (see Fig. 5), serve for playback equalization in the preamp described here. The compensation required during recording, however, is too great to be attained with a single network and is too complex to be so easily achieved.

Actually three separate networks are involved. The additive effect of these networks is shown in Fig. 3. These curves, incidentally, are provided merely to illustrate the method of equalization. They are approximate, inasmuch as they are not based on measured data.

The first network consists of C_{22}, R_1, C_{12}, and R_{10}, and peaks at approximately 350 cps, resulting in an approximation of the curve shown in Fig. 3A. This is, in effect, a frequency discriminating voltage divider network, decreasing the signal applied to the grid of the third amplifier stage at frequencies in the region of approximately 350 cycles.

C_{13} and R_{27} at the cathode of the second triode section peak at approximately 500 cycles and act to change the response curve to that shown in Fig. 3B.

Finally, the network consisting of L_1 and C_{11}, at the cathode of the third triode section resonates at a frequency of 16,000 cycles. This results in a sharp drop in the impedance in the cathode of that stage with an equivalent increase in gain at frequencies in the 16,000 cycle region. This produces the ultimate curve shown in Fig. 3C.

No novelty is involved in these equalizing networks, they are proven circuits commonly employed in one form or another to obtain the required effect. The individual contributions of the circuit components are outlined here merely to satisfy the typical home experimenter who would rather know how it works than merely know that it works. The over-all compensation characteristic, it will be noted, serves not only to provide the prerequisite gain at the high-frequency end of the audio band, but also provides a rising characteristic below 100 cycles which compensates for the wrap-around loss which occurs at low frequencies, where the wavelength on the recorded track exceeds the width of the head.

Circuit Description

The complexity of the record-playback preamplifier described here should not be minimized. The switching circuit, for example, requires careful wiring. Because of the extensive decoupling networks and the equalization circuits, this preamp is more difficult to wire than an ordinary audio amplifier. Properly wired, however, and properly checked out as to bias adjustments, etc., this preamp is capable of professional quality recording as well as playback.

The erase and bias-current adjusting networks are of the proper value for use with the *Dynamu* heads. If the same preamplifier is to be used with heads other than the *Dynamu* units, circuit modifications would be essential both to provide the correct erase and bias currents and the proper equalization characteristics as well.

The preamplifier is provided with its own self-contained power supply. The bias-oscillator consists of a 12AV7 tube in a push-pull oscillator circuit. The related circuit components are those calculated to provide operation in the desirable frequency range between 60 and 70 kc. In the event a higher frequency is desired, it can be obtained by decreasing the value of capacitors C_{17} and C_{18}. If other values are substituted, however, they must be substituted in pairs, maintaining a balanced condition. Linearity and good waveform are essential if distortion is to be kept to a minimum. The oscillator output is adequate to provide the required 13 ma. current to the erase head. The voltage appearing here is then attenuated by the adjustable bias control R_{18} and series resistor R_9 to provide a bias current of 0.8 ma. to the record-playback head when recording (record-playback switch in *record* position). With this switch in the *play* position, plate voltage is removed from the oscillator.

The possibility of employing bulk-erasure in certain modes of operation will be of interest, particularly with respect to stereophonic recording which will be discussed in the next part. It should be noted that the erase head, although a part of the deck itself, forms the inductive load for the erase-bias oscillator. In the event that the preamp is to be used with a deck which is not equipped with an erase head, a 30 mhy. choke (or a spare erase head) must be substituted as a loading coil. This may necessitate some juggling of the resistance value of R_9 to provide the desired bias current of 0.8 ma. to the record head. This bias current value, incidentally, is one of the critical prerequisites to full-fidelity recording.

Two input connectors are shown. The first of these must be a shorting-type microphone connector and is used primarily for recording from a microphone. A total gain of 62 db is provided after equalization; sufficient to justify use of a professional type microphone. The high-level input provides a gain of 32 db and serves for recording from flat-response, radio, phono, or similar sources. Equalization is fixed in either case. The gain, or recording level, is adjusted by means of R_{28}.

The playback preamplifier output is derived from the third amplifier stage, a portion of which supplies a voltage through C_{10}. This voltage is rectified by a germanium diode CK705, and applied to the eye indicator tube which serves as the recording level indicator. The same recording signal voltage is fed also to the output jack where it is available for "earphone" monitoring of mike pickups, or may be used to drive the music system in the conventional manner when recording from radio, phono, etc.

With the record-playback switch in the play position, the shielded cable from the record head is switched to the input of the preamp. At the same time, the recording compensation networks are removed from the active circuits and the single network consisting of R_{38} and C_6 is substituted. This provides the 6 db-per-octave attenuation shown in Fig. 2B.

Resistor R_3 and capacitor C_{21} at the input to the first amplifier stage are wired directly at the grid and cathode tube socket terminals and provide an effective filter for the 60-cycle buzz which can result from the sync pulses transmitted by television stations. This particular type of interference is easily differentiated from power supply

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ripple or hum because of its raspy sound. It succumbs easily, however, to the filtering effect of this simple network.

Mechanical Construction

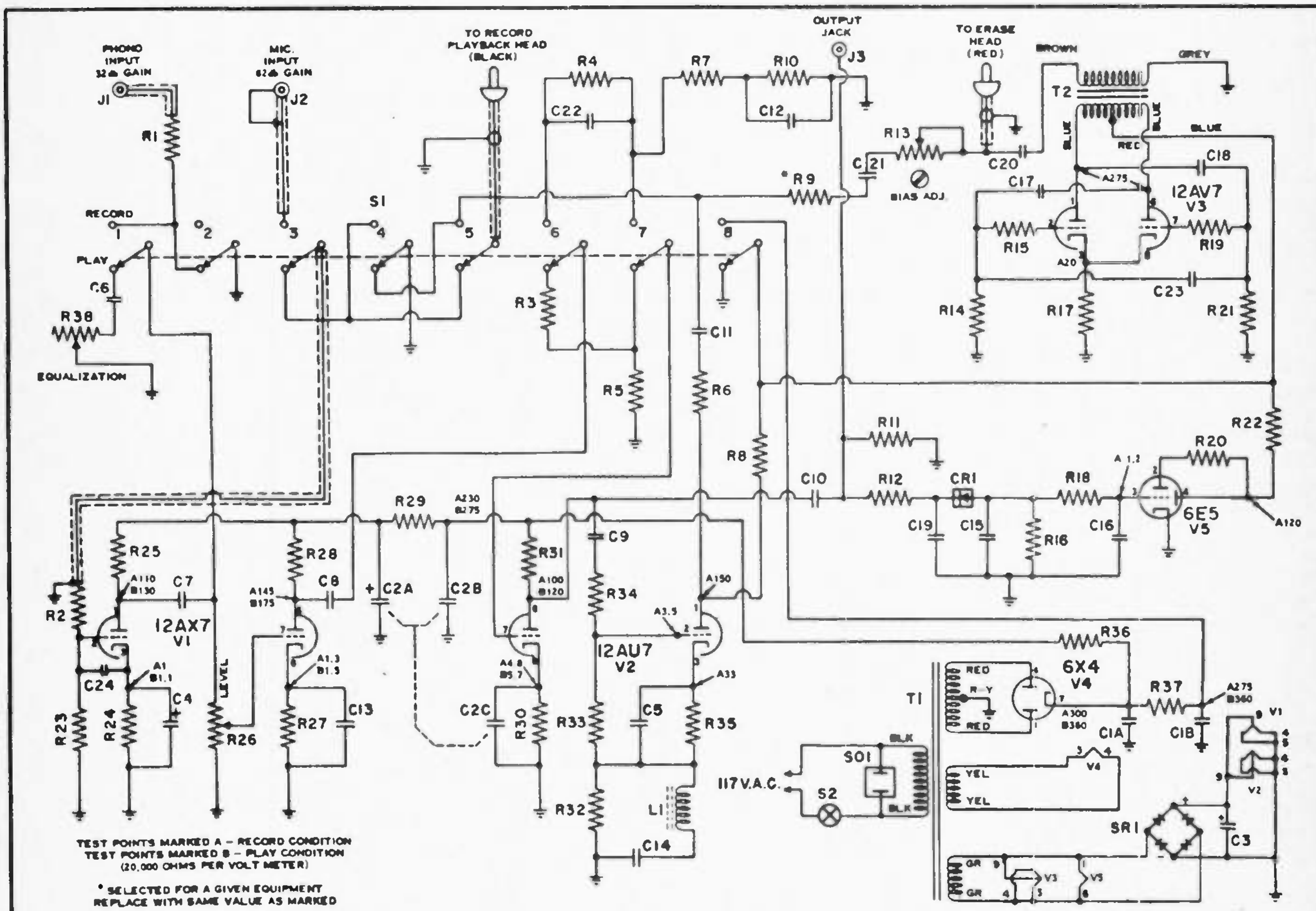
Box-type construction was dictated by the requirement for minimum size and the over-all requirement for a shape which would lend itself to cabinet installation along with the tape deck. A view of the unit with cover removed is shown in Fig. 4. The power supply and the not-so-critical erase-bias oscillator are contained in one end, the critical amplifier stages are at the other end, as far from the power supply as possible. The middle section of the box is utilized for the less critical wiring associated with the 6E5 indicator tube, the a.c. switch, gain and equalization controls, and the record-playback switching circuitry.

This switch consists of two decks with four switch sections per deck. The switch is special to the extent that it is ordinarily available only with ½-inch spacing between decks. New screws, shaft, and spacers must be added to provide 1-inch spacing, this, in order to eliminate the possibility of spurious oscillation.

All low-level signal leads must be run with *rubber covered* shielded wire. Be certain that the shielding is not inadvertently grounded to the chassis at any point, since this would nullify the effectiveness of the single ground point. This ground is completed at a silver-plated screw midway between the two input connectors. The silver-plated screw is not an essential. It is mentioned only to point up the necessity for thorough and careful handling of the ground circuits. Cutouts at the corners of the end chassis permit wires to be run from the center section to the end chassis.

The erase head and record-playback head leads are rubber covered shielded wires, terminated with plugs which fit

Fig. 5. Complete schematic diagram of the record-playback preamplifier and the companion bias oscillator unit.



R_1, R_7, R_{10}, R_{12}, R_{28}, R_{31}—100,000 ohm, ½ w. res.
R_2—1000 ohm, ½ w. res.
R_3, R_5, R_{11}, R_{14}, R_{21}, R_{22}, R_{25}—220,000 ohm, ½ w. res.
R_4—330,000 ohm, ½ w. res.
R_6, R_9—56,000 ohm, ½ w. res.
R_8—47,000 ohm, 1 w. res.
R_{13}—50,000 ohm log taper pot (Bias Adj. Control)
R_{15}, R_{19}—22,000 ohm, ½ w. res.
R_{16}—4.7 megohm, ½ w. res.
R_{17}, R_{37}—2200 ohm, 1 w. res.
R_{18}, R_{20}, R_{23}—1 megohm, ½ w. res.
R_{24}—2700 ohm, ½ w. res.
R_{26}—250,000 ohm audio taper pot
R_{27}, R_{30}—3300 ohm, ½ w. res.
R_{29}—68,000 ohm, 1 w. res.
R_{32}—10,000 ohm, ½ w. res.
R_{33}, R_{34}—470,000 ohm, ½ w. res.
R_{35}—1800 ohm, ½ w. res.
R_{36}—33,000 ohm, 1 w. res.
R_{38}—5000 ohm linear taper pot
All fixed resistors to be ± 10%
C_{1A}-C_{1B}—15/15 µfd., 475 v. elec. capacitor
C_{2A}-C_{2B}-C_{2C}—30/30/20 µfd., 250/250/25 v. elec. capacitor
C_3—1000 µfd., 15 v. elec. capacitor
C_4, C_{13}—25 µfd., 25 v. elec. capacitor
C_6, C_7, C_8, C_9, C_{10}—.05 µfd., 400 v. capacitor
C_{11}—.25 µfd., 400 v. capacitor
C_{12}—.1 µfd., 200 v. capacitor
C_{13}, C_{14}, C_{15}, C_{16}—.01 µfd., 200 v. capacitor
C_{17}, C_{18}—500 µµfd., 400 v. ceramic capacitor ±10%
C_{19}—500 µµfd., 400 v. disc ceramic capacitor
C_{20}—750 µµfd., 400 v. disc ceramic capacitor
C_{21}—250 µµfd., 400 v. disc ceramic capacitor
C_{22}—100 µµfd., 400 v. disc ceramic capacitor
C_{23}—1000 µµfd., 400 v. disc ceramic capacitor
C_{24}—50 µµfd., 400 v. disc ceramic capacitor
T_1—Power trans. 250-0-250 v. @ 15 ma.; 6.3 v. @ .6 amp.; 6.3 v. @ 1.2 amp.*
T_2—Bias oscillator coil (Viking Part No. D501)*
CR_1—1N66/CK705 germanium diode (Raytheon)
SR_1—Selenium rectifier bridge, 300 ma., 25 volts r.m.s. max. input, bridge center hole mounting
L_1—10 mhy. iron-core choke*
S_1—Dual-section selector switch, 4-poles per sec., non-shorting type, 1" spacing between sections
S_2—S.p.s.t. rotary switch
SO_1—Female a.c. receptacle, chassis-mounting type
J_1, J_3—Phone tip jack
J_2—Mic. jack, single-contact, insulated type
V_1—12AX7 tube
V_2—12AU7 tube
V_3—12AV7 tube
V_4—6X4 tube
V_5—6E5 tube
** Note: All parts thus indicated are special and may be obtained, along with an escutcheon of the type shown in the photograph, from Viking of Minneapolis, 3520 E. 43rd Street, Minneapolis 6, Minn.*
Miscellaneous parts:
1—Escutcheon
1—Male a.c. receptacle, round pins, chassis-mounting type
1—117-volt line cord with female receptacle molded on

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the phono jacks used for head connections at the back of the deck. These, incidentally, present a troublesome soldering job for the home builder. The issue is circumvented by buying a 36-inch patch cord, complete with molded-on plugs and cutting it near-center to provide two leads of proper length. With 70 kc. bias this length should be not more than 18 inches. The shielded leads are anchored at the back of the preamp chassis with a flat cable retaining clip.

Keep resistor and capacitor leads short. Use sleeving wherever danger of shorting exists. Don't use tie-points with ground lugs. Instead run grounds to an insulated tie-point and thence to the common ground. During construction, note the location of signal and oscillator leads. You may have to redress them slightly during the final checkout.

Testing of the unit after wiring should logically begin with a check of voltages appearing at the test points indicated on the schematic diagram. Note that readings marked "A" are obtained in the *record* condition and readings marked "B" are taken in the *playback* condition.

Using an oscilloscope, check to see that the erase-bias oscillator is operating. The frequency, using the D501 transformer, should be between 60 kc. and 70 kc.

The 6E5 indicator tube should be operative, but fully open when the switch is in the record position and the gain control at maximum. If the eye is partially or fully closed, oscillation in the amplifier stages is indicated.

Plug in the erase and record head shielded leads at the proper jacks on the deck and clip a ground lead from the preamp chassis to the deck. The erase head jack should be grounded to the deck. The record head jack should not be grounded. During this adjustment procedure the cover should be removed from the deck permitting access to the head connections.

To determine the erase head current, insert a 100-ohm resistor in series with the head at the deck. With the preamp in the *record* condition, an a.c. voltage reading of 1.2 to 1.4 volts should be obtained across this resistor. Too low a reading indicates that the bias frequency is too high. Change capacitors C_{17} and C_{18}, if necessary.

Place the same 100-ohm resistor in series with the record-playback head. Adjust R_{13} to provide a reading of 0.8 volt. Change the value of series resistor R_9 if necessary to obtain this exact value.

Now, place the *record-playback* switch in the *play* position and put on a music tape or other recorded tape. Adequate listening volume should be provided, with high impedance phones plugged in at the output jack or with this jack connected to the music system.

No azimuth adjustment of the record-playback head is required unless it has been loosened at any time, or removed from the bracket. In that case, loosen the head retaining nut slightly and alternately adjust the two azimuth alignment screws until maximum output is obtained. Actually, azimuth alignment should be done using a very-high-frequency continuous-tone tape and an oscilloscope.

The erase head is not critical as to alignment. If it has just been installed, align it by plugging in the record-playback lead at the erase head jack, use a continuous tone or music tape and merely rotate the head for maximum volume. Since the erase head has a very wide gap, very little high-frequency response will be obtained.

The equipment is then ready for use as a recorder. With phono, radio, or mike inputs the eye indicator should close entirely as the gain is advanced. The best fidelity and dynamic range is provided at a setting that causes the eye to close not more than half way on peaks. Distortion will occur if the eye closes entirely.

If noticeable hum is encountered, check all ground leads carefully. Check to be certain that the record head jack is not grounded at the deck, and that the erase head is grounded. With a short clip lead, momentarily ground the cable shields and ground tie-points throughout the preamp. For minimum hum, the level control should be set fully clockwise during playback.

One more precaution, be sure to place the *record-playback* switch in the *play* position after finishing a recording. An erase head has neither conscience nor judgment about what it erases.

As pointed out in the article, the unit described is a commercial product. For those who would rather purchase an assembled instrument, it is available from *Viking*. It is known as the RP61 and is priced at $77.50. For those who need an erase head for conversion to the two-head unit (Fig. 1B) it is available for $7.65. The three-head unit (Fig. 1C) is priced at $27.00 complete.

Stereophonic Operation

Part 4. Concluding article covers conversion of basic deck to permit recording and reproduction of stereophonic tape.

BASED on questions asked at various Audio Fairs and stereo sound demonstrations, any discussion of stereophonic sound should begin with a clarification of the terms "stereophonic sound" and "binaural sound," which are often used interchangeably. Actually, they represent slightly different recording techniques. Through common usage, however, one generic term "stereophonic sound," is rapidly becoming applicable to either.

Specifically, binaural recording applies to recordings wherein two microphones are spaced a distance equivalent to that separating the human ears, and separated by a disc or sphere roughly equivalent to the size of the human head. In many cases, coupling equivalent to the auditory coupling between the human ears is actually introduced between the microphones in an effort to obtain the truest binaural effect. A binaural system in this strict sense of the word is shown in Fig. 1A. As will be seen, the output of the left-hand and right-hand microphones is recorded on separate tracks, is played back in turn through separate amplifiers, and is listened to using independent earphones powered by the left-hand and right-hand amplifiers respectively. Thus, it is apparent that the left and right ears, respectively, will hear the program material with the exact time-relationship and balance with which it arrived at the microphones.

A binaural tape recorded in this manner can be played back over a dual-

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channel amplifier system, but using speakers spaced at some given distance instead of the earphones shown, and much of the same listening quality will be preserved. In other words, binaurally recorded tapes are adaptable for use with stereophonic loudspeaker systems.

A complete stereophonic system is diagrammed in Fig. 1B. In stereophonic recording the microphones are spaced fairly far apart, the exact distance depending upon the size of the musical group being recorded and, to some extent, the hall accustics and other factors. The audio channels are kept entirely separate as in binaural recording, and spaced speakers are used for playback. Here again, earphones can be used for playback without adversely affecting the program reproduction, although there is no valid reason for doing so since the recording process did not involve this discrete spacing.

Whether a given recording is binaural or stereophonic in origin makes little difference in the end result. Music so recorded and played back through separate high-quality music systems, takes on an auditory depth or spatial effect which lends amazing presence and realism. This is particularly true with respect to orchestral groups or symphonies having widely dispersed sound sources. It is evident even in the reproduction of single source instruments such as piano and voice, because of the reflections which, to an extent, fortify the sound as we normally hear it in an auditorium or concert hall.

Both stereophonic and binaural sound are terms which, loosely used, have become somewhat interchangeable. For the sake of brevity, both recording modes will be grouped together and considered as "stereophonic" throughout this article.

Tape is ideally suited to the recording and reproduction of stereophonic material. Present recording practice calls for the recording of two 90 mil tracks along either side of a standard ¼" tape. In monaural recording, one of these tracks is recorded in either direction. In stereophonic recording these two tracks are recorded simultaneously.

Either binaural or stereophonically recorded tracks can be mixed by paralleling the two stereophonic pickup heads or the output of the preamplifiers, applying the mixed signals to a single amplifier and speaker. Played through a single music system, such a tape effectively becomes a monaural music source, in every way equivalent to a standard monaural recording.

Conversely, a monaural tape can be played through a stereophonic system using both amplifier channels, providing what can be considered as "pseudo-stereophonic" performance, but it will not provide any of the directional sense or depth which characterizes stereophonically recorded tape. The gain, such as it is, is merely the elimination of the single-point sound source.

The "Dactron Steradapter" kit, an in-line stereo adapter for tape recorders, developed by the International Magnetic Electronics Co. of Minneapolis, Minn.

"Staggered" and "Stacked" Modes

Stereophonic tape recordings are commonly available in either of two types of recording. Fig. 2B shows the head configuration used for so-called staggered recording. Here ordinary single record or playback heads are spaced exactly 1¼" apart, one covering the upper half of a standard ¼" tape, and the other the lower half of the same tape. It follows logically that if the same head arrangement and the exact spacing of heads is employed in playback as was used in recording the material, the two channels will be heard in their original balance and time relationship. With in-line heads, as shown in Fig. 2C, the same condition applies, except that the heads instead of being separated by a finite spacing are located one above the other.

Champions of both the stacked and staggered modes of stereophonic recording and playback are not lacking. It is argued that slight differences in spacing between a given playback head assembly and that used in the original recording process will destroy the balance of the program material. Actually, the very slight difference encountered is roughly equivalent to that of moving three or four seats one way or the other in the concert hall. A perfectly valid objection to the staggered head style of recording is the fact that cut-and-splice editing of staggered tapes is almost impossible.

Stacked or, if you prefer, in-line heads, on the other hand, have one fundamental disadvantage. Because of the unavoidable close coupling between heads, the stacked head provides a very high degree of crosstalk, *i.e.*, a signal voltage appearing at a given instant in either head, sets up a ghost image in the adjacent head. This is no disadvantage in stereophonic recording or playback since the program material in both heads is almost identical. It does, however, set up an immediate obstacle to use of either one of the stacked heads for monaural recording or playback.

The actual crosstalk ratio can be minimized with effective *Mumetal* shielding, but is a considerable factor in any case; usually on the order of 30 to 45 db signal ratio. Thus, in recording, the signal impressed on the desired channel would appear as an unwanted ghost track on the adjacent channel. Similarly, in using one head section for playback from a standard half-track dual recorded tape, the program material from the adjacent channel would appear in the output at a level approximately 30 to 45 db down from the desired material.

Because of this crosstalk characteristic inherent in stacked heads, it is normal practice to provide an additional single-track head for use with monaural tapes. In the *Viking* "universal" stereophonic head assembly this additional head is placed on the lower track and spaced at the standard staggered head spacing of 1¼" as shown in Fig. 2C. Thus it serves not only for the recording and playback of monaural tapes, but for playback of staggered stereophonic tapes as well. Such a head arrangement is considered advantageous in that for the man who already owns a library of staggered recorded tapes, it provides a means for utilizing these along with those of the new tapes which are being released only in the staggered configuration. Similarly, it permits the user to choose program material on the basis of its

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merit alone, without regard to the mode used in recording.

Even though it is probably a safe assumption that the stacked or in-line head will eventually become an industry standard, staggered head assemblies presently enjoy one distinct advantage. They are generally cheaper and easier to acquire. Stacked heads are more expensive to manufacture because of the very precise three-dimensional alignment of the cores which must be accomplished within the head itself prior to encapsulation. In staggered head assemblies this spacing and azimuth alignment of the individual heads is accomplished by adjusting, individually, the position of the heads on the mounting bracket.

In previous articles of this series it was shown how the basic *Viking* FF75 deck could be changed over to an erase-record deck merely by installation of the proper head assembly components. It is possible to purchase stereophonic models of the *Viking* deck, factory assembled and tested, as well as decks with any desired head combinations. However, it is possible to convert the basic deck to any of the other modes of operation merely by adding the required head assemblies. Fig. 2A shows the simplest monaural playback head assembly available from the company. Fig. 2B shows the unit with a staggered mounting bracket and two identical half-track heads, suited for playback of staggered tapes or, using the lower head only, for playback of monaural tapes. The owner of the single-head deck can convert to this head assembly by purchasing an additional 75-1 record-playback head at a cost of $9.00 and a D340 bracket at $1.50.

Note that this head arrangement does not permit space for an erase head to the right of the offset erase-record head. In this case, the tapes must be bulk erased prior to recording.

Fig. 2C shows the "universal" stereophonic head assembly which provides for playback of stacked tapes using the stacked heads, and playback of staggered tapes utilizing the upper of the stacked heads in combination with the single offset head. Crosstalk from the unused head section in this case cannot be detected under normal conditions and can be considered as having negligible effect. The offset head scans the lower track and is therefore available for monaural recording and playback.

The owner of the basic deck can

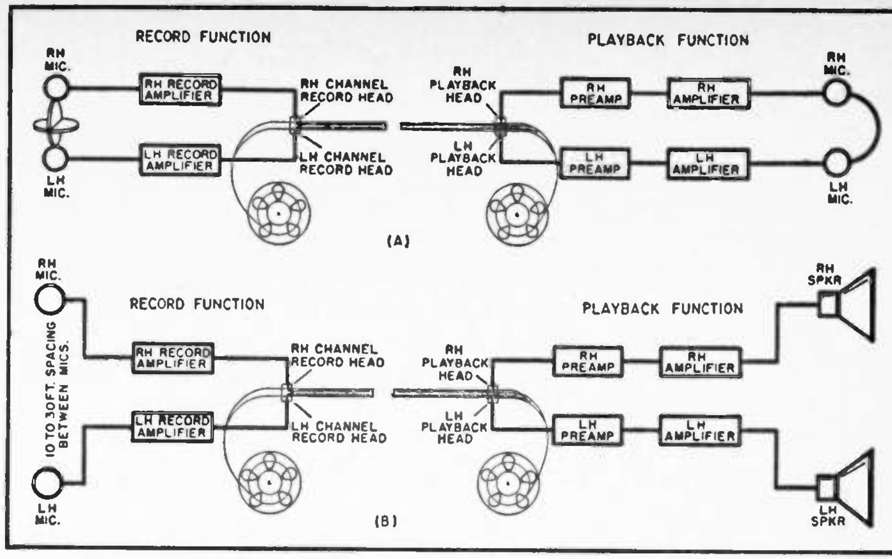


Fig. 1. Binaural (A) and stereophonic (B) systems of recording and playback.

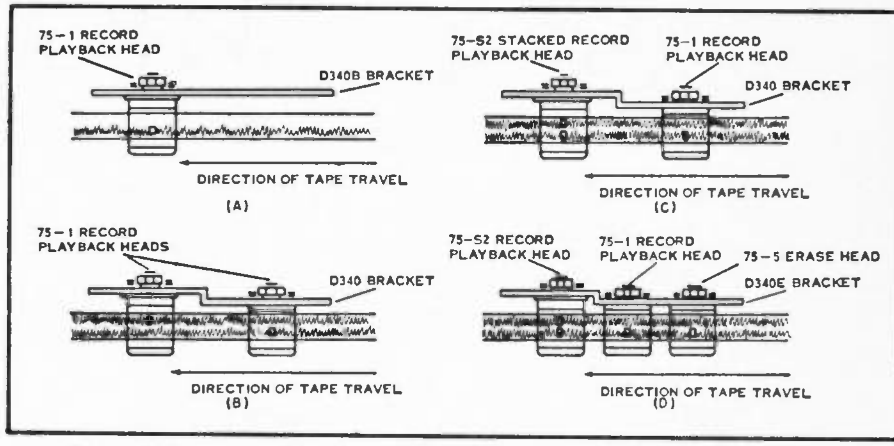
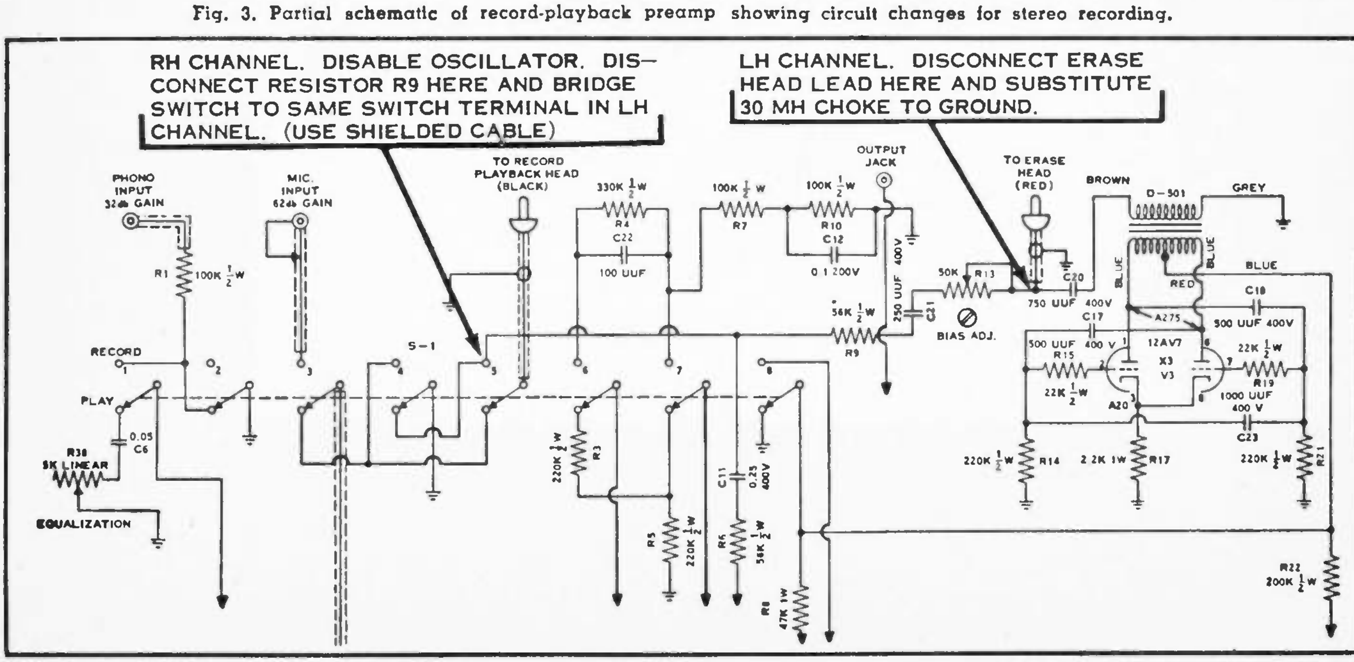


Fig. 2. Basic monaural head assembly and stereophonic derivations. See text.

Fig. 3. Partial schematic of record-playback preamp showing circuit changes for stereo recording.



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convert to this head configuration by purchasing the same D340 bracket and a 75-S2 stacked head assembly at $33.25 The original head is moved to the offset position on the bracket. A D368XA tapelifter and pressure pad assembly should be added at a cost of $4.60.

Fig. 2D shows a head configuration which has grown out of demands for a head arrangement which would permit stereophonic playback (or recording) using stacked heads and which would also permit monaural erase-record operation. To convert the basic deck to this configuration it would be necessary to add a D340E bracket ($2.30), the 75S-2 head assembly ($33.25), and a 75-5 erase head ($7.65 . The original head serves in the center, position. The tapelifter and pressure pad assembly is essential.

Using the head configuration of Fig. 2D, the record-playback preamplifier (in the previous article) is merely connected to the erase and record-playback heads in the normal manner for monaural erase-record operation. For stereophonic playback, it is preferable that two properly compensated preamplifiers be connected to the output jacks provided for the stacked heads. The record-playback preamplifier could, of course, be used in either of the stereophonic channels, if desired, eliminating the cost of one playback preamp. This, however, would involve the risk of accidentally erasing a valuable stereo tape.

As part of the installation procedure, of course, each of the heads must be carefully rotated on the axis of its mounting screw to provide maximum volume and then must be azimuth aligned, adjusting the two azimuth adjustment screws provided for each head until maximum volume is obtained. This alignment should normally be made using a high-frequency alignment tape or a music tape which has fairly consistent high-frequency output.

It should be noted that in-line heads are recommended only when used with pressure pads. This is due to the fact that a slight deformity of the tape can result in momentary loss of contact with one or the other of the head pole pieces resulting in momentarily reduced output. A tapelifter and pressure pad mechanism is available as an accessory. The pad assembly and tapelifter lifts automatically as the control is moved into the "stop," "fast forward," or "rewind" positions and holds the tape in contact with the heads when the control is in the "forward" position.

The tapelifter mechanism is desirable in any case in recording applications. Since it lifts the tape from the head in fast forward and rewind, it prevents accidental erasure of a tape in the event the operator forgets to switch the record-playback amplifier out of the "record" position upon completion of a recording.

Amplifier-Speaker Requirements

For stereophonic playback of recorded tapes, the basic preamplifier, power amplifier, and speaker system required for a monaural system can be exactly duplicated. The preamplifier function is adequately served by the *Viking* PB60, described in a previous part Power amplifiers with only a few watts of power will provide acceptable performance, but high-fidelity amplifiers capable of handling from 30 to 60 watts of power will sound immeasurably better.

It is a common misconception that a stereophonic system eliminates the need for high-quality amplifiers and reproducers. Actually, the same fundamental requirements still apply. It is true that stereo reproduction has such brilliance and such a pleasing "new" sound that even an ordinary table radio or phono reproducer can be utilized as a means to an end. However, the same program material sounds infinitely better when reproduced with solid, well-damped amplifiers and true high-fidelity speakers.

It should be remembered that well-recorded tape provides a dynamic range of over 100 to 1. It takes adequate amplifier power, a low signal-to-noise ratio, and good speakers to fully utilize this range.

Speaker arrangement is a matter of personal taste and is affected by the individual room requirements. Both speakers should be arranged along one wall or, if preferred, in corners at one end of the listening area. The distance between the speakers should be less in a small room than in a large listening area. Normally, however, a spacing of from 6 to 15 feet will prove most desirable. To be completely technical, the ideal spacing would depend upon the microphone placement during recording, the size of the musical group being recorded, and on the acoustical properties of the room in which the recording was made. Manufacturers often state the microphone placement used for a given recording, but not with the expectation that it will be cause for rearrangement of furniture.

Incidentally, there is no real necessity for using matched amplifiers and speakers for stereo reproduction. If the system is to be used for monaural as well as stereo operation, however, it is logical that the best amplifier and the most adequate speaker be placed in the left-hand position. This is for two reasons. Properly set up, the left-hand channel (lower track on tape) feeds the left-hand speaker system. Thus in using the equipment for monaural recording and playback you automatically utilize the best amplifier and speaker system.

Stereophonic Recording

The same stacked or staggered head assemblies which serve for playback may be used for stereophonic recording. This fact will be of particular interest in those fortunate areas where stereophonic FM-AM broadcasts are available. It should be noted, however, that using present *Viking* equipment or, for that matter, any recorder which does not provide full-track or dual-track erase heads, bulk erasure is essential. This has the advantage, however, of insuring minimum noise level. Bulk erasers range in price from $15.00 to $50.00 or more. In its simplest form, the bulk eraser is a heavy a.c.-powered electromagnet to which a full reel of tape is thoroughly exposed. With the current still applied, the reel is slowly removed from the field. Careful and thorough bulk erasure will reduce tape noise to the lowest possible degree and is completely acceptable for most applications.

In stereophonic recording it is advisable that both heads be biased from the same bias source. This refers to the ultrasonic bias voltage discussed in connection with the record-playback preamplifier described in previous part. If entirely independent bias oscillators are employed, as would be the case if two independent record-playback preamplifiers were used, the difference frequency of the two bias oscillators would appear on the recorded tracks as an audible beat frequency. This, of course, is the result of the crosstalk characteristic of the heads and, to some extent, would be a problem even with the isolation provided by a staggered head arrangement.

To overcome this it is necessary to supply bias to the two recording heads from only one of the record-playback preamplifiers. Circuit alterations are indicated in Fig. 3, which shows the essential portion of the *Viking* RP61 record-playback preamplifier circuit. In the right-hand channel amplifier, the bias oscillator is merely disabled by removing the 12AV7 oscillator tube, and the series resistor, R_9, is disconnected at switch S_1. That terminal on the switch is then bridged, using shielded cable, to the same switch terminal on the left-hand channel record-playback preamplifier. In thus doubling the load on the left-hand channel oscillator, it will be necessary to readjust the bias adjustment potentiometer, R_{13}. It will also be necessary to substitute a 27,000 ohm resistor for the 56,000 ohm resistor, R_9. Note, also, that since bulk erasure is used, both erase head leads should be removed. The left-hand channel lead is replaced with a 30 mhy. choke in order to provide a suitable load for the oscillator. As in monaural recording a bias current of .8 ma. should be applied to each of the record-playback heads during recording. The method for determining this critical current is described in previous article.

The RP61 record-playback preamplifier described here and in the preceding issue is, of course, adaptable to either monaural or stereophonic recording from microphones as well as from radio or phono inputs. The over-all recording amplifier gain of 60 db plus is adequate to justify use of better-than-average microphones having a frequency response characteristic of 60 to 12,000 cps or better. Dynamic microphones of this type are generally available in the $40.00 to $60.00 price range and are well worth the investment. —30—

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Recording from Tape to Tape with a Single Deck

By **JAMES A. McROBERTS**

Preserve your priceless recordings by making new tapes of the material. You can do it on your standard home machine.

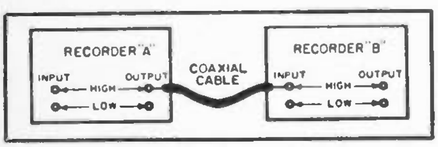

Fig. 1. Method for rerecording tape on two separate machines. Refer to article.

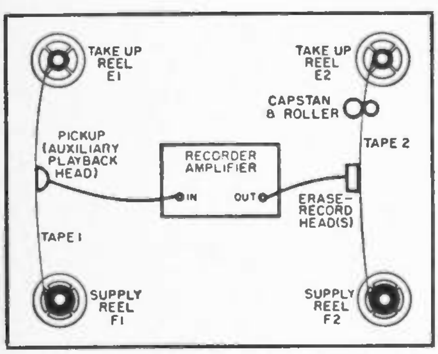

Fig. 2. Simplified diagram illustrating the basic idea of duplicating a magnetic tape utilizing a single recording machine equipped with auxiliary head. Details in text.

NUMEROUS occasions may arise making the duplication, or rerecording, of a magnetic tape desirable, or necessary, or both desirable and necessary. An irreplaceable tape, or perhaps a valuable tape becoming noisy due to wear, are common examples.

No particular problem exists if a second recorder is available having the same tape transport speed. Even the "same speed" requirement may be waived if the machine that records the tape will be used in playing it later. Of course, the recorder that picks up the old material must be similar in speed to the recording that it picks up. As Fig. 1 shows, the output of the "pickup" recorder "A" feeds the input of the "record" recorder "B" that does the job of making the new tape or rerecording. Naturally, the impedances of the two machines (output of "A" and the input of "B") must be nearly the same or an impedance matching network is needed. An additional machine may not be available, or perhaps the expense involved does not justify the purchase or rental of a second machine. This article will describe means whereby a single recorder may be used to make a duplicate recording.

Fig. 3. A horizontal plan of a dual tape transport mechanism. Tape 1 is the original tape while Tape 2 is the new recording being made of material on Tape 1.

To effect rerecording on a single machine, two problems must be solved. These are: 1. An additional tape must be transported at constant speed with supply reel braking and takeup facilities, and 2. The additional tape, the old tape from which a record is desired, must be pulled past a playback head to pick off its recorded signals.

The general idea is illustrated in Fig. 2, which outlines the problem. We will now consider, in detail, how the separate phases are solved.

Dual Tape Transportation

If we set up a recorder and thread a tape through it in the customary manner for recording, then the capstan and its roller govern the linear speed at which the recorded tape is wound on the takeup reel. Therefore, if we place another reel on top of this takeup reel, and key it to the lowermost reel, the added upper reel will pull a tape on it at the same linear speed as the bottom reel pulls its tape on to itself and winds that tape up. Thus, if the reel (bottom record takeup reel, see Fig. 3) is of the same spool diameter as the reel on top of it (the playback takeup reel), then tape 1 (the top tape being played back for rerecording) will travel at the same speed linearly as the bottom tape (on which the rerecording is being performed) which latter tape is capstan controlled. Hence, the two tapes travel linearly at the same speed.

Fig. 3 shows also a form of a brake on the supply reel for the uppermost supply reel. This brake is a piece of rather rough cardboard laid on top of the lower reel after the "new" tape has been threaded. It provides some additional braking of the top reel over the bottom. A pressure pad brake could have been arranged against this reel's rim as an alternative measure. The paper "brake" plus the braking inherent in the machine (applied to the supply reel turntable) is quite sufficient for all practical purposes.

The old or master tape is threaded past an auxiliary pickup (playback) head—more later about it—onto the

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takeup reel. See Figs. 2, 3, and 4. The takeup reel must turn at the same angular speed as the bottom takeup reel as previously mentioned. The spindle on all recorders will project far enough through the top of the bottom reel to position the top reel but will not force it to revolve. This revolution is forced by the insertion of a key in the key spaces on the standard reels. Two pieces (halves) of a toothpick form a good keying device as shown in Fig. 6. The problem of pulling tape 1 at the same speed as tape 2 is now solved completely.

The Auxiliary Head

As Fig. 3 shows, the top tape is pulled past an auxiliary playback head. This head is connected to the input of the recorder as indicated in Fig. 2. If the auxiliary head is properly chosen, no impedance matching will be required—simply plug into the high or the low impedance input to the recorder, usually the high impedance or microphone input. A matching network may be used if such is necessary.

In choosing the head, see that it has tape guides. If no guides are furnished as an integral part of the head, then they must be fashioned to hold the tape in position vertically against the head. Also, check to see that the head is either a lower track or an upper track head as the old recording(s) require.

In selecting the position for the auxiliary head, the warp must be considered. Fig. 5 illustrates correct and incorrect warping. Some types of heads do not require warping but do need pressure pads. Naturally, the pads are additional cost and trouble. If needed you must fit them in the same manner as for ordinary playback service. Preferably purchase a head that will tend to hold the tape tight against itself by the warp method.

The mountings for the head used in this particular machine were fashioned from small brackets procured at the local dime-store. The top of the machine's head cover provided a handy support, as is shown in the photo. All adjustments of the head except the azimuth adjustment could be obtained by the joints in the hardware. The azimuth adjustment was secured by the forming of the thin metal brackets—thin enough for bending but still sturdy enough so that excessive vibration would not be encountered. A wingnut is preferable to a plain nut where it can be used since it is easier to tighten or loosen.

The vertical adjustment of the head must be such that the tape will lie against it with normal tension applied; the tape guides will do very little work if this vertical adjustment is properly made. The head must be adjusted so that the tape will lie flat against the contour of the head, particularly the gap. The top of the tape should not exert more pressure against the head than the bottom of the tape; bouncing will result if this alignment is not correct. A further adjustment is making the gap at right angles to the direction of travel of the tape. Loss of volume, the loss of high note volume particularly, will result if this setting is incorrect. Monitoring the output with headphones while adjusting the head will tell the correct adjustment—the greatest volume with all other things equal is the correct adjustment point.

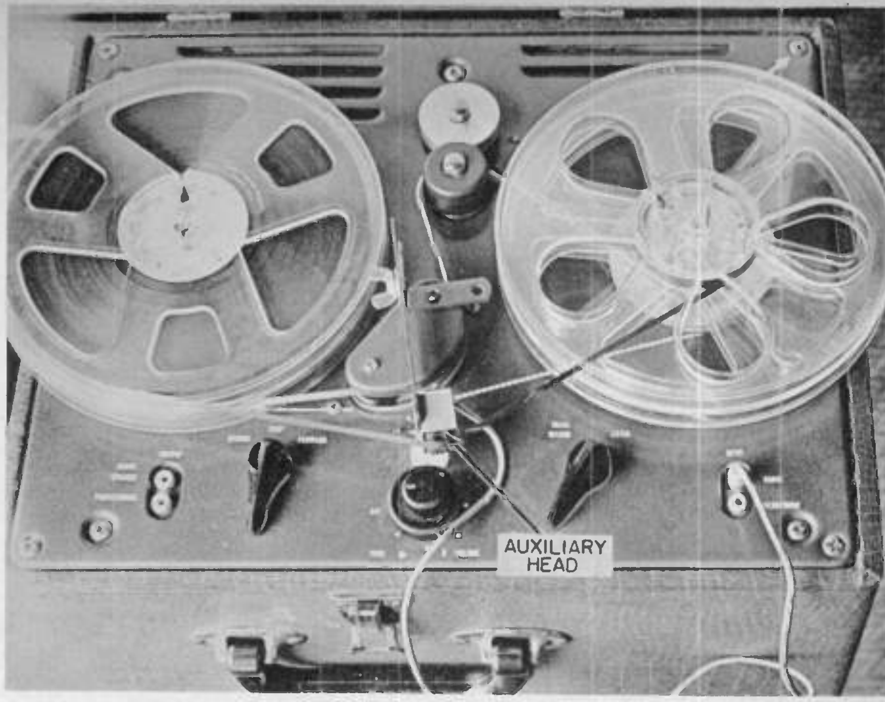


Fig. 4. Typical setup for rerecording with a single, standard tape recorder.

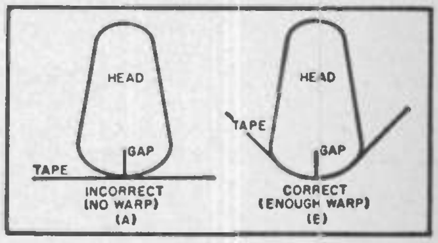


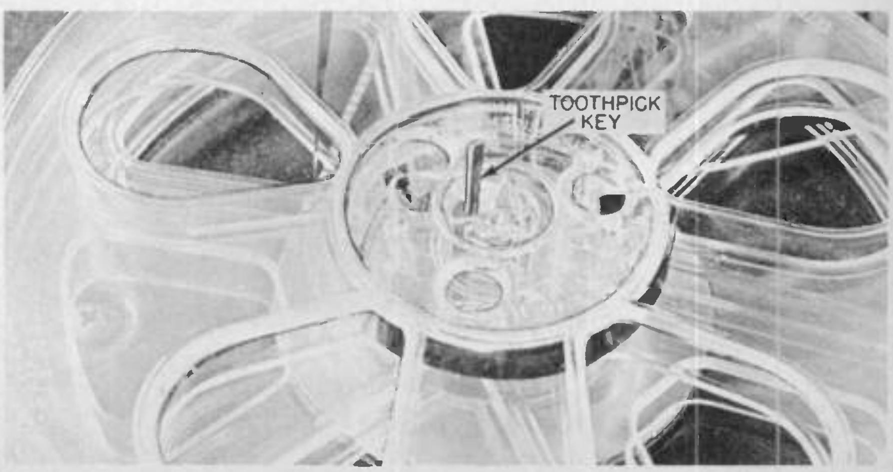
Fig. 5. The correct and incorrect warping of recording tape around playback head.

Now the sound is all electrical in the case of this type of recording. There is no guide post except the overload indicator, which may be rather crude. For this reason, the output should be monitored with a pair of headphones. Pay particular attention to the lower passages. The overload indicator will tell you if the higher or louder passages are too strong.

At the start of the recording, one hand should be placed on the record level control. The level may be too great or too little at the beginning and must be adjusted. If you do not prepare yourself, the reels may require rewinding and starting the process over again.

A standard test tape record has been rerecorded in such a manner with satisfactory results. No tonal differences could be observed between the original and the rerecording which indicated relatively constant speed. The quality will depend to an extent on the alignment of the head, which must be performed with care, as much care as the main heads of the machine. —30—

Fig. 6. Closeup showing the toothpick "key" on take-up reel. Refer to article.



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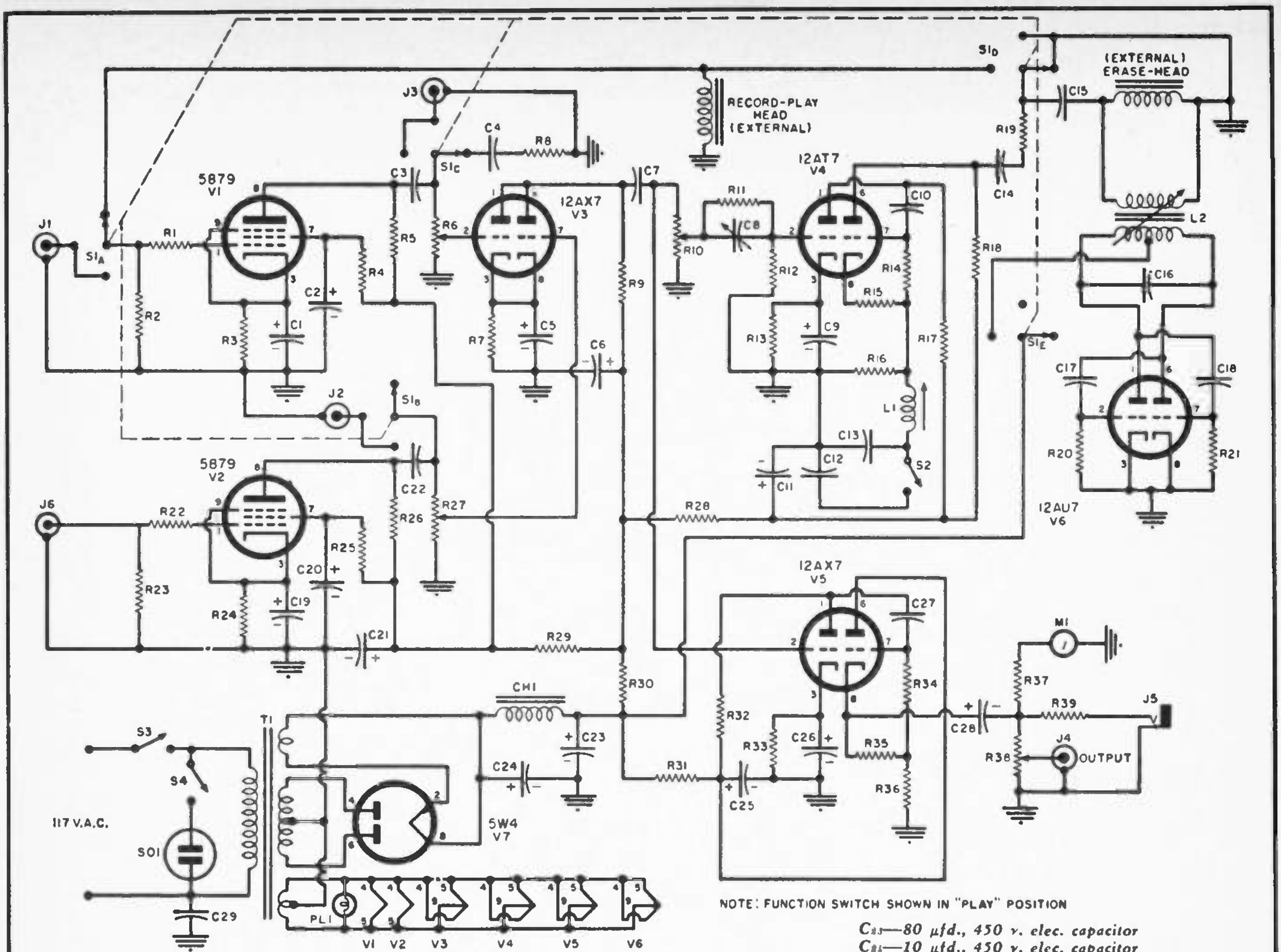

R_1, R_{22}—33,000 ohm, ½ w. res
R_2, R_{23}—2 megohm, ½ w. res.
R_3, R_{24}—2700 ohm, ½ w. res.
R_4, R_{25}—2.2 megohm, 1 w. res.
R_5, R_{26}—470,000 ohm, 1 w. res.
R_6, R_{10}, R_{27}—250,000 ohm audio pot
R_7, R_{13}, R_{37}—1500 ohm, ½ w. res.
R_8—10,000 ohm, ½ w. res.
R_9, R_{32}—47,000 ohm, 1 w. res.
R_{11}—680,000 ohm, ½ w. res.
R_{12}—33,000 ohm, ½ w. res
R_{14}—470,000 ohm ½ w. res.
R_{15}—470 ohm, ½ w. res.
R_{16}—8200 to 47,000 ohm, ½ w. res. (see text)
R_{17}—100,000 ohm, 1 w. res.
R_{18}—39,000 ohm, 1 w. res.
R_{19}—68,000 ohm, ½ w. res.
R_{20}, R_{21}, R_{39}—100,000 ohm, ½ w. res.
•R_{28}—2700 ohm, 1 w. res.
R_{29}—10,000 ohm, 1 w. res.
R_{30}, R_{31}—4700 ohm, 1 w. res.
R_{33}—3300 ohm, ½ w. res.
R_{34}—1 megohm, ½ w. res.
R_{35}—820 ohm, ½ w. res.
R_{36}—47,000 ohm, ½ w. res.
R_{38}—1 megohm audio pot
C_1-C_9-C_{19}—20/20/20 μfd., 25 v. elec. capacitor
C_2-C_{20}—10/10 μfd., 350 v. elec. capacitor
C_3, C_7, C_{22}—.1 μfd., 400 v. capacitor
C_4, C_{12}—.03 μfd., 400 v. capacitor
C_5, C_{26}, C_{28}—10 μfd., 25 v. elec. capacitor
C_6-C_{11}-C_{21}-C_{25}—10/10/10/10 μfd., 450 v. elec. capacitor
C_8—100-400 μμfd. trimmer capacitor
C_{10}, C_{27}, C_{29}—.05 μfd., 400 v. capacitor
C_{13}—.01 μfd., 400 v. capacitor
C_{14}—.25 μfd., 400 v. capacitor
C_{15}—250 μμfd., ceramic capacitor
C_{16}—.004 μfd., 600 v. capacitor
C_{17}, C_{18}—500 μμfd. ceramic capacitor
C_{23}—80 μfd., 450 v. elec. capacitor
C_{24}—10 μfd., 450 v. elec. capacitor
S_{1A}, S_{1B}, S_{1C}, S_{1D}, S_{1E}—3-pos., 5-sec. ceramic function switch (Centralab PA-2015)
S_2, S_3, S_4—S.p.s.t. bat-handle toggle switch
M_1—Rectifier type db or vu meter (see text)
L_1—20 mhy. variable powdered-iron-core coil (TV linearity or width coil can be used)
L_2—Osc. coil (Pentron 321-4 or equiv.)
T_1—Power trans., 350 v. @ 40 ma.; 5 v. @ 2 amps.; 6.3 v. @ 3 amps. (encased type)
J_1, J_6—Shorting type microphone connector (Amphenol Series 75)
J_2, J_3, J_4—Phono jack
J_5—Phone jack
CH_1—8 hy., 40 ma. filter choke (encased type)
SO_1—117 volt power-line socket
PL_1—6.3 volt pilot light
V_1, V_2—5879 tube
V_3, V_5—12AX7 tube
V_4—12AT7 tube
V_6—12AU7 tube
V_7—5W4 tube

Complete schematic diagram of the tape recorder amplifier. It may also be used as a two-channel audio mixer for p.a. applications.

lows complete mixing flexibility. Thus, two inputs are available for crystal or other high impedance microphones. Two additional inputs for high-level signals provide for FM-AM tuners, TV audio, phono signals, or other sources. With two mixer channels, independent control of each channel level is possible. It is apparent that recorded music can be introduced as "background" for a microphone pickup; cross-fades and similar professional mixing techniques are possible, to list a few of countless examples.

Referring again to the circuit, the 5879 stage now functions as one microphone preamp with S_{1A} in the "Record" position. An additional 5879 is similarly used for the second microphone preamp. Series grid resistors are added to avoid rectification of strong broadcast carriers which can occasionally prove troublesome in high-gain amplifiers. (See the short item "FM Interference" by Ken Maxwell on page 122 of the February 1951 issue of this magazine.)

The two high-level inputs are introduced through sections S_{1C} and S_{1B} of the function switch. R_6 and R_{27} serve as the mixer gain controls. These controls feed individual grids of the mixer tube, the common plate load producing electronic mixing without interaction or crosstalk. The output of the mixer is parallel-fed to two independent output amplifiers.

The 12AX7 amplifier-cathode follower circuit now functions as the monitor amplifier. A recording level meter and a headphone output jack are added to the cathode follower for this application. Notice that only the mixer gain controls determine the level supplied to the meter and headphones. Additionally, the normal amplifier output is available with an output gain control affecting this output signal only.

A 12AT7 is used for the record amplifier section. A screwdriver-adjusted takeup reel. See Figs. 2, 3, and 4. The takeup reel must turn at the same angular speed as the bottom takeup reel as previously mentioned. The spindle on all recorders will project far enough through the top of the bottom reel to position the top reel but will not force it to revolve. This revolution is forced by the insertion of a key in the key spaces on the standard reels. Two pieces (halves) of a toothpick form a good keying device as shown in Fig. 6. The problem of pulling tape 1 at the same speed as tape 2 is now solved completely.

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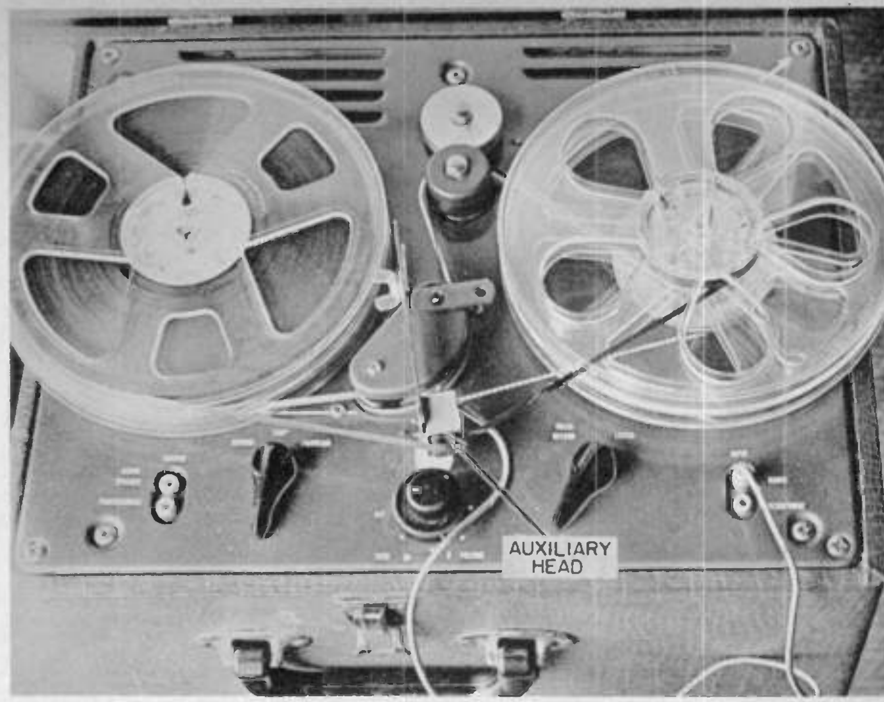

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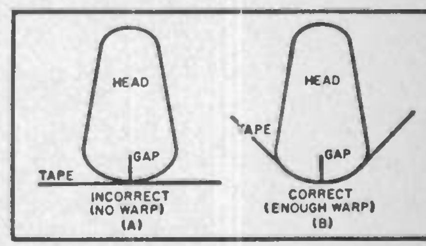

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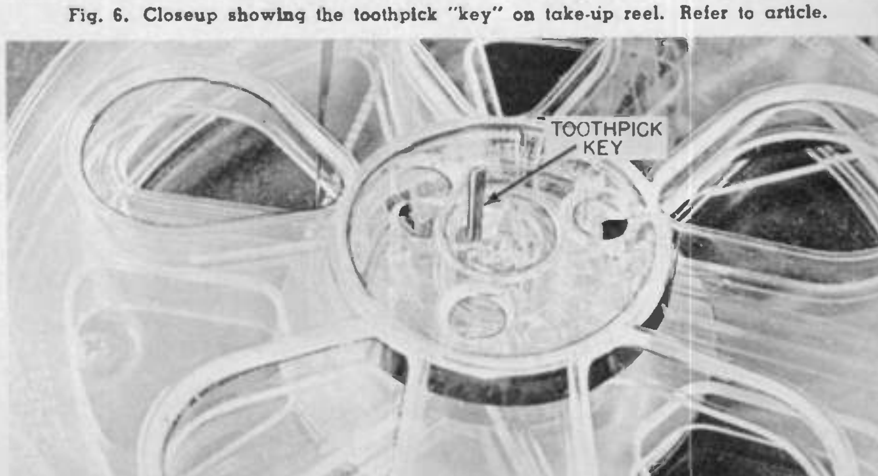

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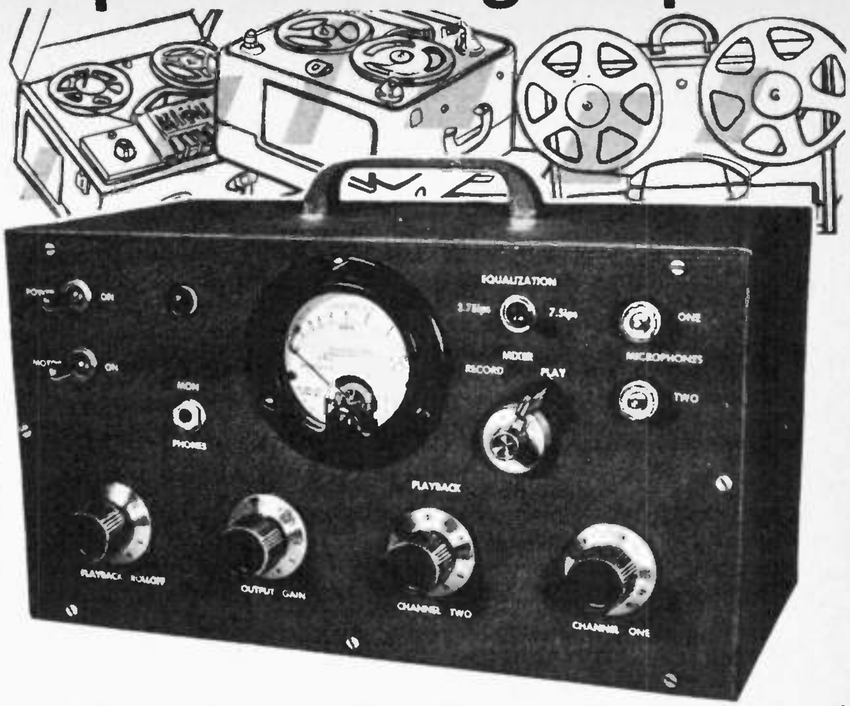
A Professional Tape Recording Amplifier for Home Hi-Fi Systems

By
MAURICE P. JOHNSON
Facilities Engr., Station WAAM



Over-all view of the author's home-built "professional-type" tape recorder amplifier designed to improve home music systems.

The design and construction of a versatile recording and playback amplifier for use with inexpensive tape decks to produce quality tapes at low cost. Audio mixer is provided.

IN A SHORT period of years, tape recorders have come to play an increasingly important role in the home audio system. A few years back, tape recorders were found only in recording studios, broadcast stations, and other professional establishments. Today, by contrast, there are recorders available in all price ranges, with the result that they are being used at all levels of audio system complexity, including the home music system.

There is little need to emphasize the convenience of recording as offered by the tape recorder in conjunction with the home audio outfit. The advantages of recording broadcasts, such as those aired by fine-music FM stations, for future use are obvious. The sentimental aspects of recording the voices of children, friends, or others need not be expounded. In short, it is apparent that a tape recorder is a valuable adjunct to any home sound system.

It is, of course, possible to buy commercial recorders in many price ranges and of varying degrees of excellence. However, there are many audio fans who enjoy constructing their own high-fidelity systems. From such a constructor's point of view, there are two general assemblies in the tape recorder. The tape transport mechanism is a mechanical device requiring close tolerances and careful fits, as well as ingenious design. As such, it is not easily duplicated by the home constructor. Fortunately, tape transport mechanisms of good quality are readily available so that the job of assembling a custom tape recorder can be confined to the construction of a suitable amplifier. Such an amplifier will be discussed in this article. While the unit described is capable of providing high-fidelity performance, emphasis has been directed toward a design involving a relatively small cash outlay.

The design of the amplifier is predicated, to a large degree, by the recording heads and transport mechanism involved. An inexpensive *Pentron* TM- 56 mechanism was selected as typical of the commercially available tape transports. Naturally, one of the others might have been equally satisfactory.

Specifications on the TM-56 include 3.75 and 7.5 inches-per-second tape speeds, and a flutter of 0.3% at 7.5 ips. Speed stability, wow, and flutter determine the mechanical limits of tape fidelity, with values below 0.5% considered necessary for quality results. The inexpensive mechanism compares favorably with more expensive units in this respect. One professional broadcast tape recorder, as an example, is rated at 0.2% r.m.s. combined flutter and wow at 7.5 ips speed. Thus, it is seen that the mechanical abilities of the inexpensive transport need not be a limiting factor in the design of a high-fidelity tape outfit.

The heads furnished in production models of the TM-56 have removable pole pieces, so that single-track or dual-track heads may be interchanged as desired. Single-track heads were used during development work on the project, so that tapes could be recorded on the TM-56 and played on an *RCA* RT-11A broadcast recorder for quality and equalization checks.

A combined record-playback head with a gap of .0005" is used, thereby becoming the limiting factor on the frequency response. Such heads as the *Brush* "Redheads" with .00025" gaps, or *Dynamu* heads with .00015" gaps are theoretically capable of more extended high-frequency response. However, with the existing heads on the *Pentron,* and proper equalization, it should be possible to obtain reasonably flat response from 50 to over 10,000 cps, at the 7.5 ips speed. This range compares quite favorably with professional machines operating at this speed. Distortion-free response beyond 10 kc. makes the use of professional speeds of 15 or even 30 ips almost essential. Other factors such as beats and intermodulation between high audio frequencies and the supersonic erase and bias oscillator serve to complicate the design for extremely high frequency response.

In general, at 7.5 ips speed, the overall characteristic of the head without equalization is as shown on next page. At this speed, it will be noticed that peak response is obtained in the vicinity of 1 to 3 kc. Because the head is a magnetic device, lower frequencies fall off in direct proportion to frequency, the slope being independent of the tape speed. Another way of stating the same thing is to say that the response falls off at a rate of 6 db per octave.

At high frequencies, however, the roll-off is determined to a large degree by tape speed as well as the head gap. The individual head thus enters

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into the design problem. Frequency response curves of magnetic heads are covered in detail in such texts as Oliver Read's "Recording and Reproduction of Sound," or "Magnetic Recording" by S. J. Begun, for those interested in a more complete analysis. It should be noted, however, that in this case the relative levels at 50 cps and at 10 kc. are approximately 20 db below the response at 2 kc. This unequalized head response indicates the nature of equalization which must be introduced in the amplifier.

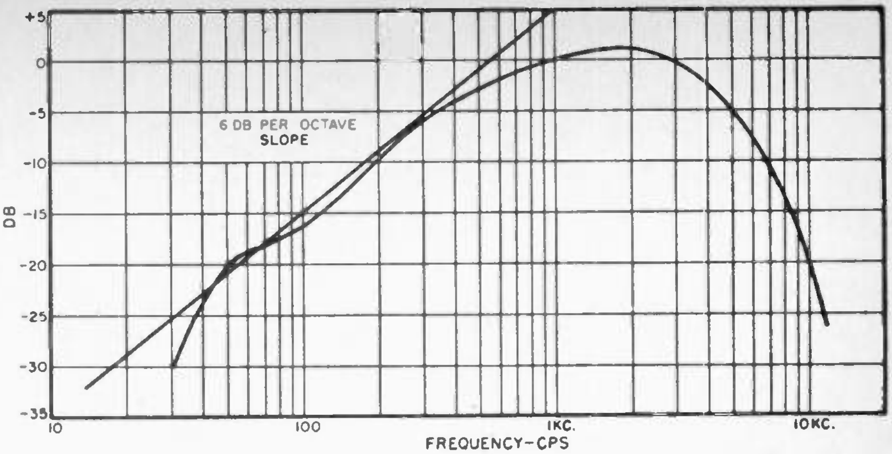
Progressing to the amplifier itself, now that the nature of the problem has been considered, it was decided that the design should be as flexible and versatile as practical, in order to cope with almost any recording situation which might arise. To this end, the amplifier was designed for three definite purposes: namely, a record amplifier, a playback amplifier, with the third application of mixer amplifier for public address work included without adding complexity to the circuit. Several stages have been made multi-purpose by incorporating a five-section transfer "function" switch.

Discussing the playback amplifier first, it is necessary to provide a high gain circuit, with low noise and hum levels to amplify the feeble tape signals to levels sufficient to feed into the usual home high-fidelity audio system. Bass boost must be included, with a turnover at 1000 cps, and a maximum boost of 20 db at 50 cps. Examination of the circuit will reveal four stages devoted to this function.

The record-playback head is connected to a low-noise pentode preamplifier stage, with function switch S_{1A} in the "Play" position. This 5879 stage is designed for maximum gain, in the interests of a high signal-to-noise ratio. The preamplifier stage is coupled to the playback gain control, R_8. Section S_{1C} of the function switch inserts the bass boost equalizer, consisting of C_1 and R_5 shunted across the preamp plate load. The equalized signal is further amplified by a triode mixer stage, then applied to the following 12AX7 output tube. One triode is used for additional voltage gain, while the remaining half of the dual triode serves as a cathode follower to furnish a low impedance output at a signal level of over one volt.

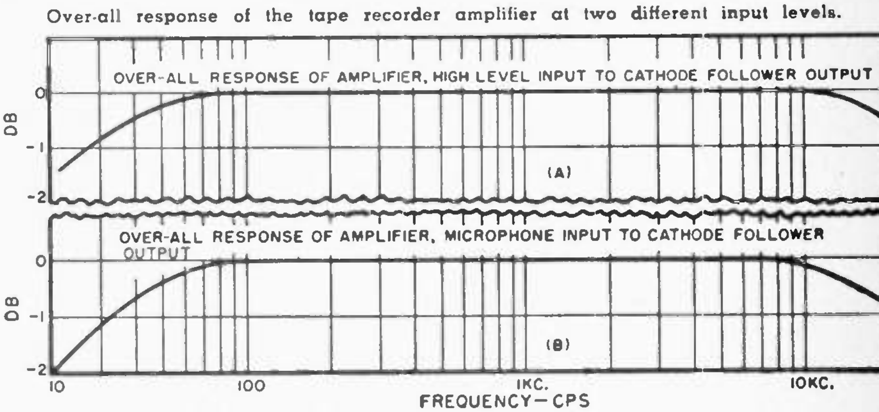
It should be noted that the only frequency compensation in this entire chain is that produced by the shunt equalizer, C_1 and R_5. In fact, with the equalizer removed, the response of the playback amplifier is flat within ± 1 db from 20 to 20,000 cps. Such convenient methods of equalization have the advantages of easy adjustment or modification to conform to the needs of a particular head. For example, the values of the shunt equalization components given are correct only for the .0005" gap recording head. If it is desired to use a head with a .00025" gap, the value of C_1 should be reduced to .01 or .02 μfd. If a head with a gap of .00015" is employed, C_1 should be further reduced to .005 or .008 μfd.

The playback amplifier is straightforward, but the record amplifier becomes a bit more complex. The first aspect to be considered concerns the input facilities to be provided. Some simple recorders provide a single mixer channel, but greater flexibility is a decided asset. Two mixer channels are obviously needed. A high-level and a low-level input for each channel al-



Recording head characteristics shown at 7.5 ips tape speed with no equalization provided. This curve is based on data supplied by the Pentron Corporation and is taken with the Pentron 9T-3M head and the 3-M Company's "Scotch" #111-A tape.

Over-all response of the tape recorder amplifier at two different input levels.



Rear view shows 2 separate input jacks (left) and 2 paralleled output jacks (center). The Jones connectors carry signals to the tape heads and a.c. power to the tape deck.

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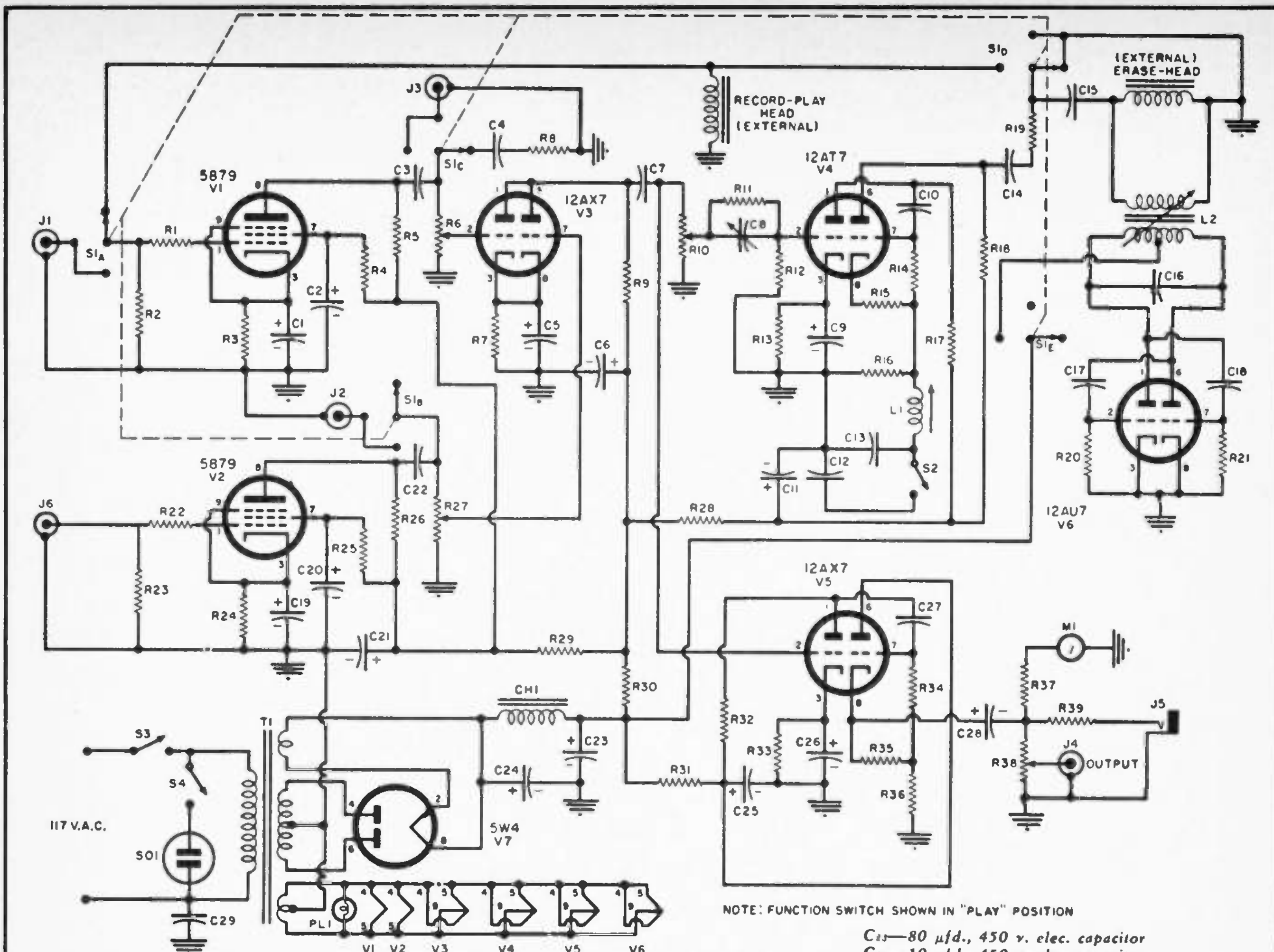

R_1, R_{22}—33,000 ohm, ½ w. res
R_2, R_{23}—2 megohm, ½ w. res.
R_3, R_{24}—2700 ohm, ½ w. res.
R_4, R_{25}—2.2 megohm, 1 w. res.
R_5, R_{26}—470,000 ohm, 1 w. res.
R_6, R_{10}, R_{27}—250,000 ohm audio pot
R_7, R_{13}, R_{37}—1500 ohm, ½ w. res.
R_8—10,000 ohm, ½ w. res.
R_9, R_{32}—47,000 ohm, 1 w. res.
R_{11}—680,000 ohm, ½ w. res.
R_{12}—33,000 ohm, ½ w. res
R_{14}—470,000 ohm ½ w. res.
R_{15}—470 ohm, ½ w. res.
R_{16}—8200 to 47,000 ohm, ½ w. res. (see text)
R_{17}—100,000 ohm, 1 w. res.
R_{18}—39,000 ohm, 1 w. res.
R_{19}—68,000 ohm, ½ w. res.
R_{20}, R_{21}, R_{39}—100,000 ohm, ½ w. res.
R_{28}—2700 ohm, 1 w. res.
R_{29}—10,000 ohm, 1 w. res.
R_{30}, R_{31}—4700 ohm, 1 w. res.
R_{33}—3300 ohm, ½ w. res.
R_{34}—1 megohm, ½ w. res.
R_{35}—820 ohm, ½ w. res.
R_{36}—47,000 ohm, ½ w. res.
R_{38}—1 megohm audio pot
C_1-C_9-C_{19}—20/20/20 μfd., 25 v. elec. capacitor
C_2-C_{20}—10/10 μfd., 350 v. elec. capacitor
C_3, C_7, C_{22}—.1 μfd., 400 v. capacitor
C_4, C_{12}—.03 μfd., 400 v. capacitor
C_5, C_{26}, C_{28}—10 μfd., 25 v. elec. capacitor
C_6-C_{11}-C_{21}-C_{25}—10/10/10/10 μfd., 450 v. elec. capacitor
C_8—100-400 μμfd. trimmer capacitor
C_{10}, C_{27}, C_{29}—.05 μfd., 400 v. capacitor
C_{13}—.01 μfd., 400 v. capacitor
C_{14}—.25 μfd., 400 v. capacitor
C_{15}—250 μμfd., ceramic capacitor
C_{16}—.004 μfd., 600 v. capacitor
C_{17}, C_{18}—500 μμfd. ceramic capacitor
C_{23}—80 μfd., 450 v. elec. capacitor
C_{24}—10 μfd., 450 v. elec. capacitor
S_{1A}, S_{1B}, S_{1C}, S_{1D}, S_{1E}—3-pos., 5-sec. ceramic function switch (Centralab PA-2015)
S_2, S_3, S_4—S.p.s.t. bat-handle toggle switch
M_1—Rectifier type db or vu meter (see text)
L_1—20 mhy. variable powdered-iron-core coil (TV linearity or width coil can be used)
L_2—Osc. coil (Pentron 321-4 or equiv.)
T_1—Power trans., 350 v. @ 40 ma.; 5 v. @ 2 amps.; 6.3 v. @ 3 amps. (encased type)
J_1, J_6—Shorting type microphone connector (Amphenol Series 75)
J_2, J_3, J_4—Phono jack
J_5—Phone jack
CH_1—8 hy., 40 ma. filter choke (encased type)
SO_1—117 volt power-line socket
PL_1—6.3 volt pilot light
V_1, V_2—5879 tube
V_3, V_5—12AX7 tube
V_4—12AT7 tube
V_6—12AU7 tube
V_7—5W4 tube

Complete schematic diagram of the tape recorder amplifier. It may also be used as a two-channel audio mixer for p.a. applications.

lows complete mixing flexibility. Thus, two inputs are available for crystal or other high impedance microphones. Two additional inputs for high-level signals provide for FM-AM tuners, TV audio, phono signals, or other sources. With two mixer channels, independent control of each channel level is possible. It is apparent that recorded music can be introduced as "background" for a microphone pickup; cross-fades and similar professional mixing techniques are possible, to list a few of countless examples.

Referring again to the circuit, the 5879 stage now functions as one microphone preamp with S_{1A} in the "Record" position. An additional 5879 is similarly used for the second microphone preamp. Series grid resistors are added to avoid rectification of strong broadcast carriers which can occasionally prove troublesome in high-gain amplifiers. (See the short item "FM Interference" by Ken Maxwell on page 122 of the February 1951 issue of this magazine.)

The two high-level inputs are introduced through sections S_{1C} and S_{1B} of the function switch. R_6 and R_{27} serve as the mixer gain controls. These controls feed individual grids of the mixer tube, the common plate load producing electronic mixing without interaction or crosstalk. The output of the mixer is parallel-fed to two independent output amplifiers.

The 12AX7 amplifier-cathode follower circuit now functions as the monitor amplifier. A recording level meter and a headphone output jack are added to the cathode follower for this application. Notice that only the mixer gain controls determine the level supplied to the meter and headphones. Additionally, the normal amplifier output is available with an output gain control affecting this output signal only.

A 12AT7 is used for the record amplifier section. A screwdriver-adjusted

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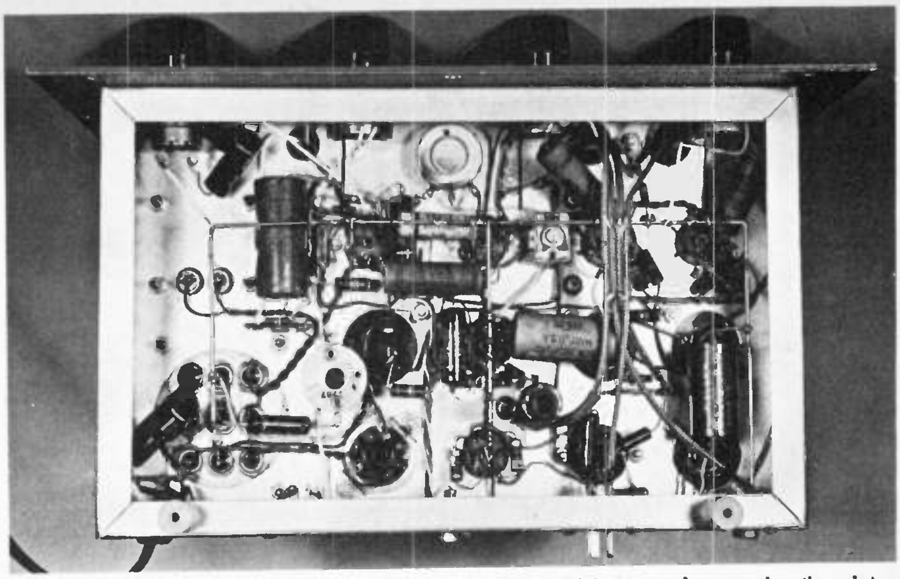


Under chassis view of amplifier. Note ground bus which is used as major tie-point.

gain control, R_{10}, determines the level fed to the recording head, and is used to coordinate the level read by the meter and that being supplied to the head. The two triodes of the 12AT7 are connected as conventional voltage amplifiers, plus the addition of high frequency pre-emphasis to produce the necessary tape equalization. The parallel combination of R_{11} and C_8 provide high frequency boost. The capacitor acts as a high-pass filter, with an inherent slope of 6 db per octave. The shunt resistor limits the loss at low frequencies. The variable capacity permits the turnover to be adjusted.

Additional high peaking is accomplished by an *LC* circuit added to the cathode of the output stage. The stage normally operates with some degeneration, because of the configuration of R_{14}, R_{15}, and R_{16}. The coil and capacitor form a series circuit in shunt with R_{16}, thus producing considerable gain at the resonant frequency.

Because high-frequency response is determined by the individual head, the high-frequency equalization is made variable in three ways. As stated, capacitor C_8 determines the turnover frequency of the high boost circuit, hence it affects the response in the vicinity of 2 to 5 kc. The inductance of the cathode coil determines the frequency of resonance. Normally, it is tuned to approximately 10 kc., but the variable iron core permits shifting this frequency as needed to control the response at the extreme high frequency end. The shunt resistor across the resonant circuit (R_{16}) controls the amplitude of the peaking. Larger values of resistance give a higher "*Q*" with consequent greater amplitude at resonance, but at the expense of increased loss in the normal gain of the stage. Likewise, smaller values of resistance will reduce the peaking, and increase the over-all stage gain. For example, with R_{16} at 8200 ohms, there will be a boost of about 8 db at 10 kc. This should suffice for very narrow gap heads. A value of 47,000 ohms will boost 10 kc. about 15 db, and will be adequate for .0005" head gaps.

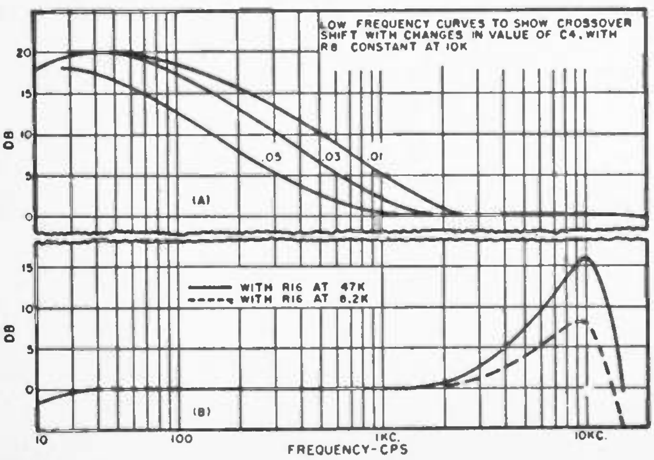
It should be pointed out that the total high peaking is limited by the permissible distortion. This is based on the fact that the harmonic distortion is increased by the high-frequency boosting required for equalization as compared to that inherent in a normally flat system. One additional means of high boost, which may be included, is that of resonating the record head at the desired high frequency by connecting a suitable capacity across the head, if needed.

It is suggested that any individual constructor make some frequency runs on the head used and adjust the compensation networks as necessary to obtain desired flat response. Of course, the tape should be recorded with the amplifier under discussion so that the over-all equalization can be checked. With proper adjustment of these variables which determine the equalization, it should be possible to record any good microgroove record and produce playback quality which is indistinguishable from the record itself.

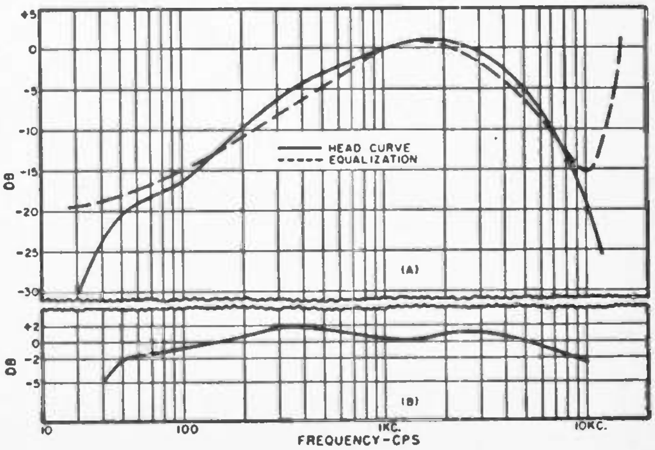
Quality recording should, of course, be done at the 7.5 ips speed. However, the 3.75 ips speed is available for less critical applications, such as spoken material. Switch S_2 is closed at the slow speed, to add additional capacity to the series resonant circuit, thereby lowering the frequency of resonance, to produce boost to help compensate for the increased roll-off at this speed.

The recording output stage is coupled to the record head through the resistor R_{19}, passing through section S_{1D} of the function switch. Also necessary for tape recording, of course, is a suitable source of bias and erase voltage. This requires a supersonic oscillator which produces a symmetrical a.c. waveform. A 12AU7 standard push-pull oscillator is used for this purpose. This simple oscillator will produce better than 40 db erasure at normal tape saturation. The bias voltage is a factor in the response and distortion of the recorded tape and should be rather carefully determined. This can be varied by choice of the value of C_{18}. The voltage should be at least that which produces maximum level when a 1 kc. signal is recorded on the tape. Professional machines obtain less distortion by using higher bias levels, as great as twice this value.

Low-frequency equalization characteristics of the playback amplifier (A) and high-frequency equalization characteristics of the recording amplifier (B). See discussion in text.



(A) A comparison of head characteristics and over-all equalization curves. (B) The resultant over-all record-playback transfer curve of "professional-type" recorder amplifier.



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Which tape type are you?

Whatever your tape recording requirements, there's a "SCOTCH" Brand Magnetic Tape to fit your own special needs. A complete line to choose from – six famous tapes – and each gives you the protection of built-in dry lubrication to reduce recorder head wear.

"SCOTCH" Brand guards your machine from harm by impregnating these tape coatings with silicone, the dry lubricant. Continually relubricating the tape's surface, this silicone causes "SCOTCH" Brand Magnetic Tapes to pass smoothly and evenly over your recorder's head. Head wear is appreciably reduced, "wow", "flutter" and "squeal" are almost eliminated. Hear the difference yourself. Try one of these popular tapes on your machine soon.

I WANT WEATHER BALANCE! "SCOTCH" Brand Extra Play Magnetic Tape No. 150 is made for use in extremes of temperature and humidity. Ideal for all-weather outdoor recording. Made with super-tough polyester backing, this tape gives you 50% extra playing time. Super-potent oxide extends 150 Tape's frequency at high end, with more uniform response and crisper tones.

I WANT HIGHER FIDELITY! Boon for the musician and music lover is "SCOTCH" Brand High Output Magnetic Tape No. 120. With this tape you enjoy music with greater dynamic range, freedom from distortion on signal peaks and 133% more output for brilliant sound reproduction. A must for hi-fi fans. Acetate backing.

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I WANT ECONOMY! All-purpose "SCOTCH" Brand Magnetic Tape No. 111 gives you just that — flawless sound reproduction at low cost. Has acetate backing and "SCOTCH" Brand's unique built-in dry lubrication. Recommended for all general recording needs.

I WANT EXTRA PLAYING TIME! Enjoy new freedom from reel change with "SCOTCH" Brand Extra Play Magnetic Tape No. 190. Get 50% more tape on a standard-size reel — as much recording time as 1½ reels of standard tape — plus brilliant sound, thanks to a new, high potency oxide coating.

I WANT SUPER STRENGTH! Tough tapes are "SCOTCH" Brand Magnetic Tape Nos. 111AM and 120AM. Their recording characteristics are similar to Nos. 111 and 120, but these super-strong tapes are coated on weather-balanced 1½ mil polyester backing — the toughest made. Especially suited for irreplaceable recordings.

PICK THE MAGNETIC TAPE BEST FOR YOUR NEEDS

Tape Number and Description	Stability	Strength	Long Wave Length Output	Short Wave Length Output	Recording Time
111 General Purpose	Good	Very Good	Good	Good	Normal
111 AM General Purpose	Best	Best	Best	Good	Normal
120 High Output	Good	Very Good	Best	Good	Normal
120 AM High Output	Best	Best	Best	Good	Normal
190 Extra Play	Good	Good	Good	Best	Extended
150 Extra Play Weather Balanced	Best	Very Good	Good	Best	Extended

The term "SCOTCH" and the plaid design are registered trademarks for Magnetic Tape made in U.S.A. by MINNESOTA MINING AND MFG. CO., St. Paul 6, Minn. Export Sales Office: 99 Park Avenue, New York 16, N. Y. © 3M Co., 1957

3M

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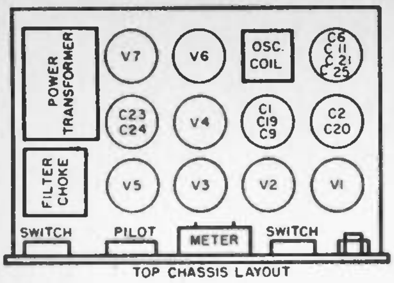


Component layout in top-chassis view.

For the usual adjustment, the erase voltage will be approximately 180 volts at 35-50 kc. oscillator frequency, with the recording bias approximately 100 volts. The final section of the function switch, S_{1F}, applies plate voltage to the oscillator tube when needed for recording.

The third use for the amplifier is as a two-channel mixer for public address applications. This feature was included without extra circuitry by making the function switch a three-position affair. For mixer uses, the mid-position of the switch acts to disable the erase oscillator and record amplifier output, while activating all input connections. Output is taken from the monitor amplifier. The headphone monitor position and level meter are both useful for this application and a signal of over one volt can be furnished to a p.a. power amplifier. The low impedance output of the cathode follower permits separating the mixer and the power amplifier by many feet without frequency attenuation.

Construction

The developmental amplifier, as constructed, is shown in the photographs. In this case, an aluminum chassis, 11 x 7 x 2 inches in size with front panel measuring 13 x 7 inches, proved to be of adequate size. All operating controls were brought out to the panel, with emphasis on a reasonably symmetrical layout. Microphone inputs are also available at the front panel, fitted with *Amphenol* series 75 shorting type connectors. The shorting connectors serve to ground the unused inputs to prevent hum pickup or open grid howls.

The a.c. power switches for the amplifier and the tape transport motor are arranged together with the pilot light and monitor phone jack at the upper left of the panel. The level indicating meter is at the center, with the function switch, equalization switch, and microphone input connectors filling the space to the right on the front panel.

The gain controls are evenly spaced along the lower section of the panel. From right to left, the controls are: mixer gain for channel one, mixer gain for channel two, monitor output gain control, and finally, a playback roll-off control. This latter control is used to reduce high-frequency response when playing noisy tapes, if desired. It consists of a one-megohm tone control, with switch, and a .01 µfd. capacitor in series, connected between grid and ground of the 12AX7 cathode follower stage. The switch is included to remove the control completely from the circuit when not in use. However, the roll-off was found unnecessary in the great majority of cases; consequently, it is not included in the circuit diagram.

The exact tube and component placement on the chassis is not entirely obvious from the photographs, and reference to the figure will clarify this.

This layout proved satisfactory from a wiring standpoint. However, a few precautions might be pointed out to assist those persons who may wish to duplicate the unit. A ground bus is used, with all grounded components going to this bus rather than to the chassis. The bus, in turn, is tied to the chassis at one point only to help prevent ground loop difficulties and hum problems.

The power transformer and filter choke were of the metal encased type and no hum field troubles were encountered from this end. Shielded wire was used for all input leads and those going to the function switch.

The encased power transformer and choke can be approximated in the *UTC* hermetic sealed "H" series. The H-81 transformer and the H-71 choke will be suitable. Where economy is desired, at the expense of some shielding, the *Merit* P-3150 transformer and C-2976 choke will suffice.

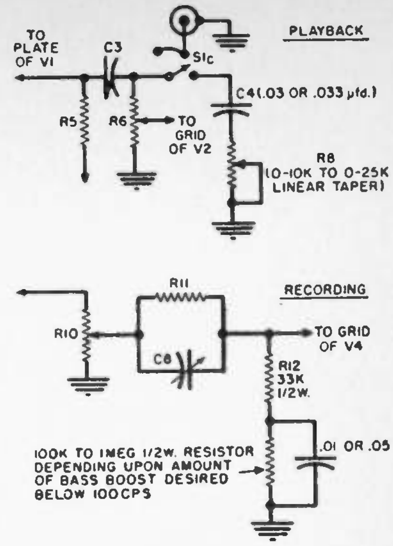
The *J. W. Miller Company,* Los Angeles 3, Calif., makes an excellent oscillator coil for 60 kc. operation, which is catalogued as the No. 1887-A.

Radiation from the erase oscillator might prove troublesome, and it might be wise to assemble this stage in a small "Minibox" or other shielded subassembly which may then be attached to the main chassis.

Input and output leads are dressed close to one another because of the single record-reproduce head. The switching system disables all circuits whenever they are unused and no difficulties were experienced.

The recording level is monitored by a rectifier-type decibel meter. A surplus telephone-type db meter was used and proved satisfactory. It was necessary to build out the meter slightly to reduce its loading effect on the circuit. One of the small inexpensive vu meters currently available for recording would be better suited, particularly since the meter ballistics are adapted for audio work.

Connections for the a.c. as well as the erase and record head are made with small *Jones* plugs at the rear chassis lip. A separate ground lead should be used between the chassis and the motor frame. High level inputs, as well as the output connections, are made with conventional *RCA* phono connectors. The amplifier is mounted in a metal cabinet for protection and, more important, for complete shielding.



The upper diagram shows a modification of the circuit for improved playback equalization. The lower diagram will yield improved recording equalization.

The frequency response of the original design with *Pentron* heads and the original equalization will be within ±2 db from 50 to 10,000 cps, and may be acceptable for many uses. However, in the light of recent trends, the equalization may be made to conform more closely to the NARTB standards by a few minor changes.

Some boost in the extremely low bass is added below 100 cycles *during recording* in the NARTB technique. This can be accomplished by adding an *RC* boost circuit between the low end of R_{12} and ground. This is shown as the .01 µfd. capacitor and shunting resistor in the schematic diagram.

To correct the playback curve to conform to the NARTB standard, the value of C_4 should be fixed at .03 or .033 µfd. and resistor R_8 replaced with a variable control, with linear taper, range of 0-10,000 or 0-25,000 ohms. This control may be located on the panel as the extreme left control marked, in the original front panel photo, "Playback Roll-off." This will replace the original roll-off control which was not found necessary, as explained in the article.

This playback equalization circuit will provide better adjustment of the curve slope from 100 to 10,000 cycles and, in particular, permits adjusting the high end to closer conformity with the NARTB curve. The NARTB standard shows the boost at 50 cycles to be about 23 db above the level at 1 kc., and 10 kc. to be about 10 db down referred to 1 kc. This is a total slope change of about 33 db over this range. With R_8 made variable, the over-all slope varies from about 24 to 35 db change over this region.

The cathode boost and adjustable C_8-R_{11} network will permit adequate adjustments to the high-frequency recording curve to match the new standard.

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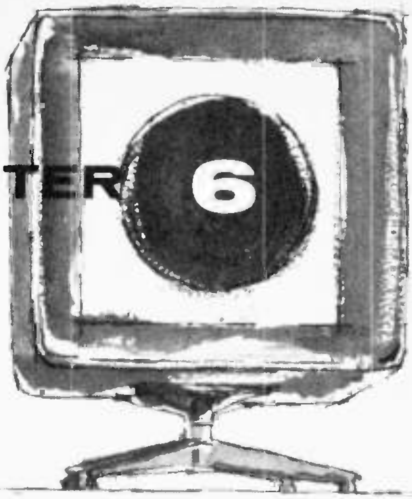

CHAPTER 6

STEREOPHONIC SOUND

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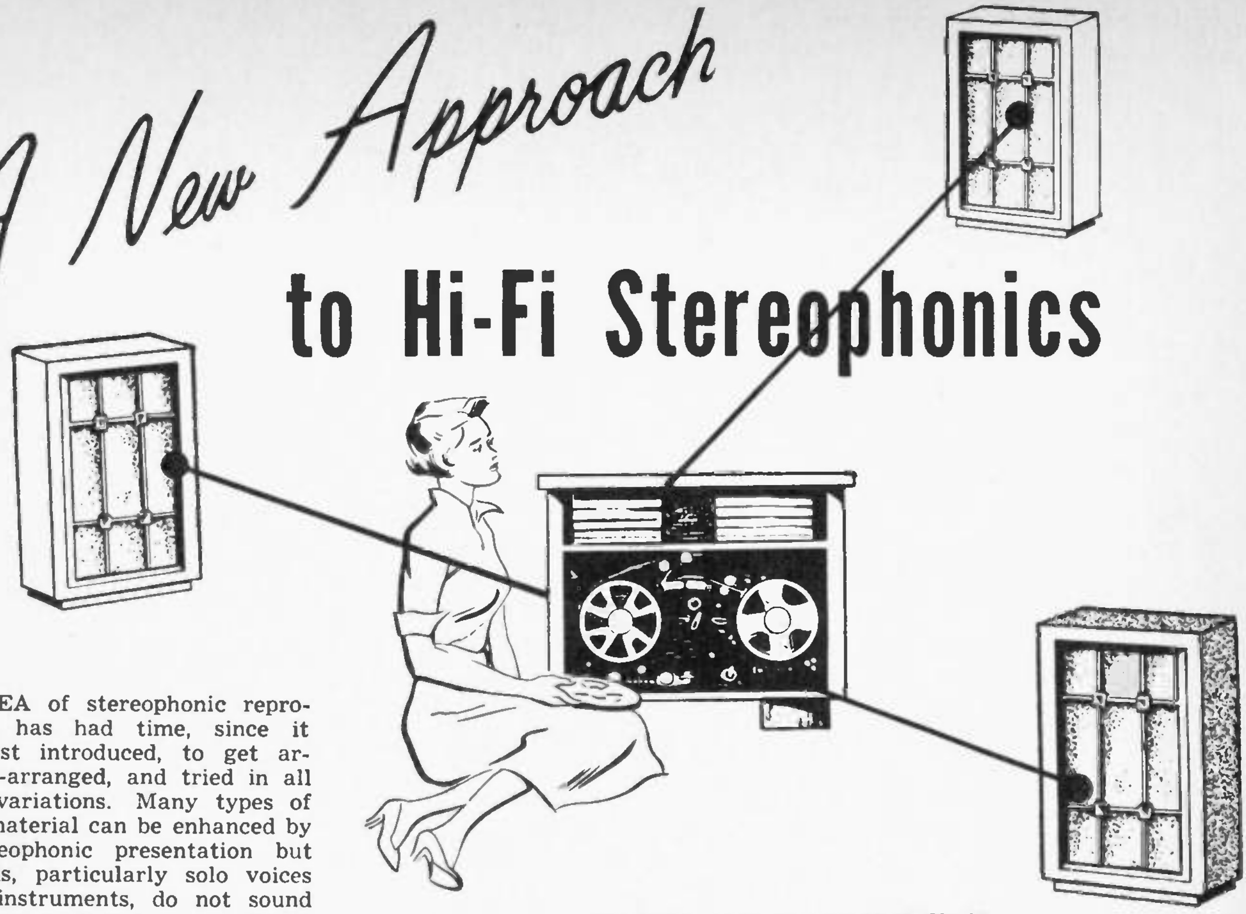
A New Approach to Hi-Fi Stereophonics



By **NORMAN H. CROWHURST**

Details on a new system for recording and reproducing tape which provides a spatial effect by filtering.

THE IDEA of stereophonic reproduction has had time, since it was first introduced, to get arranged, re-arranged, and tried in all kinds of variations. Many types of program material can be enhanced by good stereophonic presentation but some kinds, particularly solo voices and solo instruments, do not sound quite natural in this medium. So let us consider what we really want a stereophonic system to do for us and how various approaches to the problem achieve or fall short of the objectives.

What Should Stereo Do?

The arguments leading to the introduction of stereophonic sound are well known by now: the fact that we have two-ear or binaural perception whereas a microphone or loudspeaker system using only one channel can only convey monaural perception. In making listening comparisons between live program material and reproduced sound, a big difference can be noted between the original sound in live program material and the reverberant sound in the building. However, as soon as the same program material is recorded and reproduced, this difference disappears and the reverberant sound becomes confused with the original sound.

The fact that our binaural perception enables us to detect separately the direction of the original sound and the direction of the reverberant echo in the original production is quickly understood. The problem is to overcome this deficiency in reproduced sound. Another thing, related to this, that we expect stereophonic sound to do for us, is to enable us to pinpoint solo sources of sound, such as a singer's voice or a solo violin, and at the same time to appreciate a widespread source of sound, such as a full orchestra.

Some even claim that stereophonic sound should enable us to pinpoint the individual instruments in a full orchestra. People who expect this of stereophonic reproduction should try sitting in an average concert hall and attempting the same thing with a live performance. They will find that this particular requirement is not as simple as it seems. You certainly get an impression of a widespread source of sound but, while the whole orchestra is playing, it is difficult to locate the precise position of the violins and wind instruments by ear alone. This being the case, it seems somewhat unreasonable to expect a stereophonic system to do for us what the live performance fails to do.

Two methods of attacking this problem are already well known, the stereophonic system and the binaural system. In the binaural system a dummy head, with two microphones in the positions normally occupied by human ears, is placed in the auditorium at a position which is intended to represent the listener's head. The program material picked up by these two microphones is then recorded on separate channels and, in due course, is played to the listener who hears the sound in headphones connected to these separate channels.

Limitations of Binaural

The limitations of this system are obvious. In the first place listening with headphones is quite unlike listening with one's ears free, whatever the quality of the headphone reproduction, and quite apart from the discomfort of wearing headphones for long periods of time. It is true that a remarkable sense of direction is achieved this way because the phasing of sound presented to the two ears is identical with the phasing and intensity of sound received at the microphones in the dummy's head. So the impression of direction, and various other effects such as reverberation coming from different walls of the building, are accurately conveyed.

A further limitation exists in the fact that the dummy's head remains still during the recording, whereas the listener may wish to turn his head slightly, as he would during a live performance, to concentrate his at-

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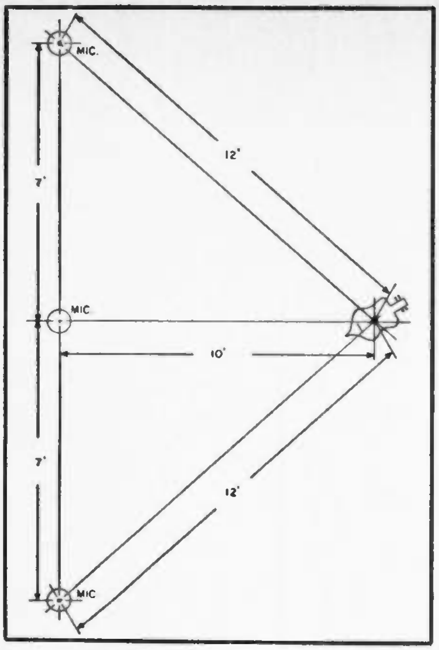


Fig. 1. How multi-microphone pickup of a solo instrument produces an unnatural effect. The side microphone, being only a little farther from the source, will pick up almost the same level. When the effect of studio reverberation is taken into account, the difference in level is negligible.

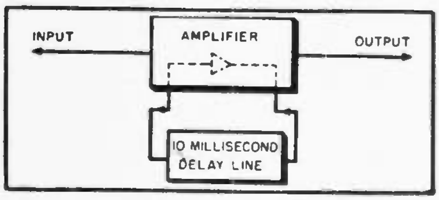


Fig. 2. Block diagram of electronic "artificial reverberation" that has been claimed to produce the same effect as stereophonic reproduction with a single channel. See article for a discussion of technique.

tention on a particular instrument or a particular source of sound. When he turns his head, of course, the dummy does not, and so the sound field rotates with his head instead of remaining as it was, which is quite an unnatural experience.

Broadly speaking, the stereophonic approach appears to have more practical potentialities although it may lack some of the capability of realism possessed by binaural approach. The stereophonic approach aims at recreating a directional sound field so the listener is free to move his head or occupy different positions just the same as he was in the original production. The weaknesses of the stereophonic system appear in the problems imposed to recreate the original sound field.

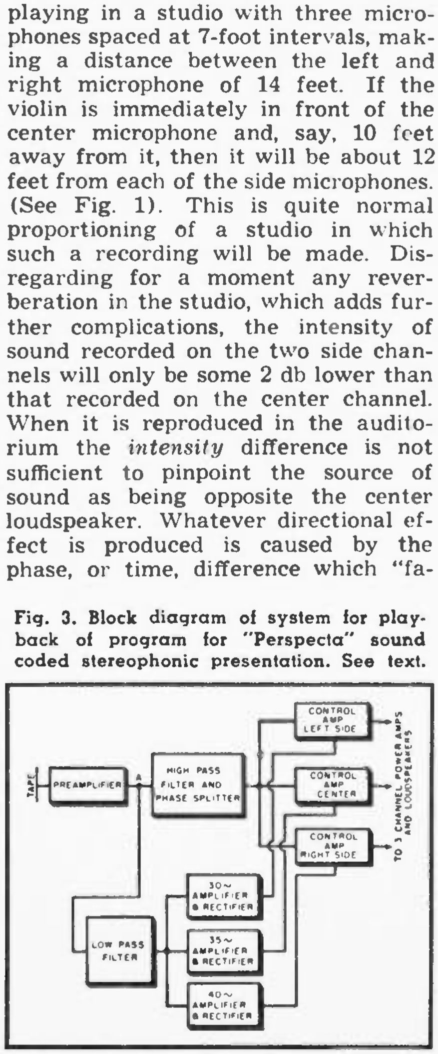
The classic presentation of the idea represents the recording studio with a line of microphones all along one wall, each microphone picking up sound at that position on the wall and recording it on a separate channel. The playback end then reproduces through a line of loudspeakers on a corresponding wall of the auditorium and the theory is that the sound wave incident at the wall of the studio will be reproduced at the corresponding wall of the auditorium. Having made this assumption, which is not quite correct as we shall see in a moment, the theory then goes on to say that we don't need a large number because, in fact, a smaller number produces quite a good approximation. The usual number is three. The latter fact has been experimentally verified.

Limitations of Classic Stereo

The fallacy in this multiple channel approach rests in the fact that the incident sound wave at each microphone possesses directionality, but the microphone must be either omnidirectional or directional. Whichever is used, it cannot be sensible to the *direction* in which the incident sound wave strikes. Similarly the corresponding loudspeaker on the wall of the auditorium cannot reproduce the wave with directional discrimination. This means that the directional properties of the original sound waves are not reproduced, only the relative phase differences at selected points along the wall. Pursuing this argument, it is readily seen that complete reproduction of the directional characteristics of the original sound field is quite impossible. This is why stereophonic reproduction fails to satisfy on some kinds of program material.

Suppose, for example, a violin is playing in a studio with three microphones spaced at 7-foot intervals, making a distance between the left and right microphone of 14 feet. If the violin is immediately in front of the center microphone and, say, 10 feet away from it, then it will be about 12 feet from each of the side microphones. (See Fig. 1). This is quite normal proportioning of a studio in which such a recording will be made. Disregarding for a moment any reverberation in the studio, which adds further complications, the intensity of sound recorded on the two side channels will only be some 2 db lower than that recorded on the center channel. When it is reproduced in the auditorium the *intensity* difference is not sufficient to pinpoint the source of sound as being opposite the center loudspeaker. Whatever directional effect is produced is caused by the phase, or time, difference which "favors" the center speaker. But despite this, the unnatural similarity in level from three sources gives the impression of a larger-than-normal violin.

Fig. 3. Block diagram of system for playback of program for "Perspecta" sound coded stereophonic presentation. See text.



An expanded solo human voice sounds even more unrealistic and it is this property of the conventional stereophonic presentation that seriously mars its realism.

It has been shown that the apparent *breadth* of stereophonic presentation is due principally to the presence of multiple sound sources. On more than one occasion, during a demonstration of stereophonic sound, the loudspeakers have been changed over so as to operate from a single channel and on at least one occasion the demonstrator was oblivious to this fact! He went on discussing the merits of reproducing sound in its correct intensity and phase relationships, quite unaware that each loudspeaker was, in effect, reproducing the same intensity and phase relation.

It should be noted, however, that changing from a single channel presented over three speakers, to proper stereophonic over the same speakers, makes an audible difference and that the change is undoubtedly an improvement. It gives an illusion that more closely resembles the original, especially in its directional effects.

Past experience has taught us that much of our appreciation is psychological: the first radio transmission was "almost lifelike", because we *could* recognize the voice. When something better came along, we realized the deficiency of what we had hitherto thought good. That story has been repeated over and over. But meanwhile, our ears do a very good job of overlooking the deficiencies!

Besides comparing single channel with stereophonic, over the same speaker grouping, some have experimented with dividing the frequency spectrum and delivering either different parts of the spectrum or different response characteristics to each speaker. This too can improve realism *in some instances*. The snag with this idea is that an arrangement that improves the presentation of one segment of the program material is detrimental to another.

Besides pointing up the weakness of multiple channel reproduction, the comparisons just discussed also reveal a possibility for improvement in two directions—cost and realism. With regard to cost, since a single channel reproduced over multiple sources *can* sound almost as realistic as separate channels reproduced over the same loudspeakers, there appears to be little point in using multiple channel recording.

How, then, shall we improve realism? The principal thing we want stereophonic sound to do is to discriminate between different kinds of sound and sound coming from different directions. With the aid of multiple loudspeakers, I suggest—and I am aware that this will sound like heresy—this can best be achieved by varying the

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combined intensity of the different units used.

For a solo, for instance, it is obviously desirable to reproduce the point source of sound over one loudspeaker suitably placed to represent the sound. For a full orchestra with its full-bodied wide area source, the three loudspeakers reproducing the same channel simultaneously will give a very good impression of realism. More of this idea in a moment.

"Single Channel" Stereo

There is one more system that we should discuss. Various methods are adopted to add artificial reverberation to a single-channel presentation. The latest recurrence of this idea appears in a system that re-inserts the same program material electronically after introducing a time delay on the order of 10 milliseconds (see Fig. 2). The purpose of the 10-millisecond delay is to introduce this much equivalent reverberation electronically. The resulting sound, which is then reproduced over a single-channel loudspeaker system, is alleged to possess the characteristics of this much reverberation, added to the original program material and the claim is made that the reproduction is much more "alive".

This reminds one of various claims made for microphones and loudspeakers which have a peaky response. Usually such products are advertised as adding a "crispness" or "liveness" to the reproduction. This is a euphemism, calling attention to the emphasis placed on the particular peak frequencies. It is quite true that such reproduction does add character to the original program material. Sometimes by careful selection of program material this added character may appear to improve the reproduction, but more often it will spoil it by producing a repeated resonance that can become annoying. The result depends, to some extent, on the spectral distribution of sound in the program material.

Our "electronic reverberation system" does practically the same thing —only more so. To see just why, we need to appreciate the difference between the electronic remixing and the acoustic intermingling of reverberation that takes place in a room: it's all a matter of phase. The point to realize is that the question of phase involved in acoustics *is not only a time difference, it is also a space difference*. The reverberant components of the wave are traveling in different directions, to which our ears are sensitive, at least in the original performance. On the other hand, using the electronic mixing it is only possible to include the effect of *time* difference in the phase of the recombined signals. No account can be taken of direction.

This means that, assuming our time delay is exactly 10 milliseconds, or one hundredth of a second, frequencies having a period in exact multiples of one hundredth of a second will become additive and produce peaks, while frequencies which are odd half multiples of this interval will partially cancel due to the fact that the phase relationship is opposing. If the two signals are recombined at exactly the same level, frequencies at these latter points would disappear entirely, but by remixing at slightly different levels the response will become wavy as shown in Fig. 7.

The addition of this electronic stereophonic device will modify the frequency response from its original level characteristic to the wavy or peaky one shown in Fig. 7, which naturally will sound more "live," in the same way that microphones and loudspeakers having a peaky response are said to be more "live." This is certainly no approach to true stereophonic reproduction. We must use a system capable of giving a phase difference in *space*.

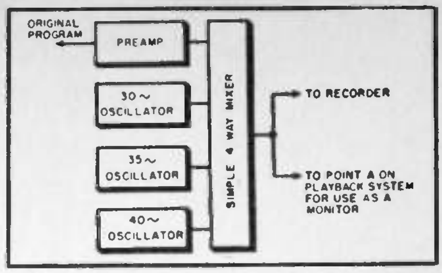


Fig. 4. Equipment, in addition to that of Fig. 3, required for preparing "Perspecta" presentation from original program material.

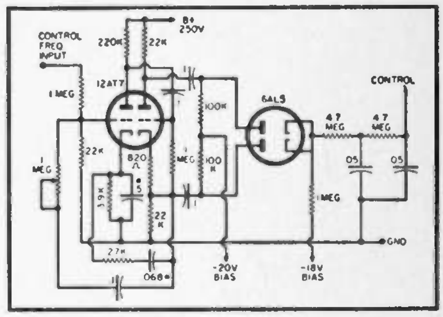


Fig. 5. Control frequency amplifiers (one required for each frequency). The starred capacitors are trimmed to tune the amplifier to the required frequency, 30, 35, and 40 cycles. The 1 megohm resistor is used to adjust the gain of the amplifier. If it is turned too far, the amplifier will oscillate at the control frequency for which it is set. Control frequency inputs parallel on the control frequency output of Fig. 10. Control connects to control stage.

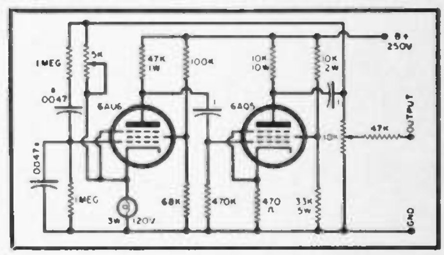


Fig. 6. Control frequency oscillator for use in system of Fig. 4. Starred capacitors are trimmed to give required frequency for each channel. The 5000 ohm variable resistor is set to give minimum harmonic (very important) by turning up until oscillation commences, then backing off, until minimum condition of stable oscillation is found. The 10,000 ohm pot can be used as a control for recording the control frequencies in preparing the composite program material. Refer to article.

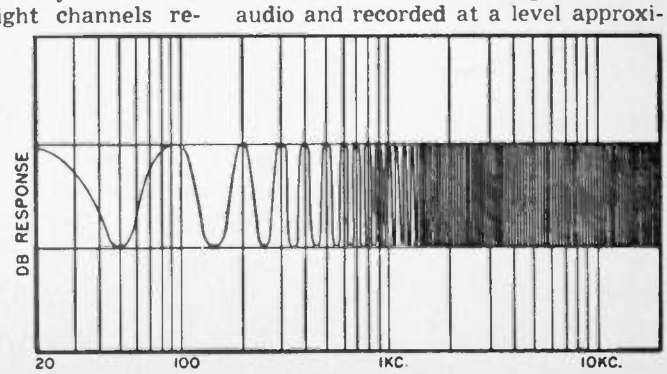
Coded Stereophonic

Utilizing a completely different principle, which achieves a synthetic stereophonic sound from single channel, is a system developed for the motion picture industry under the tradename *Perspecta* sound. This system utilizes a fact, mentioned a little earlier in this article, to produce what seems to promise the best approach to stereophonic realism thus far achieved. With *Perspecta* sound, the channels not in use when a solo performance is being presented, are reduced to a level sufficiently below the one being used not to be heard at all. On other occasions the single-channel sound can be applied to the individual loudspeaker systems in varying combinations of intensity so as to achieve any effect desired.

At this point, many readers will probably react the way I did when the system was first described to me: "This is a 'poor man's' stereophonic, enabling economy in recording, but it cannot possibly sound as good." That it can and does produce more realistic sound than the so-called "true" stereophonic is something that has to be heard before it can be believed.

The system, as used for motion pictures, employs three loudspeaker systems positioned in the usual manner behind the motion picture screen. Distribution of the single-channel audio is controlled by three subaudible control frequencies. In the case of the *Perspecta* sound system, the frequencies are 30, 35, and 40 cycles for the left, center, and right channels respectively. These control frequencies are mixed with the single channel audio and recorded at a level approxi-

Fig. 7. The reason why the arrangement of Fig. 3 sounds more "live". This is the kind of frequency response it produces. The relative db values will depend on the levels at which delayed signal is mixed with the undelayed signal.



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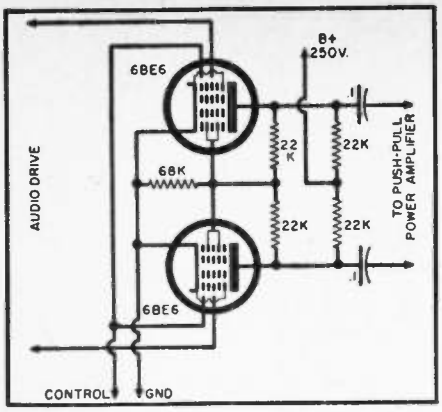


Fig. 8. Control stage (3 required). Audio drive is taken from the circuit of Fig. 10 the three units being paralleled at this point. Control comes from the appropriate circuit of Fig. 5. Refer to the article.

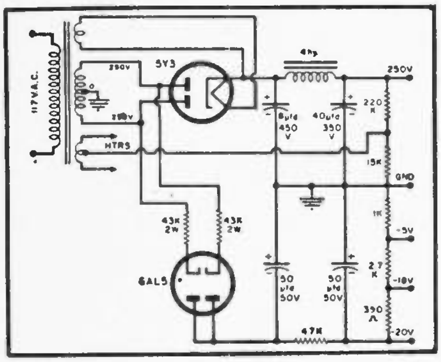


Fig. 9. Power supply circuit to provide all operating voltages needed for Figs. 5, 6, 8, 10.

mately 20 db below the peak level of the audio. This ensures that, if the same sound track is reproduced over a normal theater system, the control frequencies are inaudible. In the *Perspecta* reproducer, however, filters remove these control frequencies and ensure that they do not get into the reproducing chain at all.

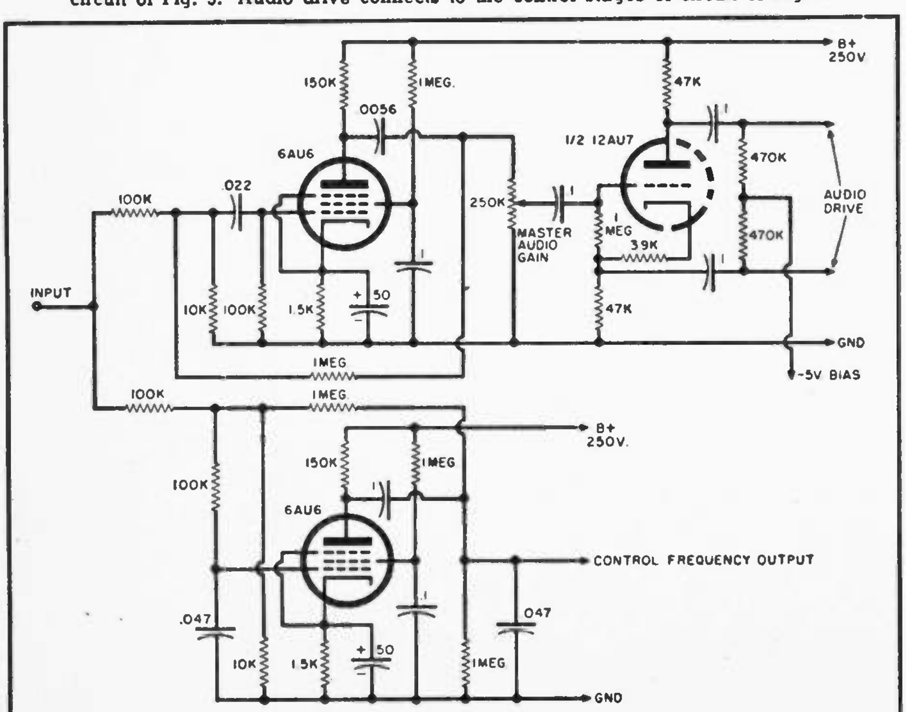
Fig. 3 is a block diagram of the system used for playback, while Fig. 4 shows the additional features necessary for making the original recording. The control frequencies are separated, amplified, rectified, and then used to control variable-gain stages feeding the three separate channel loudspeakers. This means that only three separate power amplifier and loudspeaker systems are necessary—not three complete amplifier chains, so there is a considerable saving in cost, especially the cost of recorded material, because the whole recording is made on a single channel.

In the system used for the motion picture business the bandpass channels provided for control frequencies are fairly complicated in design, because of the necessity for accommodating variations in the control frequency. This is necessary because of differences in speed at which the projector may run, because of power line variations throughout the country, which may upset the intended synchronous running speed.

A Home Version

In order that you can hear it for yourself, and try making your own recordings, data is given so the reader can make up the essential electronic parts of the system. The complicated filtering provided for theater use is unnecessary, because your own tape recorder will probably run at a consistent speed. This means that a simpler circuit arrangement can be used for the elements making up the system and Figs. 5, 6, 8, 9, and 10 show schematics for the various blocks in the diagrams of Figs. 3 and 4. From this information any reader should be able to put together a system of his own.

Fig. 10. Low- and high-pass filters and phase splitter for the circuits of Figs. 3 and 4. The composite program is connected to the terminal marked "input." Control frequency output connects to the three control frequency inputs on the circuit of Fig. 5. Audio drive connects to the control stages of circuit of Fig. 8.



The procedure will then consist of first making a recording of the program material that you wish to arrange for this presentation. If desired, material already recorded on discs can be used, the program being taken directly from the disc, or alternatively an FM broadcast can be recorded on tape, and later re-recorded with these control frequencies on. Whichever program source is used, it should first be obtained in some recorded form so that it can be played through and notations made as to the desired effects. After the sound has been played through a time or two on a single channel and you have made up your mind as to what changes you want to add with the *Perspecta* system, you are now ready to record in the control tracks on your tape recorder.

For this kind of presentation the general method is as follows: adjust the control frequencies so as to *anticipate* any particular feature of the program. For example, if a solo singer or solo instrument is going to come in, decide which loudspeaker you will use to feature this solo and bring this one to full intensity by fading up its control frequency so that the solo appears to be coming from this point as a center. If there is background music, you adjust the control frequency level for the remaining two speakers to operate at a little lower level than the main speaker, which gives a general spread but concentrates the attention on the loudspeaker with the soloist. It is surprising how realistic this kind of presentation can be. Then, when a full orchestra comes in, all loudspeakers are operated at equal level.

The extraordinary thing, until you think a little more about the principles of stereophonic sound, is that the change caused by adjusting the distribution of sound prior to the solo part, drum break, violin, or whatever it may be, is not noticed until the solo actually begins. The fact that the general distribution of the sound changes around does not impress itself. Nor is one readily conscious of the fact that the dominant background music also comes from the same loudspeaker as the solo, unless the other loudspeakers are cut out entirely, so as to give the effect of single channel once again.

The reason behind the improved sense of realism that can be achieved by this method, which at first strikes one as being completely contrary to true principles, is to be found in the fact that the sense of direction, which our hearing can give to sounds, is based principally upon *transient* sounds.

In point of fact, the best demonstrations of so-called true stereophonic have been faked to do the same thing. The gain in the separate channels is "panned" to emphasize the desired ef-

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fect and the impression received is not just due to simple simultaneous three-channel communication.

If you carefully analyze your sense of direction, you will find that it is extremely difficult to tell the direction of a continuous tone, whether it be organ music, orchestral music at full volume, or an oscillator, when reproduced over a loudspeaker. Any form of continuous tone, or group of continuous tones reproduced in music, sets up a whole pattern of standing waves in the room, as shown in Fig. 11. This means that the intensity pattern throughout the room is extremely complicated, and you will find that, if you move your head in a room where a loudspeaker is reproducing a tone from an oscillator, the apparent direction of the sound source changes quite quickly with a small head movement. It is difficult to identify the sound with the loudspeaker emitting it.

However, as soon as you put some form of more live program material through the same loudspeaker you are

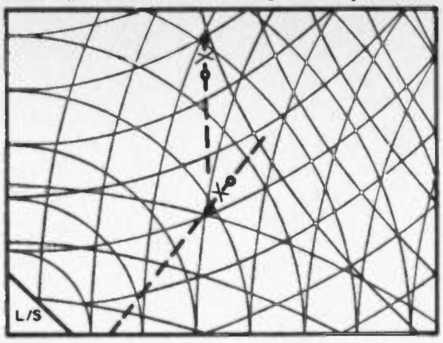

Fig. 11. Standing wave pattern set up by single tone reproduced from a loudspeaker in a room. At the positions marked "X," the dotted lines indicate the apparent direction of source which can be adjudged by the points of maximum and minimum instantaneous pressure, indicated by solid and open circles. See the article.

much more conscious of its position. You will find, by careful analysis, that this is due to the transient nature of the program material. It is with the sharp wavefronts that you identify directions and not the continuous tones which follow them. This fact is what enables the *Perspecta* principle to operate so effectively to produce what proves to be, in fact, a more satisfactory illusion of stereophonic sound than the classic method employing three or more channels.

It would seem likely that this system, or one using a similar principle, will ultimately be accepted as providing the best and most economical approach to realism in reproduced sound. As it will be some time before such a system is developed commercially and recorded material is available, doubtless some readers will want to try it for themselves, following the method outlined. —30—

Answers to readers' questions about the "Perspecta" coded stereo system

AFTER the previous article appeared in RADIO & TV NEWS, we received a steady flow of letters with questions about the various aspects of the coded stereo system.

Some readers wrote in asking for complete details, wanting chassis size, layouts, details of components, parts lists, blueprints, etc., for the complete system. Others have wanted to get details about the theater system that was mentioned in the article.

The theater system has been quite extensively used and whatever data is available for release to anyone outside (or inside) the motion picture industry, can be obtained by applying to *Perspecta Sound, Inc.*, 711 5th Ave., New York, 22, N. Y.

At the outset it should be made clear that the system, some details of which were suggested in the previous article, had not been built. The theater system works perfectly well, using the same principles. Individual parts for the system, schematics for which were given in the previous article had been built, sometimes for other purposes. But the entire system had not been put together as a coded stereo system. The data put forward was given for the benefit of people who want to experiment and thus help forward the development of such a system, either for their own interest or as a contribution toward ultimate progress for all of us.

Question 1. How do you make sure none of the low audio frequencies operate the control, and that the control frequencies are not heard?

This question arises out of calling the control frequencies "sub-audio." The "sub-" here is used in two senses: it means the control frequencies are lower in *frequency* than the band normally used for amplifying audio, at least in this system; and also that they are below the *level* that will allow them to be audible. Additionally, high-pass and low-pass filters are used in the system to separate the frequencies, so as to insure that not enough residual audio gets into the control circuits to upset their action, on the one hand, and that the control frequencies do not get through into the audio channels and produce an audible tone on the other hand.

In the theater system, the control frequencies of 30, 35, and 40 cycles are at least 20 to 26 decibels below the maximum audio level. This, however, is to allow for the possibility of a *Perspecta* track being played in a theater not equipped for this kind of sound, and which might therefore reproduce these frequencies at their recorded intensity. This margin proves adequate to insure the control frequencies are inaudible, even without filtering.

But when filtering is provided to eliminate the control frequencies from the audio, they can be operated at a somewhat higher level without risk of this kind of interference.

Some are probably going to object here that 30, 35, and 40 cycles are definitely within the audio range, so *how can they be called sub-audio?*

In application to theater systems it was found that most theaters failed to reproduce anything appreciably below 70 cycles. Consequently, nothing is lost by introducing a high-pass filter that cuts off everything *below* 70 cycles, so we can use frequencies down there for control purposes.

In the living room we have a further factor to help us in the same direction Unless your living room is at least a 50 foot cube (which is rather large for a living room!) you cannot possibly hope to develop even one complete wave of 20 cycle frequency in the room. It is true such a frequency might be audible *if* reproduced, but the theory about reproducing it is very different from practice.

Despite the stress that has been laid on requiring response down to 20 cycles, satisfactory fidelity for the living room can be achieved easily by reproducing down to 50 cycles. This only requires a living room with the dimension of a 20 foot cube to produce one complete wave. Most living rooms are even smaller than this!

Question 2. How do you set up the control frequency oscillators and tune the control frequency amplifiers?

The circuits originally given showed the components which set the frequency marked with an asterisk. These have to be adjusted by trying different values until the frequency comes out right.

The first thing to do is to set up the control frequency oscillators. The frequency for 30 cycles can be found by means of an oscilloscope. If you don't have an oscilloscope it would be good to arrange to borrow one for an evening, or some suitable time, from your local service shop. Many service shops have an oscilloscope which they seldom use and will be glad to lend if they are satisfied you know how to use it.

Getting a 30 cycle frequency is very easy with an oscilloscope, using the 2 to 1 Lissajous pattern. Forty cycles gives a 3 to 2 pattern. This is using line frequency as a time base. Almost all modern oscilloscopes have a mutiple switch for selecting horizontal deflection: the internal time base, the line time base, or an external connection. Using the "line" position of this switch, and applying the output from your control frequency oscillator to the normal vertical scope input, will produce the Lissajous patterns shown in Fig. 1 when the frequencies are correct.

Adjusting the values of the capacitors will alter the rate at which the pattern moves until you can get a

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steady or stationary pattern. Then you have the right frequency.

Having set up the 30 cycle and 40 cycle control frequency oscillators, the next step is to set up the corresponding control frequency amplifiers. This can be achieved by a similar method. Turning the gain adjustment up to the point where they oscillate, the frequency is adjusted in the same manner. Then the gain adjustment is turned down until they just do not oscillate. Finally set the gain of each control frequency amplifier so that the same input from the corresponding control frequency oscillator gives you the same output voltage from each.

Now you have a tool for setting up the 35 cycle frequency, which is not quite as easy as the 30 and 40 cycles, because there is not a convenient Lissajous pattern. Adjust the frequency of the 35 cycle oscillator until it gives an equal output from both the 30 cycle and 40 cycle control frequency amplifiers, showing there is equal leak-through each way. (This output should be considerably less than when the correct frequency of 30 or 40 cycles is applied to its own amplifier, of course.)

Finally you can set up the 35 cycle control frequency amplifier by aligning it until it gives a maximum output on the 35 cycle oscillator.

This is quite a convenient way of setting up the frequencies without needing to have "reference standards," which is the method used for the motion picture system.

Question 3. How about using more than three channels to get greater realism, say 4 or 5? What control frequencies should I use?

If you want to extend to, say, five channels, the best frequencies to choose would be down to 20 and 25 cycles. If you want six you can also go up to 45 cycles. The 45 cycle Lissajou pattern, that would be used on the oscilloscope to align the frequency, is fairly simple, as shown in Fig. 2. The 20 cycle pattern is easy, but 25 cycles is not. The author suggests making the 20 cycle oscillator and amplifier first and then fitting the 25 in as a midway point between 20 and 30, in the same way as 35 was fitted in between 30 and 40.

Using a combination ranging from 20 to 40 cycles would have the advantage that would not encroach any more on the audio frequency range than the three channel system. As you might run into difficulties in satisfactory recording of 20 cycles, it may be a good thing to use this channel for the position of the loudspeaker that you consider least important.

Question 4. Is it possible to add electronic reverberation effect with this system?

This question touches on the vast potentialities of a system of this type. As originally set up, this coded system just produces an equivalent of three-channel stereo. But it would be equally possible to utilize additional control channels for the purpose of adding artificial reverberation, in variable quantity, to suit the program. This opens a new vista of versatility in reproduction. Possibly, if you use five channels, the extra two, utilizing maybe the frequencies 20 and 25 cycles for control purposes, could be used to control reverberation distributed by two more speakers in a position behind the listener from the main three speakers.

This reverberation could use a second pickup. In the case of tape, this is quite easy to do, by putting a second pickup head so the tape passes it a fraction of a second later than the existing one.

The output from the second head could be delivered to another preamplifier and the control frequencies picked off by the first head applied to control the output from the second preamplifier feeding the reverberation loudspeakers. A block diagram of this idea is shown in Fig. 3.

Question 5. Can the control frequency oscillators and amplifiers be made so the frequency can be adjusted with a trimmer, instead of having to be set by a trial and error method?

Trimmer capacitors only come in values up to about 2000 micromicrofarads (.002 microfarad), but the tuning capacitors for these circuits are in the region of .1 microfarad. So any of the available trimmer capacitors would be totally inadequate as a means of frequency adjustment.

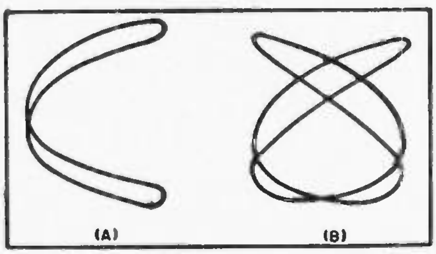


Fig. 1. (A) shows a 30-cycle Lissajous figure. (B) shows a 40-cycle pattern.

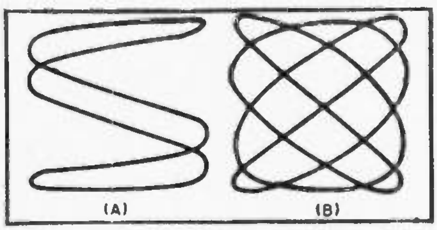
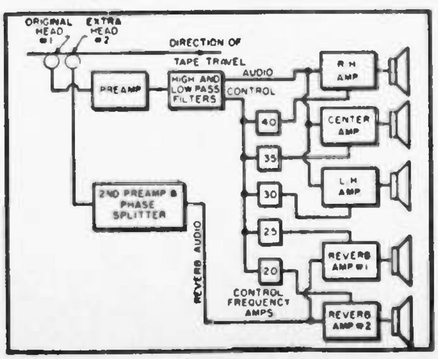


Fig. 2. (A) shows a 20-cycle Lissajous figure. (B) shows a 45-cycle pattern.

Fig. 3. Block diagram showing method employed to introduce reverberation.



The only way to overcome this would be to use one of the phase shift oscillators where the frequency is adjusted by varying one or more of the components in the circuit. I have developed a simple circuit for this purpose which gives quite good waveform and requires a lower plate current supply than the original circuit. This helps with the design of the power supply for the system.

Question 6. Are there any steps that can be taken to reduce the cost of this system, or to improve its performance?

This question really summarizes what many readers have asked in different ways, and with different specific suggestions. Editorial limitations prevent my going into any detail to answer these here. The high-voltage supply, as given in the original article, was only just sufficient for the requirements of the system as described. But several readers are interested in extending the system to include ideas of their own, and then they run into difficulties in getting enough "B+" supply conveniently.

Accordingly, to help this problem, I have gone over most of the sections of the system and redesigned them using different tubes and circuits, making very small plate current demand. As it is not practical to devote the large amount of space that would be required, we are preparing this information so it can be sent out on mimeographed sheets, to give the readers who are interested in pursuing the matter further the information they will need.

It still is not practical to give complete constructional details—chassis layout, drillings, detailed parts lists, and data of that nature, because of the variety of ways in which individual readers want to build this thing. But the data offered in this way will be sufficient for any reader who can work from a schematic, and will include details on setting up the complete system, to make it work, and answers to many questions that have been asked about different aspects of the system, that we do not have space to answer here. As a reader service, we are making this data available for a nominal cost of $1. This should be sent directly to the author at 150-47 14th Rd., Whitestone 57, N. Y.

Some of these changes will reduce cost. But it is still an experimental system. The real possibilities in cost reduction would come if and when it is designed for the consumer market. Then the whole system would be packaged into one unit, like a TV chassis, and the author foresees the *possibility* of this taking place in a unit that will not cost very much more than a single channel system of today. Coded stereo recordings need cost *no* more than the equivalent program time in single channel. But whether this all comes about is not entirely a matter of "engineerability," so we cannot prognosticate in this realm. —30—

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An Improved "3D" Converter

Top chassis view of the simplified "3D" converter. This improved version incorporates a self-contained power supply which will also operate a separate tuner or preamp to be used with this system.

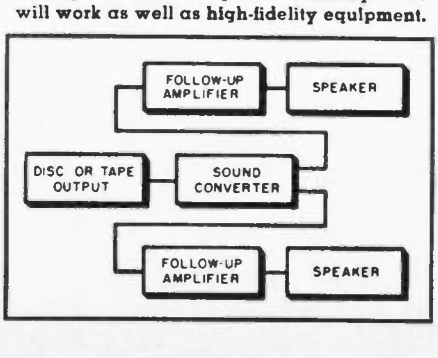
By **JOSEPH CHERNOF**

A simplified and improved circuit

SINCE publication of the author's article, "The Poor Man's 3D Converter," in the November, 1954 issue of Radio & Television News, a number of interesting comments have been received from readers who have constructed and experimented with the unit.

On the basis of this material and other work, a new unit has been developed which has proved to be a considerable improvement over the original. An internal power supply has been added and provisions have been made for utilizing the unit to supply power to a separate preamp or tuner in a custom installation. Separate audio inputs are provided to accommodate both low-and high-level audio signals. The bass predominant channel gain has been reduced to bring it more in line with that of the treble predominant channel. Feedback has been added to the input preamp stages to improve their frequency response and linearity. Low-frequency performance of the unit has been improved by using larger coupling capacitors. Capacitor C_6 has been made larger to provide a smoother crossover for the bass and treble channels.

Fig. 1. Block diagram of the complete "3D" system. Although dual output amplifiers and speakers are required, ordinary units will work as well as high-fidelity equipment.



The original "3D" converter was designed to provide the audio experimenter with a means of capturing the illusion of depth and liveness associated with true stereophonic or binaural sound at a fraction of the cost and complication involved in assembling either one of these two systems.

The "3D" converter attempts to convert ordinary monaural sound, as supplied from phono pickup or tape, to simulated stereophonic sound. This is accomplished by electronically separating the single sound input channel into two separate output channels, sufficiently different in phase and frequency structure from each other to simulate, say, the sound output of a symphonic orchestra as picked up by two separate microphones on a live recording stage.

The two sound output channels from the converter are followed by two separate external amplifiers and speaker systems so that recombination of the synthetic stereophonic sound takes place only at the ears of the listener as is the case with true binaural or stereophonic sound.

A block diagram of the complete "3D" system is shown in Fig. 1. While dual output amplifiers and speakers are required, very ordinary units used in this system will give results superior in most respects to a high quality monaural sound system. One possibility would be to use the audio section of a good console radio or TV set for the second sound channel.

The schematic diagram of the improved unit is shown in Fig. 2, with typical operating voltages indicated on it. The first two stages utilize both sections of a 12AX7 twin-triode in a conventional preamp circuit. Ample gain is available for a magnetic cartridge or other low-level signal source. No attempt is made to incorporate any specific means of equalization since a complete "3D" system will have a multitude of tone and volume controls on the various units to accomplish this.

The output of the preamp stages is fed through the master gain control, R_9. High level input signals from a crystal or ceramic cartridge or tuner are fed in at this point. In this connection, the jumper between terminals 2 and 3 of J_1, a 4-terminal *Jones* barrier strip, is removed. From the master gain control, the amplified input signal is coupled to the grids of triode sections V_{2A} and V_{2B} which comprise the bass and treble predominant channels respectively. Each channel has its own gain control.

The phase and frequency structure of the original signal are modified by means of selective feedback in each of the two channel amplifiers to provide the desired stereophonic effects. The separate outputs of V_{2A} and V_{2B} are coupled to individual cathode-fol-

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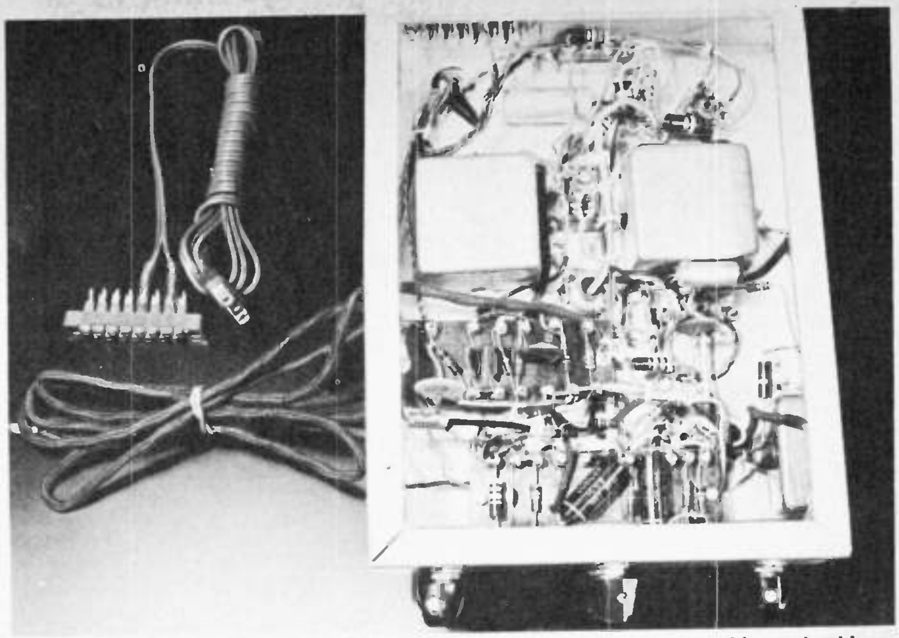
lower triode sections, V_{3A} and V_{3B}, to provide isolation and also to permit the follow-up amplifiers to be located at some distance from the converter.

The power supply is conventional in design, using a 6X4 rectifier and a *Triad* R-7A power transformer. Very heavy filtering and decoupling networks are used. Since the total "B" current drain of the unit is very low, less than 6 ma., filter resistors are used rather than chokes.

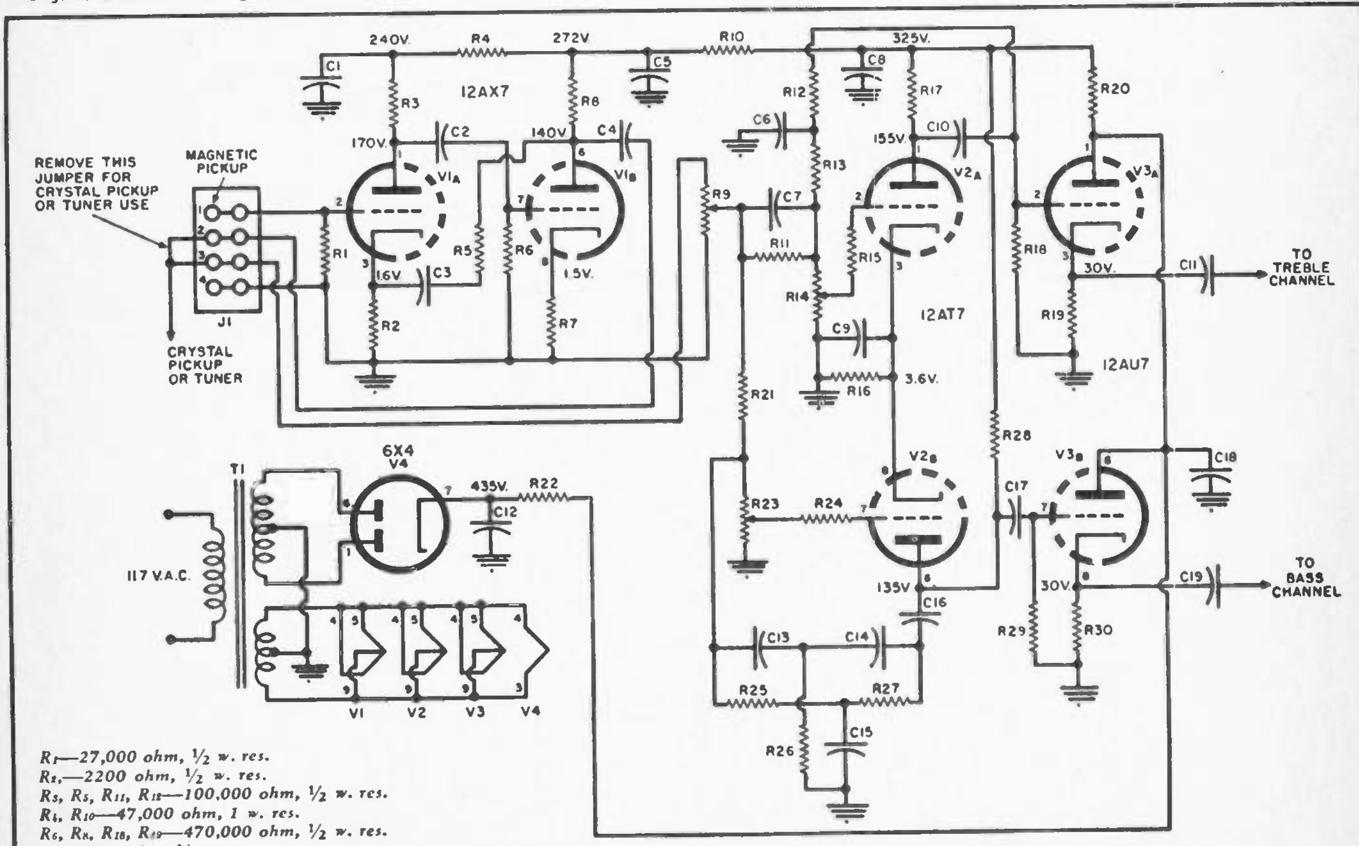
As shown in the accompanying photos, a 15-pin male *Winchester* connector is mounted on the rear of the unit. The a.c. power is supplied through the associated female connector. This arrangement allows the unit to plug into a complete sound assembly, with a common "on-off" switch for several units. It also provides a handy outlet for "B" or filament power to be taken out of the unit to operate an external preamp or tuner as well as a very flexible means for connecting input signals into the unit for a custom installation where all wiring is to be concealed. Conventional parts and construction are used throughout except that disc ceramic capacitors are used instead of tubulars to conserve space.



Under chassis view of unit. Construction is conventional and no problems should be encountered by the builder. Disc ceramic capacitors are used to conserve space.

Readers who have wondered whether or not stereophonic sound is for them can make the experiment with this equipment without investing a fortune in stereophonic head and pickups, dual amplifiers, and double speaker systems. While no claims are made that this "3D" converter is anything but a *simulated* stereophonic system, it will provide a fairly good idea of the *type* of performance that you can expect from a true stereo setup. —30—

Fig. 2. Schematic diagram and parts list covering an improved "3D" converter. Standard, easily obtainable parts are used throughout.



R_1—27,000 ohm, ½ w. res.
R_2—2200 ohm, ½ w. res.
R_3, R_5, R_{11}, R_{12}—100,000 ohm, ½ w. res.
R_4, R_{10}—47,000 ohm, 1 w. res.
R_6, R_8, R_{18}, R_{29}—470,000 ohm, ½ w. res.
R_7—4700 ohm, ½ w. res.
R_9, R_{14}, R_{23}—500,000 ohm audio taper pot
R_{13}, R_{17}, R_{28}—220,000 ohm ½ w. res.
R_{15}, R_{19}, R_{24}, R_{30}—47,000 ohm, ½ w. res.
R_{16}—2000 ohm, ½ w. res.
R_{20}—22,000 ohm, 2 w. res.
R_{21}—680,000 ohm, ½ w. res.
R_{22}—15,000 ohm, 2 w. res.
R_{25}, R_{27}—2.2 megohm, ½ w. res.
R_{26}—10,000 ohm, ½ w. res.
C_1—.5 μfd., 400 v. capacitor
C_2, C_4—.047 μfd., 200 v. capacitor
C_3—.005 μfd., 400 v. disc ceramic capacitor
C_5, C_8—2 μfd., 600 v. bathtub capacitor
C_6, C_{16}—.03 μfd., 400 v. disc ceramic capacitor
C_7—250 μμfd., 400 v. disc ceramic capacitor
C_9—25 μfd., 50 v. bathtub capacitor
C_{10}, C_{11}—.01 μfd., 400 v. disc ceramic capacitor
C_{12}, C_{18}—8 μfd., 450 v. elec. capacitor
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C_{15}—.02 μfd., 400 v. disc ceramic capacitor
C_{17}, C_{19}—.1 μfd., 400 v. tubular capacitor
T_1—Power trans. 300-0-300 v. @ 50 ma.; 5 v. @ 2 amps.; 6.3 v. c.t. @ 2 amps. (5 volt winding not used. Triad R-7A)
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V_1—12AX7 tube
V_2—12AT7 tube
V_3—12AU7 tube
V_4—6X4 tube

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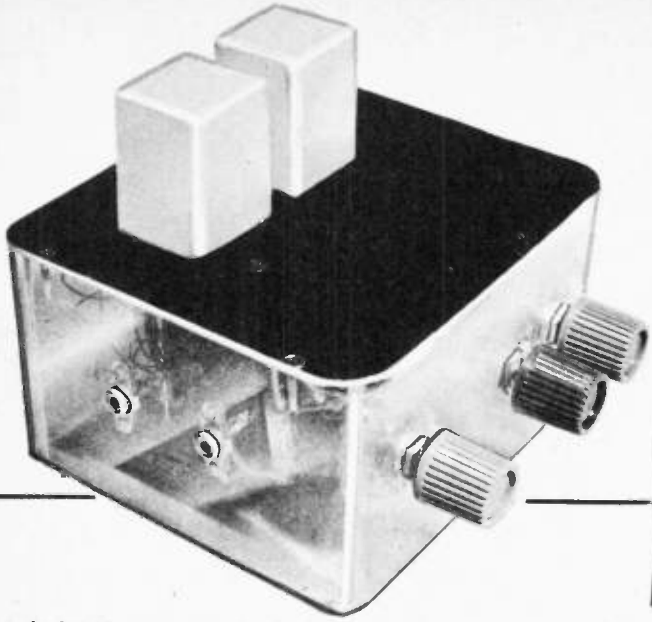

All Transistor Hi-Fi Amplifier

By
HUGH R. LOWRY
Application Specialist
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Completely battery operated, this combination preamp, for magnetic cartridges, and power amplifier will provide quality reproduction. Although the power output is quite low, only about three-quarters of a watt, the unit's output is sufficient for ordinary home listening.

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Recently the *General Electric Company* announced a complete line of low cost audio transistors with very close control on the parameters so that complete interchangeability is possible, even in push-pull output stages which formerly required high priced matched pairs. With these new transistors and present circuit design techniques, it is now practical to build a completely transistorized high-fidelity amplifier of sufficient output for ordinary home listening. The complete amplifier, including magnetic cartridge preamplifier and tone controls, is shown in Fig. 1. No attempt was made toward extreme miniaturization in the author's model, but the compactness typical of transistorized equipment is obvious. The circuit diagram of the complete unit is shown in Fig. 5.

One of the most popular pickups among high-fidelity equipment owners is the *G-E* "variable reluctance" unit and this preamplifier circuit was designed to work with it. To obtain proper high-frequency equalization, the manufacturer recommends that the cartridge be loaded down with 6200 ohms and that 18 db of bass boost be used to compensate for recording characteristics. This combination will compensate for the RIAA characteristic, which is used by virtually all record manufacturers, and be sufficiently close compensation to give acceptable performance with older recordings. In the case of this circuit, proper cartridge loading is accomplished by means of resistor R_1. This 3900-ohm resistor, operating in conjunction with the low input impedance of the first transistor, V_1, provides a circuit with proper response.

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If a wide range of transistor current gains is to be tolerated with stable performance at high temperatures, it is necessary to use the circuit shown in Fig. 3A. In this type of biasing, the emitter current (and hence collector current) is essentially determined only by the voltage at the base and the resistance in series with the emitter. Although this type of biasing is extremely stable, good low-frequency response requires a very large bypass capacitor around the emitter resistor, which adds considerably to the cost and little to the performance. Since this transistor hi-fi amplifier was not intended to be operated under extreme environmental conditions and the new *G-E* transistors have close control on current gain, it is possible to use a simpler form of biasing as shown in Fig. 3B. In this type of biasing the base current is determined by the collector voltage and the resistance between collector and base. This type of biasing is extremely degenerative. For example, consider the situation where the collector current increases, causing a drop in collector voltage. A decrease in collector voltage reduces the current flowing into the base which, in turn, reduces the collector current, thereby counteracting the original increase in collector current.

It might appear at first glance that this type of biasing would not only be degenerative for d.c. drifts, but would also degenerate the signal being amplified. This would be true if the a.c. load impedance was high in comparison with the 18,000 ohm collector resistor.

If the load is the input to another transistor, as is usually the case, the input impedance of the following transistor is low compared to the collector resistance and little a.c. degeneration takes place. Since tubes are voltage-operated devices, the usual method of bass boost is to use voltage feedback from the second plate to first cathode.

The input to the first transistor is almost a current source (low impedance cartridge plus large series resistance), and it is better to use current feedback rather than voltage feedback. A signal proportional to the output current is taken from the small resistor in the emitter of the second lower triode sections, V_{3A} and V_{3B}, to provide isolation and also to permit the follow-up amplifiers to be located at some distance from the converter.

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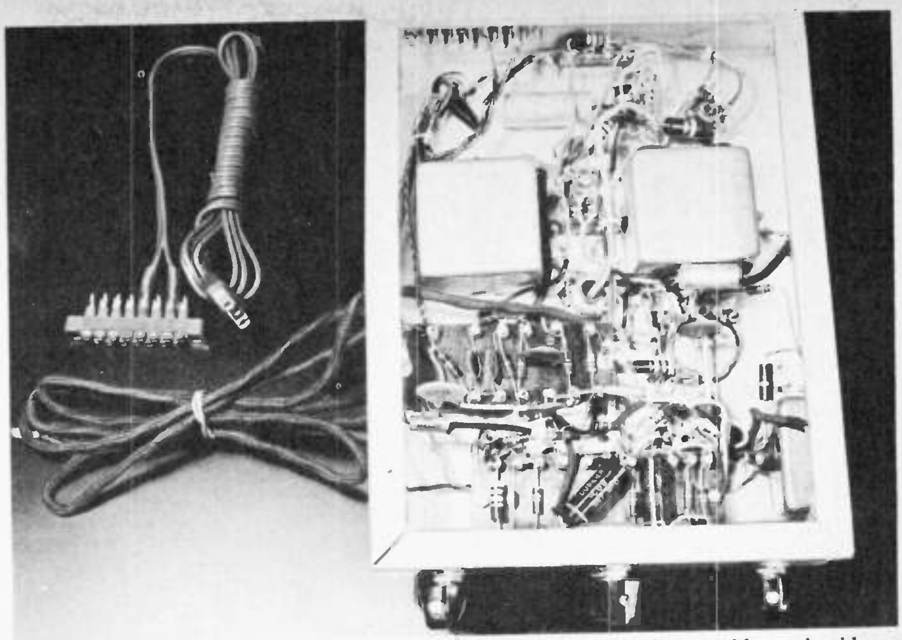

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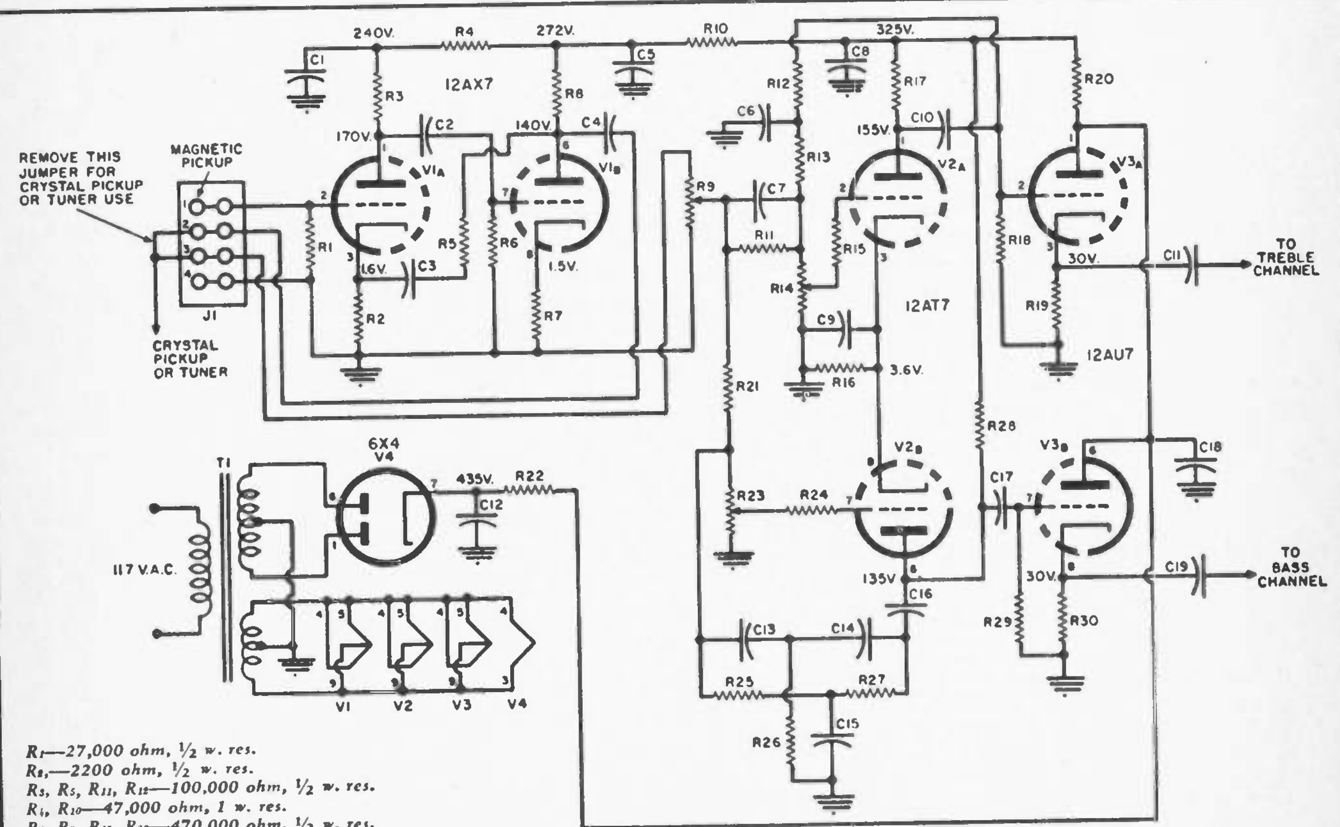

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R_3, R_5, R_{11}, R_{12}—100,000 ohm, ½ w. res.
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R_6, R_8, R_{18}, R_{29}—470,000 ohm, ½ w. res.
R_7—4700 ohm, ½ w. res.
R_9, R_{14}, R_{23}—500,000 ohm audio taper pot
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R_{21}—680,000 ohm, ½ w. res.
R_{22}—15,000 ohm, 2 w. res.
R_{25}, R_{27}—2.2 megohm, ½ w. res.
R_{26}—10,000 ohm, ½ w. res.
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C_3—.005 µfd., 400 v. disc ceramic capacitor
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J_1—4-terminal barrier strip (Jones)
V_1—12AX7 tube
V_2—12AT7 tube
V_3—12AU7 tube
V_4—6X4 tube

A Two-Tube Binaural Receiver

By JAMES J. BROPHY

Build this inexpensive, two-tube receiver which provides an interesting "3-D" effect with AM-FM station simulcasts.

IN MANY cities regularly scheduled stereophonic sound programs are broadcast over standard AM stations for one channel and the station's FM affiliate for the other channel. Program listings in the local newspaper generally identify these broadcasts. Using such program material, the experimenter can easily experience the thrill of "3-D" sound with simple equipment costing far less than the two-channel tape systems now becoming popular.

Standard AM and FM receivers can be used if they are placed several feet apart facing the listener. For the maximum effect, however, earphones should really be used so that each ear hears only one channel. If one does not care to provide earphone outputs on his receivers, the simple two tube set diagrammed can be constructed. While it is far from high fidelity, the "presence" generated by the stereophonic effect must be heard to be fully appreciated.

Wearing an inexpensive magnetic headset (ours dates from crystal set days), the listener can literally "point" to the various solo instruments in an orchestra, notice that the string section is down front, while the percussion group is in the rear, etc. Occasionally one gets the feeling that he is sitting in the middle of the orchestra itself. If the headset is reversed left to right, the relative positions of the various instruments interchange. This "presence" effect gives the simple system a high-fidelity sound it can't possibly have.

As shown in the circuit diagram, the receiver employs a superregenerative slope detector for FM and a regenerative receiver for AM, each followed by a single stage of audio amplification. Dual triodes are used, so that the tube complement totals two, exclusive of power supply. The circuit is quite non-critical and reasonable departures from components and tube types given are permissible. Our set was constructed in its entirety from parts supplied by the junk box.

The circuit is conventional in every respect, except perhaps for the use of both a regenerative and superregenerative detector in the same envelope. We used this arrangement out of curiosity to see how well it might work and absolutely no difficulty was encountered. There is no apparent interaction between the two detectors and no interference between them. Obviously, separate tubes could be used to simplify the parts layout problem somewhat.

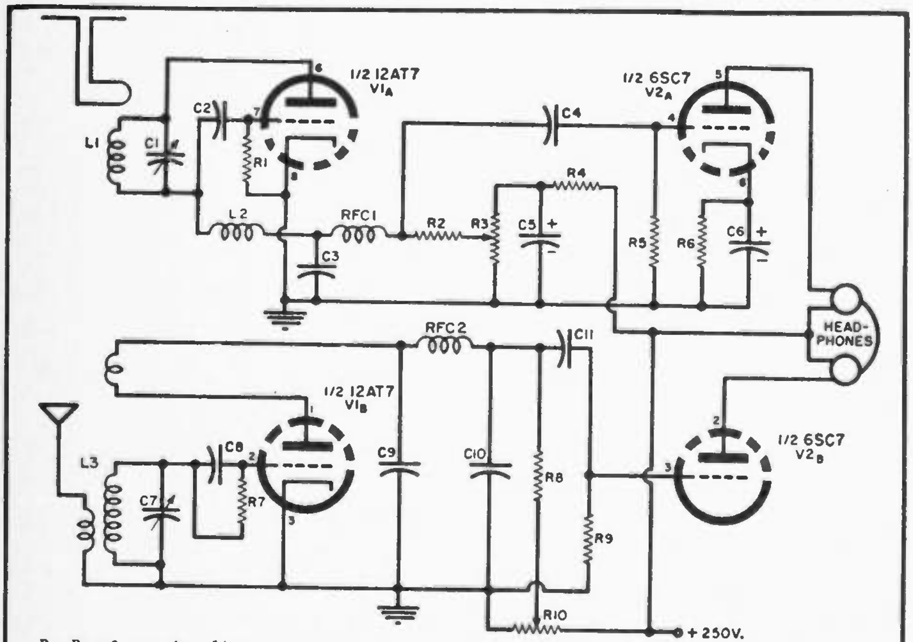
Superregeneration is controlled by variation of plate voltage of the detector. Some adjustment of the one-turn antenna coupling loop should be made to make this control operate smoothly. The set can be spotted in the FM band by squeezing or stretching the self-supporting coil. As usual, r.f. leads should be kept as short as possible, but all other wiring may be any convenient length. An outdoor dipole antenna is recommended for best results.

Plate feedback in the regenerative detector is supplied by a tickler winding on a standard broadcast coil. As usual, the proper polarity of feedback voltage must be obtained in order to make the detector oscillate. The tickler connections should be interchanged if proper operation is not achieved on the first try. Here, again, plate voltage variation controls regeneration. A short outside antenna will be sufficient in most instances.

The audio stages require little comment other than that magnetic earphones must be used in order to provide a d.c. path for the plate current. Individual volume controls in the grid circuits may be desirable to balance the sound levels of the two channels, but the regeneration controls accomplish this reasonably well. Another frill might be a switching arrangement in the outputs so that one could listen to either channel with both ears. Comparison of the effect obtained in this case with that given by the stereophonic connection is most remarkable.

Demonstration of this little set to both technical and non-technical listeners generally elicits some comment as "how can all that music come from two tubes?" The slight effort required to construct the receiver is more than repaid the first time the "3-D" effect is heard. —30—

Complete schematic of receiver designed to be used with magnetic headphones to provide a true binaural effect. Author built his receiver using available junk box parts.



R1, R7—2 megohm, 1/2 w. res.
R2—25,000 ohm, 1/2 w. res.
R3, R10—50,000 ohm, 2 w. linear-taper pot
R4—25,000 ohm, 1 w. res.
R5, R9—500,000 ohm, 1/2 w. res.
R6—680 ohm, 1/2 w. res.
R8—50,000 ohm, 1 w. res.
C1—35 μμfd. var. air capacitor
C2—50 μμfd. mica capacitor
C3—.005 μfd., 400 v. capacitor
C4, C11—.01 μfd., 400 v. capacitor
C5—8 μfd., 250 v. elec. capacitor
C6—10 μfd., 25 v. elec. capacitor
C7—365 μμfd. var. air capacitor
C8—250 μμfd. mica capacitor
C9, C10—.001 μfd., 400 v. capacitor
L1—4 t. #12 wire, spacewound on 1/2" dia. form with 1 t. coupling loop (See text)
L2—25 t. #30 d.c.c. wire, closewound on 1 megohm, 1 w. res.
L3—Standard broadcast coil with tickler winding of 10 t. #30 d.c.c. wire, closewound
Headphones—See text
RFC1—30 mhy. choke
RFC2—2.5 mhy. choke
V1—12AT7 tube *V2—6SC7 tube*

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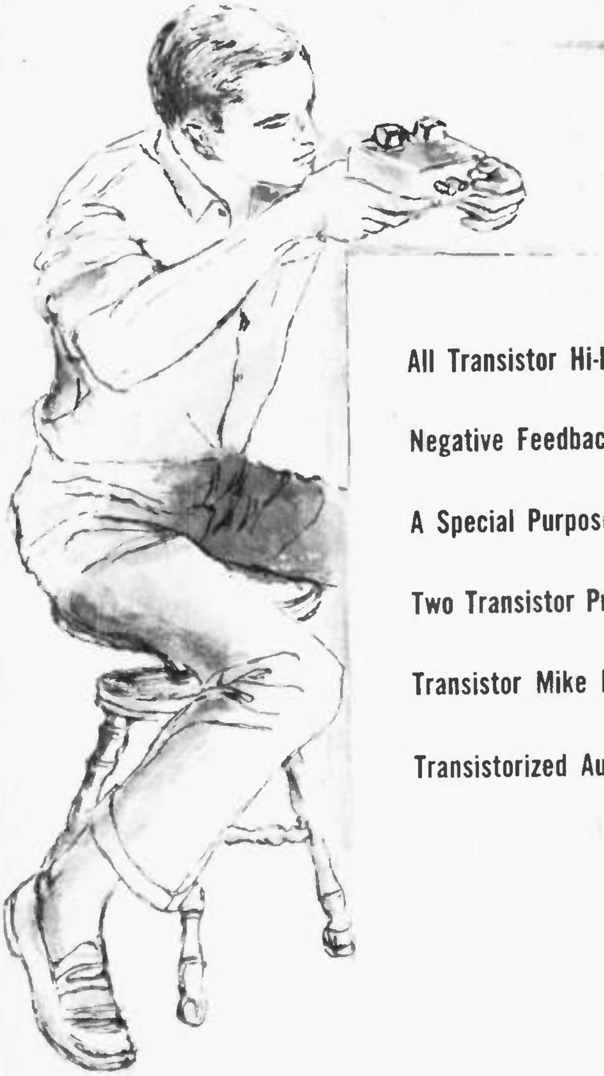
CHAPTER 7



TRANSISTOR HI-FI

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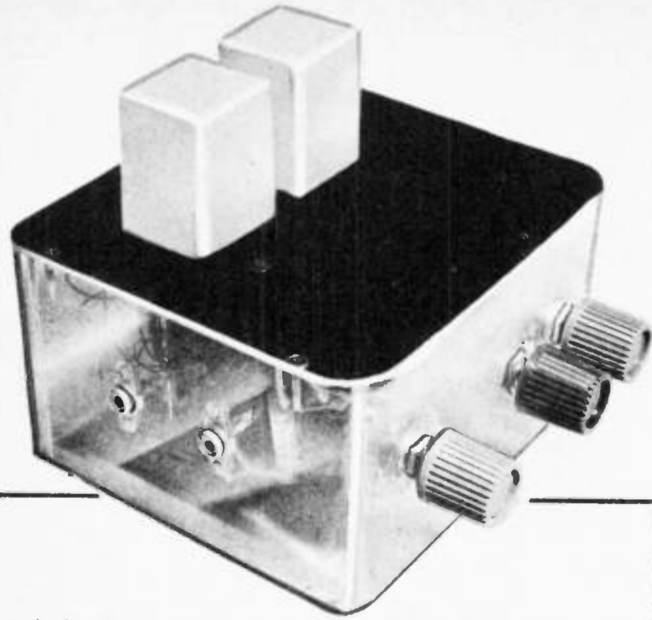
Fig. 1. The completely assembled transistorized amplifier housed in its Plexiglas case. From left to right the dials on the front control the volume, bass, and treble. The jacks, from left to right, are for output to an 8-ohm speaker system and input from a General Electric variable reluctance pickup cartridge.



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The preamplifier consists of two *RC*-coupled, grounded-emitter amplifiers with frequency selective feedback to give the required bass boost and turnover frequency. There are many possible methods of biasing transistor amplifiers, each with its own advantages and disadvantages.

If a wide range of transistor current gains is to be tolerated with stable performance at high temperatures, it is necessary to use the circuit shown in Fig. 3A. In this type of biasing, the emitter current (and hence collector current) is essentially determined only by the voltage at the base and the resistance in series with the emitter. Although this type of biasing is extremely stable, good low-frequency response requires a very large bypass capacitor around the emitter resistor, which adds considerably to the cost and little to the performance. Since this transistor hi-fi amplifier was not intended to be operated under extreme environmental conditions and the new *G-E* transistors have close control on current gain, it is possible to use a simpler form of biasing as shown in Fig. 3B. In this type of biasing the base current is determined by the collector voltage and the resistance between collector and base. This type of biasing is extremely degenerative. For example, consider the situation where the collector current increases, causing a drop in collector voltage. A decrease in collector voltage reduces the current flowing into the base which, in turn, reduces the collector current, thereby counteracting the original increase in collector current.

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transistor and fed back through a frequency sensitive network to the base of the first transistor. The performance of the preamplifier is shown in Fig. 2, and it can be seen that it is comparable in every respect to conventional tube preamplifiers.

As a matter of interest, the .05 μfd. capacitor was replaced with a 6 μfd. unit and the response of the preamp was found to be ±0.25 db from 20 to 20,000 cycles. Over-all response to the voice coil jack was within about ±2 db over the same frequency range.

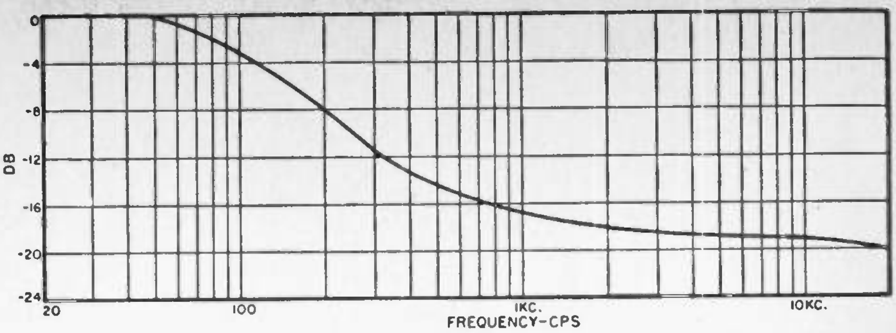


Fig. 2. The frequency response of the transistorized preamplifier unit.

Tone Controls

It might be argued that perfect reproduction of music only requires that the amplifier compensate for recording characteristics and no additional frequency compensation is necessary. In practice, however, it is necessary to include tone controls so that differences in loudspeakers, room acoustics, listening levels, and individual preferences can be accommodated.

The tone controls used in this amplifier provide independent bass and treble control without interaction and both have a 1000 cycle turnover frequency. The action of the tone controls is easily understood if they are considered as current transfer networks rather than voltage transfer networks as in vacuum tube amplifiers. The output current from the second transistor goes to the volume control where part of it is shunted to ground and the rest goes to the junction of the 0.02 μfd. and 0.2 μfd. capacitors and the center arms of the potentiometers. At 1000 cycles the equivalent circuit of the tone controls is very simple, as shown in Fig. 4A. At this frequency, the current is divided so that 10/11ths of the current is shunted to ground and 1/11th goes on to the next transistor. The low-frequency equivalent circuit for the "bass boost" condition is shown in Fig. 4B. With the movable arm of the potentiometer near the top, the 0.02 μfd. capacitor is bypassed and more of the current is shunted into the 10,000 ohm resistor as the impedance of the 0.2 μfd. capacitor rises at low frequencies.

The high-frequency equivalent circuit of the tone control is shown in Fig. 4C for the "treble cut" condition.

Depending on the potentiometer setting, most of the higher frequencies will be shunted to ground as compared to a 1000 cycle signal. With the potentiometer arm at the top, the higher frequency current would bypass the 10,000 ohm resistor and a treble boost would be achieved. An exact analysis of this type of tone control is somewhat complicated since the performance depends on the volume control setting and the input impedance of the following transistor stage. Listening tests indicate that the performance is quite adequate and the frequency characteristics at an average volume control setting are shown in Fig. 6. It will be noted that the tone controls are not absolutely symmetrical but this is of little consequence since the extreme boost or cut settings are seldom used.

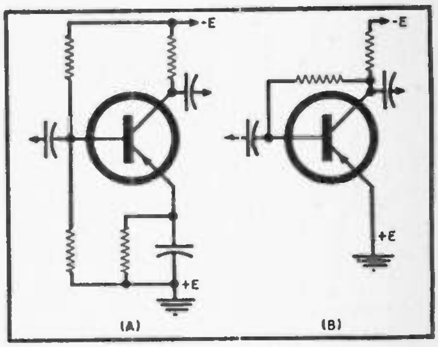


Fig. 3. (A) Biasing method suitable for extreme environmental conditions and (B) for use when unit is operated in normal places.

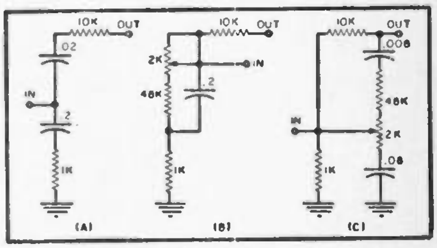
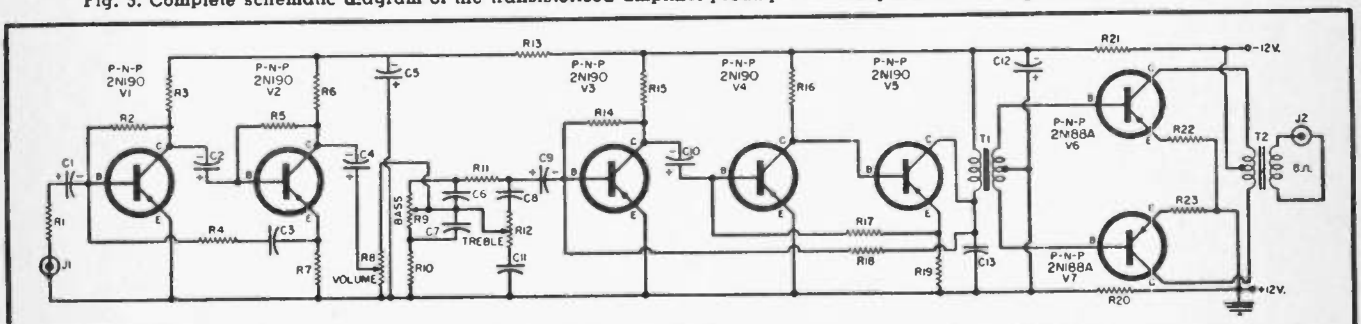


Fig. 4. (A) A 1 kc. equivalent circuit. (B) low-frequency equivalent circuit, and (C) the equivalent circuit at high frequencies.

Driver and Output Stages

The next three transistors comprise a linear amplifier and driver stage for the power amplifier. The biasing is much the same as for the preamplifier stage with the exception that the load resistances are lowered to accommodate the larger current swings at the collector. Negative feedback is used from the collector of the driver stage

Fig. 5. Complete schematic diagram of the transistorized amplifier-preamp unit. All parts are readily available at distributors.



R_1—3900 ohm, ½ w. res.
R_2, R_5, R_{14}—220,000 ohm, ½ w. res.
R_3, R_6, R_{15}, R_{16}—18,000 ohm, ½ w. res.
R_4—4700 ohm, ½ w. res.
R_7—220 ohm, ½ w. res.
R_8—100,000 ohm, 2 w. audio taper pot.
R_9, R_{12}—50,000 ohm, 2 w. linear taper pot.
R_{10}, R_{13}—1000 ohm, ½ w. res.
R_{11}—10,000 ohm, ½ w. res.
R_{17}—160,000 ohm, ½ w. res.
R_{18}—5 megohm, ½ w. res.
R_{19}—500 ohm, ½ w. res.
R_{20}—5 ohm, ½ w. res.
R_{21}—100 ohm, ½ w. res.
R_{22}, R_{23}—10 ohm, ½ w. res.
C_1, C_9—25 μfd., 12 v. elec. capacitor
C_2, C_4, C_{10}—5 μfd., 12 v. elec. capacitor
C_5—.05 μfd. capacitor
C_3, C_{12}—50 μfd., 25 v. elec. capacitor
C_6—.02 μfd. capacitor
C_7—.2 μfd. capacitor
C_8*—.008 μfd. capacitor
C_{11}*—.08 μfd. capacitor
C_{13}—.0047 μfd. capacitor

*These are design values. .01 μfd. and .1 μfd. can be used for C_8 and C_{11} respectively without seriously affecting performance.

J_1, J_2—Phono jack

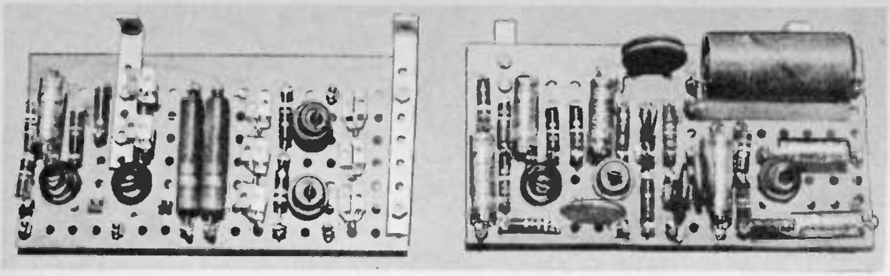
T_1—Interstage trans., 5000 to 3000 ohms centertapped (Argonne Transformer Type AR-500)

T_2—Output trans., 125 ohms centertapped to 8 ohms (Argonne Transformer Type AR-501)

V_1, V_2, V_3, V_4, V_5—"p-n-p" junction transistor (General Electric 2N190)

V_6, V_7—"p-n-p" junction transistor (General Electric 2N188A)

World Radio History



Top view of early version of amplifier (left) and preamp-tone control chassis (right).

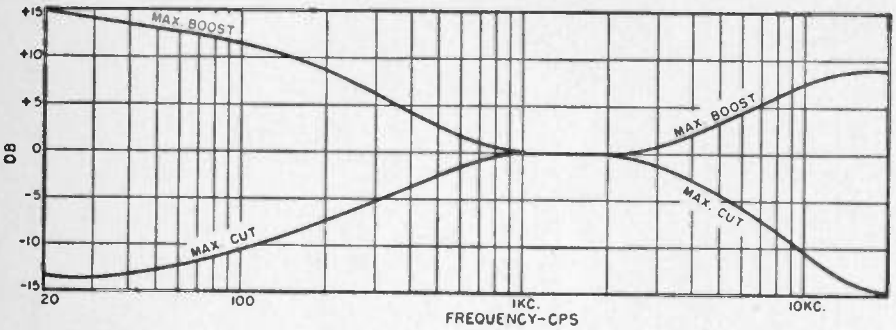


Fig. 6. Frequency response of the tone controls used with transistorized amplifier.

to the base of the third transistor. This type of feedback makes the output voltage proportional to the current flowing into the third transistor, which is precisely what is desired since the tone controls were designed from a current transfer basis. Feedback applied to the base also lowers the input impedance of the third transistor so that it is small compared to the 10,000 ohm resistor in series with the signal. A low input impedance, controlled by the feedback network, helps to make the amplifier performance independent of transistor variations.

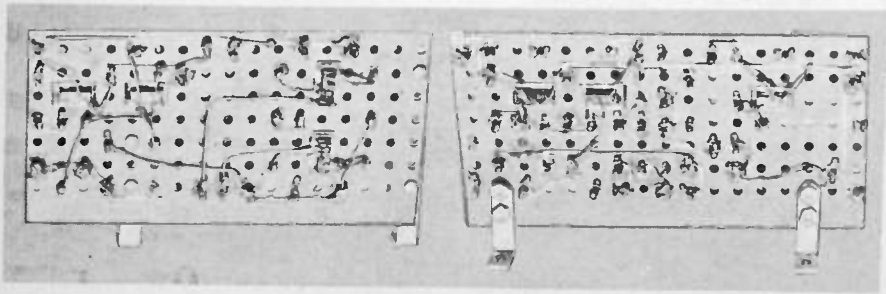
The driver transformer acts as a phase inverter and impedance match for the push-pull output stage. In order to prevent crossover distortion in a transistor push-pull class B amplifier, it is necessary to supply a slight forward bias to the transistors. This bias is provided for by the currents of the various transistors flowing through R_{20}, a 5-ohm base biasing resistor. The 10-ohm resistors in the emitters of the transistors provide a slight amount of degeneration and also prevent thermal runaway due to heating of the transistors. Under "no-signal" conditions the current drain of the output stage should be 3 to 4 ma.

One aspect of transistor output stages that sometimes confuses designers familiar with vacuum tube circuits is that no attempt is made to match the output transformer impedance to the transistors. A little thought will soon indicate the reason for this. If, as in this amplifier, it is desired to obtain undistorted peaks of two watts from a 12-volt supply, the peak current drawn by each transistor must be 333 ma. In order to get 333 ma. from a 12-volt source, the load on each transistor must be 12 volts/333 ma. or 40 ohms. The collector-to-collector load will be four times this, or 160 ohms. Therefore the impedance of the output transformer is determined only by the supply voltage and the desired output power.

If the transformer were designed to match the approximately 30,000 ohm output impedance of the transistors, maximum efficiency would be obtained but the undistorted output power would be limited to a few milliwatts.

The transistors used in the output stage are designed to have a gain which is constant from low collector currents up to collector currents of 0.2 or 0.3 ampere, so that there will be no distortion of the amplified signal. These transistors are measured and classified at the factory so that any two transistors with the same RETMA type number will give excellent performance in push-pull circuits.

Bottom view of chassis. Use of punched boards facilitates the wiring operation.



At the present time there are very few transformers on the market adequate for high-quality transistor amplifiers. Most so-called transistor transformers are designed with miniaturization as an objective so that the frequency response is severely limited. The driver and output transformers used in this amplifier were developed by the *New England Transformer Co.* (distributed by *Argonne*) specifically for this application, and these transformers (or those having equivalent quality) should be used for best results.

In a class B amplifier the current drain varies with the output signal so a well regulated power supply is a "must" for this amplifier. A very convenient source of 12 volts for experimental work can be obtained from a car battery or some of the miniature storage batteries now available. Alternatively, a regulated 12-volt supply working from the power line may be used.

Construction

One extremely desirable feature of transistor circuitry is that the layout is noncritical since all circuits operate at a low impedance level and the transistors are small and can be mounted in any position. The author chose to construct the circuits using pre-punched terminal cards and ratchet terminals available from the *Alden Products Company,* Brockton, Mass. The components (including transistors) are mounted on one side of the boards and the wiring is done on the other side. All the circuits used in the amplifier were constructed on two 2⅛" x 4¼" punched terminal cards. These cards were fastened by brackets (also available from *Alden*) to the main chassis which holds the transformers and potentiometers. The advantages of this method of construction are that it is rapid, rugged, and the final chassis has the "professional" look of printed wiring. Transistor sockets were used in the original design, since it was desired to make interchangeability tests. Sockets are not really necessary, however, and the transistors might just as well be soldered directly into the circuit since each transistor is backed up by a full year's warranty when used within ratings.

Other possible methods of construction include conventional point-to-point wiring on a metal chassis, use of etched circuit boards, and wiring on terminal boards. The exact choice of layout and size will depend on the location of the amplifier. The compact size and negligible heat dissipation of this amplifier make it possible to install the amplifier in a speaker cabinet, bookshelf, or even in the base of a record changer.

Although small in size, this amplifier definitely is not small in performance. It may not be the amplifier to rattle the windows of the houses in the next block, but the output power and quality leave little to be desired for ordinary listening in your apartment or home. —30—

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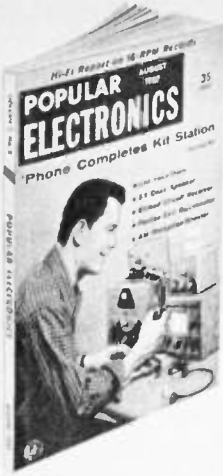

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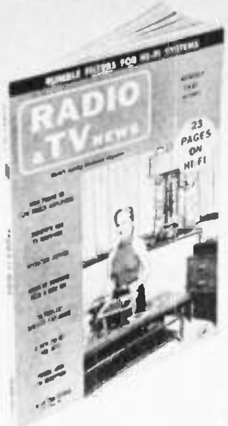

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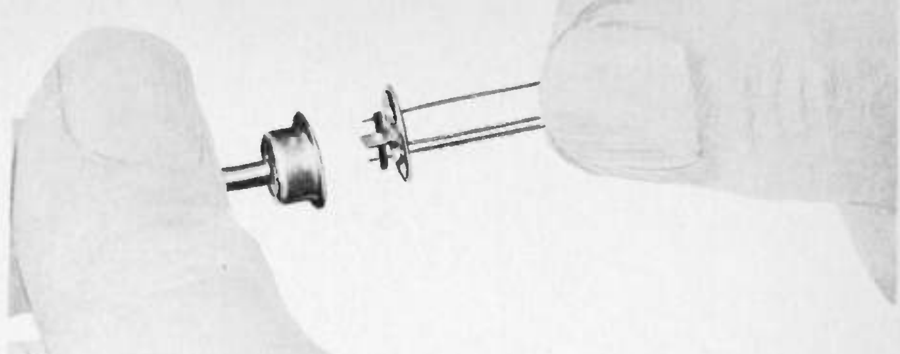
A view of the widely available General Electric 2N107 "p-n-p" junction transistor showing the internal construction.



Negative Feedback Transistor Amplifiers

By **HUGH R. LOWRY**
Manager, Application Engineering
Semiconductor Products Dept.
General Electric Co.

Complete design data for transistor negative feedback audio amplifiers with high-fidelity characteristics.

VACUUM-TUBE audio amplifiers incorporating negative feedback have always been a popular item with the electronic hobbyist since there is a wealth of material in the literature giving design procedures and showing typical circuits which can be modified if desired.

This situation is not true with transistor audio amplifiers because, until recently, most of the engineering effort has gone into developing circuits that will produce the maximum available gain from a transistor. This type of circuitry gives unsatisfactory results from transistor audio amplifiers since the frequency response tends to be limited, the distortion high, and the performance extremely dependent on the characteristics of the individual transistor.

This article will describe how to design simple transistor negative feedback audio amplifiers that have characteristics good enough for even the "golden eared" group of audio enthusiasts. Before getting into actual circuits, however, it would be best first to review briefly a few basic facts about transistor circuits.

The first important point about transistors is that their input impedance is quite low. As shown in Fig. 1, the input impedance is approximately $Z_{in} = (1 + \beta)\, h_{ib} \approx (1 + \beta) \frac{30}{I_e}$ where β is the grounded-emitter current gain. *Beta* (β) is normally in the range of 30 to 100 making the input impedance roughly 1000 to 3000 ohms. With such a low input impedance, a transistor makes an excellent current adder. If two signals are fed from a source impedance high in comparison to the input impedance, the input current into the transistor will be essentially equal to the sum of these two currents.

The second fundamental is the concept of a current divider as shown in Fig. 2. If a current source is fed into a network consisting of an impedance Z_1 and Z_2, the output current will be equivalent to $\frac{Z_1}{Z_1 + Z_2}$. An example of these two principles is the analysis of the low-frequency response of the circuit shown in Fig. 3.

Here a transistor biased in the conventional manner is being fed from a transistor amplifier with a 3900 ohm load resistance. The equivalent circuit for low frequency may be approximated as shown on the right on Fig. 3. This equivalent circuit consists of a current source which is feeding into the parallel combination of a gener-

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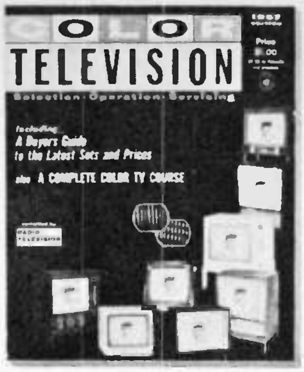

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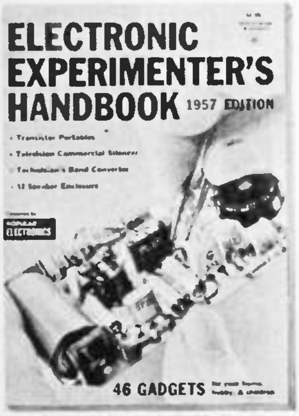

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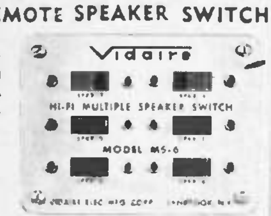

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ator resistance and input impedance of the transistor. Since the input impedance of a transistor is equal to $(1 + \beta)$ times the impedance of the emitter circuit, the emitter resistance must be multiplied by $(1 + \beta)$ and the bypass capacitor will be divided by $(1 + \beta)$. The formula for the transfer function is shown in Fig. 4 and it can be seen that if $R(1 + \beta)$ is much larger than the generator resistance, the low-frequency 3 db point is determined only by the generator resistance, current gain, and the emitter bypass capacitor. In other words, the emitter bypass capacitor is not bypassing the emitter resistance, but is bypassing a resistance equal to the generator resistance divided by $1 + \beta$. Fig. 5 shows the calculated frequency response of this circuit and it can be seen that even with a 25 μfd. bypass capacitor, the low-frequency response is down 3 db at 150 cycles.

The third fundamental is the concept of the gain of a servo type amplifier. This is shown in Fig. 6. Assuming that the feedback network, denoted by H, does not load the circuit, the gain is equivalent to G divided by $1 + GH$. With loading included, the gain is equal to $\frac{1 - H}{H}\left[\frac{GH}{1 + GH}\right]$.

These fundamentals will now be applied to the problem of high-fidelity audio amplifiers. To start off with, consider the case where a transistor is desired as a linear amplifier to work into a conventional vacuum-tube amplifier. It is also assumed that a variable reluctance pickup is to be used, that the supply voltage is 50 volts, and also that the emitter bypass capacitor be eliminated in order to obtain the best possible low-frequency response. A circuit that would meet these specifications is shown in the diagram Fig. 7.

The biasing is done with a resistance from collector to base and with the circuit shown, the collector-to-emitter bias voltage will be about 2.5 volts. The items of interest would be the current gain, the voltage gain, and the input resistance. If the circuit is analyzed as a servo-amplifier, assuming that the feedback current and input current are added without interaction, the equivalent circuit is shown in Fig. 8. The feedback factor will be equal to the load resistance divided by the load resistance plus the feedback resistance. Putting this into the formula of a servo-amplifier with loading, the current gain is seen to be essentially the feedback resistance divided by the load resistance. At this point, the readers who are familiar with transistor circuitry are probably wondering how good these approximations are since, as is well known, a transistor is a bi-lateral device, there will be a forward transmission through the feedback loop as well as other considerations that must be included in an exact analysis. The degree of approximation can be seen from a comparison of the input resistance using the approximate and exact formulas as are shown in Fig. 10A.

The approximate input resistance is equal to the current gain plus one times h_{ib}. Using this formula, the input resistance comes out 90 ohms. The exact expression for input resistance for this type of circuit is given in Shea's "Principles of Transistor Circuits" and is shown on the right hand side of Fig. 10A. A numerical evaluation of this formula gives the input impedance as 86.5 ohms as compared to 90 ohms from the approximate formula. In order to obtain the correct high-frequency roll-off with a reluctance cartridge, it should be loaded down with about 6200 ohms. Thus the circuit shown in Fig. 7 has 6200 ohms in series with the signal and it is obvious that the difference between 86.5 and 90 ohms is of little significance in the design of this amplifier. The expression for voltage gain for this amplifier is shown in Fig. 10B and is approximately equal to the feedback resistance divided by the input resistance which in this case is about 16. One final point concerning this type of amplifier is that a radio-frequency type transistor such as the *G-E* 2N168 should be used because the output capacitance of a so-called *p-n-p* audio transistor is very high and if an audio transistor were used with a 50,000 load resistance, the high-frequency gain would begin to fall off at roughly 5000 to 10,000 cycles.

The simple amplifier, just described, is not suitable when followed by another transistor amplifier. If the next stage is a transistor, the input impedance of the following stage is very low and there would be essentially no a.c. feedback in the amplifier. There will, of course, be d.c. feedback to stabilize the operating point.

A configuration suitable for transistor amplifiers is shown in Fig. 9. This circuit consists of two *RC*-coupled grounded emitter stages with negative current feedback from the emitter of the second transistor to the base of the first transistor. It is desirable to linearize the current gain with feedback since the input (reluctance pickup plus 6200 ohms series resistance) is essentially a current source. If the feedback capacitance is on the order of 5 μfd. the frequency response will be flat down to 20 cycles. By using a small capacitor, however, it is possible to effectively decrease the feedback at low frequencies and produce a bass boost. To compensate for the RIAA recording characteristics, the response of the preamplifier should have a 500-cycle turnover frequency with the response leveling out again near 50 cycles.

Using the servo-amplifier analogy, an analysis of the configuration of Fig. 9 yields the expression of Fig. 11. The turnover frequency is determined by the time constant of the feedback circuit and the low-frequency response plateau is a function of the open loop current gain. The performance of an actual circuit is shown in Fig. 12 and it can be seen that it is comparable to conventional vacuum-tube circuits.

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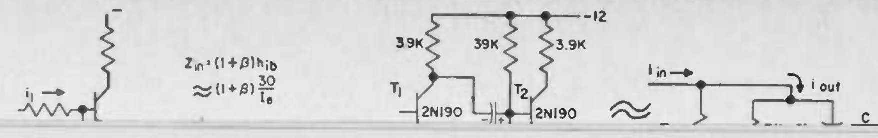

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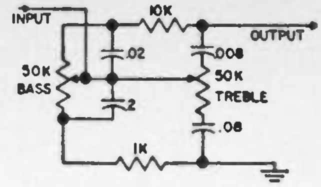

Fig. 19

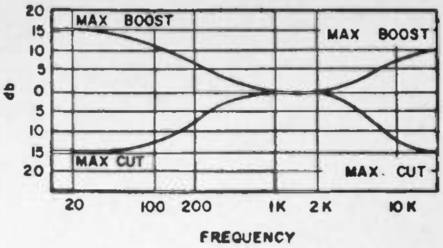

Fig. 20

ference between transistor and tube circuits being that current feedback is used with transistors and voltage feedback with tubes. The use of current feedback in transistor amplifiers requires very few extra components and the performance advantages far outweigh the time taken in design.

REFERENCES

1. *Lo, Endres, Zawels, Waldhauer & Cheng: "Transistor Electronics," Prentice-Hall.*

2. *Lowry, Hugh R.: "All Transistor Hi-Fi Amplifier,"* RADIO & TELEVISION NEWS, *November 1956.* —30—

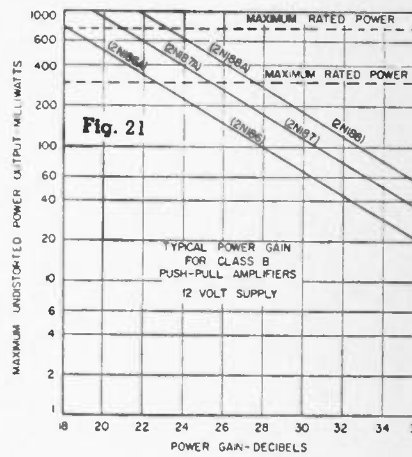

Fig. 21

Two charts shown here indicate power gains that are obtained with commonly used transistors at various power outputs with a 12-volt supply. The chart above is Class B push-pull amplifiers and the chart below is for class A single-ended circuits. Note higher output powers available from those transistors with "A" designations.

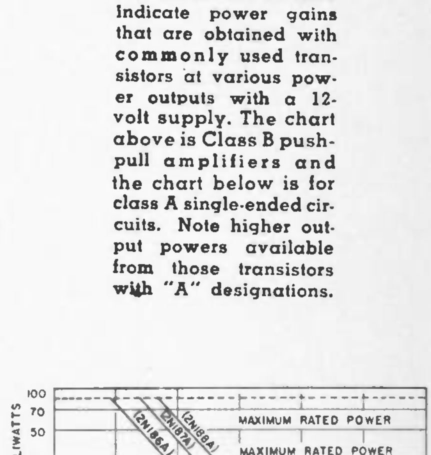

Fig. 22

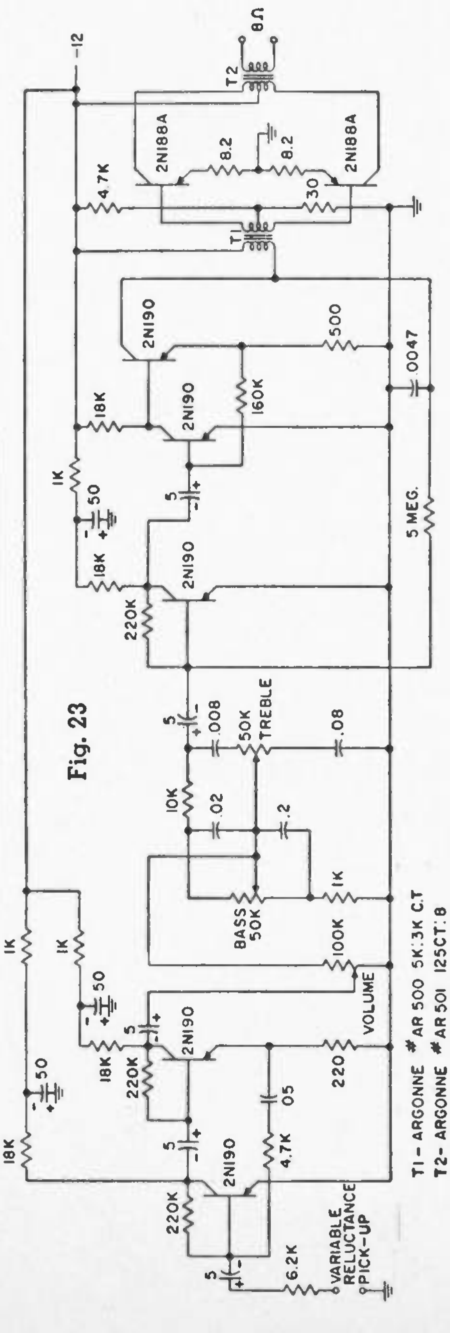

Fig. 23

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formulas as are shown in Fig. 10A.

The approximate input resistance is equal to the current gain plus one times h_{ib}. Using this formula, the input resistance comes out 90 ohms. The exact expression for input resistance for this type of circuit is given in Shea's "Principles of Transistor Circuits" and is shown on the right hand side of Fig. 10A. A numerical evaluation of this formula gives the input impedance as 86.5 ohms as compared to 90 ohms from the approximate formula. In order to obtain the correct high-frequency roll-off with a reluctance cartridge, it should be loaded down with about 6200 ohms. Thus the circuit shown in Fig. 7 has 6200 ohms in series with the signal and it is obvious that the difference between 86.5 and 90 ohms is of little significance in the design of this amplifier. The expression for voltage gain for this amplifier is shown in Fig. 10B and is approximately equal to the feedback resistance divided by the input resistance which in this case is about 16. One final point concerning this type of amplifier is that a radio-frequency type transistor such as the *G-E* 2N168 should be used because the output capacitance of a so-called *p-n-p* audio transistor is very high and if an audio transistor were used with a 50,000 load resistance, the high-frequency gain would begin to fall off at roughly 5000 to 10,000 cycles.

The simple amplifier, just described, is not suitable when followed by another transistor amplifier. If the next stage is a transistor, the input impedance of the following stage is very low and there would be essentially no a.c. feedback in the amplifier. There will, of course, be d.c. feedback to stabilize the operating point.

A configuration suitable for transistor amplifiers is shown in Fig. 9. This circuit consists of two *RC*-coupled grounded emitter stages with negative current feedback from the emitter of the second transistor to the base of the first transistor. It is desirable to linearize the current gain with feedback since the input (reluctance pickup plus 6200 ohms series resistance) is essentially a current source. If the feedback capacitance is on the order of 5 μfd. the frequency response will be flat down to 20 cycles. By using a small capacitor, however, it is possible to effectively decrease the feedback at low frequencies and produce a bass boost. To compensate for the RIAA recording characteristics, the response of the preamplifier should have a 500-cycle turnover frequency with the response leveling out again near 50 cycles.

Using the servo-amplifier analogy, an analysis of the configuration of Fig. 9 yields the expression of Fig. 11. The turnover frequency is determined by the time constant of the feedback circuit and the low-frequency response plateau is a function of the open loop current gain. The performance of an actual circuit is shown in Fig. 12 and it can be seen that it is comparable to conventional vacuum-tube circuits.

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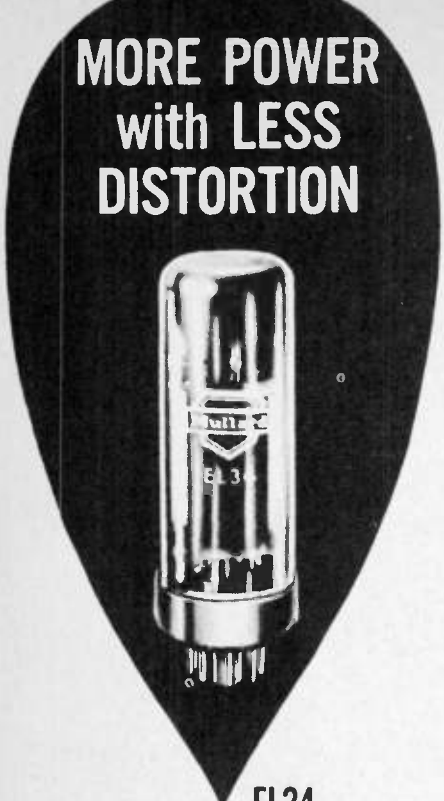

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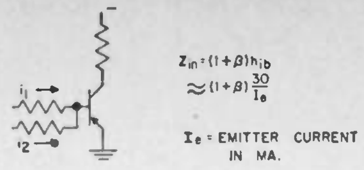


Fig. 1

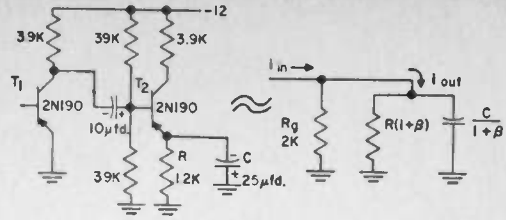
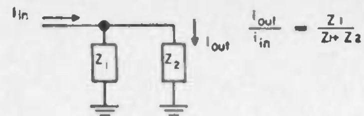


Fig. 3



← Fig. 2

By adding a resistor in shunt with the capacitor, a more positive control over the low-frequency plateau can be obtained. For example, for a 500-cycle turnover frequency the impedance of the capacitor at 500 cycles should be equal to R_2. A 50-cycle low-frequency plateau is obtained if the resistor shunting C is made equal to $9R_2$.

The biasing methods used in the amplifiers described thus far are suitable for *RC*-coupled amplifiers where the collector voltage is considerably smaller than the supply voltage. A problem arises, however, on the biasing of a driver stage which is transformer-coupled to the output stage. With a transformer load in the collector circuit, a biasing resistor from collector to base is no longer d.c. degenerative since the d.c. resistance in the collector circuit is small. One possible circuit for a small signal amplifier followed by a driver stage is shown in Fig. 13A. This type of biasing will stabilize the collector current of the driver stage, but a large bypass capacitor must be used for good low-frequency response as explained earlier. A better type of circuit is shown in Fig. 13B. The base of the driver stage is directly connected to the collector of the preceding stage and the emitter current of the driver stage is determined by the voltage at the base and the resistance in the emitter circuit. For the circuit shown, the bias current in the driver stage would be 4 ma. The biasing resistor to the base of the first stage is connected to the emitter of the driver stage rather than to the collector of the first transistor. This type of biasing by current feedback makes the bias points more stable than if each transistor were biased separately as in Fig. 13A. The reduction in the number of components by using direct coupling is also quite apparent. Perhaps the most important advantage of the direct-coupled circuit of Fig. 13B is that there are no capacitors to produce phase shift at low frequencies and possible instability. With this direct-coupled circuit it is possible to add another amplifier stage and use feedback around all three stages. A typical circuit of this type is shown in Fig. 14. The resistor from the collector of the driver stage to the base of the first stage makes the output voltage proportional to the input current which is very desirable. It is not necessary to use a d.c. blocking capacitor with the 5-megohm feedback resistance since the d.c. flowing through the resistor will not appreciably affect the bias point.

If the driver transformer has a 5000 ohm primary impedance, the over-all current gain of the amplifier will be 5 megohms/5000 ohms or 1000 as calculated by the formula in Fig. 8. Depending on the frequency response of the transformer, it may be necessary to add a capacitor from the collector of the last transistor to ground in order to prevent high-frequency oscillations. High-frequency oscillations occur because the high-frequency phase shift produced by three cascaded stages is enough to make the feedback at high frequencies positive rather than negative unless precautions are taken.

Class B Push-Pull Outputs

The design of driver and push-pull stages depends on the desired power output so it is necessary to start with the final stage and work backwards. In the majority of applications, the output power is specified so a design will usually begin at this point. The circuit of a typical push-pull class B output stage is shown by Fig. 18.

The voltage divider, consisting of resistor R and the 47-ohm resistor, gives a slight forward bias on the transistors to prevent crossover distortion. Usually about 1/10 of a volt is sufficient to prevent crossover distortion and, under these conditions, the no-signal total collector current is about 3.0 ma. The 8.2 ohm resistors in the emitter leads stabilize the transistors so they will not go into thermal runaway when the junction temperature rises to 60° C. Typical collector characteristics with a load line are shown in Fig. 15. It can be shown that the maximum a.c. output power without clipping using a push-pull stage is given by the formula:

$$P_{out} = \frac{I_{max} E_c}{2}$$

Since the load resistance is equal to $R_L = \frac{E_c}{I_{max}}$ and the collector-to-collector impedance is four times the load resistance per collector, the output power is given by the formula:

$$P_o = \frac{2 E_c^2}{R_{c-c}}$$

Thus, for a specified output power and supply voltage the collector-to-collector load resistance can be determined. For output power on the order of 50

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$$\frac{i\ OUT}{i\ IN} = \frac{R_g}{R_g + R(1+\beta)} \left[\frac{1 + 2\pi RCf}{1 + \frac{2\pi R_g RCf}{R_g + R(1+\beta)}} \right]$$

$$\frac{i\ OUT}{i\ IN} \approx \frac{R_g}{R(1+\beta)} \left[\frac{1 + 2\pi RCf}{1 + \frac{2\pi R_g}{1+\beta} Cf} \right] \quad \text{FOR } R(1+\beta) \gg R_g$$

Fig. 4

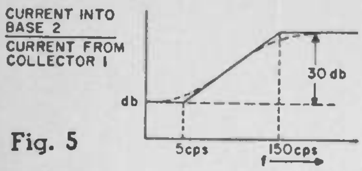

Fig. 5

mw. to 750 mw., the load impedance is so low that it is essentially a short circuit compared to the output impedance of the transistors. Thus, unlike small signal amplifiers, no attempt is made to match the output impedance of transistors in power output stages.

The power gain is given by the formula

$$Power\ Gain = \frac{P_{out}}{P_{in}} = \frac{I_o^2 R_L}{I_{in}^2 R_{in}}$$

Since $\frac{I_o}{I_{in}}$ is equal to the current gain, *beta,* for small load resistance, the power gain formula can be written as:

$$P.G. = \beta^2 \left(\frac{R_{c-c}}{R_{b-b}} \right)$$

where: R_{c-c} = collector-to-collector load resistance

R_{b-b} = base-to-base input resistance

β = grounded-emitter current gain

Since the load resistance is determined by the required maximum undistorted output power, the power gain can be written in terms of the maximum output power by combining equations (the previous equations) to give:

$$P.G. = \frac{2\beta^2 E_o^2}{R_{b-b} P_{out}}$$

Class A Driver Stages

For a required output power of 250 mw., the typical gain for a push-pull output stage would be on the order of 23 db. Thus the input power to the output stage would be about 1 to 2 mw. The load resistance of a class A driver stage is then determined by the power that must be furnished to the output stage and this load resistance is given by the equation:

$$R_L = \frac{E_c^2}{2 P_o}$$

For output powers on the order of a

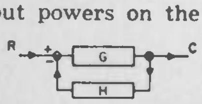

Fig. 6

WITHOUT LOADING $\frac{C}{R} = \frac{G}{1+GH}$

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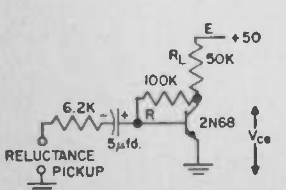

$V_{ce} \approx \frac{ER}{R + \beta R_L}$

$V_{ce} \approx 2.5V$

Fig. 7

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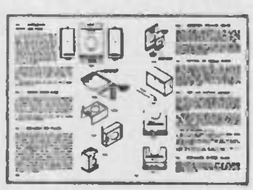

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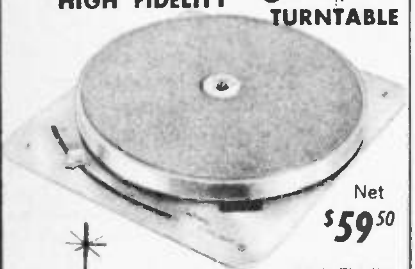

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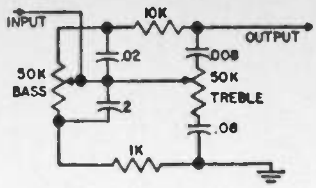

Fig. 19

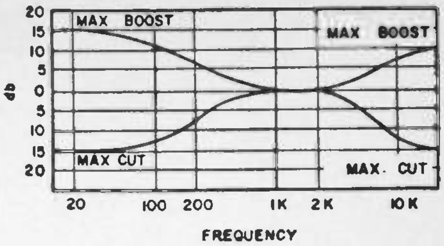

Fig. 20

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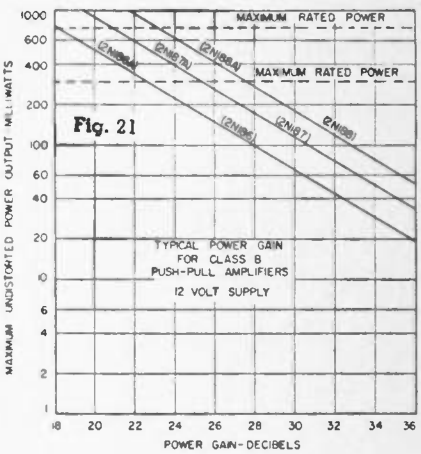

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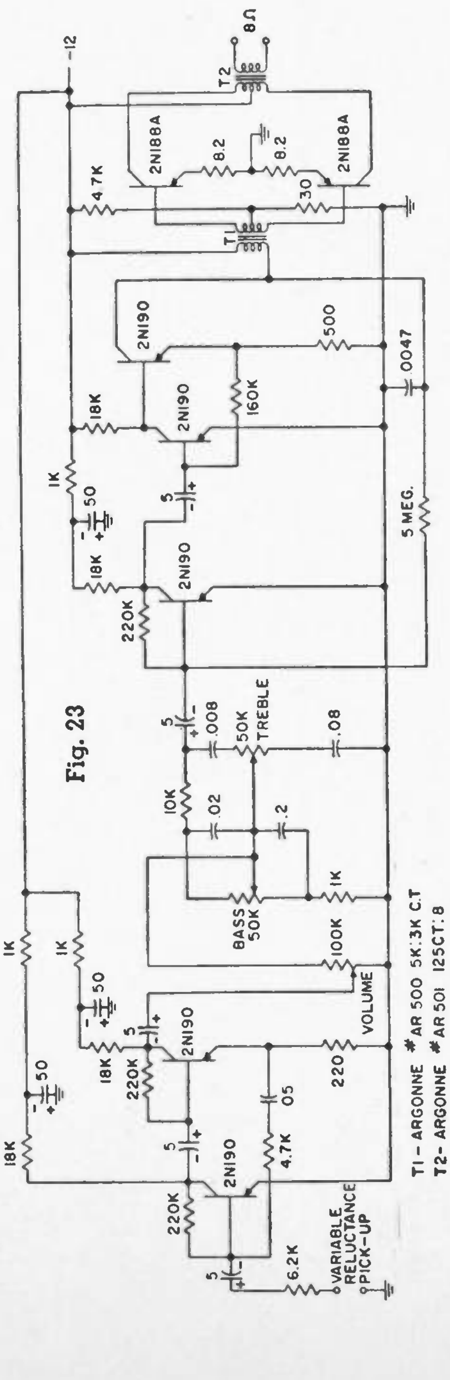

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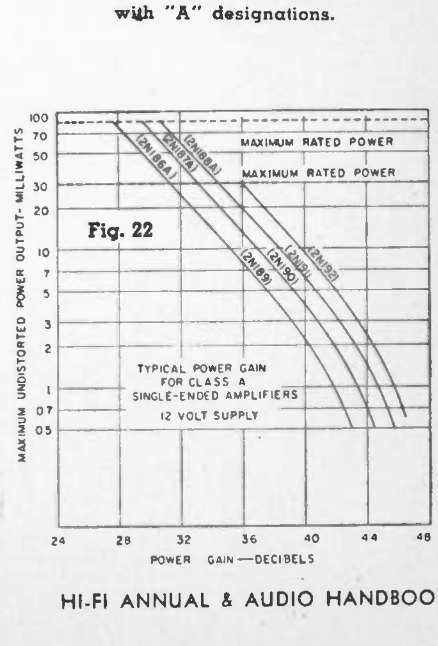

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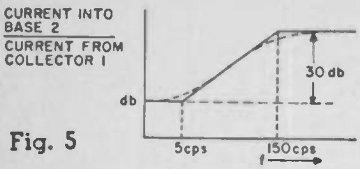

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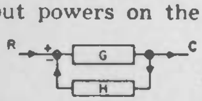

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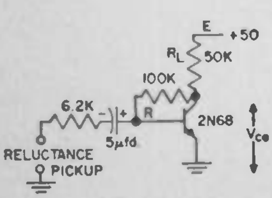

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Fig. 7

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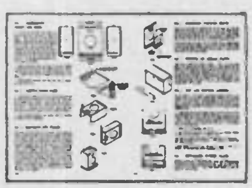

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few milliwatts, the load resistance is not negligible in comparison to the output impedance of the transistors, therefore, more exact equations must be used to determine the power gain of a class A driver stage. From four-terminal network theory, after making appropriate approximations, it can be shown that the voltage gain is given by the formula:

$$A_V = \frac{R_L}{h_{ib}}$$

where: h_{ib} = grounded base input impedance.

The current gain is given by the formula:

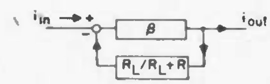
$$A_I = \frac{\alpha}{1 - \alpha + R_L h_{ob}}$$

where: h_{ob} = grounded base output admittance

The power gain is the product of the current gain and the voltage gain, thus unlike the formula for high power output stages, there is no simple relationship between required output power and power gain for a class A driver amplifier.

Design Charts

Figs. 16, 17, 21, and 22 are design charts for determination of transformer impedances and typical power gains for class A driver stages and class B push-pull stages. Their use can be best understood by working through a typical example. It will be assumed that it is desired to design a driver and push-pull amplifier capable of delivering 1 watt with a 12-volt supply. Using Fig. 16, for 1 watt of undistorted output power, the required collector-to-collector load resistance is 200 ohms. From Fig. 21 using a typical 2N187A, the power gain is 20 db. In numerical terms, a power gain of 20 db is 100. Therefore, the required input power to the driver stage would be: $P_{in} = 1000/100$ or 10 mw. From Fig. 17, for 10 mw. of undistorted output power, the load resistance is over 5000 ohms. From Fig. 22, assuming a 2N191 driver transistor, the power gain is 38 db. The typical power gain of the two stages using a 2N191 driver and 2N187's in the output would be 58 db. The secondary impedance of the driving transformer should be 3000 ohms center-tapped. The secondary impedance of the output transformer should be selected to match the impedance of the load.



$$\frac{i_{out}}{i_{in}} = \frac{\beta}{1+\frac{R_L}{R}(1+\beta)}$$

$$\frac{i_{out}}{i_{in}} \approx \frac{R}{R_L}$$

Fig. 8

APPROXIMATE

$$R_{in} = (1+A_i)h_{ib} = \left(1+\frac{R}{R_L}\right)h_{ib} = \left(1+\frac{100}{50}\right)30$$

$$R_{in} = 90\,\Omega$$

EXACT

$$R_{in} = \frac{(R+R_L)h_{ib}}{h_{ib}+R_L+R(1-a)} = \frac{(150)(30)}{52.03}$$

$$R_{in} = 86.5\,\Omega$$

(A)

$$A_V = A_I\,\frac{R_L}{R_{in}}$$

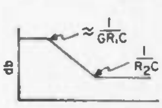
$$A_V \approx \frac{R}{R_L}\,\frac{R_L}{R_{in}} \approx \frac{R}{R_{in}}$$

$$A_V \approx \frac{100K}{6.2K} \approx 16$$

(B)

Fig. 10 ↑

$$\frac{i_{out}}{i_{in}} = \frac{G\,[1+2\pi R_2 C f]}{1+C2\pi f\,[R_2+GR_1]}$$



G = OPEN LOOP CURRENT GAIN

Fig. 11

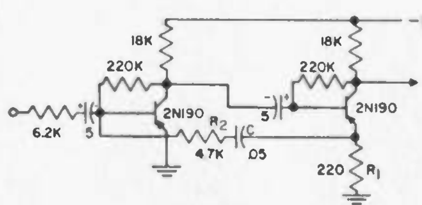
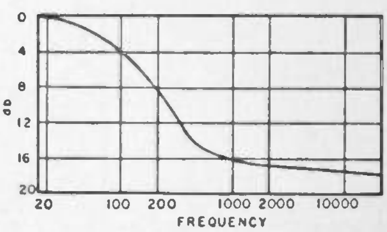


Fig. 9

Fig. 12 →



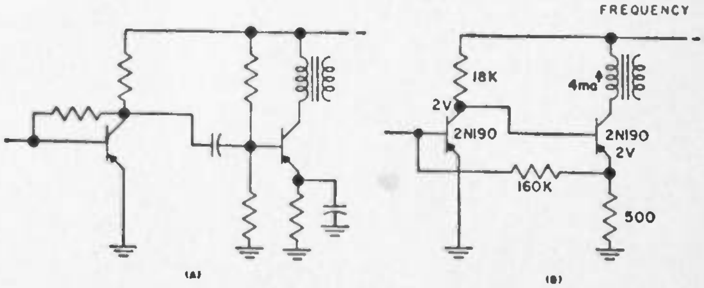


Fig. 13

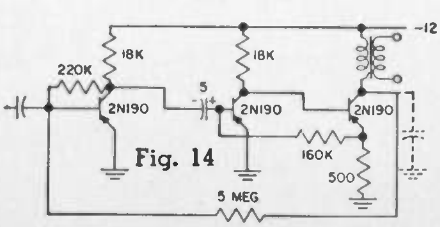


Fig. 14

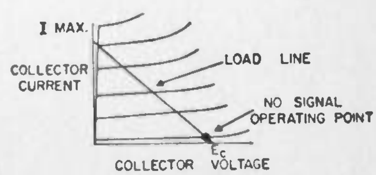


Fig. 15

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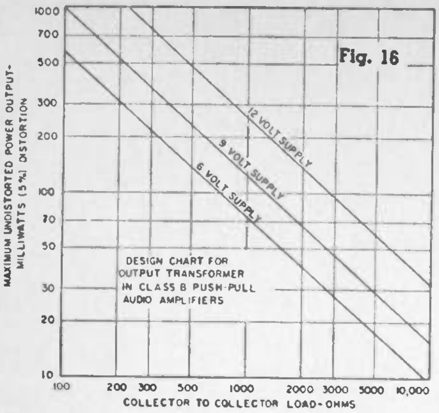


Fig. 16

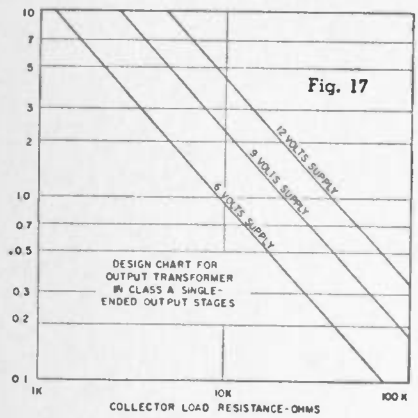


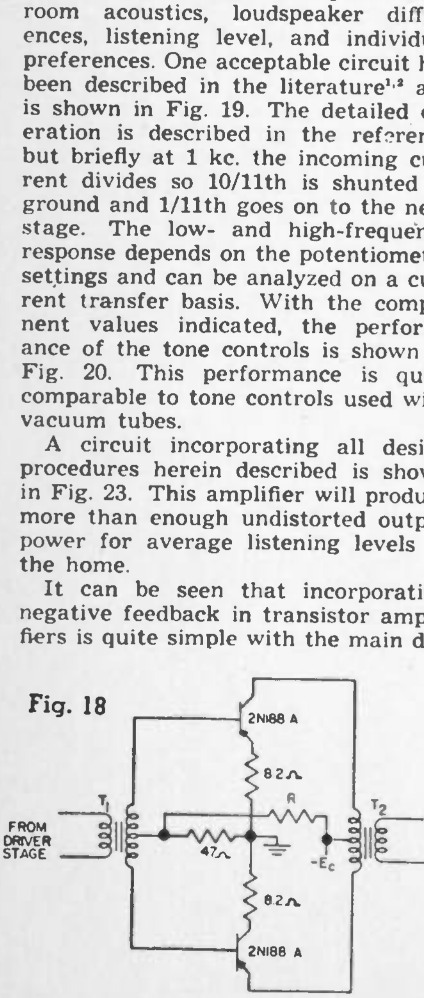
Fig. 17

A necessity for any hi-fi system is a set of tone controls to compensate for room acoustics, loudspeaker differences, listening level, and individual preferences. One acceptable circuit has been described in the literature[1,2] and is shown in Fig. 19. The detailed operation is described in the reference but briefly at 1 kc. the incoming current divides so 10/11th is shunted to ground and 1/11th goes on to the next stage. The low- and high-frequency response depends on the potentiometer settings and can be analyzed on a current transfer basis. With the component values indicated, the performance of the tone controls is shown in Fig. 20. This performance is quite comparable to tone controls used with vacuum tubes.

A circuit incorporating all design procedures herein described is shown in Fig. 23. This amplifier will produce more than enough undistorted output power for average listening levels in the home.

It can be seen that incorporating negative feedback in transistor amplifiers is quite simple with the main dif-

Fig. 18



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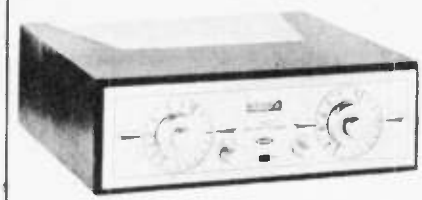

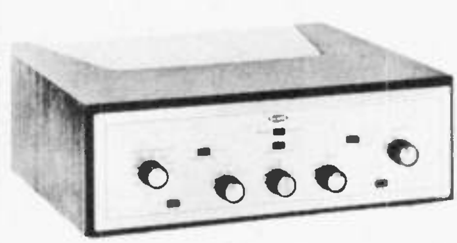

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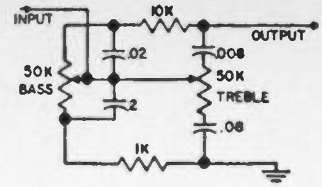


Fig. 19

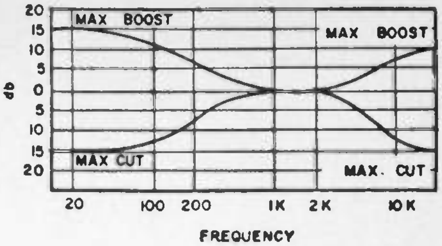


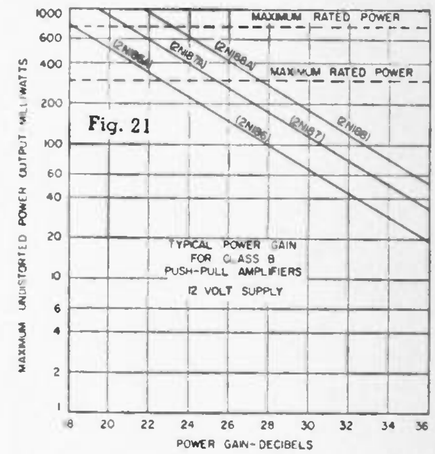
Fig. 20

ference between transistor and tube circuits being that current feedback is used with transistors and voltage feedback with tubes. The use of current feedback in transistor amplifiers requires very few extra components and the performance advantages far outweigh the time taken in design.

REFERENCES

1. Lo, Endres, Zawels, Waldhauer & Cheng: "Transistor Electronics," Prentice-Hall.

2. Lowry, Hugh R.: "All Transistor Hi-Fi Amplifier," RADIO & TELEVISION NEWS, *November 1956.* —30—



Two charts shown here indicate power gains that are obtained with commonly used transistors at various power outputs with a 12-volt supply. The chart above is Class B push-pull amplifiers and the chart below is for class A single-ended circuits. Note higher output powers available from those transistors with "A" designations.

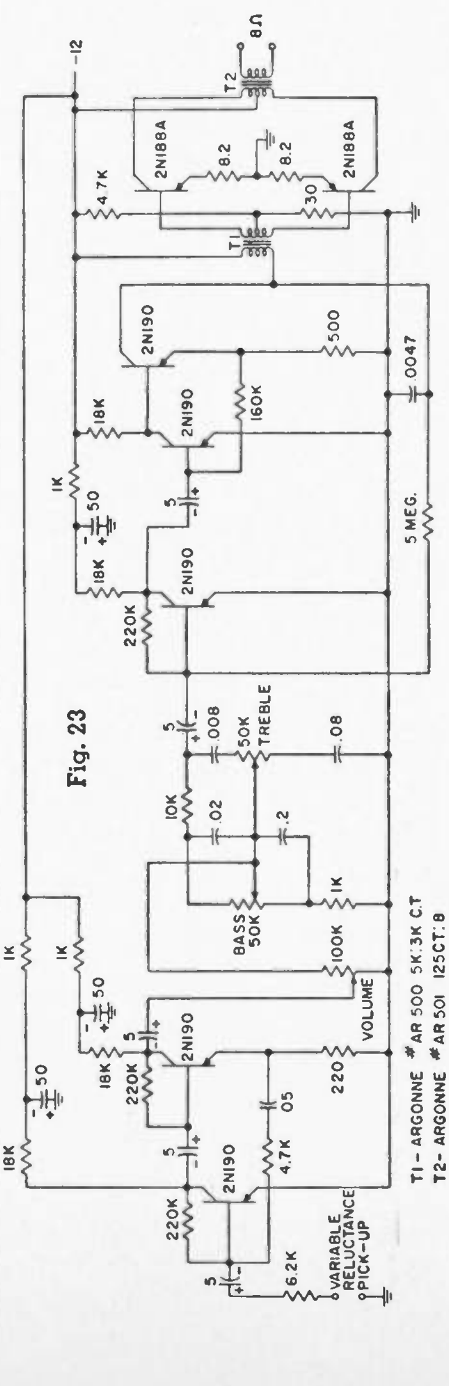
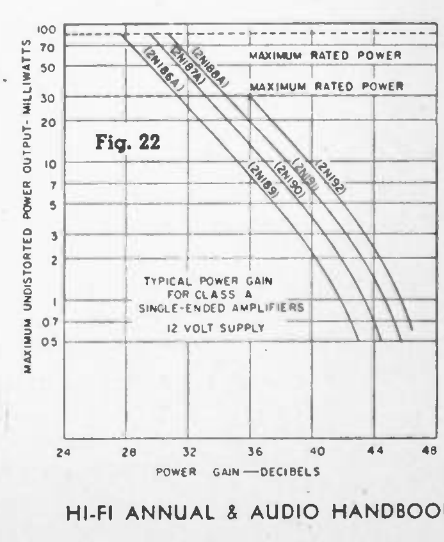


Fig. 23



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A TWO-TRANSISTOR PREAMPLIFIER

THOSE who have been looking for a compact preamp to be used with magnetic pickups will be interested in the circuit diagrammed in Fig. 1. This unit can also be used as a preamp in conjunction with the transistorized power amplifier whose construction details are given on page 138.

This unit was specifically designed for the hobbyist or the experimenter who wants to obtain some basic knowledge of transistor operation. Obviously, quality-wise, it does not compare with vacuum-tube type preamplifiers.

Although its frequency response does not follow any of the more commonly-used recording characteristics, it will provide fairly good performance when used as a preamp between a magnetic cartridge and many power amplifiers.

One problem arises, however, when this circuit is used with a wide-band power amplifier and that is with regard to background noise due to the transistors. If the power amplifier incorporates a tone control so that the high-frequency response can be cut off, transistor noise can be reduced to a negligible figure. Some means of varying the volume output is necessary.

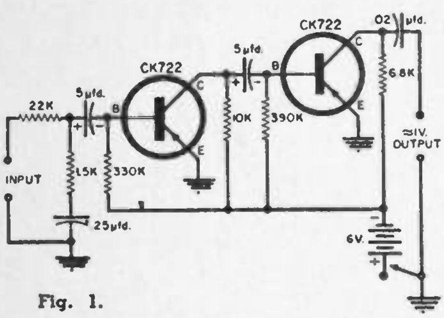

Fig. 1.

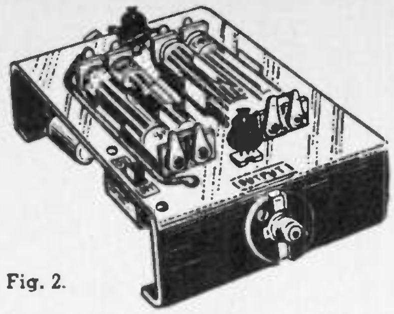
Fig. 2.

All of the component parts required for the preamp can be mounted on an aluminum chassis measuring as little as 3" x 4" x 1" if parts are selected carefully with an eye to small size. By building the unit in a small space, the preamp can be installed under the tone arm of many record playing setups. Hum pickup is, in most cases, sufficiently low to permit this installation technique if desired.

All of the parts needed to build this preamp are standard and may be obtained at any well-stocked parts house. If the builder wishes to take advantage of the convenience of a packaged circuit, Lafayette Radio, 100 Sixth Ave., New York 13, N. Y. is offering the preamp as a kit (KT-71) for $8.95 complete. -30-

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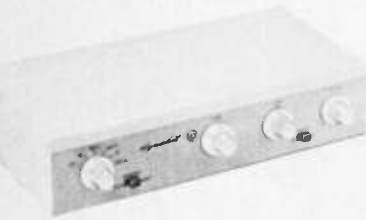

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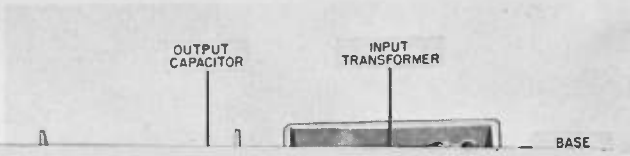

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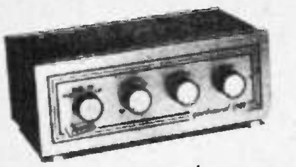

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Conversely, if the reader desires high-temperature operation but needs less gain, he can reduce the load resistance to 6000 or 7000 ohms.

If six volts is not convenient, the preamps will operate with good results from 4.5 volts or even 3 volts.

The preamps here are limited in their quality of reproduction mainly by the components. Distortion is low, since the transistor is operated in a low-distortion area. That is, it is low right up to the point of clipping, where it will clip sharply.

The frequency response is determined by the transformer and the capacitors. The size of the capacitors has been determined to allow adequate low-frequency response.

The amplifiers have been used for tape recording with good results, and for public address work with equally gratifying results. The author believes that you'll be pleased with these simple, useful floating-base mike preamps. —30—

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The circuit itself is built around three of the General Electric 2N107 junction transistors of the "p-n-p" type. The amplifier is powered by four 1½ volt penlite cells and operates in push-pull class B. Frequency response is from 100 to 8000 cps with an over-all power gain in excess of 50 db.

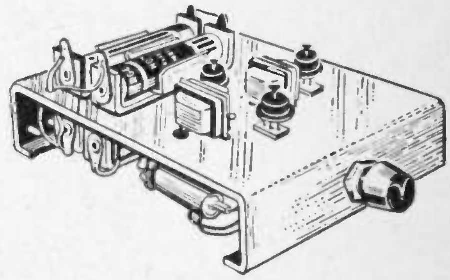
The three transistors are connected in a grounded-emitter type circuit which provides sufficient power to operate a good quality 4" to 6" PM speaker with a 3.2 ohm voice coil.

The transformers required for this construction are of the type especially designed for transistor circuitry. An important feature of such components is their high inductance and low d.c. resistance. Input impedance of this circuit is 1000 ohms. For high impedance inputs a matching transformer is required. For magnetic inputs a preamplifier is needed.

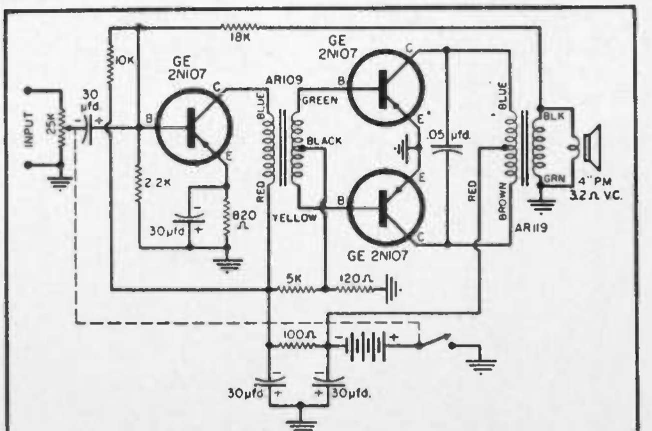
Since a feedback loop is employed, the necessary precautions in regard to phasing must be employed. Follow the color coding of the transformers carefully. Should oscillation occur after completion, reverse the black and green wires of the output transformer. The transformers, which are the only special components required to build this amplifier, are manufactured by Argonne Electronics Mfg. Co.

For those who wish to avoid the trouble of shopping for individual components, Lafayette Radio, 100 Sixth Avenue, New York 13, New York has a kit, Model KT-69, available for $17.95.

The primary purpose of this kit is for the experimenter, student, or anyone who wants to obtain some working knowledge of transistor operation. For all practical purposes, considering the price, a vacuum tube-type power amplifier would be more advisable. —30—



↑ Over-all view of the transistorized audio amplifier. Here, it is built breadboard style on a chassis measuring 3" x 4" x 1", although any arrangement of parts will do.



← Complete schematic diagram of audio amplifier. It uses three of the new G-E 2N107 transistors and two specially-designed transistor type transformers.

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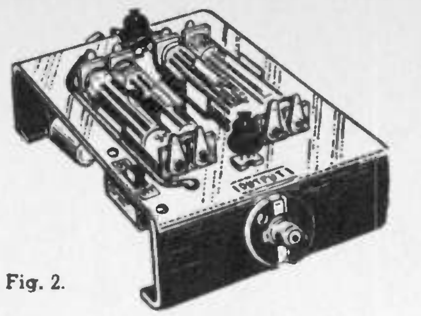
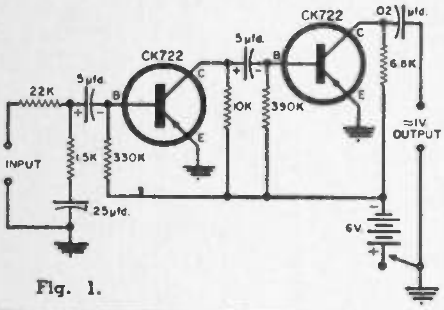


Fig. 2.



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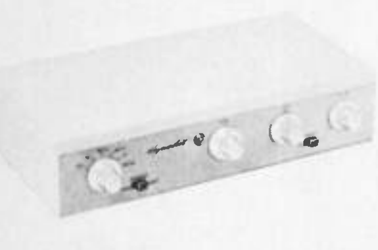

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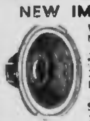

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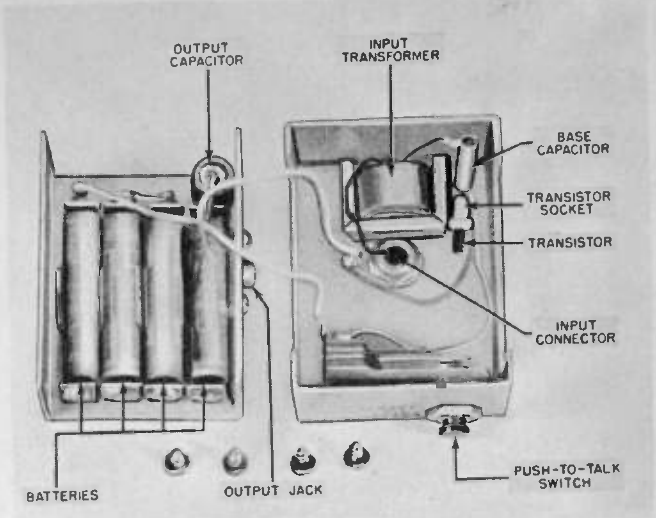


Fig. 4. Disassembled view of transistorized crystal microphone preamplifier.

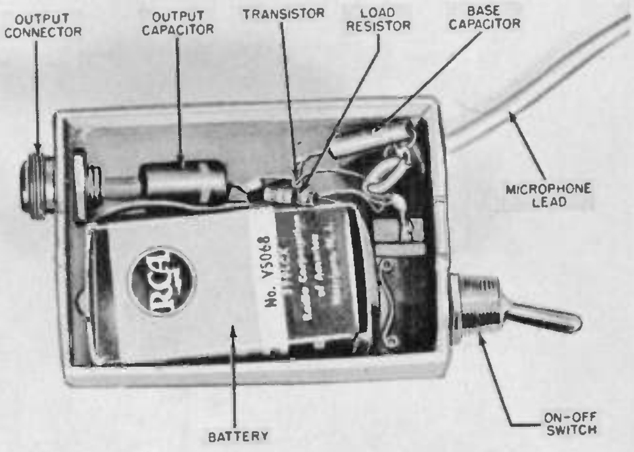


Fig. 5. Underside view of the transistorized dynamic mike preamp.

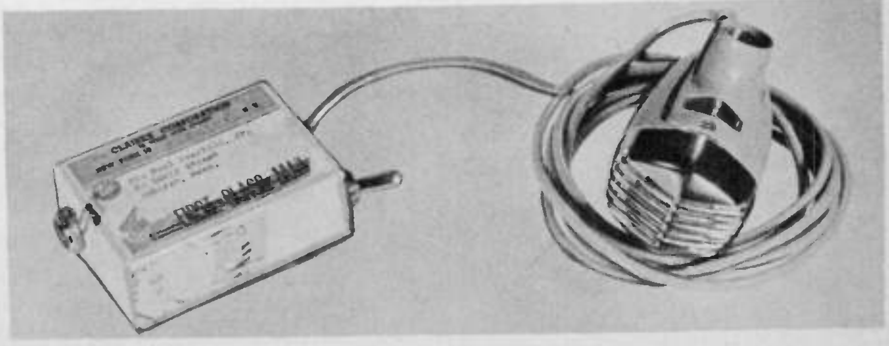
This type of circuit unfortunately has one disadvantage. If your transistor doesn't have the right amount of cut-off current, you may find yourself operating too close to either end of the load line, and thus clipping will occur. Much the same thing will happen if the input capacitor is leaky. Also, operating the transistor considerably above room temperature will increase the cut-off current and push the bias point into saturation.

Because of this last condition, the amplifiers described here should not be run above about 100° F.

The 2N105 transistor was chosen for these amplifiers because of the exceptional uniformity from transistor to transistor. The cut-off current does not vary much from one unit to the next, in contrast with other transistor models. In the author's experience, other, more popular, lines of transistors have not only showed very bad uniformity from one transistor to the next, but also have aged rather rapidly. Since aging always involves changes in cut-off current and gain, it is clear that these transistors are to be avoided.

It may be that recent vacuum baking and hermetic sealing techniques

Fig. 6. A cardboard chassis was used for dynamic mike preamp.



A TWO-TRANSISTOR PREAMPLIFIER

THOSE who have been looking for a compact preamp to be used with magnetic pickups will be interested in the circuit diagrammed in Fig. 1. This unit can also be used as a preamp in conjunction with the transistorized power amplifier whose construction details are given on page 138.

This unit was specifically designed for the hobbyist or the experimenter who wants to obtain some basic knowledge of transistor operation. Obviously, quality-wise, it does not compare with vacuum-tube type preamplifiers.

Although its frequency response does not follow any of the more commonly-used recording characteristics, it will provide fairly good performance when used as a preamp between a magnetic cartridge and many power amplifiers.

One problem arises, however, when this circuit is used with a wide-band power amplifier and that is with regard to background noise due to the transistors. If the power amplifier incorporates a tone control so that the high-frequency response can be cut off, transistor noise can be reduced to a negligible figure. Some means of varying the volume output is necessary.

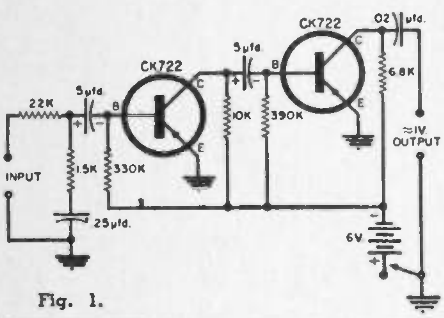


Fig. 1.

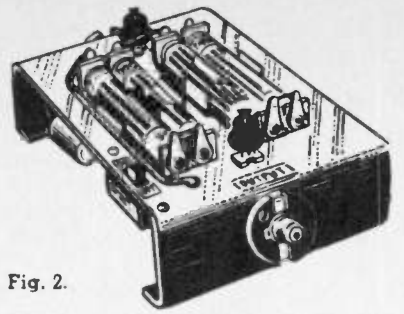


Fig. 2.

All of the component parts required for the preamp can be mounted on an aluminum chassis measuring as little as 3" x 4" x 1" if parts are selected carefully with an eye to small size. By building the unit in a small space, the preamp can be installed under the tone arm of many record playing setups. Hum pickup is, in most cases, sufficiently low to permit this installation technique if desired.

All of the parts needed to build this preamp are standard and may be obtained at any well-stocked parts house. If the builder wishes to take advantage of the convenience of a packaged circuit, Lafayette Radio, 100 Sixth Ave., New York 13, N. Y. is offering the preamp as a kit (KT-71) for $8.95 complete. -30-

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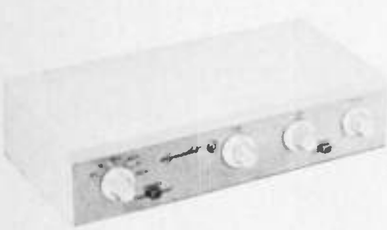

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Now, two years later, the AR-1 is widely accepted as a bass reference standard in both musical and scientific circles. There is general understanding of the fact that, due to the patented acoustic suspension design, the small size of the AR-1 is accompanied by an advance in bass performance rather than by a compromise in quality.

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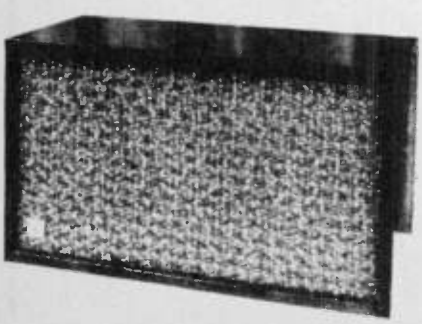
The AR-2 is the first application of the acoustic suspension principle to a low-cost speaker system. Prices are $89 in unfinished fir cabinet, $96 in mahogany or birch, and $102 in walnut.

We would like to suggest, as soberly as we invite comparison between the AR-1 and any existing bass reproducer, that you compare the AR-2 with conventional speaker systems which are several times higher in price. No allowances at all, of course, should be made for the AR-2's small size, which is here an advantage rather than a handicap from the point of view of reproducing quality.



Literature is available on request.

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Transistor Mike Preamps

Fig. 1. Crystal microphone preamplifier.

By **PAUL PENFIELD, JR.**

Two simple, easily built preamplifiers for connecting a crystal or dynamic microphone to your hi-fi system.

OFTEN the hi-fi enthusiast has some reason or other to connect a microphone to his hi-fi set-up. Some preamplifiers have provision for crystal mike input, but few can take a medium- or low-impedance mike. For this reason many are stymied. A good, cheap, simple mike preamp would be welcomed by these people.

Transistors seem a "natural" for this application, since there is no need for bulky, fragile, hot vacuum tubes, no hum problems, no need for high plate voltage, and no wasteful filament power. Accordingly the two preamps described here were designed and built. Several models of each have been built and used, and the design is considered free from flaws. The circuit is so simple that few if any mistakes can be made in wiring it, even by the most inexperienced constructor. The parts are small enough to fit almost anywhere.

Since a battery is used there must be some "on-off" switch. The reader is free to design his own "on-off" switch, terminal arrangement, and physical layout, to suit his own needs. Two of the amplifiers built by the author are shown here, but don't feel restricted to these types of housing or connections. Any reasonable layout will do. Be sure only to keep the input lead shielded.

The Circuit

The circuit is particularly simple because of the bias arrangement, known as a "floating base bias." Note that, because of the 10 μfd. capacitors in each of the two circuits (Fig. 2) no direct current flows through the base. At first thought it might appear as if we would be biasing the transistor at cut-off, but this is not so. Actually the collector is passing current just equal to $(\beta + 1)I_{co}$ where β is the grounded-emitter current gain, and I_{co} is the transistor cut-off current. This is made a little clearer by the collector family of the transistor, Fig. 3. Actually, Fig. 3 is not accurate for the 2N105, the transistor used, but was drawn merely to point out the action involved.

In Fig. 3, the current axis has been expanded considerably, so we can investigate the transistor at very low bias currents. Note, first, that the line corresponding to zero base current is *not* as low as we can go. If we send the base a bit positive, it will send a small current *into* the base. Normally, the base of a p-n-p transistor is biased negative, so current can normally flow *out*.

World Radio History

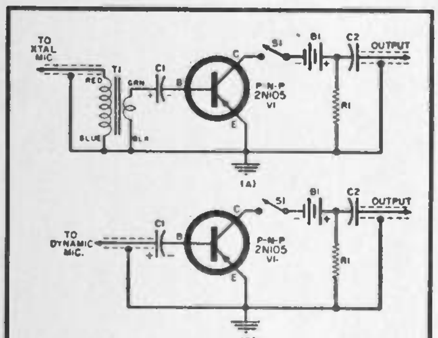
There is nothing particularly magical about the value zero for base current. We can get useful amplification on either side of the zero line.

The lowest point we can go on Fig. 3 (the bottom line) corresponds to a collector current of just I_{co}. At this value, the base current is the same, and the emitter passes no current. Because of the physics of the collector junction, no smaller current than I_{co} can be passed with reasonable voltage biases.

Fig. 3 corresponds to a very small portion of the transistor family—note that collector currents are in the range under half a milliampere. For many purposes, the difference between cut-off (the lowest we can go) and zero base current can be ignored—however not here.

A load line of 10,000 ohms has been placed on Fig. 3, showing the operation of the amplifier. A six-volt battery is used as a supply. The operating point is determined by the load line and the line of zero base current.

This explains the basis of the floating base bias. The d.c. collector current is just the cut-off current times approximately the transistor current gain, or $I_{co}(\beta + 1)$ with the base left floating. The signal comes into the base *via* a capacitor in each circuit, and varies the bias point slightly on either side of the zero base current point. The result is an amplified signal appearing across the load, as represented in Fig. 3 by moving slightly up and down the load line.



R1—10,000 ohm, ½ w. res.
C1—10 µfd., 6 v. elec. capacitor
C2—.05 µfd., 200 v. capacitor
S1—S.p.s.t. toggle switch
T1—Input trans. (Argonne AR-100, see text)
B1—6 volt battery (see text)
V1—2N105 "p-n-p" transistor (see text)

Fig. 2. Floating base bias preamps for (A) crystal and (B) dynamic microphones.

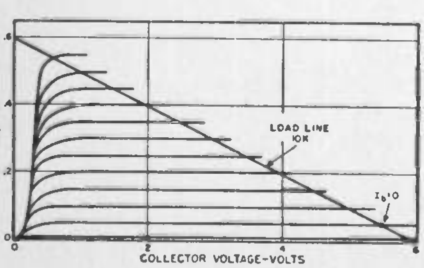


Fig. 3. View of collector family, showing gain below curve for zero base current.

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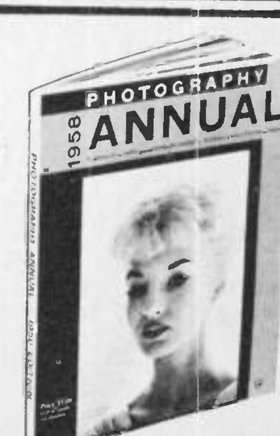

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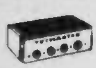

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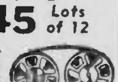

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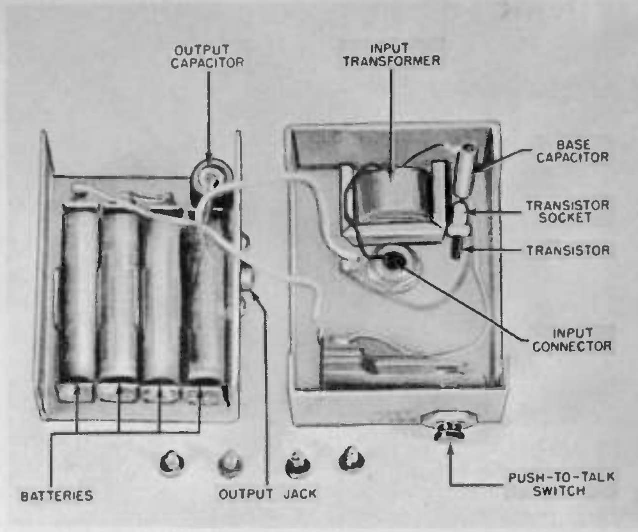


Fig. 4. Disassembled view of transistorized crystal microphone preamplifier.

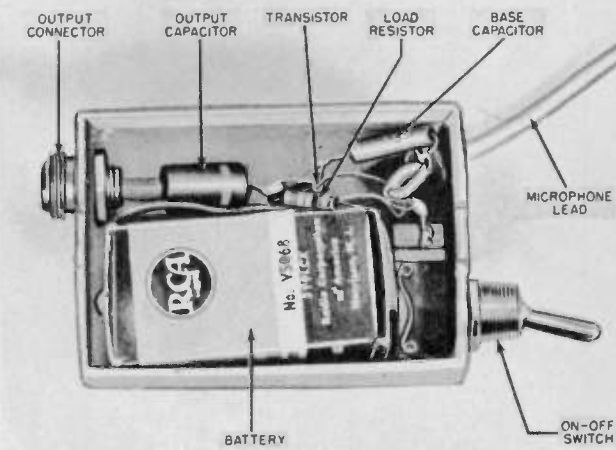


Fig. 5. Underside view of the transistorized dynamic mike preamp.

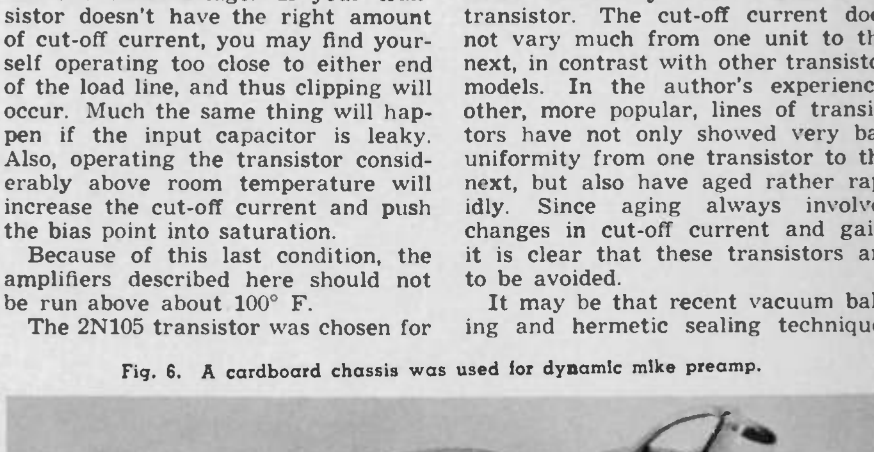
This type of circuit unfortunately has one disadvantage. If your transistor doesn't have the right amount of cut-off current, you may find yourself operating too close to either end of the load line, and thus clipping will occur. Much the same thing will happen if the input capacitor is leaky. Also, operating the transistor considerably above room temperature will increase the cut-off current and push the bias point into saturation.

Because of this last condition, the amplifiers described here should not be run above about 100° F.

The 2N105 transistor was chosen for these amplifiers because of the exceptional uniformity from transistor to transistor. The cut-off current does not vary much from one unit to the next, in contrast with other transistor models. In the author's experience, other, more popular, lines of transistors have not only showed very bad uniformity from one transistor to the next, but also have aged rather rapidly. Since aging always involves changes in cut-off current and gain, it is clear that these transistors are to be avoided.

It may be that recent vacuum baking and hermetic sealing techniques

Fig. 6. A cardboard chassis was used for dynamic mike preamp.



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will eliminate the aging headache by driving out the offending oxygen and water vapor, but still the manufacturing tolerances must be considered. The 2N105 is one of few transistors available today that meets the strict requirements as to cut-off current and gain uniformity demanded by the floating base amplifier.

Construction

As mentioned earlier, the physical layout of the amplifiers is up to the reader. The only requirement is that the input from the mike be shielded to keep down hum due to stray 60-cycle fields. The amplifier itself, running from a battery, is otherwise completely free from hum.

Figs. 1 and 4 show views of one of the author's units, designed specifically for the *Astatic* JT-30 crystal mike shown with it. Fig. 2A is the schematic of this amplifier.

The input connector not only makes electrical contact but holds the microphone as well. A standard male chassis connector was used with a female-adapting ring. A push-to-talk switch was used, because of the intended use of the amplifier.

The transformer (*Argonne* AR-100) and the battery holder (as well as the other parts) are available from *Lafayette Radio,* 165-08 Liberty Ave., Jamaica 33, N. Y. All the parts required fit very easily into the box shown (*ICA* type 29435).

The dynamic mike preamp (Fig 2B) was wired directly to the mike (*Lafayette* type PA-19). See Figs. 5 and 6 for details. It was made in a mailing box of cardboard; the holes for the toggle switch and output connector were punched out with a knife. This was strictly a "rush" construction job, and no care was taken in trying for neat wiring, etc. Nevertheless, the unit gave good performance until the case broke from wear. For a permanent amplifier, the author recommends something slightly stronger than cardboard!

With some microphones it might be possible to mount the transistor and other small parts right inside the mike case, eliminating all hum problems. In some installations, it may be practical to fasten the amplifier to the mike stand with some kind of clip. Alternatively, it could be built right inside the high-fidelity installation, even drawing power from the amplifier.

The 10,000 ohm load resistor in the circuit was chosen to provide adequate amplification with little chance of having trouble with high-temperature operation. On some microphones of low sensitivity, the preamp described here may not have enough gain. The author has not had any trouble along this line, but microphones have varying sensitivities, and possibly the reader has one partially dead, or otherwise insensitive. In that case, the load resistor can be increased up as high as 30,000 ohms with a little sacrifice in high-temperature operation.

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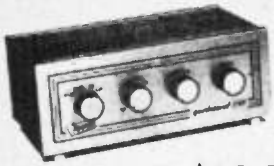

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Conversely, if the reader desires high-temperature operation but needs less gain, he can reduce the load resistance to 6000 or 7000 ohms.

If six volts is not convenient, the preamps will operate with good results from 4.5 volts or even 3 volts.

The preamps here are limited in their quality of reproduction mainly by the components. Distortion is low, since the transistor is operated in a low-distortion area. That is, it is low right up to the point of clipping, where it will clip sharply.

The frequency response is determined by the transformer and the capacitors. The size of the capacitors has been determined to allow adequate low-frequency response.

The amplifiers have been used for tape recording with good results, and for public address work with equally gratifying results. The author believes that you'll be pleased with these simple, useful floating-base mike preamps. —30—

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The circuit itself is built around three of the General Electric 2N107 junction transistors of the "p-n-p" type. The amplifier is powered by four 1½ volt penlite cells and operates in push-pull class B. Frequency response is from 100 to 8000 cps with an over-all power gain in excess of 50 db.

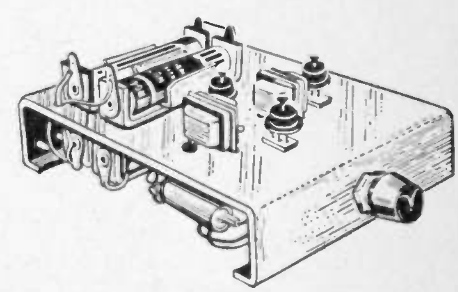
The three transistors are connected in a grounded-emitter type circuit which provides sufficient power to operate a good quality 4" to 6" PM speaker with a 3.2 ohm voice coil.

The transformers required for this construction are of the type especially designed for transistor circuitry. An important feature of such components is their high inductance and low d.c. resistance. Input impedance of this circuit is 1000 ohms. For high impedance inputs a matching transformer is required. For magnetic inputs a preamplifier is needed.

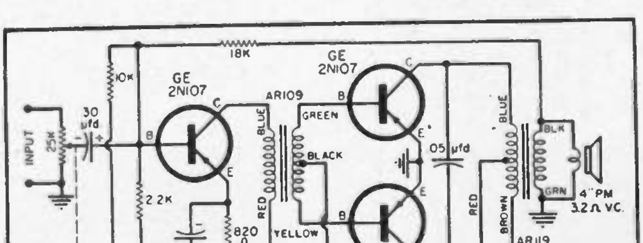
Since a feedback loop is employed, the necessary precautions in regard to phasing must be employed. Follow the color coding of the transformers carefully. Should oscillation occur after completion, reverse the black and green wires of the output transformer. The transformers, which are the only special components required to build this amplifier, are manufactured by Argonne Electronics Mfg. Co.

For those who wish to avoid the trouble of shopping for individual components, Lafayette Radio, 100 Sixth Avenue, New York 13, New York has a kit, Model KT-69, available for $17.95.

The primary purpose of this kit is for the experimenter, student, or anyone who wants to obtain some working knowledge of transistor operation. For all practical purposes, considering the price, a vacuum tube-type power amplifier would be more advisable. —30—



↑ Over-all view of the transistorized audio amplifier. Here, it is built breadboard style on a chassis measuring 3" x 4" x 1", although any arrangement of parts will do.



← Complete schematic diagram of audio amplifier. It uses three of the new G-E 2N107 transistors and two specially-designed transistor type transformers.

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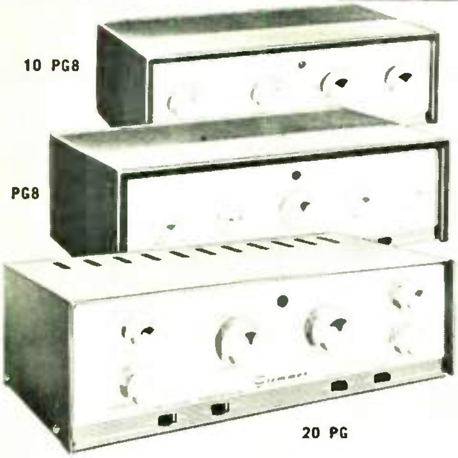

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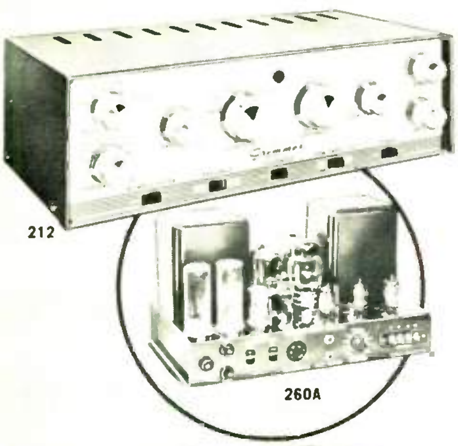

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