Transactions of the I·R·E

Professional Group on Audio

A Group of Members of the I. R. E. devoted to the Advancement of Audio Technology

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The Institute of Radio Engineers

I.R.E. PROFESSIONAL GROUP ON AUDIO

The Professional Group on Audio is a Society, within the framework of the I.R.E., of Members with principal Professional interest in Audio Technology. All members of the I.R.E. are eligible for membership in the Group and will receive all Group publications upon payment of prescribed assessments.

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A CONVERSATION ON LOUDSPEAKER ENCLOSURES*

Daniel W. Martin
The Baldwin Company
Cincinnati, Ohio

In this invited Editorial, written in a humorous vein, Dr. Martin indicates some of the sources of frustration of the Audio Engineer
— Editorial Committee —

Every engineer in the audio profession is at some time asked by a non-technical friend for advice on the construction of an ideal enclosure for a loudspeaker of somewhat dubious background and performance. One has two professional choices. He can pose as an oracle and write a complete prescription as if there were an unique solution. Or, he can try the educational approach and explain a few basic principles of cabinet design. Although the latter course is basically more honest, it does have its difficulties, as illustrated in the following conversation.

AL: "Say, Paul, I just picked up a fifteen-inch speaker at a bargain sale down at Joe's Radio Shop. It only cost me a dollar fifty-nine."

PAUL: "What kind did you get, Al?"

AL: "I don't know what it is, and there's no label on it, but I recalled that you were a member of the IRE Professional Group on Audio, so I came over to get some advice on how to build a really good cabinet."

PAUL: "My first advice would be to get a really good loudspeaker unit."

AL: "I'm afraid that would cost too much. This one will probably be good enough. I want to build just the right cabinet for a fifteen-inch speaker. Can you tell me exactly how to do it?"

PAUL: "There's no unique solution to your problem. It's something like building a house. The architect's solution depends upon the size and shape of the lot, the location of the street, the contour of the ground, the type of occupants and what they do, and the choice of materials which are both suitable and economical. The architect would want to know how large a lot you have. And I would ask, 'Do you have ten cubic feet or more available for the cabinet?'

AL: "Ten cubic feet? That sounds awfully big."

PAUL: "I know, but ten cubic feet isn't as large as it sounds. A cube twenty-six inches on an edge would give you the volume I suggest. Of course, you wouldn't want all of the dimensions the same, for acoustical and functional reasons, or even for appearance."

AL: "Why not?"

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* Manuscript received August 27, 1951.

PAUL: "Well, the acoustical result would be to bunch the box resonances together instead of spreading them along the frequency scale. This is a principle followed in the acoustical design of small rooms. Again there is no magic, unique solution, but I'd suggest dimensions of 20 x 26 x 33 inches as a starting point. You may deviate from those dimensions somewhat, and if you don't like the proportions of this combination, I'll calculate another good combination to approximate the proportions you'd prefer."

AL: "I don't understand this sort of approach. When I buy a vacuum tube, the tube manual gives circuit values to use. Why aren't your recommendations exact?"

PAUL: "There are several reasons, Al. In the first place, the tube manual values are not sacrosanct. They apply to only one set of operating conditions. Besides, the values suggested by the manual correspond to a particular type of tube. You didn't tell me anything about your loudspeaker, except that it has a fifteen-inch diameter. You wouldn't expect to find in the tube manual a set of suggested values for components in the circuit of just any triode. Some of the loudspeaker manufacturers now provide, with their units, an information sheet recommending cabinet volume, shape, and even complete design data on a cabinet known to give good performance with that loudspeaker."

AL: "The dimensions you gave are close to what I had in mind. I can put the record player, a radio, and some shelves for record albums in the cabinet, and use the space to advantage. Maybe I can leave enough space for a secret compartment."

PAUL: "Some of the things you would put in the compartment might rattle, Al, and you have misunderstood the purpose of the space behind the loudspeaker. If you fill up all of the space, the loudspeaker cone will be stiffened acoustically and the low-frequency response will be lost."

AL: "Maybe it would be better not to have a cabinet if it would make the speaker sound worse."

PAUL: "No, the loudspeaker without some sort of baffle wouldn't give you much low-frequency response either. Let's go back to the ten cubic feet of enclosed space behind the cone."

AL: "I can't use that much. I don't have enough material."

PAUL: "Oh, you already have the material. What kind did you get?"

AL: "It's a large sheet of quarter-inch fibre-board that was left over when my neighbor was insulating his attic."

PAUL: "I'm afraid it won't be rigid enough for good low-frequency response, Al. You should get some three-quarter inch plywood."

AL: "I have some, but I'm going to use it for a ping-pong table. Well, thanks, Paul, for all of your advice. It's been very helpful. I never would have thought there was so much theory to building a speaker cabinet. I'll let you know how it works when I get it built. Thanks again."

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JOE: "Al, did you ever hook up that speaker you bought at my annual bargain sale last year?"

AL: "Yes, would you like to hear it?"

JOE: "Say, that sounds great! What kind of a cabinet did you use?"

AL: "Paul helped me design it, I never did get it completely finished. I think he said to make the front panel three foot square. I finished that much and mounted the speaker, then set it on edge back there in the corner against the walls. It sounded pretty good when I hooked it up, so I just made a triangular top panel to pile things on, and left it there. Everyone likes it because it has so much bass. Paul certainly was a big help to me in the cabinet design. He's a nice guy, and what a brain! I couldn't understand him most of the time. He kept talking about low-frequency response."

THE TAPESCRIPTS COMMITTEE OF THE IRE-PGA*
Presents the Following Titles of Recorded Technical Papers

Andrew B. Jacobsen
University of Washington
Seattle, Washington

These papers are recorded on magnetic tape at $7\frac{1}{2}$ inches per second on 7-inch reels, and slides are $3\frac{1}{4} \times 4$ inches. Material will be sent express collect and must be promptly returned, prepaid.

"A Single Ended PP Audio Amplifier," Arnold Peterson and D.B. Sinclair, General Radio. 50 minutes. Proc. I.R.E., January, 1952.

"A Sound Survey Meter", Arnold Peterson, General Radio. 20-minute abstract. Proc. I.R.E., February, 1952.

"Microphones for Measurement of Sound-Pressure Levels of High Intensity over Wide-Frequency Ranges", J.K. Hilliard, Altec Lansing. 20-minute abstract. Proc. I.R.E., p. 214; February, 1952.

"A Method for Measuring the Changes Introduced in Recorded Time Interval by a Recorder Reproducer", J.F. Sweeney, Department of Defense. 15 minutes.

"An Instrument for Measuring the Time-Displacement Error of Recorders", E.N. Dingley, Jr., Department of Defense. 15 minutes. Proc. I.R.E., p. 214; February, 1952. (Dingley and Sweeney papers should be used together).

"Application of Electric-Circuit Analogies to Loudspeaker Design Problems", B.N. Locanthi, California Institute of Technology. 30-minute abstract. Proc. I.R.E., p. 214; February, 1952.

"Model 2-C Idiosynchrovisor", J.M. Henry and E.R. Moore, Boston, Mass. 11 minutes. A nonsensical satire on technical papers and jargon of specification writing. Will help to enliven dry technical meetings.

"A Block Diagram Approach to Network Analysis", T.M. Stout, University of Washington, Seattle, Washington. 30 minutes. Abstract as follows:

By treating voltages and currents as signals which may be applied to blocks representing network impedances or admittances, the block diagram techniques now used to analyze control systems can be applied in strictly electrical network problems. The basic rules are presented and a number of applications to ladder networks, parallel and lattice networks, and simple amplifiers are given. Advantages of the procedure include simplification of the labor required in solving network problems, reduced possibility of mistakes, and an increased appreciation of the role of various circuit elements in determining transient or frequency response.

When using recorded papers, the program chairman should preview material and be prepared to answer questions. Address requests to: Andrew B. Jacobsen, Electrical Engineering Department, University of Washington, Seattle 5, Washington.

* Received July 21, 1952.

P.G.A. PEOPLE

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From 1936 until mid-1951 he was associated with the Stromberg-Carlson Company, Rochester, New York; first as Engineer in the Telephone Laboratory; then from 1941 through 1946 as Assistant to the Vice-President in charge of Engineering; and from 1946 to 1951 as Chief Engineer of the Sound Equipment Division. He has recently accepted a position as Chief Engineer of the Caledonia Electronics and Transformer Corporation in Caledonia, New York.

Mr. Angevine is a Senior Member of the IRE, and was one of the founders and first Chairman of the IRE Professional Group on Audio. He has also served as Chairman of the Rochester Section and on the Committee on Audio Techniques. He is Chairman of the Sound Equipment Section of the Engineering Department of the Radio-Television Manufacturers Association, a Member of the Acoustical Society of America, a Member of the American Institute of Electrical Engineers, and a Member of the Rochester Engineering Society. He is the author of several papers on relays, sound equipment, and intercommunication systems.

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SPEAKERS AND TRANSMISSION OF SOUND WAVES*

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Summary

Considerations of this discussion involve the over-all design requirements for coupling devices to the air medium. The requirements are considered from the standpoint of the medium itself rather than reverse from the standpoint of the motor mechanism. Considering the requirements for linearity and coupling to this medium, certain speaker designs are evolved. Test speakers utilizing this information have been made which should bring speaker performance much closer to the quality of presently available amplifiers.

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The fundamental purpose of a sound reproducing system is to cause an air pressure wave to exist in the exact form and amplitude of the original, or which would exist if the observations were made at the position of the microphone.

To build a system approaching this ideal requires consideration of the following:

(a) The nature of hearing or how the ear hears.
(b) The intensity of the sounds, both instantaneous and average.
(c) Frequency or frequencies of the sounds.
(d) The characteristics of the medium carrying the sound (the air).
(e) The environment: the effect of the rooms, etc.

Involved in the reproduction of sound waves are additional factors of consideration such as:

(f) Coupling to the air medium.
(g) Resonance characteristics of the acoustic system, including the enclosure and/or horns.
(h) Frequency modulation and Doppler distortion.
(i) Directivity characteristics of speakers.
(j) Compression characteristics of air and acoustical distortion effects.

A great deal has been written about the nature of speech and music, the intensity and average power values, by Bell Laboratories and many other organizations. Conservation in communication requires a study and application of the facts concerning speech and music to provide the best service for the required cost. This means that many communications systems have purposely compromised and limited the transmission to effectively restrict the frequency range and the volume range. Distortion has been permitted. The limits to which the practice has gone is a function of the development

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* Presented at the Southwestern IRE Conference, May 17, 1952, in Houston, Texas. Manuscript received, August 18, 1952.

of the art and the cost of extended accomodation.

Since better equipment and more experience have been available, we find the old standards are no longer acceptable to the critical ear, especially if a truly improved system has become available for him to listen to. It seems reasonable, therefore, that despite certain limitations in the transmission media, a re-investigation into the requirements for faultless transmission might be worth while.

Accordingly, this discussion describes factors concerning the media through which the sound travels and determines from the limitations of these media what must be done to satisfy distortion limits as indicated by the critical ear in order to develop the illusion that the real sound source is present.

There has been a great deal written about the method or physics of hearing. The ear is a remarkable device having astounding accommodation and sensitivity. For instance, the ear at middle frequencies can hear a steady-state signal 10^{-13} acoustic watts power per square foot and can accommodate a steady sound level 140 db above threshold, or 10 watts per square foot at the ear. This is a power range of 10^{14}, or 100,000,000,000,000 (one hundred trillion) to 1. It will accommodate considerably more than this power value if the duration of the sound is short, perhaps 100 or 1,000 times as much. For instance, one who is hammering on steel or hard surfaces generates peaks of this higher order. The ear absorbs the sound wave in the fluid of the inner ear. It therefore acts as an integrator, or sums the energy under the curve, and can therefore stand much more intensity for short nonrepetitive periods than for continuous periods. All the known auditory curves showing "pain or feeling" contours are based on continuous sine wave tones for the average ear. The possibilities of greater tolerance for short duration peaks, such as are found in speech and music, are very real indeed.

This facility of the ear has been a great asset to the communications field because through the suggestion of sounds present, the ear and mind will seek to fill the gaps and help give the illusion of reality by reaching out and listening for the notes not present at the original volume, but this is not accomplished without loss or fatigue. Only with a balanced reproduction of adequate sound level, free of distortion, does one realize how great is the loss of an inadequate system. It is almost impossible to describe why "bi aural" sounds more realistic; however, it is apparent instantly when you listen. A third channel adds still more realism since it permits the illusion of depth as well as the two-dimensional concept.

The musical instruments and other sound devices determine the design requirements of speakers and amplifiers alike, and for realism, the reproduced sounds must be at equal intensity at the ear, as would occur if the listener were at the position of the microphone. This does not mean that the absolute sound power must be the same in the home as it is in the auditorium since there is a room gain to be considered, and the directivity index of the speaker system reduces the power requirement. Also, the listener is usually closer to the speaker in his home than he is in the auditorium. The inefficiency of speakers, as will be seen, however, does almost completely compensate for the gain factors and therefore the power requirement for the speakers is pretty well determined by the power capacity of the individual instruments for medium places.

Since the ear does not recognize loudness on the basis of the peak intensity of the sound signals, it is found that very little average power is required to produce adequate sound loudness. However, since the peaks of speech and music are from 200 to 900 times the average value, as shown by Sacia[4] and others, it is necessary to provide capacity in both speaker and amplifier to handle these peaks if true reproduction and low distortion are to be expected.

A bass drum delivers an acoustic peak power of some 100 to 200 watts, even down to 20 cycles. A cymbal crash delivers from 24 to 48 acoustic peak watts up to 15,000 to 20,000 cycles. The other musical instruments vary in peak intensity from a few milliwatts to 30 or 40 peak acoustic watts.

Since the musical and other instruments create sounds of substantial intensity at all audible frequencies, it is obvious that amplifiers and speakers alike must handle all the frequencies of the audio spectrum. Nearly everyone agrees that this is necessary.

Now, as to the methods of transmission, the coupling to the air and transient response — this has been perhaps the most underemphasized phase of the transmission of sound energy, yet a great deal is known about it. The air is a compressible medium. There are definite pressure limits if acceptable linearity is not to be exceeded. Coupling requirements do vary with frequency and the amount of air that it is necessary to move is more than 1,000 times at 20 cycles than at 20 kc for equal power outputs. "Doppler" distortion may be the major annoying distortion if too low and too high a frequency is present on the same surface, or the over-compression of the air may introduce 70 to 80 per cent distortion.

The design of an adequate system is one of making the best compromise of conflicting factors to satisfy the over-all design requirements. It is believed feasible to design a practical system having no more than 1 or 2 per cent distortion at the desired operating levels.

Now let us see what must be done so as not to exceed these limits of distortion,first considering the physics of the air, and then some aspects of speaker design to deliver the proper power at low distortion.

Since the air is the medium through which the sound must travel, the pressure wave generated in the air and transmitted to the ear is usually developed by some motion of an electromechanical device known as the loudspeaker. A recent unique development utilizes the ionized air to generate this motion without the aid of a diaphragm. Other speaker types consist of a compressed air column or flow which is modulated to generate sound waves. Regardless of the form or the nature of the source, a certain pressure must be developed to generate a specified sensation level in the air at a given distance.

Fig. 1 shows the acoustic power required from a point source at specified distances to develop the indicated intensities in db above threshold —

-10^{-16} watts per square centimeter, or 10^{-13} watts per square foot, approximately.

It will be seen that 100 acoustic watts are required to deliver +120 db above threshold at only 10 feet from the point source, and that 450 watts (acoustic) are required for the same level at 20 feet. It is not unrealistic to consider this as a minimum peak sound level for design purposes. These figures of acoustic power requirements may be reduced somewhat by factors of room gain and directivity characteristics. These will be discussed late.

At higher frequencies where accelerations are higher, the air appears to be more solid, or of a more rigid medium than it does at low frequencies. Consequently, very much smaller areas for the high frequency speaker are required for satisfactory radiated acoustic powers. If a very small speaker is utilized at the low frequencies, then an amazing amount of amplitude is required for the same power as radiated at the very high frequencies. As an example, it is found that the density of air times the sound velocity divided by the radiation resistance of speaker per unit area ($\rho c/R_a = 1$) will just equal 1 at some certain frequency and all frequencies above that. Below this critical frequency, the radiation falls off approximately 12 db per octave and the amplitude required below this point is inversely proportional to the square of the frequency. Whereas, above this frequency, the amplitude varies inversely with the frequency. It is desirable that areas of speakers satisfy this condition of $\rho c/R_a = 1$.

Fig. 2 shows that approximately 252 square feet of diaphragm would be required to satisfy this condition at 20 cycles. A 15-inch speaker satisfies this condition down to a frequency of 275 cycles. It is evident, therefore, that some compromise must be made to make something practical and sufficiently efficient. As can be seen from the above, the low frequencies in audio reproduction systems have been pretty small compared to the actual needs, and various gadgets of various kinds have been resorted to in an attempt to raise the efficiency at the low end, and for the most part, with only minor success.

The situation is similar to that of radiating antennas and corresponds to the aperture effect of an antenna. It can be said that there is no real substitute for aperture , either in loudspeakers or in antennas. However, good efficiency is obtained from antennas roughly 30° in height, and the same general logic would apply to loudspeakers. Consequently, investigations were conducted to determine amplitude requirements for speakers having practical areas.

One approach to solving the low frequency problem has been to utilize a horn of suitable length and mouth cross sectional area. The difficulty in horn design is that if it is driven from a 12- to 15-inch speaker, it should have a long axial length and a large mouth opening in order to satisfy the requirements at the lowest frequencies.

As a matter of comparison, the size of the mouth opening for an ideal horn and the area in square feet for the direct radiator diaphragm are equal for the same coupling effect to the air. It is obvious that such a horn would be enoumous in size, not likely to become part of the living room. Most considerations of horns or box enclosures assume absolutely rigid enclosing surfaces. These surfaces, if not rigid, tend to radiate sound as if they were the diaphragm, and because of the large area, such surfaces sometimes radiate more power to the air than does the projected diaphragm directly. This in itself would not be such a bad effect, except that the enclosing structure causes a delay in the reaction time of the sound wave and produces a hangover effect which continues the sound beyond that which is in the signal; in a word, contributes to the transient distortion.

This effect is much more serious than most people consider and is the main reason that nearly all such devices have a boxy or horn sound which is caused by the delay distortion of the chamber. A rigid horn or enclosure for a small speaker would require almost a 2- to 3-inch wall of concrete or other suitable material to make this enclosure the ideal that is assumed for it in speaker calculations.

Because of the above considerations, it is concluded that only with a large diaphragm having good coupling to the air and lowest possible mass could we hope to achieve an ideal coupling with a good transient characteristic from the low frequency speaker.

The amplitude required for speakers of practical dimension is shown in Fig. 3. It is apparent that reasonable sizes of speakers for low frequency operation can be realized with practical excursion requirements. It is apparent, too, that from 4 to 16 square feet is desirable for most applications if frequencies of 20 to 40 cycles are to be reproduced at desired levels.

Since low frequency sounds are more nearly sine wave in form, the loudness is greater for the same peak values because of the greater area under the curve or power content. Therefore, the desired intensity time function is satisfied at lower absolute peak powers.

"Doppler" distortion is another characteristic on which very little has been written, but which is certainly of major concern. Thousands of man hours have been put on the design of amplifiers,and units have been produced which have a very high degree of linearity both from a single frequency standpoint and intermodulation standpoint, while there is still serious intermodulation distortion in loudspeakers. This particular type of distortion is a form of frequency modulation and results from the presence of both low and high frequencies on the same diaphragm. The greater amplitude of the speaker due to the low frequency signal tends alternately to compress the high frequency wave and to stretch it out, which causes "Doppler" distortion. This "Doppler" distortion (see Fig. 4) of a 15-inch speaker can exceed the energy content of the high frequency signal if, for one instance, such frequencies as 30 cycles and 10,000 cycles are present at the same time. In other words, this distortion in practical commercial speakers might very well be 200 or 300 per cent, or more.

As will be seen in Fig. 5, for 1 per cent Doppler distortion, the 15-inch speaker can deliver 1 acoustic watt at 20 cycles and operate up to 32 cycles only. If a higher frequency is placed on this diaphragm carrying the low frequency component, then this distortion will exceed 1 per cent and goes up very rapidly. Further study of this diagram indicates that for a circular speaker carrying 20 cycles and delivering 1 acoustic watt, and with a frequency of 100 cycles also present, the diameter of a circular unit must be roughly 3.6 feet to keep the "Doppler" distortion at 1 per cent. In the lower diagram, the opposite consideration is shown for the same 1-watt output and 1 per cent "Doppler" distortion. The high frequency unit is considered to have 20,000 cycles on it, and for the various diameters, shows the lowest frequencies that it is possible to place on that diaphragm. It is evident from the above that without regard to any particular make or configuration of speaker unit, these distortion figures suggest the use of at least three sets of speakers operating at different portions of the audio spectrum. And here again we have evidence of the justification for very large diaphragm areas compared to what is normally considered necessary today for the low frequency unit. It turns out in practice that somewhere between 4 and 15 square feet of diaphragm is desirable, depending on the environment.

At the high end of the spectrum, the size of the speaker elements can be made quite small compared to the problem for the low frequency end. However, the mass of the voice coil and diaphragm becomes the real limit, and care is necessary to insure response at the upper limits of audibility.

Many speaker systems utilize a so-called "tweeter", which consists of a diaphragm operating into a restricted throat and emanating into an exponential or catenoidal horn of some form, usually with a honeycomb, or some baffle type of distribution. It is pointed out by many researchers on speaker design that the main merit of this horn is its relatively high efficiency, brought about primarily by the compression of air in the chamber immediately ahead of the diaphragm. This compression may exceed the linear pressure variation limit of the air and, under most circumstances, results in exceedingly high distortion, particularly if frequencies of widely separated values are present at the same time. Fig. 6 shows the distortion from such a speaker working into an infinite horn and delivering 1 watt per square centimeter of throat area.

The power output versus distortion per cent is shown for various ratios of the horn cutoff frequency to the frequency under consideration. For instance, if the tweeter is to cover from 500 to 20,000 cycles, this ratio is 100 to 1. It is foreseeable that for peak powers which occur in practice, this type tweeter is not very satisfactory for high quality sound reproduction unless the range and power are limited. There is, of course, a further disadvantage which has to do with the Doppler distortion mentioned above, especially when the tweeters are to cover wide frequency ranges.

Another characteristic is one concerning the environment, or the effects of rooms. It will be seen from Fig. 7 that in ordinary speech and music there is a substantial gain in a small room if the optimum reverberation time exists in that room. It must be said, however, that these gains are predicted on measurements made with sustained notes, and the reverberation time of rooms is determined by the decay rather than the build-up time. It is not known whether the full gain is realized for the short duration peak pulses of speech and music, because the peak has occurred and decreased to zero long before the room build-up time has taken place; therefore, the useful room gain might be considerably less than predicted by this method alone.

Doing a little arithmetic, we see that for speakers of probable efficiencies 2 per cent = $-20 \log_{10} 1/50 = -1.7 \times 20 = -34$ db. Room gain (see Fig. 7) for a room $12 \times 18 \times 10 = 2,160$ cubic feet, or $+25$ db, approximately. Directional gain due to corner of room, etc., $\equiv +8$ db, or total $= +33$ db. Therefore,power required into the speaker equals the power required in open space consideration plus 1 db (see Fig. 1). Now, this is a small room, and it is not difficult to see why very substantial amplifier powers are required for the adequate sound system even in the home.

It is interesting to note that independent checks by others[1] have indicated that the amplifier power requirement for reproduction of large orchestras varies from 20 watts to several kilowatts, depending on the size of the rooms.

Fig. 8 shows the calculated "Doppler" distortion for a speaker arrangement utilizing ten different elements. The "tweeters" are direct radiation 2-inch speakers arranged to distribute the frequencies above 2,500 cycles resulting in very nearly 180° distribution. Two 8-inch units were used for the intermediate frequencies. Four square feet were used for low frequency speaker.

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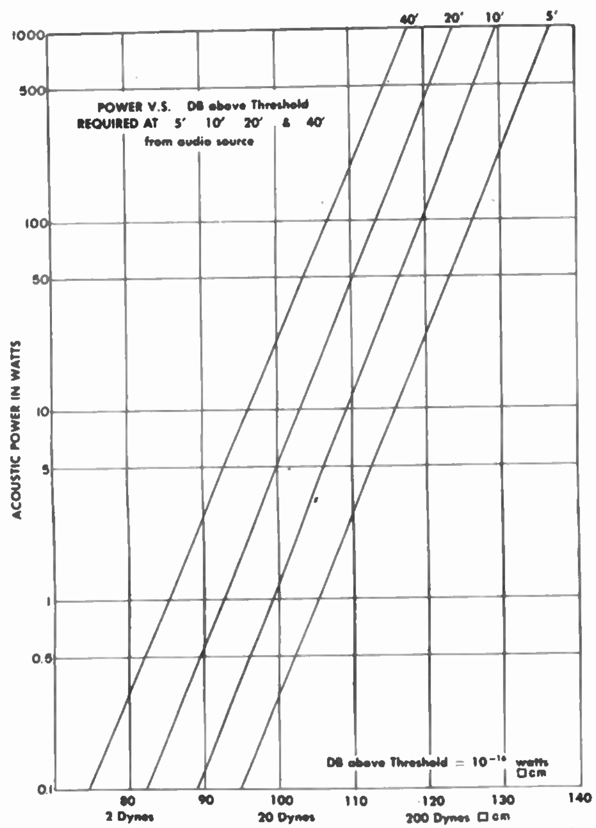


Fig. 1

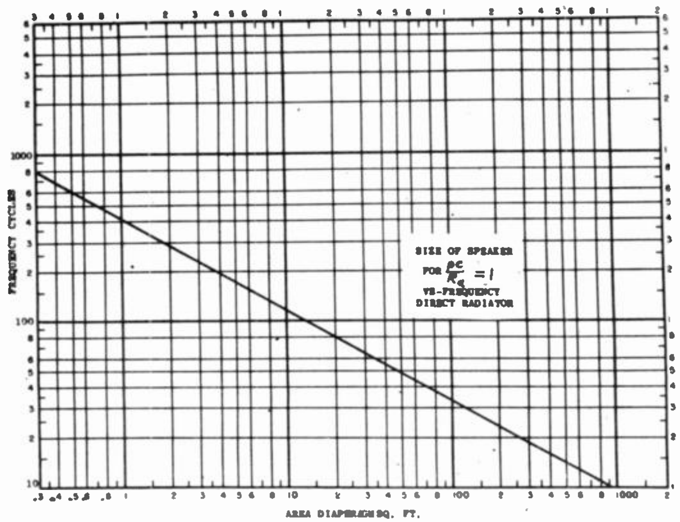


Fig. 2

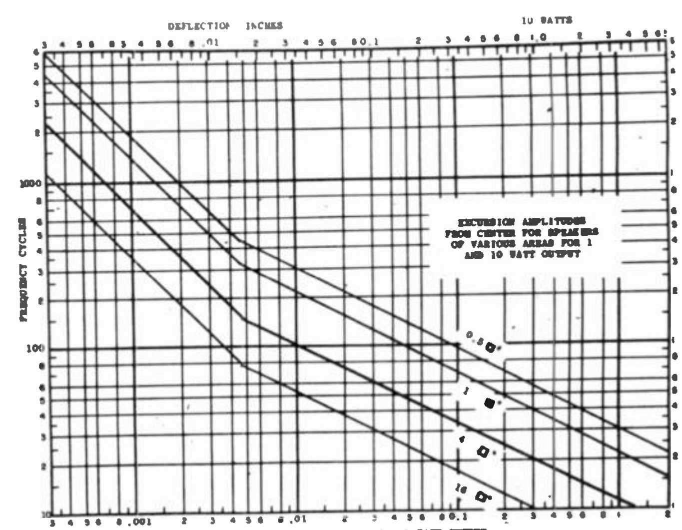
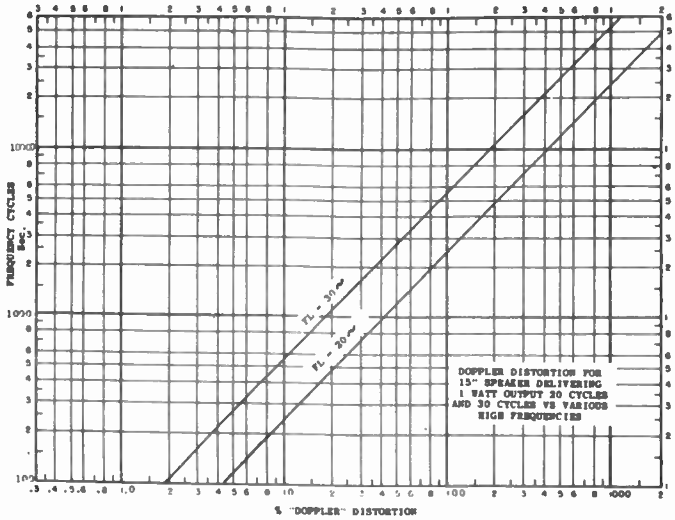
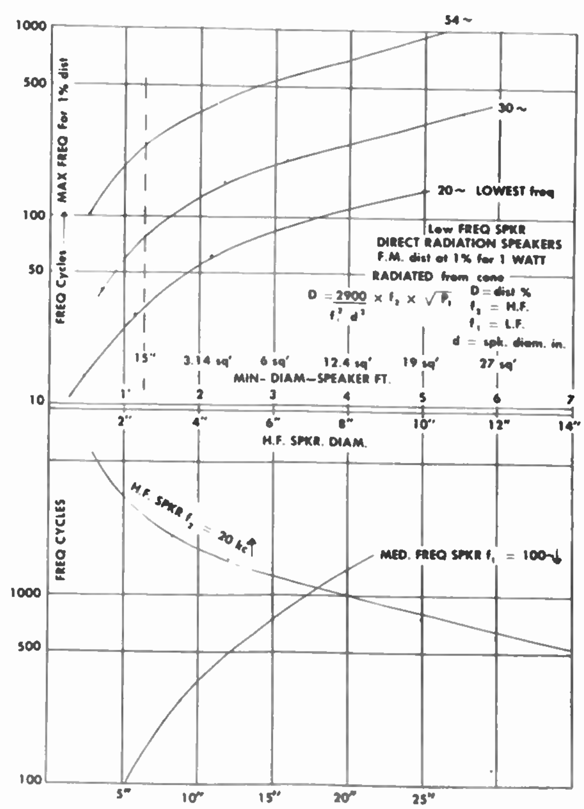


Fig. 3



FREQUENCY CYCLES Sec.
% "DOPPLER" DISTORTION
FL = 30~
FL = 20~
DOPPLER DISTORTION FOR 15" SPEAKER DELIVERING 1 WATT OUTPUT 20 CYCLES AND 30 CYCLES VS VARIOUS HIGH FREQUENCIES

Fig. 4



FREQ Cycles → MAX FREQ for 1% dist
54~
30~
20~ LOWEST freq
Low FREQ SPKR
DIRECT RADIATION SPEAKERS
F.M. dist at 1% for 1 WATT
RADIATED from cone
D = 2900/(f₁² d²) × f₂ × √P₁
D = dist %
f₂ = H.F.
f₁ = L.F.
d = spk. diam. in.
15" 3.14 sq' 6 sq' 12.4 sq' 19 sq' 27 sq'
MIN- DIAM—SPEAKER FT.
H.F. SPKR. DIAM.
FREQ CYCLES
H.F. SPKR f₂ = 20 kc
MED. FREQ SPKR f₁ = 100~

Fig. 5

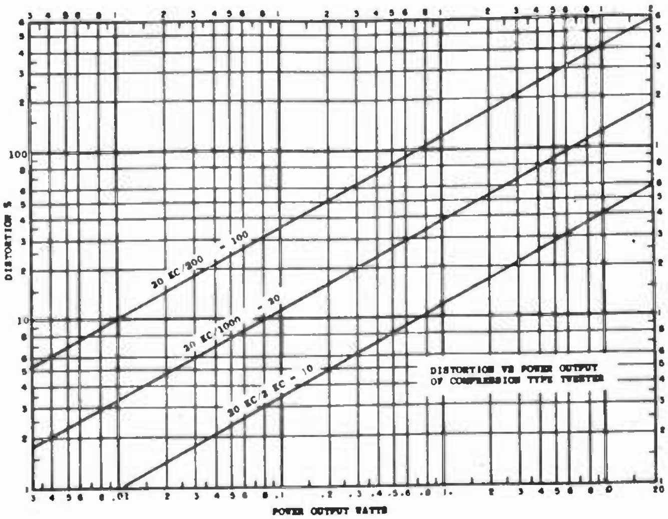


Fig. 6

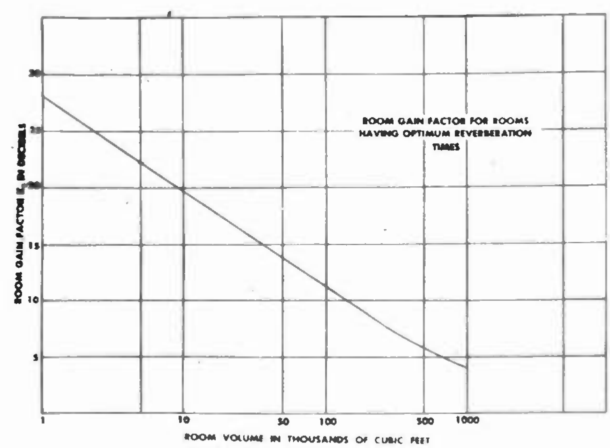


Fig. 7

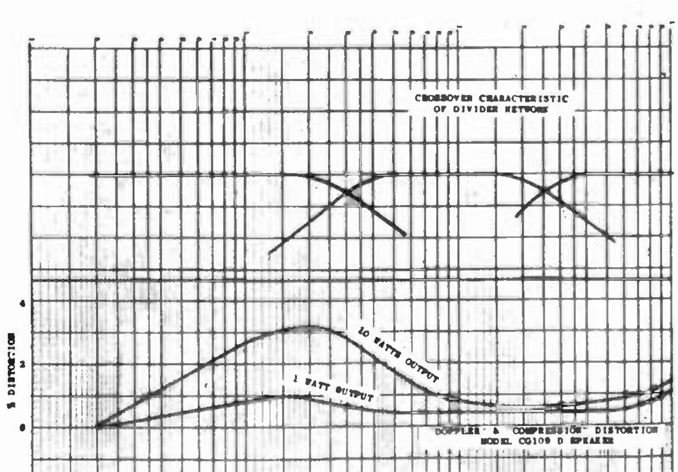


Fig. 8

INTERFERENCE EFFECTS IN MAGNETIC RECORDING HEADS*

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Chicago, Ill.

Summary

Irregularities in the frequency response of magnetic recording heads may be understood by considering the head to be made up of a multiplicity of gaps which, in playback, act as sources of flux with respect to the coil. The output from these sources differs in phase and amplitude and causes the irregularities, which are interference patterns. These sources of flux are the main gap and the edges of the pole pieces, which exist in all heads. In addition, in overlap-type heads, which are composed of laminations at right angles to the direction of tape travel, the spaces between the laminations act as sources of flux. When the outputs of these three different types of sources are added algebraically, a frequency response curve is obtained which is in good agreement with the experimental curve. By choosing the number of gaps and/or the spacing between them, the irregularities in the frequency response may be reduced, or various predetermined frequency responses may be approximated.

Introduction

The frequency response of a magnetic tape recording and playback head shows certain departures from smoothness, particularly at the lower frequency end. These irregularities are of interest in themselves, in addition to the practical improvement in the heads which may be achieved by an understanding of the reasons for their presence. This paper is an analysis of the reasons for these irregularities, and a discussion of means of controlling them.

Method of Analysis

The irregularities are almost entirely a function of the playback head only. This is shown in Fig. 1, where two heads are used -- a "smooth" response head and a head whose response has been purposely made as irregular as possible. In Fig. 1(a), the irregular head is used both for recording and for playing back its own signal, and, as can be seen, the result is irregular. In Fig. 1(b), the same with the smooth head, and the response is smooth. In Fig. 1(c), the response is recorded by the irregular head and played back on the smooth head, and it can be seen that the response is smooth, although there is evidence of slight irregularities. Fig. 1(d) shows that the response, when recorded on the smooth head and played back on the irregular head, is irregular. This simplifies not only the problem, but also the presentation, since hereafter, the behavior of the playback head only is of interest.

* Presented at the National Electronics Conference, Chicago, Ill.; September 29, 1952. Manuscript received September 3, 1952.

In the playback process, it is helpful to think of gaps as sources of flux which take magnetic flux from the tape and supply it to the coil. It has been found that in addition to the main gap, there are two other types of sources of flux in tape heads, and that these sources interfere with the output of the main gap and cause the irregularities. These sources of flux are: (a) The edges of the pole pieces (edge gaps).[1] (b) In tape heads in which the laminations are at right angles to the direction of tape travel, the spaces between these laminations act as sources of flux. These will be called auxiliary gaps. Since these sources of flux are displaced from the main gap, there will be a phase difference between their output and the output of the main gap. Their output will add to the main gap output at some frequencies, subtract at others, or, in general, interfere with the output and produce an irregular response.

Fig. 2(a) is a diagram of a type of tape recording and playback head. The two magnetic pole pieces (0.010 inch thick) are separated from each other by a metallic nonmagnetic spacer, which is commonly 0.0005 inch thick. This forms the main gap. In addition to the main gap, the head shown in Fig. 2(a) contains two auxiliary gaps (the spaces between the laminations) and two edge gaps. Fig. 2(b) shows a modification of the head in 2(a) in which there is only one auxiliary gap. Fig. 2(c) shows still another type in which the relative position of the gaps is changed. Fig. 3 is a section of the head shown in 2(a) The main gap is at A, the two auxiliary gaps at B, and the two edge gaps at C.

Let us first consider the analysis of a case where there are symmetrical edge gaps alone, displaced from the main gap by a distance b, shown in Fig. 4. Let the tape traveling over the head have impressed upon it a magnetic signal of wavelength λ, and be traveling across the head with velocity $v = x/t$, where x is the displacement.

The output from the main gap is:

$$e = A \cos 2\pi x/ \qquad (1)$$

where A is a variable which includes in it the 6-db per octave rise at low frequencies due to the velocity characteristic of the magnetic circuit and the decline in output at the high frequencies (short wavelengths) due to the gap effect.

Now, the output from the edge to the right is

$$e_2 = -C \cos 2\pi(x+b)/\lambda \qquad (2)$$

since the source is located at the point $(x + b)$. The minus sign in front of the term requires a word of explanation. It is due to the fact that there is an inherent 180° phase reversal at the edges, as shown in Fig. 4. It can be seen that an elemental magnet placed at the edge causes flux to flow through the coil in a different direction than a similarly poled magnet placed at the main gap.

Similarly, the output from the edge to the left is

$$e_3 = -C \cos 2\pi(x-b)/\lambda \qquad (3)$$

and the total output is the sum of (1), (2), and (3), or

$$e_{out} = A \cos 2\pi x/\lambda \; - C \left[\cos 2\pi(x+b)/\lambda + \cos 2\pi(x-b)/\lambda \right]. \qquad (4)$$

Summing:

$$e_{out} = \cos 2\pi x/\lambda \left[A - 2B \cos 2\pi b/\lambda \right]. \qquad (5)$$

This last equation is the useful result since it gives the frequency response in terms of the ratio of b, the pole width, to λ, the wavelength on the tape. This result is shown in Fig. 5, which is the actual frequency response of a head with pole pieces 0.020 inch in width (in the direction of tape travel) plotted at 3.75 ips.

The calculated first minimum for the above set of constants occurs at 167 cycles, which can be seen to be in good agreement with the observed value. At 15 ips, with $\frac{1}{4}$-inch pole faces, the first minimum occurs at 60 cps.

It should be pointed out that one method of minimizing this effect is to place the main gap unsymmetrically between the edges. Then the separate responses of the edges do not add to each other in phase. Also, by causing the tape to leave the pole pieces gradually, the magnitude of the effect may be decreased.

The frequency response of a head in which there are two 0.010-inch laminations on either side of the main gap (as shown in Fig. 2(a) is shown in Fig. 6(a). These laminations are arranged at right angles to the direction of travel.

Using an exactly similar method as that used for the two laminations on each side case, the response of the head may be written; let the lamination width be d, as shown in Fig. 3.

$$e_A = A \cos 2\pi x/\lambda \qquad (6)$$

$$e_B = B \cos 2\pi (x-d)/\lambda + B \cos 2\pi(x+d)/\lambda \qquad (7)$$

$$e_C = -C \cos 2\pi(x-2d)/\lambda \; -C \cos 2\pi (x+2d)/\lambda \qquad (8)$$

Summing these and performing some trigonometric manipulation yield

$$e_{out} = \cos 2\pi x/\lambda \left[A + 2B \cos 2\pi d/\lambda - 2C \cos 4\pi d/\lambda \right]. \qquad (9)$$

Note that this gives one term due to the edges plus a term due to the auxiliary gaps.

Fig. 6(c) shows the calculated response which is (9) plotted on a base of a smooth response head, i.e., a head in which the pole pieces are infinitely long. To obtain Fig. 6(c), A was arbitrarily given the value 1, B given $\frac{1}{4}$, and C given 1/8. This curve may be compared with Fig. 6(a), which is the actual curve obtained with a head of this type.

Another configuration of interest is one in which there are two 0.010-inch laminations on one side of the gap, and one 0.010-inch lamination on the other (for brevity, called 2 and 1, as opposed to the previous 2 and 2). This configuration is shown in Fig. 6(b). With this configuration, it is possible to obtain a smoother response than in the symmetrical case. The theoretical response of the heads, using the same notation and method as previously, is

$$e_{out} = \cos 2\pi x/\lambda \left[A + (B-C) \cos 2\pi d/\lambda - C \cos 4\pi d/\lambda \right]. \tag{10}$$

The actual response is shown in Fig. 6(b). It will be noted that the "holes" in the responses do not coincide as they do in the symmetrical case shown above in Fig. 6(a) and a substantially smoother frequency response is the result. Fig. 6 shows the response of the 2 and 2 structure in 6(a) as compared with the 2 and 1 structure in 6(b).

It is possible to calculate the response of any configuration of laminations, varying both in number and width, by this method, and many interesting combinations are the result. For example, the configuration in which three laminations are placed on one side of the main gap, and only one on the other (as shown in Fig. 2(c)) gives, both theoretically and practically, almost as smooth a response as the 2 and 1 structure used commercially.

It is also possible to build heads that have responses that are peaked to certain predetermined frequencies, for applications in telemetering, for example, in which certain discrete frequencies are recorded.

Conclusion

It is possible to obtain an analytical expression for the irregularities in the response of a tape head, making two assumptions: (a) the edges act as gaps; (b) spaces between the laminations act as gaps. The analytic expression obtained using these assumptions is in good agreement with empirical results, and may be used to predict the output of modifications of the basic lamination stack. In particular, it is possible, by varying the lamination width, number of laminations, and the spacing between laminations, to obtain responses which are peaked at certain frequencies, and/or have holes at certain frequencies.

Acknowledgment

I would like to acknowledge the helpful and stimulating conversations on the subject which I have had with other members of the Tape Recording Section of Shure Brothers, Inc., in particular, Mr. Lee Gunter, Jr., and Mr. Thomas W. Phinney.

Reference

[1] Clark and Merrill, "Field measurements on magnetic recording heads," Proc. I.R.E., vol. 35, pp. 1575-1579; December, 1947.

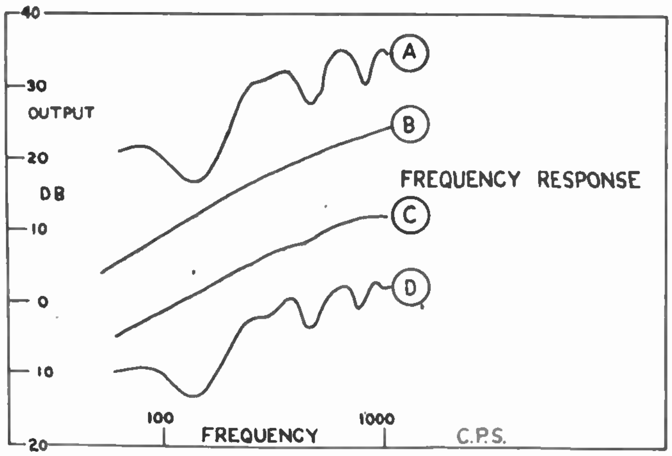

Fig. 1

Frequency response curves which show that the interference irregularities occur in the playback head only. See text.

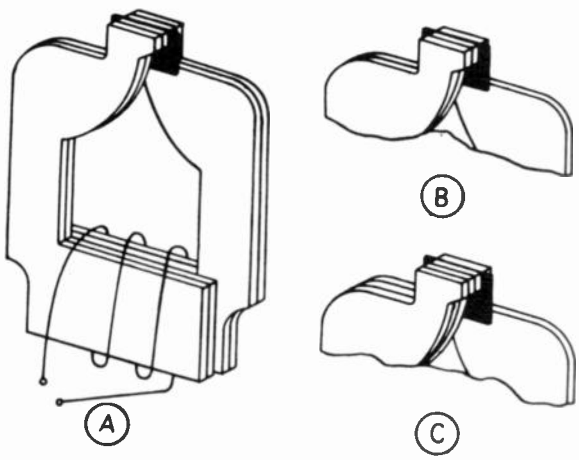

Fig. 2

Structure of the "2 and 2", "2 and 1", and "3 and 1" tape heads.

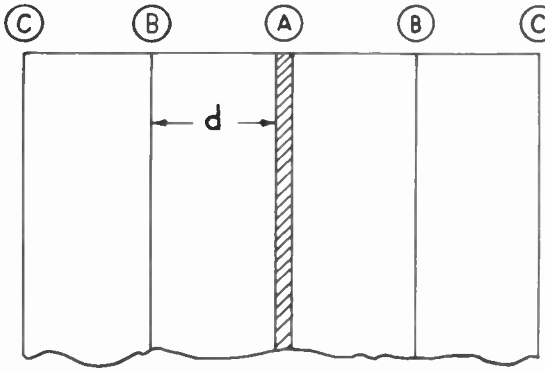

Fig. 3

Diagram showing sources in a "2 and 2" tape head.

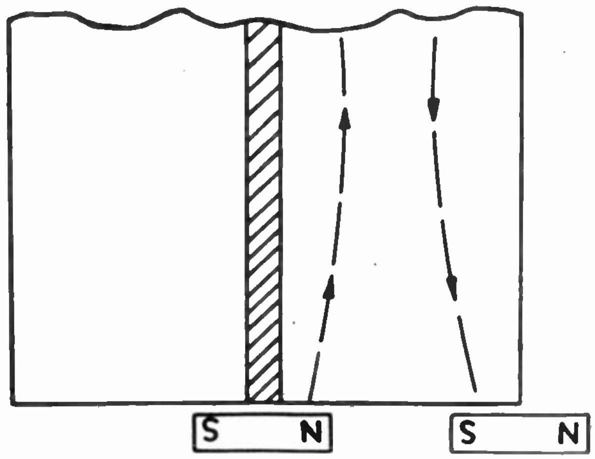

Fig. 4

Diagram showing a single pole piece on each side of the main gap.

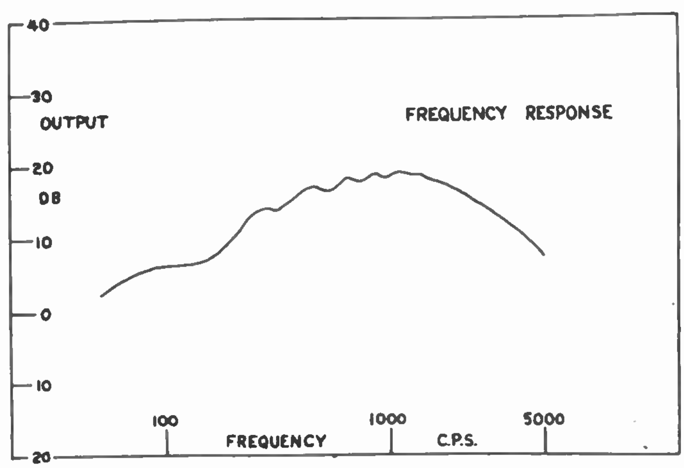

Fig. 5

Frequency response of the head shown in Fig. 4.

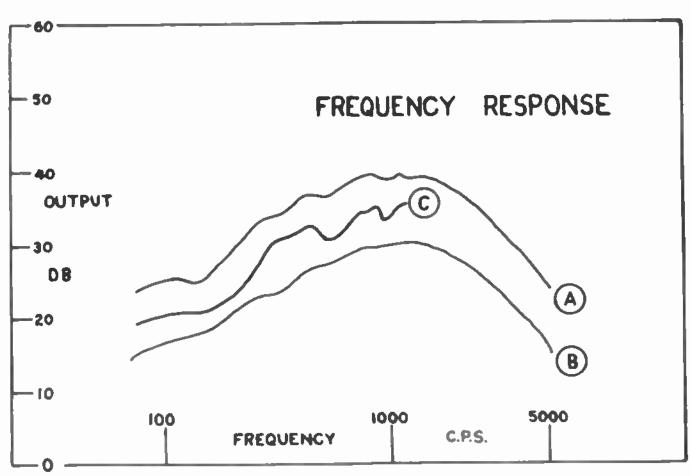

Fig 6

Frequency response of (a) "2 and 2" structure measured; (b) "2 and 1" structure measured; (c) "2 and 2" structure calculated.

INCREASED AUDIO WITHOUT SPLATTER*

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Summary

It is not generally realized, especially in cases where the Class B modulating transformer has poor regulation, that a higher ratio of audio output in the positive direction can be obtained if the secondary of the transformer has an additional asymmetric load placed on it through the use of a diode tube and appropriate resistor (see attached sheet).

If the negative voltage excursion of the Class B transformer secondary just equals the plate supply voltage, then the positive voltage excursion cannot be as far because in the first instance there is no load across this secondary at the end of the negative voltage excursion and consequently no voltage drop, while in the second instance there is a maximum load across the secondary as the voltage swings in the positive direction with its consequent IR voltage drop, and this subtracts from the positive swing.

Placing an extra load on the transformer secondary through the use of a diode and series resistor, so that this combination allows loading only on the negative voltage swing, provides a means for preventing the negative voltage excursion from exceeding the positive voltage excursion to any degree desired.

Adjustment of the resistor value can be such that the voltage and modulating power output of the Class B transformer secondary is even greater in the positive direction than in the negative direction, with consequent greater modulation effects without, however, exceeding 100 per cent modulation.

A summation of (E_1I) and (E_2I) will show that the requirement of a 50 per cent modulating power for 100 per cent modulation factor is not changed. However, it is interesting to note that there is a 16 per cent load on the Class B transformer secondary during the negative voltage excursion and an 84 per cent load on it during the positive voltage excursion with respect to the polarity of E_1. Thus, there already exists an asymmetric loading of the Class B transformer secondary during an audio cycle.

The addition of the extra load proposed in "B" during the negative voltage excursion, if of proper value, results in nearly perfect symmetry of loading of the Class B transformer secondary during the negative as well as the positive voltage excursions.

****

* Presented at the IRE Southwestern Conference, Houston, Texas; May 17, 1952. Manuscript received June 30, 1952.

Let us consider a Class C RF stage that is to be amplitude modulated. This Class C RF stage is capable of a linear plate current increase with a linear plate voltage increase from a value that may be considered to be the dc plate voltage E_b to twice this voltage, $2E_b$. Thus the plate input will go from X watts to 4X watts. Consequently, the stage being linear, it can be considered to represent a constant load R, determined by E_b/I_b, I_b being the average plate current, and E_b the plate voltage E_1.

If we now add the modulating transformer to the Class C RF stage in the usual manner, shown in A of Fig. 1, we can consider the secondary of the modulating transformer E_2 to be the equivalent of E_1 in voltage output, since 100 per cent modulation is to be achieved. By definition, 100 per cent modulation is achieved when $E_1 - E_2 = 0$, $I_b = 0$, and the output from the Class C RF stage is zero. Any time that $-E_2$ is greater than E_1, overmodulation occurs, no current flows through the Class C RF stage since it is a unidirectional valve, and the carrier power is interrupted.

Interruption of the carrier power can result in spurious side bands that interfere with other uses of those frequencies, and are to be avoided. These spurious frequencies are also generated when negative clipping is resorted to unless a filter is inserted in the circuit to round off the sharply clipped edges. However, it is usually impossible to re-establish the original wave form; thus, some distortion is invariably present in the modulated output. The only reason for negative clipping and subsequent filtering is the desire on the part of the radio amateur for an increased modulated carrier output during the positive voltage swing of the modulation transformer secondary while preventing the modulation transformer secondary voltage swing in the negative direction from exceeding the normally applied dc plate voltage, and consequently preventing the generation and emission of spurious frequencies.

It occurred to the writer that it should be possible to increase the emission in the positive direction of modulation without the need for clipping in the negative direction of modulation, thereby preventing both overmodulation and audio distortion. As an aid in visualizing the action of the modulation transformer secondary and its effect on the Class C RF stage, an analysis of the several instantaneous values was made, and the results shown as curves on cross section paper.

Several interesting items not previously detailed in text books became clear and obvious. The separation of the total power output due to the instantaneous sum of the dc plate and the ac modulation transformer secondary voltages into their component factors E_1 and E_2 is clearly shown in Fig. 1. It is also shown quite clearly that while the full $-E_2$ voltage is developed during the negative voltage generation of the modulation transformer secondary equal to the dc plate voltage E_1, there actually is no load on either of these two generators. Only when the full $+E_2$ voltage is developed during the positive voltage generation of the modulation transformer secondary equal to the dc plate voltage E_1 are these separate generators fully and equally loaded.

The load curves for generators E_1 and E_2 are clearly a function of their instantaneous voltages and the instantaneous current I in the circuit system. Obviously, E_1 is constant, I varies from 0 to 2I, and E_2 varies from 0 to the value of E_1 additive or subtractive. The load curve for generator E_1 is therefore a straight line between 0 and 200 per cent load, while the load curve for generator E_2 is a curve in the positive load direction from 0 to 200 per cent and a curve in the negative load direction from 0 to a maximum of 25 per cent load and continuous toward zero load at its maximum negative voltage swing. The sum of these two generator loads is, of course, $I\ (E_1 \pm E_2) = I\ ^2R_1 + I\ ^2R_2 = (E_1 \pm E_2)^2/R$.

We have now brought R, R_1, and R_2 into our analysis, and it is interesting to note that the resistances across generators E_1 and E_2 vary as the sum of $E_1 \pm E_2$ varies. When the modulating voltage is zero, the system R obviously is the value E_1/I and represents the Class C stage load at the average carrier point shown in the graph. However, when the generator voltage E_2 equals generator voltage E_1 and is additive, the resistance R divides into R_1 and R_2, and each generator sees $\frac{1}{2}R$. This can be more readily understood if we analyze Figs. 2(a) through 2(c). When E_2 equals E_1 but is subtractive, I equals zero, generator E_1 sees a positive infinity resistance, and generator E_2 sees a negative infinity resistance and the sum therefore equals zero. However, the sum of the separate resistances that generators E_1 and E_2 see is always equal to the value R at all other points on the varying $E_1 \pm E_2$ voltage curve.

Having noted that the generator E_2 sees no load when its voltage equals the voltage of E_1 but is subtractive and sees $\frac{1}{2}R$ when its voltage equals E_1 but is additive, it follows that no voltage drop occurs in the first instance due to the losses that occur in the modulating transformer secondary winding, constituting its own IR drop, and since this IR drop subtracts from the positive voltage swing, if we can still equal E_1 at this point, we must somehow prevent the negative excursion from exceeding E_1 by the amount of the IR drop, otherwise we exceed 100 per cent modulation.

We have now come to the crux of the whole analysis. We require a means to produce an IR drop in the modulation transformer secondary when its voltage swing is in the negative direction to equal the IR drop when its voltage swing is in the positive direction. A diode tube with a resistor in series, connected across the modulating transformer secondary so that current flows in this auxiliary circuit only when the voltage swing is in the negative direction, is therefore indicated. The value of the resistance we have already determined to be $\frac{1}{2}R$, while its watt rating should be $I^2R/4$.

The loading on the modulation transformer secondary when its voltage swing is in the negative direction will now be as indicated by the dotted line in Fig. 1. It will be noted that the modulator system now has a symmetrical loading during the negative as well as the positive voltage swing, resulting in reduced second-harmonic distortion, full modulation on the positive voltage swing, and prevention of overmodulation on the negative voltage swing.

Table I

$\%E_1$	$\%E_2$	$\%E_1 + E_2$	$\%I$	$\%R_1$	$\%R_2$	$R = R_1 + R_2$ %
100	0	100	100	100	0	100
100	20	120	120	83.3	16.7	100
100	40	140	140	71.4	28.6	100
100	60	160	160	62.5	37.5	100
100	80	180	180	52.4	47.6	100
100	100	200	200	50.0	50.0	100
100	-20	80	80	125	-25	100
100	-40	60	60	166	-66	100
100	-60	40	40	250	-150	100
100	-80	20	20	500	-400	100
100	-100	0	0	00	-00	0

Table II

Generator 1				Generator 2			
%E_1	%I	%Po			%E_2	%I	%Po
100	0	0			0	100	0
100	20	20			20	120	24
100	40	40			40	140	56
100	60	60			60	160	96
100	80	80			80	180	144
100	100	100			100	200	200
100	120	120			-20	80	-16
100	140	140			-40	60	-24
100	160	160			-60	40	-24
100	180	180			-80	20	-16
100	200	200			-100	0	0

Table III

%$E_1 + E_2$	%I	%Po
100	100	100
120	120	144
140	140	196
160	160	256
180	180	324
200	200	400
80	80	64
60	60	36
40	40	16
20	20	4
0	0	0

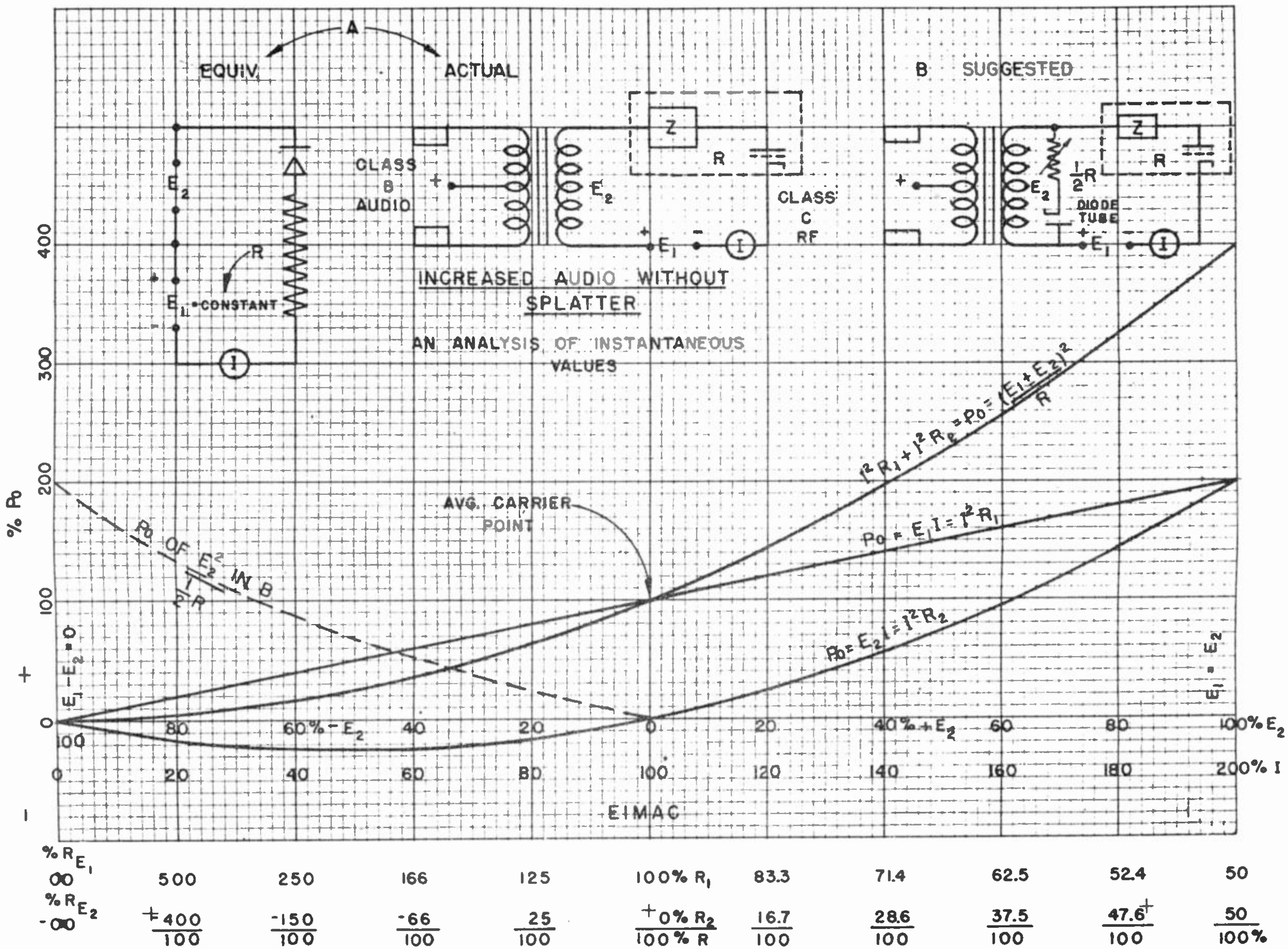
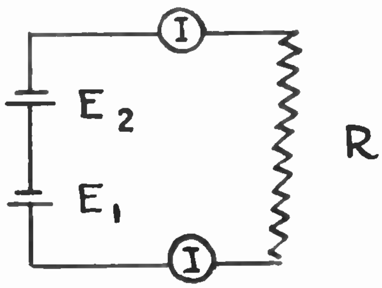

A
EQUIV.
ACTUAL
B SUGGESTED
CLASS B AUDIO
CLASS C RF
DIODE TUBE
E_1 = CONSTANT
INCREASED AUDIO WITHOUT SPLATTER
AN ANALYSIS OF INSTANTANEOUS VALUES
AVG. CARRIER POINT
$I^2R_1 + I^2R_2 = P_0 = \frac{(E_1+E_2)^2}{R}$
$P_0 = E_1 I = I^2 R_1$
$P_0 = E_2 I = I^2 R_2$
P_0 OF E_2^2 IN B $\frac{1}{2}R$
$E_1 - E_2 = 0$
$E_1 = E_2$
% Po
60% − E_2
40% + E_2
100% E_2
200% I
EIMAC
% R_{E_1}
% R_{E_2}
100% R_1
+0% R_2
100% R

Fig. 1



$RI = E$

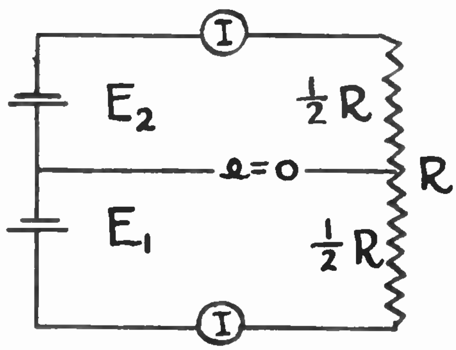
(a)

$E_1 = E_2$

$E = E_1 + E_2$

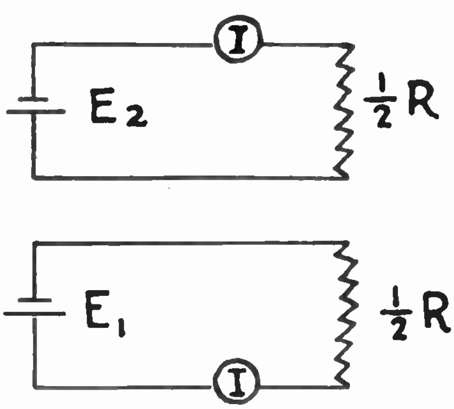
$R = \dfrac{E_1 + E_2}{I}$

I = Constant



(b)

At $\frac{1}{2}$ R exists a zero voltage point with respect to the connection between E_1 and E_2. I has not changed.



$\frac{1}{2}RI = E_2$

$\frac{1}{2}RI = E_1$

(c)

The two circuits can be separated without changing I, and each generator sees but $\frac{1}{2}$R.

Fig. 2

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