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IRE TRANSACTIONS® on AUDIO

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The Editor's Corner

You have not seen guest editorials in previous issues for one good reason—we could not get anybody to write them. So, this first month, we wrote some on a do-it-yourself basis. In addition to the short-sermon type of editorial, we would like to see a few which deal with the lighter side of audio engineering. If any pop into your mind, wrap them up and send them to the editor, so that others will enjoy them.

The Red Aluminum Jar is Full of Nuts

The old Model 50 wire recorder, built during the war, would operate at rewind speed comparable to its forward speed. By disconnecting the muting switch we could listen to music backwards, as well as to backward talk. This backward talk had many interesting (but not useful) applications. We convinced one stenographer that we had a translating machine. We recorded her voice in English, and by flipping the lever, turned it into Japanese.

Someone discovered that if he imitated the backward talk, and recorded this gibberish, it made sense when played in reverse. A few of us acquired a handy vocabulary of swear words, although the accent was always wrong, until you learned how to snarl backwards properly. Our champion was Jack Kemp who could dash off things like "Stun vuh luf si raj munimoola dare ed," and have it come back clearly "The red aluminum jar is full of nuts."

It is interesting that duplicates of tape recordings are now regularly recorded while running backwards, from a master which plays in reverse. The copy record is thus in a ready-to-play condition on its spool, without a rewind operation.

Negative Feedback—Something for Nothing

In the era BH (before high-fidelity), many enlightened radio engineers, in their spare time, would install negative feedback around the output pentode of inexpensive radio receivers for themselves and for their associates. Some of my friends had favorite circuits; one, I remember, required adding only a single resistor from output plate to driver plate.

We generally noticed a real improvement in quality. Distortion went down, frequency response was flattened. Also we got rid of shrieking caused by inductive load of the loudspeaker at high frequencies. Here was a case of getting something for (almost) nothing.

I remember a discussion among ourselves in which we wondered why receiver manufacturers did not avail themselves of these benefits. It could not be cost, because usually more parts were eliminated than added. It might be inadequate gain, but any set we ever ran across had more than enough reserve. Possibly it was inertia, resistance to change, or even lack of knowledge. One of us thought that there must be patents which were not available to the industry.

Almost two decades have passed. Inexpensive audio is still found in high-priced equipment, notably television receivers. With few exceptions, the circuits have not changed. Can anyone shed light on the true reason why negative feedback is not universal in output pentode circuits?

Complete Cycle

"Handiest thing I've seen for a long time." My young assistant held the new gadget so I could see it. He continued to enthuse, "Someone could introduce these to labs and make a lot of money. They are especially good when you have to make a lot of soldered connections—like when you're wiring a chassis. . . . You know, these darned soldering guns we've been using . . . heavy . . . clumsy . . . got to heat them up every time you make a connection."

I looked up, saw a once familiar instrument, and sighed. Before the war we had an American Beauty at every bench. She would be on all day. Heated up the lab summer and winter. Used to burn our knuckles if we did not watch out. When we did need her, she was usually corroded.

Then came an invention. Cold all the time, but heated in five seconds whenever you wanted. Could lay on the bench all day. No bother. No maintenance. No corrosion. Pretty soon that is all we had—soldering guns. The new generation did not know anything else. No wonder Bob was excited when he rediscovered a soldering iron.

Marvin Camras

PGA News

CHAPTER NEWS

Chicago, Ill.

Murray Crosby was the featured speaker at a Chicago Section meeting held Friday, January 9, 1959, in the Western Society of Engineers Building. The Long Island inventor spoke on: "Performance Characteristics of FM Multiplex Stereo Transmission." According to *Scanfax* of January, 1959:

"Murray G. Crosby received his B.S.E.E. from the University of Wisconsin in 1927, and an E.E. degree in 1943.

"For 20 years after his graduation from college, he was research engineer for RCA Laboratories, at Riverhead, L. I., N. Y. After leaving RCA Laboratories, he was associated with Paul Godley Co., consulting engineers, for three years. In 1948 he set up his own business, Crosby Laboratories, Inc., which now is located in Syosset, L. I., N. Y.

"Mr. Crosby is a Fellow of the Institute of Radio Engineers, Fellow of the American Institute of Electrical Engineers, and a licensed professional engineer in New York state.

"He is the author of about 20 technical articles on frequency and phase modulation, and holds some 180 patents in that field.

"Abstract of the general paper: Measurements on typical FM tuners in conjunction with a stereo FM multiplex adapter are given which show the signal-to-noise ratio characteristics of the main channel, the subcarrier channel, and the stereo channel.

"These data are given for the standards which provide the optimum performance of an FM station as a stereo broadcasting station and also for the proposed systems which apply two subcarrier channels, one of which provides stereo information and the other a subsidiary communications channel.

"Recommended standards with respect to program crest factor, pre-emphasis tolerance and modulation monitoring are given for the compatible system using optimum standards."

For February 13, 1959, the Chicago Section has scheduled a paper on "Electronic Telephone Switching," by R. W. Ketchledge of Bell Laboratories.

Dayton, Ohio

John S. Stanton, *Chairman 1958–1959*, reports the following formal meetings:

October 2, 1958—"Special Problems Encountered in Doppler-Shift Tracking of Earth Satellites," Lloyd Root, Director of Research, George Behm and Co. For demonstration, tape recordings of Sputnik and U. S. signals were played. The attendance was 60.

November 13, 1958—"Measurements and Demonstration of the Roberts Recorder," E. O. Valentine and J. S. Stanton, WADC and Monsanto. The attendance was 50.

The Dayton Chapter also joined with the Section to present Cyril N. Hoyler from RCA who spoke on "Electronics in Solids, Space and Sound" on December 11, 1958. As announced in the December, 1958 *Waveguide:*

> An event awaited with considerable interest takes place December 11 when RCA's Cyril N. Hoyler gives his famous presentation on "Electronics in Solids, Space and Sound."
>
> The program, a tremendous success everywhere it has been presented, is the highlight of the joint IRE-AFCEA dinner meeting, December 11. It is also the annual Section Ladies Night.
>
> One of the new electronic developments of great promise for the future is the RCA Electronic Music Synthesizer, developed under the direction of Dr. H. F. Olson. This electronic system, operating from coded information fed into its circuits by means of perforated tape, can generate any tone that can be achieved by the human voice or any conventional instrument, as well as totally new tones never heard before.
>
> Mr. Hoyler demonstrates the principles of the synthesizer with special equipment, and plays several selections that have been electronically synthesized in the style of various instruments.
>
> As a musical prelude to Mr. Hoyler's presentation, RCA stereophonic disc recordings are played during the half hour before the start of the demonstration lecture.

The Dayton Chapter also holds informal meetings in members' homes. The first such meeting for 1958 was a listening session on "Stereo-Tape vs Disk" presented by Donald Spangler, Monsanto, on September 18, 1958, with an attendance of 10.

San Francisco, Calif.

As announced in the *San Francisco Guide*, October, 1958, the new PGA officers are:

Don Broderick, *Chairman*

Don Broderick, Hewlett-Packard Co., *Chairman.* A research and development engineer in the laboratory of Hewlett-Packard Co., Broderick was formerly a test engineer with that company following his work with Boeing Airplane Co. in Seattle. His present work is principally concerned with power supplies and pulse circuits and devices.

Born in Chico, Calif., in 1928, he attended high school in Redding and enlisted in the Navy in 1945 to serve as electronic technician's mate.

On his discharge he attended the University of California, being graduated with the B.S. in electrical engineering in 1950. He has since done graduate work in electrical engineering at Stanford. He is a member of AIEE.

Lambert Dolphin, *Vice-Chairman*

Lambert Dolphin, Jr., Stanford Research Institute, *Vice-Chairman.* Born in Shoshone, Idaho, Dolphin received the A.B. in physics from San Diego State College in 1954. His program of study included considerable work in electronics as well as general physics, with a minor in mathematics. He has since completed two years of graduate work in physics at Stanford University with a minor in electrical engineering.

Joining Stanford Research Institute in June, 1956 as a research physicist in the special techniques group of the Radio Systems Laboratory, he has been engaged in radio propagation studies. He is a member of Sigma Phi Sigma.

Larry Johnson, *Secretary-Treasurer*

Lawrence W. Johnson, Hewlett-Packard Co., *Secretary-Treasurer*. Beginning his career as a research assistant in the nuclear research center at Carnegie Institute of Technology, Johnson was concerned with design of equipment for nuclear research, having obtained the B.S. in physics at the same institution. In the following year he earned the M.S. in electrical engineering.

His next post was as section head for special products with the York division of Bendix Aviation Corp., York, Pa. This involved automatic test gear for missile use. Since 1956 he has been a development engineer at H-P on design and production liaison for cathode-ray oscilloscopes.

A native of Washington, D. C. Johnson served with the U. S. Army from 1944 to 1946, during which time he spent six months studying electrical engineering at Stanford. His memberships include Sigma Xi and Tau Beta Pi.

October 27, 1958—"Report on Loudspeaker Research Being Conducted at Ampex Audio," Ward Widener, Ampex Audio. The attendance was 35. The latest Ampex loudspeakers were demonstrated, and a tour was made of the Ampex plant, including their anechoic chamber.

November 19, 1958—"FM Stereo Multiplex Transmission," Murray Crosby, President, Crosby Laboratories. A joint meeting with the Professional Group on Broadcast Transmission Systems.

Syracuse, N. Y.

November 6, 1958—"CBS-Television New York Tape Facilities," was presented by K. B. Benson and P. E. Fish, CBS, at a meeting held in WSYR-TV studio. Attendance was 200. An Ampex videotape unit was demonstrated by A. J. Eicholyer, Chief Engineer.

Twin Cities, Minn.

A PGA Chapter is being organized by Robert L. Sell of the Audio Development Co.

ANNOUNCEMENTS

The IRE National Convention Record, particularly that of the Spring National Convention in New York and of WESCON, will no longer be sent automatically to members of professional groups. Instead, we plan to publish PGA convention papers in these Transactions. We believe this will result in earlier publication of papers which are submitted well in advance of presentation; and the papers will appear in better form. In the past, an author of a convention paper ran into a situation where his paper could not be accepted by Transactions because it was distributed to PGA members in photocopy form as part of the Convention Record. This resulted in justice not being done to some of the most important papers and authors.

These Transactions are always pleased to receive manuscripts to be considered for publication, whether or not the material will be presented at a convention. Please send them in care of the Editor. Some advantages of publishing in Transactions are:

1) It is the official journal on audio of the Institute of Radio Engineers and its Professional Group on Audio.

2) It is a recognized scientific journal with a reputation for highest publication standards. It is carried by scientific libraries throughout the world, and its articles are reviewed in *Science Abstracts*.

3) It is published by the same staff and has the same high-quality printed "make-up" as PROCEEDINGS OF THE IRE.

4) Reprints of every article are made available to authors in any quantity, at a nominal cost.

5) Papers are reviewed by specialists. Often they come across minor revisions or corrections, which are suggested to the author before publication. This insures that the printed paper is relatively error-free, and one in which the author can take pride.

6) TRANSACTIONS are received by approximately 5000 engineers who are members of the IRE as well as of the Professional Group on Audio.

7) Papers may be republished in other journals such as *Electronics*, *Tele-Tech*, *Audio*, etc.

8) Authors become eligible for the following IRE awards, usually made each year: Browder J. Thompson Memorial Prize, Harry Diamond Memorial Award, Editor's Award, PGA Award ($100.00), and PGA Senior Award ($100.00).

Performance of Enclosures for Low-Resonance High-Compliance Loudspeakers*

JAMES F. NOVAK†

Summary—**This paper analyzes the low-frequency behavior of the vented loudspeaker enclosure and the pressure-tight closed box. Inherent interrelationships of the speaker-amplifier system Q, efficiency and response balance are discussed. A method is described for design of small enclosures with very low-resonance high-compliance loudspeakers.**

The author concludes that the vented enclosure can have greater acoustic output for a given amount of distortion, lower total harmonic, intermodulation, and transient distortion than a completely closed box of similar size.

List of Symbols

A_d = effective speaker diaphragm area—meters².
A_p = area of vent—inches².
B = air gap flux density—webers/meter².
C_{MS} = total mechanical compliance of the suspension—meters/Newton.
C_{MR} = total mechanical compliance of the enclosure volume—meters/Newton.
E_g = amplifier open circuit voltage—volts.
f, f_L, f_H, f_r = input frequency, lower critical frequency, upper critical frequency, enclosure resonant frequency—cps.
$F = BLE_g/(R_g+R_e)$ = driving force.
$g = \omega/\omega_S$ = forced frequency ratio.
$h = \omega_P/\omega_S$ = tuned frequency ratio.
l_d = duct length—inches.
L = voice coil conductor length—meters.
M_M = total speaker moving system mass—kilograms.
M_{MD} = diaphragm and voice coil mass—kilograms.
M_{MR} = total air load mass—kilograms.
M_P = air mass of vent—kilograms.
ω_L = lower critical frequency—radians/second.
ω_M = upper critical frequency—radians/second.
ω_0 = system resonant frequency—radians/second.
$\omega, \omega_S, \omega_P$ = input frequency, speaker resonant frequency, enclosure resonant frequency—radians/second.
P, P_S, P_P = complex rms sound pressure, complex rms sound pressure from speaker diaphragm, complex rms sound pressure from vent—microbar.
$Q_S = \omega_S M_M/R_M$ = speaker mesh Q.
$Q_P = \omega_P M_P/R_P$ = port mesh Q.
R_M = total mechanical resistance in speaker mesh—MKS mechanical ohms.
R_{MP} = total vent resistance—MKS mechanical ohms.
R_{MS} = mechanical resistance of suspension—MKS mechanical ohms.
R_P = total mechanical resistance in vent mesh—MKS mechanical ohms.
ρ = density of air—kilogram/meter³.
R_g = amplifier internal resistance—ohms.
r = average distance of observation point from diaphragm and/or port—meters.
R_e = voice coil resistance—ohms.
S_S = speaker suspension stiffness—Newtons/meter.
S_B = stiffness of enclosure volume—Newtons/meter.
t = vent wall thickness—inches.
U, U_S, U_P = complex rms volume velocity, complex rms volume velocity of speaker diaphragm, complex rms volume velocity of vent—meter³/second.
v_S = speaker diaphragm velocity—meters/second.
v_P = vent velocity—meters/second.
x, x_S, x_P = amplitude, diaphragm amplitude, vent amplitude—meters.
$x_{ST} = F/S_S$ = static deflection.
V_B = volume of enclosure—inches³.
V_D = volume of duct—inches³.

* Manuscript received by the PGA, January 13, 1959. This paper was presented at the Audio Eng. Soc. Convention, New York, N. Y., October 3, 1958, and is published through courtesy of the AES.
† Jensen Manufacturing Co., Chicago, Ill.

Introduction

The subject of enclosure design for direct-radiator loudspeakers has received much attention in recent years. There is no other subject so controversial, perhaps as a result of a general lack of understanding of the basic principles of operation of enclosures. The current trend is toward smaller enclosures, lower speaker resonances, and better performance claims. Trade journals tell of "all new enclosures," "revolutionary concepts," and "totally new principles of acoustics," when in reality there is a close identity with enclosure systems described long ago in well-known classics on acoustics. Actually there has been no basically new type of enclosure developed in this decade, although much worthwhile effort has been devoted to refinement and improvement of existing basic types.

The objectives of this paper are to select one of the currently popular enclosures, analyze its low-frequency behavior, discuss its limitations, and find a means of improving its performance.

The pressure-tight closed box using a high-compliance low-frequency speaker was selected for two reasons: 1) the design is based on sound engineering principles, and 2) the ever increasing popularity of this enclosure may alone have justified the expenditure of the time.

The Pressure-Tight Closed Box

In order to proceed intelligently, it is necessary to define certain design criteria for low-frequency reproduction. A slight problem arises here because all loudspeaker and enclosure designs must at the final stages of development be based on subjective judgments by people as to what constitutes "good quality." Because of this, it is not possible to be too specific in defining the design criteria. A reasonable amount of research into this problem disclosed that most listeners preferred a flat response to frequencies as low as about 40 cps. They also preferred speaker systems with low harmonic distortion and little or no transient distortion. Low-efficiency speaker systems were generally frowned upon because in most cases this meant forced obsolescence of existing amplifiers.

It will be shown later that it will be necessary to sacrifice over-all efficiency in order to extend the low-frequency response. It now remains to determine the maximum allowable efficiency loss. That a maximum limit on efficiency loss must be established becomes apparent when one considers three factors: 1) the amplifier economics, 2) the necessity for adequate reserve power to handle peaks without overload, and 3) the deterioration of amplifier characteristics with time. This last factor is tied in closely with the second.

Observations have shown that VU meter readings of one watt or so are about as high as reached in normal home listening. It should be possible, therefore, to justify a loss of 4 to 5 db of efficiency before noticing distortion on peaks from a good 10 to 12 watt amplifier. A loss of 10 db would be too excessive for a 10 to 12 watt amplifier as there would then be no reserve to handle peaks.

From the evidence just stated it is possible to set up the design criteria.

1) The response must be flat to as low a frequency as possible.
2) The total harmonic distortion must be as low as possible.
3) There must be no transient hangover.
4) The efficiency must be great enough to permit the use of a 10 to 12 watt amplifier.

The first design criterion, the extension of response to as low a frequency as possible, ordinarily would mean that the value of enclosure compliance would be made as large as possible so that the suspension compliance determined the resonant frequency. Because the small box precludes this approach, the speaker compliance is made as large as practical. The enclosure compliance then determines the resonant frequency. Although it is not possible to obtain a resonant frequency much lower than about 60 cps in this manner, the distortion characteristics are somewhat improved as a result of the good linearity of the enclosure compliance.

The resonant frequency can be lowered an additional 10 to 15 per cent by completely filling the box with loosely packed fiberglass, kapok, or cellufoam.[1] The resonant frequency decreases because the compressions become isothermal. This means that the velocity of sound decreases from about 344 to 291 m/sec. A 1 to 2 db loss in efficiency results, however, because the resistive component of box impedance is increased.

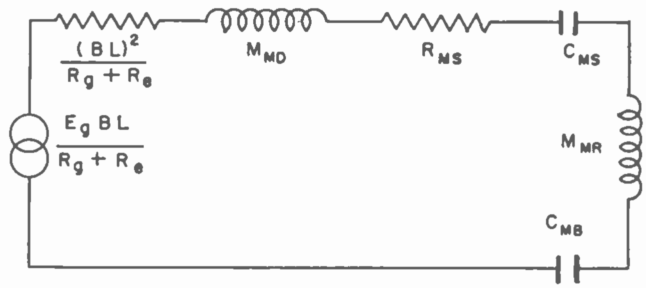


Fig. 1.

The mechanical equivalent circuit, Fig. 1, and equations of motion of the closed box-speaker system are well known. The steady-state solution yields the equation describing amplitude which when multiplied by the angular frequency and effective speaker area becomes the equation for volume velocity. Once volume velocity is known, it is a simple matter to obtain the sound pressure output.

The plot of amplitude vs frequency, Fig. 2, reveals that the amplitude increases many times in the region of resonance and below when Q becomes greater than 0.5. Suspension nonlinearities play an important part in determining the amount of total harmonic distortion at large amplitudes even though the box compliance supplies the major part of the restoring force. The result of a nonlinear suspension is the production of odd order harmonics with the third harmonic being predominant. Because the amplitude of a direct radiator speaker is inversely proportional to frequency squared below the region of ultimate radiation resistance, greatest distortion will occur at the low frequencies.[2]

Fig. 3 is a plot of the sound pressure response of a closed box-speaker system. Note that a Q of 0.5, corresponding to critical damping and best transient response, does not give the flattest output down to the lowest frequency possible. For flat response, Q should approximately equal unity. This conflicts with the first and third design criteria.

The only factor determining the transient response

[1] L. L. Beranek, "Acoustics," McGraw-Hill Book Co., Inc., New York, N. Y., 1st ed., p. 220; 1950.

[2] H. F. Olson, "Elements of Acoustical Engineering," D. Van Nostrand Co., Inc., New York, N. Y., 3rd ed., p. 186; 1958.

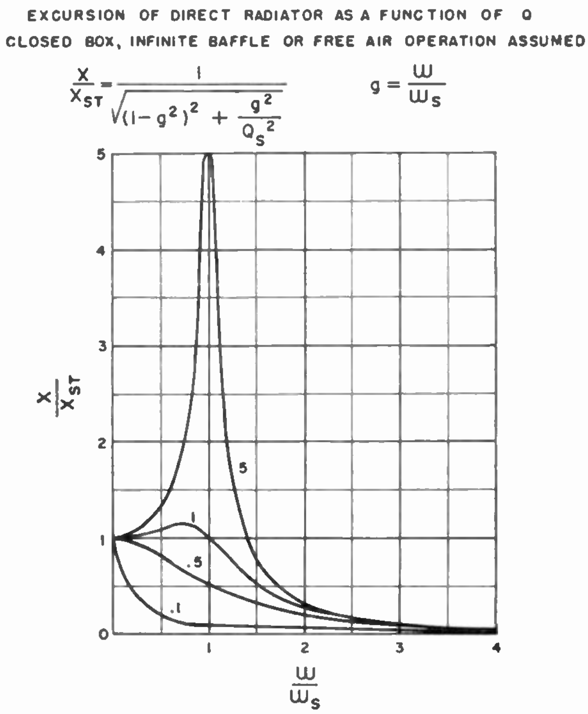


Fig. 2.

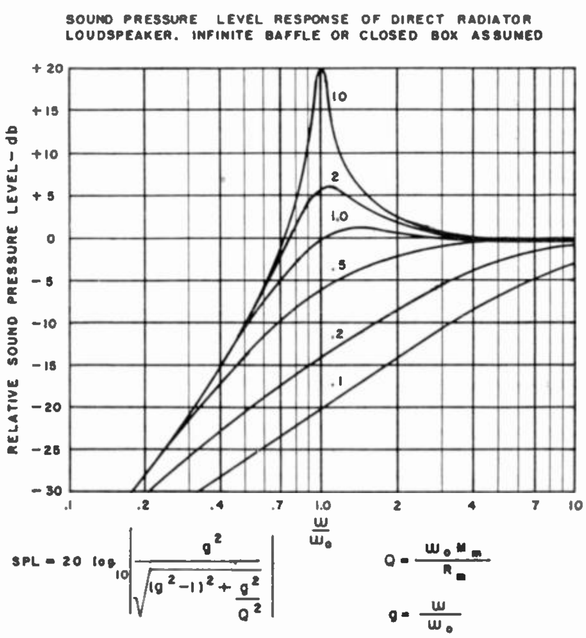


Fig. 3.

and low-frequency performance is the amount of damping. The damping can be changed by proper choice of amplifier damping factor or by adjustment of the flux density in the magnetic circuit. The popular belief that a large value of air stiffness in a small closed box increases speaker damping is erroneous. Damping is a function of resistances in the system.

The factor exerting the greatest influence on Q (damping) and low-frequency output is the product of magnetic field strength and voice coil conductor length commonly known as the BL product. Decreasing the BL product will actually increase efficiency in the region of resonance but an efficiency loss results at frequencies above resonance. It now becomes apparent that the pressure-tight closed box cannot fulfill adequately all four design criteria. It will be necessary to compromise transient response and over-all efficiency because increases in Q are usually achieved by an increase in moving system mass, a decrease in BL product or both.

The Vented Enclosure

A review of other known speaker enclosures suggests that the vented enclosure when used with a high-compliance speaker could produce a sound pressure response at least as good as that of the completely closed box and with certain advantages.

A search of the literature pertaining to vented enclosure design reveals that although equivalent circuits and equations for calculating Helmholtz resonance and location of the three critical frequencies were thoroughly developed, apparently no method has been published for calculating the response shape. Beranek's excellent chapter on enclosures[3] describes a method for obtaining the relative sound pressure level at the three critical frequencies. But the critical frequencies become widely separated when the box is small; and because the speaker system operates as a simple doublet at the lowest critical frequency, it is no longer possible to describe adequately the response shape because useful output occurs at only two of the three points, the middle and upper critical frequencies.

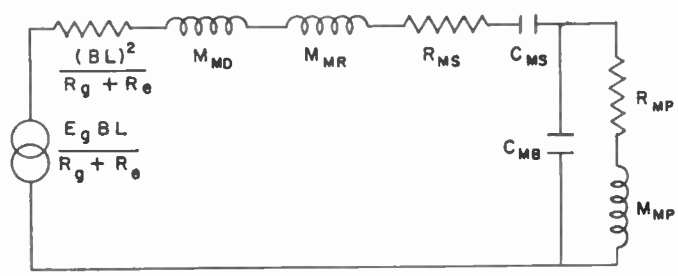


Fig. 4.

The mechanical equivalent circuit of the vented enclosure, Fig. 4, can be solved with the methods used for the closed box case. An exact solution becomes extremely complex because of the mutual coupling between the speaker and vent and the presence of resistance in both meshes. The vent resistance, however, can be ignored because the Q of the vent mesh is usually 20 to 40 times greater than the speaker mesh Q. The effects of mutual coupling can also be ignored since gains are insignificant when the two piston areas are very much different.[4]

[3] Beranek, *op. cit.*, pp. 241–258.

[4] S. J. Klapman, "Interaction impedance of a system of circular pistons," *J. Acoust. Soc. Amer.*, vol. 2, pp. 289–295; January, 1940.

The error resulting from these simplifications becomes significant only for large enclosures and then only in the region of speaker resonance when the vent Q is less than 10.

A vent in the closed box adds a second degree of freedom to the system causing a redistribution or repartition of the resonant frequency and damping of the speaker. The original speaker resonance is replaced by two new resonances, one near the closed box resonance and one substantially below the speaker resonance.

Damping at these two resonances is greater than for the closed box case.[5]

The vent acts as an acoustical mass which resonates with the compliance of the enclosure volume at a particular frequency. The equivalent circuit indicates that if the enclosure is tuned to the speaker resonance, the impedance becomes a maximum and is resistive at this frequency so that speaker amplitude should be greatly reduced. Maximizing the equation for sound pressure output with respect to enclosure resonance reveals that enclosure resonance must indeed be equal to speaker resonance for maximum over-all output to the lowest frequency possible. This condition of tuning is assumed for the remainder of this paper although it may be desirable to tune the enclosure to a higher frequency if increased output is desired in the region of the upper critical frequency. The sound pressure output of the system now becomes a function of two variables, the speaker Q and the ratio of enclosure stiffness to speaker suspension stiffness. This ratio can be thought of as a coefficient of coupling between the speaker and port meshes.

Low-frequency output is increased and the low-frequency cutoff is lowered when the stiffness ratio is decreased. Although output varies directly with speaker Q as in the closed box case, it will be shown that a Q less than 0.5 will give a flat response whereas the closed box requires a Q of approximately unity.

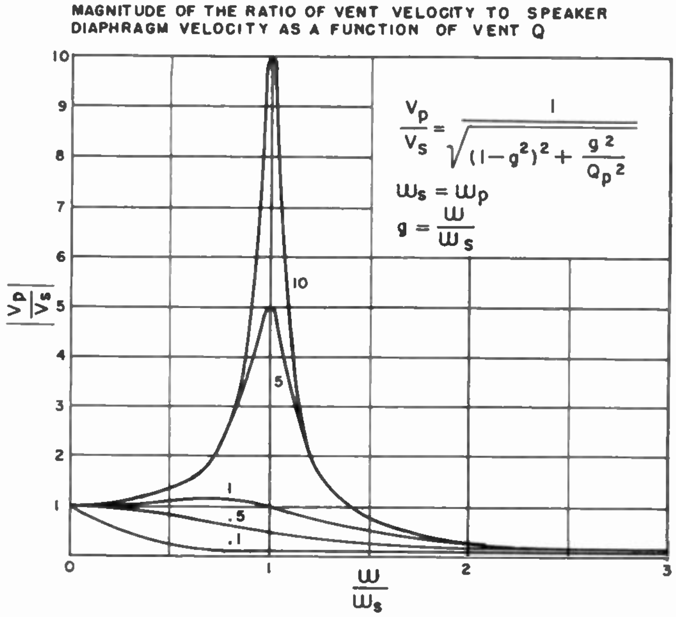


Fig. 5.

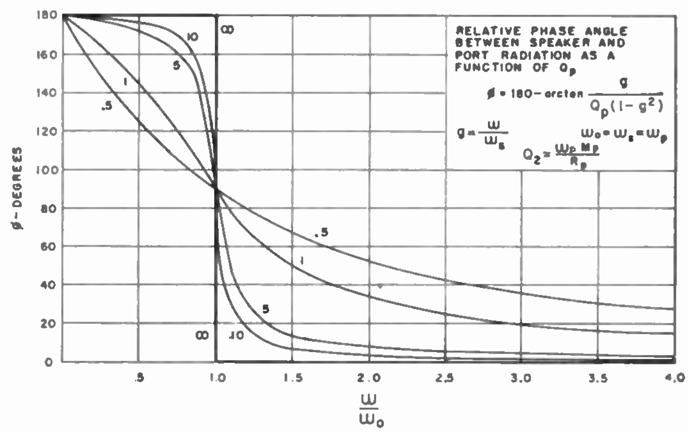


Fig. 6.

Fig. 5, a plot of the ratio of vent velocity to speaker diaphragm velocity, shows that for normal vent Q's (greater than 10) the acoustic power radiated from the vent predominates over that radiated from the speaker diaphragm for about one-half of an octave above and below speaker resonance. It is observed, however, that total radiation decreases rapidly below speaker resonance. This occurs because at frequencies below speaker resonance the volume velocity in the vent becomes out of phase with the speaker diaphragm volume velocity. Fig. 6 is a plot of the relative phase angle between vent and speaker diaphragm volume velocity. Although vent and diaphragm radiation are in quadrature at speaker resonance for all values of vent Q, the transition to out-of-phase or in-phase operation is very rapid when the Q is greater than 5.

The gradual phase shift occurring when the vent Q is less that 0.5 may lead to the erroneous conclusion that low values of vent Q can lower the low-frequency cutoff. An extension of low-frequency cutoff cannot occur because vent power output diminishes rapidly as Q becomes equal to or less than unity (see Fig. 5). The performance then approaches the closed box performance. This is predicted by the equations of motion which reduce to those of a closed box when either the vent Q or area are allowed to approach zero.

The diaphragm amplitude becomes a minimum at speaker resonance as opposed to the maximum which occurs in the closed box. Fig. 7 shows the amplitude variations of a speaker diaphragm in a vented enclosure for a stiffness ratio of 7 as a function of speaker Q. A

[5] J. B. Crandall, "Theory of Vibrating Systems and Sound," D. Van Nostrand Co., Inc., New York, N. Y., 2nd ed., pp. 62–63; 1927.

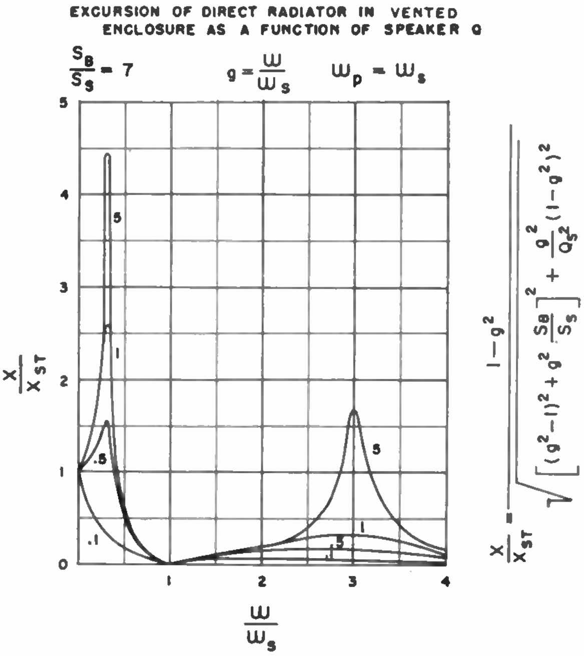


Fig. 7.

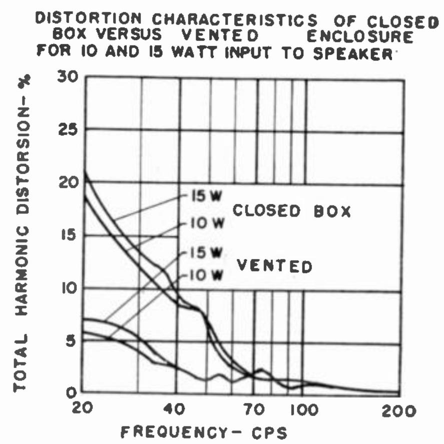


Fig. 8.

stiffness ratio of 7 corresponds to a 12-inch high-compliance speaker in an enclosure volume of about $2\frac{1}{4}$ cubic feet. The diaphragm amplitude remains very uniform down to speaker resonance for critically or overdamped speaker operation. Because the vent mesh consists of linear elements, the harmonic and intermodulation distortion are greatly reduced.[6]

Fig. 8 shows the distortion characteristics of a $2\frac{1}{4}$-cubic-foot closed vs vented box-speaker system. The speaker used in these measurements was a Jensen P12-NF high-compliance woofer operating with a Q of 0.5.

Fig. 9 illustrates the dependence of flat response on proper enclosure tuning. Fig. 10 is the experimental data verifying the theoretical data of Fig. 9. Figs. 11–13 show the theoretical sound pressure response of the vented enclosure with stiffness ratios of 1, 3, and 7, respectively, as a function of speaker Q. The experimental

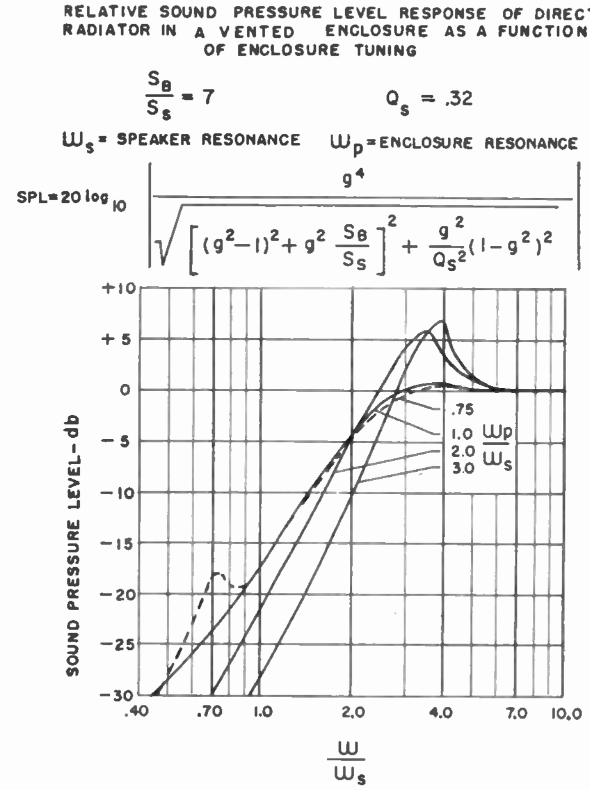


Fig. 9.

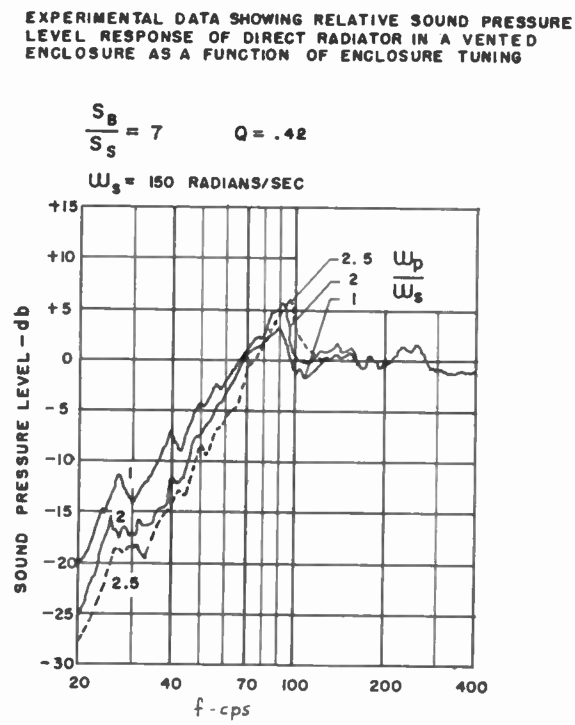


Fig. 10.

[6] H. S. Knowles, "Loudspeakers and room acoustics," in "Henry Radio Engineers Handbook," McGraw-Hill Book Co., Inc., New York, N. Y., pp. 760–761; 1950.

RELATIVE SOUND PRESSURE LEVEL RESPONSE OF DIRECT RADIATOR IN VENTED ENCLOSURE AS A FUNCTION OF SPEAKER Q

$$SPL = 20 \log_{10} \left| \frac{g^4}{\sqrt{\left[(g^2-1)^2 + g^2 \frac{S_B}{S_S}\right]^2 + \frac{g^2}{Q_S^2}(1-g^2)^2}} \right|$$

$\frac{S_B}{S_S} = 1$

$\frac{\omega_L}{\omega_s} = .62$

$\frac{\omega_H}{\omega_s} = 1.62$

$\omega_s = \omega_p$

$Q_s < Q_p$

$g = \frac{\omega}{\omega_s}$

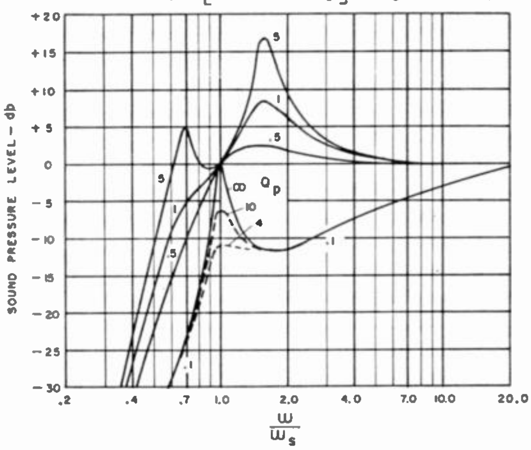


Fig. 11.

RELATIVE SOUND PRESSURE LEVEL RESPONSE OF DIRECT RADIATOR IN A VENTED ENCLOSURE AS A FUNCTION OF SPEAKER Q

$\frac{S_B}{S_S} = 7$ $\omega_s = \omega_p$ $Q_s < Q_p$ $\frac{\omega_L}{\omega_s} = .34$ $\frac{\omega_H}{\omega_s} = 3$

$$SPL = 20 \log_{10} \left| \frac{g^4}{\sqrt{\left[(g^2-1)^2 + g^2 \frac{S_B}{S_S}\right]^2 + \frac{g^2}{Q_S^2}(1-g^2)^2}} \right|$$

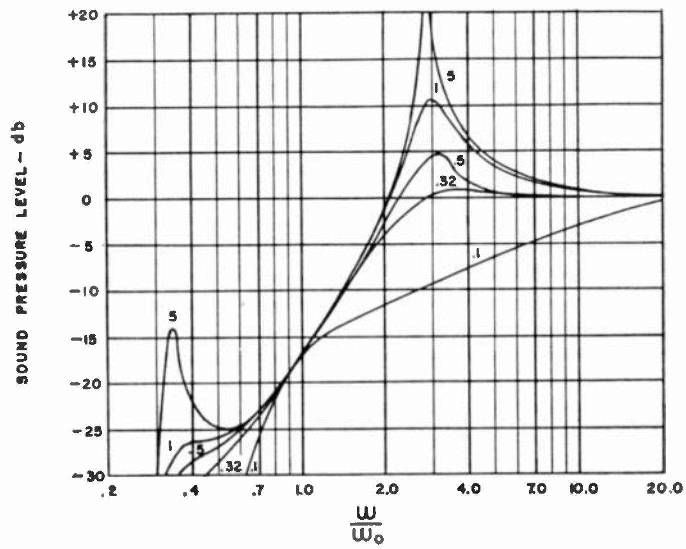


Fig. 13.

RELATIVE SOUND PRESSURE LEVEL RESPONSE OF DIRECT RADIATOR IN VENTED ENCLOSURE AS A FUNCTION OF SPEAKER Q

$\frac{S_B}{S_S} = 3$ $\frac{\omega_L}{\omega_s} = .458$ $\frac{\omega_H}{\omega_s} = 2.19$ $\omega_s = \omega_p$ $Q_s < Q_p$ $g = \frac{\omega}{\omega_s}$

$$SPL = 20 \log_{10} \left| \frac{g^4}{\sqrt{\left[(g^2-1)^2 + g^2 \frac{S_B}{S_S}\right]^2 + \frac{g^2}{Q_S^2}(1-g^2)^2}} \right|$$

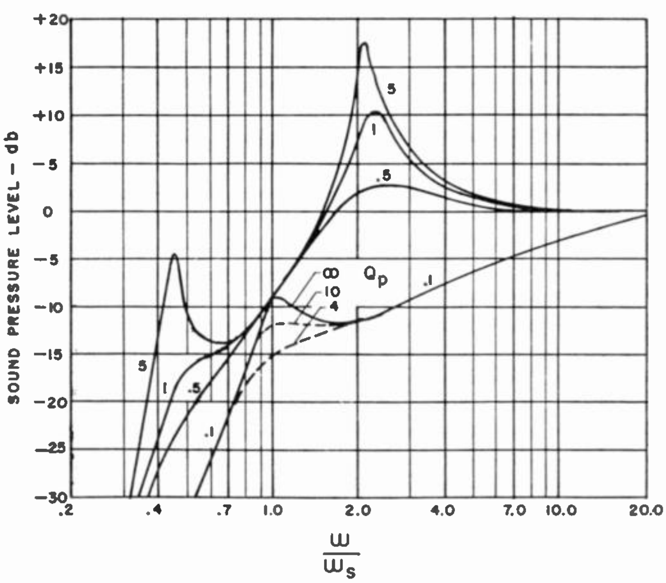


Fig. 12.

EXPERIMENTAL DATA SHOWING RELATIVE SOUND PRESSURE LEVEL RESPONSE OF DIRECT RADIATOR IN A VENTED ENCLOSURE AS A FUNCTION OF SPEAKER Q

$\frac{S_B}{S_S} = 7$ $\omega_s = \omega_p = 151$ RAD./SEC.

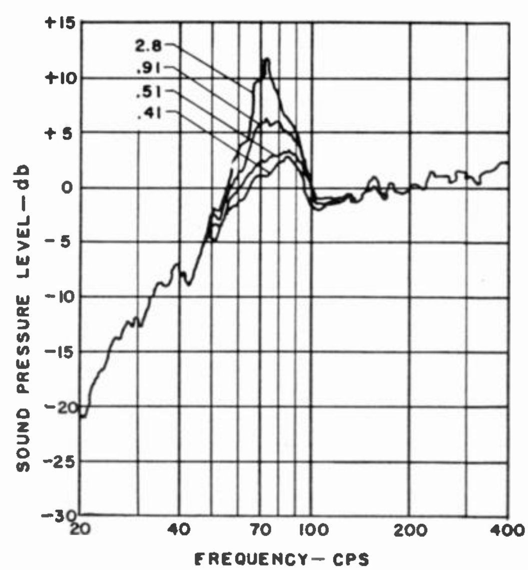


Fig. 14.

data in Fig. 14 verify the theoretical data of Fig. 13 and substantiate the validity of the theoretical equations. These response curves certainly demonstrate that a large box still outperforms a small box. An important result of this data is that flat response is in all cases obtained with a speaker Q less than 0.5. The data also show that the acoustic output from small enclosures is only slightly improved by venting. Consider the case of a stiffness ratio of 7 and assume a speaker resonant frequency of 20 cps.

Fig. 13 shows that flat response is maintained to 54 cps for a speaker Q of 0.32 with the 10-db down point occurring at 28 cps. The same stiffness ratio applied to a closed box would increase speaker resonance to 60 cps. Fig. 3 shows that a speaker Q of 1 will give flat response to 60 cps with the 10-db down point occurring at 32 cps. The increase in output from 28 to 60 cps in the vented enclosure is barely 1 db. The transient response, harmonic and intermodulation distortion are, however, greatly improved. In cases where the enclosure volume can be more generous, venting offers considerable gains in output.

Conclusions

Many considerations lead to the conclusion that a vented enclosure can do everything the closed box can do, and with certain advantages. But honest comparison of the two must include all factors, not just those favoring one system.

Important points against venting which could result in poor performance if proper consideration is not given them are:

1) Drop in output below the resonant frequency is 18 db per octave compared to 12 db per octave for the closed box. It is commonly thought that the transient effects resulting from a cutoff as sharp as 18 db per octave can be bothersome.

2) Very small enclosure volumes coupled with a very low speaker resonant frequency may require a vent so small as to be ineffective because of excessive viscous losses in the vent. The operation is then about the same as that of the closed box. In extreme cases of this sort, rectification has been observed in the air flow through the vent. This caused a displacement in the dynamic center of diaphragm amplitude. The resulting distortion was greater than for the closed box.

3) Mistuning or incorrect speaker Q can exaggerate the output in the vicinity of the upper critical frequency. This can cause an aggravated booming sound generally too high in frequency to give even a good impression of false bass.

The optimum system goal as defined by the four design criteria must overcome these limitations. This can be accomplished in the following manner:

The speaker resonant frequency must be low enough that the 18 db per octave slope occurs below 20 cps. Subjective listening tests did not indicate the presence of any troublesome transients. An acceptable resonant frequency would be between 20 and 30 cps.

The vent must have small enough resistance so that the volume velocity is not impeded seriously. Volume velocity will be independent of vent area so long as viscous losses are minimized. If the vent area becomes too small, a larger area with a duct must be used.

The amplifier must have a damping factor sufficient to maintain at least critical damping in the speaker mesh. Speaker Q's ranging from 0.3 to 0.4 give best results.

Having disposed of the limitations by proper design, the vented enclosure has definite and important advantages over the closed box. All exist in the octave or octave and one half in the region of speaker resonance. They stem from the one characteristic differentiating the vented enclosure from the closed box—better diaphragm loading in the region where the box is an active element.

The vent relieves the diaphragm of much of the necessity to move. The reduced diaphragm amplitude is not at the expense of sound output because the vent also assumes the task of radiation of sound energy. The advantages of the vented enclosure become: 1) lower harmonic distortion because the linear vent operation and reduced diaphragm amplitude minimize the effects of speaker nonlinearities, 2) reduced intermodulation distortion because of reduced diaphragm amplitude, 3) improved transient response because the speaker is at least critically damped, 4) greater acoustic output for a given amount of distortion, and 5) less of the deliberate efficiency loss for purposes of response leveling is required because the speaker must be operated at a lower Q.

The Q can be decreased by increasing the speaker efficiency. It has been found that the efficiency averages at least 3 db better, for equivalent response trend, than would have been allowable with a closed box system.

The author believes that in spite of its simple appearance the design of a vented enclosure is sufficiently difficult that it should not be attempted by the layman unless he is exceptionally well informed and has adequate test facilities.

Appendix

A piston whose diameter is less than one third wavelength is essentially nondirectional at low frequencies. It can, therefore, be approximated by a hemisphere whose rms volume velocity is equal to the product of voice coil velocity and effective speaker cone area,

$$U_c = v_c A_d.$$

The magnitude of rms sound pressure at a distance r (in the far field) from the speaker is[7]

[7] Beranek, *op. cit.*, p. 188.

$$|P| = \frac{|U_c| f\rho}{r}. \tag{1}$$

The equations describing sound pressure response of the closed and vented enclosure can be obtained from the equations of motion for the two systems by substituting for volume velocity in (1).

Closed Box

The equation of motion obtained from Fig. 1 is

$$(-M_M\omega^2 + S_M + jR_M\omega)x = F. \tag{2}$$

Eq. (2) is solved for x. The equation describing amplitude becomes

$$x = \frac{x_{ST}}{\sqrt{(1-g^2)^2 + \frac{g^2}{Q_S^2}}}. \tag{3}$$

The expression for voice coil velocity is obtained by multiplying (3) by the angular frequency.

$$v_S = \omega x. \tag{4}$$

Volume velocity is obtained by multiplying (4) by the effective speaker cone area.

$$U_S = A_d\omega x. \tag{5}$$

The sound pressure level is obtained by substituting (5) into (1).

$$P = \frac{A_d\omega x_{ST} f\rho}{r\sqrt{(1-g^2)^2 + \frac{g^2}{Q_S^2}}}. \tag{6}$$

A reference volume velocity is defined by

$$U_{\text{ref}} = \frac{x_{ST}\omega A_d}{g^2}. \tag{7}$$

This is the actual volume velocity above resonance under the special condition that the expression under the radical in (3) is proportional to g^4; *i.e.*, the diaphragm is completely mass-controlled,

$$\left((g^2-1)^2 \gg \frac{g^2}{Q_S^2}\right).$$

A reference sound pressure is defined for low frequencies by substituting (7) into (1).

$$|P_{\text{ref}}| = \frac{x_{ST}\omega A_d f\rho}{g^2 r}. \tag{8}$$

An expression for relative sound pressure level (*SPL*) in decibels is obtained by taking the ratio of (6) and (8).

$$\text{SPL} = 20 \log \left| \frac{g^2}{\sqrt{(g^2-1)^2 + \frac{g^2}{Q_S^2}}} \right|. \tag{9}$$

Vented Enclosure

The equations of motion obtained from Fig. 4 are

$$\left.\begin{aligned} (-M_S\omega^2 + S_S + S_B + jR_S\omega)x_S - S_Bx_P &= F \\ -S_Bx_S + (-M_P\omega^2 + S_B + jR_P\omega)x_P &= 0 \end{aligned}\right\}. \tag{10}$$

Eq. (10) is solved for x_S and x_P.

Amplitude

In order to simplify the expressions, Q_P is assumed to equal infinity. The resulting error is very small because the typical vent mesh Q is 20 to 40 times greater than the speaker mesh Q. The speaker and vent amplitudes become

$$x_S = \frac{x_{ST}(h^2-g^2)}{\sqrt{\left[(g^2-1)(g^2-h^2) - g^2\frac{S_B}{S_S}\right]^2 + g^2\left[\frac{h^2-g^2}{Q_S}\right]^2}} \tag{11}$$

$$x_P = \frac{x_{ST}h^2}{\sqrt{\left[(g^2-1)(g^2-h^2) - g^2\frac{S_B}{S_S}\right]^2 + g^2\left[\frac{h^2-g^2}{Q_S}\right]^2}}. \tag{12}$$

The volume velocities are obtained exactly as in the closed box case:

$$U_S = A_d\omega x_S \tag{13}$$

$$-U_P = -A_d\omega x_P. \tag{14}$$

The total sound pressure is obtained by substituting (13) and (14) into (1) and adding the two pressures. A negative sign is used for U_P because, except for the phase shift introduced in the vent mesh, the radiation from the back of the speaker cone is 180° out of phase with the front radiation. Using the concept of reference sound pressure, the relative sound pressure level in decibels is

$$\text{SPL} = \left|\frac{P_S - P_P}{P_{\text{ref}}}\right| = 20 \log \left| \frac{g^4}{\sqrt{\left[(g^2-1)(g^2-h^2) - g^2\frac{S_B}{S_S}\right]^2 + g^2\left[\frac{h^2-g^2}{Q_S}\right]^2}} \right|. \tag{15}$$

This allows the calculation of response for any condition of tuning.

Maximizing (15) with respect to enclosure tuning shows that maximum output at speaker resonance is obtained when the enclosure is tuned to speaker resonance, *i.e.*, $h=1$. The expression for relative sound pressure level becomes

$$\text{SPL} = 20 \log \left| \frac{g^4}{\sqrt{\left[(g^2-1)^2 - g^2 \dfrac{S_B}{S_S}\right]^2 + \dfrac{g^2}{Q_S{}^2}(1-g^2)^2}} \right|. \quad (16)$$

Designing the Enclosure

The enclosure volume should be based on the maximum amount of space available. The object is to get the lowest possible value of S_B/S_S. It is necessary to know only two factors, S_B/S_S and Q_S, in order to calculate the response.

Determination of S_B/S_S

First measure free air speaker resonance, denoting this by f_1. Then install the speaker in the unvented enclosure and measure resonance again. Denote this by f_2. The stiffness ratio is then given by

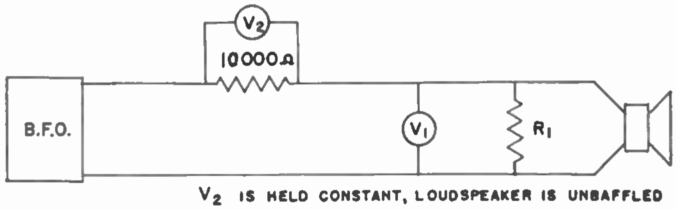
$$\frac{S_B}{S_S} \doteq \left[\frac{f_2}{f_1}\right]^2 - 1.$$

Determination of Speaker Q

Q_S is determined from the velocity curve as described in Fig. 15. The width, Δf cps, is measured between the points of the curve on either side of the resonance peak where the voltage is 3 db down (0.707) from the maximum voltage.

The area of the vent is obtained from

$$f_r = 2155 \sqrt{\frac{A_p}{V_B(t + 0.96\sqrt{A_p})}}. \quad (17)$$



(1) RESONANT FREQUENCY

VARY OSCILLATOR FREQUENCY UNTIL V_1 IS MAXIMUM.

(2) LOADED Q

(A) OPEN CIRCUIT R_1. OBTAIN SHAPE OF VELOCITY CURVE BY PLOTTING AGAINST FREQUENCY THE QUANTITY

$$e = V_1 - \frac{R_{VC}V_2}{10000} \qquad Q_1 = \frac{f}{\Delta f}$$

(B) INSERT R_1 (ABOUT 80 OHMS). PLOT NEW VELOCITY CURVE

$$Q_2 = \frac{f}{\Delta f}$$

(C) DETERMINE MECHANICAL RESISTANCE, R_M, FROM

$$R_M = (R_1 + R_{VC})\left(\frac{Q_1}{Q_2} - 1\right)$$

(D) LOADED Q IS GIVEN BY

$$Q_L = \frac{R_g + R_{VC}}{R_g + R_{VC} + R_M} Q_1$$

$$Q_L = \frac{R_{VC}}{R_{VC} + R_M} Q_1 \quad \text{IF} \quad R_g \ll R_{VC}$$

R_{VC} = VOICE COIL RESISTANCE – OHMS

R_g = GENERATOR INTERNAL RESISTANCE – OHMS

GENERATOR INTERNAL RESISTANCE CAN BE OBTAINED FROM

$$\text{DB REGULATION} = 20 \log_{10} \frac{R_g + R_L}{R_L}$$

R_L = LOAD RESISTANCE – OHMS

Fig. 15.

The enclosure should be tuned to the resonant frequency of the speaker. It is permissible to tune to a higher frequency if increased output is desired in the region of the upper critical frequency. In some cases, the output in the region of the upper critical frequency may tend to peak up. Peaking can be minimized by increasing the damping on the speaker by increasing amplifier damping factor. A negative damping factor may have to be used in some instances. If the damping factor cannot be changed, the enclosure should be tuned lower than speaker resonance. While this will reduce the peak, it will also result in some losses in output lower in frequency.

Since the vent behaves as a simple source at low frequencies, the amount of power radiated is independent of the vent area for any given volume velocity. It is permissible, therefore, to use any value of vent area so long as the desired enclosure resonance is obtained.

The use of resistive loading over the vent or the use of a series of small holes distributed over a large area (another way of adding resistive loading) is generally not recommended. An important reason for using a vented enclosure is that the loudspeaker produces far less distortion in the octave above speaker resonance than would be the case if the box were closed. Adding resistance to the vent will reduce the power radiated and will increase the speaker diaphragm amplitude and distortion. The vent area should not be allowed to be less than about 4 square inches. If (17) indicates an area less than this value, the area should be increased arbitrarily and a duct (installed behind the vent) is used to tune the enclosure properly. The expression for resonance now becomes

$$f_r = 2155 \sqrt{\frac{A_p}{(V_B - V_D)(l_d + 0.96\sqrt{A_p})}}. \quad (18)$$

The enclosure should be lined with a 2-inch thickness of fiberglass on at least three sides to eliminate any normal modes. The approximate location of the upper and lower critical frequencies is obtained from

$$f_{L,H} = (0.707) f_s \sqrt{2 + \frac{S_B}{S_S} \pm \sqrt{4\frac{S_B}{S_S} + \left[\frac{S_B}{S_S}\right]^2}}. \quad (19)$$

The response shape can be calculated from either (15) or (16), depending on whether or not the enclosure is tuned to speaker resonance. The calculations can become rather tedious but unfortunately there is no short cut.

A Sliding Class-A Audio Output System*

JOSEPH A. WORCESTER†

Summary—**A single-ended transistor output stage is described in which the low battery drain and high-power output characteristics of push-pull Class B are achieved by a sliding bias produced by the signal.**

IT has been stated that the basic activity of an engineer is to try to do something for a dollar that any damn fool can do for two dollars. It is somewhat in this vein of homely endeavor that the present paper is directed, namely, to approach the operational advantages of push-pull Class B operation with a single output transistor. While this project has not resulted in full equivalence to push-pull Class B, it has displayed sufficient superiority over straight Class A and approximated push-pull Class B from the dual standpoints of battery drain and power output to merit inclusion in at least one production transistor receiver of the pocket variety, the General Electric P745.

The reference to Class B has been rather loosely applied since most so-called transistor Class-B output systems are in reality Class AB. In other words, to avoid the possibility of distortion at low output levels, a small amount of quiescent current is allowed to flow, which means that for the first couple of milliwatts of output power, the operation is Class A and changes to Class B for larger output levels.

This circuit is too familiar to require further elaboration, but it might be desirable to point out in Fig. 1 the basic components required. It will be noted that these include: 1) an input transformer, 2) an output transformer, 3) two transistors, and 4) quiescent biasing resistors. To this fundamental complement, stabilization and decoupling networks may be added as required.

The basic concept of the circuit to be described is one of economy in that, if the type of operation of the circuit just shown can be obtained with a single transistor, not only will the cost of one transistor be saved, but also the rather expensive input transformer will no longer be essential as a coupling device.

Therefore, if we start with a conventional Class-A stage and reduce the fixed bias until the collector current drain is reduced to quiescent proportions and then add a variable opening bias derived from the output audio signal, Class-B type of operation has been essentially achieved.

Fig. 2 shows the schematic diagram of this circuit. Quiescent current conditions are established by the resistors R_1 and R_2. It will be noted that the primary and secondary of the output transformer are stacked to provide maximum ac voltage for bias production. This ac voltage is applied through capacitor C_1 to a germanium diode. The rectified ac is filtered by capacitor C_2 and the resulting negative dc is used to open up the transistor in accordance with the syllabic content of the output signal.

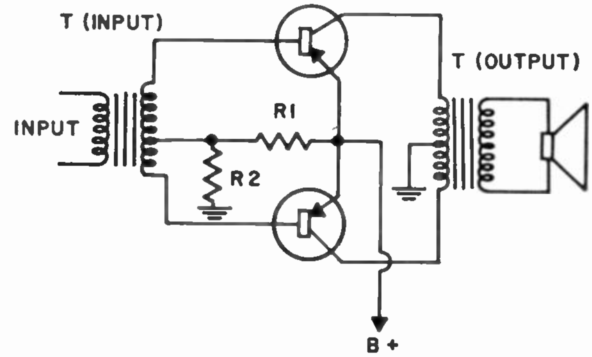


Fig. 1.

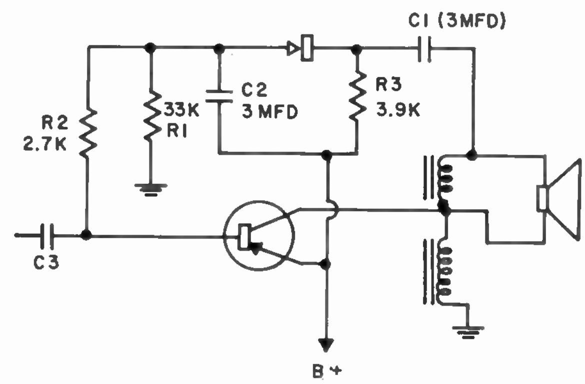


Fig. 2.

The proper operation of this device requires some attention to design details. The possible failing that occurs to nearly everybody at first blush is the probability of observable distortion caused by the first few cycles of a syllable which produce the opening bias and which, of course, would be clipped to some extent by the transistor still operating under quiescent conditions. In practice, however, distortion from this cause has not been troublesome. About the only precaution to observe is to make the time constant of the diode output circuit (R_2C_2) just large enough to prevent feedback of low-frequency audio signals, but not greatly larger than this, so that the bias circuit operates without observable delay from a syllabic standpoint.

It is important that the diode be biased so that it starts to conduct just before the maximum undistorted output permitted by quiescent current alone is ap-

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proached. If the diode operates later than this, distortion, of course, will result and, if it opens substantially sooner, the transistor will be opened more than is required to handle the audio signal and excessive battery drain will be experienced.

The proper bias on the diode can be obtained by juggling the total resistance used to provide quiescent current for the transistor between resistors R_1 and R_2. For instance, if R_1 is made relatively smaller and R_2 increased, the anode of the diode will be moved closer to ground potential and the cutoff bias will be increased since the diode cathode returns to $B+$ through R_3.

The proper distribution of the quiescent biasing resistance between R_1 and R_2 is an initial circuit design consideration and, once determined, further factory adjustment or selection will not be found necessary to account for variations in transistors and diodes. Such is not the case, however, for resistor R_3. By varying R_3, it is possible to regulate the percentage of the total control voltage developed by the diode that is utilized to open the transistor. The larger this resistor is made, the smaller the potential applied to the transistor. This adjustment is important since the gain of production transistors varies over such wide limits. For a low-limit transistor the ac voltage in the collector circuit will be low and nearly all the developed control potential may be required to provide sufficient opening bias. This would necessitate a low value of R_3. On the other hand, very high gain units will produce control bias greatly in excess of requirements and R_3 will have to be increased substantially.

In practice, a selection from several values is made in the factory to provide the proper battery drain at maximum 10 per cent distortion output.

In conclusion, the writer would like to acknowledge the importance of the contributions made toward the practical utilization of this circuit by R. L. Miller of the transistor radio product engineering operation.

The Delta Sound System for Television Receivers*

ROBERT B. DOME†

Summary—**The delta sound system for television receivers incorporates the three features of AM compression, high-level audio output FM detection, and fundamental AM cancellation in a circuit having cost advantages.**

Design information is presented for the system to enable the circuit designer to specify component parts and to predict the performance. AM compression in the range of 12 to 24 db and a peak-to-peak audio output voltage of about 60 volts with cancellation of the fundamental component of the undesired AM are obtainable with readily available low-cost tubes and circuit components.

Introduction

THE development of the delta system was undertaken in an effort to make available a sound system that would be less expensive than the ratio detector system employed in television receivers. The name "delta" has been given to the system for the sake of having a single word to identify the system. The system itself consists of four principal parts:

1) A 4.5-mc pentode amplifier fed from a take-off circuit connected or coupled to the plate circuit of the receiver video frequency amplifier,
2) A diode AM compressor connected to or coupled to the plate circuit of the 4.5-mc pentode amplifier,
3) An FM detector consisting of a discriminator circuit and a triode operating as a power detector (sometimes called a bias detector or a plate-bend detector), with provision for cancelling out residual fundamental frequency AM,
4) A volume control, audio power amplifier, and loudspeaker.

The three features that distinguish this system are therefore:

1) AM compression,
2) High-level audio output FM detection,
3) Fundamental frequency AM cancellation.

These three points may be associated with the three sides of Δ, hence the selection of the name "delta" for short.

An analysis of the ratio detector system showed that expensive items were the 6T8 triple diode high mu triode tube, the elaborate triple winding discriminator transformer, and the electrolytic condenser required in shunt with the detector self-bias resistor. The ratio detector has good AM rejection characteristics, and the output is balanced for AM. Its output is rather low, however, because of the relatively low impedance of the detector system. On account of this it is necessary to add a stage of audio frequency amplification between the detector and the audio power amplifier.

Development

It was evident that there was a possibility of obtaining some savings if the detection and amplification could be accomplished in one electron stream. A single and relatively inexpensive triode, operated as a power detector would provide high audio output levels for relatively low RF input levels, and so it was decided to employ a triode for this purpose. But a single triode of itself can provide single-ended output only, and so AM present in the FM wave would also be detected. Therefore means of balancing out the AM audio had to be provided. Even though balancing could be obtained by some means, the AM rejection feature of the ratio detector would not be achieved unless something else was done. It was therefore decided to employ a diode as an AM compressor to obtain AM rejection. When this was done, it was found that as a by-product of the diode compressor, some AM audio of suitable phase and amplitude which could be introduced into the triode detector system to obtain cancellation of the fundamental frequency component of the audio frequency of the AM became available.

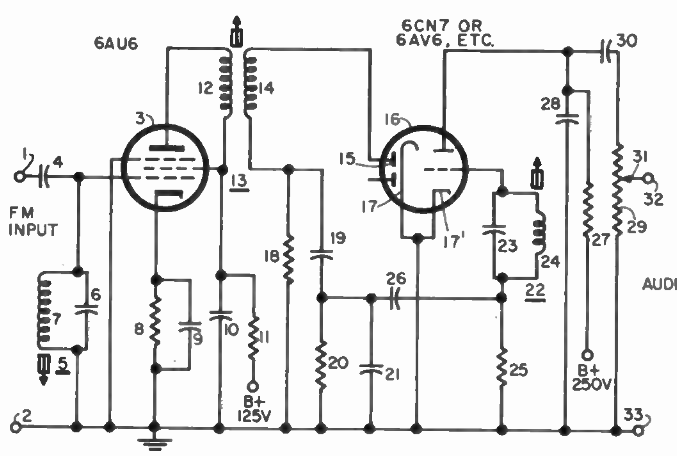

Fig. 1—Schematic diagram of delta sound system.

When these functions are combined, a single circuit such as shown in Fig. 1, results. The 4.5-mc center carrier frequency FM sound signal is applied at the input terminals 1, 2, the latter being grounded. This signal is applied to the control grid of a pentode amplifier tube 3 by a connection from terminal 1 through a coupling condenser 4. The grid-to-ground shunt tuned circuit 5 consists of a condenser 6 in shunt with tunable inductor

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7 which is adjusted so as to maximize the 4.5-mc voltage applied to the tube. The cathode of the pentode is connected to ground through a bias resistor 8, and this resistor is by-passed for RF by a condenser 9. The suppressor grid of the pentode is connected either to ground or to the cathode and is shown here to be connected to ground. The screen grid is by-passed to ground by a condenser 10, and the screen is connected to $B+$ through an isolation filter resistor 11. The anode of the pentode is connected to the screen grid through primary winding 12 of a transformer 13. The secondary winding 14 of transformer 13 is closely coupled to the primary. This may be achieved by making use of bifilar construction, for example. A single magnetic core tunes this transformer and the associated tube and circuit capacitance to 4.5 mc. The upper or hot terminal of secondary 14 is connected to one of the two diode anodes 15 in vacuum tube 16. Tube 16 consists of two diodes and a high-mu triode all within one envelope. The cathodes for the three electron streams may be a common cathode or the cathodes may be physically separated as shown by 17 and 17′. The lower terminal of secondary 14 is connected to ground through a resistor 18. A by-pass circuit around resistor 18 consists of an audio frequency by-pass condenser 19 in series with a relatively low resistance in the form of a resistor 20, the lower end of the latter being grounded. An RF condenser 21 is shunted around resistor 20. The cathodes 17 and 17′ are connected together and are grounded. The grid of the triode is connected to the upper end of a tank circuit 22 consisting of a capacitor 23 in shunt with an adjustable inductor 24. This circuit is tuned to a frequency slightly higher than 4.5 mc as will be explained later. The low end of tank 22 is connected to ground by a grid leak resistor 25 and is also connected to the junction between capacitors 19 and 21 by an audio frequency coupling condenser 26. The triode anode is connected to $B+$ through a resistor 27. The triode anode is by-passed to ground for RF by condenser 28. The capacitance of condenser 28 is chosen so as to provide an appropriate de-emphasis characteristic for the audio. The load circuit for the triode consists of a volume control potentiometer 29. One terminal of potentiometer 29 is grounded. The other terminal is connected to the triode anode by a coupling and blocking condenser 30. The arm 31 of potentiometer 29 is connected to an output terminal 32. Audio output is taken between terminals 32 and 33, the latter being grounded. In a television receiver terminal 32 may be the control grid of the audio power amplifier output tube.

Briefly the circuit operates as follows: the pentode amplifies the incoming 4.5-mc signal and when the RF voltage between grid and ground is of sufficient magnitude the tube will act as a limiter of AM. The output of the pentode is acted upon by the diode circuit so as to reduce the percentage modulation of any AM present. The amount of compression depends on several factors, discussed later, but values of compression in the range from 12 db to over 24 db have been measured. The resultant RF wave, now only lightly modulated in amplitude, excites the grid-cathode elements of the triode. This excitation is received as follows: condenser 21 is purposely chosen so that some RF exists across it as part of the tuned circuit current in circuit 13. The voltage across condenser 21 is transferred to the triode grid through condenser 26 and tank circuit 22 which, in conjunction with the capacitance between the triode grid and ground, forms a frequency selective or discriminator circuit whereby the amplitude of the RF signal at the grid is a function of RF. The grid-cathode region of the tube rectifies this RF voltage causing a bias to develop as the result of grid current flow through grid leak 25. This voltage is stabilized by the relatively large condenser 26 so that the triode can act as a power detector for RF envelope changes at audio frequency rates. Now any residual AM, not removed by the diode compressor, will also be detected by the triode. For example, during a positive going AM cycle, the plate current of the triode will increase, causing a *negative-going* audio pulse to appear at output terminal 32. Cancellation of this effect is achieved as follows: the same positive-going AM cycle at the diode rectifier will cause a negative-going audio pulse to appear across resistor 20 in series with condenser 19. Since the triode grid is coupled to this point by condenser 26, the grid receives a negative-going pulse. The triode amplifies this in the normal manner of an amplifier causing the plate current to decrease or causing a *positive-going* audio pulse to appear at output terminal 32. Thus by choosing the value of resistor 20 suitable to provide just the right amount of cancelling voltage, the fundamental frequency component of the disturbing AM may be removed or reduced to a low level.

Diode Compressor

The function of the diode is to compress the undesired AM that may be present on the incoming FM signal. The principle upon which compression depends is that of a dynamic change of the RF load line in the pentode plate circuit. Fig. 2 shows a schematic diagram of the elements making up the diode compressor together with the operational characteristics. Curve 1 represents the peak RF output voltage e obtainable as a function of pentode RF grid voltage, e_g, when the diode is made inoperative by turning off its heater supply. This load is the usual tuned circuit loss load R_0. Curve 2 represents the peak RF voltage obtainable when the diode is permitted to rectify so that the additional load of supplying the diode plate loss and the power dissipated in the dc load resistance R is placed in shunt with the initial circuit loss resistance R_0. Curve 3 represents the dc voltage obtained across load resistor R.

Suppose there is no AM present and the input is e_{g1} as shown by the dotted line which intersects curves 3 and 2 in Fig. 2. If a sudden negative pulse of AM occurs, e_{g1} drops. The rectifier will continue to rectify until the output voltage reaches the dc voltage stored across condenser 19. To find where this occurs on curve 1, draw the horizontal line (shown by long dashes) from the intersection of the dotted line with curve 3 until it intersects curve 1. Now drop a perpendicular (a dot-dash line) from the last named intersection to the e_g axis at e_{g2}. Thus it is deduced that the downward modulation capability of the diode compressor before it becomes ineffective is given by

$$m = \frac{e_{g1} - e_{g2}}{e_{g1}} = 1 - \frac{e_{g2}}{e_{g1}}. \tag{1}$$

Meanwhile, the change in level of the voltage e across the tank 13 is the voltage difference between curves 2 and 3 along the dotted line. If the downward modulation by this change is called M, then

$$M = 1 - \eta \tag{2}$$

where η = diode rectification efficiency factor.

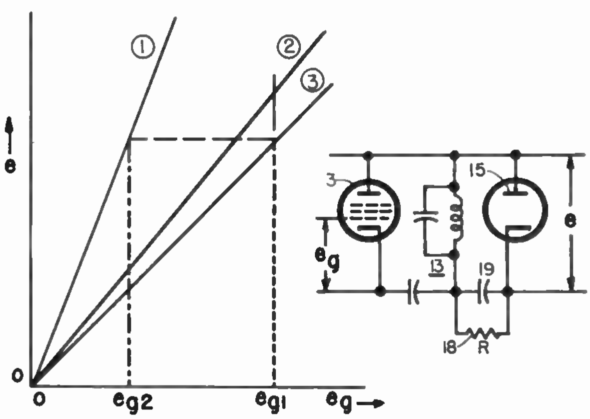

Fig. 2—Schematic diagram of diode compressor and operational characteristics.

The modulation compression factor, expressed as a number greater than unity, is thus

$$\nu = \frac{m}{M} = \frac{1 - \dfrac{e_{g2}}{e_{g1}}}{1 - \eta}. \tag{3}$$

Thus if $\nu = 5$, it means that the residual modulation is one-fifth of the uncompressed modulation. An examination of (3) shows that the amount of compression increases as η improves or approaches unity.

Now the numerator of (3) contains e_g terms. To express these in terms of circuit constants would be more useful. Since the value of e is the same all along the dash line of Fig. 2,

$$e_{g2} g_m R_0 = e_{g1} g_m \eta R_L \tag{4}$$

where g_m = tube mutual conductance, R_L = total load resistance effective for curve 2, and R_0 = load resistance effective for curve 1.

Solving (4) for e_{g2}/e_{g1}

$$\frac{e_{g2}}{e_{g1}} = \frac{\eta R_L}{R_0}. \tag{5}$$

Substituting this into (1),

$$m = 1 - \frac{\eta R_L}{R_0} \tag{6}$$

and thus (3) becomes

$$\nu = \frac{1 - \dfrac{\eta R_L}{R_0}}{1 - \eta}. \tag{7}$$

In order to make ν as high as possible, the ratio of R_L/R_0 should be made as small as possible. This calls for as high an R_0 as possible, and as low an R_L as possible.

Now R_L can be expressed in terms of η, R_0, and the dc load resistance R_{dc}. An asymtotic value for R_L, good for reasonably high values of η, is given by

$$R_L = \frac{R_0 R_{dc}}{R_{dc} + \dfrac{4\eta^2 R_0}{1 + \eta^2}}. \tag{8}$$

η may be expressed in terms of the diode internal plate resistance r_p and R_{dc} by the following asymptotic equation good for values of $\eta > 0.75$, or where $(r_p/R_{dc}) < 0.04$.

$$\eta = 1 - \left(\frac{r_p}{0.3 R_{dc}}\right)^{2/3}. \tag{9}$$

Now multiply the term inside the parentheses by R_0/R_0, and rearrange (9) to read

$$\eta = 1 - \left(\frac{1}{0.3} \cdot \frac{r_p}{R_0} \cdot \frac{R_0}{R_{dc}}\right)^{2/3}. \tag{10}$$

Eqs. (8) and (10) may now be substituted into (6) and (7) to yield

$$m = 1 - \left[1 - \left(\frac{1}{0.3} \cdot \frac{r_p}{R_0} \cdot \frac{R_0}{R_{dc}}\right)^{2/3}\right] \left\{\frac{1}{1 + \dfrac{4R_0}{R_{dc}} \left\{\dfrac{\left[1 - \left(\dfrac{1}{0.3} \cdot \dfrac{r_p}{R_0} \cdot \dfrac{R_0}{R_{dc}}\right)^{2/3}\right]^2}{1 + \left[1 - \left(\dfrac{1}{0.3} \cdot \dfrac{r_p}{R_0} \cdot \dfrac{R_0}{R_{dc}}\right)^{2/3}\right]^2}\right\}}\right\} \tag{11}$$

and

$$\nu = \frac{m}{\left[\dfrac{1}{0.3} \cdot \dfrac{r_p}{R_0} \cdot \dfrac{R_0}{R_{dc}}\right]^{2/3}}. \tag{12}$$

Fig. 3 shows a plot of R_{dc}/R_0 vs R_0/r_p for various values of m, using (11). Fig. 4 shows a plot of ν expressed in db vs R_0/r_p for various values of m, using (12). The values for R_{dc}/R_0 for use in (12) in plotting Fig. 4 are obtained from Fig. 3.

These curves may be used as follows, illustrated by an example:

Given:

A diode with an $r_p = 1500$ ohms

Tuned circuit impedance = 150,000 ohms

Protection desired to $m = 0.5$.

1) Calculate R_0/r_p. In the example $R_0/r_p = 100$.

2) Determine R_{dc}/R_0 from Fig. 3. In the example, at $R_0/r_p = 100$ and at $m = 0.5$, $R_{dc}/R_0 = 2.15$, whence $R_{dc} = 2.15$ (150,000) = 322,000 ohms.

3) Determine ν from Fig. 4. In the example, at $m = 0.5$ and $R_0/r_p = 100$, $\nu = 18$ db or 7.94 times.

4) The solutions are valid because

$$\frac{r_p}{R_{dc}} = \frac{1500}{322{,}000} = 0.00466$$

which is less than 0.04.

The time constant of the compressor is determined by the capacitance of C_{19} and its discharging resistance, R_{dc}. The discharge equation is

$$e = E_0 \epsilon^{-\alpha t} \tag{13}$$

where E_0 = initial voltage across R_{dc}

$$\alpha = \frac{1}{Rc}.$$

Now the terminal value of $\epsilon^{-\alpha t}$ is seen to be $1 - M$, and for small values of M,

$$\epsilon^{-\alpha t} = 1 - \alpha t \text{ or } \alpha t = M. \tag{14}$$

Thus

$$\frac{t}{R_{dc}C_{19}} = M \text{ or } C_{19} = \frac{t}{MR_{dc}}. \tag{15}$$

Now the time t for downward modulation is one fourth of a cycle at a frequency f, so

$$t = \frac{1}{4f} \text{ and hence } C_{19} = \frac{1}{4fMR_{dc}} = \frac{\nu}{4fmR_{dc}}. \tag{16}$$

Now using the figures given in the example and assume $f = 90$ cps,

$$C_{19} = \frac{7.94}{4(90)(0.5)(322{,}000)} = 0.137\ \mu\text{fd}. \tag{17}$$

Discriminator

A discriminator circuit may be very elaborate as in the ratio detector, or it can be relatively simple, as in the case of the circuit selected for the delta detector.

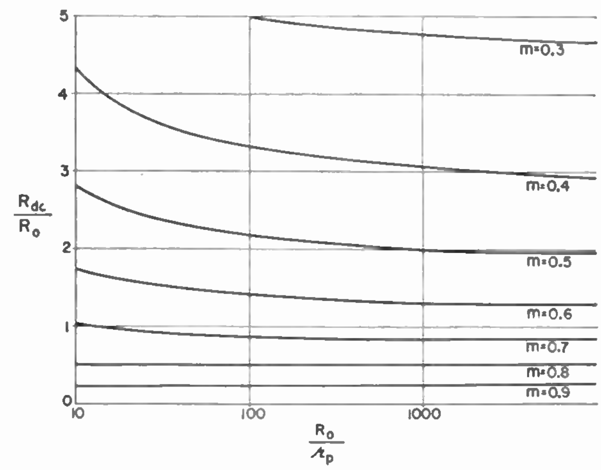

Fig. 3—Plot of R_{dc}/R_0 vs R_0/r_p for various values of m.

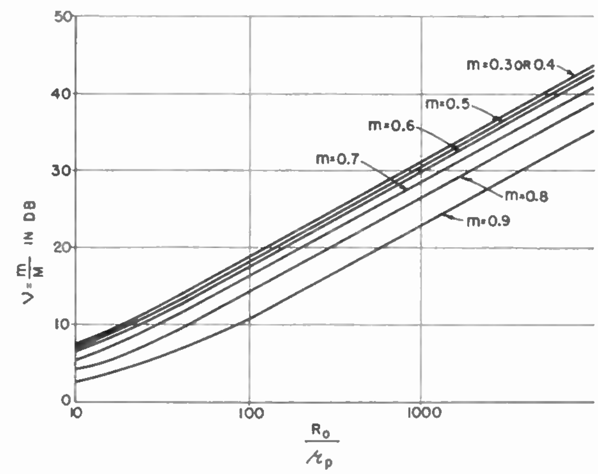

Fig. 4—Plot of ν vs R_0/r_p for various values of m.

This discriminator consists of two condensers and one inductance, and since one of the two condensers is the input capacitance to a tube, the physical parts of the discriminator reduce to only one condenser and one inductor. The RF circuit is shown in Fig. 5. A source of FM waves e_1 feeds a network consisting of a shunt tuned tank circuit L_1C_1 in series with the tube input capacitance C_2. A resistance R is shown across L_1C_1 to represent the effective resistance of the tank caused by the tank circuit losses. The ratio of e_2/e_1 may now be derived making these assumptions:

$$a = \frac{C_1 + C_2}{C_1} \text{ or } C_1 = \frac{C_2}{a - 1} \tag{18}$$

$$L_1C_1 = \frac{1}{\omega_0^2} \tag{19}$$

$$Q = \frac{R}{\omega L_1} \tag{20}$$

where there is derived

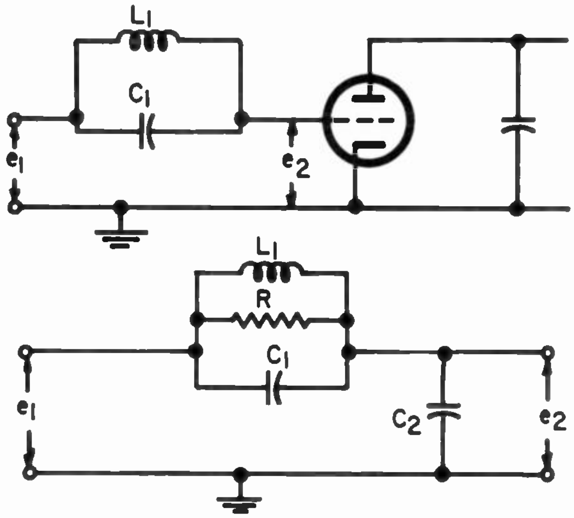

Fig. 5—Discriminator circuit.

$$\left|\frac{e_2}{e_1}\right| = \sqrt{\frac{Q^2\left(1 - \frac{f^2}{f_0{}^2}\right)^2 + 1}{Q^2\left(1 - \frac{af^2}{f_0}\right)^2 + 1}}. \tag{21}$$

Fig. 6 shows plots of $|e_2/e_1|$ for various Q's for $a = 1.04$. Translated to 4.5-mc terms, a variation of 0.011 is equivalent to 50 kc, the maximum peak-to-peak deviation. A reasonably linear section of the $Q = 50$ or $Q = 100$ curve may be used centered at about 0.986.

In the discriminator circuit, wider separation of peaks is obtainable by increasing the factor a. Flatter (and at the same time slightly wider peak separation) slopes are obtainable with lower Q values. By way of example, if the triode in a 6AV6 tube is being used, the input capacity including tube, socket, and strays is about 6.2 $\mu\mu$fd. With $a = 1.04$, then, tank condenser C_{23} of Fig. 1 should be

$$C_{23} = \frac{6.2}{a - 1} = \frac{6.2}{1.04 - 1} = 155\ \mu\mu\text{fd}. \tag{22}$$

The tank is tuned to

$$f_0 = \frac{4500\text{ kc}}{0.986} = 4563\text{ kc}. \tag{23}$$

The fixed bias for the power detector may be obtained from suitable negative voltage elsewhere in the receiver, or it may be self-generated by grid-cathode rectification. A grid leak of 10 megohms is suggested. The audio coupling condenser in the order of 0.05 to 0.1 μfd will keep the distortion at low audio frequencies at reasonably low levels. A condenser of 0.01 μfd on the other hand appears to be inadequate.

The discriminator circuit driving voltage developed across condenser 21 in Fig. 1 is best found by trial. The size of condenser 21 depends on the RF current flowing

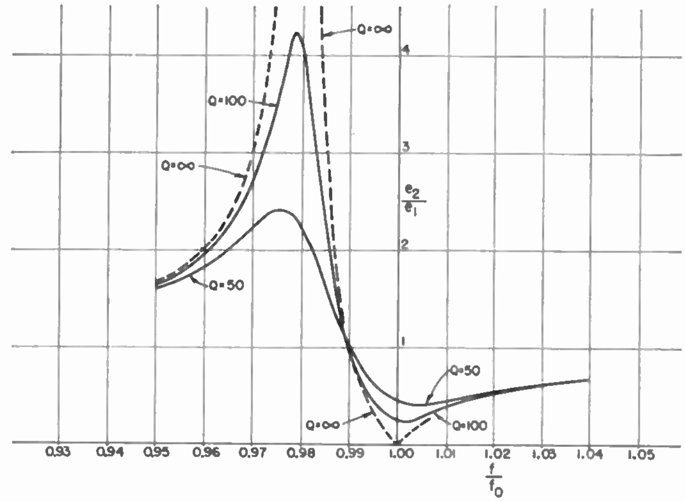

Fig. 6—Frequency response of detector network.

through it, the plate voltage on the triode, the amplification factor of the triode, and the shunting effect of resistor 20. Condenser 21 will usually lie in the range from 50 to 500 $\mu\mu$fd.

AM Cancellation

The fundamental component of the undesired AM is cancelled by returning the grid leak by-pass condenser through resistor 20 to ground instead of directly to ground. The value for resistor 20 will vary depending upon other circuit values and tube characteristics but the range of values usually places resistor 20 somewhere between 500 and 2000 ohms. It is suggested in the design stage that a 3000-ohm rheostat be inserted in the circuit for resistor 20 and that it be adjusted for best AM rejection or fundamental frequency cancellation.

De-Emphasis

The de-emphasis time constant may be adjusted by selecting the proper capacitance for condenser 28 which by-passes the triode anode to ground. There are three resistances in parallel to form the resistive component of the de-emphasis network, namely, the volume control 29, the plate coupling resistor 27, and the internal plate resistance of triode 16. In a typical example, assuming resistor 27 is one megohm, potentiometer 29 is 500,000 ohms, and the tube plate resistance is 200,000 ohms, the total shunt resistance becomes 125,000 ohms. If the de-emphasis time constant T is 75 μsec, then the value for condenser 28 is

$$C_{28} = \frac{T}{R} = \frac{75 \times 10^{-6}}{125{,}000} = 600\ \mu\mu\text{fd}. \tag{24}$$

Detector Output Level

With a $B+$ voltage of 250 volts for the supply voltage for the detector tube and with a frequency deviation of ± 25 kc, a peak-to-peak maximum audio voltage of about 60 volts may be expected. The voltage depends on various circuit parameters so that it may be as low as 40 volts or as high as 90 volts depending upon the particular design.

This level of voltage is sufficiently high for a conventional audio power output tube to be driven directly from the detector without the necessity of employing an intermediate audio frequency amplifier stage between the detector and the output tube.

Conclusions

Design data have been presented to enable the circuit designer to specify component parts of the system and to predict the performance.

AM compression in the range from 12 to 24 db and a peak-to-peak audio frequency output voltage of about 60 volts with cancellation of the fundamental component of the undesired AM is obtainable with readily available low cost tubes and circuit components.

Correspondence

Design of Transistor RC Amplifiers*

A recent paper by Murray[1] pointed out that the prediction of emitter-current variations with temperature could not be done accurately based solely on a knowledge of the I_{c0} temperature dependency. The method suggested by Murray of a semigraphical approach using $V_c - I_c$ curves at the temperature extremes gives good results but is a rather tedious procedure and does require the production of the V_cI_c curves at the temperature extremes of the problem.

It is possible to obtain a simple design procedure based on I_{c0} variations but accounting for an error in the I_e change predicted solely on the I_{c0} change.

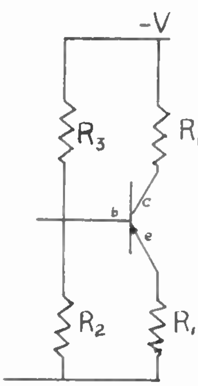

Fig. 1.

For the circuit in Fig. 1 define:

$$\frac{\delta I_e}{\delta I_{co}} = S_I \qquad \frac{\delta I_e}{\delta V_{be}} = S'$$

$$\therefore \frac{\delta I_e}{\delta T} = S_I \frac{dI_{co}}{dT} + S' \frac{dV_{be}}{dT}. \tag{1}$$

It can be shown for the configuration in Fig. 1 that

$$S_I = \frac{G_1}{G_2 + G_3 + G_1(1-\alpha)} \doteq \frac{G_1}{G_2 + G_3} \tag{2}$$

$$S' = -S_I(G_2 + G_3) \doteq -\frac{1}{R_1}. \tag{3}$$

Also

$$\frac{\delta V_{be}}{\delta T} \doteq -\frac{0.785 - V_{be}}{T} \text{ for germanium} \tag{4}$$

$$\frac{\delta I_{co}}{\delta T} \doteq 9060 \frac{I_{co}}{T^2} \text{ for germanium.} \tag{5}$$

Substituting (2)–(5) into (1),

$$\frac{dI_e}{dT} = \frac{0.785 - V_{be}}{R_1 T} + S_I \frac{9060 I_{co}}{T^2}. \tag{6}$$

The relative magnitude of these two factors will depend on the temperature range of interest.

To find the error in stability factor introduced by considering S_I alone,

$$\frac{\Delta I_e(\text{actual})}{\Delta I_e(S_I)} = \frac{0.785T - V_{be}T + 9060 I_{co} R_1 S_I}{S_I 9060 I_{co} R_1}$$

$$= 1 + \frac{0.785 - V_{be}}{S_I 9060 I_{co} R_1} T$$

$$= 1 + \text{error} \tag{7}$$

from which

$$S(\text{actual}) = (1 + E)S_I. \tag{8}$$

For germanium units we may assume average values of

$$\frac{0.785 - V_{be}}{T} \doteq 0.00212 V/°C,$$

$$I_{co} \doteq K_{co}\epsilon^{-9060/T},$$

$$I_{co} \doteq 5\mu a \text{ at } 30°C,$$

from which $K_{co} = 505 \times 10^5$.

Further we may define $E' = ES_IR_I$ which will give a general error independent of S_IR_1 from which E may be calculated for a specific case from which

$$E' = 47.3 \times 10^{-16} T^2 \epsilon \frac{9060}{T}.$$

A plot of this function is shown in Fig. 2.

The practical use of this assumes that a figure for S_I has been arrived at based on the I_{co} variation over the temperature extreme and the allowable I_e change.

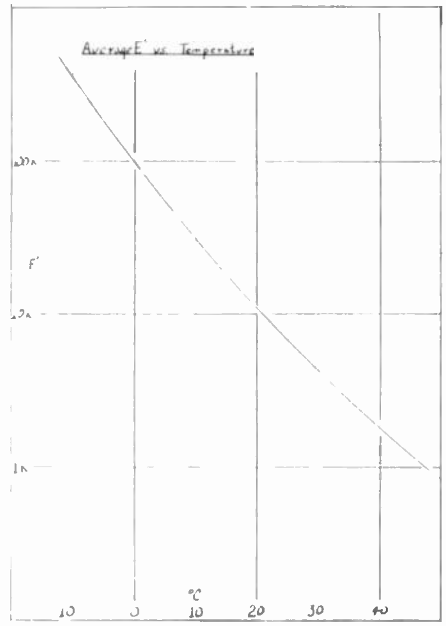

Fig. 2.

Suppose this $S_I = 10$ and the temperature range to be 0°–50°C. An average E' (at 25°C) = 6800. Assuming $R_1 = 1000$ gives $E = 0.680$. If one desires an S(actual) of 10 one must design for $10 = S_I(1 + 0.680)$ or $S_I = 5.95$.

The design of the bias network can be carried out based on methods outlined by Shea,[2] but using the corrected value of S_I.

It has been found that for the units tested, the maximum error in predicted stability was 15 per cent.

It might be worth observing in conclusion that if a really serious stability problem exists, the use of linear stabilization methods is not the most economical.

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* Received by the IRE, December 11, 1958.

[1] R. P. Murray, "Design of transistor RC amplifiers," IRE Trans. on Audio, vol. AU-6, pp. 67–76; May–June, 1958.

[2] R. F. Shea, "Transistor Circuit Engineering," John Wiley and Sons, Inc., New York, N. Y.; 1957.

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James F. Novak (M'55) was born on September 5, 1926, in Oak Park, Ill. He served in the Naval Reserve during World War II working as a technician with a radome development group at the Naval Air Materiel Center in Philadelphia, Pa. In 1952 he received the B.S. degree in electrical engineering from the Illinois Institute of Technology, Chicago.

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Mr. Novak is a member of the Acoustical Society of America, the Chicago Acoustical and Audio Group, and the Institute of Physics.

❖

Joseph A. Worcester was born on April 6, 1909 at Middletown, N. Y. He received the E.E. degree from Cornell University, Ithaca, N. Y. in 1930.

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After graduation Mr. Worcester worked at Bell Telephone Laboratories, New York, N. Y., from 1930 to 1932. He has been with the General Electric Company, Utica, N. Y., from 1935 to date. His current position is consulting engineer in advance circuits, housewares and radio receiver division.

❖

Robert B. Dome (A'27–M'58–SM'43–F'48) was born on October 12, 1905, in Tell City, Ind. He received the B.S. degree from Purdue University, West Lafayette, Ind., in 1926, and the M.S. degree in electrical engineering from Union University, Schenectady, New York, in 1929.

In 1926 he joined the General Electric Company in Schenectady as a student engineer in the Radio Engineering Department and became engineer-in-charge at the South Schenectady Development station in 1928 where he was engaged in the development of high-power short and long wave transmitting equipment and antennas. In 1934 he transferred to the Radio Receiver Department and became consulting engineer for the Television Receiver Department in 1943, which is his present position.

R. B. Dome

Mr. Dome has published a number of papers and holds a large number of patents in the radio and television fields. He is a member of Sigma Xi and RESA. He has also served on the monochrome and color television NTSC committees and has served continuously on either the Papers or Editorial Review Committees of IRE since 1940.

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