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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 TRANSACTIONS of the I.R.E. Professional Group on Audio

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Editorial Committee

ELECTRONIC MUSIC - PAST, PRESENT AND FUTURE*

Earle L. Kent
C. G. Conn, Ltd.
Elkhart, Indiana

In keeping with the PGA policy of including, from time to time, subjects outside of the strict boundaries of Technology, we have invited Dr. Kent to submit his very fine review of Electronic Music. Future developments in Audio techniques, and, therefore, this subject is bound to be of interest to all PGA members. — Editorial Committee.

In ancient time the Greek musical theorists applied the word "organic," as a general term, to instrumental music. For several centuries the term organ has been applied to musical instruments, wherein the tones are generated by pipes blown by suitable means with air under pressure. The principle involved in obtaining sound from a pipe with wind pressure dates back to about 284 to 246 B.C. By 100 A.D. the organ embraced about three octaves, and water was used to obtain steady wind pressure. During this long period the organ was gradually improved, and electricity has played an increasing role in recent years. First, electrically driven blowers replaced the man-power required to furnish the air pressure and then electromagnets replaced the manually operated air valves to greatly reduce the physical effort required to play the organ.

For the past 55 years various attempts have been made to produce musical tones electrically. The first instrument of note that produced tones electrically was built long before vacuum tube amplifiers were available, and it comprised over thirty carloads of equipment. Since that time several hundred patents have been issued on instruments that produce musical tones electrically, and great strides have been made in electronic circuits and components. At present there are a number of electronic musical instruments on the market and activity in this field is growing at an exponential rate.

There are several factors that work against making improvements in musical instruments. In fact, improvements in musical instruments are up against the most formidable combination of conservatism confronting any product. Between the manufacturer and the consuming public stand several conservative forces. They are the teacher, the composer, the music publisher, the musician, and the listening public.

During the past 75 years the Conn company has witnessed cases of inertia that are almost unbelievable. The flute used today is hardly distinguishable from the flute as it came from Boehm's hands in 1847, but only in the past

*Part of paper presented before the Chicago Section, Institute of Radio Engineers on December 19, 1952. Manuscript received January 23, 1953.

forty years has it won undisputed and universal acceptance. Similar stories can be told about the clarinet, cornet, french horn, and other musical instruments. Some improvements have made the grade after many years of struggle, but many have not. For example, the Conn company developed the mezzo-soprano saxophone in F about 30 years ago. Many thought that the E♭ alto was not a good lead voice for the saxophone choir because it was really an alto and pitched too low for a lead. The B♭ soprano was difficult to play. The F saxophone seemed to be the answer. After building the instrument, its beauty of tone and ease of playing was recognized at once, but there were no F parts in orchestrations. We worked closely with music publishers to persuade them to include parts for this instrument in their music, but their answer was that they would do so when there were a sufficient number of F saxes being played to warrant it. You can still hear an F sax now and then in some dance bands, but after a ten year struggle with the publishers we gave it up.

Many examples could be given about the teachers who refuse to teach students on a so-called "non-standard" instrument, the superstitions of directors and musicians who all want something new so long as it isn't different, and even the listening public who put the brakes on progress by their apathy and even antagonism toward the new instruments and new effects. New and unfamiliar sounds are not generally liked.

The obstacles met by the traditional instruments have been multiplied for the electronic instruments. In 1936 the American Guild of Organists published the statement that in all-electric instruments, the basic tone is necessarily devoid of all color. The various orchestral instruments have been imitated by the flue pipes and reeds of the pipe organ with varying degrees of fidelity. When an attempt is made to create these orchestral tones by electronics, we find that organists do not like too realistic duplication of the tones of the original instrument, but that they insist the tone qualities must be those with which they have become familiar on the pipe organ. We can create new and beautiful tonal colors not found on pipe organs, but when organists hear these tones they are confused and displeased. It is possible to do things with electronics that pipe organ builders have tried in vain to do, and yet, when we do these things they are opposed by organists.

In spite of these restrictions and handicaps, electronic musical instruments are steadily making progress. There are several good reasons for this progress. The electronic servants that serve mankind so well in so many different ways are quite capable of producing musical tones. It is taken for granted that electronics should serve in recording, reproducing, amplifying, and transmitting music, so it is not unreasonable to expect it to serve in the actual creation of music. In fact, it is now possible to produce beautiful tones electronically in solo type or in organ type instruments that have greater versatility than was ever dreamed of in traditional instruments. When tones are produced electronically the musician does not need to exert any energy pounding, blowing, or scraping, to produce the tones, but can devote his entire attention to the control of the tone to make music. When a beginner starts to learn to play most of the traditional instruments, he must spend considerable time in developing muscles, muscular control, and techniques in order to make a tone that does not sound very discouraging to him and to all who have to listen to it. With an electronic musical instrument, the beginner can make a variety of beautiful tones from the beginning, and can start at once to learn to play the right note at the right time with the proper expression.

Electronic musical instruments of the organ type have become an accepted fact to the general public and are enjoying widespread application in liturgical, concert, and entertainment music.

These instruments are the only twentieth century musical instruments; and they are opening new horizons in music.

A demonstration of the versatility and scope of the electronic organ was shown in the presidential nominating conventions here last summer. The International Amphitheatre, where the conventions were held, had no organ; and the Connsonata was selected to fill the need for this type of music. The seating capacity is twelve thousand. In addition to the permanent seating space on the main floor and balcony, there is a clear central arena 123 x 236 feet with a clear height of 73 feet. The floor space alloted for the organ was less than 45 square feet, but this was more than enough since room remained for several chairs. To install a pipe organ, with sufficient power to dominate, supplement, or override the sound level, occasioned by the customary enthusiastic demonstrations of so many people in so large a room, would not be practicable on a temporary basis, and impossible in the installation time and space allowed.

The two basic functions in musical instruments are the generation of musical tones and the control of those tones. The generation of good tones in a practicable manner is relatively easy due to the advanced state of electronic circuits and components. Loudspeakers have generally been the weak link here because the general run of radio type loudspeakers have not been adequate for the stiff requirements. The high peak power, the 32 to 12 or 15,000 cycle range, and the sustained nature of the tones, all demand good transient response, good frequency response, and low distortion. However, speakers are now available that give good results if properly used.

The control of the tone includes: the starting and stopping of the tone (the envelope of the attack and release are quite important), the quality of timbre of the tone, the tremulant, the dynamics or loudness, the location of the source of sound (including echo effects) and the formant of the tones. Solo and ensemble effects must be considered, and it is important that the organist "feel at home" at the console and find the controls where he is accustomed to finding them.

The organ type instrument is the grandest and most magnificent of all instruments invented by human genius. The organ challenges the total capacity of the organist's mind and body in the production of music, but it does not give him the intimate control over the tone that is possible with a monophonic or solo type instrument; so we must not neglect this phase of electronic instruments, and more will be said about that later.

The Connsonata uses plate keyed Hartley oscillators for the tone generators. There is one oscillator for each key and pedal plus a few extra that are keyed through couplers that operate beyond the range of the keyboard. The frequency vibrato is produced by introducing a low frequency sine wave at the grid of the tube. A flute-like tone is produced at the tap on the tank coil, and a pulse tone is produced across a resistor in the cathode circuit. By mixing these two basic ingredients and by means of filters for formant control, we produce a basic organ tone — the diapason, the oboe, the vox humana, the chorus reed or organ trumpet, and others. One test of an organ is to determine how the four basic tone families sound in ensemble. First, the diapason group, second the flutes, third the strings, and finally the reeds.

Organs are called upon to play concert music: they are used extensively in churches for liturgical music, they are used for novelty effect, particularly on the radio, and they must be able to play music for general entertainment or theatrical work.

While electronic organs are now generally accepted by the general public in all types of organ application, the same is not true for electronic solo-type instruments. They have not supplemented conventional musical instruments in the way that they can and should. It is possible to build solo electronic instruments that fulfill all musical requirements, but it will take time to overcome the inertia that prevents a good instrument from reaching the fellow who needs it badly. Small school bands or orchestras often have but one or perhaps none of certain instruments in their organization. For example, perhaps the group has one good oboeist or french horn player, and he graduates leaving no one to play those parts. A good electronic instrument could play any of the missing parts well enough to be a great help. Perhaps one section is weak and needs reinforcement to obtain proper balance. An electronic instrument could do it. The teacher who must teach several instruments he does not play could play the parts for the student if he had a versatile electronic instrument that would closely simulate the instruments he is teaching. There are many other instances where electronic solo instruments can substitute for traditional instruments. If they can do a good job as a substitute, it will gradually become apparent that they can stand on their own merits in serious music. I have talked to progressive directors who feel that this need can and will be filled satisfactorily with electronic musical instruments. Of course, one fellow retorted that he never expected to see a band marching down the football field with power cords dragging behind. That would be out of the question, but with the advent of the transistor and other improvements in small effecient electronic components, we may some day see an all electronic band without the cords dragging behind.

Technological advances in electronic circuits and components will bring improvements in electronic musical instruments just as fast as people will tolerate them, and it is already evident that these changes will come faster than they have on the traditional instruments.

In addition to the improved electronic instruments of the solo and organ type, I believe the future will find still different instruments in use. These instruments will further expand the possibilities in music by doing what is impossible on other instruments. For example, I started the development of such an instrument, which I call an electronic music box, as a spare time project. This instrument was described at the National Electronics Conference last year.[1] Percy Grainger, a noted musician and composer, came to Elkhart to see my music box and was very enthusiastic about its possibilities, since he had composed music having such irregular rhythm and scales that no one could play it. He surprised me by saying that the greatest handicap to music is the musicians hands. Of course, Mr. Grainger is a visionary rebel, and some musicians would take offense at that statement, but what I have in mind would not hurt musicians. I do not believe that actors have suffered any loss because Walt Disney is able to produce motion picture fantasies by animated cartoons. He has just expanded the motion picture field, and I believe that something like my music box will produce musical fantasies for recordings, and the like, that could never be produced through manual manipulations. Lack of time has made my progress slow in completing the development of the music box, but I am still working on it and perhaps you'll hear more about it later.

These new devices will not only remove the limitations in tonal production, as do the other electronic musical instruments, but will remove the limitations of the musician's hands. They will make it possible for a composer to produce his music directly without going through interpreters.

1. Kent, Earle L., "An Electronic Music Box," Transactions of the IRE-PGA. Proceedings of the Naticnal Electronic Conference, Vol. 7, February, 1952.

REPORT OF SOUTHWESTERN I.R.E. CONFERENCE AND ELECTRONICS SHOW San Antonio, Texas -- February 5, 6 and 7, 1953

B. B. Bauer
Shure Brothers, Inc., Chicago 10, Illinois

Following the pattern of growth of electronics in the South and Southwest, the 1953 Southwestern IRE Conference was bigger and better than ever. Registration exceeded 1100 and two parallel sessions of technical papers and over 100 exhibits provided an interesting and busy time for all those who attended.

The session on Audio was held on Saturday morning, February 7, under the chairmanship of Dr. Richard E. Lane of the University of Texas. Approximately 225 members attended this session. Four papers were presented as follows:

1. "Electron Beam Reproducing Head for Magnetic Tape Recording" -- Dr. A. M. Skellett and Dr. Lawrence E. Loveridge, National Union Radio Corporation, and Mr. J. Warren Gratian, Stromberg-Carlson Company.

   Present reproducing heads for magnetic tape depend upon the threading of flux through a coil. The output is proportional to the frequency of recording and to the speed of the tape. The electron tube reproducer guides the flux from the tape into a cathode-ray type of tube where it deflects the beam in accordance with the absolute magnitude of the flux. Thus, the output is dependent neither on the frequency nor the speed of the tape. In addition to this, the new head can be made many times as sensitive as the old ones. This development has been sponsored by the Bureau of Ships of the United States Navy under Contract NObsr-57452.

2. "Status of Military Research and Development in Acoustics and Audio" -- Mr. Paul Weber, Bureau of Ships, Navy Department, Washington, D. C.

   The fields of acoustics and audio engineering are finding many new military applications within the Army, Navy and Air Force. Continued progress is also being made toward improving the performance of existing audio systems and techniques to better meet increasingly stringent operational requirements. Some recent developments and proposed projects in these fields will be discussed. Problems which are as yet unsolved will be mentioned. The procedure established within the Department of Defense for coordinating all research and development in the broad field of acoustics-in-air will be described.

3. "Acoustical Damping of Loud Speakers" -- Mr. B. B. Bauer, Shure Brothers, Inc., Chicago, Illinois

   Acoustical damping can be applied advantageously to loudspeakers. The advantages of acoustical damping as a replacement for, or as an adjunct to the electrical damping are discussed. Equivalent circuits for the loudspeaker and the enclosures, and simplified theoretical and experimental design methods are described. Transient performance of the electrical circuit and of the loudspeaker is demonstrated.

4. "The Fluid Sound Phonograph Pickup" -- Mr. Bruce D. Eytinge, Institute of Inventive Research, San Antonio, Texas

A new type of transducer has been developed wherein the resistance of a column of conducting fluid is varied by mechanical motion which is made to increase or decrease the cross-sectional area of the column. Experimental phonograph pickups using this principle show extremely good frequency response at the extreme lows, and output voltages comparable with crystal transducers. Other properties of these units are discussed.

A spirited discussion which ensued caused your reporter to almost miss his plane out of San Antonio.

Arrangements are being made to include some of these papers in the coming issue of TRANSACTIONS.

Two of the highlights of the Conference were the toastmastering of Mr. Trevor H. Clark, of the Southwest Research Institute, and the humor of Dr. C. P. Boner, of the University of Texas, who was the Banquet Speaker.

REPORT OF THE SECRETARY-TREASURER

Marvin Camras
Armour Research Foundation, Chicago 16, Illinois

Our membership, our income, and our treasury balance continue to grow. At the end of 1952, we had a total of 2031 members, compared to 794 at the end of 1951. Our financial statement at the end of the year compares very favorably with the figures for the previous period and for the previous year.

	Period Ending Dec. 31, 1952	Period Ending Sept. 30, 1952	Period Ending Dec. 31, 1951
RECEIPTS			
Assessments	$4864.00	$3976.00	$1530.00
Other Income	961.61	769.21	101.00
Matched Funds	2579.00	1500.00	1000.00
Total Income	$8404.61	$6245.21	$2631.00
EXPENSES	3959.59	3273.09	1190.77
BALANCE ON HAND	$4445.02	$2972.12	$1440.23

REPORT OF THE ACTIVITIES OF THE TAPESCRIPT COMMITTEE

A. B. Jacobsen
University of Washington
Seattle, Washington

The Tapescript Committee has procured and prepared for distribution a number of technical papers primarily on the subject of audio. Recordings were made of Convention papers, in general, and offered to interested groups.

On the whole, Convention papers have not proved as satisfactory as a specially prepared paper. The best example of the specially prepared paper is "Germanium — The Magic Metal," which was produced by the General Electric Company for the IRE especially to determine the feasibility of this medium of distribution of technical papers. Two copies of this paper are currently being distributed and a total of between 25 and 30 showings have been made. Response to this paper, and the manner of distribution, has been good. "Germanium — The Magic Metal" runs approximately 40 minutes and has about 75 slides in color. Narration is by two General Electric engineers who are directly associated with the subject.

Standards for Tapescripts should be set and it is felt that full track at $7\frac{1}{2}$ inches per second tape speed, on 1200 foot rolls, is the most universal sound recording medium, and with $3\frac{1}{4}$ x 4 slides, 35 mm slides, or single frame 35 mm film, should be desirable standards. Besides the slides and the sound recording, a copy of the script is very important, and a summary of suitable length for meeting announcements and notices, should be available to the program chairman. A few references to applicable literature would be of considerable value to the person who must attempt to answer questions at the end of the recorded presentation.

We believe that there is a great need for specially prepared, high quality recorded technical papers for those groups who would otherwise be unable to procure live papers. The Tapescript Committee has been aided by many individuals and concerns in carrying out this preliminary phase of the program.

ANALYSIS OF A SINGLE-ENDED PUSH-PULL AUDIO AMPLIFIER*

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Summary

This paper deals with a theoretical circuit analysis of a single-ended push-pull audio amplifier. Linear tube characteristics and small signals are assumed. The problem of impedance matching is first discussed. The properties and the requirements of a satisfactory driver stage are analyzed, and an output stage using an impedance matching output transformer is discussed. The effects of the plate-to-ground capacitance of the driver stage on the frequency-gain characteristic and of the choice of the tube and circuit parameters are analyzed. The paper concludes with some experimental results indicative of the inherent properties of the amplifier.

In a recent article by Peterson and Sinclair,[1] special circuits have been suggested by which a high fidelity, low distortion audio amplifier can be constructed without using output transformers or in which the requirements of an output transformer can be greatly simplified. The present article will give a theoretical analysis of the basic circuit used by Peterson and Sinclair. It will be observed that many desirable characteristics of this amplifier circuit can be deduced from this analysis.

Equivalent Circuit of the Basic Single-Ended Push-Pull Amplifier Circuit

The basic circuit of a single-ended push-pull amplifier is shown in Fig. 1a. Assuming class-A operation with resistive load, and neglecting the effects of interelectrode capacitances, an equivalent circuit of the phase inverting driver stage and the single-ended push-pull output stages is shown in Fig. 1b. With the circuit constants, tube parameters, and loop currents as indicated, one may write the following simultaneous equations:

$$\begin{aligned}
&I_1(2R + r_{p1} + \mu_1 R) + I_2 R_L - I_3 R_L = \mu_1 E_1 \\
&I_1 \mu_2 R - I_2 R_L + I_3(r_{p2} + R_L) = 0 \qquad (1)\\
&-I_1(r_{p2} + \mu_2 R) + I_2(r_{p2} + R_L) - I_3 R_L = 0
\end{aligned}$$

*Presented at the Audio Session of the IRE Convention in Long Beach, California on August 29, 1952. Manuscript received December 22, 1952.

From which the individual loop currents can be evaluated. The net current flowing through the load resistance R_L is

$$I_3 - I_2 = \frac{-\mu_1 E_1 (2\mu_2 R + r_{p2})}{R_L(4R + 2\mu_1 R + 2\mu_2 R + 2r_{p1} + r_{p2}) + r_{p2}(2R + \mu_1 R + r_{p1})} \qquad (2)$$

The power output P_o is given by $P_o = (I_3 - I_2)^2 R_L$. The power sensitivity, which is defined as the ratio of the power output to the square of the input voltage is then given by

$$\text{Power sensitivity} = \frac{P_o}{E_1{}^2} \quad \frac{\mu_1{}^2 A^2 R_L}{R_{p2}^2 \left[1 + \frac{R_L}{r_{p2}} (2+A\right]^2} \qquad (3)$$

$$\text{where} \quad A = \frac{2\mu_2 + r_{p2}/R}{2 + \mu_1 + r_{p1}/R}$$

The condition for maximum power sensitivity by varying the load resistance can be obtained by taking the derivative of Equation (3) with respect to R_L and equating to zero. The optimum load required is thus

$$R_L \text{ (for maximum power sensitivity)} = \frac{r_{p2}}{2 + A} \qquad (4)$$

It can easily be proved that Equation (4) also represents the source impedance of the amplifier. The value of this maximum power sensitivity is given by

$$\frac{P_o}{E_1{}^2} \max = \frac{u_1^2 \; A^2}{4\, r_{p2}\, (2 + A)} \qquad (5)$$

To get high power output, A should not be small. But consideration of negative feedback (which will be discussed in the next section) puts restrictions on the choice of the tube parameters, and indicates that A should be small.

The ratio of power output to the maximum power possible is given by

$$\frac{P_o}{P_o \max} = \frac{4 \frac{R_L}{r_{p2}} (2 + A)}{\left[1 + \frac{R_L}{r_{p2}} (2 + A)\right]^2} \qquad (6)$$

Equation (6) is calculated for various values of A and is plotted in Fig. 2 as the ratio of $\frac{r_{p2}}{R_L}$ is varied. It is seen that even for a wide variation of A, the matching conditions permit r_{p2} to have a value of from 2 to 3 times that of R_L.

The above discussion on maximum power sensitivity made use of linear analysis with resistive load. However, the actual maximum un-distorted power output, which is limited at one extreme by grid current and at the other extreme by plate current cutoff, will occur at higher load resistance values. The consideration of plate and screen-grid dissipation also favors higher load resistance.

The Phase-Inverting Driver Stage

Information concerning the behavior of the driving stage can be deduced by solving for the current I_1 in Equation (1). Thus

$$I_1 = \frac{-\mu_1 E_1}{r_{p1} + (2 + \mu_1)R + R_L(2\mu_2 R + r_{p2})/(r_{p2} + 2R_L)} \quad (7)$$

Equation (7) is in a form similar to that of an amplifier employing negative current-feedback of one considers the term $r_{p1} + (2+\mu_1)R$ as the internal resistance R_i and $(2\mu_2 R+r_{p2})R_L/(r_{p2}+2R_L)$ as the effective load resistance R_e of the driver. $\mu_1 E_1$ is the fictitious voltage.

The voltage across R_e is the actual voltage output of the single-ended push-pull output stage. The driving voltages for that stage is derived from the voltage drops through the resistances R. Thus the a.c. voltage across the plate-to-cathode of the phase-inverting driver supplies both the driving as well as the output voltages. This puts a restraint on the D.C. plate voltage supply for the driver. A high D.C. plate voltage greater than the sum of these a.c. voltages is needed in order that serious non-linear distortion can be avoided.

The negative feedback that exists between the input and output of the amplifier stages can be expressed in terms of a feedback factor β, the ratio of the feedback voltage available at the grid terminal of the driver stage to the output voltage across the load. Thus

$$\beta = (2 + r_{p2}/R_L) \,/\, (2\mu_2 + r_{p2}/R) \quad (8)$$

The inherent feature of this negative current feedback is to maintain the current I_1 practically independent of load changes. This would mean that the driving voltages for the output stages could also be maintained unchanged, a very desirable characteristic of the amplifier if we can achieve it. To so design this stage would require

$$R_i \gg R_e$$

or

$$A/(2+r_{p2}/R_L) \ll 1 \tag{9}$$

and for a matched load, Equation (9) becomes

$$A/(4+A) \ll 1 \tag{10}$$

To satisfy the requirement indicated in Equations (9) or (10) would call for a small value of A. But a smaller A means reduced output voltage and power output. A compromise should always be worked out between these contradictory requirements.

A proper choice of A gives some guidance for the choice of tube parameters and circuit constants. For a typical operation, if $\mu_1 \gg 1$, $r_{p1} \cong R$ and $R > r_{p2}$, then with a matched load for maximum power sensitivity

$$A \cong 2\mu_2/(3+\mu_1) .$$

and

$$\beta \cong 4/2\mu_2$$

Thus, if μ_2 is small, μ_1 is large; A is very small and β is comparatively large. A larger β helps to reduce distortion while large power output is still possible if the input signal is large.

A choice yielding a larger value of μ_1 is always desirable. The larger value of μ_1 allows larger power output while still keeping the reasonably small value for A which is required for constant I_1 operation. It also gives a larger driving voltage for exciting the power stage.

The Power Output Stage

In the basic circuit of this amplifier (Fig. 1a), if one assumes identical tube characteristics; same d.c. plate voltage per tube, and equal and opposite grid swings, the a.c. plate currents add in the load in exactly the same way as in a conventional push-pull circuit and the distortion-cancelling feature is retained. The d.c. plate currents are, however, in series for the plate supply.

The need for a high voltage plate supply (from 800 to 1000 volts) with good regulation to operate this amplifier is, to a certain extent, an objection to most experimenters. If only a moderate plate supply is available, a circuit suggested by Peterson and Sinclair[2] containing an output transformer, is recommended. A modified version of their circuit is shown in

Fig. 3. It uses an output transformer with separate primary windings. Here the d.c. plate currents of the two tubes flow through the two halves of the primary windings in opposite directions so that the problem of d.c. saturation is not acute. The tubes are in parallel across the d.c. plate supply. The signal currents flow through the two halves of the primary windings which are connected in parallel by the by-pass capacitors. The leakage reactance between the halves of the primary windings plays no important part in switching transients[3] and the requirements of the output transformer are not as critical as they were in conventional push-pull circuits.[2]

The linear analysis used thus far can be extended to include the impedance matching transformer provided class A_1 operation in triodes is assumed. Assuming an ideal transformer, equivalent circuits of the output stage of a single-ended push-pull amplifier with impedance matching output transformer are represented in Fig. 4. Fig. 4a can be reduced to Fig. 4b, which shows that as far as the varying components are concerned, the tubes are in parallel. Fig. 4c is another version of Fib. 4b.

Figures 4b and 4c are very similar to the equivalent circuit of a conventional push-pull circuit used for power considerations alone. Thus a simple relationship between these two types of push-pull circuits can be deduced. For impedance matching the single-ended push-pull circuit will need an output impedance which is one-fourth of the plate-to-plate impedance of a conventional push-pull circuit.

The concept of the composite tube which is so helpful in the analysis of conventional push-pull circuits when the path of operation extends beyond the linear region of the plate characteristics can be borrowed to advantage. Thus the analysis of the single-ended push-pull circuit can be extended to include the operation as class AB_1, although graphical analysis is required. To extend the operation beyond class AB_1 into class AB_2 or class B operations is not feasible since the flow of grid current will affect the biases unless special circuits can be arranged.

The Effect of the Plate-To-Ground Capacitance of the Driver Stage on the Gain-Frequency Characteristics of the Amplifier

As was first pointed out by Peterson and Sinclair, the capacitance from plate to ground of the driver stage is going to affect the gain of the amplifier at higher frequencies. With normal circuit and tube parameters, this effect will first be noticed at video frequencies. The previous analysis can be extended to include this effect. In Fig. 1b, if a capacitor C (plate-to-ground capacitance) is connected across the points marked x-x and a loop current is assigned to the loop on the right hand side of the capacitor, one has a new equivalent circuit for this analysis. As the currents flowing through resistances R are different now, we may expect different driving voltages on the grids of the output tubes. This difference is, however, small if the shunting capacitance is small or the frequency in consideration is not too large. By writing the simultaneous equations of the loops, and solving the currents as before, one obtains, for a sinusoidal input signal, the net current through the load resistance as

$$I_3-I_2 = \frac{-\mu_1 E_1\left[\mu_2 + \frac{1}{j\omega CR}\ (2\mu_2 + r_{p2}/R)\right]}{R_L\left\{(1+\mu_1+\frac{r_{p1}}{R})\left[(2+\mu_2+\frac{r_{p2}}{R}+\frac{r_{p2}}{R_L})\right]+\frac{1}{j\omega CR}\left[2(\mu_1+\mu_2+2+\frac{r_{p1}}{R})+\frac{r_{p2}}{R_L}(2+\mu_1+\frac{r_{p1}}{R}+\frac{R_1}{R})\right]\right.} \tag{11}$$

And thus the ratio of the voltage gains at higher and lower frequencies can be found by taking the ratio of Equation (11) to Equation (2), and is given by

$$K_h/K_L = \frac{\dfrac{\mu_2}{2\mu_2+r_{p2}/R} - j\dfrac{1}{\omega CR}}{\dfrac{(1+\mu_1+\frac{r_{p1}}{R})(2+\mu_2+\frac{r_{p1}}{R}+\frac{r_{p2}}{R_L})}{(2+u_1+\frac{r_{p1}}{R})(2+\frac{r_{p2}}{R_L}+\frac{2\mu_2+r_{p2}/R_L}{2+\mu_1+r_{p1}/R})} - j\dfrac{1}{\omega CR}} \tag{12}$$

For abbreviation, let us define the following terms:

$\omega_o = 1/CR$, or $f_o = \frac{\omega_o}{2\pi} = \frac{1}{2\pi RC}$, the upper reference frequency,

which is determined by the RC time constant of the resistance R and the plate-to-ground capacitance C of the driver stage.

$\nu = \omega/\omega_o = \frac{f}{f_o}$, the frequency ratio

$$N = \frac{\mu_2}{2\mu_2 + \frac{r_{p2}}{R}},$$

and

$$M = \frac{(1+\mu_1+r_{p1}/R)(2+\mu_2+\frac{r_{p2}}{R}+\frac{r_{p2}}{R_L})}{(2+\mu_1+\frac{r_{p1}}{R})(2+\frac{r_{p2}}{R_L})+2\mu_2+\frac{r_{p2}}{R_L}} \tag{13}$$

Then the ratio of the voltage gains becomes

$$K_h/K_L = (1+jN\nu)/(1+jM\nu) \tag{14}$$

The magnitude of this gain ratio can be expressed in terms of decibels gain as

$$20 \log_{10} K_h/K_L = 20 \log_{10} \sqrt{1 + N^2 \nu^2} - 20 \log_{10} \sqrt{1 + M^2 \nu^2} \qquad (15)$$

and the phase angle is $\phi = \tan^{-1} N\nu - \tan^{-1} M\nu$ (16)

The factors M and N in the previous equations together with the factors A and β mentioned in Sections I and II set up a practical limitation for the choice of the circuit and tube parameters. To make the factor A reasonably small, but still seeking a sizable power output with adequate negative feedback, one would choose a large μ_1, while μ_2 and r_{p2} are kept reasonably small. Again, let $R \cong r_{p1}$ and $r_{p1} > r_{p2}$, and for a matched resistive load, Equation (13) becomes

$$N \cong \frac{\mu_2}{2\mu_2} = \frac{1}{2}$$

$$M \cong \frac{(2+\mu_1)\left[4 + \mu_2\left(1 + \frac{2}{3+\mu_1}\right)\right]}{4(3 + \mu_1 + \mu_2)} \qquad (17)$$

It is then permissible to assign different values of μ_1 and μ_2 and calculate the range of M that is of practical interest. Fig. 5 is the plot of these calculations. It is noticed that within practical considerations r_{p2} and μ_2 should not be very large to give a small A, hence it leaves little choice for the M value if μ_1 is to be as large as permissible.

The choice of M and N affects the frequency response of the amplifier. This can be seen from a calculation of the gain-frequency characteristic computed for different assumed values of M and N. Let M = 1.5, N = 0.5, Equations (15) and (16) are computed and plotted as the solid-line curve in Fig. 6. The three discontinuous dotted straight-line segments are the asymtotical plots of the gain characteristic with a first break at $\nu_1 = 0.667$ and a second break at $\nu_2 = 2.0$. If f_o is chosen as one million cps, then from Fig. 6, the amplifier shows a 2.55 decibel loss at 667 kc and a 7 decibel loss at 2mc. At much higher frequencies, the loss will be 9.54 db. Increasing the value of M moves the first break point toward the left. Thus for M = 2, N = 0.5, the break points will be at $\nu_1 = 0.5$ and $\nu_2 = 2.0$ (dot-dash line) respectively. The amplifier will then have a loss of 2.75 db at 500 kc and 12.04 db at higher frequencies. Thus the flat portion of the gain-frequency curve is reduced. It can also be shown that reducing the value of N moves the second break point and the phase shift curve toward the right. The difference in gain between lower and higher frequencies also increases

Experimental Results

Fig. 4 is the actual circuit used in the experiment. Separately adjustable biases for the power tubes were provided. With a pair of 616GA's as output tubes and one section of a 6SN7 as driver, maximum power output of 28 watts was obtained. For this operation, 400 volts were used for plate supply and -30 volts for the grid biases. The plate circuit efficiency thus obtained was 31%. In another operation, using -35 volts as bias, the plate efficiency increased to 35.3% with a power output of 22.7 watts. Frequency response in each case was flat to within two decibels from 30 cycles per second to 200,000 kilocycles per second. A Western Electric output transformer 134C was used in the output system for impedance matching purposes.

Several experimental results are worth mentioning. Fig. 7 is a plot of the power output against load resistances for two different operating voltages. Signal input in both cases is maintained constant to a level such that practically no distortion is observed in a cathode-ray oscilloscope over the entire range of the load variation. The curves show well defined maximum at certain resistances which coincide with the values computed from one-fourth of the plate-to-plate resistance for a conventional push-pull circuit. Fig. 8 shows the variation of the driving voltage across the grid-to-cathode of the output tubes for two different operating conditions. In both cases, the driving voltages decrease as the load resistance is increased. The drooping driving voltage characteristic is the main cause for the drooping power output curve in Fig. 7 at larger load resistances. Larger output power can be obtained at these larger load resistances by increasing the input signal.

A curve for the output voltage is also plotted along with the driving voltage. It increases with increasing load resistance. The sum of this voltage and two times the grid driving voltages constitute the total a.c. voltage available across the driver tube. With increasing load resistance, the re-division of the total a.c. voltage among the effective resistance (the output voltage) and the resistances R (the driving voltages) is the cause of these variations. As the aim of this circuit design is to obtain large output power, the condition for maintaining constant driving voltage (small A) mentioned in our theory is not used.

Acknowledgement

The author wishes to express his appreciation for the valuable discussions carried out during the course of this investigation with Professor E. L. Chaffee of Harvard University and Dr. D. B. Sinclair and Dr. A. Peterson of General Radio Company, Cambridge, Massachusetts.

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1. A. Peterson and D. B. Sinclair, "A Single-Ended Push-Pull Audio Amplifier," Proc. IRE, Vol.40, pp.7-11; January, 1952.

2. See Peterson and Sinclair, loc.cit.

3. A. P. T. Sah "Quasi-Transients in Class-B Audio-Frequency Push-Pull Amplifiers," Proc. IRE, Vol.24, pp.1522-1541; November, 1936.

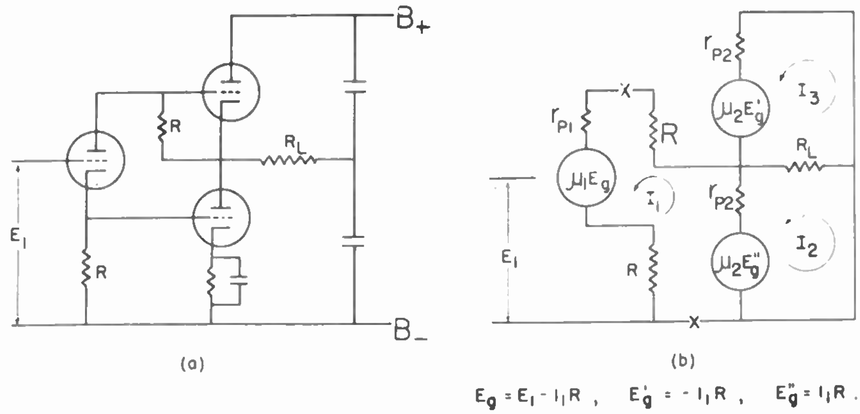

Fig. 1 (a) The basic circuit and (b) the equivalent circuit of a single-ended push-pull audio amplifier.

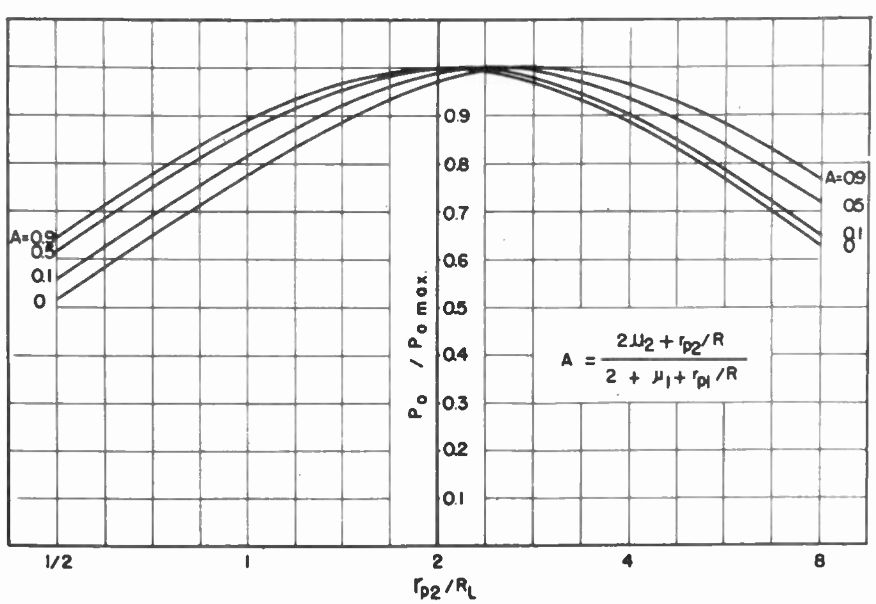

Fig. 2 Power sensitivity as a function of load resistance.

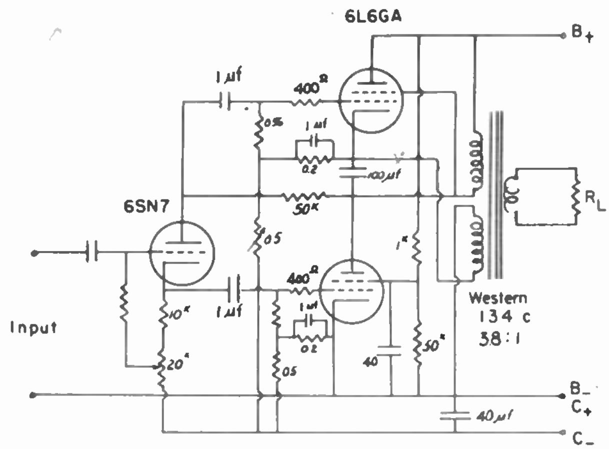

Fig. 3

Method of using an impedance matching transformer to put the output tubes in parallel across the dc plate supply taken from the paper by Peterson and Sinclair with modification to adjust the grid bias of the output stages independently.

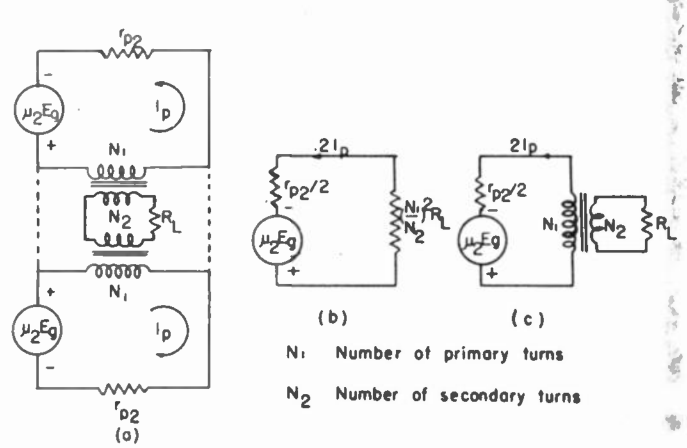

Fig. 4

Equivalent circuits of the output stage of a single-ended push-pull amplifier with impedance matching output transformer used to connect the output tubes in parallel across the dc plate supply.

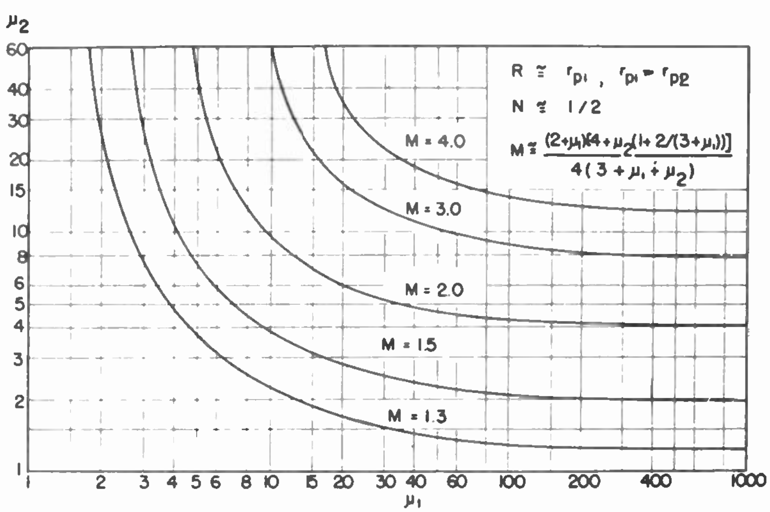

Fig. 5

Range of tube parameters for practical considerations.

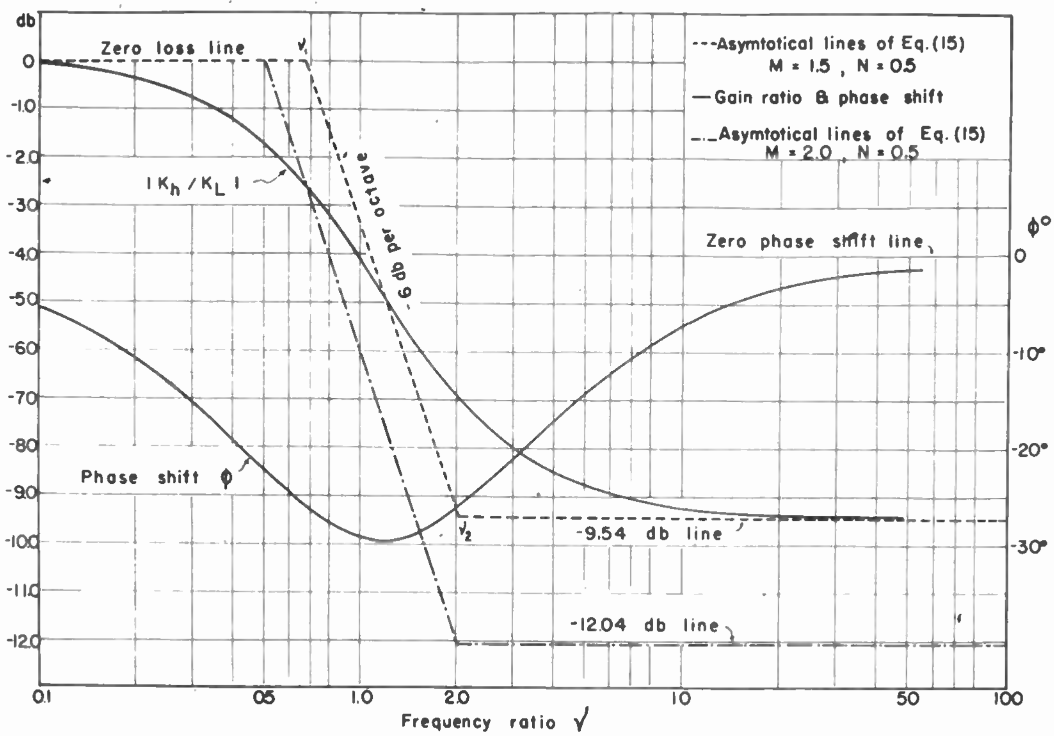

Fig. 6

Magnitude and phase of the gain ratio as a function of the frequency ratio.

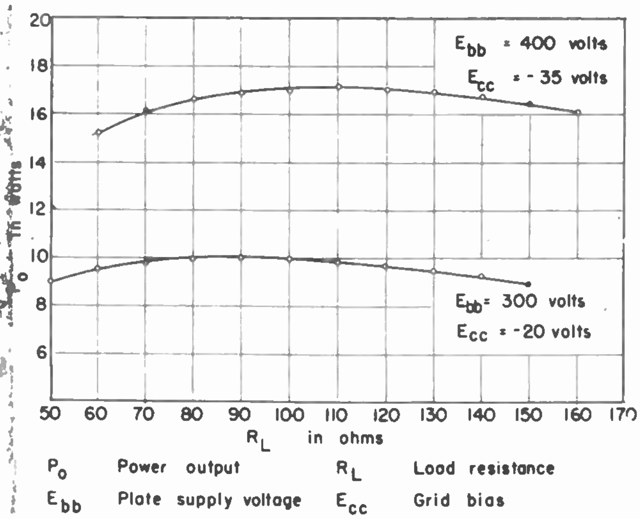

Fig. 7
Power output - load resistance curves for constant signal input.

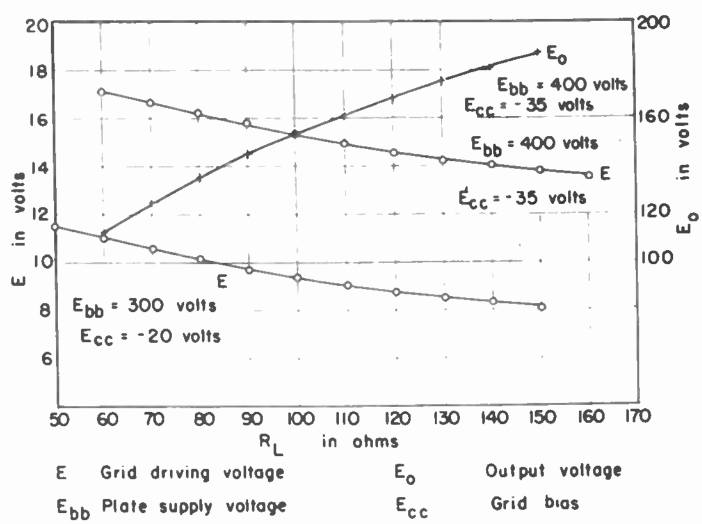

Fig. 8
Driving voltage and output voltage - load resistance.

EQUALIZATION OF MAGNETIC TAPE RECORDERS FOR AUDIO AND INSTRUMENTATION APPLICATIONS*

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As a starting point for discussion, let us consider equalization requirements in terms of an "Ideal System" without losses. An "Ideal System" would consist of the following. A constant current amplifier connected to a record head; the record head producing a constant flux vs. frequency on the tape, and a playback head picking up the signals from the tape and feeding an amplifier whose amplification drops off at the rate of 6 Db. per octave across the band. The 6 Db. per octabe playback amplifier is required because the output from the ideal playback head is directly proportional to frequency when reproducing a constant flux vs. frequency signal. Unfortunately, a number of factors are present which make a practical system differ considerably from this ideal.

First, there are several possible losses associated with the record and reproduce heads. The electrical losses are produced by eddy current and hysteresis effects in the core and capacity effects in the windings. In a well designed and constructed head, these effects are not a problem in the audio range, but do enter the picture somewhat in instrumentation recorders operating in the 100 Kc. region.

The high frequency loss due to the length of the playback gap usually determines the upper frequency limit of the recorder. When the effective length of this gap equals the reproduced wave length, a null or cancellation will occur. The loss in decibels at other wave lengths on the tape is equal to

$$20 \text{ Log } \frac{\sin \pi \, \theta/\lambda}{\pi \, \theta/\lambda}$$

where θ is the effective playback gap length and λ is the reproduced wave length. From this expression a theoretical loss of 4 Db. will occur when a wave length of .0005" is reproduced with a head incorporating a gap length of .00025". This would be the case at 15,000 cycles and a tape speed of $7\frac{1}{2}$ inches per second. The writer has found that reproduce heads with this gap length can be built with surprising uniformity, making reproduction down to .0005" wave length achievable in production equipment.

Heads incorporating larger gaps, although desirable from the standpoint of output, are seriously limited in high frequency range at the lower tape speeds.

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The record head gap length does not bear the same relation to the frequency characteristic as that of the playback head. The magnetization left on the tape is determined primarily by the trailing edge of the gap and not by its length. The sharpness and, of course, straightness of this edge are paramount. The record head gap length is important from other aspects, however. An excessively small record gap would produce a decaying field through the oxide layer on the tape. This decaying field in turn would increase low frequency distortion since the oxide layer in direct contact with the gap would have to be recorded at a relatively high level to compensate for the only partially magnetized undercoat of oxide. The wider the record gap, the greater the demagnetization effect on the highest recorded frequencies. Therefore, excessively wide gaps are to be avoided. .001" to .002" has been found to be the most acceptable range. The so called demagnetization effect is created by partial erasure of a high frequency recorded signal at the time of recording by the high frequency bias field.

The remaining losses in a practical tape system are associated with the tape itself. The oxide particle size and uniformity of dispersion are large factors in the high frequency characteristic of a tape. Large particles or poor dispersion will result in reduced area of contact with the head gap. This reduced area of contact has little effect while recording and reproducing long wave lengths, but will drastically effect high frequency performance. Paper base tapes are unsatisfactory in this respect as the roughness of the paper, and subsequently the oxide coating, prevent proper contact with the head surface. Coating thickness is another factor which has an indirect bearing on the frequency characteristic. As the coating thickness is decreased, the bias requirement will decrease. Consequently, high frequency demagnetization will be reduced and high frequency record efficiency will increase. Unfortunately, low frequency distortion will increase with decreased coating thickness so reduction of coating thickness beyond a certain point is impractical.

Another factor, of course, is the characteristics of the oxide itself. Lower demagnetization losses can be accomplished on some of the new tapes which incorporate oxides selected for low bias requirements and high output. This higher performance oxide permits less coating thickness for the same low frequency output level as compared with previous professional tapes.

Because of variation in performance which is possible due to the tape, it is best to equalize a recorder with a sample of tape known to be a "centerline" of the tape manufacturers' tolerances and to use only tape of that manufacture of a known equivalent.

Let us now consider the problems of equalizing the audio recorder. The equalization of audio recorders at a given tape speed requires a study of the noise and distortion characteristics of the system. A good approach to the problem is to evaluate the system with the tape equalization arbitrarily divided in record and playback. The established energy distribution curves for speech and music serve as a rough guide as to maximum permissible record equalization at any frequency. A noise spectrum analysis will insure that the noise is fairly distributed over the pass band. If this is not the case, the playback equalization should be adjusted until fairly even distribution of noise exists. The record equalization can then be altered to compliment the playback characteristic. These measurements can be weighed on the basis of ear sensitivity, and a better noise characteristic thereby obtained if discretion is used in the amount of correction applied. Such changes in the equalization, based on ear sensitivity, should be carefully checked by listening tests on wide range equipment.

The record system must now be studied from the standpoint of distortion and overload at all frequencies in the pass band. If the record equalization is not greater than the amount required to compliment the energy distribution in speech or music, the overall distortion characteristic will be found satisfactory. If the equalization requirements are greater than that which can be tolerated on this basis, three possibilities exist. The first would be to lower the record level and thereby compromise the signal-to-noise ratio. The second would be to chance the system running into overload distortion at frequencies excessively pre-emphasized. The third would be to lower the record equalization to acceptable limits and raise the post equalization at the expense of signal-to-noise ratio in the raised spectrum. At the professional primary and secondary speeds of 15 and 7½ inches per second, these compromises are unnecessary for full range recording from 30 to 15,000 cycles. The full dynamic range of the tape is therefore available.

The record curves indicated in Fig. 1 and the playback curves indicated in Fig. 2 were established in conjunction with Ampex heads and M.M.M. type 111 tape, construction 5 RBA. These heads display negligable magnetic and electrical losses in the pass band. The playback head gap length is .00025". The bias was adjusted to the point of maximum record efficiency while recording a .015" wave length (1 Kc. at 15 inches per second). The overall response achievable under these conditions is as follows:

At	30"/Sec.	±2Db, 50 to 15,000 cycles
	15"/Sec.	±2Db, 30 to 15,000 cycles
	7½"/Sec.	±2Db, 40 to 15,000 cycles
	3¾"/Sec.	±2Db, 40 to 7,500 cycles

The playback curves are easily accomplished by connecting a vacuum tube operating as a constant current generator to a capacitive load whose reactance equals the generator impedance at 65 cycles.

The 30 inch curve with the exception of the low frequency departure is the characteristic required to compensate the "Ideal System." The slight low frequency departure from the ideal curve was found desirable for the elimination of low frequency thermal effects in playback amplifier input tubes operating at low levels. This departure is made up for by a slight rise in the low frequency playback head characteristic brought about by its physical dimension and by 2½ Db. boost in the record amplifier at 50 cycles. A resistor of such value to effect a time constant of 50 microseconds has been placed in series with the 6 Db. per octave condenser to produce the desired high frequency characteristic at 15 and 7½ inches. The 3-3/4 inch curve is accomplished by a relatively larger resistor effecting a time constant of 200 microseconds. The 15" per second record curve is such that possibility of overload does not exist for the most sever audio requirements. At 7½" per second the record curve is considerably steeper than the 15" curve and reaches 17 Db. at 10 Kc. Listening tests conducted with material recorded on equipment adjusted to this characteristic have shown it to be entirely satisfactory for high fidelity recording. This is the case because of the energy distribution encountered in normal speech and music, and because of a characteristic, of the tape, to compress the high frequency, high intensity peaks occasionally encountered, without appreciable distortion. Sound already pre-emphasized for special effects or from highly resonant microphones might present overload problems at 7½" which, of course, would not occur at the 15" speed. The overall response of a typical Ampex 300 or 403 recorder can be adjusted to +1 Db. from 50 cycles to 15 Kc. at both 7½" and 15" speeds. Slightly wider specifications are advertised to allow manufacturing tolerance and insure the average machine being well within its specifications.

The background noise on a high quality professional recorder, using the tape characteristic described, ranges from 60 to 64 Db. below 3% harmonic distortion. This is true at 7½", 15", and 30" per second. The noise at 3-3/4" per second is approximately 10 Db. higher. The point of approximately 1% harmonic distortion has been found most desirable for operating level and is approximately 6 Db. below the 3% distortion point.

Instrumentation recorders fall into two general categories as pertaining to the subject under discussion.

Pulse systems and carrier systems are in the first category. These systems do not require equalization for the tape system. The second category contains the conventional magnetic recorders employing high frequency bias and recording a band width within the range of 100 cycles to 100 Kc. These recorders incorporate similar electronic systems to audio recorders except for the distribution of equalization. The intelligence recorded on such instruments is usually of a nature that the energy level is fairly uniform over the pass band. This requires a record characteristic with a uniform overload and saturation characteristic in respect to frequency. An unequalized constant current amplifier driving the record head and producing essentially a constant flux recording best suits this requirement. Equalization required for flat overall response is therefore placed in the playback amplifier.

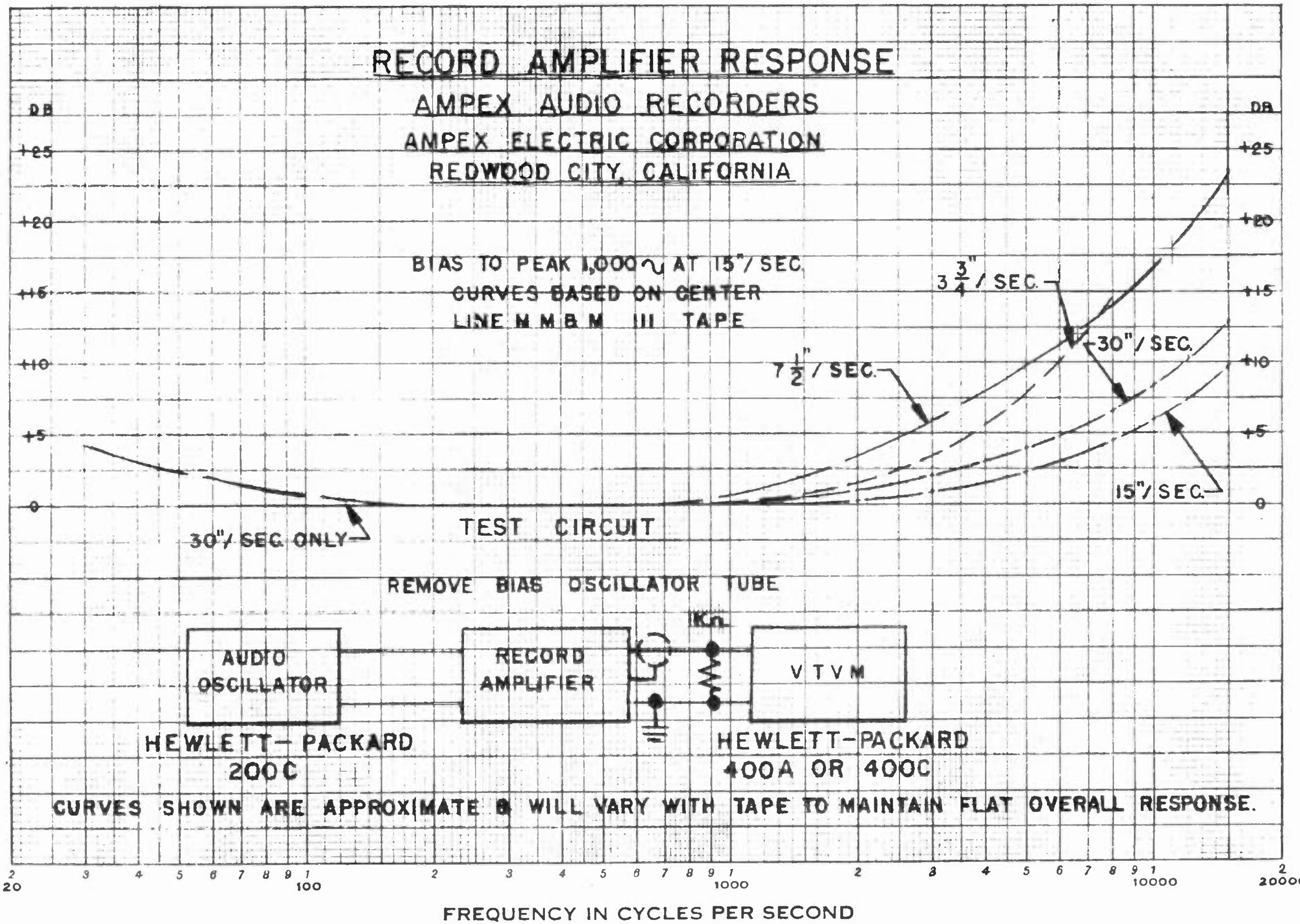
RECORD AMPLIFIER RESPONSE
AMPEX AUDIO RECORDERS
AMPEX ELECTRIC CORPORATION
REDWOOD CITY, CALIFORNIA
DB
+25
+20
+15
+10
+5
0
BIAS TO PEAK 1,000 ~ AT 15"/SEC.
CURVES BASED ON CENTER
LINE M M B M 111 TAPE
3 3/4"/SEC.
30"/SEC.
7 1/2"/SEC.
15"/SEC.
30"/SEC. ONLY
TEST CIRCUIT
REMOVE BIAS OSCILLATOR TUBE
1KΩ
AUDIO OSCILLATOR
RECORD AMPLIFIER
V T V M
HEWLETT-PACKARD 200C
HEWLETT-PACKARD 400A OR 400C
CURVES SHOWN ARE APPROXIMATE & WILL VARY WITH TAPE TO MAINTAIN FLAT OVERALL RESPONSE.
20
100
1000
10000
20000
FREQUENCY IN CYCLES PER SECOND

Fig. 1

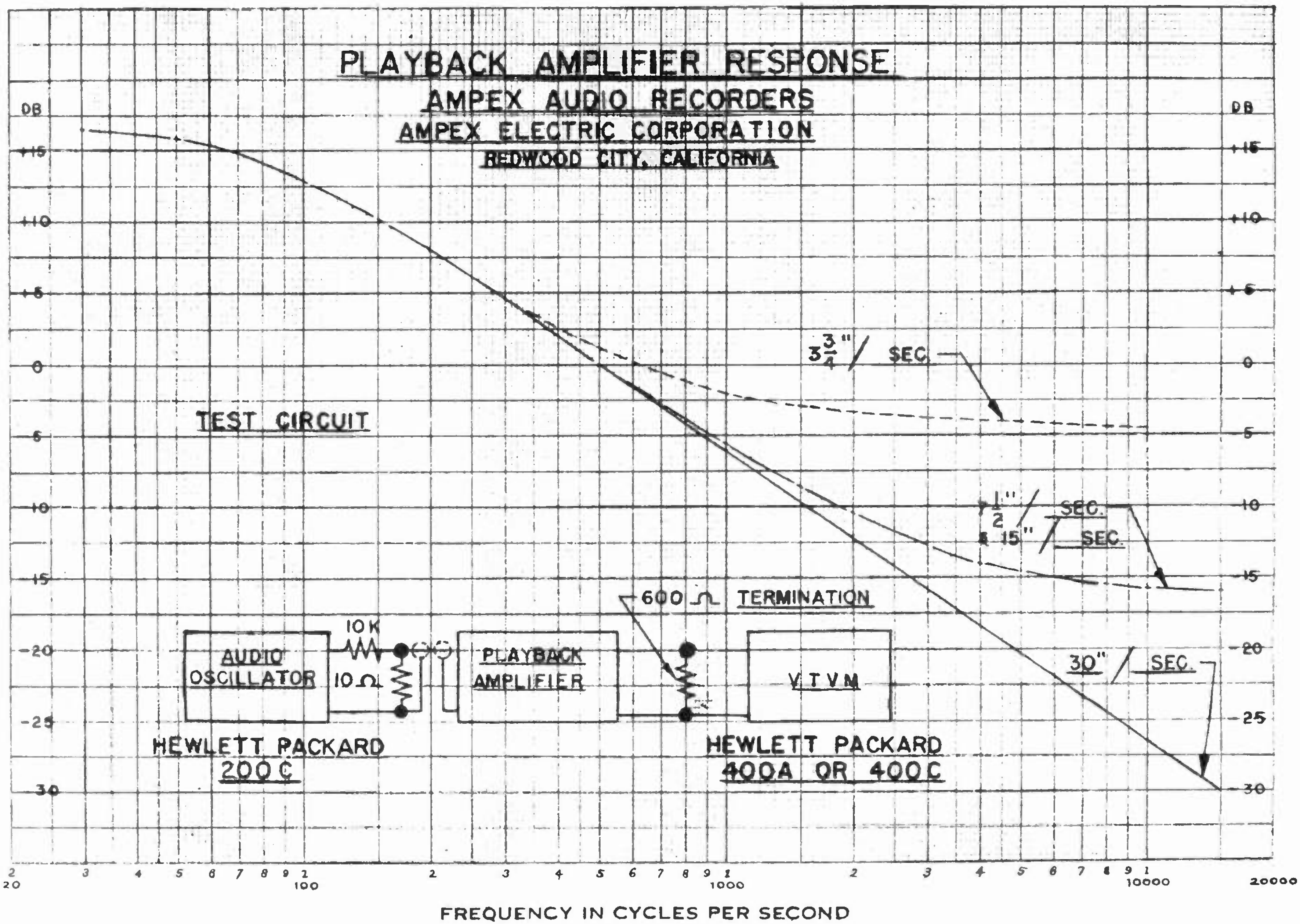
PLAYBACK AMPLIFIER RESPONSE
AMPEX AUDIO RECORDERS
AMPEX ELECTRIC CORPORATION
REDWOOD CITY, CALIFORNIA
DB
+15
+10
+5
0
-5
-10
-15
-20
-25
-30
3¾"/SEC.
7½"/SEC.
& 15"/SEC.
30"/SEC.
TEST CIRCUIT
AUDIO OSCILLATOR
10K
10Ω
PLAYBACK AMPLIFIER
600Ω TERMINATION
VTVM
HEWLETT PACKARD 200C
HEWLETT PACKARD 400A OR 400C
20
100
1000
10000
20000
FREQUENCY IN CYCLES PER SECOND

Fig. 2

WHY FIGHT GRID CURRENT IN CLASS B MODULATORS?*

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ABSTRACT

Grid voltage waveform distortion ranks with output transformer inadequacies among Class B audio amplifiers deficiencies. This article deals with the problem of grid voltage waveform distortion and suggests the use of low plate resistance low mu triode tubes as a solution. By avoiding operating conditions which result in grid current, the major source of distortion is eliminated. Direct coupled driving circuits which encourage large amounts of inverse feedback are readily employed under such conditions. A typical amplifier and its measured characteristics is used for illustration.

1. Why Is This Subject Important?

Class B operation of push-pull amplifiers is one of the best ways known today of obtaining high power with reasonable efficiency in the audio frequency spectrum. In order to keep the harmonic distortion to a tolerably low value, careful attention must be paid to the design of the output transformer and the grid circuit details. Proper transformer design has been expounded by several authors. Grid circuit design, on the other hand, has been more or less by rule of thumb.

2. Why Is Grid Current an Important Cause of Distortion?

Figure 1 illustrates the grid voltage plate voltage relationship in Class B operated vacuum tubes. It will be noted that the two lefthand curves are for a medium mu triode. The extreme left curve shows typical operation with intermediate values of grid voltage. The maximum value of grid voltage is less than the zero bias voltage so that the grid is always in the negative grid region. Plate voltage swing is nominal, being considerably less than the maximum capability of the tube. This operation naturally will be very inefficient because the minimum value of plate voltage is always rather large. The middle curve, on the other hand, shows a quite acceptable plate voltage swing which brings the efficiency to reasonable values but requires that the grid be driven into the positive region to do so. Please note that a small curve is shown which is designated grid current. Grid current

*Presented at the Audio Session of the IRE Convention in Long Beach, California on August 29, 1952. Manuscript received January 23, 1953.

starts flowing at the instant the grid voltage passes the zero line and moves into the positive grid region. It is during this interval when grid current flows that the driver is called upon to deliver power. If the driver voltage regulation is anything but perfect, the voltage will drop below what it should be during this grid current interval. The dotted curve shows a possible modification of the voltage waveform as a result of this grid current. It should be noted that this also affects the plate voltage swing. These points are the distortion which is common to this form of operation. The only way to operate under these circumstances without excessive distortion is to make the driver's impedance so low that its voltage regulation is essentially perfect. This, of course, is very wasteful of driver power and tends to offset the improved amplifier efficiency.

The curve on the right, which is designated as a low mu triode, illustrates a different approach. Here a tube is selected that is capable of efficient operation without the necessity of operating in the positive grid region. Grid voltage is shown always in the negative region. Even so, the plate voltage swing reaches a very satisfactory minimum value, which thus insures reasonably good efficiency, during the negative half of the cycle. Since it is never necessary for the grid to be driven positive, no grid current flows and therefore no power is required from the driver. The driver in this case is simply a voltage amplifier. It is true that the driver voltage required may be considerably higher than that required by a medium mu triode. This voltage swing is relatively easy to obtain since no power is required. It is only when a combination of voltage swing and power output are required that driver design becomes a serious problem.

3. What Are the Characteristics of an Ideal Audio Power Amplifier?

A power amplifier which approaches the ideal is one in which the grid to cathode impedance remains constant throughout an entire cycle of driving voltage. This might be some finite value in which case there would be a constant proportionality between the driving voltage and grid current, or it might be infinite. Unfortunately, there are no suitable vacuum tubes which present a constant finite input impedance. Therefore, we must choose a tube with infinite input impedance. This means that the grid must never be driven positive with respect to the cathode so that grid current never flows. Such a tube obviously never dissipates any power from the driving source and thus has a second advantage. This we shall set aside as characteristic --

a. No driving power required

In order that reasonable efficiency may be realized, it is important that the minimum plate voltage be reasonably low at the peak of the cycle. Since this peak occurs when the grid is at its maximum swing in the positive direction which is identically the zero grid voltage line, it is important that the tube have low plate resistance at this instant. Therefore, a second characteristic of a good audio power amplifier is --

b. Low plate resistance

A low plate resistance tends to make a constant voltage generator out of the amplifier which, for modulator service where the load is determined by the Class C amplifier loading, is a distinct advantage.

4. What Are the Particular Requirements of Tubes for This Service?

In a triode tube the relationship between the important tube parameters is expressed as mu = $G_M R_P$. It is apparent from this that to have low plate resistance the tube must have a high transconductance and a low mu factor. Generally speaking, then tubes which best meet this requirement will be low mu.

A few examples will illustrate the points just made.

Figure 2 is a typical plate current plate voltage characteristic of a triode of medium or high mu. It will be noted that the zero bias line has a rather flat slope. A typical load line has been superimposed to show that driving the grid only to the zero bias is a very inefficient use of the tube. The load line must be extended (dotted portion) into the positive grid region to realize any efficiency.

Figure 3 is a typical plate voltage plate current characteristic of a low mu triode. It will be noted here that the superimposed load line represents relatively low minimum plate voltage values when it intersects the zero bias line. The extra dotted lines illustrate a range of suitable load impedance values. It is to be noted that the plate voltage swing changes very little as the load impedance is changed.

Figure 4 is a typical tetrode characteristic curve. Here it is apparent that by careful choice of the load impedance very high plate circuit efficiency is obtained. It will be noted, however, that small changes in plate load impedance cause serious changes in plate swing. In many cases this is a serious disadvantage. What is not shown in these curves are the screen current variations. Screen current flows in pulses quite like grid current in positively driven grid circuits. Its peak value depends considerably on the minimum plate voltage during a cycle. Extremely good regulation of the screen voltage supply is therefore a must for proper operation of tetrodes in Class B amplifiers.

5. What Circuit Advantages Does Class B_1 Operation Provide?

(Class B_1 operation is defined as an operating condition where the plate current flows for one-half cycle, while grid current never flows for even a fraction of a cycle. The subscript 1 indicates this grid current limitation. A subscript 2 refers to operation wherein grid current flows during at least part of a cycle.)

Since there is no grid current to contend with, the coupling circuits between the driving amplifier and a Class B_1 modulator can be simple and of the type common to ordinary voltage amplifiers. Resistance capacitance circuits can replace the usual Class B driving transformer. Figure 5 is a circuit which illustrates this fact.

Figure 6 is a modification of the schematic shown in Figure 5 wherein a direct coupled circuit is substituted between the driver and the power amplifier. An input stage and a feedback circuit are also shown. This arrangement has very marked advantages where inverse feedback is to be employed. It will be noted that there is only one capacitor in the entire feedback loop which is almost a certain guarantee of proper low frequency phase versus feedback loop response characteristics. There can be no low frequency motor boating with this circuit, no matter how much inverse feedback is used. It will be noted also that the high frequency feedback loop response can be readily controlled by the strategic choice of plate load resistance values and rather simple response shaping circuits.

It will be noted that the driver is operated with its cathode negative to ground. This allows the voltage drop in the driver load resistor to be used directly as the bias and signal voltage for the modulator. Although a triode driver is shown, a tetrode may be used to considerable advantage to obtain higher voltage gain. The screen connection is shown dotted.

Typical characteristics of a modulator using the basic circuit shown in Figure 6, but with three 8C25 tubes in parallel on each side of the push-pull circuit (a total of six in all) are as follows:

Tubes

First Stage -- 12AX7	**Plate Voltage -- 5 KV**
Second Stage -- Push-pull 4-250A	**Input Level -- +10 DBM**
Third Stage -- (6) - 8C25	**Inverse Feedback -- 20 DB**
Power Output -- 25 KW	

Figure 7 shows the inverse feedback loop response for a 50 to 15,000 cycle passband. Both ideal and actual response curves are shown for stable operation with 26 DB of feedback and actual operation with 20 DB.

Figure 8 shows a typical response curve taken from the rectified RF of a 35 KW transmitter which was plate modulated by the modulator described. The high frequency cut-off shown is produced by a low pass filter following the modulator unit.

Typical values of distortion measured from the same rectified RF source are shown in this same figure. It will be noted that from 100 to 3,000 cycles the distortion is less than 1%, while between 60 and 15,000 cycles it is well below 2%.

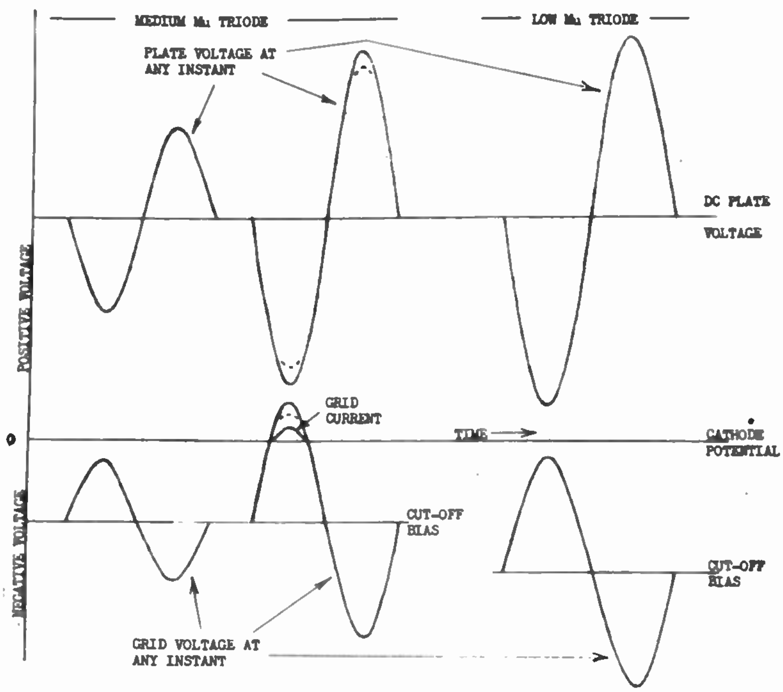

Fig. 1

Typical instantaneous plate and grid voltage relationships for Class B operated tubes.
Left and middle curves are for medium mu, while right curve is for low mu triodes.

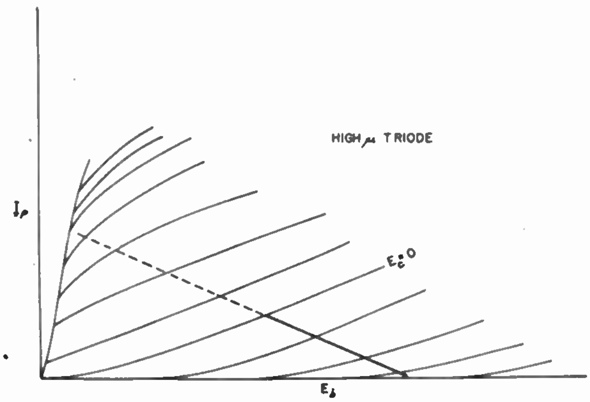

Fig. 2 Typical plate characteristic curves for a high mu triode with a possible load line superimposed.

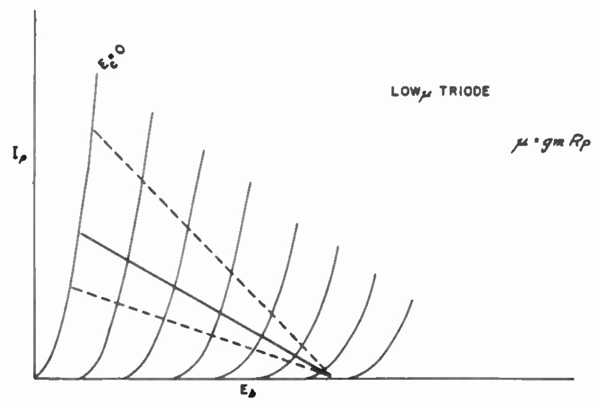

Fig. 3 Typical plate characteristic curves for a low mu triode with several possible load lines superimposed.

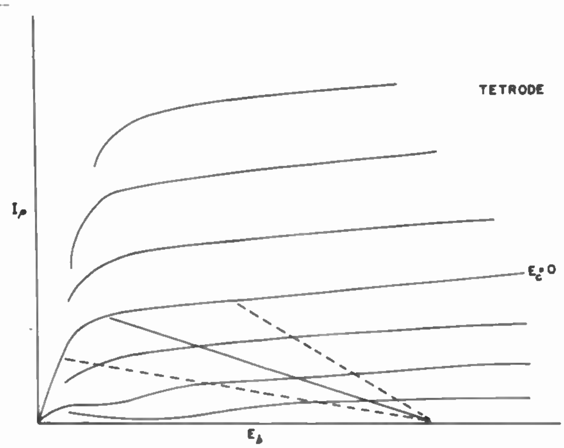

Fig. 4

Typical plate characteristic curve for a tetrode tube with several possible load lines superimposed.

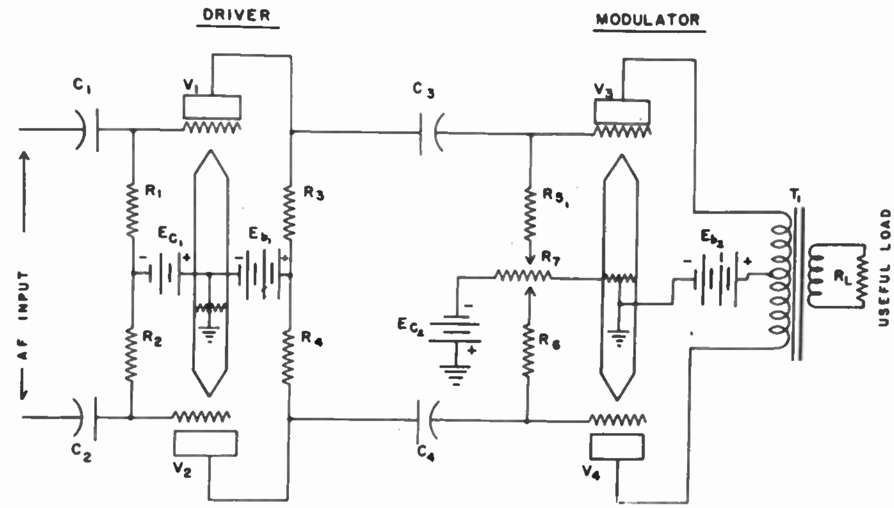

Fig. 5

Convential driver modulator arrangement for Class B_1 operation.

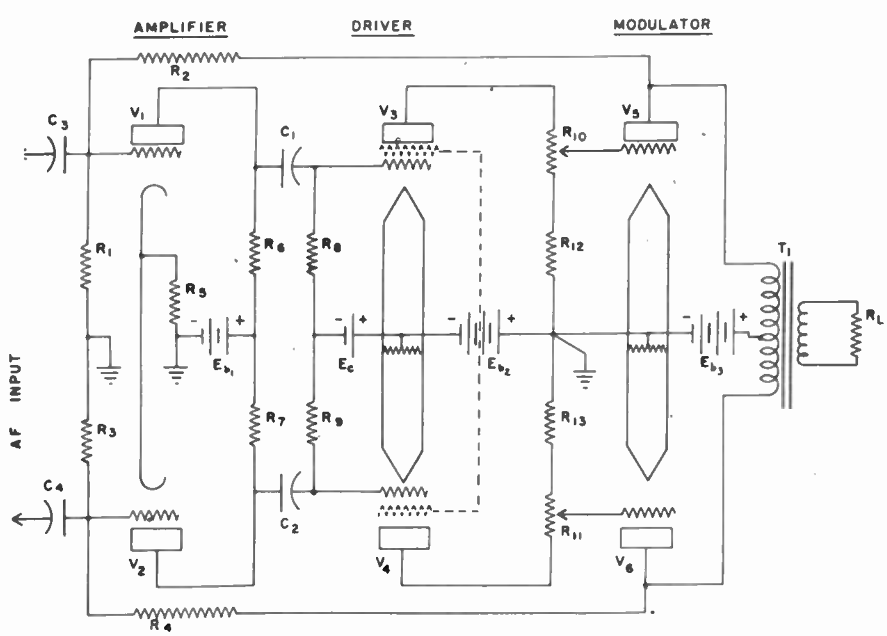

Fig. 6

Direct coupled driver modulator circuit arrangement with improved performance.

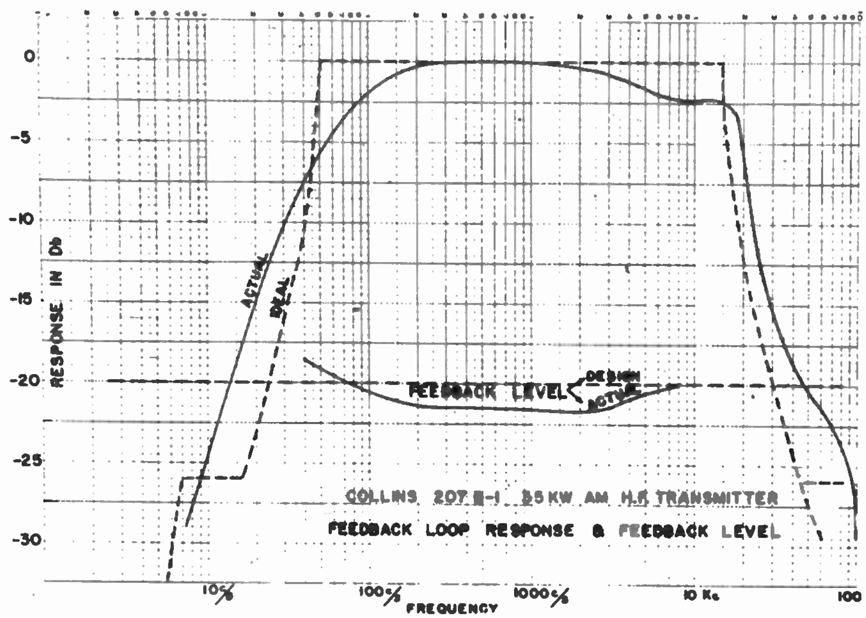

Fig. 7

Typical inverse feedback loop response showing the ideal curve for a 50 to 15,000 cycle passband and a solid curve depicting the curve actually attained.

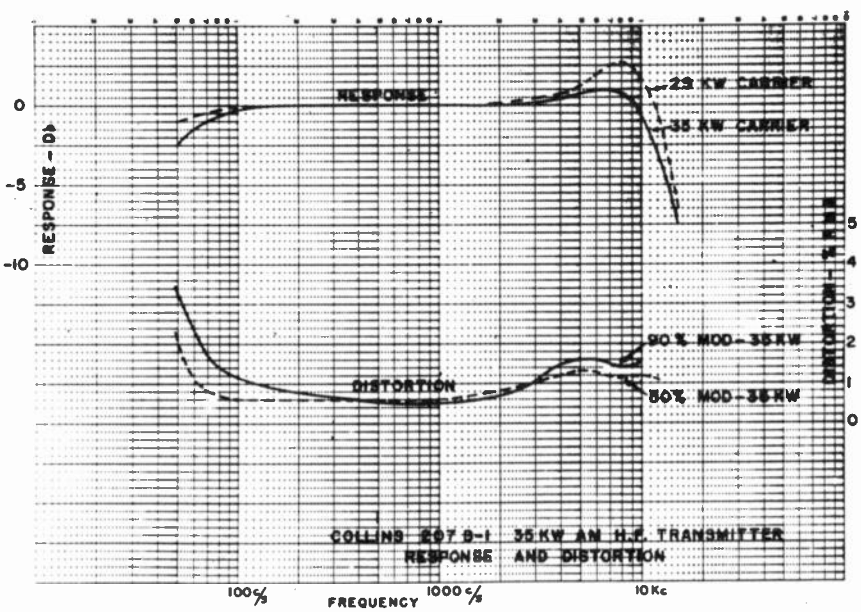

Fig. 8

Typical response and distortion data measured at the output of a 35 KW AM transmitter employing Class B_1 modulators in the direct coupled circuit described in this paper.

MICROPHONE SENSITIVITY CONVERSION CHART

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In the current technical literature, microphone sensitivities are often specified in different systems. This makes comparison rather difficult. The accompanying nomogram gives the relationship between three systems of ratings most commonly used. They are:

1. Open Circuit Voltage Response

The equation for open circuit voltage response is

$$S = 20 \log_{10} \frac{E}{p} \text{ db}$$

where E is the open circuit voltage, and p is the free-field sound pressure; expressed in db relative to 1 volt per microbar (1 dyne/cm^2).

2. Open Circuit Power Response

The equation for available power response is

$$S_p = S_v - 10 \log_{10} R + 44 \text{ db}$$

where R is equal to the nominal impedance at some specified frequency, usually 1000 cps. This equation gives the available power sensitivity in db relative to 1 milliwatt for a 10 microbar sound pressure.

3. RTMA Sensitivity Rating

The RTMA microphone rating G_M is defined as

$$G_M = S_v - 10 \log_{10} R_{MR} - 50 \text{ db}$$

where R_{MR} is defined as the RTMA "microphone rating impedance".

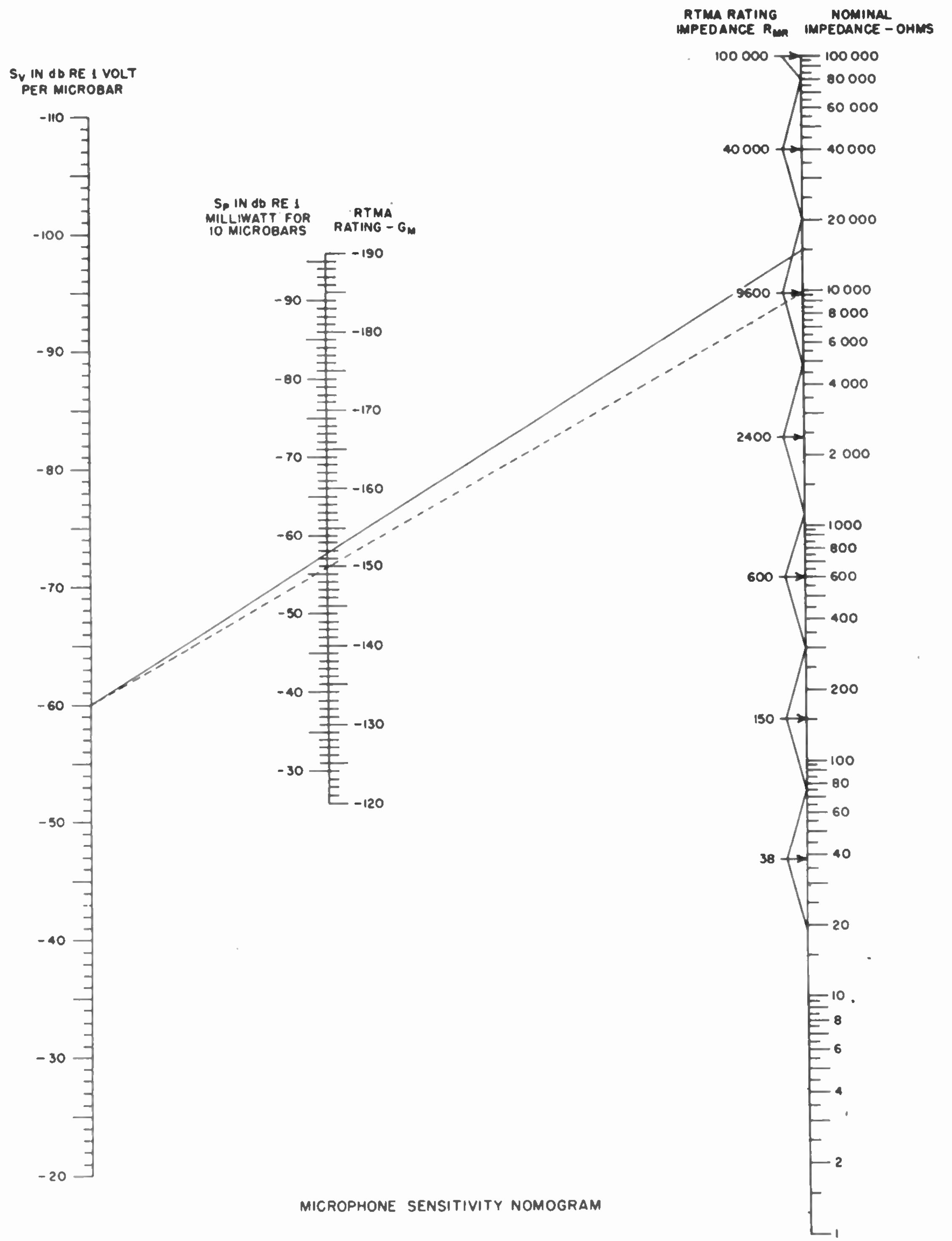
For a given value of nominal impedance, R_{MR} is indicated by arrows in the center of the range of impedance values covered by the triangular sections on the nomogram. For nominal impedances above 100,000 ohms, the rating impedance of 100,000 ohms is used. Microphone rating G_M is not defined for nominal impedances below 19 ohms. The value of G_M obtained from the above expression is essentially the available power in db relative to 1 milliwatt for 0.0002 dynes/cm^2.

The conversion of one rating to another is a simple but tedious problem of numerical computation and can be facilitated by the use of the accompanying nomographic chart. To illustrate the use of the nomogram, suppose we have a microphone with an open circuit voltage sensitivity of -60 db re 1 volt per microbar, and with a nominal impedance of 15,000 ohms. The solid line indicates that the power sensitivity is -58 db re 1 milliwatt for 10 microbars. To find G_M we first note that 15,000 ohms lies between 4,800 and 20,000 ohms. The microphone rating impedance is therefore 9,600 ohms and the dashed line indicates that G_M is -150 db. As a general rule R_{MR} for high impedance crystal microphones is 100,000 ohms.

The nomogram can also be used to determine the open circuit voltage sensitivity of microphones undergoing impedance transformation. First determine the power sensitivity for the original impedance and then pivot about this point until aligned with the new impedance. The new open circuit voltage may then be read on the left hand stem. This will be in error by the loss introduced by the transformer, which is usually small.

REFERENCES

RMA Standard SE-105, "Microphones for Sound Equipment".
ASA Standard Z24.1, "Acoustical Terminology".



MICROPHONE SENSITIVITY NOMOGRAM

Fig. 1

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