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IRE PROFESSIONAL GROUP ON AUDIO

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The Editor's Corner

THIRD PERSON PASSIVE

WHEN something has been done in the same way for a long time we may suppose: A) that it is the very best way, it meets a need fully, it solves a problem adequately; or B) that we are in a rut. Our engineering professors told us, "Reports are to be written in third person, passive voice." We studied out of textbooks and read scientific articles written in this manner. There is no question about it: third person passive is socially acceptable in scientific circles. Any other style may encounter lifted eyebrows.

Let us examine proposition A. The accepted form allows concentrated presentation of facts. The impersonal manner lends an air of scientific objectivity. There is minimum distraction from the subject matter. On the other hand, few scientists look forward to reading TRANSACTIONS with the same enthusiasm as the *Saturday Evening Post.* Many differences between these periodicals could be pointed out, but one thing is immediately evident. The *Post* does not require engineering style in its articles.

If a scientific paper is inherently difficult to understand, can't we render it more palatable by an interesting mode of presentation? Has it ever been tried? In earlier days scientific writing had yet not become standardized. Galileo wrote his "Mechanics" in the manner of Plato's dialogues, a question and answer conversation between teacher and pupil with a "heckler" thrown in to ask paradoxical questions that brought out unusual sidelights. Benjamin Franklin's accounts of his classic experiments on electricity, written in narrative style, make fascinating reading even today. More recently, a few government agencies became alarmed at the utter unintelligibility of their own questionnaires and forms. After careful study, they evolved instruction sheets written in second person singular which are relatively understandable, compared to their predecessors.

We suspect that some day a brave scientist who is also a gifted writer will depart from the standardized style when he publishes his findings. When others notice how much more interesting and instructive his paper becomes, they might follow in his footsteps. In the meantime, I notice that my monthly report is overdue. I guess I'll use third person passive.

MARVIN CAMRAS

PGA News

ANNOUNCEMENTS

WESCON PAPERS DEADLINE SET FOR MAY 1

Authors wishing to present papers at the 1959 Western Electronic Show and Convention technical sessions to be held in San Francisco, August 18–21, must submit them by May 1. Required are 100–200 word abstracts, together with complete texts or additional detailed summaries, which should be sent to the Chairman of the Technical Program: Dr. Karl R. Spangenberg, c/o WESCON, 60 West 41 Avenue, San Mateo, Calif.

This year there will be an important innovation. The IRE WESCON CONVENTION RECORD will be made available at the Convention. Convention authors will be expected to submit complete manuscripts by July 1, prepared for the RECORD in accordance with special instructions which will be sent at the time the paper is accepted.

Authors will be notified of acceptance or rejection by June 1.

Nominations

Dr. Harry F. Olsen, Chairman of the Nominating Committee, announced the following PGA slate for 1959–1960: For Chairman and Vice-Chairman: S. J. Begun, A. B. Bereskin, J. K. Hilliard, P. B. Williams. For members of the Administrative Committee: R. W. Benson, M. Copel, P. C. Goldmark, I. Kerney, J. R. MacDonald, W. B. Snow.

Acoustical Society Meeting

On May 14, 15, and 16, 1959, the Acoustical Society of America will meet at Chateau Laurier in Ottawa, Ontario, Canada. Invited papers, symposia, and contributed papers have been arranged for the technical program. Information may be obtained from the secretary, Wallace Waterfall, 335 E. 45 Street, New York 17, New York.

Stuttgart Congress

Fifteen invited papers are planned for the Third International Congress on Acoustics in Stuttgart, Germany, September 1 to September 8, 1959. Subjects covered are: Psychoacoustics, Ultrasonics, Room Acoustics, Speech, Noise, Propagation, Communication Theory, etc. Speakers will be from France, Germany, Sweden, Holland, Japan, England, Russia, Switzerland, and U.S.A. Contributed papers are invited, and should be offered to the secretary by May 1. The secretary of the Congress is Dr.-Ing. E. Zwicker, Stuttgart, Breitscheidstrasse 3.

PGVC Conference

The 10th National Conference of the Professional Group on Vehicular Communications is seeking material for its technical program on December 3 and 4, 1959, in St. Petersburg, Florida. If you are interested, please write to J. R. Neubauer, RCA, Building 1–4, Camden 2, N. J.

Coming Papers

The following papers are in process and should appear in subsequent issues of IRE TRANS. ON AUDIO:

Absolute Amplitudes and Spectra of Certain Musical Instruments and Orchestras, by L. J. Sivian, H. K. Dunn, and S. D. White, of Bell Telephone Laboratories. The classic paper by this name first appeared January, 1931, in the *J. Acoust. Soc. Am.*, and has served as a basis for design of audio components that are used in musical reproduction, especially pre-equalization and post-equalization systems. The data in the original paper was difficult to interpret directly because of the form in which peaks in the different frequency bands were presented. Dr. Dunn has overcome this objection by converting his graphs to read directly in db, so in its republished form the information will be more useful than ever.

Magnetic Tape Recording with Longitudinal or Transverse Oxide Orientation, by Richard F. Dubbe of Minnesota Mining and Mfg. Co. This paper compares the performance of magnetic tape when the recording field is in the same direction as the orientation of the oxide particles in the coating, and when it is perpendicular to the orientation in magnetic disc and video recording.

On the Response and Approximation of Gaussian Filters, by J. Klapper and C. M. Harris, of Columbia University. Multi-channel audio spectrum analyzers have been made in which each channel has a transmission characteristic which resembles a Gaussian probability curve. Advantage of such filters are described.

Study of a Two Channel Ceramic Transducer for Use in Stereo Phonograph Cartridges, by C. P. Germano of Clevite Electronic Components. Analysis and performance of experimental transducers are given.

A Frame-Grid Audio Pentode for Stereo Output, by J. L. McKain and R. E. Schwab of Sylvania Electric Products, Inc. Construction and operating characteristics of a new type of dual pentode are presented.

Wide Screen Stereo, by P. W. Klipsch of Klipsch and Associates. Experiments with a three speaker system are described.

CHAPTER NEWS

Albuquerque—Los Alamos

October 9, 1958—A panel discussion and demonstration were held of stereo disk vs tape.

December 11, 1958—"The Status of Stereo Multiplexing," Jack Hopperton, KHFM. Included a demonstration of stereophonic FM reception.

Chicago

January 9, 1959—"Performance Characteristics of FM Multiplex Stereo Transmission," Murray G. Crosby, Crosby Labs.

San Diego

The San Diego Chapter has had a successful 1958 season according to Kenneth R. O'Neal, 1958 Chairman. Mr. O'Neal reports, "We have attracted some outstanding speakers from neighboring areas, which I believe is one of the keys to a successful meeting and hence to increased attendance. Our average attendance at meetings increased from an average of 15 per meeting to an average of 35 per meeting.

The new 1959 Chairman, Hal R. Brokaw reports:

A most excellent talk was delivered by Mr. W. M. Turner, of the Arnoux Corporation, Santa Barbara, on January 15, 1959, describing the development of "Servosound."

Mr. Turner's discussion was delivered ad lib, since work on the project continued up to the time of his departure. The equipment was strictly laboratory, but the performance was outstanding. The speaker assembly was finished three days before the meeting. In spite of lack of preparation, however, he held the interest of all present far over the normal time.

In the past, there have been many approaches to the problem of stabilizing the performance of amplifiers and eliminating the distortion, by various applications of inverse feedback. These have been from one stage to over-all feedback, even including the voice coil in the feedback loop. "Servosound" has gone a step further, by including the paper of the cone in the feedback loop. Thereby, those harmonics caused by cone breakup are fed back and cancelled out of the acoustic output.

The surface of the cone is metallized and charged with a polarizing voltage. The electrical analog of the cone output is then coupled back to the input of a direct coupled transistor amplifier, which has a peak capability of 100 watts. The soundness of this approach, for which a patent is in process, is attested by the range and smoothness of response, and the excellent performance on sharp transients with no hangover. Three speakers were used, in one cabinet, of $4.00 quality; but the performance put more expensive systems to shame. These were only 8-inch size, in a small cabinet, but bass and high response were better than is usual; (down only 5 db at 35 cycles)—but particularly noticeable was the clean mid range. Speaker or other resonance was not perceptible, as is common in many of the latest systems with resonated bass.

Several recordings with difficult passages were played to demonstrate the cleanness and transient response of the system. Those familiar with the recordings agreed they heard new tonal expressions and harmonies that were masked before. To demonstrate, the feedback loop was opened to show the improved performance. There was no doubt that "Servosound" is an outstanding contribution to the art of reproduction of recorded sound.

Prior to the talk, the new officers for 1959 were introduced by Ken O'Neal, retiring chairman. Hal R. Brokaw is the new chairman and Lowman Tibbals, the vice chairman.

Previous meetings held in 1958 by the San Diego Chapter were:

June 3—"Description of Video Tape Recorders," Frank Gonsalves and Brian Trankle, Ampex.

November 20—"Performance Characteristics of FM Multiplex Stereo Transmission," Murray G. Crosby, Crosby Labs.

December 18—"Channel Separation in Stereo Reproduction," Odin Thaanum, Stromberg-Carolson.

San Francisco

Lawrence W. Johnson reported in San Francisco's December, 1958 issue of *The Grid:*

The October meeting of the Professional Group on Audio featured a talk by Ward Widener of Ampex Audio on the loudspeaker development program under way there. This was followed by a tour of the Sunnyvale plant.

Widener first presented a historical background, revealing the manner in which the Ampex Corporation found it desirable to set up a separate operation for consumer audio products. This need developed from Ampex participation in Cinemascope and Todd-AO theater sound-system design.

Early development work based on some Jim Lansing designs had taken place in a small Los Angeles installation. This work was transported in 1956 to a temporary Cupertino location, where Ampex Audio was starting up, having inherited the A series tape recorders and the 620 series of amplifier-speaker units. The present plant in Sunnyvale was occupied in early 1957.

The loudspeaker development work was aimed at producing the units needed for use in highly exacting deluxe home installations selling in the $500 to $2600 range. Widener discussed the areas of improvement in the LF and mid-range speakers which were evolved and presented especially interesting information on his work on hemispherical HF loudspeakers.

These are dome-shaped radiators without horns, backloaded, employing magnet structures similar to those used in the LF and mid-range units. Careful design and construction—including a fiberglass-filled back cavity—have led to the high performance result. A wide angular dispersion for the usually difficult high frequencies is a built-in feature of this design. Widener demonstrated with an a-b test a principle of growing importance, namely, that such a wide dispersion is indispensable in convincing reproduction of recorded sound.

The plant tour presented the group with an unparalleled opportunity to view high-quality audio equipment in various stages of construction. Probably the most interesting part of the tour was the chance to enter the Ampex Audio anechoic chamber, which has played a most implrtant part in the development work described earlier.

The meeting concluded—as it had begun—with an informal ques tion period and more ear-evaluation of the completed instruments."

In the same December, 1958 issue of *The Grid*, R. J. Newman and George Spelvin reviewed the November meeting:

According to Murray G. Crosby, principal speaker at the November joint session of the San Francisco Chapters of PGBTS and PGA, it seems probable that the Australians will be first to develop a satisfactory system of national standards for stereophonic FM broadcasting.

Approximately seventy gathered at the Shadows Restaurant at San Mateo to hear the details of the compatible system developed by the speaker who is president of the Crosby Laboratories, Inc., at Syosset, N. Y. Giving credit for the original work on two-channel FM transmission to E. H. Armstrong, Crosby pointed out that his own scheme provides refinements which permit the reception of a balanced audio program to monophonic receivers on the main channel—this being the chief significance of the term "compatible."

Before going into the details of the system, Crosby first addressed himself to a concept which has recently had considerable currency: namely, that only one wide-range channel is necessary to stereo because of the fact that the human ear loses directional interpretation above 1000 cps. Citing references and reading extracts from publications of both Dr. Harry F. Olson and Dr. Harvey Fletcher, Crosby pointed out that the stereo effect depends upon considerations of phase, amplitude, and sound quality and that the restricted range concept relates only to phase—ignoring the other two. He expressed the feeling, not only that a balanced frequency response in both channels was essential to good stereo but also that since the stereo art is in a formative stage, restrictive standards would undoubtedly place limitations on its fullest development.

Sum and Difference

Referring to a fuller discussion of the system delivered before the October 1957 AES National Convention and published in the Journal of the AES for April, 1958, Crosby presented the transmitter and receiver circuits shown here. (See Figs. 1–3.)

Channels A and B are mixed so that the sum $(A+B)$ is fed to the main FM transmitter channel while the difference $(A-B)$ is fed to the sub-carrier generator. Both are modulated in the main FM modulator. At the receiver, $(A+B)$ appears at the output of the receiver discriminator and $(A-B)$ at the output of a sub-carrier receiver. A phase inverter adds and subtracts these two signals and separates the original two channels.

Advantages include the facts that the monophonic listener receives a balanced signal which is the sum of the two input channels, and that there is a 6-db improvement in sub-carrier signal-to-noise ratio.

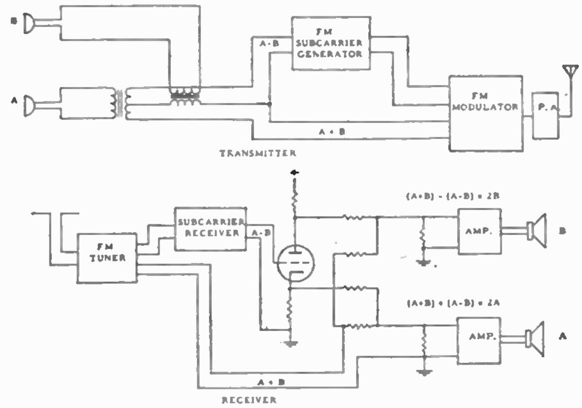

Fig. 1

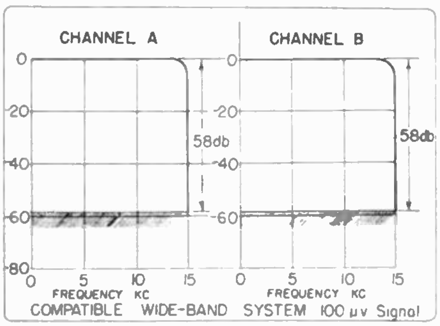

Fig. 2

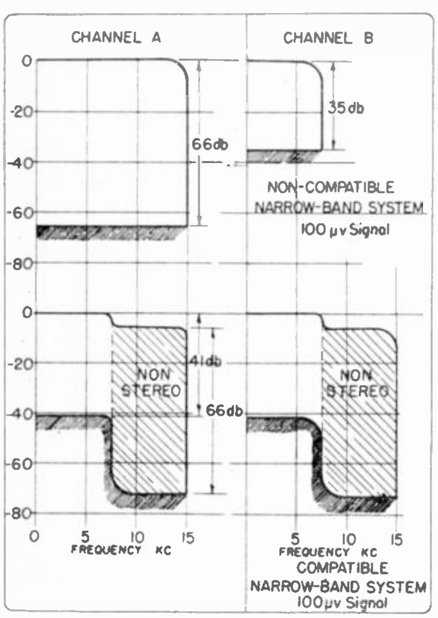

Fig. 3

Against Point-to-Point

Although Crosby expressed a personal opinion, seconded by some of the broadcast people in the audience, that FM broadcasters would be better off to keep away from common-carrier operation, nevertheless, he stated that the compatible system was capable of providing both stereo and non-stereo sub-channels, the characteristics of which are included in the accompanying plot where signal-to-noise ratio is the ordinate.

The chief problem, he felt, in non-stereo multiplexing would be channel crosstalk since the requirements for foreign program material are more severe than those for stereo.

Committee at Work

Discussing the outlook for FCC action on stereo, Crosby referred to the formation by Dr. W. R. G. Baker, within the Electronics Industry Association, of a national stereophonic radio committee analogous to the NTSC, also headed by Baker, who is now engineering director of the EIA. Other industry representatives presently on the committee include W. J. Morlock, General Electric Company; A. V. Loughren, Airborne Instruments Laboratory; C. Graydon Lloyd, GE; J. E. Young, RCA; V. M. Graham of EIA; Peter Goldmark of CBS; and Crosby himself.

Syracuse

William S. Bachman, Director of Engineering for Columbia Records, spoke on "Stereophonic Disc Recording" at a section meeting sponsored by the PGA at the Syracuse Museum of Fine Arts on December 2, 1958. Attendance was 150. Coffee and doughnuts were served as refreshments. The meeting announcement read as follows:

Did you think stereophonic disc recording is still a long way from your own living room?

You may be surprised to hear that the sales of stereophonic records by one of the largest record manufacturers are already exceeding his sales of 33⅓ rpm records! In just two short years since the introduction of the new industry—standardized 45-45 stereo disc system, a revolution has occurred in the home phonograph field. Today there is hardly a manufacturer of high-fidelity audio equipment who is not featuring a full line of stereophonic disc reproducing equipment.

The stereophonic disc recording has moved into an area more rapidly than has the stereophonic tape which has been available at least five years.

The technical story of the new stereophonic disc is extremely interesting. This story is told by our speaker, William S. Bachman of Columbia Records, Inc. Bachman is without question one of the foremost authorities in the world on all phases of disc recording. In addition, he has that rare knack of explaining and analyzing complicated technical phenomena in a way which makes them easy to understand.

His talk on December 2 will cover very broadly the whole field of stereo disc technology, from studio pickup techniques, including microphone arrays and microphone placement, to home amplifier and loudspeaker systems. The cutting of stereophonic records, both vertical-lateral and 45°-45°, and the correlation of cutting techniques with studio pickup procedures will also be discussed. A portion of his talk will deal with the production of stereophonic discs, including such important considerations as tracing distortion, choice of tip radius, tracking-angle distortion, and channel separation.

At the close of Mr. Bachman's talk, we will have a demonstration of stereophonic disc recording, representative of the latest techniques in both stereo discs and stereo disc reproducing equipment. The demonstration will be conducted by William W. Dean, Manager of Engineering, General Electric's High Fidelity Audio Components Section.

Twin Cities

A charter meeting of the newly organized Twin Cities Professional Group on Audio was scheduled for Wednesday, March 18, 1959, in the auditorium of Murphy Hall on the University of Minnesota campus. Mr. Joseph Maupin of Minneapolis-Honeywell was invited to speak on "High Power Linear Amplifiers Using Tetrode Transistors."

The PGA Chapter

This newsletter, now in its fourth consecutive year of publication, was issued January 26, 1959, to officers of PGA Chapters, by Chapters Committee Chairman J. Ross MacDonald. *The PGA Chapter* summarizes chapter activities, statistics, etc., and provides a clearing house for ideas that help improve chapter meetings. Dr. MacDonald welcomes any suggestions or chapter news that may be offered.

Audio Amplifier with Reduced Plate Dissipation*

ROBERT B. DOME†

Summary—**An audio frequency amplifier arrangement is described which results in decreased plate dissipation. The arrangement is suitable for either class A or class B amplifiers. The scheme employed is to feed the grid of the amplifier an auxiliary signal as well as the desired audio frequency signal. The auxiliary signal is at a super-audible frequency and is automatically adjusted so that its peaks in the positive direction never cause the peak plate current to exceed the maximum current peak of the desired signal, nor shall its amplitude be so high as to cause clipping of the auxiliary signal on its negative peaks; in other words, the added wave should not affect the *average* value of the current generated by the desired signal. The plate circuit has its regular low-frequency load, but it also has a load circuit tuned to the super-audible frequency. The super-audible output may be dissipated as heat in a resistor.**

Calculations have been made which show that for a given input and output at low frequencies, the maximum plate dissipation of a class A amplifier may be reduced to as low as 41 per cent of the maximum dissipation attained in conventional class A operation, and that the maximum plate dissipation of a class B amplifier may be reduced to as low as 50 per cent of the maximum dissipation attained in conventional class B operation.

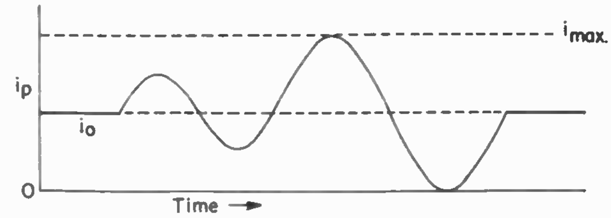

Fig. 1—Conventional class A amplifier plate current.

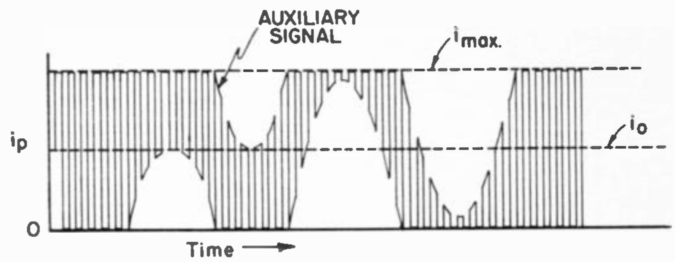

Fig. 2—Class A amplifier plate current with auxiliary signal added.

Introduction

There is an old adage which when paraphrased says that a way to keep a boy away from dissipation is to give him something to do whether or not the job assigned is useful. The same philosophy appears to hold in the case of audio frequency amplifiers. In a class A amplifier of conventional design, for example, in the absence of an input signal, all of the plate input power is dissipated on the plate. When an input signal is applied to the tube, useful output is obtained in an external circuit and the plate dissipation is reduced correspondingly. The scheme to be described, then, is one of providing some additional input to the tube on all occasions except at the instant when the tube is being driven to its peak possible plate current excursion. The additional input must not interfere with the low frequency desired signal, so a super-audible frequency may be chosen for the auxiliary signal.

Class A Amplifier

The class A amplifier is typified by its constant average plate current. Fig. 1 shows two cycles of audio frequency following a no signal region. A dc plate current meter will read a steady value of i_0 whether the audio is present or not since the average current is constant. Fig. 2 shows the same two cycles with an auxiliary signal added to give the amplifier something to do at all times except when it is called upon to deliver its maximum possible plate current i_{max}. It will be observed that the auxiliary signal has been adjusted so that it rides evenly in a positive and negative direction about the low frequency signal as an axis so as not to disturb the average low-frequency current values.

The plate circuit of the class A amplifier will have two load circuits, one for the auxiliary frequency and one for the low audio frequency as shown in Fig. 3. The low frequency is fed to a loudspeaker through a conventional transformer whose primary is by-passed for the auxiliary super-audible frequency, while the load for the super-audible frequency is shown as a resistor R bypassed for low frequencies by a small inductance L.

The wave shape of the auxiliary signal may be sinusoidal or it may be a square wave. The plate circuit for the auxiliary signal may be tuned for the sinusoidal wave or for the fundamental component of the square wave, or it may be very broad in response so as to resemble a video frequency amplifier and produce essentially square waves of auxiliary signal from the square wave grid excitation. The degree of plate dissipation reduction depends on the choice of these factors and in general improves with the use of square waves.

Fig. 4 shows the results of the calculation for the performance of a class A amplifier operated with an auxiliary super-audible square wave at the grid and with the plate circuit auxiliary signal load broad-banded to produce a square wave plate voltage at the super audible frequency. This curve has been calculated for a

$$k = \frac{E_{bb} - E_{bmin}}{E_{bb}}$$

* Manuscript received by the PGA, December 18, 1958.
† General Electric Co., Electronics Park, Syracuse, N. Y.

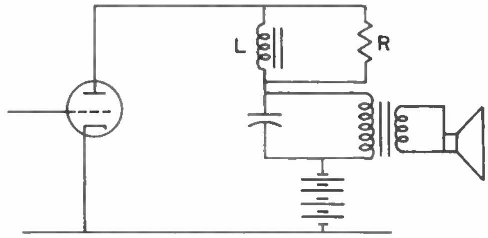

Fig. 3—Plate circuit of low plate dissipation class A amplifier.

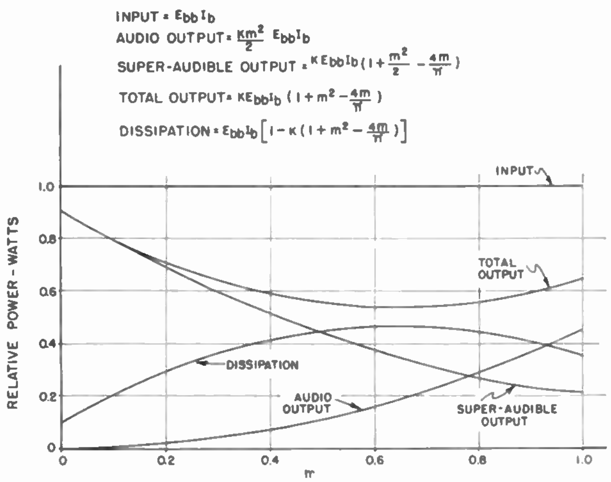

Fig. 4—Class A amplifier with auxiliary super-audible square wave. Grid signal and plate broad-banded, for $k=0.9$.

of 0.9, and shows the results as a function of the amplitude of the low frequency signal m, as it may be adjusted from no audio signal at $m=0$ to the maximum audio signal at $m=1$.

It will be observed that in conventional class A amplifiers, the maximum dissipation is equal to the full input and occurs at $m=0$; however, when the auxiliary signal is used, the maximum dissipation occurs at $m=0.636$, and is equal to 0.4635 times the input power.

A set of curves could be prepared showing the *maximum* dissipation as a function of k, the 0.4635 point being the point for $k=0.9$. Fig. 5 shows what may be expected of a class A amplifier operated in four different ways, namely:

1) As a conventional class A amplifier.
2) As an amplifier with auxiliary sine wave grid signal and plate tuned to the fundamental frequency of the auxiliary signal.
3) As an amplifier with auxiliary square wave grid signal and plate tuned to the fundamental frequency of the auxiliary signal.
4) As an amplifier with auxiliary square wave grid signal and plate circuit broadly tuned to accept all of the harmonics of the square wave.

The data in Fig. 5 are presented as maximum watts dissipation per watt of input as a function of a factor k, where k is defined as

$$k = \frac{E_b - E_{b\min}}{E_b}. \tag{1}$$

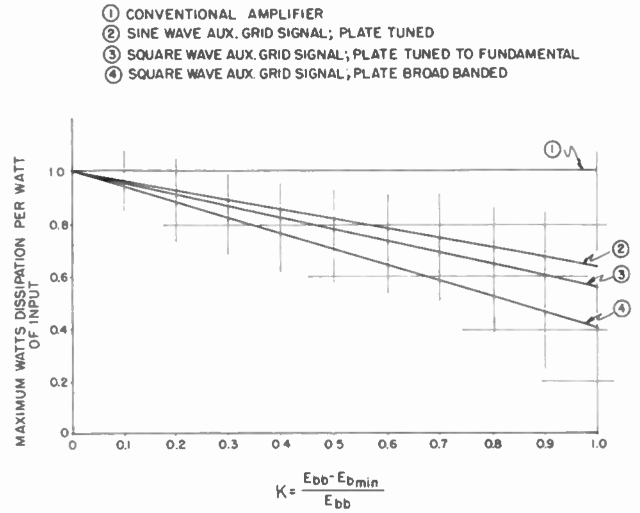

Fig. 5—Maximum dissipation vs k for class A amplifier operated under four different conditions. Input constant at 1.0 w.

It is seen that the plate dissipation can be reduced to as low as 40.5 per cent of the conventional and that even when $k=0.91$, the dissipation is only 46 per cent of conventional. Now a k of 0.91 is quite easily achieved in commercially available tubes; for example, the familiar 6V6GT attains a k of 0.91 when operated at its rated value of $E_b=315$ v, $E_{sc}=225$ v and $i_0=34$ ma, for the plate voltage $E_{b\min}$ is approximately 28 v at $E_{c1}=0$ and $I_b=68$ ma. Thus

$$k = \frac{315 - 28}{315} = 0.91. \tag{2}$$

The plate input is 10.7 w, and if this is called the rated dissipation, the maximum dissipation may be reduced to 46 per cent of this, or to $10.7 \times 0.46 = 4.93$ w.

The data for plotting the $k=1$ point of curve 2 of Fig. 5 were obtained as follows:

The output voltage at the desired low frequency may be expressed as

$$e_1 = mE \sin \mu t \tag{3}$$

where

m = amplitude factor lying between 0 and 1.0
$\mu = 2\pi f_a$
f_a = audio frequency.

The *added* signal e_2 thus has the following form for the interval corresponding to the positive cusp of the desired signal (*i.e.*, where μt varies from 0 to π)

$$e_2 = E(1 - m \sin \mu t) \cos \omega t \tag{4}$$

where

$\omega t = 2\pi f_r t$
f_r = super-audible frequency.

Now the root mean square value of the desired signal is

$$\underset{\text{rms}}{e_1} = \frac{mE}{\sqrt{2}} \tag{5}$$

while the root mean square value of the *added* signal is

$$e_{2\,\mathrm{rms}} = \frac{E}{\sqrt{2}}\sqrt{1+\frac{m^2}{2}-\frac{4m}{\pi}} \text{ (by integration).} \tag{6}$$

Assuming the load resistors to be the same for each frequency and equal to R, the useful audio power output is given by

$$W_{01} = \frac{E^2m^2}{2R} \tag{7}$$

while the power output for the added signal is

$$W_{02} = \frac{E^2}{2R}\left[1+\frac{m^2}{2}-\frac{4m}{\pi}\right]. \tag{8}$$

Thus the *total* output power is the sum of W_{01} and W_{02}, or is

$$W_{0t} = \frac{E^2}{R}\left[\frac{1}{2}+\frac{3m^2}{4}-\frac{2m}{\pi}\right]. \tag{9}$$

Now the total dc power input to the plate system is

$$W_{in} = \frac{E^2}{R}. \tag{10}$$

Thus the total dissipation is given by $W_{in} - W_{0t}$, or by

$$W_{diss} = \frac{E^2}{R}\left[\frac{1}{2}-\frac{3m^2}{4}+\frac{2m}{\pi}\right]. \tag{11}$$

The data for W_{diss} may now be tabulated for a range of values of m from $m=0$ to $m=1$, as given in Table I, assuming the input is 1 w. Maximum dissipation occurs at $m=0.4$ and is 0.635 w for an input of 1.0 w. This 0.635 value is therefore plotted as the point for curve 2 at $k=1$. Since in a class A amplifier the dissipation varies linearly with k, and the dissipation is 100 per cent of the input when $k=0$, a straight line may be drawn to complete curve 2.

TABLE I

W_{diss} VS m FOR CLASS A AMPLIFIER WITH AUXILIARY SIGNAL SINUSOIDAL AT GRID AND PLATE

m	W_{diss}
0	0.500
0.1	0.556
0.2	0.597
0.3	0.623
0.4	0.635
0.5	0.631
0.6	0.612
0.7	0.579
0.8	0.529
0.9	0.466
1.0	0.368

Eq. (11) may, therefore, be generalized to the form

$$W_{diss} = E_{bb}I_b\left[1-k\left(\frac{1}{2}+\frac{3m^2}{4}-\frac{2m}{\pi}\right)\right]. \tag{12}$$

The value for m which yields maximum dissipation may be found by differentiation of (12) with respect to m, setting the differential equal to zero, and solving for m. Thus

$$\frac{dW_{diss}}{dm} = 0 = \frac{6m}{4}-\frac{2}{\pi} \tag{13}$$

and

$$m_{\text{for } W_{diss\,max.}} = \frac{4}{3\pi} = 0.425. \tag{14}$$

In a similar way, the equation for plotting the dissipation as a function of m and k for the case of a square super-audible auxiliary grid wave but with the plate auxiliary circuit tuned to the fundamental component of the square wave to produce essentially a sinusoidal plate voltage at the auxiliary frequency, is

$$W_{diss} = E_{bb}I_b\left\{1-k\left[\frac{2}{\pi}+\left(\frac{1}{2}+\frac{1}{\pi}\right)m^2-\frac{8m}{\pi^2}\right]\right\}, \tag{15}$$

and by differentiation,

$$m_{\text{for } W_{diss\,max.}} = \frac{8}{\pi^2+2\pi} = 0.495. \tag{16}$$

Also, the equation for plotting the dissipation as a function of m and k for the case of a super-audible square auxiliary grid wave but with the auxiliary plate circuit broad-banded to cause the plate voltage wave form to be essentially square at the auxiliary frequency, is

$$W_{diss} = E_{bb}I_b\left[1-k\left(1+m^2-\frac{4m}{\pi}\right)\right], \tag{16}$$

and by differentiation,

$$m_{\text{for } W_{diss\,max.}} = \frac{2}{\pi} = 0.636. \tag{17}$$

CLASS B AMPLIFIER

The class B amplifier is typified by its zero signal plate current being essentially zero and by the positive and negative half cycles of audio frequency plate currents being taken care of by separate tubes. Fig. 6 shows how the plate voltages from one plate to the other appear when the auxiliary signal is added. Note that this signal is different from the class A case in that no auxiliary signal is present when the low frequency signal is zero.

Again, the wave shape of the auxiliary signal may be sinusoidal or square at the grids, and may be sinusoidal or square at the plates depending upon the input signal and upon how the plate circuit is tuned for the auxiliary signal. Fig. 7 shows how the maximum dissipation is reduced when the auxiliary signal is added to the system as a function of k and under the following conditions:

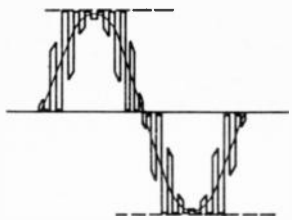

Fig. 6—Class B amplifier plate voltage with auxiliary signal added.

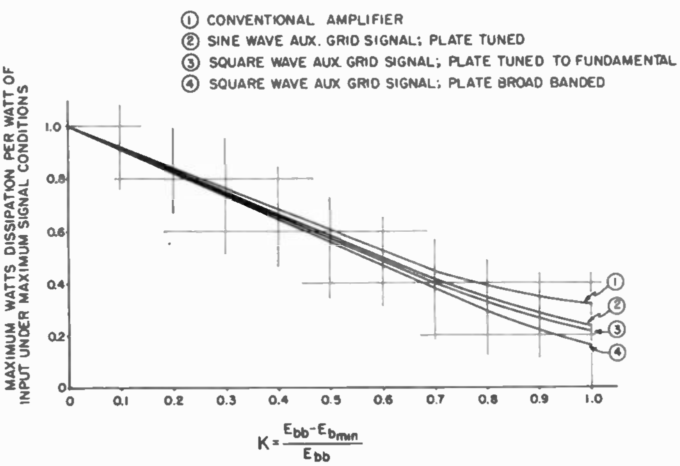

Fig. 7—Maximum dissipation vs k for class B amplifier operated under four different conditions. Input constant at 1.0 w under maximum signal conditions.

1) As a conventional class B amplifier.
2) As an amplifier with auxiliary sine wave grid signal and plate tuned to the fundamental frequency of the auxiliary signal.
3) As an amplifier with auxiliary square wave grid signal and plate tuned to the fundamental frequency of the auxiliary square wave.
4) As an amplifier with auxiliary square wave grid signal and plate broadly tuned to accept all of the harmonics of the square wave.

The data of Fig. 7 are presented as maximum dissipation vs k for a constant input power of 1 w under maximum signal conditions. It should be pointed out that maximum dissipation may occur at something less than maximum signal input.

It is seen that at $k=1$, the maximum dissipation may be cut in half, and that for $k=0.90$, a value usually attainable, the maximum dissipation may be cut to 63 per cent of the maximum dissipation under conventional operation.

Increased Output Power

The use of the auxiliary signal immediately shows up as a benefit in reducing the plate dissipation without altering in any respect the input power or the useful output power.

Since the dissipation has been reduced, a possibility also exists of increasing the input power and in therefore obtaining more useful output power while operating the tubes at no more than rated dissipation. This possibility

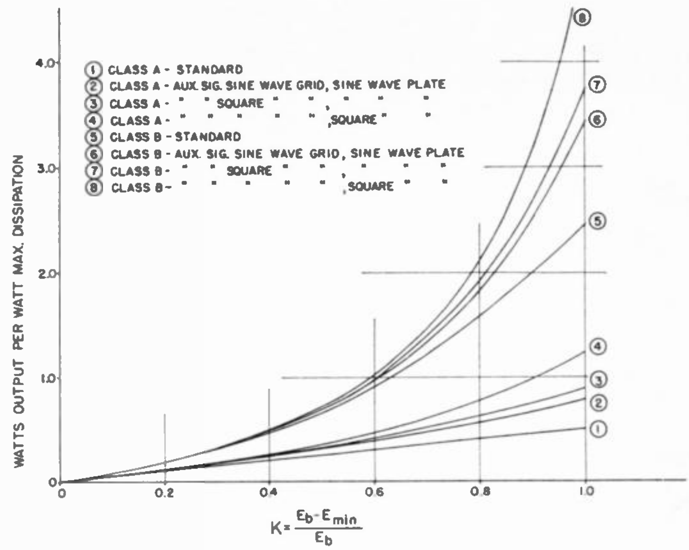

Fig. 8—Power output per watt of maximum rated plate dissipation for class A and class B amplifiers operating under conventional and various auxiliary super-audible signal conditions.

exists whenever the dc plate voltage or dc plate current or both may be increased within the operating limits of the tube. Fig. 8 has been prepared to show these possibilities. The data are displayed in the form of curves showing the maximum power output obtainable per watt of rated maximum plate dissipation as a function of k. For example, if the total rated dissipation of a pair of tubes operated class B is 250 w and $k=0.9$, in conventional class B operation, referring to curve 5 of Fig. 8, the power output obtainable is 2 times the dissipation rating, or $2\times 250=500$ w. Now if the auxiliary signal is added, using square waves at the grid and broad banding the plate for the auxiliary signal, the output available at $k=0.9$ from curve 8 is 3.18 times the dissipation rating; or $3.18\times 250=797$ w an increase of 297 w. The question arises, of course, as to whether the increased input is permissible, and this point must be investigated for the particular tube under study.

Consider, for example, a pair of type 810 triodes operated class B. Each tube is rated at 125 w plate dissipation under CCS operating conditions, with a maximum dc plate voltage of 2500 v, and with a maximum dc plate current of 250 ma. The peak allowable plate current is therefore $0.25\pi=0.785$ a, which may be attained with a $E_{b\min}$ of 160 v, so with $E_b=2500$ v, a k of 0.936 is available. Now referring to Fig. 8, curve 5, the watts output under conventional class B operation is

$$W_0 = 2.15 \times 250 = 537 \text{ w.} \tag{18}$$

The power input is equal to

$$W_{in} = \frac{4W_0}{k\pi} = \frac{4 \times 537}{0.936\pi} = 730 \text{ w.} \tag{19}$$

Since

$$\frac{E_b}{I_b} = \frac{2500}{0.25} = 10{,}000,$$

and since

$$2E_bI_b = 730,$$

solving the equations simultaneously yields an operating voltage and current of

$$E_b = 1910 \text{ v} \tag{20}$$

and

$$I_b = 191 \text{ ma per tube.} \tag{21}$$

Both of these values are less than the maximum permissible values so this operating condition is permissible.

Now suppose an auxiliary square wave signal is added and the plate circuit is broad banded for the auxiliary signal. Curve 8, Fig. 8, shows that the useful output may be 3.7 times the rated dissipation. Thus

$$W_{0s} = 3.7 \times 250 = 925 \text{ w.} \tag{22}$$

The input has been increased by the ratio of curve 1 to curve 4 of Fig. 7, or by the ratio of 0.34 to 0.20, or

$$W_{in} = 730\left(\frac{0.34}{0.20}\right) = 1241 \text{ w.} \tag{23}$$

The two equations,

$$\frac{E_b}{I_b} = 10000 \tag{24}$$

and

$$2E_bI_b = 1241, \tag{25}$$

may be solved simultaneously for E_b and I_b, yielding

$$E_b = 2491 \text{ v} \tag{26}$$

and

$$I_b = 249.1 \text{ ma per tube.} \tag{27}$$

Both of these values are permissible values, so this operating condition is satisfactory from a voltage and current standpoint.

Similar calculations may be performed for the square wave grid wave and plate circuit tuned to the fundamental where it will be found that under permissible limits,

$$E_b = 2260 \text{ v} \tag{28}$$

$$I_b = 226 \text{ ma per tube} \tag{29}$$

$$W_{in} = 1020 \text{ w} \tag{30}$$

$$W_{0s} = 750 \text{ w.} \tag{31}$$

This latter value of 750 w should be readily obtainable with practical circuits; the 925 w figure is theoretically obtainable if the super-audible plate circuit is truly broad banded. If, however, some of the higher harmonics of the super-audible frequency are attenuated, something less than 925 w would be obtained, approaching as a limit the 750 w figure when only the fundamental component is left.

Amplifier Input Circuit

Numerous circuits may be devised for producing the required grid excitation voltages, and so only one of many possibilities will be given. The circuit shown in Fig. 9 is suitable for the class A amplifier. Two diodes D_1 and D_2 have their cathodes connected together. A common resistor R_1 connects the cathodes to ground. Each anode is provided with a resistive impedance to ground and each anode is arranged to be biased into conduction by potentiometers connected to a B plus source. The input signals consisting of a super-audible wave and the desired audio signal are fed through a blocking condenser to one anode. The output signal is taken from the other anode through a suitable blocking condenser. The circuit is adjusted so that with a constant amplitude super-audible frequency applied, potentiometer P_1 is set to just clip the negative-going half-cycle of the auxiliary wave, while potentiometer P_2 is set to just clip the positive-going half-cycle of the auxiliary wave. When the audio frequency signal is applied, it will be found that the output will be of the desired form as shown in Fig. 2. Maximum permissible audio input signal will be that level which at its peaks just reduces the auxiliary signal to zero at those points. The amplitude of the auxiliary signal should be sufficient to cause the plate current of the amplifier tube being fed to go through the excursion of zero plate current to the plate current at zero grid bias voltage.

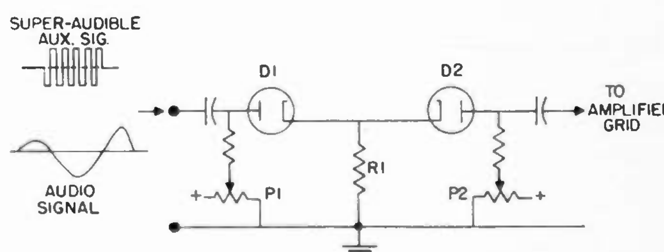

Fig. 9—Grid excitation circuit for class A amplifier with auxiliary signal.

Conclusion

Calculations and curves have been presented to show how it is possible to reduce the plate dissipation in audio frequency amplifiers by the introduction of an appropriate super-audible auxiliary signal. Illustrations were given for both class A and class B amplifiers, and it was shown that in the case of a push-pull class B amplifier employing a pair of type 810 tubes, the useful power output could be increased from the conventional figure of 537 w to 750 w, an increase of almost 40 per cent in output power, all within the dissipation ratings of the tubes.

Three-Channel Stereo Playback of Two Tracks Derived From Three Microphones*

PAUL W. KLIPSCH†

Summary—Playback of two-track stereo source material with a derived center channel offers accurate reproduction of the original stereo geometry, and requires very simple implementation. Essentially this two-track three-channel stereo depends on the principle that if two microphones are properly placed relative to each other and to the plane of the sound source, their combined output is that of a single microphone in the center, which output may be recovered by recombination of the two tracks. When a physical third microphone in the center is employed to feed the two tracks, its recovery from two tracks depends on relative polarity and amplitudes and in one recombination method the center microphone could be cancelled instead of being reproduced. An all-pass network may be used to shift the phase of one track so that on recombination the physical third microphone is always recovered, regardless of the manner in which it was mixed into the two tracks. The all-pass network produces 90° phase shift at only one frequency, but by choice of this frequency and the expectancy of additive polarity of original mixing, the center channel is recovered with excellent acuity, based on tests similar to those of Steinberg and Snow. Experimental recovery of a center output channel from a single center microphone feeding the two tracks, with flanking input signals zero, resulted in a center track output which was substantially indistinguishable from a normal monophonic reproduction.

DERIVATION of a third stereo channel from two sound tracks has been described.[1] The simplest and most effective implementation for this is shown in Fig. 1. The original paper showed the use of three amplifiers, but since there are essentially two sound tracks it seems superfluous to use more than two amplifiers. Lest the idea appear to be a sort of bootstrap lift, the basic philosophy should be remembered as expressed in the Summary. Also one may consider the situation analogous to the phantom circuit of wire telephony in which three conversions may take place over two physical pairs of wires. Localization of sound, or reproduction of geometry, as tested by having observers plot the apparent locations of sound, is nearly as accurate with the two-track three-channel "phantom" stereo as it was when observers plotted the original sounds without the intervening electroacoustic system.[2] Owing to the wider speaker spacing, the playback geometry more nearly resembles that of three channels derived from three tracks.[3]

Mixing a third (middle) microphone into two stereo sound tracks has been a concept for a long time, and its practice is liable to become more widespread. Steinberg and Snow[3] mentioned it in 1933. Frazier[4] stated he had experimented with it. Some Mercury stereo tapes contain a reference on the box cover to the effect that a center microphone was mixed into the two tracks.

The simplest way to achieve the "mix" is to use a stereo tape machine with dual inputs, microphone, and bridging; the two flanking microphones of some high-level type like the Stephens C2-OD4 are inserted in the bridging input, and one microphone in the center is fed to both microphone inputs, the mixing being accomplished by the respective input gain controls. The middle microphone may be fed preponderantly to either track, or balanced so each track gets the same amount.

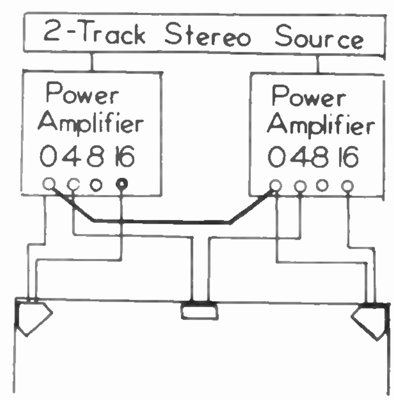

Fig. 1—Basic means for deriving a center channel from 2 sound tracks. (See Klipsch.[1]) Note that if the flanking speakers are connected to 16-ohm taps and the center speaker is between 4-ohm taps, the center speaker receives $\frac{1}{2}(A-B)$ where A and B are the voltages in the independent tracks and the flanking voice coils.

In the case of unequal amounts, one need not worry about what the center speaker does. But if equal amounts are mixed, the center speaker is either additive or substractive, and in the latter case the center speaker puts out none of the center microphone input.

This (perhaps rare) event can be obviated by shifting either output 90° in the center of the spectrum of interest. In this case, the center speaker receives the two signals as $a \pm b < 90°$ or 1.4 times as much voltage, (with respect to what each track contributes), twice as much power or 3 db up—enough to insure the "focusing" effect of the center unit in spite of the fact that

* Manuscript received by the PGA, July 21, 1958; revised manuscript received, January 16, 1959.

† Klipsch and Assocs., Hope, Ark.

[1] P. W. Klipsch, "Stereophonic sound with two tracks, three channels by means of a phantom circuit (2PH3)," *J. Audio Eng. Soc.*, vol. 6, pp. 118–123; April, 1958.

[2] P. W. Klipsch, "Wide Screen Stereo," paper and demonstration performed at Bell Telephone Labs., November 21, 24, 1958. Submitted for publication.

[3] J. C. Steinberg and W. B. Snow, "Symposium on auditory perspective—physical factors," *Trans. AIEE*, vol. 53, no. 1, pp. 12–17; January, 1934.

[4] J. Frazier, personal communication.

each flanking speaker is carrying a similar signal component. Fig. 2 shows a suitable network for feeding the center channel.

Actually, the center speaker being tapped down on the output transformers may be zero db or 3 db down instead of 3 db up; the actual taps used are determined by experiment and will differ with environment. Arnold Auditorium at the Murray Hill Bell Telephone Laboratory required the lowest transformer taps and a 3-db pad loss to maintain balance; the 463 West Street Laboratory Auditorium required about a 6-db higher output for the center speaker to keep central area events "in focus."

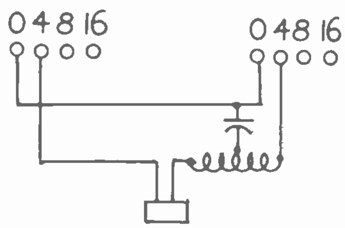

Fig. 2—All-pass phase-shifting network to make the 2 sources 90° out of phase with each other at a selected mid-frequency.

Ordinarily, the center channel receives a -6-db signal from each track so the resultant is -3 db. With equal speaker efficiencies it would be expected that a monophone signal from one track would be heard between the center and flanking speaker and at two thirds the distance from the center speaker to the flanking speaker. When testing the geometry of reproduction with two-track stereo and three-channel playback, the proper placement of two microphones enables the entire space between flanking speakers to be filled with sound and central area events satisfactorily focused. The center microphone would be employed to heighten the center effects, for example, a soloist.

Theory

The phase shifting of one of the two tracks assures that the center microphone, if used, will be reproduced through a center speaker regardless of its polarity in the two tracks at the time of recording.

Needless to say, the playback of a two-track recording made with two microphones will not be affected by the additional phase shift of less than π radians. Since in the middle spectrum—say 1700 cycles, a wavelength is 8 inches, and a microphone spacing of 100–200 inches would represent a relative phase shift of 12–25 wavelengths, or 4,500° or 9,000°—another 180 more or less would be negligible.

The circuit of Fig. 2 is essentially the simplest form of an all-pass network in which the propagation constant $\Gamma = a + j\beta = 2\tanh^{-1}\sqrt{Z_1/_2Z}$ where Z_1 and Z_2 are the impedances of a lattice derived from the network shown. Since the phase shift β is small at low frequencies and approaches 180° at very high frequencies and since there is only one at which the phase shift is 90°, this particular frequency should be properly chosen. At 1700 cycles, there are several octaves within which the signal addition is satisfactory. Actually the center speaker needs mainly to reproduce above 300 cycles. Lower frequencies are filled in from the flanking corner full-range speakers being fed full range signals.

Experiment

A tape was cut with a single microphone mixed equally into the two tracks on a stereo tape machine with accurate head alignment, and the center channel null was quite sharp when the two playback tracks were balanced. The same tape was played on another machine in which the two heads were not precisely "stacked." (They were in azimuth alignment so response was good to 15 kc, but the gaps were not precisely colinear, *i.e.*, there was a small "delay.") The result was not a sharp null, but the chest tones were thinned out.

Using the first machine with the better null, the speech was restored to full intelligibility and tonality by the network.

With the outside channels going, the focus was retained in the center channel, whereas without the phase shifting network, the center channel exhibited its null.

Use of this network is no guarantee that a microphone phasing will not someday be employed on which the network will fail to work. But any normal center microphone mixing circuit which places a signal S on track A and either plus or minus S on track B will play back through the center channel with enough power to be retained in focus.

This idea is expressed in anticipation of use of three or more channels to feed two tracks from which three channels should always be recoverable.

The use of two-track machines to make the master tape is expected to continue. Some portable equipment using two tracks is quite adequate for making superb masters. But where feasible, three or more tracks are liable to be used and the tracks mixed later to form the two tracks. This enables "second guessing" on mixing levels. Regardless of how the recording was made, with two or three mikes and/or two or three channels, the middle channel is always recoverable. It is hoped that the third or middle microphone will not be overdone to the extent of bringing the soloist to an undue focus of presence.

All the tapes so far obtained, *i.e.*, Mercury, RCA-Victor, Stereophony, Livingston, Audio Fidelity, Klipschtapes and others have not needed the phase correcting circuit to recover the center channel, all having been "focused" by the introduction of the third channel, and not having been altered audibly by the phasing network. Thus the Mercury tape which employed the third microphone was recorded in such a way that the third channel stayed in focus whether played as a sum, difference or 90° phase shift.[5]

[5] Coming recently to attention here are cases of stereo records made with a center microphone in which the center speaker cancels a soloist when fed a difference signal, a particular case being Capital SWAL 9032. Here a phase shift or polarity reversal is definitely needed to reproduce the center channel. (This new data was received by the PGA, May 4, 1959.)

It is anticipated that not all future tapes will be so recorded; indeed, the easy way to apply the third microphone is simply by sum or difference in which difference or sum playback would cancel it out. The 90° network would be the universal way to play it back; if the recordist then tries to satisfy both the sum and difference playbacks and makes the 90° playback ineffective, all he has to do is reverse the playback network and shift the other track.

Unilateralized Transistor Amplifier*

L. M. VALLESE†

***Summary*—A transistor audio amplifier with input impedance independent of the load and adjustable continuously up to a value of the order of $r_m/2$ is described. The amplifier consists of a unilateralized hook common collector configuration and has large power gain, very low output impedance. Its noise figure and dependence upon temperature, frequency, bias, and load are discussed.**

Introduction

THE design of a transistor amplifier with high input impedance, low noise figure, and large power gain is one of the difficult problems of the present status of the art. Various circuits are available which provide large input impedance: for example, the common collector amplifier with load resistance R_L has input impedance

$$R_{\text{in}} = \frac{r_c R_L (1 + \alpha_{cb})}{r_c + R_L(1 + \alpha_{cb})} \cdot \tag{1}$$

On the other hand, in the case of the compound transistor[1,2] [Fig. 1(a)], if, for simplification of formulas, the parameters of the two transistors are considered to be identical, the input impedance is

$$R_{\text{in}} = \frac{r_c R_L (1 + \alpha_{cb})^2}{r_c + R_L(2 + \alpha_{cb} + \alpha_{cb}^2)} \cdot \tag{2}$$

The assumption of identical parameters is introduced for convenience but is not valid when the bias points of the two transistors are different. The "bootstrap" common collector configuration[3] [Fig. 1(b)], with the

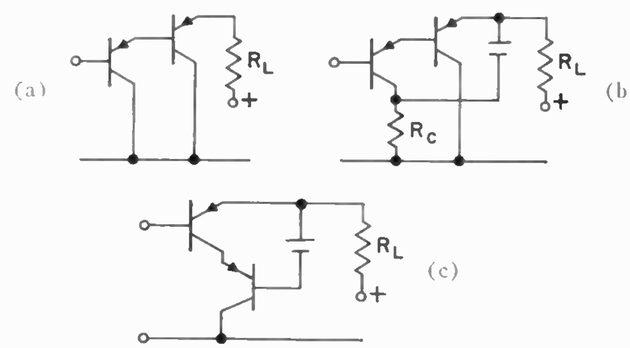

Fig. 1—Transistor amplifiers with high input impedance: a) compound connection, b) bootstrap connection, c) cascode connection.

same assumption on the transistor parameters, has input impedance

$$R_{\text{in}} \cong \frac{r_c R_L'}{r_c + R_L'} (1 + \alpha_{cb} + \alpha_{cb}^2) \tag{3}$$

where $R_L' = R_c R_L/(R_c + R_L)$. Finally the "cascode" common collector configuration[3] [Fig. 1(c)] has input impedance

$$R_{\text{in}} \cong \frac{r_c R_L}{r_c + R_L} (1 + \alpha_{cb}). \tag{4}$$

Examination of these expressions shows that the (1) and (2) are always less than r_c, while (3) and (4) may be larger than r_c provided that R_L is sufficiently large. One common characteristic of these configurations is that the input impedance is strongly dependent upon the load impedance variations and that a large value of R_L is required to obtain a rather large value of R_{in}.

When a large constant value of R_L is not available, the previous configurations cannot be used conveniently. In such a case, using unilateralization, it is possible to obtain a circuit with input impedance independent of the load. The amplifier illustrated in the following

* Manuscript received by the PGA, November 25, 1957; revised manuscript received, January 2, 1959.

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[1] S. Darlington, Patent No. 2,663,806, Bell Tel. Labs., Murray Hill, N. J.

[2] A. E. Bachman, "Noise figure of the Darlington compound connection for transistors," IRE Trans. on Circuit Theory, vol. CT-5, pp. 145–147; June, 1958.

[3] R. A. Stampfl and R. A. Hanel, "Transistor amplifier with extremely high input impedance," *Proc. NEC*, vol. 11, pp. 67–73; October, 1955.

is a three terminal, transformerless unilateral common collector circuit[4] which has high power gain and input impedance adjustable up to values of the order of $r_m/2$. Its noise figure is minimum for a source resistance of the order of 1000 Ω.

Theory

Consider the amplifier configuration shown in Fig. 2. If the parameters of the two transistors are assumed to be identical for simplification of formulas, it is found that, when the resistance R_e has the value

$$R_e \cong r_m - r_c/2, \tag{5}$$

the amplifier is unilateralized, and its input impedance is

$$R_{\text{in}} \cong r_m/2. \tag{6}$$

The high value of R_e given by (5) is impractical for usual applications; however, if a resistor R_{be} is added between base and ground as shown in Fig. 3, it is found that unilateralization is realized when the following relation is satisfied;

$$\frac{R_e R_{be}}{R_e + R_{be}} \cong \frac{r_m}{2}\left(1 + \frac{R_{be}}{R_e + R_{be}}\right) - \frac{r_c}{2}. \tag{7}$$

Correspondingly, the input impedance is still $R_{\text{in}} = r_m/2$. Eq. (7) may be satisfied with relatively small values of R_e and R_{be}. In fact, solving (7) for R_{be} and letting $r_d = r_c - r_m$, one finds

$$R_{be} = \frac{r_d R_e}{r_m - r_d - 2R_e}. \tag{8}$$

If $R_e \ll r_d$, this equation reduces to

$$R_{be} \cong R_e/(\alpha_{cb} - 1). \tag{9}$$

The output resistance of the unilateralized configuration is

$$R_{\text{out}} = r_e + r_b/2, \tag{10}$$

and the maximum available power gain is

$$A_{P\max} = r_m/(8r_e + 4r_b). \tag{11}$$

In Fig. 3, a resistance R_{bc} shunting the two collectors is also shown. When this resistance is finite, the condition of (7) is modified as follows;

$$\frac{R_e R_{be}}{R_e + R_{be}} = \left(r_m\left(1 + \frac{R_{be}}{R_e + R_{be}}\right) - r_c\right)\frac{R_{bc}}{r_c + 2R_{bc}}, \tag{12}$$

and the corresponding input impedance becomes

$$R_{\text{in}} \cong \frac{r_m R_{bc}}{r_c + 2R_{bc}}. \tag{13}$$

Hence by means of R_{bc}, it is possible to control continuously the value of the input impedance.

[4] L. M. Vallese, "Unilateralized common collector transistor amplifier," *Proc. NEC*, vol. 13, pp. 55–68; October, 1957.

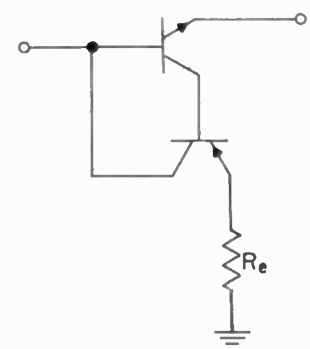

Fig. 2—Unilateral common collector amplifier.

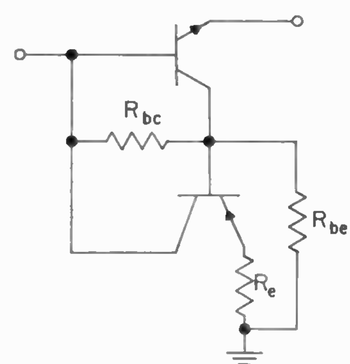

Fig. 3—Unilateral common collector amplifier with low values of the unilateralizing resistors.

Experimental Verifications

For the purpose of experimental investigation, the circuit of Fig. 4 was built. In this case the amplifier was fed with a current source realized by means of transistor T3; in practice, a similar result may be obtained using a large resistance in place of transistor T3. A capacitance C_e was shunted across R_e to extend the frequency range of unilateralization. Finally resistance R_e' was included for test purposes, but should be zero in the final design circuit.

The condition of unilateralization was determined applying a 2 mv, 1 kc source at the output terminals and measuring the reverse voltage gain. In Fig. 5, the values of A_v^- (reverse voltage gain), of R_{in} and of R_{out} are plotted as functions of R_{be}. Taking R_e respectively 0, 3 $k\Omega$, 30 $k\Omega$, the corresponding unilateralizing values of R_{be} were found to be respectively 25 Ω, 75 Ω, 1000 Ω. The values of the input and of the output impedances were found to be independent of the particular combination R_e, R_{be} used for unilateralization. Since a resistance $R_{bc} = 50\ k\Omega$ was used, the input impedance turned out to be of the order of 30 $k\Omega$.

The dependence of the input impedance upon the load resistance is illustrated in Fig. 6 for the case of $R_e = 3000\ \Omega$, taking as parameter R_{be}. It is seen that, when unilateralization is realized, R_{in} remains constant as R_L is varied, but that a minimum critical value of R_L exists, under which the power gain increases to such value that the amplifier becomes unstable, unless

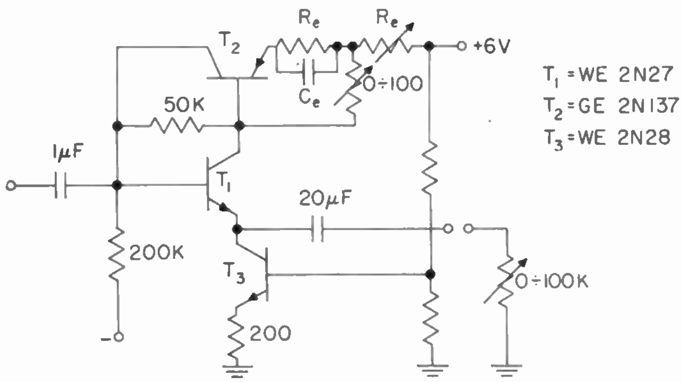

Fig. 4—Circuit used for experimental verifications.

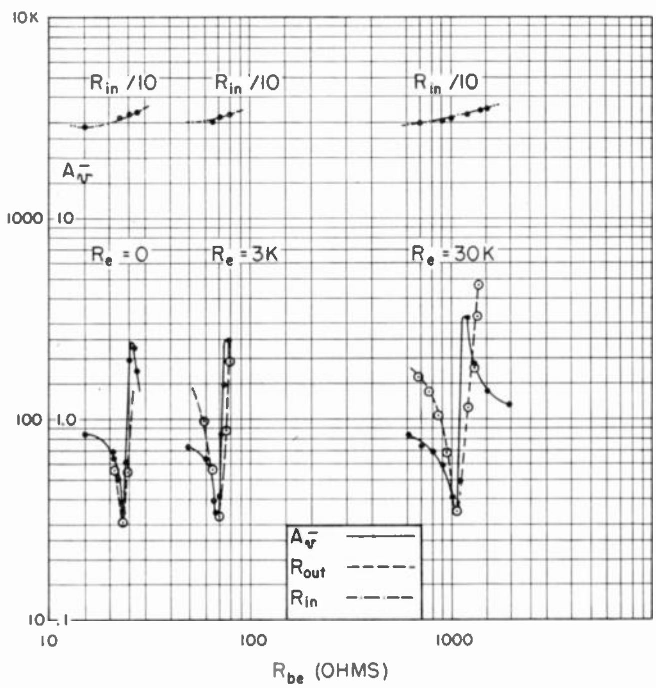

Fig. 5—Values of A_v^-, R_{in}, and R_{out} at 1000 c for different R_e, R_{be} values.

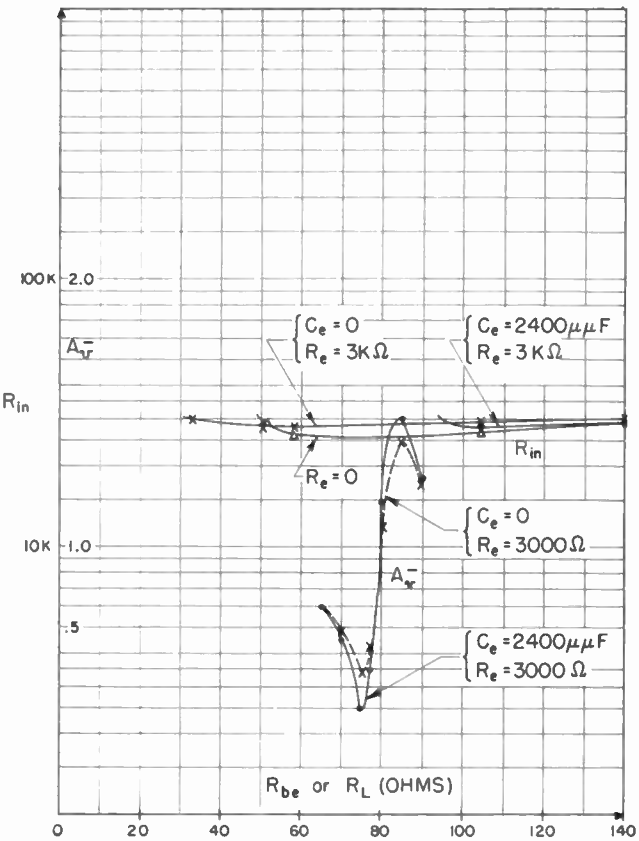

Fig. 6—Values of R_{in} vs R_L and of A_v^- vs R_{be}.

proper shielding of the output circuit is provided. In the same Fig. 6, the reverse voltage gain is plotted as function of R_{be}.

The advantage of the use of a constant current dc supply is illustrated in Fig. 7, where the reverse voltage gain and the input resistance are plotted as functions of the input bias voltage. The results are compared with those obtained when the transistor T3 of Fig. 4 is replaced with a 2000 Ω resistance. It is seen that, when constant current source is used, the dynamic range of the amplifier is approximately doubled.

In Fig. 8 the dependence of the amplifier characteristics upon frequency is investigated. It is seen that, using a capacitor of 2400 μμF in parallel with R_e, a frequency range of about 10 kc is obtained. Of course, the actual bandwidth obtainable depends on the cutoff frequencies of the transistors used and can be extended to video frequencies without difficulties.

Finally the variations of R_{in}, R_{out}, A_v^- as functions of R_{be} are illustrated in Fig. 9.

The temperature behavior of the unilateralized amplifier has been investigated placing the entire amplifier in an appropriate oven. For the case of $R_e = 3000$ ohms, $R_{be} \cong 75$ ohms (at room temperature) it has been found that, in order that unilateralization be maintained, the value of R_{be} should decrease as the temperature is increased, with a constant rate of about 2 per cent per degree C in the temperature range from 30°C to 50°C. Such a variation can be realized by means of thermistors.[5] With Silicon transistors greater thermal stability is obtained.

The noise figure of the amplifier may be computed by standard procedure, considering emitter and collector noise generators and assuming an appropriate correlation coefficient for the generators of the same transistor. If, for simplification purposes, the small resistances r_e and r_b are neglected, and if the unilateralized condition is considered, the equivalent circuits of the amplifiers of Fig. 2 and of Fig. 3 are found to be identical (Fig. 10). Indicating with $\bar{v}_{nE}$ the noise voltage of the unilateralizing resistance (*i.e.* R_e in the case of the circuit of Fig. 2 and $R_eR_{be}/(R_e+R_{be})$ in the case of the circuit of Fig. 3) the following expression is found for the noise figure:

[5] L. M. Vallese, "Temperature stabilization of transistor amplifiers," *Commun. and Electronics*, no. 26, p. 379; 1956.

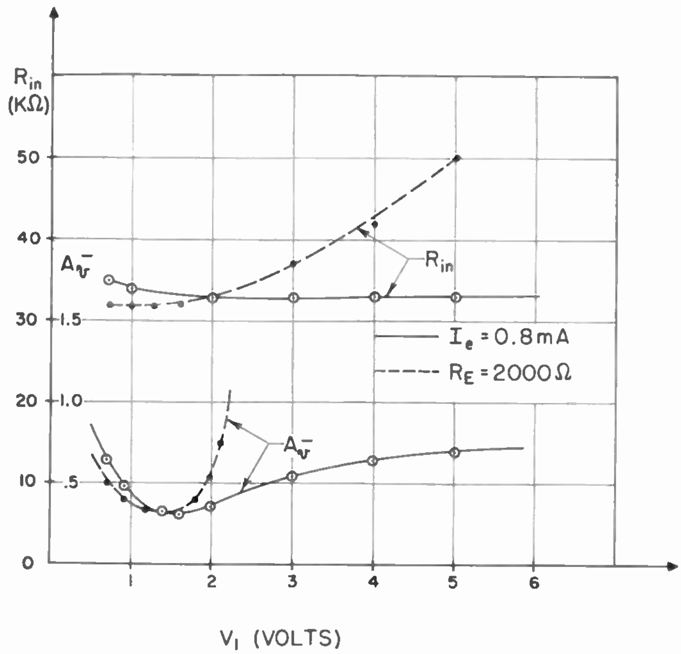

Fig. 7—Values of A_v^- and of R_{in} vs input bias at 1000 cps.

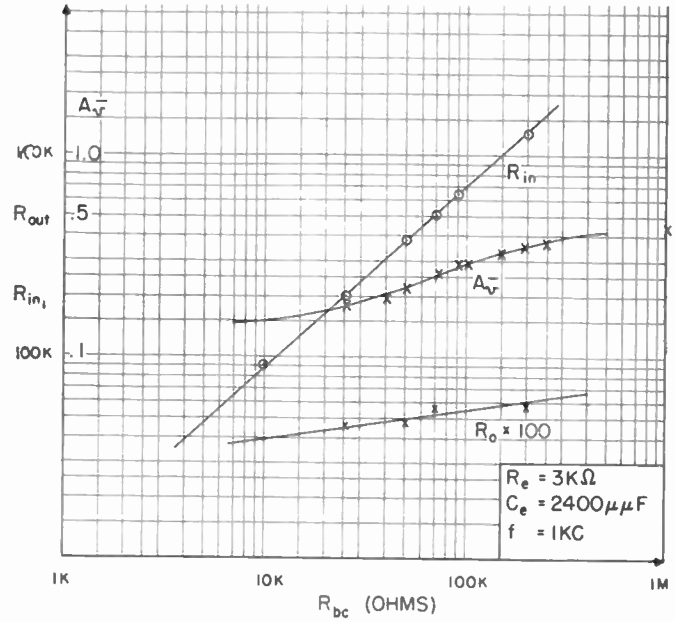

Fig. 9—Dependence of unilateralization upon R_{be} at 1000 c.

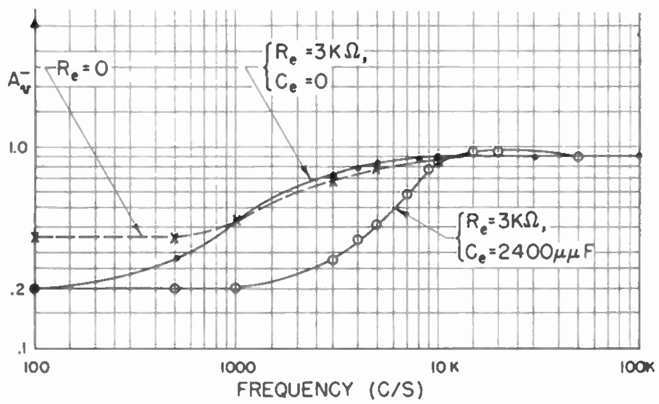

Fig. 8—Dependence of unilateralization upon frequency.

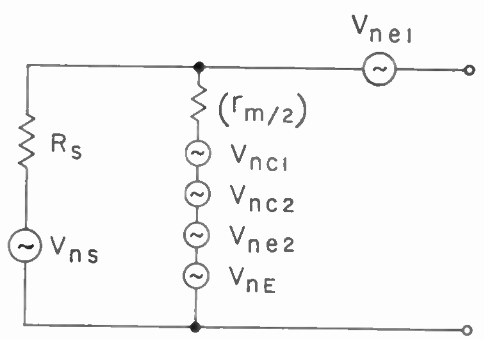

Fig. 10—Equivalent circuit of the unilateralized common collector amplifier with noise generators.

$$F \cong 1 + \frac{\bar{v}_{ne1}^2\left(1 + \frac{2R_s}{r_m}\right)^2 + \left[\bar{v}_{ne2}^2 + \bar{v}_{nE}^2 + 4(\bar{v}_{nc1}^2 + \bar{v}_{nc2}^2)\right]\frac{4R_s^2}{r_m^2}}{4kTBR_s}. \tag{14}$$

In this relation the correlation coefficients have been neglected. The corresponding optimum value of source resistance is found to be approximately,

$$R_{s\,\mathrm{opt}} \cong \frac{r_m}{2\left[1 + \frac{\bar{v}_{ne2}^2 + \bar{v}_{nE}^2 + 4(\bar{v}_{nc1}^2 + \bar{v}_{nc2}^2)}{\bar{v}_{ne1}^2}\right]^{1/2}}. \tag{15}$$

From these equations it appears that the noise figure is lower for the configuration of Fig. 3, because the corresponding $\bar{v}_{nE}$ is smaller, and that it decreases as r_m increases.

Conclusion

The experimental verifications described confirm the usefulness of the unilateralized common collector amplifier, although they are not indicative of optimum performance obtainable. It is expected that greatly improved performance is obtained with low noise high frequency transistors. The circuit is useful because it is transformerless, and has common ground terminal between input and output. Its input impedance may be adjusted continuously up to a value of the order of $r_m/2$ and is independent of the load resistance. Hence, the amplifier appears particularly convenient for cases in which the load resistance is not constant and the input impedance is to be reasonably high.

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