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IRE PROFESSIONAL GROUP ON AUDIO

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IRE TRANSACTIONS® ON AUDIO

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The Editor's Corner

INVENTIONS

"Did you notice Sam lately," asked Bob as he set his food tray on the table. "He acts kind of lovelorn."

"He's got an invention," answered Bill. "Every good engineer makes one sooner or later. Sam's got it bad. Wants to quit his job and start a company."

"I tried to find out what it's about," Bill continued, "but he won't tell until he files a patent. Says it will revolutionize the dry-cleaning industry."

"Dry-cleaning industry." Bob almost dropped his hamburger. "Sam is a loudspeaker designer. How does he get to dry-cleaning?"

"Who knows," said Mike. "Maybe he does use loudspeakers. He could adjust the frequency until it shook the dirt right out of your clothes while you were wearing them. Stereophonically, yet!"

Everyone ignored this explanation.

Bill philosophized as he sipped his soup, "When the grand invention gets you, it's like being in love. Everyone reacts differently. Some day-dream of yachts and empires. Some are secretive; others use it as a conversation piece. Some fall into and out of different inventions every week, while others have one that lasts a lifetime. Some are disillusioned after a sad experience, and become invention-haters."

"If love is blind, so are inventors," observed Bob, finishing his beefstew. "At least I can't see what most of them see in their inventions."

"I hope it's a simple one," said Mike. "You know our engineer's motto: 'The easy things take forever. The difficult take a bit longer.' That's not the way the public relations department says it, but we know better."

"Speaking of simple inventions," said Bill as he scooped his ice cream sundae, "I know someone whose uncle put the bend in the hairpin. Made millions out of it. This guy had his own hairpin factory. Efficient operation. Come to think of it, he probably would have done just about as well even without the invention."

"It's a matter of timing," Bob interrupted. "There was a generation of hair styles which didn't use hairpins. No matter how ingenious the idea, the patent would have expired without paying for itself."

"That's often the case." Mike lit his pipe. "I seem to recall statistics—to the effect that of hundreds of good ideas only a few per cent are considered important enough to go to the trouble of applying for a patent. Of all the patents applied for, only a fraction are allowed. And of the ones that finally issue, maybe ten per cent barely pay for themselves, and less than one per cent make a good profit. It's really discouraging."

"The trouble with inventions," said Bill, "is that most of the simple useful things have already been thought of; sometimes exactly as you thought of them, sometimes in a different field. You apply for a patent on a fantastically different can opener, and the patent examiner shows you that the same principle was used in 1908 in Sweden on a mining machine. All you have left are features that you wouldn't have bothered with, if you had known in the first place."

"There is a better chance of obtaining a complicated patent. But in such cases there may be a dozen other ways to accomplish the same result and no one cares to use your patent."

"Then, as I understand it," said Mike as he drew deeply on his pipe, "you could classify inventions into two categories."

"Class I: Inventions of ingenuity, simplicity, and wide basic importance, but on which you can't obtain a good patent because it resembles something which has been done before."

"Class II: Inventions which give you strong iron-clad patent protection on something that nobody in his right mind would want to manufacture."

"You are a blankety-blank pessimist," Bob declared. "But anyway, times have changed and the day of the individual inventor is past. It has been said, 'Nowadays the frontiers of science are inexorably advanced by well-organized research teams, superbly equipped, adequately financed, and backed by able management.' "

"You noticed who is saying this, of course," answered Bill. "None other than 'able management.' It helps them obtain the 'adequate financing.' "

"I used to be on the research team of a large toy company. We were organized to the hilt, and turned out some amazingly ingenious things, regularly—something new every week. One day I found out where the best ideas were coming from. There was an old man who used to come in the side door carrying a paper shopping bag. They said he was an Austrian refugee who worked in his own basement. He didn't speak English well, and was shy. He never was introduced to any of the fellows on the research team. But every time he visited the research director, sure enough, the next day we would have a new project."

"Then there is truth to the old motto," said Mike. " 'Behind every successful research team, there is an intelligent individual.' I just made up this motto, in case it doesn't sound familiar."

"Well, anyway," Bill piled his dishes on the tray, "your chances of winning on a patent are about as good as a ticket on the Irish Sweepstakes. It looks as if the only sure winner is the 'house.' It gets a small but guaranteed percentage every time. Patent attorneys are paid regardless, although it must be conceded that they do work for their money."

"When history is written," said Bob, "it will be noted that you have hit upon an all-pervading truth. In the stock exchange, brokers work for 'peanuts', but in the long run they do better than most successful investors."

"I read a book on 'Photography for Fun and Profit,' where the author told in great detail how to make outstanding pictures and where to sell them. He forgot that the most profit is made by dealers who sell supplies to photographers, and here you don't need any photographic talent."

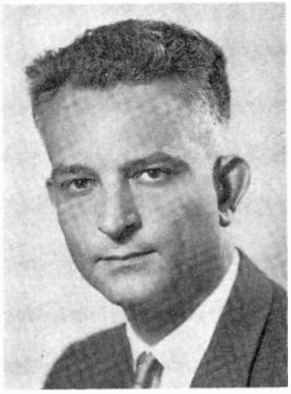
Mike smiled. "You, in turn, have forgotten the most successful percentage-collector of all: our government."

"There goes the bell. Lunchtime is over. Let's go back and console poor Sam."

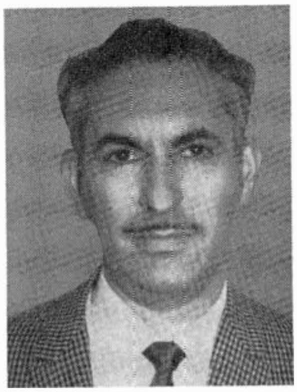
Marvin Camras, *Editor*

National Officers of the PGA, 1962-1963

R. W. Benson
Chairman, 1962–1963



F. A. Comerci
Vice Chairman, 1962–1963



M. Copel
Secretary-Treasurer, 1962–1963

Robert W. Benson (M'52) was born in Grand Island, Neb., on January 21, 1924. He received the B.S.E.E., M.S.E.E., and the Ph.D. degrees from Washington University, St. Louis, Mo., in 1948, 1949, and 1951, respectively.

From 1948 to 1954 he was a Research Associate with the Central Institute for the Deaf, St. Louis, Mo.; from 1954 to 1960 he was with the Armour Research Foundation, Chicago, Ill., as Assistant Director of the Physics Division, having previously served as supervisor of the acoustics section. During the past two years he served as Professor of Electrical Engineering at Vanderbilt University, Nashville, Tenn. He is currently President of Robert W. Benson & Associates, Inc., Nashville, Tenn., a firm devoted to research in the fields of electronics and physics. He has been active in all phases of acoustics including electroacoustics, physiological acoustics and room acoustics, as well as working in the fields of ultrasonics.

Dr. Benson is a Fellow of the Acoustical Society of America, a member of Sigma Xi, an editorial consultant to Industrial Research, a member of CHABA, a consultant to the American Academy of Ophthalology and Otolaryngology and the American Dental Associaton, and is a Past Chairman of the St. Louis Section of IRE.

Frank A. Comerci (SM'55) was born in Newark, N.J., on January 18, 1920. He received the B.S.E.E. degree from Newark College of Engineering, in 1943.

From 1943 to 1946 he served in the U. S. Army as a Communications Officer, installing and maintaining cryptographic speech communications systems. He joined the Rangertone Corporation in 1946, where he worked on the design of the first high-quality magnetic-tape recorder built in the United States. In 1947 he became affiliated with the Navy Material Laboratory, Brooklyn, N.Y., and was in charge of their Acoustics and Communications Section from 1950 to 1959. He was later employed by Audio Devices, Inc., Glen Brook, Conn., as Senior Electronic Engineer. At present he is Manager of the Magnetics Department of CBS Laboratories, Stamford, Conn., and has responsibility for fundamental and applied research on magnetic materials and magnetic recording techniques. He has written several papers on magnetic recording and flutter.

Mr. Comerci is a member of the Acoustical Society of America and the Audio Engineering Society, serving on the Editorial Board of the *Journal of the Audio Engineering Society* for several years, and is a member of the Sound Committee of the Society of Motion Picture and Television Engineers. He is chairman of the IRE Recording and Reproducing Committee and the Flutter Subcommittee and serves as IRE representative to ASA Section Committee Z-57 on Sound Recording.

Michel Copel (M'53–SM'57) was born in Paris, France, in 1916. He received the B.S. degree in 1935, and the E.E. degree in 1937, both from the Conservatoire National des Arts et Metiers in Paris. He also attended New York University.

From 1942 to 1946 he was engaged in the design and development of military loudspeaker equipment as Chief Design Engineer of University Loudspeakers. From 1946 to 1948 he was Senior Engineer at Dictograph Products, Inc. Since 1948 he has been engaged in investigations, developments and evaluations of audio communication equipment at the Naval Material Laboratory, Brooklyn, N.Y., where he has written a large number of technical reports on his

E. E. DAVID, JR.
Administrative Committee, 1962–1965

D. F. ELDRIDGE
Administrative Committee, 1962–1965

J. F. NOVAK
Administrative Committee, 1962–1965

work. He has headed the Acoustics Group since 1955.

Mr. Copel is a member of the Acoustical Society of America. He has served as Chairman of the Ways and Means Committee of PGA and Organizer of the Audio Sessions at the 1955, 1956, and 1961 IRE Conventions. He is currently participating in Standards work in the IRE and the American Standards Association, as Member and Chairman of several technical committees. In 1961 and 1962 he joined the U. S. delegation to the meetings of the International Electrotechnical Commission and the International Standards Organization.

Edward E. David, Jr. (A'48–M'56–SM'58) was born in Wilmington, N.C., on January 25, 1925. He received the B.E.E. degree from the Georgia Institute of Technology, Atlanta, in 1945 and the Sc.D. degree from the Massachusetts Institute of Technology, Cambridge, in 1950. During the latter period he was a research assistant at the M.I.T. Research Laboratory of Electronics working with microwave vacuum tubes and noise theory.

He joined Bell Telephone Laboratories, Inc., Murray Hill, N.J., in 1950, and worked subsequently in underwater sound and speech communication. He is presently Director of the Visual and Acoustics Research Laboratory in which capacity he is concerned with sensory communication, in particular, perception, coding and human information processing. He has been co-author of two books on the physics and physiology of speech communication, "Man's World of Sound" and "Waves and the Ear," both published by Doubleday and Company, New York, N.Y. His active interest in acoustics is evidenced by publications in the *Journal of the Acoustical Society of America*, the PROCEEDINGS of the IRE, and *Scientific American*.

Mr. David is a Fellow of the Acoustical Society of America. He has received professional awards from Eta Kappa Nu, the Georgia Institute of Technology, and the Summit, New Jersey, Junior Chamber of Commerce.

Donald F. Eldridge (A'50–M'55–SM'60) was born in Passaic, N.J., on January 30, 1929. He received the B.S.E.E. degree from Lehigh University, Bethlehem, Pa., in 1949.

He then joined the Boeing Airplane Company, Seattle, Wash., where he engaged in work covering many phases of dynamic data acquisition and reduction. In 1956 he became affiliated with the Research Division of Ampex Corporation, Redwood City, Calif., where he did research on many aspects of magnetic recording. His last position there was as head of the Magnetics Department of the Ampex Corporate Research Division, from which he resigned in December, 1960. He is presently Vice President and Technical Director of Memorex Corporation, Palo Alto, Calif.

James F. Novak (S'50–M'55–SM'61) was born in Oak Park, Ill. on September 5, 1926. He received the B.E.E.E. degree from the Illinois Institute of Technology, Chicago, in 1952.

He joined Jensen Manufacturing Company in 1952 as an engineer in the Advanced Engineering Group concerned with the design and development of horns, drivers and enclosures. He became Senior Design Engineer in 1956 and now directs a group responsible for direct radiator and loudspeaker enclosure design and research on diaphragm materials and various items related to electroacoustical transducers in general. He has presented and had published a number of technical papers and has been granted a number of patents.

Mr. Novak is a member of the A.S.A., A.E.S., and C.A.A.G.

PGA Awards for 1961

The Awards Committee of the Professional Group on Audio is proud to announce the following awards for the year 1961.

IRE-PGA Achievement Award

J. ROSS MACDONALD

To honor a member of the PGA, who, over a period of years, has made outstanding contributions to audio technology documented by papers in IRE publications. A certificate and $200 have been presented.

IRE-PGA Senior Award

W. H. BEAUBIAN AND H. B. MOORE

For the paper "Perception of Stereophonic Effect as a Function of Frequency," published in IRE Transactions on Audio, Vol. AU-8, pp. 144–153; September–October, 1960. A certificate and $100 award have been presented.

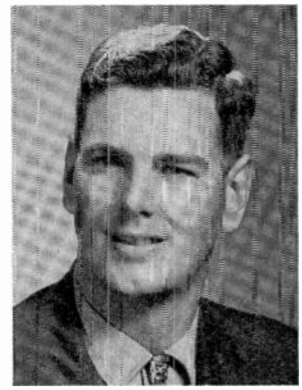
IRE-PGA Award

WILLIAM H. PIERCE

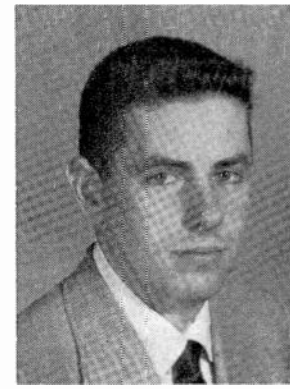
For his paper "The Use of Pole-Zero Concepts in Loudspeaker Feedback Compensation," published in IRE Transactions on Audio, Vol. AU-8, pp. 229–234; November–December, 1960. A certificate and cash award have been presented.

J. R. Macdonald

W. H. Beaubien



H. B. Moore



W. H. Pierce

J. Ross Macdonald, (S'44–A'48–SM'54–F'59) was born in Savannah, Ga., on February 27, 1923. He received the B.A. degree in physics from Williams College, Williamstown, Mass., and the S.B. degree in electrical engineering from The Massachusetts Institute of Technology, Cambridge, both in 1944. After teaching at the M.I.T. Technical Radar School and serving as a radar officer in the USNR, he returned to M.I.T. in 1946 and received the S.M. degree in electrical engineering in 1947. Upon completion of further graduate study in physics at M.I.T., he spent two years at Oxford University, England, as a Rhodes Scholar from Massachusetts, receiving the D.Phil. degree in Natural Philosophy (physics) in 1950.

For two years research in experimental and theoretical physics at Armour Research Foundation was carried out under his direction. He then spent a year's leave of absence at Argonne National Laboratory working on solid state physics problems. He joined Texas Instruments Incorporated, Dallas, in 1953 and is presently Director of the Corporate Physics Research Laboratory. In addition, as Clinical Associate Professor of Medical Electronics at the Southwestern Medical School of the University of Texas, Dallas, he consults in the fields of physics, chemical physics and electronics.

Dr. Macdonald is a Fellow of the American Physical Society, the AAAS, and a member of Phi Beta Kappa and Sigma Xi. He was selected for the PGA Senior Paper Award in 1957.

William H. Beaubien (M'58) was born in Toledo, Ohio, on December 31, 1923. After serving in the U. S. Air Force as a B-29 navigator in World War II, he graduated from the University of Toledo in 1948, with the B.S. degree in engineering, magna cum laude.

He joined General Electric Company, Syracuse, N.Y., in 1948, and participated in various kinds of advance electronic product development. After being named manager of Electronic Traffic Control Design Engineering for the Outdoor Lighting Department in 1954, he was designated Manager of Advance Product Development Engineering for the Radio Receiver Department, Utica, N.Y., in 1957.

Mr. Beaubien is a member of the Audio Engineering Society.

Harwood B. Moore was born in Providence, R. I., on March 9, 1928. He received the B.S. degree in electrical engineering from the University of Buffalo, N.Y., in 1953 and continued graduate studies at Northeastern University, Boston, Mass., from 1954 to 1955.

In 1953 he joined General Electric Company, Lynn, Mass., where he gained experience in electronic control and magnetic product design engineering. He is presently employed as Advance Product Development Engineer in the Radio Receiver Department.

Mr. Moore is a registered professional engineer in the state of New York and a member of the Audio Engineering Society.

William H. Pierce was born in Washington, D.C., on July 10, 1933. He received the B.A. degree from Harvard University, Cambridge, Mass., in 1955.

From 1955 to 1958 he served as an Officer in the Atlantic Fleet destroyer force. Since 1958 he has been a student in the Electrical Engineering Department at Stanford University, Stanford, Calif., where he received the M.S. degree in 1959.

Mr. Pierce has been a National Science Foundation Fellow in 1959-1960 and 1960-1961, while studying for the Doctorate degree.

PGA News

SPECIAL ANNOUNCEMENT
AMENDMENT OF PGA CONSTITUTION

The following changes in the PGA Constitution have been approved by our Administrative Committee and by the Professional Groups Committee. They will become effective thirty days after publication in these TRANSACTIONS, unless ten per cent of the Group Members file objections within thirty days of publication.

The revision of the Constitution and Bylaws for the Professional Group on Audio is proposed to achieve the following.

1) Have the Vice Chairman automatically become Chairman of the Group.
   Thus he will have had at least one year of experience with the Group. Under the present rules, the Chairman could be elected without any prior service on the Administrative Committee or knowledge of the functional details of the Group's activities.
2) The Chairman shall be a member of the Administrative Committee but without vote except to break a tie.
3) The retiring Chairman shall remain as a member of the Administrative Committee *ex officio* and without vote for a period of one year upon completion of his term of office.

CONSTITUTION ARTICLES AS REVISED
ARTICLE IV

MANAGEMENT AND ELECTION OF OFFICERS

(SUBSTITUTE THE FOLLOWING FOR SECTIONS 1, 2, 5, AND 7)

Section 1: The Group shall be managed by an Administrative Committee consisting of the Chairman and nine additional members of the Group, one of whom shall be the Vice Chairman (Chairman Elect) elected by the Group membership.

Section 2: The terms of office of the elected members of the Administrative Committee excluding the Chairman and Vice Chairman (Chairman Elect) shall be three years. The retiring Chairman shall remain as a member of the Administrative Committee *ex officio* and without vote for a period of one year upon completion of his term of office. The number of members necessary to bring the Committee to full strength shall be elected each year by the Group membership.

Section 5: The office of Chairman shall not be elective, but shall be filled automatically by the Vice Chairman at the expiration of the normal term of office. Nominees for the office of Vice Chairman (Chairman Elect) shall be a) from among the members of the Administrative Committee who are willing to run; b) any other group member nominated by the Administrative Committee; c) any other group member nominated by the nominating committee and approved by the Administrative Committee; d) any other Group member petitioned by 25 members of the Group.

Section 7: The Vice Chairman (Chairman Elect) shall be a voting member of the Administrative Committee. The Chairman shall be a nonvoting member except to break a tie.

ARTICLE V

POWERS, PRIVILEGES, AND DUTIES

(SUBSTITUTE THE FOLLOWING FOR SECTION 3)

Section 3: The Vice Chairman (Chairman Elect) shall assume the duties and have the powers, duties, privileges, and responsibilities of the Chairman during the latter's absence or incapacity and shall in general assist the Chairman in fulfilling his duties. He shall become Chairman upon the death, resignation, or permanent incapacity of the Chairman and the Administrative Committee is empowered to fill the vacancy of Vice Chairman (Chairman Elect) thus created.

MISCELLANEOUS

The following changes are to make the wording of other articles consistent with the above objectives, or to correct typographical errors in the previous Constitution.

Article I, Section 3, line 1—omit *the;* line 3, change *held* to *hold.*

Article III, Section 2, line 5—change *their* to *its.*

Article IV, Section 6, line 1—omit *Chairman and;* insert (*Chairman Elect*) after *Vice Chairman;* line 6, insert (*Chairman Elect*) after *Vice Chairman.*

Article V, Section 2, line 11—omit *and of the Professional Groups Committee;* Section 6, line 1, omit *newly elected* and insert (*Chairman Elect*) after *Vice Chairman.*

As previously indicated in these TRANSACTIONS (see May-June issue, page 61) the changes, drafted by H. E. Roys, Chairman of the IRE-PGA Committee on Constitution and Bylaws, have been pending for more than a year. It is hoped that they can be in effect at the time of our next election. Revisions of our Bylaws consistent with these changes have also been approved by the Administrative Committee. The latest Constitution and Bylaws will appear in a PGA Directory issue that is now being compiled.

The Photo-Magnetic-Electric Effect in Germanium as a Pickup Means for Magnetic Tape*

J. F. BANZHAF, III†, H. S. KATZENSTEIN‡, AND D. E. BECKERMAN†

Summary—The results of a study to determine the feasibility of using the PME effect in semiconductors as a pickup means for information stored on magnetic tape are presented, as well as pertinent information on the performance of a completed device. Because the output voltage of the device is proportional to the product of the instantaneous magnetic flux and the light intensity, reproduction of frequencies down to zero cycles per second is theoretically possible, as well as scanning speeds approaching the speed of light. A partial study of the factors affecting the PME voltage in germanium is also presented.

Introduction

In recording information on magnetic media, the tape or wire is polarized with a magnetization proportional to the instantaneous signal. The recovery of this information from the polarized medium is generally effected by passing the tape in front of a magnetic circuit which is linked by a multiturn coil. The motion of the tape past the poles of the magnetic circuit produces a changing magnetic flux in the circuit. The voltage induced in the coil is proportional to the rate of change of the magnetic flux. In the recording and reproduction of single-frequency sinusoidal signals, this proportionality to rate of change imposes no difficulty. Because the rate of change of sinusoidal signals is also sinusoidal, no distortion results. However, in the case of signals involving a wide range of frequencies, there is a loss in signal amplitude inversely proportional to the frequency of the signal. This loss necessitates correcting networks to recover the low-frequency output of the system. The ability to reproduce very low frequencies is thus severely limited by the amount of spurious hum and noise present in the system.

An upper limit on frequency also exists where the wavelength of the signal recorded on the magnetic tape is of the same size as the magnetic gap in the pickup head. The upper frequency limit can be raised by decreasing the magnetic gap of the pickup head or by increasing the tape transport speed. The former method is limited by those practical considerations involved in any miniaturization while the latter method is limited by considerations of tape wear and mechanical problems of high-speed tape transport.

* Received July 18, 1962; revised manuscript received July 27, 1962. The work described in this paper was done for Olympic Radio and Television, Division of the Lear-Siegler Corporation.
† Data and Controls Division, Lear-Siegler Corporation, Long Island City, N. Y. Formerly with Olympic Radio and Television.
‡ Solid State Radiation Incorporated, Los Angeles, Calif.

Several methods have been proposed to extend the inherent frequency limitations but each method has associated disadvantages. These methods include AM modulation of a carrier signal by a slowly varying signal (inefficient in terms of tape utilization), vibrating or rotating pickup heads (severe mechanical problems and limitations), frequency modulation (inefficient tape utilization), and other hybrid arrangements.

This paper is a report on the results of a study to develop a practical pickup device whose output would be directly proportional to the instantaneous magnetic flux rather than proportional to the rate of change of this flux. Such a device would have no low-frequency limitations because the amplitude of the output signal would be independent of the relative motion between the tape and the pickup device. Thus it would have an inherently uniform low-frequency response. The magnetoresistive, Hall, and photo-magnetic-electric effects in semiconductors were examined and the latter proved most satisfactory for the intended purpose.

The Photo-Magnetic-Electric Effect

Fig. 1 shows a parallelepiped of semiconductor material,

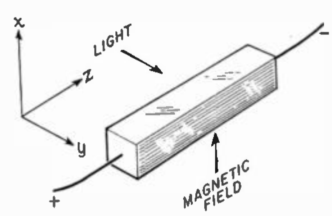


Fig. 1—Semiconductor crystal.

illuminated on the negative Y face with light of a conveniently chosen wavelength. The bombardment of photons creates hole-electron pairs on the negative Y face which diffuse in the positive Y direction. A magnetic field is applied to the sample in the positive X direction. The holes and electrons moving perpendicularly to the magnetic field experience a force which is perpendicular to both the field and the direction of diffusion, and is in an opposite direction for each type of carrier. Thus a separation of charges is effected and a potential is created between the ends of the crystal to which leads are attached. This phenomenon might also be described as a modified Hall effect in which the original movement of the carriers is caused by stimulated diffusion rather than by current flow. Over a certain range the PME voltage is proportional to the product of the magnetic field intensity and the illumination intensity, although both tend to show saturation effects [1]–[4].

Experimental Methods

All samples used in these experiments were parallelepipeds of n-type germanium attached to lucite strips with rubber cement or beeswax. At first, leads were attached by copper-plating the ends of the germanium and soldering thin wire leads, using ordinary solder and a low heat pencil-tip soldering iron. In the experiments with crystals 2 mils thick, the crystal breakage rate using this method was so high that instead the leads were painted on using conducting silver paint over the copper plating. Twisted wire leads or shielded cable were used wherever possible to eliminate stray pickup, and every effort was made to shield the sample from stray magnetic fields. A filtered, optically focused, Fisher microscope illumination light operated on dc was used for all experiments.

A Hewlett Packard Model 425A dc Micro Volt-Ammeter was used for all dc measurements, and a Ballantine Model 300 ac VTVM was used to amplify the ac signal to drive a Tektronix- type 531 oscilloscope with a Type 53/54C plug in unit. Shielded cable was used wherever necessary. No practical method of light intensity measurement was available, but the light intensity may be assumed to be constant during any single experiment unless otherwise indicated.

Results

The first experimental consideration was the linearity of the PME voltage with respect to small fields and the relative magnitudes of output voltages obtainable from small germanium samples. Fig. 2 (next page) indicates the PME voltage obtained from a 3 ohm-cm sample of germanium, 6 by 120 by 120 mils. Typical values for 40 ohm-cm etched samples 40 by 40 by 300 mils were 0.5 to 1 mv in a field of 120 gauss.

The desire to maximize the output voltage from the germanium sample encouraged an investigation of the effects of surface preparation. Theory suggests that the PME voltage would be maximized in a sample with low front-surface recombination and high back-surface recombination. A 15-second etch on the front (towards the light) surface of a germanium sample which previously had an undetectable (<0.02 mv) PME voltage produced a voltage output of 0.15 mv. Fig. 3 gives a clearer indication of the effects of etching the front surface of a germanium sample on its PME voltage.

The investigators also felt that the PME voltage might increase as the thickness of the crystal was decreased. If we consider a crystal of great thickness, it is plain that a large portion of the sample is not utilized in the production of the PME effect because the mean carrier diffusion length is limited by volume recombination and thus serves only as a shunt, lowering the output voltage of the device. Fig. 4 shows the results of grinding away the surface and reducing the thickness of the sample. The germanium slab was encapsulated in epoxy resin and the surface was ground down with smooth sandpaper. The face of the sample was etched for a uniform period after each grinding. The nonuniformity of the curve may have been due to minute cracks in the sample due to grinding.

One further consideration is saturation with respect to light. Fig. 5 shows the PME voltage at various field intensities with respect to the intensity of illumination.

An experimental model of a germanium PME pickup device was constructed by mounting a sample, 2 by 20 by 375 mils, of n-type 60 ohm-cm germanium on a thin sheet of transparent plastic. The face on which the light is to shine had been etched as previously described before being cemented to the plastic. The other face had been rubbed with sandpaper. The ends of the crystal were copper plated and wire leads were attached with conducting silver paint. The entire device was sprayed with a thin coating of plastic for protection (see Fig. 6).

The PME voltage of the sample under a powerful light source is presented in Fig. 7 as a function of static magnetic field. To evaluate the response of the device to signals recorded on magnetic tape, a special wide-track recording was made over the entire 250-mil face of a magnetic tape using a modified flyback transformer as a write head. The magnetic field within the gap was approximately 500 gauss rms and recordings were made at 50, 100, and 150 cps at 7½ ips. Under high light intensity and using pressure pads to keep the tape in intimate contact with the germanium, the output from the crystal was 2 mv peak-to-peak. This output was the same amplitude at 3¾ or 7½ ips or when the tape was moved slowly by hand. The experimenters felt that the higher frequency response of the device was limited by the sample width and orientation problems and so no other tests were performed.

Conclusions

The experimental model has demonstrated that the use of the PME effect in semiconductors appears feasible for the reproduction of signals recorded on magnetic tape. Additional development is needed but no problems exist which have not been surmounted in the allied fields of transistors and standard coil pickup heads. The PME pickup device should extend and exceed the restrictions heretofore placed on conventional pickup heads.

The most salient characteristic of a flux-responsive pickup device is its ability to extract magnetic information from a magnetic record independent of the state of motion existing between the pickup and the recording medium. Thus the head may be used for the reproduction of very-low frequency signals, down to and including dc. In extracting information from a magnetic medium, a large array of PME pickup devices may be used and only those particular pickups from which information is desired are

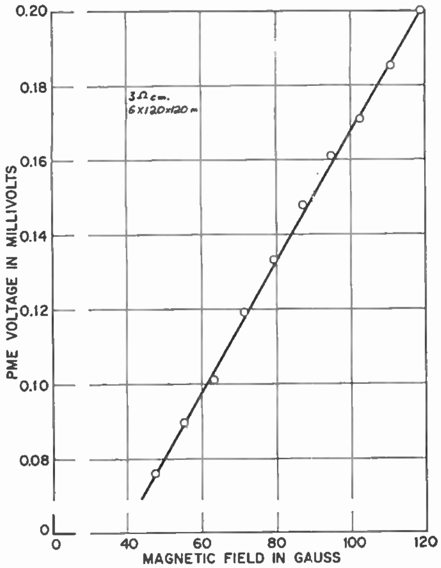


Fig. 2—PME voltage vs magnetic field.

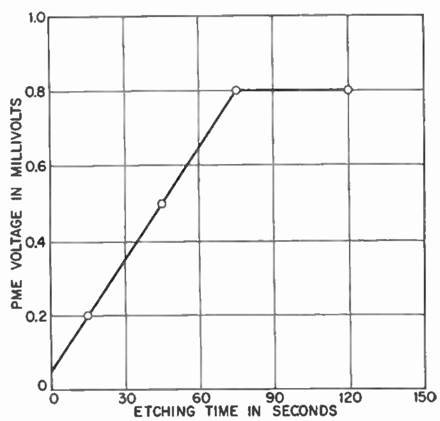


Fig. 3—PME voltage vs etching time.

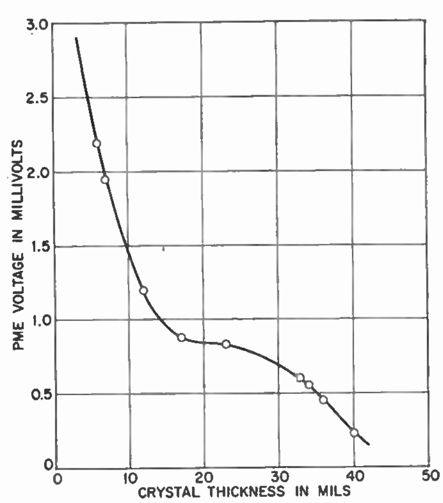


Fig. 4—PME voltage vs crystal thickness.

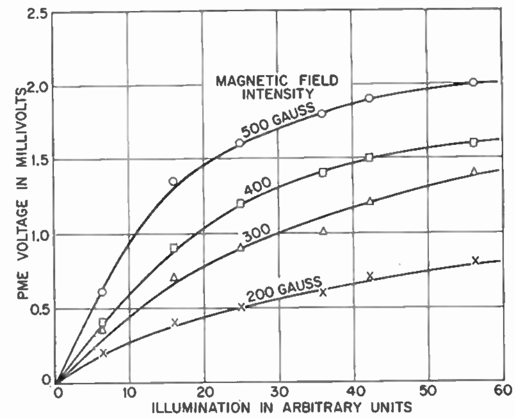


Fig. 5—PME voltage vs light intensity.

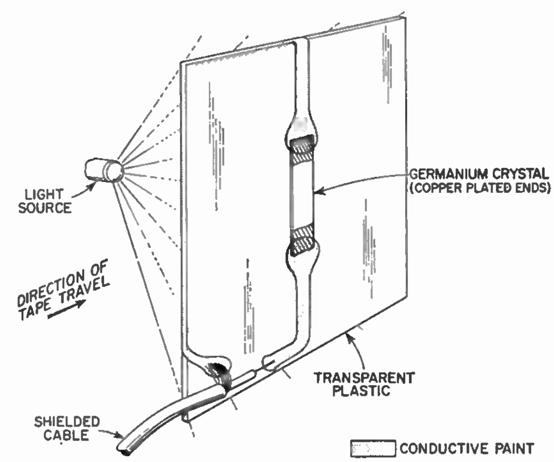


Fig. 6—Experimental pickup device.

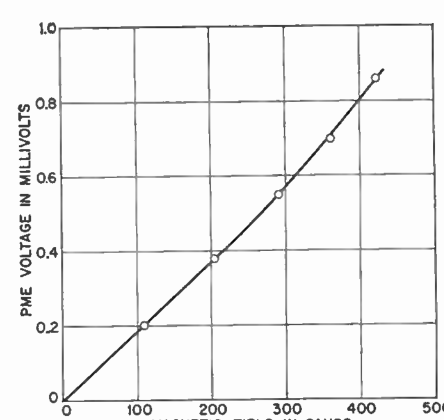


Fig. 7—PME voltage vs magnetic field of experimental pickup device.

actuated by supplying them with light energy. Thus, by using a method of rapidly scanning with a light beam, extremely high frequency signals may be read without the correspondingly high-speed mechanical scan of the medium. Still another application is the modulation of the light beam to produce a signal which is the product of two independent signals.

Acknowledgment

The authors would like to thank H. Sullivan for his many helpful suggestions, and B. Parzen for his help and encouragement. The writers are also endebted to L. Fletcher, of Darby and Darby, for his assistance in handling the patent applications, and to M. Jacobs.

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Audio-Radar Monitor Recorder*

A. A. GOLDBERG†, SENIOR MEMBER, IRE

Summary—The audio-radar monitor recorder is designed to continuously monitor and record 16 audio channels and one radar display on a single reel of magnetic tape. The tape is 2 in wide and moves at the speed of 15/16 ips. Each reel has a time capacitance in excess of 24 hours.

The radar recording is accomplished by means of a quantizing system that separates the video signal into 14 low-frequency channels. The PPI radar display is recorded out to a 30 mile range. Range definition between 0 and 20 miles is 0.2 miles and between 20 and 30 miles is 0.33 miles. The azimuth resolution is 1°. During playback, a PPI display, complete with range marks, is shown on the face of a scope.

Stationary magnetic-head and slow-speed tape assure trouble-free operation and long life.

Introduction

Radio and radar are the voice and eyes of the approach air traffic controller. These sounds and blips represent precious cargo; no errors can be afforded here. It is in the interest of all concerned that the audio-radar situation be continuously recorded—just in case!

Major airports like New York International presently record 12 audio channels but have no facility for recording the approach radar picture. The lack of a radar recording proved to be a handicap at the inquiry into the recent crash of two planes over New York City.

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† CBS Laboratories, Stamford, Conn.

Video recorders are characterized by high-speed moving parts and a prodigious appetite for magnetic tape. Twenty-four hour a day operation would be a difficult and costly enterprise. The proposed CBS Laboratories audio-radar monitor recorder avoids these problems by recording a quantized picture rather than the direct video.

Aircraft beacon transponder returns cannot be accommodated on the recorder to be described. This, however, is not an inherent system limitation. Beacons could be recorded but at the expense of fewer audio channels.

The audio-radar monitor recorder incorporates the folling features:

1) The monitor can record 16 channels of audio and one radar picture on a single tape so that they can be time correlated when played back.

2) Tape reels with a greater than 24-hour capacity can be used. Thus the recorder will require reloading only once a day.

3) System reliability is high and maintenance low.

Description

Direct radar video recording requires a bandwidth of approximately five megacycles. Existing video recorders use 2-in wide tape running at 15 ips, or a tape area of 30 square inches per second. This high consumption of tape, as well as the maintenance problem imposed by high-speed moving parts, has discouraged the use of radar monitor recording.

Examination of the display on the radar scope discloses that the targets themselves require narrow bandwidths if the range definition is limited to a reasonable value. Quantizing target information makes it possible to record the radar on a tape area of approximately 1 square inch per second. It is this small tape area consumption that makes the CBS Laboratories audio-radar monitor recorder system practical.

The ASR-3 surveillance radar rotates at 13 rpm, or 84° per second. The antenna lobewidth is approximately 3°. Thus, each target occupies 36 msec in time (see Fig. 1). The 20-mile radar range is normally used for approach control but the 30-mile range is occasionally employed, especially with the newer and faster aircraft.

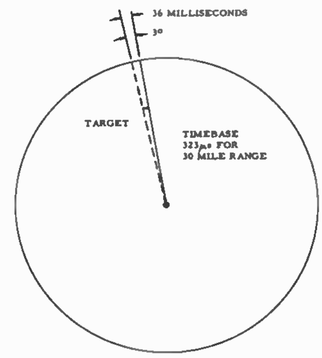

Fig. 1—ASR-3 PPI scope.

Radar Recorder

The radar recorder operates as follows: The 20-mile range is recorded in 1 per cent increments, range definiton being 0.2 miles. The range between 20 miles and 30 miles is recorded in 3 per cent increments with a definition of 0.33 miles. With the radar set to a 30-mile range, the radar video signal is electronically commutated into 100 separate, sequential signals during the first 20 miles of range, and 30 sequential signals during the 20-to-30-mile ranges. This is done during the 323 μsec time-base that the 30-mile radar requires.

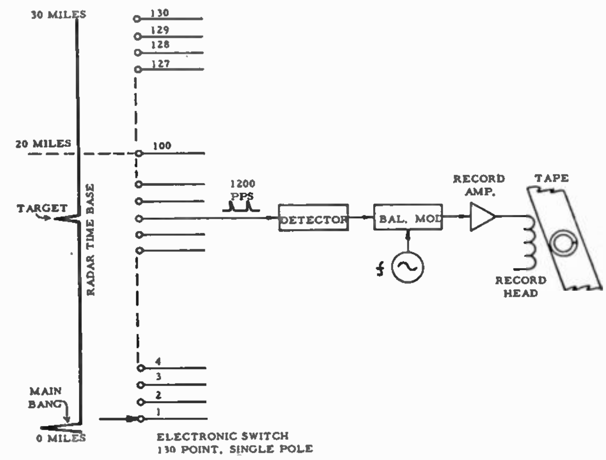

Fig. 2—Basic radar recording system.

Referring to Fig. 2, the main bang of the radar initiates the electronic sweep which sequentially keys a signal into 130 points within the 323 fsec. A target return causes one of the points to deliver a 1200-pps signals which is the radar pulse rate. These pulses are detected and the output of the detector is a broad pulse of 36 msec in duration which is the time that the echo is written on the PPI scope. The detector output pulse causes the balanced modulator to pass a sinusoidal burst from oscillator f which is recorded on one track of the magnetic tape. In effect, the radar echo has been quantized into a low-frequency audio signal that can be recorded on slow-moving magnetic tape.

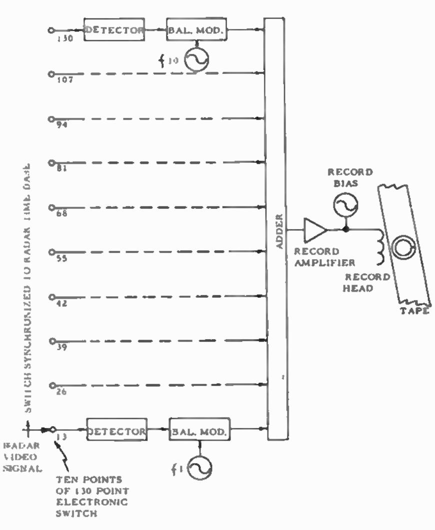

Fig. 3—One of 13 radar recording channels.

By a system of frequency multiplex the 130 range increments are recorded on 13 magnetic tape channels, one of which is shown in Fig. 3. Ten range increments are recorded on each magnetic track. Since it is intended that the tape be used to record audio up to 4000 cps, the oscillator frequencies in Fig. 3 are restricted to below 4000 cps. The lowest frequency is that which can include at least 10 cycles within 36 msec or approximately 280 cps. Frequencies f1 to f10 should not be closely harmonically related and are: 300, 500, 700, 1100, 1300, 1600, 1800, 2200, 2600 and 2900 cps. Outputs of all 10 balanced modulators are linearly added and recorded on one magnetic track. In order to prevent intermodulation, an ac-biased magnetic recording system is used and each frequency burst is 20 db below maximum record level. The signal-to-noise ratio is adequate due to narrow-band audio filters which discriminate against noise during playback.

All 13 records channels are similar. Oscillators f1 through f10 are stable tuning fork devices and are used in common for the other 12 record channels. In the event of a magnetic tape dropout in one channel, every thirteenth range point will momentarily disappear; this is less objectionable than experiencing the disappearance of a large contiguous block of range points.

Reproducer

Fig. 4 represents one of 13 playback channels. The playback head signals is amplified and separated into frequencies f1 through f10. Each tone is detected and the resultant broad pulses are fed to one point of an electronic switch. The 130-point electronic switch sequentially passes each detector signal to a common video amplifier and then to the intensity grid of a long persistence PPI scope. The deflection yoke of the scope is energized by a 1200-cps sawtooth current which is synchronized with the playback electronic switch. Simultaneously, the playback scope deflection yoke rotates mechanically at the same speed and phase as the original radar antenna. The resulting display closely approximates the original radar.

A fourteenth recording channel is used to synchronize the rotation of the playback scope deflection yoke with the radar antenna. Minor tape speed variations will not affect system azimuth accuracy because the target and synchronization tracks are on one tape. The synchronizing channel also contains a brightening pulse at zero degrees as an aid to checking the azimuth accuracy. The playback electronic switch need not be synchronized to the original radar either in phase or repetition rate.

Original radar range circles can be reproduced but may obscure targets superimposed upon them. To prevent this from happening, the radar range circles are recorded and reproduced only in alternate quadrants of alternate azimuth sweeps. Thus, a target that may normally be obscured by a range marker can be clearly seen on alternate azimuth sweeps. This is shown in Fig. 5.

The electronic radar map is not recorded because of its greater bandwidth requirements. It can, however, be reinserted during playback or it can take the form of a plastic overlay on the face of the playback scope.

Audio Recording

CBS Laboratories has demonstrated that magnetic recording densities of 8000 cycles per inch per second are practical. The audio-radar monitor records 3000 cps at a tape speed of 15/16 ips. This is 3200 cycles per inch per second and excellent reliability can be predicted.

Sixteen separate audio channels, each capable of 200- to 3000-cps response are available in the audio-radar monitor recorder. The radar recording requires 14 parallel tracks and these, added to the 16 audio tracks, make a total of 30 magnetic tracks on the 2-in wide tape.

The audio reproducer contains four audio playback channels that can be connected via pushbutton to any of the 16 audio playback magnetic heads. Means are provided to mix these four signals into a single channel if desired.

Magnetic Tape

The magnetic tape is 2 in wide, 7000 ft long, and records for 25 hours. Total tape thickness is 0.9 mil; it includes a 0.6 mil tensilized polyester backing and a 0.3 mil thick coating. Unlike video tape, the magnetic domains are

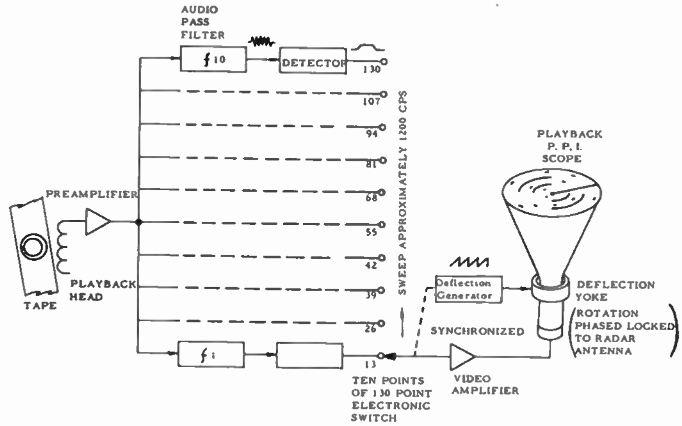

Fig. 4—Playback channels.

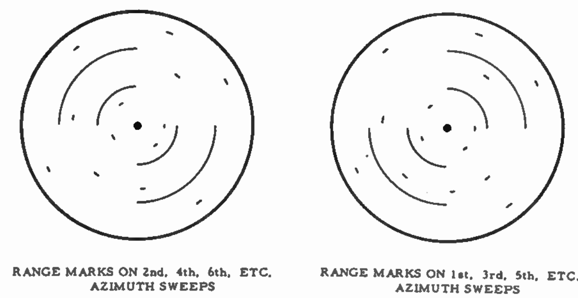

Fig. 5—Range markers on a playback scope.

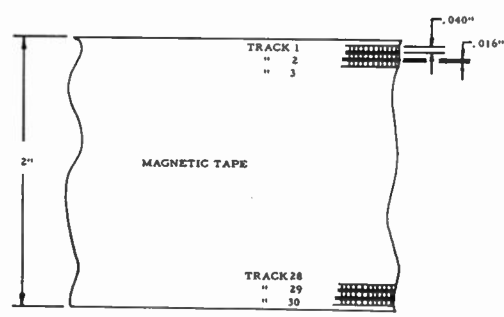

Fig. 6—Magnetic tape track specification.

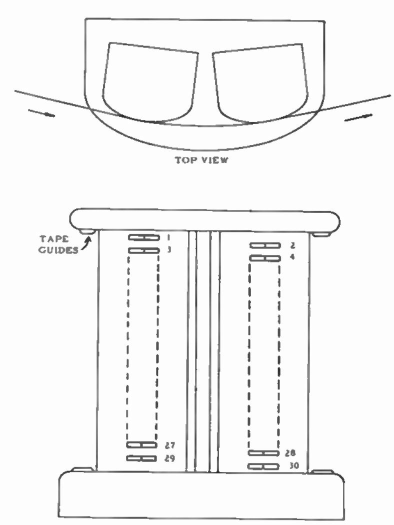

Fig. 7—Magnetic head.

longitudinally oriented. Each 12½ in-diameter reel of tape can be used thousands of times without loss of quality. The specification of magnetic tape tracks is shown in Fig. 6.

Magnetic Heads

Fig. 7 shows the geometry of the record and playback magnetic heads. Two staggered stacks of 15 heads each are used. All 30 heads are prealigned and securely mounted to a single base; a multiconnector provides convenient connection to the electronic circuitry.

The record head gaps are approximately 5 microns wide and the playback head gaps are approximately 1.5 microns wide. The absence of any moving heads or pole pieces in this recording system results in excellent reliability and low maintenance.

Apparatus

The audio-radar monitor recorder system consists of two main components; one recorder and one reproducer.

Recorder Apparatus

Each recorder occupies a standard 19-in wide rack which is approximately 6½ ft tall. As shown in Fig. 8, the tape deck occupies 3 ft of vertical space; the balance being used for electronic circuitry.

The recorder tape deck, as seen in Fig. 8, consists of a supply reel, supply idler, record heads, capstan, take-up idler and take-up reel. Separate torque motors and mechanical brakes are used for both the supply and take-up reels. A hysteresis motor insures constant capstan speed. Although the tape deck has provisions for rapid rewind, it is intended that the recorded tape be stored on the take-up reel and rewound on the reproducer only in the event of playback. An automatic device that does not depend on any signal from the magnetic tape will warn when the tape is within one hour of being exhausted. Additional alarms will be sounded if for any reason the tape stops moving or breaks.

Fig. 9 is a symbolic diagram of the 30-channel recorder circuit. All channels are identical. Inputs are bridged across 600-ohm lines, 0 db (±10 db). AGC amplifiers effectively maintain constant record level with input variations of 10 db, control action being accomplished without audible distortion.

One 50-kc bias oscillator is used in common for all channels. Each channel contains a bias amplifier to prevent interaction between record channels.

A decibel meter may be switched into one of three modes: input level, record level and bias level. A 30-position switch connects the meter to any one of the record channels.

The first 16 channels are allocated to audio recording and the incoming audio lines are connected to inputs 1 to 16.

Channels 17 through 30 are allocated to radar recording and these inputs receive signals from the radar quantizer which is shown in Fig. 10 (next page).

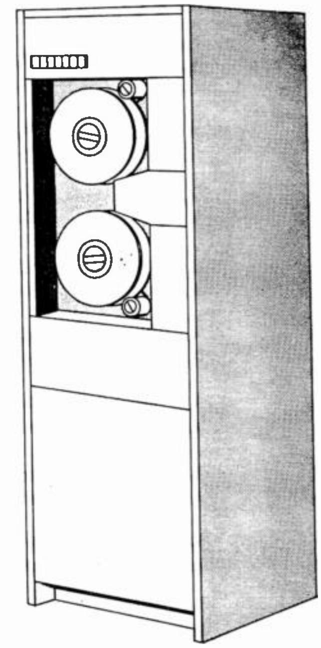

Fig. 8—Recorder.

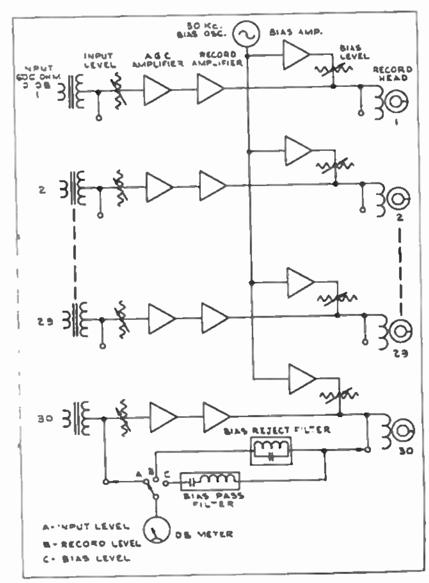

Fig. 9—30-channel recorder.

Radar Quantizer

The radar quantizer converts the radar information to a form which can be recorded in 14 channels of the recorder. Referring to Fig. 10, the following signals are required from the radar:

1) Video
2) Range markers
3) Time base
4) Azimuth servo
5) 0° azimuth pulse.

The radar video is amplified and sent to an adder where it is summed with the processed range markers and the 0° azimuth pulse. The summed signal feeds the pole of the

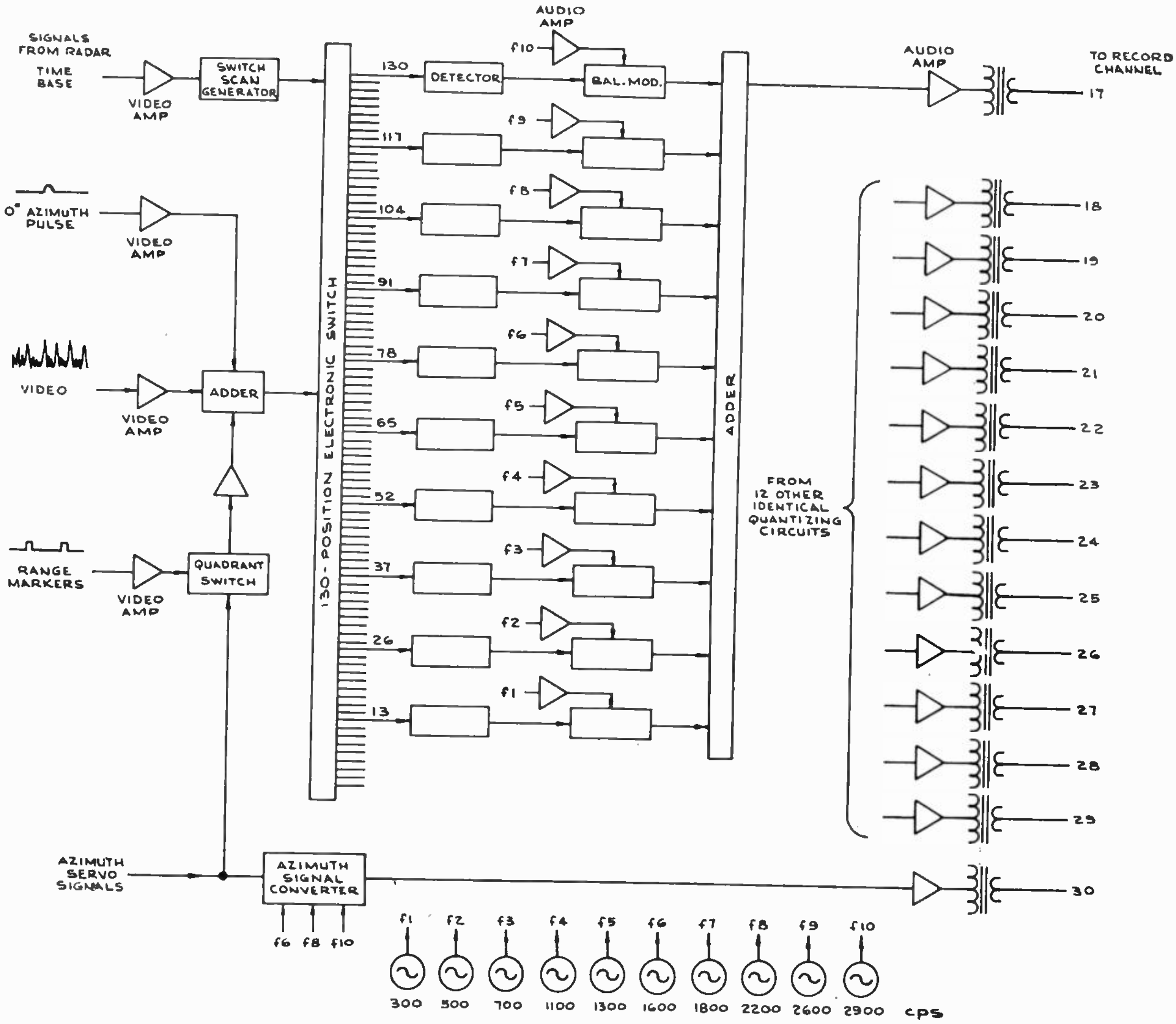

Fig. 10—Radar quantizer.

130-position electronic switch. The electronic switch is actuated from the radar time base signal via a switch scan generator. (Only one of the 13 quantizing channels is shown.) Every thirteenth position of the electronic switch feeds a separate detector that converts the 1200-pps signal to 36-msec pulses. These pulses are fed to a balanced modulator and gate the audio tone "on." The audio tones are derived from oscillators f1 through f10 and amplified in separate audio amplifiers. All ten balanced modulator outputs are combined in an adder and the resultant signal is amplified and fed to a 600-ohm, 0-db line.

The other positions of the electronic switch excite 12 other identical quantizing channels. The single set of oscillators (f1–f10) is common to all 13 channels.

The azimuth servo signal controls a quadrant switch that gates the range markers "on" and "off" as previously described. The azimuth servo signal also feeds an azimuth signal converter which changes the azimuth signal into tones for recording on the thirtieth channel of the recorder. Frequencies F6, F8 and F10 are used for this conversion.

The outputs of the radar quantizer are 14-audio lines at 600-ohm impedance and 0-db level.

Reproducer

The reproducer consists of two main elements, the audio section and the radar section. Both are shown symbolically in Fig. 11.

Audio Section: Tape tracks 1 through 16 are played by their respective magnetic heads and amplified through separate preamplifiers. The 16 preamplifier outputs are connected to a push-button matrix that allows any combination of audio channels to be switched into any combination of transcribe lines, A, B, C, and D. The transcribe-line signals are amplified and appear at the four 600-ohm, 0-db lines, across which is bridged a mixing switch that combines the A, B, C, D lines for earphone listening, loudspeaker listening, and for transcribing.

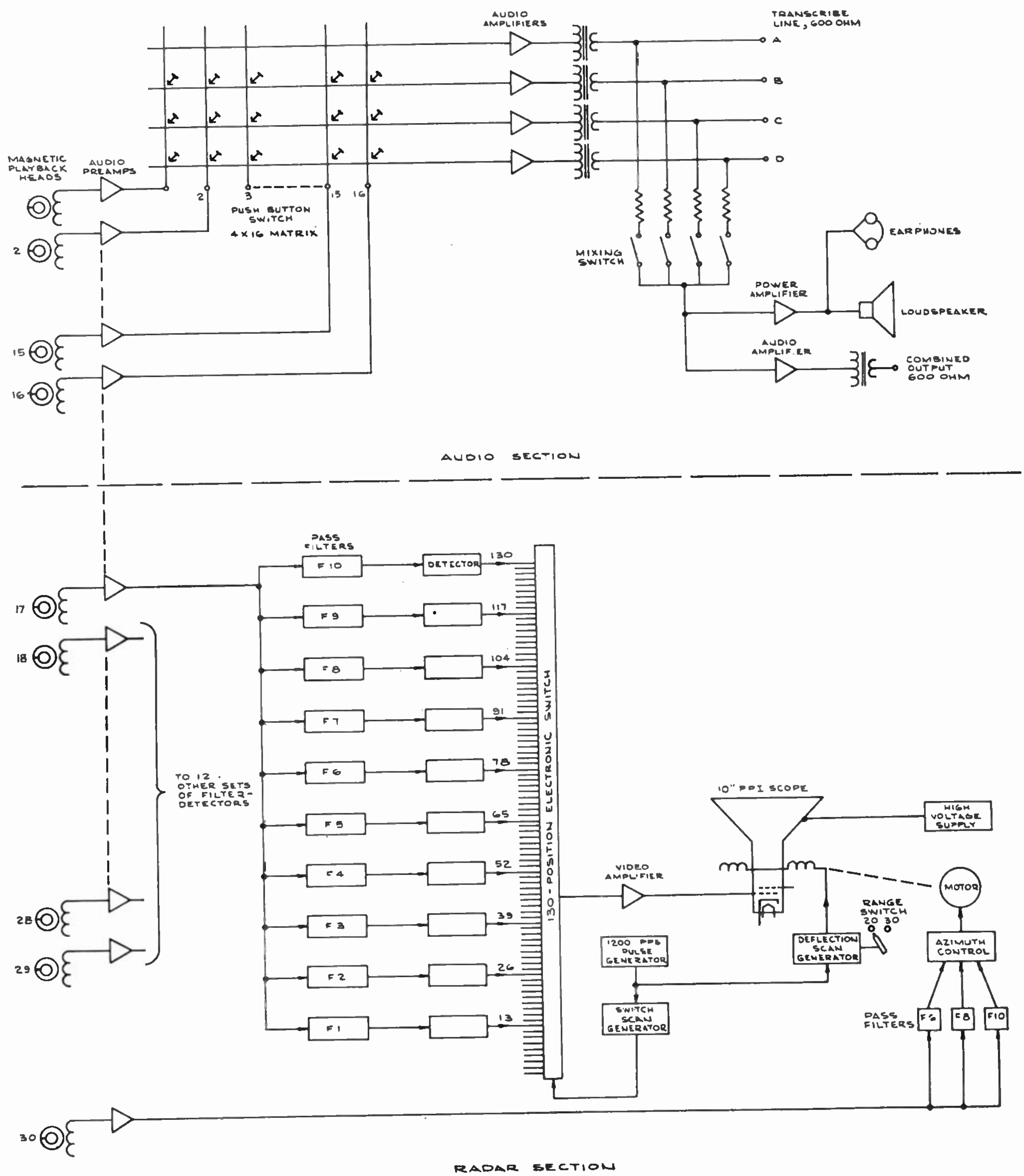

Fig. 11—Reproducer block diagram.

Radar Section: Tape tracks 17 through 30 are played by their respective magnetic heads and amplified in separate preamplifiers. With the exception of channel 30, each preamplifier feeds an audio filter detector network consisting of 10 resonant pass filters F1 through F10 and 10 detectors. The combined audio tones on each playback channel are separated by the audio filter, and the detector generates a 36-msec pulse for each target. The output of each of the 10 detectors is connected to one associated position of the 130-position electronic switch. As this switch is scanned at a 1200-cps rate, the output of the switch resembles the original radar video signal. The video signal is amplified and applied to the intensity grid of a 10-in PPI scope.

A 1200-pps free-running pulse generator excites the switch scan generator as well as the deflection scan generator for the PPI scope.

The preamplifier output of channel 30 is separated through filters f6, f8, and f10 for reconstituting the azimuth signal of the original radar. This azimuth signal controls the motor that rotates the deflection yoke of the PPI scope.

The PPI scope high voltage is generated by the customary high voltage supply.

Mechanical Design

The reproducer is self-contained and is shown in Fig. 12. The reels of tape are threaded on the vertical tape deck and the operator sits in front of the console and looks down at the PPI scope as though he is operating a radar. All controls for operating the tape deck functions, as well as electrical circuit controls, are handy to the operator and are designed in accordance with relevent human engineering considerations. A loudspeaker is located above the tape deck. Large caster wheels and a guide handle are furnished for convenience in moving the reproducer from one area to another.

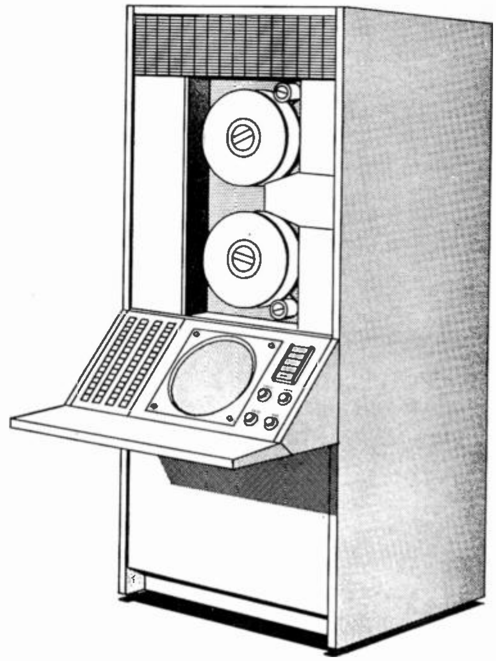

Fig. 12—Reproducer.

Conclusion

The audio-radar monitor recorder provides a convenient, reliable and inexpensive means for making a permanent record of the air traffic control situation. Although the system has not been built, experience and study has proved its feasibility. New requirements for recording the transponder returns can be accommodated in future designs.

Acknowledgments

B. B. Bauer, Vice President of CBS Laboratories, contributed to the study. The cooperation of J. E. Grambart, ARDS, Federal Aviation Agency, is gratefully acknowledged.

Power Dissipation in Class B Amplifiers*

LEONARD BAKER†, MEMBER, IRE

Summary—A linear treatment is developed which will predict the conversion efficiency, power output, and power dissipation for sinusoidal Class B push-pull operation of an imperfect device such as a silicon power transistor. Normalized expressions for conversion efficiency, power outputs, and power dissipation as functions of the ratio of load to dc saturation resistance are determined. The results are of particular interest to amplifier designers. Limiting cases are considered in order to indicate the established expressions for the ideal device. Calculated and measured data agree well enough to justify the use of the theory for practical purposes.

Introduction

PAPERS ON Class B amplifiers present graphical methods for solving design problems followed by the use of cut-and-try methods to obtain the desired results. A load line is placed on the output plane and computations are made for the maximum power output and maximum power dissipation, but computations for one load line cannot lead to conclusions of a general nature. The literature usually indicates that maximum power dissipation occurs at 40.5 per cent or 100 per cent of maximum power output. The following analysis shows that the maximum ratio of power dissipation to maximum power output is a continuous function of the load resistance and dc saturation resistance. The maximum power dissipation will occur between 40.5 per cent and 100 per cent of maximum power output when the ratio of load resistance to dc saturation resistance varies from infinity to zero.[1]

Basic Power Equations

At the present state of the art, devices of high power gain and good linearity are available. With high power gain we may neglect the signal input power. We must then consider the power input from the dc supply, the power dissipated in our output devices and the power delivered to the load.

To reach general conclusions it is best to have a reference which, in this case, will be the maximum power delivered to the load in the system under study. The maximum value of the load power is due to the device boundaries. For linear operation, the signal is driven up to the active-saturation-region boundary of the output devices. If every power quantity of interest is normalized with respect to the maximum power output, we can make the following statement:

$$\frac{P_{d(\text{two})}}{P_{0_{\max}}} = \frac{P_{\text{in}}}{P_{0_{\max}}} - \frac{P_0}{P_{0_{\max}}}. \qquad [1]$$

In (1) the normalized power output is considered the independent variable and the normalized power dissipation is considered the dependent variable. Since the maximum power output is a constant for a particular system, (1) can be considered as a functonal relationship between the power dissipation of both output devices and the power delivered to the load. We seek an attack such that the normalized power input will be an explicit function of the normalized power output and the ratio of load resistance to dc saturation resistance. This ratio is considered as a variable parameter.

Derivation of the Normalized Power Dissipation

The power input is the dc supply voltage multiplied by the average value of the rectified output current. For a sine wave the average value of the rectified output current is $(2/\pi)I_0$, where I_0 is the peak current in the load.

We will take the second ratio of the right-hand side of (1) and present it in a different form, representing a more general characteristic of the amplifier. Our first step is to define the load resistance R_L and the dc saturation resistance R_s. For push-pull transformer-type output stages, the load is defined as the reflected resistance appearing across one-half of the primary of the output transformer. For single-ended push-pull output stages, the load is the output side of the capacitor to ground.

Fig. 1 shows the dc saturation resistance R_s on the output volt-ampere plane. The saturation resistance R_s is defined as the reciprocal of the slope of a line which is the secant connecting the origin of the output volt-ampere plane and the intersection of the load line with the active-saturation region boundary. If the active-saturation region boundary is a linear function, the secant connecting the two points merges into the boundary. The saturation resistance as used here is defined as the "dc saturation resistance."

The power input then is

$$P_{\text{in}} = E_{cc}\left(\frac{2}{\pi} I_0\right). \qquad (2)$$

* Received July 18, 1962.

† Engineering Staff, Tele-Dynamics Division, American Bosch Arma Corp., Philadelphia, Pa.

[1] L. Baker, "Design of Class B Amplifiers Using Imperfect Devices," M.S. thesis, Moore School of Elec. Engrg., Univ. of Pennsylvania, Philadelphia, Pa.; 1959.

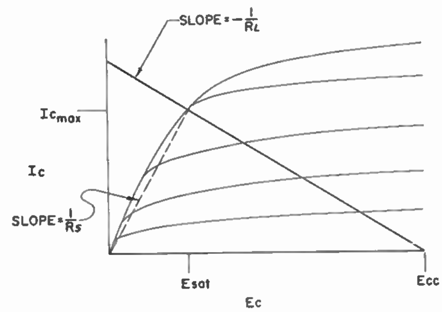

Fig. 1—DC saturation resistance on the output volt-ampere plane.

For a sine wave, the maximum power developed across a resistive load is

$$P_{0_{max}} = \frac{(E_{0_{max}})(I_{0_{max}})}{2}. \qquad (3)$$

$I_{0_{max}}$ is the maximum value that the peak current in the load may assume in the circuit under consideration. Similarly, $E_{0_{max}}$ is the maximum value that the peak voltage across the load may assume in the circuit under consideration. It is most important to note that E_0 and I_0 are defined as the peak output voltage and peak output current and are functions only of the signal level. Whereas, $E_{0_{max}}$ and $I_{0_{max}}$ are respectively the maximum (peak) output voltage and maximum (peak) output current and are functions only of the active-saturation region boundary line.

The maximum output voltage across the load E_{0max} can be written as a difference of two voltages. For linear operation we restrict the output voltage to the linear active region. The maximum output voltage is the supply voltage E_{cc}, minus the saturation voltage E_{sat}, required to ensure linearity.

Thus,

$$E_{0_{max}} = E_{cc} - E_{sat}. \qquad (4)$$

The maximum output current is also set at the limit of the active-saturation region boundary to ensure linear operation. Substituting (4) in (3) and then dividing (3) into (2) will yield an expression which describes the ratio of dc power input to maximum power output in terms of currents and voltages.

Therefore,

$$\frac{P_{in}}{P_{0_{max}}} = \left(\frac{4}{\pi}\right)\left(\frac{I_0}{I_{0_{max}}}\right)\left(\frac{E_{cc}}{E_{cc} - E_{sat}}\right). \qquad (5)$$

In (5) we choose the supply voltage E_{cc} as a fixed quantity. The output current I_0 is a variable. The maximum output current $I_{0_{max}}$ and the saturation voltage E_{sat} are boundary value variables. The second ratio at the right-hand side of (5) is suitable for modification as an explicit function of the normalized power output. This is done by realizing that the ratio of I_0 over $I_{0_{max}}$ is the normalized peak current in the load. This is related to the normalized power output as follows:

$$\frac{I_0}{I_{0_{max}}} = \sqrt{\frac{P_0}{P_{0_{max}}}}. \qquad (6)$$

The normalized power input is expressed as an explicit function of the normalized power output by substituting (6) in (5):

$$\frac{P_{in}}{P_{0_{max}}} = \left(\frac{4}{\pi}\right)\left(\frac{E_{cc}}{E_{cc} - E_{sat}}\right)\sqrt{\frac{P_0}{P_{0_{max}}}}. \qquad (7)$$

We can consider the supply voltage to be dropped across the load resistance, and the dc saturation resistance permits a mathematical relationship between the saturation voltage and the supply voltage via the voltage division rule

$$E_{sat} = \frac{R_s}{R_L + R_s} E_{cc}. \qquad (8)$$

We express (8) in a more useful form and then substitute (8) in (7):

$$\frac{P_{in}}{P_{0_{max}}} = \frac{4}{\pi}\cdot\left[\frac{\left(\frac{R_L}{R_s}\right)+1}{\left(\frac{R_L}{R_s}\right)}\right]\cdot\sqrt{\frac{P_0}{P_{0_{max}}}}. \qquad (9)$$

The normalized power input expressed in (9) consists of three factors. The first factor is a constant which is the average value of a rectified sine wave divided by the mean-square value of a sine wave. The second factor is a property of the system and not the waveform and may be expressed in various ways. The form used by the writer is arbitrary. The third factor of (9) depends on the amount of drive applied to the amplifier. Insertion of (9) in (1) provides the normalized power dissipation of the output devices as a function of the normalized power output:

$$\frac{P_d(\text{two})}{P_{0_{max}}} = \frac{4}{\pi}\cdot\left[\frac{\left(\frac{R_L}{R_s}\right)+1}{\left(\frac{R_L}{R_s}\right)}\right]\cdot\sqrt{\frac{P_0}{P_{0_{max}}}} - \frac{P_0}{P_{0_{max}}} \qquad (10)$$

where

$$0 \leq \frac{P_0}{P_{0_{max}}} \leq 1$$

and

$$0 < \frac{R_L}{R_s} < \infty.$$

A plot of (10) is shown in Fig. 2.

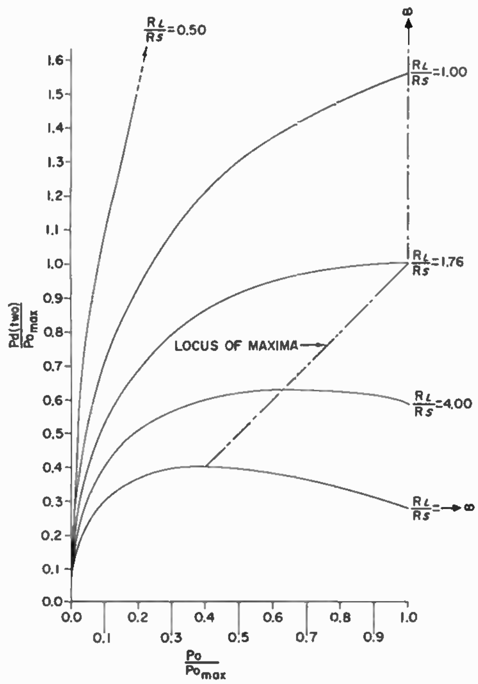

Fig. 2—Normalized power dissipation.

The limits for the normalized power output for (10) are between zero and one due to the normalization used. For purposes of analysis we must admit that the load resistance and saturation resistance can, individually, assume any value from zero to infinity. This determines the limits for the ratio of the load and dc saturation resistance. This ratio R_L/R_s is considered a variable parameter of (10).

The Maxima Functions

Again, it is emphasized that the maximum power output is a constant for any given amplifier. With this in mind we may consider (10) as a relationship between the power dissipation of the output devices and the power delivered to the load. We immediately raise certain questions. What is the maximum normalized power dissipation? What is the normalized power output for the occurrence of maximum normalized power dissipation? Clearly, the answers are useful criteria, from the power consideration point of view, for Class B amplifier design during sinusoidal excitation. To provide the answers to these questions involves the "maxima determination" of a function which has a bounded independent variable (which would be expected from a physical problem) and a variable parameter which is positive and unbounded. This means that we must check the end points of the functions for a maxima in addition to the "in-between points." The function must be checked in this manner for all values of the variable parameter (ratio of load to dc saturation resistance).

Taking the derivative of the normalized power dissipation expressed in (10), with respect to the normalized power output, and setting it equal to zero will yield the normalized power output at maximum normalized power dissipation:

$$\left.\frac{P_0}{P_{0_{\max}}}\right]_{\mathrm{Max}\{P_{d(\mathrm{two})}/P_{0_{\max}}\}} = \left(\frac{2}{\pi}\right)^2 \left[\frac{\left(\frac{R_L}{R_s}\right)+1}{\left(\frac{R_L}{R_s}\right)}\right]^2. \qquad (11)$$

Since the normalized power output can never be greater than unity, there must be a limit to the usefulness of (11). Setting the normalized power output equal to, or less than, unity will lead us to the least value of load-to-dc saturation resistance for which (11) is still valid. We now set the right-hand side of (11) equal to or less than unity:

$$\left(\frac{2}{\pi}\right)^2 \left[\frac{\left(\frac{R_L}{R_s}\right)+1}{\left(\frac{R_L}{R_s}\right)}\right]^2 \leq 1. \qquad (12)$$

With a little manipulation we are led to the least value of the variable parameter that permits (11) to hold. The maximum value of the variable parameter is infinity and the value of the breakpoint is 1.76. The region of validity for (11) is now established.

$$\left.\frac{P_0}{P_{0_{\max}}}\right]_{\mathrm{Max}\{P_{d(\mathrm{two})}/P_{0_{\max}}\}} = \left(\frac{2}{\pi}\right)^2 \left[\frac{\left(\frac{R_L}{R_s}\right)+1}{\left(\frac{R_L}{R_s}\right)}\right] \qquad (13)$$

where

$$1.76 \leq \frac{R_L}{R_s} < \infty.$$

The ratio of load resistance to dc saturation resistance has been so established that it can take a position anywhere from zero to infinity. What, then, is the expression for the normalized power output at maximum normalized power dissipation when the variable parameter lies between 1.76 and zero? If we insert values of load-to-dc saturation resistance less than 1.76 into (11), we obtain normalized power outputs at maximum normalized power dissipations that exceed unity. This is inconsistent with the maximum limit of the normalized power output which can never exceed unity. Consequently, the normalized power output at maximum normalized power dissipation equals unity when the ratio of load-to-dc saturation resistance exists between 1.76 and zero. The following statements can now be made:

$$\left.\frac{P_0}{P_{0_{\max}}}\right]_{\operatorname{Max}\{P_{d(\text{two})}/P_{0_{\max}}\}} = \begin{cases} 1 & 0 < \dfrac{R_L}{R_s} \le 1.76 \\ \left(\dfrac{2}{\pi}\right)^2 \left[\dfrac{\left(\dfrac{R_L}{R_s}\right)+1}{\left(\dfrac{R_L}{R_s}\right)}\right]^2 & 1.76 \le \dfrac{R_L}{R_s} < \infty \end{cases} \quad (14)$$

A plot of (14) is shown in Fig. 3.

The maximum normalized power dissipation follows by substituting (14) in (10). With a little manipulation the following statements can now be made:

$$\operatorname{Max}\left[\frac{P_{d_{(\text{two})}}}{P_{0_{\max}}}\right] = \begin{cases} \dfrac{4}{\pi}\left[\dfrac{\left(\dfrac{R_L}{R_s}\right)+1}{\left(\dfrac{R_L}{R_s}\right)}\right] - 1 & 0 < \dfrac{R_L}{R_s} \le 1.76 \\ \left(\dfrac{2}{\pi}\right)^2 \left[\dfrac{\left(\dfrac{R_L}{R_s}\right)+1}{\left(\dfrac{R_L}{R_s}\right)}\right]^2 & 1.76 \le \dfrac{R_L}{R_s} < \infty \end{cases} \quad (15)$$

A plot of (15) is shown in Fig. 4.

An Equality

Observations of (14) and (15) indicated that the normalized power output at maximum normalized power dissipation was equal to the maximum normalized power dissipation when the ratio of load to saturation resistance is between 1.76 and infinity. An equality can now be made:

$$\left[\frac{P_0}{P_{0_{\max}}}\right]_{\operatorname{Max}\{P_{d(\text{two})}/P_{0_{\max}}\}} = \operatorname{Max}\left[\frac{P_{d_{(\text{two})}}}{P_{0_{\max}}}\right] = \left(\frac{2}{\pi}\right)^2 \left[\frac{\left(\dfrac{R_L}{R_s}\right)+1}{\left(\dfrac{R_L}{R_s}\right)}\right]^2 \quad (16)$$

when

$$1.76 \le \frac{R_L}{R_s} < \infty.$$

The Conversion Efficiency

The conversion efficiency is an important quantity since it is an index to the merit of a system. The ratio of power delivered to the load to power into the amplifier, from the supply, is by definition, the conversion efficiency

$$\eta = \frac{P_0}{P_{\text{in}}}. \quad (17)$$

Normalizing the numerator and denominator of (17) with respect to the maximum power output and substituting (9) in (17) will express the conversion efficiency in a different manner:

$$\eta = \frac{\pi}{4}\left[\frac{\left(\dfrac{R_L}{R_s}\right)}{\left(\dfrac{R_L}{R_s}\right)+1}\right]\sqrt{\frac{P_0}{P_{0_{\max}}}} \quad (18)$$

where

$$0 \le \frac{P_0}{P_{0_{\max}}} \le 1$$

and

$$0 < \frac{R_L}{R_s} < \infty.$$

A plot of (18) is shown in Fig. 5.

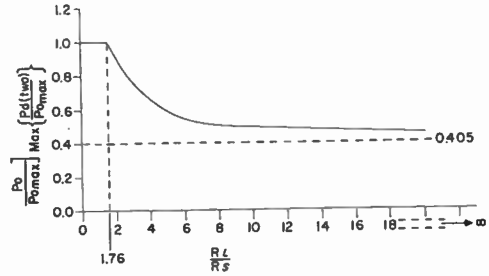

Fig. 3—Normalized power output for maximum normalized power dissipation.

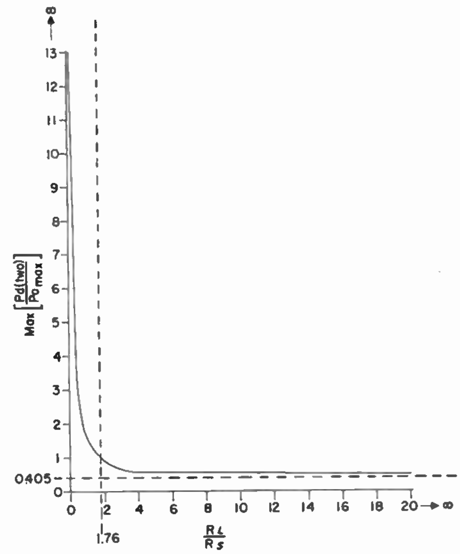

Fig. 4—Maximum normalized power dissipation.

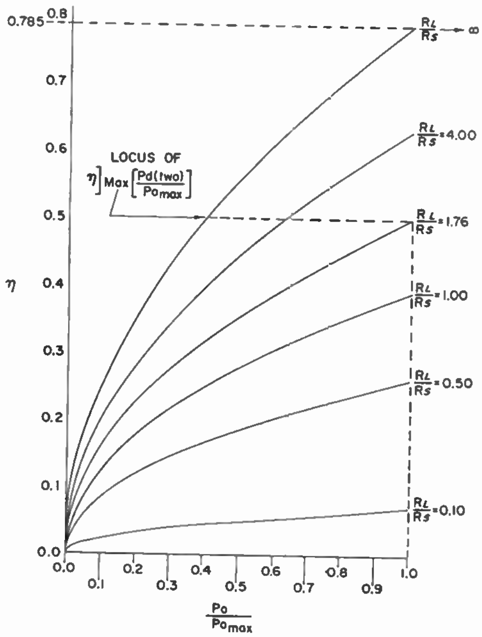

Fig. 5—Conversion efficiency.

The conversion efficiency at maximum normalized power dissipation is obtained by substituting (14) into (18):

$$\eta\Big]_{\mathrm{Max}\{P_{d(\mathrm{two})}/P_{0_{\max}}\}} = \begin{cases} \dfrac{\pi}{4}\left[\dfrac{\left(\dfrac{R_L}{R_s}\right)}{\left(\dfrac{R_L}{R_s}\right)+1}\right] & 0 < \dfrac{R_L}{R_s} \le 1.76 \\ 0.5 & 1.76 \le \dfrac{R_L}{R_s} < \infty. \end{cases} \qquad (19)$$

From (19) we realize that during maximum power dissipation the power output is one-half the power input when the ratio of load resistance to dc saturation resistance is greater than, or equal to, 1.76. This is consistent with the equality demonstrated in (16).

Limiting Cases

An ideal device, from our point of view, is one whose saturation resistance is zero. Equivalently, the ratio of load resistance to saturation resistance approaches infinity. With the use of L'Hospital's Rule, we let the limit of the variable parameter, in all our equations, approach infinity. Our expressions then become statements pertaining to the ideal device. This means that the signal can swing from the voltage axis to the correct axis of the output volt-ampere plane. If the load resistance is large compared to the dc saturation resistance, the following equations which are arrived at in the limit are good approximations to the situation encountered in practice when using very low dc saturation resistance devices.

Using (10):

$$\lim_{\frac{R_L}{R_s}\to\infty} \frac{P_{d_{(\mathrm{two})}}}{P_{0_{\max}}} = \frac{4}{\pi}\sqrt{\frac{P_0}{P_{0_{\max}}}} - \frac{P_0}{P_{0_{\max}}} \qquad (20)$$

Using (14):

$$\lim_{\frac{R_L}{R_s}\to\infty} \frac{P_0}{P_{0_{\max}}}\Big]_{\mathrm{Max}\{P_{d(\mathrm{two})}/P_{0_{\max}}\}} = \left(\frac{2}{\pi}\right)^2 \qquad (21)$$

Using (15):

$$\lim_{\frac{R_L}{R_s}\to\infty} \mathrm{Max}\left[\frac{P_{d_{(\mathrm{two})}}}{P_{0_{\max}}}\right] = \left(\frac{2}{\pi}\right)^2. \qquad (22)$$

Using (18):

$$\lim_{\frac{R_L}{R_s}\to\infty} \eta = \frac{\pi}{4}\sqrt{\frac{P_0}{P_{0_{\max}}}} \qquad (23)$$

Using (19):

$$\lim_{\frac{R_L}{R_s}\to\infty} \eta\Big]_{\mathrm{Max}\{P_{d(\mathrm{two})}/P_{0_{\max}}\}} = 0.5 \qquad (24)$$

Eqs. (20) and (22) show that the normalized power dissipation increases as the normalized power output increases up to a maximum of 0.405. Thereafter, the normalized power dissipation decreases with increased normalized power output until the normalized power dissipation is 0.28 when the normalized power output is unity. Eq. (21) indicates that the normalized power output at maximum normalized power dissipation is equal to 0.405. The conversion efficiency shown in (23) indicates that the ratio of power output to power input is proportional to the root of the normalized power output. At maximum power output the conversion efficiency is 0.785. Eq. (24) demonstrates that the power output is one-half the power input for the condition of maximum dissipation. Since the maximum normalized power dissipation for ideal devices during sinusoidal operation is 0.405, this means that the maximum power output can approach 4.95 times the maximum power dissipation on one output device.

Greatest Value for Maximum Power Output

We have been using the maximum power output as our reference. Given a fixed power supply voltage E_{cc} and active-saturation region boundary, we can determine the greatest value for maximum power output in terms of the supply voltage and the dc saturation resistance.[2] The condition for which this occurs is also of interest. The maximum power output across a resistive load when the signal is a sine wave is

$$P_{0_{\max}} = \tfrac{1}{2} E_{0_{\max}} I_{0_{\max}} = \tfrac{1}{2}(E_{cc} - E_{sat}) I_{0_{\max}}. \qquad (25)$$

We consider $I_{0_{\max}}$ as a function of E_{sat}. Then we take the first derivation of (25) with respect to E_{sat}:

$$\frac{dP_{0_{\max}}}{dE_{sat}} = \frac{1}{2}\left[(E_{cc} - E_{sat}) \frac{dI_{0_{\max}}}{dE_{sat}}\right] - I_{0_{\max}}. \qquad (26)$$

The conditions for realizing the maximum power output can be found by setting (26) equal to zero. We will state this condition in terms of load conductance:

$$\frac{1}{R_L} = \frac{I_{0_{\max}}}{E_{cc} - E_{sat}} = -\frac{dI_{0_{\max}}}{dE_{sat}} = -\frac{1}{r_s}. \qquad (27)$$

This means that the load conductance should equal the negative of the boundary slope at the intersection of the load line and boundary between the active and saturation regions. If the boundary is a linear function, the ac saturation resistance r_s is equal to the dc saturation resistance R_s.

This upper limit for the maximum power output is due to the device boundaries. Since care is taken to operate just to the boundary between the active and saturation regions, a given voltage is made unavailable for appearing across the load. This is equivalent to a generator dropping the unavailable voltage internally. In general, the device output impedance and saturation resistance differ. They may be considered the same for a triode tube and different for a pentode tube or junction transistor.

It must be remembered that the volt-ampere plane is partially bounded. All real devices have additional constraints placed upon them which completely bound the plane. For various practical reasons most devices have ratings of maximum voltage and maximum current. In transistors, the voltage is restricted by a breakdown value. The current is restricted by beta crowding in high current regions which causes distortion.

[2] K. A. Macfadyen, "Modifications of the push-pull output stage Part I," *Wireless Engr.*, vol. 12, p. 532; October, 1935.

In addition, as the load resistance approaches the saturation resistance, the dissipation increases. The highest device dissipation may prevent achieving the greatest value for maximum power output; or if it can be reached, the highest operating ambient temperature is greatly limited by the high dissipation.

Design Results

The results of a design using imperfect devices are shown in Fig. 6. The imperfect devices, which are type 2N389 silicon transistors, are called imperfect mainly due to a nonzero saturation resistance. Each transistor is capable of dissipating 85 watts at room temperature.

$$\frac{R_L}{R_s} = \frac{20}{4.5} = \frac{4.5}{1} \left(\begin{array}{l} R_s \text{ measured with a transistor} \\ \text{curved tracer} \end{array}\right). \qquad (28)$$

The maximum power output was 50 watts.

$P_{d_{(two)}}$ measured 31 watts.

From (15):

$$\text{Max}\left[\frac{P_{d_{(two)}}}{P_{0_{\max}}}\right] = \left[\frac{2}{\pi}\right]\left[\frac{\left(\frac{R_L}{R_s}\right) + 1}{\left(\frac{R_L}{R_s}\right)}\right]^2 \qquad (29)$$

since

$$1.76 \le 4.5 < \infty$$

then

$$\left.\frac{31 \text{ watts}}{50 \text{ watts}}\right]_{\text{measured}} \doteqdot 0.405\left[\frac{4.5 + 1}{4.5}\right]^2_{\text{calculated}} \qquad (30)$$

$$0.65\Big]_{\text{measured}} \cong 0.609_{\text{calculated}}$$

The agreement at the maxima is reasonable.

From Fig. 6 it is evident that some deviation exists

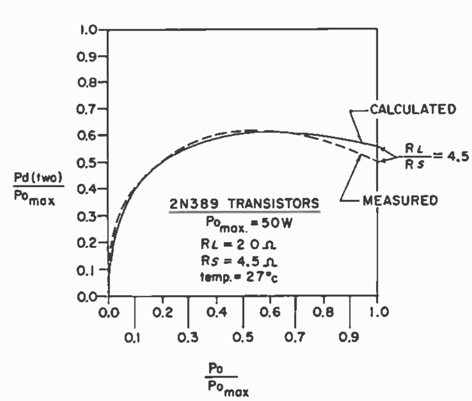

Fig. 6—Calculated and measured normalized power dissipation.

at high power output. This is due to distortion. The theory is a linear one and assumes that the load has a pure sinusoidal current flowing through it. As would be expected, the signal deviates from a pure sinusoid resulting in a departure from the theory, but at low power levels the theoretical and measured curves agree since low distortion is encountered.

Conclusion

Present aero-space programs demand that Class B amplifiers not be over-designed due to weight considerations. In addition, the cost of output transistors generally increases with maximum allowable power dissipation at a given temperature.

The various theoretical curves and other data agree well enough with measured information to justify their use for practical purpose.

Appendix

Symbols

E_c—Collector-to-emitter voltage, volts.
E_{cc}—Collector supply voltage, volts.
$E_{0_{max}}$—Maximum output voltage, peak volts.
E_{sat}—Saturation voltage, volts.
$I_{c_{max}}$—Maximum collector current, peak amps. } Same
$I_{0_{max}}$—Maximum output current, peak amps. }
I_c—Collector current, peak amps. } Same
I_0—Output current, peak amps. }
Max[$P_{0_{max}}$]—Greatest value for maximum power output, watts.
Max[$I_{c_{max}}$]—Greatest value for maximum collector current, peak amps.
Max[$P_{d(two)}$—Maximum power dissipation for both transistors, watts.
η—Conversion efficiency, ratio of power output to power input.
$P_{0_{max}}$—Maximum power output, watts.
$P_{d(two)}$—Power dissipation for both output transistors, watts.
P_i—Power input from dc supply, watts.
R_L—Load resistance, ohms.
R_s—DC saturation resistance, ohms.
r_S—AC saturation resistance, ohms.

A Stereophonic Ceramic Pickup Cartridge for Two-Gram Tracking*

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Summary—The pickup described will track with a vertical force of 2 grams, on simple but properly designed tone arms. It has a mechanical compliance 6 cm × 10^{-6}/dyne and an effective mass of about 1.4 mg. Output is 0.35 volt. A combination of optimum rubber properties and viscous damping provide a response free of resonances.

The design is well suited for quantity production for the home phonograph market. Another paper describes successful tracking of this cartridge in commercially available record changers.

In a pickup destined for quantity production, several design factors must be met in order to maintain uniform high performance. It is important that the two ceramic elements be correctly positioned during assembly. In the present design this is done by means of octagonal rubber element mounts which are placed into two semioctagonal channels on mating interfaces of two plastic housing parts.

In any two-element stereophonic pickup design, a "coupler" is needed to join the two elements to a single stylus arm. Design of the coupler has important bearing on distortion, output level, channel separation, and mechanical impedance. Coupler requirements in the present design have been met by proper configuration and choice of material.

Another important design factor is the means provided for pivoting the stylus level arm. For lateral motion, at the stylus, compliance must be high for lateral motion in two degrees of freedom. Rotational compliance, about the lever axis, must be low in order to avoid undesirable rotation of the lever arm. There must be no lateral freedom at the pivot point. Longitudinal compliance of the level arm must be low in the direction of drag from the record groove to avoid frequency modulation effects.

The stylus lever mounting arrangement described achieves these design requirements in a simple assembly. A light tube, comprising the lever arm, is flattened at its pivoted end in a manner to produce tapered shoulders. These shoulders are brought to bear against a hole in a small rectangular metal enclosure which is filled with a plastic material having viscoelastic properties. The potting arrangement damps any lever resonances and provides a small restoring force to position the lever for engaging the coupler during change of styli.

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Introduction

DESIGN requirements for phonograph pickups are well known. To restate them briefly, a pickup must convert mechanical energy imparted to a stylus from the record groove into corresponding electrical energy. Mechanical impedance at the stylus point must be as low as possible in order to minimize the perpendicular tracking force needed to play the record and to reduce the consequent wear of stylus and groove. The electrical output must be as free as possible of any form of distortion of the groove information. To a stereophonic system one must add another important requirement: the pickup must respond independently to signals of the two stereophonic channels. Ideally, each pair of pickup output terminals should contain signals derived from its associated groove channel only, and not from the other. The ratio of magnitudes of signals derived from successive equal modulation of the wanted and unwanted channels is called "channel separation," and is commonly expressed in decibels.

It is historically interesting and significant that the tracking force started at several ounces, or over 100 grams, in the early days of phonograph reproduction and has now dropped to about 6 to 9 grams in commercially built home phonographs. The more costly pickups currently used by high fidelity enthusiasts track at about 2 grams.

This paper describes a high fidelity stereophonic ceramic pickup cartridge which is capable of 2 grams operation on commercial record changers of good design and is intended primarily for the home phonograph field. In addition to meeting exacting performance standards, this cartridge produces sufficient output voltage for use with home phonograph amplifiers. An accompanying paper describes a tone arm of unique design, which protects records and overcomes problems attending tripping of record changers at 2-gram tracking force.

Design Principles

Many transducer systems are available to the pickup designer, and much has been written about their relative merits. Pickup design is essentially a mechanical problem. It may be said that the best transducer system is the one whose design leads to the closest fulfillment of the various important performance requirements. The piezoelectric transducer has long emerged as the best system for pickups intended for home phonographs because of its high output and convenient electrical equalization. The pickup to be described employs piezoelectric ceramic elements of the lead zirconium titanate type.

A ceramic element has a high mechanical impedance which must be converted to the required low impedance at the stylus point. Ordinarily this is done by mounting the element in elastic supports, by compliant connection to the stylus, or by a mechanical lever between stylus and ceramic element. Any of these may be used singly or in combination. The lever is to be preferred because it is a mechanical transformer; that is, it lowers mechanical impedance as the square of the lever ratio, while electrical output decreases as the first power. However, high-order lever ratios are difficult to design and control in manufacture. Practical limits are ratios of about 3 or 4 to 1.

Most ceramic stereophonic pickups use two transducer elements, one for each of the two program channels. Since there is but one stylus, some form of hinge or coupler is needed to connect each ceramic element to the stylus for independent motion. Ideally, if a coupler transmits each channel information to its respective transducer element only, separation will be infinite. In some stereophonic pickups, separation decreases with increasing frequency and sometimes is reduced to zero or even takes on negative values. At low frequencies, channel separation is determined by the geometry of the mechanism. At high frequencies, slight unbalances caused by lack of symmetry become important. In the pickup to be described, excellent separation is maintained throughout the audio-frequency spectrum.

Design of the Pickup

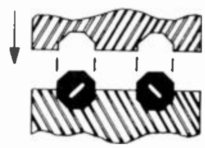
Mechanical impedance transformation from ceramic elements to stylus is accomplished by a combination of compliant element mounts, a compliant coupler and a lever between stylus and coupler. Emphasis has been placed on a design that would permit precise placement of elements using modern line assembly techniques. Ceramic element mounts are octagonal in cross section, as shown in Fig. 1(a). The octagon configuration results in a unique combination of angles. Ceramic elements are placed parallel to a pair of sides, perpendicular to another pair, and 45° to the remaining four sides. Both mounts are clamped between two plastic shells containing complementary semi-octagonal grooves with small interference fit between grooves and mounts. This provides the required 45° orientation. Another advantage of the octagon shape is that two sides of the mount are perpendicular to the shell interfaces and therefore are aligned with the direction of assembly. Once the parts are correctly assembled by means of an assembly fixture, there is little likelihood that any distortion of element position will occur. This arrangement is depicted in perspective by Fig. 1(b). The material for the mounts possesses some viscoelastic properties which provide partial damping of the system.

During development much attention was directed to the coupler. Several materials and some novel configurations were evaluated. The final design is the configuration shown in Fig 2. The material is a viscoelastic formulation similar to the one used for the element mounts. Two slightly undersized cavities are provided in the coupler for engaging the free ends of the ceramic elements. Linkages from each element converge to form a yoke which will engage the stylus lever arm. The coupler is fairly compliant and contributes. to the final compliance of the pickup. The stylus lever arm is placed at the intersection of the two perpendiculars to the elements, as shown. Transverse deflections of the linkages cause torsional forces at the elements, which do not generate any output. Compliance of the linkages is relatively high in the transverse direction, thus further reducing any undesired "cross-talk" forces at the nondriven element.

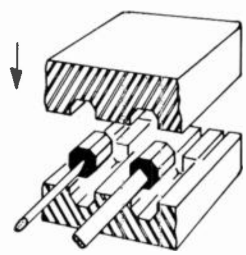
The requirements for a stylus lever arm in a stereophonic pickup are described with the aid of the diagram in Fig. 3. The lever is pivoted at point A, carries a stylus which engages the groove at point B, and engages the coupler at point C. The force transformation ratio is (l_2/l_1), and the impedance transformation ratio is $(l_2/l_1)^2$.

The lever must perform the following functions:

1) Any motions of the stylus in the surface b must be translated into equivalent motion at the surface c.

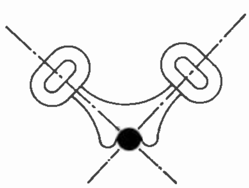


(a)

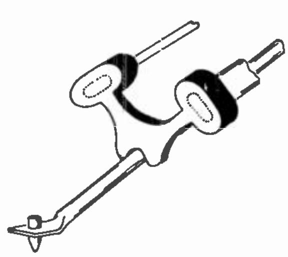


(b)

Fig. 1—Mounting arrangement for ceramic elements.



(a)



(b)

Fig. 2—Coupler configuration.

2) The stylus extends a distance from the lever arm to provide clearance between the lever and the record. This creates a moment arm that tends to produce rotational motion around the axis A-B, instead of the desired translational motion at C. For this reason, the lever itself must be axially stiff and the pivot at A must prevent rotational motions.

3) The frictional drag between the stylus and the groove creates a force pulling at the pivot. This drag varies with groove modulation so that there is a tendency to produce an effect known as longitudinal distortion if the lever is permitted to move. Therefore, the pivot at A must prevent motions in the longitudinal direction.

Various attempts to solve these multiple problems led to the design depicted by Fig. 4. Two stylus levers are provided back-to-back in a turret arrangement which is rotated 180° to change styli. A turret pivot post is attached to a formed metal part comprising a small box and two

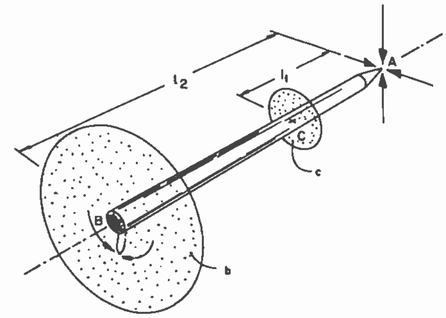


Fig. 3—Diagram to depict stylus lever design requirements.

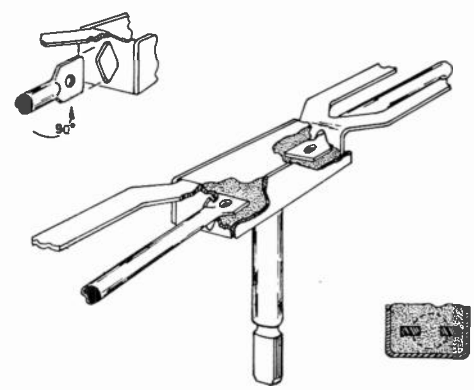


Fig. 4—Details of the turret stylus assembly and lever arms.

extensions that surround the lever arms to afford some protection. Both ends of the box are pierced with "keyhole" openings as shown. Lever arms are made of thin walled aluminum alloy tubing, one end of which is flattened and pierced to receive a stylus. The other end is also flattened in the form of a spade with flaring shoulders between tube and spade.

In manufacture, the turret is placed in a produciton fixture. The lever arms are inserted in their respective openings by aligning the spades and keyholes as shown. Both tubes are then rotated 90° to make them captive. In the production fixture, means are provided for holding the lever arms in place. A small spring force (in the fixture) pulls the lever arms outwardly so that the flaring shoulders are in contact with the inner walls of the box. As the next operation, the box is filled completely with a vinyl dispersion in a liquid plasticizer. At this stage, the material is a fluid of medium viscosity. After a short period at an elevated temperature, the compound fuses to a soft but durable elastomer. A small hole is provided through the spade end. This becomes filled with the elastomeric compound and provides a large increase in rotational stiffness about the lever axis without decreasing the transverse compliance at the stylus. The entire double stylus assembly is now locked together in a position determined by the production fixture.

Fig. 5 depicts the cartridge in vertical cross section. The octagonal channels extend beyond the rubber element mounts and terminate in a larger opening that surrounds the coupler. The pickup is only partially damped as it has

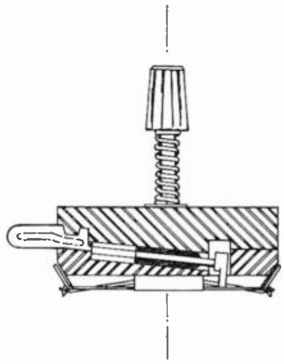


Fig. 5—Vertical cross section of the assembled cartridge.

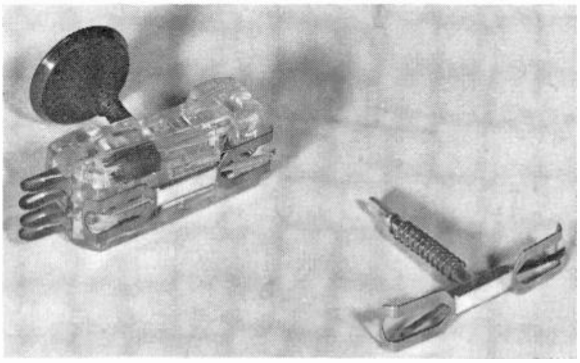


Fig. 6—Photograph of the cartridge and turret stylus assembly.

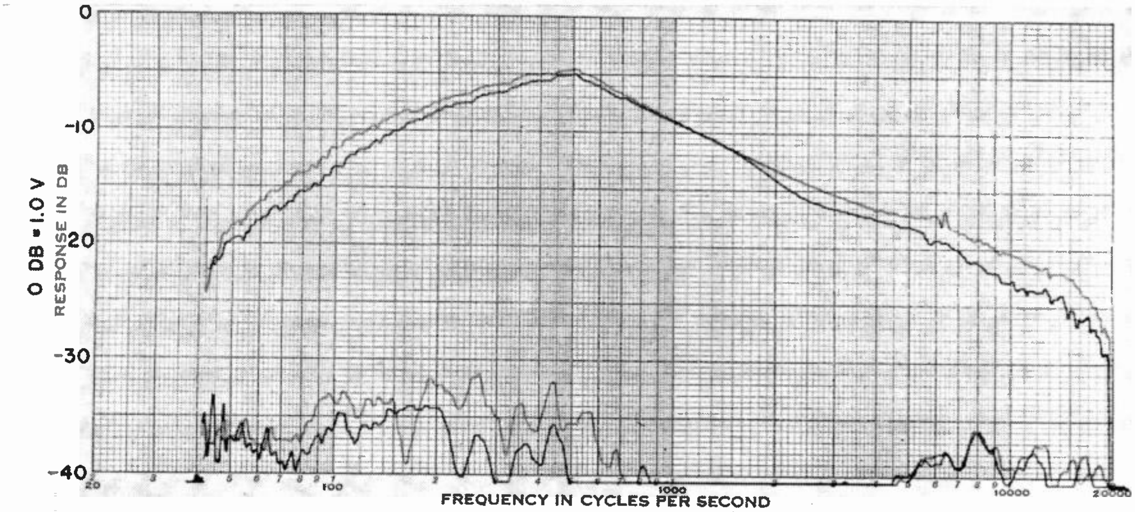


Fig. 7—Frequency response and separation, left and right channels. Output—0.35 volts/3.54 rms velocity, a 1 kc. Compliance—6×10^{-6} cm/dyne. Effective mass—1.5 mg. CBS Laboratories test record STR-100: shunt capacitance—125 mmf; resistive load—1 megohm; tracking force—2 grams.

been described to this point. Complete damping is obtained by injecting a measured amount of stable noncreeping grease around the elements and within the octagonal channels.

Fig. 6 is a photograph of the complete pickup with a detached turret assembly shown below the cartridge. Replacement of the stylus assembly is a simple fingertip operation without tools.

Fig. 7 depicts frequency response and separation for both left and right channels. These characteristics were automatically plotted using the CBS Laboratories STR-100 stereophonic test record in conjunction with a General Radio type 1521-A graphic level recorder. From low frequencies to 1 kc, pickup response is flat after correcting for the load resistance used in the measurements. Above 1 kc, response generally follows the constant velocity characteristic of the record with a slow and gradual rise to about 15 kc after which there is a slight roll-off to 20 kc. Some values of separation taken from the curves are: 25 db at 100 cps, 35 db at 1 kc, 19 db at 10 kc, 18 db at 15 kc, and 14 db at 20 kc. Compliance is 6×10^{-6} cm/dyne, effective mass at the stylus is 1.5 mg. Output is 0.35 volt at 1 kc for an RMS velocity of 3.54 cm/second.

Acknowledgment

These cartridges are currently being manufactured by the Zenith Radio Corporation where they will be used with the Zenith "Micro Touch" tone arm. The authors are indebted to D. Knight, Chief Audio Engineer, and other members of the Zenith staff, for their participation in the production engineering stages of this project. The photographs and performance data given herein are based on production cartridges supplied by Zenith.

The authors also acknowledge early design contributions by D. P. Doncaster and G. W. Sioles, of CBS Laboratories, and J. Van Leer, formerly of the Laboratories.

A Stereo Tone Arm for Tracking at Two Grams on a Record Changer*

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Summary—**A tone arm, capable of tracking phonograph records at 2 grams, for use in record changers is described. The low tracking force is accomplished by introducing an additional degree of freedom in the arm. An equivalent circuit analysis of the new design demonstrates its greater stability in comparison with conventional systems. Such matters as alignment of pivots for minimum frequency modulation and reduced damage to records resulting from accidental scraping are discussed.**

Introduction

It HAS LONG been recognized that very low tracking forces in phonograph pickups will minimize wear of the record and styli. Another advantage of low tracking force is the reduction in wavelength loss. Two factors tend to place a limit on the minimum tracking force which can be used. One is the mechanical impedance of the pickup and the other is the stability of the pickup-tone arm system when subjected to jarring. A tracking force of approximately 2 grams is usual in high-priced manual players which are handled carefully by relatively skilled people. However, in most home phonographs tracking force is found to be above 6 grams to insure stability when subjected to vibration. This paper describes a tone arm configuration which was developed to allow tracking at 2 grams in automatic record changers with a low mechanical impedance pickup described in a companion paper.[1] As an additional benefit the new tone arm protects records by preventing damage due to accidental scratching or dropping of the tone arm.

Description of Tone Arm

Fig 1(a) shows a schematic representation of the tone arm design. The pickup is held in a carriage which is pivoted about a horizontal axis near the stylus tip. A trailing spring is attached to the overhung portion of this carriage and provides the force required to take up the net unbalanced force at the stylus tip, as well as isolate the cartridge from the tone arm. The vertical pivot for the arm is located behind the horizontal pivot and a weight is placed at the rearmost part of the arm to statistically balance it in the horizontal plane. A second spring, disposed as shown in the sketch, is required to reduce the unbalanced vertical force at the stylus tip to 2 grams.

* Received April 20, 1962. Presented at the IRE International Convention, New York, N. Y., March 28, 1962.
† CBS Laboratories, Stamford, Conn.

[1] A. L. DiMattia, E. Kaulins, and B. B. Bauer, "A Stereophonic ceramic pickup cartridge for two-gram tracking," this issue, pp. 145–148.

Equivalent Circuit Representation of the Mechanical System

Fig. 1(b) shows the analogous circuit of the mechanical constants of the tone arm for the vertical plane. The first inductance represents the mass of the centridge, the second the mass of the tone arm; both of these masses being referred to the stylus tip (all rotational mechanical reactances are referred to the stylus tip as either compliances or masses). Three capacitances are shown; the first is analogous to the compliance of the stylus, the second represents the compliance of the first spring, and the third, the compliance of the second spring. The mechanical impedance plot for the circuit is shown in Fig. 1(c) on the assumption that the mechanical resistances are small. It may be noted that there are two resonances and two antiresonances.

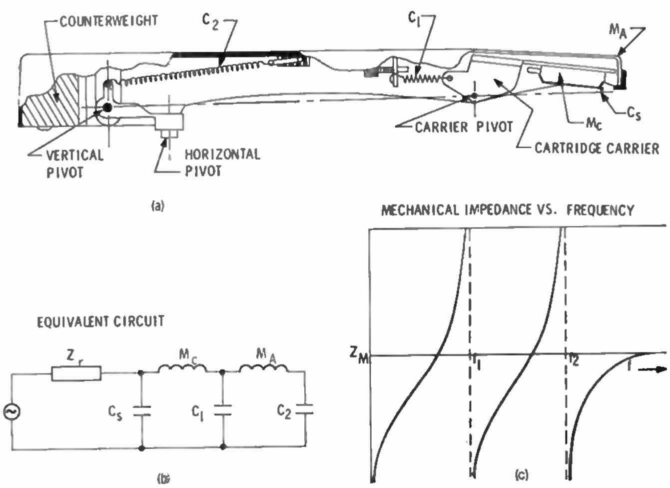


Fig. 1—(a) Schematic representation of tone arm. (b) Equivalent circuit. (c) Impedance plot vs frequency.

It is interesting to compare this impedance characteristic with that of the conventional arm with respect to stability considerations. In a conventional arm there would be only one antiresonance frequency, occurring at an intermediate frequency to the ones shown, defined by the stylus compliance and the arm mass. In the arm under consideration, the antiresonance occurring at the higher frequency is caused by the compliance of the stylus and the mass of the cartridge and its carrier, the latter modified by the other elements in the circuit. Since the compliance of the first spring in the equivalent circuit is much higher than the stylus compliance, and the mass of the arm is much larger than the mass of the cartridge and its carrier, the remaining part of the circuit is effectively uncoupled. Thus, for all practical purposes, the upper impedance maximum is a result of the stylus compliance with the mass of the cartridge and carrier. The impedance at the antiresonance frequency is equal to the mass divided by the product of the compliance and resistance, for small values of resistance. Because of the low mass of the cartridge plus carrier the impedance at this frequency is considerably lower than it would be otherwise. Velocities arising from jarring of the equipment produce smaller forces at the stylus tip than would otherwise be the case with an attendant increase in tracking stability. Naturally, the frequency response would be affected if this antiresonance were allowed to occur in the desired transmission range and a high compliance stylus is needed to produce a satisfactorily low antiresonance with the small allowed mass.

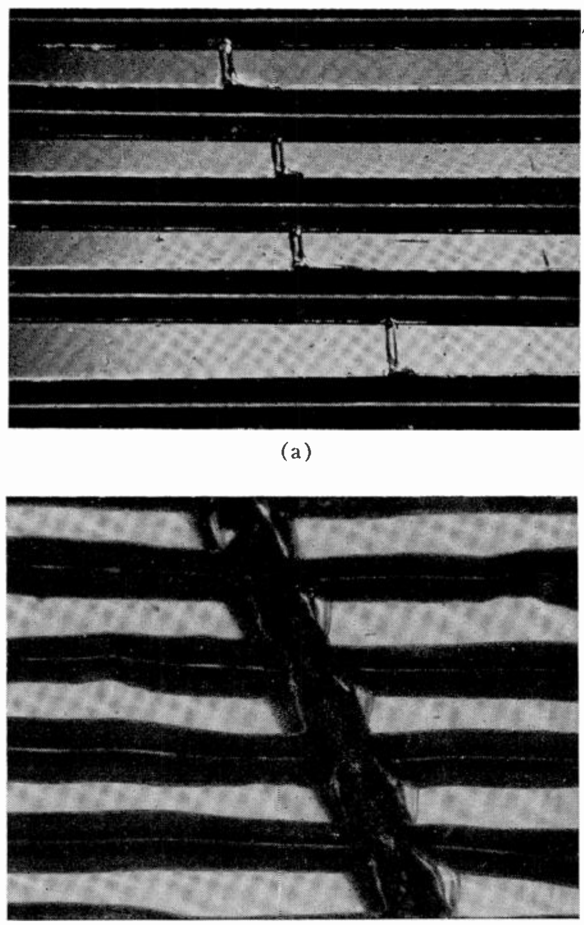
We now consider the lower maximum in the impedance which arises from the parallel resonance of the trailing spring compliance with the tone arm mass, whose reactance is reduced slightly by the compliance of the second spring. The compliance of the second spring is greater than that of the first, so that the antiresonance we are observing is effectively that between the mass of the tone arm and the compliance of the trailing spring. This antiresonance is considerably lower in frequency than that defined by the stylus compliance and the arm mass because, as previously noted, the first spring has a higher compliance than the stylus compliance. The impedance at this antiresonance is inversely proportional to the compliance and with unchanged arm mass, the impedance at antiresonance is lower than would normally be the case in accordance with the relation noted earlier. In short, by replacing the single impedance maximum of the conventional tone arm by two maxima with lower mechanical impedance, the force tending to throw the stylus out of the groove when any part of the system is subjected to impulsive velocities has been reduced. It will be noted that whereas the impedance tends towards zero at the higher frequencies it approaches infinity as the frequency goes to zero. This might raise the question as to whether the system is truly stable when subjected to shocks. The answer to this is that the reactance depicted is extremely small and does not become large until a very low frequency—in fact so low that the expected spectrum of disturbing amplitudes lies above the range where forces would be developed sufficient to cause mistracking. This has been verified by experience.

Alignment of Pivots

With the cartridge pivot so close to the stylus, the playing of a warped record will result in frequency modulation or "wow," which by proper design is made negligible. The problem is not severe for a gentle warp since there is relatively little differential motion between the tone arm and the cartridge. This can be seen by viewing the equivalent circuit for the system as comprising the compliance of the trailing spring C_1, in parallel with the mass of the tone arm m_A. For a gently warped record this circuit is excited below the resonance frequency and the tone arm motion is much greater than that of the cartridge. If the warp is more abrupt, approaching an impulse, the equivalent circuit is excited at its resonance frequency and both the cartridge and tone arm will be displaced in proportion to the relative resistance in each branch. The location of the cartridge pivot in relation to the stylus, and the angle the line joining these points makes with the record become important if "wow" is to be prevented. This is clear from the geometry. Since the stylus moves on an arc, the component of motion in the direction of the groove is increased when the pivots are not on a line parallel to the record. In practice, a slight downward angle is used so that the pivot will clear the record.

Scratchless Feature

The method of elastically mounting the cartridge produces a performance advantage in regards to protection of records and fragile stylus tips. In a conventional arm, an accidental downward thrust on the arm produces displacement of the stylus and a resultant force proportional to this displacement which may be sufficient to plastically deform the record material. Frequently such accidents produce a lateral motion of the arm and result in scratching of the record; and in more unfortunate cases the stylus tip



(a)

(b)

Fig. 2—(a) Photomicrograph of a disc scratched by the described arm compared (b) with a conventional tone arm.

is broken. Because the compliance of the elastic mounting is larger than that of the stylus, the force between stylus and record is lessened for such occurrences with a corresponding reduction in the amount of damage done. In the design described the maximum force which can exist between the stylus and the record is approximately 5 grams, at which point the cartridge has retracted sufficiently into the arm to no longer be vulnerable.

An example of the relative damage to a record with this arm compared with conventional designs can be seen in Fig. 2. These photomicrographs show the width of groove embossed into the disc when the arm was deliberately caused to "skate" across the record.

ACKNOWLEDGMENT

The authors express their thanks to I. Delin, A. L. Di Mattia and D. Doncaster for their technical contributions in the development of the device.

Correspondence

Comments on "Musical Transfer Functions and Processed Music"*

May I refer to two processing techniques to supplement my discussion of processed music:[1]

Processing Stereo from a Single-Channel Recording

Complex filtering techniques are now in use.[2] The aim is reconstruction of nontransduced stereophonic information. Provided the placement of the musical instruments in the original sound field is known, partial retrieval of stereophonic information is compatible with the theorems of information theory. If the objective is simulation of stereophonic liveness rather than authentic reconstruction of the original sound field, even a simple-channel split with separate tonal controls may evoke remarkable stereophonic illusion.

Processing Music for the Stage

For better theatrical effects, especially in the opera, the performances of orchestra, vocal soloists and choir may be processed to simulate different localizations and to emphasize dynamics.[3]

Finally, I wish to supplement my considerations concerning *automatic performance scoring*. Analog techniques are confined to the time-intensity domain. For scoring performers' intonational standards in the melodic[4] or harmonic[5] musical contexts, digital techniques are required.

Scoring of a simple melodic sequence of tones suggests the following scheme:

Preparatory Step: The theoretically correct frequency of each tone is translated into the binary number system and punched into a separate card (standard deck).

First Step: Preparation of a tape recording of the performance at fast speed.

Second Step: Playback of the tape at a low speed. An analog-to-digital converter translates into the binary language the fundamental frequency in real time of each performed tone. Each tone is represented by a separate punched card (experimental deck).

Third Step: The digital computer is programmed to perform the following tasks on the basis of the two decks:

1) Computation of the differences between performed and theoretical frequencies; tabulation of these differences in the temporal sequence of the tones;
2) Computation of the average difference and its standard error;
3) Computation of average differences and standard errors relative to subsets of cards grouped for the various musical intervals.

Fourth Step: Conversion of the results from the binary system into logarithmic cents.[6]

It is of interest to recall that from digitally punched tapes, the RCA music synthesizer can record and reproduce music.[7] Accordingly, there is a theoretically unlimited reversible process between performance of music and its scoring.

Drifting of the musical tempo vs metronomic time raises more complex tasks which, as yet, computers may not handle.

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* Received July 18, 1962. At the USN Medical Research Laboratory, Groton, Conn. The opinions expressed in this article are not necessarily the official views of the U.S. Navy.

[1] A. G. Pikler, "Musical transfer functions and processed music," IRE TRANS. ON AUDIO, vol. AU-10, pp. 47–52; March-April, 1962.

[2] J. A Somer, "Toscanini in stereo," *High Fidelity*, pp. 40–41; March, 1961.

[3] D. W. Martin, "Supplementary sound for opera," *J. Acoust. Soc. Am.*, vol. 1, no. 1, pp. 25–33; 1962.

[4] P. C. Greene, "Violin Performance with Reference to Tempered, Natural and Pythagorean Intonation," *Ia. St. Mus. IV*, University of Iowa, Iowa City, pp. 232–251; 1937.

[5] J. F. Nickerson, "Intonation of solo and ensemble performance of the same melody," *J. Acoust. Soc. Am.*, vol. 21, pp. 593–595; September, 1949.

[6] "Stroboconn Tables," C. G. Conn, Ltd., Elkhart, Ind.; 1950.

[7] H. F. Olson and H. Belar, "Electronic music synthesizer," *J. Acoust. Soc. Am.*, vol. 27, pp. 595–612; May, 1955.

Comments on "Early Experiences with Stereotape Recording"*

As comment on "The Editor's Corner,"[1] some quite early experiments on auditory perspective, which I made in 1916 while with the Navy as Expert Radio Aide for Aviation, U.S.N., at Pensacola, Florida, may be of interest.

I was working with sound-powered microphones of the "Baldwin," balanced-armature type, for pilot-observer communication on the open-cockpit seaplanes then in use for pilot training. The same units were used as receivers with, as noise-excluding ear cushions, telescope gunsight *eye* shields as the only thing available. The receivers were mounted in the earflaps of leather helmets.

The noise, of course, was at an extremely high level and came mostly from the unmuffled exhausts of the Curtiss OX-2, 100-h.p. motor, only a few feet distant, as measured by a noisemeter I had contrived, which measured noise in terms of a 1-kc signal audibility necessary for reception through those blanketing noises via unshielded Baldwin headphones.

I soon used *two* such sound-powered microphones in series with the two receivers of the dual headphone used by the student pilot. One of these was for the instructor, the other for neutralizing noises, and located in a paddle-like, short board, adjacent to the talking unit. It was, of course, oppositely phased. This helped a lot, but most for a particular orientation of the paddle, with both microphones equidistant from the engine, for *in*-phase incoming exhaust sounds.

I then eliminated one microphone and removed the back cup of the bakelite shell enclosure to expose the back side of the (mica) diaphragm, leaving the threaded portion for the cap, which also was reduced to a mere ring. This proved a great improvement by keeping back and front side received noise radiation in-phase and equal-level when, especially, the diaphragm was edge-on towards the motor.

Later, to relieve diaphragm overloading due to high-level shouting by the talker, so that he himself could hear what he was saying through these noises, as well as to improve articulation, I replaced the 400-cps diaphragm with a much stiffer one of 2 kc. With this arrangement the reception was really excellent; in fact, with the motor idling, whispered instructions were intelligible. Still later I used a diffraction-back shield to change the directional characteristics from bi- to unilateral, and patented the various arrangements.[2]

While this may be interesting with respect to the history of anti-noise and directional microphones, it is only the setting for the acoustic perspective experiments. Intrigued with the idea of that subject I connected four of these Baldwin receivers in a dual circuit, one microphone for each earphone of the receiver station, the microphones being a foot or so apart. It was then that I experienced that strange sensation of the received sound rotating around from one ear to the other in the *back* of my head, as another person moved away from one microphone to the other, seemingly coming from behind instead of in front of me at varying directions.

Much later, in fact on November 10, 1939, I applied some of these principles to an electronic piano. Two pickup strings (*i.e.*, series) were used, one tapering gradually in spacing from the piano strings from extreme bass to extreme treble. For the other series of pickups the taper was reversed, that is, close at the treble and rather widely spaced at the bass end. The voicing adjustments of this pickup produced uniform, pianistic sound levels throughout the pitch compass. Two separate amplifiers and speakers were used, the speakers being laterally separated by about 5 feet.

With this arrangement the output sounds were spacially separated, the bass tones appearing to come from one side and the treble tones from the other. By separating the speakers much farther the apparent size of the piano, laterally, was much increased. If one were placed *above* the other, the piano, with one's eyes closed, seemed to be standing on one end!

These effects, where loudspeakers instead of individual channels were used, were by no means as pronounced, as Mr. Camras also has observed, and I agree with his conviction that the present ballyhoo over stereophonic radio and phonographs is very much overdone. It is far, very far, from what we hear directly and binaurally from the sounds around us.

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* Received July 18, 1962.

[1] M. Camras, IRE TRANS. ON AUDIO, vol. AU-10, pp. 29–31; March-April, 1962.

[2] B. F. Meissner, Patent No. 1,507,081; September 23, 1949.

Contributors

Leonard Baker (S'52-M'56) was born in Boston, Mass., on June 14, 1931. He received the B.S. degree in electrical engineering from the University of Miami, Coral Gables, Fla., in 1956, and the M.S. degree in electrical engineering from the Moore School, University of Pennsylvania, Philadelphia, in 1959. He is now doing graduate work for the Ph.D. degree at the latter university.

With four years of service in the U.S. Army Signal Corps, he joined the Radio Corporation of America, Camden, N. J., in 1956, as a development engineer in transistorized circuitry. Typical projects in which he was engaged included the development of gates, flip-flops, differentiators, and pulse shapers; data reconstitution circuits; an audio amplifier for operation at 150°C; and a nuclear reactor safety amplifier. Three years later, in 1959, he became associated with the National Company in Malden, Mass., where he developed transistorized circuitry for atomic beam clocks. In 1960 he joined the Tele-Dynamics Division of American Bosch Arma Corporation, Philadelphia, Pa., as Project Engineer and has since been responsible for the design and development of phase-shift and multivibrator subcarrier oscillators for telemetry systems.

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In 1957, he became a civilian research assistant for the U. S. Army Signal Corps Laboratories, Fort Monmouth, N.J., where he did research in semiconductors physics including studies of injection in reduced-area contacts and lifetime measurement. In 1959 he joined the staff of Olympic Radio and Television, Long Island City, N. Y., as a Research Engineer. In their Military Research Laboratories he investigated semiconductor gyrators, tunnel-diode logic devices including a one-tunnel-diode flip-flop, crystal-controlled tunnel-diode oscillators, and various low-noise solid-state frequency multipliers. He also assisted in the development of a cloud height radar and Project Tall Tom and has served as a consultant on varactor devices.

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Benjamin B. Bauer (S'37-A'39-SM'44-F'53) was born in Odessa, Russia, on June 26, 1913. He received the B.S. degree in industrial electrical engineering from Pratt Institute, Brooklyn, N. Y., in 1932, and the E.E. degree from the University of Cincinnati, Ohio, in 1937.

In 1936 he joined Shure Brothers, Inc., Chicago, Ill., manufacturers of microphones and electronic components, becoming Chief Engineer in 1940 and Vice President in 1950. In 1957 he joined CBS Laboratories, Stamford, Conn., as Head of Audio and Acoustics. In 1958 he became Vice President of the Laboratories, in charge of the Acoustics, Magnetics, and Instrumentation Research Departments. He has the over-all responsibility for research and development in the areas of acoustics and electroacoustical transducers, magnetic transducers and media, electronic and mechanical instrumentation, and specialized audio products.

Mr. Bauer is a Fellow of the Acoustical Society of America and the Audio Engineering Society and a member of the Society of Photographic Engineers and Scientists. He is a founder and past National Chairman of the IRE Professional Group on Audio, former Editor-in-Chief of the IRE TRANSACTIONS ON AUDIO, and a 1955 recipient of the IRE-PGA "Achievement Award." He is a member of Tau Beta Pi, Sigma Xi, and Eta Kappa Nu, and was, in 1961, awarded the Eta Kappa Nu "Award of Merit."

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From June to December of 1958 he was employed by the Polarad Electronic Company, New York, N. Y., where he served as a Reliability and Components Engineer for missile checkout equipment. In 1959 he joined the staff of Olympic Radio and Television, Long Island City, N. Y., where he has served since as Research Engineer and Project Engineer. He developed a hygrometer using the Peltier effect, developed and standardized a transistor nor logic system, and did research and development on Sensors for the AN/FMQ-5 Automatic Weather Station. He has also been Project Engineer on a digital barometer and on a high-precision frequency comparator.

Alfred L. DiMattia (A'45) was born in Boston, Mass., on May 10, 1910. He studied at the University of Massachusetts Extension.

In 1942 he entered the U. S. Government's Electroacoustic Laboratory at Harvard University, Cambridge, Mass., where his work included design and development of headphones, noise-cancelling and other types pf microphones, and transducers for sound-powered communications. From 1945 to 1947, he was Chief Engineer with the William J. Murdock Company, Chelsea, Mass., and from 1947 to 1959, Senior Development Engineer with the Dictaphone Corporation, Bridgeport, Conn. In 1959 he joined CBS Laboratories, Stamford, Conn. As Section Head, Acoustical Transducers, his efforts have been in investigation and development of military pressure and noise-cancelling microphones, laboratory standard microphones, vibration detection transducers, and environmental testing techniques and apparatus for specialized transducers. As consultant in medical electronics in the fields of heartbeat transducers and phonocardiography, he has been affiliated with Hartford Hospital, Hartford, Conn., and the Sanborn Company, Waltham, Mass. He is a Registered Professional Engineer in the state of Connecticut.

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Abraham A. Goldberg (M'53-SM'57) was born in New York, N. Y., on March 21, 1919. He majored in physics at Columbia University, New York, N. Y.

During World War II he trained in radar technology at the Navy Radio Material School, later becoming an instructor there. He has been a staff member of CBS Laboratories, Stamford, Conn., since 1946. Among his wide range of activities at the Laboratories, he has been responsible for the design of color television systems and studio equipment, video and audio tape recording systems, ultrasonic control devices, and broadcast network signaling systems. In the position of Branch Manager, he directs the work being undertaken in research and development in Instrumentation, Audio Circuitry, and Electromechanisms. He has lectured in diverse fields of his interest such as color television, stereophonic AM transmission and reception, magnetic tape performance analysis, and magnetic recording systems.

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Henry S. Katzenstein was born in Shreveport, La., on January 9, 1927. He received the B.A. and M.S. degrees from the University of Connecticut, Storrs, in 1952, and the Ph.D. degree in physics from the University of Connecticut in 1954.

From 1948 to 1950 he served as a Research Technician at the Institute of Radiobiology and Biophysics of the University of Chicago, Ill., where he designed and supervised the construction of experimental apparatus. He worked as a Research Associate for the University of Connecticut in connection with experimental work on Time-of-Flight Neutron Spectrometry Mass Spectroscopy, Self-Quenching Geiger-Mueller Counters, and atmospheric measurements using a pulsed searchlight. From 1954 to 1957 he did original work on classified projects in pulsed Doppler surveillance and fire control radar systems for the M.I.T. Lincoln Laboratories, Lexington, Mass., and experimental investigation of line-of-sight propagation effects. In 1957 he joined the staff of Olympic Radio and Television, Long Island City, N.Y., as Director of Research, where he originated research in ultrasonics and semiconductor devices and applications and supervised projects including a cloud height radar and various data-handling systems. In 1961 he became Director of Engineering for Olympic Radio and Television. In 1962 he joined Solid State Radiations, Inc., Los Angeles, Calif., as President.

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Edward Kaulins was born in Riga, Latvia, on November 17, 1925. He attended the Technical Institute of Riga, Latvia in 1943, and the Bridgeport Engineering Institute, Stamford, Conn., from 1956 to 1958.

From 1950 to 1952 he was with the Electric Indicator Company, Springdale, Conn., and from 1952 to 1958 with Ralph W. Biggs Company, Stamford, Conn., a subsidiary of Hamilton Watch Company, working on the design, fabrication, and assembly of miniature components and structures. He has been a member of the staff of CBS Laboratories, Stamford, Conn., since 1948. In the capacity of designer, his work has been in the development of specialized transducers, including phonographic pickup cartridges, microphones, bio-medical sensors, and other electroacoustical devices.

George W. Sioles (M'52) was born in Peekskill, N. Y., on October 15, 1924. He received the B.S. degree from Columbia University, New York, N. Y., in 1949, and has taken graduate work there and at New York University, N. Y.

He has a broad background in engineering and electroacoustics, with specialized experience in theoretical acoustics, psycoacoustics and loudspeakers. He was Project Engineer for navigation training devices with the U. S. Naval Training Devices Center, Port Washington, N. Y. 1950 to 1953. From 1953 to 1959, he was with University Loudspeakers, Inc., White Plains, N. Y. As Engineering Manager, he was responsible for that organization's research, development, design, and production. He joined CBS Laboratories, Stamford, Conn., in 1959. As Section Head, Acoustics Research, his responsibilities include the technical direction of theoretical and applied work in psychoacoustics, stereophonic recording and reproduction, studies of acoustics of bounded space, and short-wavelength disk recording.

Mr. Sioles is a member of the Acoustical Society of America and the Audio Engineering Society.