IRE Transactions on AUDIO

Volume AU-5 JANUARY-FEBRUARY, 1957 Number 1

Published Bi-Monthly

•

•

TABLE OF CONTENTS

PGA NEWS

CONTRIBUTIONS

PUBLISHED BY THE

Professional Group on Audio

IRE PROFESSIONAL GROUP ON AUDIO

The Professional Group on Audio is an organization, within the framework of the IRE, of members with principal professional interest in Audio Technology. All members of the IRE are eligible for membership in the Group and will receive all Group publications upon payment of an annual assessment of $2.00.

IRE TRANSACTIONS® on AUDIO

Published by the Institute of Radio Engineers, Inc., for the Professional Group on Audio at 1 East 79th Street, New York 21, New York. Responsibility for the contents rests upon the authors, and not upon the IRE, the Group, or its members. Individual copies available for sale to IRE-PGA members at $0.45; to IRE members at $0.65, and to nonmembers at $1.35.

PGA News

PGA CHAPTER ACTIVITIES

Albuquerque, N. M.

A. D. Pepmueller reports that plans for coming meetings include the demonstration of sound equipment by various manufacturers, enabling members to determine which equipment is best suited to individual needs. The speakers will deliver general information on new improved sound equipment and specific information on technical aspects of interest to audio engineers.

Baltimore, Md.

A petition for the formation of a PGA Chapter in Baltimore was approved by the IRE Executive Committee. An ad hoc committee has been appointed consisting of Kenneth W. Betch, John L. Markwalter, and George N. Webb. Mr. Betch is serving as acting chairman until elections are held in June to coincide with the sections' elections.

Boston, Mass.

Roger H. Praeger, M.I.T., Cambridge 39, Mass. is the 1956–1957 Chairman of the Boston Chapter PGA. Please modify your list of Chapter Chairmen accordingly.

Dayton, Ohio

The following is an abstract of the paper, "Needed: Uniform Standards in the Acoustics Industry," presented by Louis S. Hoodwin at the December meeting of the Dayton Section. Mr. Hoodwin is Chief Engineer, Loudspeakers, Electro-Voice, Inc., Buchanan, Mich.

"What is high fidelity reproduction? How should the performance of a loudspeaker be measured? How should a loudspeaker be designed to obtain a desired performance? According to much of the information published by equipment manufacturers, there is a great diversity of opinion as to the answers to these questions. The answers often vary according to the special interests of those who are responding without regard to physical laws or ethics.

"In order to have a standard by which to judge objectively the performance of high-fidelity systems, three characteristics of reproduced sound should be taken into consideration. They are frequency response at the position of the listener, distortion, and spatial configuration.

Weakest Link

"Of these, the most important to the conventional present day hi-fi system is frequency response; and since the loudspeaker is the weakest link in the reproducing chain, the frequency response of the loudspeaker is of the utmost importance. According to much of the present advertising literature, this problem has been solved, since it is implied that there are numerous systems available which will provide flat frequency response from pedal organ tones to beyond the upper limits of audibility. Since most of these 'perfect' systems sound entirely different, the general listening public can only conclude that either someone is publishing misleading information or that there is something mystical about the reproduction of sound which cannot be explained.

"The third requisite, proper spatial configuration, is being given intensive study in binaural and stereophonic systems. The usual published frequency response data on loudspeakers shows only the response on the axis and does not mention the polar distribution of the sound. This data cannot be used to predict the frequency at the position of the listener unless the listener is located on the axis. For most loudspeaker systems, this is an unusual listening location. Many of the frequency response claims can only be met if response that is 15 db below the average of the response curve is considered significant.

"Some of the greatest abuses of physical laws are found in the claims made for loudspeaker enclosures. In this field, it appears as though anything can be claimed as long as the public is gullible enough to believe it.

Standards Vary

"What is the answer to these problems? Objective standards of performance should be applied to the acoustical field just as they are applied to other fields. There will always be difference among listeners just as there are differences in color preferences among decorators.

"When a person buys red paint, he expects to get red paint. Similarly, when a person buys a loudspeaker system that is advertised to reproduce down to 40 cps, he should receive one that produces down to 40 cps. If the acoustics industry does not soon begin to police itself, it may find itself policed by governmental agencies to protect the consumer."

Long Island, N. Y.

We have been advised that a new chapter is being formed in Long Island. John D. Meehan was appointed temporary chairman and tentative arrangements had been made to have a meeting in Dave Dettinger's home. We wish this group the best of luck.

Milwaukee, Wis.

William I. Alpert, 3870 North 57th St., Milwaukee 16, Wis., is the 1956–1957 chairman of the Milwaukee Chapter PGA. Please modify your list accordingly.

Phoenix, Ariz.

Andrew B. Jacobsen who has been acting chairman of the Phoenix Chapter has been replaced by the new chairman: Zoeth McFaul, Culver Electronic Co., 231 North 1st Ave., Phoenix, Ariz. Please modify the list of Chapter Chairmen accordingly.

San Diego, Calif.

Larry La Zelle reports that Stan Sessions of NEL spoke on the subject of "Is 50 Watts Necessary?" at the November 16 meeting held at the U. S. Naval Electronics Laboratory. Mr. Sessions demonstrated the relative merits of power in amplifiers with respect to the "listener effect."

San Francisco, Calif.

At a meeting held at the Ampex Corporation, 934 Charter St., Redwood City, John Leslie, Chief Audio Engineer of the Ampex Corporation, discussed "Tomorrow's Music Today—A New Stereophonic Home Music Center." A demonstration accompanied the lecture.

Syracuse, N. Y.

On November 14, the Syracuse Chapter of PGA held a meeting at Dellomorte's Restaurant. At this meeting R. A. Miller of the Bell Telephone Laboratories spoke on "Automatic Telephone Answering as an Audio Facility." Mr. Miller's talk was devoted to a description of the problems encountered in designing an Automatic Telephone Answering Set from a circuit point of view, and covered the problems encountered in embodying the circuit into a physical piece of apparatus to be installed by the average telephone installer without the benefit of an audio facilities background and used by the average public, in many cases not even skilled in the use of business machinery.

A description was given of a special version of a magnetic recording medium developed for this application, "magnetic rubber," and some of the experiences with this medium. A resume was also given of the operating characteristics which were achieved after the manufacture of some 40,000 Automatic Telephone Answering Units.

A few minutes were devoted to some of the uses of Transcribed Message Services in the Bell plant, and the audience participated in giving the speaker suggestions as to attributes which they would like to see in Transcribed Message Services as typified by Automatic Telephone Answering Service.

William W. Dean, Special Products Engineering, General Electric Co., Syracuse, N. Y. is the new 1956–1957 Chairman of the Syracuse Chapter. Please modify your list of Chapter Chairmen accordingly.

PGA CHAIRMAN VISITS SOUTHWESTERN CHAPTERS

IRE-PGA National Chairman Martin devoted parts of his vacation in November to visits to four of the IRE-PGA Chapters in the Southwest. At each meeting chapter operations were discussed, and Dr. Martin gave a demonstration lecture entitled "The Generation and Radiation of Electronic Organ Tone."

In Houston, Texas the meeting was held at the Research Center of the Humble Oil and Refining Company on November 5 with W. C. Wrye, Jr. as chairman.

At San Antonio, Texas, Bill Case presided at the Chapter meeting held in the Physics Laboratory of St. Mary's College on November 6.

The Phoenix Section IRE held its meeting at the Desert Sun Hotel on November 20, with Mike Eberhard, Section Chairman, presiding. Andy Jacobsen (National Vice-Chairman and Acting-Chairman of the Phoenix Chapter) introduced the speaker. At Phoenix the local Baldwin dealer, Roles Piano Company, supplied an organ and Leonard Leigh for a program of background music during the Section dinner.

Chairman Jim Palmer of the Albuquerque Chapter conducted the meeting on November 21 in the auditorium of the Public Service Company of New Mexico.

The coincidence of three of the four dates chosen (to fit the Chairman's itinerary) with times when the population would be vitally interested in other matters had little apparent effect upon attendance. The well-known enthusiasm and hospitality of those living in the "wide, open spaces" was well demonstrated by our members living in the Southwestern cities. Chairman Martin recommends to all prospective audio travelers the pleasant experience of scheduled visits to these Chapters and localities.

PGA-SPONSORED SESSIONS AT NATIONAL CONVENTION

SESSION 20

Tuesday, March 19
2:30–5:00 P.M.

Sert Room, Waldorf-Astoria
HIGH FIDELITY AND HOME MEASUREMENTS

Chairman: FRANK H. SLAYMAKER, *Stromberg-Carlson Co., Rochester, N. Y.*

20.1 "Intermodulation Distortion: Its Measurement and Evaluation," Arnold Peterson, General Radio Co., Cambridge 39, Mass.

20.2 "Testing High-Fidelity Amplifiers in the Home," William W. Dean, General Electric Co., Syracuse, N. Y.

20.3 "Disk and Magnetic Tape Phonograph Systems," Walter H. Erikson, Radio Corporation of America, Camden, N. J.

20.4 "Improved Low-Frequency Loudspeaker Performance," S. Zuerker, General Electric Co., Syracuse, N. Y.

20.5 "A Low-Pressure Phonograph Cartridge," W. E. Glenn, General Electric Research Laboratory, Schenectady, N. Y.

20.6 "A High-Fidelity Phonograph Reproducer," Benjamin B. Bauer and Lee Gunter, Shure Brothers, Inc., Evanston, Ill.

SESSION 27

Wednesday, March 20
10:00 A.M.–12:30 P.M.

Astor Gallery, Waldorf-Astoria
MAGNETIC RECORDING

Chairman: MARVIN CAMRAS, *Armour Research Foundation, Chicago 16, Ill.*

27.1 "An Approach to Quantitative Methods for Evaluation of Magnetic Recording System Performance," Clarence B. Stanley, Ampex Corp., Redwood City, Calif.

27.2 "The Application of WOW and Flutter Compensation Techniques to FM Magnetic Recording Systems," Robert L. Peshel, Ampex Corp., Redwood City, Calif.

27.3 "Design of Instrumentation Magnetic Tape Transport Mechanisms," Konrad Schoebel, Ampex Corp., Redwood City, Calif.

27.4 "The Reverbetron," P. C. Goldmark and J. M. Hollywood, CBS Laboratories, New York 22, N. Y.

27.5 "A Multichannel Transducer for Magnetic Recording," Harold A. Johnson, Shure Brothers, Inc., Evanston, Ill.

SESSION 35

Wednesday, March 20
2:30–5:00 P.M.

Astor Gallery, Waldorf-Astoria
SPEECH ANALYSIS AND AUDIO AMPLIFIERS

Chairman: HARRY F. OLSON, *RCA Laboratories, Princeton, N. J.*

35.1 "A Demonstration of the Representation of Speech by Poles and Zeros," Sze-Hou Chang and Ralph Bach, Jr., Northeastern University, Boston, Mass.

35.2 "A High-Efficiency Speech Amplifier," Herbert Sullivan and Joseph Nelson, David Bogen Co., Inc., Paramus, N. J.

35.3 "Fifty Watt High-Quality Transistor Audio Power Amplifier," Alexander Bereskin, University of Cincinnati, College of Engineering, Cincinnati 21, Ohio.

35.4 "Low Noise Transistor Microphone Amplifier," James J. Davidson, RCA Victor Radio & Victrola Div., Camden 2, N. J.

35.5 "Circuit Considerations for Audio Output Stages Using Power Transistors," R. E. Minton, Radio Corporation of America, Somerville, N. J.

WITH OTHER ACOUSTICAL AND AUDIO SOCIETIES

The September, 1956 issue of the *Journal of the Acoustical Society of America* contained numerous papers of interest to the audio technologist.

S. S. Stevens of the Psycho-Acoustic Laboratory, Harvard University treats the subject of "Calculation of the Loudness of Complex Noise." This is a further study of the subject presented by Dr. Stevens in 1955 and described in this column shortly thereafter. By means of direct loudness matches Dr. Stevens shows that the loudnesses in octave bands can be combined according to the formula $S_t = S_m + 0.3(\Sigma S - S_m)$, where S_t is the loudness (in sones) of the total noise, S_m is the loudness of the loudest band, and ΣS is the sum of the loudnesses of all the bands. The loudnesses of the octave bands can be determined from measurements of their sound pressure levels by means of a chart based upon a new determination of the equal loudness contours for bands of noise. Other charts and formulas are presented for half- and third-octave bandwidths.

Leo Beranek deals with "Criteria for Office Quieting Based on Questionnaire Rating Studies." In order to discover what are the maximum noise levels that office personnel find acceptable and what their reactions are to noisy offices, a survey was carried out at a large air base. A questionnaire composed of 15 rating scales was administered to 190 people scattered over 17 different locations on the base. The rating scales allowed the workers to assess such things as the "noiseness" of their environment and to appraise the effect of noise on various aspects of their work, such as their ability to converse or to use the telephone. Dr. Beranek is president of Bolt, Beranek, and Newman, Inc., acoustical consultants.

Walter Spieth of the Operational Applications Laboratory, Air Force Cambridge Research Center, writes on "Annoyance Threshold Judgments of Bands of Noise." Annoyance threshold judgments were obtained by exposing an individual to noise for three minutes and asking him to adjust the intensity to the level which, if any louder, would annoy him if it were present most of the time where he was working.

Charles T. Morrow and Homer I. Sargeant from the Hughes Aircraft Company wrote an article on "Sawtooth Shock as a Component Test." The shock effect of a terminal-peak sawtooth of acceleration obtained by a drop onto a lead pellet has been investigated experimentally using the shock spectrum as the criterion of severity. It is shown that this spectrum obtained by analysis of the acceleration-time data can be made to approximate closely over a wide frequency range that obtained theoretically for an ideal sawtooth. The smoothness of the spectrum which results from the extreme asymmetry of the pulse shape makes the sawtooth a particularly attractive standard pulse shape for shock test in the laboratory.

Harry F. Olson of RCA Laboratories, Princeton, N. J. describes "Electronic Control of Noise, Vibration and Reverberation." Existing passive materials are inadequate to cope with many problems in the control of noise, vibration, and reverberation. It is only within quite recent times that active systems have been given consideration for the control of sound. The electronic sound absorber is an example of an active-type sound absorber. The electronic sound absorber consists of a microphone, amplifier, and loudspeaker connected in an inverse feedback manner. The electronic system reduces the effective acoustical impedance in the vicinity of the absorber. As a consequence, the electronic sound absorber may be used in the manner of a conventional sound absorber or as a zone-type sound reducer. The electronic vibration reducer consists of a sensor amplifier and driver connected either in negative or positive feedback fashion. The electronic vibration reducer may be used to isolate vibrating machines or to reduce the vibration of machines.

Leo M. Levine and Joseph Hershkowitz of the U. S. Naval Material Laboratory describe a new method for "Measurement of Noise Canceling Effectiveness of Microphones." The noise canceling effectiveness of a microphone is determined by measuring the random noise discrimination as a function of frequency under conditions simulating actual usage, and then using this characteristic to compute the articulation index for an ideal system containing the microphone, when operating under specified noise and speech conditions. Results so obtained are shown to correlate with those of articulation tests for three microphones evaluated for three noise types at different noise levels.

D. H. Howling has an article on "Noise in Magnetic Recording Tapes." A theoretical and experimental investigation is made into the mechanism of noise in magnetic media as used in magnetic recording. It is shown that the basic noise in a wow and flutter free system arises from the fact that the domain size in metals or the particle size of oxides is finite, and improvement in the lower limit of noise can only be made by reduction of particle or domain size. The effect of impressed ac and dc magnetization on the noise level is also examined.

The issue contains its usual excellent "References to Contemporary Papers on Acoustics," by Robert N. Thurston, and "Review of Acoustical Patents," by Robert W. Young.

The April, 1956 issue of the *Journal of the Audio Engineering Society* which arrived recently, contains a number of interesting articles.

Theodore Lindenberg of Pickering and Company, Inc., describes "The Isophase Loudspeaker." The article describes a push-pull electrostatic loudspeaker featuring large diaphragm area, which makes possible low-velocity sound propagation per unit area.

J. Marshall and D. Mogen of the Minneapolis-Honeywell Regulator Co., describe "Transistor Audio Power-Output Stages." The characteristics of the common-base, common-emitter, and common-collector circuits are analyzed for their advantages and disadvantages for audio work.

Hiroshi Amemiya of the University of Tokyo describes "An Output-Transformerless Amplifier." The output stage is single-ended, and the tubes are connected in series. The cathode-coupled phase-inverter driver is used to drive the output stage. An analysis of the driver circuit is followed by a description of the amplifier and its performance. With 20 db of negative feedback, frequency response is flat to 30 kc, down only 1.5 db at 100 kc. A power output of 3 watts is obtained into a 600-ohm load with an input signal of 0.8 volt. Intermodulation distortion is only 0.7 per cent at 3 watts.

Other articles deal with precision recording lathes, audio console design, disks for measurements of disk-reproducer performance, and quality control of loudspeakers.

Ben B. Bauer

High-Speed Duplication of Magnetic Tape Recordings*

JOHN M. LESLIE, JR.†

Summary—**The high-speed duplication of previously recorded magnetic tapes has made the tape radio network practical. For example, each of the two major radio networks in Mexico duplicates over 19,000 seven-inch reels of taped programs per month for their several hundred affiliated stations. Since the tapes are erasable and wire program circuits are seldom used, these networks operate very economically, and the quality of their recorded programs is unimpaired by losses in transmission.**

Likewise, the adoption of Automatic Programming Systems for radio broadcasting will create a demand for syndicated magnetic tape programs and tape libraries.

High-speed magnetic tape duplicators are also utilized for the mass production of tape recordings for entertainment, education, churches, libraries for the blind, reviews of professional publications, and sales information for large field sales organizations.

The Ampex Model S-3200 tape duplicator with ten slaves will produce up to ten copies at one time of 15, 7½, or 3¾ ips master recordings at 60 ips. The copies can be recorded to be reproduced either at the speed of the master or at half speed; *i.e.*, 3¾ ips copies from a 7½ ips master. The duplicates may be either half-track, double-track, full-track, or two-track stereophonic.

Comparisons are made of the degradation of the signal-to-noise ratio, frequency response, and distortion with respect to the number of generations of the duplicates. With optimum adjustment of the equipment, it is very difficult to identify the fourth generation copy from the master recording.

INTRODUCTION

THE GROWTH of the magnetic recording industry has created a demand for an economical method for producing quantities of copies of prerecorded magnetic tapes. This is accomplished by playing the master tape at 60 ips and by recording the duplicates on a number of slave recorders also operating at high speed. Thus, if ten slave recorders are utilized, ten copies of a half-hour 7½ ips master tape can be made in less than four minutes. Allowing for setup time, such a duplicator installation is capable of producing approximately 120 half-hour copies per hour, a production speed-up ratio of more than 60:1 compared to a conventional dubbing arrangement.

Audio recordings made on ¼-inch tape at speeds of 30, 15, 7½, or 3¾ ips and recorded full-track, half-track, double-track, or two-channel stereophonic may be duplicated by this system. Both single-track and dual-track duplicates may be made to play back at the same speed as the master, or single-track duplicates at half speed since a choice of 60 ips or 30 ips speed is provided for the master reproducer and the slave recorders. The playback speed of a duplicate is a function of the ratio of the speed of the master reproducer to the speed of the slave duplicator, as well as the speed at which the master tape was recorded.

* Manuscript received by the PGA, October 5, 1956. Reprinted from IRE TRANS., PGBTS-4, pp. 1–6; March, 1956.

† Audio Div., Ampex Corp., Redwood City, Calif.

DESCRIPTION OF THE TAPE DUPLICATOR

The Ampex Model S-3200 tape duplicator is shown in Fig. 1. The left-hand console contains the master playback unit which utilizes two half-track heads and two independent equalized amplifier channels. The equipment rack contains the master recording amplifier, a stabilized high-frequency tape bias supply, and a master control panel. The recording amplifier and the bias supply are capable of feeding twenty recording heads; hence, the ten slave recorders each utilize two half-track heads. At the right of the picture are three slave recorders. The slaves each have separate upper and lower half-track heads and one full-track head. No erase heads are provided since the tape used for duplicating is bulk erased.

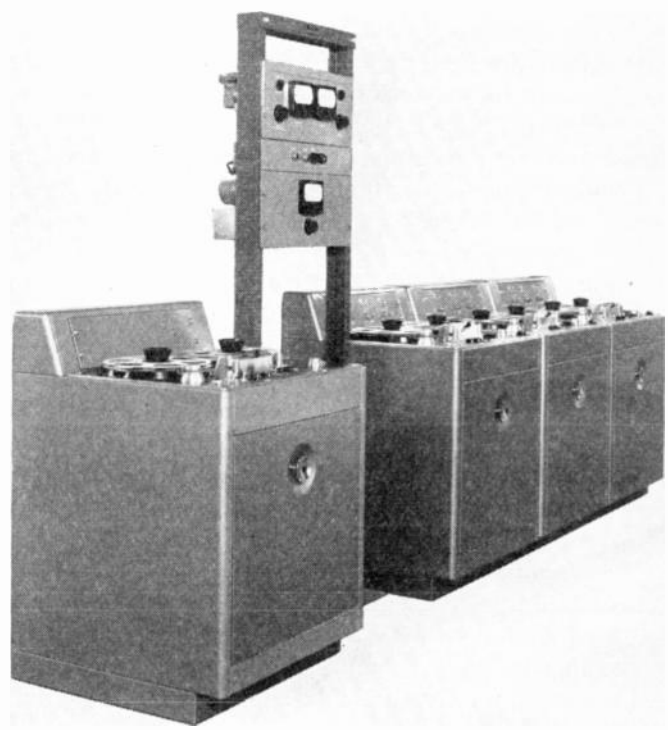

Fig. 1—The Ampex Model S-3200 tape duplicator. The left-hand console contains the master playback unit which includes a two-channel equalized playback amplifier. The rackmounted equipment from top to bottom includes a two-channel recording amplifier with equalization and gain controls, a master control panel, and a high-frequency bias supply. Three slave recorders are shown at the right.

The tape transports of the duplicator system utilize a hysteresis synchronous motor for the capstan drive and induction motors for the turntables. Any reel size up to 14 inches in diameter can be accommodated. Toggle switches corresponding to small or large reel sizes are provided to adjust the torque of the takeup, and rewind motors to provide appropriate tape tension for the size of the reels being used. The controls and arrangement of the tape transport are similar to those of the Ampex Model 300 recorder, and the tape transports may be mounted either in consoles or on equipment racks.

Performance Considerations

Since the duplicator will operate at a tape speed of 60 ips, its over-all frequency response must be flat from 400 cps to 120,000 cps. Constant flux recording is utilized since it is imperative in duplicating tapes to produce copies which have the same flux characteristic vs frequency as the original recording. Hence, a 15-kc signal on a 15-ips master tape will be reproduced and rerecorded as a 120-kc signal, and a 50-cps signal will be reproduced and rerecorded as a 400-cps signal when the duplicator is operated at 60 ips, or eight times the speed of the original recording. When the master playback machine and the slaves are operated at the same speed, the copies of any master recorded at $3\frac{3}{4}$, $7\frac{1}{2}$, 15, or 30 ips, will reproduce at the same speed as the original. If it is desired to make half speed copies; *i.e.*, $3\frac{3}{4}$ ips copies from $7\frac{1}{2}$ ips masters, the master playback will be run at 60 ips and the slaves will be operated at 30 ips. Provision is made in the master recording amplifier for the proper pre-emphasis to match the equalization curves for 2:1 speed reduction. The frequency response of a $3\frac{3}{4}$ ips copy will not be as good as that of the $7\frac{1}{2}$ ips master because the tape is moving only half as fast. Hence, the $7\frac{1}{2}$ ips master may have frequencies recorded out to 15 kc, but the $3\frac{3}{4}$ ips duplicate will be limited to approximately 7.5 kc response.

However, duplicates made at the same speed as the master will not be degraded in frequency response.

If the characteristics of the tape that is being used for duplicating are known, it is possible to adjust the bias current for optimum frequency response and distortion. The bias supply has a control and a meter for setting the bias simultaneously for all the slaves, and each slave head is provided with vernier controls for adjusting its bias and recording currents.

There is always the question of how well the duplicates compare to the master. This can be answered best by direct quantitative comparison of frequency response, distortion, and signal-to-noise ratio of an original tape recording to that of successive generations.

The frequency response of an original recording compared to that of its seventh generation copy is shown in Fig. 2. The curves illustrate the importance of flat frequency response in recording equipment, since any departure from the reference line is multiplied by the number of generations. They show also that with properly adjusted duplicating equipment even the seventh generation should sound excellent to the ear.

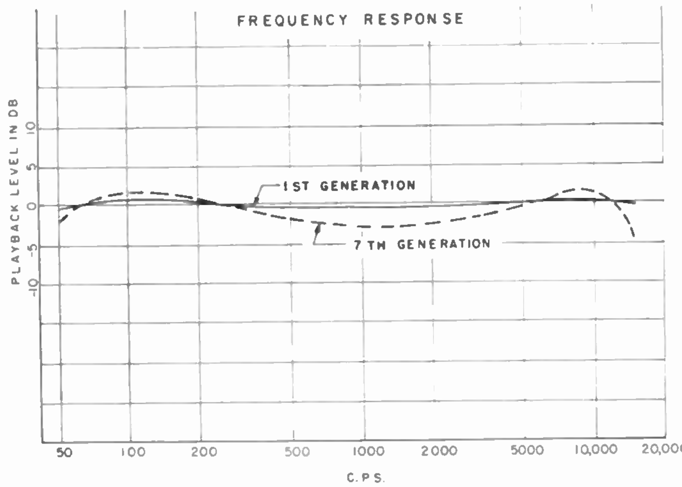

Fig. 2—Comparison of frequency response of an original recording with that of a seventh generation copy. Each generation was copied from the previous generation using the same playback and record machines which had been carefully adjusted and aligned for optimum results with the tape used.

A graph of distortion vs the number of generations of duplication of a recorded tape is shown in Fig. 3. The predominant distortion in magnetic recording is third harmonic. During the process of copying or duplicating an original recording, the third harmonic distortion reaches a maximum of approximately 2.25 per cent at the fourth generation. However, only the most critical ear will be able to detect the difference in distortion between fourth generation copy and the original when music or speech is being compared.

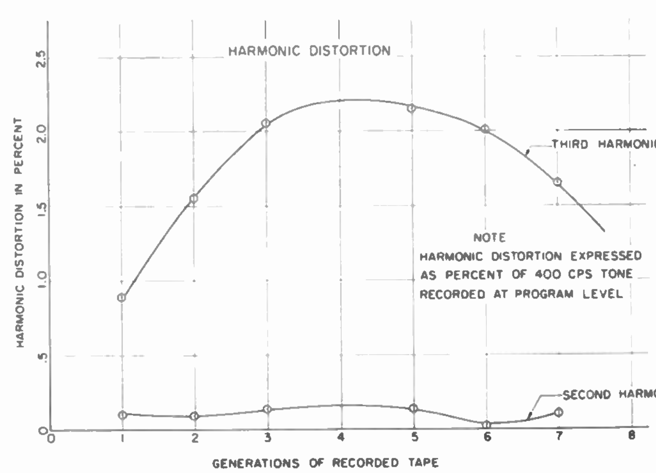

Fig. 3—Comparison of harmonic distortion vs the number of generations of the copies.

A graph of the increase in noise level vs the number of generations of duplication of a recorded tape is shown in Fig. 4. The noise increases 3 db each time the number of generations doubles; *i.e.*, the first copy (2nd generation) has 3 db higher background noise than the

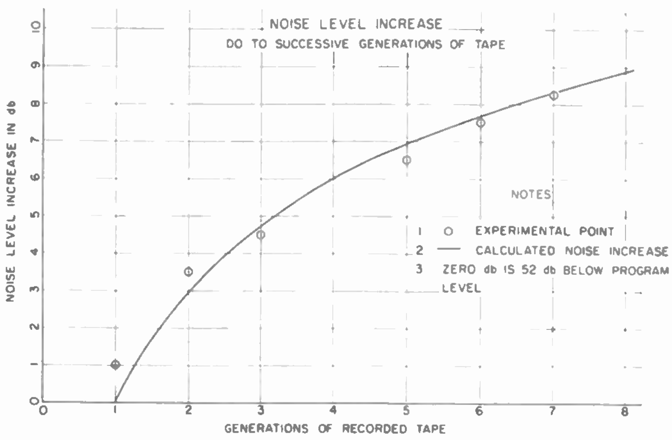

Fig. 4—Comparison of the increase in noise level vs the number of generations of the copies.

original, while the 4th generation has only 3 db more noise than the 2nd generation.

The data for Figs. 2, 3 and 4 was obtained from playback and recording machines which had been carefully adjusted for optimum performance with the particular tape which was used.

For broadcast work it is unusual to use generations beyond the third from the original. Therefore, in practice, the signal-to-noise ratio of a duplicate can be expected to be degraded less than 5 db when compared to the master recording.

Magnetic Tape Networks

How practical is magnetic tape duplication in broadcasting? Two radio networks in Mexico rely upon it for the majority of their program service to their affiliates. One of the networks, Radio Programas de Mexico, operates three duplicator systems with a total of eleven slaves in Mexico City, and two systems with one slave each in Guadalajara and Monterrey. They duplicate approximately 19,000 seven-inch reels of tape per month. Their duplicator installation in Mexico City is operated 22 hours per day and it serves three regional networks, one in Central America and two in the Republic of Mexico. An illustration of the rate of growth of this network is the fact that the number of programs duplicated quadrupled during 1955.

Radio Cadena Nacional in Mexico City has grown from 15 duplicates per year in 1951 to 143,000 per year in 1955. This network utilizes two duplicator systems with a total of eleven slaves.

The tapes are prepared for a two-week delay and are mostly dramatic and musical programs. Transportation of the tape is by parcel post and express, utilizing every available method of transportation. Since good wire circuits are not generally available and their cost would be prohibitive, the tape network is an ideal solution to Latin American network broadcasting. In addition, the tape programs are not restricted in response due to transmission losses, and the tape programs are capable of providing live studio quality.

Automatic Programming

As the automatic programming systems become more prevalent, transcription libraries and syndicated radio program services duplicated on tape will become available for broadcasting. These program tapes will be provided in half-hour program segments with subaudible control tones and with time allowed for local tape-recorded announcements to be automatically inserted in the programs in the desired sequence.

The Ampex Model S-3380 Automatic Programming System, which can utilize such program material, is shown in Fig. 5.

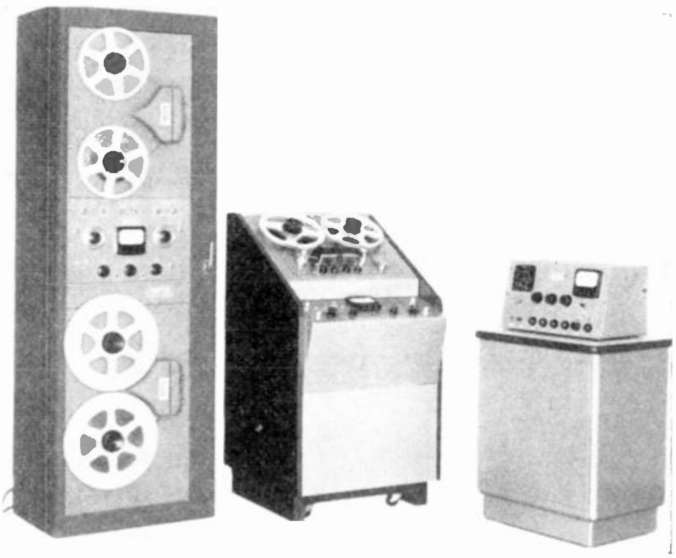

Fig. 5—The Ampex S-3380 Automatic Program System. The top tape transport is utilized for station announcements, spots, etc., and the lower tape transport supplies the program. Twenty-five cycle tones recorded after the announcements and program selections control the switching circuits of the Program Control panel at the center. This system is also adaptable to utilize automatic record players. The console recorder in the center is an Ampex Model 350 for recording local announce tapes and programs. The Record console at the right is a three-channel audio mixer with complete remote control facilities for the Model 350, a 25-cps tone generator for the control tones, and an integrating time clock which measures the total time of the announcement recording.

The operation of the system is as follows. The tape transport (Model 450) at the top of the rack is the "announce machine," and the similar unit at the bottom is the "program machine." At the beginning of, let us say, an eight A.M. show, the machine on the bottom plays the first musical selection or theme, after which the "program machine" automatically stops and simultaneously starts the upper machine which reproduces the local announcements. At the completion of the first announcement, the "announce machine" stops and the operation is automatically switched back to the "program machine." It is a flip-flop type of operation with tight cueing features. At 30 seconds before the half-hour period, the Program Control Unit in the center of the rack takes control; it has a built-in clock mechanism which automatically starts the "announce machine" for the station break and perhaps a time signal. For a short period both machines may actually be running because the moment the timer takes control, the "program machine" is automatically faded down, and the station break announcement over-rides the music—just like many live disk jockey shows. It shounds very realistic since the listener is unaware of the automatic switching or the 25-cps control tones which are filtered out. This type of operation gives correction of true time every half-hour automatically throughout the complete period of operation of the equipment.

The flip-flop operation is not limited to just the two Ampex Model 450's; it can be extended to include a multirecord changer in place of the "program machine." It also can be used to flip-flop automatically on the multiples of half-hours between network and local programs. It can be applied also to other Ampex solenoid operated tape reproducers.

The maximum time of continuous operation with the Automatic Programming System, with standard 2.2 mil tape, is 12 hours; if thin base tape is used, 18 hours of time is available.

Program material for this equipment can be provided in several ways. The independent broadcaster can begin by putting the Automatic Programming System to work for only a few hours a day, and thereby he can free station personnel to start building a tape program library. In a period of three months one independent has recorded his entire record library on tape.

A network can easily set up a small group of people to build a program library of master tapes. These tapes can be duplicated on a high-speed duplicator and then be supplied to the stations in the network where they will be combined with the local announce tapes or local live announcers. In this way, the local listener always hears the local announcers, even though the program tape had been made at some distant location.

Other Uses for Tape Duplicators

Since 1947, over 1,000,000 tapes recorders of all types have been sold in the United States. Over 900,000 of these recorders are designed for nonprofessional use such as in homes, schools, and churches. The large demand for prerecorded tapes for nonprofessional use is being supplied by tape duplicates which now compete favorably with LP records in the retail market.

An "Audio Digest" on tape is now available to the medical profession. These tapes contain a new commentary prepared for medical journals. Subscribers to this service can listen to the tapes while relaxing, thereby saving many hours of research and reading. Such tapes are mass-produced on high-speed duplicators, and are mailed to subscribers on five-inch reels.

Over 28 states maintain libraries of educational tapes for use in schools and a number of religious organizations operate tape duplicating facilities to supply the needs of religious education.

Tape libraries for the blind which provide readings and dramatizations of books and magazines, as well as music, are also established in many cities.

Conclusion

To summarize briefly, the high-speed magnetic tape duplicator is capable of producing a copy of a twin-track 12-hour program tape for the Ampex Automatic Programming System in 24 minutes, and with additional slaves it can produce up to ten copies in the same period of time. The high quality of the tape allows broadcasters to present to the listener both better tone quality and a greater variety of announcers and local personalities. And all of this is possible with reduced operating costs and no impairment of program service.

Compensation Networks for Ceramic Phonograph Reproducers*

BENJAMIN B. BAUER†

Summary—**This paper examines the theory and practice of compensation networks for ceramic phono-reproducers. It shows that even though the RIAA Recording Characteristic is defined in terms of velocity, an amplitude-responsive reproducer may be used to match this characteristic in an ideal manner.**

* Manuscript received by PGA, November 12, 1956. Portions of this paper were presented at the National Electronics Conference, October, 1956.

† Shure Brothers, Inc., Evanston, Ill.

Introduction

THE PURPOSE of this paper is to examine the theory and practice of compensation networks for ceramic phono-reproducers and to dispel some erroneous ideas about them. The recording characteristic of the Radio Industry Association of America is defined in terms of velocity. Therefore, some people have surmised that a velocity-responsive reproducer, rather than an amplitude-responsive reproducer is required to match it in an ideal manner. That such an assumption is unwarranted will become clear in the course of this paper, with the resulting conclusion that the battle of transducing principles must remain on technological rather than on theoretical grounds.

The fact that amplitude-responsive reproducers are more adapted to the RIAA characteristic than are the velocity-responsive reproducers is evident from Fig. 1.

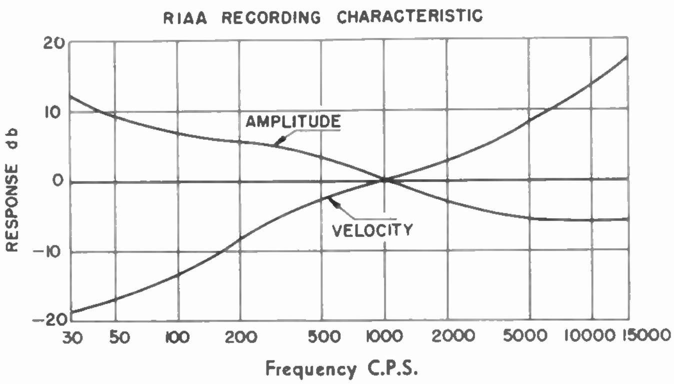

Fig. 1—Record Industry Association of America standard characteristic plotted in terms of amplitude and velocity.

This figure shows the standard RIAA recording characteristic plotted in terms of velocity and of amplitude. It is seen that a velocity-responsive reproducer will require a total compensation of 36 db to equalize to "flatness" from 30–15,000 cps while an amplitude-responsive reproducer will require a total of 18 db, and this fact should dispose of the practical aspects of the problem. This paper is intended to treat the theoretical arguments also. In a customary arrangement of an amplitude-responsive ceramic reproducer connected to a grid resistor, a practical compensation circuit is formed of a series resistor and a capacitor connected across the terminals of the reproducer. Whereas it has been known for sometime that this arrangement provides adequate compensation it has not been heretofore proven, to our knowledge, that it actually does theoretically meet the requirements of the RIAA recording characteristic. This is shown to be the case and the analysis yields the means for calculating the necessary circuit constants for an ideal amplitude-responsive reproducer. One unsolved problem which will be left for others is to prove that the practical circuits currently used with magnetic and dynamic reproducers are equally able theoretically to meet the requirements of the RIAA standard.

Methods for compensating nonideal amplitude-responsive reproducers have been treated in a previous paper on this subject.[1]

[1] B. B. Bauer, "Engineering considerations of ceramic phonograph pickups," IRE Trans., vol. Au-4, pp. 94–98; July–August, 1956.

The Problem

The details of the problem are illustrated in Fig. 2. At the upper left-hand corner there are three networks which define the standard RIAA characteristic. The standard is specified as the algebraic sum of the ordinates of three individual curves which conform to the admittances of these three networks expressed in db.

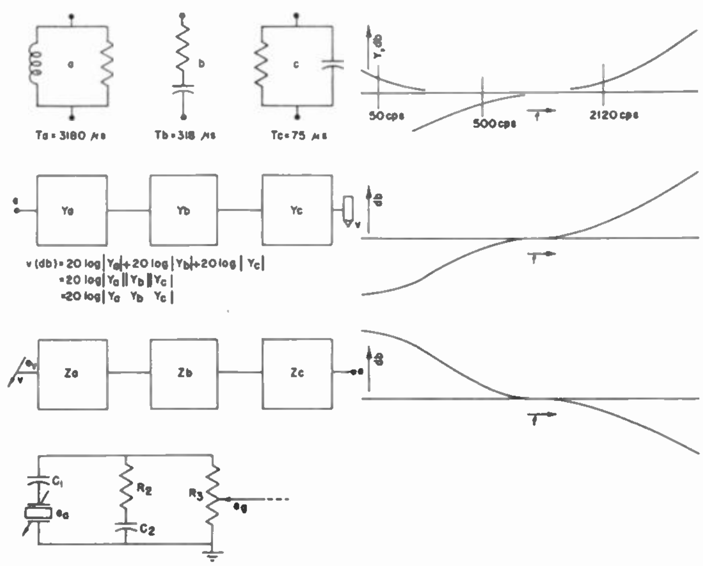

Fig. 2—The problem of compensation of phonograph reproducers to comply with standard RIAA characteristic.

Whereas this definition is ambiguous insofar as the term db is technically applicable to current, voltage, or power ratios, the intent of the standard becomes clear by examination of the published curve shown in Fig. 1. The central network labeled "b" has a time constant T_b of 318 microseconds which corresponds to a 500 cycle turnover. The left-hand network with a time constant T_a of 3180 microseconds provides a low-frequency pre-emphasis with a turnover of 50 cps. The right-hand networks with a time constant of T_c of 75 microseconds provides a high-frequency pre-emphasis with a turnover at 2120 cps. As shown in the schematic diagram immediately below the circuits, the voltage "e" to be recorded is modified, in effect, by three networks having transmission characteristics proportional to $|Y_a|$, $|Y_b|$, and $|Y_c|$ and thence, applied to a record cutter imparting to the stylus a velocity proportional to the product of the absolute values of the three admittances. The over-all velocity-frequency characteristic obtained in this manner is shown at center-right and it is the familiar RIAA recording characteristic.

To obtain a theoretically correct playback characteristic with a velocity-responsive reproducer the output of the reproducer must be modified by three networks having a transmission characteristic proportional to the impedance of the above three networks or an equivalent thereof. These networks will define a transmission characteristic shown at the lower right-hand side of the figure which is a precise counterpart of the RIAA recording characteristic.

In the instance of a displacement-responsive reproducer as shown in the lower left-hand corner, Fig. 2 having an internal capacitance C_1 the compensation network long ago discovered by insight and experiment consists of a series resistor and capacitor connected across the terminals of the reproducer. The question is: "How to prove (or disprove) that the compensation provided by this circuit is fully compatible with the RIAA recording characteristic"?

To answer this question it is necessary first to develop an analytical expression for velocity function of the RIAA recording characteristic, which from its definition consists of the product of three absolute values of the individual admittances, $|Y_a|$, $|Y_b|$, and $|Y_c|$. This presents a rather sticky algebraic situation involving roots of the sums of square terms. Our insight tells us that it might be equivalent, instead, to use the absolute value of the product of the three complex functions Y_a, Y_b, and Y_c and this may be proven correct by anyone willing to spend time on the algebraic work envolved. The problem is reduced to the following steps:

1) Obtain an analytic expression for the RIAA recording characteristic in terms of velocity and of amplitude.
2) Obtain the transmission characteristic of the compensation circuit in question.
3) Compare the two expressions to evolve the required circuit relationships.

Transmission Requirements

Step 1) may be readily achieved by remembering the equations for time constants and then rewriting the individual admittances in terms of time constants. Since $T_a = L_a/R_a$, $T_b = R_bC_b$, and $T_c = R_cC_c$, the following expressions are obtained:

$$Y_a = \frac{1}{R_a} + \frac{1}{j\omega L_a} = \left(\frac{1}{R_a}\right)\left(1 + \frac{1}{j\omega T_a}\right) \tag{1}$$

$$Y_b = \frac{1}{R_b + \dfrac{1}{j\omega C_b}} = \left(\frac{1}{R_b}\right) \times \frac{1}{1 + \dfrac{1}{j\omega T_b}} \tag{2}$$

$$Y_c = \frac{1}{R_c} + j\omega C_c = \left(\frac{1}{R_c}\right)(1 + j\omega T_c). \tag{3}$$

Velocity v is equal to an arbitrary constant k multiplied by the three admittances resulting in the following expression:

$$v = kY_aY_bY_c = (k/R_aR_bR_c)\,\frac{(1 + 1/j\omega T_a)(1 + j\omega T_c)}{1 + 1/j\omega T_b}$$

$$= K\,\frac{1/T_a + 1/T_c + j\omega(1 - 1/\omega^2 T_a T_c)}{1 + 1/j\omega T_b}. \tag{4}$$

In the right-hand term all the constant terms have been combined into a new arbitrary constant K.

Amplitude is obtained by dividing velocity by $j\omega$ with the following result:

$$a = v/j\omega = K\,\frac{1/j\omega T_a + 1/j\omega T_c + (1 - 1/\omega^2 T_a T_c)}{1 + 1/j\omega T_b}. \tag{5}$$

A network to compensate a velocity-responsive reproducer will require a transmission which is the inverse of (4). A network to properly compensate an amplitude-responsive reproducer will require transmission which is the inverse of (5).

This latter expression is the one which will be used for analysis of networks for ceramic reproducers.

Analysis of Ceramic Pickup Networks

A schematic diagram of an ideal ceramic reproducer and compensation network is shown in Fig. 2. The equivalent open circuit voltage is denoted by e_a in series with the combined capacity of the transducer element and the leads C_1. The output voltage e_0 is developed across the grid resistor R_3. Across this combination we connect a series R_2 and C_2. The transmission characteristic of this circuit is as follows:

$$\frac{e_0}{e_a} = \frac{1 + \dfrac{1}{j\omega C_2 R_2}}{\dfrac{1}{j\omega}\left\{\dfrac{1}{C_1R_3} + \dfrac{1}{\dfrac{C_1C_2}{C_1+C_2}}\right\} + \left(1 - \dfrac{1}{\omega^2 C_1C_2R_2R_3}\right)}. \tag{6}$$

To be sure this is not the simplest possible form of this expression, but rather one that is obtained by suitable manipulations to simplify the subsequent analytical work. The required transmission function is obtained by inverting (5) as shown below:

$$\frac{K}{a} = \frac{1 + \dfrac{1}{j\omega T_b}}{\dfrac{1}{j\omega}\left(\dfrac{1}{T_a} + \dfrac{1}{T_c}\right) + \left(1 - \dfrac{1}{\omega^2 T_a T_c}\right)}. \tag{7}$$

Here again the $j\omega$ terms were rearranged to simplify analysis. If (6) and (7) are compared, then it is evident that:

$$T_b = C_2R_2. \tag{8}$$

At this point one must avoid yielding to the temptation of equating T_a to C_1R_3 and T_c to $C_1C_2R_2/(C_1+C_2)$: instead the complete terms containing ω and ω^2 coefficients must be equated as follows:

$$\frac{1}{T_a} + \frac{1}{T_c} = \frac{1}{C_1R_3} + \frac{1}{\dfrac{C_1C_2R_2}{C_1+C_2}}, \tag{9}$$

$$T_aT_c = C_1C_2R_2R_3. \tag{10}$$

Eqs. (8), (9), and (10) include four network parameters, namely, C_1, C_2, R_2, and R_3. If any one of these terms is known, the remaining three may be calculated. For example, given the capacity C_1 of the ceramic element and connection leads, farads, and remembering that $T_a=3180$; $T_b=318$; $T_c=75\times10^{-6}$ seconds, the following values are found for the remaining electrical terms:

$$R_3 = T_aT_c/T_bC_1 = 750 \times 10^{-6}/C_1 \text{ (ohms)} \tag{11}$$

$$R_2 = \frac{R_3}{\dfrac{T_a}{T_b}+\dfrac{T_c}{T_b}-\left(1+\dfrac{T_a}{T_b}\dfrac{T_c}{T_b}\right)} = R_3/6.88 \text{ (ohms)} \tag{12}$$

$$C_2 = C_1\left[\frac{T_b}{T_a}+\frac{T_b}{T_c}-\left(1+\frac{T_b}{T_a}\frac{T_b}{T_c}\right)\right] = 2.92\,C_1 \text{(farads)}. \tag{13}$$

Conclusion

The expressions resulting from the preceding analysis are applicable to ideal amplitude-responsive reproducers and they may be used for obtaining initial design for practical cases. Many practical reproducers deviate from the ideal and means have been described to compensate and test these practical reproducers.[1]

It has been shown that the amplitude-responsive reproducers can be made to conform to the RIAA characteristic by a simple compensation means in a theoretically ideal manner and the theory of compensation for amplitude-responsive reproducers has thus been established on a firm basis.

Evaluation of High-Powered Outdoor Sound Systems*

ROBERT W. BENSON†

Summary—**Outdoor installations of high-powered sound systems have been made for the purpose of communicating over large areas from systems located on tall buildings and on airplanes. In order to evaluate the performance of these systems it is necessary to use actual speech materials rather than perform simple physical measurements. Airborne systems are affected greatly by the Doppler shift in frequency which cannot be accounted for in a physical evaluation of a system and reflections from buildings introduces echoes for which it is impossible to calculate the effect upon intelligibility. Speech materials have been used to determine both the intensity levels as a function of distance and angle, as well as the intelligibility of the system for various power levels. The results of these studies lead directly to the design of more efficient communication systems. The application of the results of two studies is shown for the design of optimum systems.**

THE DESIGN of a sound reinforcement system is based upon requirements for increasing the intensity of words delivered to a listening position without affecting the intelligibility of those words. The major reason for increasing the intensity is because of insufficient power to overcome the background noise at the listening position. In certain instances, consideration must be given to factors other than that of maintaining adequate signal to noise ratios. The installation of sound reinforcement equipment in auditoriums and other indoor locations makes it necessary to give consideration to conditions which produce acoustic feedback. Problems concerning reverberation and echoes also exist in the indoor installation. In outdoor installations, however, the problem of maintaining an adequate signal to noise ratio is mainly dependent upon the total power available and the angular distribution of energy. If the total power rating of the system can be determined, then the major design problem is to obtain suitable directional properties of the loudspeaker system.

Several studies have been made involving the prediction of performance of speech communications systems by the measurement of the physical performance of the over-all system. It has been shown that adequate correlations exist between physical performance and actual speech intelligibility measures[1] on systems having nonuniform frequency response characteristics and containing unwanted noise. These techniques have been applied to over-all communications systems, including telephone transmission and radio-telephone communications. Limited use of this information has been used to predict speech intelligibility in auditoriums which use sound reinforcement systems. A complicating factor in the application of the simple procedures is the handling of the effects of reverberation and echoes which may be produced by the acoustical environment. In these cases, the amount of signal is usually adequate with respect to

* Manuscript received by the PGA, November 15, 1956. Paper presented at National Electronics Conference, October, 1956.
† Armour Research Foundation, Chicago, Ill.

[1] L. L. Beranek, "The design of speech communication system," Proc. IRE, vol. 35, pp. 880-890; September, 1947.

the noise, but the actual speech signals are blurred by the acoustical environment. This blurring effect is not accounted for in the simple methods of predicting speech intelligibility.

Sound reinforcement systems for outdoor communication are also difficult to design solely on the basis of adequate signal to noise ratios. In a fixed installation it is necessary to account for the atmospheric conditions which cause sounds to fluctuate quite markedly while propagating over relatively short distances. In addition, the background noise usually encountered is very unsteady and quite unpredictable as a function of time. Another factor is the location of major obstructions which act as reflectors for the sound and cause pronounced echoes. These various factors influence not only evaluation procedures but the original design of an outdoor sound reinforcement system.

Another application is the use of sound systems which are carried by airplane for the communication of voice signals from air to ground. The design, as well as the performance, of these systems is influenced by the above factors and, in addition, to those effects caused by the motion of the sound source with respect to the observer on the ground. This results in the familiar Doppler shift in frequency causing an additional effect which is not accounted for in simple procedures used for the evaluation of communications systems.

An outdoor system is usually required to provide adequate signal levels over a relatively wide listening area. In many instances the amount of power available is limited by factors of either economy, weight, or size. The installation of outdoor sound systems for athletic events held in large stadiums are almost routinely installed on the basis of previous experience and then adjusted over a period of time until satisfactory performance is obtained. It is desirable in certain installations to provide a straight-forward method of evaluation of such systems in order to provide an economic, as well as a satisfactory solution to the communication problem.

Three different systems have been designed, installed, and evaluated, for which it was desirable to obtain optimum performance on the basis of weight and size or economy. The first two of these systems are airborne sound systems which are used for communicating speech signals from air to ground from an airplane passing overhead. These two systems are required to provide adequate signals so that an observer on the ground may understand a message for as long as one minute. Since the entire sound system must be mounted on an airplane, both size and weight are important factors in the design of these systems. One of these systems was studied for the Federal Civilian Defense Agency. This system, due to the large quantities contemplated, must, therefore, also be considered from the standpoint of initial cost and operating costs. The third system studied was a ground installation of a sound system for the purpose of communicating signals over large areas of a city and was also supported by the Federal Civilian Defense Agency. If this method of communicating to the public can be shown feasible and desirable, many similar installations would be contemplated throughout the country and, therefore, economic factors are of the utmost importance. These three systems, therefore, require that sufficient information be obtained in order to provide an economical solution to the problem so that similar applications may be made.

The specific components, both electronic and electroacoustic, used in these studies are of little importance. The amplification necessary to obtain adequate signal levels can be obtained by many varieties of commercial electronic amplifiers. The loudspeaker characteristics, both in power-handling capacity and directional characteristics, can be obtained by the use of various commercial components. The original system, however, was designed and constructed of components which would allow for the utilization of a maximum amount of output power. Later systems based upon the information obtained in the original studies could then be selected on the basis of actual cost, weight, and size, providing their over-all electroacoustic performance met the requirement for adequate communication.

The first requirement for the design of each of the three systems was to provide adequate directional characteristics for the loudspeaker array. Although it is possible to design a loudspeaker horn system with a given radiation pattern, it was more practical to use the combination of multiple horn units which would provide for variable radiation pattern characteristics. The characteristic of one individual horn-type loudspeaker is shown in Fig. 1. Using this as a basic unit, graphical methods were used to determine the first system to be tested for use on aircraft for which the speed and altitude capabilities are known. A desired message time was decided upon and from these factors the optimum radiation pattern for maintaining constant signal level on the ground was determined. Four loudspeakers having radiation patterns shown in Fig. 1 could be combined to produce near optimum radiation characteristics. The pattern predicted from the graphical method and the actual measured characteristic of the loudspeaker system are shown in Fig. 2. The actual characteristic of the loudspeaker system is determined by an actual flight test of the loudspeaker system when corrected for atmospheric attenuation and for the position and speed of the aircraft (shown in Fig. 3). This system is the first attempt at the design of a system which would provide optimum performance. Similar procedures were used for the design of another airborne, as well as the installation of a loudspeaker system in a city. The directional characteristics desired for the installation in a city were designed for optimum coverage of an area which was square in shape, having the loudspeaker installation in the center. Provisions were to be made for adequate coverage of one square mile. The loudspeakers chosen had approximately a 90° pattern horizontally and a

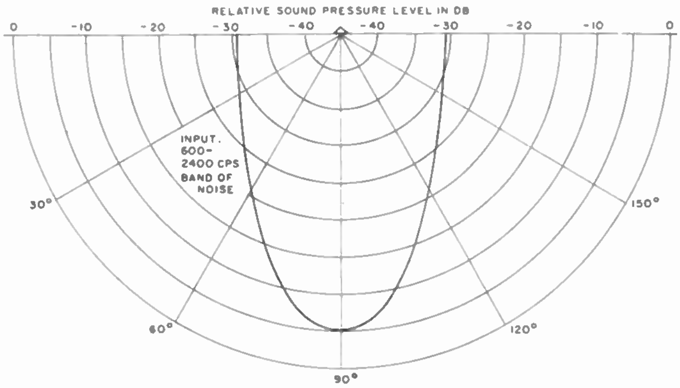

Fig. 1—Directional response of single loudspeaker unit.

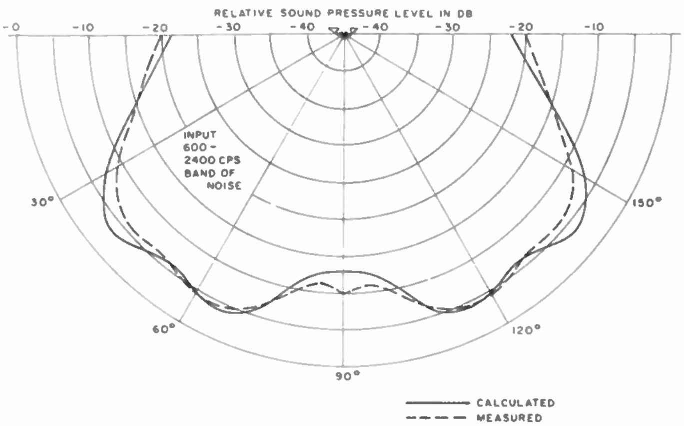

Fig. 2—Calculated and measured response of four loudspeaker units separated by 25°–50°–25°.

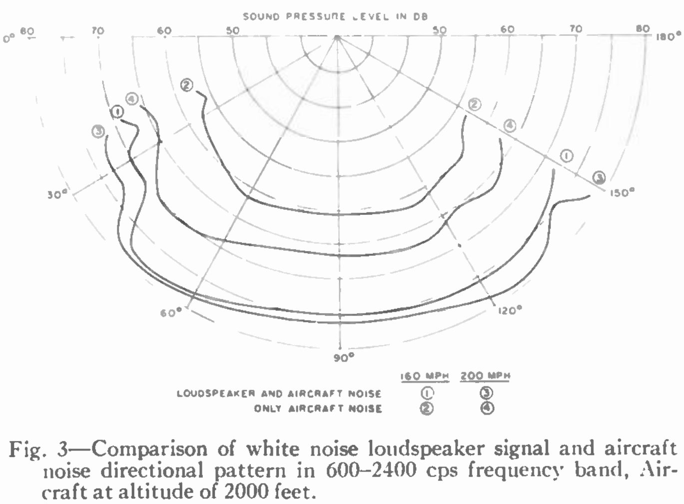

Fig. 3—Comparison of white noise loudspeaker signal and aircraft noise directional pattern in 600–2400 cps frequency band, Aircraft at altitude of 2000 feet.

total of eight speakers were used vertically in a line to provide a vertical pattern of approximately 5°. The maximum power in the airborne systems was determined by the capacity of the aircraft so far as weight and size of the installation. The power of the ground installation was determined by calculations indicating that which would provide adequate signal levels throughout the area of one square mile. The systems had over-all power ratings of 2000 watts and 300 watts for the two airborne systems and 3200 watts for the ground installation.

On the basis of information contained in the literature, it was decided that the frequency range of the system would be adequate for speech purposes for the band of frequencies from 600 to 2000 cycles per second. The loudspeaker selected for this purpose maintained its maximum efficiency throughout this frequency range. It is well known that most of the energy in speech occurs below the frequency of 600 cycles per second. The amount of increase in gain for words used in the evaluation of these systems can be determined by analyzing data available on the spectral distribution of speech energy.[2]

An analysis of the various test words used in the study provided information concerning the number of words which could be amplified providing certain filter characteristics were chosen. These increases in gain appear in Fig. 4 (next page) for various filter characteristics. Fig. 5 indicates both the spectrum level of speech through the various amplification systems and the actual filter characteristics plotted on a relative gain basis showing the additional amplification available through the speech range. An analysis of the data shown predicted the measured values for the average gain available to speech signals for various high pass filter characteristics. This data is plotted as a function of frequency in Fig. 6. The effect of not only eliminating low-frequency energy, but also eliminating high-frequency energy, is shown in Fig. 7. The graph illustrates that for a low-frequency cutoff of 600 cps very little is gained for reducing the high-frequency cutoff. This figure illustrates that merely by eliminating those frequencies below 600 cycles per second it is possible to increase the average gain for each syllable by approximately 8 db. In other words, that part of speech energy which is important for communication purposes can be delivered to the loudspeaker at a level 8 db higher than similar unfiltered speech.

Two factors are necessary to describe a system so far as its ability to provide adequate communication. The first requirement is that the system provide adequate levels so that the signal may be detected. The second requirement is that those signals which are sufficiently loud to be detected must also arrive at the listening position with small enough distortion so that they remain intelligible. For systems having adequate power capacity it is possible to determine the sensitivity of the system by physical measurements. When the system may be complicated by adverse signal to noise ratios or fluctuations in either signal or noise, the ear is a better detector for the determination of the sensitivity of the system. The intelligibility of words even if they provide adequate levels is only predictable by physical measurements if the system is free from such complex factors as fluctuating signals, fluctuating noises, and frequency

[2] R. W. Benson and I. J. Hirsch, "Some variables in audio-spectrometry, *J. Acous. Soc. Amer.*, vol. 25, pp. 499–503; May, 1953.

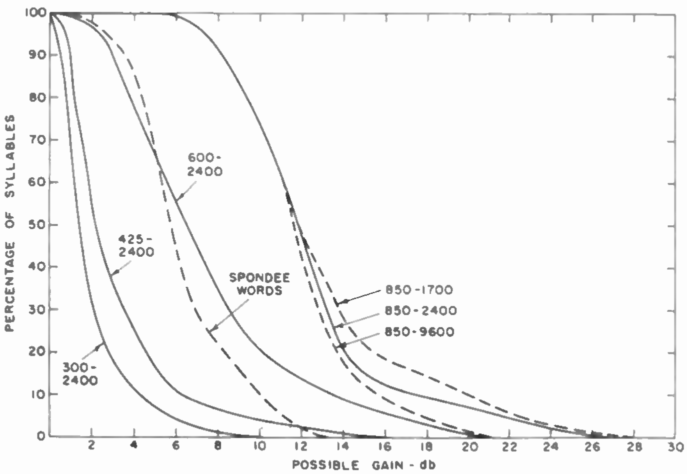

Fig. 4—Relative number of syllables which may be amplified a given amount after filtering.

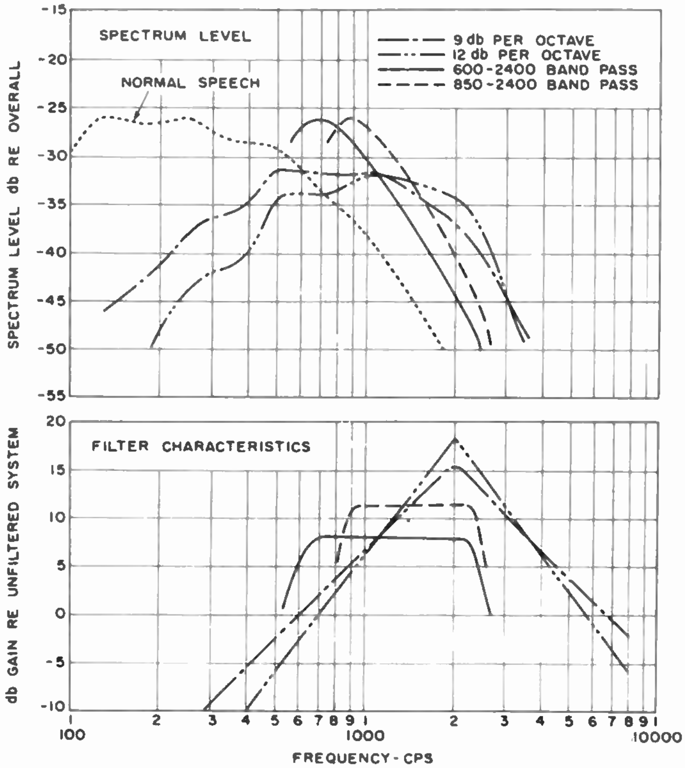

Fig. 5—Spectrum level of speech after passing various filters.

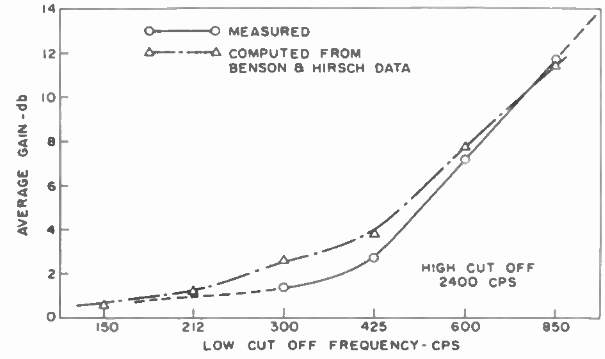

Fig. 6—Mean speech system gain increase with low cutoff filter.

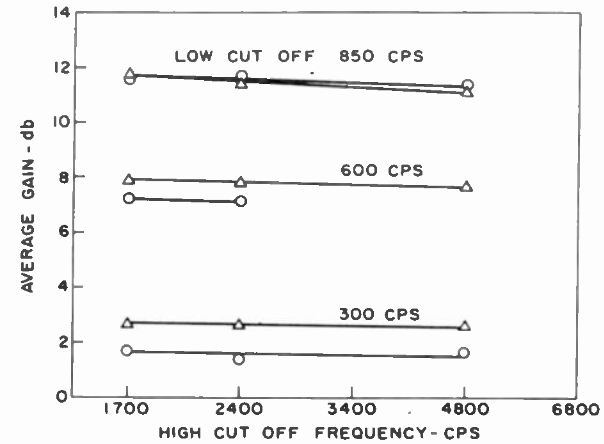

Fig. 7—Mean speech system gain increase with high cutoff filter.

shifts. Each of the above three systems was, therefore, evaluated by the use of speech materials which have been designed to measure sensitivity and intelligibility. By measuring the sensitivity and intelligibility of the systems as a function of power input to the loudspeaker system one is able to determine, first of all, if the total power is adequate and, secondly, if the angular distribution of power is optimum.

The testing procedure is based upon similar concepts used in the evaluation of an individual's hearing capabilities. Basically the objective is to vary the level of the signal until the intensity of the individual speech signals is just great enough to provide intelligible speech. In order to provide a statistical figure of performance, it is necessary to have numerous subjective opinions. In order to do this, tape recordings were made which would produce a source of signal as a function of time. These recordings were then played over the sound system while another tape recording was made of the transmitted speech signals. For the airborne sound systems the listening microphone was fixed in a location below the airplane's path. For the sound system located in a city, the position of the listening microphone was moved to various locations in order to provide an indication of the coverage obtained by the system. The recordings made during any of the tests could then be replayed at a level corresponding to the actual level of the signals to many subjects at a more convenient time and in the laboratory.

The first series of measurements upon the systems was to determine the input power to the system which would just provide sufficient signal for adequate listening. In order to obtain this information, the power input to the loudspeaker system was reduced from the maximum power rating of the system to an inaudible level. The evaluation of the system for these various power inputs would then indicate the amount of excess signal which is available in the over-all system. In the case of the airborne loudspeaker systems, when the amount of excess power is plotted as a function of airplane position, one is able to determine if the directional characteristics of the loudspeaker system have been designed to produce constant listening conditions for the desired message time. If a plot of the power required to produce intelligible speech shows a constant level as a function of the position of the plane for the period of time which is desired, then the radiation pattern of the loudspeaker system is correct. If it is not, one is able to correct any deficiency of the loudspeaker system by adjusting the radiation pattern. The amount of increase in background noise which can be tolerated during the operation of the system is determined by the amount of reduction in power of the system required to produce no intelligible speech.

Fig. 8 indicates a plot of the power required to produce intelligible speech from an aircraft passing overhead as a function of the position of the aircraft. Similar plots can be made for any system, including fixed installations, to indicate the requirements for the distribution of energy. Once this information is available the radiation pattern of the loudspeaker system may be designed.

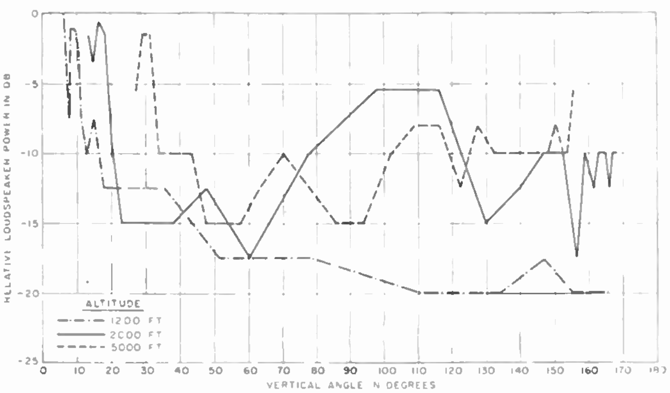

Fig. 8—Relative loudspeaker power needed to achieve 50 per cent spondee word intelligibility on ground. Airspeed of 160 mph.

The initial requirements, therefore, are to have a system which will provide adequate signals over the entire area to be covered and perform an evaluation of the system to determine the power requirements for each location of interest. This information may then be used to redesign a system which provides equal listening conditions at every location. These measurements include all factors which may influence the intelligibility of the system. Factors such as fluctuating signals, fluctuating background noises, echoes, and frequency shifts are all combined to affect the power requirements for producing intelligible speech.

It is possible that a system which provides adequate listening levels fails to transmit intelligibility. The basic speech words which are selected for the above measurements are, therefore, chosen to be affected least by various distortions which may be introduced. The first series of measurements are designed for the determination of sensitivity of the system so that it can be determined that adequate listening levels are available throughout the entire area. Words such as these have been developed for the measurement of hearing sensitivity and are commonly referred to as "spondee" words.[3] Another group of words are also available which are extremely sensitive to various distortions. They allow for a direct indication of the ability of the system to convey intelligibility. These words which are representative of everyday speech can then be used to predict the success of the over-all system. The evaluation using these words may be conducted after it has been determined that the system provides adequate listening levels. The system is then operated at a power sufficiently in excess of the minimum required to assess the various distortions introduced by reflecting surfaces, frequency shifts, and other phenomena occurring. The results of this information are not likely to influence the design of the system from the standpoint of electroacoustic components. This information will determine limits on operating conditions of the system. In the case of airborne systems, limits will be found for the speed of the aircraft, that factor which produces the Doppler effect, and altitudes for which the transmission of sound fluctuates sufficiently to produce unintelligible speech. In the case of ground installations, it is possible to assess the importance of various obstacles which produce echoes and reverberant listening conditions which, again, limit the use of a system.

The evaluation of outdoor systems by the above procedures of using actual speech materials allows for the proper design of an over-all sound reinforcement system as well as information giving the limiting conditions of operation of the system.

[3] I. J. Hirsch, "The Measurement of Hearing," McGraw-Hill Book Co., Inc., New York, N. Y.; 1952.

Contributors

Benjamin B. Bauer (S'37–A'39–SM'44–F–'53) graduated in industrial electrical engineering from Pratt Institute in 1932, and received the E.E. degree from the University of Cincinnati in 1937.

B. B. BAUER

He joined Shure Brothers, Inc., in 1936, became chief engineer in 1940, Director of Engineering in 1944, and Vice-President in 1948.

Mr. Bauer is former National Chairman and Editor-in-Chief of IRE TRANSACTIONS ON AUDIO, and is presently Secretary-Treasurer of the PGA. He is a Fellow of the Acoustical Society of America and of the Audio Engineering Society, and is an associate editor of the Journal of the Acoustical Society of America. He is a member of Eta Kappa Nu, Tau Beta Pi, and Sigma Xi.

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

R. W. BENSON

Dr. Benson was formerly head of the Physical Acoustics Section of the Central Institute for the Deaf in St. Louis, He is now assistant manager of the Physics Research Department of Armour Research Foundation of the Illinois Institute of Technology.

Dr. Benson holds membership in the Acoustical Society of America.

❖

John M. Leslie, Jr. (SM'56) was born in Springfield, Mo., on June 22, 1921. His education at the University of California was interrupted by service in the Navy during World War II, when he was an instructor in radio and radar at Treasure Island. He received the B.S. degree in electrical engineering in 1949.

J. M. LESLIE, JR.

Mr. Leslie was associated with Bendix Radio and Television Corporation in Towson, Md., as a field engineer covering the territories of northern California and Nevada. In 1950 he joined Ampex Corporation where he now holds the position of chief audio engineer. He has participated in the development of both the broadcast studio quality audio recorders and multichannel, fm carrier, data recorders.

He is a member of S.M.P.T.E. and A.E.S.

INSTITUTIONAL LISTINGS (Continued)

JENSEN MANUFACTURING COMPANY, 6601 South Laramie Ave., Chicago 38, Illinois
Loudspeakers, Reproducer Systems, Enclosures

KNOWLES ELECTRONICS, INC., 9400 Belmont Ave., Franklin Park, Illinois
Miniature Microphones and Receivers, Special Recorder and Audio Devices

JAMES B. LANSING SOUND, INC., 2439 Fletcher Dr., Los Angeles 39, California
Loudspeakers and Transducers of All Types

THE MAICO COMPANY, INC., 21 North Third St., Minneapolis 1, Minnesota
Audiometers, Hearing Aids, Magnetic Recording Devices

SHURE BROTHERS, INC., 222 Hartrey Ave., Evanston, Ill.
Microphones, Pickups, Recording Heads, Acoustic Devices

SONOTONE CORPORATION, P.O. Box 200, Elmsford, New York
Ceramic Phonograph Cartridges

STROMBERG-CARLSON COMPANY, Special Products Div., 1700 University Ave., Rochester 10, New York
High-Fidelity Equipment; Intercommunication, Public Address, and Sound Systems; Electronic Carillons

TELEX, INC., Telex Park, St. Paul 1, Minnesota
Subminiature Magnetic Receivers, Microphones, Hi-Fi Headsets, Encapsulation, Packaged Circuitry

UNITED TRANSFORMER COMPANY, 150 Varick St., New York, New York
Transformers, Filters, and Reactors

UNIVERSITY LOUDSPEAKERS, INC., 80 South Kensico Ave., White Plains, New York
Public Address and High Fidelity Loudspeakers

Charge for listing in six consecutive issues of the TRANSACTIONS—$75.00.
Application for listing may be made to the Technical Secretary, Institute of Radio Engineers, Inc., 1 East 79th Street, New York 21, N.Y.

INSTITUTIONAL LISTINGS

The IRE Professional Group on Audio is grateful for the assistance given by the firms listed below, and invites application for Institutional Listing from other firms interested in Audio Technology.

ALLIED RADIO CORPORATION, 100 N. Western Ave., Chicago 80, Illinois
Everything in Radio, Television, and Industrial Electronics

ALTEC LANSING CORPORATION, 9356 Santa Monica Blvd., Beverly Hills, California
Microphones, Speakers, Amplifiers, Transformers, Speech Input

AMPEX CORPORATION, 934 Charter St., Redwood City, California
Magnetic Tape Recorders for Audio, Video, and Instrumentation Application

AMPLIFIER CORPORATION OF AMERICA, 398 Broadway, New York 13, New York
Battery-Operated Magnetic Tape Recorders, Bulk Tape Erasers, Amplifiers, Power Supplies

AUDIOPHILE RECORDS, Saukville, Wisconsin
High Quality Disc Recordings for Wide Range Equipment

BALLANTINE LABORATORIES, INC., Fanny Rd., Boonton, New Jersey
Electronic Voltmeters, Decade Amplifiers, Voltage Calibrators, Multipliers, Shunts

THE DAVEN COMPANY, 530 West Mount Pleasant Ave., Route 10, Livingston, New Jersey
Attenuators, Potentiometers, Resistors, Rotary Switches, Test Equipment

ELECTRO-VOICE, INC., Buchanan, Michigan
Microphones, Pickups, Speakers, Television Boosters, Acoustic Devices

FAIRCHILD RECORDING EQUIPMENT CO., Whitestone 57, New York
Audio Amplifiers, Moving Coil Cartridges, Transcription Arms, Turntables, Professional Disc Recorders

FREED TRANSFORMER COMPANY, INC., 1718 Weirfield St., Brooklyn 27, New York
Transformers, Reactors, Filters, Magnetic Amplifiers, and Laboratory Test Equipment

(Please see inside back cover for additional listings)