IRE Transactions on AUDIO

Volume AU-4 NOVEMBER-DECEMBER, 1956 Number 6

Published Bi-Monthly

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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

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PGA News

PGA CANDIDATES FOR 1957

Listed below are the biographies of all candidates who appear on the 1957 ballot of the PGA, which is now being mailed, for the offices of Chairman, Vice-Chairman, and three Administrative Committee member posts.

Michael Copel was born in Paris, France, in 1916. He received the B.S. degree in 1935 from the University of Paris and the E.E. degree in 1937 from the Conservatoire National des Arts et Métiers in Paris. Mr. Copel 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 Laboratories. He was senior engineer at Dictograph Products, Inc. from 1946 to 1948.

Since 1948 M. Copel has been engaged in investigations, developments, and evaluations of audio communication systems at the Naval Material Laboratory in Brooklyn, N. Y., where he holds the position of supervising electronics scientist (electro-acoustic).

Mr. Copel is a member of the Acoustical Society of America and the IRE. He is presently serving on the IRE Electro-Acoustic Committee and the American Standard Association Committee Z24 W22 on Loudspeaker Measurements. In addition, Mr. Copel is Chairman of the PGA Ways and Means Committee. He organized the Audio Sessions of the 1955 and 1956 National Conventions.

Herbert Heller was born May 27, 1923 in Vienna, Austria. He received the degree of Bachelor of Science in electrical engineering with honors from Case Institute of Technology, Cleveland, Ohio, in 1950.

His activities include experiments in stereophony with the Cleveland Orchestra; 2-channel, 1955–1956, and 3-channel, 1956–1957. He is presently senior project engineer in the Transducer Section of Clevite Research Center in Cleveland.

Mr. Heller was Chairman of the Regional PGA Program Committee in the Midwest during 1954–1956. He was organizer of the Cleveland PGA in 1953, and was Chairman of the group during 1954–1955.

Harry F. Olson attended the University of Iowa where he received the B.E. degree in 1924, the M.S. degree in 1925, the Ph.D. degree in 1928, and the professional degree of E.E. in 1932.

Dr. Olson, who is presently Director of the RCA Acoustical and Electromechanical Research Laboratory, Princeton, N. J., joined that company in 1928. He holds more than 60 patents on devices and systems and is the author of more than 70 professional papers. His books include "Dynamical Analogies," "Elements of Acoustical Engineering," and "Musical Engineering."

He is a Fellow of the Society of Motion Picture and Television Engineers, and a member of the American Physical Society, IRE, the Acoustical Society of America, and the Audio Engineering Society. He is past president of the Acoustical Society of America.

For his contributions in the audio field, Dr. Olson received the John H. Potts Medal of the Audio Engineering Society in 1952, the Samuel L. Warner Medal of the Society of Motion Picture and Television Engineers in 1955, and the John Scott Medal in 1956.

H. E. Roys received the B.S. degree in electrical engineering from the University of Colorado in 1925. He was with the General Electric Company in Schenectady, N. Y., doing developmental work on radio transmitters and receivers until 1930, when he transferred to RCA at Camden, N. J. In 1931, he became associated with the Phonograph Section. Since then, he has been active in the disk and magnetic recording field. From 1941 to 1946, he was located in Indianapolis working mainly on disk recording and reproducing problems. Returning to Camden in 1946, he participated in the development of "45" and LP records. He became more active in the magnetic recording field and recently participated in the development of wideband recording equipment. In July, 1956, he returned to Indianapolis as Manager of Engineering, RCA Victor Record Division.

Mr. Roys is a member of Tau Beta Pi and Eta Kappa Nu, and has approximately twenty published papers to his credit. He is a member of IRE Recording and Reproducing Committee and served as Chairman during 1950–1952. He has served on NARTB, CCIR, and ASA committees and was recently appointed Chairman of the RETMA Phonograph Combinations and Home Recording Devices Committee. He is a Fellow of AES, the Acoustical Society of America, and IRE. He served as Chairman of the Philadelphia Section of PGA during 1954–1955.

Walter T. Selsted, a graduate of the University of California, received the B.S. degree in electrical engineering in 1944. The same year he was employed by the University's Radiation Laboratory for work on Manhattan Project. In 1946 he became chief engineer for Pacific Broadcasting Company. Mr. Selsted was consultant to Ampex Corporation in 1948, when the company decided to manufacture tape recorders. In 1949 he became chief engineer, and in 1955 director of Research for Ampex.

Mr. Selsted is a member of IRE and SMPTE. He holds many patents in the magnetic recording field.

Frank H. Slaymaker, born April 22, 1914, attended the University of Nebraska and received the B.S. degree in electrical engineering in 1941 and the E.E. degree in 1946. He joined Stromberg-Carlson as a professional engineer in 1941, becoming senior professional engineer in 1950, and chief engineer in the Sound Equipment Division in 1951. At present, he is Manager of Electroacoustics in the Research and Advanced Development Department.

Mr. Slaymaker is a member of Sigma Tau, Sigma Xi, and Pi Mu Epsilon, and also of the SMPTE, IRE, (Audio Techniques Committee), and the American Standards Committee on Standardization of Carillon Bells. He is a registered Professional Engineer in New York, and holds several patents. His articles in the audio field have appeared in four leading professional journals.

Weiant Wathen-Dunn was born in New York, N. Y., on April 27, 1912. He attended Wesleyan University, receiving the B.A. degree in 1934 and the M.A. degree in 1936. He taught physics and mathematics in Robert College, Istanbul, Turkey, during 1936–1940, and taught physics at Williston Academy, East Hampton, Mass. during 1940–1941.

In 1941 he joined the Naval Research Laboratory in Washington working on problems of airborne sound and calibrated sound recording. Since 1951 he has been employed with the Air Force Cambridge Research Center, Bedford, Mass., working on a project on transmission of digitized information, and on another for reducing channel capacity required to speech transmission.

Philip B. Williams was born in Ancon, Canal Zone, in 1916. After receiving the B.A. degree in physics and mathematics, he studied engineering at RCA Institutes and Illinois Institute of Technology. He was engaged in AM broadcast and early fm engineering at Station KBTM, Jonesboro, Ark., and Zenith Radio Corporation, and joined Jensen Manufacturing Company for work in audio and acoustics in 1942. Mr. Williams became senior development engineer in 1950, and in that capacity was responsible for advanced development of Commercial and military transducers. As Jensen's chief engineer, he directs electroacoustics research and transducer development.

Mr. Williams is a council member of Chicago Audio and Acoustic Group, a member of the Audio Engineering Society, the American Institute of Electrical Engineers, and a Senior Member of IRE. He is past Chairman of the Chicago chapter of the PGA, past News Editor, Scanfax, and Vice-Chairman of the Chicago IRE Publicity Committee. His industry activities include IRE Electroacoustic, RETMA Speaker, and RETMA High Fidelity Committees. He has presented various papers to professional groups, and has published articles on loudspeakers and enclosures. He is presently Program Chairman of the PGA.

PGA CHAPTER ACTIVITIES

Cleveland, Ohio

Carmen P. Germano, 540 E. 105 St., Cleveland, Ohio is the 1956–1957 Chairman of the Cleveland Chapter PGA. Please modify the list of Chapter Chairmen, published in the September–October issue, accordingly.

Dayton, Ohio

The following 1956–1957 officers have been announced for the Dayton PGA Chapter:

T. P. Mountz, Jr.
Chairman

Edward Lazur
Vice-Chairman

Albert Parker
Secretary

Albert Peters
Program

Philadelphia, Pa.

N. Johnson reports a dinner and meeting held by the Philadelphia PGA Chapter on May 19. Attendance at the dinner was 103 and at the meeting 120. The meeting was held at the Leeds and Northrup plant at North Wales, Pa. The speakers of the evening were G. E. Beggs, Jr., E. H. Rogge, G. H. Semblin, F. A. Cline, Jr., and W. P. West, all of the Leeds and Northrup Company.

Following a brief introduction by E. H. Rogge, during which he outlined the scope of the new plant's operation, the other gentlemen described the design and construction of the "Safetemp System." The name is a coined one for the Leeds and Northrup night-lighting and public address system. The talks were very interesting and the trip through the plant was well received.

At the annual business meeting held in conjunction with this meeting the following officers were elected for the 1956–1957 year:

Lowell Good, RCA
Chairman

C. D. O'Neal, Philco
Vice-Chairman

R. C. Dorris, Gawler-Knoop
Secretary

On October 10, this Chapter saw a new film (September, 1956) showing the complete story of modern records from the recording session in Boston's Symphony Hall to the pressing and packaging of the finished product.

George Marek commented on the selection of artists and repertoire for classical and popular music and on record distribution. A. A. Pulley and H. G. Roys spoke on the engineering aspects of the record business. All three of these men are associated with the RCA Victor Division of the Radio Corporation of America. George Marek is Vice President of the Record Albums Department. A. A. Pulley is Administrator of General Recording and H. G. Roys is Manager of Engineering. The meeting was held at the WCAU Studios in Philadelphia.

San Diego, Calif.

Larry La Zelle reports that meetings were held on May 17 and on June 21. The May 17 meeting was held at the recording section of the U. S. Electronics Laboratory in San Diego. The program was built around a tour of the recording section facilities. The manner in which sound is added to a motion picture was demonstrated using some of the guests as actors. Immediate playback was available, interlocked with rerun of the picture to permit the actors to see how they made out. The June 21 meeting was a Ladies Night and the subject was Electronic Music. A demonstration was given of the Hammond Organ with Plectro and Harp Celeste. This program was enjoyed by all.

San Francisco, Calif.

At a meeting held on September 18, in the Physics Building at Stanford University, Dr. Vincent Salmon of the Stanford Research Institute gave a talk on "The SRI Modulated Air-Stream Loudspeaker."

Syracuse, N. Y.

On September 12 the Syracuse PGA Chapter sponsored a joint meeting with the IRE Syracuse Section. The meeting was held at Syracuse Museum of Fine Arts. At this meeting Jerry B. Minter, President of the Components Corporation of Denville, N. J., spoke on "Recent Developments in Recording and Reproduction of Music." Mr. Minter discussed some of the basic limitations and faults of past recording and reproduction systems. He demonstrated a new type of amplifier and speaker system offering more realistic dynamic range and better distribution of sound through a room. A stereophonic demonstration of the equipment was included.

WITH OTHER ACOUSTICAL AND AUDIO SOCIETIES

The July, 1956 issue of the *Journal of the Acoustical Society of America* contains 37 papers and 14 Letters to the Editor many of which will be of interest to the members of the IRE Professional Group on Audio. The bulk of the issue is devoted to the papers of the Speech Communication Research Symposium held at the Navy Electronics Laboratory in San Diego on November 21–22, 1955.

Among the most notable papers is, "Articulation Reduction by Combined Distortions of Speech Waves," by our National Chairman, D. W. Martin and co-authored by R. L. Murphy and Albert Meyer, all of the Baldwin Piano Company, Cincinnati, Ohio.

"Previous investigators have shown that speech waves can undergo any one of a number of severe forms of distortion in low ambient noise levels without serious reduction of word articulation. There are well-known notable exceptions (*e.g.*, center clipping). However, it is not enough to avoid these exceptional forms of distortion. In this study it has been demonstrated that combinations of speech-wave distortions, which individually are quite innocuous with regard to word articulation, can be devastating in their combined effect, even in the absence of serious noise."

T. J. Schultz of the Acoustic Laboratory, Harvard University published a paper on "Acoustic Wattmeter" which is a device giving pointer indications of acoustic intensity over a 50-db range at frequencies up to 10,000 cps. The signals from a pair of pressure-sensitive condenser microphones, mounted back-to-back in a space roughly the size of a dime, are treated electronically to provide pressure and velocity signals which are then multiplied and averaged in a specially moving coil dynamometer to give precise pointer readings.

Joseph Hershkowitz of the U. S. Naval Material Laboratories, Brooklyn, New York in a Letter to the Editor describes a probe tube for the WE640AA Condenser Microphone which provides a smooth, readily equalized, frequency response to 9000 cps.

The issue contains a number of other papers of outstanding interest to Audio Technologists, and it has its usual excellent References to Contemporary Papers on Acoustics by Robert N. Thurston of the Bell Telephone Laboratories, and its comprehensive review of Acoustical Patents by a panel of outstanding reviewers under the Chairmanship of Robert W. Young of the U. S. Navy Electronics Laboratory in San Diego, California. An added feature is the complete program of the joint meeting of the Acoustical Society of America with the Second International Congress on Acoustics held in Cambridge, June 17–21, 1956.

The January, 1956 issue of the *Journal of the Audio Engineering Society* was just received, and it contains 7 papers which will be of interest to members of the PGA.

K. N. Stevens of the Acoustical Laboratory, M.I.T., has reviewed "Synthesis of Speech by Electrical Analog Devices."

Charles A. Wilkins of the David Bogen Company treats the "Control of Amplifier Source Resistance." "General equations are developed for the design of any type of circuit utilizing current-voltage feedback control of amplifier source impedance, whether it be fixed, semi-variable, or continuously variable."

Walter O. Stanton of Pickering and Company describes a program to design a high quality pickup which is a study of various factors taken into account in the design of the Pickering Fluxvalve Pickup.

Several papers deal with the properties, applications, and measurements of magnetic tapes.

Benjamin B. Bauer

Terminated Horn Enclosures*

WILLIAM E. GLENN†

Summary—The characteristics of a finite exponential horn terminated in a physically realizable impedance have been calculated on an IBM 650 computer. Some of the results of the calculations and some experimental results of tests on such a terminated horn are presented.

Introduction

THE ADEQUATE reproduction of low frequencies in music is a major problem in high-fidelity systems. Below 100 cycles the size of the speaker system can be considered to be small compared to wavelength. To radiate low frequencies with a directly radiating speaker requires a large speaker, large enclosure, and a high speaker cone excursion.

Horn-type enclosures have become quite popular in recent years as a method of increasing the distortion-free radiation in low frequencies. The transformer-like action of a horn allows more sound power radiated from the system for a given speaker excursion. Or, for a given sound output there is much less distortion. Added impetus was given to horns by Klipsch. His use of the volume between the back of the enclosure and the wall gives an effective enclosure volume of about twice the actual volume.

In the past, analyses of horn enclosures have generally been based on the data of an infinite horn or a horn whose mouth is large compared to wavelength. In practical systems this analysis does not apply. At low frequencies, the size of the mouth of the horn is necessarily very short compared to wavelength. Consequently, the acoustical impedance of the air load does not match the impedance of the mouth of the horn. A condition then results which is somewhat analogous to that of a transmission line which is not terminated in its characteristic impedance. Since the energy is not all absorbed at the mouth, some of it reflects back down the horn and tends to either reinforce or cancel the original signal. This results in large fluctuations, with changing frequency, in the impedance at the throat of the horn. Fluctuations in the horn throat impedance result in nonuniform frequency response and in high speaker excursion at some frequencies with its associated distortion.

A method of greatly reducing these variations is to terminate the mouth of the horn in its characteristic impedance. This can be done by inserting sound-absorbing padding. As in the transmission line, if the acoustical resistance of the padding is equal to the characteristic impedance of the mouth of the horn, no energy will reflect back down the horn and there will be no fluctuations in impedance at the throat. Adding absorbing material does not necessarily mean less sound output. It will lower "resonant" peaks in the frequency response, but by referring to Fig. 1 it can be seen that it actually increases the output at very low frequencies.

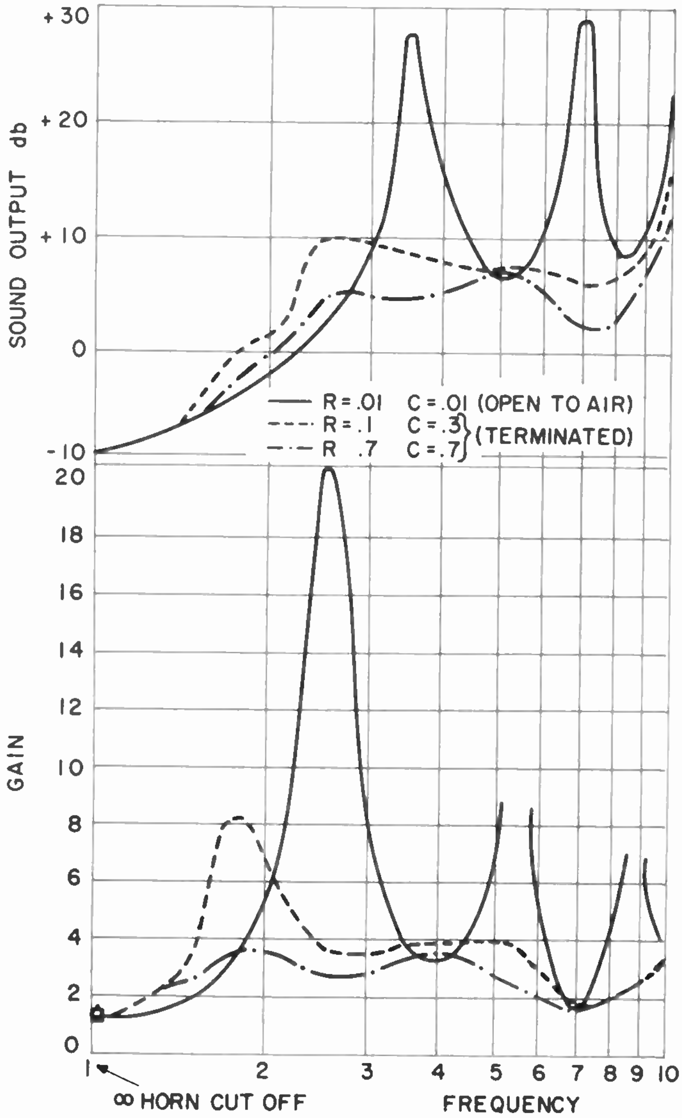

Fig. 1.

Theory

The equations for a finite terminated exponential horn were computed on an IBM 650 computer. A more general shape than a simple exponential was first computed but, after assigning factors of merit to the enclosures, it was found that the exponential was about optimum.

The input impedance of an infinite exponential horn becomes reactive near cutoff. This reactive component

* Manuscript received by the PGA, June 7, 1956.
† General Electric Res. Lab., Schenectady, N. Y.

can be approximated by lumped acoustical elements. Freely hanging cloth has a resistive component due to the viscosity of the air passing through it and a reactive component due to the mass of the cloth. In the computations this type of termination was assumed. The impedances were normalized such that one was the high-frequency characteristic impedance of the mouth of the horn. The reactive element C of the termination was unity if it had an impedance of unity at the horn cutoff frequency.

The equation for the throat impedance of a finite exponential horn is[1]

$$Z_1 = \frac{R\cos(\beta L + \theta) + j(X\cos(\beta L + \theta) + \sin\beta L)}{-X\sin\beta L + \cos(\beta L - \theta) + jR\sin\beta L}$$
$$= R^1 + jX^1$$

where

$$R = \frac{r}{r^2 + \dfrac{1}{C^2x^2}}$$

$$X = \frac{\dfrac{1}{Cx}}{r^2 + \dfrac{1}{C^2x^2}}$$

$$x = \frac{\lambda c}{\lambda}$$

λc = horn cutoff frequency

λ = wavelength considered

r = normalized terminating resistance

c = normalized terminating reactance (corresponds to mass)

$$\beta = \frac{2\pi}{c}\sqrt{x^2 - 1}$$

$$\theta = \tan^{-1}\frac{1}{\sqrt{x^2 - 1}}$$

L = length of horn.

The factors that we would like to determine are the frequency response and gain of the enclosure. The gain will be defined here as the factor that the excursion of the speaker cone is reduced for the same sound radiated as in an infinite baffle.

It can be shown that the gain G of an enclosure including the radiation from the unloaded side of the driver is

$$G = \left| \sqrt{\frac{R^1 A_m}{RA_t}} - 1 \Big/ \frac{360 L x^0}{\lambda_c} \right|$$

where

A_m = the mouth area of the horn

A_t = the throat area of the horn.

If the driver produces a constant force regardless of frequency the normalized sound output 0 for an enclosure small compared to wavelength will be

$$0 = \frac{Gx}{Z_1}.$$

A large number of enclosures were computed and factors of merit were assigned such that for a given performance the factor would be inversely proportional to the volume of the enclosure. The enclosure design turned out not to be particularly critical but the one that seemed to be of most interest was a simple exponential with a ratio of mouth to throat area of about 6 to 1. The theoretical frequency response and gain are plotted for different amounts of termination in Fig. 1.

Experimental Results

An enclosure was constructed as shown in Fig. 2. The enclosure was designed such that the reactance of the speaker compliance would cancel the reactance of the horn throat impedance near cutoff and give an effective cutoff that is slightly lower than that obtained from the calculated frequency response curves. The amount of termination used was strictly empirical. The terminating material was about five pounds of burlap. The frequency response and gain is shown in Fig. 3. There is reasonably good agreement between the theoretical and measured curves. The measured gain was lower than was expected. There are several possible contributing factors to this:

1) The presence of folds in the horn;
2) Transmission loss through terminating material;
3) The finite length of the terminating material;
4) Compliance of the walls of the horn.

The horn enclosure has a flat frequency response over a frequency range of about 3 to 1. The horn must, therefore, be cut off sharply at a high frequency and allow the speaker to radiate directly above this frequency. This can be done by lumping the terminating material in two places in the horn. The frequency responses of the horn terminated in this manner, the unterminated horn, and the speaker in an infinite baffle are shown in Fig. 3. With multiple termination the velocity of propagation down the horn was found to be about 15 per cent slower than in space. This allows one to design the horn this much shorter for a given cutoff frequency but the horn cross-sectional area must be increased by the same factor in order to compensate for the change in impedance.

[1] H. F. Olson, "Elements of Acoustical Engineering," D. Van Nostrand Co., New York, N. Y., p. 105; 1947.

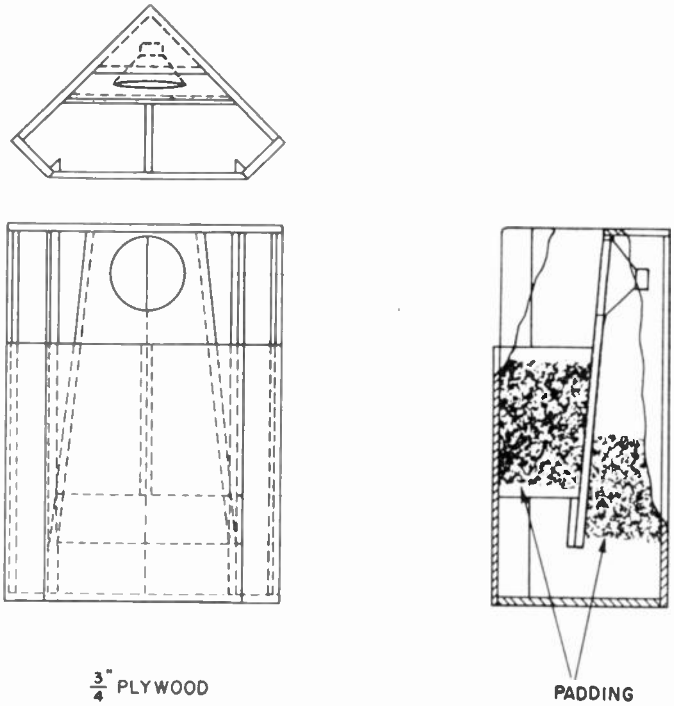

Fig. 2.

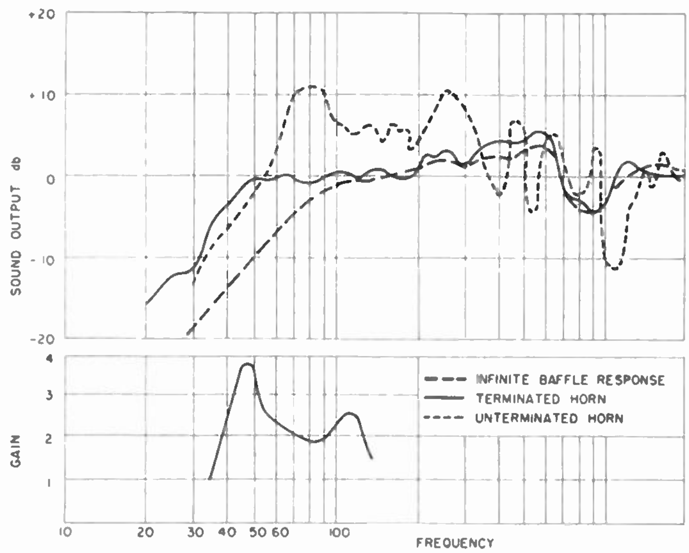

Fig. 3.

The design parameters for this particular horn are given in Table I. The horn throat area is chosen such that the output of the horn at low frequencies will match that of the speaker at high frequencies. Now that the problem has been set up for the computer it is a simple matter to determine the theoretical frequency response, gain, design parameters, and several other factors of any speaker enclosure which can be expressed as one or two exponential sections in cascade with an arbitrary termination of the form used.

It may be concluded from the above analysis that horn type speaker enclosures can be greatly improved by the use of proper termination. There is need for more work on the choice and placement of a terminating material and on understanding and possibly removing the discrepancy between the experimental and theoretical gain curves.

TABLE I

Design Parameters for an Exponential Horn with a Mouth to Throat Area Ratio of 6 to 1 and a Termination of $r=1$, $c=3$

Speaker Size (Inches)	Effective Cone Mass (Grams)	Effective Horn Cutoff	Minimum Enclosure Volume (Cubic Feet)		Horn Throat Area (Square Feet)		Horn Length (Feet) (Without Annulling)	
			Without Reactance Annulling	With Reactance Annulling	Theory	Practice	Theory	Practice
8	8	45	5	4.3	0.18	0.21	7.7	6.7
12	19	45	13	11.0	0.62	0.72	7.7	6.7
15	55	30	26	23.0	0.75	0.86	11.5	10.0

An Experimental 9000-Watt Airborne Sound System*

D. W. MARTIN†, A. MEYER†, R. K. DUNCAN†, AND E. C. BROXON†

Summary—**An experimental 9000-watt speech announcing system AN/AIC-11(XA-1) was developed for installation in a B-26 aircraft. The system was used for studies of direct communication through the atmosphere to ground personnel from aircraft operating at relatively high altitude. The equipment consisted of a turbine generator type of auxiliary power unit; three 3000-w amplifiers, each driving a separate twin-horn loudspeaker; signal preparation, control, and monitoring units; a loudspeaker mounting frame which rotates the loudspeakers and supports two of the twin horns outboard from the fuselage; and magnetic tape recorders.**

INTRODUCTION

THE EXPERIMENTAL 9000-w airborne sound system to be described was intended for the investigation of direct acoustical transmission of information to listeners on the ground 6000 feet below an aircraft moving at a speed of 160 mph. There are various tactical, operational, psychological, and rescue purposes for which successful transmission under these conditions would be useful.

Small public address systems (having an audio power of about 100 w) have at times been used successfully in small civilian aircraft for advertising purposes. They have generally operated over small towns where the ambient noise level is low and the area of coverage is small, simply by circling over the target area at minimum engine power, airspeed, and altitude, and directing the loudspeaker horn toward the center of population.

When larger noisier areas are to be covered more rapidly, or when higher altitude and speed are operationally necessary, it becomes much more difficult to obtain satisfactory speech transmission through the medium and adequate sound level at the ground. The reasons for this difficulty have been given in a separate paper,[1] which reports data obtained from the system described here and earlier systems. Important factors include turbulence and gross inhomogeneity in the medium (resulting in random fluctuations in level), the effect of source motion in speeding up the fluctuations, multipath interference, normal attenuation with distance, high-frequency absorption in the medium and Doppler effect. The effect of combined controlled distortions of these types upon the intelligibility of speech was investigated experimentally in the laboratory and has been reported in detail elsewhere.[2]

Operating conditions for the equipment were influenced greatly by the type of aircraft which had been selected. The choice of B-26 had been made because of tactical reasons and the availability of the aircraft for the purpose at the time the project began. An announcing system of lower power, which had been used in a B-26 with limited success in Korea, was intended to be improved or superseded by this project. The end result sought was "good intelligibility for at least 30 seconds at any point within $\frac{1}{4}$ mile of the aircraft's flight path projection on the ground."

SYSTEM PLANNING

Preliminary calculations and measurements on the earlier system indicated that its efficiency was good; that its intelligibility (as recorded several feet below the loudspeakers during flight) was high; that the intelligibility of speech received at ground level was submarginal; and that the power-delivering capacity needed to be increased by an order of magnitude if possible. By installing a new type of gas turbine, which had recently been developed by the Solar Aircraft Company, it appeared possible to obtain sufficient primary power for a 9000-w audio system.

The only high-power audio amplifiers available in the power range needed for this project were many times too heavy for airborne operation. Therefore it was necessary to develop a power amplifier specifically for the purpose. For reasons given later it was decided to divide the power amplification system into three parts.

Fig. 1 is a block diagram of the experimental system. Starting, monitoring, and stopping the Solar gas turbine was accomplished by the operator through the use of an engine control box. The starter motor for the turbine was used as a dc generator after the engine has reached operating speed. This change in function was accomplished by the dc power control box. Direct current was supplied to a 60-c inverter for 110-v alternating current for the tape recorder-playback unit and the test equipment power outlets in the control cabin. Twenty-eight volts dc from the turbine generator was also supplied through the amplifier power control panel to the filament circuits of the three 3000-w amplifiers. Small amounts of dc power were also used for the signal preparation unit and for the motors of the loudspeaker rotator.

* Original manuscript received by the PGA, September 17, 1956. Work done under government contracts AF33(038)-23313 and AF33(616)-2320.

† Baldwin Piano Co., Cincinnati 2, Ohio.

[1] D. W. Martin, R. K. Duncan, and R. L. Murphy, "Vertical Atmospheric Transmission from a Moving Sound Source," paper presented at the Second Int'l. Congress on Acoustics in conjunction with the 51st Meeting of the Acoust. Soc. of Am., Cambridge, Mass.; June, 1956.

[2] D. W. Martin, R. L. Murphy, and A. Meyer, "Articulation reduction by combined distortions of speech waves," *J. Acoust. Soc. Am.*, vol. 28, pp. 597–601; July, 1956.

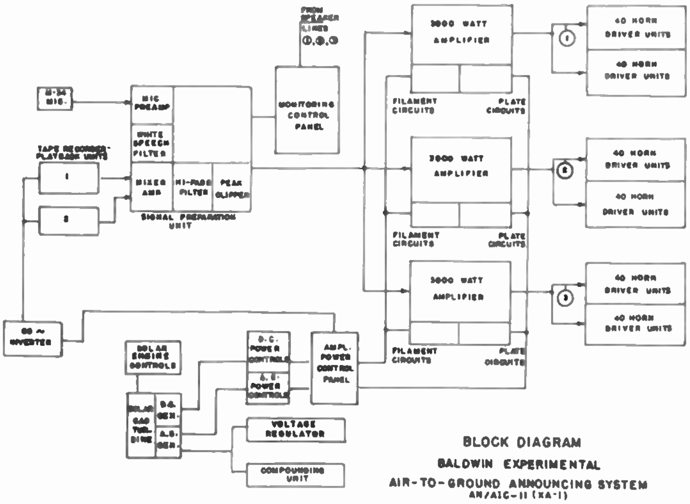

Fig. 1—Experimental 9000-watt airborne sound system—block diagram.

A 400-cycle 15-kva Eclipse-Pioneer alternator was mounted on the turbine. Associated with this was a voltage regulator and a compounding unit. AC power to the power amplifiers was controlled through relays actuated by switches on the amplifier control panel. The output from each amplifier was supplied to two banks of 40 horn-driver units each, each bank driving a separate half of a twin horn. The central twin horn was mounted in the under side of the bomb bay, and the other two twin horns were mounted outboard. Either prerecorded tape material or direct speech from the operator using an M-34/AIC microphone could be supplied to the power amplifiers through the signal preparation unit. A standard Air Force preamplifier and dynamotor of the AN/AIC-10 type were mounted in the signal preparation unit. The output of the preamplifier passed through a special electrical filter which equalized the transmitted speech spectrum to a uniform speech spectrum, so that the maximum practical pre-emphasis of high-frequency energy would be supplied to the power amplifiers. Both tape and microphone signals passed through a high-pass filter which protected the loudspeakers, and a peak clipper which protected the amplifiers. The monitoring control panel permitted the operator to check signal levels at all necessary points in the signal preparation unit, and to monitor the output of each power amplifier both by vu meter and by headset listening.

System Development

Auxiliary Power Unit

The aircraft engine generators could not supply the necessary primary power. Extra pads on each engine were not suitable for mounting alternators of the required capacity, and conventional auxiliary power units having sufficient capacity were much too heavy and large to be considered for installation in the B-26 aircraft. A Solar type M-1 gas turbine engine (Mars 50 hp) had recently become available. In addition to the 15 kva, 28-v dc generator included in the M-1 equipment, an Eclipse-Pioneer type 28EO3-3-B alternator was driven by the turbine engine. Fuel was obtained from the reserve fuel tank of the aircraft. The entire auxiliary power plant was installed in the rear turret compartment after the turret was removed. Installation of the turbine (and of other equipment described in this paper) was accomplished by WADC personnel. The Solar equipment performed well for the duration of the project with only normal maintenance and care.

Signal Preparation Unit

An obvious correction for the high-frequency losses in the atmosphere would be to pre-emphasize the speech spectrum at high frequencies. However, the atmospheric losses are far greater than the amount of droop in a normal speech spectrum. To pre-emphasize the high-frequency components beyond a uniform speech spectrum condition[3] would be a poor compromise, because it would necessitate lowering the amount of power transmitted by the system in the frequency range where transmission is best. In order to achieve the uniform speech spectrum condition, an equalizer filter was designed having a response characteristic inverse to the speech spectrum transmitted by the M-34/AIC microphone from a typical speaking voice. This equalizer was installed in the signal preparation unit, just following the microphone preamplifier, and ahead of the mixer amplifier stage, so that it would have no effect upon prerecorded tape material.

Both speech signal and tape playback pass through a 400-cps high-pass filter which prevents the loudspeakers from being driven excessively below horn cutoff frequency. This was followed by a symmetrical peak clipper. The threshold of peak clipping was adjusted to protect both the power amplifiers and the loudspeaker groups from peak overload.

Fig. 2 is a schematic diagram of the signal preparation unit. Not shown are the microphone (and its press-to-talk switch), the microphone-recorder selector switch, the monitoring headset, the vu meter, and the meter circuits and selector switch. The standard Air Force interphone components are indicated by blocks only. The speech spectrum equalizer is shown at upper right, the mixer amplifier at lower left, the high-pass filter at lower center, the peak clipper at lower right, and a separate monitoring headset amplifier at upper left (which amplifies from meter-circuit level up to a level suitable for listening in aircraft noise). Note that metering and monitoring connections are provided at various circuit points.

[3] D. W. Martin, "Uniform speech-peak clipping in a uniform signal-to-noise spectrum ratio," *J. Acoust. Soc. Am.*, vol. 22, pp. 614–621; September, 1950.

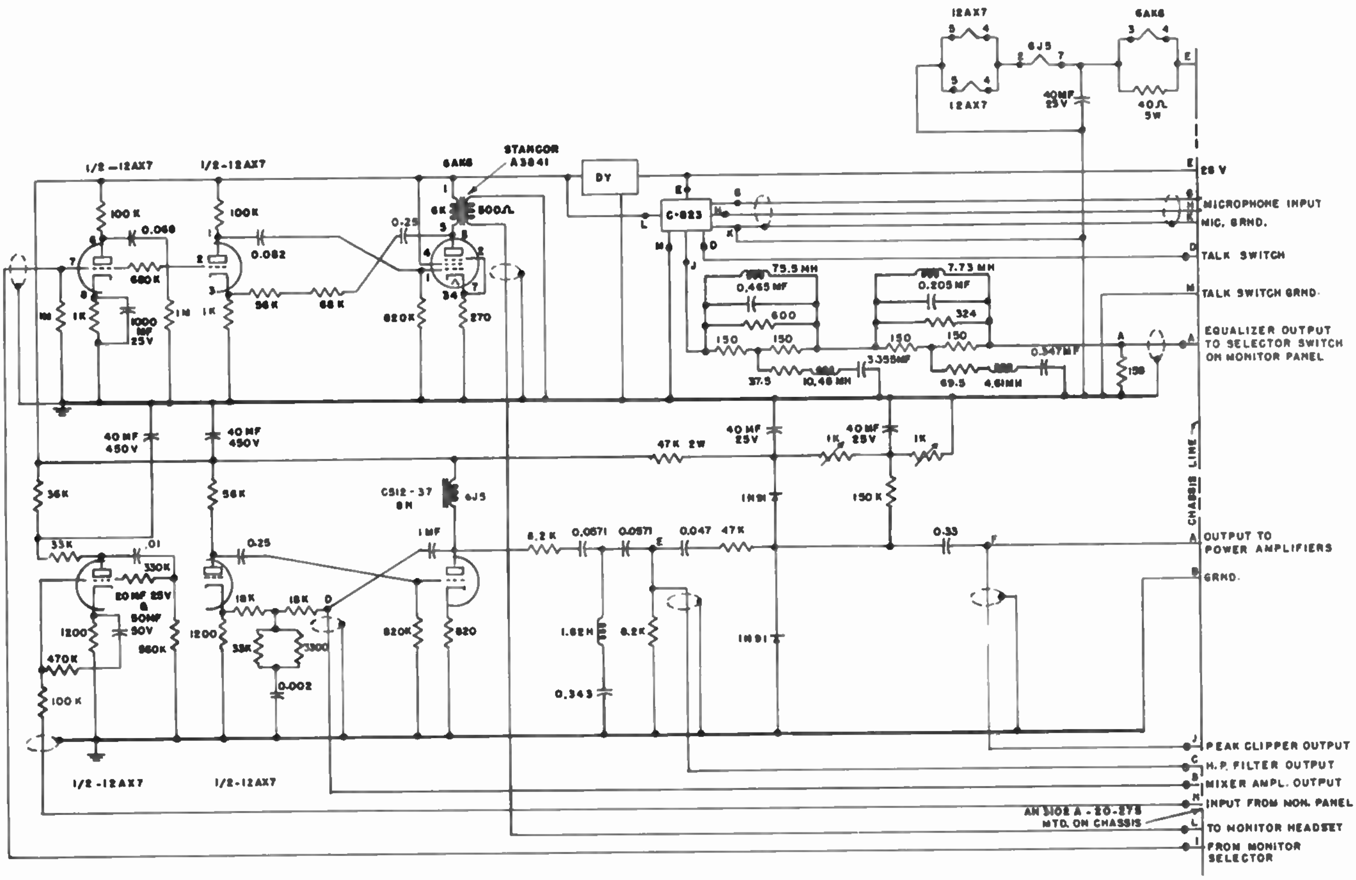

Fig. 2—Signal preparation unit—schematic diagram.

3000-Watt Amplifier Unit

For maximum electrical dependability, and because it appeared physically necessary to divide the loudspeaker system, it was decided to divide the 9000-w system into three equal parts. Initially, triodes were decided upon for the output stage of the power amplifier, because of the low source impedance obtainable even without feedback. It was not certain that feedback could be used at all without complicated correction to insure unconditionally stable operation. Tube type 304TH was chosen because of the very low driving power required (according to published characteristics), and the small number (four) of tubes required for 3000-w power-delivering capacity, minimizing parasitic problems. For the 3-kv plate supply for the output stage four 872A's in a bridge circuit with a swinging choke (5–25h) and a 2 mfd filter capacitance provided more than adequate supply and filtering. A schematic of the 3000-w amplifier is shown in Fig. 3 (opposite). Each half of the output transformer consisted of 295 turns of No. 20 heavy Formvar. Each half of the secondary consisted of twelve parallel turns of No. 12 heavy Formvar. The laminations consisted of a five-inch stack of No. 29 Allegheny Ludlum Audio A, El-19. The over-all dimensions of the transformer were 10×7×5 inches. After the first model was constructed it was found that considerably increased driving power would be required. This was confirmed by revised specifications from the manufacturer of the output tubes. A new type of audio power amplifier,[4] which had been developed at Baldwin for possible commercial use, was adapted to the needs of the driving amplifier stage. The commercial amplifier circuit developed by Bereskin was adapted to give 70 w, using two 807's with a specially designed bifilar output transformer for driving the 304TH's, and a special 400-cps power transformer.

After these changes it was determined that the power output capability was 3200 w, and the 400–4000 cps response at this level was flat to within plus or minus 0.1 db. However, there was considerable distortion caused by variation in the bias voltage moving the operation slightly into class C at full power. Installation of a triode bias control tube reduced this variation, so that at 3000 w the harmonic distortion was reduced to 10 per cent.

2.5 μh chokes were inserted into each plate lead in the output stage, and were completely effective in stabilizing the operation. Hum and noise voltage at the 8-ohm output was less than 0.25 v rms, compared to 155 v at 3000 watts output. In the actual installation the hum voltage was somewhat higher because of system wiring. Only a few db of feedback could be applied while maintaining stability, so none was used in the final model. All power switching was accomplished by relays installed on the underside of the chassis.

[4] A. B. Bereskin, "A high-efficiency, high-quality audio-frequency power amplifier," IRE Trans., vol. AU-2, pp. 49–60; March, 1954.

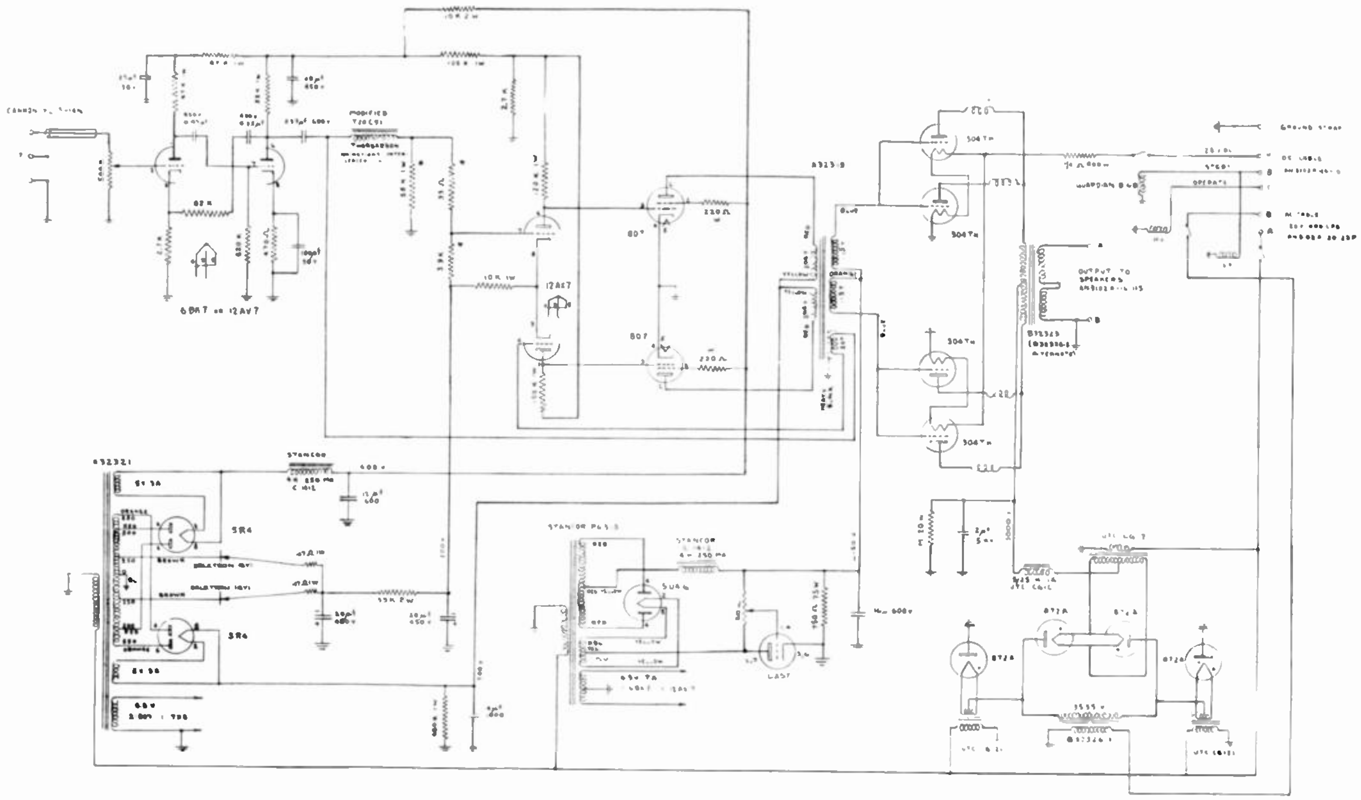

Fig. 3—3000-watt power amplifier—schematic diagram.

Twin-Horn Loudspeakers

The large acoustical power-delivering requirement necessitated the use of horn-driver units having maximum power-delivering capacity per pound of driver mass. Random noise (500–4000 cps band) was used in a series of life tests on individual driver units at various input powers. The sound pressure output level was monitored to determine the relative power-delivering capacity of each type of driver which appeared to meet the requirements. The University LB-35 unit (having a metal voice coil form with serrated edges) was adopted on the basis of these tests. This type of unit (when suitably horn loaded) operated for periods of several hours at 37 w. Higher powers did not yield corresponding increases in sound pressure level. The units failed due to heating after one-half hour at 45 w. On the basis of these tests the number of driver units required per 3000-w amplifier was estimated to be 80, or a total of 240 units in the system. At this level, however, loudspeaker distortion is high at high frequencies.

The desirability of connecting all of these driver units to a single horn (to minimize interference effects in the directional pattern) was recognized from the beginning. However, a number of other considerations prevailed, both theoretical and practical. It was recognized that nonlinear distortion would be a factor, and it was desired to keep this to a reasonable percentage if possible. At high frequencies the distortion is a function of the low-frequency cutoff of the horn,[5] and it is advisable to place this cutoff (determined by the rate of horn flare) near the lowest frequency that it is planned to reproduce from the horn. The rate of flare chosen in the design corresponds to a low-frequency cutoff of 400 cps.

Mechanical layouts were made for the throat block on which horn units would be mounted. In the design, it was necessary to keep the driver units as close together as physical dimensions would practically permit; to make the horn length within the throat block short, so that the rapid rate of flare would not cause the horn cross-section dimensions to exceed a wavelength where the short horn sections join at the throat of the horn extension, at the highest frequency to be reproduced; and to keep the design practical from a maintenance standpoint.

Fig. 4 shows the throat block adopted, in two sectional views. Half of the block is shown in cross section through the horn throats, and the other half is shown through the solid block between throats. The horn sections were of uniform width inside the throat block, with the exponential expansion occurring in planes at right angles to this view. Rail construction, commonly used in the piano industry, was adopted in order to simplify the construction. Driver units were mounted on aluminum bars running lengthwise of the throat block. Terminals for loudspeaker wiring were mounted on brass strips running lengthwise between rows of driver units. The throat block itself was constructed of wood rails of triangular cross section, between which were glued curved pattern blocks with suitably tapered edges to give the desired horn contour. In a design for produc-

[5] M. Y. Rocard, "Sur la propagation des ondes sonores d'amplitude finie," *C. R. Acad. Sci., Paris*, pp. 161; January, 1933.

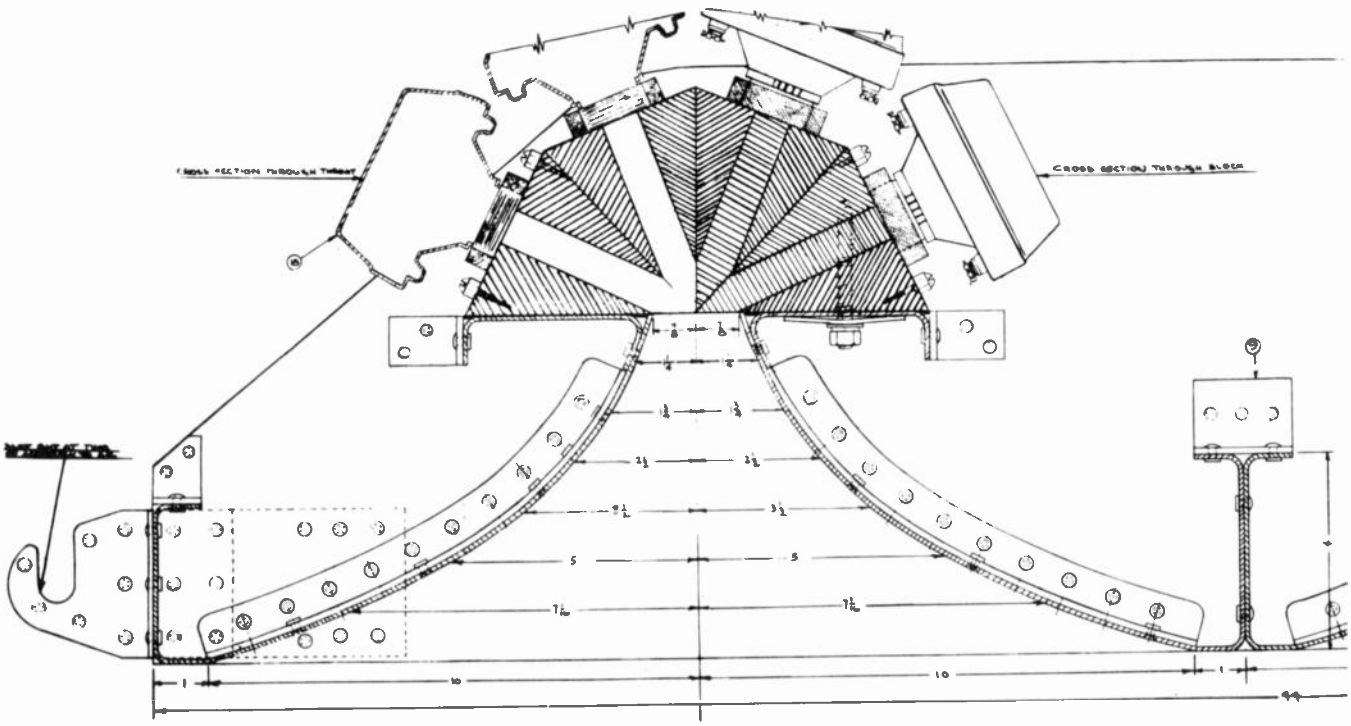

Fig. 4—Half twin horn—cross section.

tion, the throat blocks would be cast, of course, but in a research model system the wooden construction was expedient, light in weight, and economical considering the small quantity required.

The associated horn was fabricated of riveted sheet aluminum in the aircraft assembly department of the Baldwin factory. Horn mouth width was only a little greater than the over-all width of the throat block and driver unit assembly, and provided each unit with a 45 to 1 ratio between the area of horn mouth and throat, entirely adequate for efficient loading.

It was the design intent to make the angle of coverage small in the plane at right angles to the flight path, because the width of the coverage area specified was only one-half mile. This dictated that the horn mouth should have its long dimension across the fuselage.[6] Yet the fuselage width was sufficient to permit a throat block only ten driver units in length. Extending the horn outboard on each side of the fuselage a similar distance, which was almost as far as space between the fuselage and the engines would permit, provided only an array of 30 by 4 or 120 driver units, one-half of the number needed for the power. Consequently, it was decided to use a twin horn in each of the three locations, each half of each horn to have the design shown in Fig. 4. The end pieces of the horn were common to both halves, and were designed with the loudspeaker mounting frame and rotator assembly in mind. A perspective view of a twin horn (Fig. 5) shows the driver units, end plates, and mechanical bracing. Fig. 6 is an on-axis view which shows the detail of the throat block pieces which provided the blend of the small horn sections into the larger horns.

The driver units were connected in a series-parallel arrangement giving each twin-horn loudspeaker a nominal impedance of 8 ohms. In an initial test one side of the system was grounded. The large potentials between the voice coil and the grounded case of the units connected farthest above ground potential caused arcing and failure of a number of units. A center-grounded arrangement was then adopted. This change plus a 300-v breakdown test on all driver and replacement units reduced loudspeaker unit failures to a minimum. Experience with this problem emphasized the importance of ease of unit replacement in a loudspeaker system of this size, and the development of a good maintenance check for loudspeaker units. The most practical method of checking driver units, with the aircraft on the flight line, was to measure the current into each driver unit with a "clip-on" ammeter, while a steady audio signal was supplied from each power amplifier. Ear plugs plus a headset are needed for aural protection in this test.

Response and impedance characteristics of one of the three twin-horn loudspeaker systems are shown in Fig. 7. The response-frequency characteristic is rather uniform throughout the frequency range of 400 to 3700 cps. The nominal impedance of 8 ohms is a fairly conservative value in this same frequency range.

[6] H. F. Olson, "Elements of Acoustical Engineering," D. Van Nostrand Co., Inc., New York, N. Y., p. 25; 1940.

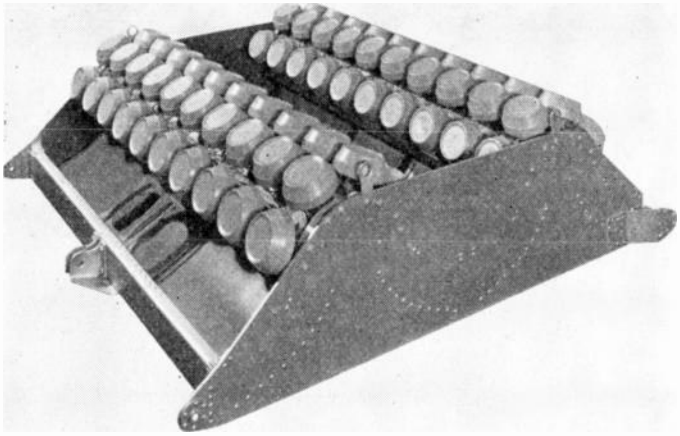

Fig. 5—Twin-horn loudspeaker—perspective view.

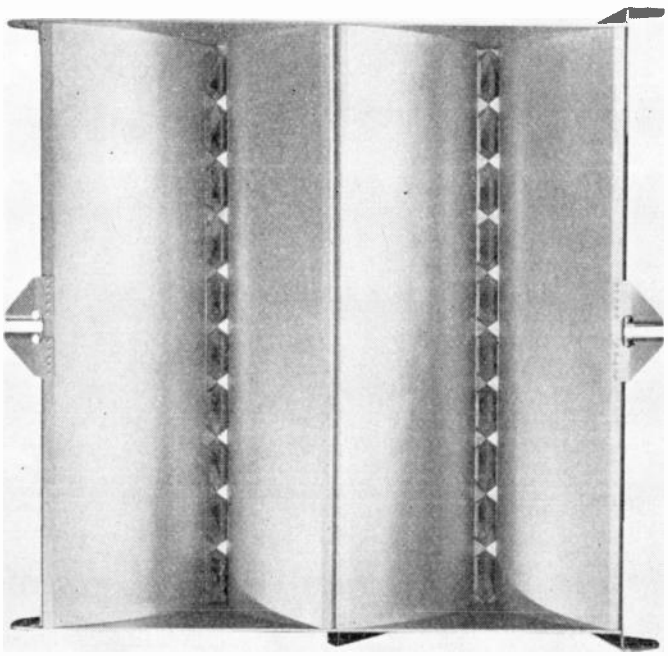

Fig. 6—Twin-horn loudspeaker—axial view.

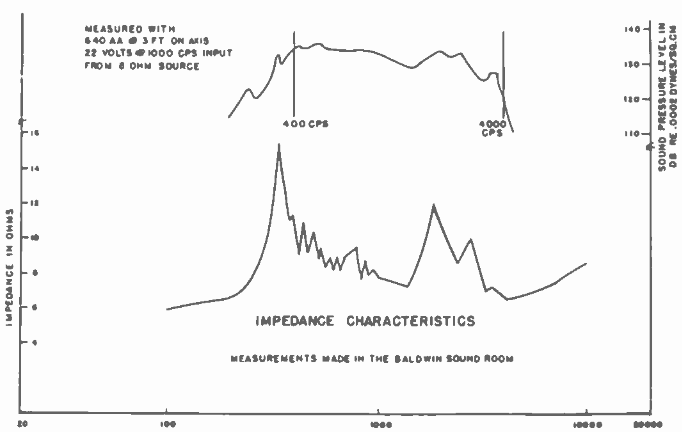

Fig. 7—Twin-horn loudspeaker—response and impedance frequency characteristics.

Fig. 8 (next page) shows the polar characteristics of a twin horn in its plane of symmetry (transverse to aircraft flight path) at a distance of 12 feet at frequencies of 400, 500, 1000, 2000, 3000, and 4000 cps. This is the plane in which a narrow beam had been the design aim. It is apparent that this aim was achieved. Addition of the other two twin horns in line in this same plane increases the directivity even further.

Fig. 9 shows the directional characteristics in the plane of the horn flare (containing the aircraft flight path). The directivity pattern in this plane is much broader, as desired, but because two horn sections were used, the directivity patterns are considerably more nonuniform than would be ideal for the purpose.

Loudspeaker Rotator

Time of effective transmission of speech had been recognized as one of the greatest factors limiting the usefulness of airborne announcing systems. Although the loudspeaker system was designed to have a fairly broad directivity pattern in the plane of the line of flight, it was decided to mount the loudspeaker system on a motor driven rotator, in order to extend the period of transmission. It was not feasible from an acoustical standpoint to rotate the horn system in such a manner that part of it would be inside the fuselage when rotation occurred. The relatively small clearance between the fuselage and the ground before take-off and after landing made it impractical to mount the loudspeaker system below the aircraft. Consequently a system was devised in which the rear edge of the loudspeaker system would be low at the beginning of the rotation cycle; then it would be raised until the loudspeaker was horizontal; finally the front edge would be lowered to complete the cycle. Screw jacks driven by dc motors controlled by relays were used to accomplish the rotation cycle. A control panel in the pilot's compartment provided visual indication of the status of the rotator. The automatic cycle was completed in approximately 30 seconds which doubled the effective period when highly directional loudspeakers were used.

System Installation and Evaluation

Fig. 10 is a greatly expanded view of the equipment operator's compartment. Actually this space was extremely crowded. The installation of all major equipment items is shown in phantom in Fig. 11. Also shown is a wind-screened monitoring microphone installed under the central loudspeaker, with which the output of the system could be measured or monitored at any time during flight. The aircraft is shown in Fig. 12 during an actual flight test, when the monitoring microphone was lowered on cables at various distances below the loudspeaker system, in order to evaluate the intelligibility of the speechwave as a function of distance. At the time of this test the outboard loudspeakers had not yet been

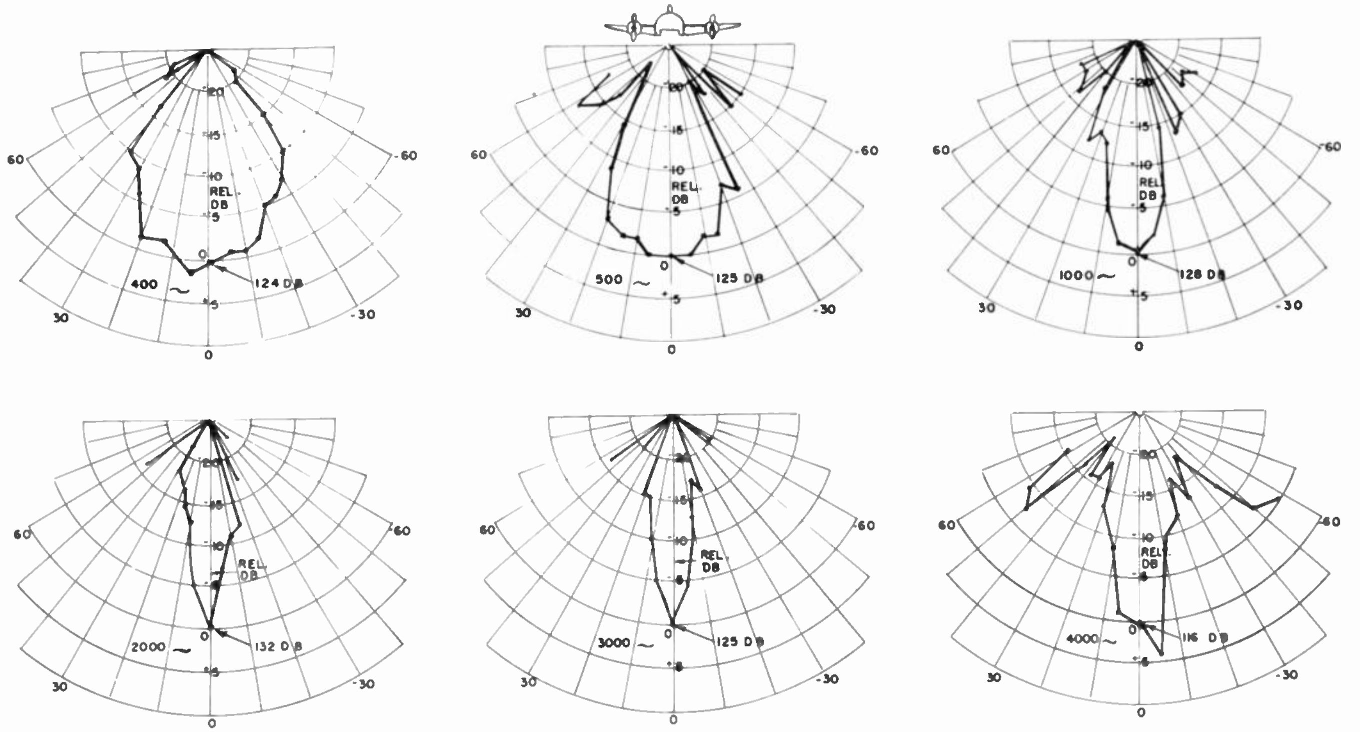

Fig. 8—Twin-horn loudspeaker—directional characteristics in the plane of symmetry.

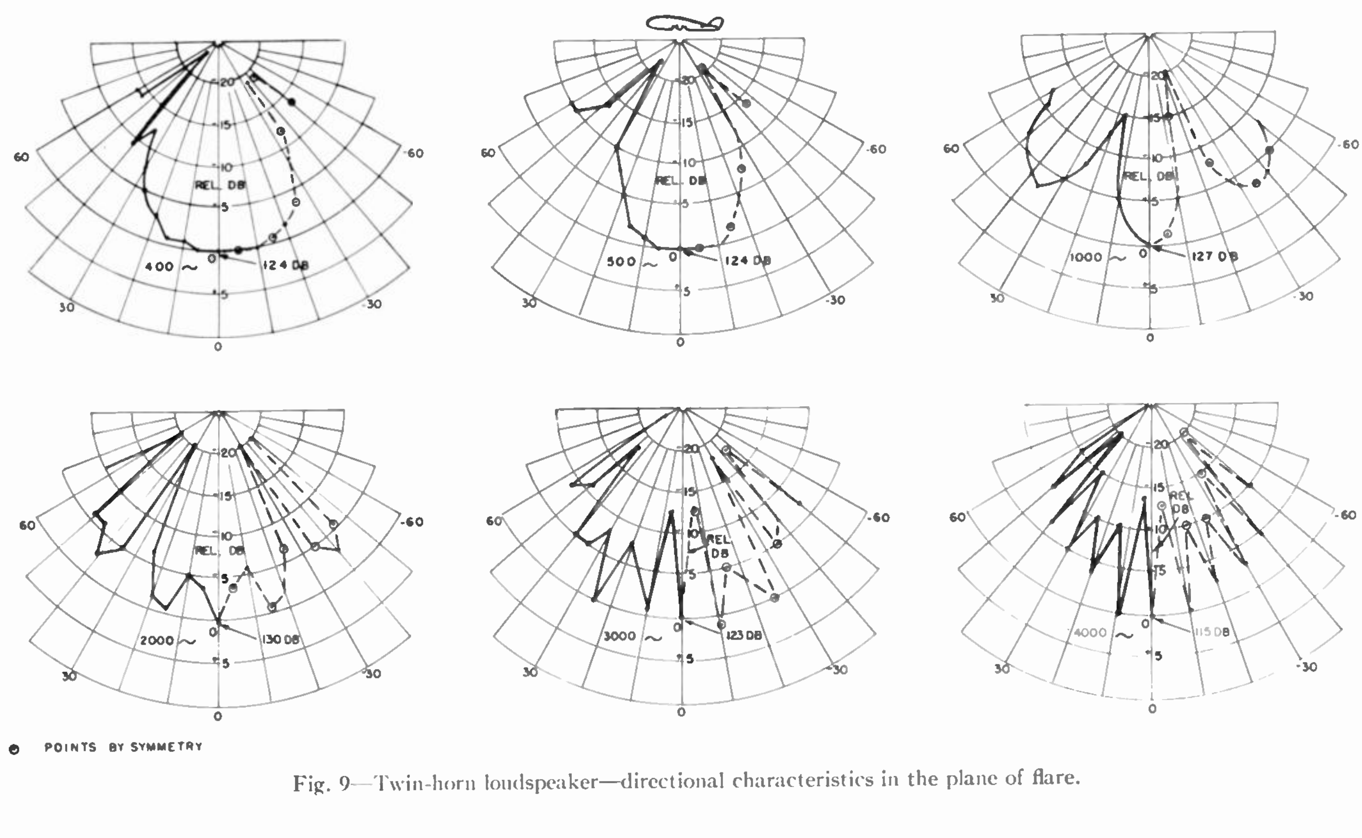

Fig. 9—Twin-horn loudspeaker—directional characteristics in the plane of flare.

installed (the nose wheel was not retracted because one of the supporting cables passed through this space).

Extensive flight tests over the rural Baldwin Ancor plant established that the increased power and improved directivity pattern increased the transmitted speech level by approximately 10 db. It was also found that reducing the air speed to 120 mph increased the average transmitted level approximately 6 db more. Width of coverage was measured and recorded by microphones located $\frac{1}{4}$ mile on each side of the projected flight path. Without the loudspeaker rotation the number of words per pass correctly received was increased nearly 3 to 1

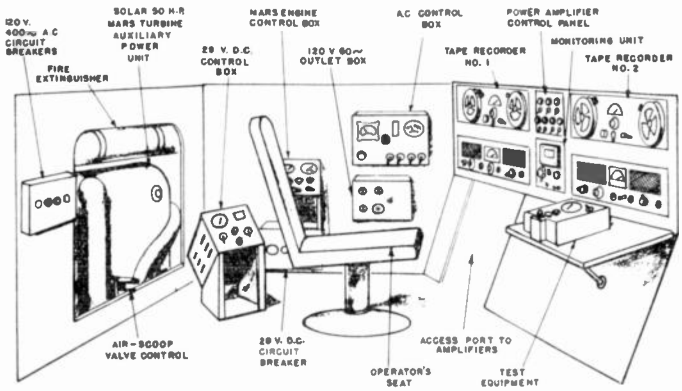

Fig. 10—Expanded view of equipment operator's compartment.

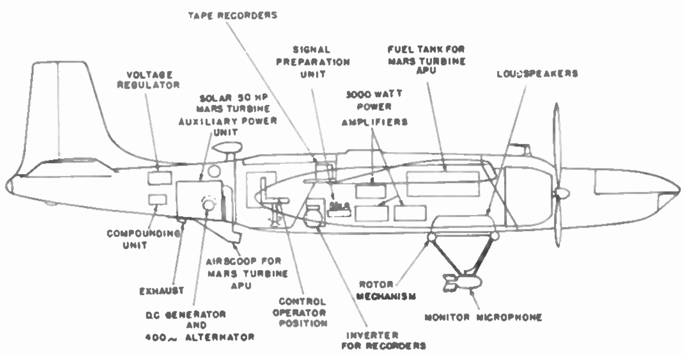

Fig. 11—Location of major equipment items in B-26 aircraft.

Fig. 12—B-26 in flight with sound system and suspended microphone.

over the earlier system, and more than 4 to 1 when the aircraft speed was slowed down.

Although the combination of improved equipment and lower airspeed greatly improved word reception, it should not be assumed that really good communication was finally achieved. Rather the performance was increased from "sub-marginal" to "marginal." Special techniques which were found useful in improving the performance further were the prerecording of messages on tape, directing the loudspeaker toward the target, the use of selected vocabulary and, particularly, redundancy. For listeners within $\frac{1}{4}$ mile off the projected flight path, reception of 15-second messages on basic topics repeated several times during the same pass, were received quite well. Air-to-ground transmission from high altitude at speeds in excess of 100 mph (but not greater than 200 mph) was concluded to be feasible under favorable weather conditions, although word intelligibility is low by conventional standards.

Proposed Improved System

During the latter part of the project two major equipment developments occurred which could be applied to a future design for production. Both of these developments reduce the space and weight requirements, and together they permit a revised installation layout which, according to preliminary estimates, would permit restoration of the armament complement of the aircraft.

Single-Flare Horn

Although it still seemed impractical, from a loudspeaker maintenance standpoint, to construct a throat block with more than four horizontal rows of driver units, the two throat blocks from a twin horn were adapted to be mounted on a wooden single-flare horn model shown in Fig. 13 (next page). A possible disadvantage of this arrangement is the necessity for a lower cutoff frequency (approximately 240 cps), tending to increase the distortion at high frequencies. The response characteristic of the single-flare horn was somewhat rougher on the axis than for the twin horn, but this irregularity was minor compared to that of the twin-horn response curves at angles off the axis.

As expected, the directional characteristics in the plane of symmetry resembled closely those for the twin horn. Fig. 14 demonstrates the expected improvement in smoothness and broadness of directivity pattern in the plane of horn flare. This type of horn is recommended for future systems if a practical means can be devised for servicing the loudspeaker units, such as by hinging the horn flares near the throat block to give access to the lower rows of units. With such a broad directional characteristic, the loudspeaker rotator could be eliminated.

Bereskin Amplifier

During the completion of the development and construction of the 3000-w power amplifiers previously described, it was decided to attempt circuit development for an improved type of amplifier which could be used to advantage in a follow-up design. This circuit development has been described elsewhere.[7] The laboratory model constructed, operating from separate laboratory power supplies, showed great promise in reducing the circuitry, the number of tubes, the size and weight. It also provided the nominal audio power at very low distortion. Possible disadvantages were the necessity for forced-air cooling (which might be provided easily in an airborne system) and increased plate potential for the

[7] A. B. Bereskin, "A 3000-watt audio power amplifier," IRE Trans., vol. AU-4, pp. 37–41; March, 1956.

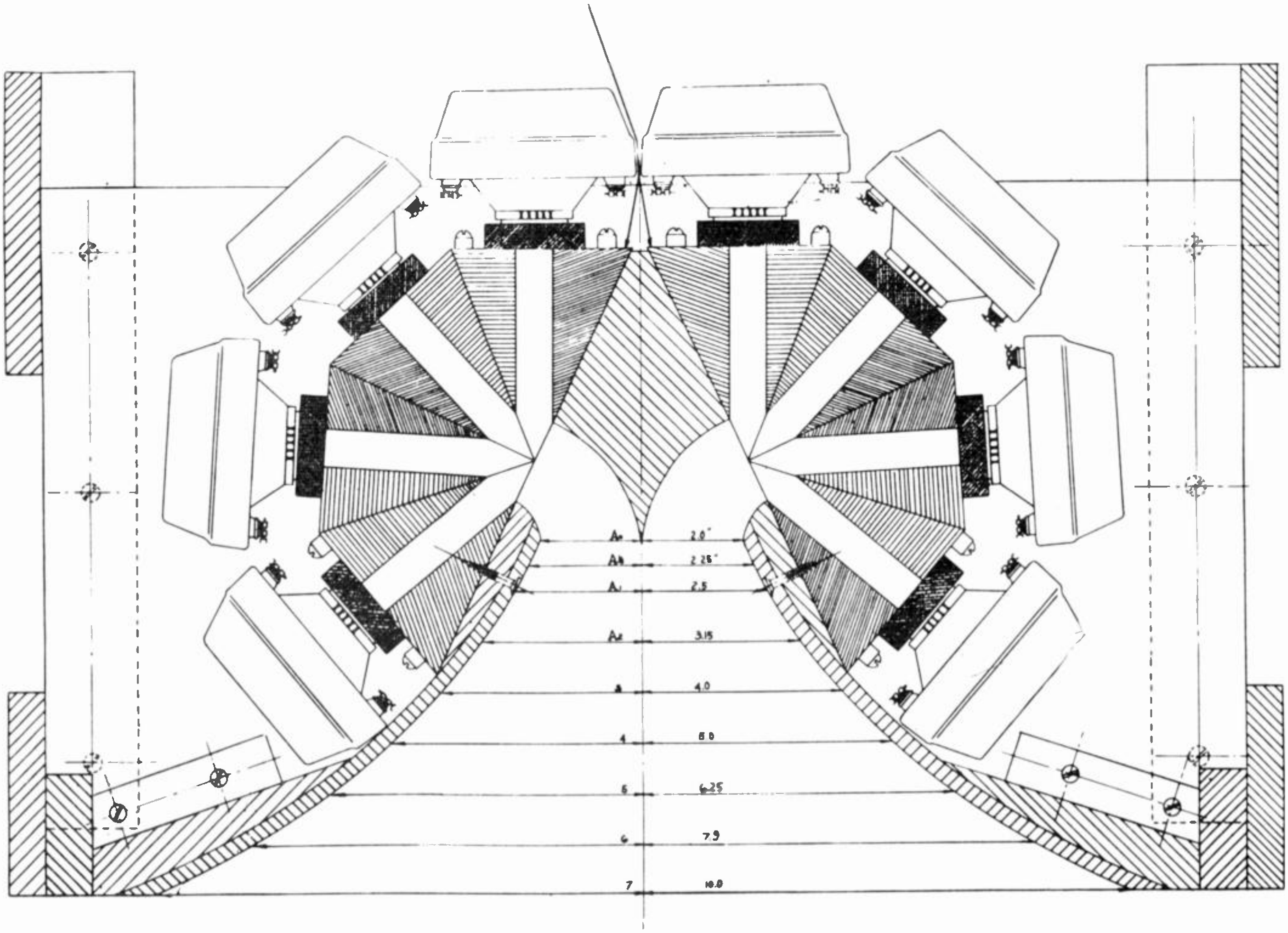

Fig. 13—Experimental single-flare horn cross section.

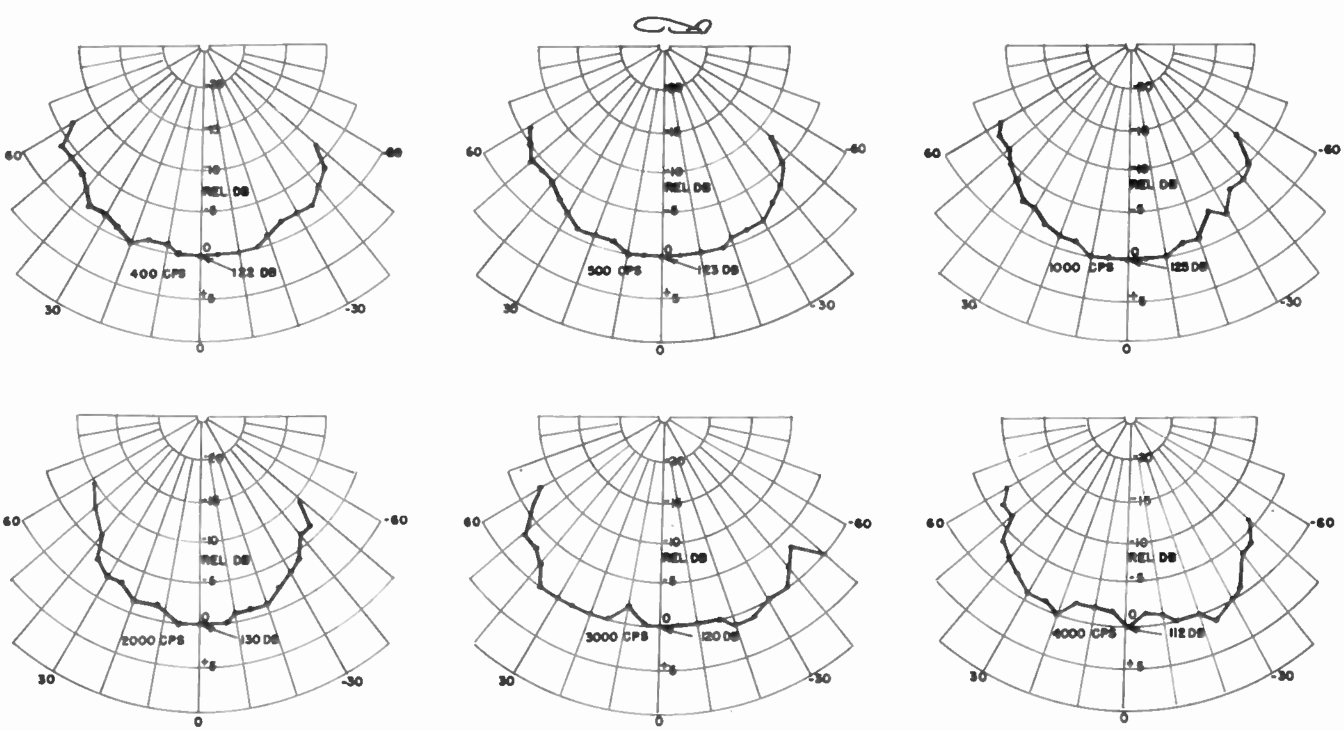

Fig. 14—Single-flare horn—directional characteristics in the plane of flare.

output tubes. The circuit employed a pair of 4-1000A tubes operating Class B_1 push-pull with a 5-kv plate supply voltage and 1.25-kv screen voltage. The driver stage used a pair of 6AU6 tubes. Consideration of this amplifier for any future airborne systems of this type is strongly recommended.

Proposed System

Through reduction of the front-to-back dimension of the loudspeaker system, using the single-flare horn, and reduction of the power amplifier dimensions, it would be possible to install the auxiliary power unit in the same compartment now occupied by the loudspeakers and amplifiers. Proximity of all the major equipment items would facilitate integration of controls into a more compact package. It would also permit restoration of aircraft armament to spaces now required for audio equipment and auxiliary power unit installation. Reduction in size and weight, and integration of the type proposed might also provide a system which could be installed more easily in aircraft of different types.

Acknowledgment

This development was done in close cooperation with WCLNE3, Communication and Navigation Laboratory, Wright Air Development Center. E. Lazur, L. N. Theroux, and Capt. Richard Biles were directly active in the government phases of the entire project. R. Murphy of Baldwin and W. L. Lane and others of WADC operated airborne equipment during flight tests.

On the Phasing of Microphones*

BENJAMIN B. BAUER†

Summary—**Correct phasing of microphones is most important when two or more microphones are connected simultaneously to a single transmission system. The phasing of all gradient and of some phase-shift microphones is reversed for rewardly arriving sound waves. In this paper the phase-frequency characteristics of most common microphones are described; methods for predicting or ascertaining the phasing of microphones are given; and a system is proposed for experimentally determining the absolute phasing of an unknown microphone.**

Introduction

The phasing of microphones is not of the same manifest importance to the user as some of the other performance characteristics, such as frequency-response or sensitivity. Anyone who has dealt with sound systems knows that reversal of leads of a balanced input amplifier does not alter the performance of the system; moreover it is known that the character of common musical tones is not patently altered by the phase relationship of the individual harmonic components. Why then is phasing of any interest at all? There are two principal reasons: 1) when two or more microphones which are used in proximity of each other and feed into a common system are out-of-phase there is evidence of cancellation of output, especially at low frequency; and 2) the waveform of speech sounds tends to be dissymmetrical and the phasing of a microphone may have a bearing upon the degree of modulation of certain amplitude-modulated transmitters.

* Original manuscript received by the PGA, February 20, 1956; revised manuscript received by the PGA, September 17, 1956.
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To explain the existence of effect 1) refer to Fig. 1(a): two microphones M_1 and M_2 are connected through two amplifiers to a common line L. If a sound source S_1 is placed in the plane of symmetry between the two microphones, and assuming that the microphones are connected so that the electrical impulses add, then the response-frequency of the system will be the same as that of either microphone; this response is represented by the straight line S_1 in Fig. 1(b). If the sound source is placed in position S_2, the phase of the sound pressure at both microphones is not significantly different at low frequency because the wavelength of sound is long. At high frequency, however, the difference in phase becomes significant, and cancellations will be encountered whenever the differences of path between S_2 and the microphones is equal to odd multiples of $\frac{1}{2}$ the wavelength of sound. The response-frequency characteristic of the system under this condition is indicated in Fig. 1(b) by the dash-line labeled S_2.

Consider next the system in which the microphones are connected "out of phase." This condition is shown diagrammatically in Fig. 1(c). For a source placed in position S_1 in a plane symmetrical with respect to the microphones, the voltages generated by both microphones (if they are identical) will be equal and opposite resulting in complete cancellation at all frequencies. For a source placed in position S_2 there will be cancellation at low frequency; at high frequency, each time the difference in path length between S_2 and the microphones equals an odd multiple of half the wavelength of sound there will be an additive condition, resulting in a re-

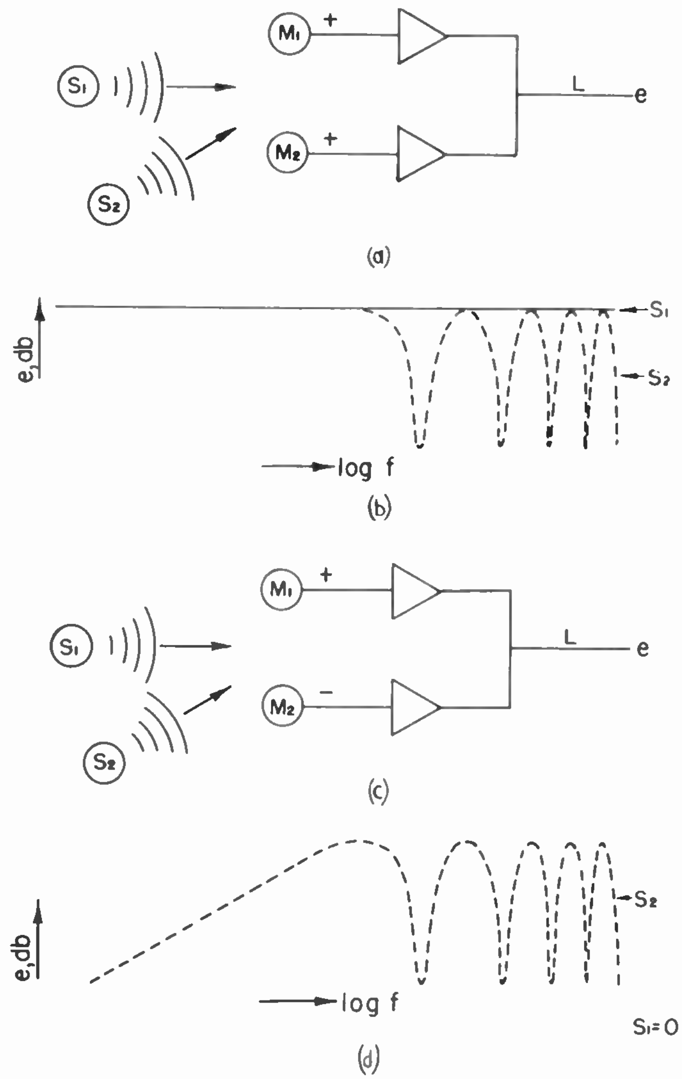

Fig. 1—Performance of two microphones connected in-phase and out-of-phase to a single transmission system.

sponse-frequency characteristic indicated by the dash-line labeled S_2 in Fig. 1(d).

Since there are situations where two or more microphones are required (as on a speaker's podium), it is evident that the system in Fig. 1(a) and 1(b) is vastly superior to the one in Fig. 1(c) and 1(d). Therefore, it is important to understand the factors influencing the phasing of microphones and to provide means for establishing correct phasing.

Effect 2) appears to be of concern in special situations pertaining to transmission of speech, and it will not be discussed here in any detail.

Phase-Frequency Characteristic of Microphones

One of the properties attributable to an ideal microphone might be the generation of an electric wave corresponding in every way to an acoustic wave, including the coincidence of phase between the sound pressure phasor and the output voltage phasor. Practical microphones depart from this ideal by a considerable margin. Let us get an idea of the type of phase-frequency characteristic encountered in the more common types of microphones.

To study the behavior of microphones we will draw equivalent electrical circuits in which the sound pressure P will be assumed to correspond to a voltage; acoustic volume velocity u to a current; acoustic mass of a diaphragm or other element M_A to an inductance; acoustic compliance C_A to a capacitance. With these similarities in mind it will be possible to draw impedance and phase diagrams of sound pressure drops in the various component parts of a microphone, much as one draws impedance and phase diagrams of an electrical circuit.

The analyses in this section have been greatly simplified, for the sake of brevity; it is believed, however, that the conclusions given are typical and representative of what may be commonly found in practice.

Capacitor (Condenser) and Piezo-electric Microphones

A capacitor microphone consists of a thin conducting diaphragm which forms the cover of a box. Within the box there is an insulated stationary electrode spaced from the diaphragm by about 1/1000 inch to form a small capacitor C (about 50 $\mu\mu f$). This capacitor is charged by a dc polarizing voltage E_0 through a very high resistance so that during any small interval of time the charge $Q=CE_0$ remains constant. The resonant frequency of the diaphragm is near the upper end of the audio spectrum. Damping is obtained from a thin film of air between the diaphragm and the back electrode. The microphone and its associated electrical circuit is shown in schematic cross section in Fig. 2(a).

The equivalent electrical circuit of the acoustical elements is shown in Fig. 2(b) in very simplified form. The diaphragm system is shown as an acoustic mass M_A, acoustic compliance C_A, and acoustic resistance R_A in series actuated by the sound pressure P. The voltage e_0 generated by the microphone owing to a sound pressure P is in direct proportion to the incremental change in capacitance stemming from the incremental displacement of the diaphragm. That displacement, moreover, is related to the volume velocity of sound u multiplied by the acoustic capacitive reactance of the compliance C_A and a constant. Therefore, $e_0=ku/j\omega C_A$. In drawing phase diagrams for the condenser microphone the phase-displacement between the generated voltage e_0 and the sound pressure P will be indicated by the angle between P and $u/j\omega C_A$.

The open circuit voltage-frequency characteristic of the microphone is indicated in Fig. 2(c) in solid line, and it will be seen that it is flat at low frequency and exhibits a damped resonant rise at a high frequency f_0 with a drop thereafter. From the electrical side, as shown in Fig. 2(d), the capacitor microphone appears as a generator of voltage e_0 in series with a capacitor C and a resistor R (the latter is composed of the charging re-

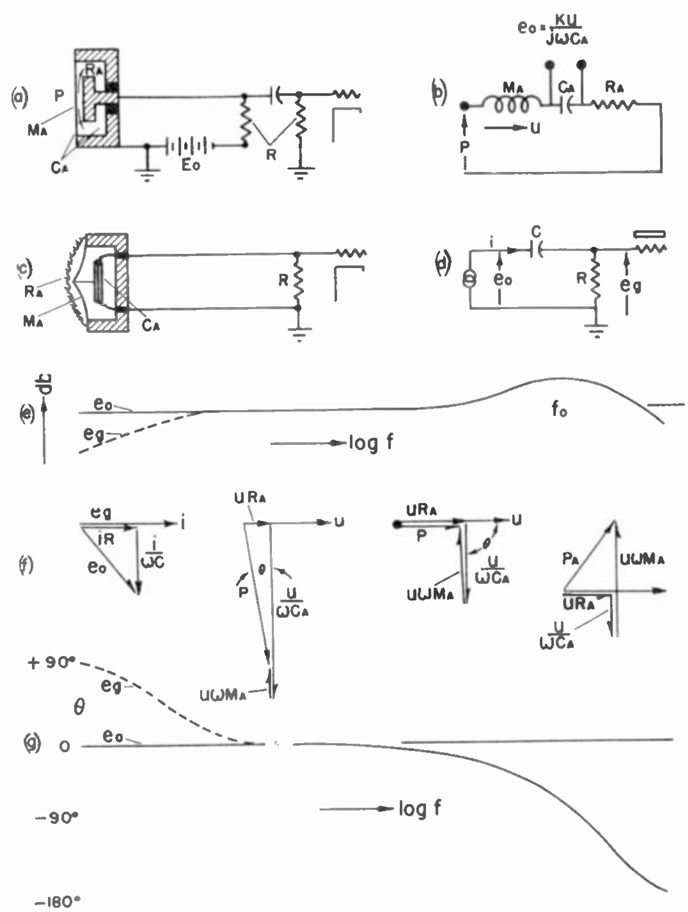

Fig. 2—Phase-frequency characteristic of capacitor and piezoelectric microphones.

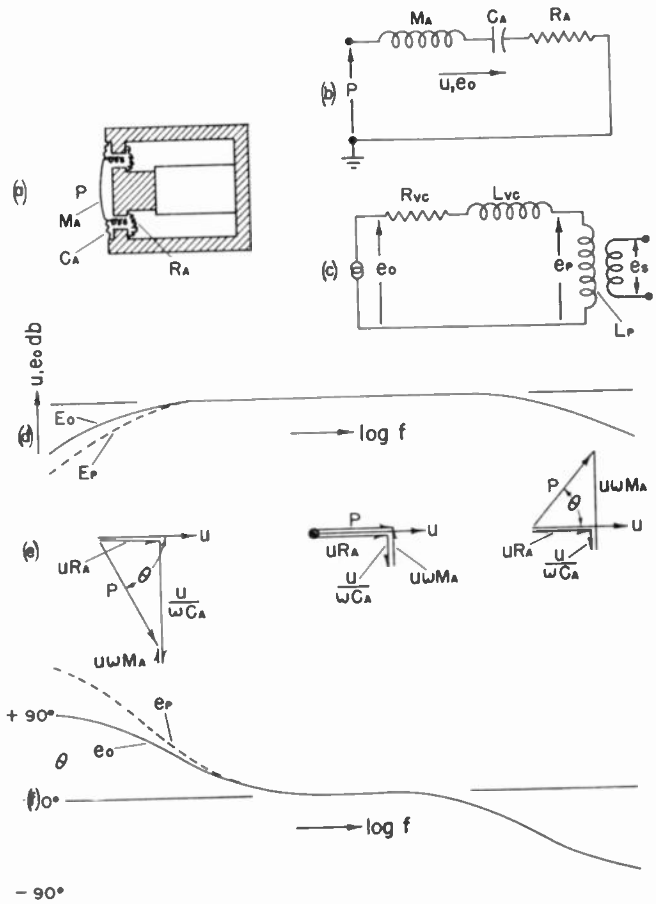

Fig. 3—Phase-frequency characteristic of moving coil microphone.

sistor and the grid resistance in parallel). At low frequency the reactance of C becomes comparable with R resulting in a drop of output. This effect is shown in dash-line in Fig. 2(e).

Phase diagrams for the capacitor microphone are shown in Fig. 2(f). P and $u/j\omega C_A$ are nearly in phase at mid-frequency; at resonance, however, the capacitive reactance cancels the inductive reactance, and P and $u/j\omega C_A$ are in quadrature; hence the electrical signal lags behind the applied pressure by 90°. Above resonance the electrical signal approaches a 180° lagging phase-shift as the frequency increases. The complete phase angle-frequency characteristic is shown in Fig. 2(g). At very low frequency the electrical phase shift becomes important; the voltage vector e_g leads the open circuit voltage vector e_0. This is shown by the dash-line in Fig. 2(g).

The piezoelectric microphone shown in cross section in Fig. 2(c) has capacitive internal impedance and a behavior rather similar to that of a condenser microphone, and its phase angle-frequency characteristic is also similar to that exhibited by Fig. 2(g).

Moving Coil Microphone

A simplified moving coil microphone is shown in cross section in Fig. 3(a). The voice coil is attached to a diaphragm which constitutes the cover of a box or enclosure. The diaphragm and coil define an acoustic mass M_A. The exposed side of the diaphragm is actuated by the sound pressure P, while the enclosed side is coupled to a damping resistance R_A. The air trapped in the box and the edge stiffness of the diaphragm are the principal elements in determining the compliance of the system C_A.

A simplified equivalent acoustical circuit is shown in Fig. 3(b). The open circuit voltage $e_0 = Blv$ is proportional to the volume velocity of the diaphragm u. In the usual type moving coil microphone the damping resistance R_A is made preponderantly higher than the reactances of the mass and compliance factors resulting in a highly damped action over most of the frequency range. The open circuit voltage response curve is shown in Fig. 3(d) as a solid line. The response is uniform over the midrange and tends to exhibit a drop at the low and the high frequency end.

In practical microphones, these dropping ends are lifted, in a manner of speaking by means of acoustic resonators, but these will not be included here.

Another factor which causes a change in response characteristic of a moving coil microphone is a transformer. The electric circuit of the microphone and transformer is shown in Fig. 3(c). It can be shown that the primary inductance L will determine the transmission when its reactance becomes comparable to or less than the voice coil impedance $R_{vc}+j\omega L_{vc}$. The effect of the transformer upon frequency response is shown in dash-line.

The phase-frequency characteristic of the microphone is analyzed by the use of phase diagrams shown in Fig. 3(e). It is seen that in midrange, when the reactance due to the mass M_A and compliance C_A are small compared to the damping resistance R_A, u and P are near in phase. At low frequency the acoustic impedance becomes predominantly compliant and the velocity u (and hence the voltage vector e_0) is in a leading relationship with respect to the applied sound pressure P. At high frequency the acoustic impedance becomes predominantly inertant, u and e_0 lag behind P. At low frequency the transformer introduces another element of lead, while acoustic resonators mentioned previously introduce further elements of phase shift at high and low frequency. The approximate phase-frequency characteristic is shown in Fig. 3(f).

Gradient (Velocity) and Phase-Shift Microphones

The gradient and the phase-shift microphones have elements of similarity, as shown in Fig. 4(a) and 4(b). The gradient microphone is composed of a ribbon suspended between pole-pieces which form a baffle, establishing a path difference d between the front and the back of the ribbon. A frontal sound wave traveling with a velocity c reaches the rear part of the ribbon d/c seconds after the front; this corresponds to a phase shift angle of $\omega d/c$ radians. The phasor relations for the frontally arriving sound waves are shown in Fig. 4(c), where P_1 represents the front pressure and P_2 represents the rear pressure, establishing the difference P_1-P_2 which is the effective pressure causing the motion of the ribbon. The ribbon has a mass M_A and a compliance C_A and is resonated at a low frequency being usually damped by the acoustic resistance of the dust screens, R_A. Throughout most of the frequency range the mass reactance is predominant, and the volume velocity through the ribbon u, and hence the open circuit voltage e_0, lag behind P_1-P_2 by 90°. This places u and e_0 approximately in phase with P_1. At resonance the response rises, and as the frequency diminishes, it again drops as shown in the frequency response characteristic in Fig. 4(e).

One important difference between pressure microphones and gradient microphones should be noted: pressure microphones operate because of the pressure of a sound wave at a point in space. Gradient microphones operate due to a difference of pressure in a sound wave at two or more points in space. Therefore, the phase characteristic of pressure microphones is independent of the direction of arrival of sound waves, while the phase characteristics of gradient microphones is dependent upon the direction of arrival of sound waves. This effect can be seen in Fig. 4(d) which portrays phase relations for rewardly arriving sound waves. It will be noticed that relative phase positions of P_1 and P_2 have been reversed, and the resulting volume velocity through the ribbon u is opposite to that encountered in Fig. 4(c).

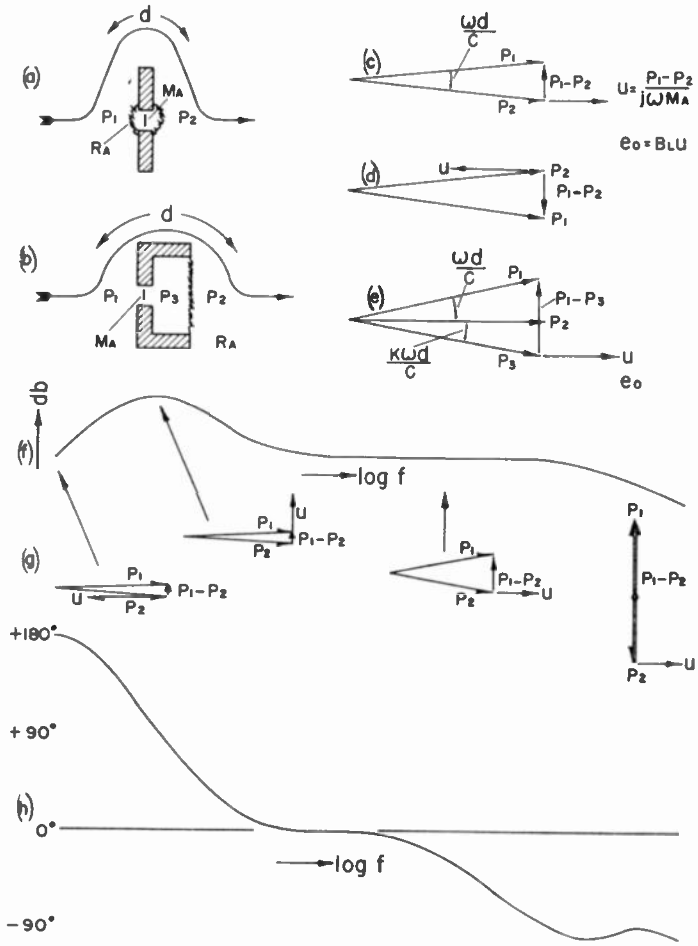

Fig. 4—Phase-frequency characteristic of gradient and phase-shift microphones.

The phase-frequency characteristic can be approximated by use of phase diagrams as in Fig. 4(f). The velocity u and the open circuit voltage e_0 are in phase with the sound pressure P at midrange. At high frequency there is a lag in the open circuit voltage; the action at the extreme high end is complicated by diffraction effects and is not readily analyzed by simple methods. At resonance, the velocity u is controlled by the resistance R_A, hence, it is in phase with P_1 and P_2, and thus leads P_1 by 90°; below resonance the ribbon is controlled by C_A and its velocity leads P_1-P_2 (and thus leads P_1) by an angle which approaches 180°. The transformer, the voice-music switch, and the grille enclosing the microphone have further important bearing upon the phase of the voltage e_0.

The phase-shift microphone is acoustically somewhat similar to a gradient microphone, with the exception of the fact that an acoustic phase-shift network is added which adds a component $k\omega d/c$ to the phase-shift $\omega d/c$. The phase diagram is shown in Fig. 4(c); it is seen that u (and hence e_0) are basically in phase with the pressure P_1 in a manner similar to that occurring in a gradient microphone. It may be shown, likewise, that the deviation from the in-phase condition at the low and the high-frequency end of the spectrum are similar to those which occur in the gradient microphone. For rewardly arriving sound waves the volume velocity may be in-phase-with or out-of-phase-with the sound pressure depending upon whether the constant k is greater or lesser than unity.

Microphones of the gradient and phase-shift type are subject to another change in phase relationship when they are exposed to a diverging sound field, such as occurs close to a person's mouth or in close vicinity of a small loudspeaker. This factor will be discussed further in the section entitled "Experimental Determination of Phasing."

Determination of Phasing by Designer

Anyone who has visual or physical access to the transducer or the working mechanism of the microphone can, in most instances ascertain the phasing of the instrument with comparative ease and reliability. The approach is different with different types of microphones.

Phasing of Condenser Microphones

The inward motion of the diaphram of a condenser microphone owing to a positive increment of pressure caused by a sound wave produces an increase in capacitance. Since the charge $Q = CE$ tends to remain constant, the increase in capacitance causes a decrease of the voltage across the electrodes of the microphone. Therefore, the inphase terminal of a condenser microphone is the terminal connected to the negatively charged electrode. Most condenser microphones employ a built-in preamplifier, and any phase-shift or reversal in the preamplifier will affect the polarity of the microphone.

Phasing of Piezoelectric Microphones

The phasing of a piezoelectric microphone is dependent upon the orientation of the molecules in the element, which is not known beforehand. However, most crystal microphones are sufficiently sturdy so that a light pressure may be applied to the apex of the diaphragm and the polarity of the resulting voltage may be readily recognized by observing the motion of the trace of an oscilloscope connected to the microphone. Fig. 5(a) shows the resulting patterns when the inphase terminal of the microphone is connected to the "high" terminal of the oscilloscope. It goes without saying that the phasing of the oscilloscope itself must be ascertained, as by use of a battery and a voltage divider.

Phasing of Dynamic Microphones and of Gradient Microphones

As in the case of the crystal microphone, the polarity of some dynamic microphones may be determined by gently depressing the diaphragm inwardly and observing the direction of motion of the oscilloscope trace. The shape of the traces for correctly phased microphones is shown in Fig. 5(b). Some dynamic and all ribbon microphone transducers are too delicate to be handled in this manner; as an alternative procedure it is possible to ascertain the polarity by causing the diaphragm or ribbon to move electrically by carefully applying a small external dc current of known polarity to the moving element and observing its motion. The following rule is applicable: The inphase terminal of a dynamic, magnetic, or a ribbon microphone transducer is that terminal to which the positive terminal of a battery must be connected to cause the element to move away from the observer facing the front of the microphone. Sufficient resistance must be connected in the circuit to avoid damage to the element. The rule as stated above will encompass the fact that the phasing of a nondirectional microphone is independent of the orientation of the microphone, while that of a gradient microphone will depend upon the direction of sound incidence. The phasing of transformer used with the microphone, if any, will obviously affect the phasing of the microphone.

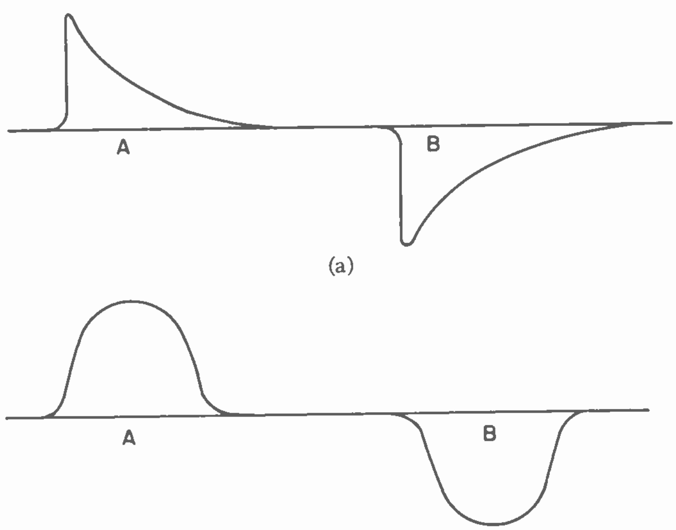

Fig. 5—(a) Pulse produced by mechanical actuation of a piezoelectric microphone. (b) Pulse produced by mechanical actuation of a moving coil microphone. A—pressure applied; B—pressure released.

Experimental Determination of Phasing

Usually the microphone element is not accessible for inspection and the user may not wish to run the risk of damaging the microphone by opening the case and handling the moving element. Therefore, it becomes desirable to develop a method for experimental determination or verification of the phasing of a microphone. We will not dwell at length on the easiest, and most commonly used method of comparing the phasing of an unknown microphone with one whose phasing is known to be correct. The two microphones are set side by side and connected to a common system and individual loudness levels are adjusted to equality. When listening to both units simultaneously over one loudspeaker system, any evidence of cancellation will indicate an out-of-phase condition of the unknown microphone. However, to measure the phasing of any one microphone it is best to provide a sound source capable of generating a sound wave of known phase.

Phase of Sound Pressure at the Diaphragm of a Loudspeaker

In Fig. 6(a) is shown a loudspeaker mounted in an infinite baffle and connected to an oscillator through a resistance R having a value considerably higher than the impedance of the voice coil. The current through the voice coil is, then, in phase with the voltage at the terminals of the oscillator. The approximate equivalent circuit for this system, viewed from the acoustical side is shown in Fig. 6(b). The sound pressure P developed across the acoustical load of the medium confronting the loudspeaker is given by the approximate equation

$$P = \frac{BlE/RA}{1 + (1/A^2)(j\omega M_D + 1/j\omega C_D)(1/R_A + 1/j\omega M_A)} \qquad (1)$$

where

- B is the flux density in the air gap
- l is the length of conductor in the air gap
- E is the voltage across the terminals of the oscillator
- R is the series resistance
- A is the area of the diaphragm
- M_D is the mass of the diaphragm
- C_D is the compliance of the diaphragm
- R_A is the equivalent parallel acoustic resistance of the medium
- M_A is the equivalent parallel acoustic inertance of of the medium.

Examination of (1) shows that, if the loudspeaker is driven at a frequency which is substantially above the resonant frequency of the cone, *i.e.*, if $j\omega M_D \gg 1/j\omega C_D$, and furthermore if the operating frequency is such that $1/R_A \ll 1/j\omega M_A$, then the following equation is obtained:

$$P = \frac{BlE/RA}{1 + (M_D/M_A A^2)} \qquad (2)$$

and hence, the pressure P will be in phase with the applied voltage E. For an ordinary 8-inch loudspeaker having a fundamental resonant frequency of about 60 cps this condition is reasonably well attained at frequency of 150–300 cps. Such a speaker may be used as a generator of sound pressure of known phase for measurements of microphone polarity.

Experimental Determination of the In-Phase Terminal of a Pressure Microphone

The experimental set-up is shown in Fig. 7. The loudspeaker may be installed in a simple baffle, or it may be even suspended freely without any baffle whatsoever.

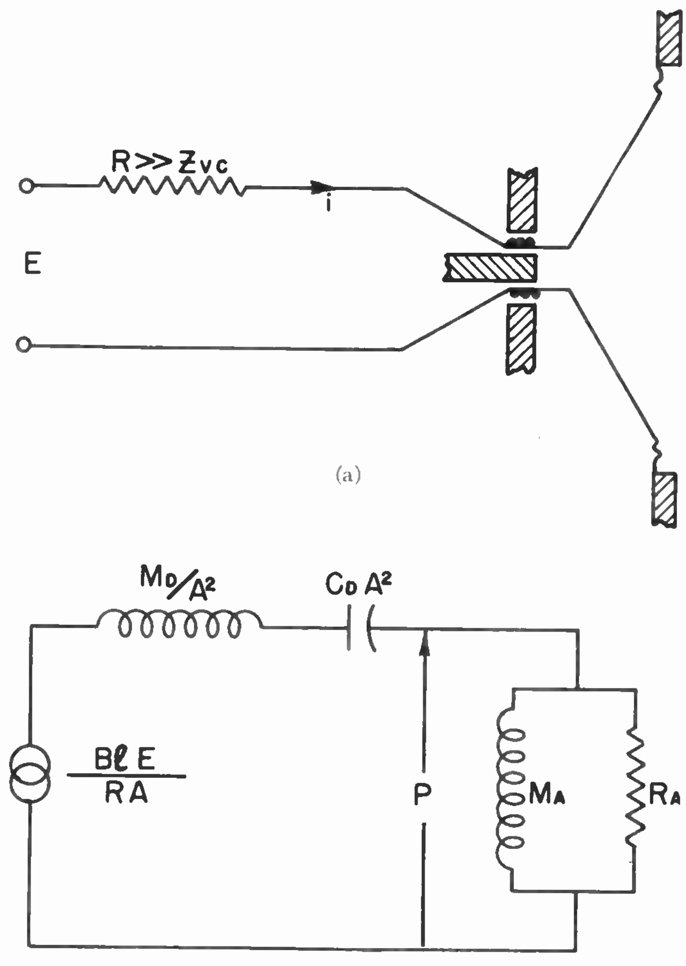

Fig. 6—(a) Loudspeaker mounted in an infinite baffle. (b) Equivalent circuit of a loudspeaker driven from a high impedance source.

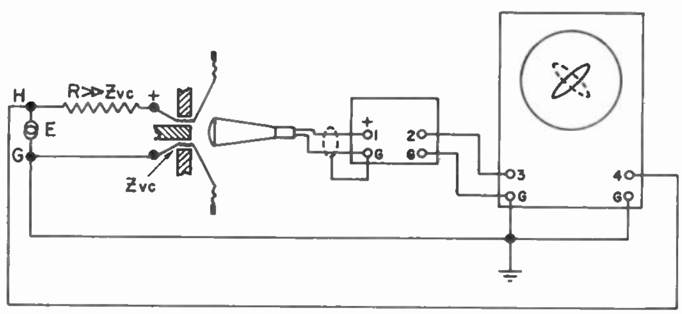

Fig. 7—Experimental determination of phasing of a pressure microphone.

The procedure found satisfactory is as follows:

1) Check the proper phasing of the oscilloscope and amplifier. To do this, connect one terminal of the oscillator to ground; connect the other (H) terminal to the "hot" terminals 1 and 4. The trace on the oscilloscope should be a line slanting from lower left to upper right.

2) Ascertain the in-phase terminal of the loudspeaker. This is done by connecting a battery across the voice-coil in such a manner that the cone moves toward the microphone; the terminal connected to the positive terminal of the battery is the in-phase terminal of the loudspeaker. Connect the resistance R between the H

terminal of the oscillator and the in-phase terminal of the loudspeaker. R should be 5 times the voice-coil impedance or more.

3) Adjust the oscillator output for suitable acoustic output from the loudspeaker. Connect the microphone to the amplifier and position the diaphragm of the microphone as close as possible to the surface of the vibrating cone. Check the orientation of the oscilloscope trace. If the trace is in the form of a slanted line or ellipse with its major axis oriented from lower left to upper right, then the in-phase terminal of the microphone is the terminal connected to the "hot" terminal of the amplifier (1). The given relationship should remain over a range of frequencies—perhaps 100 to 400 cps.

Experimental Determination of the In-Phase Terminal of a Gradient Microphone

When the procedure outlined above is applied to a gradient (velocity) microphone it is noted that the trace is in the form of a circle. This is explained in Fig. 8. The sound pressure developed at the cone diminishes rapidly with distance, and hence the pressure at the back of the ribbon P_2 is significantly smaller than the pressure at the front P_1. The pressure difference $P_1 - P_2$ is nearly in phase with P_1; the velocity u and hence the voltage e_0 lags with respect to $P_1 - P_2$, and thus also with respect to P_1 by about 90°. To obtain a meaningful measurement, it is necessary to shift the phase of the voltage applied to the horizontal deflection plates by about 90°. This can be done by means of a phase shift

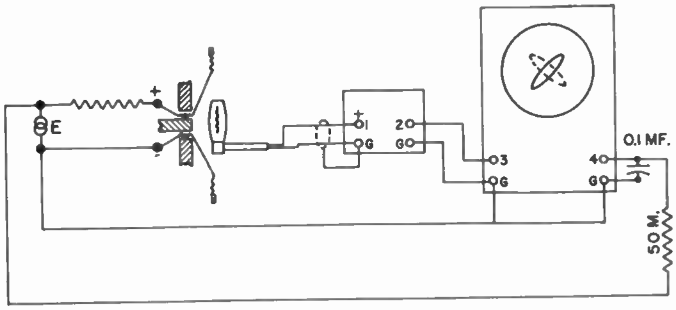

Fig. 8—Experimental determination of phasing of a gradient or phase-shift microphone.

network consisting, for example, of a 50m ohm resistor and a 0.1 uf condenser connected as shown in Fig. 8. Except for this detail, the experiment can be performed exactly the same as described above. Similar considerations apply to phase-shift microphones.

Conclusion

It has been shown in this paper that the phase relationships of all microphones under consideration follow a pattern which is similar. Therefore, proper phasing is important even if two dissimilar microphones are used. The principal problem with polarity is encountered when two or more identical microphones are oppositely connected resulting in severe cancellation because of identity of the opposing phase relationships. Methods have been shown for determining the polarity during the design of the microphone and for checking the polarity of unknown microphones.

The AF Anechoic Chambers at Cherry Hill*

M. S. CORRINGTON†, R. L. LIBBEY†, AND S. V. PERRY‡

***Summary*—The RCA Victor Division Laboratories at Cherry Hill, New Jersey, include two "sound proof" rooms designed for acoustic measurements on television and radio receivers, phonographs, and loudspeakers. These rooms are located on the second floor of a steel and concrete building where headroom is limited. Each room is of the double-shell masonry construction. The outer shell consists of the concrete floor and ceiling of the building, and concrete-block walls. The inner shell is a complete masonry box weighing about 40 tons. It is completely isolated by steel springs under its floor and above its ceiling. This construction saved greatly needed headroom.**

Access to the inner chamber is through separate doors in each shell, specially designed and built for this application. Ventilation is provided through long multitubular treated ducts, spring suspended and isolated by felt and rubber.

The internal acoustic treatment consists of Fiberglas wedges on the walls and ceiling and flat Fiberglas padding on the floor. This treatment is designed to produce the maximum absorption of sound in the available space. The over-all performance is adequate for making acoustic measurements at distances up to several feet from the source.

* Manuscript received by the PGA, September 10, 1956. Throughout this paper the term anechoic chamber refers to a sound measuring room. The word "anechoic" is from the Greek an-echoic and refers to the fact that sound reflections or echoes in these rooms have been nearly eliminated.

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‡ RCA Victor Radio and "Victrola" Division, Cherry Hill, N. J.

WHEN ACOUSTICAL measurements are made on sound waves, any reflections or echoes due to nearby objects will change the sound field and cause errors in the results. Echo-free measurements can be obtained in a large, unobstructed, outdoor field. However, unwanted sounds from airplanes, birds, or automobiles often make accurate outdoor data difficult to obtain. Wind and inclement weather may hamper or prevent the work. To eliminate these difficulties of outdoor measurements, it is necessary to approximate ideal free-field conditions in a special room. The extent to which such a room simulates actual free-field conditions depends upon its size, construction, and acoustical treatment.

Walls Must Absorb the Sound

A large room with sound-absorbing walls is desirable for an anechoic chamber. As the size of the room is decreased, the path difference between the direct and reflected waves decreases and the two waves become more nearly equal in intensity. This causes larger errors due to the addition and subtraction of the reflected wave.

In a small room, acoustical treatment can be used to make the room approximate the conditions of free space. The only limitation is that enough acoustical absorption be used on the walls. The action of acoustical absorbing material can be analyzed using transmission-line theory. Air can be thought of as a transmission line for sound. In the case of an open field, this air transmission line is infinitely long and there are no reflections. In an anechoic chamber the transmission line becomes finite in length. If the proper kind and amount of material can be used to match the impedance and terminate the line, no reflections will occur.

Outside Noises Excluded

Construction of an anechoic chamber must also provide isolation from external mechanical vibration and airborne noise. If the chamber is placed in a building where other activities are going on, there may be considerable unwanted noise and vibration. Airborne noise can be excluded by the use of heavy, nonporous masonry construction. Double walls provide additional isolation by giving a "room within a room." The inner room can be completely suspended upon springs to prevent the transmission of external mechanical vibrations.

Isolation must also be considered when installing signal cables, power, lighting, and air conditioning. The air conditioning duct may permit unwanted sound to enter if it is not properly designed. It must be well damped to prevent the entire duct from resonating as a pipe. The air passage in the duct must also be treated acoustically to prevent inflowing air from causing noise.

An anechoic chamber must be accessible and convenient. It must be relatively easy to move large television or "Victrola" cabinets in and out of the room. It should be possible to place microphones anywhere in the room without undue effort or risk of damage. It is often difficult to provide this accessibility and still keep the isolation of the inner chamber.

Chambers Are on Second Floor

Most sound rooms are mounted on solid foundations which are deep in the ground. At Cherry Hill it was necessary to build the rooms on the second floor of a large building, with very limited headroom between the concrete floor and ceiling slabs. It was necessary to isolate against mechanical vibrations in the building, and to design for the greatest possible headroom within the chamber. Two rooms were built, one for the Television Division and one for the Radio and "Victrola" Division.

Double-Shell Construction

Both anechoic chambers are of the double-shell construction, as shown by Figs. 1 and 2. The outer shell of each consists of four concrete block walls which stand on the concrete floor slab of the building, and extend to the concrete roof slab. Air leakage through the mortar joints and blocks is sealed off by a one-inch-thick nonporous coating of cement plaster on the inner sufaces.

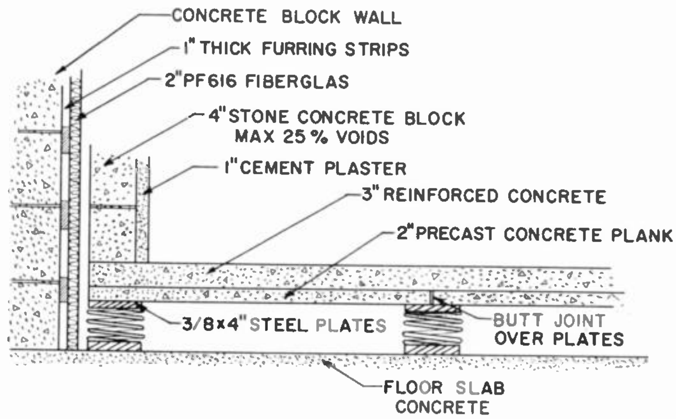

Fig. 1—Details of floor construction.

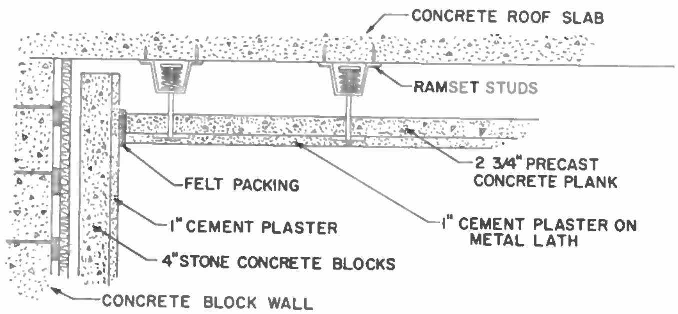

Fig. 2—Details of ceiling construction.

The inner surfaces of the outer walls and the roof slab are lined with two inches of Fiberglas PF616[1] mounted on and spaced from the wall by one-inch-thick furring strips. This lining is used to absorb whatever sound may get into the space between walls. This reduces the sound transmission to the inner wall.

Inner Shell Mounted on Springs

The inner shell of each chamber is a complete masonry box isolated from the outer shell and from the building by steel springs, as shown by Figs. 1 and 2. The spring details are shown by Fig. 3(a) and 3(b).

[1] Manufactured by Owens-Corning Fiberglas Corp.

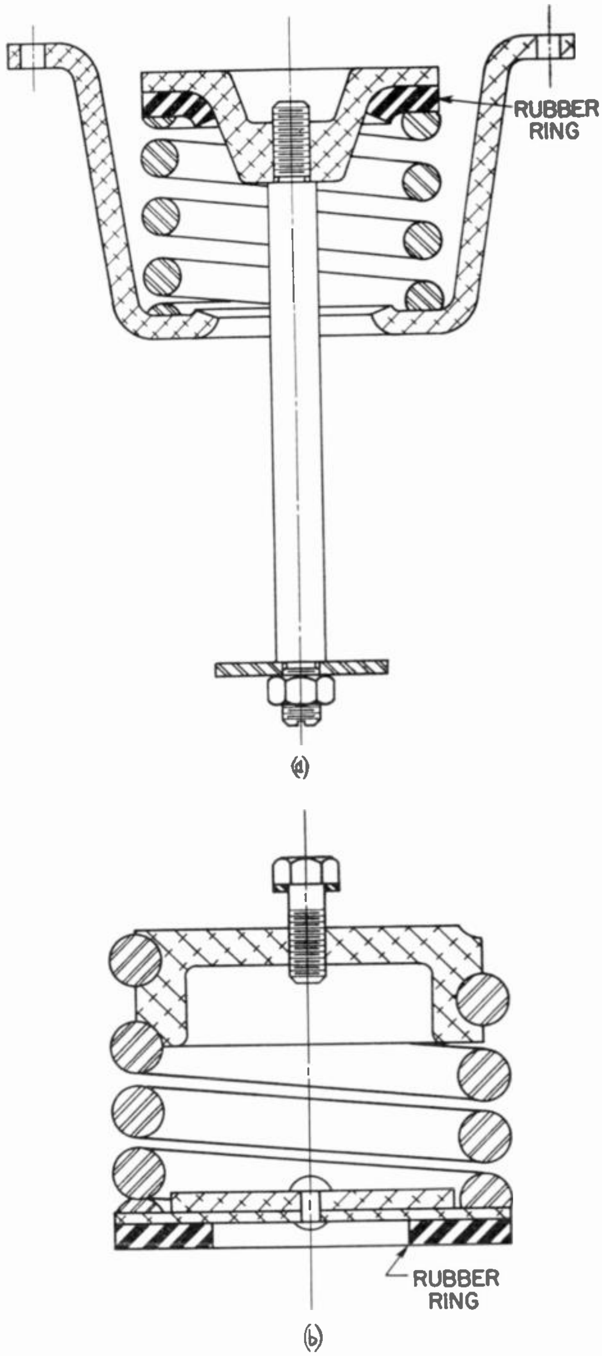

Fig. 3—(a) Ceiling spring. (b) Floor spring.

The springs reduce the transmission of mechanical vibrations from the building into the walls of the inner shell, and thence into the air inside the inner chamber. These springs, when loaded by the completed room, each have a static deflection of one-half inch, thereby giving the structure a natural frequency of mechanical vibration of 4.4 cycles per second in the vertical direction. The theoretical maximum reduction of vibration which can be produced by this system in the audible range is 26 db at 20 cps, and increases 12 db per octave at higher frequencies.

This amount of isolation is attained if the damping is zero. In order to approach this theoretical maximum isolation no resistive (or frictional) damping was employed in the spring suspension system. Such a system could build up excessive amplitudes of motion at frequencies near its natural resonance (in this case, between 3 cps and 6 cps) if subjected to sufficiently great and prolonged exciting forces. In this case it was assumed that no such forces would exist, an assumption that has so far been verified by experience.

Spring Details

The side walls and the floor of the inner chamber are all supported as a unit by one set of compression springs located under the floor, as shown in Figs. 1 and 3(b). Each spring unit is made of $\frac{1}{2}$-inch spring-steel rod wound into a helix of four inches outer diameter. There are suitable metal end caps at each end of the helix, and a $\frac{1}{4}$-inch thick rubber pad under the lower end cap. The rubber pad reduces the transmission of unwanted high frequencies through the spring and thence into the floor slab. The over-all loaded height of these units is $3\frac{1}{4}$ inches. These spring units carry a working load of 625 pounds each. The floor and ceiling springs were designed and manufactured for the job by Robinson Aviation Inc. of Teterboro, New Jersey.

The 4-inch outer diameter of the spring helix was chosen to give good mechanical stability horizontally, and to give the completed structure approximately the same natural period horizontally that it has vertically.

In the anechoic room used by the Television Division the weight of the floor slab is supported by 45 of these spring units spaced uniformly approximately 38 inches (center-to-center) under its entire area. The weight of the side walls is supported by 57 more of these units spaced approximately 15 inches apart under the floor slab around its outer edge. Each row of springs is bolted to a steel bar $\frac{3}{8}$ by 4 inches laid flat, to properly distribute the load into the concrete slab.

The Radio and "Victrola" Division room is slightly larger, and additional springs were used for the added weight.

Floor Slab and Walls

The floor slab is a 3-inch thick layer of reinforced concrete poured over 2-inch thick precast concrete planks, laid on the steel bars. This design facilitated construction since no wood forms were required under the concrete to support it during the setting period. The side walls are of concrete block and mortar construction, lined with a one-inch layer of cement plaster to seal off all leakage through the mortar or blocks. The total weight of the floor slab and walls is about 32 tons.

Ceiling Details

The ceiling of the TV Division room is very similar in construction to the floor slab, and is illustrated in Fig. 2. It does not rest on top of the side walls as in ordinary construction, but is suspended from the roof slab by 45 steel spring units uniformly spaced over the entire ceiling area (each carrying 300 pounds). This avoids deep wall-to-wall beams, saving greatly needed head room. There is a ½-inch space all around between the suspended ceiling and the walls which is packed tightly with felt to reduce sound leakage. The total weight of the ceiling (including the internal acoustical treatment), is about 7 tons.

Room Dimensions

The floor of the inner chamber is 8⅞ inches above the floor slab of the building, and the lower surface of the ceiling is 7¼ inches below the roof slab of the building. The inside head room before installation of the acoustic treatment was almost 9 feet, 2 inches. This is felt to be very good, since the head room of the building is only 10 feet, 6 inches. The floor dimensions of this room (before acoustic treatment) were 21 feet, 9 inches×20 feet, 1 inch.

The anechoic chamber used by the Radio and "Victrola" Division is similar in construction to the room described above, but differs in floor dimensions which are 21 feet, 7 inches×24 feet, 7 inches.

Air Conditioning System

The ventilation system was designed to supply an adequate volume of air at sufficiently low velocity to prevent wind noise, through a duct system designed for minimum leakage of noise from outside sources. Conditioned air from the distribution system of the building is brought into each anechoic chamber through a relatively long (14 feet) multitubular duct or plenum chamber, as illustrated in Fig. 4. Each tube of this duct is completely lined with 1-inch thick Fiberglas PF616 held in place by metal clips and by wire screen on the inside of the Fiberglas.

The group of tubes is held together with metal bands on the outside. The entire unit and suspension is completely coated with a nonhardening caulking compound applied irregularly to break up and damp out any panel resonances. The entire assembly is suspended from the building roof slab by steel spring and rubber "isolators" designed to have ½-inch deflection under load, the same as the inner chamber and its ceiling.

The duct assembly is located in the space between walls, and passes through the outer wall through a felt-lined opening. It is connected to the air distribution duct system of the building through a neoprene flexible connection to prevent the direct conduction of outside noises through the metal ductwork. The inner end of the duct opens into the inner chamber through a felt-lined opening, located near one corner of the chamber, diagonally opposite from the doorway. The exhaust is through the doorway, which is kept open at all times, except when measurements are in process. When the door is closed, the air flow is automatically stopped during measurements. This system has proved to be quite adequate in actual operation.

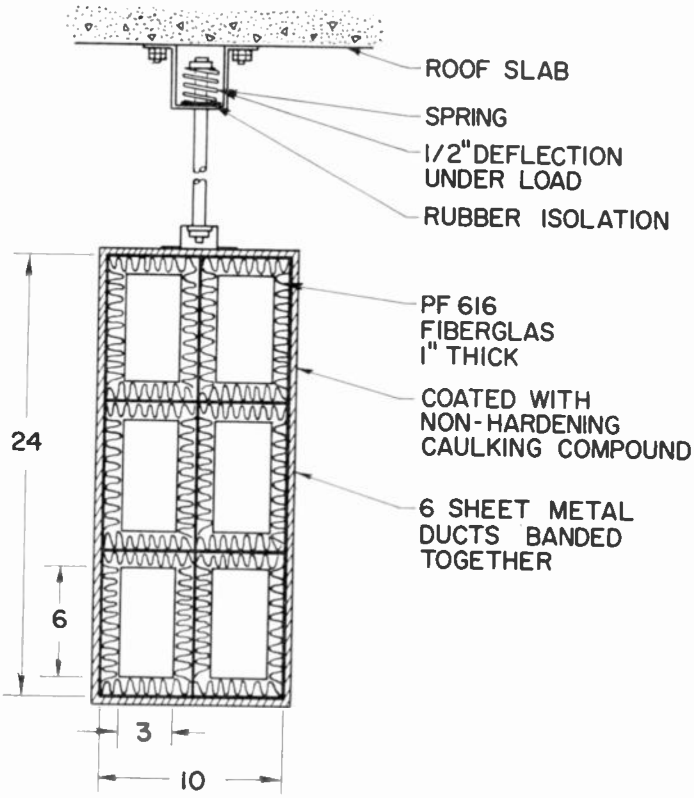

Fig. 4—Duct construction.

Soundproof Doors

The doors of the inner and outer chambers were designed and built for these rooms by the Munchhausen Soundproofing Company, Inc., of New York. They are shown in the photograph of Fig. 5, and their cross section is illustrated in the sketch of Fig. 6. They are of wood-frame and flush-panel construction, so made that the inside panel and frame and the outside panel and frame are separated from each other by a layer of felt. The inner and outer panels are held together by a dovetail construction so that no nails or other fasteners pass through the felt from one panel to the other.

When closed, the side and top leakage is sealed off by closely fitting double-gaskets of highly compliant and resilient material. The bottom is closed by two felt "drop seals" which drop when the door is closed and raise up when it is opened. Each door is closed by an ordinary door check. As a safety precaution there is no latch or lock. The inner door is rated at 35 db attenuation, and the outer door at 45 db attenuation.

Wedge Design

The internal acoustical treatment of the walls and ceiling consists of Fiberglas wedges designed in accord with the curves of Beranek and Sleeper,[2] and made by The Eckel Corporation, Cambridge, Mass. The wedges were designed to absorb nearly all energy above 125 cps.

[2] L. L. Beranek and H. P. Sleeper, Jr., "The design and construction of anechoic sound chambers," *J. Acoust. Soc. Amer.*, vol. 18, pp. 140–150; July, 1946.

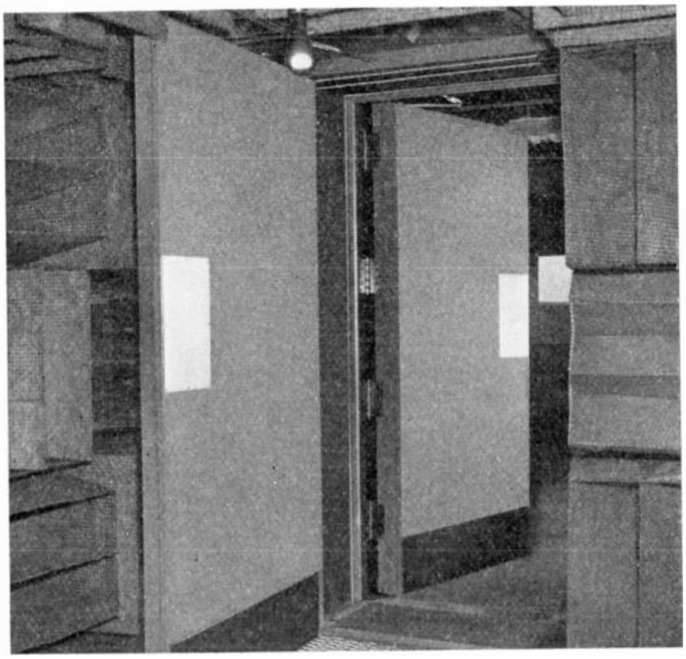

Fig. 5—A view from the interior of the TV Division Anechoic Chamber showing the double-door, soundproof construction which employs a dead-air space in between (total depth of doors and air space is about 2 feet).

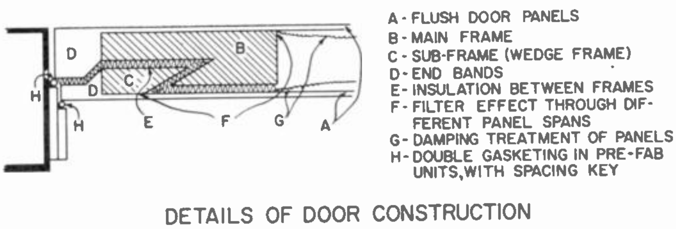

Fig. 6—Details of door construction.

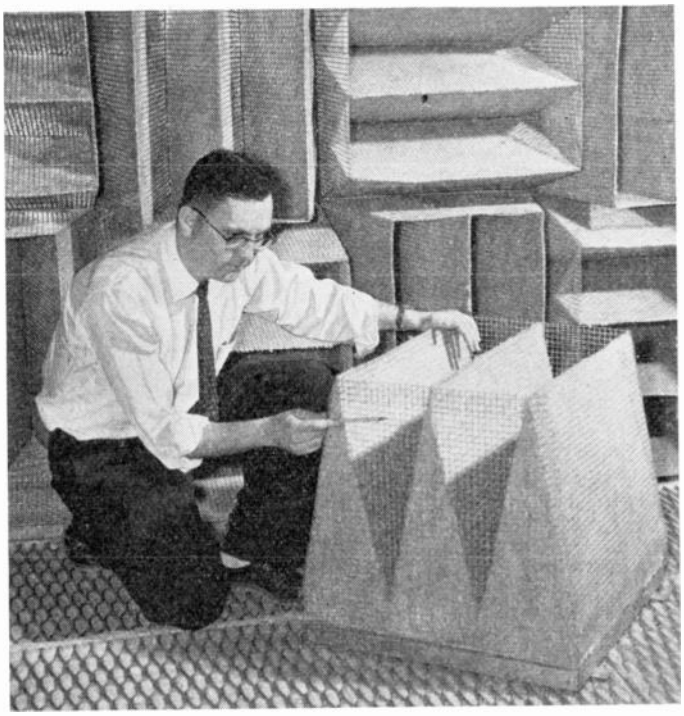

Fig. 7—Robert Libbey of the Advanced Development Section is pointing to a cube-shaped Fiberglas wedge unit like those used in the construction of the TV Division Anechoic Chamber. About 200 such units (approximately 2 feet×2 feet×2 feet) were used.

Below this point the absorption decreases with frequency. Because of headroom limitations, the wedges were built in cubes two feet on a side as shown by the photograph of Fig. 7. Each one consists of an angle-iron base with lugs for fastening to the furring strips behind them. Each wedge is covered with ½-inch mesh hardware cloth as shown. The completed room, with wedges in place, is shown by Fig. 8 (next page).

Floor Covering

The available headroom in the rooms prevented the use of wedges on the floor. This, of course, limited both the absorption of low frequencies and the isolation from external noise and vibrations. However, by careful design it is believed that optimum results were obtained from the available space.

Several types of PF Fiberglas were measured in a 4-inch acoustical transmission line to determine the absorption and the acoustical impedance. A study of the impedance as a function of frequency indicated that the maximum absorption for a layer 5 inches thick was obtained by using 1 inch of PF613 on the concrete floor covered with 1 inch of PF616, followed by 1 inch of PF613 and 2 inches of PF616. An expanded metal grating was used as a walk above the Fiberglas.

Performance of the Anechoic Chambers

The performance of an anechoic chamber is dependent upon the way it is used. In the low-frequency region reflections may produce standing waves and the intensity of these standing waves will change from one part of the room to another. Likewise, the effect of the reflections depends upon the size, number, and spacing of the sound sources. The anechoic chambers described herein are used to check instruments with several sound sources (loud speakers) as often as they are used to check on a single source in a large baffle.

Inverse-Distance Law

If the room is free from reflections, the sound pressure from a simple point source will decrease inversely with the distance from the source. That is, the pressure will decrease two to one (or six decibels) each time the distance from source to microphone is doubled. Any deviation from the inverse-distance law gives an indication of the performance of the room at the frequency of the signal.

The graph in Fig. 8 shows the performance of the room as measured with a 12-inch loudspeaker in a small box. This is an accurate copy of a normalized set of curves taken in the Television Division anechoic chamber. They have been shifted the proper amount to allow for the inverse-distance law. For a point source in a perfect room the curves should coincide. At the low-

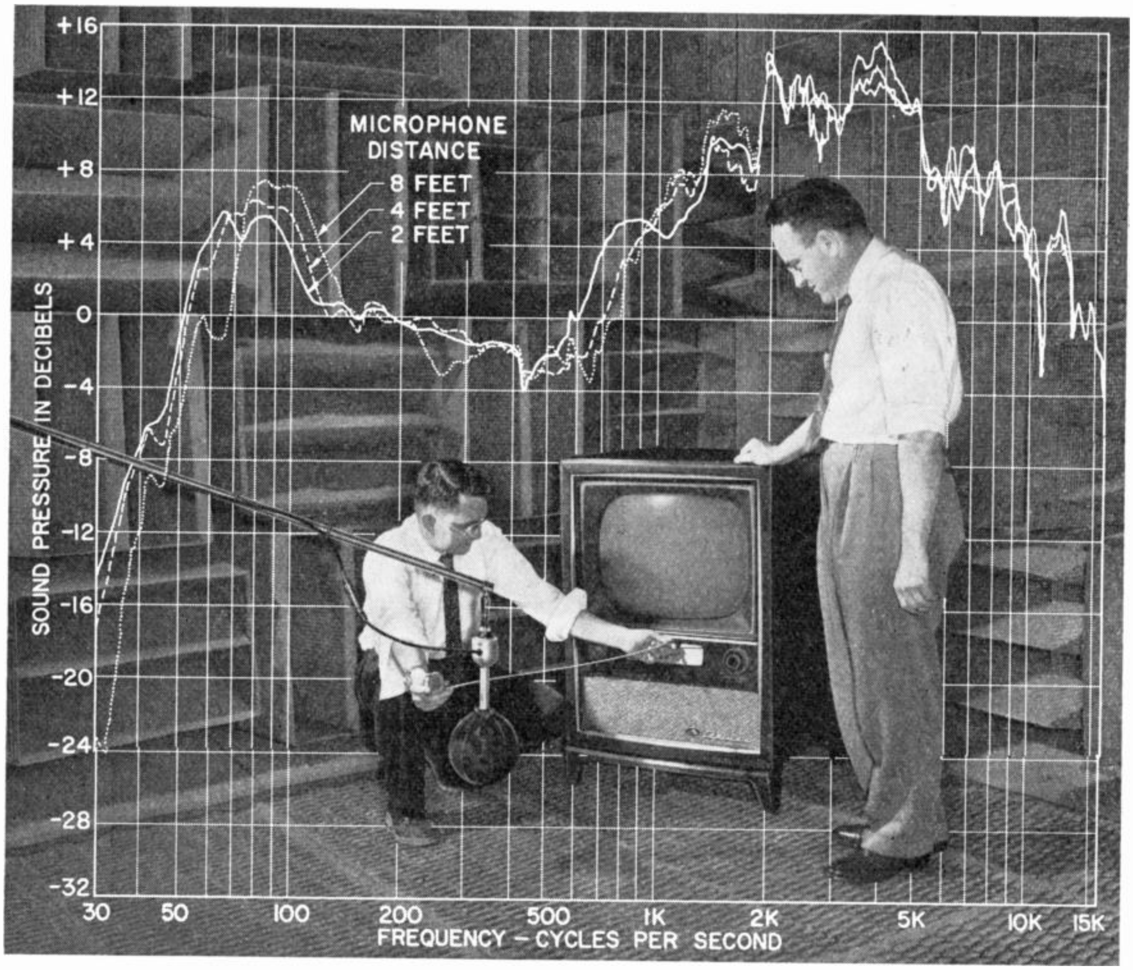

Fig. 8—Curves showing the variation of sound pressure with distance are printed over a photo which shows wedges and floor of TV Division's Anechoic Room. R. L. Libbey (left) and M. S. Corrington are adjusting the distance of the microphone from a TV receiver in preparation for acoustic measurements.

frequency end the results are within $2\frac{1}{2}$ db of ideal. In the mid-frequency range the accuracy is very good. At high frequencies the deviations are due to the changing plane of the source.

Often it is impossible to establish the exact plane of the source at high frequencies. For instance, for a 12-inch extended-range loud speaker several parts of the cone may be radiating if the frequency is above 10,000 cps.[3] In this case, the *exact* source of sound may move with frequency. Thus, it might appear that the inverse relationship is not holding.

[3] M. S. Corrington and M. C. Kidd, "Amplitude and phase measurements on loudspeaker cones," Proc. IRE, vol. 39, pp. 1021–1026; September, 1951.

Conclusion

Two anechoic chambers have been built at Cherry Hill on the second floor of the Engineering Building. They are so well isolated from the rest of the building that the ambient noise is too low to measure on the most-sensitive scale of a General Radio Sound Level Meter. They are completely air conditioned and are well suited to acoustical research and development work on radio and television receivers.

Letters to the Editor

Membership-Income-Expense Graph*

(*Letter from Mr. Angevine*)

While I was very much interested in B. B. Bauer's graph[1] on income and expense per member, I do not agree with the editorial introduction. This introduction suggests that the IRE Professional Group on Audio has only been operating since 1952. Actually the group was formed during 1948 and sponsored a session on audio at the IRE National Convention in March, 1949. I am sure Ben Bauer knows this since he was a member of the organizing committee.

Apparently many of our group membership and national officers are under a similar impression as to the age of the Audio Group. While I may be mistaken on this I believe we were the first professional group organized, or at least we were among the first two or three whose constitution and by-laws were approved by the committee of professional groups.

O. L. Angevine, Jr.
Caledonia Electronics and
Transformer Corp.

* Received by the PGA, September 5, 1956.

[1] B. B. Bauer, "Membership-Income-Expense Graph," IRE Trans., vol. AU-4, p. 84; July-August, 1956.

(*Author's comment*[2])

You are entirely right about the fact that the Professional Group on Audio was formed much prior to 1952. My graph goes back to 1952 since we had meager finances prior to that date and accurate information is lacking.

I sympathize with your regret that the heading of the graph is not clear about the age of the Group. Possibly when I redraw the graph, as I intend to, the abscissa should extend back to 1948 even if no entries are made.

Benjamin B. Bauer
Evanston, Ill.

[2] Received by the PGA, September 16, 1956.

Contributors

For a photograph and biography of Benjamin B. Bauer, see page 4 of the January–February, 1956 issue of the IRE Transactions on Audio.

❖

Errett C. Broxon was born at Akron, Ohio, on February 4, 1915. He received the Bachelor of Science degree in aeronautical engineering from the University of Cincinnati, Cincinnati, Ohio, in 1938.

Mr. Broxon has engaged in aircraft engineering at Curtiss Wright Company, Emerson Electric Company, and the Baldwin Piano Company (Aircraft Division).

In 1956 Mr. Broxon was appointed supervisor of the Engineering Services Section of the Baldwin Piano Company.

❖

Murlan S. Corrington (SM'49–F'54) was born on May 26, 1913 in S. Dakota. He received the B.S. degree in E.E. in 1934 from the South Dakota School of Mines and Technology, and the M.S. degree in 1936 from Ohio State University. From 1935 to 1937 he was a graduate assistant in the Physics Department of Ohio State University. In 1937 he joined the Rochester Institute of Technology where he taught mathematics, mechanics, and related subjects.

M. S. Corrington

Since 1942 Mr. Corrington has been engaged in mathematical engineering in the Advanced Development Section of the RCA Victor Television Division. He is presently manager of Audio, Acoustics, and Antennas for the Section.

Mr. Corrington is a member of Sigma Pi Sigma, the Acoustical Society of America, the Society for Industrial and Applied Mathematics, and is Chairman of the Philadelphia Section of IRE. He has written many technical papers in the fields of audio, fm, and circuit theory and is the author of two textbooks.

❖

Robert K. Duncan was born in Kokomo, Ind., on October 12, 1919. He received a B.S. degree in electrical engineering at Purdue University in 1941. During World War II he served in the Naval Reserve, working on sonar and other special Navy equipment.

In 1946 he joined RCA Victor Division as Acoustical Design Engineer specializing in microphone design. In 1949 he went to Capehart-Farnsworth where he engaged in radar system development.

R. K. Duncan

Since 1952 Mr. Duncan has been in the acoustical research section of the Baldwin Piano Company, specializing in acoustical transducer research and development, and acoustical measurement. He is a member of the Acoustical Society of America and the AIEE.

❖

William E. Glenn was born in Atlanta, Ga. on May 12, 1926. He received the B.E.E. degree from Georgia Institute of Technology in 1946, the M.S. degree in 1949, and the Ph.D. degree in 1952 in electrical engineering from the University of California in Berkeley.

W. E. Glenn

Dr. Glenn is a member of the Applied Physics Section of the General Electric Research Laboratory. He worked with nuclear radiation detecting instruments as an officer in the Navy in 1947. He continued this work concurrently with his graduate work at the University of California Radiation Laboratory in Berkeley until he joined the staff of the General Electric Research Laboratory in 1952. There he has been engaged in work on electro-acoustic and electro-optic transducers.

Dr. Glenn is a registered professional engineer in the state of California and a member of Sigma Xi and Eta Kappa Nu.

❖

Robert L. Libbey (A'53) was born on April 17, 1927 in Denver, Colo. He received the B.S. in E.E. degree from the University of Wyoming in 1950 and a Master's degree in speech and dramatics two years later. As an undergraduate and graduate student he was supervisor of Radio at the University of Wyoming; Chief Engineer of radio station KOWB, Laramie, Wyo., and Chief Engineer of radio station KCVO, Missoula, Mont.

R. L. Libbey

In 1952 Mr. Libbey joined RCA as a specialized trainee. Since the completion of his training program he has been a member of the advanced development section of the RCA Victor Television Division, where he has worked on magnetic recording and miniature transistor circuits. He is now engaged in audio and acoustic development and is responsible for the acoustic approval of all tv receivers.

❖

For a photograph and biography of Daniel W. Martin, see page 6 of the January–February, 1956 issue of the IRE Transasctions on Audio.

❖

Albert Meyer (S'56–A'51–M'52–SM'55) was born on January 18, 1922, in Cincinnati, Ohio. He received the E.E. degree in 1948 and the M.S. degree in electrical engineering in 1950, both from the University of Cincinnati, Cincinnati, Ohio.

During World War II, he served in the U.S. Naval Reserve, working on radar and communication equipment. While at the University of Cincinnati he served as teaching fellow in electrical engineering.

Since 1950 Mr. Meyer has engaged in electronic circuit development at the Baldwin Piano Company, in the Acoustical and Electronic Research Sections.

❖

Sydney V. Perry (A'35–VA'39) is a native of England, and was graduated from Queen's University at Kingston, Ontario, Canada in 1923. He then joined the Westinghouse organization, which was at that time affiliated with RCA; and he has been a member of the RCA organization ever since. He has been engaged in the development and design of phonograph pickups, microphones, loudspeakers, loudspeaker cabinets and enclosures, and entire audio and acoustic reproducing systems intended for use in radio, phonograph, and television instruments.

S. V. Perry

Mr. Perry is the holder of several patents on acoustic systems and devices.

Index to

IRE TRANSACTIONS

ON

AUDIO

Volume AU-4, 1956

IRE Professional Group on Audio Combined Index for 1956

Use of the Index

The combined indexes which follow are of three types. The first type is simply a compilation of Tables of Contents of IRE TRANSACTIONS ON AUDIO and the Audio portion of the CONVENTION RECORD. Next is an Author Index. The third type is an Analytic Subject Index.

The volume system of indexing began with the first issue of 1953. Volume AU-3, issues 1, 2 . . . 6, contain all IRE-PGA publications for the year 1956 and the Audio part of Section 7 of the 1956 CONVENTION RECORD. In the Tables of Contents, this section is numbered CR-4-7, and inserted chronologically between issues AU-4-4 and AU-4-5, IRE TRANSACTIONS ON AUDIO.

In both the Author Index and the Analytic Subject Index references are made to publications by the issue and page number in the volume (*e.g.*, AU-4-2, 32).

The Analytic Subject Index lists titles under the appropriate classifications in the series shown below. In many cases valuable material is published under titles which cannot be fully descriptive of all the material which the paper covers. It is for this reason that some titles are listed under various classifications, some of which may seem inappropriate to the title itself. It is hoped that this will increase the probability of finding quickly most of the information available under a particular classification.

Classification of Subjects

1. IRE-PGA
   - 1.1 General
   - 1.2 Constitution and By-Laws
   - 1.3 National and Regional Meetings
   - 1.4 Chapters
   - 1.5 Membership
   - 1.6 Transactions
   - 1.7 People
2. Bibliographies, Reviews, Standards, Tapescripts
   - 2.1 Bibliographies
   - 2.2 Reviews
   - 2.3 Standards
   - 2.4 Tapescripts
3. Sound Systems
   - 3.1 General
   - 3.2 Stereophonic, Binaural, and Spatial Effects
   - 3.3 Military
4. Microphones
   - 4.1 General
   - 4.2 Condenser
   - 4.3 Magnetic
   - 4.4 Crystal
   - 4.5 Moving—Coil
   - 4.6 Ribbon
   - 4.9 Special Microphones
5. Amplifiers
   - 5.1 General
   - 5.2 Preamplifiers and Voltage Amplifiers
   - 5.3 Power Amplifiers
   - 5.4 Frequency-Range Dividing Networks
   - 5.5 Transistorized Amplifiers
   - 5.9 Special Amplifiers
6. Loudspeakers
   - 6.1 General
   - 6.2 Direct-Radiator Units
   - 6.3 Horn-Driver Units
   - 6.4 Horns, Enclosures, Baffles
   - 6.9 Special Types
7. Disk Recording and Reproduction
   - 7.1 General
   - 7.2 Disks
   - 7.3 Recording
   - 7.4 Pickups and Tone Arms
   - 7.5 Pre-emphasis and Postequalization
   - 7.9 Special Mechanical Recorders
8. Magnetic Recording and Reproduction
   - 8.1 General
   - 8.2 Tape and Wire
   - 8.3 Recording and Erasing
   - 8.4 Playback
   - 8.5 Pre-emphasis and Postequalization
   - 8.9 Special Magnetic Recorders
9. Acoustics
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   - 9.2 Room Acoustics
   - 9.3 Sound Waves and Vibrations
   - 9.4 Speech
   - 9.5 Music
   - 9.6 Hearing
   - 9.7 Psychoacoustics
10. Broadcast Audio
11. Audio Measuring Equipment and Techniques
12. Electronic Musical Instruments

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ANALYTIC SUBJECT INDEX FOR 1956 IRE-PGA PUBLICATIONS

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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.

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PERMOFLUX CORPORATION, 4900 West Grand Ave., Chicago 39, Illinois
Loudspeakers, Headphones, Cee-Cors (Hipersil Transformer Cores)

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

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THE MAICO COMPANY, INC., 21 North Third St., Minneapolis I, Minnesota
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THE TURNER COMPANY, Cedar Rapids, Iowa
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UNITED TRANSFORMER COMPANY, 150 Varick St., New York, New York
Transformers, Filters, and Reactors

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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 Chairman of the IRE-PGA Ways and Means Committee, Michel Copel, 156 Olive St., Huntington, Long Island, New York.