IRE Transactions on AUDIO

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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

REPORT OF THE IRE-PGA AWARDS COMMITTEE, 1955–1956

The committee voted to make no nominations this year for the Browder Thompson Award. It was decided to make the official period for award considerations the year ending July 1 of the year preceding that in which the awards are made.

After consideration of audio contributions to PROCEEDINGS, TRANSACTIONS, and CONVENTION RECORD, the Awards Committee recommends the following nominations for IRE-PGA Awards in 1956:

PGA Achievement Award
Harry F. Olson

PGA Senior Award
William B. Snow

PGA Award
J. Barry Oakes

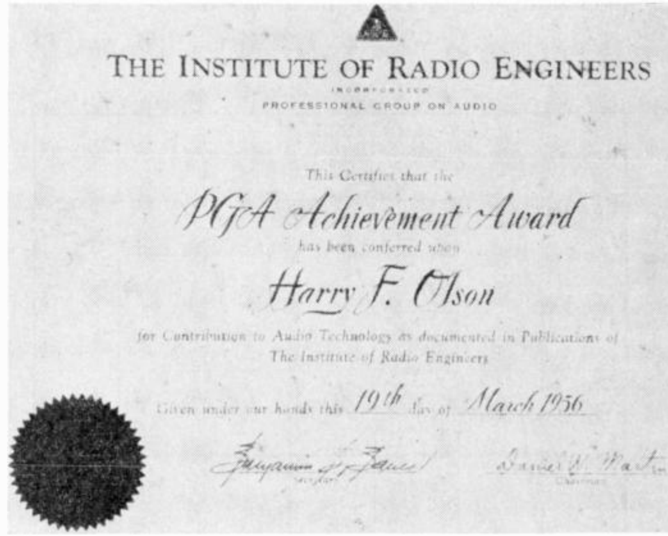
With regard to point three of the minutes of the October 4, 1955, meeting of the Administrative Committee, the Awards Committee recommends that the following announcement be made in the IRE STUDENT QUARTERLY, and also be sent to all faculty representatives of IRE student chapters:

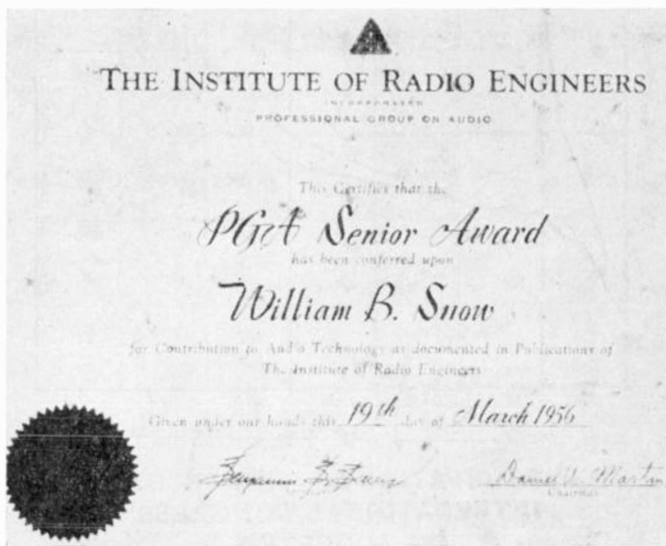
"The IRE Professional Group on Audio announces the establishment of an annual Student Papers Competition in Audio. All undergraduates are eligible. Papers must be related to audio, and must be submitted to the PGA Awards Committee by June 1. One award of one hundred dollars will be made, and will be announced at the fall meeting of the IRE-PGA. All papers submitted will also be considered for possible publication in IRE TRANSACTIONS ON AUDIO."

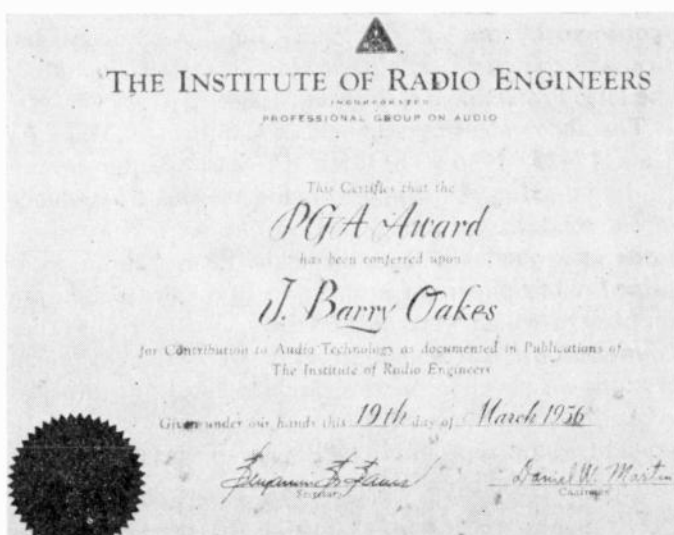
DANIEL W. MARTIN
Chairman, Awards Committee

PGA AWARD CERTIFICATES

The certificates opposite were presented to Harry F. Olson, William B. Snow, and J. Barry Oakes as described in the Awards Committee Report. For the nature and amount of these awards refer to the March-April, 1955, and July-August, 1955, issues of the IRE TRANSACTIONS ON AUDIO.

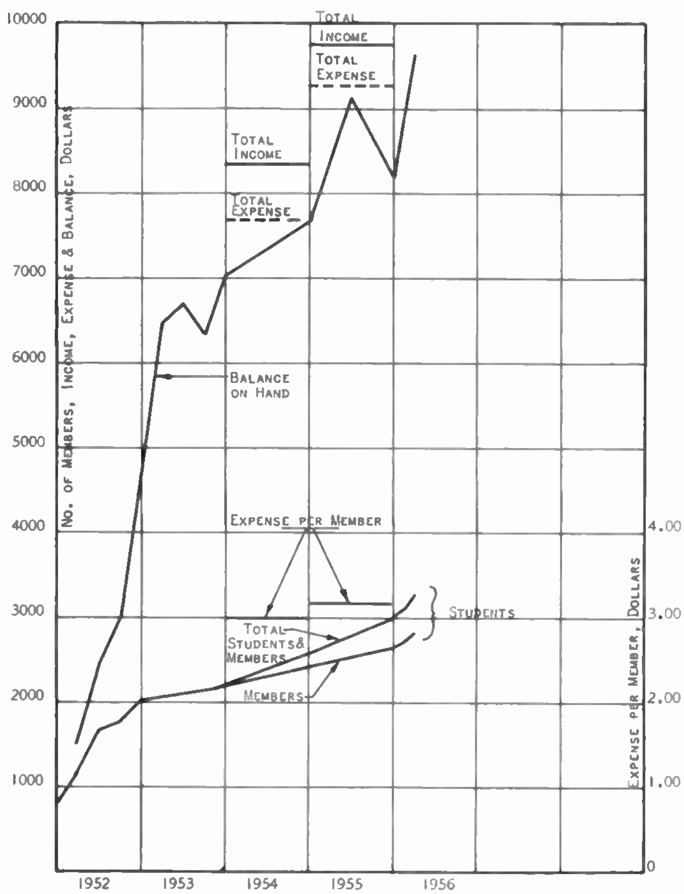

THE INSTITUTE OF RADIO ENGINEERS
PROFESSIONAL GROUP ON AUDIO
This Certifies that the
PGA Achievement Award
has been conferred upon
Harry F. Olson
for Contribution to Audio Technology as documented in Publications of The Institute of Radio Engineers
Given under our hands this 19th day of March 1956


THE INSTITUTE OF RADIO ENGINEERS
PROFESSIONAL GROUP ON AUDIO
This Certifies that the
PGA Senior Award
has been conferred upon
William B. Snow
for Contribution to Audio Technology as documented in Publications of The Institute of Radio Engineers
Given under our hands this 19th day of March 1956


THE INSTITUTE OF RADIO ENGINEERS
PROFESSIONAL GROUP ON AUDIO
This Certifies that the
PGA Award
has been conferred upon
J. Barry Oakes
for Contribution to Audio Technology as documented in Publications of The Institute of Radio Engineers
Given under our hands this 19th day of March 1956

MEMBERSHIP-INCOME-EXPENSE GRAPH

The following graph, prepared by B. B. Bauer, shows the manner in which the membership, the expense per member, and the total income and expense has varied during the life of the IRE Professional Group on Audio.



PARTICIPATION IN THE SECOND INTERNATIONAL CONGRESS ON ACOUSTICS

(Organized by the Acoustical Society of America and sponsored by the International Commission on Acoustics and various participating organizations including the IRE Professional Group on Audio.)

The above meeting was held in Cambridge, Mass. on June 17–23, 1956. The IRE Professional Group on Audio joined in the support of this meeting as evidence of our solidarity of interest with other similar organizations in acoustics and audio fields. The program contained many papers of interest to PGA membership, a number of which were authored by visitors from other countries. Of particular interest were the technical sessions on psychoacoustics, architectural and musical acoustics, electroacoustics, transducers, and loudspeakers and sound reproduction. Papers of specific interest included the following:

"Principle Considerations on the Artistic Qualities of Musical Sound," *Erich Thienhaus.*

"Some Problems of Reproduction and Perception of Electronic Tape Music," *Werner Mayer-Eppler.*

"Acoustical and Electrical Considerations in Approaching Facsimile Reproduction of the Symphony Orchestra," *Walter T. Selsted and Ross H. Snyder.*

"Audibility of Nonlinear Distortion in Sound Transmission Systems," *A. V. Rimsky-Korsakov.*

"Transient Response of Loudspeaker Paper Diaphragms," *L. Alons.*

"Some Instrument Aids in Use in Western Germany for Checking Professional Standards in Audio Equipment," *Wilhelm Franz.*

"High Fidelity Philosophy from a European Standpoint," *J. Rodrigues de Miranda.*

"A Distributed Loudspeaker," *Sipko L. Boersma.*

"The Effect of a Negative Impedance Source on Loudspeaker Performance," *Richard E. Werner.*

"Learning, A Major Factor Influencing Preferences for High Fidelity Reproducing Systems," *Roger E. Kirk.*

"The E. M. I. Stereo Recording and Reproducing Systems," *G. F. Dutton.*

PARTICIPATION IN THE NATIONAL ELECTRONICS CONFERENCE

This year the PGA is again participating in the NEC by sponsoring an Audio Session. Since the specific time and location of the session had not been decided at the time this issue went to press, the Program of the NEC should be consulted for further details. The NEC this year will be held at the Hotel Sherman in Chicago from October 1–3.

PGA CHAPTER ACTIVITIES

Boston, Mass.

On May 24 Professor Frederick V. Hunt presented an illustrated historical essay on the "Origins of Electroacoustics." Prof. Hunt's treatment of the background and early days of electroacoustic transduction made a stimulating evening for all who attended.

Born in Ohio, "Ted" Hunt took the B.A. and B.E.E. degrees at Ohio State, after which he trekked to Cambridge to complete the roster with a Master's and a Doctorate in Physics at Harvard. He is now Rumford Professor of Physics and Gordon McKay Professor of Applied Physics at the older of Cambridge's two large institutions of higher learning. During the war Dr. Hunt was director of the Harvard Underwater Sound Laboratory, and in 1947 he received the Presidential Medal for Merit. He is a Fellow of the American Academy of Arts and Sciences, the Physical Society, the IRE, and the Acoustical Society of America. He is an honorary member of the Audio Engineering Society.

Chicago, Ill.

According to a release from the Armour Research Foundation of the Illinois Institute of Technology in Chicago, Dr. Robert W. Benson, who has been supervisor of the acoustics design section since 1954, has been named assistant manager of the physics research department. Dr. Benson will be responsible for three acoustic research sections: acoustic design, sound and vibration control, and the Riverbank Acoustical Laboratory.

Prior to joining the Foundation, Dr. Benson was in charge of physical research at the Central Institute for the Deaf, St. Louis. He also was on the staff of Washington University, St. Louis, as assistant professor in the electrical engineering department, assistant professor in electrical research at the dental school, and instructor in the audiology department.

Dr. Benson is a member of the Acoustical Society of America, IRE, and Sigma Xi. He also served as chairman of the National Noise Abatement Symposium in 1955.

San Francisco, Calif.

On May 22 the San Francisco Chapter held a joint meeting with the San Francisco Section of the Audio Engineering Society. At this meeting Charles P. Ginsburg, chief video engineer of the Ampex Corporation, spoke on the new "Videotape Recording System." In this system a high-speed rotating-head assembly permits transverse recording and reproduction of the full 4-mc video bandwidth on a 2-inch-wide tape.

In announcing the achievement of what has been widely considered "the impossible," it was stated that Ampex has under way the development of a system for recording programs in full color as a logical extension of this development.

Charles Ginsburg joined Ampex as development engineer in 1952 and has been responsible for various projects developing both audio and instrumentation magnetic tape recording equipment.

From 1947–1952 he was associated with KQW in San Jose and from 1942–1947 with Associated Broadcasters as operations engineer. During 1940–1941, he was with Electrical Communication Company of San Francisco and McCune Sound Services as development engineer.

Mr. Ginsburg attended the University of California and San Jose State and received the A.B. degree in Mathematics with a minor in electrical engineering.

WITH OTHER ACOUSTICAL AND AUDIO SOCIETIES

The March, 1956, issue of the *Journal of the Acoustical Society of America* contains 18 papers on theoretical and applied acoustics. While few of these may be classified as being directly in the field of audio technology, many of them will be of interest to those who wish to further their acoustical knowledge.

Three papers deal with the propagation of sound in tubes and through fluid saturated porous solids.

Two of the papers describe the biological effect of high intensity sound.

Three papers deal with thermoacoustic effects.

G. J. Thiessen describes the "Performance of a Type B Fog Horn."

C. M. Davis and S. F. Ferebee offer a paper on "Dynamic Magnetostrictive Properties of Alfenol." Alfenol is the new aluminum-iron alloy which combines a high permeability with extreme hardness.

R. M. Hoover presents a paper on "High Power Operation of a Magnetostrictive Transducer."

The remaining papers are directed to psychoacoustics and theory.

The issue contains its usual excellent references to contemporary papers on acoustics by R. N. Thurston and a "Review of Acoustical Patents" by R. W. Young.

BENJAMIN B. BAUER

Letters to the Editor

Efficiency and Power Rating of Loudspeakers*

We have read with a great deal of interest the above article.[1]

We are especially interested in a graph shown in Fig. 6 in which the Electro-Voice CDP Compound Diffraction Projector was compared to the University LB35 and SA30 drivers mounted on an SMH horn.

We believe that several errors may have been made in this survey and therefore wish to clear up a few points which presently bother us. First, the CDP is rated at an input power of 25 watts and it would seem to us that this unit has been compared with a 30-watt driver unit in the University family. We acknowledge that absolute power ratings are not necessarily a function of efficiency, but we must point out that the University drivers have one-pound magnets whereas the Electro-Voice CDP has a magnet weight of 8 ounces. It seems to us that if this comparison were reduced to an absurdity and the *same difference* of weights maintained that a driver unit having an 8-ounce magnet would be infinitely more efficient than one having a magnet weight of zero ounces!

Another interesting thing is that we cannot find a correlation between the presently accepted RETMA sensitivity rating method and that found by Benson for efficiency.

If we assume that doubling magnet weight in a driver of identical structure increases efficiency 3 db, and indeed this is the case if the structure does not saturate, we will see that the Electro-Voice CDP will compare quite well with the other driver units under consideration, and indeed surpass them at certain points in the frequency response spectrum. We honestly feel quite strongly about this point since we have been intimately concerned with relative efficiencies of loud-speaker systems for quite some time, and though we admit that comparison for comparison's sake is quite interesting, if the analysis is to be a rigorous one, then weighting factors must be applied for disparateness of mechanical and magnetic construction as well as for electrical losses in the system.

CULLEN H. MACPHERSON
Manager
Electro-Voice, Inc.
Buchanan, Mich.

* Dated March 2, 1956.
[1] R. W. Benson, vol. AU-4, pp. 19–23; January–February, 1956.

Rebuttal*

I can see the concern about the graphs shown in Fig. 6, interpreted as indicating that your loudspeaker, as tested in this study, may look inferior to other loudspeakers also tested in the same study. As pointed out in my article, the measurement of the efficiency of loudspeakers is a desirable characteristic so far as the design of indoor sound systems is concerned. There is no indication in the article that it is desirable to use the highest efficiency unit available, but I did indicate that whatever the efficiency is, it would be helpful in calculating the performance of a sound system.

As is well known, the performance of a loudspeaker is very seldom described by a measure of efficiency and certainly the choice of a loudspeaker, with the exception in specialized military uses, is almost never determined by the specific efficiency of a unit. Such factors as quality of reproduction, directional characteristics, and cost are more important. Even if efficiency becomes an important factor, it is important to interpret the value of a loudspeaker on either a weight vs sound output basis, or a cost vs sound output basis. I had no intention in the article to discuss either of these factors and only wanted to point out that efficiency measurements can be made quite easily and are certainly important not for the selection of a loudspeaker, but for the calculation of performance of an over-all sound system.

I have no disagreement with the comparisons attempted in your letter and hope that this letter will clarify the original intent of my article.

R. W. BENSON
Armour Res. Foundation of Ill. Inst. of Tech.
Chicago, Ill.

* Dated March 15, 1956.

Bells, Electronic Carillons, and Chimes*

The above paper[1] presents a subject which has been of intriguing interest to me during the past 25 years of research in this and related fields. In fact, I did consulting work for Liberty Carillons (electronic) which Stromberg-Carlson Co. absorbed some years ago.

There are several important details of the tone generators for electronic carillons which Slaymaker has not explained, and which may interest those who have read his paper.

Unlike the exactly-integral, frequency relationships among the partials of continuously-maintained, musical-tone generators, percussively excited, fixed-free, uniform-section rods, such as those used in some of these instruments, have a very much different partial spectrum. Slaymaker's musical staff hardly shows this clearly for engineers; besides, it cannot indicate frequencies closer than 1/12 octave. Table I compares, on a relative frequency basis, the spectrum of a normal, musical, sustained tone, with that of an impulsively excited, fixed-free rod.

TABLE I

Partial	I	II	III	IV	V	VI	VII	VIII	IX	X
Normal Tone	1	2	3	4	5	6	7	8	9	10
Fixed Free Rod*	1	6.27	17.55	34.39	56.9	85	118.5	157.2	202	253

* The values for the rod are approximate.

It is seen that, frequency-wise, the lower partials of the rod are quite far apart, for the lowest numbered partials, and gradually crowd together more for those of higher number.

If all of these partials, down to I, were used in the output tone, nothing like a bell sound would be produced, since their relative amplitudes vary roughly inversely with the absolute frequency.

Hardly anything except partial I would be heard, it being more or less six times the amplitude of II, over 17 times that of III, and so on, depending on the exciting and pickup points along the rod.

In order, therefore, to pull the partials of the output tone closer together in frequency, and reduce the great disparity of their amplitudes, it has long been customary to use a high pass filter in the audio amplifier input, to remove at least partials I and II of the rod. The relative frequencies of the remaining partials, from III upward, then become Table II.

TABLE II

I	II	III	IV	V	VI	VII	VIII
1.	1.96	3.26	4.85	6.78	9.	11.53	14.5

That is, partial III of the rod now becomes partial I of the output tone, and so on.

While 1. to 1.96 is not quite an octave, we thus have approximately the first two components of a harmonic series, to provide some semblance of an aural pitch determinant. All of the other partials, except the seventh, are out of a normal Fourier series. However, III, which is 3.27 times I in frequency, is not too far removed from a C-to-the-second-A-above-harmony (which has a ratio of 3.36) to be usable, if not ideally harmonious. The higher partials must generally be inharmonic to make what our ears recognize as a bell tone.

In order to prevent the out-of-tuneness, which Slaymaker describes, it is very desirable that three or four among the lowest partials fall exactly in a true harmonic series, else the tone pitch is easily confused and indefinite.

To clarify the pitch characteristic of bell tones, one manufacturer uses two independent, round rods per note, and Stromberg-Carlson uses one dual-mode rod of rectangular section. The pitches of the two vibration series of such generators is chosen to weight the pitch of their output tone on one, rather

* Dated March 3, 1956.
[1] F. H. Slaymaker, vol. AU-4, pp. 24–26; January–February, 1956.

than two or three possible pitch judgments. I have used both of these plans for many years to produce true bell tones with a very definite aural pitch characteristic. I have also patented rod type, bell-tone generators of other types for achieving these same objects. One of these involves a free end load; another involves one or more attached vibrators for adding other desired frequency components; a third involves critically-curved rods; a fourth involves synthesis of all of the desired components, each with the desired damping rate and amplitude, by use of the fundamental components only of a plurality of much smaller, reed type vibrators which are simultaneously excited.

It has long been customary to use fixed-end grooving, such as Slaymaker describes, for improvement in bell tone quality, and not alone in electronic carillons. Rod type clock chimes of this type have been in use for at least 50 years, perhaps more. The effect of this grooving is to bring the lower partials into still closer harmony. In clock chimes and orchestral chime tubes, the cutting off of the lowest partials is inherent, because the air coupling for these very low frequencies is insufficient for appreciable sound radiation. It may be added that this cutoff action operates differently for these bell tone generators than it does for generators of a true harmonic series of partials.

In the latter, even though many of the lower partials are removed, the aural pitch remains as the first partial of the whole series, because the beats between any two adjacent partials is always equal in frequency to that of the removed first partial—and therefore supplies the fundamental subjecttively. This occurs also for the second partial, which is thus produced by beat action between partials separated by one partial of the series, and so on, for the other cutoff partials.

There is an important limitation of such rod type generators as to usable pitch compass. When this, in octaves, exceeds the 2.79 (about 1.5 octave) frequency ratio, between partials II and III of the lowest pitch rod, partial II of the rod will become more and more audible as the compass is increased, because, on an absolute frequency basis, it has risen above the filter cutoff point. Being inharmonic, and lower in pitch than the desired fundamental of the output tone, it yields an unpleasant growl-like component to the output tone. Additionally, the pitch becomes confused because the strike tone pitch, higher than the overtone in loudness and frequency, is heard first, while the low, growl-tone is heard last, because its damping rate is lower.

This is an interesting and baffling subject. One wishes to make the bell tone as useful, solo wise and chordally, as other musical tones but, if this is carried too far, the ear says it is no longer a bell tone. If one goes back in the other direction too far, he may get a bell tone, but its pitch becomes indefinite, and it is unusable chorally, because of multiple clashes among the partials of the elements of the chord. So the design must become a never-ideal compromise.

Benjamin F. Miessner
Miessner Inventions, Inc.
Morristown, N. J.

Rebuttal*

Miessner's position among the pioneers in electronic musical instrument development is unique in this country and I was very pleased to see his comments on my paper. As he points out, many of the features which are used in electronic carillons are old. For example, necked, as well as weighted rods, were used on clock gongs long before the development of electronic musical instruments. Miessner, and his fellow pioneers, used some of these features when they made many important developments. In a similar manner, we may have used some old features when we developed our patented musical instrument using rods that have rectangular cross sections. I was interested to note that some of Miessner's patents were among those which were cited by the Patent Office as being of historical significance when they allowed our patent (No. 2,690,091).

There is a baffling aura of mysticism about bell tones and the psychological aspects of pitch determination which makes the study of bell tones very tantalizing. Miessner reiterates, in somewhat more complete form by using two additional tables, the information I had given on the inharmonic relationship between the frequencies of the various partials and the wide spacing of the first, second, and third modes of vibration in a vibrating rod. His observations concerning the different effects of varying the low frequency cutoff for rod tones and those that are made of a harmonic series are interesting and quite pertinent. In my paper, I confined my attention to the partials that are actually reproduced in an electronic carillon.

Although Miessner shows the ratio between the third and fourth modes of vibration to be less than an octave, it is possible, by using a rod with a neck at the clamping point to make the ratio between the third and fourth modes an exact octave instead of some 35 cents shy of an octave as it would be with no neck. The hum tone and strike tone of Table II are the third and fourth modes obtained from vibration perpendicular to the wide side of the rectangular rod and the octave can be made as true as it can be measured on the chromatic stroboscope.

Frank H. Slaymaker
Stromberg-Carlson, Inc.
Rochester, N. Y.

* Dated April 23, 1956.

List of Published Standards that May Be Applied to High Fidelity Equipment

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W. E. Kock (PGA)
Henry Kogel (SMPTE)
C. J. LeBel (AES, MRIA, and IRE 19)
R. C. Moyer (IRE 19)
A. P. G. Peterson (IRE 19)
A. A. Pulley (AES)
H. E. Roys (IRE 19)
L. Thompson (IRE 19)
A. F. Zuckerman (RETMA)

INTRODUCTION

FIDELITY in reproduction of sound is depreciated by many factors such as limited frequency range, nonlinear distortion, flutter, and other characteristics. In commercial equipment these depreciating factors can be minimized only by compromise. The compromises are often interrelated. At this stage of the art no one is competent to indicate how much weight should be given to each of the factors and there may be factors still unknown which depreciate the fidelity of reproduction. Accordingly the degree of fidelity designated as "High Fidelity" is not herein defined.

The Committee felt it would be useful at this time to collect all known published standards for the use of those who wish to evaluate systems of varying fidelity. Some of these, particularly IRE Standards, define measuremental procedures while others apply to numerical values and are to be applied as guides in the rapidly changing art. This Committee has listed these standards, but has not attempted to scrutinize or criticize them in detail. Therefore, it does not guarantee their accuracy or does it necessarily confirm the numerical values particularly in the standards of the commercial associations.

1.00 IRE

Standard		*Price*
1.01	**54 IRE 3. S1.** Standards on Audio Techniques: Definitions of Terms, 1954. Reprinted from PROCEEDINGS OF THE IRE, July, 1954.	$0.50
1.02	**53 IRE 3. S2.** Standards on American Recommended Practice for Volume Measurements of Electrical Speech and Program Waves, 1953. Adopted by ASA. (ASA C16.5—1954). Reprinted from PROCEEDINGS OF THE IRE, May, 1954.	0.50
1.03	**56 IRE 3. S1.** Standards on Audio Techniques: Methods of Measurement of Gain, Amplification, Attenuation, Loss and Amplitude-Frequency Response. Reprinted from PROCEEDINGS OF THE IRE, April, 1956.	0.80
1.04	**42 IRE 6. S1.** American Recommended Practice for Loudspeaker Testing. Adopted by ASA. (ASA C16.4—1942).	0.40
1.05	**51 IRE 6. S1.** Standards on Electroacoustics: Definitions of Terms, 1951. Reprinted from PROCEEDINGS OF THE IRE, May, 1951.	1.00
1.06	**47 IRE 17. S1.** Standards on Radio Receivers: Methods of Testing Frequency-Modulation Broadcast Receivers, 1947. Adopted by ASA. (ASA C16.12—1949).	.50
1.07	**48 IRE 17. S1.** Standards on Radio Receivers: Methods of Testing Amplitude-Modulation Broadcast Receivers, 1948. Adopted by ASA. (ASA C16.19—1951).	1.00
1.08	**49 IRE 17. S1.** Tests for Effects of Mistuning and for Downward Modulation. 1949 Supplement to 47 IRE 17. S1. Adopted by ASA. (ASA C16.12a—1951). Reprinted from PROCEEDINGS OF THE IRE, December, 1949.	0.25
1.09	**52 IRE 17. S1.** Standards on Receivers: Definitions of Terms, 1952. Reprinted from PROCEEDINGS OF THE IRE, December, 1952.	0.60

Standard		*Price*
1.10	**53 IRE 19. S1.** Standards on Sound Recording and Reproducing: Methods of Measurement of Noise, 1953. Reprinted from PROCEEDINGS OF THE IRE, April, 1953.	$0.50
1.11	**53 IRE 19. S2.** Standards on Sound Recording and Reproducing: Methods for Determining Flutter Content, 1953. Adopted by ASA. (ASA Z57.1—1954). Reprinted from PROCEEDINGS OF THE IRE, March, 1954.	0.75

2.00 ASA

Standard		*Price*
2.01	**C16.4—1942.** Loudspeaker Testing. (42 IRE 6. S1).	$0.50
2.02	**C16.5—1954.** Volume Measurements of Electrical Speech and Program Waves. (53 IRE 3. S2).	0.50
2.03	**C16.12—1949, C16.12a—1951.** Methods of Testing Frequency-Modulation Broadcast Receivers, with Supplement. (47 IRE 17. S1, 49 IRE 17. S1).	1.25
2.04	**C16.19—1951.** Methods of Testing Amplitude-Modulation Broadcast Receivers. (48 IRE 17. S1).	1.00
2.05	**Z57.1—1954.** Method for Determining Flutter Content of Sound Recorders and Reproducers. (53 IRE 19. S2).	0.75
2.06	**Z24.1—1951.** Acoustical Terminology.	1.50
2.07	**Z22.51—1946.** Method of Making Intermodulation Tests on Variable Density 16 mm Sound Motion Picture Prints.	0.25
2.08	**PH22.52—1954.** Cross Modulation Test 16 mm Variable Area Photographic Sound.	0.25
2.09	**Z24.11—1954.** Method for Free Field Secondary Calibration of Microphones.	0.50
2.10	**Z24.4—1949.** Method for Pressure Calibration of Laboratory Standard Pressure Microphones.	0.75

3.00 RETMA

Standard		*Price*
3.01	**SE-101-A.** Amplifiers for Sound Equipment.	$0.25
3.02	**SE-103.** Speakers for Sound Equipment.	0.30
3.03	**SE-104.** Engineering Specifications for Amplifiers for Sound Equipment.	0.25
3.04	**SE-105.** Microphones for Sound Equipment.	0.90
3.05	**TR-107.** Electrical Performance Standards for FM Broadcast Transmitters.	0.25
3.06	**REC-134.** Magnetic Recorders-Conditions for Measurements and Definitions.	0.30
3.07	***REC-146.** Lateral Disk Recording Characteristic.	0.25
3.08	**TR-105B.** Audio Facilities for Radio Broadcasting Systems.	0.35

Note: Your attention is called to SE-8-5287-3 "Tentative RETMA Standard: Amplifiers for High Fidelity Equipment," which may be made available in revised form at a later date.

4.00 AES

Standard		*Price*
4.01	***TSA-1-1954.** Standard Playback Characteristic for Lateral Disk Recording.	No charge

* The recording and reproducing characteristics of these are substantially equivalent.

5.00 NARTB

Standard		*Price*
5.01	***Supplement No. 2 to NAB (NARTB).** "Engineering Handbook" (Fourth Edition, 1949), NARTB Recording and Reproducing Standards. (June, 1953).	$1.00

6.00 RIAA

Standard		*Price*
6.01	*Standard on Recording and Reproducing Characteristic.	No charge

7.00 MRIA

Standard *Price*

(Standards of MRIA may become available later)

Note: Reference is also made to British Standard No. 1928: 1955, "Gramophone Records, Transcription Disc Recordings, and Disc Recording Equipment" and to Armour Research Foundation, Bulletin No. 92, Magnetic Recorder Licensee Service, "Magnetic Recording Standardization."

LIST OF SOURCES OF STANDARDS WITH ADDRESSES

The Institute of Radio Engineers, Inc.,
1 East 79 Street,
New York 21, New York.

American Standards Association,
70 East 45th Street,
New York, New York.

Radio-Electronics Television Manufacturers Association,
11 West 42nd Street,
New York, New York.

Audio Engineering Society,
P. O. Box 12,
Old Chelsea Station,
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National Association of Radio & Television Broadcasters,
1171 North Street,
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Record Industry Association of America,
1 East 57th Street,
New York 22, New York.

Magnetic Recording Industry Association,
444 Madison Avenue—Room 1011
New York 22, New York.

The Use of Transistors in Airborne Audio Equipment*

VICTOR P. HOLEC†

***Summary*—The need for light weight, low power consumption, reliable audio amplifiers in airborne intercommunication systems led to the development of a new series of amplifiers to meet these requirements. Careful evaluation of the influence of temperature on operating points and circuit stability is an essential part of obtaining a satisfactory and reliable design.**

EARLY IN 1955 it became apparent that in order to meet the commercial airlines' demand for lighter weight and physically smaller equipment which would perform the same functions and to the same degree of reliability as in present units, careful consideration would have to be given to the use of transistors.

Since at that time (and yet today), silicon transistors capable of dissipating the required power levels were not readily available, and the equipment ambient operating requirements were specified to be in the region of 70°C., it was obvious that in order to use germanium junction transistors, a particularly detailed study of the possible circuitry would be required. As a later development, it was desired to provide a light weight unit capable of providing output power in the range of 2–3 watts for operation of cockpit speakers or for passenger compartment speakers for airborne wired music systems. The resulting unit is the 346A-1 airborne interphone and isolation amplifier shown in Fig. 1.

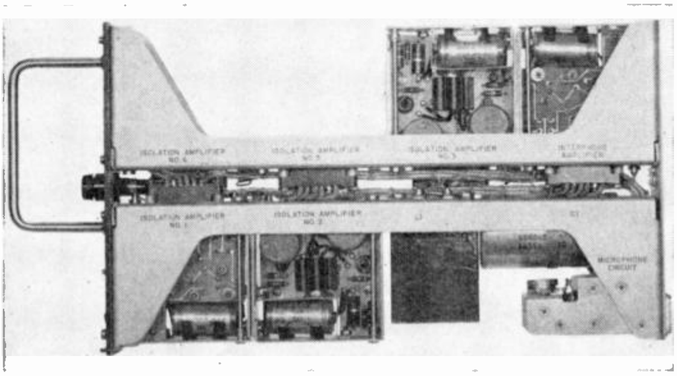

Fig. 1—Side view of 346A-1 airborne interphone and isolation amplifier.

It was found possible to utilize germanium transistors for these applications to provide completely reliable operation. A secondary design objective was that of producing a unit having the absolute minimum number of high cost components. An idea of the extent to which this goal was achieved may be seen by reference to Figs. 2 and 3.

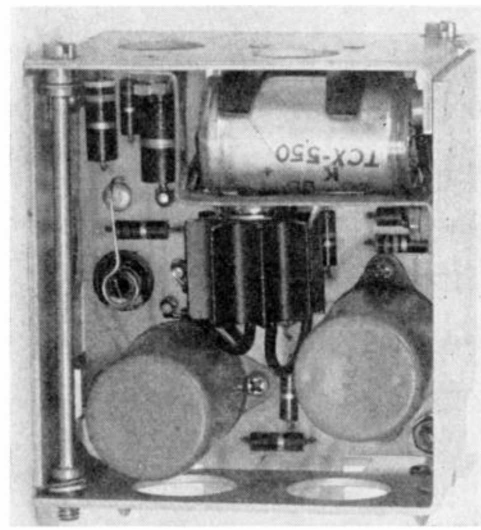

Fig. 2—150-mw amplifier, type 356C-1.

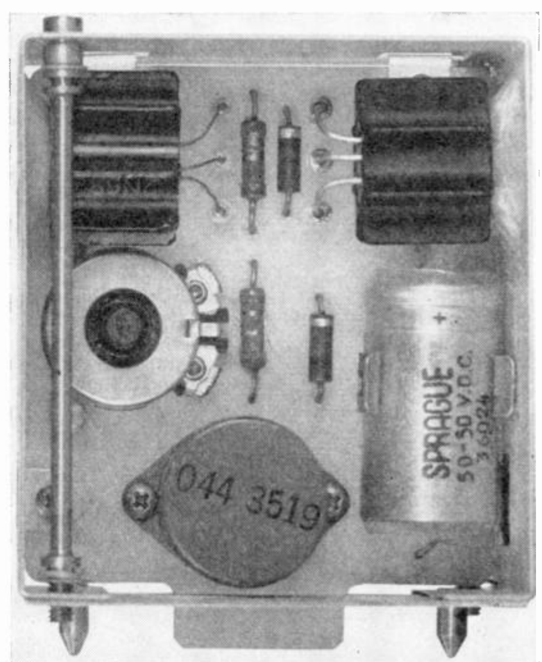

Fig. 3—2-watt amplifier, type 356D-1.

In the case of the 150-mw pre-amplifier shown in Fig. 2, only 9 resistors, 2 electrolytic condensers, and 2 low cost transistors, plus 2 simple transformers are required to provide all circuit functions. In the case

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of the 2-watt amplifiers, shown in Fig. 2, even fewer parts are required—amounting to 4 resistors, 1 potentiometer, 1 electrolytic condenser, 2 power transistors, and 1 interstage transformer. In both cases printed wiring boards containing an integral printed plug have been used to keep manufacturing costs to an absolute minimum.

One additional consideration worthy of note in connection with the use of transistors in airborne audio equipment is that of the power supply filtering required, since it is quite common to operate these units directly from the 27.5-volt aircraft supply. Due to the low audio level of the amplifiers employed, and the necessity for providing the various required bias voltages for the operating points from one voltage source, it became essential that adequate filtering be provided. In the case of the low-level preamplifiers shown in Fig. 1, this additional filter required was mounted on the frame of the $\frac{1}{4}$ atr size unit in which the pre-amplifiers were mounted.

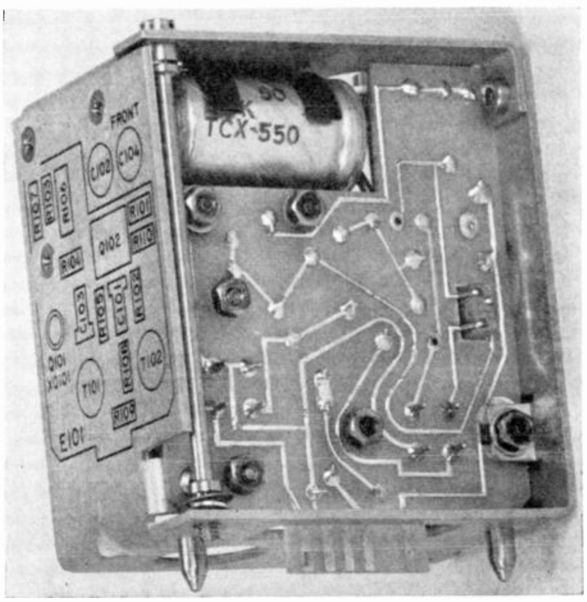

Fig. 4—Printed wiring and plug 150-mw amplifier.

Detailed Description of Units

The unit shown in Fig. 2 and Fig. 4, the 356C-1 isolation and interphone amplifier, is a unit having a nominal gain of 40 db operating from the low-level input to provide approximately 150-mw output power. Its normal use in an aircraft intercommunication system is to amplify a low-level signal (resulting from the mixing of the outputs of several communications receivers into a combining network) back to the original signal level so that it may be fed to the headphones of the various crew members. It can also be used as an interphone amplifier by connecting ahead of it simple RC circuitry to provide microphone current for a carbon microphone.

The 2-watt speaker amplifier, shown in Fig. 3, operates from a low impedance driving source which provides about 100-mw drive and has an output power of 2–3 watts at an impedance level in the range of 15–30 ohms. The amplifier may be directly connected to a speaker voice coil.

The 356C-1 amplifier is a 2-stage transistorized circuit employing *p-n-p* transistors. The input stage employs a GE type 2N45 transistor and the output stage employs a Sylvania 2N68 power transistor. The input impedances encountered in such amplifier stages are of a low value; therefore, the coupling capacitors between stages are necessarily of relatively high value. By reference to the circuit diagram, Fig. 5 (next page), it will be seen that the first coupling capacitor is 15 microfarads and the second is 20 microfarads. The amplifier input impedance is a nominal value of 100 ohms. By reason of its being such a low value, no input transformer is required. This alone represents both a cost and weight saving. The input voltage divider serves only to attenuate the input signals so that the specified nominal 40-db gain is obtained from the unit.

The input signal is then coupled to the base of the type 2N45 transistor. The base operates at a negative potential with respect to the emitter and receives audio feedback from the output circuit through R-110. Note again the usual high value of by-pass capacitor required to prevent degeneration at that point. The 2N45 collector is operated in a relatively high impedance circuit by employing a transformer as an interstage coupling device. It has been found more satisfactory to use a transformer than RC coupling techniques due to the wide impedance transformation required. The output of the interstage transformer is fed to the base of the type 2N68 output transistor.

The biasing potential for the base is obtained by voltage divider action and results in the divider being operated positive with respect to ground. The emitter is bypassed to ground (again by a 50-microfarad condenser) to reduce audio degeneration. The 2N68 collector, with the output transformer as its load, operates negative with respect to its base. Negative feedback, as mentioned earlier, in the form of a connection from the output winding back to the base of the input transistor, is employed to reduce the distortion.

The 356D-1 is a unit designed for providing a 2-watt output to drive a loudspeaker. This loudspeaker can be located either in the cabin as a substitute for the conventional headphones, or can be used alone or in conjunction with other units to provide a wired music type of installation for the passenger compartment. With reference to Fig. 6, it will be seen that the schematic is quite simple and straightforward.

A potentiometer is provided in the input circuit to adjust the over-all gain of the amplifier. The output stage is operated effectively as two individual separate Class B audio amplifiers, each one consisting of a transistor, a bias resistor, and a stabilizing resistor. In opera-

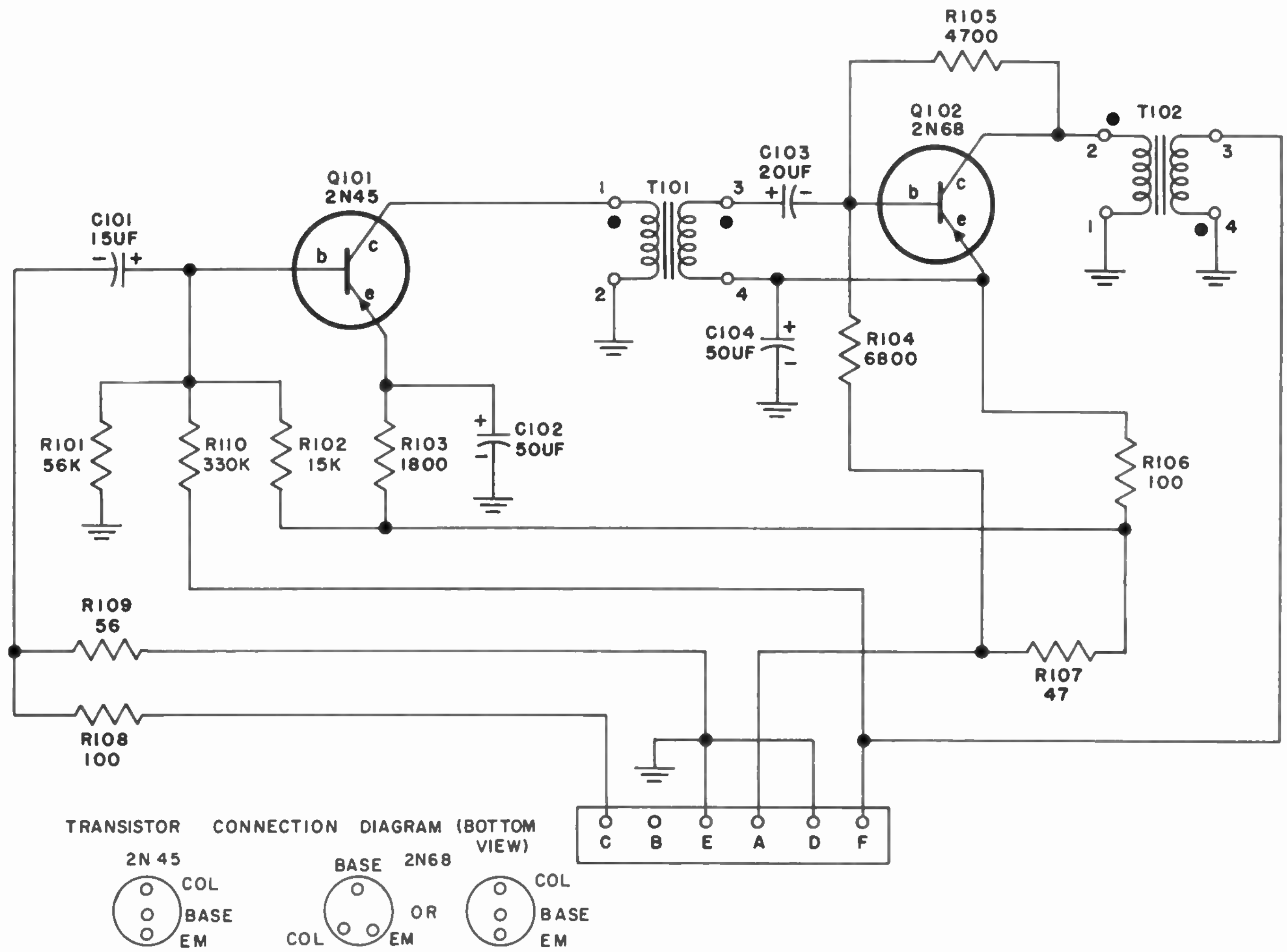

Fig. 5—2-watt amplifier schematic diagram.

tion each of the transistors is biased near "cut-off" by the biasing system. It can readily be seen that the two transistors are effectively connected in series for dc (in conjunction with their emitter stabilizing resistors across the 28-volt supply). The output capacitor is connected at the junction and is therefore charged to a nominal +14-volt value.

In a practical sense the transistor on one half of the audio cycle will try to add to this 14-volt charge, and on the other half will try to discharge the charged capacitor. One half of the audio cycle Q202 is cut off, but Q201 conducts between emitter and collector, and supplies the output capacitor with a pulse of positive current which flows through the load. On the other half of the audio cycle, the other transistor Q201 is cut off, but Q202 conducts, and "short circuits" the output capacitor C201, and attempts to discharge it through the load. The resulting audio signal is applied to the loudspeaker transformer or voice coil. Normally it can be directly connected to the voice coil since there is no practical difficulty in obtaining a voice coil with 15 to 30-ohm impedance.

Resistors R201 and R202 also provide negative feedback and stabilize the operating points of the transistors during temperature changes. Approximately 70 to 120-mw input is required to produce the rated 2-watt output. It was in the development of this circuit that considerable experience was gained in the course of attempting to develop a circuit and circuit values that would be stabilized over the wide temperature range required, while keeping in mind the previously mentioned design criteria of low cost and weight reduction.

It is felt that this amplifier represents a significant advance in the state of the art on the 2-watt level. It should be specifically noted that the transformer used is not an output transformer but an input transformer and is needed primarily because of the requirement for providing 2 out-of-phase signals to drive the 2 transistors.

Some comment might well be made on the mechanical considerations in packaging the 2 units. It can be seen by reference to Figs. 2, 3, and 4 that careful effort has been made to get as wide an area of the frame in contact with the 2N68 units to provide the best pos-

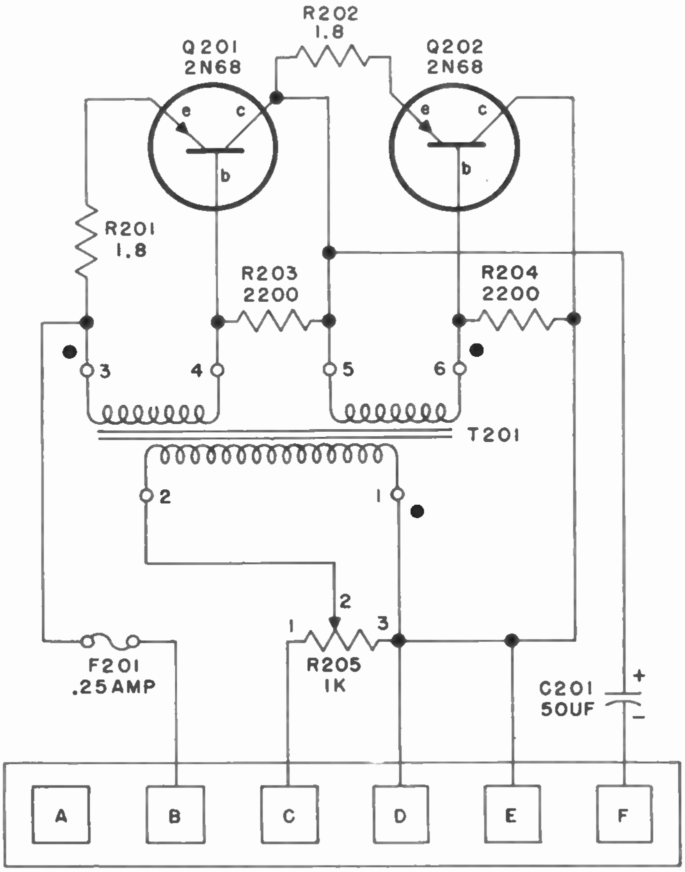

Fig. 6—150-mw amplifier schematic diagram.

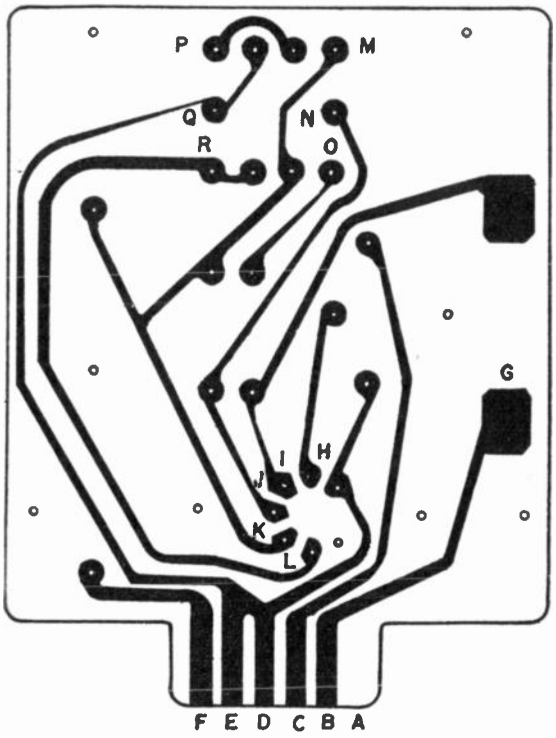

Fig. 7—Printed board detail.

sible thermal band to the heat sink. This is particularly important since the transistors operate within a few degrees of the point of the manufacturers' recommended maximum rating. Extensive laboratory tests, temperature cycling, and flight tests have well established, we believe, that this type of operation is satisfactory. In addition, in the case of the 2-watt amplifier, the operation is essentially that of a Class B unit in which very little current is drawn at zero signal conditions. This also helps to provide an additional safety margin for the power transistors.

Both units employ printed wiring, because it is the most convenient type of mounting as well as being a low cost manufacturing technique. The detail of the printed wiring board of the 2-watt amplifier is shown in Fig. 7. Use is made of the printed card to also provide the function of the connecting plug by simply running the printed wiring out to the edge of the board.

Additional protection for the contact portion of the board is provided by a rhodium finish on top of the basic copper foil. The board material itself is of the epoxy fiberglass and has proven to be exceptionally tough. While admittedly more expensive than the paper base phenolics, its use can be justified, because the additional cost is so little on a small board, and also because of the greatly reduced loss due to scrap and breakage in handling or use.

The construction employed in these units is that employed in the Collins modular type of design, each unit being built to fit a series of standard sizes and dimensions. In particular, these two units measure $3\times3\frac{1}{4}\times2$ inches, approximately, and are designed to fit in a $\frac{1}{4}$ atr frame which will accommodate up to 6 of these units plus a space for power supply connections and filtering.

Several of these units have been successfully flight tested by commercial airlines for periods of several months with no significant change in performance. That they represent a definite improvement in performance over previous units is borne out by the fact that it is no longer necessary to supply the power to these units from the "radio bus," which is energized only when the aircraft is being prepared for flight or actually in flight, and can be in constant operation for all ground checks and maintenance operations. Further, transistors have, in effect, an unlimited life compared to the greatly deteriorated life of vacuum tubes used in previous units when the units were operated on standby for long periods or were subjected to numerous on-off cycles.

Engineering Consideration of Ceramic Phonograph Pickups*

BENJAMIN B. BAUER†

Summary—**Performance of ceramic pickups is compared to Rochelle salt and magnetic pickups. Whereas voltage-temperature characteristics of barium titanate ceramics and Rochelle salt crystals are relatively constant over a range of temperatures, ceramics exhibit a more stable capacity vs temperature characteristic than does Rochelle salt, and are not subject to damage due to arid and tropical conditions.**

The performance of piezoelectric pickups and magnetic pickups is analyzed with respect to the standard recording characteristic. It is concluded that crystal pickups are outstanding when high output is the principal requirement, where quality requirements are moderate, and climatic conditions are benign. Ceramic pickups are the logical choice when quality and economy are both important or where climatic conditions are severe or when magnetic induction is a problem. Current magnetic or dynamic pickups are indicated when the available amplifying equipment, or the present-day public opinion are the principal factors.

Introduction

PRIOR TO THE introduction of barium titanate ceramics a few years ago two types of phonograph pickups were generally available, namely Rochelle salt crystal and magnetic (or dynamic). The crystal has a high output and requires relatively little compensation to match the recording characteristic of records, but it is adversely affected by heat and by dry or humid weather. The magnetic has low output and requires considerable compensation to match the recording characteristic of records, but it is relatively free of atmospheric effects. This stability of the magnetic pickup has made it a likely candidate for the field of high fidelity, whereas the Rochelle salt crystal was destined to serve those applications where low cost is the principal aim. On the other hand ceramic pickups are free of the adverse atmospheric limitations of crystal pickups and retain a considerable advantage over magnetic pickups with respect to sensitivity, response-frequency characteristic, and freedom from magnetic induction; they stand out therefore, when both performance and cost are important.

The first part of this paper is devoted to the description of the design features of one series of ceramic pickups. The second part deals with the environmental characteristic of ceramics. The last part offers some recommended circuits for obtaining faithful record reproduction with ceramic pickups.

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† Shure Bros., Inc., Evanston, Ill.

Ceramic Pickups

As shown in Fig. 1, a typical ceramic element consists of two thin wafers of barium titanate with a metal vane in between. The barium titanate wafers are soldered to the metal vane to form a sandwich and provided with electrodes. The assembly is subjected to a strong dc polarizing voltage of about 100 v/mil applied in such manner that the wafers are oppositely polarized. Upon removal of this dc voltage, it is found that the element exhibits piezoelectric properties. During operation a bending stress is applied as in the right-hand view, one wafer being subjected to tensile stress, and the other to compressive stress; since they are oppositely polarized the potential generated across the wafers is additive. One of the electrodes will have acquired a positive potential and the other a negative potential.

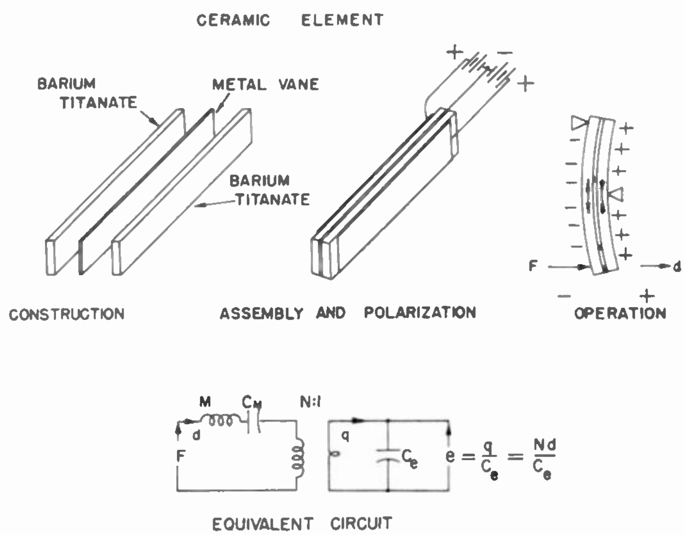

Fig. 1—Construction, polarization, and equivalent circuit of a ceramic element.

The equivalent circuit of the ceramic element is shown in Fig. 1 with the mechanical end at left and the electrical end at right. On the mechanical side, the element has an effective mass M in series with a compliance C_m. The ideal N:1 transformer represents the electromechanical coupling. On the electrical side the unit defines a capacitor C_e. Mechanical displacement d causes a corresponding flow of charge $q = Nd$ which charges the capacitor C_e with a voltage $e = q/C_e = Nd/C_e$. Thus displacement d corresponds to a given generated voltage e across the terminals of the pickup. Hence, the ceramic element is a displacement responsive device. By contrast, it will be remembered, a magnetic transducer is a velocity responsive device.

The typical ceramic element used in pickups is about ½ inch long, 1/16 inch wide and has a total thickness of 0.025 inch. The effective mass M is approximately 20 milligrams. While relatively simple pickups have been made by attaching a needle tip to one end of these elements, it is evident that such pickups will impose an unduly high mass load upon the groove. The needle-point mass in a pickup should not exceed a very few milligrams. To reduce the effective mass we must use a mechanical transformer. A lever is just such a transformer. Fig. 2 shows a system of two levers used in the Shure twin-lever pickup cartridges. The ceramic element is clamped at one end and is connected at the other end to a light metal coupler, which forms the first lever.

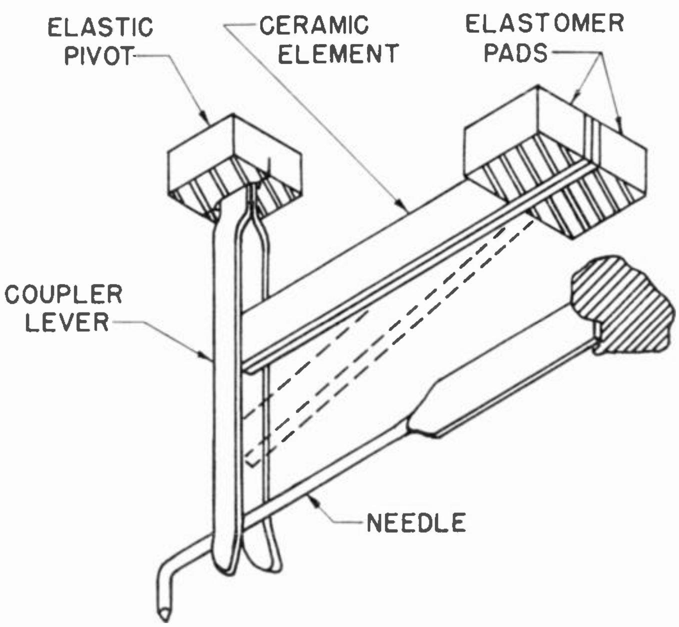

Fig. 2—A system of two levers used in the Shure twin lever pickup cartridges.

At the lower end the coupler has a saddle capable of receiving the needle or stylus. The needle forms the second lever. By choosing the position of the ceramic element relative to the coupler, and by altering the length of the needle, effective leverage ratio may be varied from 2:1 to 4:1 resulting in mechanical impedance transformation of 4–16:1 respectively. This choice of leverage ratio makes the difference between ceramic pickups which produce a high output with moderate performance and those which produce a moderate output with high grade performance.

The internal assembly of the twin-lever pickup is shown in Fig. 3. In the lower portion are the two case halves of the pickup, each provided with two support pads for the ceramic element. The element together with the coupler is positioned in the left case half. Contact is made with the electrodes by means of metal tabs extending over the rear support pads. The upper view shows the two case halves assembled with the auxiliary hardware. The saddle end of the coupler can be seen in this view.

Two needle assemblies are shown in the center. The left-hand side assembly shows that the needles are flattened to form a pivot. The right-hand assembly shows a block of semiviscous material positioned around the needles which provides the required damping. The needle assembly is held by a thumbscrew and may be changed without the use of tools. As shown in the upper right-hand side of Fig. 3, one stylus at a time engages the coupler. The dimensions of the styli can be visualized by comparing them with the 1-inch straight pin.

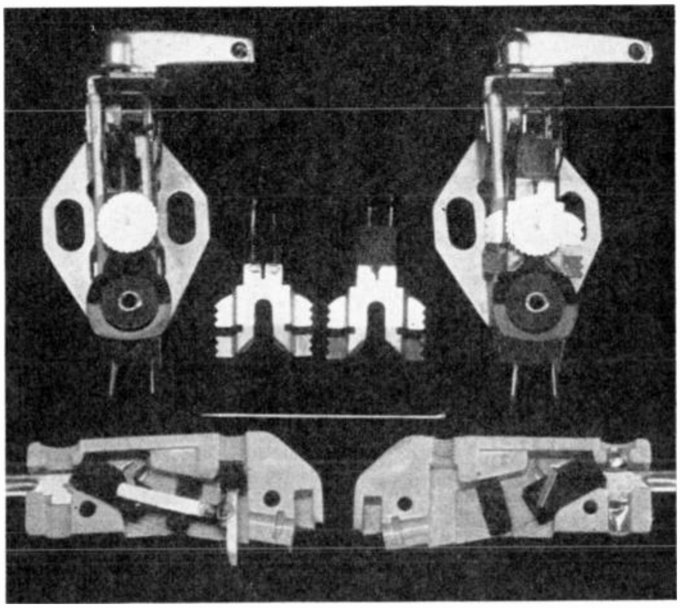

Fig. 3—Assembly details of PC6 phonograph pickup cartridge.

To shift from one needle to the other the arm at front of the pickup is flipped 180°. This lifts the transport mechanism and causes the first needle to be disengaged from the saddle, moves the second needle in position and lowers it into the place formerly occupied by the first.

The perspective view of this pickup cartridge is shown in Fig. 4. A variety of mounting ears and a choice of front or side flipper may be provided to suit individual requirements.

Another twin-lever cartridge, namely the PC20, designed in cooperation with Dr. Goldmark of CBS Laboratories with a mobile phonograph player in mind, is shown in Fig. 5. The construction of this cartridge is similar to that previously described, except that a single 0.0003-inch radius needle is used.

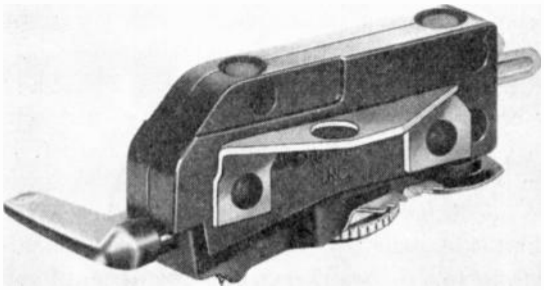

Fig. 4—PC6 pickup cartridge for home phonographs.

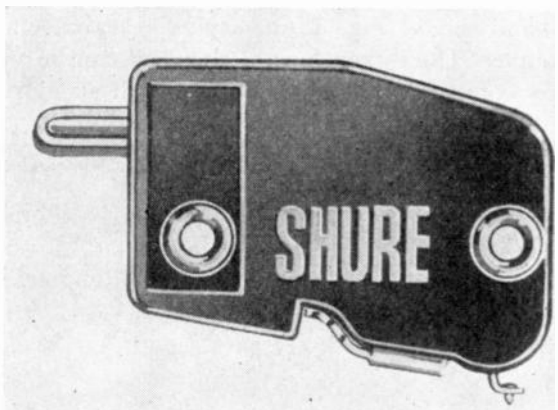

Fig. 5—PC20 pickup cartridge for mobile phonograph player.

Environmental Performance

Because the output of ceramic pickups is not much different from that of certain crystal pickups, some people think that a ceramic element is merely a different kind of a Rochelle salt crystal; to correct this impression, one need only to compare the environmental properties of ceramic elements and of Rochelle salt. In Fig. 6, at the bottom is shown the output vs temperature characteristics of a barium titanate ceramic element and that of a Rochelle salt element when driven by a constant force. It is evident that throughout the range of room temperatures (*i.e.*, 60°–100°F) both elements have constant output voltage-temperature characteristics. The ceramic elements, however, will operate at a temperature of 140°F and exhibit no significant permanent deterioration when subjected to 175°F, whereas the Rochelle salt elements begin to undergo certain irreversible changes above 120°–125°F and melt at about 135°F. In an automobile parked in the hot sun temperatures of 140°–150° may be attained. Hence, in the mobile player described by Dr. Goldmark ceramic elements were found to be satisfactory whereas Rochelle salt elements could not be used.

Another significant property is the capacity-temperature characteristic of ceramic and Rochelle salt elements which is shown at the top of Fig. 6. Here, it is seen that the capacity of the ceramic element used in pickups is relatively constant over the usual range of room temperature (*i.e.*, 60°–100°F). On the other hand, a Rochelle salt element has very severe variations in capacitance right at room temperatures. Therefore, the performance of a ceramic pickup will remain relatively constant whereas that of a Rochelle salt pickup is apt to vary significantly, even at room temperature, depending upon termination impedance.

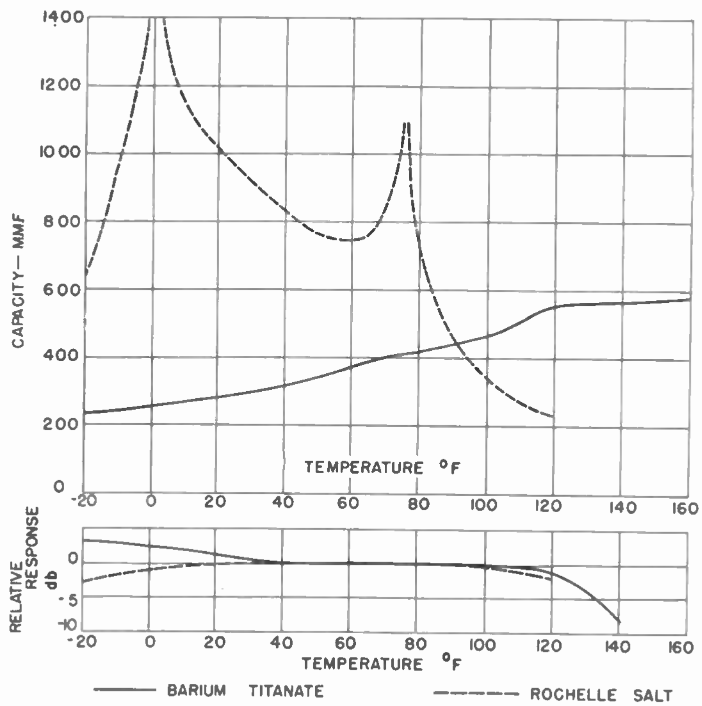

Fig. 6—Capacity and sensitivity vs temperature characteristics of barium titanate and Rochelle salt elements.

Another consideration is the performance under conditions of high or low humidity. Ceramic pickups suffer no damage as do the Rochelle salt crystals when subjected to humidity and dryness. Because of this environmental stability ceramic pickups became a serious contender with the magnetic pickups in the field of high fidelity.

Connection of Ceramic Pickups

Connection of ceramic pickups to obtain optimum performance must be done in the light of the present day recording characteristic. In Fig. 7 the rising solid line represents the recording characteristic of the Record Industry Association of America in terms of velocity. If a theoretical magnetic or a dynamic pickup is used with a *flat* amplifier playing records recorded with the standard characteristic then the frequency response will rise in accordance with the velocity curve and the system will sound thin and tinny. To obtain an over-all *flat* frequency response, the bass must be boosted approximately $18\frac{1}{2}$ db at 30 c and the highs must be dropped $17\frac{1}{2}$ db at 15,000 c, constituting a total equalization of 36 db from 30–15,000 cps.

To visualize the performance of a ceramic pickup, we must replot the IRAA recording characteristic in terms of amplitude and refer it to 0 db at 1000 cps. To do this, we divide the ordinates of the velocity curve by the frequency in kc and obtain the downwardly-slanting line, labeled *amplitude*. This, then, will be the frequency response of a theoretical ceramic pickup, playing records recorded with the standard characteristic. The compensation required for *flat* frequency response is less than that needed with a magnetic or dynamic pickup being as follows: A reduction of 12 db at 30 cycles and a rise of 6 db between 4 and 15 kc; or a total equalization of 18 db, which is half of that required for the magnetic pickup.

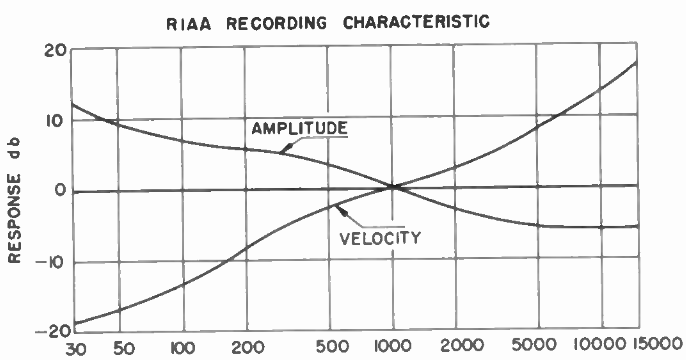

Fig. 7—Radio Industry Association of America recording characteristic in terms of amplitude and velocity.

Here again, nature appears to favor the ceramic pickup. You will remember (Fig. 6) that the ceramic has a capacitance of about 400 $\mu\mu$f. The shielded lead from pickup cartridge to amplifier may add another 200–300 $\mu\mu$f to the ceramic capacity, giving a total capacity of around 650 $\mu\mu$f. The reactance of 650 $\mu\mu$f at 30 cps is 8 megohms; hence if the pickup is loaded with a volume control of about 2–3 megohms, the low frequency end will drop approximately the required 12 db. Likewise, by suitable choice of supporting pads for the ceramic element and needle damping material, it is possible to realize the 6 db rise required in the upper frequency range. When these conditions are suitably fulfilled, the ceramic pickup requires practically no additional compensation to reproduce the standard recording characteristic in adequate manner.

To verify the above statement by using a test record it is necessary first to derive the frequency response on a particular record which corresponds to the desired playback characteristic. This is done by subtracting the recording characteristic from the velocity-frequency calibration of the test record. Referring to Fig. 8, the

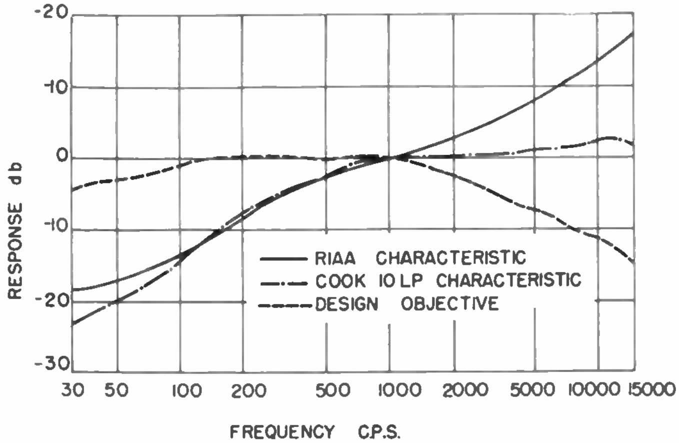

Fig. 8—Derivation of required design objective curve for RIAA response on Cook 10LP record.

dot-dash line is the velocity-frequency calibration of the Cook 10LP test record. The solid line is the RIAA recording characteristic. We subtract the RIAA characteristic point by point from the record calibration, obtaining the dash line which is the desired design objective playback characteristic. In Fig. 9 is shown the

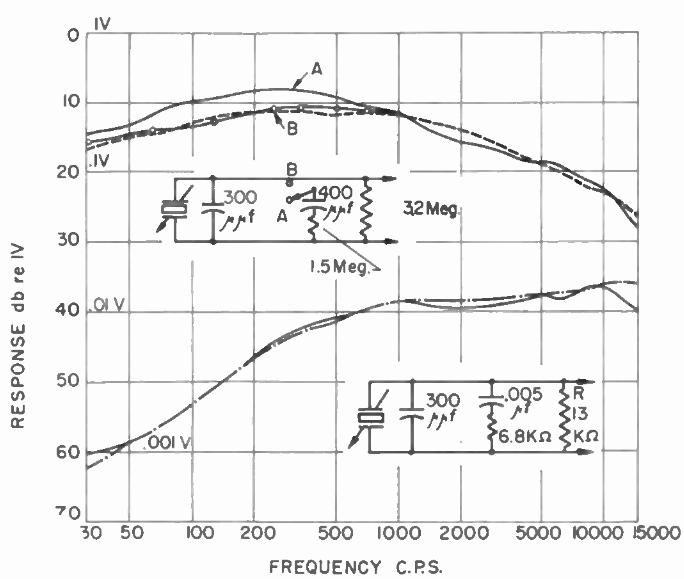

Fig. 9—Connection of PC6 pickup for high and low impedance circuits.

performance of one of the pickups previously described, namely the PC6 ceramic pickup, played on Cook 10LP record. The design objective curve is shown in dotted line. The solid line shows that when this pickup is connected by a 300 $\mu\mu$f cable to a 3-megohm input resistor (which can be a volume control), the design objective curve is well matched. A somewhat better match at low frequency is obtained if a series combination of 400 $\mu\mu$f condenser and 1.5 megohm resistor is connected across the pickup, the resultant being shown by the circle-line. To endow this pickup with a response similar to a magnetic pickup we must differentiate its output with respect to frequency; this is done by connecting a 13,000 ohms resistor across the terminals of the pickup. A still better match is obtained by adding a series combination of 0.005 $\mu\mu$f condenser and 6.8 k-ohm resistor across the terminals of the pickup. The response of the pickup and network is shown in solid line below, and compared to the velocity-frequency characteristic of Cook 10LP record shown in dot-dash line. This response is undistinguishable from that of the better grade magnetic or dynamic pickups.

At this point, the economic advantage of a ceramic pickup becomes quite evident, since a magnetic pickup requires 45 db more gain at 30 cps and 25 db more gain at 1000 cps to equal the output of the ceramic connected to a high-impedance circuit. This comparison does not take cognizance of the possibility of using transformers, but rather it is a comparison based on a practical circuit design situation.

Conclusion

I have attempted to describe the ceramic pickup in its proper perspective relative to the Rochelle salt crystal pickup and the magnetic pickup. One must be careful of generalizations, but the following summary seems to fairly represent the current state of the art: Crystal pickups are outstanding when high output is the principal requirement, where quality requirements are moderate, and climatic conditions are benign. Ceramic pickups are the logical choice when quality and economy are both important or where climatic conditions are severe or when magnetic induction is a problem. Current magnetic or dynamic pickups are indicated when the available amplifying equipment, or the present day public opinion are the principal factors. Within this generalization there will be many exceptions based upon the specific equipment under consideration.

There is little doubt, however, that ceramic pickups are helping the audio technologist to lower the cost of high fidelity so that it can become available to a wider circle of consumers. In so doing he is fulfilling one of the main virtues of his calling.

Acknowledgment

I am grateful to various members of the engineering division of Shure Brothers, Inc., for assembling the data presented in this paper. The temperature characteristics of piezoelectric elements came from the notebooks of J. Medill. The PC6 pickup was perfected by L. Gunter assisted by E. Seeler and the networks of Fig. 9 were developed by R. Anderson.

Stereo Reverberation*

R. VERMEULEN†

Summary—**Investigating the reasons why reproduced music gives an impression different from that which a listener receives during a concert, it was found that the distribution of the sound over the room is essential. Although stereophonic reproduction can give a sufficiently accurate imitation of an orchestra, it is necessary to imitate also the wall reflections of the concert hall, in order that the reproduction may be musically satisfactory. This can be done by means of several loudspeakers, distributed over the listening room, to which the signal is fed with different time-lags. The diffused character of the artificial reverberation thus obtained seems to be even more important than the reverberation time. Likewise, when a live orchestra is playing in an acoustically unsatisfactory hall (*e.g.*, a theater), the diffuseness of the sound field and the reverberation time may both be improved by picking up the music by means of a directional microphone and repeating it through loudspeakers with different retardations. The audience does not experience the improvement consciously and ascribe it to the orchestra playing better. The performers, however, are aware of the change in the acoustics as making the hall more playable.**

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† Philips Res. Labs., Eindhoven, Netherlands.

Introduction

The acoustic properties of a hall are determined by the behavior of the sound waves, in particular those which are reflected from the walls. The importance of the reflected sound can be well understood if one has listened to an open-air speech made without an amplifier installation, or if one imagines an open-air performance by a symphony orchestra without this aid.

A simple calculation will show that in a hall at even quite a small distance from the orchestra, the direct sound can sometimes be weaker than the reflected sound. At a distance r from a sound source of power P, the energy density of the direct sound is equal to $P/(4\pi r^2 c)$, where c is the velocity of sound. In a hall with a volume V and reverberation time T (defined as the time in which the sound intensity decreases by 60 db, after the source has ceased radiating), the energy density of the indirect sound[1] is $PT/(13.8\ V)$. These two energy densities are equal at a distance $r_0=(1.1V/cT)^{1/2}$. Values of r_0 corresponding to various practical values of V and T are

$V=100$	1000	10,000 m^3
$T=0.7$	1.0	1.5 s
$r_0=0.7$	1.8	4.5 m

At distances greater than r_0 the indirect sound predominates. It is seen that this can occur at distances of only a few meters.

The qualities required of a hall for speech and for music are quite different. In a theater, the intelligibility is of primary importance. If speech is to be clearly understood, the reflected sound must reach the audience with so little delay that it reinforces the direct sound but does not overlap the sounds which follow. For the latter, the persistence of the preceding sound must be regarded as *background noise*, which adversely affects intelligibility. As a rough guide, one can say that all sound which reaches the audience within 50 milliseconds can be regarded as useful sound.[2] Erwin Meyer has formulated the idea of *clearness* or *definition*,[3] which he defines as follows:

$$\frac{\int_0^{50\text{ msec}} p^2(t)dt}{\int_0^{t_1} p^2(t)dt}$$

where p is the sound pressure, t is the time (measured from the moment at which the source is silenced), and t_1 is a time much greater than 50 msec.

For a concert hall, on the other hand, the first requirement is not intelligibility, but a fine, full tone. Here it is much more difficult to specify the requirements. For speech, the reverberation must be accepted as an inevitable, disturbing accompaniment to the useful sound, because it simply is not possible to silence the sound suddenly after 50 msec. For music we know that the reverberation time not only may be, but must be, longer. The optimum value is clearly dependent on the nature of the music. It is often seen that a composer has consciously taken into account the acoustics of the space (church, concert hall, room) where he wanted his music to be played. The inclination to sing in the bathroom can probably be largely attributed to the long reverberation time of this acoustically *hard* room.

It is becoming increasingly clear that the reverberation time is not the only property governing the suitability of a hall for musical performances. One might even conjecture that here, too, reverberation is merely an inevitable subsidiary effect. Just as important, or perhaps even more so, is the *diffuseness* of the sound (and possibly also the nature of the fluctuations of the reverberation).

To study these phenomena more precisely, we attempted to produce an artificial diffuse reverberation in the laboratory by means of distributed loudspeakers which repeated the music played, with controllable intensity and lag. This experiment appeared to improve the acoustics to such an extent that we ventured to take the bold step of using this artificial reverberation to make a theater suitable for concerts. We propose that this artificial diffuse reverberation be called *stereo reverberation*.

Our first installation was in the Philips Theatre at Eindhoven, whose acoustical properties as a theater were very satisfactory as a result of rebuilding in 1935, but which left much to be desired as a concert hall. In addition to this theater, a hall known as the "Gebouw voor Kunsten en Wetenschappen" (Arts and Sciences Hall) in The Hague, is now fitted with a permanent installation for stereo reverberation.[4]

Principle of the Installation for Stereo Reverberation

The Delay Wheel

The principle of the stereo reverberation installation may be explained with the help of Fig. 1. Controllable time lags are obtained by means of magnetic recording and playback. Magnetic material such as is used for magnetic tape is coated on the rim of a wheel (delay wheel).[5] The music is recorded on this material via a microphone and a recording head. A number of playback heads—in the final apparatus four, in experimental types six (Fig. 2) or more (Fig. 3)—are mounted around the circumference of the wheel and connected via separate channels to loudspeakers. These are installed in various places in the hall: on the ceiling, along the balustrade of the balcony, in a lighting cornice, in *dead* corners under the balcony, etc. Between the last play-back head and the recording head is an erasing head, which ensures that the magnetic layer is blank on returning to the recording head.

[1] See, for example, A. Th. van Urk, "Auditorium acoustics and reverberation," *Philips Tech. Rev.*, vol. 3, pp. 65–73; March, 1938.
[2] *Ibid.*, pp 72–73.
[3] E. Meyer, "Definition and diffusion in rooms," *J. Acoust. Soc. Amer.*. vol. 26, pp. 630–636; September, 1954.

[4] Demonstrations of stereo reverberation have also been given at the first ICA Congress on Electro-Acoustics, (*Acustica*, vol. 4, pp. 301; February, 1954) at Gravesano in Switzerland (at the invitation of the conductor, W. Scherchen; see his book "Musik, Raumgestaltung und Electroakustik," Arsviva Verlag, Mainz, 1955) and at the 3rd "Tonmeistertagung" of the Nordwestdeutsche Musikakademie Detmold (see D. Kleis, *Elektron. Rdsch.*, vol. 9, pp. 64–68; February, 1955). Similar tests, but done in the open air, are described by H. S. Knowles, *Acustica*, vol. 4, pp. 80–82; 1954.
[5] Others have also constructed a similar equipment, but have not created diffuse reverberation with it; see H. Schiesser, "Einrichtungen zur Erzeugung kunstlichen Nachhalls," *Funk und Ton*, vol. 8, pp. 361–368; July, 1954 and P. Axon and co-workers, "Artificial reverberation," *J. IEE*, vol. 1, pp. 368–371; June, 1955.

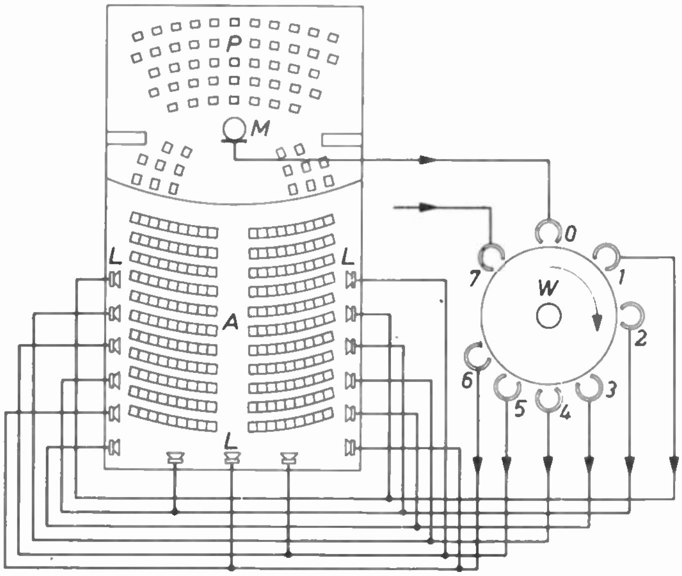

Fig. 1—Installation for simulating indirect sound with various time lags. A auditorium. P platform. M microphone. W delay wheel, coated on the edge with magnetic material suitable for magnetic sound recording. O recording head, 1 . . . 6 play-back heads. 7 erasing head. L loudspeakers.

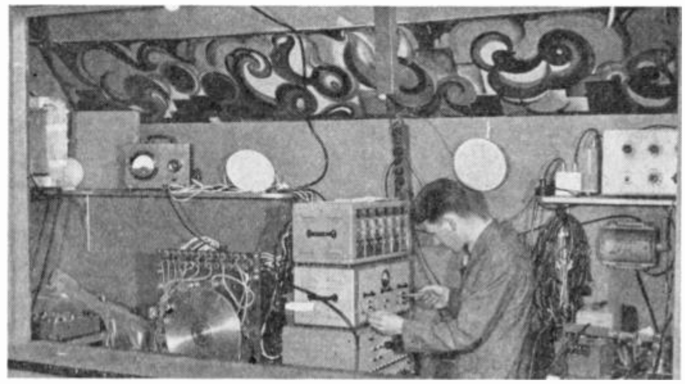

Fig. 3—Stereo reverberation installation in the "Gebouw voor Kunsten en Wetenschappen" (Arts and Sciences Hall), The Hague, in experimental form. The delay wheel was then fitted with ten play-back heads, of which six could operate simultaneously. It was possible to switch over rapidly from one set of play-back heads to another.

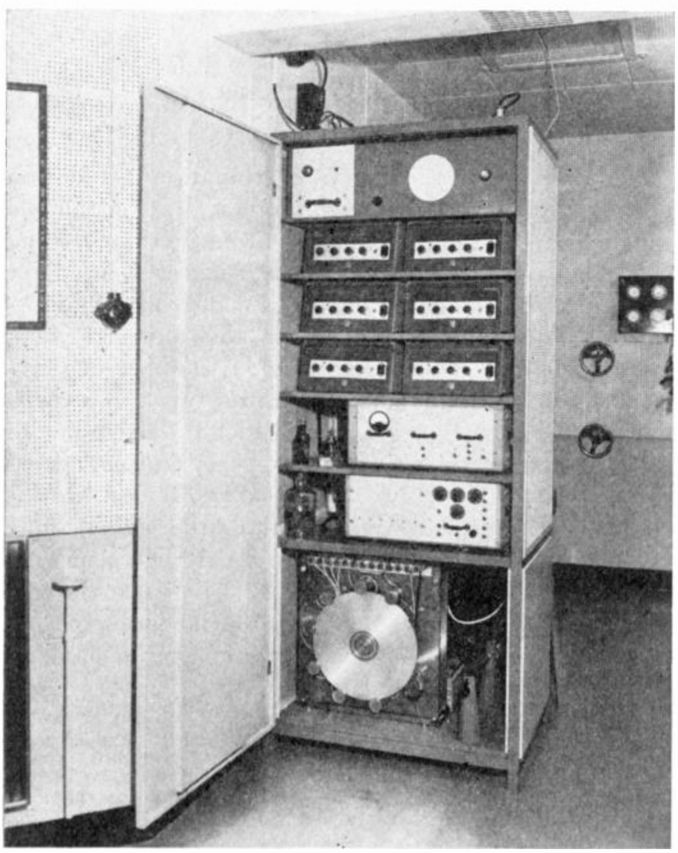

Fig. 2—Stereo reverberation installation in the Philips Theater at Eindhoven. At the bottom of the cabinet is the delay wheel (see Fig. 1); above it are the amplifiers.

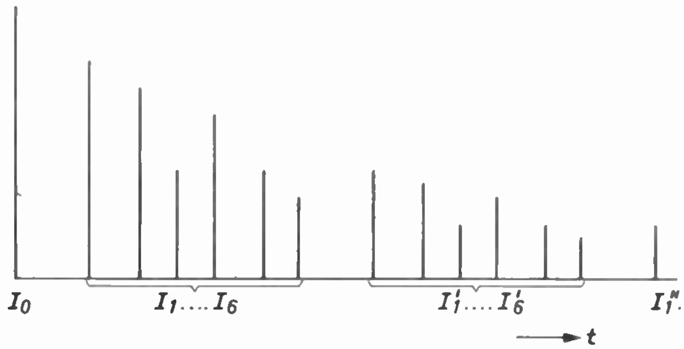

Fig. 4—If a report (pulse I_0) is produced in a hall with stereo reverberation, the six loudspeakers deliver during one revolution of the delay wheel the six reports $I_1 \cdots I_6$. If electrical feedback is applied from the sixth play-back head to the recording head, the loudspeakers give a second series of six reports ($I_1' \cdots I_6'$), a third, and so on. (Acoustic feedback is here neglected.)

Let us consider the case of a sharp report produced in the hall (pulse I_0, Fig. 4). After a certain transit time this reaches the microphone and then the artificial indirect sound begins. The latter, if we are using six play-back heads, consists in the first place of six successive reports from the loudspeakers ($I_1 \cdots I_6$). The intensity of each of these reports can be adjusted at will by the gain controls; the time intervals are also under control, by the spacing of the heads round the wheel and by its speed of revolution.

For three reasons, however, the listener hears more than these seven (1+6) reports: first, because to each play-back head, several loudspeakers are connected, which are dispersed throughout the hall and are at different distances from the listener, so that the transit times are different; second, because the walls reflect both the original sound and those coming from the loudspeakers; and third, because the report from each loudspeaker is picked up by the microphone, once more recorded on the wheel and, though attenuated, reproduced sixfold.

All these effects contribute to the fact that the original report is followed by a large number of others, so that the reverberation time can achieve considerable values. The third effect mentioned above—the feedback from the loudspeakers to the microphone—even involves the danger that one particular note continues to sound too long and in the extreme case howls back continuously; therefore this effect must be curtailed as much as possible (we shall return to this point presently). This acoustic feedback can be advantageously replaced by electrical feedback, in which a chosen fraction of the signal from the last play-back head is fed back to the recording head. This records the signal afresh: in our example, a second series of six reports then ensues ($I_1' \cdots I_6'$ in Fig. 4); this series is once more recorded and a third series follows, and so on. By adjusting the fraction of the output signal which is fed back, each series can be attenuated to any given degree, so that the reverberation time can be chosen at will.

Calculation of the Effect of the Stereo Reverberation on the Acoustic Properties of the Hall

The following calculation shows, for a simple case, the effect which stereo reverberation has on the acoustic properties of the hall. Suppose that the energy density in the hall is $E(t)$. Then $-VdE/dt$ is the rate at which acoustic energy in the hall diminishes. In a hall without stereo reverberation, this must be equal to the power absorbed by the walls,[6] which is proportional to E, say equal to αE. Stereo reverberation supplies extra power proportional to the energy density at some time τ ago: $\beta E(t-\tau)$. We can thus construct the equation

$$-V\frac{dE}{dt} = \alpha E(t) - \beta E(t-\tau).$$

A solution to this is

$$E = E_0 \exp(-mt)$$

where

$$mV = \alpha - \beta \exp(m\tau).$$

The quantity m is inversely proportional to the reverberation time T, and equal to $13.8/T$. For small values of $m\tau$, an approximate value of $\exp(m\tau)$ is given by $1+m\tau$. Thus we have

$$m \approx \frac{\alpha - \beta}{V + \beta\tau}.$$

From this it is seen that increasing the strength of the stereo reverberation (β) has the same effect as decreasing the absorption (α) or increasing the volume (V) of the hall. An increased lag τ also gives the effect of a hall of larger volume; we shall return later to this point.

The Microphones

The above calculation was based on the assumption that one could speak of *the* energy density in the hall. In many cases the total energy (sum of potential and kinetic energy) is indeed fairly evenly distributed throughout the hall. With standing waves, however, as is well known, the potential and kinetic energy alternate with each other, and this means that the sound pressure at constant frequency changes sharply from place to place, and, conversely, that at a particular place the sound pressure is very dependent on the frequency. The microphone which picks up the signal which is fed back to the hall via the delay wheel and the loudspeakers, responds only to the pressure at the spot. The factor β in our calculation therefore varies sharply with the frequency, and the reverberation time, which is inversely proportional to $\alpha-\beta$, varies also and to a much greater degree. If β as a function of the frequency shows a peak, then increasing the amplification will cause the note at which the peak occurs to go on sounding for a long time, while for most other frequencies, the reverberation time is not yet lengthened appreciably. With even greater amplification, the note fails to decay at all (howl-back: $\alpha-\beta$ has become negative).

The obvious solution is to try to suppress the peak by a filter in the microphone channel. However, there are so many peaks in the frequency characteristic of a hall, and these peaks are so sharp, that the suppression of all of them would be a hopeless task, particularly since the peaks are modified by all changes made in the hall or on the platform. It is worthwhile, nevertheless, to attenuate those frequency ranges in which the highest peaks occur in such a way that increase of amplification causes a number of widely separated frequencies to howl back at the same time.

In this connection it can be argued that electrical feedback via the delay wheel is a better means of obtaining a long reverberation time than acoustic feedback via the hall.

In a system with delayed feedback, the feedback signal and the input signal exhibit, in general, a relative phase difference which is proportional to the product of frequency and time lag. If there are a great number of feedback channels (as there invariably are in practice, with acoustical feedback), and if we suppose that they all make equal contributions to the input signal, then in general the contributions will show fairly random phase differences, so that the average resultant increases as the root of the number of feedback channels. There will be, however, one or more frequencies for which all the contributions are nearly in phase with each other. For these frequencies the resultant will be a maximum and equal to the algebraic sum of the contributions, and thus proportional to the number of feedback channels. This reasoning makes it clear that the ratio of the maximum value of the resultant (which occurs at a particular frequency and limits the maximum amplification) to the average value (at other frequencies) increases with the root of the number of feedback channels. In the case of a microphone and loudspeakers in the same hall, this number is very large, and it is to be expected that a situation can easily arise in which one note continues to sound for a long time. With feed-

[6] See, for example, the article referred to in footnote 1, where it is shown that the proportionality factor α is equal to $1/4\alpha Ac$, where α is the average absorption coefficient and A the area of the walls.

back via the delay wheel, we are using only one feedback channel (from the last playback head to the recording head) and the above-mentioned danger is thus much less. It would be possible to introduce more feedback channels, *e.g.*, from one or more of the preceding playback heads to the recording head. Experiments have shown that this is undesirable, and this is partly explained by the above considerations.

Instead of a microphone which responds to the average sound density in the hall, the other extreme would be one which responds exclusively to sound coming directly from the source and is insensitive to sound from the hall.[7] The danger of notes howling back would then be completely averted. This situation can be approximated closely enough by using a microphone having a sharp directional characteristic. An arrangement for achieving this, a so-called *line microphone*, consists of a group of ten condenser microphones mounted at equal intervals along a rod rather more than a meter in length (Fig. 5). By suitably positioning this line microphone above the orchestra, we can ensure that the loudspeaker signal at the microphone, although still making an important contribution, no longer dominates. If it is not possible to cover the orchestra adequately with one line microphone, two may be used (Fig. 6).

Fig. 5—The stage in the "Gebouw voor Kunsten en Wetenschappen" in The Hague. The line microphone (rod with ten condenser microphones) is indicated by two arrows. (Other microphones visible in the photograph were for broadcasting and had nothing to do with stereo reverberation.)

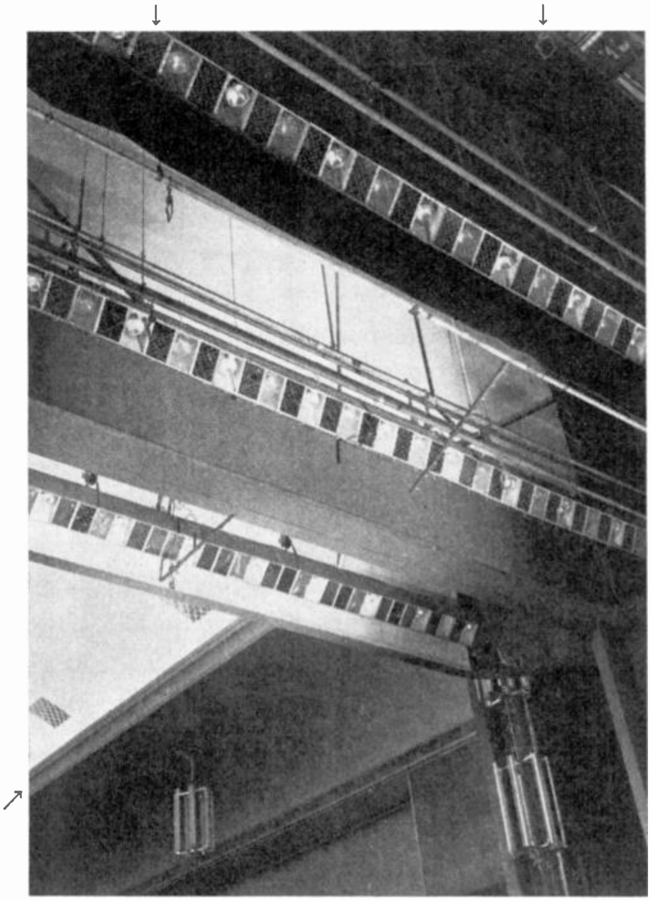

Fig. 6—In the Philips Theater at Eindhoven two line microphones are used. The loudspeakers are mounted in a concealed position behind the lighting cornice indicated by the oblique arrow at the left. (compare Fig. 9.)

The use of one or more extra microphones can be desirable in order to strengthen weak instruments in the orchestra. Thus, for example, in the Philips Theater at Eindhoven the organ was rather weak in comparison with the orchestra and choir in the annual performance of Bach's St. Matthew Passion; it appeared to be an improvement when this instrument was boosted by a microphone of its own (Fig. 7). With such cases in mind, the stereo reverberation system was provided with several microphone channels with mixing facilities. These should, however, be used with the utmost restraint, for the sound engineer must never encroach upon the conductor's prerogative for the balance of the instruments in the orchestra.

The Loudspeakers

As already suggested, the diffuseness of the sound is perhaps even more important than the lengthening of the reverberation time. Diffuseness can be obtained by

[7] This occurs, for example, in the case of music reproduced in a hall, when it is (or was) actually performed elsewhere. We return to this point again at the conclusion.

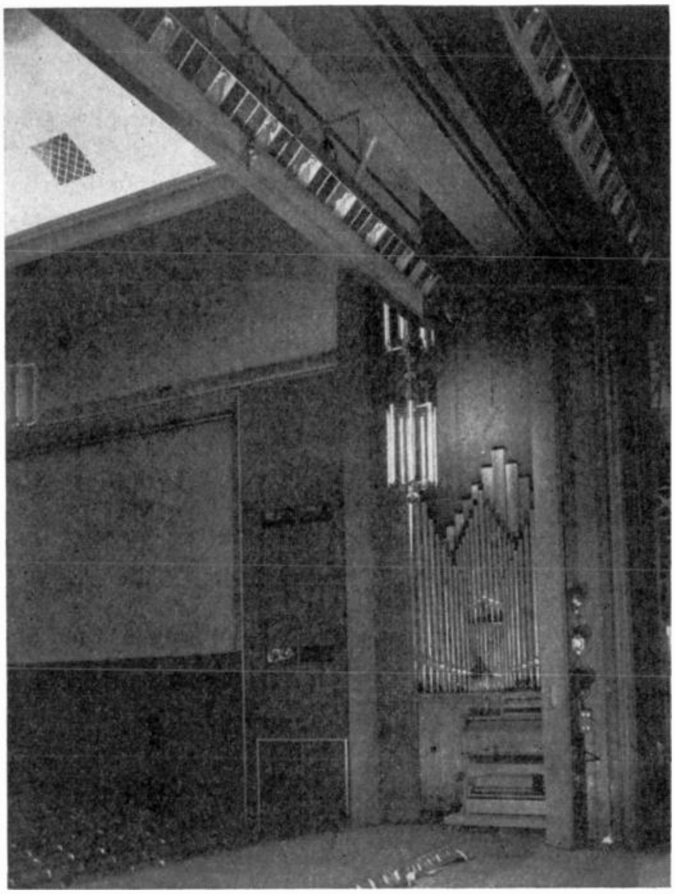

Fig. 7—The small organ in the Philips Theater can be reinforced through an amplifier channel fed from a separate small microphone, which can be seen just above the largest organ pipe.

Fig. 8—In the "Gebouw voor Kunsten en Wetenschappen" the loudspeakers are mounted along the edge of the upper balcony and (not visible in the photograph) under the balconies.

dispersing the loudspeakers over the hall (Fig. 8) and connecting them to the various play-back heads. The wiring is simplified if all the loudspeakers belonging to one group (fed from the same play-back head) are connected in series. In the case of four play-back heads, one four-core cable is run around the hall, balconies, etc; where a loudspeaker is installed, the appropriate core is cut and the speaker connected in series (Fig. 9).

The distribution of the loudspeakers between the various play-back heads should be done as randomly as possible. The only restriction is that the audience should never get the impression that the sound comes from the loudspeakers. We shall now try to explain further the general lines to follow in order to avoid this impression.

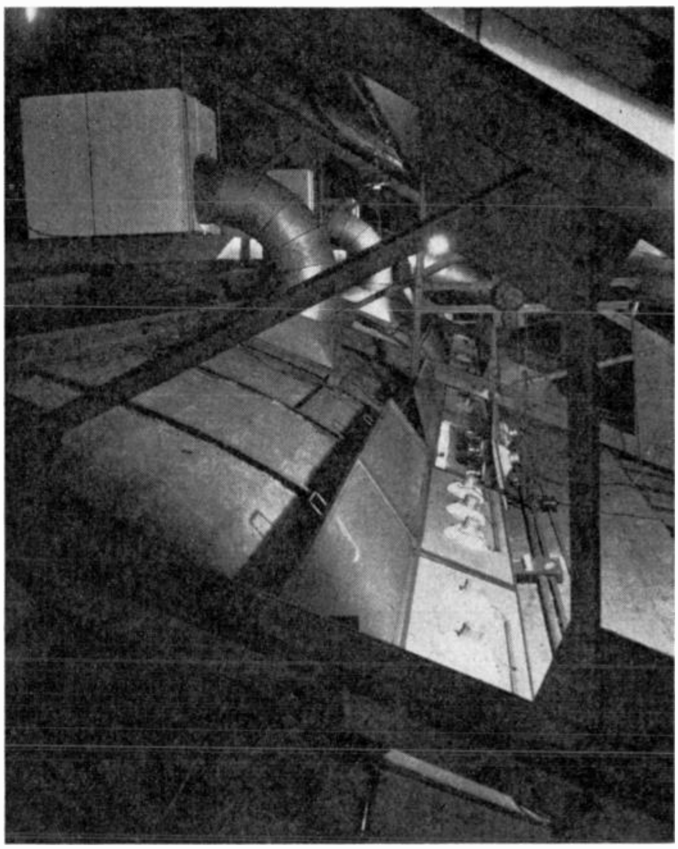

Fig. 9—The loudspeakers of the stereo reverberation installation in the Philips Theater are mounted in groups of two or three in the panels which cover the lighting cornice (see Fig. 6). One of the panels carrying two loudspeaker systems has been lifted up.

We shall use some results of the work of K. de Boer on stereophony.[8] In fact, we are here dealing with an analogous problem. The condition that nobody in the hall may consciously hear music coming from the loudspeakers, means that at all points in the hall the *sound image* must appear to be located in the orchestra. This latter can be regarded as one of the two sound sources in a stereophonic installation, one of the loudspeakers being the other. For the sake of simplicity we assume that the observer is in the plane of symmetry of the two

[8] For a recapitulation of the principles of stereophony, with references to the literature, see R. Vermeulen, "A comparison between reproduced and 'live' music," *Philips Tech. Rev.*, vol. 17, pp. 171–177; December 19, 1955–1956.

sources (if he is otherwise situated, the values to be mentioned presently must be modified by suitable amounts). We then know that he will locate the virtual sound source (the *sound image*) in the orchestra if 1) the sound from the orchestra and that from the loudspeaker arrive at the same time but the intensity of the orchestra is at least 10 db above that of the loudspeaker, or 2) orchestra and loudspeaker are equally loud but the sound from the orchestra arrives 2 msec earlier. These are the two extremes; for intermediate cases, as far as the position of the sound image is concerned, a lag of 1 msec in one sound is compensated by an increased intensity of 5 db, according to the almost linear relationship[9] shown in Fig. 10. The sound image is still located in the orchestra even when the intensity of the orchestra is 5 db less than that from the loudspeaker, provided the sound from the orchestra arrives 3 msec earlier.

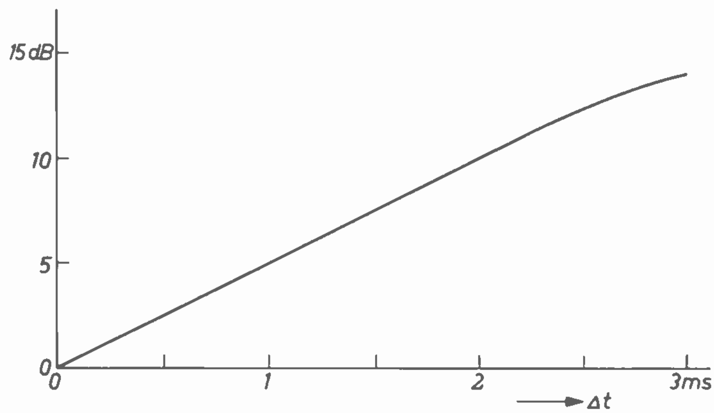

Fig. 10—Differences in intensity (in db) plotted against the phase differences Δt which produce the same angular displacement of the sound image.

Another condition is that for no listener may the sound from the loudspeakers arrive with so great a time lag that it is no longer experienced as reverberation of the orchestra, but as a separate echo. This means that for no observer may the first loudspeaker signal which reaches him arrive more than 50 msec after the direct sound from the orchestra. This value corresponds to that found earlier when investigating the maximum time interval during which, in speech, the indirect sound contributes to the intelligibility (see Introduction). Recent investigations[10] have confirmed this value and also that the indirect sound may be stronger than the direct without disturbing the location of the sound image.

[9] K. de Boer, "Stereophonic sound reproduction," *Philips Tech. Rev.*, vol. 5, pp. 107–114; April, 1940.

[10] H. Wallach, E. B. Newman, and M. R. Rosenzweig, "The precedence effect in sound localization," *Amer. J. Psych.*, vol. 62, pp. 315–336; July, 1949.

G. Meyer and G. R. Schodder, "Über den Einfluss von Schallrückwürfen auf Richtungslokalisation und Lautstarke bei Sprache," *Nachr. Akad. Wiss. Göttingen*, IIa, pp. 31–42; 1952.

On the basis of these data we can set down these rules for a stereo reverberation installation:

1) No member of the audience may receive sound from any loudspeaker before the direct sound has reached him.
2) Nowhere may the first loudspeaker sound arrive more than 40 msec after the direct sound (a safety margin of 10 msec has been deducted from the limiting value of 50 msec).
3) The intensity of the loudspeakers may nowhere be more than 5 db above that of the direct sound[11] (apart from the diffuseness required, this is another reason why many dispersed loudspeakers must be used).

If the same hall is also to be used for plays or lectures and improvement of the intelligibility is desirable, the stereo reverberation installation can also make itself useful for this purpose. One must then ensure that no sound is repeated later than 40 msec after the direct sound, and no feedback should be applied.

Directions for Operation of the Installation

Complying with the rules listed above does not necessarily ensure that the installation works satisfactorily. Because the theory lacks sufficient experimental backing, one should beware of clinging to unfounded preconceived opinions.

An example will illustrate this. In one of our first experiments with stereo reverberation, it was thought that the artificial reverberation should be built up of as many repetitions as possible, in order to obtain the smoothest possible exponentially decreasing intensity. It appeared, in fact, that though this did give the impression of a long reverberation time, this was by no means accompanied by the feeling of being in a large hall—rather that of a small *hard* room such as a bathroom. To suggest a large space, it was necessary to increase the time interval between the echoes and to make the reverberation not at all smooth. By careful adjustment in a laboratory room of about 1000 m^3 volume, we could create the acoustic impression of being in a cathedral.

One general piece of advice relating to electroacoustical intervention in musical performances is that the engineer must show considerable restraint. He must suppress his desire to make the effect *striking* and take care that he does not exaggerate. His is a thankless task. Whenever his work is recognized for what it is, he will be reproached, and the better he does his work, the more natural the result will appear and the less thanks he can expect. The highest praise he can receive is probably the simple verdict: ". . . the orchestra sounded much better."

As we have said, not only the Philips Theater at Eindhoven, but also the Gebouw voor Kunsten en

[11] Meyer and Schodder, *ibid.*, Fig. 8.

Wetenschappen in The Hague, is fitted with a stereo reverberation installation. On November 30, 1954, this installation (then still provisional) had its public debut during a concert given by the Residentieorkest. After the concert, the effects which can be achieved were demonstrated more emphatically.

The verdict on the operation during the concert varied from favorable to very favorable. Some people, however, found it difficult to observe the effect consciously. This is illustrated by a remark from a musician in the audience, "It was remarkable that one could not conciously hear that the installation was in operation; one only felt, or experienced it. Only during the demonstration after the concert did I become consciously aware of it, and convinced that my observations during the concert were not imaginary."

Though the improvement may be appreciated only unconsciously by some of the audience, it is another matter for the performing musicians, both members of the orchestra and soloists; they experience the stereo reverberation very clearly and consciously as making the hall more *playable*. This undoubtedly contributes to the attainment of a higher artistic level.

We feel that it is an important milestone in the development of electroacoustics that leading musicians not only permit microphones and loudspeakers and all that goes with them, in the concert hall, but actually welcome their help.

Other Applications of Stereo Reverberation

Stereo reverberation will undoubtedly find other applications apart from the conversion of theaters, acoustically speaking, into concert halls. However, the apparatus is so complicated that these applications will be limited to the professional field for the time being.

One obvious application is in broadcasting studios. Here, stereo reverberation can be a means of adjusting the acoustics of the studio to the nature of the music or the play to be performed. This can, of course, be done afterwards by adding artificial reverberation to the microphone signal; the echo chamber is a device often used for this purpose. But this deprives the musicians of the stimulus of good acoustics and if there is an audience in the studio, they too miss the full effect. Stereo reverberation can overcome these difficulties.[12]

Another application that springs to mind is in the cinema. Here stereo reverberation can be obtained in the manner described above, with the difference that the microphone is discarded and the recording head on the delay wheel is now fed with the signal derived from the sound track of the film. It is even simpler if the cinema is fitted for a system like "CinemaScope," *i.e.*, fitted with loudspeakers around the hall and projectors suitable for films with more than one sound track; the direct and the delayed diffuse sound could then be recorded beforehand on these tracks, so that the cinema has no need to be equipped with the delay wheel.

Finally we may mention the *duplication of concerts*,[13] that is, the stereophonic reproduction in one or more *overspill* halls of a concert given elsewhere. Clearly, diffuse reverberation in the *overspill* halls can considerably increase the musical value of the program presented.

Conclusion

A shortage of good concert halls means that music is often presented in a hall which is less suitable for this purpose, for example, in a theater. Such halls can be given better acoustics for music by providing artificial diffuse reverberation. An installation is described with which such *stereo reverberation* can be provided electroacoustically. It is based on a so-called delay-wheel, the rim of which is coated with a material suitable for magnetic sound recording. A recording head records the music performed. A number of play-back heads (say, four or six) around the circumference of the rotating wheel pick up the recorded sound with predetermined time lags, and separately feed a group of loudspeakers, mounted throughout the hall. The intensity of each group is independently controllable, and the time lags can be regulated by the speed of the wheel. The sound is picked up by one or two *line microphones*, each consisting of ten condenser microphones with strongly directional characteristics, to reduce the possibility of continuous *howl-back*.

It has been shown that such an installation can produce a great improvement in the musical acoustics of a hall. Some of the audience experience this only unconsciously, but the performing musicians are strongly aware of it as making the hall more *playable*.

Finally, some other possible applications of stereo reverberation are discussed: in broadcasting studios, in the cinema and in the *duplication* of concerts.

[12] According to a private communication from Dr. J. J. Geluk, Head of the Dutch Broadcasting Union Labs., the Union is planning to build two large broadcasting studios (7500 m^3 each) in which the principles of stereo reverberation will be applied.

[13] R. Vermeulen, *Philips Tech. Rev.*, vol. 10, pp. 169–177; 1948–1949. R. Vermeulen and W. K. Westmijze, *Philips Tech. Rev.*, vol. 11, pp. 281–290; 1949–1950.

Contributors

For a photograph and biography of Benjamin B. Bauer, see page 4 of IRE Transactions on Audio, Volume AU-4, Number 1, January–February, 1956.

❖

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V. P. Holec

Mr. Holec is a member of the Military Amateur Radio Service.

❖

R. Vermeulen was born in July, 1897, in 's-Hertogenbosch, the Netherlands. In 1920 he received a Mechanical Engineering degree and in 1923 an Electrical Engineering degree. These were both obtained at the Technical University of Delft.

R. Vermeulen

In 1923, Mr. Vermeulen joined the Philips Research Laboratories at Eindhoven in the Netherlands. He became active in the field of electro-acoustical research, and in 1947 became Director of Acoustical Research. At present he is a Scientific Adviser to the laboratory.

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Wide Range Corner Horn Loudspeaker Systems, Corner Horns

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

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

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

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

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

THE TURNER COMPANY, Cedar Rapids, Iowa
Microphones, Television Boosters, Acoustic Devices

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

UNIVERSITY LOUDSPEAKERS, INC., 80 South Kensico Ave., White Plains, New York
Manufacturers of 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.