Transactions of the I·R·E

Professional Group on Audio

A Group of Members of the I. R. E. devoted to the Advancement of Audio Technology

PGA-10 NOVEMBER - DECEMBER, 1952

TABLE OF CONTENTS

The Institute of Radio Engineers

I.R.E. PROFESSIONAL GROUP ON AUDIO

The Professional Group on Audio is a Society, within the framework of the I.R.E., of Members with principal Professional interest in Audio Technology. All members of the I.R.E. are eligible for membership in the Group and will receive all Group publications upon payment of prescribed assessments.

Annual Assessment: $2.00

The TRANSACTIONS of the I.R.E. Professional Group on Audio

Published by the Institute of Radio Engineers, Inc. for the Professional Group on Audio at 1 East 79 Street, New York 21, New York. Responsibility for the contents rests upon the authors and not upon the Institute, the Group, or its Members. Individual copies available for sale to I.R.E.-P.G.A. Members at $0.70; to I.R.E. Members at $1.05 and to nonmembers at $2.10.

THE PHILADELPHIA IRE-PGA CHAPTER

W. G. Chaney, Secretary
Philadelphia Chapter IRE-PGA

The idea of an Audio Chapter of the IRE in Philadelphia was made public in February, 1952 and was timed to take advantage of a paper being given to the Section by Dr. Leo Beranek. A census of the membership indicated some 75 individuals were interested in joining the Group in addition to the 88 who already belonged.

Plans were temporarily shelved pending conclusion of a highly successful Symposium on Transistor Electronics until late spring. It was felt that this delay would not be serious as plans could then be worked out for full season starting in the fall.

Accordingly, on May 28, 1952, under the auspices of the Executive Committee of the Philadelphia Section, a meeting was held at which Dr. J. G. Brainerd, Vice-Chairman of the Section invited a number of prominent audio people in the area to participate. The outcome of this meeting was the appointment of an administrative committee whose duty would be to obtain recognition of the Philadelphia Chapter, PGA and initiate its activities during the 1952-53 season.

The officers for the first year were selected by the Administrative Committee and are as follows:

Chairman:	Herbert K. Neuber
Vice-Chairman:	Murlan S. Corrington
Secy.-Treas.:	William G. Chaney

The original plans were to schedule five papers during the season. The topics were to be based on the outcome of a questionnaire mailed to the entire section membership. Response to this questionnaire would also provide considerable assistance in a drive to recruit members for the Chapter. The present list of members and interested individuals now exceeds 250.

The first meeting will be October 23, at which time Mr. J. A. Maurer will speak on the subject of "Recording Sound on Film". The meetings will be held in the Auditorium of Radio Station KYW who have kindly agreed to the use of their facilities for as many meetings as desired.

The original plan for five meetings was changed when the section decided to sponsor a Symposium on Audio consisting of six papers spread out over the season. Only three meetings are now planned for this season.

A CERAMIC VIBRATION PICKUP*

E. V. Carlson
Shure Brothers, Inc.
Chicago 10, Illinois

Abstract

The peculiar properties of barium titanates lend themselves to the construction of an acceleration sensitivity transducer which responds primarily to translational components along one direction. A transducer of this type has been constructed using a bimorph crystal element and is described. A calibration by the reciprocity method utilizing three of these transducers is practical, and the calibration techniques are discussed.

Devices utilizing practically every known method of converting mechanical motion into an electrical duplicate have been constructed and used as a means for studying vibration. The principal methods of conversion which have held a dominant position in recent years employ either magnetic fields or piezoelectric crystals. These means are used to transform a small portion of the mechanical energy of a vibration into electrical energy which is readily conveyed to a convenient observation station or which can be made into a permanent record for study later. In the observation of geological motions the magnetic field has been the principal basis for the instrumentation. Here either a magnetic field is caused to move in relation to a coil of wire in which a voltage is induced or an armature is caused to modulate the magnetic field linking a pickup coil. On the other hand, in the industrial field the piezoelectric crystal has found a wide spread usage.

For many years there have been commercially available several crystal vibration pickups of a general utility type. They were designed to use a Rochelle salt crystal piezoelectric element arranged so as to be sensitive to motion. Throughout most of their operational range they yield a voltage proportional to the acceleration applied.

---

*Presented at the National Electronics Conference, September 29, 1952, in Chicago, Illinois. Manuscript received October 10, 1952.

The approximate construction of such a transducer is shown in Fig. 1.[1] This device is caused to function when the motion to be examined is coupled to the housing so that the entire unit is vibrated. Inside the housing, mounted on three of its four corners, is a square Rochelle salt crystal element. This sensitive element is formed by bonding together two properly oriented slabs of the active material. When the crystal is forced into motion at its three supported corners inertia acts over the unsupported portions tending to distort it into a "saddle" shape. For purposes of approximate analysis the distorting force can be replaced by a single force (F) acting at the free corner of the crystal. As the crystal is distorted we find in one slab tensions produced in the direction of one diagonal while compressions are being produced in the other diagonal. A similar condition but of the opposite sense occurs in the other slab. The axis of the crystals have been so arranged that one potential is developed on the exterior surfaces and the opposite potential is developed at the interface of the slabs. Electrodes are provided for collecting these potentials which are made available externally by a cable passing out through the housing.

At frequencies reasonably below the resonant frequency of the crystal the output voltage is directly proportional to the acceleration. At frequencies well above the resonant frequency the output would be proportional to displacement, if the secondary modes of vibration could be sufficiently suppressed. In practice useful measurements is possible up to about one half octave below the resonance. To obtain a signal proportional to velocity or to displacement an electrical integrator as shown in Fig. 2. is useful.

The commercial devices of this type have in general weighed about one half a pound, had there first resonance in the neighborhood of 1,500 cps, had an output impedance equivalent to the impedance of a 10,000 micromicrofarad condenser and have had a sensitivity of about one volt per G of acceleration.

There were some practical limitations inherent to these units:

1. The Rochelle salt crystal is not a rugged chemical combination, it melts at the rather low temperature of 130° F, it is deliquescent in atmospheres of high humidity, and it will lose its water of crystalization in atmospheres of low humidity. Exposure to any one of these three conditions produces a permanent deterioration of the sensitive element.

2. The electrical capacitance is a marked function of temperature. Since the capacity of the crystal and the capacity of the connecting cable form a capacitive voltage divider the instability of this capacity complicates the application where long cables are necessary and where absolute measurement is desired.

3. The vibration pickups of this construction are also sensitive to one component of rotational acceleration as well as to a component of translation. This can lead to erroneous results if care is not taken in the application.

The use of polarized polycrystaline Barium Titanate in place of the Rochelle salt as the sensitive element has some practical attractions:

1. The Barium Titanate material can withstand temperatures in the neighborhood of 200° F without losing its piezoelectric properties. It is not permanently damaged by high humidities or even water, however, it may be temporarily affected by surface leakages and some precautions are required. It is unaffected by dry atmospheres.

2. The dielectric constant of Barium Titanate, and consequently the electrical impedance is a function of temperature. The variations are minor when compared to those found in Rochelle salt. Fig. 3. illustrates this difference.

3. The principal undesirable characteristic of Barium Titanate crystals is that for a similar resonant frequency and output impedance it will exhibit a sensitivity nearly 20 db lower than that of Rochelle salt.

A vibration pickup utilizing Barium Titanate can be similar in construction to one using Rochelle salt. There is need for one fundamental modification in the construction because of a difference in the piezoelectric behavior of the two materials. Where in the Rochelle salt transducer the element was supported on three corners and the acceleration acted upon the fourth, in the Barium Titanate unit the crystal is mounted at all four corners and the acceleration acts upon the central portion of the element.

The ceramic material is isotropic in the plane of the slabs, which is normal to the direction of polarization. If it were compressed along one diagonal and tensed along the other as it is done in the Rochelle salt element the effects would tend to cancel and there would be little resultant output voltage. When the four corners are supported both diagonal directions are treated similarly and there is mostly tension in one slab at the time there is mostly compression in the other slab. With this more symetrical treatment rotations about the center of the element produce forces which tend to cancel in their electrical effects and the unit is not sensitive to rotation. As the housing is accelerated the inertia of the central portion causes the crystal to distort in a "dished" fashion.

A variety of schemes have been devised to obtain calibration data on vibration pickups. We have used several; however, for the last ten years we have obtained measurements of sensitivity by attaching the unit to an electrically driven tuning fork which was equipped with an eyepiece to measure the mechanical displacement. Frequency response data was obtained by mounting the unit on a crystal actuator together with a test cell which was resonant well above the range of frequencies to be explored.(1)

The possibility of using the reciprocity techniques has for many years seemed like a desirable solution to the calibration problem. The reciprocity method of calibration is a technique by which the sensitivity of a transducer can be determined mainly by electrical measurements.

The equipment necessary will be found in most laboratories which in itself is a strong recommendation. The reciprocity method has at its roots the reciprocal requirement that the transfer impedance from the electrical terminals to the mechanical terminal be the same as the transfer impedance from the mechanical terminal to the electrical terminals. If energy is neither lost nor gained as an action passes from the electrical to the mechanical system or vica-versa the method is likely to apply. In the case of these piezoelectric materials the necessary conditions are sufficiently satisfied. In one form this type of calibration envolves making three measurements of electrical transfer impedance and knowing the value of the mechanical coupling impedance.

The derivation of the mathematics of making this type of calibration has been presented amply in the literature.[2] For the present purpose, two equations are sufficient.

$$20 \log S_1 = 10 \log M - 20 \log \omega - 10 \log C_1 + \frac{T_{12} + T_{31} - T_{32}}{2} \qquad (1)$$

$$M = \frac{\dfrac{\Delta M}{E}}{\dfrac{E}{\Delta m} - 1} \qquad (2)$$

Equation 1. expresses the sensitivity (S_1) of the reversible unit in decibles below 1 volt for and acceleration of one meter/second/second. The numbers T_{12}, T_{31}, and T_{32} are ratios of the voltage applied to the motor unit (denoted by the first subscript) to the voltage derived from the receiver unit (denoted by the second subscript) expressed in decibles. C_1 is the electrical capacity of the first or reversible unit in farads, and M is the coupling mass in kilograms. If T_{31} and T_{32} are interchanged a calibration of Unit No. 2 (S_2) is obtained.

The coupling impedance is treated as if it were a simple mass and its value is adjusted as a function of frequency in accordance with Equation 2. This method will function satisfactorily if dissipative portion of the coupling impedance is small compared to the mass reactance of the coupling impedance. The value of M can be determined with sufficient accuracy in some instances by direct weighing of the apparatus, however, a dynamic measurement is desirable as the frequency is increased into the neighborhood of the first resonance. The dynamic measurement of the effective coupling mass is accomplished by adding an incremental mass (ΔM) to the assembly and noting the effect on the transmission.

A convenient arrangement for making this type of calibration is outlined in Fig. 4.

Three of the pickups are bolted together with a provision for adding an incremental mass. One of the units is across the input of an attenuator along with an audio oscillator. The output from one of the other two units is fed to the high impedance input of a preamplifier, which is followed by a bandpass filter and an indicating meter. The output of the attenuator is substituted for the output of the driven pickup and the attenuator is adjusted to provide the same indication on the meter. Four measurements are made, T_{12}, T_{31}, T_{32} and T_{12} with the incremental mass added. An additional measurement of C_1 at the same temperature and frequency is required and there is sufficient information to obtain calibrations on units #1 and #2.

In a group of three units it is possible to make six measurements of transmission and three measurements of capacity. This data will provide four calculations of sensitivity on each unit using different combinations of data. Table 1. is a tabulation for four such calculations on a single unit throughout the temperature range from 15° to 170°.

The frequency response of another unit has been calculated the four ways the six measurements can be combined. This data appears in Table 2. The data is in good agreement with itself and yields a result approximately 1 db more sensitive than the calibration obtained at 60 cycles per second on the tuning fork calibrator.

When we first attempted to perform this experiment in 1940 using the Rochelle salt pickups the results were not nearly so encouraging. At that time there were two major pitfalls: The capacity of the motor unit was a function of voltage, and the sensitivity to rotational motion tended to excite rotational motions. The ceramic pickup has sensibly overcome these difficulties as the electrical capacity is only slightly dependent on the applied voltage, they do not tend to excite rotations, and they are mechanically well adapted to coupling together in a reliable fashion. It appears that the reciprocity calibration using more or less standard laboratory equipment is very practical.

References

1. Benjamin B. Baumzweiger (Bauer), "Application of Piezoelectric Vibration Pick-Ups to Measurement of Acceleration, Velocity and Displacement", Journal of the Acoustical Society of America, vol. 11, No. 1, January, 1940, p. 303.

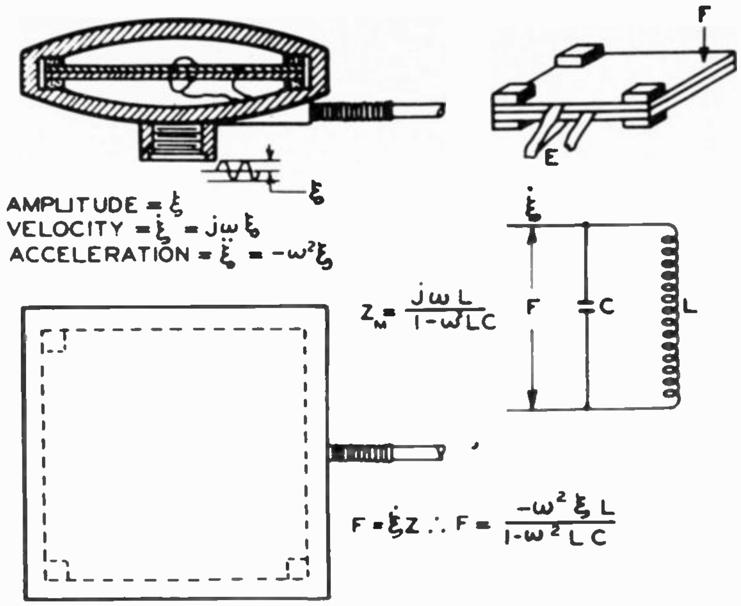
2. Mark Harrison, A. O. Sykes, and Paul G. Marcotte, "The Reciprocity Calibration of Piezoelectric Accelerometers", Journal of the Acoustical Society of America, Vol. 24, No. 4, July, 1952, p. 384.

Table 1.

Temperature °F	Sensitivity - db re 1 volt per meter per second per second			
15	- 43.8	- 43.9	- 43.7	- 43.9
36	- 43.2	- 43.3	- 43.3	- 43.2
55	- 43.4	- 43.3	- 43.3	- 43.4
69	- 43.5	- 43.5	- 43.5	- 43.6
84	- 43.8	- 43.8	- 43.8	- 43.8
89	- 43.8	- 43.8	- 43.8	- 43.9
105	- 43.9	- 43.9	- 44.0	- 44.0
132	- 44.3	- 44.4	- 44.3	- 44.3
152	- 44.7	- 44.7	- 44.6	- 44.6
170	- 45.0	- 45.1	- 44.9	- 45.0

Table 2.

Frequency cps	Sensitivity - db re 1 volt per meter per second per second			
250	- 45.4	- 45.5	- 45.2	- 45.1
350	- 45.3	- 45.3	- 45.0	- 45.0
500	- 45.0	- 45.0	- 44.6	- 44.7
700	- 44.7	- 44.6	- 44.3	- 44.3
1000	- 43.1	- 43.1	- 42.7	- 42.6
1500	- 39.6	- 39.6	- 39.3	- 39.2
1800	- 33.9	- 33.9	- 33.6	- 33.5
2000	- 27.5	- 27.7	- 27.4	- 27.4
2200	- 25.1	- 25.6	- 25.6	- 25.6
2400	- 30.8	- 30.8	- 30.9	- 31.2



INERITIA-TYPE PIEZO-ELECTRIC VIBRATION PICKUP
AND ITS APPROXIMATE EQUIVALENT CIRCUIT

Fig. 1

SCHEMATIC DIAGRAM OF CONTROL BOX

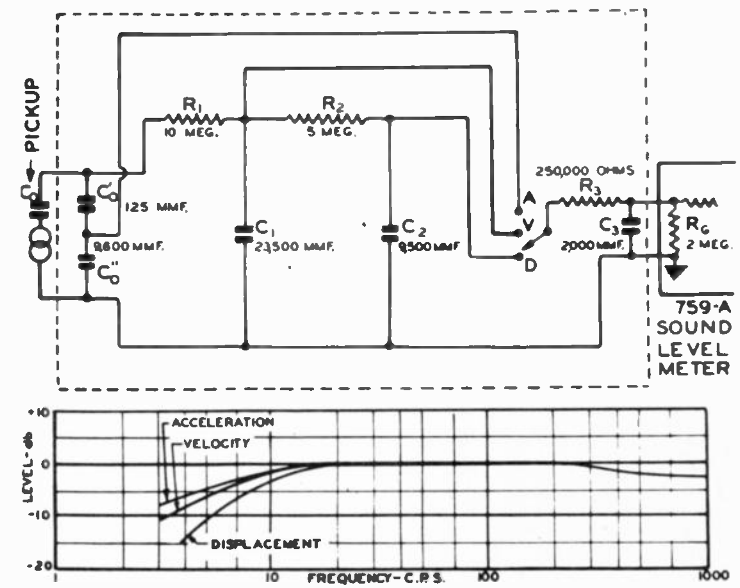


Fig. 2

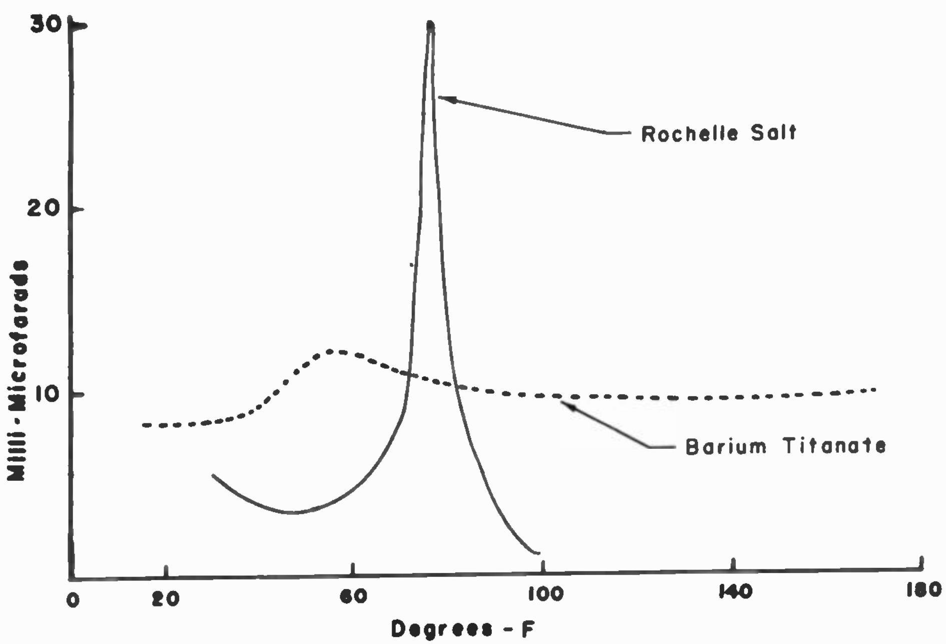


Fig. 3

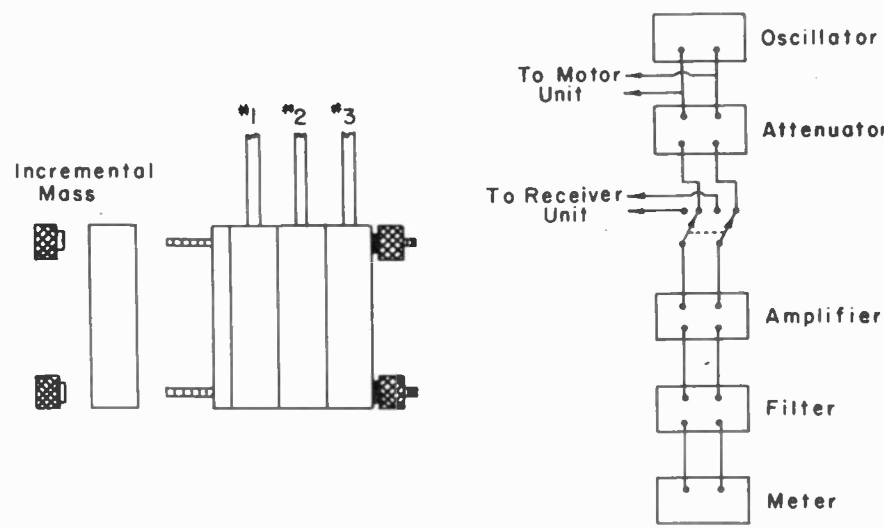


Fig. 4

ENGINEERING CONSIDERATIONS IN THE USE OF MAGNETIC RECORDING HEADS*

Lee Gunter, Jr.
Shure Brothers, Inc.
Chicago 10, Illinois

The design engineer or the experimenter contemplating magnetic recording for the first time is immediately confronted with questions regarding the proper use of magnetic recording heads and the circuitry to be employed in connecting these heads to amplifiers or to radio receivers. The purpose of this discussion is to describe simple basic concepts and circuits which may be used as a start of such design work.

Tape heads have three functions: recording the signal on the tape, playing back the signal, and erasing the previous recording. A basic problem is to supply suitable power and amplifier characteristics to perform these functions. Other basic problems are: recording and playback head alignment, corrective networks, and reduction of hum pickup.

General Considerations

Magnetic recording tape exhibits a non-linear magnetization curve similar to all magnetic materials. Without suitable means to overcome this effect, a distorted recording would result. This distortion can be reduced by adding a second current in the recording head along with the audio signal to be recorded. This current is called a bias current. Bias current can be a direct current, or it can be a high frequency ultrasonic current.

Since erasure of the recording tape is generally done just prior to recording, and since the condition of the tape after erasure also affects the recording and biasing process, the type of erasure must be given consideration along with the type of bias.

Signals on tape may be erased with a permanent magnet or with a D.C. current applied to erase head windings, or with a high frequency or ultrasonic current. Ultrasonic erasure demagnetizes the tape leaving it in a magnetically neutral condition. High frequency erase is generally employed in the highest quality applications. Permanent magnets or D.C. erase saturates all portions of the tape thereby eliminating the signal and leaving the tape in a magnetically saturated condition, although in some instances special heads are made to achieve a neutral condition. It is possible to employ a high frequency bias in recording on tape which has been erased by

*Manuscript received 10/9/52.

either D.C. or high frequency. When using D.C. bias, it is generally the practice to use D.C. erase, since a saturated state of the recording tape may be overcome by introducing the D.C. bias of opposite polarity.

General Bias Considerations

In both high frequency A.C. bias and D.C. bias, the bias current necessary for the particular application is dependent upon the characteristics of the recording tape and also upon the type of equipment and application for which it is intended. The type and magnitude of bias affects the distortion, signal to noise ratio, and, to some extent, the high frequency extension of the recording. The typical values for bias to be used with Shure tape heads as shown on the individual data sheets have been selected in accordance with RTMA specification REC-134 and should produce less than 2% distortion. In using high frequency bias, an increase in bias current above that shown as nominal bias has the effect of decreasing distortion in recording and also decreasing the high frequency extension. Naturally, under these conditions, a compromise must be made between distortion and high frequency extension. When employing D.C. bias, an increase or decrease in bias from the nominal value increases distortion in recording. The D.C. bias current is also dependent upon the magnetic properties of the recording tape. In addition to the foregoing factors, the use of A.C. bias produces a lower noise level than does the use of D.C. bias. This has led to a preferred use of A.C. bias in high quality recording.

Ultrasonic Erase and Bias

Fig. 1 shows an oscillator circuit which may be used as a source of high frequency A.C. erasing and biasing. This oscillator is designed to operate at 25 kcps and will give ample voltage and power to both erase and bias the Shure TR-5 series heads or the TR-16 record head and TE-2 erase head. The coil "L" is wound on a powdered iron core 1/4" diam. x 1-1/16" long using 750 turns of No. 25 Formex or enamel wire and tapped at 150 turns and 300 turns. Additional taps may be added to provide a variable voltage. The frequency of the bias is usually chosen to be about five times the highest audio frequency to be recorded. The oscillator frequency may be varied by increasing or decreasing the 0.003 mmf condenser.

Fig. 2 shows the connections of a Shure combination erase and record-playback head of the TR-5 series in a typical recording circuit using high frequency erase and bias. The 500 mmf trimmer capacitor adjusts the bias current and the 120 k resistor provides for constant current audio signal. The recommended average audio current and bias current is specified on the data sheet for the individual heads.

Permanent Magnet Erasure

Tape can be erased by means of a permanent magnet placed close to, or in actual contact with, the tape. The dimensions and orientation of the magnet are best determined by experiment. Mechanical means must be provided to move the magnet away from the tape when rewinding and playing back.

Direct Current Erase and Bias

Fig. 3 shows a Shure combination erase and record-playback head of the TR-5 series in a typical recording circuit using D.C. erase and D.C. bias. The erase and bias currents are shown provided by batteries, but a well-filtered "B" supply is satisfactory. The head must be connected with the proper polarity, as shown, otherwise, the output will be distorted and low in level.

Response Correction

The typical response frequency characteristic of a tape recording head rises at the rate of 6 db per octave in the lower frequency range and falls off fairly rapidly after the turnover point. The frequency of this turnover point is dependent upon the tape speed, and characteristics of the tape, and the tape recording and playback head.

In some applications in voice and communication recording, the constant current recording and uncompensated playback response frequency characteristic shown on the typical data sheet give quite satisfactory results. In applications where more uniform response is desired, compensation networks in recording and playback are generally used.

For the most part, compensation has been accomplished in two very general ways: (1) High frequency and low frequency boost in both recording and playback -- this process being the simplest form when the same amplifier is employed in recording and playback inasmuch as there is no necessity to provide switching means on the compensation network; and (2) High frequency correction in both record and playback and low frequency correction in playback only.

Means of attaining high and low frequency boost in amplifier circuits are well-known to the experimenter. Care should be taken to avoid the addition of capacitances and resistances directly across the terminals of the head in either playback or recording. Since the head is principally an inductance, the addition of circuit elements across the head may seriously alter its characteristics. Fig. 4 shows the recommended connections of the Shure TR-5 series or TR-16 type head to the high gain playback amplifier.

An example of a flexible interstage network is shown in Fig. 5. When recording, the input terminals are connected to the microphone and the output terminals to the recording head circuit; in playing back, the input terminals are connected to the playback head and the amplifier output connected to the audio output system. The circuit as shown incorporates high frequency emphasis in recording and playback and low frequency correction in playback only.

Reduction of Hum Pickup

Consideration should be given early in the stages of design toward circuit arrangement and isolation to provide minimum hum pickup in playback. This can best be accomplished by keeping the magnitude of the hum field at the playback head as low as possible. It is good practice to experimentally determine the best position for hum producing components such as motors and transformers which will produce the minimum hum pickup at the head. Although a substantial reduction in hum pickup is accomplished by enclosing the playback head in a high permeability shield, it is important that severe hum producing components be carefully located. It is quite often found that orienting the position of a power transformer by experiment can accomplish as much hum reduction as may be realized by additional expensive shielding.

Mechanical Operation Considerations

When mounting the tape recording head, head alignment means, tape guiding means, and the path of tape travel over the head, must be considered. It is of utmost importance that the playback gap be aligned with the recording gap or serious high frequency losses occur in playback. The mounting and control on playback head should allow for the playback gap to be adjusted perpendicular to the line of motion of the tape to within $\pm 1/4^{\circ}$ without changing the position of the recording head reference to the tape track. The adjustment may be made by adjusting for maximum output on standard alignment tape.

Patent Considerations

While magnetic recording is an old art, there are many patents covering various phases of this art, and care should be exercised to avoid infringement of any valid patents. We cannot guarantee that by following instructions and information set forth above the user will not come within the scope of certain valid patents.

OSCILLATOR CIRCUIT

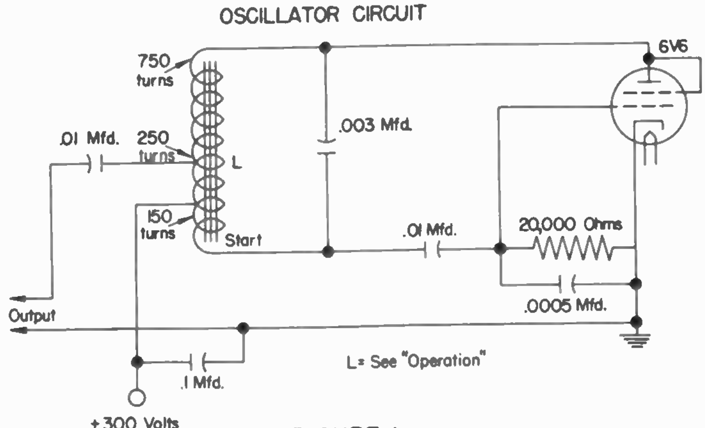


FIGURE 1

RECORD CIRCUIT USING A.C. BIAS AND ERASE VOLTAGE

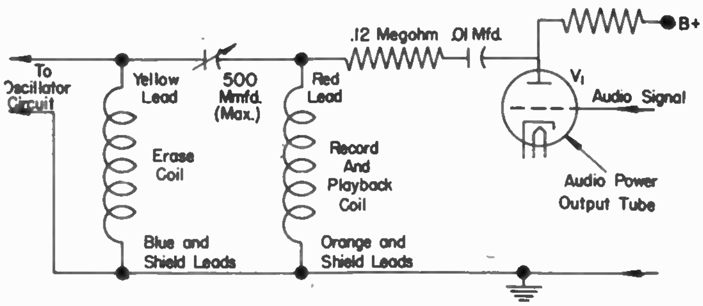


FIGURE 2

RECORD CIRCUIT USING D.C. BIAS AND ERASE VOLTAGE

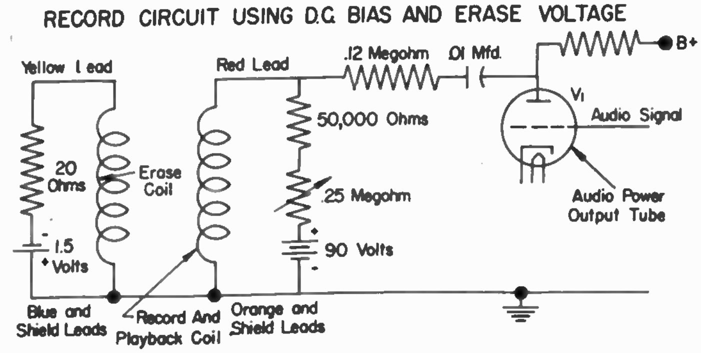


FIGURE 3

PLAYBACK CIRCUIT

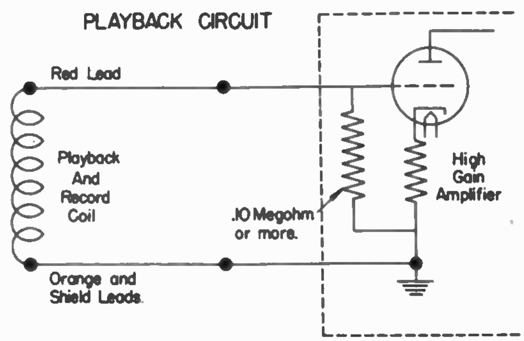


FIGURE 4

INTERSTAGE COMPENSATING NETWORK

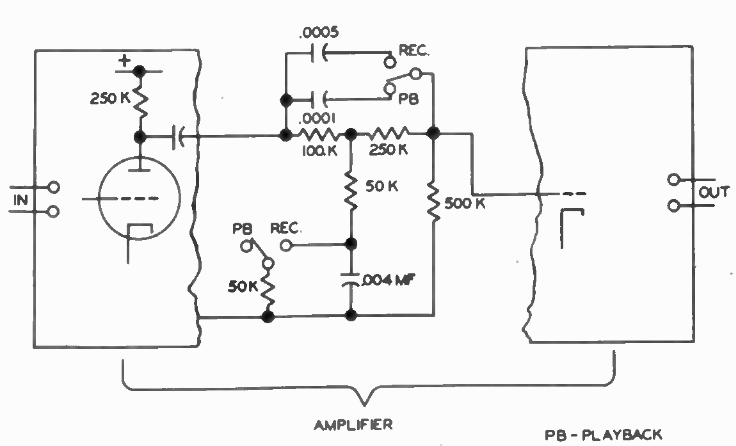


FIGURE 5

DIRECT MEASUREMENT OF THE EFFICIENCY OF LOUDSPEAKERS BY USE OF A REVERBERATION ROOM*

H. C. Hardy, H. H. Hall, and L. G. Ramer
Armour Research Foundation of Illinois Institute of Technology
Chicago, Illinois

Abstract

If a loudspeaker has an axis of symmetry, its total radiated power, and therefore its efficiency, is computed by calculations from free-field data taken at various angles. The objections to this method are: (1) the calculations are very tedious; (2) loudspeakers as used in typical baffles do not have an axis of symmetry; and (3) elimination of phase effects requires measurements at a comparatively large distance outdoors or in a large anechoic room. The problem can be more directly approached by integrating the total power output in a large reverberation room such as the ones at the Foundation's Riverbank Acoustical Laboratories or at the Bureau of Standards. Data will be presented on the results of such measurements and the care that must be taken to obtain random sound conditions in the measurement room will be discussed. When a loudspeaker has been given a primary calibration by such a measurement, it may be used as a secondary standard in a smaller reverberant space, for example, in the manufacturer's laboratory.

I. Introduction

This paper will discuss a subject about which there has been some speculation and many exaggerated claims, namely the electroacoustic output efficiency of loudspeakers. For many years there has been considerable agitation for acoustic scientists and engineers to establish standards for measurement of power output and efficiency. Such standards have been slow in being codified because there is (1) considerable disagreement on what should be measured, and (2) the conventional methods are too complex and time consuming, particularly in the usual case where the speaker enclosure does not have an axis of symmetry.

No attempt will be made here to elaborate on which properties need to be measured except to state there is considerable interest in knowing the total acoustic power output. This is particularly true when the speaker is to be used indoors. Outdoors, the response on axis, or within some small angle on each side of the axis, is the chief criterion. Indoors, it should be remembered that at some distance (around five feet for small reverberant rooms and

*Presented at the National Electronics Conference, September 29, 1952, in Chicago, Illinois. Manuscript received October 8, 1952.

50 feet for large non-reverberant rooms) more energy arrives at a point in the room from the multiple reflections from the walls than from the directly radiated sound. In this case the total power output, especially as a function of frequency, becomes the main criterion.

This paper will present a relatively simple method for integrating the power output by the use of a reverberation room. No particular originality is claimed for this method. Although it does not appear to have been published, it is known to have been used at least in an elementary manner by other experimenters.

II. The Conventional Method of Measuring Output Power

The power output of a loudspeaker is conventionally computed from the directivity index which in turn is computed from the measured response of the loudspeaker at various angles in an anechoic room.[1] It is assumed that the measuring microphone is far enough away that the energy flux flows radially. The power output in watts, P, is then given by the equation

$$P = \frac{4\pi p_{ax}^2 r^2}{Q\rho c \times 10^7} , \quad (1)$$

or the power level above a milliwatt by

$$PL = L_{ax} + 20 \log r - DI - 119.2 , \quad (2)$$

where

p_{ax} = sound pressure level in microbars on the axis

r = distance from loudspeaker to microphone in centimeters

ρc = specific acoustic impedance of the air = 41.4 cgs units

L_{ax} = sound pressure level re 0.0002 microbar on axis in a free field

Q = directivity factor

DI = 10 log Q = directivity index.

The directivity factor is defined as

$$Q = \frac{4\pi p_{ax}^2}{\int_0^{2\pi}\int_0^{\pi} p^2(\theta,\varphi) \sin\theta \, d\theta \, d\varphi} , \quad (3)$$

The spherical integration must be made over all space in which the loudspeaker radiates energy (see Fig. 1). The difficulty in measuring the output power comes in obtaining this integral which must usually be calculated numerically. Also, it is usually assumed that the speaker has an axis of symmetry; otherwise the integration becomes too difficult to be practical. In this case

$$Q = \frac{1}{\frac{\pi}{2n} \sum_{n=1}^{n} \frac{p^2(\theta_n)}{p_{ax}^2} \sin \theta_n}, \tag{4}$$

where n is the number of points around a half-circle that measurements are made. Usually measurements are made every ten degrees which means that measurements and calculations must be for 18 points for every frequency at which a value is desired.

Before discussing other difficulties with this method, it can be pointed out that because of the sin θ term in the integration, most of the power is not radiated along the axis but at an angle from the axis of between 40 and 60 degrees at low frequencies and smaller at higher frequencies. At the angle of maximum radiation diffraction effects are usually quite pronounced and the jagged response curve may introduce considerable uncertainty in the integration.

These and other disadvantages of obtaining the power radiated by a loudspeaker by free-field measurements and numerical integration can be summarized as follows:

1. The method is only practical when applicable to loudspeakers in enclosures which have a symmetrical axis. This is true of only a very limited number.

2. In order that the above mathematical expressions hold, the measuring microphone must be far enough away from the loudspeaker that the intensity vector is normal to a sphere with its center at the point of rotation of the loudspeaker. In practice this means that the distance r above must be several times the largest dimension of the loudspeaker. Therefore, often measurements must either be made outdoors or a very large anechoic room must be available. This also often means that it may be very difficult to get sufficient power output to measure the radiation of the loudspeaker off its axis at these distances.

3. The jagged response curve either due to cone breakup, horn or cavity resonance, or diffraction effects makes it difficult to estimate the radiation at any given angle.

4. The measurements and calculations are rather tedious so that only a limited number of frequencies are computed in practice.

This paper will describe the reverberation method which has been developed to overcome these limitations.

III. Theory of the Reverberation Method of Measuring Power Output

The reverberation room method of measuring sound output consists of integrating the total power output in an enclosed reverberant space where the energy flow is random and sound energy density is known to be substantially constant so that a measurement of sound pressure anywhere in the room is related directly to the total integrated power radiated by the source. Such rooms have been built for measurement of the sound absorption of acoustical materials and several exist in this country. Figure 2 is a photograph of the interior of the reverberation room at the Riverbank Acoustical Laboratories, operated by Armour Research Foundation.

In general, a large reverberation room will not necessarily have a random sound field, especially at low frequencies. It is necessary that several precautions be taken in order that the energy density be made substantially constant throughout the room. These will be discussed below.

When a source of energy is introduced into such a room, the sound energy at a point in the room rapidly reaches an equilibrium where energy is absorbed at the same rate that it is introduced. This absorption occurs at the surfaces of the walls and, at high frequencies, in the air due to its molecular absorption. The power emitted to the room is $\underline{P}$. The power absorbed in the room is the sound intensity $\underline{I}$ multiplied by the total absorption $\underline{a}$, so that

$$P = Ia \quad . \tag{5}$$

The total absorption in the room is measured as in sound absorption measurements by observing the rate of decay of sound in the room. This is essentially a constant for the particular room used. In general, it is related to the decay time, $\underline{T}$, by Sabine's equation[2]

$$a = \frac{0.049\ V}{T} \text{ (British units)}, \tag{6}$$

or

$$a = \frac{0.00161\ V}{T} \text{ (cgs units)}, \tag{6a}$$

where $\underline{V}$ is the volume of the room, and $\underline{T}$ is the time for sound to decay 60 db.

No apparatus is available to measure directly the sound intensity $\underline{I}$ since most microphones are sound pressure measuring devices. In a truly random reverberant space the reverberant sound pressure $\underline{p}_{rev}$ is related to the intensity (watts per cm^2) by[3]

$$I = \frac{p_{rev}^2}{4\rho c \times 10^7} \tag{7}$$

where $\underline{\rho}$ is the density of the air, and $\underline{c}$ the velocity of sound.

The power output of a sound source in such a room is, therefore,

$$P = \frac{p_{rev}^{2} a}{4\rho c \times 10^{7}} . \tag{8}$$

In decibels the power level above one milliwatt becomes

$$PL = L_{rev} + 10 \log a - 106.5 , \tag{9}$$

where L_{rev} is the reverberant sound pressure level in decibels referred to 0.0002 microbar and a is the absorption in sabins (British units).

The only limitations on the above simple theory are that (1) the test room has low enough absorption so that Sabine's equation is valid, and (2) the sound energy in the room is sufficiently random so that equation (7) above holds. The latter assumption would not be true for most rooms. Recent experimentation on reverberation rooms[4] indicates that the latter is very close to being true provided that certain important steps are taken to randomize the sound in the room. These are:

1. A warble tone is used to excite many modes of vibration of the room.

2. A large vane (see Fig. 2) is rotated slowly and thus effectively causes the room shape to be changed constantly.

3. Diffusing cylinders or other means are used to further diffuse the sound. (See Fig. 2.)

4. There is enough absorption present and randomly scattered so that room resonances are not sharply tuned.

All these criteria are maintained in the Riverbank reverberation chamber and most of the other large reverberation rooms used by standard laboratories to measure acoustic absorption. These laboratories also have ready means available to find the acoustic absorption present. The acoustic absorption at Riverbank as a function of frequency is given in Table I.

Table I

Frequency (cps)	125	250	500	1000	2000	4000	8000
Absorption (sabins)	97	96	99	101	116	160	250

Below 1000 cps, the absorption will practically be constant. At higher frequencies, a large contribution is made up of molecular absorption in the air and this will vary with the absolute humidity.

IV. Technique of Measurement

The speaker to be measured is placed in the room usually near one of the corners. The microphone or microphones are placed near another corner so that the distance of separation is such that the reverberant level in the room is much greater at the microphone position than the direct sound from the loudspeaker.

The loudspeaker is connected to its source of power and, if efficiency measurements are desired, this connection follows the technique for measuring constant available power. With the vanes rotating and frequency warbled, the driving frequency is swept very slowly over the range to be measured. Such a sweep takes approximately 40 minutes to run from 20 to 10,000 cps, during which the vane makes 40 half-revolutions.

The signal picked up by the calibrated 640AA condenser microphone is fed into a graphic level recorder. Portions of such a record between 100-200 cps, 600-1200 cps, and 4000-8000 cps are shown in Fig. 3. The fluctuation in level, 2 db at high frequencies and as much as 5 db at low frequency, have been minimized by the warble and vane rotation. From such a trace it is not difficult to draw an average curve to within a decibel.

Although these curves indicate gradual changes in power level with frequency, it should be remembered that the frequency scale is greatly elongated and large changes of power level with frequency occur in most loudspeakers.

In order to obtain the reverberant sound pressure level, it is necessary to know the random calibration of the microphone. For a microphone with an active surface as small as the 640AA microphone, the random calibration follows very closely the 90-degree incidence calibration up to 3000 cps. The random incidence calibration is only 1 db higher at 5000 cps and 2 db higher at 10,000 cps.

Two corrections with frequency are therefore needed to get the acoustic power from the sound pressure level: (1) the random microphone calibration curve and the room absorption curve. In Fig. 4 are plotted the two correction curves. Actually the two corrections tend to compensate at high frequencies so that the total correction with frequency varies only by $\pm$ 1 db up to 8000 cps.

V. Correlation of the Two Methods

Experiments were undertaken to compare the numerical integration and room integration method of measuring power output. A special cabinet was built which provided reasonable axis of symmetry. It consisted of a closed box 32 x 32 x 16 inches, with the loudspeaker opening in the center of one of the 32 x 32-inch sides. A 12-inch loudspeaker and an 8-inch speaker with adapter were used. Both speakers were measured at a distance of 10 feet in the anechoic room. The on-axis response curve, and the power output curve for the 12-inch loudspeaker as measured in the reverberation room are shown in Fig. 5. Similar curves are shown for the 8-inch speaker in Fig. 6. The sound pressure level is referred to 0.0002 microbar and the power level to one milliwatt. The circular points are the data computed from the free-field response measured around the loudspeaker. Nearly all the points are within $\pm$ 1 db.

In both experiments, the speakers were driven by the constant available power method[5] and the input power available was 2 watts. The efficiency of the two speakers as measured in the reverberation room at a few frequencies are given in Table II.

Table II

Efficiency of Two Typical Loudspeakers

Frequency (cps)	Efficiency (Percent) 12-inch Speaker	8-inch Speaker
70	0.1	0.1
100	0.4	0.5
200	1.1	1.1
400	1.1	1.1
700	0.7	2.2*
1000	0.5	1.0
2000	0.3	0.6
4000	0.3	0.3
7000	0.1	0.4
10000	0.02	0.01

*Resonant peak

It will be noticed that these efficiencies, of the order of one percent, are much lower than those generally stated for loudspeakers. However, the calculations usually given are computed from the axial response, ignoring the directivity factor which in the mid-frequency range is of the order of 6 db.

Another measurement of an 8-inch speaker in a smaller cabinet is given in Fig. 7. These measurements were made several years ago by Mr. J. E. Ancell and Mr. D. E. Bishop of our organization. Again the check is good.

The power response of several other large loudspeakers is shown in Figures 8, 9, and 10. Notice that large differences are measured in the power response frequency curves which are significant to the acoustic designer.

VI. Industrial Use of the Reverberation Power Measurement Technique

Reverberation rooms are not commonly available for output power level measurements as indicated above. However, if the loudspeaker manufacturer or loudspeaker purchaser wishes to obtain such data in his own plant and has a fairly large empty room available, the following procedure is recommended. One loudspeaker of a particular type may be calibrated in a reverberation room by the method outlined above. Using this as a secondary standard, other similar loudspeakers may be measured by a substitution method by using a variation of the techniques outlined above. An ordinary sound level meter can be used to measure the sound energy as a warbled tone is applied to the speaker. Point by point or graphic recording techniques can be used to obtain the data.

VII. Conclusions

The reverberation room method of integrating the power output of a loudspeaker has been brought to the stage of development where it can be applied to commercial loudspeaker testing. Its convenience, accuracy, dependability, and suitability for presenting the complete output power data should make it a useful tool for establishing standards and obtaining design information.

The authors are grateful for the help and advice of other members of the acoustic staff of the Armour Research Foundation and particularly to Mr. James E. Ancell who helped in the earlier development of this method.

References

1. Beranek, L. L., Acoustic Measurements, John Wiley and Sons, Inc.(1949), p680f.

2. Morse, P. M. Vibration and Sound, 2nd ed. McGraw Hill, 1948, p. 387.

3. Ibid, p. 414.

4. Hardy, H. C. and Tyzzer, F. G., Experimentation on the Theory of the Reverberation Method of Measuring Sound Absorption. Jour. Acoust. Soc. Am. 24, 115A.

5. Beranek, loc. cit. p. 664 f.

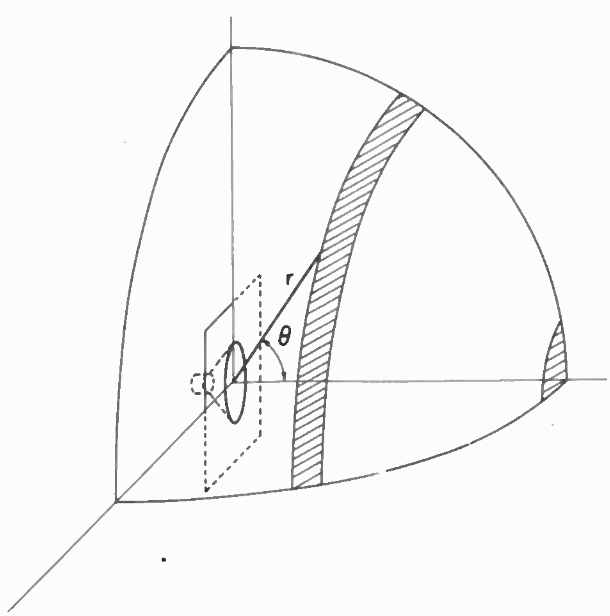


Fig. 1 - Coordinate system used for spherical integration. The origin is located at the effective center of loudspeaker cone.

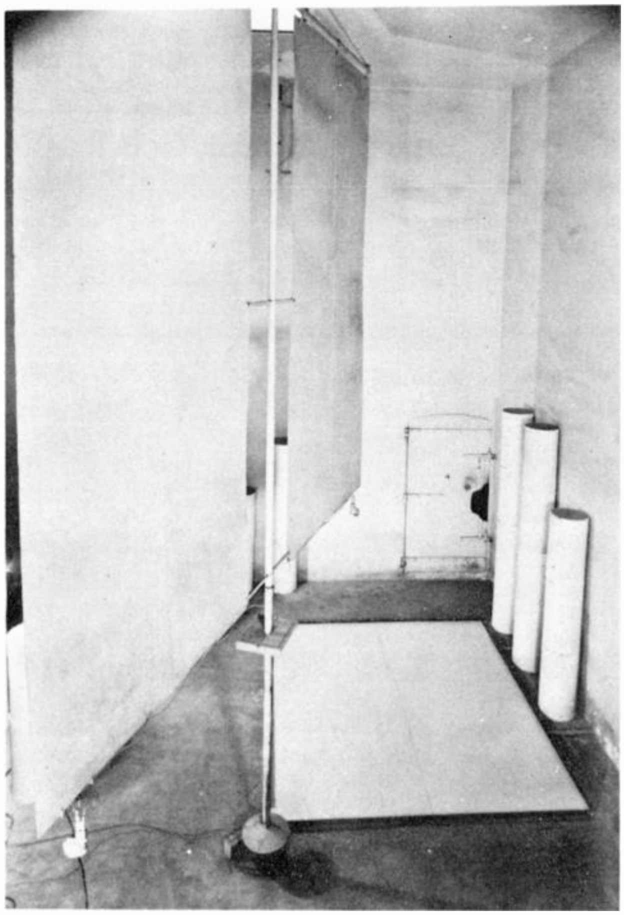


Fig. 2

Interior view of Riverbank reverberation chamber showing diffusing pillars and rotating vanes.

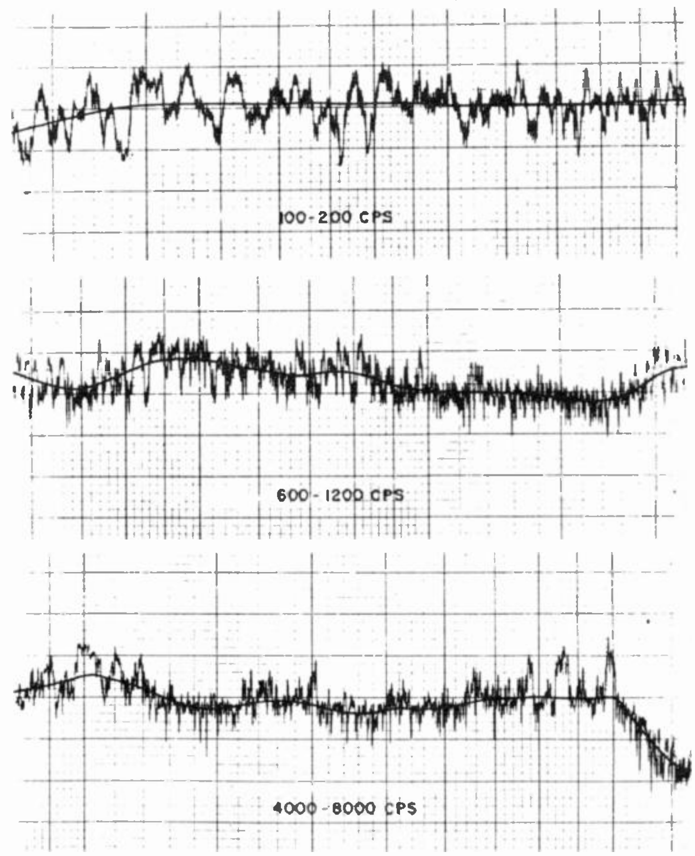


Fig. 3

Sections of chart from graphic level recorder.

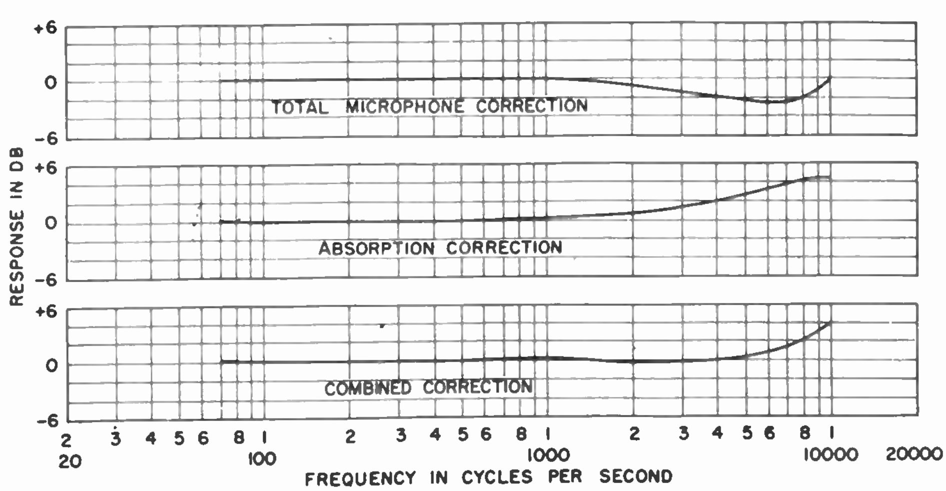


Fig. 4

Curves showing corrections applied to microphone terminal voltage to obtain reverberation level.
(Top) Microphone and preamplifier calibrations.
(Center) Correction for room absorption.
(Bottom) Combined correction curve.

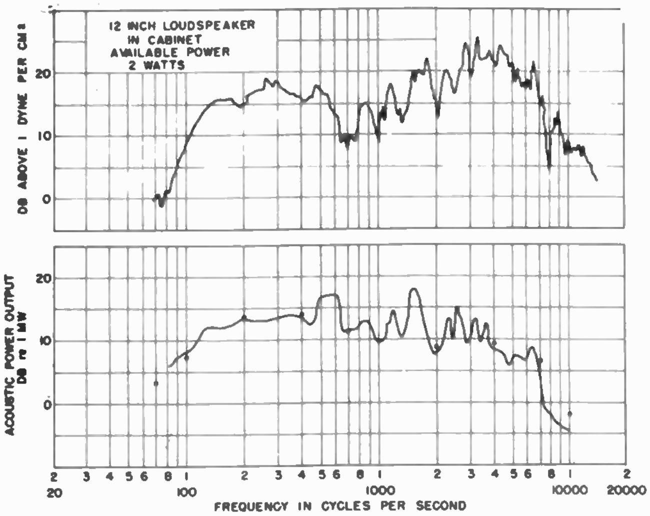


Fig. 5

(Top) Free-field response on axis of 12-inch loudspeaker at a distance of 10 feet.
(Bottom) Acoustic power output of same loudspeaker as measured in the reverberation room.

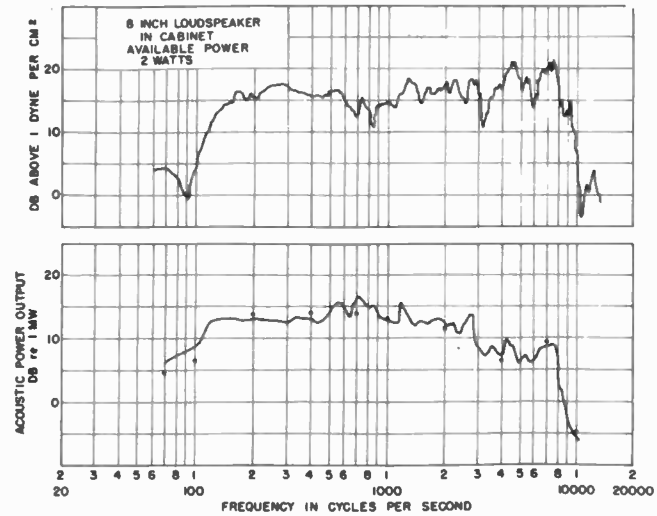


Fig. 6

(Top) Free-field response on axis of 8-inch speaker at a distance of 10 feet.
(Bottom) Acoustic power output of same speaker as measured in the reverberation room.

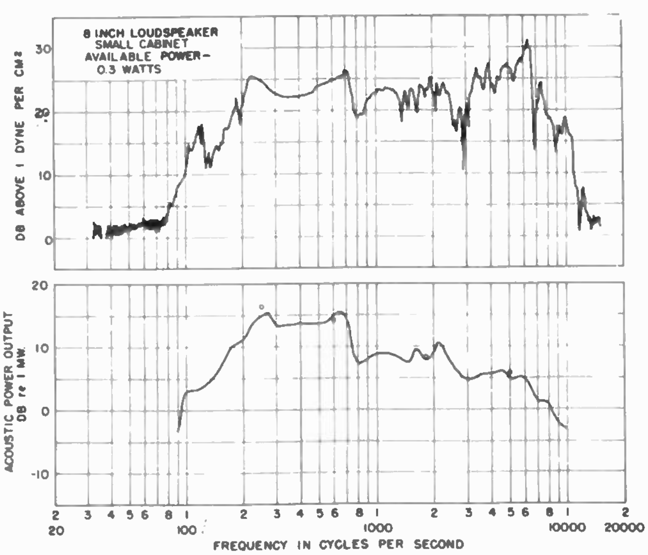


Fig. 7

(Top) Free-field response on axis of an 8-inch speaker in small cabinet at a distance of 3 feet.
(Bottom) Acoustic power output of same speaker as measured in the reverberation room.

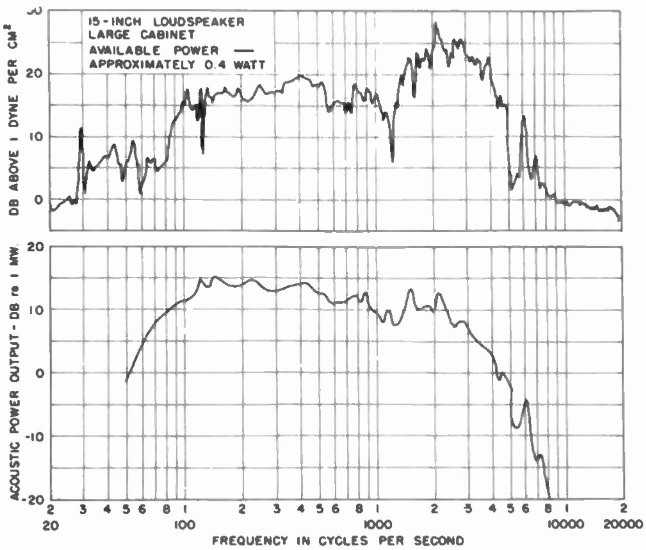


Fig. 8
(Top) Free-field response on axis of a 15-inch speaker in large cabinet at a distance of 9 feet. (Bottom) Acoustic power output of same speaker as measured in the reverberation room.

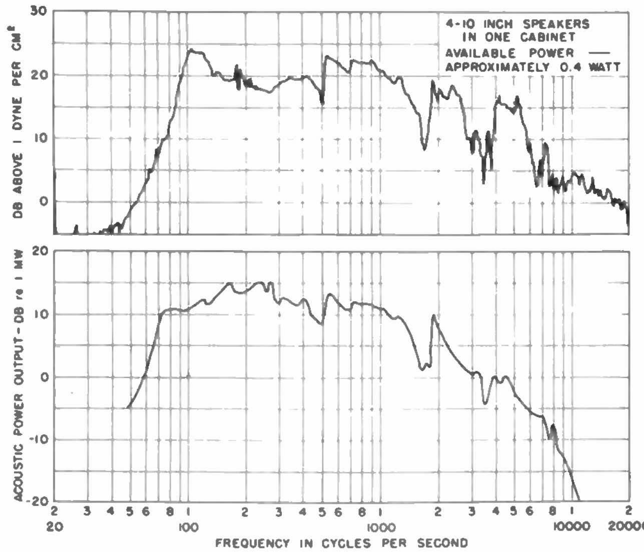


Fig. 9
(Top) Free-field response at a point midway between the axes of four 10-inch speakers in a large cabinet at a distance of 9 feet. (Bottom) Acoustic power output of same speakers as measured in the reverberation room.

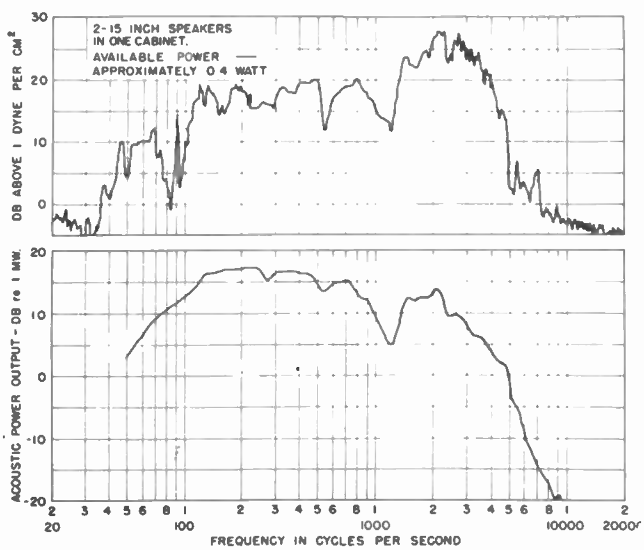


Fig. 10

(Top) Free-field response at a point midway between the axes of two 15-inch speakers in a large cabinet at a distance of 9 feet. (Bottom) Acoustic power output of same speakers as measured in the reverberation room.

MAGNETIC TAPE RECORDING DEMAGNETIZATION FOR SIMPLE CYCLIC EXCITATIONS*

O. William Muckenhirn
University of Minnesota
Minneapolis 14, Minnesota

The resultant active remanent magnetic flux associated with an element of magnetic tape upon completion of the recording process depends upon the entire previous magnetic history of the element. If the tape has been "erased" by a good alternating current erasing device before recording occurs, then, for engineering purposes, the tape may be considered as "demagnetized" and its magnetic history considered as beginning from this demagnetized state. Since the intelligence is transferred from electrical form to magnetic form by means of the recording head and its associated air gap, it follows that the magnetic field distribution of the air gap plays a predominant role in the magnetic history of the tape during recording.

Some effects of the air gap field distribution for longitudinal recording are presented in the series of oscillograms of figure 1 which shows the remanent flux for different recording head cyclic excitations. For the purposes of this paper, one cycle of excitation is defined by the operation of increasing the recording head current from zero to some arbitrary value in one direction and returning it to zero.

The oscillograms were obtained as follows. With the unmagnetized tape in a recording position but stationary with respect to the recording head, the recording head current was varied manually from zero through a cyclic pattern as specified for each oscillogram. In this manner the section of tape in the effective magnetic air gap region experienced a magnetic excitation in accordance with the gap field distribution and the cyclic excitation of the recording head. The remanent magnetic flux of this tape section was then measured by passing the tape across a playback head in the usual playback fashion. The output signal of the playback head was first amplified, then integrated in order to obtain a proportional to the remanent flux, and finally applied to a Tektronix Type 511-A oscilloscope on which the oscillograms were photographed. Since no changes in adjustments were made during the measurements or photography, the oscillograms show relative magnitudes. For further detail on equipment and procedure, reference is made to "Recording Demagnetization in Magnetic Tape Recording" Proc. of I.R.E., August 1951, pp. 891-97.

*Manuscript received October 8, 1952.

Figure 1(a) shows the remanent flux produced by increasing the recording head current from zero to + 3.0 milliamperes and returning to zero. The shape of the trace results from the air gap field distribution for this single cycle of excitation together with the non-linear saturation and hysteretic properties of the tape. In each of the remaining oscillograms two cycles of excitation were applied; the first to a maximum positive value of 3.0 milliamperes and the second in the reverse or negative direction to successively larger values. The peak values for the second cycle in figures 1 (b), (c), (d), (e) and (f) are 1.2, 1.6, 1.8, 2.0 and 3.0 ma. respectively. In figures 1 (b), (c) and (d) the demagnetizing action of the second cycle is significant only at the center of the air gap where the exciting field is most intense. In the outer edges of the gap field the reverse excitation of the second cycle is not sufficiently large to produce any perceptable demagnetization. In figures 1 (e) and (f) however this is no longer the case with the result that the entire remanent flux appears in the reverse direction.

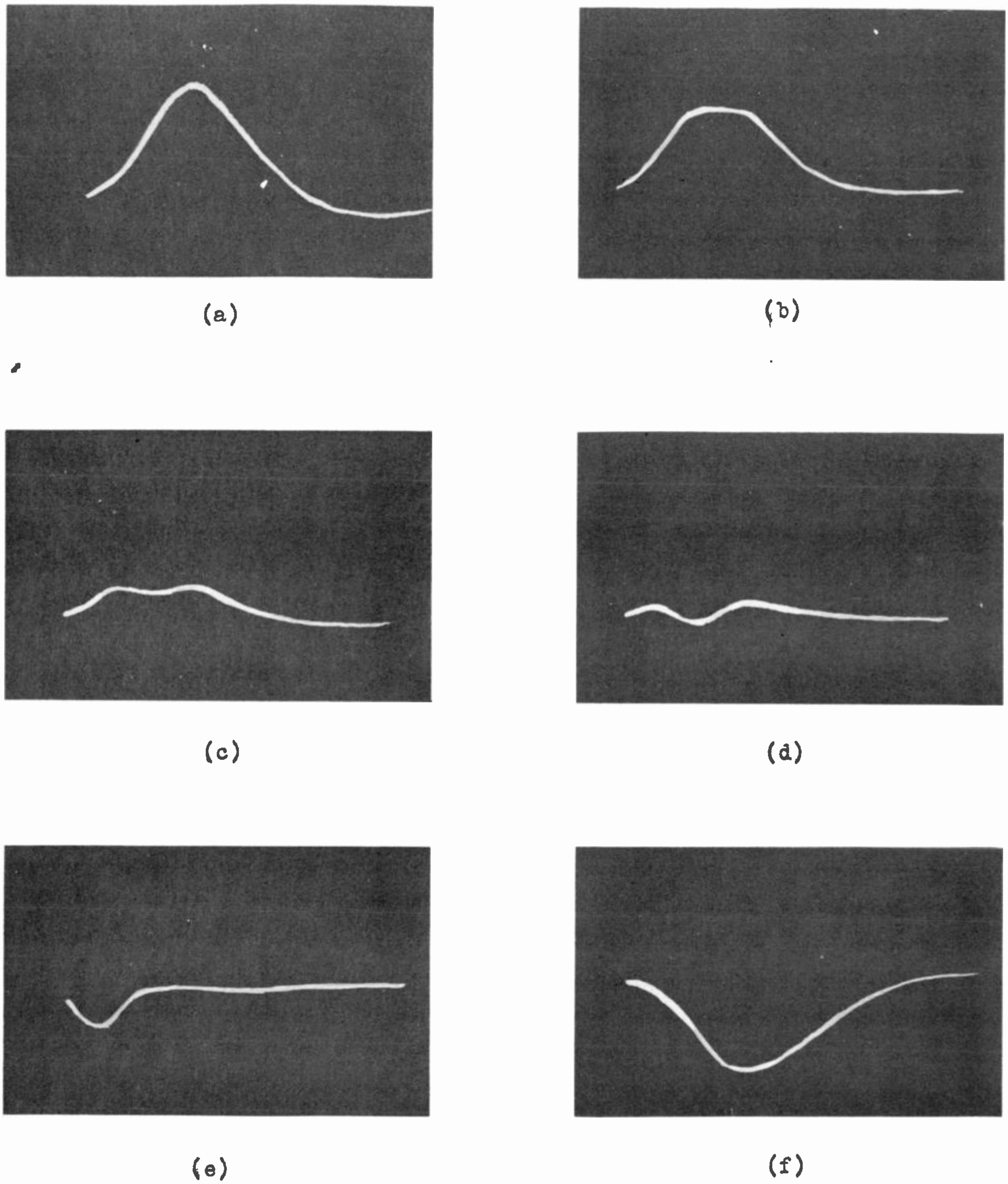


Figure 1. Oscillograms showing the remanent flux distribution for six different cyclic excitations of the recording head. The recording head current was varied as follows: (a) from 0 to + 3.0 to 0 ma.; (b) from 0 to + 3.0 to 0 to - 1.2 to 0 ma.; (c) from 0 to + 3.0 to - 1.6 to 0 ma.; (d) from 0 to + 3.0 to 0 to - 1.8 to 0 ma.; (e) from 0 to + 3.0 to 0 to - 2.0 to 0 ma.; and (f) from 0 to + 3.0 to 0 to - 3.0 to 0 ma.

INSTITUTIONAL LISTINGS (Continued)

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

ATLAS SOUND CORPORATION, 1443 - 39th Street, Brooklyn, New York
Loud Speakers, Public Address, Microphone Supports, Baffles

THE ASTATIC CORPORATION, Harbor and Jackson Streets, Conneaut, Ohio
Microphones, Pickups, TV-FM Boosters, Recording Heads, Acoustical Devices

AMPEX ELECTRIC CORPORATION, 934 Charter Street, Redwood City, California
Magnetic Tape Recorders for Audio and Test Data

AMPERITE COMPANY, INC., 561 Broadway, New York 12, New York
Ribbon Microphone, Dynamic Microphone, Kontak Microphone, Delay Relays

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

ALLIED RADIO, 833 West Jackson Blvd., Chicago 7, Illinois
Everything in Radio, Television, and Industrial Electronics

Charge for listing in six consecutive issues of the TRANSACTIONS - $25.00.
Application for listing may be made to the Secretary-Treasurer of the PGA,
Marvin Camras, Armour Research Foundation, Chicago 16, Illinois.

INSTITUTIONAL LISTINGS

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

UNIVERSITY LOUDSPEAKERS, INC., 80 South Kensico Avenue, White Plains, N.Y.
Manufacture of Public Address and High Fidelity Loudspeakers

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

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

SHURE BROTHERS, INC., 225 West Huron Street, Chicago 10, Illinois
Microphones, Pickups, Recording Heads, Acoustic Devices

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

McINTOSH LABORATORIES, INC., 320 Water Street, Binghamton, New York
Wide-Range Low-Distortion Audio Amplifiers

MAGNECORD, INC., 360 North Michigan Avenue, Chicago 1, Illinois
Special and Professional Magnetic Tape Recording Equipment

JAMES B. LANSING SOUND, INC., 2439 Fletcher Drive, Los Angeles 39, California
Loud Speakers and Transducers of all Types

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

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

THE DAVEN COMPANY, 191 Central Avenue, Newark 4, New Jersey
Attenuators, Potentiometers, Resistors, Rotary Switches, Test Equipment

CINEMA ENGINEERING COMPANY, 1510 West Verdugo Avenue, Burbank, California
Equalizers, Attenuators, Communication Equipment

THE BRUSH DEVELOPMENT COMPANY, 3405 Perkins Avenue, Cleveland 14, Ohio
Piezoelectric, Acoustic, Ultrasonic, and Recording Products; Instruments

BERLANT ASSOCIATES, 4917 West Jefferson Blvd., Los Angeles 16, California
Magnetic Tape Equipment for Audio and Instrumentation Recording

(Please see inside back cover for additional names)