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Professional Group on Audio

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

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SENSITIVITY OF MICROPHONES TO STRAY MAGNETIC FIELDS*

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One of the purposes of PGA is to acquaint all members with the special problems and techniques existing in various branches of Audio Technology. In this invited editorial, the author deals with an important problem to which insufficient attention has been given heretofore.-- Editorial Committee.

In addition to the usual attributes of a microphone which are commonly measured,such as sensitivity, response-frequency characteristics and directional properties, there are secondary attributes which may be of equal importance in specific applications. Most significant are the following: Sensitivity of the microphone to stray magnetic fields of low frequency, sensitivity to wind and sensitivity to mechanical shock. The purpose of the following discussion is to describe a possible standard method for evaluating one of these factors, namely: the sensitivity of microphones to stray magnetic fields of low frequency, such as are commonly referred to as hum fields.

Electrodynamic transducers, and all types of microphones in which a coupling transformer is included as a part of the microphone, are sensitive in some degree to hum fields. The evaluation of this sensitivity has always been of some importance for microphones used in Broadcast applications, and of late it has become increasingly important where microphones are used in Television programming, because of the number, strength, and closeness of the hum sources.

Hum fields may originate from any device operating on alternating current or from the incident wiring, and of course the strength of the source and the relative proximity to the microphone are factors which are of equal importance. The most likely sources of stray fields are Motors, Power transformers, Voltage regulating transformers, Fluorescent light-fixtures, Electric clocks, Wiring incident to high power lighting, Power supplies, and Amplifiers with self-contained power supplies. Some sources are a serious handicap because of their strength and others because of their closeness to the microphone.

Within the microphone and associated circuit, excluding the microphone preamplifier, there are several hum sensitive elements as follows: the microphone cable, the microphone transformer, the internal wiring, the moving conductor and compensating reactor if any.

The problem is two-fold. First, it is necessary to have a standard hum source which will allow various microphones to be compared with regard to their sensitivity to hum fields, and second, equipment is required to properly evaluate hum fields in microphone locations in order that the performance of a microphone may be predicted with reasonable accuracy.

Hum Excitation Equipment

A Large diameter coil of small cross section is best for this purpose because of the uniformity of the field close to the center and the resulting non-critical positioning of the microphone during testing. The field intensity at the center of the coil may be calculated from the following:

*Manuscript received October, 17, 1952.

(1) $H = \frac{.2\pi NI}{r}$

H = magnetizing force (Oersteds)
N = number of turns
I = current (Amperes)
r = radius (Centimeters)

For practical reasons, values of H. between 0.1 and 0.5 Oersted are most suitable. Values lower than 0.1 are likely to approximate ambient fields in magnitude and values above 0.5 pose difficulties from heating of the coil and acoustic noise. A 60 cycle power source is most convenient for the coil excitation because of its availability and the similarity of the resulting field to those usually encountered in practice.

The measurement is quite simple. The microphone is placed at the center of the coil and oriented until a maximum output is indicated on a voltmeter whose input impedance is high enough to assure that the open circuit voltage is being measured. If no such meter is available a voltage substitution method may be used. The sensitivity of the microphone to hum is then expressed as follows:

(1) $G_H = (20 \log_{10} \frac{E_H}{H} - 10 \log_{10} R_{MR}) -50$ db

where the reference values are 0.001 watt and a field of 0.0002 Oersteds.

The value of G_H is without practical significance because the criterion of the performance of the microphone is the signal-to-noise ratio. This is obtained as follows:

(2) $G_{MH} = (G_M - G_H)$ db

where G_H is as expressed above and
$*G_M = (20 \log_{10} \frac{E_p}{p} - 10 \log_{10} R_{MR}) -50$ db
(G_M = Microphone Sensitivity)

G_{MH} then reduces to:

(3) $G_{MH} = (20 \log_{10} \frac{E_p}{p} - 20 \log_{10} \frac{E_H}{H})$ db

In the above equations,
E_H = the open circuit hum voltage
H = field strength
E_p = the open circuit signal voltage
p = sound pressure dynes $/cm^2$
R_{MR} = microphone rating impedance

A suitable exciting coil is sketched in Figure 1.

Evaluation of Hum Fields

Hum fields may be very easily evaluated for a given location by means of an exploring coil and a voltage indicating device. The coil used should be air core so that the magnetic field is not disturbed. For such a coil:

(1) $E_s = \frac{Nd\phi}{dt} 10^{-8}$

where
N = number of turns
ϕ = flux through the coil
E_s = open circuit voltage due to the stray field

(2) $\phi = A_c B \sin \omega t$

where

B = flux density = H for air

A_c = Area of coil

(assuming a sinusoidal variation for B)

* See RTMA Standard SE-105 Microphones for Sound Equipment

(3) $\frac{d\phi}{dt} = A_c \omega B \cos \omega t$

dropping time function and substituting (3) in equation (1).

(4) $E_s = NA_c \omega B \times 10^{-8}$

(5) $B = H = \frac{E_s \times 10^8}{NA_c \omega}$

If E_s is measured in RMS volts B and H will also represent RMS values.

In order to obtain the maximum value of the field, readings are taken for three mutually perpendicular axes. Then,

(7) $H_t = \sqrt{H_x^2 + H_y^2 + H_z^2}$

Where H_x, H_y and H_z represent the field strength along the three axes.

H_t is then referred to a zero level of .0002 Oersteds.

(8) Field level $= 20 \log_{10} \frac{H_t}{.0002}$

Figure II shows the schematic arrangement of a field measuring set. The number of turns on the pickup coil will depend upon the sensitivity of the voltmeter and the strength of the fields to be measured. If an electronic voltmeter is used, the pickup coil must be kept far away enough from it to assure that the field due to the voltmeter is not contributing to the results.

Typical Results

Hum fields encountered are not entirely 60 cycles as can be seen from the analysis shown in Figure III, IV & V. Since the effectiveness of a given value of H is proportional to frequency for both the hum measuring coil and for most microphones tested, the effect of assuming the entire hum voltage measured to be 60 cycles results in a correct signal-to-noise voltage ratio. On the basis of correlation with actual listening to such a signal there may be some merit in considering rating the microphones on a 120 or 180 cycle field.

Hum Levels in Typical Locations

Voltage Regulating Transformer (10 ft.)	+ 26.7 db
Recording Studio	+ 16.2 db
Broadcast Studio	+ 15 db
Broadcast Control Room	+ 14 db
Fluorescent Fixture (80 watt) (36 inch distance)	+ 18 db

G_M & G_H for Microphones

	G_H	G_M
RCA - Type 77-D Polydirectional Microphone	- 139 db	-151 db
RCA - Type 44-BX Velocity Microphone	- 129	-149
RCA - Type BK-1A Pressure Microphone	- 116	-145
RCA - Type BK-4A Starmaker	-139	-153

From the above data the signal to noise ratio may be predicted for any given location if the sound pressure level and hum field levels are known. The following is an example of such a calculation.

G_M for type 77-D Microphone	- 151 db
Sound Pressure Level (assumed)	+ 94 db
Output Level from Microphone	- 57 db
G_H for Type 77-D Microphone	- 139 db
Hum Level in Typical Location	+ 16 db
Hum Level from Microphone	- 123 db
Signal-to-Hum ratio $G_{MH} = (G_M - G_H)$ or	+ 66 db

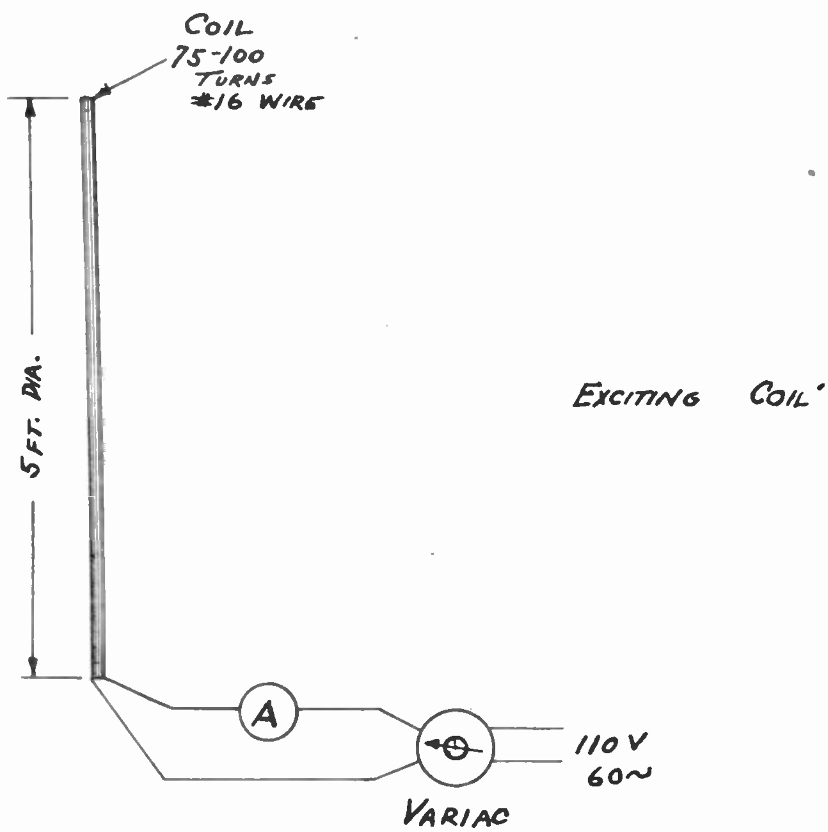
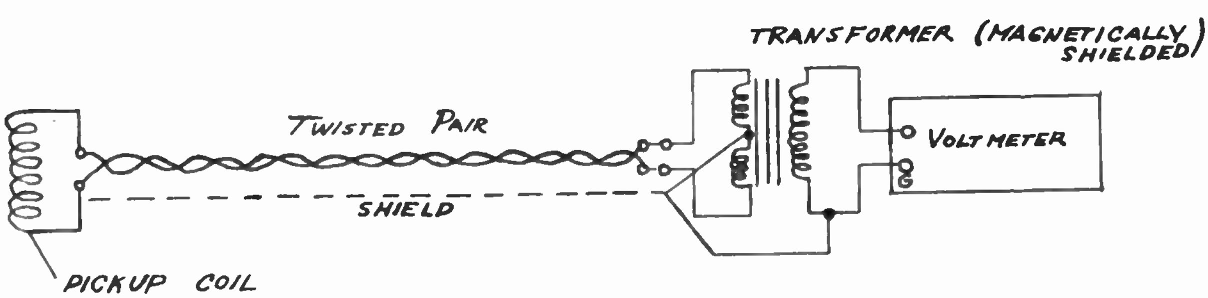


Fig. 1



FIELD MEASURING EQUIPMENT

Fig. 2

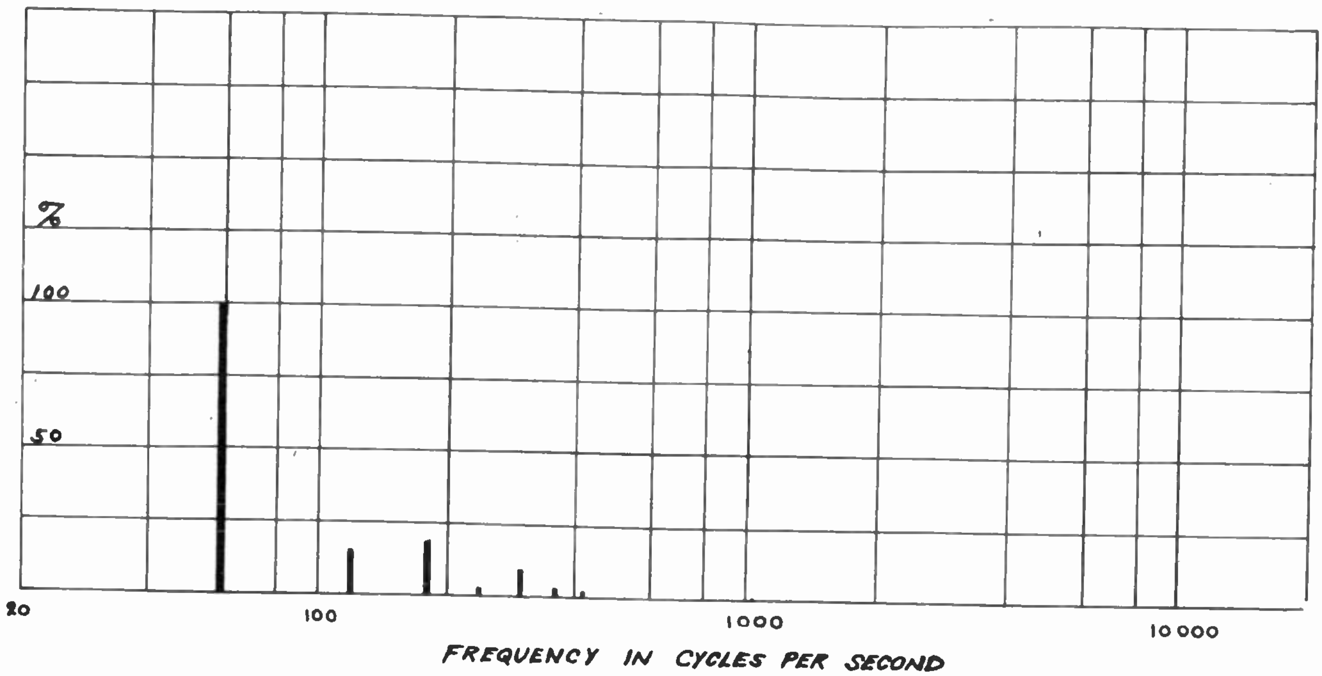


Fig. 3 - Field Analysis - Typical Broadcast Studio.

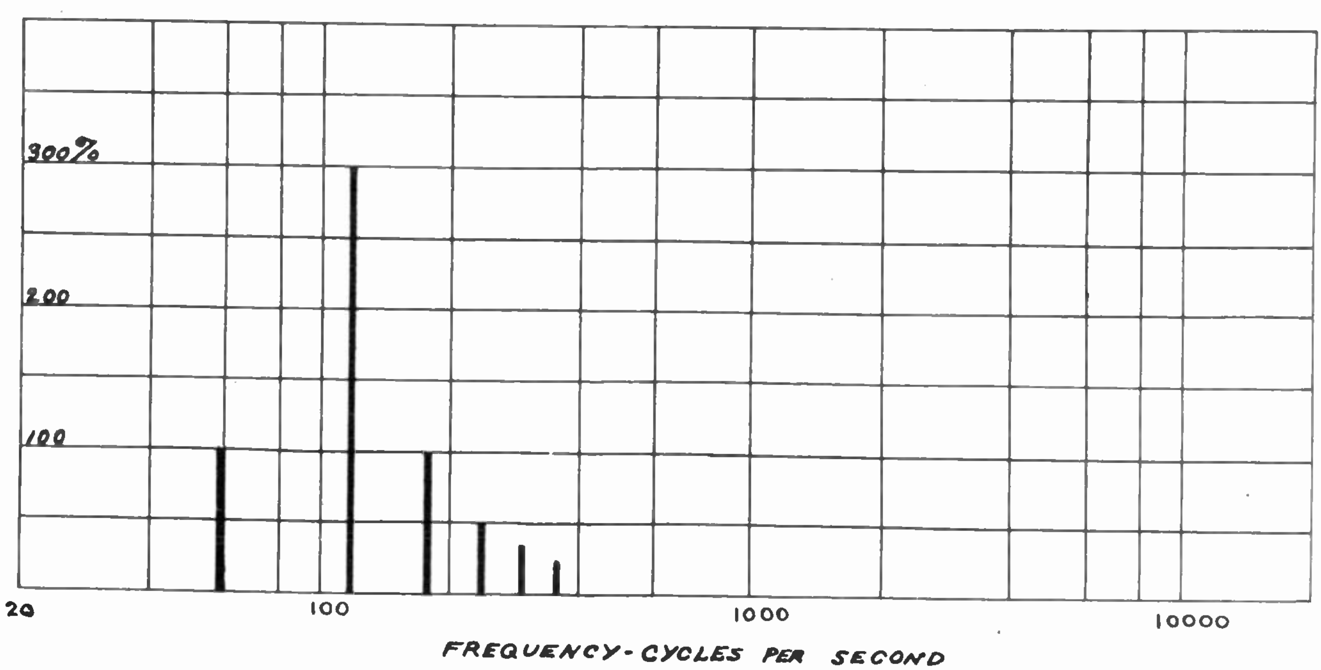


Fig. 4 - Field Analysis Near Power Amplifier.

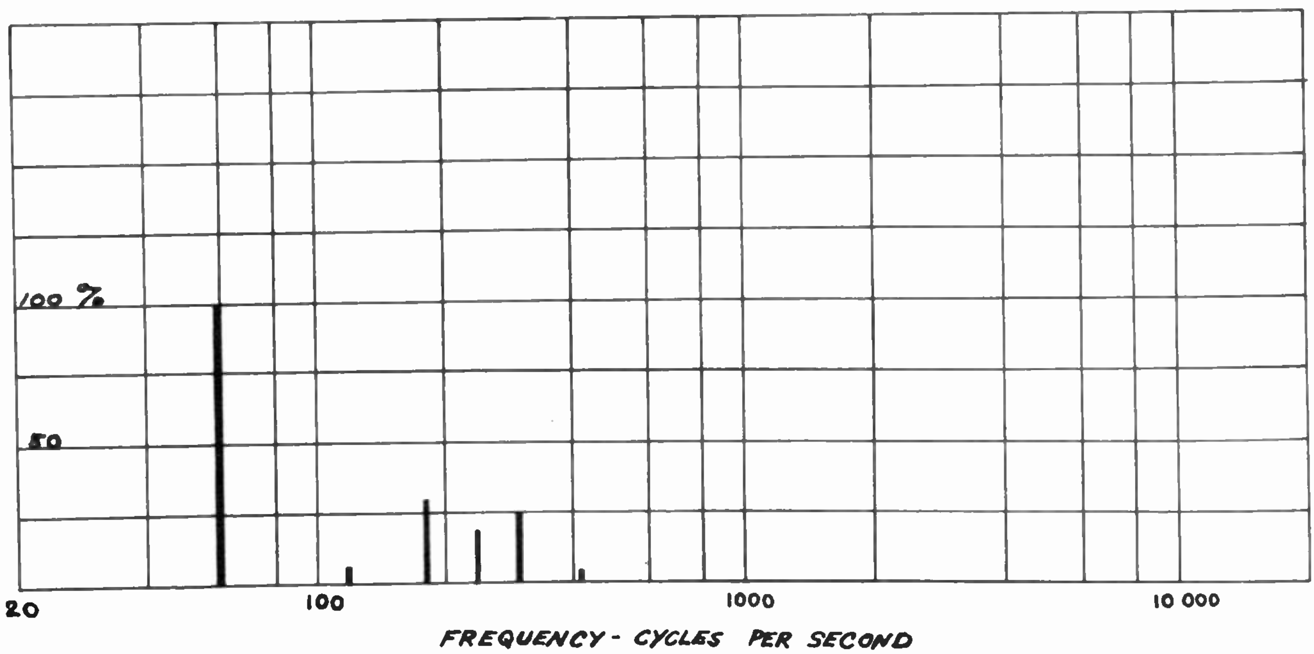


Fig. 5 - Field Analysis Near 80 Watt Fluorescent Fixture.

THE CHICAGO CHAPTER PGA

R. E. Troxel, Chairman
Chicago Chapter PGA
Shure Brothers, Inc., Chicago 10, Illinois

The Chicago Chapter of the IRE Professional Group on Audio is now entering its second year of existence. Members in the Chicago area total 111. In the interest of providing a service to its members, the local group has an interesting program outlined for the next few months. This program, coupled with the activity already past, will provide a very enjoyable year.

The first Technical Program of the 1952-53 year was presented on September 26. Mr. B. B. Bauer, Vice-President--Engineering, Shure Brothers, Inc. was the speaker. The meeting was composed of a technical paper and a demonstration regarding "Electroacoustic Analogies for the Radio Engineer".

Early in October the group members were given the opportunity of attending the several acoustical and audio papers given at the National Electronics Conference in Chicago. This Audio Session was arranged by the IRE Professional Group on Audio and the following five papers were presented:

"High Power Audio Amplifiers" by L. F. Deise and H. J. Morrison

"Analog for Loudspeaker Design" by J. J. Baruch and H. C. Lang

"A Ceramic Vibration Pickup" by E. V. Carlson

"Direct Measurement of the Efficiency of Loudspeakers by Use of a Reverberation Room" by H. C. Hardy, H. H. Hall and L. G. Ramer

"Interference Effects in Magnetic Recording Heads" by A. H. Mankin

All of the above papers have appeared in the TRANSACTIONS of the IRE-PGA.

December brings up a paper by Mr. O. C. Bixler, Chief Engineer, Magnecord, Inc. on "A Practical Binaural Recording System" -- a subject which has created much interest among the recording enthusiasts.

In February, Dan W. Martin of The Baldwin Company is tentatively scheduled to present a paper with a demonstration on the subject "Enhancement of Music by Reverberation". At this program the Chicago Acoustical and Audio Group will meet with the IRE-PGA Chapter. A very interesting meeting is anticipated.

The year's program will be completed in April when the Chicago PGA Chapter will meet with the Acoustical and Audio Group and enjoy a paper by Winston E. Kock, Director of Acoustics Research, Bell Telephone Laboratories on "Recent Work in Acoustics at Bell Telephone Laboratories".

AN ANALOGUE FOR USE IN LOUDSPEAKER DESIGN WORK*

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I. Introduction

There is, of course, nothing novel in the concept of utilizing electrical equivalent circuits in the design or analysis of mechanical systems. Any system whose behavior is described by a set of linear differential equations with constant coefficients of second degree or less can generally be represented by a common electrical analogue. Usually, the designer is satisfied with using the analytical techniques developed by the electrical engineer in the analysis of the equivalent circuit of his own problem. Many problems, however, wherein parametric variations are to be investigated, require some more facile method for detecting the change in a variable produced by a given parametric change.

It has often occurred to electrical engineers and others that in preference to performing the extensive analytical computations necessary to determine the effects of parametric change the circuit might be tested electrically. Such simple proportional analogues (usually called simulators) have been constructed and used for problems varying from the analysis of stress in an airplane wing to the transmission loss of a multiple-layer wall structure. Indeed, many laboratories throughout the country have available to them simulators of varying degrees of complexity which are used in everyday analysis of loudspeakers and loudspeaker enclosures. These analogues are used to study such effects as change in enclosure dimensions, change in cone stiffness or mass, and other easily controllable parameters.

II. Simple Equivalent Circuit

Fig. 1. shows the equivalent circuit of an ordinary loudspeaker mounted in an infinite baffle. The circuit contains several approximations, but is quite valid for the frequency region below that frequency where the perimeter of the diaphragm becomes equal to one wavelength. Actually, this is the main region of interest for loudspeaker analysis by means of equivalent circuits. Above this frequency, it is generally found that certain anomalous effects such as cone breakup and assymmetric modes prevent the representation of the loudspeaker as a simple lumped-constant circuit. The generator shown in the diagram represents the voltage applied to the loudspeaker terminals multiplied by a simple constant determined by the voice-coil-magnet structure and the electrical resistance. The first resistive element R_E represents the damping effect produced by both the internal generator resistance and the resistance of the loudspeaker voice coil. The inductor represents the acoustic mass of the diaphragm and voice coil, the condenser, the acoustic compliance of the suspension and spider. The next resistor represents the actual acoustical damping present at the points of suspension. The parallel representation of the radiation mass and resistance

*Presented at the National Electronics Conference, September 29, 1952, in Chicago, Illinois. Manuscript received October 24, 1952.

shown to the right of the diagram is valid, as stated before, for the low frequency region, and is even approximately valid for the region above this cutoff frequency.

In this equivalent circuit, U, the volume velocity, is analogous to current, while the pressure across a branch is analogous to voltage. Evidently then, the voltage across the inductive elements is proportional to the acceleration of the moving system, the pressure across resistive elements is proportional to the velocity of the system, and the pressure across a capacitive element is proportional to the displacement of the system. The sound pressure produced by such a loudspeaker at a distant point in a free field or in a room is proportional to square root of the power dissipated in the resistive radiation element.

Since this power is equal to $U_1{}^2R_{AS}$, the pressure at a distant point is given by $U_1\sqrt{R_{AS}}$. R_{AS}, however, is simply (for a piston in an infinite baffle) $1.4\rho/S_D$ where S_D is the area of the diaphragm. Hence the pressure is proportional to U_1/d_d where d_d is the diameter of the diaphragm. For a given diameter diaphragm, a relative pressure response curve may be obtained by simply measuring U_1.

III. The Vented Enclosure

When the loudspeaker of Fig. 1 is mounted in a vented enclosure, the circuit diagram shown in Fig. 2. is a useful one for analysis purposes. In this circuit, the compliance of the box is indicated by the capacitor C_{MB} while the port or ports and their mechanical components are indicated by the part of circuit to the right of the points marked XX. For the analysis of an unvented enclosure the circuit may be broken at the points XX.

Generally, in both vented and unvented enclosures, the representation of the coupling constant of the box by a simple capacitor is an unsatisfactory approximation. Beranek has shown in a 1950 paper before the N.E.C. that the actual reactance which the box presents to the rear side of the loudspeaker may be somewhat better represented by a series MC circuit than a simple capacitor as shown in Fig. 2. This series M is shown dotted in its proper position. No simple value is given for M_4 since, as illustrated by Beranek this value of the inductance depends largely on the ratio of loudspeaker size to the size of the box face. For large loudspeakers in small boxes the inductor becomes fairly unimportant while for small loudspeakers in large boxes the inductor may indeed contribute a prominent part to the impedance presented to the back of the loudspeaker diaphragm.

The sound pressure produced in a room by a vented enclosure is dependent no longer simply on U_1 but now on the vector difference between U_1/d_1 and U_2/d_2 where d_1 and d_2 are the diaphragm and port diameters respectively. This simple vector substraction presupposes of course that the distance between the port and loudspeaker is small compared to a wavelength in the frequency range of interest. For free-field analysis, the difference in path length from observor to the port or observor to the speaker must be small compared with a wavelength. Actually, no coupling is shown in Fig. 2. between the front side of the loudspeaker cone and the port. Various investigators are at present working with suitable coupling circuits to represent accurately this external coupling in the region of interest. Most of the work to date, however, has shown that, for conventional designs at least, the effect of this external coupling is small because of the high impedance of the internal cavity.

Fig. 3 represents diagrammatically an analogue which has been used with some success for analysing the behavior of loudspeakers mounted in vented enclosures. All the elements in the circuit are adjustable. The resistors representing the mechanical resistance of the speaker and the mechanical resistance of the port are used in order to secure voltage takeoffs proportional to U_1 and U_2 respectively. The isolating amplifiers labeled G operate into a typical summing circuit, the output of which is fed to an automaticfrequency response recorder. The frequency response recorder is geared to the oscillator and simple and rapid response determinations for the system represented by the analogue are easily obtainable. The purpose of the osciliscope across AA will be explained in a moment.

Available in conjunction with the computing equipment are many additional components which may be connected at various points in the circuit. A typical example of this is a simple RLC series circuit connected between the points shown as XX in Fig. 3. This circuit has been investigated as the analogue of a perforated resonant enclosure contained in the original vented enclosure. Such a device may be designed to reduce the resonant peak responsible for generating the typical closed box "boom" effect and to reduce the upper made peak in vented enclosures.

IV. Distortion Detection

As the use of the analogue was extended a severe limitation was observed, which made its usefulness in some problems fairly marginal. Vented enclosures, for example, have often achieved notoriety because of the ease with which they produce distortion at the lower frequencies. This distortion is generally attributable to two factors. Either the suspension behaves in a nonlinear fashion under large excursions or the voice coil wanders out of the linear region of the magnetic field. We are, of course, excluding the type of distortion produced by loose bolts, scraping wires, and a voice coil which is incorrectly centered. For the time being, we will consider only the simple type of nonlinear distortion produced by excessive cone excursion.

In general, the cone excursion may be represented by the voltage across the condenser C_{MS}. Idealy, the effect of this nonlinearity of the remainder of the circuit can be shown by introducing a nonlinear circuit element which limits the current U at a predetermined level of V_c. Actually such an arrangement is somewhat difficult to achieve in practice. One requires a condenser having a capacitance which may be reduced to a small value by an excess of voltage across it. In general, however, the major problem is to avoid distortion rather than to study its effect on the remainder of the circuit. A circumstance under which this is not the case will be described later.

The circuit shown in Fig. 4 has proven quite useful for rapidly analysing the effects of additional elements and meshes on the frequency at which distortion occurs and the distortion level of a loudspeaker. It is a simple clipping circuit whose output is shown on the osciliscope and which is connected across points AA. The wave-form on the 'scope takes on the characteristic flat top of a distorted displacement wave. Actually, of course, this circuit does not decrease the volume velocity at a given level of voltage but increases it. It nonetheless yields a very striking and simple method for detecting the presence of distortion in the eventual loudspeaker. Since the addition of the circuit shown in Fig. 4 the ordinary frequency analysis curves have been used to show up the point at which distortion will start. The transients introduced in the remainder of the circuit by the clipping action show up on a level recorder plot as sharp spikes, and indicate immediately the location of the distortion frequency. Adjustment of the potentiometers shown in Fig. 4 permits one to compensate the circuit for various measured displacement distortion characteristics of given loudspeakers.

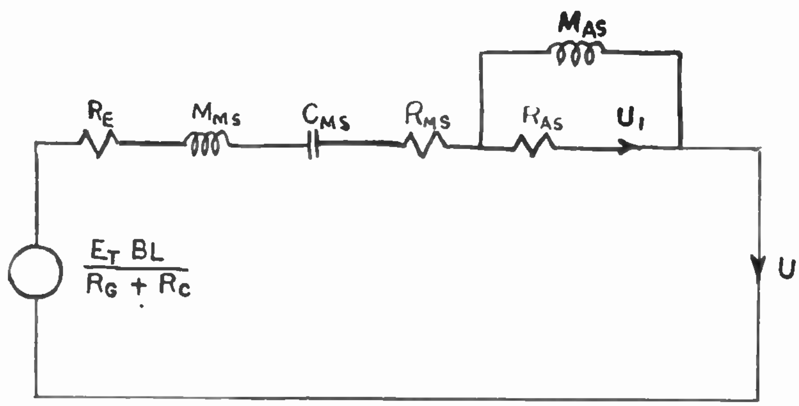
V. Distortion Analysis

There is, as mentioned earlier, a type of loudspeaker construction where the distortion produced by the loudspeaker must be reproduced in the circuit itself in order to secure an accurate knowledge of the behavior of the loudspeaker enclosure. A typical example of such a case is the loudspeaker shown in Fig. 5. Here the loudspeaker is enclosed in a box. The only point of egress for its radiation is through the small ports. The equivalent circuit of Fig. 3 may be used to represent this arrangement provided that the radiation elements in the left hand mesh (R_{AS} and M_{AS}) are eliminated and the size of the condenser C_{MS} is reduced to account for the added stiffness presented by the air cavity behind the speaker. In this case, the only volume velocity effective in producing a sound pressure at a distance is U_2. Note, however, that any distortion present in the radiation from the loudspeaker reaches the listener after having passed through an L-C filter. Thus the relative levels of the distortion components have been altered by the time they reach the observer. Evidently, in order to analyze such a system, it is necessary that the distortion components in U produced by both displacement and magnetic nonlinearities be reproduced faithfully in their effect on U_2. Thus, for such an enclosure-filter combination we require a nonlinear element which will reduce U as the voltage across C_{MS} reaches some critical value.

An amplifier having capacitive feedback around it presents an impedance, when properly connected, which looks like a capacitor whose value is linearly dependent on the gain of the amplifier. At present, a saturable amplifier is under construction at the laboratory for use in such a feedback circuit. The device, whose behavior is being studied as a bachelor's thesis, will it is hoped, present a capacitive impedance which is linear over most of its range but which drops very rapidly as the input voltage is increased. The substitution of such a feedback amplifier for C_{MS} will permit more accurate and more detailed analyses on the types of distortion produced by loudspeakers and the effects of the loudspeaker mounting system on the amount of this distortion perceivable to the listener. The amplifier is being designed for both symmetric and asymetric clipping to permit investigations of even as well as odd harmonic distortion.

The entire simulator described in this paper is constructed of toridal inductors, paper capacitors, and decade resistors. In order to improve the

Q of the inductors, and in order to achieve a sensible impedance level, the values of L, R, and C calculated using the CGS system for loudspeakers are multiplied by 10^4, 10^5 and 10^{-6} respectively. This transformation increases the impedance level of the circuit by a factor of 10^5, and decreases the time scale by a factor of 10. Thus the twenty cycle response of the loudspeaker is measured by feeding the equivalent circuit analogue with a two hundred cycle signal. Unless such time-scale and impedance-scale changes are made, the equipment becomes extremely bulky. To provide for more complicated circuits and more complex enclosures, the analogue contains eight variable inductors, twelve variable resistors, and ten variable capacitors. Should the needs of the art outstrip the facilities of the analogue computor, more parts may be easily added in the remaining space. Isolating amplifiers are included, to facilitate inspection of various voltages throughout the circuit. It is to be hoped that eventually transient analyses as well as frequency analyses will be performed with the computor's aid.



$$R_1 = \frac{B^2 L^2}{R_e + R_c}$$

Fig. 1 - Equivalent circuit of a loudspeaker mounted in an infinite baffle.

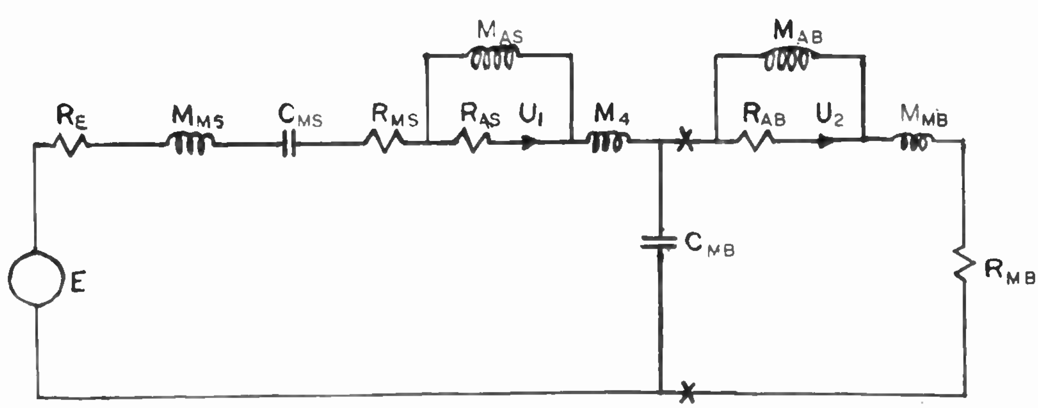


Fig. 2 - Equivalent circuit of a loudspeaker mounted in a vented enclosure.

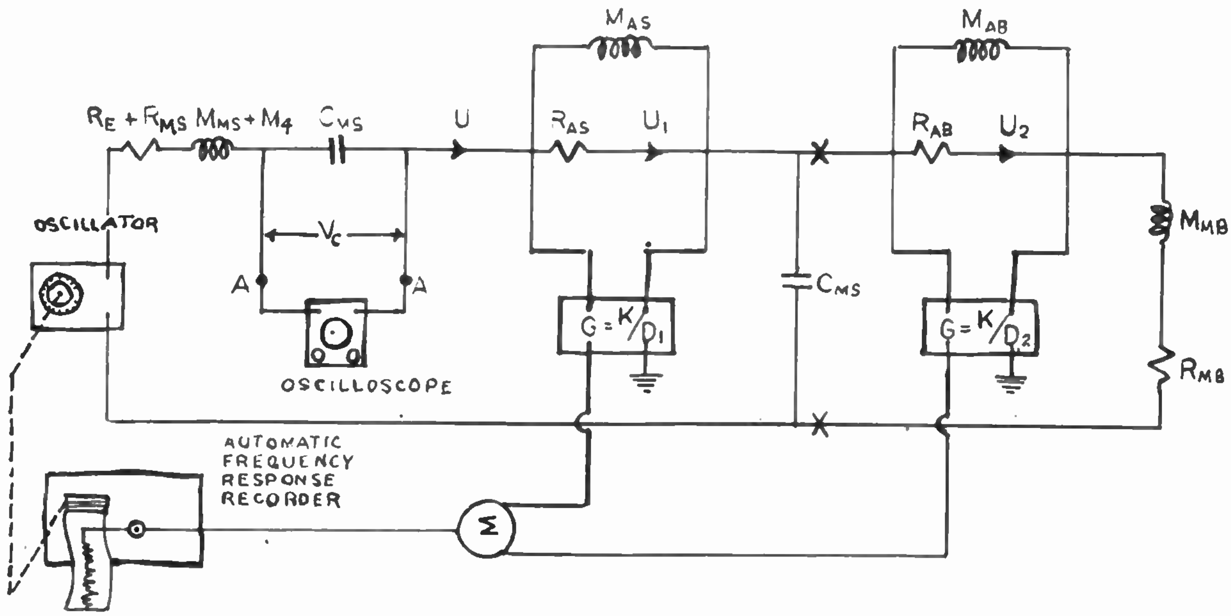


Fig. 3 - Schematic diagram of analogue computer.

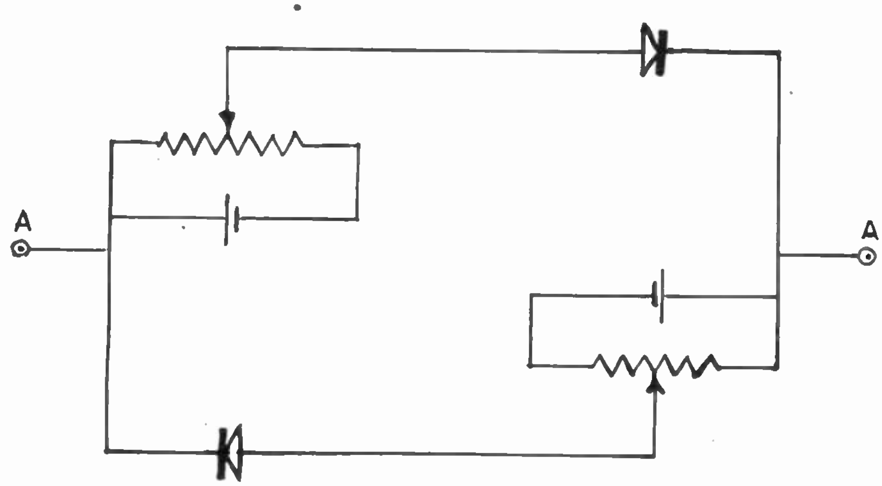


Fig. 4 - Circuit used for introducing amplitude distortion.

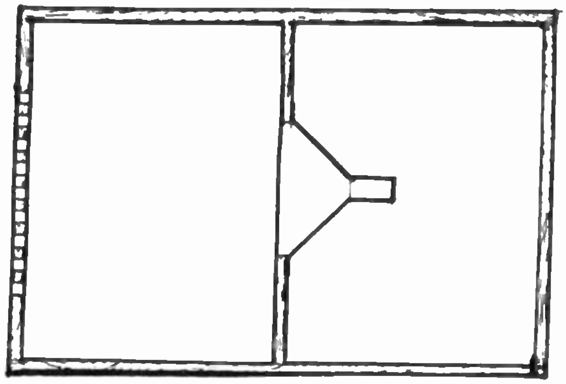


Fig. 5 - Sectional views of completely enclosed loudspeaker.

A PRACTICAL BINAURAL RECORDING SYSTEM*

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SUMMARY

A practical binaural recording unit was designed and manufactured to extend the present day high quality sound recording-reproducing equipment development so as to utilize some of the benefits inherent in a stereophonic system.

A review of theoretical factors involved in binaural sound recording and reproduction is presented, with a description of the technical equipment developed to provide a high quality binaural system consistent with reasonable overall equipment cost. Some novel problems and effects encountered in the development program, as well as experiences with binaural recording techinques in the fields of radio transmission, court room recording, and test instrumentation, are described.

BINAURAL HEARING - DESIGN OBJECTIVE

Binaural hearing ability has been granted to most living creatures in order that they may be well adapted to fitting into a three-dimensional world and to provide them with sufficient protective mechanisms to increase their chances of survival. Binaural hearing is a valuable sense which furnishes the listener with not only direction of sound origin and a measure of the distance to the source, but also a general perception of the environmental surroundings in which the listener finds himself when the sound is emitted.

Present day high quality sound systems are nearing the peak of monaural perfection. It was believed that considerable public interest could be developed if an enhanced method of listening were provided. It was therefore decided to produce a simple, practical, low-cost commercial binaural recording-reproducing system.

THE MECHANISM OF AUDITORY PERSPECTIVE

It is to be noted that a human's ears and brain constitutes a directional computing system based upon their phase and amplitude sensitivity. This dual system has a sensitivity versus frequency cross-over area determined as follows: the average low frequency ear phase sensitivity is from the lowest frequency of sound detection up to approximately 800 to 1000 cycles per second. The sensitive transducers of each ear furnishes data to the brain computing system, which allows a perception of sound origin or directivity by binaural phase comparison over this frequency range. Above this frequency range, the wave lengths of sound are so short that the ear phase discrimination falls off rapidly and no comparison above 1000 cycles is possible. The amplitude of a low frequency sound below 800 to 1000 cycles per second is nearly equal at both ears since these long wave lengths readily pass around the cranium obstruction without amplitude loss. This means that

*Presented at the Audio Session of the IRE Convention in Long Beach, California on August 29, 1952. Manuscript received December 22, 1952.

the effective ear-mental sensitivity to amplitude differential falls off rapidly below 1000 cycles per second. In the region above 800 to 1000 cycles per second, the amplitude sensitivity of each ear becomes important. The physics of sound propagation is such that frequencies above 1000 cycles are attenuated by passing around the cranium. The ears' amplitude sensitivity range continues up to the highest frequency perception limit, which allows directional computation by amplitude comparison between the two ears. The amplitude sensitivity range of the ear is of course defined by the dynamic volume range shown in the standard Fletcher-Munson hearing curves, modified by the room masking level (Reference #1).

The overall computing system for direction is therefore quite effective from the lowest frequency to the highest frequency of sound perception. Figure 1 shows diagrammatically that high frequencies strike the nearest ear to the sound source with full amplitude, but effectively pass by the cranial obstruction without striking the far ear. There is a loss in the order of 30db at 10 kilocycles between the near and the far ear for sounds originating on the axis of the ears. Low frequency sound striking the near ear passes readily around the cranial obstruction so that there is only a 3db loss between the near ear and the far ear.

It may be shown that the portion of the normal auditory perspective due to phase sensitivity, is related to the distance between the human ears. Under the conditions that an observer has a between the ears distance of 6.78" and with a speed of sound in air of 1130 ft/sec., then the maximum frequency f, that the ears may compare phase on, without redundancy, is equal to a half wave length λ ; which is the distance between the ears.

$$\text{If } \lambda = 6.78 \times 2/12, \text{ then } f = \frac{1130}{\lambda} =$$

$$\frac{1130}{6.78} \times 2/12 = 1000 \text{ cycles per second}$$

This then is the maximum possible frequency for binaural phase detection. Average sound sources possess frequencies both below and above the cross-over range of 800 to 1000 cycles. The listener therefore locates the direction of such a combination sound source by both phase and amplitude methods. This enables one to derive a very accurate angular localization.

Two methods are also available for measuring the distance from the listener to the sound source. Since the phase shift of sound of a given frequency is a function of both angular location as well as distance to the binaural listener, a measure of the distance to the sound source is available to the mental computer when the source to listener distance is small. Experience in listening results in an ability to measure the distance to a sound source; the mental computing mechanism calculates the distance by comparing the ratios of the amplitude and the phase of the direct sound to reverberant sound reaching the ears. This is, of course, dependent upon room acoustics. Experience in a given room, or "awareness" of its characteristics allows the ratio to be measured automatically.

LISTENING METHODS

The human being, who has been given this wonderful sense of binaural hearing, achieves a false auditory perspective when monaural recordings are played back. Re-creation of an original sound field varying in amplitude and phase is accomplished by means of stereophonic recording and reproduction. This term "stereophonic" was developed to describe a system for re-creating at any plane in space, the passage of an original sound field with both correct amplitude and phase relationships. Many years of research and experimentation have shown that a very satisfactory re-creation of sound could be achieved through the use of a recording system utilizing three microphones and three sound tracks, which is then reproduced at a later time through a sound system terminating in three loud speakers. The mechanisms of such a system are currently well known, however, because of costs and technical complexities, they are not in present use to any great commercial extent. True stereo sound is quite difficult in technical achievement, and is comparatively costly in the forms which have been demonstrated to the public to date. Both stereo and binaurally reproduced sound have the characteristic of apparently placing the listener in the original sound field. It is this psychological effect which contributes so much to the realism of the reproduced sound.

Earphone sound possesses more binaural enhancement and apparent fidelity than does binaurally reproduced loud speaker sound due to the inherent sound isolation given to the ears by earphones. Experiments and tests with binaural loud speaker reproduction were carried out, and while it was found that some stereo deterioration occurred, it was possible through careful direction and placement of the loud speakers to achieve a considerable improvement over monaural reproduction. Excellent loud speaker listening can be accomplished by spacing the two reproducer loud speakers and the listener at the corners of an isosceles triangle, of perhaps 8 to 12 feet on a side. A somewhat larger listening audience may be accomodated by spacing with as much as 25 foot per triangle side; however, as the distance is increased, the effective sound directivity decreases, so that binaural listening suffers due to the lack of isolation between the sound channels.

Listening tests with such a system brought to light an interesting phenomenon due to the mental correlation of the strictly random background noise pulses ("hiss") present in the separate sound channels. The result of the mental correlation was that focused listening attention was directed toward apparent spacially localized noise sources. This resulted in the subjective raising of the background noise level and a coarsening of the apparent noise source. The random nature of the original white noise was effectively disturbed by false phase and amplitude coincidence, as correlated by the brain to produce apparently localized sources of noise.

SPACIAL DISTORTION

Spacial distortion, a form of distortion which is rarely mentioned and seldom commented upon, is the spacial distortion of a spacially disposed multiple sound source. Focused listening attention may be directed toward spacially disposed sources so as to sort out sound as much as 13db below the general noise background so as to secure intelligibility. Listening tests on monaurally reproduced sound shows that it is difficult to concentrate to the point of detecting the intelligence content of a single sound present in a generally noisy background.

A SIMPLE BINAURAL SYSTEM

Figure 2 shows a binaural sound system complete from sound origination in microphones, through amplifier including pre-equalizer, recorder, tape, playback amplifier including post equalization, loud speaker system, and listener. Microphone placement techniques for binaural recording vary somewhat at the present due to notable divergencies of opinions. This is in part caused by characteristic variations of both microphones and loud speakers, as well as the effect of the acoustics of both the pickup point as well as the reproduction room.

The simplest binaural system for obtaining correct results should obviously use microphones whose pickup patterns approach that of a normal, human ear. This can be accomplished by using so-called non-directional or semi-directional microphones, which actually do have some directional pattern, and mounting them pointing slightly outward on either side of an acoustic septum, which represents the cranium obstruction. This microphone system is the equivalent of a theoretical binaural listener and faithfully picks up sound for binaural storage in a tape recording system; the signal reaching each microphone being stored separately. The techniques of modern tape recording are such that a number of commercial machines are currently using half the area of 1/4" plastic base magnetic tape for high quality speech and music recording. This means that each half of the tape can be used for storing the information derived from a single microphone and amplifier channel, which allows a true binaural recording system to be developed.

A dual amplifier system was built to record or reproduce using this mechanism. When the recorded material is rewound and played back through the two amplifiers, which are also maintained as entirely separate channels, and the outputs are fed independently to speakers, a complete binaural recording and reproduction system is obtained. A second method of audible reproduction is provided for by connecting each reproducing channel to a single earphone in an especially designed binaural headphone set.

THE TAPE TRANSPORT

The development of a binaural tape transport from a standard Magnecord PT63-A was possible, because this basic unit incorporates an assembly of three magnetic heads. The tape passes in succession over an erase head, recording head, and a tape monitor head. The full track recording and tape monitor heads were simply replaced with half-track recording heads arranged to record on opposite edges of the tape. This, of course, sacrificed the facility of monitoring from the tape while recording. However, this was not thought to be a serious loss because of the reliable nature of magnetic recording. It was possible to add the 60kc recording bias circuit of the second recording head in series with the erase and the first recording head without materially changing the circuit impedance and upsetting the 60kc bias oscillator circuits. This was allowable because the main impedance of this series circuit is the erase head, the recording bias winding being a very small impedance. Using this arrangement, it was therefore only necessary to supply proper pole pieces and to reconnect the tape transport's internal wiring in order to accomodate the second recording channel. The existing plug and receptacle arrangements were such as to automatically maintain channel identity between tape transport and amplifier units. The half-track recording pole pieces were made by cutting away slightly more than half of the standard Mumetal pole pieces and soldering into place an equivalent size brass insert to fully support the tape.

THE AMPLIFIER UNIT

A new portable dual record-reproduce amplifier unit was designed, incorporating the characteristics of existing recording equipment amplifiers, except that miniaturization techniques were employed in order to package this unit in the same space as previously occupied by a single channel amplifier. The special features of this dual amplifier unit include individual illuminated VU meters for each channel, individual channel gain controls, and a single (dual) overall master gain control which simultaneously controls the gain of both channels, a special binaural headphone receptacle, and a single panel mounted monitor speaker with a unique volume control. This volume control is so arranged that the speaker is off when the control is at its center position. Maximum loud speaker volume for the one channel is obtained with clockwise control rotation, and maximum volume for the other channel is obtained with counter-clockwise rotation.

The individual amplifier tube lineup consists of two 5879's followed by a dual triode 12AX7, the second half of which is an inverter driving a pair of push-pull 6AQ5 tubes. A multiple section shielded selector-switch switches the equalization and input-output connections simultaneously for both amplifiers in order to change the unit function from record to playback. A full wave selenium rectifier provides DC filament power for the input tubes. The two independent 10 watt amplifier outputs are provided with nominal impedances of 4 and 16 ohms, as well as a 600 ohm balanced connection at plus 4dbm. Pre-and-post equalization is used in order to yield a flat response from 50 cycles to 15kc, plus or minus 2 db at 15 inches per second tape speed. An optional equalization facility is provided to allow operation at 7-1/2 inches per second with a 50 cycle to 7.5kc plus or minus 2db response. A signal to noise ratio of 50db may be achieved with this equipment. A 35db signal to cross-talk ratio between channels caused by magnetic coupling occurs at 50 cycles, but drops with frequency increase until it is below the tape noise at around 100 cycles per second.

Since accurate binaural localization depends to a considerable degree upon amplitude comparisons, a means of electronic balancing of both the recording and reproducing circuits is provided through the use of a calibration button which introduces a 60cps signal simultaneously into the first stage of each of the amplifiers. The individual channel gain controls may then be adjusted to yield equal VU meter readings. When recorded, this calibration signal allows balancing of the playback amplifier gains in a similar manner.

EQUIPMENT APPLICATIONS

The first binaural experimental units were built for an automobile manufacturer for laboratory and field use. Additional units were built for demonstration use to acquaint the public with this new medium and to "feel-out" the possible market applications of the equipment. In the first public demonstrations of binaurally reproduced music and sound, it was not possible to present the technical usefulness of this device and to poll the research workers properly since such a large group of music lovers invariably gathered so as to completely prevent adequate demonstration of the equipment to technical personnel. It has therefore been necessary to carry out considerable specialized work investigating the different fields of application. These endeavors are described in the following paragraphs:

POLICE WORK

Police and Secret Service Departments have begun to make use of binaural recording techniques for surreptitious recording since this method overcomes all accepted methods of masking voice intelligibility. A monaural recording system cannot overcome background noises, the running of water, the turning up of radio volume, etc. A binaural system permits spacial location of the masking source and allows focused listening attention to be directed to the intelligence source so as to achieve intelligibility under all of these conditions. It has been found that it is possible to obtain complete transcripts from recordings made under conditions where previously nothing useful could be obtained.

COURT ROOM REPORTING

Court reporting is an exceedingly important application of binaural recording equipment which assures accurate court records, including making a positive identification of persons in the court room. A study of monaural court reporting has been carried out by Mr. Ray Hurst (Reference # 6), who has clearly shown that court records are often at variance with what actually transpires because court clerks are unable to follow testimony fast enough to accurately transcribe it as it is presented. Often the clerk may hear something wrong and can also be guilty of making obvious mistakes. On one occasion, to our knowledge, it has been necessary to reverse a written court record which occurred due to a stenographic error. We have followed up this original monaural recording work by making binaural recordings in the State of Wisconsin Circuit Courts. The results achieved in recording actual court room procedures were more than gratifying. By properly disposing the microphones, excellent binaural recordings were made, which resulted in 100% intelligibility on playback, even when as many as three people were talking at once., e.g., the State's Attorney, the defense attorney, and the regular court reporter, who was inquiring concerning something which he had not heard well.

Experimental court room set-ups for the microphone locations resulted in placing one slightly in front of the intersection of the Bench and the witness's chair, and the other at the corner of the counsel table about 15 feet from the first microphone. This located the judge and witness near one microphone, while the two attorneys' positions were relatively close to the second microphone. The usual court room distractions went on throughout the recordings; extraneous noises included the building radiator noises, rattling of papers by court reporter and judge, coughing of people throughout the room, etc. The excellent results obtained have caused other courts to begin preliminary studies of this medium.

RECORDING OF HEARINGS

Another use of the binaural recording method is found in recording the proceedings of large Commissions where discussions may be originated from any position throughout a large group of people. Tests were made in hearings before the Public Service Commission of Wisconsin, in a room,perhaps, 25 x 50 feet in size. Speeches came from all parts of the room. The microphones were placed in one end of the room about 15 feet apart, near to the presiding member of the Commission. It was found in these tests that complete understanding was had during playback of every speech made throughout the Appearances with the exception of speeches which camefrom the court reporter. This man did not speak plainly, and could not be understood in the room at the time when the original recordings were made. All speeches, even those coming from the rear of the room, showed a high degree of intelligibility, which was not found when only monaural reproduction of the recordings was made.

SYMPHONIC AND ORCHESTRAL RECORDINGS

Excellent recordings have been made of large University bands; experimental recordings have been made of the University of Wisconsin band as well as of the University of Illinois band. An interesting occurance took place during a reproduction of one of the recordings which had just been completed in a large music hall. One of the caretakers approached, slipped on a headphone set and with a very startled look, wheeled around and stared at the empty stage. The "three-dimensional" listening effect had fooled him into believing that the band was still on the stage. Such involuntary reactions are a tribute to the effectiveness of binaural recording. From a music lover's standpoint, the improvement in realism with binaurally reproduced music is the most important improvement factor of such a system. Indications to date are that the additional complexity and cost of such a system as is described here are very acceptable in view of the results obtained.

RADIO BROADCASTING

In order to test public acceptance of this "new medium", experiments have been conducted utilizing simultaneous broadcasts over radio stations having both AM and FM facilities. Spot announcements and newspaper advertisements giving careful instructions to listeners were provided throughout the days preceding the broadcast concerning the fact that separate microphones were to be fed into the AM and FM channels so as to achieve a binaural effect. Instructions were issued telling the listener how to set-up his AM and FM receivers for the best listening effect. Very gratifying results have been achieved on radio stations WGN and WGNB, in Chicago. Sufficient interest in this "new medium" has been stirred up so that AM-FM binaural broadcasts have also been carried on experimentally by WJR in Detroit, WGAR in Cleveland, WQXR in New York, and elsewhere. It has been found that considerable listening enhancement in the home may be had using a quality FM receiver for one channel and a small "kitchen variety" AC-DC set for the AM channel. The reaction of music lovers to the improvement has been astounding. The only additional facilities required by a radio station for such a binaural broadcast are the use of a second microphone and separate amplifiers for the separate channel inputs to their respective AM and FM facilities. Since most radio stations have these facilities already, no additional expenditure is required for a direct binaural broadcast. Delayed broadcasts, and "canned" music of course may be handled directly by the binaural recorder described in this article.

In tests conducted in conjunction with a broadcast of the Chicago Philharmonic Orchestra, it was found that the introduction of individual reverberation chambers into the separate binaural channels detracted considerably from the quality of the transmission, to the point of almost completely destroying the binaural effect. The directorial staff assigned to production were in the habit of using reverberation for enhancement of their regular monaural symphony broadcasts. However, experimentation demonstrated that there was no question but that the binaural effect provided sufficient additional enhancement to far make up for the loss of the questionable reverberant enhancement.

EDUCATIONAL PROGRAMS

Demonstrations of the usefulness of this medium in audio-visual education programs have already disclosed that the third dimensional realism is of considerable assistance in the critical analysis associated with speech classes, band and choir practicing, dramatics, etc. The auditory "liveness" inherent in stereo localization is a major step forward in this field. Several well known choirs and orchestras now use this system as an accepted rehearsal tool.

RESEARCH APPLICATIONS

For the majority of commercial applications, binaural's usefulness lies in the information identifying field where information is normally obscured or masked by a multiple sound background as reproduced by a monaural system. One of the current field uses of the binaural recording system is by a prominent automobile manufacturer, who has standardized experiments with the equipment to assist in judging noise factors in newly designed automobiles. Tape editing allows ready A-B testing, so as to allow judging between automobiles with a critical view toward improvement in design as changes are made.

Considerable research (Reference # 7) has been carried out to improve the "muddy" sounding emphasized bass that results from monaural recordings of engine noises. Deisel engines as well as conventional gasoline automobile engines,both indoors and out-of-doors, were tested yielding the same un-realistic sounding recordings with monaural systems. Road rumble recorded during automotive road testing with a monaural system seemed to come from all directions thus effectively obscuring the test information. Binaural recording overcomes both these effects and through the realism and assignment of sound direction allows evaluation testing to be carried out.

A non-binaural laboratory use of the equipment is dual channel recording of simultaneous information and the recording of separate commentary during a single channel information recording.

CONCLUSION

In the short time since the introduction of the commercial binaural recorder, it has already proven its usefulness. The simplicity of the system developed no doubt has contributed to its wide spread acceptance.

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5. J. P. Manfield, A. W. Colledge, and R. T. Friebus, "Pickup for Sound Motion Pictures (including Stereophonic)", Journal of the Society of Motion Picture Engineers, p. 666, June, 1938.
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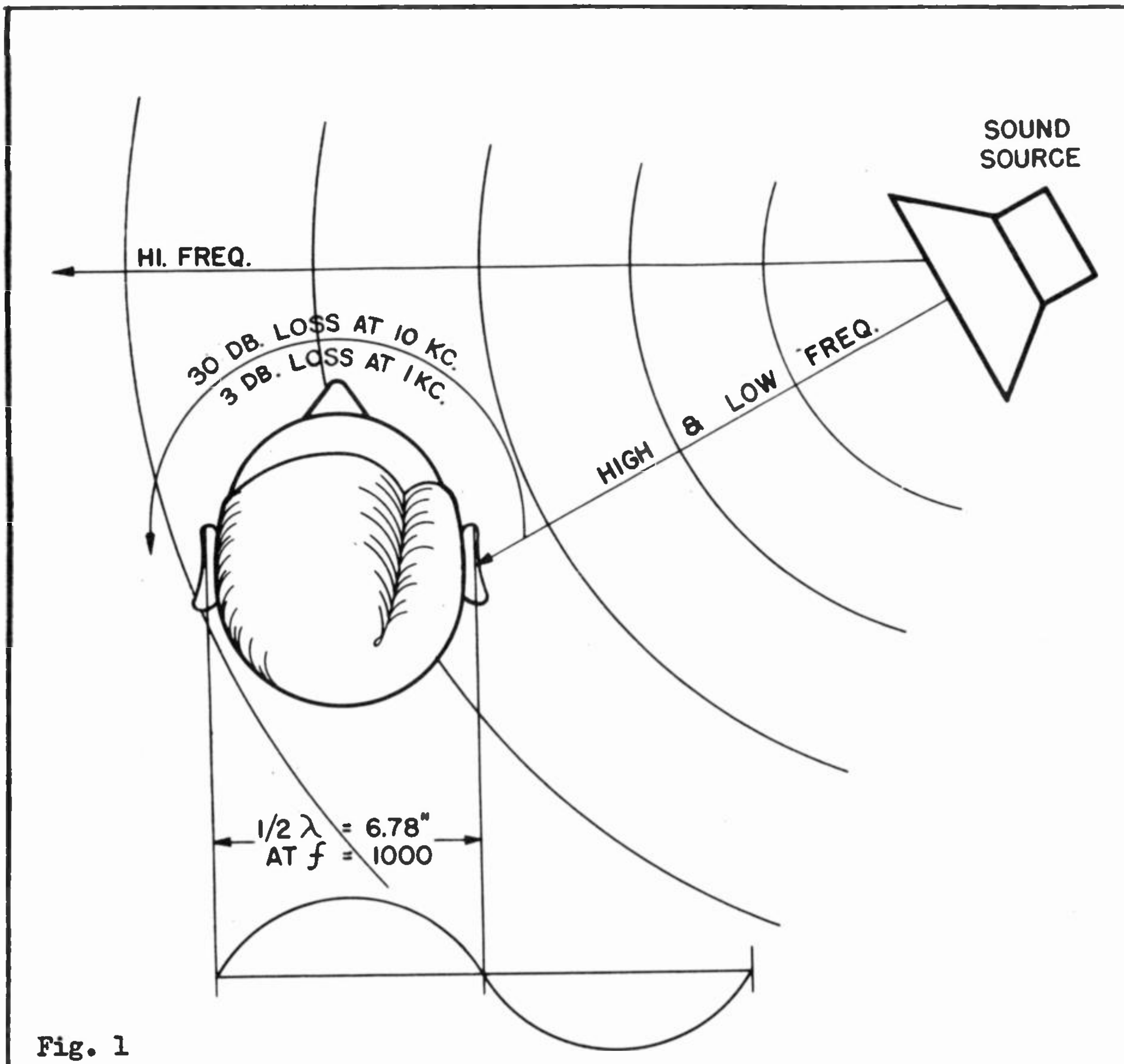

Fig. 1

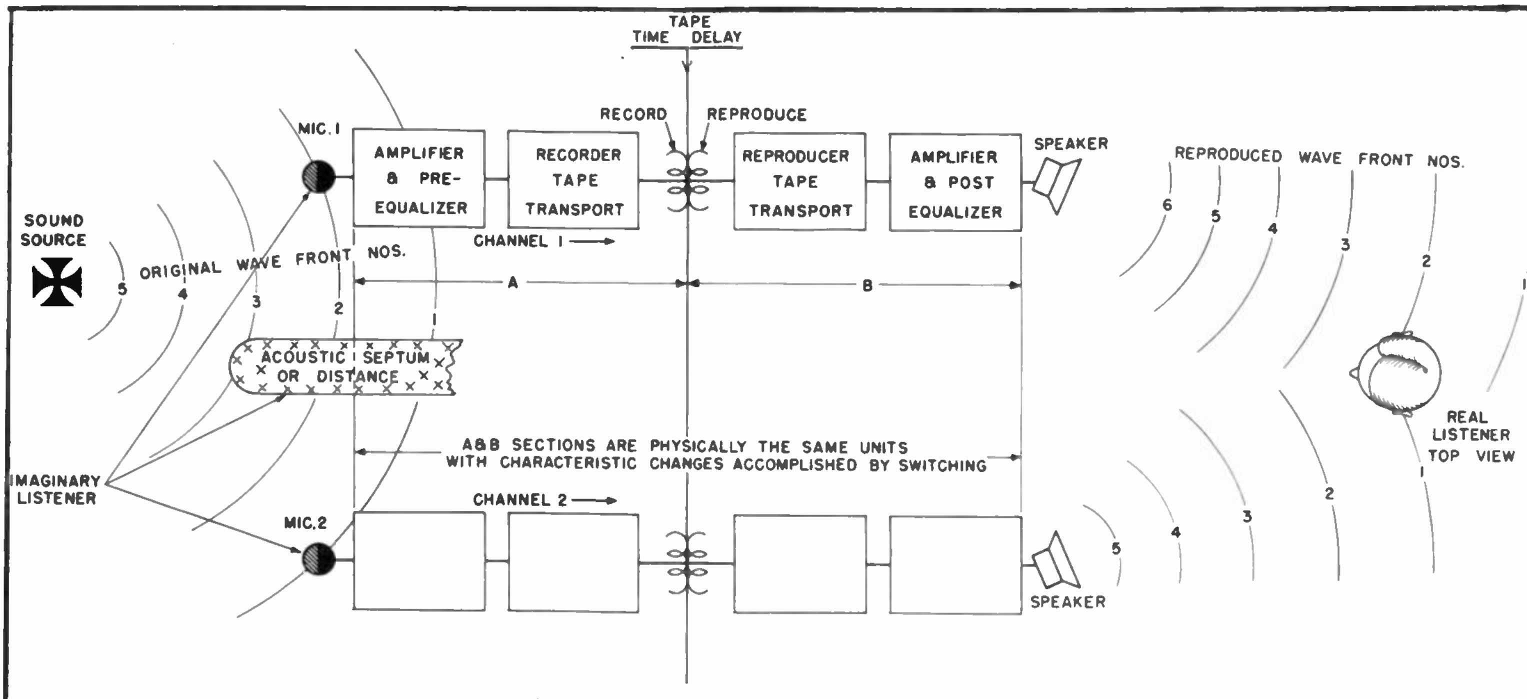

FIGURE 2 NOTE THAT APPROXIMATELY THE SAME SOUND PHASE EXISTS AT THE REAL LISTENERS POSITION AS AT THE IMAGINARY LISTENERS POSITION. THE SLIGHT DIFFERENCE REPRESENTS THE REPRODUCTION ERROR INHERENT IN MICROPHONE & SPEAKER RELATIONSHIPS.

SEMINAR ON ACOUSTICS FOR RADIO ENGINEERS

The growth of Audio Technology during the past decade has greatly increased the number of IRE members interested in Audio. Many members have entered this field only recently. Others, busy with everyday tasks, have not had a chance to keep up with the advances in Audio outside of their immediate field of specialization.

To serve these members more effectively, the PGA has organized a Seminar in Acoustics which will be held during the IRE Spring Convention. This Seminar will provide an opportunity for the old and the new members to exchange ideas regarding fundamental principles and the latest techniques with authorities in various fields of Audio.

The Seminar will occupy one full day of sessions at the Spring National Convention in New York on Wednesday, March 25. The subjects and the speakers will be as follows:

1. "Fundamental Theory" -- Leo L. Beranek, M.I.T., Cambridge, Massachusetts

2. "Microphones" -- Harry F. Olson, R.C.A., Princeton, New Jersey

3. "Loudspeakers" -- Hugh S. Knowles, Industrial Research Products, Inc., Franklin Park, Illinois

4. "Phonograph Reproducers" -- Benjamin B. Bauer, Shure Brothers, Inc., Chicago, Illinois

5. "Tape Recording" -- Marvin Camras, Armour Research Foundation, Chicago, Illinois

6. "Studio Acoustics" -- Hale J. Sabine, Celotex Company, Chicago, Illinois

PGA Chairman Jordan J. Baruch of M.I.T. will act as moderator.

All IRE members interested in Audio Technology are invited to attend and will be assured of an interesting and instructive session.

REPORT OF THE SECRETARY-TREASURER

Marvin Camras
Armour Research Foundation, Chicago 16, Illinois

In keeping with the growth of PGA, our income has increased steadily. The balance in our treasury has grown even after the rising publication expenses were deducted. Our financial status at the close of the last three reported periods is:

	Period Ending September 30, 1952	Period Ending June 30, 1952	Period Ending March 30, 1952
RECEIPTS			
Assessments	$3976.00	$3554.00	$2248.00
Other income	769.21	533.60	175.44
Matched funds	1500.00	1000.00	1000.00
Total Income	$6245.21	$5087.60	$3423.44
EXPENSES	3273.09	2651.33	1905.49
BALANCE ON HAND	$2972.12	$2436.27	$1517.95

Total membership increased from 1653 at the end of June, to 1787 at the end of September. Institutional listings, a major source of our income, is at an all time high of 21 companies.

PGA PEOPLE

OTTO C. BIXLER is Director of Engineering and Research at Magnecord, Inc. His responsibilities include the development of commercial magnetic recording equipment as well as Government research projects.

Previously, Mr. Bixler was associated with Airesearch Manufacturing Company as an electrical development engineer on aircraft and guided missile applications of special electronic equipment. Before that he was with Western Electric in the Electrical Research Products Division, where he served as Systems Engineer on electronic equipment. This work included both optical and magnetic recording projects as well as the design of control equipment. He transferred to this position from Western Electric's Radio Division where he was Senior Engineer on radar fire control and search systems as well as sonar systems. Prior to this time Mr. Bixler was engaged in engineering cost and valuation work for The Southern California Edison Company Limited.

Mr. Bixler is an active member of the Chicago Chapter PGA.

INSTITUTIONAL LISTINGS (Continued)

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

ATLAS SOUND CORPORATION, 1443 - 39th Street, Brooklyn, New York
Loudspeakers, 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.
Manufacturers 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 & Professional Magnetic Tape Recording Equipment

JAMES B. LANSING SOUND, INC., 2439 Fletcher Drive, Los Angeles 39, California
Loudspeakers 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, Attentuators, 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)