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-8 July, 1952

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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.

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

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REPORT TO P.G.A.

Jordan J. Baruch, Chairman
IRE Professional Group on Audio

The P.G.A. is extremely happy to note that the Institute has taken recognition of some of our outstanding people in the field of Audio by awarding them the grade of Fellow. These awards, which were made at the Spring meeting, and which were reported in the PROCEEDINGS for May, 1952, are made annually in recognition of outstanding contributions to the state of art. A list of the Fellows from the field of Audio and their citations follows:

S. J. Begun	"in recognition of his contributions to the field of magnetic recording"
L. L. Beranek	"for his contributions in research, teaching, and administration in the fields of acoustics and speech communication"
M. Camras	"in recognition of his contributions to the field of magnetic recording"
H. E. Hartig	"for his achievements as a teacher, his research in the field of acoustics, and his contributions to the underwater sound program during World War II"
J. K. Hilliard	"for his contributions in the field of motion picture, audio engineering, and in the advancement of standards"
H. H. Scott	"for his contributions to acoustic measurement and reduction of noise in audio reproduction"

To these six people we wish to extend our heartiest congratulations for their award and our heartiest thanks for the advancements and achievements which they have brought to our field.

As mentioned on Page 3 of the May TRANSACTIONS, the Administrative Committee approved a constitutional amendment to increase the size of the Administrative Committee from six to nine members. The Institute has now approved this proposed amendment and it remains to be submitted to the membership for ratification.

Included with this issue, you will find a printed ballot. If you approve the above resolution, please indicate this on the ballot. If you disapprove this resolution, please mark the space so designated. In either case, please mail the ballot in accordance with the printed instructions.

The proposed amendment has been approved by the Administrative Committee with the view of securing more complete representation for the various parts of the country, fields of interest, and types of specialization. Should the resolution be approved, the constitution will be altered where necessary and the three new members of the Administrative Committee will be appointed for terms of three years, two years, and one year as soon as possible.

Replies have now been received from most of the committee chairmen of the PGA, and indications are that during the coming year our committees will be chaired by essentially the same people who chaired them during the past year. Their able work in contributing to the growth of the PGA will therefore continue for another year.

DO YOU AURALIZE?*

Daniel W. Martin
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Cincinnati 2, Ohio

In this interesting Technical Editorial, the PGA Midwestern Editor treats certain acoustical concepts and terminology. By special arrangement with the IRE Editorial Department and the "Journal of the Acoustical Society of America", this editorial is being published simultaneously in both publications. The author will welcome comments from our readers regarding this Editorial.
— Editorial Committee

During the past ten to fifteen years the term "audio" has developed in usage from strictly scientific and engineering parlance into a word understood and even used by the proverbial "man on the street". Many factors have contributed, such as the following:

1. More widespread use of audiometers in the selection of hearing aids.
2. Increased emphasis upon the use of audio equipment in education and training.
3. The formation of organizations to serve the professional interests of engineers specializing in audio equipment and its use.
4. Entire publications on audio topics and other publications devoting a section to audio.
5. The growth of a new type of hobbyist known as an "audiophile" (a lover of recorded sound).
6. The distinction made in television program credit lines, between the "video engineer" and the "audio engineer".

Another group of terms, which antedates "audio" in widespread usage and acceptance, includes "acoustics", "acoustic", and "acoustical". The new American Standard Acoustical Terminology, sponsored by the Acoustical Society of America in co-operation with the Institute of Radio Engineers, makes a distinction in the usage of "acoustic" and "acoustical", similar to that previously adopted for "electric" and "electrical". There would appear to be little reason for ambiguity between "audio" and "acoustic" as long as "audio" pertains to the means by which the electric counterpart of sound is amplified, recorded, played back, modified, monitored, or transmitted. Of course, the sound itself is "acoustic", and the devices by which the electric waves are converted to acoustic waves are "electroacoustic" transducers.

It is in that part of the process known as "audition", or hearing, that the terminology becomes more difficult to clarify. An "audience" composed of "auditors", occupying a space called an "auditorium", uses the "auditory sense for the reception of "audible" sounds. Yet another group of terms, less widely used but well known, includes "aural" and "auricle", "aurist".

* Manuscript received April 9, 1952.

Perhaps the phonetic similarity of "aural" and "oral" has discouraged the use of the former term, but there is an important audible distinction which specialists in sound should be capable of producing consistently. (Pronounce "au" as in Austria and "o" as in old). It may be too late to suggest the combination "aural-visual" as a substitute for "audio-visual", but the writer finds "aural-visual" more euphonious and somehow more suggestive of the educational process. Perhaps for the "aural-visual" method there should be "audio-video" equipment. There might be some slight objection to "aural" as implying the existence of an aura, but this is hardly more serious than the implication of a vision by the term "visual".

These remarks are introductory to the main purpose of this editorial, which is to suggest what the writer believes to be a new and useful verb, to "auralize", and its companion noun "auralization". These words would be the counterpart in hearing to the terms "visualize" and "visualization" in seeing. In other words, "to auralize" would be to form a mental impression of sound not yet heard. Auralization is much more commonly practiced than has generally been recognized. Recent research in physiology reveals that people move the muscles of the vocal mechanism involuntarily and almost imperceptibly as they read silently. Surely some auralization must accompany these reflex actions of "vocalization". Why else would one so enjoy reading poetry silently?

In the field of music the composer auralizes the composition while writing, the conductor auralizes as he scans the manuscript before the rehearsal, and the instrumentalist or vocalist auralizes while examining the written music before the time comes for performance. Many musicians have difficulty playing music in a key which differs in pitch from that shown on the printed sheet because the sound elicited differs in pitch from that auralized. This writer, for example, finds it much easier to transpose a familiar melodic line by playing in accordance with transposed auralization than by shifting an arbitrary number of semitones from the notes printed on the music.

Recording engineers, program directors, and audio monitors auralize (or need to) for the typical ultimate listener. In other words, through practice one should learn to auralize a modified sound while listening to an original or monitored sound, in order to compensate for differences in listening means and environment. Aids to auralization need to be developed for the audio monitor.

Listening tests, whether for musical instruments, loudspeakers, records, or audio systems, require auralization for ultimate validity. It is an art to be practiced. Do you auralize?

PROPER CARE OF TEST RECORDS*

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In accordance with the policy of the PGA, the TRANSACTIONS endeavors to publish material of tutorial nature dealing with the latest advances in audio theory and technology as expounded by eminent authorities in the field. This invited editorial will prove to be of great interest to all persons responsible for the custody and use of records. — Editorial Committee

A well made test record, such as one of the frequency records used for calibration of reproducing equipment, is a precise measuring device. In order to utilize such records to their fullest capabilities, they must be afforded the same care as other precision equipment. The following paragraphs point out some of the "do's" and "don'ts" which should be followed.

Storage Temperature

When properly stored, all types of phonograph records are capable of withstanding reasonable temperatures. Long periods at temperatures up to 120°F do them no harm if they are properly supported.

Excessively high temperatures will, of course, permanently damage records. Such temperatures may be reached if records are stored near registers or radiators, or if they are exposed to direct sunlight. Records left on a window sill or in a parked automobile, for example, can be seriously damaged by the heat absorbed from direct sunlight. A lamp bulb placed too near records can also be a dangerous source of heat.

Storage at excessively low temperatures does not harm phonograph records. If records have been allowed to reach zero temperature, merely bring them back to normal room temperature slowly and no harm will result. Care should be exercised in handling records while they are excessively cold since they may be slightly brittle until they are warmed up again.

Position during Storage

In general, records should never be stored in such a position that their own weight or pressure of other objects will tend to warp them out of shape. Any plastic material, such as a phonograph record, is subject to warping or bending if it is left under a mechanical stress for a sufficiently long time.

Ten- and twelve-inch records, either 78 or 33 1/3 rpm, should be stored vertically on edge in their original containers or in suitable storage albums. Such records should never be piled flat, stored at an angle, or left on the

*Manuscript received April 10, 1952.

supports of a changing mechanism.

"45" records may be stored either vertically or piled flat. The raised label area of these records, combined with their extremely light weight, makes flat storage quite practicable. This is often a convenience when the records must be stored in cabinets designed for 10- and 12-inch records. Vertical storage is, of course, much more convenient when appropriate space is available. Like 10- and 12-inch records, the "45's" should never be allowed to lean at an angle against the end of a storage cabinet.

Dust and Dirt

Records should be protected from accumulation of lint, cigarette ashes, or dust of any kind. Normally, the sleeves or folders supplied with the records will provide adequate protection while the records are in storage cabinets. Dirt and dust will collect on the records when removed from the cabinets. Extra care in handling test records will pay good dividends in lengthened record life.

Generally, records should not be removed from their folders until they are actually placed on a turntable to be played, and they should be returned to their folders immediately after use. In addition, care should be taken to avoid leaving fingerprints or other oily smears on the record playing surface. Such deposits tend to cause dust and lint to stick to the record surface and work into the grooves where it is difficult to remove.

A word about the effect of dirt and dust on test records should emphasize the necessity for keeping records as clean as possible. In general, the mere presence of loose dust and lint on a playing surface will not seriously interfere with high quality reproduction or add noise because the dust is usually brushed aside by the stylus as the record is played. Serious noise in the form of clicks or "static" will result, however, if a dusty record is scuffed or rubbed against another record or against an envelope with some pressure applied. Microscopic particles have become imbedded in the record compound or have scratched the groove walls sufficiently to form slight projections or holes which cause noise. Once a record has been damaged in this manner, no amount of cleaning will restore its original quiet reproduction.

If a record collects a film of dust or dirt, it should be thoroughly cleaned before playing or storing. In cleaning, it is desirable to use a soft cloth which has been dipped in a dilute room temperature solution of a soapless detergent such as "Dreft", "Tide", "Vel", etc. Such solutions should be kept away from the record label. If a detergent solution is not readily available, use a soft cloth which has been dampened with plain cool water.

Under certain temperature and humidity conditions, records will acquire relatively high charges of static electricity while being removed from envelopes, slipped on a turntable, or wiped with a brush or dry cloth. Such charges make the records much more likely to pick up harmful dust and dirt. If left alone, the charge will gradually dissipate into the air, but as pointed out earlier, the accumulated dust may cause considerable damage to the records in the meantime. The static charge may be removed by wiping the

the record surface with a damp cloth, but this will not prevent the build-up of a charge the next time the record is used. Use of the soapless detergents in cleaning records leaves an invisible film on the surface which has a definite antistatic effect for a short period of time. For a more lasting, but not necessarily permanent effect, any of the currently available antistatic preparations for disc records may be used.

Record grooves are extremely accurately prepared surfaces which must be maintained in their original condition if the highest fidelity of reproduction is to be obtained. In cleaning records, therefore, it is essential to use care in rubbing or wiping the surfaces so that the grooves will not be scratched or scuffed. All wiping should be in the direction of the grooves and not across the grooves. Rough brushes, dusty cloths, or excessive pressure in wiping may all produce serious scratches.

Records should never be cleaned with alcohol, naphtha, or other household cleaning fluids. Many of these cleaning materials will dissolve some of the record surface. The record cannot be damaged by cool water, as recommended above.

Record Wear

Even with the best of care, test records eventually wear out in usage. While an exact analysis of the causes of wear is not possible, it is evident that dust particles are one of the leading factors. It is not difficult to envision a gouge or tear in a groove wall which was started by a dust particle between groove wall and stylus, and which is spread and extended by subsequent passages of the stylus.

Actual plastic flow of the record material is, in some cases, a cause of record wear. The mechanical stresses to which the groove wall is subjected by the stylus may be sufficient to deform the modulations and produce a distorted signal. On test records, this is most likely to occur on the high frequency bands.

In some cases, the friction between stylus and groove wall develops sufficient heat to allow roughening and tearing of the surface film of plastic. This is not likely to be an important factor in test record wear, however. I It becomes important only in the case of heavy pickups.

A fourth cause of record wear is the chipping away (failure in shear) of portions of the groove wall. This seldom happens in the case of flexible vinyl records, but is the primary cause of wear on rigid "shellac type" pressings. The dust developed on such records offers visible evidence that material has been bodily removed from the groove.

The symptoms of wear are: (a) an increase in background noise; (b) an increase in distortion; (c) a change (it may be either an increase or a decrease) in output. These symptoms can best be detected by comparison of the suspected record with a new pressing of the same record. It is desirable that test records in routine use be inspected in this manner at regular intervals so that pressings which have changed sufficiently from their original characteristics as to be no longer useful may be discarded and replaced.

It should be noted that mere storage does not change the characteristics of a test record. Thus a record well stored for two or three years may still be a "new pressing" so far as its usefulness in measuring is concerned.

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REPORT ON FOURTH SOUTHWESTERN IRE CONFERENCE*

R.M. Brougher
Houston Section IRE
Houston, Texas

The fourth Southwestern IRE Conference held in Houston, Texas, May 16-17, was finally here (after nine months of labor) and activities began with opening remarks by Gerald K. Miller, General Conference Chairman, a notice of welcome by Sidney A. Martin, Chairman of the Houston Section, a keynote address by Dr. Donald B. Sinclair, President of the IRE.

Following the presentation of student papers from Rice Institute and the University of Houston, there were technical sessions on Instrumentation and Geophysics, Communication and Navigation, Components, Broadcast and TV, Audio and Instrumentation, and Microwave Communications with a total of thirty-one papers presented.

The Audio session included the following papers and was held under the guidance of Lewis B. Erath as Technical Chairman:

"Microphone Directivity", by B.B. Bauer, Shure Brothers, Inc.

"Recent Development in Magnetic Tape Recording", by Harrison Johnson, Ampex Electric Corporation.

"Ionic Loudspeakers", by J.C. Axtell, Southwest Research Institute.

"New Developments in Loudspeaker Design", by Frank H. McIntosh, McIntosh Engineering Laboratories.

"Sub-Audio Time Delay Circuit", by C.D. Morrill, Goodyear Aircraft Corporation.

Other papers of particular interest to PGA members were "An Audio Circuit with Splatter Suppression", by John Reinartz, Eitel-McCullough Inc., and "Mechanical and Electrical Design Considerations in Speech Input Systems of the Highest Fidelity", by Norbert L. Jochem, Gates Radio Company.

Many exhibitors contributed to the success of the conference, and the items displayed were far too numerous to mention. However, some of the audio items of particular interest were the following:

The Gulf Coast Electronics demonstration room included a wide variety of standard high fidelity components, some of the newer items being custom built exponential horns and cabinets, the Stevens 500D direct drive amplifier used with the 500-ohm woofer and tweeter speakers mounted in a Klipsch cabinet, a laboratory model of the McIntosh speaker system, the Grommes preamplifier, Magnecorder equipment including the binaural, and ample supply of good demon-

* Received May 26, 1952.

stration tapes and records, all set up for comparative listening.

The Sound Sales and Engineering Company demonstration room included RCA sound equipment, custom built consoles for schools and industrial uses, intercommunication and paging equipment, and a demonstration of television projected on a 6- x 8-foot screen.

In the Ampex Electric Corporation booth was a multichannel tape recorder with demonstrations of AM and FM recording of geophysical data.

The Gates Radio Company demonstration room included their own transcription tables, the new line of plug-in amplifiers, a dual channel console, and numerous other components such as Ampex, Magnecorder, the Magnecordette, Garrard changers, Partridge transformers, Gray pickups, KT-66 tubes, Jensen speakers and enclosures.

Other articles of interest such as ADC transformers, Clarkstan equipment, Permoflux speakers, Radio Craftsman units, Concertone recorders, Browning tuners, Triad transformers, Webster Teletalk equipment, Rekocut turntables, Pickering and Audak cartridges, University speakers, MB vibration pickups, miniature components, and practically all makes of test equipment were on display by the various exhibitors along with TV, instrumentation, industrial, communication, and component items.

During the conference it was noticed that Dr. William N. Rust, Jr., Director of Region 6, was quite busy shaking hands and chatting with his many acquaintances. Mr. William C. Copp, with his attractive IRE booth and full of wit and humor (as is usual) passed out good advice, applications for membership in the IRE, information regarding the Professional Groups, and even sold Don Sinclair an emblem. By the way, during Dr. Sinclair's stay in Houston, he was declared a naturalized citizen of the Empire of Texas, was given a certificate to prove it, and was awarded a Texas hat. Conversations overheard included one in which Frank McIntosh was encouraged either to make his transformers available (you can get one now with each amplifier) or to produce a kit for the people who like to say, "I built this".

The conference closed with a banquet address by Commander T.A.M. Craven of Washington on "TV Spectrum Allocations". Cash awards of $25.00 for first place, $15.00 for second place, and paid-up Associate memberships in the IRE for both were given for the two best student papers. The following newly elected officers of the Houston Section were announced: Harvey T. Wheeler, Chairman; Karl O. Heintz, Vice-Chairman; James K. Hallenburg, Secretary; Floyd S. Phillips, Treasurer.

All of us look forward to another successful Region 6 Conference planned for San Antonio, Texas, in February, 1953.

PGA PEOPLE

EDMOND GRANGER DYETT, JR.

Born March 30, 1928, in Guatemala, C.A. Attended elementary and high school in New York state. Trip winner in Westinghouse Science Talent Search and Valedictorian of senior class, 1944.

Attended Massachusetts Institute of Technology, 1944-1947, majoring in Physics and Acoustics. Received B.S. in Physics in 1947.

Employed as Research Technician by M.I.T. Acoustics Laboratory during summer of 1947 and joined Hermon Hosmer Scott, Inc., Cambridge, Mass. as Development Engineer in fall of 1947. At present, heading the Production and Purchasing Department.

Joined the Institute of Radio Engineers in 1948 and assisted in formation of the Audio Chapter in Boston in 1949. Served as Vice-Chairman of this Chapter in 1950-1951 and as Chairman in 1951-1952.

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MICROPHONE DIRECTIVITY*

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The auditory habits of humans are dependent to a great extent upon the experience of binaural hearing, which, aided by eyesight, helps us to localize and concentrate upon the desired sources of sound. One is compelled unconsciously to transfer this experience to the microphone -- but this often leads to disappointing results. A conventional "pressure" microphone is a monaural device and what it perceives can be roughly approximated by plugging up one ear and closing both eyes. When this is done, the desired sounds are often lost amidst the surrounding noise. To aid with the pickup of the desired sounds, directional microphones have been developed which favor sounds arriving from certain directions. These directional microphones serve partially to overcome the handicap of the monaural system by permitting us to concentrate upon the desired sounds and to reject undesired sounds. In this paper, we propose to review microphone directivity and the utilization of directional microphones -- in the light of recent developments.

The choice of a microphone by the purchaser depends upon several factors -- technical, financial, and aesthetic; an important consideration is the mode of the moment. At this writing, the "new look" in microphone fashions is the elongated miniature microphone. Unfortunately, as is often the way with fashions, the choice of the mode of the moment does not necessarily indicate compliance with sound physical principles. Disregard of directional properties of miniature microphones has led many users to disappointing results. It is well to remember, therefore, that while miniaturization "per-se" is not bad, it is not good -- unless directional properties are kept in the foreground.

DIRECTIONAL PROPERTIES

To review the directional performance of pressure microphones, Fig. 1a shows the polar patterns of the A.S.A. Standard Type L microphone and preamplifier.[1] This microphone was introduced by the Bell Telephone Laboratories during the early part of World War II as a laboratory sound pressure standard. It is the forerunner of all present-day omnidirectional "miniature" microphones, and shares with them its directional properties. Only one surface of the diaphragm is exposed to sounds, and, therefore, the directional properties are dependent solely upon the ability of sound pressure to reach the diaphragm front. The diameter of the microphone is only 0.936 inches, and sound pressure from all directions can reach it very well indeed. As a result of this small diameter, this microphone is quite non-directional, or "omnidirectional". The directivity patterns at 250 and 1000 cps do not deviate from a perfect circle by more than 1/2 db. At 4000 cps, the response begins to drop from the side and the rear; nevertheless, it is still within 3 db of being uniform. It is

*Presented at the Southwestern IRE Conference, May 17, 1952, in Houston, Texas.

evident that this type microphone will receive all sounds about equally, whether they arrive directly from the performer, from the audience, or after reflection from the room boundaries. The type of performance exhibited by an omnidirectional microphone is desirable in certain applications -- such as the measurement of sound level or intensity of noise, or in orchestral pickup in an acoustically "dead" studio where it may be desired to retain the maximum of remaining reverberant quality. However, sound engineers have discovered that omnidirectional pickup does not provide sufficient control over undesired sounds. This accounts for the continued popularity of the earlier type semi-directional microphones in many broadcasting and communications applications.

The polar patterns of a typical semi-directional microphone are shown in Fig. 1b. Sounds with frequency of 250 cps and below are received by the microphone with almost complete uniformity; however, beginning with 1000 cps and above there is considerable directivity. This directivity is caused solely by the diffractions of sound around the microphone case when the wavelength is comparable to, or smaller than the dimensions of the case, and, therefore, directivity increases with frequency. At 4000 cps, for example, directivity is very pronounced. Semi-directional microphones are useful in separating high frequency noises, such as applause, footsteps, etc. from the voice of the commentator. Several manufacturers have provided snap-on "directional baffles" for small pressure microphones to increase their directivity.

Advances in broadcasting and public address techniques demanded an increasingly greater control over reverberation. Therefore, new means had to be discovered to improve directivity of microphones at low frequency without a corresponding increase in size. The earliest of the successful attempts in this direction was the invention of the velocity microphone by Dr. Harry Olson in 1932.[2] Velocity microphones owe their directivity to the "gradient" of sound pressure, that is, to the space-rate change of sound pressure in the direction of propagation. Many improvements have been made in their design during the past two decades; however, the basic operating principles have remained unchanged.

To review these principles briefly, a gradient-operated microphone is shown in schematic cross section in Fig. 2. The unit has two elongated pole pieces which provide an intense magnetic field, and a thin aluminum ribbon, free to float back-and-forth in this field as a result of the difference of sound pressures at its sides. We define a distance d, which represents the equivalent distance of sound travel between the front and the back of the ribbon when arriving from a 0° or frontal direction. To the first approximation, this distance may be assumed to be equal to the shortest air path from the center of one side of the ribbon to the other. As the sound travels from right to left, the pressure at the back p_2 will be delayed, or lag behind the pressure at the front p_1 by the time interval d/C_v, where C_v is the velocity of sound. This interval corresponds to a phase angle equal to $2\pi fd/C_v$ radians, where f is the frequency. If the sound arrives from an angle Θ, the effective difference in paths equals $d \cos \Theta$, and the phase shift angle is correspondingly decreased. When $\Theta = 90^\circ$, the time of arrival at the front and the back is identical, resulting in equal and opposite pressures, i.e., no output from the microphone.

In graphical form, the pressure relationship is shown by the vector diaphragm, where, for the zero degree incidence, $p_1(0°)$ and p_2 are separated by the angle $2\pi fd/C_v$. For other angles of incidence -- choosing p_2 as reference vector -- p_1 will move along the dashed arc in accordance with the cosine law. The resultant R will diminish as the angle Θ increases until it will vanish at the 90° incidence. Thereupon, there is a phase reversal and the resultant increases to a negative maximum at 180°. The output of the microphone is proportional to the magnitude of this resultant pressure difference. Plotting this resultant in polar coordinates, the typical figure-8 polar pattern is obtained.

Despite the many valuable characteristics of gradient microphones, often a microphone is desired which is unidirectional or most sensitive to sounds arriving from one direction. This property can be obtained in more than one way. We shall describe, however, the way which is used today more than any other, and which employs acoustical phase shift networks.[3]

A phase shift microphone is shown in schematic cross section in Fig. 3. For a 0° incidence, let the sound travel from right to left and impinge upon the diaphragm with a pressure $p_1(0)$. The sound travels an additional distance d before reaching the rear entrance ports; therefore, the pressure p_2 at these ports is delayed by the interval d/C_v, corresponding to a phase shift $2\pi fd/C_v$ radians. To this point, the operation is identical with that of the gradient microphone. However, in contrast with the gradient operated structure, a new element is now introduced, in the form of an acoustical phase shift network shown schematically by the doubly cross-hatched section. This network is designed to shift the phase of sound entering the port, so that the pressure p_3 acting internally upon the diaphragm will be of the same magnitude as p_2 but shifted in phase by an angle $m(2\pi fd/C_v)$. Therefore, the two phase angles are related by the constant of proportionality m.

If sound proceeds from a direction Θ, the effective distance between the front of the diaphragm and the rear entrance ports becomes d cos Θ, corresponding to a phase shift $(2\pi fd/C_v) \cos \Theta$. Again choosing p_2 as a fixed reference vector, the vector $p_1(\Theta)$ will shift along the dashed arc in accordance with the cosine law curve at right. The resultant pressure upon the diaphragm is the difference between $p_1(\Theta)$ and p_3. Complete cancellation will not occur at 90° but rather at some angle between 90° and 180°, when vectors $p_1(\Theta)$ and p_3 become coincident. As the source of sound keeps rotating around the microphone, the phase of the resultant force is reversed and rises to a negative maximum at the 180° incidence. The magnitude of R_Θ plotted against the angle Θ in polar coordinates produces the familiar graph of a limaçon defined by the equation $F(\Theta) = (1 - k) + k \cos \Theta$. Here, k is the quotient of the maximum phase shift angle due to the external gradient divided by the maximum total phase shift angle, and it is given by the equation:

$$k = 1/(1 + m) \qquad (1)$$

For the special case where m = 1, i.e., the internal phase shift is equal in magnitude to the maximum external phase shift, k = 1/2, and the polar pattern is that of a cardioid.

One should keep in mind that the external phase shift varies directly with the frequency. Therefore, the internal phase shift must also increase with frequency to maintain the directional pattern at all frequencies. To fulfill this condition, the phase shift network is designed to provide an internal phase shift which is proportional to frequency throughout the important working range of the instrument. The theory of phase shift network design for use with unidirectional microphones has been described elsewhere[3] and need not be restated for the purposes of this paper.

SMALL DIRECTIONAL MICROPHONES

In Fig. 4 is shown a small gradient type element. The magnetic structure consists of a soldered assembly of two Alnico magnets, two soft iron pole pieces, and a ribbon mounting frame. The assembly is 2-1/2 inches long, 7/8 inch wide, and 1/2 inch deep. The ribbon element is approximately 1 inch long, 1/16 inch wide, and 0.0001 inch thick, and it is suspended in the air gap so that it is free to move to-and-fro with a clearance of approximately 0.003 inch at either side. The ribbon is suitably insulated from the frame, and the electrical connections are made to the terminals at either end. To protect the ribbon from dust, a fine mesh screen is installed at both sides, before magnetizing. The unit is mounted in the case shown in Fig. 5. Access to both sides of the ribbon is achieved through perforated metal grilles. The lower part of the case contains the low impedance-to-line transformer, and a switch for selecting either the low, medium, or high impedance winding. The case is mounted on a cable connector shell through a rubber bushing. The shell contains a frequency modifying choke and a voice-music switch.

A relatively small unidirectional phase shift microphone cartridge is shown sectionalized in Fig. 6. The front of the diaphragm is freely exposed to sound. After travelling around the pole piece, the pressure enters through the slit formed by the voice coil and the bobbin, which is part of the phase shift network, into the chamber at the rear of the diaphragm, and thereupon it is by-passed through the acoustical screens into the inner volume of the magnet and into the rear chamber. An appropriate choice of acoustical constants of the slits, screens and volumes produces a phase shift proportional to frequency throughout the useful frequency range of the network. The polar pattern is a cardioid at all important frequencies.

The unit is approximately 1-5/8 inches square and 2-1/4 inches long. It is mounted in the case shown in Fig. 7 with the front removed using elastic rubber mounts. The multi-impedance voice coil-to-line transformer is located below the unit and the switch is conveniently accessible at the rear base portion. The profile of the case as seen by the audience is approximately 2 inches wide. The case is 3 inches deep, and is attached to a stand type connector similar to the previous model.

IMPLICATIONS OF DIRECTIVITY

The ability of a directional microphone to favor reception of frontal sounds is described by a number called the Directivity Factor.[4] This term is defined as the ratio between the power transmitted by the microphone owing to frontal sound and the power transmitted owing to random sounds of equal intensity. The method employed to determine the Directivity Factor can be made clear by reference to Fig. 8. Let the sounds from the performer impinge directly upon the front of the microphone and note the power output. Next, remove the direct sound and let the noises and other undesired sounds, as from excessive reverberation, etc. fall upon the microphone, from all directions at random but with the same average intensity as before. What is the power output due to the undesired sounds? In the case of the omnidirectional microphone, the answer is simple. The microphone being equally sensitive in all directions, the output due to the noise is the same as that due to the desired sounds. Therefore, the Directivity Factor is unity. This is precisely why omnidirectional microphones are useful in the measurement of ambient noise.

In the case of a directional microphone, the answer is a bit more complicated.[5] Assume an imaginary sphere of unit radius around the microphone. The area of this sphere is known to be 4π. The area of a zone located at an angle θ from the axis of symmetry and having a width $d\theta$ is $2\pi \sin\theta\, d\theta$. All directions of sound arrival being equally probable, the fractional contribution to the average intensity by sounds transmitted through this area is $(2\pi \sin\theta\, d\theta)/4\pi$. If the fractional angular voltage response of the microphone is given by the function $F(\theta)$, then the fractional angular power response is given by the function $F^2(\theta)$. The fractional contribution to the total power output due to sounds penetrating through this annular area will then be $F^2(\theta)\ (2\pi \sin\theta\, d\theta)/4\pi$, and the total power output from the microphone in terms of the normal incidence power can be obtained by integrating this expression from $0°$ to π. The Directivity Factor is the reciprocal of this integral, and it is given, therefore, by the equation:

$$\text{D.F.} = 2/\int_0^{\pi} F^2(\theta) \sin\theta\, d\theta \qquad (2)$$

This equation is applicable only to microphones which have an axis of symmetry coincident with the direction of maximum sensitivity, which is the case with the Limaçon family of directivity patterns. The Directivity Factor for the Limaçon family is shown in Fig. 9, in terms of k, the fractional contribution of the gradient component as given in Eq. 1. At both extremes are shown the parents of the family, the pressure pattern (circle) with a Directivity Factor equal to 1, and the gradient (cosine or figure-8) pattern, with a Directivity Factor equal to 3. Mixed in equal contributions, for $k = 0.5$ is the cardioid, also with a Directivity Factor equal to 3; and two other recognized members of the family, the hyper-cardioid[6] for $k = 0.75$, which has the highest Directivity Factor equal to 4, and the super-cardioid,[7] the most unidirectional member of the family, with practically nil response at the rear and with a Directivity Factor of 3.75. It is evident that all microphones with polar patterns from the cardioid to the gradient are highly effective in favoring sounds on the axis against sounds arriving from random directions. The choice between them depends upon operational conditions and the acoustic configuration.

A most useful corollary of the Directivity Factor graph is a deduction of the distance from the performer to the microphone for equal signal-to-random noise ratio. This we call a "Distance Factor". The sound energy density roughly varies inversely with the square of the distance from the mouth of the performer. Therefore, the Distance Factor is the square root of the Directivity Factor. The Distance Factor for the Limaçon family is also plotted in Fig. 9. It is seen that for equal signal-to-noise ratio, microphones having directional characteristics from cardioid to gradient may be used at distances from the performer between 73% and 100% greater than is possible with the omnidirectional microphones. Formerly, the Distance Factor was considered to have a bearing principally upon the freedom of motion of a performer in front of the microphone. It is now known, however, that this factor has an important bearing upon the unobtrusiveness of microphones.

UNOBTRUSIVENESS OF MICROPHONES

We shall attempt to answer the question: "How unobtrusive is a miniature microphone?" At first, this question sounds trite. However, consider the following query: Will a 1-inch diameter microphone placed at 1 foot from the performer partially hide his face for a greater portion of the audience than a 2-inch diameter microphone placed at 2 feet? When put in this fashion, the question takes on a different aspect. The geometry of this situation is shown in Fig. 10. The large circle represents the head, and the small circle, the microphone. It is evident that all persons in the audience located within the angle γ will see the microphone in front of all or a portion of the face of the performer. This will not be the case for those outside the angle γ. The geometry of this situation is identical with that of a solar eclipse where γ is the angle of the penumbra. γ is given by the equation $\sin \gamma/2 = (a + b)/(a + b + 2D)$, D being the distance between the two bodies and a and b the diameters of the bodies. Assuming a head with average diameters of 7 inches, families of curves have been drawn in Fig. 10 showing the angle of the penumbra as a function of distance for microphones of various diameters. These curves bring out interesting facts: An omnidirectional microphone with a diameter of 1 inch placed at 10 inches from the performer will partially obstruct the view of the audience seated within an included angle of 34°. If the microphone could be reduced down to nothing so that only the stand tubing were to remain, the view of a 30° segment of the audience still would be obstructed. However, if the same 1-inch microphone, by virtue of being endowed with directional properties, could be moved out to a distance of 17 to 20 inches, the view of only a 21° segment of the audience would be obstructed. A relatively large microphone of 3-inch diameter at the greater distance will impede the view of only a 25° segment of the audience, which is considerably less than will be caused by the small microphone placed at 10 inches.

Similar results are obtained by repeating this analysis for various distances from the performer. It will be seen in every case that a small omnidirectional microphone obstructs the view of a greater segment of the audience than a considerably larger microphone with cardioid or cosine characteristics. Therefore, for minimum obtrusiveness, microphones which are directional as well as small are needed.

CONCLUSIONS

Let us summarize our findings and state some conclusions. The trend toward miniaturization is definitely good. Miniature omnidirectional microphones have many virtues. They are useful for noise measurement purposes. When small and light they can be easily worn on the lapel of the speaker. They are useful for round table discussions, for orchestral sound pickup and similar applications in relatively "dead" studios where it is desired to preserve all the available reverberation. Semi directional microphones are valuable for general announce purposes, and commentary in sports and public events where it is desirable to create the general presence of the crowd, and yet to decrease some of the high frequency background noises. Directional microphones are indicated when it is desired to greatly reduce ambient noise, emphasize or de-emphasize certain instruments in an orchestra, control feedback in public address systems, reduce reverberation and hangover in live studios, and permit the performer to stand at a greater distance from the microphone. Unobtrusiveness of microphones is also improved by directional properties. The choice of pattern will depend upon the source and location of reflections and undesired sounds, as well as upon the experience and preference of the users.

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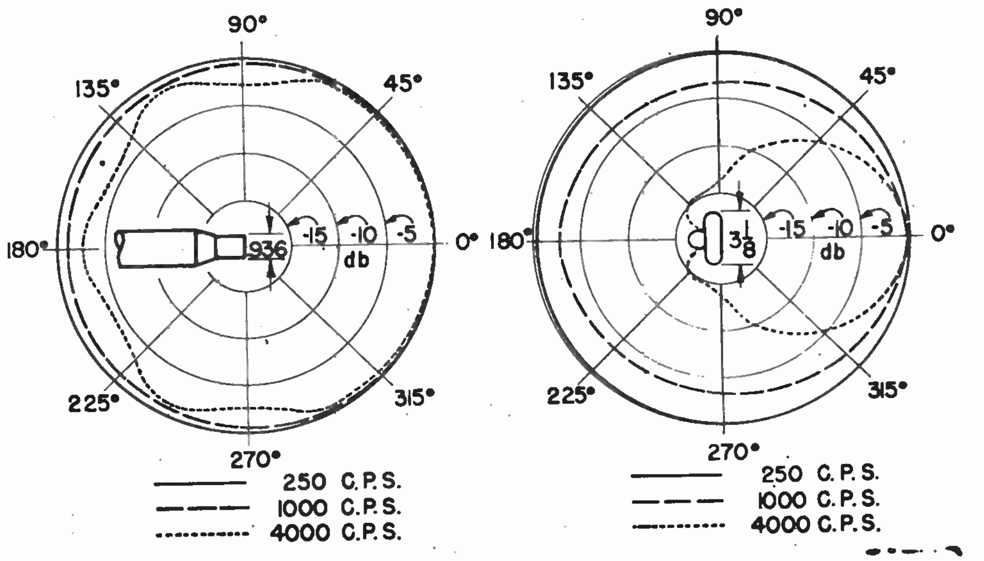


Fig. 1 - (a) Polar patterns of an omnidirectional microphone;
(b) Polar patterns of a semidirectional microphone.

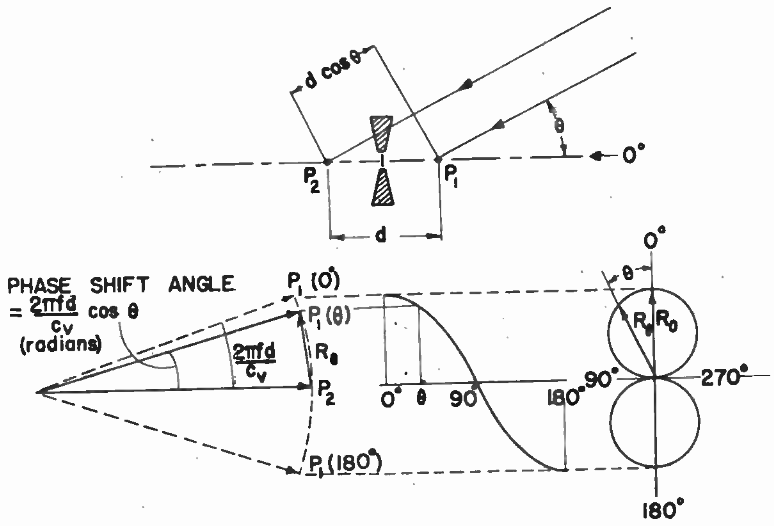


Fig. 2 - Operating principle of a gradient microphone.

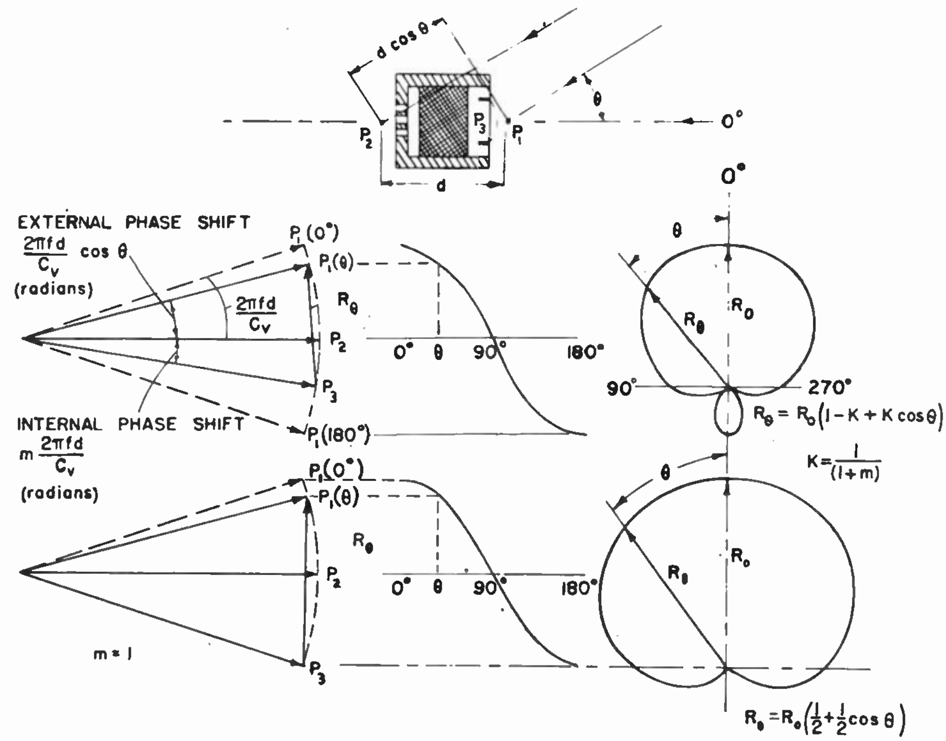


Fig. 3 - Operating principle of a phase shift microphone.

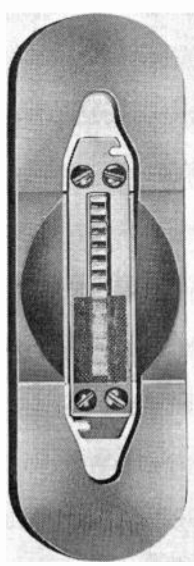


Fig. 4

A small gradient type element.

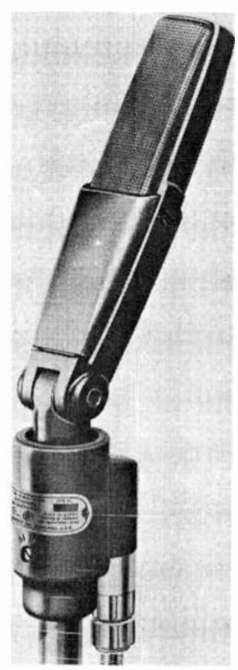


Fig. 5

Miniature gradient microphone.

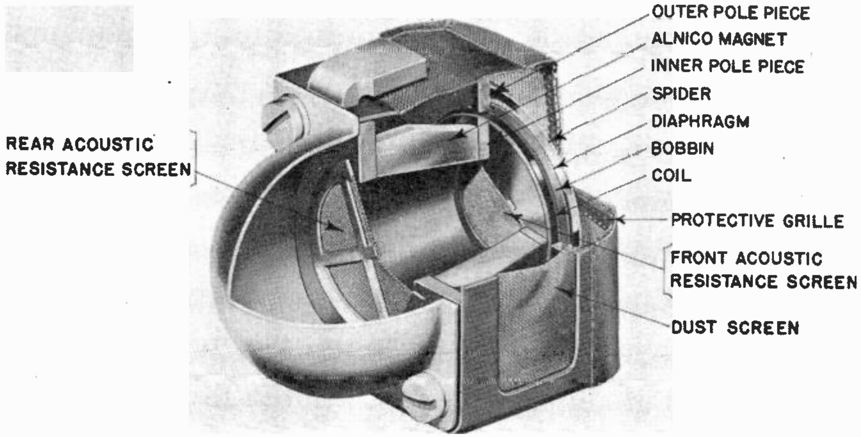
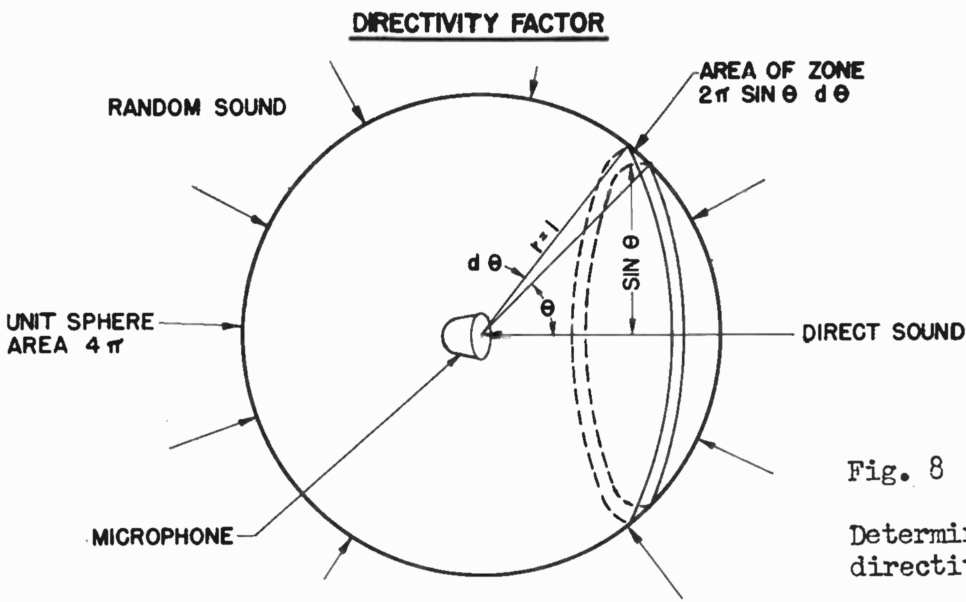


Fig. 6

Section of unidirectional phase shift microphone element.



FRACTIONAL ANGULAR VOLTAGE RESPONSE = $F(\theta)$

FRACTIONAL ANGULAR POWER RESPONSE = $F^2(\theta)$

$$\frac{\text{POWER OUTPUT CAUSED BY DIRECT SOUND}}{\text{POWER OUTPUT CAUSED BY RANDOM SOUND}} = \frac{1}{\int_0^\pi F^2(\theta)\,\frac{2\pi \sin\theta \, d\theta}{4\pi}} =$$

$$\text{DIRECTIVITY FACTOR} = \frac{2}{\int_0^\pi F^2(\theta) \sin\theta \, d\theta}$$

Fig. 8

Determination of directivity factor.

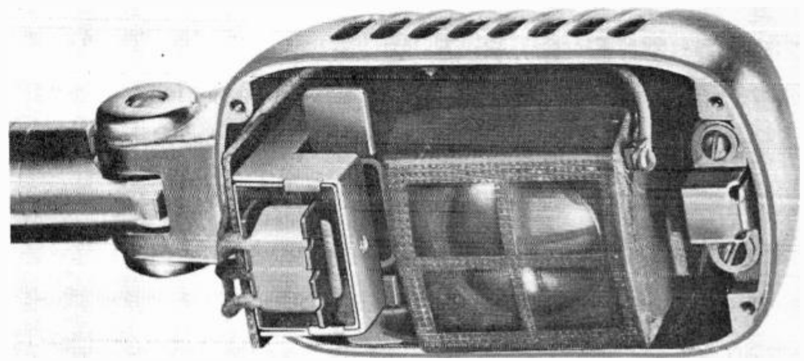


Fig. 7

Internal details of a small unidirectional microphone.

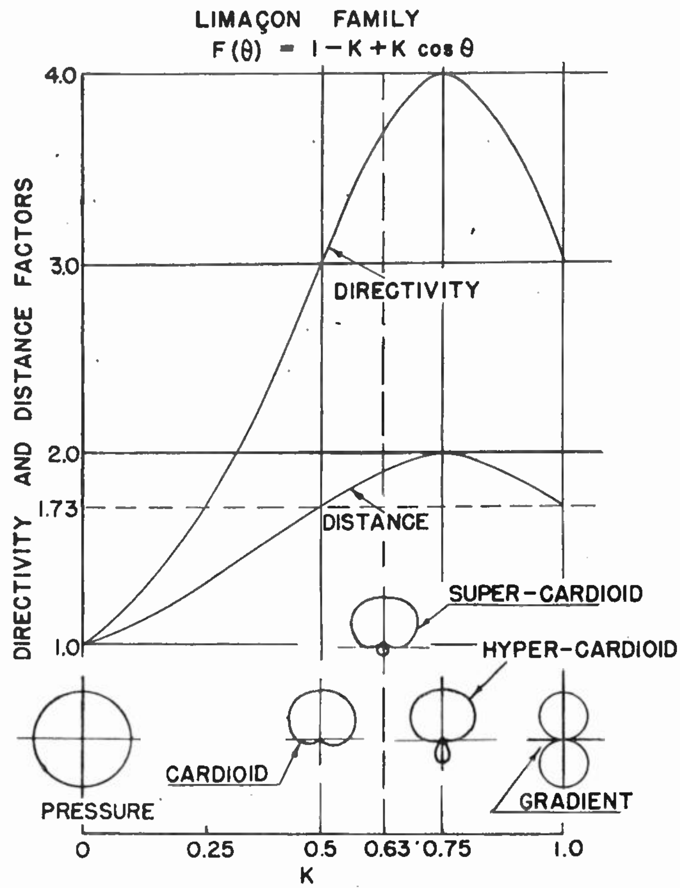


Fig. 9

Directivity and distance factors for the Limaçon family of polar patterns.

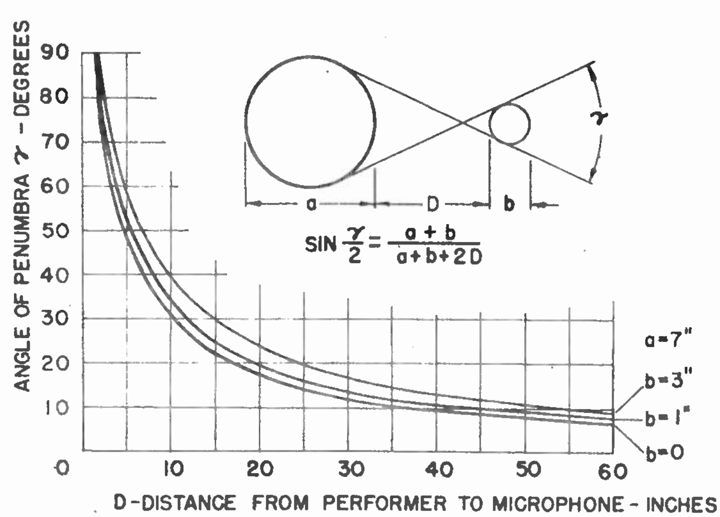


Fig. 10 - The angle of microphone penumbra.

IONIC LOUDSPEAKERS*

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The conventional audio reproduction system has for years consisted of an audio amplifier driving a diaphragm loudspeaker. The loudspeaker has, on occasion, been connected to a horn in order to give more efficient coupling to the air, but the diaphragm mechanism has been used almost exclusively. It is clear that any system which may be proposed to either supplement or complement this time-proven method must either eliminate some of the associated acoustical problems, or be far less expensive to construct. It is the purpose of this paper to present a system which certainly offers promise of meeting the first requirement, and may eventually meet the second. This system has been devised largely through the intensive efforts of Dr. Siegfried Klein of Paris, France, who conceived the idea in the early 1940's, and through whose research the present form of the device was evolved.

Some years ago, while studying the characteristics of radioactive materials, Dr. Klein constructed an equipment, diagrammed in Fig. 1, for the measurement of the ionization produced by the incidence of fast charged particles on various gases. The contemporary theory at that time predicted that the current flowing between the electrodes should be a defined function of the electrode separation, the amount of radioactive material, and the gas characteristics.[1]

When the experimental data had been gathered, an unexpected phenomenon was evident. The currents measured were far greater than those predicted. These were quickly explained away when it was suggested that the electrons created by the initial bombardment were being accelerated by the collector field and were effecting further ionization. The cascade effect was thus proposed to explain the newly observed characteristic.

Now, it is well known that the total ionization produced by a fast charged particle is essentially independent of the density of the medium being traversed, but is rather very nearly constant.[2] However, since Dr. Klein's device had exhibited effects not ascribable to the primary ionization of the theory, there was reason to suspect the validity of the density effect as well. An obvious method of test was, of course, to change the gas pressure in the system, thus altering the density of the gas, which is precisely what was done. Much to the surprise of the experimenter, very small changes in pressure resulted in relatively large changes in the system current. It is now supposed that the effect of the density (or pressure) change upon the collector current was brought about by the change of mean free path of the primary ions; when a reduction in pressure, resulting in an increase in free path, reduced the number of secondary ions released during the primary ion transit time.

Having been well schooled in acoustics, Dr. Klein immediately recognized that the pressure variations in a sound wave were of such a magnitude that

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they should be effective in modulating the system current. Tests rapidly proved, unfortunately, that the current changes produced were too small to be of value, since the random noise in the system had a very high level, and as a result, the signal-to-noise ratio was far too small.

Reasoning that the primary source of noise lay in the source of ionization, Dr. Klein endeavored to devise a means of producing at once a heavier ion current and a reduced noise level. His acquaintance with Richardson's classical work on positive ion emission from incandescent metal surfaces led to the construction of the equipment diagrammed in Fig. 2.[3] The device constructed by Dr. Klein consisted of a platinized metal cylinder, heated internally, situated coaxially with a cylindrical collector electrode, the platinized cylinder being maintained at a positive potential with respect to the collector. Early models were coated with electrolytically deposited platinum black which was known to emit positive ions when heated. The susceptibility of such surfaces to "poisoning" led to the incorporation of modified materials, the most successful of which was a proprietary mixture of precipitated platinum, precipitated iridium, aluminum phosphate, and graphite. The proportions used were entirely empirical, and no explanation has been found for the inclusion of the auxiliary materials other than that they seem to result in greater surface area for the platinum. The formula exhibits the peculiar property of being very resistant to "poisoning".

Properly energized, the platinized cylinder was found to emit very heavy ion currents — currents of the order of 4 microamperes per mm^2 of active surface.[4] When it is realized that the projected surface area corresponding to the active area is a small fraction of a square millimeter, the ion currents emitted from a small projected surface can be quite large. Furthermore, the noise level was found to be no higher than that exhibited by the first model, which produced much smaller ion currents. The sensitivity to pressure change exhibited by the first model was also observed in this unit, but in a much greater degree. It was suggested that the pressure effect was more pronounced because of the greatly enhanced ionization.

Properly coupled, the device seemed to work as a fairly good microphone. It was entirely reasonable to expect that it should also function as an acoustic generator. In order to determine the validity of the prediction, the electrical connections were modified to the extent that the voltage applied between the emitter and the connector was made variable at a rate corresponding to audio frequencies. When tested, the unit emitted an audio signal, but the level was just above the audible threshold. It thus appears, on first examination, that the device does not obey the reciprocity law, as its sensitivity as a microphone has been found to be considerable. That this conclusion may be in error is evident, however, for the efficiency of coupling to the outside medium is very low because of the bad acoustic geometry of the device.

An extended effort was made to determine the validity of the reciprocity law for the device, and eventually it was found that reciprocity did hold, after all. In the course of the investigations, the element was introduced into one end of a conical cavity, and it was noted that the coupling was much improved. Extrapolation of the experimental cavity to an exponential horn was an obvious step.

Up to this point, no effort had been made to direct the investigation to the production of a unit suitable for commerce, as the only major interest to this time had been the source itself. The device appeared sufficiently promising by this time that Dr. Klein decided to develop the acoustic generator as a separate device. However, the experimental design was not particularly good acoustically, and certainly was not conveniently constructed, so that modifications in the design were undertaken.

It is not clear what the next step in Dr. Klein's development was, for no records have been made available to the author. It is known that some years of intermediate research were needed to bridge the gap between the old and the new models of the device. It is also known that the hairpin heated emitter was discarded for a system utilizing radio-frequency heating, though the platinum compound emitting agent was retained.

In lieu of a chronology, the following explanation is offered. It has long been known that the high-voltage dc corona in air was accompanied by heavy ion densities in the region of the corona.[5] Anyone who has experimented with such corona discharges also knows that the emission is accompanied by considerable noise, usually of very high audio frequencies. It was thus apparent that the use of dc corona discharges would not accomplish the desired end. However, the noise level is considerably reduced by the use of high RF voltages, as anyone acquainted with RF arc discharges can confirm.

Now, one of the unusual characteristics of the radio-frequency discharge in air is the fact that only one electrode need be energized to produce the arc flame. Since the excitation is primarily a capacitance effect, the earth serves as the second electrode, without direct coupling. However, the asymmetric ground plane resulting from the use of the earth causes nonuniformities in the arc flame which make its application to this problem complicated.

Early in 1950, Dr. Klein constructed the instrument diagrammed in Fig. 3.[6] The operation of the device can be described as essentially a modulated corona discharge located in the throat of a horn. The inclusion of the horn is, of course, necessary if the small frontal area of the sound source is to be coupled to the air with any degree of efficiency. It will be noticed that the device consists of a platinized cylindrical surface projecting into the throat of a small horn, a re-entrant cavity containing a solid electrode, most conveniently made of platinum and entirely surrounded by a vacuum jacket. The inclusion of the individual elements was dictated by considerable investigation of the system characteristics. It was found that the platinized cylindrical surface provided an excellent source of the ions needed to maintain a stable corona discharge, while the dielectric heating produced by the radial field between the re-entrant solid electrode and the cylindrical ground surface was sufficient to raise the platinized surface to the emitting temperature, provided the system was thermally insulated by the surrounding vacuum jacket. It has been found that the vacuum jacket is not an absolute necessity, but the demands of efficiency dictate that it be retained, for its omission requires that considerably more energy be fed to the corona to maintain stability. The associated electronic driving gear was of conventional design, and incorporated no special circuits or components.

It is to be noted that the entire working unit was constructed of fused quartz, a very difficult medium out of which to fabricate anything so complicated as the unit pictured. Unfortunately, no other material has been found suitable because of the stringent requirements placed upon the construction medium. A measurement of the temperatures involved shows that the surface temperature of the platinized surface is of the order of 1,000° C, while the inner surface of the vacuum jacket is several hundred degrees cooler. It is known that certain glasses will withstand the thermal stresses involved, and several experimental units have been constructed. However, none were satisfactory because of the tendency for glasses containing metal salts to become conducting at elevated temperatures. It was found that so long as the glass units were cooled, they performed properly, although poorly. However, upon allowing the inner jackets to assume equilibrium temperatures, the corona flame became erratic and the glass was very quickly punctured, which destroyed the entire unit. Pure fused quartz, on the other hand, appears to retain its insulating properties at the temperatures involved, and operation is entirely satisfactory.

Evaluation showed that the unit was quite satisfactory from an operational viewpoint, but is is easily seen that the construction of the device is quite complicated. In view of the fact that no one has yet devised a machine for working fused quartz, the problem was even more acute, for production of the units involved the location of numbers of expert quartz workers. Since the entire globe boasts only a few hundreds of these artisans, it is obvious that their abilities must be very efficiently utilized. An experienced glass worker pointed out the fact that the most difficult of the operations involved in the construction was the insertion of the axially symmetric quartz cylinder, and that if this could be elininated, the constructional difficulties would be considerably reduced.

As a purely experimental attempt to simplify the unit, a modified version was constructed, as shown in Fig. 4. The unit is similar to the previously described device in all respects but one -- the quartz cylinder. The modified version of the device consists of the fused quartz horn surrounded by the evacuated jacket with a small platinum sphere located at the origin of the horn. The platinum lead is made removable to facilitate cleaning and modification. And, though the principle of operation is much the same as that of the previously described unit, the appearance of the corona is much different, for, instead of originating over an extended surface, the entire corona now emanates from the surface of the small platinum sphere, as shown in Fig. 5. As before, the only material found to provide satisfactory operation has been fused quartz. The operating characteristics of the experimental unit were so satisfying that Dr. Klein concluded that further research need only be directed to the designing of commercial units of this type. As a result, the present form of the device is almost identical to this experimental unit. Even now there is a considerable art connected with proper design of the unit, for the greater portion of available design data has been empirical in nature, so that drastic modifications result only from accidents or intuitive concepts.

The electronic gear associated with the unit is conventional, consisting of an ordinary audio power amplifier feeding the screen circuit of a conventional push-pull RF oscillator (see Fig. 6). Experience has shown that proper operation of the unit demands not less than 10 kv RF at 27 mc, with an average current of about 2 milliamperes; thus the corona dissipation is of the order of 20 watts. This power figure should not be construed as a limit, for a number of units have been operated at much lower levels — down to $\frac{1}{2}$ watt, and others have been driven at 1 kilowatt. The only factor to be remembered is that optimum efficiency for each average power level demands modified design characteristics. To eliminate the problems inherent in transmitting such high RF potentials, the power is fed at low voltage along a coaxial cable from the oscillator to the driver-transformer located at one end of the transducer. Here all the elements can be well shielded and hazards are eliminated.

It has been found that none of the operating conditions are critical, and the only criterion that must be met is that the conditions must be such that the corona never extinguishes. This necessitates that the per cent modulation be kept low, the actual values depending upon the remaining parameters. The units have been successfully operated on RF power with frequencies as low as 400 kc and as high as 75 mc. However, it has been found that, at low frequencies, the corona is quite noisy, and at high frequencies the dielectric heating of the quartz walls causes excessive energy loss. The frequency now used was chosen simply because of the availability of components.

Although the actual mechanism of the operation of the transducer is not understood, several patterns have been proposed. One explanation which has met with some favor describes the mechanism as a fluctuating ion cloud. It is supposed that, since observation shows the length of the corona discharge to be a function of the applied voltage, the total number of ions in the corona is also a function of the applied voltage. However, it is postulated that the volumetric density of the ion cloud is essentially constant, so that the effect of a changing voltage is to modify the total volume of the ion cloud in a manner which closely follows the applied voltage. The alternately expanding and contracting ion cloud which would result form the application of a modulated voltage would act as a volumetric piston, causing similar motion of the air column in the throat of the horn. Evidently, the magnitude of the voltage excursion would determine the amplitude of the pressure variation in the air column, and, if the modulation lay in the audio band, it would also determine the intensity of the sound input.

Thus, the volumetric piston of ions can be supposed to replace the solid piston or diaphragm of the conventional loudspeaker. However, the substitution offers numerous advantages, the most important of which is the elimination of resonances in the driving mechanism. Since the ions making up the cloud are almost weightless, the inertia of the cloud to modulation is extremely small, and may, in most cases, be neglected. Experiment has shown that the effect of the ion mass does not enter until frequencies of the order of 1,000 mc are reached, where the energy of the field is, in large part, converted to excitation of the vibrational states of the gas molecules. However, acoustic energies at such frequencies are of little value at present, and cannot be successfully transmitted in gases because of the extreme attenuation. In the range from a few cycles up to well above 40 kc, the response of the transducer faithfully follows the excitation. It is thus evident that a loudspeaker utilizing the element can be constructed without recourse to special design characteristics supposed to eliminate or reduce unwanted response. It is important, however, as in all audio reproduction equipment, that the associated electronic gear introduce a minimum of distortion, for the unit is inherently incapable of correcting these difficulties. It should be evident that the difficulties produced by the nonlinearity of the gaseous medium in regions of high intensity levels cannot be eliminated by the use of the ionic loudspeaker. However, the distortion introduced into conventional systems by air loading are notably absent from the audio output of the Ionophone.

Because of the necessity for high voltage operation, it is not likely that the Ionophone, in its present stage of development, will replace the conventional loudspeaker. It does, however, offer promise of being an ideal high fidelity transducer, and in all likelihood, will find considerable application in sound systems where high audio levels are desirable, since few limitations on the attainable power output exist.

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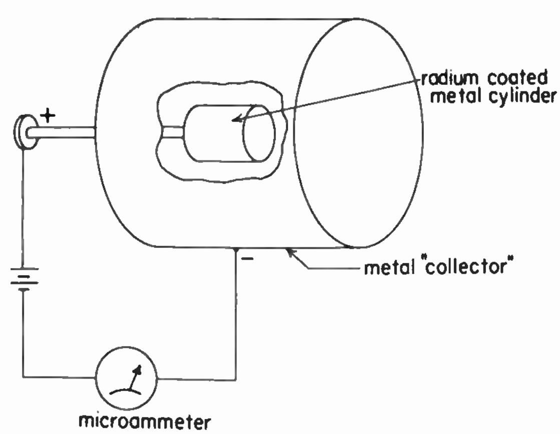

Fig. 1 - Original ionic microphone.

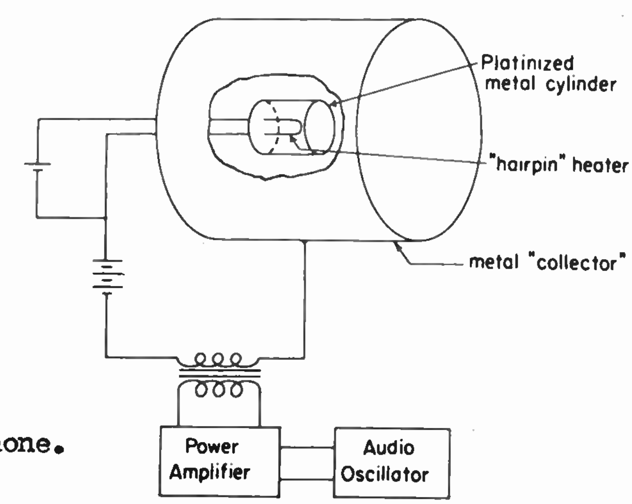

Fig. 2 - Experimental ionophone.

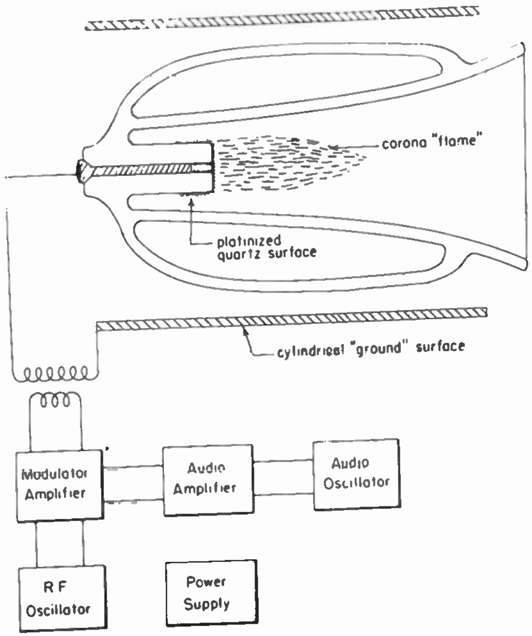

Fig. 3 - First successful ionophone.

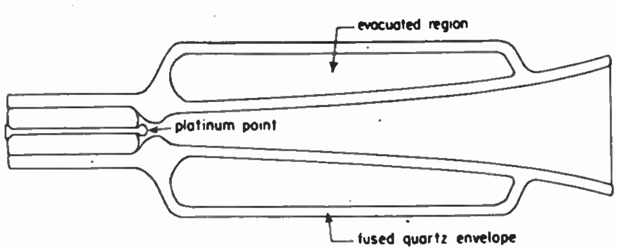

Fig. 4 - Typical ionophone unit.

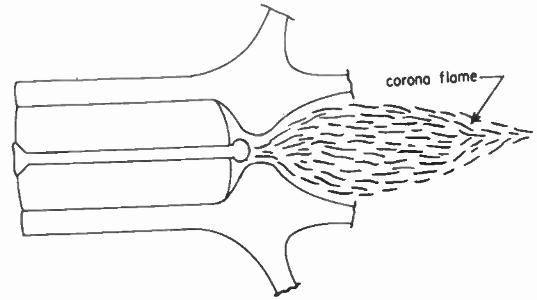

Fig. 5 - Typical corona appearance.

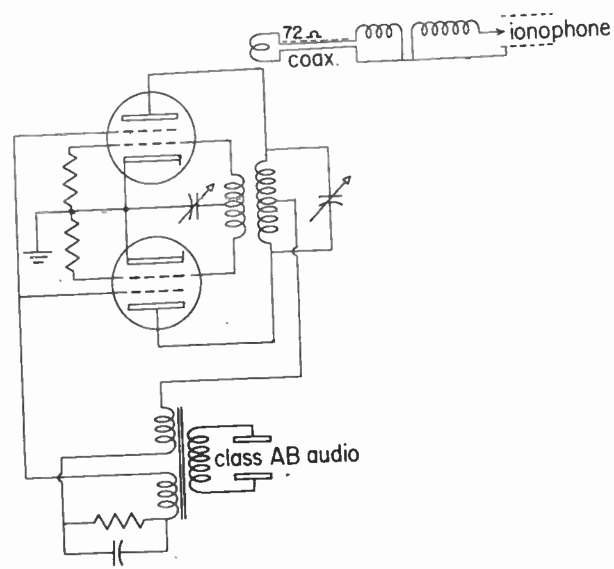

Fig. 6 - Oscillator.

A NEW AUDIO CONTROL CONSOLE FOR TELEVISION AND RADIO*

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In the design of a line of TV studio equipment, a major item is the audio control console. In this time of different types of broadcast services, it is desirable that a console should be capable of meeting the operating requirements of both TV and radio without slighting either service.

Today's broadcast engineers want their audio consoles designed and constructed in the simplest, most dependable manner. Both TV and radio people must keep their maintenance costs down. In addition, TV studio operators would prefer to spend the maximum amount of their time on the picture side of their operation and not have to bother too much with the audio.

The requirements for radio audio consoles are quite well known. When a TV audio console is being designed, many new ideas must be given consideration.

Input Requirements

TV usually requires more mixers, not only because there are added projector inputs, but because the TV operation is inherently more complicated and it becomes confused if there is need for too much input switching during the program. The wide use of magnetic recording, both in radio and in TV, also contributes to the need for more inputs.

TV's pre-amp requirements vary greatly. Fewest are needed to do the master control job, while a TV studio might require a pre-amp for every mixer. This parallels radio somewhat, but the variations in number of pre-amps in different TV installations are generally greater. This leads to the conclusion that some means of installing just the number of pre-amps required for a given job would be very helpful. This problem may be solved by the use of plug-in amplifiers, not only for pre-amps, but for all console requirements.

Single Versus Dual Outputs

TV also has less need for dual output two-channel consoles because there are fewer dual program originations and multiline feeds. On the other hand, many TV audio men still like two program channels for greater protection against equipment failures and for the greater operating flexibility they permit, especially with regard to auditioning, previewing, and cueing without tying up the program monitor channel.

The one-versus-two output dilemma may be resolved by including in the console certain of the second-channel components that will serve a useful function whether the console is feeding a second phone line or not.

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* Manuscript received June 9, 1952.

By using some of the components of the second output channel, it is possible to obtain submaster control of grouped inputs, and to superimpose this material on the outgoing program channel. Inputs may also be routed via this control to a power amplifier feeding a separate speaker in the studio for cue or accompaniment purposes.

It is most convenient and flexible to have two mixer buses, booster amplifiers, and master gain controls included in the basic design. In the presence of these facilities, it is frequently worth while to add the second line amplifier and VU meter to get full dual outputs.

Remote Facilities

TV remote facilities generally bring the program loop into the audio console and the order loop or loops direct to the intercom system. Single line remotes generally are not favored since there is so much traffic during the show.

Thus, a straight TV console could eliminate remote line talkback, override, and cue facilities. However, if the console is to be used in radio service, there will be a demand for the facilities that permit a single loop to be used for program, order, and cue. In either TV or radio, it is handy to be able to route incoming lines to either program or telephone circuits. When separate program and order loops are used, the routing keys may be used to reverse the lines in case of trouble.

Talkback

The booth to studio communication in TV goes on during the show, so it is provided by separate intercom equipment. However, the console may frequently be used for loudspeaker talkback into the studio before the show so the usual radio type talkback circuits will prove useful for both applications. With talkback, of course, go the usual speaker and warning-light relays.

Cabinetry

Today's audio console should have a cabinet that matches TV studio equipment, for there will be many installations with audio and TV equipment side by side. Some TV installations and most all those in radio mount the console by itself so it must function properly and look well under these circumstances.

The audio console is frequently made more useful by the addition of extra facilities housed in a separate cabinet. Therefore, the basic design must be such that extension cabinets may easily be added. Examples of extra facilities include additional mixers, output switching systems, patching facilities, recording panels, etc.

A New Audio Console Design

The new General Electric Type BC-11-A Audio Console has been planned to meet these requirements. The design approach started with the turret cabinet which had been designed to match the General Electric TV studio line. This cabinet was planned in two lengths, the shorter, $19\frac{1}{4}$ inches to take rack mounting panels, and the longer cabinet, just twice this length, or $38\frac{1}{2}$ inches. This latter size looked attractive. The front panel size of 38 x 14 inches was found to mount all of the necessary controls easily (see Fig. 1).

Height and depth were investigated and found to be adequate for audio console design (see Fig. 2). Since the input requirements varied so much, the use of removable amplifiers looked attractive. Since compact, high quality plug-in amplifiers for preamp, line amp, and monitor amplifier service were available, methods of mounting were investigated.

It was concluded that an adequate number of GE Type BA-1-F Preamplifiers and Type BA-12-C Program/Monitor Amplifiers could be installed in the turret, mounted as shown in Fig. 2. Insertion and removal is accomplished from the front with the panel hinged down. The cabinet may be mounted against a wall or window when desired.

Connections

For external connections, plugs similar to those used on the amplifiers were used. Receptaoles for these are mounted just below those for the amplifiers. Access to the lower row of receptacles may be had through the bottom of the cabinet, for external wiring, or through from the front, when the panel is opened and amplifiers removed.

For the first time, a console is available that offers a control panel and cabinet with plug-in mounting facilities to take a standard line of plug-in amplifiers already widely used in rack mounted installations. Of course, the control panel and cabinet only may be used with the same General Electric Plug-In Audio Amplifiers mounted remotely in a rack, or the Desk Unit may operate with any manufacturer's rack-mounted amplifiers.

Provision for plug-in external connections makes installation simple. The cables and plugs may be assembled and wired before the console is available. Then installation consists of placing the desk unit over the cable ducts and plugging the cables in by reaching through the open front. The amplifiers may then be inserted and the console is ready to go on-the-air.

All receptacles for plug-in amplifiers and external connections are mounted just a few inches behind the front panel so that they are accessible for maintenance. This type of construction was also adopted to make it easy for the user to change the arrangement of receptacles and wiring to suit his individual needs. Fig. 3 gives the rear view of the receptacle mounting and shows the location of all amplifiers and customer connections. There is no wiring in the back of the turret cabinet.

Provision is made for easy insertion and removal of all plugs on both amplifiers and external connections. Amplifier plug insertion is facilitated by the proximate location of a turned up lip on the amplifier chassis and a large hole in the top of the receptacle brackets. These devices are located so as to permit the use of a pair of slip-joint pliers to apply extra insertion pressure where necessary. When extra force is needed to remove an amplifier, a screw-driver blade may be used as a wedge between the side of the recepatcle bracket and the lip on the amplifier. Accesss holes in the mounting shelf permit insertion of a screw-driver blade for easy removal of external connection plugs.

Where there is a demand for external patching facilities for the audio console, its plug-in amplifiers may be rack mounted adjacent to the jack field. Connections to the console may be plugged into the amplifier receptacles thus picking up amplifier source and load circuits. This arrangement of console, amplifiers, and jacks reduces wiring between console and rack by 50 per cent over consoles with special internally mounted amplifiers.

When the amplifiers are retained in the console, they may easily be wired to external jack fields by making the external connections at the receptacles and at easily installed terminal strips. Instant access to all connections make on-the-job changes very simple.

Audio Circuits

The circuit arrangement of the General Electric Type BC-11-A Audio Console is shown in Fig. 4. The first seven mixers are wired to take preamplifiers. All or none of these may be used, as fits the particular installation. If it should be desirable to use nine input preamplifiers, the extra ones may be installed in the last two spaces at the opposite end of the cabinet. These preamps could be wired direct to the mixers or to positions on the #8 and #9 mixer keys.

Ahead of mixers #6 and #7 are telephone keys permitting selection of projector inputs or turntable preamplifiers. These preamplifiers may be used for additional studio microphones when reproducer amplification is supplied elsewhere. That wide input flexibility is provided will be realized from inspection of the diagram.

Four telephone key switches are provided ahead of the remote mixer #8 and one switch ahead of the network mixer, #9. Every effort is made to use standard components and open wiring and assembly so that users can modify the circuit of the BC-11-A to meet the special requirements that are bound to be present in almost every station.

In a console with provision for varying input and mixer facilities, some method of varying the marking must be devised. In this console, the panel markings for the input key switches and the mixers are covered with transparent plastic strips. When it is desired to depart from the markings furnished, paper strips marked as desired can be inserted between the plastic and the panel.

Over each mixer is a write-in strip. This is made of satin finish aluminum and can be written on with pencil or crayon and then erased. These strips provide a place to mark "piano", "trumpet", etc. over the appropriate mixers and to change the marking for each show.

Color coded control knobs are also provided to facilitate the operation of the console. Two blue mixer knobs are associated with the blue handled turntable/projector selector keys. The red mixer knob is associated with the four remote red handled key switches, and the white mixer knob with the white handled network selector key switch.

Inspecting the program circuit, it will be found that standard design methods have been followed incorporating a booster amplifier, master gain control, and line amplifier. This latter has ample power output to feed the phone lines through the 6-db loss deltawye dividing network without any clipping of peaks, even under momentary overload conditions.

The three line outputs may be used as follows: L1A may feed the transmitter through the regular phone line, L1B may feed the transmitter through spare phone line, and L2 may, when activated, feed another station, a recorder, or network, etc. To get an output from line 2, a takeoff from the monitor amplifier is provided, or a second line amplifier may be plugged into the console. In this latter case, full two-channel operation is provided.

For VU meter indication of the program peaks, either one or two VU meters may be installed. As shipped, the BC-11-A has one VU meter. To add a second VU meter, a meter similar to the one shipped with the console is obtained. Then the two meters may be installed using the two-VU meter plate furnished a as a part of each console. The second VU meter range selector may be a fixed pad of the desired loss. For instance, if the second VU meter is to run at + 14 VU to indicate line peaks of + 8 VU at the output of the 6-db line pad, then a series resistor of 3,600 ohms ahead of a 10-db 3900/3900 T pad should feed the meter. This will give the proper loss and provide standard 600-ohm VU meter impedances of 7,500 ohms across the line and 3,900 ohms feeding the meter.

The monitor channel may be switched to monitor program output or it may be used to preview mixer bus #2. This latter bus is provided with a booster amplifier the same as bus #1. The output of #2 booster feeds the monitor amplifier selector switch and the submaster gain control. This control may be used to feed mixer bus #2 back into bus #1, thus providing for submaster gain control of grouped inputs.

When the BC-11-A Audio Console is equipped with two line amplifiers, the submaster control becomes channel 2 master gain control for full two-channel program operation.

The output of booster amplifier #2 is brought out for connections to an external monitor amplifier to feed cue material into the studio when desired. This studio accompaniment feed can provide recorded music to cue dancers, provide special effects, etc. By closing the submaster switch, this material may be superimposed in the regular program.

The monitor amplifier and program amplifier are both Type BA-12-C Amplifiers. These are equipped with Hi-Lo gain switch which changes the feedback from about 6 db in the Hi gain position (rated gain 71 db) to about 21 db in the Lo gain position (rated gain 56 db). The single line diagram levels were obtained with the program amplifier in Lo gain and the Monitor Amplifier in Hi gain position. The program channel distortion rating is kept within 1 per cent by the use of the full feedback. In the monitor channel, advantage is taken of the higher gain, yet the distortion rating is under 3 per cent. The 6 db of monitor feedback provides an internal output impedance of the BA-12-C sufficiently low to insure good speaker damping.

Control Circuits

The first five mixer key switches are provided with a shielded control section. This is normally wired to the last receptacle on the right end of the console for use with a plug-in FA-45-A Relay Chassis. This chassis contains two relays for control of speakers or warning lights. It has coils that operate from 24 V dc or 115 V ac. Power supply for relays must be supplied by the user.

Since the BC-11-A control circuits are not tied down to any particular types of relays or power supplies, other relay assemblies may be used or the user may wire in relays of his own choice. Another usable relay assembly is the GE Type FA-20-A which operates from 48 V dc. This is a flat plate chassis which mounts in a wall pull box and controls speakers and light circuits.

There will be many installations where 115 V ac may be used to operate the relays since the shielded key contacts provide adequetely low cross talk from 60 cycles except where noise levels must be kept unusually low. When line voltage supply is used to drive the relays, it should be wired through a fused isolation transformer.

Another way to eliminate the need for relays and their dc relay supply is to interrupt the speaker feeds directly on the key contacts. If 16-ohms impedance or lower, one side grounded, is used, cross talk is again at a tolerable low value. To facilitate voice coil impedance connections, the monitor output is equipped to feed 16-, 8-, and 3.2-ohm loads, in addition to 600 ohms. This output may be run balanced or unbalanced, but the latter is recommended for speaker cut-off when interrupting only one side of the circuit, as is done in the BC-11-A and accessory relays.

Either the line relay power or direct voice coil circuit switching are simplifications that may prove satisfactory at many studios. It is not advised that warning-lamp power be switched directly on the key contacts. The control contacts on the first five mixer keys may be wired by other arrangements to suit a variety of requirements. The philosophy of furnishing standard components that may be easily rewired to suit local situations has been followed as much as possible in the design of this console.

Power Supply

The power supply furnished with the BC-11-A is a single Type BP-10-B plug-in unit, with matching tray Type FA-22-F. This supply may be rack mounted using a GE Type FA-23-A Shelf or it may be mounted in a wall cabinet or in any out-of-the-way place that is convenient to the user. The BP-10-B is a simple capacitor input filter unit with a stage of LC filter feeding the load. Two 5Y3 tubes are operated in parallel as full wave rectifiers.

Use of such a simple supply is made possible by the design of the plug-in amplifiers, each of which will tolerate reasonably high ripple because of its tertiary feedback hum cancelling circuit. This type of feedback also minimizes B+ cross talk and motorboating.

The BP-10-B incorporates an output voltage adjusting resistor which may be set to provide the 300 volts output regardless of the number of amplifiers used in the desk unit. The power supply also provides +30 volts filament bias which is needed to minimize hum output from the tubes.

The BC-11-A incorporates a power transfer relay for automatic cut-over to a second power supply in case of B+ failure in the regular supply. In this case, two BP-10-B supplies would be used, and could conveniently mount on one FA-23-A Shelf.

When two program amplifiers are used, the B+ drain exceeds the rating of one BP-10-B. In this case, the second emergency power supply may be used to feed program channel 2, as shown in Fig. 5. This arrangement automatically transfers all preamps to the second power supply in case of trouble in #1 supply. It provides for normal two-channel operation, and complete operation through channel 2 in an emergency.

The General Electric Type BC-11-A Audio Console can be purchased with or without amplifiers and power supply. In the latter case, the Type BC-12-A Desk Unit is specified. With amplifiers it starts as a Model 4BC11A7 with one input preamp and ends up as a Model 4BC11A7 with seven input preamps. All BC-11-A models also include two BA-1-F booster preamps, two BA-12-C Program/Monitor Amplifiers, one BP-10-B Power Supply, and one Tray Type FA-22-F. Each component part is complete with one set of tubes.

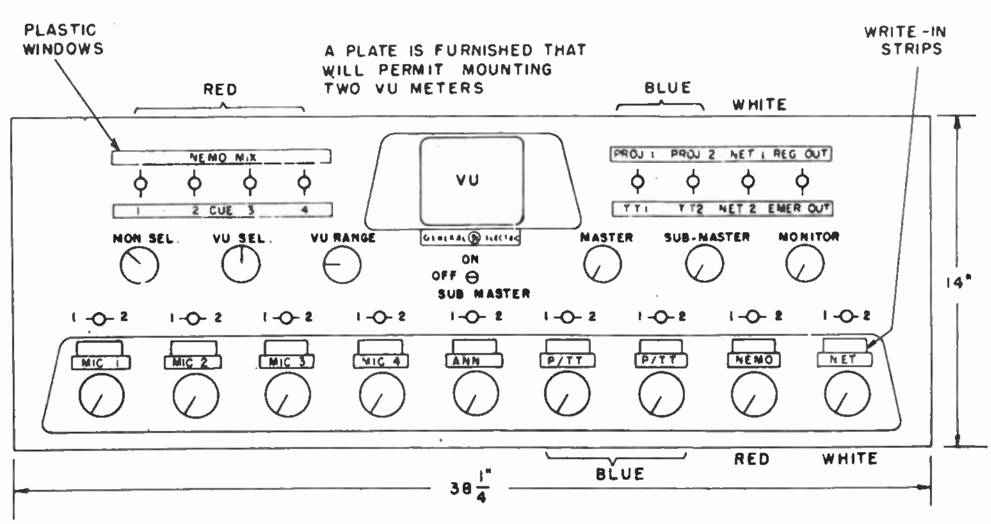

Fig. 1

Front panel and controls, audio console, General Electric Type BC-11-A.

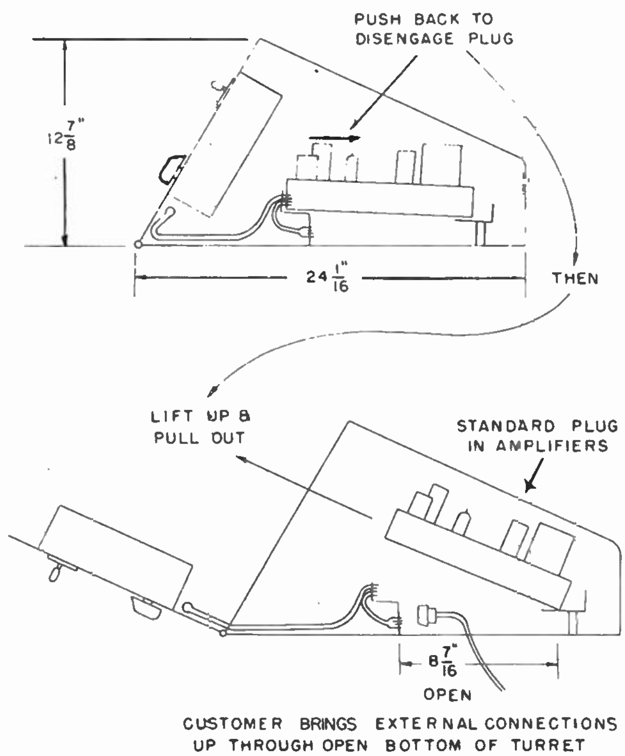

Fig. 2

Positioning of plug-in amplifier, audio console, General Electric Type BC-11-A.

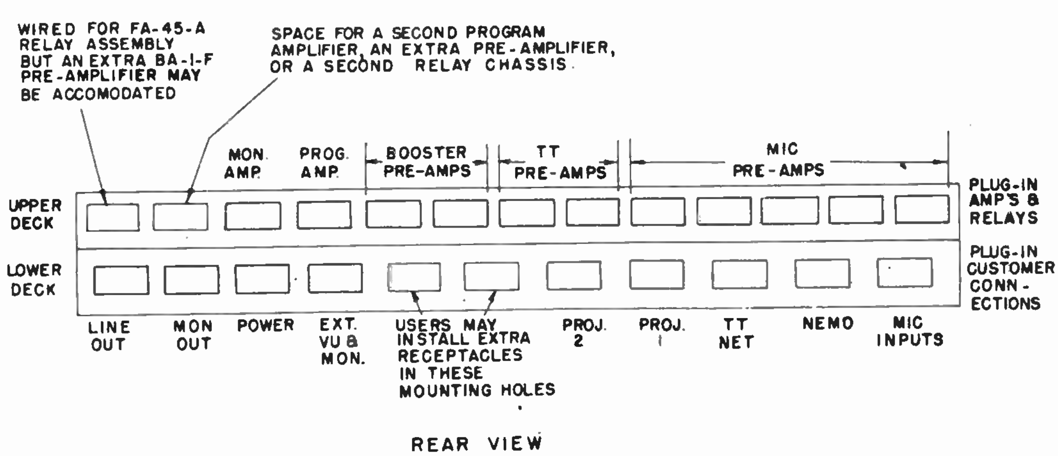

Fig. 3

Arrangement of amplifiers and plug connections, audio console, General Electric Type BC-11-A.

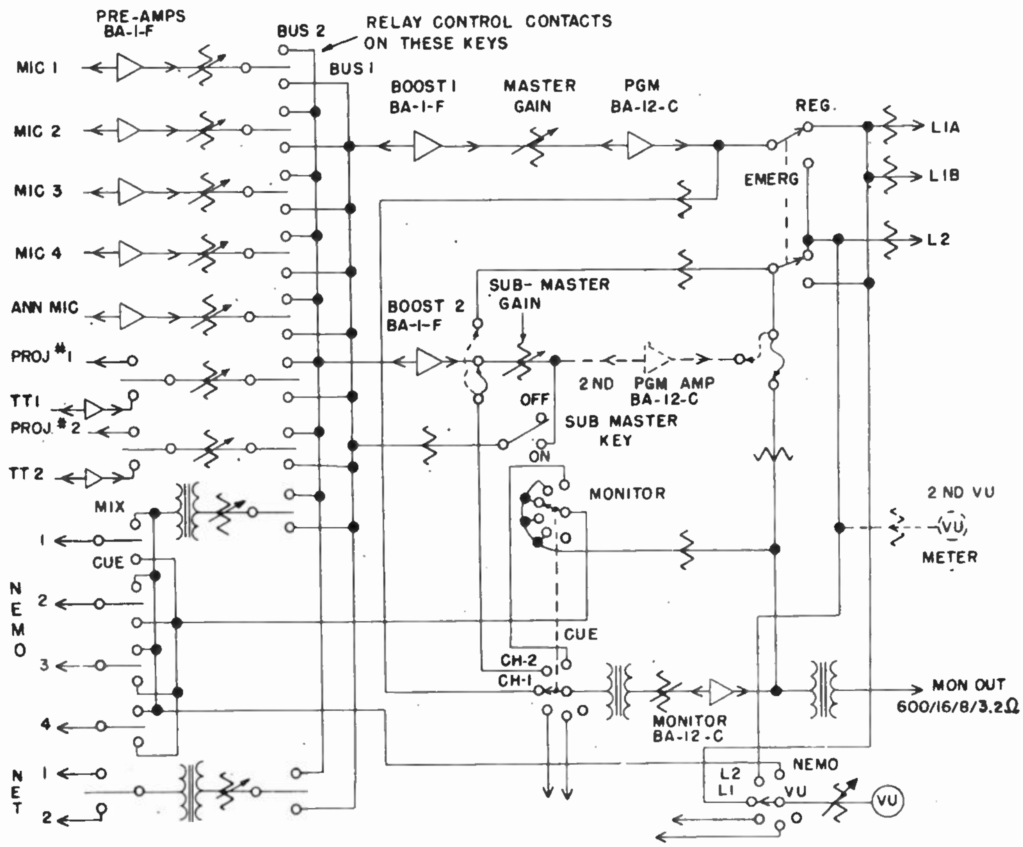

Fig. 4 - Line diagram of General Electric Type BC-11-A audio console.

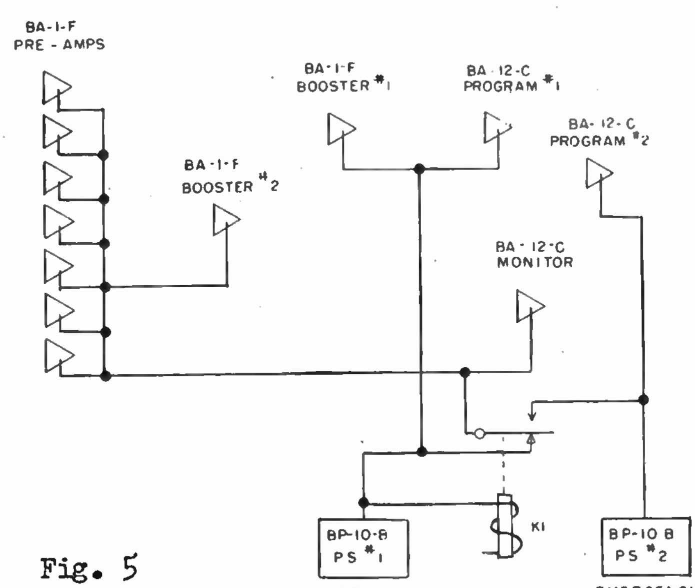

Fig. 5

Power distribution for dual-channel operation, audio console, General Electric Type BC-11-A.

A PHASE AND GAIN METER*

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In accordance with the PGA policy to encourage student activities, the TRANSACTIONS is glad to publish this paper presented in the Student Papers Competition and awarded a prize by the Houston Section of the Institute of Radio Engineers on May 15, 1952. Readers of TRANSACTIONS will find this material novel and interesting.
— Editorial Committee

Introduction

The Phase and Gain Meter described here is a device permitting rapid and accurate measurement of the phase shift and gain of an audio amplifier or other three-terminal network.

Gain is calculated from measurement of input and output voltages and phase is determined by vector relations in the input, output, and phase-shift voltage triangle. Simple adjustments of the meter controls permit both phase and gain to be read directly from the recalibrated scale of a conventional vacuum tube voltmeter. Convenience and accuracy are provided by arrangement of the controls and the ease of adjustments.

Description

When gain and phase shift measurements are made on an audio amplifier, a considerable amount of time is often used in connecting the equipment and the process of taking data is long and tedious. Vacuum tube voltmenters are used to measure input and output voltages and a cathode-ray oscilloscope is usually used to indicate phase shift. Results of such phase measurements are likely to be inaccurate and are difficult to obtain at best. This Phase and Gain Meter was constructed to provide greatly improved accuracy of phase measurement and greater ease of use.

As usual, gain is determined by measurement of input and output voltages and calculation of their ratio. Often, it is possible to normalize the input to 1.00 volt or some decimal multiple of this in which case the significant figures in the gain ratio can be read directly in the measured output voltage. A single voltmeter is used in this particular device for all measurements, and the different voltages to be measured are chosen by a selector switch. Once set up, it is not necessary to change any connections in the circuit except by use of this selector switch.

Phase shift is determined by a vector triangle method. The input voltage is used as a reference and a voltage divider (output potentiometer) is

* Manuscript received June 9, 1952.

adjusted to provide another voltage equal in magnitude to this input but in phase coincidence with the output voltage. This normalized output voltage vector and the input voltage vector must both terminate on a circular locus, as shown in Fig. 1. If the voltage between the normalized "high" potential input and output terminals is measured, this reading determines the magnitude of the chord V_ϕ connecting the termini of these two vectors. Thus, by simple measurements of magnitudes, the angular difference in input and output voltages can be obtained.

It is not necessary to employ any graphical techniques to determine the magnitude of this phase shift. Inspection of Fig. 1 shows that the perpendicular bisector of this phase voltage vector bisects the angle of phase shift when input and output voltages are normalized. Therefore:

$$V_\phi / V_{in} = 2 \sin (\theta/2) \tag{1}$$

and the scale of the voltmeter can be marked off directly in degrees phase shift.

One of the difficulties in interpretation which arises is distinguishing between leading or lagging phase shifts. This difficulty can be avoided by adding a capacitive network which can be switched across the normalized output voltage. This network will introduce a lag in the circuit when the lead-lag switch is closed and will, in effect, rotate the output voltage clockwise.

With the phase reversal switch in the "off" position, and with a leading phase angle, the capacitive network will cause a drop in the phase voltage because the clockwise rotation effect will tend to move the original output voltage closer to the reference voltage. With the phase switch in the "off" position, and with a lagging phase angle, the capacitive network will cause an increase in the phase voltage because the clockwise rotation effect will tend to move the original output away from the reference voltage. Uncertainty exists, however, near 180° phase shift because the shunting effect of the capacitive network may overshadow the change in phase voltage due to the introduced phase shift. This uncertainty has been avoided by addition of a phase reversal network.

Accuracy of phase measurement using this method is dependent upon the accuracy of adjusting and reading magnitudes of voltages. The accuracy to be expected is proportional to the cosine of ½ the angle of phase shift, and so would be poor in the vicinity of 180°. The region about 180°, however, is most important in amplifier phase measurements. Accuracy here can be improved considerably by use of a phase reversing network so that the supplement of the phase shift angle is measured. This is used for phase shifts in the range from about 95° to about 265°.

Phase reversal is obtained by applying a portion of the output voltage of the test unit to a single-stage amplifier. The output from this phase inverting amplifier then is compared to the input voltage for phase measurements. Since any phase shift in this stage would be added to that of the test unit, it has been designed so as to keep this phase shift small. To accomplish this, the plate load resistor is made small, cathode feedback is used, and direct coupling is employed. This is feasible because of the small input signal needed and near-unity gain desired. The regulated power supply used to provide the necessary plate and screen potentials of this amplifier is responsible for most of the moderate bulk of the unit.

A vacuum tube voltmeter should be used to measure these voltages so as to present a high input impedance. Some consideration was necessary to assure that the meter would not introduce significant error, especially in phase measurements. Any shunting across the output potentiometer would result in inaccurate normalization of the output voltage so the total resistance of this potentiometer was put at $\frac{1}{2}$ megohm. This was considered sufficiently small since the large input impedance of the vacuum tube voltmeter is shunted across only a small portion of this total resistance.

Most vacuum tube voltmeters have a sizable capacity from the "low potential" terminal to ground, while the impedance from "high potential" terminal to ground is several megohms. The measurement of the "phase" voltage is taken with the low terminal connected to the input terminal and the high terminal connected to the output terminal. By connecting in this way, the larger "low-terminal-to-ground" capacitance is shunted across the low resistance input potentiometer where it will cause negligible change in circuit conditions. Reversal of voltmeter connection has been found to yield phase shift readings too small at high frequencies while error is usually negligible for the connections specified. This error can be reduced further, if necessary, by use of a smaller input potentiometer.

Operation

1. Connect a variable frequency audio oscillator to "input" terminals of the meter and to the input terminals of the amplifier under test.

2. Connect the output terminals of the amplifier under test to "output" terminals of the meter.

3. Connect a vacuum tube voltmeter to the "meter" terminals with proper regard to polarity.

4. Adjust frequency to desired value.

5. Make adjustments and take readings thus:

Switch Position	Adjust	Read
(a) 1	Audio Oscillator output	Desired input voltage (usually decimal multiple of unity)
(b) 2	None	Output voltage (gain)
(c) 3	Output Pot.	Normalized output voltage. This is set equal to input voltage.
(d) 4	None	Phase voltage or phase shift (if so calibrated)

(e) 5	Output Pot.	Adjust to equal output voltage of position 2 (use if gain less than unity)
(f) 6	None	Phase voltage or phase shift (if gain less than unity)

6. Push lead-lag switch to determine sense of phase shift.

Table I

Voltage	Switch Position			
	off		on*	
	> 90°	< 90°	> 90°	< 90°
Decrease	Lead	Lead		Lag
Increase	Lag	Lag		Lead

* If the phase angle magnitude is between 95° and 180°, throw the phase reversal switch to the "on" position.

Conclusion

The phase and gain meter is very useful in the measurement of phase and gain of an audio amplifier. With the phase reversal network and the lead-lag control incorporated in the meter, it is relatively easy and accurate to determine the phase shift through an amplifier or three-terminal network.

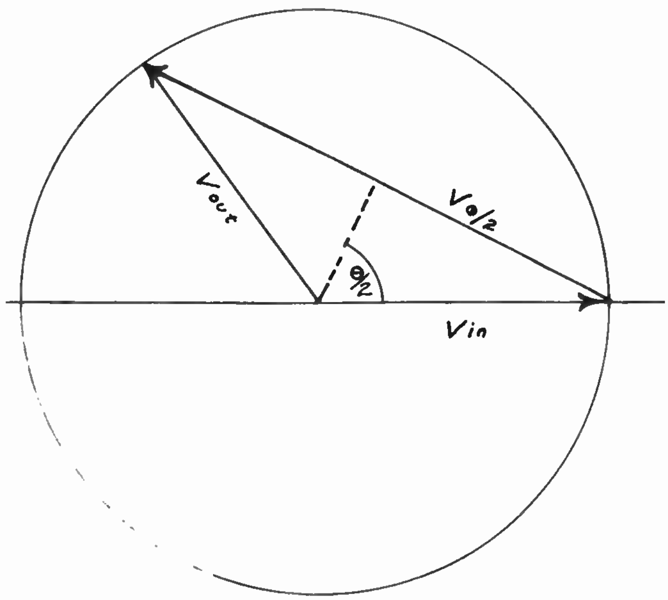

Fig. 1

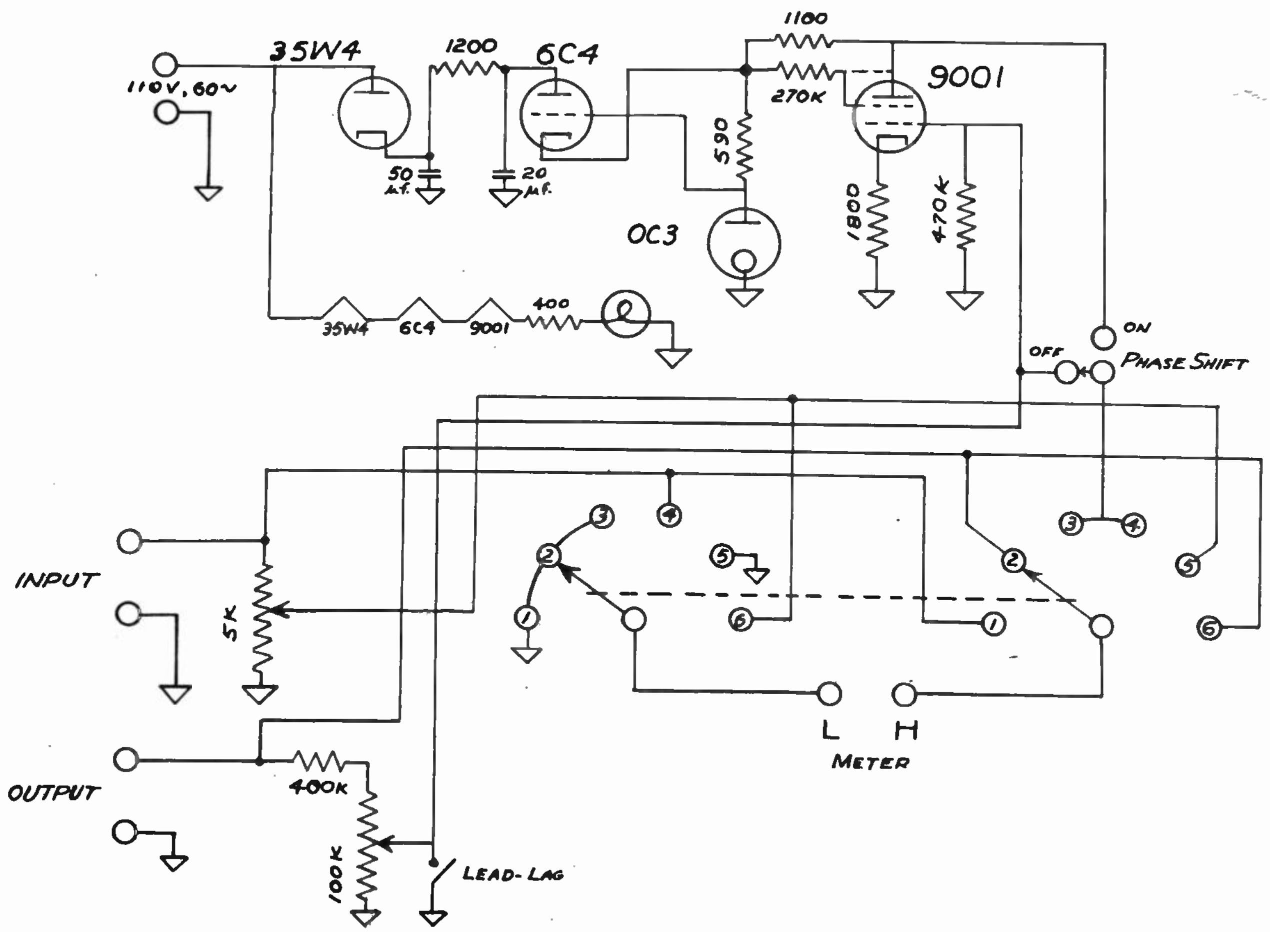
35W4
1200
6C4
1100
270K
9001
110V, 60~
50 µf.
20 µf.
590
1800
470K
OC3
35W4
6C4
9001
400
ON
OFF
PHASE SHIFT
INPUT
5K
OUTPUT
400K
100K
LEAD-LAG
L
H
METER

Fig. 2

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Transactions of the I·R·E

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

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A CONVERSATION ON LOUDSPEAKER ENCLOSURES*

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The Baldwin Company
Cincinnati, Ohio

In this invited Editorial, written in a humorous vein, Dr. Martin indicates some of the sources of frustration of the Audio Engineer
— Editorial Committee —

Every engineer in the audio profession is at some time asked by a non-technical friend for advice on the construction of an ideal enclosure for a loudspeaker of somewhat dubious background and performance. One has two professional choices. He can pose as an oracle and write a complete prescription as if there were an unique solution. Or, he can try the educational approach and explain a few basic principles of cabinet design. Although the latter course is basically more honest, it does have its difficulties, as illustrated in the following conversation.

AL: "Say, Paul, I just picked up a fifteen-inch speaker at a bargain sale down at Joe's Radio Shop. It only cost me a dollar fifty-nine."

PAUL: "What kind did you get, Al?"

AL: "I don't know what it is, and there's no label on it, but I recalled that you were a member of the IRE Professional Group on Audio, so I came over to get some advice on how to build a really good cabinet."

PAUL: "My first advice would be to get a really good loudspeaker unit."

AL: "I'm afraid that would cost too much. This one will probably be good enough. I want to build just the right cabinet for a fifteen-inch speaker. Can you tell me exactly how to do it?"

PAUL: "There's no unique solution to your problem. It's something like building a house. The architect's solution depends upon the size and shape of the lot, the location of the street, the contour of the ground, the type of occupants and what they do, and the choice of materials which are both suitable and economical. The architect would want to know how large a lot you have. And I would ask, 'Do you have ten cubic feet or more available for the cabinet?'

AL: "Ten cubic feet? That sounds awfully big."

PAUL: "I know, but ten cubic feet isn't as large as it sounds. A cube twenty-six inches on an edge would give you the volume I suggest. Of course, you wouldn't want all of the dimensions the same, for acoustical and functional reasons, or even for appearance."

AL: "Why not?"

* Manuscript received August 27, 1951.

PAUL: "Well, the acoustical result would be to bunch the box resonances together instead of spreading them along the frequency scale. This is a principle followed in the acoustical design of small rooms. Again there is no magic, unique solution, but I'd suggest dimensions of 20 x 26 x 33 inches as a starting point. You may deviate from those dimensions somewhat, and if you don't like the proportions of this combination, I'll calculate another good combination to approximate the proportions you'd prefer."

AL: "I don't understand this sort of approach. When I buy a vacuum tube, the tube manual gives circuit values to use. Why aren't your recommendations exact?"

PAUL: "There are several reasons, Al. In the first place, the tube manual values are not sacrosanct. They apply to only one set of operating conditions. Besides, the values suggested by the manual correspond to a particular type of tube. You didn't tell me anything about your loudspeaker, except that it has a fifteen-inch diameter. You wouldn't expect to find in the tube manual a set of suggested values for components in the circuit of just any triode. Some of the loudspeaker manufacturers now provide, with their units, an information sheet recommending cabinet volume, shape, and even complete design data on a cabinet known to give good performance with that loudspeaker."

AL: "The dimensions you gave are close to what I had in mind. I can put the record player, a radio, and some shelves for record albums in the cabinet, and use the space to advantage. Maybe I can leave enough space for a secret compartment."

PAUL: "Some of the things you would put in the compartment might rattle, Al, and you have misunderstood the purpose of the space behind the loudspeaker. If you fill up all of the space, the loudspeaker cone will be stiffened acoustically and the low-frequency response will be lost."

AL: "Maybe it would be better not to have a cabinet if it would make the speaker sound worse."

PAUL: "No, the loudspeaker without some sort of baffle wouldn't give you much low-frequency response either. Let's go back to the ten cubic feet of enclosed space behind the cone."

AL: "I can't use that much. I don't have enough material."

PAUL: "Oh, you already have the material. What kind did you get?"

AL: "It's a large sheet of quarter-inch fibre-board that was left over when my neighbor was insulating his attic."

PAUL: "I'm afraid it won't be rigid enough for good low-frequency response, Al. You should get some three-quarter inch plywood."

AL: "I have some, but I'm going to use it for a ping-pong table. Well, thanks, Paul, for all of your advice. It's been very helpful. I never would have thought there was so much theory to building a speaker cabinet. I'll let you know how it works when I get it built. Thanks again."

****

JOE: "Al, did you ever hook up that speaker you bought at my annual bargain sale last year?"

AL: "Yes, would you like to hear it?"

JOE: "Say, that sounds great! What kind of a cabinet did you use?"

AL: "Paul helped me design it, I never did get it completely finished. I think he said to make the front panel three foot square. I finished that much and mounted the speaker, then set it on edge back there in the corner against the walls. It sounded pretty good when I hooked it up, so I just made a triangular top panel to pile things on, and left it there. Everyone likes it because it has so much bass. Paul certainly was a big help to me in the cabinet design. He's a nice guy, and what a brain! I couldn't understand him most of the time. He kept talking about low-frequency response."

THE TAPESCRIPTS COMMITTEE OF THE IRE-PGA*
Presents the Following Titles of Recorded Technical Papers

Andrew B. Jacobsen
University of Washington
Seattle, Washington

These papers are recorded on magnetic tape at $7\frac{1}{2}$ inches per second on 7-inch reels, and slides are $3\frac{1}{4}$ x 4 inches. Material will be sent express collect and must be promptly returned, prepaid.

"A Single Ended PP Audio Amplifier," Arnold Peterson and D.B. Sinclair, General Radio. 50 minutes. Proc. I.R.E., January, 1952.

"A Sound Survey Meter", Arnold Peterson, General Radio. 20-minute abstract. Proc. I.R.E., February, 1952.

"Microphones for Measurement of Sound-Pressure Levels of High Intensity over Wide-Frequency Ranges", J.K. Hilliard, Altec Lansing. 20-minute abstract. Proc. I.R.E., p. 214; February, 1952.

"A Method for Measuring the Changes Introduced in Recorded Time Interval by a Recorder Reproducer", J.F. Sweeney, Department of Defense. 15 minutes.

"An Instrument for Measuring the Time-Displacement Error of Recorders", E.N. Dingley, Jr., Department of Defense. 15 minutes. Proc. I.R.E., p. 214; February, 1952. (Dingley and Sweeney papers should be used together).

"Application of Electric-Circuit Analogies to Loudspeaker Design Problems", B.N. Locanthi, California Institute of Technology. 30-minute abstract. Proc. I.R.E., p. 214; February, 1952.

"Model 2-C Idiosynchrovisor", J.M. Henry and E.R. Moore, Boston, Mass. 11 minutes. A nonsensical satire on technical papers and jargon of specification writing. Will help to enliven dry technical meetings.

"A Block Diagram Approach to Network Analysis", T.M. Stout, University of Washington, Seattle, Washington. 30 minutes. Abstract as follows:

By treating voltages and currents as signals which may be applied to blocks representing network impedances or admittances, the block diagram techniques now used to analyze control systems can be applied in strictly electrical network problems. The basic rules are presented and a number of applications to ladder networks, parallel and lattice networks, and simple amplifiers are given. Advantages of the procedure include simplification of the labor required in solving network problems, reduced possibility of mistakes, and an increased appreciation of the role of various circuit elements in determining transient or frequency response.

When using recorded papers, the program chairman should preview material and be prepared to answer questions. Address requests to: Andrew B. Jacobsen, Electrical Engineering Department, University of Washington, Seattle 5, Washington.

* Received July 21, 1952.

P.G.A. PEOPLE

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From 1936 until mid-1951 he was associated with the Stromberg-Carlson Company, Rochester, New York; first as Engineer in the Telephone Laboratory; then from 1941 through 1946 as Assistant to the Vice-President in charge of Engineering; and from 1946 to 1951 as Chief Engineer of the Sound Equipment Division. He has recently accepted a position as Chief Engineer of the Caledonia Electronics and Transformer Corporation in Caledonia, New York.

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SPEAKERS AND TRANSMISSION OF SOUND WAVES*

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Summary

Considerations of this discussion involve the over-all design requirements for coupling devices to the air medium. The requirements are considered from the standpoint of the medium itself rather than reverse from the standpoint of the motor mechanism. Considering the requirements for linearity and coupling to this medium, certain speaker designs are evolved. Test speakers utilizing this information have been made which should bring speaker performance much closer to the quality of presently available amplifiers.

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The fundamental purpose of a sound reproducing system is to cause an air pressure wave to exist in the exact form and amplitude of the original, or which would exist if the observations were made at the position of the microphone.

To build a system approaching this ideal requires consideration of the following:

(a) The nature of hearing or how the ear hears.
(b) The intensity of the sounds, both instantaneous and average.
(c) Frequency or frequencies of the sounds.
(d) The characteristics of the medium carrying the sound (the air).
(e) The environment: the effect of the rooms, etc.

Involved in the reproduction of sound waves are additional factors of consideration such as:

(f) Coupling to the air medium.
(g) Resonance characteristics of the acoustic system, including the enclosure and/or horns.
(h) Frequency modulation and Doppler distortion.
(i) Directivity characteristics of speakers.
(j) Compression characteristics of air and acoustical distortion effects.

A great deal has been written about the nature of speech and music, the intensity and average power values, by Bell Laboratories and many other organizations. Conservation in communication requires a study and application of the facts concerning speech and music to provide the best service for the required cost. This means that many communications systems have purposely compromised and limited the transmission to effectively restrict the frequency range and the volume range. Distortion has been permitted. The limits to which the practice has gone is a function of the development

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of the art and the cost of extended accomodation.

Since better equipment and more experience have been available, we find the old standards are no longer acceptable to the critical ear, especially if a truly improved system has become available for him to listen to. It seems reasonable, therefore, that despite certain limitations in the transmission media, a re-investigation into the requirements for faultless transmission might be worth while.

Accordingly, this discussion describes factors concerning the media through which the sound travels and determines from the limitations of these media what must be done to satisfy distortion limits as indicated by the critical ear in order to develop the illusion that the real sound source is present.

There has been a great deal written about the method or physics of hearing. The ear is a remarkable device having astounding accommodation and sensitivity. For instance, the ear at middle frequencies can hear a steady-state signal 10^{-13} acoustic watts power per square foot and can accommodate a steady sound level 140 db above threshold, or 10 watts per square foot at the ear. This is a power range of 10^{14}, or 100,000,000,000,000 (one hundred trillion) to 1. It will accommodate considerably more than this power value if the duration of the sound is short, perhaps 100 or 1,000 times as much. For instance, one who is hammering on steel or hard surfaces generates peaks of this higher order. The ear absorbs the sound wave in the fluid of the inner ear. It therefore acts as an integrator, or sums the energy under the curve, and can therefore stand much more intensity for short nonrepetitive periods than for continuous periods. All the known auditory curves showing "pain or feeling" contours are based on continuous sine wave tones for the average ear. The possibilities of greater tolerance for short duration peaks, such as are found in speech and music, are very real indeed.

This facility of the ear has been a great asset to the communications field because through the suggestion of sounds present, the ear and mind will seek to fill the gaps and help give the illusion of reality by reaching out and listening for the notes not present at the original volume, but this is not accomplished without loss or fatigue. Only with a balanced reproduction of adequate sound level, free of distortion, does one realize how great is the loss of an inadequate system. It is almost impossible to describe why "bi aural" sounds more realistic; however, it is apparent instantly when you listen. A third channel adds still more realism since it permits the illusion of depth as well as the two-dimensional concept.

The musical instruments and other sound devices determine the design requirements of speakers and amplifiers alike, and for realism, the reproduced sounds must be at equal intensity at the ear, as would occur if the listener were at the position of the microphone. This does not mean that the absolute sound power must be the same in the home as it is in the auditorium since there is a room gain to be considered, and the directivity index of the speaker system reduces the power requirement. Also, the listener is usually closer to the speaker in his home than he is in the auditorium. The inefficiency of speakers, as will be seen, however, does almost completely compensate for the gain factors and therefore the power requirement for the speakers is pretty well determined by the power capacity of the individual instruments for medium places.

Since the ear does not recognize loudness on the basis of the peak intensity of the sound signals, it is found that very little average power is required to produce adequate sound loudness. However, since the peaks of speech and music are from 200 to 900 times the average value, as shown by Sacia[4] and others, it is necessary to provide capacity in both speaker and amplifier to handle these peaks if true reproduction and low distortion are to be expected.

A bass drum delivers an acoustic peak power of some 100 to 200 watts, even down to 20 cycles. A cymbal crash delivers from 24 to 48 acoustic peak watts up to 15,000 to 20,000 cycles. The other musical instruments vary in peak intensity from a few milliwatts to 30 or 40 peak acoustic watts.

Since the musical and other instruments create sounds of substantial intensity at all audible frequencies, it is obvious that amplifiers and speakers alike must handle all the frequencies of the audio spectrum. Nearly everyone agrees that this is necessary.

Now, as to the methods of transmission, the coupling to the air and transient response — this has been perhaps the most underemphasized phase of the transmission of sound energy, yet a great deal is known about it. The air is a compressible medium. There are definite pressure limits if acceptable linearity is not to be exceeded. Coupling requirements do vary with frequency and the amount of air that it is necessary to move is more than 1,000 times at 20 cycles than at 20 kc for equal power outputs. "Doppler" distortion may be the major annoying distortion if too low and too high a frequency is present on the same surface, or the over-compression of the air may introduce 70 to 80 per cent distortion.

The design of an adequate system is one of making the best compromise of conflicting factors to satisfy the over-all design requirements. It is believed feasible to design a practical system having no more than 1 or 2 per cent distortion at the desired operating levels.

Now let us see what must be done so as not to exceed these limits of distortion,first considering the physics of the air, and then some aspects of speaker design to deliver the proper power at low distortion.

Since the air is the medium through which the sound must travel, the pressure wave generated in the air and transmitted to the ear is usually developed by some motion of an electromechanical device known as the loudspeaker. A recent unique development utilizes the ionized air to generate this motion without the aid of a diaphragm. Other speaker types consist of a compressed air column or flow which is modulated to generate sound waves. Regardless of the form or the nature of the source, a certain pressure must be developed to generate a specified sensation level in the air at a given distance.

Fig. 1 shows the acoustic power required from a point source at specified distances to develop the indicated intensities in db above threshold —

-10^{-16} watts per square centimeter, or 10^{-13} watts per square foot, approximately.

It will be seen that 100 acoustic watts are required to deliver +120 db above threshold at only 10 feet from the point source, and that 450 watts (acoustic) are required for the same level at 20 feet. It is not unrealistic to consider this as a minimum peak sound level for design purposes. These figures of acoustic power requirements may be reduced somewhat by factors of room gain and directivity characteristics. These will be discussed late.

At higher frequencies where accelerations are higher, the air appears to be more solid, or of a more rigid medium than it does at low frequencies. Consequently, very much smaller areas for the high frequency speaker are required for satisfactory radiated acoustic powers. If a very small speaker is utilized at the low frequencies, then an amazing amount of amplitude is required for the same power as radiated at the very high frequencies. As an example, it is found that the density of air times the sound velocity divided by the radiation resistance of speaker per unit area ($\rho c/R_a = 1$) will just equal 1 at some certain frequency and all frequencies above that. Below this critical frequency, the radiation falls off approximately 12 db per octave and the amplitude required below this point is inversely proportional to the square of the frequency. Whereas, above this frequency, the amplitude varies inversely with the frequency. It is desirable that areas of speakers satisfy this condition of $\rho c/R_a = 1$.

Fig. 2 shows that approximately 252 square feet of diaphragm would be required to satisfy this condition at 20 cycles. A 15-inch speaker satisfies this condition down to a frequency of 275 cycles. It is evident, therefore, that some compromise must be made to make something practical and sufficiently efficient. As can be seen from the above, the low frequencies in audio reproduction systems have been pretty small compared to the actual needs, and various gadgets of various kinds have been resorted to in an attempt to raise the efficiency at the low end, and for the most part, with only minor success.

The situation is similar to that of radiating antennas and corresponds to the aperture effect of an antenna. It can be said that there is no real substitute for aperture , either in loudspeakers or in antennas. However, good efficiency is obtained from antennas roughly 30° in height, and the same general logic would apply to loudspeakers. Consequently, investigations were conducted to determine amplitude requirements for speakers having practical areas.

One approach to solving the low frequency problem has been to utilize a horn of suitable length and mouth cross sectional area. The difficulty in horn design is that if it is driven from a 12- to 15-inch speaker, it should have a long axial length and a large mouth opening in order to satisfy the requirements at the lowest frequencies.

As a matter of comparison, the size of the mouth opening for an ideal horn and the area in square feet for the direct radiator diaphragm are equal for the same coupling effect to the air. It is obvious that such a horn would be enoumous in size, not likely to become part of the living room. Most considerations of horns or box enclosures assume absolutely rigid enclosing surfaces. These surfaces, if not rigid, tend to radiate sound as if they were the diaphragm, and because of the large area, such surfaces sometimes radiate more power to the air than does the projected diaphragm directly. This in itself would not be such a bad effect, except that the enclosing structure causes a delay in the reaction time of the sound wave and produces a hangover effect which continues the sound beyond that which is in the signal; in a word, contributes to the transient distortion.

This effect is much more serious than most people consider and is the main reason that nearly all such devices have a boxy or horn sound which is caused by the delay distortion of the chamber. A rigid horn or enclosure for a small speaker would require almost a 2- to 3-inch wall of concrete or other suitable material to make this enclosure the ideal that is assumed for it in speaker calculations.

Because of the above considerations, it is concluded that only with a large diaphragm having good coupling to the air and lowest possible mass could we hope to achieve an ideal coupling with a good transient characteristic from the low frequency speaker.

The amplitude required for speakers of practical dimension is shown in Fig. 3. It is apparent that reasonable sizes of speakers for low frequency operation can be realized with practical excursion requirements. It is apparent, too, that from 4 to 16 square feet is desirable for most applications if frequencies of 20 to 40 cycles are to be reproduced at desired levels.

Since low frequency sounds are more nearly sine wave in form, the loudness is greater for the same peak values because of the greater area under the curve or power content. Therefore, the desired intensity time function is satisfied at lower absolute peak powers.

"Doppler" distortion is another characteristic on which very little has been written, but which is certainly of major concern. Thousands of man hours have been put on the design of amplifiers,and units have been produced which have a very high degree of linearity both from a single frequency standpoint and intermodulation standpoint, while there is still serious intermodulation distortion in loudspeakers. This particular type of distortion is a form of frequency modulation and results from the presence of both low and high frequencies on the same diaphragm. The greater amplitude of the speaker due to the low frequency signal tends alternately to compress the high frequency wave and to stretch it out, which causes "Doppler" distortion. This "Doppler" distortion (see Fig. 4) of a 15-inch speaker can exceed the energy content of the high frequency signal if, for one instance, such frequencies as 30 cycles and 10,000 cycles are present at the same time. In other words, this distortion in practical commercial speakers might very well be 200 or 300 per cent, or more.

As will be seen in Fig. 5, for 1 per cent Doppler distortion, the 15-inch speaker can deliver 1 acoustic watt at 20 cycles and operate up to 32 cycles only. If a higher frequency is placed on this diaphragm carrying the low frequency component, then this distortion will exceed 1 per cent and goes up very rapidly. Further study of this diagram indicates that for a circular speaker carrying 20 cycles and delivering 1 acoustic watt, and with a frequency of 100 cycles also present, the diameter of a circular unit must be roughly 3.6 feet to keep the "Doppler" distortion at 1 per cent. In the lower diagram, the opposite consideration is shown for the same 1-watt output and 1 per cent "Doppler" distortion. The high frequency unit is considered to have 20,000 cycles on it, and for the various diameters, shows the lowest frequencies that it is possible to place on that diaphragm. It is evident from the above that without regard to any particular make or configuration of speaker unit, these distortion figures suggest the use of at least three sets of speakers operating at different portions of the audio spectrum. And here again we have evidence of the justification for very large diaphragm areas compared to what is normally considered necessary today for the low frequency unit. It turns out in practice that somewhere between 4 and 15 square feet of diaphragm is desirable, depending on the environment.

At the high end of the spectrum, the size of the speaker elements can be made quite small compared to the problem for the low frequency end. However, the mass of the voice coil and diaphragm becomes the real limit, and care is necessary to insure response at the upper limits of audibility.

Many speaker systems utilize a so-called "tweeter", which consists of a diaphragm operating into a restricted throat and emanating into an exponential or catenoidal horn of some form, usually with a honeycomb, or some baffle type of distribution. It is pointed out by many researchers on speaker design that the main merit of this horn is its relatively high efficiency, brought about primarily by the compression of air in the chamber immediately ahead of the diaphragm. This compression may exceed the linear pressure variation limit of the air and, under most circumstances, results in exceedingly high distortion, particularly if frequencies of widely separated values are present at the same time. Fig. 6 shows the distortion from such a speaker working into an infinite horn and delivering 1 watt per square centimeter of throat area.

The power output versus distortion per cent is shown for various ratios of the horn cutoff frequency to the frequency under consideration. For instance, if the tweeter is to cover from 500 to 20,000 cycles, this ratio is 100 to 1. It is foreseeable that for peak powers which occur in practice, this type tweeter is not very satisfactory for high quality sound reproduction unless the range and power are limited. There is, of course, a further disadvantage which has to do with the Doppler distortion mentioned above, especially when the tweeters are to cover wide frequency ranges.

Another characteristic is one concerning the environment, or the effects of rooms. It will be seen from Fig. 7 that in ordinary speech and music there is a substantial gain in a small room if the optimum reverberation time exists in that room. It must be said, however, that these gains are predicted on measurements made with sustained notes, and the reverberation time of rooms is determined by the decay rather than the build-up time. It is not known whether the full gain is realized for the short duration peak pulses of speech and music, because the peak has occurred and decreased to zero long before the room build-up time has taken place; therefore, the useful room gain might be considerably less than predicted by this method alone.

Doing a little arithmetic, we see that for speakers of probable efficiencies 2 per cent $= -20 \log_{10} 1/50 = -1.7 \times 20 = -34$ db. Room gain (see Fig. 7) for a room $12 \times 18 \times 10 = 2{,}160$ cubic feet, or +25 db, approximately. Directional gain due to corner of room, etc., $\equiv +8$ db, or total = +33 db. Therefore, power required into the speaker equals the power required in open space consideration plus 1 db (see Fig. 1). Now, this is a small room, and it is not difficult to see why very substantial amplifier powers are required for the adequate sound system even in the home.

It is interesting to note that independent checks by others[1] have indicated that the amplifier power requirement for reproduction of large orchestras varies from 20 watts to several kilowatts, depending on the size of the rooms.

Fig. 8 shows the calculated "Doppler" distortion for a speaker arrangement utilizing ten different elements. The "tweeters" are direct radiation 2-inch speakers arranged to distribute the frequencies above 2,500 cycles resulting in very nearly 180° distribution. Two 8-inch units were used for the intermediate frequencies. Four square feet were used for low frequency speaker.

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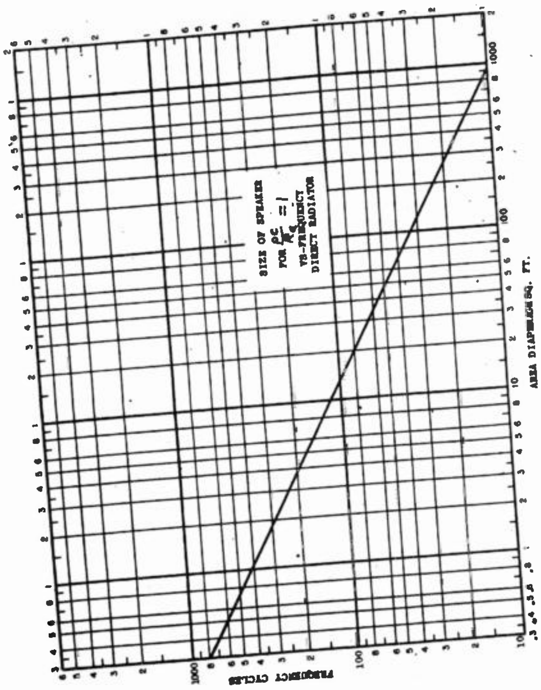


Fig. 2

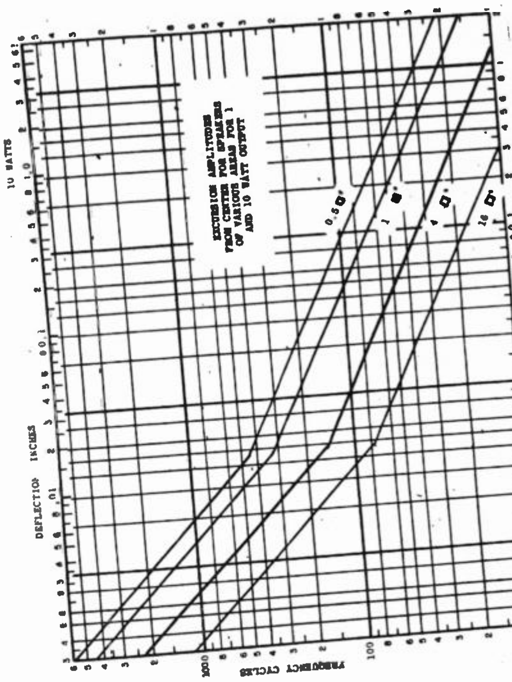


Fig. 3

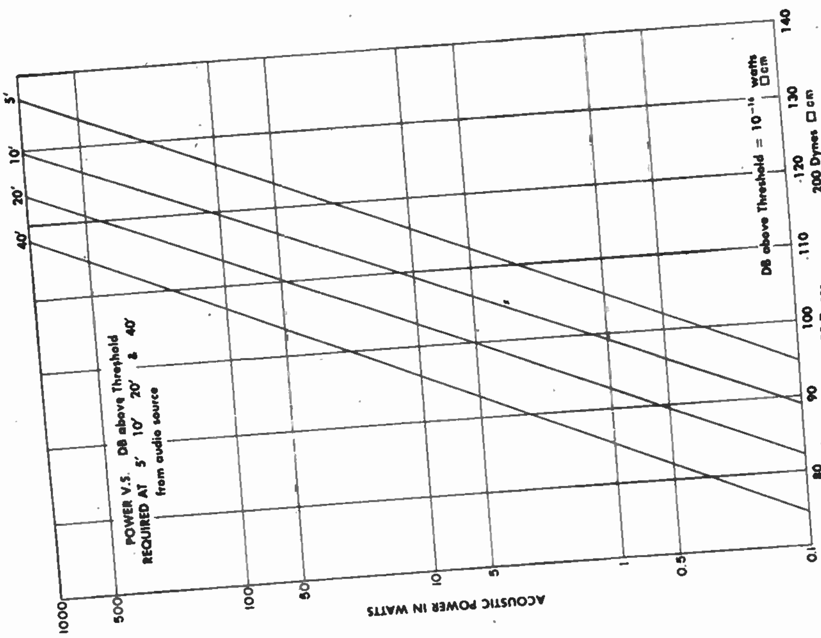
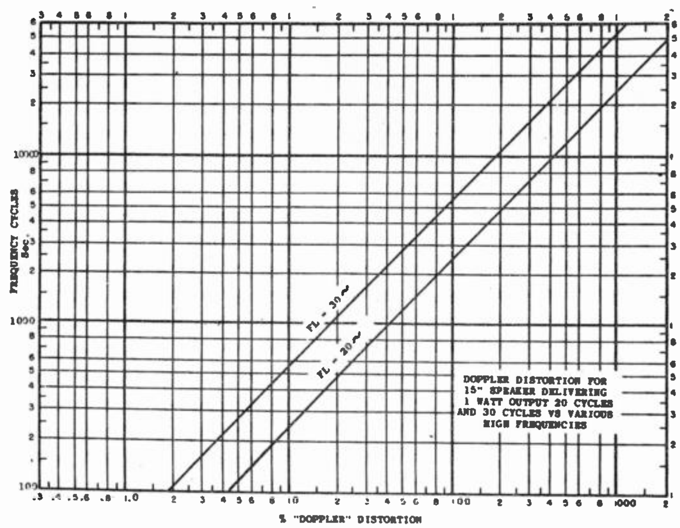


Fig. 1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% "DOPPLER" DISTORTION
FL = 30~
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DOPPLER DISTORTION FOR 15" SPEAKER DELIVERING 1 WATT OUTPUT 20 CYCLES AND 30 CYCLES VS VARIOUS HIGH FREQUENCIES

Fig. 4

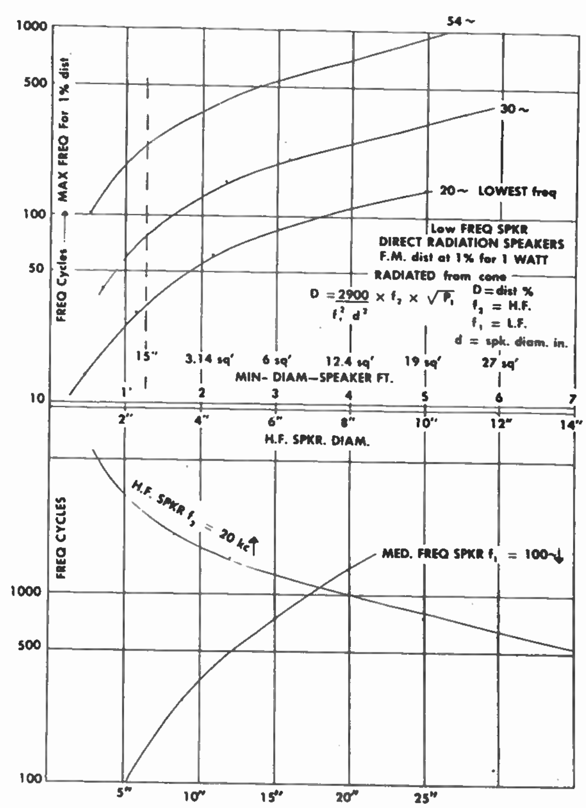
FREQ Cycles —— MAX FREQ For 1% dist
54 ~
30 ~
20 ~ LOWEST freq
Low FREQ SPKR
DIRECT RADIATION SPEAKERS
F.M. dist at 1% for 1 WATT
RADIATED from cone
D = 2900/(f1² d²) × f2 × √P1
D = dist %
f2 = H.F.
f1 = L.F.
d = spk. diam. in.
15"
3.14 sq'
6 sq'
12.4 sq'
19 sq'
27 sq'
MIN- DIAM—SPEAKER FT.
H.F. SPKR. DIAM.
FREQ CYCLES
H.F. SPKR f2 = 20 kc
MED. FREQ SPKR f1 = 100~

Fig. 5

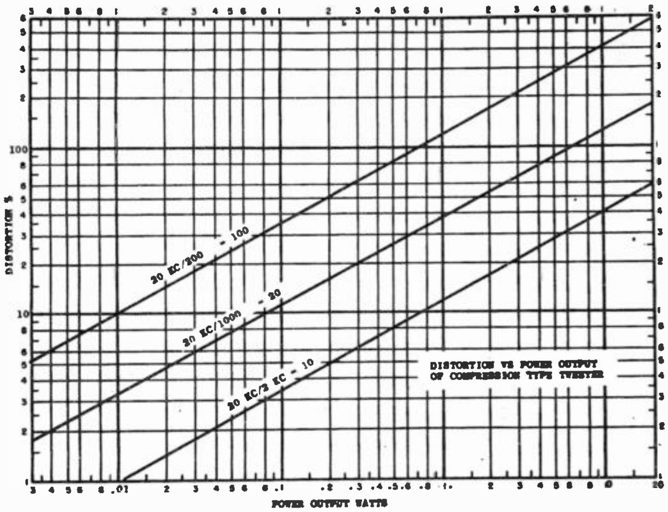

Fig. 6

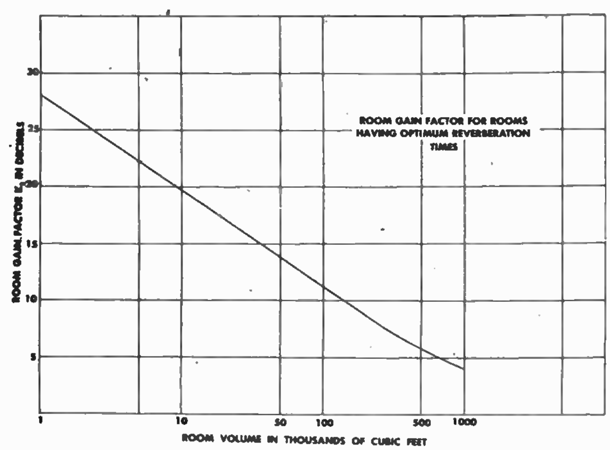

Fig. 7

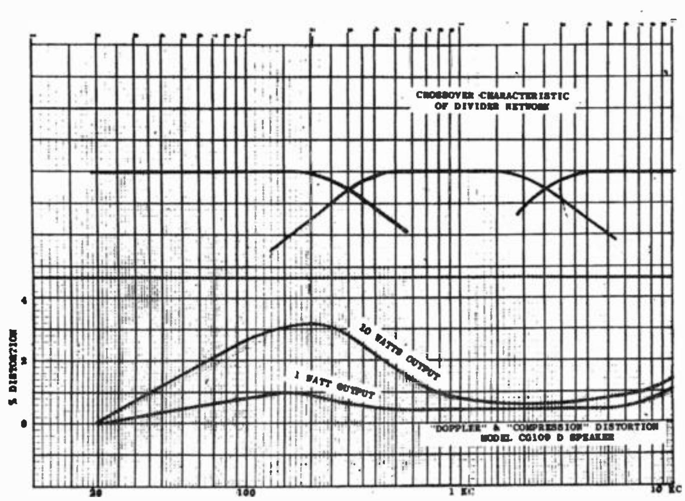

Fig. 8

INTERFERENCE EFFECTS IN MAGNETIC RECORDING HEADS*

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Summary

Irregularities in the frequency response of magnetic recording heads may be understood by considering the head to be made up of a multiplicity of gaps which, in playback, act as sources of flux with respect to the coil. The output from these sources differs in phase and amplitude and causes the irregularities, which are interference patterns. These sources of flux are the main gap and the edges of the pole pieces, which exist in all heads. In addition, in overlap-type heads, which are composed of laminations at right angles to the direction of tape travel, the spaces between the laminations act as sources of flux. When the outputs of these three different types of sources are added algebraically, a frequency response curve is obtained which is in good agreement with the experimental curve. By choosing the number of gaps and/or the spacing between them, the irregularities in the frequency response may be reduced, or various predetermined frequency responses may be approximated.

Introduction

The frequency response of a magnetic tape recording and playback head shows certain departures from smoothness, particularly at the lower frequency end. These irregularities are of interest in themselves, in addition to the practical improvement in the heads which may be achieved by an understanding of the reasons for their presence. This paper is an analysis of the reasons for these irregularities, and a discussion of means of controlling them.

Method of Analysis

The irregularities are almost entirely a function of the playback head only. This is shown in Fig. 1, where two heads are used -- a "smooth" response head and a head whose response has been purposely made as irregular as possible. In Fig. 1(a), the irregular head is used both for recording and for playing back its own signal, and, as can be seen, the result is irregular. In Fig. 1(b), the same with the smooth head, and the response is smooth. In Fig. 1(c), the response is recorded by the irregular head and played back on the smooth head, and it can be seen that the response is smooth, although there is evidence of slight irregularities. Fig. 1(d) shows that the response, when recorded on the smooth head and played back on the irregular head, is irregular. This simplifies not only the problem, but also the presentation, since hereafter, the behavior of the playback head only is of interest.

* Presented at the National Electronics Conference, Chicago, Ill.; September 29, 1952. Manuscript received September 3, 1952.

In the playback process, it is helpful to think of gaps as sources of flux which take magnetic flux from the tape and supply it to the coil. It has been found that in addition to the main gap, there are two other types of sources of flux in tape heads, and that these sources interfere with the output of the main gap and cause the irregularities. These sources of flux are: (a) The edges of the pole pieces (edge gaps).[1] (b) In tape heads in which the laminations are at right angles to the direction of tape travel, the spaces between these laminations act as sources of flux. These will be called auxiliary gaps. Since these sources of flux are displaced from the main gap, there will be a phase difference between their output and the output of the main gap. Their output will add to the main gap output at some frequencies, subtract at others, or, in general, interfere with the output and produce an irregular response.

Fig. 2(a) is a diagram of a type of tape recording and playback head. The two magnetic pole pieces (0.010 inch thick) are separated from each other by a metallic nonmagnetic spacer, which is commonly 0.0005 inch thick. This forms the main gap. In addition to the main gap, the head shown in Fig. 2(a) contains two auxiliary gaps (the spaces between the laminations) and two edge gaps. Fig. 2(b) shows a modification of the head in 2(a) in which there is only one auxiliary gap. Fig. 2(c) shows still another type in which the relative position of the gaps is changed. Fig. 3 is a section of the head shown in 2(a) The main gap is at A, the two auxiliary gaps at B, and the two edge gaps at C.

Let us first consider the analysis of a case where there are symmetrical edge gaps alone, displaced from the main gap by a distance b, shown in Fig. 4. Let the tape traveling over the head have impressed upon it a magnetic signal of wavelength λ, and be traveling across the head with velocity $v = x/t$, where x is the displacement.

The output from the main gap is:

$$e = A \cos 2\pi x/ \tag{1}$$

where A is a variable which includes in it the 6-db per octave rise at low frequencies due to the velocity characteristic of the magnetic circuit and the decline in output at the high frequencies (short wavelengths) due to the gap effect.

Now, the output from the edge to the right is

$$e_2 = -C \cos 2\pi(x+b)/\lambda \tag{2}$$

since the source is located at the point $(x + b)$. The minus sign in front of the term requires a word of explanation. It is due to the fact that there is an inherent 180° phase reversal at the edges, as shown in Fig. 4. It can be seen that an elemental magnet placed at the edge causes flux to flow through the coil in a different direction than a similarly poled magnet placed at the main gap.

Similarly, the output from the edge to the left is

$$e_3 = -C \cos 2\pi(x-b)/\lambda \tag{3}$$

and the total output is the sum of (1), (2), and (3), or

$$e_{out} = A \cos 2\pi x/\lambda \; - C\left[\cos 2\pi(x+b)/\lambda + \cos 2\pi(x-b)/\lambda\right]. \tag{4}$$

Summing:

$$e_{out} = \cos 2\pi x/\lambda \left[A - 2B \cos 2\pi b/\lambda\right]. \tag{5}$$

This last equation is the useful result since it gives the frequency response in terms of the ratio of b, the pole width, to λ, the wavelength on the tape. This result is shown in Fig. 5, which is the actual frequency response of a head with pole pieces 0.020 inch in width (in the direction of tape travel) plotted at 3.75 ips.

The calculated first minimum for the above set of constants occurs at 167 cycles, which can be seen to be in good agreement with the observed value. At 15 ips, with $\frac{1}{4}$-inch pole faces, the first minimum occurs at 60 cps.

It should be pointed out that one method of minimizing this effect is to place the main gap unsymmetrically between the edges. Then the separate responses of the edges do not add to each other in phase. Also, by causing the tape to leave the pole pieces gradually, the magnitude of the effect may be decreased.

The frequency response of a head in which there are two 0.010-inch laminations on either side of the main gap (as shown in Fig. 2(a) is shown in Fig. 6(a). These laminations are arranged at right angles to the direction of travel.

Using an exactly similar method as that used for the two laminations on each side case, the response of the head may be written; let the lamination width be d, as shown in Fig. 3.

$$e_A = A \cos 2\pi x/\lambda \tag{6}$$

$$e_B = B \cos 2\pi(x-d)/\lambda + B \cos 2\pi(x+d)/\lambda \tag{7}$$

$$e_C = -C \cos 2\pi(x-2d)/\lambda \; -C \cos 2\pi(x+2d)/\lambda \tag{8}$$

Summing these and performing some trigonometric manipulation yield

$$e_{out} = \cos 2\pi x/\lambda \left[A + 2B \cos 2\pi d/\lambda - 2C \cos 4\pi d/\lambda\right]. \tag{9}$$

Note that this gives one term due to the edges plus a term due to the auxiliary gaps.

Fig. 6(c) shows the calculated response which is (9) plotted on a base of a smooth response head, i.e., a head in which the pole pieces are infinitely long. To obtain Fig. 6(c), A was arbitrarily given the value 1, B given $\frac{1}{4}$, and C given 1/8. This curve may be compared with Fig. 6(a), which is the actual curve obtained with a head of this type.

Another configuration of interest is one in which there are two 0.010-inch laminations on one side of the gap, and one 0.010-inch lamination on the other (for brevity, called 2 and 1, as opposed to the previous 2 and 2). This configuration is shown in Fig. 6(b). With this configuration, it is possible to obtain a smoother response than in the symmetrical case. The theoretical response of the heads, using the same notation and method as previously, is

$$e_{out} = \cos 2\pi x/\lambda \left[A + (B-C) \cos 2\pi d/\lambda - C \cos 4\pi d/\lambda \right]. \qquad (10)$$

The actual response is shown in Fig. 6(b). It will be noted that the "holes" in the responses do not coincide as they do in the symmetrical case shown above in Fig. 6(a) and a substantially smoother frequency response is the result. Fig. 6 shows the response of the 2 and 2 structure in 6(a) as compared with the 2 and 1 structure in 6(b).

It is possible to calculate the response of any configuration of laminations, varying both in number and width, by this method, and many interesting combinations are the result. For example, the configuration in which three laminations are placed on one side of the main gap, and only one on the other (as shown in Fig. 2(c)) gives, both theoretically and practically, almost as smooth a response as the 2 and 1 structure used commercially.

It is also possible to build heads that have responses that are peaked to certain predetermined frequencies, for applications in telemetering, for example, in which certain discrete frequencies are recorded.

Conclusion

It is possible to obtain an analytical expression for the irregularities in the response of a tape head, making two assumptions: (a) the edges act as gaps; (b) spaces between the laminations act as gaps. The analytic expression obtained using these assumptions is in good agreement with empirical results, and may be used to predict the output of modifications of the basic lamination stack. In particular, it is possible, by varying the lamination width, number of laminations, and the spacing between laminations, to obtain responses which are peaked at certain frequencies, and/or have holes at certain frequencies.

Acknowledgment

I would like to acknowledge the helpful and stimulating conversations on the subject which I have had with other members of the Tape Recording Section of Shure Brothers, Inc., in particular, Mr. Lee Gunter, Jr., and Mr. Thomas W. Phinney.

Reference

[1] Clark and Merrill, "Field measurements on magnetic recording heads," Proc. I.R.E., vol. 35, pp. 1575-1579; December, 1947.

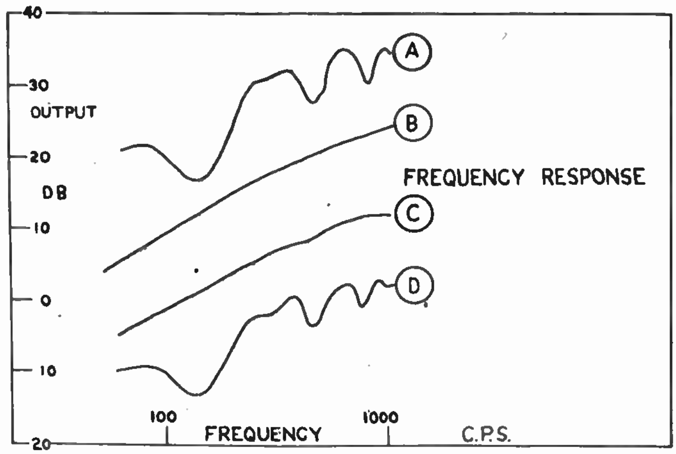


Fig. 1

Frequency response curves which show that the interference irregularities occur in the playback head only. See text.

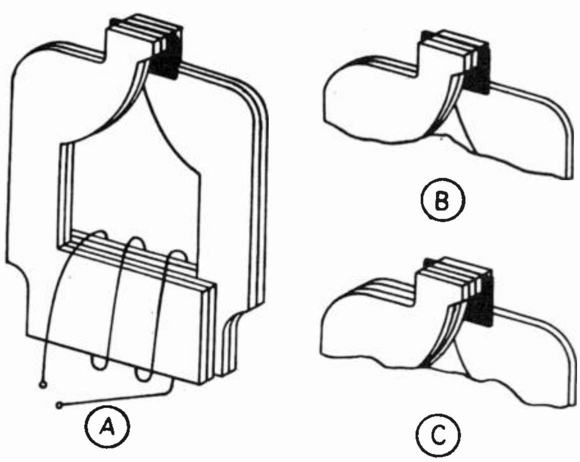


Fig. 2

Structure of the "2 and 2", "2 and 1", and "3 and 1" tape heads.

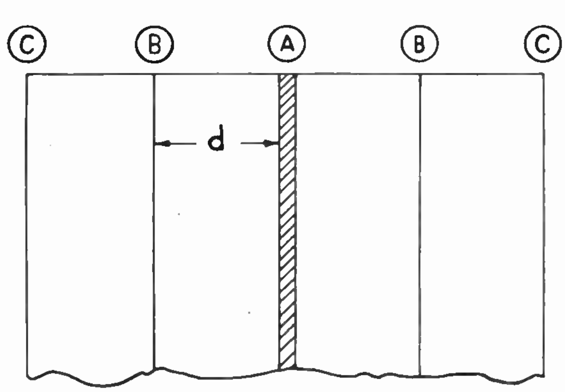


Fig. 3

Diagram showing sources in a "2 and 2" tape head.

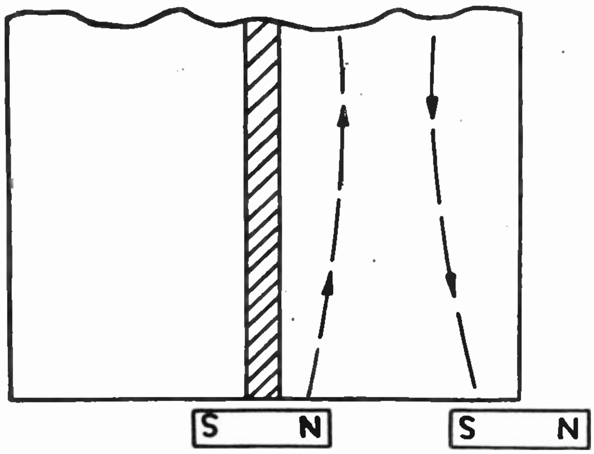


Fig. 4

Diagram showing a single pole piece on each side of the main gap.

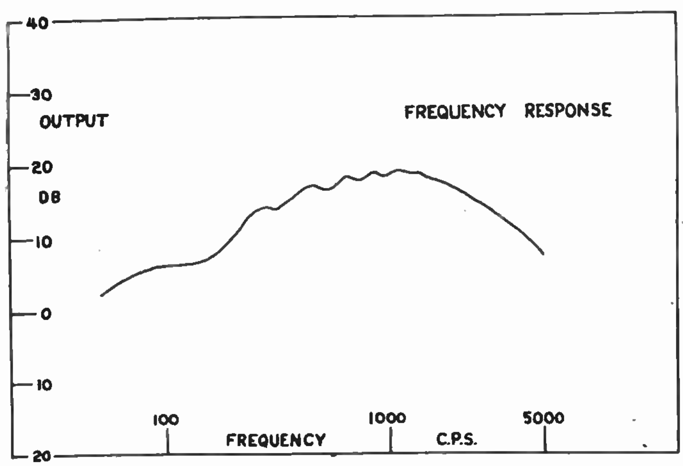


Fig. 5

Frequency response of the head shown in Fig. 4.

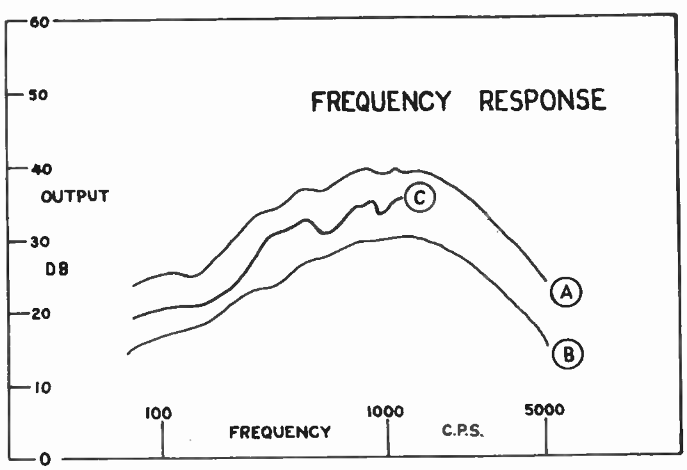


Fig 6

Frequency response of (a) "2 and 2" structure measured; (b) "2 and 1" structure measured; (c) "2 and 2" structure calculated.

INCREASED AUDIO WITHOUT SPLATTER*

John L. Reinartz
Eitel-McCullough, Inc.
San Bruno, Calif.

Summary

It is not generally realized, especially in cases where the Class B modulating transformer has poor regulation, that a higher ratio of audio output in the positive direction can be obtained if the secondary of the transformer has an additional asymmetric load placed on it through the use of a diode tube and appropriate resistor (see attached sheet).

If the negative voltage excursion of the Class B transformer secondary just equals the plate supply voltage, then the positive voltage excursion cannot be as far because in the first instance there is no load across this secondary at the end of the negative voltage excursion and consequently no voltage drop, while in the second instance there is a maximum load across the secondary as the voltage swings in the positive direction with its consequent IR voltage drop, and this subtracts from the positive swing.

Placing an extra load on the transformer secondary through the use of a diode and series resistor, so that this combination allows loading only on the negative voltage swing, provides a means for preventing the negative voltage excursion from exceeding the positive voltage excursion to any degree desired.

Adjustment of the resistor value can be such that the voltage and modulating power output of the Class B transformer secondary is even greater in the positive direction than in the negative direction, with consequent greater modulation effects without, however, exceeding 100 per cent modulation.

A summation of (E_1I) and (E_2I) will show that the requirement of a 50 per cent modulating power for 100 per cent modulation factor is not changed. However, it is interesting to note that there is a 16 per cent load on the Class B transformer secondary during the negative voltage excursion and an 84 per cent load on it during the positive voltage excursion with respect to the polarity of E_1. Thus, there already exists an asymmetric loading of the Class B transformer secondary during an audio cycle.

The addition of the extra load proposed in "B" during the negative voltage excursion, if of proper value, results in nearly perfect symmetry of loading of the Class B transformer secondary during the negative as well as the positive voltage excursions.

****

* Presented at the IRE Southwestern Conference, Houston, Texas; May 17, 1952. Manuscript received June 30, 1952.

Let us consider a Class C RF stage that is to be amplitude modulated. This Class C RF stage is capable of a linear plate current increase with a linear plate voltage increase from a value that may be considered to be the dc plate voltage E_b to twice this voltage, $2E_b$. Thus the plate input will go from X watts to 4X watts. Consequently, the stage being linear, it can be considered to represent a constant load R, determined by E_b/I_b, I_b being the average plate current, and E_b the plate voltage E_1.

If we now add the modulating transformer to the Class C RF stage in the usual manner, shown in A of Fig. 1, we can consider the secondary of the modulating transformer E_2 to be the equivalent of E_1 in voltage output, since 100 per cent modulation is to be achieved. By definition, 100 per cent modulation is achieved when $E_1 - E_2 = 0$, $I_b = 0$, and the output from the Class C RF stage is zero. Any time that $-E_2$ is greater than E_1, overmodulation occurs, no current flows through the Class C RF stage since it is a unidirectional valve, and the carrier power is interrupted.

Interruption of the carrier power can result in spurious side bands that interfere with other uses of those frequencies, and are to be avoided. These spurious frequencies are also generated when negative clipping is resorted to unless a filter is inserted in the circuit to round off the sharply clipped edges. However, it is usually impossible to re-establish the original wave form; thus, some distortion is invariably present in the modulated output. The only reason for negative clipping and subsequent filtering is the desire on the part of the radio amateur for an increased modulated carrier output during the positive voltage swing of the modulation transformer secondary while preventing the modulation transformer secondary voltage swing in the negative direction from exceeding the normally applied dc plate voltage, and consequently preventing the generation and emission of spurious frequencies.

It occurred to the writer that it should be possible to increase the emission in the positive direction of modulation without the need for clipping in the negative direction of modulation, thereby preventing both overmodulation and audio distortion. As an aid in visualizing the action of the modulation transformer secondary and its effect on the Class C RF stage, an analysis of the several instantaneous values was made, and the results shown as curves on cross section paper.

Several interesting items not previously detailed in text books became clear and obvious. The separation of the total power output due to the instantaneous sum of the dc plate and the ac modulation transformer secondary voltages into their component factors E_1 and E_2 is clearly shown in Fig. 1. It is also shown quite clearly that while the full $-E_2$ voltage is developed during the negative voltage generation of the modulation transformer secondary equal to the dc plate voltage E_1, there actually is no load on either of these two generators. Only when the full $+E_2$ voltage is developed during the positive voltage generation of the modulation transformer secondary equal to the dc plate voltage E_1 are these separate generators fully and equally loaded.

The load curves for generators E_1 and E_2 are clearly a function of their instantaneous voltages and the instantaneous current I in the circuit system. Obviously, E_1 is constant, I varies from 0 to 2I, and E_2 varies from 0 to the value of E_1 additive or subtractive. The load curve for generator E_1 is therefore a straight line between 0 and 200 per cent load, while the load curve for generator E_2 is a curve in the positive load direction from 0 to 200 per cent and a curve in the negative load direction from 0 to a maximum of 25 per cent load and continuous toward zero load at its maximum negative voltage swing. The sum of these two generator loads is, of course, $I\ (E_1 \pm E_2) = I\ ^2R_1 + I\ ^2R_2 = (E_1 \pm E_2)^2/R$.

We have now brought R, R_1, and R_2 into our analysis, and it is interesting to note that the resistances across generators E_1 and E_2 vary as the sum of $E_1 \pm E_2$ varies. When the modulating voltage is zero, the system R obviously is the value E_1/I and represents the Class C stage load at the average carrier point shown in the graph. However, when the generator voltage E_2 equals generator voltage E_1 and is additive, the resistance R divides into R_1 and R_2, and each generator sees $\frac{1}{2}R$. This can be more readily understood if we analyze Figs. 2(a) through 2(c). When E_2 equals E_1 but is subtractive, I equals zero, generator E_1 sees a positive infinity resistance, and generator E_2 sees a negative infinity resistance and the sum therefore equals zero. However, the sum of the separate resistances that generators E_1 and E_2 see is always equal to the value R at all other points on the varying $E_1 \pm E_2$ voltage curve.

Having noted that the generator E_2 sees no load when its voltage equals the voltage of E_1 but is subtractive and sees $\frac{1}{2}R$ when its voltage equals E_1 but is additive, it follows that no voltage drop occurs in the first instance due to the losses that occur in the modulating transformer secondary winding, constituting its own IR drop, and since this IR drop subtracts from the positive voltage swing, if we can still equal E_1 at this point, we must somehow prevent the negative excursion from exceeding E_1 by the amount of the IR drop, otherwise we exceed 100 per cent modulation.

We have now come to the crux of the whole analysis. We require a means to produce an IR drop in the modulation transformer secondary when its voltage swing is in the negative direction to equal the IR drop when its voltage swing is in the positive direction. A diode tube with a resistor in series, connected across the modulating transformer secondary so that current flows in this auxiliary circuit only when the voltage swing is in the negative direction, is therefore indicated. The value of the resistance we have already determined to be $\frac{1}{2}R$, while its watt rating should be $I^2R/4$.

The loading on the modulation transformer secondary when its voltage swing is in the negative direction will now be as indicated by the dotted line in Fig. 1. It will be noted that the modulator system now has a symmetrical loading during the negative as well as the positive voltage swing, resulting in reduced second-harmonic distortion, full modulation on the positive voltage swing, and prevention of overmodulation on the negative voltage swing.

Table I

$\%E_1$	$\%E_2$	$\%E_1 + E_2$	$\%I$	$\%R_1$	$\%R_2$	$R = R_1 + R_2$ %
100	0	100	100	100	0	100
100	20	120	120	83.3	16.7	100
100	40	140	140	71.4	28.6	100
100	60	160	160	62.5	37.5	100
100	80	180	180	52.4	47.6	100
100	100	200	200	50.0	50.0	100
100	-20	80	80	125	-25	100
100	-40	60	60	166	-66	100
100	-60	40	40	250	-150	100
100	-80	20	20	500	-400	100
100	-100	0	0	00	-00	0

Table II

Generator 1					Generator 2		
$\%E_1$	%I	%Po			$\%E_2$	%I	%Po
100	0	0			0	100	0
100	20	20			20	120	24
100	40	40			40	140	56
100	60	60			60	160	96
100	80	80			80	180	144
100	100	100			100	200	200
100	120	120			-20	80	-16
100	140	140			-40	60	-24
100	160	160			-60	40	-24
100	180	180			-80	20	-16
100	200	200			-100	0	0

Table III

$\%E_1 + E_2$	%I	%Po
100	100	100
120	120	144
140	140	196
160	160	256
180	180	324
200	200	400
80	80	64
60	60	36
40	40	16
20	20	4
0	0	0

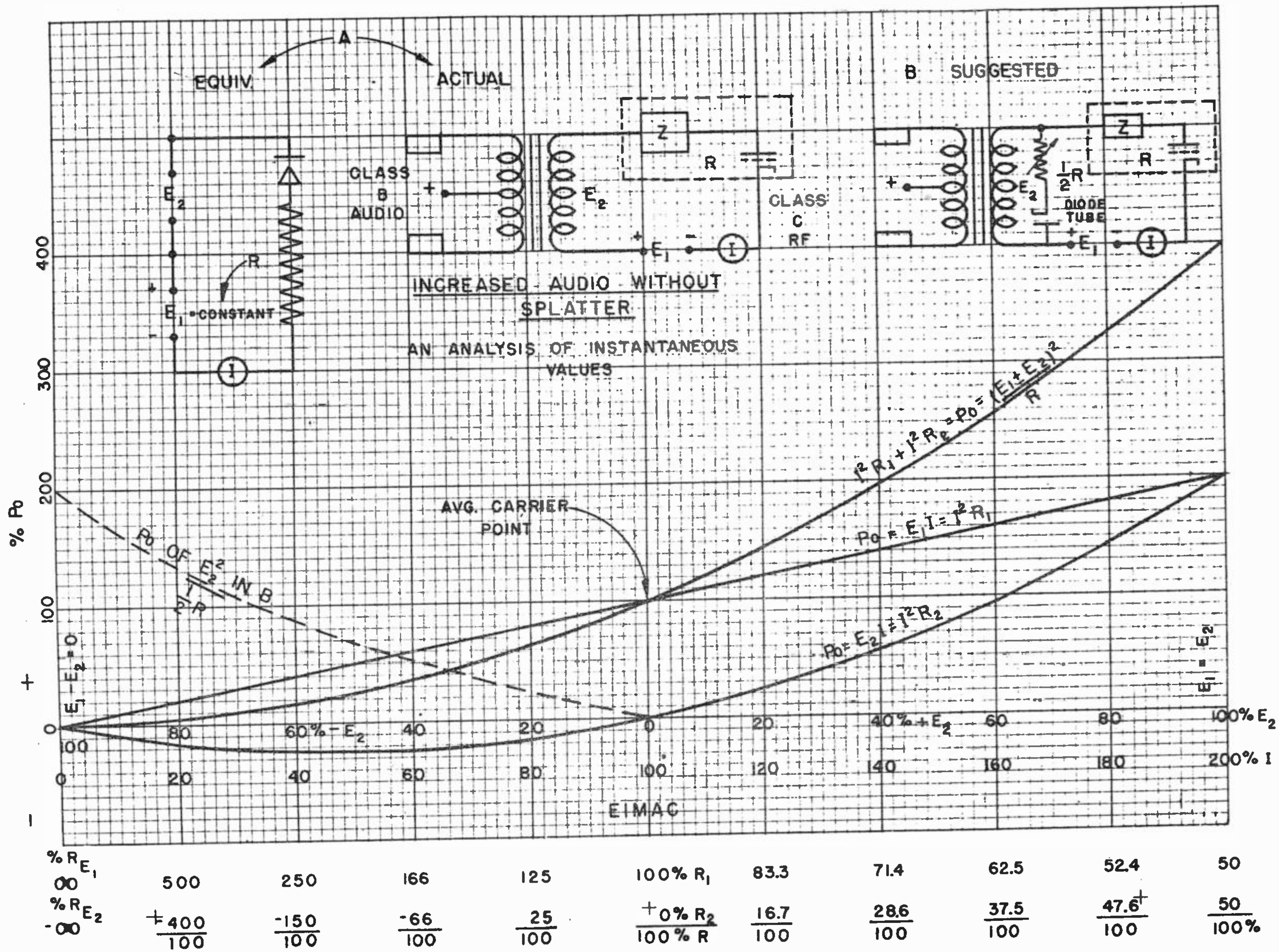
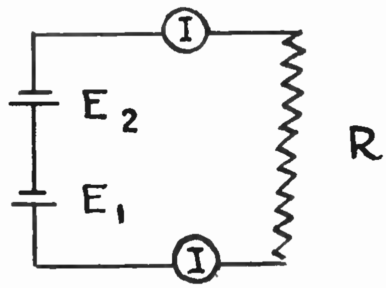

A
EQUIV.
ACTUAL
B SUGGESTED
CLASS B AUDIO
CLASS C RF
DIODE TUBE
INCREASED AUDIO WITHOUT SPLATTER
AN ANALYSIS OF INSTANTANEOUS VALUES
AVG. CARRIER POINT
E₁ = CONSTANT
Po OF E₂² IN B ½R
I²R₁ + I²R₂ = Po = (E₁+E₂)²/R
Po = E₁I = I²R₁
Po = E₂I = I²R₂
E₁ - E₂ = 0
E₁ = E₂
% Po
EIMAC
% R_E₁
% R_E₂
100% E₂
200% I
100% R₁
+0% R₂
100% R

Fig. 1



$RI = E$

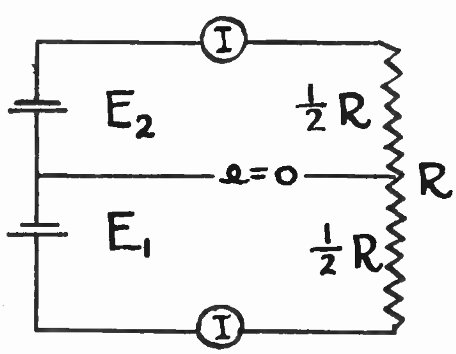
(a)

$$E_1 = E_2$$

$$E = E_1 + E_2$$

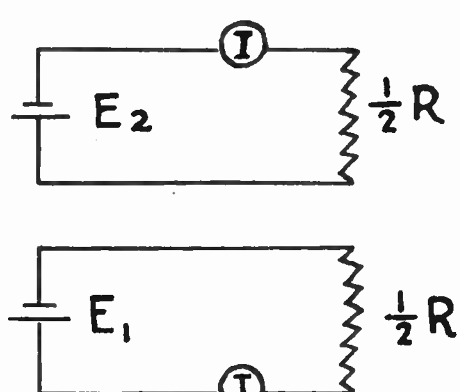
$$R = \frac{E_1 + E_2}{I}$$

I = Constant



(b)

At $\frac{1}{2}$ R exists a zero voltage point with respect to the connection between E_1 and E_2. I has not changed.



$\frac{1}{2}RI = E_2$

$\frac{1}{2}RI = E_1$

(c)

The two circuits can be separated without changing I, and each generator sees but $\frac{1}{2}$R.

Fig. 2

INSTITUTIONAL LISTINGS (Continued)

INSTITUTIONAL LISTINGS

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(Please see inside back cover for additional names)

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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

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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*

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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.

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

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

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.

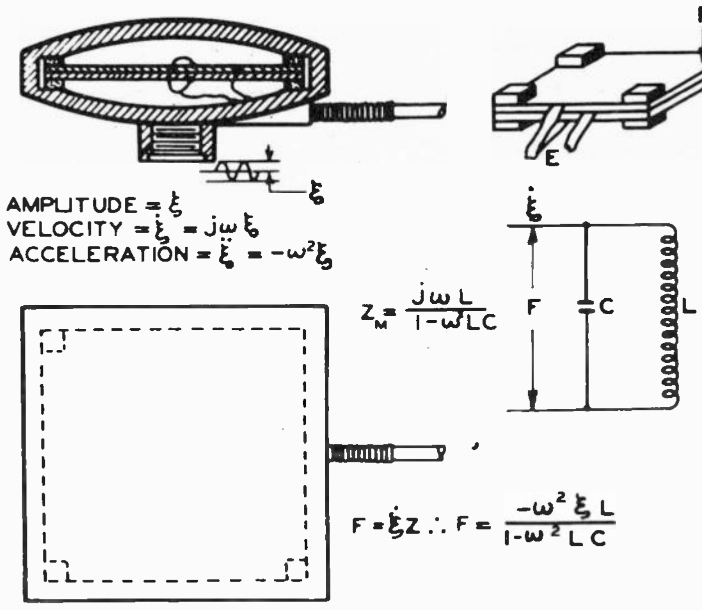
2. Mark Harrison, A. O. Sykes, and Paul G. Marcotte, "The Reciprocity Calibration of Piezoelectric Accelerometers", Journal of the Acoustical Society of America, Vol. 24, No. 4, July, 1952, p. 384.

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

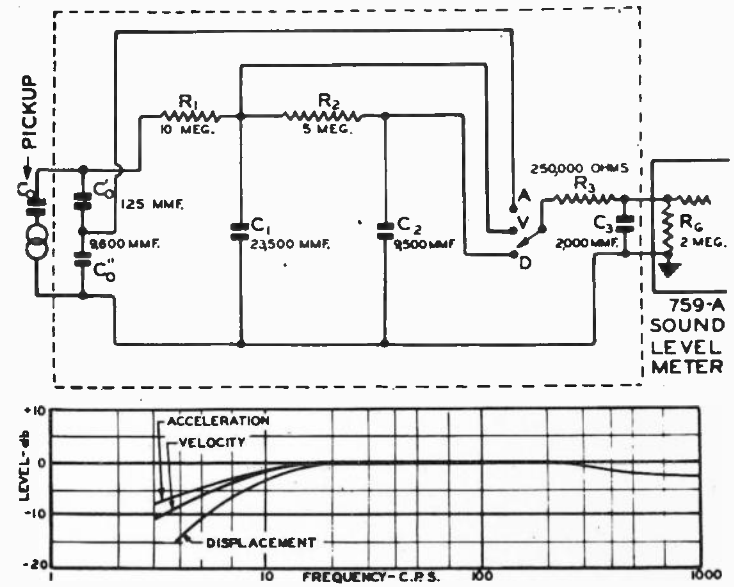


Fig. 2

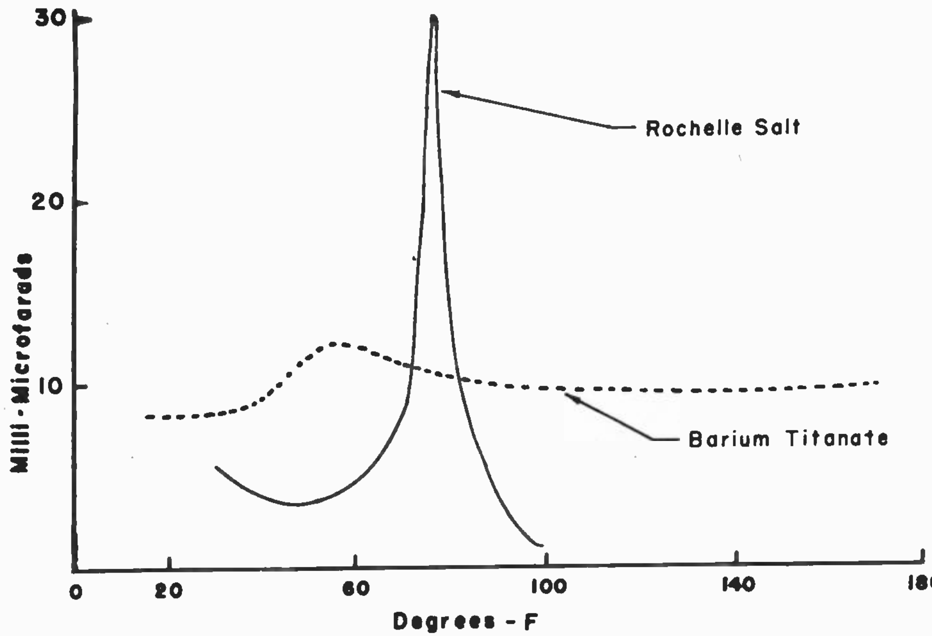


Fig. 3

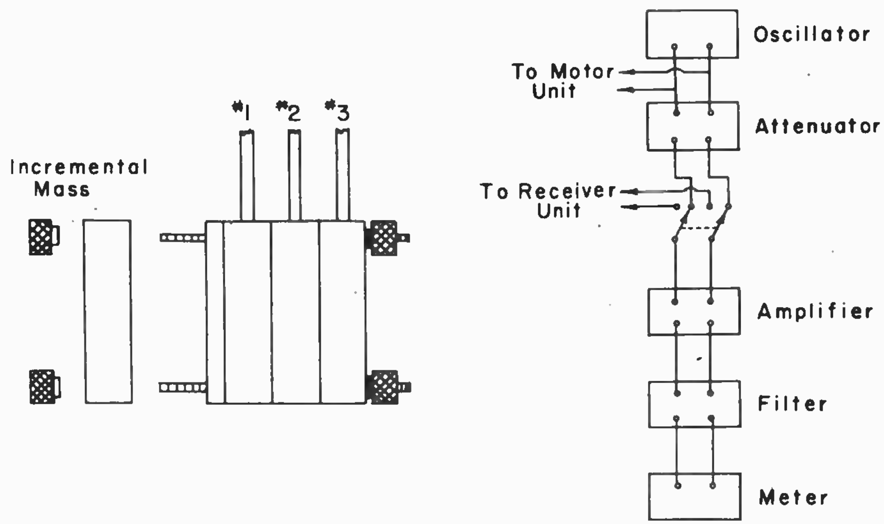


Fig. 4

ENGINEERING CONSIDERATIONS IN THE USE OF MAGNETIC RECORDING HEADS*

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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

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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

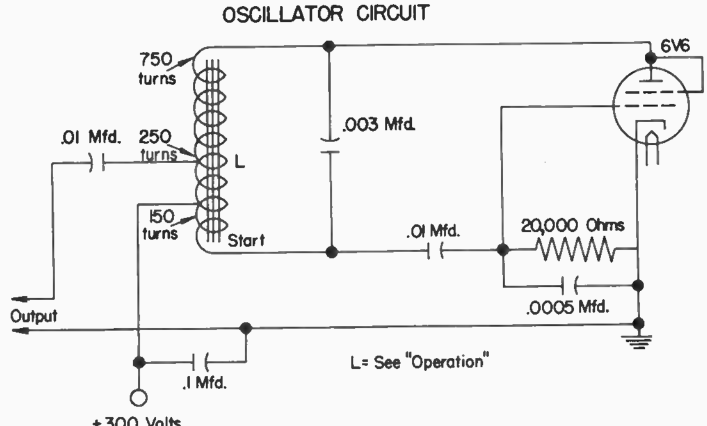

FIGURE I

RECORD CIRCUIT USING A.C. BIAS AND ERASE VOLTAGE

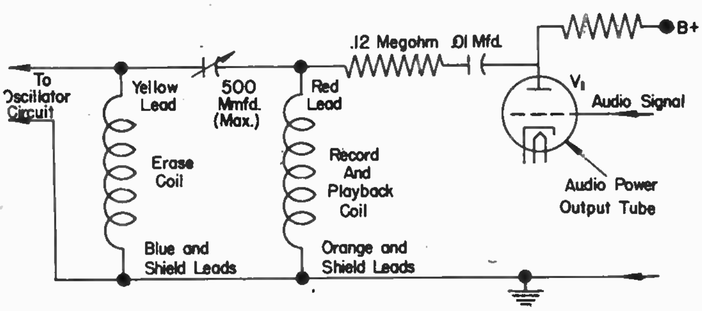

FIGURE 2

RECORD CIRCUIT USING D.C. BIAS AND ERASE VOLTAGE

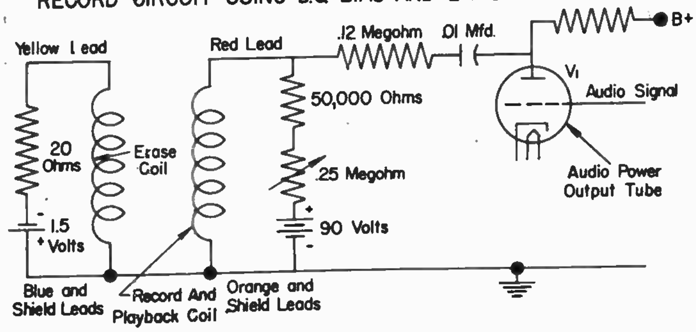

FIGURE 3

PLAYBACK CIRCUIT

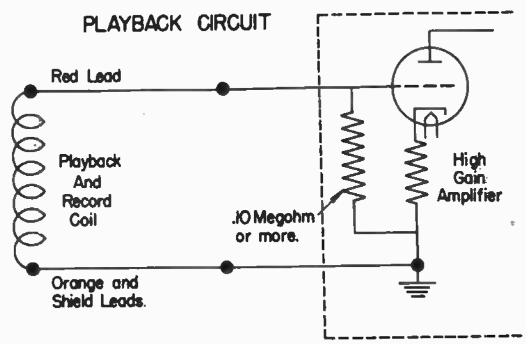

FIGURE 4

INTERSTAGE COMPENSATING NETWORK

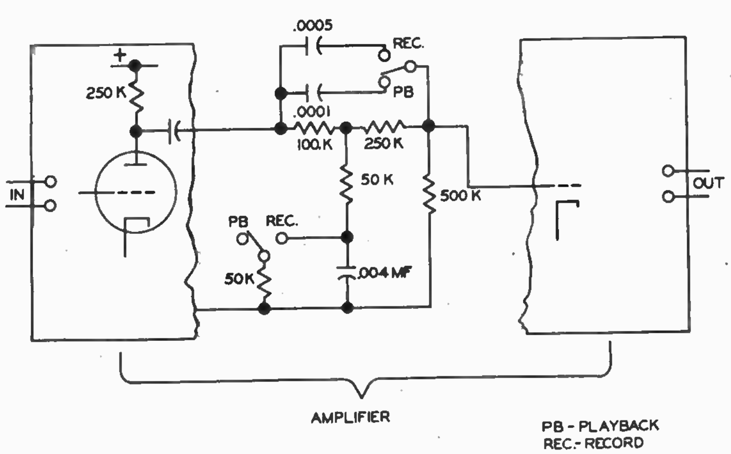

FIGURE 5

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

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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

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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

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

or the power level above a milliwatt by

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

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

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

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

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

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]

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

or

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

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

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

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.

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

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

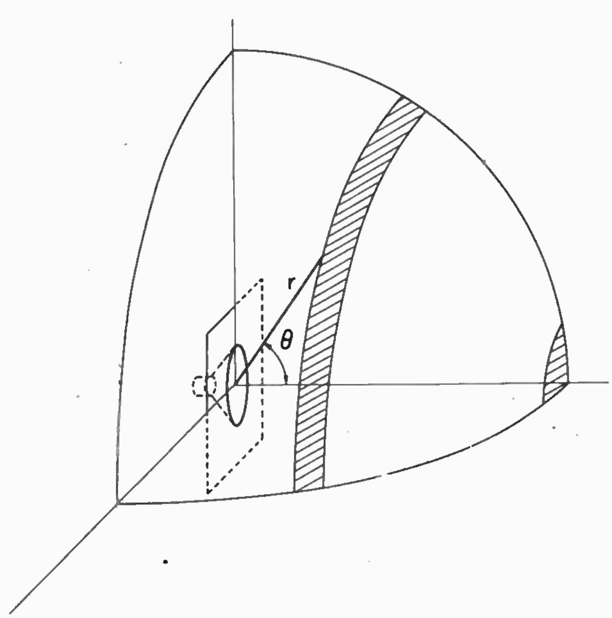

Fig. 1 - Coordinate system used for spherical integration. The origin is located at the effective center of loudspeaker cone.

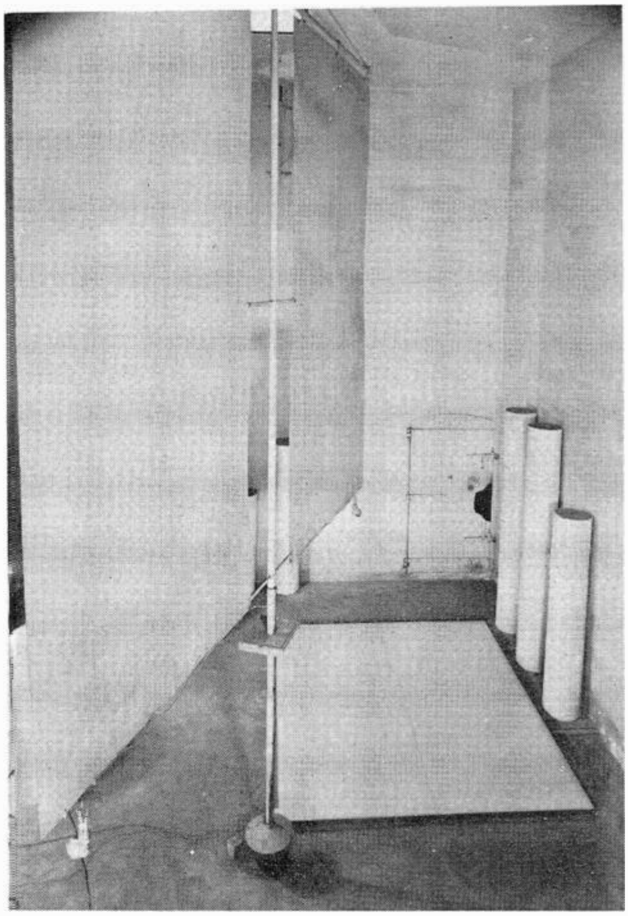

Fig. 2

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

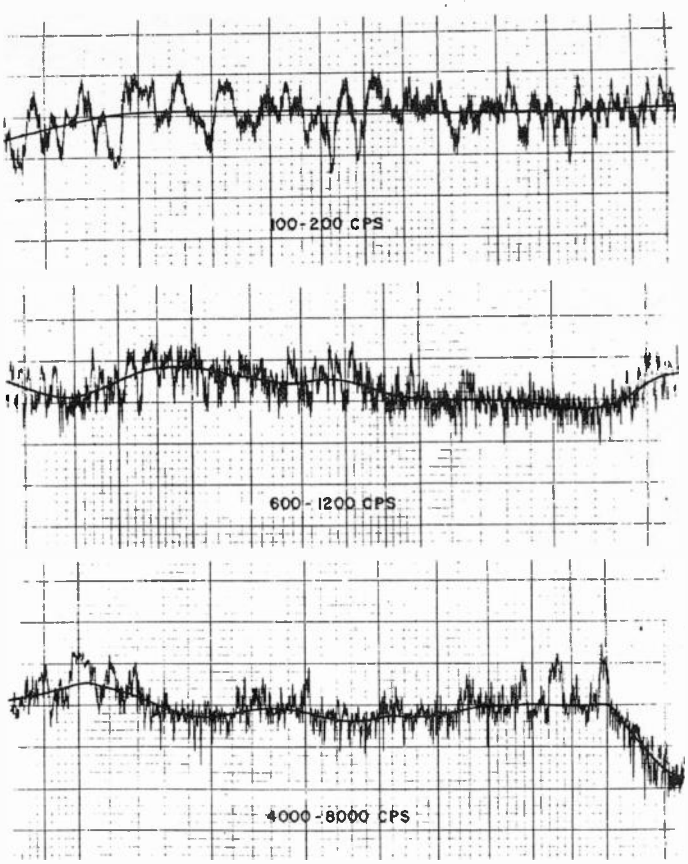

Fig. 3

Sections of chart from graphic level recorder.

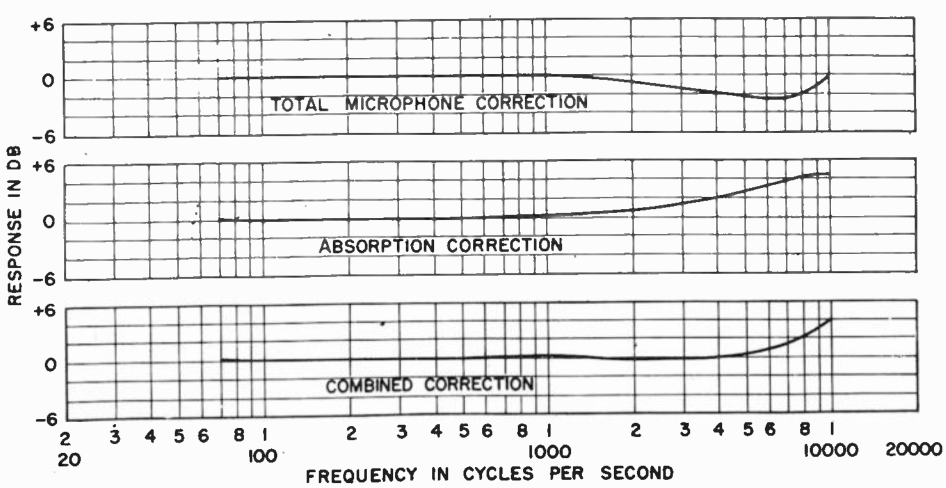

Fig. 4

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.

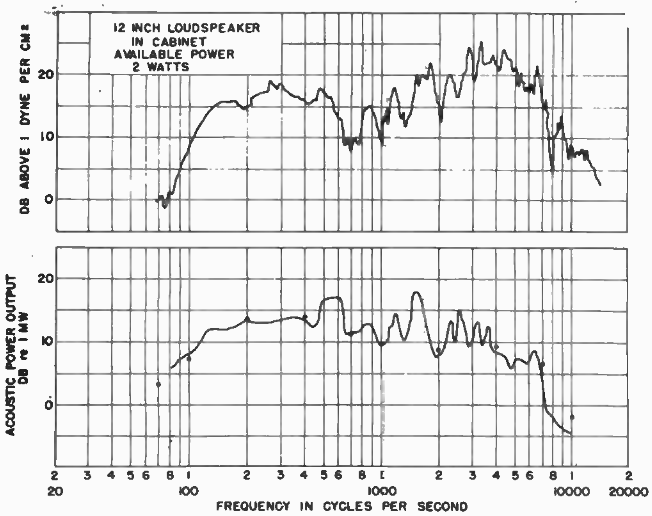

Fig. 5

(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.

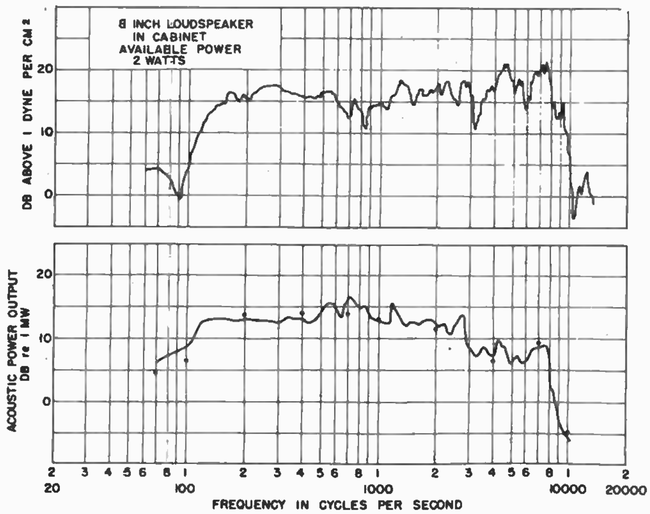

Fig. 6

(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.

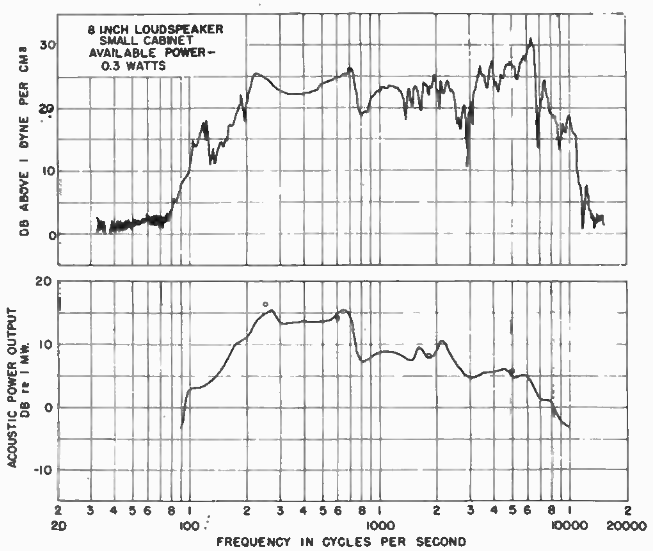

Fig. 7

(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.

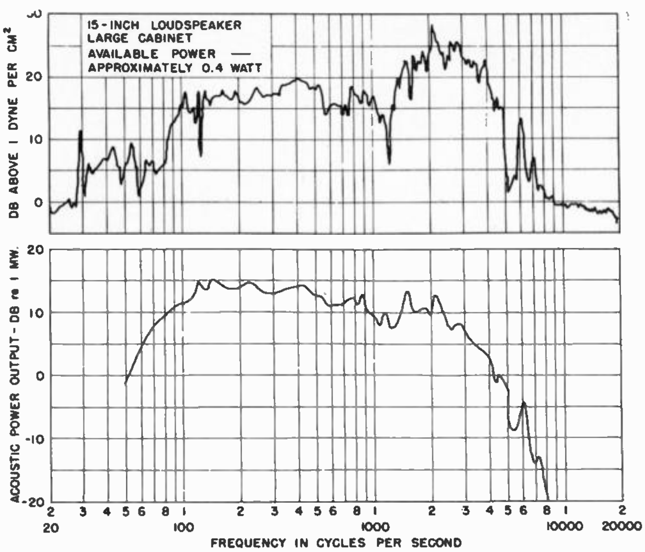

Fig. 8

(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.

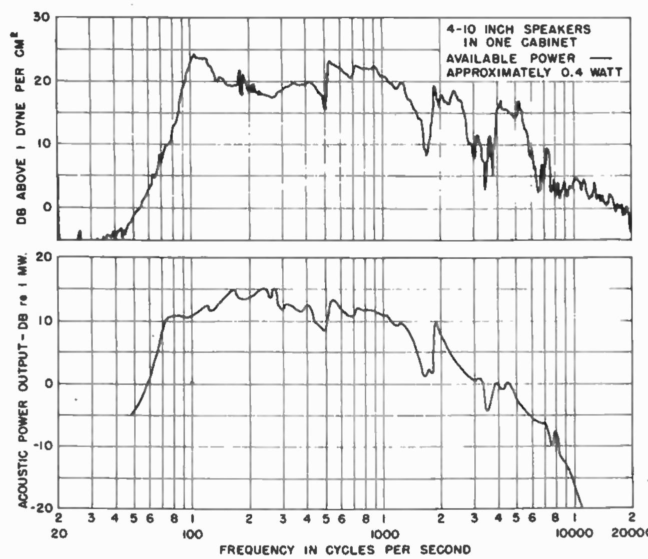

Fig. 9

(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.

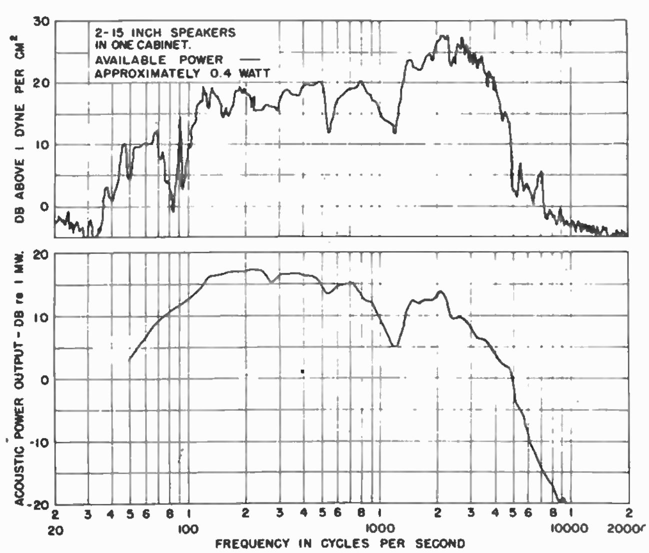

Fig. 10

(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*

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Minneapolis 14, Minnesota

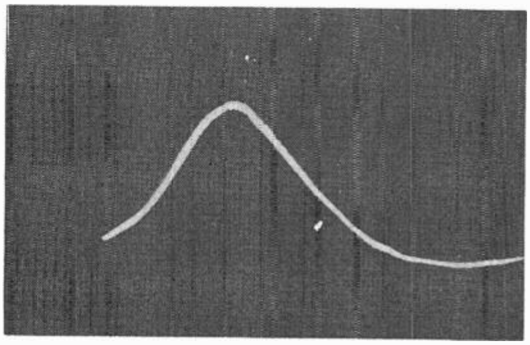
The resultant active remanent magnetic flux associated with an element of magnetic tape upon completion of the recording process depends upon the entire previous magnetic history of the element. If the tape has been "erased" by a good alternating current erasing device before recording occurs, then, 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.

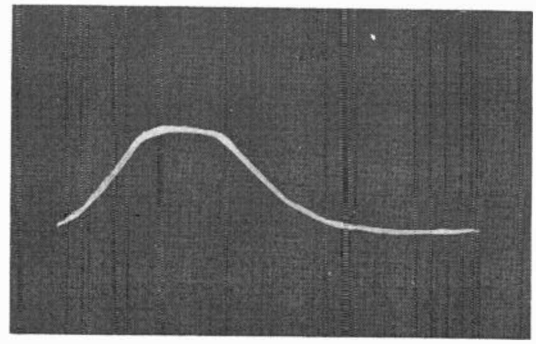
The oscillograms were obtained as follows. With the unmagnetized tape in a recording position but stationary with respect to the recording head, the recording head current was varied manually from zero through a cyclic pattern as specified for each oscillogram. In this manner the section of tape in the 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.

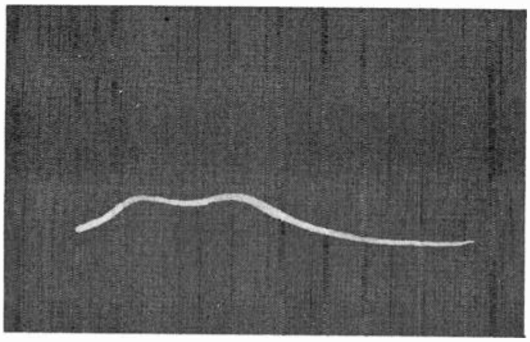
Figure 1(a) shows the remanent flux produced by increasing the recording head current from zero to + 3.0 milliamperes and returning to zero. The shape of the trace results from the air gap field distribution for this single cycle of excitation together with the non-linear saturation and hysteretic properties 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.



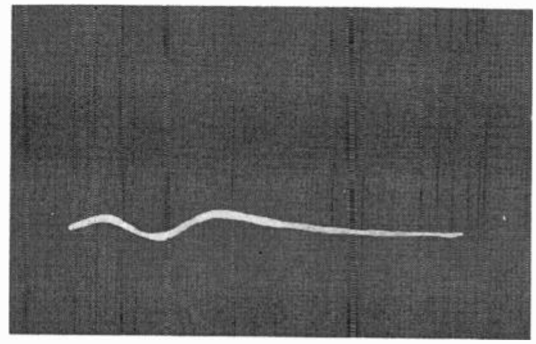
(a)



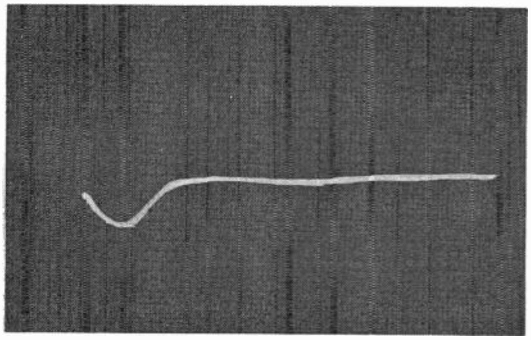
(b)



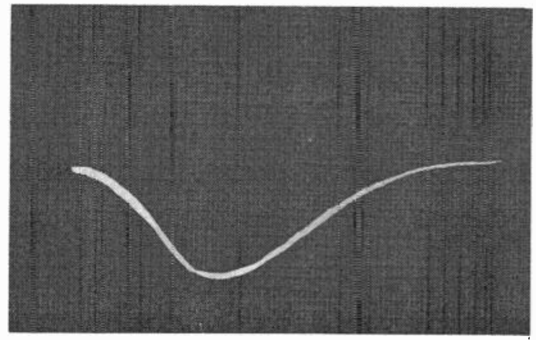
(c)



(d)



(e)



(f)

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)

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(Please see inside back cover for additional names)

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January — February, 1953

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The Institute of Radio Engineers

I.R.E. 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*

L. J. Anderson
RCA Victor Division of RCA
Camden, New Jersey

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

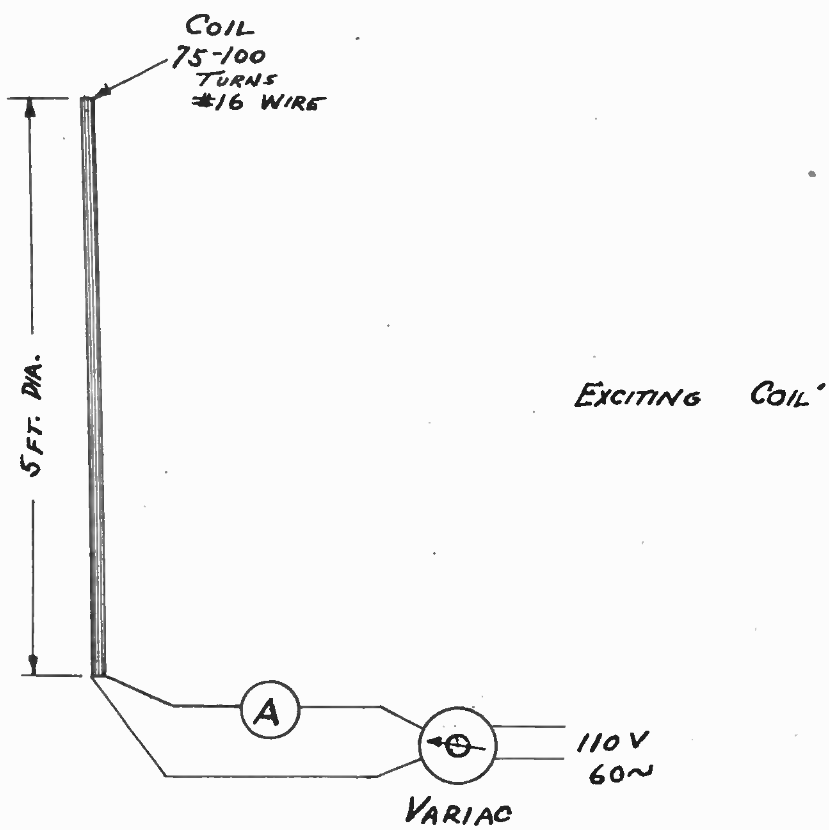
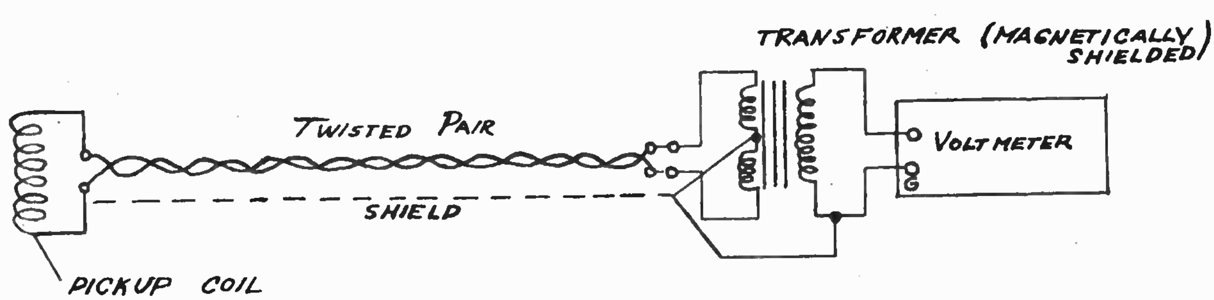


Fig. 1



FIELD MEASURING EQUIPMENT

Fig. 2

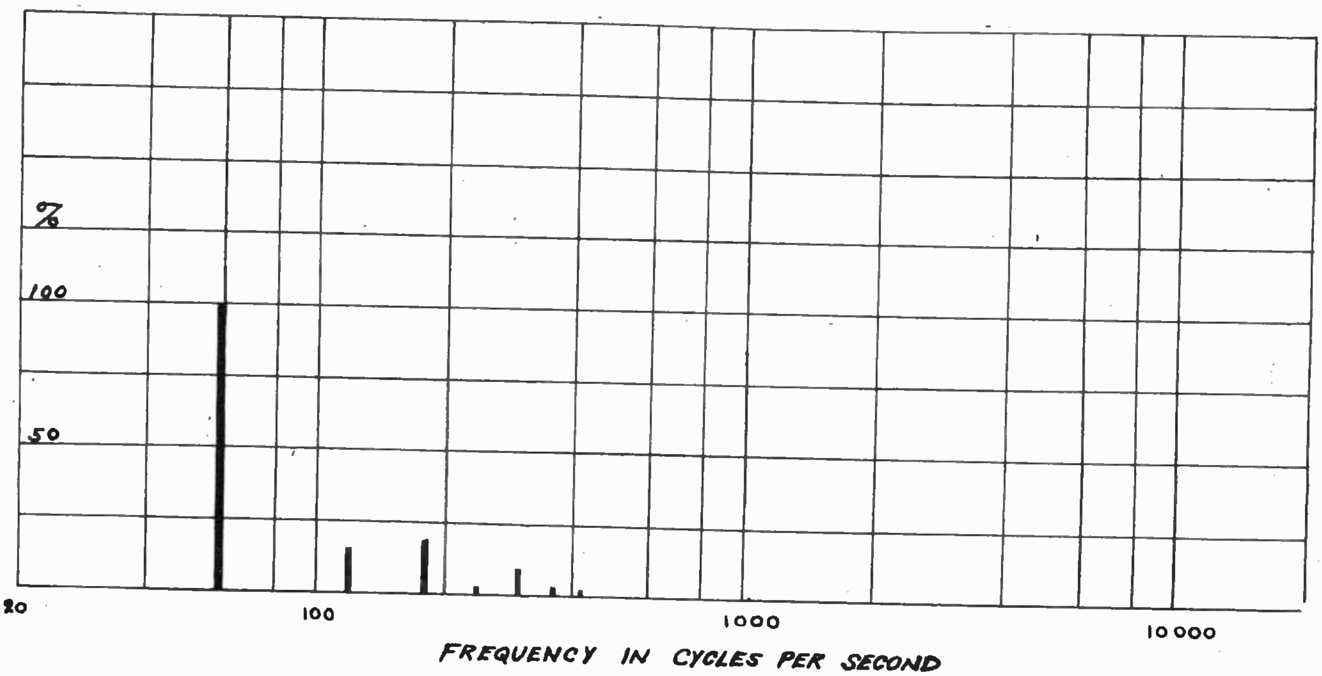


Fig. 3 - Field Analysis - Typical Broadcast Studio.

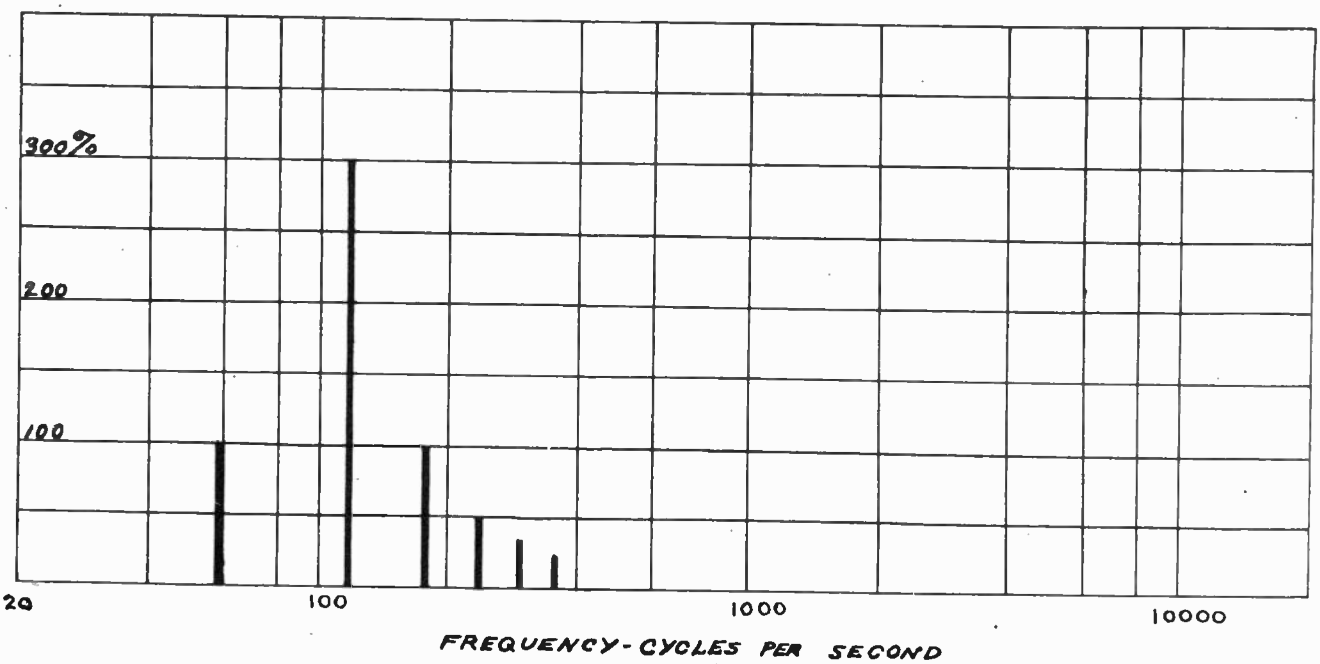


Fig. 4 - Field Analysis Near Power Amplifier.

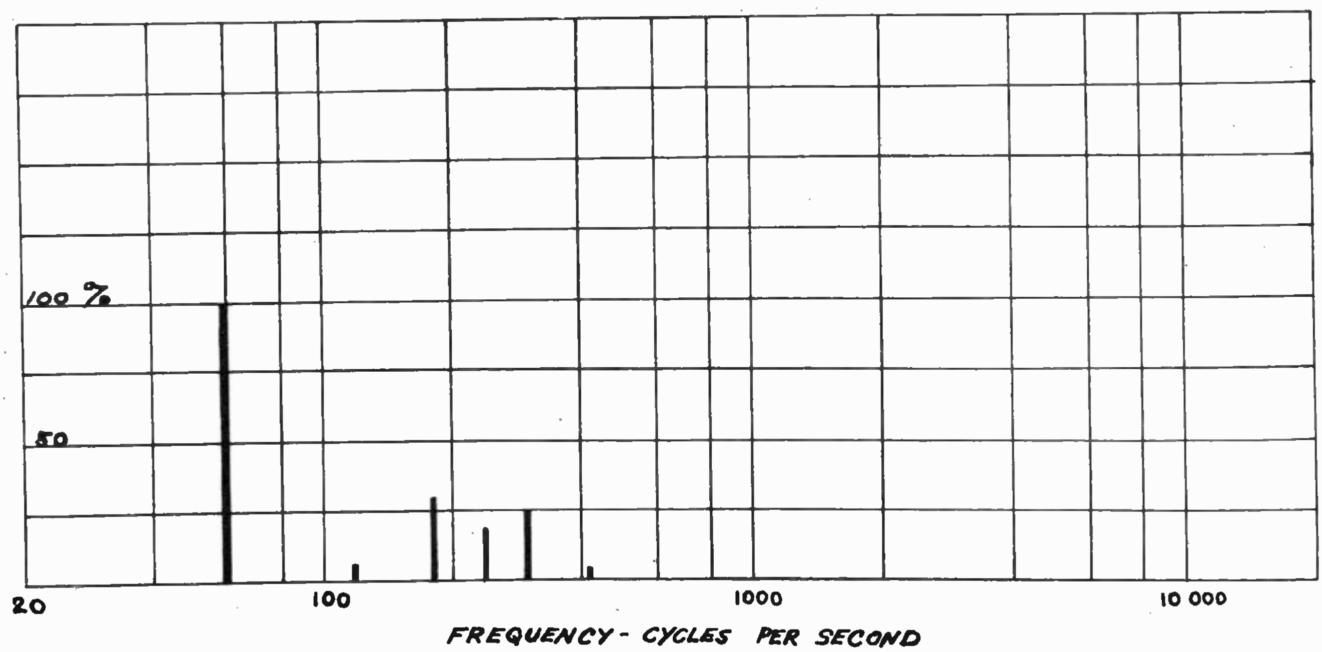


Fig. 5 - Field Analysis Near 80 Watt Fluorescent Fixture.

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

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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.

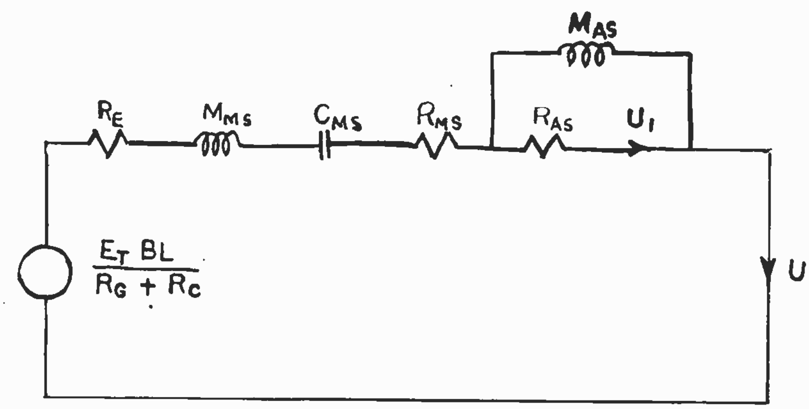
V. Distortion Analysis

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.



$$R_1 = \frac{B^2 L^2}{R_G + R_C}$$

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

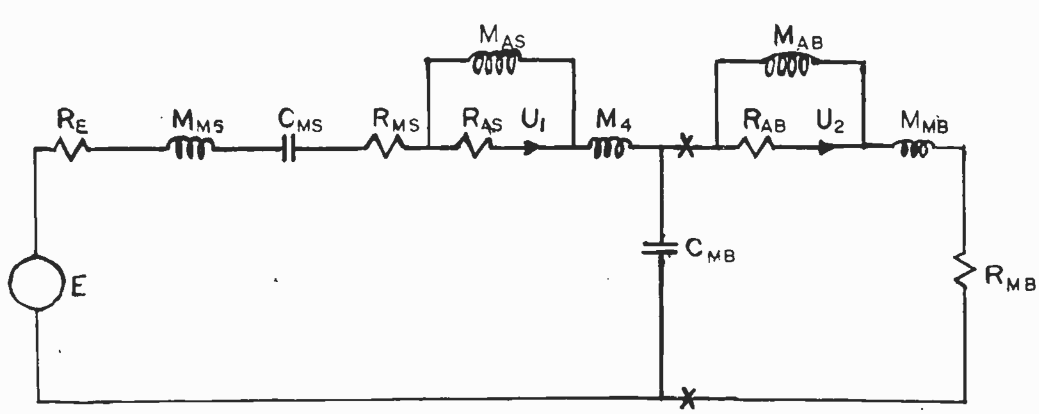


Fig. 2 - Equivalent circuit of a loudspeaker mounted in a vented enclosure.

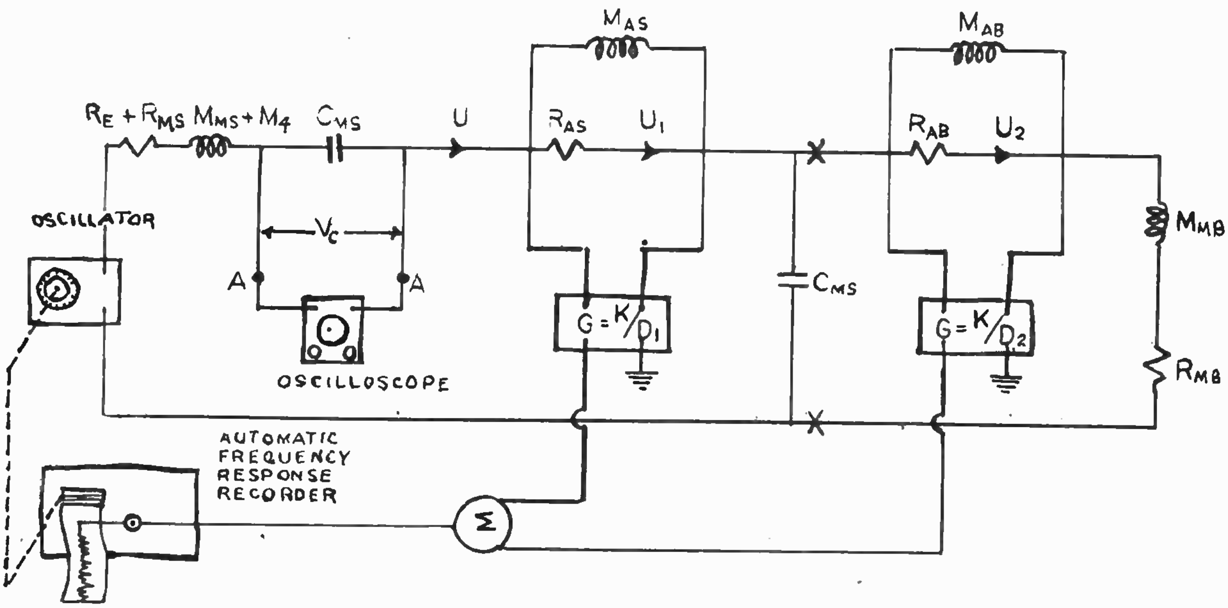


Fig. 3 - Schematic diagram of analogue computer.

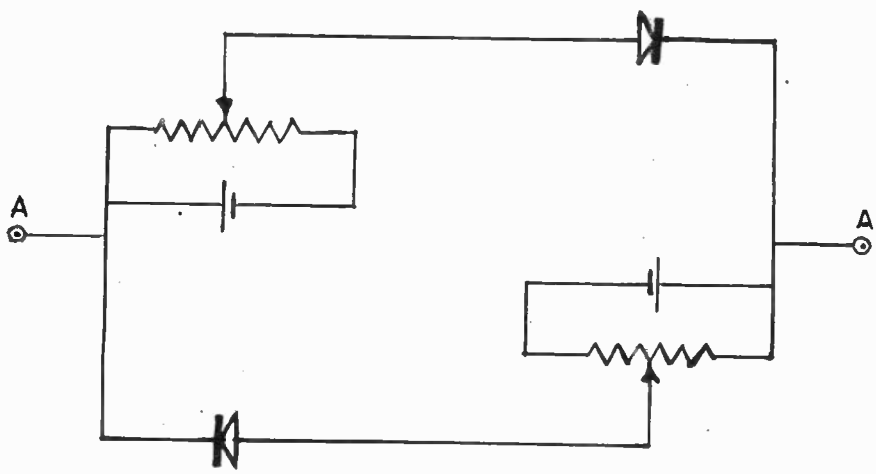


Fig. 4 - Circuit used for introducing amplitude distortion.

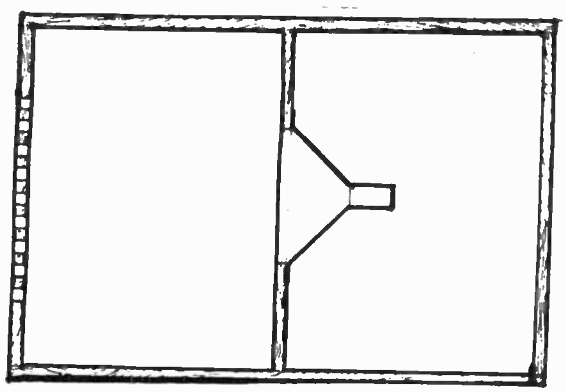


Fig. 5 - Sectional views of completely enclosed loudspeaker.

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

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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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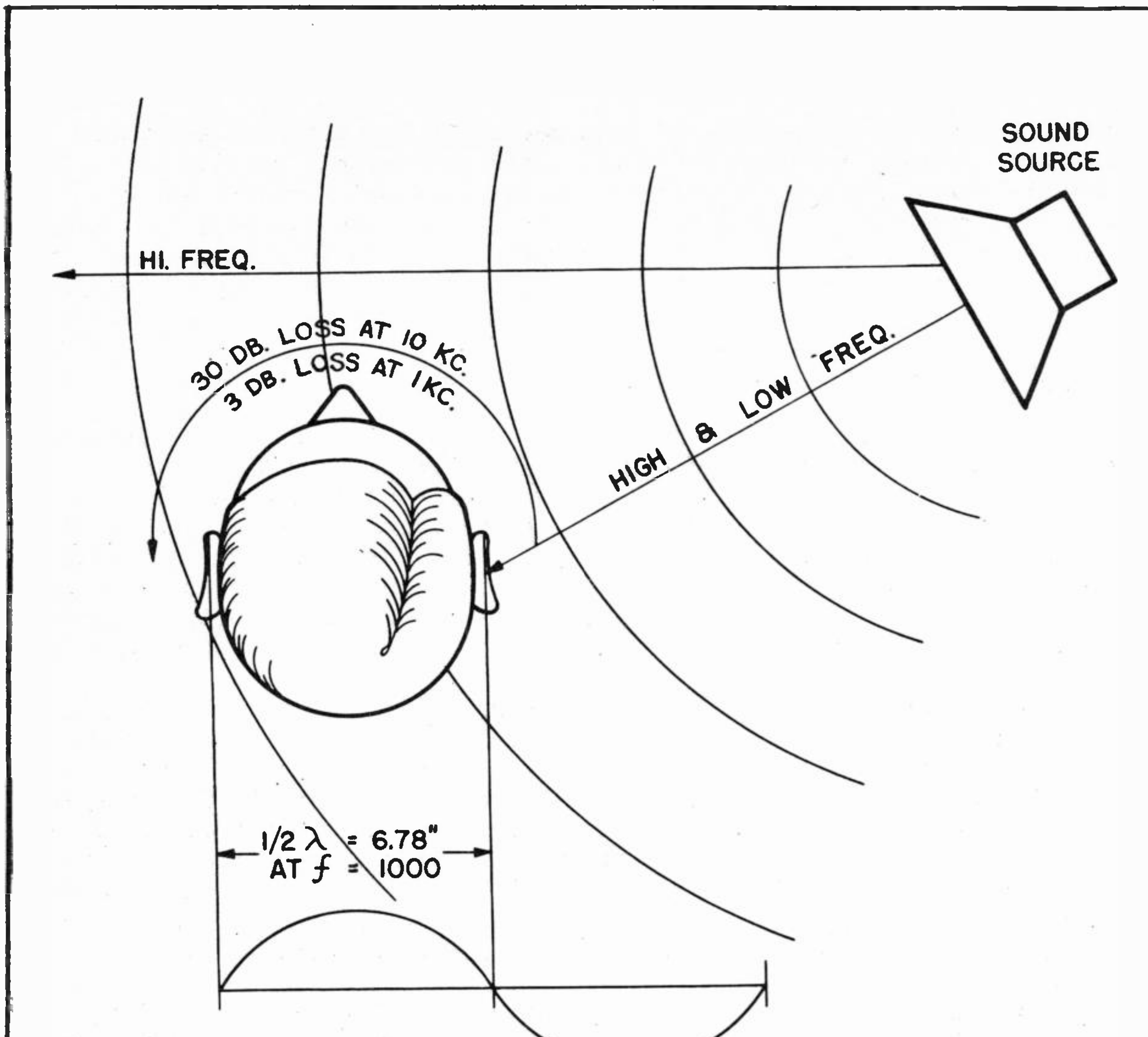

Fig. 1

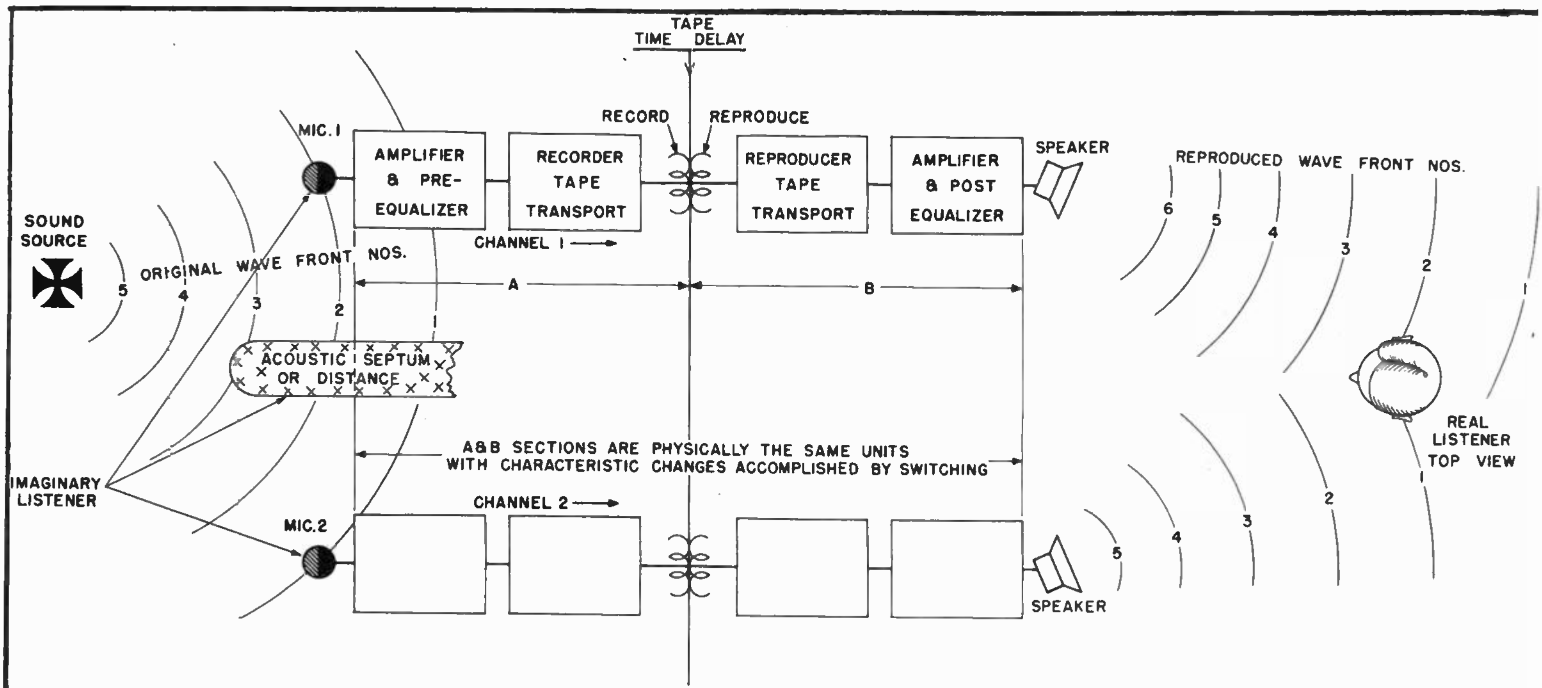

FIGURE 2 NOTE THAT APPROXIMATELY THE SAME SOUND PHASE EXISTS AT THE REAL LISTENERS POSITION AS AT THE IMAGINARY LISTENERS POSITION. THE SLIGHT DIFFERENCE REPRESENTS THE REPRODUCTION ERROR INHERENT IN MICROPHONE & SPEAKER RELATIONSHIPS.

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)

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

March — April, 1953

Published Bi-Monthly

Volume AU-1 **Number 2**

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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

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

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ELECTRONIC MUSIC - PAST, PRESENT AND FUTURE*

Earle L. Kent
C. G. Conn, Ltd.
Elkhart, Indiana

In keeping with the PGA policy of including, from time to time, subjects outside of the strict boundaries of Technology, we have invited Dr. Kent to submit his very fine review of Electronic Music. Future developments in Audio techniques, and, therefore, this subject is bound to be of interest to all PGA members. — Editorial Committee.

In ancient time the Greek musical theorists applied the word "organic," as a general term, to instrumental music. For several centuries the term organ has been applied to musical instruments, wherein the tones are generated by pipes blown by suitable means with air under pressure. The principle involved in obtaining sound from a pipe with wind pressure dates back to about 284 to 246 B.C. By 100 A.D. the organ embraced about three octaves, and water was used to obtain steady wind pressure. During this long period the organ was gradually improved, and electricity has played an increasing role in recent years. First, electrically driven blowers replaced the man-power required to furnish the air pressure and then electromagnets replaced the manually operated air valves to greatly reduce the physical effort required to play the organ.

For the past 55 years various attempts have been made to produce musical tones electrically. The first instrument of note that produced tones electrically was built long before vacuum tube amplifiers were available, and it comprised over thirty carloads of equipment. Since that time several hundred patents have been issued on instruments that produce musical tones electrically, and great strides have been made in electronic circuits and components. At present there are a number of electronic musical instruments on the market and activity in this field is growing at an exponential rate.

There are several factors that work against making improvements in musical instruments. In fact, improvements in musical instruments are up against the most formidable combination of conservatism confronting any product. Between the manufacturer and the consuming public stand several conservative forces. They are the teacher, the composer, the music publisher, the musician, and the listening public.

During the past 75 years the Conn company has witnessed cases of inertia that are almost unbelievable. The flute used today is hardly distinguishable from the flute as it came from Boehm's hands in 1847, but only in the past

*Part of paper presented before the Chicago Section, Institute of Radio Engineers on December 19, 1952. Manuscript received January 23, 1953.

forty years has it won undisputed and universal acceptance. Similar stories can be told about the clarinet, cornet, french horn, and other musical instruments. Some improvements have made the grade after many years of struggle, but many have not. For example, the Conn company developed the mezzo-soprano saxophone in F about 30 years ago. Many thought that the E^b alto was not a good lead voice for the saxophone choir because it was really an alto and pitched too low for a lead. The B^b soprano was difficult to play. The F saxophone seemed to be the answer. After building the instrument, its beauty of tone and ease of playing was recognized at once, but there were no F parts in orchestrations. We worked closely with music publishers to persuade them to include parts for this instrument in their music, but their answer was that they would do so when there were a sufficient number of F saxes being played to warrant it. You can still hear an F sax now and then in some dance bands, but after a ten year struggle with the publishers we gave it up.

Many examples could be given about the teachers who refuse to teach students on a so-called "non-standard" instrument, the superstitions of directors and musicians who all want something new so long as it isn't different, and even the listening public who put the brakes on progress by their apathy and even antagonism toward the new instruments and new effects. New and unfamiliar sounds are not generally liked.

The obstacles met by the traditional instruments have been multiplied for the electronic instruments. In 1936 the American Guild of Organists published the statement that in all-electric instruments, the basic tone is necessarily devoid of all color. The various orchestral instruments have been imitated by the flue pipes and reeds of the pipe organ with varying degrees of fidelity. When an attempt is made to create these orchestral tones by electronics, we find that organists do not like too realistic duplication of the tones of the original instrument, but that they insist the tone qualities must be those with which they have become familiar on the pipe organ. We can create new and beautiful tonal colors not found on pipe organs, but when organists hear these tones they are confused and displeased. It is possible to do things with electronics that pipe organ builders have tried in vain to do, and yet, when we do these things they are opposed by organists.

In spite of these restrictions and handicaps, electronic musical instruments are steadily making progress. There are several good reasons for this progress. The electronic servants that serve mankind so well in so many different ways are quite capable of producing musical tones. It is taken for granted that electronics should serve in recording, reproducing, amplifying, and transmitting music, so it is not unreasonable to expect it to serve in the actual creation of music. In fact, it is now possible to produce beautiful tones electronically in solo type or in organ type instruments that have greater versatility than was ever dreamed of in traditional instruments. When tones are produced electronically the musician does not need to exert any energy pounding, blowing, or scraping, to produce the tones, but can devote his entire attention to the control of the tone to make music. When a beginner starts to learn to play most of the traditional instruments, he must spend considerable time in developing muscles, muscular control, and techniques in order to make a tone that does not sound very discouraging to him and to all who have to listen to it. With an electronic musical instrument, the beginner can make a variety of beautiful tones from the beginning, and can start at once to learn to play the right note at the right time with the proper expression.

Electronic musical instruments of the organ type have become an accepted fact to the general public and are enjoying widespread application in liturgical, concert, and entertainment music.

These instruments are the only twentieth century musical instruments; and they are opening new horizons in music.

A demonstration of the versatility and scope of the electronic organ was shown in the presidential nominating conventions here last summer. The International Amphitheatre, where the conventions were held, had no organ; and the Connsonata was selected to fill the need for this type of music. The seating capacity is twelve thousand. In addition to the permanent seating space on the main floor and balcony, there is a clear central arena 123 x 236 feet with a clear height of 73 feet. The floor space alloted for the organ was less than 45 square feet, but this was more than enough since room remained for several chairs. To install a pipe organ, with sufficient power to dominate, supplement, or override the sound level, occasioned by the customary enthusiastic demonstrations of so many people in so large a room, would not be practicable on a temporary basis, and impossible in the installation time and space allowed.

The two basic functions in musical instruments are the generation of musical tones and the control of those tones. The generation of good tones in a practicable manner is relatively easy due to the advanced state of electronic circuits and components. Loudspeakers have generally been the weak link here because the general run of radio type loudspeakers have not been adequate for the stiff requirements. The high peak power, the 32 to 12 or 15,000 cycle range, and the sustained nature of the tones, all demand good transient response, good frequency response, and low distortion. However, speakers are now available that give good results if properly used.

The control of the tone includes: the starting and stopping of the tone (the envelope of the attack and release are quite important), the quality of timbre of the tone, the tremulant, the dynamics or loudness, the location of the source of sound (including echo effects) and the formant of the tones. Solo and ensemble effects must be considered, and it is important that the organist "feel at home" at the console and find the controls where he is accustomed to finding them.

The organ type instrument is the grandest and most magnificent of all instruments invented by human genius. The organ challenges the total capacity of the organist's mind and body in the production of music, but it does not give him the intimate control over the tone that is possible with a monophonic or solo type instrument; so we must not neglect this phase of electronic instruments, and more will be said about that later.

The Connsonata uses plate keyed Hartley oscillators for the tone generators. There is one oscillator for each key and pedal plus a few extra that are keyed through couplers that operate beyond the range of the keyboard. The frequency vibrato is produced by introducing a low frequency sine wave at the grid of the tube. A flute-like tone is produced at the tap on the tank coil, and a pulse tone is produced across a resistor in the cathode circuit. By mixing these two basic ingredients and by means of filters for formant control, we produce a basic organ tone — the diapason, the oboe, the vox humana, the chorus reed or organ trumpet, and others. One test of an organ is to determine how the four basic tone families sound in ensemble. First, the diapason group, second the flutes, third the strings, and finally the reeds.

Organs are called upon to play concert music: they are used extensively in churches for liturgical music, they are used for novelty effect, particularly on the radio, and they must be able to play music for general entertainment or theatrical work.

While electronic organs are now generally accepted by the general public in all types of organ application, the same is not true for electronic solo-type instruments. They have not supplemented conventional musical instruments in the way that they can and should. It is possible to build solo electronic instruments that fulfill all musical requirements, but it will take time to overcome the inertia that prevents a good instrument from reaching the fellow who needs it badly. Small school bands or orchestras often have but one or perhaps none of certain instruments in their organization. For example, perhaps the group has one good oboeist or french horn player, and he graduates leaving no one to play those parts. A good electronic instrument could play any of the missing parts well enough to be a great help. Perhaps one section is weak and needs reinforcement to obtain proper balance. An electronic instrument could do it. The teacher who must teach several instruments he does not play could play the parts for the student if he had a versatile electronic instrument that would closely simulate the instruments he is teaching. There are many other instances where electronic solo instruments can substitute for traditional instruments. If they can do a good job as a substitute, it will gradually become apparent that they can stand on their own merits in serious music. I have talked to progressive directors who feel that this need can and will be filled satisfactorily with electronic musical instruments. Of course, one fellow retorted that he never expected to see a band marching down the football field with power cords dragging behind. That would be out of the question, but with the advent of the transistor and other improvements in small effecient electronic components, we may some day see an all electronic band without the cords dragging behind.

Technological advances in electronic circuits and components will bring improvements in electronic musical instruments just as fast as people will tolerate them, and it is already evident that these changes will come faster than they have on the traditional instruments.

In addition to the improved electronic instruments of the solo and organ type, I believe the future will find still different instruments in use. These instruments will further expand the possibilities in music by doing what is impossible on other instruments. For example, I started the development of such an instrument, which I call an electronic music box, as a spare time project. This instrument was described at the National Electronics Conference last year.[1] Percy Grainger, a noted musician and composer, came to Elkhart to see my music box and was very enthusiastic about its possibilities, since he had composed music having such irregular rhythm and scales that no one could play it. He surprised me by saying that the greatest handicap to music is the musicians hands. Of course, Mr. Grainger is a visionary rebel, and some musicians would take offense at that statement, but what I have in mind would not hurt musicians. I do not believe that actors have suffered any loss because Walt Disney is able to produce motion picture fantasies by animated cartoons. He has just expanded the motion picture field, and I believe that something like my music box will produce musical fantasies for recordings, and the like, that could never be produced through manual manipulations. Lack of time has made my progress slow in completing the development of the music box, but I am still working on it and perhaps you'll hear more about it later.

These new devices will not only remove the limitations in tonal production, as do the other electronic musical instruments, but will remove the limitations of the musician's hands. They will make it possible for a composer to produce his music directly without going through interpreters.

1. Kent, Earle L., "An Electronic Music Box," Transactions of the IRE-PGA. Proceedings of the National Electronic Conference, Vol. 7, February, 1952.

REPORT OF
SOUTHWESTERN I.R.E. CONFERENCE AND ELECTRONICS SHOW
San Antonio, Texas -- February 5, 6 and 7, 1953

B. B. Bauer
Shure Brothers, Inc., Chicago 10, Illinois

Following the pattern of growth of electronics in the South and Southwest, the 1953 Southwestern IRE Conference was bigger and better than ever. Registration exceeded 1100 and two parallel sessions of technical papers and over 100 exhibits provided an interesting and busy time for all those who attended.

The session on Audio was held on Saturday morning, February 7, under the chairmanship of Dr. Richard E. Lane of the University of Texas. Approximately 225 members attended this session. Four papers were presented as follows:

1. "Electron Beam Reproducing Head for Magnetic Tape Recording" -- Dr. A. M. Skellett and Dr. Lawrence E. Loveridge, National Union Radio Corporation, and Mr. J. Warren Gratian, Stromberg-Carlson Company.

   Present reproducing heads for magnetic tape depend upon the threading of flux through a coil. The output is proportional to the frequency of recording and to the speed of the tape. The electron tube reproducer guides the flux from the tape into a cathode-ray type of tube where it deflects the beam in accordance with the absolute magnitude of the flux. Thus, the output is dependent neither on the frequency nor the speed of the tape. In addition to this, the new head can be made many times as sensitive as the old ones. This development has been sponsored by the Bureau of Ships of the United States Navy under Contract NObsr-57452.

2. "Status of Military Research and Development in Acoustics and Audio" -- Mr. Paul Weber, Bureau of Ships, Navy Department, Washington, D. C.

   The fields of acoustics and audio engineering are finding many new military applications within the Army, Navy and Air Force. Continued progress is also being made toward improving the performance of existing audio systems and techniques to better meet increasingly stringent operational requirements. Some recent developments and proposed projects in these fields will be discussed. Problems which are as yet unsolved will be mentioned. The procedure established within the Department of Defense for coordinating all research and development in the broad field of acoustics-in-air will be described.

3. "Acoustical Damping of Loud Speakers" -- Mr. B. B. Bauer, Shure Brothers, Inc., Chicago, Illinois

   Acoustical damping can be applied advantageously to loudspeakers. The advantages of acoustical damping as a replacement for, or as an adjunct to the electrical damping are discussed. Equivalent circuits for the loudspeaker and the enclosures, and simplified theoretical and experimental design methods are described. Transient performance of the electrical circuit and of the loudspeaker is demonstrated.

4. "The Fluid Sound Phonograph Pickup" -- Mr. Bruce D. Eytinge, Institute of Inventive Research, San Antonio, Texas

   A new type of transducer has been developed wherein the resistance of a column of conducting fluid is varied by mechanical motion which is made to increase or decrease the cross-sectional area of the column. Experimental phonograph pickups using this principle show extremely good frequency response at the extreme lows, and output voltages comparable with crystal transducers. Other properties of these units are discussed.

A spirited discussion which ensued caused your reporter to almost miss his plane out of San Antonio.

Arrangements are being made to include some of these papers in the coming issue of TRANSACTIONS.

Two of the highlights of the Conference were the toastmastering of Mr. Trevor H. Clark, of the Southwest Research Institute, and the humor of Dr. C. P. Boner, of the University of Texas, who was the Banquet Speaker.

REPORT OF THE SECRETARY-TREASURER

Marvin Camras
Armour Research Foundation, Chicago 16, Illinois

Our membership, our income, and our treasury balance continue to grow. At the end of 1952, we had a total of 2031 members, compared to 794 at the end of 1951. Our financial statement at the end of the year compares very favorably with the figures for the previous period and for the previous year.

	Period Ending Dec. 31, 1952	Period Ending Sept. 30, 1952	Period Ending Dec. 31, 1951
RECEIPTS			
Assessments	$4864.00	$3976.00	$1530.00
Other Income	961.61	769.21	101.00
Matched Funds	2579.00	1500.00	1000.00
Total Income	$8404.61	$6245.21	$2631.00
EXPENSES	3959.59	3273.09	1190.77
BALANCE ON HAND	$4445.02	$2972.12	$1440.23

REPORT OF THE ACTIVITIES OF THE TAPESCRIPT COMMITTEE

A. B. Jacobsen
University of Washington
Seattle, Washington

The Tapescript Committee has procured and prepared for distribution a number of technical papers primarily on the subject of audio. Recordings were made of Convention papers, in general, and offered to interested groups.

On the whole, Convention papers have not proved as satisfactory as a specially prepared paper. The best example of the specially prepared paper is "Germanium — The Magic Metal," which was produced by the General Electric Company for the IRE especially to determine the feasibility of this medium of distribution of technical papers. Two copies of this paper are currently being distributed and a total of between 25 and 30 showings have been made. Response to this paper, and the manner of distribution, has been good. "Germanium — The Magic Metal" runs approximately 40 minutes and has about 75 slides in color. Narration is by two General Electric engineers who are directly associated with the subject.

Standards for Tapescripts should be set and it is felt that full track at $7\frac{1}{2}$ inches per second tape speed, on 1200 foot rolls, is the most universal sound recording medium, and with $3\frac{1}{4} \times 4$ slides, 35 mm slides, or single frame 35 mm film, should be desirable standards. Besides the slides and the sound recording, a copy of the script is very important, and a summary of suitable length for meeting announcements and notices, should be available to the program chairman. A few references to applicable literature would be of considerable value to the person who must attempt to answer questions at the end of the recorded presentation.

We believe that there is a great need for specially prepared, high quality recorded technical papers for those groups who would otherwise be unable to procure live papers. The Tapescript Committee has been aided by many individuals and concerns in carrying out this preliminary phase of the program.

ANALYSIS OF A SINGLE-ENDED PUSH-PULL AUDIO AMPLIFIER*

Chai Yeh
University of Kansas
Lawrence, Kansas

Summary

This paper deals with a theoretical circuit analysis of a single-ended push-pull audio amplifier. Linear tube characteristics and small signals are assumed. The problem of impedance matching is first discussed. The properties and the requirements of a satisfactory driver stage are analyzed, and an output stage using an impedance matching output transformer is discussed. The effects of the plate-to-ground capacitance of the driver stage on the frequency-gain characteristic and of the choice of the tube and circuit parameters are analyzed. The paper concludes with some experimental results indicative of the inherent properties of the amplifier.

In a recent article by Peterson and Sinclair,[1] special circuits have been suggested by which a high fidelity, low distortion audio amplifier can be constructed without using output transformers or in which the requirements of an output transformer can be greatly simplified. The present article will give a theoretical analysis of the basic circuit used by Peterson and Sinclair. It will be observed that many desirable characteristics of this amplifier circuit can be deduced from this analysis.

Equivalent Circuit of the Basic Single-Ended Push-Pull Amplifier Circuit

The basic circuit of a single-ended push-pull amplifier is shown in Fig. 1a. Assuming class-A operation with resistive load, and neglecting the effects of interelectrode capacitances, an equivalent circuit of the phase inverting driver stage and the single-ended push-pull output stages is shown in Fig. 1b. With the circuit constants, tube parameters, and loop currents as indicated, one may write the following simultaneous equations:

$$\begin{aligned}
&I_1(2R + r_{p1} + \mu_1 R) + I_2 R_L - I_3 R_L = \mu_1 E_1 \\
&I_1 \mu_2 R - I_2 R_L + I_3(r_{p2} + R_L) = 0 \qquad (1)\\
&-I_1(r_{p2} + \mu_2 R) + I_2(r_{p2} + R_L) - I_3 R_L = 0
\end{aligned}$$

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

From which the individual loop currents can be evaluated. The net current flowing through the load resistance R_L is

$$I_3 - I_2 = \frac{-\mu_1 E_1 (2\mu_2 R + r_{p2})}{R_L(4R + 2\mu_1 R + 2\mu_2 R + 2r_{p1} + r_{p2}) + r_{p2}(2R + \mu_1 R + r_{p1})} \qquad (2)$$

The power output P_o is given by $P_o = (I_3 - I_2)^2 R_L$. The power sensitivity, which is defined as the ratio of the power output to the square of the input voltage is then given by

$$\text{Power sensitivity} = \frac{P_o}{E_1^2} \quad \frac{\mu_1^2 A^2 R_L}{R_{p2}^2 \left[1 + \frac{R_L}{r_{p2}} (2+A)\right]^2} \qquad (3)$$

$$\text{where} \quad A = \frac{2\mu_2 + r_{p2}/R}{2 + \mu_1 + r_{p1}/R}$$

The condition for maximum power sensitivity by varying the load resistance can be obtained by taking the derivative of Equation (3) with respect to R_L and equating to zero. The optimum load required is thus

$$R_L \text{ (for maximum power sensitivity)} = \frac{r_{p2}}{2 + A} \qquad (4)$$

It can easily be proved that Equation (4) also represents the source impedance of the amplifier. The value of this maximum power sensitivity is given by

$$\frac{P_o}{E_1^2} \max = \frac{u_1^2 \; A^2}{4 \, r_{p2} \, (2 + A)} \qquad (5)$$

To get high power output, A should not be small. But consideration of negative feedback (which will be discussed in the next section) puts restrictions on the choice of the tube parameters, and indicates that A should be small.

The ratio of power output to the maximum power possible is given by

$$\frac{P_o}{P_o \max} = \frac{4 \frac{R_L}{r_{p2}} (2 + A)}{\left[1 + \frac{R_L}{r_{p2}} (2 + A)\right]^2} \qquad (6)$$

Equation (6) is calculated for various values of A and is plotted in Fig. 2 as the ratio of $\frac{r_{p2}}{R_L}$ is varied. It is seen that even for a wide variation of A, the matching conditions permit r_{p2} to have a value of from 2 to 3 times that of R_L.

The above discussion on maximum power sensitivity made use of linear analysis with resistive load. However, the actual maximum un-distorted power output, which is limited at one extreme by grid current and at the other extreme by plate current cutoff, will occur at higher load resistance values. The consideration of plate and screen-grid dissipation also favors higher load resistance.

The Phase-Inverting Driver Stage

Information concerning the behavior of the driving stage can be deduced by solving for the current I_1 in Equation (1). Thus

$$I_1 = \frac{-\mu_1 E_1}{r_{p1} + (2 + \mu_1)R + R_L(2\mu_2 R + r_{p2})/(r_{p2} + 2R_L)} \qquad (7)$$

Equation (7) is in a form similar to that of an amplifier employing negative current-feedback of one considers the term $r_{p1} + (2+\mu_1)R$ as the internal resistance R_i and $(2\mu_2 R+r_{p2})R_L/(r_{p2}+2R_L)$ as the effective load resistance R_e of the driver. $\mu_1 E_1$ is the fictitious voltage.

The voltage across R_e is the actual voltage output of the single-ended push-pull output stage. The driving voltages for that stage is derived from the voltage drops through the resistances R. Thus the a.c. voltage across the plate-to-cathode of the phase-inverting driver supplies both the driving as well as the output voltages. This puts a restraint on the D.C. plate voltage supply for the driver. A high D.C. plate voltage greater than the sum of these a.c. voltages is needed in order that serious non-linear distortion can be avoided.

The negative feedback that exists between the input and output of the amplifier stages can be expressed in terms of a feedback factor β, the ratio of the feedback voltage available at the grid terminal of the driver stage to the output voltage across the load. Thus

$$\beta = (2 + r_{p2}/R_L) \ / \ (2\mu_2 + r_{p2}/R) \qquad (8)$$

The inherent feature of this negative current feedback is to maintain the current I_1 practically independent of load changes. This would mean that the driving voltages for the output stages could also be maintained unchanged, a very desirable characteristic of the amplifier if we can achieve it. To so design this stage would require

$$R_i \gg R_e$$

$$\text{or} \qquad A/(2+r_{p2}/R_L) \ll 1 \tag{9}$$

and for a matched load, Equation (9) becomes

$$A/(4+A) \ll 1 \tag{10}$$

To satisfy the requirement indicated in Equations (9) or (10) would call for a small value of A. But a smaller A means reduced output voltage and power output. A compromise should always be worked out between these contradictory requirements.

A proper choice of A gives some guidance for the choice of tube parameters and circuit constants. For a typical operation, if $\mu_1 \gg 1$, $r_{p1} \cong R$ and $R > r_{p2}$, then with a matched load for maximum power sensitivity

$$A \cong 2\mu_2/(3+\mu_1) \; .$$

$$\text{and} \qquad \beta \cong 4/2\mu_2$$

Thus, if μ_2 is small, μ_1 is large; A is very small and β is comparatively large. A larger β helps to reduce distortion while large power output is still possible if the input signal is large.

A choice yielding a larger value of μ_1 is always desirable. The larger value of μ_1 allows larger power output while still keeping the reasonably small value for A which is required for constant I_1 operation. It also gives a larger driving voltage for exciting the power stage.

The Power Output Stage

In the basic circuit of this amplifier (Fig. 1a), if one assumes identical tube characteristics; same d.c. plate voltage per tube, and equal and opposite grid swings, the a.c. plate currents add in the load in exactly the same way as in a conventional push-pull circuit and the distortion-cancelling feature is retained. The d.c. plate currents are, however, in series for the plate supply.

The need for a high voltage plate supply (from 800 to 1000 volts) with good regulation to operate this amplifier is, to a certain extent, an objection to most experimenters. If only a moderate plate supply is available, a circuit suggested by Peterson and Sinclair[2] containing an output transformer, is recommended. A modified version of their circuit is shown in

Fig. 3. It uses an output transformer with separate primary windings. Here the d.c. plate currents of the two tubes flow through the two halves of the primary windings in opposite directions so that the problem of d.c. saturation is not acute. The tubes are in parallel across the d.c. plate supply. The signal currents flow through the two halves of the primary windings which are connected in parallel by the by-pass capacitors. The leakage reactance between the halves of the primary windings plays no important part in switching transients[3] and the requirements of the output transformer are not as critical as they were in conventional push-pull circuits.[2]

The linear analysis used thus far can be extended to include the impedance matching transformer provided class A_1 operation in triodes is assumed. Assuming an ideal transformer, equivalent circuits of the output stage of a single-ended push-pull amplifier with impedance matching output transformer are represented in Fig. 4. Fig. 4a can be reduced to Fig. 4b, which shows that as far as the varying components are concerned, the tubes are in parallel. Fig. 4c is another version of Fib. 4b.

Figures 4b and 4c are very similar to the equivalent circuit of a conventional push-pull circuit used for power considerations alone. Thus a simple relationship between these two types of push-pull circuits can be deduced. For impedance matching the single-ended push-pull circuit will need an output impedance which is one-fourth of the plate-to-plate impedance of a conventional push-pull circuit.

The concept of the composite tube which is so helpful in the analysis of conventional push-pull circuits when the path of operation extends beyond the linear region of the plate characteristics can be borrowed to advantage. Thus the analysis of the single-ended push-pull circuit can be extended to include the operation as class AB_1, although graphical analysis is required. To extend the operation beyond class AB_1 into class AB_2 or class B operations is not feasible since the flow of grid current will affect the biases unless special circuits can be arranged.

The Effect of the Plate-To-Ground Capacitance of the Driver Stage on the Gain-Frequency Characteristics of the Amplifier

As was first pointed out by Peterson and Sinclair, the capacitance from plate to ground of the driver stage is going to affect the gain of the amplifier at higher frequencies. With normal circuit and tube parameters, this effect will first be noticed at video frequencies. The previous analysis can be extended to include this effect. In Fig. 1b, if a capacitor C (plate-to-ground capacitance) is connected across the points marked x-x and a loop current is assigned to the loop on the right hand side of the capacitor, one has a new equivalent circuit for this analysis. As the currents flowing through resistances R are different now, we may expect different driving voltages on the grids of the output tubes. This difference is, however, small if the shunting capacitance is small or the frequency in consideration is not too large. By writing the simultaneous equations of the loops, and solving the currents as before, one obtains, for a sinusoidal input signal, the net current through the load resistance as

$$I_3-I_2 = \frac{-\mu_1 E_1\left[\mu_2 + \frac{1}{j\omega CR}\;(2\mu_2 + r_{p2}/R)\right]}{R_L\left\{(1+\mu_1+\frac{r_{p1}}{R})\left[(2+\mu_2+\frac{r_{p2}}{R}+\frac{r_{p2}}{R_L})\right]+\frac{1}{j\omega CR}\left[2(\mu_1+\mu_2+2+\frac{r_{p1}}{R})+\frac{r_{p2}}{R_L}(2+\mu_1+\frac{r_{p1}}{R}+\frac{R_1}{R})\right]\right.} \tag{11}$$

And thus the ratio of the voltage gains at higher and lower frequencies can be found by taking the ratio of Equation (11) to Equation (2), and is given by

$$K_h/K_L = \frac{\frac{\mu_2}{2\mu_2+r_{p2}/R} - j\frac{1}{\omega CR}}{\frac{(1+\mu_1+\frac{r_{p1}}{R})(2+\mu_2+\frac{r_{p1}}{R}+\frac{r_{p2}}{R_L})}{(2+u_1+\frac{r_{p1}}{R})(2+\frac{r_{p2}}{R_L}+\frac{2\mu_2+r_{p2}/R_L}{2+\mu_1+r_{p1}/R})} - j\frac{1}{\omega CR}} \tag{12}$$

For abbreviation, let us define the following terms:

$\omega_o = 1/CR$, or $f_o = \frac{\omega_o}{2\pi} = \frac{1}{2\pi RC}$, the upper reference frequency,

which is determined by the RC time constant of the resistance R and the plate-to-ground capacitance C of the driver stage.

$\nu = \omega/\omega_o = \frac{f}{f_o}$, the frequency ratio

$$N = \frac{\mu_2}{2\mu_2 + \frac{r_{p2}}{R}},$$

and

$$M = \frac{(1+\mu_1+r_{p1}/R)(2+\mu_2+\frac{r_{p2}}{R}+\frac{r_{p2}}{R_L})}{(2+\mu_1+\frac{r_{p1}}{R})(2+\frac{r_{p2}}{R_L})+2\mu_2+\frac{r_{p2}}{R_L}} \tag{13}$$

Then the ratio of the voltage gains becomes

$$K_h/K_L = (1+jN\nu)/(1+jM\nu) \tag{14}$$

The magnitude of this gain ratio can be expressed in terms of decibels gain as

$$20 \log_{10} K_h/K_L = 20 \log_{10} \sqrt{1 + N^2 \nu^2} - 20 \log_{10} \sqrt{1 + M^2 \nu^2} \tag{15}$$

and the phase angle is $\phi = \tan^{-1} N\nu - \tan^{-1} M\nu$ (16)

The factors M and N in the previous equations together with the factors A and β mentioned in Sections I and II set up a practical limitation for the choice of the circuit and tube parameters. To make the factor A reasonably small, but still seeking a sizable power output with adequate negative feedback, one would choose a large μ_1, while μ_2 and r_{p2} are kept reasonably small. Again, let $R \cong r_{p1}$ and $r_{p1} > r_{p2}$, and for a matched resistive load, Equation (13) becomes

$$N \cong \frac{\mu_2}{2\mu_2} = \frac{1}{2}$$

$$M \cong \frac{(2+\mu_1)\left[4 + \mu_2\left(1 + \frac{2}{3+\mu_1}\right)\right]}{4(3 + \mu_1 + \mu_2)} \tag{17}$$

It is then permissible to assign different values of μ_1 and μ_2 and calculate the range of M that is of practical interest. Fig. 5 is the plot of these calculations. It is noticed that within practical considerations r_{p2} and μ_2 should not be very large to give a small A, hence it leaves little choice for the M value if μ_1 is to be as large as permissible.

The choice of M and N affects the frequency response of the amplifier. This can be seen from a calculation of the gain-frequency characteristic computed for different assumed values of M and N. Let M = 1.5, N = 0.5, Equations (15) and (16) are computed and plotted as the solid-line curve in Fig. 6. The three discontinuous dotted straight-line segments are the asymtotical plots of the gain characteristic with a first break at $\nu_1 = 0.667$ and a second break at $\nu_2 = 2.0$. If f_0 is chosen as one million cps, then from Fig. 6, the amplifier shows a 2.55 decibel loss at 667 kc and a 7 decibel loss at 2mc. At much higher frequencies, the loss will be 9.54 db. Increasing the value of M moves the first break point toward the left. Thus for M = 2, N = 0.5, the break points will be at $\nu_1 = 0.5$ and $\nu_2 = 2.0$ (dot-dash line) respectively. The amplifier will then have a loss of 2.75 db at 500 kc and 12.04 db at higher frequencies. Thus the flat portion of the gain-frequency curve is reduced. It can also be shown that reducing the value of N moves the second break point and the phase shift curve toward the right. The difference in gain between lower and higher frequencies also increases

Experimental Results

Fig. 4 is the actual circuit used in the experiment. Separately adjustable biases for the power tubes were provided. With a pair of 616GA's as output tubes and one section of a 6SN7 as driver, maximum power output of 28 watts was obtained. For this operation, 400 volts were used for plate supply and -30 volts for the grid biases. The plate circuit efficiency thus obtained was 31%. In another operation, using -35 volts as bias, the plate efficiency increased to 35.3% with a power output of 22.7 watts. Frequency response in each case was flat to within two decibels from 30 cycles per second to 200,000 kilocycles per second. A Western Electric output transformer 134C was used in the output system for impedance matching purposes.

Several experimental results are worth mentioning. Fig. 7 is a plot of the power output against load resistances for two different operating voltages. Signal input in both cases is maintained constant to a level such that practically no distortion is observed in a cathode-ray oscilloscope over the entire range of the load variation. The curves show well defined maximum at certain resistances which coincide with the values computed from one-fourth of the plate-to-plate resistance for a conventional push-pull circuit. Fig. 8 shows the variation of the driving voltage across the grid-to-cathode of the output tubes for two different operating conditions. In both cases, the driving voltages decrease as the load resistance is increased. The drooping driving voltage characteristic is the main cause for the drooping power output curve in Fig. 7 at larger load resistances. Larger output power can be obtained at these larger load resistances by increasing the input signal.

A curve for the output voltage is also plotted along with the driving voltage. It increases with increasing load resistance. The sum of this voltage and two times the grid driving voltages constitute the total a.c. voltage available across the driver tube. With increasing load resistance, the re-division of the total a.c. voltage among the effective resistance (the output voltage) and the resistances R (the driving voltages) is the cause of these variations. As the aim of this circuit design is to obtain large output power, the condition for maintaining constant driving voltage (small A) mentioned in our theory is not used.

Acknowledgement

The author wishes to express his appreciation for the valuable discussions carried out during the course of this investigation with Professor E. L. Chaffee of Harvard University and Dr. D. B. Sinclair and Dr. A. Peterson of General Radio Company, Cambridge, Massachusetts.

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1. A. Peterson and D. B. Sinclair, "A Single-Ended Push-Pull Audio Amplifier," Proc. IRE, Vol.40, pp.7-11; January, 1952.

2. See Peterson and Sinclair, loc.cit.

3. A. P. T. Sah "Quasi-Transients in Class-B Audio-Frequency Push-Pull Amplifiers," Proc. IRE, Vol.24, pp.1522-1541; November, 1936.

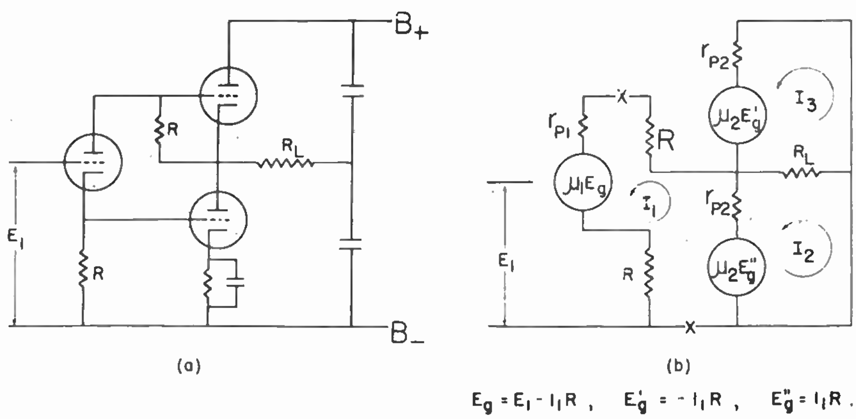

Fig. 1 (a) The basic circuit and (b) the equivalent circuit of a single-ended push-pull audio amplifier.

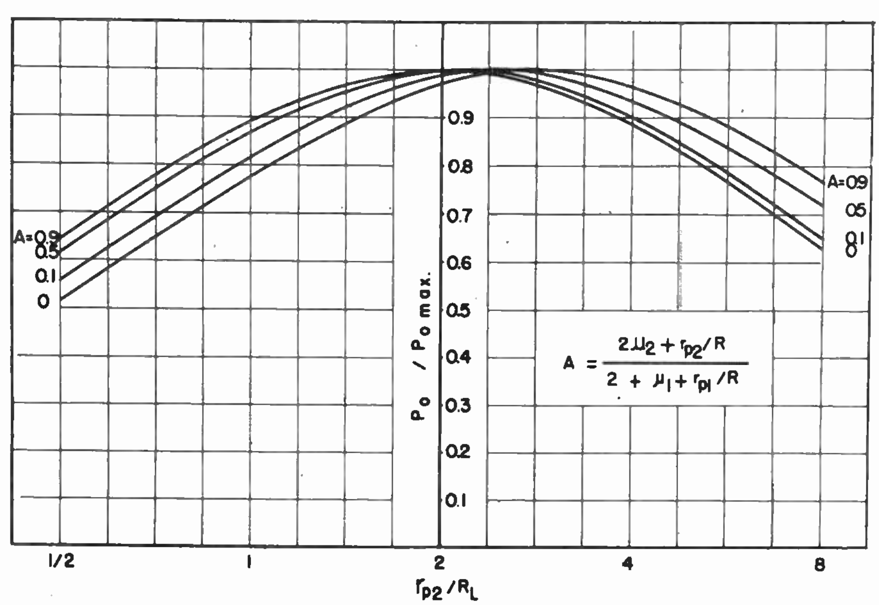

Fig. 2 Power sensitivity as a function of load resistance.

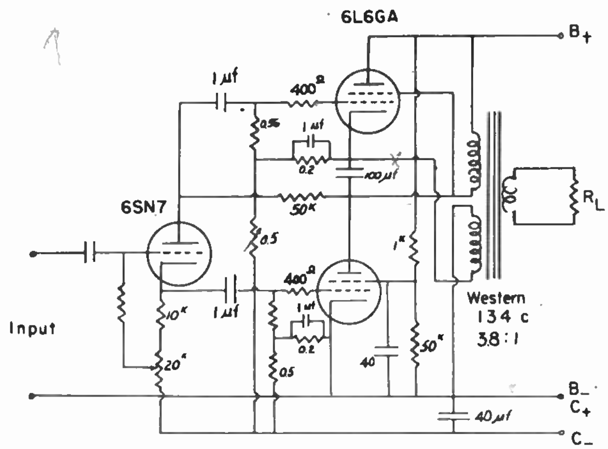

Fig. 3

Method of using an impedance matching transformer to put the output tubes in parallel across the dc plate supply taken from the paper by Peterson and Sinclair with modification to adjust the grid bias of the output stages independently.

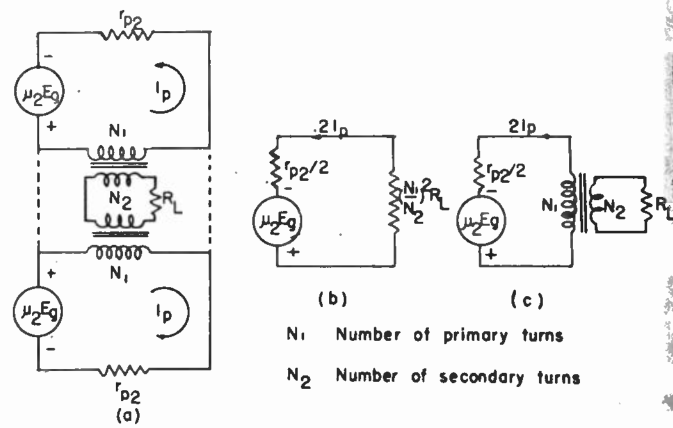

Fig. 4

Equivalent circuits of the output stage of a single-ended push-pull amplifier with impedance matching output transformer used to connect the output tubes in parallel across the dc plate supply.

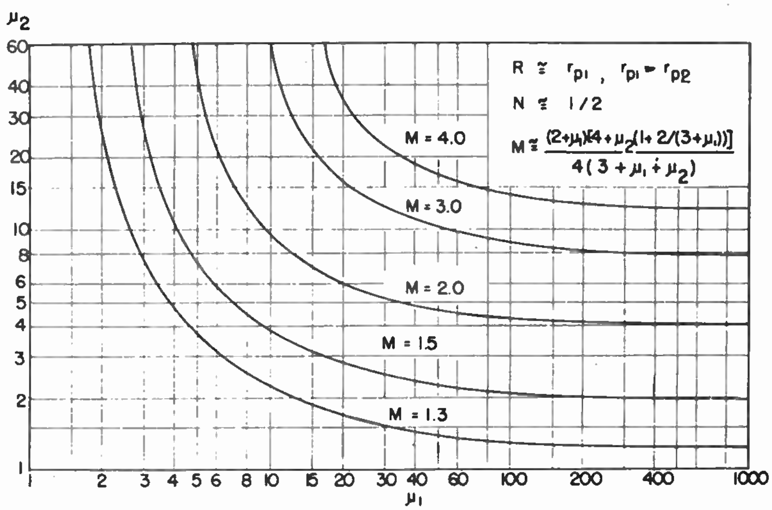

Fig. 5

Range of tube parameters for practical considerations.

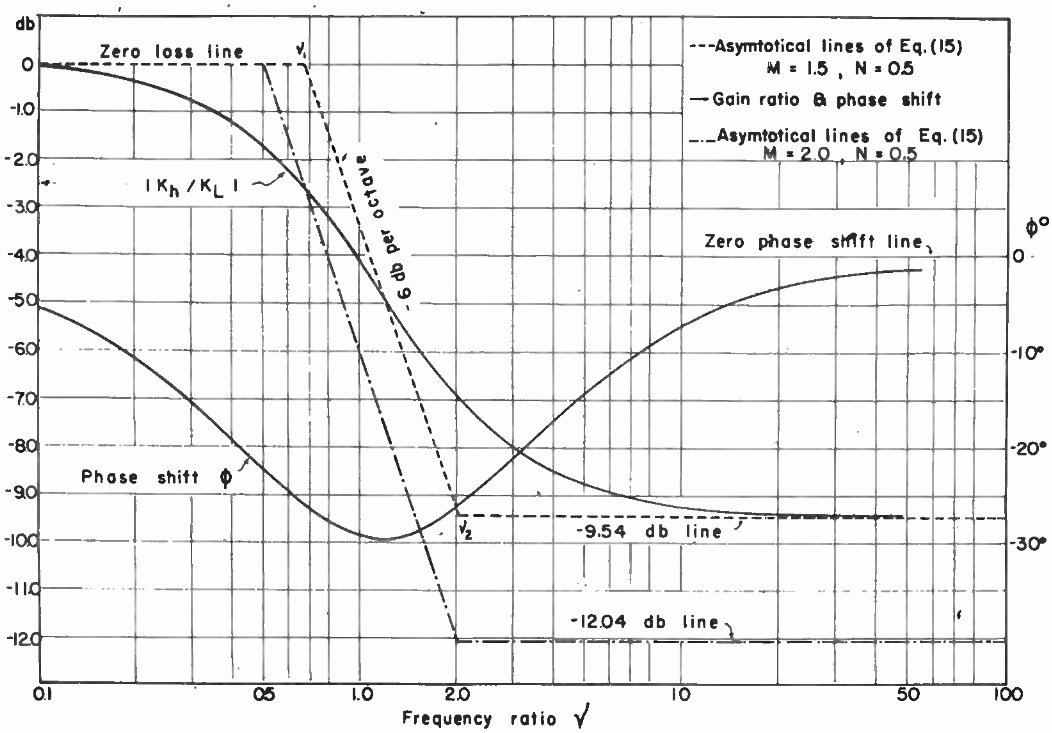

Fig. 6

Magnitude and phase of the gain ratio as a function of the frequency ratio.

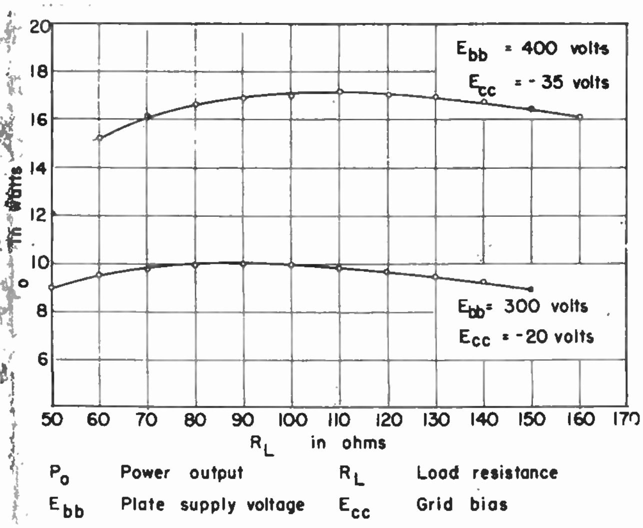

Fig. 7
Power output - load resistance curves for constant signal input.

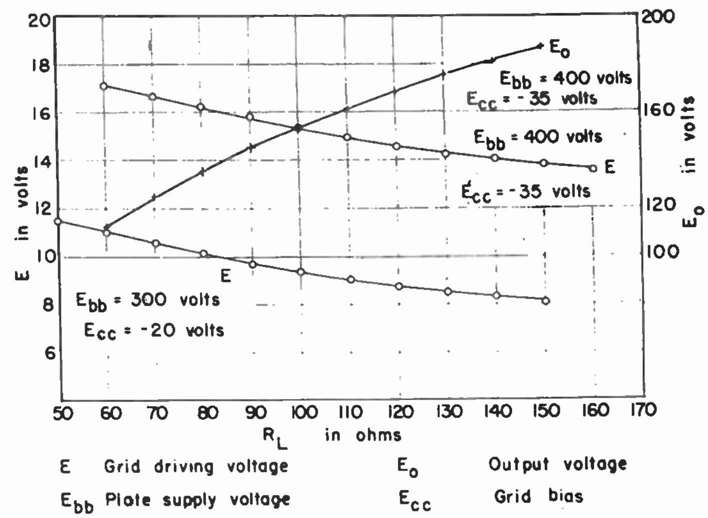

Fig. 8
Driving voltage and output voltage - load resistance.

EQUALIZATION OF MAGNETIC TAPE RECORDERS FOR AUDIO AND INSTRUMENTATION APPLICATIONS*

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As a starting point for discussion, let us consider equalization requirements in terms of an "Ideal System" without losses. An "Ideal System" would consist of the following. A constant current amplifier connected to a record head; the record head producing a constant flux vs. frequency on the tape, and a playback head picking up the signals from the tape and feeding an amplifier whose amplification drops off at the rate of 6 Db. per octave across the band. The 6 Db. per octabe playback amplifier is required because the output from the ideal playback head is directly proportional to frequency when reproducing a constant flux vs. frequency signal. Unfortunately, a number of factors are present which make a practical system differ considerably from this ideal.

First, there are several possible losses associated with the record and reproduce heads. The electrical losses are produced by eddy current and hysteresis effects in the core and capacity effects in the windings. In a well designed and constructed head, these effects are not a problem in the audio range, but do enter the picture somewhat in instrumentation recorders operating in the 100 Kc. region.

The high frequency loss due to the length of the playback gap usually determines the upper frequency limit of the recorder. When the effective length of this gap equals the reproduced wave length, a null or cancellation will occur. The loss in decibels at other wave lengths on the tape is equal to

$$20 \text{ Log } \frac{\sin \pi \, \theta/\lambda}{\pi \, \theta/\lambda}$$

where θ is the effective playback gap length and λ is the reproduced wave length. From this expression a theoretical loss of 4 Db. will occur when a wave length of .0005" is reproduced with a head incorporating a gap length of .00025". This would be the case at 15,000 cycles and a tape speed of 7½ inches per second. The writer has found that reproduce heads with this gap length can be built with surprising uniformity, making reproduction down to .0005" wave length achievable in production equipment.

Heads incorporating larger gaps, although desirable from the standpoint of output, are seriously limited in high frequency range at the lower tape speeds.

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The record head gap length does not bear the same relation to the frequency characteristic as that of the playback head. The magnetization left on the tape is determined primarily by the trailing edge of the gap and not by its length. The sharpness and, of course, straightness of this edge are paramount. The record head gap length is important from other aspects, however. An excessively small record gap would produce a decaying field through the oxide layer on the tape. This decaying field in turn would increase low frequency distortion since the oxide layer in direct contact with the gap would have to be recorded at a relatively high level to compensate for the only partially magnetized undercoat of oxide. The wider the record gap, the greater the demagnetization effect on the highest recorded frequencies. Therefore, excessively wide gaps are to be avoided. .001" to .002" has been found to be the most acceptable range. The so called demagnetization effect is created by partial erasure of a high frequency recorded signal at the time of recording by the high frequency bias field.

The remaining losses in a practical tape system are associated with the tape itself. The oxide particle size and uniformity of dispersion are large factors in the high frequency characteristic of a tape. Large particles or poor dispersion will result in reduced area of contact with the head gap. This reduced area of contact has little effect while recording and reproducing long wave lengths, but will drastically effect high frequency performance. Paper base tapes are unsatisfactory in this respect as the roughness of the paper, and subsequently the oxide coating, prevent proper contact with the head surface. Coating thickness is another factor which has an indirect bearing on the frequency characteristic. As the coating thickness is decreased, the bias requirement will decrease. Consequently, high frequency demagnetization will be reduced and high frequency record efficiency will increase. Unfortunately, low frequency distortion will increase with decreased coating thickness so reduction of coating thickness beyond a certain point is impractical.

Another factor, of course, is the characteristics of the oxide itself. Lower demagnetization losses can be accomplished on some of the new tapes which incorporate oxides selected for low bias requirements and high output. This higher performance oxide permits less coating thickness for the same low frequency output level as compared with previous professional tapes.

Because of variation in performance which is possible due to the tape, it is best to equalize a recorder with a sample of tape known to be a "centerline" of the tape manufacturers' tolerances and to use only tape of that manufacture of a known equivalent.

Let us now consider the problems of equalizing the audio recorder. The equalization of audio recorders at a given tape speed requires a study of the noise and distortion characteristics of the system. A good approach to the problem is to evaluate the system with the tape equalization arbitrarily divided in record and playback. The established energy distribution curves for speech and music serve as a rough guide as to maximum permissible record equalization at any frequency. A noise spectrum analysis will insure that the noise is fairly distributed over the pass band. If this is not the case, the playback equalization should be adjusted until fairly even distribution of noise exists. The record equalization can then be altered to compliment the playback characteristic. These measurements can be weighed on the basis of ear sensitivity, and a better noise characteristic thereby obtained if discretion is used in the amount of correction applied. Such changes in the equalization, based on ear sensitivity, should be carefully checked by listening tests on wide range equipment.

The record system must now be studied from the standpoint of distortion and overload at all frequencies in the pass band. If the record equalization is not greater than the amount required to compliment the energy distribution in speech or music, the overall distortion characteristic will be found satisfactory. If the equalization requirements are greater than that which can be tolerated on this basis, three possibilities exist. The first would be to lower the record level and thereby compromise the signal-to-noise ratio. The second would be to chance the system running into overload distortion at frequencies excessively pre-emphasized. The third would be to lower the record equalization to acceptable limits and raise the post equalization at the expense of signal-to-noise ratio in the raised spectrum. At the professional primary and secondary speeds of 15 and 7½ inches per second, these compromises are unnecessary for full range recording from 30 to 15,000 cycles. The full dynamic range of the tape is therefore available.

The record curves indicated in Fig. 1 and the playback curves indicated in Fig. 2 were established in conjunction with Ampex heads and M.M.M. type 111 tape, construction 5 RBA. These heads display negligable magnetic and electrical losses in the pass band. The playback head gap length is .00025". The bias was adjusted to the point of maximum record efficiency while recording a .015" wave length (1 Kc. at 15 inches per second). The overall response achievable under these conditions is as follows:

At	30"/Sec.	±2Db, 50 to 15,000 cycles
	15"/Sec.	±2Db, 30 to 15,000 cycles
	7½"/Sec.	±2Db, 40 to 15,000 cycles
	3¾"/Sec.	±2Db, 40 to 7,500 cycles

The playback curves are easily accomplished by connecting a vacuum tube operating as a constant current generator to a capacitive load whose reactance equals the generator impedance at 65 cycles.

The 30 inch curve with the exception of the low frequency departure is the characteristic required to compensate the "Ideal System." The slight low frequency departure from the ideal curve was found desirable for the elimination of low frequency thermal effects in playback amplifier input tubes operating at low levels. This departure is made up for by a slight rise in the low frequency playback head characteristic brought about by its physical dimension and by 2½ Db. boost in the record amplifier at 50 cycles. A resistor of such value to effect a time constant of 50 microseconds has been placed in series with the 6 Db. per octave condenser to produce the desired high frequency characteristic at 15 and 7½ inches. The 3-3/4 inch curve is accomplished by a relatively larger resistor effecting a time constant of 200 microseconds. The 15" per second record curve is such that possibility of overload does not exist for the most sever audio requirements. At 7½" per second the record curve is considerably steeper than the 15" curve and reaches 17 Db. at 10 Kc. Listening tests conducted with material recorded on equipment adjusted to this characteristic have shown it to be entirely satisfactory for high fidelity recording. This is the case because of the energy distribution encountered in normal speech and music, and because of a characteristic, of the tape, to compress the high frequency, high intensity peaks occasionally encountered, without appreciable distortion. Sound already pre-emphasized for special effects or from highly resonant microphones might present overload problems at 7½" which, of course, would not occur at the 15" speed. The overall response of a typical Ampex 300 or 403 recorder can be adjusted to +1 Db. from 50 cycles to 15 Kc. at both 7½" and 15" speeds. Slightly wider specifications are advertised to allow manufacturing tolerance and insure the average machine being well within its specifications.

The background noise on a high quality professional recorder, using the tape characteristic described, ranges from 60 to 64 Db. below 3% harmonic distortion. This is true at 7½", 15", and 30" per second. The noise at 3-3/4" per second is approximately 10 Db. higher. The point of approximately 1% harmonic distortion has been found most desirable for operating level and is approximately 6 Db. below the 3% distortion point.

Instrumentation recorders fall into two general categories as pertaining to the subject under discussion.

Pulse systems and carrier systems are in the first category. These systems do not require equalization for the tape system. The second category contains the conventional magnetic recorders employing high frequency bias and recording a band width within the range of 100 cycles to 100 Kc. These recorders incorporate similar electronic systems to audio recorders except for the distribution of equalization. The intelligence recorded on such instruments is usually of a nature that the energy level is fairly uniform over the pass band. This requires a record characteristic with a uniform overload and saturation characteristic in respect to frequency. An unequalized constant current amplifier driving the record head and producing essentially a constant flux recording best suits this requirement. Equalization required for flat overall response is therefore placed in the playback amplifier.

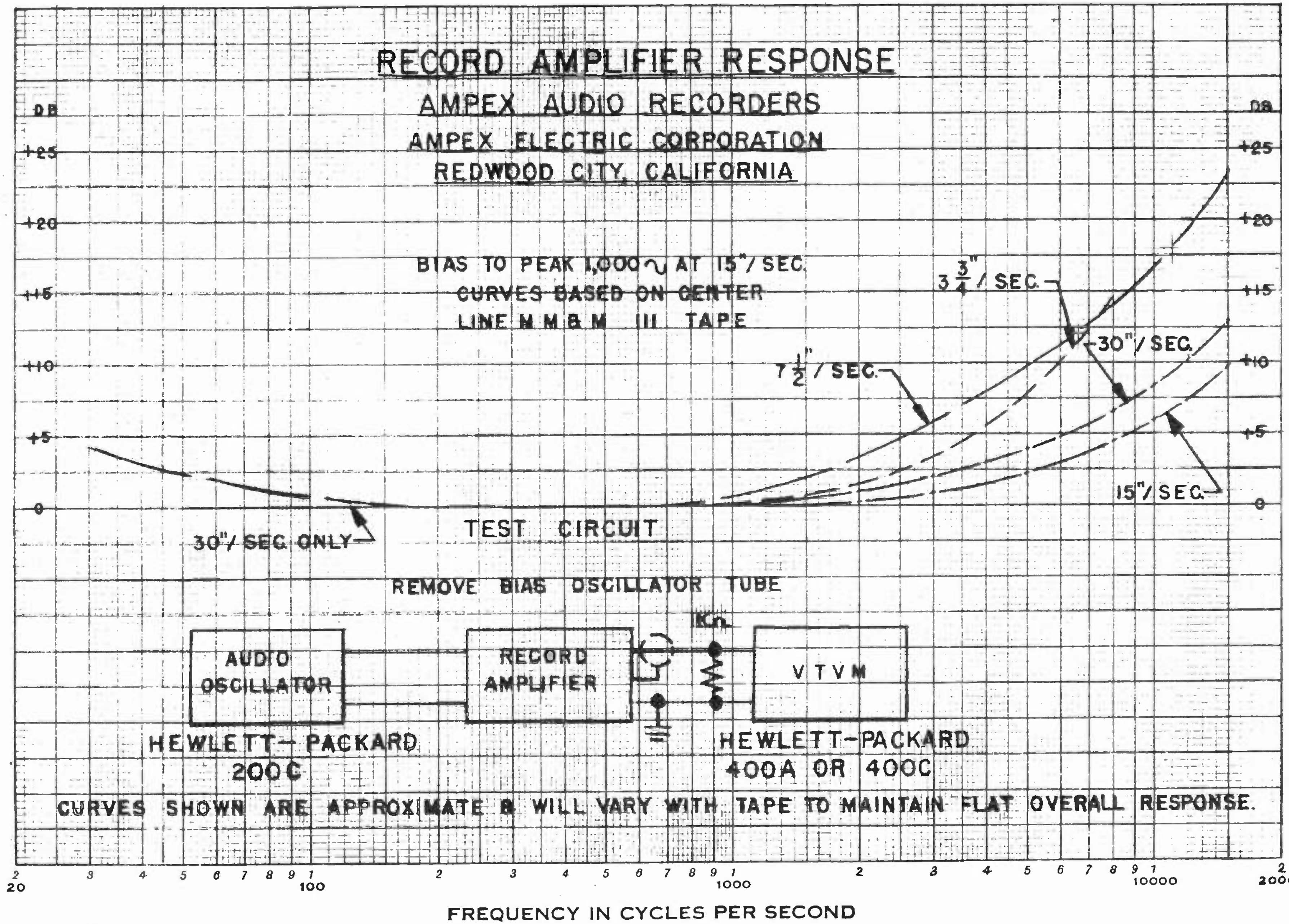
RECORD AMPLIFIER RESPONSE
AMPEX AUDIO RECORDERS
AMPEX ELECTRIC CORPORATION
REDWOOD CITY, CALIFORNIA
BIAS TO PEAK 1,000 ∿ AT 15"/SEC.
CURVES BASED ON CENTER
LINE M M & M 111 TAPE
DB
+25
+20
+15
+10
+5
0
3 3/4"/SEC.
30"/SEC.
7 1/2"/SEC.
15"/SEC.
30"/SEC. ONLY
TEST CIRCUIT
REMOVE BIAS OSCILLATOR TUBE
AUDIO OSCILLATOR
RECORD AMPLIFIER
1KΩ
V T V M
HEWLETT-PACKARD 200C
HEWLETT-PACKARD 400A OR 400C
CURVES SHOWN ARE APPROXIMATE & WILL VARY WITH TAPE TO MAINTAIN FLAT OVERALL RESPONSE.
20
100
1000
10000
20000
FREQUENCY IN CYCLES PER SECOND

Fig. 1

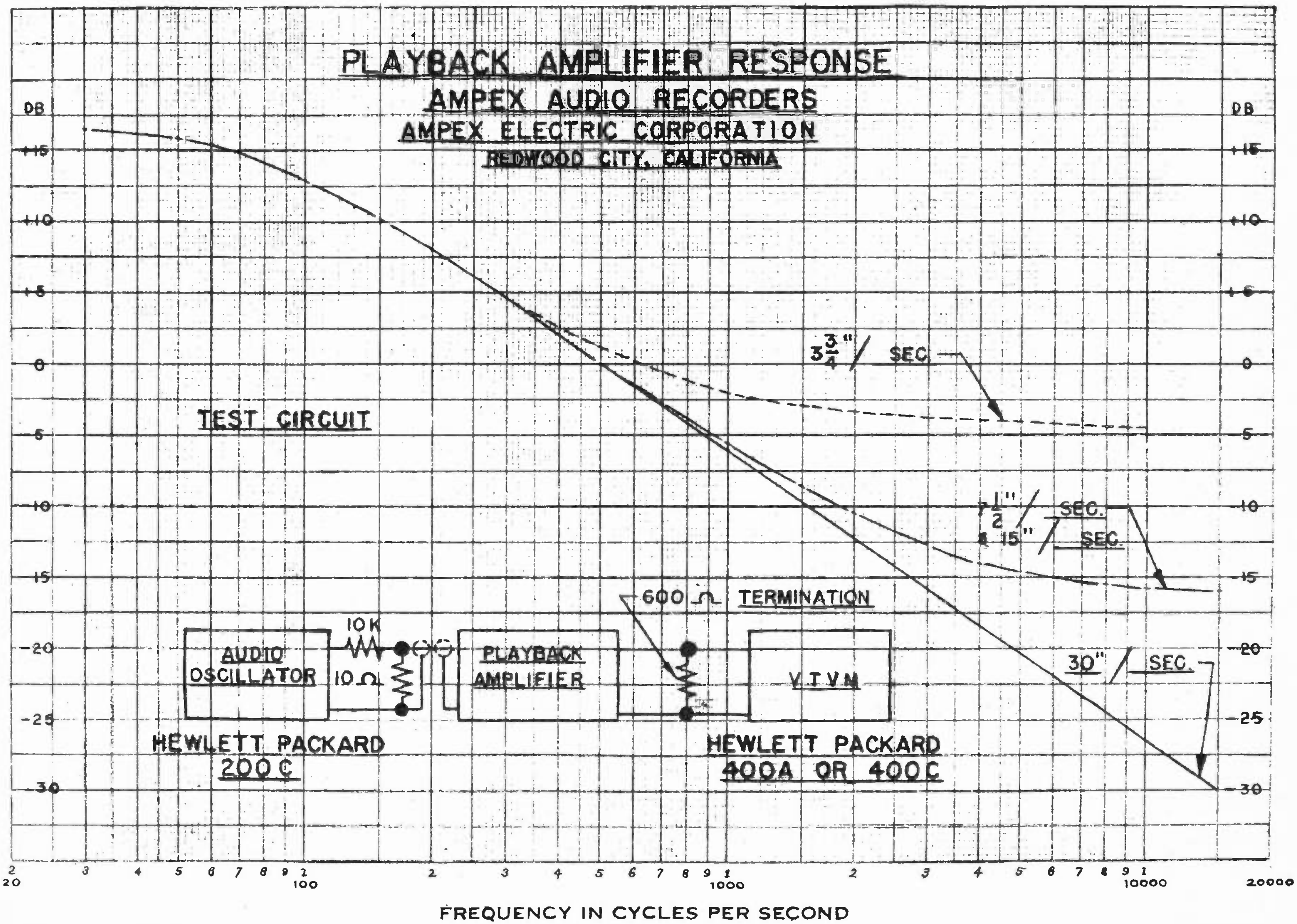
PLAYBACK AMPLIFIER RESPONSE
AMPEX AUDIO RECORDERS
AMPEX ELECTRIC CORPORATION
REDWOOD CITY, CALIFORNIA
DB
+15
+10
+5
0
-5
-10
-15
-20
-25
-30
3 3/4" / SEC.
7 1/2" / SEC.
& 15" / SEC.
30" / SEC.
TEST CIRCUIT
600 Ω TERMINATION
10 K
10 Ω
AUDIO OSCILLATOR
PLAYBACK AMPLIFIER
V T V M
HEWLETT PACKARD 200 C
HEWLETT PACKARD 400A OR 400C
20
100
1000
10000
20000
FREQUENCY IN CYCLES PER SECOND

Fig. 2

WHY FIGHT GRID CURRENT IN CLASS B MODULATORS?*

J. L. Hollis
Collins Radio Company
Cedar Rapids, Iowa

ABSTRACT

Grid voltage waveform distortion ranks with output transformer inadequacies among Class B audio amplifiers deficiencies. This article deals with the problem of grid voltage waveform distortion and suggests the use of low plate resistance low mu triode tubes as a solution. By avoiding operating conditions which result in grid current, the major source of distortion is eliminated. Direct coupled driving circuits which encourage large amounts of inverse feedback are readily employed under such conditions. A typical amplifier and its measured characteristics is used for illustration.

1. Why Is This Subject Important?

Class B operation of push-pull amplifiers is one of the best ways known today of obtaining high power with reasonable efficiency in the audio frequency spectrum. In order to keep the harmonic distortion to a tolerably low value, careful attention must be paid to the design of the output transformer and the grid circuit details. Proper transformer design has been expounded by several authors. Grid circuit design, on the other hand, has been more or less by rule of thumb.

2. Why Is Grid Current an Important Cause of Distortion?

Figure 1 illustrates the grid voltage plate voltage relationship in Class B operated vacuum tubes. It will be noted that the two lefthand curves are for a medium mu triode. The extreme left curve shows typical operation with intermediate values of grid voltage. The maximum value of grid voltage is less than the zero bias voltage so that the grid is always in the negative grid region. Plate voltage swing is nominal, being considerably less than the maximum capability of the tube. This operation naturally will be very inefficient because the minimum value of plate voltage is always rather large. The middle curve, on the other hand, shows a quite acceptable plate voltage swing which brings the efficiency to reasonable values but requires that the grid be driven into the positive region to do so. Please note that a small curve is shown which is designated grid current. Grid current

*Presented at the Audio Session of the IRE Convention in Long Beach, California on August 29, 1952. Manuscript received January 23, 1953.

starts flowing at the instant the grid voltage passes the zero line and moves into the positive grid region. It is during this interval when grid current flows that the driver is called upon to deliver power. If the driver voltage regulation is anything but perfect, the voltage will drop below what it should be during this grid current interval. The dotted curve shows a possible modification of the voltage waveform as a result of this grid current. It should be noted that this also affects the plate voltage swing. These points are the distortion which is common to this form of operation. The only way to operate under these circumstances without excessive distortion is to make the driver's impedance so low that its voltage regulation is essentially perfect. This, of course, is very wasteful of driver power and tends to offset the improved amplifier efficiency.

The curve on the right, which is designated as a low mu triode, illustrates a different approach. Here a tube is selected that is capable of efficient operation without the necessity of operating in the positive grid region. Grid voltage is shown always in the negative region. Even so, the plate voltage swing reaches a very satisfactory minimum value, which thus insures reasonably good efficiency, during the negative half of the cycle. Since it is never necessary for the grid to be driven positive, no grid current flows and therefore no power is required from the driver. The driver in this case is simply a voltage amplifier. It is true that the driver voltage required may be considerably higher than that required by a medium mu triode. This voltage swing is relatively easy to obtain since no power is required. It is only when a combination of voltage swing and power output are required that driver design becomes a serious problem.

3. What Are the Characteristics of an Ideal Audio Power Amplifier?

A power amplifier which approaches the ideal is one in which the grid to cathode impedance remains constant throughout an entire cycle of driving voltage. This might be some finite value in which case there would be a constant proportionality between the driving voltage and grid current, or it might be infinite. Unfortunately, there are no suitable vacuum tubes which present a constant finite input impedance. Therefore, we must choose a tube with infinite input impedance. This means that the grid must never be driven positive with respect to the cathode so that grid current never flows. Such a tube obviously never dissipates any power from the driving source and thus has a second advantage. This we shall set aside as characteristic --

a. No driving power required

In order that reasonable efficiency may be realized, it is important that the minimum plate voltage be reasonably low at the peak of the cycle. Since this peak occurs when the grid is at its maximum swing in the positive direction which is identically the zero grid voltage line, it is important that the tube have low plate resistance at this instant. Therefore, a second characteristic of a good audio power amplifier is --

b. Low plate resistance

A low plate resistance tends to make a constant voltage generator out of the amplifier which, for modulator service where the load is determined by the Class C amplifier loading, is a distinct advantage.

4. What Are the Particular Requirements of Tubes for This Service?

In a triode tube the relationship between the important tube parameters is expressed as mu = $G_M R_p$. It is apparent from this that to have low plate resistance the tube must have a high transconductance and a low mu factor. Generally speaking, then tubes which best meet this requirement will be low mu.

A few examples will illustrate the points just made.

Figure 2 is a typical plate current plate voltage characteristic of a triode of medium or high mu. It will be noted that the zero bias line has a rather flat slope. A typical load line has been superimposed to show that driving the grid only to the zero bias is a very inefficient use of the tube. The load line must be extended (dotted portion) into the positive grid region to realize any efficiency.

Figure 3 is a typical plate voltage plate current characteristic of a low mu triode. It will be noted here that the superimposed load line represents relatively low minimum plate voltage values when it intersects the zero bias line. The extra dotted lines illustrate a range of suitable load impedance values. It is to be noted that the plate voltage swing changes very little as the load impedance is changed.

Figure 4 is a typical tetrode characteristic curve. Here it is apparent that by careful choice of the load impedance very high plate circuit efficiency is obtained. It will be noted, however, that small changes in plate load impedance cause serious changes in plate swing. In many cases this is a serious disadvantage. What is not shown in these curves are the screen current variations. Screen current flows in pulses quite like grid current in positively driven grid circuits. Its peak value depends considerably on the minimum plate voltage during a cycle. Extremely good regulation of the screen voltage supply is therefore a must for proper operation of tetrodes in Class B amplifiers.

5. What Circuit Advantages Does Class B_1 Operation Provide?

(Class B_1 operation is defined as an operating condition where the plate current flows for one-half cycle, while grid current never flows for even a fraction of a cycle. The subscript 1 indicates this grid current limitation. A subscript 2 refers to operation wherein grid current flows during at least part of a cycle.)

Since there is no grid current to contend with, the coupling circuits between the driving amplifier and a Class B_1 modulator can be simple and of the type common to ordinary voltage amplifiers. Resistance capacitance circuits can replace the usual Class B driving transformer. Figure 5 is a circuit which illustrates this fact.

Figure 6 is a modification of the schematic shown in Figure 5 wherein a direct coupled circuit is substituted between the driver and the power amplifier. An input stage and a feedback circuit are also shown. This arrangement has very marked advantages where inverse feedback is to be employed. It will be noted that there is only one capacitor in the entire feedback loop which is almost a certain guarantee of proper low frequency phase versus feedback loop response characteristics. There can be no low frequency motor boating with this circuit, no matter how much inverse feedback is used. It will be noted also that the high frequency feedback loop response can be readily controlled by the strategic choice of plate load resistance values and rather simple response shaping circuits.

It will be noted that the driver is operated with its cathode negative to ground. This allows the voltage drop in the driver load resistor to be used directly as the bias and signal voltage for the modulator. Although a triode driver is shown, a tetrode may be used to considerable advantage to obtain higher voltage gain. The screen connection is shown dotted.

Typical characteristics of a modulator using the basic circuit shown in Figure 6, but with three 8C25 tubes in parallel on each side of the push-pull circuit (a total of six in all) are as follows:

<u>Tubes</u>

First Stage -- 12AX7	Plate Voltage -- 5 KV
Second Stage -- Push-pull 4-250A	Input Level -- +10 DBM
Third Stage -- (6) - 8C25	Inverse Feedback -- 20 DB
Power Output -- 25 KW	

Figure 7 shows the inverse feedback loop response for a 50 to 15,000 cycle passband. Both ideal and actual response curves are shown for stable operation with 26 DB of feedback and actual operation with 20 DB.

Figure 8 shows a typical response curve taken from the rectified RF of a 35 KW transmitter which was plate modulated by the modulator described. The high frequency cut-off shown is produced by a low pass filter following the modulator unit.

Typical values of distortion measured from the same rectified RF source are shown in this same figure. It will be noted that from 100 to 3,000 cycles the distortion is less than 1%, while between 60 and 15,000 cycles it is well below 2%.

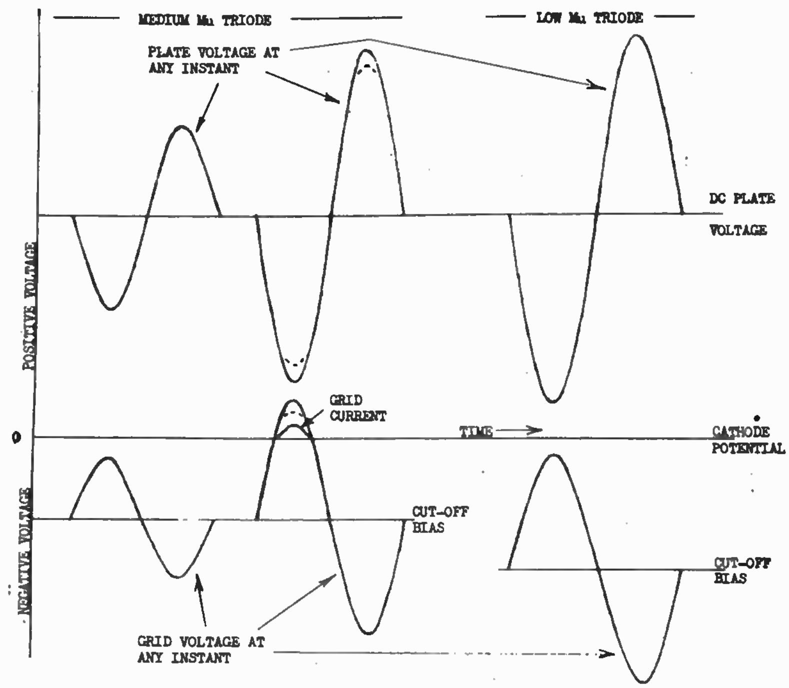

Fig. 1

Typical instantaneous plate and grid voltage relationships for Class B operated tubes. Left and middle curves are for medium mu, while right curve is for low mu triodes.

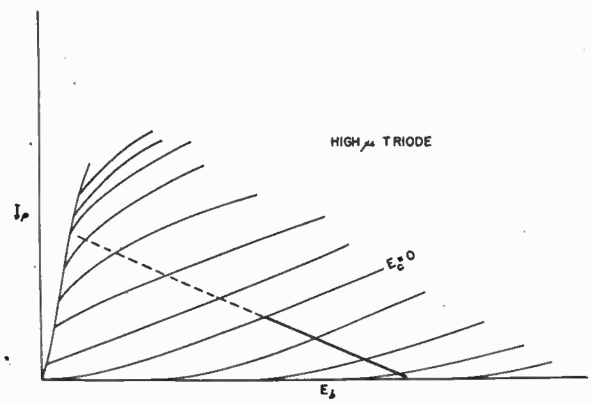

Fig. 2 Typical plate characteristic curves for a high mu triode with a possible load line superimposed.

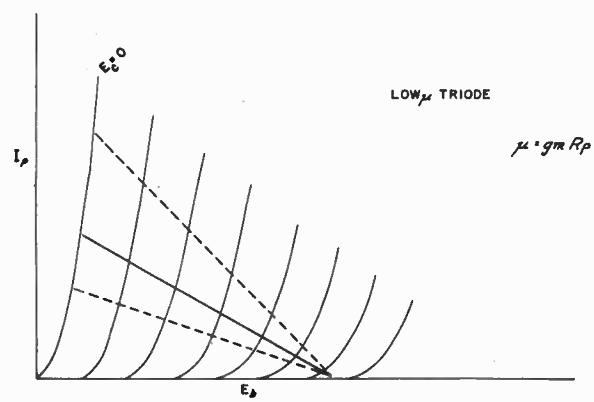

Fig. 3 Typical plate characteristic curves for a low mu triode with several possible load lines superimposed.

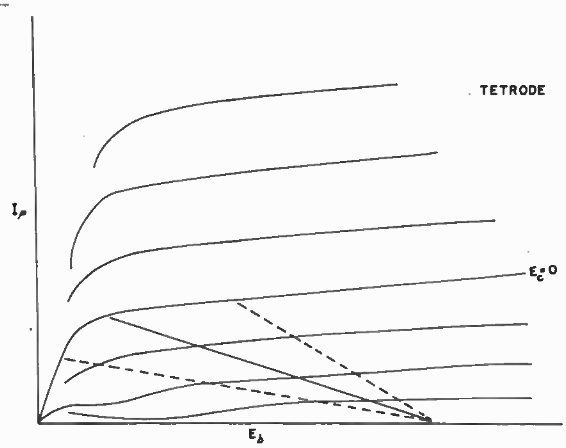

Fig. 4

Typical plate characteristic curve for a tetrode tube with several possible load lines superimposed.

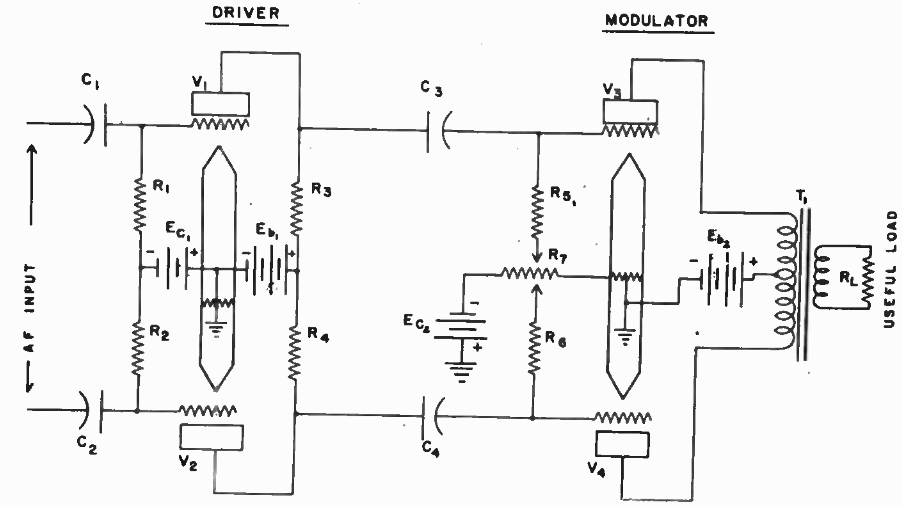

Fig. 5

Convential driver modulator arrangement for Class B_1 operation.

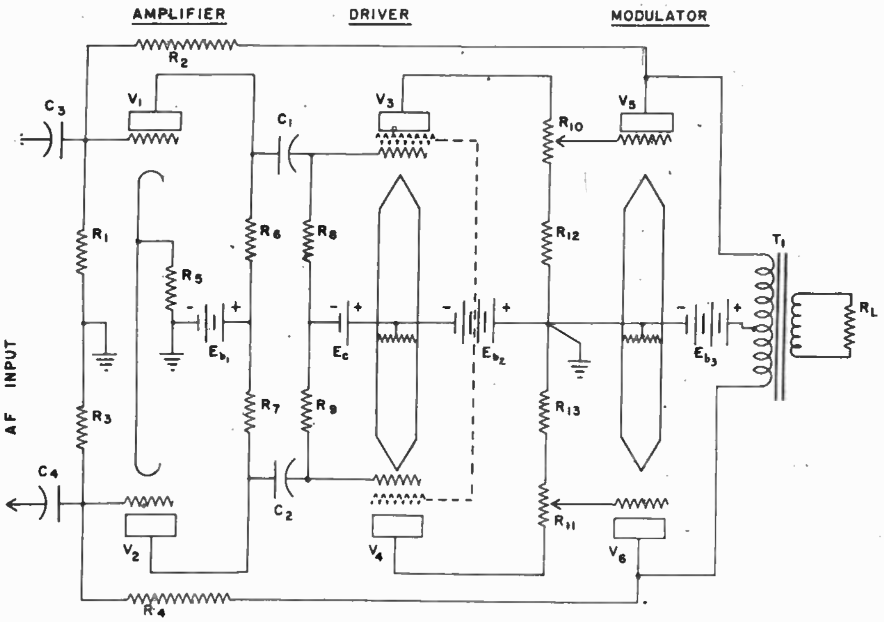

Fig. 6

Direct coupled driver modulator circuit arrangement with improved performance.

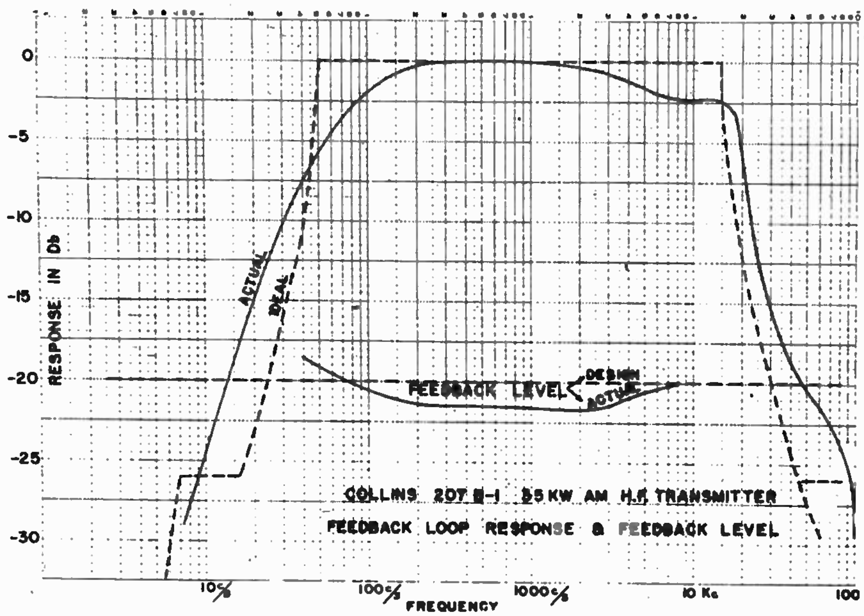

Fig. 7

Typical inverse feedback loop response showing the ideal curve for a 50 to 15,000 cycle passband and a solid curve depicting the curve actually attained.

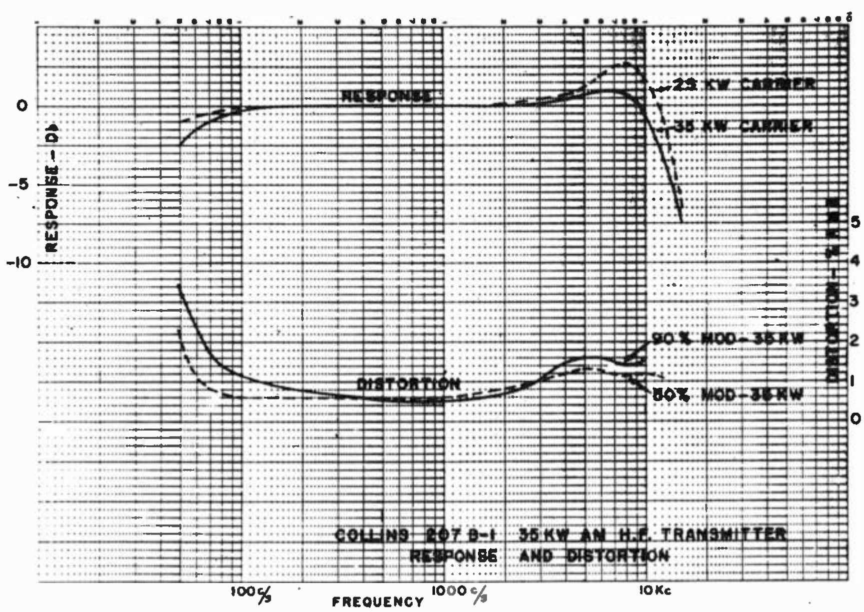

Fig. 8

Typical response and distortion data measured at the output of a 35 KW AM transmitter employing Class B_1 modulators in the direct coupled circuit described in this paper.

MICROPHONE SENSITIVITY CONVERSION CHART

Leo Rosenman
Shure Brothers, Inc., Chicago 10, Illinois

In the current technical literature, microphone sensitivities are often specified in different systems. This makes comparison rather difficult. The accompanying nomogram gives the relationship between three systems of ratings most commonly used. They are:

1. Open Circuit Voltage Response

   The equation for open circuit voltage response is

   $$S = 20 \log_{10} \frac{E}{p} \text{ db}$$

   where E is the open circuit voltage, and p is the free-field sound pressure; expressed in db relative to 1 volt per microbar (1 dyne/cm^2).

2. Open Circuit Power Response

   The equation for available power response is

   $$S_p = S_v - 10 \log_{10} R + 44 \text{ db}$$

   where R is equal to the nominal impedance at some specified frequency, usually 1000 cps. This equation gives the available power sensitivity in db relative to 1 milliwatt for a 10 microbar sound pressure.

3. RTMA Sensitivity Rating

   The RTMA microphone rating G_M is defined as

   $$G_M = S_v - 10 \log_{10} R_{MR} - 50 \text{ db}$$

   where R_{MR} is defined as the RTMA "microphone rating impedance".

For a given value of nominal impedance, R_{MR} is indicated by arrows in the center of the range of impedance values covered by the triangular sections on the nomogram. For nominal impedances above 100,000 ohms, the rating impedance of 100,000 ohms is used. Microphone rating G_M is not defined for nominal impedances below 19 ohms. The value of G_M obtained from the above expression is essentially the available power in db relative to 1 milliwatt for 0.0002 dynes/cm^2.

The conversion of one rating to another is a simple but tedious problem of numerical computation and can be facilitated by the use of the accompanying nomographic chart. To illustrate the use of the nomogram, suppose we have a microphone with an open circuit voltage sensitivity of -60 db re 1 volt per microbar, and with a nominal impedance of 15,000 ohms. The solid line indicates that the power sensitivity is -58 db re 1 milliwatt for 10 microbars. To find G_M we first note that 15,000 ohms lies between 4,800 and 20,000 ohms. The microphone rating impedance is therefore 9,600 ohms and the dashed line indicates that G_M is -150 db. As a general rule R_{MR} for high impedance crystal microphones is 100,000 ohms.

The nomogram can also be used to determine the open circuit voltage sensitivity of microphones undergoing impedance transformation. First determine the power sensitivity for the original impedance and then pivot about this point until aligned with the new impedance. The new open circuit voltage may then be read on the left hand stem. This will be in error by the loss introduced by the transformer, which is usually small.

REFERENCES

RMA Standard SE-105, "Microphones for Sound Equipment".
ASA Standard Z24.1, "Acoustical Terminology".

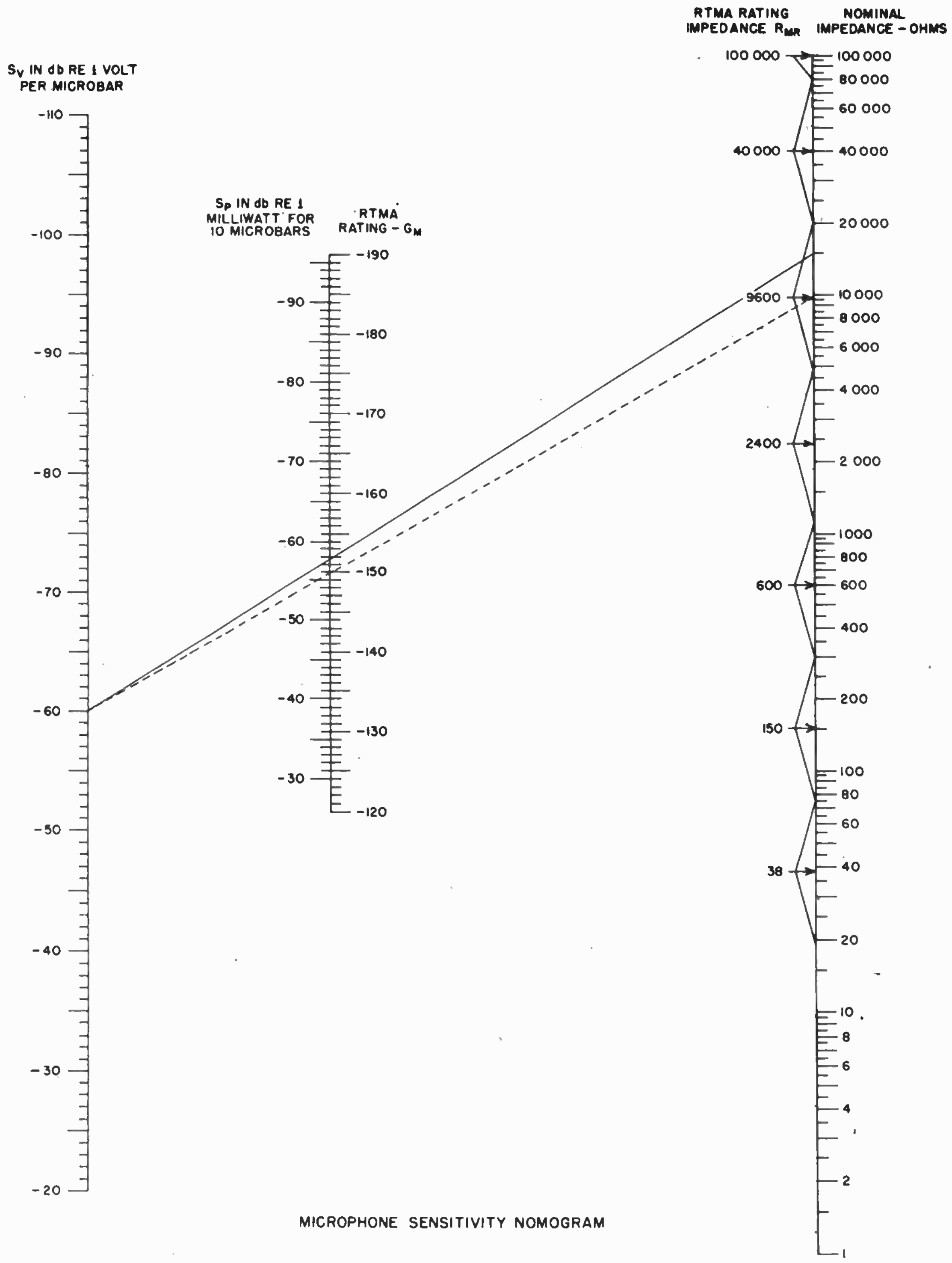

Fig. 1

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Marvin Camras, Armour Research Foundation, Chicago 16, Illinois.

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Loudspeakers and Transducers of All Types

(Please see inside back cover for additional names)

Transactions of the I·R·E

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

May-June, 1953

Published Bi-Monthly

Volume AU-1 Number 3

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IRE PROFESSIONAL GROUP ON AUDIO

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

Annual Assessment: $2.00

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PGA -- 1953-54

Marvin Camras
Armour Research Foundation
Chicago 16, Illinois

This year has seen drastic political changes in major world governments, and also in the PGA. Elected and appointed officers are:

Marvin Camras, Chairman 1953-54 (Armour Research Foundation, Chicago, Illinois)

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A. M. Wiggins, 1953-54 (Electro-Voice, Inc., Buchanan, Michigan)

An informal luncheon meeting was held Wednesday, March 25, to acquaint old and new committee members. As a result of the discussion it was decided that we continue to issue TRANSACTIONS OF THE IRE-PGA on a bi-monthly basis. It was also decided to study income and expenses so that the annual assessment can be revised. However, the assessment of $2.00 will continue until further action by the administrative committee.

PGA has grown so remarkably in the last year that we ought to take inventory of member's interests and affiliations. A survey by postcard is planned. The results will help the officers and editorial staff to serve PGA better, and to make this the best year in our history.

EDITORIAL COMMITTEE REORGANIZATION

Daniel W. Martin
The Baldwin Company
Cincinnati 2, Ohio

For the past two years the IRE-PGA has been very fortunate to have Mr. Benjamin B. Bauer, Shure Brothers, Inc. as editor-in-chief of its TRANSACTIONS. As a matter of fact Mr. Bauer, with only token assistance from the PGA Editorial Committee and more substantial publication assistance from IRE headquarters, has transformed TRANSACTIONS of the IRE-PGA from an irregular, mimeographed newsletter to a regularly scheduled, technical publication with well defined format, policies and procedures. The Editorial Committee wishes to thank Mr. Bauer for his successful efforts and for his willingness to continue as editorial advisor and an active member of the committee, while serving as the new Secretary-Treasurer of IRE-PGA.

The reorganization of the Editorial Committee includes not only new and additional personnel, but also a revised plan of operation. The regional editorship idea is not to be abandoned completely, but will be retained as part of the new plan. We will attempt to distribute some of the editorial functions, and to assign subject-matter classifications within audio to individual committee members, while each member of the committee will serve as a regional editor in the particular region centered about his geographical location. Members of IRE-PGA will know whom to contact on a regional basis by referring to the addresses of the editorial committee members listed on the inside cover.

The rapid growth of IRE-PGA is believed to have resulted in large measure from IRE membership interest in the advancement of audio technology through timely publication of the results of fundamental research, of engineering development and design, and of experience gained in the use of audio systems. It is the pleasant responsibility of individual IRE-PGA members to be active in these matters, as well as interested. When some new principle or device or concept has been discovered, or when something well known has been re-examined to new advantage, the membership of IRE-PGA will certainly appreciate a first chance to hear about it through TRANSACTIONS of IRE-PGA. Policies on republication of material elsewhere, previously explained in these columns, will continue.

CINCINNATI CHAPTER, IRE-PGA, 1952-53

The Cincinnati Chapter of the IRE-PGA has completed its second yearly program of monthly meetings and technical papers, under the leadership of the following officers for 1952-53:

Chairman: Roy E. Kolo, Cincinnati and Suburban Telephone Company
Vice-Chairman: E. M. Jones, The Baldwin Company
Secretary-Treasurer: Albert Meyer, The Baldwin Company

The Audio Chapter sessions were regularly held at the headquarters building of The Technical and Scientific Societies Council, immediately following the Cincinnati Section IRE meetings. This arrangement has proved beneficial to both section and chapter. The chapter has enjoyed the whole-hearted support of the section in all of its activities. The following technical papers were given, generally with demonstrations and aural-visual aids.

"The Technique of A-B Listening Tests", Daniel W. Martin, The Baldwin Company.

"British Audio", H. A. Hartley, London, England

"Quiet, Please", Robert Holden, E. C. Decker Company

"Historical Development of the Loudspeaker", Wm. H. Breunig, The Baldwin Company

"Fifty-Thousand Hands", M. E. Strieby, American Telephone & Telegraph Co. (joint IRE-AIEE)

"Magnetic Structures for Audio and Acoustical Devices", C. A. Maynard, Indiana Steel Products Co.

"Carillonic Bells", George J. Schulmerich, Schulmerich Electronics, Inc.

At the April meeting the following officers were elected for 1953-54.

Chairman: E. M. Jones, The Baldwin Company
Vice-Chairman: W. W. Gulden, Cincinnati and Suburban Telephone Company
Secretary-Treasurer: J. P. Goode, Cincinnati Time Recorder Company

STATUS OF MILITARY RESEARCH AND DEVELOPMENT IN ACOUSTICS AND RADIO*

Paul J. Weber
Bureau of Ships, Department of the Navy
Washington 25, D. C.

Summary

The fields of acoustics and audio engineering are finding many new military applications within the Army, Navy and Air Force. Continued progress is also being made toward improving the performance of existing audio systems and techniques to better meet increasingly stringent operational requirements. Some recent developments and proposed projects in these fields are discussed. Problems which are as yet unsolved are mentioned. The procedure established within the Department of Defense for coordinating all research and development in the broad field of acoustics-in-air is described.

A. Military applications of audio

The allied fields of acoustics and audio are playing a variety of important roles in the military operations of the Army, Navy and Air Force. Typical examples are the following:

1. Voice Communications

The most familiar military application of audio is its use for voice communications.

In the Army, ground units are linked together by field telephones. Figure 1 shows a soldier wearing one of the latest Army sound powered telephone handsets. Special features of this handset (shown in Figure 2), include a hand-operated generator for signalling another station, a buzzer with volume control for receiving an incoming call, and a visual signal for silent calls.

In the Navy, communication between key stations aboard ship is handled almost entirely by sound powered telephones and intercommunicating units (see Figure 3) which provide selective two-way amplified voice communication. A typical control station operator on a Navy submarine is pictured in Figure 4. Note the extent to which he is wired for sound -- a headset, two sound powered telephone handsets, a radio telephone handset for exterior communication, a microphone for one-way broadcast to a special information circuit and an intercom unit.

Small hand-held battery powered public address sets called electric megaphones are used aboard ships to facilitate such operations as fueling at sea and small boat handling.

*Presented at the Southwestern IRE Conference, February 7, 1953, in San Antonio, Texas. Manuscript received February 12, 1953.

This same device is used by the Army's Military Police. The megaphone may be detached from the amplifier and used like the familiar cheer-leader's hand megaphone, while the amplifier is carried in the other hand or set upon the deck.

Larger portable self-powered public address sets are used on beaches during landing operations for controlling the movements of boats, vehicles and personnel.

2. Sound Ranging

A less familiar application of acoustics by the Army is for locating enemy artillery and other weapons from the sound produced by the firing of such weapons. Although this is an old technique, having been employed during World War I, sound ranging still offers some real advantages over other more modern methods. Sound ranging equipment is relatively simple and does not reveal its existence or location to the enemy.

3. Recording

An application of audio which is growing rapidly in scope and importance throughout all three of the services is the use of recorders and recording techniques for the recording and subsequent reproduction of voice, sound and other signals -- both within and beyond the audio frequency range. In addition to the usual uses of voice recorders, there are many applications of a strictly military nature in which recording techniques serve as a valuable tool for signal analysis.

4. Psychological Warfare

An application of audio, which has received new and expanded emphasis since the outbreak of hostilities in Korea, has been in the field of psychological warfare. High powered public address systems mounted in aircraft, and on tanks and other armored vehicles, have proven particularly effective when used for such purposes as:

(1) Directing surrender proposals to by-passed enemy units and to isolated pockets of resistance;

(2) Broadcasting propaganda designed to lower the morale of the enemy and to encourage desertions and surrender.

B. Recent developments

Some of the more significant developments which have been made recently are the following:

1. New Aircraft Audio System

The most ambitious military postwar development in audio has been the development by the Air Force of a completely new audio system of high intelligibility interphone equipment for military aircraft. The interphone system of a military airplane provides the means by which the individual crew members of a multiplace airplane converse with each other. The system also provides terminal equipment for the various radio receivers and transmitters employed for plane-to-plane and plane-to-ground communications. The high acoustic noise levels in today's planes has seriously reduced the voice communication effectiveness of World War II equipment. A completely new line of audio equipment components had to be developed. This resulted in a dramatic improvement in intelligibility under the most adverse conditions encountered in modern military aircraft. The improvement was brought about by a combination of the following means:

(1) Carbon microphones with their inherent distortion, instability, and limited frequency response were replaced by moving coil dynamic microphones;

(2) The microphone was designed for pressure gradient operation to discriminate against background noise, particularly the lower frequency components of propeller aircraft noise. A rubber mouth shield was added to give additional noise discrimination against high frequency noise components encountered in jet aircraft (Figure 5). Better noise insulation was built into the earphone cushions;

(3) Earphone receivers of the moving diaphragm magnetic type were replaced by moving coil dynamic receivers;

(4) The design of both the microphone, receivers, and the associated amplifier circuitry included means for eliminating the effect of altitude variation on the performance of the system. This is better appreciated when we consider the enormous range over which the acoustic impedance of the air medium itself varies from sea level to an altitude of 40,000 feet;

(5) Speech clipping, signal compression and frequency equalization were all employed in the amplifier circuits to decrease signal strength variations and to improve the overall signal-to-noise ratio in the ears of the listener;

(6) A great deal of attention was paid in designing the earphone cushions to provide maximum comfort for the wearer, permitting the wearing of headphones without fatigue for long periods of time.

2. Flight Deck Announcing

The military are the biggest users of high powered audio systems. The largest known concentration of audio power exists on board a large Navy aircraft carrier. Each carrier of the MIDWAY class has installed seven audio power amplifiers, each capable of delivering 1,000 watts of audio output power. Three of these seven amplifiers are used to provide through-ship sound coverage. Nearly 1,000 loudspeakers are used for this purpose. The other four kilowatts of audio power are fed to a small number of super-power loudspeakers (see Figure 6), usually referred to as "bull horns," which cover the flight deck with sound, permitting speech and alarm signals to be directed to pilots and to plane handlers engaged in flight operations. These loudspeakers are mounted on the island structure, the only structure which projects above the level of the flight deck. The system produces a minimum sound intensity level of 110 decibels over the entire area of the flight deck, which is approximately 1,000 ft. long by 100 ft. wide. This represents a length of two city blocks and an area of $2\frac{1}{2}$ acres.

On the new aircraft carrier which the Navy is building, referred to as the FORRESTAL class, there will be no island structure. The new carriers will truly be flat tops. Sound coverage of the flight deck, which is considerably larger than previous flight decks, is still an operational requirement, in spite of the fact that there is no longer any place to mount the "bull horns." To solve this problem, the Navy has under development a new compact high-powered loudspeaker of compound horn design having an extended frequency range for maximum speech intelligibility in the presence of the higher noise levels which are expected; and having a wide sound dispersion angle to permit uniform coverage of the flight deck when using a large number of such speakers spaced at regular intervals along the edges of the flight deck.

3. Voice Plane

Another development employing high-powered audio equipment in connection with aircraft is the so-called "voice plane" (Figure 7) being used currently in Korea for psychological warfare purposes. It is a medium bomber fitted with four 500 watt amplifiers, each feeding a multiple-driver horn type of loudspeaker (Figure 8) mounted in the bomb bay (Figure 9). These planes are capable of delivering a message on the ground of approximately thirty seconds duration when flying at an altitude out of accurate range of small weapons fire. Other smaller systems are designed for ground use on tanks and armored vehicles (Figure 10). Still smaller systems are carried and set up by hand in front areas for broadcasting tactical propaganda (Figure 11).

4. Low Frequency Recording

Some significant achievements have been made in the development of new recording techniques. The results of one Navy-sponsored development was recently reported.* The electron beam reproducing head for magnetic tape recorders permits the reproduction of extremely low frequency signals and preservation of signal wave form without the complexities of equalization and phase correction, and without the use of frequency modulation usually employed for this purpose.

5. Improvement of Testing Techniques

A minor phase of military research, but one worth mentioning, is the effort that goes into the investigation and development of test techniques and methods of measurement. It must be borne in mind that the military departments are required by law to buy their technical equipment and supplies under open competitive bidding procedures using specifications based upon performance requirements. In the field of audio, the state of the art in some areas has not yet progressed to the point where it is possible to completely define the performance of an item by objective tests alone. Anyone familiar with the efforts of the Institute of Radio Engineers, Radio and Television Manufacturers' Association or of the American Standards Association to establish standard test methods and values for loudspeakers will confirm this fact. At the present time three projects of this type are being pursued by the Navy's Material Laboratory at the New York Naval Shipyard. We buy what we call noise cancelling microphones, and yet we have no satisfactory way of measuring the noise cancelling properties -- except by using actual talkers and listeners in an artificial noise field. One of these projects, then, concerns itself with seeking objective means for evaluating the effectiveness of a microphone in discriminating against background acoustical noise; and of finding a method for predicting its performance in any given noise situation using data obtained from objective tests. A second project is concerned with a basic investigation of non-linear distortion, its causes, its effects, its measurement and acceptable limits for various audio components including loudspeakers. A third project is concerned with a similar investigation of flutter and wow in recording and reproducing systems. We believe the results of these investigations will contribute to and advance the state of the audio art. This type of work is unclassified, and it is planned to report the results obtained at open meetings of technical societies.

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* "Electron Beam Reproducing Head for Magnetic Tape Recording" by Dr. A. M. Skellett and Dr. Lawrence E. Loveridge, National Union Radio Corp., and Mr. J. Warren Gratian, Stomberg-Carlson Co. -- Presented by Dr. Skellett at the Southwestern IRE Conference in San Antonio, Texas, Feb. 7, 1953.

C. Unsolved Problems

And now, a word about some of the problems which still lie ahead of us, and which suggest future research and development work.

1. Sound Propagation in Air

One of the most important unsolved problems is the lack of adequate fundamental data and understanding of the propagation of sound in air under non-homogeneous conditions. What little data is available covers only ground-to-ground (or horizontal) propagation of sound through air. There is a dearth of data covering air-to-ground (or vertical) sound propagation. Much needs to be known about the effects of varying atmospheric conditions and terrain on sound transmission. There are a number of military applications for which this information is needed, one of which is the psychological warfare problem already described. Typical questions which must be answered regarding the operation of loudspeaker systems from aircraft are the following:

(1) What are the audio power requirements and optimum system characteristics (such as frequency response) for successful sound transmission to the ground?

(2) How should loudspeakers be mounted and baffled on the aircraft for optimum performance; and how does the speed of the aircraft affect this problem?

(3) How can we predict the performance on the ground which we will get under a given set of atmospheric conditions?

2. New Techniques of Sound Amplification and Reproduction

The demand for audio power in airborne loudspeaker systems is already in excess of that which can be satisfied with a reasonable size and weight of equipment mounted in the plane -- using the audio techniques which are known today. The solution to this problem requires discovery of radically new techniques of audio amplification and efficient conversion from electrical to acoustical power. One possible new approach to the loudspeaker problem which should be explored is one wherein the audio amplifier can act as a valve to control or modulate the flow of energy drawn from some source other than the amplifier itself. This is intended to apply in a much broader and more general sense than the one example of a "modulated air stream loudspeaker."

3. Wideband Recording

In the field of recording, all three departments are actively investigating methods for distortionless wideband recording and reproduction. Immediate applications exist for recorders capable of recording and playing back signals of frequencies up to five megacycles. The number and scope of such applications will be limited mainly by the complexity, size and weight of the system developed. The simpler, the smaller, and the lighter they can be made, the more uses they can be put to. In some cases research programs in wholly unrelated fields are being retarded awaiting the availability of recording equipment capable of recording signals at least up to 100 kilocycles without phase shift, drift or frequency instability.

D. Army-Navy-Air Force Coordination

Following this discussion, one might well ask, "With so many parallel applications of audio and acoustics within the three departments, is there not a great deal of duplication and overlapping?" With so many different groups and laboratories, so widely separated geographically, each attacking similar problems but from their own local viewpoints, there might be some doubts as to whether this could be otherwise. The following comments will describe the way in which "unification" is working within the Department of Defense in the areas of acoustics and audio.

1. Research and Development Board

The IRE Proceedings recently contained an article* which described the work of the Research and Development Board of the Department of Defense. This board is the agency charged with coordination of the research and development program carried on by the Army, Navy and Air Force. The RDB is made up of committees and panels, staffed by part-time military and civilian experts, each concerned with a specific field of science or a certain type of weapon.

2. Allocation of Responsibility to a Single Service

Certain technical areas exist in which all three services are equally active, and in which the operational requirements for equipment and systems are very similar. Acoustics-in-air is such an area. I use the term acoustics-in-air to differentiate the audio phases of acoustics from the underwater or sonar phases. In a few cases such as this, the RDB has recognized the desirability of providing a mechanism for closer and more continuous coordination and guidance of the research and development program on a day-to-day basis, than could be achieved through the normal workings of committees and panels which meet infrequently. What

* "Research and Development for National Defense" - by Edwin A. Speakman - Proceedings of the I.R.E., July, 1952, p. 772.

the RDB has done in these cases is to assign primary responsibility to one of the three departments for the coordination, supervision and guidance of the research and development work in all three departments.

Such an allocation of responsibility was made in the field of acoustics-in-air to the Department of the Navy. The Navy's Bureau of Ships is the agency which has been chosen to do this job. Thus in effect, the Bureau of Ships has been deputized by the RDB to coordinate the acoustics and audio research and development for all laboratories and offices and bureaus within the Army, Navy and Air Force.

3. Solving the Problem of Effective Communications

What specific actions are being taken by the Bureau of Ships under the RDB allocation to achieve effective coordination between the three departments in this field?

First we must consider the big problem in any large organization. Coordination is only as good as the internal communication. In the Department of Defense the normal channels for communication or flow of information within and between the services, follow the established organizational and functional lines. This creates a real problem of delivering timely information to the man at the bench and the man with the slide rule, who wants and needs to know. What does finally reach him is often bound together with so much other unrelated information, and carries so high a security classification, as to discourage its use. Principal emphasis has therefore been placed upon establishing well-oiled channels of communication so that all engineers and scientists working in acoustics are kept informed of what the others are doing.

4. Periodic Bulletin

Several steps have been taken to increase and speed up the vital flow of information. One of these has been to publish a periodic "Acoustics-in-Air Bulletin" containing news and notes on new projects, proposals, reports and current status of work within the various fields of military acoustics.

The style of the bulletin is informal in nature, to make easy reading for busy people. The bulletin is mailed directly to all key personnel within the three departments who are concerned with any phase of acoustics research or development.

5. Semi-annual Symposium

Effective as this and other measures have been for transmitting the written word expeditiously, they cannot do the whole job. There is still an appreciable delay before the results of a research and development project find their way into writing -- and some indeed are not written down at all. This problem has been solved in part by conducting a Department of Defense acoustics-in-air symposium twice a year, usually held in conjunction with the meetings of the Acoustical Society of America for the convenience of those who may wish to attend both functions. Here papers are presented and discussed by the engineers and scientists of the Department of Defense. Timely data and information are exchanged. New ideas and fresh approaches to common problems are stimulated by these group discussions and interchanges. The security classification of the meetings is set high enough to permit free discussion of all aspects of the problems, including actual military applications.

6. Appraisal of Results

It may be too early to fully appraise the results of these efforts. It can be said, however, that in our field of acoustics there has been firmly rooted a spirit of teamwork between the three services and an awareness of the other fellow's problems. Those doing business with the military departments today will not find the rivalry or wasteful competition between services which was a frequent complaint about the military in the days before unification.

A recent example will illustrate the degree of inter-service cooperation which has been achieved. This is the case of the airborne loudspeaker system shown in Figure 8. This equipment was procured through the Army's Signal Corps by the Navy's Bureau of Aeronautics under the direction of the Army's Chief of Psychological Warfare. It was installed by the Air Force in a B-26 bomber. It was flown by Air Force pilots in Korea, while being operated by Army psychological warfare personnel to broadcast Chinese and Korean propaganda messages to the enemy. When certain deficiencies were discovered in the performance of this equipment, the Navy's Bureau of Ships set up and directed a project at its Material Laboratory in the New York Naval Shipyard for a thorough test of the equipment and an investigation of the means which would be required to correct these deficiencies.

The intent of the foregoing remarks was to give some insight into the uses of audio in the Department of Defense, some of the advances which have been made, some of the problems we still face, and some of the measures we have adopted to obtain maximum return on our investment of defense dollars and maximum utilization of our technical and scientific manpower.

Fig. 1

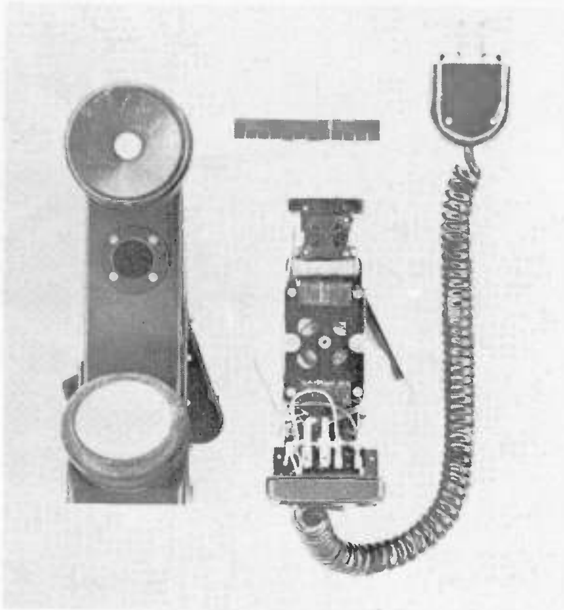

Fig. 2

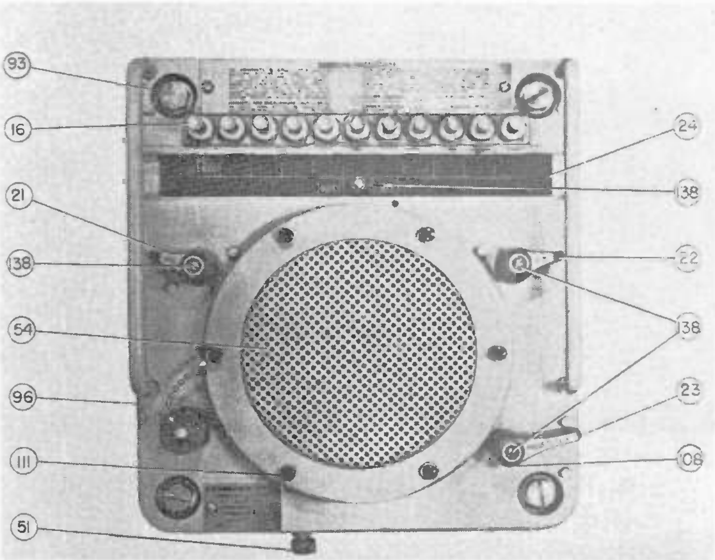

Fig. 3

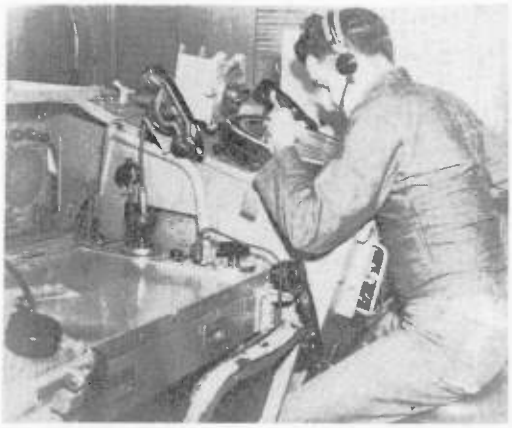

Fig. 4

Fig. 5

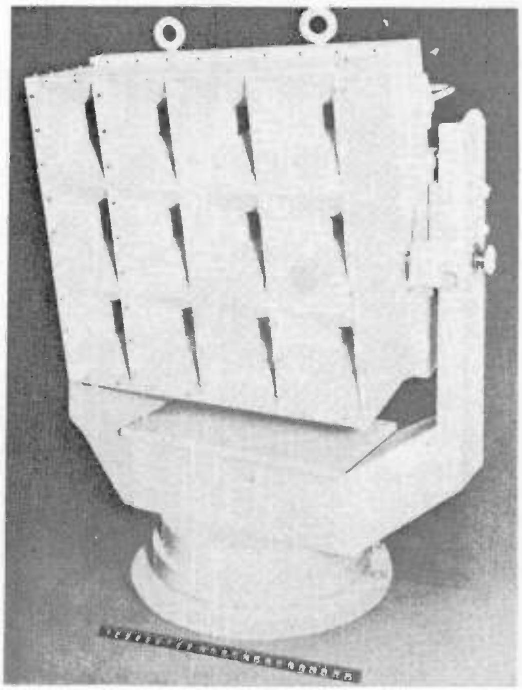

Fig. 6

Fig. 7

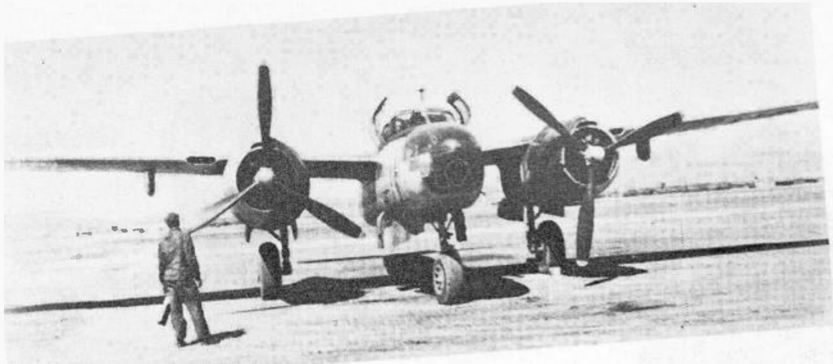

Fig. 8

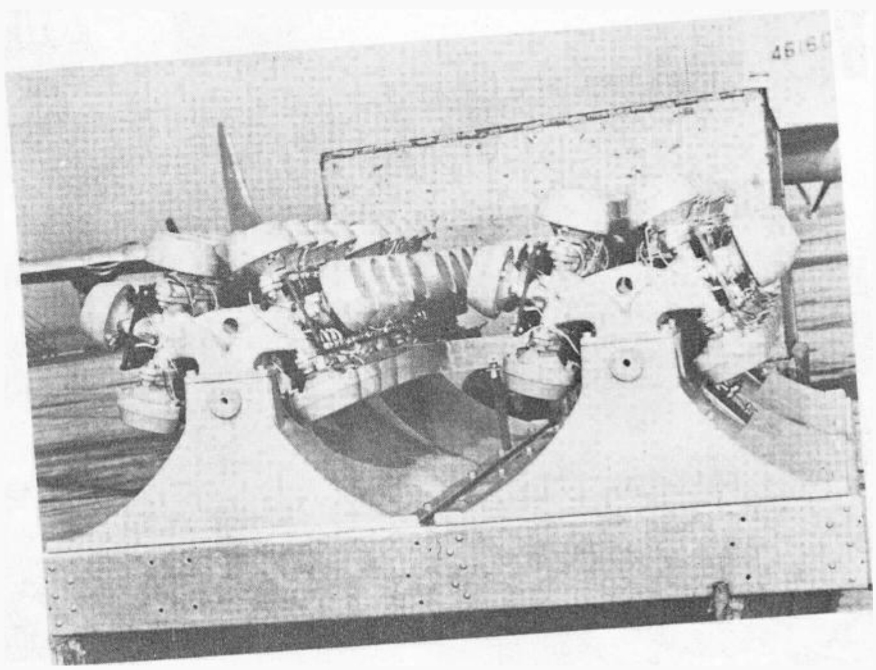

Fig. 9

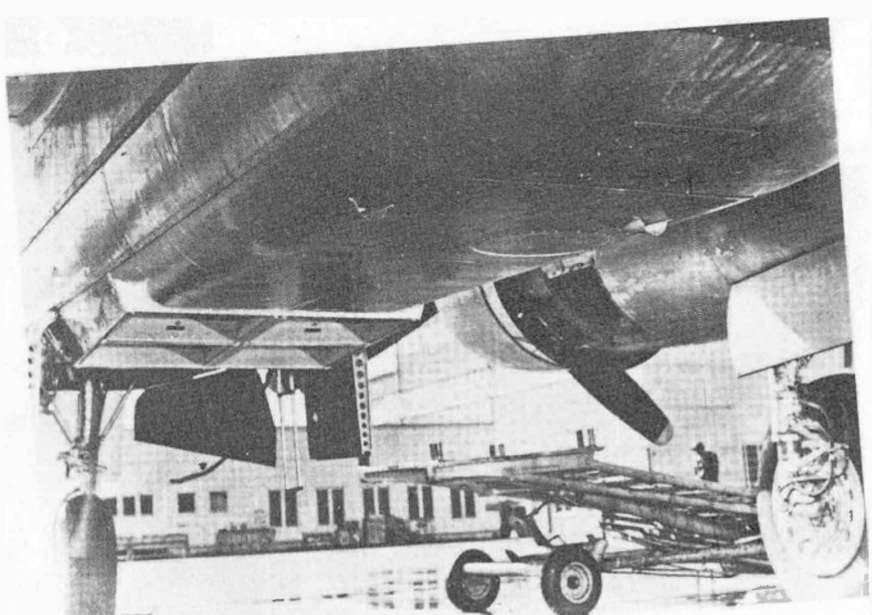

Fig. 10

Fig. 11

LOUDSPEAKER DEVELOPMENTS*

Paul W. Klipsch
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If there is ever a culminating design for a loudspeaker it should be a corner type, for this affords the maximum performance for a given bulk of space requirement.

The year 1953 has shown signs of approaching this culmination more closely than any time in the past. One manufacturer's production of corner speakers has run its serial numbers into 5 figures, which is still short of any suggestion of market saturation.

To trace the development of the corner speaker entails listing a short technical bibliography (See appendix). But to trace some of the basic and major developments in loudspeakers entails digging around in material which by current scientific standards is ancient. Every acoustic worker is aware of Rayleigh's Theory of Sound, but may well have overlooked the complete analysis therein of the direct radiator speaker to be invented 46 years later. The principle of images is probably at least 20 centuries old, and as applied to acoustics may be almost as old (surely echoes must have been noticed almost as many centuries ago as optical reflections), but it appears to have been as late as 1929 that the reflections produced by the room walls at a corner were first gropingly sought as a means of improving loudspeaker performance.

The beginnings of the corner speaker would be hard to find; perhaps it would be fair to assume the beginnings involved the use of a single wall, or the idea of a baffle. The first practical use of the baffle appears to have taken place in the late 1920's, but the true origin goes back farther, at least to the aforementioned Rayleigh's mathematical development.

In the present practical form, the corner horn speaker of conservative bulk appears to have originated in 1940 with this writer's patent application and subsequent publications which are included in the bibliography of this paper.

It was in 1877 that Lord Rayleigh published his Theory of Sound. The mathematical treatment of a direct radiator in an infinite baffle was treated under the index item "plate, vibrating circular, effect of air upon." For the reference conscious, the development starts on page 162 of Volume 2, Dover reprint of 1937.

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*Presented at the Southwestern IRE Conference, February 7, 1953, in San Antonio, Texas. Manuscript received February 20, 1953.

Forty-six years after Rayleigh's publication in 1877, the direct radiator loudspeaker was invented by Chester W. Rice and Edward W. Kellogg; this was reported in Proc. IRE in 1923. This dynamic cone speaker development appears to have been first exploited in the Radiola 28, the AC model of which used a fieldcoil dynamic speaker of about an 8-inch diameter, without baffle. The expression, "without baffle," should be especially noted. It is significant that, even in the mid-twenties, science had become so diversified that workers in one field of applied science could not find the pure or applied science writings in other fields. Hence the inventors of the direct radiator, analyzed 46 years before, could be excused if they were unaware of the analysis and significance of Rayleigh's assumptions, which included the baffle or wall with a hole in it within which the piston vibrated.

Actually it was in the late twenties that the workers in the loudspeaker field began to awake to the beneficial effects of a "baffle" (and note again the quotes around "baffle"). Still later, the early thirties, the applications began to appear, and at the present time, in 1953, we still observe the 20-year old influence of the loudspeaker in the hole in the wall.

Olson quotes J. B. (Jim) Lansing as the author of the combined bass cone and coaxial center-fire horn tweeter (Jour. Soc. Mot. Pict. Eng. 46, 3, 1946). This arrangement appears to have been first devised by A. A. Crawford in 1929, when an archive model was made from sketches prepared by Crawford. This speaker type, as a drive system, began to gain popularity after World War II, and is particularly applicable to certain types of corner speakers. Twenty-four years old in 1953, this coaxial speaker idea is only beginning to exert its proper influence in both corner and non-corner applications.

The first speaker system involving a plurality of speakers to reproduce several tonal ranges is variously attributed to a number of workers, depending on the author. Hilliard gives credit to Shearer in 1934; the Crawford patent application of 1928 antedates that by half a decade. The tools for the accomplishment were developed to a state of practicality with the Bostwick tweeter described in the Jour. of Acous. Soc. of America in 1930. But the ground work was laid much earlier. Rice and Kellogg were trying to obviate the expense and bulk of horn speakers in their work leading to the cone speaker, but their IRE paper of 1923 described experiments using 3 horn speakers each working in a portion of the audio spectrum. They observed that this 3-horn speaker sounded better than any design of single horn they had been able to try. The multiple horn speaker reached the highest state of development when applied to corner speaker systems.

As for corner horns, the first patent to describe a potentially wide range performance with conservative bulk was issued to Sandeman in 1934. This was for a trihedral corner horn to be reinvented by Gilson and Andrea in 1951, almost exactly 17 years later as the Sandeman patent was expiring. This discussion is limited to optimum speaker systems, or those wherein maximum performance per unit bulk is approached. This excludes an early horn invention, but since this appears to be the first corner horn it is necessary to mention the Weil patent of 1931.

This writer's contribution began with a corner horn of dihedral symmetry, as contrasted with the Sandeman trihedral structure. Patent application was filed in 1940. Since then improvements in the bass system, and developments in associated treble horns were found necessary to achieve an optimum speaker system.

The year 1952 saw the beginnings of recognition of corner horns. Previously, only this writer and the Electro-Voice Company had exploited the corner horn design. But 1952 saw a number of independent designs, as well as near and remote copies of basic corner-horn structures.

The basic idea of the corner horn speaker depends on the fact that the mirror image formed by a wall doubles the effective radiation area, and enables a relatively small radiating surface to generate a longer wavelength with practical efficiency; the multiple reflections of 3 mutually perpendicular walls in a room corner increase the effective radiating area by a factor of 2 cubed, or 8 times.

The basic horn idea may be applied in two ways: first, as a direct radiator with corner horn back loading, and second, as a series of 2 or more horns at least one of which is an optimized corner type. In the first, the speaker is essentially a direct radiator, but the back of the cone radiator is corner horn loaded to enhance the extreme bass range. In this application it is essential that some special means be provided to hold the bass radiation to a restricted efficiency so it will be a comparable level of loudness with the treble radiation from the front of the cone. This is usually accomplished with some form of acoustic filter having a low-pass characteristic.

In the second form of application of the corner horn, the efficiency is maintained at as high a value as possible, and the middle and extreme treble ranges are radiated with horns exhibiting efficiencies comparable with the bass corner horn. Note that the word "comparable" rather than "equal" is used. It is possible, feasible, and practicable to design bass or woofer horns with efficiencies of the order of 60 to 90 per cent. If we define efficacy as the product of efficiency times the ability to absorb power from a mismatched generator impedance, the woofer efficiency attainable is at least 40 per cent in optimum designs, which means than an acoustic output of 4 watts may be obtained from a 10 watt amplifier. But considering the tweeter, the ultimate efficiency in the absence of deleterious resonant peaking is of the order of 10 per cent at ten thousand cycles and the efficacy will be still lower. In other words, the wide-range multiple-horn system will exhibit a response which is at least 6 decibels down, or perhaps ten or more, at ten kc referred to the woofer range.

The multiple horn, therefore, appears at first glance to be at a disadvantage. However, it is to be noted that the response of a properly designed multiple horn system droops smoothly, and can be readily equalized. This is in contrast to the simpler radiators which exhibit serious peaks and valleys of response which could hardly be equalized to an over-all flat response. It should be hurriedly stated that such equalization should be prior to the final amplifier stage, and not in the form of attenuator pads between amplifier and speaker units.

In the first class of corner horn designs, in which a cone speaker is used as a direct radiator with the back radiation utilized through a corner horn, there are several commercial forms being exploited. The idea is believed to have been first described in this writer's U. S. Patent issued in 1943. The first to be produced was the Electro Voice Regal in 1950, quickly followed by the now popular Aristocrat, the diminutive Baronet and the well-named Royal. Klipsch and Associates introduced the Rebel in the summer of 1950, and a modified improvement in 1951. In late 1952 the Gately "Super-Horn" appeared as a commercial offering which enjoys the novelty of not being a near-copy of earlier widely published corner horn designs.

The corner horn as a back-loading for the direct radiator is the most economical approach to an extended bass range, and the application is apt to be applied to an excess degree. The performance of speaker systems in past years has been deficient in the bass range, and a bass-hungry public has insistantly demanded more bass to the extent that manufacturers have been able to make an easy dollar by feeding an extra dozen decibels in the range below 100 cycles. There is beginning to be recognized a revolt in the ranks of discriminating listeners; the rumbling bass with a lack of transient resolution still finds wide popularity but there is a growing section of the public that demands that percussions have some semblance to realism, and a minority even demands that a drum sound like a drum. The result is that the better designs of corner-horn back loading of direct radiators are being recognized for their controlled response whereby the bass efficiency is limited to a level matching the direct radiator efficiency in the middle frequency range derived from the front of the cone.

In the second class of corner horns, the performance is entirely different; remember the corner-horn back-loaded direct-radiator exhibits the efficiency of a direct radiator, typically of the order of 2 to 5 per cent. The multiple corner horn, on the other hand, is designed to give efficiencies of the order of 50 per cent or more, with the idea that as efficiency is increased the distortion is decreased.

Obviously if the bass horn efficiency is of the order of 50 per cent, it is desirable for other frequency ranges to be reproduced with comparable efficiencies. It has been a major accomplishment to attain decent efficiencies in the middle and upper frequency ranges, while retaining satisfactory pressure response, polar pattern and freedom from extraneous zonal and sectorial diaphragm motions. Electro-Voice has gone so far as to use 4 separate speakers; Klipsch and Associates favors 3 speakers as being less subject to ground-plane reflection interference. The difference is academic technically, though there are economic factors which favor the choice of 3 or 4 speakers, depending on whether one manufactures drivers and buys horns, or manufactures horns and buys drivers.

Tracing the history of the loudspeaker, and particularly the modern types, may appear to be of academic interest only. But a perusal of the subject in depth will reveal an important truth. The corner speaker is fundamentally the most efficient and efficacious. It is the one type which will offer the most performance with least bulk. It is the only means by which the longer wavelengths of music, such as the 16-foot organ pipe tones, can be radiated by structures of size tolerable for use in the living room.

The principle of the corner speaker is so fundamental that it should be the stock in trade of every radio engineer, acoustic specialist, and, for that matter, every one who even merely listens to a record player. If one remembers that the speaker is one of the weak links in the acoustic chain, and that the link is strenghthened by a factor of eight by the wall reflections at a corner, it should be evident that a dollar value increase by a factor approaching 8 is available by taking advantage of the corner. The time should soon arrive when architects will be designing for corner speakers, instead of showing customers house plans with wall speaker installations which were an obsolete fad in the mid 1930's.

APPENDIX

(Bibliography of corner loud speakers)

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Sandeman, U. S. Patent 1,984,550 (1934).

Weil, U. S. Patent 1,820,996 (1931).

Klipschorn is the trademark of the Klipsch-designed speaker system built by Klipsch and Associates and protected by the patents listed below:

2,238,023	2,537,141
2,310,243	D 163,700
2,373,692	and pending applications

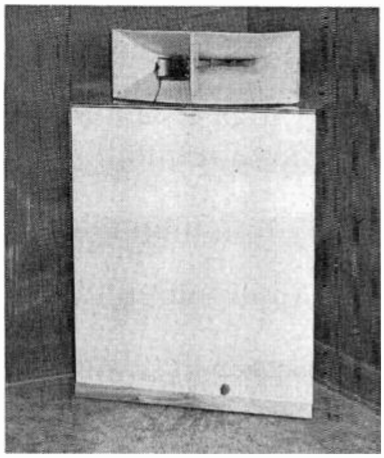


Fig. 1

Modern corner 3-way horn speaker system.
Note "tweeter" nested coaxially with "Squawker," both on top of "woofer."

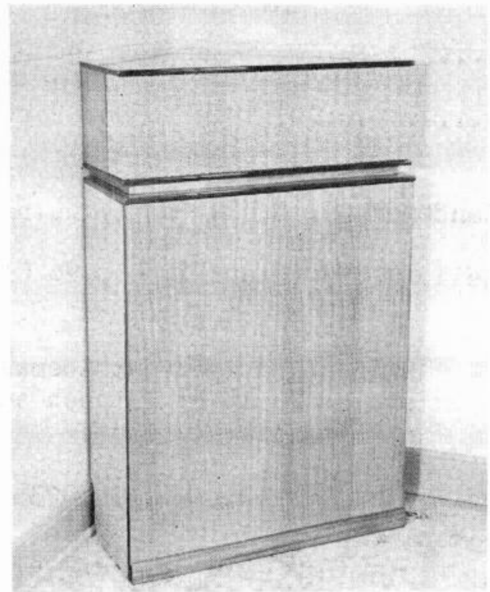
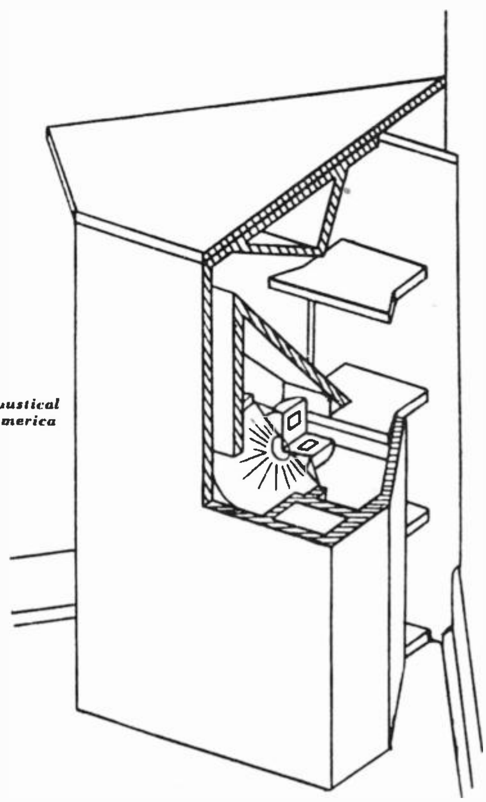


Fig. 2

Mahogany-finished version for home use; functionally same as system shown in Fig. 1.



Courtesy Acoustical Society of America

Fig. 3 Cut-away view of corner horn woofer.

ACOUSTIC DAMPING FOR LOUDSPEAKERS *

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The fundamental resonance of loudspeakers is recognized by many as a source of annoyance. Usually this resonance can be damped electrically by suitable selection of the amplifier impedance.[1] What is less well-known is that damping can also be achieved by acoustical means incorporated into the loudspeaker or the enclosure. This paper deals with the theory and methods for providing acoustic damping.

Electrical damping requires the use of a low impedance amplifier. When this type of source is not available, and in the absence of other damping means, performance of the system may be seriously impaired. For example, a typical home radio or record player has a pentode output stage without inverse feedback. This type of output stage is known to provide a source of high impedance. A loudspeaker driven from this source will exhibit a resonant condition which may cause poor transient response or "hang-over" and it may be responsible for the acoustic feedback in record players. Acoustic damping may be found helpful in this instance.

Another use for acoustic damping will be found in the design of high-fidelity systems, where frequent attempts are made to improve damping by lowering the amplifier impedance to a value approaching zero. There is a limit to the amount of damping which may be obtained in this manner.[2][3] Furthermore, electrical damping per se is not very effective in eliminating the resonance in cabinets with reflex ports. Acoustic damping can readily provide such additional damping as may be required.

Much effort and circuitry has been devoted to attempts to obtain damping from the electrical side. By contrast, the use of acoustic damping has been given little attention. In this paper we outline a simplified theory of acoustic damping for loudspeakers and enclosures. To provide a rational basis for the design of acoustic damping the acoustic constants of the loudspeaker and the enclosure must be known. Thence, an equivalent electrical circuit can be set up and the damping resistance may be determined experimentally by adjusting the electrical circuit constants. Keeping this approach in mind, first we derive the resonant frequency equations of a loudspeaker in a flat baffle and in an enclosure. Next, from these two equations we determine the acoustic mass and compliance of the loudspeaker cone. Thirdly, we set up an equivalent electrical circuit and determine the required damping resistance. And finally we build the acoustic damping into the enclosure and test its acoustic performance.

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II. EQUIVALENT CIRCUITS OF LOUDSPEAKERS AND ENCLOSURES

The equivalent circuit of a loudspeaker installed in an "infinite" flat baffle is shown in Fig. 1. In actual practice, of course, the baffle need not be infinite, but only considerably larger than the loudspeaker. The acoustic mass of the moving system (consisting of the voice coil and the piston diaphragm) is represented as an inductance L_{AM}; the acoustic compliance of the elastic suspension is shown as a capacitor C_{AM}. These terms can be calculated from measurements of the resonant frequency, as shown later in this paper. The driving pressure due to the voice-coil force is represented by a constant voltage e_o. We assume that the loudspeaker is driven from a high impedance source. Therefore, the damping factor reflected from the electrical side Z_{AP} equals zero. Either side of the piston is confronted by the acoustic impedance of the medium, Z_{AP}. This impedance is shown grounded because, from the acoustical point of view, it has only one available driving point terminal.(4)

It will suffice our purpose to represent Z_{AP} by a parallel combination of an inductance L_{AP} and a resistance R_{AP}.(5) These elements can be calculated in terms of the properties of the medium and the effective radius of the piston r:

$$L_{AP} = \rho/\sqrt{2}\pi r = 0.00027/r \quad \text{grams-cm}^{-4} \text{ "Acoustic henrys"} \tag{1}$$

$$R_{AP} = \rho c_v/\pi r^2 = 42/\pi r^2 \quad \text{dynes-sec-cm}^{-5} \text{ "Acoustic ohms"} \tag{2}$$

where, r is measured by the perpendicular distance from the axis to the mid-point of the flexible annulus of the cone, cm.

ρ is the density of air, grams per cc = 0.0012 for normal atmosphere.

c_v is the velocity of sound in air, = 34,400 cm per sec. in normal atmosphere.

These equations are given for circular pistons; however, they may be used for pistons of other shapes, in terms of circular pistons of equivalent area.

R_{AP} predominates at high frequency; L_{AP} predominates at low frequency. Loudspeaker resonances occur at low frequency, and hence, R_{AP} can be neglected in the resonant frequency equation:

$$(2\pi f_1)^2(L_{AM} + 2L_{AP})C_{AM} = 1 \tag{3}$$

Resonant frequency can be measured by connecting the voice-coil to a variable frequency oscillator through a high series resistance (to approach a constant current condition) and measuring the voice-coil voltage with a high-impedance voltmeter. At the resonant frequency the voice-coil voltage is a maximum.

When a loudspeaker is installed in a shallow cabinet with open back, its resonant frequency is essentially the same as with a flat baffle of similar dimension.

Next, consider the loudspeaker in an enclosure as shown in Fig. 2. The simplest enclosure is a closed box with a hole of an appropriate size to accommodate the loudspeaker. The design of such enclosure has been treated in detail by Beranek[6] and others. Our purpose is limited to providing its equivalent circuit elements.

To minimize reflections at high frequency, the enclosure may be lined with sound absorbing material such as Fiberglas or Ozite. These materials, when mounted against the walls of the enclosure, offer a negligible amount of sound absorption at low frequency and contribute little to the damping. Therefore, the low frequency impedance confronting the piston is predominantly reactive, and it can be represented by an inductance L_{AB} in series with a capacitance (or compliance) C_{AB}. For rectangular boxes, L_{AB} is given by the semi-empirical equation:

$$L_{AB} = \frac{4}{\pi^2} \frac{\rho \ell}{A_B} \left(\frac{A_B}{A_P}\right)^{2/3} \text{ grams-cm}^{-4} \text{ ("acoustic henrys")} \qquad (4)$$

The equation for acoustic compliance is well-known.[7]

$$C_{AB} = \frac{V}{\rho c_v^{\,2}} = \frac{V}{1.41} \times 10^{-6} \text{ cm}^5\text{-dyne}^{-1} \text{ ("acoustic farads")} \qquad (5)$$

where ρ is the density of air - 0.0012 grams per cc for normal atmosphere

A_B is the cross-sectional area of the box, sq. cm.

ℓ is the depth of the box, cm.

A_P is the effective area of the piston, sq. cm.

V is the volume of the box, cu. cm.

Equation (4) holds quite well under these conditions:

1. The loudspeaker is about equally spaced from all the points at the inside periphery of the box.

2. The ratio of the box area A_B to the effective piston area A_P is not in excess of about 10:1.

3. The depth ℓ is about $0.7 \sqrt{A_B}$.

The use of the word "about" is intended to signify that the conditions indicated can be violated considerably without undue error.

The air-load impedance at the front of the cone can be assumed to be given approximately by equations (1) and (2). The resistive component may be neglected in the equation for resonant frequency, as before. The resonant frequency f_2 of this system is contained in the equation:

$$(2\pi f_2)^2(L_{AB} + L_{AM} + L_{AP}) \frac{C_{AM}C_{AB}}{C_{AM} + C_{AB}} = 1 \tag{6}$$

The resonant frequencies measured in the baffle (f_1) and in the box (f_2) enable us to calculate the acoustical constants of the loudspeaker. From equations (6) and (3) the following expressions are derived:

$$L_{AM} = \frac{1 - (2\pi f_2)^2 C_{AB}(L_{AB} + L_{AP}) + 2(2\pi f_1)^2 L_{AP}C_{AB}}{4\pi^2(f_2^{\,2} - f_1^{\,2})C_{AB}} \quad \text{gram-cm}^{-4} \text{ ("acoustic henrys")} \tag{7}$$

$$C_{AM} = \frac{1}{(2\pi f_1)^2(L_{AM} + 2L_{AP})} \quad \text{cm}^5\text{-dyne}^{-1} \text{ ("acoustic farads")} \tag{8}$$

Having thus determined all the important circuit elements of the loudspeaker and the simple enclosure, we are ready to set up the equivalent electrical circuit and determine the damping resistance. Before proceeding, however, we must digress briefly to introduce the subject of acoustic resistance.

III. ACOUSTIC RESISTANCE

Acoustic resistance is encountered when sound is made to flow <u>through</u> thin crevices or slits. Consequently, any broad surface member may be converted into an acoustic resistance by perforating it with small holes, slits, etc., which are calculable by well-known equations.[8] Cloth, felt, Ozite, etc. mounted on a suitable supporting member in the path of sound flow constitutes a simple and inexpensive method for providing good acoustic resistance. The acoustic resistance of these materials cannot be calculated, but it can be measured easily by causing air to flow at a known rate through a given area of material and measuring the pressure drop.

An instrument designed to perform this measurement is shown in Fig. 3. The material is held between two disks which provide an aperture having a known area. The rate of air flow is adjusted by means of the lower gauge which indicates the pressure drop across a standard slit. The pressure drop across the material is shown on the upper gauge which is calibrated to read the specific acoustic resistance of the material in acoustic ohms per sq. cm.

Cloth is an especially satisfactory material and it can be selected to provide any desired specific resistance. For example, the open weave fabrics designated "Ninon" and "Georgette" have resistances from 1/4 to 5.0 ohms per sq. cm. Most sateens, broadcloths, etc. have resistances between 5 and 75 ohms per sq. cm.; sailcloths and similar tightly woven fabrics have resistances upward of 75 ohms per sq. cm.

Usually the acoustic resistance element should have an area about equal to the area of the piston, or greater; this is to avoid constriction of the air flow which would have the effect of added acoustic mass. The specific resistance of the cloth to be employed is easily determined by multiplying the required acoustic damping resistance R_{AC} by the chosen area of the element in sq. cm. The cloth is supported by cementing to a heavy wire mesh or perforated grille to avoid motion in a diaphragm-like fashion. When a perforated grille is used, the effective area is, of course, the net area of the openings. The resistance may be adjusted experimentally, as by blocking some of the holes of the supporting member by means of Scotch tape or cement.

IV. ACOUSTIC DAMPING

Acoustic damping may be applied to loudspeakers and enclosures in many ways. Some of these are shown in Fig. 4. In general, the damping resistance must be coupled to the cone by a sufficiently small enclosure. In this manner, a good deal of the air displaced by the diaphragm is forced through the acoustic resistance resulting in a damping action. In the simplest case, A, the small enclosure, is the sole enclosure and the damping is placed at the back or one of the sides of the box. This small enclosure may be attached to a large baffle, as shown in B, to improve low frequency response. At C, a large enclosure is used, the small enclosure being provided within the larger enclosure. D is similar to C, except that a damped port is provided for reflex action. At E, a port has been added to the enclosure of A; and finally, F shows an arrangement similar to D in which both the speaker and the port derive a measure of damping from a single acoustic resistance element. Many other combinations are possible.

When an acoustic resistance element confronts the external medium, as in A or B for example, the air particles moving through the element are confronted with an acoustic impedance. The approximate value of this impedance may be estimated through the use of equations (1) and (2).

Since our purpose is to describe acoustic damping techniques rather than to evaluate the merit of various designs, only the simplest arrangement, namely that shown in 4-A, will be treated in detail. An experimental speaker system utilizing this arrangement is shown in Fig. 5. It consists of a medium-priced 8" loudspeaker installed in an enclosure with inside dimensions 14 x 14 x 9 inches deep. This enclosure may well be typical of those used in a medium-sized radio or P.A. system. The enclosure has interchangeable backs; one of them is solid, and the other is provided with an 11" diameter hole for installation of the acoustical damping resistance. The choice of this resistance by electrical circuit analysis is to be described.

First, the acoustic constants of this system were determined by the steps previously mentioned. The constants are shown in Table III.

Table III

Measured resonant frequency in box, with back removed	f_1 = 62 cps
Measured resonant frequency in box with hard back in place	f_2 = 98 cps
Measured effective radius of the piston . .	r = 8.5 cm
Acoustic mass due to air-load, from equation (1)	L_{AP} = 0.032 x 10^{-3}
Acoustic resistance due to air-load, from equation (2)	R_{AP} = 0.19
Acoustic mass of the closed box, from equation (4)	L_{AB} = 0.028 x 10^{-3}
Acoustic compliance of the closed box, from equation (5)	C_{AB} = 20.4 x $10^3 \mu f$
Acoustic mass of the piston, from equation (7)	L_{AM} = 0.147 x 10^{-3}
Acoustic compliance of the piston, from equation (8)	C_{AM} = 31 x $10^3 \mu f$
Acoustic mass due to air-load upon damping screen, from equation (1)	L_A = 0.019 x 10^{-3} henrys
Acoustic resistance due to air-load upon damping screen, from equation (2)	R_A = 0.07 ohms

The equivalent circuit using these constants is shown in Fig. 6. The acoustic values are shown in parenthesis. Some of the acoustical elements have a much lower impedance than the electrical components available in the laboratory; in the equivalent circuit, therefore, the impedance of all the constants was multiplied by the factor 2000. The switch S_1 at position A portrays the action with the back removed, because the fluid from the back of the cone is able to enter the external medium. The switch in position B (open circuit) portrays the action with the solid back, since the fluid displaced by the cone is unable to enter the external medium. The switch at C causes the insertion of an adjustable resistance R_{AC} which portrays the acoustical damping resistance due to cloth when the damping back of the box is used. A simplifying assumption is made that the interaction between the back and the front radiation has no effect upon damping and that the air-load impedance at the back of the box remains the same with and without the damping resistance.

The optimum damping resistance was chosen by observing the transient response of the system. A steep pulse from a General Radio Strobotac was applied to a small (10 ohm) resistor "e" inserted in the circuit.

The Strobotac was adjusted to fire about 10 pulses per second. The transient potential developed across Z_{AP} was observed on the screen of an oscilloscope. (The circuit was rearranged to permit the single-ended Strobotac and Oscilloscope to be properly grounded.) Varying degrees of damping were obtained by adjusting R_{AC}. Some of the resulting transients are shown in Fig. 7. "A" is obtained with the switch at A to represent the open-box condition; "B" is obtained with the switch at B to represent the solid back condition; "C" is taken with the switch at C to represent the condition resulting from adjusting R_{AC} to 100 ohms. The corresponding acoustic resistance is 100/2000 = 0.05 ohms. If a greater degree of damping were desired, some of the remaining circuit elements would require alteration.

To provide 0.05 ohms acoustic damping in the loudspeaker enclosure, the damping cloth was mounted on a perforated metal screen with a total net area of 200 sq. cm. Cloth having a specific resistance of 10 acoustic ohms was used.

The actual acoustic performance of the system was tested with the aid of the Strobotac generator connected to the loudspeaker (Fig. 8). A high resistance was inserted in the circuit to simulate the action of a pentode output stage (without inverse feedback). The sound pressure generated immediately in front of the loudspeaker cone was detected by means of a Sound Level Meter Microphone. A flat amplifier was used to connect the microphone to the oscilloscope. Transients were observed with the open box (A), with solid back (B), and with damping back (C). The resulting transients are shown in Fig. 9. "A" is the open-box response; "B" is the response with the solid back attached; under these two conditions a hang-over tone is clearly heard together with the pulses. "C" is the response with the damped back attached. The pulses are heard clearly without the hang-over tone. These oscillograms and listening tests confirm the results predicted by the equivalent circuit.

V. ELECTRICAL VS. ACOUSTIC DAMPING

To compare the electrical damping with the acoustic damping, the following experiment was performed. The hard back was installed and the loudspeaker was connected to the pulse generator as before -- except that a small resistor was connected across the loudspeaker terminals to simulate the effect of low amplifier output impedance. The input was readjusted to compensate for the power loss in the resistor. The resulting transient is shown in Fig. 10A. This confirmed the fact that the action of the electrical damping is similar to that of the acoustic damping, although, in this instance, not quite as effective. Next, the solid back was again replaced with the damped back and the acoustic impedance was slightly adjusted. The resultant transient is shown in Fig. 10B. In this instance, combining acoustic damping with electrical damping resulted in a near aperiodic response which is considered by many as ideal for the reproduction of percussion sounds.

With regard to frequency response, acoustic damping generally should have an effect similar to that of electrical damping. Response may be affected, however, by the specific arrangement of acoustic damping. It is interesting, therefore, to compare response curves for the system which we have described under similar conditions of damping -- electrical vs. acoustic. The measuring set-up for this purpose is shown in Fig. 11. The response measurements were made on the axis at three feet from the cone, with a constant generator voltage and suitable series resistance to simulate the source impedance. The resulting curves are shown in Fig. 12. Only the low frequency is affected by damping, and, hence, the curves stop at 1000 cps. The solid curve is for the loudspeaker in the acoustically damped enclosure and high source impedance. The dash-curve is for the loudspeaker in the open back enclosure and a low source impedance. The dotted curve is for the loudspeaker in the enclosure with the solid back and also a low source impedance. It would appear that the damped-back curve might be chosen as the best compromise because it is the smoothest and the easiest to equalize. It must be reiterated that the system under study is not intended to qualify as a "high-fidelity" system, but rather to represent the conditions which might exist in a home radio or a public address system.

Of further interest is the performance of the acoustically damped loudspeaker as a function of the source impedance. The response curves for high and low impedance source are shown in Fig. 13, and they are quite alike. This may be accounted for by the fact that the voice-coil impedance remains fairly constant owing to the acoustic damping. This comparative immunity from source-impedance effects is not to be minimized, since it provides the audio designer with a degree of freedom in the choice of equipment.

VI. CONCLUSION

Transient response of loudspeakers and enclosures can be effectively controlled by acoustic damping. Furthermore, the response-frequency characteristic of the loudspeaker system need not be adversely affected, and it actually may be improved. Loudspeakers with acoustic damping may operate from high-impedance amplifiers without "hang-over". Performance characteristics become largely independent of the amplifier impedance. Acoustic damping may be designed in a straightforward manner by ascertaining the acoustical constants and using standard experimental techniques of equivalent circuit analysis. We conclude, therefore, that acoustic damping for loudspeakers merits far more serious consideration than it has had heretofore.

ACKNOWLEDGMENT

In the preparation of this paper, the author wishes to acknowledge with thanks the able assistance of Mr. Leo Rosenman of Shure Brothers, Inc., who has been of immeasurable help with the planning and execution of these experiments.

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(2) Salmon, V., "Coupling of the Speaker to the Output Stage", NEWSLETTER of the IRE-PGA, Vol. 3, No. 1, January, 1952, p.5.

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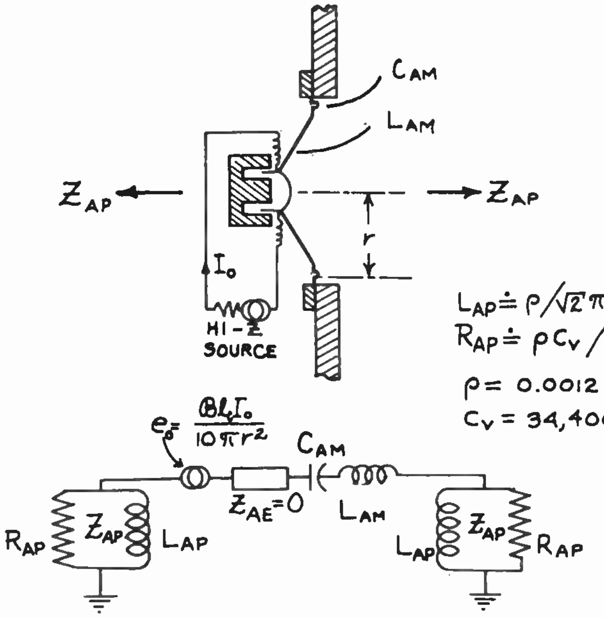
(7) For example, in text of reference (1), p. 89.

(8) For example, in text of reference (1), p. 87.



$$L_{AP} \doteq \rho/\sqrt{2}\pi r \qquad (1)$$

$$R_{AP} \doteq \rho c_v/\pi r^2 \qquad (2)$$

$\rho = 0.0012$ gm/cc

$c_v = 34{,}400$ cm/sec

$$(2\pi f_1)^2 (L_{AM} + 2L_{AP}) C_{AM} = 1 \qquad (3)$$

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

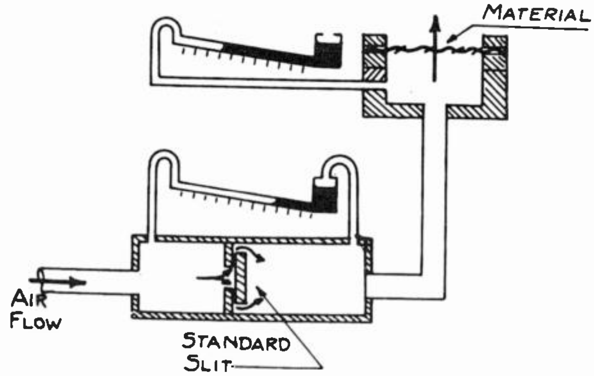
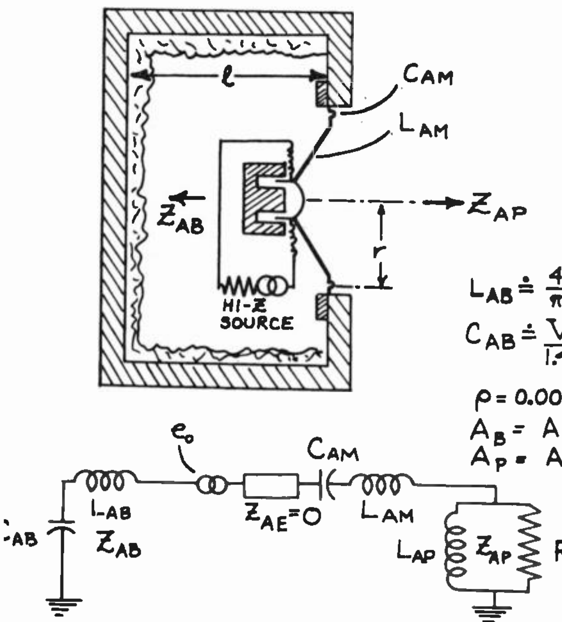


Fig. 3. Schematic arrangement of acoustical resistance meter



$$L_{AB} \doteq \frac{4}{\pi^2}\frac{\rho l}{A_B}\left(\frac{A_B}{A_P}\right)^{\frac{2}{3}} \qquad (4)$$

$$C_{AB} \doteq \frac{V}{1.41}\times 10^{-6} \qquad (5)$$

$\rho = 0.0012$ gm/cc

A_B = Area of Box, cm^2

A_P = Area of Piston, cm^2

l = Depth, cm

V = Volume of Box, cc

$$\left\{ \begin{array}{l} A_B/A_P \lesssim 10 \\ l \doteq 0.7\sqrt{A_B} \end{array} \right\}$$

$$(2\pi f_2)^2 (L_{AB} + L_{AM} + L_{AP}) \frac{C_{AM} C_{AB}}{C_{AM} + C_{AB}} = 1 \qquad (6)$$

Fig. 2. Equivalent circuit of a loudspeaker mounted in an enclosure

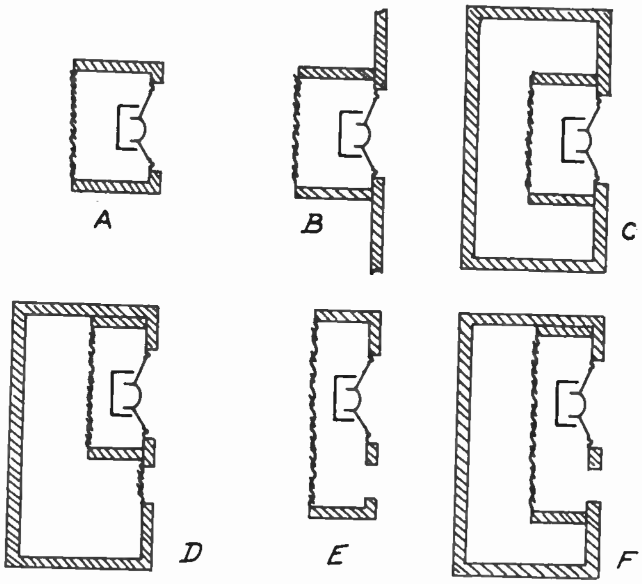


Fig. 4. Some of the means for applying acoustic damping to loudspeakers

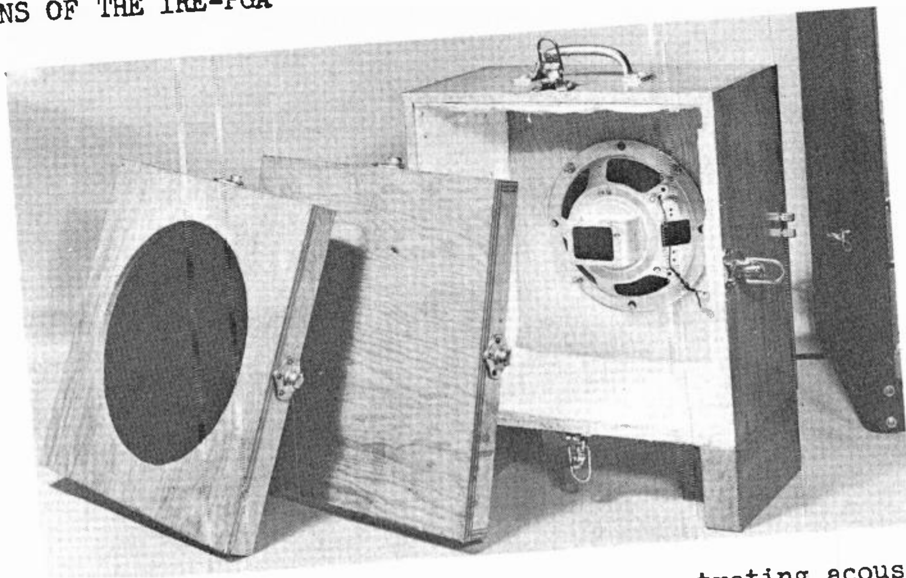


Fig. 5 Experimental loudspeaker system demonstrating acoustic damping

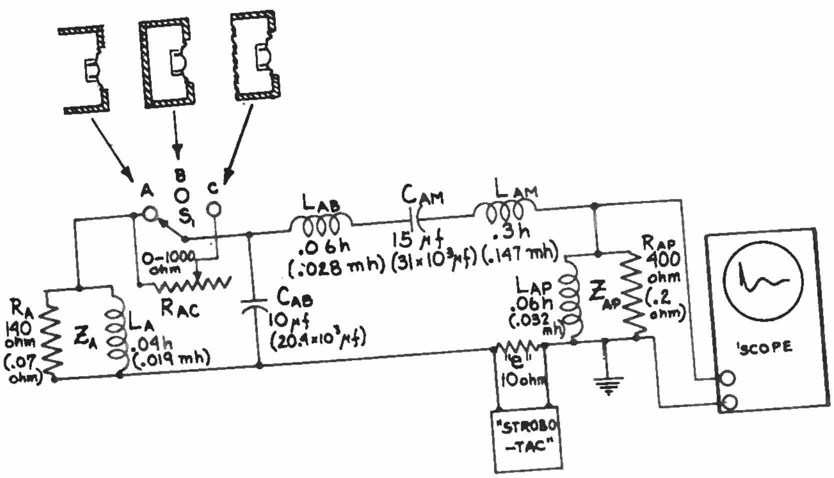


Fig. 6 Equivalent circuit of experimental loudspeaker system

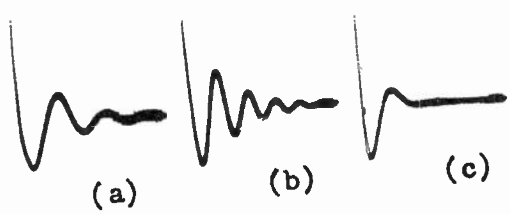


Fig. 7 Transients obtained with equivalent circuit of Fig. 6

(a) simulating open box
(b) simulating closed box
(c) simulating damped box

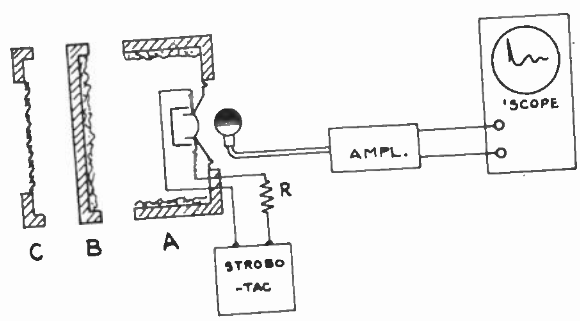
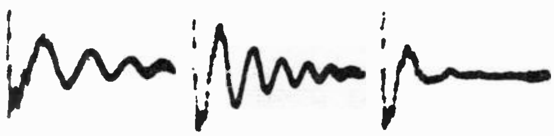
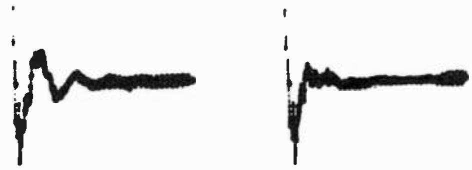


Fig. 8 Schematic arrangement for studying acoustical transients



(a) (b) (c)

Fig. 9 Transients obtained by acoustic measurements
(a) open box
(b) closed box
(c) damped box



(a) (b)

Fig. 10 Transients obtained by acoustic measurements
(a) closed box with electric damping
(b) acoustically damped box with electric damping

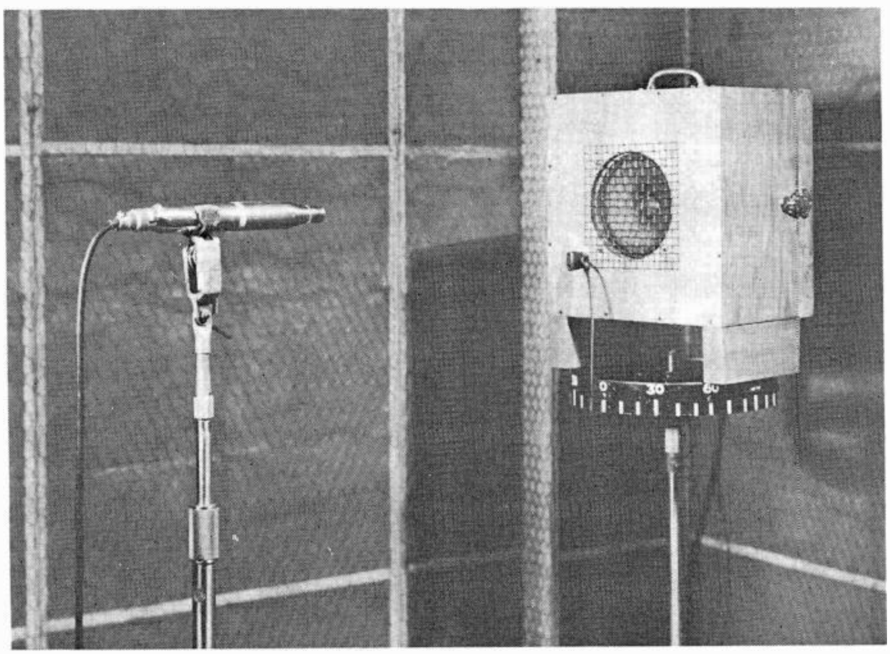


Fig. 11 Setup for frequency response measurements

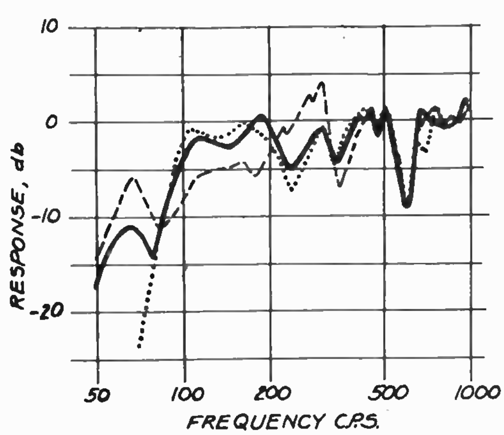


Fig. 12 Frequency response --
Dashed line - open box
Dotted line - closed box
Solid line - damped box

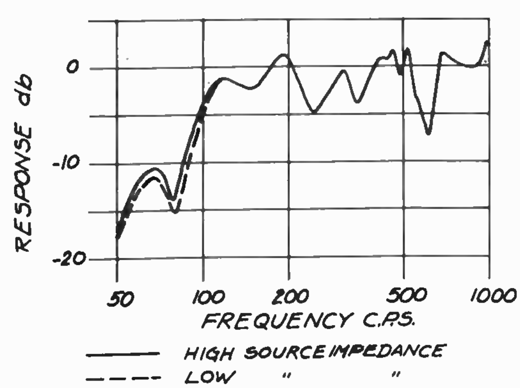


Fig. 13 Response of acoustically damped loudspeaker as a function of source impedance

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Professional Group on Audio

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

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IRE PROFESSIONAL GROUP ON AUDIO

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LOUDSPEAKER IMPEDANCE

Vincent Salmon
Stanford Research Institute
Stanford, California

This invited technical editorial by the Vice-Chairman of IRE-PGA treats a subject small in scope but of great importance of audio, in a comprehensive and yet fundamental manner. — *Editorial Committee*

INTRODUCTION

In most fields of specialization, the time soon comes when the meanings of terms must be standardized so that all workers may use a common technical language. The initial efforts of technical lexicographers usually run smack into the illogical but accepted usages which have grown up in each field. It then becomes necessary to define old, as well as additional new, terms so as to avoid ambiguity, and then to hope that these precisely defined terms will win some degree of acceptance. Such a situation now exists in the use of the term impedance. In this note we shall discuss it from the point of view of the loudspeaker, and shall comment on some of the proposed new terms and their usefulness. Most of the discussion applies specifically to moving coil loudspeakers, but the concepts are easily transferred to other types.

SPEAKER IMPEDANCE AND SPEAKER RATING IMPEDANCE

What do we do with a loudspeaker? Ordinarily, it is connected to a signal source for operating or for testing. In order to predict the performance during operation, it is necessary to know, among other things, the electrical impedance characteristics of both source and loudspeaker as a function of frequency, acoustical environment, and perhaps power level. On the other hand, engineering tests on loudspeakers are usually made under idealized conditions in which the test source is a constant voltage in series with a pure resistance. In order to relate the two situations, it is necessary to idealize the varying impedance characteristics of a loudspeaker to a single number, which somehow represents an averaged behavior. To get a clue as to how this is done, let us first discuss the operation of a loudspeaker under actual conditions of use.

When a loudspeaker is to be connected to an ordinary amplifier, the first question is usually, "To what terminals should it be connected?" Obviously, this implies an impedance which is a single figure, not the electrical impedance which varies with frequency. In essence, we want numbers on both amplifier output and speaker input terminals such that when terminals bearing the same number are connected together, then the combination will deliver *rated* performance. Accordingly, this magic single number may be called the *rating* impedance of the loudspeaker. The term "speaker impedance" by itself, we then take to be simply the complex electrical impedance, expressed in magnitude-angle or resistance-reactance form, as a function of frequency. What, then, is the relation between "speaker impedance" and "speaker rating impedance"?

The loudspeaker is a load on the driving amplifier, and as such should permit the amplifier to deliver to it the maximum power which is consistent with prescribed distortion limitations. In actuality, the speaker impedance and the spectrum of the audio signal are functions of frequency, and conditions are highly transient. If an ideal transformer of variable turns ratio were interposed between the amplifier and the loudspeaker, then it would be found that the optimum transformer setting for maximum distortion-limited power would depend on the type of signal. Extremes of signal types are represented by those at the output of electric organs and paging systems. The single value of rating impedance commonly quoted applies specifically to a wide range audio signal with spectrum maximizing between 200 and 800 cps. In actuality, the rating impedance is best expressed as a range rather than a single number, because the optimum is so broad. In practice a two-to-one range is acceptable; a loudspeaker with an 8-ohm rating impedance may be connected to amplifier outputs marked from 5.6 to 11.3 ohms. The performance at the impedance extremes will be degraded only at extremes of frequency and power.

AMPLIFIER OUTPUT IMPEDANCE

At this juncture, the term output impedance as applied to amplifiers deserves discussion. It is this writer's opinion that the term should never be used unqualified, since it refers to a position, and does not state in which direction the impedance is to be measured. Facing the amplifier, the amplifier internal output impedance is seen; and facing the load, the amplifier load impedance is seen.

ESTIMATING RATING IMPEDANCE

To return to loudspeakers, it is seen that the rating impedance may be obtained from the loudspeaker impedance if the signal spectrum is used to get a properly weighted average of the impedance-frequency characteristic. For ordinary audio signals, the following rule-of-thumb procedures may be followed for estimating the rating impedance for wide-range, moving-coil loudspeakers; all dc measurements are at the voice coil. In the first procedure, measure the dc resistance of the voice coil. The rating impedance will be about 20 per cent greater for direct-radiator loudspeakers, and about 40 per cent greater for horn loudspeakers. In a second procedure, ac measurements of the magnitude of the impedance are utilized. For direct-radiator loudspeakers, locate the minimum impedance in the 200-800 cps range and add 10 per cent. For horn loudspeakers, the rating impedance is the average value of the magnitude of the impedance taken near the center of the pass band. In view of the two-to-one range in the standard rating impedances of 4, 8, and 16 ohms, these estimates should be close enough. It must again be emphasized that these rough rules apply only to ordinary wide-range, moving-coil loudspeakers; they will be quite different, for example, for loudspeakers designed for battle announce systems.

SPEAKER TEST SOURCE IMPEDANCE AND SPEAKER INPUT

In testing loudspeakers, the rating impedance has another function in the calculation of the input to the loudspeaker. We note first that when an amplifier is properly loaded for maximum output at, say, 2 per cent harmonic distortion, the impedances of the driving source and of the load are rarely equal. Thus amplifiers usually do not operate with a matched load in the sense that the impedance at the output are the same in both directions. The appropriate measure of loudspeaker input is hence not the maximum power available from the source, but rather is the power the source can deliver to its rated load, within prescribed distortion limitations. In testing a loudspeaker we therefore use as a source an adjustable voltage (which is held constant during the test) in series with a fixed resistance which equals the amplifier internal output impedance in the mid-range; this is the test source impedance. To set the power available to the loudspeaker, replace the loudspeaker by a resistor of value equal to the rating impedance. Then vary the adjustable voltage until the desired power is absorbed by the rating impedance. When the loudspeaker is re-inserted into the test circuit, it will actually absorb a different power, but by keeping the source emf constant, conditions of use will be simulated in an idealized fashion.

REGULATION

It is a common tendency nowadays to employ so much negative voltage feedback in amplifiers so that the loudspeaker is driven from essentially a zero impedance source. Whether or not this is desirable is another matter, on which many words have been spilled but for which few facts have been marshaled. Here we shall merely introduce what seems to be a useful measure of the amplifier internal output impedance. First measure the open circuit voltage across the amplifier output, keeping within the linear region. Then connect the rated load across the terminals. The ensuing drop in voltage, expressed in decibels, is termed the regulation. For example, the regulation of a triode output stage is normally 3.5 db. For beam power tubes without feedback, 15 db is perhaps a lower limit. With feedback, values as low as 0.2 db have been stably attained. A matched source has a regulation of 6 db, whereas for a constant voltage source the regulation is zero db. The importance of the regulation is this: at extreme frequencies the loudspeaker impedance usually is very large compared with the rating impedance and hence the loudspeaker approaches an open-circuit load condition. The regulation then represents the maximum increase in voltage across the loudspeaker (and hence the rise in output), expressed in decibels to be expected as a consequence of the rise in impedance. Of course, the presence of reactive components in the actual source and load may somewhat alter this output increase, but the regulation is seen to furnish a good basis for estimating the behavior of a given loudspeaker with different output stages. Also, this points out the necessity for staging the source regulation in giving results of loudspeaker tests, since it is one of the important factors to be considered in comparing results of tests on different loudspeakers.

IMPEDANCE DISTORTION

Amplifiers are commonly tested and rated with a resistive load. Thus, the path of operation in the equivalent plate current-plate voltage family of tube curves is a straight line, assuming a perfect output transformer. However, long before the effect of finite transformer primary inductance is felt at low frequencies, the speaker will have become a load having considerable reactance. Consequently, the path of operation will become an approximate ellipse instead of a straight line. Because this ellipse will extend into regions where the tube characteristics are not so well behaved, there will result a species of distortion for which the term "impedance-distortion" is suggested. In practice it is almost certain that for well designed and stable amplifiers this distortion will be of minor consequence, owing to the vastly greater amount arising from mechanical nonlinearity and magnetic field inhomogeneity in the loudspeaker. Still, it might be instructive to construct a dummy load simulating the loudspeaker for use in amplifier measurements at low frequencies.

CONCLUSIONS

Five impedances are associated with the operation and testing of loudspeakers: the complex speaker impedance; the rating impedance; the test source impedance; the amplifier internal output impedance; and the amplifier load impedance. From these values it is possible to make the proper connections for operation and for testing, and to calculate the power available to the loudspeaker. The rating impedance is not a very sharply defined quantity, and it may be estimated in a relatively simple manner in most cases of interest. The regulation of the test source should always be stated in giving results of tests; it aids in estimating loudspeaker performance at extremes of frequency.

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PGA COMMITTEE APPOINTMENTS

It is a pleasure to announce the following committee appointments:

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We invite any interested PGA members to help these committees. Please communicate with the appropriate chairman or with me.

Marvin Camras
Chairman, IRE-PGA

NEW IRE–PGA CHAPTERS FORMING

Welcome to the new San Diego Chapter of IRE-PGA! Although there was a news item concerning the possible formation of a chapter in San Diego several issues back, the San Diego Chapter is now "official", and we hope to have some news from them in a future issue.

A petition for a new chapter in Houston has been received, and may be acted upon before this issue of *Transactions* of the IRE-PGA is printed.

PGA now has chapters in the following cities:

Albuquerque	Kansas City
Boston	Los Angeles
Chicago	Milwaukee
Cincinnati	Philadelphia
Detroit	Seattle
San Diego	

If the IRE Section in your vicinity has a group of members (or potential members) interested in audio, please communicate with Robert E. Troxel, Chairman of the Committee on Chapters, (Shure Brothers, Inc., 225 W. Huron St., Chicago 10, Illinois).

IRE–PGA AT 1953 NATIONAL ELECTRONICS CONFERENCE

Once again the IRE-PGA is cooperating with the National Electronic Conference by organizing an audio program for the 1953 NEC meeting in Chicago, September 28, 29 and 30. This session has regularly been the fall meeting of the Professional Group on Audio for several years. Mr. R. J. Tinkham has been appointed by Marvin Camras as Chairman of the Program Committee for IRE-PGA.

Another excellent program is expected. Make your plans to attend.

ROOM ACOUSTICS*

Hale J. Sabine
The Celotex Corporation
Chicago, Illinois

This basic summary of room acoustics was prepared by Dr. Sabine specifically for radio engineers. The active discussion which it stimulated at the national IRE Convention indicated its potential general interest to the membership of IRE-PGA.—*Editorial Committee.*

The foregoing papers have described the action of various electrical, mechanical, and electroacoustic links which may form a more or less complex chain between a source of sound and its reception by the ear. The present discussion will be concerned with the transmission of sound through the air in a room and the ways in which the signal may be modified by the acoustical characteristics of the room. In the cases to be discussed, the source of sound in the room may be a live voice or musical instrument, or may be reproduced by a loudspeaker. It may be received either by the ear or by a microphone for further transmission.

As the simplest possible case, we may consider transmission in a room in which reflection of sound waves is essentially nonexistent. This would be equivalent to the open air, or a so-called "free field". Under these conditions the sound signal is transmitted from source to receiver with no modifications except for simple attenuation due to the decrease in amplitude of the wave as it spreads out from the source. As given by the inverse square law, this attenuation amounts to 6 decibels for each successive doubling of distance from the source, or 20 db per ten-fold increase of distance, and is essentially independent of frequency. Over large distances, high audio frequencies are attenuated somewhat more than low frequencies, due to absorption of sound energy by the air itself. The amount depends on frequency and relative humidity. A 10,000 cycle tone is attenuated 9 db per hundred feet of travel at 20 per cent relative humidity, and 4 db at 50 per cent. In any room this directly transmitted sound is always present as a component of the total sound field, and its intensity or energy density at the receiver depends only on the acoustic power output and directivity of the source and on the distance of the receiver from the source. The energy density of the direct sound in ergs per cubic centimeter is given by

$$E_D = \frac{W}{4\pi D^2 c} \cdot \frac{4\pi}{\Omega_1} . \qquad (1)$$

where W = acoustic power output of source,
D = distance from source,
c = velocity of sound, and
Ω_1 = effective solid angle of radiation of source, in steradians.

For a nondirectional source,

$$E_D = \frac{W}{4\pi D^2 c} \qquad (2)$$

In a typical room sound waves are reflected back and forth many times between the walls, floor, and ceiling, with some loss of energy by absorption at each reflection as shown in Figures 1 and 2. If a steady source is maintained, the rapid accumulation of reflected waves quickly sets up a level of sound energy which is essentially uniform throughout the room and remains constant until the source is cut off. This steady state level is determined by the point at which the rate of energy loss by absorption at the room surfaces equals the rate at which sound energy is supplied by the source. From this it can be shown that the steady state energy density is directly proportional

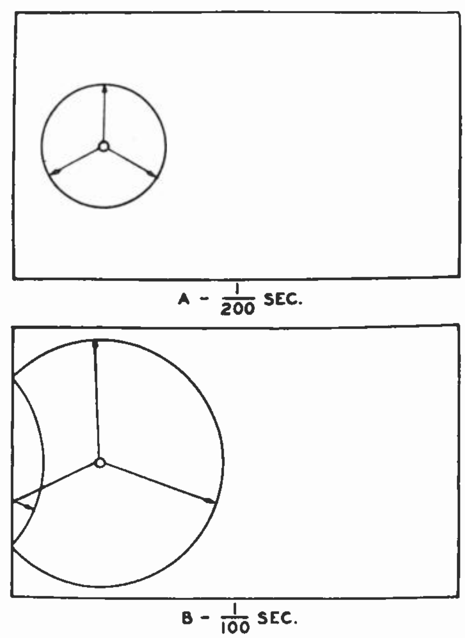


Fig. 1 – Multiple reflection of a single wave front in a closed room.

*Presented as part of the Seminar, "Acoustics for the Radio Engineer", organized by the IRE Professional Group on Audio, IRE National Convention, New York, N.Y., March 25, 1953. Manuscript received June 3, 1953.

to the acoustic power output of the source and inversely proportional to the room absorption, which depends on the area and average sound absorptivity of the room surfaces and furnishings. The value of the reflected energy density is given approximately by

$$E_R = \frac{4W}{ac} \tag{3}$$

The room absorption a, in *sabins*, is the figure obtained by multiplying the area in square feet of each room surface by its sound absorption coefficient, or *absorptivity*, taking the sum of these products, and adding in the absorption supplied by furnishings, seats, and

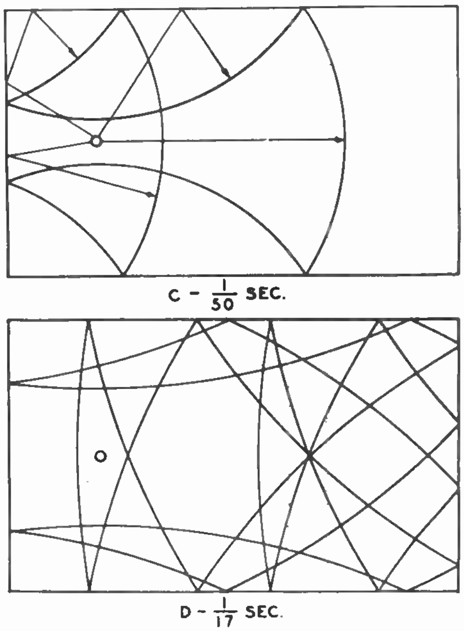


Fig. 2 – Multiple reflection of a single wave front in a closed room.

persons in the room. One sabin, or absorption unit, is defined as the equivalent of one square foot of perfectly absorptive surface, such as an open window. The absorption coefficient of a surface is defined as the per cent of incident sound energy absorbed by the surface. Sound waves are assumed to strike the surface equally from all possible angles. Ordinary hard interior finish materials such as concrete, plaster, wood, glass, etc., are excellent sound reflectors, having absorption coefficients of only 1 to 5 per cent. Carpets and drapes are more or less effective absorbers, depending on frequency. Upholstered seats have absorptions of 1 to 4 sabins each, and a seated person has an absorption of about 4 sabins. Architectural acoustical materials, designed as sound absorbent interior surfaces, have coefficients which may approach 100 per cent at certain frequencies, and in general have coefficients, averaged over the frequency range, of 50 to 90 per cent, depending primarily on thickness.

Sound absorption is fundamentally the transformation of the vibratory energy of air molecules in a sound wave to the random molecular motion of heat energy. This can only take place through friction or viscosity; that is, by *damping*. In the case of porous material, which is the most common type of sound absorber, a sound wave readily enters the material, where the vibratory energy is dissipated by friction of the air molecules against the pore walls or fibers of the material. This results in the reflection of a wave with diminished sound energy, as shown in Figure 3. Sound can also be absorbed by impervious diaphragms, such as thin plywood over an air space. The two basic requirements for effective absorption in this manner are, first, that the diaphragm be light enough and backed by a deep enough air space that it can move relatively freely under incident sound pressure, and second, that there be a frictional or viscous element of the proper value in the system for developing the required transformation to heat energy. This element can be either the internal bending viscosity of the diaphragm or a porous material in the air space.

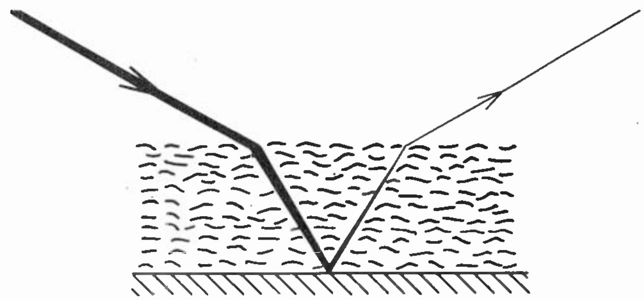
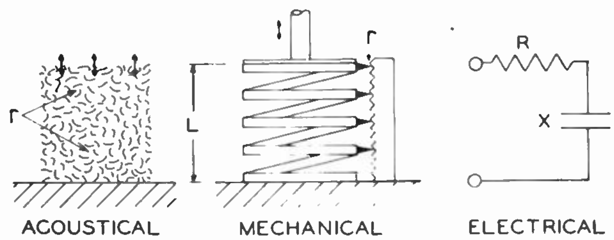


Fig. 3 – Absorption and reflection of sound by a porous material.

The electrical analog of sound absorption is a transmission line having a purely resistive characteristic impedance, representing the air outside the material, terminated by a complex impedance, representing the surface of the material. If the electrical reflection coefficient R of such a line termination is expressed in terms of a power absorption coefficient, $\alpha = 1 - R^2$, its value will be numerically equal to the absorption coefficient of the corresponding acoustical impedance (for normal wave incidence). The simplest type of sound absorber is a porous material on a rigid backing. The acoustic impedance of the surface of such a material is essentially the analog of a resistance and a capacitive reactance in series, as shown in Fig. 4. The resistance corresponds to the frictional resistance offered by the material to the penetration of air, and the capacitive reactance to the stiffness reactance of the air volume contained between the outer surface and the rigid backing of the material. Covering the material with a perforated facing is equivalent to adding a series inductive reactance to the equivalent electrical circuit, and causes an absorption peak at the resonant frequency. (See Figure 5.)



$$R = 1/3\, r L$$

$$X = -\frac{\rho c^2}{2\pi f P L}$$

Fig. 4 – Mechanical and electrical analogs of sound absorption by a homogeneous porous material of thickness L, porosity P, and specific flow resistance r. (H. J. Sabine, "Sound absorption and impedance of acoustical materials," *Jour. Soc. Mot. Pic. Eng.)* vol. 49, no. 3, pp. 262-278; September, 1947. Reproduced by permission.)

Complete transfer of power to the load in the electrical circuit, corresponding to complete sound absorption, will take place only when the load impedance is a pure resistance equal to the characteristic line impe-

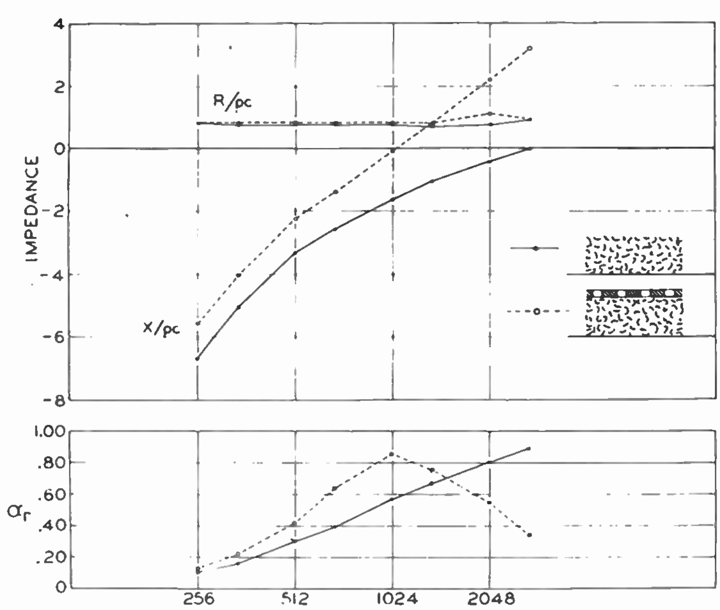


Fig. 5 – Acoustic impedance and absorption coefficient of one-inch porous material with and without perforated facing. (Reproduced by permission from same source as Fig. 4.)

dance. Since the impedance of acoustical materials in general contains a reactive component, as they are normally manufactured and installed, their absorption will vary with frequency.

Neither the ear nor a microphone responds directly to sound energy, but instead to sound pressure (or particle velocity in the case of a velocity microphone), therefore, it is important to know how sound pressure is distributed with a steady source being maintained. The total sound pressure at a given point depends on the vector summation of the random pressure amplitudes and phases of all of the reflected waves passing through that point. The pressure will therefore fluctuate markedly from point to point in a stationary series of peaks and dips termed a "standing wave pattern" or "interference pattern". Successive peaks or dips when excited by a single frequency source are spaced roughly one-half wavelength apart in any direction, the spacing in feet being approximately equal to 560 divided by the frequency. The difference in pressure level between maxima and minima may be as much as 20 or 30 db, but with increasing absorptivity and irregularity of the room surfaces the pattern becomes much less pronounced and also less regular. Interference peaks and dips may be easily heard in any room in which a steady, moderately high-frequency tone is sounding, by stopping one ear and moving the head slowly over a distance of a few feet. A sound level meter will register similarly.

The dips in a standing wave pattern are sometimes referred to as "dead spots", a term which is misleading as far as over-all room acoustics is concerned. For every frequency which would produce a pressure minimum at any given point, there is on the average another frequency which would produce a maximum at the same point. Normal sound sources, such as speech or musical instruments, produce many frequencies at once which are constantly changing. Interference effects therefore tend to average themselves out under actual conditions and there is no evidence of fixed dead spots attributable to interference *per se*. This is shown very clearly in Figure 6,

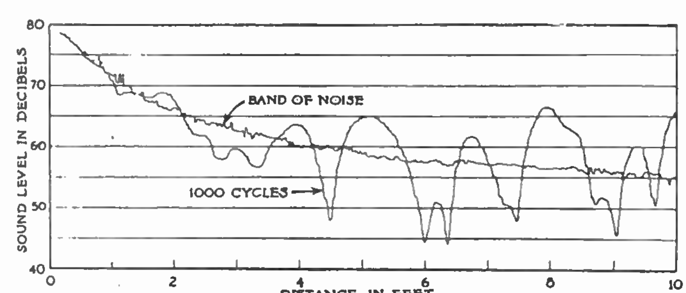


Fig. 6 – Sound pressure level vs. distance from a loudspeaker in a room (Reproduced by permission from V. O. Knudsen and C. M. Harris, "Acoustical Designing in Architecture," John Wiley and Sons, Inc., New York, N.Y., 1950; p. 138.)

in which the interference pattern is virtually eliminated by substituting a continuous band of frequencies one octave wide for a single frequency tone. This figure incidentally illustrates why a band of frequencies rather than a single fixed tone should be used in measuring the average sound pressure level in a room. Interference effects are further minimized by binaural listening, since the chances are against pressure minima coinciding with the positions of both ears. As a possible exception to the above statements, it may be noted that at very low frequencies of less than 100 cycles, where wavelengths are 10 feet or more and interference patterns correspondingly broad, definite areas of poor response to certain bass tones may occur, particularly in small rooms.

Since any change in frequency will cause a change in the spacing of the interference pattern, it can be seen that the sound pressure at a fixed point, due to the reflected wave system, will fluctuate with frequency. Each peak in the pressure-frequency curve is due to an individual resonant response or so-called "normal mode of vibration" of the volume of air enclosed in the room. Roughly, a resonance peak may be thought of as occurring whenever a reflected wave after traveling once around a room (or between two opposite surfaces) is exactly in phase with itself at its starting point.

The response curve of a typical room of living room or studio size and moderate absorptivity is shown in Figure 7. The fluctuations shown are due largely to room resonances, and only secondarily to variation of loudspeaker output. At low frequencies the peaks are clearly defined and quite broad, but at higher frequencies they become more and more closely spaced. In a larger room the low frequency peaks would also be more closely spaced. Since the peaks are caused by reflection, it follows that increasing the absorptivity of the room surfaces should smooth out the response curve. This occurs to a marked degree, as shown by Figure 8, in which the room absorption is increased four times.

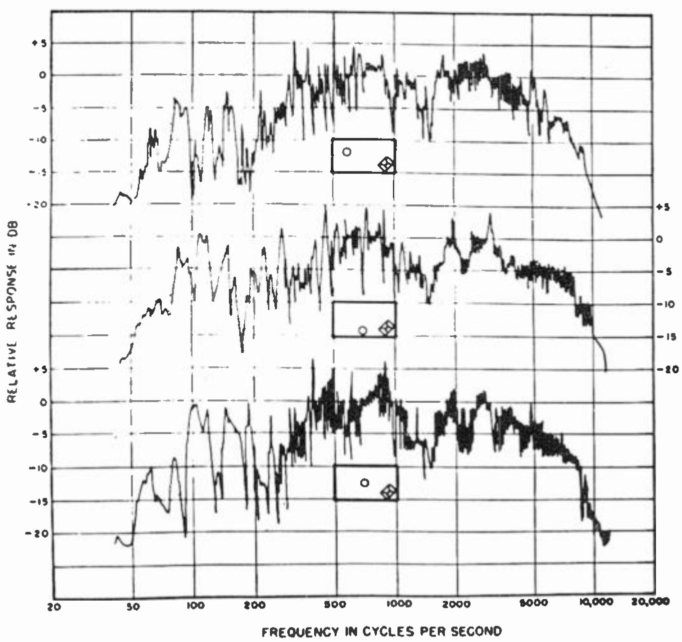


Fig. 7 – Variation of sound pressure level with frequency at a fixed point in a room. (Reproduced by permission from Jensen Technical Monograph No. 1, "Loud Speaker Frequency-Response Measurements," published by Jensen Manufacturing Co., Chicago, Illinois.)

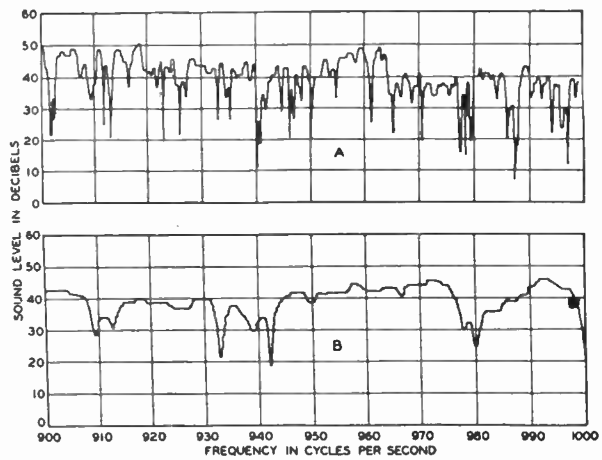


Fig. 8 – Variation of sound pressure level with frequency in a room with low absorption *A* and high absorption *B*. (E. C. Wente, "The characteristics of sound transmission in rooms," *Jour. Acous. Soc. Amer.*) vol. 7, p. 123; October, 1935. Reproduced by permission.)

The frequency irregularity of a room can be easily heard if one ear is closed and a pure tone is varied slowly in frequency. However, in listening to speech and music the frequency peaks are so uniform and so closely spaced that they are not distinguishable as such by the ear. Exception must be made again to the case of low frequencies in small rooms where the irregularities are so broad that they can be readily detected in normal listening or microphone pickup. For example, successive pedal notes of an electronic organ might differ noticeably in audible level.

As far as hearing conditions are concerned, then, we can largely ignore both the spatial and frequency fluctuations of sound pressure due to interference effects, and consider only the average sound pressure which, when squared, will be a direct measure of the reflected sound energy density. As pointed out previously, its steady state level is uniform throughout the room (assuming a generally rectangular shape). The total energy density at the point of pickup with a steady source sounding is the numerical sum of the energy density of the reflected sound and that of the directly transmitted sound. The ratio of direct to reflected sound energy density is, approximately,

$$\frac{E_D}{E_R} = \frac{a}{16\,\pi D^2} \qquad (4)$$

where a = room absorption in sabins, and
D = distance from source in feet.
There difference in level in decibels is

$$L_D - L_R = 10 \log_{10} \frac{a}{16\,\pi D^2} \qquad (5)$$

If either the source or the receiver, or both, has a directional radiation or response characteristic, the foregoing expression is modified to

$$L_D - L_R = 10 \log_{10} \frac{a}{16\,\pi D^2} + 10 \log_{10} \frac{4\pi}{\Omega_1} + 10 \log_{10} \frac{4\pi}{\Omega_2} \quad (6)$$

where Ω_1 and Ω_2 are the effective solid angles of radiation and response of the source and receiver, respectively, in steradians.

This relation (Equation 5) is plotted in Figure 9 as a function of distance from the source for the indicated values of room absorption. Both source and receiver are assumed to be nondirectional. If either one were directional, the direct sound in relation to reflected would be correspondingly higher, by the amounts given in (6).

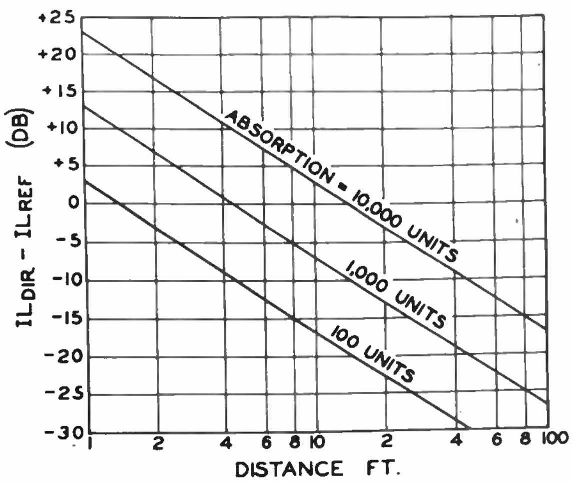


Fig. 9 – Level of direct sound (IL_{dir}) above or below generally reflected sound (IL_{ref}) in relation to distance from source and number of absorption units in the room.

Figure 9 can be used to estimate at what distance from the source the reflected sound will become the major component in rooms of various sizes. If it is assumed that the direct component becomes negligible when it falls to 10 db below the reflected sound, it will be seen that this occurs at only about 5 feet from the source in a room with 100 sabins absorption. This would be typical of an average living room. In a room with 1000 sabins, which would be a small auditorium with an audience of 150 to 200, or a 25 by 40 foot radio studio, the reflected sound would become predominant at about 15 feet from the source. A room with 10,000 sabins would be a fairly large auditorium or concert hall filled to a capacity of 1500 to 2000 people. In this case the reflected sound would not become predominant until about 50 feet from the source.

If either the source or pickup is directional, the above distances will be increased by a factor of $\sqrt{4\pi/\Omega}$. This is equivalent to saying that the effect of a directional source or pickup is the same as moving closer to the source.

One important application of these figures is in determining what effect the absorption-frequency characteristic of the room will have on the sound transmitted to various receiving positions. At points far enough from the source that the total level is due mostly to reflected sound, the frequency characteristic will be essentially that of the room absorption. Close to the source, the variation of absorption with frequency will have very little effect on the transmission-frequency characteristic. The equation relating the total level, in decibels, to distance and room absorption, assuming non-directional source and receiver, is

$$L_T = L_D + 10 \log_{10} \left(1 + \frac{16\,\pi D^2}{a}\right) \qquad (7)$$

Since the direct sound L_D is transmitted without frequency discrimination, this equation can be used to determine the effect of variation of absorption a with frequency on the total level for various distances D. In a well-designed room the overall variation in absorption over the audible frequency range will be of the order to 2 to 1, which is a deviation of only 3 db from a flat room response curve. The effect of non-uniform absorption can be further reduced by a directional source or receiver.

Loudspeakers are used as sound sources principally to reinforce the natural voice in large auditoriums, and to reproduce recorded sound, as in motion picture theatres. It is generally desirable that the amplified sound be as close to a uniform level as possible over the seating area. This can be accomplished either by distributing a number of low level speakers around the room so as to blanket the seating area, or by utilizing the directional pattern of a single loudspeaker near the stage. As illustrated in Figure 10, the speaker is oriented so that the drop in level away from the axis is compensated by the closer distances.

The entire discussion has thus far been confined to transmission through the room of sound from a steady, continuous source which does not vary in strength, frequency, or frequency distribution, such as a sustained note from a musical instrument without tremolo or vibrato. The effects of room characteristics on the audible tone quality of such a sustained sound are actually quite minor and difficult to detect, and are apparent only in subtle differences in the harmonic structure of a musical tone due to the interference pattern, and possibly to high frequency attenuation due to air absorption or non-uniform room absorption. The sounds of music, and especially of speech, however, are in general constantly changing and of relatively short duration. The transient responses of the room to these changes have a very marked and unmistakable effect on sound quality and hearing conditions.

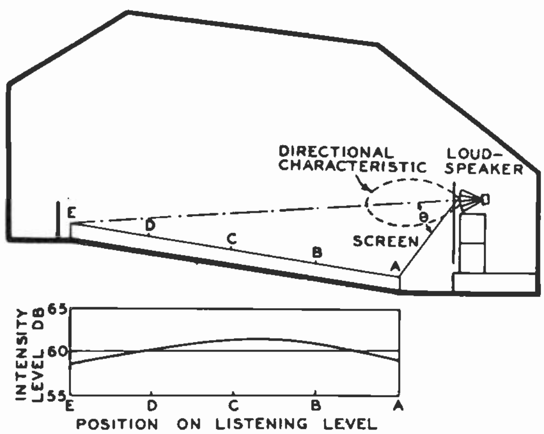


Fig. 10 – Arrangement of the loud speaker for a sound motion picture reproducing system in a theater. (Reproduced by permission from H. F. Olson, "Elements of Acoustical Engineering," D. Van Nostrand Company, Inc., New York, N.Y., 1947; p. 409.)

The most important index of the transient response of a room is its reverberation time. Reverberation is defined as the prolongation and dying out of sound energy in a room after the source is stopped, due to the continued travel and absorption of reflected waves. The sound energy decay follows as exponential curve, which when plotted on a decibel scale against time is a straight line. The reverberation time T of a room is defined as the number of seconds required for a decay of 60 decibels. It is a definite acoustical property of the room which depends only on its volume V in cubic feet and total absorption a in sabins, in accordance with the formula

$$T = \frac{0.05\, V}{a} \tag{8}$$

Since the absorption normally varies with frequency, the reverberation time will be frequency dependent in inverse ratio.

It is important to note that the reverberant sound decay starts from the level of the reflected sound and not necessarily from the total level, as illustrated in Figure 11. That is, if the receiver position is close enough to the source that the direct sound predominates with the source on, there will be an instantaneous drop to the reflected sound level when the source is stopped, after which the decay takes place at the rate given by the reverberation time. At more distant positions where the direct sound is much less than the reflected, the decay starts from the total level with no discernible sudden drop when the source is cut off. The extent of this initial drop-off of direct sound is probably the most important factor in determining "acoustical perspective", or the ability to estimate the distance of a sound source from the point of pickup by listening alone, either directly or through a microphone and loudspeaker. As pointed out

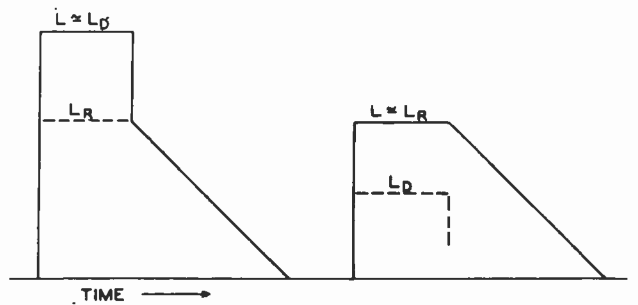


Fig. 11 – Diagram showing that reverberant sound decay starts from level of reflected sound (L_R) and not from level of direct sound (L_D).

above, a directional source or microphone will cause the distance to be estimated as less than it actually is. It is this same effect which makes it possible for directional loudspeakers to partially overcome the effects of excessive reverberation on speech intelligibility. Good results have been obtained in this manner in very large cathedrals, where tremendous volumes and lack of available treatment areas make it impossible to control reverberation sufficiently with acoustical materials.

The term "liveness" has been used to denote the overall subjective effect of reflected sound and reverberation in altering the quality of the directly transmitted sound, and numerous attempts have been made to attach a numerical figure to this property. Maxfield and Albersheim have proposed a "liveness constant" for a given room and pickup location which is proportional to the reverberation time and to the ratio of reflected to direct sound at that location. It is open to question, however, as to whether both of these factors can be properly combined into a single constant. For example, in a room having a four-second reverberation time, a point could be found close enough to the source that the ratio of reflected to direct energy would be 1/4. At this point, therefore, there would be a drop in level of 6 db before the start of the slow decay given by the four-second reverberation time. In another room having a one-second reverberation time, a point could be found distant enough from the source that the reflected energy would equal the direct; that is, their ratio would be 1. In this case there would be a 3 db drop when the source was stopped, but this would be hardly perceptible when followed by the rapid one-second decay time. The liveness constant would be the same for the two cases, but there would be a marked difference in the ear's judgment of room effect. It is especially important to consider the initial drop and the following decay rate separately when adding reverberation artificially to program material either by an echo chamber or electronically.

Generally speaking, the reverberation time must be held within certain limits in order to prevent confusion and overlapping of successive speech sounds, with resulting loss of intelligibility, and to impart a pleasing quality to the rendition of music. The choice of reverberation time representing optimum hearing conditions has been covered extensively in the literature and in standard texts on architectural acoustics, and no detailed discussion will be attempted here. It can be stated that acceptable reverberation times may range from 0.5 to 3 seconds, depending on frequency, room size, the type of sound, that is, whether speech, music, live, or reproduced, and on whether pickup is by the two ears (binaural) or by a microphone (monaural). See Figure 12. It can also be stated that deviations of plus or minus 20 per cent from recommended values will cause no serious harm to the quality of hearing conditions.

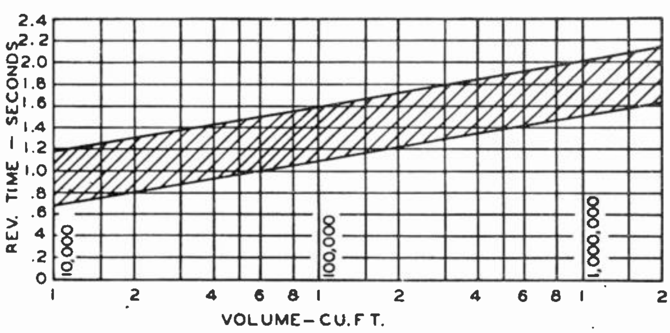


Fig. 12 – Range of acceptable reverberation times at a frequency of 500 cycles for auditoriums of various sizes. Range includes various uses of room. (Reproduced by courtesy of Acoustical Materials Association).

The effect of the frequency variation of reverberation time on hearing conditions may require some explanation. A reverberation curve which is flat above 1000 cycles and gradually rising to about twice the 100 cycle value at 50 cycles is generally considered acceptable. This curve can be justified theoretically on the basis that it causes all frequency components of a complex sound to maintain the same relative loudness to each other throughout the decay. Furthermore, such a curve is fairly typical of most ordinary listening rooms and auditoriums, and can therefore be considered as representing "naturalness" in room quality. Deviations from this ideal curve are objectionable only insofar as they produce "unnatural" sound quality. As a rule, the most readily observed frequency irregularities are excessively long reverberation times at the extreme low and high frequencies, which produce room responses which can be described as "boomy" and "hissy", respectively. Differences in decay rate with frequency are also more easily heard when the average reverberation time is short than when it is long. As a practical matter, reverberation-frequency characteristics cannot always be completely controlled, as for instance in an auditorium with full audience, where the curve is largely fixed by the absorption of the people, and at very high frequencies by the absorption of the air. In broadcasting or recording studios, on the other hand, the reverberation-frequency curve can be adjusted as desired by proper choice of acoustical treatment.

Reverberation as heard by the ear or a microphone has thus far been considered as a smooth, continuous trail of sound dying out to inaudibility. Actually, the decay curve of sound pressure level fluctuates more or less widely about an average straight line, due to the interference pattern. The amount of fluctuation is much less for a mixture of frequencies than for a single frequency tone, and is also reduced by irregular room surfaces, where the dimensions of the irregularities are generally comparable to the wavelength.

A certain amount of fluctuation in decay is common to all rooms and is therefore considered desirable on the basis of "naturalness". Artificial reverberation should therefore provide for some decay irregularity.

Another type of transient room response is associated with the sound which reaches the ear or microphone after only one or two reflections, as shown in Figure 13. These

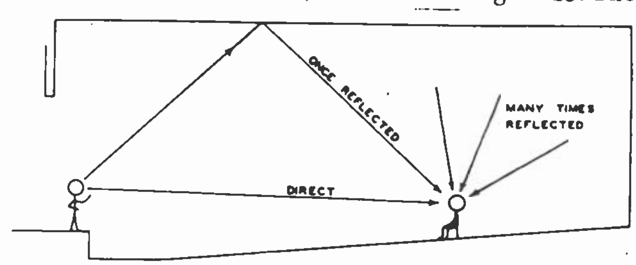


Fig. 13 – Components of sound received by a listener in an auditorium.

arrive immediately after the direct sound, and since they are at a high level and account for comparatively large changes in level at the beginning of the general decay, they may strongly affect the quality of hearing conditions. Probably the most familiar of these effects is an echo, which is defined as a single reflection which can be heard as a distinct repetition of the direct sound. In order for a single reflection to be heard as an echo, it is necessary first that the reflected sound arrive at the ear or microphone at least 1/17th second later than the direct sound, or in other words, that it travel a path from source to listener at least 65 feet longer than the path traversed by the direct sound. If the time lag is shorter than this, the reflection cannot be distinguished by ear, and will instead act as a reinforcement of the direct sound. Secondly, echoes are audible only if the reverberation time is short enough that the sound of the echo is not obscured by the general reverberation. Echoes in auditoriums are most often caused by rear walls and excessively high ceilings, since it is these surfaces which make possible the longest path difference between direct and reflected sound when the source is on the stage.

When caused by flat surfaces, echoes are considered undesirable insofar as they cause a distraction to attentive listening. Concave room surfaces, however, such as curved rear walls or domed ceilings, produce converging or focused reflections which result in greatly intensified echoes. Under the worst conditions, namely, where the ear or microphone is close to a strong sound focus, as shown in Figure 14, and the room is large enough to allow sufficient time lag between direct and reflected sound, echoes may be so severe that speech is unintelligible and listening to music is intolerable. To avoid such difficulties, extended curved surfaces should be laid out so that their centers are far removed from any sound pickup position (see Figure 15), or else should be broken up by large irregularities having dimensions of several feet. Heavy acoustical treatment is helpful, but not completely effective in severe cases.

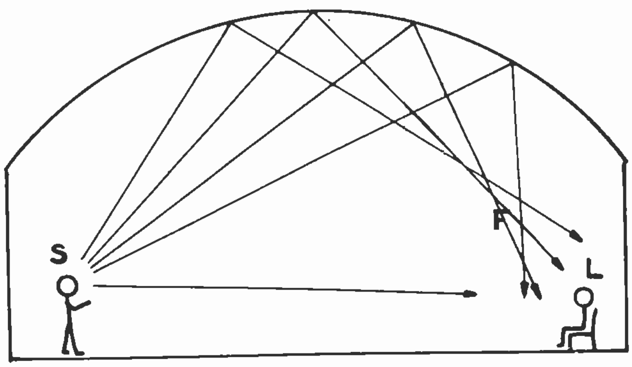


Fig. 14 – Diagram showing focused reflection from a concave ceiling.

In designing an auditorium, the shaping of room surfaces and the location and distribution of the acoustical treatment required for reverberation control can be utilized to some extent to take advantage of the first reflections. In general, absorption should be placed on those surfaces which produce the longest delayed first reflections to most listening positions. Such surfaces would be rear walls and excessively high ceilings. In theatres and auditoriums where direct or reproduced speech is at least partly involved, it is desirable to shape the walls and ceilings near the stage so that, as far as possible, first reflections will arrive at each listening position with a path difference of less than about 65 feet and from a direction close to that of the direct sound. These measures result in a marked reinforcement of the direct sound and a heightening of the auditory impression that all of the sound is coming from the stage. The latter effect is especially important in sound motion picture presentation. Obviously, surfaces which provide useful first reflections should not be acoustically treated, or at most lightly and in partial areas.

It is particularly important in a concert hall or scoring stage that the performers be surrounded with reflective surfaces on the stage itself. This is vitally necessary in permitting the conductor as well as each member to hear the group as a whole with the proper blending and balance.

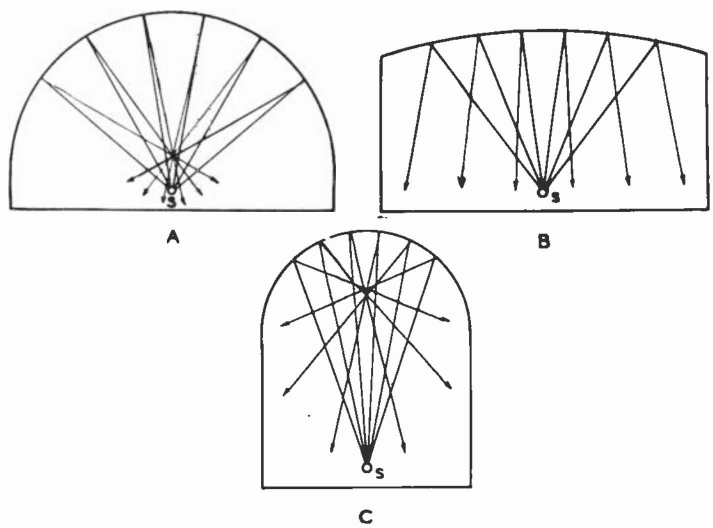


Fig. 15 – Reflection from concave ceilings of various radii. Center of curvature near floor line *A* produces most harmful effects. Conditions *B* and *C* are relatively harmless.

In auditoriums using loudspeakers for reinforcement of live speech, heavy rear wall absorption is of considerable help in preventing acoustic feedback to the microphone, and consequent "howling".

Control of first reflections is effective only in fairly large rooms, such as recording and audience broadcast studios and auditoriums seating at least a few hundred people. In smaller rooms, such as living rooms and small studios, all reflections arrive in such rapid succession that special control of first reflections produces no effects that can be readily appreciated by the ear.

The phenomenon of flutter echo should also be mentioned in connection with room design. This is a multiple reflection back and forth along a single line between two perfectly parallel reflecting surfaces. It can only be detected when the source and pickup are both on the same line perpendicular to the surfaces, and then only when the general room reverberation is short. As the name suggests, it is heard as a buzzing or ringing effect which is undesirable to the extent that it is disturbing or out of place. A flutter can also occur between two diagonally opposite room corners. It can be eliminated by putting the surfaces even very slightly out of parallel, or by making one or both of them absorptive. Serious trouble due to flutter is apt to occur only in small broadcast studios where the foregoing precautions have not been observed.

It is hoped that this brief outline of the essential principles of room acoustics will be helpful to those interested in this field. Reference to the literature and texts as well as personal experimentation and critical listening will establish a background of useful experience for handling the more subtle and involved problems which may be encountered.

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THE UNIAXIAL MICROPHONE*

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Summary—A small unidirectional microphone has been developed with the following features: Maximum sensitivity along the axis of the microphone; a high ratio of electrical output to size; a sharper directivity pattern than a cardioid; a directivity pattern that is independent of the frequency; a blastproof vibrating system. The high discrimination which this microphone exhibits to sounds which originate from the sides and rear makes it particularly suitable for long distance sound pickup in radio, television, sound motion pictures, and sound reinforcing systems.

INTRODUCTION

The general trend in microphones for use in television is in the direction of smaller and more unobtrusive units. One of the first high-quality units of this type was the small "Bantam" velocity microphone.[1] Another example is the "Starmaker," a nondirectional pressure ribbon microphone.[2] However, for boom use and other long distance pickup applications a directional microphone is required. The "77D" unidirectional microphone is almost universally used for these applications. This microphone was developed almost ten years ago. Over this period, research and development work has been carried on with the objective of improving the acoustic and magnetic systems of microphones. An example, incorporating some of these developments, is the "Starmaker" microphone referred to above. A review of the unidirectional microphone indicated that it would be possible to reduce the size and improve the directivity. Furthermore, a new requirement was a blastproof vibrating system capable of withstanding blast from guns, pistols and small explosions. This review also indicated that a blastproof feature could be incorporated. The vibrating system which appeared to be the most suitable for obtaining the above enumerated features was one similar to that used in the Starmaker. In the unidirectional form this system has been termed the uniaxial microphone because the maximum sensitivity corresponds to the axis of the system. It is the purpose of this paper to describe the uniaxial microphone.

THEORY

The motor selected for this microphone is shown in Figure 1. Among the advantages of this motor is the simple and efficient magnetic structure. Simplicity is accomplished by the use of a small number of easily machined parts. High magnetic efficiency is obtained due to the small leakage inherent in magnetic designs of this type.

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[1] L. J. Anderson and L. M. Wigington, "The Bantom Volicity Microphone," *Audio Eng.*, vol. 34, p. 13; January, 1950.

[2] H. F. Olson and J. Preston, "Unobtrusive Pressure Microphone," *Audio Eng.*, vol. 34, p. 18; July, 1950.

The next consideration is a blastproof system. A typical sound wave produced by the firing of a gun is shown in Figure 2. The total time depends upon the type of gun. It appears to range from 1/20 to 1/40 of a second from a 0.45- to a 0.22-caliber gun. The low-frequency components of this wave are not reproduced through the complex chain of elements which constitute the sound

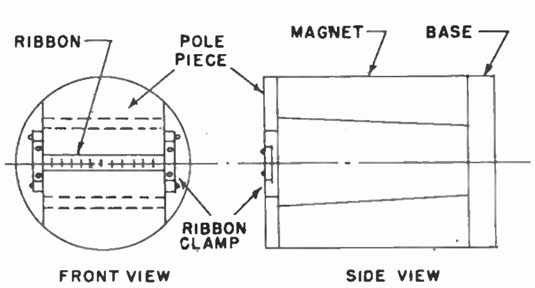

Fig. 1 – Front and side views of the motor of the uniaxial microphone.

channel in recording or broadcasting. However, these low-frequency components in the blast pulse produce the large deflections in the ribbon and stress it beyond the elastic limit and thereby introduce a permanent deformation in the ribbon. What is required is a system which will reduce the low-frequency amplitude of the sound pressure delivered to the ribbon.

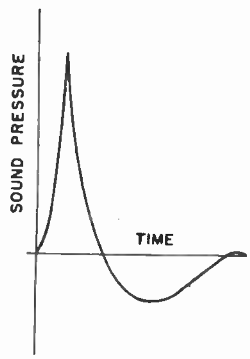

Fig. 2 – A graph of the sound wave produced by the firing of a gun.

The sound pressure delivered to the ribbon in the low-frequency range can be reduced by the use of an acoustical resistance placed over each side of the ribbon as shown in Figure 3. The acoustical curcuit is also shown in Figure 3. The components of the acoustical impedance of the elements of the system shown in Figure 3 are shown in Figure 4. These components are as follows:

The acoustical resistance is

$$r_A = r_{A_1} + r_{A_2}, \tag{1}$$

where r_{A_1} = acoustical resistance of the screen on the front of the ribbon,

r_{A_2} = acoustical resistance of the screen on the back of the ribbon.

The positive acoustical reactance of the ribbon is given by

$$X_{AM} = 2\pi f M, \tag{2}$$

where M = inertance of the ribbon.

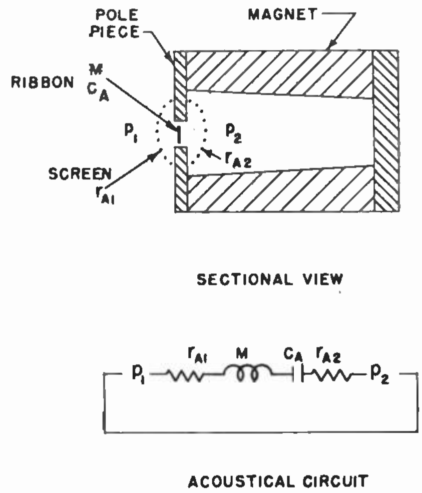

Fig. 3 – Sectional view and acoustical circuit of a simplified vibrating system of a uniaxial microphone to show the effect of the blast baffles. In the acoustical circuit: p_1 is the sound pressure on the front of the microphone; r_{A_1} and r_{A_2} the acoustical resistances of the blast baffles; M and C_A the inertance and acoustical capacitance of the ribbon; and P_2 the sound pressure on the back of the microphone.

The negative acoustical reactance of the ribbon is given by

$$X_{AC} = \frac{1}{2\pi f C_A}, \tag{3}$$

where C_A = acoustical capacitance of the ribbon.

The volume current in the system of Figure 3 is given by

$$\dot{X} = \frac{p_1 - p_2}{r_A + jX_{AM} + jX_{AC}}, \tag{4}$$

where p_1 = pressure on the front of the ribbon, and
p_2 = pressure on the back of the ribbon.

The driving pressure, $p_1 - p_2$, is proportional to the frequency. The reason is as follows: The magnitudes of p_1 and p_2 are the same. However, the phase difference between p_1 and p_2 is proportional to the frequency. Therefore, in the frequency range in which the distance between the front and back is small compared to the wavelength, the difference in pressure between the front and back will be proportional to the frequency.

Referring to Figures 3 and 4 it will be seen that the low-frequency response will be attenuated by the addition of the acoustical resistances. The response–frequency characteristics with and without the acoustical

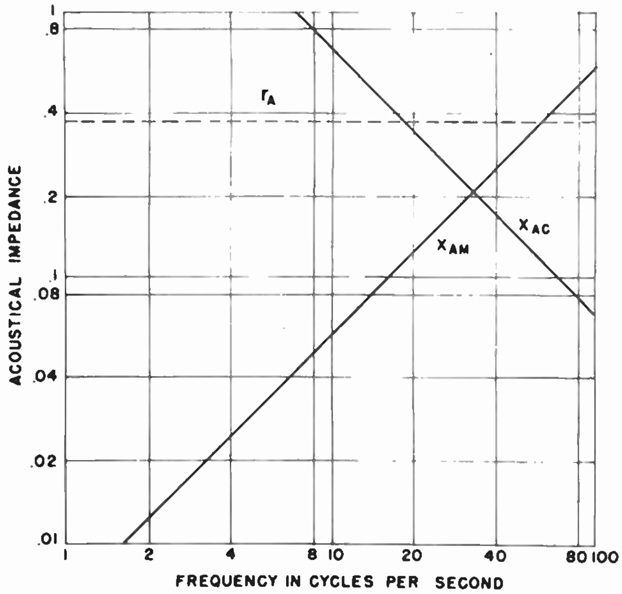

Fig. 4 – The components of the acoustical impedance of the system of Fig. 3. r_A is the total acoustical resistance, x_{AM} the positive acoustical reactance, and x_{AC} the negative acoustical reactance.

resistances are shown in Figure 5. These characteristics show the high attenuation in the low-frequency region due to the addition of the acoustical resistance.

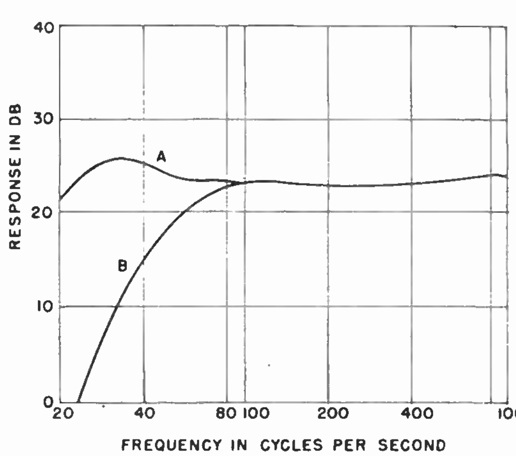

Fig. 5 – Response frequency characteristic of the system of Fig. 3: A. Without acoustical resistance; B. With acoustical resistance.

A sectional view of the complete vibrating system is shown in Figure 6. The magnetic system is similar to the one shown in Figure 1. The ribbon is connected to the damped folded pipe or labyrinth by means of a connector which couples the rectangular cross-sectional area at the ribbon to the circular cross-sectional area at

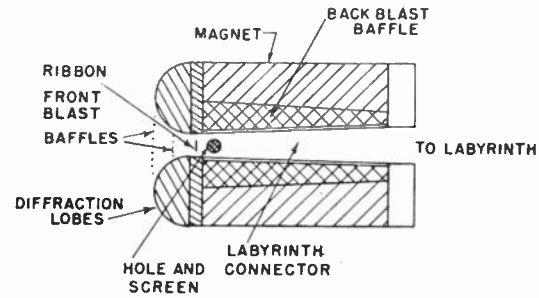

Fig. 6 – A sectional view showing the elements of the motor of the uniaxial microphone.

the labyrinth. The connector is provided with two holes, one on each small side near the connection to the ribbon. These two holes provide the essential portion of the phase-shifting acoustical network so that the directional pattern will be of the unidirectional type. The front face of the microphone is equipped with two lobes. The lobes perform three functions: the reduction of the deleterious effects of diffraction, the accentuation of the high-frequency response, and the support of the blast baffles. The front of the microphone is equipped with two blast baffles held in place by the lobes. The side of the microphone is equipped with a single blast baffle. The action of the different elements and the complete microphone will be described in the discussion which follows.

The front face of the microphone is circular. Therefore, the response will be nonuniform in the high-frequency range due to the diffraction effects produced at the face of the cylinder. The sound-pressure response-frequency characteristic at the center of the cylinder for normal incidence of the impinging sound wave of uniform sound pressure in free space is shown in Figure 7A. The response-frequency characteristic of the system shown in Figure 6 operating as a pressure microphone and without the lobes is shown in Figure 7B. The deviation of this response-frequency characteristic from that of Figure 7A is due to the fact that the ribbon covers a length of about one inch on the surface rather than a single point at the center. The response-frequency characteristic of the system of Figure 6 operating with the lobes is shown in Figure 7C. The addition of the lobes reduces the effects of diffraction. The net result is a smoother response-frequency characteristic. The lobes also act as a small horn which accentuates the response in the extreme high-frequency range.

A sectional view and acoustical network of the complete microphone are shown in Figure 8. It is of the single-ribbon type in which the back of the ribbon is coupled to a damped pipe and an inertance in the form of an aperture in the pipe.

The action of the microphone can be obtained from a consideration of the acoustical network. The sound pressure on the open side of the ribbon may be written

$$p_1 = p_0 \epsilon^{j\omega t}, \tag{5}$$

where p_0 = amplitude of the pressure, in dynes, per square centimeter,

ω = $2\pi f$,

f = frequency, in cycles per second,

t = time.

The sound pressure acting upon the air load of the aperture and the aperture may be written

$$p_2 = p_0 \epsilon^{j(\omega t + \phi_1)}, \tag{6}$$

where ϕ_1 = phase angle between the pressure p_1 and the pressure p_2.

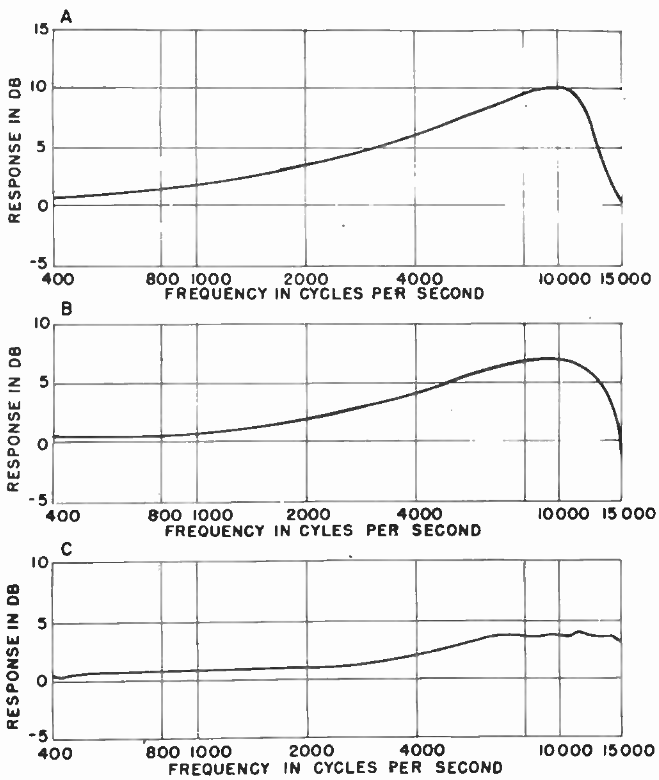

Fig. 7 – A. The response at the center of a cylinder; B. The response of the uniaxial microphone without the diffraction lobes; C. The response of the uniaxial microphone with the diffraction lobes.

Referring to Figures 6 and 8 it will be seen that there is a cavity between the magnets filled with damping material. This cavity is coupled to the aperture as shown in the acoustical network. This cavity is also acted upon by a sound pressure which may be written

$$p_3 = p_0 \epsilon^{j(\omega t + \phi_2)} \tag{7}$$

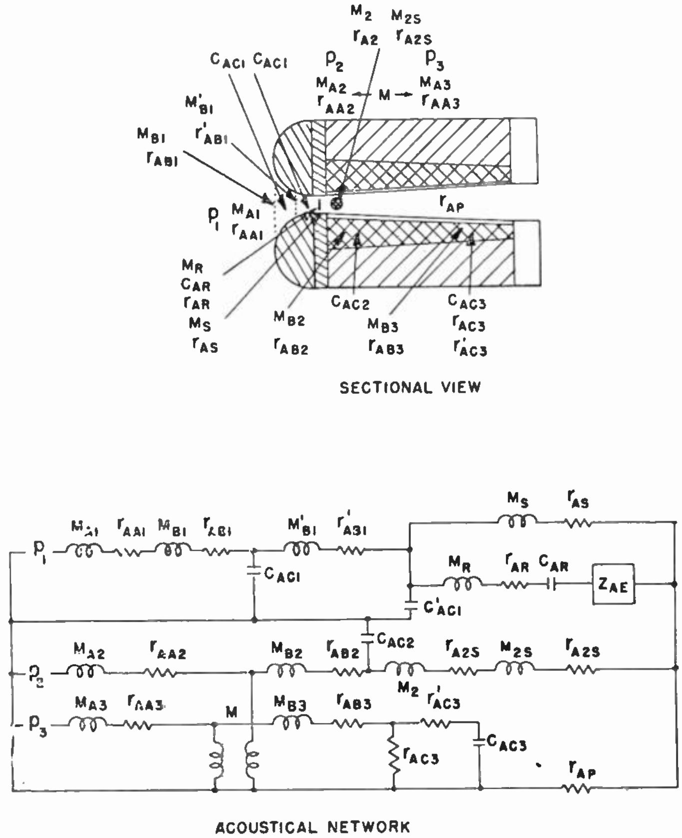

Fig. 8 – Sectional view and acoustical network of a uniaxial microphone. In the acoustical network: p_1 is the sound pressure on the front of the microphone; M_{A_1} and r_{AA_1} the inertance and acoustical resistance of the air load on the front of the microphone; M_{B_1}, r_{AB_1}, M'_{B_1}, and r'_{AB_1} the inertances and acoustical resistances of the blast baffles on the front of the microphone; C_{AC_1} and C'_{AC_1} the acoustical capacitances of the volumes between the blast baffles; M_S and r_{AS} the inertance and acoustical resistance of the slit between the ribbon and pole pieces; M_R, r_{AR}, and C_{AR}, the inertance, acoustical resistance and acoustical capacitance of the ribbon; Z_{AE} the acoustical impedance due to the electrical circuit; p_2 the sound pressure at the apertures in the labyrinth connector; M_{A_2} and r_{AA_2} the inertance and acoustical resistance of the air load at the apertures of the labyrinth connector; M_{B_2} and r_{AB_2} the inertance and acoustical resistance of the blast baffles on the side of the microphone; C_{AC_2} the acoustical capacitance of the volume behind the blast baffle; M_{2S} and r_{A_2S} the inertance and acoustical resistance of the screen covering the hole in the labyrinth connector; M_2 and r_{A_2} the inertance of the hole in the labyrinth connector; r_{AP} the acoustical resistance of the labyrinth; p_3 the sound pressure at the damped cavity between the magnets; M_{A_3} and r_{AA_3} the inertance and acoustical resistance of the air load upon the damped cavity; M_{B_3} and r_{AB_3} the inertance and acoustical resistance of the blast baffle over the damped cavity; C_{AC_3}, r_{AC_3} and r'_{AC_3} the acoustical capacitance and acoustical resistances of the cavity between the magnets; M the coupling between the cavity and the apertures.

The phase angles ϕ_1 and ϕ_2 are a function of the angle of the incident sound as follows:

$$\phi_1 = \Phi_1 \cos\theta, \tag{8}$$

and

$$\phi_2 = \Phi_2 \cos\theta, \tag{9}$$

where θ = angle between the normal to the surface of the ribbon and the direction of the incident sound wave,

Φ_1 = constant phase angle for the pressure p_2, and

ϕ_2 = constant phase angle for the pressure p_3.

The coupling M between the aperture and the damped cavity is complex function of the dimensions and configuration of the cavity and the relation of the cavity to the aperture. In addition, the coupling is a function of the direction of the incident sound. Under these conditions the acoustical network of Figure 8 is extremely

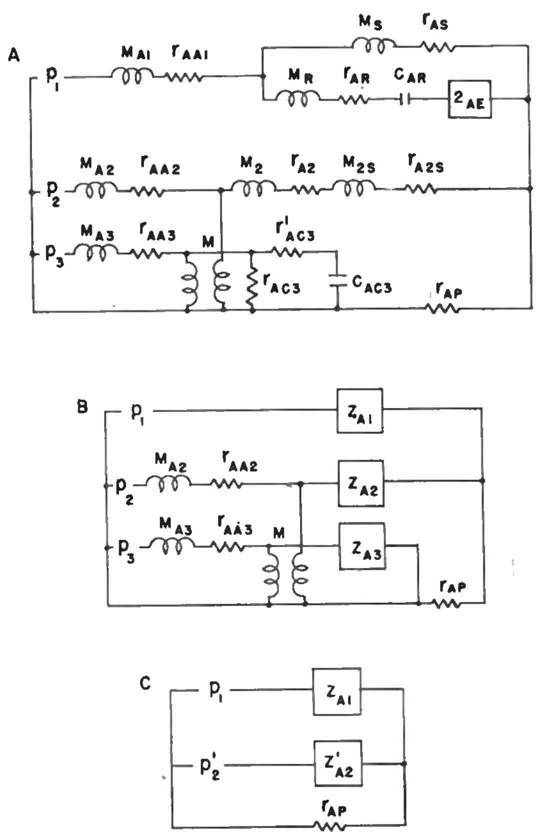

Fig. 9 – A. Acoustical network of Fig. 8 with the blast baffles removed; B. Acoustical network of A with the equivalent lumped acoustical impedances; C. The approximate equivalent acoustical network of B with the mutual coupling system and associated acoustical impedances replaced by a single pressure and acoustical impedance.

complex. If the blast baffles are removed, the acoustical network is reduced to the one shown in Figure 9A. A further simplification can be obtained by lumping the elements into the impedances as shown in Figure 9B. The acoustical network of Figure 9B can be reduced to the acoustical network of Figure 9C in which the pressure

$$p'_2 = f_1(p_2)\, f_2(p_3)\, f_3(\theta), \tag{10}$$

and the acoustical impedance

$$z'_{A_2} = f_4(z_{A_2})\, f_5(z_{A_3}). \tag{11}$$

In the acoustical network of Figure 9C, the volume current in the acoustical impedance z_{A_1} due to pressure p_1 is

$$\dot{X}_1 = \frac{p_1(z'_{A_2} + r_{AP})}{z_{A_1} z'_{A_2} + z_{A_1} r_{AP} + z'_{A_2} r_{AP}}. \tag{12}$$

The volume current in the acoustical impedance z_{A_1} due to the pressure p'_2 is

$$\dot{X}_2 = \frac{p'_2 r_{AP}}{z_{A_1} z'_{A_2} + z_{A_1} r_{AP} + z'_{A_2} r_{AP}}. \tag{13}$$

The resultant volume current in the acoustical impedance z_{A_1} is the difference between the volume currents

$$\dot{X}_R = \dot{X}_1 - \dot{X}_2. \tag{14}$$

If the volume current through the slit between ribbon and the pole is negligible, the volume current $\dot{X}_R$ is the volume current of the ribbon.

The velocity of the ribbon is

$$\dot{x}_R = \frac{\dot{X}_R}{A_R}, \tag{15}$$

where A_R = area of the ribbon.

The voltage generated by the motion of the ribbon is given by

$$e = Bl\dot{x}_R, \tag{16}$$

where B = flux density in the air gap, and l = length of the ribbon.

The performance of the microphone can be predicted from (5) to (16) inclusive. The directivity pattern of the uniaxial microphone shown in Figure 6, under a given set of constants, can be approximately expressed as

$$e = \left(K \quad 0.3 + 0.7\cos\theta\cos\frac{\theta}{3}\right), \tag{17}$$

where K = sensitivity constant of the microphone.

The directivity pattern of the uniaxial microphone obtained from (17) is shown in Figure 10.

It is interesting to compare the directional patterns of the uniaxial microphone with the conventional unidirectional microphone. If the damped cavity is omitted, then

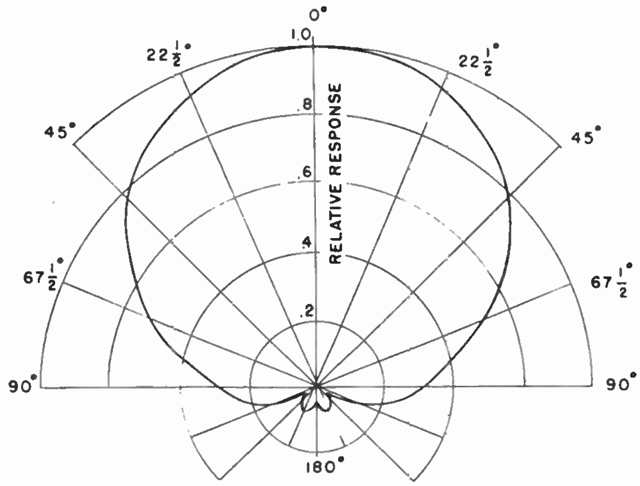

Fig. 10 – Theoretical directional pattern of the uniaxial microphone.

the microphone becomee the conventional unidirectional type with the acoustical network of Figure 11A. The acoustical network of Figure 11A can be reduced to the acoustical network of Figure 11B.

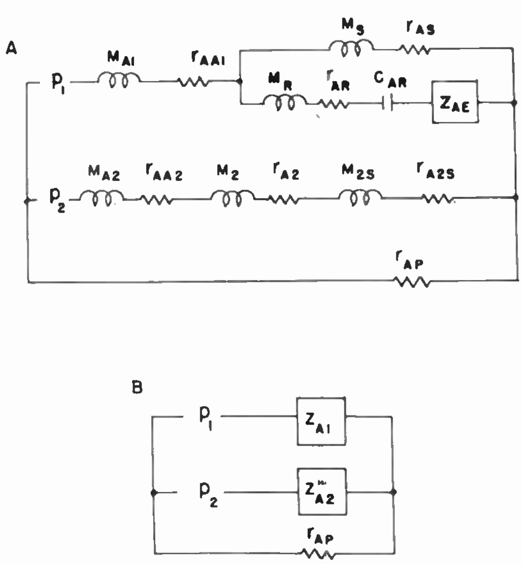

Fig. 11 – A. The acoustical network of the acoustical system of Fig. 9 with the damped cavity omitted; B. The equivalent acoustical network of A with the equivalent lumped acoustical impedances.

The resultant volume current through the acoustical impedance z_{A_1} is given by

$$\dot{X}_R = \frac{p_1 (z''_{A_2} + r_{AP}) - p_2 r_{AP}}{z_{A_1} z''_{A_2} + z_{A_1} r_{AP} + z''_{A_2} r_{AP}} . \qquad (18)$$

The voltage generated by the ribbon can be obtained from (15), (16), and (18). The directivity pattern of this microphone is given by

$$e = K\ (a + b \cos \theta) ,$$

where K = sensitivity constant of the microphone, a and b = limaçon constants, $a + b = 1$.

A few of the directivity patterns which can be obtained from this microphone are shown in Figure 12.

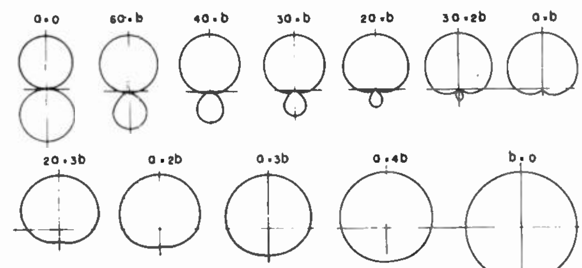

Fig. 12 – A few of the single infinity of directional characteristics obtainable with the polydirectional microphone.

Comparing Figures 10 and 12 it will be seen that it is not possible to obtain, in the conventional microphone, the same directivity in the front hemisphere as that in the uniaxial microphone without sacrificing high discrimination in the rear hemisphere. For example, referring to Figure 12, the directivity pattern of the unidirectional microphone in the front hemisphere for $3a = b$ is approximately the same as the uniaxial microphone in the front hemisphere. However, the discrimination in the rear hemisphere for the unidirectional microphone for the same constants is only 6 decibels for 180 degrees as compared to 26 decibels for the uniaxial microphone.

PERFORMANCE CHARACTERISTICS

The response frequency characteristics of the uniaxial microphone for the angles 0°, 45°, 90°, 135° and 180°, where 0° corresponds to the axis of the microphone are shown in Figure 13. The average polar directivity pattern of the uniaxial microphone is shown in Figure 14.

In order to show the effect of the damped cavity upon the directivity pattern, the cavity was covered with thick sheets of copper. The directivity pattern obtained under these conditions is shown in Figure 15. The directivity pattern under these conditions is slightly broader in the front hemisphere. Also, a large lobe has now appeared in the rear hemisphere. The directivity pattern shown in Figure 15 is not satisfactory as a unidirectional microphone because of the inadequate discrimination in the rear hemisphere. With the cavity covered, the holes in the connector were reduced in size until the

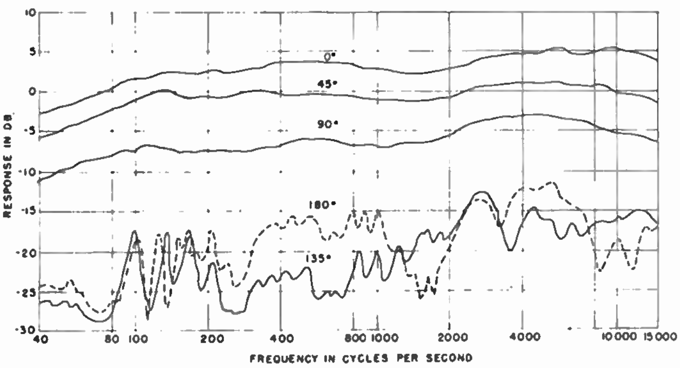

Fig. 13 – The measured response frequency characteristics of the uniaxial microphone sound incident at 0°, 45°, 90°, 135° and 180°. 0° corresponds to the axis of the microphone.

cardioid pattern shown in Figure 16 was obtained. This is the typical pattern for a unidirectional microphone with high discrimination against sound coming from the rear. Now the directivity pattern is much broader in the

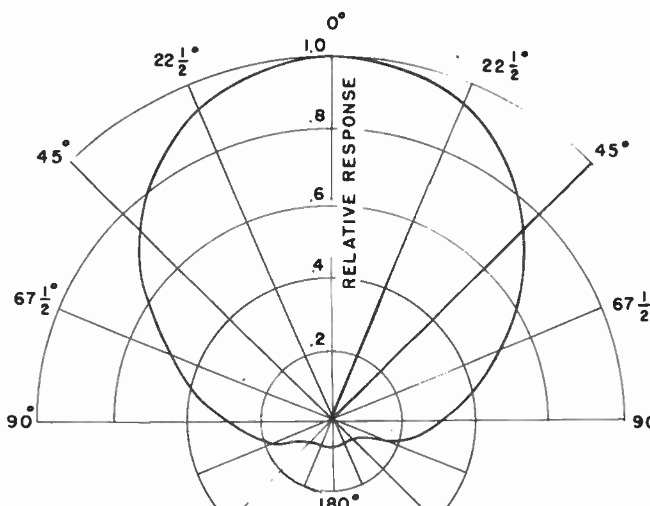

Fig. 14 – The average of the measured directional patterns of the uniaxial microphone.

front hemisphere. The response of a unidirectional microphone with a cardioid directivity pattern to random sounds, with all directions equally probable and of equal strength, is 1/3 that of a nondirectional microphone. The response of the uniaxial to random sounds is 1/5 that of a nondirectional microphone. This means that the response of the uniaxial microphone to random sounds is 60 per cent that of a microphone with a cardioid pattern. From the standpoint of sound pickup distance, the uniaxial microphone will operate at 30 per cent greater distance for the same reverberation or undesirable sounds or noise.

Tests have been made of the effectiveness of the blastproofing. The uniaxial microphone will stand the blast of a 0.45 pistol firing blanks at a distance of 5 feet indoors, with the direction of firing at right angles to the

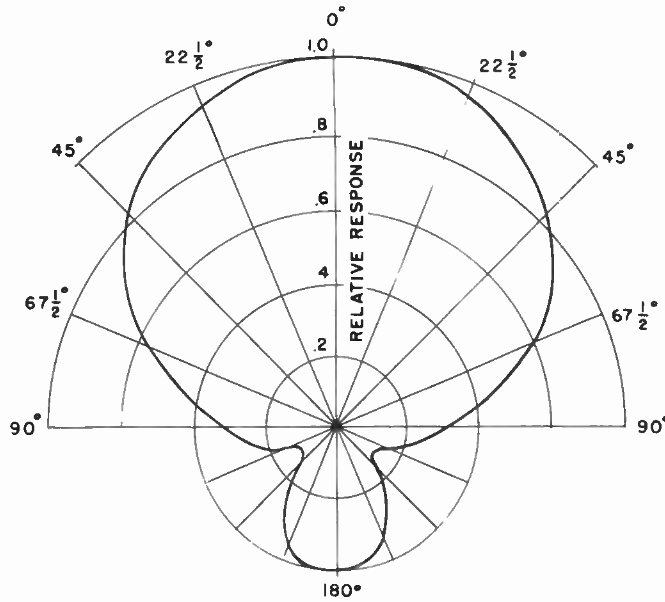

Fig. 15 – The average of the measured directional patterns of a conventional unidirectional microphone set for the pattern $e = .3 + .7 \cos \theta$.

microphone. It will withstand the same blast at a smaller distance outdoors. However, a distance of 5 feet should be ample for practically all conditions of use because the effective operating pickup distance of the uniaxial microphone is greater than the conventional unidirectional microphone.

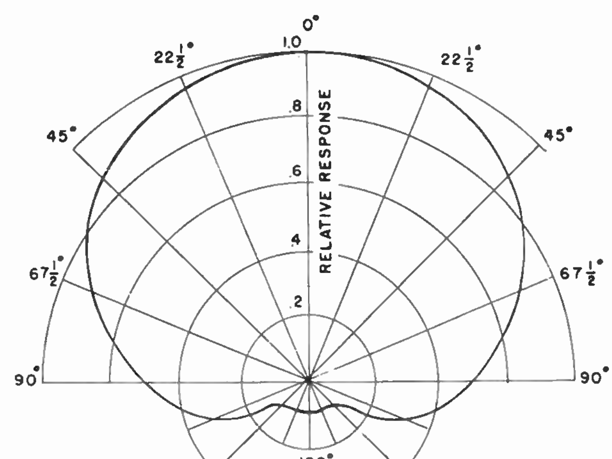

Fig. 16 – The average of the measured directional patterns of a conventional unidirectional microphone set for a cardioid pattern.

APPLICATIONS

One of the applications for this microphone is the pickup of sound in television where the microphone is mounted on a boom and kept out of the picture. The directional properties of the microphone are particularly suitable for this application because the discrimination against unwanted sounds which originate at the sides and the rear is very high. A photograph of the cradle which has been developed for this microphone for use on a boom is shown in Figure 17. The cradle design provides an underslung mounting of the microphone so that the microphone is the lowest part of the system. This condition allows for minimum distance between the source of sound and the microphone.

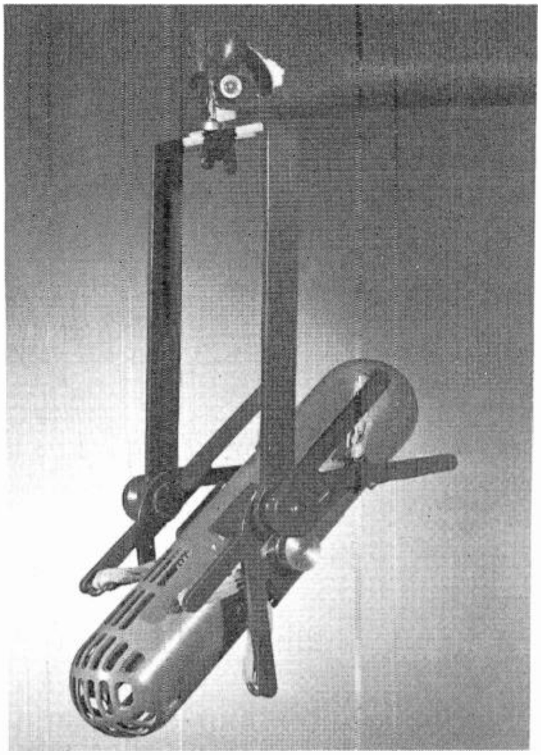

Fig. 17 – The uniaxial microphone mounted in a cradle for boom operation.

Other applications for this microphone are the pickup of sound in television where the microphone is seen in the picture. Either floor or desk stands are used for these applications. A desk stand is shown in Figure 18. The uniaxial microphone is particularly suited for these applications because the maximum response occurs on the axis. Under these conditions the projected area of the microphone is a minimum as viewed by the camera.

Fig. 18 – The uniaxial microphone mounted on a desk stand.

The uniaxial microphone is also useful for standard broadcast and sound reinforcing applications where the unidirectional microphone has already been established as the solution for sound pickup problems. It has been extensively field tested, including use at the 1952 political conventions.

INSTITUTIONAL LISTINGS (Continued)

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

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

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

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

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

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

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,
B. B. Bauer, Shure Brothers Inc., 225 W. Huron St., Chicago 10, 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.

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

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

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

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

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

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

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

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

CINEMA ENGINEERING CO., Division Aerovox Corp., 1100 Chestnut St., Burbank, California
Equalizers, Attentuators, Communication Equipment

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

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

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

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

(Please see inside back cover for additional names)

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Professional Group on Audio

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

September-October, 1953

Published Bi-Monthly

Volume AU-1 **Number 5**

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The Institute of Radio Engineers

IRE PROFESSIONAL GROUP ON AUDIO

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

Annual Assessment: $2.00

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Professional Group on Audio

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PHILADELPHIA CHAPTER CONDUCTS SYMPOSIUM ON AUDIO*

William G. Chaney
Bell Telephone Company of Pennsylvania
Philadelphia, Pennsylvania

With the Philadelphia Chapter of the P. G. A. struggling to gain a recognized place in local I. R. E. activities, the idea of a symposium seemed to be highly in order. First, it would stimulate interest in the Chapter and encourage membership. Second, it would bring prominent speakers to Philadelphia to present various phases of the subject in a cohesive fashion. There was ample proof that audio papers were in demand by the phenomenal attendance at scattered section meetings over the past few seasons.

It was found that the Communications Division of the New York Section A.I.E.E. was also planning to offer a series of six papers on audio. There seemed to be no reason why the two should not parallel each other and by working out a cooperative schedule it was possible to have three of the speakers present their papers in both places.

An offer of joint sponsorship was accepted by the Philadelphia Section of the A.I.E.E. This was done to promote better relations between the two organizations and to get the increased attendance from their support.

The Symposium on Audio consisted of six papers presented at the rate of one a month from November 1952 to April 1953. A list of the speakers and a brief resume of the talks will give an idea of the scope.

1. Mr. E. W. Kellogg, formerly RCA: "The History of Recorded Music from the 19th Century to the Present."

   Mr. Kellog, one of the pioneers in the loudspeaker and disc recording fields described the methods of disc recording as they developed to the present state. The paper was supplemented by demonstrations of antique recordings including several wax cylinders played back on machines of the same vintage. The reproducing equipment was borrowed from RCA's historical library.

2. Winston E. Kock, Director of Acoustics Research, Bell Telephone Laboratories: "The Physics of Music and Hearing."

   Dr. Kock discussed the physical concepts of hearing and also the loudness and frequency range for optimum listening enjoyment. The remainder of the talk dealt with electrical means of synthesizing orchestral instruments. A film, "Action Pictures of Sound," was shown.

3. Frank H. Slaymaker, Chief Engineer Sound Equipment Division, Stromberg-Carlson Company: "Performance Criteria of Loudspeakers."

   Mr. Slaymaker described some of the many ways available to determine the relative merits of loudspeakers. A large part of the talk was concerned with a discussion and demonstration of effects due to resonances in the loudspeaker system.

4. Emory Cook, President, Cook Laboratories: "Recording and Reproducing Techniques."

   Mr. Cook placed special emphasis on the subject of "binaural" recording both from the recording and play-back viewpoint. He demonstrated many of his binaural records and played a Livingston dual-track pickup.

5. Lowell H. Good, Supervisor, Advanced Development Section, RCA: "Audio Amplifiers."

   Mr. Good brought the series back to a more practical footing with a down-to-earth discussion of audio amplifiers. He demonstrated some of the commonly experienced faults such as peak clipping, poor transient response, intermodulation and overdriving.

6. H. H. Scott, President, Herman Hosmer Scott, Inc. — "Component Integration of Sound Systems."

   Mr. Scott concluded the series with a summation of the characteristics of all the integrated units in a versatile sound reproducing system. He discussed impedance matching, tone controls, operating levels and equalization.

Registration and attendance at the meetings lived up to the most optimistic expectations. A total of 267 persons registered for the entire series and the attendance for the six meetings was slightly greater than 1600.

Observations made during the progress of the Symposium may be of interest to other groups.

1. Papers presented once a month tend to lose continuity because of the long gap.
2. The papers must be technical enough to have the listener leave with the feeling that he has learned something useful.
3. Speakers should be selected for speaking ability as well as for technical knowledge.
4. Demonstrations are invaluable, both for illustrating points and relieving monotony in a two-hour lecture.
5. In determining attendance, publicity is of first importance, prominent speakers rank second, and convenience of hour and location third.

* Manuscript received July 29, 1953.

REPORT ON TAPESCRIPTS COMMITTEE PGA*

Andrew B. Jacobsen, Chairman
Tapescripts Committee, IRE-PGA

The activities of the Tapescript Committee had covered two years, starting out in a very modest manner with papers given at Conventions. The last year has been bolstered materially by specially prepared papers produced by General Electric and Bell Telephone Laboratories. Such papers provide excellent program material for those Sections and Chapters who find it difficult to obtain speakers to appear on their programs.

It has been found that only a small percentage of convention papers is satisfactory for this type of presentation, and even this small percentage would be better if specially recorded. It is not necessary that the tapescripts be an elaborate production; but it *is* essential that they be presented by the author, with a series of well-organized slides to accompany the author's talk. In general tapescripts should be along lines of fairly general interest. In rare cases a very specialized tapescript is justified, but it should be kept in mind that the material will be presented to student Chapters and small Sections, Professional Group Chapters, and industrial group meetings.

To date there have been three outstanding tapescripts – a convention paper by Peterson and Sinclair of General Radio, presented by Peterson; "Germanium – The Magic Metal," produced by General Electric; and "Fundamentals of Acoustics" by Kock of Bell Telephone Laboratories. "Germanium – The Magic Metal" is rather an elaborate production using 78 slides in color. W. H. Doherty of Bell Telephone Laboratories has produced a series of seven tapescripts. All of these tapescripts have had wide distribution, and it is hoped that other organizations will see fit to produce equally outstanding material for future distribution.

The technical aspects of tapescripts are really very simple. At present, the best standard seems to be 7½ inches per second, full track magnetic tape on 1200 foot reels. Slides for distribution should be 2" x 2". It has become apparent in the distribution of this material that one copy can be utilized only on the average of twice a month.

The PGA is appropriating funds to provide prepaid transportation to the organization using the tapescript, and the organization will be expected to return it by the cheapest method. The PGA is also undertaking the production of standardized copies of future tapescripts which may be forthcoming. An effort will be made to contact people in the electronics and radio industries to obtain outstanding tapescript material for distribution, and further efforts will also be made to acquaint groups with tapescript material as it becomes available.

*Manuscript received June 22, 1953.

PGA BRIEFS

An *Awards Committee* has been appointed by Marvin Camras, Chairman of IRE-PGA. The chairman of the Awards Committee is Mr. John K. Hilliard and members are Messrs. John A. Kessler, D. W. Martin, Arnold Peterson and Vincent Salmon.

Additional recent appointments to the IRE-PGA *Nominations Committee* include Messrs. Frank Lennert, Hugh Knowles and Leo L. Beranek.

The semi-annual meeting of the *IRE Professional Group on Audio* was scheduled for 2:00 P.M. Monday, September 28, 1953, in the Ruby Room of Hotel Sherman, Chicago, Illinois. The *National Electronics Conference* session on audio was scheduled for the morning of the same day at the same hotel. The date of these meetings is approximately the same as the intended date of this issue of TRANSACTIONS of the IRE-PGA. A full account of the meetings will be carried in the next issue.

A Symposium on Measurement of Sound System Performance was held at the *IRE Western Convention and Electronic Show,* held in San Francisco, August 19-21. The symposium was conducted by Dr. Vincent Salmon, vice-chairman of IRE-PGA. Material from the symposium is expected to appear soon in TRANSACTIONS of the IRE-PGA. An additional session on audio was conducted by Roy Long, Stanford Research Institute.

Welcome to the Houston Chapter of the IRE-PGA! The new chapter was officially approved at the July 7th meeting of the Executive Committee of the Institute. Houston is the twelfth chapter of IRE-PGA. Mr. L.A. Geddes has been elected chairman of the chapter for the coming year.

The Chicago Chapter IRE-PGA has the following new officers:

Chairman:	Philip B. Williams, Jensen Manufacturing Company
Vice-Chairman:	R. Lee Price, Magnecord, Inc.
Secretary:	Theodore S. Pryst, Shure Brothers, Inc.

The fourth annual national Noise Abatement Symposium will be held at Armour Research Foundation on October 23 and 24. Further information may be obtained from Mr. G. L. Bonvallet, Armour Research Foundation, Chicago, Illinois.

The Acoustical Society of America is conducting its fall meeting at Case Institute in Cleveland, October 15-17.

Dr. W. O. Muckenhirn has accepted the chairmanship of the PGA *Ways and Means Committee.*

The Department of Defense is holding a symposium on Magnetic Recording in Washington, D.C. on October 12 and 13, 1953. Attendance is restricted to those directly engaged in some phase of magnetic recording development or application. Information may be obtained from: Department of the Navy, Bureau of Ships (Attn: Code 565E), Washington 25, D.C.

RTMA SE–8 COMMITTEE ON HIGH FIDELITY SOUND SYSTEMS*

Frank H. Slaymaker
Stromberg-Carlson Company
Rochester, New York

The RTMA Engineering Department activities in the field of high fidelity audio equipment are being handled by a special committee (SE–8). This committee met for the first time on October 30, 1952 in New York City. It was formed in an effort to dispel some of the mysteries that have enveloped high fidelity sound, whether one regards it from the point of view of either a business or a hobby. In accordance with the general purposes of all RTMA committees, one of its aims is to encourage the use, by various manufacturers of high fidelity equipment, of comparable terms to describe and rate their equipment. A long range objective of the committee is to find some common basis for specifically defining the term *high fidelity* itself. An immediate and more readily obtained objective, however, is to set up industrial standards on the means used to interconnect tuners, amplifiers, tape recorders and record players, as well as to establish mutually acceptable standards on the output levels and gains required for the various types of equipment.

At present, the non-technical high fidelity fan is never sure whether he can connect a Brand "A" record player to Brand "B" preamplifier to Brand "C" power amplifier to Brand "D" loudspeaker, or whether it is possible to connect another type of tape recorder, television chassis, or radio tuner into his amplifier and have the system work. He is not even sure whether the connectors will mate properly.

The need for such a committee had been recognized for some time, and at the 1952 RTMA Fall Meeting held in Syracuse in October, the Sound Equipment Section laid the plans for its organization and secured as its chairman Mr. Frank H. Slaymaker, Chief Engineer of the Sound Equipment Division of Stromberg-Carlson Company. This committee was designated SE–8. A subcommittee (SE–8.1) to specialize in the investigation and potential standardization of sound levels and inter-unit connectors is under the direction of Chairman H. H. Scott. As part of a plan to establish a definite program of activities for SE–8, it was decided to determine the present state of the art in the matters of interconnectors and operating levels. Therefore, the subcommittee, SE–8.1, recently prepared and distributed throughout the high fidelity industry, a comprehensive questionnaire covering these and other items. This survey of the field will be used in establishing the work schedules of several other subcommittees which will be established soon to cover all phases of the work. Some of these other problems, noted by the Committee as requiring future action, are as follows:

1. The problem of how to rate the output characteristics of an amplifier is to be studied.
2. A task force is to be set up to investigate the problems associated with UL tests and listings.
3. The record playback characteristics are to be investigated, along with associated matters. A task force appointed by the chairman has made initial studies here.
4. Ratings applicable to amplifier with variable turnover frequencies is another problem for study.
5. The possibilities of securing the cooperation of other groups in the matters of definitions, and correlating the definitions already established, is being determined together with methods of measurement of the various quantities involved. The IRE, SMPTE, NARTB, AES, and the Motion Picture Research Council are being consulted here.
6. The standardization of certain physical dimensions in the sound equipment field would also seem to promote better interchangeability.

The committees are staffed by competent engineering representatives of recognized manufacturers of high fidelity amplifiers, tuners, tape recorders, loudspeakers and associated equipment. At the present time, the committee welcomes representatives of any other high fidelity manufacturers having pertinent engineering topics to discuss.

Additional information concerning future meeting dates and locations may be obtained from the RTMA Engineering Office.

*Manuscript received July 27, 1953.

TECHNICAL COMMITTEE WORK ON AUDIO-FREQUENCY MEASUREMENTS*

Arnold Peterson
General Radio Company
Cambridge, Massachusetts

Among the IRE Technical Committees, there are three which are directly concerned with audio problems. These committees are the following: Audio Techniques (#3), Electroacoustics (#6), and Sound Recording and Reproducing (#19). Some of the other committees, such as Circuits (#4) and Electron Devices (#7), are so broad in scope that their work is of interest to audio engineers as well as to all other radio engineers. One of these committees of broad scope, namely, Measurements and Instrumentation (#25), has a subcommittee, Audio-Frequency Measurements (#25.4), which is working on audio problems. This subcommittee was the most recently formed of any of the committees dealing with audio. As an initial step, therefore, it surveyed the existing technical committee work pertaining to its field of interest. Since this survey can be useful in bringing standardization work to the attention of audio engineers, this report summarizes the results of that survey.

The organizations and committees listed below are those working in the field of audio-frequency measurement or fields closely related to it. The current standards applying to the field of audio-frequency measurements which have been prepared by these committees are listed. When a standard includes a specification of an audio test procedure, it is also listed.

I. *Institute of Radio Engineers*

Committee 3. Audio Techniques.
- 53 IRE 4 S1 Volume Measurements of Electrical Speech and Program Waves (see ASA C16.5-1942)

Committee 6. Electroacoustics

Committee 19. Sound Recording and Reproducing
- 53 IRE 19 S1 Methods of Measurement of Noise.

Subcommittee 25.4 Audio-Frequency Measurements

II. *American Institute of Electrical Engineers*

Joint Subcommittee on Electronic Instruments.
Special Communications Application Electroacoustics Subcommittee.

III. *American Standards Association*

A. IRE sponsored standards.

1. C16.4 – 1942 Loudspeaker Testing.
2. C16.5 – 1942 Volume Measurements of Electrical Speech and Program Waves.

B. AIEE sponsored standards.

1. C39.1 – 1951 Electrical indicating instruments.

C. Acoustical Society of America sponsored standards.

1. Z24.1 – 1951 Acoustical Terminology.
2. Z24.2 – 1944 Noise Measurement.
3. Z24.3 – 1944 Sound Level Meters for Measurement of Noise and Other Sounds.
4. Z24.4 – 1949 Pressure Calibration of Laboratory Standard Pressure Microphones.
5. Z24.5 – 1951 Audiometers for General Diagnostic Purposes.
6. Z24.7 – 1950 Test Code for Apparatus Noise Measurement.
7. Z24.8 – 1949 Laboratory Standard Pressure Microphones.
8. Z24.9 – 1949 Coupler Calibration of Earphones.
9. Z24.10–1953 Octave-Band Filter Set for the Analysis of Noise and Other Sounds.
10. Z24.12–1953 Pure-Tone Audiometers for Screening Purposes.
11. Z24.14–1953 Measurements of Characteristics of Hearing Aids.

D. Society of Motion Picture and Television Engineers sponsored standards.

1. Z22.51–1946 Intermodulation Tests on Variable Density 16-Millimeter Sound Motion Picture Prints.
2. Z22.52–1946 Cross-Modulation Tests on Variable Area 16-Millimeter Sound Motion Picture Prints.

IV. *Radio-Television Manufacturers Association*

REC-124	Output Transformers for Radio Broadcast Receivers; February, 1949.
REC-125-A	Phonograph Pickups; July, 1949.
REC-128	Standard Frequency Test Record; May, 1949.
REC-134	Magnetic Recorders; Conditions for Measurement and Definitions; August, 1949.
SE-101-A	Amplifier for Sound Equipment; July, 1949.

*Manuscript received July 31, 1953.

SE-103	Speakers for Sound Equipment; April, 1949.
SE-104	Engineering Specifications for Amplifiers for Sound Equipment; May, 1949.
SE-105	Microphones for Sound Equipment, August, 1949.
SE-106-A	Sound Systems; December, 1951.
TR-105-B	Audio Facilities for Radio Broadcasting Systems; November, 1949.
TR-118	Cable Connectors for Audio Facilities for Radio Broadcasting; December, 1949.
TR-121	Audio Transformers for Radio Transmitters; January, 1951.
TR-122	Audio Reactors; January, 1951.
TR-130	Basic Requirements for Broadcast Microphone Cables; July, 1952.

V. *National Association of Broadcasters*

N.A.B.	Recording and Reproducing Standards for Mechanical, Magnetic and Optical Recording and Reproducing; April, 1949.
N.A.B.	Engineering Handbook; 4th Edition, 1949.

VI. *Audio Engineering Society*

POWER CAPACITY OF LOUDSPEAKERS*

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SUMMARY – A method of determining the power capacity of a loudspeaker is submitted which is based upon the departure from linearity of acoustic output versus electrical input curves using octave bands of thermal noise. The method is being considered for inclusion in a new Standard which is being prepared by the Committee on Electroacoustics of the IRE, W. D. Goodale, Chairman. Comments of interested readers are solicited.

INTRODUCTION

The Electroacoustic Committee of the Institute of Radio Engineers is presently writing a new Standard for Loudspeaker Testing. One of the most elusive criteria for which the Committee would like to write a recommended measurement procedure is power handling capacity. The lack of a realistic standard for the determination of this quantity is a continual source of embarassment to specification writers. It is the function of this paper to report a method which has been used only to a limited extent, but which has shown promise as a simple and useful method for defining a realistic measure of the ability of a loudspeaker to reproduce any type of program material without objectionable distortion.

METHOD

The proposed method defines power handling capacity in terms of a power input spectrum and a sound pressure output spectrum which are related on the basis of a specified departure from linearity of output as a function of input. The results of such a test on a particular loudspeaker are shown in Fig. 1.

The procedure is as follows:

Bands of thermal noise are used for input material. The frequency limits of the bands are defined by the proposed ASA Standard Z-24, 10/272 for Octave Band Filter Sets of July 23, 1951 which is in agreement with MIL-S-3151 of March 10, 1950 and the Aircraft Industries Association, Communication of Sept. 27, 1946. The "thermal" noise used in these tests has uniform energy per cycle and a peak factor of about 12 decibels which is the ratio of peak power to RMS power. These bands of noise are fed, preferably through a calibrated attenuator, to a power amplifier which must have adequate power capacity to deliver the necessary peak power to the loudspeaker without distortion. A calibrated microphone is placed at a convenient distance on the axis of the loudspeaker and the output of the microphone is observed on a calibrated voltmeter. The acoustic environment of the microphone is not important for this test, if only the power input spectrum is required.

For each band of noise a curve of sound pressure output versus electrical power input is plotted. These curves are shown on the left of Fig. 1 and they are linear within the usual operating range of a loudspeaker. When overload is approached the output does not increase linearly with the input, so that the curve bends over. At

* Manuscript received July 6, 1953.

any point the departure of the curve from linearity can be determined, and is used as a measure of acceptable distortion. The degree of departure which is used to define a limit for power capacity can be chosen arbitrarily. For the loudspeakers which the writer has tested, the point of one db departure from linearity conforms with a significant aural observation.

It is characteristic of distortion in general that the output energy occupies a wider spectrum than the input energy. Exactly this effect can readily be heard when an octave band of sound energy causes a loudspeaker to overload. The sound energy is heard to spread out into a much broader spectrum. In the case of loudspeakers which the writer has tested, this phenomenon occurs quite abruptly at an input level which corresponds to one decibel departure from linearity. This value has therefore been selected as a suggested standard.

The output-input curve is plotted for each band of noise as shown on the left in Fig. 1. For each band the point of one decibel departure is located. When using a calibrated attenuator it is usually not necessary to actually plot each curve to locate this point. From these points the two curves on the right of Fig. 1 can be plotted The curve labeled "Maximum Output Level" is found by simply projecting each point of one db departure over to its corresponding frequency band. The curve of "Maximum Input Level" is found by plotting the abscissa of each point from the linearity curves over its corresponding frequency band.

The resulting curves clearly indicate an all-important fact, namely that the power capacity of a given loudspeaker cannot in general be stated in terms of a single number. The power capacity is clearly dependent upon the frequency spectrum of the program material. For example, if one is given the octave-band analysis of a certain electrical program signal, this spectrum can be compared with the maximum input level spectrum to determine the region of the spectrum at which overload will first become apparent. It also becomes clear how a spectrum can be modified, for example, by equalizing a speech signal, to make the maximum effective use of the loudspeaker's capacity.

It should be recognized that the power limitation of a loudspeaker may be determined by other factors, such as voice-coil heating or structural damage. However, in general, for a well designed unit, such limits will be well above those defined by distortion.

The signal used for these tests has a 12 db peak factor, hence the instantaneous peak power capacity for the unit shown in Fig. 1 is 12 db higher than the figures on the curve. However it is well to recognize that the limit might occur at a different level if a sinusoidal signal, or a warble tone is used. It is believed that thermal noise is a particularly appropriate signal for the intended purpose because the peak factor is very similar to that of typical program material.

The method of determining power capacity which has been described has the advantage of giving a quantitative result from a relatively simple and rapid method, using input material which is similar to program material in peak factor. Limited experience with the method indicates that it correlates well with listening tests.

It is urged that all those who have a direct interest in Measurement Standards for Loudspeaker Performance consider and use this method, reporting their experience to the IRE Electroacoustic Committee, in the hope that a considered opinion may be derived as to its worthiness for inclusion in the forthcoming Standard.

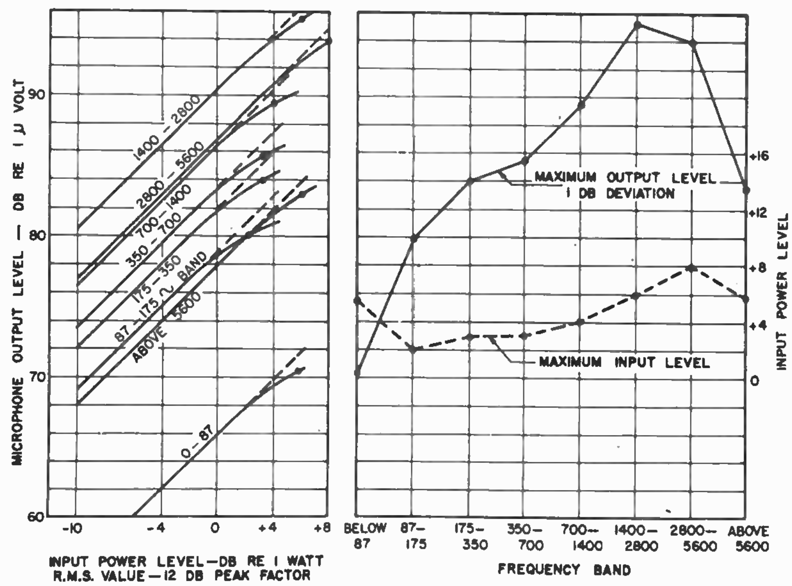

Fig. 1 — Determination of the power capacity of loudspeakers.

SUBJECTIVE LOUDSPEAKER TESTING*

Harry F. Olson
RCA Laboratories
Princeton, New Jersey

INTRODUCTION

A subjective test of a loudspeaker involves a determination of some of the performance characteristics by direct listening to the loudspeaker operating under controlled program and environment conditions. Listening tests play an important part in research, development, and commercialization of loudspeakers. Listening tests range in scope from exceedingly simple comparison tests to elaborately controlled and conducted tests. In view of the importance of subjective testing of loudspeakers, it appears desirable to standardize the procedure. However, discussions with those familiar with establishment of standards indicate that the subject of loudspeaker listening tests has not been carried to the point where standards can be established. On the other hand, it seems timely and appropriate to outline some of the salient procedures used in listening tests of loudspeakers. Accordingly, it is the purpose of this paper to describe listening tests of loudspeakers.

LOUDSPEAKER ENVIRONMENT

The listening test of a loudspeaker should be conducted in the environment for which it was designed to operate. Specifically, a loudspeaker designed for home type radio receivers, phonographs, and television receivers should be tested in a room with dimensions and acoustics similar to those of the average living room in the home. A loudspeaker for an automobile radio receiver should be tested in an automobile. A loudspeaker for a sound motion picture system should be tested in a typical theater. A loudspeaker for a public address, sound reinforcing, or paging system should be tested under typical surroundings for these systems.

LOUDSPEAKER HOUSING, PLACEMENT AND MOUNTING

The horn, baffle, housing, or cabinet for loudspeaker listening tests should be similar to those used under actual operating conditions. The placement and mounting arrangement in the test environment should correspond to those used in actual installations.

SIGNAL SOUND LEVEL

The signal sound level produced by a loudspeaker in a listening test should correspond to the sound level under actual operating conditions in the field. The use of the proper level is very important in determining the balance of high, mid, and low frequencies, the distortion, the transient response, etc., under actual operating conditions. The upper sound levels in the description which follows do not necessarily represent the upper power capabilities of the systems. The signal sound level will be somewhere between 65 and 75 db for a radio receiver, phonograph, or television receiver operating in a typical or average living room. The signal sound level will be somewhere from 65 to 100 db for an automobile radio receiver. The signal sound level will be between 76 to 85 db for speech reproduction, and 75 to 95 db for music reproduction for a sound motion picture reproducing system operating in a theater. The signal sound level for a public address, sound reinforcing, or paging system will vary over wide limits depending upon the application. To summarize, the signal sound level of the test should correspond to the level under actual operating conditions.

AMBIENT NOISE SOUND LEVEL

The ambient noise under which the listening test is conducted should correspond to the ambient noise encountered under actual conditions. This involves two main factors, namely, the sound level and spectrum of the ambient noise. For example, the average ambient noise sound level in the average living room is 42 db. The average ambient noise sound level in a theater is also 42 db. In an automobile, the ambient noise sound level depends upon the speed, open or closed windows, the road, etc. In public address, sound reinforcing or paging applications, the noise sound level will vary over wide limits. This must be taken into account, and the noise conditions under which the equipment will be operated must be simulated in the listening tests. It is important that the spectrum of the noise encountered under the actual operating conditions should be simulated in the subjective tests as well as the noise level.

SIGNAL OR PROGRAM MATERIAL

The signal or program material used in listening tests should be similar to that encountered in the field. This is not so for the material presented under "Frequency Range" and "Power Handling Capacity." A radio or television receiver should be operated from typical broadcast or television transmitters. Under certain conditions it may be necessary to use the equivalent of a radio or television transmitter, as, for example, a modulated signal generator. A phonograph should be operated from

*Manuscript received July 22, 1953.

typical commercial records. A sound motion picture reproducing system should be operated from typical sound motion picture film. Sound reinforcing systems for use with music should be tested with musical program material. Public address and paging systems should be tested with speech as the program material.

REFERENCE SYSTEMS

Almost all listening tests on loudspeakers are conducted by comparing the loudspeaker under test with a reference loudspeaker. The reference system is, in general, a loudspeaker which is similar to the loudspeaker under test. The loudspeakers should be placed behind a light-opaque, sound-transparent curtain so that it is impossible to identify the loudspeakers by sight. A suitable indicator should show which loudspeaker is operating at any time. In general, the procedures in most listening tests are not formalized because the tests are conducted to determine the engineering and commercial aspects of rather small changes in design. If a jury type procedure is used, secret ballots should be taken of the preference. Statistical methods should be employed in planning and conducting such jury tests.

RELATIVE LOUDNESS EFFICIENCY

The relative loudness efficiency of a loudspeaker is determined from a loudness balance. High quality transformers should be used to match each loudspeaker to the appropriate impedance. In some cases it is desirable to include the driving means in determining the efficiency, because this is important in any practical design. The input to the loudspeakers should be adjusted so that the loudness levels of all loudspeakers are the same. The attenuation required to adjust to the same loudness gives a measure of the relative loudness efficiency. In these tests, the observers should move around to different locations to insure that no advantages are given to any loudspeaker due to a better listening location. For the same reason, the locations of the loudspeakers should also be interchanged.

RELATIVE DIRECTIVITY

The relative directivity of a loudspeaker is determined by listening at observation points removed from the axis. In order to reduce the effect of the difference in the angle during a comparison, the following precautions should be observed: Only two loudspeakers should be used at a time. The loudspeakers should be placed as close together as possible. The position of the two loudspeakers should be interchanged during the test. In determining the relative directivity, listening tests should be conducted along different angles with respect to the axis. This test indicates the loss in loudness level and frequency discrimination for observation points removed from the axis.

FREQUENCY RANGE

The approximate frequency ranges of loudspeakers may be determined from listening tests by employing program material which has a wider frequency range than the loudspeaker under test in combination with calibrated high and low pass filters introduced between the program source and the loudspeaker. It is very important that the program material contain adequate frequency components in both the high and low frequency ranges and thereby insure reliable results. The approximate frequency range can be determined by noting the settings of the filters for which there is no appreciable frequency discrimination, as determined by the quality of reproduction. The filters should have at least three cut-off steps per octave.

POWER HANDLING CAPACITY

The power handling capacity of a loudspeaker may be determined by employing a low distortion program source capable of overloading the loudspeaker without introducing distortion in the program source which is fed to the loudspeaker. The frequency range of the system which feeds the loudspeaker should be restricted by means of filters to correspond to that of the loudspeaker under test. The power level at which the distortion becomes intolerable may be considered to be the power handling capacity of the loudspeaker. In this connection intolerable distortion depends upon the application in which the loudspeaker is to be used. This requires a high order of judgment by the listener.

The test outlined above for determining the power handling capacity may appear to be oversimplified in view of the many factors involved. For example, the power handling capacity of a loudspeaker may be determined by failure of the diaphragm, the suspension system, the voice coil structure and heating of the voice coil. Of course, all these forms of failure will be manifested as intolerable distortion.

Again it should be emphasized that the crux of this test is the determination of what is considered intolerable distortion.

RESPONSE FREQUENCY CONTOUR

In most completely integrated systems, such as radio and television receivers and phonographs, there are distinct economic and technical advantages in employing components which individually do not exhibit a uniform response frequency characteristic but taken collectively do exhibit a uniform response frequency characteristic. In these applications, listening tests are very useful in checking the objective measurements for a proper balance of the frequency characteristic. This type of listening requires great skill obtained through practice. A reference system which is known to be acceptable is almost a necessity in tests of this type.

NON-LINEAR DISTORTION

Loudspeakers are used with other components in a sound reproducing system. Therefore, in a properly integrated system the limitations upon the allowable non-linear distortion of each element depends upon the allowable distortion of the system as a whole. For example, it would be technically and economically unsound to use a wide-range, high-quality loudspeaker in a reproducing system in which the components in the remainder of the system were of much lower quality. The quality of the loudspeaker required for the application can be determined from listening tests of loudspeakers of various degrees of quality. In this way it is possible to determine the loudspeaker which introduces distortion of such magnitude as to be perceptible above the distortion of the remainder of the system.

GENERAL ASPECTS

In most conventional, mass-produced, complete sound reproducing systems, technical compromises must be made in order to obtain a product which is commercial from economic considerations. The principal factors which are involved from a subjective standpoint are frequency range, response frequency contour, directivity, non-linear distortion, power handling capacity, and noise. For example, the objectionable effects of non-linear distortion and noise are reduced as the high frequency cutoff is reduced. On the other hand, some of the naturalness of a restricted frequency range system can be regained by a change of the response frequency contour. Listening tests are very useful for obtaining a practical compromise between these factors.

AIR-CORE COIL DESIGN FOR CROSSOVER NETWORKS *

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Cincinnati, Ohio

Much has been written on the several various crossover network circuits in general use, including equations for computing the values of the required elements. Insofar as the inductors are concerned, however, not much is said regarding how to achieve the desired value. Some information has been published by University Loudspeakers, Inc. The following investigation was made to determine the proper wire size and the optimum dimensions for an inductor to have a specified Q and inductance L. (Q in this analysis involves only the dc portion of the resistance.) An equivalent method for specifying the problem is, of course, in terms of inductance L and maximum allowable dc resistance.

A simplified equation for inductance of an air-core inductor is given by Wheeler.[1]

$$L = \frac{2 \times 10^{-7} A^2 n^2}{3A + 9B + 10C}$$

where L = inductance in henrys
n = number of turns
A = mean turn diameter, inches
B = winding width, inches
C = radial winding depth, inches

The number of turns for a random wound coil is

$$n = BC\phi$$

where ϕ = turns possible/square inch. This quantity is tabulated in Terman's *Radio Engineers' Handbook*. Thus, the general equation for inductance is

$$L = \frac{2 \times 10^{-7} A^2 B^2 C^2 \phi^2}{3A + 9B + 10C}$$

The dc resistance of the winding is

$$R = \left(\frac{\pi A}{12}\right)\left(BC\phi\right)\left(\frac{\rho}{1000}\right)$$

where ρ = ohms/1000 ft.

Based upon this dc resistance the coil Q is

$$Q = \frac{\omega L}{R} = \frac{2.4 \times 10^{-3} ABC\phi\omega}{\pi\rho(3A + 9B + 10C)}$$

For a frequency of 1000 cps

$$Q_{1kc} = \frac{4.8\, A\, B\, C}{(3A + 9B + 10C)} \; \frac{\phi}{\rho}$$

This function is to be maximized, with the three variables of dimension subject to the constraining equation of inductance. Before doing so, however, several

* Manuscript received August 17, 1953.

[1] H. A. Wheeler, "Simple inductance formulas for radio coils," *Proc. I.R.E.* vol. 16, p. 1398; October, 1928.

simplifications may be made. The wire function $\frac{\phi}{\rho}$ is essentially monotonic (the values of ϕ are based upon average commercial practice) so there is no optimum wire size. Furthermore, much simplicity is gained by normalizing the dimensional variables.

$$Q_{lkc} = \frac{4.8\,\phi}{270\,\rho}\;\frac{(3A)\;(9B)\;(10C)}{3A + 9B + 10C}$$

$$L = \frac{2\times 10^{-7}\,\phi^2}{(270)^2}\;\frac{(3A)^2\;(9B)^2\;(10C)^2}{3A + 9B + 10C}$$

and upon putting

$$\alpha = 3A$$
$$\beta = 9B$$
$$\gamma = 10C$$

the equations may be expressed as

$$Q' = \frac{Q_{lkc}\,270\,\rho}{4.8\,\phi} = \frac{\alpha\beta\gamma}{\alpha+\beta+\gamma}$$

$$L' = \frac{(270)^2\,L}{2 + 10^{-7}\,\phi^2} \quad \frac{\alpha^2\,\beta^2\,\gamma^2}{\alpha+\beta+\gamma}$$

Now Q' is maximized, using the method of Lagrange for constrained maxima, by forming the partial derivative equations

$$Q'_\alpha - \lambda L'_\alpha = 0$$
$$Q'_\beta - \lambda L'_\beta = 0$$
$$Q'_\gamma - \lambda L'_\gamma = 0$$

where λ is the Lagrange multiplier.
For α there results

$$\frac{(\alpha+\beta+\gamma)\,\beta\gamma - \alpha\beta\gamma}{(\alpha+\beta+\gamma)^2} - \lambda\,\frac{(\alpha+\beta+\gamma)\,2\alpha\,\beta^2\gamma^2 - \alpha^2\beta^2\gamma^2}{(\alpha+\beta+\gamma)^2} = 0,$$

and since both

$$\alpha+\beta+\gamma \neq 0,$$

and

$$\beta\gamma \neq 0,$$

this simplifies to

$$\beta + \gamma - \lambda\alpha\beta\gamma(\alpha + 2\beta + 2\gamma) = 0$$

Because of the symmetry of the equations L' and Q' the other partial derivative equations are obtained *mutatis mutandis*.

$$\beta + \gamma - \lambda\alpha\beta\gamma\,(\alpha + 2\beta + 2\gamma) = 0$$
$$\alpha + \gamma - \lambda\alpha\beta\gamma\,(2\alpha + \beta + 2\gamma) = 0$$
$$\alpha + \beta - \lambda\alpha\beta\gamma\,(2\alpha + 2\beta + \gamma) = 0$$

The multiplier is readily obtained by addition

$$2\,(\alpha + \beta + \gamma) - \lambda\alpha\beta\gamma 5\,(\alpha + \beta + \gamma) = 0$$

$$\lambda = \frac{2}{5\,\alpha\beta\gamma}$$

so these equations simplify to

$$-2\alpha + \beta + \gamma = 0$$
$$\alpha - 2\beta + \gamma = 0$$
$$\alpha + \beta - 2\gamma = 0$$

from which it is found

$$\alpha = \beta = \gamma$$

and since

$$L' = \frac{\alpha^2\,\beta^2\,\gamma^2}{\alpha+\beta+\gamma} = \frac{\alpha^6}{3\alpha} = \frac{\alpha^5}{3}$$

so

$$\alpha = \beta = \gamma = \sqrt[5]{3L'}$$

$$3A = 9B = 10C = \sqrt[5]{\frac{(270)^2\,3L}{2\times 10^{-7}\,\phi^2}}$$

$$Q' = \frac{\alpha\beta\gamma}{\alpha+\beta+\gamma} = \frac{\alpha^2}{3}$$

$$Q_{lkc} = \frac{4.8\,\phi\alpha^2}{270\,\rho\;3} = \frac{4.8\phi}{810\,\rho}\left(\frac{3\,(270)^2\,L}{2\times 10^{-7}\,\phi^2}\right)^{2/5}$$

In the following equations, L is in millihenrys.

$$\delta = 3A = 9B = 10C = 64.23\,\left(\frac{L}{\phi^2}\right)^{1/5}$$

$$Q\max_{lkc} = 24.45\,\frac{\phi^{1/5}}{\rho}\,L^{2/5}$$

$$n = 45.84\;\phi^{1/5}\,L^{2/5}\ \text{turns}$$

$$l = 257\,\frac{L^{3/5}}{\phi^{1/5}}\quad \text{feet}$$

$$R = \frac{0.257\;L^{3/5}\,\rho}{\phi^{1/5}}\quad \text{ohms}$$

Since Q has been maximized for a given value of inductance and frequency, the winding resistance is therefore minimized, and the winding length is also minimum.

Fig. 1 is a plot of the dimension equation and the Q equation. Fig. 2 is a plot of the turns equation and the length equation. Fig. 3 is a plot of the resistance equation. In all three figures wire size is the parameter. The following examples illustrate the use of these curves, and the values are as read from them.

OPTIMIZED COIL DIMENSIONS FOR MAX. Q
AND MAXIMUM Q AT 1KC

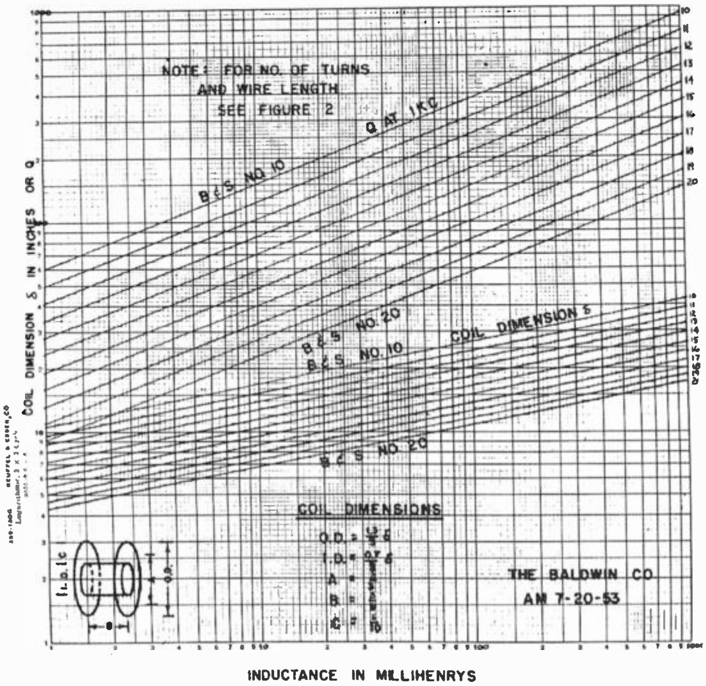

FIGURE 1

(*A*) A 30 mh inductor is to have a Q of at least 100 at 1 kc. From Fig. 1, B & S #14 (or larger) is required. Using #14, also from Fig. 1

$$\delta = 15''$$
$$A = 5''$$
$$B = 1\ 2/3''$$
$$C = 1.5''$$

NO. OF TURNS AND WIRE LENGTH FOR AIR CORE
INDUCTORS HAVING OPTIMUM DIMENSIONS
FOR MAXIMUM Q

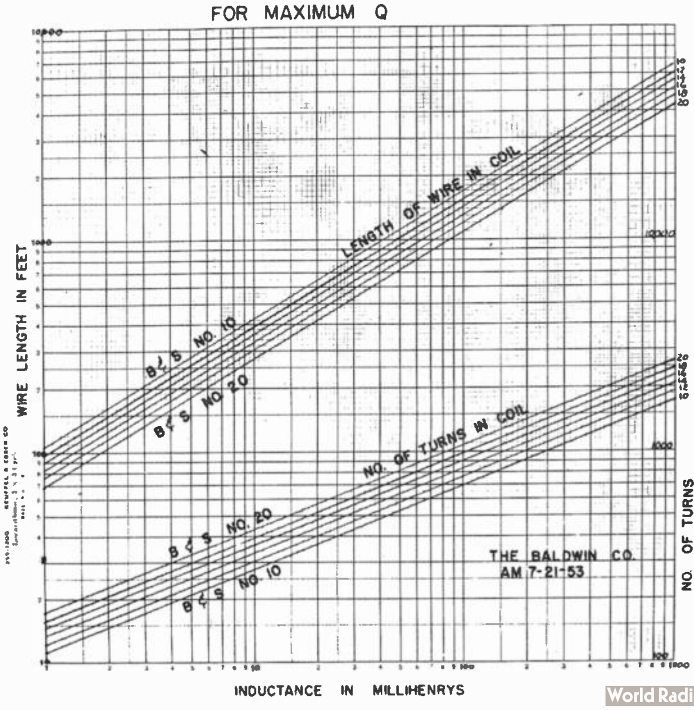

FIGURE 2

$$O.D. = 6.5''$$
$$I.D. = 3.5''$$
$$Q = 110$$

From Fig. 2

$$l = 680 \text{ ft.}$$
$$n = 520 \text{ turns}$$

and from Fig. 3

$$R = 1.7\,\Omega$$

(*B*) A 10 mh inductor is to have a dc resistance of less than 0.5 ohm.

From Fig. 3, B & S #10 will result in $R = 0.42$ ohm. From Fig. 2

$$l = 420 \text{ ft.}$$
$$n = 280 \text{ turns}$$

DC RESISTANCE OF AIR CORE INDUCTORS HAVING
OPTIMUM DIMENSIONS FOR MAXIMUM Q

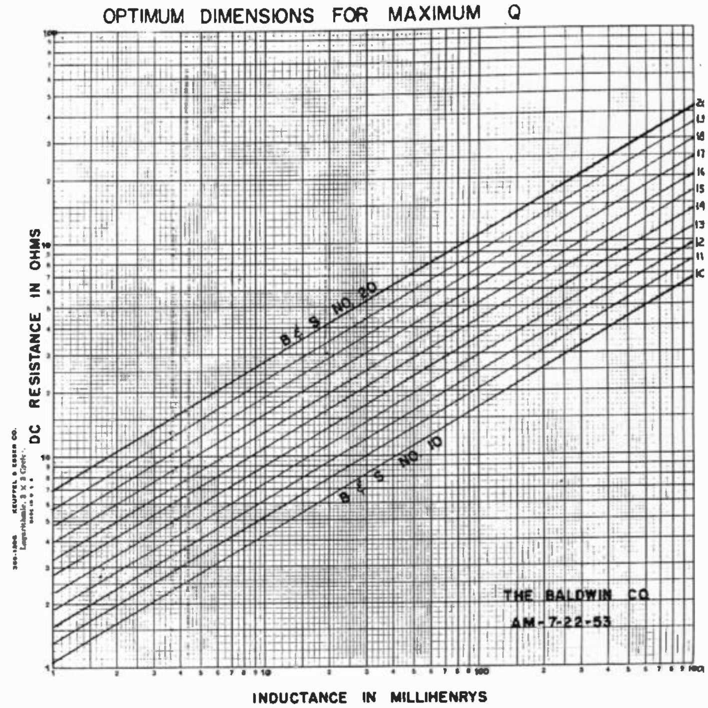

FIGURE 3

From Fig. 1

$$Q_{1kc} = 150$$
$$\delta = 17''$$
$$O.D. = \frac{1.3}{3} \times 17 = 7.37''$$
$$I.D. = \frac{0.7}{3} \times 17 = 3.97''$$
$$B = \frac{17}{9} = 1.89''.$$

INSTITUTIONAL LISTINGS (Continued)

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AMPEX ELECTRIC CORPORATION, 934 Charter Street, Redwood City, California
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THE ASTATIC CORPORATION, Harbor and Jackson Streets, Conneaut, Ohio
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EDUCATIONAL LABORATORIES, INC., 1625 Connecticut Avenue, Washington 9, D. C.
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ELECTRO-VOICE, INC., Buchanan, Michigan
Microphones, Pickups, Speakers, Television Boosters, Acoustic Devices

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

(Please see inside back cover for additional names)

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

November - December, 1953

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Volume AU-1 Number 6

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The Institute of Radio Engineers

IRE PROFESSIONAL GROUP ON AUDIO

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

Annual Assessment: $2.00

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IRE - PGA QUESTIONNAIRE DATA

R. E. Troxel, Chairman
IRE - PGA Committee on Chapters

The IRE-PGA obtained very informative results in its survey of the membership by the questionnaire postal card. Slightly over 600 cards have been returned at this writing, which represents approximately 28% of the total membership.

The summary of the "main interest" column is tabulated below.

	Number of Votes		
	1st Choice	2nd Choice	3rd Choice
High Fidelity	105	89	87
Design	98	70	53
Amplifiers	68	104	70
Hobbyist	59	26	48
Magnetic Recording	56	64	70
Broadcasting	47	18	20
Acoustics	44	53	40
Phonograph Reproduction	39	54	57
Loud Speakers	39	49	64
Microphones	11	32	21

Nearly all of the PGA Chapters were represented in the survey. A tabulation of members reporting from the the various chapters is shown below.

Albuquerque, New Mexico	1
Boston, Massachusetts	24
Chicago, Illinois	23
Cincinnati, Ohio	8
Houston, Texas	7
Kansas City, Missouri	3
Los Angeles, California	12
Milwaukee, Wisconsin	7
Philadelphia, Pennsylvania	30
San Diego, California	8
Seattle, Washington	7

Also, of those returning the cards, 87 were members of the Audio Engineering Society, 58 were members of the Acoustical Society of America, 42 were members of the Society of Motion Picture and Television Engineers, and 9 were members of the Chicago Acoustical and Audio Group.

Of special interest to the PGA Chapters Committee was the significant number of members desiring to become affiliated with a local chapter if one were organized and promoted. Cities showing special interest in this respect are New York City; Washington, D.C.; Toronto, Canada; Baltimore, Md.; Dayton, Ohio; Columbus, Ohio; and Memphis, Tennessee. At this date, some of these Sections are already contemplating organization of PGA Chapters.

All of the above cities have sufficient PGA Members to support local Chapters. The IRE-PGA Chapters Committee will cooperate readily and supply organizational information to any Section or person desiring to promote a new PGA Chapter. Please write to Robert E. Troxel, Chairman, Committee on Chapters, 225 West Huron Street, Chicago 10, Illinois.

REPORT ON STUDY OF CONCURRENT MEMBERSHIP

A. M. Wiggins, Chairman
IRE - PGA Committee on Membership

Recently a comparison has been made of membership lists of the several American societies interested primarily in audio and acoustics. The purpose of the study was to determine whether the membership of the three organizations was largely the same or independent. The results of the study were intended to influence the IRE-PGA policies on publication and other matters. The study reveals the following approximate data:

Total IRE-PGA membership 2000
IRE-PGA members also in Acoustical Society (5.4%) 108
IRE-PGA members also in Audio Engineering (3.4%) 68

All of these figures are subject to continuous revision. In fact the data from the recent IRE-PGA questionnaire do not yield exactly the same indication as the data here. In either case, however, it is clear that the current membership of IRE-PGA is chiefly independent of the two neighboring societies and, consequently, that there is not a substantial overlapping of activities.

Membership in the Professional Group on Audio is a concomitant of membership in the Institute of Radio Engineers. Each Professional Group has a field of interest approved by the Institute Committee on Professional Groups. According to the Constitution for the Professional Group on Audio, Article III, Section I, "The Field of Interest of the Group shall be the technology of communication at audio frequencies and of the audio-frequency portion of radio-frequency systems including the acoustic terminations and room acoustics of such systems and the recording and reproduction from recordings at audio frequencies, and shall include scientific, technical, industrial or other activities which contribute to this field, subject, as the art develops, to additions, subtractions, or other modifications directed or approved by the Institute Committee on Professional Groups."

A clear understanding by the membership of the purpose and organizational plan of the professional group system, data on the relative independence of the membership of IRE-PGA from neighbor societies, and the above definition of the field of interest of IRE-PGA, should assist the membership of IRE-PGA, not only in our own activities and operations but also in friendly cooperation and exchange of technical information with neighbor societies.

ANALYSIS OF READER INTEREST IN TRANSACTIONS

Daniel W. Martin, Chairman
IRE - PGA Editorial Committee

Ten percent of the PGA questionnaire cards returned contained comments and suggestions concerning TRANSACTIONS of the IRE-PGA. These specific, helpful responses from our readers are sincerely appreciated. Together with the larger body of general information contained in the "main interest" data (tabulated elsewhere in this issue), they will influence the policies, plans, and actions of the Editorial Committee.

The most frequent general suggestion was for further expansion of the technical papers section of TRANSACTIONS of the IRE-PGA. The Editorial Committee concurs, and has been working diligently toward that end. The rather slender appearance of the September-October issue was not indicative of our future outlook, and can be attributed to a combination of the following factors:

1. Publication of recent IRE-PGA papers in the CONVENTION RECORD.
2. A new, more compact format.
3. Seasonal reduction.
3. Postponed publication of manuscripts at hand pending necessary approvals.

During the current rapid commercial expansion of the audio field it is increasingly difficult for many of our most experienced authors in audio to find time for the preparation of technical papers. There is greater pressure for "commercial" papers than before and, in addition, they are probably easier to prepare. Your Editorial Committee is trying to meet this problem by more systematic solicitation from known potential authors (which we hope will not lose us friends). At the same time we want to encourage the newer people in the audio field to take advantage of the possibilities of publication of technical papers in TRANSACTIONS of the IRE-PGA. Don't wait to be asked to submit a paper!

Reader comments were much more complimentary than derogatory. If this does not express your viewpoint, "say it where it counts" – to us.

Several have asked why some of the papers from the 1953 Convention have not been published. In three sessions eleven papers were given. All of these were solicited for publication in the CONVENTION RECORD, and five made the deadline. Two more have since been published in TRANSACTIONS. Three authors declined to publish tutorial material which they had presented only by request, and which they had borrowed almost entirely from information published in their own textbooks and handbooks.

Requests were received for abstracts of papers published elsewhere. Your attention is called to the first section of the Abstracts and References department of PROCEEDINGS of the IRE, which, although incomplete, is very helpful along these lines. An attempt is already being made to obtain translations of some foreign audio papers of merit and interest for republication in TRANSACTIONS. Additional suggestions, and especially translations, are welcome. As a matter of information for engineers interested in patents as a source of technical information, many audio patents are reviewed regularly in the JOURNAL of the ACOUSTICAL SOCIETY.

A request was received for a cumulative index for TRANSACTIONS of the IRE-PGA. We are pleased to include such an index in this issue, covering all IRE-PGA material to date. A request for information on audio standards has perhaps been answered already by the current summary in the September-October issue.

With regard to style of technical papers, suggestions for higher standards for articles accepted for publication were matched by requests to "keep it down to earth." These two requirements are not necessarily contradictory.

Suggestions on subjects for future papers were varied. Many were helpful and will be followed up. For obvious reasons we should not attempt to answer the requests for articles on customer guidance, in the purchase of specific brands of equipment. However, the submission of papers similar in approach to the Technical Editorial "Selecting a Loudspeaker," written by Dr. H. F. Olson, is welcomed.

A brief analysis has been made of the types of technical papers and editorials published to date by IRE-PGA, including the present issue. A correlation of the analysis data with the main interest data from the post-card survey follows.

Totaling the first, second, and third place votes for each of the ten categories used for "main interest" yields the second column below. This gives a somewhat different order than is obtained from first choice votes alone, and is justified on the basis that a majority of the readers have a diversified interest in audio which a single choice cannot properly identify.

Each of the technical papers and editorials was classified in the same categories. The results are in column three. In some cases two categories applied. Eight audio papers did not fit into any of the categories listed. Three of these were on electronic music and the others were on audio measurements and measuring equipment.

Main Interest	Total 1st, 2nd, 3rd choice votes	Number of Technical Papers
High Fidelity	281	3
Amplifiers	242	5
Design	221	7
Magnetic Recording	190	8
Loudspeakers	152	16
Phonograph Reproduction	150	7
Acoustics	137	6
Hobbyist	133	1
Broadcasting	85	3
Microphones	64	7

It is interesting to note that all categories are at least represented. The strongest categories of interest are all matched by a reasonable number of papers with the exception of the first one, High Fidelity. Admittedly this is a difficult topic because it is so general. It is subject to various interpretations, and hard to discuss without being proprietary. However technical papers on this subject have been requested recently from two of the most active engineers in the field of high-fidelity system manufacture. If they respond, their papers will receive prompt publication. A comprehensive paper on power amplifiers, and two papers on low-level amplifier design problems are scheduled for publication in forthcoming issues.

In conclusion, a quotation from an editorial in the May-June 1953 issue:

"It is the pleasant responsibility of individual IRE-PGA members to be active in these matters, as well as interested. When some new principle or device or concept has been discovered, or when something well known has been re-examined to new advantage, the membership of IRE-PGA will certainly appreciate first chance to hear about it through TRANSACTIONS of the IRE-PGA."

PGA BRIEFS

A meeting of the IRE-PGA Administrative Committee was held in the Ruby Room of the Hotel Sherman in Chicago on September 28. Among the items discussed were the following:

1. Methods of nomination.
2. Report of the nominating committee (publication postponed until acceptances are received).
3. Liaison between Professional Groups and Technical Committees working in the same general area.
4. Annual assessment.
5. Postcard Survey of Membership (reported elsewhere in this issue).
6. Survey of Overlapping Membership (reported elsewhere in this issue).

A Papers Procurement Committee has been formed which has the following membership:

Arnold Peterson, Chairman
General Radio Company
Cambridge 39, Massachusetts

Melvin Chun
Southwest Research Institute
San Antonio, Texas

M. S. Corrington
RCA Victor Division
Camden, New Jersey

W. Ryland Hill, Jr.
University of Washington
Seattle, Washington

W. M. Ihde
General Radio Company
Chicago, Illinois

F. R. Miller
Pacific Telephone and Telegraph Co.
Portland, Oregon

V. Salmon
Stanford Research Institute
Stanford, California

W. R. Thurston
General Radio Company
New York, New York

The new Chairman of the IRE-PGA Program Committee is Dr. Winston E. Kock, Director of Acoustics Research at Bell Telephone Laboratories. This committee is in charge of the PGA session at the IRE Convention in March, 1954.

The theme of the IRE-PGA program for the 1954 IRE Convention in March will be "High-Fidelity." Day-time sessions and a panel discussion in the evening are being planned. Papers already lined up for the program are in the following fields of PGA interests: microphones, power amplifier, loud speakers, reproduction of music, psycho-acoustics.

The Société des Radioélectriciens, 10 Avenue Pierre Larousse, Malakoff (Seine), in cooperation with other French technical societies, is planning to hold a Congrès International, de L'Enregistrement Sonore (International Meeting on Sound Recording), April 5-10, 1954. Any PGA members planning to attend will please so advise the president of IRE, Mr. J. W. MacRae.

The new address of the Chairman of the IRE-PGA Tapescripts Committee is

Mr. Andrew B. Jacobsen
1802 N. 47th Pl.
Phoenix, Arizona

Mr. Jacobsen is now associated with the Phoenix Research Laboratory of Motorola, Inc.

The new Publicity Chairman for the Chicago Chapter IRE-PGA,

Mr. Wallace H. Coulter
3023 West Fulton St.
Chicago, Illinois

has given a list of meetings, including those recently held, and those scheduled for the near future, as follows:

October 16 – "Stereophonic Sound," William B. Snow, Consultant in Acoustics, Los Angeles, California

November 20 – "An All Transistor Hearing Aid," Stanley K. Webster, Beltone Hearing Aid Co., Chicago

December 16 – "Recent Advances in Compact Loudspeaker Housing," Daniel Plach, Jensen Mfg. Co., Chicago

January 15, 1954 – "Audio Problems Peculiar to Television," George Ives, WBKB, Chicago (Demonstration Trip).

The Dayton, Ohio, IRE Section is engaged in organizing an IRE-PGA Chapter under the guidance of Mr. A. B. Henderson, Vice-Chairman, Dayton Section.

HOUSTON CHAPTER IRE-PGA

L. A. Geddes, Chairman
Baylor University College of Medicine
Houston, Texas

In response to the current enthusiasm for high fidelity sound, the executive committee of the Houston Section of the IRE sponsored two audio meetings. The purpose of the meetings was to examine the interest of the membership with a view to ultimately setting up a chapter of the Professional Group on Audio in Houston.

The first meeting held in November 1952 was unusually well attended. The program consisted of explanations and demonstrations of high fidelity reproducing equipment belonging to various local enthusiasts and members.

Following the demonstrations, the meeting adjourned with an informal get-together. Various members of the audience participated in tests permuting and combining the various pieces of apparatus and evaluating the results.

The success of the first meeting stimulated the executive committee to petition the IRE for permission to form a local chapter of the Professional Group on Audio. However, prior to the formal recognition of the Houston chapter, another meeting was sponsored with a program similar in nature to that of the first.

The second meeting held on May 19th included an election of officers. It was the aim to fully activate the chapter after formal recognition by IRE headquarters.

The program of the second meeting included first a detailed description of folded horn enclosures by D. P. Carlton of Humble Oil Company. Following the descriptions, demonstrations were given of various models of the horns described.

The second part of the program consisted of a PGA tapescript entitled "A Single-Ended Push-Pull Amplifier" by Peterson and Sinclair of General Radio. The tapescript was accompanied by the projection of slides pertinent to the explanation of the operation of the amplifier.

Following the tapescript, Mr. Roy Brougher, IRE Meetings Chairman, demonstrated a modified version of the Single-Ended Push-Pull Amplifier. The demonstration was well received, and much interest was shown in the circuit details.

A message repeater was then demonstrated by Jim Hallenberg, IRE Section Secretary. The message repeater consisted of a miniature tape recorder which repeated any dictated message of less than one-hundred seconds duration.

The Houston chapter of the Professional Group on Audio received recognition from National headquarters July 7th, and plans to hold regular meetings on Audio and allied subjects.

A MINIATURE MICROPHONE FOR TRANSISTORIZED AMPLIFIERS*

B. B. Bauer
Shure Brothers, Inc.
Chicago, Illinois

SUMMARY – A new magnetic microphone has been developed for use in hearing aids and communication and similar equipment which requires a small, light, medium-impedance microphone. The new microphone has a diameter of 1 inch and a thickness of 3/8 inch, and it weighs 9 grams. The sensitivity is high enough to override the relatively high noise level of transistors in the frequency range most useful for voice transmission.

The recent trend toward miniaturization and the use of transistors has created a demand for a microphone suitable for use in portable transmitters, mobile equipment, hearing aids, and the like, with the following combination of characteristics:

1. Small size and weight
2. High output
3. Frequency range suitable for intelligible transmission of speech
4. Satisfactory match to a load in the vicinity of 1000 ohms.
5. Ruggednes and reliability

In appraising these characteristics against the available methods of transduction, it was decided that they would be best fulfilled by a magnetically balanced moving armature structure.

A modified moving armature structure previously used in several military and civilian applications was adapted for this purpose. This structure has had a long production history, but it has not been previously described outside of its U.S. Patent.[1]

The principal structural differences between a conventional moving armature transducer and the new modified form are shown in Fig. 1. The conventional transducer, at the left, has a voice coil which is contained within the pole piece structure. The to-and-fro motions of the armature set up a differential flux through the armature and pole pieces as shown by the dash arrows. This conventional structure is beset by two handicaps: first, the ac flux component travels a relatively long path through the pole pieces which are highly biased by the dc flux component. Secondly, the relatively large volume of the coil surrounded by the pole pieces results in a substantial amount of leakage flux which, in turn, requires the use of heavy cross-section of iron and a large magnet, whereby the size and weight of the structure are adversely affected.

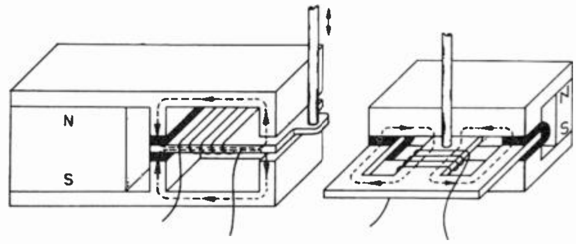

Fig. 1 – Comparison of conventional moving armature structure (left) with the new structure.

In the new unit, at the right, the coil is moved outside the magnetic structure. Therefore, the volume embraced by the pole pieces can be decreased, resulting in less leakage and allowing for a smaller magnet and thinner pole pieces. Additionally, the ac flux path in the pole pieces is shortened considerably.

The magnetic circuit of the new structure is shown schematically in Fig. 2. Stationary gaps at the two side legs taken jointly have a reluctance R_o which is the same as the steady state reluctance R_o of the gap at the moveable center leg. Assuming, for the sake of simplicity,

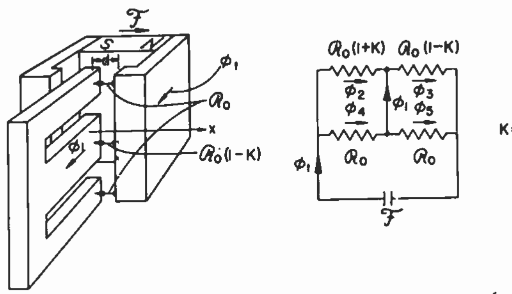

Fig. 2 – Approximate equivalent electric circuit of magnetic structure.

that all the reluctance resides in the air gaps, the equivalent electrical circuit of the magnetic structure is shown at the right hand side. R_o's form the arms of a bridge. ϕ_t represents the total flux due to the magneto-motive force F. ϕ_1 is the flux through the center leg, which in actual practice is surrounded by the voice coil. It is evident, that ϕ_1 is O in the "at rest" position of the armature when the reluctances of the air gaps are equal. When the center leg of the armature moves a distance x to the right, and defining $k = x/d$, the reluctance of the left gap becomes $R_o(1 + k)$ and that of the right gap becomes $R_o(1 - k)$. ϕ_1 can be calculated by the following equations:

$$\phi_1 = \frac{F\,(2 - k)}{2R_o\,(2 - k^2)} \qquad (k = x/d) \qquad (1)$$

*Manuscript received June 17, 1953. Also published in Journ. Acoust. Soc. Amer., vol. 25, p. 867; September, 1953.

[1]B. B. Bauer, U.S. Patent 2,454,425, November, 1948.

$$\phi_3 = \frac{F\,(2+k)}{2R_o\,(2-k^2)} \tag{2}$$

$$\phi_1 = \phi_3\ \phi_2 = \frac{F}{R_o}\cdot\frac{k}{(2-k^2)} \approx \frac{F}{2R_o}\,k = \frac{F}{2R_o}\,\frac{x}{d}$$

$$(x << d) \tag{3}$$

From (3), it is seen that for small motions of the armature, ϕ_1 is directly proportional to the armature displacement. The voltage induced in the coil equals:

$$e = n\frac{d\,\phi_1}{dt} \times 10^{-8} = n\,\frac{F\,10^{-8}}{2R_o d}\,\frac{dx}{dt} \text{ volts} \tag{4}$$

From (4), one might assume that the sensitivity of the structure can be increased indefinitely by increasing the polarizing flux, or decreasing the air gap distance d. It is well-known, however, that this is not the case because of mechanical instability which develops above a given level of polarization. Let us analyze approximately the nature of this instability. The forces F_2 and F_3 at the left and right air gaps of the center leg may be calculated from the equation

$$F_n = \phi_n^2/8\pi A$$

Substituting ϕ_2 and ϕ_3 into this expression, the following is obtained:

$$F_2 = \frac{\phi_2^2}{8\pi A} = \frac{F^2\,(2-k)^2}{32\pi A R_o^2\,(2-k^2)^2} \text{ dynes} \tag{5}$$

$$F_3 = \frac{\phi 3^2}{8\pi A} = \frac{F^2\,(2+k)^2}{32\pi A R_o^2\,(2-k^2)^2} \text{ dynes} \tag{6}$$

and

$$F_1 = -(F_3 - F_2) = -\frac{F^2\,x}{16\pi A d R_o^2\left(1-\dfrac{x^2}{2d^2}\right)^2} \text{ dynes} \tag{7}$$

The net force F_1 is negative because it is opposed to the direction of armature displacement. This gives rise to a negative stiffness S_1 which can be obtained by differentiating F_1 with respect to x. With the armature centrally disposed, the following result is obtained:

$$S_1 = \frac{dF_1}{dx} = -\frac{F^2}{16\pi A d R_o^2} \text{ dynes/cm.} \tag{8}$$

Whenever the negative magnetic force F_1 exceeds the mechanical restoring force of the armature, or the negative stiffness S_1 exceeds the positive stiffness of the armature, a condition of instability will result. It is found that the negative stiffness S_1 is related to the voltage generated e by a factor of proportionality which includes the polarizing flux ϕ_t. Therefore, sensitivity and instability go hand-in-hand, and hence, all the factors of the magnetic transducer must be carefully balanced in a manner which must be left to the judgment and the skill of the designer.

The component parts of the microphone are shown in Fig. 3. At the upper left is the E-shaped armature which is made of high permeability material. The two non-magnetic spacers and the Y-shaped drive unit are attached

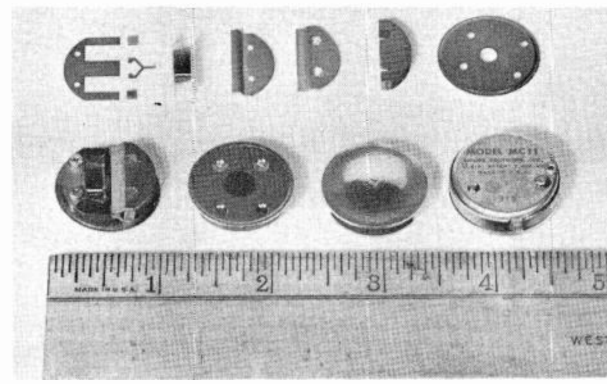

Fig. 3 – Component parts and assembly of magnetic microphone.

to the end of the armature legs after installation of the bobbin. Thereupon, the armature, the two pole pieces, and the magnet are assembled to the circular base plate. At the lower left is the completed motor. Turning it over, we see attached a piece of damping fabric and the drive unit protruding through a hole in the fabric. Lastly, the diaphragm is attached and cemented to the drive unit, and the motor assembly is placed in an aluminum case. The complete structure is 1" diameter, ⅜" thick, and it weighs approximately 10 grams.

The cross section of the microphone is shown at left in Fig. 4 and at right is the equivalent circuit. The sound pressure P_1 enters the front chamber through a

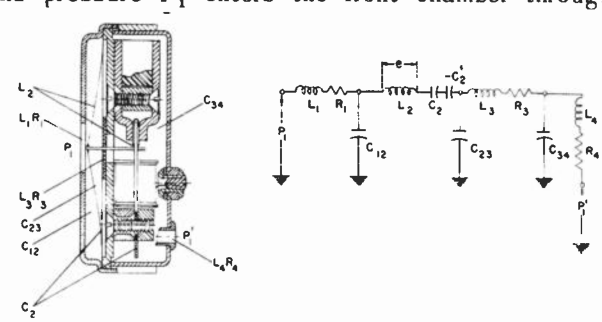

Fig. 4 – Equivalent electrical circuit of the acoustical structure of the new microphone (classical or impedance analogy).

hole in the front cover which defines a resistance R_1 and an inductance L_1. The chamber has an acoustic compliance C_{12}. The diaphragm and the armature have a joint acoustic mass L_2 and acoustic compliance C_2. To this, we must add the acoustic compliance due to the negative magnetic pull $-C_2' = A^2/S_1$, where A is the diaphragm area. The motion of the diaphragm compresses the air in the chamber between the diaphragm and the mounting plate which defines the compliance C_{23}. This causes a flow of fluid through the fabric in the center hole defining an inductance L_3 and a resistance R_3 into the back volume which has a compliance C_{34}. A Thuras type resonator tube defining L_4 and R_4 is used to couple this latter compliance with the outside pressure P_1', and it serves to improve the low frequency response.

The particular unit here described was designed to meet a specification which requires a rise of response at about 3 kc, low frequency response to 400 cps, combined with a minimum volume. Speaking in general terms, the mesh $L_1R_1C_{12}$ provides a resonant rise at 3 kc. The mesh $R_4L_4C_{34}$ provides a rise at around 400 cps, and the central portion of the network provides a slight hump at 1500 cps. The response frequency characteristic typical of production units is shown in Fig. 5. Because the impedance of the voice coil is inductive, the open circuit response has been adjusted to provide a generally rising characteristic so that when the unit is connected to a 1000 ohm load, a slightly rising over-all characteristic is obtained. The output into 1000 ohms is approximately 70 db re 1 volt/microbar which is equivalent to 10^{-10} watts for a 1 microbar signal. This is as much sensitivity as is obtained from many conventional microphones several times the size of this particular unit.

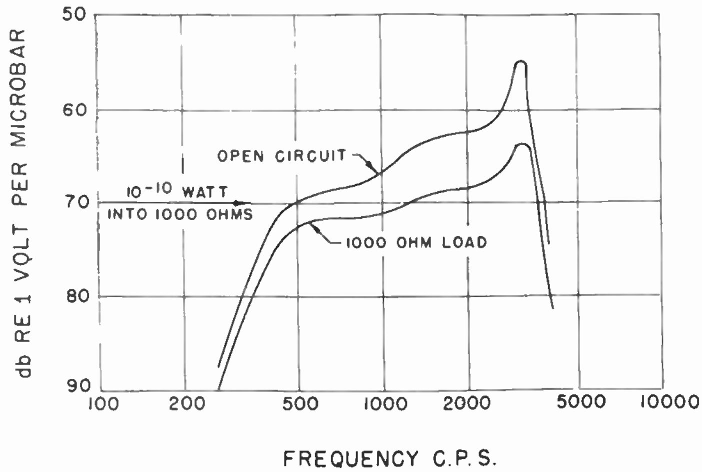

Fig. 5 – Response-frequency characteristic of the new microphone.

CONCLUSION

By suitable proportioning of the magnetic circuit, and by taking advantage of resonant effects in acoustic meshes, it is possible to construct a magnetic microphone which is small, light, stable, and with excellent sensitivity in the frequency range most important for voice transmission.

ACKNOWLEDGMENT

The new miniature magnetic microphone was developed with the participation of Mr. E. V. Carlson and Mr. R. Carr.

A MINIATURE PIEZOELECTRIC MICROPHONE*

John Medill
Shure Brothers, Inc.
Chicago, Illinois

SUMMARY – A new Rochelle salt microphone has been developed for use as a secondary standard in production testing, sound level measurements, high quality transmission of sound, and similar applications. The diameter of the new microphone (1¼ inch) is half the diameter of similar units in current use. Frequency response, directional characteristics, and moisture protection have been improved without a significant loss in sensitivity.

The practical study of acoustics requires that measurements be made. Whether the specific problem at hand involves frequency-response determinations, production testing, ambient noise-levels, reverberation measurements, or any of a host of other possibilities, some type of microphone is required. Some measurements, of course, require a degree of precision which can be obtained only by the use of a primary standard microphone. But such microphones are expensive, and must be very carefully handled and maintained, if their calibrations are to remain accurate.

However, for many applications, a secondary standard microphone is quite adequate. Among the desirable attributes of such a microphone are: (1) uniform frequency response; (2) high sensitivity; (3) high acoustic impedance; (4) small size; and (5) freedom from electrostatic and electromagnetic induction. Other important characteristics are portability, and relative low cost. For many years these requirements have been met by the Shure 98-98 microphone, which has been an accepted secondary standard, and has proved a valuable tool in this field. The purpose of this paper, though, is to introduce the type 98-99 microphone, which embodies many improvements over the 98-98.

*Manuscript received June, 19, 1953. Also published in Journ. Acoust. Soc. Amer., vol. 25, p. 864; September, 1953.

In Fig. 1 can be seen a photographic comparison of the two microphones, with the newer 98-99 on the right. The smaller size and lighter weight of the new model make it easier to handle and use, in addition to the advantage of its reduced diffraction effects. Model 98-99 also has an improved frequency-response and greater protection against the effects of humidity, which make this model even more useful than was its predecessor.

Fig. 1 – Comparison of 98-98 and 98-99 microphones.

Construction of the 98-99 "cartridge" is shown in Fig. 2. At the extreme left the foil-wrapped crystal, in its mounting bracket, can be seen inside the "cartridge" case, with the drive-arm extending upward from the unsupported corner of the crystal. This crystal is a Rochelle-salt torsion bimorph, and is supported on three corners by its mounting bracket. The second view shows the diaphragm in position, with the drive-arm projecting through

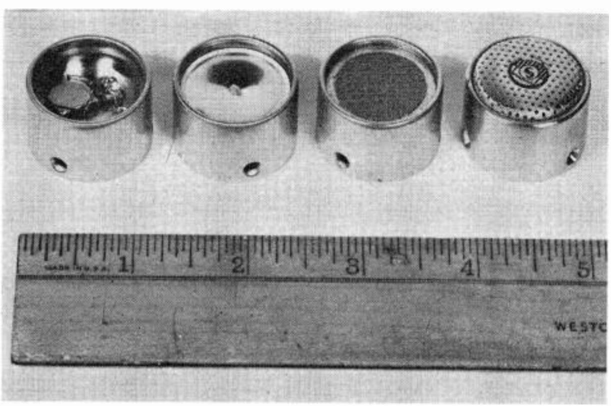

Fig. 2 – Construction of 98-99 microphone.

its center. Around the outer edge of the diaphragm is a spacer, which provides clearance between diaphragm and damping plate. This damping plate, with damping cloth in place, is shown in the third view, while, at the extreme right, a completed "cartridge", with grille, is shown.

It should be noted that the design of this microphone is considerably different from the usual crystal microphone, intended for public address and home use. The "cartridge" is a heavy die-casting and the damping plate is of heavy gauge metal, especially intended to eliminate resonances within the frequency range of the instrument.

Without the damping plate, the resonant frequency of the system is about 5 kc. Since a "flat" response is desired, the damping plate is provided with an appropriate resistance, coupled into the diaphragm by the small volume between diaphragm and damping plate. Mechanical constants of the crystal, diaphragm, damping cloth, and the volume between damping plate and diaphragm, are the principal factors which determine the frequency-response of the microphone.

A diagrammatic cross-section and an equivalent circuit of the 98-99 "cartridge," in terms of acoustical constants, is shown in Fig. 3. The diaphragm mass and compliance are respectively M_d and C_d, while those of the crystal are signified by M_c and C_c. The volume of air at the rear of the diaphragm has a compliance C_r. The

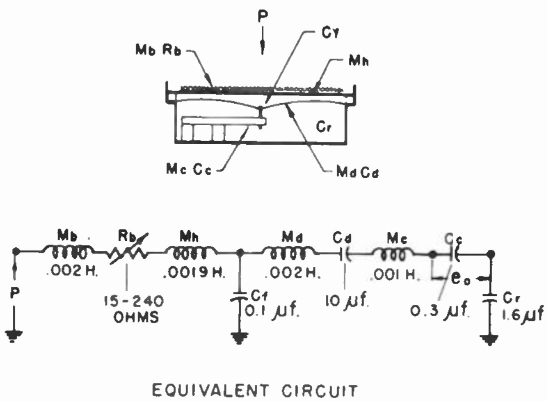

Fig. 3 – Equivalent circuit of 98-99 microphone.

stress developed in the crystal is represented by the voltage drop E_o across the condenser C_c. If this system were not damped in some manner, it would exhibit a sharp peak at 5 kc. To flatten this peak, the damping plate is installed, providing a damping impedance, and the sound pressure P acts upon the diaphragm through this damping plate. The symbols M_b and R_b refer to the acoustic mass and resistance of the damping cloth. M_h is the mass of air in the damping plate holes, together with end effects. C_f represents the acoustic compliance of the air in front of the diaphragm. By altering R_b, any degree of damping may be obtained.

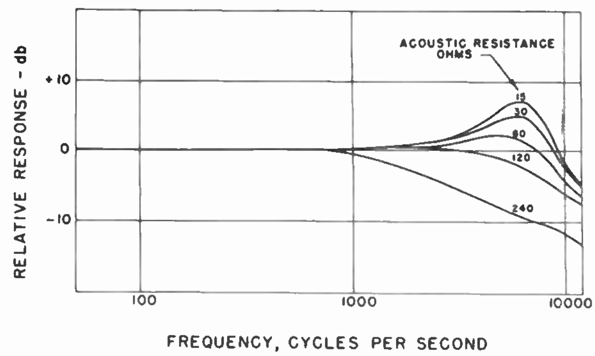

Fig. 4 – Effect of damping-resistance, R_b.

A family of transmission curves, run on an electrical analog of this microphone, is shown in Fig. 4. The numbers 15 through 240 refer to the equivalent damping resistance in acoustical ohms. From the curves can be seen the effect on frequency-response of the various damping resistances used. For instance, with only 15 ohms of damping resistance, the resonant peak is about 8 db high, while a resistance of 60 to 120 ohms provides a substantially flat curve. Actually, the 98-99 model uses damping of 120 ohms; however, the response curve is flatter than that shown in Fig. 4 because of the effects of diffraction, which are best determined by experimenting with the model.

In Fig. 5 is shown the actual frequency-response of the 98-99 microphone, for different angles of sound incidence.

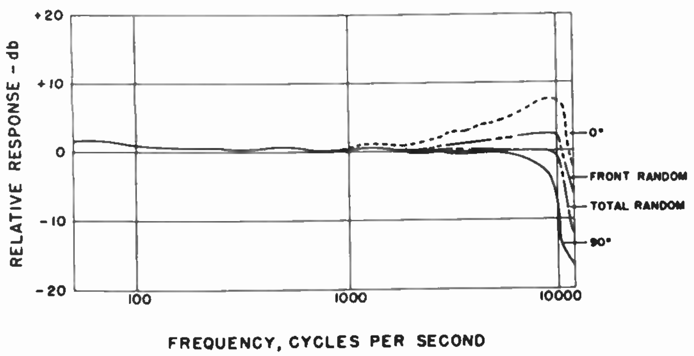

Fig. 5 – Typical response-frequency characteristic of 98-99 microphone.

The dotted line represents the axial response of the unit and exhibits the typical rise at higher frequencies due to diffraction and reflection, while the solid line indicates response at 90° incidence. Also shown, however, are curves of front random incidence response, and of total random incidence response. The front random efficiency is as defined by American Standard Z24.3-1944, Sound Level Meters, etc., and integrates over only the front hemisphere. This response is indicated by the next-to-highest curve. On the other hand, the curve representing total random efficiency is integrated over both the front and rear hemispheres, and hence gives a more accurate indication in a random sound field.

Protection of Rochelle salt crystals from effects of extremes of himidity has always been a problem. In the older Model 98-98, protection of the crystal is accomplished by the use of a special wasing process, which, though complicated, does provide a satisfactory life, even under conditions of high humidity. More complete humidity protection is supplied in the new 98-99 microphone, since it uses a further improvement, a foil-wrapped crystal. A summary of extensive moisture-proofing tests is shown in Fig. 6. In these tests, the crystals, with various types of treatments, were exposed to a constant humidity of 93%, at a temperature of 100° Fahrenheit. This relative humidity was obtained in sealed chambers containing a saturated solution of ammonium dihydrogen phosphate, with an excess of the salt in solid phase. This is a very

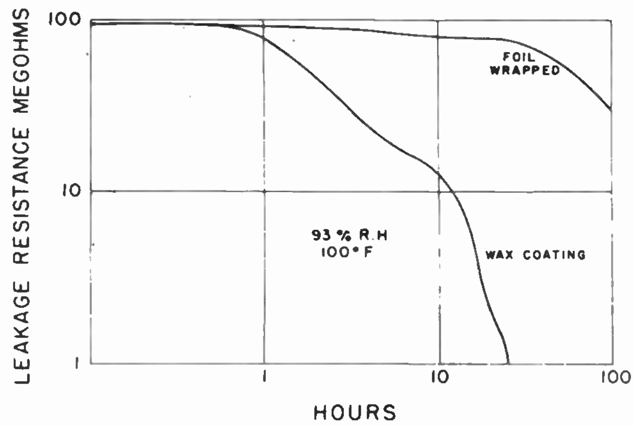

Fig. 6 – Comparison of moisture-protection methods.

rigorous method, and it represents a greatly accelerated life test. The wax-protected crystals averaged about 30 hours before reaching a leakage resistance of 1 megohm. The foil-wrapped crystals, after 100 hours exposure, still had leakage resistance in the order of 35 megohms.

Measurements of high sound intensities require that the microphone output be linear with respect to the sound level. Fig. 7 shows the linearity of the type 98-99 microphone, carried out to 10,000 microbars, or 154 db

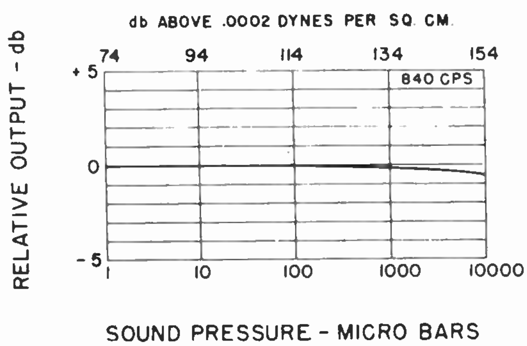

Fig. 7 – Linearity of 98-99 microphone.

above 0.0002 dynes per square centimeter. It will be noted that the departure from linearity is 1/2 db, which approaches the limit of accuracy of our measurements.

An interesting variation of the 98-99 is the general purpose type 777 microphone shown in Fig. 8. As can be seen in the photographs, a somewhat longer handle has been provided, plus several accessories which adapt it for use on a stand. There are some other differences which are desirable from the standpoints of the fields of application of a general purpose microphone. These differences are in such things as diaphragm thickness, damping, and frequency-response. To give a slight "lift" to

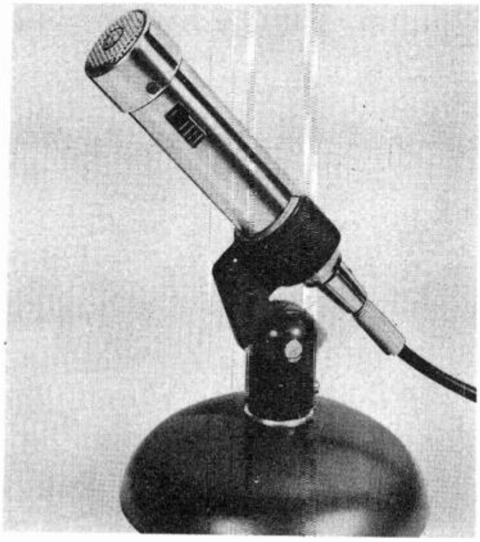

Fig. 8 – Model 777 "Slim-X" microphone, with swivel-type stand.

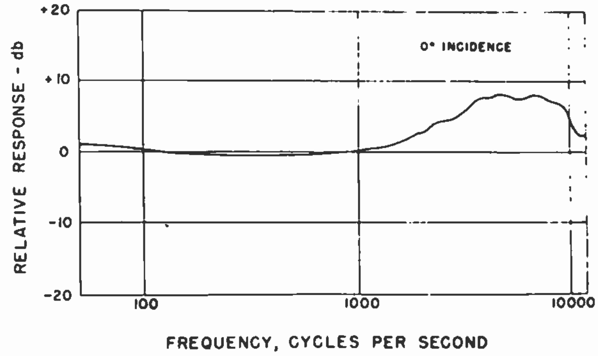

Fig. 9 – Typical response-frequency characteristic of 777 microphone.

the high frequencies, the damping resistance has been reduced to 70 ohms. This results in a rise at high frequencies which has been found of value in public address and communications uses. This is shown in Fig. 9, which is a typical axial response curve of the Model 777 microphone.

ACKNOWLEDGMENT

The author wishes to express his appreciation to B. B. Bauer for his many helpful suggestions in the preparation of this paper.

A LOW-FREQUENCY SELECTIVE AMPLIFIER*

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SUMMARY – A selective feedback amplifier which is useful at sub-audio frequencies is described. It is shown that it has the same characteristic as a simple resonant circuit and that the selectivity is improved by an increase in gain. The measurements on an actual amplifier are presented; they show close agreement with the theory.

INTRODUCTION

At audio and sub-audio frequencies it is highly impractical if not impossible to construct a frequency selective amplifier using inductances. The methods actually used at present utilize either a null network (Wien bridge, parallel T, bridged T, etc.)[1,2] in a negative feedback circuit or a regenerative amplifier[3,4,5] on the verge of oscillation. The first requires inconveniently large components for very low frequencies and the second is extremely sensitive to normal variations in amplifier gain. This paper described a selective amplifier which uses components of reasonable size and which is considerably more stable than a regenerative amplifier.

THEORY

Consider the familiar parallel resonant circuit of Fig. 1. Its driving point impedance is

$$Z = \frac{pL}{p^2 LC + pL/R + 1} \tag{1}$$

* Manuscript received August 24, 1953.

** Now with the Ralph M. Parsons Company, Pasadena, California.

[1] G. E. Valley and H. Wallman, "Vacuum Tube Amplifiers," McGraw Hill Book Co., Inc., New York, N.Y., pp. 384-408; 1948.

[2] E. A. G. Shaw, "A Tunable Audio Frequency Amplifier of variable Selectivity, "*Journ. Sci. Instr.*, vol. 27, pp. 295-298; November, 1950.

[3] C. C. Shumard, "Design of high-pass, low-pass, and band-pass filters using RC networks and direct current amplifiers with feedback," *RCA Rev.*, vol. 11, pp. 534-564; December, 1950.

[4] O.G. Villard, Jr.; "Independent control of selectivity and bandwidth," *Electronics*, vol. 24, pp. 121-123; April, 1951.

[5] C. H. Miller, "RC amplifier filters," *Wireless Eng.*, vol. 27, pp. 26-29; January, 1950.

Substituting $w^2_o = 1/LC$ and $Q_o = R/w_oL = Rw_oC$ then (1) becomes

$$Z = \frac{pL}{p^2/w^2_o + p/w_oQ_o + 1} \qquad (2)$$

where $p = jw$, w_o is the resonant (angular) frequency and Q_o is the resonant Q.

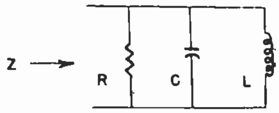

Fig. 1 – Parellel resonant circuit.

Now consider the voltage gain of the circuit in Fig. 2.

$$K = E_o/E_i = \frac{-A_1A_2T_1p}{(1 + A_1A_2)T_1T_2p^2 + (T_1 + T_2)p + 1} \qquad (3)$$

where $T_1 = R_1C_1$ and $T_2 = R_2C_2$. Also, either A_1 or A_2 (not both) can have the negative sign associated with it.

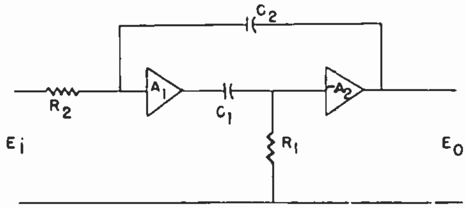

Fig. 2 – Block diagram of selective amplifier.

Notice that (3) has exactly the same form as (2). This means that the gain of the circuit in Fig. 2 has exactly the same frequency characteristic as the impedance of the circuit in Fig. 1. That is, K exhibits the same resonance curve as Z.

By direct analogy to (2), we set

$$w^2_o = \frac{1}{(1 + A_1A_2)T_1T_2} \qquad (4)$$

and

$$Q_o = \frac{\sqrt{T_1T_2(1 + A_1A_2)}}{T_1 + T_2} \qquad (5)$$

If Q_o is differentiated with respect to T_1 (or T_2) and the derivative is set equal to zero, it is found that Q_o has a maximum when $T_1 = T_2$. Setting $T_1 = T_2$ in (5) we get

$$Q_{o\ max} = \frac{\sqrt{1 + A_1A_2}}{2} \qquad (6)$$

Thus, the selectivity of the amplifier is determined by its gain. To obtain high selectivity, one uses a high gain amplifier and the only practical limits on obtainable selectivity are the practical limits on gain of dc amplifiers.

For maximum selectivity and also convenience, we take $R_1 = R_2 = R$ and $C_1 = C_2 = C$. Also, let $A_1A_2 = A$. Then (3) becomes

$$K = \frac{-ARCp}{(1 + A)R^2C^2p^2 + 2RCp + 1} \qquad (7)$$

and (4) becomes

$$w_o = \frac{1}{RC\sqrt{1 + A}} \qquad (8)$$

Equation 6 becomes

$$Q_{o\ max} = \frac{\sqrt{1 + A}}{2} \qquad (9)$$

and the gain at resonance

$$K_o = -A/2 \qquad (10)$$

It can be seen from (8) that resonant frequency is dependent upon gain. Because of this, it is possible to get low resonant frequencies merely by making the gain large. On the other hand, the frequency will change as the tubes age – this could be highly unsatisfactory in any application requiring high frequency stability. However, frequency varies as a function of the square root of gain which means that the percentage change in frequency will be less than half of the percentage change in gain. This seems to assure adequate stability for many practical purposes.

DESIGN CONSIDERATIONS

The output impedances of the two amplifiers, A_1 and A_2, must be small compared to R. (Equation 3 is written with the assumption that the output impedance is zero.) This situation is identical to that encountered in ordinary RC coupling: if the grid resistor is too small, the gain of the amplifier is reduced. In the selective amplifier, unnecessary loss of gain should be avoided because of the dependence of all characteristics upon gain.

As with all feedback amplifiers, the possibility of high frequency oscillations must be taken into account. If there are only one or two stages in the amplifier, then oscillations are unlikely. However, if there are three or more stages, oscillation is almost certain and some sort of corrective network is required.

PRACTICAL RESULTS

Fig. 3 shows the schematic of a selective amplifier built according to these principles and Fig. 4 shows the resulting frequency response. The A_1 Amplifier is a cathode follower and thus has a gain of about unity. The A_2 amplifier has a gain of about -650 and has essentially

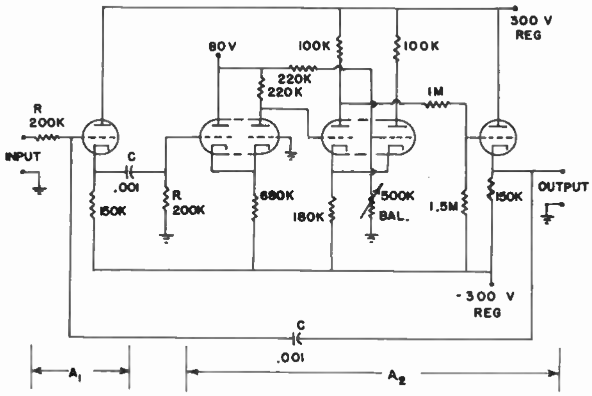

Fig. 3 – Schematic of selective amplifier (All tubes are 12AX7's).

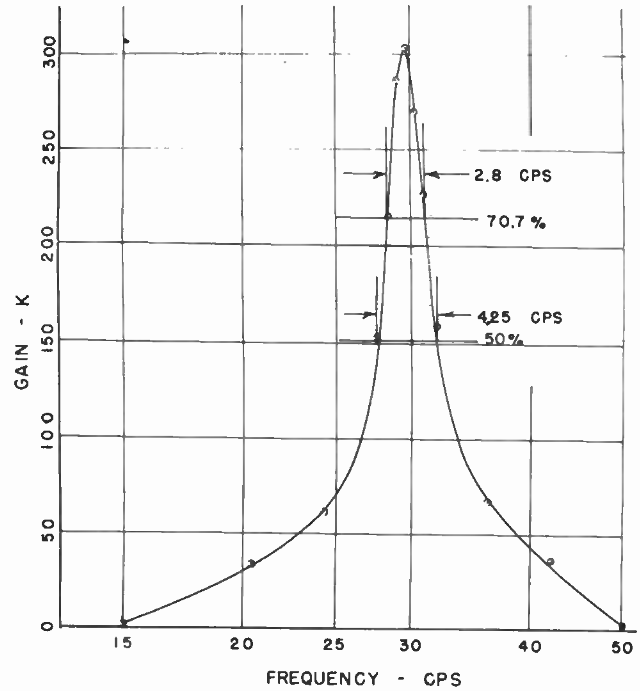

Fig. 4 – Frequency response of selective amplifier.

two stages so that there is no oscillation problem. Cathode followers are used to keep output impedances low.

The values of R and C are 200,000 ohms and 0.001 mf respectively. Since the overall gain is 650, one would expect a Q of 12.5, a resonant gain of 325, and a resonant frequency of 31 cps. The measured Q is 10.5, the measured gain is 303 and the resonant frequency is 29.5 cps. Although these values are close enough to the theory for most practical purposes, the discrepancies are not readily explained.

CONCLUSIONS

A simpler amplifier could certainly be devised. If, for example, amplifier A_1 were to have high gain and A_2 were a cathode follower, most of the difficulties of a dc amplifier (drift, output voltage level, etc.) would be blocked out by the condenser. However, the amplifier shown was readily available so it was used.

The resonant frequency may easily be lowered considerably by using larger values of R and C. For example, if C is increased to 0.01 mf, the resonant frequency would be about 3 cps. Thirty cps was chosen only because measurements are fairly simple to make at that frequency.

To obtain higher selectivity, it would seem to be advisable to cascade several networks of moderate Q rather than to construct a dc amplifier of extremely high gain. This is partly due to the great difficulties encountered with high gain dc amplifiers but also, since the selectivity is only proportional to the square root of gain, one rapidly reaches the point of diminishing returns when increasing gain.

ACKNOWLEDGMENT

The author wishes to take this opportunity to acknowledge the advice and assistance of Dr. G. H. Fett of the University of Illinois in the preparation of this paper.

STEREOPHONIC RECORDING EQUIPMENT*

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SUMMARY – The normal individual has no more than two ears (unfortunately, sometimes only one, putting him at a stereophonic disadvantage), and so it was at first erroneously believed that only two channels need be used for stereophonic reproduction. Earphone listening, binaurally, has certain deficiencies, except for certain subjective listening studies. Loudspeaker reproduction reaches more people at a given time, and corrects some of the deficiencies of listening through earphones. However, room acoustics problems, not present when earphones are used, enter here. An optimum minimum of three channels for loudspeaker listening has been established by comparative listening tests. Electrical phasing, time phasing, and balanced loudness between channels are necessary for the maximum realization of the benefits of a stereophonic system. Minimum harmonic and intermodulation distortion obviously aids any system. Transmitted band width has an effect on realism also.

It has been stated that the distinguishing characteristic of man is his ability to communicate his thoughts to another. Man has progressed through the successive stages of communicating over distances from whispering, talking, yelling, wig-wagging, heliographing, telegraphing, telephoning, wirelessing and aural-broadcasting to televising. It is inevitable that now, since he has transmitted instantaneous pictures, he should go back a bit to do something more than merely transmit intelligence. We come, then, to the refinements. The Session on Binaural Broadcasting concerns itself with a means of reproducing sound more faithfully than before.

The title of the session uses the adjective "binaural." This paper uses the adjective "stereophonic." For a clearer understanding of the question perhaps some lexicography is in order. "Binaural" is defined by Websters' Unabridged Dictionary as "Having or relating to two ears; involving the use of both ears." This, then, is purely a subjective term relating to one's innerself. "Stereophonic" is defined as "Giving a three-dimensional effect of auditory perspective; of sound reproduced." This word, then, is an objective term, relating to something perceived externally to one's self, like perhaps a stereoscopic picture. "Binaural" is from the Latin, meaning "two-eared." "Stereophonic" is from the Greek, meaning "solid sound." Perhaps we should say that we use our *binaural* sense to perceive sound *stereophonically*, or in auditory perspective.

In fact, this very term, *auditory perspective*, was used throughout the six papers,[1] written by members of the Bell Telephone Laboratories, reporting their research preparatory to transmitting the sound of the Philadelphia Symphony in auditory perspective from Philadelphia to Washington, D.C. in 1933. The orchestra performed in Philadelphia, and the sound was picked up by three microphones, transmitted over three separate channels, and reproduced over three separate loudspeaker systems on the stage of Constitution Hall. This transmission was the demonstration of the results of their research into this fascinating business of increased realism in reproduction.

Just now are we becoming publically aware of the possibilities of stereophonic reproduction. This time lag is probably due to several factors. 1933 was a depression year. Such demonstrations cost money. Constitution Hall, though packed for the occasion, could accommodate only a fraction of the total population who might be interested. Only since World War II have we had the means at hand of easily, cheaply, and simply recording more than one channel at a time, by the use of tape recording. It should be pointed out that multi-channel tape recordings of music pre-dated the recent interest in multi-channel broadcasting. Some of the first of these recordings were made at the Summer Music Camp at Interlochen, Michigan in 1951. Subsequently, two-channel broadcasts were made in Chicago and New York. More recently Standard Oil of California sponsored a series of two-channel delayed broadcasts at San Francisco. Currently NBC has a weekly two-channel program at Chicago billed as stereophonic. Several other AM-FM broadcasts have been made in various parts of the country.

The tape recordings at Interlochen were made with two closely-spaced microphones (about 8 inches apart) on two separate channels and were to be listened to by means of earphones. A switch was provided to feed one of the two channels to both earphones, ignoring the other channel, to simulate monaural conditions for comparison purposes. When the switch was thrown to give the two-channel effect, it was startling, to say the least. One seemed to be in the midst of the orchestra. And it recalled to mind the two-eared, hydrophone listening used aboard submarines in World War I, and the Bell Laboratories Oscar exhibit at the World's Fair in Chicago in 1933.

*Manuscript received September 14, 1953. Simultaneous publication with AIEE, with permission. Presented at AIEE meeting, Atlantic City, June 16, 1953.

[1] *Elect. Eng.*, vol. 53, no. 1; January, 1934.

This exhibit is now housed at Chicago's Museum of Science and Industry. To refresh your memory, Oscar was a life-sized dummy with two microphones hidden in his head at the normal ear locations. He was enclosed in a glass-walled room, so that all could see the lecturer walk around Oscar. Several pairs of earphones were available to observers outside the room. They wore these and could both see and hear where the lecturer walked with respect to Oscar. The various auditors seemed acoustically to be where Oscar stood. But the curious point was that Oscar had his back to the audience. This, the lecturer explained, was because, as he walked behind Oscar while talking, he appeared aurally to be walking behind the auditor. And when he walked in front of Oscar, he still appeared to be walking *behind* Oscar! This was a paradox. The eyes said "yes," but the ears said an emphatic "*no.*"

Now this condition was again evident when one listened by means of earphones to an orchestral recording. The sound which should have been in front of the listener was perceived to be all around him, or behind him. Thus we came to the inescapable conclusion that we should reproduce the sound by means of a multiple loudspeaker system, which would definitely help the auditor to localize the sound in front of himself where it logically should be. Here was this subjective "binaural" listening, as contrasted to the objective "stereophonic" reproduction.

Recalling the earlier experimental work back of the Philadelphia-to-Washington transmission, we found that the researchers had explored many of the parameters involved. Using separate transmitting and listening rooms, they had tried listening tests using two, three, four and more separate channels. They also tried dividing the output of two microphones to three loudspeakers, and of three microphones to two loudspeakers. Phasing and loudness tests were made. The following conclusions were reached:

1. One channel gave no lateral information, but *did* yield some depth information, due to loudness changes and increased reverberation as microphone-to-source distance increased.
2. Two channels gave fair lateral judgment, but poor depth determination.
3. Three channels gave good depth information and good lateral information by filling in at the center.
4. Four or more channels gave but little more information than the three-channel system, laterally or depthwise. Therefore they concluded that the three-channel system was the optimum minimum number of channels to be used for best results.
5. Borrowing from another channel (i.e. three microphones to two loudspeakers, or two microphones to three loudspeakers) was only a little better than the two-channel system and not nearly so good as the three-channel system.
6. Proper phasing of all channels was a necessity to avoid cancellation effects.
7. Varying the relative loudness between channels made the source appear to move toward the louder reproducing point.
8. Microphone placement must be in relation to loudspeaker placement.

Ampex began the commercial production of multi-track tape recorders about three or four years ago for those interested in recording telemetered information from guided missiles and the like. These equipments provide any number of simultaneous channels up to fourteen. It was a simple step, therefore, to make stereophonic tape recorders with two or more channels. In reviewing the literature, and our prospective customers' needs, we have built both two-track and three-track machines. The two-track units have been used primarily in the industrial field, where one is definitely planning to use the equipment for the purely investigative purposes of localizing and subjectively analyzing sound sources, or making sounds subjectively appear more realistic. This is accomplished primarily by means of two earphones, and only rarely by means of two loudspeakers. One track is often also used for quantitative measurement, while the other is used for running commentary. The three-track units have been used primarily where sound is to be reproduced by means of a loudspeaker system. Obviously, one may use only two tracks if he wishes to use earphones. When using loudspeakers, the problem of listening-room acoustics is all important, and must be taken into consideration, but this is beyond the scope of this paper.[2]

In recording multi-channel sound for later reproduction, we have not only verified all of the findings of the Bell Labs. experiments of twenty years ago, but we have also run into one other problem. Since the Bell Labs. experiments were all concerned with instantaneous transmission of the original sound, time lag effects between channels were not factors. A recorder, basically, is a time storage device. In delaying time we find that we must delay all of the channels by the same amount. Because they are recorded side by side on the same tape, one expects that they must all be reproduced at the same time. However, we have found that any deviation of spacing, longitudinally with respect to the direction of tape motion, introduces small constant time differences between channels. This has the effect in reproduction of making the apparent location of a given source deviate from its actual original position. The angle of deviation increases as the time differential increases, and can be calculated. Therefore, all of the heads must have the smallest possible error in parallel alignment. Some manufacturers both record and reproduce using the same heads for the two functions. Here, obviously, there is no time error. But should such a recording be played on another machine, the head spacing can not be different from the first without running afoul of this time-phase shift.

[2] R. J. Tinkham, "Binaural or stereophonics," *Audio Eng.*, January, 1953.

Ampex uses two separate sets of heads and amplifiers, one set for recording, and one for playing back the same record an instant after it has been made. This method has obvious monitoring advantages during the recording of a program. It also has the problem for the manufacturer, not the user, of making it necessary to build high accuracy into the head spacing. This problem and the one of inductive cross-talk between adjacent channels (40 db) have been worked out.

DESCRIPTION OF EQUIPMENT

The two- and three-track Ampex systems are housed in three luggage cases. One case contains the mechanical equipment involved in tape motion, the second contains the requisite number of combined record/playback amplifiers, each with separate controls and meter. The third case contains the power supplies for the amplifiers. All parts, optionally, may be mounted in a standard 19" relay rack.

The new series 350 Ampex mechanical assembly is mounted on a 15¾ x 19" panel. Each reel turntable is driven by its own torque motor, and each motor is equipped with a solenoid actuated brack, self-energizing in the proper direction.

All ball-bearing mounted flywheel, driven by the passage of the tape around a drum mounted on the same shaft and above the panel, stabilizes the tape motion as it approaches the head assembly. The head assembly holds, from left to right, the full-track erase head, the two- or three-track playback head. The heads are suitably shielded in mu-metal cans, with covers which fall into place as the hinged portion of the assembly is lowered into place after tape threading. The tape is accurately guided on and off the heads by means of pyrex glass guides spaced as far apart as the width of the tape at the entrance and exit of the assembly. The tape is pulled over the heads by means of a pinch drive which consists of a capstan and a rubber capstan idler roller. The capstan is the accurately ground shaft of a dual-speed hysteresis motor, and is capable of driving the tape at either seven and one-half inches or fifteen inches per second, with a timing accuracy of ± 3.6 seconds in thirty minutes of recording time. The capstan motor is also equipped with a flywheel. Motor speed is changed by a toggle switch. All mechanical functions — fast forward, fast rewind, stop, and playing speed forward — are controlled by means of momentary contact push-buttons. Electrical memory is furnished by dc actuated relays. Therefore the entire system may be remotely controlled from a separate push-button station. Ampex Speed Lock equipment may also be plugged into the mechanism to make possible the synchronizing of the equipment with motion picture camera and projector equipment to within one/one-thousandth of a second accuracy. This is an electrical arrangement and requires no additional heads or mechanical modification to the equipment.

Each separate amplifier channel has both a recording and reproducing amplifier, with appropriate switching, so that either the incoming signal on that channel, or the signal just recorded on the tape for that channel, may be monitored. Each standard four-inch VU meter may also be switched between the recording amplifier and the playback amplifier. Erase and bias voltages for each channel may also be read on this meter through appropriate switching. The recording amplifier has a choice of three inputs: low-impedance microphone, balanced, and unbalanced bridge input. The playback amplifier delivers + 8 VU across a balanced or unbalanced 600 ohm line output. An internal 600 ohm load resistor may be cut in or out by means of a switch. Suitable gain controls are provided for each input and output. Erase and bias current is obtained from a push-pull oscillator using a toroidally wound coil, and is tuned to approximately 100 kilocycles. This oscillator is mounted integrally on one of the amplifiers. Its output is also fed to separate buffer amplifiers associated with each of the other channels. This eliminates the possibility of beats which would exist if each channel had its own oscillator.

The power supplies are straightforward. They are mounted separately in order to reduce hum pickup and to divide the weight into convenient packages.

On the three-track system, the tracks are each 0.040 inches wide, and are separated by blank tape space of about 0.050 inches. All three tracks lie side by side on standard tape of one-quarter inch width. With such narrow tracks, some signal-to-noise ratio is sacrificed. A restricted bandwidth would give less background noise. While it has been stated that we can tolerate a more restricted bandwidth when using multi-track reproduction to get same subjective feeling of presence as compared to a mon-aural system, it is also true that the full auditory band width will given even more realism to the sound.

USE OF THE INDICES

Daniel W. Martin, Chairman
Editorial Committee IRE - PGA

The cumulative indices which follow are of three types. The first type is simply a compilation of Tables of Contents of TRANSACTIONS of the IRE-PGA, preceded by a list of the newsletters and other publications which came before the publication of TRANSACTIONS. Next is an Author Index. The third type is an Analytic Subject Index.

From the standpoint of indexing it is unfortunate that a consistent numbering system for volumes and pages could not have been adopted at the start. Absence of such a system has pointed up the necessity for these cumulative indices. It is planned that future volumes will adopt the conventional consecutive page-numbering system, which will simplify both indexing and the reference use of the publication.

The earliest technical papers circulated among the membership of the IRE Professional Group on Audio, and the separate NEWSLETTERS carried no identifying numbers other than dates. Late in 1951 several technical papers were issued as separate TRANSACTIONS of the IRE-PGA, and these were assigned PGA numbers PGA-1, PGA-2, etc. Early in 1952 the NEWSLETTER was absorbed into TRANSACTIONS, and the latter became a regular bimonthly publication, continuing to use the same numbering system through PGA-10, the final issue of 1952. Because most of the previously unnumbered publications predate this numbering system, a series of *Roman* numerals has been assigned to them in the Tables of Contents which follow. Thus PGA-I is the first paper circulated by IRE-PGA to its membership back in 1950, and PGA-1 is the first paper published as an official TRANSACTIONS, and made available also outside the membership of IRE-PGA.

The volume system of indexing began with the first issue of 1953. Volume AU-1, issues 1, 2, ... 6, contain all IRE-PGA publications for the year except Section 3 of the CONVENTION RECORD. In the Tables of Contents this section has been assigned a number CR-1-3, and inserted chronologically between issues AU-1-3 and AU-1-4 of the TRANSACTIONS of the IRE-PGA.

In both the Author Index and the Analytic Subject Index references are made to publications by the Roman or Arabic numbers of the PGA series, or by the issue number in volume AU-1, plus the page number in the issue. In the future this can be simplified to the volume number and the page number in the volume (e.g., AU-2,56).

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

Classification of Subjects

1. IRE-PGA
    - 1.1 General
    - 1.2 Constitution and By-Laws
    - 1.3 National and Regional Meetings
    - 1.4 Chapters
    - 1.5 Membership
    - 1.6 TRANSACTIONS
    - 1.7 People
2. Bibliographies, Reviews, Standards, Tapescripts
    - 2.1 Bibliographies
    - 2.2 Reviews
    - 2.3 Standards
    - 2.4 Tapescripts
3. Sound Systems
    - 3.1 General
4. Microphones
    - 4.1 General
    - 4.2 Condenser
    - 4.3 Magnetic
    - 4.4 Crystal
    - 4.5 Moving – Coil
    - 4.6 Ribbon
    - 4.9 Special Microphones
5. Amplifiers
    - 5.1 General
    - 5.2 Preamplifiers and Voltage Amplifiers
    - 5.3 Power Amplifiers

5.4 Frequency – Range Dividing Networks
5.9 Special Amplifiers

6. Loudspeakers

6.1 General
6.2 Direct – Radiator Units
6.3 Horn-Driver Units
6.4 Horns, Enclosures, Baffles
6.9 Special Types

7. Disc Recording and Reproduction

7.1 General
7.2 Discs
7.3 Recording
7.4 Pickups and Tone Arms
7.5 Preëmphasis and Postequalization
7.9 Special Mechanical Recorders

8. Magnetic Recording and Reproduction

8.1 General
8.2 Tape and Wire
8.3 Recording and Erasing
8.4 Playback
8.5 Preëmphasis and Postequalization
8.9 Special Magnetic Recorders

9. Acoustics

9.1 General
9.2 Room Acoustics
9.3 Sound Waves and Vibrations
9.4 Speech
9.5 Music
9.6 Hearing

10. Broadcast Audio
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Analysis of Reader Interest in TRANSACTIONS, Daniel W. Martin, AU-1-6, 2, November-December, 1953.

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Otto C. Bixler, AU-1-1, 24, January-February, 1953.

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Reviews of Audio Publications, Kenneth W. Goff, PGA-7, 10, May, 1952.

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Technical Committee Work on Audio-Frequency Measurements, Arnold Peterson, AU-1-5, 4, September-October, 1953.

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Report of Tapescripts Committee, Andrew B. Jacobsen, PGA-6, 12, March, 1952.

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Report of the Activities of the Tapescripts Committee, Andrew B. Jacobsen, AU-1-2, 8, March-April, 1953.

Report on Tapescripts Committee, Andrew B. Jacobsen, AU-1-5, 2, September-October, 1953.

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Microphone Directivity, Benjamin B. Bauer, PGA-8, 10, July, 1952.

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Coupling the Speaker to the Output Stage, Vincent Salmon, PGA-VIII, January, 1952.

Analysis of a Single-Ended Push-Pull Audio Amplifier, Chai Yeh, AU-1-2, 9, March-April, 1953.

Why Fight Grid Current in Class B Modulators?, J. L. Hollis, AU-1-2, 26, March-April, 1953.

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Coupling the Speaker to the Output Stage, Vincent Salmon, PGA-VIII, 5, January, 1952.

Over-all Audio Performance of Commercial Radio Receivers, W. B. Clark, PGA-7, 11, May, 1952.

Speakers and Transmission of Sound Waves, Frank H. McIntosh, PGA-9, 6, September-October, 1952.

Direct Measurement of the Efficiency of Loudspeakers by Use of a Reverberation Room, H. C. Hardy, H. H. Hall, and L. G. Ramer, PGA-10, 14, November-December, 1952.

Acoustic Damping for Loudspeakers, Benjamin B. Bauer, AU-1-3, 23, May-June, 1953.

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Subjective Loudspeaker Testing, Harry F. Olson, AU-1-5, 7, September-October, 1953.

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Application of Electric Circuit Analogies to Loudspeaker Design Problems, B. N. Locanthi, PGA-6, 15, March, 1952. (Addendum: PGA-7, 46, May, 1952.)

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A Conversation on Loudspeaker Enclosures, Daniel W. Martin, PGA-9, 1, September-October, 1952.

An Analogue for Use in Loudspeaker Design Work, Jordan J. Baruch and Henry C. Lang, AU-1-1, 8, January-February, 1953.

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Design Interrelations of Records and Reproducers. H. I. Reiskind, PGA-5, February, 1952.

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The Application of Damping to Phonograph Reproducer Arms, William S. Bachman, PGA-II, 1951.

Wear of Phonograph Needles, Benjamin B. Bauer, PGA-1, December, 1951.

Phonograph Reproduction, Benjamin B. Bauer, CR-1-3, 3, June, 1953.

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Sound System for Plenary Hall of United Nations General Assembly Building, C. W. Goyder and L. L. Beranek, CR-1-3, 26, June, 1953.

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Do You Auralize?, Daniel W. Martin, PGA-8, 2, July, 1952.

A Practical Binaural Recording System, Otto C. Bixler, AU-1-1, 14, January-February, 1953.

Stereophonic Recording Equipment, Russell J. Tinkham, AU-1-6, 13, November-December, 1953.

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Over-all Audio Performance of Commercial Radio Receivers, W. B. Clark, PGA-7, 11, May, 1952.

Do You Auralize?, Daniel W. Martin, PGA-8, 2, July, 1952.

Proper Care of Test Records, L. A. Wood and R. C. Moyer, PGA-8, 4, July, 1952.

A New Audio Control Console for Television and Radio, W. W. Dean, PGA-8, 28, July, 1952.

Increased Audio without Splatter, John L. Reinartz, PGA-9, 22, September-October, 1952.

A Practical Binaural Recording System, Otto C. Bixler, AU-1-1, 14, January-February, 1953.

Why Fight Grid Current in Class B Modulators, J. L. Hollis, AU-1-2, 26, March-April, 1953.

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Measuring the Stability of Sonic Recorders, E. N. Dingley, Jr., PGA-7, 20, May, 1952.

A Method for Measuring the Changes Introduced in Recorded Time Intervals by a Recorder-Reproducer, J. F. Sweeney, PGA-7, 24, May, 1952.

A Sound-Survey Meter, Arnold Peterson, PGA-7, 30, May, 1952.

Microphones for the Measurement of Sound Pressure Levels of High Intensity over Wide Frequency Range, J. K. Hilliard, PGA-7, 38, May, 1952.

A Phase and Gain Meter, Stephen F. Condon, PGA-8, 37, July, 1952.

A Ceramic Vibration Pickup, E. V. Carlson, PGA-10, 2, November-December, 1952.

Direct Measurement of the Efficiency of Loudspeakers by Use of a Reverberation Room, H. C. Hardy, H. H. Hall, and L. G. Ramer, PGA-10, 14, November-December, 1952.

Sensitivity of Microphones to Stray Magnetic Fields, L. J. Anderson, AU-1-1, 1, January-February, 1953.

Loudspeaker Impedance, Vincent Salmon, AU-1-4, 1, July-August, 1953.

Technical Committee Work on Audio-Frequency Measurements, Arnold Peterson, AU-1-5, 4, September-October, 1953.

Power Capacity of Loudspeakers, Paul S. Veneklasen, AU-1-5, 5, September-October, 1953.

12. Electronic Musical Instruments

An Electronic Music Box, Earle L. Kent, PGA-4, December, 1951.

Imitation of Natural Sounds, Daniel W. Martin, PGA-VII, 7, November, 1951.

Electronic Music – Past, Present, and Future, Earle L. Kent, AU-1-2, 1, March-April, 1953.

INSTITUTIONAL LISTINGS (Continued)

KLIPSCH AND ASSOCIATES, Box 64, Hope, Arkansas
Wide Range Corner Horn Loudspeaker Systems, Corner Horns

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

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

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

SHALLCROSS MANUFACTURING COMPANY, 524 Pusey Avenue, Collingdale, Pennsylvania
Audio Attenuators, Precision Resistors, Rotary Switches, Test Instruments

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

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

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

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

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,
B. B. Bauer, Shure Brothers Inc., 225 W. Huron St., Chicago 10, 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.

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

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

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

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

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

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

CINEMA ENGINEERING CO., Division Aerovox Corp., 1100 Chestnut St., Burbank, California
Equalizers, Attentuators, Communication Equipment

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

EDUCATIONAL LABORATORIES, INC., 1625 Connecticut Avenue, Washington 9, D. C.
Custom Audio-Visual Installations; Recording, Projection and High Fidelity Music Equipment

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

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

(Please see inside back cover for additional names)

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

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The Institute of Radio Engineers

IRE PROFESSIONAL GROUP ON AUDIO

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

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SAN DIEGO CHAPTER ACTIVITIES

The San Diego Chapter of IRE-PGA has been conducting an active program of meetings for a year and a half. There were four meetings held before the formal approval of the chapter. The first meeting was held on April 29, 1952, where an interesting talk on microgroove recording was presented by Chester Boggs of Century Engineers Inc. A second meeting was held on July 29, 1952, when three short papers were presented on the subjects of disc recording, amplifiers and loudspeakers. Two of these papers were presented by members of the chapter and one was a PGA tapescript. The third meeting was held November 5, 1952, when Mr. Gales and Mr. Webster of the Navy Electronics Laboratory, Psychophysics Branch, gave a demonstration lecture on "Subjective Factors in Evaluating Audio Systems."

At a business meeting held in January it was decided to hold approximately four meetings in 1953. It was further decided to have a different member put in charge of each meeting. The first of these meetings was held on February 17 with Charles Miller as chairman. The subject of this discussion meeting was "Low Cost Instrumentation for the Evaluation of Audio Systems." The second 1953 meeting was held in July with Matt Brady as program chairman. A paper was given by J. P. Maxfield entitled "The Simulation of Spatial Distribution of Reproduced Sound." One week later the chapter met at the home of F. X. Byrnes for a business meeting. Up to this time no formal organization of the chapter existed. The ground work for the first meeting had been carried out by Stan Sessions, and at that meeting a working committee consisting of Stan Sessions, Matt Brady, Royal Burkhardt and Francis Byrnes had been formed with Mr. Byrnes as acting chairman. This informal organization was replaced when a regular slate of officers was elected, as follows:

David G. Bailey,	Naval Base, Chairman
Charles N. Miller,	NEL, Vice-Chairman
Cdr. William B. Bernard,	NEL, Secretary-Treasurer

On October 3 in a suite at the San Diego Hotel a demonstration meeting featuring high-fidelity audio equipment was held. The demonstration ran from 9 A.M. to 9 P.M. in order to permit everyone to have a chance to see and hear the equipment. The attendance, which was estimated at well over 500, was so large that during the afternoon and evening it was frequently difficult to see the equipment for the people. The meeting which was arranged by the chapter chairman, Dave Bailey, was very well publicized in the newspaper and on radio and television. The large attendance made it difficult for the PGA members to make a subjective evaluation of the equipment being demonstrated, as originally planned. Future demonstrations of this type should provide for a special session for members only, so that the principal purpose of the meeting can be fulfilled. The last chapter meeting of the year was a tour of the complete studio and transmitter facilities of radio station KSDO.

PHILADELPHIA CHAPTER EVENTS

1. At the first meeting on October 19, 1953, Mr. E. D. Nunn, President of Audiophile Records, presented a very interesting talk on "The Quest for High Fidelity – From a Hobbyist's Point of View." After a discussion of the early experiments, methods of approach and recording techniques and equipment, demonstrations of high quality equipment were made covering a frequency range in excess of nine octaves.
2. Dr. William L. Everitt of the University of Illinois gave a lecture with demonstrations on "A Method of Time Compression or Expansion of Speech" at the regular IRE section meeting on November 5, 1953. He explained how the ear is faster than the mouth; words can be understood more rapidly than they can be spoken. The device can produce speech either with the original frequency spectrum in a shorter time (speaker seems to be talking more rapidly) or the information can be transmitted in the same time over a narrower frequency band. Time or frequency expansion can also be obtained.
3. On February 18, 1954 the Audio Chapter will hear Dr. Daniel W. Martin, The Baldwin Company, discuss "The Enhancement of Music by Reverberation." He will use magnetic tape recordings to demonstrate the effects of natural reverberation upon organ and piano tones. A new loudspeaker with built-in reverberation will be demonstrated. The added reverberation improves the tone of an electronic organ installed in a small room.

Murlan S. Corrington, Chairman

SUMMARY OF AVAILABLE TAPESCRIPTS

Andrew B. Jacobsen
Phoenix Laboratory, Motorola, Inc.
Phoenix, Arizona

Tapescripts are recorded technical papers with slides, intended to extend the author's coverage to a larger audience. Ideally, we would like a personal presentation with slides, demonstrations and exhibits. The ideal can be approached through tapescripts by means of the author's recorded voice, continuous slide presentation and a duscussion leader who has previewed the paper, and studied the references.

Technical standards have been set for IRE-PGA tapescripts: sound recorded at 7½ inch per second on ¼ inch tape, full track on 7 inch reels; slides to be 2" x 2" cardboard mounts in black and white or color. A script and references, as well as an abstract, should be available.

Tapescripts are now available for loan, at no cost except for return postage, to any IRE-PGA group interested. They should be ordered ahead, so as to be available several days before use. This permits the discussion leader to preview the material. The Tapescripts Committee of the Professional Group on Audio has funds to provide copies and to ship the material. You may order one or more papers for preview at a program committee meeting.

The following papers are available from the PGA Tapescripts Committee Chairman, A. B. Jacobsen, 1802 North 47th Place, Phoenix, Arizona.

"Method for Time or Frequency Compression – Expansion of Speech", Grant Fairbanks, W. L. Everitt, and R. P. Jaeger, University of Illinois. Available February 1, 1954.

"Magnetic Recording", Marvin Camras, Armour Research Foundation of Illinois Institute of Technology. Discusses fundamentals of wire and tapes; heads; bias; circuits; equalization; present problems; and future developments. Thirty minutes.

"Phonograph Reproduction", B. B. Bauer, Shure Brothers, Inc. Present trends and discussion of design factors affecting record wear and quality of reproduction. One hour.

"Push Pull Single Ended Audio Amplifier", Arnold Peterson and D. B. Sinclair, General Radio Company. A convention paper presented by Dr. Peterson. Twenty-five minutes. 3¼ x 4 slides.

"Sound Survey Meter", Arnold Peterson, General Radio Company. A convention paper. Twenty minutes. 3¼ x 4 slides.

"Microphones for High Intensity and High Frequencies", John K. Hilliard, Altec Lansing Company. A convention paper. Twenty minutes. 3¼ x 4 slides.

"The Ideo-Synchronizer", J. M. Henry and E. R. Moore, Boston Bell. A humorous satire on technical writing, specifications, and engineering. Good for mixed audience. Twelve minutes.

Other tapescripts are available elsewhere, including the following:

"Germanium the Magic Metal". A technical paper on the preparation of Germanium, theory of diodes and transistors, and construction details. Seventy-eight color strip slides, sound on 16 inch, 33⅓ discs. Forty-five minutes. This may be obtained from J. H. Sweeney, General Electric Company, Electronics Park, Syracuse, N.Y.

The following Bell Laboratories Experimental Tapescripts may be obtained from W. H. Doherty, Bell Telephone Laboratories, Inc. Murray Hill, New Jersey.

Tapescript Number	Title	Author
1*	"Experiments with Linear Prediction in Television"	C. W. Harrison
2*	"Statistics of Television Signals"	E. R. Kretzmer
3*	"Efficient Coding"	B. M. Oliver
4	"The Physics of Music and Hearing"	W. E. Kock
5	"Transistors in Negative Impedance Circuits"	J. C. Linvill
6	"A Junction Transistor Tetrode for High Frequency Use"	R. L. Wallace, Jr.
7	"Some Circuit Properties of "Junction Transistors"	L. G. Schimpf
8	"Ferroelectric Storage Devices"	J. Reid Anderson

*These papers were published in the July 1952 issue of the Bell System Technical Journal.

PGA BRIEFS

A group of thirty-four members of the Cleveland Section of the IRE has petitioned for the formation of a Cleveland Chapter of the IRE Professional Group on Audio. The petition has been approved by the Executive Council of the Cleveland Section, and has been forwarded to IRE Headquarters.

Any IRE-PGA members planning to attend the International Meeting on Sound Recording, to be held in Paris April 5-10, and sponsored by The Société Des Radioélectriciens, 10 Avenue Pierre Larousse, Malakoff (Seine), can obtain further information from IRE Headquarters or from IRE-PGA. Early registration will be necessary for suitable arrangements to be made.

The program of the Chicago Chapter IRE-PGA has had several recent changes. The December 16 paper was changed to "The Design and Construction of Low Frequency Horns," Dan Plach and Karl Kramer, Jensen Mfg. Co. The January 15 paper was postponed. A paper entitled "High Fidelity", by John Clark of El-Rad Mfg. Co., has been scheduled for February 19.

Mr. Philip B. Williams, chairman of the Chicago Chapter IRE-PGA, has recently been promoted to the position of Chief Engineer of the Jensen Mfg. Co.

The Cincinnati Chapter IRE-PGA recently enjoyed hearing former Cincinnati resident Dr. W. E. Kock, Director of Acoustics Research, Bell Telephone Laboratories, give a paper, "The Physics of Music and Hearing," at a joint meeting of the IRE, AIEE, and the Engineering Society of Cincinnati.

An interesting program on "High-Fidelity" is being arranged for the 1954 IRE convention in March, under the chairmanship of Dr. W. E. Kock. The complete program of the convention, including 100-word abstracts of all papers, will be published in the March issue of Proceedings of the I.R.E. Make your plans to attend.

On November 17, 1953, Professor A. B. Bereskin of the University of Cincinnati, and an electronic consultant of The Baldwin Company, addressed the Cincinnati Section IRE on the subject, "A High Efficiency-High Quality Audio-Frequency Power Amplifier." The amplifier was demonstrated in a music reproduction system. Following the section meeting the IRE-PGA chapter participated in a demonstration test prepared by officers of the chapter, entitled "How Much Distortion Can You Hear?" A special tape recording of signals containing various measured amounts of harmonic and intermodulation distortion was played over a low-distortion reproducing system in randomized paired comparisons. Attendance at the chapter meeting was at an all-time high. Results of the test were reported at the December IRE meeting, and will appear in a future issue of Transactions Of The IRE-PGA. The recording may be converted into a tape-script.

The latest news before this issue is printed concerns a petition from the Phoenix Section of IRE asking for the formation of a Phoenix Chapter of the Professional Group on Audio.

DEPARTMENT OF DEFENSE SYMPOSIUM ON MAGNETIC RECORDING

A symposium on magnetic recording was held on October 12 and 13, 1953, in the Department of Interior Auditorium, Washington, D. C., under the auspices of the Department of Defense. Attendance was open to all with a direct interest in development, manufacture or professional application of magnetic recording and playback. The sponsor of the meeting was the Department of the Navy, Bureau of Ships, as part of its allocation to provide coordination of Acoustics-in-Air research projects within the Department of Defense.

The program opened with an introduction, entitled "Purpose and Plan of the Symposium" by the session chairman, Dr. S. J. Begun, President of Clevite-Brush Development Company. Other session chairmen were: Mr. Marvin Camras, Armour Research Foundation, and chairman of the IRE Professional Group on Audio; Mr. Robert H. Carson, Sound Division, Naval Research Laboratory; Mr. Lynn C. Holmes, Research Department, Stromberg-Carlson Company; and Mr. Paul J. Weber, National Security Agency. The Institute of Radio Engineers was a co-sponsor of the symposium. The Washington IRE representative on the Papers Committee was Mr. Henry J. Meisinger, U.S. Recording Company. Some of the papers presented at the symposium appear in this issue of TRANSACTIONS of the IRE-PGA. Others are being solicited for future issues. The unclassified part of the program follows:

Paper No.	AUTHOR	TITLE
1	Dr. W. W. Wetzel	"Certain Advantages of Ferrite Over Mu-Metal Magnetic Recording Heads"
2	Dr. L. E. Loveridge	"A Vacuum Tube for an Electron Beam Magnetic Reproducing Head"
3	Mr. J. W. Gratian	"Core Structure for the Electron Beam Magnetic Reproducing Head"
4	Messrs. S. M. Rubens A. B. Bergh	"A Magnetostatic Reading Head"
5	Mr. W. R. Boenning	"Performance Characteristics of Magnetostatic Reproducing Equipment"
6	Mr. C. E. Williams	"Playback of Magnetic Recordings Through Transistor Amplifiers"
7	Dr. John G. Frayne	"Components and Mechanical Considerations for Magnetic Sound on 35 mm. Film"
8	Mr. O. C. Bixler	"Basic Mechanical Considerations for Tape Transport Systems"
9	Mr. J. E. Johnston	"Mechanical Factors Governing Tape Coatings, Backings"
10	Mr. G. L. Davies	"Magnetic Recorders for Data Recording Under Adverse Environments"
11	Mr. Walter T. Selsted	"Improved Performance of Magnetic Recording System for Precision Data"
12	Mr. E. W. Darcy	"Present Status of Measurement Techniques, Equalization Systems, and Calibrated Recordings in 16mm. Magnetic Recording"
13	Mr. Walter H. Erikson	"Tape Testing on a Comparison Basis"
14	Messrs. F.A. Comerci, S. Wilpon, and R. Schwartz	"Characteristics of Recent Commercial ¼" Magnetic Tapes"
15	Mr. Frank Radocy	"Some Notes on Problems Encountered in the Use of the Standard Reference Tape"
16	Messrs. F. Comerci, S. Wilpon, and R. Schwartz	"A Standard Magnetic Tape Recording for Standardizing the Characteristics of Navy-Recorder-Reproducers"
17	Mr. Frank G. Lennert	"Equalization of Magnetic" Tape Recorders, and General Recorder Performance Tests"
18	Mr. J. D. Bick	"Methods of Measuring Surface Induction of Magnetic Tape"

RCA RECORDS COLOR TELEVISION ON MAGNETIC TAPE

The recording of television pictures on magnetic tape in color and in black-and-white was publicly demonstrated for the first time on December 1, 1953 by RCA at its laboratories at Princeton, N.J. Although the video tape equipment is still in the development stage, it was estimated that it would be ready for commercial use in about two years.

This development is of particular interest to the Professional Group on Audio because it is an adaptation of audio equipment and techniques to the video field. The seven man team of research engineers responsible for the development of the equipment consisted of Dr. Harry F. Olson (a frequent contributor to TRANSACTIONS of the IRE-PGA) and William D. Houghton, who head the development program, and Maurice Artzt, J. T. Fischer, A. R. Morgan, J. G. Woodward and Joseph Zenel.

The demonstration featured a simultaneous side-by-side comparison of a live color television program fed directly to one receiver and at the same time recorded on tape and played back instantly to a second receiver. The program originated in NBC studios in New York City and was beamed by microwave radio across the 45-mile path to Princeton where it was received and recorded.

The equipment used in the demonstration utilized a single magnetic tape ½-inch wide on which five parallel channels were recorded, one for each of the three primary color signals, one for the synchronizing signal and one for the sound signal.

The four-minute program was recorded at a tape speed of 30 feet per second on a 17-inch reel. Work is now under way towards developing a reel 19 inches in diameter which will carry a 15-minute program.

Magnetic tape recording of television pictures offers several advantages over the present method of recording on film. Since a recorded tape requires no further processing, it may be played back immediately and eliminates the time and expense associated with developing and processing film. Moreover, unlimited number of magnetic tape recordings can be made quickly. In addition, recorded tapes can be demagnetized and used many times over.

Since the tape can be re-used it is estimated that recording a color television program on tape would cost one-twentieth as much as a film recording, and to record a black and white program would cost only one-fifth as much as the film recording.

AN ACOUSTIC LENS AS A DIRECTIONAL MICROPHONE*

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Murray Hill, N. J.

SUMMARY – An acoustic lens combined with a conical horn can be used to obtain a highly directional microphone without some of the disadvantages of the parabolic microphone. The directional characteristics can be calculated satisfactorily if one assumes that the horn provides uniform flooding of the lens aperture.

INTRODUCTION

Many directional microphones,[1] of which the well-known "cardioid" is an example, have a medium directivity which does not vary appreciably with frequency over a wide band. More highly directional microphones include line microphones,[2,3] high-order gradient microphones,[4] and parabolic reflector microphones,[5,6,7] all of which are subject to variation of gain and directivity with frequency.

The development of convenient acoustic lenses[8,9] makes possible another highly directional microphone, similar to the parabolic one, in which the transducer is placed at or near the focal point of an acoustic lens and the intervening space enclosed by a conical horn.

PARABOLIC REFLECTOR vs. LENS-HORN

Parabolic-reflector and lens focussing have already been compared by Kock[10] in connection with microwave antennas. This comparison holds true in the acoustic case, since the behavior of the devices is essentially the same for sound waves as for microwaves, with the exception of polarization effects. The parabolic reflector microphone has the following disadvantages:

(a) The transducer, placed at the focus, blocks a portion of the beam (Fig. 1b).
(b) There is a finite sensitivity in the backward direction due to the diffraction of sound around the paraboloid into the transducer.
(c) There are strict requirements on the parabolic surface for good focussing.

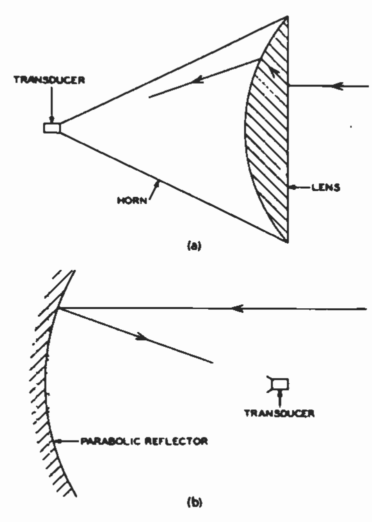

Fig. 1 – (a) The lens-horn microphone.
(b) The parabolic microphone.

*Manuscript received July 27, 1953. Also published in the November, 1953 *Jour. Acous. Soc. Amer.*

[1]For a comprehensive review of directional microphones see H. F. Olson, "Elements of Acoustical Engineering", D. Van Nostrand Company, Inc., New York.

[2]Olson, H. F., *Proc. I.R.E.*, vol. *27*, 7, p. 438; 1939.

[3]Mason and Marshall, *Jour. Acous. Soc. Amer.*, vol. 10, 3, p. 206; 1939.

[4]Olson, H. F., *Jour. Acous. Soc. Amer.*, vol. 17, p. 192; 1946.

[5]Hanson, O. B., *Jour. Acous. Soc. Amer.*, vol. 3, 1, p. 81; 1931.

[6]Dreher, Carl, *Jour. Soc. of Mot. Pic. & Telev. Eng.*, vol. 16, 1, p. 29; 1932.

[7]Olson and Wolff, *Jour. Acous. Soc. Amer.*, vol. 1. 3, p. 410; 1930.

[8]Kock and Harvey, *Jour. Acous. Soc. Amer.*, vol. 21, p. 471; 1949.

[9]Kock and Harvey, *Bell Sys. Tech. Jour.*, vol. 30, p. 564; 1951.

[10]Kock, W. E., *Proc. I.R.E.*, vol. *34*, p. 828; 1946.

The lens-horn microphone (Fig. 1a) is not subject to disadvantages (a) and (b), and since the arrangement of the lens elements is not very critical,[11] disadvantage (c) is practically eliminated. For acoustic applications, an additional disadvantage of the parabolic reflector derives from the need to cover a wide frequency range. The directional sensitivity of the transducer cannot be adjusted to flood the reflector aperture properly for a wide frequency range, and consequently appreciable "spillover" sidelobes are observed at the lower frequencies. In the lens-horn microphone this problem is avoided as the horn acts as an effective shield at all frequencies.

DIRECTIONAL CHARACTERISTICS OF A LENS-HORN

Fig. 2 shows a lens-horn combination using a slant-plate delay lens.[11] This lens has a 29-inch aperture and a 30-inch focal length. Directional patterns of this combination were taken, some examples of which are shown in Fig. 3. These patterns were taken with the lens-horn

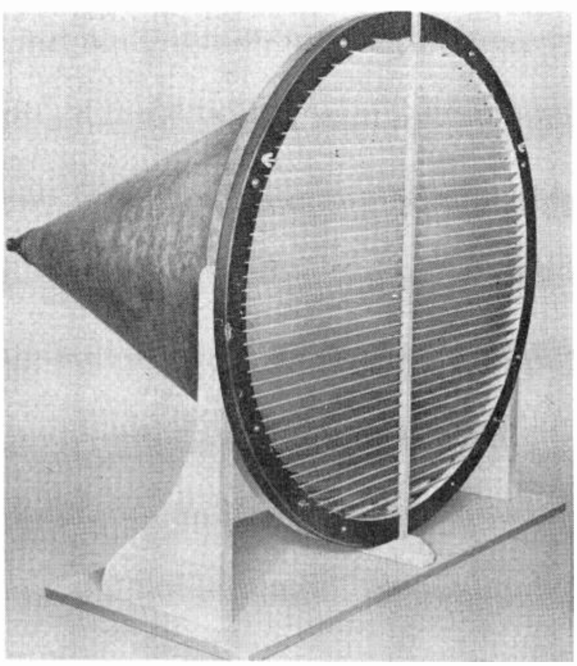

Fig. 2 – A lens-horn combination using a slant-plate lens.

acting as a radiator, and, by the principle of reciprocity, are the same as would be obtained with the combination used as a microphone. The angular width of the beam at the half-power points has been calculated from diffraction theory,[12] assuming that the lens aperture is uniformly flooded by the horn. The result is

$$\sin \frac{\theta_3 db}{2} = \frac{1.61}{\pi} \cdot \frac{\lambda}{D} \qquad (1)$$

where θ_{3db} is the angular width of the beam at the half power points,

λ is the wavelength of the sound waves,

D is the lens diameter.

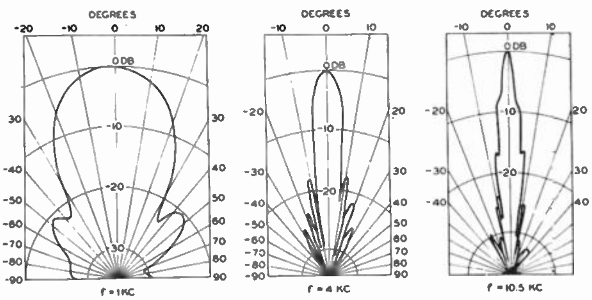

Fig. 3 – Directional characteristics of the lens-horn combination of Fig. 2.

For the lens-horn combination shown in Fig. 2, equation (1) is represented by the solid line in Fig. 4, which is to be compared with the experimental results shown by the vertical bars in the same figure. The lengths of the bars

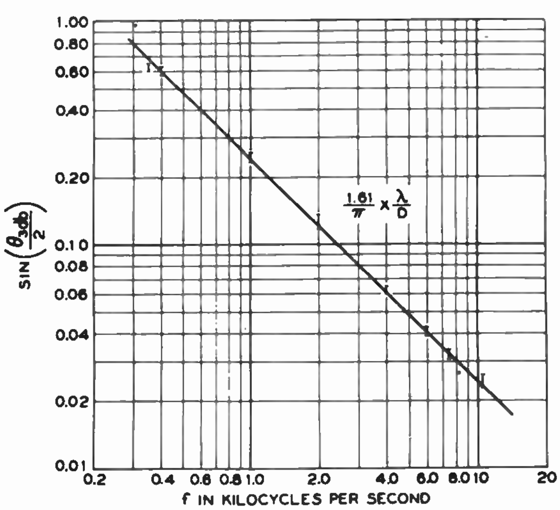

Fig. 4 – Theoretical and experimental beam widths for the lens-horn combination of Fig. 2.

represent an estimate of the errors in measurement of the beam widths. The directional patterns were taken at a finite distance of 27 feet from the lens, and ought not to correspond exactly to the calculated characteristics, which refer to an infinite distance. The agreement between experimental and theoretical beamwidths shown in Fig. 4, however, indicates that at a typical distance at which such a microphone might be used the uniform-flooding assumption is fairly good.

[11] Kock, W. E., *Proc. I.R.E.*, vol. 37, p. 852; 1949.

[12] See, for example, P. M. Morse, "Vibration and Sound", McGraw-Hill Book Company, New York.

It has been pointed out[5,13] that the gain and directivity can be made less dependent on frequency by defocussing (i.e., moving the transducer away from the focal point of the lens or paraboloid). In particular, if one defocusses by moving the transducer closer to the lens, the resulting characteristic corresponds to a diverging beam. The width of the beam resulting from this divergence is independent of frequency, while that due to diffraction varies with frequency. Thus by sacrificing overall sharpness of the beam one can make its width less dependent on frequency. At the same time the gain of the device becomes less dependent on frequency as well.

[13]See reference in footnote [1], p. 213. (1st Edition.)

METHOD FOR TIME OR FREQUENCY COMPRESSION-EXPANSION OF SPEECH*

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Urbana, Illinois

The purposes of this paper are to outline a method for compression and expansion of speech, to describe the device employed in the method, and to demonstrate by means of recordings the results of the method at this experimental stage.

Until comparatively recently we had not been aware of the fact that several approaches to the problem similar to ours had previously been made by other experimenters. We have now learned that our method, although developed independently, resembles in certain features of theory and details the earlier work of French and Zinn[1], Gabrilovitch[1], Haase[1], Gabor[2], Vilbig[3], and, perhaps, others.

Fundamentally, the process depends upon the fact that the duration of the average speech element or phoneme of live connected speech, such as *ah* or *s* or *r*, exceeds the minimum duration necessary for perception by a listener, or exceeds the minimum time necessary for sampling the essential phonemic qualities of the speech element in question. This minimum duration has been the object of a psychophysical study by Peterson[4] and of theoretical calculation by Gemelli and Pastori[5]. The excess duration may be referred to as *temporal redundancy*, which term we suggest as a useful specification at the experimental level when spoken language is in question.

The dimensions of the problem are clearly not only those of engineering, but also those of psychophysics. In this paper we confine ourselves to the method. A psychophysical program is in progress and its results will be reported separately.

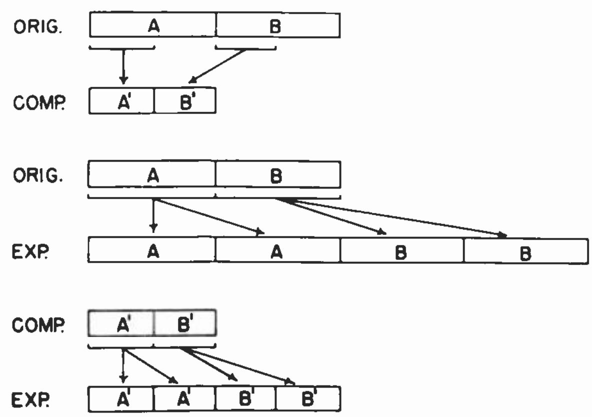

Fig. 1 — Theory of time compression and expansion by sampling.

For purposes of explanation assume two different phonemes, A and B, which are of equal duration and joined without interruption as shown (Fig. 1). Assume that A' and B' are valid samples of A and B, and that each is of adequate duration for perception. Assume that samples A' and B' are extracted from A and B and abutted in time as shown without discontinuity, and that $A - A'$ and $B - B'$ are discarded. If, now, A', B' is reproduced, the time will be shorter than the original A, B, but the phonemes should be perceptible.

When this proposition was advanced several years ago by the first author it was validated for connected speech by cutting and splicing magnetic tape at arbitrary points, without regard to the phonemes. It was discovered

*Revised manuscript received October 30, 1953. This is the substantial equivalent of a paper presented at the 1953 National Convention of the I.R.E. in New York City, Reprinted with minor changes from the Convention Record, Part 8 — Information Theory, pp. 120 - 124. Because this paper included a demonstration, references to the latter have been retained in the published form.

**Now at Bell Telephone Laboratories.

[1]Cited in Gabor.

[2]Gabor, D., *Jour. Inst. Elec. Eng.*, vol. 93, Part III, pp 429-457; 1946; vol. 94, Part III, pp. 369-386; 1947; vol. 95, Part III, p. 39; 1948; vol. 95, Part III, pp. 411-412; 1948.

[3]Vilbig, F., *Jour. Acous. Soc. Amer.*, vol. 22, pp. 754-761; 1950; vol. 24, pp. 33-39; 1952.

[4]Peterson, G. E., PhD. Dissertation, Louisiana State University; 1939.

[5]Gemelli, A. and Pastori, G., "L'Analisi Elettroacustica del Linguaggio," Milan, Italy, pp. 149-162; 1934.

that substantially more than 50 per cent of the total time of connected speech could be discarded by this means without destroying intelligibility. That is, $A - A'$ could exceed A'. At about the same time, Garvey and Henneman independently used the same cutting-and-splicing method to compress isolated words and found similar results.

In the case of expansion, assume that phonemes A and B are caused to be repeated, as in the middle portion. If A, A, B, B is reproduced, the time will be longer and the auditory effect, given the above assumptions, should be that of prolongation of A and of B.

Finally, assume that A and B are first compressed to A' and B', and then expanded to A', A', B', B' as shown at the bottom. Here the original time for A and B has been restored. A and B have been reconstructed from A' and B'.

arrows. Entering the device, the tape is directed by means of rollers over a Magnecord erase head, and then over a fixed Magnecord record head where the input is temporarily recorded. Passing over another roller, the tape then descends to a revolving playback head assembly enclosed in a mu-metal box, where signal recorded on the loop is scanned. Next the tape passes to the drive capstan, around a roller, and, finally, over a Brush permanent magnet erase head.

The revolving head assembly consists of a brass drum with four Brush playback heads equally spaced around its periphery. The output of the heads is taken off by means of a slipring-brush unit. The circumferences of both drum and capstan are 7.64". Drum and capstan are mounted on shafts supported in sleeve bearings at the back of the panel. Massive flywheels are also mounted on the shafts. The two units are driven by twin $^1/_{15}$ hp DC

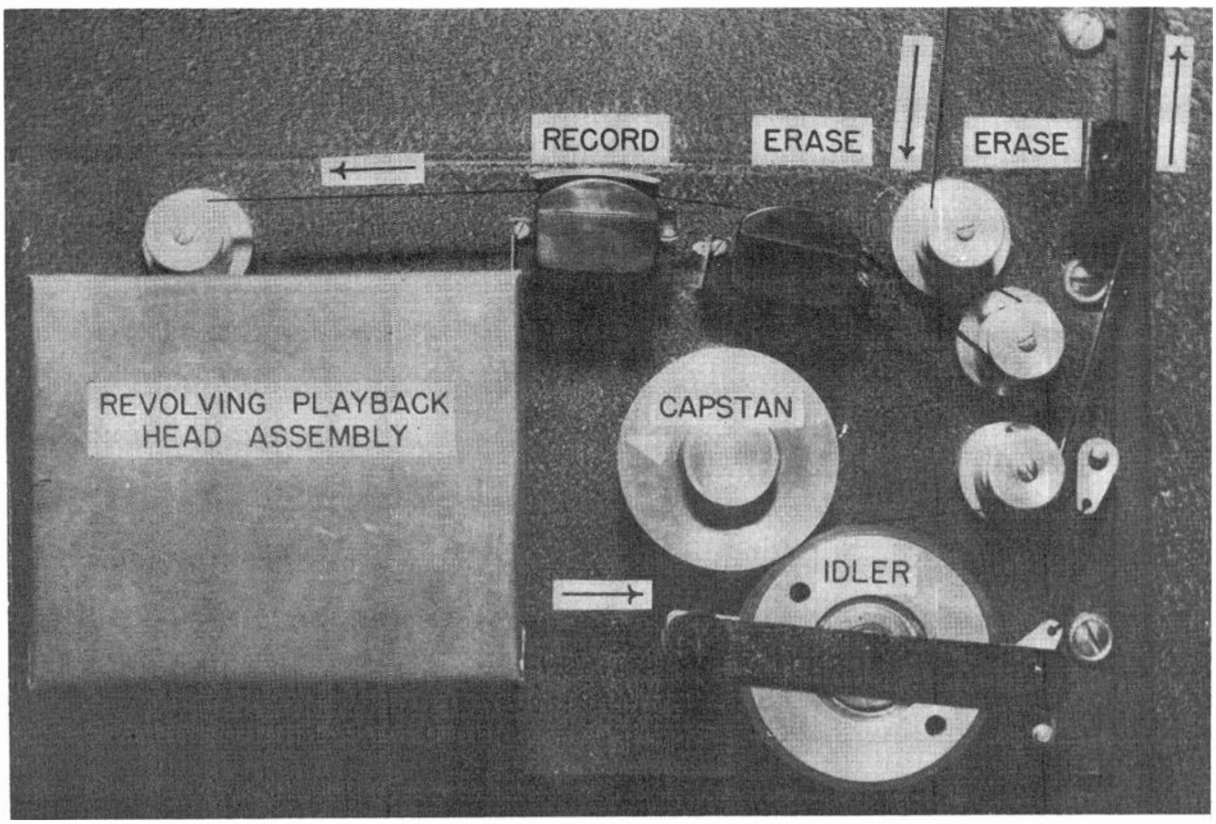

Fig. 2 – Apparatus.

Fig. 2 shows a photograph of the essential part of an experimental model of a device for compression or expansion along the lines of such a theory. Basically, the device is a continuous loop magnetic tape recorder, mounted at the bottom of the rack containing the other components. The tape loop, approximately 12 feet long, rises along the right edge of the rack to a pulley under slight spring tension at the top. Its pathway is shown by

[6]Garvey, W. D. and Henneman, R. H., Air Force Tech. Rep. #5917; 1950.

Bodine motors with independent speed controls by means of GR Variacs. Speeds are measured with a GR Strobotac.

The remaining components are conventional. An independent Magnecorder PT6-A is used for storage and playback. This has been modified for continuously variable speed reduction and furnished about a 15 to 1 range of tape velocities.

In Fig. 3 operation of the revolving head assembly is shown at the left. The four playback heads are identified by letters. The tape passes over the drum and is in contact with ¼ of its circumference, or a distance equal to

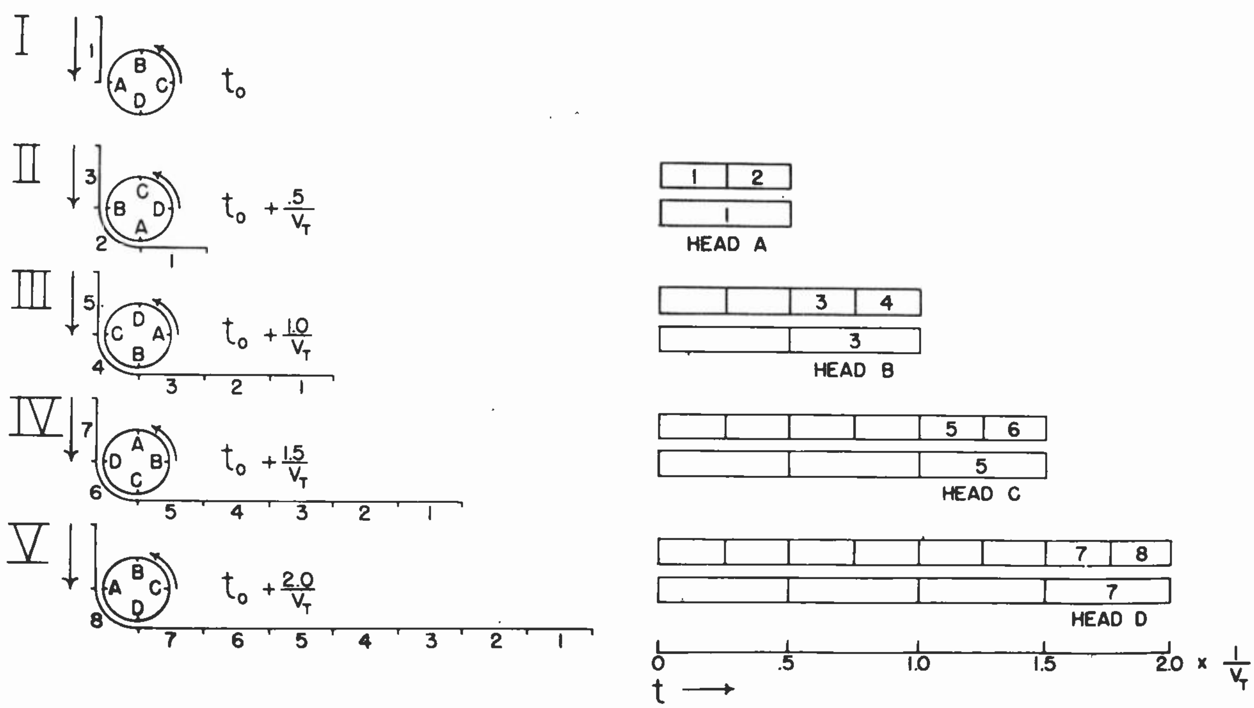

Fig. 3 — Compression process.

the peripheral distance between any two adjacent playback heads. The tape is retained by flanges around the drum periphery. Tape direction is constantly counter-clockwise. In the compression application the direction of drum rotation is also counter-clockwise. Under load the top tape velocity is approximately 190 in/sec. The top peripheral drum velocity is about 225 in/sec.

For purposes of explanation the tape is divided into hypothetical numbered segments, each equal to the distance between heads. The relative positions of tape and heads are shown at representative times. The diagram shows 50 per cent time compression as an example.

In Part I segment 1 is shown at t_0 when it first comes into contact with the drum. At this time it is intercepted by head *A*, which is moving in the same direction. If the drum were stationary, reproduction would be one-for-one. If its velocity were equal to the tape, no signal would be reproduced. Between times I and II, however, head *A* moves through ¼ of a revolution. During the same interval tape segments 1 and 2 pass the 9 o'clock point where head *A* was at t_O. As a result, head *A* reproduces segment 1 during that interval. The *effective* tape velocity is V_T - V_H. In the example diagrammed V_H equals $V_T/^2$ which equals the effective velocity. Therefore, the frequencies of segment 1 as reproduced by head *A* are divided by 2.

At time II head *A* is at 6 o'clock and head *B* is at 9 o'clock, while segment 2 lies between them in contact with the drum. Head *A* is about to leave the drum, while head *B* is about to begin reproducing segment 3. Accordingly, although there is no discontinuity, segment 2 is not reproduced by any head. The remaining diagrams show how the process continues, the odd-numbered segments being reproduced at reduced frequency and the even-numbered segments being discarded. It is evident that various durations of either reproduced or discarded segments can be realized by varying the absolute and relative velocities of tape and head, and that a range of sampling frequencies and compression ratios can thus be produced.

The output of the device with respect to time is diagrammed at the right. Between times I and II, for example, segment 1 is reproduced by head *A* in the time necessary for both segments 1 and 2 to pass a point. Head *B* then reproduces segment 3, etc. The final yield is segments 1,3,5,7. When these segments are stored at a given speed and then reproduced at an appropriately higher speed, their original frequencies are restored and the elapsed time is shortened.

With respect to duration the odd-numbered segments are termed *sampling intervals;* the even-numbered segments *discard intervals.* The reciproval of their summed durations is the *sampling frequency.* One hundred times the discard interval divided by the sum of the two intervals will be termed the *compression percentage.* Since sampling is periodic the ratio applies also to the total message time, and describes the percentage by which that total time has been reduced.

Assuming that the process results in intelligible speech, it becomes evident that the processed message may be transmitted over a system with smaller bandwidth than originally necessary. The capacity of a conventional transmission link for handling simultaneous messages will be a function of the amount of compression, or frequency division.

Fig. 4 is a similar diagram for expansion. Here the drum bearing the playback head revolves in a direction opposite to that of the tape. The illustrative example shows the condition when these velocities are equal. The effective velocity is equal to their sum.

At t_O, shown at I, segment 1 is in contact with the drum between heads *A* and *D*. During the next interval head *D*, as it moves from 6 o'clock to 9 o'clock, will reproduce both segments 1 and 2 and then leave the tape. At that time it will be replaced by head C, which has moved to the 6 o'clock position to intercept the tape at the beginning of segment 2, and which will reproduce segments 2 and 3 during its sweep. The result, shown at the right, is that between times I and II, while segments 1 and 2 are passing the 6 o'clock point, segments 1,2,2,3 are reproduced. The rest of the figure shows how this process continues.

Since the effective tape velocity has been increased by the opposite movement of head and tape, frequency multiplication has been incurred. The original frequencies are restored by reproducing the processed message in an appropriately longer time. One hundred times the amount of time thus added divided by the original time is the *expansion percentage*. In the diagram this equals 100 per cent.

Fig. 5 summarizes the various stages in compression. The comparative times and frequencies are indicated at the bottom. In an original time T_O and with original frequencies f_O, the input is recorded on the loop at the velocity V_T and scanned by the revolving head unit moving in a positive direction at $V_T R_C$. This yields the compressed frequency f_C shown at the bottom. Simultaneously the compressed signal is stored at a recording tape velocity which will be taken as V_R. This recording is reproduced at a later time at the higher tape velocity shown, in the relative time indicated, and with f_O restored. The following recordings will illustrate this.

In this and the other recordings you will "hear" repetitions of a semi-nonsense test sentence which provides a rigorous test of the system. The sentence contains one and only one example of every American phoneme, with exception of the unstressed neutral vowel as in the first syllable of the word *away*, which occurs three times.

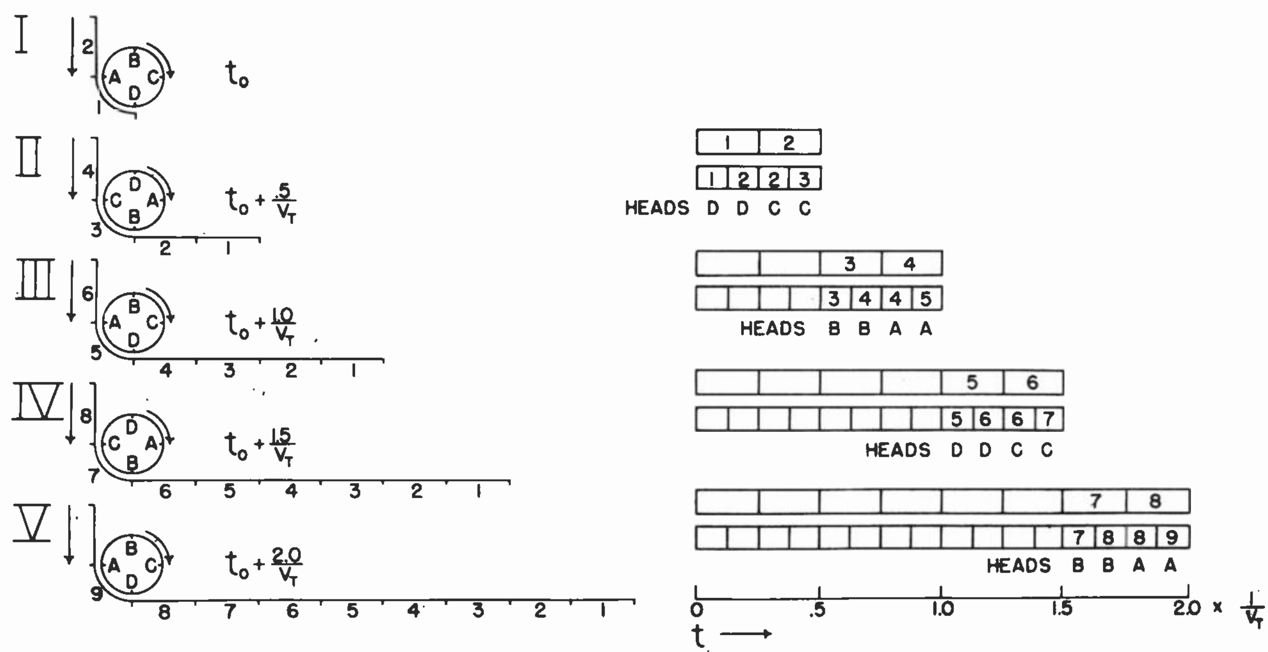

Fig. 4 – Expansion process.

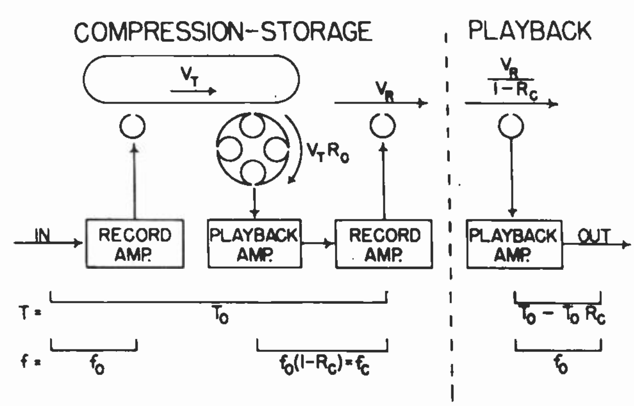

Fig. 5 – Method of time compression.

'Recording 1. Compression. Original message: We hasten the boy off my garage path to show which edge young owls could view. Frequency division 1.25. No time compression. Sampling frequency 10: (sentence). Time compression 20%: (sentence). Test out.'

Next you will "hear" the perceptual effects of various degrees of compression.

'Recording 2. Time compression series. Sampling frequency 10. Compression 10%: (sentence). Compression 30%: (sentence). Compression 50%: (sentence). Test out.'

'Recording 3. Time compression series. Sampling frequency 20. Compression 50%: (sentence). Compression 70%: (sentence). Compression 90%: (sentence). Test out.'

You will have noted that the smaller values of compression affect intelligibility and perceived speed of talking very little. Although both factors are perceptibly affected as compression is increased, you can observe that intelligibility persists with surprisingly large compression percentages.

sound. This occurs when the interval repeated exceeds the duration of one phoneme. This is a size limitation in our experimental model and not a limitation of the method.

Fig. 7 shows a system which involves the following: (1) compression, (2) transmission of the compressed message, (3) expansion of the compressed message. The steps are carried on simultaneously with two units. A transmission link, undiagrammed, is inserted between the two at the arrow. Velocities, times and frequencies are labeled.

The process is illustrated in the next recordings. First you will hear the original message. Then you will hear the transmitted message with frequency division. Finally you will hear the message as received after reconstruction by means of expansion and corresponding frequency multiplication. Eighty per cent of the message was discarded before transmission and the final message

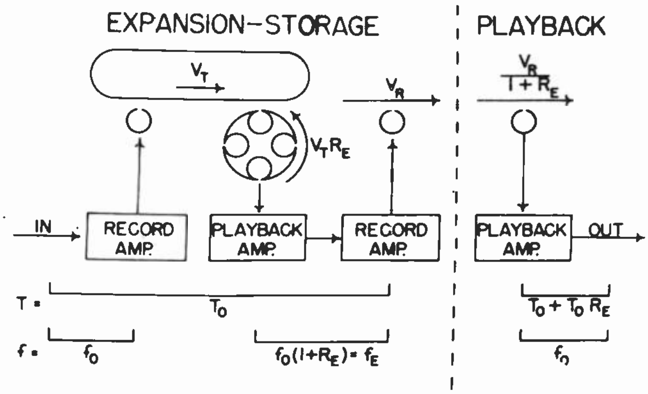

Fig. 6 — Method of time expansion.

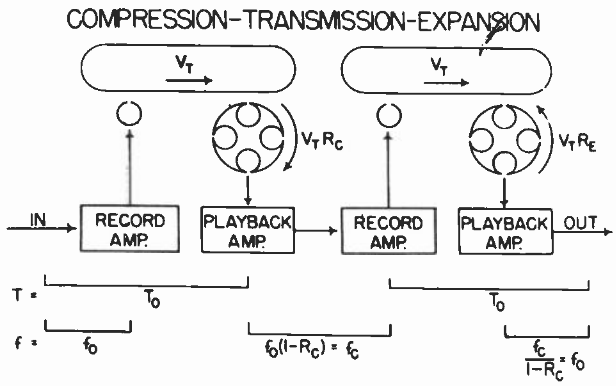

Fig. 7 — Method of frequency compression — transmission — expansion.

Fig. 6 is a similar diagram for speech expansion. Head movement is negative with respect to the tape, and equals $V_T R_E$. In the original time the original frequencies are multiplied by 1 plus R_E, yielding f_E as stored. The message is then reproduced at the lower velocity shown, f_O being restored with the time expansion. The next recording illustrates the three stages.

'Recording 4. Expansion. Original message: (sentence). Frequency multiplication 1.2. No time expansion. Sampling frequency 10: (sentence). Time expansion 20%: (sentence). Test out.'

We will now illustrate the perceptual effects of expansion. The expansion percentage will be progressively increased.

'Recording 5. Expansion series. Sampling frequency 10. Expansion 10%: (sentence). Expansion 30%: (sentence). Expansion 50%: (sentence). Test out.'

'Recording 6. Expansion series. Sampling frequency 33.3. Expansion 50%: (sentence). Expansion 70%: (sentence). Expansion 90%: (sentence). Test out.'

Note that small percentages did not affect the perceived speed of talking very much, and that the details of speech became more readily heard as expansion increased. Toward the end you may have heard an echo-like

as you hear it was reconstructed from the 20 per cent fragment that remained. To help you appreciate the last point we will also "play" at the end a recording in which the original frequencies are restored by accelerated playback without time expansion.

We present this next recording with some hesitation and we hope you will not be disappointed. It was made on an experimental model of the device. Its main purpose is to validate the theory and demonstrate potential feasibility. (You will "hear" considerable noise and distortion. Some of this can be eliminated fairly readily, but part of it is inherent in the method and will need to be counteracted.)

The important thing, however, is that the final output is intelligible at all when bandwidth reduction is by a factor of 5 and compression is 80 per cent.

'Recording 7. Compression — transmission — expansion. Original message: (sentence). Transmitted message. Original time. Frequency division 5. Sampling frequency 60: (sentence). Restored message. Original time. Frequency multiplication 5. Sampling frequency 16: (sentence). Time compression 80%: (sentence). Test out.'

Apart from its theoretical interest, the method appears to have several practical applications. For one thing, the smaller compression and expansion ratios should be useful in the programming of rebroadcast speeches in radio, since they furnish 'tailormade' time without the audience's knowledge. A saving of 10 minutes per hour is completely realistic. Conversely, and we advance this suggestion with diffidence, thinking of commercials, more intelligence can be communicated to an audience in a given amount of time.

Straightforward compression by larger amounts should be useful wherever high-speed communication is crucial, as in certain military situations. Expansion should facilitate branches of study such as experimental phonetics and linguistics where auditory analysis is important.

Finally, of course, the method gives promise as an approach to the long-standing problem of bandwidth reduction.

In conclusion we should like to "play" two more recordings. The first of these is self-explanatory.

> 'Recording 8. In order to demonstrate that the method is inherently practical, the recorded explanatory materials in connection with the recordings that you have heard today, as well as these words that you are hearing now, were all compressed by 10%. Test out.'

We are frequently asked if the method applies to women's voices, fast articulation, or music. The next recording illustrates its use with all three.

> 'Recording 9. Vocal music. Rosemary Clooney. *Come On-a My House*. Columbia record Number 39467. 78 rpm, shellac. Compression series. No time compression: (music). Compression 30%. Sampling frequency 20: (music). Compression 60%. Sampling frequency 40: (music).'

The final recording shows the effect upon music which has already been processed to make it ultra-fast. In the first section you will hear a portion of the original. The second section demonstrates 30 per cent compression.

> 'Recording 10. Compression series. Les. Paul. *Lover*, by Rodgers and Hart. From Capitol LP record Number H226, *The New Sound*. No time compression: (music). Compression 30%. Sampling frequency 20: (music). Test out.'

A DEVICE FOR TIME EXPANSION USED IN SOUND RECORDING*

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In volume three, 1948, of *Funk and Ton* there was described a device for finding certain recorded sound elements on recorded tape. To do that, the tape was usually standing still and the playback head was rotating. Gunka and Lippert have mentioned that this kind of device could be used not only for editing but also for sound analysis. Actually an article was found in American technical literature where a similar device was used for a tape recorder which was very similar to the German "Magnetophon".

The magnetic tape recorder "Magnetophon-Special" produced by AEG used rotating heads. That recorder was developed at the beginning of the last war and manufactured on a large scale. In that recorder, the time expansion device was called a "Zeitdehner". The basic principles will be discussed further in this article.

Modulated recording mediums for different sound recording systems, such as disc, film or tape, never hold the recorded frequency by themselves. By measuring the distance between two maximum groove amplitudes, or maximum opaque for film, or maximum magnetization on tape, we can determine the wave length λ For the frequency, we have to know the velocity for the recording medium

$$f = v/\lambda \tag{1}$$

Usually the velocity for recording medium remains the same in the recording and playback processes. Any unwanted change in velocity, caused by speed variations will cause variations in the reproduced sound pitch.

By increasing the speed of recording medium carrying a recorded message, we can transmit the message more cheaply, because the time is reduced to use cable line and the charge will be less. Furthermore, there will be a need to slow down recorded dictation for the typist. By slowing down the speed in the usual fashion, the intelligibility will be changed and it would result in a complete misunderstanding.

The velocity v in (1) represents the relative velocity between recording medium and recording or playback head element. Usually, the recording or playback head will not

EDITORIAL NOTE. – Because of the widespread interest in the preceding paper by Fairbanks, Everitt and Jaeger, and related work in this country, a translation has been made of a German paper by Dr. H. Schiesser which appeared in *Funk Und Ton*, vol. 3, no. 5, 1949. This translation is published with the permission of *Funk Und Ton*. The illustrations have been reproduced from photostatic copies *of the original publication.*

be in motion. Gunka and Lippert described a device where the recording medium stands still, and the playback head element moves.

In general, it could be arranged that the recorded medium and the playback head element both are in motion. Velocity for the recorded medium is v_t and for playback element v_a. Recording done by a relative velocity v will give us the same pitch in playback at *two* different velocities for the recording medium, depending on the relative direction of motion between medium and playback head element.

$$v_t = v \pm v_a \tag{2}$$

As long as the pitch in playback is determined by the relative velocity between recording medium and playback head, the tempo by which the recorded information can be reproduced is determined by the velocity of the playback gap moving along the recording medium.

From a design standpoint, the playback gap striking area on tape is determined and the tempo for reproduction depends upon the speed of the recording medium.

Expanded playback requires diminishing the velocity of recording medium v_t with respect to velocity that was used in the recording process. To simplify the problem, a stationary head was used in the recording process, and in that case the velocity for recording medium is equal to the relative velocity between medium and recording device.

The expansion factor will be

$$d = v/v_t \tag{3}$$

The above-mentioned relation represents the ratio between the time for recording and reproduction.

For d smaller than 1, it appears as a time compression. Fig. 1 curves, where, by a chosen certain amount of compression or expression d, there can be found the corresponding velocity for recording medium in playback and the two velocities $\dot{v}_a$ for the playback head rotational motion, to get the same pitch as it was recorded.

The resulting velocities will be multiples of the recording velocity V. Only in one special case $d = 1$, the motional velocity for the gap along the medium will be the same as the relative velocity between medium and playback device. And only in that case is it possible to have uninterrupted continuity in playback. It can be seen that a time compression in playback by keeping the original pitch, could be done only by partially sampling the recorded information from the tape. For time expansion we have partly to overlap the recorded information by our playback gap striking area or by introductory interruptions. By introducing interruptions or by repeated (overlapped) reproduction, the samples of the separate recorded signal elements and the interruptions should be short so as to avoid introducing too much discontinuity so as to be out of range where human hearing system distinguishes sound separations. That principle used here is similar to human optical inertial that is encountered in moving pictures.

The interrupted "take off" procedure is done periodically by using a rotating or oscillating scanning device. The disc recording is rather difficult, but it is easier in magnetic or photoelectric recording.

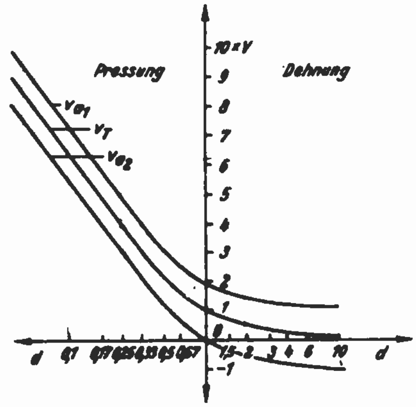

Fig. 1 – Velocity for recording medium v_t and the playback device v_a as a multiple of velocity used by recording vs. compression expansion factor (d).

The gap must move periodically along the recording medium at a velocity v_a and displacement h. At the end of determined displacement h, the gap must jump back to starting point and repeat the same procedure. By photoelectric recording, that motion can be reached by using a rotating spiral opening in the path of the light beam.

For magnetic tape recording, it is more convenient to use a rotating device with many gaps distributed around a rotating drum.

The wrap angle for tape around the playback head with n number of gaps, must be wider than $2\pi/n$ and there must be provisions that allow only one effective gap in contact with the tape in this sector.

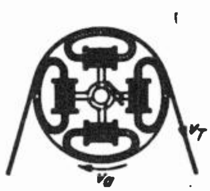

Fig. 2 – The basic principles of a rotating device, using four heads 90° apart and electric current commutator.

Fig. 2 represents the rotating head arrangement for AEG tape recorder. There are four independent heads mounted on a base. Every one is a closed ring head by Schuller with a proper gap and core made of high permeability material. Magnetic potential developed in the gap creates an EMF in the coil. The lead ends from coil are going to a collector lamination, and then from collector to the input of playback amplifier. By this collector arrangement, after 90° rotational displacement of the gap, the point of pickup returns to the beginning of the sector. The very low current through collector causes some troubles, and that is why a "magnetic switching" arrangement will be better. (The induced voltage into the coil is around 1 mv).

Fig. 3 represents this so-called "magnetic switching" device basically. This in rotational motion is only a cylinder made of hi-μ material. The cylinder or drum has four gaps distributed evenly around this surface. The magnetic flux lines come from the effective gap through a part of the drum, and then by the shortest path through a wide overlapped gap into the pole piece with the coil.

The playback effective gap displacement h will be the distance between two gaps. For one playback element

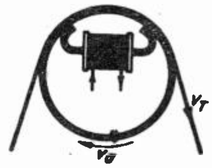

Fig. 3 – The basic principles of rotating playback head by using four playback gaps 90° apart and magnetic commutator.

the distance that the effective gap strikes the medium is L. L is different from h, because the tape is in motion.

$$L = hv/v_a \tag{4}$$

The sequence for separate "picking ups" for the two corresponding spots on the tape will be e.

$$l = hv/v_a d \qquad (d = \text{compression-expansion factor})$$

For $d = 1$, $L = l$, indicating that the distance between two corresponding spots contacted by two sequentially following "picking-ups" will be the length of the partial element on tape. The playback will follow with an uninterrupted continuity. For d smaller than 1, (meaning time compression) there will be introduced dead spots between two played back elements. For d larger than 1, (meaning time expansion) there will be overlaps of played back elements; every separate spot on tape gets reproduced d-times.

Fig. 4 represents time-displacement diagram. The horizontal axis represents the displacement of the playback gap on the recorded medium. An example will be for $d = 4$, and $d = 0.5$, the playback head is rotating in one direction for one diagram and in the opposite direction for the other diagram. The gap displacement h is in both cases the same. The projection of arrows on horizontal direction will represent the length L by which the playback gap strikes the tape. The distance between the arrow points will represent 1. The slope of the arrows is reversed proportional to the velocity v_a of the playback device. The slope of the center line of arrows is revised proportional to the velocity v_t of the recording medium. As it can be seen from examples on b and d, the separate elements get played back reversed since the motion of medium and playback device are in the same direction. That way we get larger amounts of separate short elements; by using an opposite motion for the head and tape, there will be less separate elements, but longer.

The number of elements will be greater when d is smaller. To get a better continuity for listening, the separate elements should be many and small, and that is why for time compression it is necessary to have the same motional directions for both the medium and the playback device. In that case, the separate elements get reproduced backwards, but that doesn't matter since the elements are small enough.

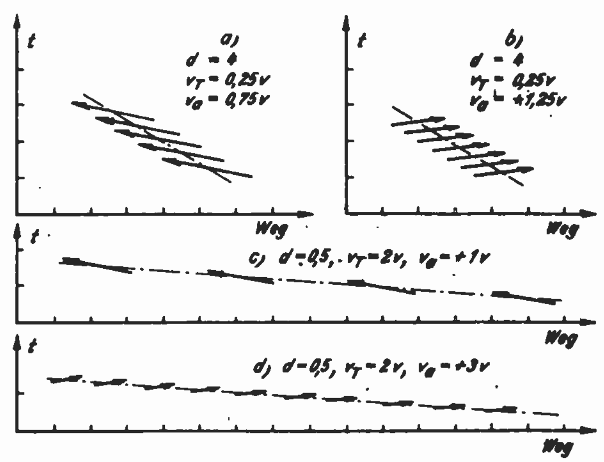

Fig. 4 – Time-displacement diagram for interrupted playback for $d = 4$ (expansion) and $d = 0.5$ (compression) for both velocities v_a for the playback device.

A tape recorder with rotating head assembly for the most unfavorable condition in playback which was represented in Fig. 4, diagram b, and uses a gap displacement $h = 2.5$ cm and recording tape speed of 75 cm/sec. The separate element length in playback is $L = 2$ cm. That corresponds to a 26.6 millisecond recording or it will be ¼ syllable.

That length of the separate element is short enough to give us a sufficient continuity in playback and the elements could be played back backwards without introducing noticeable distortions.

By the above discussed example, 38 elements per sec. were played back with the listener having the impression of discontinuity. It is supposed that much greater expansion would not be necessary in general use. For special occasions where we have to use a much greater expansion, the tape speed has to be increased in recording.

CONCLUSIONS

The previously discussed article should explain the problems in using an expansion or compression for sound recording. This process is possible in the present recording systems used for magnetic or photoelectric recordings. The time expansion or compression is used for dictating machines, for more economical use of cable lines in communication and for scientific purposes especially for phonetic research.

MECHANICAL COMPONENTS FOR HANDLING MAGNETIC RECORDING TAPE*

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GENERAL DESIGN OBJECTIVES

The object of this paper is to discuss all the component elements in the tape path which contribute to the tape handling and to the characteristics of the recording which are dependent, to a great extent, upon the mechanical tape transport unit.

The objective of the design of a mechanical tape transport is to provide means of storage for magnetic recording tape both before and after recording and to translate the tape uniformly at a controlled and even speed past a point where the tape may be erased, recorded or reproduced. An additional design objective is to provide such flexibility of mechanical design as to allow fully automatic control when desired, and to incorporate safety features and means for remote control. The result of the mechanical design should yield safe mechanical and environmental conditions for the magnetic recording medium.

GENERAL MECHANICAL CONFIGURATIONS

One of the usual tape transport configurations is shown in Fig. 1 as a bare panel view, and in Fig. 2 as a completed unit. The tape pays off from one reel, passes over a non-rotating alignment guide which is combined with a spring loaded compliance arm, then passes over a tape stabilizer roller before being fed to the head assembly where the erasing, recording and reproducing are done; following the head assembly, the tape is pulled and its motion metered by passing between the capstan driven at a constant speed and a pressure roller with a rubber rim; following the capstan tape puller, the tape is fed over another non-rotating tape guide combined with a compliance arm and fed to the take-up reel.

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In some tape machines, the capstan is placed at the input to the head assembly. Either arrangement may be used except that it is believed some inherent advantages accrue in having the capstan beyond the head assembly as follows: 1) Should the tape break beyond the capstan or should the tape take-up reeling mechanism fail, the recording may be safely made even though the tape piles up on the floor. A floor pile-up of tape may be readily cleared by reeling the tape up carefully without disturbing the pile. 2) A faster start is possible with a tape puller. The pressure roller may be engaged with the continuously moving capstan, thereby bringing the tape from standstill up to its normal speed in less than a tenth of a second.

The compliance arm following the tape puller is used to take up the momentary starting slack until the take-up reeling mechanism has a chance to get started. Some machines use a transient surge of power in order to overcome the reel starting inertia and to prevent throwing a tape loop during the starting period.

TAPE STORAGE MECHANISMS

The most generally used form of tape storage is that of a simple reel. Other means of storing tapes are: 1) by passing the tape over a number of spools which may be distributed about a panel, or 2) by feeding the tape into a tape tank just wide enough to receive the tape and within which tank the tape makes a number of folds back and forth upon itself. The tape is usually fed from the top and pulled out from either the side or bottom of the tank. 3) by using a spiral endless loop device where each layer of tape slides on the adjacent layer. These devices are utilized in endless loop service wherein it is not required to store the length of tape required in the usual recording operations. Their application is limited to tape lengths of several hundred feet.

TAPE REELING MECHANISMS

The storage of tape on reels is the usual mode of operation in most recorders. Nationally, we have standardized upon 7", 10½", and 14" diameter spools. It is to be

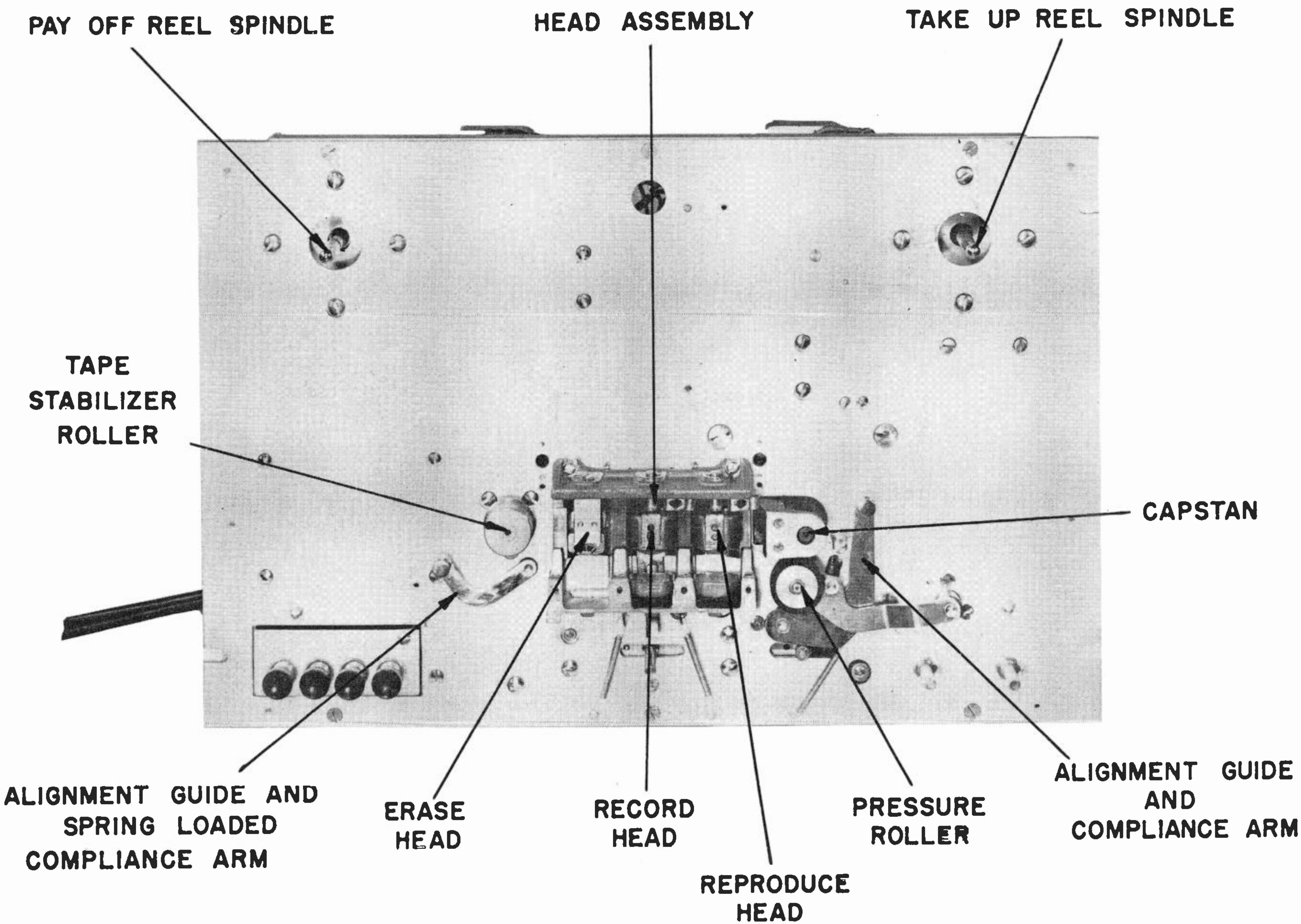


Fig. 1 – Tape transport – bare panel view.

noted that the position of the drive slots in the reels have been the subject of tacit national standardization in order to obtain reels which are interchangeable between tape machines. The most common tape width is, of course, ¼" at the present time.

In the usual design of tape handling mechanisms a spindle is provided with a fixed flange onto which the tape reel can be slipped. The reel spindle is usually mounted on some sort of bearing assembly behind the front panel of the recorder and may be attached directly to a torque motor shaft to save the cost of extra bearings. The spindle shaft is controlled in one of several ways in order to put either back or forward tension upon the tape for pay-off and take-up reels respectively. This holds the tape taut with from 2 to 8 ounce tape tension so that it does not flop loosely over the recording heads.

There are three basic reeling control means in use at the present time. 1) Friction disc clutches which are adjustable in order to control the spindle drag and consequently the tape back tension. The usual felt-to-brass clutch is designed from the formula

$$T = FN \frac{(r_1 + r_2)}{2} \mu .$$

where T = torque
F = axial force
N = number of clutch surfaces
μ = coefficient of friction = .3 for felt against brass
r_1 and r_2 = the inner and outer clutch radii

2) Torque motors, (Fig. 3), operating at full voltage during high speed reeling and at reduced voltage during normal forward operation of the machine in order to provide both back torque and forward take-up torque; motor braking for the pay-off reel may also be applied by means of energizing with dc the windings of an ac motor.

3) Servo-controls which operate through servo motors or magnetic clutches. These latter are the more elaborate and costly systems, and operate through sensing tape tension and correcting the reel torque accordingly.

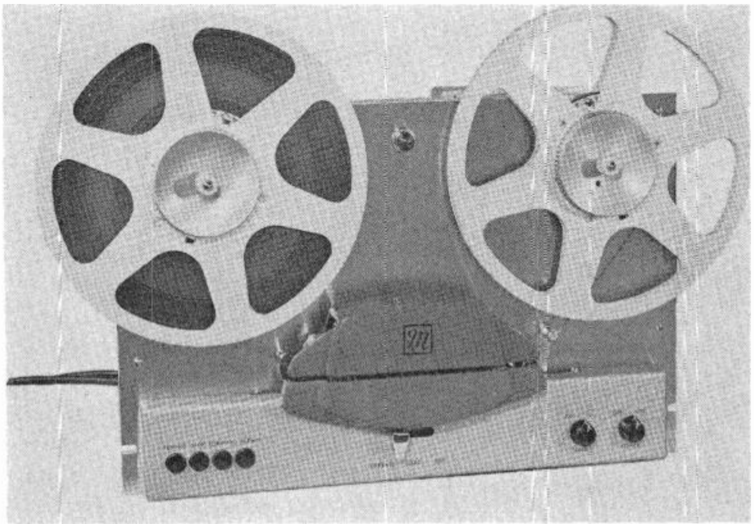


Fig. 2 – Tape transport completed – panel view.

These schemes have been used to obtain essentially constant tape tension. It is to be noted that in the first two of the three control modes mentioned above, constant torque rather than constant tape tension is obtained. It should also be noted that the true design objective is to obtain constant tape tension over the magnetic record/reproduce heads, and thus maintain a minimum disturbance to the capstan pull so that minimum flutter or wow result. The tape tension force F is related to the torque, of course, by the simple formula:

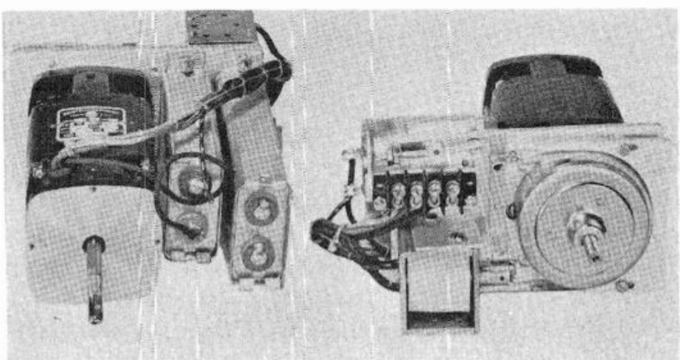


Fig. 3 – Bi-directional torque motors with condensers and band brake assemblies.

$$F = \frac{\text{Torque}}{\text{Radius of tape pay-off}}$$

Since the torque is essentially constant with either torque motors or any other good constant torque device, it will be seen that the tape back tension varies as the ratio of the outer reel to the tape hub diameter. The tape back tension desired is usually in the neighborhood of 2 to 8 ounces and the tape take-up tension about 6 to 20 ounces. Typical torque motor curves are shown in Fig. 4.

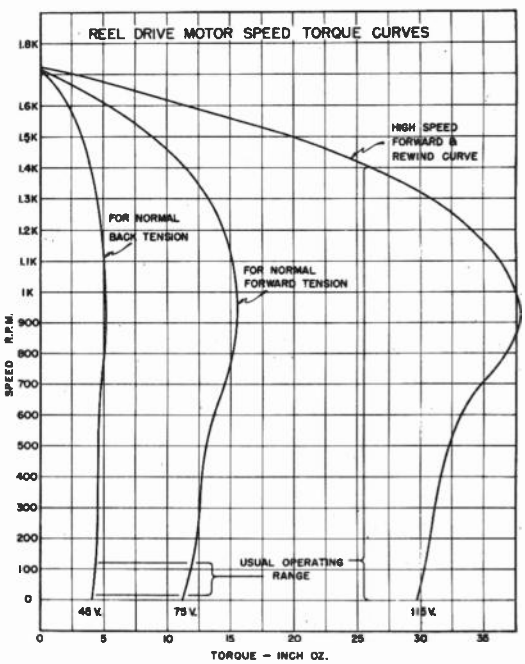


Fig. 4 – Typical torque motor speed-torque curves.

In most commercial tape reels today, an effort has been made to minimize this diameter variation. Maximum to minimum tape pay-off ratios are now usually held to approximately two to three, although a few years back this ratio was as high as five. (See Fig. 5.) To date, no really satisfactory simple automatic mechanical device has been devised to compensate for this diameter change. In the motion picture industry, where large diameter ratios and heavier, wider "tapes" are used; tape follower arms have been installed which continuously measure the diameter of the reel and adjust the reel torque accordingly to obtain essentially constant tape tension. When ¼" wide tape is used, mechanical clearances are such that this device is tricky to use. The fragile nature of the ¼" wide tape, only .002 thick, also militates generally against the use of the mechanical follower arm. Devices depending on the weight of the tape to determine tape tension have also been used, but for only vertical operation of the machine.

High quality professional machines may use any of the three reeling control methods listed above, but it is felt that the simplest and most dependable method of reel control is through a torque motor.

	LATEST REEL DIMENSIONS				PLAYING TIME	
NOMINAL REEL SIZE	TAPE I.D.	TAPE O.D.	O.D./I.D. RATIO	ACTUAL PLASTIC TAPE LENGTH	7.5"/SEC.	15"/SEC.
7" O.D.	2 3/4"	6 7/8"	2.5	1250 FT.	30 MIN.	15 MIN.
10½" O.D.	5"	10"	2	2500 FT.	60 MIN.	30 MIN.
14" O.D.	5"	13 3/4"	2.75	5400 FT.	120 MIN.	60 MIN.

Fig. 5 – Present day reel pay-off ratios and playing time.

When a motor is used, the torque characteristics through every degree of rotation of the motor are important since it is desired to maintain constant tape tension independent of the angular position of the reel. Some problems have been encountered in this respect since it has been found that motors may be built for "constant" running torque, but that during slow speed operation (15 to 60 RPM) within one revolution the torque may vary as much as 25% in extreme cases. In one investigation of a quantity of torque motors, it was found that variations in the air gap and in the electrical properties of the rotor caused motor torque variations within one revolution as shown in Fig. 6, Curve A. In a cooperative program with a motor manufacturer, this torque variation was reduced to that shown in

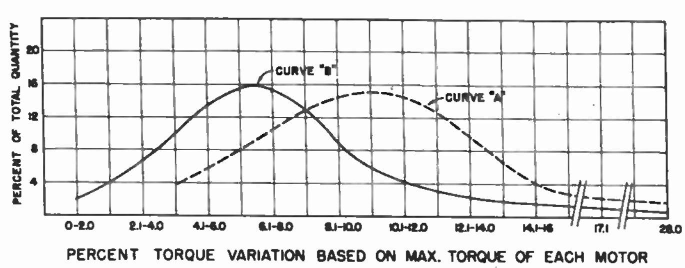


Fig. 6 – Torque motor torque variation within one revolution before (A) and after (B) quality improvement.

Fig. 6, Curve B. The effect of the high torque variation was to insert a flutter rate, which is to say, a tape speed variation rate in the order of one or two cycles per second. Reduction of the torque variation was sufficient to improve recorder performance appreciably and to allow other tape handling elements to further improve the constancy of tape speed.

TAPE STABILIZATION AND MOTION FILTERING

The tape stabilizers located on either side of the head assembly form a portion of the filter system which prevents any reeling or reel mechanism irregularities from affecting the smoothness and straightness of tape travel over the heads in order to hold flutter and signal drop-outs to a minimum while maintaining adequate tape wrap around the head gap magnetic structure to obtain good low frequency response. These stabilizers take the form of flywheel loaded pulleys over which the tape passes. The tape wrap around these pulleys must be made sufficient so that, with the tape to pulley surface coefficient of friction, no slippage can occur in normal operation. Further, one of these inertia or flywheel loaded pulleys (preferably the one following the heads) is utilized as the main tape drive element and is therefore designated as the capstan which is motor driven either directly or through some speed reducing means. Both the stabilizer roller and the drive-motor-stabilizer assembly require close tolerance precision honed-oilite (±.00015 dia. tolerance) or precision ball bearings. It becomes of importance to remove heavy flywheels during shipping in order to avoid Brinelling of the bearing races or bending of the flywheel shafts, which could cause flutter later.

Typical tolerances on a capstan with a nominal diameter of .2367" moving at 600 rpm to achieve 7½" per second tape motion are as follows: diameter tolerance ±.0002; run-out tolerance ±.00005. Such accuracies assure 7½" per second tape speed flutter below 0.15 per cent and tape timing with an accuracy of ± 3 seconds in one-half hour of recording. The other flywheel loaded pulley is driven by the tape itself.

If A~C~ torque reeling motors are used, it is the usual thing to expect a certain amount of 60 cycle and 120 cycle hum to appear in the tape tension. The stretch or compliance of the tape between the tape reel and the spring loaded compliance arm acts with the tape stabilizer inertias to absorb this 60 cycle and 120 cycle variation in tape tension, and to completely eliminate this effect from reaching the head recording area.

Examining the tape, the tape compliance arm and both stabilizers as a filter, it may be said that we have a low pass filter. It is desirable to push the filter cut off down to as low a frequency as is practicable, considering the physical size of the flywheel used in the stabilizer. This is desirably below one cycle and in practice can be easily held to about ¼ to ½ cps. An illustrative calculation follows:

Considering the pay-off reeling mechanism, both stabilizers and the first compliance arm for a practical case, we then have:

$$F_R = \frac{1}{2\pi\sqrt{IC}} \text{ and } C = \frac{\theta}{F}$$

where

F_R = low pass filter resonance (near cut off) frequency in c p s

I_1 = tape driven stabilizer rotary inertia

$= \frac{wt}{2}(R^2 + r^2)$

$= 2.47 \times 10^{-4}$ slug ft²

I_2 = motor driven stabilizer inertia

$= 2.25 \times 10^{-3}$ slug ft²

C = equivalent arm compliance

ϕ = arm deflection in radians = .75 radians

F = arm deflection torque $= \frac{1}{192}$ lb ft.

Then for the tape driven stabilizer:

C = .75 x 192 = 144 radians per lb ft

$$FR_1 = \frac{1}{2\pi\sqrt{.000247 \times 144}} = 0.85 \text{ cps}$$

This cut-off frequency filters out the 60 cps and 120 cps flutter components. The tape couples the motor drive stabilizer to the driven stabilizer so that even lower frequencies are filtered out by the combination of both stabilizers:

$$F_R\,(1+2) = \frac{1}{2\pi\sqrt{.000247 + .00225) \times 144}}$$

$$F_R\,(1+2) = 0.267 \text{ cps}$$

This last value is very satisfactory. The actual mechanical arrangement and simplified electrical analogue is shown in Fig. 7.

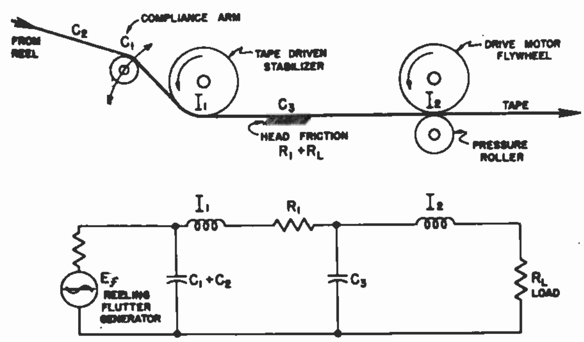


Fig. 7 – Tape transport simplified mechanical-electrical analog schematic. Hi-frequency flutter source at heads neglected.

Present day professional tape recorders are capable of maintaining flutter between 0.05 and 0.3 per cent. The weather may actually enter into the flutter performance of the tape recorder, since stickiness of the tape affects its travel over the heads and other elements in the tape path. Tape is both thermoplastic and humidity plastic. Sticky tape may cause chatter, which is to say variable speed or flutter.

TAPE DRIVES

Over the years that recordings have been made on a moving medium, various driving methods have been used. The modern tape recorder draws from all previous experience and knowledge and has used many various drives and speed reducers. Since for speeds below 600 rpm, overly large motors are necessary, it is desirable to have the drive motor running at 600 rpm or some speed in excess of 600 rpm. A speed reducing scheme is then normally used in order to obtain adequate capstan torque and the low speed from a small drive motor. Standard motor speeds used have been 600, 900, 1200, 1800 and 3600 rpm. General practice has established about 5 pounds available capstan pull to do a satisfactory tape driving job. The actual average pull on the tape is much less than this, about 6 to 8 ounces.

For a professional or industrial type recorder, it is usual to select a synchronous drive motor in order to minimize actual tape speed errors. Since tape has no sprocket holes to eliminate slippage, the design philosophy is to make all other parts of the drive mechanism as accurate as possible, thus making the major speed error that of the actual physical tape slippage itself. In the smaller fractional horsepower ratings a straight synchronous motor is not too efficient insofar as physical size, starting torque, and running torque are concerned. Therefore, the trend has been toward the use of hysteresis synchronous motors in designing these mechanisms. Hysteresis synchronous motors have the drawback of running hottest when lightly loaded, thus making heat dissipation a problem.

Speed reduction methods that have been used (see Fig. 8) are:

(a) Gear reductions with their usual complications of gear tooth modulation and worm irregularity modulation. These effects can be minimized in design by the use of helical gears and by using multiple thread worms. Gears and worms are difficult to handle, insofar as flutter elimination is concerned, because they generate low frequency flutter which most easily passes through the low pass filters discussed previously under TAPE STABILIZATION and MOTION FILTERING.

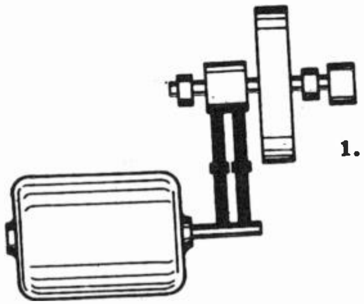
(b) Puck speed reducing systems, where a rubber rimmed puck reduces the speed between the motor shaft and a driven hub, which in turn directly drives the capstan and reduces the motor speed in the ratio of approximately the drive motor shaft diameter to the drive hub diameter. This puck reduction in speed must be corrected for an error which is the result of the difference of indentations in the rubber puck by the drive motor shaft, and the larger driven hub shaft. The puck is maintained at the normal puck optimum wedge angle of 114°. This angle is that which is sub-tended by the driven hub and motor shaft with the puck hub considered as the apex of the angle. Absolute looseness of the puck support is essential in order to assure self-centering action with its resultant uniform drive. Some recorders use two parallel pucks in order to assure uniformity of drive. A special form of puck may be used which is essentially a rim drive. The puck effect is obtained by putting a rubber rim around the driven hub and resting the drive motor shaft against this rim. An alternate system is to place a small rubber drive tire around the motor shaft, which in turn rests against a solid driven wheel. This latter configuration is not as satisfactory as the others because the speed error is greater in this case due to the rubber indentation in the small motor roller causing a greater error than a similar size indentation in a larger driven hub.

(c) A toothed compliance belt, which is an efficient and accurate speed reducing means, but which produces flutter within itself at both the belt rotation rate and the belt tooth rate. It is then necessary to utilize a mechanical filtering system beyond the driven hob to eliminate these two flutter rates. This system is somewhat more complex and more difficult of deriving satisfactory results than those previously mentioned.

(d) An endless woven flat belt. Due to the compliance of such a belt, this method is ideal insofar as it eliminates flutter components which might come from the motor. However, because of the vagaries of woven materials and because of belt slippage, the belt drive average speed control is not as precise as some of those previously mentioned. In the usual professional machine wherein high accuracy of tape timing is desired, it is then necessary to evolve a more complicated drive speed control system wherein servo elements are used to control or meter the tape. This is usually handled through recording of a standard control frequency on the tape so that when it is played back, the reproducer can be controlled to play with exactly the original timing. This standard recorded control signal has taken the form of superimposed 60 cycle modulation, a high frequency carrier 60 cycle modulated, and printed patterns on the reverse side of the tape from the oxide surface.

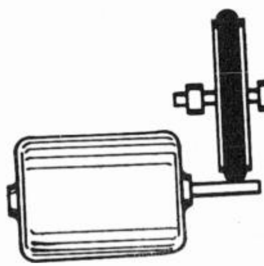
(e) A direct drive system wherein the tape is driven directly by an accurately ground area located on a shaft extension of the motor. This is perhaps the simplest accurate drive system possible, but requires careful design in order to reduce the flutter effects caused by shaft run out, bearing and housings tolerances, and vibration due to being directly coupled to the motor. This design is capable of good results if as low a speed motor as is practicable is used to prevent reducing the tape drive diameter of the capstan to too small a physical size. Experience has shown that diameters below approximately 0.2 inch should not be used. Too small a drive diameter is not only physically weak and subject to bending to

RUBBER INTERMEDIATE IDLER ROLLER OR PUCK REDUCERS

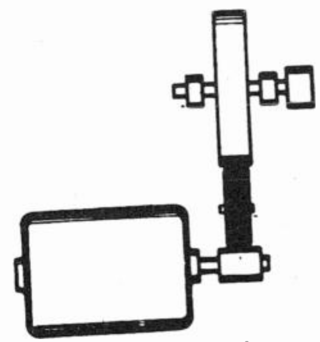


1. CAPSTAN HUB SHAFT DRIVE (Used in Magnecorder PT6-A)

Two independent pucks drive hub on capstan shaft . . . permit full flywheel action to minimize flutter. Provides constant contact and least possible slippage.

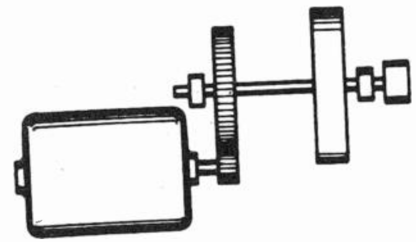


2. RUBBER TIRE FLYWHEEL DRIVE

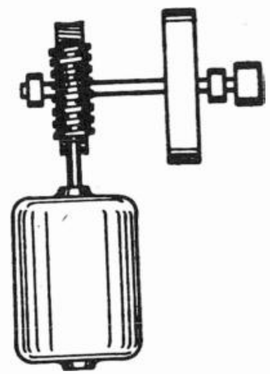


3. FLYWHEEL RIM DRIVE

GEAR REDUCERS

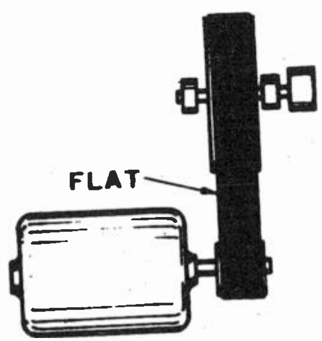


SPUR, HERRINGBONE OR HELICAL GEAR DRIVE

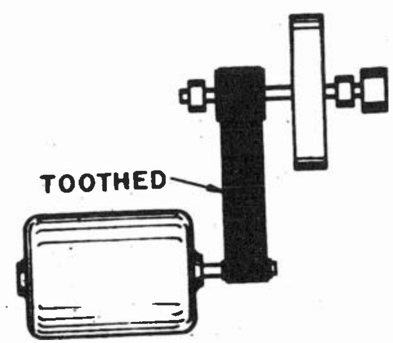


WORM AND WORM GEAR DRIVE

BELT REDUCERS



1. FLYWHEEL RIM DRIVE (BELT)



2. CAPSTAN SHAFT HUB DRIVE (BELT)

Fig. 8 – Speed Reducer Systems.

cause flutter, but should any oxide deposits from the tape build up on the capstan, their effect in speed deviation (once per capstan revolution) is greatly exaggerated.

TAPE GUIDANCE

Proper tape guidance first assures uniform wind on both take-up and rewind reels. Secondly, it assures the tape of a uniformly straight linear motion with no snaking or side to side motion over the heads. Since ¼" tape is normally held to slitting width tolerances of +0 -.002 inches, it is seen that the slot or guides which control the tape movement from side to side may be fairly accurate. Too much control, that is to say, too tight a pressure on either side of the tape will result in tape buckling so that an added means of flutter could be induced in a machine or poor tape contact with the heads might be obtained.

It is usual to establish the compliance arm tape guides to control reel winding and in addition to use some means in the head area to achieve control over the tape movement from side to side. Tape slither or skating from side to side causes the tape transverse axis to be no longer parallel to the longitudinal axis through the head area (to which the head gaps are aligned at 90° during the so-called azimuth alignment). This, of course, results in momentary head to tape misalignment so that high frequencies are not reproduced at normal amplitudes. In some cases where tape is not properly controlled, the high frequency output near the upper limit of the tape recorder may vary as much as 20 or 30 db. This amplitude variation should be held to one or two db at the most. Surface and contour squareness of recording heads, pulleys, guides, etc., are critical in order to minimize tape skating.

CONTROL FEATURES AND MISCELLANEOUS

The control features of a recorder are aimed at making all conditions of tape travel as automatic as possible so that they are completely controlled by the transport with a minimum of control effort expenditure by the operator. Thus, after loading the machine with tape, the operations of normal forward, high speed forward and high speed rewind, should be available to the operator at the touch of a button. This requires mechanical linkages, relays, solenoids, etc., in order to provide for fail-safe operation and safe handling of the tape under all conditions without tape breakage.

Some means of keeping the tape under constant tension between the reels must be built into the machine not only in normal and high speed operations, but under braking conditions from high speed to stop as well. This requires that torque differences for all conditions of operation be established between the take-up and pay-off reels and particularly during the stopping conditions from high speed operation. This latter problem may be solved either by dynamic braking or mechanical braking of the pay-off reel at a slightly higher rate than the take-up reel.

The mechanical braking can be achieved by making use of the differential braking of a band brake. One end of the band is fixed while the other end is moved by either a mechanical or electrical device to apply braking. A fail-safe method is to operate the brake with a spring arrangement and to release it with a solenoid. This provides safe braking of the machine should the power fail, as well as providing for normal operation. (Fig. 3) A 180° band, felt lined, ½" wide brake of 2¾ inch diameter can provide a direction differential of 4 in. oz. with 7 in. oz. and 11 in. oz. total braking torque in the two directions of rotation respectively.

Another essential control feature of a tape transport is to lift the tape clear of the heads during high speed forward or rewind operation in order to save wear on the heads and tape. This may be accomplished by either solenoid or mechanical linkage operation inter-locked with the forward and high speed control functions. There are two design possibilities; one is to lift the tape from the heads, during high speed operation; the other is to physically wrap the tape around the heads during normal operation and to allow it to fall away during high speed conditions. Either function is quite satisfactory.

When full solenoid control is designed into a tape transport, it is relatively easy to provide for remote control with remote lights indicating the operating state of the mechanism. Since a large amount of tape can be spoiled because of tape breakage during high speed operation and because a remote operator usually does not have direct view of the reels during normal operation, it is standard to install a "tape break" switch in conjunction with one of the moving compliance arms in order to bring the machine rapidly to a stop should the tape break. With a tape puller capstan the tape break switch is installed between the pay-off reel and the capstan; this insures continuity of program should the tape fail beyond the capstan.

Mechanical and electrical interlock functions must be provided to assure operational safety so that an operator will be forced to go through the "stop" condition from "high speed" conditions. This is necessary since if the tape is moving fast when the pressure roller grips the tape against the positive metering capstan, the tape is usually broken.

An important feature which is too often not considered is the desirability of building into a precision tape mechanism adequate cooling. A blower will dissipate to the surrounding atmosphere the heat of the torque motor, drive motors, and other electrical elements. Over an extended operating time, the operating temperatures of the front panel and tape handling parts may come excessively close to the nominal 105°F upper tape limit which is set to minimize the tape stickiness unless cooling provision is included.

CONCLUSION

The overall design of a tape transport for a professional tape recording system is very complex, full of many compromises and special problems, involving not only design, but the physics of machine shop practice in order to turn out a device which may be manufactured readily with uniformly good operating characteristics.

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1. Frayne and Wolfe, "Sound Recording", John Wiley and Sons, New York.
2. Olson, Harry F., "Dynamical Analogies", D. Van Nostrand Company, New York.

ELECTRON BEAM REPRODUCING HEAD FOR MAGNETIC TAPE RECORDING*

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and
J. Warren Gratian
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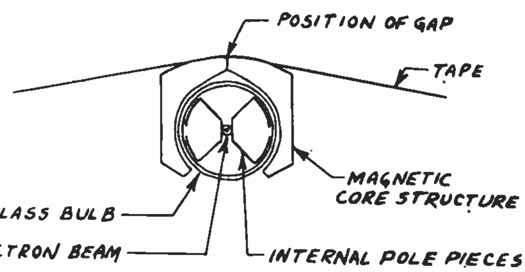
INTRODUCTION

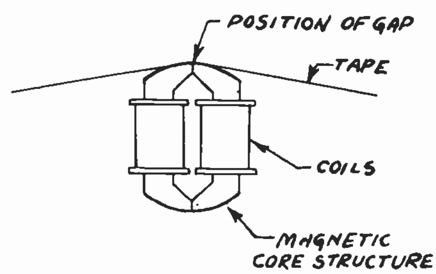
Since the time of Poulson's invention[1] of magnetic recording over 50 years ago only one principle of reproduction has been employed. This is the generation of a varying electric current in a coil by the variations of magnetic flux from the tape. The flux has usually been guided from the tape into the coils by elements of magnetic material although a simple coil has been used with the recording wire or tape passing through it.

A disadvantage of this kind of reproduction lies in the fact that the output signal is proportional to the *rate of change* of the flux in the tape rather than to the instantaneous values. This gives a frequency characteristic that increases from zero linearly until limited by the gap size.[2] And since in recording, the flux is directly proportional to the input signal the output from a conventional head is not a true reproduction of what was recorded. As is well known, equalizing networks must be incorporated in the reproducing circuits to restore the balance between low and high frequencies so that the reproduction will sound like the recorded signal.

In the new pickup head,[3] described herein, the magnetic flux in the tape is guided by a magnetic structure into a tiny cathode ray tube where it deflects the electron



(a)



(b)

Fig. 1 – (a) Section through new head.
(b) Old head.

beam in proportion to the instantaneous magnitude of the flux. Thus, unlike in present equipment, the magnitude of the output signal is independent of the recorded fre-

*This paper was presented at the Meeting of the Institute of Radio Engineers on February 7, 1953 at San Antonio, Texas. Reprinted from *Electronics*, October 1953, with permission.

[1]V. Poulson, Patent No. 661,619, November 13, 1900.
[2]S. J. Begun, "Magnetic Recording," Murray Hill Books, Inc., New York, p. 81; 1949.
[3]A. M. Skellett, Patent No. 2,165,307.

quency and of the speed of the tape. In fact if the tape is moved very slowly or even stopped altogether the amplitude of the output signal is not decreased. Furthermore, the tube acts as a deflection type of amplifier, and as a result the output voltages are many times those from conventional heads.

Fig. 1 (a) shows a sectional view of the new type of reproducing head for comparison with the old type shown in Fig. 1 (b). It contains the conventional gap in contact with the tape from which the flux is guided by the magnetic core structure through the glass walls of the tube. The internal magnetic pole pieces carry it into the deflecting region where it produces a magnetic field that deflects the electron beam.

THE TUBE

The tube not only has to provide the electron beam and associated elements but also its design must be optimized for best performance of the complete head. The design of the internal pole pieces for example is dictated as much by the relation to the external structure as by the deflection requirements of the beam.

Fig. 2 shows a cut-away view of the tube. The miniaturized electron gun at the left sends a beam of electrons between the pole pieces toward the split target. The deflection of the electrons is at right angles to the flux and hence parallel to the pole piece faces. This property of magnetic deflection is advantageous in increasing the sensitivity since the pole pieces never get in the way of the deflected beam. The pole pieces are made of thin moly permalloy because of its suitable magnetic properties. The other metal parts of the tube are non-magnetic stainless steel.

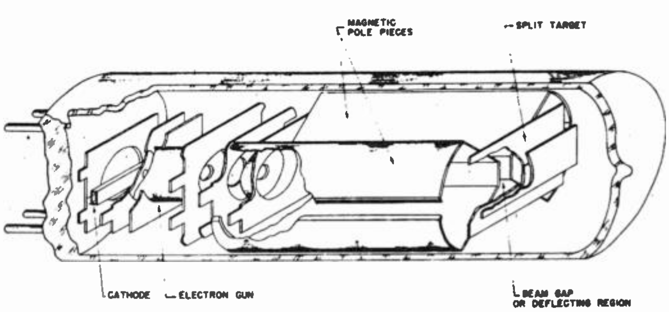


Fig. 2 – Cut-away view of tube.

Fig. 3 shows the basic circuit connections. The small plate located just behind the slit between the target plates is grounded to repel those electrons that pass through the slit. It repels them equally to the two plates and thus reduces the effective slit width. For zero flux and hence zero deflection the beam is equally split between the output plates of the target and hence the output voltage across the load resistors shown in Fig. 3 is zero. In operation the beam swings back and forth between the two plates. When the flux is negative it swings in one direction and when positive in the other. When the beam is deflected the target currents are no longer equal and hence a net output voltage is developed across the two load resistors.

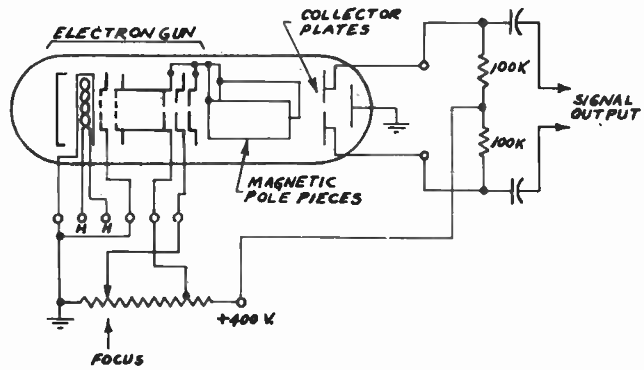


Fig. 3 – Circuit for tube.

Fig. 4 is an output characteristic. It is the output voltage measured across the output plates for different values of flux in the beam-gap. Load resistors of 100,000 ohms were used in series with each plate as shown in Fig. 3. Note that it is linear over a range of plus or minus one gauss. This range is about 50 times greater than the maximum flux variation from standard ¼-inch magnetic tape when used with the complete head under typical operating conditions.

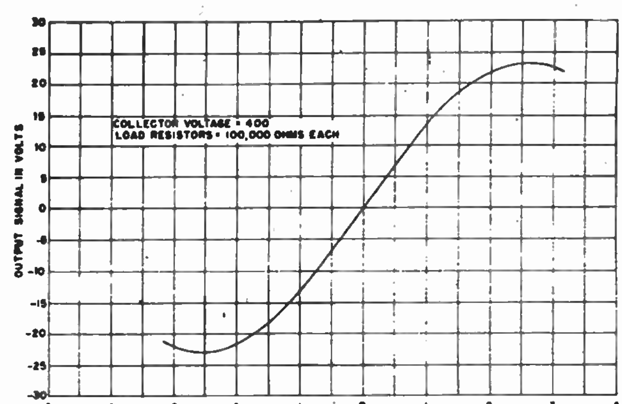


Fig. 4 – Beam-gap flux in gausses.

Fig. 5 shows the internal structure of the tube and Fig. 6 is a photograph of the finished tube in its present form. The tube is made in a T5½ miniature bulb with a standard jumbo miniature stem so that it can be used in a standard 9 pin tube socket.

THE MAGNETIC STRUCTURE

While the primary function of the magnetic structure is the guidance of the flux from the tape into the tube

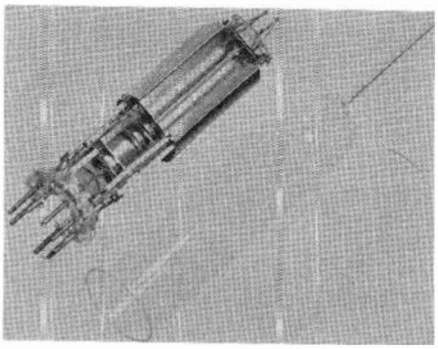


Fig. 5 – Tube structure and bulb before sealing.

many other considerations must be taken into account in order to arrive at the most efficient design. The design shown in Fig. 1 (a) while representing a practical design was abandoned early in the work. The following considerations led to the adoption of the strip-type structure shown in Fig. 7.

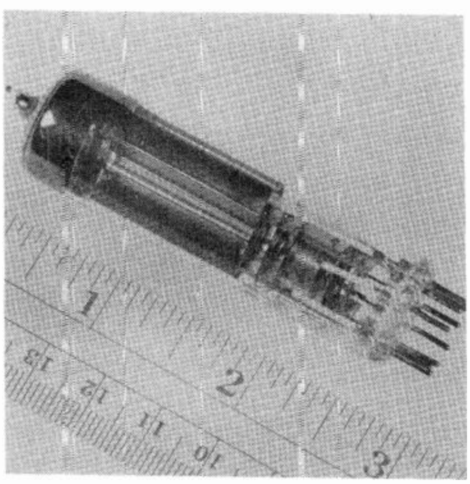


Fig. 6 – Finished tube.

Contrary to the conditions of the conventional pickup which has a very low reluctance path, the magnetic gap caused by the deflecting region in the tube introduces a very high reluctance. In consequence, the reluctance of the elements in the magnetic circuit becomes negligible in comparison with the beam gap reluctance and the magnetic elements may therefore be reduced to the smallest practical dimensions without appreciable loss of flux. A single lamination of Mu metal 0.014 inch thick proves to be adequate.

For experimental work it became easy to fabricate this strip-type structure by using hysol thermo setting casting resin which had good adhesion to the metal and which was very easy to handle. A molybdenum spacer .0003 inch thick was used at the pickup gap.

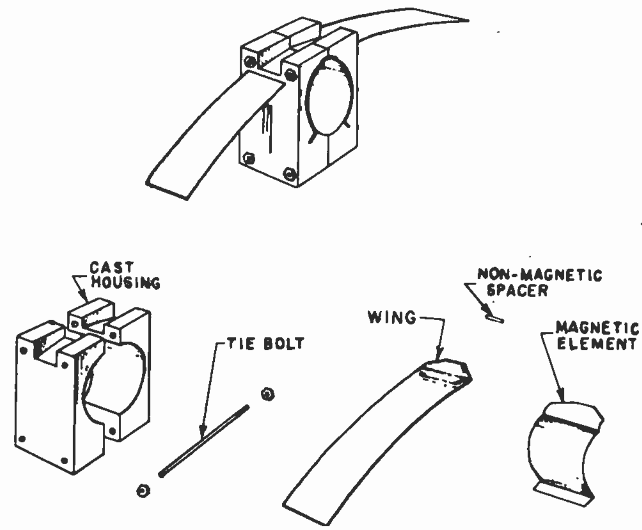


Fig. 7 – Assembly and exploded view of typical experimental winged core model.

Fig. 8 is a photograph of the completed head. The wings, which extend out in the direction of the tape and increase the effective length of the magnetic core, act to extend and flatten the low-frequency response of the pickup.

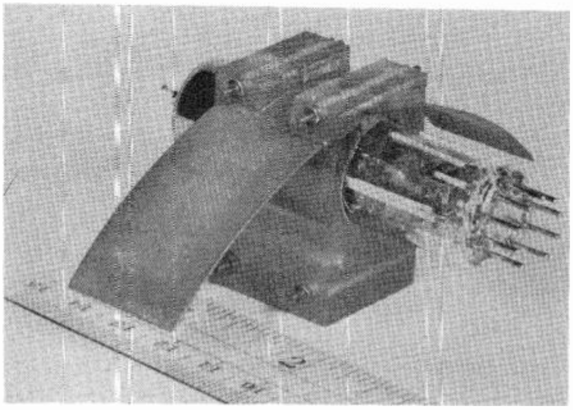


Fig. 8 – Complete head.

SENSITIVITY

No simple method was available to determine the flux density at the electron beam. A method was developed in which a calibrated vibrating probe was used to measure the flux density in the gap between a pair of dummy pole pieces mounted in proper relation to the external magnetic structure. With a saturated tape recording, it was found that a field intensity of approximately 0.04 gauss was available for deflection of the electron beam. (Reluctances and leakage factors of the various portions of the magnetic circuit were determined by additional measurements and calculations. Using these values, it was shown that the total flux in the beam gap should approximate 8% of that available from the recording. The cross-section area of the magnetic tape coating was approximately 1/1200 that of the beam gap. Thus it was shown that a saturated recording on a typical tape having a retentivity of 600 gauses should provide a beam-gap flux density of 600 x 0.08 x 1/1200 = 0.04 gauss, which is in agreement with the earlier measurements.)

Tube sensitivity was determined by measuring the tube voltage output with a known applied magneto-motive force. The beam-gap flux density was calculated from the known mmf and previously determined magnetic circuit parameters, thus permitting the tube sensitivity to be expressed in terms of tube voltage output per gauss of beam-gap flux density. With the sensitivity of the tube equal to 15 volts per gauss and a beam-gap flux density of 0.04 gauss, a maximum output voltage of 0.6 peak plate-to-plate was expected. Tests with recordings on tape confirmed these figures within experimental error.

These figures indicate the high sensitivity of the cathode ray tube. It will give satisfactory output voltages on field strengths from one-tenth to one-hundredth the strength of the earth's magnetic field.

FREQUENCY CHARACTERISTICS

Curve A of Fig. 9 gives the frequency response of the new head without any equalization. Curve B, shown for comparison, is the frequency characteristic of a conventional pickup head also without equalization. It should be noted that the scales are quite different for

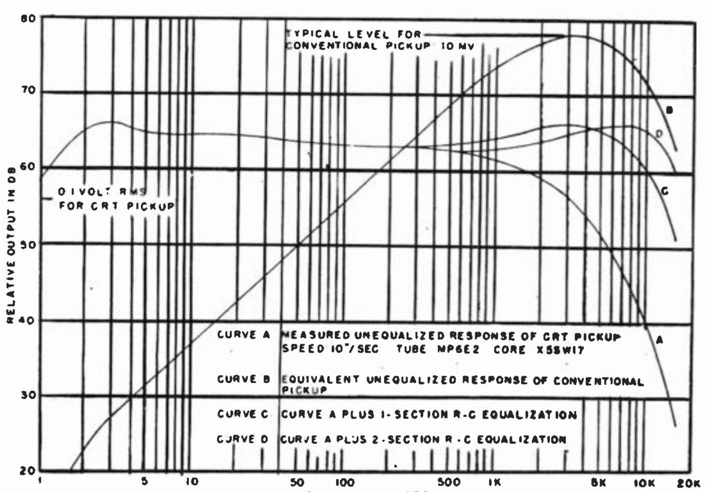


Fig. 9 – Comparison of crt and conventional pickups.

Curves A and B. For example, the maximum output obtained by the conventional head is only 10 millivolts whereas the level of most of the frequency range of the new head is about two-tenths of a volt. These curves were taken with conventional longitudinal recordings and demonstrate the superior low frequency performance of the new head.

At the upper frequency end of the curves the deterioration in output is caused by the so-called "gap effect" which comes about because the gap itself is comparable in length with the wavelength of the recorded pattern. The curves show that this effect is more serious with the new type of head. The reason is that the steeply ascending curve of the conventional head partly compensates for this gap effect whereas the flat characteristic of the new head has no compensating feature.

Calculation indicates that a very simple, single section R-C equalizing network used in conjunction with the new head will modify the characteristic to Curve C. A two section R-C filter will give an even better effect as shown by Curve D.

PATTERN OF MAGNETIZATION OF THE TAPE

All of the discussion given above was based upon conventional longitudinal recording on the tape. Yet for very low frequencies, i.e., for long wavelengths in the tape this is not the optimum type of recording pattern. For the long wavelength the gap has access only to the leakage flux near the center of the elementary magnet and therefore as the head is moved slowly over a long recording, of a square wave for example, the response will be a maximum at the ends and a minimum at the center of the square pulse.

Either perpendicular or transverse magnetization would be more suitable, but both of these suffer at the high frequency end. The work to date has been restricted pretty well to longitudinal recording since the over all response was more important than the very low frequency end of the spectrum.

Work is being carried on, however, on perpendicular recording, i.e., magnetization through the tape and indications are that this may be made to give much superior results at the low frequency end without too great a loss at the high frequency end of the spectrum.

APPLICATIONS

This new type of head offers advantages over conventional heads in a number of uses. In spite of the excellent results obtained with well equalized conventional recording of music, it is believed that ultimately the quality obtained with the new head will surpass that possible with conventional types.

There are certain commercial and military applications where it is desirable to record very low frequencies, dc levels or pulses without distortion and for these the new head is ideally suited whereas the old type will not perform adequately without considerable complexity of the apparatus, e.g., the dithering head, frequency modulation for dc recording, etc.

The inherent amplification of the tube and the cheaper equalization circuits required by the new head gives rise to the hope that a simpler, cheaper magnetic tape recorder may be possible. This, of course, because of the economic advantages, would further popularize and widen the field of magnetic tape recording.

ACKNOWLEDGMENT

This development has been sponsored by the Bureau of Ships of the United States Navy under Contract NObsr-57452. The tube has been under development at National Union Radio Corporation and the magnetic head structure at Stromberg-Carlson Company.

INVESTIGATION OF CORE STRUCTURES FOR THE ELECTRON-BEAM REPRODUCING HEAD IN MAGNETIC RECORDING*

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SUMMARY – The relative advantages of perpendicular versus longitudinal recording have been re-evaluated in terms of the electron-beam pickup. It is shown that dc response can be obtained only with perpendicular or transverse recording, but because of other considerations longitudinal recording appears to be more generally practicable, even though the resulting low-frequency response limit depends upon the extent to which the physical dimensions of the core may be increased and the speed of the recording medium may be reduced. Measurement techniques have been devised to permit study of the magnetic circuit parameters which influence over-all pickup sensitivity; associated studies show the advantages of a core structure having unconventional configuration and physical dimensions. Procedures recently presented by other writers for the investigation of factors contributing to high-frequency loss in conventional heads have been tested with the electron-beam pickup; tape-coating thickness appears to be responsible for a decibel loss approximately equal to the total contributed by all other factors. Relatively simple strip-type core structures for use with the electron-beam tube have been developed – over-all performance data under typical recording conditions show an output of the order of 0.2 volt and an equalized response range of approximately one to 15,000 cycles per second, plus or minus three decibels, at a tape speed of 10 inches per second.

INTRODUCTION

The electron-beam magnetic reproducing head described in a recent paper[1] represents a radical departure from the conventional pickup. Its potential outstanding characteristics are wide-range frequency response and high sensitivity. Realization of wide-range response and high sensitivity is dependent to a large extent upon the configuration of that portion of the core structure which is external to the pickup tube itself. Objectives in the investigation of this core structure have been:

(1) To obtain as high an upper limit of frequency response as is practicable at a given tape speed.

(2) To obtain as low a lower limit of frequency response as is practicable without impairing the upper limit and without resorting to very large core structures.

(3) To obtain as smooth a response as is practicable between the lower and upper limits.

(4) To obtain as high sensitivity as possible without serious sacrifice in high-frequency response.

The purpose of this paper is to present a comprehensive discussion of those factors which affect the performance of the core structure in combination with the electron-beam tube. Experimental data concerning the separation of losses in response at short wavelengths will be presented as well as data showing the effects of shielding and variation in the parameters of the magnetic circuit of the pickup. Magnetic-recording phenomena associated with signals of long recorded wavelengths and the relative advantages of perpendicular versus longitudinal recording will be discussed. Curves will be presented to illustrate the extent to which the objectives have been met in a practical design.

INITIAL CONSIDERATIONS

A typical circuit for the electron-beam tube is shown in Fig. 1. In operation, a beam of electrons is

*Manuscript received December 3, 1953. An abridged version of this paper was delivered at the Department of Defense Symposium on Magnetic Recording, October 12-13, 1953, Washington, D.C.

[1]Skellet, A. M., Loveridge, L. E., and Gratian, J. W., "Electron beam reproducing head for magnetic tape recording," presented at IRE Meeting, San Antonio, Texas, Feb. 7, 1953; published in *Electronics*, vol. 26, No. 10, pp. 168-171; Oct. 1953; and elsewhere in this issue.

projected from a miniaturized electron gun through the gap between a pair of magnetic pole pieces to a pair of collector plates. The electron beam is deflected toward one or the other of the two collector plates depending upon the direction of the flux through the beam gap. A push-pull output voltage is thus provided across the two load resistors.

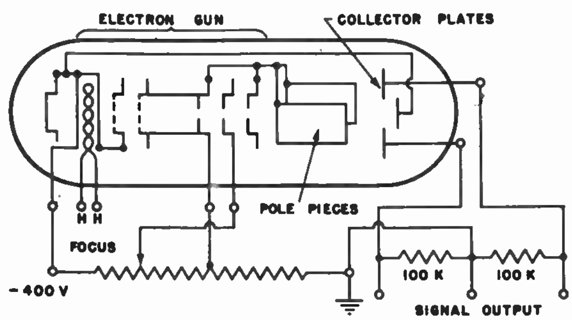


Fig. 1 – Electrical circuit for electron-beam tube.

When used in a reproducing head for magnetic recording the tube is fitted with an external core structure which may be of the form indicated in Fig. 2. The complete magnetic circuit in this case is similar to that of a ring-type head. However, practical design requires that special consideration be given to the high reluctance of the beam gap and the space occupied by the tube envelope, which is between the external core and the tube pole pieces. Assuming the recorded tape to act substantially as a source of constant flux, it is seen that the available signal flux after entering the pickup core divides between two primary paths. The portion of the flux which is conducted to the beam gap within the tube is used to deflect the electron beam; the portion of the flux which is shunted through the front gap of the core is wasted. An extremely short front-gap length is necessary since the high-frequency response of the pickup is limited by the ability of the front gap to resolve signals having a wave-length of a fraction of a mil. However, to provide maximum over-all pickup sensitivity, the reluctance of the front gap must be large compared with the reluctance through the useful flux path which includes the beam gap.

The electron-beam tube provides a voltage output which is proportional to the flux through the tube, in contrast to the conventional pickup for which the voltage output is proportional to the time rate of change in flux through the core. Consequently, if the flux induced in the electron-beam pickup were independent of the wave-length of recorded signals, the frequency response would be flat to zero cycles per second. As is generally known, however, the response of conventional longitudinal-type pick-ups falls at a rate exceeding six decibels per octave as the recorded wavelength becomes increasingly larger than the physical dimensions of the pickup. Previous work by other writers[2,3] has shown theoretically that the response should approach a slope of 18 db per octave for signals of very long recorded wavelength. Since no published experimental verification was known, the curve of Fig. 3 was obtained through the use of high tape

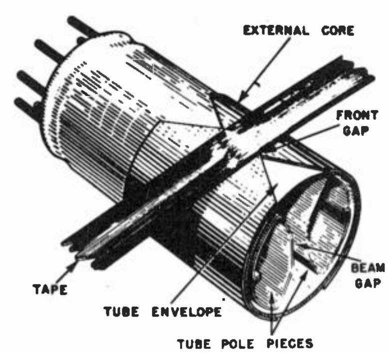


Fig. 2 – Cut-away view showing magnetic elements.

speed, an amplifier having a very low noise level and a ring-type reproducing head having an over-all length of approximately ⅜ inch. Fig. 3 confirms the 18 db per octave slope for a conventional longitudinal-type pickup;

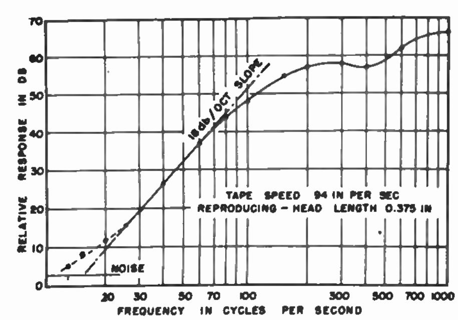


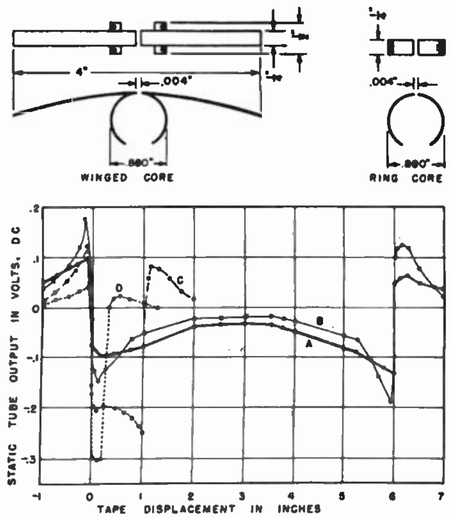
Fig. 3 – Low-frequency response for a conventional reproducing head.

[2] Otto Kornei, "Frequency response of magnetic recording." *Electronics*, vol. 20; August, 1947.

[3] Donald L. Clark and Lynn L. Merrill, "Field measurements on magnetic recording heads," *Proc. I.R.E.*, vol. 35, no. 12; December, 1947. (In subsequent unpublished work the authors found that (8) of the published paper could be expanded in series form to predict that response should vary at a rate of 18 db per octave when the recorded wavelength is much greater than that of the over-all core length.)

the electron-beam tube in combination with a longitudinal-type core structure should therefore produce a response which reaches a falling rate of 12 db per octave at recorded wavelengths which are much greater than the physical length of the core.

In contrast to the characteristics of the longitudinal pickup as outlined above, the flux induced in a perpendicular-type pickup is independent of recorded wavelength for recorded signals of very long wavelength.



CURVE	CORE	RECORDED PULSE LENGTH
A	WINGED	6"
B	RING	6"
C	RING	1"
D	RING	.250"

Fig. 4 – Pulse response of longitudinal type core.

For comparison there is shown in Fig. 4 the pulse response of two longitudinal pickups and in Fig. 5 the pulse response of a perpendicular pickup. In all three cases, a unipolarity rectangular pulse was recorded and the reproduced output was read, point-by-point, on a dc meter as the recorded tape was drawn slowly over a pickup comprised of the indicated core and an electron-beam tube. Curves B, C, and D of Fig. 4 show the response for pulses of three different lengths when reproduced with the aid of a ring core having an outside diameter of approximately 0.88 inch. Curve A shows the improvement in response which results from the use of a winged core which has an over-all length of four inches. The superior long-wavelength response of a perpendicular core, as shown in Fig. 5, is evident.

Although the perpendicular core in combination with the electron-beam tube is seen to offer possibilities which deserve further study, especially for applications in which dc response is required, further development of the longitudinal core was undertaken first for the following reasons:

(a) Existing equipment can be adapted to accommodate the longitudinal electron-beam pickup with no fundamental change in the recording portion of the equipment.

(b) The effective gap of the longitudinal pickup can be made several times shorter than that of the perpendicular pickup, thus providing maximum high-frequency response at minimum tape speed.

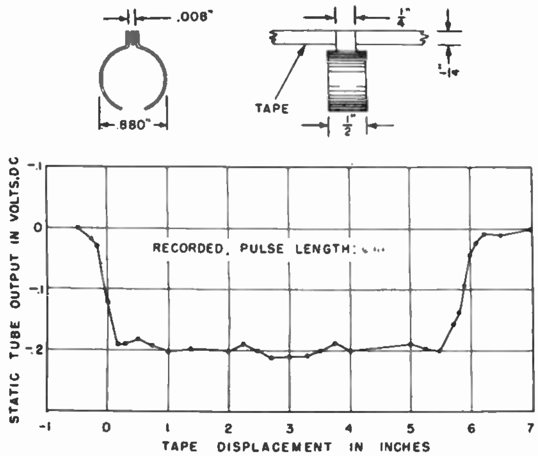


Fig. 5 – Pulse response of perpendicular type core.

(c) The longitudinal core places a limit only on maximum recorded wavelength, not on minimum frequency. The inherent ability of the tube to respond to static fields is not impaired.

(d) A low-frequency response limit of the order of one cycle per second can be provided using conventional tape speeds and practicable pickup dimensions.

(e) The low-frequency response limit can be lowered as far as desired with no loss in output by reducing tape speed.

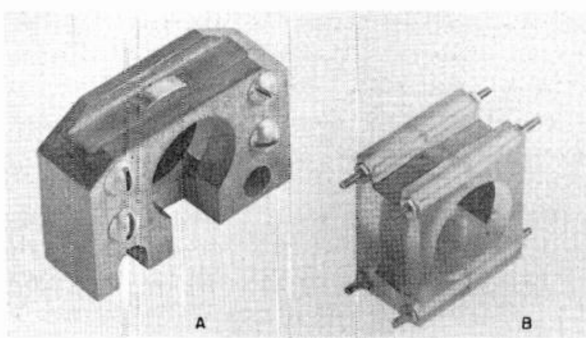


Fig 6 – Experimental ring-type core models.

EXPERIMENTAL RING CORES

The photograph of Fig. 6 shows two core models which were particularly useful in initial studies. The magnetic core of model A consists of two C-shaped half sections built up of eight-mil thick laminations to an over-all core width of 0.128 inch. The pole faces are undercut as in conventional pickups to provide a front-gap pole-face height of approximately 30 mils. In contrast, each half section of the magnetic core of model B is composed of a pair of eight-mil thick by ¼ inch wide strip-type laminations concentric with the axis of the tube to provide a total strip thickness of 16 mils. The inner laminations are set back 0.015 inch from the edges of the front gap, leaving an effective front gap pole-face height of eight mils as determined by the thickness of the outer lamination alone. The magnetic core is cast in a thermo-setting resin to complete the structure.

Fig. 7 shows response curves as a function of wavelength for core model A in combination with a tube model having ½ inch long pole pieces. Curves A, B and C for front-gap lengths of 4.8, 0.72 and 0.24 mils, respectively, show the manner in which sensitivity decreases and relative high-frequency response improves as gap length is reduced.

The cause of the hump in the response at two inches recorded wavelength is believed to be associated with the fact that the length of the tube pole pieces is appreciably greater than the external core width. Recorded signals having a wavelength which is several times that of the spacing between the recording medium and the pole pieces act to induce an appreciable component of flux in the tube pole pieces with no assistance from the external core of the pickup. (Experimental evidence confirming this is shown in Curve C of Fig. 8.) The external field for signals of shorter recorded wavelength, however, is concentrated nearer the tape and must be conducted to the tube pole pieces by the external core of the pickup. In effect, the front gap is more closely coupled to the flux source for signals of short recorded wavelength than for signals of longer wavelength. Increasing the shunt effect of the front gap therefore tends to reduce pickup sensitivity more for signals of short and moderate wavelengths than for signals of long wavelength.

For Curve D of Fig. 7, conditions were identical to those for Curve C except that eight-mil thick Mumetal shims having a width several times that of the core were inserted in the clearance space between the tube and the external core. The increase in sensitivity at short wavelengths is due to the decrease in core-to-pole-piece reluctance which permits a larger percentage of the available input flux to be conducted to the beam gap. The small change in sensitivity at long recorded wavelengths results from the tendency of the shims to shield the pole pieces from the direct influence of signal fields of long recorded wavelength. With the addition of the shims which become an effective part of the external core, the total flux induced in the core by long wavelength signals may be greater than that induced without the shims. However, all components of flux which enter the core, regardless of their point of entry as determined by the signal wavelength, are then attenuated approximately equally by the effects of the shunt front-gap reluctance and series core-to-pole-piece reluctance before reaching the beam gap; hence, a flatter response curve results.

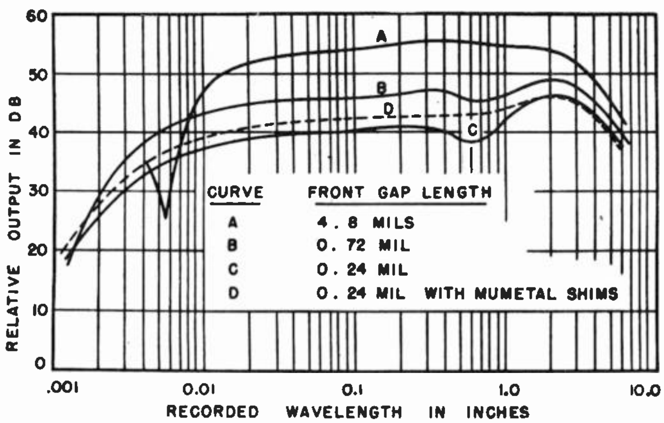
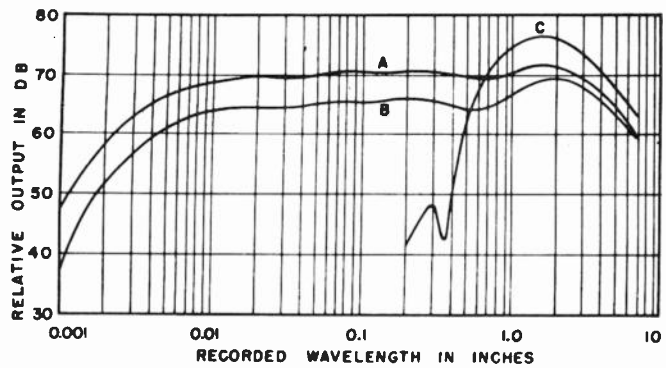


Fig. 7 – Wavelength response curves for core model A.



CURVE	CORE	GAP LENGTH	POLE THICKNESS
A	B	0.39 MIL	8.0 MILS
B	C	0.44 MIL	14.0 MILS
C	NONE	———	———

Fig. 8 – Wavelength response curves for electron-beam tube with two different cores and for tube alone.

Considerations which made a strip-type core as represented by model B seem especially well suited for use with the electron-beam tube are the following:

(a) The necessity of providing an aperture for passage of the electron beam in the tube precludes the provision of a very low-reluctance path for useful flux as is possible in the conventional pickup. Since thin strips of high-permeability alloy provide reluctances which are negligible in comparison with the beam-gap reluctance, a heavier cross-section is of no advantage.

(b) The strip may be made sufficiently thin to restrict eddy-current loss to a tolerable value.

(c) With the availability of casting resins which have good adhesion to metal, it is possible to provide satisfactory support for the critical edges of the front gap.

(d) Cores of unconventional size and configuration may be more readily produced in strip form.

Curves A and B of Fig. 8 show the response of two strip-type cores with a given tube model. Core model C is similar in construction to the previously described model B except that the core is formed of non-laminated strip having a thickness of 14 mils and the measured gap length is 0.44 mil versus 0.39 mil for B. Curve A for model B shows higher sensitivity and more uniform response. The higher sensitivity is due primarily to a smaller front-gap pole face area. The improved high-frequency response is due partially to a shorter front gap and partially to reduced eddy-current loss. The flatter low-frequency response is associated with the more favorable ratio of front-gap reluctance to beam-gap reluctance plus core-to-pole-piece reluctance.

Curve C of Fig. 8 shows the response of the electron-beam tube with no external core. As may be seen by comparing Curves A, B and C, the tube pole pieces, which are one inch long for this tube model, determine the shape of the over-all response curve at long wavelengths. The primary consideration which led to the use of relatively long-pole pieces is the following: If a given input flux is assumed and if the beam-gap reluctance is assumed to be shunted by a much smaller front-gap reluctance, the magnitude of the beam-gap flux will be proportional to pole-piece length and the beam-gap flux density will be independent of pole-piece length. With a given beam-gap flux density, the tube output is approximately proportional to pole-piece length.

Further advantages are obtained with the use of the long tube pole pieces in combination with an external core of corresponding width. First, the reduced reluctance between the tube pole pieces and the external core results in increased pickup sensitivity. Second, as shown later, increased core width tends to flatten and extend the low-frequency response of the pickup.

EXPERIMENTAL WINGED CORES

Prior to the work described herein, a winged-core structure for the purpose of extending the low-frequency response of conventional pickups had been successfully developed at Stromberg-Carlson. In the work to be described, means of adapting the winged core to the special requirements of the electron-beam pickup were investigated and an extended study of factors which influence low-frequency response was carried as far as time permitted. In addition to generally recognized factors such as over-all core length, core contour and shield dimensions, core width was found to be important.

Experimental core models of the form shown in Fig. 9 were used in tests concerning the effect of core dimensions. The construction of these models is similar to that previously described for model B except for the increased width and thickness of the magnetic elements and the fact that the outer lamination is formed with a larger radius to provide wings extending in the direction of the tape travel. The C-shaped elements which contact the tube are one inch wide and equal to the corresponding dimension of present tube pole pieces. The thickness of the formed magnetic elements is 14 mils. The effective width of the experimental cores was varied by means of auxiliary Mumetal plates fastened to the original wings as shown in the photograph of Fig. 9.

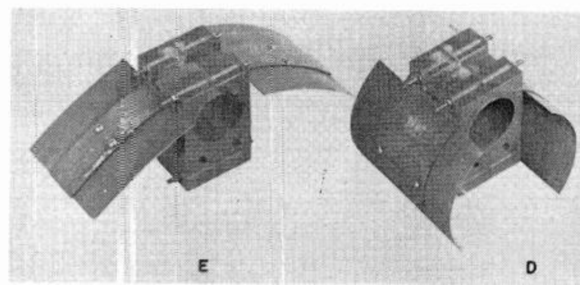


Fig. 9 – Experimental winged core models.

The curves of Fig. 10 show the manner in which long-wavelength response varies with changes in wing radius. For these data the width of the wings was ¼ inch and the over-all length of the wings before bending was four inches. The pickup was centered in a shield box having major dimensions of 1½ inches by 2¾ inches by 10 inches. Under these conditions a radius of approximately three inches was found to provide the best compromise between maximum response at seven inches recorded wavelength and smoothest response between 0.7 and 7 inches recorded wavelength.

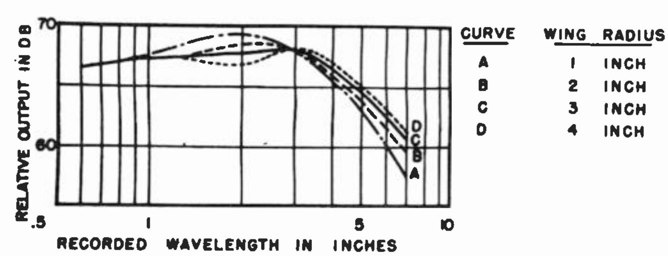


Fig. 10 – Effect of wing radius on long-wavelength response of shielded, wing core model.

Considerable study was given to the fact that means could not be found whereby long-wavelength response could be extended in direct proportion to the over-all length of the core structure. This is an important consideration because the physical length of the winged cores tends to limit their application. On the basis of response as shown in Fig. 8 for a close-fitting core having an outside diameter of approximately ¾ inch, it seems that it should be possible to provide response which is down no more than six decibels at a recorded wavelength of approximately 12 inches when using a core structure having an over-all length of two inches.

The closest approach to that condition was obtained with core model D. With extensions as shown in Fig. 9, the wings have an effective width of two inches and an outside diameter of two inches throughout an arc of approximately 270 degrees. Curve A of Fig. 11 shows that the response of this core, unshielded, is down six decibels at approximately eight inches recorded wavelength. From these and other data it was concluded that increased arc length and wing width provide smoother extended low-frequency response.

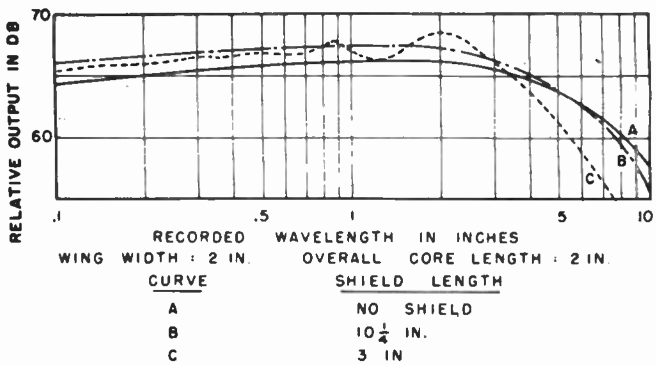


Fig. 11 – Wavelength response curves for core model D showing effects of variation in shield dimensions.

Curve A of Fig. 12 shows the long-wavelength response of core model E which has an over-all length of 3⅞ inches as compared with two inches for model D discussed above. As may be seen by comparing Curve A of Fig. 12 with Curve A of Fig. 11, the long-wavelength response of model E extends further, but is less smooth than that of model D.

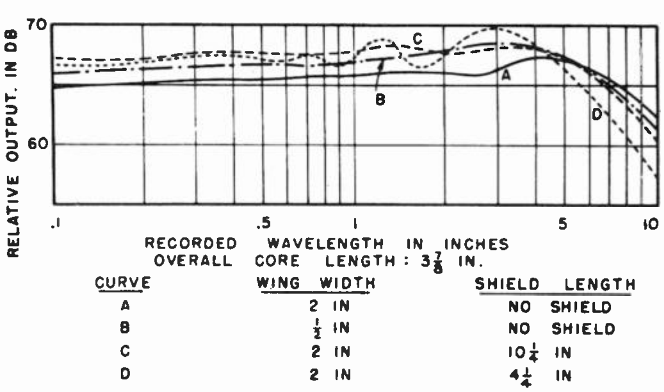


Fig. 12 – Wavelength response curves for core model E showing effects of variation in shield dimensions.

Comparison of Curves A and B of Fig. 12 show clearly the improvement in long-wavelength response which results from an increase in the wind width of core model E. The remaining curves of Figs. 11 and 12 with tabulated data show in detail the manner in which the long-wavelength response of core models D and E varies with changes in shield dimensions.

EDDY CURRENT LOSS

Fig. 13 shows the loss in high-frequency response caused by eddy currents for three of the experimental core models discussed above. Curve C for the winged core which is formed of 14-mil thick Mumetal strip shows a high-frequency loss of three decibels at 10 kc. Curve A for the core having a stack of conventional laminations and Curve B for the core having a pair of concentric laminations show less loss. Although the loss for the strip-type cores is appreciable, it represents only a small portion of the total attenuation at the upper frequency limit when conventional tape speeds are used. In view of the simplicity of the strip core, it seems advisable to consider the use of more complex laminated structures only for special applications requiring higher upper-frequency limits.

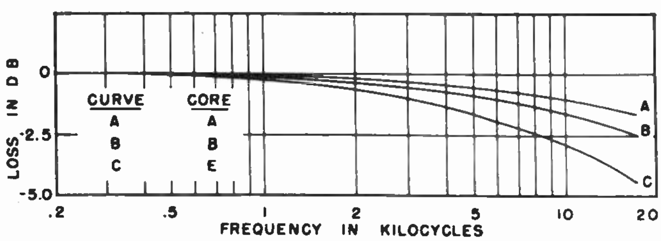


Fig. 13 – Response loss due to eddy currents.

OVER-ALL RESPONSE OF WINGED CORE MODEL OF FINAL DESIGN

The final winged core model is identical to the experimental model E, shown in Fig. 9, except for the substitution of a solid wing having a width of one inch. The complete electron-beam pickup and the rear half of the shield box as mounted on a conventional tape loop drive mechanism are shown in Fig. 14. The dial which shows in the photograph behind the shield permits tape speed to be adjusted over a range of 10:1. A set of conventional erase, record and playback heads is located at the lower center of the front panel. Circuits and controls for mixing recording bias and signal are located at the left end of the drive unit.

The unequalized frequency-response curve for one of the final core models using a tape speed of 10 inches per second and constant-current recording is shown in Curve A of Fig. 15. The response, as shown for a pickup enclosed in a shield having dimensions of 2⅞ inches by

2¾ inches by 10 inches, is flat within plus or minus three decibels from 1.2 to 1600 cycles per second. The lower response limit is extended to approximately 0.9 cps when no shield is used. The general effect of shield dimensions for a similar core model is shown in Fig. 12.

Curves A and C of Fig. 15 show that high-frequency equalization of approximately 20 db at 10 kc is required to obtain response flat within plus or minus three decibels from 1.2 to 10,000 cps using the cathode-ray-tube pickup. With this amount of high-frequency

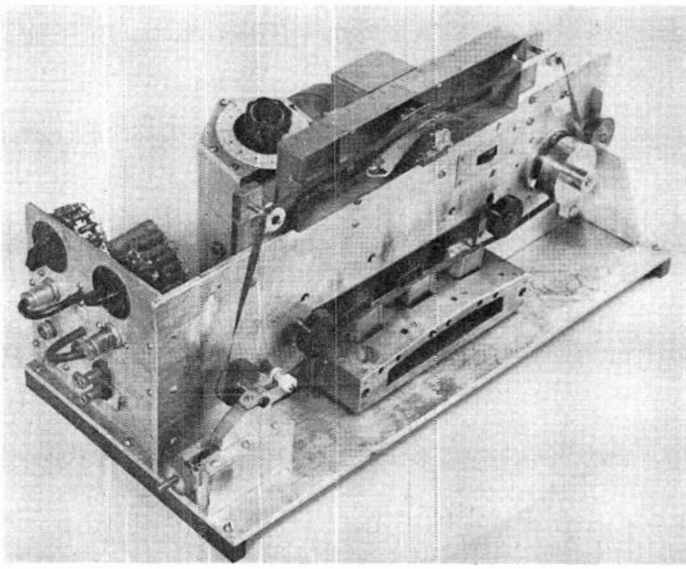


Fig. 14 – Tape drive unit with electron-beam tube pickup.

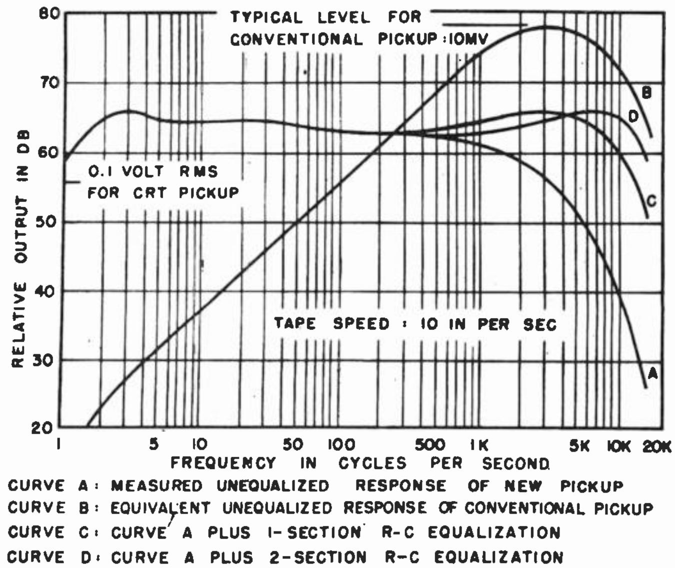


Fig. 15 – Comparison of electron-beam and conventional pickups.

Curve B of Fig. 15 shows the effect of superimposing the cathode-ray-tube pickup response on a frequency characteristic which rises at six decibels per octave, as would be obtained with a conventional pickup of comparable dimensions for which the output voltage is proportional to the time derivitive of flux. It should be noted that the differentiation of flux in the conventional pickup produces, in effect, equalization which greatly improves the apparent high-frequency response of the pickup. With the addition of equivalent high-frequency equalization it will be seen that the high-frequency performance of the cathode-ray-tube pickup is comparable with that of high-quality conventional pickups.

With the addition of a single-section R-C high-frequency equalizer, the calculated equalized response is flat within plus or minus three decibels from 1.2 to 10,000 cycles per second, as shown in Curve C of Fig. 15. With the addition of a dual-section R-C high-frequency equalizer, the calculated equalized response is flat within plus or minus three decibels from 1.2 to 15,000 cps, as shown in Curve D. In both cases the lower response limit is extended to approximately 0.9 cps when no shield is used. The equalized response curves were calculated assuming the use of high-frequency equalizer sections of conventional form with the ratio of resistances sufficiently great to allow the equalizer response to rise at six decibels per octave to the highest frequency shown.

equalization, the signal-to-noise ratio exceeds 40 db. By contrast, from Curve B it will be seen that approximately 60 db of low-frequency equalization would be required to obtain the same range of response with the conventional pickup. At one cycle per second the output of a conventional high-impedance pickup is of the order of 10 microvolts when a high recording level is used. The required equalization cannot be provided without excessive loss in signal-to-noise ratio.

A general comparison for the over-all frequency characteristics of the electron-beam pickup with winged core and that of the conventional pickup may be summarized in the following manner: With no external equalization, the response of the electron-beam pickup is flat within plus or minus three decibels over a recorded wavelength range having high and low limits in a ratio exceeding 1500:1. The corresponding range for a conventional pickup is less then 15:1. With typical equalization the range of the electron-beam pickup is approximately 15,000:1. The typical equalized range for high-quality commercial equipment using a conventional pickup is 300:1.

CAUSES OF HIGH-FREQUENCY LOSS

Fig. 16 shows a breakdown of high-frequency losses as determined for one of the final core models. The data are plotted versus recorded wavelength since three of the four factors considered are a function of recorded wavelength. The procedure used is essentially that presented by R. L. Wallace[4] in a recent paper. The heavy curve shows the-over-all measured response. Circled points are calculated or measured values representing the individual losses. The level of zero loss is assumed to be that of the measured response at 0.1-inch recorded wavelength, at which point the effect of low-frequency irregularities becomes negligible.

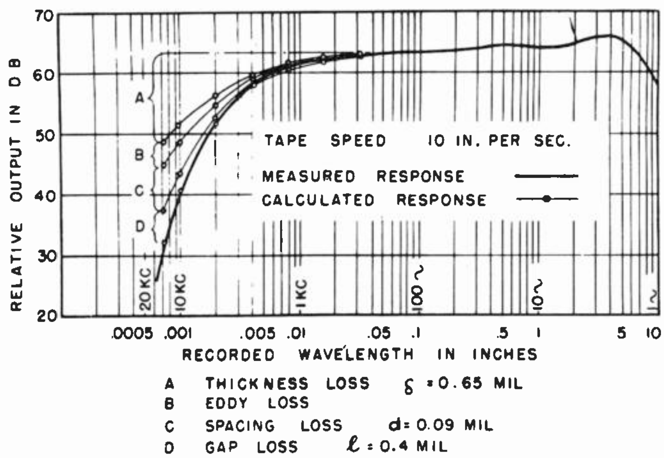


Fig. 16 – Comparison of experimental and calculated response of final pickup model.

The loss associated with the coating thickness of the recording medium was calculated from the following expression as given by Wallace:

$$\text{Thickness loss} = 20 \log_{10} \frac{2\pi\delta/\lambda}{1-e^{-2\pi\delta/\lambda}}$$

where δ = coating thickness

λ = recorded wavelength.

The loss in response attributed to eddy currents actually includes all losses which are a function of frequency only. These losses are believed to be due primarily to eddy currents. To determine this loss, a small coil is wrapped closely around the core of the pickup at the gap; the coil is then driven with constant current throughout the frequency range of interest and the voltage output of the electron-beam tube is measured.

Spacing loss is attributed to a lack of perfect contact between the recording medium and the core of the pickup. The loss was calculated from the following formula as given by Wallace:

$$\text{Spacing loss} = 54.6\, d/\lambda \text{ db}$$

where d = effective spacing between the core and the recording medium

λ = recorded wavelength.

Since it is impossible to measure the spacing, Wallace's procedure was to attribute to spacing the loss which remained after subtracting all known losses as determined by other means. Since the characteristic shapes of the various loss functions differ appreciably, the fact that the calculated points fall close to the measured over-all response is considered good evidence of the validity of the procedure.

It is interesting to note that the minimum effective spacing achieved by Wallace in his work was 0.23 mil as compared with the 0.09 mil determined in this work. The flexible recording medium used in this work would be expected to allow better contact than could be achieved with a plated drum such as was used by Wallace. In this connection it was found that relative high-frequency response varies appreciably with tape wear. In order to obtain reproducible results, measurements were made using slightly burnished tape loops which were found to provide reasonably consistent results for an appreciable period of time. Gap loss was determined from the well-known expression:

$$\text{Gap loss} = 20 \log_{10} \frac{\pi l/\lambda}{\sin \pi l/\lambda}$$

where l = effective gap length,

λ = recorded wavelength.

The nominal thickness of the gap spacer was 0.30 mil and the physical gap length as measured under a microscope was approximately 0.40 mil. The effective gap length,[5] which is usually assumed to be 50 to 100 per cent greater than the physical length, could not be determined by inspection of response data at higher frequencies. Therefore, in arriving at the values shown in Fig. 16, both the effective gap length and effective spacing were treated as unknowns. Several values of each unknown were tested in the respective loss expressions to determine what values were required to give a total loss best matching that shown by the measured response. Combinations other than those shown were found to result in an appreciable departure from the measured response.

In evaluating the results shown in Fig. 16 the following points are of particular interest:

R. L. Wallace, Jr., "The reproduction of magnetically recorded signals," The Bell Sys. Tech. Jour., vol. XXX, no. 4, Part II; October, 1951.

[5] S. J. Begun, "Magnetic recording," Murray Hill Books, Inc.,; p. 85.

(a) Thickness loss appears to be the most important single factor contributing to the total high-frequency attenuation. Two different formulas[4,6] have been recommended for the computation of thickness loss. Measurements were made on available tape samples having different coating thicknesses in an effort to determine whether either of the proposed formulas could be verified experimentally but results were not conclusive. Wallace's formula, which gives considerably lower loss, was used in calculating thickness loss for Fig. 16.

(b) Recording demagnetization[6] was assumed to be negligible for the recording conditions of these measurements. Recording bias current was determined by a somewhat unconventional procedure. Since signal-to-noise specifications for the pickup were stated in terms of that signal level which produces three per cent total harmonic distortion, curves of output voltage and recording current versus bias current for constant three per cent total harmonic distortion with a 400 cps signal were first determined. Curves of output versus bias at higher signal frequencies were then obtained using the recording-current values which were determined at 400 cps. From these curves the bias which provided best over-all performance was found to be approximately two milliamperes as compared with 3.4 ma when determined by conventional procedures, and it corresponds approximately to that bias which gives maximum high-frequency output. Specific changes in the performance characteristics and operating conditions which resulted from the change from 3.4 to 3.0 ma bias are as follows: 400 cps distortion, unchanged at three per cent; relative response at one mil recorded wavelength, +4.2 db; 400 cps output, -0.9 db; recording-bias and recording-signal current, -4.6 db and -3.8 db, respectively.

(c) The assumed level of zero loss is somewhat arbitrary. The general slope of the low-frequency response varies with shield dimensions, core length, core width and front-gap length. For example, a change in wing width from one-half inch to two inches was found to produce a drop in response which varies progressively from zero decibels at one mil recorded wavelength to approximately 2½ db at the wavelength of maximum response. Hence, an increase in wing width appears to reduce the high-frequency attenuation relative to the assumed level of zero loss.

(d) Further study of the procedure for evaluating the specific sources of high-frequency attenuation appears worthwhile.* The data originally presented by Wallace, and the supplementary data presented herein for a pickup of unconventional size and general design both show a good correlation between total measured loss and the sum of the deduced individual losses. However, in view of questions which have been raised during the later tests it seems quite likely that further work might show an appreciably different distribution of losses. If the data as presented are correct, a recording medium having a thinner magnetic coating could be used to very good advantage with the electron-beam pickup.

DETERMINATION OF BEAM-GAP FLUX DENSITY AND TUBE SENSITIVITY

In the early stages of the tube development an estimate of beam-gap flux density under typical recording conditions was required. The equipment which was used in the experimental verification of initial estimates of beam-gap flux density is shown in Fig. 17. As shown, a

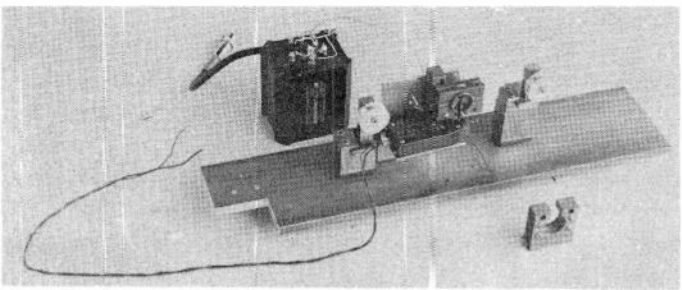


Fig. 17 – Apparatus used in measurement of beam-gap flux density.

[6]O. William Muckenhirn, "Recording demagnetization in magnetic tape recording," *Proc. I.R.E.*, vol. 39, no. 8; August 1951.

*Comparison of these results with those published by E. D. Daniel, since the preparation of this paper, are of particular interest. (*The Proc. Inst. Elec. Eng.*, vol. 100, Part III, no. 65, pp. 168-175; May, 1953.) Fig. 10 of that paper shows a recording loss, at 1.5 mils recorded wavelength, of 13 db which approximately equals the corresponding loss, exclusive of reproducing gap and eddy-current loss, as shown in Fig. 16 of this paper. However, Daniel attributes only six decibels of this total loss to effects associated with the thickness of the magnetic medium and the remainder to other effects such as self-demagnetization. Daniel's analysis which takes into account non-uniform magnetization through the medium seems more sound fundamentally than one which assumes uniform magnetization. On the other hand, the following assumptions which appear to be implied in Daniel's work may cause appreciable error and seem worthy of further study: a) The "effective susceptibility versus bias-field-strength curve" for a signal of short wavelength represents the performance of an infinitesimal surface layer rather than the average performance of a layer of substantial thickness. b) The shape of the susceptibility curve for a surface layer is substantially identical to that for a layer located deeper within the magnetic tape coating. c) The magnitude of the susceptibility curve at any given depth within the medium depends only on the magnitude of the field intensity in the plane through the center of the recording gap; it is therefore independent of the shape of the recording field at the trailing edge of the gap[3], i.e., the associated variations in recording demagnetization[7] at different signal wavelengths and at different depths within the tape coating are negligible.

pair of tube pole pieces are mounted in their proper relative positions in a phenolic dummy structure simulating the tube. An experimental core model is mounted in its normal position with respect to the tube pole pieces. In use, the core structure is excited by drawing prerecorded tape over the idlers and core. The resulting beam-gap flux density is measured by means of a small rectangular pickup coil located in a probe which is vibrated by a crystal cutting head. The pickup, which contains no magnetic material, was initially calibrated in a known field of relatively high intensity. The usable sensitivity of this equipment when limited by a pickup coil of given size, was found to be approximately 5000 times that of a light-beam type fluxmeter having a constant of 400 flux linkages per millimeter. Measurements on the earliest core models showed that beam-gap flux density under given simulated recording conditions was of the order of one per cent of the flux density obtained in conventional pickup cores having comparable front-gap lengths.

When electron-beam tube models became available they were found to offer the most convenient means of testing experimental core models, but an independent test to determine tube sensitivity was also required. The following procedure proved satisfactory: A magnetic yoke, of the form shown in the foreground of Fig. 17, is slipped over the tube and centered on the tube pole pieces in the position normally occupied by the external core structure. A known direct current is applied to the coil of the yoke and the change in plate-to-plate tube voltage is measured. The tube sensitivity in volts per gauss is then determined from the tube voltage output per unit of yoke current divided by the yoke constant in gausses per unit of yoke current.

type fluxmeter and an accurately formed search coil. A flux density of approximately 200 gausses is required for this measurement, whereas the electron-beam tube is normally operated with a beam-gap flux density of less than 0.1 gauss. The yoke calibration was shown to be constant throughout the range of 200 to 0.1 gauss through measurements with the more sensitive, but somewhat less convenient, vibrating probe which was previously described.

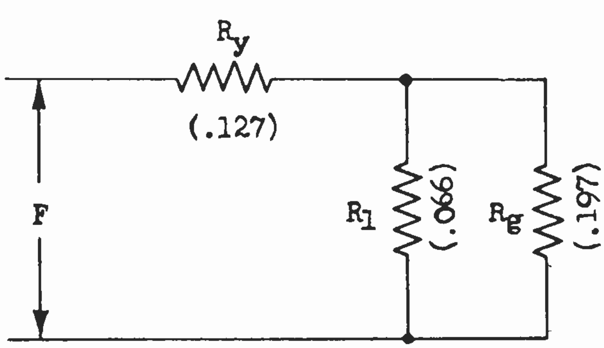
EFFECT OF PARAMETERS OF THE MAGNETIC CIRCUIT ON OVER-ALL PICKUP SENSITIVITY

The parameter of the magnetic circuit which was found most difficult to determine was the effective leakage reluctance of the tube pole pieces. Because of the complex shape of these structures and the large ratio of leakage to gap flux, field-mapping techniques provide only a poor approximation. The procedure which was used with reasonable success involved the measurement of the beam-gap flux density produced by a given yoke magneto-motive-force as explained below.

The magnetic circuit of the calibrating yoke in combination with the tube pole pieces may be represented as shown below. (Fig. 18) Beam-gap flux density for a given yoke current was determined by measurement as previously described to be

$$B_g = 278\, I \text{ gauss,}$$

where I = yoke current in amperes.



F = Applied yoke mmf

R_y = Total effective yoke-to-pole-piece reluctance

R_l = Effective pole-piece leakage reluctance

R_g = Beam-gap reluctance

Fig. 18 – Magnetic parameters of calibrating yoke and tube pole pieces.

The yoke constant applies only to the combination of yoke and given tube pole pieces. To determine this constant, the pole pieces are dummy-mounted in their proper positions and the yoke is substituted for the core shown in Fig. 17. With a known yoke current, beam-gap flux density is measured through the use of a light-beam

For a yoke having a coil of 100 turns,

$$F = 126\, I \text{ gilberts.}$$

Assuming all beam-gap fringing flux to be included in the pole-piece leakage flux, the area and reluctance of the beam gap as calculated from the actual gap dimensions are, respectively

$$A_g = 0.644 \text{ cm}^2$$
$$R_g = 0.197 \text{ cgs unit.}$$

With allowance for fringing, the total yoke-to-pole-piece reluctance as calculated from the effective gap dimensions is $R_y = 0.127$ cgs unit. Solving for R_1 in the circuit of Fig. 18, with substituted values as determined above, gives

$$R_1 = \frac{A_g B_g B_g R_y}{F \cdot B_g A_g (R_g + R_y)} = 0.066 \text{ cgs unit.}$$

It is interesting to note that R_g = 3.0 R_1 and that pole-piece leakage flux is therefore 3.0 times as great as the flux which exists in the beam gap.

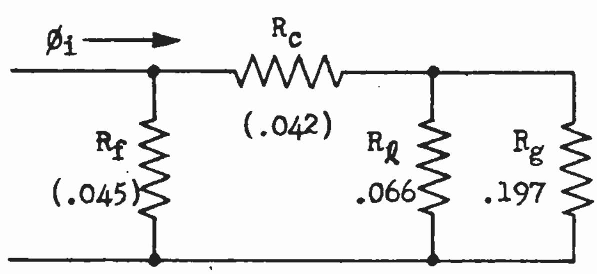
Using the constants determined above, the magnetic circuit of the external core in combination with the tube pole pieces may be represented as shown in Fig. 19. With allowance for fringing, the total core-to-pole-piece reluctance as calculated from the effective gap dimensions is

$$R_c = 0.042 \text{ cgs unit.}$$

Assuming the tape to act as a source of constant input flux, ϕ_i, the circuit of Fig. 19 may be solved to show that

$$\phi_i = 12.1\,\phi_g$$

where ϕ_g = the flux in the beam gap.



R_c = Total effective core-to-pole-piece reluctance

R_f = Effective front-gap reluctance

Fig. 19 – Magnetic parameters of core and tube pole pieces.

If ϕ_i is assumed to be equal to the flux developed within the recording medium its value may be determined from the following dimensions and characteristics of the recording medium:

Track width = 0.125 inch

Coating thickness = 0.65 mil

B_R = 600 gausses

Hence ϕ_i = 0.31 line and the beam-gap flux density for a saturated recording is:

$$B_g = \phi_g/A_g = \phi_i/12.1\,A_g = 0.040 \text{ gauss.}$$

The sensitivity of the tube at the time of the following correlation was

$$S = 14.9 \text{ volts per gauss.}$$

Hence, the expected tube voltage output for a saturated recording is:

(1) $V = B_g S = 0.040 \times 14.9 = 0.60$ volt, peak, plate-to-plate. Tests on ten core models showed an average output at 100 cps of 0.20 $\sqrt{2}$ volt, peak, plate-to-plate for a recording level of approximately six decibels below saturation. (It is important to note that the output level relative to saturation was determined on a peak basis by reading relative outputs on an oscilloscope. Readings on a vacuum-tube voltmeter were found to introduce an appreciable error, probably because of the difference in signal waveform at saturation and at lower levels.) Thus for a saturated recording, the average voltage output for the ten core models becomes:

(2) $V = 2 \times 0.20\sqrt{2} = 0.57$ volt, peak, plate-to-plate. The calculated output of (1) checks the experimentally determined output of (2) well within limits of accuracy of the procedures involved.

ALTERNATE, WINGLESS, STRIP-TYPE CORE

A brief study has been made of the performance of a wingless core model, F, which is similar to the previously described model B except that the core width is one inch. The outside diameter of the magnetic structure is approximately 0.75 inch. Loss in high-frequency response due to eddy currents was measured as 1.5 db at 10 kc and 3.0 db at 20 kc.

The measured unequalized response for this core in combination with the electron-beam tube is shown as the solid-line curve of Fig. 20. The dotted-line curve shows the calculated high-frequency response assuming the use of two ideal R-C equalizer sections having half-power frequencies one octave apart. The dashed curves show the manner in which the low-frequency response hump may be reduced by means of an auxiliary plate positioned a small distance above the reproducing gap. The two dashed curves correspond to two different positions of the plate and provide an indication of the degree of control which is possible. These measurements were made with the pickup enclosed in a 10½-inch-long shield box. Additional tests showed that the shield length may be reduced to three inches without introducing low-frequency irregularities exceeding plus or minus one-half decibel.

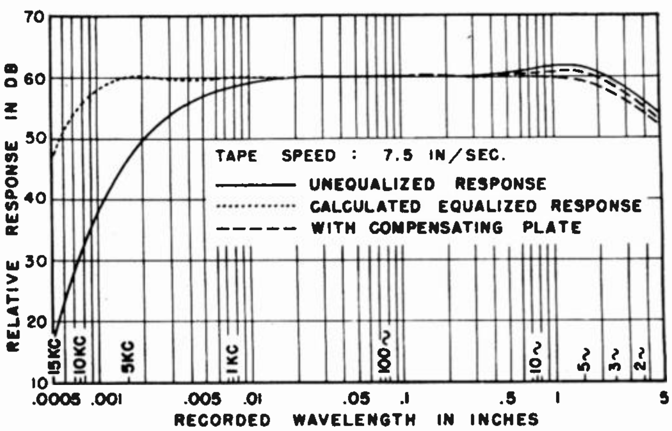


Fig. 20 – Wavelength response for core model F.

From these curves it appears possible to provide equalized response which is flat within plus or minus one decibel from approximately three to 7500 cps at a tape speed of 7½ inches per second using a commercial tape and a core having an outside diameter of approximately 0.75 inch. Under the same conditions the equalized response is flat within plus or minus three decibels from approximately two to 10,000 cps. At a tape speed of 15 inches per second an equalized response of approximately three to 20,000 cps, plus or minus four decibels, would be obtained.

ACKNOWLEDGEMENTS

This work has been sponsored by the Bureau of Ships of the U. S. Navy under Contract NObsr-57452. The tube has been under development at National Union Radio Corporation and the magnetic pickup structure at Stromberg-Carlson Company. I wish in particular to express my appreciation to Lynn C. Holmes, Director of Research at Stromberg-Carlson, for his interest and counsel concerning the work presented in this paper; to A. W. LaBeouf who made most of the measurements; and to Norman Cole and W. B. Latchford for their cooperation on the mechanical design and construction of models and special test equipment.

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The Institute of Radio Engineers

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A NOTE ON NOISE IN AUDIO AMPLIFIERS*

H. J. Woll and F. L. Putzrath
RCA Victor Division
Camden, New Jersey

The limitations of background noise present a universal problem to designers of high-gain amplifiers. This technical editorial was invited for the purpose of summarizing these limitations for readers of TRANSACTIONS of the IRE-PGA. It happens that the conclusions are not in complete accord with another paper in this same issue.** The papers are being published simultaneously so that readers may have the benefit of both points of view.

— Editorial Committee

There are a number of noise sources in audio amplifiers such as:

a. Thermal noise in the input coupling circuit
b. Shot noise
c. Partition noise in pentodes
d. Flicker noise
e. Ballistics and microphonics
f. Pops
g. Hum

Hum will not be considered here although it is a serious problem and much can be said concerning its elimination. Ballistics, microphonics, and pops are matters of tube design and selection rather than circuit design and will not be considered either.

This paper will be primarily concerned with flicker, shot, and partition noise and the effects of the input coupling network. An attempt will be made to outline the requirements of the input circuit and to discuss the magnitude of second stage noise in conventional configurations.

Noise is generated in the resistive component of any impedance by the thermal agitation of the electrons. The magnitude of the thermal noise voltage of a resistance in temperature equilibrium is:

$$e = \sqrt{4kTBR} \text{ volts rms}$$

where

k = Boltzmann's constant
T = temperature in degrees Kelvin
R = resistance in ohms
B = bandwidth in cycles per second

A certain signal-to-noise ratio is available from the source and is determined by the signal strength and the magnitude of thermal noise. Using an amplifier, it can be approached but never exceeded. Noise figure is a convenient way of expressing the amount by which an amplifier deteriorates the signal-to-noise ratio available from the source. Noise figure may be defined as the available signal-to-noise ratio at the source divided by the available signal-to-noise ratio at the amplifier output. An equivalent definition is that noise figure is the ratio of the total noise power at the output of the amplifier to that component of the noise output power which is due to thermal noise in the source impedance. Noise figure is costumarily expressed in db.

Shot noise is generated because the plate current of a tube is not continuous, but consists of discrete charges which are numerous enough to very closely approximate a continuous current. Partition noise in pentodes is similar to shot noise but is caused by the division of current between the plate and the screen. Flicker noise is caused by local fluctuations of emissivity of the cathode. These sources of noise may be lumped and their effect duplicated by an equivalent voltage generator. This generator is commonly represented by a resistor, R_{eq}, in series with the grid of the tube and is chosen to be of such a value that the thermal noise voltage generated by it is equal to the sum of the shot, partition, and flicker noise voltages referred to the grid circuit. Thus:

$$R_{eq} = R_{shot} + R_{part.} + R_{flicker}$$

Shot noise and partition noise are independent of frequency and their equivalent noise resistances are generally considered to be:

$$R_{shot} = \frac{2.5}{g_m}$$

$$R_{part.} = \frac{20\, I_{screen}}{g_m\, I_{cathode}}$$

In the audio frequency band, flicker noise is much greater than either of the above two noise sources. It is a function of frequency and thus its equivalent noise resistance, $R_{flicker}$, is also. For any particular frequency characteristic an integrated $R_{flicker}$ can be found that is a constant.

The noise spectrum of a typical low noise triode with a coated cathode is shown in Fig. 1. In this tube, R_{shot} is 1500 ohms and $R_{flicker}$ integrated over a 12 kc flat bandwidth is 6000 ohms.

Consider the problem of determining the noise figure of an audio amplifier stage. Identical grounded-grid and grounded-cathode triodes with the same source re-

*Manuscript received January 29, 1954.
**R. Lee Price, "The Cascode as a low noise audio amplifier."

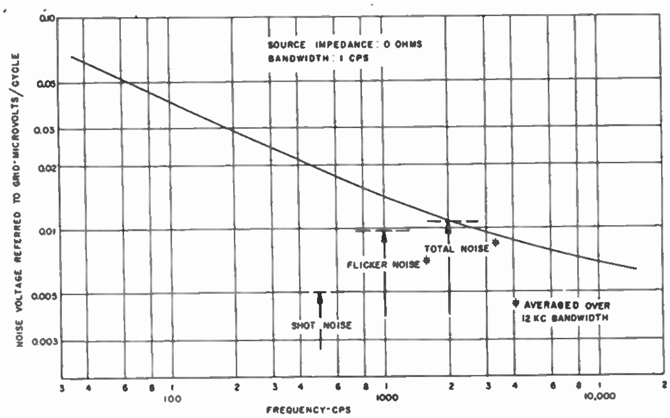

Fig. 1 – Triode tube noise vs. frequency.

sistance, R, are shown in Figs. 2 and 3. The tube noise is referred to the grid circuit and is represented by R_{eq}, which is a fictitious generator of voltage $e = \sqrt{4kTBR_{eq}}$. The tubes are then considered to be ideal amplifying devices. Since noise figure is:

$$F = \frac{\text{total noise power at the output}}{\text{that component of output noise due to thermal noise in the source}}$$

$$= \frac{4kTBR_s + 4kTBR_{eq}}{4kTBR_s} = 1 + \frac{R_{eq}}{R_s}$$

The interesting fact is that the noise figures of the grounded-cathode and grounded-grid stages are identical. Thus, although the input resistance of a grounded-grid stage might be under 500 ohms, a high source resistance

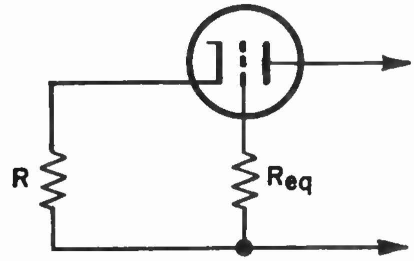

Fig. 2 – Grounded-grid equivalent circuit.

of perhaps 30,000 ohms is required to obtain a good noise figure just as in the case of the grounded-cathode connection.

It is to be noted that the above expressions represent *first* stage noise figures. The noise figure of an amplifier represents the deterioration of signal to noise ratio by *all* the stages in the amplifier. If the first stage gain is high, the succeeding stages do not contribute appreciable noise and the noise figure of the amplifier is about the same as that of the first stage. This is generally the case in audio amplifiers.

On the other hand, an amplifier with a grounded-grid input tube must be operated from a low source resistance to obtain appreciable first stage gain. Hence this amplifier is either operated from a high resistance source and suffers from noise contributed by the second stage, or is operated from a low resistance source and has a poor first stage noise figure, or some combination of the two.

Excepting ballistics, microphonics, hum, and pops, the following general conclusions can be drawn:

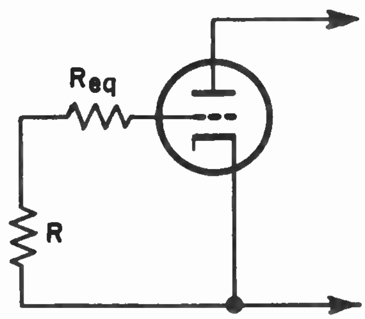

Fig. 3 – Grounded-cathode equivalent circuit.

Noise figure improves as the source impedance increases and zero db. noise figure can be approached in practice. (A low-noise triode with 22,000 ohm source resistance has a 1.0 db noise figure. See Fig. 4). As a result, efficient input transformers greatly improve the noise figure of a system with a low impedance source such as magnetic pickups and microphones.

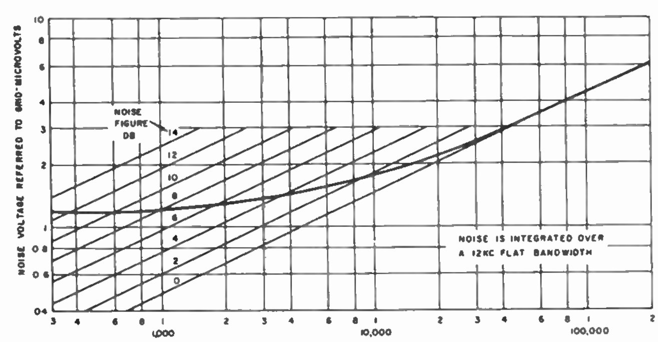

Fig. 4 – Noise vs. source resistance.

If the source impedance is low and bandwidth requirements or other conditions permit, the noise figure can be improved by paralleling input tubes increasing the g_m and thereby lowering R_{eq}.

It is to be noted that noise figure, *per se*, is meaningless. To express the performance of an amplifier, one must specify the noise figure with a given source resistance.

The noise figure of a stage is independent of the configuration, i.e., whether the stage is grounded-cathode or grounded-grid. Generally the gain of the first stage is high enough so that second and later stage noise make only a small contribution to the total.

Thus it can be generalized that cascaded grounded cathode stages at audio frequencies will give as good or better noise performance than other possible circuit configurations. In addition this configuration is most advantageous from a practical point of view – such as heater B+ supplies.

APPENDIX

A typical amplifier employing two cascaded grounded cathode stages is shown in Fig. 5. R_s and L_s are respectively the series resistance and inductance of

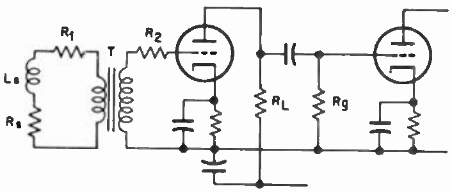

Fig. 5 – Typical two-stage amplifier.

the source. T is an input transformer with a turns ratio of n, a primary resistance of R_1, and a secondary resistance of R_2.

The equivalent circuit is shown in Fig. 6 where

$$L = n^2 L_s$$

$$R = n^2 R_s$$

$$R_t = n^2 R_1 + R_2$$

$$R_i = \frac{R_g R_L}{R_g + R_L}$$

The equivalent noise resistors of the two stages are represented by R_{eq^1} and R_{eq^2} respectively.

Examination of the noise figure of this amplifier will be made at point A. The noise figure at the input of the first ideal tube is:

$$F_1 = \frac{4kTB(R + R_t + R_{eq1})}{4KTBR} = 1 + \frac{R_t}{R} + \frac{R_{eq1}}{R}$$

The noise voltage due to the interstage coupling circuitry and the second tube is:

$$E_i = \sqrt{4kTB}\sqrt{\left(\sqrt{R_i}\,\frac{R_p}{R_p + R_i}\right)^2 + \left(\sqrt{R_{eq^2}}\right)^2}$$

$$= \sqrt{4kTB}\sqrt{\frac{R_i}{\left(1 + \dfrac{R_i}{R_p}\right)^2} + R_{eq^2}}$$

where R_p is the plate resistance of the first tube. Dividing the above expression by the first stage voltage gain, G, this noise voltage is referred to the amplifier input so that the ratio of the ideal to actual noise powers due to the source and due to the interstage circuitry is:

$$F_2 = \frac{\dfrac{R_i}{\left(1 + \dfrac{R_i}{R_p}\right)^2} + R_{eq^2}}{G^2 R_s}$$

The over-all noise figure of the two stages up to point A is then:

$$F = F_1 + F_2 = 1 + \frac{R_t}{R} + \frac{R_{eq^1}}{R} + \frac{1}{G^2}\,\frac{\dfrac{R_i}{\left(1 + \dfrac{R_i}{R_p}\right)^2} + R_{eq^2}}{R}$$

The above formula might be construed to mean that it would be desirable to let the source impedance have as high a resistive component as possible. However, an increase in source resistance must be accompanied by a corresponding increase in signal voltage, as is realized with an input transformer or with a "high impedance" magnetic playback head.

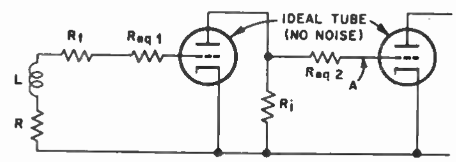

Fig. 6 – Equivalent circuit of two-stage amplifier.

A typical amplifier using a 12AY7 twin triode might have the following constants:

n = 28.3
L_s = 2 mh
R_s = 1 ohm
R_1 = 8 ohms
R_2 = 6000 ohms

$$R_{shot}\ 1 = R_{shot}\ 2 = \frac{2.5}{1660 \times 10^{-6}} = 1500 \text{ ohms}$$

$R_{flicker}1 = R_{flicker}\ 2$ = 6000 ohms
R_L = 100,000
R_g = 1,000,000 } R_i = 100,000 ohms
R_p = 25,000 ohms
G = 30

then:

$R_{eq^1} = R_{eq^2}$ = 7500 ohms
n^2 = 800
L = 1.6 hy
R = 800 ohms
R_t = 12,400 ohms

and

$$F = 1 + 15.5 + 9.4 + 0.02 = 25.9$$

or an equivalent ratio of 14.1 db.

From the above example it can be seen that the noise in this system is over 14 db worse than that which could have been obtained with an ideal amplifying device. A substantial noise contribution is made by the resistance in the input transformer, yet a negligible amount is contributed by the interstage coupling network and the second tube.

A preferred arrangement that would eliminate the noise contribution of the input transformer could be obtained by the use of a "high impedance" head. Thus:

$n = 1$

$L_s = L = 1.6$ hy

$R_s = R = 800$ ohms

$R_1 = R_2 = R_t = 0$ ohms

other values as above.

Under these conditions

$$F = 1 + 9.4 + 0.02 = 10.4$$

or an equivalent ratio of 10.2 db.

Even here, the second stage noise contribution is negligibly small.

HOUSTON PGA CHAPTER

L. A. Geddes, Chairman

On Thursday, December 10, 1953, the Houston Chapter of the Professional Group on Audio held its second meeting of the season at the Haliburton Oil Well Company Auditorium in Houston, Texas. The meeting was well attended. Only standing room was available after 8:30 P.M. The attendance was made up of 6 PGA members, 12 IRE members, and 42 guests.

The program commenced with a sound film on the manufacture and testing of magnetic tape. The process described in the film was that used by Audio Devices Inc., of New York. It is similar to the production processes carried out by other manufacturers.

Following the film, a talk, "Binaural Sound Principles and Practices," was given by Harry Keep of Gulf Coast Electronics Company. The requirements of the system were outlined and was followed by a description of recording and playback equipment. A short summary of the present status of the art and the possibilities of binaural reproduction for home use, was also given.

A demonstration recording of a ping-pong game on binaural tape was reproduced through a Magnecord Recorder, McIntosh Amplifier, and two Bozak Loudspeaker Enclosures. The tape ably illustrated the principles and listening possibilities inherent in the Binaural System.

The latest Emory Cook "Sounds of our Times" twin track disc recordings were then played through the system. The level of the two Binaural Channels was altered from *zero* to *maximum* to illustrate the illusion of presence and spatial distribution obtained with this method of reproduction.

The meeting formally adjourned at 10:30 P.M. and was followed by an informal session of more recorded music on an experimental *Request-from-the-Audience* basis.

HOW MUCH DISTORTION CAN YOU HEAR?

E. M. Jones, Chairman
Cincinnati Chapter IRE-PGA

"How Much Distortion Can You Hear?" was the subject of the November 17, 1953 meeting of the Cincinnati Chapter IRE-PGA. A specially prepared series of recorded tones was played as a demonstration test of percentages of harmonic and intermodulation distortion which can be detected. The preparation of the tape recording was a cooperative effort of the executive committee consisting of E. M. Jones, The Baldwin Company; W. W. Gulden, Cincinnati and Suburban Telephone Company; J. Parke Goode, National Sound Service. They were assisted by D. W. Martin of The Baldwin Company. The demonstration also included several live A-B comparisons from the oscillators and the distorting amplifier which had been used to make the tape.

The test series consisted of 15 different sample tones having known harmonic or intermodulation distortion, compared with "undistorted" signals. In each case the distorted signal and "undistorted" signal were presented five successive times in randomized paired comparisons. For each time, the listener was asked to choose whether the "undistorted" signal came first or second. The percentage distortion and the correct answers were announced after each group of five comparisons.

The test tones were sinusoidal signals of approximately 100 and 1000 cps presented singly (for harmonic distortion) or combined (for intermodulation distortion). The distortion was artificially created by using as a voltage amplifier a push-pull pentode amplifier with feedback removed. The amount of distortion was controlled by varying the input signal while keeping the output constant by means of an output attenuator. The same signal, bypassed around the amplifier, served for the "undistorted" tone. However, when the two signals were combined for intermodulation distortion, the larger, low-frequency signal was still passed through the distorting amplifier. Thus the same harmonic distortion of the low-frequency signal would be present, and the only clue to the difference between the distorted and "undistorted" cases would be the intermodulation distortion. The tape recorder produced some measurable distortion, although recording and playback were well below the levels recommended. The power amplifier (described elsewhere in this issue) and the loudspeaker system (exponential HF and LF horns) were operated at a power of less than one watt. Acoustical measurements on the amplifier-loudspeaker output at this power showed negligible distortion.

The percentage distortion in the tones played was measured with a wave analyzer at the input to the power amplification system. For the single tones the percentage harmonic distortion was computed by dividing the square root of the sum of the squares of the second and third harmonics by the fundamental. For the combined tones, the intermodulation distortion was computed by dividing the square root of the sum of the squares of the 800, 900, 1100 and 1200 cps components by the amplitude of the 1000 cps signal (the 100 cps signal being four times as great).

It must be emphasized that the actual percentages of distortion reaching a listener could vary greatly above or below the computed values, because of unavoidable variations in frequency response of both the loudspeaker system and the auditorium. A listener might be located where there was considerable acoustic cancellation at 1000 cycles and reinforcement at the adjacent frequencies. For this listener, the percentage intermodulation distortion would be greatly exaggerated.

Seventy-six test forms were turned in. A listener was considered able to detect the distortion if he guessed correctly at least four out of the five comparisons. With this definition, 19% of the listeners would "detect" the distortion by pure chance.

A summary of the results follows:

a. *Harmonic distortion of 1000 cps:* 87% of the listeners detected 11% distortion; 67% detected 4%; 42% detected 2.1%; 35% detected 0.9% distortion. "Undistorted" signal had 0.3% distortion in most cases.
b. *Harmonic distortion of 100 cps:* 42% of the listeners detected 10.5 distortion; 34% detected 4.3%; 43% detected 2.5%; 35% detected 1.7% distortion. "Undistorted" signal had .8% distortion.
c. *Intermodulation distortion of 100 and 1000 cps (on tape):* 85% of the listeners detected 22% distortion; 58% detected 11.5%; 37% detected 3.9%, and 28% detected 3.7% distortion. The "undistorted" signal had 1.5% intermodulation distortion.
d. *Intermodulation distortion of 100 and 1000 cps (live):* 42% of the listeners detected 5% distortion; 29% detected 2.2% and 21% detected 1.3% distortion. The "undistorted" signal had negligible intermodulation distortion.

Tabulation of the data according to average scores, or percentages of listeners getting five correct out of five attempts, gave similar distributions relative to the chance and perfect values.

It was concluded that most of the distortion percentages presented could be detected by a significantly large percentage of listeners in an auditorium, under the conditions of this particular experiment, i.e. simple, steady, sinusoidal tone presented immediately before or after a comparison tone that is identical, except having less distortion.

It is planned to add commentary to the tape used at this meeting for further use as a tapescript for IRE-PGA chapters.

PGA BRIEFS

Final approval was given by the IRE Executive Committee on January 5 for the establishment of IRE-PGA Chapters at both Cleveland, Ohio and Phoenix, Arizona.

Mr. Stanley K. Webster of the Beltone Hearing Aid Company gave a talk before the Cincinnati Chapter IRE-PGA on January 19, "An All Transistor Hearing Aid". The performance of the Beltone transistor hearing aids was described and the circuit designed by minimizing variations in transistor performance was explained.

We are informed that the Audio Section of the CONVENTION RECORD will be distributed to IRE-PGA members in the same manner as last year. This will be in addition to a regular May-June issue of TRANSACTIONS of the IRE-PGA. An effort will be made to obtain for later publication in TRANSACTIONS any papers which do not make the deadline for the CONVENTION RECORD. The May-June issue will contain the results of the IRE-PGA elections and audio news from the convention.

THE "VAGABOND" WIRELESS MICROPHONE SYSTEM*

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SUMMARY – A cable-less microphone system has been developed. Designed for general public address use, the system utilizes induction coupling between the transmitter and receiver, and requires no license. The subminiature transmitter is completely contained in a stick type microphone housing. It consists of a microphone, a five-tube, printed circuit frequency-modulated transmitter, a self-contained antenna inductor, and batteries, and weighs less than one pound. The system gives excellent performance with operating areas up to 5000 square feet. The development of the system and of the subminiature transmitter is discussed and the performance of the induction system is evaluated.

I. THE FUNDAMENTAL PROBLEM – MAINTENANCE OF ACOUSTICAL COUPLING

Since the first use of electrically amplified sound, audio engineers have been faced with the problem of minimizing the pickup of extraneous sound by the microphone. The usual approach to the problem is to keep the microphone and the sound source in close proximity. This may be comparatively simple if the sound source to be amplified has a fixed location.

However, if the source is mobile, as is frequently the case, the problem of maintaining this proximity can be a very difficult one. In the case of an individual performer, several approaches to the problem are possible.

A microphone on a fixed stand can be used. This approach may fail unless the performer has had training in microphone technique. In many situations, the enforced immobility of the performer greatly reduces the utility of the system and the value of the performance.

Alternatively, a microphone can be mounted on a boom and a trained technician so manipulate the microphone as to follow closely the sound source. This approach is satisfactory for motion picture and television work, but completely unsuitable for general public performance from the standpoint of expense as well as for obvious aesthetic reasons.

A method frequently used in public address and entertainment work is to attach the microphone to the user or to have him carry it. The most serious defect in this technique is the encumbrance caused by the microphone cable. The user must move about, dragging the cable after him and avoiding entanglements.

A solution to the problem which eliminates these difficulties is the use of a miniature radio transmitter which can be carried or worn by the performer. It was for this purpose that the Shure "Vagabond" System was designed.

*Manuscript received December 29, 1953. Presented at the National Electronics Conference, Chicago, Ill., September, 1953.

II. VARIOUS APPROACHES TO THE WIRELESS MICROPHONE PROBLEM

When we set out to design a wireless microphone system, two basic avenues of approach lay before us. The first was to design a high frequency system with a large operating range which would adequately cover Madison Square Garden, or Soldiers Field. Such a system must necessarily be relatively high powered and hence have short battery life. It would definitely require a Federal License. This would greatly restrict, or even prohibit its use in the very applications which we wished to reach, namely, general public address work, and theater and night club entertainment. Furthermore, those frequency assignments which might be obtained are available on a non-exclusive basis and will be subject to greater and greater interference as time goes on. A final disadvantage of the licensed system was the strict technical requirements imposed by law. These requirements would make the design of a miniature transmitter more difficult and would certainly increase its cost.

The second approach available to us was to design an induction system which would have a relatively restricted operating range, but would require no Federal licensing and hence be available to any and all potential users. If such a system could be developed to give adequate performance, it would offer excellent operating economy through prolonged battery life, and it would have to meet only those technical requirements dictated by satisfactory operation. Since it was felt that a practical induction system could be developed, that approach was chosen for the "Vagabond" system.

III. THE FUNDAMENTAL DESIGN OF THE TRANSMITTER UNIT

Having decided upon a non-licensed or induction system, other fundamental design decisions had to be made. These were the choice of the physical form of the transmitter, the modulation system to be employed, and the operating frequency of the system.

The Physical Form of the Transmitter

Two physical forms were considered for the transmitting unit. One form consisted of a pocket-size case, containing the transmitter and batteries, which could be worn concealed about the person of the user, with a separate lapel-type microphone attached to the transmitter with a cable. The second form considered was that of a stick-type case which would contain the microphone, antenna, transmitter, and batteries.

The stick form was adopted, as it offered several advantages. Being completely self-contained, the stick mike could readily be handed from one person to another, or it could be placed in a microphone stand and used conventionally. The performer would not have to "dress" himself in the microphone. Having no interconnecting cables, the self-contained unit would not be subject to wear and tear and would require less maintenance. It was also felt that the stick form, being quite similar to several conventional microphones in appearance and use, would place performers at ease and hence would be more readily accepted. The only manifest disadvantage of the stick form was its inability to be concealed easily on the person. This was felt to be a minor consideration, while the ability of one microphone to be used by several performers in immediate succession represented an economic advantage as it would replace several wearable units.

Choosing the Modulation System

Two primary considerations led to the choice of frequency modulation for the "Vagabond" System. Since the coupling between the transmitting and receiving antennas in a wireless microphone system will vary greatly, some means of assuring constant over-all audio gain is imperative. A frequency modulation system was chosen as the simplest means of achieving this goal.

Secondly, the desire to obtain the best possible signal-to-noise performance, with a transmitter of limited power, again indicated the use of a frequency modulation system. A modulation index of at least five was set as a goal.

Choosing the Carrier Frequency

The carrier frequency chosen for the "Vagabond" system was a compromise of opposing factors.

To obtain reasonable performance from a miniature transmitting antenna and to accommodate the bandwidth required by the frequency modulation system, it was desirable to use the highest possible carrier frequency. Furthermore, a study of the literature on the occurrence of noise throughout the radio spectrum indicated that less interference could be expected at the higher frequencies.

On the other hand, the Federal Communications Commission regulation governing unlicensed transmitters limits the radiated field strength in inverse proportion to the carrier frequency, thus dictating the use of the lowest possible frequency.

A final consideration was the minimization of interference from other transmitters operating on the same frequency. This obviated the use of the Broadcast, Amateur, Police, and Loran bands which lie between 0.5 and 2.0 megacycles. In view of all these factors, a carrier frequency of approximately 2.1 mc was chosen.

IV. THE MECHANICAL AND ELECTRICAL DESIGN OF THE TRANSMITTER UNIT

The Transmitting Antenna

Since the feasibility of the entire stick design depended on obtaining a satisfactory miniature antenna, the development of such an antenna was undertaken. Calculation and experiment indicated that within practical limits the size of the transmitting antenna would not affect the performance of an induction system. Since any antenna could be reduced to an equivalent magnetic dipole, the only effect of size variation was to change the power required to produce the desired field strength. A satisfactory ferrite-core transmitting inductor was developed which was three inches long and weighed two ounces.

Transmitter Circuit Design

Having achieved a practical ferrite antenna, a transmitter circuit was developed to operate from a 30-volt hearing-aid battery and a 1.3-volt mercury cell. (See Fig. 1.) Five subminiature tubes were used. The transmitter circuit is divided into two sections: a two-tube audio section and a three-tube radio frequency section.

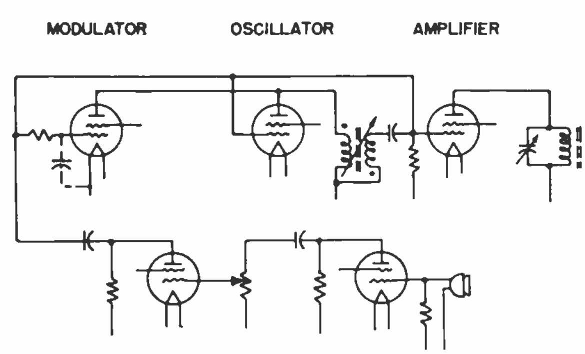

Fig. 1 – Transmitter schematic diagram.

Two tetrode voltage amplifiers are cascaded in the audio section to obtain a gain of 55 db at 1000 cycles. A miniature volume control between the two stages allows the gain to be adjusted for the desired degree of modulation. While the two-stage audio amplifier alone showed no tendency toward regeneration without any decoupling networks, motorboating at low frequencies occurred when it was connected to the reactance modulator, due to modulation of the plate supply voltage. To eliminate this motorboating, it was necessary to add a decoupling filter in the first audio stage and to restrict the low frequency response of the audio amplifier as well.

In order to obtain the best possible signal-to-noise ratio in the over-all system, an 80 microsecond pre-emphasis was used in the transmitter and a corresponding de-emphasis in the receiver. The problem of obtaining sufficient audio gain together with the necessary pre-emphasis was solved by providing a microphone cartridge of special design, with a response which very closely approximated the desired pre-emphasis curve. (See Fig. 2.)

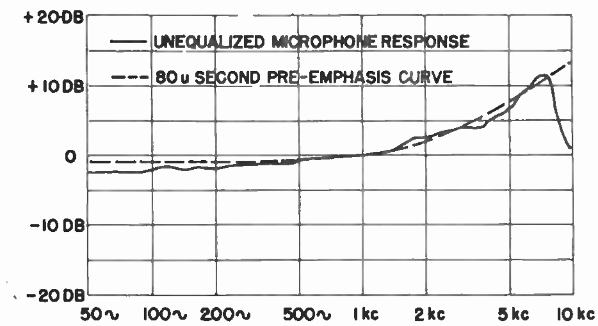

Fig. 2 – Microphone response curve.

An omnidirectional ceramic microphone cartridge was used. The choice of an omnidirectional unit was in conformity with present day trends in stick microphone design.

The radio frequency section of the transmitter consists of a self-controlled oscillator, a reactance modulator, and a radio frequency amplifier.

After trying many circuit arrangements utilizing only two tubes, the RF power amplifier stage was added. When only the two tubes were used, the oscillator necessarily had to operate at a higher power level, as its plate load was the antenna circuit. This arrangement required excessive modulator plate current to produce the two per cent frequency deviation required.

In order for a reactance modulator to cause a frequency shift of 1%, it must produce a reactive current which is approximately 2% of the magnitude of the circulating current in the oscillator tank circuit. If the oscillator tank circuit has a Q of 50, a typical value in this application, the reactance modulator must draw an average ac current equal to that of the oscillator and must be capable of 100% modulation of that current without distortion. This means that the modulator must draw three or four times as much plate current as the oscillator. This was obviously impractical, and it was necessary to use a low power oscillator followed by a power amplifier. A further advantage of such an arrangement was the elimination of carrier frequency shift because of antenna detuning caused by hand capacity.

A second circuit feature is the method of feeding signal and bias to the reactance modulator grid. Two operational difficulties are substantially reduced by this circuit arrangement. They are: first, the amplitude modulation of the oscillator due to variable loading by the reactance modulator, which produces distortion; and second, the carrier frequency shift which occurs as the batteries run down.

By obtaining the ac grid voltage for the reactance modulator from the secondary of the oscillator plate transformer, (connected out of phase with the primary), it is possible to use a single rc phase-shifting network and obtain a phase difference between the modulator grid and plate voltages which exceeds 90 degrees. The use of a modulator grid voltage which is more than 90 degrees out of phase with the plate voltage causes the reactance modulator to appear as a reactance in parallel with a negative resistance. This negative resistance will vary as the voltage on the modulator grid varies, and, with proper circuit adjustment, can be made to compensate for the variable loading of the oscillator by the modulator grid circuit. This minimizes the amplitude modulation of the oscillator and reduces the over-all distortion. Using the reactance modulator circuit shown, it has been possible to obtain excellent modulator performance together with adequate rf amplifier grid driving voltage at a reasonable total plate current. (See Fig. 3.)

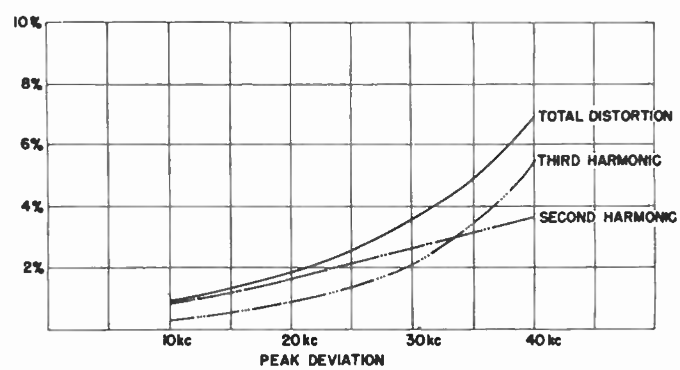

Fig. 3 – Transmitter distortion characteristic.

Since the use of any sort of automatic frequency control in the transmitter was out of the question due to the complex circuitry involved and the minute space available, it was necessary to design an oscillator-modulator circuit which would have sufficient frequency stability to meet the requirements of the system. This stability was achieved by means of the modulator grid bias circuit shown. The bias is derived from the self-rectified voltage developed by the oscillator and amplifier grids. As the plate and filament batteries run down in use, the transconductance of the reactance modulator will tend to be reduced and cause the carrier frequency of the transmitter to shift. However, the circuit arrangement used causes the dc grid bias of the modulator to be changed in such a fashion that its transconductance variation is greatly reduced.

Construction of the Printed Circuit Chassis

The entire transmitter circuitry, excluding the antenna and tubes, occupies a volume of about one cubic inch. The chassis casting, which contains eight condensers, eleven resistors, a volume control, five subminiature tube sockets, and a powdered iron core oscillator coil, is a cylinder one inch in diameter and 1.3 inches long. To obtain such extreme miniaturization, it was necessary to use rather unorthodox construction. (See Fig. 4.)

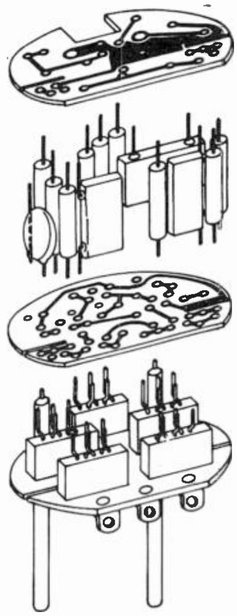

Fig. 4.

Two printed circuit plates form the fundamental structure of the transmitter chassis. They are made by a technique which produces a conductive pattern on both sides of a phenolic plate and on the inside surface of the holes through the plate. It is thus possible to produce cross connections between the two sides of the phenolic plate, automatically, and without the aid of eyelets. Since one printed circuit plate, with an area of less than one square inch, contains 54 holes for tube socket pins and component leads, the use of eyelets for making cross connections would be impossible. This is further emphasized by the fact that the center to center spacing of the tube socket pins is only 0.100 inch. A very important advantage of printed circuits of this type is the fact that it is possible to do all soldering on the upper side of the plate and still make connections to the pattern printed on the under side. In this assembly, the soldering of the 25 tube socket connections could be done in no other way.

Assembly of the chassis is begun by mounting the tube sockets in a phenolic plate which serves to position them and which carries three terminals for connection to the gain control. The lower printed circuit plate is then placed over the tube socket pins and soldered to them from the top side. Next, the various resistors and condensers are soldered to the top side of the lower plate. Finally, the upper printed circuit plate is fitted over the leads from the components, and the leads are cut off and soldered to the printed circuit. Again, connections are made to the printed pattern on the under side of the top plate by soldering on the upper side. (See Fig. 5.)

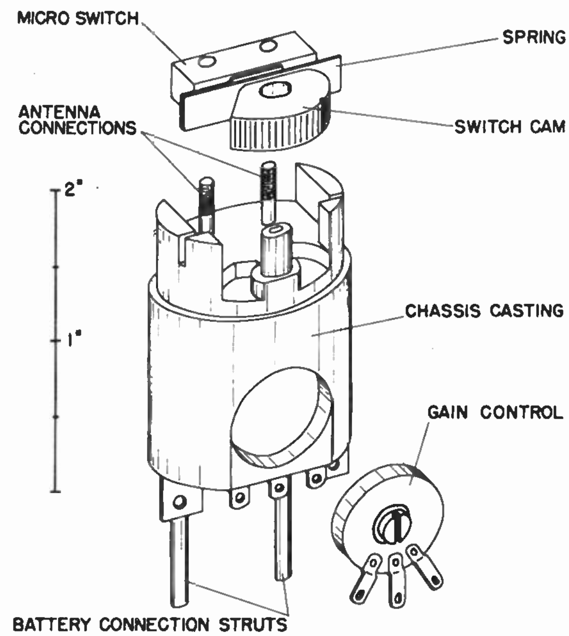

Fig. 5 – Transmitter chassis assembly.

After the printed circuits and components are assembled and inspected, the entire unit is placed in a mold and filled with a casting resin. Upon setting, this resin forms a rigid, moisture-proof mechanical assembly which cannot be damaged by vibration or shock, and which is readily attached to the other portions of the transmitter. An additional advantage of such an embedded circuit is the immobilization of all leads and components which eliminates any possibility of detuning due to mechanical motion.

After the chassis assembly is cast in resin, the gain control, filament circuit microswitch, and battery contact assembly are attached. (See Fig. 6.) The addition of the antenna and microphone completes the transmitter

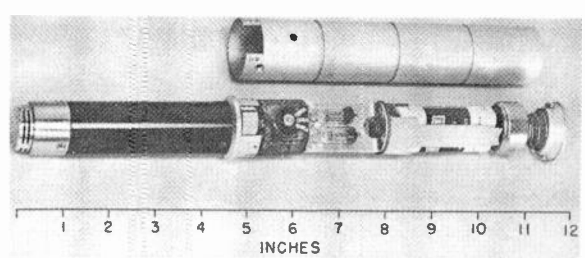

Fig. 6 – Complete transmitter chassis.

chassis assembly. A tubular case is attached to the transmitter chassis with three screws after the tubes are installed. The batteries are inserted in the bottom of the case and the battery cap is snapped into place. Connection to the junction between the filament and plate batteries is made automatically by the battery contact assembly.

Casting an entire circuit in plastic is certainly a case of "burning your bridges behind you," as it is virtually impossible to repair such a unit. However, any properly designed assembly which was good before casting, should be good after casting, and should remain good indefinitely. Component failure is very unlikely since the casting resin provides protection from corrosion and mechanical damage which are the chief causes of failure in very low power circuits.

It should be added, however, that a circuit which is to be cast must be designed to make allowance for the increased distributed capacitance caused by the dielectric properties of the casting resin. To simulate this dielectric effect during development work, sample assemblies were immersed in cottonseed oil, which closely duplicated the dielectric constant of the casting resin used. This proved to be an extremely useful experimental technique. Fig. 7 is a view of the completed transmitting unit.

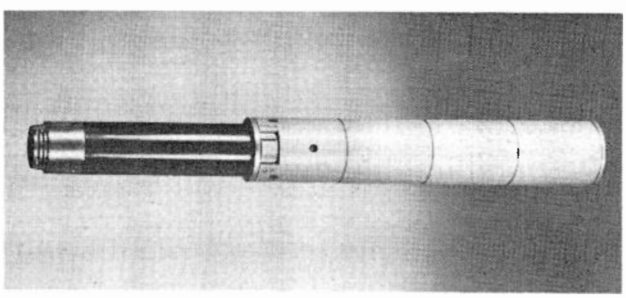

Fig. 7 – Over-all view of transmitter.

V. THE VAGABOND RECEIVER

The high performance superheterodyne receiver designed for the Vagabond System has a pentode, tuned radio frequency amplifier with a bandwidth of 150 kilocycles. A pentode mixer with a separate triode oscillator is used and automatic frequency control is provided by a pentode reactance modulator. The wide-band intermediate frequency amplifier employs two pentode amplifier stages, two cascaded triode limiters and a Foster-Seeley discriminator. The output of the discriminator is fed through a gated, cathode follower triode to the audio output terminals of the receiver. A triode DC amplifier is provided in the carrier-operated squelch circuit.

To aid in tuning the Vagabond transmitter, the receiver incorporates a double target tuning eye controlled by the outputs of the discriminator and the first limiter. By means of this tuning eye, it is possible to tune both the oscillator and antenna circuits of the transmitter. A three position switch on the receiver shifts the frequency of the receiver oscillator in 50 kilocycle steps to provide three operating channels. These may be used when more than one Vagabond System is to be operated simultaneously in the same vicinity or when strong interference is experienced on any one channel. Since the radio frequency amplifier has a 150 kilocycle bandwidth, no retuning of the receiver is required.

VI. CONCLUSION

The Vagabond System has been installed in a number of locations and some evaluation of the induction system can be made. It has provided ample quality for general public address and entertainment work with operating areas of 500 to 5000 square feet depending upon local interference conditions and the nature of the particular application. The audio response and signal-to-noise ratio have proved sufficient for radio broadcasting purposes in some studio installations.

The objective of developing a transmitting unit comparable in size and weight to current stick type microphones has been met. The transmitting unit weighs less than one pound and is 1¼ inches in diameter and 12 inches long. The life of a set of batteries is 40 operating hours. The battery cost is about five cents an hour.

Although the operating range of the induction system is limited, there are many applications where it has great value and where it alone is available.

VII. ACKNOWLEDGMENTS

The author wishes to express his gratitude to his fellow employees for the generous assistance, especially to Mr. John Knox, Assistant Project Engineer, Mr. J. S. Knechtsberger, Project Mechanical Engineer, and Mr. J. W. Medill, who developed the special microphone cartridge. Deep appreciation is also extended to Mr. B. B. Bauer, Vice-President – Engineering, and to Mr. E. V. Carlson, Development Engineer, for their many contributions to the project.

A HIGH EFFICIENCY – HIGH QUALITY AUDIO FREQUENCY POWER AMPLIFIER*

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The development of the power amplifier discussed in this paper was undertaken with the objective of providing a large amount of good quality audio power in a small package at relatively low cost. The size of an amplifier package and its cost are dependent largely on the efficiency of operation and the power sensitivity of the output stage. Beam power tubes satisfy these two requirements more readily than triodes. The quality of the audio power is improved by the use of push-pull operation and large amounts of feedback. The use of a suitable amount and type of feedback with beam power tubes eliminates the advantage of inherent low output impedance obtained with triodes.

In Class A or Class AB operation the dynamic characteristics of the two tubes may be matched to obtain a reasonably linear characteristic, and while feedback is desirable its use may sometimes be avoided. In Class B operation it is not possible to match dynamic characteristics to get linear operation, and therefore a large amount of feedback must always be used. The problems encountered in Class B operation are considerably more severe than those occurring in Class A and Class AB operation, and the question naturally arises as to whether this type of operation is worth the extra effort required. This question can be resolved rather conclusively by the following example:

Two *6L6* tubes operating Class A_1 push-pull with 30% efficiency, and having an allowable total plate dissipation of 40 watts, would be capable of developing an output power of $\frac{30}{70} \times 40 = 17.1$ watts.

The same two 6L6 tubes operating Class B_1, push-pull with 60% efficiency would be capable of developing $\frac{60}{40} \times 40 = 60$ watts. The possibility of getting 60 watts of output power in Class *B* operation for the same investment in power tubes which produced only 17.1 watts in Class *A* operation provides a real incentive to solve the severe problems encountered in Class *B* operation. The advantage of high efficiency operation also reflects advantageously on the size of the power supply necessary to operate the power amplifier.

The basic power amplifier circuit shown in Fig. 1 is capable of employing large amounts of feedback with good stability. This circuit has been operated with 36 *db* of feedback without showing any trace of instability. However, since the driving voltage required under these conditions is too great, the circuit is normally used with only 24 *db* of feedback. In this circuit the 12AX7 tube is used as a phase inverter – amplifier – driver stage. It is direct-coupled to the beam power output tubes and the

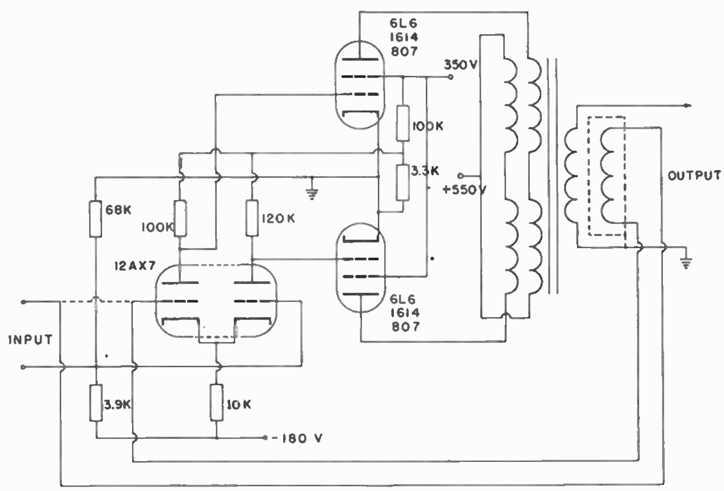

Fig. 1 – Basic Bereskin power amplifier circuit.

operation is essentially Class B_1 since the high impedance driver stage is incapable of driving the power tube grids positive. An added advantage of the direct-coupled driver is that it eliminates the possibility of blocking due to excessive input signal. The two output-tube cathodes are returned to ground and therefore any combination of screen and plate supply voltage may be used. The values shown on the diagram will keep both the screen and plate dissipation below the rated values for full signal Class B_1 operation with either the 1614 or the 807 tubes. The bias for the output tubes is supplied by the direct-coupled phase inverter – amplifier – driver and is normally adjusted to produce a zero signal plate current of about 15 ma per tube. The high value of cathode resistance employed makes the driver circuit fundamentally stable. A test with 6 different 1614 tubes and 12 different 12AX7 tubes, of different manufacturers and chosen at random, produced a zero-signal plate current variation ranging from 10 to 25 ma per tube. The full signal operation was substantially independent of the choice of 12AX7 and beam power output tubes. The feedback winding used is electrostatically shielded from the secondary but very closely coupled to it. The elimination of the electrostatic shield greatly reduces the amount of feedback that can be used successfully.

*Manuscript received February 2, 1954. Paper delivered at Cincinnati IRE Section, November 17, 1953.

One of the major problems associated with Class *B* operation is due to the energy stored in the leakage reactance between the two primary windings. A. P. Sah[1] showed how this stored energy gave rise to a conduction transfer notch which must be eliminated before Class *B* operation can be used successfully. Several different winding schemes will reduce the leakage reactance below the critical value but the most successful one is that of using a bifilar winding for the two primary sections.

The use of bifilar windings introduces new problems which were not previously important. One of these problems is that appreciable voltage may exist between the adjacent wires of a bifilar winding and sufficient insulation must be provided to withstand this voltage. Adequately insulated wires are now available commercially so this problem is seldom a serious one. A second and more significant problem is that appreciable capacitance exists between the adjacent wires in a bifilar winding and that charging current must be supplied to this capacitance before any voltage can be developed between the wires. This charging current must usually be supplied through the output stage tubes and is one of the major factors limiting the high frequency power-delivering capacity of an amplifier.

This problem can be understood more readily by examining the circuit of Fig. 2A. In this circuit the bifilar primary has been separated into two sections and the

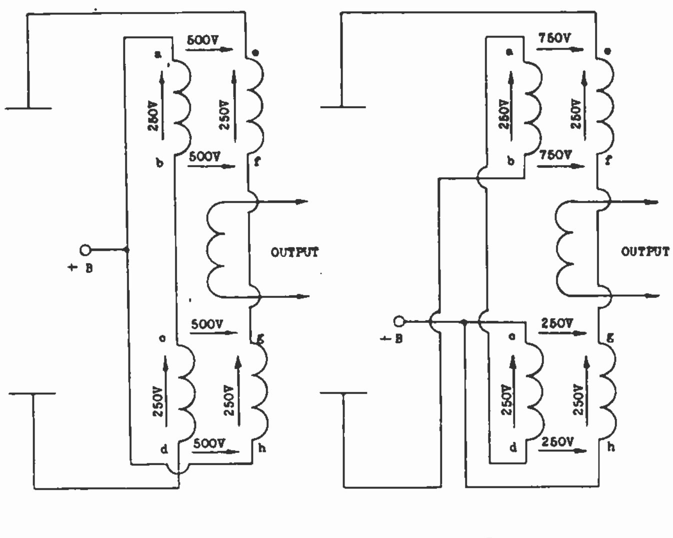

Fig. 2 – Primary interwinding voltage relations.

secondary has been sandwiched between them to keep the leakage reactance between the primaries and secondary at a low value. The four primary sections have been symmetrically interconnected and there is negligible dc voltage between these windings. If a peak signal voltage of 250 v is assumed on each of the primary sections, as shown by the vertical arrows, it is observed that this will give rise to a peak signal voltage of 500 v between all adjacent points on the bifilar winding as shown by the horizontal arrows. The undesirable feature here is that before this voltage can appear between the two primary windings the interwinding capacitance must be suitably charged, and the charging current must flow through one of the two tubes. An experimental transformer of this type, wound with adjacent-wire layer bifilar winding, using No. 28 Heavy Formvar wire, was found to have a capacitance of 0.045 microfarads between the two primary windings. A peak charging current of 1.5 amperes is required to charge this capacitance at 10KC with a sine wave voltage of 500 volts peak across the primaries. An ordinary tube used in a circuit of this type would normally supply only about 0.30 amperes peak so that at 2 Σkc the peak current capacity of the tube would be required to furnish the charging current for the interwinding capacitance. The amplifier's power-delivering capacity to a resistance load would therefore be down 3 db at 2 Σkc.

A natural step at this point is to consider the possibility of using a different interconnection of the primary sections to reduce the interwinding voltages, thereby reducing the charging current required for the interwinding capacitance. Fig. 2B shows a different connection of the primary sections. The voltage between the lower sections of the bifilar winding has been reduced to 250 v, but the voltage between the upper sections has been correspondingly increased to 750 v. The total interwinding charging current is the same as before but the insulation burden on the upper section is greater than it was before. As a matter of fact, further sectionalization and reconnection of the primary does not reduce the charging current problem and will, in most cases, increase the burden on the insulation.

A different type of interconnection is shown in Fig. 3. In this case half of the sectionalized bifilar

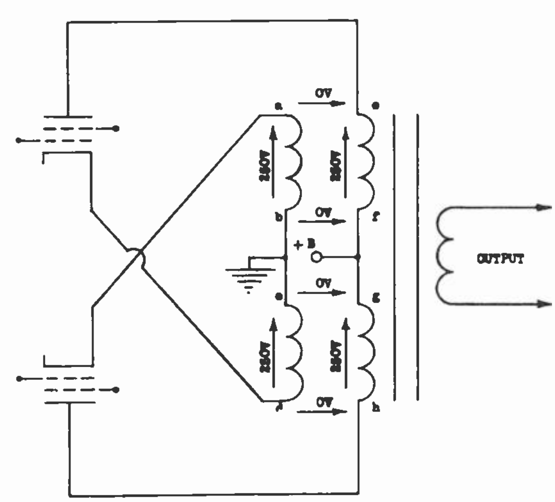

Fig. 3 – Basic McIntosh power amplifier.

[1]A. Pen-Tung Sah, "Quasi-transients in Class B audio-frequency push-pull amplifiers," *Proc. I.R.E.*, vol. 24, no. 11, pp. 1522-1541; November, 1936.

primary winding is connected in the plate circuit, as before, and the other half, with proper consideration for the signal polarities, is connected in the cathode circuit. Now points b, c, f, and g are all at zero ac signal potential. For signal conditions the same as those considered before, when a is instantaneously positive by 250 v with respect to b, e is positive by the same amount with respect to f. This means there will be zero ac signal voltage between points e and a. The same is seen to be true for points h and d and for all other adjacent points on the two bifilar windings. We now require zero charging current for the primary interwinding capacitance regardless of the value of this capacitance. Since point h is at the same ac signal potential as d, but at a much higher dc potential, it could be connected to the screen of the upper tube to supply a constant screen-cathode voltage for this tube. In a like manner point e could be connected to the screen of the lower tube. The screen and plate supply voltages in this circuit will be equal unless special circuitry is provided to make them different. This circuit also requires a grid drive voltage greater than 50% of the output transformer primary voltage. When this circuit is combined with a suitable driving circuit and feedback networks it becomes the McIntosh Power Amplifier[2] and is capable of delivering 50 w of high quality power over an exceptionally large frequency range.

A different solution to the problem is provided by the basic Sinclair-Peterson[3] circuit shown in Fig. 4A and its transformer coupled equivalent, using beam power tubes, in Fig. 4B. In the transformer coupled case both

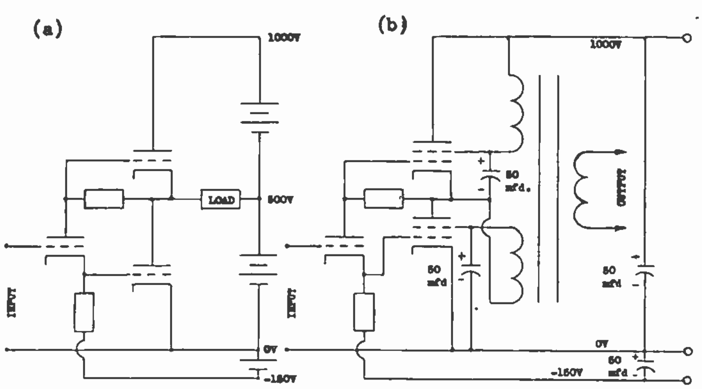

Fig. 4 – Basic Sinclair – Peterson power amplifier.

primary sections of the transformer work at all times so they need not be bifilar wound. These circuits present an appreciable burden to the phase inverter-driver for, if a peak primary or load voltage of 450 volts is assumed, it is seen that the phase inverter-driver plate supply voltage fluctuates between 1100 volts and 200 volts during the course of one cycle. Suitable points are available in Fig. 4B for supplying the screen voltages of the beam power tubes. A disadvantage of this connection is that the plate and screen supply voltages are forced to be equal unless special additional circuitry is used. The basic Sinclair-Peterson circuit has been incorporated in two complete power amplifier circuits which have been described in the literature[4,5]. In these circuits the screens are fed through suitable voltage dropping VR tubes to insure that the screen voltage will be less than the average plate supply voltage. The published data on these amplifiers indicate that their performance is substantially the same as that of the McIntosh amplifier mentioned previously. Neither the McIntosh nor the Sinclair-Peterson circuits use feedback around the output transformer.

The design of the transformer used in the circuit of Fig. 1 represents another solution to the problem introduced by the primary interwinding capacitance. In the general case of two isolated parallel circular wires an increase of the spacing between the surfaces from 10% to 20% of the diameter of the wire will reduce the capacitance between these wires by approximately 30%. An increase of this spacing from 10% to 100% of the diameter of the wire will reduce the capacitance by approximately 70%.

In the transformer winding even though we are not dealing with two isolated parallel wires but with many wires in a small space, the same general principles apply. Each wire will have capacitance to the two wires on each side of it in the same layer and also to the wires in the layers above and below it. The capacitance between wires in the same layer can be cut in half by transposing the two wires of the bifilar wire at every turn. The capacitance between wires in adjacent layers will not be modified by this process. In the non-transposed winding, assuming the same spacing between layer centers that exists between adjacent wire centers in a layer, and uniform dielectric material, the capacitance between the wires in the layers accounts for approximately two thirds of the total capacitance, while the capacitance between wires in adjacent layers accounts for one third of the total capacitance. Since transposition of the wires can be expected to cut in half the capacitance between wires in a layer without disturbing the capacitance between wires in adjacent layers, it should reduce the total capacitance to two thirds of its original value. The use of insulating materials such as cotton, varnish, and wax, and the accumulation of moisture will all tend to increase the capacitance. A mechanism has been developed which automatically transposes the bifilar winding at every turn so that this type of winding is no more

[2] F. H. McIntosh & G. J. Gow, "Description and analysis of a new 50 watt amplifier circuit," *Audio Eng.*, vol. 33, no. 12, pp. 9-11, 35-40; December, 1949.

[3] A. Peterson and D. B. Sinclair, "A single ended push-pull audio amplifier," *Proc. I.R.E.*, vol. 40, no. 1, pp. 7-11, January, 1952.

[4] A. P. Peterson, "A new push-pull amplifier circuit," *The General Radio Experimenter*," vol. 26, no. 5, pp. 1-7; October, 1951.

[5] H. W. Lamson, "A high power toroidal output transformer," *The General Radio Experimenter*, vol. 26, no. 6, pp. 5-8; November, 1951.

difficult to make than any ordinary bifilar winding.

An alternative to the transposed bifilar winding is a random wound bifilar winding. The random winding is not as consistent as the transposed winding but appears on an average to produce an increase in the capacitance of approximately 15%. Other than this the two windings are equivalent.

The introduction of space between the adjacent primary wires will tend to increase the leakage reactance between the two primary sections. The safety factor provided by the bifilar type of winding has so far been sufficiently large to avoid the appearance of the conduction transfer notch.

Fig. 5 is a schematic diagram of the output transformer used to make the tests discussed in the remainder of this paper, and Fig. 6 shows the coil buildup that was used. Fig. 6 is drawn in proper vertical scale but the horizontal scale has been modified to show the relative positions of the windings without showing the true coil width. This transformer winding was designed to be used with two grain-oriented Hipersil C Cores (Moloney ME-31 Hipercores or equivalent). The nominal impedance levels were intended to be 4, 8, and 16 ohms but the optimum levels obtained with this transformer were 4.63,

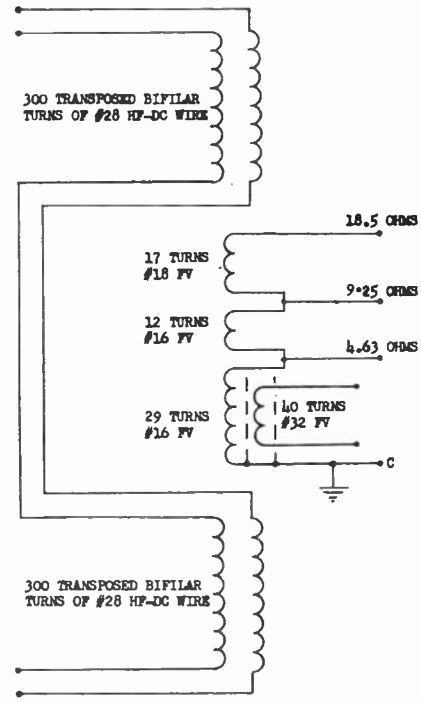

Fig. 5 – Output transformer winding arrangement.

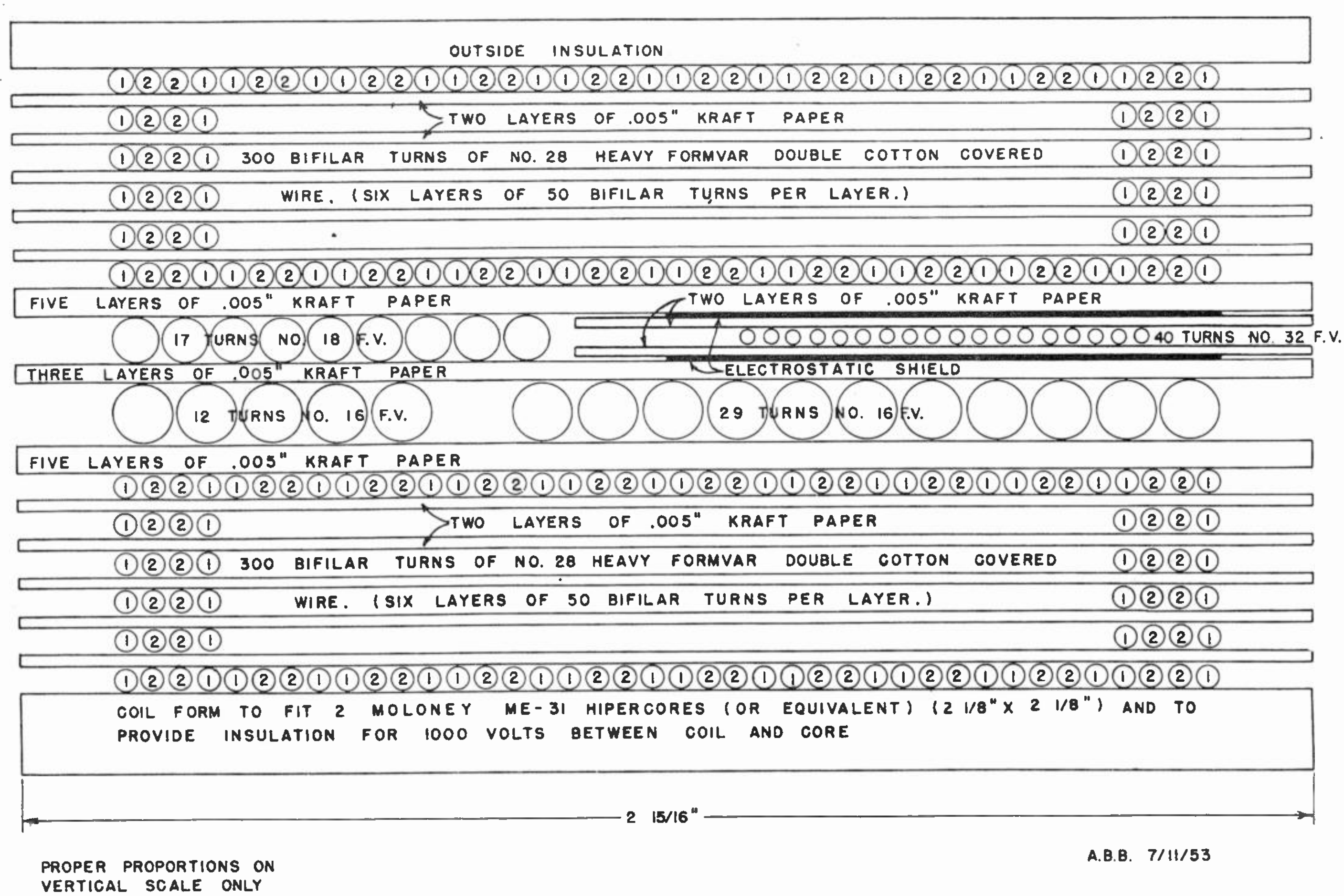

Fig. 6 – Output transformer coil buildup for Bereskin 50 watt 1614 tube power amplifier.

9.25, and 18.5 ohms. An optimum value resistor was used on the 9.25 ohm tap in all of the succeeding tests.

The transformer of Fig. 6 had a primary interwinding capacitance of 0.010 microfarads when it was vacuum impregnated with GE Type 9700 clear baking varnish and baked for the prescribed amount of time. This transformer was potted and has maintained the same value of capacitance since that time. Another transformer winding differing from this one only in the fact it had an ordinary layer bifilar winding had a primary interwinding capacitance of 0.015 mf. A third transformer winding in which No. 28 quadruple Formvar wire was used with the same spacing between wire centers and an ordinary layer bifilar winding had a primary interwinding capacitance of 0.012 microfarads. Additional paper was used between layers in this case to achieve the same spacing between wire centers in adjacent layers. This last winding has been found to have less tendency to pick up moisture from the atmosphere than the other two windings although potting eliminates this trouble in all cases.

In order to obtain the same low frequency power delivering capacity with an ordinary non grain-oriented core it is necessary to increase both the core cross section and the number of turns by about 25%. The combined effect of these increases is that the primary interwinding capacitance increases by approximately 50%.

The circuit of Fig. 1 was used with the transformer of Fig. 6 with transformer coupled input. The input transformer had an electrostatic shield and this shield was connected to ground. All filaments were operated from a common filament supply, one end of which was connected to ground. With well regulated screen and plate supply voltages, the residual hum in the output was 96 db below 50 w. For this particular amplifier, without any special attempt to obtain good balance, a ripple voltage of 42 volts inserted in series with the plate supply, or 9 v inserted in series with the screen supply, was necessary to bring the residual hum level in the output to 80 db below 50 w.

The power delivering capacity of this amplifier was tested by setting the input at the value necessary to produce 2% distortion in the output. Fig. 7 shows the results of this test. In the range below 30 cs the output was limited by the inability of the 1614 tubes to supply adequate magnetizing current to the transformer. Between 30 and 3000 cs the output was limited by peak clipping due to the inability of the 12AX7 tubes to drive the 1614 tube grids positive. Above 3000 cs the output was limited by the inability of the 1614 tubes to supply the charging current required by the primary interwinding capacitance. At the low end the amplifier has a drop-off rate of 9db/octave while at the high end it approaches a drop-off rate of 6 db/octave.

Most of the power in speech, song, and music is contained in the fundamental tones with frequencies below 3000 cs. The power levels of the higher frequency fundamental tones and of the harmonics of the lower frequency fundamentals drop off at a greater rate than the power-delivering capacity of this amplifier. It will be shown in the appendix that the power-delivering capacity of this amplifier is fully adequate for all audio frequency signals.

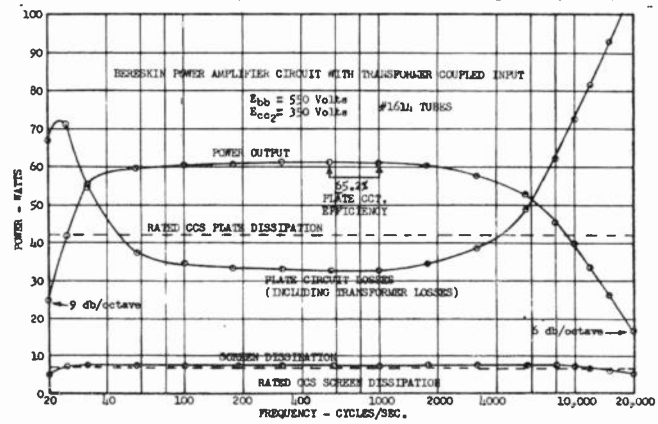

Fig. 7 – Two per cent distortion – power relations.

The amplifier develops its full power output of 60 watts over most of the middle frequency range with total plate circuit losses, including transformer losses, considerably lower than the rated CCS values. The screen dissipation exceeds the rated CCS value by about 7%, but this is only because the amplifier is being over driven to obtain the required 2% distortion. A reduction of 3 w in the output power, over most of the range, brings the harmonic distortion below 1% and the screen dissipation safely below the rated CCS value. The maximum plate circuit efficiency occurred in the 500-1000 cs. range and was 65.2%. This value includes the output transformer losses, and is ramarkably close to the ideal value of 78.5% for Class B operation, which does not include the transformer losses.

Since transformer coupled input is not usually available for amplifiers of this type a circuit including a preamplifier and power supply was developed and is shown in Fig. 8. This circuit was designed to deliver 50 w of high quality power over most of the middle frequency range. Since the output stage is very insensitive to ripple in the screen and plate supply circuits, very simple power supply filter circuits were adequate. Filter chokes were not necessary in either the plate or screen supply circuits. A single 5U4–G rectifier tube, operating within the manufacturer's ratings, was adequate to supply the power required by the plate circuits of the 1614 tubes. One 6X4 tube is used to supply the power required by the preamplifier and the screens of the 1614 tubes, while another 6X4 tube is used to supply the negative voltage required by the 12AX7 tube. The two 6X4 tubes could be replaced by a single rectifier tube with separate cathodes and plates, used as two single phase half-wave rectifiers, but the tubes available cost more than the two 6X4 tubes put together and, in addition, full-wave rectification in

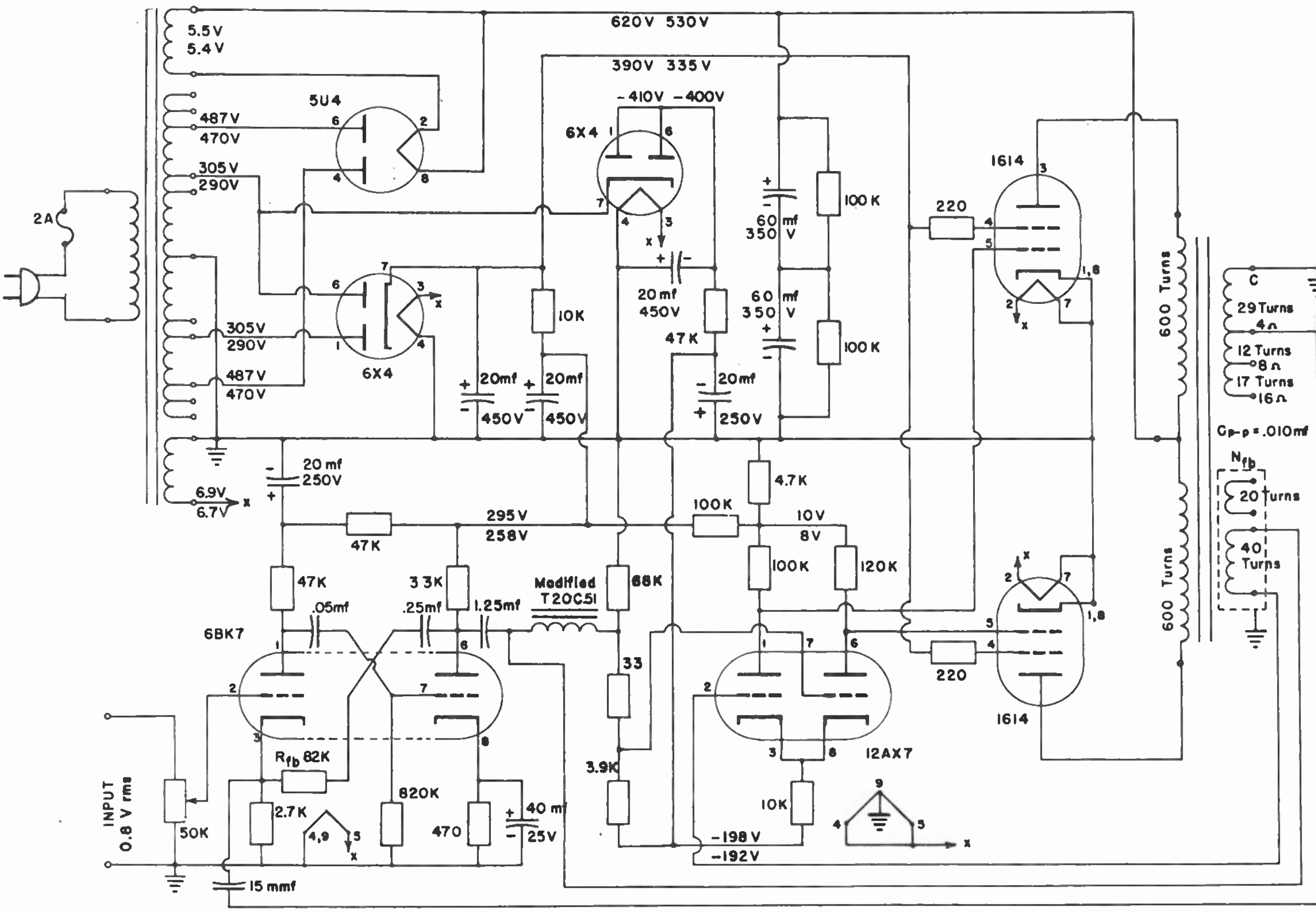

Fig. 8 – Bereskin 50 watt 1614 tube power amplifier.

the screen supply permits the use of a lower value of filter capacitance. The voltages indicated at various locations on the diagram are the measured no-load and full-load values.

The preamplifier consists of a two stage resistance-capacitance coupled amplifier with feedback between the second plate and the first cathode. This feedback provides good wave shape and low output impedance on the preamplifier. The preamplifier is coupled to the 12AX7 grid with a 1.25 μf capacitor and a modified Thordarson T20C51 choke. The modification consists in interleaving the laminations of this choke. A low dc resistance is necessary in this circuit because the 12AX7 has appreciable grid current when the grid voltage becomes more positive than −1 volt and this grid current must not be allowed to change the bias relations of the phase inverter. This coupling circuit has a low Q resonance between 10 and 15 cs.

Feedback from the secondary of the transformer is incorporated in a manner similar to that used before, but additional overall feedback has been incorporated from the 4.63 ohm tap to the first cathode in the preamplifier. It was found that a complex Bridged T network produced the best high frequency square wave response but that the square wave response was perfectly adequate when a simple 15 $\mu\mu$f capacitor was substituted in the overall feedback circuit. This capacitor has no effect on the low frequency response but reduces the tendency of the amplifier to ring slightly with sharp rise time square wave inputs. The photographs of Fig. 9 show the manner

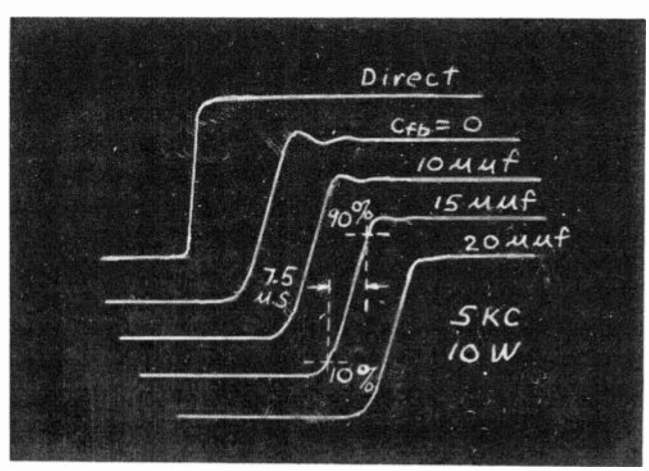

Fig. 9 – Leading edge analysis of 5 KC 10 watt square wave.

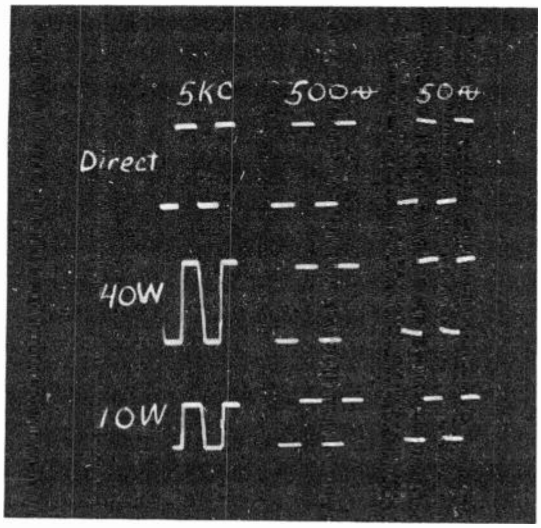

Fig. 10A – Amplifier square wave response.

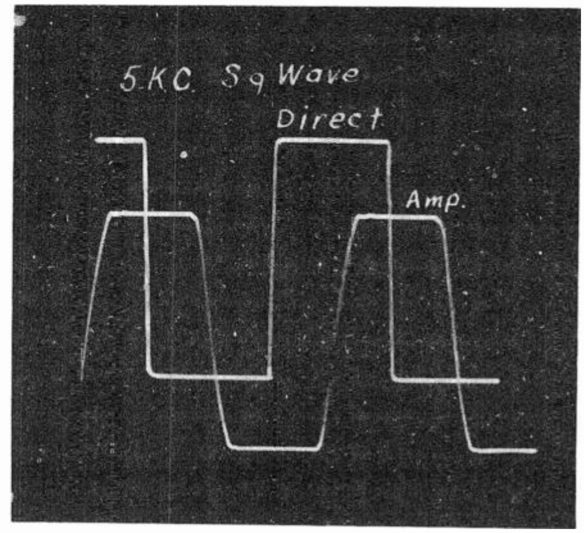

Fig. 10B – Enlarged view of amplifier response to a 40 watt 5 kc square wave.

in which the ringing, following the leading edge of a 10 w - 5 kc square wave, is modified by varying the value of this capacitor. The rise time of the leading edge is approximately 7.5 μs between the 10% and 90% points and the ringing frequency is approximately 100 kc. The complete square wave response at 50, 500, and 5000 cs. and power levels of 10 and 40 w is shown on Fig. 10A together with the output of the square wave generator connected directly to the CRO. A greatly expanded view of the 40 w - 5000 cs case is shown on Fig. 10B. The gain control settings were not changed during these tests.

The results of tests made to determine the best balance between the various types of feedback are shown in Fig. 11. In this figure curves 1 and 2 have inadequate feedback turns to correct for Class B operation at low power levels. The Bridged T overall feedback network produced 6 db feedback at operating frequencies so that curve 1 is lower than curve 2 at high power levels. Curve 3 has much less low level distortion than curve 2 because of the additional feedback turns but it requires more drive from the preamplifier and therefore has higher distortion than curve 2 at high power levels. Curve 4 uses approximately 6 db additional feedback in the preamplifier and its distortion is quite satisfactory at both low and high power levels. The 15 $\mu\mu f$ overall feedback has no effect at these frequencies. The values of feedback turns and preamplifier feedback resistance used here represent a practical compromise between low input signal and low output distortion. Additional reduction in distortion could be obtained by increasing the number of turns on the feedback winding and reducing the value of the feedback resistor in the preamplifier. Both of these changes would increase the input voltage required to produce full output power. The conditions specified for curve 4 are the ones shown in the circuit diagram of Fig. 8 and are the ones used in all succeeding tests. Fig. 12 shows the curves of plate circuit losses (in-

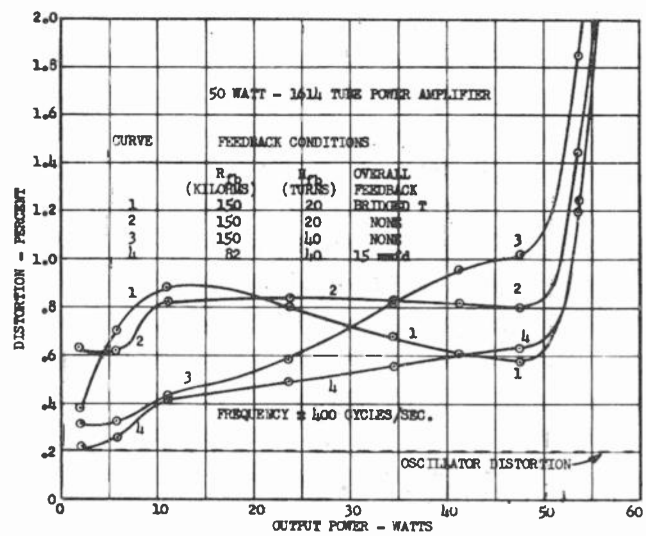

Fig. 11 – Feedba... – distortion characteristics.

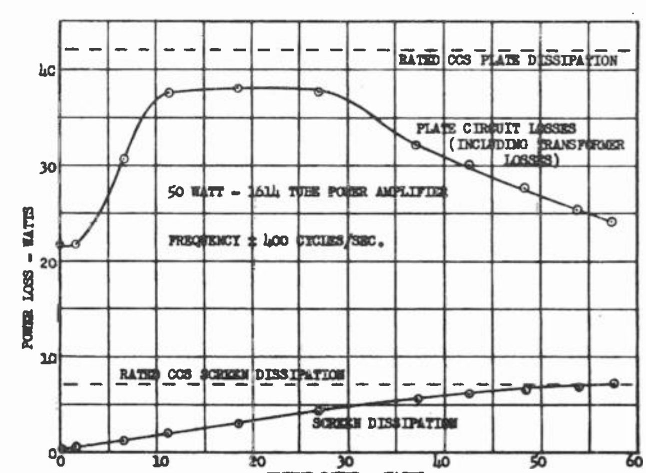

Fig. 12 – Power loss characteristics.

cluding transformer losses) and screen dissipation as the output power is varied. The plate circuit losses are less than the rated CCS values for all operating conditions. The screen dissipation becomes equal to the rated CCS value at the highest power levels shown but is less than the rated value at lower levels.

It can be seen from Fig. 11 that once the amplifier starts over-loading, the distortion increases very rapidly. For comparison purposes a 2% distortion level was considered a satisfactory standard value for determining the power delivering capacity of the amplifier. 1% distortion would have been adequate at 400 cs, where the oscillator distortion was only 0.2%, but not at 20 and 20,000 cs where the oscillator distortion was 0.8% and 0.6% respectively. Fig. 13 shows typical 2% distortion wave shapes at 20, 500, and 10,000 cs, for this amplifier.

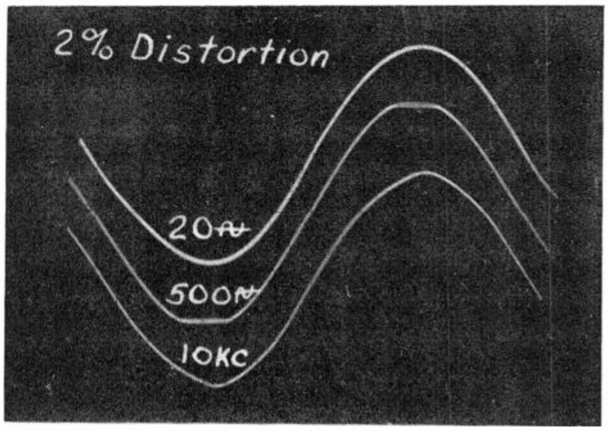

Fig. 13 – Two per cent distortion wave shapes.

The 2% distortion power handling capacity together with the corresponding plate circuit losses and screen dissipation are shown in Fig. 14. The highest plate

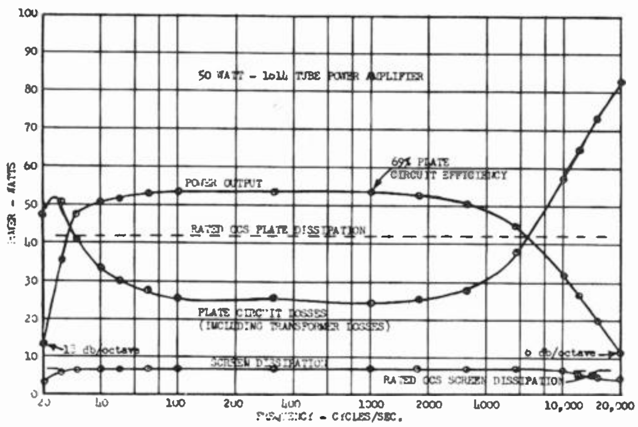

Fig. 14 – Two per cent distortion – power relations.

circuit efficiency, including transformer losses, occurred at 1000 cs and was 69%. The plate circuit losses are below the rated CCS value for the tubes alone and the screen dissipation is equal to the rated CCS value over most of the range. It should be emphasized that this curve represents the power delivering capacity of the amplifier and not the linearity of response with frequency variations. The frequency response characteristics, together with the power delivering capacity curve, plotted to a db scale, are shown in Fig. 15. It can be seen from

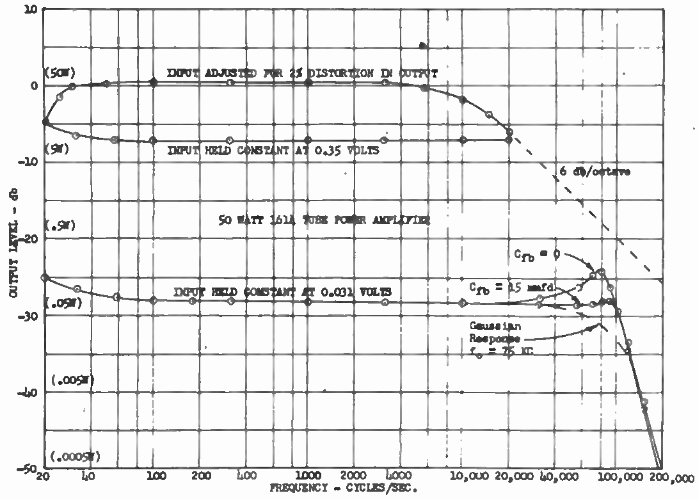

Fig. 15 – Frequency response characteristics.

this diagram that as long as the operating level is below the 2% distortion curve the response is perfectly flat between 100 and 20,000 cs. Below 100 cs the response rises slightly due to the series resonance in the impedance coupled circuit. The amount of this rise can be controlled by modifying the values of the coupling capacitor and choke. At low levels the response above 20,000 cs depends on the amount of feedback capacitance used. The curve for C_{fb} = 0 is seen to rise to a maximum at about 85 kc, and then to drop off very rapidly. The curve for C_{fb} = 15 micromicrofarads is seen to be almost perfectly flat to 95 kc, after which it also drops off very rapidly. A curve for Gaussian response with a −3 db point at 75 kc has also been shown for comparison purposes. Gaussian response would produce minimum rise time consistent with zero overshoot. It is seen that the ringing frequency of approximately 100 kc corresponds closely to the region of maximum deviation of the actual response characteristic from the Gaussian response. This also shows why the use of C_{fb} = 15 micromicrofarads reduced the ringing amplitude obtained with a square wave input signal.

Fig. 16 shows the results of an intermodulation distortion test using a 4:1 combination of 60 and 1500 cs. As is customary, the resulting distortion is plotted as a per cent of the smaller of the two signals. The distortion values shown here should be divided by 5 if they are to be compared with the harmonic distortion values discussed previously. The values shown are acceptable up to at least 112% peak-to-peak equivalent input. An intermodulation distortion test was also performed for a 4:1 combination of 60 and 15,000 cs. and the results are shown in Fig. 17. The purpose of this test was to find out if the high frequency roll off of the power handling capacity had any appreciable effect on the intermodula-

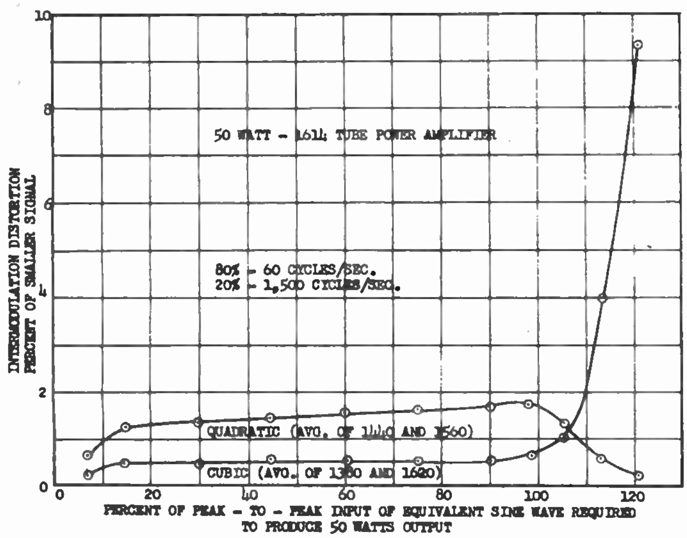

Fig. 16 – Intermodulation distortion characteristics.

tion distortion. The values obtained in this test were slightly higher than those shown in Fig. 22 but the performance is acceptable up to at least 105% peak-to-peak equivalent input.

Good transient response of a loudspeaker requires a low output impedance source. The manner in which the output impedance of this amplifier varies with frequency is shown in Fig. 18. The output impedance remains relatively constant with frequency and is about 10% of the nominal impedance of the tap. This output impedance was relatively independent of the current used to make the test.

A short circuited secondary test was performed on the amplifier to determine its susceptibility to accidental

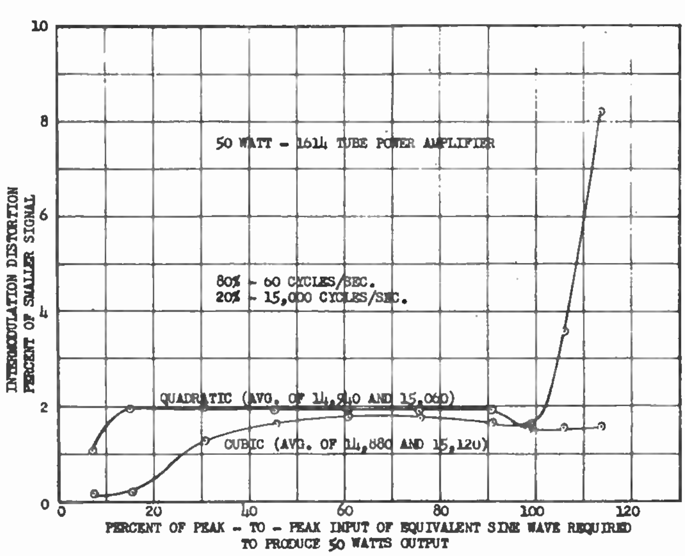

Fig. 17 – Intermodulation distortion characteristics.

damage. The results of this test are shown in Fig. 19. The screen dissipation of the tube reaches 4.5 w at relatively low signal levels but remains below 5 w even

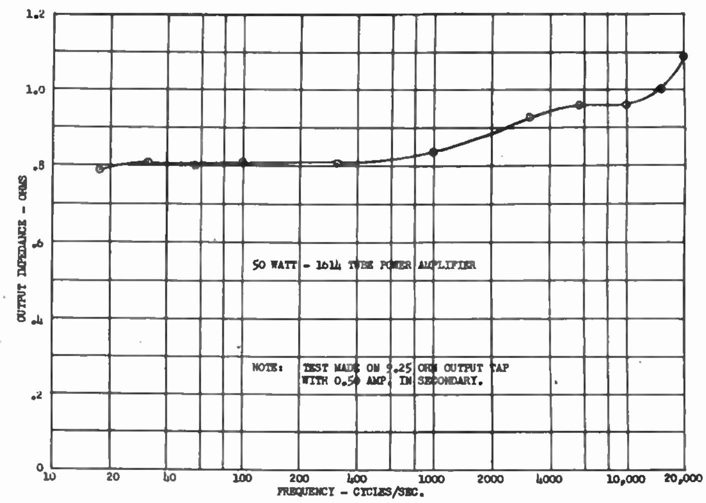

Fig. 18 – Output impedance characteristic.

with 120% of normal full load signal. With the same signal the plate circuit input approaches 120 w. This is considerably greater than the rated value but the plate structure is quite rugged and capable of handling these powers for short periods of time. The screen grid of the tube is quite fragile compared to the plate structure so it is fortunate in this situation that it only has to handle about 70% of its rated dissipation. This amplifier was operated for 10 minutes at the highest signal condition shown on the graph. At the end of the ten minute period the short circuit was removed and the amplifier operated normally. No damage could be detected in any part of the amplifier or its tube complement.

The amplifier can also operate with full signal applied under open secondary conditions. The only noticeable change is that the output voltage rises by about ten per cent when the load is disconnected. This

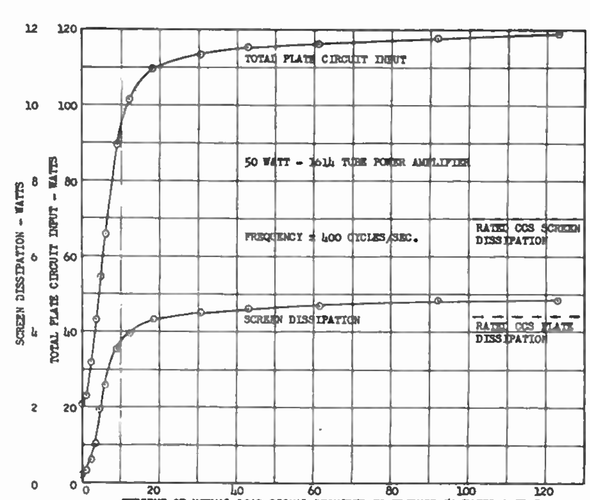

Fig. 19 – Short circuited output – power relations.

control is due to the close coupling used between the secondary and the feedback winding.

The residual hum in the complete amplifier is slightly greater than that measured in the basic amplifier with transformer coupled input. For the complete amplifier the residual 120 cycle hum is about 80 db below 50 w. The residual 60 cycle hum is about 66 db below 50 w. Due to its lower frequency, the 60 cycle hum is no more serious than the 120 cycle hum and both have been found to be completely negligible in most cases. The 60 cycle hum is picked up inductively by the unshielded modified T20C51 choke from the power transformer. If further reduction of the 60 cycle hum is desirable and the amplifier is operated remotely from other equipment, this choke can be oriented for minimum 60 cycle pickup. Approximately 20 db reduction has been obtained by this means but a clumsy mounting position was required for the choke. If the equipment is operated in proximity to other equipment whose relative position may change from time to time, a better solution would be to use a fully shielded choke. It should be emphasized, however, that in most cases further hum reduction is not necessary.

Fig. 22 is a photograph of a developmental model of the Bereskin 50 watt - 1614 Tube Power Amplifier. The output transformer is shown mounted on the chassis in its normal position but not potted.

APPENDIX

On the basis of the characteristics shown in Fig. 15, the 2% distortion power-delivering capacity of this amplifier is down approximately 6 db at 20,000 cycles while the low level response is flat to approximately 100,000 cycles. Theoretical and experimental investigations on both a qualitative and quantitative basis were made to determine the adequacy of this power-delivering capacity. It is the purpose of the appendix to discuss these investigations.

Sivian, Dunn, and White[6] and others have shown that most of the power in speech, song, and music is contained in the fundamental tones with frequencies below 3000 cs. The power levels of the higher frequency fundamental tones and of the harmonics of the lower frequency fundamentals decrease rapidly as the frequencies become greater than 3000 cs. Curve D in Fig. 20 is an adaptation of data presented by Sivian, Dunn, and White with regards to a 75 piece orchestra playing 4 different types of musical selections. In the original data the audio frequencies were divided into suitable but not equal bands, and for each of these bands the instantaneous peak power was recorded. This was done for each of the four musical selections and for many other sources of sound. The adaptation of this data consisted in taking the peak instantaneous power required in each of the four musical selections and calling this value 0 db. All four musical selections produced 0 db in the 250-500 cs band. The value plotted in any of the other bands was chosen from the musical selection in which this band had the minimum deviation from its own 0 db level. This curve is believed to be a close approximation to the most severe requirements for an audio frequency power system and has therefore been called the "MAXIMUM SEVERITY" composite. Curve E has been obtained by converting the

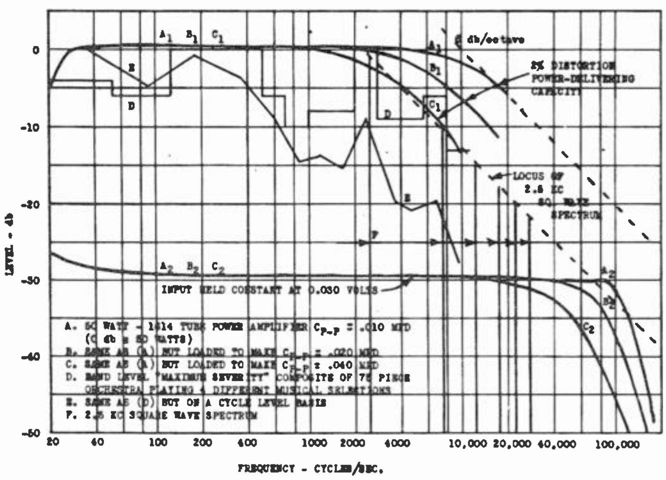

Fig. 20.

data represented by curve D to a cycle level. Sound is in general an integrated composite of many frequencies and these two curves show the relative contributions to be expected, under very severe conditions, from various frequency bands and ranges.

The 50 Watt – 1614 Tube Power Amplifier lends itself readily to the control of its power-delivering capacity without appreciably affecting its low level frequency response or its middle frequency distortion characteristic. Curve A_1 in Fig. 20 is the 2% distortion power-delivering capacity of the amplifier. Curves B_1 and C_1 are experimental 2% distortion power-delivering capacity curves for this same amplifier when it was loaded with .005 microfarad and .015 microfarad capacitors respectively between each of the 1614 tube plates and B+. For curve A the total primary interwinding capacitance is .010 microfarads, for Curve B it is .020 microfarads, and for Curve C it is .040 microfarads. Curves A_2, B_2, and C_2 are the corresponding low level frequency response characteristics obtained with a constant input voltage. The 400 cs harmonic distortion for all three cases was identically the same as that of Curve 4 in Fig. 11. The loading capacitors were connected with a rotary switch which made it possible to switch from any one condition to any other condition without going through the remaining condition.

Several full orchestra passages were recorded from LP records to tape with a Magnecord Tape Recorder operating at 15 inches/second. The tapes were cut into

[6]Sivian, Dunn, and White, "Absolute amplitudes and spectra of certain musical instruments and orchestras," *Jour. Acous. Soc. Amer.*, vol. 11, no. 3, pp. 330-371; January, 1931.

four foot lengths which were spliced into rings so that they could be played over and over with a one second dead interval in between to permit switching of the amplifier capacitance load. The Magnecord output was used as an input to the power amplifier and was also connected to the horizontal deflection system of a cathode-ray oscilloscope. The vertical deflection system of the cro was connected to the power amplifier output and the various gain controls were adjusted for a 45° trace on the face of the cro. Distortionless operation was characterized by a straight diagonal trace with slight tendency towards an ellipse due to the higher audio frequencies. Middle frequency distortion was characterized by horizontal extensions at the tips of the diagonal lines. High frequency distortion was characterized by small loops, similar to musical half note marks, at the tips of the diagonal lines.

For each section of tape the input level to the power amplifier was adjusted so that the amplifier just failed to clip peaks at the highest level passage of this section when the condition of Curve A was used. Switching the amplifier to the conditions of Curves B and C produced correspondingly larger indications of high frequency distortion. Reductions in input level ranging between ½ and 1 db were necessary to eliminate the distortion for the condition of Curve B. Reductions in input level ranging between 2 and 3 db were necessary to

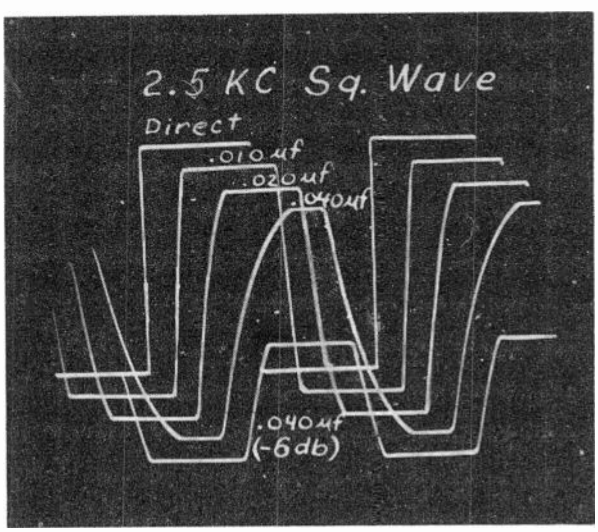

Fig. 21 – Effect of capacitance loading on square wave response of amplifier.

eliminate the distortion for the condition of Curve C. In all cases the distortion shown by the condition of Curve A was of the middle frequency peak clipping variety. Listening tests made at full level for these conditions showed in general that a slight difference could be detected between the conditions of Curves A and C but there was no conclusive agreement as to which represented the better operating condition. In the region of 2500 cycles/second Curve B is seen to be approximately 1 db below Curve A and Curve C is seen to be approximately 3 db below Curve A. This same region is seen to correspond to humps in both Curves D and E. These humps are probably exaggerated by the manner in which the "Maximum Severity" data was adapted from the original data but there is ample evidence that the 2000 to 2800 cycle/second band makes an appreciable contribution to the power content of the composite signal. Beyond this band Curves A_1 and B_1 safely override the band level "Maximum Severity" composite curves. All three of the Curves A_1, B_1, and C_1 are seen to drop off at a lower rate than the cycle level "Maximum Severity" composite in the high frequency range.

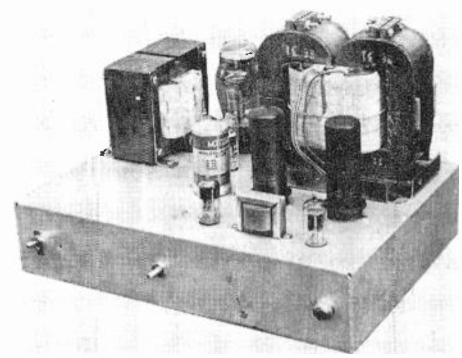

Fig. 22

This amplifier was used in listening tests with several combinations of full frequency range equipment and LP records containing full orchestra passages. In some of these tests the input level was adjusted to just fail to produce peak clipping during the most severe full orchestra passages with the condition of Curve A. In the remainder of the tests a high level was maintained but this level was below the peak clipping region. The transition between conditions A, B, and C was noiseless and in no case was the audience able to detect any difference between the three conditions.

The spectrum of a 2500 cs square wave, having a total power of 0 db, is shown up to the 11th harmonic by the arrows marked F in Fig. 20. The components above 27.5 kc have been left out to avoid confusion in the diagram but their effect can be understood by recognizing that only odd harmonics are present and that the spectrum level of a square wave drops off at a 6 db/octave rate. Several of the harmonics of this wave are in the audible frequency range and the drop-off rate of the harmonics does not differ too much from the band level and cycle level "Maximum Severity" composite characteristics. The behaviour of the amplifier conditions to a full level square wave input signal could be used for comparison purposes to interpret the high frequency power delivering capacity of other power amplifiers.

The oscillograms in Fig. 21 were obtained by the application of a 2.5 kc square wave to the amplifiers corresponding to Curves A, B, and C in Fig. 20. The input signal used in the upper 4 waves was equal to that required to produce 50 w output at 250 cs. The power present in the output was slightly less than 50 w because of the sloping sides of the output waves. The input signal used for the lowest wave was 6 db below that used for the other curves. The top curve is used for reference and represents the application of the output of the square wave generator directly to the cro with the same gain control settings that were used in all of the other tests.

The rise times between the 10% and 90% points are tabulated below:

C_{P-P}	Signal Level	Rise Time (10% to 90%) (microseconds)	(% of period)
(microfarads)			
.010	0 db	22	5.5
.020	0 db	44	11.0
.040	0 db	92	23.0
.040	-6 db	36	9.0
(direct from sq. wv. gen. to CRO)	0 db	2.5	.63

The wave for condition (C_{P-P} = .040 mfd) shows a relatively large amount of rise time for full signal operation but this rise time is reduced to less than half its original value when the signal level is reduced by 6 db. All of the waves are characterized by an extremely small amount of ringing which is an indication of good low level frequency response.

It appears from the tests discussed previously that the condition of Curves B (C_{P-P} = .020 mfd) represents a transition range where it is possible to detect incipient loss of high frequency program material with instruments but the loss involved is still insufficient to be detected from listening tests. Although further investigations will be made, it appears reasonable at this point to assume that an amplifier will have adequate high frequency power-delivering capacity if it can reproduce a 2.5 kc square wave at full signal level with a rise time of less than 40 microseconds between the 10% and 90% points of the wave. To avoid the possibility of audible intermodulation components due to the combination of ringing and high audio frequencies, the ringing amplitude should be relatively small and the ringing frequency should be relatively high.

It has been shown that the amplifier discussed in this paper has adequate high frequency power-delivering capacity for all normally encountered audio frequency signals. The middle and low frequency power-delivering capacity and the transient response are all excellent. The harmonic and intermodulation distortion are uniformly low and could be further reduced at the expense of requiring additional input signal. All of these performance features have been achieved with a structure which is compact and relatively inexpensive and capable of withstanding unusual abnormal conditions, such as full signal input with short circuited load, without damage to the amplifier itself.

THE CASCODE AS A LOW NOISE AUDIO AMPLIFIER*

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The improvement of signal-to-noise ratio has always been a major problem to engineers engaged in the development and design of high quality audio equipment. Faced with the limitations of the maximum signal level obtainable from transducers together with the minimum accompanying amplifier noise level, improvements in signal-to-noise ratios over the years have been made slowly by improvements in transducers and amplifiers. For example, a microphone with a maximum output signal level of -55 dbm obviously cannot achieve a signal-to-noise ratio greater than 60 db with an amplifier whose equivalent input noise level is -115 dbm.

Noise level is a particularly serious problem in recording. The development of modern professional quality tape recording has greatly increased the signal-to-noise ratio which is generally available in recorded form.

Even here noise is still a problem, for even when it is 50 to 60 db below the signal, a careful listener can detect it. The output from multi-channel and narrow track heads is restricted by the available track width and the resulting signal to noise ratio is then usually limited by the amplifier input stage noise level. Most of the recent work on the improvement of signal-to-noise ratio has been in the direction of increased signal output by

*Manuscript received February 9, 1954; See also H. J. Woll and F. L. Putzrath, "A note on noise in audio amplifiers" in this issue. These two papers are being published simultaneously so that readers may have the benefit of both points of view. — Editorial Committee.

the development of higher output tapes and heads. Some work also has been done on the decrease of noise level by the development of tapes with better uniformity, bias oscillators of improved waveform, and magnetic pickup heads with reduced microphonism or magnetostrictive noise. The improvement which can be obtained by these methods, however, is limited by the tube noise of the input stages of high gain amplifiers. Most of the recent work in reducing the equivalent input noise level of audio amplifiers has been confined to refinements in the design of existing circuits and components. Some degree of improvement in input stage noise is sometimes obtained by the use of selected tubes, the selection being made either by the tube manufacturer who gives the selected tubes a special type number, or by the user on a trial and error basis. It is obvious, however, that tube selection is not a completely satisfactory answer to the low noise amplifier problem.

In the present development, a completely different approach was used. Techniques from outside the audio field were reduced to their basic form and then modified to give the desired results at audio frequencies. Work done during the war by Wallman and others resulted in the development of the grounded cathode triode grounded grid triode, or "Cascode" amplifier circuit. Wallman has shown[1,2] that this amplifier gives the lowest noise figure obtainable at the present state of the art. It has been used with considerable success in the improvement of the noise figure of high frequency circuits used in radar, television, and many other RF applications. More recently, it has been used to reduce the noise level and to increase the dynamic range of video input amplifiers[3,4] for television cameras.

The first use for the grounded cathode triode in series with a grounded grid triode was in a dc amplifier in a voltage stabilizer.[4,5] The low noise feature of the present circuit is, however, entirely unconnected with this original use. An investigation of the basic form of the "cascode" circuit given by Wallman[5] and shown in Fig. 1 shows that it is not the exclusive property of the high frequency worker. In fact, the elimination of tuning and neutralizing circuits which are unnecessary in audio applications, results in considerable simplification over the forms usually seen.

As shown in Fig. 1, the basic "cascode" circuit consists of a grounded cathode triode in series with a grounded grid triode. Additional coupling elements are not needed between the two triodes and the same dc flows through both tubes. This "cascode" connection combines the advantages of the grounded cathode triode with those of the pentode while minimizing their disadvantages.

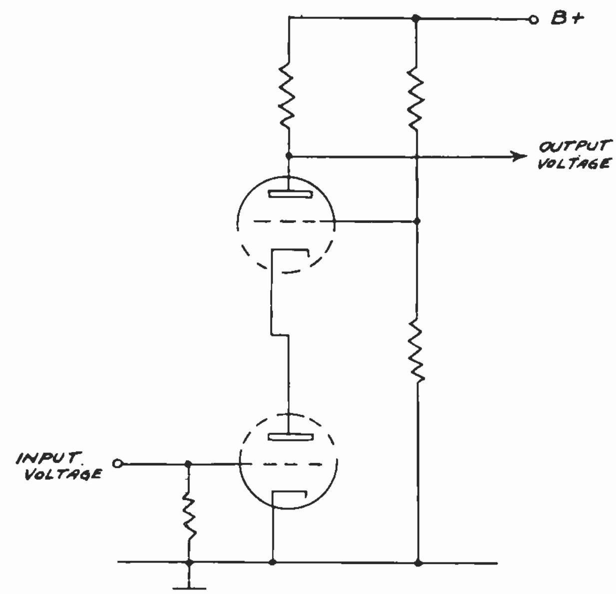

Fig. 1 – Basic form of cascode circuit.

THEORY OF OPERATION

The operation of the audio frequency "cascode" amplifier can perhaps best be undersood by considering briefly the relative advantages and disadvantages of grounded cathode pentodes, grounded cathode and grounded grid triodes.

The grounded cathode pentode amplifier has the advantages of high gain, and low input capacity due to the isolation of the plate circuit from the control grid by the screen grid. For these reasons, pentodes are often used in audio input stages in spite of their disadvantages of a noise level which is three to five times higher than that of a triode. The grounded-cathode triode has moderate available power gain with an output conductance of $1/r_p$ and a low equivalent input noise level.

The grounded grid triode has a large input conductance and a rather low available power gain.

The ideal triode input circuit should have all the improvement in input stage noise figure over the pentode that is theoretically possible, with the contribution of second and later stages to the noise figure no greater than with pentode input stages of the same bandwidth. Therefore, the stage gain should be of the same order of magnitude as that of a pentode. In addition, the circuit should be stable, with low input capacity and effective isolation of the input circuit from the output circuit.

Out of the nine combinations possible with two triode input tubes, using various combinations of grounded cathode, grounded grid, and grounded plate, the only

[1] Valley and Wallman, "Vacuum Tube Amplifiers," McGraw-Hill Book Company, New York, pp. 656-660; 1948.
[2] Wallman, Macnee, and Gadsden, "Low noise amplifier," *Proc. I.R.E.*, vol. 36, no. 6, pp. 700-708; June, 1948.
[3] K. B. Benson, "Modified preamplifier improves movie telecasts," *Electronics*, vol. 26, no. 12; pp. 166-169; December, 1953.
[4] "Application note on Type 6198 Vidicon television camera tube," published by the Tube Department, RCA.
[5] Valley and Wallman, ibid., p. 440.

circuit which meets all of the above requirements is the grounded cathode triode in series with a grounded grid triode or "cascode" circuit.

With this circuit arrangement, the upper tube has a fixed dc grid potential at ac ground which tends to hold the lower triode plate potential fixed, but still permits its current to flow in a load resistor. If e_{p_1} were really held constant, the current gain of the lower triode would be g_m and the voltage gain from e_{g_1} to e_{p_2} would be $-g_m R_L$ as in a pentode. Thus, the behavior is similar to a pentode, with the advantage that no screen current with its accompanying partition noise is present. The grid of the upper tube, being bypassed to ground, has an effect on stage behavior similiar to that of a screen grid. The output conductance of the first tube is of the same order of magnitude as the optimum source conductance for the second tube, so that the full available power gain of the grounded cathode triode is utilized. The cushioning effects of the two space charges are in series, no physical coupling resistances are needed, and effective isolation of input and output circuits and low input capacity also result. The reduction of capacitive loading of the input grid circuit due to the "Miller Effect" which is present in conventional triodes makes possible higher interstage coupling circuit gain.

It has been pointed out[3,6] that the use of the series connected cascode circuit results in considerable reduction of cross modulation distortion as compared with pentode circuits.

Fig. 2 shows the equivalent circuit of the "cascode" amplifier using two identical triodes. The gain of this amplifier may be derived as follows. Around the closed loop of Fig. 2 we obtain:

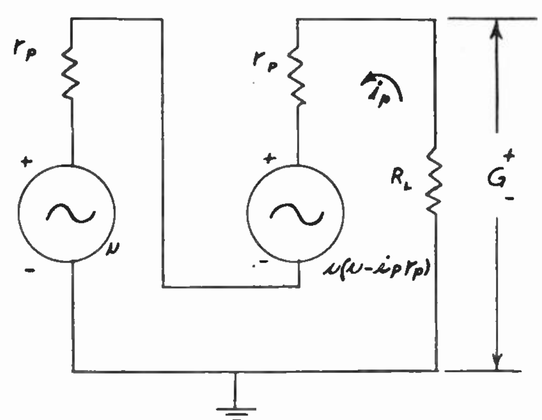

Fig. 2 – Equivalent circuit of cascode amplifier.

$$\mu + \mu\,(\mu - i_p r_p) = (2 r_p + R_L)$$

$$\text{Gain} = i_p R_L = \frac{\mu\,(1+\mu)}{1 + \frac{r_p}{R_L}\,(2+\mu)}$$

[6]R. M. Cohen, "Use of new low noise twin triode in television tuners," *RCA Rev.*, vol. 12, no. 1; March, 1951.

When analyzing this circuit, it should be remembered that since the same current flows in both triodes, the two tubes are not independent of each other but must be considered as a single stage. It has been pointed out[7] that tube noise is best represented by a constant current generator between plate and cathode.

As an experimental check on the action of the grounded grid triode in increasing the gain and decreasing the noise of the lower triode, the gain and noise figure of the arrangement of Fig. 3 was compared with the circuit of Fig. 4, all other circuit conditions being

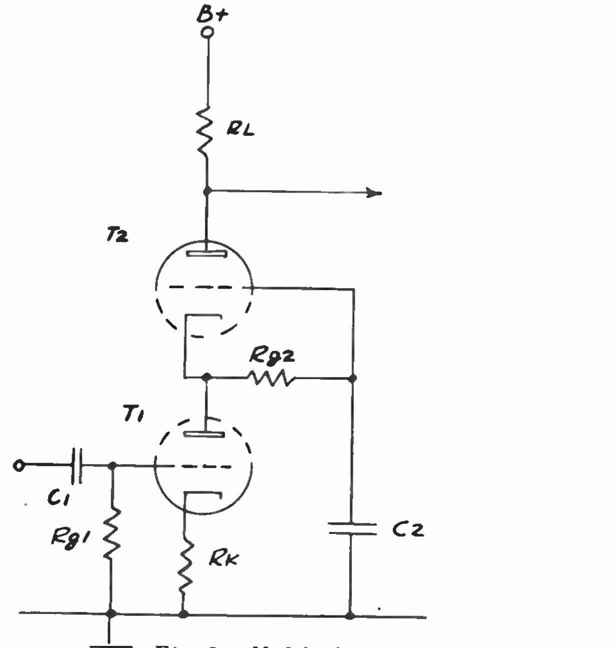

Fig. 3 – Modified cascode amplifier.

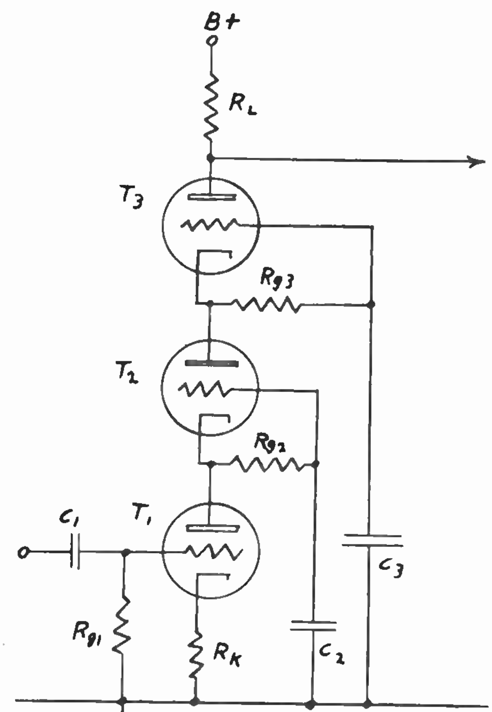

Fig. 4 – Cascode with two grounded grids.

[7]Valley and Wallman, Ibid., pp. 562-563.

equal. The addition of the third grounded grid triode in series with the other two resulted in a substantial further reduction of noise and increase in stage gain, thus demonstrating the effectiveness of the grounded grid tube.

PRACTICAL APPLICATIONS

In practical applications of the basic circuit of Fig. 1, it is soon found that due to variations in tube characteristics, the bias on the grounded grid triode must be adjustable to maintain it at a slightly negative potential with respect to its cathode. A modification which overcomes this critical adjustment is shown in Fig. 3. This modification consists of the use of grid leak or contact potential bias on the upper triode to replace the voltage divider of Fig. 1. The value of grid to cathode resistor is chosen so as to maintain the grid-cathode bias potential at approximately one volt negative. AC ground potential is maintained on g_2 by condenser C_2. With this biasing method, tube variations have very little effect on the biasing of either triode.

Figs. 5 and 6 show two possible methods for applying negative feedback to this amplifier stage. In Fig. 5, feedback is applied around one stage only. Fig. 6 shows

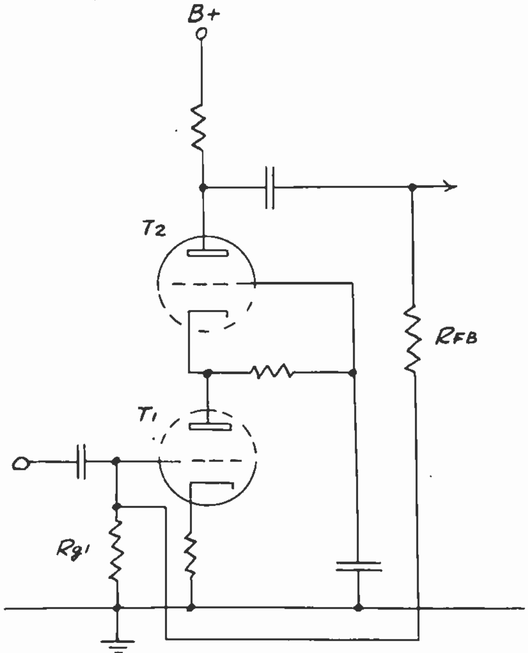

Fig. 5 – Cascode with feedback over one stage.

negative feedback applied over two stages. R_k also provides cathode bias and degeneration for the input stage. The use of negative feedback apparently has little effect on the noise figure of the amplifier. Feedback does, however, tend to reduce microphonic noise and increase the available dynamic range for a given distortion level.

It is important in the construction of a low noise amplifier of this type to use only the highest quality components and to exercise extreme care in parts location and layout so as to avoid undesired noise pickup due to leakage currents, stray magnetic and electric fields, etc. It has been the author's experience that it is advisable to use low noise resistors such as the deposited film type. Resistances composed of carbon granules usually generate noise considerably in excess of thermal agitation noise, due to fluctuation in contact resistance between adjacent granules. It is also helpful if the resistors can be kept in fairly stable thermal equilibrium by avoiding high ambient and operating temperatures.

As is the case with other types of low level amplifiers, it is advisable to operate the filaments from dc to minimize hum problems.

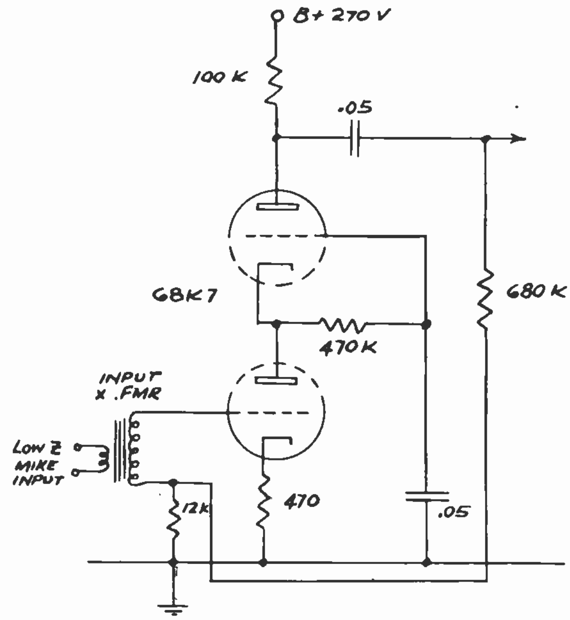

Fig. 6 – Cascode with feedback over two stages.

PERFORMANCE

In evaluating the performance of the cascode circuit at audio frequencies, comparative measurements were made between the circuit of Fig. 7 and several commercially available low noise amplifiers. Noise generator techniques were used as well as absolute level measurements and we were able to correlate results between the two methods quite closely. The noise figure measurements were made using a temperature limited diode noise generator.[8] With the "cascode" amplifier, noise figures as low as 1.5 were obtained which is 1.8 db from the thermal agitation noise in the input grid. The equivalent input noise level as obtained by dividing the output noise by the amplifier gain was found to be approximately -127 dbm over a 15 kc band which agrees quite well with the noise figure measurement. Input levels up to -20 dbm were possible without exceeding 1% total harmonic distortion.

These measurements showed an improvement in

[8] Terman and Pettit, "Electronic Measurements," 2nd Ed., McGraw-Hill Book Co., New York, pp. 372-373; 1952.

noise figure of from two to twelve db between the "cascode" and the conventional types of low noise amplifiers. The exact amount of improvement will depend on the

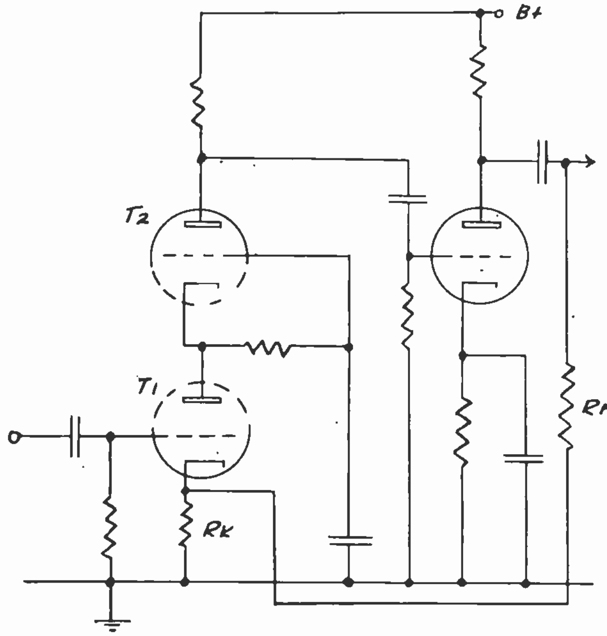

Fig. 7 – Microphone amplifier using cascode circuit.

type of tubes used and the frequency characteristic involved. Tube noise in the audio range consists of low frequency noise due to cathode "flicker," medium frequency noise due mainly to microphonics, and uniformly distributed noise due to "shot effect".

Pentodes have an additional noise known as "partition noise" which is due to the random fluctuation of the distribution or partition of cathode current between plate and screen and which generally overrides the other types of noise just mentioned. The final selection of tubes will depend to a large extent on which of these types of noise it is desired to minimize.

An additional practical consideration is the fact that with the "cascode" circuit, relatively large stage gains may be obtained with medium mu tubes which are apparently less critical with respect to microphonics than are the high mu triodes and pentodes ordinarily used in amplifier input stages. No increase in the number of circuit components and parts is required over a pentode stage and as a general rule, the double triodes which are suitable for cascode use are less expensive than are low noise pentodes.

In recent months, several new types of dual triodes specifically designed for cascode circuit applications have appeared. These types include 6BK7, 6BQ7, 6BZ7 and GL-6386. In addition to these types, good results have also been obtained using the more conventional types of dual triodes such as the 12AU7.

CONCLUSION

The work which has been done in adapting the "cascode" amplifier to audio frequency service has resulted in the successful development of a stable, low-noise, high-gain input stage, having low distortion, which requires only a minimum number of component parts and which does not require the use of expensive or especially selected tubes. An improvement in noise figure of from two to twelve db has been obtained with this circuit as compared with conventional input amplifier circuits.

The "cascode" audio amplifier is being used in the new Magnecord M-80 series of tape recorders with considerable success. Consistently lower noise has resulted than with the special low noise tubes used previously and it has not been necessary to resort to an undue amount of tube selection to maintain amplifier noise specifications in production.

The author wishes to acknowledge the valuable assistance and suggestions given by Professor A. H. Wing of Northwestern University and Mr. W. F. Boylan of Magnecord, Inc, which contributed greatly to the success of this development.

AN ALL-TRANSISTOR HEARING AID*

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The device to be described uses three *p-n-p* alloy-junction transistors. It is well known that, in the manufacture of this type of transistor, it is necessary to have pure germanium to 1 part in a billion. Producing purities of this order were unknown up to three years ago. In this state of purity, the crystal lattice structure has four valence electrons. Controlled amounts of "impurity" elements are than added to the germanium. If antimony having five valence electrons is added, the material becomes *n* type, i.e., the current flow takes place due to a movement of electrons. If gallium or indium having three valence electrons is added, the material becomes *p* type, and current flow is described as being due to a movement of positive holes.

In the *p-n-p* junction type transistor the *n* material forms the base connection and the layers of *p* material on each side provide the terminals called emitter and collector. As with vacuum tubes, there are three basic methods of operation, namely; — grounded base, grounded collector, and grounded emitter. For hearing aid application, the grounded emitter is more useful than either of the others, because greater gain is obtained, and only one battery supply is necessary.

The microphone used is a balanced magnetic armature type. It has a sensitivity of approximately 10^{-10} watts at 1000 cps when excited by a sound field of 1 microbar (74 db re.0002 dyne/cm^2). This type of microphone has an impedance of about 1000 ohms, which is an excellent match for the transistor input circuit used. Because the microphone is magnetic it is much more stable with respect to temperature and humidity than the Rochelle salt microphones which were used previously with high-impedance input circuits of the vacuum-tube type. The microphone is capacitively coupled to the base of the transistor with a 4 microfarad electrolytic condenser.

Much progress has been made recently in the design of very small and very high quality electrolytic condensers for transistor circuit applications. The particular condenser used is a tantalum type hermetically sealed in a metal can. These condensers are practically impervious to effects of temperature and humidity, at least over the range anticipated in hearing aid use.

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Many claims and counter-claims have been made regarding transistors and their ability to perform properly and uniformly. In some cases charges have been made against the transistor which have resulted from limited understanding of circuit application information. As with vacuum tubes there are many different types with different characteristics. If the circuit engineer becomes familiar enough with transistors to develop circuitry which takes full advantage of the transistor characteristics, the transistor can be controlled as carefully as vacuum tubes. It may be temperature-stabilized so that, over a range of temperature from 40° to 120°F, the change in current will be very small. Thus, over a wide range of temperatures, a stable control of noise characteristics can be achieved, plus a stable control of battery operating current and battery cost. This stabilization can also provide a control of less than 1 db change in gain over the temperature range of 40° to 120°F.

The transistor in the first stage of the hearing aid is temperature-stabilized by means of a voltage-divider network, which maintains a rather constant voltage at the base of the transistor. Further stabilization is accomplished by proper choice of the emitter resistance. The value of the resistors used in the voltage-divider network are dependent upon the characteristics of the transistor, and upon the supply voltage used.

It is necessary, for example, to make the current through the bleeder resistors large as compared with the normal base current of the transistor. There is also a critical ratio for the center point of the divider connected to the base. This ratio changes with different transistor characteristics, and also with different supply voltages.

This particular circuit was designed for Raytheon's CK.718 PNP type junction transistor. The emitter resistor value and the voltage divider ratio and value may be predetermined in design for a given supply voltage, so that it is not necessary to select transistors for the circuit. The collector circuit is transformer coupled to the next stage. The reflected primary impedance is 20,000 ohms while the secondary impedance is 1000 ohms. Particular care and attention must be given to the transformer design, so that the dc primary resistance will be suitable for the type of transistor used and for the supply voltage, in order to give the proper dc voltage at the collector.

The transformers used in the Beltone Concerto instruments are magnetically shielded in mu-metal cans in order to eliminate electrical hum pickup caused by fluorescent lights, power lines, motors and such devices. The mu-metal cans serve also as container into which the transformer is potted, so that the effects of moisture and mechanical shock are practically eliminated. The mu-metal cans even improve the efficiency and frequency response.

The problem of tone control design is quite different in a transistor circuit than in vacuum tube hearing-aid design. Between the secondary of the first transformer and the second transistor a series resistor, shunted by a capacitor, has been added to provide the tone control action. Operation with the tone-control switch in the shorted position gives the "flat" response or "full-tone" condition. When this switch is opened, the parallel resistor-capacitor network accentuates the passage of high-frequency tones and gives the effect of suppressing low tones. This noise suppressor is intended primarily to enable the user to adjust the frequency response of the instrument to his environmental needs. When the user is in a noisy location, such as a restaurant or office, or exposed to street traffic noise, he will probably prefer the number two, or suppressor, position. In this way many of the annoying background noises will be eliminated. On the other hand, when he is in a quiet environment such as in his own home he will prefer to switch to the number one position in order to get the full range of tonal quality available in the instrument.

The volume control is a 10,000 ohm potentiometer controlling the input signal to the transistor. Because impedance reflected from the output stage to the input stage has an influence on transistor circuits (that is not present with tube design), the value for the emitter resistor changes with different stages. The second stage is also temperature-stabilized by use of a proper voltage-divider network and emitter resistance. The output or collector of this second transistor is again transformer-coupled to the output stage. Control of the dc current in the power-output stage is provided by the selection of the proper base resistor for the particular supply voltage and transistor being used.

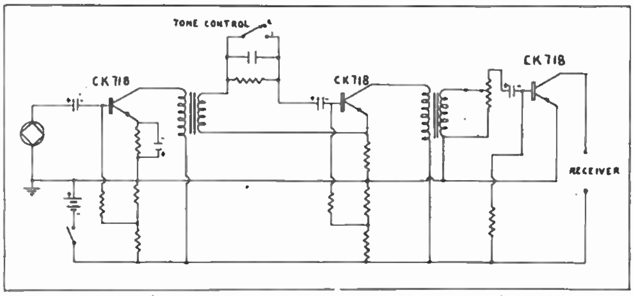

Fig. 1 – Triode tube noise vs. frequency.

The hearing aid "receiver" or earphone is magnetic. It is connected directly in series with the collector circuit of the output stage. This transducer has a nominal impedance of 1000 ohms. Again particular attention must be given to the design of the coil, so that the DC resistance of the receiver winding will be proper for the transistor and for the supply voltage used.

Great flexibility is provided in the basic circuit (See Fig. 1). More than one supply voltage may be used. When this is done, the model is given another number or color code. Reference to Fig. 2 will disclose the extremely wide range of maximum acoustic power levels available in the Concerto instrument.

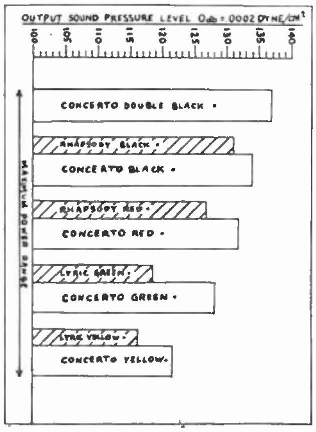

Fig. 2 – Grounded – grid equivalent circuit.

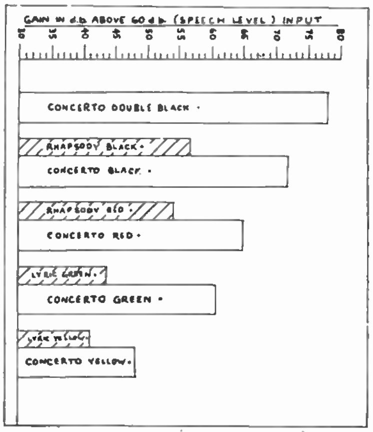

Fig. 3 – Grounded – cathode equivalent circuit.

It should be mentioned that in all our previous experience with vacuum tube hearing aids, it has been necessary to choose an instrument which has enough gain and power available, when the batteries are discharged to "end-point." The gain and power with a fresh battery was considerably more than that available when the battery had reached end-point. It has always been necessary, therefore, to choose a fitting which would have enough reserve gain and power to provide for a reasonable drop in battery voltage.

A conservative estimate of loss in acoustical power between a fully charged "B" battery, and one which has reached normal end-point of ⅔ voltage, is a loss in maximum power level of six decibels. At the same time this drop in battery voltage was responsible for a loss of at least 12 db, conservatively speaking, in gain. Fig. 2 shows the power available in the various color-code combinations of the Concerto model. Here the design has incorporated stair steps at acoustic power level starting at 122 db for the Yellow Dot, going to 128 db for the Green, 132 db for the Red, 134 db for the Black, and 137 db for the Double Black Dot. This may be compared in Fig. 2 with the power available with reasonable battery life in the Lyric-Rhapsody line of vacuum tube aids.

In Fig. 3 is shown the extremely wide range of fitting available in the various color dots. Here again, the design has purposely incorporated stair-step progressions of acoustical gains which are available and reference to Fig. 3 will show the comparable gain which was available with the Lyric-Rhapsody (Vacuum-tube) line, considering normal battery drop. The gain and power data are shown at 1000 cps for simplicity and for comparison to previous instruments for which similar data have been published. All of the gain and power data shown for the Concerto Model in Figs. 1 and 2 are based upon the use of a standard microphone and a T-3 receiver.

hearing aid to work at top efficiency with the maximum performance in gain, power, battery economy, and minimum distortion, it is necessary to design for a predetermined battery voltage. If it is desired to provide temperature stabilization, it is necessary to use a completely different stabilizing circuit at one voltage than for another. Moreover, the input and output impedances of the transistor change radically with a change in supply voltage. Completely different circuitry is required for optimum performance at different supply voltage. Resistance tolerances are held to plus or minus one-percent in the instrument. This is necessary for optimum transistor performance.

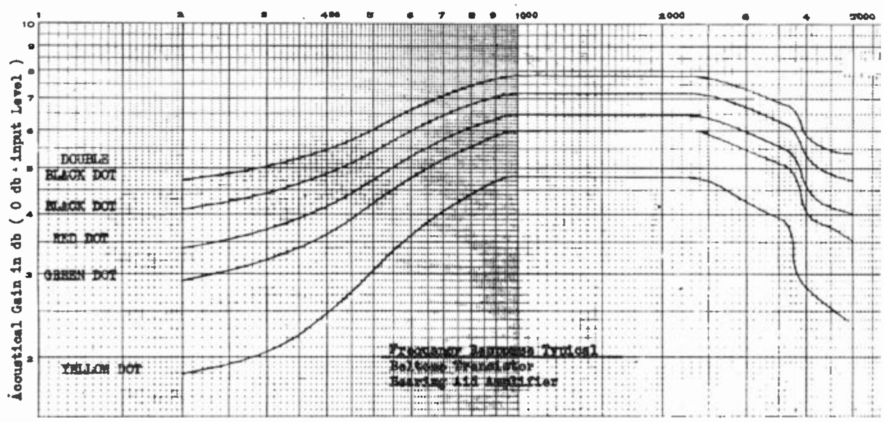

Fig. 4 — Noise vs. source resistance.

The yellow and green dot instruments have circuits which have been designed for 1.25 volt operation. This provides a low-gain, low-power instrument; and a medium-gain, medium-power instrument. The red, black, and double-black dot instruments are designed for a supply voltage of 2.5 volts. This provides a high-gain, high-power instrument; a very high gain, high power; and a special instrument for extreme (high loss, high tolerance) cases.

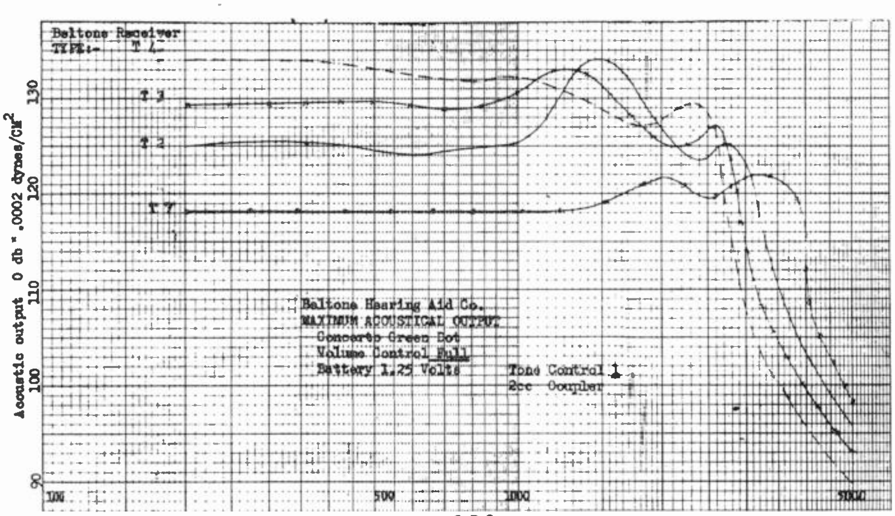

Fig. 5 — Typical two stage amplifier.

Additional flexibility is available in the maximum output, as well as the frequency-response, by selection of one of several receivers (earphones), each with different characteristics. Many factors influence the circuit design of a transistor hearing aid. In order for a transistor

The transistor completely eliminates the necessity for a "B" battery. This ends the constant expense of buying "B" batteries and the nuisance of changing them frequently. Battery costs for the Concerto model, used by a person with average hearing loss, are as low as $2.00 for a full year. Because most users of vacuum-tube aids with average losses are accustomed to paying $40.00 to

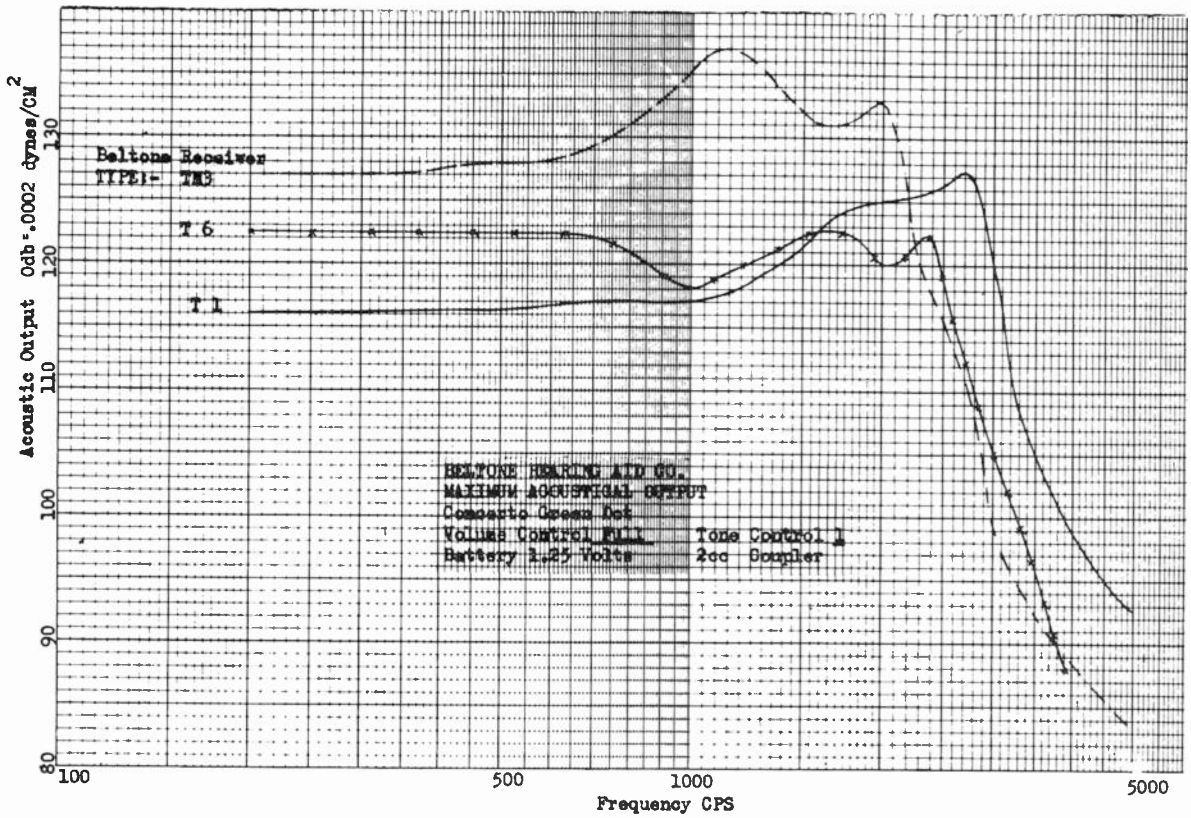

Fig. 6 – Equivalent circuit of two stage amplifier.

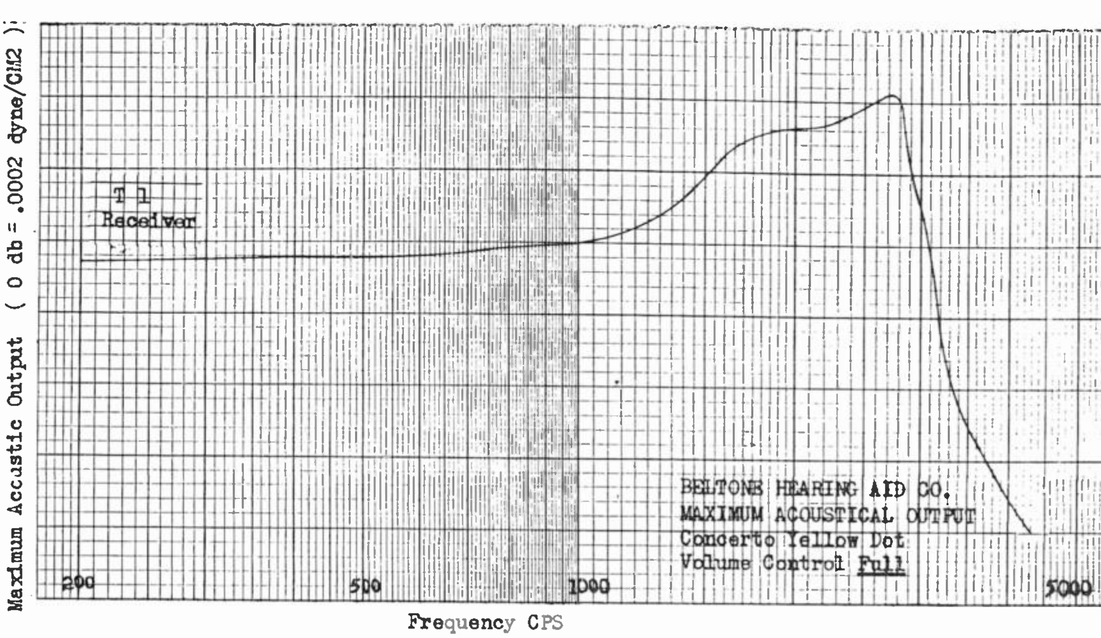

Fig. 7.

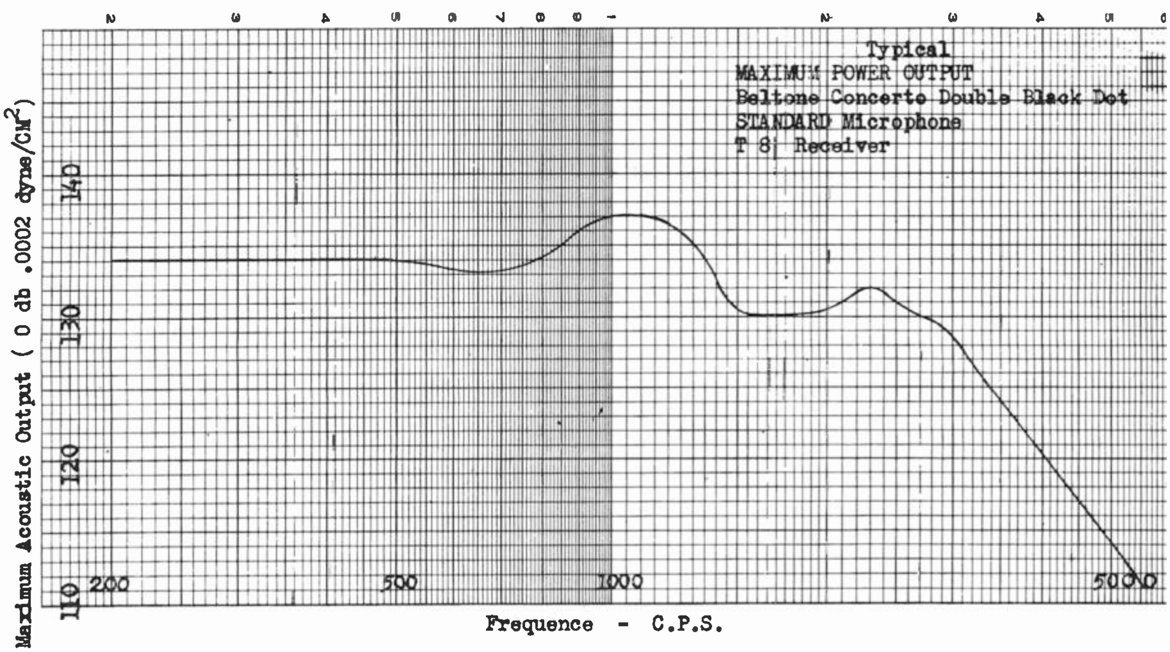
Typical
MAXIMUM POWER OUTPUT
Beltone Concerto Double Black Dot
STANDARD Microphone
T 8 Receiver
Maximum Acoustic Output (0 db .0002 dyne/CM²)
140
130
120
110
200
500
1000
5000
Frequence - C.P.S.

Fig. 8.

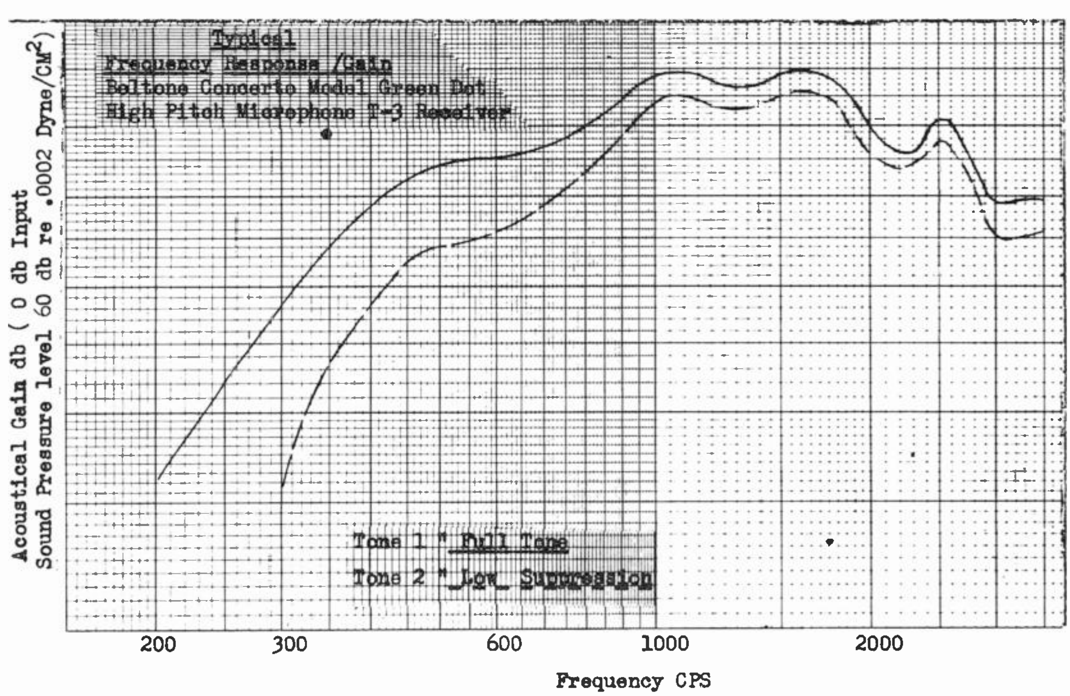
Typical
Frequency Response /Gain
Beltone Concerto Model Green Dot
High Pitch Microphone T-3 Receiver
Tone 1 " Full Tone
Tone 2 " Low Suppression
Acoustical Gain db (0 db Input
Sound Pressure level 60 db re .0002 Dyne/CM²)
200
300
600
1000
2000
Frequency CPS

Fig. 9.

$60.00 a year for batteries, this tremendous saving is one of the transistor's most powerful appeals. Even people with very severe losses can be helped to hear for less than one-cent an hour.

The instrument uses a mercury type battery which has a practically constant voltage discharge curve. This gives the instrument uniform gain and power throughout the life of the battery. The voltage of the mercury battery, when it is completely discharged, drops off very rapidly (in a matter of a few minutes).

The total current drain of the 1.25 volt circuit is approximately 1.2 milliamperes. This gives more than seven-hundred hours of life from a battery which has a retail cost of about 30¢. In the case of the very high power instrument where the 2.5 volt battery is used, the total current drain is approximately 2.4 milliamperes, and this gives an average of 160 hours life from a battery costing 60¢ retail.

Reference to Fig. 4 will show the relative frequency response and gain of the various Concerto instruments using the standard microphone and T-3 receiver. Note the wide frequency range and the stair-step levels of sensitivity starting at 48 decibel gain for the Yellow Dot, and increasing to 78 decibel gain for the Double Black Dot.

The Concerto model instrument has the microphone mounted on the front cover. This simplifies replacement in the field. The Concerto model is available with four different front combinations:

1. Standard Front—This permits the use of any one of three microphones. It includes the tone control, but it does not have the Phone Clarifier nor the Fashioner. Note: The telephone clarifier is particularly applicable to transistor circuitry because the coil used for telephone pickup matches into the transistor circuit with a much greater efficiency than was possible when a coil of this type was matched into a vacuum tube circuit.

2. A front cover is available which has both the microphone and the Phone Clarifier.

3. A third front cover is available which uses the Fashionear only. It does not have the Phone Clarifier, but has an internal microphone and the Fashionear microphone socket, so that the Fashionear may be plugged in. When the cord is removed from the front cover, the internal microphone is reconnected.

(The same microphone is used in the Fashionear as is mounted internally in the instrument. Thus the same performance, gain and frequency response are available, whether the internal microphone or the Fashionear microphone is used.)

Figs. 5 and 6 show the wide range of flexibility in the fitting, when the various receiver types are used. All of the curves shown in Figs. 5 and 6 are Green Dot Concerto with a "standard" microphone.

Fig. 7 shows the frequency-response and maximum power of the Yellow Dot Concerto using the midget T-1 receiver. Fig. 8 shows the power output curve of the Double Black Dot with a T-8 receiver.

Fig. 9 is the frequency-response of the Green Dot instrument with a "high-pitch" microphone and a T-3 receiver.

Experimental work is in progress which indicates that eventually grown-junction *n-p-n* silicon transistors will be feasible for use in hearing-aid applications. This may well lead to completely new circuit approaches, and perhaps to a combination of *p-n-p* and *n-p-n* in complementary circuits.

INSTITUTIONAL LISTINGS (Continued)

KLIPSCH AND ASSOCIATES, Box 64, Hope, Arkansas
Wide Range Corner Horn Loudspeaker Systems, Corner Horns

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

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

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

SHALLCROSS MANUFACTURING COMPANY, 524 Pusey Avenue, Collingdale, Pennsylvania
Audio Attenuators, Precision Resistors, Rotary Switches, Test Instruments

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

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

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

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

Charge for listing in six consecutive issues of the TRANSACTIONS—$25.00. Application for listing may be made to the Chairman of the IRE-PGA Ways and Means Committee, Prof. O. W. Muckenhirn, Department of Electrical Engineering, University of Minnesota, Minneapolis 14, Minn.

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.

(Please see inside back cover for additional names)

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

May-June, 1954

Published Bi-Monthly

Volume AU-2 Number 3

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The Institute of Radio Engineers

IRE PROFESSIONAL GROUP ON AUDIO

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

Annual Assessment: $2.00

TRANSACTIONS of the I • R • E®
Professional Group on Audio

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CHAIRMAN'S REPORT 1953-54
IRE PROFESSIONAL GROUP ON AUDIO

Marvin Camras
Armour Research Foundation
Chicago 16, Illinois

In 1953-54 PGA reached new levels in all its activities. Five percent of all IRE members now belong to the PGA. By the end of 1953 our paid membership increased to 2147, as compared to 1681 the year before. Our treasury balance for the same period went up even more rapidly: from $4445 to $7035. New chapters were organized in San Diego, Phoenix and Cleveland, giving us a total of thirteen chapters.

Transactions of the PGA has continually improved under the editorship of Dan Martin. *Transactions* is by far the most important activity of our national organization. It ties together the local chapters, and the members at large. It gives comprehensive coverage of audio technology and rapid publication of original work.

In 1953, Bob Troxel of our Chapters Committee and Alf Wiggins of our Membership Committee took inventory of members' affiliations and interest. The results (published more completely in the November-December, 1953 *Transactions*) showed that only about 5% of our group were members of the *Acoustical Society of America*, and only about 3% belonged to the *Audio Engineering Society*. The fields of main interest were "High Fidelity," "Design," and "Amplifiers." Our policies in regards to programs and publications will be influenced by these facts.

The year was an active one for Tapescripts. Andy Jacobsen made available about a half dozen new ones, to give a selection from about sixteen programs. Tape recordings by the original author plus lantern slides give a realistic presentation of important audio papers that are not otherwise available. Tapescripts may be obtained free of charge by any IRE chapter of section.

A very successful convention program was organized by Win Kock, Chairman of the Program Committee. It consisted of a High Fidelity session, a General Audio session, and a seminar on High Fidelity by outstanding authorities in the field.

Finances were studied by Ben Bauer's committee, and it appears that we will be able to expand our program considerably in '54-'55 without exceeding our budget and with no change in annual assessments.

The Awards Committee headed by John Hilliard proposed awards as follows:

(1) For an outstanding development in audio.
(2) For continued contributions to audio.
(3) For an outstanding paper in the field of audio.

Awards will be made annually, or less often at the discretion of the committee.

The Administrative Committee proposed changes in the PGA Constitution and By-laws to facilitate nominations, and also to allow a wider choice of candidates for Chairman and Vice-chairman when holdover members decline to run for these offices. The changes are now in the process of approval and ratification.

During 1954-55 the national IRE-PGA plans to expand communications among local chapters by circulating a regular bulletin describing activities, speakers, etc. that are part of different chapters' programs. These bulletins will be a source of ideas for local program committees.

As a long term project our secretary is compiling a "Manual of Procedure" for PGA committee members. The "Ways and Means Committee" procedure has been finished. The manual will preserve continuity of committee work, and will ease the expediting tasks of the chairman.

We will also try to have our elections and tenure of office such that the newly elected officers can be at the annua' Administrative Committee meeting in March, and can take over the latter part of the meeting. In this way we can have a good start on the program for the coming year.

I'd like to express my appreciation to PGA committee members for their excellent cooperation. Our editor, Dan Martin, and our secretary, Ben Bauer, deserve special commendation for the many hours they have devoted and continue to devote to PGA.

TREASURER'S REPORT 1953-54
IRE PROFESSIONAL GROUP ON AUDIO

B. B. Bauer
Shure Brothers, Inc.
Chicago, Illinois

At the present time, the PGA is in excellent financial shape, with a cash balance of $7,035.68 as of December 31, 1953. The financial growth of PGA can be seen by reviewing the balance as periodically reported by Headquarters.

Date	Balance
March 30, 1952	$1,517.95
June 30, 1952	2,436.27
September 30, 1952	2,972.12
December 31, 1952	4,445.02
March 31, 1953	6,482.84
June 30, 1953	6,701.26
September 30, 1953	6,344.76
December 31, 1953	7,035.68

During the above period PGA membership has been rapidly increasing, and at the present time it stands at approximately 2100 paid-up members, or about 5% of the Institute membership. While our records are incomplete, there is evidence that the rate of growth has diminished. We assume that PGA membership will continue to expand at a somewhat slower rate than heretofore. The rate of expansion will depend on PGA activity, quality of publication, services provided to chapters, and similar factors. A careful and sound financial policy is essential.

Since the income of PGA consists mainly of assessments, matched funds and institutional listings, all of which constitute prepaid funds, the financial policy of PGA must continue to be sound and conservative. Therefore, the Administrative Committee has decided to continue a $2.00 annual assessment throughout the fiscal year, beginning with April 1, 1954. This will insure financial stability and fulfillment ot our commitments.

1953 - 1954 REPORT OF THE IRE-PGA EDITORIAL COMMITTEE

Daniel W. Martin
The Baldwin Piano Company
Cincinnati 2, Ohio

Last April the Editorial Committee was reorganized and enlarged to six members in accordance with the By-laws of IRE-PGA. An attempt has been made to decentralize some of the editorial functions, such as procurement of technical papers, technical editorials and chapter news, the editing of papers in specialized subjects, and preparation of institutional listings. The attempt has been partially successful, and gives promise of greater success in the coming year. Our publication continues to grow, making further decentralization of the editorial work imperative.

In six bimonthly issues of TRANSACTIONS of the IRE-PGA (including March-April, 1954) and the AUDIO section of the 1953 CONVENTION RECORD, the Professional Group on Audio has contributed twenty-nine technical papers and technical editorials, plus twenty-four smaller items including technical news, chapter news reports, and editorials. In addition, Cumulative Indices arranged by chronology, by author, and by subject classification, were prepared for all PGA publications through 1953. It is planned to include cumulative indices for the year in each November-December issue in the future.

An improved format was first used in the July-August 1953 issue. The combination of neater appearance and more efficient usage of page space seems to be universally appreciated. At the suggestion of the editorial offices of IRE the committee devised a PGA symbol for use on the cover of TRANSACTIONS of the IRE-PGA. The matter was referred to the Committee on Professional Groups which, it is reported, has taken negative action on such symbols.

In connection with the questionnaire sent to the membership of PGA, an analysis was made of reader interest in TRANSACTIONS. This was published in the November-December 1953 issue.

Because of the length of the publication cycle, the news in TRANSACTIONS is sometimes delayed. This has probably been the chief disadvantage of combining the NEWSLETTER with the TRANSACTIONS. The committee will attempt to work out an arrangement with Mr. Gannett, managing editor of IRE, on insertion of late news and reduction of the overall cycle on publication.

Currently the flow of technical papers is just sufficient. A greater flow would be welcomed, especially on the engineering and technical aspects of high-fidelity systems and components. The policy of prompt publication of technical papers has continued. The absence of a tangible backlog, resulting from this policy, in combination with the usual summer lull and the absorption of 1953 convention papers by the CONVENTION RECORD, created a problem in mid-summer 1953. Since then attempts were made to obtain agreement on converting the Audio section of the 1954 CONVENTION RECORD into a May-June issue of TRANSACTIONS of the IRE-PGA. This is not a problem with other professional groups which publish less frequently and, in some cases, rather irregularly. Consequently none of the proposals considered was adopted. There will be a regular May-June issue as before. Whether a summer lull occurs again in 1954 will depend in part upon the percentage of audio convention papers which meet the CONVENTION RECORD deadline. (Three of the 1954 convention papers had already been promised to TRANSACTIONS before the 1954 IRE convention program was solicited.)

The new editor of IRE, Dr. Pierce, has requested abstracts of all TRANSACTIONS papers, for publication in PROCEEDINGS of the IRE. PGA will put this practice into effect with the May-June issue of TRANSACTIONS.

Many excellent technical papers have been received during the past year. Procurement efforts related to papers presented at the Department of Defense Symposium on Magnetic Recording were particularly successful. However, very few unsolicited papers are submitted by the PGA membership. Procurement of additional papers is the major 1954 goal of the TRANSACTIONS of the IRE-PGA.

The chairman wishes to thank each member of the committee for excellent cooperation during the past year.

PGA SESSIONS AT IRE NATIONAL CONVENTION

The three Audio sessions on March 23 at the 1954 IRE National Convention in New York City, were attended by capacity audiences. The two daytime sessions were held in the Jade Room of the Waldorf-Astoria Hotel and the Audio Seminar — "High Fidelity in Audio Engineering" took place in Marconi Hall, Kingsbridge Armory. All of these sessions were organized by the Professional Group on Audio. PGA is grateful to the Program Committee, Dr. Winston E. Kock, Chairman, for excellent planning and followup.

The morning session included three demonstrations. A large-area condenser microphone suspended over the speaker was used to good advantage as a public-address microphone throughout the morning sessions. Although its physical resemblance to a shower bath (pointed out by one of the authors of the paper) was the subject of levity, the effectiveness of the microphone directivity was quite impressive. In a second demonstration, recordings of piano and organ tones under various conditions of reverberation were played in an environment which, fortunately, was made highly anechoic by the acoustically absorptive influence of such a large number of audio engineers. The third demonstration employed the Haas effect. In addition to demonstrating the effect it was also possible to mystify a large portion of the audience by causing a miniature loudspeaker to give the aural illusion of copious bass response.

Extended question periods after the technical papers were evidence of the high interest created throughout the day. The authors should be commended for both the timeliness of their remarks and for good presentation.

Before the Audio Seminar in the evening, a brief annual business meeting of the Professional Group on Audio was held. Chairman Camras announced the results of the recent election as follows: Chairman — Vincent Salmon, Stanford Research Institute, Stanford, California; Vice-Chairman — M. S. Corrington, RCA Victor Division, Camden, N.J.

New members of the Administrative Committee: (3 year term) F. H. Slaymaker, Stromberg-Carlson Co., Rochester, New York; D. W. Martin, The Baldwin Piano Company, Cincinnati 2, Ohio, and W. D. Goodale, Jr., Bell Telephone Laboratories, Murray Hill, N.J. Thanks for their recent efforts on behalf of IRE-PGA was extended to the retiring members of the Administrative Committee who are the following: J. J. Baruch, A. M. Wiggins, and E. Uecke. Reports by the Chairman, Secretary-Treasurer, and Editor followed.

The Audio Seminar brought together an eminent group of veterans in the audio field whose work contributed greatly to "high-fidelity" long before the use of the term became widespread. The brief formal summaries given by each member of the panel are expected to appear along with the other audio convention papers in the CONVENTION RECORD. (Certainly if any of these papers miss the final deadline of the CONVENTION RECORD every effort will be made to obtain them for subsequent publication in TRANSACTIONS.) Unfortunately those readers who were absent from the meeting will not be able to enjoy the unrehearsed question and answer session and the humorous repartee of Chairman Knowles. Questions from the audience concerned such subjects as the relative merits of omni-directional and directional microphones for high quality sound pickup, of domestic and foreign condenser microphones, of direct-radiators and corner-horns for home installations of high fidelity equipment, and of the low-frequency response of loudspeaker enclosures of various sizes. Discussion was proceeding at maximum pitch when the session was ended by the curfew of the building attendants.

PGA BRIEFS

Vincent Salmon
Chairman IRE-PGA

Thanks to Marvin Camras from IRE-PGA for a fine job of chairmanship during the past year.

Congratulations to Dr. Vincent Salmon, the new Chairman of IRE-PGA. His PGA experience as Vice-Chairman and on various committees provides excellent background for his new responsibility.

The Chicago Chapter of IRE-PGA had a joint meeting with the Broadcast and Television Chapter on February 19th. John Clark of EL-RAD Manufacturing Company, Chicago, Illinois, spoke on the subject, "High Fidelity." A meeting in April, held jointly with the Circuit Theory Chapter, had as speaker B. B. Bauer, Shure Brothers, Inc., and Secretary-Treasurer of IRE-PGA. The following summary of Mr. Bauer's paper, "Equivalent Circuit Analysis," was provided by the Chapter reporter, Walter Coulter:

> "The conceptual basis underlying equivalent circuit analysis dates back to the time of Volta, Ohm and Ampere. Mathematical analysis of a-c circuits in terms of mechanical equations has been expounded by Rayleigh in the latter half of the 19th century. Over two decades ago it was shown that a dual system of analogies exists, which–while differing from the classical concept–does possess advantages in some instances. Current concepts of equivalence between mechanical, acoustical, and electrical circuits include the definition of "terminals," the similarity between the equations of d'Alembert and Kirchoff and the use of ideal transformer couplers for points of connection."

Mr. John K. Hilliard of Altec Lansing Corp. and a member of our Editorial Committee, is recruiting audio papers for the Western Electronic Show and Convention (WESCON) at Los Angeles, August 25-27. Authors should contact him as soon as possible.

Approximately 40% of all the PGA members are located in areas already served by a PGA Chapter. An additional number of over 40% is distributed among areas which are potential Chapter locations. Thus over four-fifths of the PGA membership could have the benefit of local PGA activities. Although the average size of a PGA Chapter at the end of 1953 was 60 members, the median size was 31 members. Already this information is somewhat out of date, because the Cleveland Chapter which reported a membership of 21 in December, 1953, has expanded at a rapid rate during the last few months. The membership of PGA Chapters in percentage of total Section membership, ranges from 6% to 13%.

Mr. Philip B. Williams, chief engineer of Jensen Manufacturing Company has been appointed chairman of the IRE-PGA program committee for the coming year.

Dr. Pierce, new Editor of the IRE, has requested author abstracts on all technical papers published in TRANSACTIONS. The abstracts will be published in PROCEEDINGS of the IRE. Abstracts from TRANSACTIONS of the IRE-PGA will be supplied for this purpose, starting with this issue.

Our new IRE-PGA Chairman begins his administration with the following Editorial Note to the membership:

> "Under its previous administrations, the IRE-PGA has evolved from a lusty infant into an adventurous adolescent. It now shows some signs of an asymptote in its current growth, but there is no let-up in the vigor of its development. My principal job will be to help guide and integrate activities which have already been gotten under way."

In his report as Retiring Chairman, Marvin Camras has noted some of these activities. To handle all of them properly, we shall need more volunteers who are willing to contribute their time and technical competence. Such volunteers are the backbone of any technical group. I hope that Committee Chairman (to be announced in the July-August TRANSACTIONS) will find plenty of willing hands to help them.

One way in which we all can help is in preparing or procuring for publication additional audio papers of professional caliber. One easy way to do this from an informal oral paper is to have it taped, transcribed and corrected. In fact, development of a procedure for accomplishing this would be useful to all other Groups. Let's have some comments on this idea.

CLEVELAND CHAPTER HAS STEREO SOUND SYMPOSIUM

Albert Preisman
Chapter News Editor

Mr. Herbert H. Heller, Chairman of the Cleveland, Ohio Chapter, sends a glowing report of the activities in this new IRE-PGA Chapter. In January 1954, the Cleveland Chapter was officially recognized with a membership of 40 members. At this time, as we go to press, the Cleveland Chapter is the fifth largest in the country with nearly 100 members and applicants. For their next two meetings they plan to have Norman Pickering, designer of a well-known pickup, and John Nigro, designer of a complex audio amplifier (3-channel stereo-dynamic with over 20 tubes) for home reproduction. Their tentative plans for the fall include a public-reaction research program with 1 to 7 channel sound production using Cinerama Sound track and a colored light display similar to a color organ.

The following is a brief synopsis of the Cleveland Chapter's March 18th Symposium, called the STEREO SOUND SYMPOSIUM which was presented to a capacity audience in Tomlinson Hall. More than 200 came who had to be turned away, as every available chair was taken and there was not even S.R.O!

1. "'The Physics of Music and Hearing" by Dr. Robert L. Hanson, of the Bell Telephone Laboratories started the meeting. Dr. Hanson said, "Recent advances in the capabilities of electronic equipment and in sound recording techniques have caused wide interest in 'High Fidelity' systems. The extent to which such systems produce 'High Faithfulness' for the listener depends upon other factors, such as the acoustical characteristics of the recording studio and listening room, the number, types and positions of the studio microphones, hearing acuity of the listener, ambient noise conditions, etc." These and other factors were discussed by Dr. Hanson, with special emphasis on their bearing on monaural, binaural, and stereo sound reproduction.

2. "Recording and Reproducing Problems of Stereo Sound," the second paper, by Mr. Rulon Biddulph, touched on the famous papers written by Bell Labs scientists, which to date form the most comprehensive foundation of our knowledge on stereophonic sound. (Since the publication of these papers, Mr. Biddulph has continued his work in this field, as well as psycho-acoustics, visible speech techniques and early magnetic recording.)

3. Pioneer, Progress and Custom Classics exhibited a comprehensive line of binaural receiving and playback equipment on TV and at the symposium.

4. The demonstrations included startling recordings and sound effects. Station WDOK prepared a demonstration tape from excerpts from the A-V Tape Libraries' new binaural catalog and from Cook records. The station's Ampex stereo tape recorder, which was flown in just in time for the symposium, brought out the best in stereophonic techniques.

5. Stan Anderson of the Cleveland Press reported, "The Cleveland Chapter of the IRE-PGA demonstrated binaural, or stereophonic sound over WCEL, WHK, and WERE during the weekend of their Symposium. The stations used TV, AM, and FM facilities. Music came through as if we were sitting in a down-front row at Severance Hall, swelling realistically throughout the room. The demonstrations were preceded by questions about the technical aspects of binaural sound and why the system has advantages over the current hi-fi reproduction. Instrumental in the planning of this Symposium are Dr. S. J. Begun of the Clevite-Brush Development, Herbert H. Heller of Bird Electronic Corp., Ralph H. Delany, WHK's chief engineer, and Harold Brinkman, chief engineer of WXEL.

The Cleveland Press and other publications gave wide-spread attention to the PGA's first Symposium, and all concurred in the report that this was THE binaural demonstration, hi-fi event of the year. Thirteen selections from five tapes, all made by Audiosphere in Italy by the Florence May Festival Orchestra of 97 pieces were played over WERE, probably for the first time over the air. The station has since broadcast several experimental programs made on a Magnecorder at the Cleveland Play House and the famous Karamu Theater. Station WDOK plans a continuing series of binaural broadcasts to start in May with the PGA chapter cooperation. In addition to these activities, several other stations commenced live binaural and monaural Hi-Fi programming in the wake of the PGA's Stereo Sound Week.

Dr. Begun and Mr. Heller expressed their gratitude to the many individuals and organizations who helped to make Cleveland's first Sound Symposium such an outstanding success. Some of those who gave much of their time and effort are: Bill Piwonka, Herb Farr and Carlton Paul of Pioneer, Ray Dehn of Custom Classics, Bob Waldo and Charlie Friedman of Progress—all for equipment; Harold Brinkman of WXEL, Ralph DeLany of WHK and Ed Stevens of WERE for airtime and engineering staff; Cook Labs, Audiosphere and A-V Tape Libraries, and Ken Hamann of Cleveland Recording-WDOK for original material.

TECHNICAL COMMITTEE 19
SOUND RECORDING AND REPRODUCING

Murlan S. Corrington
RCA Victor Division
Camden, New Jersey

The Sound Recording and Reproducing Committee of the IRE met at Headquarters on November 20, 1953 and on January 29, 1954. The Standards on "Methods for Determining Flutter Content, 1953" were approved and have now been published in the *IRE Proceedings*, vol. 42, pp. 537-541; March, 1954. The Annual Review reports were completed by Mr. Sherman Fairchild and have been submitted for publication in the April 1954 issue of the *Proceedings*.

A preliminary copy of the "Proposed Standards on Intermodulation Distortion: Definitions and Procedures for Measurement" was submitted by Dr. A. Peterson, and is now being revised.

Mr. Lincoln Thompson, Chairman of the Subcommittee on Mechanical Reproducing, reports that progress is being made on measurement procedures for disk frequency records.

The Comité Consultivo Internacional de Radio Comunicaciones, (C.C.I.R.), Task Group, which is setting international standards, reported that the disk recording characteristics presently recommended by C.C.I.R. and the present Standards of the American disk recording industry are in essential agreement except for differences in the time constants of the middle and upper audio frequency regions. These differences result in about 3 db less high frequency tip-up for the C.C.I.R. method.

A similar condition exists with reference to the magnetic recording characteristics. The time constant for the playback system for use with the C.C.I.R. characteristic is 35 microseconds, while that for The National Association of Radio and Television Broadcasters, (N.A.R.T.B.), characteristic is 50 microseconds.

Since it is desirable that these standards be the same, an attempt will be made to reconcile the differences.

VISIBLE SPEECH – ROTARY FIELD COORDINATE – CONVERSION ANALYSER*

Friedrich Vilbig
Air Force Cambridge Research Center
Cambridge, Massachusetts

For several years there has been active interest in research on visual recognition means for elements of speech communication. This paper describes a method devised for studies in this new research field. – Editorial Committee.

SUMMARY – A process is discussed, in which a rotary field is generated by the speech frequency spectrum. The rotary field deflects the electron current of an oscilloscope, so that the typical polar coordinate pictures appear on the screen. These are substantially independent of fundamental frequency. Successive pictures can be produced by conversion of these polar coordinate pictures into cartesian coordinate pictures. In order to accomplish this one employs a disc with a radial slit, rotating at a speed ν, and a glass prism, rotating at a speed of $\nu/2$. Selected parts of the screen picture can be traced through regulation of the intensity, e.g. by a selected range of the frequency spectrum. The cartesian pictures are thus also simplified, so that a "reading" of speech symbols becomes possible. In addition to its use for speech analysis, this process may also be employed to produce polar and cartesian pictures of brain waves, etc. Moreover, polar coordinate maps can be converted into cartesian coordinates and vice versa.

PRINCIPLE OF THE ROTARY FIELD ANALYSIS

If we charge the two vertical deflection plates of a cathode-ray oscilloscope with a voltage $a = A \sin \omega t$ and the two horizontal with an equal voltage $b = A \cos \omega t$ which has the same frequency and amplitude but a phase shift of 90°, then the electron beam is influenced by the electric rotary field, generated in the area of the deflection plates and describes a circular trace on the screen of the tube. We must assume here, that the horizontal (k_{hor}) and the vertical (k_{vert}) sensitivity of deflection of the two pairs of plates are equal; therefore, that $k_{hor} = k_{vert} = k$. The deflection from the center point, i.e., the radius of the circular trace (Fig. 1) is

$$r = \sqrt{(kA \sin \omega t)^2 + (kA \cos \omega t)^2} = kA.$$

Since

$$\tan \Phi = \frac{a}{b} = \frac{kA \sin \omega t}{kA \cos \omega t} = \tan \omega t$$

the momentary center angle is

$$\Phi = \omega t.$$

* Manuscript received January 29, 1954.

The electric rotary field may now consist of several frequencies instead of a single frequency. The corresponding frequencies on the two pairs of plates are equal in amplitude, merely undergoing a phase shift of 90°. The deflection voltages are then:

$$a = A_1 \sin\omega_1 t + A_2 \sin\omega_2 t + A_3 \sin\omega_3 t + \cdots = \sum_i A_i \sin\omega_i t$$

$$b = A_1 \cos\omega_1 t + A_2 \cos\omega_2 t + A_3 \cos\omega_3 t + \cdots = \sum_i A_i \cos\omega t$$

and thus the momentary deflection of the curve traced on the screen is

$$r = k\sqrt{\left(\sum_i A_i \sin\omega_i t\right)^2 + \left(\sum_i A_i \cos\omega_i t\right)^2}.$$

$$= k\sqrt{(A_1 \sin\omega_1 t + A_2 \sin\omega_2 t + A_3 \sin\omega_3 t + \cdots)^2 + (A_1 \cos\omega_1 t + A_2 \cos\omega_2 t + A_3 \cos\omega_3 t \cdots)^2}.$$

$$= k\sqrt{[A_1^2 + A_2^2 + A_3^2 + \cdots] + 2\{[A_1A_2 \cos(\omega_1 - \omega_2)t + A_1A_3 \cos(\omega_1 - \omega_3)t + \cdots] + [A_2A_3 \cos(\omega_2 - \omega_3)t + \cdots] + \cdots}$$

The momentary center angle is

$$\phi = \text{arc tan}\frac{A_1 \sin\omega_1 t + A_2 \sin\omega_2 t + \cdots}{A_1 \cos\omega t + A_2 \cos\omega_2 t + \cdots}$$

$$= \text{arc tan}\frac{\Sigma (A_i \sin\omega_i t)}{\sum_1 (A_i \cos\omega_i t)}.$$

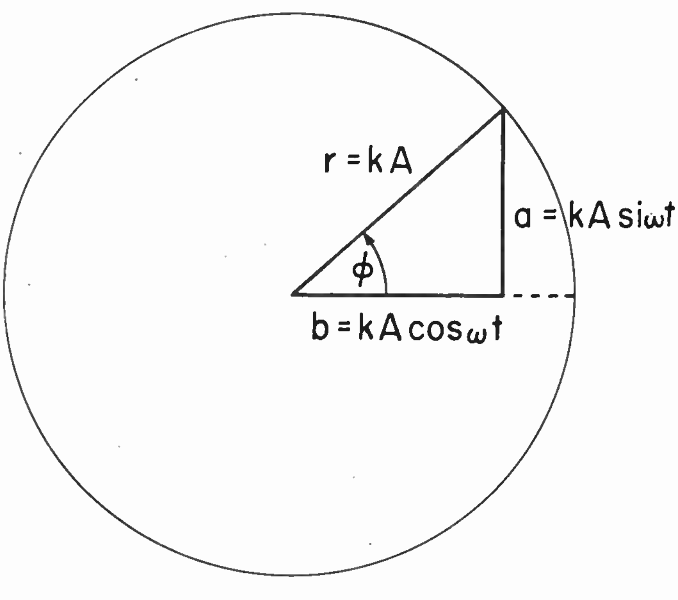


Fig. 1 – Center deflection of the electron beam of an oscilloscope in the case of a simple rotary field deflection.

The screen picture will then remain still only if the frequencies ω_1, ω_2, ω_3 ··· are harmonically related. Fig. 2 gives screen reproductions for single harmonics 2ω, 3ω or 4ω respectively, which are present in addition to the fundamental frequency ω. The pictures remain very

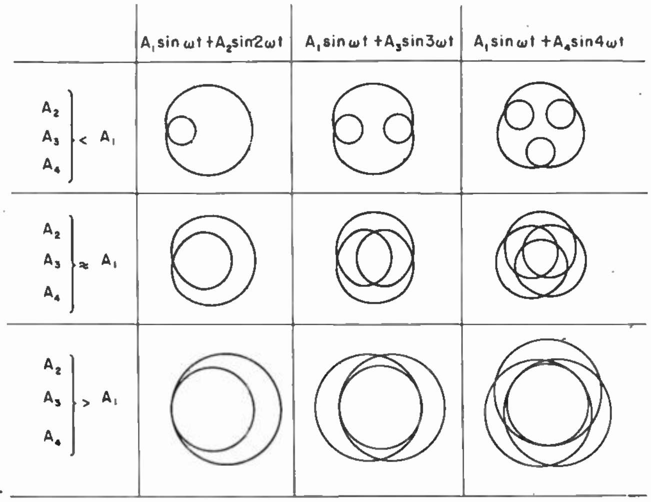


Fig. 2 – Polar coordinate pictures with rotary field deflection by a frequency spectrum, consisting of one fundamental frequency and a 2nd, 3rd and 4th harmonics respectively.

clear, as long as the amplitude of the harmonic is small compared to the amplitude of the fundamental frequency (line 1). The amount of small loops = $n - 1$, if n is the number of the harmonic. The pictures become more complicated with an increase in the amplitude of the harmonic (line 2 and 3). These complications are even more pronounced, if the number of harmonics, simultaneously present, is increased. For every given harmonic pattern e.g., for different vowels, we always obtain a typical picture which is independent of fundamental frequency. In other words, we obtain very similar pictures, e.g., for the same vowel, etc., though spoken by different people.

GENERATION OF A ROTARY FIELD BY A FREQUENCY SPECTRUM

Of particular importance in the generation of such polar coordinate pictures is the exact formation of a rotary field by 90° phase shift of every single frequency of the spectrum. For this case, the respective amplitudes may *not* be changed. Four-pole circuits offer only an approximate solution of this problem, wherein the difficulties increase with the width of the frequency band. However, the problem can be accurately solved for a wide frequency range, using a method[1] developed by the author for a single side-band modulation circuit. For example, a frequency spectrum $\sum_i A_i \sin\omega_i t$ is modulated in a push-

[1] F. Vilbig, "Experimentelle Untersuchung der Verschiebung eines theoretisch beliebig grossen Frequenzbandes um einen bestimmten Phasenwinkel," TFT 27 Heft 12 S. 560; 1938.

pull or in a ring modulator RM (Fig. 4) with a carrier frequency $C \sin at$, where the carrier frequency is cancelled out in the modulator output. The upper side-band

$$-\sum_i b_i \cos (a + \omega_i)t$$

is now transmitted through a high-pass filter HP and fed to demodulators DM I and DM II. The amplitudes of the upper side-band frequencies are b_i. At the input of amplifier V, we have carrier frequency a. At its output, two component voltages $c \sin at$ and $c \cos at$, with equal amplitudes but with a phase difference of 90°, are tapped off at a circuit consisting of a resistor R and a capacitance C and led to demodulators DM I and DM II.

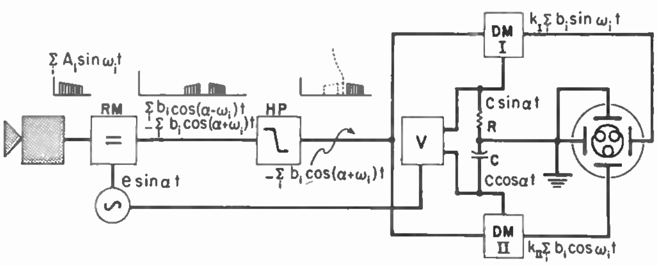


Fig. 3 – Polar coordinate pictures of vowel [a] and [i].

The demodulation of

$$c \sin at - \sum_i b_i \cos (a + \omega_i)\, t$$

then gives

$$1)\ K_1 \sum_i b_i \sin \omega_i t$$

and the modulation of

$$c \cos at - \sum_i b_i \cos (a + \omega_i)\, t$$

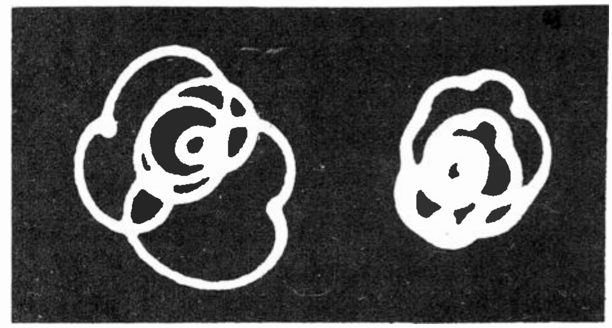
results in

$$2)\ K_{II} \sum_i b_i \cos \omega_i t$$

K_I and K_{II} are constants which are given by the characteristic curves of the demodulators. Assuming, that the two demodulators have the identical characteristic curves ($K_I = K_{II} = K$), we obtain at the outputs an exact 90° phase shift for all frequencies *without* any change in the amplitude relations. (Any other phase shift of the frequency spectrum can also be obtained, if we adjust the phases of the two carrier frequency components differently.)

CONVERSION OF THE POLAR COORDINATE PICTURES INTO CARTESIAN COORDINATE PICTURES (COORDINATE CONVERSION)

Polar coordinate representation of speech processes e.g., vowels, show interesting typical pictures (Fig. 3). The particular advantage is that it is substantially independent of the fundamental frequency. However, we do encounter a disadvantage in that only a single photograph can be made, e.g. for a given vowel. Photographs of the transition phenomena are thus almost impossible. We can eliminate this disadvantage by converting the polar coordinate pictures into cartesian coordinate pictures.



a b

Fig. 4 – Circuit diagram for production of a rotary field by a frequency spectrum.

Fig. 5 depicts a disc S with a radial slit. This is placed before the screen of the oscilloscope. With the disc rotating, we only pick up a punctiform section of the picture, cf. Fig. 2. A 45° prism, which can also rotate, is set up between the disc and a plane. The portion of the picture, passing through the slit, is projected through a lens L onto the plane. If we first hold the glass prism still and rotate the slit through the angle $+\phi$, its picture rotates through the angle $-\phi$. If, on the other hand, the

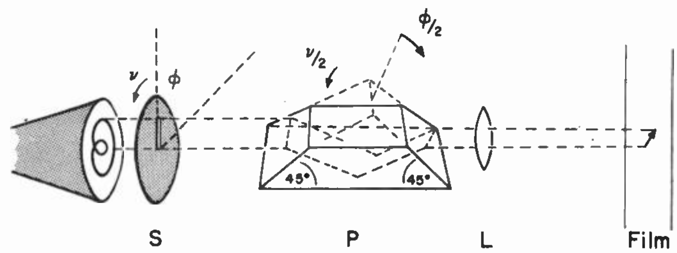


Fig. 5 – Arrangement for converting a polar coordinate picture into a cartesian coordinate picture.

disc remains motionless, while the prism rotates through the angle $+\phi/2$, the picture, passing through the slit, rotates through the angle $+\phi$ and thus once more assumes its original position. If we rotate the disc with the angular velocity ν, and the glass in the same direction with velocity $\nu/2$, then the projection of the picture remains motionless. In this motionless picture (through the slit) we see that the points move during slow rotation of the disc. The points correspond to the intersections of the rotating slit with the polar coordinate picture. We can then set up a movie camera in the plane of the projection. If we run the film at constant speed, we obtain a picture in cartesian coordinates, converted from the polar coordinate picture. Fig. 6a gives the laboratory construction of the entire apparatus. The rotating disc (with slit) may be seen enlarged in Fig. 6b. The rotating glass prism appears in Fig. 6c.

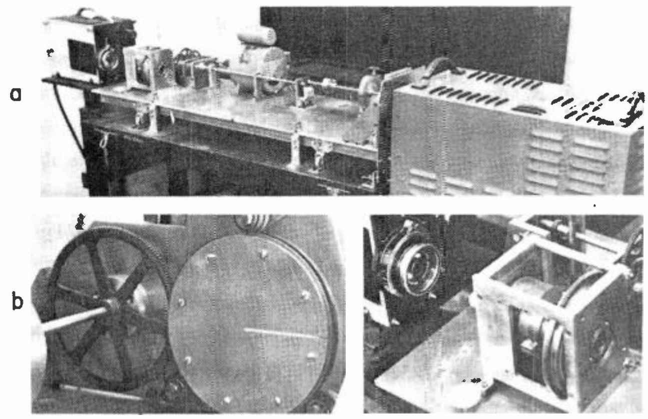


Fig. 6 – Construction of the apparatus for coordinate conversion; a–Total view of the apparatus; b–Rotating slit; c–Rotating prism.

the rotating electron beam during the reproduction of the polar coordinate picture. The effect cannot be completely eliminated by a corresponding choice of the time of afterglow; but it can be lessened. The polar coordinate pictures become more complicated, if the spectra contain more harmonics. This then becomes true also for the corresponding cartesian coordinate picture. We can naturally adjust the scale of the time axis and thus the length of the cartesian coordinate picture by choice of the film feed, so that we obtain an optimal picture impression. Moreover, the speed of disc rotation (ν) and of the prism ($\nu/2$) is chosen such, that changes in the speech phenomena can be detected. However, the speed of rotation is also chosen, so that we may analyse a speech sound, while it is spoken, within a small number of revolutions.

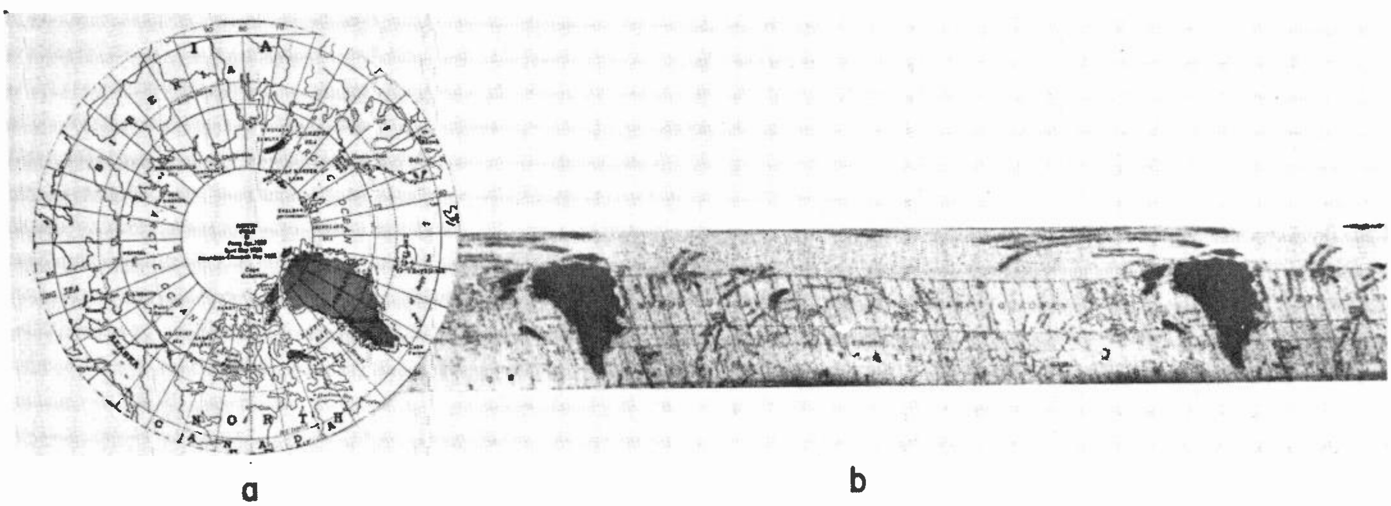


Fig. 7 – Coordinate conversion of a map of the North Polar Zone: a–Given in polar coordinates; b–Given in cartesian coordinates.

There exists a possible additional use for the instrument, which will also facilitate our understanding the procedure involved. We may replace the oscilloscope picture by a photograph of the northern polar zone in polar coordinates (Fig. 7a), which is transilluminated from behind. The optic quality of the laboratory instrument prototype must be improved, yet we can clearly recognize the details of the map in the cartesian picture (Fig. 7b). The picture repeats with every rotation of the slit or semirotation of the prism.

Fig. 8 shows the conversion from oscilloscope pictures in polar coordinates (line 1) into cartesian coordinates pictures (line 2). "A[1]" corresponds to a spectrum, composed of a fundamental frequency (500 cps) and the 4th harmonic (2000 cps). "B[1]" corresponds to a somewhat more complicated spectrum, which consists of the frequencies 500, 1000 and 2000 cps. "C[1]" is the vowel [*a*] and "D[1]" the sibilant [*s*]. In the cartesian coordinate representations of these pictures (line 2), we note that a part of the curves appear dotted. We may trace this to a stroboscopic effect between the rotating slit and

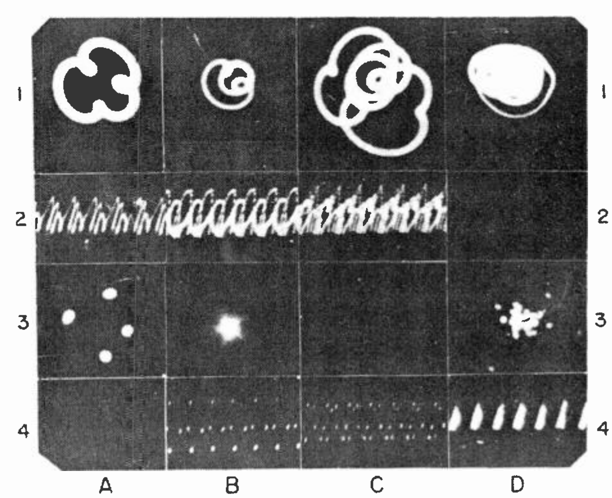


Fig. 8 – Coordinate conversion of polar oscilloscope pictures: Series A. Frequency spectrum 500 and 2000 cps; B. Frequency spectrum 500, 1000 and 2000 cps; C. Vowel [a]. D. Sibilant [s]. Line 1 – Polar coordinate picture; 2–Cartesian pictures corresponding to the pictures of line 1; 3–Polar coordinate pictures with regulated electron beam; 4–"Simplified" cartesian pictures corresponding to the pictures of line 3.

There exists an additional possibility for simplifying the pictures. We can regulate the intensity of the beam current of the oscilloscope by a range of the spectrum. In line 3 of Fig. 8, the polar coordinate picture is produced with the entire speech spectrum. The beam current is, however, simultaneously regulated by that range of the spectrum lying above 1500 cps. This hipass spectrum is selected by a filter and led to the Z-axis blanking grid of the oscilloscope. The polar coordinate pictures seem thereby divided into individual points. This is particularly effective, e.g., in the case of the sibilant [*s*]. *Without* regulation of the intensity the polar coordinate picture appears indistinct, similar to a woolen ball (Fig. 8-D1). The same screen with intensity regulation is similar to a picture of a star cluster (Fig. 8-D3). If we convert the polar coordinate pictures with regulated intensity into cartesian coordinate pictures (line 4 of Fig. 8), then we obtain a considerable simplification as compared with the unregulated pictures, cf. line 2 of Fig. 8.

DYNAMIC AMPLIFIERS FOR PHONOGRAPHIC REPRODUCTION*

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SUMMARY – In dynamic amplifiers for phonographs, the gain versus frequency characteristics change in accordance with the varying nature of the impressed audio signal. This paper outlines the basic principles of design of such amplifiers for expansion and for background noise reduction purposes. Especial reference is made to the fundamental patent literature in this field.

INTRODUCTION

A dynamic amplifier may be defined as one for which the gain vs frequency characteristics are a function of the nature of the input signal. Dynamic compressor and expander amplifiers for audio purposes usually are devices for which the gain changes are substantially the same for all frequencies, while the amount of change is in accordance with the change of acoustical value of the input signals. That is, the gain control device must give suitable weight to the acoustical value of the various tonal constituents of the signal, and must discriminate especially against the control by very low frequencies to which the ear is least sensitive. Selective dynamic amplifiers for producing much greater gain changes in some frequency ranges than in others are of interest for reduction of background noises by providing high gain for the ranges containing noises only when there are signals present suitable for masking the noises.

While the idea of dynamic compression is related to the concept of automatic volume control for radio receivers, the idea of dynamic expansion appears to have originated with the desire to reduce background noises in the wire communication art. Thus a patent to Affel[1] shows broadly how the gain of an amplifier may be increased in accordance with the signal volume at the input of the amplifier, by use of rectification and bias control on the grid of the amplifying tube. A later patent to Gerlach[2] shows specific applications to sound systems, and features compressors and expanders using instantaneous waveform distortion methods in contrast with the Affel method. A patent to Keller[3] is chiefly concerned with the noise reduction properties of dynamic expansion circuits. Other patents cover the early thinking along the lines of dynamic amplifiers for a variety of purposes. None of these appear to show the refinements necessary to permit construction of a phonograph system that would be musically satisfactory.

The improvements in phonograph records brought about by the introduction of electrical methods of recording resulted in a corresponding investigation by John Hays Hammond, Jr. as to improvements in electrical methods of reproducing. Using a phonograph amplifier which enabled symphonic records to be reproduced with original volume, Mr. Hammond concluded that in general, the records were deficient in dynamic range, and were insufficiently free from background noises. As a result, attention was directed to the development of expanders to increase the dynamic range and to the development of selective dynamic amplifiers to reduce the background noises without impairment of the musical quality. The various steps are recorded in a number of basic patents[4] which form the background of the recent commercial devices in this field.

* Manuscript received February 10, 1954.

[1] H. A. Affel, U.S. Pat. 1,574,780.

[2] E. Gerlach, U.S. Pat. 1,767,790.

[3] Keller, U.S. Pat. 1,784,839.

[4] J. H. Hammond, Jr., U.S. Pats. 1,979,034; 1,979,035; 1,979,036; 1,998,620; 2,008,701; 2,008,705; 2,008,707; 2,008,710; 2,008,825; 2,009,229; and others.

DYNAMIC EXPANDER AMPLIFIERS

The important features of dynamic expanders may be illustrated by Fig. 1, in which tube T_1 drives push-pull pentodes T_2 and T_3 which are gain controlled by signals originating prior to tubes T_2 and T_3 in the amplifier chain. In this example, the pentodes[5] are "suppressor and screen" modulated by low-frequency, subaudible-frequency and dc modulating components derived from an electrical evaluation of the acoustical level corresponding to the signal input. This is accomplished by use of amplifiers T_4 and T_5 which drive rectifiers T_6 and T_7 which supply a modulating signal to the suppressors of T_2 and T_3. The screen modulation is automatic because of the resistance-coupled amplifier effect which causes the second grids to follow the potential changes of the third grids without phase reversal. Electrical controls or adjustments of the modulation characteristics include the "action" control which determines the amount of acoustically evaluated signal required to make a given change of gain, a "timing" control by which the action may be slow or fast, and a "range" control by which the amount of change of gain may be regulated. These controls are shown independent but may be interlinked for purposes of simplification. Thus for speech purposes, the timing should be set relatively fast, the range should be wide and the action should be strong. The push-pull operation is used to prevent the modulating signal from passing directly into the modulated output. Resistors across the output transformer are necessary for fidelity purposes. A switch may be used for test purposes to connect the suppressor grids of the dynamic tubes to the cathodes to reduce the system from the operating condition to that of a normal amplifier for position N, which is substantially the same as would be produced by setting the range and action controls at zero. A dynamic expander of this type can provide expansion of the order of 30 db, with an output sufficient to drive push-pull triode power-amplifier tubes.

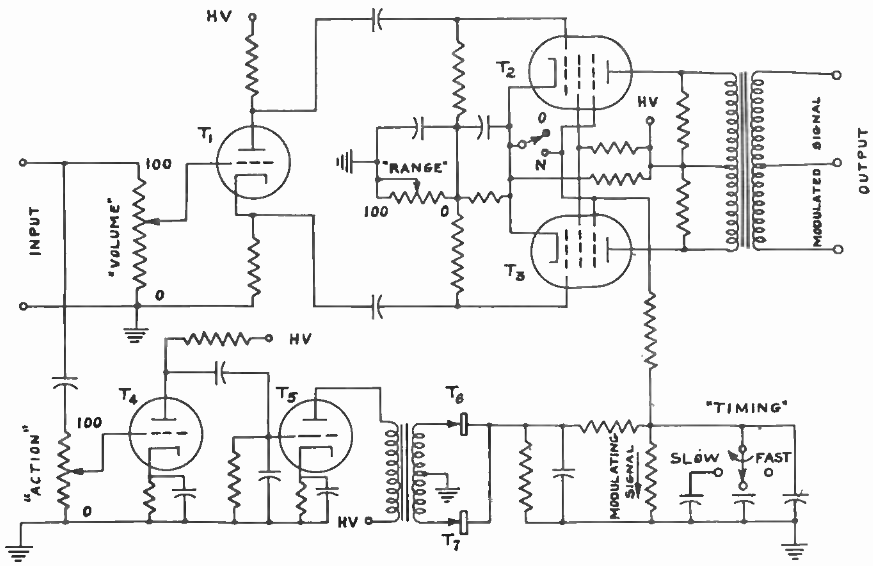


Fig. 1 – Dynamic expander, using pentode tubes.

DYNAMIC AMPLIFIERS FOR LOW-POWER PHONOGRAPHS

When the maximum output level of a phonograph is to be rather low, as in home instruments adjusted at less than ten percent of their full power output, it is generally conceded that the dynamic range cut into records is usually sufficient. Under such conditions, a dynamic amplifier should be designed to yield improved fidelity of reproduction with lowered ratio of noise to signal when averaged over the playing time of the record. Dynamic expanders can provide a reduction of background noise without appreciable expansion of the signal by concentrating the expansion in the low level of signal plus noise. This works out especially well for speech records, and with the action, range and timing controls properly set, background noises between words and syllables can be greatly reduced without appreciable alteration of the speech signal. However for musical recordings, it is usually impossible to set the timing both so slow as to provide smooth dynamic operation and yet so fast that the background noise will be quickly quenched whenever the signal is abruptly terminated.

The over-all ratio of signal to noise, averaged over the playing time of the record, can be greatly increased by providing that the high-frequency transmission gain shall vary in accordance with the total strength of the signal and noise in the high-frequency range. Let us suppose that a standard phonograph has been adjusted as to volume, bass and treble response to give the most

[5] E. S. Purington, U.S. Pat. 2,096,759.

pleasing effect to an individual listener, and that an automatic scratch-suppression device can be cut into the lead from the phono pickup to the phono amplifier. This device may be designed to yield zero insertion loss on all audio frequencies of interest when the device is wide open in response to a signal with sufficient high-frequency content. When there is no signal, or when the signal is mainly in the low and medium-frequency range, the device should produce very considerable insertion loss in the high-frequency range. A device so designed will reduce very considerably the average ratio of noise to signal, will reduce somewhat the average output volume, and will reduce slightly the average fidelity of reproduction. The individual listener may then choose to advance the volume and treble controls to secure equal or greater average volume, and equal or greater average fidelity, but with equal or less average ratio of noise to signal. It is in this manner that selective dynamic amplifiers can influence the listener to adjust the controls to utilize more fully the tonal range already provided in records, and the volume output already provided in the reproducer.

Selective dynamic amplifiers may be based upon any of four different electrical principles, a few of which have already been incorporated into commercial equipment. These principles are illustrated by Figs. 2 to 5, providing in general a low-pass filtering action when the signal plus noise in the attenuated range is weak, with automatic removal of the filtering action as the signal in the attenuated range becomes strong. The circuits shown provide for automatic reduction of the "scratch" noise. "Rumble" reduction could be secured by substituting high-pass filtering action, with removal of the filtering action by signals in the low-frequency attenuated range. Automatic and independent reduction of both scratch and rumble may be secured by sending the signal and noise through an automatically variable rumble filter and an automatically variable scratch filter in series.

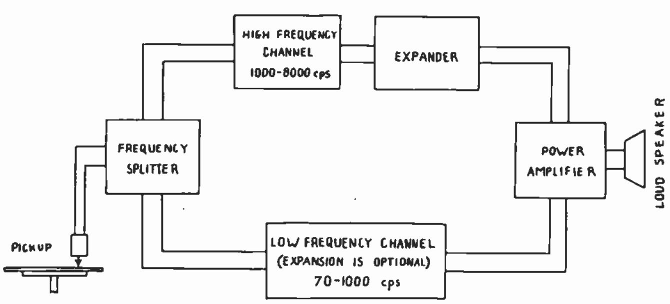


Fig. 2 – Selective expanding amplifier (dual channel method).

PARALLEL CHANNEL METHOD

One of the earliest methods of producing a selective dynamic amplifier[6] is shown in Fig. 2, where the entire signal from a phonograph pickup is split into two or more frequency channels by conventional filtering methods. Each channel may be individually treated dynamically, and the outputs may be combined or separately applied to loudspeakers. In the example indicated, for low-volume phonographs there would be little or no expansion in the low-frequency range, but a considerable degree of expansion, say up to 20 db, in the high-frequency range. Thus high-frequency signals get through the system only when they are present in sufficient quantities to mask the background noises in the high-frequency range. Ideally this is an excellent method and the degree of perfection achievable is limited only by the number of channels used, and the skill in lining them up to permit smooth transitions of musical passages in changing from one frequency range to another. By use of this system, it should be possible to play any of the sustained tones of a standard test record without background noises associated with frequency ranges different from that of the tone being played. It has already been applied commercially especially in a two-channel system[7] with the high-frequency channel using a special type expander which of necessity limits the expanded frequency range to one octave.

TRANSFORMER LOADING METHOD

The second method of selective dynamic expansion covered by Fig. 3, provides for an L-C type low-pass filter with variable characteristics[8], using the principle that loading the secondary of a transformer reduces the effective primary impedance. Here the filtering action, with low-frequency signal components only present, is produced by resonating the transformer to a frequency near the upper cutoff of the sound system, and including the primary between the sound source and the load, which may be a volume control such as the input volume control of the expander of Fig. 1. The loading of the secondaries is in a push-pull manner by two dynamic tubes T_1 and T_2 which pass no current and therefore have infinite internal resistances when there are only low-frequency signals present in the system. These low-frequency signals pass through to the volume control with little or no attenuation, but high-frequency signals in the scratch range are attenuated unless there are signals also in the high-frequency range which cause the secondaries to be loaded. This control is accomplished by a rectifier driver T_3 which receives high-frequency signals from the source and grid modulates the tubes T_1 and T_2 by low-frequency and dc components produced by rectification, using T_4. The change of bias in the dynamic tubes is limited by unidirectional conductor T_5.

This method has worked out well for a period of twelve years when operating into an expander of the type of Fig. 1, in the high-power phonograph installation of the Hammond Museum. The transformer has to be of rather low impedance to avoid capacitance difficulties, and therefore must work between rather low impedance termina-

[6] J. H. Hammond, Jr., U.S. Pat. 2,008,825.

[7] H. F. Olson, in "Electronics," p. 118, December, 1947.

[8] E. S. Purington, U.S. Pat. 2,096,760.

tions. Moreover the amount of current and the variations in amount of current required by the dynamic tubes, and the use of a special transformer is a consideration of importance.

This principle of changing the filtering characteristics by varying the feedback of an electronic system is the basis of most current commercial noise suppression circuits. One of the most popular of such devices, developed by Scott,[11] combines both rumble and scratch suppression in a single intertube coupling network, with the control of the passage of the low and the high-frequency signals through the network from take-off points also on the intertube network. A more conservative procedure for producing similar or improved performance would be to provide the rumble and scratch suppression in successive and independent intertube coupling systems.

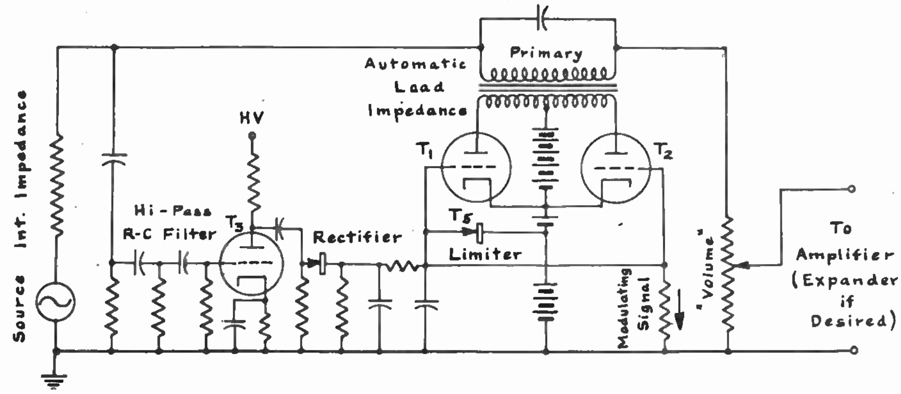


Fig. 3 – Selective expanding amplifier (transformer loading method).

FEEDBACK METHOD

The third method of Fig. 4 is a more recent development[9] based upon the feedback principle originally developed for use in frequency-modulation transmitter systems. Here a resistance coupled amplifier T_1 has a pentode tube T_2 in parallel between the plate of T_1 and ground. This tube T_2 is an impedance shunt on the tube$_1$ and its external load, and the transmission properties may be varied by controlling the value of the effective internal impedance of T_2. Broadly this is the same line of thought as used by Hammond[10], except that the present system uses a feedback capacitor from plate to grid of T_2 so that the effective impedance of T_2 is mainly a capacitance, resulting in giving the system a frequency characteristic. This effective capacitance shunt across the output of T_1 is present when there are only signals of low frequency present. But when signals of high frequency are also present, the capacitance is reduced to substantially zero by use of the rectifier driver T_3 and the rectifier T_4, which produces a low-frequency and dc signal to cut off the reactance tube current. In design it must be noted that the system is not of a push-pull nature, and that an abrupt change of space current of the reactance tubes caused by the modulation process would result in a transient pulse to the output volume control. This may be reduced to some extent by using separate feed resistors to the plates of T_1 and T_2, and reducing the low-frequency coupling from T_2 to the output volume control without impairing the coupling from T_1. If a bias change limiter is not provided, similar to T_5 of Fig. 2, care must be taken in the design so that the tube T_2 will not be driven far beyond cutoff by the rectifier, thereby preventing prompt restoration of T_2 to normal capacitance value when the high-frequency signal ceases to mask the background noises.

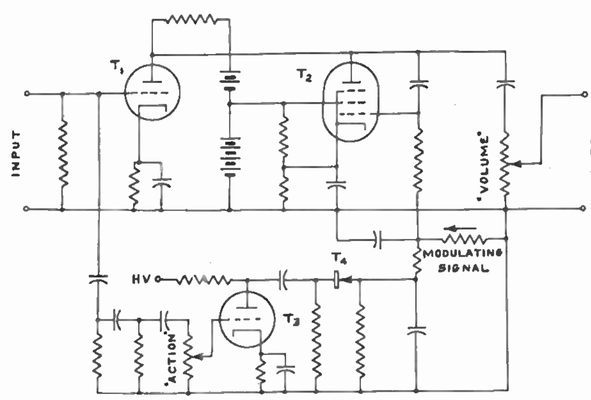


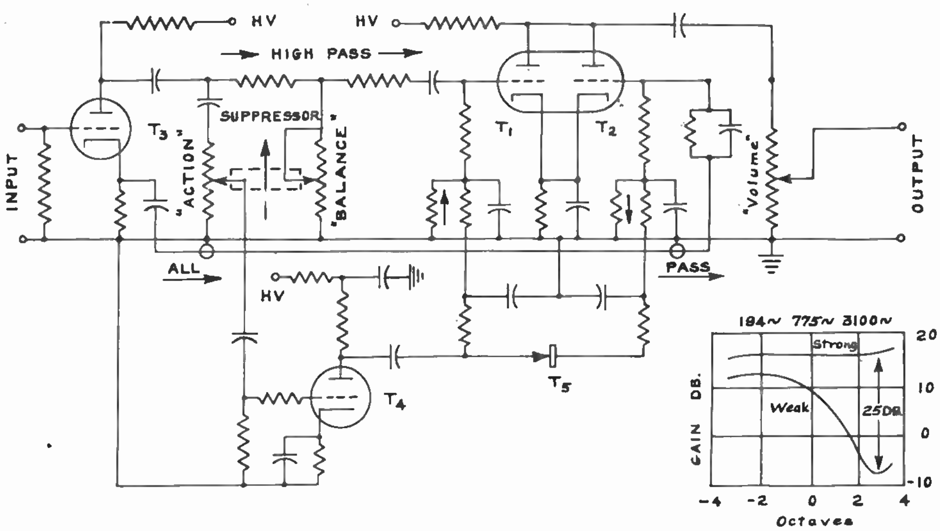
Fig. 4 – Selective expanding amplifier (reactance tube method).

BALANCE METHOD

A final fourth method of producing a selective dynamic amplification is shown in Fig. 5. This uses a balance principle[12] by which the transmission is made weak for high-frequency signals unless the input spectrum is sufficiently strong at high frequencies to remove the balance. In this circuit the triodes T_1 and T_2 are the dynamic tubes, which are modulated in opposite or push-pull senses by modulating signals, in such a manner that the sum of the two plate currents remains practically constant during the

[9] G. F. Devine, U.S. Pat. 2,369,952.
[10] J. H. Hammond, Jr., U.S. Pat. 1,998,618.
[11] H. H. Scott, in "Electronics," p. 96, December, 1947.
[12] E. S. Purington, U.S. Pat. 2,589,133.

modulation. This prevents transient surges to the output volume control. The grids of T_1 and T_2 are excited out of phase for high frequencies in the scratch range, but T_1 is not excited by the low-frequency signals which operate T_2. The balancing is accomplished by use of a phase inverter tube T_3, with the plate of T_3 connected to the grid of T_1 using a high pass RC filter, and with the cathode of T_3 connected to the grid of T_2 by an all-pass coupling. By a shunt capacitor on the high-frequency channel, or by a series capacitor in parallel with a resistor in the all-pass channel, the system may be brought into partial or complete balance at a frequency near the frequency cutoff of the pickup, power amplifier and loudspeaker system.



Fib. 5 – Selective expanding amplifier (balance method).

The over-all effect is to produce a transfer characteristic from input to output of the amplifier corresponding to an "m-derived" low-pass filter. The modulation is applied by rectifier driver T_4 and rectifier T_5 in a sense to reduce the transfer through T_1, thereby upsetting the balance and allowing T_2 to supply the entire signal to the output volume control without appreciable discrimination. It will be noted that there are two controls for this dynamic amplifier in addition to the "volume" control. These controls are "action" and "balance," and these may be engineered as a dual control marked "suppressor." When set at zero, there is no signal applied to T_1, and there is no dynamic action. As the control is advanced, the system comes into more perfect "balance" for high frequencies, subject to the restoration of fidelity of transmission by removal of the signal delivered by T_1 by the operation of the "action" circuit. This control may be adjusted to give slight overbalance for maximum setting, so that the action is not effective to produce change of transmission until the high frequency content is especially strong. This system gives a slight amount of expansion on low frequencies, but this is necessitated by the requirement of push-pull modulation to minimize thump, and in practice the expansion effect is in the proper direction and is generally beneficial. By use of high impedance triodes, such as 6SL7GT, it is possible to use a fixed crystal rectifier for T_5. Limiters of the grid swing of T_1 and T_2 may be dispensed with by choosing the constants of the rectifier driver circuit properly to give grid and plate-limiting action.

DYNAMIC AMPLIFIER FOR HIGH-POWER PHONOGRAPHS

When a selective dynamic amplifier such as in Figs. 2 to 5 is operated through an expander such as in Fig. 1, the over-all gain vs frequency characteristic of the system is a dual function of the nature of the signal input. Thus the selective dynamic amplifier is controlled especially by a restricted range of the input frequencies while the dynamic expander amplifier is controlled by the acoustical value of the output signals of the selective dynamic amplifier. Phonograph circuits using this dual control arrangement are believed to be best for very high-power installations. The selective dynamic amplifier should emphasize especially the scratch reduction, because the acoustical expander usually provides sufficient reduction of rumble and hum disturbances at levels where these effects would be harmful.

A dynamic amplifier for medium-power phonographs which combines both expansion and selective dynamic amplification in one single amplifier stage[13] is shown in Fig. 6. This is based upon a previous patent[14] in the field of selective RC electronic circuits, but by a different point of view it combines a dynamic expander amplifier as in Fig. 1 with a selective dynamic amplifier as in Fig. 4, except that both operations are effected by the same rectifier control. Or by a still different point of view it combines "horizontal" expansion with "vertical"

[13] E. S. Purington, U.S. Pat. 2,557,009.
[14] E. S. Purington, U.S. Pat. 2,082,097.

expansion. For simplicity, the control arrangement is shown symbolically, and it will be understood that the actual electronic control serves to drive the space current of tube T_1 from cutoff to normal bias and at the same time to drive the space current of tube T_2 in an opposite sense from normal bias to cutoff. This arrangement minimizes the transient effect upon the output voltage due to the changes of space currents in the dynamic tubes. Tube T_1 operates, broadly speaking, as the dynamic expander tube for "vertical" expansion; while T_2 operates, broadly speaking, as the selective dynamic tube for "lateral" expansion. T_2 should not be considered a "reactance" tube, but rather as an "impedance" tube, since it is a the tube T_1, and high conductance and very considerable feedback effect for T_2. As the acoustical level of the input signal is increased, T_1 approaches normal conductance for Class A operation and T_2 approaches cutoff, so that the system operates with substantially flat frequency characteristics at high gain. Curves are given showing how the gain vs. frequency characteristics are made to depend upon the control operation, measured in terms of the dc value of the space current for T_1.

This system of control provides sufficient undistorted output for driving a pentode output tube such as 6V6, and is perhaps the simplest method of producing 20 db or more reduction of noise in the entrance and exit grooves of commercial records, without impairment of tonal quality when signals are present. It will be noted that the "time constant" of the system, which will be determined by the rectifier driver and rectifier output circuit, is not the same for the high and the low-frequency ranges which carry most of the noise disturbances, as for the medium frequencies which carry most of the signal energy. Thus while the plate current of tube T_1 is dropping from 4.5 ma to 0.2 ma (and of course the plate current of tube T_2 is increasing from 0.2 ma to 4.5 ma), there is a drop of say 16 db in the gain of the system for 775 cps, while during the same period of time there is a drop of say 34 db for 3100 cps in the scratch range. This gives in effect a much more rapid decay rate for high-frequency signals than for medium-frequency, and partly compensates for the fact that this system does not use a dual control system for expansion and for selective dynamic amplification.

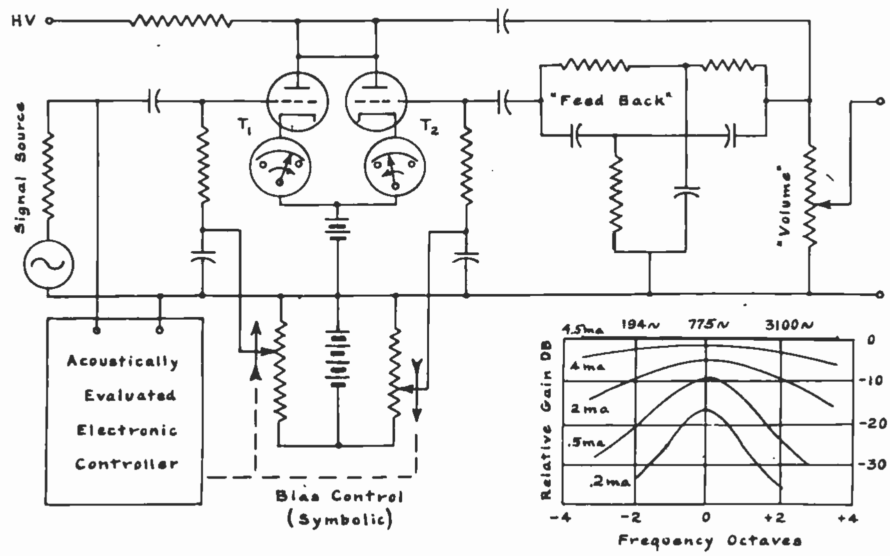


Fig. 6 – Dynamic amplifier with horizontal and vertical expansion.

triode and since the feedback from plate to grid is a a complex function of frequency. A low-pass RC network paralleled by a high-pass RC network is used for feedback purposes, and the grid voltage of T_2 is degeneratively in phase with the plate voltage both for very low and for very high frequencies, for which the capacitors are either of nearly infinite impedance or nearly zero impedance. This arrangement is not that of a "Wien-Bridge," since the cutoff for the low-frequency channel can be much lower than the cutoff for the high-frequency channel. For medium frequencies, the system may be slightly regenerative because the impedance of T_2 has a negative resistance component, but with low-impedance triodes (such as in 6SN7GT tubes) there is no danger of oscillation and the circuit is perfectly stable. The degeneration is greatest for high frequencies, but some amount of degeneration for low frequencies is desirable. In operation, the initial bias values for the triodes T_1 and T_2 may be set to give small space current for T_1, say 0.2 ma, and a correspondingly large space current for T_2, say 4.3 ma. This results in low conductance and weak gain for

CONCLUSIONS

In conclusion, this paper covers the broad circuit arrangements for dynamic amplifier design for phonograph reproduction. The choice of methods to be used for any specific application will depend upon matters other than the actual possibilities. Presumably devices of the type shown in Figs. 2 to 5 will suffice for small table model or small console type phonographs. For high-power installations, the system shown in Fig. 6 will work out somewhat better because of the combined vertical and lateral expansion effects. However, it is considered that one of the circuits shown in Figs. 2 to 5 operating into an expander (Fig. 1), or into a dynamic amplifier (Fig. 6) with less feedback effect, may prove most satisfactory. At any rate, it would appear that the design of a dynamic amplifier for phonograph purposes is ceasing to be an art, and is becoming more a matter of detailed engineering.

COMPONENTS AND MECHANICAL CONSIDERATIONS FOR MAGNETIC SOUND ON 35 MM FILM*

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SUMMARY — The art of sound recording in the motion picture field had reached a high professional status before magnetic recording began to receive consideration by the radio and television industries. This article reviews this status and discusses the relative merits of 35-mm sprocket-hole film and ¼ inch tape with respect to the requirements of the motion picture industry. The article traces the applications of the magnetic recording medium to single-track, multi-track and stereophonic recording, including composite magnetic sound and picture film. The discussion includes a description of the associated apparatus and its conformity to the established practices of the industry.

It is probably not generally appreciated by engineers outside the motion picture field that sound recording in the motion picture industry had reached a high professional status before magnetic recording began to receive consideration by the radio and television industries. Since 1928 the motion picture industry throughout the world has been equipped with photographic recording equipment, and improvements both in equipment and in recording techniques have been made constantly since that time. Not only were the studios equipped with photographic recording facilities but they had also brought to a high state of development, laboratory editing, dubbing and re-recording techniques built around the photographic sound medium. The question, therefore, was why should the motion picture industry change over to the magnetic medium when they seem to have had a highly developed system of recording on film. This situation undoubtedly accounts for the relatively slow adoption of magnetic recording in the motion picture industry as compared to the rapid use of the medium in the broadcast and allied fields. Gradually, however, the benefits accruing from the use of magnetic recording became obvious to the industry — such benefits including lower operating costs, better sound quality and simplified stage techniques.

Today, the motion picture industry is close to being 100% equipped with magnetic recording equipment for both production recording on the stage and for other special purposes, such as scoring and re-recording. The fact that the magnetic track was invisible and could not be "read" by the editors was, and still is, to some extent a stumbling block in the use of magnetic recording for editing and cutting purposes. It was difficult to break down the practices of a lifetime for many editors in switching over to the new medium. For this reason, it has been common practice to transfer the accepted "takes" of the original production recordings made on magnetic film to photographic tracks by re-recording or electric copying. These tracks might take the form of negatives from which prints could be made in a normal fashion, or in some instances they might be direct-positive photographic tracks. The positives made in either manner could then be used for the normal editing, cutting and re-recording procedures which had been established over a period of years for the photographic medium.

While ¼" tape has become the almost universal medium for magnetic recording in the radio, phonograph and television industries, it has found relatively little favor in the motion picture industry. The synchronized tape recording machines, which have been described in the literature,[1] are in limited use but represent only a negligible share of the film footage used in the motion picture industry. There are several reasons why this situation exists. In the first place, the motion picture industry did not wish to scrap its rather expensive 35 mm photographic film recorders for the newer and untried medium, preferring to have them modified for recording on 35 mm magnetic coated film. In the second place, the problem of synchronizing sprocket-hole sound film in the recorder and sprocket-hole picture film in the camera had

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[1] See references listed at the end of this paper.

been successfully worked out and motor systems had been devised to make this a simple and foolproof operation. As a result, there was considerable skepticism of the feasibility of any attempt to synchronize a tape without sprocket holes but having a superimposed carrier as a means of providing the exact degree of synchronization required. Further, over a period of years the problems of pulling 35 mm sprocket-hole film with a high degree of uniformity of film motion past the line of translation have been largely overcome with a result that by the time magnetic recording made its appearance, professional photographic recorders were available on the market with speed variations well under 0.1%.[2] Also, such auxiliary studio devices as film editing machines, rewinds and synchronizers used standard 35 mm type sprockets; and the projection rooms used to project the daily sound and picture films were, of course, also equipped with sprocket type film pulling mechanisms. The conversion or replacement of these equipments by synchronized ¼" tape devices would have meant considerable capital outlay. As a result of all these considerations, the motion picture industry seems to be definitely committed to the use of 35 mm magnetic film or, in some cases, split film of 17½ mm dimension. Magnetic tape is nominally only 2.2 mils thick and "print through" of the signal occurs between adjacent windings in a reel at a sufficient level to be objectionable in the motion picture technique. This is not a problem with the thicker 35-mm film.

The first use of magnetic recording in the motion picture studios was largely made on converted photographic recorders, and Fig. 1 shows a typical photographic recorder thus modified.[3] In this type of recorder the line of trans-

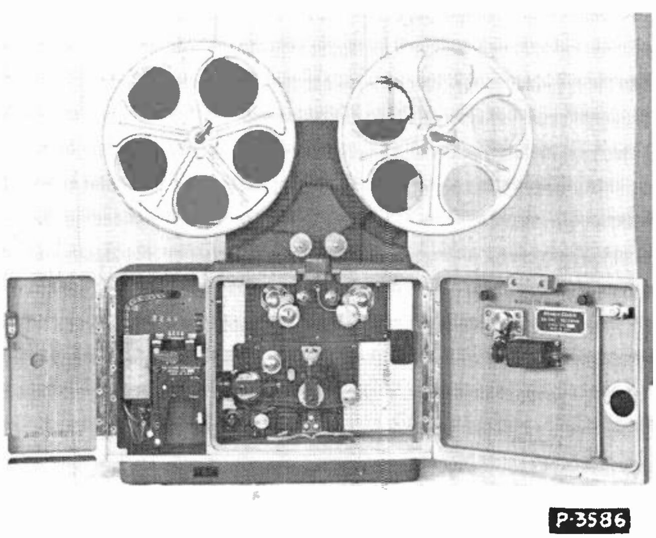

Fig. 1 – Photographic recorder modified for magnetic recording.

lation for the photographic medium was at the film recording drum. At this point in the film path, the filtering of undesirable speed variations, whether due to gears, sprockets, or rollers, etc., was at a maximum and the constancy of speed at a corresponding optimum. It was natural, therefore, to mount the magnetic head so that the line of translation for it would correspond to that for the photographic medium. In Fig. 2, there is shown a magnetic head mounted in this drum position – the head being mounted inside the film loop formed by the film as it passes around the drum, the magnetic coating being, of course, on the inside of this loop and in direct contact with the magnetic head.

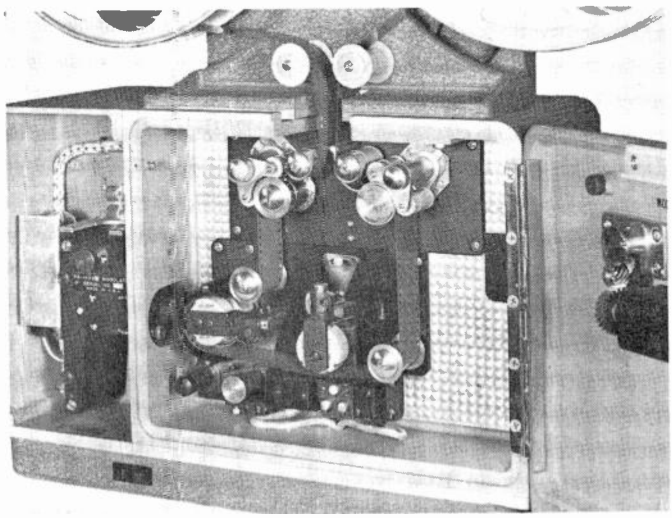

Fig. 2 – View of recorder showing magnetic head mounting.

By mounting the head in this position, it was found that the constancy of magnetic film speed was comparable to that of the earlier photographic type of recording, with the single exception that irregularities of motion due to the passage of the film over the magnetic head added considerably to the amount of flutter in the higher flutter rates. Thus, for example, for all flutter rates normally found below the sprocket hole frequency of 96 cycles, comparable performance was found between the magnetic and photographic methods. At 96 cycles and higher rates, the magnetic film showed a considerably higher degree of flutter disturbance. This was found on examination to be partially due to the polygoning of the film adjacent to the sprocket hole areas and partly due to the rubbing effect of the film as it passed over the magnetic head. Since the whole area between the sprocket holes of 35 mm film was available for the magnetic sound track, the first of these problems was met by removing the track to a considerable distance from the sprocket holes. Thus, the earlier single track magnetic recorders utilized a film location about 135 mils in from the inside edge of the sprocket holes. This appeared to eliminate the 96 cycle disturbances referred to above. Later, however, when the industry evinced an interest in multiple tracks on 35 mm film, it was found necessary to move the outside tracks closer to the sprocket holes. The compromise[4] finally adopted for 3 tracks on 35 mm film was as shown in Fig. 3. This shows that a 50 mil separation was provided between the magnetic tracks and the nearest sprocket holes. Admittedly, this meant some increase in 96 cycle sprocket hole modulation[5] but at such a level as not to interfere noticeably with the quality of the recording.

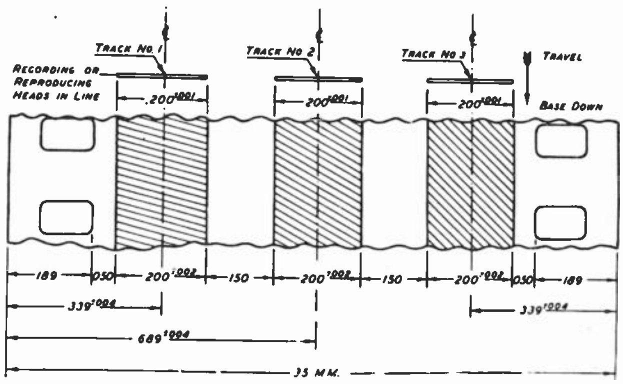

Fig. 3 – Magnetic film-track standards.

In introducing magnetic recording into the motion picture studios, the design engineers were undoubtedly influenced by the already existing photographic recording channels which in turn had proven quite satisfactory to the motion picture industry. Thus, while it was common practice in the ¼" tape equipment industry to supply, at least in the initial stages, one instrument embodying all the electronic as well as the mechanical components, this violated a long established practice with the motion picture industry. The studios have always employed separate recorders embodying only the mechanical and optical elements necessary for exposing the photographic film. The mixing unit has always been a separate and usually a highly portable unit and the remainder of the channel equipment, including amplifiers, power supplies and other accessories, has usually been mounted in separate cabinets or racks or boxes, and in the case of portable equipment is usually mounted in light-weight trucks. The physical operation of the magnetic recorder from its associated electronic circuits, both recording and monitoring, influenced to some extent the design of the magnetic heads. Thus, while high impedance heads are common practice in ¼" tape recording machines where only short cable runs are necessary from recording and monitor amplifiers, the motion picture industry generally accepts low impedance heads of the order of 2 or 3 mh. This permits operation at a considerable distance over low impedance circuits from the recording equipment and, if necessary, from the monitoring pre-amplifiers. An exploded view of a typical magnetic recording or reproducing head is shown in Fig. 4.

Another practice in which the motion picture magnetic film recording differs from tape recording is in the almost complete absence of electronic erase heads from the recording machines. This is largely due to the fear of the motion picture industry that such an erase head might well result in the accidental erasure of material which had been obtained at a very great cost, and could only be replaced in many instances with the expenditure of an equal amount of money. This led to the development of

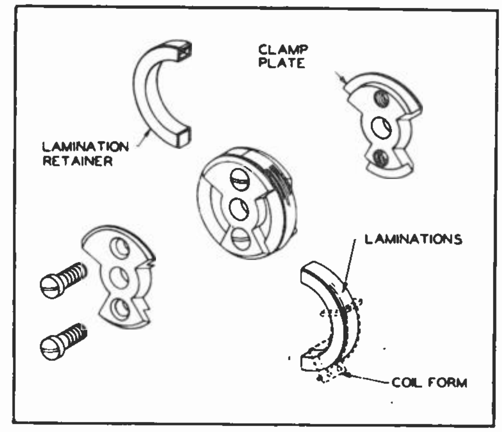

Fig. 4 – Exploded view of typical magnetic head.

bulk erasure equipment and although commercial bulk erasers were made available, most of the motion picture studios have seen fit to develop their own particular way of accomplishing this result. A typical magnetic bulk eraser is shown in Fig. 5. This is one of the simpler non-automatic types and requires manual movement of the

Fig. 5 – Typical bulk eraser.

film through the erasing field – the film passing through a total of 3 times with the roll being rotated 20° before each pass. In general, it has been found that such a type of eraser is very satisfactory and seems to result in less noise than is found with the use of electronic type erasing.

The modification of existing photographic recording equipment to provide magnetic recording facilities was followed by the development of magnetic recording systems. The principal components of a typical system are shown in Fig. 6. They consist of a recorder, a two-channel mixer and a power supply. The recorder is provided

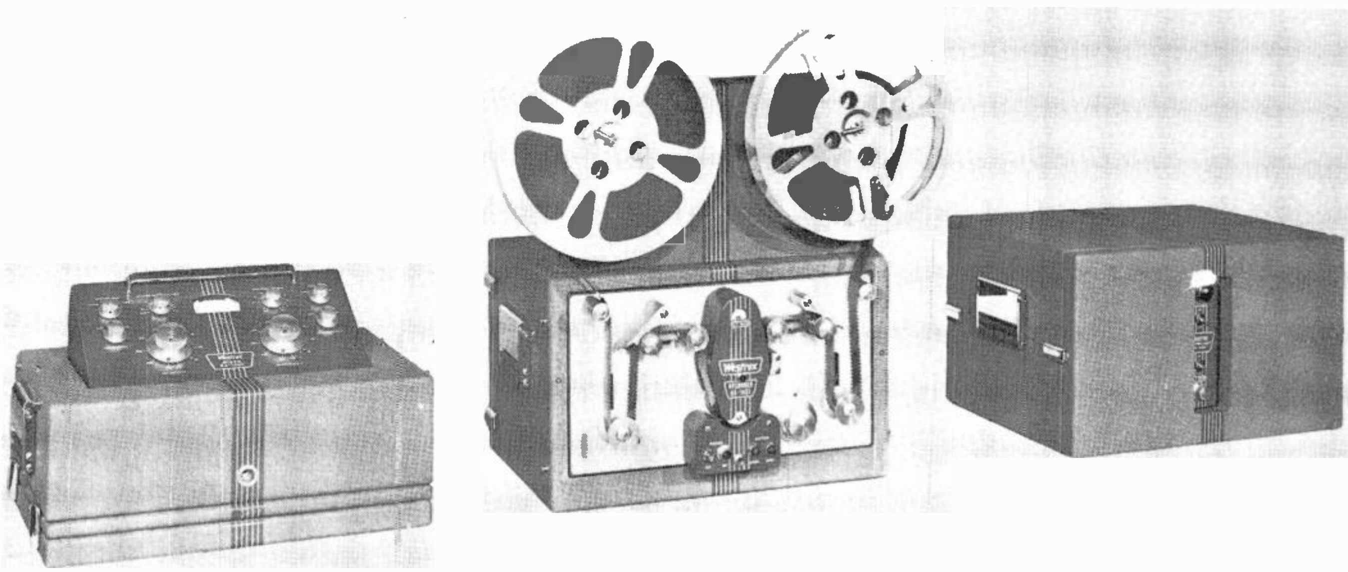

Fig. 6 – Typical magnetic recording system.

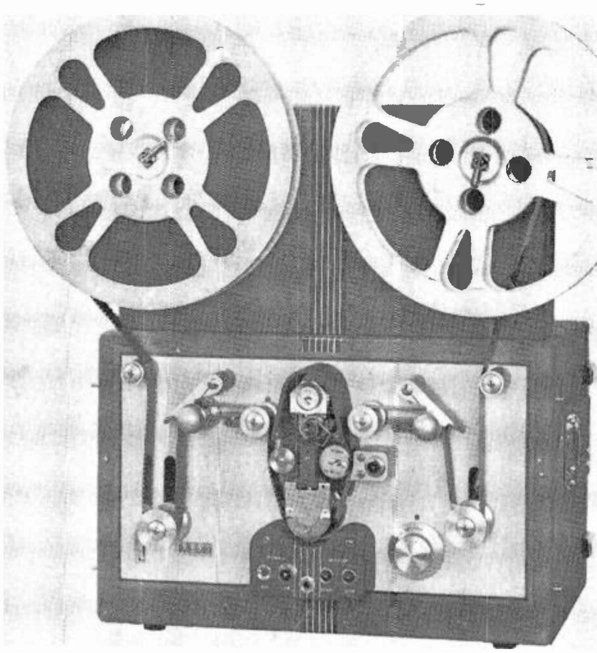

Fig. 7 – Magnetic recorder with front cover and cover plate removed.

with high-quality film-monitoring facilities which can also be used for reproduction from magnetic sound track using either the monitoring or the recording magnetic head.

Fig. 7 is a view of the magnetic recorder with front cover and the cover plate over the magnetic head assembly removed. Simplicity of operation and flexibility to meet various studios' operating procedures have been given special attention. A signal light indicates the proper film threading loop for optimum filtering of the film drive. A

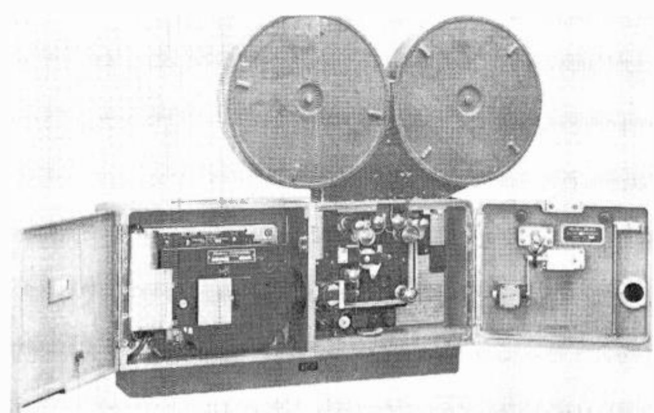

Fig. 8 – Dual photomagnetic recorder.

selector dial automatically sets the drive conditions and circuit connections for recording, reproducing from either magnetic head or high-speed rewind. Either film reel can be operated in either direction to accommodate a studio's practice with reference to the direction in which they wind their film. Shock rollers absorb the initial increase in film tension at the start to insure a minimum of film wear.

Another interesting medium employed in the studios is the combination of photographic and magnetic recording on a single film. This has not attained universal acceptance but is employed in a few of the Hollywood studios. In this case the magnetic coating appears in the form of a stripe placed on the film base, the film itself being coated with a standard photographic emulsion. A recorder equipped to provide simultaneous photographic and magnetic recording is shown in Fig. 8. The purpose of this type of recording is to provide a photographic track which may be used for the inspection of the daily prints and for editing and cutting purposes. In this case, normal studio practices for photographic editing, cutting, etc., are employed. Since the photographic and magnetic modulations are co-linear, the film may be cut straight across and spliced together and maintain proper synchronization with the picture film. The magnetic track on such a film is normally employed only for re-recording or dubbing purposes, since it is generally accepted that the quality of recording from a magnetic track is superior to that of a photographic track. If this type of equipment had been made available in the early stages of conversion from photographic to magnetic recording this technique might have become more widely adopted. However, since magnetic editing and cutting from so-called Magnastripe[6] film is becoming quite popular in the studios, the future of this type of recording is decidedly limited. The same might be said for the practice discussed above of transferring from magnetic to photographic recording for editing purposes. With the conversion of magnetic editing machines and the introduction of new editing machines[7], such as shown in Fig. 9, the editing of magnetic films has been made quite simple; hence, in the not too distant future the use of the photographic medium in the motion picture studios for all production and studio operations will undoubtedly become less in evidence and the whole field will be taken over by the magnetic medium.

The use of multiple tracks posed the problem of minimizing cross-talk between adjacent tracks. Cross-talk even in minute quantities becomes a very important matter especially when separate intelligences are recorded on each of the individual tracks. For example, triple track film might have a dialogue sequence on track no. 1 and a music sequence on track no. 2. At some time during the production of the motion picture, it might prove desirable to erase the dialogue on track 1 and replace it with a somewhat different version, or even with a different language. For these reasons, it is quite obvious that cross-talk of the original dialogue into the music track would be very undesirable. It was found that most of this cross-talk occurred at lower frequencies or longer wavelengths, due apparently to the spreading of the magnetic fields of the longer magnets at the lower frequencies. A very ingenious device, shown in Fig. 10, illustrates how this problem of cross-talk was overcome. This involved the use of what has come to be known as decouplers[8] which are mounted between the individual sections of the multiple track head. These strips of mu-metal are deliberately placed so as to introduce an out-of-phase signal from one head into the other and of sufficient value to cancel out the induced cross-talk signal. It has been found that with the use of these decouplers, the cross-talk can be reduced effectively to a value approximating 60 db below that of the fully modulated signal.

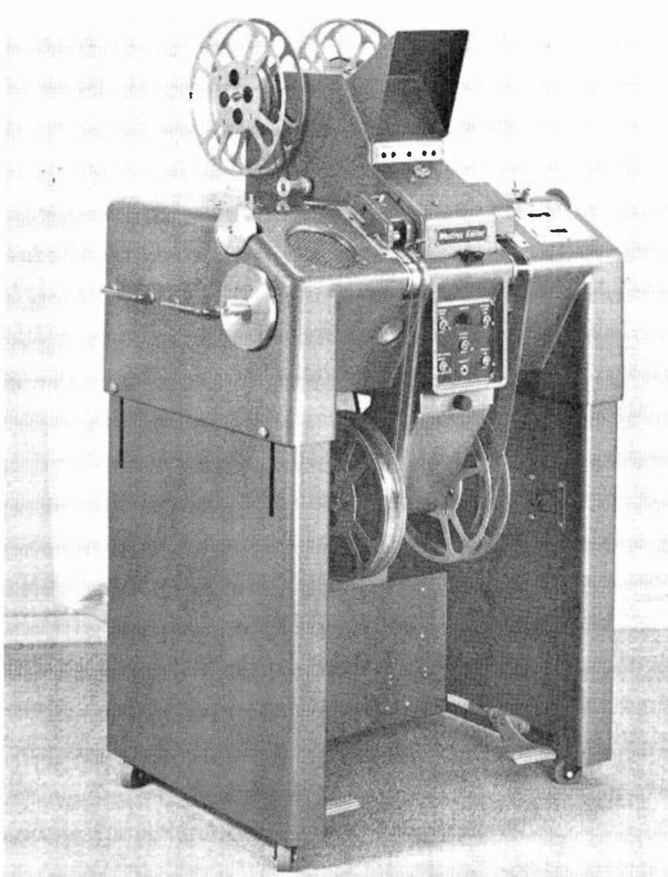

Fig. 9 – Westrex film editer.

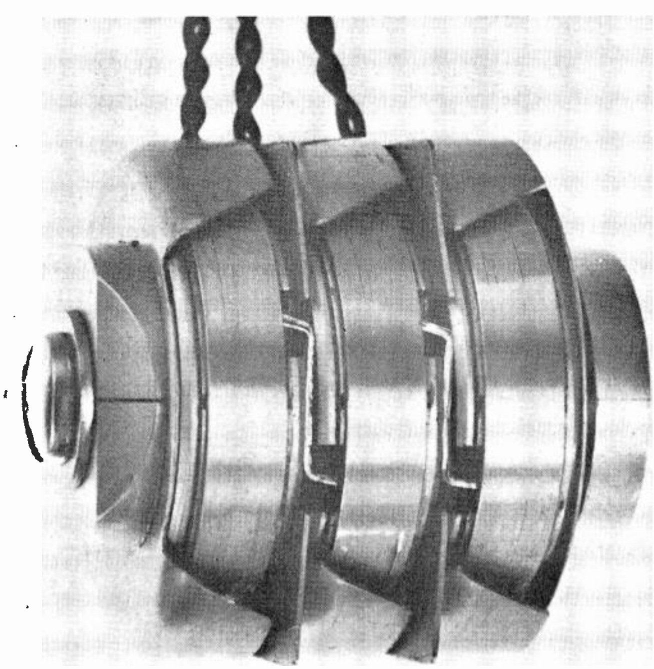

Fig. 10 – Triple-track magnetic head.

This reduction is a function of frequency as shown in Fig. 11, curve 1, but when the ear-weighting characteristic is superimposed on the cross-talk curve, effective cross-talk reduction is found over the entire useful audio spectrum, as shown in curve 2.

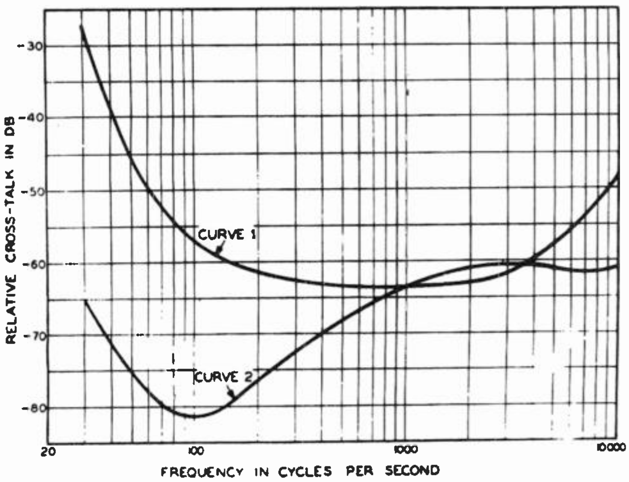

Fig. 11 – Crosstalk as a function of frequency.

The use of the multiple tracks on 35 mm film necessitated a modification of the film drive mechanism outlined previously in this paper. It was no longer possible to mount a multiple track head at the recording drum. As a result, the double flywheel type of drive such as shown in Fig. 12 was developed for this type of operation. The basic elements of the Davis Drive[9] almost universally used in sprocket type film pulling mechanisms was retained in this newer double flywheel type of mechanism. In this case the magnetic heads, both recording and monitoring, are mounted on a plate between the two drums. It was found that when the moment of inertia of the two drums and their associated flywheels equals that of the single drum and flywheel, equally good flutter performance is obtained with this type of drive. In fact, the double flywheel type of drive showed an improvement over the single flywheel in that the polygoning was considerably reduced due to the much lesser curvature of the film as it passed over the magnetic heads. In the type of machine shown in the illustration, the film wrap around each magnetic head is of the order of 30°, thus providing a large area of contact with the permalloy core. This in turn reduces wear and increases the useful life of such a head. With this arrangement the performance on all three tracks is identical and when the decouplers referred to above are installed in such a recording head, a signal-to-noise ratio in excess of 60 db is obtainable.

This multiple track recorder was originally designed for scoring and dubbing work in the motion picture studios before the recent revived interest in stereophonic recording.[8] With the introduction of Cinerama, which has 7 stereophonic sound tracks on a standard 35 mm film, the industry became quite conscious of the entertainment value of stereophonic sound. The triple track recorder described above was obviously made to order for this type of recording and when 20th Century-Fox decided to add stereophonic sound to their proposed Cinemascope pictorial presentation, this machine was immediately used

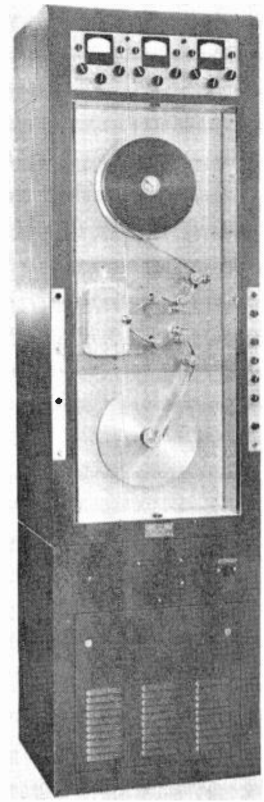

Fig. 12 – Triple-track recorder.

for the stereophonic production scoring and dubbing recording operations. However, in its original form it was not suitable for portable type of operation so that a new or rather a modified single channel portable recorder was developed to provide stereophonic recording. Such a portable recorder is shown in Fig. 13. This recorder incorporates, in addition to the film pulling mechanism and the magnetic sound heads, the necessary electronic items for its operation. These include a bias oscillator which furnishes bias current for the 3 tracks and 3 monitor amplifiers as well as the various operating controls. This recorder when associated with the mixer shown in Fig. 14 provides the necessary electronic items for a complete production stereophonic channel.[10] The power supplies required for such a channel are housed in a separate container. For stereophonic recording, the cross-talk reduction of 60 db previously mentioned is not necessary in view of the natural cross-talk between the microphone pickups on the stage. With the elimination of the decouplers, cross-talk reduction of the order of 40 db is possible and this seems to be quite satisfactory for stereophonic recording.

With recording machines of this type and re-recording machines of the earlier type described above, the motion picture studios were in a position to do production stereophonic recording and carry on the re-recording operations in the same medium.

In order to get the stereophonic sound into the theatre, the first attempt made during the early part of

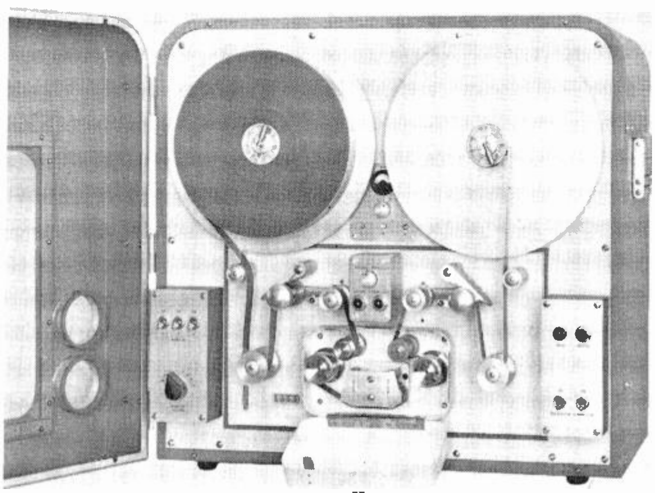

Fig. 13 – Portable stereophonic recorder.

1953 was to supply a separate 35 mm magnetic film on which were recorded three 200-mil magnetic tracks. This film was projected over a separate sound dummy, such as shown in Fig. 13, located in the projection booth, provision being made for interlocking this machine with one of the projectors for ordinary flat pictures or with both projectors for 3-D pictures. This was accomplished

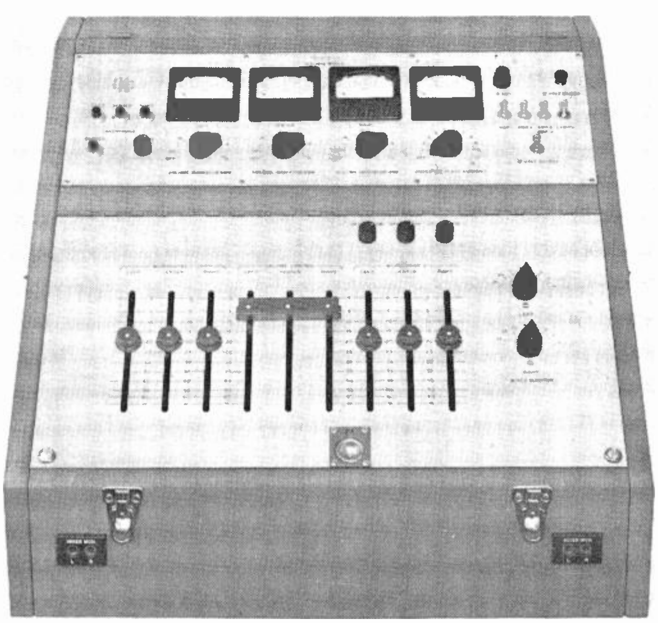

Fig. 14 – Stereophonic mixer.

by mounting 2-pole Selsyn motors tied in together electrically, each unit being driven through a reduction belt from the driving motor on each mechanism. This arrangement worked reasonably satisfactorily and many pictures were projected throughout the country during the year using this arrangement.

The economic difficulties as well as the technical problems of supplying 2 or 3 films to theatres in order to put on a complete show led to the development of single film methods which embody many of the entertainment features of these pictures. Thus, during 1953 the Cinemascope process was brought to complete development and introduced into the theatres. In this process there are several innovations. For the first time in the history of motion pictures a single film carries both the picture and the associated stereophonic sound tracks. These sound tracks are provided by striping certain otherwise unused areas of the picture film with narrow magnetic coatings. Four such coatings in all are supplied – two being outside the sprocket holes and two inside and adjacent to them. A diagram of the Cinemascope film showing the location and dimensions of the magnetic stripes and of the recorded track areas is shown in Fig. 15. The three 50-mil tracks provide the stereophonic signals to the three

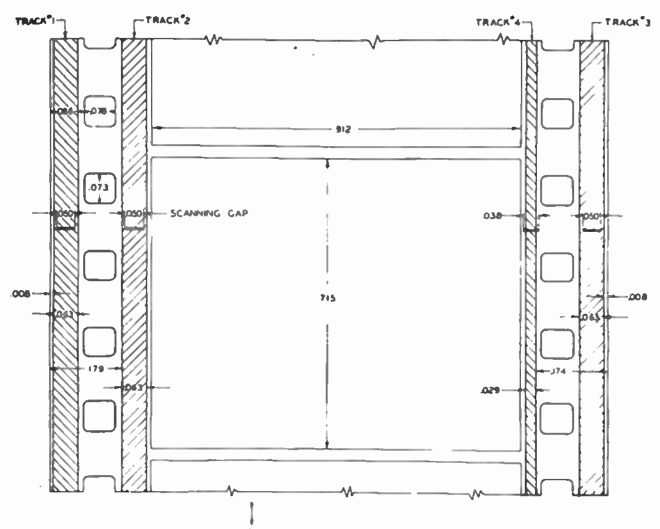

Fig. 15 – Proposed standard for Cinemascope sound track.

speakers behind the screen. The narrow track which is 29 mils wide provides a signal for auditorium speakers as well as a control signal for cutting these speakers in and out. Another item of novelty in this film is a new sprocket hole design. The standard sprocket hole has been abandoned in favor of a narrower one in order to provide more room for the sound tracks. The old sprocket hole which was 0.110" wide has been reduced to 0.078" wide, the height being reduced at the same time from 0.078" to 0.073". The change of sprocket hole means, of course, a change of all existing sprockets in theatre reproducing equipment.

In order to be able to run this Cinemascope film in the motion picture theatres throughout the world, it was decided to design a new sound head specifically for playing these tracks. Since this new sound head is located on top of the projector housing, it has been denoted variously as a penthouse, button-on or sandwich head. A typical one is shown in Fig. 16. This means that the practice, which has been adopted since the introduc-

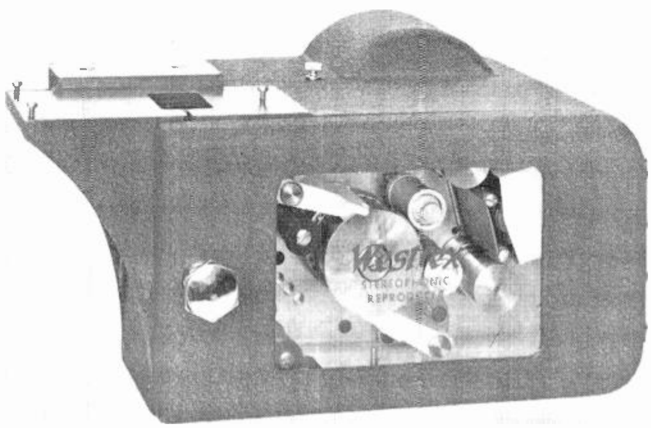

Fig. 16 – Theatre reproducer for Cinemascope film.

tion of sound pictures, of advancing the sound track start mark ahead of the picture by 20 frames to permit scanning the sound track in a special head located beneath the projector has had to be abandoned. For the Cinemascope film the start mark has been retarded by 28 frames. An item of interest about this new Cinemascope sound head is that the sprocket is film-driven rather than being used as a means of pulling the film through the machine.[11] The film is pulled by the upper sprocket in the picture projector and the sprocket in the penthouse head is simply used as a means of maintaining a loop of a certain length. Otherwise, the film drive in this penthouse head is quite similar to that used in the recorders and re-recorders described previously in this paper.

In reproducing the 4 Cinemascope sound tracks in the theatre, 4 pre-amplifiers are used. These are usually mounted in a box on the front wall of the projection booth. The frequency response of these amplifiers is shown in Fig. 17. This characteristic incorporates the customary

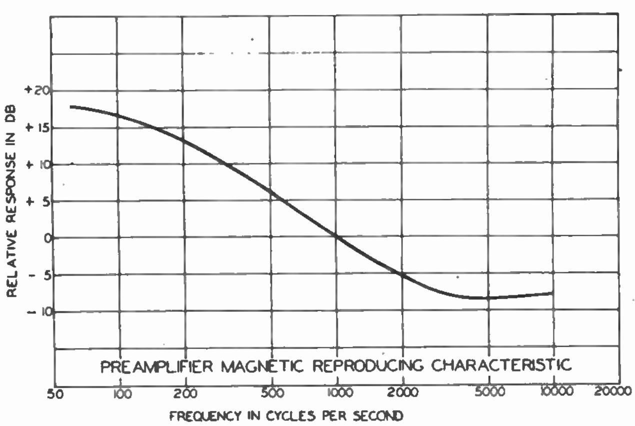

Fig. 17 – Pre-amplifier magnetic reproducing characteristic for Cinemascope sound track.

6 db per octave magnetic reproducing characteristic in addition to low and high-frequency post-emphasis. The latter amounts to about 6 db at 50 cycles and 4 db at 8 kc – these values corresponding to recorded complementary pre-emphasis at the same points in the frequency spectrum. The resulting over-all frequency response is essentially flat from 50 cycles to 8 kc with a loss of about 3 db at 10 kc.

The loudspeakers used behind the screen are the standard two-way theatre horn systems with the dividing network cross-over frequency located at 500 cycles. A photograph of a typical 3-horn installation is shown in Fig. 18. The low frequency units used in these speakers are usually standard 16" paper cones driven by permanent-magnet-actuated drivers and are mounted in a combination horn baffle. The high frequency units are mounted above the low frequency units and to insure proper distribution of the more directive high frequencies, either multi-cellular-type horns or acoustic lenses are employed. In order to provide good coverage in theatres, especially those having balconies, more than one high

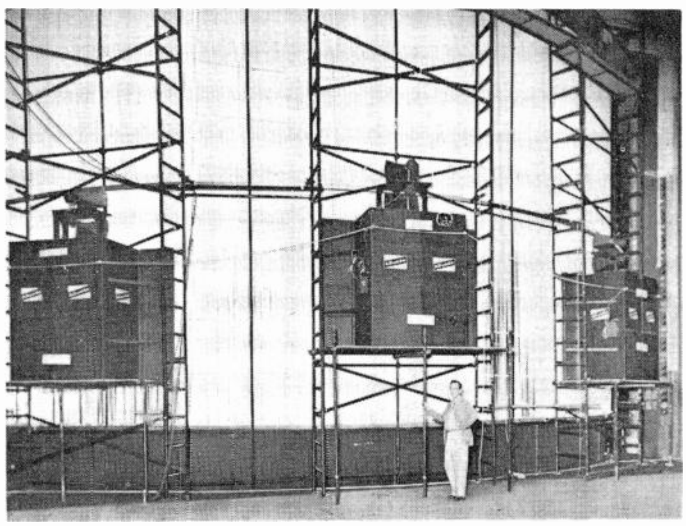

Fig. 18 – Stereophonic loudspeaker installation.

frequency unit may be used in order to direct the sound uniformly throughout the auditorium. The loudspeakers are usually placed at the center and at the right and left of the screen – the centerline of the side speakers being located at a distance from the outer edge of the screen which is equal to one-third the screen width. This has been found to give good distribution when employed with screens up to 65' wide, such as employed for Cinemascope projection in large theatres. It is characteristic of most commercial high frequency units that there is considerable fall-off in high frequency response above 8 kc. At the present time, no attempt is made to equalize electrically for this fall-off in response. In fact, it is generally necessary to provide some high frequency roll-off to secure satisfactory reproduction of dialogue in many auditoriums. In other words, to date the motion picture industry has not succeeded in putting what might be called high fidelity sound into theatres even with magnetically recorded tracks and with stereophonic sound. The situation, however, is considerably improved over single photographic track systems where a very rapid attenuation of the high-frequency response above 7500 cycles has been standard practice.

The advent of magnetic tracks in theatres has created the problem of how to make many duplicates from a master recording. So long as the photographic medium was employed, the prints were made from a photographic sound negative – the printing process being usually carried on simultaneously with that of printing the picture. Since no acceptable method has been developed for making contact magnetic prints, electrical copying of such prints is necessary at the present time. In order to accomplish this, a single multi-track reproducer using a 4-track magnetic master film is used to feed signals to a bank of multi-track magnetic recording machines. A photograph of a typical installation is shown in Fig. 19.

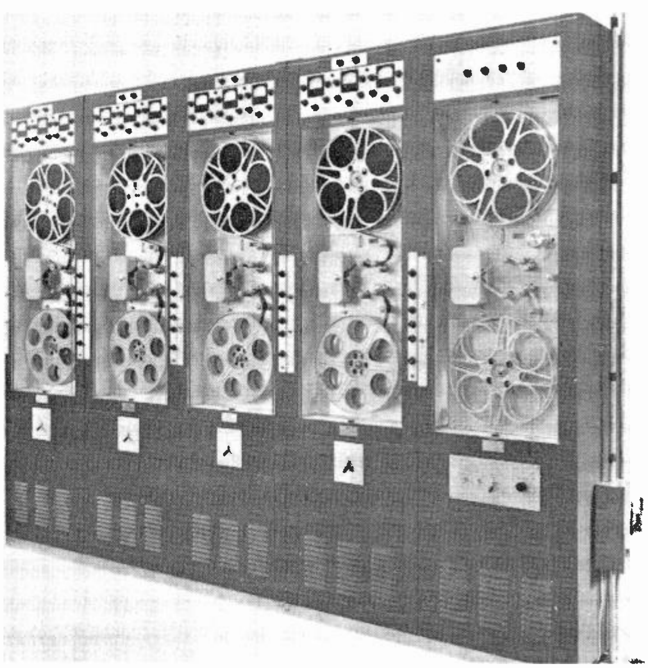

Fig. 19 – Multitrack electrical printer.

This shows one reproducer and 4 recorders. In order to accelerate production, this group of machines may be operated above the normal motion picture film speed of 90' per minute and speeds up to 180' per minute are permissible. With increased speed and the use of a number of recorders, the magnetic transfer process can be highly accelerated.

The story on Cinemascope tracks would not be complete without a brief mention of the method of applying the magnetic material to the Cinemascope picture prints. In this case the finished photographic print is run through what is known as a striping machine, a view of which is shown in Fig. 20. A solution carrying the finely ground magnetic oxide is fed from a hopper through 4 nozzles which lay down the 4 tracks previously discussed. Since these tracks are in the liquid state, the film must be passed through a dry box similar to that used in film developing machines before the material can be used in the printing machines described above. A variation from this type of coating machine is one in which very thin strips of previously coated base are laminated onto the picture film. The latter method has not attained practical acceptance yet in the industry and as far as is known, the only system currently employed in commercial practice is that similar to the one provided by the machine shown in the last figure.

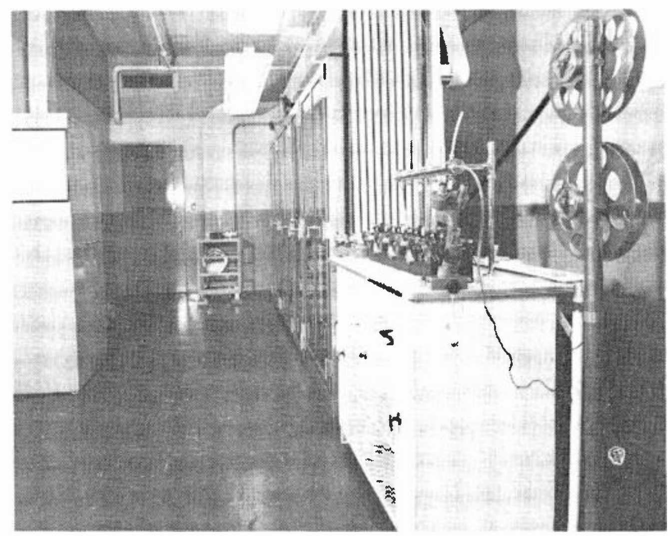

Fig. 20 – General view of magnetic film striping machine. (Courtesy Warner Brothers Studios)

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A LOUDSPEAKER ACCESSORY FOR THE PRODUCTION OF REVERBERANT SOUND*

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SUMMARY – Organ music produced in small rooms having little natural reverberation can be enhanced by the addition of artificial reverberation. A direct method of adding the aftersound at the electro-acoustic transducer itself is described. This is in contrast to the more conventional reverberation systems employing driving transducers, time-delay means, pickup transducers, and mixing and amplifying circuits. Multiply resonant helical mechanical delay lines store the energy and radiate it at a later time. In one model the coupling to the transducer is mechanical, and in the other model acoustical coupling provides a number of practical and acoustical advantages.

INTRODUCTION

Organs have traditionally been heard in large reverberant rooms. This has affected organ composition and even the technique of playing. Organists prefer to have the organ console far enough away from the organ tone-chamber for listening to the combination of organ and room. In a sense they are playing the room as well as the instrument. This can easily be demonstrated.[1]

Organ music, whether pipe, electronic, or some other type, is enhanced by reverberation. In recent years organs have become economically feasible for use not only in small churches, chapels, and studios having very short reverberation periods, but in homes where parlor size and furnishings preclude effective reverberation.

The first reverberation means used on organs having electric output,[2] was similar in concept to the reverberation systems employed for recording and broadcasting purposes. Such systems consist of a driving transducer, a multiple time-delay means, one or more pickup transducers, and mixing and amplifying circuits. The devices described in this paper eliminate much of this complication and, consequently, have a decided economic advantage to the domestic user.

One type[3] consists of an otherwise conventional direct-radiator loudspeaker, coupled mechanically to helical coils of wire mounted in front of the cone. Vibrational energy absorbed by the reverberation coils, directly from the motion of the loudspeaker voice-coil, is transmitted back to the loudspeaker cone at a later time, and radiated as reverberant sound.

In a second type,[4] similar reverberation coils are coupled mechanically to a separate cone, which is smaller than the loudspeaker cone and mounted concentrically in front of it. Acoustical coupling provides

* Manuscript received April 7, 1954.
** Now at the Physics Department, University of Cincinnati.

[1] Daniel W. Martin, The Enhancement of Music by Reverberation, *Conv. Rec. IRE*, 1954, part 6, p. 4.

[2] L. Hammond, U.S. Patent 2,230,836; 1941.
[3] Armand F. Knoblaugh, U.S. Patent applied for.
[4] Daniel W. Martin, U.S. Patent applied for.

the driving force for the reverberation cone-coil combination, which is an accessory to the conventional loudspeaker.

It is ironic that acoustical physicists and engineers, after years of effort aimed at dispelling the old notion that wires strung through auditoriums would improve the acoustics, would use wire wound compactly into coils in order to produce a simulation of large-room acoustics.

THEORETICAL CONSIDERATIONS

Helical coils of wire were used twenty-five years ago by Wegel,[5] in mechanical speech-wave transmission lines for time-delay purposes. Resistance termination was used purposely in order to prevent wave reflection and resonance effects. The velocity of the wave transmitted by compressional vibration of the end of the coil is given by the equation

$$V = \frac{d}{\pi D^2}\left(\frac{E_s}{2\rho}\right)^{1/2} \qquad (1)$$

in which

V = propagation velocity in turns/second
d = diameter of the wire in cm
D = mean diameter of the helix in cm
E_s = shear modulus of the material, dynes/cm²
ρ = density of the material, grams/cm³

For infinitely long coils the characteristic impedance in mechanical ohms is given by the equation

$$Z = \frac{\pi d^3}{4D}\left(\frac{E_s \rho}{2}\right)^{1/2} \qquad (2)$$

The impedance of a partially damped coil of finite length varies with frequency quite sharply from this value, because of multiple resonances. The frequencies of the normal modes of vibration of the coil are

$$f = \frac{nV}{4N} \qquad (3)$$

where

f = frequency in cps
n = an odd integer when the coil end is clamped, and an even integer when the coil end is free
N = number of turns in the helix

The "room" which the reverberation coil simulates is of course one-dimensional, neglecting torsional and transverse modes. Consequently the density of normal frequencies along the frequency axis is uniform, in contrast to the three-dimensional rectangular room, in which the distribution follows the well-known square law[6] function to a first approximation. This is not too serious a disadvantage in the frequency range where a real room becomes replete with normal frequencies, because organ music typically contains complex tones, some harmonics of which will lie near enough to normal frequencies of the coil to cause excitation. Actually some torsional and transverse modes are also excited in practice.

At a high frequency (corresponding to the reciprocal of V), where a wavelength approximates a single turn of the helix, the compressional wave starts to disappear from the scene. Although this sets an upper limit on the range of normal operation, some irregular reverberation response is obtained at higher frequencies where, in fact, much of the radiation occurs from the coils itself instead of the cone.

[5] R. L. Wegel, U.S. Patent 1,852,795; 1932.

[6] Morse, "Vibration and Sound," pp. 291-297, McGraw-Hill Book Co., 1936.

DIRECT COUPLED MODEL

The main features in the construction of the direct-coupled model are shown in Fig. 1. A perforated metallic driving-dome, cemented to the cone near the voice-coil form, replaces the conventional paper dust-cap of the loudspeaker. A Rivnut expanded within the central hole of the dome provides connection for the machine-screw coupling the dome to the driven end of a group of four reverberation coils. In some models two of the coils are rigidly terminated at a cast bronze support ring, and the other two are terminated in a dynamically free, "hairpin" connection to the ring for static support. This permits both even and odd-numbered modes of vibration to be excited.

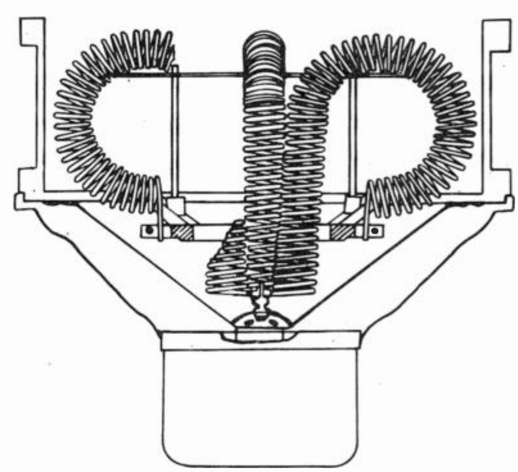

Fig. 1 — Direct-coupled reverberation loudspeaker.

Nylon cords, tied from points along the coils to small metal posts projecting from the support ring, minimize extraneous low-frequency transverse modes and provide some support for the coils under abnormal condition of shock during local transport. The coils are rigid enough to support themselves and to maintain the preformed shape shown, during normal operation. A cylindrical wooden housing surrounds the coils and connects the loudspeaker rim to the mounting baffle. By curving the coil axes, the over-all increase in depth is held to less than six inches for a loudspeaker of the fifteen-inch size.

In the design shown the combined mechanical impedance of the group of coils matches approximately the mechanical impedance of the driving loudspeaker between 300 and 400 cps. At the natural frequencies of coil vibration the coil impedance is very high, reducing the steady-state response of the loudspeaker locally at these frequencies. Between these frequencies the normal loudspeaker response-frequency characteristic is retained with a moderate reduction in over-all sensitivity. Because of the reduction in sensitivity the direct-coupled model is always used in combination with another conventional loudspeaker, which dominates the steady-state organ tone.

The lowest frequency of resonance in this design is approximately ten cps for the coils clamped at one end. The lowest corresponding frequency for free-end coils is twenty cps. Thus the frequencies of available compressional modes are multiples of ten cps, giving a rather dense distribution of resonances for musical excitation. If the coil had a much lower first frequency the time delay for the first reflection might be excessive in musical selections involving tones of short duration, giving the effect of a distinct echo.

If the coil had a much higher frequency for the first mode, it would give a sparser distribution of resonances and consequently, would decrease the probability of excitation of each mode. Thus the ten-cycle separation appears to be a good compromise.

The reverberation period measured for this design, using warble-tone excitation, averages approximately four seconds in the frequency range below 1200 cps. Fig. 2 shows a simple comparison of reverberation curves, one of which (A) was recorded at 500 cps in a large music hall, and the other (B) near the reverberation loudspeaker in a small anechoic room. This is not to imply that the over-all effects were identical, but it does show a degree of similarity.

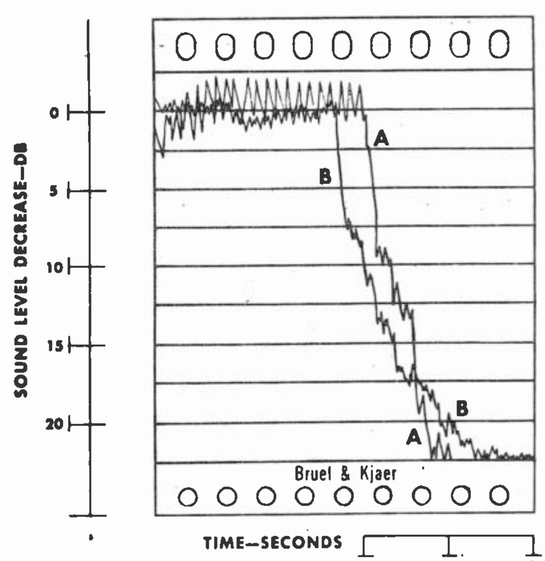

Fig. 2 – Comparison of sound decay curves in (A) large music hall and (B) from reverberation loudspeaker.

ACOUSTICALLY COUPLED MODEL

Fig. 3 shows the general construction of an acoustically coupled model which possesses several acoustical and practical advantages over the direct-coupled model. A separate cone is mounted coaxially in front of the conventional cone, and serves to support and drive the inner end of the reverberation coils. The outer cone also radiates the reverberant sound. At the lower frequencies it is rather transparent acoustically. The driving dome is perforated to transmit the higher frequencies, which normally are radiated chiefly from the central part of the inner cone. Direct sound is also radiated from the annular opening around the edge of the outer cone.

Because the entire reverberation cone-coil structure is supported on the cylindrical shell enclosure, this assembly may be considered an accessory unit which is separable from the conventional loudspeaker which drives it. This is an advantage for shipping, and simplifies the addition of the simulated reverberation feature to existing loudspeaker installations.

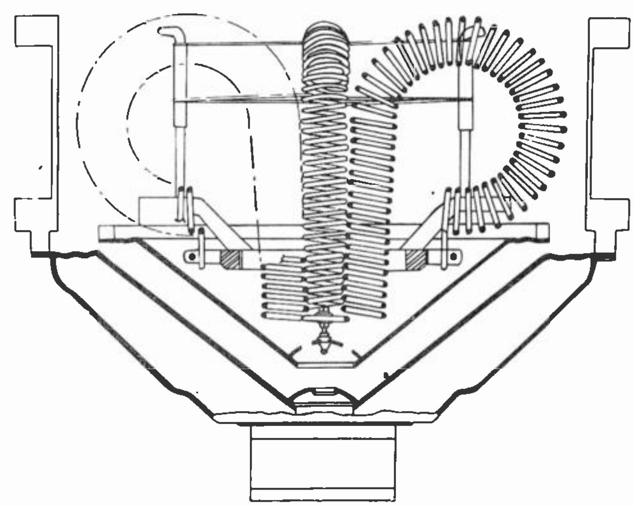

Fig. 3 – Acoustically-coupled reverberation accessory for loudspeakers.

Loudspeaker voice-coil suspensions are designed for high compliance along the axis of motion, and low compliance in the plane normal to this axis. It is the latter factor which permits the direct-drive model to partially support the weight of the driving end of the coil-structure when the axis is horizontal. The direct-drive model has not been used with the axis vertical, because the cone would gradually yield along the axis, displacing the voice-coil from its normal position in the gap. In the acoustically-driven model either vertical or horizontal mounting is possible without reaction upon the voice-coil.

The acoustically-coupled model does not require an additional conventional loudspeaker to supplement its steady-state acoustic output. The economic advantage is obvious. Fig. 4 compares the steady-state response-frequency characteristic on the axis of a fifteen-inch loudspeaker on a large flat baffle, with and without the reverberation accessory. The response below fifty cps is unaffected. This can be attributed to the annular opening around the outer cone, which prevents the stiffness of the outer assembly from raising the principal cone-resonance frequency. From fifty to fifteen-hundred cps the response changes are small and variable in sign. Above 1500 cps there is an average loss of approximately five db on the axis, but part of this results from the diffusion of the high-frequency sound waves, a condition considered desirable for organ music.

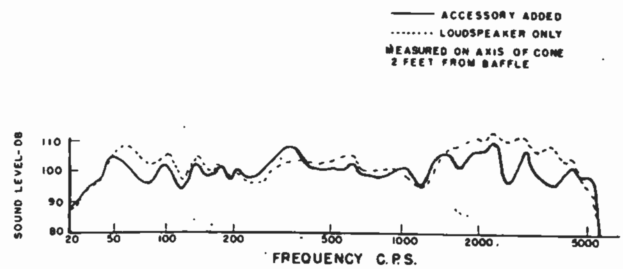

Fig. 4 – Steady-state response-frequency characteristics of a fifteen-inch loudspeaker with and without the reverberation accessory.

CONCLUSION

A simulated reverberation effect can be provided for organs radiating electroacoustic output in non-reverberant spaces, by the simple addition of a loudspeaker accessory of one of the types described.

ACKNOWLEDGMENT

The encouragement and suggestions of Mr. John Jordan, and the assistance of Mr. R. K. Duncan in the development of the acoustically-coupled model, are gratefully acknowledged.

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COMMENT ON FLUTTER STANDARDS*

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This invited technical editorial, by a pioneer in the development of loudspeakers and sound recording, and reproducing equipment, is related to a standard published in the March, 1954 issue of PROCEEDINGS of the IRE, pp. 537-541. However it is a plea for futher fundamental research on the subject. — *Editorial Committee.*

The March 1954 issue of the IRE PROCEEDINGS carries a set of definitions and specifications, under the title "I.R.E. Standards on Sound Recording and Reproduction: Methods of Determining Flutter Content." This standard, designated at the time as Z57.1/68 of the American Standards Association, was approved October 15, 1953 by the Standards Committee of the I.R.E. It had received the approval, earlier in 1953, of the Society of Motion Picture and Television Engineers, and on March 16, 1954 was approved as an American Standard.

The need for standards with respect to use of terms related to flutter, methods of measurement, and manner of specification, was recognized by the Sound Committee of the S.M.P.E. (Now S.M.P.T.E.) which in 1947, under the chairmanship of Dr. John G. Frayne, drew up proposed standards, which were published in the Journal of that Society in August 1947.

That Fall a committee of the American Standards Association, designated as Z57, was formed to work out standards in the entire field of sound recording and reproduction. Mr. George Nixon of NBC was chairman. The undertaking was sponsored jointly by S.M.P.E. and IRE. At a meeting of the Z57 committee in October 1947, the writer, who was one of the members representing the S.M.P.E., was asked to act as chairman of a subcommittee to recommend standards for measurement of distortion in sound recording and reproduction. Flutter would obviously come within this assignment. One of the first results of the undertaking was that, as thoughts on various phases of the subject of distortion measurement began to crystallize, the writer prepared a paper for the S.M.P.E., setting forth his reflections. The paper was published in the November 1948 Journal, under the title, "Proposed Standards for the Measurement of Distortion in Sound Recording." In that paper are given proposals for measurement of signal-to-noise ratio, which it is still hoped will eventually receive consideration and be used, with modification if need be. The part of the paper dealing with flutter is a statement of the considerations justifying the choice of a "root-mean-square" figure for per cent flutter, rather than peak or average. There was also a comment on the proposed "Flutter Index."

Unaware, at the time, of the manner in which a proposed standard would have to be handled before it could be adopted by A.S.A., the writer prepared a new draft of "Flutter" specifications, based on the Sound Committee proposal, but departing wherever a change would more nearly represent his own ideas and those of members of his A.S.A. subcommittee. It then developed that for acceptance by S.M.P.E. the concurrence of the original Sound Committee would be needed. To that end, a special committee, Dr. Frayne, Messrs. R.R. Scoville and J. K. Hilliard, was appointed to represent the Sound Committee. There followed a protracted correspondence and a number of revisions, but all around approval was finally reached, with the specifications as given in the Z57.1/68, the one considered and approved recently by the I.R.E. Standards Committee.

The item about which there was most correspondence is that which appears in Appendix 1 of Z57.1/68, namely "Flutter Index." This term, with definition, and the formulas essentially as they now appear, were included in the proposed standards published in the August 1947 Journal, with the purpose of giving readers the benefit of what had been learned about flutter perception thresholds, in an important series of tests in the Bell Telephone Laboratories. The results of those tests had been published previously in a paper on "Analysis of Sound Film Drives" by Alberscheim and MacKenzie in the November 1941 S.M.P.E. Journal. They were reprinted in the August 1947 Proposed Standards, and are shown here as Fig. 1.

In the Standards Z57.1/68, the definition states that "Flutter Index is a measure of the perceptibility of frequency modulation of a single tone." But the suggested formulas are required to complete the definition. The formulas given under Flutter Index in effect describe it as the measured flutter, multiplied by a factor which varies inversely as the threshold, using as threshold the values shown in the curves of Fig. 1. For simplicity's

*Manuscript received April 5, 1954.

sake, only approximations to the inverse threshold relations are attempted.

In the tests on which these curves are based, the observers were in what has been described as a moderately live room, and each observer indicated (as the magnitude of the frequency variations was changed) when he could just distinguish the frequency-modulated tone from a steady tone of the same average frequency.

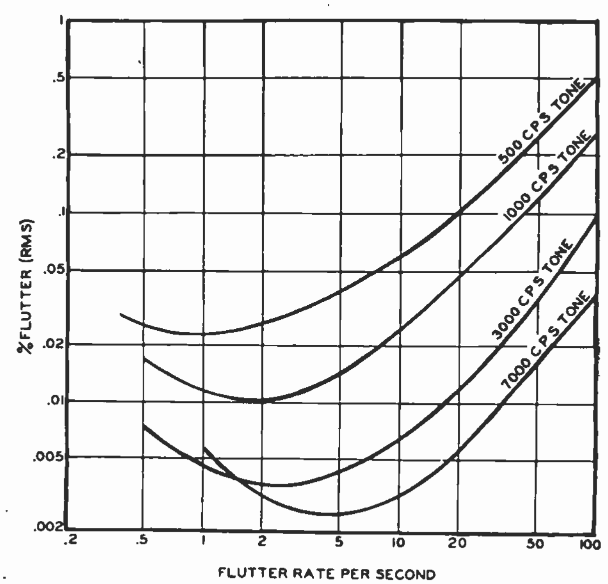

Fig. 1 Minimum perceptible per cent flutter for oscillator tones in small auditorium.

The thresholds were determined for 500, 1000, 3000 and 7000 cycle tones and for modulation or flutter rates from one-half to 100 cycles per second. Each of the family of curves in Fig. 1 shows the perception threshold, for the test tone indicated, plotted as rms per cent frequency modulation, against rate of repetition of the modulation cycle. It will be noticed that all of the curves reach minimum somewhere within the range 1 to 5 flutter cycles per second, and that in the range above 5 cycles the threshold rises almost in direct proportion to the flutter rate. The threshold was found to occur at about the same number of cycles of deviation from average frequency for the various tones, and therefore if expressed as percent flutter, varies about inversely as the frequency of the tone. Another way of describing the relations above 5 flutter cycles per second, is to say that threshold was reached at about the same total phase shift for all of the tones and flutter rates.

The ratio of the measured flutter to the threshold is taken as a "measure of perceptibility," as in the definition, and the formulas are designed to make the calculated Flutter Index directly proportional to this ratio. This is done by multiplying the measured flutter by a factor which corrects for the difference in threshold under the conditions in question (tone frequency and flutter rate) and threshold under conditions chosen as reference. The reference conditions are with a 3000 cycle test tone (the tone which is standard for flutter testing) and at a flutter rate between 1 and 5 cycles per second in which range the threshold is lowest. For these conditions the multiplying factor is unity, which makes the flutter index simply equal to the measured flutter. With a flutter rate at which the threshold is twice that at reference, the Flutter Index would be half the measured flutter.

Those who questioned the desirability of including the material on Flutter Index did not doubt the validity or significance of the tests for the conditions they represented, but feared that they might be misconstrued, as applying to flutter perception in general, as for example when listening to music. To forestall any such misinterpretation some changes in the definition and explanatory note were made, to make clear the conditions of the tests. The fear that any major error would be made was scarcely justified by any material which appears in the Standard under "Flutter Index" for no indication is there given of the absolute magnitudes of the flutter thresholds shown in Fig. 1, but only the general shape of the curves. There is some support in practical experience for the belief that with actual music the threshold is lowest for slow flutter or "wow", and is higher for rapid flutter. Therefore the effect of rate on threshold is very likely at least roughly similar to that indicated by the curves of Fig. 1. A much greater error might result if any one were to assume that the absolute values of the experimental thresholds from which the Flutter Index formulas were derived were generally applicable to program material. However nothing is said in the adopted Standard which gives any support for such an assumption, nor which indicates absolute values, so whatever misgivings may have been felt on this score, as to the desirability of publishing the material on Flutter Index, were probably groundless.

The Flutter Index relations should be helpful to those who are judging flutter by listening to test tones, to better interpret what they hear, especially if their test tones are not standard (1000 cycles, for example, is sometimes used) or if the predominant flutter is rapid. It is hoped that the presentation of this information will also serve to draw recurrent attention to the need for more information about flutter threshold.

The only other published data on frequency modulation thresholds so far as our committee knew was that given by Shower and Biddulph in a paper on the "Differential Pitch Sensitivity of the Ear," in the October 1931 Journal of the Acoustical Society of America. Shower and Biddulph were seeking only to find the rate at which the ear would be sensitive to the smallest changes, and having found a definite minimum threshold at about two cycles and a rapid rise (ear less sensitive to change) at rates above four or five cycles, they did not carry their tests much beyond that. They worked with a 1000 cycle tone using headphones. The interesting point is that the thresholds so found were about ten times higher than those shown in Fig. 1 for the same tone frequency and modulation rates.

One of the members of our A.S.A. subcommittee was Dr. Harry Schechter, at the time working at the Acoustics Laboratory at the Massachusetts Institute of Technology on a project for the U.S. Air Material Command, as part of which he had made extensive tests on flutter thresholds with monaural listening, using a group of subjects, and several test frequencies. For the frequency and range of modulation rates covered by Shower and Biddulph, the agreement was as near as would be expected, and the Schecter data constitutes an extension of the headphone investigation of flutter thresholds. Insofar as it is permissible to draw general conclusions, it appears that compared with loudspeaker listening, in addition to much higher thresholds, the headphone listening gives a less rapid rise in threshold with increase of flutter rate above 5 per second, and (based on a comparison of tests at 250 cycles and 1000 cycles) the threshold is found at about the same per cent frequency modulation for both tones; whereas with loudspeaker listening the threshold was found at a per cent frequency modulation which varies inversely as the frequency of the test tone.

It is well known that sustained tones in a moderately live room build up standing wave patterns, with regions of strong maxima and minima, and that very slight changes in frequency cause radical shifts in the positions of these regions. This, in conjunction with the much higher thresholds with earphones, gives grounds for believing that the very low thresholds found in the tests with a loudspeaker in a live room, may represent observations by intensity changes rather than actual sensitivity to frequency changes.

It would seem on first thought, that since the tests with loudspeakers were for acoustic conditions similar to those under which most music is reproduced, they would constitute a better guide as to the flutter tolerances and the manner in which the tolerance is related to flutter rate, than would tests with earphones. On the other hand it is not established that intensity fluctuations of themselves are objectionable. Certainly they do not make music sound "sour" as do slow flutter or "wows." The music of orchestras, organs and choruses comes from numerous sources at once, which are not exactly synchronized, and so must produce beats or intensity fluctuations of great complexity. Therefore when listening to such musical notes, intensity fluctuations caused by shifting standing wave patterns must, if perceived at all, be perceived in the presence of other fluctuations having what we may call "legitimate causes." Can there be much question that the threshold for the unwanted fluctuations would be considerably raised by the presence of the normal ones? It can hardly be thought that the intensity fluctuations arising from flutter would have a different quality (owing to their origin as frequency changes) and therefore be distinguishable, if the frequency changes themselves are well below thresholds for frequency changes when these are not magnified by transformation into intensity changes. The threshold without such transformation is indicated by the headphone tests.

Even the musical tones of solo instruments or voices are extremely complex. Whereas in loudspeaker listening the standing waves of a single or pure tone would establish a definite pattern of maxima and minima, the pattern for each component of a complex tone would differ from the patterns of the others, and since with the frequency changes due to flutter, the component tones would not rise and fall together, what might be expected would be a cycle of changes in quality, with perhaps little change in loudness. That threshold for such a quality cycle would be as low as the rise and fall of intensity in one component alone is doubtful. Only tests can show how much the difference might be.

When a steady tone is produced, the standing wave patterns are not established until the tone has continued for a time comparable with the reverberation time of the room. Even though in certain musical compositions notes or chords may be sustained this long, the probability of a listener's observing variations within such duration are much less than when he is given as much time as he wants. Assuming that a listener is concentrating his attention on whether there is or is not a detectable effect of flutter, his perceptive faculties must be readjusted every time the note or chord changes.

For all these reasons it seems clear that the relation between flutter thresholds under the three conditions: (a) with pure tones from a loudspeaker in a representative listening room, (b) with music in such a room, and (c) with pure tones with headphones, are largely a matter of speculation and guesswork. The writer's personal guess is that (b) will be found to be much closer to (c) than to (a). Much more experimental work is needed before we can state with confidence what amount of flutter at various rates is barely perceptible and what tolerable.

Such knowledge would be valuable to manufacturers and users of audio recording or reproducing equipment, in writing specifications with respect to flutter more intelligently, in interpreting and judging the significance of flutter measurements, and in designing and building equipment which will be acceptable but not unnecessarily expensive. It would put the control of flutter more nearly on a scientific and engineering basis, in place of the system of trying a product on the public and repenting if it is unacceptable.

In arguing that practical thresholds for flutter are probably much higher than those for steady tones, the writer is by no means seeking to relax standards. As a matter of fact the extremely low figures for threshold indicated in the steady tone and loudspeaker tests (Fig. 1) are far below the flutter values attained in even very high-grade commercial equipment. (We understand that they have recently been approached or equaled in certain tape equipment built to meet very exacting requirements not directly related to sound reproduction).

Since the one way in which reliable information can be obtained, on actual flutter threshold, with speech and various types of music, would be by a comprehensive series of tests, it is very much to be hoped that such investigations will be undertaken. They could be research projects of some of the larger companies interested in audio equipment, or they could be university projects, financed jointly by a group of interested companies. One such investigation is being carried on by the Bureau of Ships*. Publication of findings will be awaited with great interest. It is still very desirable, in view of the complexity of this important problem and the large number of observations needed for drawing reliable conclusions, that similar projects be undertaken by other organizations.

The normal approach to such an investigation would be to introduce flutter into a recording-reproducing system which is otherwise free from flutter. The last stipulation is of course unattainable. but at least the flutter must be well below threshold, and the quality must be particularly clean, and low in distortion.

Precautions would be needed to be sure that the operation which introduces the flutter does not introduce some other cyclic variation. For example, if a disk is driven at fluctuating speed, there must be no vibration to affect the pickup, or make periodic variations in pressure. If a magnetic tape system is not used it would be essential to make tests to prove that the speed modulating system does not affect the evenness of contact nor the contact pressure between tape and reproducing magnet. A photographic sound system (which is a pure amplitude system) would have the advantage that changing speed would cause no amplitude modulation, and if a deep-focus optical system is used, the scanning would be in small danger of being affected by other than longitudinal movements of the film. However, it may turn out that some of these precautions are unnecessary, and it may be possible to prove by tests with pure amplitude modulation that the accidentally introduced output voltage changes are far below threshold and can safely be neglected. If that is true, the choice of system can rest more on convenience.

Either friction drives or thread-belt systems can probably be made to give sub-threshold flutter when the modulating system is inactive. It might prove desirable to cover low flutter rates with one mechanism, and high rates with another.

Judgements will be difficult, and the number of types of program material could easily be formidable. Therefore the schedule would need to be reduced to short basic types, and flutter rates limited to a few discrete values. Something significant would be revealed, for example, if under normal listening conditions the thresholds were determined for intermittent pure tones with various off-on intervals, then with no silent periods but a sequence of several tones, the test being repeated at different note lengths. Then a similar series of tests could be made with complex tones. In a set of tests such as this, the sources could be electronic, with purely electronic (instead of mechanical) methods of modulation.

It would certainly not be in order to try to make determinations with any great precision until work has progressed to the point of proving that significant results are possible, and greater refinement justified.

The foregoing thoughts are offered in the hope that they will help interested persons to visualize what such a project might be like. It seems to the writer that even a very incomplete investigation of thresholds for music and speech would be so much better than the present sporadic observations and guesswork that an effort to get such a project started would be amply justified.

*Navy Research Project NSS-683-034(10), Bureau of Ships (Code 565).

"HIGH-FIDELITY" in MUSICAL TONE PRODUCTION?

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The tremendous surge of interest in "high-fidelity" sound during recent years has attracted the attention of many manufacturers in commercial fields peripheral to the manufacture of sound equipment. Recently (and somewhat paradoxically) several manufacturers of musical instruments have attempted to capitalize on "hi-fi" interest. For example, electronic organs are sometimes advertised as having high-fidelity amplification equipment, and the magic of the term "high-fidelity" has even been borrowed by advertisers to describe the design of a piano soundboard. Before this trend in loose terminology develops further in the field of musical tone *production*, the origin and meaning of "high-fidelity" in relation to the *re*production of musical sound should be carefully examined.

In the minds of typical discriminating consumers of both tone *production* and *re*production equipment, the term "high-fidelity" is something relatively new and rather mystic. Actually, twenty-five years ago "High-Fidelity" originated as a registered trade-mark of a loudspeaker manufacturing company which, incidentally, is one of the present commercial leaders in the "hi-fi" field. Only a few years later there were several manufacturers producing sound equipment which, in many electrical and acoustical characteristics, equalled the best sound equipment available today! This equipment was used chiefly for reproduction of sound-on-film recordings for motion pictures, and to a lesser extent in the radio broadcasting industry. Such equipment was not mass-produced in the usual sense. The lack of a mass market and the high cost of equipment production were inter-dependent, and one can hardly say which was cause and which was effect. The disc recordings available to music lovers at that time contained non-linear distortion and noise, in amounts which discouraged the use of sound equipment which was capable of reproducing the entire audio-frequency range.

During the early and middle nineteen-forties the use of magnetic recording and playback equipment, and the development of improved record manufacturing techniques and materials, completely changed the picture. Through these means the general public could then obtain recorded music which was potentially of much higher quality than before. However, full appreciation of this potential required the use of electrical and electro-acoustic equipment possessing greater frequency range, better transient response, less non-linear distortion, and greater freedom from system noise. Fortunately for the new types of records, such equipment was already developed, and the major problems of the new audio equipment industry were largely the development of mass production methods and associated simplifications in design for systems and components already available. There have been, of course, further improvements in components and in technique, within the several years since the audio "boom" began. Research and development in the audio and acoustical fields continue, with renewed interest and financial support.

The public appreciates the new musical *re*production, and interested people typically ask: "What is high-fidelity? Who invented high-fidelity? Why can one buy high-fidelity equipment at so many different price levels?" These questions reflect a belief that high-fidelity is a "thing", which a piece of equipment either has or has not.

High-fidelity is a relative term. Probably this is not easy to explain to a prospective purchaser of high-fidelity equipment. Much has been said and written on the necessity for standards in high-fidelity equipment. The difficulty (perhaps impossibility) in standardization for "hi-fi" equipment is that various engineers and manufacturers have progressed along different paths towards the achievement of improved sound fidelity. Actually progress along any one of these lines can truly be termed a step toward "high-fidelity."

One enterprising audio engineer may push along a particular path of improvement farther than anyone else has gone. In doing so, he may have had the necessity of compromise on performance factors related to other paths of improvement. Other audio engineers at the same time are developing equipment in which the compromise is made in favor of other performance factors. Superficial examination of the problem might lead to the conclusion that "high-fidelity" is a combination of all of the possible improvements which have been made individually. Such all inclusive standards would probably result in a design impossibility. At the other extreme, standardization upon the minimum performance now permitted in each of the so-called "high-fidelity" reproducing equipments would degrade the term to a meaningless level. This is why it is difficult to define "high-fidelity" for sound *re*production equipment.

There is no similar problem of ambiguity with regard to "high-fidelity" in the original production of musical sound because, by definition, the original is the highest fidelity possible. This is not to say that it is impossible to improve upon the original (although this seldom occurs in the recording and reproduction process), but Gertrude Stein's classic statement "A rose is a rose is a rose..." applies equally well to a violin or to a trumpet.

To the extent that an electronic instrument is intended to *simulate* a different instrument, the fidelity of simulation does have meaning. However, in this usage the term "fidelity" should be applied to the instrument as a whole, and not to the amplification equipment alone (as has been the case). For example, the use of amplification equipment designed for high-fidelity reproduction of recorded music on one type of electronic organ which generates no signal components at frequencies above 5000 cps, will add nothing but noise to the upper range of the spectrum. By contrast a different electronic instrument, which generates an abundance of high-frequency harmonics, may advantageously employ a "roll-off" in high-frequency response of the amplification system as part of the overall voicing of the instrument. Moreover, fidelity of simulation is by no means the sole goal in electronic musical instrument development. In the sense that an electronic instrument is a new instrument, "fidelity" has no meaning at all.

Similarly statements concerning a "hi-fi" piano sound-board should be examined carefully. Do they mean that the sound radiated by the board corresponds exactly to the vibration of the strings which excite it? A person who has listened to the vibrations of a struck string, over an amplifying system having uniform response-frequency characteristics, will understand that the resulting tone is hardly identifiable as a piano. A soundboard is part of the instrument, and helps to create the tonal effect desired. To identify it simply as a sound transmission channel is an oversimplification. To term a soundboard "high-fidelity" is misleading, in this writer's opinion. Any part of a manufactured product may be modified or improved, but this has little to do with "high-fidelity" unless the product is part of a sound *re*producing system.

Improved fidelity of musical sound reproduction can be expected to broaden popular interest in original instruments and in the music they produce. Moreover many of the scientific principles and techniques, and engineering practices and procedures, discovered and developed for sound reproduction, can advantageously be adapted to the solution of musical tone-research problems and to the engineering of new music production equipment. Indeed, this has been occurring quietly for some time, beginning even before "high fidelity" was popularized. Similarly, knowledge gained from research in the field of musical tone has benefited musical sound reproduction, and ultimately will affect it greatly. Although the purposes of these two distinct fields (development of means for *creating* music and for *re*creating music) are similar, they should not be confused.

In summary it can be said that the goal of sound reproducer research and development has been to attempt to *re*create the original sound. Presumably one-hundred per cent fidelity would be the achievement of perfect reproduction of the original sound. Manufacturers of musical instruments should think this through before paying the high compliment of imitation to their own imitators. A music *production* system has inherent advantages in quality of performance over music *re*production systems (lower system noise and distortion, for example). Surely exploitation of such inherent advantages is a more positive and lasting policy for producers of music production equipment than borrowing publicity generated by producers of music reproduction equipment.

PHILADELPHIA CHAPTER ACTIVITIES

Murlan S. Corrington, Chairman
Philadelphia Chapter, PGA

At the regular meeting on March 16, 1954, Mr. Stephen A. Caldwell, Electronics Products Division, Radio Corporation of America, Camden, N.J., gave a lecture with demonstrations on "Multichannel Sound Reproduction." It was explained how multichannel sound systems increase the listening pleasure from recorded music. He discussed the effect of the directional characteristics of the loudspeaker and of the acoustical environment of the recording and reproducing setups. Several systems were demonstrated to show how the fidelity of reproduction, the amount of reverberation, and the relative time delays are all important in the design of a high quality system.

The final meeting of the year was held on April 22, 1954 with Dr. Winston E. Kock and Mr. Floyd K. Harvey of the Bell Telephone Laboratories, Murray Hill, N.J., giving a talk and demonstration on "Polarized Airborne Sound Waves." Although sound waves have generally been considered to be purely longitudinal and therefore not polarizable, it is possible to generate transverse sound waves having a definite "plane" of polarization. Such waves should be confined in hollow tubes to remain polarized. A demonstration of these properties was given, which included the rotation of the plane of polarization by "half-wave plates," the production of circularly polarized sound waves by "quarter-wave plates," acoustical filters, and converging lenses.

The new officers who were elected at the Annual Meeting are:

Mr. H. E. Roys, Radio Corporation of America, Chairman
Mr. Edwin C. Gulick, Philco Corporation, Vice-Chairman
Mr. William L. ten Cate, Custom Sound Associate, Secretary.

CINCINNATI FIELD TRIP TO DAYTON

E. M. Jones, Chairman
Cincinnati Section, PGA

The Cincinnati Chapter IRE-PGA for its final meeting of the season made a field trip to Dayton, Ohio to hear the Sunday evening concert in Carillon Park. Members of the chapter and their families enjoyed a picnic at a nearby park before the concert. The first half of the concert was played on the large carillon. A 5500-watt audio system called the "Celestron" is installed in the same tower as the carillon. The second half of the concert was recorded music played on the Celestron system. After the concert the PGA group was given an inspection trip through the tower installation.

At an earlier meeting on April 20, the Cincinnati Chapter heard a talk on "Wow in Recordings and Measurement Thereof" by Mr. Meredith L. Young of the

General Industries Co., Elyria, Ohio. Common causes of frequency variations in phonograph turntables were discussed. The circuits of "Wow" meters for measurement of these variations were presented. A demonstration accompanied the paper in which measurements were made upon phonograph turntables having various amount of frequency fluctuation. This gave the audience an opportunity to correlate audibility of "Wow" with precise measurements made on the meters.

The new officers of the Cincinnati Chapter IRE-PGA for 1954-55 are as follows:

Chairman: Wynne W. Gulden, Cincinnati & Suburban Bell Tel. Co.
Vice-Chairman: J. Park Goode, National Sound Service
Secretary-Treasurer: Richard Lehman, The Baldwin Piano Co.

PGA BRIEFS

Members of the IRE Section in Washington, D.C. have recently petitioned for the formation of a Chapter of the Professional Group on Audio. The Chapter formation has been approved by the Executive Committee of the Washington Section, by the Administrative Committee of the Professional Group on Audio and by the Executive Committee on the Institute. A hearty welcome to this new Chapter.

The Acoustical Society of America celebrated its twenty-fifth anniversary on June 23-26 in New York City. The Acoustical Society is a part of the American Institute of Physics. Although the Professional Group on Audio and the Acoustical Society of America are organizationally separate and have only a small overlapping membership, there naturally exists a strong bond of "first-cousin" relationship between the two. Congratulations for a quarter-century of fundamental contributions to acoustics and audio.

Inadvertently in the March-April issue the primary affiliation of Professor A. B. Bereskin, author of the paper "A High-Efficiency High-Quality Audio-Frequency Power Amplifier" was omitted. Mr. Bereskin is a Professor of Electrical Engineering at the University of Cincinnati, and a part-time consultant in electronics to The Baldwin Piano Company, where he performed the research reported in this paper.

The Administrative Committee of IRE-PGA met in New York City on June 26th, in order to discuss publication methods, schedules and costs, changes in bylaws, an interchapter memo service, joint meetings, increased cooperation with IRE technical committees, and formation of new chapters.

IRE-PGA Chairman, Dr. Vincent Salmon, announces that the following appointments have been accepted for the 1954-55 period:

Secretary-Treasurer—Benjamin B. Bauer,
Shure Brothers, Inc.
Chairman Editorial Committee—Daniel W. Martin,
The Baldwin Piano Company
Chairman Program Committee—Philip B. Williams,
Jensen Manufacturing Company
Chairman Tapescripts Committee—Andrew B. Jacobsen,
Motorola Labs.
Chairman Chapters Committee—Robert E. Troxel,
Shure Brothers, Inc.
Chairman Nominations Committee—Marvin Camras,
Armour Research Foundation
Chairman Awards Committee—John Hilliard,
Altec Lansing Corporation
Chairman Papers Procurement Committee—John Kessler,
Massachusetts Institute of Technology

This is the time of year when the use of tapescripts should be scheduled for coming Chapter meetings. For listings of available tapescripts, refer to the January-February issue of TRANSACTIONS of the IRE-PGA, page 2. One additional tapescript, entitled "How Much Distortion Can You "Hear," described in the March-April issue, has been added to the available tapescripts list.

The IRE-PGA Program Committee made an effort to stimulate audio papers for the October 4-6, 1954 conference of National Electronics Conference, Inc. This effort was partially successful, resulting in several audio papers on the program, but not enough for a special audio session. Commitments for papers for this conference must be made months in advance, at a time of year when authors are planning vacations. Our thanks to the Program Committee for its effort. The Administrative Committee of IRE-PGA will hold its annual fall meeting in Chicago at the time of the NEC. Members of IRE-PGA interested in presenting matters for consideration of the Administrative Committee, are invited to do so by correspondence with the Chairman or Secretary-Treasurer.

RECORDING AND REPRODUCING COMMITTEE

Murlan S. Corrington
Vice-Chairman, Committee 19

Technical Committee 19, which is working on IRE Standards for Recording and Reproducing, met at IRE Headquarters in New York on April 9, 1954. Three new members were appointed: Ellis W. D'Arcy, Marvin Camras, and Alvin H. Willis.

The revised Scope and new name of Committee 19 is as follows:

RECORDING AND REPRODUCING COMMITTEE

1. The selection of terms and the preparation and maintenance of standard definitions for complete systems for mechanical, optical and magnetic recording, and their components which include: the recording device, the reproducing device, the recording medium, and the drive mechanism.
2. The preparation and maintenance of standards covering methods of measurement in the above fields.
3. The coordination of activities with other IRE committees, other professional societies and liaison with technical organizations engaged in allied work.

Consideration was given to cooperation with the Standards Committee of the Audio Engineering Society, which has a Subcommittee on Disk Recording. They are working on: (a) Specifications for a test record, (b) Methods for calibration of test records, and (c) Production problems and quality control. Since both committees are working on similar problems, Committee 19 is to work with them whenever possible.

A new Subcommittee will be established to deal with problems relating to (a) Magnetic Record Media, and (b) Measurement of the State of Magnetization of Magnetic Record Media as a Function of the Recorded Signal.

A letter has been received from the Department of the Navy, Bureau of Ships, dated March 29, 1954 which indicates that the Materials Laboratory of the New York Naval Shipyard, Brooklyn 1, N.Y. is studying flutter in recording systems. They have proposed several minor changes in the "Methods for Determining Flutter Content, 1953" which were published in Proc. I.R.E., vol. 42, pp. 537-541; March, 1954.

NATURAL SOUND REPRODUCTION*

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There are many factors which affect the fidelity of reproduction of voice and music. The purpose of this article is to collect some of the important considerations in such reproduction. Reasonable approximations have been used in order to have definite numerical figures to make comparisons and to establish the general order of each factor. Two sets of reported listening tests have had a definite bearing on these considerations of overall fidelity.

A most interesting series of basic listening tests was performed by Dr. Harry F. Olson[1] of the RCA Laboratories at Princeton, New Jersey. He demonstrated that the complete frequency range was preferred by listeners to a frequency range limited by an acoustic curtain cutting off above 5,000 cps. Live sources of music were provided behind this curtain in one corner of a room and the audience listened in the normal (binaural) manner. During these tests the audience was at no time informed of the exact mechanism of the test. Comparisons were made of fidelity with the hidden acoustic curtain being either arbitrarily opened or closed. Almost invariably the listeners chose the full frequency range. A conclusion reached by Dr. Olson was that other listener preferences for a restricted frequency range in *reproduced music* were probably accounted for by the distortion and deviations heard with extended frequency range in the reproducing system.

The RCA test used no reproducing equipment between the listeners and the orchestra. However, in the case of binaural reproduction, where two channels are used with two microphones separated at approximately the distance between the ears, there are important advantages over a single channel system. One of these advantages is the apparent reduction of reverberation in the studio. A pair of ears also has the characteristic of virtually eliminating distortion of certain types which may exist in the two channels separately. Further, a binaural reproducing system allows the apparent angular separation of instruments of the orchestra, so that the listener can focus his attention deliberately on a particular instrument

*Manuscript received April 15, 1954.

[1] H.F. Olson, "Frequency range preferences for speech and music," Electronics, p. 80; August, 1947.

which is angularly separated from others, even by but a few degrees. By the same token, a binaural listener is capable of discriminating against a moderate amount of noise when it originates at an angle different from the desired sound. The sum result is that binaural listening, whether directly or through binaural microphones and amplifiers, has a naturalness which is very apparent after short experience. Numerous such tests have been made with experienced musicians, who react very favorably to even poor frequency-range recordings when made on a true binaural system.

The most extensive information available on listener preference with a reproducing system was reported by Howard A. Chinn and Philip Eisenberg of the Columbia Broadcasting System.[2,3] They made a series of careful listening tests with public participation. High-quality, specially-prepared program material was employed, which was reproduced on excellent equipment in a quiet room. Audiences voted on the frequency range preferred. The results showed a definite preference for a "medium range" from approximately 75 to 7,500 cycles, rather than a wider or narrower range. The narrowest range was limited in the high-frequency direction at about 5,000 cps. The results of Olson's work compared directly with the Chinn and Eisenberg results in that the audiences desired a range *exceeding* 5,000 cps in both cases. However, because the Olson experiments were not performed with a second curtain cutting off at 7,500 cps, it cannot be predicted what the result would have been. Presumably there would still have been some preference for the unlimited range.

There are a number of factors which help to explain the results for preference of a medium-frequency range in reproducing systems. In order to explain this, the expected results from three types of equipment will be compared as follows:

A. A typical radio-phonograph, called the "usual receiver," which has a frequency range of approximately 75 to 3,500 cps with a 10" loudspeaker. Such a receiver does not cut off sharply below 75 cps, although the acoustic efficiency is dropping rapidly with decrease in frequency.

B. A "high fidelity" amplifier covering a frequency range of approximately 30 to 15,000 cps with a 14" loudspeaker with conventional baffle, and a tweeter.

C. A "medium fidelity" amplifier covering from 75 to 7,500 cycles with *three* conventional, dynamic speakers of about 16", 4" and 1" in diameter, respectively, mounted on conventional baffles. (Such a system was in constant use for comparison purposes.)

TABLE I

Frequency Range Comparisons

Comparison of:	Usual Receiver (*75-3500 cps)	High Fidelity (30-15000 cps)	Medium Fidelity (75-7500 cps)
Quality loss of each range	7% loss lows 45% loss highs	Virtually no loss	8% loss lows 8% loss highs
Network program fed directly to each audio amplifier	12% loss lows 35% loss highs	4% loss lows 25% loss highs	8% loss lows 25% loss highs
Local program through selective receiver	12% loss lows 45% loss highs	6% loss lows 45% loss highs	8% loss lows 45% loss highs
Phonograph records-average pickup	10% loss lows 45% loss highs	5% loss lows 45% loss highs	8% loss lows 45% loss highs
Phonograph records-good pickup	8% loss lows 40% loss highs	4% loss lows 8% loss highs	8% loss lows 8% loss highs
Local program—wider selectivity receiver	— —	6% loss lows 6% loss highs	8% loss lows 8% loss highs

*But does not cut off sharply below 75 cps as does the 75-7500 system.

Table I shows frequency-range comparisons with respect to the loss of low and high frequency due to the system. The percentage loss in quality of the reproduction of orchestral music is a concept used in an investigation by Snow.[4]

The figures used throughout this article have been taken from many sources, with some estimates where no data was found, or where data does not agree. Reproducing systems corresponding to the three systems below have been compared at various times, but the three systems were not constructed for direct comparison as one experiment.

[2] H.A. Chinn and P. Eisenberg, "Tonal-range sound-intensity preferences," Proc. I.R.E., vol. 33, p. 571; September, 1945.

[3] H.A. Chinn and P. Eisenberg, "Influence of reproducing system on tonal-range preferences," Proc. I.R.E., vol. 36, p. 572; May, 1948.

[4] W.B. Snow, in: Jour. Acoust. Soc. Amer., vol. 3, no. 1, pt. 1, p. 155.

The first line of Table I shows the estimated quality loss of each range due to frequency-range restriction alone. The second line concerning a network program shows such additional loss of each system if the network program line were fed directly through each audio amplifier and speaker. The loss of high frequencies in this case for the high fidelity and medium fidelity amplifiers is due to the usual practice of limiting most network programs to about 5,000 cps as their highest frequency. The next line involves a local program received by radio through a selective receiver and shows the high frequency loss contributed by conventional selectivity. This effect is minimized in the last line where the program has been received through a wider selectivity receiver on AM and this also applies to a local program received by any reasonably good FM receiver.

So far, it can be concluded that the medium fidelity amplifier is more than adequate for most network programs (assuming an upper limit at 5,000 cps). It is certainly adequate for all local programs (upper limit 10,000 cps) and for phonograph records (upper limit about 10,000 cps on carefully made records). Actually little music is available through any of the usual sources at frequencies much above 7,500 cps, except for some local FM programs which extend to 15,000 cps or on good transcriptions or live programs.

Unfortunately, there are certain noises which disturb listening. During the Chinn and Eisenberg experiments it was found that any slight noise in the reproducing system was immediately objectionable to the listeners. Noise can be classified as hum, turntable rumble, record scratch and electrical interference (such as static, pops, and clicks).

It is very interesting to observe the high-fidelity growth in popularity through 1952, 1953, and 1954 which is probably as much due to reduction of background noise as extension of frequency range. Time and again musicians have reacted very favorably to limited range reproduction free of background noise, and unfavorably to extended range with some, even slight, background noise. The satisfaction of clean silence during pauses in music must not be underestimated.

Table II shows the relative intensity of high frequency noises in amplitude for the same three amplifier and speaker arrangements. The table refers to the amount of noise which will be reproduced in comparison to the noise present within the program frequency range. It will be noted that the "usual receiver" reduces noise considerably since its frequency range is less than that of the program range in every case. However, Table I shows the frequency range penalty for so doing. It is no accident that the usual receiver is designed in this way, because noise is so objectionable. The high fidelity amplifier will produce noise outside of the program frequency range as shown in Table II. Of course, this assumes that noise does exist with the program as is usually the case. Record scratch at each frequency actually increases with frequency increase. Thus, the amounts of scratch reduction in the "usual receiver" is more than the frequency range reduction alone would indicate, and conversely the high fidelity unit has more than proportional scratch.

TABLE II

Intensity of High Frequency Noises

Comparison of:	Usual Receiver (75-3500 cps)	High Fidelity (30-15000 cps)	Medium Fidelity (75-7500 cps)
Network program directly to amplifier	Noise reduced to two-thirds	Noise tripled	Noise increased 50%
Local program through selective receiver	Noise reduced to one-third	Noise reduced to one-third	Noise reduced to one-third
Phonograph records-good pickup	Noise reduced to one-half	Noise doubled	Noise normal
Local program – wider selectivity receiver	– –	Noise increased 50%	Noise normal
Record scratch	Scratch reduced to one-third	Scratch doubled	Scratch normal

The reception of radio programs from AM radio stations, within a few miles of a receiver, is such that little high-frequency noise will be heard. FM reception will usually render noise-free programs for a greater distance, particularly under summertime conditions. Recorded music depends upon the type of record and will have clicks, pops and scratch, which on only the quietest records will be wholly acceptable.

There are other types of noise at the low frequencies and these are principally hum and turntable rumble. The harmonics of power line hum will give trouble in any case. Table III shows the effect of the 60 cycle power hum alone. Naturally, the best turntables are better than cheaper ones with respect to rumble.

Let us review the situation thus far. As far as frequency range is concerned, the medium fidelity amplifier was almost equal to the high fidelity amplifier for most available programs. High-frequency noise was much less in evidence on the medium fidelity amplifier in comparison to the high fidelity amplifier. The usual receiver was best of all for noise, but its frequency range is very limited from the standpoint of high fidelity. Lastly, the effect of hum and turntable rumble do favor the use of a medium fidelity amplifier.

TABLE III

Audibility of Hum and Turntable Rumble

Comparison of:	Usual Receiver (75-3500 cps)	High Fidelity (30-15000 cps)	Medium Fidelity (*75-7500 cps)
Power hum in receiver	some present	Negligible	Negligible
Power hum with program	Will be heard	Will be heard	Almost none
Turntable rumble	Some present	Considerable present	Almost none

*Cuts off sharply below 75 cps.

Now, let's examine the problem of reproduced distortion in Table IV. The figures for the usual receiver were taken from an average of measurements on a number of home receivers commercially produced during the year 1950. Included in the distortion is that part contributed by the loudspeaker which is very important, particularly neglected in making distortion measurements. Amplifier distortion alone is no criterion for overall performance in reproduction.

The low-frequency threshold of hearing is about 50 cps in a quiet residence with ordinary levels of reproduction. Frequencies lower than 50 cps will not be heard at ordinary levels. For this reason, also, little can be heard between 50 and 75 cps except distracting power noise at 60 cps and turntable rumbles extending up to about 70 cps. When one listens to a series of harmonics, the fundamentals will be recreated in the mind. The next to the last line of Table IV shows that the lowest notes apparently heard are often created by distortion due to the unloaded, loudspeaker-diaphragm flopping. When one listens to a series of harmonics in the music itself, the fundamentals will be recreated in the mind by this same process. The last line of Table IV shows the lowest frequency which will probably be so created by the natural harmonics in the music itself. Thus, the medium fidelity amplifier will apparently produce notes down to roughly 40 cps, which is ample. Furthermore, because it cuts off below 75 cps the cone is prevented from excessive travel which otherwise produces a serious amount of distortion. Parenthetically it may be said that the exponential horn is almost a necessity for distortionless reproduction below about 150 cps as it keeps the speaker cone properly loaded above its cutoff.

TABLE IV

Effect of Harmonic Distortion
(especially at lower frequencies)

Comparison of:	Usual Receiver (75-3500 cps)	High Fidelity (30-15000 cps)	Medium Fidelity (75-7500 cps)
Distortion tolerable – entire system (1000 cps)	10%	3% (hard to achieve)	5% (possible with care)
Distortion present (2 watts) at lowest note reproduced	25% (or more)	30% (even with care)	3% (or less)
Lowest note heard as fundamental – quiet residence	85 cps	60 cps	75 cps
Lowest note apparently heard due to distortion	60 cps	30 cps	Little bass distortion
Lowest note due to harmonic reconstruction	30 cps (approx.)	20 cps (approx.)	40 cps (approx.)

at the low-frequency region of the spectrum. This is not generally realized, but is well substantiated.[5] It is interesting to note that the usual microphone adds very little distortion to the program, entirely unlike the loudspeaker which, with its higher amplitude of motion, is so often

[5] H.F. Olson, "Elements of Acoustical Engineering," McGraw-Hill Book Co., p. 167; 1947.

Push-pull output is especially good for eliminating *even* harmonics but *not* the objectionable odd ones. Therefore, the use of a single output tube with proper caution (feedback, etc.) in design may be equally acceptable to the much touted push-pull amplifier. One of the best expensive commercial radio receivers in times past used a single output tube with very superior results.

For reasons that will be explained, it is well to divide the musical range into at least two, or possibly three parts. One output tube and its associated loudspeaker can be used for each part of the range. A suggested division for the 75 to 7500 cps range is 75-350, 350-1600 and 1600-7500 cps. Each range would thus cover a frequency ratio of about 4½ to 1. Thus, it is difficult for a fundamental in any one range to produce much fifth harmonic and much more difficult to produce higher harmonics.

The three ranges are beneficial in another important detail.

"Intermodulation" is probably the most prolific source of audible distortion. This is a much more sensitive criterion of performance than harmonic generation.[6] When the voices of a choir seem to blur unnaturally in reproduction, this is almost a sure sign of intermodulation. A very simple listening test will show the presence of distortion very strikingly. Stand about four feet from another person some ten feet from the loudspeaker. Talk in a low to moderate conversational tone to him before turning up the volume of some orchestral selection. Then stop talking and raise the volume until he signifies by raising his hand, or some other prearranged signal, that the loudness is adequate for *good* volume, neither soft nor very loud. Now try to converse in the same tones as before. If the conversation now becomes very difficult to hear, it is an indication that there is very *little* distortion. In other words, the level of the clean reproduction is actually quite high in intensity. However, if there is considerable distortion, the volume level will actually be quite low, thus interfering but little with the conversation. Several trials with various reproducing systems will show the validity of this simple test.

If a low frequency is being produced from a loudspeaker, and, at the same instant, some very much higher tone is being reproduced, there is a special type of distortion because the high tone is taking off from a moving "springboard". This causes a wobbling of the high frequency note leading to "frequency modulation" distortion.[7] With the range divided into three parts, this distortion should not exceed one per cent in any range. A large loudspeaker for the entire range, as is usual, may cause ten per cent of this type of distortion with simultaneous reproduction of 100 and 7500 cps, for example.

The matter of loudspeaker efficiency is important.[8] The efficiency of a 16" loudspeaker is best between about 60 and 800 cycles. A 4" loudspeaker works well between 150 to 2,000 cps. A 1" loudspeaker works well between 800 and 10,000 cps. The highest frequency usable is inversely proportional to the mass of the cone. The low-frequency limit is based on distortion requirements, because the amplitude of the cone becomes greater with decrease in frequency.

The three loudspeaker diameters mentioned are approximate, and could be modified without serious difficulty, such as using a 2" diameter for the high frequencies, a 6" diameter for the medium frequencies, and two 12" units for the low frequencies. If the two 12" loudspeakers are used, it is highly preferable to provide a small exponential horn to load them properly. In fact, the use of a moderate horn, such as a simple corner loudspeaker, has great advantage from the low-frequency standpoint in maintaining loudspeaker efficiency at the lowest frequency, and also in limiting excessive cone travel to prevent distortion.

TABLE V

Summary of Other Considerations

Comparison of:	Usual Receiver (12" speaker)	30-15000 (*14" & tweeter)	75-7500 (3 speakers)
Intermodulation at 2 watts	10% (3% is tolerable)	5% or less	2% (or less)
Low frequency "springboard"	10% (or more) distortion	7%	1%
Angular coverage	Poor at highs	Reasonable with expensive spkrs.	Good with inexpensive spkrs.
Efficiency	Falls off at highs	Reasonable	Good
Record storage in medium size cabinets	Space available	Some space	No space

*Conventional baffle and high frequency speaker in simple form — neither with exponential horns as in high-priced two-way systems.

[6] J.K. Hilliard, "Intermodulation testing," Electronics, p. 123; July, 1946.

[7] H.F. Olson, "Elements of Acoustical Engineering," McGraw-Hill Book Co., p. 171; 1947.

[8] H.F. Olson, "Elements of Acoustical Engineering," McGraw-Hill Book Co., p. 126; 1947.

The directional characteristics of the loudspeakers are quite important because a large loudspeaker is very directional at high frequencies.[9] A 16" loudspeaker emits sound of higher frequencies in an increasingly narrowing beam. This is not serious below 500 cps. A 4" loudspeaker has an upper non-directional limit of some 2,000 cps. A 1" loudspeaker is quite nondirectional up to 8,000 cps. Thus, the use of three conventional loudspeakers will result in a wide angle of coverage throughout the range without special horns or dispersion systems.

Table V gives a summary of the various other effects considered.

If the three loudspeakers with the medium fidelity amplifier in a living room, within the ranges suggested, produce 4, 4 and 0.1 watts maximum output respectively, they will reproduce an equivalent sound heard by a listener in the fourteenth row orchestra seat or ten feet from a piano, in fortissimo passages.[10] There is much less acoustic power available from the orchestra in the 1600 to 7500 cps range than in the two lower ranges. The method[10] of computing this is shown in Table VI. The power per cycle in each range for an orchestra is multiplied by the number of cycles in the range to arrive at the total average acoustic power in each range. The ratio of maximum average power to normal power is then multiplied times the total average to obtain the maximum acoustic power for each range. Because the loudspeakers have an efficiency of approximately five per cent, the electrical input power necessary for each loudspeaker is then computed.

Thus, 8.1 watts peak is the maximum audio power needed for a three-speaker system, which will remain within the bounds of economic design. Actually, program monitoring will reduce these peaks so that it will be safe to run such an amplifier up to one-quarter watt on average power passages. The peak passages will be at 8.1 watts which is very loud in the normal living room.

Finally, it might be said that treble tone controls are generally useful to reduce noise and should be regarded as such. In systems with distortion or noise, the treble tone control will always be used by the average listener.

The combination of reasons given helps to explain the public preference for the 75 to 7500 cps range as reported by Chinn and Eisenberg. The "usual receiver" design is also explained, in which noise and distortion compel the user to prefer a range limited to 3500 cps or less and lead him to reduce the treble response still further for normal listening.

TABLE VI

Intensity from Symphony Orchestra at 14th Row Seats

Comparison of:	Lower Range (75-350 cps)	Middle Range (350-1600 cps)	Upper Range (1600-7500 cps)
Average power per cycle in range	7.3 microwatts	1.6 microwatts	0.04 microwatts
Band width in cycles	275 cycles	1,250 cycles	5,900 cycles
Total average acoustic power in range	2 milliwatts	2 milliwatts	0.25 milliwatts
Ratio of maximum to average power	100 times	100 times	20 times
Total maximum acoustic power	0.2 watts	0.2 watts	0.005 watts
Power necessary for loudspeaker (5% efficiency)	4 watts	4 watts	0.1 watts

[9] K. Henney, "Radio Engineering Handbook," McGraw-Hill Hill Book Co., 3rd Ed., p. 895.

[10] H.F. Olson, "Elements of Acoustical Engineering," McGraw-Hill Book Co., pp. 412, 481-482; 1943.

EQUIVALENT CIRCUIT ANALYSIS OF MECHANO-ACOUSTIC STRUCTURES*

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I. INTRODUCTION

To comprehend the new, the unknown, we often fall upon the old, the known. Thus, in the time of Volta and Ohm (circa 1800) many of the conceptions of electricity were based upon similarity with the older arts, i.e., hydraulics, mechanics and heat. Volta was among the first to recognize that the phenomena grouped under the name of galvanism were a manifestation of "electricity in motion" – as contrasted with the older electrical phenomena which represented "electricity in tension." Since there was no evidence of accumulation of electricity at any point in a circuit it followed that the current could be represented figuratively by the flow of an incompressible fluid along rigid and inextensible pipes. Ohm used Fourier's analyses of the conduction of heat to derive electrical laws and he was instrumental in developing the concepts of "current" and "electromotive force."[1] Thus since the earliest days of electrical theory, electro-motive force became endowed with the attributes akin to a mechanical force of hydraulic pressure, and electric current has been thought of as being of similar nature as mechanical velocity or the velocity of fluid flow. Undoubtedly, these classical concepts form the historical basis for equivalent circuit analysis as it is known today.

The early analogies became especially important during the end of the 19th century when ac electricity was still in its infancy while the theory of vibrations and sound had been already highly developed by Rayleigh and others. It was discovered that certain differential equations developed for use with vibrating mechanical bodies were equally applicable to electrical quantities. Rayleigh was among the first to bring the subject of alternating current electricity within the scope of acoustics and to prove that similar mathematical principles were applicable to both phenomena.[2]

Much has happened during the half century which followed Rayleigh's writings to change the relative technological positions of electricity, acoustics and mechanics. The improvement in electrical circuit elements, oscillators, amplifiers, oscillographs and meters – and indeed the perfection of the "analogy" computer – have made of the electrical network a most useful and flexible analytical tool. Therefore, we find an ever-increasing tendency to turn to the analogous electrical circuits for solution of mechanical, acoustical and other problems.

After having been introduced to a subject as old and venerable as this, the reader might wonder what there is new to be said about it. Surprisingly enough, the study of analogies has had a recent spurt in activity. New uses as well as limitations of analogies are being discovered almost daily. Complex physical organizations are being put into form manageable by analogies. Definitions and terminology are being brought in line with the new developments. The object of this paper is to attempt to give the reader a broad view of the subject to acquaint him with some of the new thinking regarding analogies.

II. THE EFP AND THE IFP ANALOGIES

The reader should be cautioned at this time that the traditional concept of similarity between voltage-force-pressure and current-velocity-fluid velocity is not the only one upon which a system of analogies can be based. During the past quarter century, through the efforts of Firestone[3] and others, it has become known that it is possible to establish another consistent system of analogies based upon certain mathematical similarities between electrical current, force, and pressure on the one hand, and voltage, velocity, and fluid flow on the other. The older "classical" analogy is currently being spoken of as the "voltage-force-pressure" analogy while the newer analogy (originally called "mobility" analogy by Firestone) is often referred to nowadays as the "current-force-pressure" analogy. For brevity we call them the EFP and the IFP analogies, respectively.

It seems feasible to apply either analogy about equally well to the solution of various problems, although it is generally recognized that the EFP analogy is the more advantageous with respect to acoustic devices and electrostatic (condenser, piezoelectric) transducers, while the IFP analogy is the more adaptable to mechanical devices and electromagnetic (magnetic, magnetostrictive) transducers. (Incidentally, equivalent circuits for transducers are a field in themselves and they are not treated here.) Almost everyone doing much equivalent circuit work eventually becomes conversant with both analogies. The casual user will avoid confusion by choosing a single analogy.

* Manuscript received June 1, 1954.

[1] Firestone, "A new analogy between mechanical and electrical systems," Jour. Acoust. Soc. Amer., vol. 4, pp. 239-267; 1933.

[2] Lord Rayleigh, "The Theory of Sound," Dover Publications, N.Y., vol. 1, p. 433, etc.; 1945.

[3] W. Dampier, "A History of Science," The MacMillan Company, N.Y., p. 232, etc.; 1944.

In the writer's opinion, the classical EFP analogy is to be preferred because electrically-trained people find it the easiest to comprehend and because it has decided advantages in connection with acoustic devices which, after all, are the principal domain of the audio technologist. However, there are differences of opinion on this matter, and some authors prefer to use the IFP analogy.

The relationships which are the basis of the EFP and the IFP analogies are shown in Table I. In this table it is assumed that all circuit elements are constant and all variables are steady-state RMS values. In each box under the Electrical, Mechanical and Acoustic elements are shown the term and its commonly used symbol, name of the unit (if available) and the unit in cgs and mks systems. Before World War II practically all work in acoustics was done in the cgs system and many acousticians prefer to continue using this system. Lately, however, the mks system has gained recognition and is used by several writers.

III. ACOUSTIC STRUCTURES

Acoustic structures consist of volumes of air which, in accordance with the EFP analogy, comprise the "springy" or capacitive elements. These may be connected together by conduits such as pipes, apertures, slits, etc. Flow of air through a pipe is accompanied by an increase in kinetic energy because of the acceleration of the mass of air as it flows from the volume into the pipe and vice-versa. Therefore, a pipe or aperture defines a mass-type or inductive element. Acoustic resistance or damping is obtained through the use of capillaries, slits, or crevices, such as the interstices between the threads of cloth through which the air is caused to flow, with the accompaniment of viscous friction. Additionally, when the openings connect to the atmosphere, sound is received or radiated by the acoustic structure and the impedance of the medium becomes a part of the acoustic structure.

TABLE I

ACOUSTICAL QUANTITY	MECHANICAL QUANTITY	ELECTRICAL QUANTITY EFP ANALOGY	ELECTRICAL QUANTITY IFP ANALOGY
Sound Pressure (p) microbar newton per sq. m	Force (F) dyne newton	Voltage (E) volt	Current (I) ampere
Volume velocity (U) cu. cm per sec cu. m per sec.	Velocity (v) cm. per sec. m per sec.	Current (I) ampere	Voltage (E) volt
Volume displacement (V) cu. cm cu. m	Displacement (D) cm m	Charge (Q) coulomb	Impulse volt-sec.
Acoustic Resistance (R_A) rayl (dyne-sec-cm^{-5}) mks-rayl (newton-sec-m^{-5})	Mechanical Resistance (R_M) mech. ohm (dyne-sec-cm^{-1}) mks-mech. ohm (newton-sec-m^{-1})	Resistance (R) ohm	Conductance (G) ohm
Inertance (M_A) gram-cm^{-4} kg-m^{-4}	Mass (M) gram kg	Inductance (L) henry	Capacitance (C) farad
Acoustic Compliance (C_A) cm^5-dyne^{-1} m^5-newton^{-1}	Mechanical Compliance (C_M) cm per dyne m per newton	Capacitance (C) farad	Inductance (L) henry
Acoustic Impedance (Z_A) $Z_A = p/U$	Mechanical Impedance (Z_M) $Z_M = F/v$	Impedance (Z) $Z = E/I$	Admittance (Y) $Y = I/E$

Impedance values for acoustic network elements have been treated by various writers[4] and derivations will not be given here. The values applicable to the most common elements are shown in Fig. 1, which also includes the equivalent electrical representation. Writing the acoustical quantities P and U directly in the equivalent electrical circuit instead of the corresponding electrical quantities E and I serves to eliminate confusion. The physical constants for use with this figure are given in Table II in both the cgs and the mks systems. To illustrate the methodology of network synthesis by means of an ex-

[4] H.F. Olson, "Elements of Acoustical Engineering," D. Van Nostrand Company, Inc., N.Y., p. 86, etc.; 1947.

TABLE II

PHYSICAL CONSTANTS FOR ACOUSTIC NETWORK ELEMENTS

QUANTITY	SYMBOL	CGS UNITS	MKS UNITS
Atmospheric Pressure (Usual, at sea level)	P_0	10^6 dynes/cm^2 (microbar)	10^5 newtons/m^2
Density of air at 20°C (at sea level)	ρ	1.2×10^{-3} gram/cm^3	1.2 kg/m^3
Coefficient of viscosity for air	μ	1.8×10^{-4} gram/cm-sec (poise)	1.8×10^{-5} kg/m-sec (mks poise)
Ratio of spec. heats c_p/c_v for air	γ	1.41	1.41
Velocity of sound in air at 20°C	c_v	34,400 cm/sec	344 m/sec

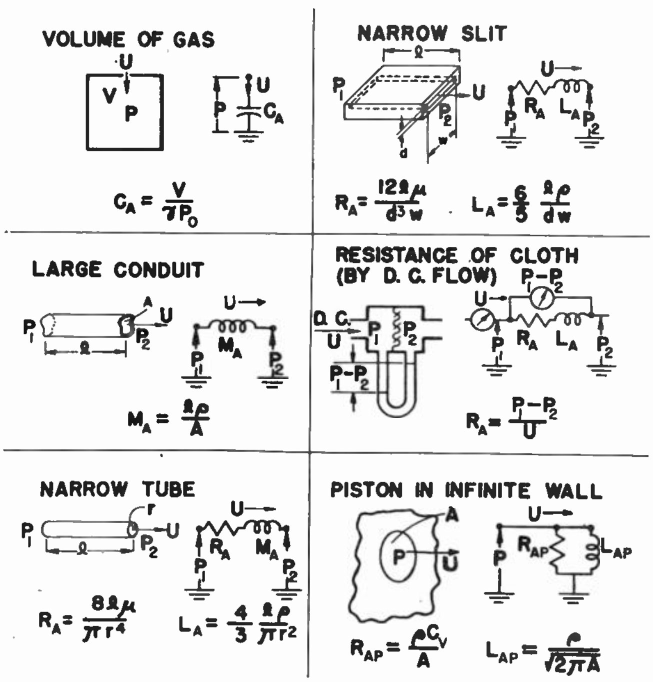

Fig. 1 – Acoustic impedance values of most common elements by the EFP analogy. For values of physical constants, see Table I.

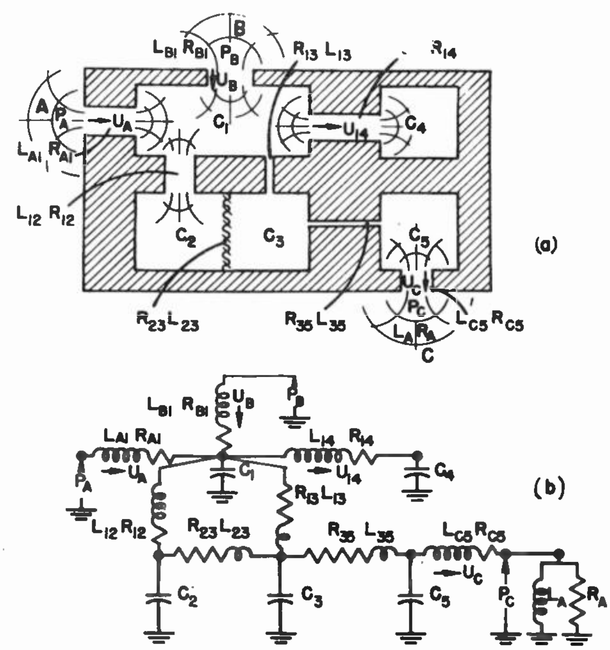

Fig. 2 – Representative acoustic structure (a); Equivalent circuit by the EFP analogy (b).

ample, an acoustic structure is shown in Fig. 2(a) and its equivalent circuit in Fig. 2b. The structure is assumed small compared to the wavelength and therefore representable by lumped impedance elements. This structure contains five volumes of air C_1 to C_5 represented in Fig. 2(b) by corresponding electric capacitors. Since the pressure throughout a volume is constant, a connection made to any part thereof is subjected to the same sound pressure. Constant pressure is tantamount to a constant potential, and hence the whole volume may be considered equivalent to an equipotential "terminal." This is analogous to the case of a capacitor consisting of an isolated sphere in space where the "other" terminal may be thought of as being a "ground."[5] Therefore, one side of each capacitor C_1 to C_5 is shown connected to "ground." One obvious consequence of the grounding of the one side of the acoustic capacitor is the proposition that the equivalent acoustic capacitances of volumes cannot be connected in "series."

[5] B.B. Bauer, "Transformer analogs of diaphragms," Jour. Acoust. Soc. Amer., vol. 23, no. 6, pp. 680-683; 1951.

At this point it is timely to make a distinction between the "terminals" and the "points of connection." Professor Firestone defines a terminal, in part, as follows: "A terminal of a specified type is the entire portion of an element or structure which is compelled to have the same value of one specified measurable quantity at any instant."[6] Thus the conduits leading into the volumes in Fig. 2(a) have different points of connection, but still are connected to a single pressure terminal.

The volumes are connected by conduits which may have the form of pipes, capillaries, slits, or cloth barriers. As may be seen in Figs. 1 and 2, conduits define acoustic inertance and resistance. Conduits of relatively large cross-section, (i.e. having minimum cross-sectional dimensions of several mm or more) are predominantly "inductive." Conduits of small cross-section, such as capillaries or thin slits, (with minimum cross-sectional dimensions of a few thousandths of an inch or less) are predominantly "resistive." Cloth, suitably mounted to prevent vibration forms an excellent acoustic resistance and it is widely used as damping element in acoustic structures. R_{23} is a cloth resistance. At each point of connection between a conduit and a volume where there is a change in cross-section, there is acceleration of the air from zero velocity to a velocity U within the conduit. The flow of air is shown by means of streamlines at the end of conduits. This acceleration causes an end-effect inertance added to inertance of the conduit. This end effect may be approximated by assuming the existence of a mass-less piston at the ends of the conduit.

From the point of view of methodology the following procedure may be followed: First, the acoustic compliances of all volumes are represented by capacitors connected to ground (or to a common buss). Next, the free terminals of all capacitors are interconnected by the inductive and resistive elements which comprise the conduits. In the equivalent circuit diagram, it is convenient to show the connections as being predominantly inductive or predominantly resistive by the relative length of the inductive and resistive components. Let the sound pressure impinging upon the entries A and B be P_a and P_b respectively. These pressures are shown as potentials to ground.[5] At C, the structure is confronted by the radiation impedance of the medium, Z_a. This impedance is represented approximately by a parallel combination of inductance L_a and resistance R_a. The pressure at the mouth of C can be obtained by multiplying the volume current U_c by Z_a.

IV. MECHANICAL ELEMENTS

The construction of equivalent circuits of mechanical structures by the EFP analogy is somewhat more complicated than that of acoustic structures. Mechanical forces and motions have magnitudes and directions, and hence, are vector quantities, while electrical potentials and currents in a circuit are scalar. To avoid complications resulting from this distinction, we restrict our analysis to mechanisms constrained to move along one axis. The mechanical network elements, or building blocks, are shown in Fig. 3. "Free" mass M can be represented by inductance L-short-circuited upon itself, and the loop current i represents the velocity v. The spring C_m has two ends which are acted upon by equal and opposite forces F. Therefore, the equivalent condenser C has two pairs of terminals acted upon by equal and opposite voltages e. The velocities of spring ends v_1 and v_2 are, in general, unequal. They are represented by currents i_1 and i_2. The compression of the spring occurs at a rate $v_1 - v_2$ which corresponds to the current flow $i_1 - i_2$ through the condenser. Similar considerations apply to the mechanical resistor R_m which is represented by the electrical resistor R_1.

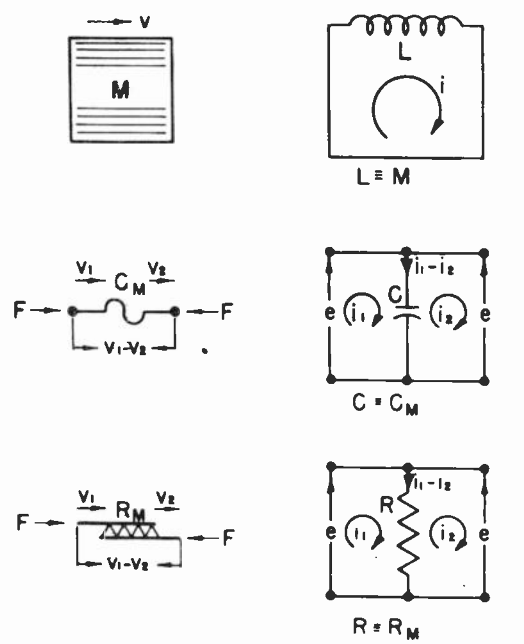

Fig. 3 – Basic mechanical network elements and equivalent electrical circuits by the EFP analogy.

The short circuited inductance is equivalent to a free mass. To represent a mass which is acted upon by a force, an emf generator must be coupled into the loop. If the motion of the mass is impeded by springs or damping elements, then means must be found to couple these impedances into the circuit. One way of achieving this is by breaking the circuit of the loop and connecting the generator or impedance to the two resulting terminals. This practice is not always possible in the equivalent analysis of mechanical structures as may be seen from the following set of examples.

[6] Firestone, correspondence with the author, 1953.

In Fig. 4(a) we have a relatively simple system consisting of three masses connected by two springs. On the right hand side is the equivalent circuit. To obtain this equivalent circuit, we have broken the loop connecting the two ends of the inductance M_1 at the left hand side, for the purpose of inserting a generator with a voltage E which represents the force F, and again, at the right hand side, for the purpose of connecting one set of terminals of the capacitor C_{12}. Similar connections have been performed elsewhere in the circuit. This procedure is almost self-evident and poses no special difficulty. A more complicated situation is shown in Fig. 4(b) which depicts four masses connected by four coupling springs.

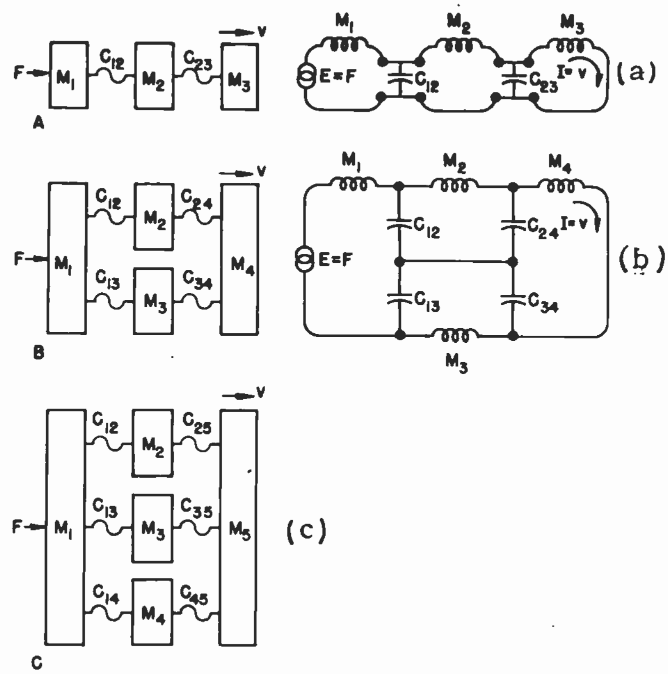

Fig. 4 – Mechanical structures of increasing complexity and equivalent circuits by the EFP analogy.

This structure is not as simply portrayed as the previous one. However, by a bit of visualization and cut-and-try methods it can be determined that the correct equivalent circuit is shown on the right hand side of the figure. While this electrical circuit has little geometric resemblance to the mechanical structure, its correctness may be checked by noticing that the sum of the voltage drops across the elements in the circuit of the inductances is the same as the sum of the forces acting upon the corresponding masses. This is an acknowledgement of the equivalence between the laws of Kirchhoff and d'Alembert.

A still more complicated structure is shown in Fig. 4(c) which shows five masses interconnected by six springs. An equivalent circuit is not given at the right hand side, because none can be found by the means discussed so far.

V. TRANSFORMER COUPLINGS

Thus, it is seen that mechanical systems of progressively increasing complexity present progressively increasing difficulty in the synthesis of the equivalent circuits by the EFP analogy.

One of the reasons is to be found in the fundamental difference between our basic concepts of mechanical elements and the corresponding electrical circuit components. As shown in Fig. 5(a), a mass is usually thought of as a rigid body obeying Newton's laws; its counterpart, i.e., an inductance, comes to mind as a coil with two free terminals. It is obvious that no current can flow through such coil. It helps to clarify our thinking to visualize a free mass as being equivalent to an inductance short-circuited by means of a loop as shown in dotted line. Every point of the circuit carries the same current, the same as every point of the mass travels with equal velocity. The mass constitutes a single velocity terminal, and the loop constitutes a single current terminal. A spring, in 5(b), has two velocity terminals, but the equivalent condenser, at right, has four points of connection. To represent a free-free spring, these points of connection must be closed with loops as shown by dash-lines. Each of these loops becomes a current terminal corresponding to the respective ends of the spring. Similar considerations apply to the mechanical resistor in Fig. 5(c). By this process the terminals and points of connection of mechanical elements become identified with the corresponding entities of electrical circuit elements.

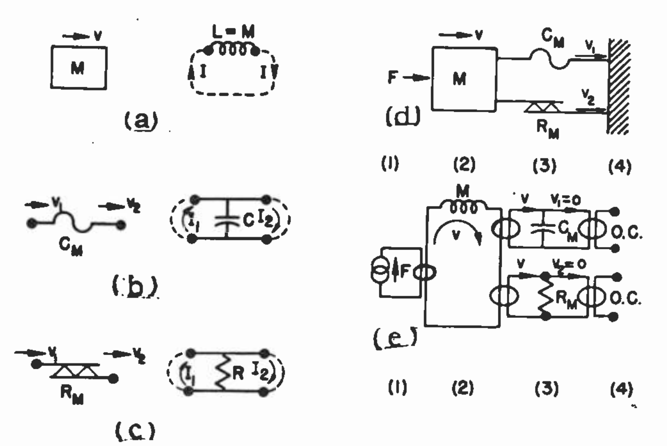

Fig. 5 – "Free" mechanical elements and transformer couplers to denote connections between mechanical elements.

Next, means must be found to couple the loops without breaking the circuit. This can be readily done by bringing the corresponding wires together and surrounding them with an ideal magnetic core of infinite permeability and zero losses, thus forming an ideal transformer of 1:1

turns ratio.[7] As an example of this type of connection, Fig. 5(d) represents a system consisting of a force (1) driving a mass (2) which is coupled by means of a spring and mechanical resistor (3) to the reference frame (4). The equivalent circuit is shown in Fig. 5(e) where constant voltage generator (1) is coupled to the inductance loop (2) which in turn is coupled to the corresponding terminals of a spring and resistance (3). The latter are terminated in the reference frame (4) which appears as an "open circuit" since v_1 and $v_2 = 0$. The ideal cores are represented by the heavy circles. Writing mechanical quantities F, V, etc. directly in the equivalent electrical circuits instead of the corresponding electrical quantites E, I, etc. serves to eliminate confusion.

Figure 5(e) suggests immediately that if a mass is made vanishingly small, the mechanical circuit will reduce to a velocity junction, and the equivalent inductance-less loop becomes an electrical current junction. By the way of example, a velocity junction is shown in Fig. 6(a) and its equivalent circuit in Fig. 6(b). The velocity

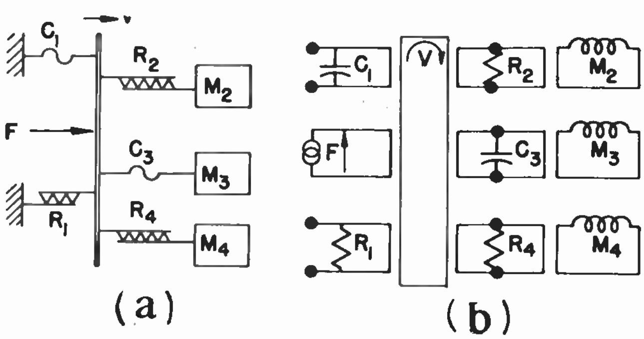

Fig. 6 – Velocity "junction" by the EFP analogy using transformer couplers.

of all terminals at the junction is v. To facilitate the drawing of the ideal transformers in equivalent circuits the magnetic core can be omitted. Instead we have adopted the convention that two adjoining parallel wires constitute the two windings of the ideal transformer.

The following section illustrates specific application of transformer couplers to the problems of equivalent circuit analysis.

VI. UTILIZATION AND REMOVAL OF TRANSFORMER COUPLINGS

As we demonstrate the use of transformer couplings, we shall also indicate methods for removing them without altering the voltage and current relationships, therefore arriving at the conventional-type analogy circuits. The use of ideal transformers adds no difficulty in the analytical treatment of circuits. However, for the purposes of experimental circuit work, it is desirable to remove the transformers, wherever possible.

Our first example is shown in Fig. 7(a). The mechanical system consists of three masses interconnected by springs and resistors in tandem. In Fig. 7(b) is the

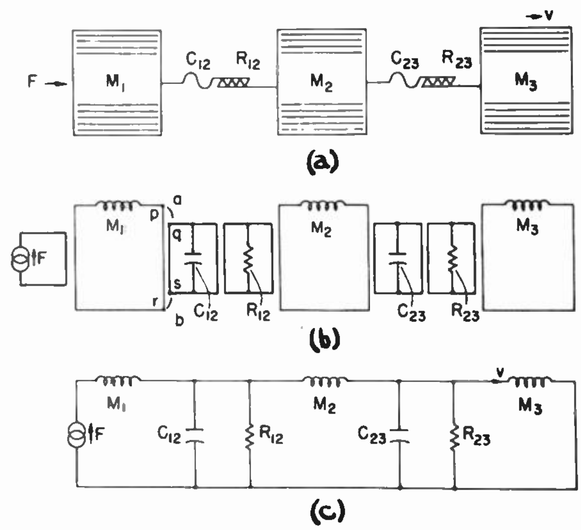

Fig. 7 – (a) Mechanical structure; (b) Equivalent circuit by EFP analogy with transformer couplers; (c) Transformer couplers removed.

equivalent circuit by EFP analogy with transformer couplings. The methodology for drawing these circuits consists first in drawing the loops constituting velocity junctions and those containing masses, and next adding the condenser and resistor elements and driving generators, with the aid of transformer couplers.

The circuit equations of Fig. 7(b) can be written as if the transformers were not present simply by remembering the voltage and current relations in an ideal transformer. In this manner, the usual mesh and junction equations can be written and solved in an ordinary manner.

For experimental circuit work, as in analogy computers, it is desirable to remove the transformers. This must be done without disturbing any voltage and current relations in the circuit. The simplest way of doing this is to select any two transformer-coupled meshes which have no *conductive* connection and connect together two adjacent points, say p and q, of the coupling transformer with a jumper "a." At that instant, points r and s become equipotential, since the primary and secondary voltages in a 1:1 transformer are identical. Therefore, r and s can be connected by jumper "b". Now the transformer $pqrs$ has no further purpose and it may be removed from the circuit. Each additional loop in Fig. 7(b) which has no conductive connection to the circuit can be connected in this manner resulting in the circuit of Fig. 7(c), which is the conventional EFP analogy circuit for the array in Fig. 7(a).

[7] B.B. Bauer, "Transformer couplings for equivalent network synthesis," Jour. Acoust. Soc. Amer., vol. 25, pp. 837-840; 1953.

A somewhat more complicated mechanical array is shown in Fig. 8(a). It consists of four masses interconnected by four sets of springs and mechanical resistors. Fig. 8(b) is the equivalent circuit by impedance analogy with transformer couplings. It is gratifying to note that the geometry of the electrical circuit is very much like the geometry of the mechanical circuit. There are 13 transformers, 12 of which can be removed if desired, as in the previous example. In Fig. 8(b) this is indicated by the jumpers placed at either side of the transformers, and it results in the circuit of Fig. 8(c). The transformer *pqrs* cannot be removed in this manner since a jumper between points p and q will short out the inductance M_3 and upset voltage and current relations. However, if we can trace an impedance-less conductive connection between the corresponding points of the windings, such as the points p and q, then these points become equipotential and may be connected together. This condition will occur if the inductance M_3 is moved from the branch p-t to the branch r-u, as shown in dotted outline. This can be done with impunity since the sum of voltage drops in the mesh t-p-r-u is not affected by the position of M_3. After M_3 has been moved to its new position, p-q and r-s can be connected as shown by the dotted jumpers a and b, resulting in the removal of the last transformer. The final circuit is given in Fig. 8(d). This is similar to the circuit in Fig. 3(b), except for the advantage of having been synthesized without the need for the use of the cut-and-try approach.

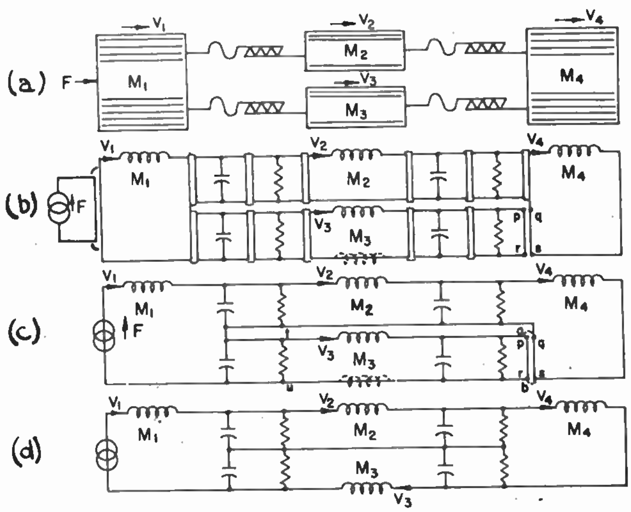

Fig. 8 – (a) Mechanical network; (b) Equivalent circuit with transformer couplers; (c) Removal of transformer couplers; (d) Final equivalent circuit.

In Fig. 9(a) is the mechanical array consisting of five masses M_1 to M_5 interconnected by three independent spring-mass-spring systems, which is the same as that shown previously in Fig. 3.

The equivalent circuit can be readily obtained by the transformer analogy as shown in Fig. 9(b). Again, we notice a geometric resemblance between the circuit and the structure, and this is a great help in drawing these circuits. Let us proceed to remove as many transformer couplers as possible. By removing, first, all the transformers between the loops with no conductive connection we successively remove all the transformers except f and g as shown by the jumpers numbered 1 to 22 inclusive. The resulting circuit appears in Fig. 9(c). The transformers f and g remain. To remove f, the inductance L moved around to the other side of the loop as shown in the previous example. As soon as this is done, the connections 23 and 24 can be made, therefore removing the transformer f. Transformer g remains and cannot be removed. The final equivalent circuit is shown in Fig. 9(d). It is evident that the user would be needlessly taxing his ingenuity if he attempted to draw the equivalent circuit of Fig. 9(a) by the conventional method of impedance analogy.[8,9]

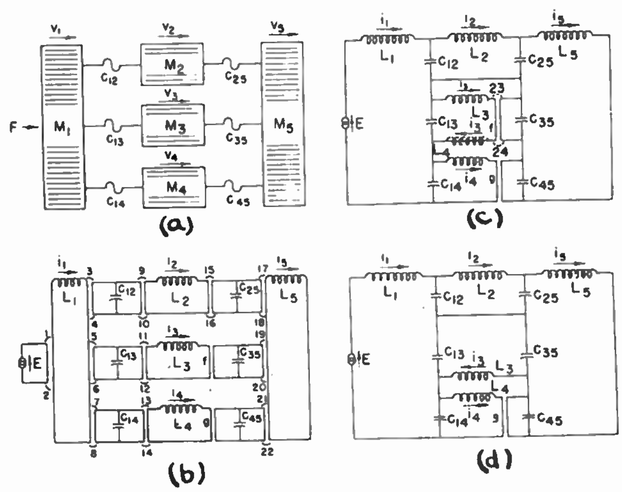

Fig. 9 – (a) Mechanical network; (b) Equivalent circuit with transformer couplers; (c) Removal of transformer couplers; (d) Final equivalent circuit. Transformer "g" cannot be removed.

VII. ANALOG FOR DIAPHRAGMS

The transfer of energy between the acoustic and the mechanical side of a structure requires the use of some sort of diaphragm. Ideal transformer couplers have been found to be of much value in representing the action of diaphragms. Each side of a piston-like diaphragm or a portion thereof of projected area A_n may be thought of as a means for transforming an actuating force F_u into a sound pressure $p_n = F_u/A_n$. At the same time, linear velocity v_n of the surface causes a volume velocity $u_n = A_n v_n$. This mechanical-acoustic transformation is seen to be analogous to the transformation of voltages and currents between the primary and the secondary

[8] A. Bloch, in: Jour. Inst. Elec. Engrs. (British), vol. 92, p. 157; 1945.

[9] A. Bloch, in: Proc. Phys. Soc. London, vol. 58, p. 677; 1946.