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

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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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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 cenvention 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 loud-speaker 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

$$r = k\sqrt{(\sum_i A_i \sin\omega_i t)^2 + (\sum_i A_i \cos\omega_i t)^2}.$$

$$= k\sqrt{(A_1 \sin\omega_1 t + A_2 \sin\omega_2 t + A_3 \sin\omega_3 t + \cdots)^2 + (A_1 \cos\omega_1 t + A_2 \cos\omega_2 t + A_3 \cos\omega_3 t \cdots)^2}.$$

$$= k\sqrt{[A_1^2 + A_2^2 + A_3^2 + \cdots] + 2\{[A_1A_2 \cos(\omega_1 - \omega_2)t + A_1A_3 \cos(\omega_1 - \omega_3)t + \cdots] + [A_2A_3 \cos(\omega_2 - \omega_3)t + \cdots] + \cdots}$$

The momentary center angle is

$$\phi = \text{arc tan}\frac{A_1 \sin\omega_1 t + A_2 \sin\omega_2 t + \cdots}{A_1 \cos\omega t + A_2 \cos\omega_2 t + \cdots}$$

$$= \text{arc tan}\frac{\Sigma (A_i \sin\omega_i t)}{\sum_1 (A_i \cos\omega_i t)}.$$

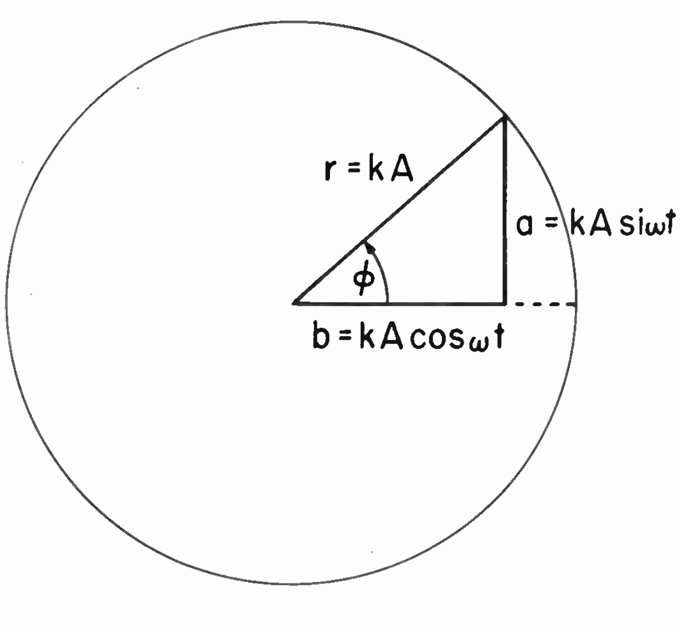

Fig. 1 – Center deflection of the electron beam of an oscilloscope in the case of a simple rotary field deflection.

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

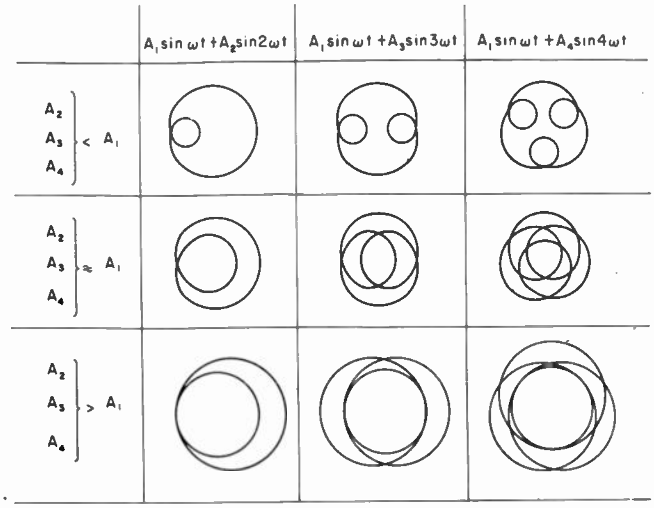

Fig. 2 – Polar coordinate pictures with rotary field deflection by a frequency spectrum, consisting of one fundamental frequency and a 2nd, 3rd and 4th harmonics respectively.

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.

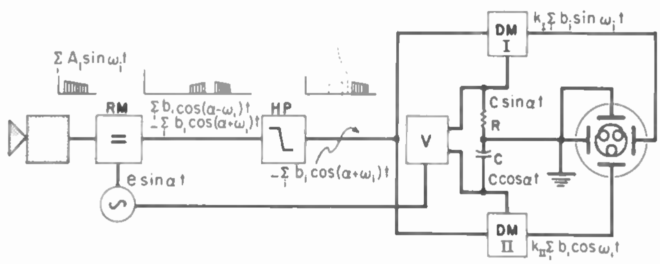

Fig. 3 – Polar coordinate pictures of vowel [a] and [i].

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.

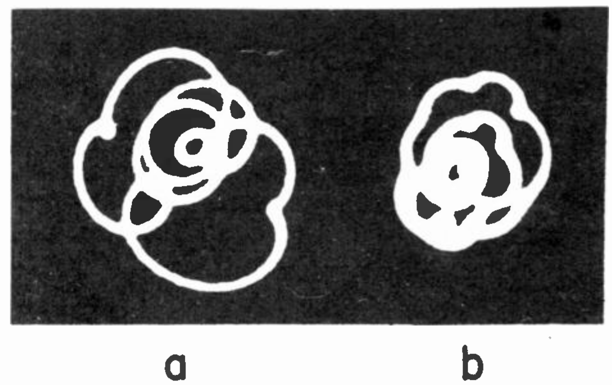

Fig. 4 – Circuit diagram for production of a rotary field by a frequency spectrum.

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

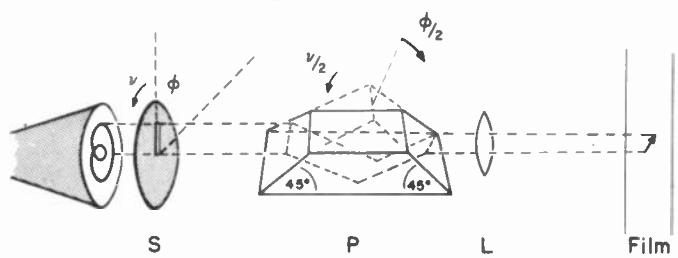

Fig. 5 – Arrangement for converting a polar coordinate picture into a cartesian coordinate picture.

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.

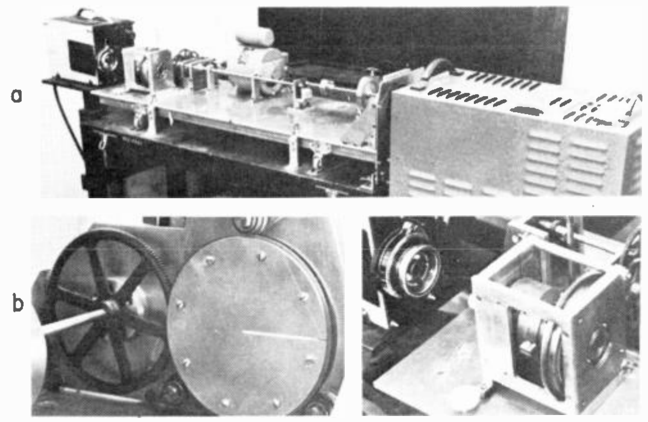

Fig. 6 – Construction of the apparatus for coordinate conversion; a–Total view of the apparatus; b–Rotating slit; c–Rotating prism.

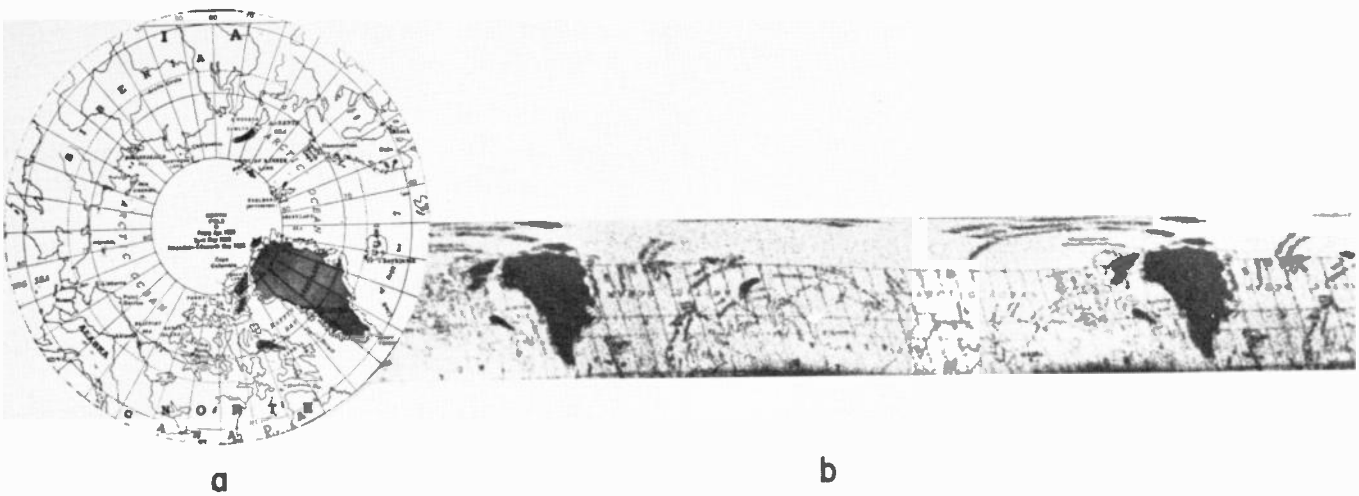

Fig. 7 – Coordinate conversion of a map of the North Polar Zone: a–Given in polar coordinates; b–Given in cartesian coordinates.

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). "A¹" corresponds to a spectrum, composed of a fundamental frequency (500 cps) and the 4th harmonic (2000 cps). "B¹" corresponds to a somewhat more complicated spectrum, which consists of the frequencies 500, 1000 and 2000 cps. "C¹" is the vowel [*a*] and "D¹" 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 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.

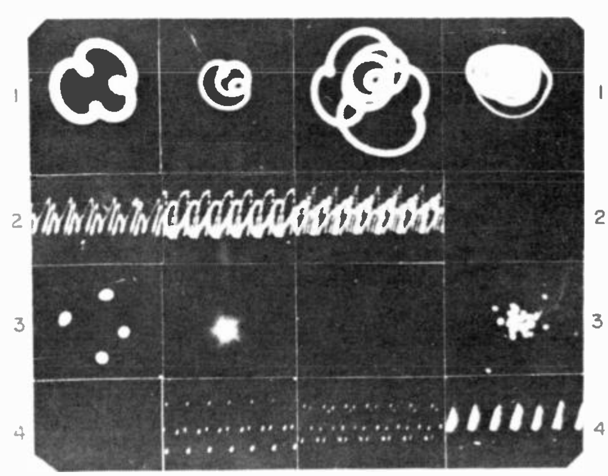

Fig. 8 – Coordinate conversion of polar oscilloscope pictures: Series A. Frequency spectrum 500 and 2000 cps; B. Frequency spectrum 500, 1000 and 2000 cps; C. Vowel [a]. D. Sibilant [s]. Line 1 – Polar coordinate picture; 2–Cartesian pictures corresponding to the pictures of line 1; 3–Polar coordinate 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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Gloucester, Massachusetts

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.

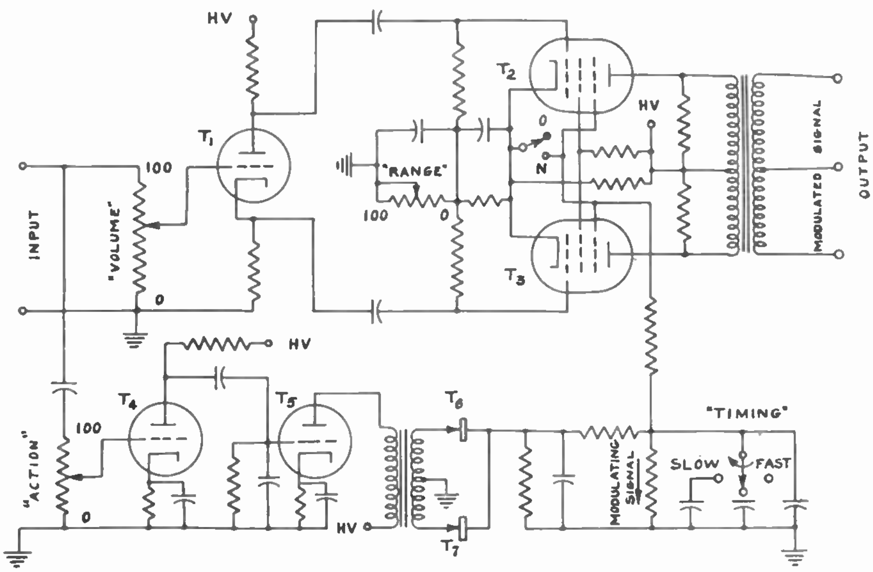


Fig. 1 – Dynamic expander, using pentode tubes.

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.

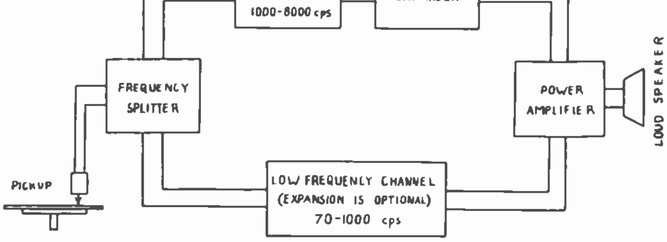


Fig. 2 – Selective expanding amplifier (dual channel method).

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, devel-

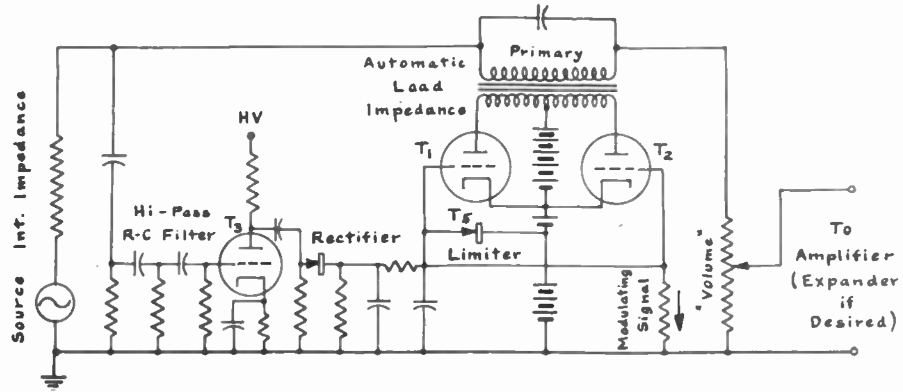


Fig. 3 – Selective expanding amplifier (transformer loading method).

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.

oped 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.

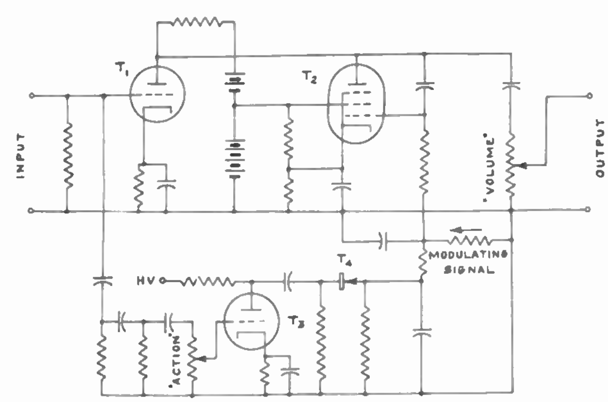


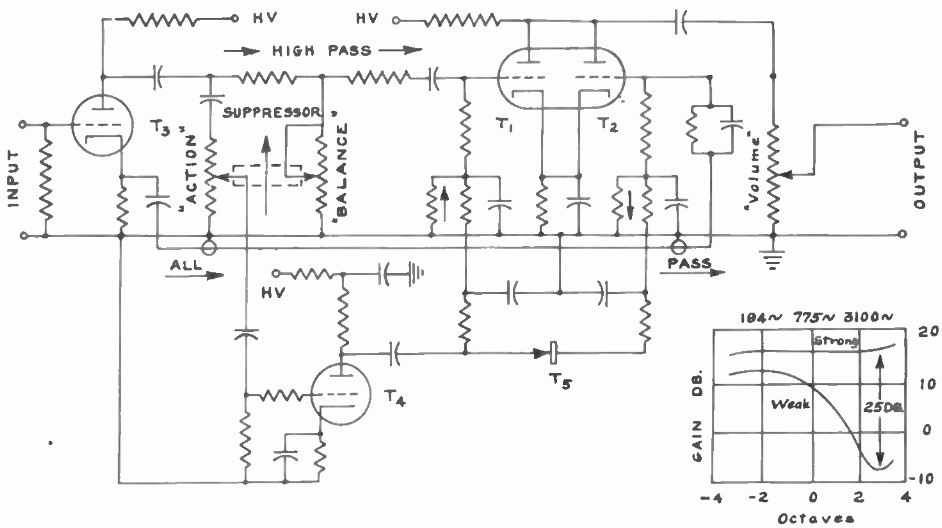
Fig. 4 – Selective expanding amplifier (reactance tube method).

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

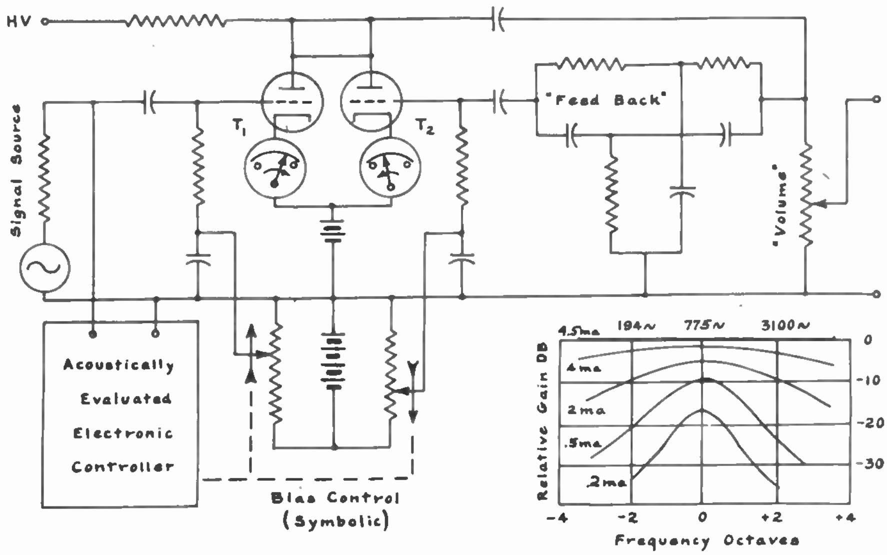


Fig. 6 – Dynamic amplifier with horizontal and vertical expansion.

triode and since the feedback from plate to grid is 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 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.

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 translation 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.

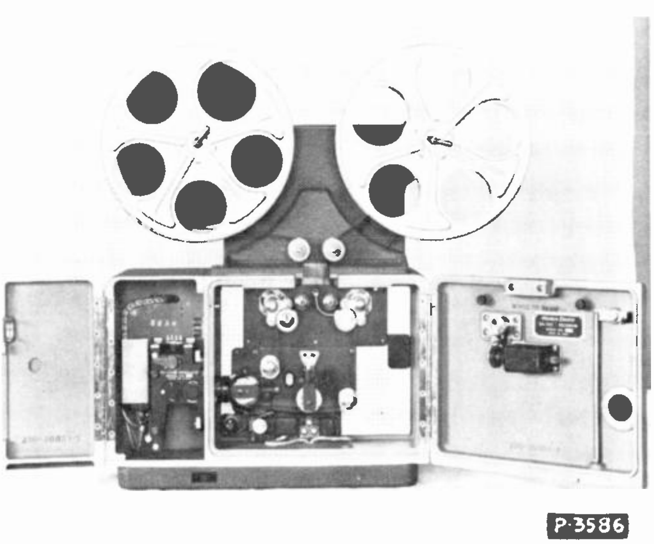

Fig. 1 – Photographic recorder modified for magnetic recording.

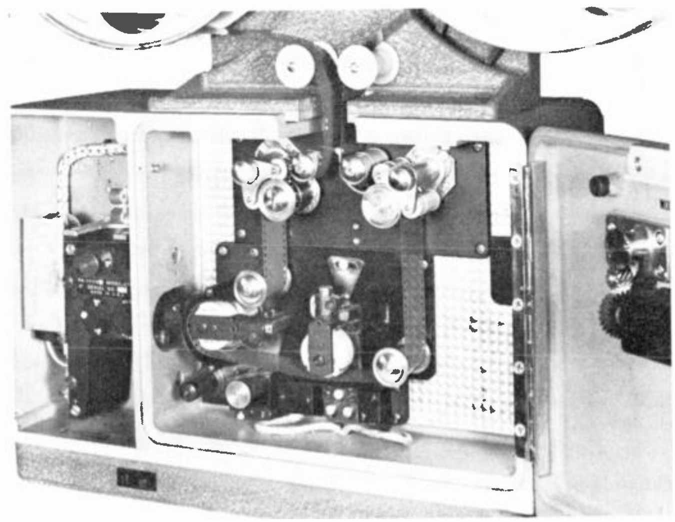

Fig. 2 – View of recorder showing magnetic head mounting.

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.

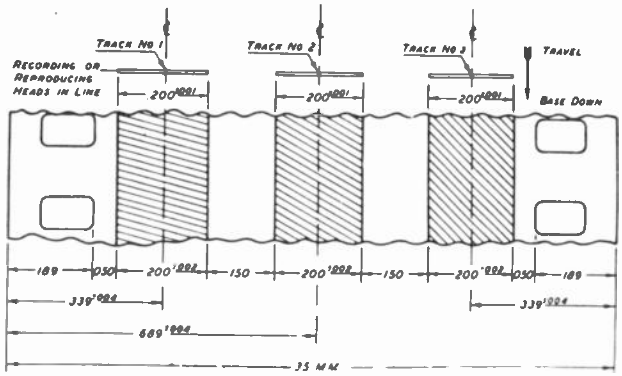

Fig. 3 – Magnetic film-track standards.

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

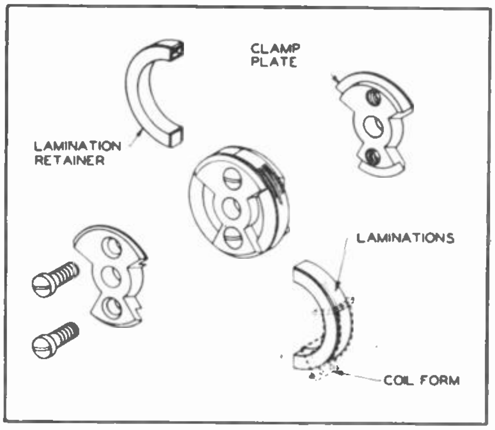

Fig. 4 – Exploded view of typical magnetic head.

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 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.

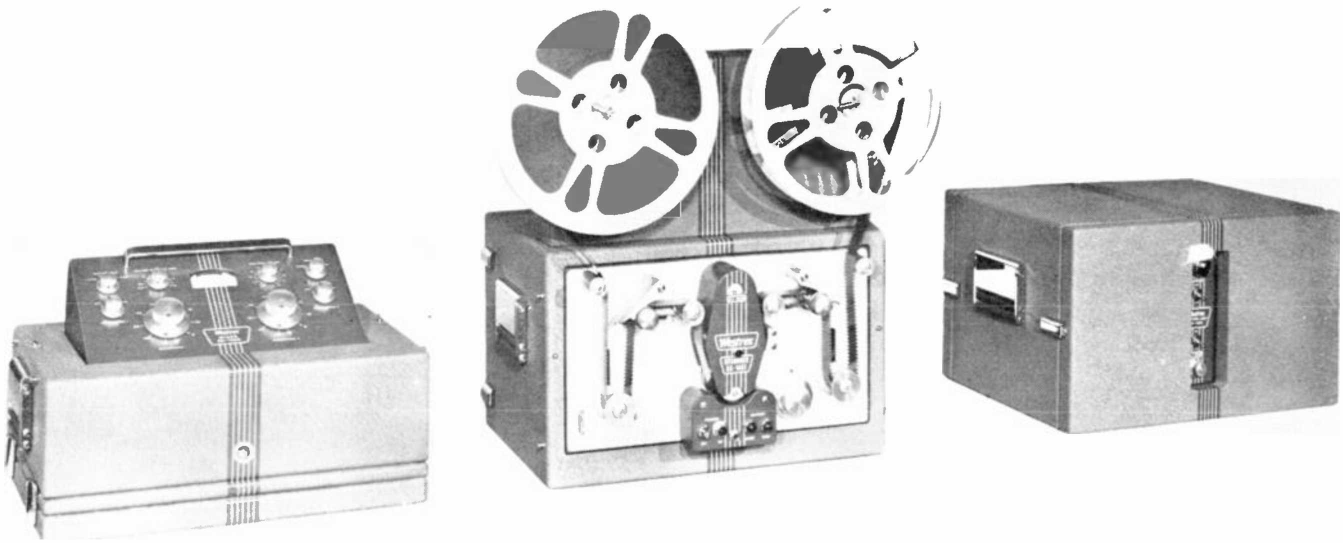

Fig. 6 – Typical magnetic recording system.

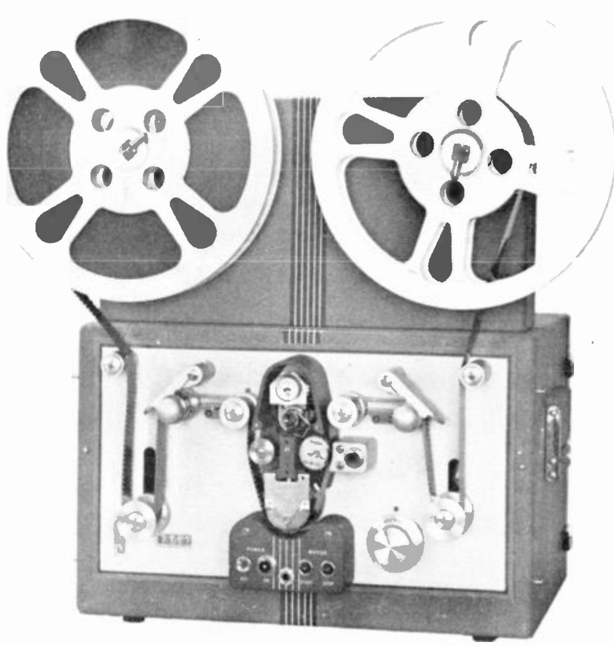

Fig. 7 – Magnetic recorder with front cover and cover plate removed.

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 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.

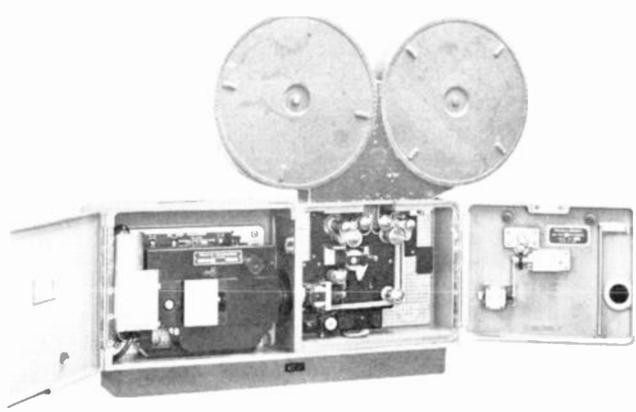

Fig. 8 – Dual photomagnetic recorder.

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.

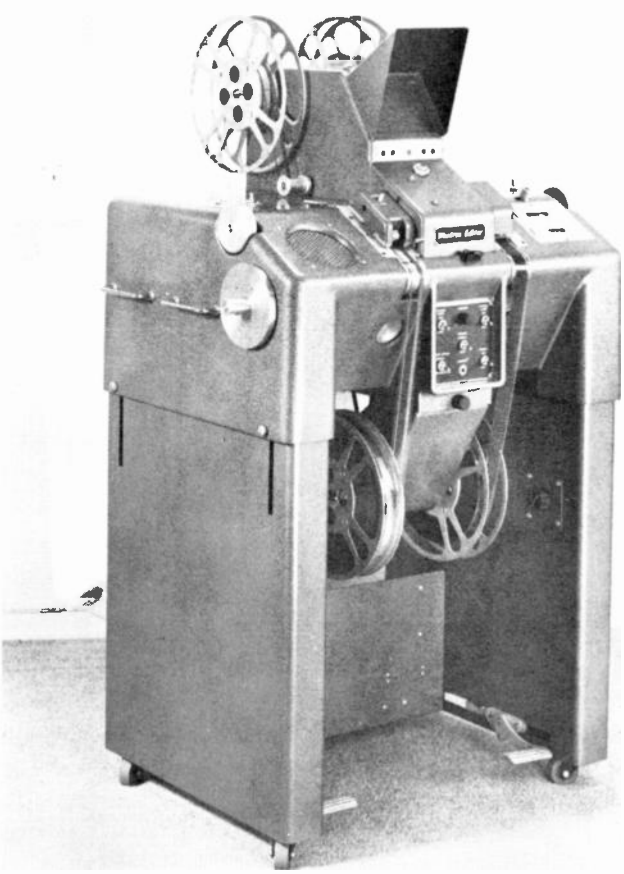

Fig. 9 – Westrex film editer.

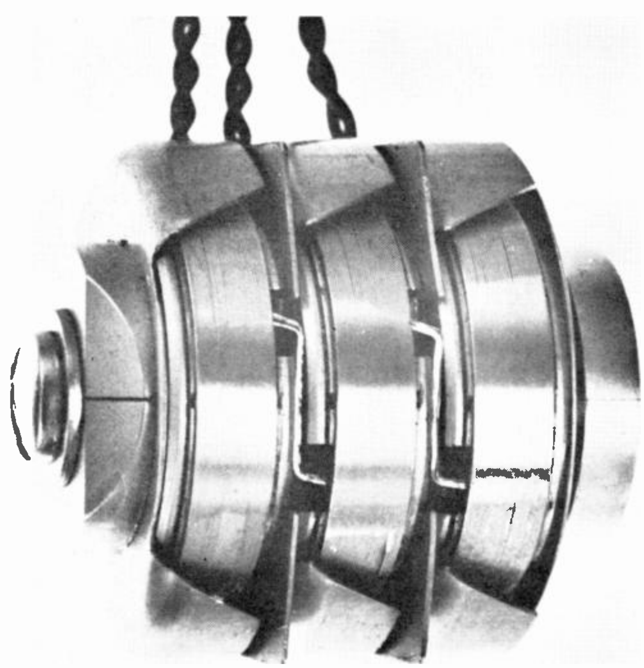

Fig. 10 – Triple-track magnetic head.

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.

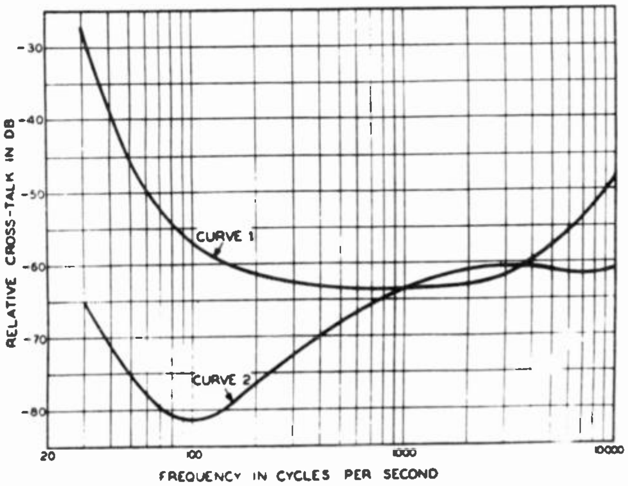

Fig. 11 – Crosstalk as a function of frequency.

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

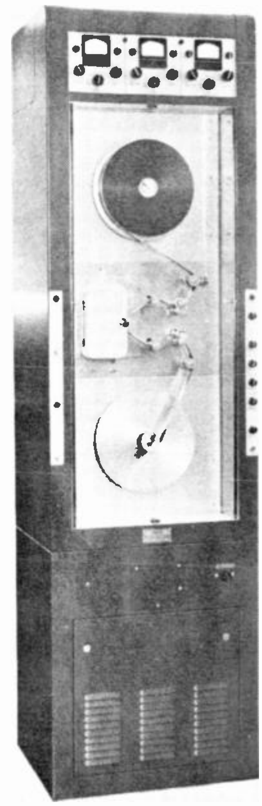

Fig. 12 – Triple-track recorder.

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 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 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.

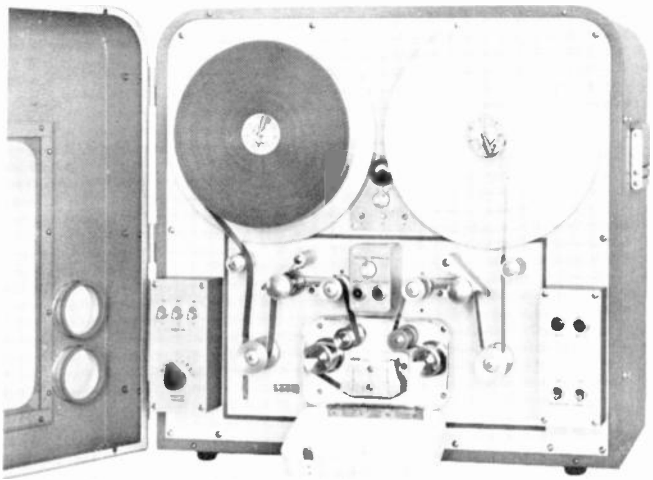

Fig. 13 – Portable stereophonic recorder.

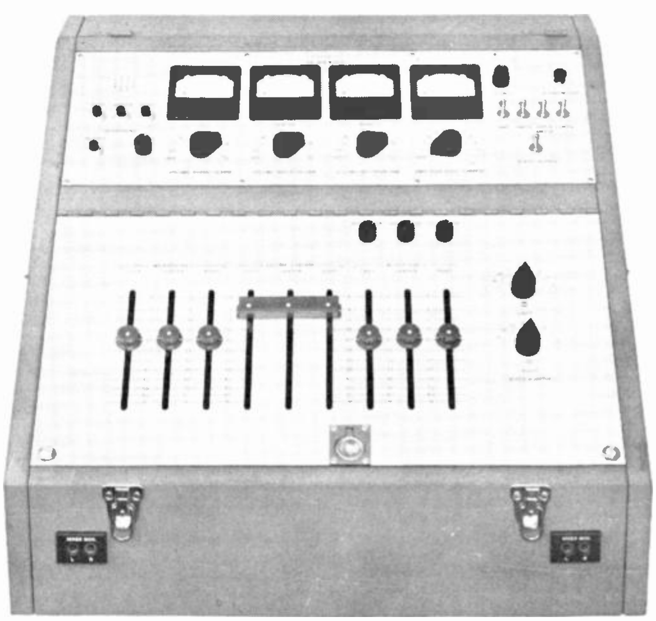

Fig. 14 – Stereophonic mixer.

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

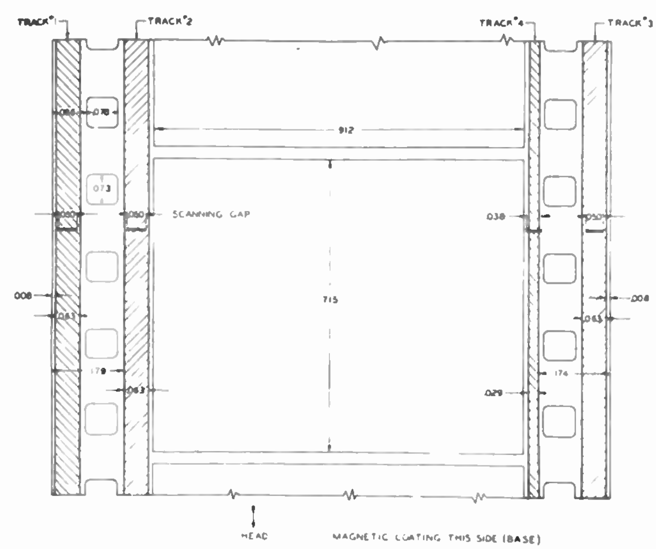

Fig. 15 – Proposed standard for Cinemascope sound track.

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 introduction 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.

Fig. 16 – Theatre reproducer for Cinemascope film.

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 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.

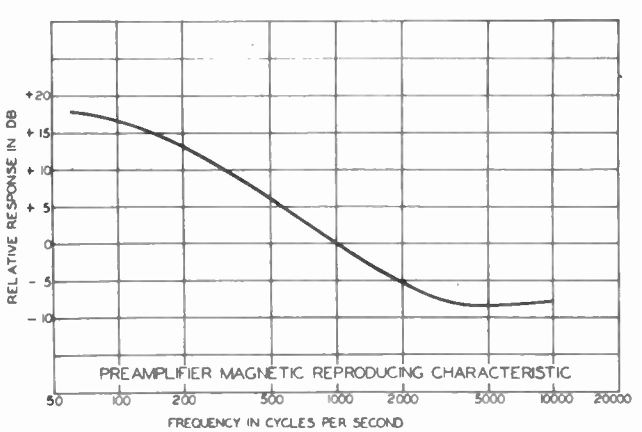

Fig. 17 – Pre-amplifier magnetic reproducing characteristic for Cinemascope sound track.

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 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

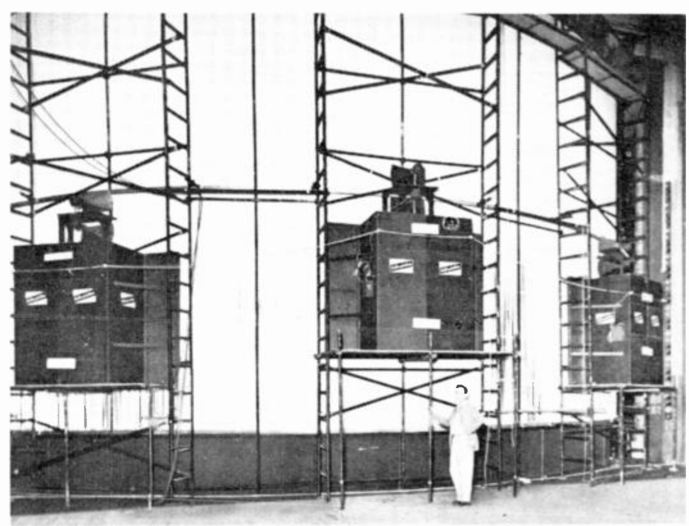

Fig. 18 – Stereophonic loudspeaker installation.

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.

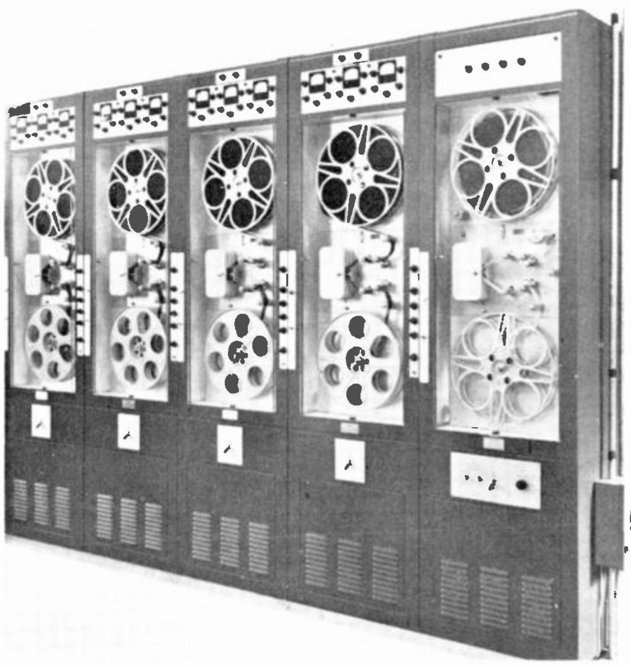

Fig. 19 – Multitrack electrical printer.

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.

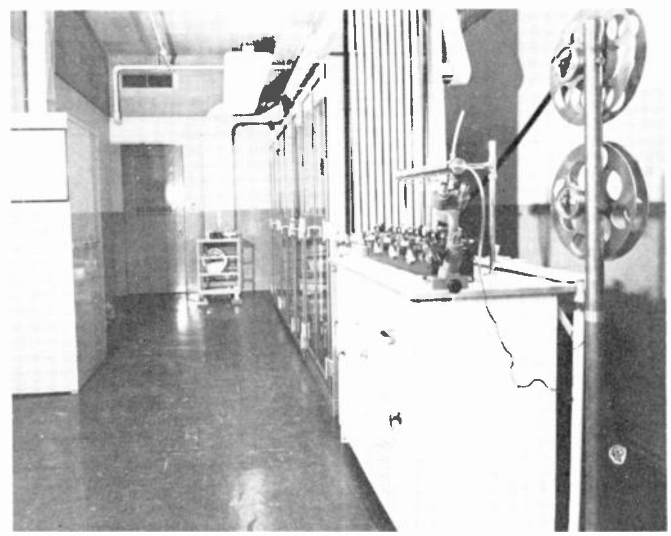

Fig. 20 – General view of magnetic film striping machine. (Courtesy Warner Brothers Studios)

REFERENCES

1. D. G. C. Hare and W. D. Fling, 'Picture-Synchronous Magnetic Tape Recording," Jour. SMPTE, vol. 54, pp. 554-566, May, 1950.

2. G. R. Crane and H. A. Manley, "A Simplified All-Purpose Film Recording Machine," Jour. SMPTE, vol. 46, pp. 465-474, June, 1946.

3. G. R. Crane, J. G. Frayne and E. W. Templin, "Supplementary Magnetic Facilities for Photographic Sound Systems," Jour. SMPTE, vol. 54, pp. 315-327, March, 1950.

4. G. R. Crane, J. G. Frayne and E. W. Templin, "Magnetic Recording on Film," Jour. SMPTE, vol. 56, pp. 295-399, March, 1951.

5. L. L. Ryder and Bruce H. Denny, "Magnetic Sound Track Placement," Jour. SMPTE, vol. 58, pp. 119-136, February, 1952.

6. Edward Schmidt, "Commercial Experience with Magna-Stripe," Jour. SMPTE, vol. 60, pp. 463-469, April, 1953, Part 2.

7. G. R. Crane, Fred Hauser and H. A. Manley, "Westrex Film Editer," Jour. SMPTE, vol. 61, 316-323, September, 1953.

8. C. C. Davis, J. G. Frayne and E. W. Templin, "Multichannel Magnetic Recording," Jour. SMPTE, vol. 58, pp. 105-118, February, 1952.

9. C. C. Davis, "An Improved Film-Drive Filter Mechanism," Jour. SMPTE, vol. 46, pp. 454-464, June, 1946.

10. J. G. Frayne and E. W. Templin, "Stereophonic and Reproducing Equipment," Jour. SMPTE, vol. 61, pp. 395-407, September, 1953, Part 2.

11. C. C. Davis and H. A. Manley, "An Auxiliary Multitrack Magnetic Sound Reproducer," to be published in Jour. SMPTE.

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

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[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.

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.

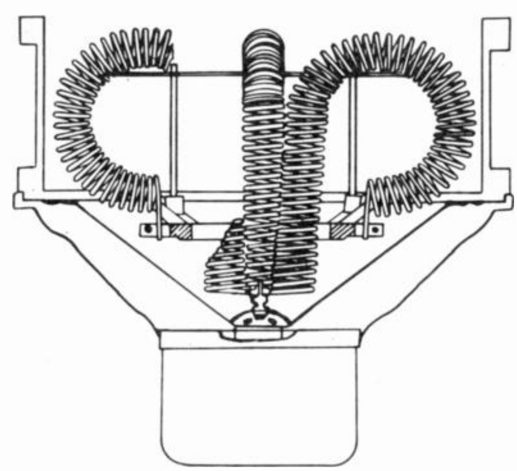

Fig. 1 – Direct-coupled reverberation loudspeaker.

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

[5] R. L. Wegel, U.S. Patent 1,852,795; 1932.

[6] Morse, "Vibration and Sound," pp. 291-297, McGraw-Hill Book Co., 1936.

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.

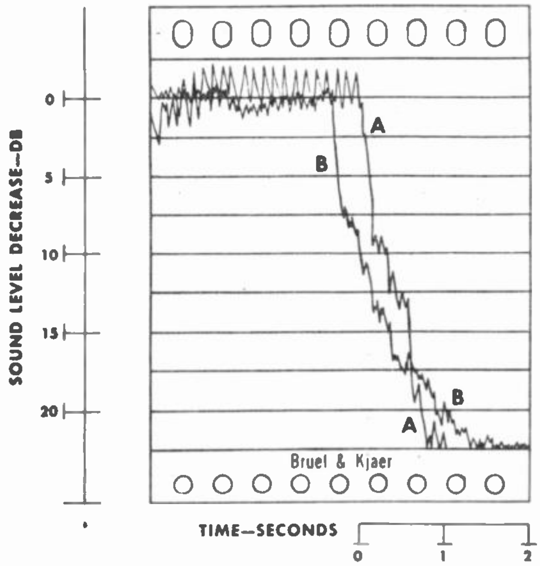

Fig. 2 – Comparison of sound decay curves in (A) large music hall and (B) from reverberation loudspeaker.

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.

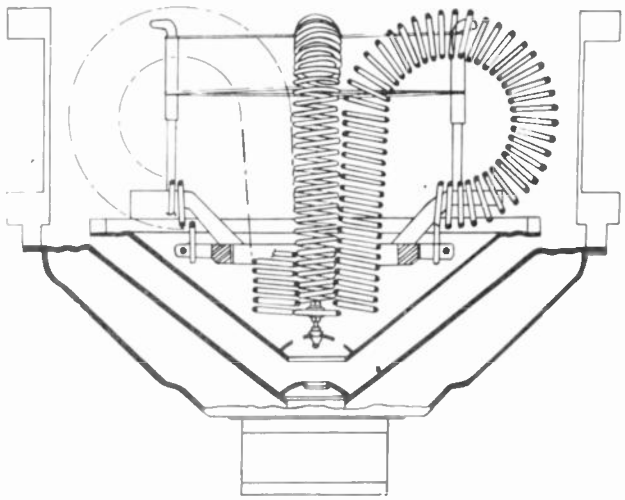

Fig. 3 – Acoustically-coupled reverberation accessory for loudspeakers.

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-

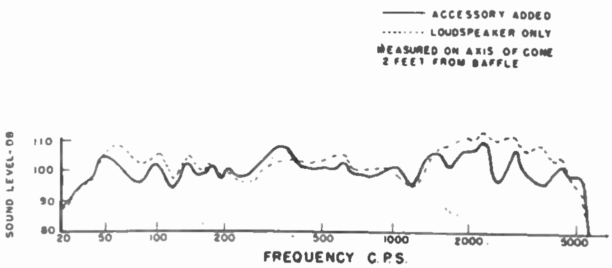

Fig. 4 – Steady-state response-frequency characteristics of a fifteen-inch loudspeaker with and without the reverberation accessory.

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.

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

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INSTITUTIONAL LISTINGS

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