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PGA NEWS

CONTRIBUTIONS

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IRE PROFESSIONAL GROUP ON AUDIO

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PGA News

MARVIN CAMRAS IS NEW EDITOR

Marvin Camras (right), new Editor of IRE TRANSACTIONS ON AUDIO, accepts responsibilities from A. B. Bereskin, who was Editor for three years. (PGA meeting March 25, 1958.)

With the May–June issue, Marvin Camras, of the Armour Research Foundation, Chicago, Ill., will become the new editor of the IRE TRANSACTIONS ON AUDIO. Alexander B. Bereskin, University of Cincinnati, Cincinnati, Ohio, who has served as editor for three years will continue as a member of the Editorial and Administrative Committees.

Mr. Camras has been a regular contributior to these TRANSACTIONS. He is a past Chairman, Secretary-Treasurer, and member of the Administrative Committee of the Professional Group on Audio.

He received the M.S. degree in electrical engineering from the Armour Institute of Technology in 1942. Since 1940, as a member of the staff at Armour Research Foundation, he has done research on projects in the electronics department, including remote control, high-speed photography, magnetostriction oscillators, and static electricity.

He contributed developments which are used in modern magnetic tape and wire recorders, such as high-frequency bias, improved recording heads, wire and tape materials, magnetic sound for motion pictures, multitrack tape machines, and binaural sound reproduction.

Mr. Camras is a Fellow of the IRE and the Acoustical Society of America, and a member of AIEE, SMPTE, AAAS, Eta Kappa Nu, Tau Beta Pi, and Sigma Xi.

ELECTION RESULTS

The following were elected to offices in the recent PGA balloting.

Chairman

F. H. Slaymaker,
Stromberg-Carlson Co.,
Rochester 21, N. Y.

Vice-Chairman

M. S. Corrington,
RCA Victor Division,
Camden 8, N. J.

Administrative Committee Members

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

John Hilliard,
Altec Lansing Corp.,
Beverly Hills, Calif.

CHAPTER NEWS

Portland, Ore.

William Marsh, Secretary-Treasurer of the Portland Section, is organizing a PGA Chapter in this area. All those interested in joining should contact him immediately at Tektronix, Inc., P.O. Box 831, Portland 7, Ore.

San Francisco, Calif.

The San Francisco Section reports a meeting on March 11. This was a panel discussion, and is the first of a proposed series of meetings to critically examine from both the musicians' and engineers' standpoint the present state of musical reproduction.

The following report of the January meeting is by Roy Long and is reprinted from the *San Francisco Grid*.

2-IN-1 AUDIO

A large turnout of 200 or so audiophiles consisting of members of the Professional Group on Audio and their guests attended the January meeting to hear Dr. John G. Frayne describing and demonstrating the Westrex Stereodisc System.

Stereophonic sound has been a favorite subject of conversation in audio circles for many years, but has only become popular in home installations since the recent advent of reasonably priced two-channel stereophonic tape machines and stereophonic recorded tapes. Many stereophonic phonograph systems have been tried, but never with the success of the system recently announced by Westrex Corporation, the audio and motion picture engineering subsidiary of Western Electric.

This system, in which the two channels are orthogonally recorded in a single groove, provides identical transmission characteristics in each channel, and complete compatibility with conventional microgroove records.

The meeting included a description of the stereodisc system, a description and display of the stereodisc recording head which utilizes a unique suspension system and employs feedback around the complete mechanical and electrical system, and a demonstration of a complete sample system for judgment of sound quality and compatibility.

WITH OTHER ACOUSTICAL AND AUDIO SOCIETIES

The January, 1958, issue of the *Journal of the Acoustical Society of America* contains 14 technical articles, six of which will be of special interest to the members of the Professional Group on Audio.

Richard V. Waterhouse, National Bureau of Standards, reports on his study of "Output of a Sound Source in a Reverberation Chamber and Other Reflecting Environments." It is important to know the output of a source as a function of position in a reverberation chamber. Expressions were derived for the sound power output of simple monopole and dipole sources as functions of source position in various reflecting environments. They were obtained by the use of the method of images and a general formula due to Rayleigh for the output of a number of simple sources. The cases of a dipole source near a reflecting plane and a simple source near a reflecting edge and corner are treated, and the effect of the bandwidth of the source is considered. The results apply when the reflectors enclose the source, as in a reverberation chamber, unless the distance in wavelengths between parallel walls is small and the absorption in the enclosure is low. Some experimental data are given, and the reverberation chamber method of measuring the power output of sources in discussed.

The standard of normal hearing is reviewed by John F. Corso, Pennsylvania State University, in a paper entitled "Proposed Laboratory Standard of Normal Hearing." In recent years, there has been a growing concern about the adequacy of the present American reference for normal hearing of pure tones. Results of earlier studies indicate that survey data tend to confirm the present zero reference, but data from laboratory studies have provided some basis for the contention that the present reference is too high. The purpose of the present study was to provide more definitive data on the threshold of hearing for pure tones through an extended laboratory investigation. Audiometric measurements were made on three groups of subjects ranging in age from 18 to 24 years inclusive. All subjects were otologically normal and showed a life history of minimal exposure to high intensity noise. Two groups were tested with ANB-HIA earphones and a 5-db step testing procedure in the method of limits; the other group was tested with Permoflux PDR-8 earphones and a 2-db step testing procedure. The results of the study indicate significant differences between the groups tested with the two types of earphones, as well as between male and female subjects. In general, excellent agreement was found with other laboratory studies. The threshold curves of the present study expressed in terms of coupler pressures are offered for consideration as a proposed laboratory standard for normal hearing.

The recent concern with noise and hearing impairment due to noise is reflected in an article "Improved Cushion for Ear Defenders" by E. A. G. Shaw and G. J. Thiessen, National Research Council, Ottawa, Can. Circumaural ear defenders require a cushion which is highly adaptable to the irregular contours of the head so as to provide a good air seal, and yet behaves like a stiff spring so as to minimize low-frequency cup vibration. These apparently contradictory requirements are met by a cushion consisting of an annulus of highly flexible yet rather inextensible material partially filled with an incompressible fluid. A theory of cushion behavior is given and is supported by experimental data. The design of a rigid cup and flexible frame suitable for use with the cushion also is discussed.

Weiant Wathen-Dunn and David W. Lipke, Air Force Cambridge Center, write "On the Power Gained by Clipping Speech in the Audio Band." For more than a decade it has been known, in a qualitative way, that the long-time average power of a speech signal in an amplitude-limited communications system can be increased materially by clipping the peaks of the speech wave and amplifying the remainder until the new peaks have the maximum allowable amplitude. This power increase can be computed from the statistical distribution for the instantaneous amplitudes of speech. Some of the available distributions have been collected and compared and, upon their showing good agreement, one of them has been used for the computation. Simple formulas show that the power increase can be neither greater than the amount of clipping nor greater than the peak factor of speech. For 24 db of peak clipping, the power gain is about 12 db.

"Direct Method of Accelerometer Calibration" is described by J. N. Brennan and J. S. Nisbet, Pennsylvania State University. Present methods for accelerometer calibration include small-amplitude sinusoidal drive, large-amplitude sinusoidal drive, and pulse-integration (ballistic pendulum) methods. The first of these is of limited use because it fails to reveal nonlinear effects and pulse saturation. The second is difficult to accomplish except at a limited number of frequencies, while the third gives very little high-frequency information. A method of direct calibration, using bonded wire strain gauges as a standard, is described. The method is valid over a very wide frequency range, up to 40 kc with present instrumentation.

"Dynamic Mechanical Stability in the Variable-Reluctance and Electrostatic Transducers" is treated by Charles H. Sherman, U. S. Navy Underwater Sound Laboratory. Large displacement behavior of the variable reluctance, and the mathematically equivalent, electrostatic, transducing mechanism was analyzed in order to evaluate the importance of mechanical instability as a limiting factor in high-power sound projectors. Previous work by Hunt *et al.* had shown the existence of static instability. This paper first considers the effect of transients on static stability, and then shows that instability also can occur under steady-state dynamic conditions even when the system is statically stable.

B. B. Bauer

The Columbia Compatible Stereophonic Record*

P. C. GOLDMARK†, B. B. BAUER†, AND W. S. BACHMAN†

Summary—A "compatible" stereophonic record is one which will reproduce full stereophonic sound when played on a stereophonic phonograph and will be indistinguishable from an LP record when played on a monaural phonograph. Such a record may be obtained by recording the sum signal S = L + R as lateral modulation and the difference signal D = L − R as vertical modulation, and suitably modifying D so that the tracking possibilities of monaural pickups are not exceeded.

OUR RESEARCH on the stereophonic record was a natural extension of the development of the LP record and the 16⅔ 7″ extra-fine groove record. The objective was to provide stereophonic sound equal to that of the best stereophonic tapes. The stereophonic LP record had to be compatible so that it could be played on any existing monaural phonograph as satisfactorily as today's standard LP record. This meant having the same sound level and quality in terms of frequency response, distortion and signal-to-noise ratio. The playing time of course would also have to be the same. It was felt that only in this way could there be a smooth transition from the current LP record to the new stereophonic LP record.

In order to understand more fully the theory underlying the development of the new record it is necessary to discuss briefly the problems of compatibility. This in turn requires an understanding of certain properties of monaural records and pickups.

Compatibility Problems

In a laterally modulated record the groove and stylus motions are mainly in the plane of the record. However, there is also a stylus motion normal to the record surface due to what is known as the "pinch effect." The origin of the pinch effect is illustrated in Fig. 1. Although the groove width and depth remain constant when measured along the radius of the disc, at the points where modulation occurs the cross section of the groove is narrowed down. Thus at P' the radial width W is the same as at the point P, but the cross section width W' at P' is diminished or "pinched" by the factor cos ϕ, where ϕ is the modulation angle. Due to the narrowing of the groove angle, the profiles of the groove and stylus require the stylus to rise at P' to accommodate this pinching of the groove. The amount of the resulting vertical motion can be 0.3-mil or more depending upon the recording level and groove velocity (or modulation angle) and stylus radius. If the stylus is unduly stiff in the vertical mode and thus not adapted to follow this up-and-down motion, gouging of the groove will occur

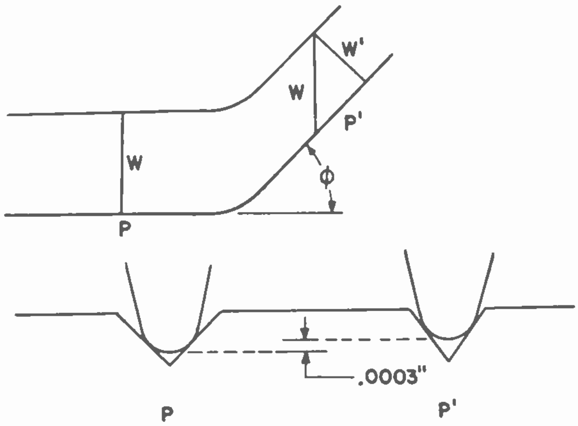


Fig. 1

and result in distorted and noisy reproduction. Another annoying effect caused by undue vertical stylus stiffness is objectionable chatter radiated from the record and the tone arm. Designers of pickups, in recognition of the importance of vertical freedom of motion, have for many years provided lateral pickups with an adequate amount of vertical compliance, which in most instances is about one- to two-thirds of the lateral compliance. Thus, most lateral pickups already have a built-in capability of handling at least 0.3-mil vertical displacement, which is then available to carry the added information required by stereophony.

The elasticity of the record material is another important factor. With a 6-gram pickup the 1-mil stylus penetrates about 2 microns into the groove wall of a vinylite record, which corresponds to about a 0.1-mil motion in the vertical direction. Therefore the total vertical displacement which any record can impose on practically all monaural pickups, without causing mistracking, is roughly 0.4-mil, or ±0.2-mil from the position of equilibrium.

Another aspect of compatibility is concerned with the manner in which stereophonic information is recorded on the disc. Since stereophony involves at least two channels, the groove must be modulated in two orthogonal directions so that one channel will not interfere with the other. Some systems of disc stereophony carry one channel as horizontal modulation and the other as vertical modulation. This results in an incompatible record because monaural pickups would reproduce only the information from the one channel which is recorded laterally.

In the new stereophonic record to be described the left and right channels are converted into a sum signal $S = L + R$ and into a difference signal $D = L - R$. The

* Manuscript received by the PGA, March 31, 1958. This paper was presented at the 1958 IRE National Convention.
† CBS Laboratories, New York 22, N. Y.

sum signal is recorded as lateral modulation and the difference signal as vertical modulation. The sum signal contains all the significant information which is presently contained on the standard LP record. The difference signal, however, carries the spatial information essential for stereophony.

To clarify the role of this difference signal in stereophonic reproduction let us take the case of two identical channels being reproduced over two separate loudspeakers. The resulting sound will appear to emerge from a single source located midway between the two loudspeakers. This in effect is monaural sound. Only when the left and the right channels differ from each other are spatial effects produced. Therefore it is the *difference* between the left and right channels when combined properly with the sum signal which produces stereophony.

A record with the $L+R$ channel recorded as lateral modulation and the $L-R$ channel as vertical modulation will be reproduced on a monaural phonograph as a conventional LP record, provided the previously mentioned restrictions as to vertical modulation limits are observed. The same record will play as a stereophonic record when reproduced with stereophonic reproducers.

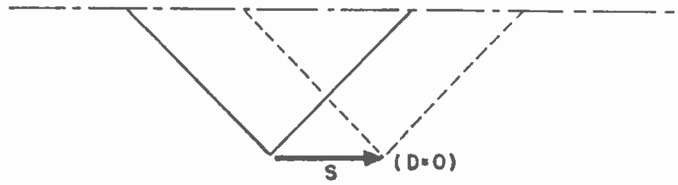


Fig. 2

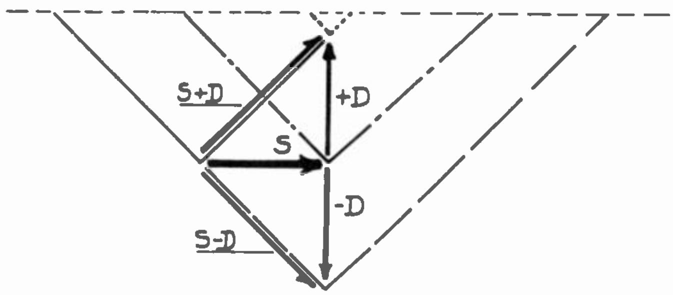


Fig. 3

Stereophonic Record

Let us first examine the type of modulation obtained when only the sum signal is recorded. The profile of such a groove is shown in Fig. 2 where the initial position of the groove is indicated in solid line (the bottom radius being omitted for simplicity). When the sum signal S is applied to the groove as lateral modulation the groove moves to the new position shown in dashed lines. In this particular case the difference information is not recorded ($D=0$), so that the apex of the groove remains in a plane parallel to the surface of the record. This leads to a standard LP record which will contain all the information of both channels except for stereophony. In Fig. 3, the difference signal has also been recorded and the effect of this D signal on compatibility will now be examined.

Let us assume that an extreme stereophonic signal is recorded with the information arriving over the left channel only, the signal from the right channel being equal to 0. In this case the sum signal will be $S=L$, and the difference signal $D=L$; thus S and D are numerically equal as represented by the vectors S and $+D$ in Fig. 3. The motion of the groove apex will follow the vector $S+D$ slanting at $+45°$. The profile of the displaced groove is shown by the small dotted triangle and it is seen that what remains of it could not guide a pickup stylus. Assuming now that the stereophonic signal is contained in the R channel only, the L channel being equal to zero, then $S=R$ and $D=-R$; therefore D is again numerically equal to S, but directed downward. The motion of the groove apex is portrayed by the vector $S-D$ slanting downward at $-45°$, and the new profile of the groove is shown in dashed lines. Note that the space occupied by this new profile is substantially greater than the space occupied by the usual monaural groove. Therefore the playing time of this record would be appreciably less than that of the LP record.

It should be noted that the system just described contains as much vertical as lateral modulation at all levels when L or R is zero. If the attempt is made to inscribe on it as much lateral modulation as is currently found on LP records, the corresponding vertical modulation will exceed the tracking capability of the current monaural pickups. It will be remembered that for compatibility the maximum vertical modulation should not exceed about ± 0.2-mil, yet it is not uncommon to find LP lateral modulation amplitude peaks reaching values in excess of ± 1-mil. Thus, to achieve compatibility the record just described would have to be recorded at a level of about 14 db lower than the standard LP level, resulting in a corresponding decrease of signal-to-noise ratio.

There exists yet another reason why this record is incompatible. In Fig. 4 the sum signal alone is represented by the arrow S and the extreme positions of the groove wall modulated by S correspond to the sidewall motion q in the normal direction. If the signal is sinusoidal then at high modulation velocities the 1-mil radius stylus may just be able to trace this modulation. If, however, a signal D is added the sidewall motion becomes q', which could be as much as twice q. The 1-mil stylus would no longer be able to trace this modulation, as the diagram at the upper left hand side shows, and therefore a considerable amount of distortion would be generated.

Because of the 45° inclination of the $S+D$ and $S-D$ vectors with respect to the horizontal, the method just described is equivalent to the so-called 45/45 system. This system poses a dilemma: either the over-all recording level of both lateral and vertical modulation has to be appreciably reduced to permit tracking with the existing monaural pickups or, if the level was maintained

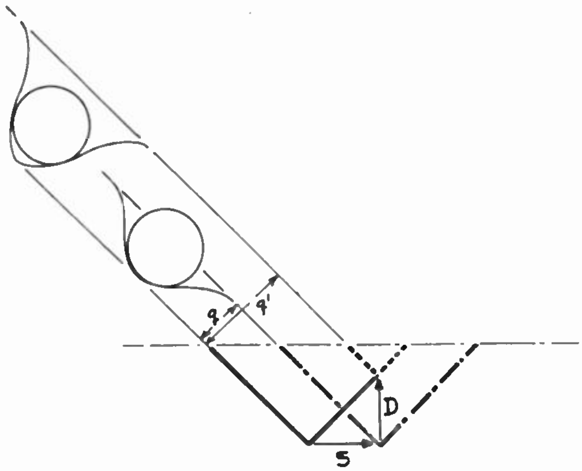


Fig. 4

as in existing LP records, the majority of existing pick-ups would not track it. Other serious problems would be the generation of distortion and the reduction of the program length. Therefore neither choice would yield a satisfactory compatible record.

The New Compatible Record

During the many years of research on stereophonic records it was the writers' objective to develop a system which would be completely compatible with existing phonographs yet would convey the required stereophonic impression to the listener. The new record system which meets these requirements is based on the following principles.

Since it was desired to produce a stereophonic record with the same sound level as the LP record which could be tracked with existing monaural pickups, the answer seemed to lie in a stereophonic record with full lateral but appreciably reduced vertical amplitude.

Studies were made of the nature of stereophonic sound radiated by two loudspeakers separated by a sufficient amount of space. The composite sound arriving at the observer was analyzed in terms of the sum $S=L+R$ and the difference $D=L-R$ signals. It was found that by far the most significant portion of the energy of the radiated sound was conveyed by the sum signal. Following this, it was determined what was the minimum amount of difference signal needed to achieve full stereophonic effect as a function of the frequency and of intensity.

The results were significant and a law was evolved according to which the difference signal was limited by varying amounts at different frequencies. An "electronic brain" was then developed (called the ASRA, for Automatic Stereophonic Recording Amplifier) capable of modifying the difference signal automatically in accordance with the established requirements. When it was inserted between the original stereo tape and a stereophonic monitoring speaker system as shown in Fig. 5, instantaneous switching of ASRA in and out of the circuit gave most critical evaluation of the modified dif-

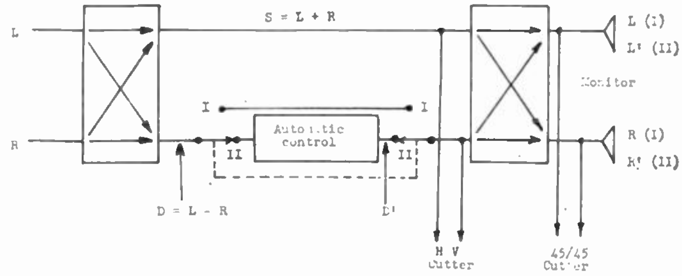


Fig. 5

ference signal. For these rigorous comparative tests the sum and difference signals were converted through matrixing into right and left channel signals, $L=(S+D)/2$ and $R=(S-D)/2$, which were then reproduced through two conventional amplifiers and loudspeakers.

If D was the original difference signal extracted from the master tape by matrixing, and D' the modified signal produced by the ASRA, when switching back and forth for comparative tests the following right and left hand signals were heard:

For D (switch position I)

$$L = \frac{S+D}{2}$$

$$R = \frac{S-D}{2}.$$

For D' (switch position II)

$$L' = \frac{S+D'}{2}$$

$$R' = \frac{S-D'}{2}.$$

With adjustments set to optimum conditions, switching between I and II gave no detectable differences. The settings of ASRA were then suitable for cutting a compatible stereophonic record from a two-track master tape. The ASRA was located between the master tape and the recording amplifiers of the stereo cutter. If the latter was the so-called 45/45 type the S and D' outputs of ASRA were rematrixed to furnish R' and L'; if a lateral-vertical type cutter was used, no rematrixing was needed and S and D' were directly supplied to the recording amplifiers.

The type of modulation obtained with the use of the ASRA is shown in Fig. 6. If, for example, $R=0$, then $S=L$, but the modified difference signal D' does not generally equal L. Thus, depending on frequency and amplitude, $D' \leq L$. At peak level of modulation the displacement of the groove apex would be as shown by the vector $S+D'$ which is directed at an angle α. For $L=0$ the displacement of the groove apex will be shown by the vector $S-D'$, directed at α. It should be noted that

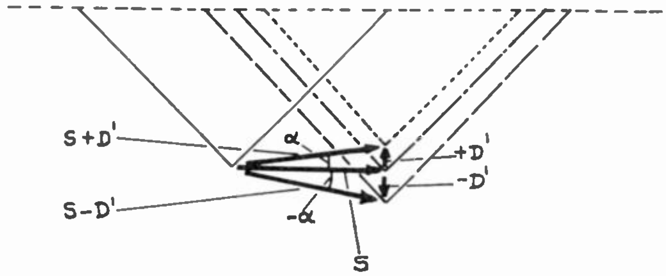


Fig. 6

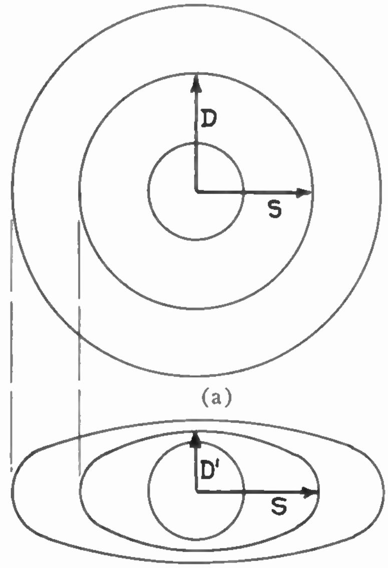
the groove profile remains similar to that of the LP record (Fig. 2) and thus there are no significant tracking or distortion problems when used with a monaural pickup.

Because α may assume a variety of values when $R=0$ or $L=0$ depending upon frequency and amplitude, the resulting method of modulation was called "elliptical modulation," where the large axis is S and the small axis D'. For small values of S and for certain frequencies the eccentricity of the ellipse is 0 or $S=D'$. This type of modulation is shown in Fig. 7(b). In the conventional stereophonic systems D and S are always equal when $R=0$ or $L=0$ and therefore the eccentricity is always 0, as shown in Fig. 7(a).

In the new stereophonic system the maximum amplitude of the vertical modulation is only about one-eighth that of the lateral amplitude, though the ratio of the two can approach unity at low levels as well as at certain frequencies. Because of the small amount of vertical modulation the wear qualities of the new stereophonic records are equal to those of the standard LP record as confirmed by tests.

Stylus wear was also investigated using the popular low-cost cartridges. Here again stylus wear was found to be the same as with standard LP records.

Any type of stereophonic pickup can be used with the new compatible stereophonic record. Diagrams of several possible connections to be used with lateral-vertical and the so-called 45/45 pickups are shown in Fig. 8.



(a)

(b)

Fig. 7

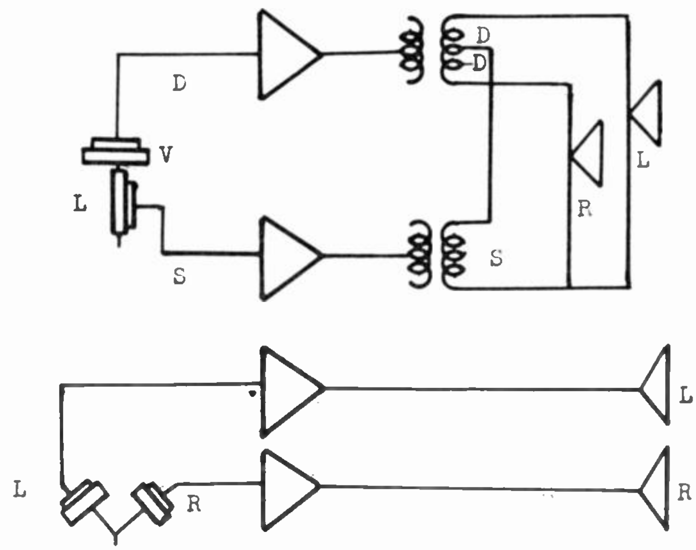


Fig. 8

Modulation Noise in Magnetic Tape Recordings*

R. LEE PRICE†

Summary—**Modulation noise remains as a major limitation on the dynamic signal-to-noise ratio obtainable in a magnetic recording system. The sources of modulation noise in magnetic tape recordings have been investigated by reducing to the smallest possible value the inherent noise of the recording and playback equipment used in the tests. Possible causes of modulation noise are then independently introduced and their effects measured. The results of these measurements indicate that when recording and reproducing equipment is properly adjusted, the major source of modulation noise is spurious amplitude modulation of the recorded signal by variations in the physical and magnetic properties of the recording tape. The amount of this modulation noise is found to increase above the value obtained with a dc signal, in proportion to the recorded signal frequency. Recommendations are made for measurement of amplitude modulation noise as a check on tape quality and on those adjustments which affect amplitude modulation noise. This measurement is in addition to the measurement of frequency modulation noise commonly known as "flutter."**

Introduction

MODULATION noise, or noise behind the signal, is a phenomenon which occurs in disc[1] and photographic records[2] as well as in tape and wire recordings. Modulation noise is defined[2] as "noise which exists only in the presence of a signal and is a function of the instantaneous amplitude of the recorded signal. (The signal is not to be included as part of the noise.)"

It has been the writer's experience in working with high quality magnetic tape-recording systems that in many cases the most obvious remaining fault is the presence of modulation noise. This modulation noise often varies in degree among different tapes, all other things being equal.

S. J. Begun[3] has this to say concerning modulation noise:

"With dc biasing (in a magnetic recording), the noise level is practically constant, independent of the magnitude of any signal which may be recorded, since with bias only it is already at about the highest value it can assume. When ac biasing is used, however, the statistical variations in the medium are, as it were, *developed* by the presence of the recorded remnant induction, and they appear as a noise component in the output which is roughly proportional to the instantaneous magnitude of the recorded signal. The signal functions as a *carrier* for the noise and is modulated by it.

"When a signal is recorded on an inhomogeneous medium using ac biasing, and is subsequently viewed on an oscilloscope during playback, the signal peaks are seen to be modulated by noise, but the points where the signal crosses the zero axis show practically no modulating. This phenomenon is known as *the noise behind the signal* or *modulation noise*, and its magnitude can be approximately predicted from measurements of the noise which appears when dc fields are superimposed upon ac biasing field and applied to a magnetically neutral medium.

"Noise behind the signal is much less objectionable to the ear than the constant noise level that appears when dc biasing is used. Moreover, since the noise goes down as the magnitude of the recorded signal decreases, ac biasing results in a much greater apparent signal to noise ratio for inhomogeneous mediums than does dc biasing. With mediums composed of powdered magnetic particles bound to a nonmagnetic base, for example, the signal to noise ratio with dc biasing may be less than 20 db, whereas for ac biasing the signal to noise ratio may exceed 60 db, even though the noise for a given value of signal may be only 20 db below the signal.

"The fact that noise behind the signal is generally less objectionable than constant noise is due to the masking property of the human ear which hears a weak tone as even weaker in the presence of a strong tone. Noise behind the signal, if the ratio exceeds about 35 db, is normally not perceptible, but it may give a somewhat *fuzzy* sound to the tones of certain instruments."

This quotation regarding the seriousness of modulation noise in its effect on the reproduced signal is concerned primarily with audio frequency applications in which the recorded signals are intended for reception by human ear. Magnetic recording has many other possible applications outside of the audio field, *e.g.*, instrumentation, data recording, etc. In applications such as these, where the maximum information handling capacity of a recording channel must be utilized, the dynamic signal-to-noise ratio, which includes modulation noise, is a limiting factor rather than the static signal-to-noise ratio which is the one usually given by equipment specifications.

L. C. Holmes[4] has written: "The background noise in the better magnetic recording systems is usually so low that the existence of other faults, such as modulation noise, or the transfer effect, is apparent because they are not sufficiently masked by the noise."

There are many more references to modulation noise in the literature which will be discussed in this paper. Those given above, however, will serve as an introduction to the problem.

Magnetic recording of sound is not a recent invention. The first wire recorder was developed by Valdemar Poulsen in Denmark before 1900. This early Telegraphone was seriously hampered by the lack of suitable amplifiers, and by the type of wire which was used as the recording medium. As amplifiers and improved magnetic materials were developed, in the 1930's magnetic recorders of better quality became possible.

When our Armed Forces occupied Germany during World War II, they found that tape recorders were in wide broadcast use there and gave results that were far superior in fidelity and signal-to-noise ratio to those obtained with our wire recorders. Present-day tape recorders have for the most part evolved from these German Magnetophones using plastic base tape with thin coatings of iron oxide.

* Manuscript received by the PGA, March 13, 1958.

† TelAutograph Corp., Los Angeles 45, Calif.

[1] E. G. Cook, "Noise modulation in recording," *Audio Eng.*, vol. 31, p. 11; December, 1947.

[2] "Standards on sound recording and reproducing: methods and measurement of noise," Proc. IRE, vol. 41, pp. 508–512; April, 1953.

[3] S. J. Begun, "Magnetic Recording," Rinehart Books, Inc., New York, N. Y., pp. 70–71; 1949.

[4] L. C. Holmes, "Techniques for improved magnetic recording," *Elec. Eng.*, vol. 68, p. 840; October, 1949.

The present superiority of magnetic tape recording over disc, wire, photographic records, etc., is due largely to the development of the iron oxide coated tapes which are used as the recording media. Recent improvements and manufacturing techniques have made possible an extremely uniform product with greatly improved signal-to-noise ratios.

Another factor which helped both the Germans and ourselves to reduce noise level and distortion on magnetic tape recordings was the use of a supersonic ac bias during recording. Although this technique was patented by W. L. Carlson of the U. S. Navy in 1927, authorities on the subject are not yet in complete agreement on the theory of operation of ac bias. Experimental work has shown, however, that for optimum results it is essential to use the correct value of supersonic current during recording, for it affects the frequency response, the distortion, and the signal-to-noise characteristics of the recordings.

In modern magnetic tape recordings, the medium on which the actual magnetic modulations are impressed is a layer of finely powdered crystalline magnetic particles evenly coated upon a paper or plastic base. This base material is normally 0.25 inch wide and 0.0015 inch thick and the magnetic coating is approximately 0.0006 inch thick. Thus the total thickness of the tape and its magnetic coating is approximately 0.0021 inch.

The coating material used in modern recording tapes is far from perfectly homogeneous. In fact, it is a heterogeneous mixture of iron oxide (Fe_2O_3) and a suitable binder. The actual composition of the coating material used and its method of manufacture and application are considered proprietary or trade secrets by most manufacturers.

Extreme care must be exercised in the manufacture of this material. Particle size should be fairly uniform since when wide variations in particle size occur, coating uniformity also varies. Nonuniform coatings contribute to variations in recorded signal amplitude and to noise, as will be brought out in this paper.

An estimate of the particle sizes required for magnetic recording tape coatings can be obtained by a consideration of the dimensions of the recorded signals.

The relationship between wave length, tape speed and signal frequency is given by:

$$\lambda = \frac{S}{F}$$

where

λ = recorded wave length

S = tape speed

F = recorded signal frequency.

Thus, when a 15,000 cps signal is recorded at a tape speed of 15 inches per second, the resulting wavelength on the tape is 0.001 inch and the distance between points of opposite polarity is one half wavelength or 0.0005 inch. If we assume that there should be at least 10 magnetic particles between opposite polarities for recording that is a good approximation to the desired waveform, a particle size of 0.00005 inch (approximately one micron) or less is indicated.

If the distribution of the magnetic material within the binder is not extremely uniform, a characteristic clumping will take place. This deposits an uneven coating having nonuniform magnetic properties as well as surface irregularities. In order for a coating of 0.0005 inch nominal thickness to be uniform within 2 per cent, it should not deviate more than approximately 0.0005×0.02 or ten millionths of an inch from constant thickness.

Both surface smoothness and uniform magnetic properties of the tape coating are necessary for low modulation noise. Surface irregularities of the base material will also affect the uniformity of the magnetic coating. Irregularities in the surface smoothness of the base will result in corresponding irregularities in the thickness and surface of the magnetic layer. It is for this reason that plastic base materials such as cellulose acetate and Mylar, a Du Pont polyester film, because of their inherently smoother surfaces, provide much lower levels of modulation noise than do paper base tapes. When paper or any other rough surface is used as a base and is covered with a magnetic coating, minute irregularities in the surface of the base will result in minute irregularities in the thickness of the magnetic layer. If the distribution of the magnetic material within the binder is not extremely uniform, a characteristic clumping of the magnetic particles will take place. This clumping deposits an uneven coating on the base material, having an irregular surface as well as nonuniform magnetic properties. These surface irregularities and magnetic nonuniformities are important sources of modulation noise as will be shown in more detail by the experimental data.

Methods

It has been established by observation and review of the literature that in good quality magnetic tape-recording systems modulation noise is much higher than system noise in level, and is thus a limiting factor on the signal-to-noise ratio which is obtainable under dynamic conditions. Since there seems to be some disagreement among authorities concerning the true sources of modulation noise, this investigation was undertaken in an attempt to separate and evaluate independently the major sources of modulation noise. If this were done successfully, it should then be possible to minimize modulation noise by using appropriate test procedures and applying corrective measures in a logical manner.

The possible sources of modulation noise which were given in the literature included:

1) Sidebands due to frequency modulation of the recorded signal by fluctuations in the tape speed at the playback head relative to that at the recording head.

2) Sidebands due to amplitude modulation of the signal by variations in head-tape spacing or contact during recording and playback.
3) Sidebands due to amplitude modulation of the signal by variable thickness and variable magnetic properties of the tape coating.
4) Magnetostriction noise in the magnetic playback head.
5) Barkhausen phenomena in the recording and reproducing heads.
6) Barkhausen phenomena in the magnetic coating of the tape.
7) Signal transfer or "print through" from layer to layer of the tape.
8) Intermodulation between the signal and the noise in other parts of the spectrum, particularly in the lower frequencies.

All these noise mechanisms are possible sources of modulation noise. However, due to the large numbers of variables involved and the lack of quantitative data, it was decided to approach this problem on an experimental basis, making use of those theoretical considerations which could be verified by experiment. This was done by reducing all the known sources of system noise and modulation noise to their lowest possible values. The suspected sources of modulation noise listed above were then introduced and measured individually, so far as was possible, to determine the contribution of each. The results of these experiments and the conclusions drawn from them are given in following sections of this report.

In an experimental investigation of this sort, it is convenient if the definitions and methods of measurement are in agreement with the industry standards which are in existence. In this way, interpretation, comparison, and correlation of the results obtained with the findings of other investigators and authorities are made easier.

In this investigation, the definitions and methods used were those given in the IRE Standards[2,5] unless otherwise stated. The principal exception in the present investigation is the definition of noise which is given in paragraph 1.2.1 of the IRE Standard[2] which defines noise:

"As applied to a sound recording and reproducing system, any output power which tends to interfere with the utilization of the applied signals except for output signals which consist of harmonics and subharmonics of the input signals, intermodulation products, and flutter and wow."

In the list of possible sources of modulation noise which is given in this report, 1) lists flutter as a possible source of modulation noise and 8) lists intermodulation between signal and system noise as a possible source of modulation noise. Therefore, in this investigation the above definition of noise will be enlarged to include all spurious output signals with the exception of those harmonically related to a sine wave recorded test signal.

[5] "Standards on electron devices: methods of measuring noise," Proc. IRE, vol. 41, pp. 890–896; July, 1953.

Three methods have been suggested in the references for the measurement of modulation noise. Several authorities, recognizing that tape having high dc bias noise level also has a high modulation noise level, have recommended that the dc bias noise level be used as a comparative measure of the modulation noise to be expected when recording with ac bias.

The IRE Standards Committee recommends a direct measurement of modulation noise, by means of an extremely narrow band rejection filter to reject the causative modulation signal, leaving the noise to be measured. This method is admittedly difficult to apply and subject to error, because of the difficulty of maintaining the speed of the recording and reproducing machines within an accuracy sufficient to insure that the signal frequency will remain within the rejection band of the filter. An additional difficulty with this method is that the modulation noise sidebands which are very near the signal frequency are unavoidably attenuated, resulting in an error in the noise measurement which depends on the filter bandwidth.

In the most recent paper which was found concerning the measurement of modulation noise, W. H. Erickson of RCA[6] recommended the use of an intermodulation analyzer for the measurement of modulation noise on the playback of a 2000 cps recorded signal. It was reported that this method of measurement gave very good correlation with listening tests.

The first step in this experimental investigation consisted of outlining the test procedures and methods to be used.

The test equipment setup for measurement of spectral noise density is shown in Fig. 1. This setup was used for all spectrum measurements of output signals and noise. A Hewlett-Packard model 300A Wave Analyzer was used as the measuring instrument with which all signal and noise voltage measurements were made across a load resistance of 600 ohms. The voltage readings obtained were plotted in decibels relative to a zero level of one volt. In determining the voltage level of random noise readings, the average reading over a period of several seconds is the one which was entered in the tabulation of data. The wave analyzer was set to its most selective position.

A feature of this test equipment setup is the method for determining the frequency to which the wave analyzer is tuned. The frequency dial of the wave analyzer gives only an approximate indication of this frequency and cannot be read to an accuracy which approaches that which is necessary for this application. As shown in Fig. 1, the wave analyzer is connected through the switch S either to output signal under measurement, or to an audio oscillator whose frequency in cycles per second is being continuously counted by a Berkeley

[6] W. H. Erickson, "Magnetic Tape Testing on a Comparison Basis," Dept. of Defense Symposium on Magnetic Recording, Dept. of the Navy, BuShips, Code 362, Washington 25, D. C., Paper No. 13, pp. 1–13.

Model 554 counter, or EPUT meter. By switching the wave analyzer to the output of this oscillator and tuning the oscillator for maximum wave analyzer indication, the resonant frequency of the wave analyzer may then be read from the reading indicated on the counter. A Ballantine ac vacuum tube voltmeter was connected in parallel with the wave analyzer as a check on the signal and noise levels and an oscilloscope gave a visual indication of the signal and noise waveforms. The decade attenuator between the Hewlett-Packard 200CD audio oscillator and the recording amplifier input was useful in changing the input level by a known number of decibels when making input level changes.

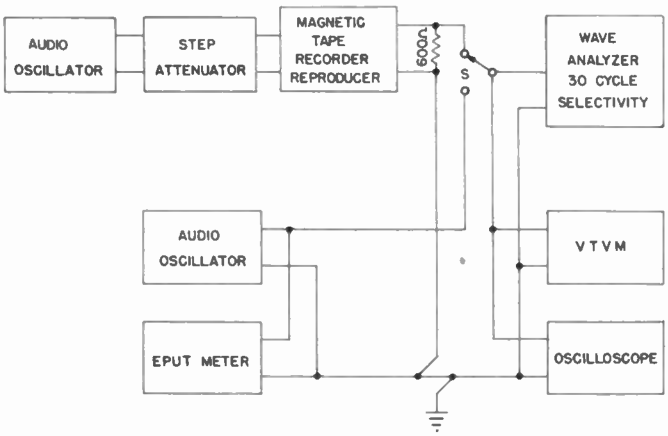


Fig. 1—Test equipment setup for measurement of spectral noise density.

Fig. 2 shows the test equipment setup for measurement of the amplitude modulation of the signal by the recording and playback processes and by the nonuniformities of the tape. As shown in this figure, the playback of a signal which was recorded at constant level is fed to an Altec Lansing TI-402 Intermodulation Analyzer. This intermodulation analyzer consists of an amplifier and a push-pull diode detector which are preceded by a high-pass filter to remove signals below the carrier frequency range. The diode detector is followed by a low-pass filter which removes frequency components above approximately 700 cps. This analyzer indicates the average amplitude modulation of rates up to approximately 700 cps on a carrier of any audio frequency above approximately 1200 cps. A panel meter on the analyzer is calibrated directly in percentage of modulation. In addition, the detected output from the low-pass filter is also available at terminals for examination with a wave analyzer or oscilloscope as in Fig. 2. The spectrum of the sidebands due to amplitude modulation is measured with the wave analyzer at the detected output of the intermodulation analyzer as shown.

The flutter meter which was used for measuring frequency modulation comprised a limiter and frequency sensitive discriminator. This discriminator is tunable to a center frequency of approximately 3000 cps and its output, which is proportional to the frequency deviation from this center frequency, is fed to an indicating panel meter and to an output jack which may be connected to an oscilloscope. The indicating meter is directly calibrated in per cent flutter and indicates the root-mean-square value of the frequency deviation at flutter frequencies between approximately 2 and 200 cps.

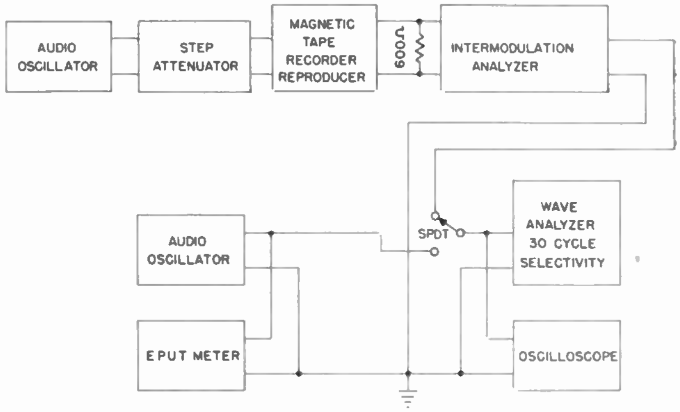


Fig. 2—Test equipment setup for measurement of amplitude and spectra of amplitude modulation sidebands.

It was not possible to measure accurately the spectrum distribution of the flutter rate signals from the discriminator output with the wave analyzer since these signals were predominately low in frequency. The usual procedure, which was the one followed here, is to observe the flutter waveform on an oscilloscope. With this arrangement, the waveform of the output signal from the discriminator in the flutter meter is observed on the oscilloscope. Comparison of this waveform with the calibration of the vertical gain and the horizontal sweep frequency of the oscilloscope makes possible a fairly good estimate of the relative amplitudes and rates of the major flutter components.

In making low level noise measurements such as from a magnetic tape playback head, it is important that amplifier noise, microphonics, outside disturbances, etc., be reduced to a level well below that of the noise being measured. In the present investigation, several means were employed to minimize these effects. The input stage noise figure of the playback amplifier was reduced from approximately 28 to a measured value of 9 (a reduction of 5 db) by the use of a cascode input stage adapter. This adapter was especially designed and constructed for this project by the author,[7] and it was used to replace the type 12SJ7 tube in the playback amplifier input stage during all the tests made in this investigation. Microphonic disturbances were minimized by the use of a low microphonic type GL-5814 premium tube in this playback amplifier input stage adapter, and by carefully shock mounting the entire amplifier chassis on very soft padding. Outside electrical disturbances were minimized by the use of a double shielded room in which all noise measurements were made. All ac power enter-

[7] R. L. Price, "Cascode audio amplifier has low noise level," *Electronics*, vol. 27, pp. 156–157; March, 1954.

ing this shielded room came from a Sola type CVH constant voltage transformer. In addition, the ac line input to the shielded room was connected through a noise suppression filter.

The tape recorder/producer used in this investigation was an Ampex Model 400. This machine was chosen because it is a typical, good quality professional machine, capable of low flutter and noise level. Also, it has a dc injection, or noise balancing control in the recording head circuit so that dc noise can be reduced to a minimum value, or set to any other desired level.

A word of explanation concerning the effect of this dc balance current is appropriate at this point. One of the advantages of magnetic recording is that the medium is inherently free of even harmonic distortion due to the symmetrical nature of the transfer characteristics. When even harmonic distortion does occur in the recording process, it is usually due to an equivalent direct current component of magnetization which prevents the recording heads from modulating the tape about the point of symmetry which corresponds to the state of demagnetization. This condition is also necessary for minimum tape noise. A direct component of magnetization may exist at the recording head due to magnetization of the head, or to stray magnetic fields from other pieces of equipment. Even if no dc current is present, a signal or bias current waveform having different positive and negative peak values will produce the same effect. That is, the unequal peaks of the bias current waveform will result in a dc component of magnetization which in turn will result in second harmonic distortion of the recorded signal and an increase of the tape noise level. The noise owing to this cause is called "dc noise."

In practice, it is not usually possible to reduce the above causes of asymmetry to zero. An alternative remedy is to pass an adjustable amount of direct current through the record head along with the high frequency bias current. The polarity and magnitude of this current can then be adjusted to neutralize dc magnetic fields from other sources. The balance current for minimum even harmonic distortion and minimum tape noise are the same since both have the same cause.

The tape recorder/reproducer used in this investigation was completely checked over and adjusted electrically and mechanically to put it in the best possible operating condition. The total indicated root-mean-square flutter was reduced to 0.07 per cent.

The harmonic distortion in the playback of a 400 cps signal, which was recorded at the established maximum or zero reference level specified by the equipment manufacturer, was slightly less than one per cent. The intermodulation between 100 cps and 7000 cps signals recorded at zero reference level and having a 1:1 ratio was approximately five per cent as indicated on the Altec Lansing TI-402 Intermodulation Analyzer. These figures and measurements agree quite closely with the specifications given by the manufacturer of the recording equipment.

In this connection, a few words concerning the necessity of frequency response equalization in audio frequency magnetic tape recording/reproducing systems are in order. A constant current amplifier connected to a recording head would, in the mid-frequency range, produce very nearly a constant magnetic flux vs frequency pattern on the tape. The playback head produces an output voltage which is the derivative of the recorded flux pattern on the tape, at frequencies below which gap effect becomes important. At these mid-range frequencies, therefore, playback equalization in effect integrates the signal on the tape. It may be said that at these frequencies the output signal and noise voltages are very nearly proportional to the flux on the tape. High frequency losses are due mainly to the fact that the length of the playback gap is no longer negligible as compared with the wavelength of the recorded signal. The loss in decibels due to the gap effect is:

$$20 \log_{10} \frac{\sin \pi\theta/\lambda}{\pi\theta/\lambda}$$

where θ is the effective playback gap length and λ is the recorded wavelength. This gap effect results in a frequency characteristic as shown in Fig. 3(A) when a constant current recording is reproduced without equalization. Because of this gap effect, it is customary to equalize, or pre-emphasize, high frequencies during recording as shown by Fig. 3(B). There is also a slight amount of pre-emphasis of low frequencies due to the effect of playback head geometry on the reproduction of long wavelengths.

The final step in preparing the recorder/reproducer for the experimental portion of this investigation consisted of demagnetizing as completely as possible the erase, record and playback heads.

The tape used for these tests, unless otherwise stated, is Minnesota Mining and Manufacturing Company's No. 109 Instrumentation Tape. All tests were made at a tape speed of 15 inches per second.

Results of Experiments and Observations

Fig. 4(B) shows the frequency distribution of the steady state or system noise which is present independently of the signal. The data for these curves was taken with the test setup of Fig. 1. The shape of this curve follows approximately that of the playback amplifier frequency response curve. There is some increase of noise at low frequencies due to "cathode flicker" in the playback amplifier input stage.

Fig. 4(D) is the noise spectrum obtained from the passage of bulk erased tape over the playback head. Bulk erased tape is tape which has been demagnetized as completely as possible by a decaying 60 cps field while on its storage reel. Bulk erasers are manufactured specifically for this purpose, and it is possible to obtain much better erasure by this means than by any other known at the present time.

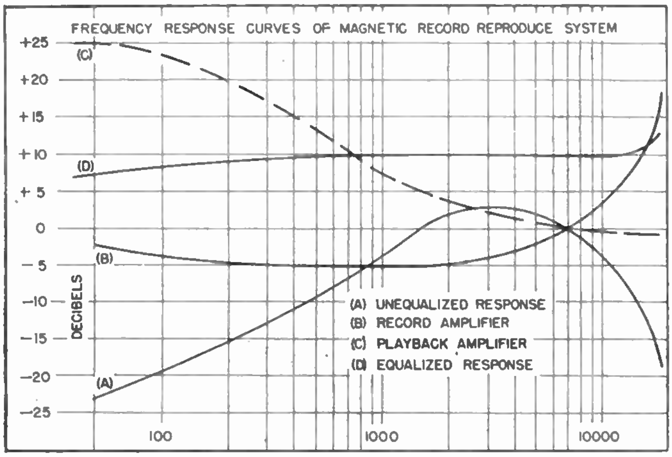


Fig. 3

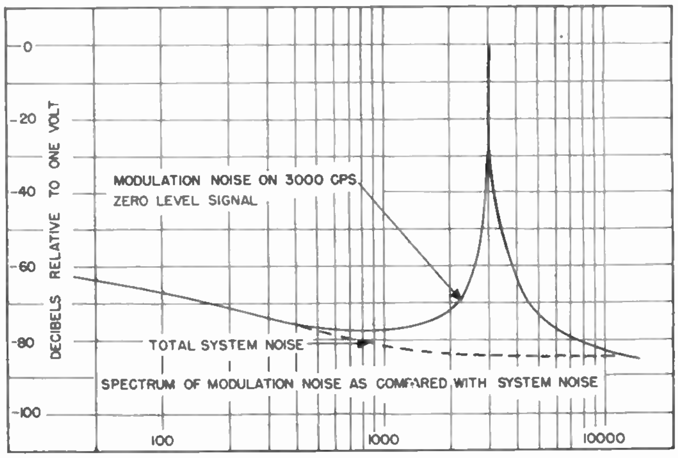


Fig. 5

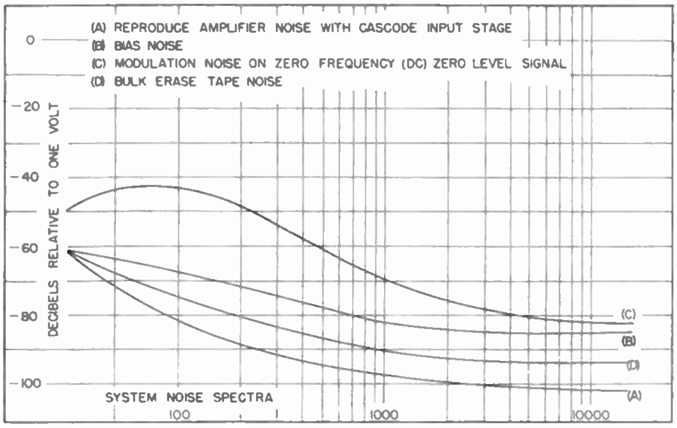


Fig. 4

Fig. 4(B) is the noise from tape which has been subjected to supersonic erase and bias fields. Even though the noise balance control in the recording head circuit was carefully set for a minimum noise level, the noise is above that from bulk erased tape. This curve demonstrates the truth of statements which were found in the literature to the effect that supersonic erase heads do not leave the tape in as nearly a neutral magnetic condition as does a bulk eraser. Comparing curve (A) with curve (B) in Fig. 4 shows that the total system noise is approximately 20 db above the playback amplifier noise at all except very low frequencies. Therefore, playback amplifier noise can probably be neglected at the frequencies in the vicinity of 3000 cps with which we will be concerned.

Curve (C) of Fig. 4 is the noise resulting from the recording and playback of a dc or zero frequency signal. Even though the playback head and amplifier cannot reproduce this dc signal, the modulation noise which results is as shown here. The value of this dc current was established by observing that an ac signal of 0.25 ma rms in the record head resulted from a zero vu input signal as indicated on the recording level meter, with the ac bias signal temporarily disabled. This zero vu level has been established by the recorder manufacturer as that which results in one per cent harmonic distortion caused by the recording processes on a test signal of 400 cps. The playback amplifier gain is fixed so that this recording level results in zero volume units output which is equivalent to 0.77 volt across 600 ohms, or one milliwatt. The dc noise balance control was then readjusted so that the total direct current in the record head winding was equal to the noise balance current plus 0.25 ma. The sum of these dc currents plus the normal ac bias current was then equivalent in effect to a zero level recorded signal of zero frequency.

In Fig. 5 is shown the frequency distribution of the output signals which were measured with the wave analyzer in the test setup of Fig. 1 during the playback of a 3000 cps recorded signal. A comparison with the system noise curve which is repeated here from curve (B) of Fig. 4 shows the increase in noise level which occurs at frequencies adjacent to the recorded signal frequency. This increase in noise level with signal level is the modulation noise which is the subject of the present investigation.

The first possible source of modulation noise which was investigated experimentally was that of intermodulation between the recorded signal and noise in other parts of the spectrum, particularly at low frequencies. A family of input-output curves is shown in Fig. 6. These are curves of playback amplifier output signal level as a function of record amplifier input signal level for low, medium, and high frequencies. A perfect system with no intermodulation would be linear with all points falling on a straight line. These curves show good linearity for recording levels up to the established maximum or zero reference level. Above this zero reference level, saturation effects take place which vary with frequency as shown.

It is possible that in a system having distortion which is independent of level, a linear input-output relationship would not be a sufficient condition for the absence of intermodulation. Therefore, as a further check on the possibility of intermodulation, an intermodulation

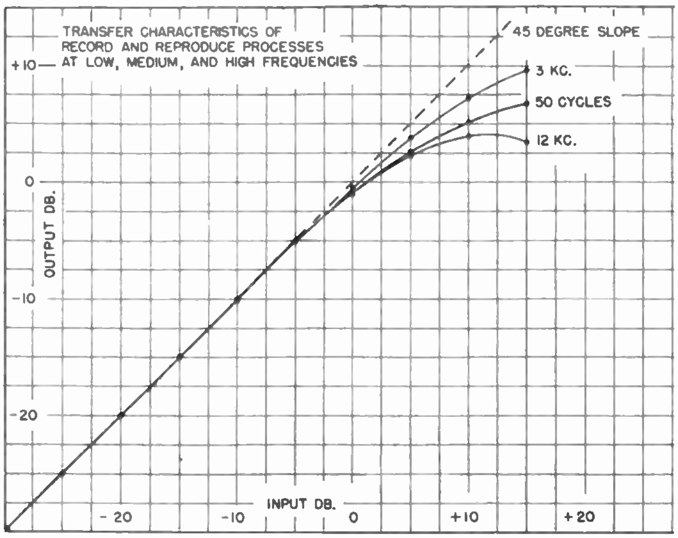


Fig. 6

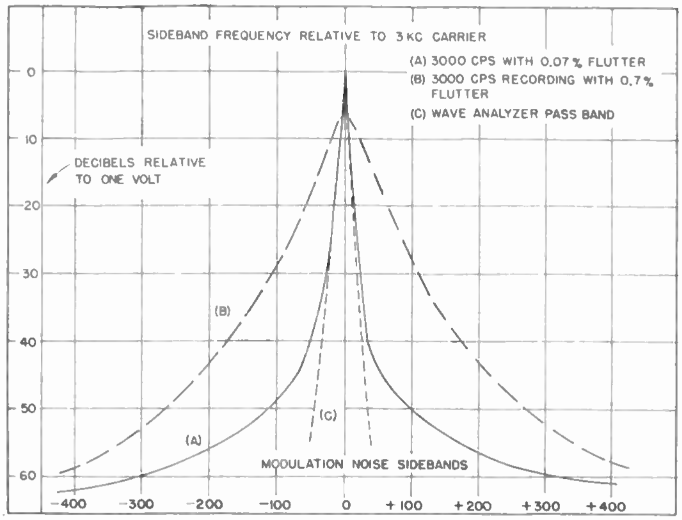


Fig. 7

analyzer was used to measure intermodulation directly. Two signals of 2000 cps and 100 cps respectively have a 1:1 ratio and were recorded at zero reference level. On playback, the resulting modulation of the high frequency by the low frequency was approximately five per cent. These two signals were then recorded again but with a 10:1 ratio of the 2000 cps to the 100 cps signal level. The indicated amplitude modulation on the high frequency signal with this 20 db signal ratio was 1.2 per cent which is equal to that due to modulation noise on the 2000 cps signal alone, intermodulation with the 100 cps signal being undetectable. The results of these tests indicate that the intermodulation between a zero level signal and system noise which is 60 db or more lower in level would not be an appreciable part of the total observed modulation noise.

The second source of modulation noise to be investigated was the effect of flutter in producing sidebands adjacent to the recorded signal, owing to frequency modulation of the signal by variations in the velocity of the tape at the playback head relative to that during recording.

Curve (A) of Fig. 7, plotted from the data for Fig. 5, shows the modulation noise in the playback of a 3000 cps recorded signal containing approximately 0.07 per cent rms flutter. The presence of modulation noise sidebands results in the broadening of the reproduced signal beyond the pass band of the wave analyzer as shown.

The data for curve (B) of Fig. 7 was taken under the same conditions as for curve (A), but with 0.7 per cent of capstan rate (20 cps) flutter introduced on playback of the 3000 cps signal. This flutter was produced by fastening small sections of adhesive tape on the capstan surface to increase its eccentricity. The waveform of the flutter was observed on the oscilloscope connected to the flutter meter output and was found to be approximately a sine wave. Curve (B) shows that with this amount of flutter, which is an increase of ten times over the former value, the amplitude of the fundamental decreases, while the sidebands immediately adjacent to the fundamental increase in amplitude. This is to be expected from frequency modulation theory. At frequencies greater than approximately 400 cps on either side of the carrier, however, the increase in modulation noise with flutter is not noticeable.

Curve (C) of Fig. 7, the pass band of the wave analyzer, is included here to indicate the effect of the resolution of the test equipment in obtaining the data for these curves.

These measurements indicate that if a tape transport mechanism introduces a relatively large amount of flutter into the tape motion, this flutter will in turn result in frequency modulation sidebands at frequencies immediately adjacent to the carrier signal frequency. When the flutter is of low percentage, as in curve (A) of Fig. 7, it is not evident that flutter contributes appreciably to the existing modulation noise since the first order sidebands of the low flutter known to be present are apparently obscured by the finite pass band of the wave analyzer. This point is discussed in detail in Appendix A.

It should be mentioned at this point that in practice there often occurs high frequency flutter of several thousand cycles per second. This type of flutter is generally caused by the longitudinal and transverse vibrations of the tape itself between the points of support and guidance, rather than by irregularities in the tape driving mechanism. Although a small amount of this type of flutter was present during these measurements, it was minimized by damping of the tape and was not of sufficient magnitude to have an appreciable effect on the total modulation noise.

The third source of modulation noise to be investigated was the effect on the reproduced signal of amplitude modulation caused by tape nonuniformities. In order to separate the sideband spectrum due to amplitude modulation from that due to frequency modulation or flutter, the test setup of Fig. 2 was used. The detector output of the intermodulation analyzer corresponds to the envelope of the amplitude variations of the playback signal. Since diode detection is used, the readings are not appreciably affected by flutter or changes in signal frequency within the operating range of the instrument. The frequency distribution of the detected output from the intermodulation analyzer was determined by means of the wave analyzer and is plotted as curve (B) in Fig. 8. The level of these rectified sidebands relative to the carrier was established by observing the intermodulation analyzer detector output when a ten per cent amplitude modulated calibration signal was supplied to the detector input. This output signal voltage corresponded to a sideband which was 26 db below the carrier level and the noise sidebands were referred to this same level.

Curve (A) of Fig. 8 is an average of the two modulation noise sidebands plotted from the data of Fig. 5. Only one sideband is shown since these sidebands are symmetrical with respect to the carrier and this sideband is plotted in terms of its frequency departure from its carrier. Curve (C) of Fig. 8, the modulation noise resulting from a zero frequency signal, is plotted from the data of curve (C) of Fig. 4 to show the similarity to the sideband of 3000 cps. Curve (D) of Fig. 8, the wave analyzer pass band, is repeated to show the resolution of the test equipment.

In an effort to reduce the amount of amplitude fluctuations in the reproduced signal, which were caused by possible variations in the contact between the tape and the recording and playback heads, felt pressure pads were improvised. These pressure pads were made by cutting squares of felt to a convenient size and mounting these inside the head shield covers of the recording and playback heads. With this arrangement, the closing of the head shield covers caused the felt pressure pads to bear against the back side of the tape as it passed over the heads, thus preventing variations in tape contact due to tape tension variations. Except for occasional transient fluctuations in signal amplitude which did not appreciably affect the average amplitude modulation readings, no reduction of modulation noise was noticed with these pressure pads in use.

The results of Fig. 8 show that the modulation noise owing to amplitude modulation, as measured by envelope detection with the intermodulation analyzer, agrees fairly well with the total measured modulation noise sidebands, as measured by the wave analyzer, except for frequencies very close to the carrier signal. Appendix A describes in detail the effects of flutter in producing side frequencies near the carrier signal frequency. At side frequencies beyond approximately 150 cps, the modulation noise is evidently due mainly to amplitude modulation sidebands, as indicated by the correlation with curve (B) of Fig. 8. The modulation noise on a dc or zero frequency signal has very nearly the same relative frequency distribution as does the modulation noise shown for a 3000 cps signal except that it is of correspondingly lower amplitude.

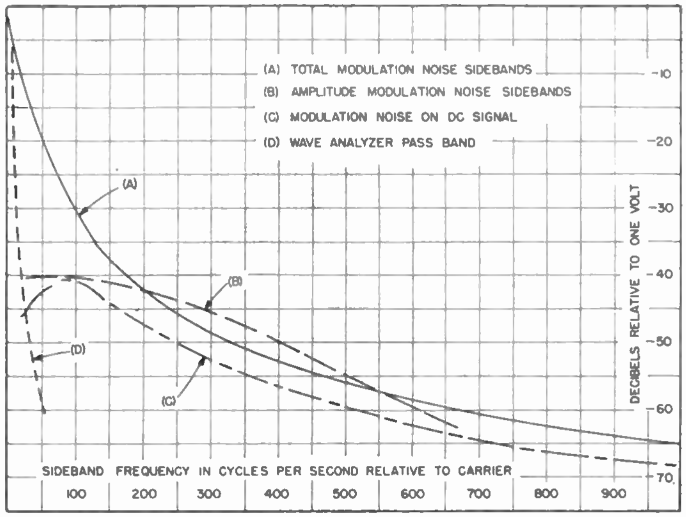


Fig. 8

Fig. 9 shows the indicated percentage of amplitude modulation as a function of recorded frequency for three grades of recording tape. These curves were drawn from data obtained with the test equipment setup of Fig. 2, with the exception that the readings taken were the indicated percentage amplitude modulation from the panel meter of the intermodulation analyzer. Measurements of the detected envelope of these sidebands with the wave analyzer showed that the relative sideband frequency distribution for all the samples of tape tested was approximately the same as shown in curve (B) of Fig. 8.

It is significant to note that with instrumentation quality tape having good uniformity, it is possible to obtain fairly low levels of amplitude modulation as compared with commercial grades of tape. Paper base tape, with its comparatively rough surface and uneven coating, has a very high level of modulation noise. Thus, it appears that this amplitude modulation is a function of the tape characteristics.

It has been found[8] that the attenuation of a signal, due to the spacing between the tape and the playback head, depends principally upon the ratio of the separation to the wavelength. Therefore, it follows that any surface irregularities that tend to interfere with the contact between the tape and the head cause corresponding amplitude variations of the reproduced signal.

In order to discover the nature of the variations to be found in commercial quality tapes, samples of several different tapes were tested for per cent amplitude modulation as a function of frequency. These tests are best summarized by the three curves in Fig. 10, which are representative of the variations which were found.

These curves are plotted on a constant frequency increment basis in order to bring out the fact that in all cases, the increase of amplitude modulation with re-

[8] Minnesota Mining and Manufacturing Co., St. Paul, Minn., "Effects of tape contact on frequency response," *Sound Talk Bulletin No. 6.*

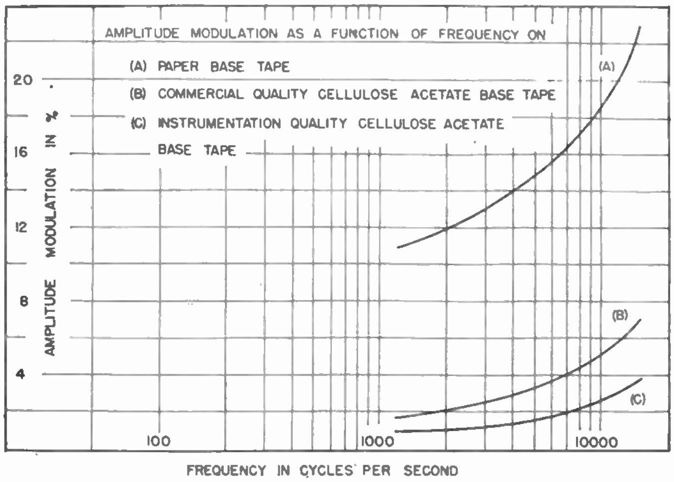


Fig. 9

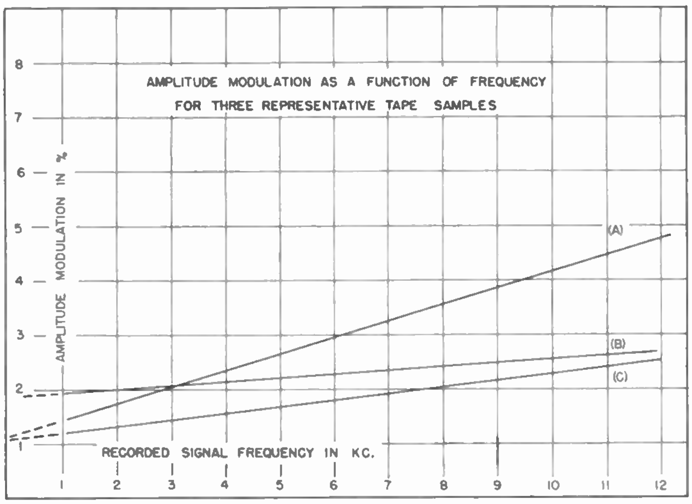


Fig. 10

corded frequency is practically a straight line. It is further noted that the modulation noise is not zero at zero frequency and that the slope of the lines differs for different tapes. Tape having low dc noise, such as sample (A), does not necessarily have low modulation noise at higher recorded frequencies. These facts would account for some of the disagreement which exists concerning modulation noise which is measured by various methods and at various frequencies.

From the experimental work reported in *Sound Talk Bulletin No. 6* it may be concluded that, since amplitude variation of reproduced signal is a function of the ratio between separation and wavelength, at zero frequency the attenuation due to spacing variations between head and tape should be negligible. The remaining causes of signal amplitude fluctuations, then, are evidently the nonuniform magnetic properties of the coating and nonuniform coating thickness. On this basis, it may be said that sample (A) in Fig. 10 has good uniformity of magnetic properties but with a rough surface which causes amplitude modulation at high signal frequencies. Sample (B) in Fig. 10 evidently has a nonuniform coating since the amplitude variations at low frequencies are rather high. The per cent amplitude modulation in this sample does not increase greatly with frequency, however, indicating that the coating surface is relatively smooth. Sample (C) is instrumentation quality tape which has good uniformity of both magnetic properties and coating surface.

Fig. 10 shows that amplitude modulation due to tape nonuniformities may be divided into the two classifications of frequency-dependent and nonfrequency-dependent amplitude modulation. If a curve of amplitude modulation as a function of signal frequency is plotted on a linear frequency scale as in the examples of Fig. 10, this curve will be very nearly a straight line and will intersect the zero frequency coordinate at the point corresponding to the percentage of nonfrequency-dependent amplitude modulation. The only test equipment needed for making this measurement is a variable frequency audio oscillator and an amplitude or envelope detector such as is commonly used to measure intermodulation in audio frequency equipment.

Tape nonuniformities which affect the spacing or contact between the tape and the heads cause an amplitude modulation of the signal which increases in percentage with signal frequency. This is true since lack of intimate contact between the tape and the playback head produces an attenuation of the reproduced signal which is inversely proportional to its wavelength on the tape. Possible causes of poor head tape contact are physical irregularities of the tape surface, improper tape tension, and dirt or dust particles which tend to lift the tape away from the head.

Tape nonuniformities which affect all recorded frequencies by an approximately equal amount are evidently due to nonuniform magnetic properties of the tape coating and improper tape guiding. Nonuniform magnetic properties may be caused by variations in the thickness of the magnetic coating, or to clumping or nonuniform distribution of the magnetic particles within the binder. Improper tape guiding may allow the edge of the tape to enter the head gaps, thus modulating the signal in accordance with the varying tape area caused by edge roughness.

The fourth suggested source of modulation noise, magnetostriction noise in magnetic playback heads, is said to be caused by mechanical excitation or vibration of the magnetic structure of the head. This vibration in turn causes a small signal or noise voltage to be produced across the signal winding of the head.

Tests were made for the presence of magnetostriction noise by passing the back or uncoated side of several bulk erased tape samples, including paper base tape, across a magnetized playback head at normal tape speed. No increase of noise due to this mechanical excitation of the head by the moving tape was noticeable above the playback amplifier noise. Therefore, it was concluded that the magnetostriction noise generated by the playback head was insignificant in the presence of the playback amplifier noise.

Another suggested source of modulation noise, signal transfer or "print through," was not encountered in the course of these tests. This was probably due to the fact that since a continuous single frequency test signal was used, the transferred signal would be of practically the same frequency as the original signal. Also, the tape was not stored for any appreciable length of time before playback. Time and storage conditions are important factors in "print through." It is felt that since this subject has been covered by L. J. Wiggin[9] there is no need for further investigation at this time.

The remaining noise mechanism to be investigated was the possible effect of Barkhausen phenomena in the magnetic structures of the recording and playback heads, and in the magnetic coating of the tape. Barkhausen noise is said to be encountered sometimes in very low level electromagnetic devices such as microphone transformers. This type of noise is believed to be due to the fact that magnetic materials are not perfectly homogeneous, but consist of domains. In a changing magnetic field these domains, acting as unit magnetic particles, give a step-like rather than a continuous change of field. The theory expressed by several of the authors listed in the bibliography is that irregularities due to these domains would give rise to modulation noise.

A test was made for the presence of Barkhausen, or domain noise, in the recording and playback heads by the following method. Two small coils of wire were placed near the gaps of the recording and playback heads respectively. Connecting the windings of these together resulted in the coupling by transformer action of a portion of the magnetic field from the recording head to the playback head. In this way, the recording and playback amplifiers and heads were operated normally except that no tape was in motion between the recording and playback heads, the link coupling replaced the tape and this coupling was adjusted to give normal playback signal level. Upon testing the signal from the playback amplifier output with the test equipment setups of Figs. 1 and 2, no measurable amount of modulation noise could be found above the playback amplifier noise level. Therefore, it was concluded that Barkhausen noise in the recording and playback heads is not a significant source of modulation noise.

No tests were made for the presence of Barkhausen noise in the magnetic coating of the tape since no method was found for separating irregularities due to domains from those due to variations in coating uniformity and thickness. Barkhausen noise results from a changing magnetic field acting on a magnetic material. The magnetic field of the tape moves with respect to the playback head but not with respect to the tape itself. The magnetic field which acts upon the tape during the recording process is high compared with those which are normally associated with Barkhausen effect. Since no Barkhausen noise was found in the heads and due to the fact that it is possible to reduce modulation noise to low values by the use of high quality tape, it is probable that Barkhausen noise is not an appreciable factor in tape coatings.

[9] L. J. Wiggin, "Magnetic print-through, its measurement and reduction," *J. SMPTE*, vol. 58, pp. 410–414; May, 1952.

Conclusions

The eight possible sources of modulation noise in magnetic tape recordings which were suggested by a study of the available literature have been investigated experimentally in an effort to discover which of these noise mechanisms actually contribute measurable amounts of modulation noise.

The results of this investigation indicate that sidebands due to modulation of the signal by the first three of these suggested sources account for practically all the total measured modulation noise present in the playback of test signals from a tape recording. The other five suggested sources of modulation noise were found to contribute little or no measurable effect on the total noise in the presence of the three major contributing causes. Although source 7), "print through," was not encountered in this investigation, the references indicate that it may be a problem under certain conditions.

In general, modulation may be classified either as time displacement (phase angle) modulation or as amplitude modulation. Time displacement modulation results from variations in or modulation of the velocity of the tape at the playback head relative to that at the recording head. Considering a recording and playback sequence as a time delay process, it follows that tape speed variations result in a variation of this time delay, generally known as "flutter" or "wow." High frequency "flutter" may be caused by longitudinal vibration of the tape, excited by friction, as it slides over the heads. Abnormal conditions, such as a sticking idler or the tape binding in the guides, may greatly increase this effect and cause it to become objectionable. Low frequency "flutter" and "wow" are commonly caused by speed fluctuations and eccentric rotation of the tape driving mechanism. Measurements made during this investigation have indicated that with a good quality tape driving mechanism, frequency modulation sidebands due to flutter may be maintained at a very low value. Typical flutter sideband amplitudes are calculated in Appendix A.

Amplitude modulation of a tape recorded signal has been found to be due primarily to nonuniformities in the physical and magnetic properties of the tape which is used as the recording medium. This amplitude modulation may be due to several causes which may be divided into the two classifications of nonfrequency-dependent and frequency-dependent variations.

Nonfrequency-dependent variations are those which are not greatly affected by signal frequency, even down to zero frequency or dc. They are evidently caused primarily by variations in magnetic properties of the tape coating such as nonuniformities in the density and dispersion of the magnetic particles. Major nonuniformities which result in loss of signal are known as "drop-outs." Improper tape guiding which allows the edge of the tape to ride within the head gap may result in amplitude modulation, corresponding to the varying tape area in the gap, caused by edge roughness.

Frequency-dependent variations increase approximately linearly with recorded signal frequency and are superimposed upon the nonfrequency-dependent variations. Frequency-dependent variations appear to be caused by nonuniformities which affect the spacing or contact between the tape and the heads. Probable causes of these variations include dust on the tape coating surface and roughness of this surface. Roughness of the coating surface may be due to roughness of the base material or to nodules caused by "clumping" of the particles within the binder.

A mathematical discussion of the effects of AM signal variations is given in Appendix B.

This investigation has brought out the fact that modulation noise is an important limitation on the signal-to-noise ratio which can be obtained under dynamic conditions in a magnetic tape recording. At the present time, measurements of modulation noise are not usually performed when evaluating the performance of magnetic tape-recording systems and no standard method of measuring modulation noise is in general use.

It is suggested that when making tests on magnetic tapes and recording systems, measurements of amplitude modulation noise be made by means of an amplitude sensitive detector, in addition to the usual measurements of flutter or frequency modulation noise by means of an FM discriminator or frequency sensitive detector. These measurements would detect both forms of modulation noise. From an analysis of these measurements, corrective measures could then be intelligently applied to minimize modulation noise.

Appendix A

Calculation of FM Sidebands

In a magnetic recording process, an electrical signal $e=f(t)$ is transformed by the recording process into a magnetic flux pattern on the tape. A sinusoidal signal would be transformed into a function of distance;

$$\phi = A \sin \frac{2\pi S}{\lambda} \qquad (1)$$

where:

ϕ = Flux density at any point
A = Peak flux density
S = Distance along the tape
λ = Recorded wavelength.

Let us assume that the tape is played back with sinusoidal flutter, where flutter is the difference in the velocity of the tape at the playback head relative to its velocity during recording.

$$v = V_0 + B \cos 2\pi FT \qquad (2)$$

where:

v = Instantaneous velocity
V_0 = Average velocity
B = Peak velocity error
F = Flutter frequency
T = Time.

$$S = \int v dt = \int (V_0 + B \cos 2\pi FT) dt$$

$$= V_0 T + \frac{B}{2\pi F} \sin 2\pi FT + \frac{B}{2\pi F} \qquad (3)$$

$$\phi = A \sin \left[\frac{2\pi V_0 T}{\lambda} + \frac{B}{\lambda F} \sin 2\pi FT + \frac{B}{\lambda F} \right]. \qquad (4)$$

Thus we have a frequency modulated signal of the form[10]

$$y = A \sin (\omega_0 t + m_f \sin \omega_m t + C) \qquad (5)$$

where:

$$m_f = \frac{B}{\lambda F}, \qquad \omega_0 = \frac{2\pi V_0}{\lambda}, \qquad \omega_m = 2\pi F$$

and

$$C = \frac{B}{\lambda F}.$$

Expanding this expression in terms of a Bessel series gives:

$$y = AJ_0(x) \sin \omega_0 t \quad \text{Recorded signal (carrier)} \qquad (6)$$
$$\left.\begin{array}{l} +AJ_1(x)[\sin(\omega_0+\omega_m)t \\ -AJ_1(x)[\sin(\omega_0-\omega_m)t] \end{array}\right\} \text{First order sidebands}$$
$$\left.\begin{array}{l} +AJ_2(x)[\sin(\omega_0+2\omega_m)t] \\ +AJ_2(x)[\sin(\omega_0-2\omega_m)t] \end{array}\right\} \text{Second order sidebands}$$
$$\left.\begin{array}{l} +AJ_3(x)[\sin(\omega_0+3\omega_m)t] \\ -AJ_3(x)[\sin(\omega_0-3\omega_m)t] \end{array}\right\} \text{Third order sidebands}$$
$$+ \cdots\cdots\cdots$$

Thus the amplitudes of the carrier and side frequencies are determined by the appropriate Bessel coefficients. From these relationships the magnitudes of the sidebands due to a given per cent flutter may now be calculated.

With capstan rate (20 cps) flutter of 0.067 per cent on a 3000 cps signal,

$$x = m_f = \frac{0.00067(3000)}{20} = \frac{2}{20} = 0.1.$$

Substituting $x=0.1$ into (6), it is found that the first order sidebands are 0.05 or -26 db and the second

[10] A. Hund, "Frequency Modulation," McGraw-Hill Book Co., New York, N. Y., 1st ed., pp. 16–17; 1942.

order sidebands are 0.0012 or −58 db below the fundamental signal amplitude. Similar figures result from the calculation of sideband amplitudes due to other flutter frequencies.

Appendix B

Calculation of AM Sidebands

If we assume that the tape is played back without flutter, (4) in Appendix A becomes

$$\phi = A \sin \frac{2\pi V_0 t}{\lambda} \tag{7}$$

or

$$\phi = A \sin \omega t \tag{8}$$

where

$$\omega = \frac{2\pi V_0}{\lambda}.$$

If the magnetic coating on the tape is not perfectly uniform, the peak recorded flux sensity, A, which would result from a given recording current will be modulated by the relation

$$A = A_0 + af(t)$$
$$= A_0[1 + m_a f(t)] \tag{9}$$

where:

A_0 = The average value of the peak flux density
a = The maximum amplitude of the modulating function
$f(t)$ = A function of time representing the noise distribution; $0 \leqq f(t) \leqq 1$
$m_a = \frac{a}{A_0}$ = Degree of amplitude modulation.

Substituting (9) into (8)

$$\phi = A_0[1 + m_a f(t)] \sin \omega t \tag{10}$$

If $f(t)$ is represented by a sum of sinusoidal components, as measured by a selective wave analyzer

$$f(t) = \sum_{k=1}^{k=N} a_k \sin \rho_k t \tag{11}$$

$$\phi = A_0 = \left[1 + \frac{1}{A_0} \sum_{k=1}^{k=N} a_k \sin \rho_k t \sin \omega t \right. \tag{12}$$

each frequency component ρ_k gives rise to a pair of sidebands $(\omega+\rho_k)$ and $(\omega-\rho_k)$ symmetrically located about the carrier frequency ω.

$$\phi = A_0 \sin \omega t \quad \text{Carrier} \tag{13}$$
$$- \frac{a_1}{2} \cos (\omega + \rho_1)t \quad \text{Upper sideband}$$
$$+ \frac{a_1}{2} \cos (\omega - \rho_1)t \quad \text{Lower sideband}$$
$$+ \cdots\cdots\cdots$$
$$- \frac{a_n}{2} \cos (\omega + \rho_n)t \quad \text{Upper sideband}$$
$$+ \frac{a_n}{2} \cos (\omega - \rho_n)t \quad \text{Lower sideband}$$
$$+ \cdots\cdots\cdots .$$

A variation of 2 per cent in the thickness of a magnetic coating having a nominal thickness of 0.0005 inches would be a variation of 0.02×0.0005 or 10^{-5} inches. Two per cent signal amplitude variation would result in sidebands of

$$\frac{a}{2} = \frac{0.02}{2}$$

or 40 db below the recorded signal level.

Acknowledgment

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Transistor Nonlinearity—Dependence on Emitter Bias Current in P-N-P Alloy Junction Transistors*

D. R. FEWER†

Summary—**A method of calculating the nonlinear behavior of class A common emitter transistor amplifiers from linear small signal measurements is given. Experimental results, obtained with a 500-milliwatt *p-n-p* alloy junction transistor, show the second and third harmonic distortion as a function of emitter bias current and driving source resistance. Distortion, calculated from small signal measurements, is shown for comparison.**

Introduction

In the early stages of the experimental measurement of transistor linearity it was apparent that the linearity of class A common emitter amplifiers was highly dependent on the bias current. This paper gives an outline of a method which predicts, at least qualitatively, the variation of distortion (second and third order) with emitter bias current. The special conditions listed below apply:

1) Common emitter class A amplifier;
2) Negligibly small load compared to the output resistance of the transistor;
3) Low frequency case;
4) Device $a \doteq 1$;
5) Considerably more linear relationship between a and I_e than between $(1-a)^{-1}$ and I_e;
6) Purely resistive driving source.

The results are based on the analysis given by W. M. Webster[1] and others.[2,3,4] The Webster paper with Fletcher's addition[2] gives an analysis of the variation of current amplification in alloy junction transistors with emitter current. A brief discussion of the analysis is given below.

Theory

As emitter current in an alloy junction transistor is increased the current transfer is observed to increase at low I_e, reach a maximum, then decrease at higher values of I_e. This nonlinear current transfer is due to the fact that the properties of the base region change when carriers are injected. Previous theory assumed that the injected carriers produced a charge density which was small compared with the density of the ionized impurity atoms in the base. This is not true at currents higher than a few milliamperes. The injected charge may exceed the ionic charge by 10 to 15 times.

The effect of injecting carriers in the base is to produce changes in

1) Surface recombination;
2) Base conductivity;
3) Volume recombination.

An electric field is also set up in the base region which produces drift in addition to the normal diffusion.

Field Effect and Surface Recombination

In the original theory the electric field developed by the injection of the charge carriers into the base region was neglected. However, in order to pass a certain hole current (assume *p-n-p* transistor), a hole density gradient is required. The condition of space charge neutrality requires an equal electron density.

Both holes and electrons will tend to diffuse in the same direction. The electrons will move until a field is set up to hold them in place against their density gradient. The same field acts in a direction to encourage hole flow and, in the limit, doubles the hole current density for a given density gradient. This, in effect, halves the surface recombination term at high currents (in the limit).

Modulation of Base Region Conductivity by Injected Carriers

When current density is high the emitter efficiency (γ) decreases and the current amplification drops with further increase in current. At high currents, the current amplification factor should vary inversely with the emitter current.

Volume Recombination

The volume recombination term shows the same dependence on emitter current as the emitter efficiency term.

The approximate equation for current gain as a function of emitter current can be written:[5]

[5] A list of symbols is given on page 44.

* Manuscript received by the PGA, October 21, 1957. This work was performed while the author was a member of the technical staff of Bell Telephone Labs., Inc., Murray Hill, N. J.

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[1] W. M. Webster, "On the variation of junction transistor current amplification factor with emitter current," Proc. IRE, vol. 42, p. 914; June, 1954.

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$$1 - a \doteq \frac{sWA_s}{D_pA} g(Z) + \frac{\sigma_b W}{\sigma_e L_e}[1 + 2Zh(Z)]g(Z)$$

Surface + Emitter
Term Term

$$+ \frac{1}{2}\left(\frac{W}{L_b}\right)^2 [1 + 2Zh(Z)]g(Z)$$

+ Volume
Term

where

$$Z = \frac{Wu_e I_e}{D_p A \sigma_b}$$

$$g(Z) = \frac{1 + p_e/N_d}{1 + 2p_e/N_d}$$

$$h(Z) = \frac{p_e}{ZN_d}.$$

At values of current where $Z > 10$ the functions $g(Z)$ and $h(Z)$ each approach the value 0.5. We may write the above equation

$$1 - a \doteq k_1 + k_2 I_e$$

where

$$k_1 = \frac{sWA_s}{2D_pA} + \frac{\sigma_b W}{2\sigma_e L_e} + \frac{1}{4}\left(\frac{W}{L_b}\right)^2$$

$$k_2 = \frac{1}{2}\left[\frac{\sigma_b W}{\sigma_e L_e} + \frac{1}{2}\left(\frac{W}{L_b}\right)^2\right]\frac{Wu_e}{D_pA\sigma_b}.$$

The values of k_1 and k_2 may be determined from the physical properties of the semiconductor material or by experimental techniques.

In order to obtain these constants experimentally a curve of $(1-a)^{-1}$ vs I_e is plotted as shown in Fig. 1.

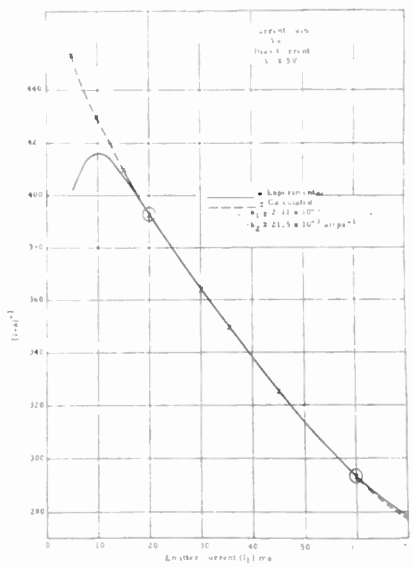


Fig. 1

Two points at the higher values of I_e are selected and the constants k_1 and k_2 are evaluated by simultaneous solution.

The equation for $(1-a)$ is substituted in the circuit equations. Consider the circuit in Fig. 2.

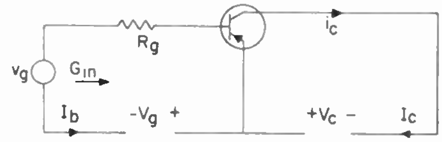


Fig. 2

We describe i_c by a power series in v_g

$$i_c = A_1 v_g + A_2 v_g^2 + A_3 v_g^3 + \cdots$$

where

$$A_1 = G_{in} h_{21}$$

$$2A_2 = G_{in}^2 h_{21}' + h_{21} G_{in}'$$

$$6A_3 = G_{in}^3 h_{21}'' + 3G_{in}G_{in}'h_{21}' + h_{21}G_{in}''$$

$$h_{21} = \frac{\partial I_c}{\partial I_b}, \quad h_{21}' = \frac{\partial^2 I_c}{\partial I_b^2}, \quad h_{21}'' = \frac{\partial^3 I_c}{\partial I_b^3}$$

$$G_{in} = \frac{\partial I_b}{\partial V_g}, \quad G_{in}' = \frac{\partial^2 I_b}{\partial V_g^2}, \quad G_{in}'' = \frac{\partial^3 I_b}{\partial V_g^3}.$$

Substituting for $(1-a)$ in the expressions for G_{in} and h_{21} and assuming $a \doteq 1$ and the derivatives of a are small with respect to the derivatives of $(1-a)^{-1}$ we may write:

$$G_{in} \doteq \frac{1}{R_g + r_b + \dfrac{r_e}{1-a}}$$

$$h_{21} \doteq \frac{1}{1-a}.$$

Let

$$R_g + r_b = R$$

and

$$r_e = \frac{kT}{qI_e}$$

$$G_{in} = \frac{\partial I_b}{\partial V_g} \doteq \frac{k_1 + k_2 I_e}{R(k_1 + k_2 I_e) + \dfrac{kT}{qI_e}}$$

$$h_{21} = \frac{\partial I_c}{\partial I_b} \doteq \frac{\partial I_e}{\partial I_b} \doteq \frac{1}{k_1 + k_2 I_e}$$

$$G_{in}' \doteq \frac{\frac{kT}{qI_e^2}[k_1 + 2k_2I_e]}{\left[Rk_1 + Rk_2I_e + \frac{kT}{qI_e}\right]^3}$$

$$G_{in}' \doteq \frac{-\frac{2kT}{qI_e^3}\left(Rk_1 + Rk_2I_e + \frac{kT}{qI_e}\right) + 3\left[(k_2R)^2 - \left(\frac{kT}{qI_e}\right)^2\right]}{\left[Rk_1 + Rk_2I_e + \frac{kT}{qI_e}\right]^5}$$

$$h_{21}' \doteq \frac{-k_2}{(k_1 + k_2I_e)^3}$$

$$h_{21}'' \doteq \frac{3k_2^2}{(k_1 + k_2I_e)^5}.$$

If the driving source voltage is $v_g = V\cos wt$, then we may write the distortion as a function of I_e

$$\frac{i_{2F}}{i_F} \doteq \frac{A_2V}{2A_1} \qquad \frac{i_{3F}}{i_F} \doteq \frac{A_3V^2}{4A_1}$$

$$\frac{i_{2F}}{i_F} \doteq \frac{V\left[\frac{kT}{qI_e^2} - k_2R\right]}{4\left(Rk_1 + Rk_2I_e + \frac{kT}{qI_e}\right)^2}$$

$$\frac{i_{3F}}{i_F} = \frac{V^2\left\{-\frac{2kT}{qI_e^3}\left(Rk_1 + Rk_2I_e + \frac{kT}{qI_e}\right) + 3\left[(Rk_2)^2 - \left(\frac{kT}{qI_e^2}\right)^2\right]\right\}}{24\left[Rk_1 + Rk_2I_e + \frac{kT}{qI_e}\right]^4}.$$

Experimental Results

Curves of

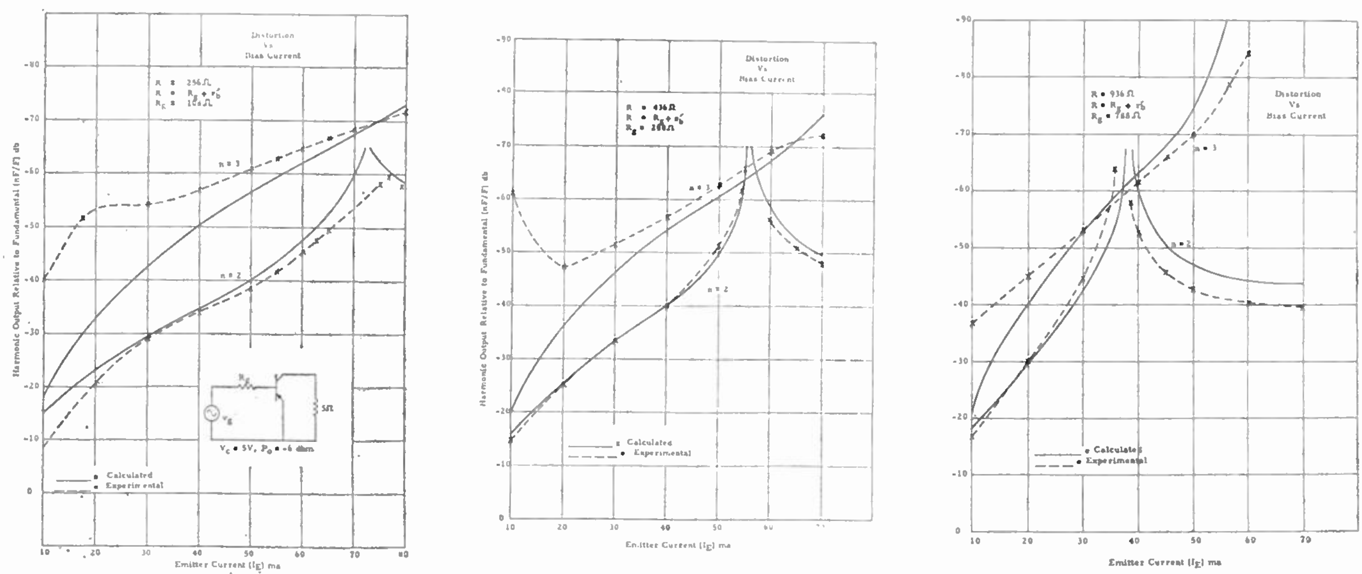
$$\frac{nF}{F} = 20\log_{10}\left|\frac{i_{nF}}{i_F}\right| \text{ versus } I_e$$

for a constant fundamental output obtained from these equations for three values of R are shown in Figs. 3(a), 3(b) and 3(c) below.

A good agreement with measured results is obtained for emitter bias currents in excess of 20 ma. Some of the difference between measured and theoretical results is due to output distortion. This is caused by the use of a measurement technique which does not allow a perfect short circuit at the output.

It is noted that the second harmonic vanishes when

$$R = \frac{kT}{k_2qI_e^2}.$$



(a) (b) (c)

Fig. 3

For a fixed R a value of emitter bias current (I_e) can be found which minimizes second order distortion. This effect is illustrated by the discontinuities in the $n=2$ curves of Figs. 3(a)–3(c).

When the second order term vanishes it can be shown that the third order term may be written

$$\frac{i_{3F}}{i_F} = \frac{-V^2}{12}\left(\frac{q}{kT}\right)^2 \frac{k_2 I_e}{k_1 + 2k_2 I_e}.$$

Conclusions

The results show that, for certain types of alloy junction transistors, the nonlinear behavior can be predicted from linear small signal measurements.

List of Symbols

s = Surface recombination velocity
W = Width of base region
A_s = Effective surface area for recombination
D_p = Diffusion coefficient for holes in the base region
A = Cross sectional area of conduction path
σ_b = Base region conductivity
σ_e = Emitter region conductivity
L_e = Diffusion length for electrons in the emitter region
L_b = Diffusion length for holes in the base region
μ_e = Electron mobility
p_e = Hole density at the emitter junction in the base region
N_d = Donor ion density in the base region.

Contributors

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In 1946 he joined Columbia Records, Inc., as director of engineering and research, and in this capacity contributed to the development and production engineering of LP records, the heated stylus disk recording technique, viscous damped reproducer arms, and program-controlled automatic groove spacing equipment. He has made numerous inventions and presented many papers on the subject of disk recording and reproduction.

Mr. Bachman has been active on numerous industry committees and is chairman of the Technical Committee of the Record Industry Association of America. He was a recent recipient of the Audio Engineering Society Berliner Award for outstanding developments in disk recording and reproduction.

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❖

Benjamin B. Bauer (S'37–A'39–SM'44–F'53) was born in Odessa, Russia on June 26, 1913. He graduated from Pratt Institute, Brooklyn, N. Y. in 1932 and received the E.E. degree at the University of Cincinnati in 1937. He joined Shure Brothers, Inc., manufacturers of electroacoustical devices, in Chicago as a co-op student in 1936, and became chief engineer in 1940 and Vice-President in 1950.

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In 1957 Mr. Bauer joined the CBS Laboratories, and was appointed a Vice-President in 1958. He has charge of audio, acoustics, and magnetics research.

Mr. Bauer has been responsible for the development of a great many electroacoustical devices such as microphones, phonograph pickups, and tape recording heads. He has over thirty patented inventions in the fields of audio and acoustics, and has published many papers in these fields.

He is a Fellow of the Acoustical Society of America and the Audio Engineering Society. He is also Associate Editor of the *Journal of the Acoustical Society of America.* In 1955 he received the Achievement Award from the IRE Professional Group on Audio, of which he was co-founder, past National Chairman, and present Secretary. He is a member of Eta Kappa Nu, Tau Beta Pi, and Sigma Xi, and has received the Eta Kappa Nu's Recognition Award.

❖

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D. R. Fewer

From 1952 to 1957 he worked on transistor circuit development at Bell Telephone Laboratories, and is now doing circuit research work at Texas Instruments, Inc., Dallas. He was co-winner of the W. R. G. Baker award in 1957.

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Peter C. Goldmark (A'36–M'38–F'43) was born in Budapest, Hungary, on December 2, 1906 and studied at the Universities of Berlin and Vienna, receiving the Ph.D. degree in physics from the latter university in 1931.

P. C. Goldmark

From 1931–1933 he was engineer in charge of television development for Pye Radio Ltd., England. In 1933 Dr. Goldmark left England for the United States and in 1936 joined the Columbia Broadcasting System as chief television engineer, later becoming director of the Research and Development Division.

The first practical color television system was developed under the direction of Dr. Goldmark in the CBS Laboratories, and on August 27, 1940, the first color telecast in history was made from the CBS television transmitter in New York.

During the war, CBS Laboratories, under Dr. Goldmark, were responsible for many military developments in the field of electronic countermeasures and reconnaissance. After the war the long-playing record was developed by Dr. Goldmark and his associates in the CBS Laboratories. Later the development of the first high-fidelity compact phonograph, the Columbia 360, followed.

In 1954 Dr. Goldmark became President of CBS Laboratories and Vice-President of CBS, Inc.

Dr. Goldmark is a Fellow of the American Institute of Electrical Engineers, the Society of Motion Picture and Television Engineers, and the British Television Society. In 1945 he was awarded a medal by the Television Broadcasters Association for his color television pioneering work, and in 1946 he received the Morris Liebmann Memorial Prize for electronic research from the IRE. He is also a visiting Professor for Medical Electronics at the University of Pennsylvania Medical School.

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R. Lee Price (A'47–M'48–SM'53) was born in Knoxville, Iowa, on April 9, 1919. He attended evening schools in the Chicago area and received the B.S. degree in electrical engineering from the University of Illinois, Urbana, in 1950. He then continued his part-time studies and received the M.S. degree in electrical engineering from Northwestern University, Evanston, Ill., in 1954.

R. L. Price

From 1941 to 1951 Mr. Price was associated with the engineering and research department of Zenith Radio Corporation. From 1951 to 1954 he was senior electronics engineer with Magnecord, Inc. From 1954 to 1957 he was employed by Donner Scientific Company as head of test instrument engineering. Mr. Price is now with the TelAutograph Corporation of Los Angeles. He has worked primarily in the fields of audio and video equipment development, electroacoustic devices, and magnetic recording.

He is a member of Eta Kappa Nu, associate member of AIEE, and is a registered professional engineer.

INSTITUTIONAL LISTINGS

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

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