Signal-To-Noise Problems and A New Equalization for Magnetic Recording of Music*

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Measuring procedures and specifications currently used in the sound recording industry are inadequate. A weighting network should be used in objective noise measurement work to evaluate developments toward a lower noise system.

We know that the ear is most sensitive in the 1 to 6 kc region, and it is possible to use pre-emphasis in this region. Better signal-to-noise ratio may be obtained in 15 ips magnetic recorders by utilizing fully their present capabilities. Subjective listening tests show that a system with the 1 to 6 kc region pre-emphasized (the AME curve) is some 7 db quieter than the same system using the NAB curve, but does not show any audible increase in distortion.

INTRODUCTION

ALTHOUGH the published signal-to-noise ratio figures for tape recording systems look impressive, a critical listening to wide dynamic range orchestral music recorded on tape reveals that tape noise is far from inaudible. Modern commercial pressings (disc recordings) are quiet enough that, with few exceptions, the tape noise can be heard above the disc noise, since disc noise is of a different character, and since the dynamic range is often limited after tape recording, before transferring to disc.

Problems Beyond Our Direct Control

The signal-to-noise problem is aggravated considerably by two situations which are entirely beyond our control. First, both studios and "Hi-Fi" fans tend to reproduce music at a volume greater than that of the original source. This, of course, also increases the audible noise level.

Second, two deficiencies of loudspeakers are apparent. No loudspeakers are truly flat. The average monitor speaker used in recording studios has a considerable deficiency in high frequency response. In an attempt to make the sound from the loudspeaker "better than being there," the producer or the recording engineer often increases the high-frequency energy electrically with an equalizer, and at least a part of this equalization is intended to compensate for deficiencies in the monitoring loudspeaker. Therefore, more high-frequency energy is present on recordings than there would be if the original monitoring could be done with a truly flat loudspeaker system. This additional high-frequency energy increases the problems that exist from high-frequency overloading at 7½ and 3¾ ips.

Another deficiency of loudspeakers involves their directional pattern. Since all speakers are more or less directive at higher frequencies, if the average sound energy in the room at high frequencies is to be kept the same as the energy at lower frequencies, the high-frequency energy directly on the axis of the speaker is higher than that in the middle frequencies, and the energy off axis is lower. If one stands on axis, as many people do, the high-frequency response (and therefore the audible noise level) is increased. Also, the noise, which is continuously present, coming from a small area seems to draw more attention to itself than if the source were a larger area.

The problems of response and directionality of the monitoring loudspeaker have been recognized by the German broadcasting companies.[1] They have established a standard acoustic response for the monitor loudspeaker *as installed in the control booth.* The speaker which they are using is said to have a spherical radiation pattern.

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[1] F. Enkel, "Experience With a New High Quality Loudspeaker for Control Booths", *Gravesano Review* 9: 107-110 (1957).

Improving the Signal-to-Noise Ratio of the Magnetic Recorder

We may look to improve the signal-to-noise ratio of the magnetic recorder in the following areas:

1. The Recording System.
   a) heads
   b) bias
   c) amplifiers
   d) type of recording (direct, FM, pulse coding, etc.)
   e) variable gain systems.
2. Tape Oxide.
3. Utilization of Available System and Tape.
   a) measurement problems
   b) equalization.

THE RECORDING SYSTEM

A study of tape equipment shows that, with the presently used direct record system, no significant improvements are to be made in the construction of heads (cores, windings or gaps) or the bias supply. The signal-to-noise ratio is not limited by amplifiers.[2]

Alternative types of recording have not been investigated.

Increased signal-to-noise ratio may be obtained through the use of variable gain systems. These systems will not be considered in this paper, as they have been extensively reported elsewhere.[3]

THE TAPE OXIDE

The various tapes manufactured in the United States have different outputs, but the signal-to-noise ratios are practically constant from one tape to another. A theoretical study of the noise characteristics of tape shows that an improvement should be possible by decreasing the particle size in the oxide.[4] Thus far, no tape is available which is interchangeable with present tapes, but quieter.

UTILIZATION OF THE AVAILABLE SYSTEM AND TAPE

Any further improvement must be made by improving the utilization of the tape with respect to the characteristics of the tape, the music, and the ear.

Signal-to-Noise Measurements

We must first see if present objective measurements are really indicative of the subjective signal-to-noise ratio of an audio recorder.[5] We must think of "signal" as the *maximum signal which can be recorded without audible distortion,* and "noise" as the *audible noise.*

Consider how we presently measure the maximum signal and the noise in an audio recorder. The measurements have been chosen so as to be easily made; little account is taken of the characteristics of the tape, the music, or the ear. (Although this paper discusses magnetic recording, most of the problems are common to the sound recording industry.)

Signal Measurement

Specifications are based on a sine wave reference signal level of 1 or 3 percent total harmonic distortion at 400 cps, but in tape recordings of music, distortion due to generated harmonics is not usually audible. What we hear when over-recording occurs are compression caused by the tape and intermodulation. The compression may be apparent in either of two ways:

First, the over-recorded signal will itself be compressed, which is to say that an increasing input level will not produce a proportionately increasing output level. (Figure 1 shows the tape output *vs* input for a 400 cps signal at 15 ips.)

Second, at high frequencies, when a signal is over-recorded, not only will the increase of input level *not* be accompanied by an increase in output level, but the output level will actually decrease with increase in input level. (Figure 2 shows the tape output *vs* input for a 12 kc signal at 15 ips with NAB equalization.) This has a secondary effect: The over-recorded high frequencies cause a reduction of output at all frequencies due to effective over-biasing by the high frequencies themselves. (Figure 3 shows the tape output of the 150 cps component of a recording of constant level 150 cps input plus variable level 12 kc *vs* the 12 kc input level, again for NAB equalization.)

Maximum program signal is measured with the VU meter; however, it is known the the VU meter takes no account of either the total peak energy (see Figure 4 for a histogram of peak factors *vs* number of occurrences, and the distribution of peak factors, from original data) or the peak energy *vs* frequency in the recorded material (see Figures 8 and 9 below), the overloading (compression) characteristics of the tape *vs* frequency (see Figures 1, 2 and 3 above), the equalization of the machine, or the audible effect of the various types of overload. Because of this, we find some studios that need the maximum signal on the tape ignoring the readings of the VU meter and simply listening to the playback of the tape, increasing the recording level until audible distortion occurs, and then slightly reducing the

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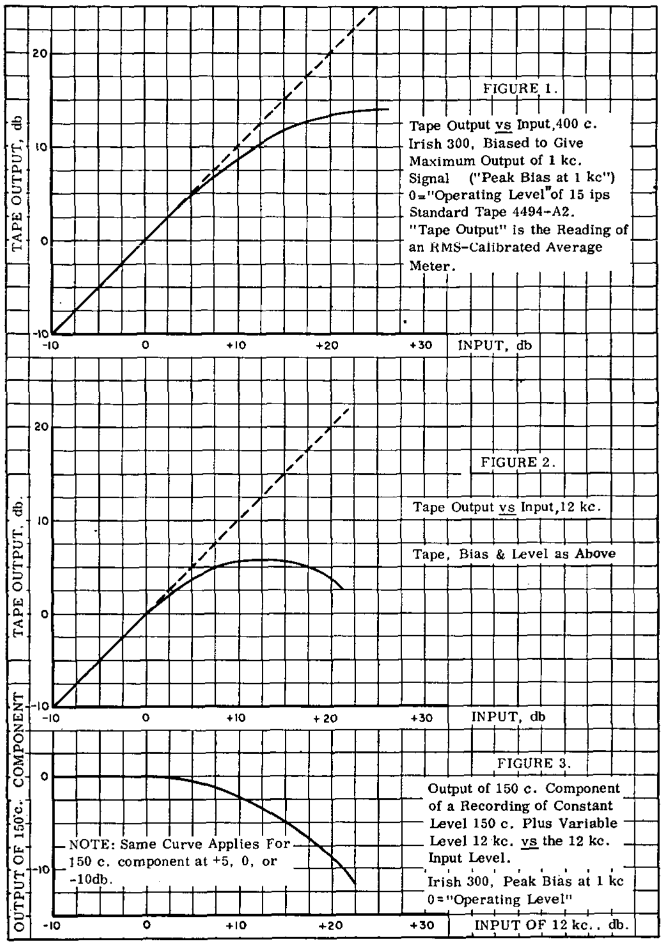
[2] The weighted noise from a properly designed playback amplifier is 10 to 15 db below the tape noise.

[3] Variable gain systems are discussed in the following references:
J. Havstad, *Information Utilization in Magnetic Recording,* Ampex Research Report No. 105 (October 1957).
R. Vermeulen and W. Westmijze, "The 'Expressor' System for Transmission of Music", *Philips Tech. Rev.* **11**: 281-290 (1950).
H. Fletcher, "The Stereophonic Sound Film System, General Theory", *Jour. SMPTE* 37: 331-352 (1941).
W. B. Snow and A. R. Soffel, "Electrical Equipment for the Stereophonic Sound Film System", *Jour. SMPTE* 37: 380-396 (1941).
H. Levinson and L. T. Goldsmith, "Vitasound", *Jour. SMPTE* 37: 147-153 (1941).
W. E. Garity and J. N. A. Hawkins, "Fantasound", *Jour. SMPTE* 37: 127-146 (1941).

[4] D. H. Howling, "Noise in Magnetic Recording Tapes", *J. Acoust. Soc. Amer.* **28**: 977-987 (1956).

[5] A discussion of these problems is given by Kellogg, "Proposed Standards for Measurement of Distortion in Sound Recording," *Jour. SMPTE* 51: 449-467 (1948).



FIGS. 1, 2, 3.

recording level, thus determining maximum level. This is an effective method, but very undesirable since there is no metering device that is accurately related to the maximum recorded level on tape.

A study of the audible effects of these compression phenomena is needed to determine quantitatively which effects are audible and at what level and frequency.[6] Using these data, we hope to be able to design a peak-reading volume-indicating meter which will have a true correlation between its reading and the audible overload.

Terman and Pettit[7] give an excellent discussion of non-linear distortion in audio frequency systems. They conclude: "The ultimate evaluation of the practical significance of non-linear distortion in audio systems must be based on listening tests. This causes the situation to be very complex, as the results of listening tests depend not only on the non-linear characteristics of the amplifier, but also upon the character of the sound being reproduced, upon the acoustics of the space in which the sound is observed, and on psychological factors which differ from person to person. The result is that there is no entirely satisfactory method of objectively defining the non-linear distortion introduced by an audio amplifier, and several different test methods are in common use."

We should also ask what level should be taken as the signal level in making signal-to-noise measurements. One can argue against any level chosen: one should not use the steady-state VU meter zero, since the program peaks always exceed zero; nor can a constant be fairly added, since the peak factors differ for each type of music; nor should the overload point of the system be used, since this varies with the type of system (tape at various speeds, or disc, etc.) and with the equalization used. No known single number tells us what the audible overload or maximum level of a system is; this is a complicated quantity which can be judged only by subjective listening tests.

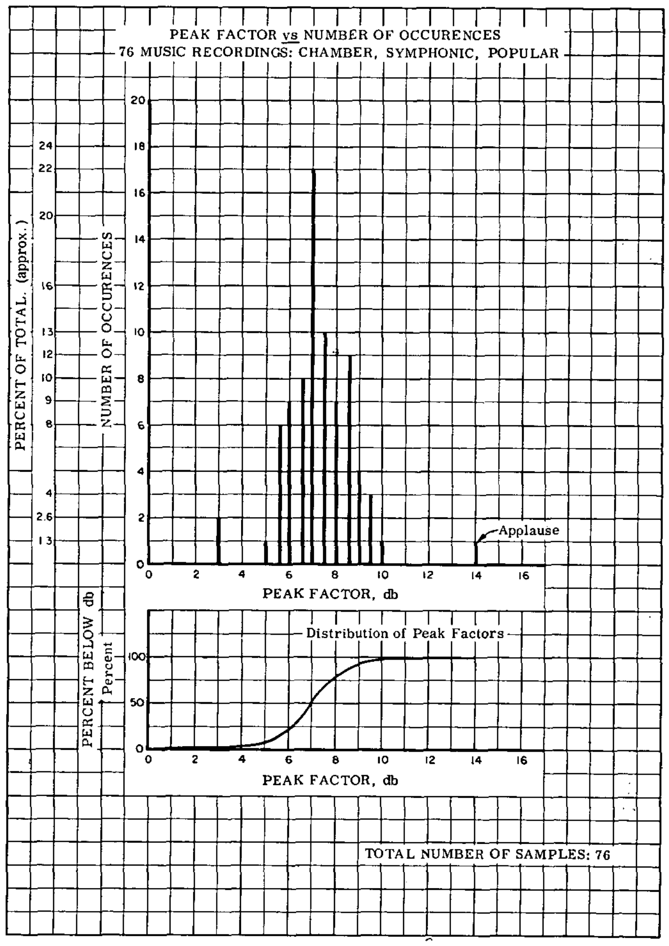


FIG. 4.

[6] Audibility of distortion is largely controlled by "masking"—the phenomenon of the ear in which presence of a lower frequency may make a higher frequency inaudible. Some studies have been done; further work is needed. See Peterson, "Intermodulation Distortion: Its Measurement & Evaluation," *IRE Convention Record,* 5 #7 (Audio): 51-56 (1957).

[7] Terman & Pettit, *Electronic Measurements,* (McGraw-Hill Book Company, Inc., New York, 1952), 2nd Ed., pp. 333-341.

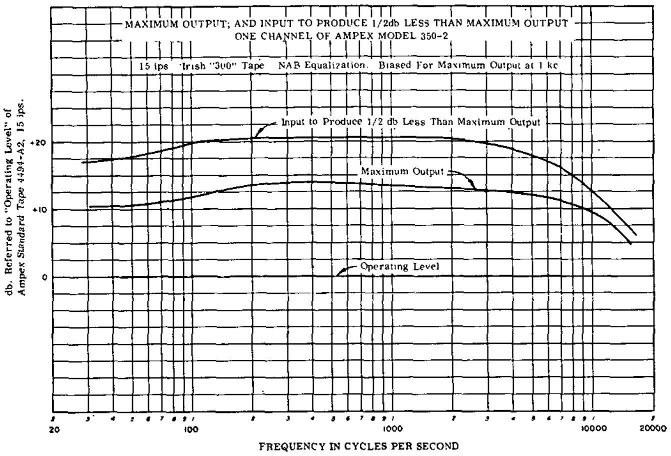


FIG. 5.

A useful piece of data in evaluating maximum program output is the maximum sine wave output of a tape recorder *vs* frequency, referred to the operating (steady-state VU meter zero) level. In addition, one should know what input signal is applied to give, say, ½ db less than maximum output.[8] The input levels should certainly never exceed that which produces maximum (saturation) output. Figure 5 shows, for example, the maximum sine wave output, and input to produce ½ db less than maximum output, for one channel of an Ampex Model 350-2, at 15 ips with NAB equalization.

One might choose to take as the lesser evil the operating (zero) level as the signal level, since this is at least a point which can be found without question or equivocation. Then one may present the data for maximum output, and the input level to produce this maximum output, to judge whether the system has adequate overload capability above this zero level. In this case, one will specify, rather than signal-to-noise ratio, a figure for noise, so many db below operating (zero) level.

The chief drawback of this type of specification is that it is common practice in film (but not disc) recording to refer the signal level to a maximum output level arbitrarily taken as 10 db above the operating level. Therefore, ratios from the signal level will give "signal-to-noise" data comparable to that for disc recording, but 10 db worse than it should be when compared to a film system using 10 db above operating level as the zero reference. This is not a problem as long as it is recognized.

Noise Measurement

Consider also the present measurement of noise. A typical specification for noise level on a tape machine is that all the frequencies between 50 and 15,000 cps are measured.

There is no relation between the ear characteristic and this common method of measuring noise. Curve 1 of Figure 6 shows system spectral noise density (in energy per cycle)[9] for one track of a Model 350-2 Ampex stereo recorder reproducing biased Irish 300 Shamrock tape at 15 ips. The energy per cycle is constant above 1,000 cps. Curve 2 shows the equipment noise (tape stopped) for the same machine.

It is impractical to try to show a comparison between system noise and hearing limits, since the level of the system noise depends on the dynamic range of the recording and on how loudly the recording is played back. The hearing limits depend on the individuals and on the noise level in the room in which the playback is made. Also, in general, the tape noise is sufficiently loud that it is not a question of whether it is audible at all, but rather how loud the various components of the tape noise sound. Therefore, we have chosen to compare the spectral noise density data for the system to an equal loudness contour of the 40 phon level (see Curve 3 of Figure 6). (If the noise is 60 to 70 db below tape recording level and the maximum sound pressure level is 100 to 110 db, then the noise level will be approximately 40 db; therefore the 40 phon level is used for comparison.) It is apparent that for the 40 phon level the ear is most sensitive to noise in the 1 to 6 kc region and that noise below 100 cps must be very great before it is audible. If we compare the system noise and equipment noise on a magnetic recorder with the equal loudness contour, we see why we hear the noise from a tape recorder as "hiss"; and why, although the audible noise increases quite noticeably as the tape is started, a flat meter will change only slightly, since it is largely reading the low frequency components of noise, which are inaudible to the ear. It is necessary in making noise measurements that are to be significant in relation to the subjective measurements of noise, to use a weighting network which has a response that is the inverse of ear response.

Weighting networks have been used for many years in the measurement of acoustic sound levels and noise measure-

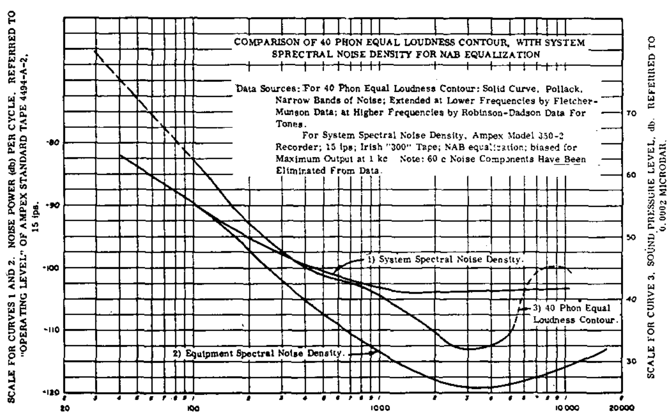


FIG. 6.

[8] See Figs. 1 and 2. The input level for "maximum output" is not a unique value, but input level for "just barely maximum" (say ½ db less than maximum) is unique. This paper does not take into account wave forms at overload; this is material for a separate study. Data here are readings from an rms calibrated average meter.

[9] See s 2.2, "Spectral Noise Density: Standards on Sound Recording and Reproducing: Methods of Measuring Noise", *Proc. IRE* **41**: 508-512, (April 1953) (Standard 53-IRE 19-S-1).

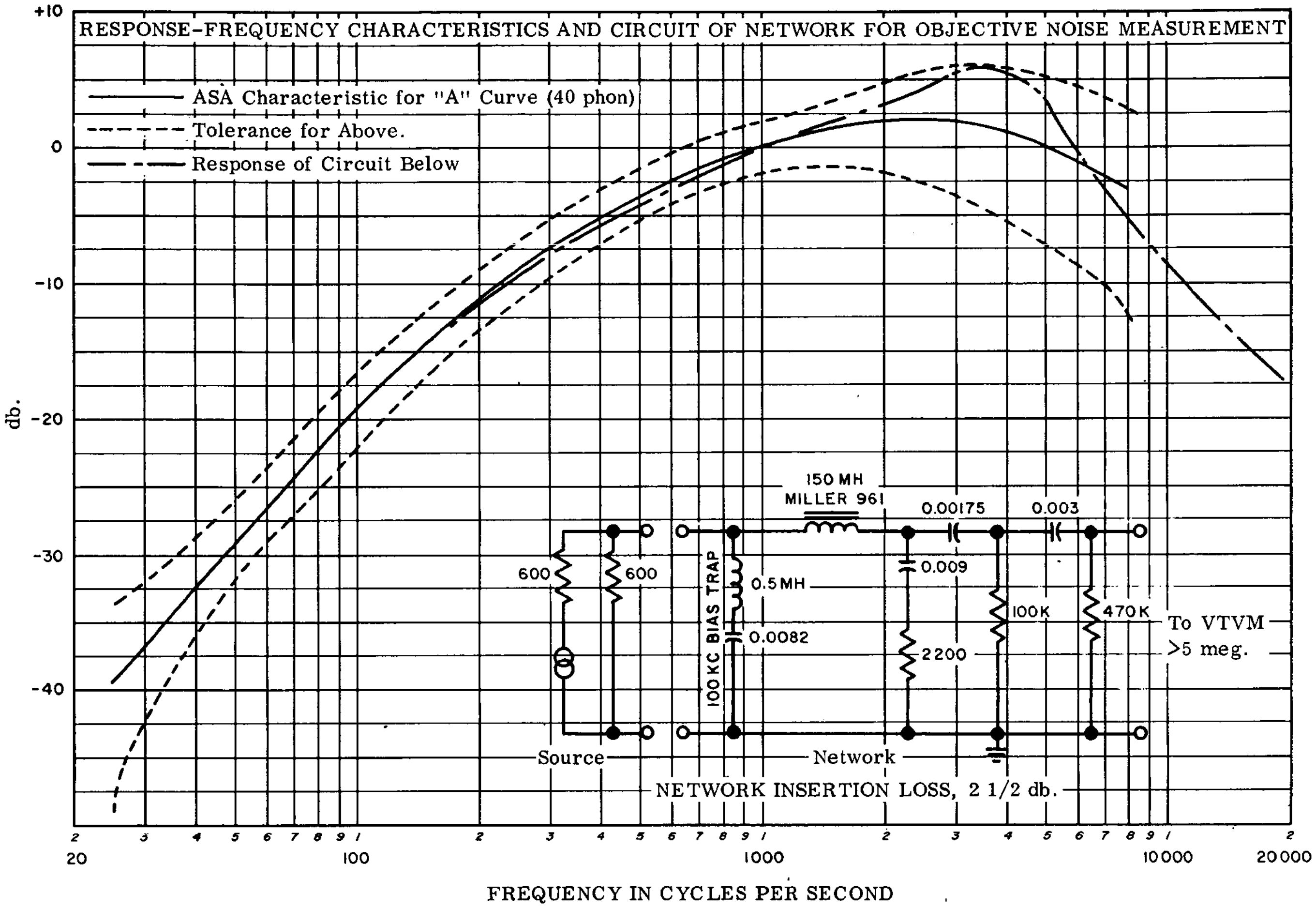


FIG. 7.

ments.[10] The use of weighted noise measurement for magnetic recorders has been proposed,[11] but has not received the acceptance that it deserves. Figure 7 shows the ASA response characteristics and tolerances for the A (40 phon) weighting curve. Also shown is the circuit and response for a weighting network which can be used from a terminated 600 ohm source into a high sensitivity vacuum tube voltmeter, for making weighted noise measurements on tape recorders, electronics, etc. This curve is within the ASA tolerance, but more closely resembles the inverse of the 40 phon ear curve than does the recommended ASA weighting response.

It is possible, then, to find a number (namely the 40 phon weighted measurement) which is closely proportional to the subjective noise level of a system. This can be given as so many db below the normal VU meter zero level.

Equalization

The limiting criteria for designing equalization into a recorder are those of making the pre-emphasis so that the system will have equal probability of over-loading at all frequencies, or of making a post-emphasis which will minimize audible noise from the system. We now want to consider how the compromise is made between these limiting criteria.

The present NAB tape equalization curves[12] are convenient curves which give constant over-all response through the tape machine, using simple networks both in record and in playback. The design at 15 ips has been very conservative with respect to overload capabilities, but the signal-to-noise ratio has not been adequate. Greater attention to the characteristics of the ear, the tape, and the music should provide a system with adequate overload characteristics and greater signal-to-noise ratio.

[10] "Sound Level Meters for Measurement of Noise and Other Sounds", Amer. Standard Z-24.3 (1944).

[11] "Standards on Sound Recording and Reproducing, Methods of Measurement of Noise", *Proc. IRE* **41**: 508-512 (April 1953) (Standard 53-IRE 19-S-1). Although they consider many noise measuring methods, only the broadband system noise method is commonly used.

[12] Lennert, "Equalization of Magnetic Tape for Audio & Instrumentation Applications," *Trans. IRE,* **AU-1 n. 2.** 20-25 (March 1953) and Snyder and Havstad, "Equalization in Direct Record. for Audio," *IRE Convention Record* 4 #7 (Audio & Broadcast): 135-141 (1956).

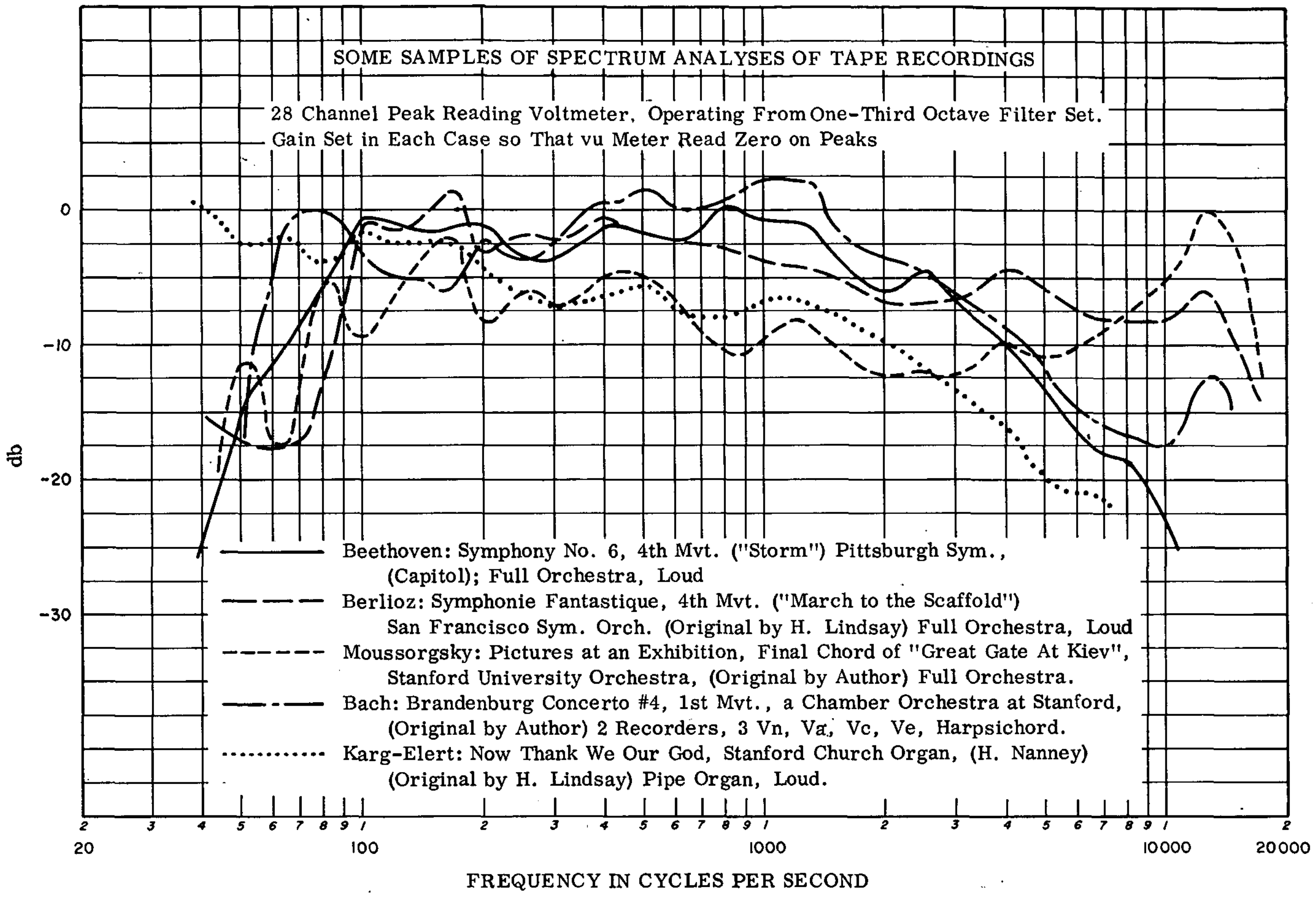


FIG. 8.

This problem has been studied previously for stereophonic sound on film systems[13] based on the Sivian, Dunn and White paper on spectra of musical instruments and orchestras.[14] This work was for a film system for theaters and it is of relatively little direct value in tape recording, but the philosophy is similar. They, like us, appear to have done little in determining the audible effects of the various types of overload which may occur in recording. We hope to investigate these effects more carefully at a later date.

The data in Sivian, Dunn and White for ratio of peak pressure to average total pressure for music, show that for the three different musical selections performed by the same group the energy distributions are considerably different. This led us to a more complete study along the same general lines. It confirmed our worst fears—one can draw a curve of peak pressure *vs* frequency quite arbitrarily and then find some selection of recorded music which will look like this, due to variations in the composition itself, the instrumentation, performers, recording studio and microphone techniques. Even without many microphones close to the instruments and high frequency equalization in the recording console, the energy in the one-third octave at 12 kc may be greater than that in the one-third octave at 400 cps, or it may be lower than 35 db less than the energy at 400 cps. Figure 8 shows some sample spectrum analyses. (There is no such thing as a "typical" spectrum analysis.) Figure 9 shows the maximum and minimum limits of peak energies in 76 recordings which all read zero on the VU meter.

Although these new studies show that some material will have considerable energy in the high frequency part of the spectrum, we note that not all material has appreciable high-frequency energy content. Since the ear is most sensitive to noise in the 1 to 6 kc region, it is desirable that the pre-emphasis be such that the overload in the 1 to 6 kc region shall occur first, but only just before overload at lower frequencies. If this criterion is fulfilled, then we will

[13] John C. Steinberg, "The Stereophonic Sound Film System, Pre- and Post-Equalization of Compander Systems", *Jour. SMPTE* **37**: 366 (1941).

[14] Sivian, Dunn & White, "Absolute Amplitudes & Spectra of Certain Musical Instruments & Orchestras", *J. Acoust. Soc. Amer.* **2**: 330 (1931), and the later discussion of this paper, Young & Dunn, "On The Interpretation of Certain Sound Spectra of Musical Instruments", *J. Acoust. Soc. Amer.* **29**: 1070-1073 (1957).

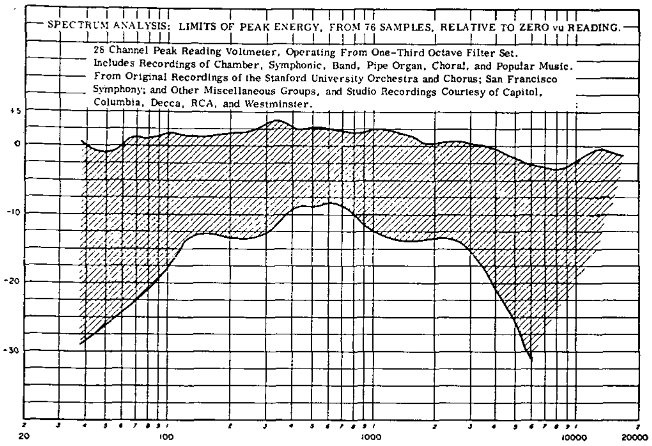
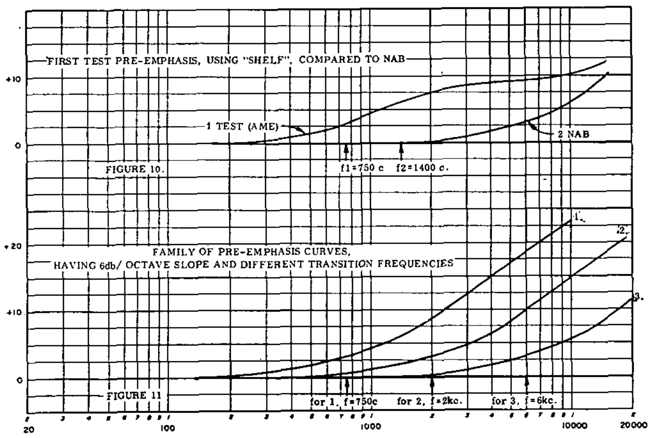


FIG. 9.

have a pre- and post-emphasis system which is as near to ideal as can be practically achieved. Obviously, no one system is ideal for all music recordings.

Figure 10, Curve 1, shows the recording pre-emphasis curve proposed for test, compared to NAB Curve 2. The shape of this new pre-emphasis curve is unusual in one basic respect: The common pre-emphasis curves rise at 6 db per octave from some transition frequency. There is only one variable, the transition frequency. This gives a family of curves as in Figure 11. For pre-emphasis at lower frequencies, the high frequency pre-emphasis is excessive. The newer curve uses two transition frequencies giving a "shelving" effect[15] (Figure 10, Curve 1). Therefore, we can control *both* the frequency at which the pre-emphasis starts and the maximum amount of pre-emphasis. The test pre-emphasis gives considerably more pre-emphasis in the 1 to 6 kc region, but essentially the same pre-emphasis in the 10 to 15 kc region. When this record pre-emphasis is used, the noise spectrum in playback would be as shown in Figure 12, Curve 2. Also on this same figure are the equal loudness curve for the 40 phon level (Curve 3) and the noise spectrum with the NAB equalization (Curve 1). The audible noise should be greatly decreased, and listening tests show that this is the case. It was necessary to attenuate the NAB channel by 7 db to have its audible noise equivalent to that of the test equalization.



FIGS. 10, 11.

[15] A slight additional boost is added above 10 kc to compensate for amplifier and record-head losses.

Figure 13 shows the data for maximum output, and input to produce $\frac{1}{2}$ db less than maximum output for the test pre-emphasis (which is the same as the finally used AME pre-emphasis), and also for the NAB pre-emphasis (from

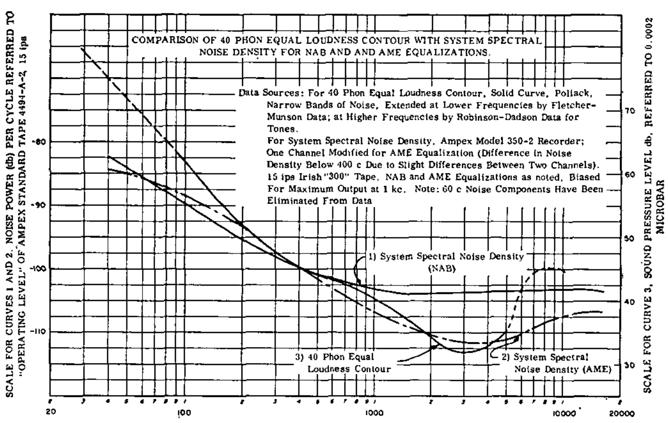


FIG. 12.

Figure 5). The two maximum outputs should differ by the amount of difference shown in Figure 10 for the two pre-emphases. (Small differences in the two channels used for making measurements cause a slight discrepancy in the

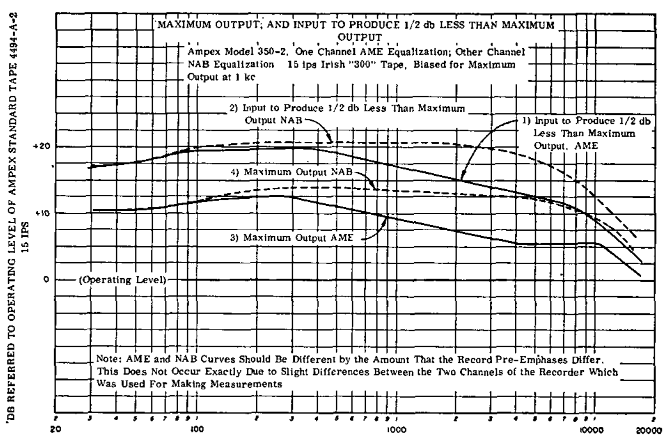


FIG. 13.

maximum outputs; likewise for the two maximum inputs.)

While we were using this first test pre-emphasis, some high frequency noise was still audible and a second test pre-emphasis was devised (see Figure 14). This was similar to the first pre-emphasis, but used an even greater pre-emphasis above 5 kc. Tests made with this pre-emphasis

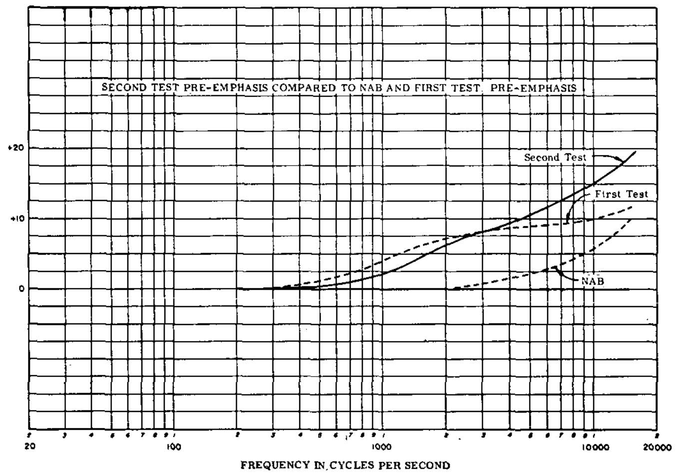


FIG. 14.

showed that severe high frequency overloading occurs before any apparent overload at lower frequencies. This was obviously too much pre-emphasis for 15 ips, but might be usable at 30 ips.

A variety of recordings have been made with the first test pre-emphasis curve (record pre-emphasis Figure 10; noise Figure 12, Curve 2; maximum output Figure 13, Curve 3); they include live recordings made of the Stanford University orchestra, the San Jose State College band, and some studio sessions of popular music at the RCA New York studios. The recordings of symphonic-type material all show appreciable improvement in the noise level with the new equalization. Neither the symphonic- nor the popular-type material had audible overload of any sort with the test pre-emphasis.

It would be very desirable to make further studies to determine what distortions are audible and how one can make a meter which correlates with these audible distortions. Since these recommended investigations will take quite a while, and since this first test pre-emphasis has worked very satisfactorily and is apparently as much pre-emphasis as could safely be used, the Professional Products Division has adopted this first test curve as their Ampex Master Equalization (AME) curve to be used for 15 ips recording of masters by studios.

The further pre-emphasis which is used in the AME curve was possible because full capabilities of the tape were not being used at 15 ips. Brief tests were conducted to see if these principles of utilization could be applied to 3¾ ips recordings, but it was quickly found that even the standard pre-emphasis curve at 3¾ ips caused noticeable overloading. Brief tests at 7½ ips indicate that it would not be possible to get better signal-to-noise ratio at 7½ ips than with the present equalization. It would, however, be possible to use slightly more pre-emphasis at 7½ ips in the 1 to 5 kc region and less pre-emphasis in the 5 to 15 kc region. In doing so, the signal-to-noise ratio would be decreased slightly, but the high-frequency overloading characteristics would be considerably improved. Further studies at 7½ ips have not been conducted.

APPLICATION

Since the AME curve represents a large deviation from the NAB standard curve, and since it appears that the only customers who are dissatisfied with the present signal-to-noise ratio are the companies making sound recordings to be released as phonograph records or pre-recorded tapes, we have felt it undesirable to propose this change of equalization for general use by broadcasters and general users. We do feel, however, that the gains are appreciable for those users who are critical in their requirements for signal-to-noise ratio.

ACKNOWLEDGMENTS

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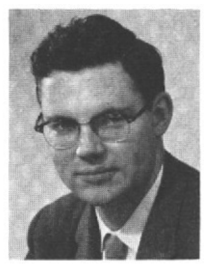
EDITOR'S NOTE: This is the first of two articles on this subject by Mr. McKnight. The second is scheduled to appear in the April, 1959, JOURNAL.



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In 1953 he worked with Ampex Corporation on the development of cinemascope-stereophonic sound equipment. After several years of other work, he returned to Ampex, where he is now Senior Engineer in the Research Division. He spent the years 1953-1956 on the engineering staff of the Armed Forces Radio Service in New York. At the same time he worked as development engineer for the Gotham and the Narma Audio Development Companies. He has always been interested in the problems of magnetic recording, specifically as they concern music.

Mr. McKnight is a member of the Audio Engineering Society and an affiliate member of the IRE.