Subject: ASPECTS OF AUDIO TESTING

I. PURPOSE

For some time, there has been a certain amount of confusion with regard to level measurements in VU, DBM, PROGRAM LEVEL, PLUS 8, TEST LEVEL, etc. This paper will resolve some of the differences of opinion with regard to these terms, and make it easier when conversing with Telephone Company personnel, and when doing "PROOFS OF PERFORMANCE"

II. BACKGROUND

A. BASIC THEORY

A DECIBEL (db) is a unit of measurement which represents a ratio of two (2) powers. It is something like a degree on a thermometer. It is usually used to indicate the difference between an input and an output, a before and after, a yesterday and today. Just as it might be said that the temperature has risen 4 degrees during the morning, so it can be said that the power in a particular circuit has increased 4 db, or 10 db, etc. Just as in using a thermometer, there arose a need to have some reference point, such as zero, this same need arises when using decibels. In the case of the thermometer, there are several reference points. There is one for the CENTIGRADE thermometer which is set at the freezing point of fresh water. There is a different zero for the FAHRENHEIT scale based on the temperature of freezing a salt solution. The KELVIN scale has still another point for its zero (probably the most logical, as well as the most useful, scientifically) based on the temperature at which all molecular motion in an object ceases. Thus, it represents "absolute Zero". There is nothing colder than this.

Just so, there are several reference points when it comes to a scale based on decibels. Acoustically, a zero reference was set at the threshold of audibility, but when it came to electronic equipment using loudspeakers to overcome the background level in the average living room, this zero was found to be too low, since based on it, the average background noise was found to be way up at 40 db on this acoustical scale. Therefore, an arbitrarily set value of 6 milliwatts of power dissipated across 500 ohms was set as the reference power, and this level became known as 0 db.

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All was well and good until it came to measuring complex wave-forms such as those produced by normal speech and music. No longer did RMS value have a meaning. The average power level of speech and music is much lower, compared to the peak value, than for a corresponding sine-wave with the same peak. Sharp, thin peaks are the rule -- greater spacing between peak pulses is produced -- in short, sinusoidal measuring techniques no longer apply. Therefore, a special meter was developed (with certain specific damping characteristics) which would respond in a definite manner to these complex wave-forms. This meter read a sort of average audio power (as well as reading the RMS value, if a sine-wave -- of 1kHz. -- was used). It required its own zero. An altogether new reference was used, namely, 1 milliwatt of power across 600 ohms, using a frequency of 1000Hz.

Since this was a meter which measured the "volume of audio", the unit of measurement became known as a "VOLUME UNIT", or VU. If a 1000Hz. sine-wave is measured by a VU meter (or VI -- Volume Indicator), the RMS value of the voltage is the result. A 1000Hz. sine-wave with power of 1 milliwatt across 600 ohms will measure 0 VU.

By Ohm's Law, we find that this represents an RMS voltage of .775 volts (across 600 ohms).

$$P = E^2/R = .775^2/600 \text{ (watts)} = .6/600 \text{ (watts)} = 1/1000 \text{ (watts)}$$

$$1/1000 \text{ (watts)} = 1 \text{ mw.}$$

About this time, it was decided, for the sake of conformity, to re-determine the standard db, and use this same 1 mw. base instead of the 6 mws. In order to distinguish between the two standards, the power ratio using 1 mw. base was designated a "dbm" (the "m" standing for milliwatt). The old designation of 0 db has practically been eliminated, and the new reference of 0 dbm is almost universally used. With a sine-wave then, VU and dbm mean the same thing. However, dbm is used ONLY with sine-wave signals, and NEVER with audio.

B. PROGRAM LEVELS

When program audio is sent through a circuit terminated in 600 ohms, the level can be read as a certain number of VU above or below the standard 0 VU which has been established. It has become more or less standard practice for ABC Radio stations to use plus 8 VU as the standard for feeding telephone lines, transmitter lines, etc. If a meter attenuator is inserted in the metering circuit, and set for plus 8, then when the meter "peaks zero", the VOLUME UNIT of that peak is in reality 8

VOLUME UNITS above the standard reference. It is called "zero at plus 8", and is, in reality, plus 8 VU. If a 1000Hz. sine-wave is used, a VU is the same size as a db, and 0 VU, and 0 dbm are then the exact same level.

Actually, the restrictions (600 ohms, 1000 cycles per second, etc.) are for the VU and not the dbm. The dbm is a unit of finite audio power, when using sine-wave signals, which has as its zero reference a power of 1 mw. This can be across any impedance whatsoever. 1 mw across a 4 ohm speaker coil, or 1 mw across a ½ meg grid resistor would each represent 0 dbm, although the voltages would be widely different. Thus, the meter would have to be calibrated in volts, and the value of the power in dbm, calculated.

The VU on the other hand is tied to a 600 ohm terminated circuit because it uses a specific meter calibrated in VU (and thus, in a sense, in power units). Actually, it is a voltmeter and will read the actual voltage across the two points being measured, but it assumes that the impedance is 600 ohms when it does the mathematics necessary and reads out in terms of power (VU). Even a meter which is calibrated in dbm, and reads dbm directly must be used in a circuit with the impedance specified for the particular meter. This is usually 600 ohms, but does not have to be. In the case of the VU meter, the impedance must always be 600 ohms, because that is one of the specifications of the meter. Thus, if a 600 ohm dbm meter is compared with a VU meter, and sine-waves are used, they will both read the same.

C. HEADROOM

Now a VU meter is so constructed and damped that a sharp audio peak can not possibly be followed by the meter. In fact, the meter presents a sort of average of the audio, and the "peak" that is seen and read on the meter is only a fraction of the actual voltage that was present to produce the peak. This produces a very important concept.

The statement has often been made that "audio peaks are higher than sine-wave peaks for the same VU meter reading". In fact, this statement is made frequently, but without a picture in the person's mind of what the real significance is. It has also been said that there is a difference of about 7 db between the two. (Some say 10 db, and some, from 8 to 14 db). In order to illustrate this, an oscilloscope was set up and pictures taken of the phenomenon under discussion. (An oscilloscope is used frequently in TV, but is extremely useful in audio work to SHOW what is being HEARD).

No hard fast rule can be laid down with regard to the real amplitude of audio peaks, because so much depends upon the tone being used, the vowel sounds, and the particular pitch or quality of a musical note. Again, it is a matter of averages. The following photographs were taken with slight time exposures, less than a second, so that there was an average of many tones and syllables in each case. First some tone was sent through some equipment so that the VU meter read zero at plus 8 VU. (This was also plus 8 dbm). The signal was fed into an oscilloscope, and the vertical gain adjusted to obtain 2 cms. of deflection. The trace was photographed. A card, cut to size, was then placed over the oscilloscope screen so that the sine-wave was completely covered. (In this way, the next trace would not obliterate the sine-wave on the photograph, since a double exposure was planned).

Audio was then picked up from the radio station, and an audio man "rode gain" as he would have for a program, aiming for a good ZERO LEVEL (at plus 8, of course). The resulting trace was then double-exposed on top of the previously photographed sine-wave. Only that portion of the audio which was greater in amplitude than the sine-wave was visible because of the card covering the center portion of the oscilloscope. The spectacular result is shown in figure 1-A.

The 2 cms. represents the reference level, and it can be seen that audio peaks cover at least a 5.6 cm. range, representing a voltage 2.8 times the voltage of the reference signal. This translates into 9 db.

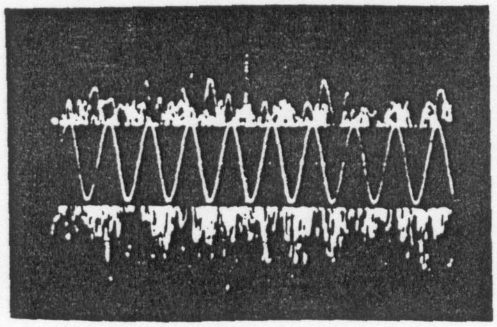

FIGURE 1-A

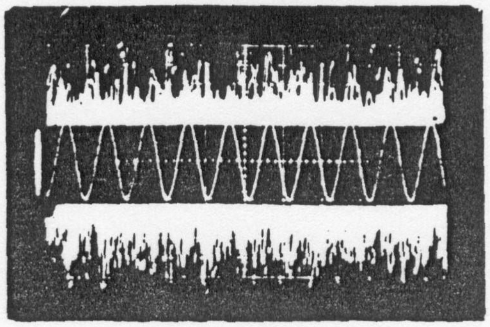

FIGURE 1-B

Figure 1-B is another picture of the same thing, but with different program material. The ratio here calculates to 10.8 db. Occasionally during this experiment, peaks as high as 14 db above reference 0 were noted, although not caught on a photo. This average value of 10 db or so is what is termed HEADROOM.

Thus, as a VU meter is read showing a tone at 0 VU, and then program audio peaks hit this same 0 VU, it must be remembered that the peaks that cause the 0 hits are in reality approximately 10 db higher in voltage, or approximately 3½ times the voltage of the steady state tone that also read 0 VU.

III. PROCEDURE

A. STUDIO PROOFS

1. GENERAL

In checking out the studio, we are primarily interested in determining the distortion produced by the system, its over-all frequency response, and how far below program level is the basic noise of the system. The first question which arises for all three tests is to determine the level at which the tests are to be made. Since the tests will be made using a steady state tone, we realize immediately that we will not be checking out the system properly unless we do something about checking at the peak-to-peak amplitude that we have found audio peaks actually run.

Since we say that there is about a 10 db discrepancy, it has become customary to run all studio checks at a level which is 10 db higher than normal program levels. This means plus 28 dbm where a normal program level would be plus 18 dbm, or plus 18 dbm where normal program level is plus 8 dbm, or -50 dbm where normal program level (microphone) would be -60 dbm, etc.

The actual test line-up of amplifiers and equipment will vary from station to station, but let us use for example Figure 2 showing a Pre-Amp for the mike, followed, after suitable attenuators for control, by a Booster Amplifier, and finally a Line Amplifier for the studio output. This would then go through an AGC Amplifier and Line Amplifier for transmission to the telephone company or the transmitter.

Assume for purposes of discussion (this also happens to be a good standard to use), that average mike level is -60 VU. Thus, to duplicate program meter readings, a tone, at the input to the Pre-Amp, of -60 dbm would be used.*

Set up a signal generator at 400 Hertz, and -60 dbm at 150 ohms output impedance, which is the usual input impedance of a Pre-Amp. Send this low-level signal into the system, and leaving the Master Volume Control where it is usually used by the audio man, set the microphone attenuator so that the output of the studio is normal program level as shown on the VU meter. (This is usually 0 at 8 VU.) This is then metered after the final Channel Amplifier, and either before or after the final splitting pad. (See Figure 2 - Next Page) Let us look at it BEFORE the final 10 db pad which normally feeds the phone line, monitors, etc. At this point then, the level will be plus 18 dbm. (After padding, the normal plus 8 will appear for transmission over the line.)

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* 0 dbm is produced by .775 volts across 600 ohms, or by .387 volts across 150 ohms. A signal 60 db less than this would be .000387 volts across 150 ohms.

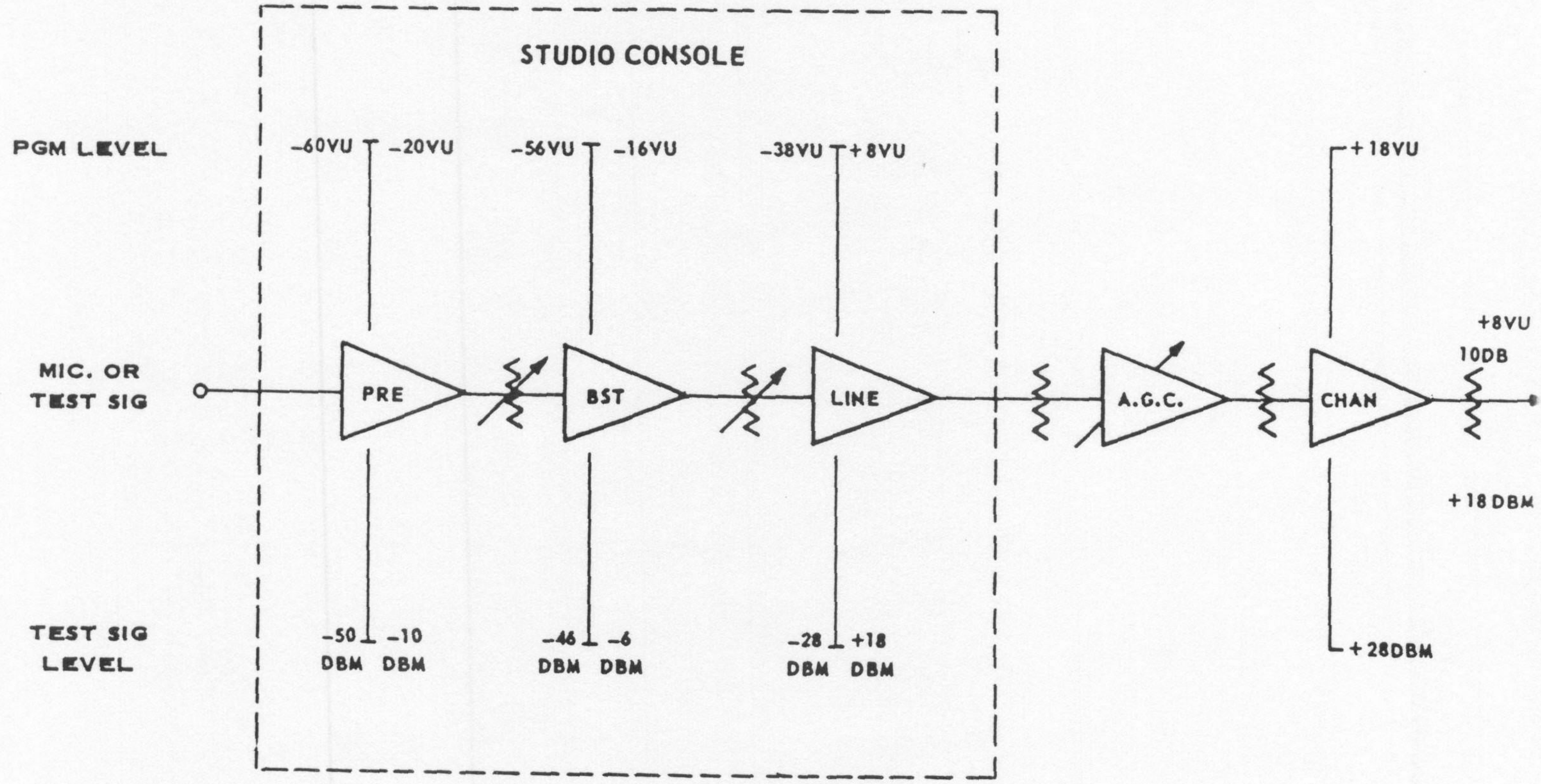
STUDIO CONSOLE
PGM LEVEL
MIC. OR TEST SIG
TEST SIG LEVEL
-60VU
-20VU
-56VU
-16VU
-38VU
+8VU
+18VU
PRE
BST
LINE
A.G.C.
CHAN
+8VU
10DB
+18DBM
-50 DBM
-10 DBM
-46 DBM
-6 DBM
-28 DBM
+18 DBM
+28DBM

FIGURE 2

NOW COMES THE BIG MOMENT! Increase the SIGNAL GENERATOR OUTPUT by 10 db (to -50 dbm)! This is to make the sine-wave peaks more closely resemble the actual audio peaks that are normally used through these circuits. See Figure 3.

The needle of the VU meter in the studio console will "wham" against the pin, but have no fears or misgivings. Do NOT change the attenuator, or remove the meter from the circuit. I have been doing this at regular intervals for several years now, and although I had strange feelings the first few times, I soon found that the meter is built to withstand this mistreatment. In fact, it can withstand this level continuously for hours. The output level at our measuring point is now plus 28 dbm. ALL STUDIO TESTS ARE MADE AT THIS LEVEL.

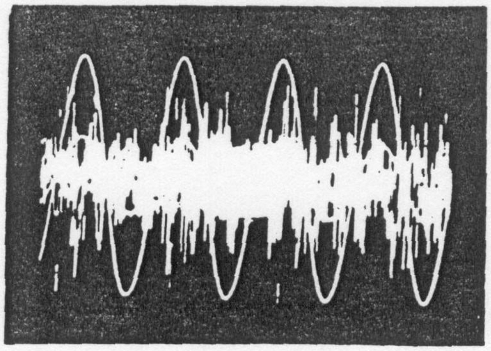

FIGURE 3-A

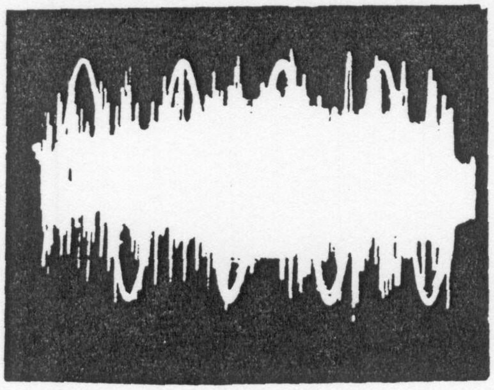

FIGURE 3-B

2. NOISE

It is not intended in this article to describe methods of making studio or transmitter proofs of performance, but rather to point out some interesting traps and phenomena that have been observed during several years of making these proofs at regular intervals.

To make a noise measurement, the reference level is taken at the plus 28 dbm point. Using this as a reference, the line is terminated at the input jack field, and using a Noise and Distortion Analyzer, the noise level is determined. If everything is working properly, the Pre-Amp will be the determining factor in the noise level. To pass studio specifications, the noise should be 65 db below the reference of plus 28 dbm. (This, of course, converts to -37 dbm.) It also means that the noise is 55 db below normal "program level."

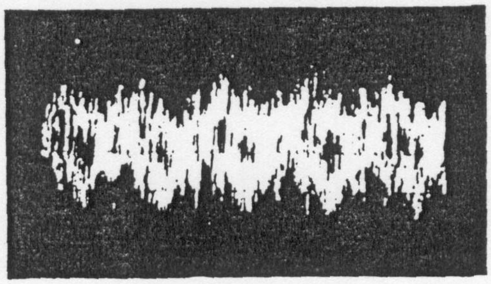

FIGURE 4-A

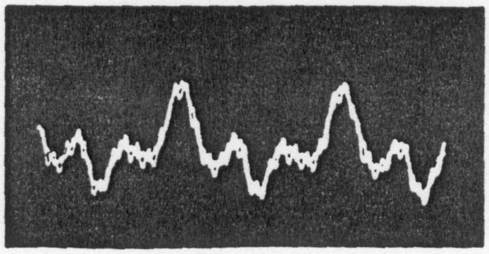

FIGURE 4-B

Figure 4-A shows the nature of the noise that is present at this low level. (In this particular case, the noise measured 72 db below reference.) The over-all pattern shows some semblance of 120 Hertz. The fine peaks, of course, are a random noise caused within the Pre-Amp first stage, and/or the input transformer.

3. DISTORTION

In making distortion measurements, again the plus 28 dbm level is used as a reference, and the Distortion Analyzer removes the original signal (400 Hertz for example), and what is left is read as distortion. Figure 4-B shows this distortion. In this case, it measures 0.9%. The basic configuration is 180 Hertz.

There is something interesting about this measurement of the distortion. Since it was difficult to understand where 180 Hertz could be originating in the studio audio equipment, I suspected that it might be coming from the Signal Generator itself. This indeed proved to be the case. The 180 Hertz signal is apparently entering the system someplace in the output transformer of the generator. There is a calibrated ladder attenuator preceding the output transformer. It was found that by inserting a 40 db pad between the generator and the audio jack field, and then increasing the signal to the output transformer by 40 db by means of the ladder attenuator, it became possible to attenuate the 180 Hertz signal, while keeping the signal strength of the signal the same, as far as the studio was concerned. Figure 5A shows the distortion signal again as shown in the last photograph. Figure 5B represents the frequency that was used (400 Hertz). Figure 5C (to the same scale as Figure 5A) shows the result after the 40 db pad was inserted. THE SIGNAL REFERENCE WAS THE SAME, but the distortion now measures only 0.15% (a decrease of 15 db). The distortion consists mainly now of third harmonic of 400 Hertz, or 1200 Hz. Thus, the studio was really cleaner than the tests at first seemed to indicate -- an interesting situation!

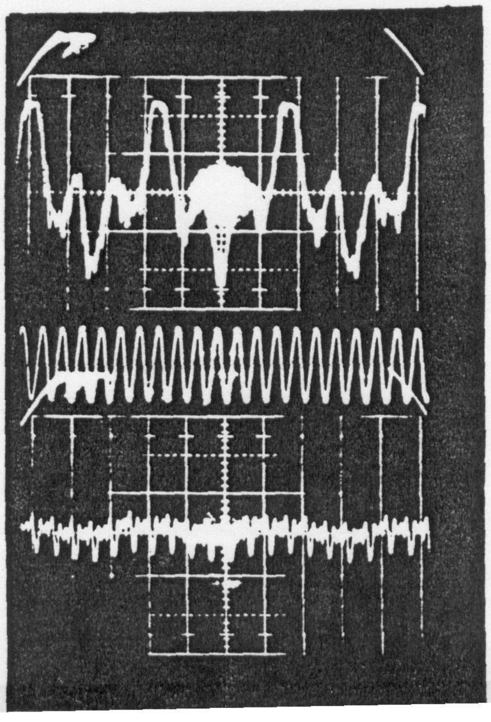

FIGURE 5-A

FIGURE 5-B

FIGURE 5-C

4. FREQUENCY RESPONSE

The only note that I would like to add regarding frequency response measurements is that the 40 db pad is frequency sensitive and so MUST NOT BE USED FOR FREQUENCY RESPONSE MEASUREMENTS. The small amount of 180 Hz. present in the generator plays no part in the frequency response measurements.

B. TRANSMITTER PROOFS

1. GENERAL

There are one or two interesting points that should be brought out when proofs are made through both the studio and the transmitter. Again, the matter of the level which which is used is a matter of importance. If the level of the signal into the studio is decreased to the program level of plus 8 dbm and sent to the transmitter, it will be found that the modulation of the transmitter will be about 45%, or approximately 7 db below 100%. The transmitter is a "peak-sensitive" device, and 100% modulation is determined by the actual peaks. Thus, a similar level sine-wave (as measured on a meter) would be about 7 db lower on peaks. Obviously, the test level must be brought up to 100% modulation. The question arises as to WHERE the level should be brought up to produce 100% modulation. The limiter at the transmitter has been disabled. Let us see what would happen if the level were to be brought up by means of the amplifier in the limiter.

Let us assume, for example, that our studio barely made its noise specification of 65 db below reference plus 28 dbm. Since we are now using a signal 10 db lower than this, or plus 18 dbm at our measuring point (plus 8 dbm at the input to the telephone line to the transmitter), the noise would now appear to be only 55 db below this new reference. If the gain of the amplifier at the transmitter is now raised to produce the 100% modulation, both signal AND noise coming from the studio are raised, and the relationship, or 55 db, remains the same.

Let us suppose further that the transmitter itself (FM) also just barely makes its specification of 60 db below reference 100% modulation. The resulting over-all noise through the entire system would then be about 53½ to 54 db below 100% modulation, and this would be far from the FCC specification of 60.

Even if two identical signals of -60 db were added, the sum would be only -56 db. (I know the books say it should be -57, but I have never found this to be the case in actual practice.) Similarly, -63 plus -60 equals about -58½, etc. Finally, when one noise signal is about 6 db or farther from the other one, there is no apparent addition. (i.e., 66 plus 60, or 68 plus 62, etc.)

This then gives a clue as to where the signal should be raised to produce the 100% modulation. With the generator in the studio set for -60 dbm, and no controls touched at the transmitter, except for disabling the limiting action of the limiter, the modulation of the transmitter should be brought up to 100% BY INCREASING THE SIGNAL GOING INTO THE PRE-AMP. It will be found that this increase is usually in the nature of about 7 db (achieved by adjustment of the ladder attenuator on the signal generator) so that the studio signal-to-noise ratio in the extreme case mentioned above would become about 62 db below the reference studio output. Thus, -62 added to -60, although still not passing specification, would be about -58 db.

THIS SHOWS THAT A BETTER THAN MINIMUM NOISE MEASUREMENT MUST BE MET IN THE STUDIO AND THE TRANSMITTER IN ORDER TO PASS THE OVER-ALL SPECIFICATIONS.

2. TESTS

This level of signal then is used throughout all the over-all tests through studio and transmitter. The 40 db pad is again used for the distortion measurements only. During the frequency response test, the signal is increased or decreased AT THE INPUT TO THE STUDIO for each frequency, to allow for the pre-emphasis at the transmitter for various frequencies.

IV. CONCLUSION

Thus, we see that certain precautions must be taken when making tests of audio equipment. HEADROOM must be allowed for and taken into consideration. The terminology VU, DBM, etc., must be carefully understood so that there is uniformity of thought and expression during tests. All in all, it is a very intriguing subject, and one in which some careful study and consideration will pay dividends in better audio, smoother operation, and greater satisfaction for the operating personnel.