Measurement Techniques and Standards

—Norman H. Crowhurst

The author explores the essentials of audio measurements from a professional point of view. Special emphasis is placed on proper termination and its importance in gain measurements. The various standards of measurement are also covered.

AUDIO MEASUREMENTS are usually regarded as comprising three basic items: gain, frequency response and distortion. A fourth dimension in audio is dynamic range, limited at one end by the maximum relatively distortion-free level and at the other by noise.

Thus, one way or another, distortion is inevitably linked with level. Usually the percentage of distortion rises as level rises, up to a point where distortion fairly suddenly becomes quite considerable—unbearable to listen to. This is the absolute upper limit of dynamic range. Preferably it should be beyond the upper working limit, so it is never reached during operation.

Below this upper level limit, most high quality equipment shows lower distortion the lower the level, but in some equipment this does not happen. Suppose, for example, that a circuit uses class B amplification, producing crossover distortion: below a certain level the magnitude of the distortion component may not diminish any further as the magnitude of the wanted signal does, so that the percentage distortion actually rises as level is reduced. Such a system sounds far from clean and would be unacceptable professionally.

Gain and frequency response are also interlinked with level and distortion, as well as being dependent on an important factor we didn't name above: impedance. All these factors tend to complicate measurements, and make the establishment of meaningful standards and techniques difficult. Let's take the various standards and look over some pitfalls in their use. This will help avoid some of the common mistakes, and also lead to the use of better methods.

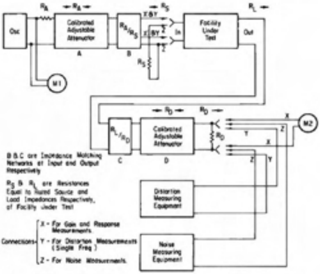


Fig. 1. Typical Measuring System for Audio Facilities (From EIA Standard: Audio Facilities for Radio Broadcasting Systems).

Gain.

Gain, frequency response and distortion tests are all made with a similar measurement set-up, according to audio standards. This consists of an audio oscillator with a metered output, fed through an attenuator and input matching network to the item of equipment being tested, which is terminated with a further matching load and output metering (fig. 1).

Most professional equipment operates at line impedance, commonly 500 ohms. So the input resistor will be 500 ohms, the attenuator will work at 500-ohms and probably, unless it is a monitor amplifier, also an output impedance of 500 ohms.

Insertion gain is defined as the change in level brought about by inserting the amplifier or other item of equipment between input and output, as compared with a direct connection (fig. 2). This is a somewhat academic definition.

If input and output impedance are the same, a direct connection will result in a 6 dB loss: voltage at the input will be double that at the "output." If the equipment is now inserted and attenuation added till the same double-value input voltage produces the same output voltage, the attentuation inserted will be equal to the insertion gain.

That's fine in theory, and it's the best we can do toward a general or universally true statement, based on standard practice. And it will be correct *if* the input and output impedances are themselves 500 ohms, as well as requiring to be matched to these impedances. But this is not always true.

For example, an amplifier may be designed to work from a 500-ohm source, but it does not provide precisely 500-ohm loading for the input at the same time. It will work satisfactorily from a 500-ohm line, because that is what it is designed to do. But its actual input impedance may be a few thousand ohms.

So using this amplifier's input as termination for a 500-ohm line, or for a 500-ohm attenuator, will result in not as much loss as the 6 dB theoretical—maybe only 1 or 2 dB. If this termination is applied to a real line, needing 500 ohms to avoid reflections, it is customary to apply a 500-ohm termination directly across the line as well. The equipment will now look out of 250 ohms, instead of the ideal 500 ohms. This addition can invalidate either gain or frequency response, or both.

At the output end, an amplifier can be designed to feed a 500-ohm (or any other desired impedance) load, without necessarily providing a source impedance of the same value. Most often, unless special measures are taken in the amplifier design to equalize these impedances, the source impedance will differ from the impedance of the load it is designed to feed into.

These two possibilities for deviation mean that insertion gain may not hold to its academic definition. Whenever source and load impedance, whether it's at the input or output end, are not equal, putting the equipment into a circuit changes the basic, theoretical 6 dB loss effective without the equipment in circuit.

The error this causes is particularly noticeable when equipment is cascaded. For example, a system may be built up of mixers, equalizers, compensators, etc., with line amplifiers or preamplifiers interspersed between these various groups of controls to equalize level losses. When the whole system is measured for over-all performance, the total gain may be considerably less or more than the sum of the parts, because of these discrepancies.

The discrepancies can best be overcome by figuring things out in a little more detail, although the standards do not call for this procedure. Know the input and output impedances, both internal values as well as the nominal, or matching values. This may require a little more measuring, but it can save some disappointments.

Frequency Response.

The same variations in impedances can invalidate frequency-response measurements. The measurement procedure calls for use of correct input and output termination. Actual systems may connect together inputs and outputs with the same rated, or nominal impedance, but in which neither has the *actual* value named. Each is merely intended to *work with* that value—say 500 ohms.

Thus, for example, an output designed to feed into 500 ohms may have a source impedance of 20 ohms, while an input designed to accept from a 500-ohm circuit may present a terminating load impedance of 2,000 ohms. Connecting these items together seems correct: two 500-ohm circuits, and input and an output, are connected together—matching is correct; but in fact, a 20 ohm source is connected to a 2,000 ohm load.

Gain as well as frequency response of both units is predicated on coupling to a true 500-ohm circuit. So, as well as changing the apparent insertion gain of both units, the frequency response of each may be different from its measured or specified performance. An input circuit designed to work from 500 ohms may work quite differently when connected to 20 ohms, as may the output connected to 2,000 ohms.

Putting in attenuator pads that reduce level by a known amount, and bring the impedance nearer to its nominal value, can minimize both these deviations (fig. 3). If you know your situation, you can correct accordingly. For example, a 500 ohm 6 dB pad would make the 20-ohm source look like about 300 ohms and the 2,000 input look like about 650 to 700 ohms—an improvement. A 10 dB pad will do much better. The loss of level must be available—amplification sufficient to compensate it.

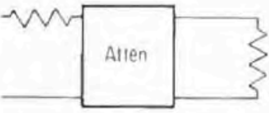
The only way to really know is to run gain and frequency response checks under all possible operating conditions, particularly extreme settings of any controls, and never to assume that the over-all performance of a system will be the sum of its parts!

Distortion.

Although errors can creep into the measurements of gain and frequency response as just discussed, unless carefully applied, these errors can be corrected fairly readily. With distortion measurements we encounter more difficult problems. While the differences already mentioned as affecting gain and frequency response can also affect distortion, there are some things even more basically deficient about distortion-measuring standards. And extra distortion due to unintentional mismatch may not be so easy to compensate out, as deviations in gain or frequency response. The only way is to carefully see that matching is correct, both ways (forward and backward).

The more basic deficiencies about distortion measurements have shown up to some extent in the variety of tests that have already been developed. The earliest, still

Fig. 2. The concept of insertion gain: the gain with the facility under test inserted is compared with the condition when a simple electrical by-pass is made, as shown here.



dB Atten	Mismatch Ratios 1.5	2	3	5	7	10	20	100
3	1.2	1.4	1.7	2	2.2	2.4	2.7	3
6	1.1	1.2	1.3	1.4	1.5	1.5	1.6	1.7
10			1.1	1.1	1.2	1.2	1.2	1.2
13	Less Than 10% Error Reflected							1.1

Fig. 3. Inserting a two-way attenuation pad to minimize mismatch due to internal impedance not being the same as the nominal or rated impedance. Such an attenuator should be of the two-way matching type (T, pi, H, box or lattice configuration). The table shows the reduction in mismatch for various attenuation values.

accepted as the basic standard, is the test with a single input frequency, to determine the generation of spurious frequencies (harmonics) in the output.

Later, recognizing some of the limitations of this test, came various intermodulation tests, using more than one frequency.

Two methods are available for the single-frequency test. Each must use the purest possible input signal tone—at least an order of magnitude better than the expected distortion content of the equipment tested: if the distortion is expected to be 0.1%, the input must have less than 0.01% harmonic present.

These two methods differ in how the output is analyzed. The classic method uses a wave analyzer, measures harmonics, usually up to the 5th, but higher if appreciable amounts are suspected present, from which the resultant distortion is calculated by taking the square root of the sum of the squares.

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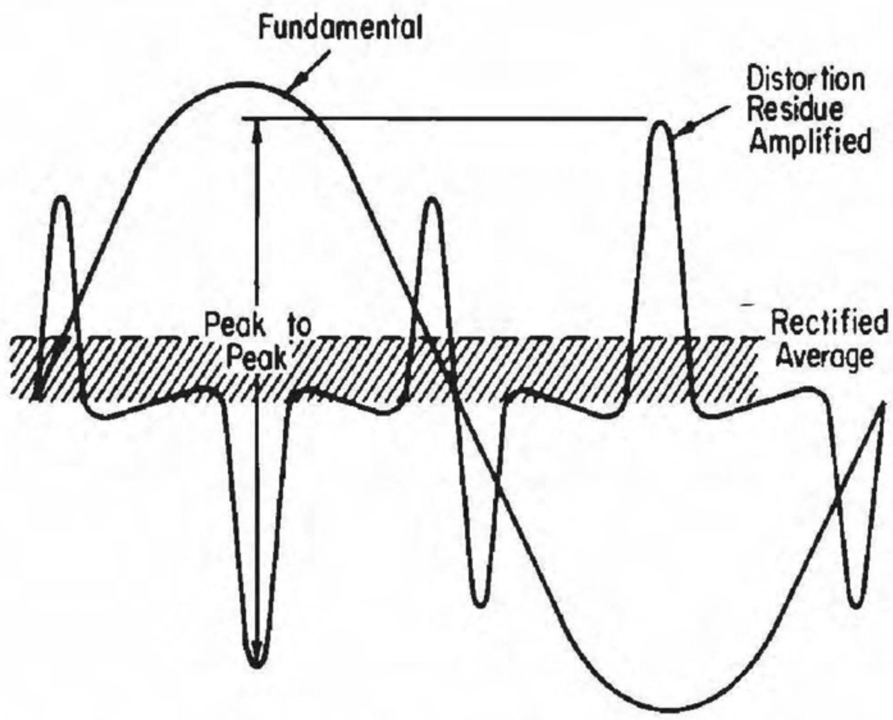


Fig. 4. Why peak reading of residual composite harmonic is more meaningful than mean reading: in this typical waveform, for a low-distortion signal, the peak value has about five times the mean value (peak-to-peak is about 10 times).

For example, if the readings are: 2nd, 0.04%; 3rd, 0.06%; 4th, 0.01%; 5th, 0.007%; the resultant is

$$0.04 \quad + 0.06 \quad + 0.01 \quad + 0.007$$
$$= 0.0016 + 0.0036 + 0.0001 + 0.000049$$
$$= 0.005349 = 0.073\%$$

The alternative may find a less accurate "rms" distortion (for whatever that is worth), with a simpler measurement method: it uses a frequency-selective bridge to null out the fundamental or input frequency, leaving the total harmonic waveform to be measured by either a mean-reading or peak-reading meter.

The harmonic residue, with the fundamental removed, will have a peak value that has been shown to be somewhat more meaningful than the average value, because the peak value represents the departure from the true waveform, which is nearer to the relative effect of the spurious sound heard (fig. 4).

So with the simplified method there is some merit to using a peak indication. But, using a wave analyzer to read individual harmonics, peak/average always has the same ratio (1.414) so there's no point to the distinction with this method. Only by reading the composite residue, with fundamental removed, can this improvement be utilized.

But there is a way to make the whole measurement easier, that many equipment manufacturers have adopted for their production test facilities, although no professionally available measuring set as yet employs it. This consists of a bridge that balances out *input* against output (fig. 5) rather than nulling out the *frequency*.

This method has several advantages: most obvious to the user, the null is far less sensitive to work with. Using frequency nulling, if frequency changes at all (even a hertz or so) balance is completely lost. Using input/output nulling, change in test frequency has comparatively little effect on balance. It is much more comfortable to use.

Another advantage is that harmonic content in the test signal does not have to be a whole order of magnitude lower than the quantity to be measured. This method allows the whole output waveform to be nulled by a component representing the whole input, leaving only spurious components due to the equipment being tested.

While nulling of harmonic components may not be perfect when the fundamental is nulled, it will usually be well within 10%, which gives the order of magnitude margin, without having to be that much "cleaner." Thus, to measure a harmonic content of 0.1%, an input signal of 0.1% harmonic is satisfactory. The harmonic in the input will null its part of the output to within 0.01% or better.

IM Distortion.

Multi-frequency distortion testing uses one of two signal forms. Most popular in this country is the *SMPTE* test, using a lower frequency with a higher one "riding" it, usually in 4:1 ratio. When intermodulation occurs, the higher frequency is modulated by the lower one, causing spurious components, usually shifted from the higher frequency by difference frequencies that are multiples of the lower frequency.

The usual method here employs filters that first remove the lower frequency from the output, then rectify the higher frequency and "smooth" it, to measure any fluctuations in its amplitude occurring at the lower frequency (fig. 6). For the kinds of distortion in this category that were common in earlier amplifiers (before feedback) this method gave a fairly good indication.

But with modern feedback amplifiers, the lower frequency can frequency modulate the upper one, as well as amplitude modulating it. The traditional method detects only amplitude modulation. Audibly, both forms are equally annoying. Searching for the spurious components with a wave analyzer and then calculating the rms resultant intermodulation would overcome this deficiency.

Alternatively, an adaptation of the input/output nulling method used for harmonic measurement would also detect both forms of intermodulation more simply.

The second kind of intermodulation test, adopted as the European or International standard, uses two higher frequencies, usually spaced by a controlled difference frequency, such as 100 Hz. Thus the input frequencies might be 5,000 Hz and 5,100 Hz. The output is then tested for content at 100 Hz, which is a distortion product.

This first-order product is due to asymmetrical distortion. And one problem of this form of test is that the product magnitude is not necessarily related to the annoyance value of the distortion it represents. And a complete test must explore for multiples of the difference frequency.

Each kind of test has some validity. There is no fixed relationship between the results of the three tests (the single-frequency harmonic and the two forms of IM test) and each is subject to interpretation, according to the method of measurement and the form of distortion present in a particular case.

For example, the difference-frequency test can be quite invalidated if there is a bass cut in the system, because the difference frequency will be seriously attenuated in the measurement. But the form of distortion could still be quite objectionable on music.

Noise.

These measurements are relatively simple, so we'll not take space to describe them in detail here. The input must be terminated with its nominal impedance, suitably shielded, to prevent spurious pickup (hum) and the output measured for noise, with or without weighting.

Whether or not weighting is used is important in comparing results obtained with other equipment. Weighting is intended to make the reading indicate more closely the audible effect of the noise. Without it, noise concentrated in the lower range may indicate much higher than it sounds, while noise concentrated in the upper frequency band may sound much higher than the indication would indicate.

Transient Performance.

The big problem in audio measurements is that the tests just described don't account for everything that happens when handling program, especially music. Most of the troubles experienced in modern equipment are related to transient handling ability. Many amplifiers that perform impeccably on steady tone test signals seem to "break up" in some way on certain parts of musical program.

One form of transient test uses tone bursts, where the signal level is switched at a lower frequency, to give opportunity to observe or measure what happens when a loud signal suddenly hits an amplifier and again how it recovers after the loud signal is removed.

An important part of this, which can be detected by tone-burst testing, but for which we may yet find better methods, is the overload recovery effect in an amplifier or system. Most tests, both with tone bursts and steady tones, keep the signals within the rated maximum level—or at least they don't exceed it.

Program often produces isolated "spikes" that do overshoot the maximum level for quite a brief moment. Some amplifiers or equipment find this really crippling—blocking for a fraction of a second after it. Others handle these spikes without trouble. While tests have been devised to demonstrate the difference, there is room for test standardization in this important area.

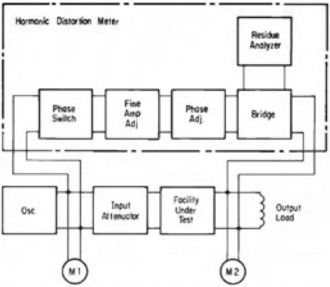
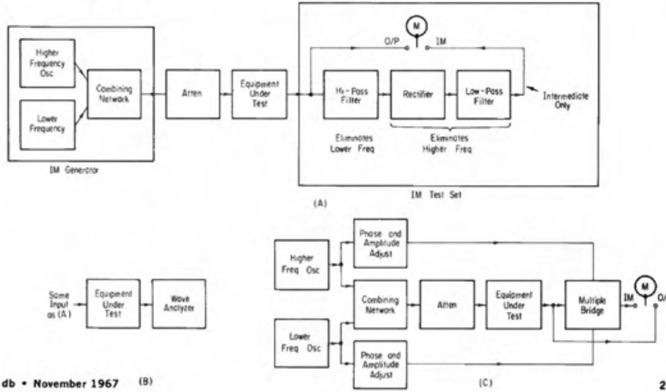


Fig. 5. An alternative measurement technique for harmonic distortion that is both easier to use, and gives greater accuracy within limitations of oscillator purity.

Fig. 6. Three methods of measuring intermodulation distortion of the SMPTE type: (a) the classic SMPTE method, which detects only amplitude intermodulation effects, fails to detect frequency modulation effects; (b) using a wave analyzer and calculating the rms value from all the spurious-frequency readings obtained, will catch what the classic method misses; (c) a simpler form of test, using bridge nulling, that catches all forms of intermodulation occurring with this form of test signal.



Letters

Measurement Techniques

The Editor:

I have just received my first issue of your new magazine entitled **db**, and permit me to congratulate you on such a fine publication. Having been in the professional audio field since 1945 (and interested in it since some time before that) I can see the need for such a publication. Most of the other audio publications (and I am sure you know them) have gravitated more and more toward the audio consumer and record reviews so that that there is a hole in the audio engineering publications field. I wish you success.

I would like to offer a few comments on the article written by Mr. Norman Crowhurst entitled *Measurement Techniques and Standards*. Mr. Crowhurst described the proper system for measuring the gain and frequency response of an amplifier, using that from EIA Standard RS-219. This in turn was derived from a paper by W. L. Black and H. H. Scott which was in **Proceeding of IRE**, October 1949 and also in **Audio Engineering Magazine** for October and November 1949.

Mr. Crowhurst makes the point that the insertion gain as measured by the EIA Standard technique may not be the same as that encountered in some other application. This is indeed true, but the fault lies *not* with the measuring system but rather with the fact that the insertion gain of an amplifier depends greatly upon the value of the resistance of the source from which it is fed. This is readily seen from an inspection of the equation for insertion gain as given in the EIA Standard. If an amplifier is measured by the fundamentally correct method of the EIA Standard, and the value of source resistance is used as will be encountered in an actual system, then everything will come out right.

Furthermore, the EIA equation takes into account that the insertion gain of an amplifier is independent of the value of load into which an amplifier feeds.

Mr. Crowhurst has also encountered problems with measuring frequency response in that measurements in say a 500 ohm system (with 500 ohm source) do not correlate with performance in a system with say a 10,000 ohm source. This effect is perfectly normal, and is due to the very great effect of source resistance upon frequency response measurements. The answer of course is that meaningful measurements are always made from a source having a value the same or near that in an actual system.

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Mr. Crowhurst Responds

Melvin Sprinkle's comments leave me puzzled as to why his attitude seems critical. My article did not criticize the EIA standard – in fact I said it was the best that could be done toward a universally true statement. As I was a member of both EIA (then RETMA) and IEEE (then IRE) committees that evolved these definitions, I could hardly say otherwise! And Mr. Sprinkle does not deny that incorrect termination affects results.

But he seems to miss the main point to which I sought to draw attention: one does not operate any amplifier between an attenuator from a tone source and an output meter under such precise measurement conditions; one operates an amplifier between other audio units.

Under these circumstances, discrepancies occur that the user may not suspect. For example, he may connect a 500 ohm output to a 500 ohm input, and expect the insertion gains of the two units to add up. This assumes that each unit, as well as requiring 500 ohm termination, also presents 500 ohm termination for the other unit. If either does not do precisely this, the result will not fulfill expectations.

I believe, too, if Mr. Sprinkle has ever made such measurements, he must have found that deviation from correct termination will invalidate specified frequency response, sometimes at least, before he goes as far as his suggested use of 10,000 ohms for a 500 ohm circuit. In some cases I have encountered, varying the termination from a nominal 500 to 350 or 700 has been enough to put response beyond specified tolerances.

As Mr. Sprinkle has been in audio so long, he must have run into such discrepancies. I can't see how he could have avoided it. Maybe it's so commonplace to him, he doesn't realize that some, who have not lived with it so long, are apt to fall into a trap he learned about a long time ago.

My own background in audio dates from 1933. When I came to the States nearly 20 years later, this kind of thing was commonplace to me, I admit. But I found plenty of American audio men for whom it was not! Hence I feel my reference to such pitfalls was justified.
Norman H. Crowhurst, P.E.