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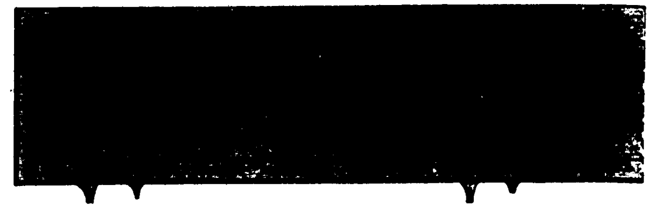
Why Do  Amplifiers Sound Different?

By NORMAN H. CROWHURST

Reasons for performance differences in audio power amplifiers having similar published specifications.

RECENTLY the opinion that the loudspeaker is the weakest link in the reproducing system and that amplifiers have progressed about as far toward perfection as it is possible to go has been widely expressed. As a basis for this conclusion, it is stated that the residual degree of various kinds of distortion present in modern amplifiers is so small as to be impossible to hear. However, many are not yet satisfied that this philosophy is true.

To illustrate this view, the following experience is by no means impossible or uncommon: two different amplifiers are compared, using the same pickup or tuner as a program source and the same loudspeaker. Both amplifiers, although of different design, use the same input and output impedances, provide the same damping factor for the loudspeaker, and give frequency responses and degrees of distortion which deviate by an acknowledged imperceptible amount—yet any discriminating listener can discern quite an appreciable difference between the sound of program played through the two amplifiers.

Why should these amplifiers sound different? A recent article on "Methods of Measuring and Specifying Audio Distortion" (August 1956 RADIO & TELEVISION NEWS) showed reasons why the same specified amount of distortion can sound different, according to the exact nature of the distortion, and pointed up the need for more precise methods of specifying such. This mostly related to the specification of distortion when clipping is involved.

But differences are noticed in the performance of amplifiers, even at levels well below the clipping point. For example, a trumpet recording is played through the two amplifiers and on one sounds quite clean while on the other there is a definite harshness about the reproduction. When the gain control is turned back the harshness becomes less noticeable, but only because the level is that much lower—it does not disappear completely, as one would expect if it were due to clipping, or an overload effect.

It became quite evident that something happens inside some amplifiers that is not adequately covered by the specifications. Incidentally, the amplifiers were checked on the same measuring equipment and both found to conform to their published specifications, which ruled out the possibility that one was not as good as it claimed.

Experimental Confirmation

Some work the author has been doing recently has verified two possible contributing causes for this kind of difference. From the results of these experiments it seems quite possible for an amplifier to perform to extremely close limits under standard test measurements and yet, with program material, the same amplifier can produce temporary or transient distortion conditions that are loud enough to be perceptible. Both these transient conditions are related to the nature of the roll-off characteristic produced by the feedback.

It is well known that, when you apply more and more feedback to an amplifier, a condition is eventually reached where the amplifier becomes unstable. This is due to the fact that, at some frequency, usually below or above the audio spectrum, the feedback becomes positive and causes oscillation. The frequency of this oscillation may be down in the region of 1 or 2 cycles or up in the region of 100 or 200 kilocycles, depending principally on which happens first.

Normally, of course, amplifiers are operated with considerably less than this amount of feedback, so they do not oscillate. Naturally, one would think that a margin of 2 to 1, or a little more, in this direction would be satisfactory to insure that the amplifier could not get unstable under any conditions. Many amplifiers have been designed with about this much margin.

This, however, overlooks certain fundamental facts that evolve from a mathematical consideration of feedback design. As this article is not written primarily for engineers, we shall refrain from going into the mathematics of such design. It is fairly easy to understand that, as we increase feedback, before the amplifier starts to oscillate, it will show a peak in the response, in the region of the frequency where it will eventually oscillate. The question is: how much must the feedback be reduced, below the amount which causes oscillation, before the peak is completely removed?

This is where the mathematics help some: in average amplifier design, we learn that the margin between oscillation and peaking, at the low-frequency end, is in the region of 18 db; while at the high-frequency end, it will be in the region of 12 to 14 db. These figures represent ratios of 8 to 1 and 4 or 5 to 1 respectively, both of which are considerably larger than the previously suggested margin of a little more than 2 to 1. These facts are illustrated in Fig. 1.

What Do Square-Wave Tests Show?

In comparatively recent times, the importance of an adequate margin at the high-frequency end has been realized. This was shown up at first by the use of square-wave testing. If there is any peaking in the amplifier response, or if the roll-off is too sharp, this will show up on a square-wave test as ringing at the corners of the square wave, as shown in Fig. 3. Many amplifier designers have, accordingly, paid attention to this feature and made adjustments to the amplifier so as to prevent this ringing. This means that high-frequency peaking must be absent from the amplifier.

However, there may not be the full 12 to 14 db stability margin, because the designers have used a trick to produce a satisfactory square wave: phase-shift capacitors associated with the feedback circuit. It's true that this method produces perfect amplification of the high-frequency end, for transients as well as steady tone, *when the amplifier is connected to a resistance load.*

Sometimes the designer has been careful to make sure that the amplifier performs reasonably well into a reactive load, but to make this test he uses for his reactance a *capacitance* across the output.

What seems to have been overlooked is the fact that most people use dynamic loudspeakers (woofers, squawkers, and tweeters) whose impedance becomes that of an *inductance* at the high-frequency end—and an inductance that gives a reactance somewhat larger than the nominal voice-coil resistance. This means that the amplifier loading is quite different from the conditions under which it is tested, as shown in Fig. 2.

The nature of the "finagle" used can be seen by a glance at the schematic: it has at least a "phase correction" capacitor across the feedback resistor, and probably has several other small-value capacitors (values given in $\mu\mu$fd., not μfd.) at various points in the circuit. This produces a satisfactory response with less than the basic 12-14 db margin, but because of this the arrangement is inevitably more critical of the correct loading on the amplifier output. This means that the use of the inductive loading provided by the loudspeaker voice coil results in a transient response which is probably worse than it would have been if the "finagle" had not been employed.

This fact accounts for the roughness in the high frequencies, observed with a number of amplifiers whose *measured* performance shows no trace of over-accentuation of the high frequencies, ringing on square waves, or distortion in this region.

Why the Struggle?

Perhaps a word is not out of place here, as to why this technique is employed. It arises principally from the current fashion for amplifiers to have a frequency response as near as possible from zero to infinity. Since zero to 20 cycles does not sound like a very big "piece," but 20 kilocycles to infinity sounds like an enormous range, the concentrated effort has been on the latter end. As a result, amplifiers have been produced with specified frequency response extending to 30, 50, 100, and even 200 kc.

While some of our high-fidelity cartoonists have suggested that such amplifiers are for the birds, this trend has generally been taken rather more seriously. Because of this, amplifier designers have been faced with the necessity of meeting specifications of this kind, dictated by the promotion or publicity departments of their companies. To get the amplifier to perform to these specifications, they have virtually had to resort to the kind of tricks we have mentioned, because the only alternative requires an output transformer whose price would be prohibitive.

What About the "Low" End?

So much for the high-frequency end. The low-frequency end seems to have escaped attention although, as we found, its effects can be disastrous with some kinds of program material.

Most amplifiers probably have a stability margin at the low-frequency end of at least 2 to 1, or 6 db, and probably as much as 12 db. But, to avoid any peaking effect at a subsonic frequency, they need a margin in the region of 18 db. Unfortunately this peak does not show up in the measured response, because it occurs in the *loop* gain characteristic and may not show up at the amplifier output, due to the low efficiency of the output transformer at this frequency.

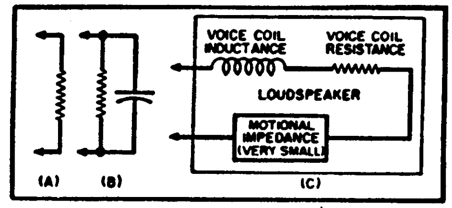


Fig. 2. (A) Common load used for testing although (B) is occasionally used. (C) Actual load offered by speaker to amplifier.

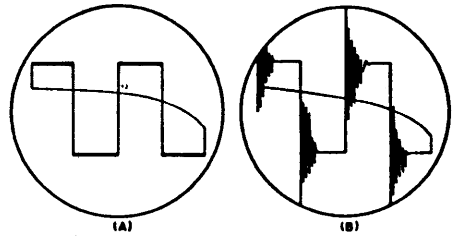


Fig. 3. (A) Good square wave applied to input and seen at output of very good amplifier. (B) A more common output waveform.

That is a rather technical distinction—just what does it mean to amplifier performance? A peak in the response anywhere means that any transient condition can cause the system to ring at this frequency. If the amplifier has any kind of peak in the region of 1 or 2 cycles, a transient condition can cause the amplifier to produce a kind of low-frequency flutter of this frequency, which may take a few seconds to die away. But what kind of transient would do this?

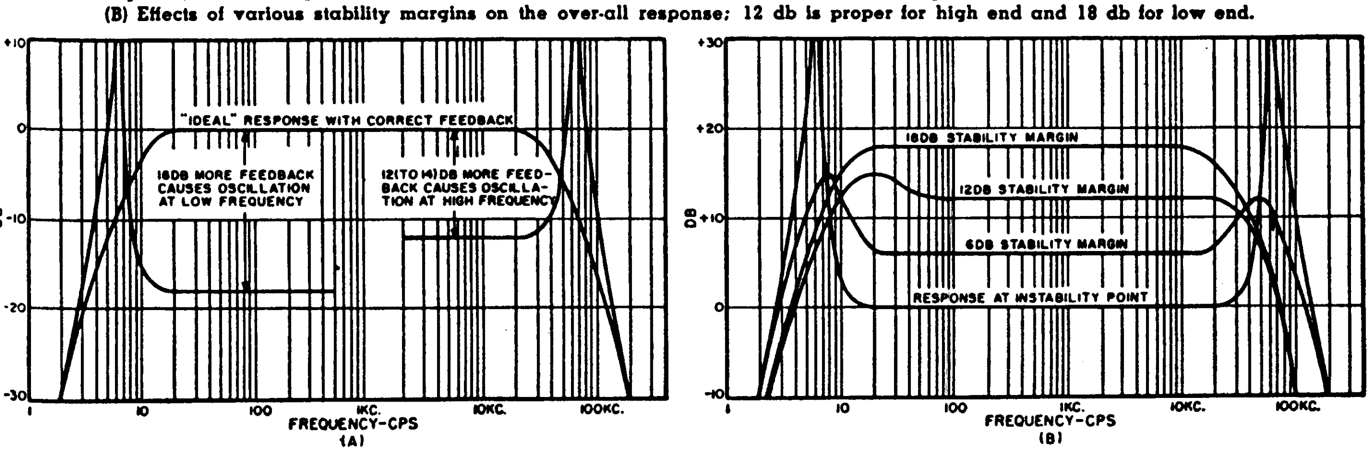
What Is a Low-Frequency Transient?

The frequency of ringing is down at one or two cycles, so the normal transient, with a sharp wavefront, will not necessarily cause this kind of ringing. The waveform that will produce it is one that possesses a momentary d.c. component. Many of these occur in practical program material.

For example, the trumpet waveform we mentioned earlier is quite asymmetrical; this means it is equivalent to an a.c. waveform, with a number of component frequencies, plus a d.c. component which offsets the waveform on one side of zero. This probably occurs due to the fact that the instrument is

(*Continued on page* 94)

Fig. 1. (A) Ideal response when the feedback is correct; part curves show instability points as feedback is increased. (B) Effects of various stability margins on the over-all response; 12 db is proper for high end and 18 db for low end.



blown and the air coming out constitutes a d.c. component. When a stringed instrument, especially a string bass, is plucked, or a percussion instrument is played, these, too, produce a momentary deflection of the waveform one way or the other from the zero line at the start of the tone.

Thus it can be seen that any of these kinds of program material can initiate the low-frequency ringing we have described.

So What Happens?

In the old-fashioned kind of amplifier, without feedback, this kind of program material will produce a momentary change of bias on each stage through the amplifier. The time taken for each bias to change will depend on the time constant, as it is called, produced by the coupling capacitor and the associated circuit resistors. In other words, a continued trumpet tone will cause the bias on each stage to re-adjust itself by some fraction and each stage will take a moment or two to settle down to its new bias value. This is illustrated in Fig. 4. The time taken for each stage to settle down will be dependent upon the time required for the coupling capacitor to change its charge: larger capacitors will take longer and smaller ones will allow the change to take place more quickly.

In a non-feedback amplifier all these changes will take effect at so slow a rate that they will not contribute any audible difference to the sound of the output. But when feedback is applied to the amplifier, all these time constants interact so as to make the amplifier *almost* into a low-frequency oscillator. It does not quite oscillate, otherwise the amplifier would be audibly unstable, but any of these transients coming along will set it into a momentary state of oscillation, which takes a few seconds to die away.

The oscillation itself is not audible, because it is only at 1 or 2 cycles and the output transformer prevents any appreciable voltage at this frequency appearing across the loudspeaker voice coil, also the loudspeaker does not produce appreciable response at this very low frequency. However, the low-frequency fluctuation occurs at measurable amplitude at some point *inside* the amplifier circuit itself.

The asymmetrical kick given by the program waveform can set up an oscillation twice as big as the effective d.c. component. This means that quite a large fluctuation can occur inside the amplifier which will not be audible outside of it.

Effect on Program

So why does it cause trouble? Because the gain of every stage in the amplifier varies with operating bias. This low-frequency fluctuation is like a periodic changing of the bias of several stages through the amplifier. Consequently the program material gets modulated at this low frequency. What we hear, then, is due to an intermodulation of the program material by this low-frequency oscillation.

If the feedback were not present (which, of course, is an impossible state to imagine, because the feedback is what is causing the oscillation), the effect most noticeable would be that the whole program would sound as if an electronic tremolo had been added. However, the presence of a large amount of feedback stabilizes the gain of the amplifier so the tremolo effect is not noticeable.

Instead, the same intermodulation that would cause a tremolo effect, but for the feedback, produces a much larger amount of IM distortion in the amplifier than occurs under static measurement conditions. This results in the harshness often observed in modern feedback amplifiers.

How All This Was Proved

These observations are not just the result of theorizing. To substantiate this, two amplifiers of conventional design were taken and modifications made to bring their designs into line with the established mathematical theory, giving the required stability margins at both ends of the frequency response to avoid peaking under any circumstances.

These changes resulted in a slight deterioration of the frequency response, but in neither instance did the response drop below 1 db at 30 cycles or 15 kc., which is still considered to be high fidelity. It is doubtful—extremely doubtful — whether a difference of 1 db at either 30 cycles or 15 kc. could possibly be heard "for itself alone." *A-B* checks were then conducted between the amplifiers, using their original circuits and the revised feedback circuits.

A difference was quite noticeable in the reproduction of program material, particularly with the kinds of program material in which, as has been discussed, there is asymmetrical waveform—when wind instruments are playing, or string instruments are played by plucking. These experiments certainly seem to have uncovered at least some of the major differences that can exist between amplifiers with equally good specifications—differences that do not show up, at any rate, in the standard method of specification. These are, in fact, defects that are not in the book! —30—

Fig. 4. (A) Asymmetric wave without isolating d.c. (B) Offsetting bias adjustment.