AUDIO UPDATE

A Distortion Primer—Part 2

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The most well-known of the "non-standard" distortions—transient intermodulation (TIM)—achieved prominence in the early 1970's, mostly through the work of Matti Otala, a Finnish engineering researcher. He rediscovered that under certain conditions, amplifiers using high levels of overall output-to-input (or *global*) negative feedback would experience input overload even though the applied signal was theoretically too small to cause such a problem. I say *re*discovered, because the effect had been documented, discussed, and solutions outlined in the early 1950's.

In a nutshell, the TIM story is this: A rapidly changing audio signal—meaning one with high-frequency components—would overdrive a feedback amplifier's input stage, while a signal of the same amplitude, but without high frequencies, would go through without problems. The overload occurred because the amp's input stage parameters assumed operation with the gain reduction of negative feedback, but the feedback signal did not get back to the input fast enough to prevent overload. A basic solution to TIM is to design for sufficient amplifier bandwidth—*before* feedback—to ensure that high-frequency signals can *slew* (travel fast enough) through the amplifier to avoid problems. Today, competent engineers can easily achieve adequate slew rate—and thereby forestall TIM—without really straining very much of their design talents.

Marketing esoteric distortions

If, as I (and others) claim, the distortion problem is essentially trivial in today's amplifiers, why all the technical papers and amplifier advertisements touting recently invented/discovered varieties of distortions and their cures? As far as I can tell, TIM and other similar obscure amplifier problems appeal to two groups: engineering academics seeking scholarly publishing credits and, especially, manufacturers and their advertising agencies.

Audio manufacturers, in an annual effort to differentiate their new products from those of their competitors, regularly discover—and eliminate—previously unrecognized sources of distortion. That reflects the need to ascribe special audible virtues to products that are really very good but, in truth, are no better than those of their competitors. If sales of a manufacturer's latest models can be enhanced by incorporating a newly developed lateral-feedback circuit to eliminate recently discovered problems of side-slip distortion, why not go for it!

In regard to those dedicated listeners who continue to hear problems or special desirable qualities in certain amplifiers, I would be willing to bet an extremely expensive set of diamond-encrusted tweeter cones that what they are hearing, for better or worse, has far more to do with minor frequency-response deviations than with any kind of old, new, or yet-to-be-discovered amplifier distortion mechanism.

Loudspeaker distortion

No one argues that loudspeakers are anywhere as distortion-free as amplifiers. In essence, a loudspeaker system is required to convert the electrical audio waveform supplied by an amplifier into an analogous three-dimensional acoustic waveform. Considering that loudspeakers do their jobs using an assortment of driven diaphragms vibrating in special boxes, the wonder (as someone once said of a chess-playing dog) is not that it does it well, but that it can do it at all!

The basic loudspeaker problem is linearity of transduction. That means that the speaker cone (and the voice coil that drives it) must move in exact accord with the audio input signal. Voice-coil motion constrained by its suspension or operating in a magnetically nonlinear portion of the voice-coil gap will generate large amounts of second- and third-order harmonic distortion. And any voice-coil movement not coupled accurately to the cone, and any cone movement not directly controlled by the voice coil, will distort the sound in some way. It's clear from test data and careful listening that in the last 15 to 20 years driver design and performance have improved dramatically.

Assessing distortion

About 15 years ago, I found myself on a business trip in England visiting the Rank HiFi speaker research and manufacturing facility. An unexpected bonus of my visit was a day-long meeting with Dr. Peter Fryer, who was then deep into an investigation of speaker distortion and all its ramifications. One of his primary objectives was to assess the audibility of the types and levels of distortion commonly produced by loudspeakers.

Since IM distortion, as mentioned last month, is generally thought of as one of the worst culprits in making things sound bad, Fryer tackled it first. A distortion generator was built that could be set to inject a calibrated amount of IM ranging from 0.1% to 10%. A virtually distortionless amplifier and speaker system were designed and built and served as the "test bed" for all of the subsequent experiments.

Much to everyone's surprise, it was necessary to crank up the IM level to 5% or 6% before it became audible on typical complex musical material, rock or classical. With simpler music, such as a solo piano, 2% IM was clearly audible. And, when sine waves were used as the test signals, IM levels of 0.1 percent could be detected under carefully controlled conditions. None of that surprised me, since the findings neatly replicated the results of some amplifier IM tests that I had been involved in four years earlier.

Fryer's next series of tests involved kinds of distortion *not* found in amplifiers. *Delayed resonance* was something that had concerned the British for years, but to my knowledge was never an issue among U.S. speaker designers. Simply described, it is the tendency of parts of a driver's cone assembly to store energy and continue to release it for some milliseconds after the original signal has ceased. It was described to me (you'll have to imagine the British accent) as "the speaker carrying on broadcasting long after the program has finished."

Unlike IM tests, the delayed-resonance research results were not easily summarized in numerical form. A basic finding was that broad low-Q resonances were far more audible than sharper peaks covering narrower bands, probably because the low-Q resonances were activated for a greater proportion of the signal. When the peak is very broad and low (a Q of less than 1), the audible effect is simply an increase in level over the affected portion of the frequency band.

The primary finding of the research was that it behooves the speaker-driver (and speaker-system) designer to eliminate resonances whenever possible, a task that has been significantly facilitated in the past decade by laser analysis of cone movement and modal vibration analysis of speaker-cabinet walls.

Doppler distortion

A distortion that excites partisan bickering among speaker-system designers, *Doppler distortion* occurs when a cone is undergoing large low-frequency excursions while simultaneously reproducing high frequencies. The theory is that the high frequencies reaching the listener's ears will be alternately compressed and stretched by the low-frequency movements of the cone. There is no question that speakers do produce Doppler distortion. The real question is: How audible is it under normal playing conditions?

A Doppler distortion generator was developed using a delay line that could be varied at a rate determined by a low-frequency signal. A total voice-coil movement of more than two inches could be simulated, certainly more than enough to simulate any real-world condition. Fryer summarized the results of days of experiments by saying that, with the possible exception of small full-range speakers (such as are found in portable radios), normal loudspeaker systems used in the home will never produce enough Doppler distortion for it to be audible.

The bottom line

As with amplifiers, most of what we hear going wrong in hi-fi speakers is in the frequency domain. However, straightening out an amplifier's frequency response is duck soup compared to the task facing a speaker designer. If an amplifier's response has a bump or a dip, a few resistors and capacitors will usually flatten it nicely. Speaker frequency response, on the other hand, involves manipulating magnetic, mechanical, and acoustic variables, in addition to the electronics of the crossover network. But improved materials and know-how continue to make the task immeasurably easier. And anyone who has done any comparison listening in the past dozen years knows that speakers continue to improve. **R-E**