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Current Drive for Dynamic Loudspeakers and the Measurement of "Power" Distortion.

by Kurt Steffensen, edited by Bill Perkins

Gentlemen:

I wonder why no one has commented on the recent article by Mr. Ragnar Lian. When I read it, I was excited and awaiting a lot of good input on the matters raised therein. It is some of the best text I have read for years. But maybe no one is really interested...

I carried out experiments some 8 years ago that led to the same conclusions; that made me develop a new method of measuring distortion. As they are rather controversial, I have not published the results yet but intend to do so within a book I am writing. (Though I might never finish it or even persuade anyone to publish it.)

I am writing this as it comes to me as I do not presently have time to edit the text.

Some 8–10 years ago I was, like others, wondering a lot about why some valve amplifiers that measured several percent of THD seemed to reproduce music a lot more naturally than did solid state amplifiers measuring 0,01% or better. This was especially evident with power amplifiers.

In order to do listening tests on amplifiers with different degrees of distortion, I introduced more or less global, negative voltage-feedback. Some of these amplifiers were already good enough to measure less than 1% THD and had frequency responses of 2-100KHz without NFB.

The subjective results were that the lower the NFB, the better and more lifelike the sound but the higher the THD!? In this respect, it did not matter if the amplifiers were valved or transistorized.

Obviously, something had to be wrong with the THD measuring methods. The true distortion of the amplifiers without NFB had to be lower to make any sense or else some weird psycho-acoustic phenomenon was consistently tricking us but I did not believe that to be the case. Many other people, including those not interested in "hi-fi" at all, heard the same things.

The human brain and hearing system have developed over at least a million years in the arena of real-life. Even the most sophisticated signal analyzers are simple toys compared with this. I became convinced that we were measuring the distortions wrong-mindedly. This was not a new thought, as others before me have taken up these same issues. But I intended to find out why or how we had, and continued to err. To foreshorten a long story, I believe that to a useful extent I succeed in moving towards a better, more coherent view.

To start; how do we measure distortion? Most commonly, we apply a continuous sine wave to the input of an amplifier and from its output we reject the excitation frequency with deep notch filters, measure the remaining "noise" and divide the constituent frequencies by the frequency of the original to yield the "order-number" of the individual harmonic components; 2nd, 3rd and so on

Music is by nature an unpredictable and extremely complex signal that will almost certainly be shown to be chaotic. Such simple, sinewave tests can hardly be expected to reveal the sonic qualities of amplifiers reproducing music.

Accordingly, a far better approach would be to use some complex signals, adjust the input and output signals to the same amplitude and subtract one from the other. Ignoring phase effects, the difference signals are distortion products. Such methodology would allow us to measure amplifier distortions with true music signals.

In real life however, this is not possible, due mainly to the phase relations through the entire circuit under test.

Nevertheless, it is interesting that this is the principle upon which the reduction of distortion by global feedback is based and I soon began to consider THD as an expression of simple "dynamic noise" or just small differences from ideal working characteristics that mean little to the reproduction of music signals.

If THD is considered as such a "noise", it is easy to comprehend why these figures can not be relied upon to show much about the merit or quality of an amplifier intended to reproduce music. But THD measuring methods using only sinewave tones seem to tell the whole story about the amplifier's ability to reproduce the specific sine tones that are analyzed. That sounds reasonable. The reason for a little doubt is that I later succeeded in building a single sine frequency with two different sines. (Inspired by "Cathode Ray" in WW - does anyone remember him?)

The common theory is that all complex signals can be resolved into a series of single, constituent, sinewave tones, the Fourier series. I agree with that approach only if the signals indeed are continuous.

It's likely not reasonable to take up any lengthy discussion of this here as it is not particularly relevant. But it does suggest that the theory behind this commonly employed method does not yield the full picture even for the simplest signals of all—the single-component sinewave. I bring this forward as another argument for the consideration of the THD method as merely a revelator of a kind of "dynamic noise". Many other aspects of an amplifier's performance more accurately indicate its ability as an accurate and, more importantly, *satisfying* reproducer of music. At this point, I would like to say that my experience has convinced me that once a distortion product accounts for less than $\frac{1}{10{,}000}$th (–80dB) of the output power of a given amplifying circuit, that contribution can safely be considered negligible.

I made two inverting, amplifier stages that produced large 2nd harmonic products. But when connecting these in cascade, the resulting THD was very low indeed. These two stages were identical in every way, I simply twice repeated the distortion or more accurately perhaps, the non-linear transfer curve.

While I understand the reason for this outcome, I simply did it to prove that two heavily distorted single-stage amplifiers can cancel the simple, sine distortions while leaving the complex distortions intact; distortions that traditional THD measurements cannot reveal.

I then re-made this two-stage amplifier—using the same tubes—but now arranged to yield much lower levels of 2nd harmonic output within each stage. When the two stages were cascaded in the same way and adjusted to the same level of THD as before, the sonic improvement was substantial. Another proof that the traditional THD method must be inadequate.

But here comes the doubt… When measuring THD in the traditional way on a single stage or a whole amplifier *without* global or any other voltage feedback, the subjective and measured results compared very well. This was only evident however in voltage amplifiers whose load was almost purely ohmic, with no reactive components being driven. Later, I will show that this is yet another proof of my "theory."

Low distortion gave high sonic quality and vice versa. That implied to me that well designed amplifiers also measured well the traditional ways. And that made sense: a good amplifier for complex signals would naturally reproduce simple sine waves very well.

I now began to investigate the relationship between measured results taken at the loudspeaker output and the subjective impressions of these. Comparing amplifiers that had a frequency response linear from a few Hz and way up into the 40-200 kHz region gave subjectively very different results. Some had *lots* of energy in the high treble while others sounded as if they rolled off by several dB up top or yielded perceptual lumps and dips in other parts of the spectrum. That seemed to be a good way to start, leaving all the complex distortions out of the picture.

Why did amplifiers behave very differently to the ear in terms of perceived frequency response when they all measured ruler straight in this regard? The first thing was to get rid of the 8Ω "test" resistor and do some measurements into a "real world" load; ie. a loudspeaker.

When starting these tests, the amplifiers that previously behaved in textbook fashion began to measure a little differently but not enough to really prove anything. Again, the more global voltage-feedback, the greater the perceived deviation from the ruler straight measurements.

I then obtained some fancy Brüel & Kjær equipment and measured the acoustic frequency response from the amplifier/loudspeaker combination. These results compared well with the subjective results and at this stage, things began to show a pattern: the more feedback, the worse the acoustic frequency response.

An idea had been brewing for some time in my head: we are, in fact, listening to the developed *energy* from our loudspeakers! The thing that drives a loudspeaker voice coil to move within its magnetically "illuminated" voice coil gap is *energy.* All the results that I have referred to up until now were derived thru the common practice of taking only *voltage* as the reference; all THDs, IMDs, TIMs etc. are taken this way. But without taking current into effect, voltage is a static factor. Only when held against the *current* is the *energy* picture clear. And, with voltage held constant, the phase, direction and magnitude of the current is entirely dependent on the reactance of the load. Current is what drives a dynamic loudspeaker and the developed energy is expressed in terms of real amp turns, ie. current that is in phase with the voltage developed across the turns in a driver's voice coil.

I then constructed a current sensing circuit that enabled me to convert the flow of current to a voltage signal that was, of course, viewable on an oscilloscope. It was exciting! Now I could see things happening at the amplifier's output that I had never seen before. A happy day that was…

Those signals compared a lot better to the subjective results. Now, feeding both the voltage and the current signals to a circuit that amplified these in relation to one another, a four-quadrant multiplier, I obtained the energy signal, I x E = W. (current x voltage = watts which equal *energy)* That energy signal related directly to the acoustically measured frequency response and to me this was a breakthrough! Yet, even though my amplifiers were better than ever, this discovery was somewhat anti-climactic as I could not draw any significant commercial advantage from the "discovery".

Many loudspeakers lack bass when driven from amplifiers with no NFB and this observation confused me for some time. But after reading Mr. Lian's article, I was fully confirmed in my thoughts and it all fits perfectly… As he says, loudspeakers are developed with voltage sources as the reference. Naturally, attempts are made to make the loudspeakers as linear as possible in relation to the constant *voltage* inputs and this leads, inevitably, to the confused and confusing state of affairs we've witnessed for decades. Nevertheless my project let me understand a lot of things much better.

Now, as my experiments show, there is no point in measuring the voltage response at the amplifier's output as this does not show much about the true signal. Neither the one actually present, nor the one needed. Measurements of the *current* signal come a great deal closer to the truth. And, somewhat paradoxically, such signals do *not* have to look like sine waves at all in order to reproduce with low distortion. This, in spite of the fact that the amplifier is feed with pure sine waves. The fact is that the voltage signal must follow the dynamic impedance of the loudspeaker, with good amplifiers consequently producing the same VA product into any load. But the signal that should really be investigated here is the energy signal, the current signal times the voltage signal. Well, it is a little more complicated than this due to the phase relationships and the fact that the loudspeaker also is a current source supplying the amplifier's output with reactive currents to be "sunk." Hence, the dynamic impedance of a loudspeaker as seen from the amplifier can easily range from several megohms to a negative impedance! Surely measuring voltage THD at the amplifier's output makes no sense. The wattage method does a much better job.

As with everything else in electronics, things can be viewed from many different vantage points The dynamic relationships that exist between loudspeakers and amplifiers goes beyond my imagination, they are unbelievably complex; in terms of load presented to the amplifier, what *is* a loudspeaker when music is playing?

There are many ways to look at this of which the Thiel and Small method is only one. As a dynamic system, too many unknowns are involved. It is hard to even define a reference and even harder to find any signal to compare it against and this can put use right back with trusty old sine waves. But if we measure the sine *power* at the output and use *that* for the THD, we are a great deal closer to a useful figure, one that relates directly to the subjective results.

While that still does not tell us all about the amplifier, difference tone distortion measurements, pulse responses and perhaps some squares or a constructed complex signals that can be reproduced, might bring us as close as we will ever come to being able to "objectively" quantify an amplifiers performance.

However when measuring THD using this method, most valve amplifiers measure quite a lot better than their solid state brethren. And just as voltage feedback is increased the distortion is also increased. That is the opposite of the traditional method. However global current feedback, was able to lower the distortion and these experiments, compared closely to the subjective results. The better the amplifier was to begin with the less did it favor feedback and the less it needed it. Poor amplifiers such as those running class B and/or ones with poor frequency response, noisy ones or ones with poorly chosen working characteristics etc., gained a lot with feedback, even if it was the voltage feedback. The Power Distortion method I have described also explains why some amplifiers sound good only with certain speakers and vice versa. It is also evident that a certain speaker cables will act differently within various combinations of equipment. I presume to conclude that it is only possible to claim an amplifier's quality or merits in combination with the particular loudspeakers with which it is judged. It might behave quite differently with other speakers...

To a lesser degree, this applies to valve amplifiers and amplifiers without global nv-f/b. This is due to their higher Z-out making them seem a little like constant-current amplifiers. Also, the low damping factor possibly plays a positive role. My conclusions were that true voltage amplifiers such as valves and FETs should be measured as such except at the amplifier/loudspeaker interface, while bipolar transistor amplifiers should be measured as current amplifiers, also at the amplifier/loudspeaker connection.

Voltage feedback taken the load or loudspeaker output terminals does not reduce distortion unless the amplifier is rather poor to begin with. It simply introduces a new "random signal" that might make things a lot worse. Current feedback from the loudspeaker output is generally a good idea however the greatest cautions should be taken due to the group delay and phase shifts.

To me it seems better to do the amplifier well enough in the first place so that no feedback needs to be provided. Local feedback, that within a stage does not seem to do any harm, but then, due to its nature, neither does it do much good.

I call the phenomena that I have unveiled "Power Distortion". All the traditional methods can readily be adapted to my method and could be then simply: THPD, or Pulse Power response, Power-Frequency, etc. As can be seen, these results of my method fit perfectly well with the conclusions of Mr. Lian even though we started from rather different directions. This can hardly be a coincidence and I believe I have demonstrated the value of both our viewpoints. My results can be reproduced with very little equipment, in any lab. If you want to reproduce my experiments but lack a frequency-linear current-sensing device, you may purchase one from me for $US25.00 These are linear from DC to 30KHz. and you will need to provide both a PSU and a subsequent voltage amplifier. The one I have developed for myself is linear from DC to 30MHz.

I wrote this post, shortly after reading Mr. Lians article, published by Thomas Dunker, but I have hesitated to mail it as I would have kept it as one of the chapters in my book. But after these last few days of posts, it just seems to be as good an occasion as any; so I decided to publish it now.

I guess I can regard this as an "acid test" and I hope that a good, fruitful debate is ahead. I don't know if this is the right forum though as no one has replied to Thomas' excellent translation of Mr. Lians posts. If you have not read that post… do so. It is *exciting.*

Sincerely,
Kurt Steffensen