Letter to the Editor

Burkhard Vogel writes:

Dear editor,

I've studied Hans van Maanen's Guest Editorial and Ian Hegglun's interesting article in Linear Audio Volume 4 and very quickly stumbled over Ian's statement that "…. no method has become popular with power amplifier designers and reviewers so far".

His statement seemed to be true on the face of it. Generally, I like to study competing theories à la Kant's advice that "there is nothing more practical than a good theory" – and then Occam's Razor will take over. It motivated me to write this Letter.

I'm not an expert in distortion matters at all and I know only a tiny part of the world-wide existing literature on that specific issue in depth, mainly the British view on it. However, I also know that 27 years ago Mr Johannes Maier from Stuttgart, Germany, published an article about his theory on the correlation between listening test results and amplifier measurement results. Not until 24 years later would this theory lead to practical applications because first the corresponding measurement approach and equipment had to be developed and tested.

During the past 3 years it was Mr Peter Schüller, the head of Stuttgart, Germany, based TESTfactory, who invented the final test approach that allows qualifying an amplifier in a way that gives a clear signal to the user whether it is free of audible add-ons (tendency: short piece of wire with amplification) or less free (tendency: creates a specific sound that adds additional information to the original signal). This would be valid for any kind of output load - purely resistive as well as complex (like a loudspeaker simulation or a real loudspeaker). Or, in other words: the measurement approach allows qualifying whether an amplifier measurement result will fully correlate with the listening test result or only part wise or not.

Shortly and in my words, the most important content of the MST (Maier-Schüller-Theory) could thus be described as follows: In addition to some other and minor important points the structure of sequence, order, and level of the nonlinear harmonics of **all** frequencies in the whole audio band play a major role in the reconciliation of listening test results and amplifier measurement results. These results are displayed by a specific measurement tool and thus can easily be assessed.

This MST was firstly presented at the Munich 2012 High-End Fair's expert forum (in German), and some weeks later on the Burosch website (also in German). Burosch is a world-wide operating company that offers a broad range of test signals for all kinds of video and audio codec purposes; this company also runs many websites that offer an additional selection of white papers on various kinds of video and audio issues.

Unfortunately, the whole lecture cannot be presented in this Letter because the many slides are all in colour and cannot be changed into the Linear Audio black and white format. However, I could convince the two inventors to formulate the MST in a way that could be published in Linear Audio in the near future – including additional findings that could not be shown at the fair.

For readers who want to acquaint themselves immediately with MST I've arranged with Mr. Schüller to translate the text of his lecture into English; this can be found at:

http://burosch.de/audio-technik/509-high-end-2012-klang-2-english.html

One of the many advantages of MST is the fact that all measurements can be performed rather elegantly by application of a top measurement instrument like the AP 2722.

I also think that Hans van Maanen's concerns and doubts about the selection of a listening team can be scattered by the two inventor's experience (together with a well known group of other long-time experienced helping colleagues). A work life long they've performed hundreds of loud-speaker and amplifier listening tests and measurements. I guess these engineers clearly know what they are talking about and there is no need for them to learn how to listen in an anechoic chamber (they have one) or in many different kinds of living rooms, nor how to listen as a scientist or as an average person.

It is very rare to find audio equipment manufacturing-independent people in the audio test community that did not change their test company during 25 years, and as a result this company has collected a huge amount of knowledge about all kinds of audio issues. By diving very deep into the loudspeaker/amplifier problems, TESTfactory is highly recognized as independent inventor of new and challenging test methods - obviously only known in Germany, as proven by Ian's literature list.

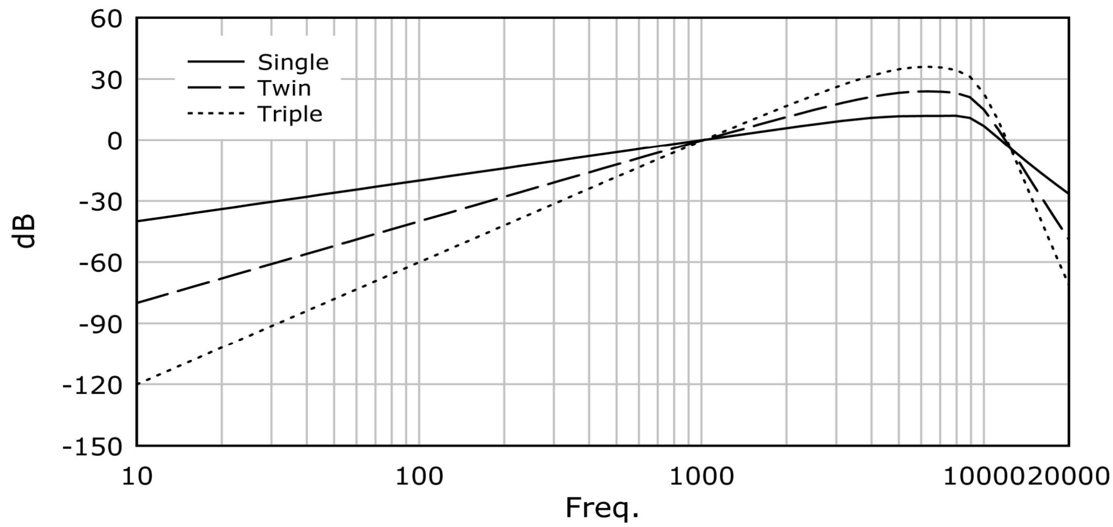
Burkhard Vogel
Stuttgart, Germany

PS. I know very well the many traps of the sub-editing process. That's why I kindly ask Mr Hegglun to check again his Fig. 6. I think it's wrong because Fig. 5 is wrong. His source [31] will give the right CCIR Standard based component values. Another source is the ITU-T J.16 paper.

Ian Hegglun replies:

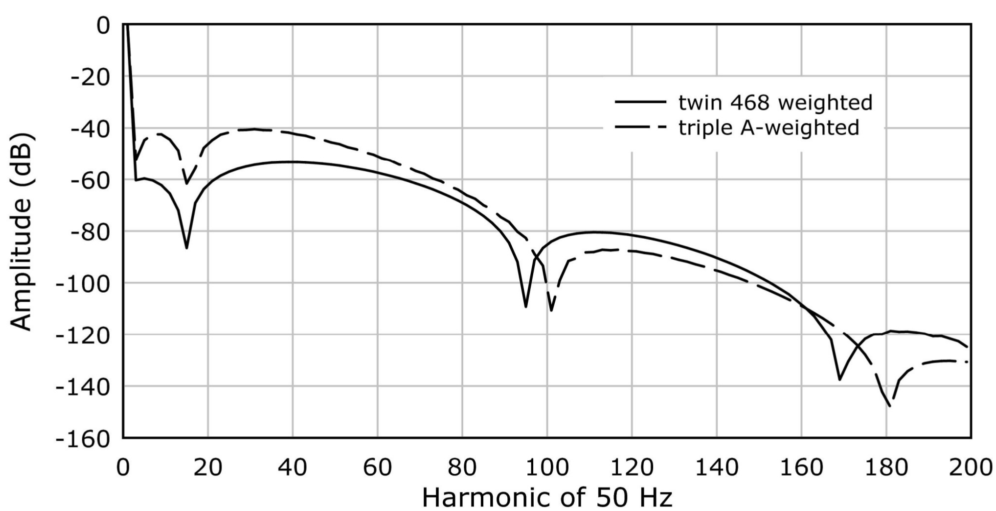
Thanks Burkhard for spotting my typo in Figure 5 circuit where L1 should be 12.88mH and C2 26.82nF. My apology. The corrected Figure 6 appears below.

Fig 6. CCIR-468 single, twin and triple filters



This changes Figure 10 slightly. The corrected Figure 10 plot appears below:

Fig 10. Voltage follower Triple-A and Twin-468 filters



One sentence on p43 needs a slight change: "...The twin 468 THD is 1.3% which is close enough to 1.9% to still be useful." should read "The twin 468 THD is 0.92% ..."

I consider the same conclusion can be made using the correct figure of 0.92% and the article is not significantly affected by this typo.

After pondering on whether it is better to use Twin-468 filters or Triple-A the Twin-468 filters consistently underweight all the harmonics by a factor of two whereas the Triple-A overweights only the first 12 harmonics (see added plot). This would allow a scale factor of two to be applied to the THD reading with Twin-468 filters to get close to an exact weighting and1/3rd less components to boot. Scaling the Triple-A reading is not recommended because harmonics above the 12th become underweighted.

On the body of Burkhard's letter - I was interested to read the translation of Peter Schüller's talk first presented at the Munich 2012 High-End Fair on the Johannes Maier and Peter Schüller Theory (MST) on the special distortion impact on amplifiers (sublink http://burosch.de/images/Schueller-lecture-03.pdf).

This raises the interesting question of what is the best harmonic structure for distortion if we cannot make it inaudible. The topic of euphonic distortion, which I did not mention in my article or references, is also a commonly raised.

An underlying assumption of my article is that a threshold exists, a level where harmonic distortion can no longer be heard. So a designer can use weighted THD measurements to try get the figure down to where it does not matter what the harmonic distortion spectra looks like. BTW Section 8 of my article showed how that can be done for a novel Class-AB topology.

I suggested a factor of safety be applied to ensure harmonic distortion is inaudible. We can still detect subtle differences even if the harmonics cannot be heard. Dr. Hans van Maanen in the guest editorial of Vol.4 mentions the masking thresholds used for MP3 are insufficient for many listeners. Robert Stuart had suggested a safety factor of around 20 and I was hoping more research will be carried out to arrive at a practical value for this for the audio industry.

How does this relate to MST? My approach and MST are mutually exclusive. If harmonic distortion can no longer be heard or detected then we do not need to be concerned with the harmonic structure of a power amplifier. Applying MST to amplifiers with distortion below the threshold of audibility may give us false negatives. It would be good to see more work in this area with alternative approaches.

The MST approach is mainly relevant to amplifiers where the distortion is high enough to be audible such as low or no negative feedback power amplifiers, to select or design for minimum unpleasant effects.

Interestingly, my article's reference 4 by Rupert Neve suggests that the 7th harmonic is the first harmonic number that needs to be made inaudible for a good amplifier because it is the first harmonic that does not form a standard musical interval. If Neve and MST are both correct then the 2nd, 3rd, 4th, 5th and 6th can exist at small levels with *minimal* deleterious effects provided these harmonics reduce monotonically and the 7th is below the threshold of detectability, say 20 times below the audible threshold. Maybe we can call this Neve+MST?

Neve+MST could explain why "...a sizeable segment of the audiophile population“ like SETs (Nelson Pass p77 Vol.4) – because the harmonic structure decreases monotonically and the 7^{th} harmonic is well below audibility with typical music material.

On euphonic distortion. The original work by Jean Hiraga on the harmonic structure of power amps was presented in English in the Hi-Fi News March and April 1977 and a critical review can be found in Stereophile "Euphonic Distortion: Naughty but Nice?" by Keith Howard (at http://www.stereophile.com/reference/406howard/index.html). Keith Howard points out that Jean Hiraga is claiming that particular patterns of distortion actually enhance fidelity, now called euphonic distortion. But is there any hard evidence available to support this claim?

Keith Howard provided various synthesized distorted files for listener tests on a web site that aimed to meet Hiraga's criteria. But positive test results have not been reported, at least that I am aware of. This does not prove that it can't be done so we are still left waiting for a positive result.

A positive result has profound implications. Some suggested that if a recipe can be found then euphonic distortion can be added using DSP to any amplifier and possibly at any stage of the reproduction chain. If euphonic distortion exists then amplifiers that are made 'bland,' because they generate no audible distortion for example, then they too can be brought back to 'life' again by adding euphonic distortion.

But in the mean time many designers including myself aim to design amplifiers that add no audible distortion, and as a fallback, to choose the best harmonic structure for the distortion that we cannot make inaudible.

Finally, I noticed in the Peter Schüller's talk that the AP analyzer 2722 appears to now remove the noise component from the THD plots so the distortion flattens out at low signal levels, eg p28 slide #27. Previously 'THD+noise' plots were generated that would rise at very low power giving a misleading indication of distortion at very low power. At last we can have distortion-only AP plots at low power levels in publications and reviews. Fantastic!

BTW in the Peter Schüller's talk p28, the text appears to have a typo in the English translation, where "k5 (red)" I think should read "k5 (blue)".

Ian Hegglun,
Australia

Hans van Maanen replies:

I have read the comments of Burkhard Vogel with great interest and I think that I have to clarify one of the statements in my Guest Editorial (Linear Audio Vol 4). When I made critical comments about the listening teams, I tried to elucidate that -in my view- gathering more or less randomly a number of people is insufficient to base far-reaching conclusions on e.g. the ability of human hearing. This did not exclude the possibility that very experienced listening teams do exist. But at e.g. AES conferences I often hear at presentations that the listening team was just a number of students and although such young people will mostly have good ears (although I fear the I-pod!), they usually lack experience. It took me tens of years with frequent (around 30 classical live concerts / year) visits to the Amsterdam Concertgebouw to get (I hope) a useful data-base of listening experiences. Including the problem that no existing audio system meets the same quality criteria as human hearing, I take the results of “scientific” listening test as described above with a large number of grains of salt. The MP-3 disaster, in which lossy compression techniques are used which were “scientifically proven” to be inaudible, is in my view a clear example of the incorrect conclusions, drawn from such listening tests.

I also read the presentation of Peter Schüller, referred to in the letter of Burkhard Vogel. Although I welcome any development to further improve our understanding of the relations between measurements and listening tests, I think the presented theory is incomplete and does not explain related problems. First of all, harmonic distortion does not come alone. As soon as a system generates harmonic distortion, it also produces e.g. intermodulation distortion.
And it is unclear (at least to the best of my knowledge) which is the worst of the two when it comes to audibility. I can imagine that when you play “simple” music (like the infamous jazz-trio), intermodulation distortion is not very disturbing or annoying compared to the harmonic distortion. But when it comes to reproducing the classical symphony orchestra, intermodulation easily leads to a “grey, misty” sound, in which details drown in the intermodulation distortion products. So the situation may then be completely opposite. This would require a lot more investigation before any final conclusions can be drawn. As far as my own experiences go, any further reduction of the amplifier's

distortion increases the audible presence of fine details in the reproduced sound. But I often listen to the classical symphony orchestra, which might be more susceptible to intermodulation distortion.

The statement that an amplifier sounds good “if and when it shows a regular and harmonic decline of its distortion spectrum” comes out of the blue. In general, such a decline is a pretty common property of distortion components: even a square wave shows this behavior. So I think this statements needs more detailed specifications to understand and to be underpinned. It also implies that the ADDITION of distortion components to acquire this property (so with a higher overall distortion figure!) would improve the sonic quality of an amplifier.
This contradicts my own experience: more distortion masks the reproduction of details as mentioned above. If the addition of distortion components would improve the sonic quality of an amplifier, it would be quite easy to build “good” amplifiers.

As a side remark: the theory of (negative) feedback stems from the twenties of the 20th century, long before the transistor was invented, so the suggestion that it was invented to improve the quality of transistor amplifiers is incorrect. It can be used to reduce the distortion products in (transistor) amplifiers, but it is by no means a “miracle cure” for all deficiencies of amplifiers.

Moving the feedback pick-up point “upstream” of the power amplifier transistors results in a signal strength dependent output impedance when the power transistors are not operating in class A. This will have an effect on the dynamic behavior of the amplifier (see also below).

The discussion about the harmonics is interesting, but it is unclear to me why the author stops at the fifth harmonic when “integral” distortions are compared (from sheet 31). Of course, I understand that the fifth harmonic of 20 kHz is outside the range of interest, but harmonics of low frequencies tend to end up in the most sensitive frequency range of human hearing, so they might be quite disturbing, especially when we are talking about harmonics which do not occur naturally in instruments. In my own experience, the presence of harmonics above the fifth, even in small amounts, often sounds annoying and tiring. Also, harmonics of harmonics may play a role (N.B. In some instruments, the upper harmonics are stronger than the first! And they get distorted too). So the actual distortion products can reach a lot further than one would expect at first sight when we look at the signals from real instruments.

Secondly, there seems to be a difference between “harmonic distortion” and “harmonic distortion”. Looking at the figures for harmonic distortion of solid-state amplifiers, valve amplifiers and loudspeaker units, one could easily conclude that the distortion of solid-state amplifiers is so low that it is completely inaudible when loudspeakers are used (but I don't know of a way to avoid these at this moment). Yet, it is not very hard to hear the effect of amplifier distortions which are several orders of magnitude smaller than those of loudspeakers. A similar remark can be made about valve vs. semiconductor amplifiers.
So there are still a lot of aspects which still are not understood and need further investigation as I don't think the theory as presented by Peter Schüller explains this.

We could start to make distortion measurements of amplifiers under more realistic conditions. The -in more than one sense- complex behavior of the loudspeaker impedance has its influence on the sonic behavior of the amplifier. A point the author does not address is the reaction of the feedback loop to the phase difference between the voltage and current when complex loads are used. The easiest way to understand this problem is to look at a zero-crossing of the output voltage. When

there is a phase difference between voltage and current, the feedback loop is forced to generate an error voltage in order to open one of the power transistors to provide the required current.
This is a distortion, which is not present when a pure Ohmic load would be used. This is the main reason why I use impedance compensation already for tens of years (see the reference below, also showing that the impedance can be a lot more constant than is shown in slide 41 of Peter Schüller's lecture). Therefore, I also support the call for loudspeakers with resistive impedance behavior, but I think the underpinning, given in the presentation of Peter Schüller is incomplete.

Another problem is the amplifier dynamics. Systems which measure nicely on steady signals can become pretty bad with dynamic signals (like you find in music). Remember that all the theory of Fourier c.s. is based on linear systems. And we are studying non-linear systems! So in order to get a good insight in the limitations of our system, we need to add tests which challenge the non-linearities. I have encountered several cases which looked pretty good with steady signals and broke down completely when some serious dynamics were used for testing.

Related to this is the statement “We all know that each power output measurement heavily depends on the exact mains voltage”. This is only true for amplifiers with non-regulated power supplies. If an amplifier has good functioning regulated power supplies, the amplifier supply voltage should be independent of the mains voltage (within, of course, reasonable limits) and thus the output power should be independent of the mains voltage. If this is not the case, the dynamic behavior of the amplifier is flawed, so the first test of an amplifier would be to verify that its output power does NOT depend on the mains voltage!

I would welcome further elucidation and explanation from both Burkhard Vogel and Peter Schüller and I am willing to elucidate my points of view to them if necessary.

Dr. Hans R.E. van Maanen,
Bleiswijk, The Netherlands

Reference:

Hans R.E. van Maanen and E.T. Zonneveld, “An Extended Model for the Impedance and Compensation of Electro-Dynamic Units and their Determination”, paper no. 3823 (P8.1), presented at the 96th AES convention, February 26 - March 01 1994, Amsterdam (The Netherlands)