Why do amplifiers sound different?

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1. Introduction

Power amplifiers are an essential part in the sound reproduction chain. And although semiconductor amplifiers have been around for over 60 years, there is still a lot of development going on. Nowadays, the distortion figures of high-end power amplifiers are very impressive (e.g. < 0.001% harmonic distortion) and easily outshine those of microphones and loudspeakers. Yet, when it comes to listening, differences are noticed between amplifiers and their distortions can be heard in spite of the use of loudspeakers with much higher distortion figures. In this note I will discuss some aspects which play a role in this –at first sight incomprehensible- phenomenon, albeit that I will address a part of the puzzle, not all noticeable differences can be explained by the points I will bring up, partly because I don't know everything there is to know and partly because not all causes have yet been identified, I think. So please see this as a contribution to the discussion, not as the final word on it. Therefore, I welcome contributions of others as "two know more than one", as an age-old Dutch expression says.

One of the basic problems is that we try to "catch" distortion in a single number. But one could pose the question whether this is feasible. To take a simple example: would the audible effect of say 1% harmonic distortion of only the second harmonic be just as noticeable or annoying as 0.1% of harmonic distortion of each of the second to the eleventh harmonic? Or be equivalent to 1% harmonic distortion of the tenth harmonic only? I don't know the answer (because I never tried such a comparison as it is rather hard to do) but there is another example: valve (tube) amplifiers are often highly rated for their musical quality, even though their distortion figures are horrible, compared to those of semiconductor amplifiers. Could there be a similarity between loudspeaker properties and valve amplifiers, distortion wise? Well, there is: both produce mostly lower harmonics (up to the fifth) with virtually no harmonics above that as is illustrated in fig. 1. Semiconductor amplifiers, however, tend to generate harmonics up to very high numbers as can be seen in fig. 2. In literature, there is agreement that our hearing tends to mask frequencies close(r) to the exciting tone than those further away. Or, in other words, the lower harmonics are easily masked by the exciting tone whereas the high harmonics are not, as is shown in fig. 3. On top of that most mechanical musical instruments generate only harmonics up to the fifth of the basic frequency, so distortion products introduce only a small change in the ratio of the harmonics, usually less than is caused by the linear distortion of loudspeakers. So it is not really surprising that components which generate only lower harmonics are not so much experienced as annoying than components which generate more higher harmonics, even at a lower level. So the distortion figure of an amplifier is in itself of little use. A spectral specification would be more useful, but is rarely given.

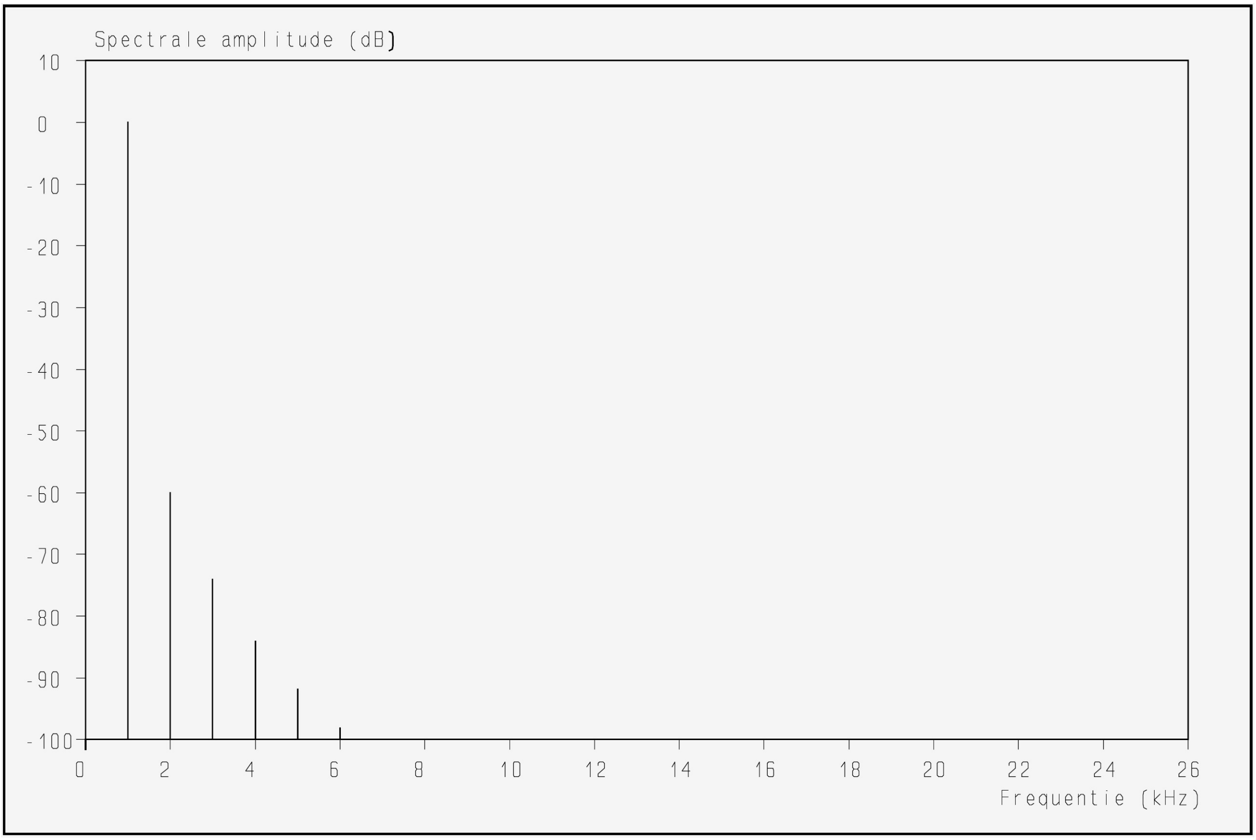

Figure 1: *Spectrum of signal from a component which generates mostly lower harmonics like e.g. loudspeakers and valve (tube) amplifiers.*

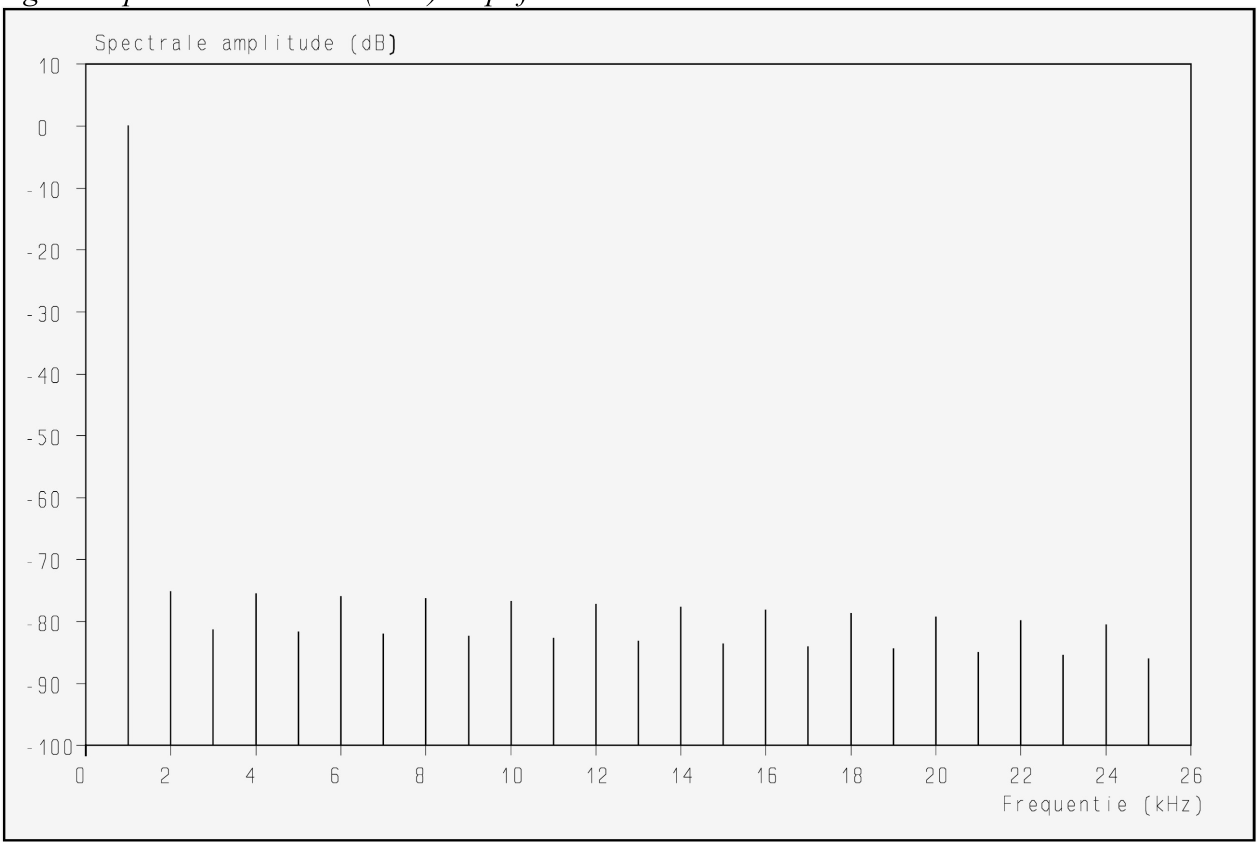

Figure 2: *Spectrum of signal from a component which generates many high harmonics like e.g. a semiconductor amplifier, albeit at a lower level.*

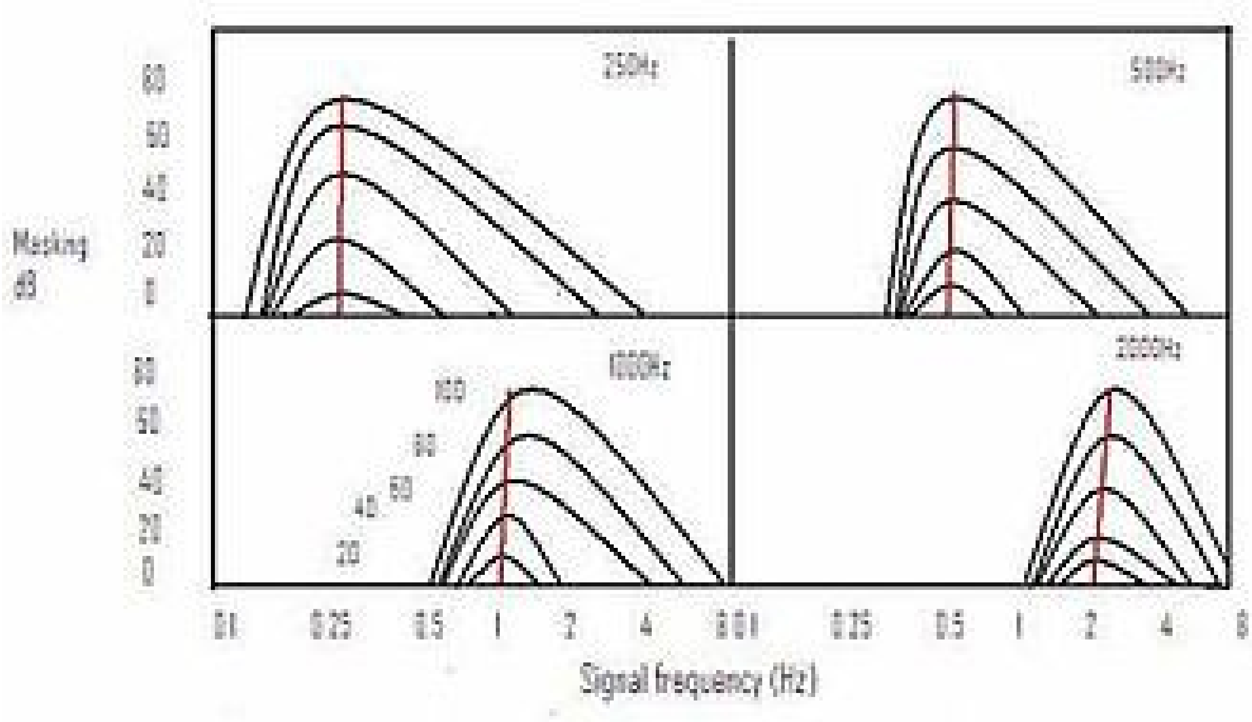

Figure 3: *Masking curves of human hearing (from the Internet). Red = exciting tone, black masking curves of different levels.*

The distortion measurement of power amplifiers is usually performed using a pure resistor as its load. This means that the output voltage and the current through the load are always in phase. However, using realistic loads like loudspeakers, this is not the case. The absolute value (modulus) of the impedance can easily vary by a factor of five (!) and the phase angle between voltage and current can exceed 45 degrees (see fig. 4) which, in general, has repercussions for the response of the amplifier and hence to its specifications. This can be understood by looking at the basic design of a power amplifier as shown in fig. 5. Take the zero crossing of the output voltage. When the current through the load is not zero because of a phase shift between the two, one of the output transistors needs to deliver this current. But this is not possible unless an error voltage at the output can open one of the output transistors via the feedback loop! So distortion is introduced and this phenomenon tends to enhance cross-over distortion which is well-known for its highly annoying properties because it has the wide spectral distribution of harmonics as shown in fig. 2. A class-A amplifier is far less sensitive to this phenomenon (simply because there is always current flowing through the power transistors), but it is very inefficient in its energy use, becomes hot and thus has a reduced lifetime. I think a better option is to apply "impedance compensation" so the load of the amplifier is far closer to the design load. This also simplifies the design of the cross-over filter and improves its properties. Such a compensation can be done as is shown in fig. 6. All in all, the ability of an amplifier to handle complex loads and the properties of the actual loading can have a major impact on the perceived quality of an amplifier, even if their specifications are identical.

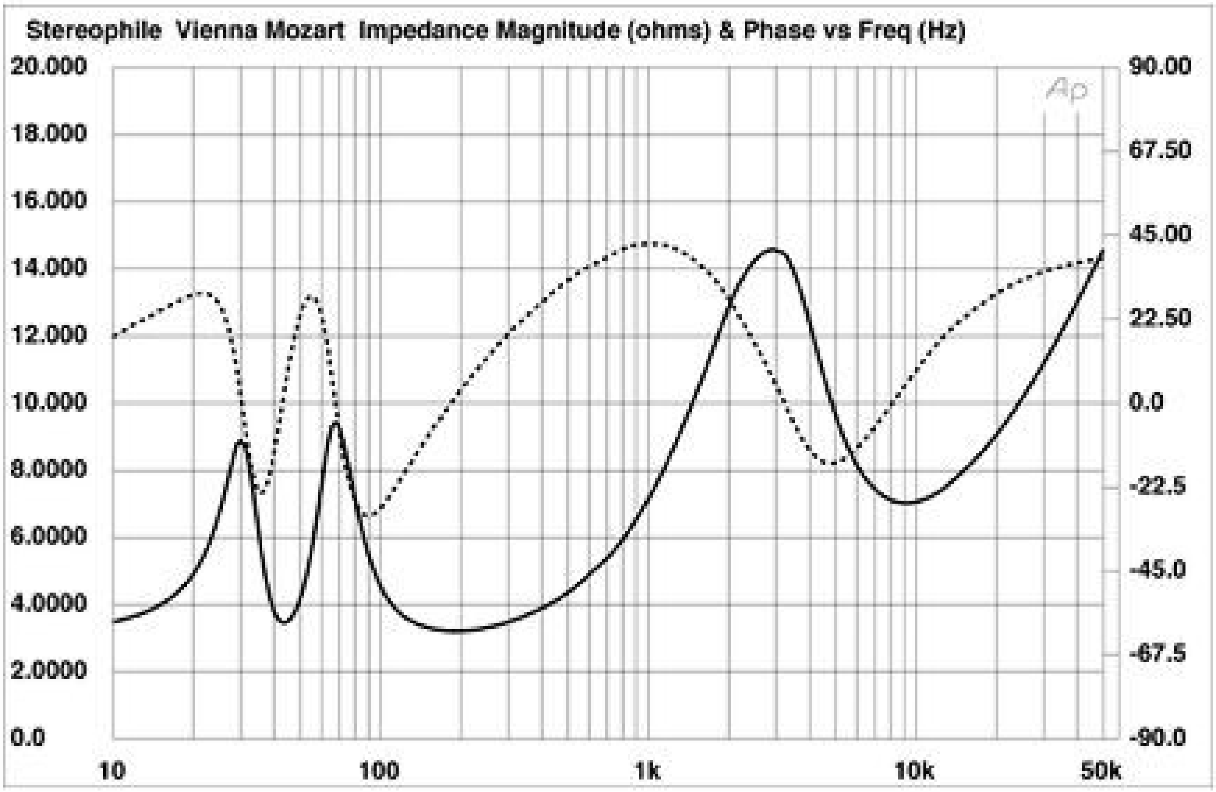

Figure 4: *Impedance as measured and published by Stereophile. Note that the impedance varies between 3 and 15 Ω, the phase angle bounces between -30 and +45 degrees.*

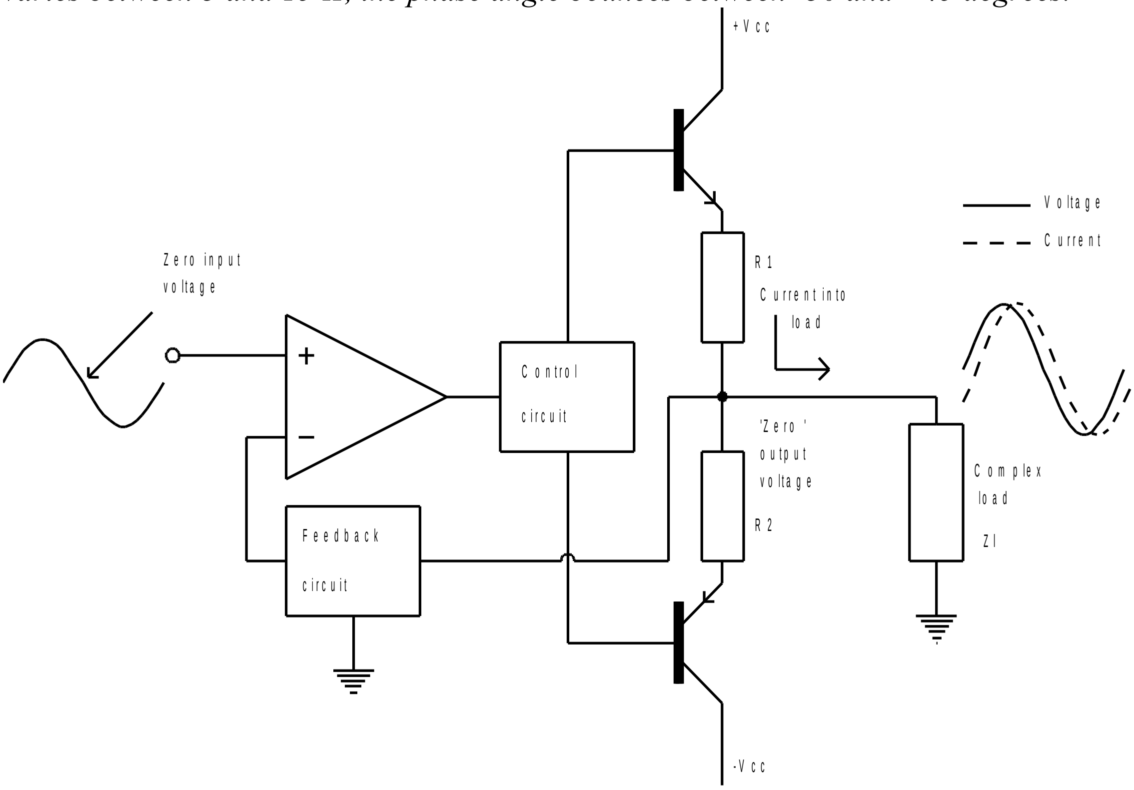

Figure 5: *Basic design of a semiconductor power amplifier. Note the necessity of an error voltage to provide current into the load when voltage and current are not in phase.*

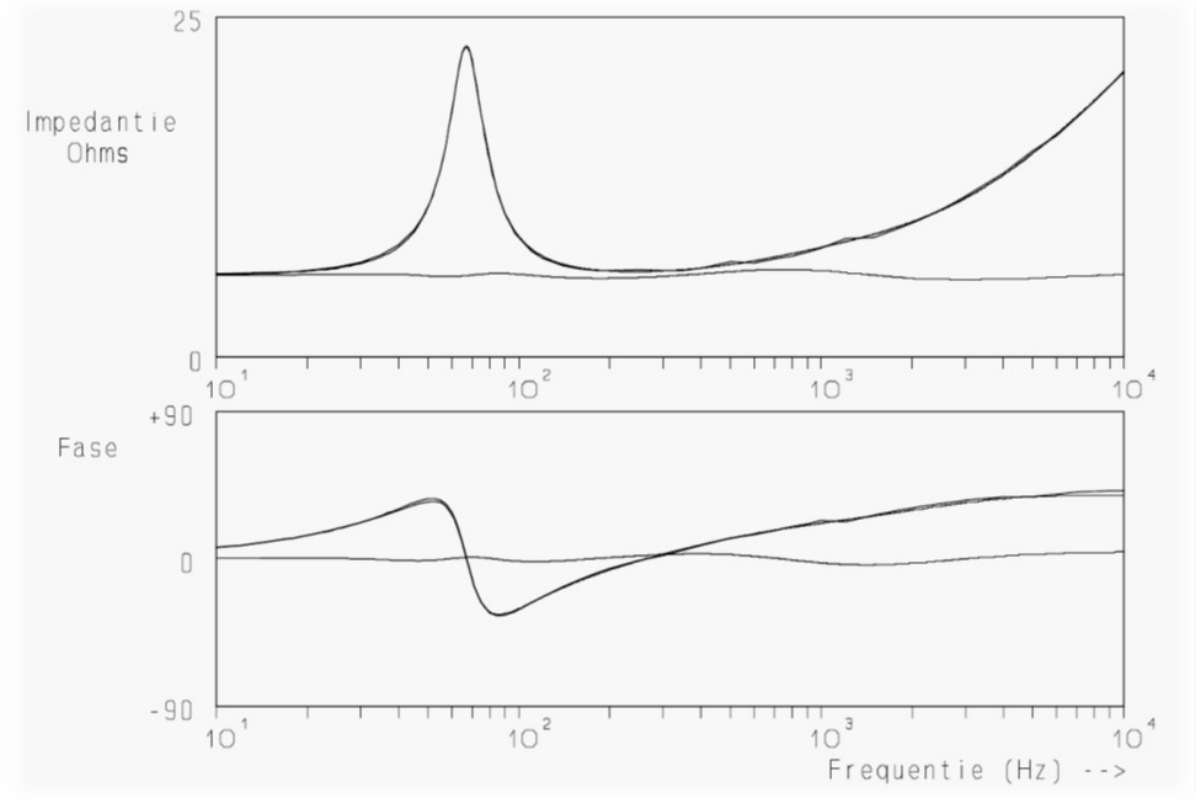

Figure 6: *Impedance compensation for electro-dynamic loudspeakers. Upper block: measured, modelled and compensated impedance, lower block the same for the phase. Original paper can be found on www.temporalcoherence.nl/papers.*

Distortion is also influenced by feedback, both “overall” and “local”. However, there are a number of misunderstandings about feedback. If the designer is not knowing what (s)he is doing, the cure might be worse than the disease. Feedback can help to further improve the quality of the amplifier, but first of all, the quality of the design needs to be as high as possible before any feedback is applied. The main reason is that it is not possible to increase the feedback factor to any desired level. Let me try to elucidate this in more detail.

Any system (including amplifiers) can become unstable (oscillate) when feedback is applied. Control theory describes the phenomena and in short, the following requirements need to be fulfilled:

- The “open loop gain” (= amplification without feedback) should decay with (preferably) -6 dB/oct.
- The “closed loop gain” (= amplification with feedback applied) can be chosen freely (albeit that in practice there are limitations because of the properties of the components), but it is recommended to limit it to say 200 kHz to avoid interference with long wave radio transmissions.
- The ratio of the “open loop” and “closed loop” gains is the feedback factor.
- The distortion is suppressed with the feedback factor.
- It is desirable that the “open loop gain”, and thus the feedback factor, is constant in the audio band up to 20 kHz.

These requirements / preferences mean that a feedback factor is limited to about 10 (20 dB) as can be seen in fig. 7 to avoid an undesirable distribution of the harmonics as is shown in fig. 2. This also means that the distortion is reduced by a factor of 10, so a low ‘starting value” of the distortion is required to bring the final distortion level to a “high-end” specification. Note that it is possible to increase the feedback factor to about 200 at 1 kHz by choosing a different time constant for the open loop decay, resulting in a 20 times lower figure of the distortion at 1 kHz (the normally used frequency at which the distortion is measured and specified) but at the expense of a high fraction of higher harmonics in the distortion product. Not really surprising is that –in general- amplifiers with a constant feedback factor over the audio range are rated higher on aspects like “musicality” and better on “harshness of sound”.

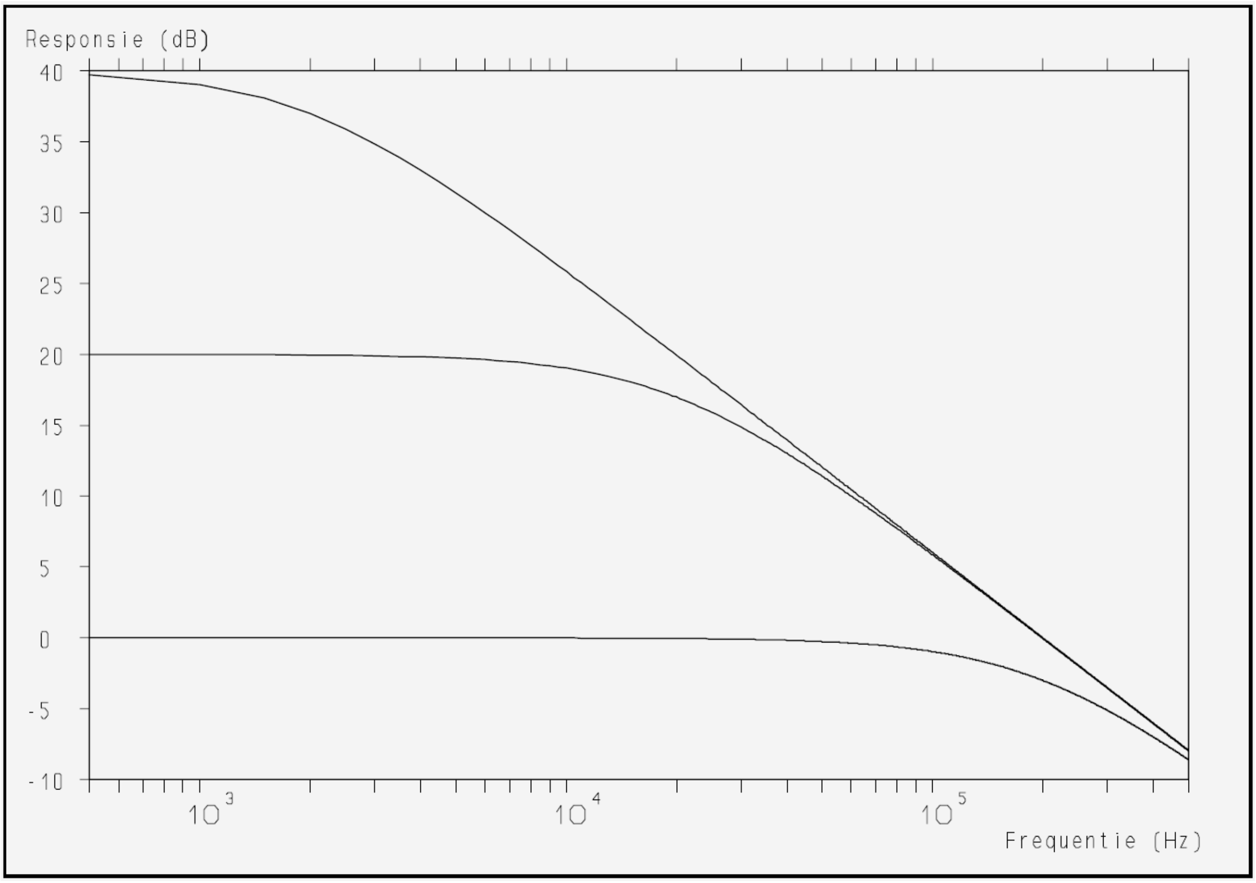

Figure 7: *Differences in open loop gains and closed loop gain. Upper trace: open loop gain with a 40 dB feedback factor, middle curve: open loop gain with a 20 dB feedback factor and lower curve: closed loop gain. Note that the higher feedback factor requires that the decay starts already in the audio band.*

The theory of feedback is usually derived by taking all parameters ideal, except for one. The result of feedback is subsequently studied and the usual textbook results obtained. However, in reality, this is not the case: it is not “or, or, or” but “and, and, and”! This changes things, especially with high feedback factors when the actual signal the amplifier has to process is rather error signal instead of the original music signal. It can easily happen that the underlying assumptions of the feedback theory are no longer valid and thus neither the outcome.

Another, often overlooked or ignored, aspect is the power supply of the amplifier. One of my bold statements is that no amplifier can be better than its power supply. The ideal power supply can provide as much current as needed without *any* change of the supply-rail voltages, neither on the short term nor the long term. Of course, this is unrealistic and every power supply will be limited in its current supply and the variations in the supplied current will reflect in the voltages of the supply rails. The question then remains: what can be heard and what not? That question is not so easy to answer, but some general remarks can be made about the issues of the power supply.

When the voltages of the power supply rails (we will assume a positive, a negative and a zero voltage rail) vary, this also means that the operating points of the different semiconductors change during the processing of the signal (read music). You probably won't notice this when you test the amplifier with a sine wave signal of constant amplitude as this will only lead to a shift in the operating points to a new equilibrium value. But with the dynamic signals from music, the operating points can keep on changing continuously and a sort of correlated low frequency "noise" is generated, in stereo amplifiers also with components from the other channel when a combined power supply is used. This can lead to a tiring listening experience without consciously noticing the "noise". You can have a look at the voltage rails with an oscilloscope either when the amplifier is processing music or with tone burst signals. The perfect power supply does not leave a trace of the signal being processed, but you might be in for an unpleasant surprise. Make sure that the amplifier has to deliver current into a load so the power supply has to work! Also, the "zero" voltage can have severe swings when the layout of the printed circuit board is non-optimal, which can give rise to similar phenomena.

The power amplification factor of a power amplifier easily amounts 10^6 to 10^7. Therefore, it is relatively easy to get parasitic feedback from the output to the input by electro-magnetic coupling This can give rise to correlated "misery" which reduces the "musicality" and detail of the reproduced sound. Such phenomena are very hard to measure, but can be heard to make the difference between a "good" and an "excellent" amplifier. Surprisingly, two amplifiers which are completely identical as far as design is concerned, but differ in the way the components and the wirings are located can sound clearly different!

It should not come as a surprise that the design of the amplifiers, used by "Temporal Coherence" considers all the above mentioned aspects and that these have been optimized to create the best sound for our ears. This has been shown by several comparison tests between other commercial amplifiers for the consumer market and our system, consisting of the control and power amplifier. But, of course, we are always willing to demonstrate this in your own environment, provided this is not too far away.

In this note, we have listed a number of aspects which can generate clearly audible differences between amplifiers with similar specifications. Although it is –at first sight- surprising that amplifiers can behave so differently, I hope it became clear from this note that there are technical issues which need to be taken into account to understand the "quality" of amplifiers. I am very well aware of the incompleteness of the list, discussed in this note, but for me it is still a continuing story of amplifier development. In the past 45 years I have discovered many different problem areas with amplifiers and I am quite sure there is more to find out. So suggestions are welcome!