Audio Engineering Society

Convention Paper

Presented at the 130th Convention
2011 May 13–16 London, UK

Signal level and frequency dependant losses inside audio signal transformers and how to prevent those.

Menno van der Veen

ir. bureau Vanderveen, Zwolle, The Netherlands, 8011PA - 21
info@mennovanderveen.nl
www.mennovanderveen.nl

ABSTRACT

In earlier work [8] a model was presented which explains that low voltage level audio signals are extra weakened when they are fed through a transformer. This extra weakening is caused by the signal level and frequency dependant inductance of the transformer. Combining this extra weakening with the threshold of hearing curves, showed that noticeable loss of micro details occurs in the frequency band from 20 Hz to 1 kHz. This paper expands the previous work with measurements on several valve amplifiers, refines the model and makes it applicable to macro signal levels close to saturation of the transformer. Also methods are given to minimize this extra weakening in transformers.

1. INTRODUCTION

Transformers sound, they say, and so true they are. As do capacitors and resistors, but that is not the aim of this paper. See [1] to [9] for more information on these matters. As a transformer designer I have a lot of experience with output transformers in valve amplifiers and found there that transformers sound because of their variable inductance, their maximum flux density of the core, their overload behavior, their leakage inductance and internal capacitances and magnet wire resistances.

In this paper I will mainly focus on the influence of the sort of iron used in the core plus the shape of the core, combined with the driving and loading impedances on the transformer and I will focus on the frequency range from 20 Hz to 1 kHz.

Imagine an output transformer in a push pull or single ended valve amplifier. The power tubes have an effective total internal resistance $\mathbf{r_i}$ and they act as an alternating voltage source $\mathbf{v_p}$ with r_i in series driving the primary of the output transformer. In [7] chapters 2 and 3, I prove that r_i certainly not is a constant, but signal level dependant, leading to DDFD (dynamic damping factor distortion). See figure-1 which demonstrates this effect.

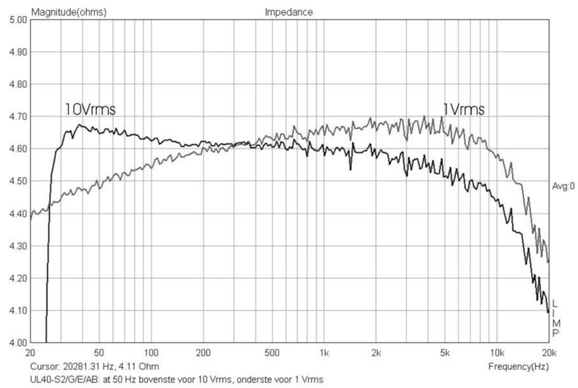

figure 1: Dynamic Damping Factor Distortion where Zout (vertical axis) of a UL40 valve amp [11] depends on the output level and the frequency.

For now I assume r_i to be constant. If I would not do that, the real problem of this paper would not be recognizable anymore because of the abundant formulas. So, lets try to keep things simple.

The secondary winding of the output transformer is loaded by a speaker $\mathbf{Z_s}$. I know that Z_s is strongly frequency dependant, but for this study it is good enough to consider Z_s to be a constant.

Also we can neglect the magnet wire resistances of the primary and secondary windings, because they represent constant losses, while I am researching variable losses caused by the core only. Let $\mathbf{N_p/N_s}$ be the turns ratio of the transformer. Then we can bring Z_s to the primary side of the transformer by means of $\mathbf{Z_{aa}} = Z_s \cdot (N_p/N_s)^2$.

Figure 1 shows that for low frequencies below 1 kHz the complete arrangement works as a voltage source driving the speaker AND the primary inductance $\mathbf{L_p}$ of the primary winding. The current that flows through L_p is called the exciting current $\mathbf{I_{ex}} = v_{Lp}/(2 \cdot \pi \cdot f \cdot L_p)$ and we wish that current to be as small as possible in order to convert all audio music energy only to the loudspeaker. This means that L_p should be as large as possible.

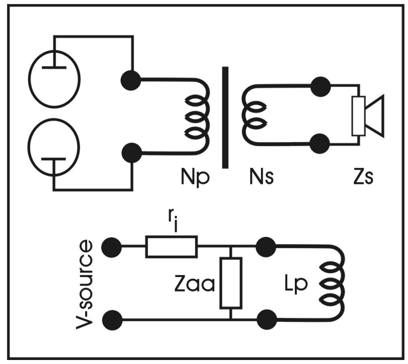

figure 2: Equivalent circuit of valve amp driving the loudspeaker Zs and Lp.

The frequency and signal level dependant part of the transfer from the primary tubes to the loudspeaker is given by formula 1.

$$H(f) = \frac{s.L_p}{r_i // Z_{aa} + s.L_p} \qquad (1)$$

The inductance of the primary winding is given by formula 2, where $\boldsymbol{\mu_o}$ is the magnetic permeability of vacuum; $\boldsymbol{\mu_r}$ is the relative magnetic permeability of the iron of the transformer core; N_p is the total number of turns of the primary winding; $\mathbf{l_c}$ is the magnetic path length inside the core; **A** is the cross sectional surface of the core; $\mathbf{l_g}$ is the width of the gap. See for more information figure 3.

$$L_p = \frac{\left[\mu_0 \cdot \left(N_p\right)^2 \cdot A\right]}{\left(l_g + \frac{l_c}{\mu_r}\right)} \qquad (2)$$

The problem I studied results from the fact that the relative magnetic permeability of the core iron is not a constant, but depends strongly on the magnetic flux density inside the core, and thus from the level of the alternating voltage that the valves send to the transformer.

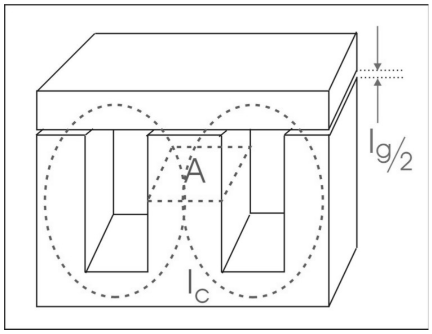

figure 3: schematics of the transformer core indicating l_c and A and the eventually present core gap l_g.

Figure 4 shows my measurements [8] on two sorts of core iron often used, and there it is clearly shown that the relative permeability is not constant. Consequently the primary inductance L_p certainly is not constant and therefore the signal transfer through the transformer is not constant.

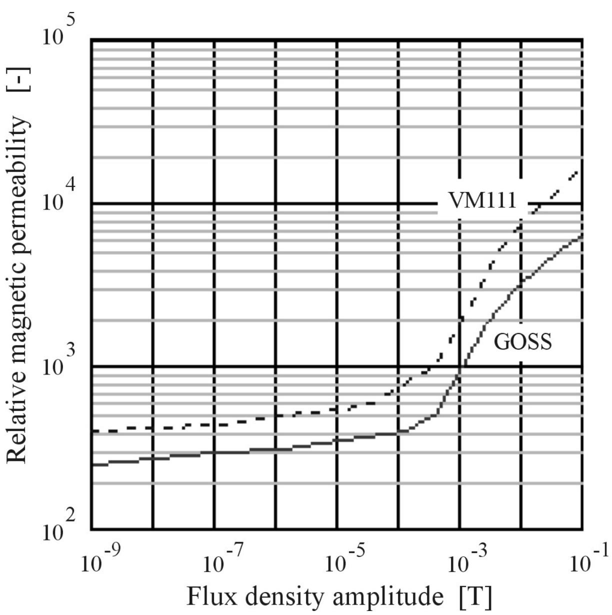

figure 4: Measurements of μ_r from 10^{-8} to 10^{-1} T; extrapolation below 10^{-8} T.
The maximum error below 10^{-3} T is 20 %.

The conversion of the driving voltage $\mathbf{V_{aa}}$ (rms value) over the primary winding to the magnetic flux density **B** (amplitude) inside the core is given by formula 3.

$$B = \frac{\left(V_{aa} \cdot \sqrt{2}\right)}{\left(2 \cdot \pi \cdot f \cdot N_p \cdot A\right)} \qquad (3)$$

Now I can formulate the problem. At normal voltage levels, say at 1 Watt power to the loudspeaker, the flux density B is rather large. Then the relative perm is large enough to make L_p large and the exiting current I_{ex} becomes negligible, no extra weakening occurs. However, at small signal levels, meaning small B-values, the relative perm becomes very small and consequently also the primary inductance L_p. Then almost all of the current will go through the almost shortcutting L_p and not through Z_{aa}, which represents the actual loudspeaker. An extra weakening will occur. We do not only meet this effect at micro signal level, but also in the region close to core saturation, at maximum output power at low frequencies. There also the relative permeability collapses.

My earlier paper [8] discusses all of this in detail, and combines it with the threshold of hearing, as shown in figure 5.

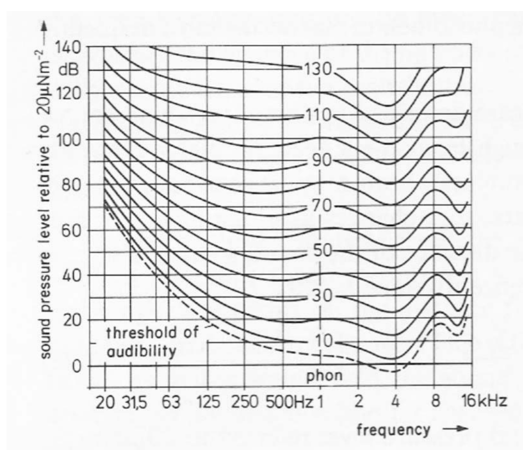

figure 5: ISO-curves of human ears

This model results in graphs which show which signals fall under the threshold of hearing caused by the permeability collapse inside the core at small signal levels. Figure 6 gives one example.

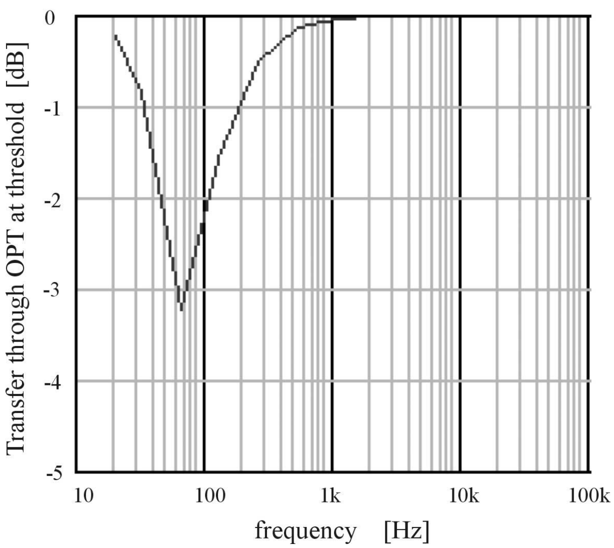

figure 6: Deviation of linear transfer through OPT at threshold, for Z_{aa} = 4 kΩ, r_i = 30 kΩ, η = 90 dB/W,m, d = 1 m, GOSS.

The fine result of this model is that it delivers very good agreement between theory and subjective observations, as explained in [8].

The model is recently precisely measured on several valve amplifiers in my lab and during the European Triode Festival 2010. Results are given and they show that the model and its description are correct.

Following I focus on how this extra weakening can be prevented by using special core steel, or by adapting the driving and loading impedances to the transformers, and/or by changing the shape of the transformer core. Also the influence of negative feedback is discussed. Design rules are given.

2. HOW TO MEASURE DEVIATIONS

Because the deviations described occur at very small (or very large) signal levels and at specific frequencies, a special measurement method had to be developed. The voltage levels can be in the nV range and consequently hum and noise will be the big enemies that make the measurement difficult. In the section following I describe my measurement setup.

2.1. THE SOUNDCARD

Nowadays we can use a good quality soundcard of a computer as a measuring device. I used a Creative Audigy 2 Z5 soundcard, in 96 kHz sampling mode with 24 bits resolution. The in- and output sensitivities of this soundcard are calibrated at 1 V_{rms} for 0 dB_{FS} (0 dB full scale). In front of the input is placed a calibrated attenuator-amplifier with an input impedance of 100 kOhm.

Detailed measurements showed that the quality (noise-floor) of the in- and output amplifiers plus low-pass filters inside the soundcard restrict the measurement range from 0 to -120 dB_{FS}.

In the frequency domain the output amplifier of the soundcard (headphone output) with its low pass filter has a restricted linear (+/- .1 dB) frequency range up to 22 kHz. So, from a few Hz to 22 kHz, the measurements are reliable in the frequency domain.

The THD of the complete soundcard, its output connected to its input, is 0.003 % at 1 kHz.

2.2. THE MEASUREMENT PROGRAM

To fully employ the goodies of this system, we need a reliable program to drive the computer. See [10] for the details of the Arta/Steps/Imp programs that I used, designed by Ivo Mateljan. His programs are of excellent quality, noticeable written by a scientist, very clear and with no nasty bugs, with detailed information about the math that is used to perform the measurements plus clear indications of the setup of the measurements to stay within the assumptions of the math applied.

2.3. THE MEASUREMENT SET-UP

In the program "Steps" the linearity function measurement is selected. There the input signal to the device under test is compared to the output signal of the DUT. The vertical axis of the resulting graph gives the amplification of the DUT (in dBV/V) as function of the input voltage (horizontal axis). In steps the input voltage changes from small to large, its range can be selected. Per measurement a frequency can be set.

The measurement is a "heterodyned measurement of sine response" where a very narrow filter is created around the selected frequency with a 1/T bandwidth. **T** is the integration time of the measurement. In my case I selected T = 200 ms, resulting in a 5 Hz bandwidth.

The DUT is a complete valve amplifier with output transformer, and it will be shown following that this measurement focuses on the OPT behavior. The amplifier is secondary loaded with a dummy load Zs. The two channel character, left and right input channels are both used, of the measurement eliminates deviations inside the soundcard. Figure 7 shows the complete measurement set-up.

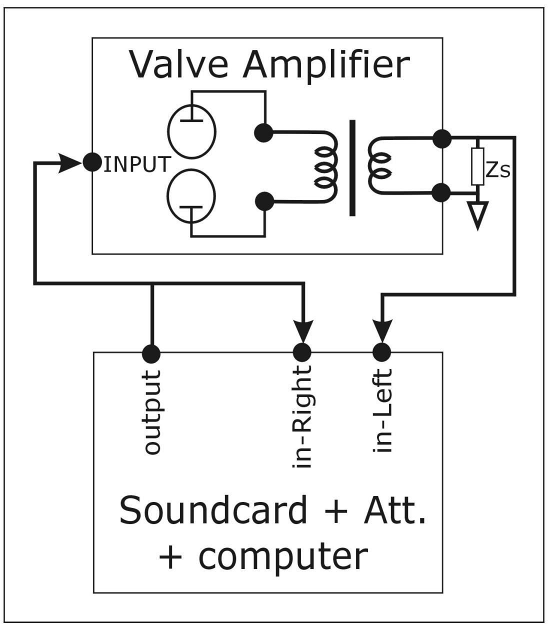

Figure 7: Measurement set-up

2.4. MEASUREMENT CONDITIONS

The math of the set-up demands absence of noise, and this condition could not fully be met with the tested valve amps. I therefore restricted the measurement floor to -80dBV at the input of the amps tested. I tested the transfer of the amps at frequencies of 1 kHz, 70 Hz, 20 Hz and 10 Hz. The test frequency of 1 kHz: because no problems are expected there, a straight line should be the result, except from overdrive or shifting of bias settings inside the valve amplifier. The test frequency of 70 Hz: earlier studies [8] have shown that there the subjective most noticeable effect will occur. With the integration time applied (T = 200 ms) the heterodyne filter is narrow enough to exclude 50 Hz mains hum. The test at 20 Hz probably will show strong core collapse at micro and macro signal level. To tease the core the hardest I also tested at 10 Hz.

In the measurement set-up the frequency range of the amp is measured as well. However, if the outcome of the measurement is a straight horizontal line, there is no core influence. Then only a constant, signal level independent, weakening is shown, caused by the coupling capacitors. If the outcome is sloped or curved, then the core collapse is the cause.

3. MEASUREMENT RESULTS

The results shown are only frequency and signal level dependant deviations from a constant transfer. The "threshold of hearing" curves are NOT applied. I research here what goes wrong inside the transformers and NOT what we can hear of it, because that part already is discussed in [8].

In my lab I tested about ten different amps, and at the ETF2010 I tested fifteen amps. They all had totally different schematics and output transformers. In spite of that, they gave more or less the same result.

See figure 8 as a clear example which enables me to explain what happens.

The upper line is the 1 kHz measurement. It is almost straight, however at larger input/output voltages the internal bias of the amp changes, causing loss of amplification (limiting) which is not caused by the transformer. This situation is found also in the 70 and 20 Hz measurements.

The second line is at 70 Hz and shows a tendency of weakening at lower input voltages. The line is sloped, so, the cause is in the transformer.

The third line measures at 20 Hz and shows more clearly the loss of amplification caused by the perm behavior in the core.
Notice that these three lines all have the same maximum, indicating a constant amplification (no influence of coupling capacitors) from 20 Hz to 1 kHz.
The fourth bottom line is measured at 10 Hz, clearly indicating perm collapse at low input levels and core saturation at high input levels, because the sloping down of the curve has shifted to the left.
Noise residuals are visible below an input voltage of -80 dBV in the 20 and 10 Hz measurements, showing the reason why I had to limit the input voltage range for a reliable result.

Combining formulas 1 to 3 with the perm measurements of figure 4 completely describes the behavior as shown in figure 8. The model presented in [8] and here is correct and its effects are noticeable in practical valve amplifiers.

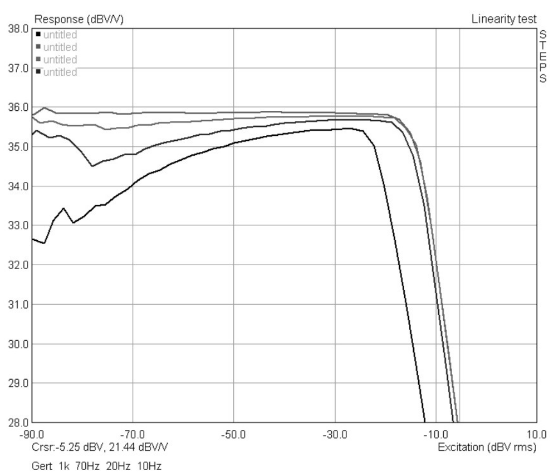

figure 8: example of core collapse.
From top to bottom: 1 kHz, 70 Hz, 20 Hz, 10 Hz.
Amplifier "GERT-RL12"; no internal negative feedback.

The next example (figure 9) is an ultra linear negative feedback amplifier and internal electronics (auto-bias module, see [11]) prevents bias shifting inside the amplifier, making more clear what happens inside the output transformer.

The 1 kHz line is straight with a small bulb at the right side because of overdrive of the tube circuitry. The 70 Hz line is little sloped, caused by perm-loss as discussed here. The 20 Hz line is further sloped and at the right side the transformer is just driven into saturation. The 10 Hz line shows this effect more pronounced.

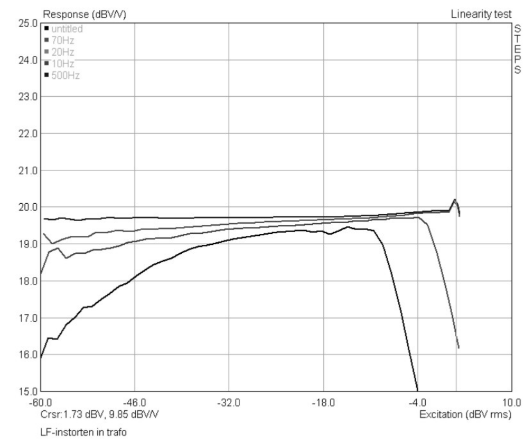

figure 9: example-2 of core collapse.
From top to bottom: 1 kHz, 70 Hz, 20 Hz, 10 Hz.
Amplifier "UL40-S2" [11] with Ultra Linear negative feedback

The third example in figure 10 shows a multi paralleled power tube amplifier with very small effective plate resistance r_i of the combined power tubes, resulting in little deviation of linearity. In this amplifier no auto-bias module was installed, bias shifting occurs on the right side. Only at 10 Hz a little deviation of linearity is noticed plus core saturating at maximum output power.

The fourth and last example (see figure 11) is an amplifier with 13 dB overall negative feedback, combined with local Super Triode [6],[7],[11] feedback . Bias-shifting is prevented with an auto-bias module. It is clearly visible that the feedback has "cured" the deviations of linearity, and only for 10 Hz the transformer core saturation is visible.

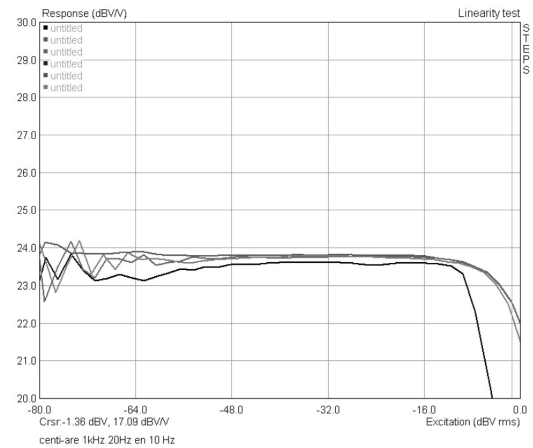

figure 10: example-3 of core collapse.
From top to bottom: 1 kHz, 70 Hz, 20 Hz, 10 Hz.
Amplifier "Centiare Tentlabs"with
multi paralleled power tubes

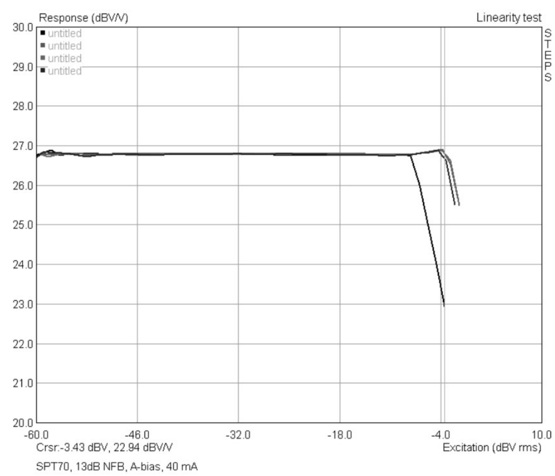

Figure 11: example-4 of core collapse.
From top to bottom: 1 kHz, 70 Hz, 20 Hz, 10 Hz.
Amplifier "SPT-70"; see [7] chapter 8

The examples given clearly show that the amount of feedback inside the amplifier influences the results measured. In the next section this positive effect will be further discussed.

4. HOW TO PREVENT LOSSES

There are several methods to prevent extra weakening and they will be discussed following.

4.1. USE r_i AND NEGATIVE FEEDBACK

Formula 1 clearly explains this method. When $r_i//Z_{aa}$ is negligible compared to $s.L_p$ in the frequency range and signal levels of interest, then the frequency dependant part of the actual transfer is unity.
Lowering r_i can be done inside the power valves: use triodes instead of pentodes. Or use local feedback around the power valves and the output transformer, like ultra linear screen grid feedback, or cathode feedback or super triode feedback. See [11] for abundant examples of such methods.
Feedback in general, being local or overall, lowers r_i by a factor of $(1+\beta\cdot A_0)^{-1}$ where β is the feedback factor and A_0 is the open loop amplification without feedback. So, feedback cures this nasty effect, or at least can bring it below the threshold of hearing.

4.2. SOLVE INSIDE THE TRANSFORMER

The core steel is the main cause of the weakening [8]. To prevent this from happening, apply high permeability steels like Metglas, HD-105-30, VM111, Isoperm, Amorphous steel and so on. Use correctly annealed steels because good annealing makes the perm larger. Also, if possible, do not vacuum impregnate the core while impregnation lowers the perm.
Another solution is to apply a gap in the core with $l_g > l_c/\mu_r$ in the range of interest. Then L_p will become smaller, but almost insensitive to the frequency and signal level of the applied voltages.
The shape of the transformer is important as well. Let a transformer have a cross sectional surface A, as is needed for the maximum power to be transferred, see formula 3. The EI-shape cores have the smallest l_c compared to C-core or toroidal or R-core. meaning that in EI-cores L_p will be the largest, see formula 2.
In order to prevent core collapse at high signal levels, one might consider to design the transformer at $0.5 \cdot B_{max}$ at full power at the lowest frequency to be transferred. This has the extra advantage that by short duration full power low frequency burst signals, the flux density in the core

will be just at B_{max} timely, preventing nasty distortion under this condition.

4.3. Designing transformers

I can't help feeling the need to express that designing transformers is an art. If you cure here, you damage there. For instance, using Metglas cures at low levels, but its overdrive behavior is sharp nasty sounding limiting. Designing for $0.5 \cdot B_{max}$ makes the transformers heavy and more expensive. Applying gaps costs money as well. And what to say about the frequency range I did not discuss here: environment of 20 kHz and larger. There leakage and internal capacitances play a leading role and then toroidal is the most favorable shape, see [1],[6].

It is not the aim of this paper to discuss all. When only focusing on the 20 Hz to 1 kHz frequency range, then the perm behavior is dominant. But the transformer designer has to consider at least the range from 20 Hz to 20 kHz and therefore shall have to find a balance between all the goodies and baddies of the different solutions. Fortunately, the customer, who buys a transformer, can cure the effect discussed by applying special valves or any kind of feedback.

5. CONCLUSIONS

In earlier work a model was created that describes the effect of magnetic permeability collapse at very low and very high flux density levels inside a transformer core. It was calculated that this collapse is subjectively noticeable as an extra weakening in the frequency range of 20 Hz to 1 kHz, taking the threshold of hearing and the sensitivity of the loudspeakers into account. In this paper a method is given how to measure this perm effect in existing valve amplifiers. Measurement and theoretical model show total agreement. Two methods are formulated to prevent the extra weakening: apply low plate resistance valves or use feedback, which has the same effect. The second method focuses on high perm steel selection and the core shape of the transformer and the linearising effect of a core gap. It is noted that the cure of weakening in the 20 Hz to 1 kHz frequency range might cause problems elsewhere in the frequency range of the transformer. In summary: perm collapse has influence on the transfer, it can be cured inside or outside the transformer, but be warned: "curing here might damage elsewhere".

6. REFERENCES

[1]: Menno van der Veen; "Theory and Practice of Wide Bandwidth Toroidal Output Transformers"; 97th AES-Convention 1994; San Francisco, preprint 3887.
[2]: Menno van der Veen; "Modeling Power Tubes and their Interaction with Output Transformers"; 104th AES convention, Amsterdam, 1998; preprint 4643.
[3]: Pierre Touzelet & Menno van der Veen; "Small signal analysis for generalized push-pull amplifier topology"; 112th AES convention, 2002 Munich, paper 5587.
[4]: Menno van der Veen & Pierre Touzelet; “New Vacuum Tube and Output Transformer Models applied to the Quad II valve amplifier"; 114th AES convention, 2003 Amsterdam, paper 5748.
[5]: Menno van der Veen; “Universal System and Output Transformer for Valve Amplifiers”; 118th AES convention, 2005 Barcelona; paper 6347.
[6]: Menno van der Veen; "Modern High-end Valve Amplifiers based on toroidal output transformers"; ISBN 0-905705-63-7, www.elektor.com
[7]: Menno van der Veen; "High-end Valve Amplifiers 2"; ISBN 978-0-905705-90-3, www.elektor.com
[8]: Menno van der Veen; "Low level audio signal transfer through transformers conflicts with permeability behavior inside their cores"; Paper 7125, 122nd AES-convention, 2007 May 5-8, Vienna Austria.
[9]: Menno van der Veen & Hans van Maanen; "Non-linear Distortions in Capacitors"; 124th AES convention, 2008 Amsterdam, paper 7500.
[10]: Ivo Mateljan; "Arta/Steps/Imp"; www.fesb.hr/~mateljan/arta
[11]: www.mennovanderveen.nl