Elektor DVD: Modern Valve Electronics

0) Welcome and structure + layout + Patrick

1) Preamplifier

a) Traditional approach – advantages and disadvantages
b) Effects of coupling and decoupling capacitors
c) Modern approach with a constant current source; designs
d) Causes of distortion in the constant current source
e) Impedance converter function and implementation
f) Cathode decoupling – yes or no?
g) ~~F~~olded Cascode to avoid ~~s~~ignal currents on the ground rail
h) Eliminating supply interference
i) Sources of additional information

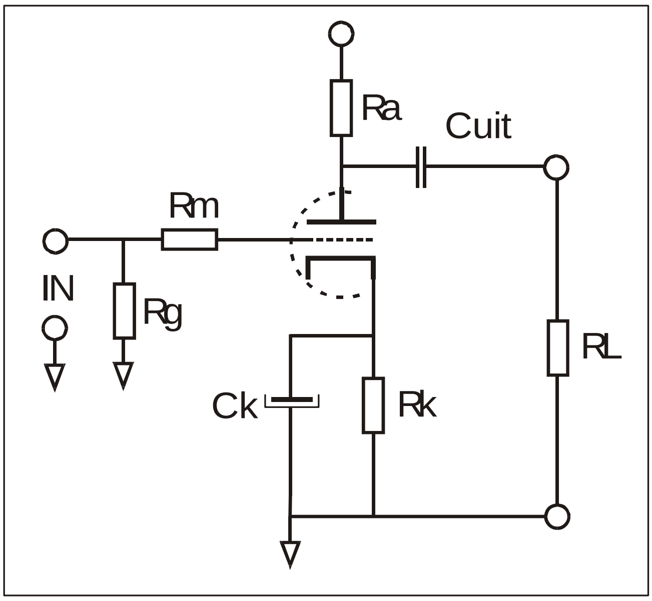
2) Output amplifier

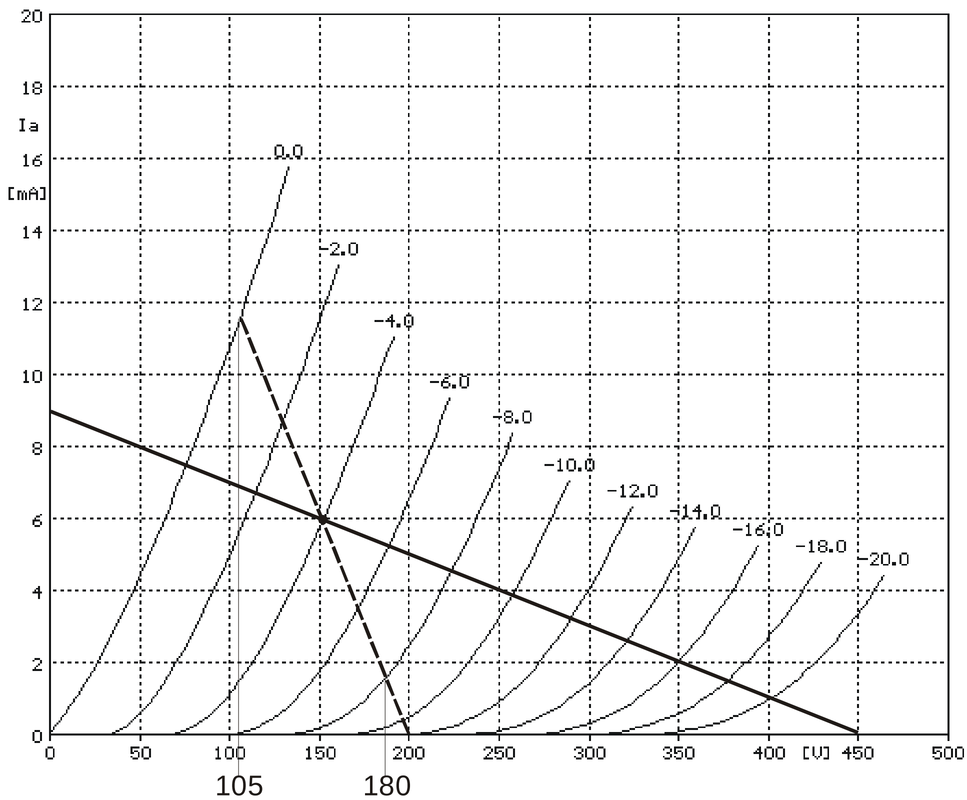
a) Function of the output transformer; basic rules
b) Output amplifier circuits (the project)
c) Disadvantages of cathode capacitors in SE amplifiers
d) Disadvantages of cathode capacitors in balanced amplifiers; remedies
e) Looking into the black box: informative measurements on the outside
f) Measuring output impedance versus frequency
g) Measuring gain linearity versus frequency and input voltage; insight into microdetail reproduction by the output transformer
h) Which types of distortion are acceptable, and which are not?

Preamplifier

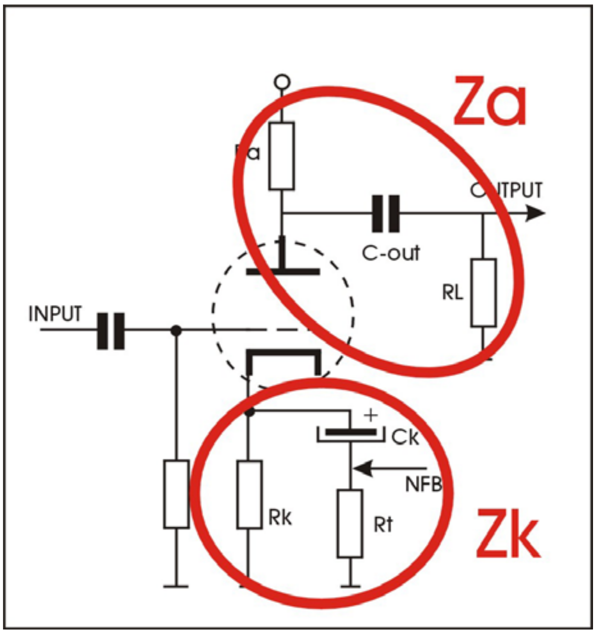
1) Traditional approach

a) Preamplifier schematic
b) Pursuing low distortion
c) Selecting the static load line
d) The reality of the dynamic load line
e) Calculating distortion (second harmonic) in the working range





f) Which cathode and coupling capacitors should you use?



$$H(s) = -\mu \cdot \frac{Z_a}{r_i + (\mu + 1) \cdot Z_k + Z_a} \cdot \frac{R_L}{R_L + Z_{C-out}}$$

$$\frac{C_k}{C_{out}} = \frac{R_a + R_L}{R_k + R_t}$$

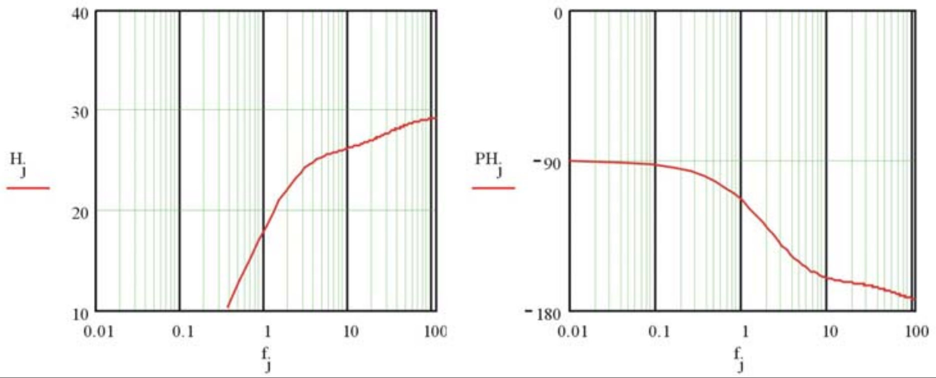
$$f_{-3\,L} = \frac{1}{2 \cdot \pi \cdot C_k \cdot R_\gamma}$$

$$R_\gamma = \frac{R_k + R_t}{r_i + (\mu + 1) \cdot R_k + R_a} \cdot \left[r_i + (\mu + 1) \cdot \frac{R_k \cdot R_t}{R_k + R_t} + \frac{R_a \cdot R_L}{R_a + R_L} \right]$$

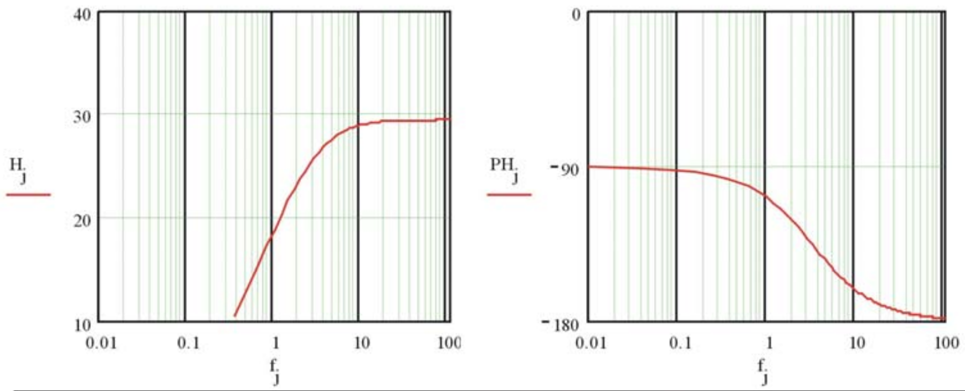
Example:

R_a = 47 kΩ, r_i = 4.51 kΩ, R_L = 470 kΩ, R_k = 680 Ω, R_t = 27 Ω,
C_{out} = 150 nF;
==> **C_k = 109 μF**; μ = 33.7 ==> **R_γ = 453 Ω** ==> **f_{-3L} = 3.2 Hz**

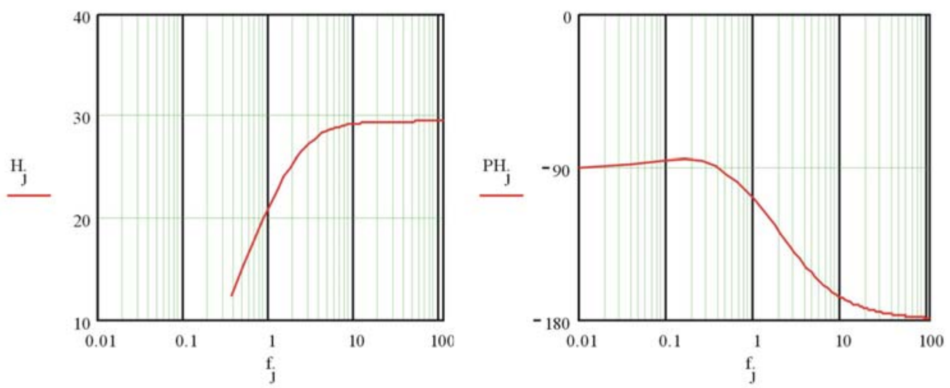
10 uF



100 uF



1000 uF

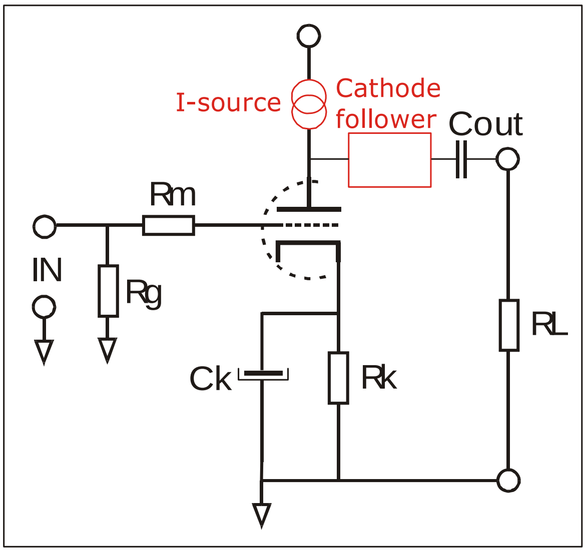


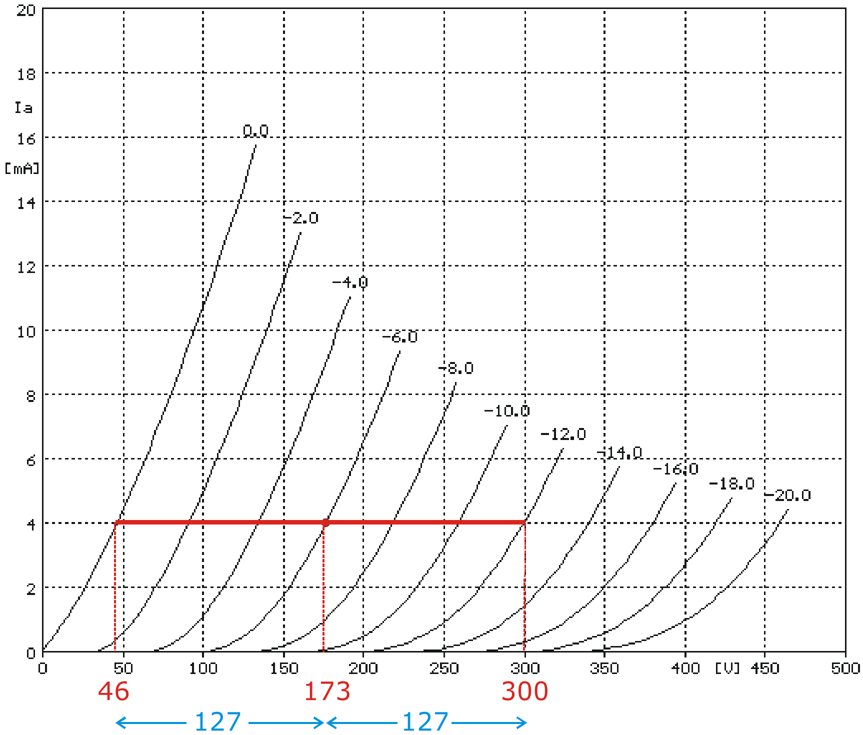
2) Disadvantages of the traditional approach

a) High THD unless a very high supply voltage is used
b) High sensitivity to supply voltage
c) Results strongly dependent on load impedance

3) Modern approach with a constant current source

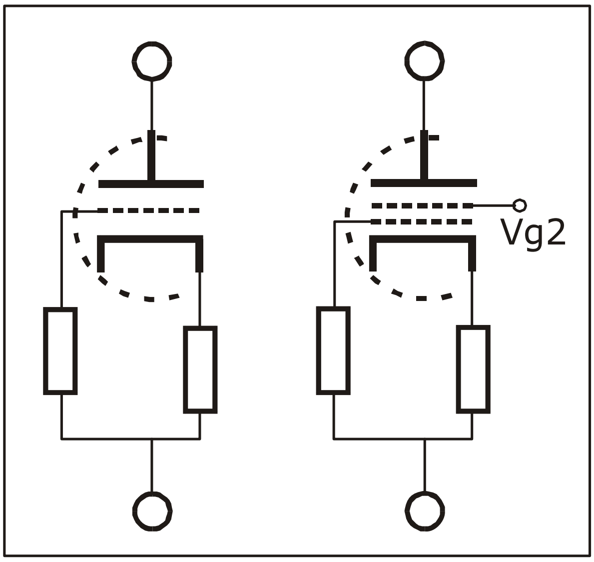
a) Horizontal load line (resulting in low THD)
b) Greater amplitude range with lower supply voltage
c) Supply interference suppression
d) Additional buffer (cathode follower, source follower or emitter follower) makes the output independent of the load resistance (R_L)





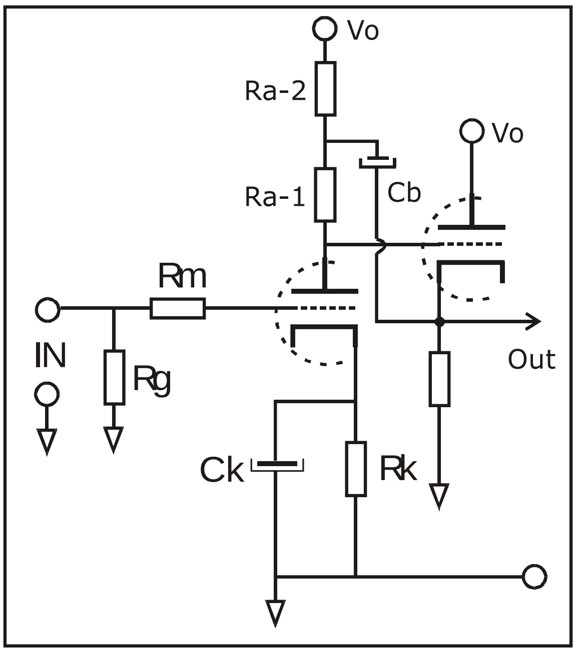
How can you make a suitable constant current source?

a) With a tube



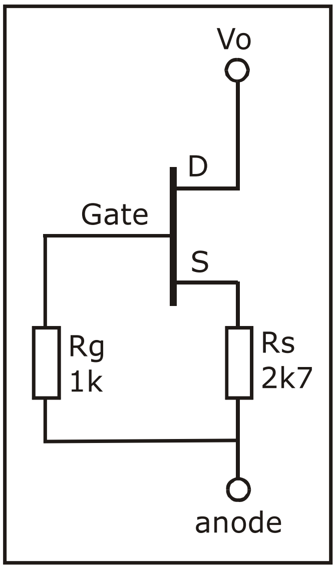
For the implementation, consider circuits such as **SRPP** or **mu follower**. There is a lot of literature available on this subject, so I will not discuss it in detail here.

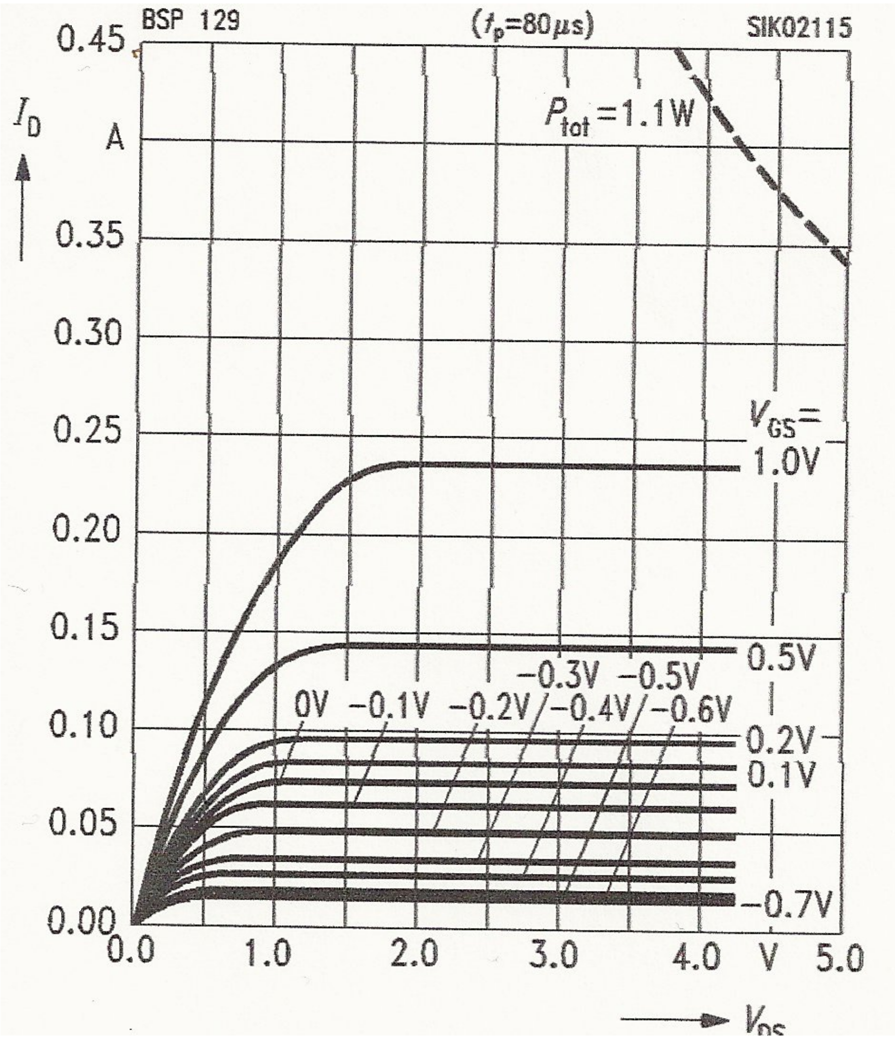
b) With an extra cathode follower plus bootstrapping



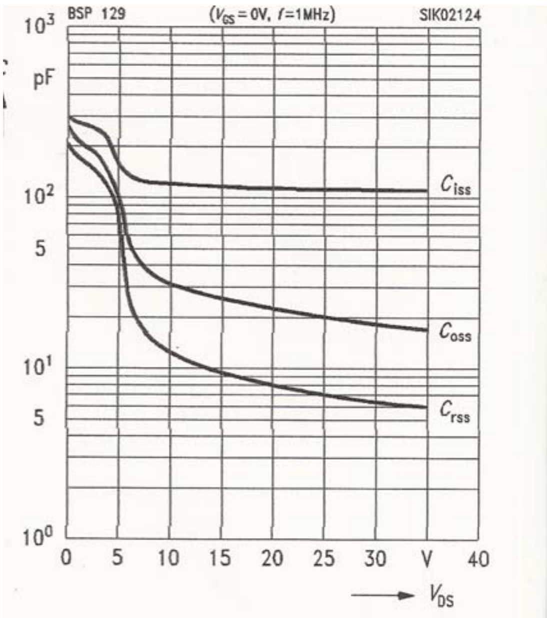
The main disadvantage of this approach is that the filament of the second tube is exposed to the high voltage on the tube's cathode, which requires special measures.
The advantage is that you kill two birds with one stone by eliminating the effect of R_L on the output.

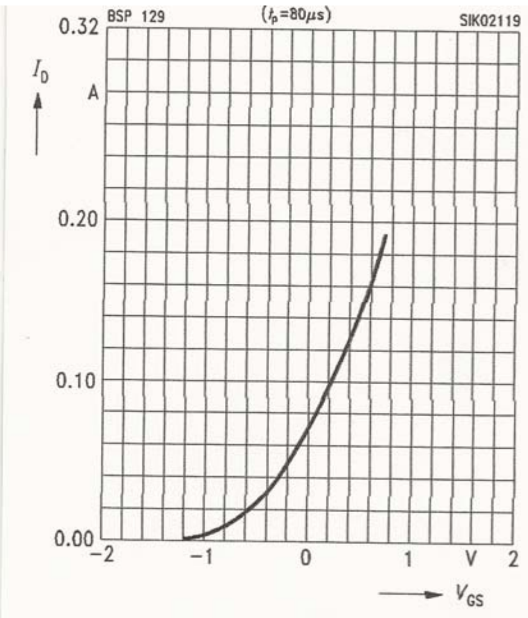
c) Constant current source with depletion-mode FET
[BSP135 (600 V) or BSP129 (240 V)**]**





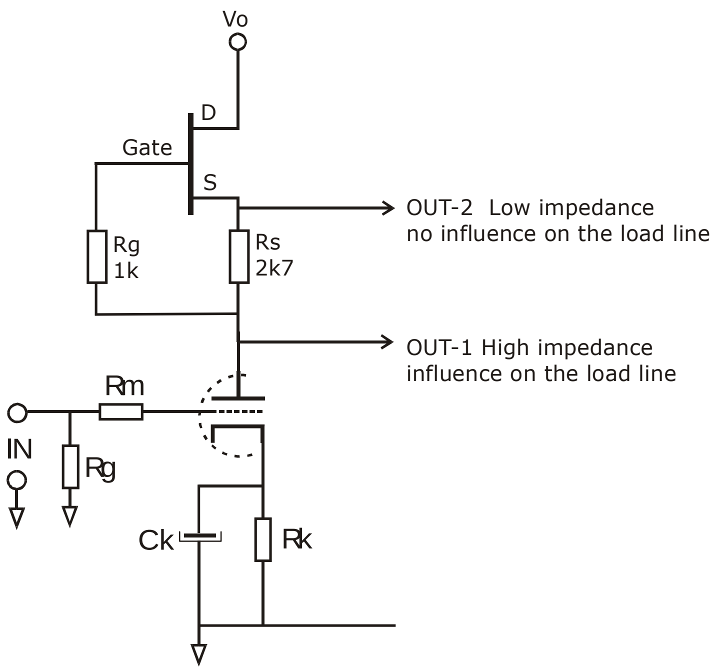
Internal capacitances can cause problems. R_g should be kept as low as possible.
The transfer characteristic is also not completely linear (see below).





See the many descriptions and discussions of CCS circuits on the Web for ways to correct these "deviations".

d) How do you couple the signal out?

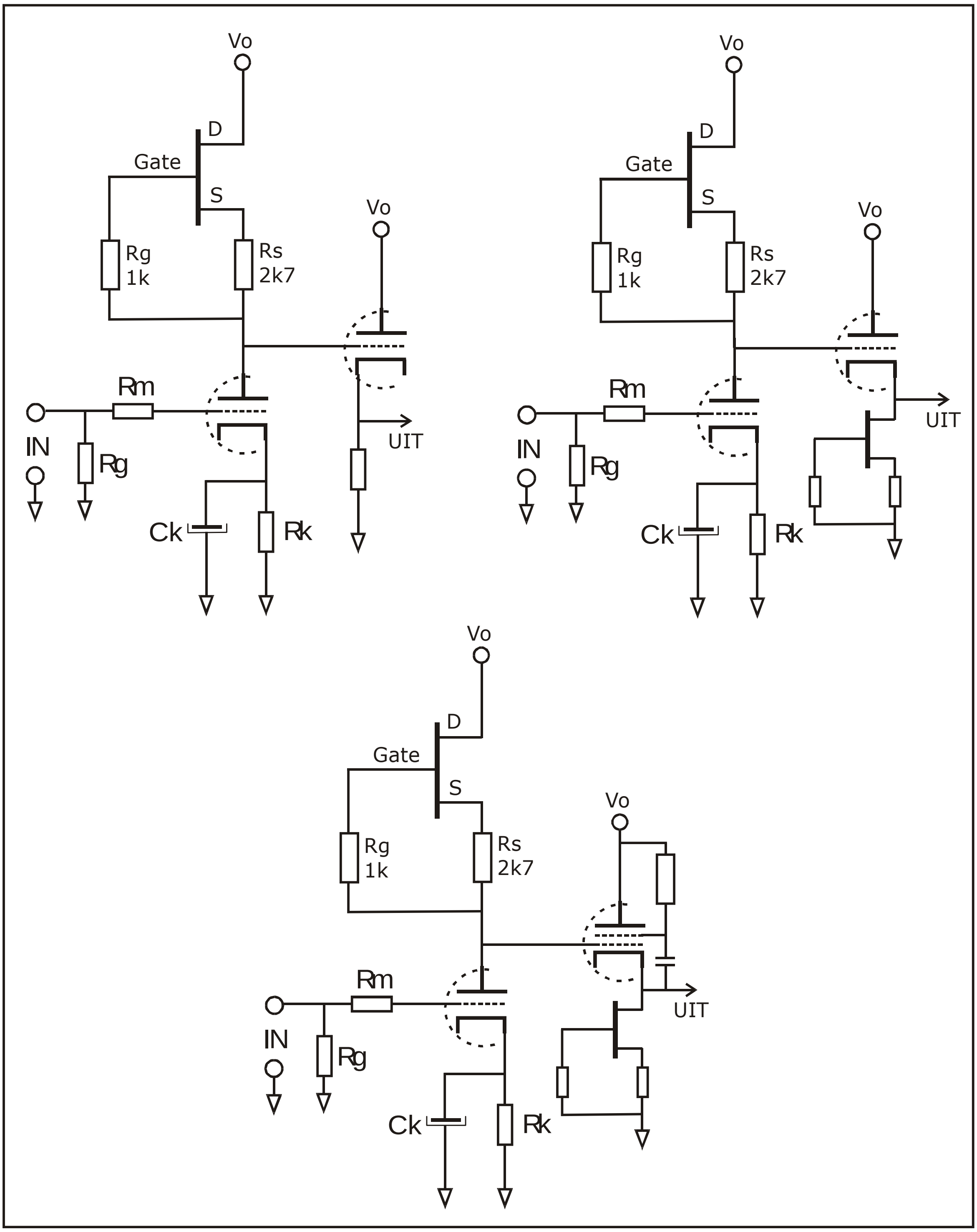


Decoupling R_k with C_k affects the IMD. It's better without C_k.

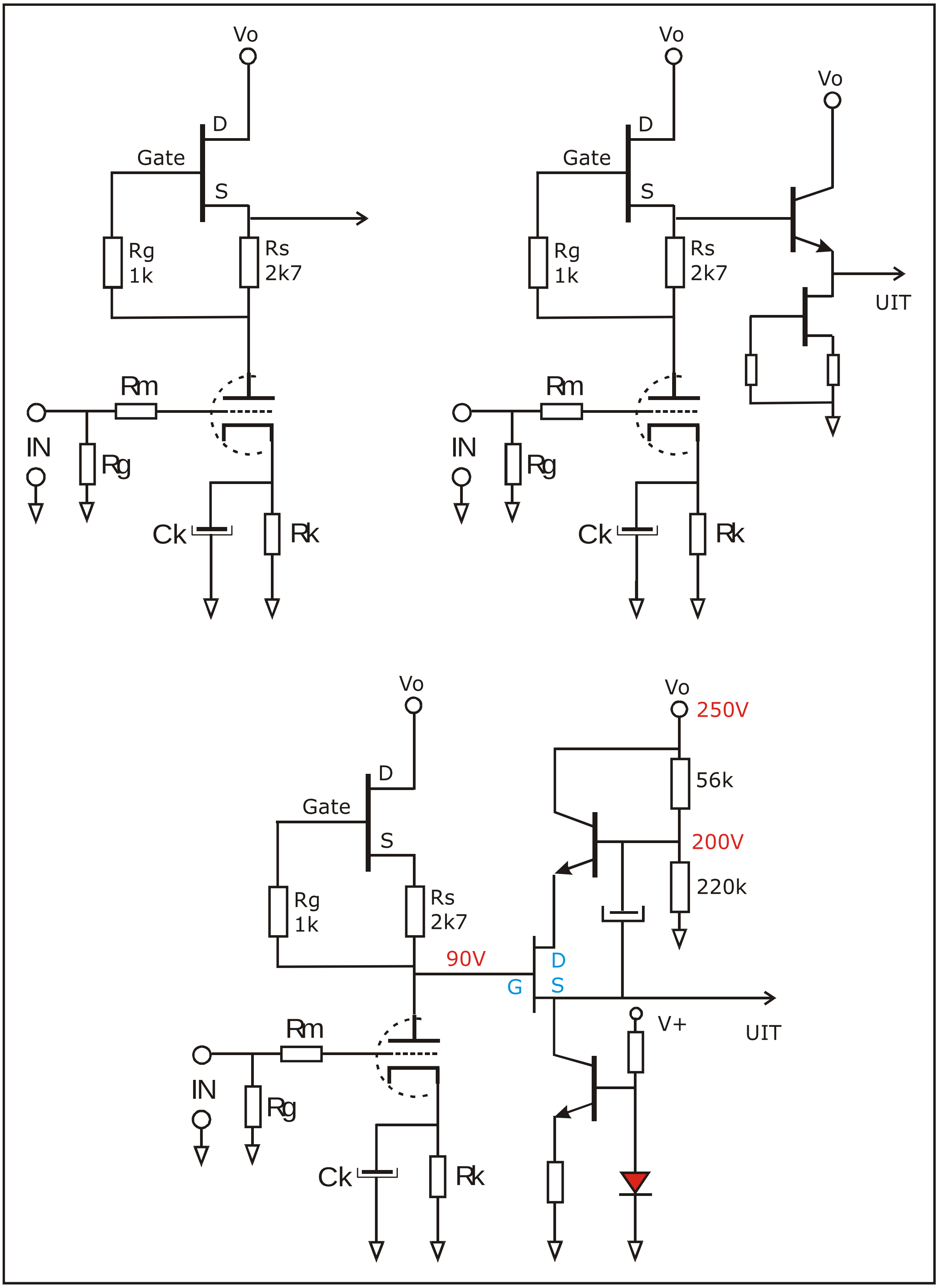
e) About impedance conversion

1) No load on the current source or the tube (otherwise the load line will not be horizontal)
2) This requires high load impedance for the current source and the tube
3) The impedance converter must have very high input impedance
4) The output impedance must be low enough to drive the load

Tube solutions

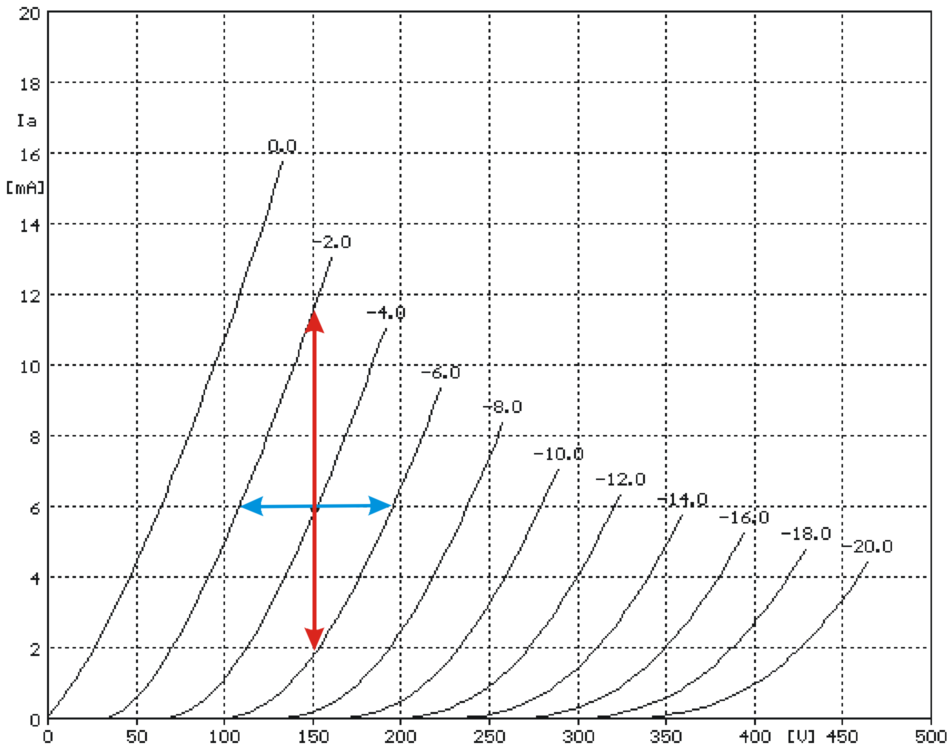


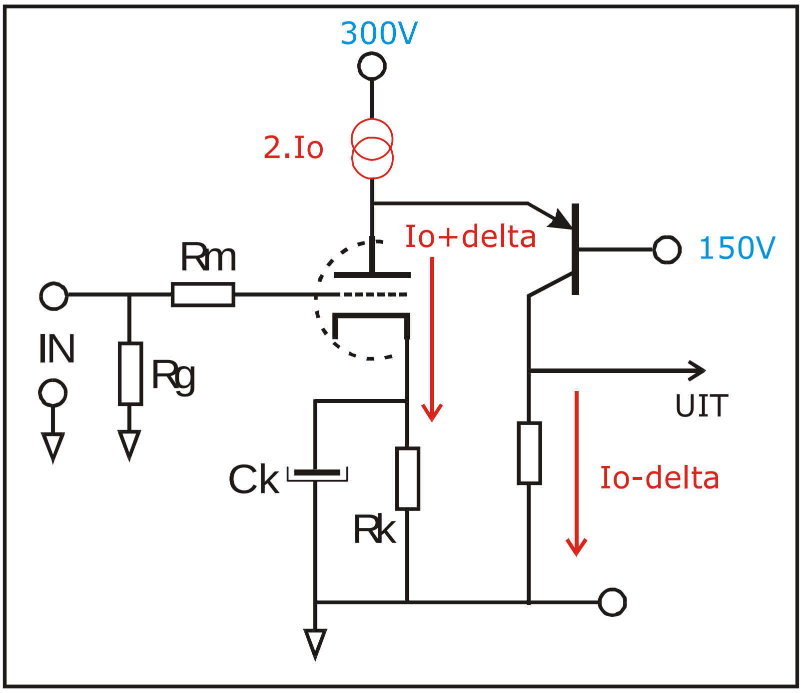
FET and bipolar transistor solutions



4) Modern approach with Folded Cascode

a) With V_a constant, the Miller capacitance effect is eliminated and the frequency range is increased.
b) A constant current source (2x I_o) eliminates supply voltage sensitivity.
c) A voltage follower reduces delta I on the ground rail to **zero**.





6) Literature and Web sources

The most informative sites are listed here. Of course there are lots more, but I know these ones and they are a source of inspiration.

www.tubecad.com: Broskie's site with fresh, modern ideas

www.pimmlabs.com: extensive descriptions of current sources

www.arcdb.ws: current sources in Audio Research amplifiers

www.mennovanderveen.nl: Dutch ==> tubesociety ==> project 2011

www.audiomagic: Peter van Willenswaard

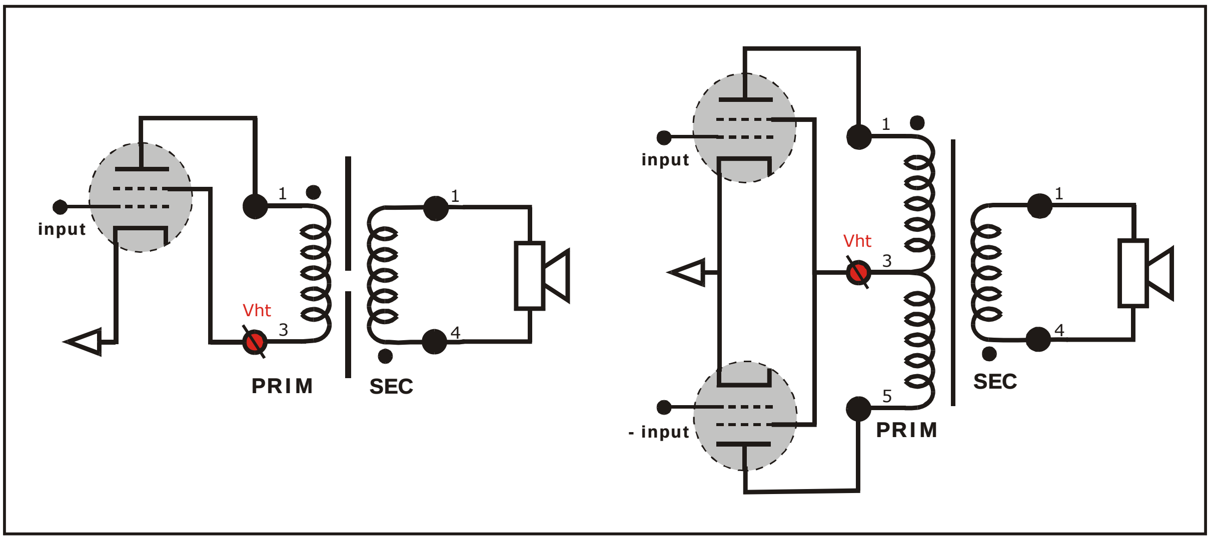
www.elektor.nl/srpp: Book by Peter Dieleman about SRPP and Mu-follower

www.tentlabs.com: Guido Tent

<u>**Output amplifier**</u>

1) Function of the output transformer

a) Converting high AC voltages on the tubes to low voltages
b) Converting low AC currents on the tubes to high currents
c) Safety – isolating high voltages
d) Very large frequency range (sound and character)
e) Avoid generation of internal distortion (sound and character)
f) Avoid loss of details (sound and character)



Brief explanation:

a) $V_s = (N_s / N_p) \cdot V_p$

b) $I_s = (N_p / Ns) \cdot I_p$

$==> Z_s = (N_s / N_p)^2 \cdot Z_p$, where $Z_p = 2 \cdot r_{ip}$ [with no feedback]

c) Minimum voltage isolation requirement: 4 kV

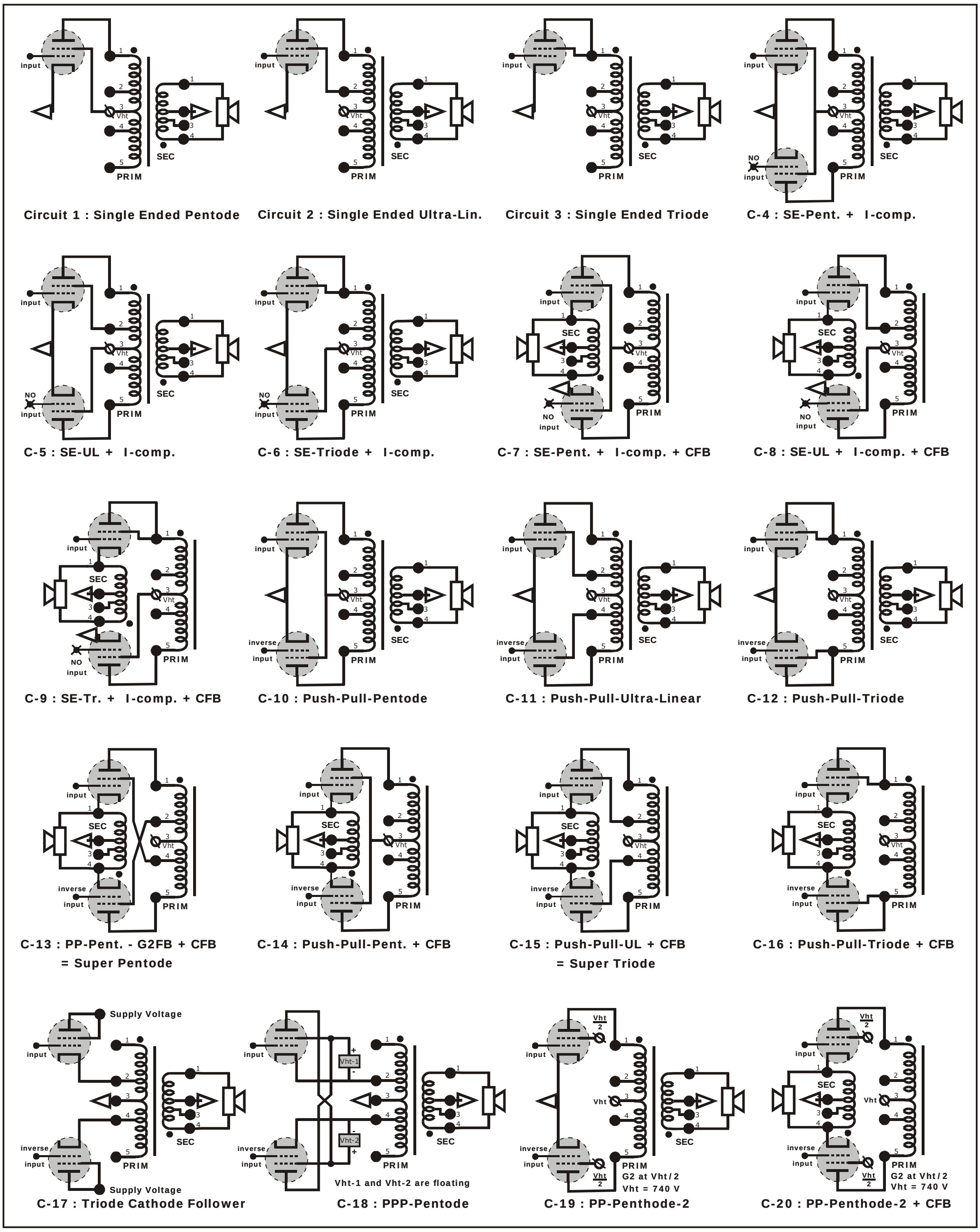
d) Low frequency rolloff: determined by L_p and Z_p
High-frequency rolloff: determined by L_{sp}, C_{ip} and Z_p

e) Core distortion occurs primarily in the low frequency range, due to non-constant relative magnetic permeability

f) Microdetails may be lost due to permeability collapse

See:
<www.mennovanderveen.nl> <publications> papers **7125** and **8360**

2) Output stage circuits (decisive for character)

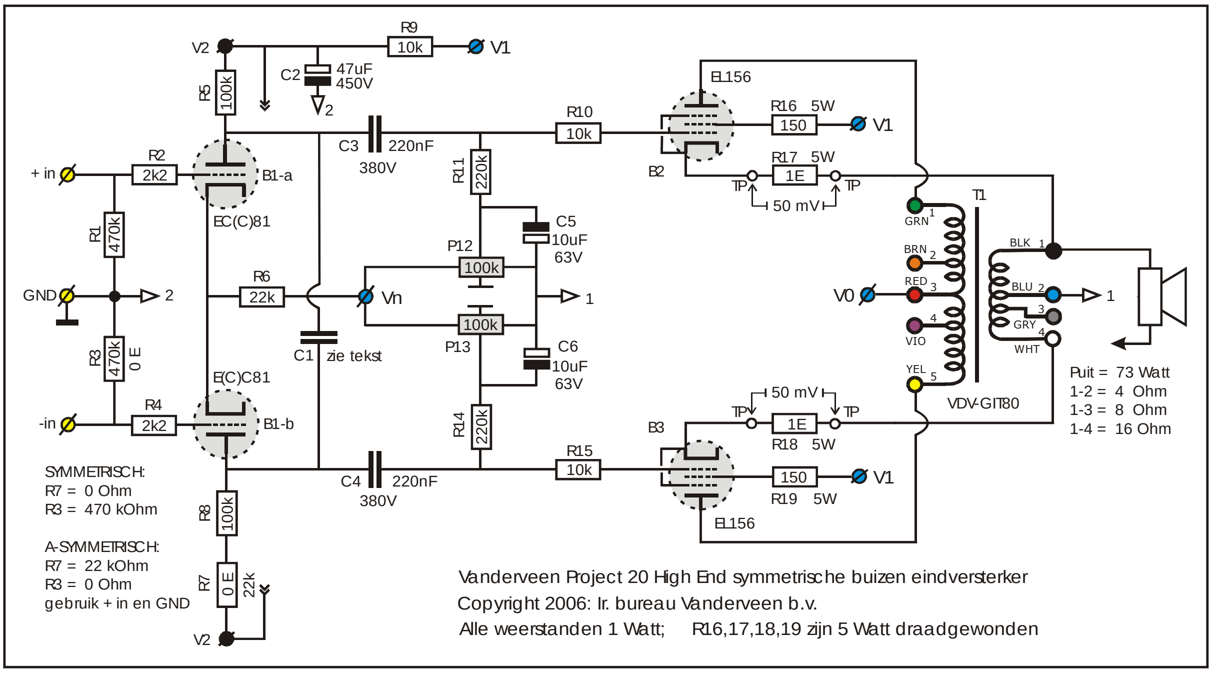


For more information see:

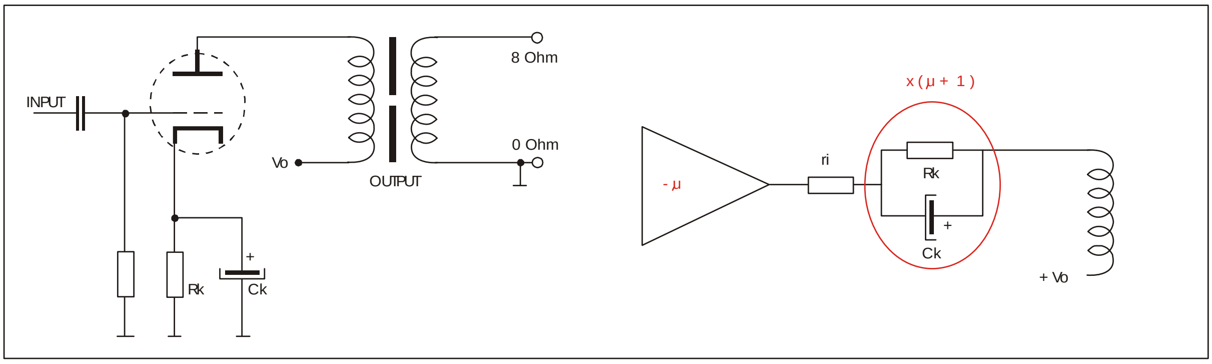
<www.mennovanderveen.nl> <**the project**>

<www.mennovanderveen.nl> <publications> paper **6347**

The Project 20 amplifier circuit (PR20-HE) is fully detailed and available as a kit. The enclosure was designed by Hans van Kastel.

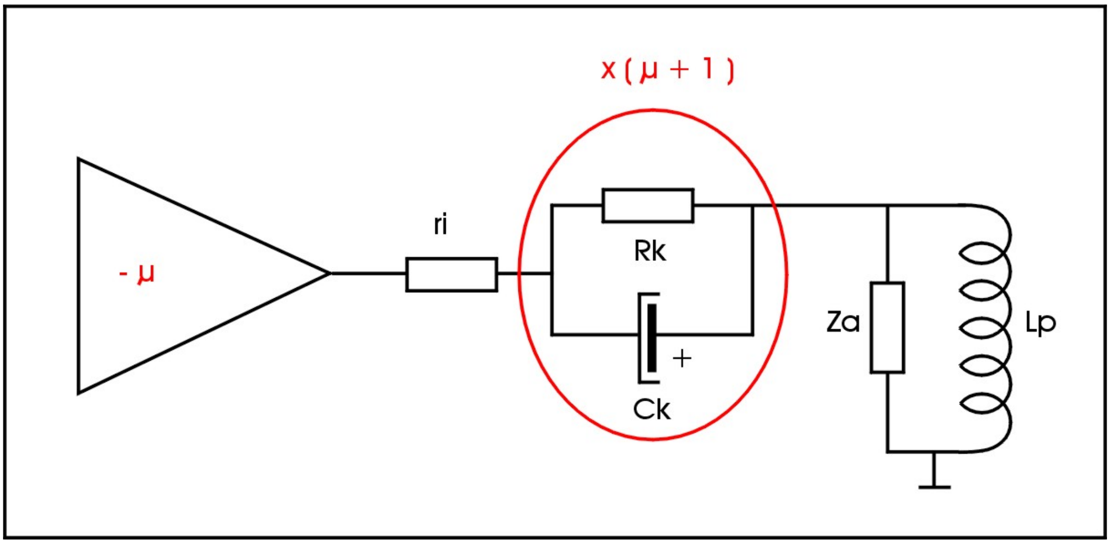


3) Disadvantages of cathode capacitors in SE amplifiers



At high frequencies C_k acts as a short circuit.

At low frequencies C_k forms a filter in combination with L_p (see below).



$$H(s) = -\mu \cdot \frac{s \cdot Z_a \cdot L_p}{s \cdot Z_a \cdot L_p + r_i \cdot \left(Z_a + s \cdot L_p\right) + (\mu + 1) \cdot \frac{R_k \cdot \left(Z_a + s \cdot L_p\right)}{s \cdot R_k \cdot C_k + 1}}$$

This is a second-order filter, and it causes extra phase shift. Especially with negative feedback, this makes the circuit prone to motorboating (slow low-frequency oscillation).

The following condition must be fulfilled to obtain a “clean” first-order filter (no effect on the sound):

$$C_k = \frac{L_p}{R_k \cdot Z_a}$$

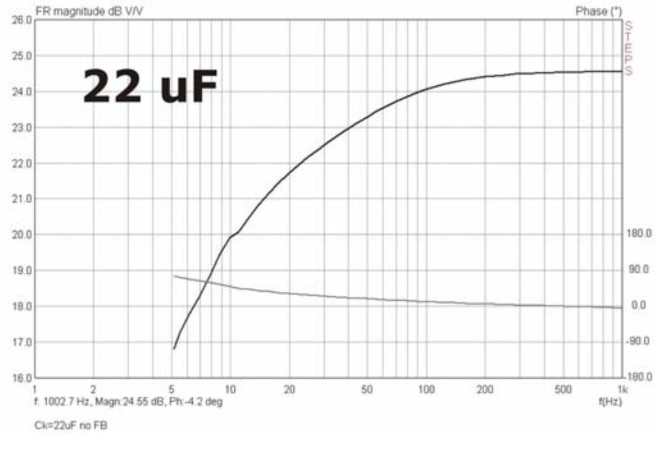
In this case the lower -3 dB frequency (f_{-3L}) is given by:

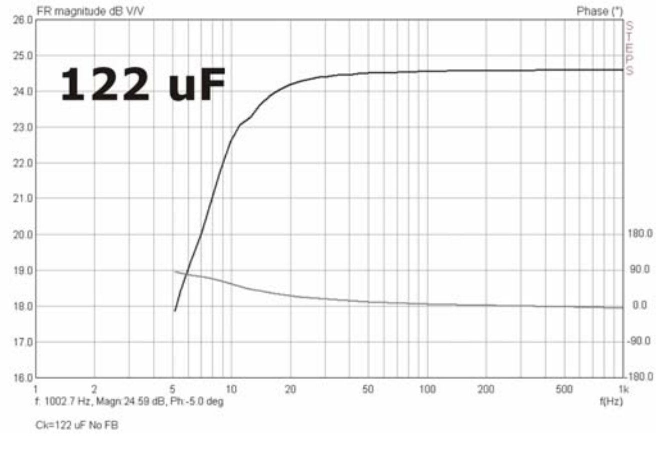
$$f_{-3\,L} = \frac{1}{2 \cdot \pi \cdot L_p} \cdot \frac{Z_a \cdot \left[r_i + (\mu + 1) \cdot R_k \right]}{Z_a + r_i}$$

The overall transfer function is now:

$$H(s) = -\mu \cdot \frac{Z_a}{Z_a + r_i} \cdot \frac{s}{s + 2 \cdot \pi \cdot f_{-3\,L}}$$

Example: L_p = 15 H, R_k = 150 Ω, Z_a = 4.5 kΩ, r_i = 680 Ω, μ = 10
==> C_k = 22 μF and f_{-3L} = 20 Hz





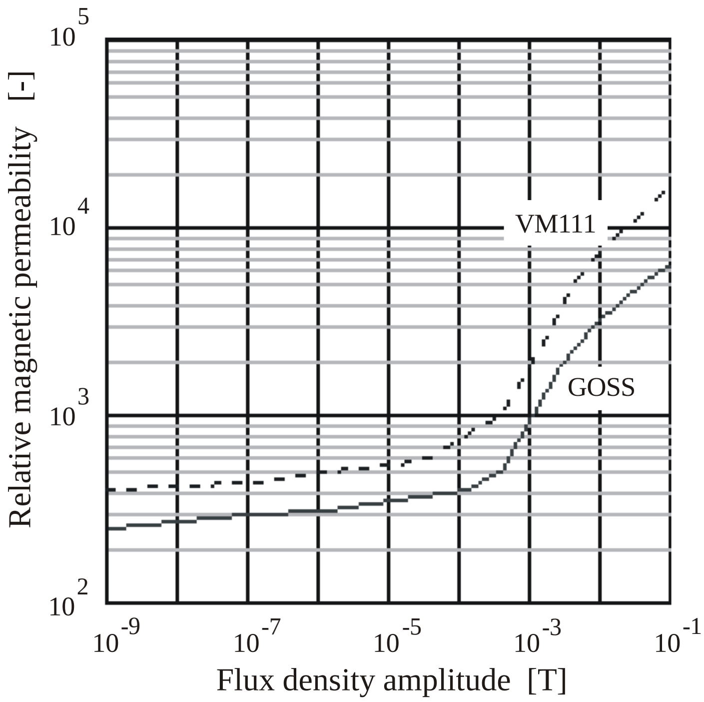
4) Disadvantages of cathode capacitors in PP amplifiers; remedies

The previous calculation is not valid with PP amplifiers.

In an SE amplifier L_p is practically constant due to the air gap l_g in the core:

$$L_p = \frac{\left[\mu_0 \cdot \left(N_p \right)^2 \cdot A \right]}{\left(l_g + \frac{l_c}{\mu_r} \right)} \qquad (4)$$

In a PP amplifier the output transformer core does not have an air gap. As a result, L_p is strongly dependent on the magnetic permeability, which means that L_p is strongly dependent on the output power.



Measured value of μ_r from 10^{-8} T to 10^{-1} T; extrapolation below 10^{-8} T.
The maximum measurement error below 10^{-3} T is 20%.

This makes it impossible to assign C_k a specific value. Consequently:

Biasing the output tubes with an adjustable negative bias voltage (fixed bias) is the best approach in this case (no additional time constant in the cathode)

An active auto-bias circuit, such as the Vanderveen / Tenlabs AutoBias circuit, can also be used for this.

5) Looking into the black box

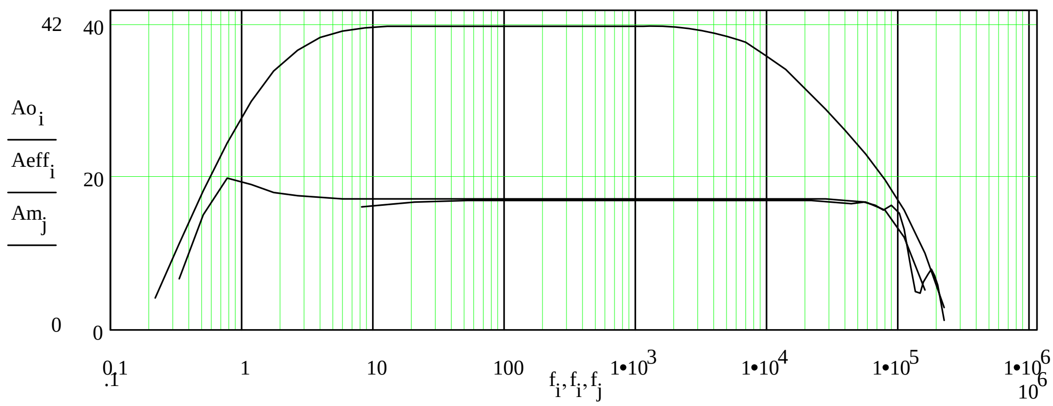
When you measure the external characteristics of an amplifier, such as the gain and frequency range, you cannot really **see** what is happening inside. Even if the frequency characteristic is a nice straight line from 3 Hz to 200 kHz, it's difficult to determine whether negative feedback is used in the amplifier, how the -3 dB corner frequencies are determined inside the amplifier, or whether the transformer is able to reproduce microdetails properly.

In this section I discuss several measurements that do give some insight into what is happening inside the black box, without requiring you to open it up or know anything about the schematic diagram.

5-a) Measuring the output impedance

Ao is the open-loop gain without negative feedback.
Ac is the closed-loop gain with negative feedback.
The difference between Ao and Ac is used to:
===> suppress distortion;
===> reduce the output impedance.

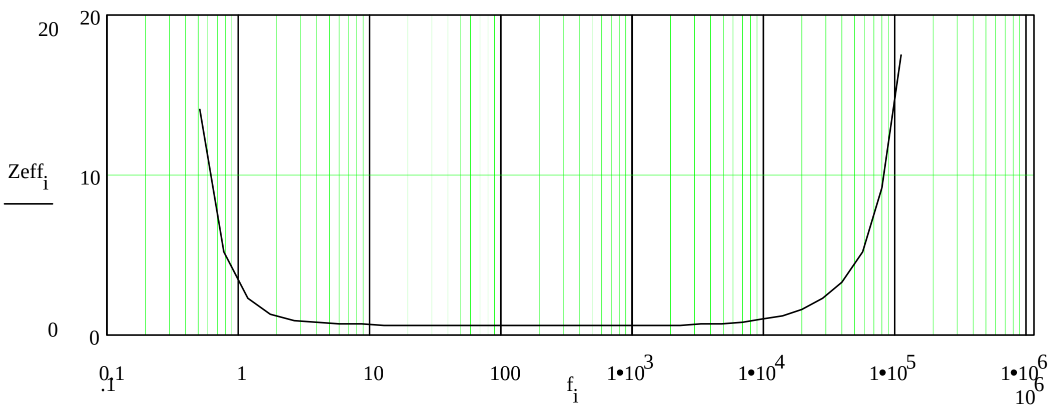
See the following example (Quad II amplifier):



Transfer functions in dB: upper curve is calculated without NFB;
lower curves are measured and calculated with NFB

Only the bottom curve (**Ac**) can be measured at the amplifier output.

The output impedance that you measure at the amplifier output is the transformed output impedance of the output tubes, reduced by the negative feedback ratio (Ac/Ao). This yields the following measurable plot:



Effective output impedance (absolute value) of the amplifier with NFB for $L_p = L_{p,max}$

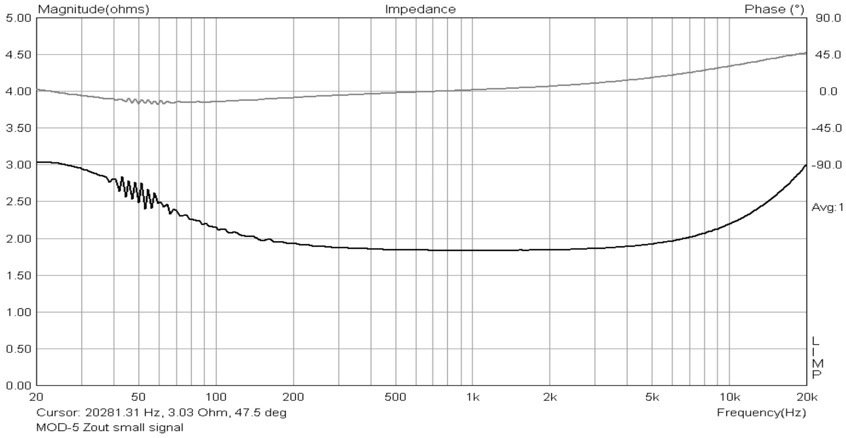
Here it is abundantly clear that the output impedance is reduced less at low frequencies (below 5 Hz) because the difference between Ao and Ac is less at low frequencies (the difference is the amount of negative feedback that acts to reduce Z_{out}). A similar effect can be seen at frequencies above 10 kHz.

Accordingly, if you measure Z_{out} versus frequency you can see directly whether the open-loop gain is constant over the measured frequency range.

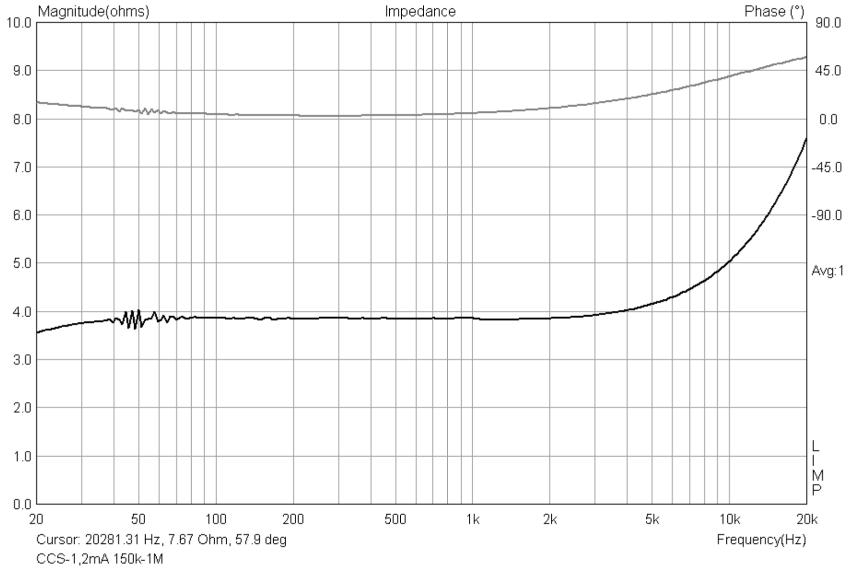
If Z_{out} versus frequency should be constant over the frequency range, for example from 20 Hz to 20 kHz, then the amplifier needs to have an open-loop bandwidth (-3 dB frequency range) of at least 20 Hz to 20 kHz. This is a very difficult requirement for tube amplifiers to meet, especially at the low-frequency end. It is usually not met, which is why tube amplifiers have relatively weak bass reproduction.

(see www.mennovanderveen.nl | Publications | paper 5748)
(see also *High-end Valve Amplifiers 2: New models and applications*, Chapters 4 and 5; ISBN 978-0-905705-90-3)

The following plot shows the measured output impedance of the Aurexx Crystal 1 amplifier:
(see www.mennovanderveen.nl | Dutch | TubeSociety | Project 2011, MOD-5)



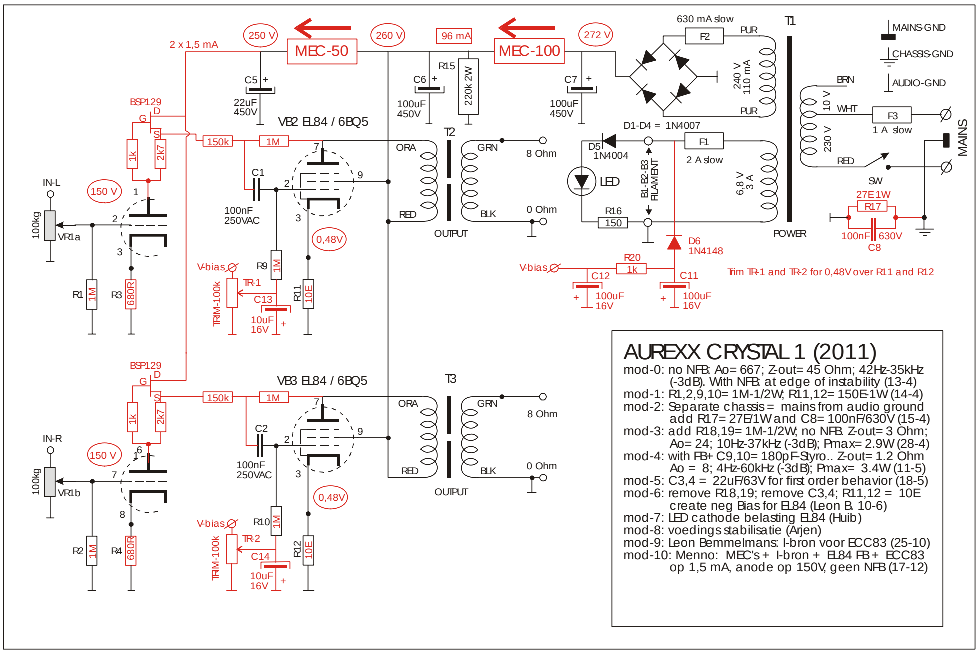
The increase in Z_{out} at low frequencies disappears if the quiescent current bias of the output tubes (in the Aurexx amplifier, for example) is replaced by an adjustable fixed bias, as shown by the following plot:



www.mennovanderveen.nl | TubeSociety | Project 2011
Mod-10 Aurexx Crystal 1

The increase in Z_{out} at high frequencies results from the effect of the Miller capacitance of the EL84 tube used in the output stage of this SE amplifier (which is effectively in parallel with the 1 MΩ negative feedback resistor for the EL84).

See the following schematic:



Aurexx Crystal 1, MOD-10

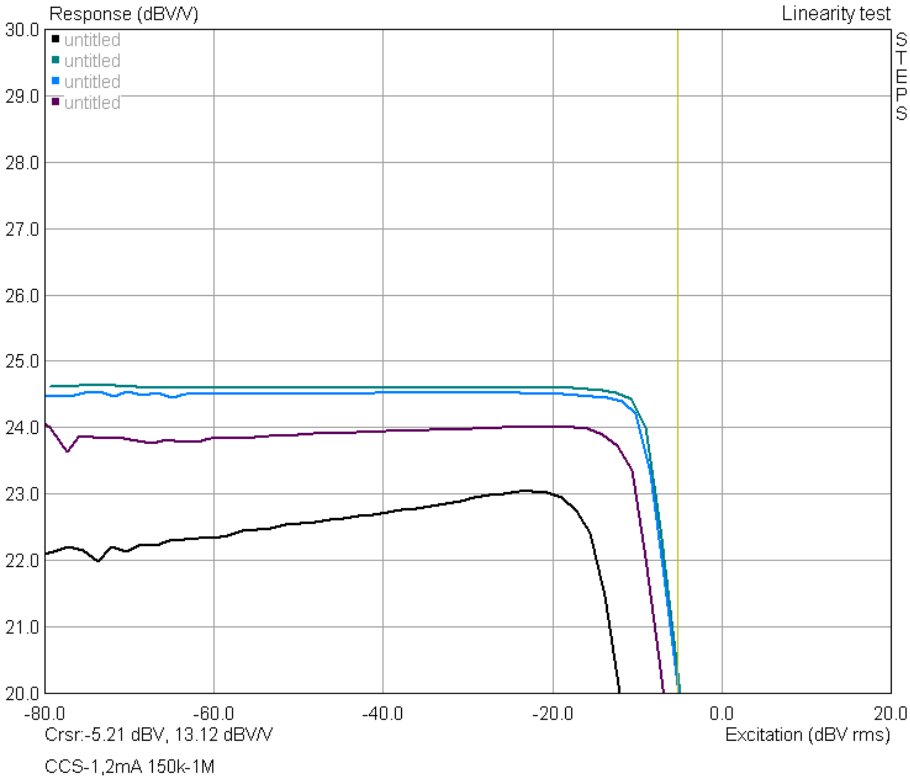
You can also see more things from the output impedance measurement:

a) Z_{out} drops even more at very low frequencies. This is because the primary self-inductance (L_p) of the output transformer starts acting like a short circuit:
$Z_p = (2 \cdot \pi \cdot f \cdot L_p) \parallel ((N_p/N_s)^2 \cdot Z_L)$, and L_p is not especially large (only 15 H); here $Z_L = 8\ \Omega$ (loudspeaker impedance)

b) At high frequencies the frequency characteristic is a straight line up to 30 kHz or more. Only the Z_{out} versus frequency curve reveals that the Miller capacitance causes the open-loop gain to drop even more at high frequencies (f_{-3H} is approximately 7 kHz under open-loop conditions).

5-b) Measuring gain linearity

You can measure the gain (V_{out} / V_{in}) versus Vin at various frequencies. The following figure shows an example of this type of measurement. Here the input voltage (in dBV) is plotted on the horizontal axis and the gain (in dBV/V = 20 log (V_{out} / V_{in}) is plotted on the vertical axis.



www.mennovanderveen.nl | TubeSociety | Project 2011,
Mod-10 Aurexx Crystal 1.
From top to bottom: 1 kHz, 70 Hz, 20 Hz, 10 Hz

Here you can clearly see that the gain is nearly constant at 1 kHz and 70 Hz. Overdrive occurs at the far right, leading to a loss of gain.

A slight variation in gain (on the edge of what is acceptable) can be seen at 20 Hz, and at 10 Hz the deviation is unmistakable. However, this falls outside our normal range of hearing, so this amplifier is not that bad.

This measurement gives you a very deep view into the metallic core of the output transformer. The variation in gain here (the slope of the 10 Hz curve) is **not** caused by the coupling capacitors. Instead, the slope results from the fact that the self-inductance L_p is relatively low with small input signals, so it acts like a short circuit. The higher the input voltage, the larger the relative permeability of the core material. This means that L_p increases with increasing signal level, so the short-circuit effect is gradually reduced and effectively disappears when Vin approaches its maximum value.

You can therefore use this measurement to determine whether the core material of the output transformer has sufficiently good characteristics to fully reproduce sound details, without having to resort to opening up the enclosure and dismantling the transformer. A lot of information about this can be found in previous publications.

(www.mennovanderveen.nl | Publications | Papers **7125** and **8360**)
(Elektor Audio Special 2010, in Dutch only)

Summary

Measuring the output impedance gives you insight into the open-loop and closed-loop gain and associated frequency ranges, and it tells you why tube amplifiers are not good at taut bass reproduction.

Measuring the gain linearity gives you deep insight into the behavior of the core material of the output transformer and shows you whether the core has sufficient permeability to reproduce microdetails correctly at all frequencies and all signal levels.

Conventional measurements, such as frequency range and distortion, do not provide this information. In particular, the method of simply making measurements at 1 kHz can be regarded as ~~totally~~ obsolete.

6) About acceptable distortion

We can distinguish between **linear** distortion (deviation from a straight-line frequency characteristic) and **nonlinear** distortion (harmonic distortion, IMD, transient distortion, etc.).

6-a) Linear distortion

There are two types of listening: technical and emotional. Each of them has its own merits. Here I want to talk about **sensitive emotional listening**. In this regard it strikes me that I find the sound more pleasant with a low damping factor than with a high damping factor. I find the latter too tight and dead, and it does not create any emotional perception, while a low damping factor stirs my emotions. What is the reason for this?

In addition to my scientific background, I have the subjective attitude of a guitar player looking for a particular sound. In this regard a straight-line frequency characteristic is entirely irrelevant to me. Instead, I regard the amplifier as a musical instrument which, in combination with the loudspeaker, produces a sound that I like.

I recently designed a new guitar amplifier that allows me to reduce the damping factor of the class-D output stage (Hypex) by using negative current feedback (causing Z_{out} to rise to 16 ohms), which produces a really nice sound. Together with the 6-inch loudspeaker (120 W), this results in a full sound stage with better articulation of deep low tones and bright high tones. How does this work?

At low frequencies approaching the resonant frequency of the speaker, the loudspeaker impedance rises sharply. At high frequencies the impedance also increases gradually due to the (uncompensated) impedance of the voice coil.

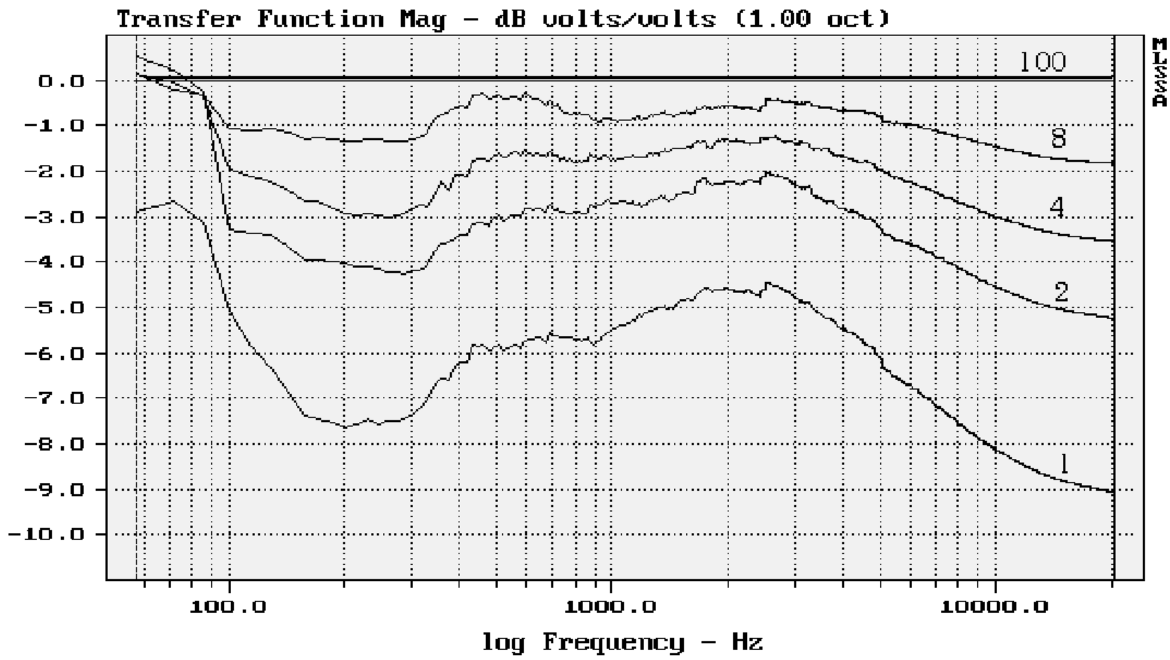
When this speaker is driven via the output impedance of the amplifier, the voltage on the speaker terminals is given by the formula:

$$V_{speaker} = V_{amp} \cdot \frac{Z_{speaker}}{Z_{uit} + Z_{speaker}}$$

If Z_{out} (= Z_{uit}) is zero, the voltage on the speaker terminals is exactly the same as the voltage V_{amp} from the amplifier.
If Z_{out} is not zero, $V_{speaker}$ depends on the frequency-dependent impedance of the loudspeaker.
In this case the impedance is high near the low-frequency resonant point of the loudspeaker, and it increases slowly at high frequencies due to the

self-inductance of the voice coil. As a result, deep low tones as well as bright tones at the top end of the frequency range are reproduced more strongly. This is exactly what you want as a guitar player (or at least what I want).

The following figure shows the results of acoustic measurements made with my MC3 loudspeaker driven with various damping factors, which support the theoretical explanation given above. You can clearly see that the acoustic radiation is strongly dependent on the impedance curve of the loudspeaker concerned.



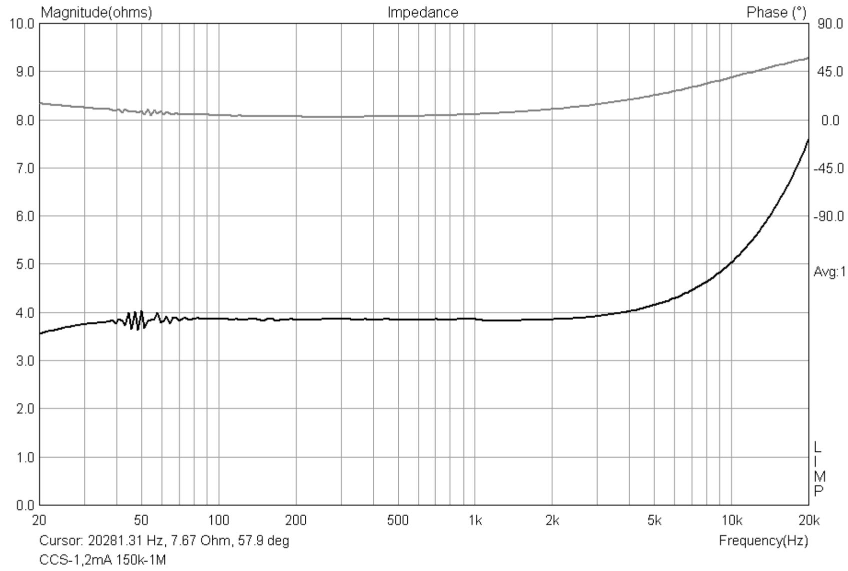
Acoustic characteristic curves of an MC3 loudspeaker driven by an amplifier with adjustable damping factor (shown at the right for each curve). The curves have been arranged vertically for ease of distinction.

What strikes me now as a high-end enthusiast is that I also apply this technique to high-end sound reproduction. There I also prefer tube amplifiers with a damping factor in the range of 2 to 4, simply because I find the resulting sound the most pleasant with my loudspeakers.

What do we care about a straight-line curve or a “straight wire” between the source and the speaker? What matters is what we hear, isn’t it?

(For technical listening, such as in a studio, the situation is completely different and rightly so, but here I am talking about listening at home.)

Another example on this subject: the previously mentioned Aurexx amplifier has a pleasantly clear sound with modification 10, which can be fully explained by the rising output impedance at higher frequencies. Have a look at the following measurement:



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Mod-10 Aurexx Crystal 1

One final remark on this subject: I have previously shown that microdetails can be lost in the output transformer. These minute details are replace by silence, so you miss important information.

Now let's consider the modern technology of mobile phones. My Samsung mobile phone with the Android operating system has a built-in noise gate that is **always** active, with the result that I can't hear the breathing of the person at the other end of the line. When I use this phone as a radio, the result is unbearable because the quiet passages in the music are artificially attenuated. This is unpleasant to listen to, and it surely is not what was intended.

I find this such a pity; it looks like technology and imagery have gotten the upper hand, while good listening has been made impossible. I'd rather have the old-fashioned noisy telephone line with a carbon microphone. I could get along with that much better (I heard more) than with modern digitally processed voice transmission. "Real" human comprehension is dramatically worse now.

I fear that this comes from ignorance – from a lack of understanding of important characteristics and conditions for "good deep listening".

Can we put this down to the "modern digital MP3 generation"?

Once again, I'm afraid the answer is yes. I experience this as a very serious step backwards and a significant loss of quality.

6-b) Nonlinear distortion

Distortion of any sort is very easy to hear and is undesirable in high-end reproduction.

Second-harmonic distortion creates a clear sound stage, but it also causes the sound stage to stick to the loudspeaker.

Third-harmonic distortion makes the sound stage grey and opaque.

Fifth-, sixth- and seventh-harmonic distortion make the sound stage shrill and irritating.

In short, minimal THD and an attenuated harmonic spectrum are highly desirable. This can be achieved by applying more and more negative feedback. The latest developments from Hypex (Ncore from Bruno Putzeys) let you see and hear how amazingly good this strategy is. They rightly follow the “straight wire” route, and we can be proud that this new technology was born in our country (and of course Belgium, as I know).

That’s all I want to say now about various types of distortion, but I have a few concluding remarks about correlation.

6-c) Distortion and relationships

Distortion can be reduced to very low levels by using negative feedback. This is a very desirable result.
Negative feedback also increases the damping factor, which I personally find less desirable.

I have noticed that I personally find that amplifiers with very low distortion (such as the Hypex Ncore) sound more pleasant when I use current feedback to artificially raise the output impedance while maintaining low distortion.

(see *High-end Valve Amplifiers 2: New models and applications*; ISBN 978-0-905705-90-3)

In this relationship you can see that negative feedback can produce both desirable and undesirable effects at the same time. Fortunately, there are means available to separate these two effects and deal with them independently.

Finally, as a result of my long experience I now find it easier to recognize relationships, and I like to experiment with these relationships. For this reason, I now prefer to talk about how things relate to each other instead of the abstract “truth”.