

# Concepts of the Diamond and Pyramide audio systems

Hans van Maanen (with many thanks to Ron Eijling and Ton Nahuijsen)



# Control amplifier

- Selection of 6 inputs using relays
- Volume- and balance control using relays and resistors (no distorting IC's in de signal path)
- Remote control
- Separation of analog and digital sections
- Distortions as low as possible for our hearing (will be discussed in more detail) --> computer modelling
- Output for interlinks of several metres length
- Output for headphone amplifier
- Output for recorder and as specified by user
- Design and programming of the digital section by Ron Eijling





# Goals

- System for the living room
- Different, special design
- Electronics integrated into systems (high WAF)
- Holistic approach: optimise the whole system
- No “sweet spot”, the same sound colour everywhere in the listening room
- Correct time response (impulse response), also for low frequencies
- Distortions as low as possible for our hearing
- Limited local and overall feedback
- High quality of the finish, freedom of choice for the customer



# Goals: choices

- “Diamond” and “Pyramide” shapes
- System with a 360° radiation pattern
- Only interlink and power cord to loudspeakers
- Active 3-way system
- “Weak” filters
- Correction of loudspeaker units in their enclosure
- Alternative suppression of distortions
- Impedance compensation of van loudspeaker units in their enclosure



# 360° radiator, why?

- Ideal loudspeaker “sighing sphere” ( $= 4\pi$  radiator)
- Radiation in horizontal plane at ears’ height ( $= 2\pi$  radiator) better than direct radiating loudspeakers
- Because then there is no “sweet spot” and/or “keyhole” effect
- Sound is the same in the whole listening room
- Indirect sound has the same (spectral) composition as the direct sound
- Therefore a more neutral, more natural sound
- Changeover between mid and high very “easy” because of the short distance between the “mirror images”
- Stereo-effect is different, it is disconnected from the loudspeakers





Reflection  
from  
tweeter

Reflection  
from  
squawker



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# Active vs. passive

- Passive: efficiency = “equalising downward”
- Less control of amplifier over the loudspeaker units because there always sits “something” between the units and the amplifier. The next slide shows an example of such a “something” .....







Example of a passive cross-over filter. Picture downloaded from Internet. **No TC product!**

# Active vs. passive

- Passive: efficiency = “equalising downward”
- Less control of amplifier over the loudspeaker units because there always sits “something” between the units and the amplifier. The next slide shows an example of such a “something” .....
- Complex interaction of filter with the impedances of the units

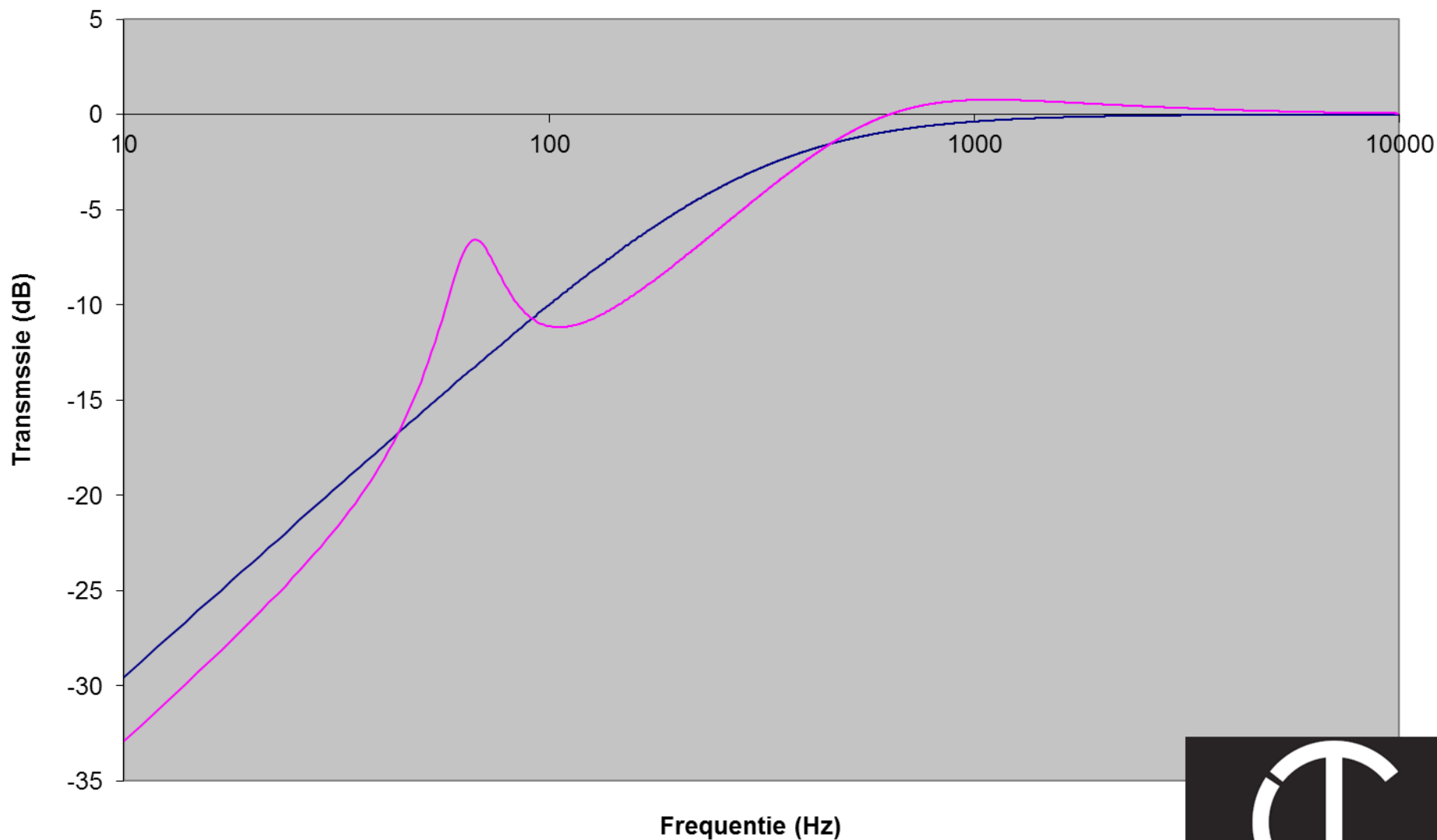


# Active vs. passive

In the next slide we see two curves. In blue the desired filter characteristic as would have been obtained with a pure Ohmic impedance of 8 Ohm. In purple the characteristic as is obtained with a more realistic impedance of a loudspeaker unit. Note the vertical scale: 5 dB/div!



# Filterkarakteristieken



— Constante impedantie

— Werkelijke impedantie



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# Active vs. passive

- Equalisation of efficiencies by resistors
- Can only be done correctly when all units are impedance compensated! And thus react as if these were pure resistors
- When we e.g. have to attenuate an “8 Ohm” unit by 3 dB, this can be done by placing a 2.34 Ohm resistor in series and a 19.4 Ohm in parallel to the unit (thus keeping the impedance 8 Ohm)
- But when the impedance varies between 6 and 30 Ohms (completely normal) this results in already 2 dB deviation
- A series resistor strongly reduces the damping factor!
- This also has negative consequences for the temporal response



# Active vs. passive

- Amplifiers are used in “wide band” mode
- Distortion products are transferred to the mid and high range units by the passive cross-over filter
- Amplitude- vs. Power addition -->  
higher peak power required  
 $4\text{ W low} + 4\text{ W mid} + 1\text{ W high} = ??\text{ W}$   
 $= 25\text{ W}$  in the peak when the amplitudes are in phase

(see also Elektor audio special nr. 5 and paper on our website)





# Active vs. passive

- Active: each unit has its own amplifier --> maximum control, minimal losses
- Filtering can be optimised for the units (in their enclosures) and their temporal response
- Amplifiers are used in “narrow band” mode --> less misery
- Misery cannot be transferred to the other units because there is no electrical connection
- Less power required because of a reduction of amplitude vs. power addition
- Power can be used for loudspeaker correction
- *Active, hurdles can be overcome which you cannot handle with passive designs!*



# Choices made

## Each audio system is a compromise!

- Choice between 2, 3 of 4-way system. With 2-way the units have to handle much at the same time, 4-ways is very hard to get time-correct because of complex filtering and differences in transit times -->

3-way system is the best compromise (we think)

- Each unit has its own “acoustic box” (compartment)
- Acoustic box can give a is time-correct response, an open enclosure cannot
- Loudspeaker unit in an acoustic box is a high pass filter, but with known properties --> correction possible



# Choices made for filters

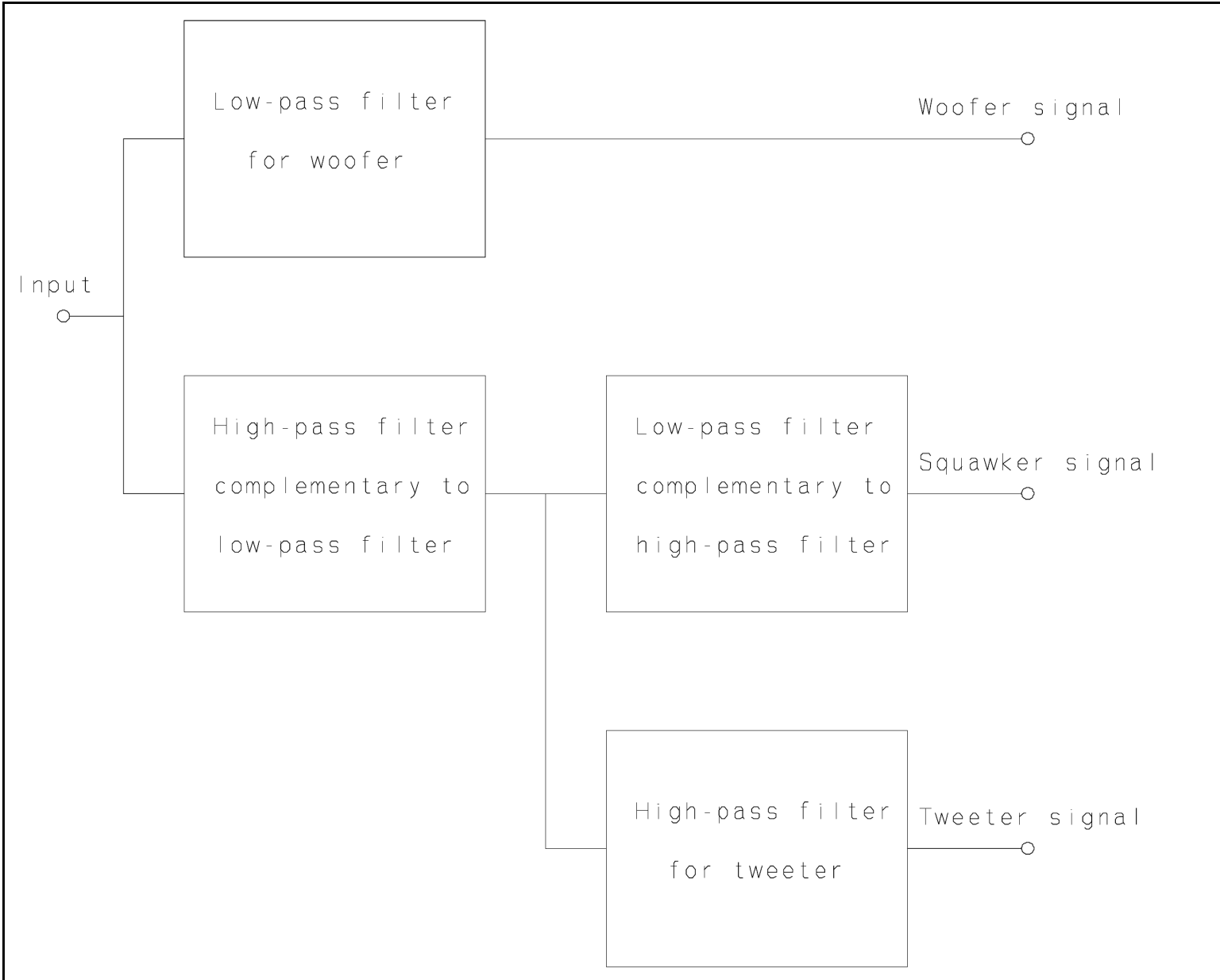
- Causal filters and correct temporal response
- “Weak” filters are then the only option left
- 2<sup>e</sup> order low pass for woofer
- 2<sup>e</sup> order high pass for tweeter
- 1<sup>e</sup> order high and low pass + correction for squawker
- For details: RE publication from 1979 on our website



# Choices made for filters

In the next slide the block diagram of the filter is shown: the sum of the output signals of the two most left-hand sided blocks is equal to the input signal, the sum of the output signals of the two most right-hand sided blocks is equal to the output signal of the lower left-hand side block. In this way, the sum of the three signals is equal to the input signal.



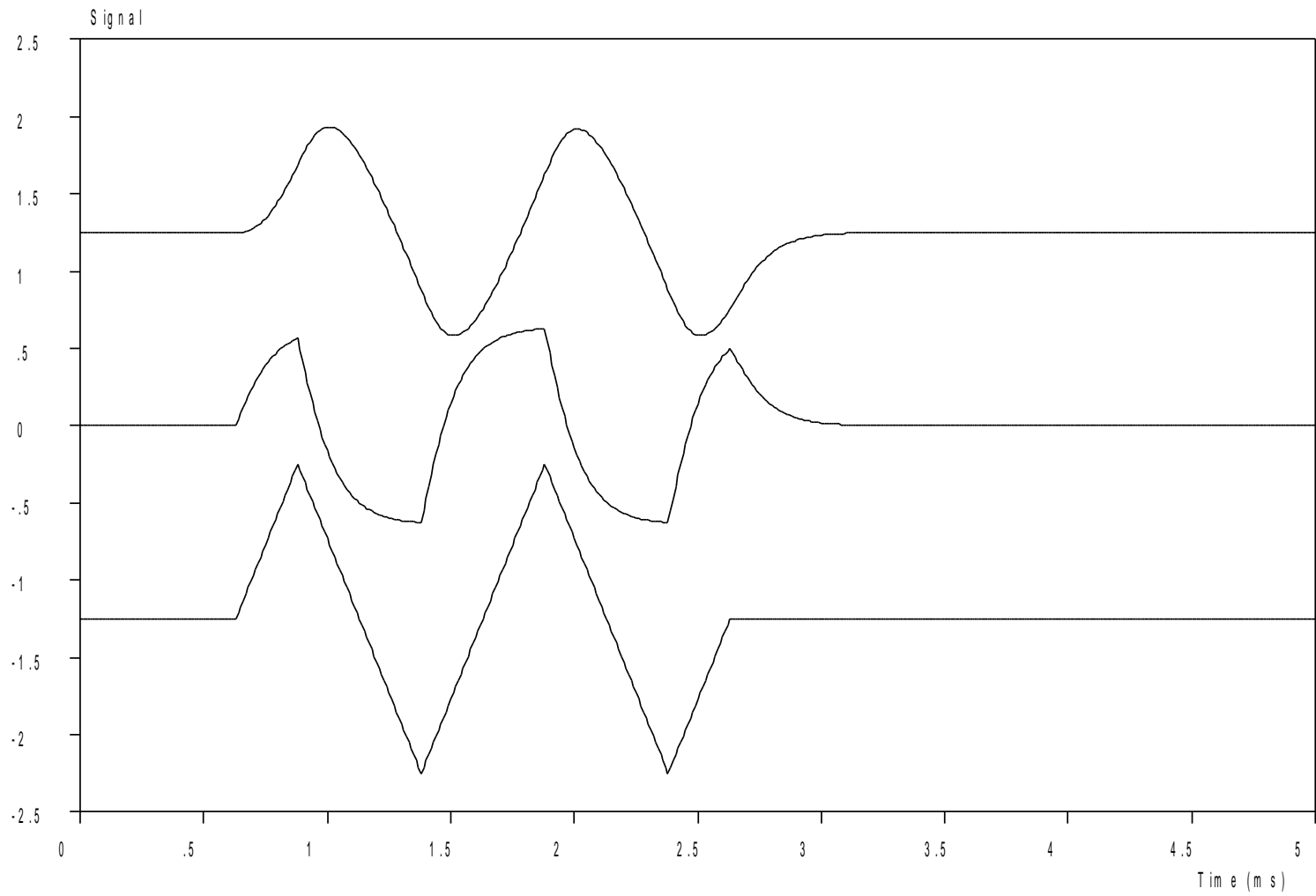


# Choices made for filters

In the next slide, the output signals of the two most left-hand blocks are shown. The upper trace is the output signal of the upper block, the middle trace is the output signal of the block below it. The lowest trace is the sum of both and is identical to the input signal.

Note that the filters are *causal*: They don't generate any output signal before any input signal is present as is possible with digital filters. All broadening in time domain (a direct consequence of the Fourier theory) occurs at the end of the signal and is not noticeable because of the normal decay of sounds.

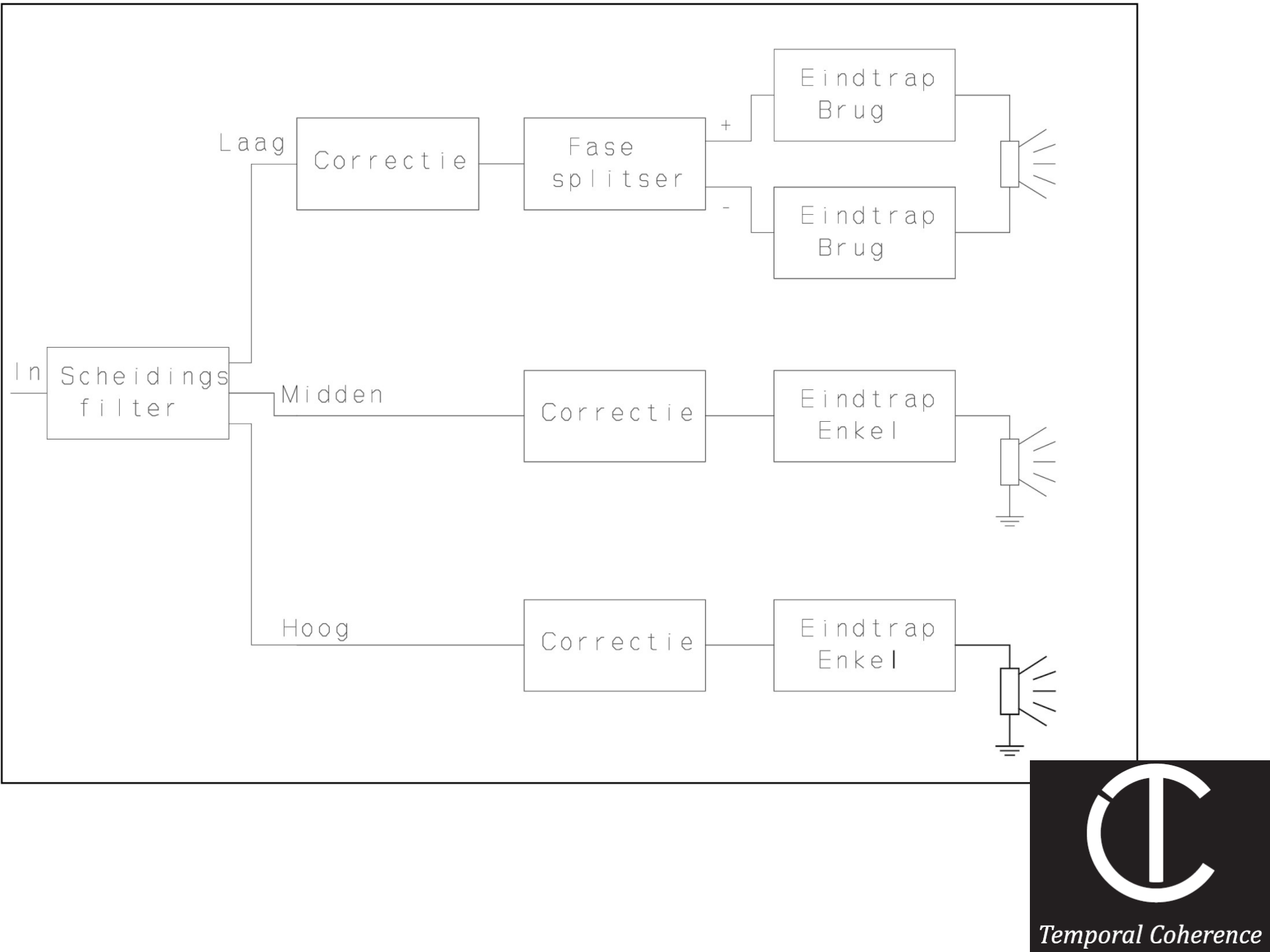




# Choices made for filters

The next slide shows the complete block diagram of the filter including the corrections for the loudspeaker units in their housing. The phase splitter (which generates a phase and an anti-phase signal) is required for the use of two power amplifiers in a bridge configuration (to make more power available to compensate the decrease in efficiency at the lowest frequencies).





# Why correction of loudspeakers?

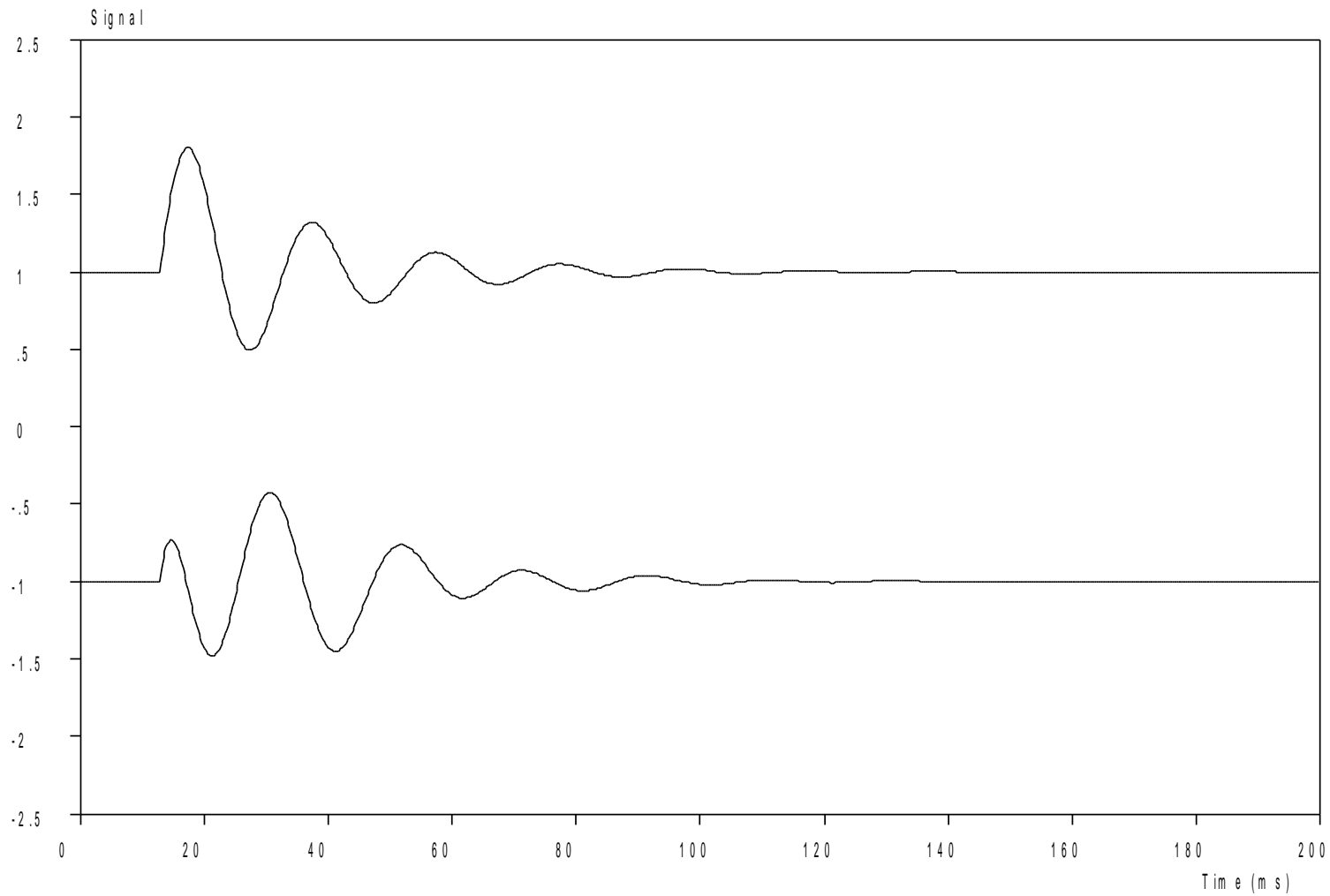
- A loudspeaker in its housing is a filter
- It influences both amplitude and phase
- Without correction no correct temporal response



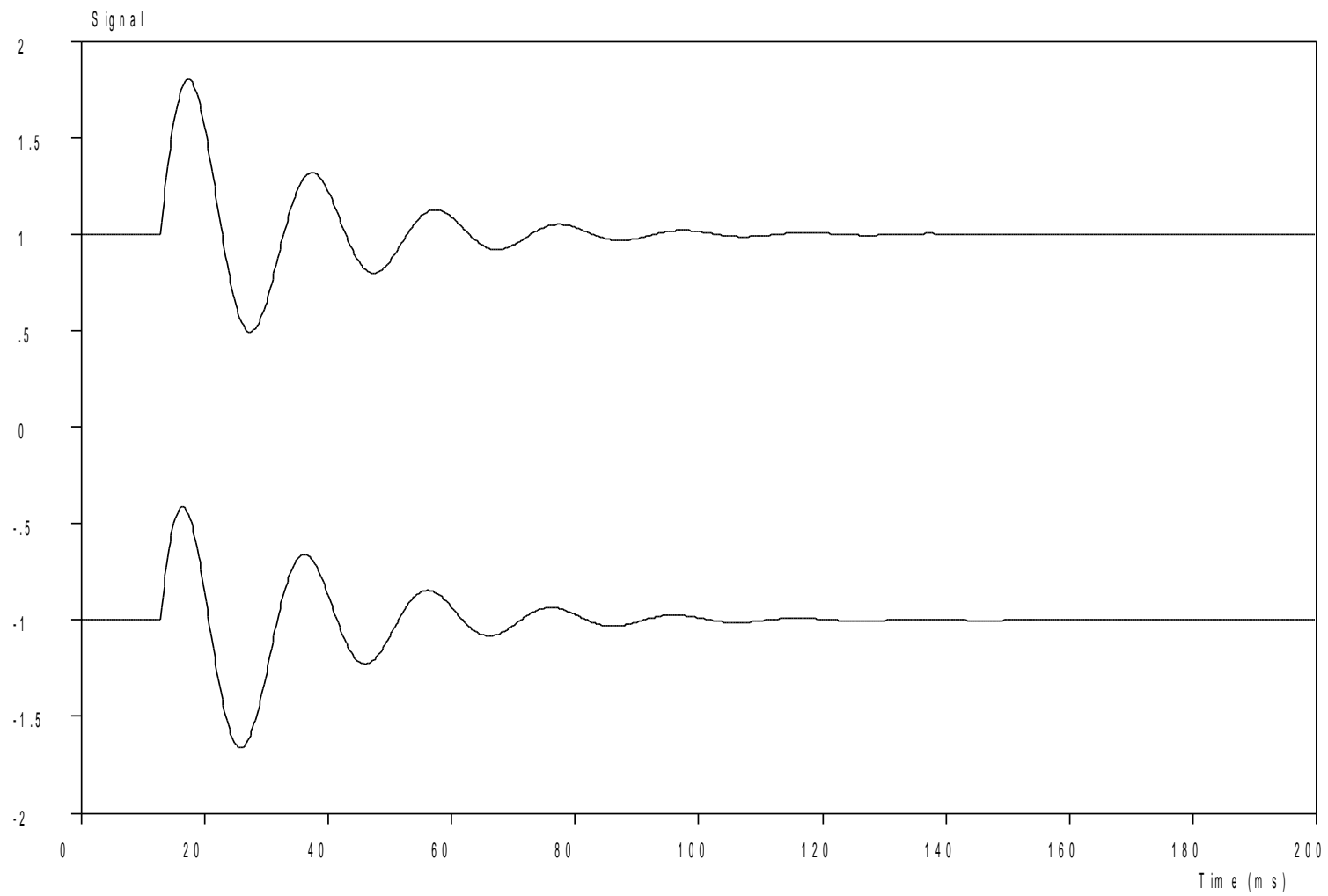
# Why correction of loudspeakers?

The next slides show the response of a base-reflex system to a simulated hit on a cattle drum (first slide) and the response of a corrected acoustic box to the same signal (second slide). Note that the base-reflex system needs time to reach its maximum strength and responds almost in anti-phase.









# Why correction of loudspeakers?

- A loudspeaker in its housing is a filter
- It influences both amplitude and phase
- Without correction no correct temporal response
- Broadening of frequency response
- Reduced efficiency remains, of course
- This costs extra power, no problem in 2014
- Certainly no objection in an in active system



# Distortions and distortions are many

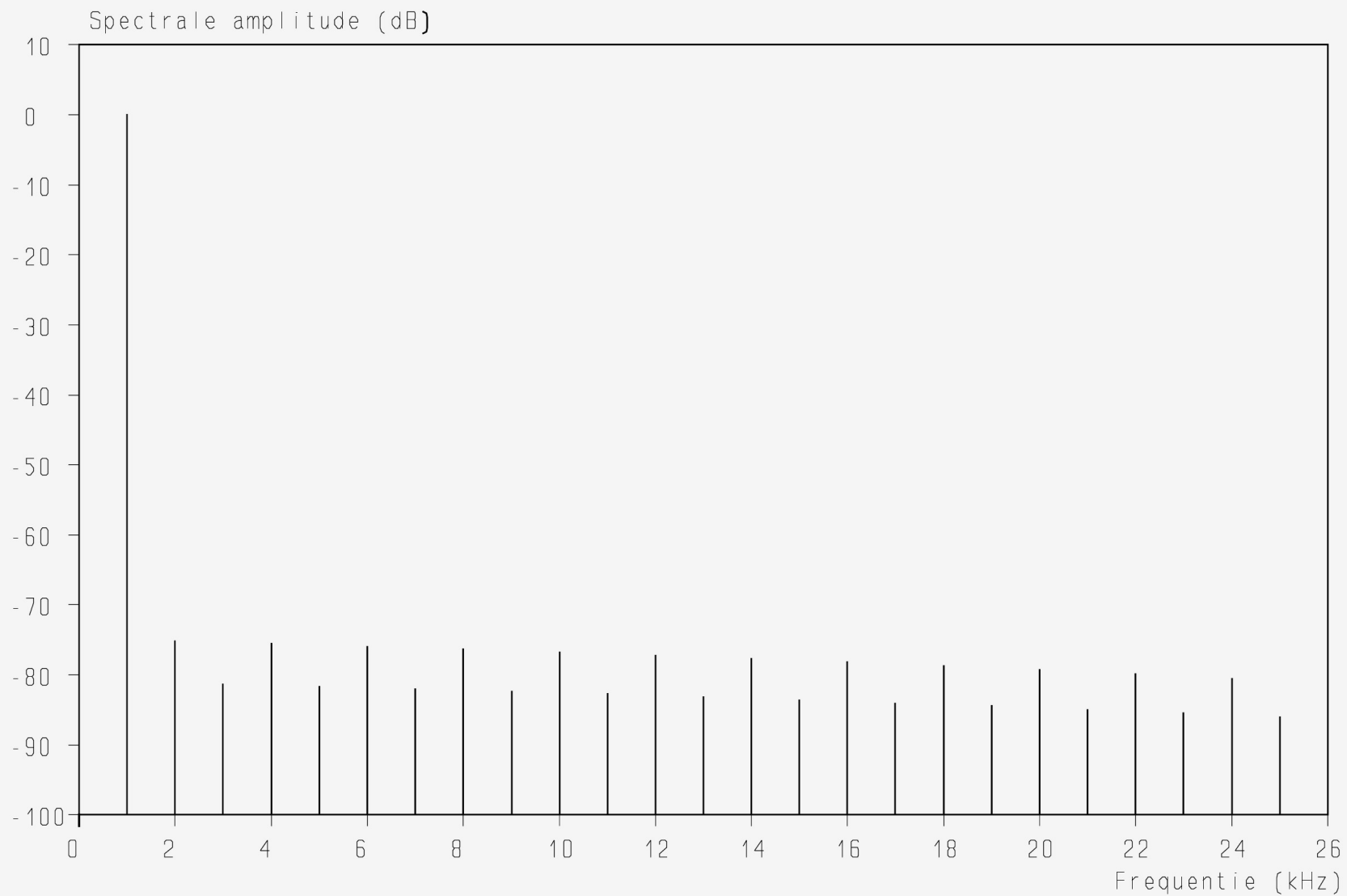
- Distortions (non-linear) are much more complex than many think and cannot be “caught” in a single number
- Example: the numbers for distortions of loudspeakers are significantly higher than those of semiconductor amplifiers. Yet, these distortions can clearly be heard using such loudspeakers
- When designing the electronics, it is necessary to take the properties of human hearing into account and use this to optimise the design
- E.g. the spectral distribution of the distortion products is important because of the masking of our hearing and the harmonics, generated by instruments



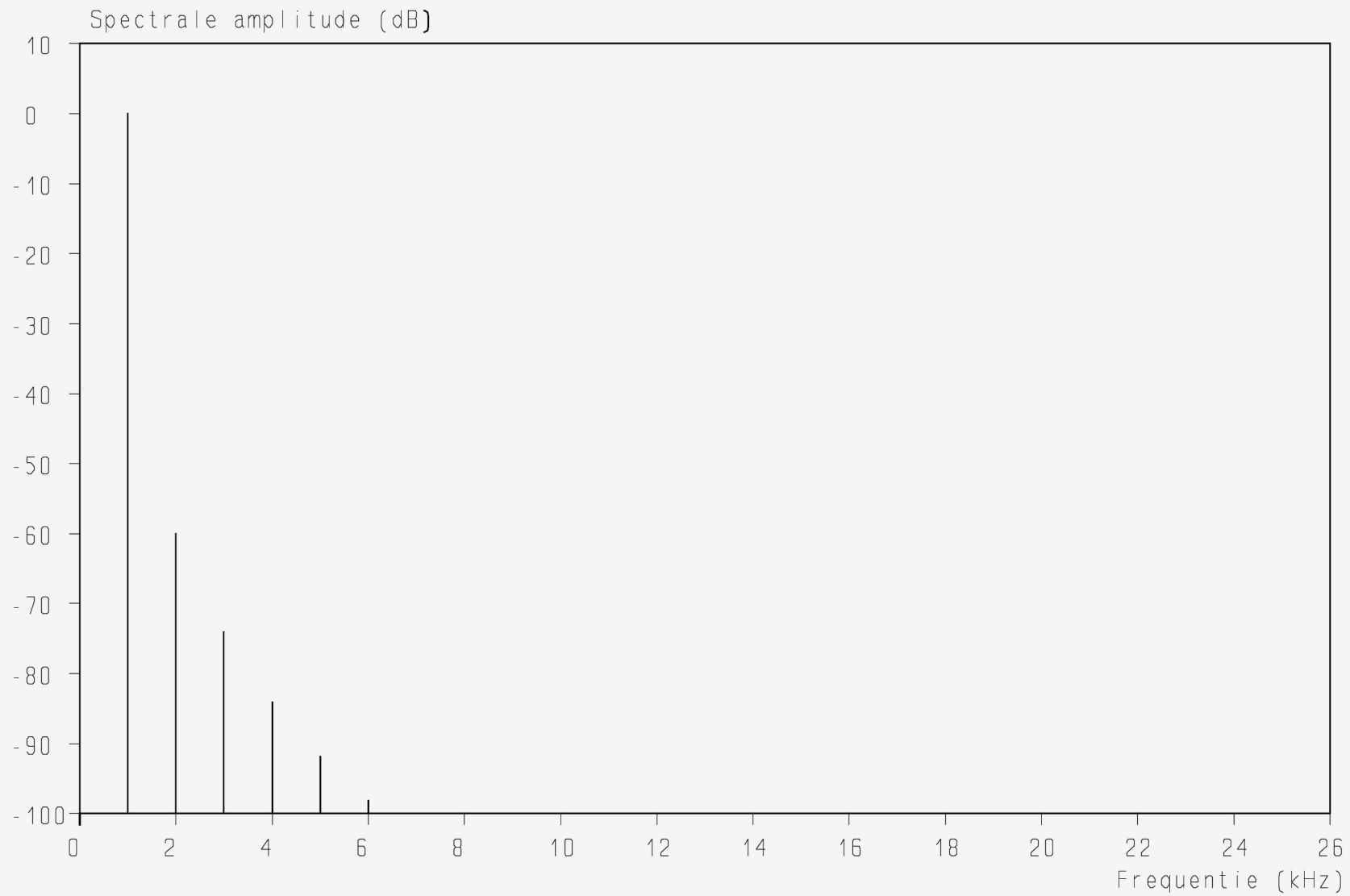
# Distortions and distortions are many

The spectral distribution of distortion products can differ strongly between circuits, loudspeakers and filters. The next two slides show two strongly different spectral distributions. In the first slide, we see a low level of the distortion products (relative to the 1 kHz tone which generates these), but continuing to high frequencies. The second slide shows distortion products which are stronger near the 1 kHz tone, but rapidly decrease with increasing frequency. Now the question is which of the two is the least disturbing for our hearing.





*Temporal Coherence*



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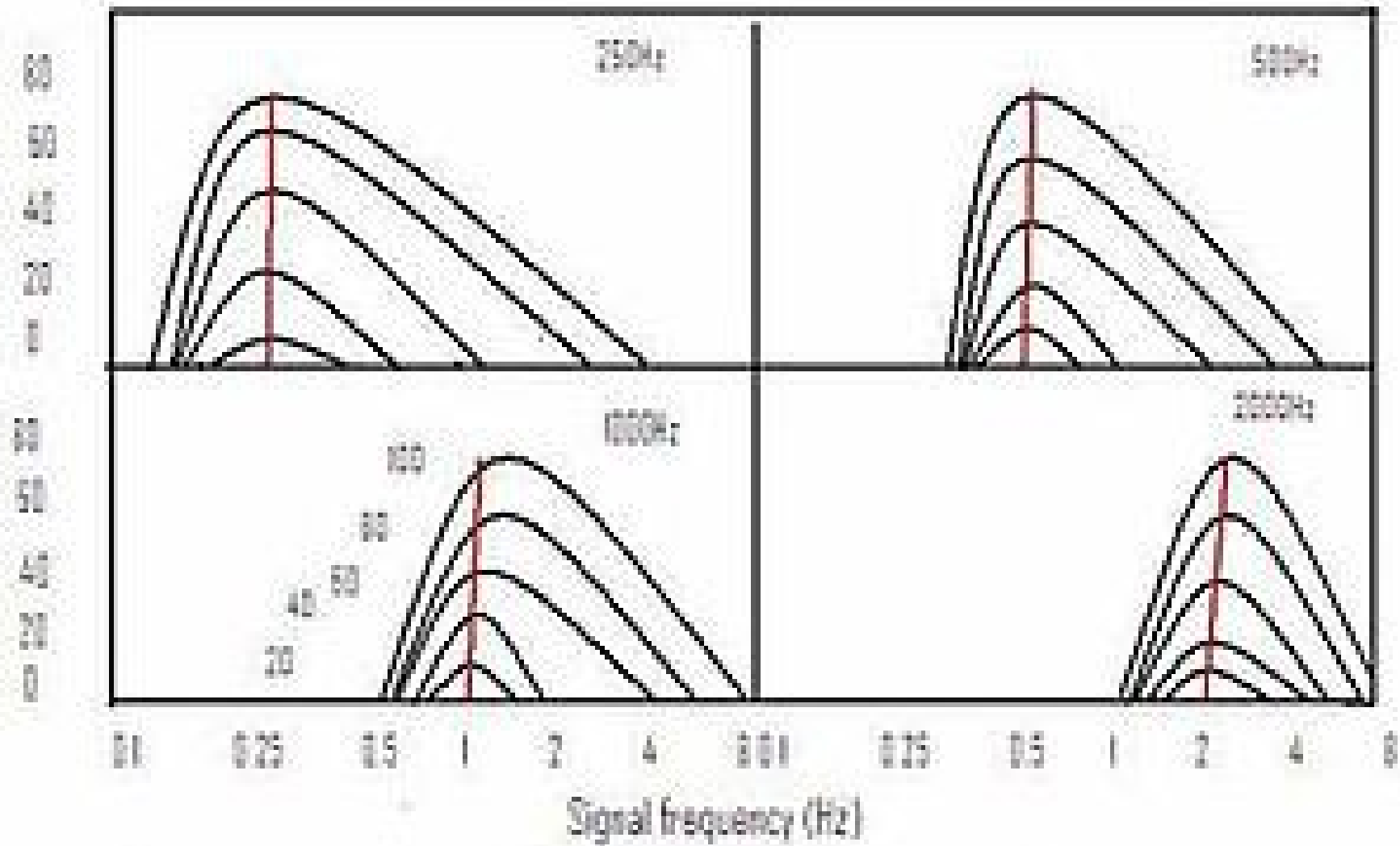


# Distortions and distortions are many

The next slide (downloaded from the internet) shows the masking of our hearing. The red line indicates the masking tone, the black curves show the masking at different levels of the masking tone. These show that harmonics, close to the masking tone, are masked much more strongly than those further away. This explains why the distortion of the second slide, albeit stronger, is less disturbing than that of the first slide.



Masking  
dB



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# Distortions and distortions are many

- Feedback is no “miracle cure”, but it can be helpful to give the “final touch”
- The feedback factor cannot be increased “ad infinitum” because of stability and bandwidth
- We will define the feedback factor as the ratio between the open-loop gain and the closed-loop gain of the amplifier
- It can be expressed as a number or in dB's



# Limitations to the feedback factor

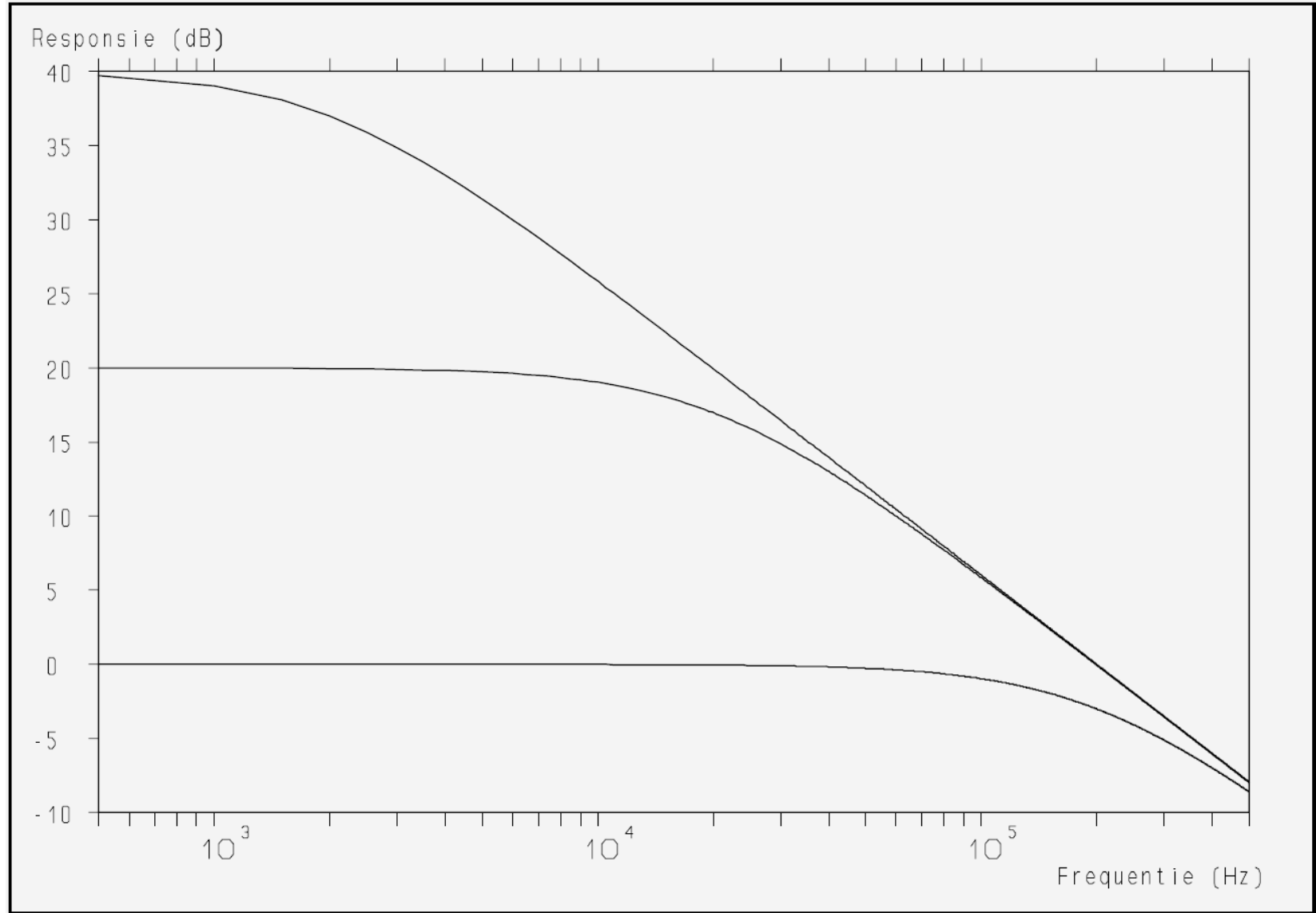
- To keep systems with feedback stable, (preferably) a slope of -6 dB/oct should be introduced into the open-loop gain
- As a consequence, bandwidth and feedback factor are coupled:  
 $\text{Bandwidth} * \text{Feedback factor} = \text{Constant}$
- You don't want a too wide closed-loop bandwidth (problems with radio signals of long and medium wave band transmitters)
- E.g. closed-loop bandwidth 200 kHz: feedback factor at 20 kHz  
 $= \text{closed-loop bandwidth} / 20 \text{ kHz} \approx 10 = 20 \text{ dB}$
- It is to be preferred to keep the feedback factor constant over the entire audio bandwidth (distribution of harmonics and intermodulation products)



# Limitations to the feedback factor

The next slide shows three graphs. The lowest one is the closed-loop gain of the amplifier, the middle curve is the open-loop gain of an amplifier with a feedback factor of 20 dB, the upper one is the open-loop gain of an amplifier with a feedback factor of 40 dB. Note that with 20 dB feedback a constant feedback factor over the entire audio range is obtained, whereas with 40 dB the feedback factor decreases from 2 kHz upward. This promotes an undesirable distribution of distortion products.





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# Limitations to the feedback factor

- Don't increase the feedback factor to a level that the amplifier is mainly processing error signals
- The theory of feedback is based on “or, or, or”, but in reality it is “and, and, and”!
- Feedback can provide excellent results as long as you know what you are doing
- Alas, often designers use it sinfully and then feedback is blamed
- High quality amplifiers require feedback in one or other form
- An in-depth analysis of designs learns that almost all the time feedback has been used, even in so-called “non-feedback” amplifiers, it is just deeply hidden



# A different way to fight distortions

- Track down the root causes of distortions and fight it at their roots
- Model calculations can be very helpful to develop circuits, to quantify distortions and to give the remaining distortions the preferred properties
- Which is precisely what we have done to a large extent to develop these novel concepts
- All our circuits have been build with discrete components to our own design (we don't use analog IC's in our products)





# A different way to fight distortions

- Distortion products should, spectrally speaking, decrease rapidly with increasing harmonic frequency
- Compare valve amplifiers and loudspeakers
- Example: Impedance compensation of loudspeakers gives a more ideal load for power amplifiers

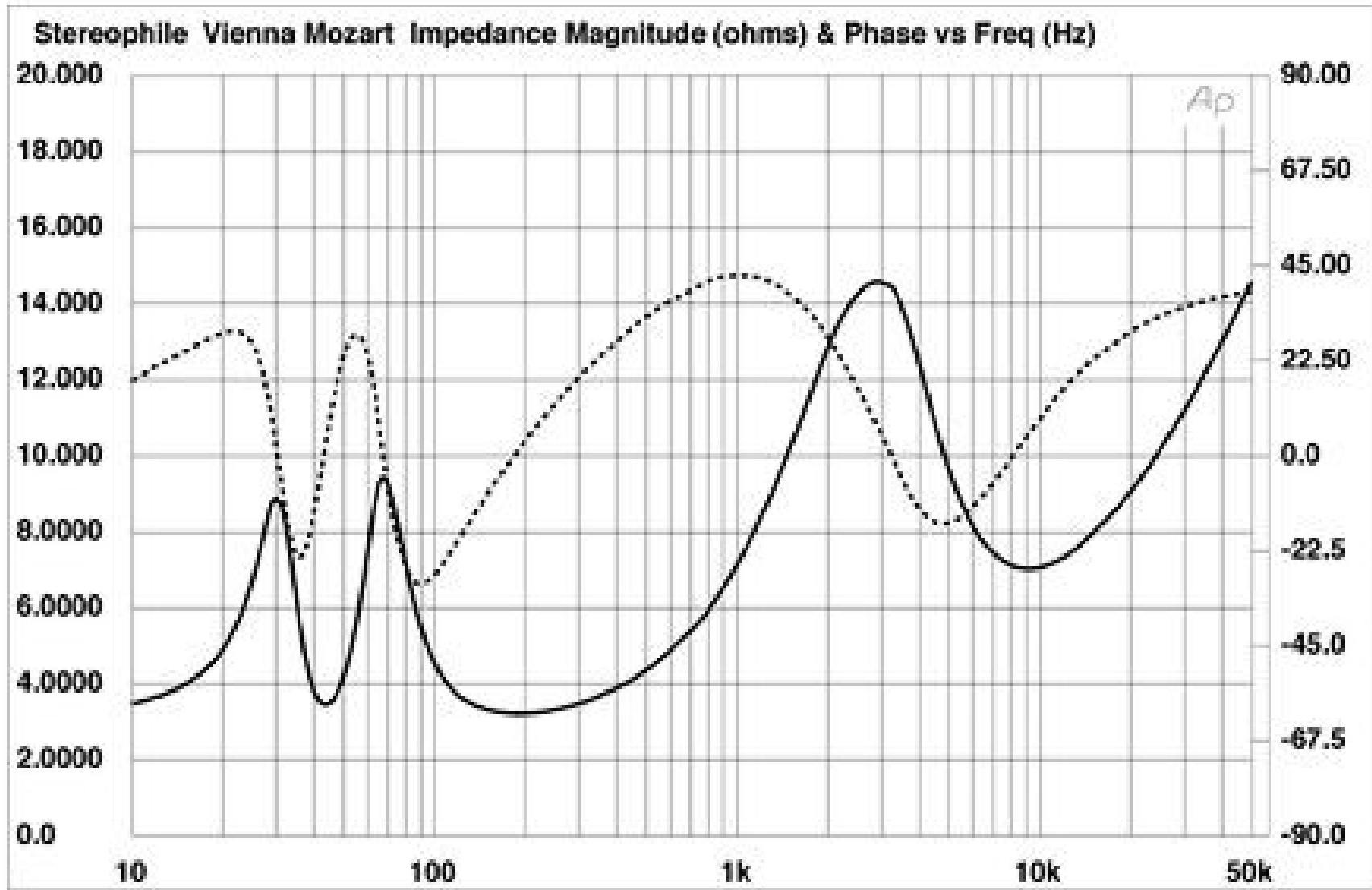


# Why impedance compensation?

- The loudspeaker impedance is (in more than one sense) a complex function of frequency

In the next slide, the measured impedance of a two-way loudspeaker system is shown. The double peak at low frequencies is caused by the port and woofer resonances of the base-reflex system, the increase towards the mid range is caused by the coil inductance of the voice coil of the woofer, the decrease above 3 kHz is the remnant of the resonance of the tweeter and the increase at the highest frequencies is the inductance of the voice coil of the tweeter. Note that the impedance varies between 3 and 15 Ohms, the phase between -30 and +45 degrees





With thanks to Stereophile

# Why impedance compensation?

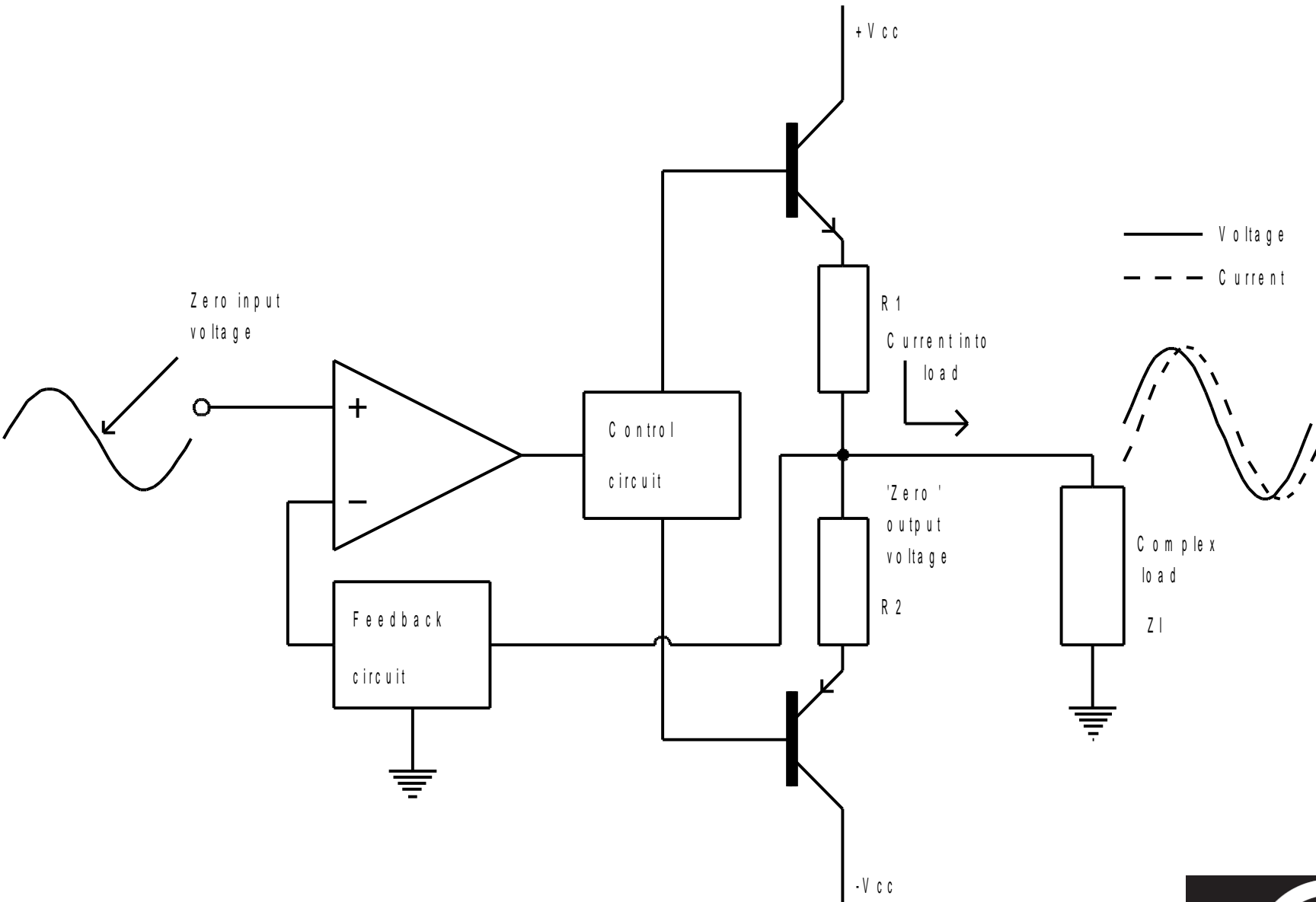
- The loudspeaker impedance is (in more than one sense) a complex function of frequency
- The output stage gets in trouble when voltage and current are not in phase -> an error voltage is required to open the power transistors

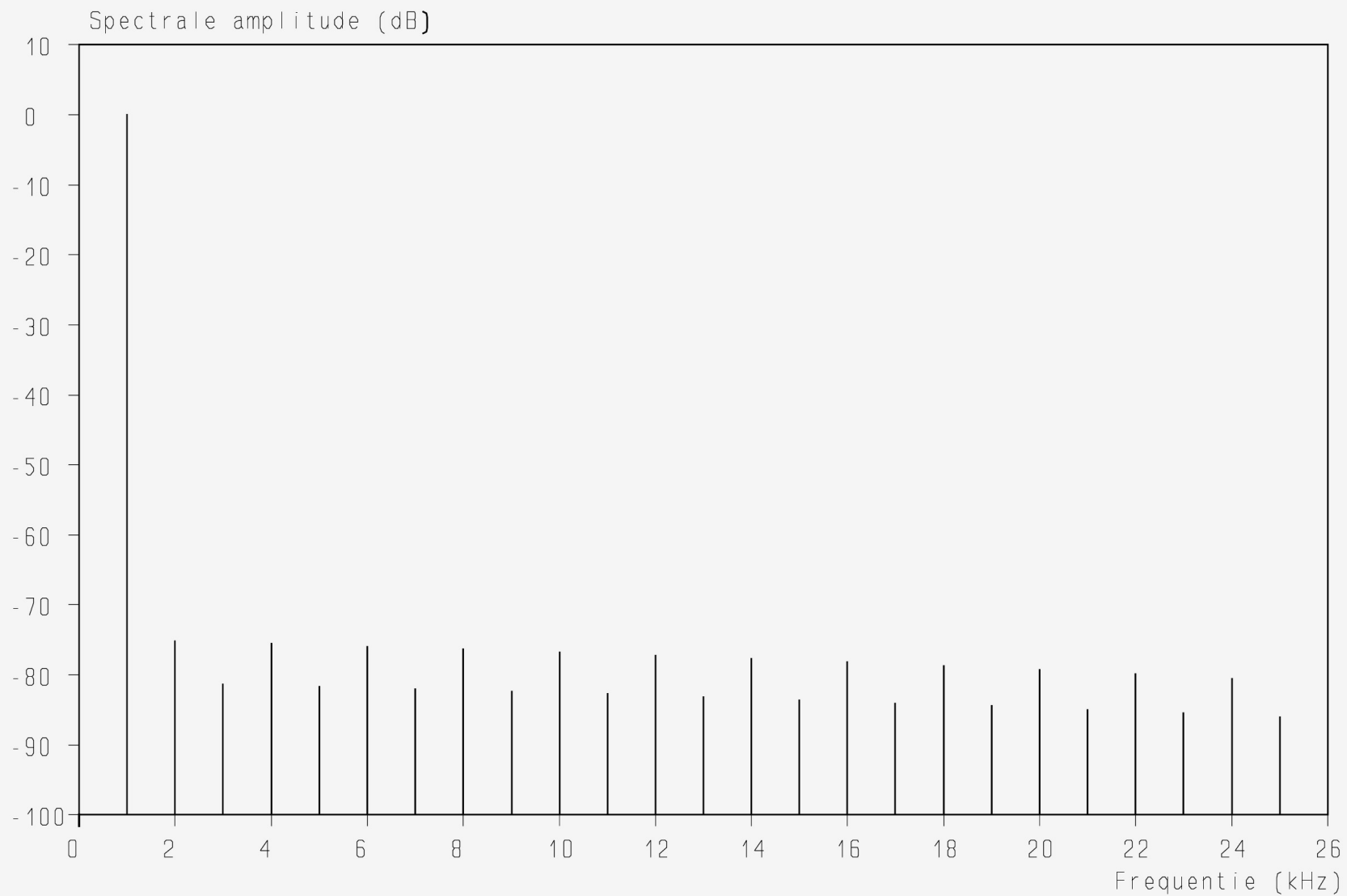


# Why impedance compensation?

The next slide shows this schematically: voltage and current are out of phase. At the voltage zero-crossing still current needs to be delivered. The transistor which should do so needs a base-voltage which can only be generated by the feedback, which requires an *error voltage* at the (loudspeaker) output. This emphasizes the cross-over distortion which has the undesirable properties like the presence of very high harmonics, as can be seen in the slide after the next one.







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# Why impedance compensation?

- The loudspeaker impedance is (in more than one sense) a complex function of frequency
- The output stage gets in trouble when voltage and current are not in phase -> an error voltage is required to open the power transistors
- Emphasis of the cross-over distortion which is very annoying to our ears because of the presence of very high harmonics
- Class A is than an option, but costly, requiring much energy and shortened lifetime of the electronics
- Tackle the problem at its root: apply impedance compensation so voltage and current remain in phase

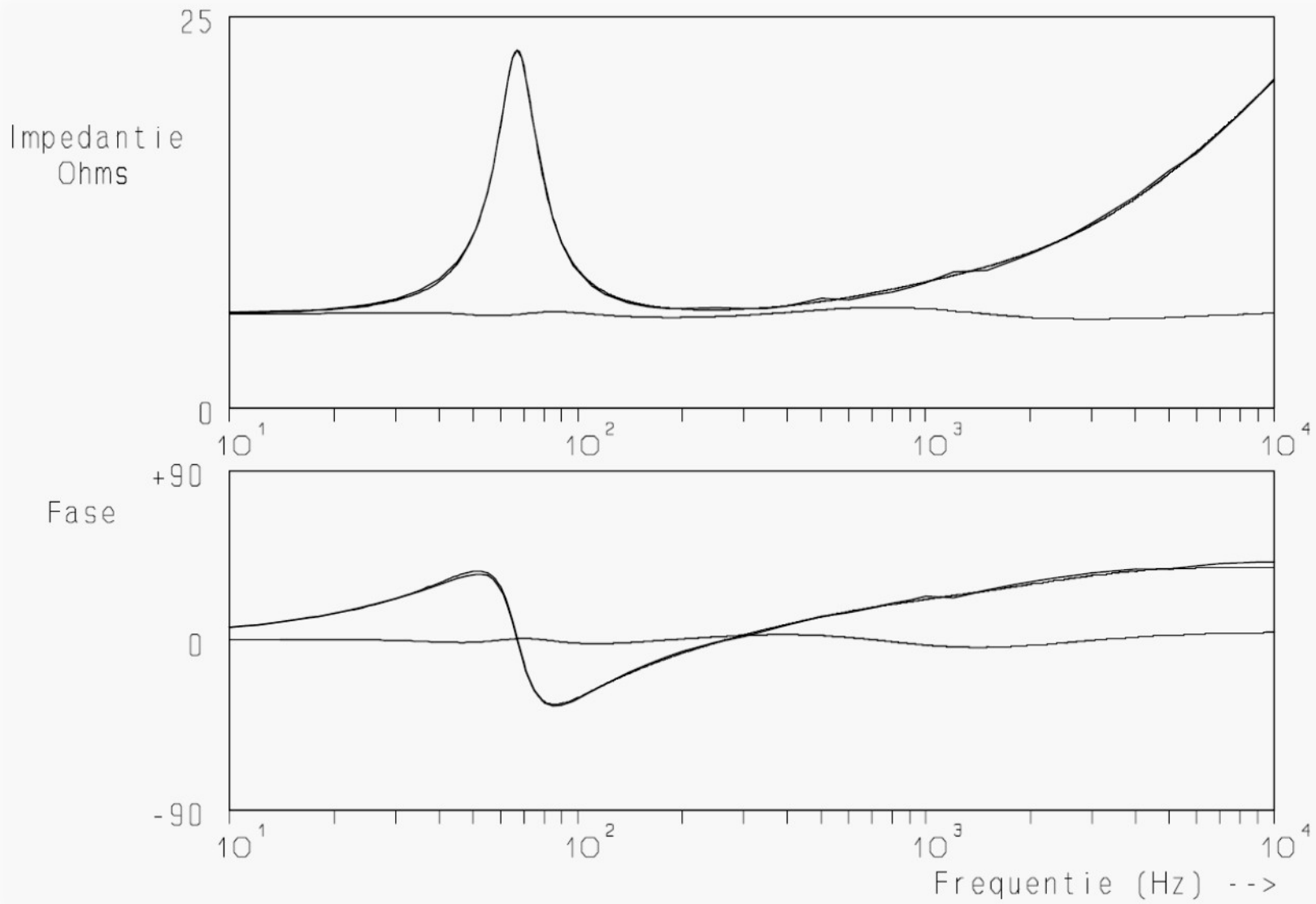




# Why impedance compensation?

In the next slide 2 times 3 curves are shown. The upper block shows the measured, the modelled and the compensated (lowest curve) impedances. The lower block shows the same for the phase angle. These curves show that impedance compensation results in a major improvement of the amplifier load.





# Choice of the loudspeaker units

- Should fit into the enclosure (especially the woofer can be problematic)
- Should produce as little misery as possible in the frequency range used
- Good impulse response
- Mid-range should be a wide-band unit
- Impedance and efficiency can be chosen freely (active system)



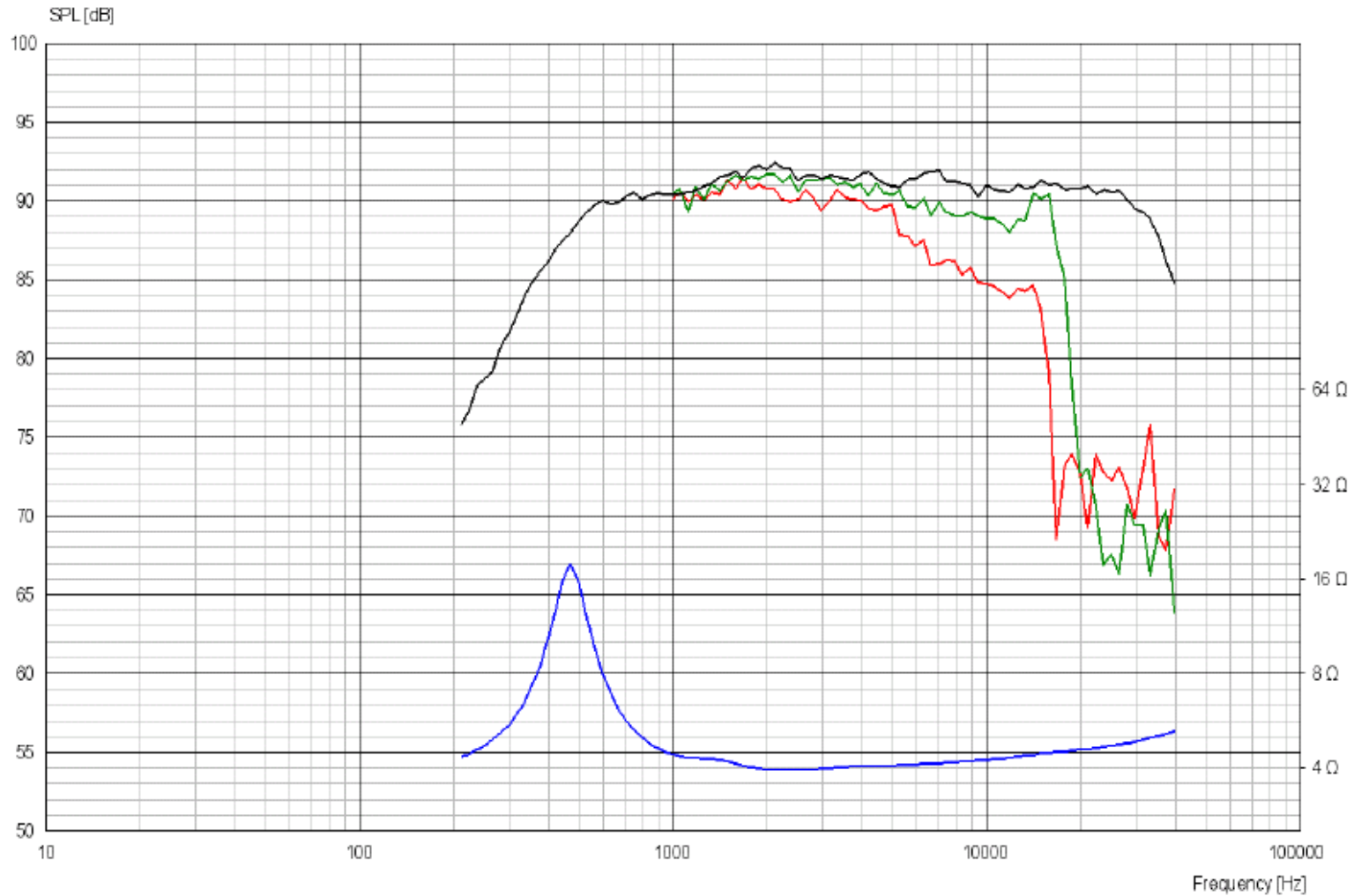
# Choice of the loudspeaker units

The next slide shows the specifications of the tweeter used by us. The absence of resonances, even above 20 kHz, is remarkable as is the reproduction to approx. 34 kHz. The response can be approximated well by a 4<sup>th</sup> order Butterworth filter and the impulse response of it is shown in the slide after the next one.

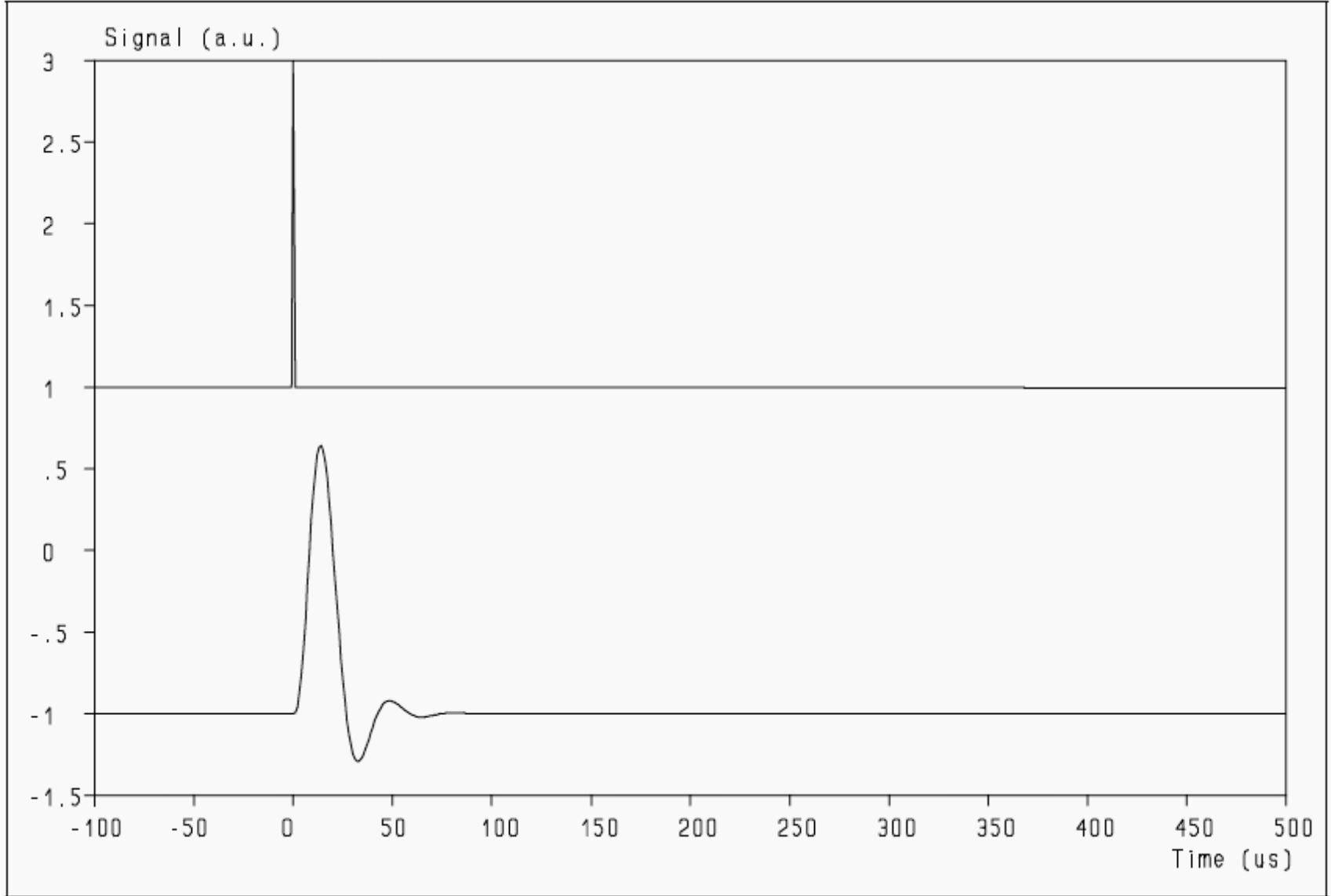


# TWEETER

D3004/662000



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Estimated “temporal decay” approx. 0,8 dB/ $\mu$ s



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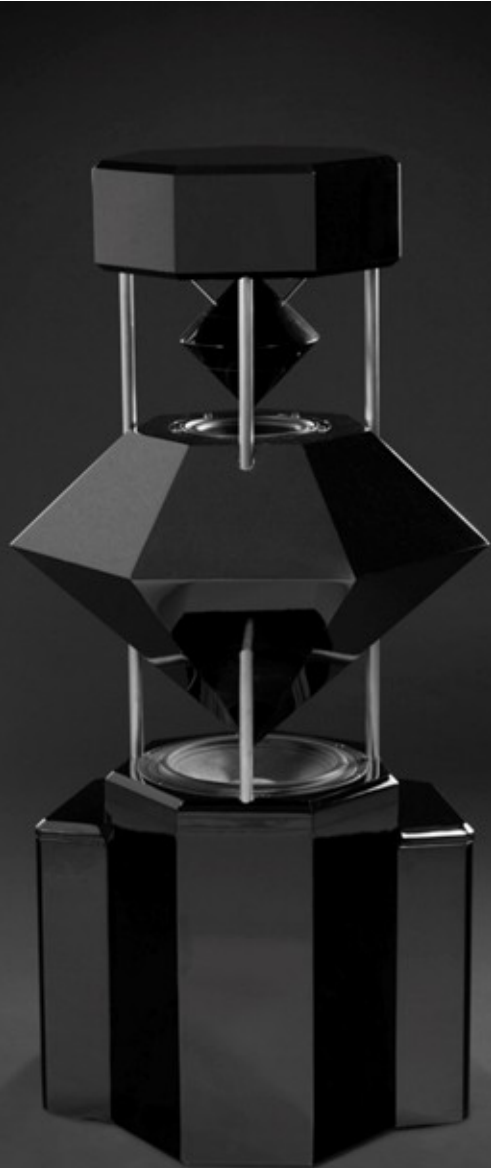
# Materialized systems

The next slide shows pictures of systems, build by us. Both use the same loudspeaker units, the same filters and compensators and amplifiers. As the “Pyramide” system is easier to build than the “Diamond” system, the price of a set of “Pyramides” is lower.



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# Two models: “Diamond” and “Pyramide”



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# Results of Siegfried Linkwitz

- Highly appreciated developer of loudspeakers
- Has come to the conclusion that the indirect sound should be the same as the direct sound: “The ideal stereo loudspeaker has a frequency independent polar response”.
- Uses dipole radiators for that purpose
- Although better than an “ordinary” loudspeaker enclosure, it is still frequency dependent (for high frequencies a ‘figure 8’ response)
- Circular radiation concept using cones is frequency independent



# Mechanical construction

- 28 – 30 mm MDF for the enclosure
- “Pyramide” internally divided in a woofer and a squawker compartment
- This creates additional stiffness and suppresses panel vibrations
- Woofer in front panel, but as the wavelength » enclosure “automatically” a circular radiation pattern is achieved
- Cones should be at ears’ height (while sitting)
- “Diamond” is the “cone” for low frequencies



# Finish and choices of customer

- Finish can be chosen by the customer: piano lacquer (any RAL-colour) or veneer of choice
- Advice not to apply veneer on “Diamond”
- “Pyramide” can very well be finished with veneer, but the pattern should show clearly at a larger distance



# Results and conclusions

- Control amplifier sounds very well, can also be applied in other systems
- Goals, as formulated at the beginning, have been achieved with the “Diamond” and “Pyramide” systems
- Using active systems, it is possible to overcome problems which cannot be tackled with passive systems
- Without the active concept, it would not have been possible to realize these systems
- The alternative way to suppress distortions works very well for our hearing
- “Spin-off” are the stereo power amplifier and the headphone amplifier





Ultra Linear Amplifier

TEMPORAL  
COHERENCE



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Free after George Orwell:

All audio systems are compromises, but some  
are more compromise than others...

Questions ?



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