THE ADVANTAGES OF "ACTIVE" OVER "PASSIVE" AUDIO SYSTEMS

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Abstract

The limitations of loudspeaker units require the use of cross-over filters to build loudspeakers which cover the entire audio band. Such filters do, however, have a number of disadvantages, which have stimulated the search for different topologies, the active cross-over filtering being the most well-known. However, the application of an active topology allows the introduction of loudspeaker unit specific corrections, which leads to a major breakthrough in sound reproduction as the frequency range can be extended, the temporal response can be improved, linear and non-linear distortions can be reduced and the headroom of the audio system can be increased.

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1. Introduction

The reproduction of sound is done by the use of different components, in very broad terms, a "source" is used (like a CD-player), a “control amplifier” (which normally includes a source selection switch, a volume control and possibly tone controls), a “power amplifier” and “loudspeakers”. Often, several components are "integrated", but the functionality is present in all systems. Traditionally, each component is intended to do its part without worrying about the others. This enables the combination of components from different manufacturers, different types etc. Although this is certainly an advantage, the disadvantage is that no possibilities exist to tackle the weakest part: the loudspeaker. Therefore, different topologies can be developed, which enable optimisation of the whole system, rather than optimisation of each component. In sec. 2, we will discuss the "traditional" and other topologies for the design of audio systems, in sec. 3, the pro's and con's of the different topologies will be discussed and in sec. 4 some conclusions will be drawn.

2. Different topologies for the design of audio systems

2.1 The "traditional" approach

The major problem that all designers of audio systems have to cope with is the loudspeaker, which is the weakest shackle of the chain. The "full-range" loudspeaker unit[i] does not exist and although electrostatic loudspeakers approach this the best, most struggle with the low frequency part of the audio range. Electro-dynamic loudspeaker units are unable to cover the whole audio range[ii] and therefore several different units are used in a loudspeaker. This, however, creates additional problems as the different loudspeaker units can only handle the frequency range these are designed for. In order to achieve the separation of the different frequency bands, a so-called "cross-over filter" is used, which separates the different frequency ranges of the output of the power amplifier to suit the different loudspeaker units. Note that because the filter components have to handle the full power, delivered by the power amplifier, these have to be large and bulky. This will create problems of its own, as we will see below.

2.2 Active cross-over systems

An alternative to the "traditional" approach is to use electronic cross-over filters to separate the different frequency bands for the different loudspeaker units at low power, followed by a separate power amplifier for each of the loudspeaker units. This allows more flexibility in the choice of the cross-over filters and eliminates the need for large and bulky components of a passive cross-over filter. This approach has some advantages over the "traditional" technique, which will be discussed below.

2.3 Active cross-over systems with response compensators

The use of active cross-over filters with separate power amplifiers for each loudspeaker unit is a bit like the early development of the motor-car: basically the horse was replaced by an internal combustion engine. It took a while before the designers realised that the internal combustion engine allowed a different design of the carriage and thus opened novel ways to improve the design. The same is true when we change over to active systems: this approach allows the use of filter types, which are basically impossible in passive cross-over filters, but that is not the full story: it also allows the use of electronic compensation of the linear deviations of the loudspeaker units which behave less-than-ideal. This is a novel way of looking at the problem of sound reproduction: the underlying concept that each component in the chain has to be "perfect" is left behind and the concept is shifted to an overall optimisation of the entire system.

2.4 Active cross-over systems with response compensators and amplifier-loudspeaker unit optimisation

Once the concept of local optimisation of each component is left behind, it is not hard to take it even one step further: the interaction between the power amplifier and the loudspeaker unit is not trivial and can lead to unwanted non-linear distortions. This can be optimised, because in an active system each power amplifier drives only a single loudspeaker unit and it is therefore possible to optimise the way these interact.

We will now go into more detail of the different topologies.

3. Discussion of the different topologies

In the following discussions, we will assume that the loudspeaker includes at least two different loudspeaker units and that filtering is required to use the units optimally.

3.1 The "traditional" topology with passive cross-over filtering

The design of a passive cross-over filter is not trivial and a large number of aspects play a role. To list the most important ones:

1. The slopes of the filter sections which are to be used (6, 12, 18 dB/oct. or even more). What can be realised and what are the consequences for the response of the loudspeaker?
2. Which cut-off frequencies should be used for the filter sections? This choice intertwines with the choice of the filter slopes.
3. The impedances of the loudspeaker units in their enclosures are not frequency-independent resistors, but show a complex behaviour. How will this affect the responses of the filter sections? Should something be done about it? What will the effect be on the impedance of the loudspeaker as a function of frequency? Will this increase the requirements for the power amplifier?

4. How much power is flowing through the components of the cross-over filter? Can I use inductors with e.g. a ferrite core or will these saturate? What kind of capacitors should be used? How will these choices influence the end result?
5. How can the efficiency of the different units be matched? Should the "average" sound pressure level (SPL) be used or rather the SPL near the cut-off frequency? Is the impedance of the unit in its enclosure constant and stable enough to use resistors to match the efficiencies?
6. How will the components of the cross-over filters influence the control of the amplifier over the loudspeaker unit behaviour?
7. What will the overall impedance of the loudspeaker be? Will this influence the properties of the amplifier?

We will now elucidate the above points and make some comments about these.

Ad 1

The temptation to use steep filters (18 dB/oct. or even more) is severe as these strongly suppress frequencies the units cannot handle well. The major, usually overlooked, disadvantage is that such filters destroy the temporal behaviour of the loudspeaker beyond repair. As a consequence, e.g. attacks, like from drums or a grand piano, are smeared in time, leading to clearly audible deficiencies in the reproduced sound. This has been discussed more extensively in ref. 1, and therefore it will not be repeated here. Another disadvantage is that steep slopes require many components, which interact, making the design complicated and sensitive for component variations (most of these components have a 10% tolerance). Also, the interaction with the complex impedance of the loudspeaker units becomes an additional complicating factor (see below).

Ad 2

The choice of the cut-off frequencies depends on a number of parameters. First of all, of course, the frequencies the unit can handle. But the choice also depends on the slope of the cross-over filter as frequencies outside the pass-band are only suppressed and not eliminated. So if e.g. a sharp resonance occurs just outside the pass-band, but the cross-over filter does not suppress this frequency sufficiently, an undesirable coloration of the reproduced sound may occur. This can only be cured by either shifting the cut-off frequency further away from the resonance or by increasing the steepness of the filter. The first could conflict with the properties of the adjacent unit (it might not be able to handle the additional frequency band), the second with the temporal response of the loudspeaker.

Ad 3

The impedance of a loudspeaker unit in its enclosure is rarely a frequency-independent resistance. The large majority of the impedances of electro-dynamic loudspeaker units behaves as is illustrated in fig. 1: at very low frequencies, it is merely the DC resistance of the voice coil, at a higher frequency a resonance occurs, caused by the mass-spring system of the moving mass and the suspension and the gradual increase at higher frequencies is caused by the inductor properties of the voice coil, nowadays reduced by eddy current damping. The corresponding phase characteristic is also shown. Such an impedance complicates the calculations of the response of the cross-over filter as it is obvious that assuming the unit has a constant impedance of 8 Ω without any phase shift is a coarse approximation of reality. The calculation of the filter characteristics can best be done if the impedance characteristic of the loudspeaker unit is modelled by a network like shown in fig. 2, taken from ref. 2. The other option is to measure the modulus and phase characteristic with a high resolution and to do the calculations of the cross-over filters numerically. But neither can be done accurately without a computer.

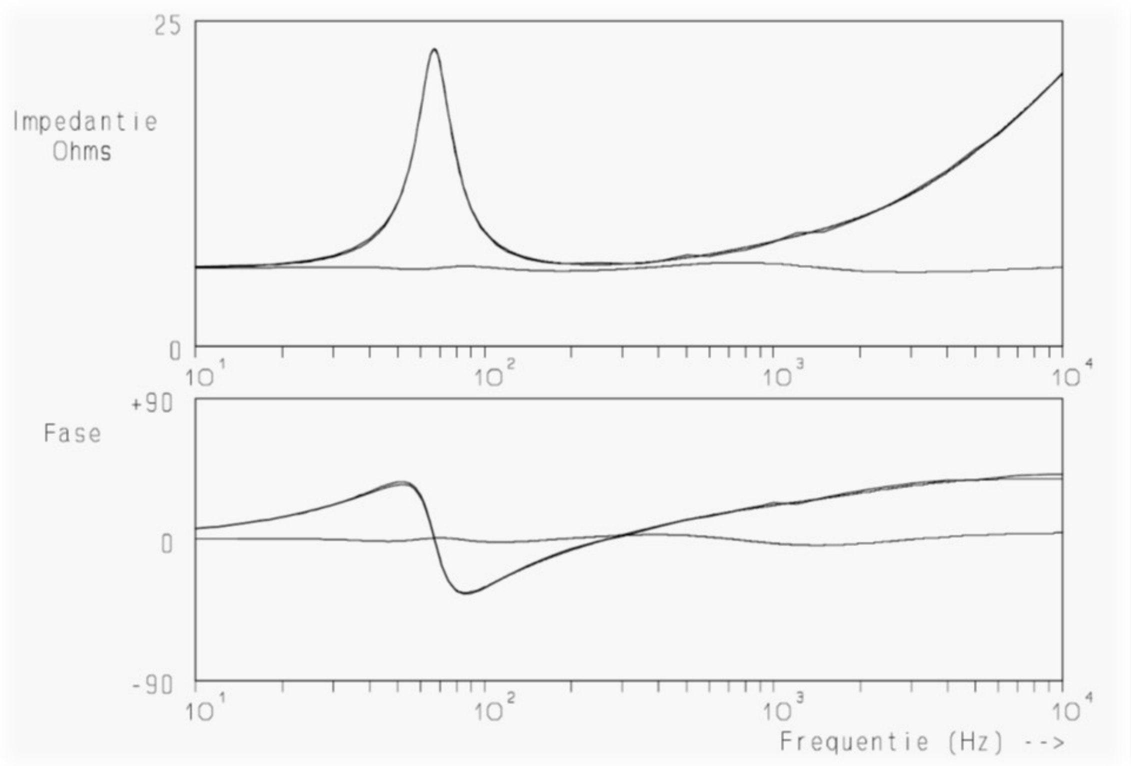

Figure 1: *Impedance (upper trace) and phase characteristics (lower trace) of an electro-dynamic loudspeaker unit. Note the resonance and the increase at higher frequencies due to the voice coil inductance. The almost horizontal lines are the properties after impedance compensation has been applied as will be discussed in sec. 3.4.*

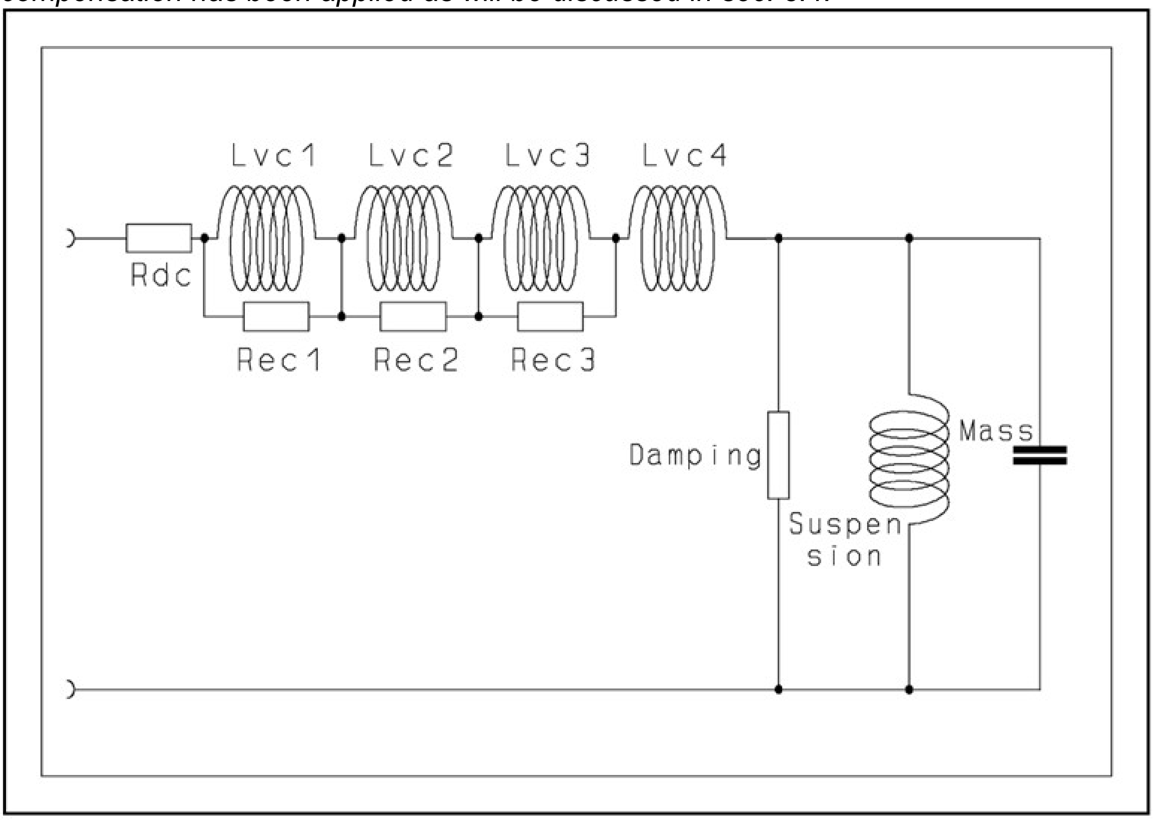

Figure 2: *Equivalent electrical model of an electro-dynamic loudspeaker unit, which can be used for the calculation of the properties of a passive cross-over filter and for impedance compensation as will be discussed in sec. 3.4.*

Ad 4

In a passive cross-over filter, the power of the amplifier is passed through the components of the cross-over filter. Although inductors and capacitors are -in theory- loss-free, this is not completely true in reality. Capacitors which have to pass large currents require a large surface area of the plates, which makes them bulky. Not all designs allow a negligible resistance of the plates and in some cases, the construction introduces inductor-properties as well. To make things worse: the large electrostatic forces, acting on the plates and the dielectric will lead to mechanical displacements which generates a non-linear response and thus lead to harmonic and intermodulation distortions. In fig. 3 the situation is shown when current is flowing through the capacitor to a loudspeaker unit. The input signal is a pure sine wave with a frequency of 1000 Hz. The current will change the charge in –and thus the voltage across- the capacitor. As a result, the capacitance changes in a non-linear way, leading to a distortion of the sine wave as can be seen in fig. 3. In fig. 4, the input signal of the high-pass section of the cross-over filter consists of a 1 kHz and an 8 kHz sine wave, 12 dB lower than the 1 kHz sine wave. The spectrum of the signal going into the tweeter is shown. The 1 kHz sine wave is attenuated by 12 dB because of the filtering properties, the spectrum shows apart from the 1 and 8 kHz input signals numerous intermodulation products due to the non-linear response of the capacitor. More details can be found in ref. 3.

Inductors are built using (mostly copper) wire, having a certain resistance, thus creating losses. On the one hand, one wants to reduce these losses, which can be done by using a (often ferrite) core in the coil, but that introduces non-linearities in the inductor properties: the inductance is current-dependent, with its most extreme behaviour when the core saturates, but also at small currents. Inductors without a core are far larger, require more length of wire and thus have more resistive losses (and are more expensive). These also generate strong magnetic fields, which can contribute to parasitic feedback. Like capacitors, coils also experience strong mechanical forces, which also can lead to non-linear responses. Usually, air coils are mechanically less rigid than coils with a ferrite core and are thus more prone to non-linear distortions. The choice of the components of a passive cross-over filter is therefore not trivial, but is critical, especially for those components which are in series with the loudspeaker units as the current through the units also flows through these components.

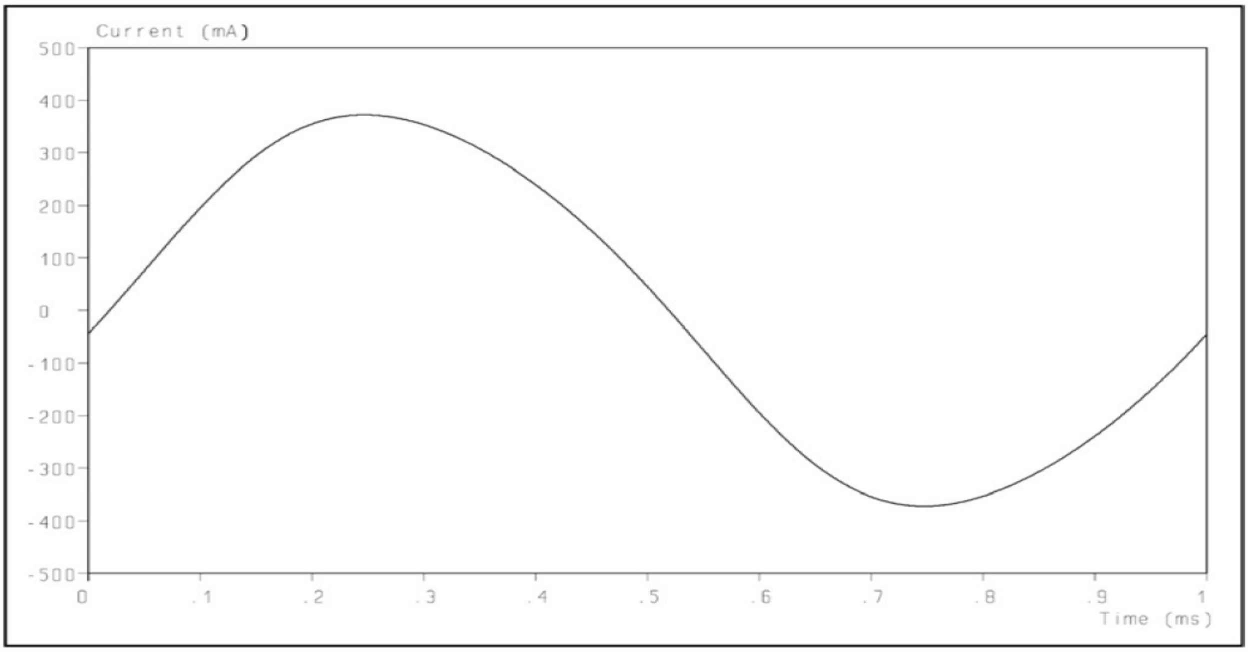

Figure 3: *The mechanical forces, acting on the plates and the dielectric of a capacitor lead to its deformation and so to harmonic and intermodulation distortion. For explanation: see text.*

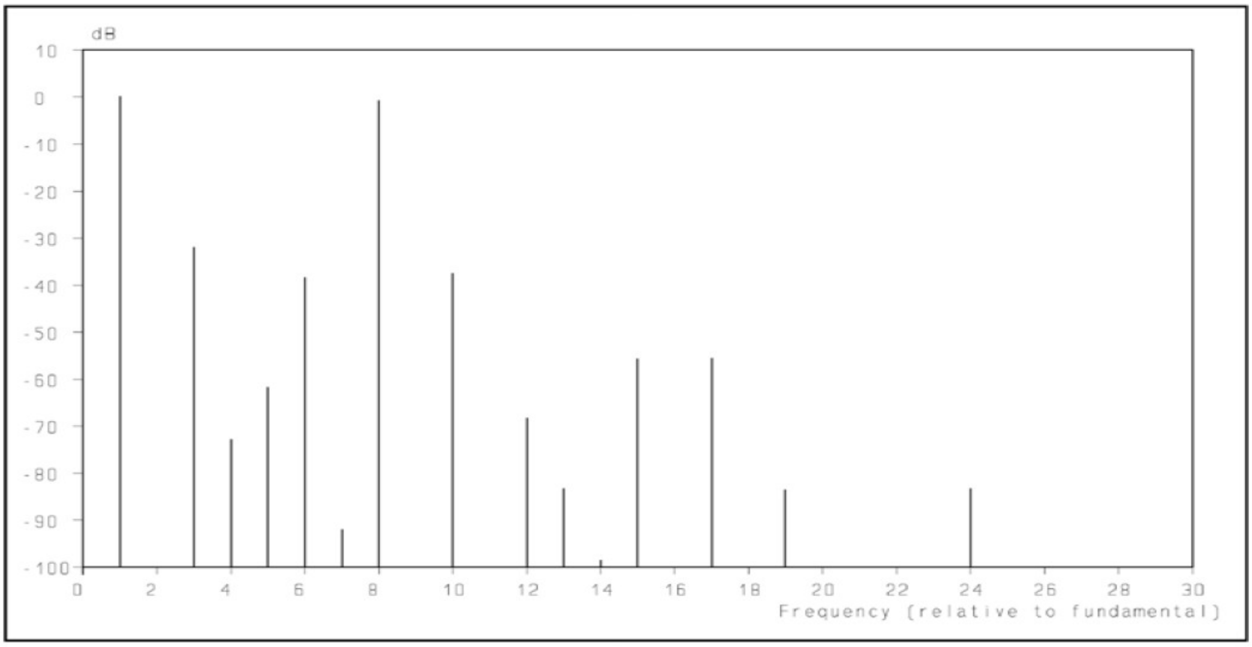

Figure 4: *Intermodulation products of the 1 and 8 kHz tones, caused by the non-linear properties of capacitors in passive filters (from ref. 3). For explanation: see text.*

Resistors are -by definition- not loss-free, which means that these should be able to handle large amounts of power. Not only because of longevity of the resistors, but also because an increase in the temperature will lead to an increase in both the average and the instantaneous value of the resistance. The average value will influence the properties of the cross-over filter, the instantaneous value will lead to non-linear distortions, especially at lower frequencies. The resistors will create, when required to adjust the efficiencies of the different loudspeaker units, a barrier between the amplifier and the loudspeaker unit, which could result in a reflection of the impedance curve of the loudspeaker unit in the response curve of the loudspeaker. See also items #5 and #6 below.

Ad 5

The choice of the different loudspeaker units depends on a number of parameters and it is very rare that the efficiencies of the different units are perfectly matching. But, of course, the first question is what the efficiency of a loudspeaker unit precisely is. As most units have a fluctuating response as a function of frequency, it is a bit arbitrary to choose the definition of the efficiency. This could e.g. be done by taking a sort of average over the frequency range of interest, but if one averages dB's, one should realise that this averaging is done over a logarithmic value, thus creating a systematic error. It is also possible to look at the frequency range near the cross-over frequency to get a smoother transition between the units. But still, in most cases, the efficiencies of the different units will have to be adjusted (read: reduced to the lowest efficiency[iii]), which can most easily be done using resistors. These should be able to handle large amounts power (see item #4 above) and basically will only work properly if the impedance of the loudspeaker unit (including the cross-over filter components!) behaves like a resistor. Which it usually does not (see fig. 1), creating the risk that the impedance curve reflects in the response curve and thus gives coloration. If the woofer is involved, the chances are high that the resonance will be insufficiently damped, leading to a "booming" sound. Note that the voice coil of the loudspeaker unit can also heat up, leading to a different impedance, which could also influence the filter characteristics.

Ad 6

The output impedance of most solid state amplifiers is very low (usually less than 0.2 Ω), resulting in a "damping factor" of 40[iv] and higher. The output impedance of valve amplifiers is often significantly larger, which can thus lead to coloration if the impedance curve of the loudspeaker is not constant (which is normally the case). But that is not the whole story: whatever the output impedance of the amplifier, the passive cross-over filter will have some of its components located between the amplifier and the loudspeaker unit terminals as can be seen in the simple example of fig. 5. As no components are ideal or perfect, the control of the amplifier over the loudspeaker units will not be perfect either (e.g. an inductor will easily have a resistance of 0.5 Ω, thus diminishing any damping factor from over 40 to 16 at best). The situation will, of course, worsen when more components are placed in series with the loudspeaker unit. Especially the presence of resistors, like those used to attenuate the response of the tweeter in fig. 5 is detrimental to the temporal response of the tweeter and it likely to lead to coloration around the resonance frequency of the tweeter and at higher frequencies (see fig. 1).

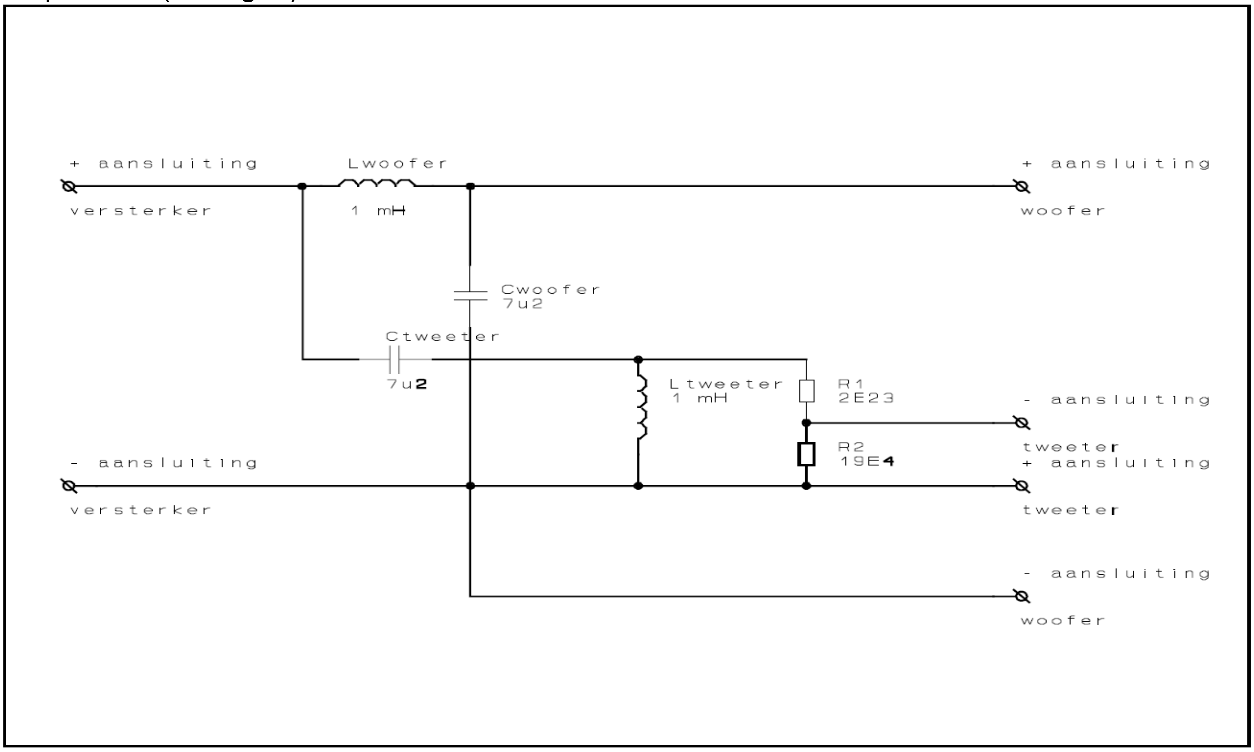

Figure 5: *Example of a simple, often applied, second order passive cross-over filter. Note that there are always components in series with the loudspeaker units.*

<u>Ad 7</u>

The loudspeaker using a passive cross-over filter will consist of a large number of components with a frequency dependent impedance. The loudspeaker units themselves are distinctively different from a resistor (see figs. 1 and 2) and the inductors and capacitors have a frequency dependence by nature. All these components interact, which, in general, will lead to a complex impedance curve of the loudspeaker. This includes phase shifts between the voltage and current at the output of the amplifier. Not all amplifiers are able to cope with complex loads, because the phase shift can lead to an error voltage in the feedback loop as is schematically illustrated in fig. 6. It might be that class AB-II amplifiers are more vulnerable to this effect and as a consequence, some people prefer class A amplifiers, albeit it that these are large energy wasters.

The above listed problems illustrate that loudspeakers, using a passive (traditional) cross-over filter, are limited in performance and have several unavoidable disadvantages. This is the main driver to study the possibilities of alternatives.

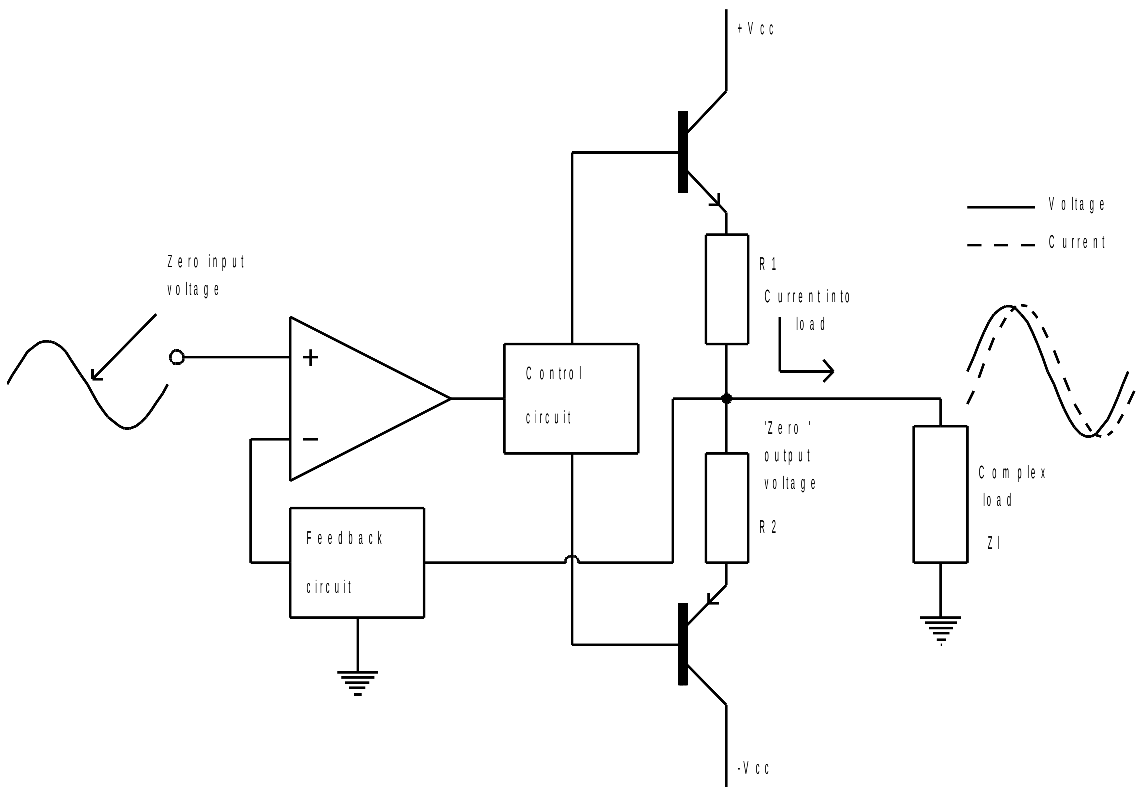

Figure 6: *Power amplifiers can give distortion when these have to drive non-ohmic loads: the requirement to deliver current at the (voltage) zero-crossing will require an error voltage which opens the power transistor via the feedback loop.*

3.2 The topology using active cross-over filters

The topology of systems, using active cross-over filters, is relatively simple: the signal from the source is split into the selected frequency bands, suited for the different loudspeaker units and each is given its own power amplifier. The major disadvantage is thus obviously the increased need for power amplifiers, albeit that the requirements can be less strict, as we will see below. There are, however a large number of advantages:

1. The filters can be more precise because the components involved are very small compared to those in passive filters (nF vs. 10 - 100 µF).
2. Losses will often have to be introduced in passive filters, which consume electrical power and thus reduce the overall efficiency, resulting in less "headroom" for the amplifier and thus for the complete system as was discussed above (see sec. 3.1, ad 5).
3. In active systems, each amplifier drives a single loudspeaker unit, which enables the optimisation of the combination of the two.
4. The "headroom" of the system is increased because the power is split over different amplifiers, thus reducing the required peak power due to amplitude addition.
5. Level adjustment between the different loudspeaker units (woofer, squawker, tweeter) can be made with a much higher precision and can be done without any mechanical work like opening the loudspeaker cabinet to access the filter.
6. The level of distortions of the amplifier is lower because the signal has a smaller bandwidth.
7. Distortion, generated by the amplifier, is no longer transferred to the squawker and tweeter (as is done by a passive cross-over filter).

Ad 1

Because of the low impedances of loudspeaker units (4 - 20 Ω), the components in passive filters need to have correspondingly large values. As the accuracy of such components is less than those of the smaller components which can be used in active filters and which are available (if desired) with 1% precision, more accurate filter properties can be realised in active filters than is generally possible with passive filters. Also, filter properties which are impossible to create with passive filters are feasible with active cross-over filters.

A note of warning should be mentioned here: although the capacitors in active cross-over filters are not as vulnerable to non-linear distortions as those in passive filters because these operate at smaller voltages and usually have a bias voltage as well, they often are sensitive to (mechanical) vibrations because of the piezo-electric effect. As these components are often mounted near or in the loudspeakers, selection of the components is also critical in active cross-over filters.

Ad 2

A topology using active filters maximizes the efficiency of the system, as each amplifier directly drives a loudspeaker unit. No attenuation using resistors to equalize the efficiency of all the loudspeaker units to the lowest is required. Adjustment can easily (and more accurately) be achieved by controlling the output signal of the filter sections.

Ad 3

In active audio systems, each loudspeaker unit is driven by a separate amplifier. In this way, the amplifier has direct control over the loudspeaker unit, contrary to passive systems in which always some components like inductors or capacitors are in series with the loudspeaker units. The combination of the amplifier & loudspeaker unit can thus be regarded as a unity and its performance can be optimised. As an example, the power, available for the different loudspeaker units can be different, but later on we will see a more sophisticated example of the possibilities.

Ad 4

When somebody is asked how much power has to be delivered by an amplifier when a low frequency tone of 4 W, a medium frequency tone of 4 W and a high frequency tone of 1 W have to be delivered continuously to a loudspeaker system, the obvious answer is 9 W and thus a 10 W amplifier would have sufficient power to do the job. **Wrong!** Because there will be a moment in time where the amplitudes of the signals will have to be added, the addition of the powers gives an erroneous result. By adding the *amplitudes*, the answer is that the peak power delivered is 25 W so this is the minimum which is required to produce the 9 W *average* output without distortion. However, in active systems, the frequencies are separated and thus this amplitude addition does occur to a lesser extent and therefore the system has effectively more "headroom" and so it is better equipped to handle peak loads. Added to this is that in passive systems, the efficiency of the different loudspeaker units will always have to be scaled down to the lowest level to avoid coloration. This problem does not occur in active systems, simply the input signal level of the power amplifier can be reduced to a lower level, but this leaves the efficiency of the loudspeaker unit at its maximum, there are no losses. This helps to increase the "headroom" of the system further.

Ad 5

Although often 3 dB is mentioned as the limit of sound level difference that a human ear can distinguish, it shows in practice that far smaller differences influence the tonal balance of an audio system. To my experience, alterations of ± 1 dB can make quite a difference in the perceived sound and adjustments to ± 0.25 dB are required to achieve the optimum tonal balance. With passive systems, such alterations are hard to make and usually require hardware modifications to the cross-over filter and the resistors which are used to adjust the levels between the different loudspeaker units. As the impedances of the loudspeaker units are not purely ohmic, the changes can easily work out in a different way than expected. With active systems, the level adjustment is very easily done by just using the level controls on the output of the filters at the input of the power amplifiers.

Ad 6

Because the amplifiers have to handle a smaller bandwidth and the amplitude level is lower, the power amplifiers generate less distortion. This is true for both the harmonic and the intermodulation distortions. This enhances the "musicality" of the system. Note that this comes in addition to the increased headroom (see item #4 above) and the deviation of the distortion products (see item #7 below).

Ad 7

The harmonic distortion products, generated by the power amplifier consist of integer multiples of the excitation signal frequency. In a passive topology, these distortion products are taken to the squawker and tweeter, which reproduce these. In an active topology, these are only fed to the loudspeaker unit which has to reproduce the excitation signal itself and it often acts as a filter (e.g. the woofer cannot reproduce all the higher harmonics of the low frequency band). This effectively reduces the audible distortion of an active audio system.

3.3 Active cross-over topologies with compensators

As is well-known, loudspeaker units are not perfect, which is the main reason why we use separate units in a loudspeaker. Only electrostatic loudspeakers approach the "full-range" ideal, but still these usually do not reproduce the low-frequency part of the audio range well due to acoustic short-circuit. When looking at electro-dynamic loudspeaker units, the units are limited on the low-frequency side by the resonance of the unit (which in itself is influenced by the enclosure of the unit) and by imperfect cone behaviour at the high-frequency side. The imperfect cone[v] behaviour can only be corrected by high-performance digital systems, which are still outside the reach of do-it-yourself audio enthusiasts, but the low-frequency behaviour is not. Using the normal specifications of loudspeaker units (e.g. moving mass, radiating surface area, compliance, Q-factors and the like), it is possible to predict the low-frequency roll-off of the unit in its enclosure, which in an acoustic box is basically the response of a second order high-pass filter. As has been described in another web-publication (ref. 1), it is possible to compensate this behaviour to extend the response to the lowest end of the audio range while maintaining a correct temporal characteristic as illustrated in figs. 7 – 9. Such a second order correction is applied to correct the woofer response.

But this approach can be extended to mid- and high-range units as well. The main reason to do this is to improve the behaviour of the units so that these will respond better in combination with the cross-over filter: if we want to maintain a correct temporal response of the loudspeaker, it is insufficient to use cross-over filters with the correct temporal characteristics, also the units need to respond close to ideal in the frequency range of interest. The latter can be improved by compensation of the non-ideal response of the loudspeaker units at the low-frequency side of their range. Fortunately, the response of the mid- and high range loudspeaker units can be approached by a first-order high-pass filter, which can easily be incorporated in the active cross-over filter design, contrary to a second order correction, as needed for the woofer which requires a dedicated compensator (ref. 1). Electronic compensation of loudspeaker units is relatively easy to achieve in active systems and it as such a step further in the development than just replacing the passive cross-over filter by an electronic version. It is an important step towards a more holistic approach in which the system as a whole is optimised rather than just the individual components.

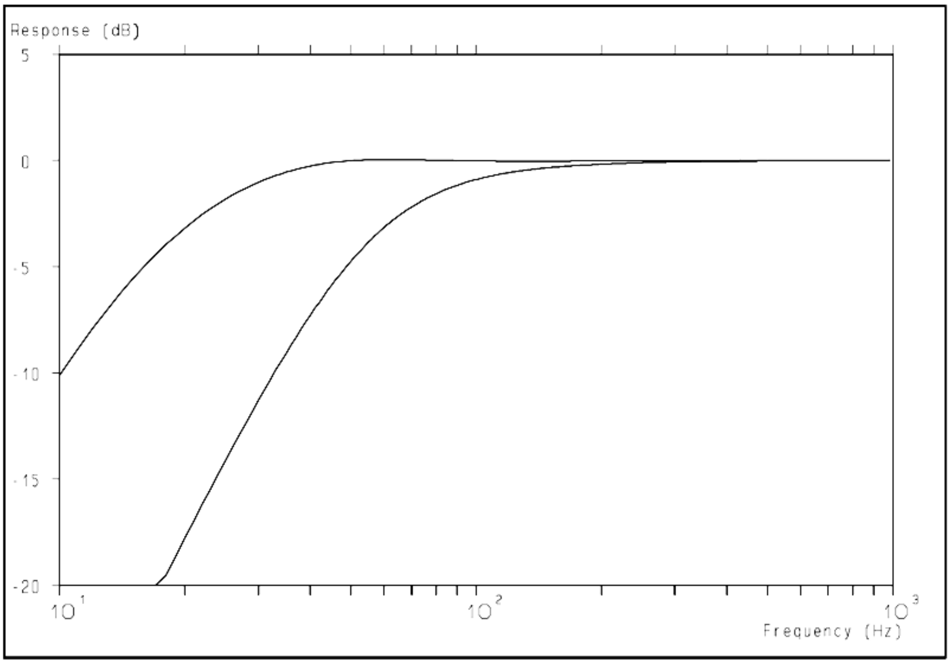

Figure 7: *The frequency response of the woofer can be extended by electronic compensation (upper trace).*

3.4 Topologies using also amplifier-loudspeaker unit optimisation

Most amplifiers are designed to deliver power into an ohmic load. Most specifications are therefore given for a specific ohmic load. As has been shown above, most loudspeakers are not a pure resistive load, but consist of a complex load, including phase shifts between voltage and current. Most amplifiers respond differently to a complex load than to a pure resistive load. Listening tests have shown beyond any doubt that a resistive load results in a more "musical" amplifier. This could be attributed to the necessity to create an error voltage to open the current dumping transistors when voltage and current are not in phase as is illustrated in fig. 6. So a further optimisation could be achieved if the load of the amplifier would be ohmic. To change the impedance of an electro-dynamic loudspeaker unit into a close to an ohmic load, a network as shown in fig. 10 can be connected in parallel to the loudspeaker unit. When properly designed, the replacement impedance is very close to ohmic as is illustrated in fig. 1. More details can be found in ref. 2.

There is, of course, no reason to limit this impedance compensation to active topologies alone. This can as well be done with passive cross-over filters also. This would simplify the design of the passive filter and probably improve its performance[vi]. And as most amplifiers respond better with an ohmic load, it would likely result in more "musical" systems as well, which would be a good selling point for the loudspeaker. Therefore I do not understand why this is not done more often. To my experience, it is very profitable to apply this impedance compensation, both in passive and in active topologies.

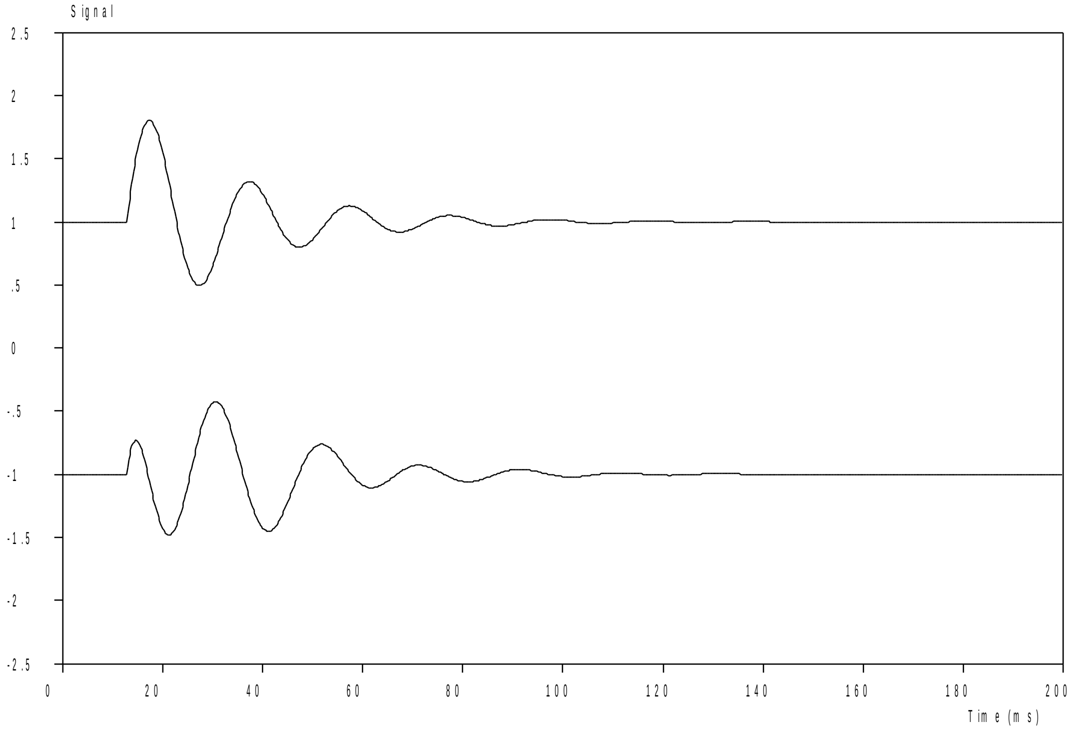

Figure 8: *The uncorrected temporal response of the woofer is degraded by the high-pass properties of the woofer in its housing. Upper trace: input signal, lower trace: response. Note the large differences at the onset of the signal and the almost 180º phase shift.*

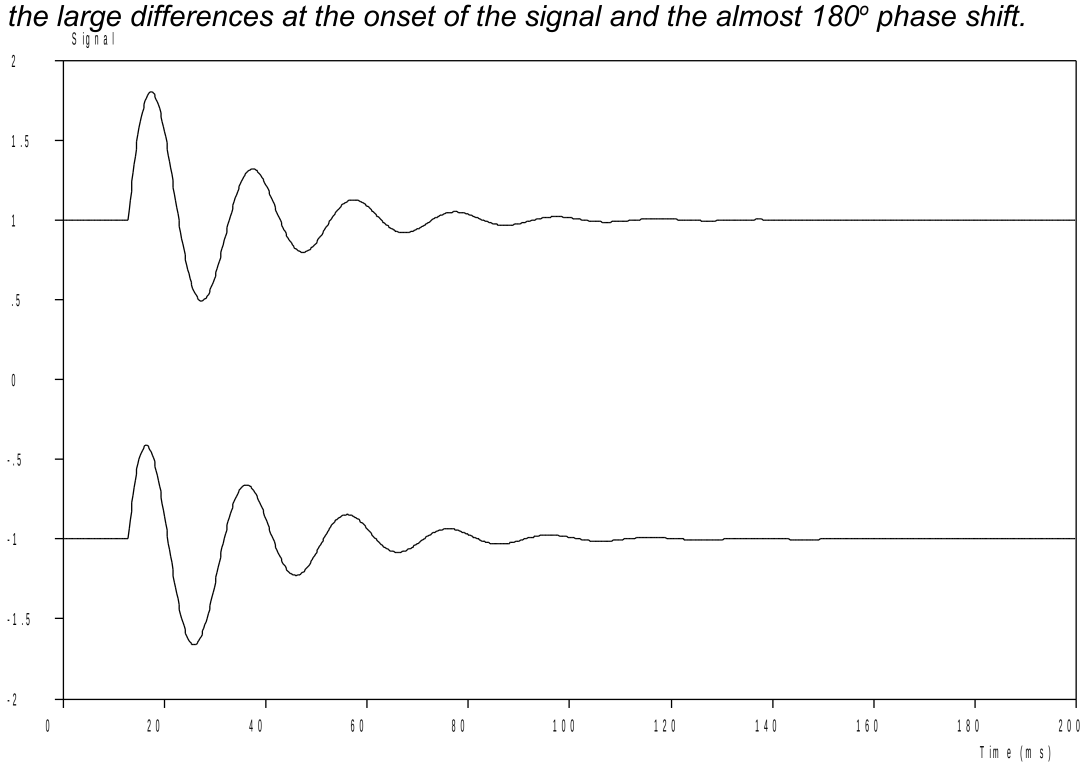

Figure 9: *The electronic correction of the frequency response of the woofer in its housing also restores the temporal response. Compare with fig. 8 and note the improvement at the onset of the signal and the virtual elimination of the phase shift.*

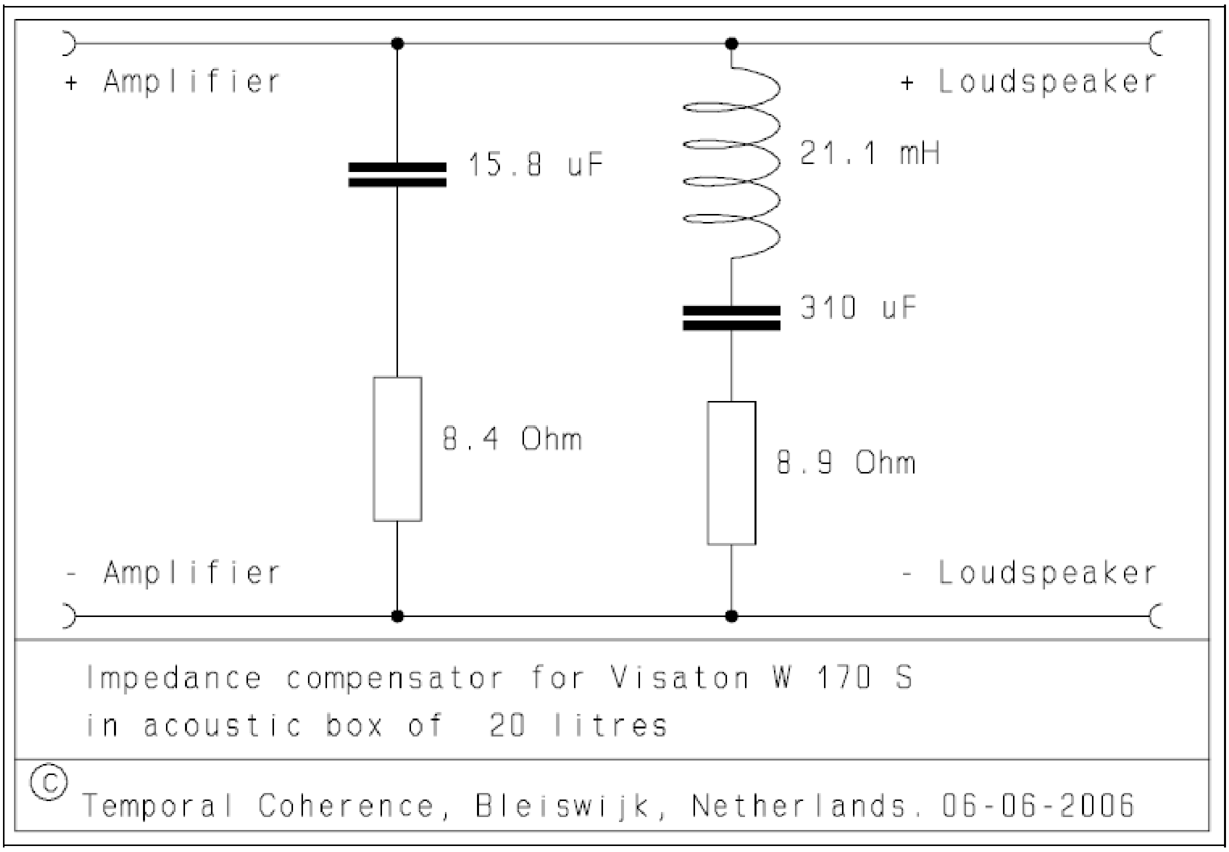

Figure 10: *Network, to be placed parallel to the loudspeaker unit in order to compensate the complex impedance characteristics. Component values depend on the loudspeaker properties in its enclosure.*

4. Conclusions

Designing audio systems using an active topology has a large number of advantages, most resulting from the possibilities to tackle the weakest link in the chain: the loudspeaker. By a more "holistic" approach, in which the whole system is optimised, novel ways can be found to extend the frequency response, improve the temporal response of the system and to reduce the non-linear distortions which reduce the "musicality" of the system. In my view, active systems, based on a holistic approach, enable a major breakthrough in sound reproduction development. The system, which I have at home and which is based on these principles, is a promising example of this approach.

References

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2. Drs. Hans R.E. van Maanen and E.T. Zonneveld, "An Extended Model for Impedance and Compensation of Electro-Dynamic Loudspeaker Units and an Algorithm for their Determination", Preprint 3823 (P8.1), presented at the 96th AES conference, Amsterdam, Netherlands, February 26 - March 01 1994.
3. Menno van der Veen and Hans R.E. van Maanen, "Non-linear distortions in capacitors", Audio Engineering Society Convention Paper, presented at the 124th Convention, May 17 – 20, 2008.

[i] With “loudspeakers”, complete systems are meant, including their enclosure, with “loudspeaker units”, the individual units, which are used to create “loudspeakers”. So a “loudspeaker” includes several “loudspeaker units”, a cross-over filter and an enclosure.

[ii] With the audio range, normally the range of 20 – 20 kHz is used, but as has been shown in another publication (ref. 1), it is required to use 15 – 50 kHz at least.

[iii] Note that 1 dB lower efficiency needs to be compensated by an increase of the amplifier output power by 26%! A 3 dB lower efficiency thus requires a doubling of the output power to achieve the same sound pressure level.

[iv] The damping factor is usually defined at the ratio of the nominal loudspeaker impedance (e.g. 8 Ω) by the output impedance of the amplifier. The value specified is sometimes above 100, but one should realise this is virtually useless as the damping of the loudspeaker is for a small part determined by the impedance of the cable between the amplifier and the loudspeaker, but to the largest extent by the properties of the components of the passive cross-over filter.

[v] This could also be a dome in case of a tweeter.

[vi] This has been done by me and the same loudspeaker units in the same enclosure sounded a lot better.