Thoughts About Crossovers

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This white paper describes some views about the application of crossovers. The goal is to show some unexpected effects and a new (?) approach for the design of crossover filters. Basically all items told here are not really new. But it seems that the idea to make the crossover filters in accordance to the good old passive crossover filters prevents from seeing the inherited problems. Luckily the use of digital crossovers allows to overcome the given limitations. Acourate offers such solutions. This is explained step by step to give insights and a better understanding.

1 The Minimum Phase Crossover Filter Story

A common way to generate crossover filters is to make similar filters like found with passive crossovers. Passive filters are made of passive elements e.g. inductors, capacitors and resistors. The crossover network built with these elements behaves like minimum phase filters. Minimum phase filters simply spoken are filters that have the shortest possible phase deviations (= delays) to achieve a desired frequency response.

As the passive crossover is a causal system the reaction on an input signal takes place <u>after</u> the signal. The best possible answer for a driver would be a Dirac pulse response. But the Dirac pulse means a flat amplitude response and we do not want to have a flat response. We want to have a flat passband where the signal passes and a stopband where the signal will be suppressed. In between there is a transition area. The transition area and the stopband have a different amplitude behaviour by definition. Thus the signal is influenced by amplitude and phase changes. The phase change means that the input signal gets delayed in the corresponding frequency range.

Let's take a view on an example. It is given as a standard 2nd order Butterworth filter (12 dB/octave or 40 dB/decade slope in the stopband) for a 3-way system. The crossover frequencies are arbitrarily selected as 150 Hz and 3500 Hz.
Remark: the filters have been created with the Scientific Filter function of Adobe Audition with a length of 65536 samples each (samplerate 48000 Hz) and been saved as dbl-files (double precision floating point format). Audition allows to read these files and to explore the given input files. The first picture shows the amplitude responses of the crossover filters:

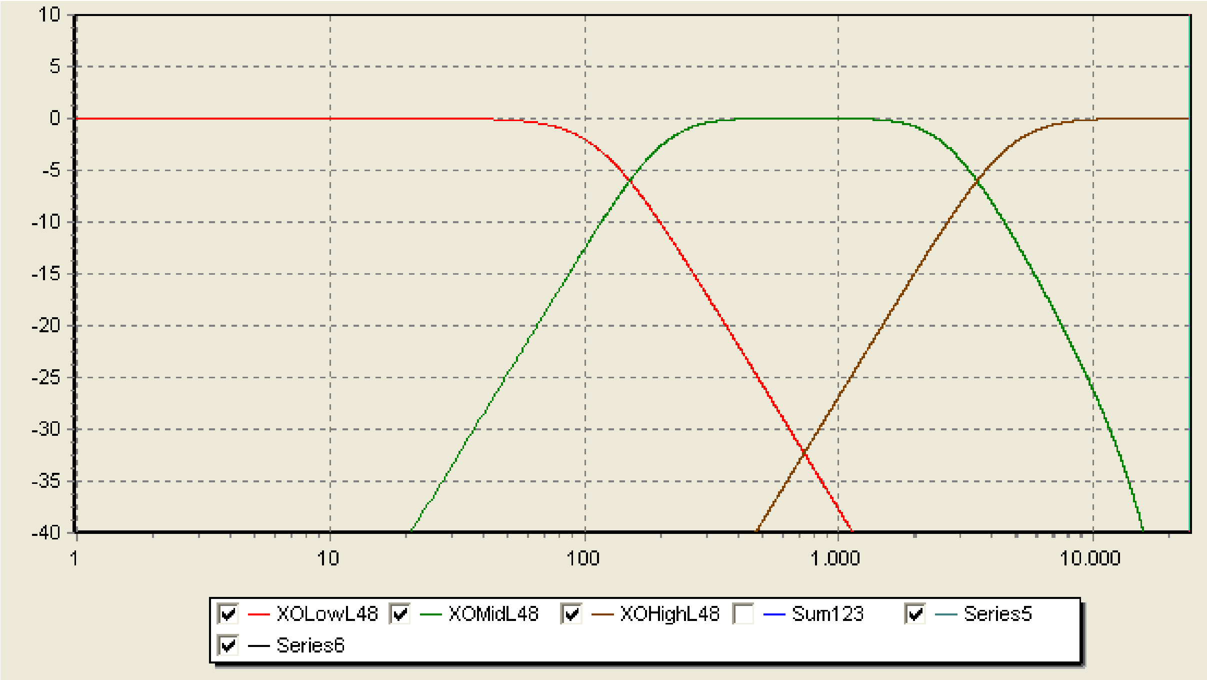

Fig. 1 Amplitude response of a Butterworth 2nd order crossover filter with lowpass 150 Hz, bandpass 150 Hz/3500 Hz and highpass 3500 Hz

The filters cross at the crossover frequencies at the -6 dB point. (Remark: The reason for the slope deviation at the right side of the bandpass is given by Adobe Audition !)

Now let's have a look on the time domain signals of the filters as created by Adobe Audition. Acourate allows the display of both time and frequency domain. It is possible to load and view 6 independant signals. The graph for each curve can be switched on/off by the appropriate checkboxes. The next picture shows the behaviour of the lowpass time domain signal.

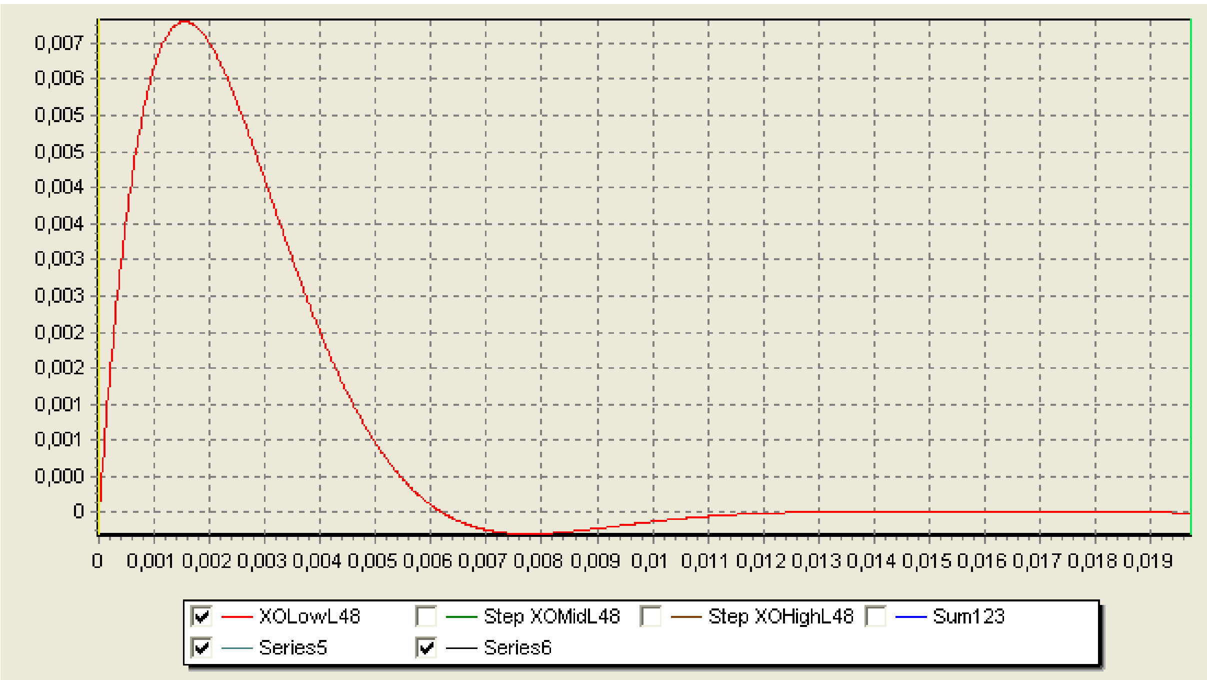

Fig. 2 Lowpass pulse response (minphase Buttherworth 2nd order 150 Hz)

The signal has a max. value of 0.0068 at sample 74 (shown by Acourate at the right side of the graph. Therefore the curve has to be selected by the Active Curve radiobuttons). Sample 74 means a time of 74/48000 = 1.54 ms.

What about the bandpass behaviour? It is shown in the next picture:

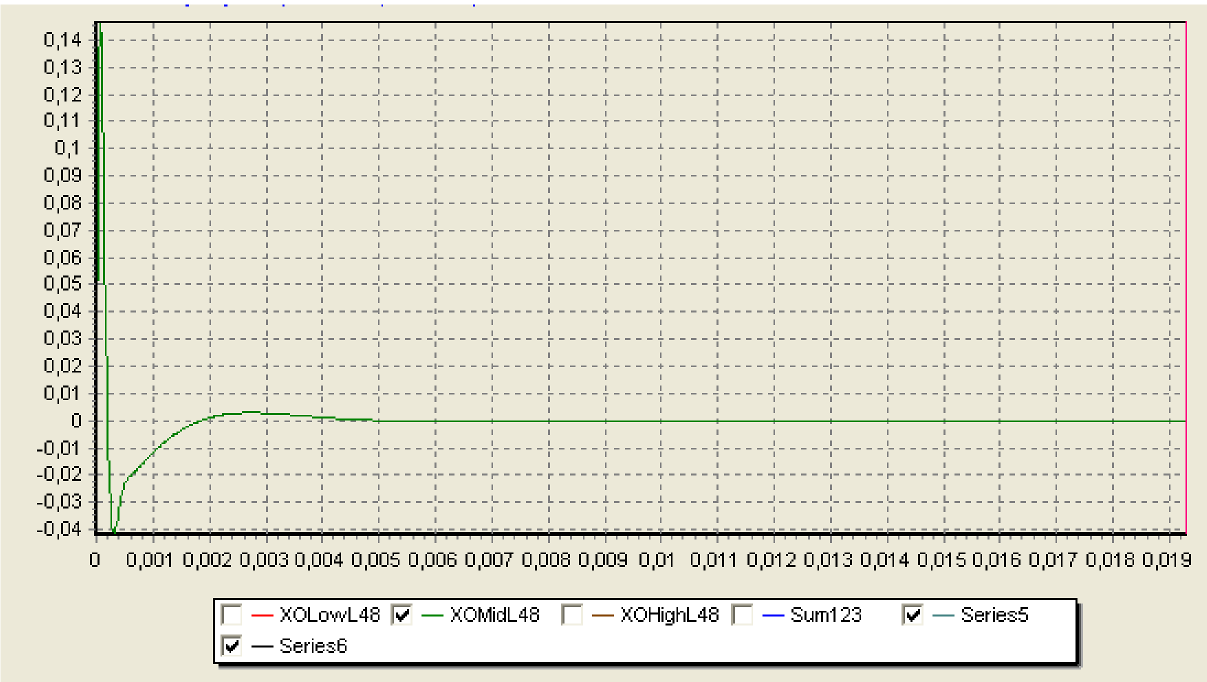

Fig. 3 Bandpass pulse response (minphase Buttherworth 2nd order 150 Hz/3500 Hz)

Acourate tells us that the max. value is 0.1459 at sample 3. The peak is at a different position.

And finally we look at the highpass curve:

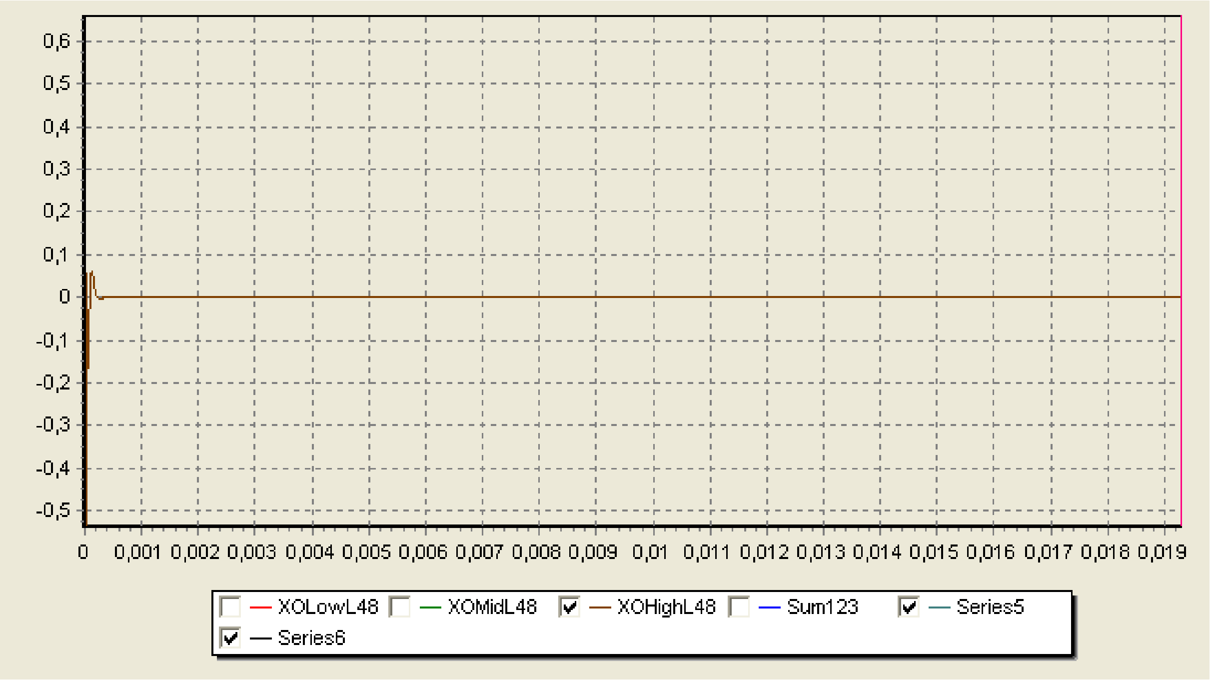

Fig. 4 Highpass pulse response (minphase Buttherworth 2nd order 3500 Hz)

The highpass peak has the value 0.6542 at sample 0. We can see that the three filters have their peaks at different times. All filters start at the time 0 but the reaction takes place at different times. For a better comparison the time axis of the three curve graphs is left unchanged.

Now let us assume that we have the best available speaker drivers. They are SO GOOD that their behaviour is described by our ideal Dirac pulse. This thought experiment allows us to study the response of the speaker where the crossover filters define the overall answer. The crossover filter response determines the result for each driver and the sum of the filters determines the speaker response. We can simply add by Acourate the three time domain signals to get the sum. And by selecting the step response checkbox of the sum curve we get a display that corresponds to the often published curves in audio magazines as a part of a speaker review:

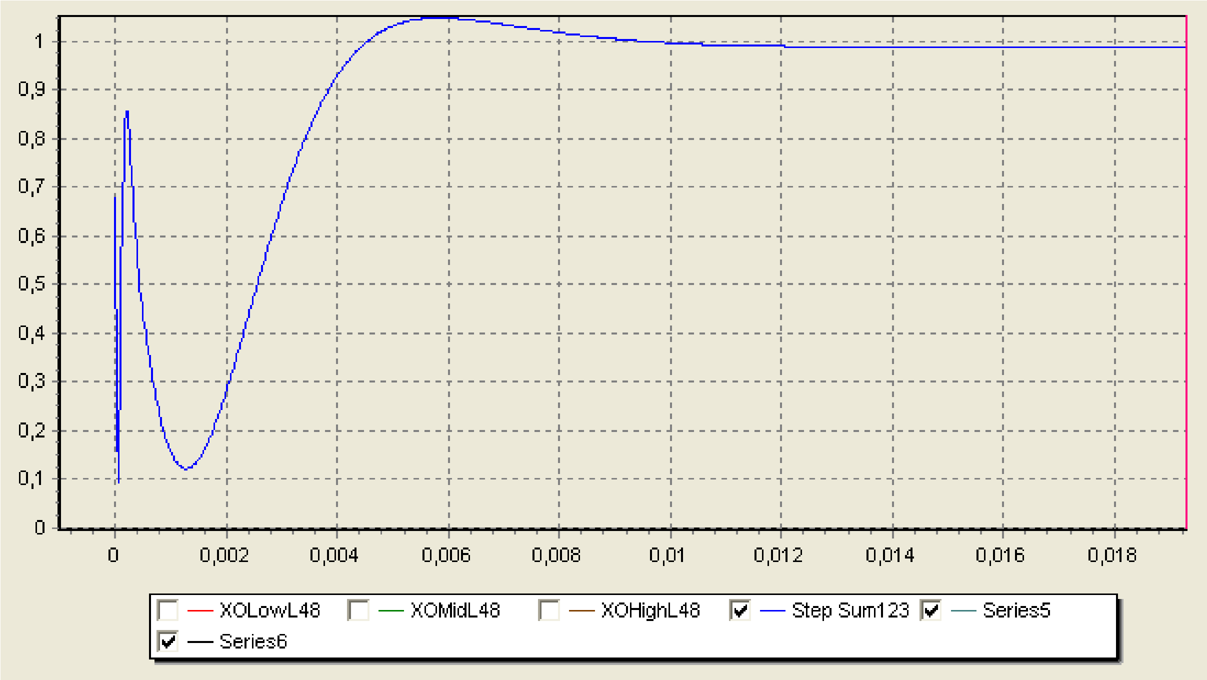

Fig. 5 Step response of sum signal (minphase Butterworth 2nd order 150 Hz, 3500 Hz)

Let us study the behaviour of the step response. First comes the tweeter (ideal tweeter + real highpass), then the mid driver (ideal midrange driver + real bandpass) and with a delay of about 6 ms the reaction of the woofer (ideal woofer + real lowpass).

It is important to understand what is happening here. Even in the case that we have our assumed ideal speaker drivers the resulting step response is determined by the minimum phase crossover filters. We can also suspect that the well-known and often published step response curves are also a result of the applied passive crossovers.

If a passive crossover is substituted by a digital crossover but the design is following the standard minphase definition then we will get NO IMPROVEMENT.

To understand a little more about the minphase behaviour let us explore a crossover of higher filter order e.g. the same Butterworth filter but instead of order 2 now with order 20. What is predictable anyway is a longer delay with higher filter orders. The more steep the slope is the longer is the filter length or simply spoken the more delay is given. This we can see in the picture 6. Doesn't the step response look really bad? No wonder why people with sensitive ears tend to prefer very low crossover filter orders. Please also not that the woofer now instead of 6 ms (2nd order) reacts at 14 ms !

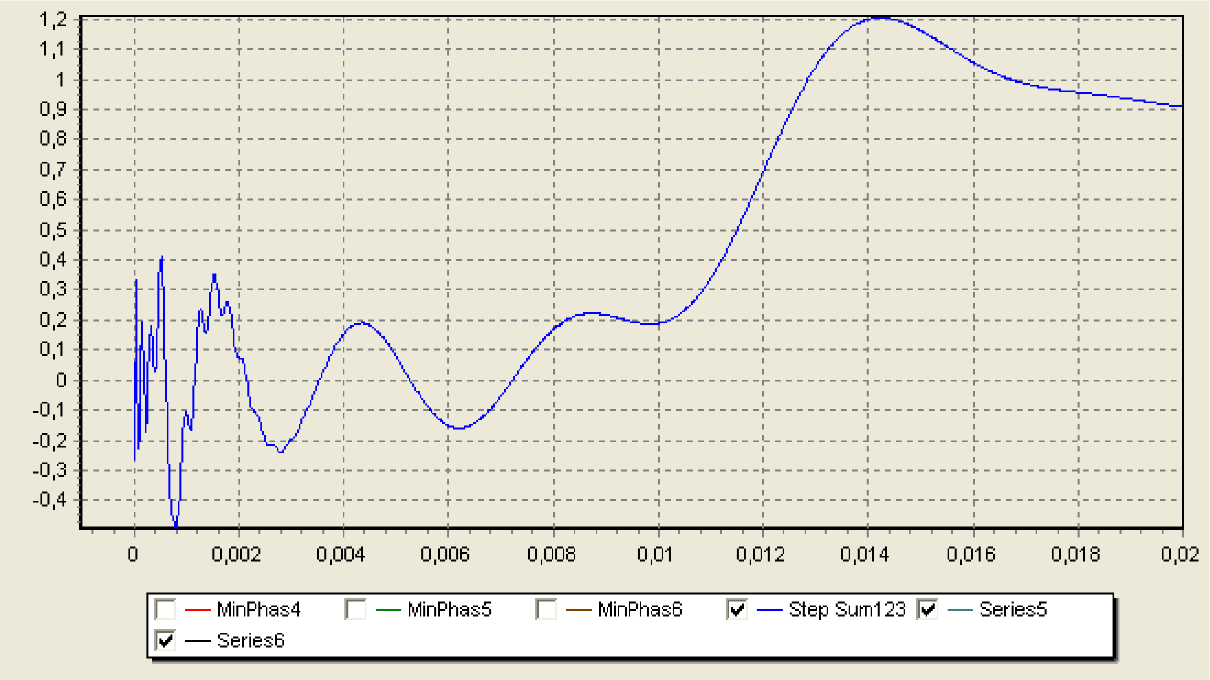

Fig. 6 Step response of sum signal (minphase Butterworth 20th order 150 Hz, 3500 Hz)

We can definitely see with the higher order minphase filter that we have to expect different expanded delays for different frequencies. And we can consider this as a serious time distortion.

Is the time distortion the only bad effect? No, for example let us take a view on the amplitude response of the sum signal of the 2nd order Butterworth crossover, given in the blue curve of picture 7. The amplitude at the crossover point is – 6 dB. This means a value of 0.5. But it seems that 0.5 + 0.5 is not equal 1.0 (0 dB). We get -8 dB to -9 dB.

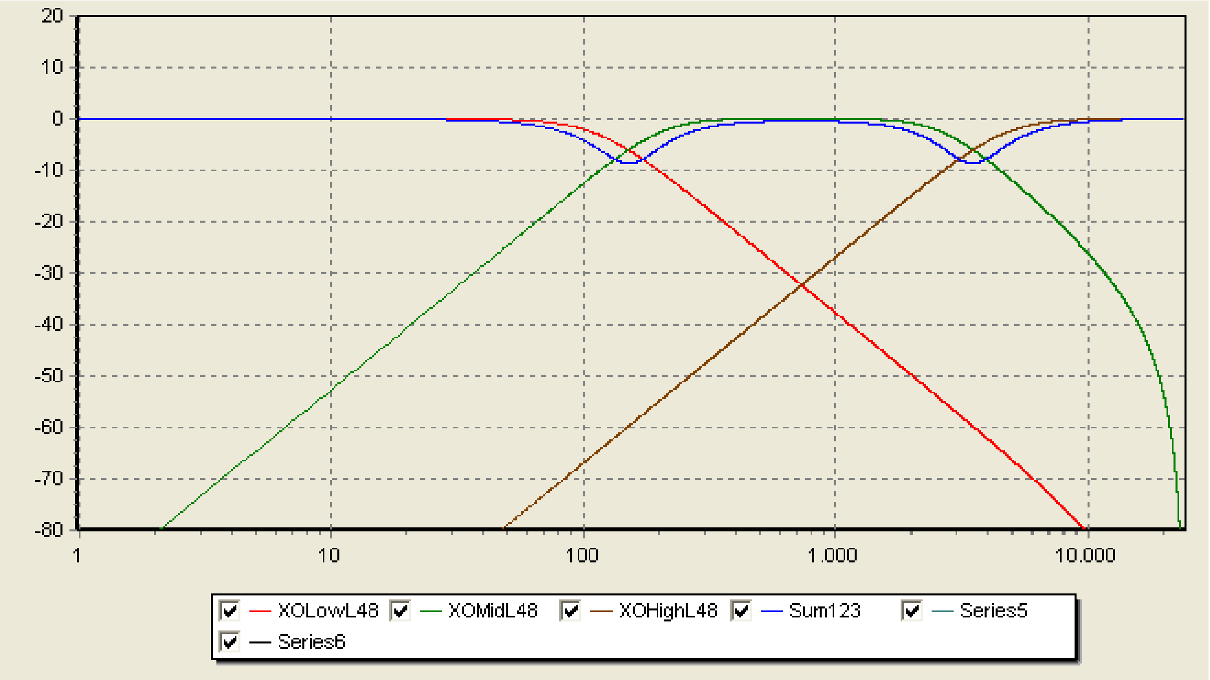

Fig. 7 Amplitude response of sum filter (minphase Butterworth 2nd order 150 Hz/3500Hz)

The reason is that we add two signals of the same frequency but with a different phase. In the best condition we can get the desired 0 dB, but in the worst condition two signals with identical frequency can even cancel each other. This is not the case here because the phase shift is not 180°. But in any case we do not get the expected straight frequency response.

Conclusion: Minphase crossovers have disadvantages. Different type of minphase crossovers have different type of disadvantages. With higher filter orders the behaviour gets worse. This has lead to the preference of low order filters. But also low order minimum phase filters result in time distortions. The different frequencies are played at different times. Is it a wonder that we have a never ending discussion about the quality of speakers? **Please note that we have discussed here only the crossovers with the assumption that the drivers and the speaker cabinet behave perfect** !

2. Delayed Minimum Phase Crossovers

The pictures 2-4 show the peaks of the pulse responses at the positions at sample 74, 3 and 0 for lowpass, bandpass and highpass. Thus they are at different positions. An idea is to apply some delay for the bandpass and the highpass to bring all peaks together. So the idea is that the sum will have the highest value at position 74 and we get closer to a Dirac behaviour. This situation is shown in the next picture. It has been created with Acourate by insertion of 71 zero samples at the begin of the bandpass time domain signal and by insertion of 74 zero samples at the start of the highpass signal.

Fig. 8 Pulse responses of bandpass and highpass with shifted peak positions

The next picture shows the resulting step response of the sum of the pulses (blue curve). Also shown are the step responses of the crossover filters. We can see that the step response still is far away from looking good.

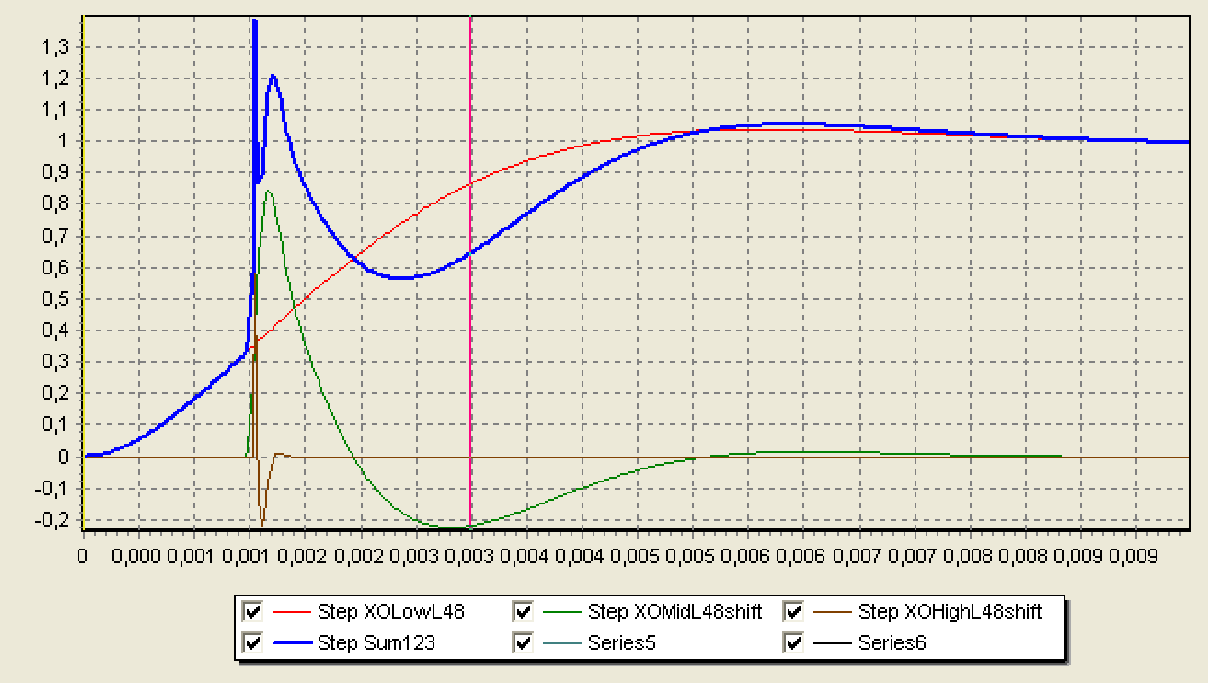

Fig. 9 Step response of added crossover filters with shifted peak positions

And we can already feel that the amplitude response also is not flat under these circumstances. In reality is has got even worse !

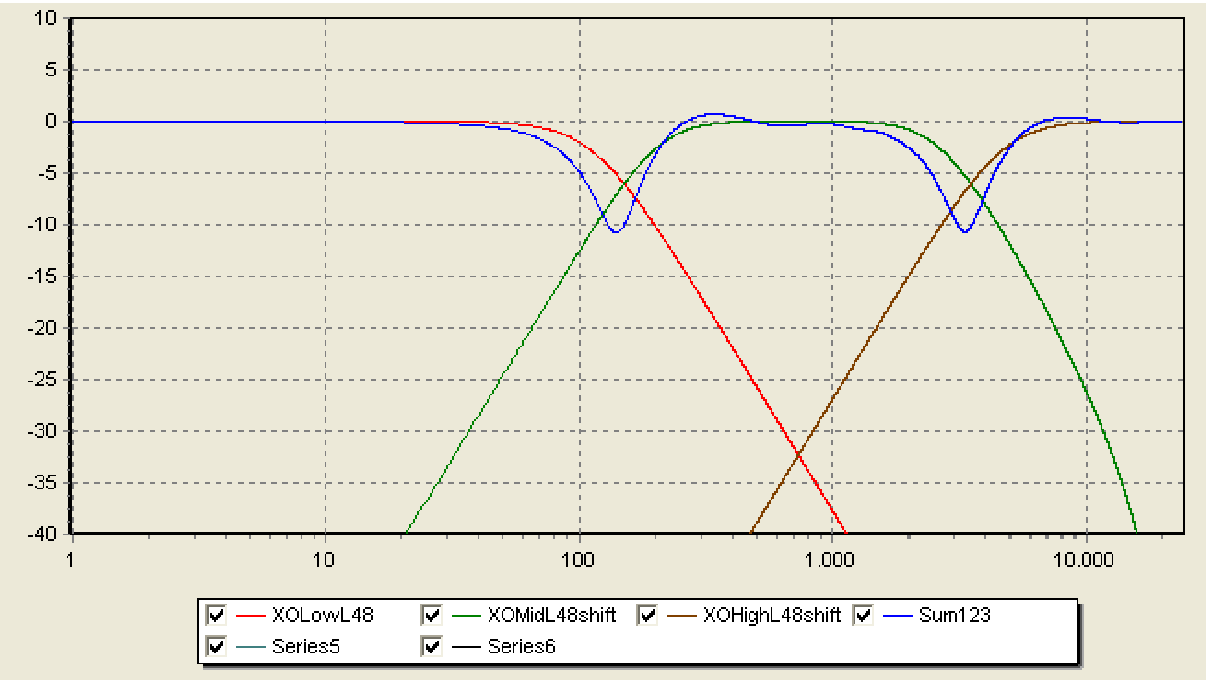

Fig. 10 Amplitude response of added crossover filters with shifted peak positions

Conclusion: The trick of simply using additional delays does not help. Of course now we can try to apply other modifications. So e.g. you can look for the proposals of Le Cléac'h. But we already feel that this will only give suboptimal solutions.

3. Linear phase crossover filters

The previous two chapters show that minimum phase crossovers will not give perfect results. So we think about another solution. What we expect is a perfect addition of the crossovers resulting in a Dirac pulse. Then we have the 1:1 transfer and our crossovers behave together like the famous piece of wire.

Let us look at 2nd order Butterworth filters with the same corner frequencies as in the previous chapters but now created by Acourate. These filters show linear phase crossover filters. Linear phase filters show a symmetric behaviour. The peak is in the center.

The centered peak of course has some important properties: we will get a delay. In our example with 65536 filter taps we will get a delay of 32768/samplerate seconds, e.g. at 48 kHz samplerate the overall delay is about 0.683 seconds. This is no problem if we have the sound without being accompanied by pictures (video). If we listen to music only the delayed playback doesn't matter. (Remark: also in case of video a solution is possible. We have to accept compromises and to take mixed phase filters. They also allow a perfect addition but we will have restrictions on the filter slopes. Another solution is to delay also the picture, but this is considered to be difficult. And finally a solution is to simply accept the minphase crossovers as the ear becomes much less critical if the eye is concentrating on the picture).

More important with a linphase filter is of course the question about the pre-echo or pre-ringing. If our ears will detect it then we will not accept linear phase filters. And this property of a linear phase filter is quite often be questioned and discussed. Let's see.

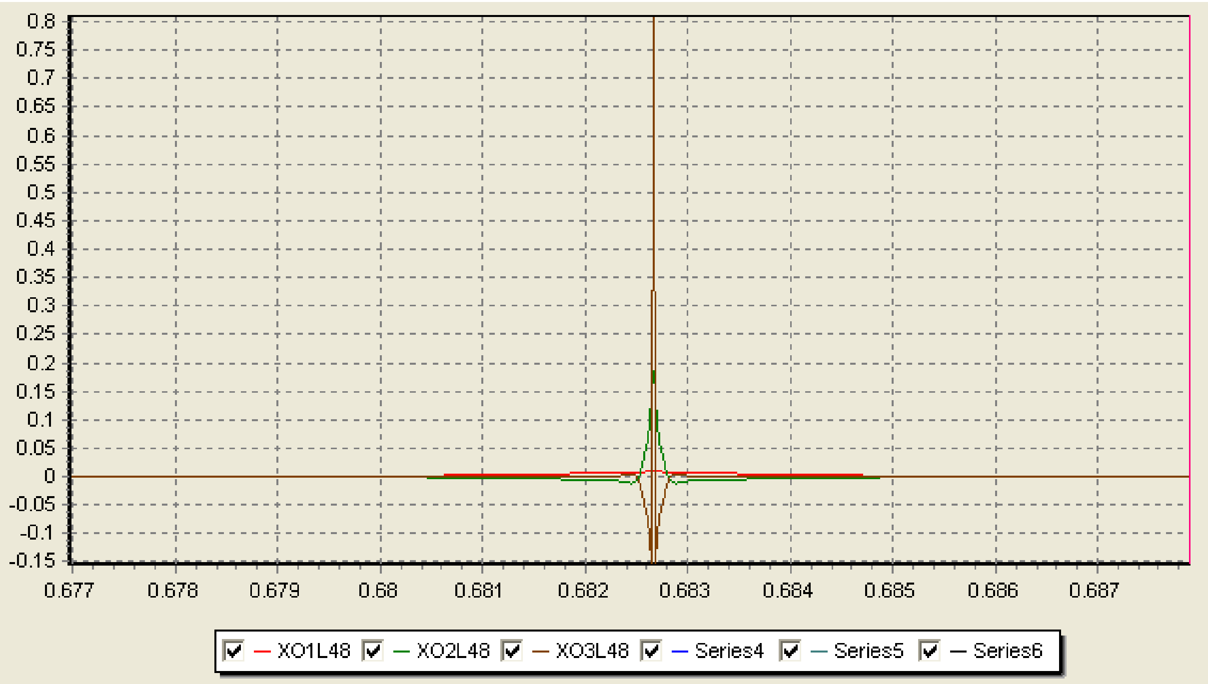

Fig. 11 Pulse responses (lowpass – red, bandpass – green, highpass – brown) of a 2nd order linphase Butterworth filter with corner frequencies 150 Hz, 3500 Hz

The amplitude response is shown in picture 12.

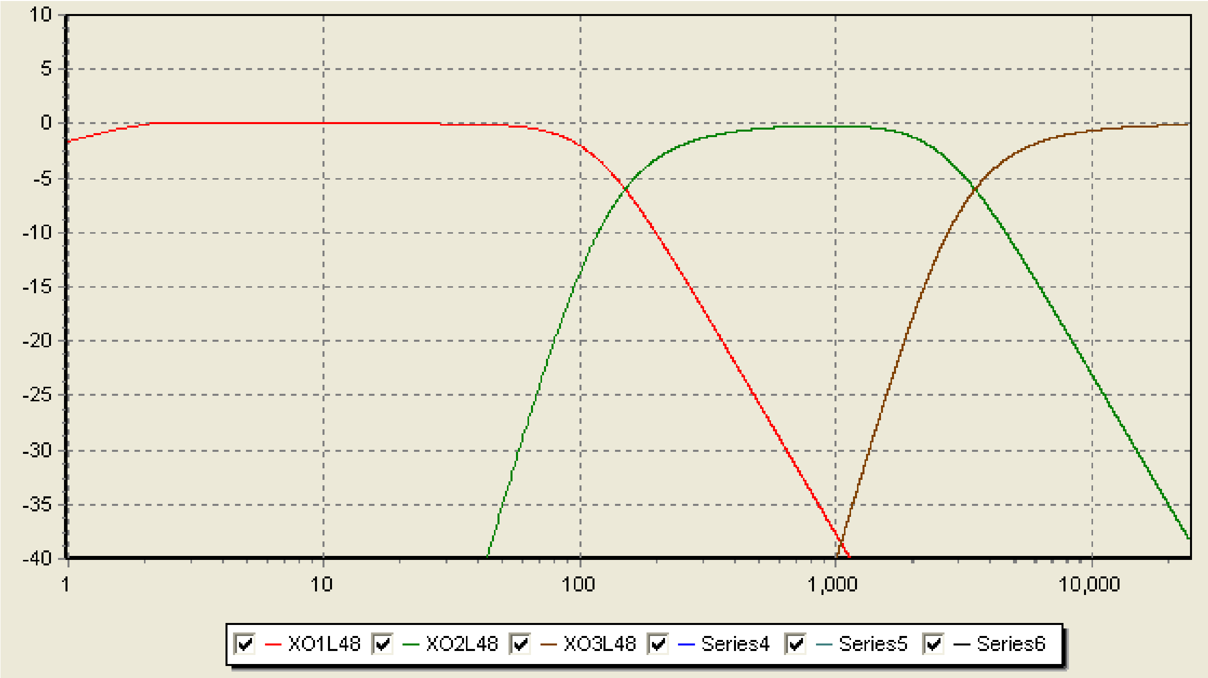

Fig. 12 Amplitude responses of the Acourate 2nd order linphase Butterworth crossovers

But now we are interested in the behaviour of all three crossovers together. How to they add together?

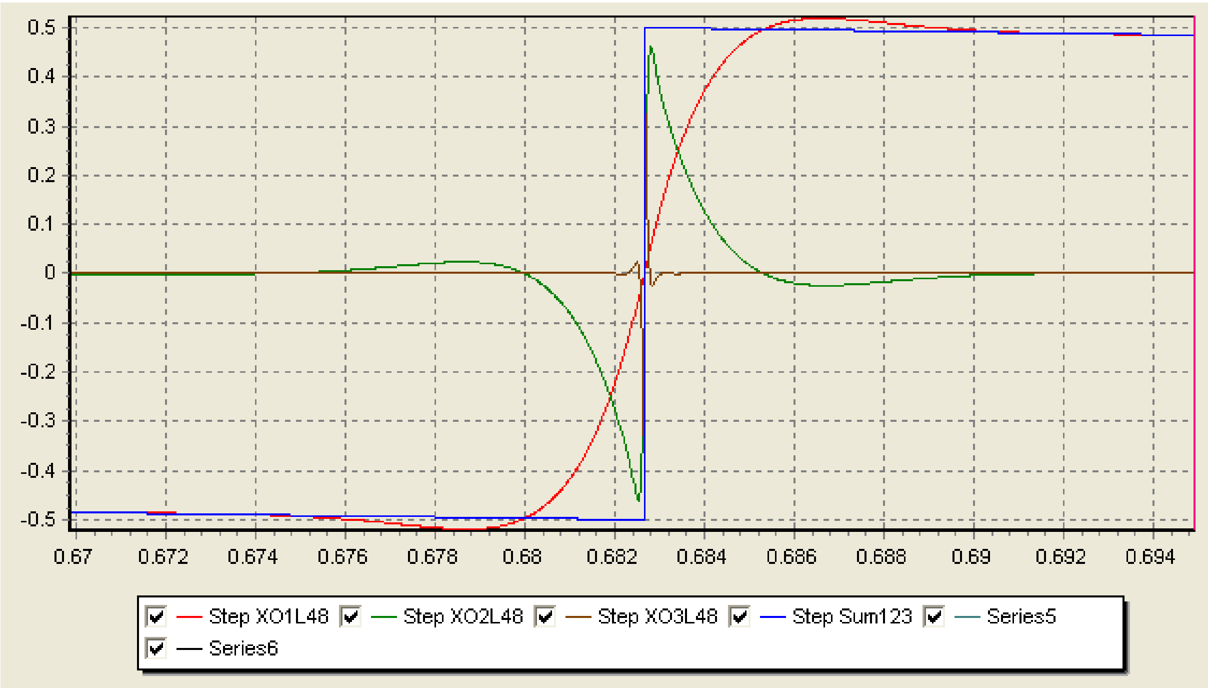

Fig. 13 Step responses of the linphase crossovers including the sum (blue curve)

The three crossovers add to a Dirac pulse and thus the step response shows the behaviour that we wish. Please note that the lowpass shows an overshoot. But it is compensated by the bandpass. And this is valid at any time. The filters compensate each other so that the sum is perfect.

If the sum is ok then also the amplitude response should look fine:

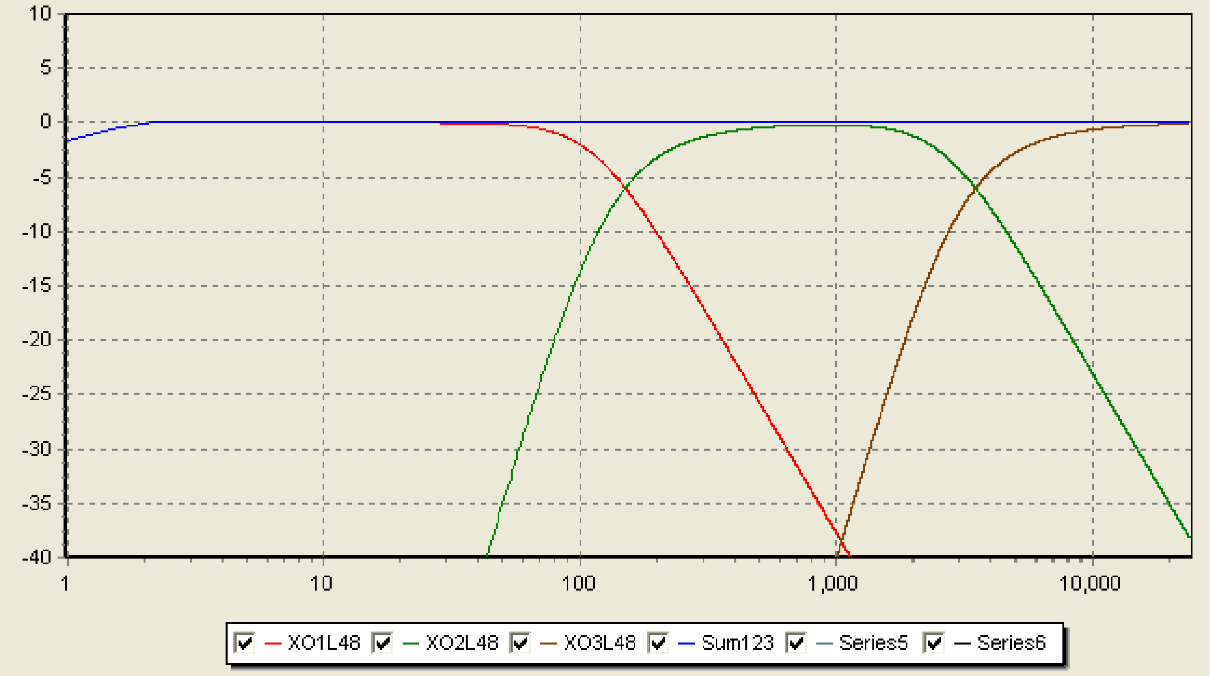

Fig. 14 Amplitude response of the added crossovers

Conclusion and discussion: With the linear phase crossovers generated by Acourate we get a perfect behaviour. The crossovers add up to a Dirac pulse. So we have reached a first goal in our speaker optimization. Based on the thought experiment with assumed perfect drivers we can concentrate on the behaviour of the crossover filters themselves. And it can be shown that linear phase filters allow us to reach the desired function.
Are Butterworth filters the only possible filters? NO. We can design filters with different shapes and they all fulfil our goal. Examples (the same corner frequencies are taken):

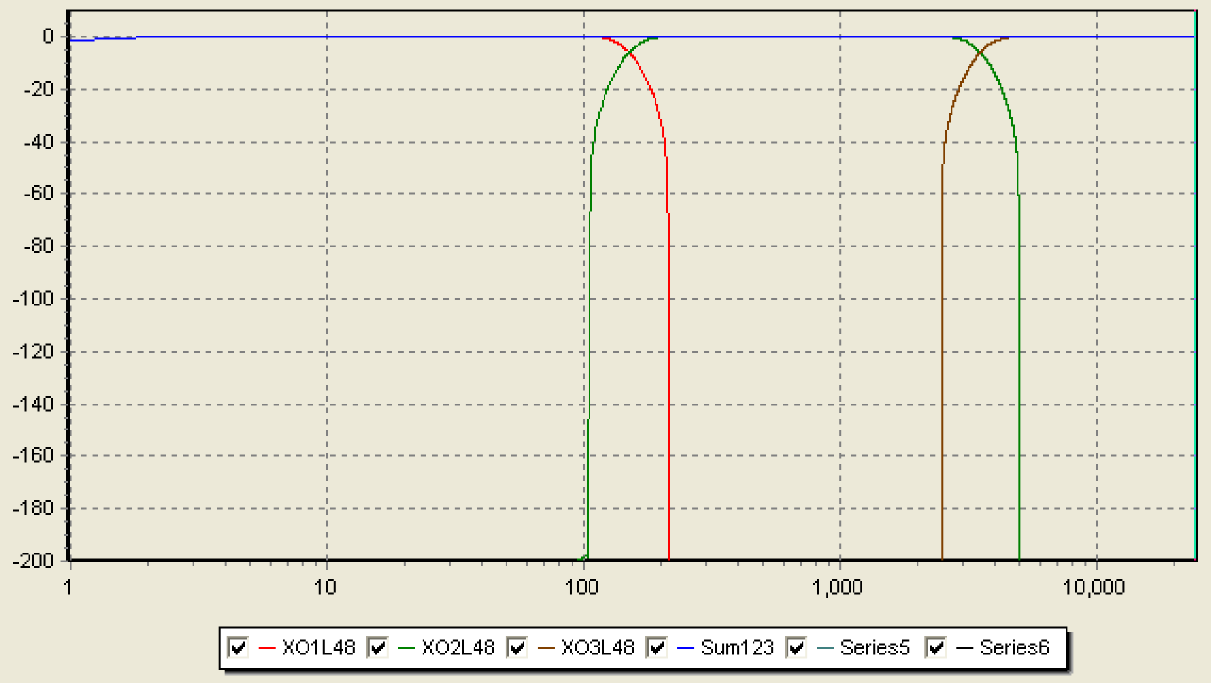

Fig. 15 Amplitude response of a 2nd order linphase Neville-Thiele crossover (please note the STEEP slopes)

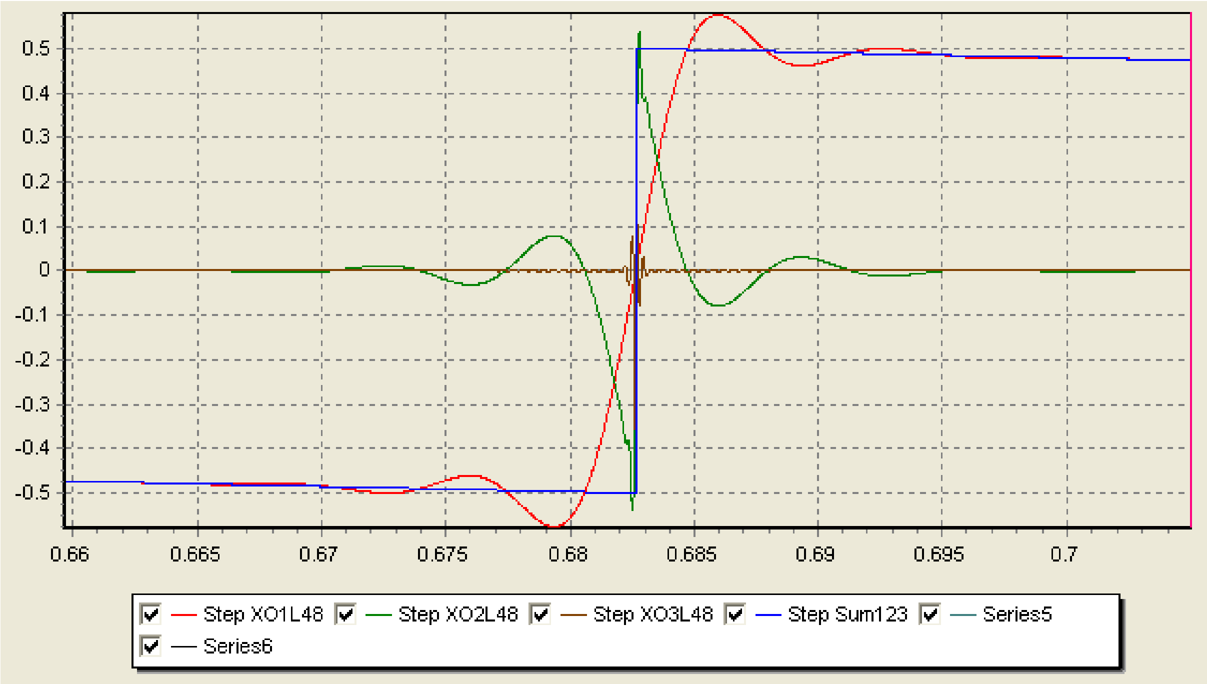

Fig. 16 Step response of the NT filter according to Fig. 15.

Of course the steep slope of the NT filter (that is perfect if we really want to separate our driver from certain frequencies, e.g. resonances of metal membranes) shows more ringing. But the sum is still perfect.

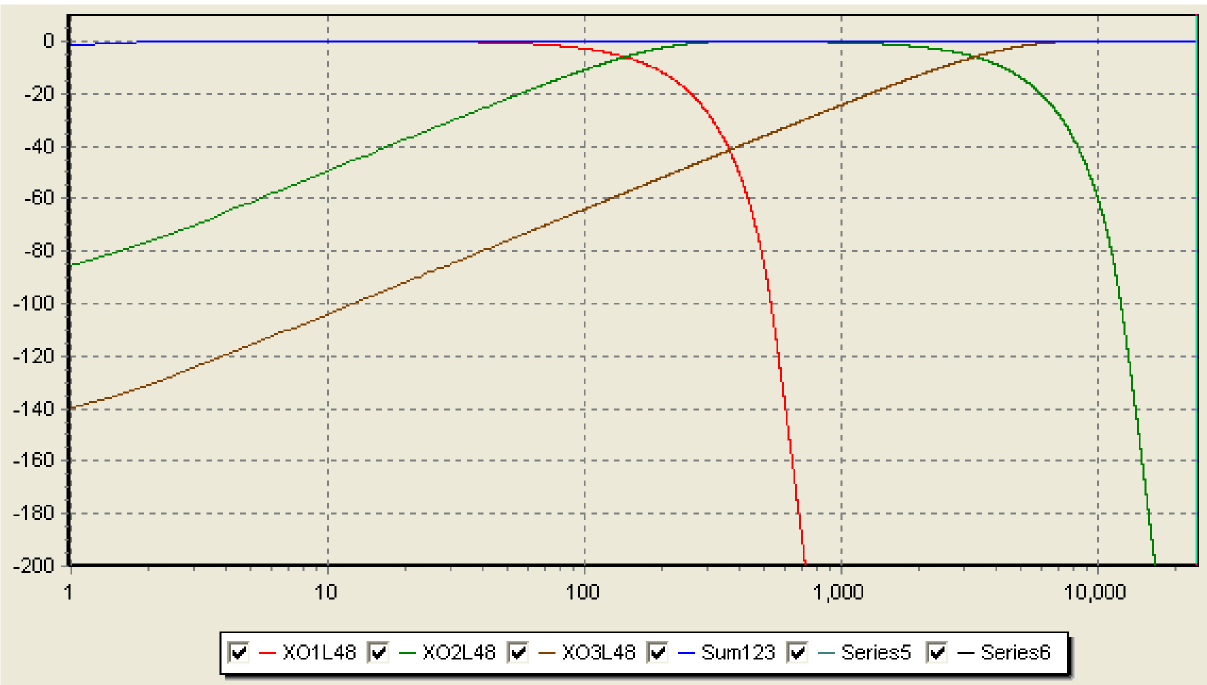

Fig. 17 Amplitude response of a R-Bessel filter of order 40.

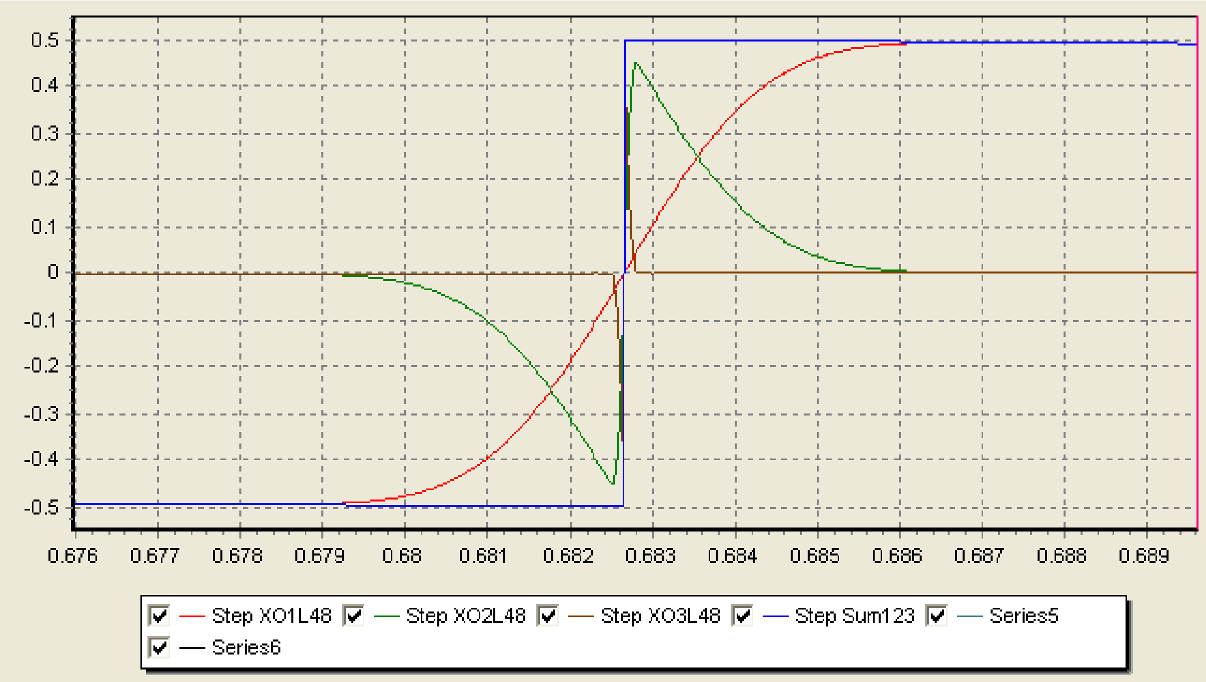

Fig. 18 Step response of the 40th order R-Bessel filter

Our example in picture 17 and 18 shows a Bessel filter of type R. This means that we have a steep slope at the right side of the crossovers. The left side always has the order 2. This is a property of the Bessel filter. A remarkable property can be seen in the step response picture. We do not have any overshoot in the lowpass despite of the high order !

How is this all achieved? Why do we always get our perfect step response?
We can simply do something that is much more difficult to achieve with passive crossovers: we can use the method of subtraction. If we subtract our first digitally generated crossover from the Dirac pulse then the rest defines a remaining part. And we can create a next crossover and subtract this again. By this way we get all our crossovers and of course they will in return add perfectly to our Dirac pulse. You should think about how to achieve this with passive analog components only.

Can the subtractive method also get applied for minphase filters?
Acourate will allow you to play with this and to study the result (Acourate is intended as an audio toolbox and not as a one button solution). You will find that you can do it but the amplitude responses will look very weird. The linear phase approach shows a much better behaviour. But it is also possible to create mixed phase filters by making some compromises (especially regarding steep slopes). The mixed phase filters have much less delay. So a delay down to about 60 ms should be possible.
If you do listen to music only then time is not important and thus linphase filters are preferable.

A final important question in this whitepaper:

All the discussion is based on the assumption of having perfect drivers. But speaker drivers are not perfect. Is the approach shown here still valid?

YES. Let us consider a bandbass. We have to expect that in the passband our driver should already behave well. In the transition area and in the stopband the driver has less influence, especially in the stopband as the driver should simply spoken get no signal to play. So the crossover has to be selected so that the good properties of the driver are used ! If the driver does not have a good behaviour we should not use it.
The goal is to select the driver's frequency range where the driver behaves well by nature (its own physical properties). Then only some small non-perfect properties have to be corrected (e.g. a

correction in the stopband is senseless). The driver has to be linearized. And the good news: Acourate supports this linearization.
The idea: if the desired crossover plus the driver plus the driver linearization behave like our ideal crossovers shown in this whitepaper then the synthesis of a good sounding speaker is no longer a fiction, it becomes reality.

