Pre- and Post Ringing Of Impulse Response

Source:

Time (Temporal) Masking

"….Simultaneous masking describes the effect when the masked signal and the masking signal occur at the same time. Human hearing is sensitive to the temporal structure of sound, and masking also can occur between sounds that are not present simultaneously.

Pre-masking is when the test tone occurs before the masking sound. Post-masking is when the test tone occurs after the masking sound. The following figure shows the time regions of pre-masking, simultaneous masking, and post-masking in relation to the masking signal.
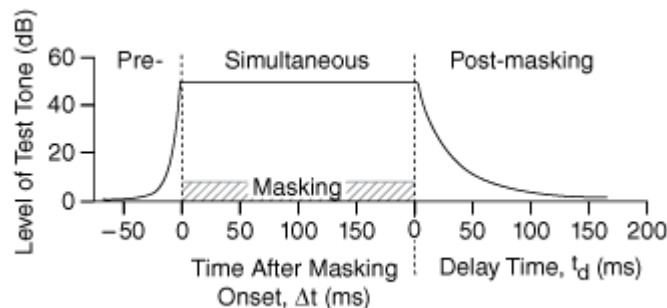

Figure 1. Slopes of temporal masking

Post-masking is a pronounced phenomenon that corresponds to decay in the effect of the masking signal. Pre-masking is a more subtle effect caused by the fact that hearing does not occur instantaneously because sounds require some time to sense. As indicated in the figure above, researchers typically can measure pre-masking for only about 20 ms.

Post-masking is the more dominant temporal effect and can be measured for 100 ms following the cessation of the masking sound. Both the threshold in quiet and the masked threshold depend on the duration of the test tone. Researchers must know these dependencies when investigating pre- and post-masking because they use short-duration test signals to perform these measurements….".

Testing Regime

Firstly, it is observable, that slopes of temporal masking in Figure 1 are plotted using lin-log scale, that is, time scale is linear in milliseconds, and level of test tone is in logarithmic scale (decibels). Since impulse responses, which we are going to examine, are typically calculated and plotted in linear scale, it is necessary to use the same vertical scale units and type – dBs.

Y decibels of a variable "X" are expressed as:

Figure 2. $Y = A * \log10(X) + B$

where A,B are suitably chosen screen display constants.

Now, in order for the slope of temporal masking display similar characteristics as on Figure 1, the "X" variable

Figure 3. $X = \exp(-T / C) + D$

Where T = time from T=T1 to T=T2, and C and D are suitably chosen constants.

I would use expressions on Figure 2 and Figure 3 to create temporal masks if we were to display impulse responses in logarithmic scale.

Calibration of the Simulation Process

I do not have mathematical formulas describing pre- and post-masking slopes on Figure 1. Therefore, I had to rely on visual inspection of Figure 1 and then adjusting coefficients A,B,C and D to obtain plots of both maskers, similar to the ones on Figure 1 (green curve on Figure 2). Please note, that pre-and post masking is shown from 0dB to 50dB on the vertical scale on Figure 1.
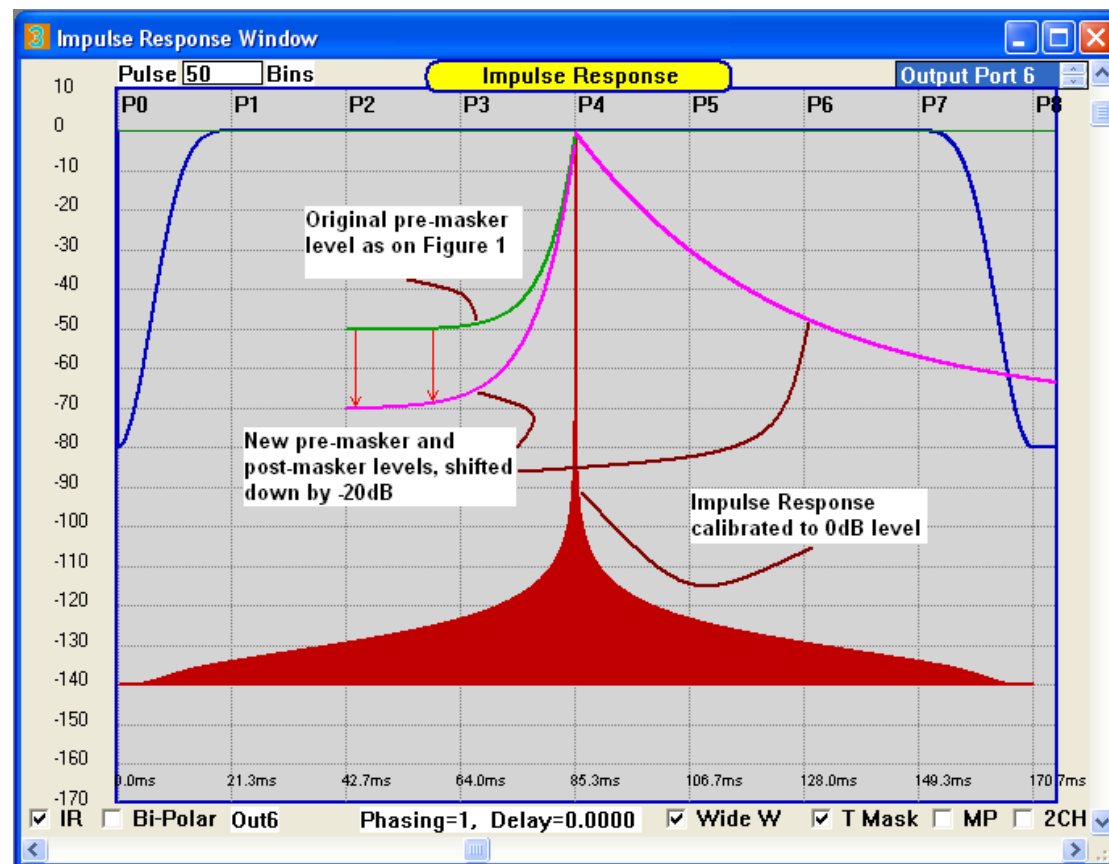

Figure 2. Calibrating Simulation Process for pure IR.

Figure 2 depicts calibration process for this simulation.

1. Impulse Response (IR) (red curve) is plotted with 10dB/div horizontal scale resolution. The IR plotted on Figure 2 was calculated from a system with a flat SPL level from within screen display limits of 10Hz-50kHz. Please note, that such system is still band-limited, but for our purpose, it is assumed flat.
2. IR of such system has a peak at 0dB level – see red peak at Figure 2. The floor of IR is clipped at -140dB level.
3. IR is now calculated and shown in logarithmic scale. Also, since the IR can be negative as it wiggles along the time scale, the negative values are shown as absolute values of the IR. (in C-language: Y = 20log10(fabs(IR))). This is why the IR looks differently from what you would typically see in MLS systems.
4. In order to increase confidence level in the simulation, I have dropped the pre- and post masker levels shown on Figure 1, by -20dB. **The new masker levels are now plotted in pink on Figure 2, and are extended over 70dB range.**

In addition, this paper extends IR of Linear-Phase (LP) and Minimum-Phase (MP) simulation into more practical area. I will also examine filter's response by creating very narrow **(2 samples width) excitation pulse** and **bipolar pulse** and feeding these pulses to both versions of the filter. From Wikipedia: http://en.wikipedia.org/wiki/Impulse_response

"…In practical systems, it is not possible to produce a perfect impulse to serve as input for testing; therefore, a brief pulse is sometimes used as an approximation of an impulse. Provided that the pulse is short enough compared to the impulse response, the result will be close to the true, theoretical, impulse response. In many systems, however, driving with a very short strong pulse may drive the system into a nonlinear regime, so instead the system is driven with a pseudo-random sequence, and the impulse response is computed from the input and output signals….".
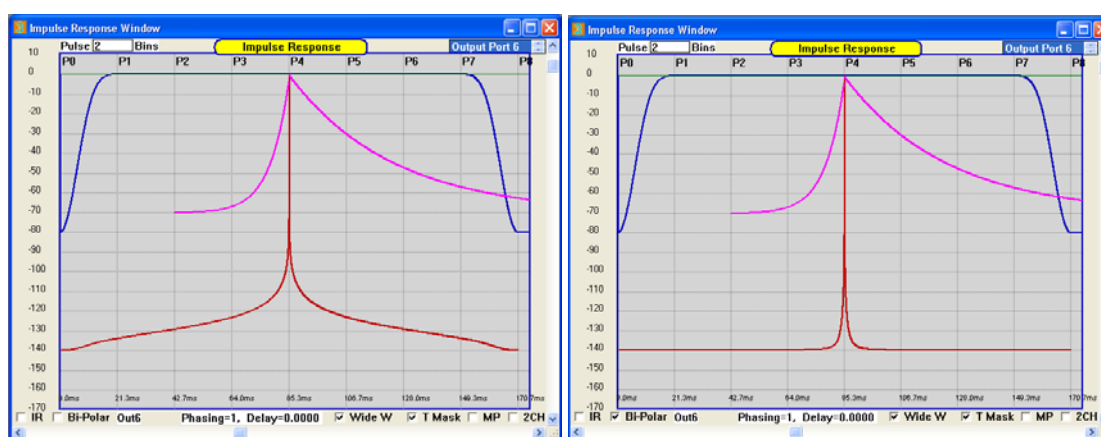

Figure 3. Calibrating Simulation for pulse   and   bi-polar pulse

It is observable on Figure 3, that "Pulse Response" (PR) of the flat system is significantly improved for bi-polar "Bi-Polar Response" (BPR), symmetrical pulse. The tails of the BPR are at the level of -140dB (the is -70dB below already reduced pre-and-post markers).

Please note, that PR simulation approach will inject twice as much energy into the filter, and BPR will inject four times as much energy into the filter. Therefore, I would expect pre- and post ringing to be worse than for pure IR – well, let's see.

Modelling process will follow the procedure below:

1. Calculate linear-phase and minimum-phase impulse responses of a selected filters.
2. Align minimum-phase impulse response with the centre of linear-phase impulse response for easy comparison.
3. Compare pre- and post ringing in both filters using pre- and post masker curves.

Impulse Response (IR) of Linear-Phase (LP) version is shown in red, and Minimum-Phase (MP) version is shown in green. Pre-masker and post-masker are shown in pink colour. MP version is time shifted, so that it's origin overlaps with centre of the LP version. This allows simple comparison of pre-and post ringing.

Pleas also note, that FFT windowing (blue curve on all figures) has been chosen to be very wide. This is deliberate, as it improves low-frequency performance of filters, exposing more of the filter's coefficients, that are responsible for low-frequency performance of the filter. If you choose to avoid this, you could select one of the standard windows, say Hanning type.

Examination of Several Audio Filters

Vented subwoofer. This situation is modelled as high-pass filter of F3dB=20Hz and slope of +24dB/oct.
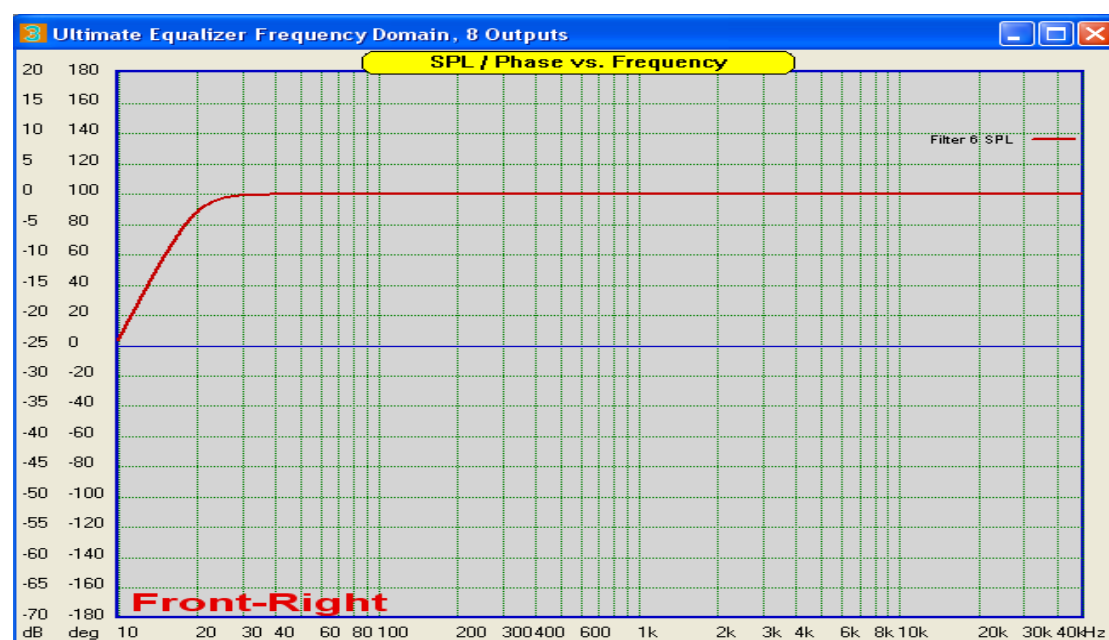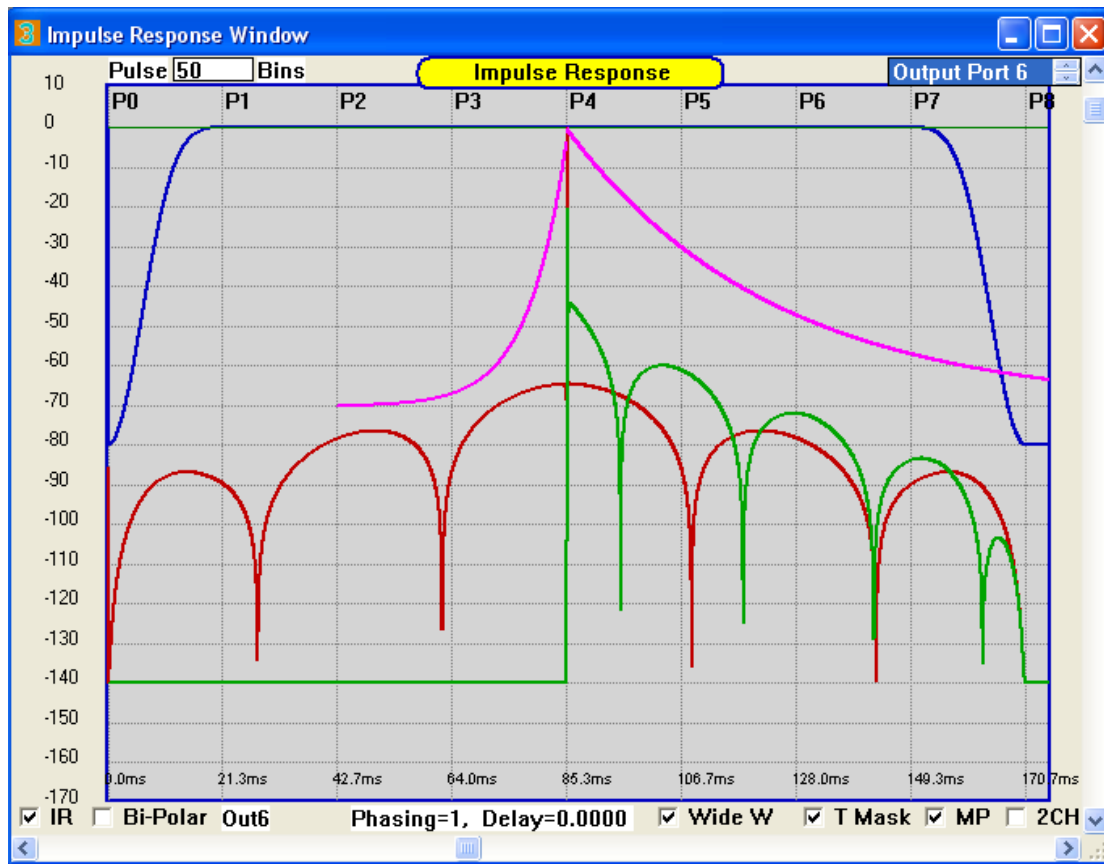

Figure 4. SPL of our vented subwoofer

Figure 5. LP (red) and MP (green) impulse responses.

It appears, that even if this subwoofer was working as a stand-alone device, neither MP nor LP processing effects would be audible.
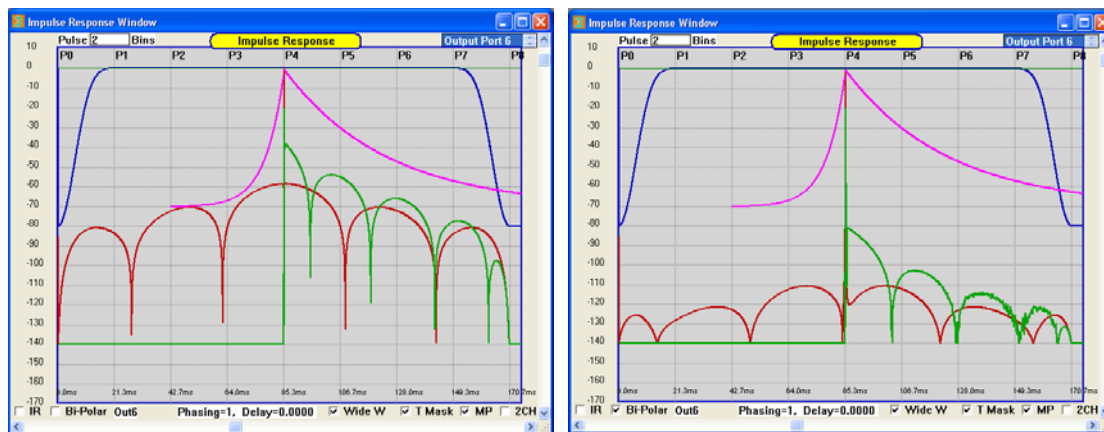

Figure 6. PR   and   BPR

It is clearly evident on Figure 6, that PR exhibits increased pre-ringing due to the pulse excitation being wider than the width of pure IR, but it's still acceptable.

Secondly, BPR excitation results in significantly lower pre-and post ringing. This is despite four times more energy injected into the filter. This is a good news as this would possibly be more realistic signal for testing. In addition, digital artefacts (or magnitude of the coefficients) of LP filter are at -110dB level and the artefacts of MP filter are at -80dB level. This is somewhat unexpected and clear win for LP filter.

Peaking EQ filter at 100Hz and Gain = +20dB, Q=10.
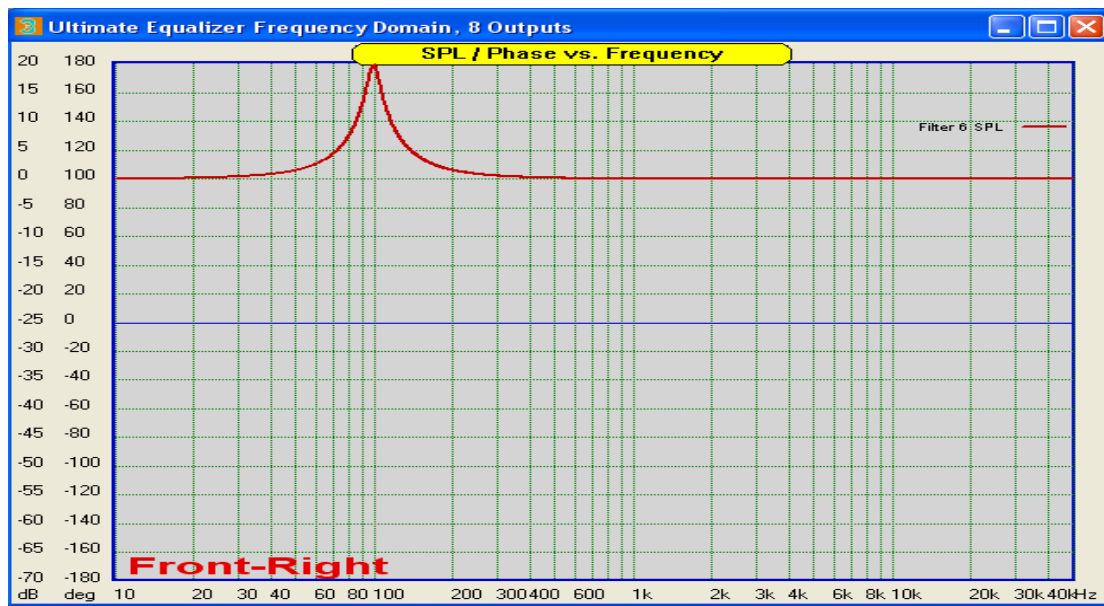

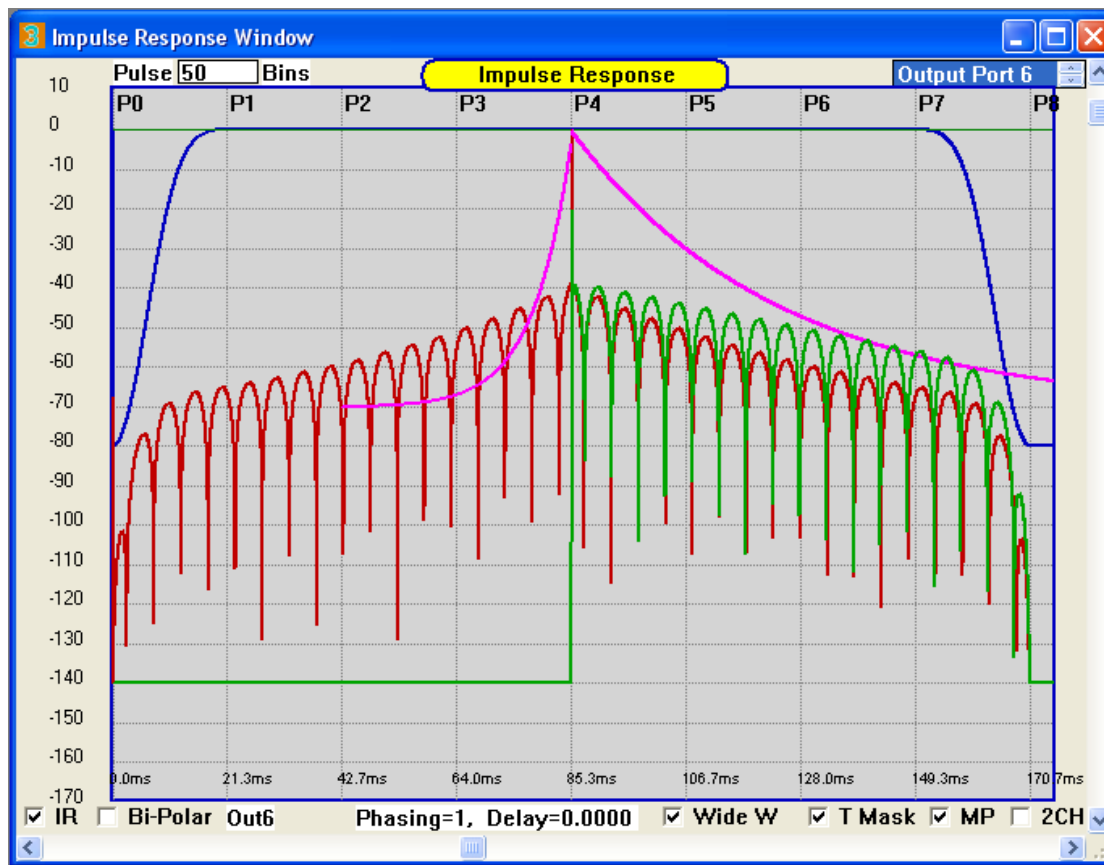Figure 7. SPL of a peaking filter.

Figure 8. LP (red) and MP (green) impulse responses.

It is very likely, that ringing in LP filter (red) would be audible. The pre-ringing clearly extends above the temporal pre-masker.
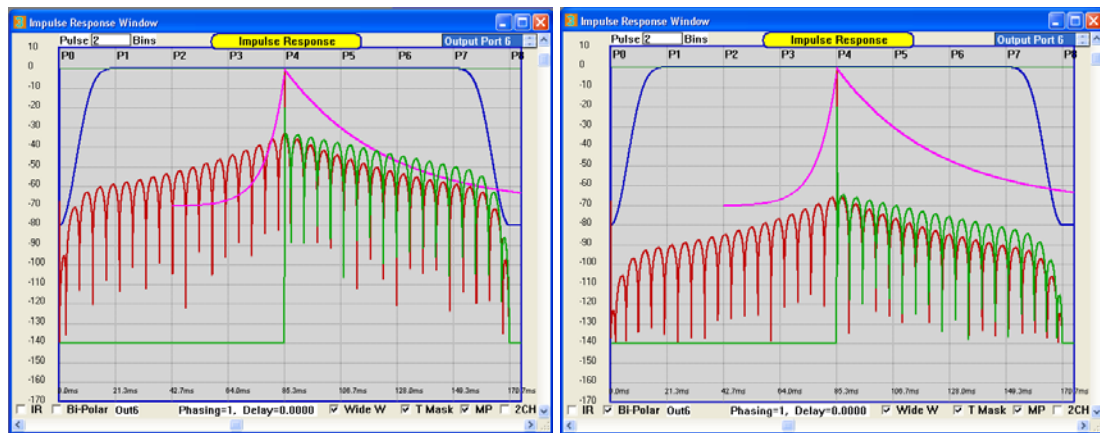
Figure 9. PR and BPR

Interestingly, as shown on Figure 9, the PR of the MP filter version, now rises above the post-marker level. Would this be audible?. The BPR is again much lower in ringing levels for both filter versions.

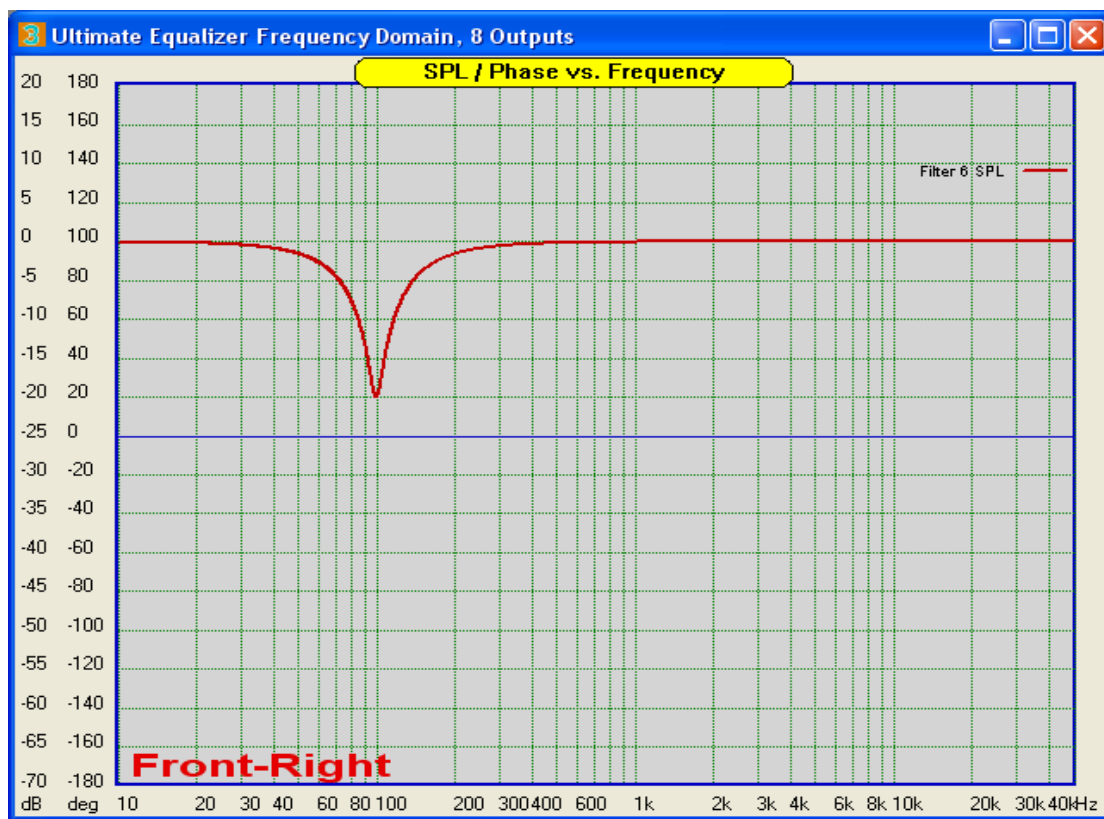Notching EQ filter at 100Hz and Gain = -20dB, Q=10.

Figure 10. SPL of a notch filter.
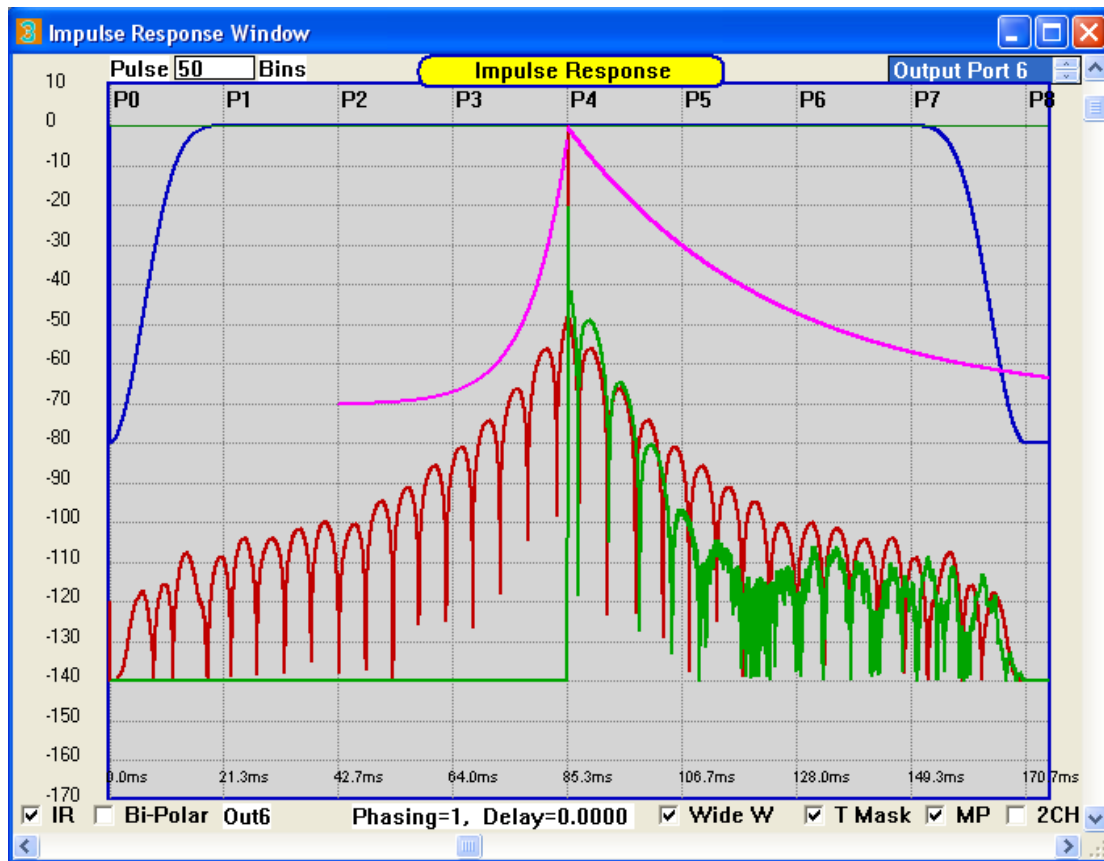
Figure 11. LP (red) and MP (green) impulse responses.

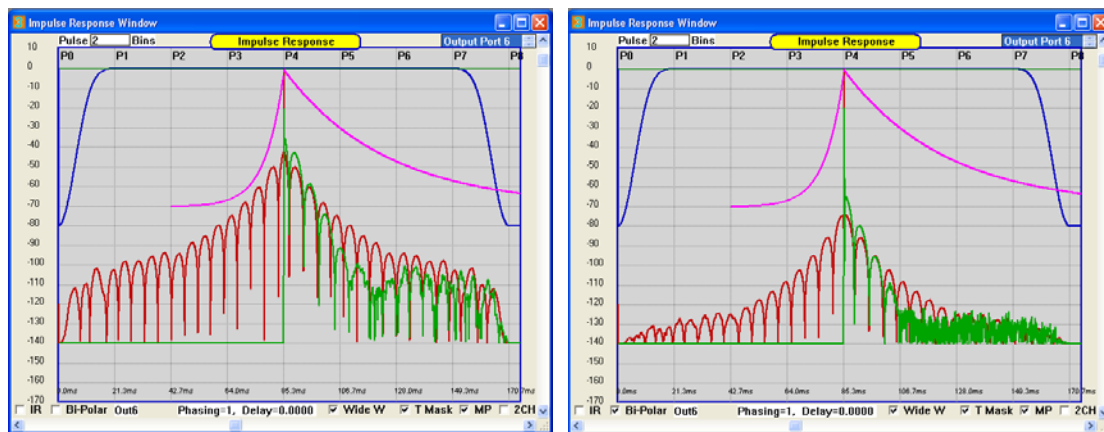It is unlikely, that ringing any in version: MP and LP would be audible.

Figure 12. PR   and   BPR

The PR and BPR are also good, and again the BPR is significantly lower in ringing.

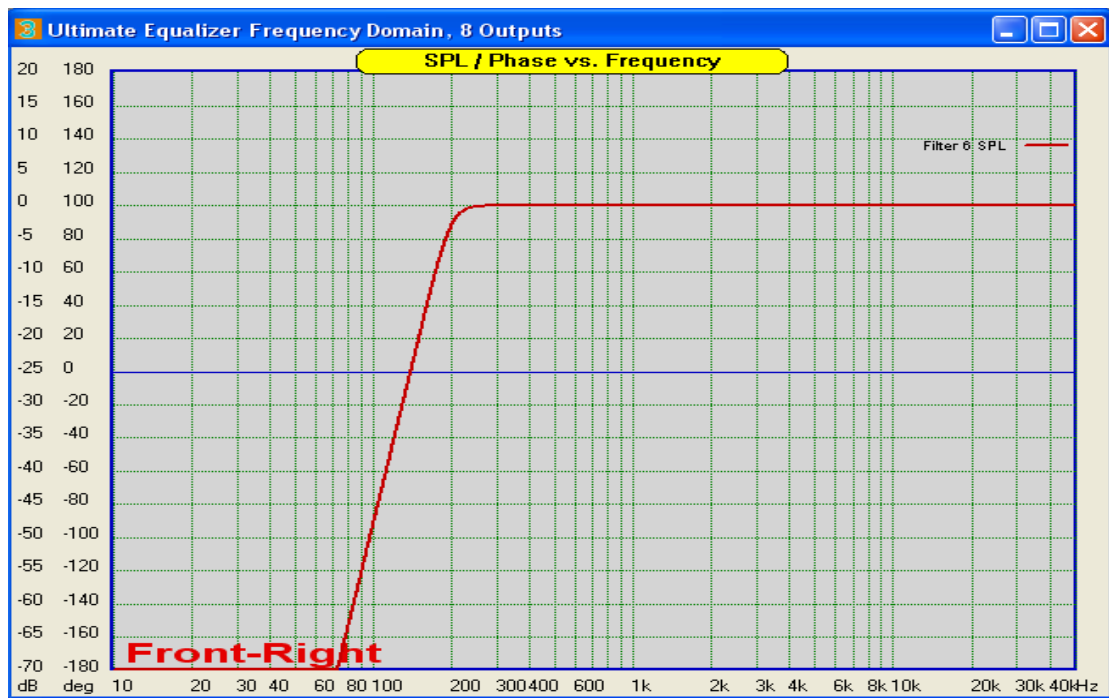High-Pass filter of F3dB=200Hz and slope of +48dB/oct.

Figure 13. SPL of a High-Pass filter

Figure 14. LP (red) and MP (green) impulse responses.

It is unlikely, that ringing any version: MP and LP would be audible. The ringing is simply too short, and falls steeply, and is masked completely.

Figure 15. PR   and   BPR

The PR and BPR are also good, and again the BPR is significantly lower in ringing.

Low-Pass filter of F3dB=1000Hz and slope of -48dB/oct.

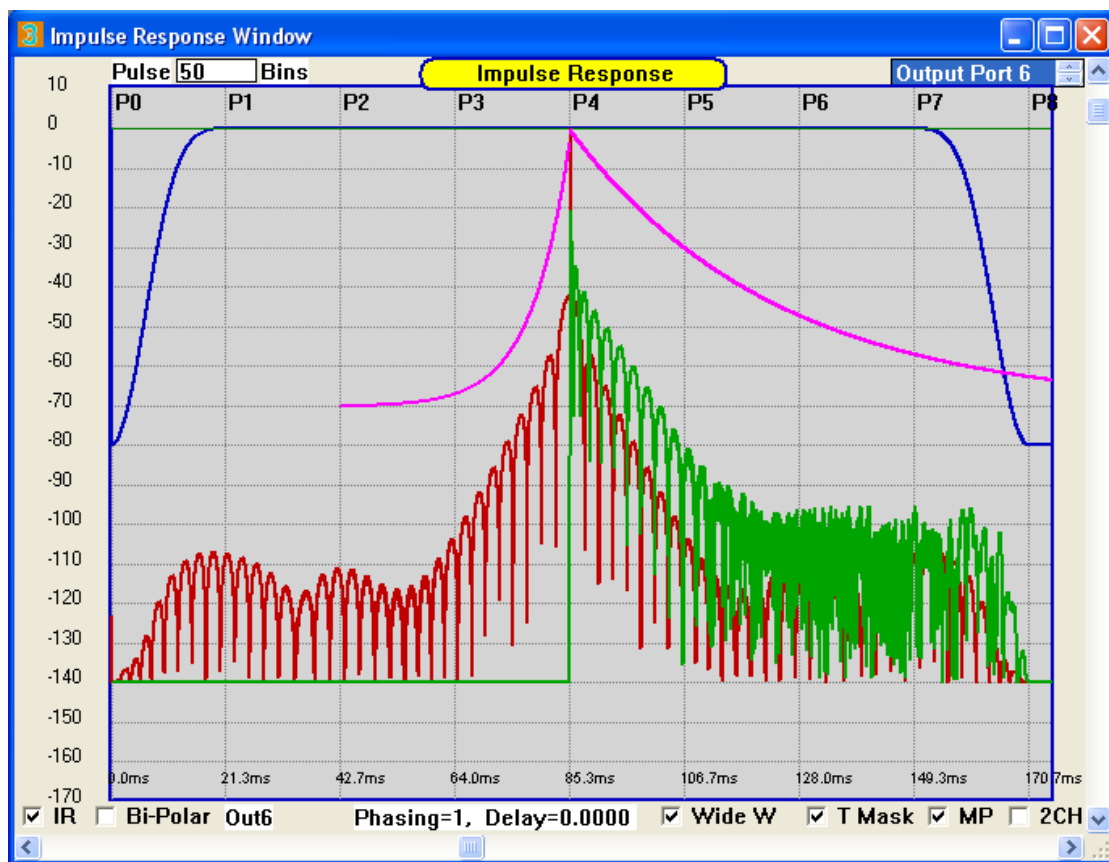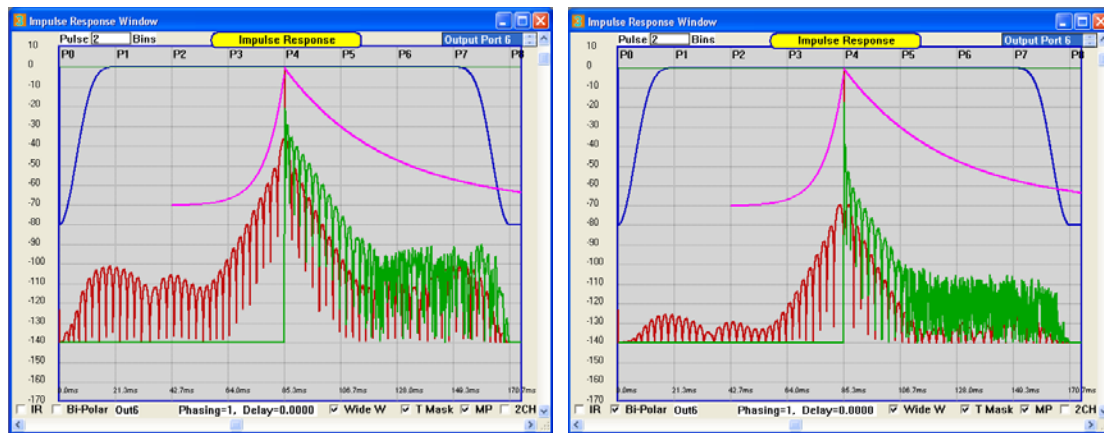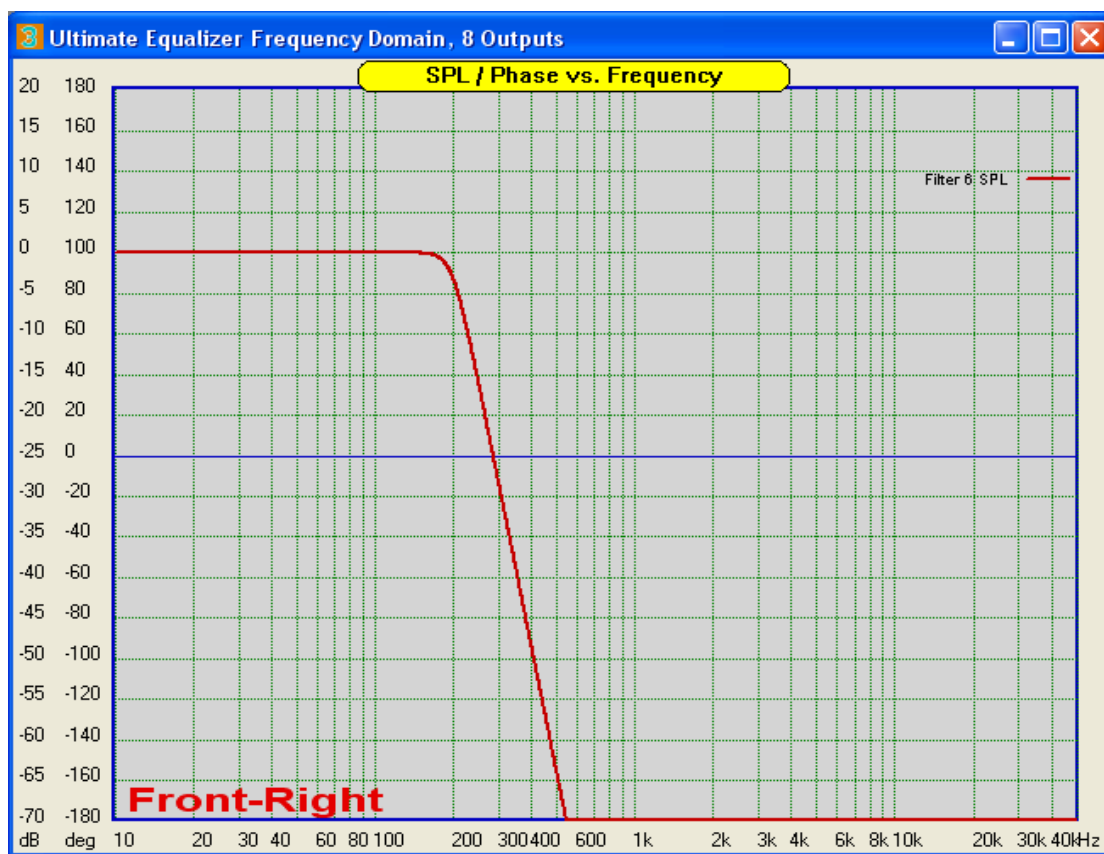

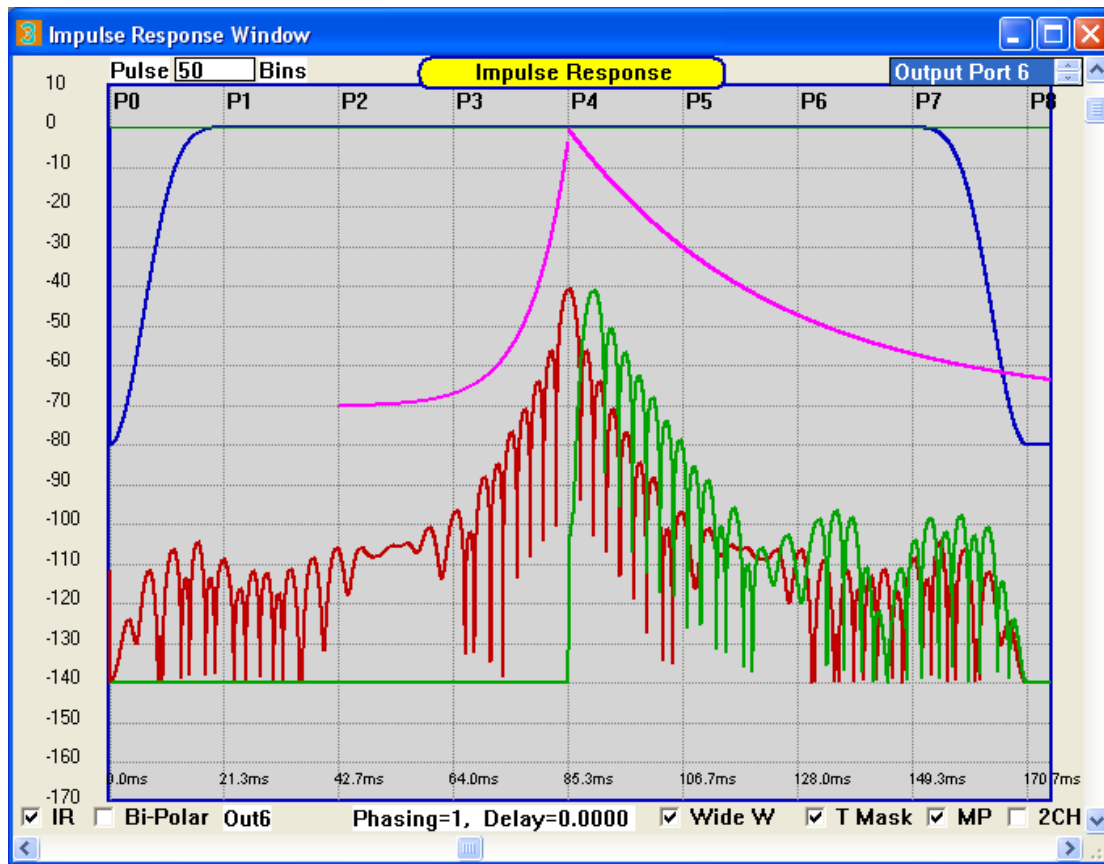Figure 16. SPL of a Low-Pass filter

Figure 17. LP (red) and MP (green) impulse responses.

It is unlikely, that ringing any version: MP and LP would be audible. The ringing is simply too short, and is masked completely.
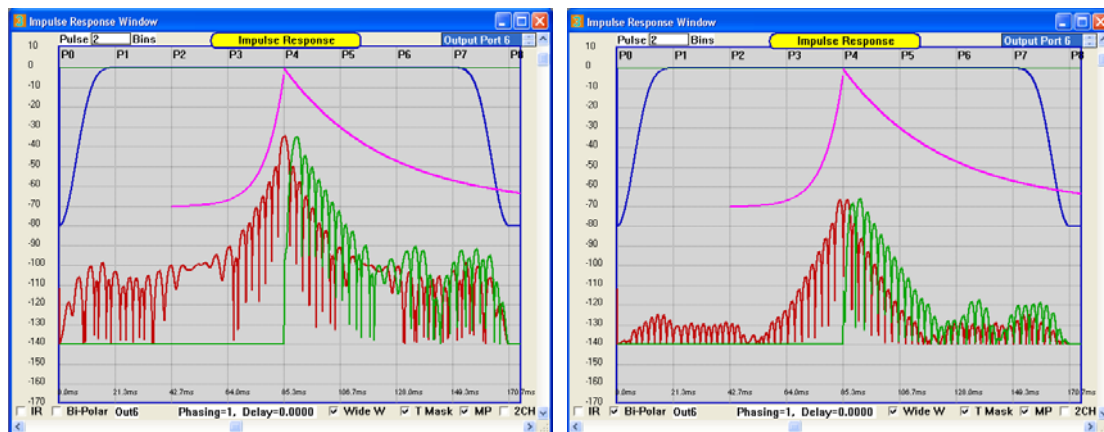

Figure 18. PR   and   BPR

The PR and BPR are also good, and again the BPR is significantly lower in ringing.

Examination of a Simple Crossover

The crossover consists of a low-pass filter with -96dB/oct slope and F3dB = 100Hz, and a high-pass filter with +96dB/oct slope and F3dB = 100Hz.
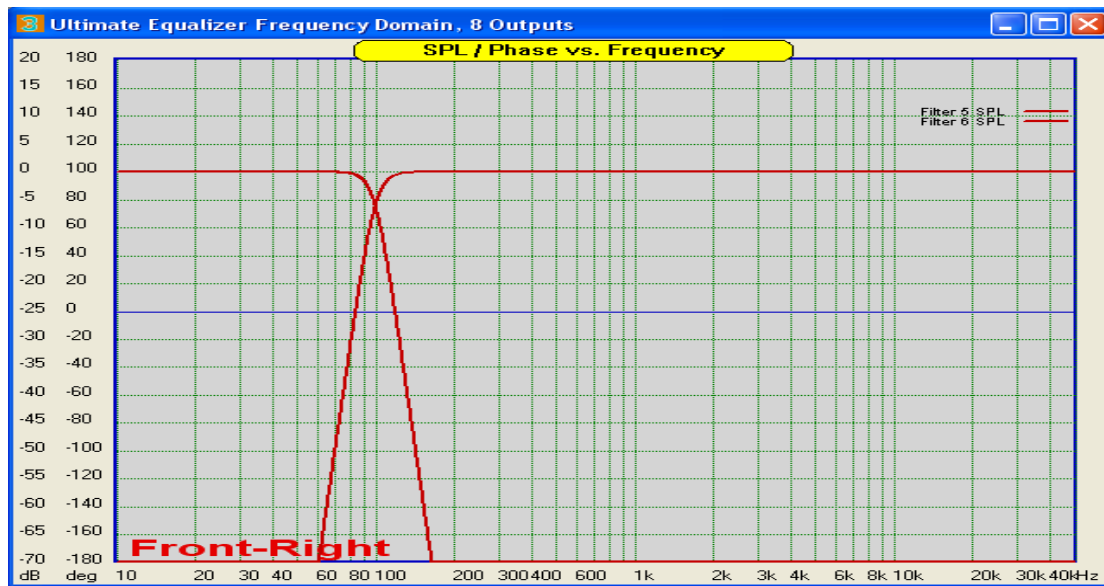

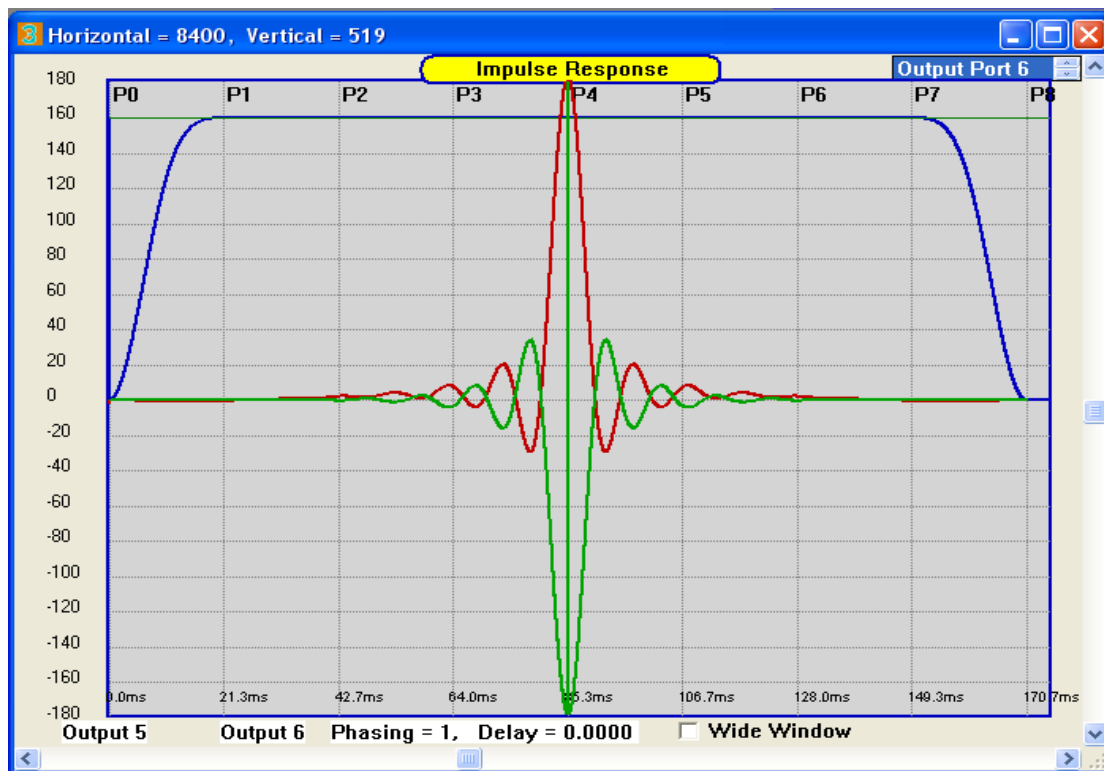Figure 19. SPL of a crossover.

Figure 20. LP (red) and MP (green) impulse responses (lin-lin scale).

It is easily observable, that adding low-pass (red) and high-pass (green) impulse responses will result in almost perfect cancellation of pre- and post-ringing, resulting in almost perfect impulse response.

MP version (shown below) of the two filters will not add to a perfect impulse response at all.
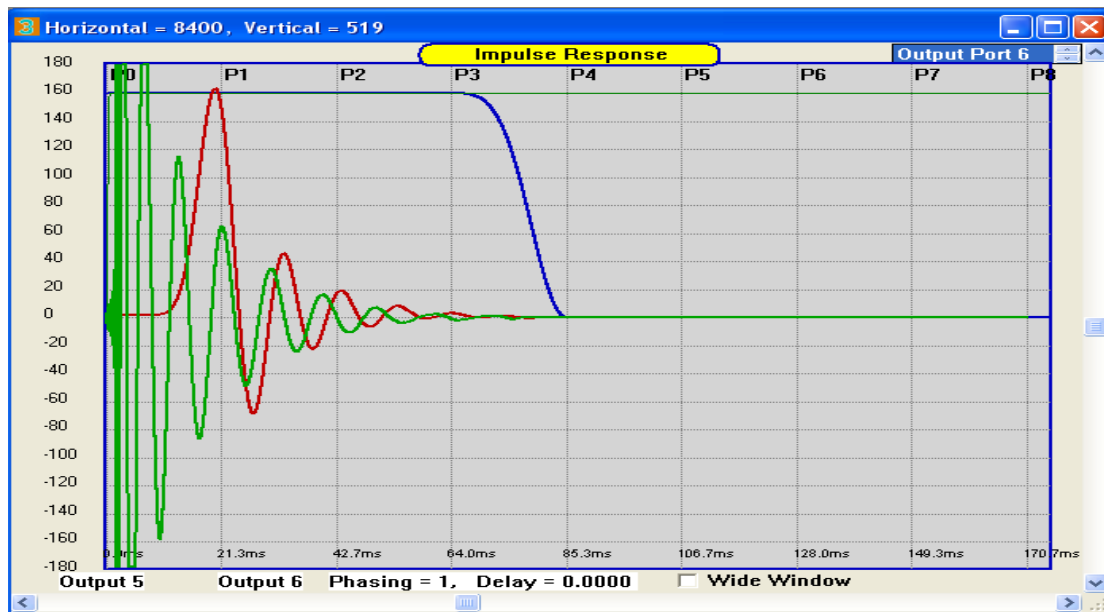

Figure 21. Impulse responses – Low-Pass (red) and High-Pass (green) of a MP version of the crossover.

Using Bi-Polar pulse for testing

When a 2ms bi-polar pulse was used for excitation for our 20Hz, +24db/oct subwoofer, the MP version (green) has done the opposite, and converted the symmetrical bi-polar pulse into a pulse with clear asymmetrical tendency, while the LP (red) version has returned a response just about perfect excitation pulse – see Figure 22 below.

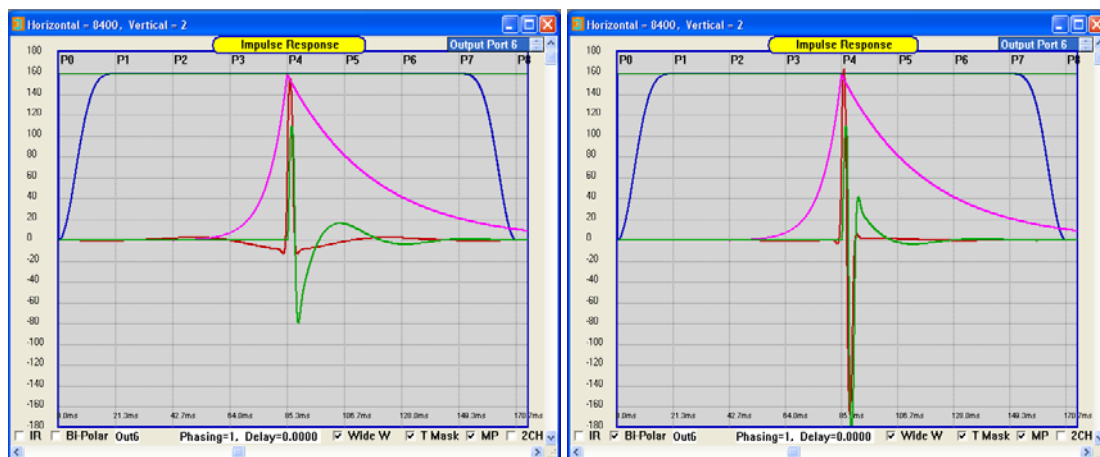Figure 22. Subwoofer response to pulse   and bi-polar pulse (lin-lin scale).

One very important issue is observable on the Figure 22 above. The bi-polar pulse used for this simulation is much closer to real-live audio signals than a one-sided, 2ms pulse (impulse). **Interestingly, when compared to pulse response, we can conclude, that left-side and right-side of the LP bi-polar pulse response is**

almost flat, and has lost it's pre-tilt and post-tilt, clearly visible on left-hand side of Figure 22.

I have actually confirmed this issue in real measurements – see Figure 23. The ringing past the pulse is due to a more distant microphone placement, so now, the mike picks some of the room reflections.

Figure 23. Bi-polar pulse in LP Mode   and   MP Mode

Please note perfect agreement between Figure 22 and the measured results above for both: MP and LP filters. I have used lin-lin scale, so that simulation model can be compared to direct CRO measurements.

However, the bi-polar nature of the pulse did not change the pre-ringing of the LP version (red) of the filter.
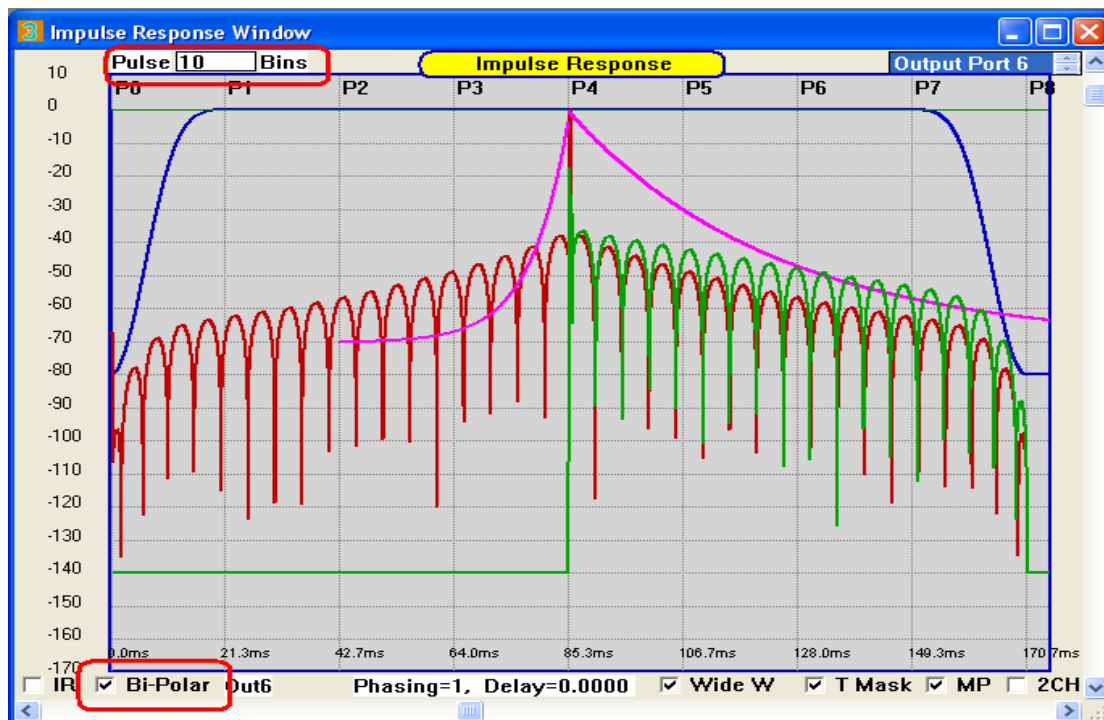

Figure 24. A 20dB peaking filter excited with a 10-sample wide bi-polar pulse.

Is Post-ringing audible at all?

During my listening tests and measurements on a large, 18" subwoofer using the 2ms pulse, the minimum-phase version delivered a more of a "thump" instead of a pop or a click. This is perhaps not surprising, as the post-ringing of the pulse extended to130ms, far exceeded the 30ms "memory effect" of the auditory system, and pushed beyond the 100ms falling edge of Temporal Masker. Here, the driver, filter and vented enclosure added it's own, combined signature – see Figure 25.

Figure 25. 5ms Impulse in Linear-Phase Mode   and   Minimum-Phase Mode

When a long, 10ms bi-polar pulse was used for excitation, the LP version still holds the characteristics of the excitation pulse very well, but the MP version had even more asymmetrical tendency.
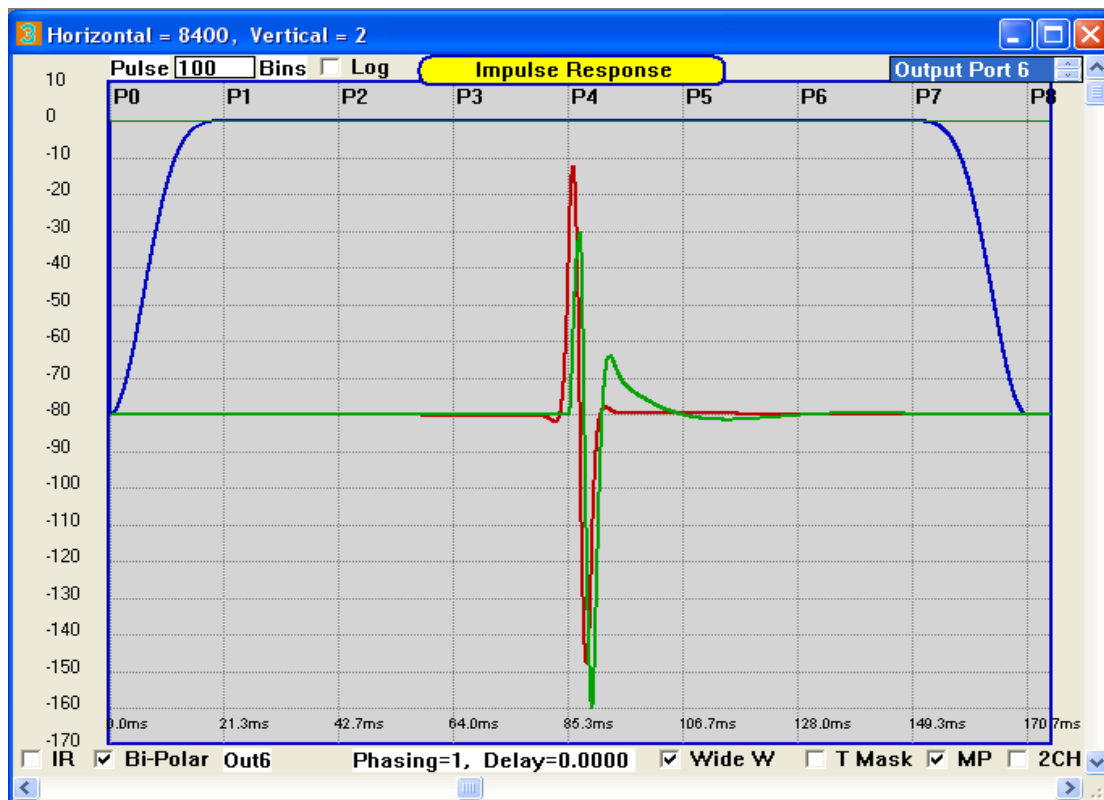

Figure 26. LP(red) and MP (green) responses to 10ms bipolar pulse (lin-lin scale).

Here is confirmation of the above in actual measurements. Once again, the ringing around the pulse is due to a more distant microphone placement, so now, the mike picks some of the room reflections.

Figure 27.10ms Bi-polar pulse in LP Mode   and   MP Mode

As before, an excellent agreement between simulated and measured responses is evident in Figures 26 and 27.

I was able to compare a subwoofer with no acoustical phase distortions (flat line phase response) to a subwoofer with minimum-phase phase characteristics (typical phase roll-off for driver + crossover systems). There is a lack of internet literature describing this exact type of tests performed on subwoofers. The only paper I was able to source, that used phase-equalized loudspeakers (but not subwoofers) was *The Audibility of Loudspeaker Phase Distortion* Preprint 2927, by Mr Richard Greenfield, Dr Malcolm Hawksford, Department of Electronic Systems Engineering, University of Essex, Colchester, England.

Conclusions

Pre-and post ringing often come into focus in perceptual coding algorithms, as explained in http://en.wikipedia.org/wiki/Pre-echo . How much does it relate to loudspeakers and crossovers?.

The IR simulations presented above are my attempt to scope the pre-ringing phenomenon, as I was interested in comments coming from recording industry about audibility of linear-phase EQ pre-ringing. Upon closer research, it was almost inevitably discovered, that the phenomenon was audible under pretty extreme EQ settings. So, this little write-up is my crude attempt to shed some light at what may be contributing to the perceived EQ pre-ringing.

It is acknowledged, that Temporal Masking is the mechanism responsible for what we hear in the presence of a masking signal. Some sources available on the internet will indicate, that pre-masking extends to 20ms. Like the curve on Figure 1. http://zone.ni.com/reference/en-XX/help/373398B-01/svaconcepts/svtimemask/ , and other http://18.7.29.232/bitstream/handle/1721.1/36465/31317829.pdf?sequence=1 will conclude, that it is most effective from 5ms. "Introduction to Digital Audio Coding

and Standards" By Marina Bosi, Richard E. Goldberg, also indicates 20ms as the typical pre-masking interval.

In the light of the information presented so far, the following conclusions, applicable to loudspeakers and crossovers can be drawn:

Conclusion 1.
Linear-Phase filters have **symmetrical coefficients** on both sides of the Impulse Response. Evident on all relevant Figures.

Conclusion 2
Typical crossover implementation: Low-Pass and High-Pass components will cancel each other's pre-ringing, **if the filters are of the same type and slope**. This is illustrated on Figure 20.

Conclusion 3
All pre-ringing artefacts, that fall into the interval of 20ms prior to the onset of the masker (and most effectively from 5ms) will not be audible due to pre-masking effect. Examples are illustrated on Figure 14 and 17.

Conclusion 4
Pre-ringing is most evident (and audible) while using peaking filters with high gain and high Q-factor, as illustrated on Figure 8 and Figure 24, and suggested by the recording industry. These should be avoided. On the other hand, notch filters are OK to use (as in RoomEQ), as illustrated on Figure 11 and 12.

Conclusion 5
Typical sounds do not have purely one-sided, pulsing nature. They tend to have more of a bi-polar attack transient envelope. In this department, LP subwoofer (or system) will outperform MP subwoofer, as evident from Figure 22 and confirmed in actual measurements on Figure 23. Also Figures 26 and 27.

Listening tests conducted on phase-linear loudspeaker (AES Preprint 2927) by Mr Richard Greenfield, Dr Malcolm Hawksford and several other expert listeners also did not reveal any pre-response issues. Example of similar comments are here:
http://tech.groups.yahoo.com/group/SoundEasyuserslist/message/4965

So, the overall conclusion from the above simulation is that, the possible audibility of the pre-ringing artefacts of Linear-Phase approach to loudspeaker design does not seem to be an issue with typical loudspeaker or crossover filter. I have been using my THX-level, Linear-Phase 5.2HT audio system for over 6 months, listening to a variety of CD/server music and watching concerts/movies in Blu-ray format.
http://www.bodziosoftware.com.au/Home_Theatre_Conclusions.pdf

No problems were observed (or heard).

Thank you for reading,

Bohdan