SPL and Pre-Ringing Comparison Within 0-60deg polar radiation

By Bohdan Raczynski

This paper attempts to quantify off-axis performance of a **DSP equalized loudspeakers**. The loudspeakers system used for the measurements are a very simple 2-way design, with a 1"/50W metal dome tweeter and 8"/90W woofers mounted on a flat baffle. Frequency response of the system was aimed at 45Hz-20000Hz, but with a DSP equalization was extended to 45Hz-30000Hz.

Figure 1. W-220P woofer and D25AG tweeter used in this project

Woofer's data

Woofer	W-220PII
SPECIFICATIONS	
Size	8
Nominal Diameter	200
RMS Power	100 W
Maximal Power	200 W
Rated Impedance	8 Ω
Sensitivity	89 dB
Frequency Range	fo-5k Hz
Basket Material	steel
Gasket Material	
Cone Material	coating with PP paper
Surround Material	rubber
Magnet Material	ferrite
Magnet Weight/Overall size	33 oz/φ126
Voice Coil Diameter	1.53" (38.8)
Former Material	aluminium
Wire	copper
Layers	two
Spider Material	cotton
Terminals Type	plug
Shorting Ring	
Magnet Structure	
Highest Recommend Crossover	≤2.5k

PARAMETERS	
Re	6 Ohm
F0	28 Hz
Qms	1.58
Qes	0.43
Qts	0.33
Zmax	27 Ohm
Sd	0.022698 m^2
Mms	26.95 g
Mmd	24.98 g
BL	8.19 T m
Xmax	6.8 mm
Cms	1187u M/N
Vas	86.87 L
SIMULATION DESIGN	
Cabinet Type	vented enclosure
Recommended Enclosure Volume	30 L
Recommended PORT of Cabinet Dimension	φ65×200
fb	37 Hz

Tweeter's data

NOMINAL IMPEDANCE	6	Ω
NOMINAL POWER (IEC 268–5)	100	W
FREQUENCY RANGE	1,5–35	kHz
SENSITIVITY (1W, 1m)	89	dB
EFFECTIVE DIAPHRAGM AREA	7,1	cm²
VOICE COIL RESISTANCE	4,6	Ω
OPERATING POWER	5	W
VOICE COIL DIAMETER	25	mm
VOICE COIL HEIGHT	1,6	mm
AIR GAP HEIGHT	2	mm
FREE AIR RESONANCE	850	Hz
MOVING MASS (incl. air)	0,3	g
FORCE FACTOR, B x l	3,3	Txm
MAGNET WEIGHT (8,5 oz)	240	g

Vifa DG25AG metal dome tweeter treatment

The DG25AG tweeter is equipped with a phase plug, designed to flatten the high-end of the frequency response. This approach works reasonably well, but there has been some criticism of "tizzy sound" attributed to this driver.

Figure 2. Vifa tweeter with the phase plug (left) and without phase plug (right).

Also, the off-axis SPL curves roll-off more smoothly without the phase plug. It has been decided to remove the phase plug from the front of the metal dome. All subsequent measurements have been conducted on the tweeter with set of figures as presented on the right.

Measurement equipment

1. All measurements were conducted using SoundEasy V18, with MLS system running at 96kHz sampling frequency.
2. DSP processor was Ultimate Equalizer V5 (UE5), also running at 96kHz.
3. Test power amplifier - LM3876, a simple 50Watt integrated design from National Semiconductor, originally had 3dB cut off at 16Hz. Amplifier was modified to lower the 3dB down to 2Hz.
4. Microphone pre-amplifier – A commercial design based on low-noise, LM833 chip. Modifications done to equalize microphone's low-end roll off.
5. Microphone – CLIO Mic01. 8.2V DC bias provided by the pre-amplifier.

6. Listening room has the following dimensions: Length = 6.5meters, width = 4.5meters and Hight = 2.6meters.

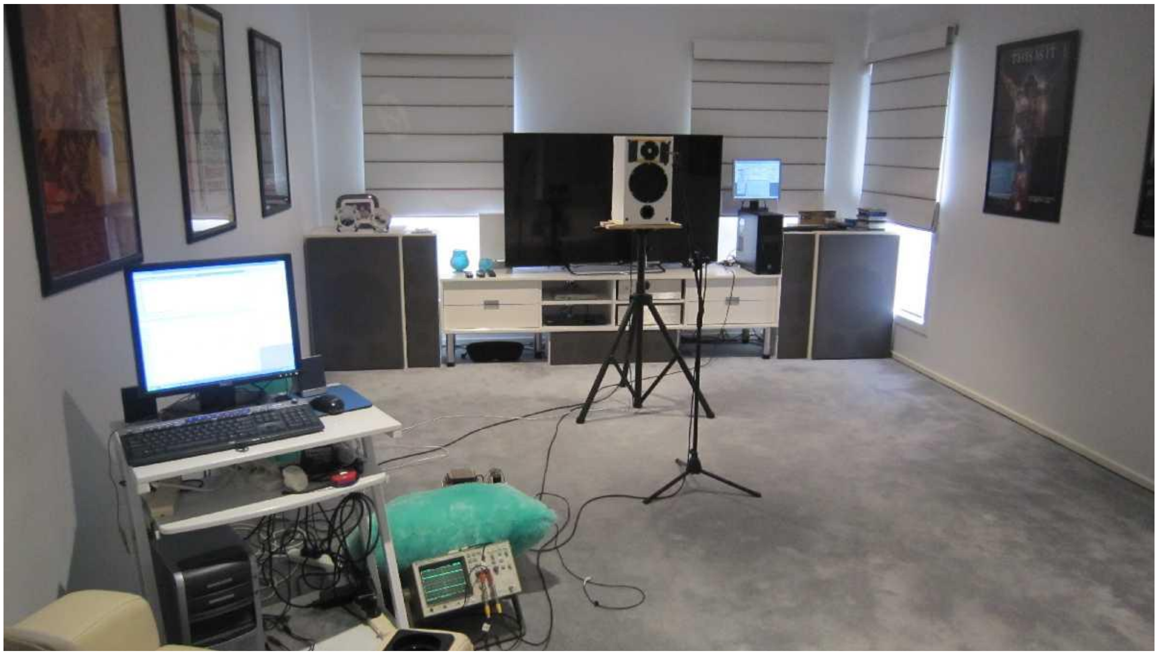


Figure 3. Overall measurement setup. Second PC (in the distance) runs UE5.

Figure 4. Close-up of the small 2-way loudspeaker.
Notice 0,15,30,45 and 60degrees direction pointer sticks.

Polar Measurement Results 200Hz-30000Hz

All curves presented below compare **residual SPL irregularities** of a system with 2kHz/24dB LR crossover with Linear-Phase and HBT equalization, against simple 2kHz/24dB, LR crossover. HBT upper limit set on UE5 was 30kHz.

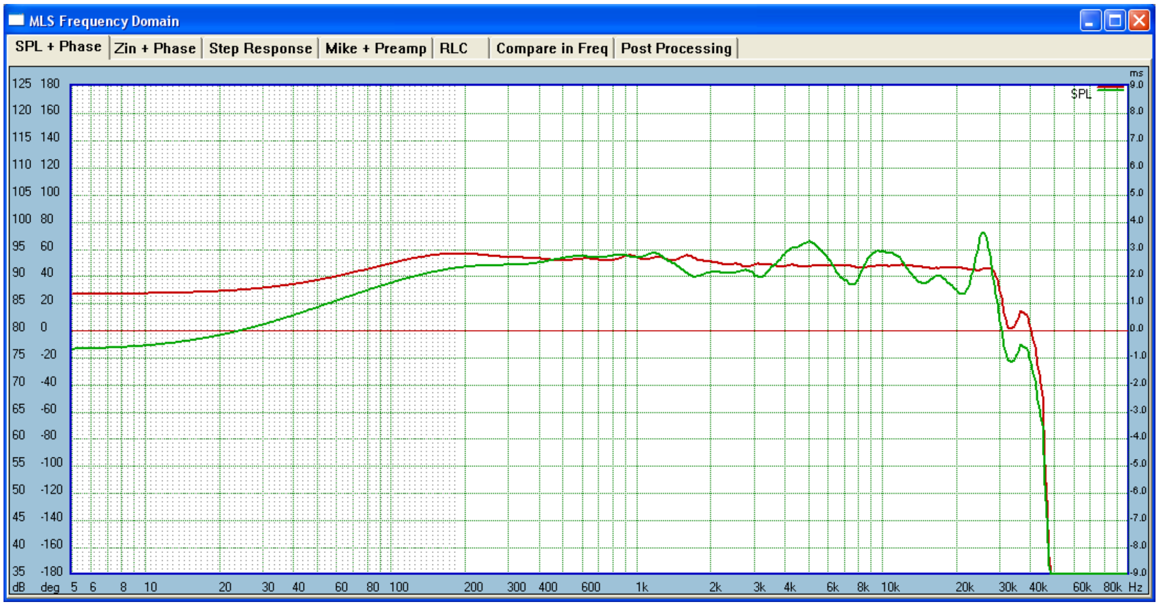


Figure 5. Mike distance = 50cm, HBT+LinPh vs. crossover only – **0deg**
Red = Linear_Phase + HBT EQ = 94dB-89dB = 5dB
Green = Minimum_Phase, No EQ = 98dB-80dB = 18dB

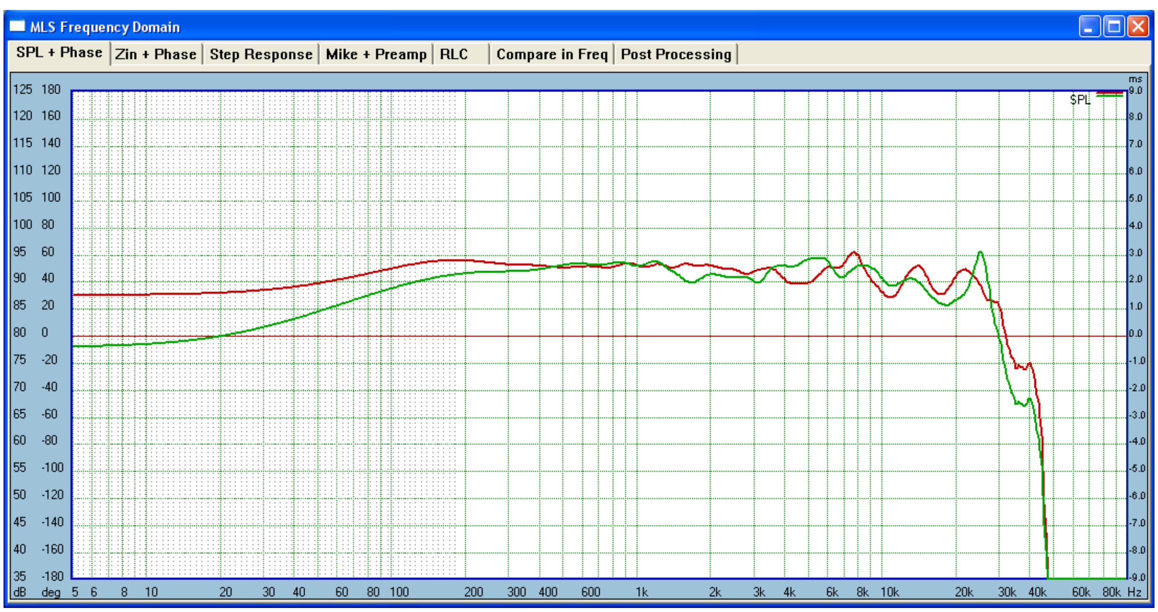


Figure 6. Mike distance = 50cm, HBT+LinPh vs. crossover only – **15deg**
Red = Linear_Phase + HBT EQ = 96dB-86dB = 10dB
Green = Minimum_Phase, No EQ = 96dB-78dB = 18dB

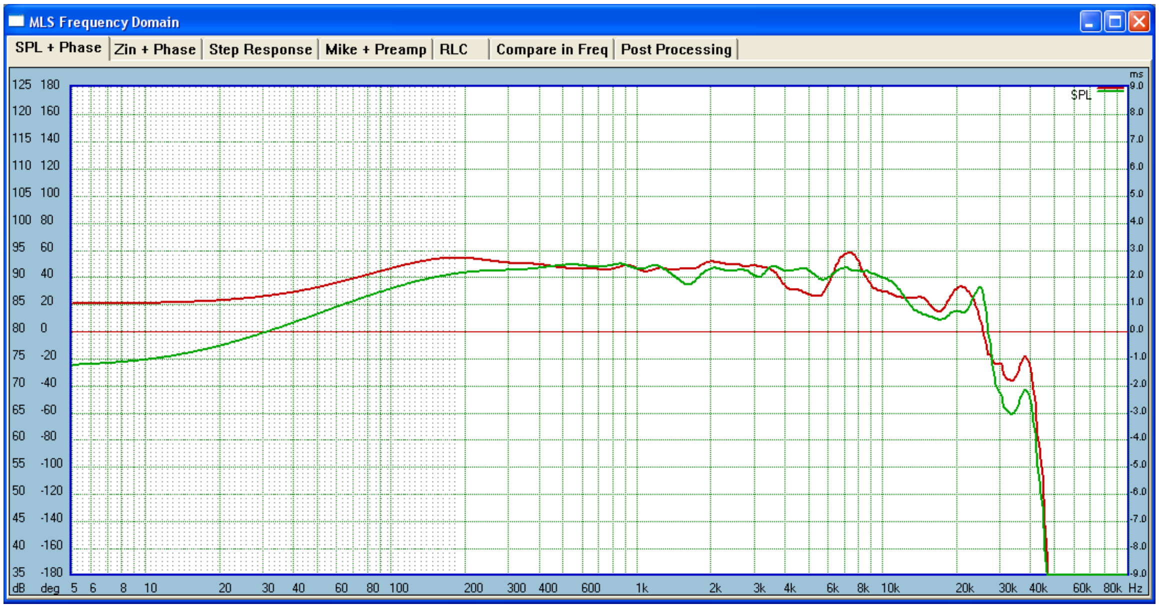


Figure 7. Mike distance = 50cm, HBT+LinPh vs. crossover only – **30deg**

Red = Linear_Phase + HBT EQ = 95dB-74dB = 21dB
Green = Minimum_Phase No EQ = 92dB-68dB = 24dB

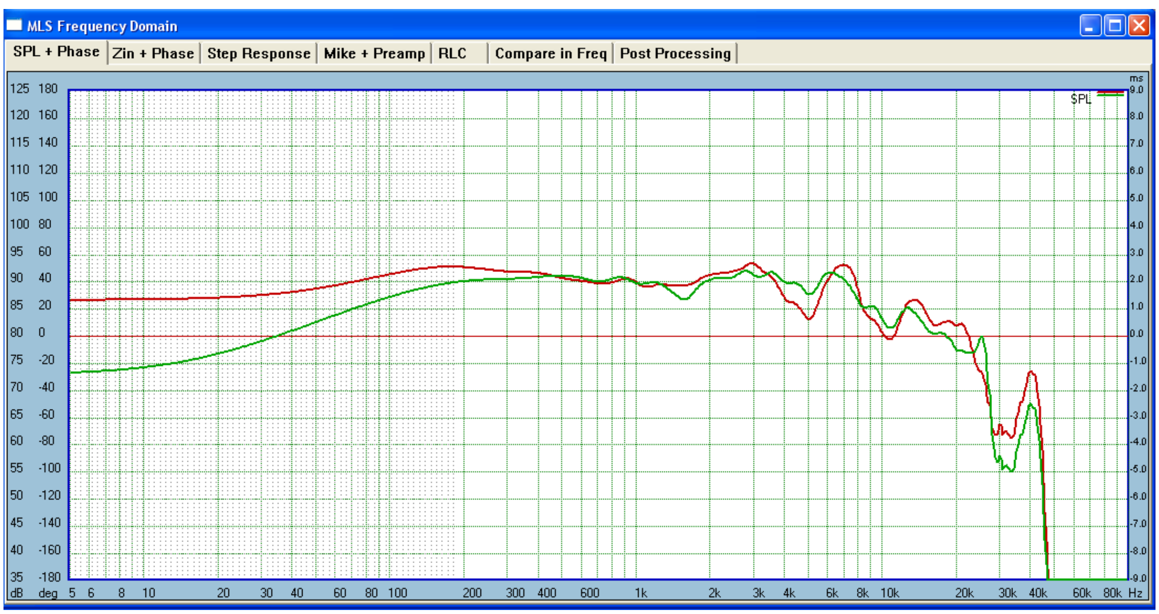


Figure 8. Mike distance = 50cm, HBT+LinPh vs. crossover only – **45deg**

Red = Linear_Phase + HBT EQ = 93dB-62dB = 31dB
Green = Minimum_Phase No EQ = 92dB-55dB = 37dB

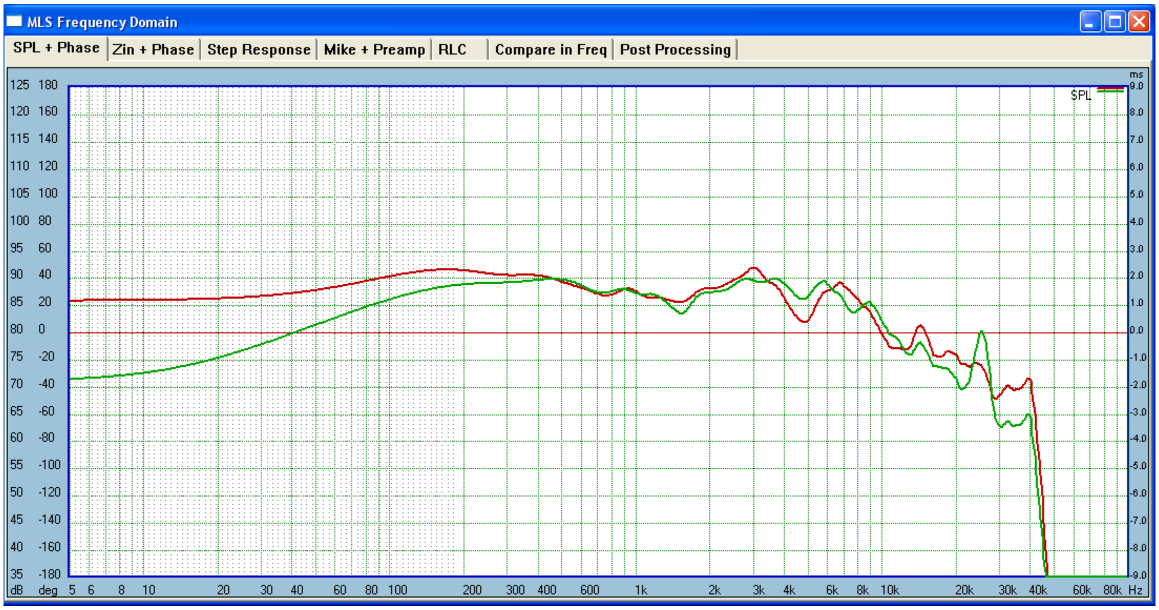


Figure 9. Mike distance = 50cm, HBT+LinPh vs. crossover only – **60deg**

Red = Linear_Phase + HBT EQ = 92dB-68dB = 24dB
Green = Minimum_Phase, No EQ = 90dB-62dB = 28dB

Two-way WTW configuration loudspeaker for Center Channel in HT

The next loudspeaker is a typical 2-way system in WTW configuration. I use this loudspeaker as center channel for the HT system. Measurement setup is shown below.

Conceptually, the front loudspeaker is a 2-way, vented system with two 8"/90Wrms woofers and 1.125"/100Wrms silk dome Dayton shown below (selected over Vifa tweeter), and making it quite a robust, medium-sized loudspeaker.

Figure 10. Overall measurement setup. Second PC (in the distance) runs UE5.

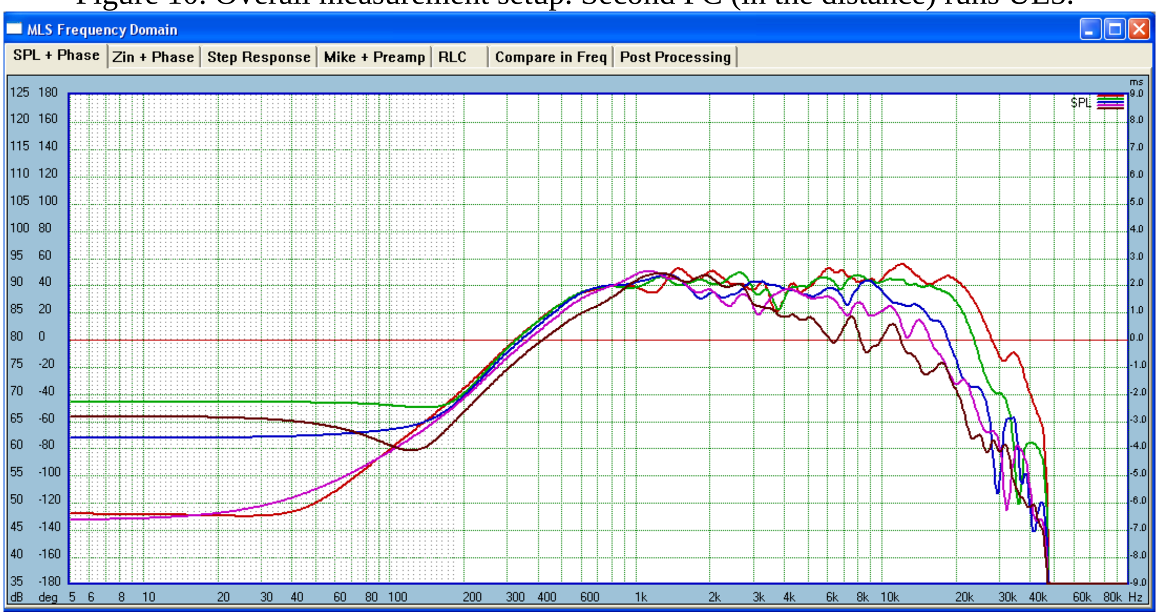
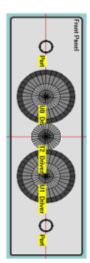


Figure 11. Raw RS28F-4 tweeter measurements (horizontal) 0, 15, 30, 45, 60 deg.

Vertical Orientation



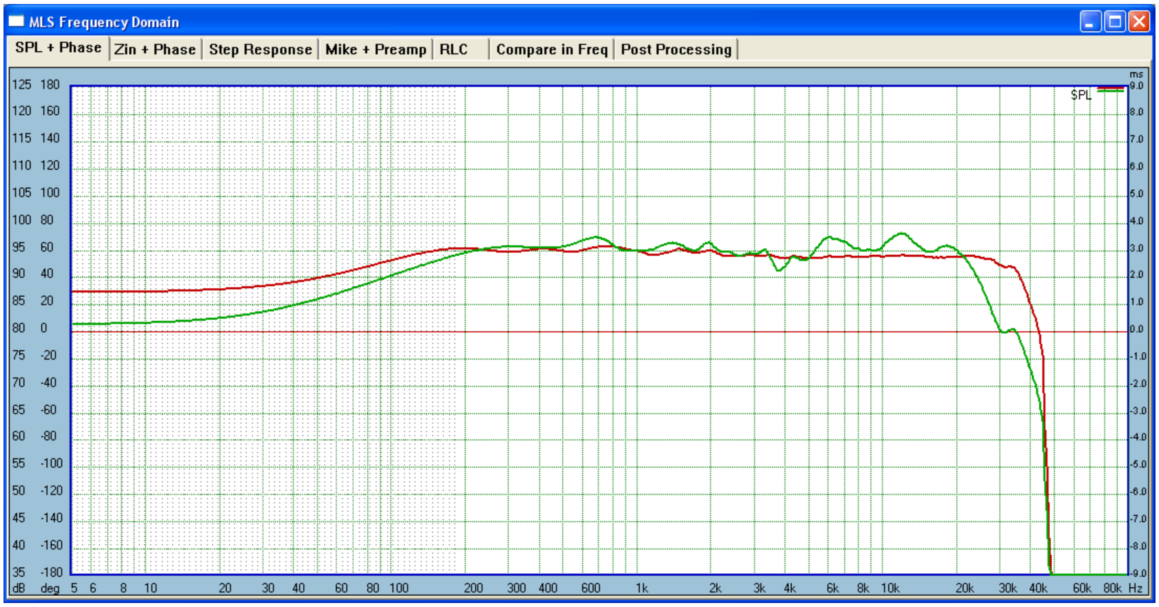


Figure 12. Mike distance = 100cm, HBT+LinPh vs. crossover only – **0deg**

Red = Linear_Phase + HBT EQ = 96dB-92dB = 4dB
Green = Minimum_Phase, No EQ = 98dB-80dB = 18dB

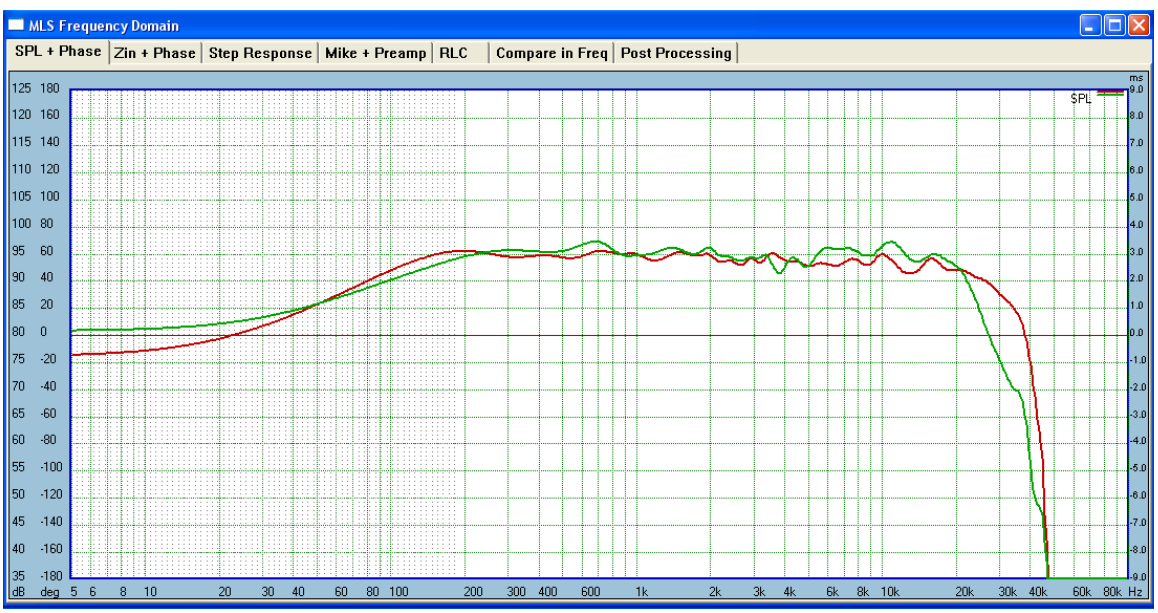


Figure 13. Mike distance = 100cm, HBT+LinPh vs. crossover only – **15deg**

Red = Linear_Phase + HBT EQ = 96dB-88dB = 8dB
Green = Minimum_Phase, No EQ = 97dB-75dB = 22dB

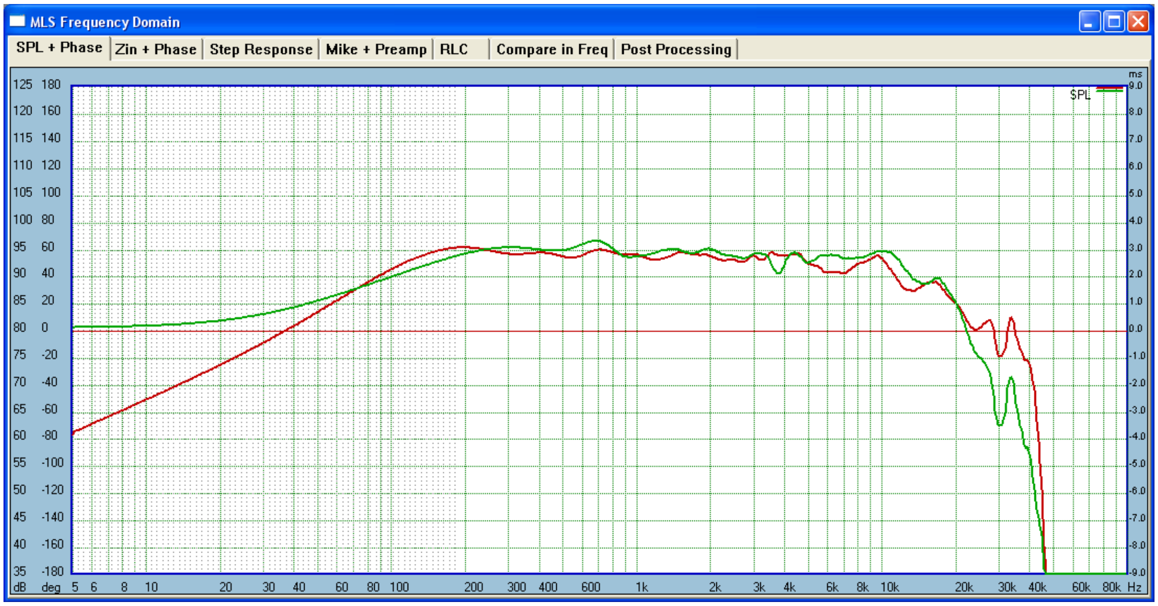


Figure 14. Mike distance = 100cm, HBT+LinPh vs. crossover only – **30deg**

Red = Linear_Phase + HBT EQ = 96dB-75dB = 21dB
Green = Minimum_Phase, No EQ = 97dB-63dB = 34dB

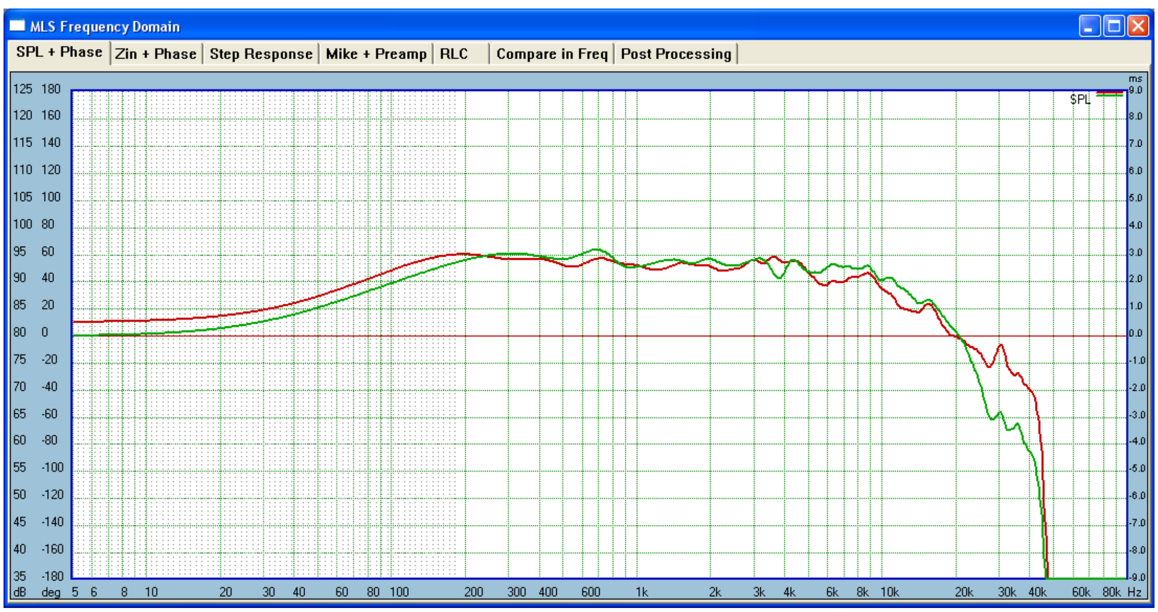


Figure 15. Mike distance = 100cm, HBT+LinPh vs. crossover only – **45deg**

Red = Linear_Phase + HBT EQ = 95dB-74dB = 21dB
Green = Minimum_Phase, No EQ = 96dB-65dB = 31dB

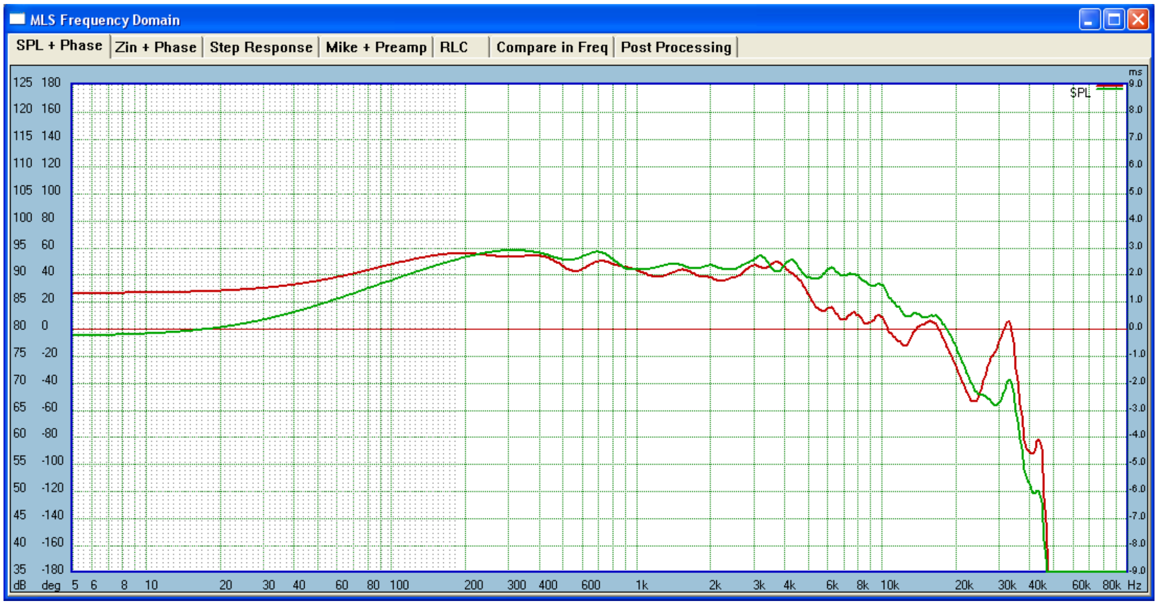
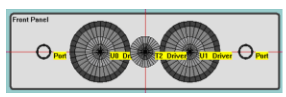


Figure 16. Mike distance = 100cm, HBT+LinPh vs. crossover only – **60deg**

Red = Linear_Phase + HBT EQ = 94dB-67dB = 27dB

Green = Minimum_Phase, No EQ = 95dB-67dB = 28dB

Horizontal Orientation



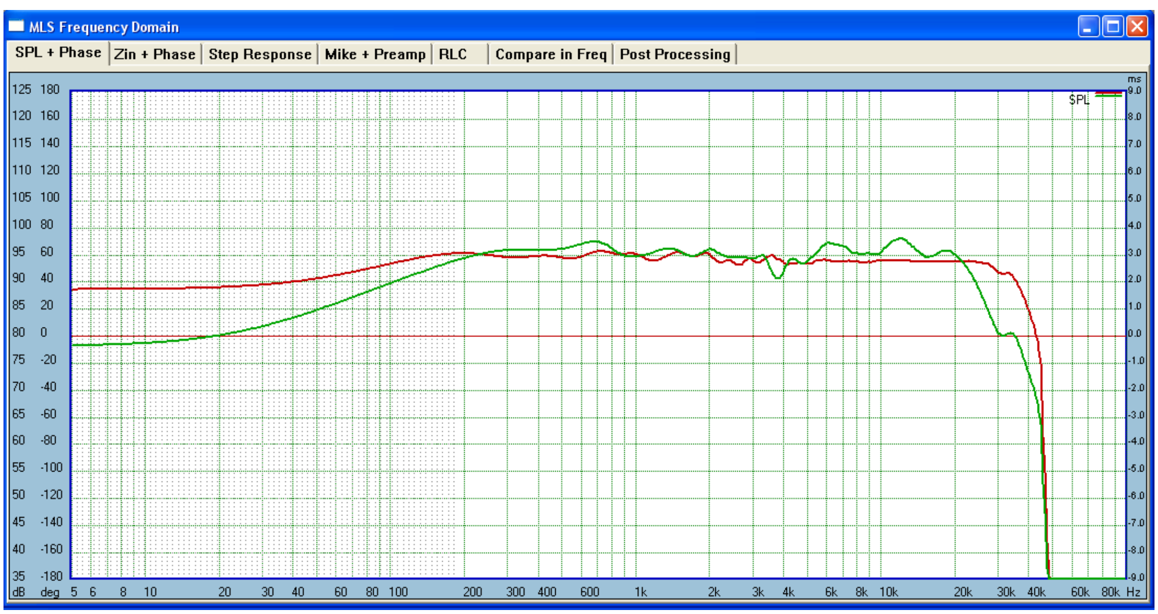


Figure 17. Mike distance = 100cm, HBT+LinPh vs. crossover only – **0deg**

Red = Linear_Phase + HBT EQ = 96dB-93dB = 3dB

Green = Minimum_Phase, No EQ = 98dB-72dB = 26dB

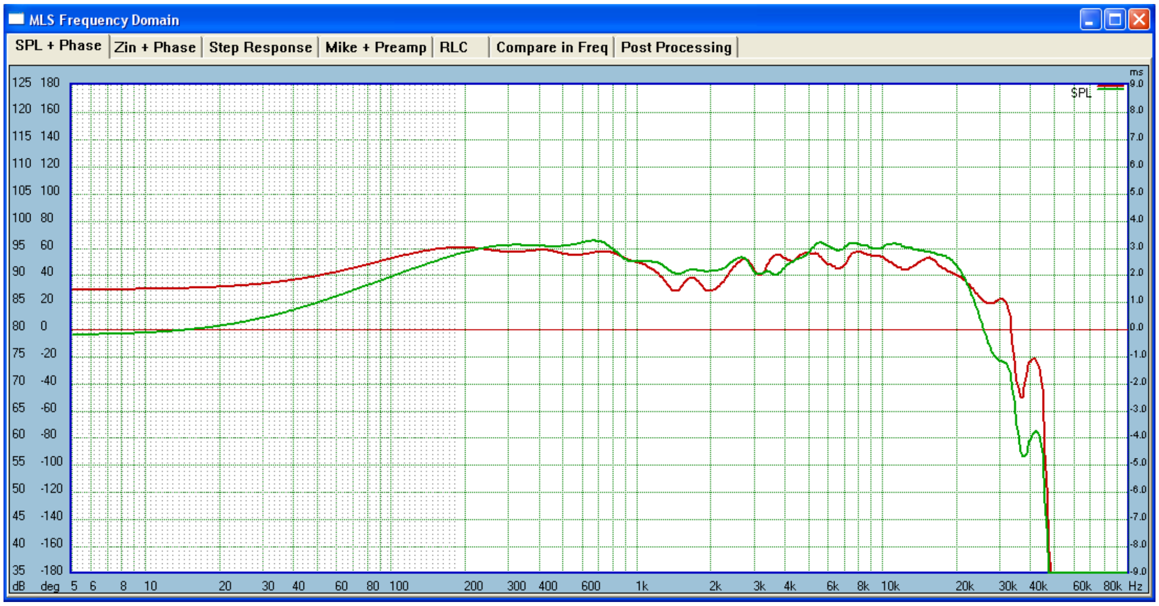


Figure 18. Mike distance = 100cm, HBT+LinPh vs. crossover only – **15deg**

Red = Linear_Phase + HBT EQ = 96dB-85dB = 11dB
Green = Minimum_Phase, No EQ = 97dB-74dB = 23dB

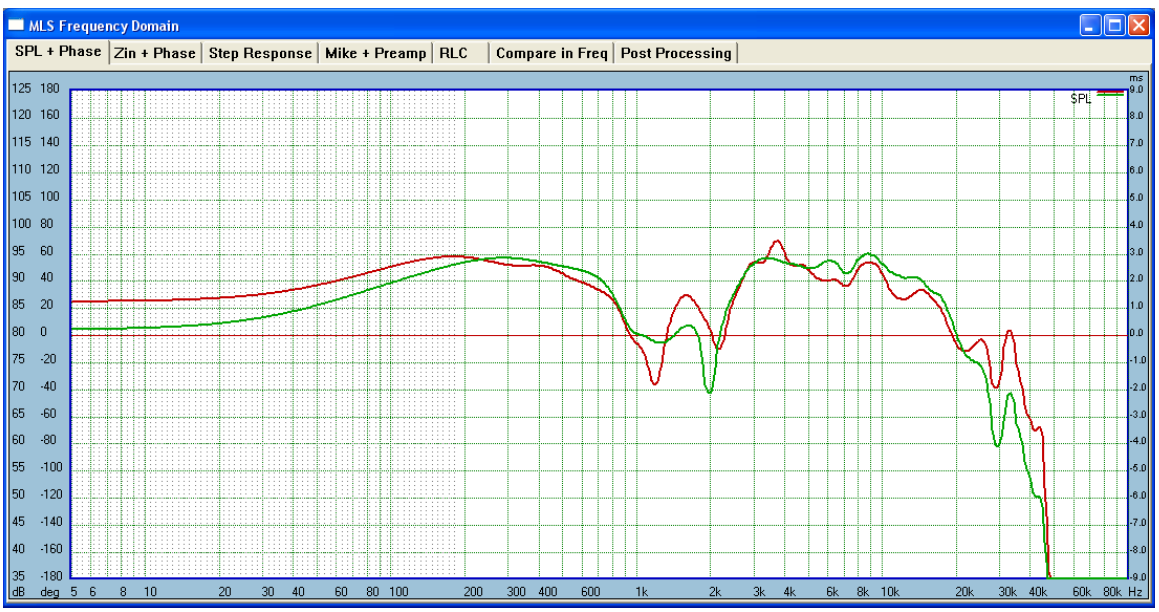


Figure 19. Mike distance = 100cm, HBT+LinPh vs. crossover only – **30deg**

Red = Linear_Phase + HBT EQ = 97dB-70dB = 27dB
Green = Minimum_Phase, No EQ = 95dB-59dB = 36dB

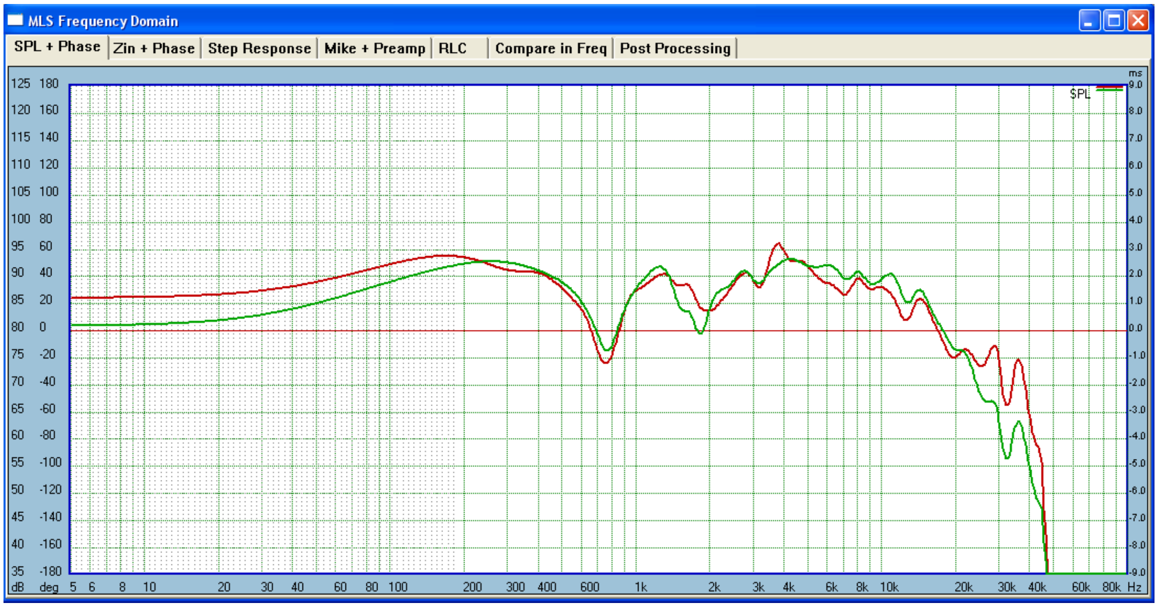


Figure 20. Mike distance = 100cm, HBT+LinPh vs. crossover only – **45deg**

Red = Linear_Phase + HBT EQ = 96dB-72dB = 24dB
Green = Minimum_Phase, No EQ = 92dB-65dB = 27dB

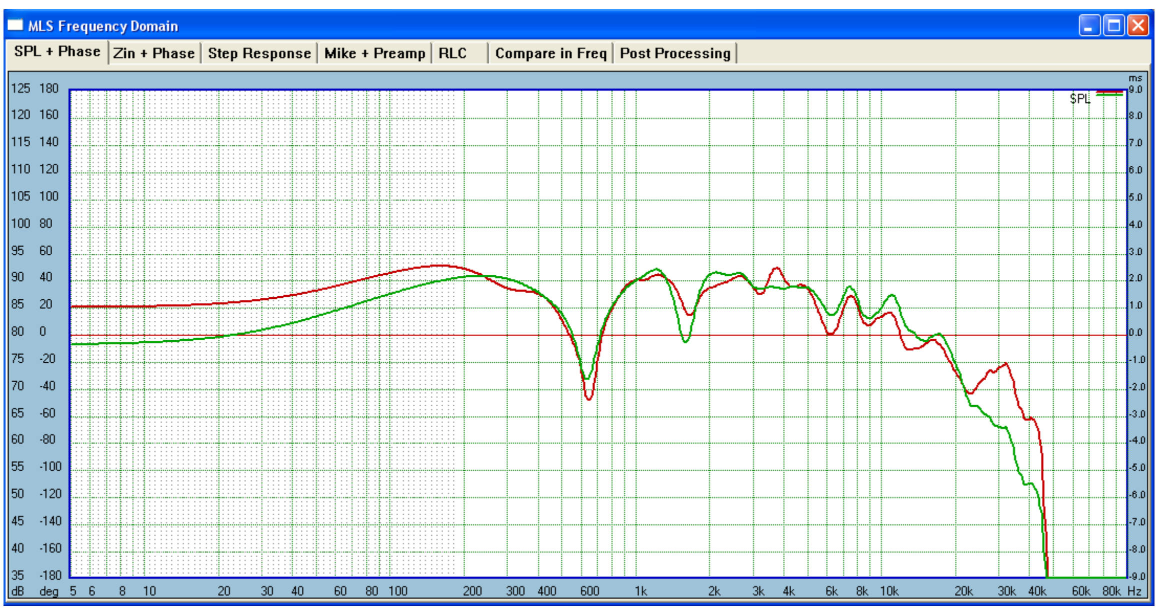


Figure 21. Mike distance = 100cm, HBT+LinPh vs. crossover only – **60deg**

Red = Linear_Phase + HBT EQ = 92dB-68dB = 24dB
Green = Minimum_Phase, No EQ = 92dB-63dB = 29dB

EQ Degradation Due to Distance

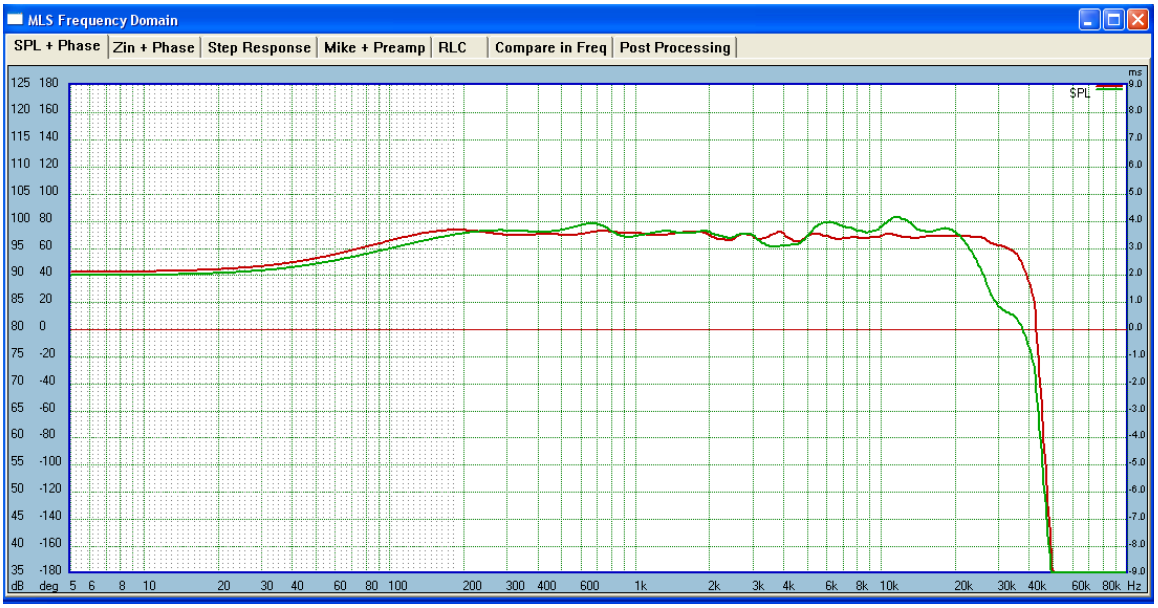


Figure 22. Mike distance = 51cm, HBT+LinPh vs. crossover only – **0deg**

Red = Linear_Phase + HBT EQ = 98dB-96dB = 2dB
Green = Minimum_Phase, No EQ = 101dB-84dB = 17dB

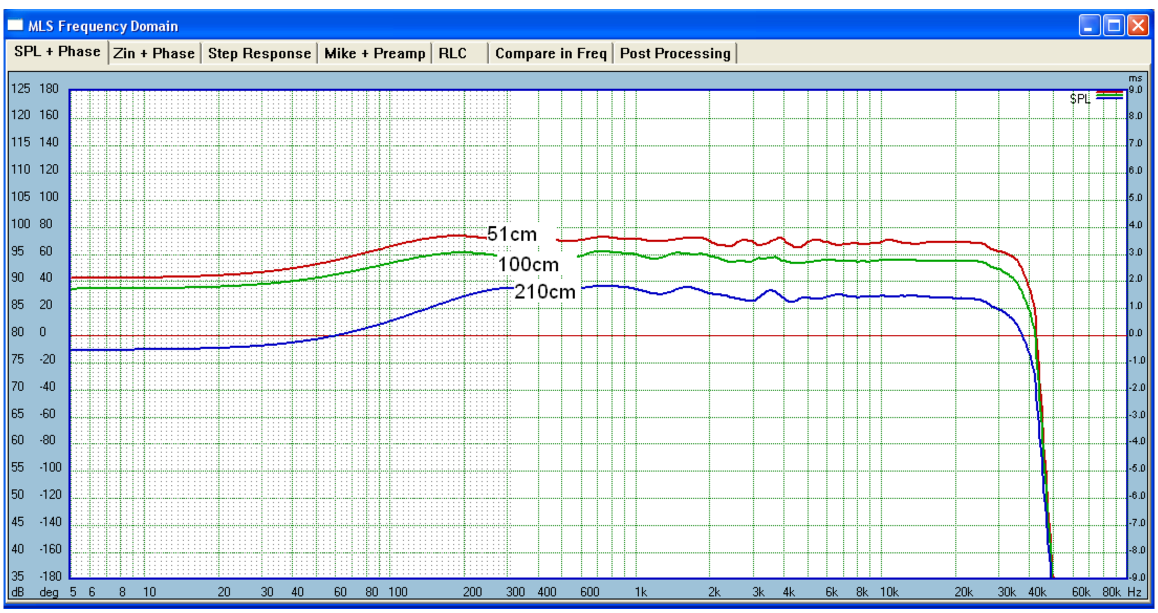


Figure 23. Mike distance = 51,100,210cm, HBT+LinPh – **0deg**

There is minimal degradation in SPL flatness due to distance. HBT was calculated for 100cm distance. This is why the green SPL curve is the flattest.

Conclusions

Loudspeakers under test were very "unsophisticated" loudspeaker systems. One with a metal dome tweeter and woofer mounted flat on the front baffle. Second one had a soft dome tweeter and two woofers. Simple 2-way crossover at 2kHz was used, and some attempt has been made to reduce diffraction – foam around tweeter driver. This is possibly the "worst-case" scenario, as a **controlled directivity driver system would be expected to perform much better in all tested scenarios**. SPL Measurements were conducted a **0deg, 15deg, 30deg, 45deg and 60deg off-axis** angles in horizontal plane. The WTW loudspeaker was measured in both planes and also evaluated at different distances on axis. It can be concluded, that:

1. SPL irregularities were smaller for the DSP-equalized loudspeaker, for all measured angles. Exceptional EQ performance is evident for on-axis measurement at 0deg – see Figure 5, 12, 23 and 24. Also the 15deg off-axis performance is significantly better with the equalized loudspeaker. For other angles, the EQ improvement in flatness is 1-4dB or better.
2. Off-axis, both loudspeakers exhibited bumps and valleys in their SPL curves. The WTW configuration measured in horizontal (as defined for HT system) orientation exhibited large degradation in SPL, regardless of EQ used or not. DSP-equalized SPL curves sometimes exhibit bumps and valleys located at different frequencies than non-equalized loudspeaker.
3. DSP equalization of loudspeaker by HBT equalization technique improves SPL linearity for off-axis angles, and provides near-perfect, flat-line SPL and phase for the "all-important", on-axis performance.
4. Equalized SPL curves exhibit good horizontal linearity below 3-4kHz, leading to the conclusion, that diffraction is the major factor contributing to linearity degradation at high frequency.

Amplitude and phase response of equalized loudspeaker extends to 30kHz, and on the low-end, is only limited by the necessity of FFT gating, to remove room reflections. See Figure 24 below.

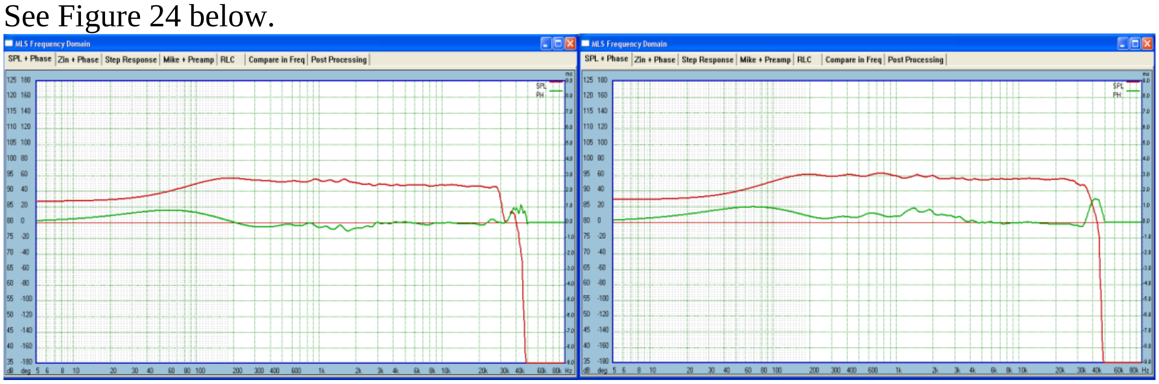


Figure 24. SPL (red) and Phase (green) of a 2-way TW (left) and WTW (right) UE5 equalized loudspeaker.

In summary: On-axis performance of a DSP-equalized loudspeaker is exceptional, and off-axis performance is still better than non-equalized driver.

Impulse Response Pre-ringing

Second issue often associated with linear-phase systems is pre-ringing of the impulse response. It can be shown, that pre-ringing of the impulse response measured on-axis will be marginal or not existent at all, as the low-pass and high-pass filters pre-ringing will cancel each other. However, the off-axis pre-ringing is sometimes being viewed as detrimental in FIR linear-phase filters. So, let's put this issue under the microscope.

Time (Temporal) Masking http://zone.ni.com/reference/en-XX/help/373398B-01/svaconcepts/svtimemask/

"….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.

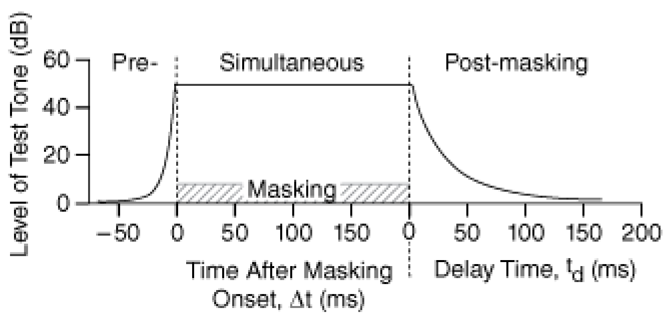
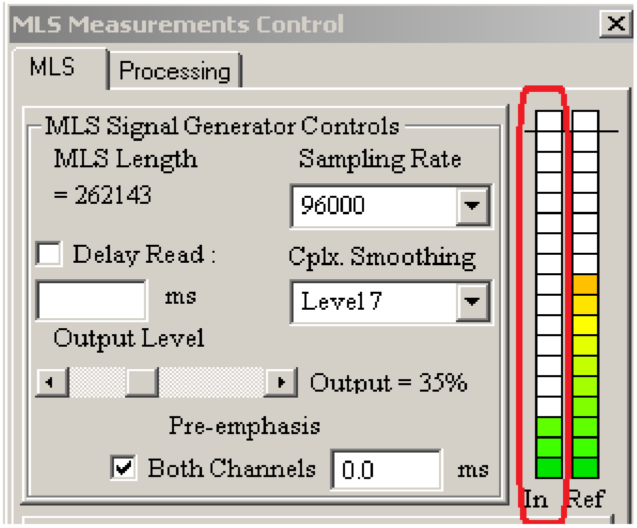


Figure 25. Slopes of temporal masking

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….".

A number of on-axis and off-axis measurements have been performed and the resulting impulse responses are presented below. The set of measurements was conducted at 1m distance, and the resulting sound level at the microphone was at quite low level (see figure below).



Normally, the lover received acoustic level would not present itself as a problem, because it can be raised to the required SPL level by adding several decibels to the resulting SPL response. However, it does affect dynamic range of impulse response presentation quite dramatically, and as a result, the dynamic range is limited to around 55-60dB by the background noise (yes, I have lost over 15dB of the dynamic range). Yet, the impulse response will be also presented in logarithmic scale (decibels), as it shows the pre-ringing decay as well.

For the sake of clarity, three figures are presented in this order:

1. Only the pre-ringing portion of the impulse response is shown, together with 20ms left-hand Blackman-Harris window. The green window clearly shows where the pre-masking effect (to the left) would cease to operate. This plot presents impulse response in linear scale.

2. The same plot transferred into Impulse Response Export dialogue and presented also in linear vertical scale. Now, the time scale is 42.7ms on the left-side of the impulse response, so the plot is twice compressed in time.

3. The same plot presented also in logarithmic vertical scale with pre- and post-ringing limits superimposed. You may expect the background noise quite visible on this plot (as discussed above). Particularly, for the 45/60deg off-axis measurements, where the total level of impulse response is further reduced and after calibration to 0dB, the noise is raised at the same time quite significantly. Time scale is 42.7ms on the left-side of the impulse response

Figure below confirms time scale between impulse response taken at 60deg off-axis and it’s plot transferred into Impulse Response Export dialogue and impulse response as measured in MLS measurement system The shape of the impulse response is identical in both instances, when plotted in the same screen resolution.

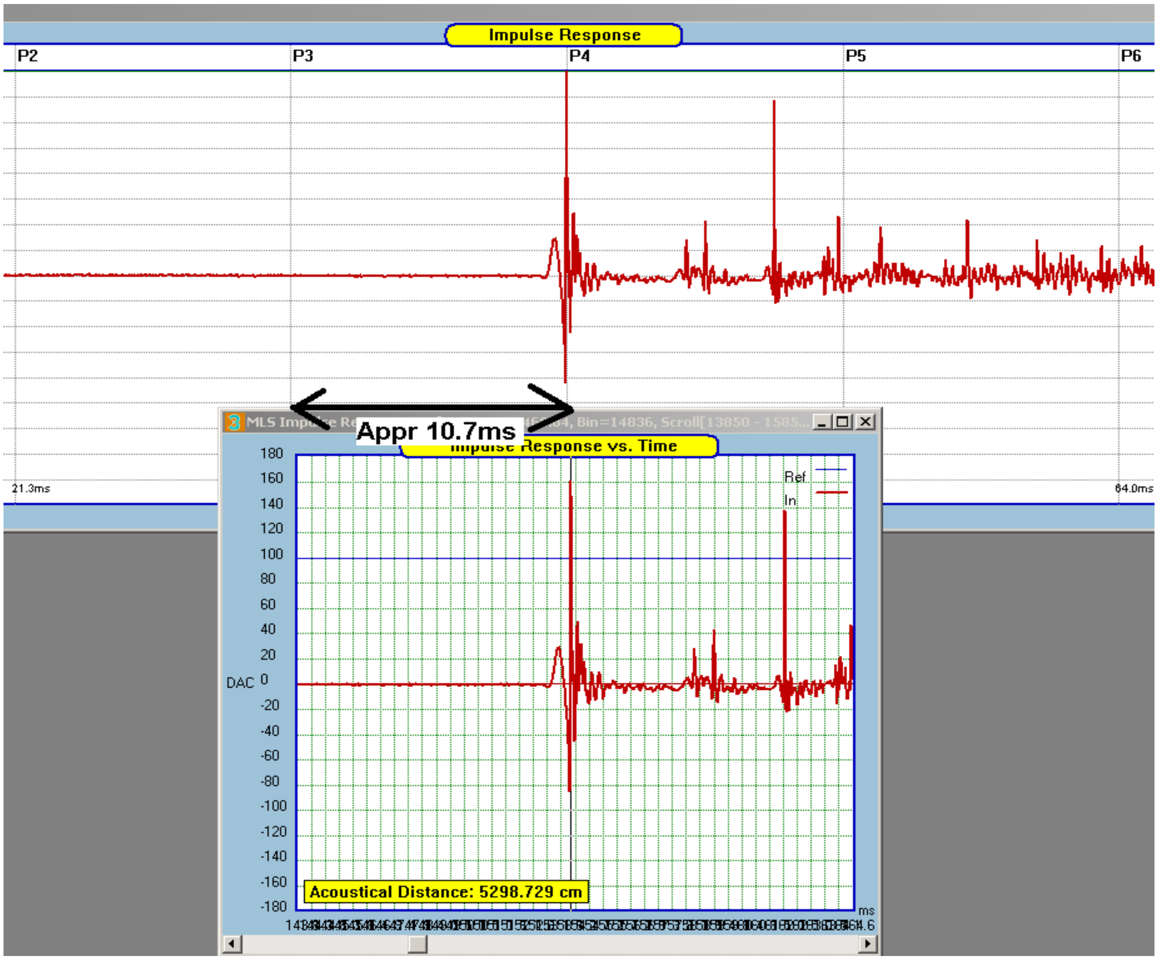
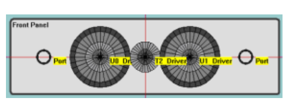


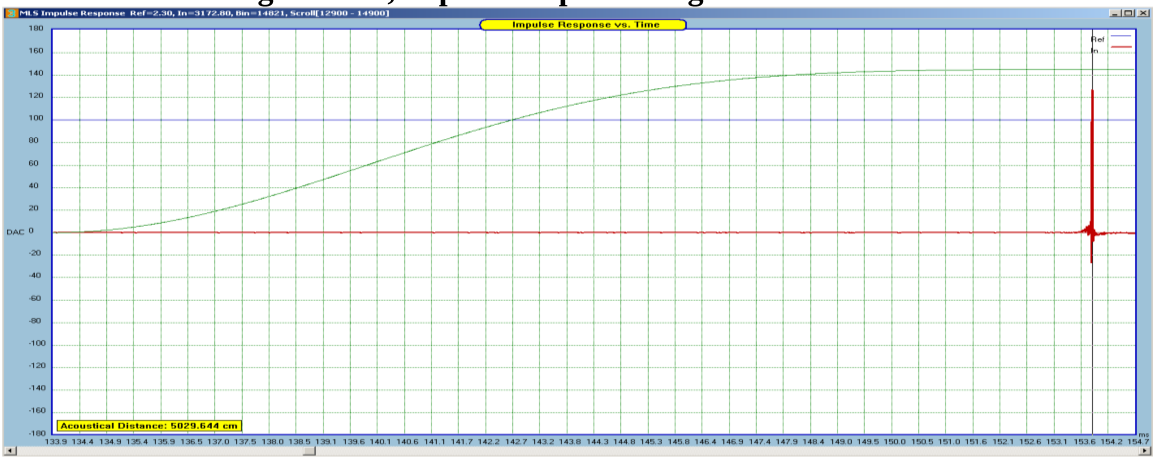
Figure 26. Time scale comparison in MLS system and Impulse Response Export dialogue

We can now proceed with the examination of impulse responses.

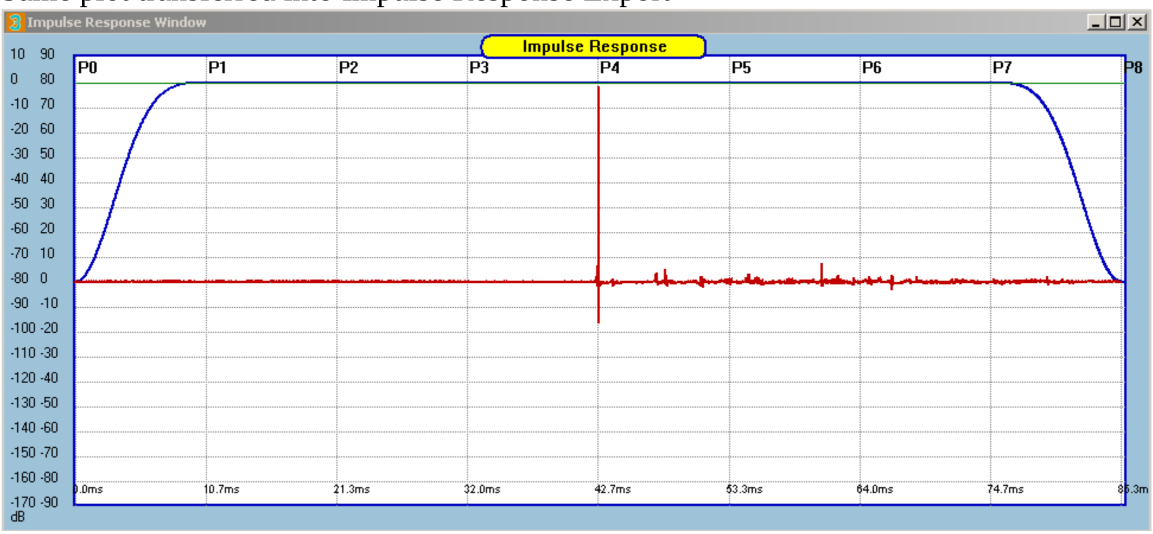
 **Vertical pattern**

The 0dB impulse response calibration was performed for the on-axis impulse response. After that, the gain can not be changed for other impulse responses measurements.

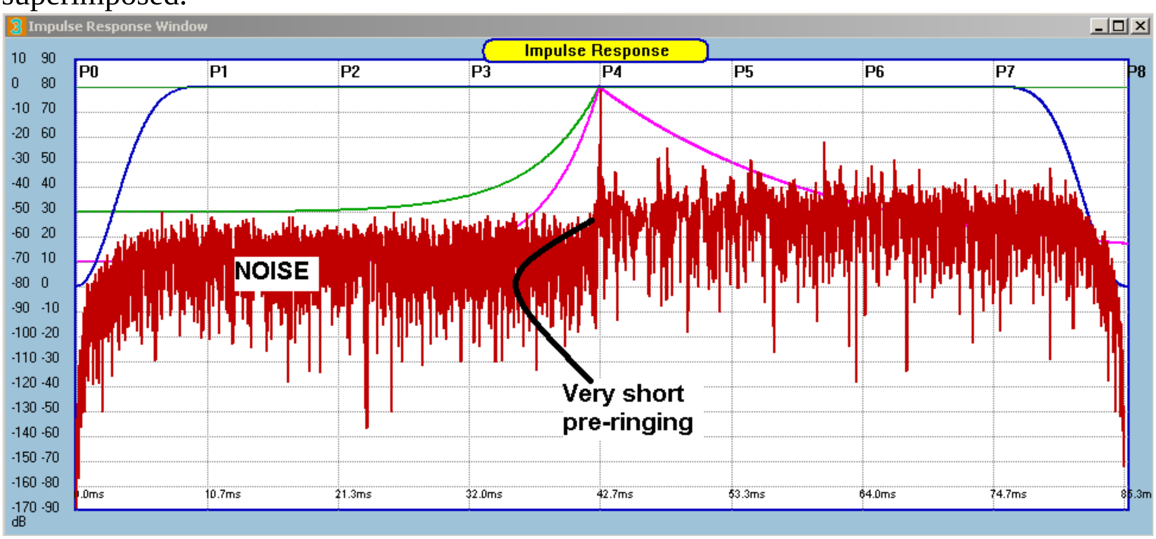
0deg on-axis, impulse response magnification = 1x



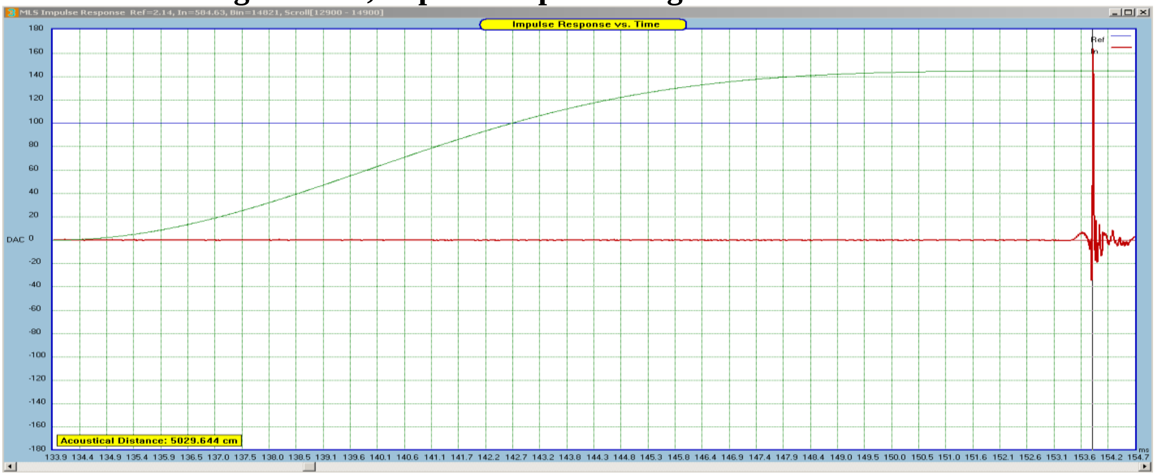
Same plot transferred into Impulse Response Export



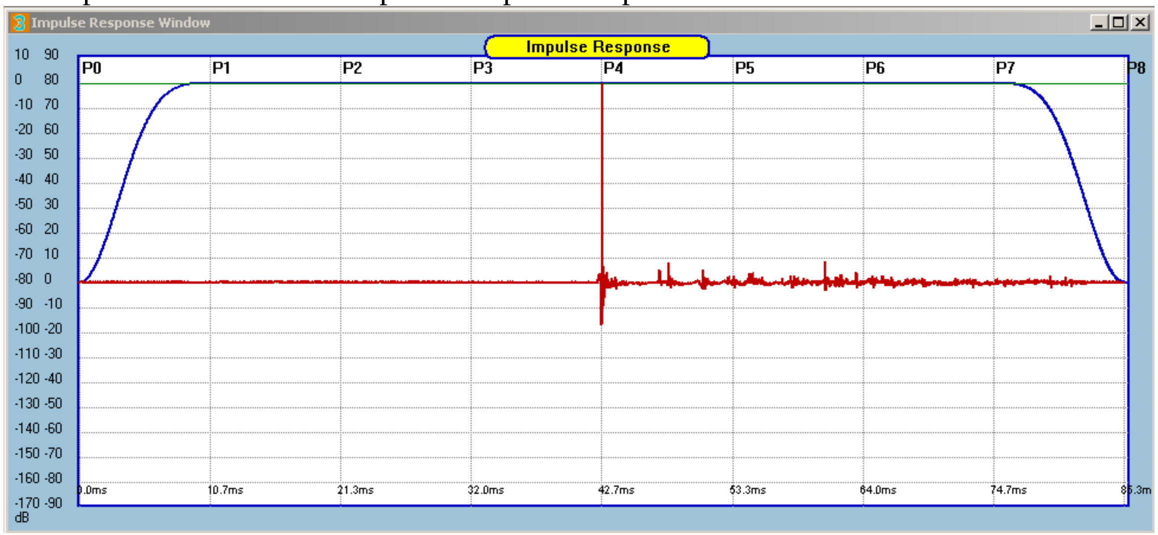
Same plot presented also in logarithmic vertical scale with pre- and post-ringing limits superimposed.



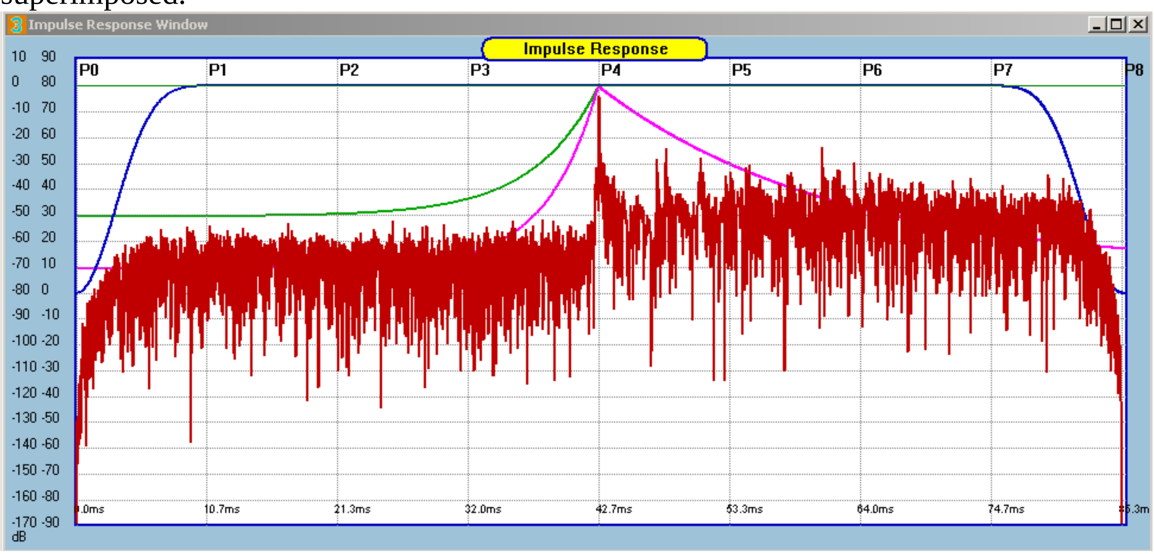
15deg off-axis, impulse response magnification = 2x



Same plot transferred into Impulse Response Export

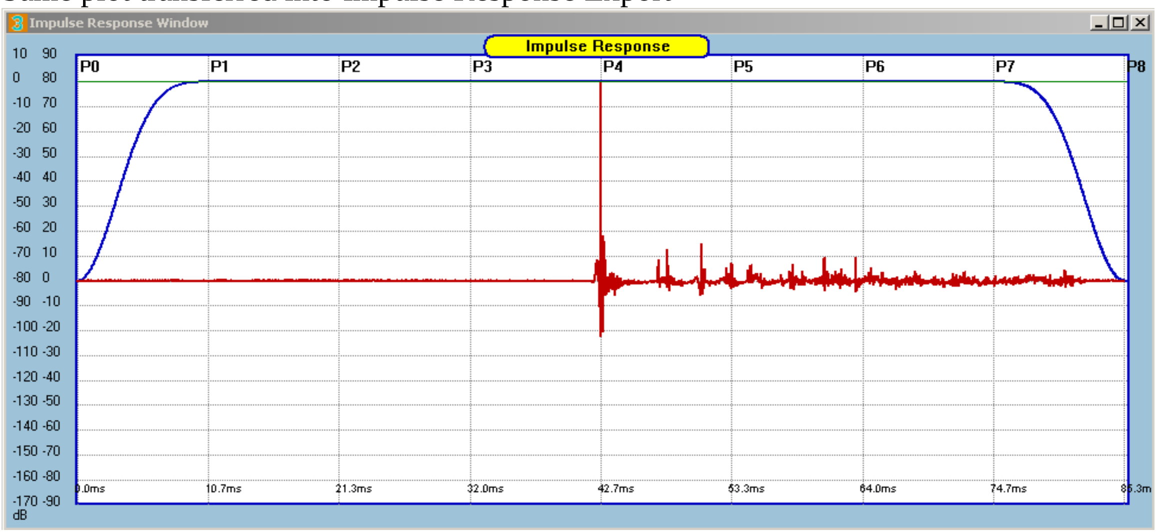


Same plot presented also in logarithmic vertical scale with pre- and post-ringing limits superimposed.

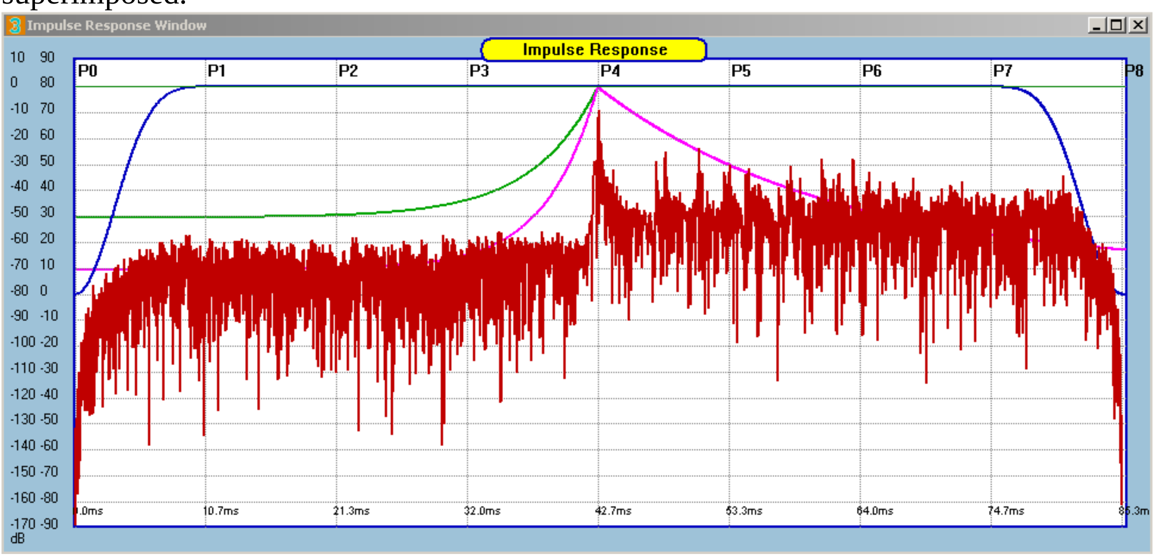


30deg off-axis, impulse response magnification = 2x

Same plot transferred into Impulse Response Export

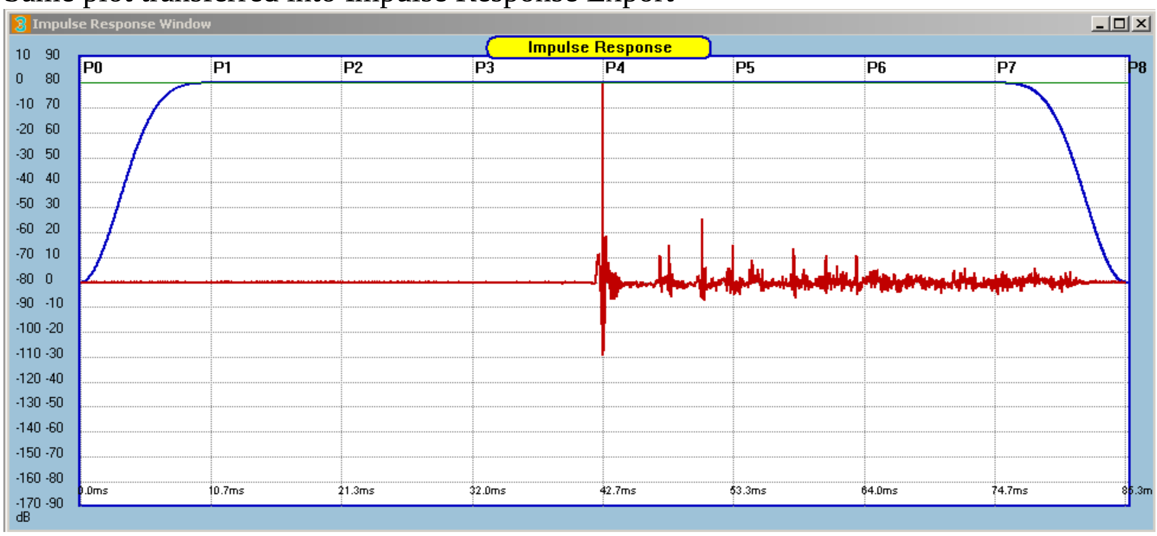


Same plot presented also in logarithmic vertical scale with pre- and post-ringing limits superimposed.

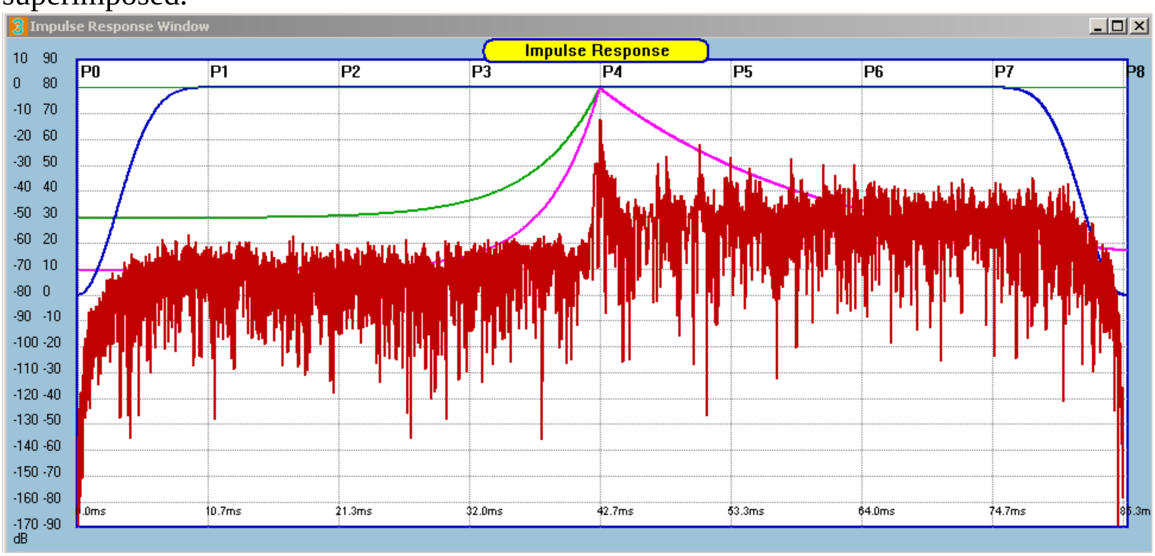


45deg off-axis, impulse response magnification = 4x

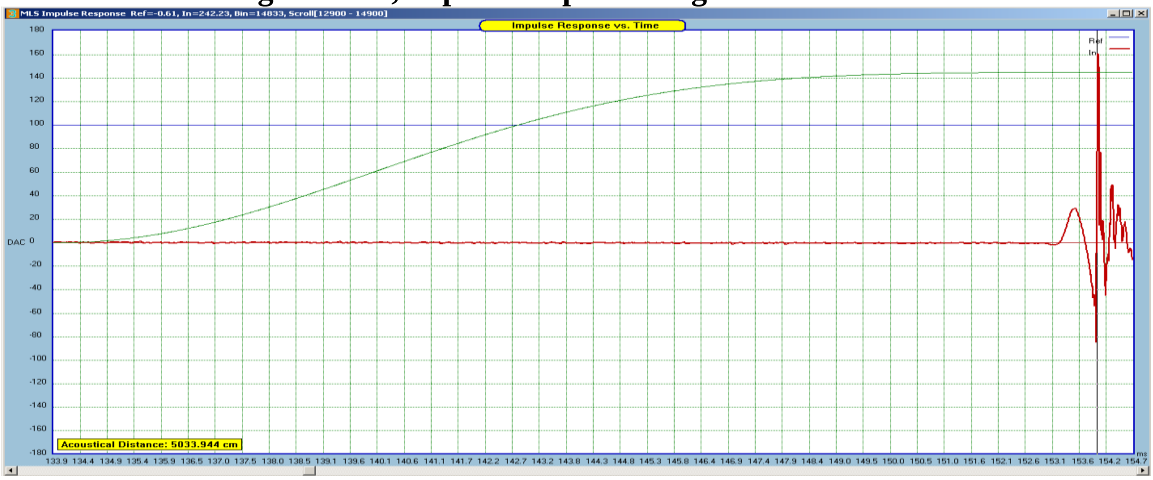
Same plot transferred into Impulse Response Export



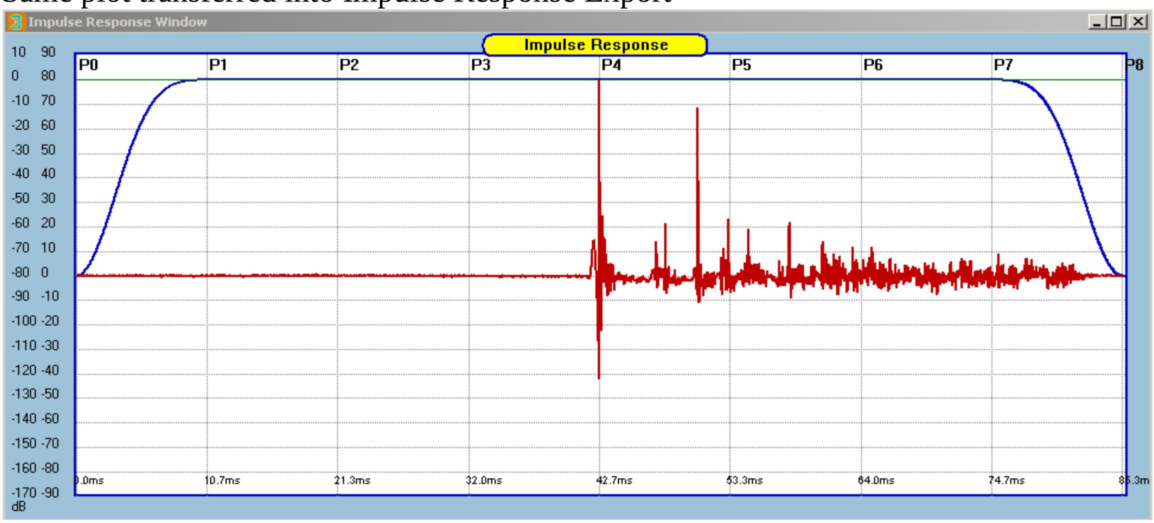
Same plot presented also in logarithmic vertical scale with pre- and post-ringing limits superimposed.



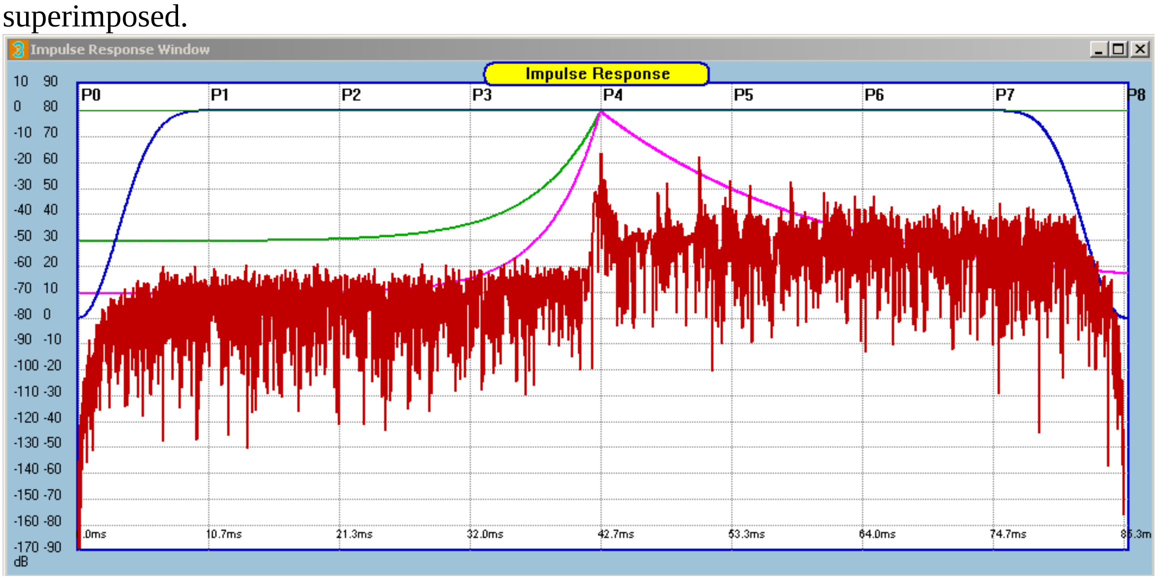
60deg off-axis, impulse response magnification = 8x

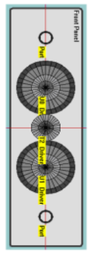


Same plot transferred into Impulse Response Export



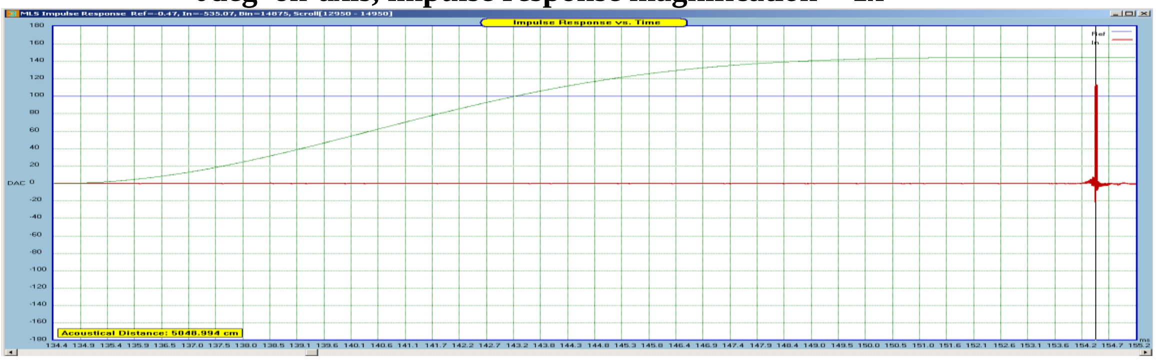
Same plot presented also in logarithmic vertical scale with pre- and post-ringing limits superimposed.



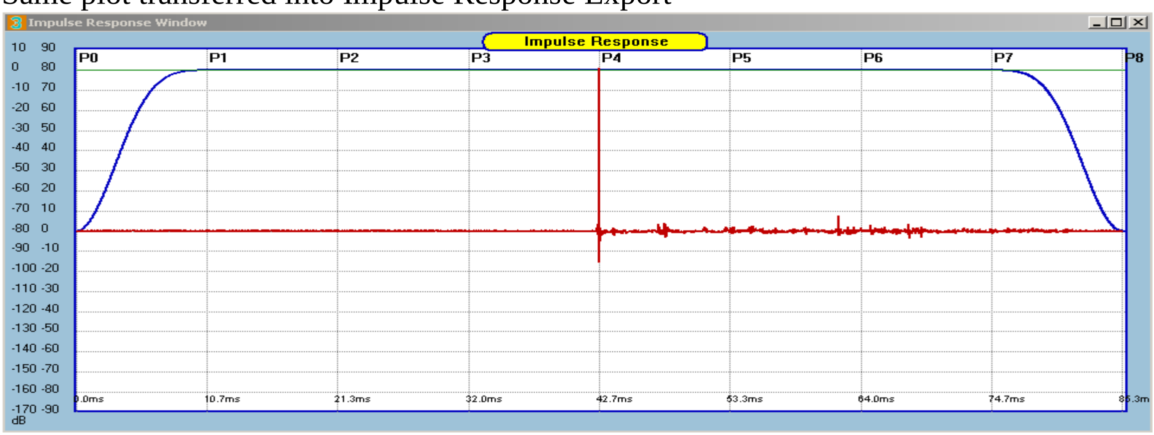


Horizontal pattern

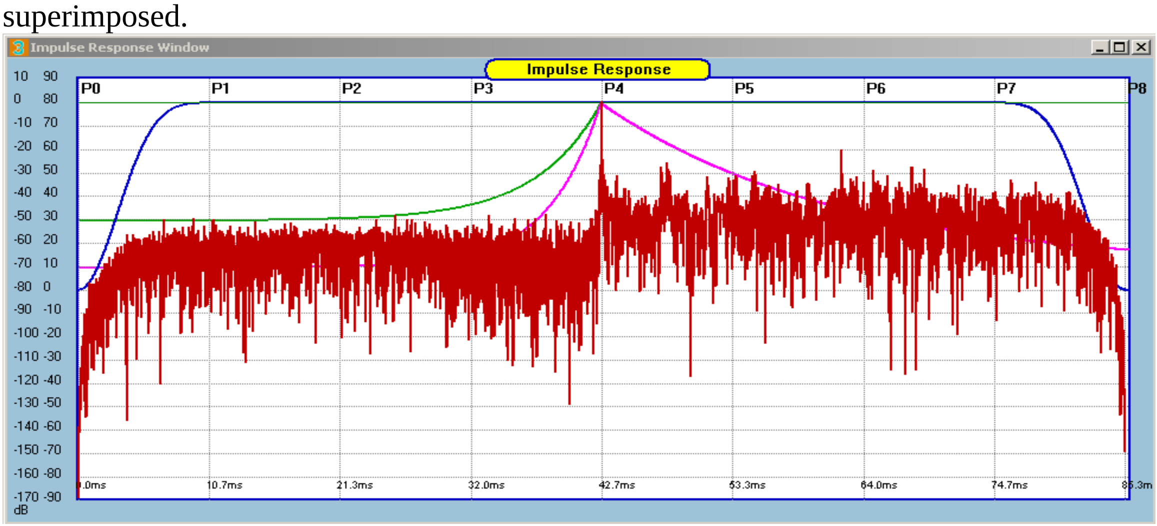
0deg on-axis, impulse response magnification = 1x



Same plot transferred into Impulse Response Export

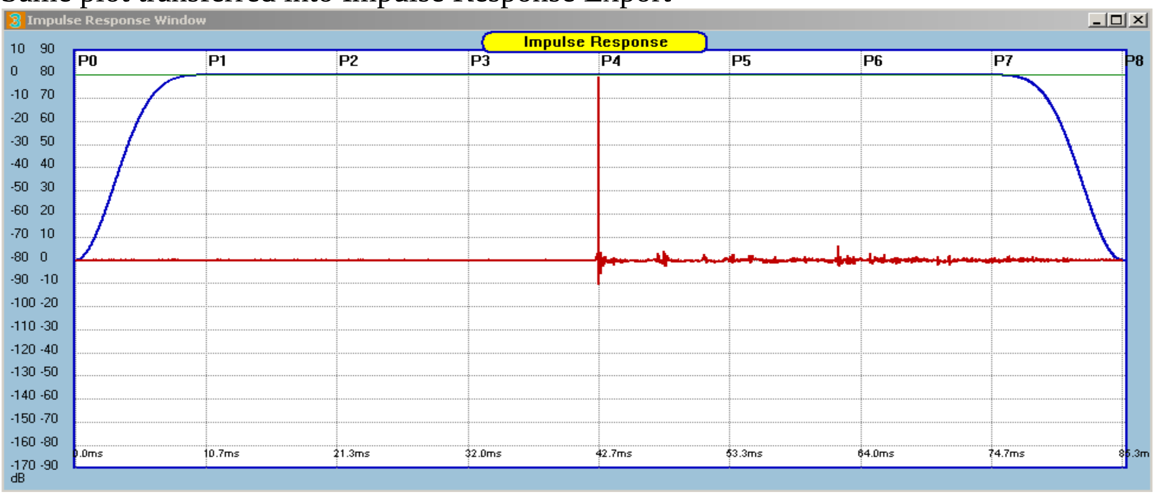


Same plot presented also in logarithmic vertical scale with pre- and post-ringing limits superimposed.

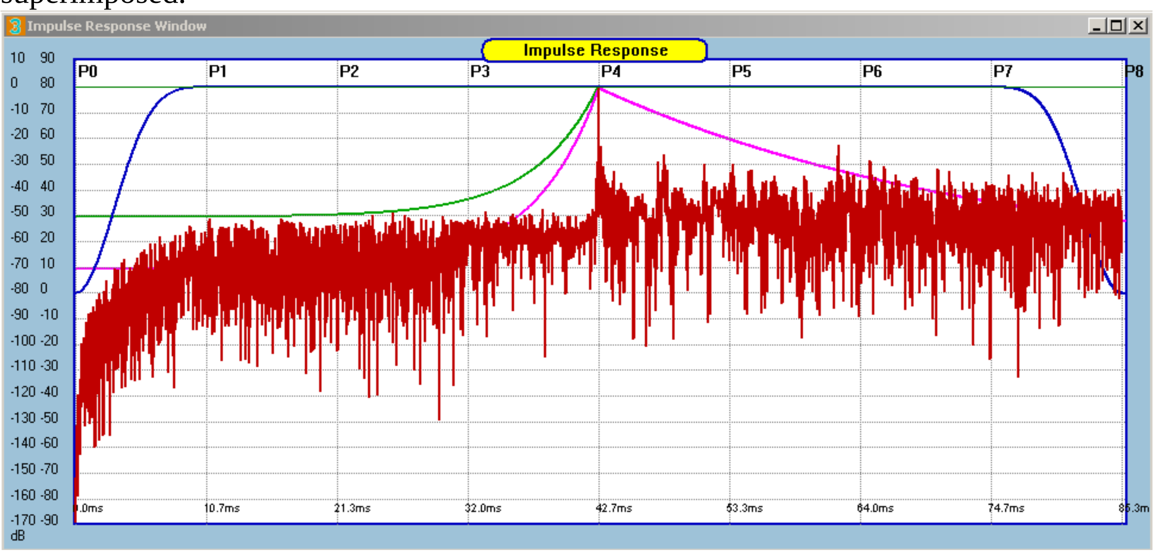


15deg off-axis, impulse response magnification = 1x

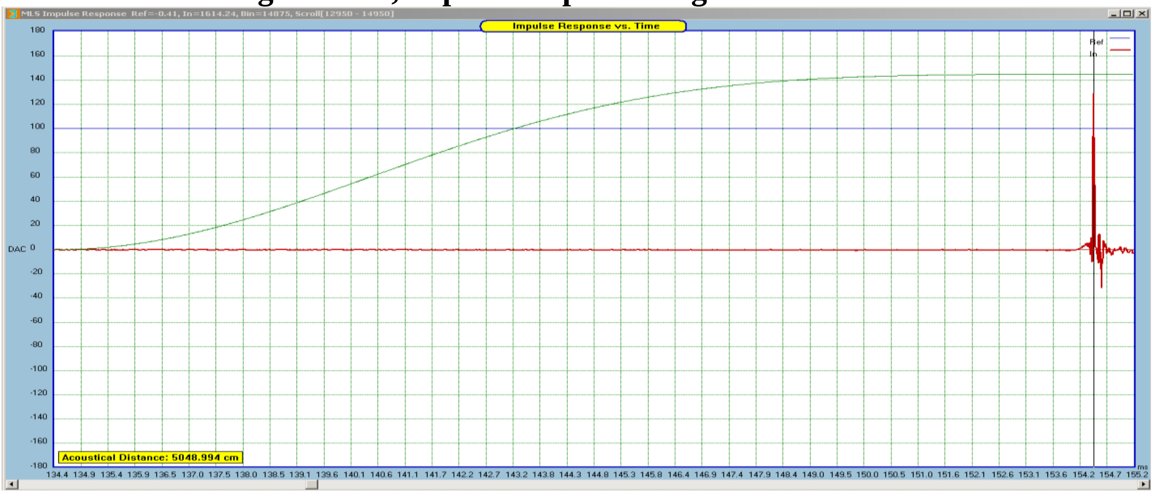
Same plot transferred into Impulse Response Export



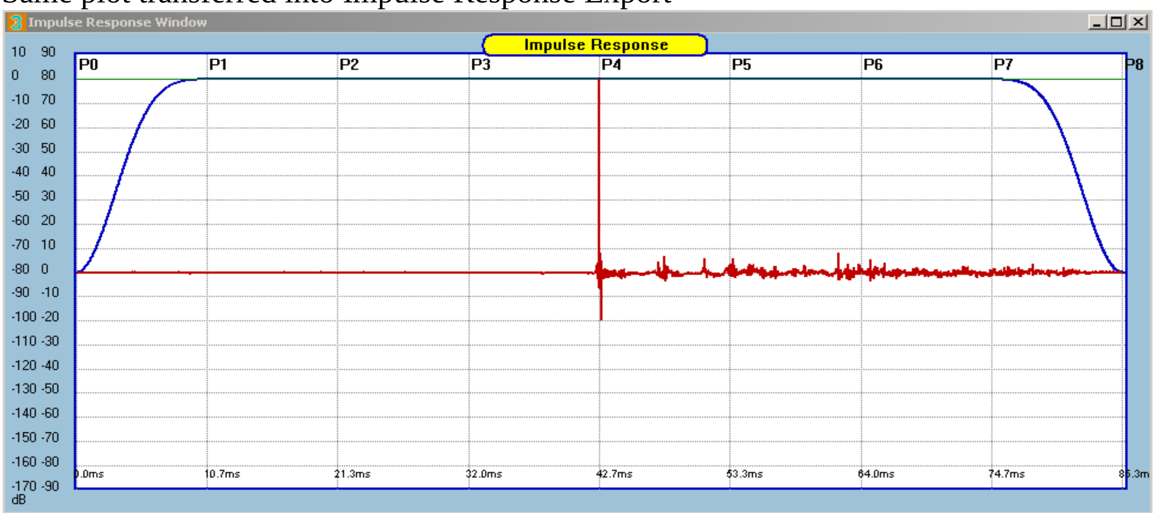
Same plot presented also in logarithmic vertical scale with pre- and post-ringing limits superimposed.



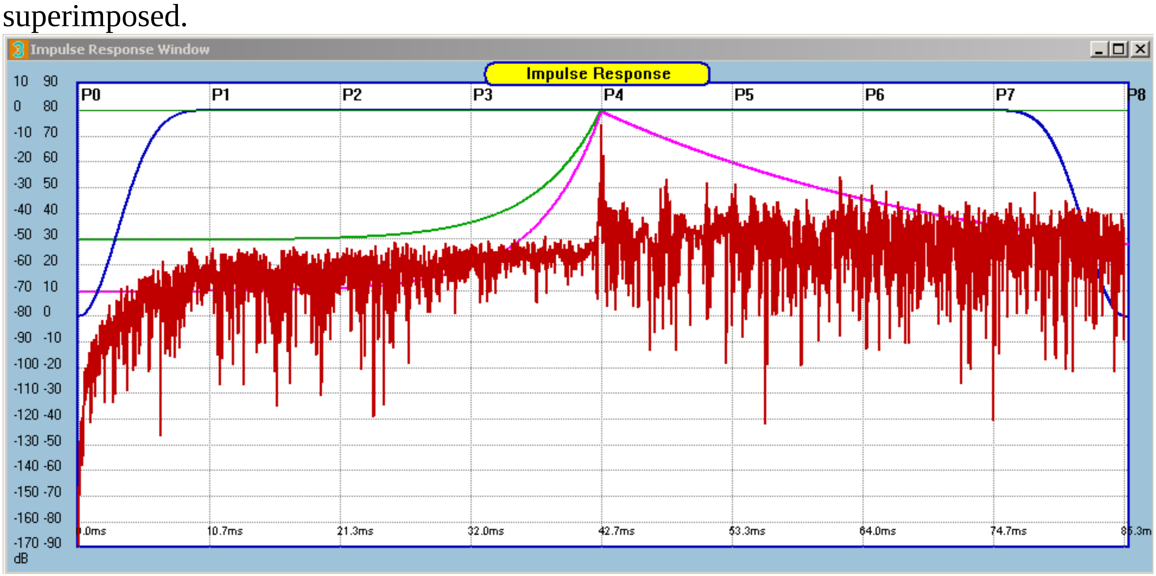
30deg off-axis, impulse response magnification = 2x



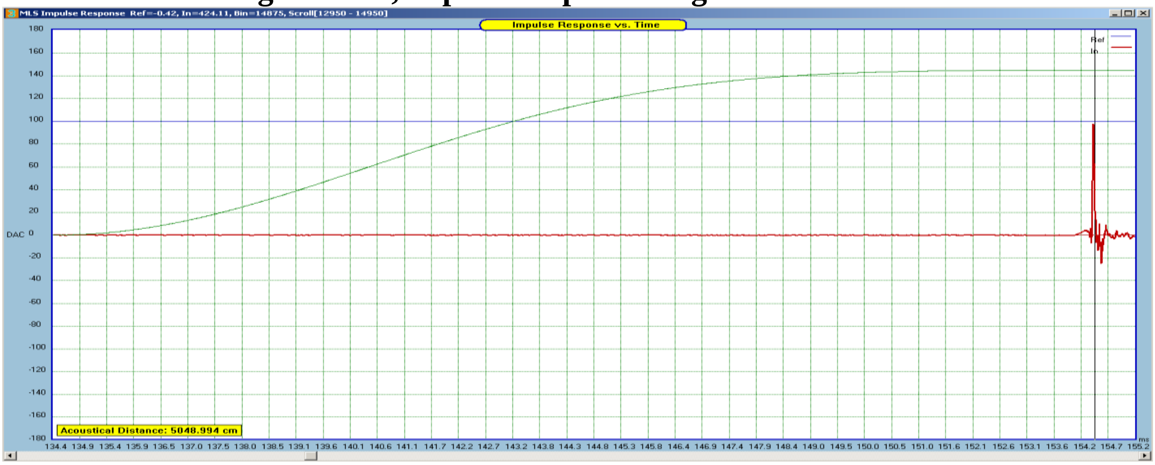
Same plot transferred into Impulse Response Export



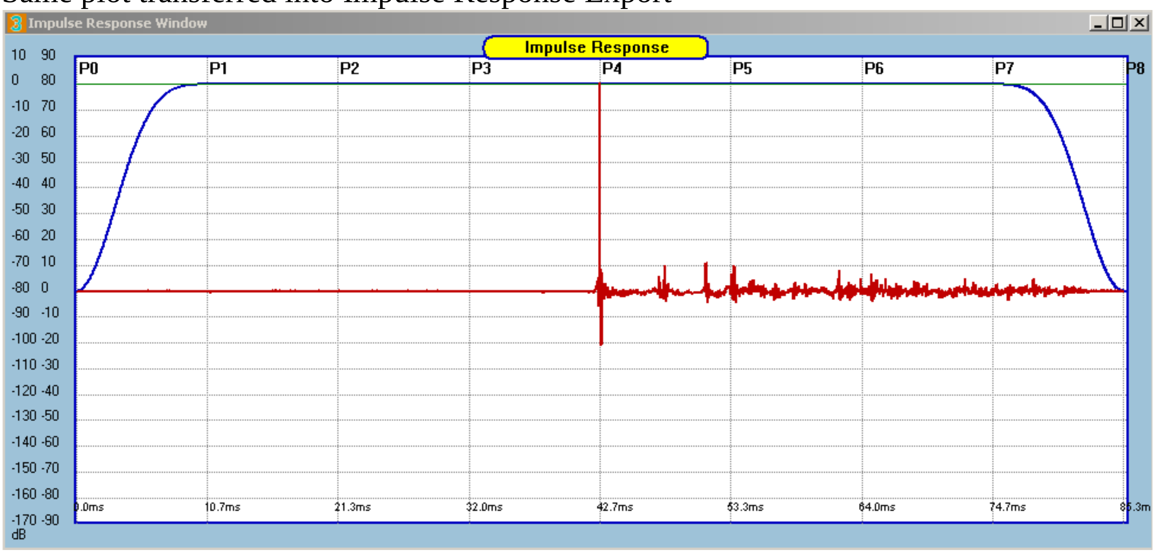
Same plot presented also in logarithmic vertical scale with pre- and post-ringing limits superimposed.



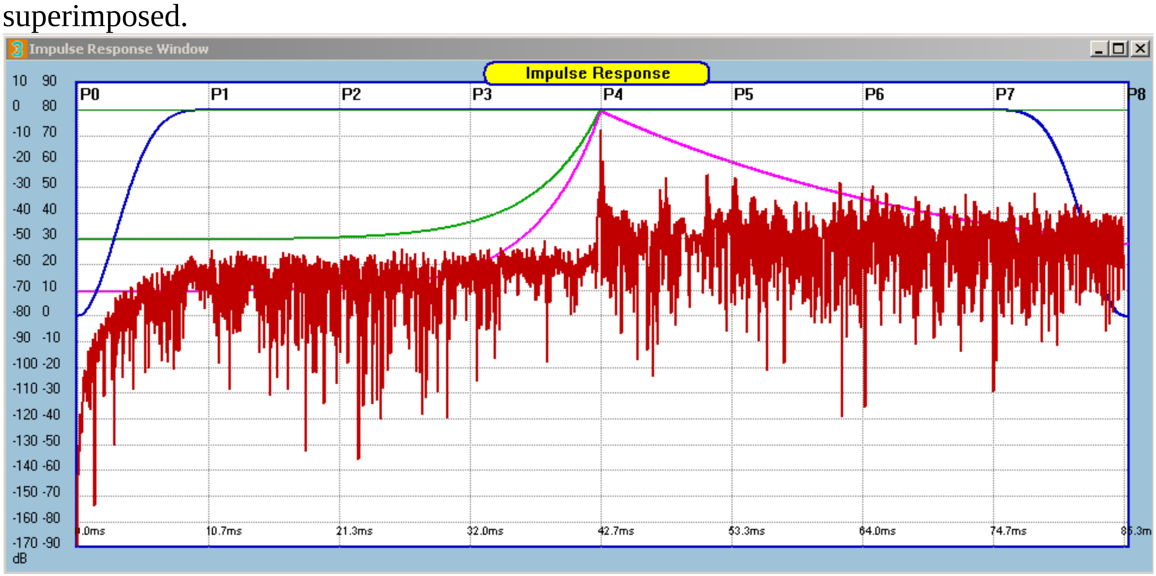
45deg off-axis, impulse response magnification = 2x



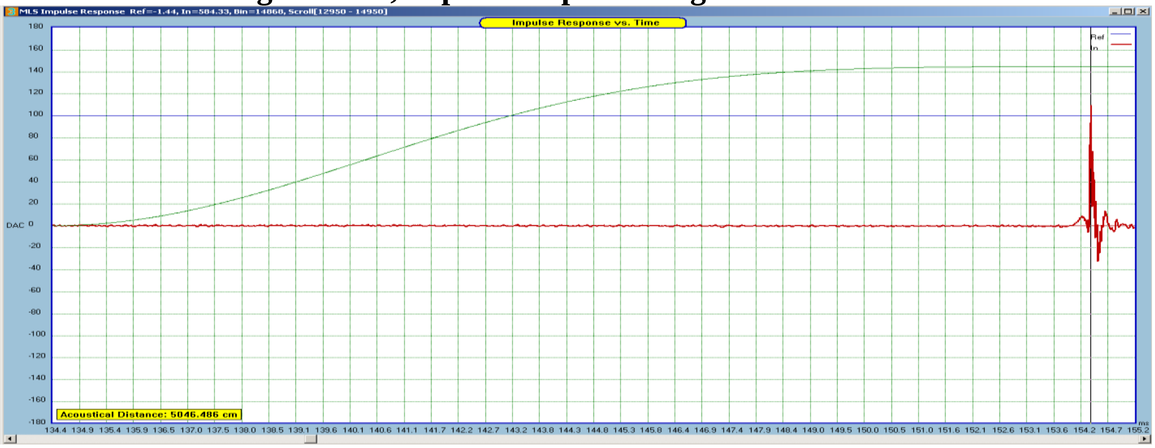
Same plot transferred into Impulse Response Export



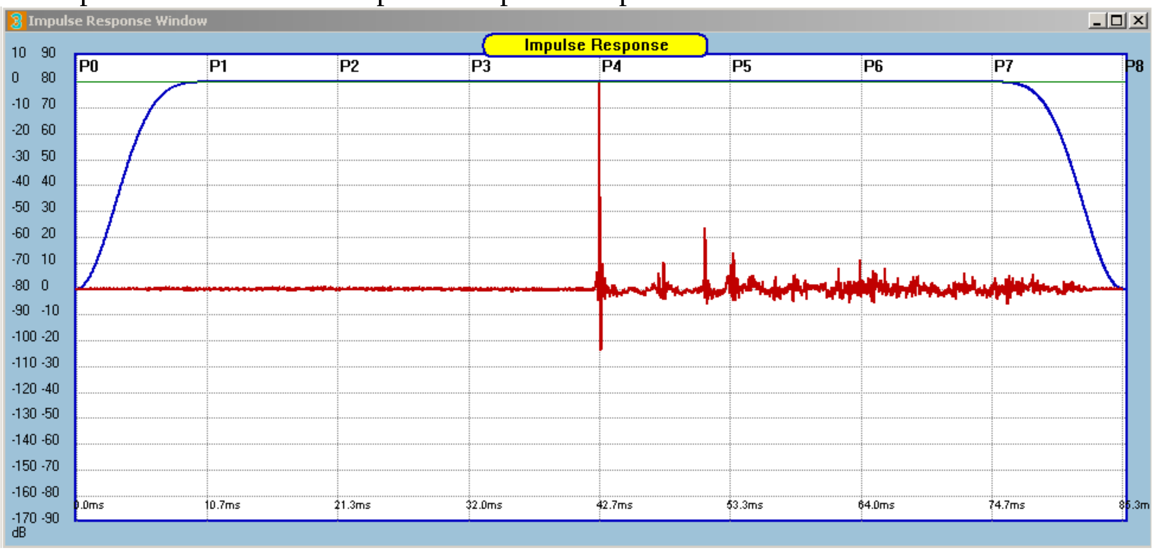
Same plot presented also in logarithmic vertical scale with pre- and post-ringing limits superimposed.



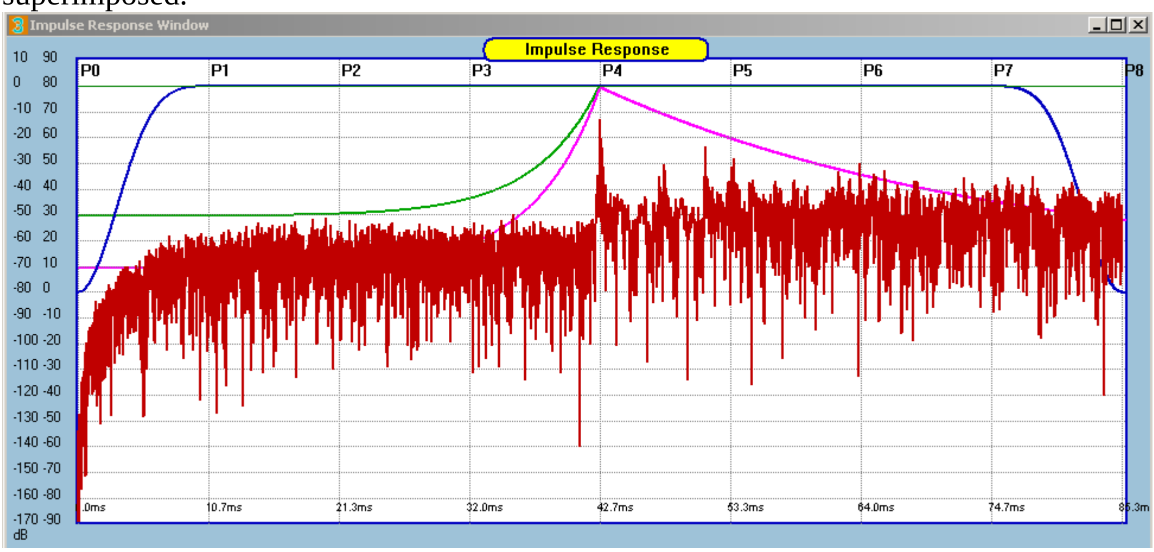
60deg off-axis, impulse response magnification = 4x

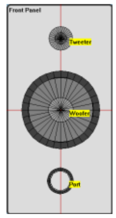


Same plot transferred into Impulse Response Export



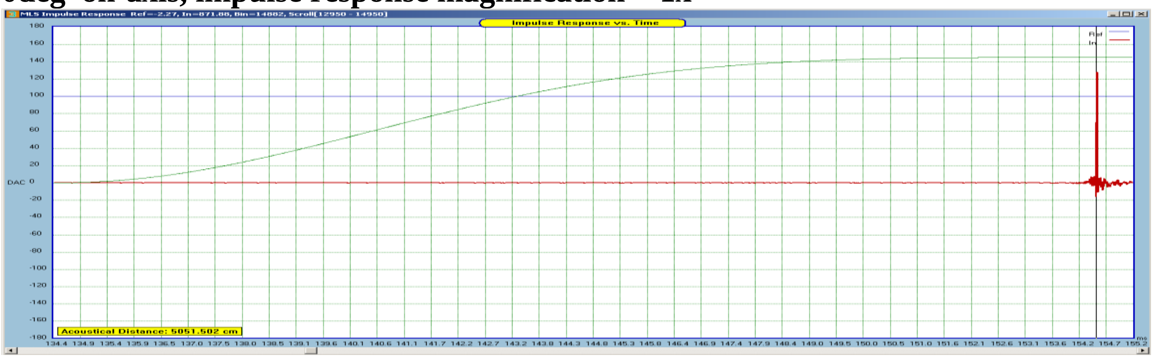
Same plot presented also in logarithmic vertical scale with pre- and post-ringing limits superimposed.



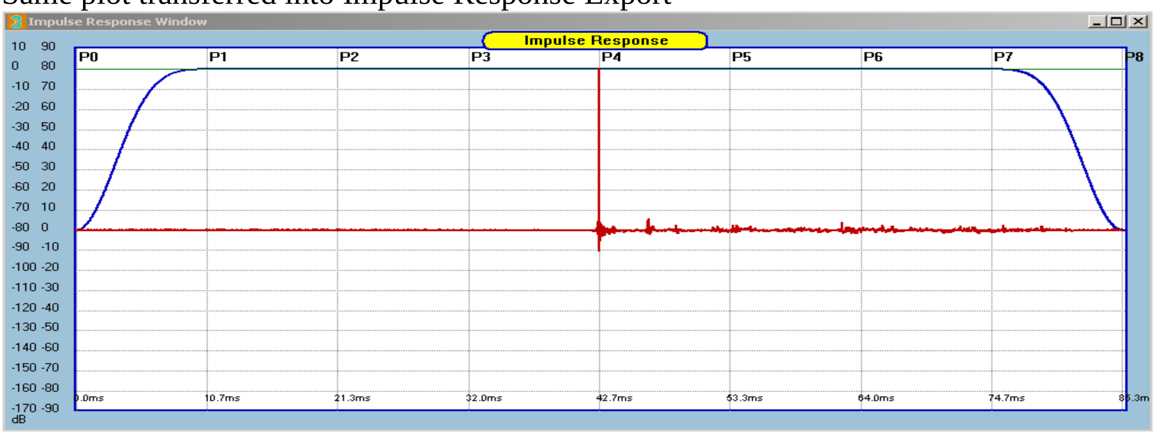


Horizontal orientation

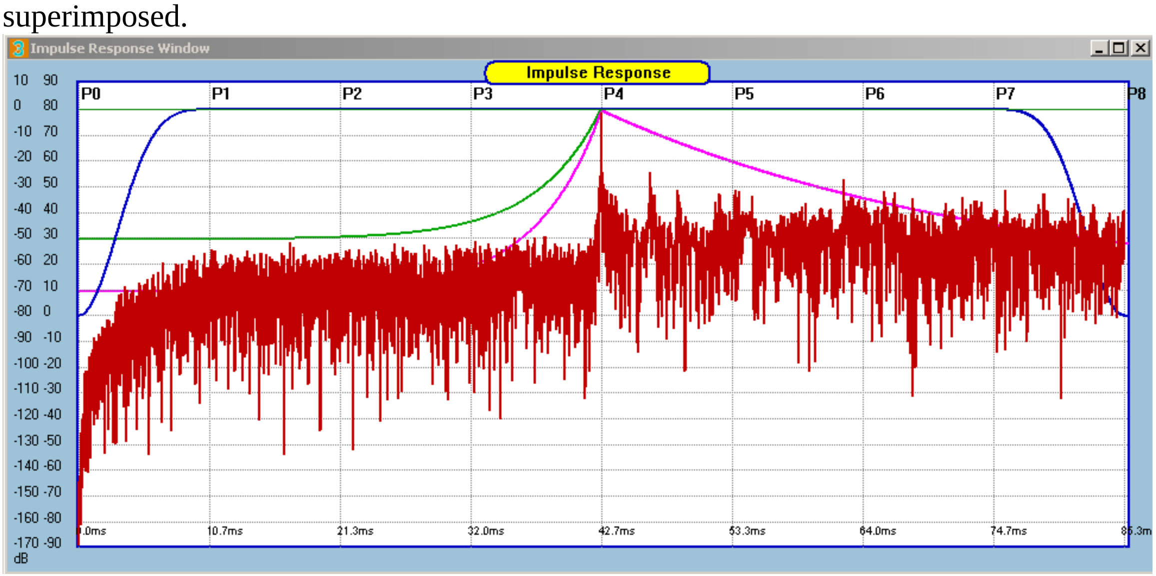
0deg on-axis, impulse response magnification = 1x



Same plot transferred into Impulse Response Export

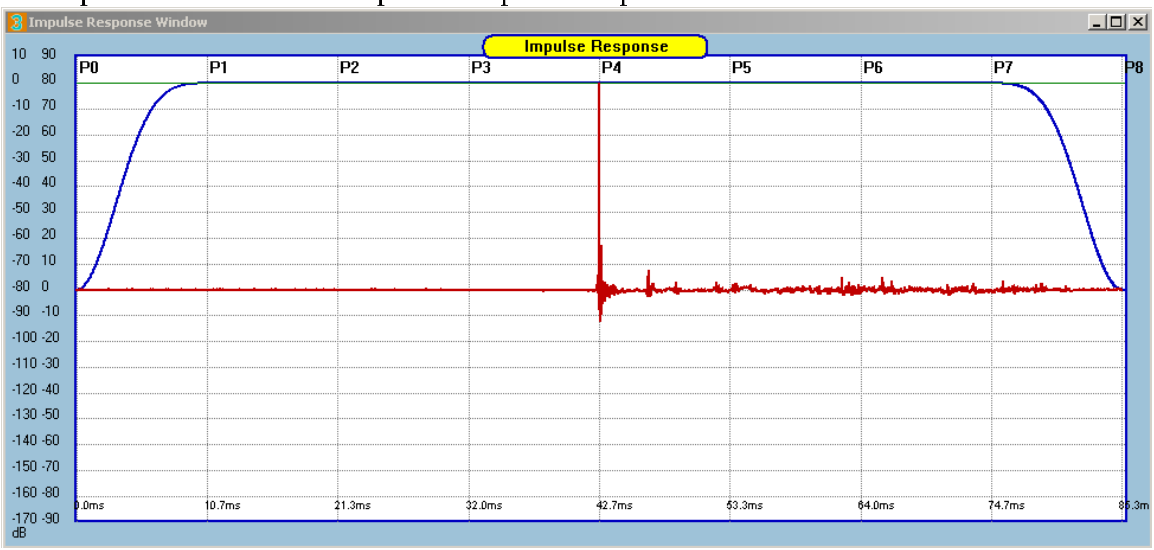


Same plot presented also in logarithmic vertical scale with pre- and post-ringing limits superimposed.

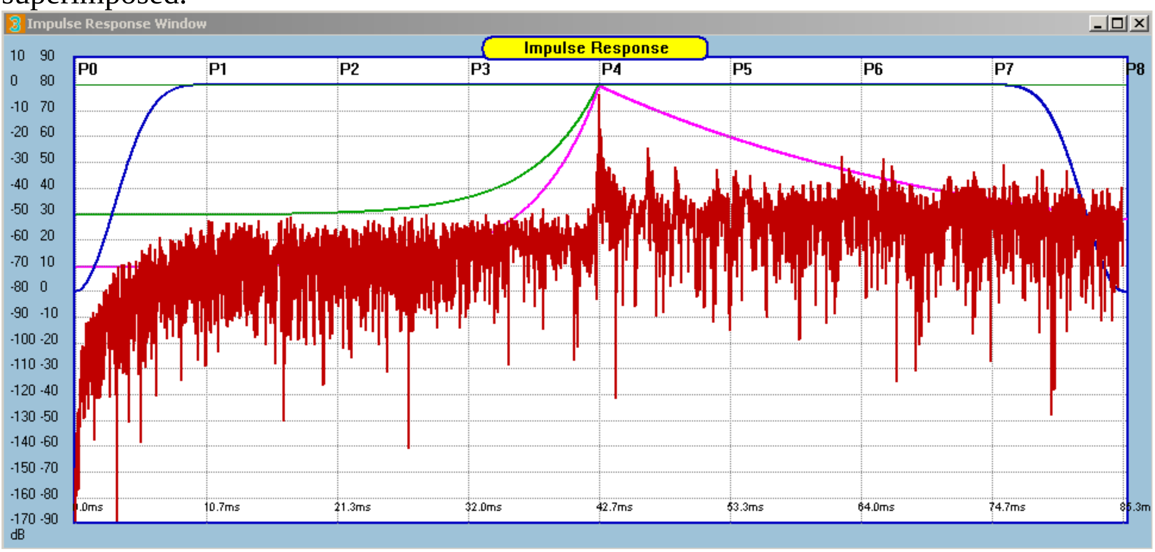


15deg off-axis, impulse response magnification = 1x

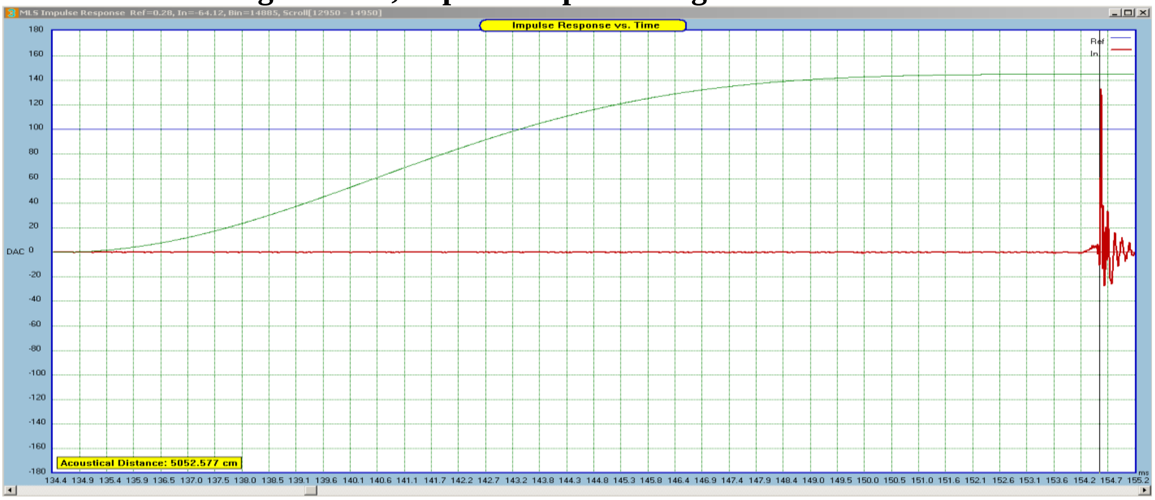
Same plot transferred into Impulse Response Export



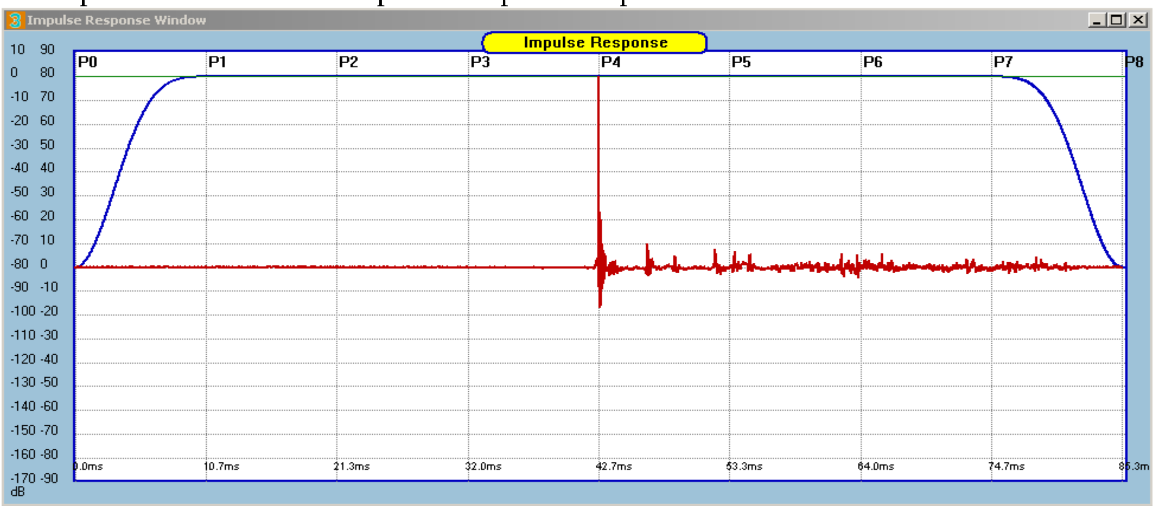
Same plot presented also in logarithmic vertical scale with pre- and post-ringing limits superimposed.



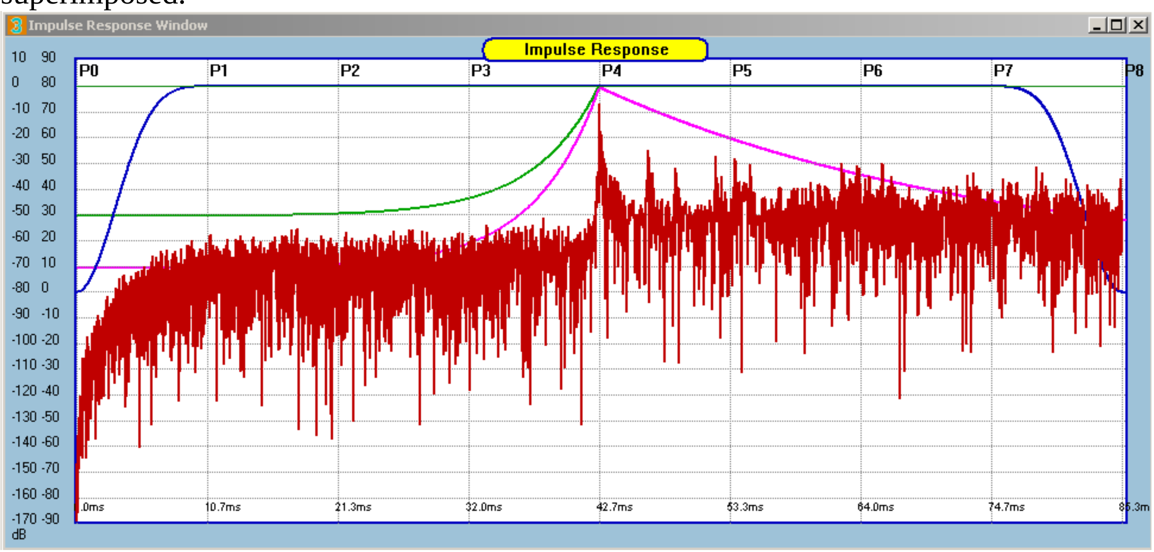
30deg off-axis, impulse response magnification = 2x



Same plot transferred into Impulse Response Export

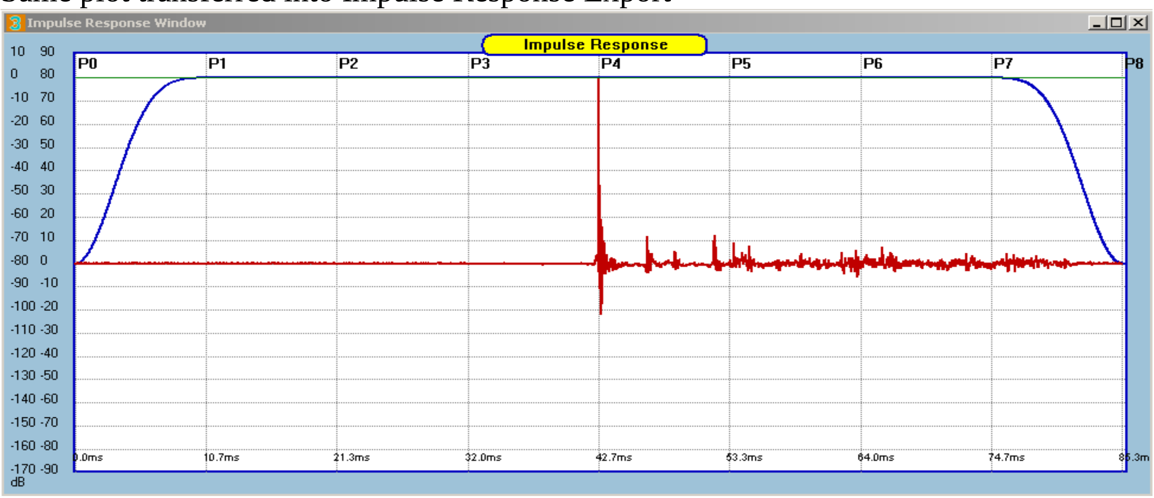


Same plot presented also in logarithmic vertical scale with pre- and post-ringing limits superimposed.

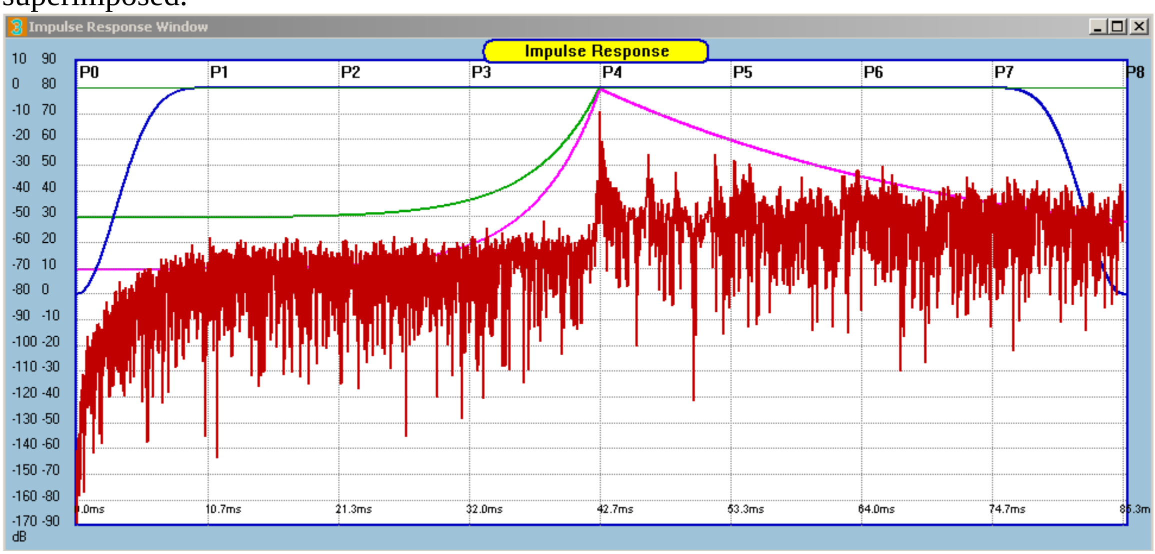


45deg off-axis, impulse response magnification = 2x

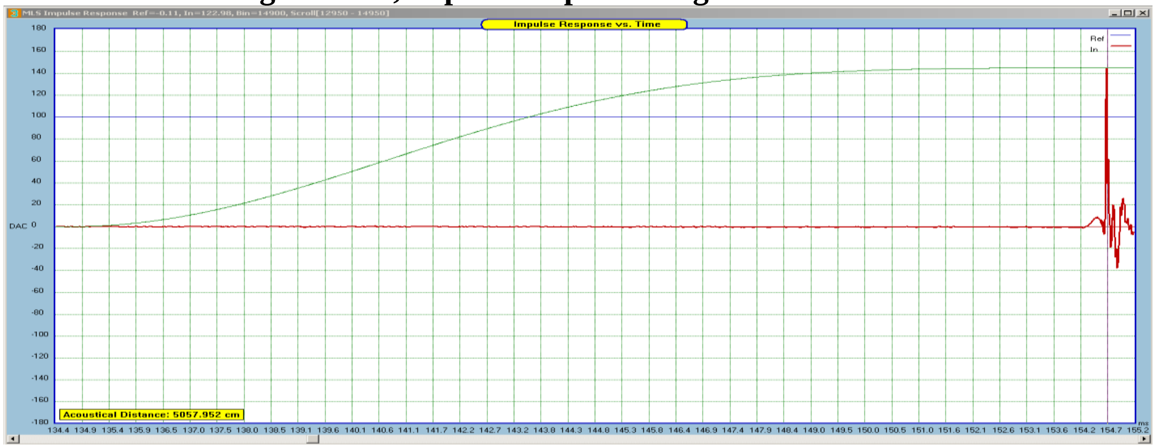
Same plot transferred into Impulse Response Export



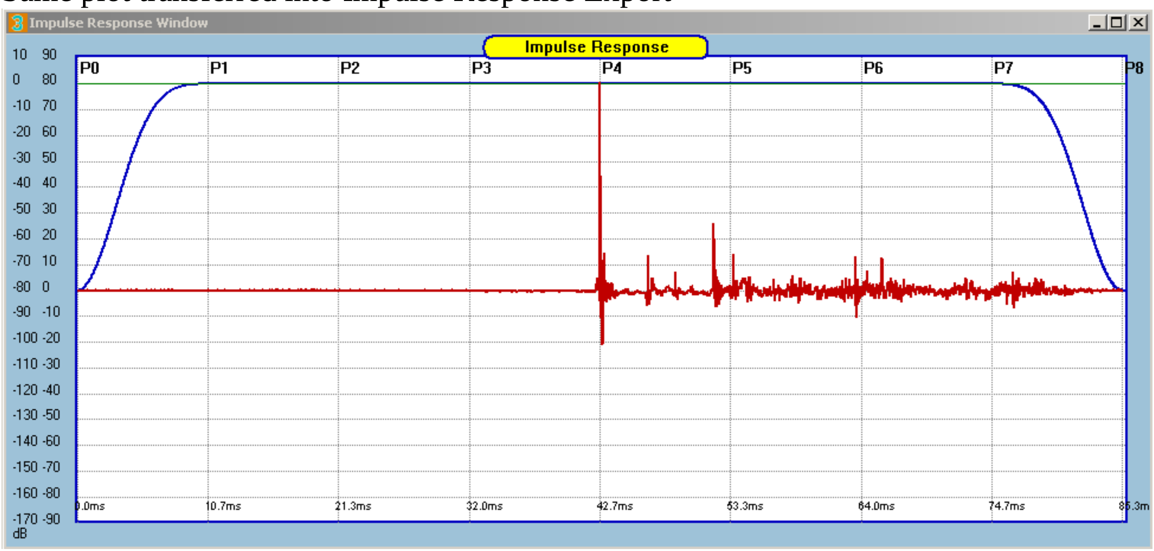
Same plot presented also in logarithmic vertical scale with pre- and post-ringing limits superimposed.



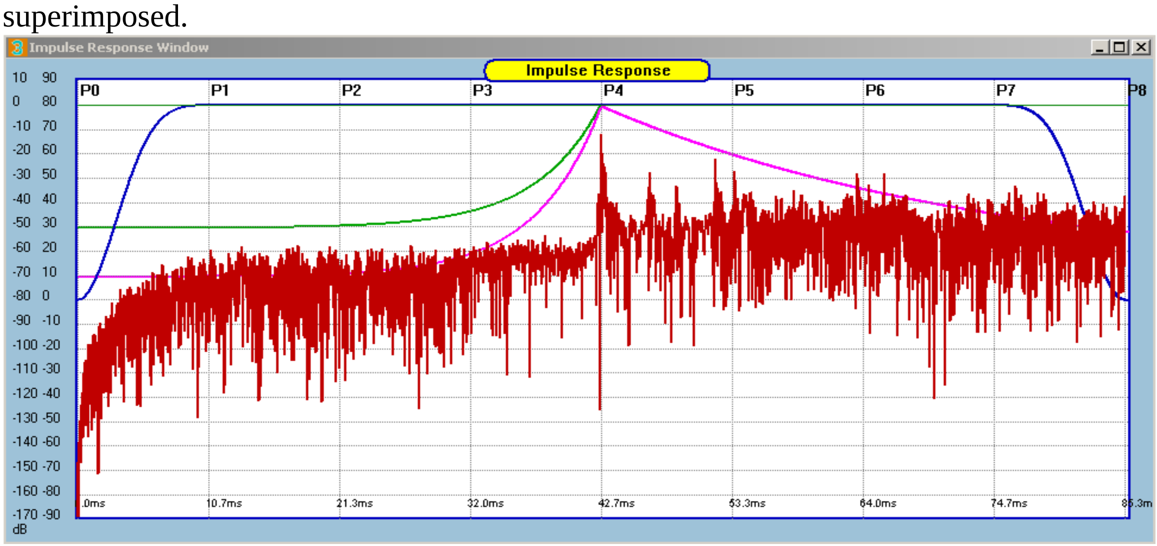
60deg off-axis, impulse response magnification = 4x



Same plot transferred into Impulse Response Export

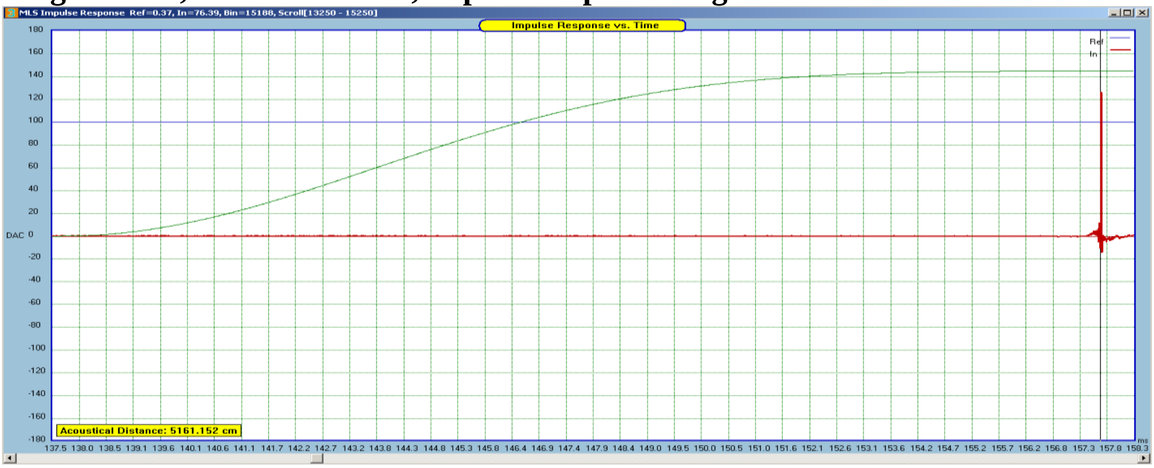


Same plot presented also in logarithmic vertical scale with pre- and post-ringing limits superimposed.

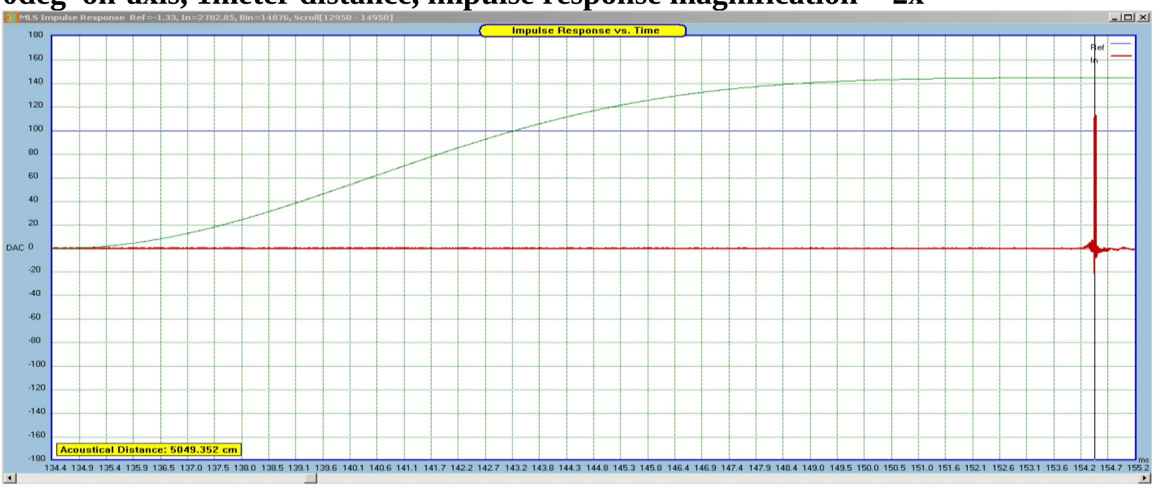


Impulse Response due to distance

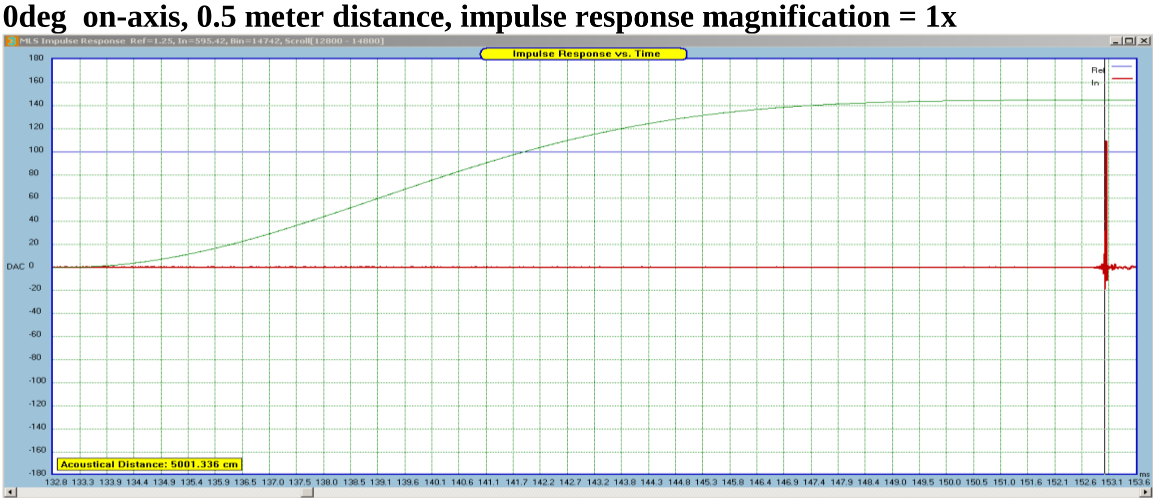
0deg on-axis, 2meter distance, impulse response magnification = 4x



0deg on-axis, 1meter distance, impulse response magnification = 2x



0deg on-axis, 0.5 meter distance, impulse response magnification = 1x



We have now examined 18 impulse responses of two different loudspeaker systems: TW configuration and WTW configuration. Both two-way loudspeakers were equipped with +/-24dB/oct Linkwitz-Riley crossovers at 2000Hz. These loudspeakers are not small in size, and would be considered to be quite realistic representatives of contemporary domestic Hi-Fi systems.

Measurements were conducted a **0deg, 15deg, 30deg, 45deg and 60deg off-axis** angles in horizontal plane for TW loudspeaker. The WTW loudspeaker was measured in both planes and also evaluated at different distances on axis.

It was shown, that:

1. Impulse response (and associated SPL) will deteriorate for off-axis measurements. This is also very true for minimum-phase measurements.

2. Impulse response will marginally differ for various distance measurements. This is also very true for minimum-phase measurements. Changes in SPL would not be noticeable.

3. Measured off-axis, the impulse response deteriorates more for bigger off-axis angles. The deteriorating (increasing) pre-ringing is evident on all these plots – this is confirming the theory. However, the pre-ringing was not extended beyond 2ms, and will easily be suppressed by the pre-masking effect up to 20ms – just as post-ringing is suppressed by the post-masking effect up to 100-150ms.

4. Both effects: pre-masking and post-masking are quite desirable in making the pre- and post-ringing of the impulse response inaudible.

5. Pre-masking effect and lack of extended pre-ringing is perhaps the main reason as to why the small, residual pre-ringing is not audible.

General Conclusions

When discussing HBT-equalized, linear-phase systems, two issues are often commented upon:

1. Hard equalization works only on-axis.
2. Linear-phase systems suffer from pre-ringing.

This paper attempts to shed some light at these two issues.

Hard equalization works only on-axis?.

Hard equalization (Inverse HBT) will perfectly flatten SPL and phase curves only for the curve, that the HBT was calculated for – which is typically on-axis. The evidence of this is clearly provided on Figure 12, where the red curve (HBT equalized) is as flat as they come.

However, examination of Figures 13, 14 and 15 would lead to the conclusion, that 15deg off-axis performance is much better for the equalized case, 30deg off-axis is still better and more extended than unequalised case, and even 45deg off-axis is still more extended curve than the unequalised case. Not to mention, that the lower the frequency, the better the equalization will be – this is due to diffraction (the major factor causing the off-axis deterioration) being reduced to zero at low frequencies. All presented curves are the in-room measurements results, therefore bass frequencies are windowed out. However, anechoic measurements would reveal, that equalized SPL at low-frequencies is much better than unequalised for all measured angels. This was evident from subwoofer measurements documented in another paper.

So, is the statement "Hard equalization works only on-axis" correct?. The answer can not be contained in one word because the measurements show, that:

1. On-axis, the equalised SPL curve was as flat as they come. So, yes, hard EQ works perfectly on-axis.
2. For angles up to +/-15degrees the equalized SPL was much better across the whole frequency range than unequalised one.
3. For angles up to +/-30degrees the equalized SPL was still better across the whole frequency range than unequalised one.
4. For larger angles the equalised SPL was not worse that the equalized one.
5. For low-frequencies (less than 2kHz for the measured systems), the equalised SPL was always better then unequalised one in Vertical Orientation of the loudspeaker.

Overall, the HBT-equalized system is performing much better than a system without the equalization, and will provide flat frequency response for the all-important first-arrival signals and improved performance at many other angles.

Linear-phase systems suffer from pre-ringing?

Yes, linear-phase systems can exhibit pre-ringing in their impulse response, but it seems to be rather difficult to make it audible.

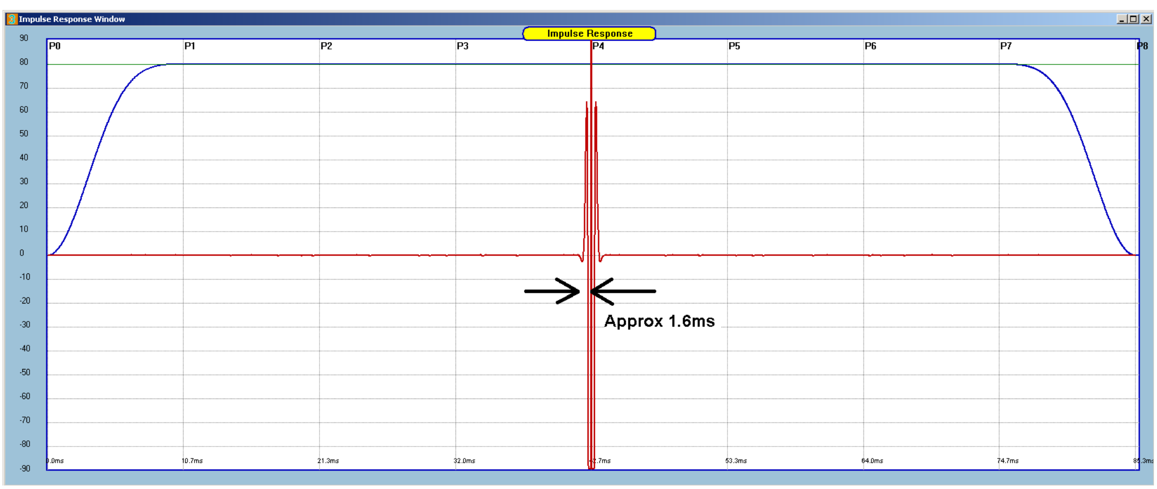
Let us not forget, that post-ringing, exhibited by all traditional, minimum-phase systems, is twice as bad as pre-ringing of the linear-phase systems, and yet, it goes unnoticed. This is because the post-masker works effectively up to 100-200ms of the impulse response duration.

Just the same, the pre-ringing is masked by the pre-masker effectively up to 20ms.

Impulse responses presented in this paper were conducted on two loudspeakers of different configurations and sizes, so the results are not unique, and are representative of contemporary style of designs. Crossover frequency for these two-way designs was set to 2000Hz and 24dB/oct LR configuration.

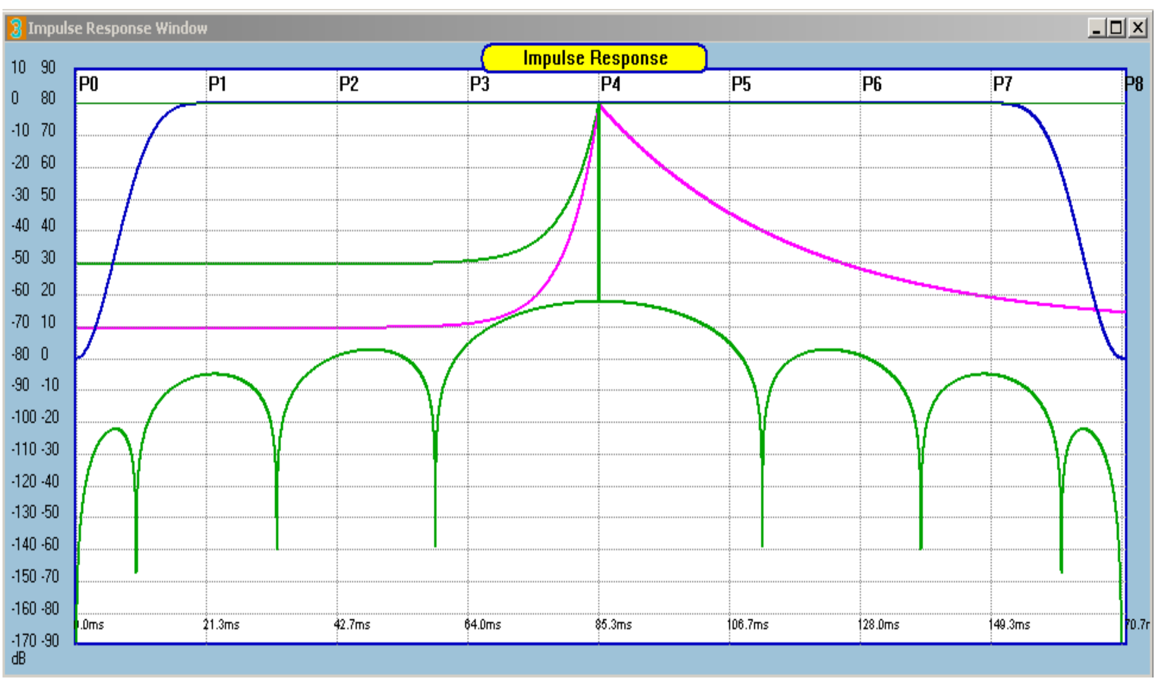
It is anticipated, that steeper slopes used (above +/-24dB/oct) would increase pre-ringing, but higher crossover frequency (2kHz) would reduce the duration of pre-ringing. In all cases for this system, the pre-ringing was not evident further away that 2ms from the peak of the impulse response and was not audible.

If we consider a ***largely magnified in amplitude***, linear-phase impulse response of a high-pass 2000Hz, 24dB/oct LR filter – see figure below, we would conclude, that pre-ringing is effectively extinguished about 1.6ms in front of the impulse response.

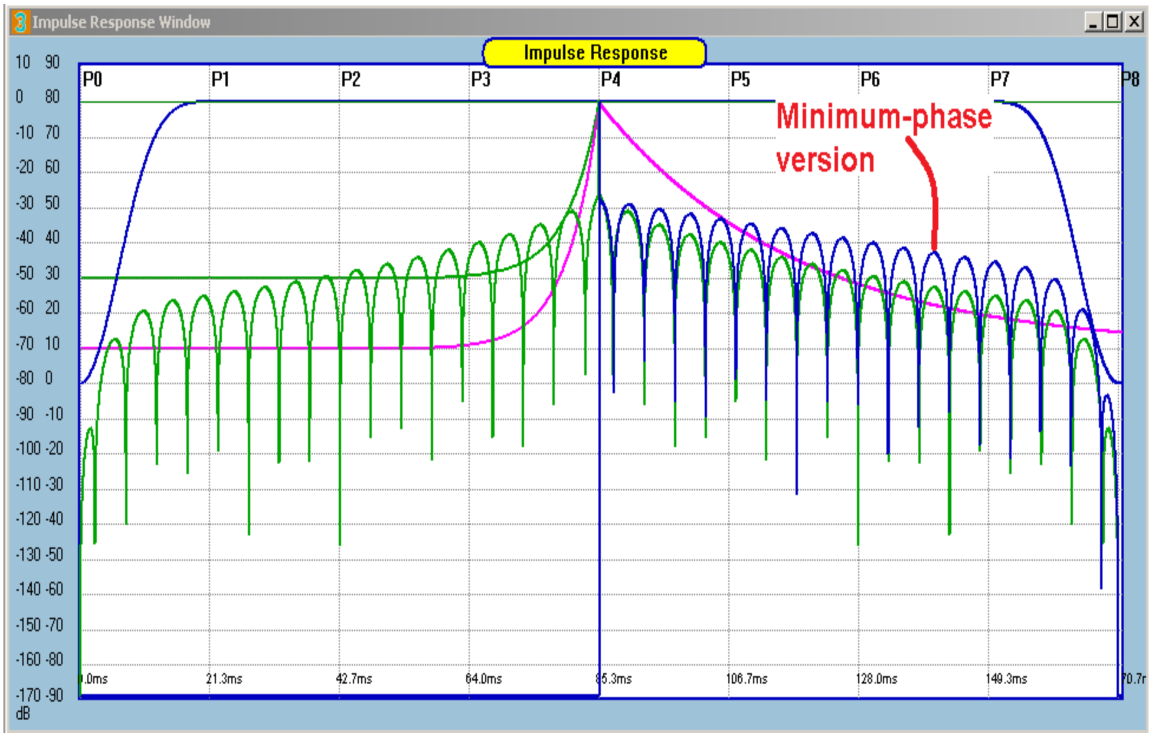


Now, for comparison, we can use 60deg off-axis, system impulse response, with 8x magnification, and conclude, that the pre-ringing is also effectively extinguished about 1.6ms in front of the impulse response. This should come as no surprise, because even that at 60deg off-axis we have lost the benefits of HP/LP impulse responses cancelling each other, the pre-ringing is actually very short in time – see figure below.

Extending this conclusion into impulse responses of filters designed for other frequencies, we find, that the 20ms pre-masker effectiveness limit will be attained by 48dB/oct HP Butterworth filter with cut-off frequency of 20Hz – see impulse response figure below, shown with impulse response in decibel scale.



Or peaking filters, such as the one depicted below. Here we have a Q-Parametric filter with gain of 30dB and Q-factor of 10.



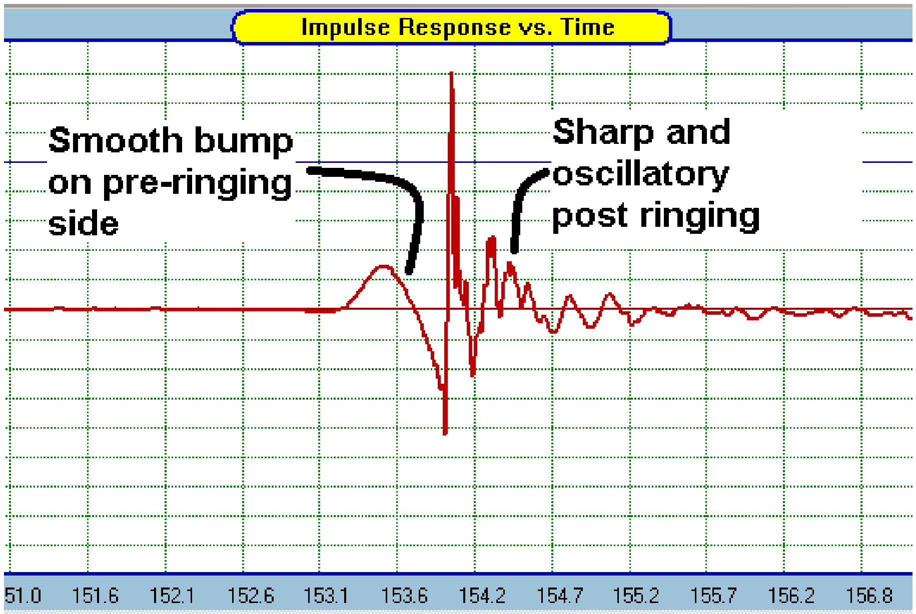
What causes off-axis pre-ringing?

System on-axis impulse response exhibits near perfect impulse response. This is because (1) pre-ringing duration is often so short, that it is masked by the pre-masker and (2) pre-ringing in low-pass channel will cancel pre-ringing in high-pass channel. However, the off-axis performance will suffer degradation. What causes this degradation?. One suggestion is that impulse responses do not add as perfectly as the on-axis summation, and one possible cause is the time-of-flight difference between woofer and tweeter radiation. This is further obscured by diffraction effects.

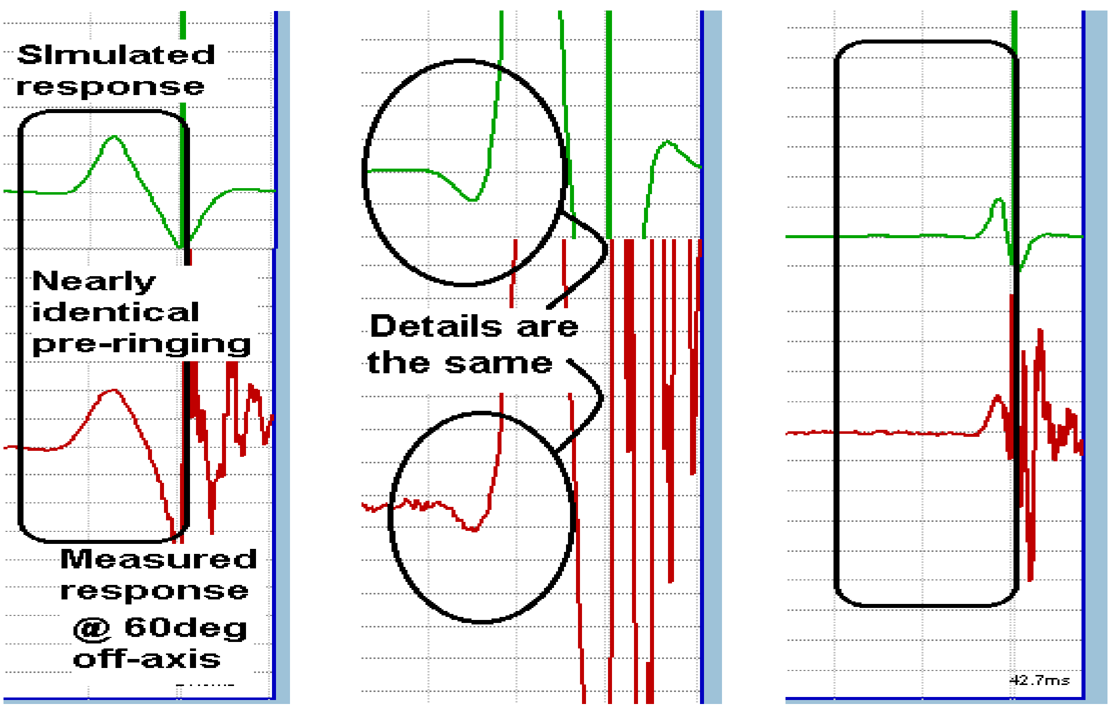
Another clue is provided by the shape of the pre-ringing section of the impulse response curve. On the figure below, it is clearly observable, that the curve is smooth and looks like a bump, as opposed to the jagged shape of the post-ringing curve.

The post-ringing curve includes some high-frequency components, as evidenced by the sharp and oscillatory nature of the curve – see figure below.

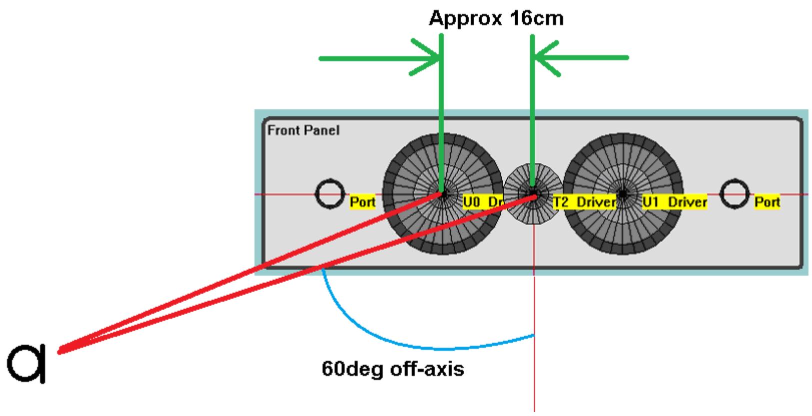
Such differentiation would indicated, that woofer and a low-pass filter are somehow more involved in exposing the pre-ringing process in this measurement and the woofer driver and it's filter are more dominant in this particular time zone. As suggested above this would happen, if the woofer signal was advanced.



Figures presented below on the left and the middle, show tweeter impulse response of the +24dB/oct LR filter delayed by 45 time samples at 96kHz in time reference to -24dB/oct LR woofer filter. This is equivalent of 0.4687 milliseconds, or 16.13 cm difference in sound travel time. The red plot on all figures below, represents measured, 60deg off-axis impulse response. The resemblance in pre-response is striking, so therefore the timing difference would pass as one of the factors contributing to pre-ringing.



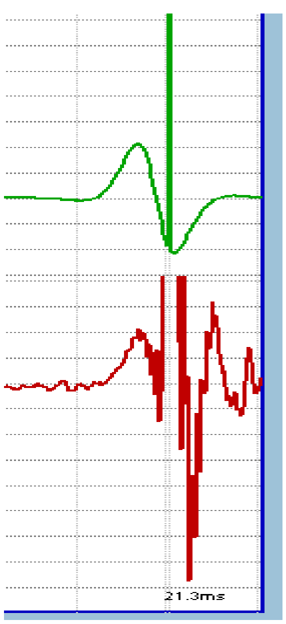
Even more so, on the middle figure above, which is the enlargement of the plots on the left. Here, we can observe, that even minute details of the simulated pre-ringing follow the measured 60deg off-axis results very well. This can not be a coincidence.

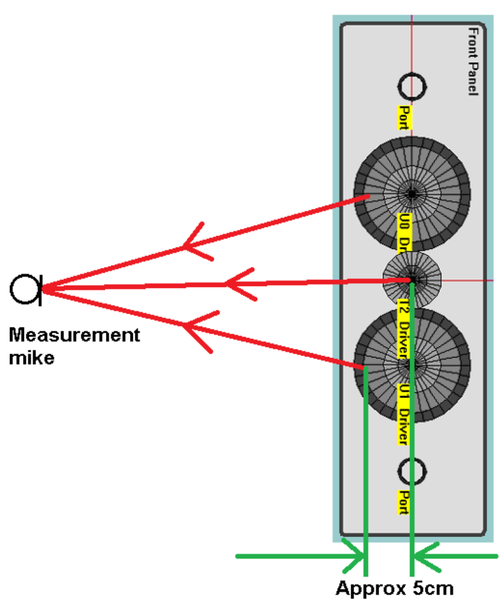


Indeed, the 60deg off-axis measurements locate the microphone approximately 16cm closer to the woofer. So the woofer will be advanced.

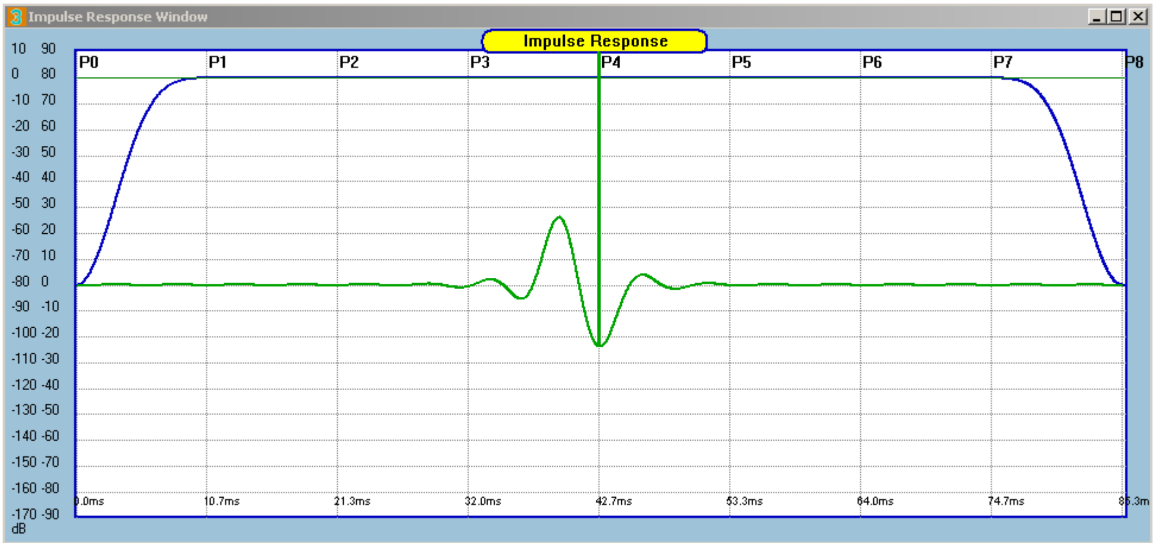
On the right-hand side, we can observe similar simulation of another two-way system with the same filters. This time the delay is only 7 samples, or 2.47cm distance difference. Once again, the resemblance in pre-response is striking, so therefore the timing difference would pass as one of the factors contributing to pre-ringing.

Measuring the WTW loudspeaker in vertical orientation, we find that woofer appears to be advanced by 0.156ms or 5.3cm due to radiation from two driver edges, amplified by diffraction at 2kHz. The red plot on the left figure below, represents measured, 60deg off-axis impulse response.

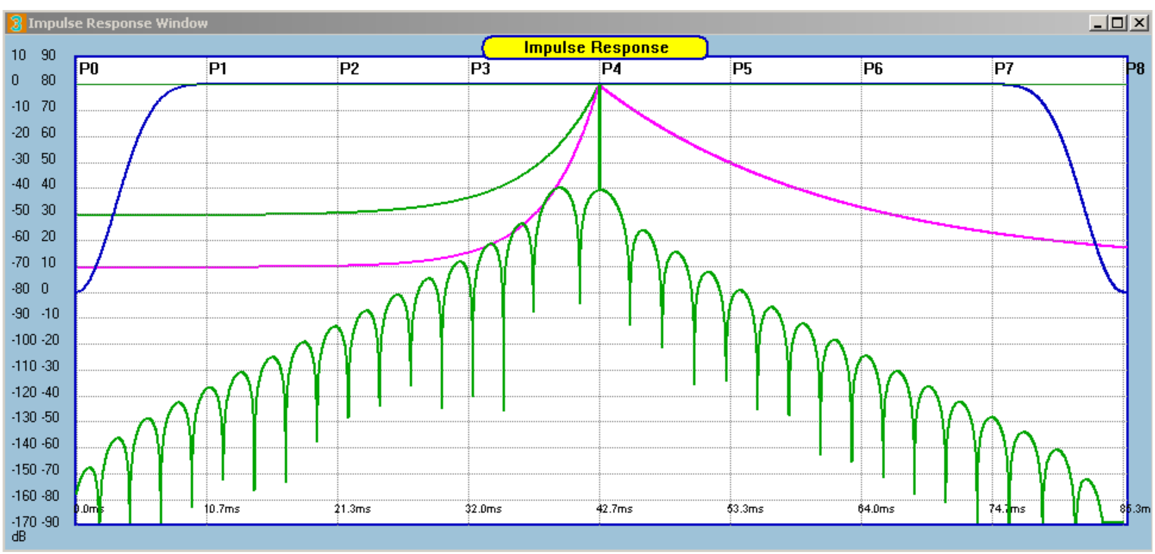




The next example is a simulation only, and shows an impulse response of a 200Hz crossover filter built using +48dB/oct Butterworth filter and -48dB/oct Butterworth filter, where the tweeter is delayed by 150 samples at 48kHz. Therefore the woofer peaks at -3.125msec to the left of the tweeter impulse response – this is clearly visible on the figures below.



This filtering arrangement would violate the pre-ringing limits of our stringent pre-masker of -70dB – see below.

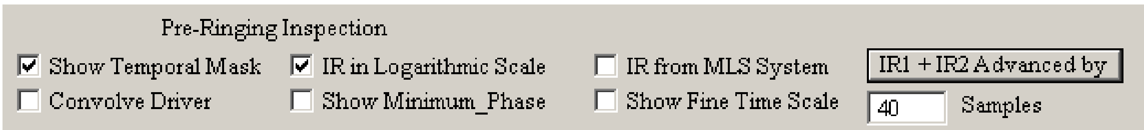


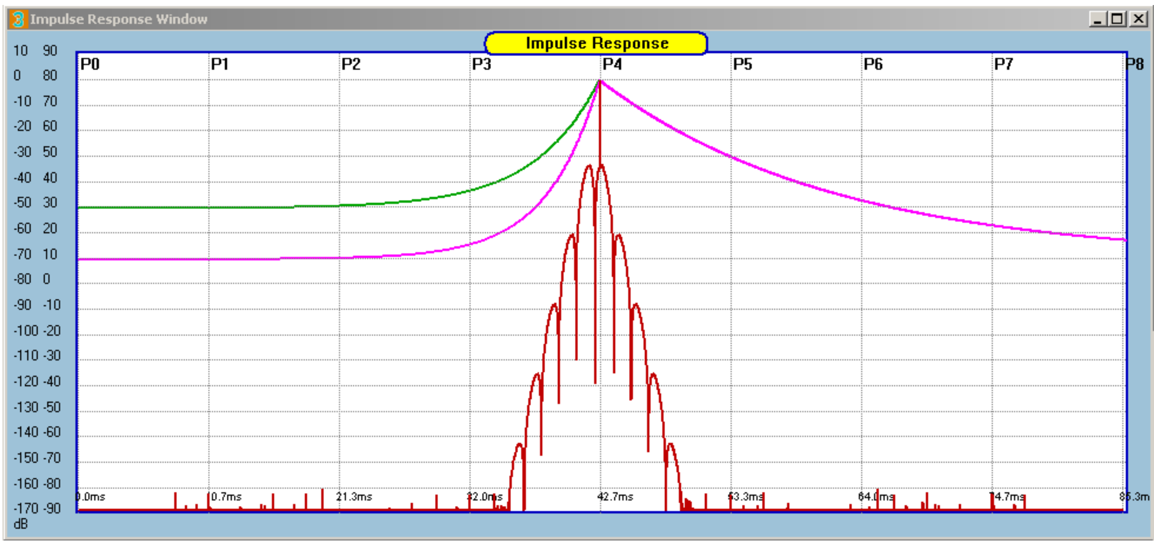
However, in terms of the location difference, this situation amounts to approximately ~1meter difference between woofer and tweeter arrival distance. This would be highly unusual to measure the system where the microphone is 1 meter closer to woofer than to the tweeter.

Finally, a simulation of a typical 3-way system, where the crossover frequencies are selected as 500Hz and 5000Hz. The filter if 24dB/oct LR design. The sampling rate is 48kHz, and impulse responses are shown in decibels.

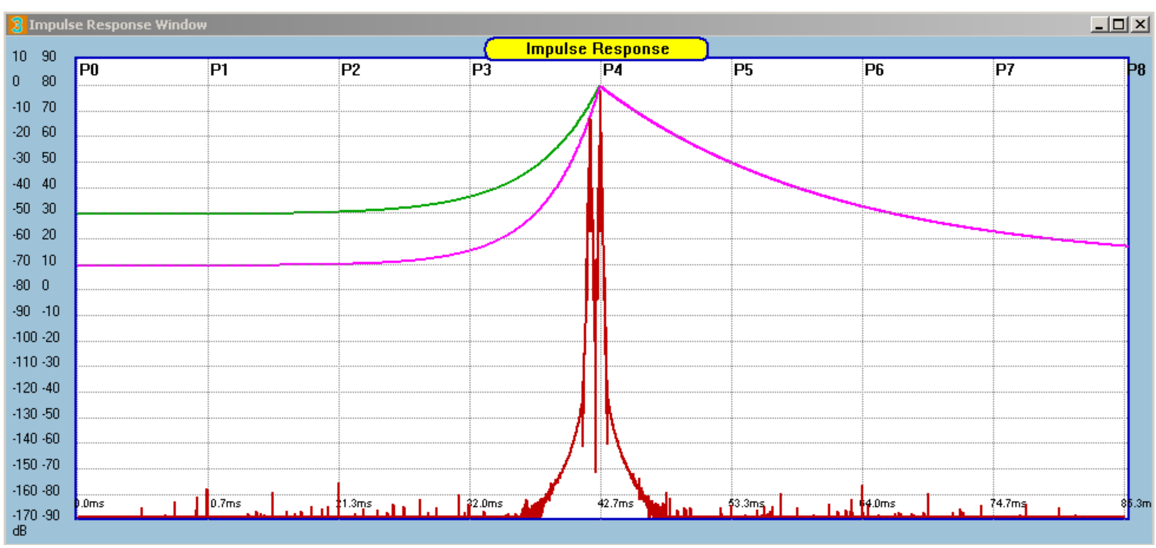
Time-of-flight distance difference between drivers is as follows:

woofer – midrange = 28.6cm = 40 samples @ 48kHz
midrange – tweeter = 28.6cm = 40 samples @ 48kHz.





Woofer and midrange



Midrange and tweeter

Interestingly, the short analysis above indicates, that midrange and tweeter will have less margin (well, almost none) under the stringent -70dB pre-masker.

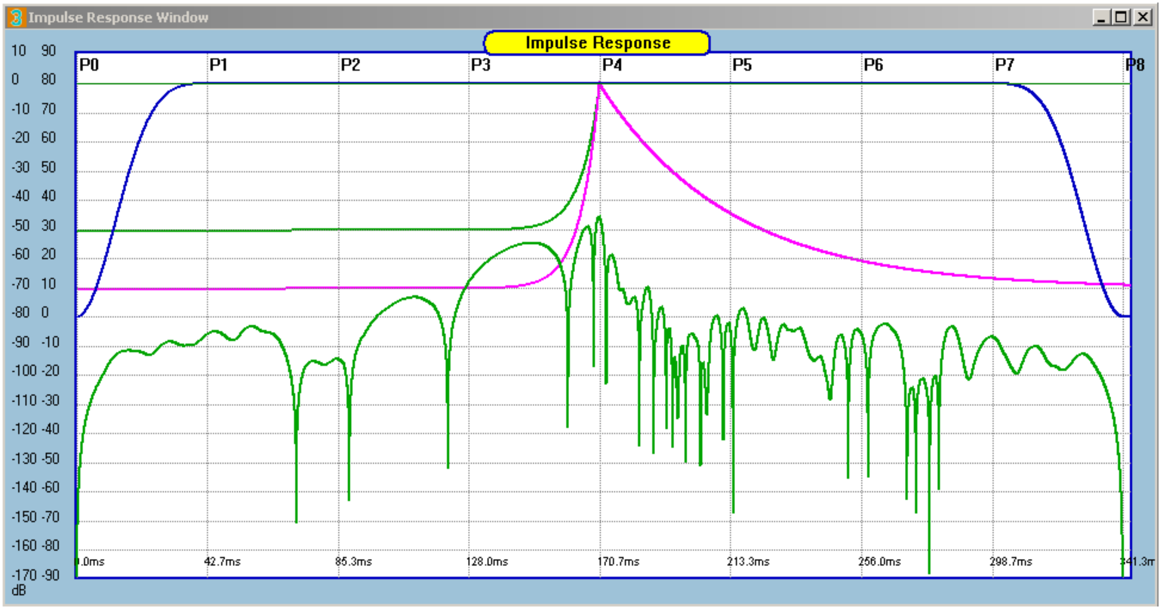
Equalized subwoofer.

Since the pre-masker and post-masker are time-limited phenomenon, it may be prudent to examine the slowest and heaviest driver in the system – the subwoofer. The driver in this example is really big, 18” McCauley subwoofer, mounted in a 300 litre venter enclosure. The equalization and filtering transfer function is depicted on the figure below.

Correction filter for subwoofer.

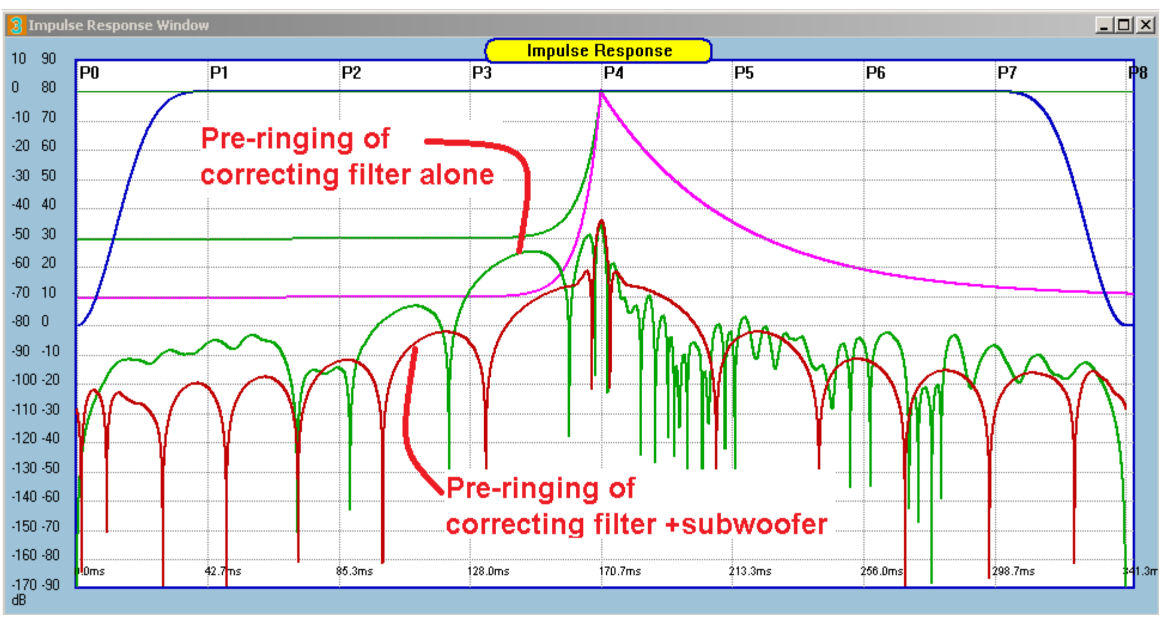
It is noticeable, that the phase response of the correction filter is inverse. This is because we are developing a linear-phase subwoofer. Also, the correction characteristics are evident below 340Hz, as the filter transfer function (-24dB/oct, LR filter) is also correcting driver’s SPL/phase below 340Hz.

Impulse response of the correction filter is shown below. Please note, that the filter alone has developed significant pre-ringing and clearly fails the -70dB pre-masker level. It is also observable, that the impulse response is very asymmetrical, not what you would expect from a linear-phase system.



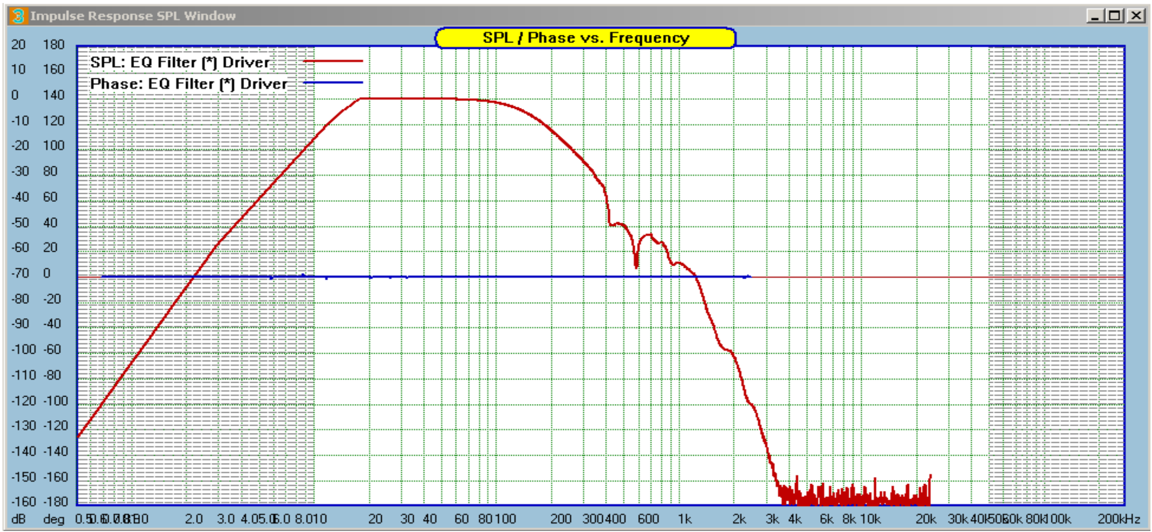
Impulse response of the correction filter (green) in decibels, 10dB/div.

This is a very common misconception, when discussing linear-phase filters. Simply because this is **not the subwoofer channel impulse response** – it's missing the actual subwoofer transfer function. Now, we can convolve the correcting filter with the actual subwoofer and then examine the resulting subwoofer channel impulse response – this is what you will be listening to.



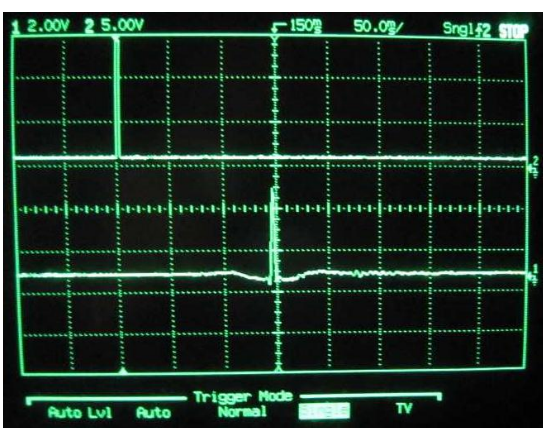
Pre-ringing of the complete subwoofer channel (red).

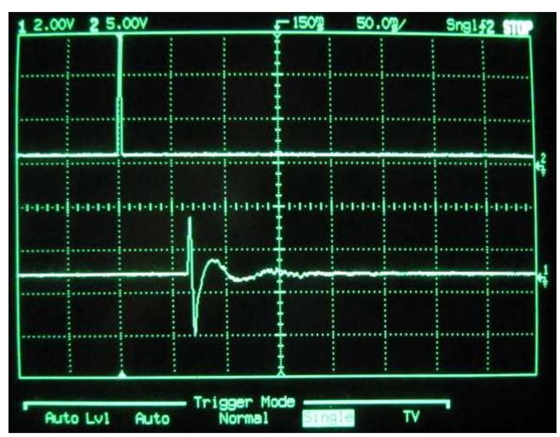
As shown on the figure above, the linear-phase impulse response (red) is now almost exactly symmetrical, and the pre-ringing has dropped by 10-20dB. This is very significant improvement, and now the total channel impulse response fits comfortably under the very strict **-70dB pre-masker and rolls-off towards timing extremes**. And here is the complete transfer function of the subwoofer channel. As you would expect, the SPL is equalized right up to 340Hz, and the phase response is now a flat line.



Complete transfer function of the subwoofer channel.

Next, we can perform some measurements on the newly equalized subwoofer.2ms-wide pulses separated by 350ms space were used as the source signal. On the 2ms pulse, the minimum-phase subwoofer 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 and far exceeded the 30ms “memory effect” of the auditory system. Here, the driver, filter and vented enclosure added it’s own, combined signature. It is also observable, that the minimum-phase version of the subwoofer has converted the clearly asymmetrical pulse into a much more symmetrical bi-polar pulse with post-ringing. This clearly visible on the screen shots below





2ms Impulse in Linear-Phase Mode and Minimum-Phase Mode

And here is the frequency and phase responses of the subwoofer.

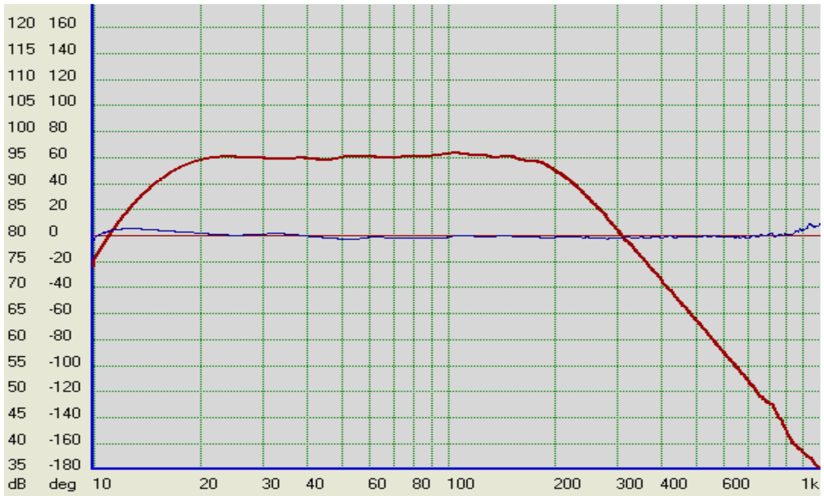


Figure 22. Frequency and phase responses of the equalized subwoofer.

The above level of performance was accomplished with the low-frequency resolution of 5.86Hz (Buffer 1024 and 48kHz sampling). When the “raw” SPL and phase do not contain rapid peaks and valleys, it is clearly possible to equalize the low-frequency driver to excellent standard.

In conclusion, pre-ringing as shown by modeling and a number of presented measurements, is typically predictable and manageable.

1. On-axis it does not exist.

2. Off-axis, in most applications, it is of such a short nature, that it is inaudible in typical domestic Hi-Fi application.

3. Filtering slopes of LP and HP filters in crossovers are in order of 12dB/oct or 24dB/oct. These numbers are quite low and such filters do not have extended pre-ringing.

4. In assessing pre-ringing, you must evaluate impulse response of a complete channel (filter + driver), rather than filter alone.

5. Extreme filtering arrangements or loudspeaker mutual mounting arrangements can lift it’s level above pre-masker attenuation. In this case, your system may need simple redesign.

Thank you for reading.
Bohdan