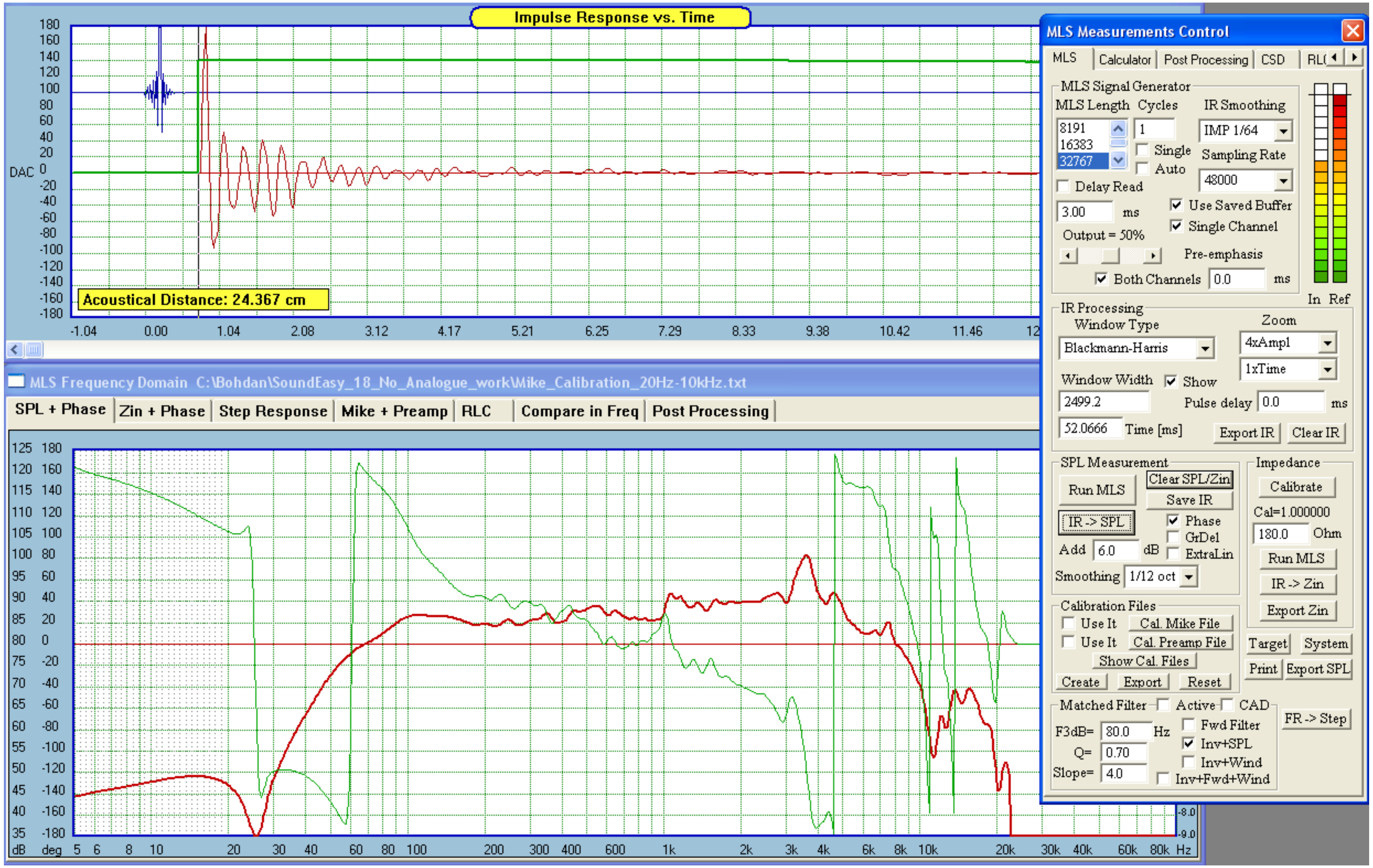
Approximate determination of minimum-phase phase response of measured loudspeaker

Correct measurement of loudspeaker phase response is a difficult process. The summary below offers some guidance for determining minimum-phase response of a loudspeaker, so that time-of-flight can be calculated for this loudspeaker, and therefore the location of acoustic centre can be established.

We start with an MLS measurement of a 12" guitar speaker SPL and phase responses – see below. **Is the measured phase correct?.**



Some ideas regarding loudspeaker's phase and AC were presented by Richard Heyser many years ago.

Determining the Acoustic Position for Proper Phase Response of Transducers*

RICHARD C. HEYSER

Tujunga, CA 91042, USA

Sound-path delay has been accurately removed from the measured phase response of a transducer when the derivative of phase response with respect to frequency approaches zero as the frequency increases without limit.

The residual nonminimum-phase all-pass component of the transducer response could only be separated from the all-pass component due to propagation time delay by investigating the asymptotic phase slope at frequencies outside the passband, which is the single procedure we are recommending in this correspondence.

REFERENCES

[1] R. C. Heyser, "The Delay Plane, Objective Analysis of Subjective Properties, Parts I and II," *J. Audio Eng. Soc.*, vol. 21, pp. 690–701 (1973 Nov.); pp. 786–791 (1973 Dec.).

[2] R. C. Heyser, "Loudspeaker Phase Characteristics and Time Delay Distortion, Parts I and II," *J. Audio Eng. Soc.*, vol. 17, p. 30 (1969); p. 130 (1969).

[3] R. C. Heyser, "Determination of Loudspeaker Signal Arrival Times, Parts I–III," *J. Audio Eng. Soc.*, vol. 19, pp. 734–743 (1971 Oct.); pp. 829–834 (1971 Nov.); pp. 902–905 (1971 Dec.).

This concept was later reviewed and expanded by Charlie Hughes.

http://www.excelsior-audio.com/Publications/Phase_Response_&_Receive_Delay.pdf

EXCELSIOR AUDIO DESIGN
& SERVICES, LLC

2130 Shadwell Court
Gastonia, NC 28056

www.excelsior-audio.com

704.675.5435 Tel
704.675.5436 Fax

Audio & Acoustical Consulting, Design and Measurement

Charlie Hughes
August 1, 2007

Phase Response & Receive Delay

In this example, a 1 kHz corner frequency is selected as the high frequency limit to illustrate a concept detailed in a paper by Richard Heyser[1]. This two page paper is highly recommended reading. Its brevity greatly belies its substance.

Heyser stated that "*If the value of time delay in the measuring medium is to be subtracted in the measurement process, as can be done for impulse, cross spectrum and TDS measurements, then the proper value of time delay has been subtracted, and the true acoustic position for the phase measurement has been obtained, when the plot of phase angle versus frequency approaches a flat horizontal line for frequencies well outside the pass band of the DUT.*" In other words we have removed the correct propagation delay when the phase response in the frequency region well above the cutoff frequency of the DUT approaches a horizontal line. This is exactly what is beginning to occur in Figure 5 at approximately 8 kHz and higher. The phase response is asymptotically approaching a horizontal line.

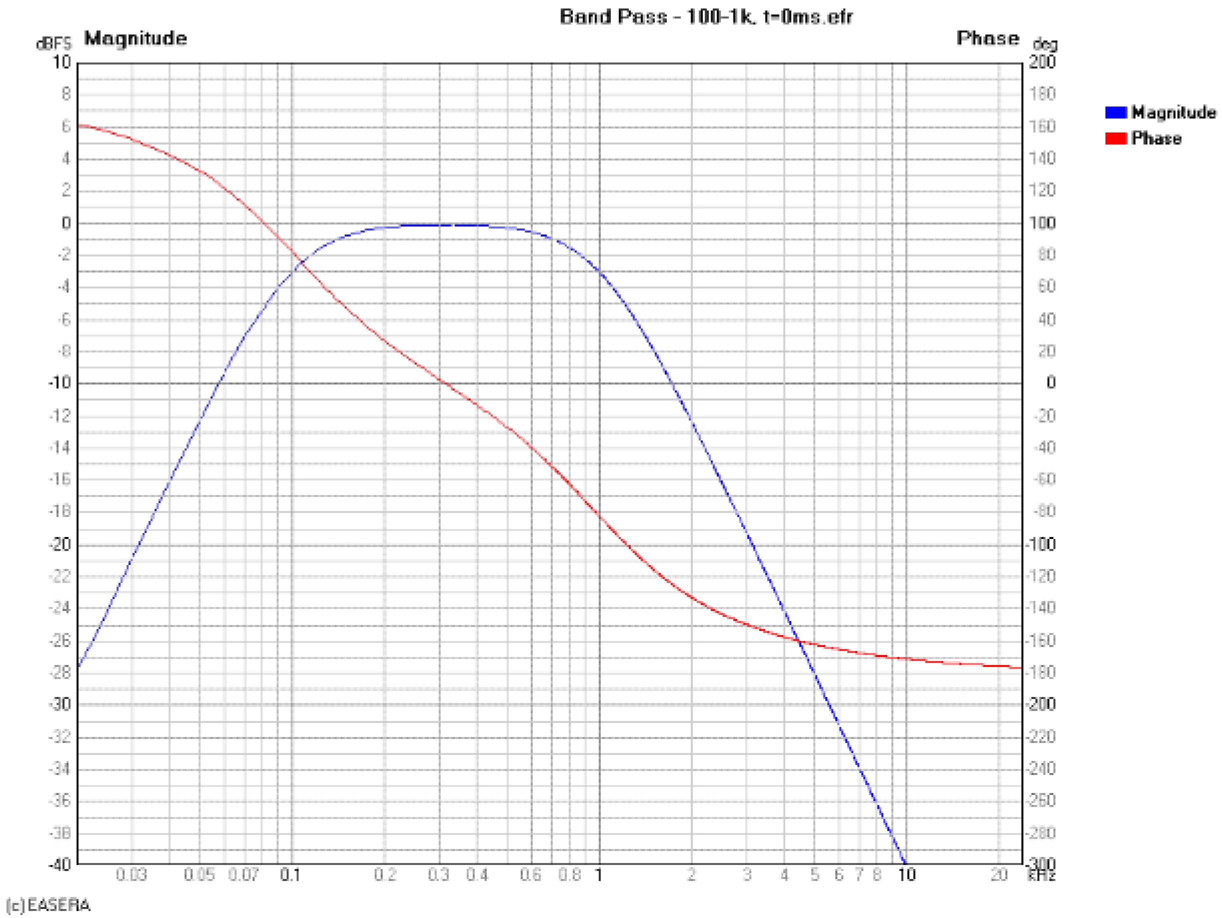
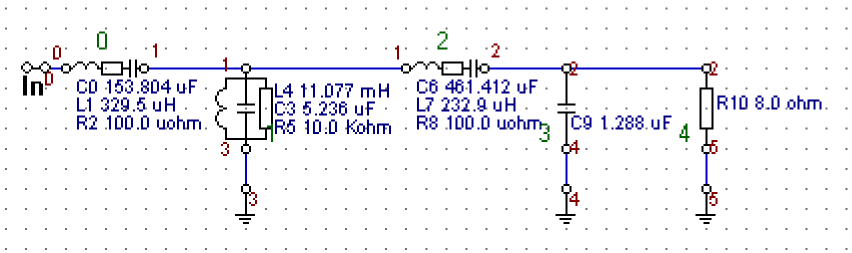


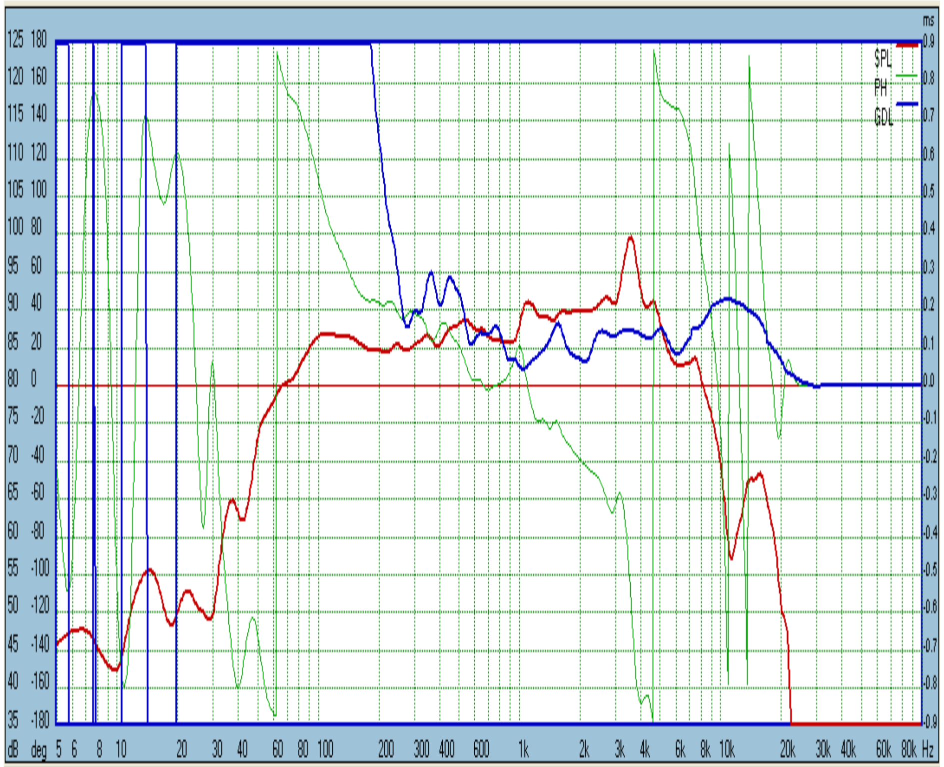
Figure 5 – Magnitude & phase response of a loudspeaker with 100 Hz – 1 kHz pass band

This is an excellent idea, but we must be evaluating the phase response at frequency far beyond the pass-band. For instance, for the 1kHz pass-band, we must examine the phase at 8 kHz. – 8 times the pass-band frequency. For a woofer with 5kHz pass-band, we'll end up looking at phase at 40kHz. For 20kHz tweeter we must examine the acoustical phase at 160kHz. This is impractical.

Another idea – examination of group delay of a "comparable" electrical filter.

Filter: +18dB/oct at 85Hz,   -24dB/oct at 6kHz -> GD = 0.1ms





It is difficult to compare electrical filter group delay to loudspeaker's group delay. There are too many ripples in the loudspeaker's group delay, making the idea of using electrical filter's group delay as a guidance impractical.

Now, let's compare measured loudspeaker's phase response with an "equivalent" electrical filter's phase response (blue curve). We can do this, as the loudspeaker is essentially a band-pass filter.

It is observable, that electrical filter approximates loudspeaker SPL very well, but the expected phase transition at 6kHz happens at 4.6kHz for the loudspeaker.

Sampling = 48kHz, Single channel, FFT window Bin = 86,

Butterworth Filter: +18dB/oct at 85Hz, -24dB/oct at 6kHz

If we re-measure the loudspeaker shifting the start of the FFT window closer to the impulse response by one bin, the 4.7kHz phase transition moves too much to the right – now at 7kHz.

FFT window Bin = 87 (20.83usec later = 1/48000)

Adding a little delay to last measurement, brings the loudspeaker phase transition at around 6kHz.

FFT window Bin = 86 - 15.0usec subtracted from “time-of-flight”. Sample time = 20.83usec

If we now use HBT (Hilbert-Bode Transform) with the same slopes as the electrical filter, we can get **excellent agreement for the phase transitions at around 6kHz**. The HBT "LP start" frequency was chosen to avoid driver's break-up region, clearly visible above 10kHz. In the break-up region, the loudspeaker driver looses it's minimum-phase characteristics.

The HBT allows you to "clean-up" measured plots by removing low-level, low-frequency noise, and undesirable break-up region from the measured curves. It also helps to validate the correctness of the measured phase response.

FFT window Bin = 86 - 15.0usec delay, (sample time = 20.83usec)
HBT: +18dB/oct at 47Hz, -24dB/oct at 10kHz

Now, we can compare all three phase responses: measured loudspeaker (with 15usec delay), electrical filter, and HBT phase response.

We can conclude, that all three phase responses are in excellent agreement – phase response was measured correctly (time-of-flight has been removed) and it is the best approximation of minimum-phase response of this loudspeaker.

1. FFT window Bin = 86 - 15.0usec delay, (sample time = 20.83usec)
2. Filter: +18dB/oct at 85Hz, -24dB/oct at 6kHz
3. HBT: +18dB/oct at 47Hz, -24dB/oct at 10kHz

For further design purposes, the HBT-calculated SPL and phase should be used.

We must avoid situations like the one depicted on the picture below – phase response measured wrongly. Without any reference (like electrical filter, or HBT), we would not even know about it.

FFT Window is way too close to IR peak.

Dome tweeter example

In the second example we will examine minimum-phase response of a popular Hi-Fi dome tweeter driver.

The dual-channel MLS system is actually designed to provide minimum-phase response of the measured driver, within the error of **+/- one sample time**. Here is how it works.

When you perform loop test, you will notice, that you will get flat phase response of the signal channel when you place the start of the FFT window at 10 samples before the peak of the impulse response – why?. This is because the reference channel is also automatically windowed with the fixed start of the FFT window also at 10 samples in front of the IR start. The loop test simply measured the true "minimum-phase" phase response of the sound card. However, each PC MLS system must be examined individually for the Reference Impulse response first.

Reference Impulse Response:

MLS Impulse Response Ref=41463.95, In=-95.36, Bin=60, Scroll[0 - 2000] Peak of 41463 at bin=60

MLS Impulse Response Ref=8584.96, In=1.29, Bin=59, Scroll[0 - 2000] Peak -1 of 8584 at bin 59.

MLS Impulse Response Ref=-3762.23, In=-1.57, Bin=58, Scroll[0 - 2000] Peak - 2 of -3762 at bin 58, the IR has gone large negative now.

For this system, bin 59 is the start of the impulse response.

Therefore, the start of the FFT window for the Reference impulse response is 10 sample times from the peak, or **9 sample times from the start of the impulse response.**

Now, we can apply the same technique to the loudspeaker measurement, and place the start of the FFT window 9 samples ahead of the start of the IR – and we'll obtain minimum-phase phase response of the loudspeaker straight away, with +/- one sample time error. To eliminate this small uncertainty error, we have to add/subtract small delay (or manipulate HBT slopes) to get the measured and HBT calculated phase into alignment.

It is important to determine the **start of the impulse response** (not the peak), as various drivers have different rising slopes of the impulse response, therefore the peak will be located at various distances from the start of the impulse. So, first we need to find the peak of the impulse response:

MLS Impulse Response Ref=-7.84, In=1886.90, Bin=339, Scroll[0 - 2000] the peak is **In = 1886.90, located at Bin=339**

Next we move the cursor to the left of the impulse response, one sample time, and each time we monitor the “In” vale.

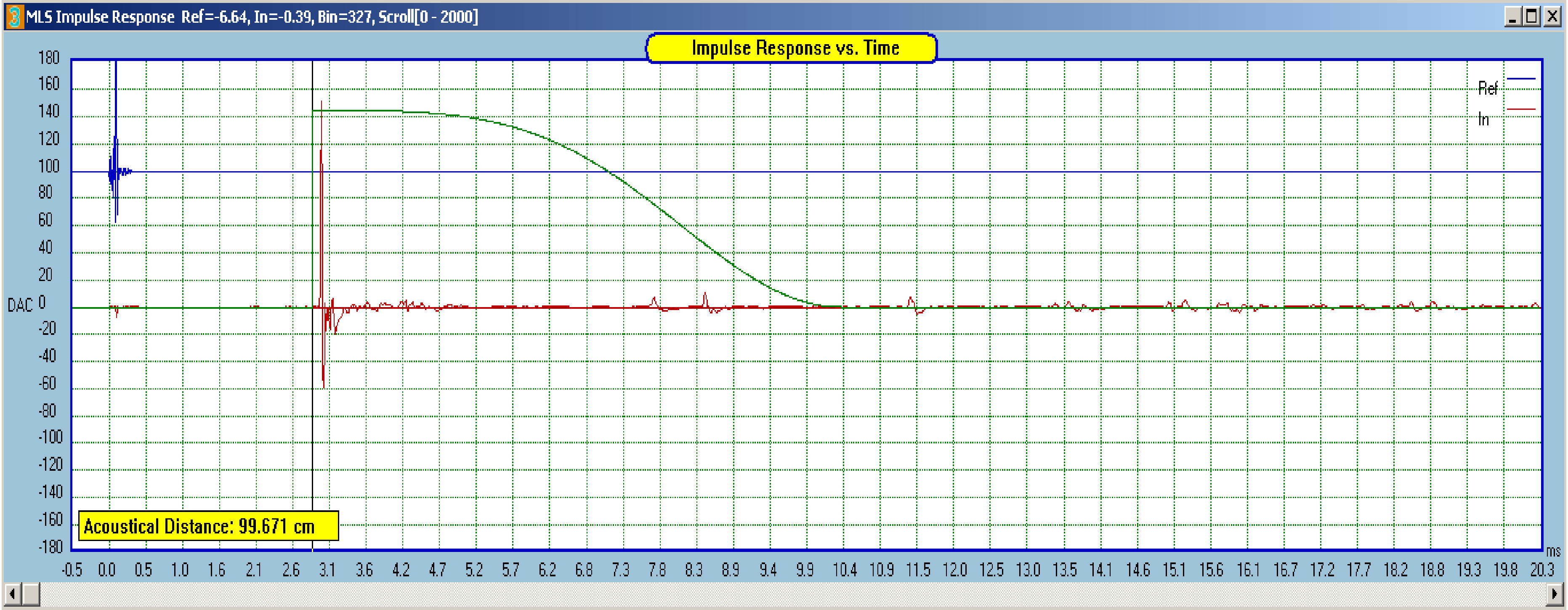
MLS Impulse Response Ref=-7.04, In=987.70, Bin=338, Scroll[0 - 2000]

MLS Impulse Response Ref=-14.45, In=147.21, Bin=337, Scroll[0 - 2000]

MLS Impulse Response Ref=-9.20, In=4.94, Bin=336, Scroll[0 - 2000] the In=4.94 and is very close to zero, therefore, we determine that the start of the impulse response is Bin=336.

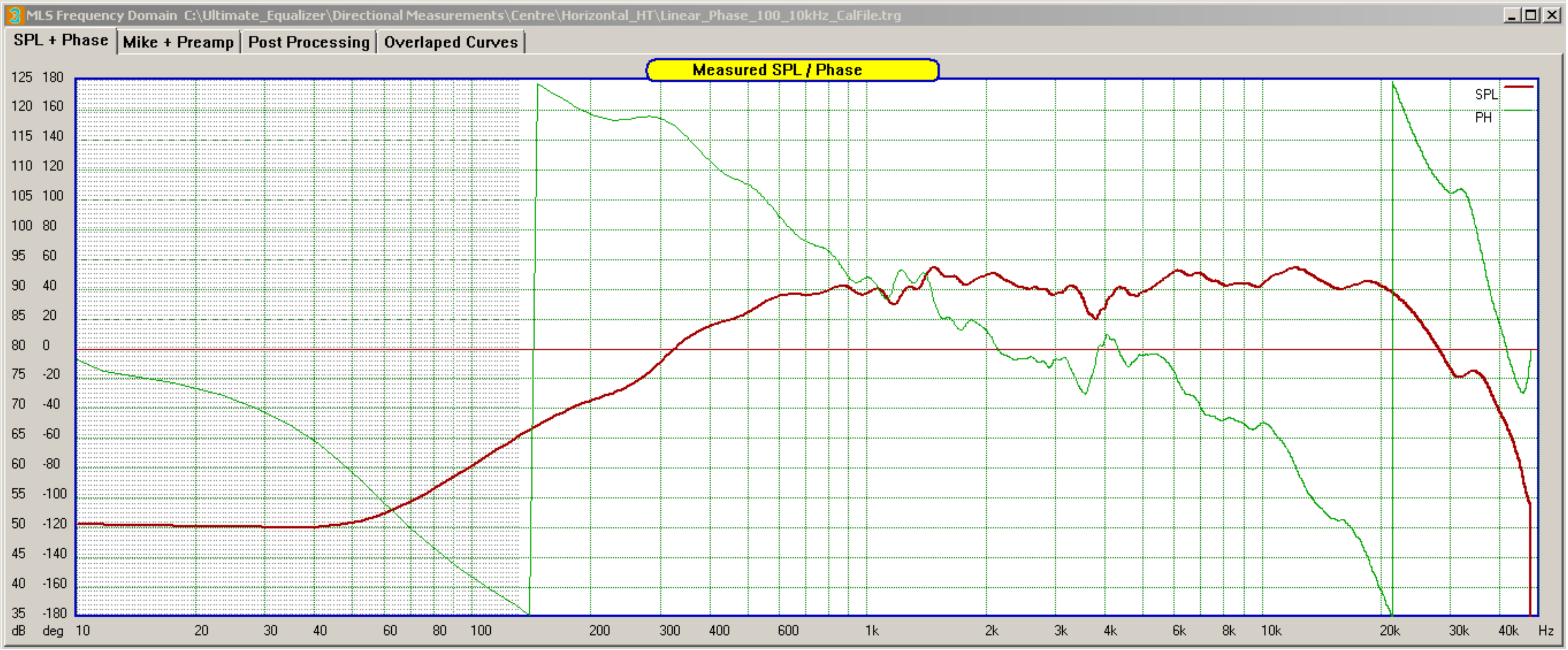
Finally, we need to move the start of the FFT window 9 sample times (for this system it is 9 sample times) to the left from the start of the impulse response, 336-9 = 327.

Now, the start of our FFT window is located at Bin = 327. See figure below.



Next, we need to obtain the SPL and phase of the driver using FFT.

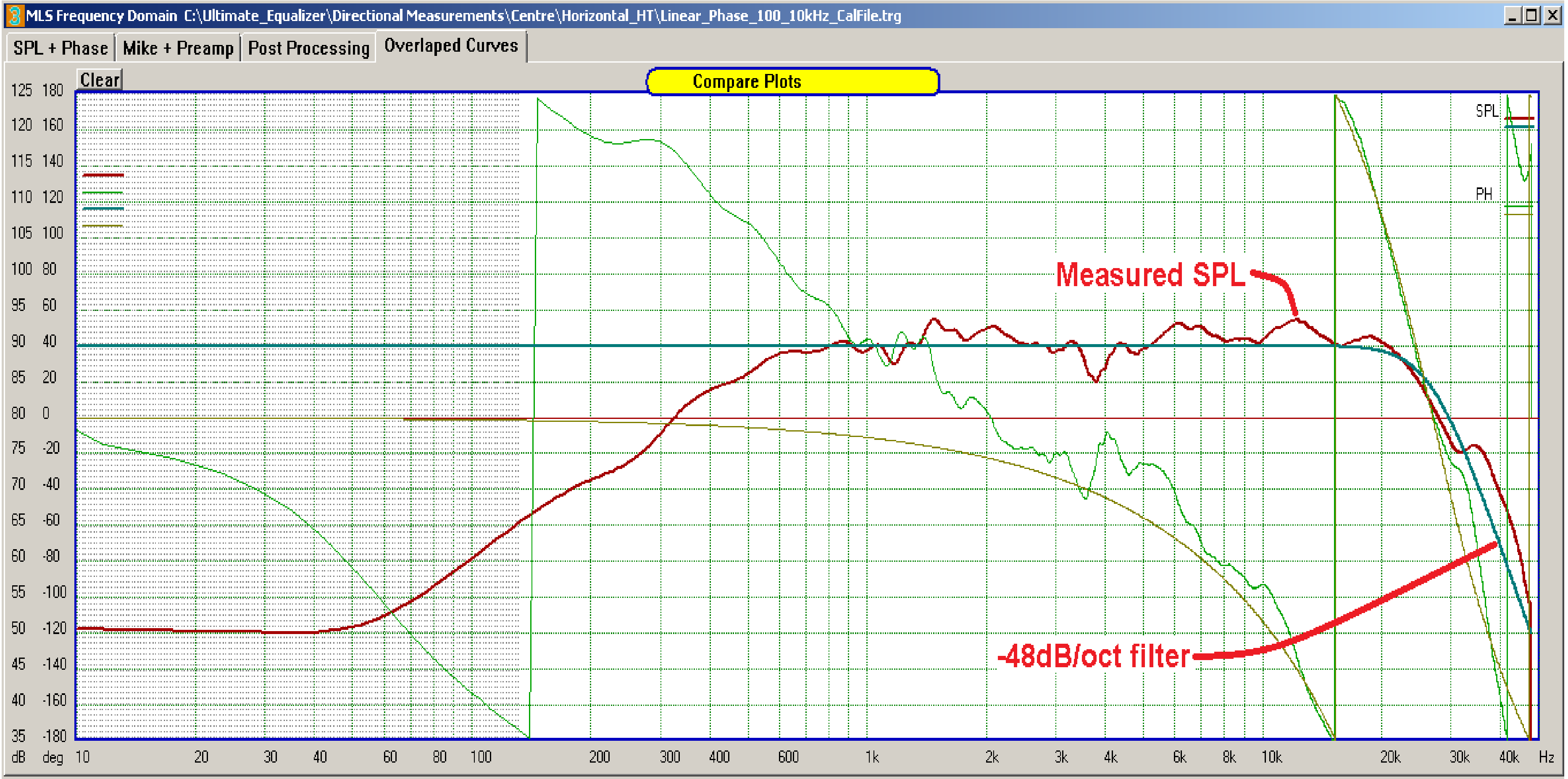
And here is the result.



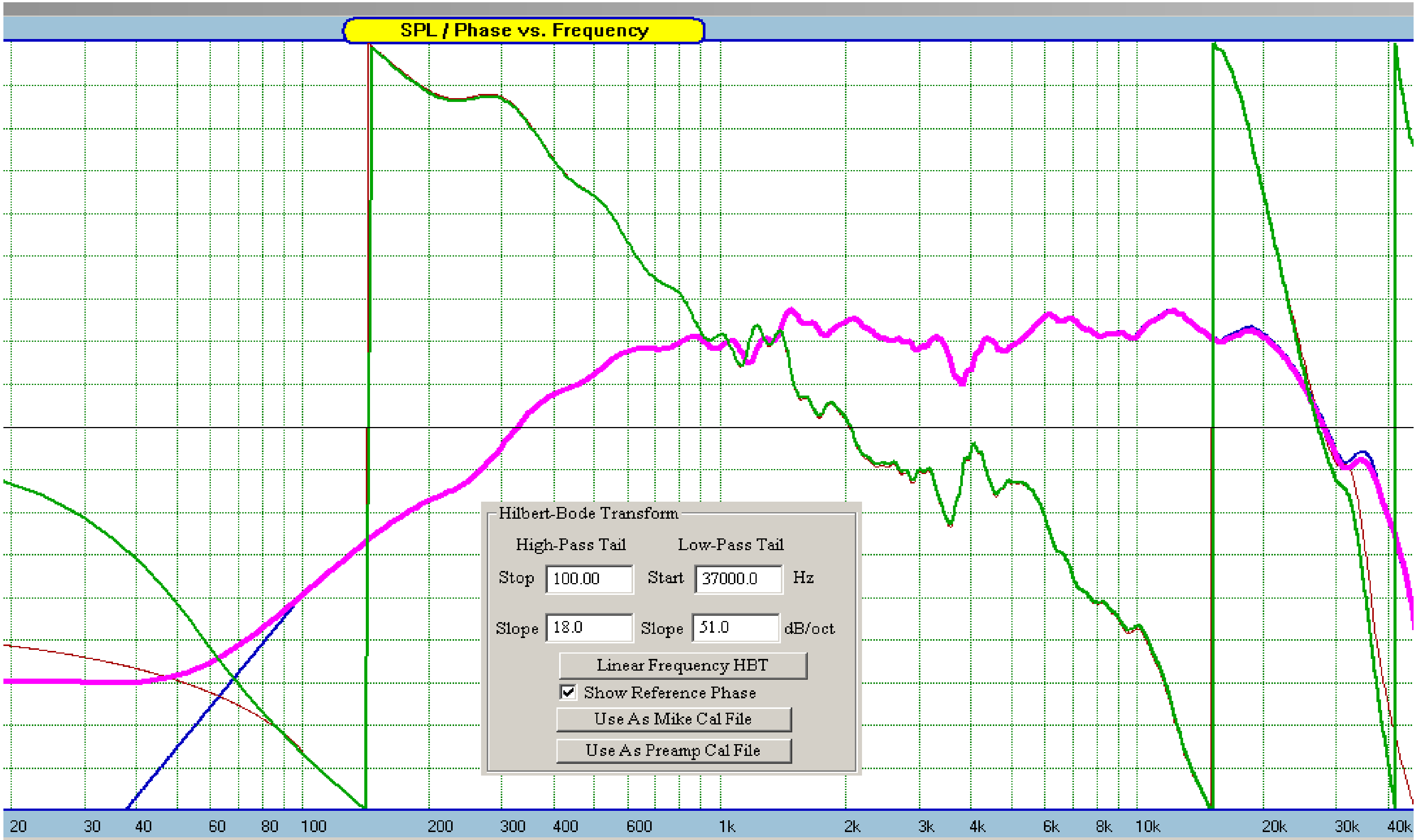
To increase the level of confidence on the phase response, I would also suggested to use a band-pass filter, comparable with the loudspeaker amplitude response. This will give you filter's phase response, which you would use as an additional guidance for the locations of the 360deg transitions of the filter and the measured phase – they should be very close.

Since this is a tweeter example, and we are only interested in finding the high-frequency tail, I use simple low-pass filter located at 27kHz with -48dB/oct slope. It is observable, that the slope of the filter is slightly slower than the measured response, so we assume **-51dB/oct as the asymptotic slope** of the measured driver SPL. Also, the 360deg transitions of both: filter and the measured SPL need to be very close.

In order the get the measured phase response phase 360deg transition to overlap filter's phase transition, a small, **12usec delay was added to the measured SPL curve**.



We can now run HBT with the high side asymptotic slope of 51dB/oct, to see how the whole picture works out.



We observe a perfect alignment of measured and HBT-derived phase responses assuming -51dB/oct asymptotic slope of the "guiding filter".
HBT SPL – blue curve
HBT Phase – red curve

There is one other bonus of the MLS measured phase – it's quite accurate at low frequencies. Measurements indicate, that dual-channel MLS system will give you minimum-phase (+/- 1.5deg error) below 200Hz straight away by placing the FFT window as described above.

Let's assume, that we have a typical 3-way system with crossover frequencies at 500Hz and 5kHz. Uncorrected phase error will be increasing with frequency, so how much phase error is equated to +/- one sample time at 5kHz?.

48kHz sampling: +/-38deg, too high, needs guiding filter and HBT correction described in this paper.
96kHz sampling : +/-18deg, slightly too high, needs guiding filter and HBT correction described in this paper.
192kHz sampling: +/-9deg, good enough for first-cut design
384kHzsampling: +/-4.5deg, good enough for first-cut design

Conclusion

It seems, that all procedures for determining phase response require some "eyeballing" of the plotted curves.

Heyser suggests visual inspection to determine where the phase response *approaches* horizontal line, so what does the "approaches" mean?.
The issue with Heyser method is that the frequency range of interest is located very high, and possibly out of range for many test setups, particularly within DIY community. Secondly, SPL at such high frequencies will be buried in environmental noise, so the phase response at such high frequencies will be very noisy, and often, it will be just noise anyway. But, if your measurement set-up is this good – go for it.

The procedure described in this paper places the focus on the frequency range wider than the band-pass range, but not wider than -30dB below the peak band-pass SPL. So, we are still confident, that we received good quality measured data points, due to good signal-to-noise ratio within the frequency range of interest.

Much of the uncertainty is associated with correctly choosing the approximating filter and HBT parameters. This is why the method is called an "approximation". However, with a bit of practice, you'll quickly learn if you have made a gross error. Remember, all three curves: measured phase, filter's phase and HBT phase must be in agreement.

In summary – Dual-channel MLS method (with correction for 48kHz and 96kHz, and no correction for 192and 384kHz), will provide sufficiently accurate minimum-phase phase response needed for first-cut design.