Automated Extraction of Minimum-Phase Phase response

By Bohdan Raczynski – July 2021

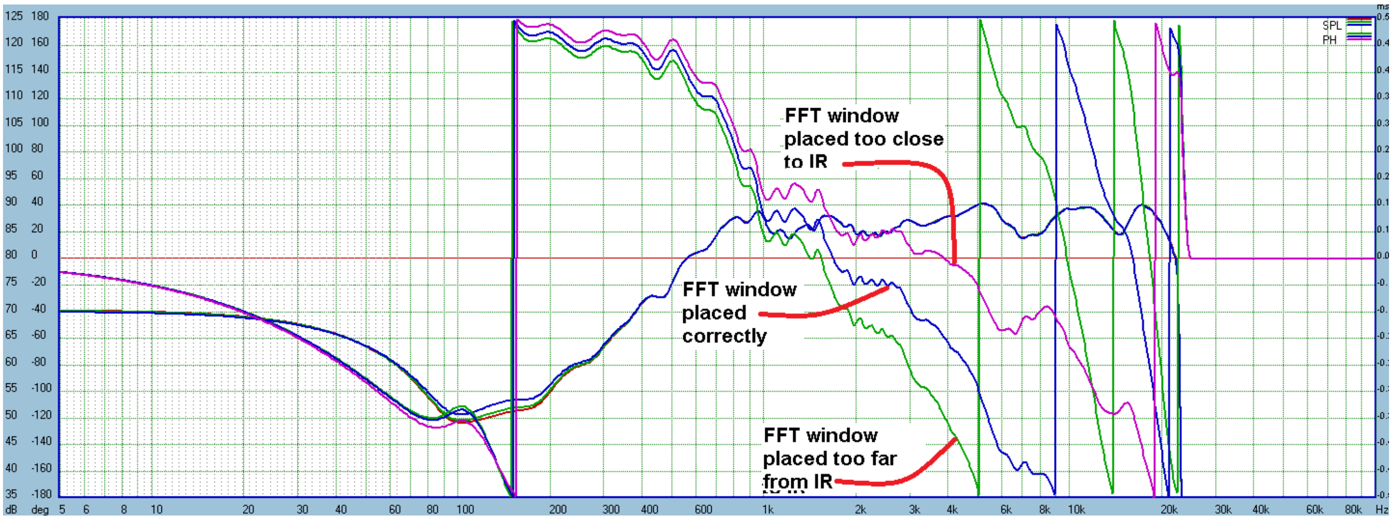
When measuring loudspeaker phase response, there is a strong possibility, that the phase will be measured erroneously. There are three distinct possibilities, typically obtained from the measurement:

1. The FFT window is placed correctly – this will result in subtracting the exact amount of time-of-flight and the "leftover" will result in **exact minimum-phase phase response**. This is the desirable option.

2. The FFT window is placed too close – this will result in erroneous phase response, which is too flat. The "leftover", which was supposed to represent the minimum-phase phase response is too shallow. Too much of the time-of-flight has been removed.

3. The FFT window is placed too far from the impulse response – this will result in the "leftover", which was supposed to represent the minimum-phase phase response, still containing small residual time delay.

The three options listed above are depicted on a measurement example below.



The previous paper introduced **Inverse Hilbert-Bode Transform** (IHBT) and provided extensive description of it's primary application. A manual method of extracting exact "minimum-phase" phase response from measured SPL curve was also given and explained in details.

URL: https://www.bodziosoftware.com.au/IHBT_White_Paper.pdf

The method required flipping between MLS measurement system screens and IHBT dialogue box several times (up to 20-30 times), before it produced the expected minimum-phase response.

This paper introduces automated method of extracting exact "minimum-phase" phase response from measured SPL curve. As the loudspeaker driver is a minimum-phase device, it's transfer function and phase response are mathematically locked together. Knowing SPL transfer function we can extract phase response from it. And conversely, knowing phase response, we can re-construct SPL transfer function from it. The re-constructed SPL transfer function will be identical to the original SPL transfer function – but ONLY if the supplied phase response was the "minimum-phase" phase response.

We can therefore supply a number of phase responses to the IHBT algorithm and compare resulting re-constructed SPL transfer functions with the originally measured SPL. The comparison is done by calculating squared error between the original and re-constructed SPL transfer functions.

Typical MLS and ESS measurement systems will both generate Impulse Response (IR) of the loudspeaker under test. The next immediate step is to apply an FFT windowing process to the recovered IR and then perform FFT function on the windowed IR. The outcome of this technique is SPL and phase response of a loudspeaker. The location of the start of the FFT window will determine the resulting phase response. The start of the FFT window can only be anchored at discrete time samples determined by signal sampling frequency. Typically, sampling will be set at 48kHz or 96kHz. For example, 48kHz sampling frequency will result in discrete time samples spaced 20.833usec apart. Therefore, corresponding phase response can only be obtained at those time samples.

Yes, our measurement system is quite coarse. The problem with this approach is, that the actual minimum-phase response may be corresponding to the FFT window placed somewhere between the discrete time samples. To overcome this shortcoming, a small positive or negative time delay can be added to the calculated phase response, and the resulting phase response is then supplied to the IHBT algorithm.

Clearly, the automated method described in this paper is a **two-stage process**. In Stage I, the FFT window is placed at 10 arbitrary, but consecutive time samples, and resulting phase responses are supplied to the IHBT algorithm. This way, we obtain 10 SPL responses and one of them will be the best match with the originally measured SPL. The time sample corresponding to this SPL curve is the starting point of Stage II.

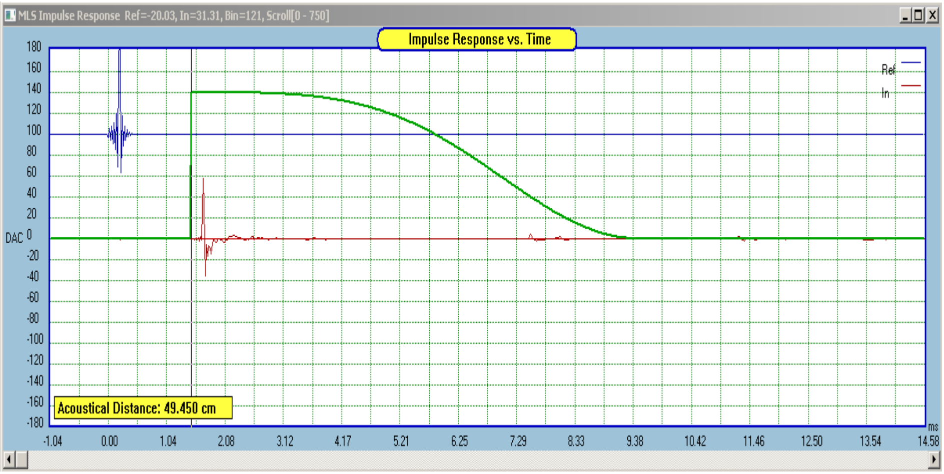
In Stage II, a small positive or negative time delay is added to the calculated phase response, and the resulting phase response is then supplied to the IHBT algorithm. Practically, the algorithm adds or subtracts ten 2usec time delays so that the new phase is recalculated across 20usec to the left and 20usec to the right of the sampling time determined from Stage I.

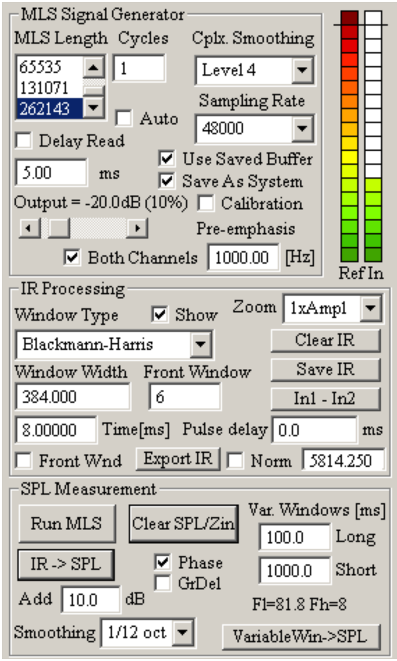
There are many options of selecting parameters driving the whole process, but for the sake of clarity of this presentation, they will not be discussed here.

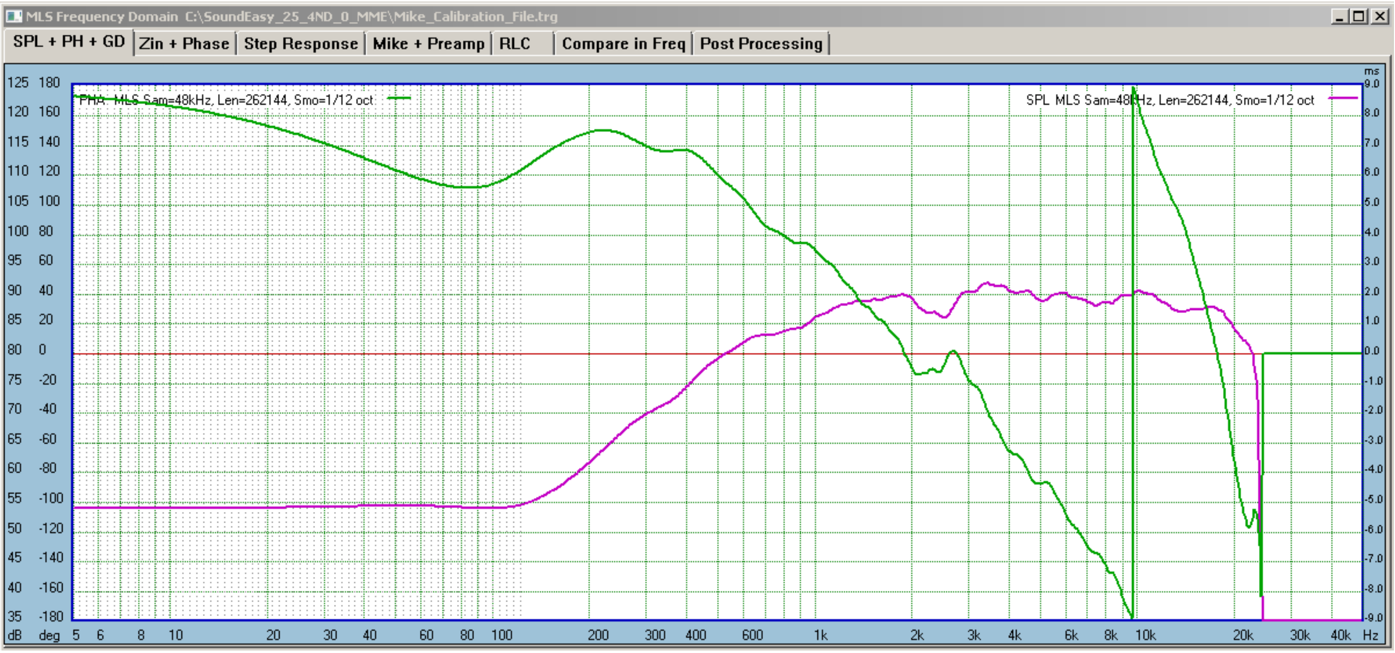
We are now in the position to present two examples of measured loudspeakers a 12” guitar driver in a vented box and a D25 tweeter. The method described above will be used to extract the exact minimum-phase phase response in both cases.

Tweeter Driver Example

From the phase extraction point of view, tweeter drivers are notoriously difficult to deal with, because there is no SPL/Phase data above the Nyquist Frequency (half the sampling frequency of 48kHz). An example of tweeter driver measured up to 24kHz is shown below. Phase response was derived with the FFT window placed at time sample "Bin 121".

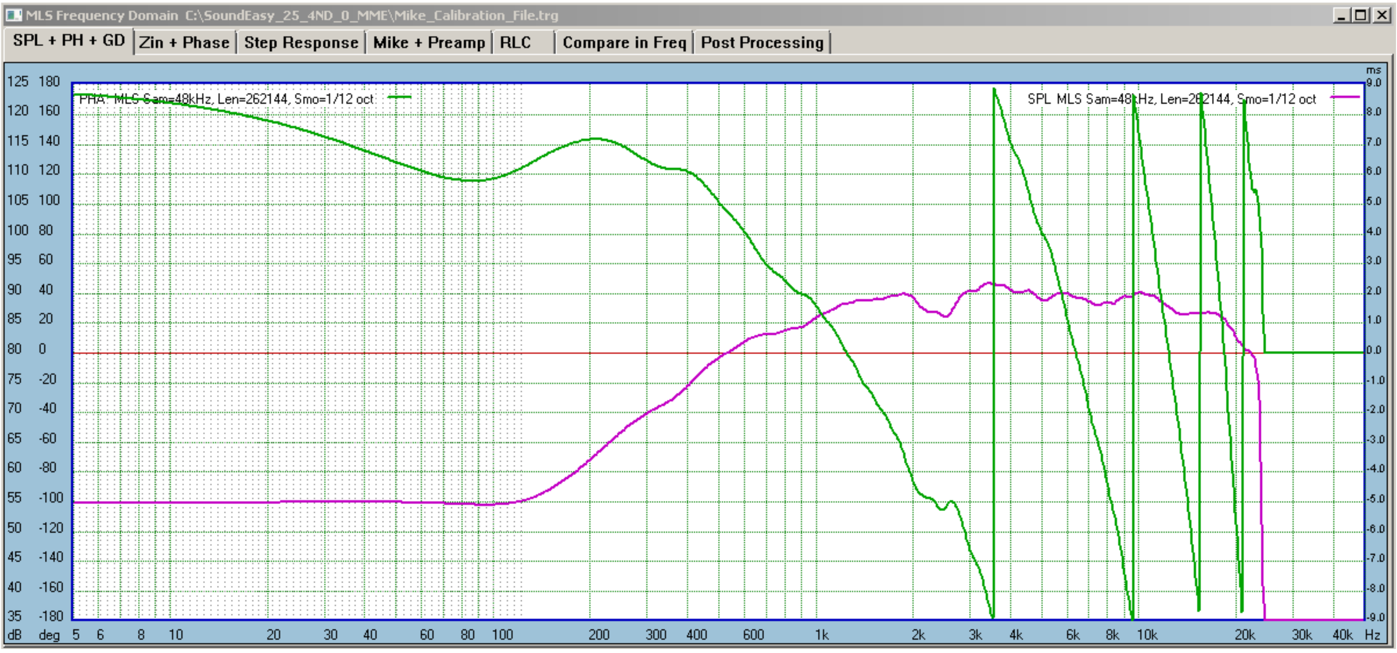




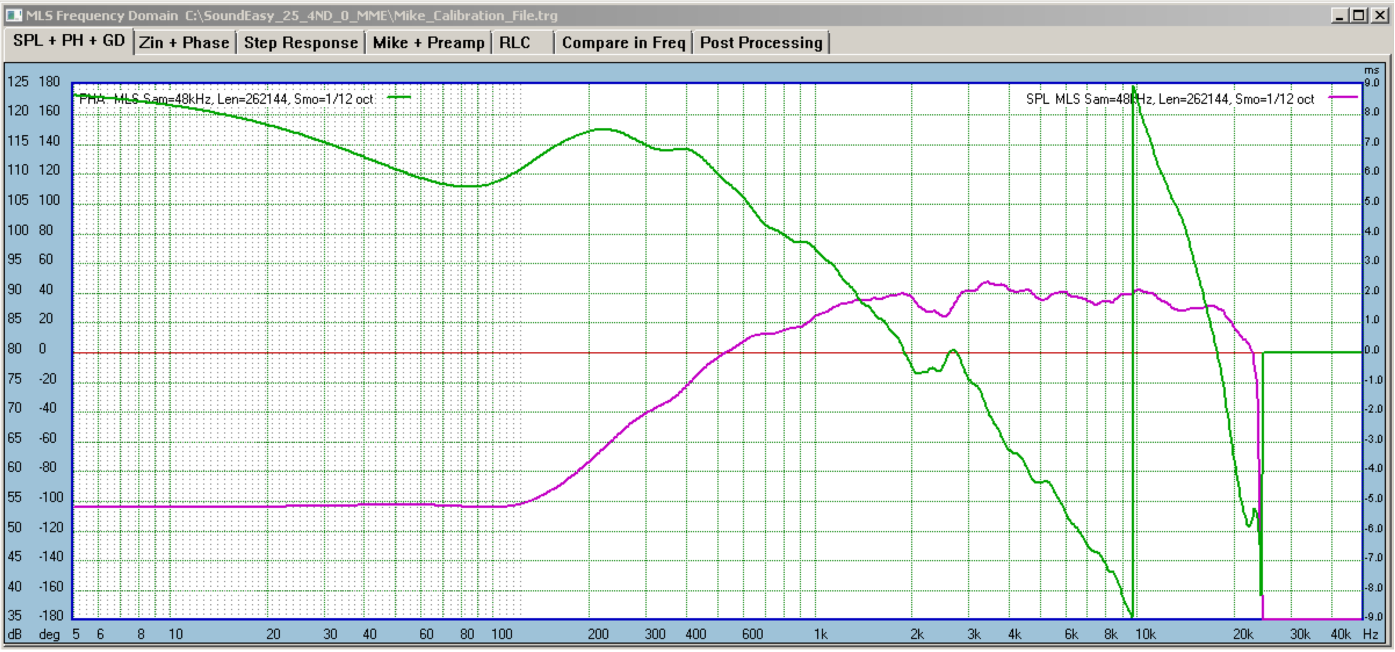


Now, we are grappling with the same question – **how to determine the correct FFT window placement**?. Shown below are three examples of FFT window placement, and corresponding phase response (green curves).

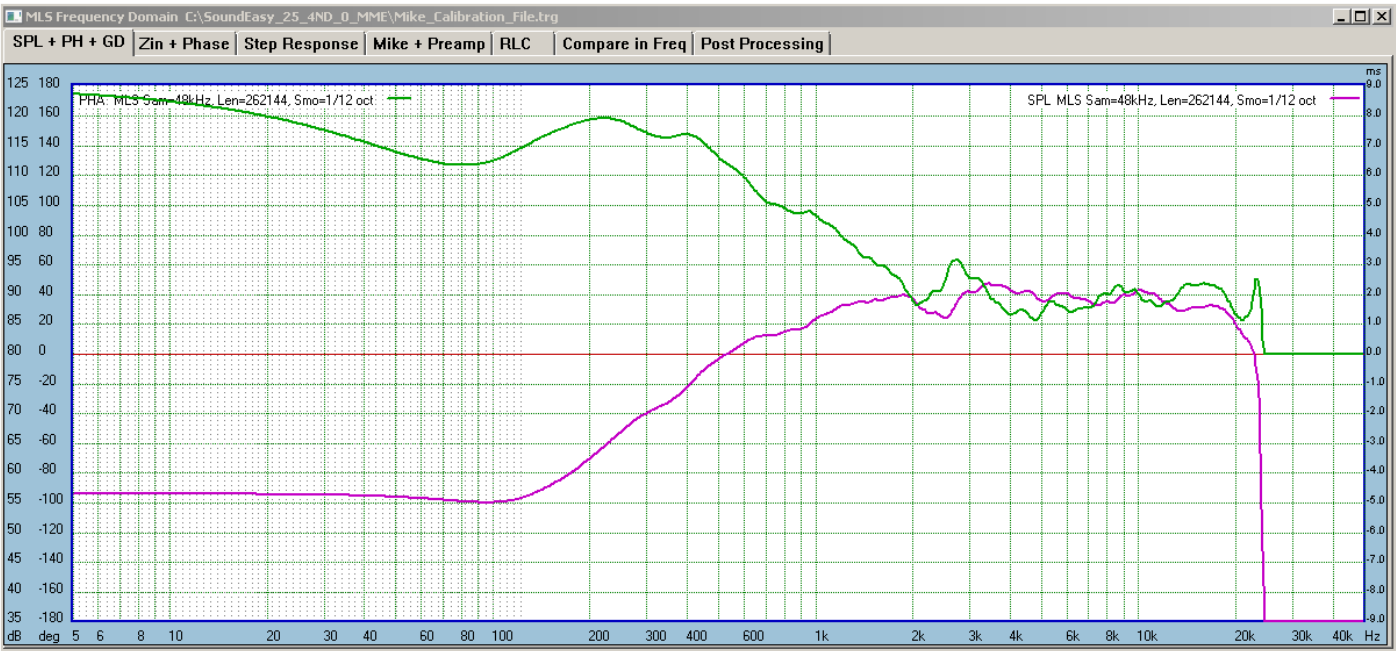
On the top picture, the FFT was placed at Bin 116, resulting in perhaps too many +180/-180 deg phase transitions. On the middle picture, the FFT window start was placed at Bin 121. Here, we only have one phase transition – but is this correct?. On the bottom picture, the FFT window was placed at Bin 124, resulting in phase response having no transitions at all.



MLS Frequency Domain C:\SoundEasy_25_4ND_0_MME\Mike_Calibration_File.trg
SPL + PH + GD
Zin + Phase
Step Response
Mike + Preamp
RLC
Compare in Freq
Post Processing
PHA MLS Sam=48kHz, Len=262144, Smo=1/12 oct
SPL MLS Sam=48kHz, Len=262144, Smo=1/12 oct
dB 125 120 115 110 105 100 95 90 85 80 75 70 65 60 55 50 45 40 35
deg 180 160 140 120 100 80 60 40 20 0 -20 -40 -60 -80 -100 -120 -140 -160 -180
ms 9.0 8.0 7.0 6.0 5.0 4.0 3.0 2.0 1.0 0.0 -1.0 -2.0 -3.0 -4.0 -5.0 -6.0 -7.0 -8.0 -9.0
5 6 8 10 20 30 40 60 80 100 200 300 400 600 1k 2k 3k 4k 6k 8k 10k 20k 30k 40k Hz



MLS Frequency Domain C:\SoundEasy_25_4ND_0_MME\Mike_Calibration_File.trg
SPL + PH + GD
Zin + Phase
Step Response
Mike + Preamp
RLC
Compare in Freq
Post Processing
PHA MLS Sam=48kHz, Len=262144, Smo=1/12 oct
SPL MLS Sam=48kHz, Len=262144, Smo=1/12 oct
dB 125 120 115 110 105 100 95 90 85 80 75 70 65 60 55 50 45 40 35
deg 180 160 140 120 100 80 60 40 20 0 -20 -40 -60 -80 -100 -120 -140 -160 -180
ms 9.0 8.0 7.0 6.0 5.0 4.0 3.0 2.0 1.0 0.0 -1.0 -2.0 -3.0 -4.0 -5.0 -6.0 -7.0 -8.0 -9.0
5 6 8 10 20 30 40 60 80 100 200 300 400 600 1k 2k 3k 4k 6k 8k 10k 20k 30k 40k Hz



MLS Frequency Domain C:\SoundEasy_25_4ND_0_MME\Mike_Calibration_File.trg
SPL + PH + GD
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RLC
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SPL MLS Sam=48kHz, Len=262144, Smo=1/12 oct
dB 125 120 115 110 105 100 95 90 85 80 75 70 65 60 55 50 45 40 35
deg 180 160 140 120 100 80 60 40 20 0 -20 -40 -60 -80 -100 -120 -140 -160 -180
ms 9.0 8.0 7.0 6.0 5.0 4.0 3.0 2.0 1.0 0.0 -1.0 -2.0 -3.0 -4.0 -5.0 -6.0 -7.0 -8.0 -9.0
5 6 8 10 20 30 40 60 80 100 200 300 400 600 1k 2k 3k 4k 6k 8k 10k 20k 30k 40k Hz

As explained in Part I article, the IHBT requires phase slopes to be attached to the phase response. There are several reasons for this.

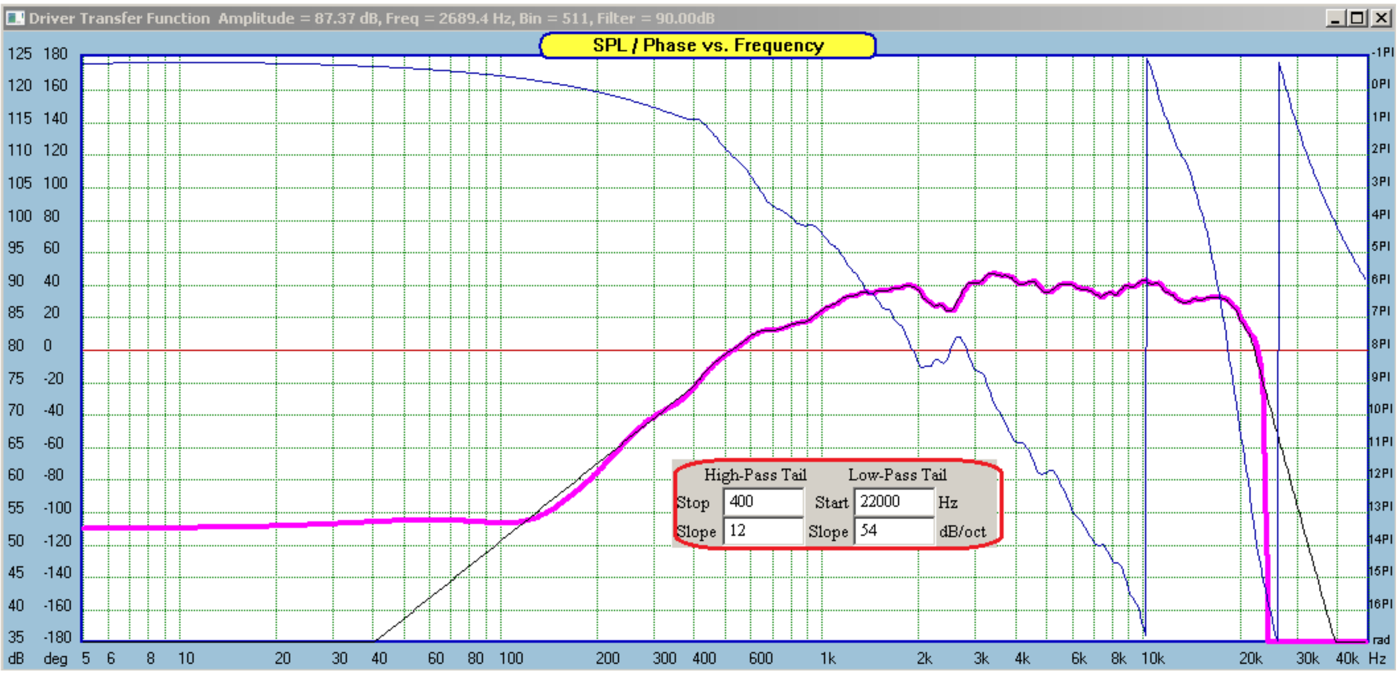
1. At high frequency range, the measured phase response is limited to 5Hz – ½ sampling frequency and does not extend beyond, to the screen limit.

2. At low frequency range, the SPL is heavily distorted by the effect of FFT windowing.

3. At both: high and low frequency ranges the measured SPL and phase responses accuracy is limited by the measurement system dynamic range. There are sound reflections, and background noises, that typically produce garbage SPL/Phase responses at the frequency fringes. This needs to be removed.

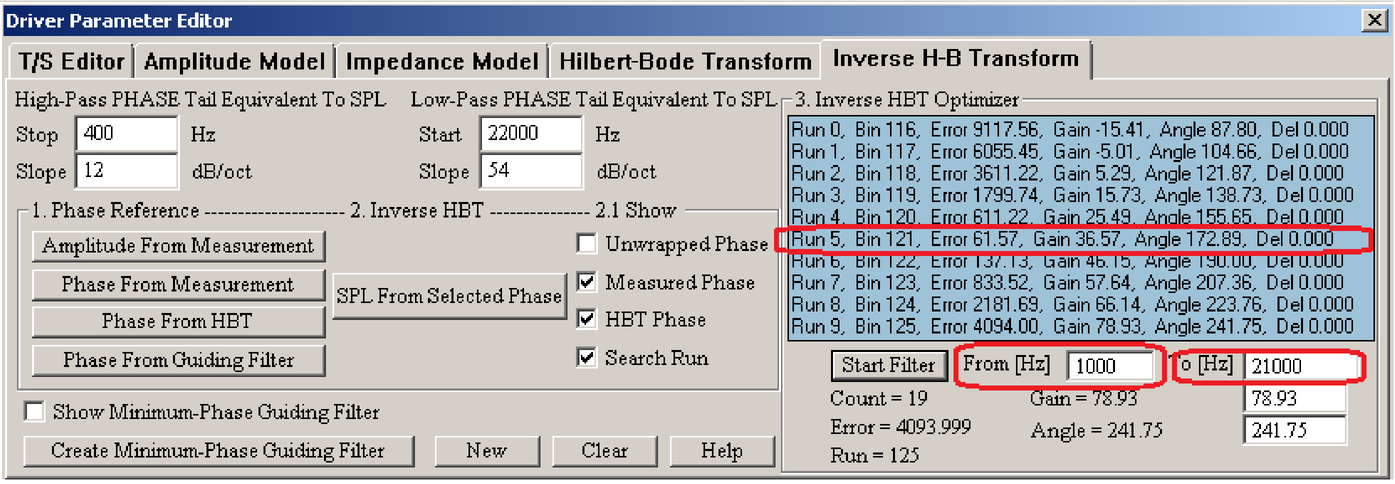
Therefore, attaching the asymptotic "phase tails" would clean up the measurements perfectly and would enable the SPL/Phase the be calculated over the whole frequency range of 5Hz-50kHz. As it was explained in Part I, the phase slopes are attached via SPL slopes mechanism – so, that the process looks more familiar to the user.

Most importantly, the SPL calculated via IHBT is not sensitive to the attached slopes, as it is calculated only between it's designated frequency points, that are different from the attachment points of the phase slopes. In our example, the phase slopes are attached at 400Hz (+12dB equivalent) and 22kHz (-54dB equivalent). While the IHBT is calculated between 1000Hz and 21kHz.



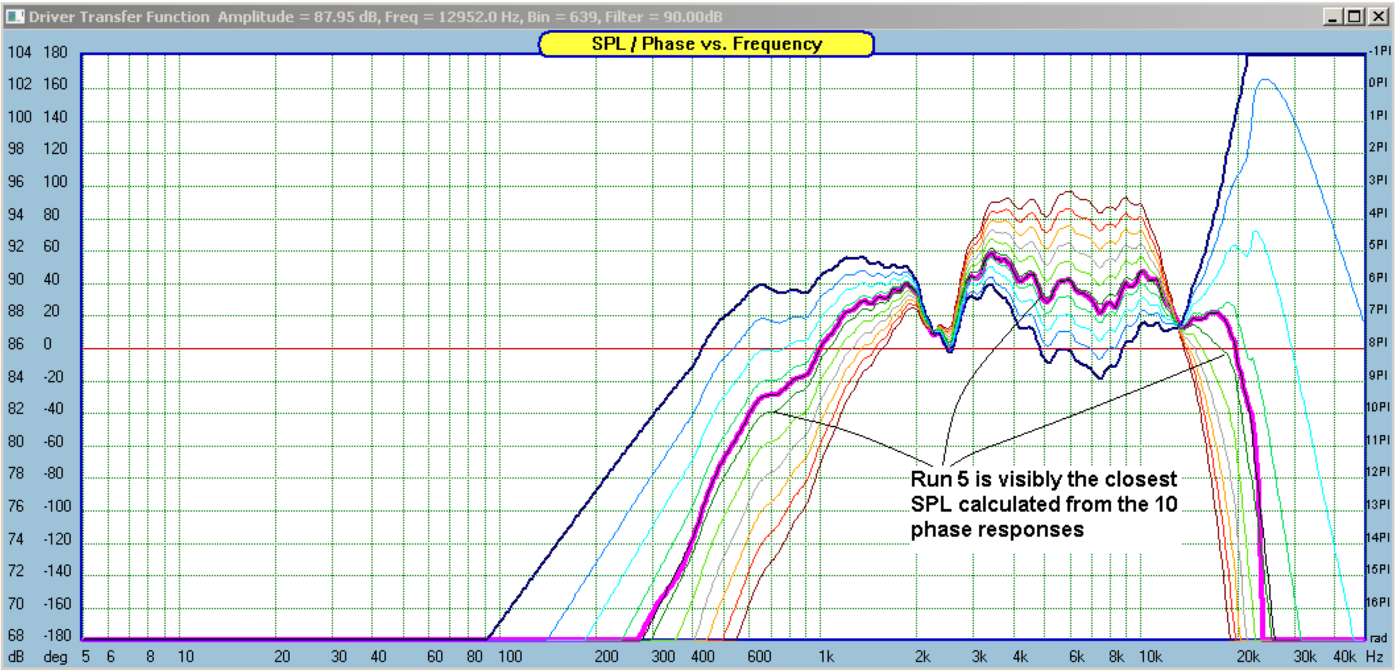
Step 1 – 10 Coarse FFT window placements

As explained before, in Stage I, the FFT window is placed at 10 arbitrary, but consecutive time samples, and resulting phase responses are supplied to the IHBT algorithm. This way, we obtain 10 SPL responses and one of them will be the best match with the originally measured SPL.



Shown below, are the 10 SPL curves re-created by the IHBT algorithm. Please note, that the vertical scale resolution has been enchanced from 5dB to 2dB in order to increase visual separation between SPL curves. The pink curve is the originally measured SPL, and it is our "reference" SPL curve.

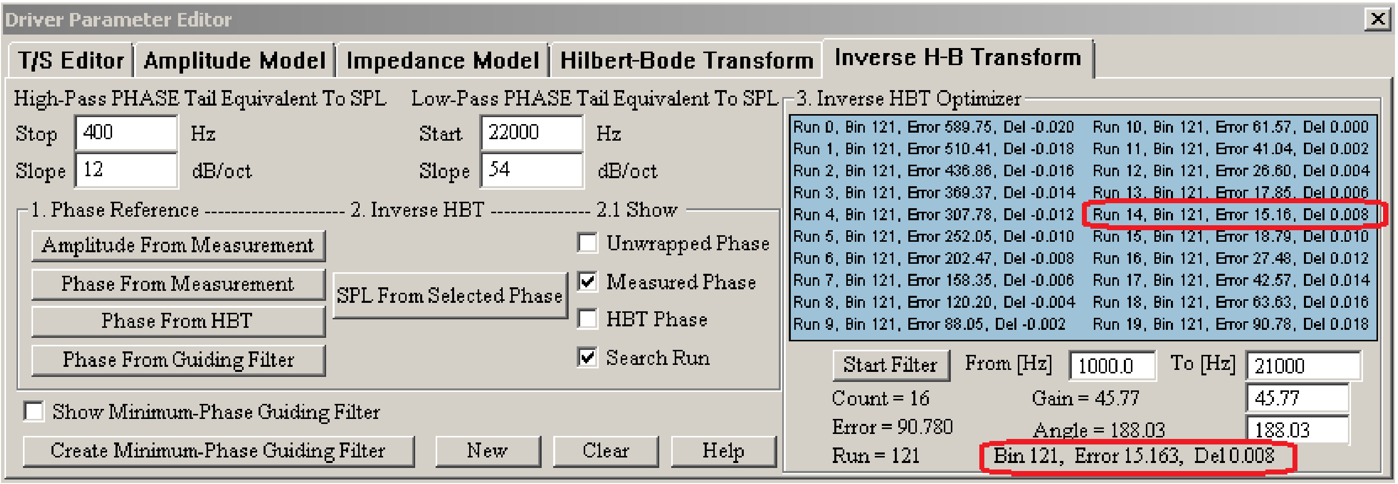
Next, the process automatically searched for the minimum "Error" value and selected Run 5 (or time sample Bin 121) as the starting point for the Stage II.



Stage II is started automatically.

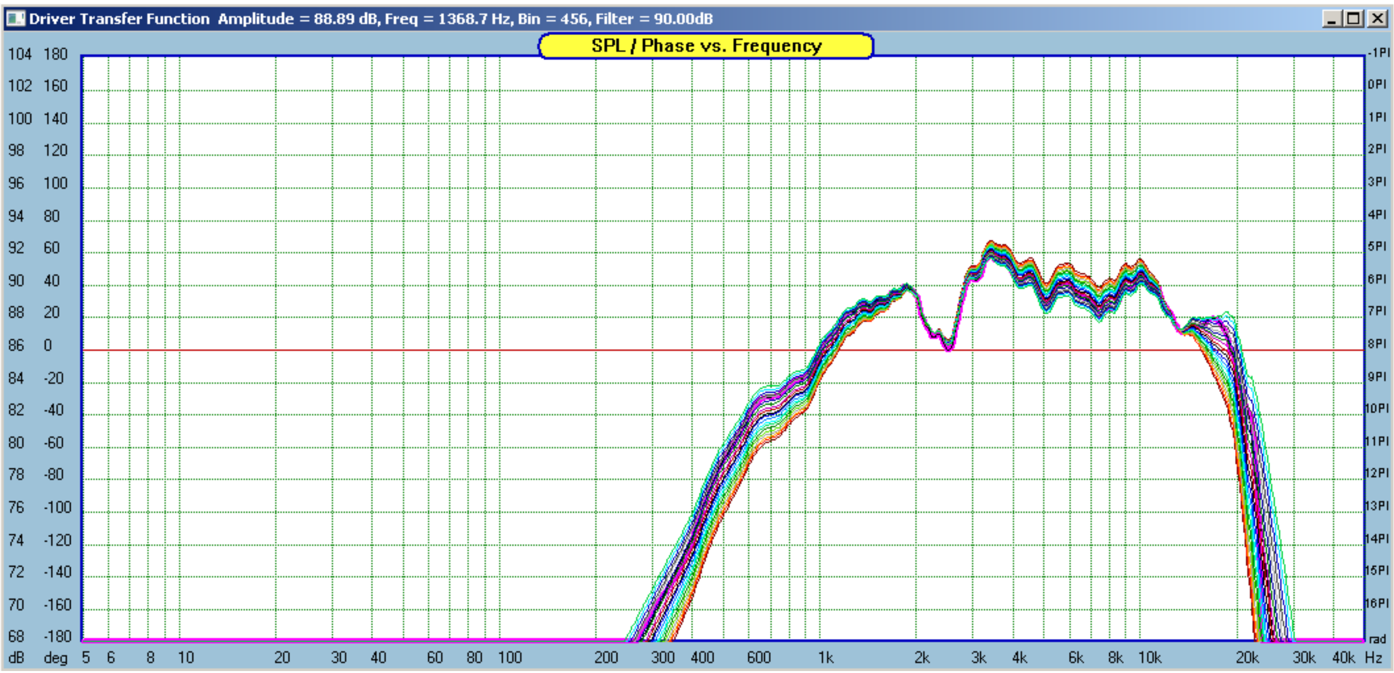
Step 2 – 20 Fine FFT (positive and negative) window delays

In Stage II, the algorithm adds or subtracts ten 2usec time delays so that the new phase is recalculated across 20usec to the left and 20usec to the right of the sampling time determined from Stage I.



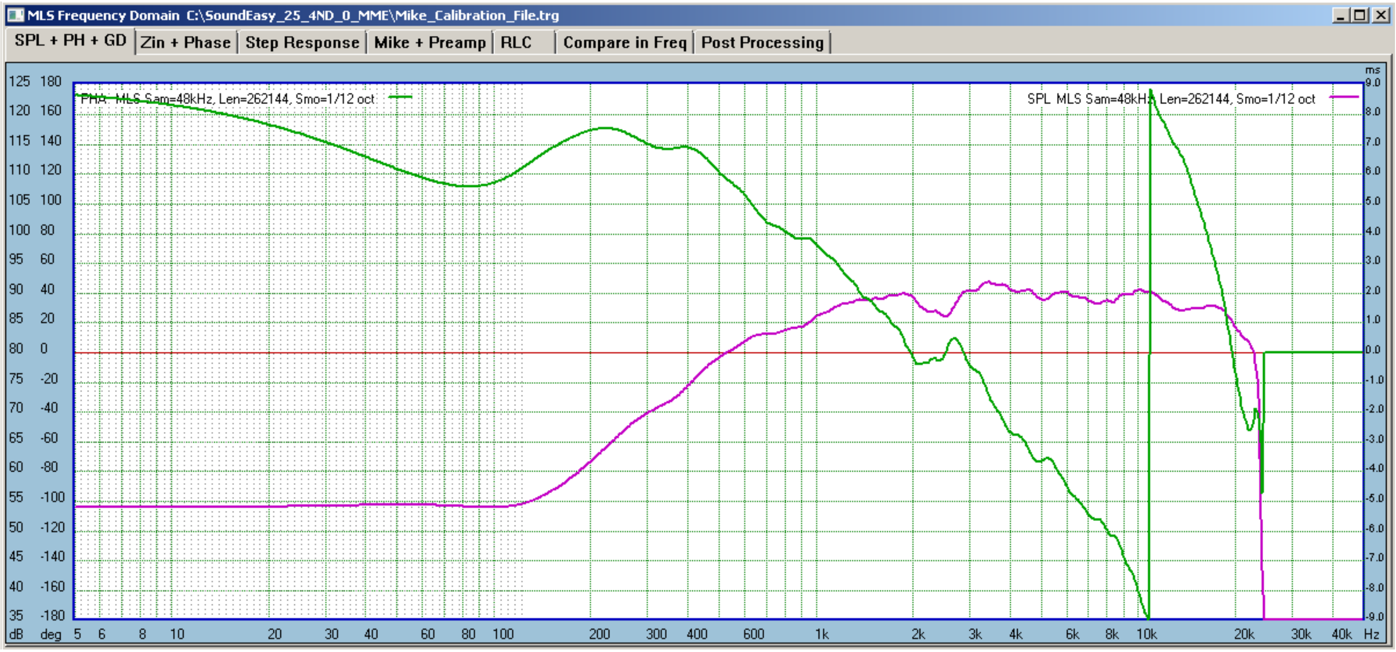
Shown below, are the 20 SPL curves re-created by the IHBT algorithm. Please note, that the vertical scale resolution has been set again to 2dB. The pink curve is the originally measured SPL, and it is our "reference" SPL curve, as before.

The process automatically searched for the minimum "Error" value and selected Run 14 (or time sample Bin 121 + Delay 0.008 ms) as the final parameters for placing the FFT window and calculating the resulting phase response

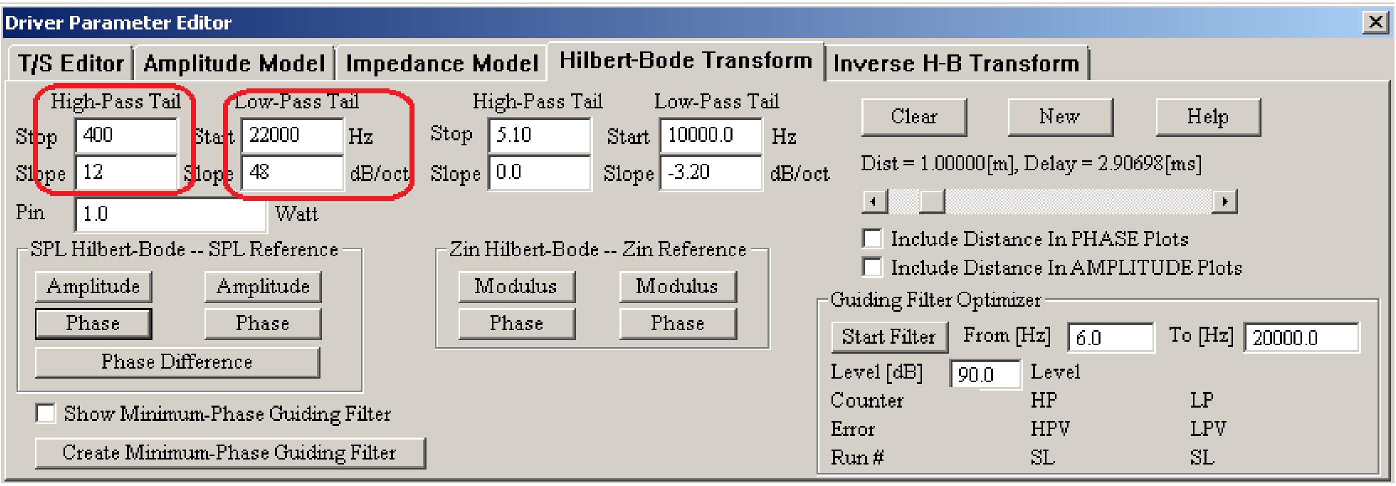


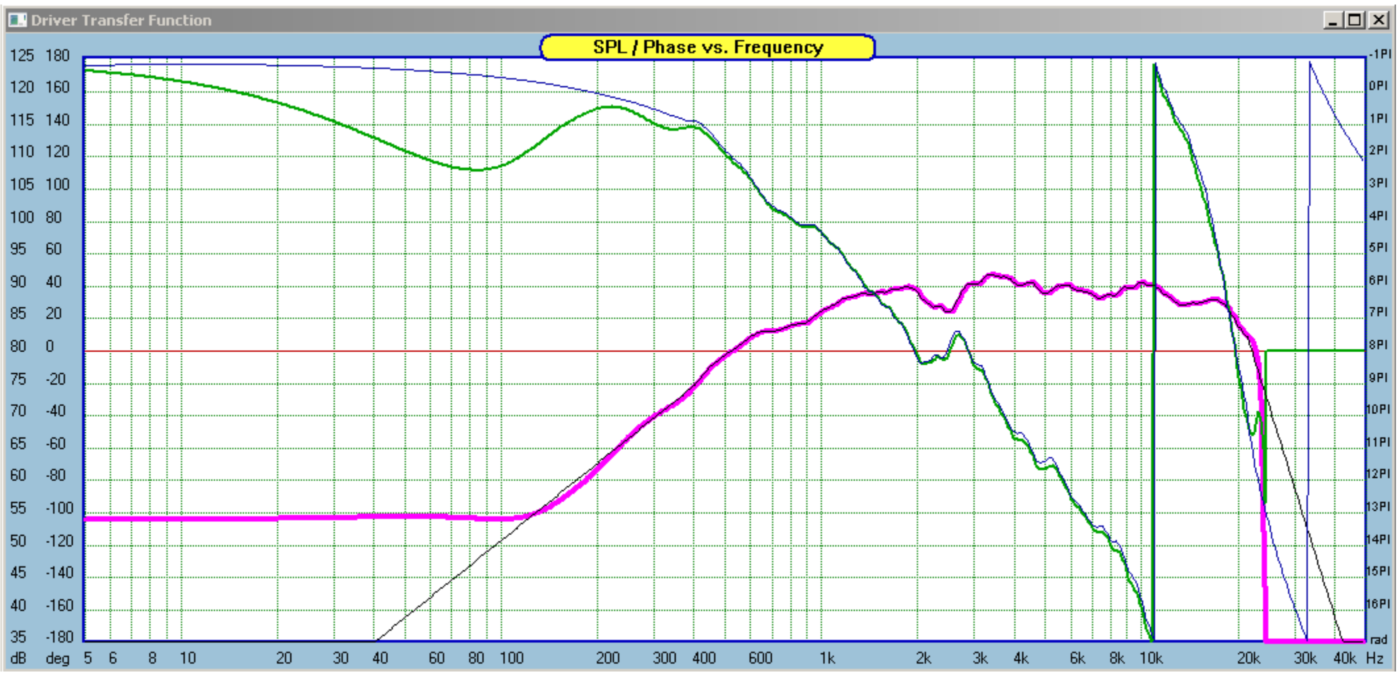
Back to MLS screen to correct FFT window placement at time sample Bin 121, with Pulse Delay of 0.008ms – see below.

Now, we obtain **correctly extracted minimum-phase response** – see below.



Having done that, the measured phase response becomes the PHASE TEMPLATE for correcting HBT settings. We can now go to HBT screen and correct HBT settings to reflect correctly measured Minimum-Phase phase response.



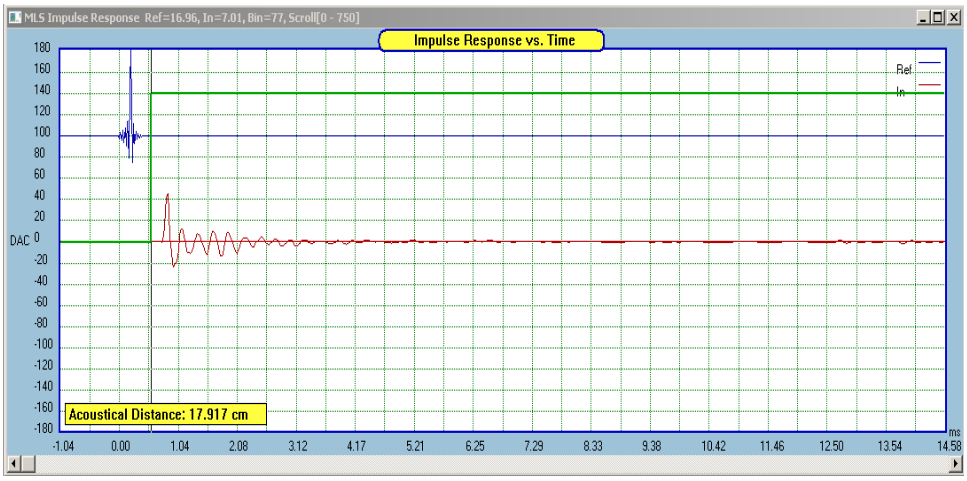


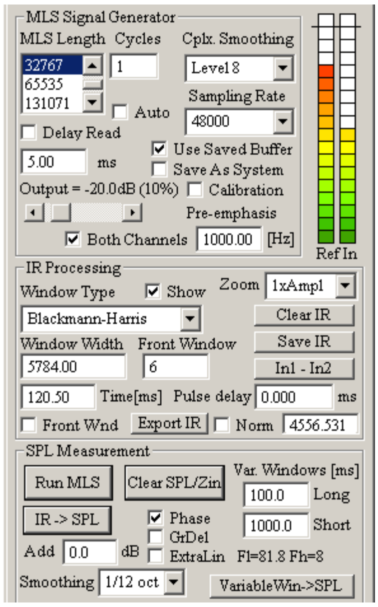
The thin blue curve is the minimum-phase phase response. Please note, that the phase response is clean and extends across the whole frequency range.

The thin black curve is the measured SPL response. Please note, that the SPL response is clean and extends across the whole frequency range.

12” Woofer Driver in Vented Box

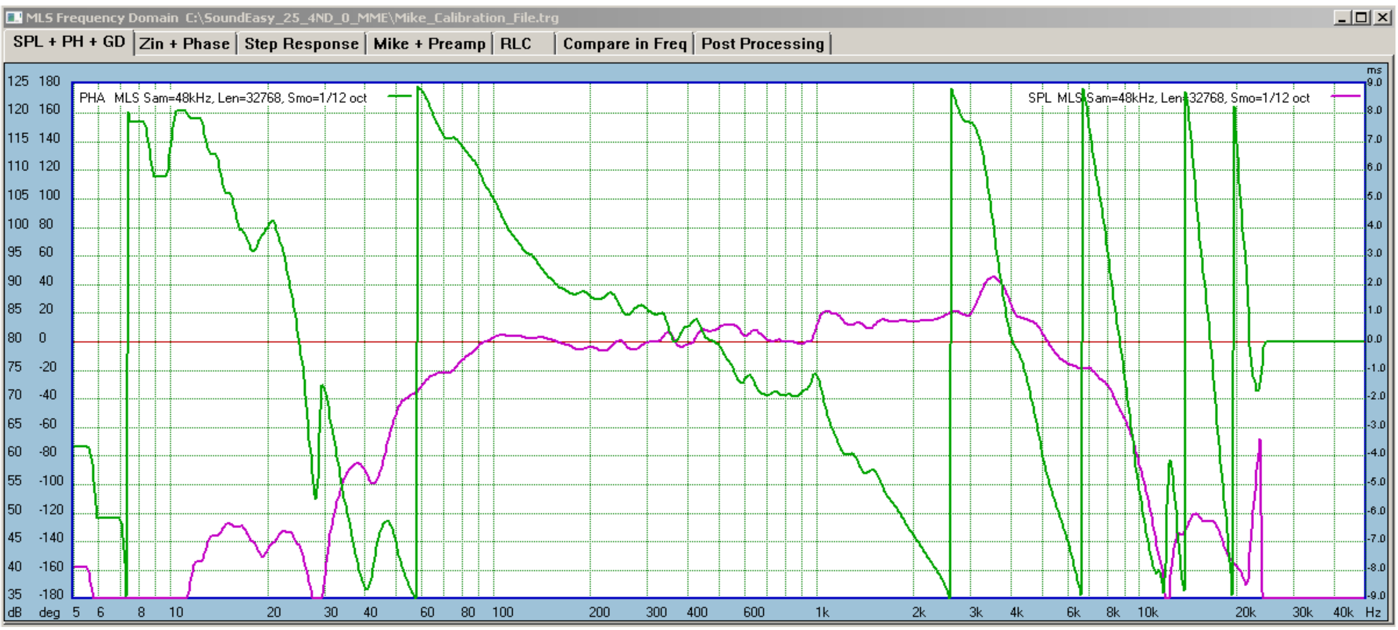
The next driver example is a 12” woofer driver measured in mid-sized room during daytime. Resulting SPL and Phase responses show room reflections and general low-level acoustic noise and reflections affecting measurement accuracy below -35dB.

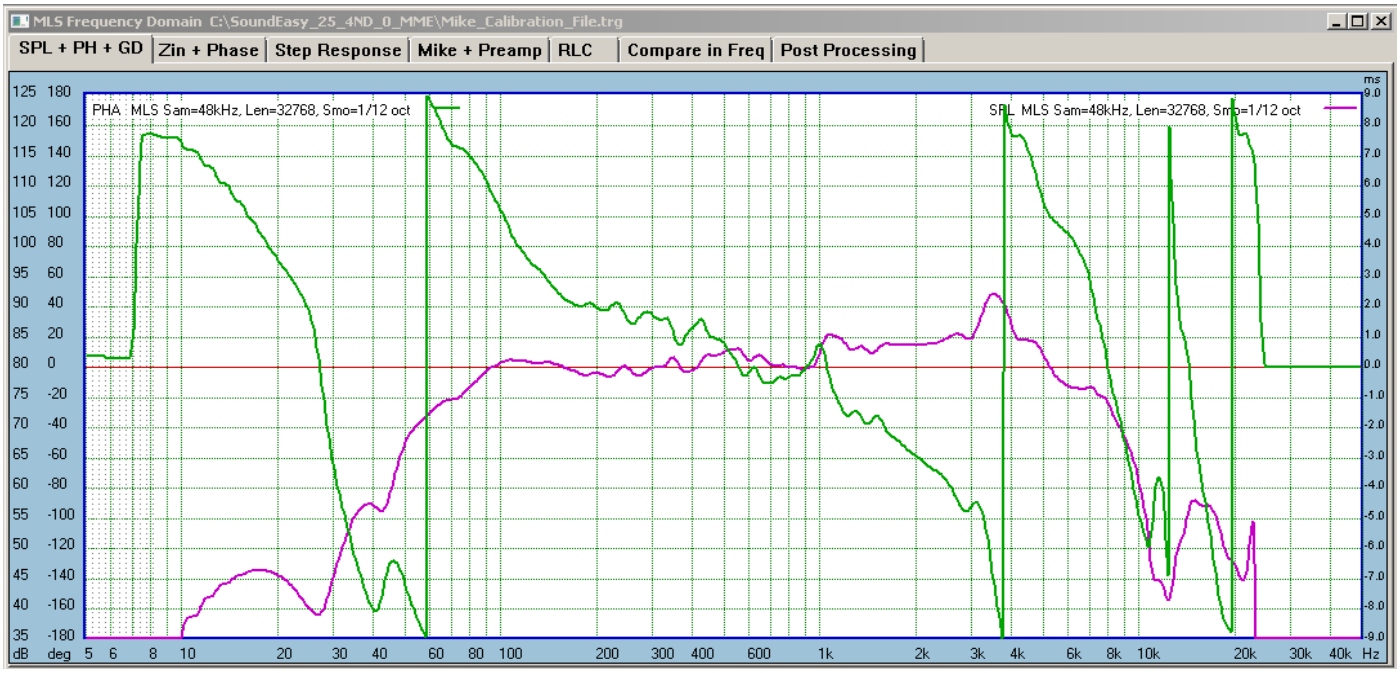


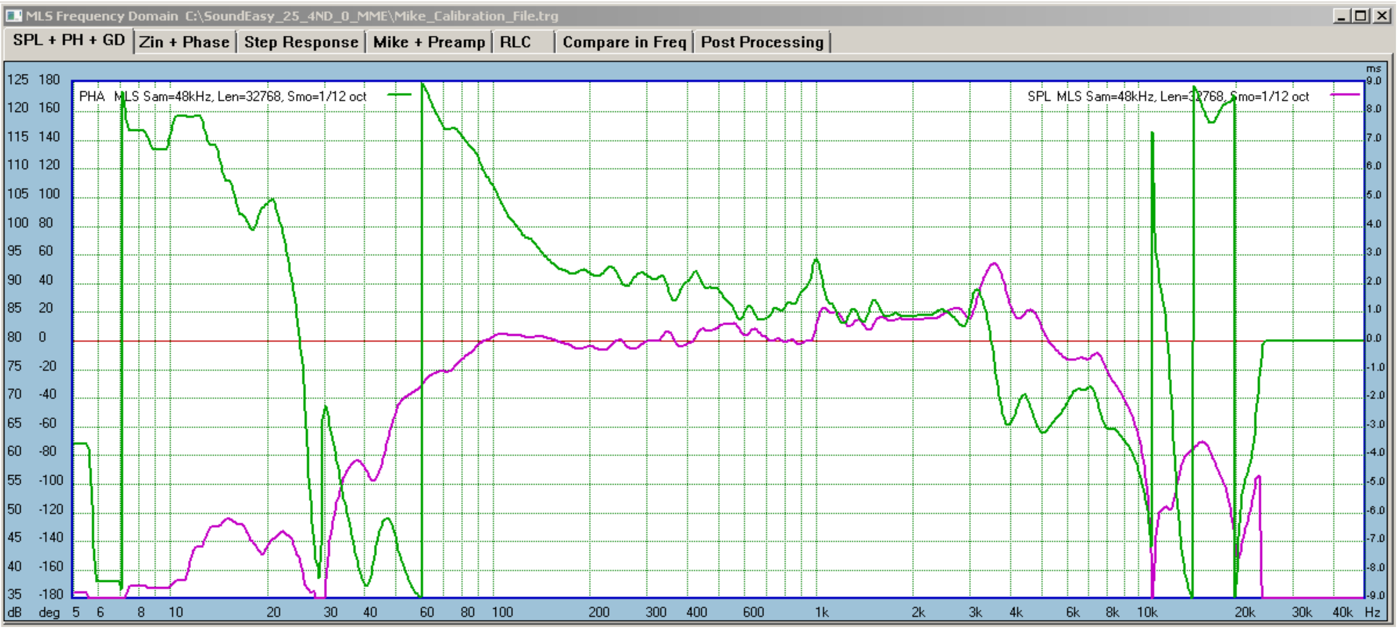


Now, we are grappling with the same question as for the tweeter – **how to determine the correct FFT window placement**?. Shown below are three examples of FFT window placement, and corresponding phase response (green curves).

On the first picture, the FFT was placed at Bin 72, resulting in perhaps too many +180/-180 deg phase transitions. On the middle picture, the FFT window start was placed at Bin 77. Here, we have fewer phase transition – but is this correct?. On the bottom picture, the FFT window was placed at Bin 81, resulting in phase response with a very few transitions.







As explained in Part I article, the IHBT requires phase slopes to be attached to the phase response. There are several reasons for this.

1. At high frequency range, the measured phase response is limited to 5Hz – ½ sampling frequency and does not extend to the screen limit.

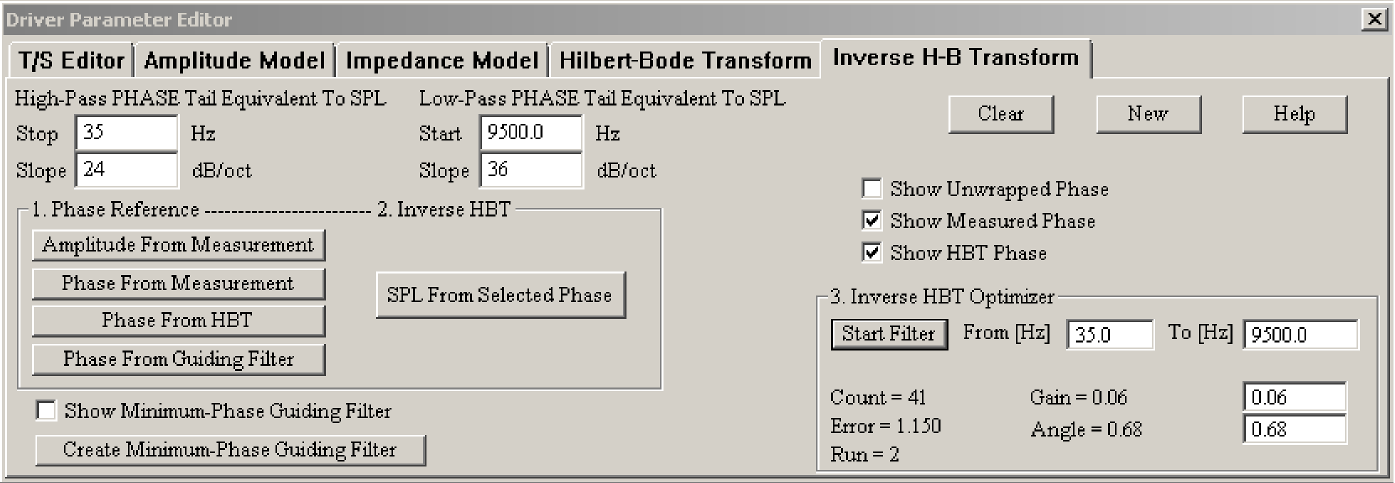
2. At low frequency range, the SPL is heavily distorted by the effect of FFT windowing and reflections.

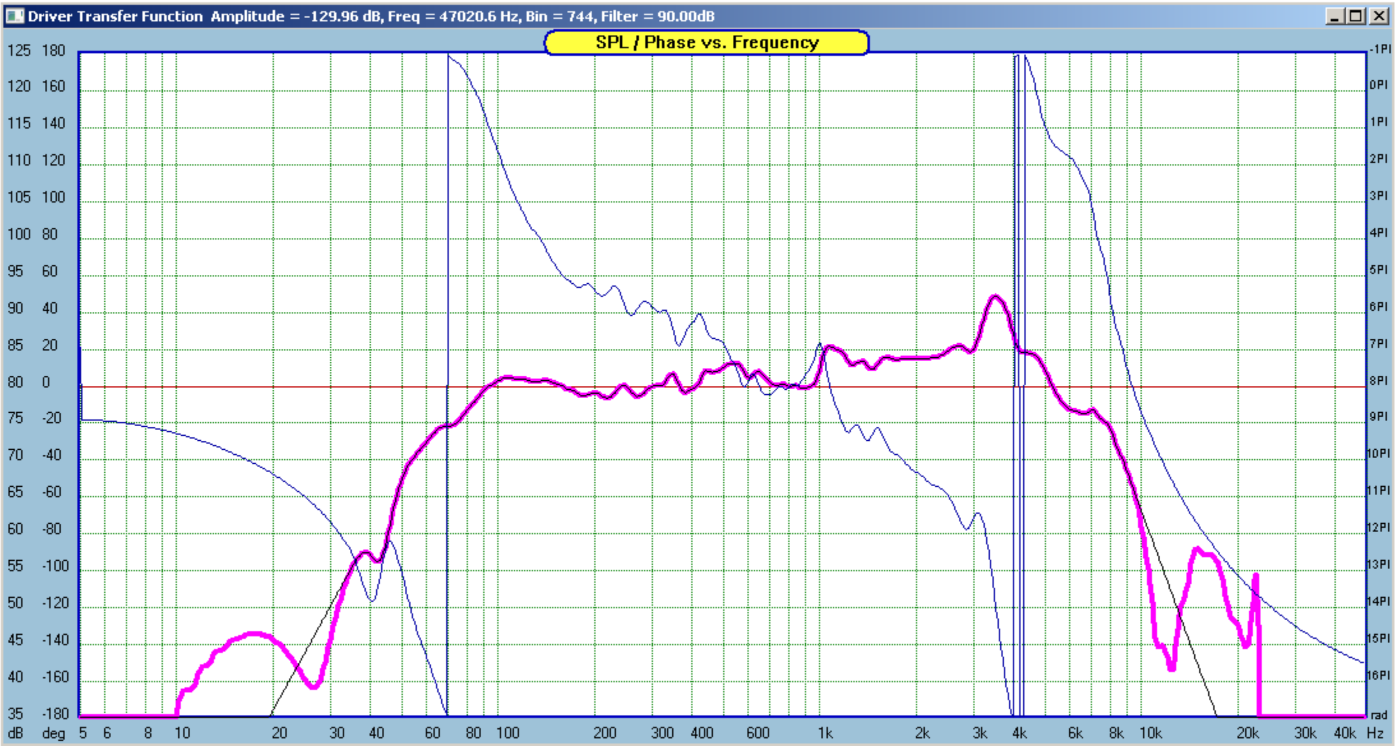
3. At both: high and low frequency ranges the measured SPL and phase responses accuracy is limited by the measurement system dynamic range. There are sound reflections, and background noises, that typically produce garbage SPL/Phase responses at the frequency fringes. This needs to be removed.

Therefore, attaching the asymptotic "phase tails" would clean up the measurements perfectly and would enable the SPL/Phase the be calculated over the whole frequency range of 5Hz-50kHz. As it was explained in Part I, the phase slopes are attached via SPL slopes mechanism – so, that the process looks more familiar to the user.

Most importantly, the SPL calculated via IHBT is not sensitive to the attached slopes, as it is calculated only between it's designated frequency points, that are different from the attachment points of the phase slopes. In our example, the phase slopes are attached at 35Hz (+24dB equivalent) and 9.5kHz (-36dB equivalent). While the IHBT is calculated between 100Hz and 9.5kHz.

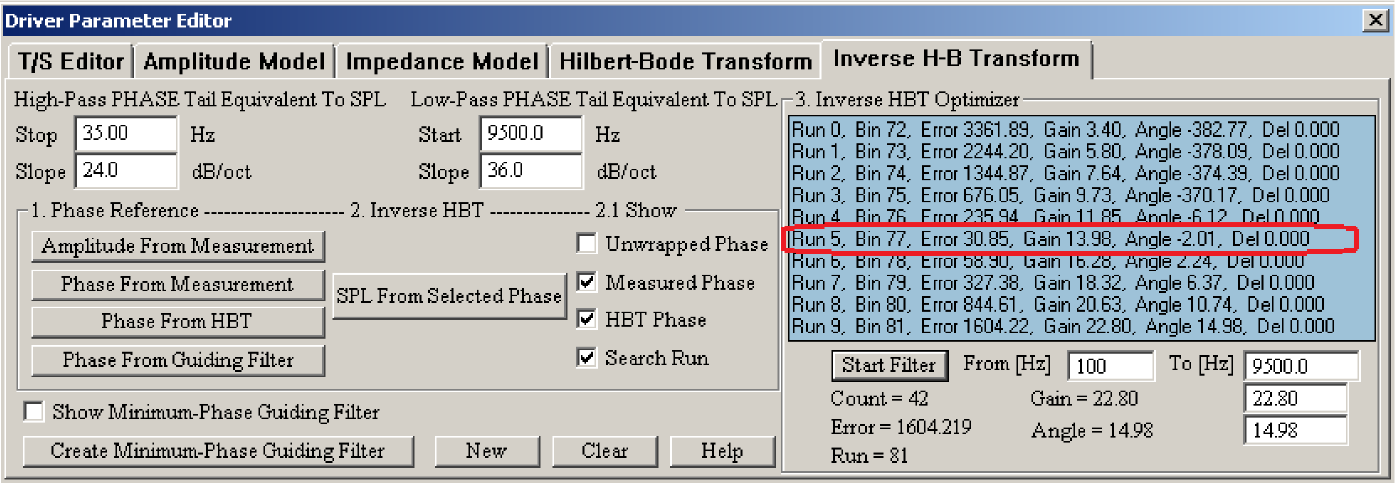
Phase from HBT using selected example slope parameters.





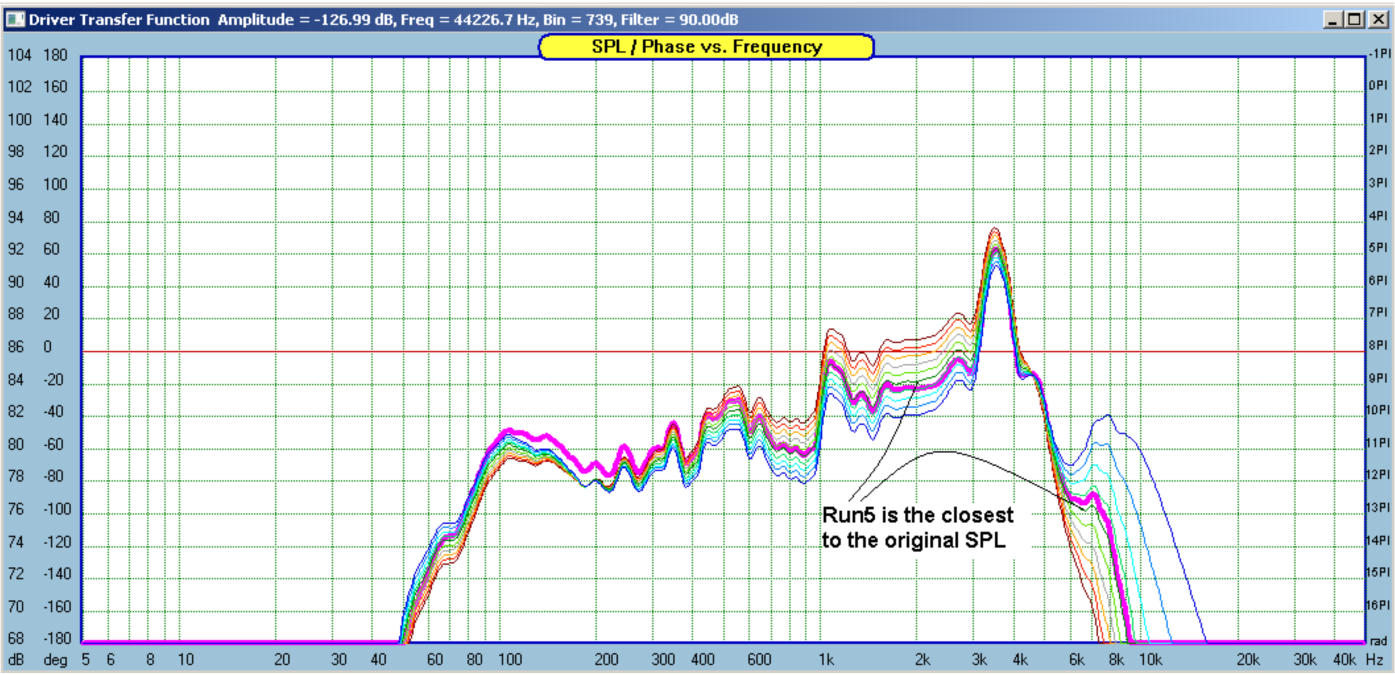
Step 1 – 10 Coarse FFT window placements

As explained in the tweeter example before, in Stage I, the FFT window is placed at 10 arbitrary, but consecutive time samples, and resulting phase responses are supplied to the IHBT algorithm. This way, we obtain 10 SPL responses and one of them will be the best match with the originally measured SPL.



Shown below, are the 10 SPL curves re-created by the IHBT algorithm. Please note, that the vertical scale resolution has been enchanced from 5dB to 2dB in order to increase visual separation between SPL curves. The pink curve is the originally measured SPL, and it is our "reference" SPL curve.

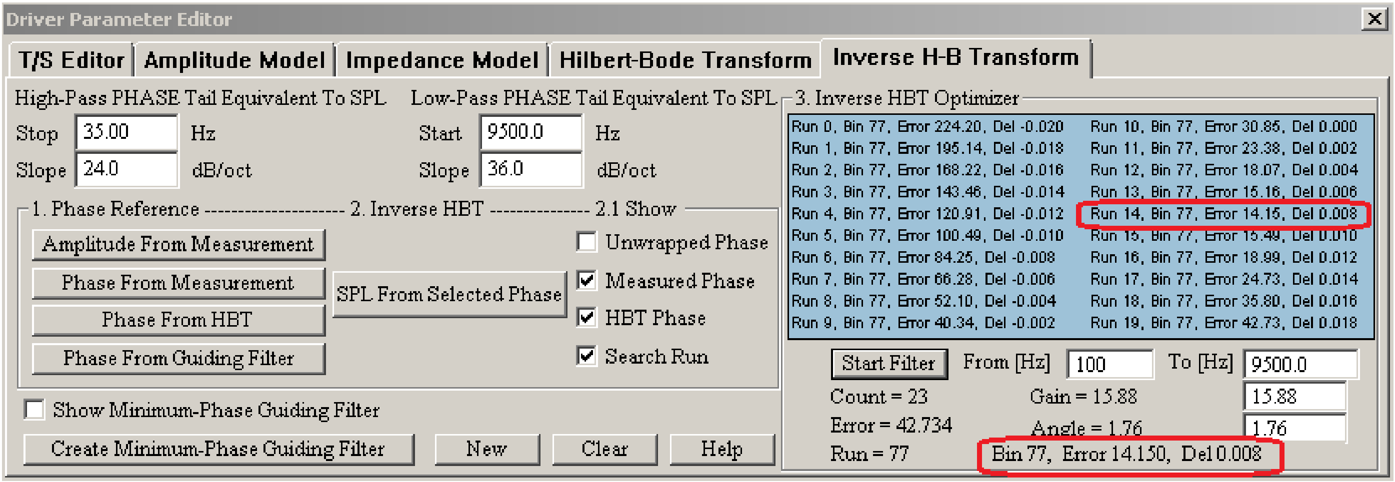
Next, the process automatically searched for the minimum "Error" value and selected Run 5 (or time sample Bin 77) as the starting point for the Stage II.



Stage II starts automatically

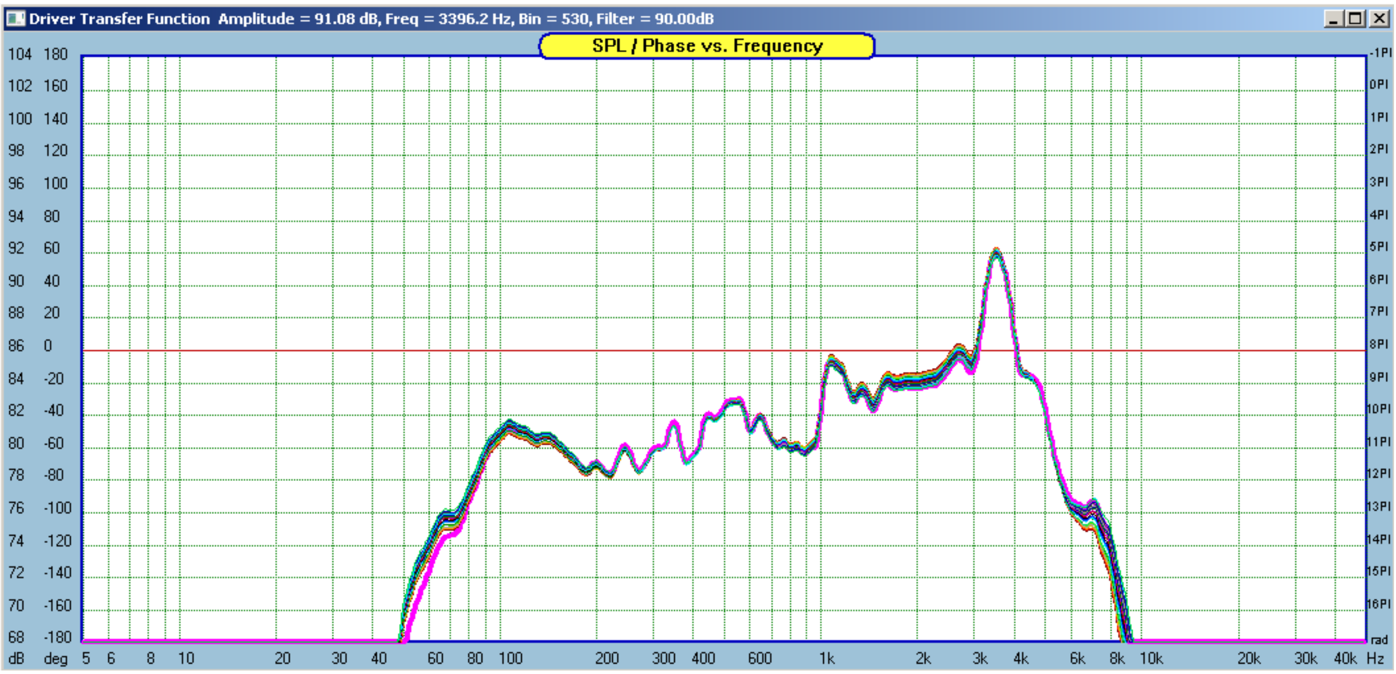
Step 2 – 20 Fine FFT (positive and negative) window delays

In Stage II, the algorithm adds or subtracts ten 2usec time delays so that the new phase is recalculated across 20usec to the left and 20usec to the right of the sampling time determined from Stage I.

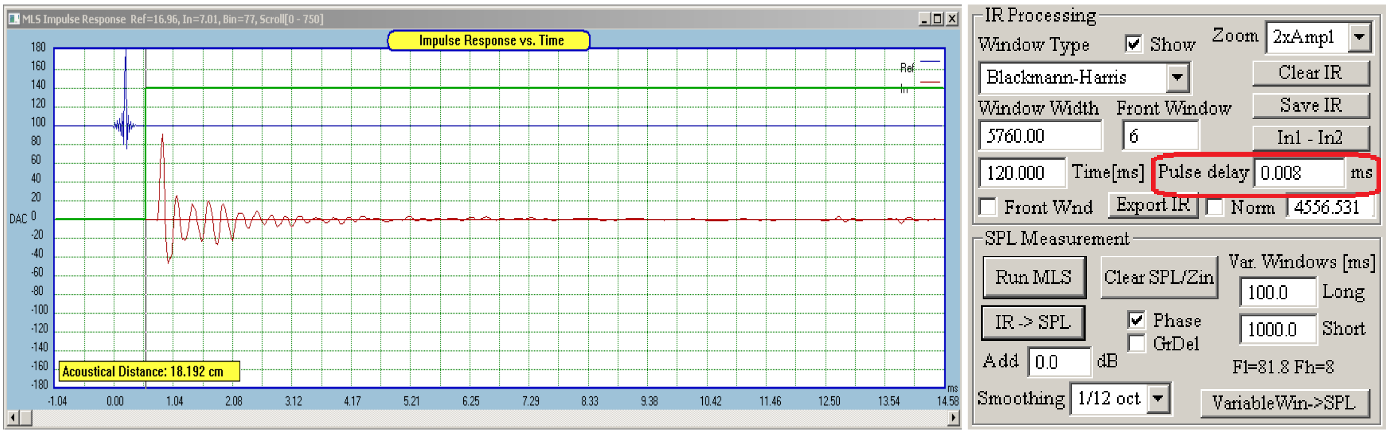


Shown below, are the 20 SPL curves re-created by the IHBT algorithm. Please note, that the vertical scale resolution has been set again to 2dB. The pink curve is the originally measured SPL, and it is our "reference" SPL curve, as before.

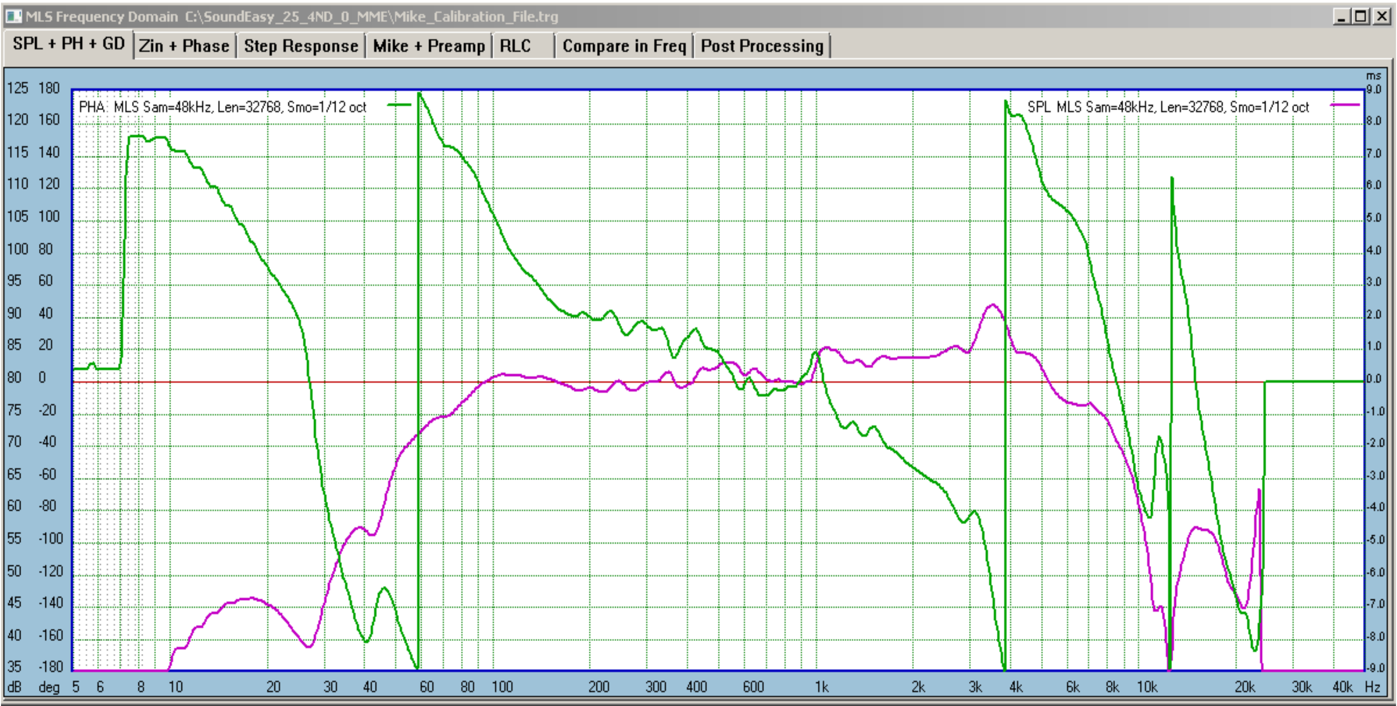
The process automatically searched for the minimum "Error" value and selected Run 14 (or time sample Bin 77 + Delay 0.008 ms) as the final parameters for placing the FFT window and calculating the resulting phase response



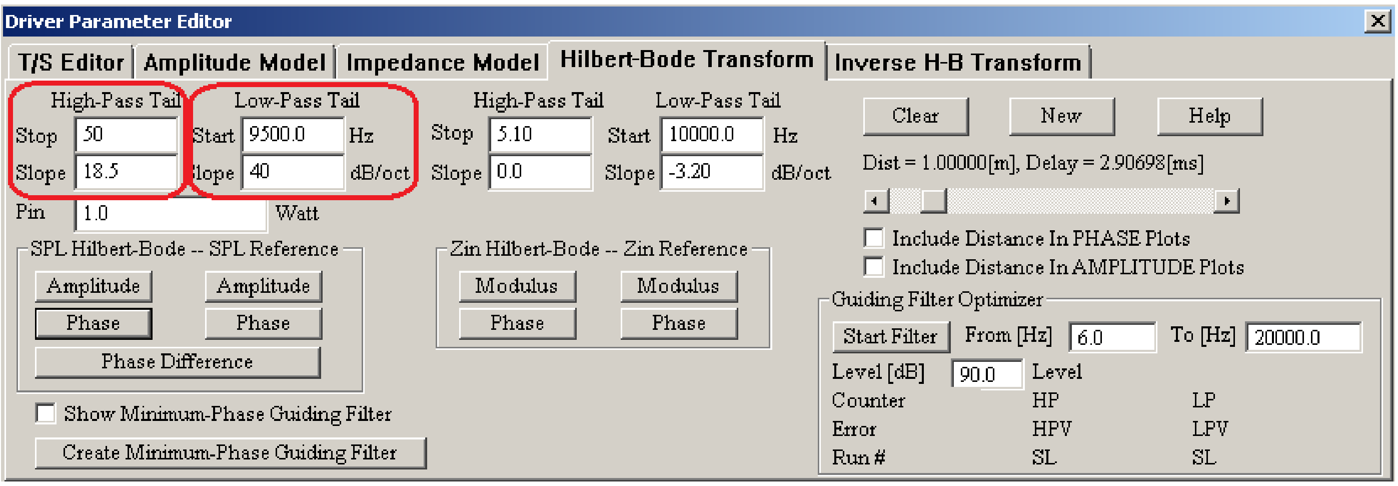
Back to MLS screen to correct FFT window placement at time sample Bin 77, with Pulse Delay of 0.008ms – see below.

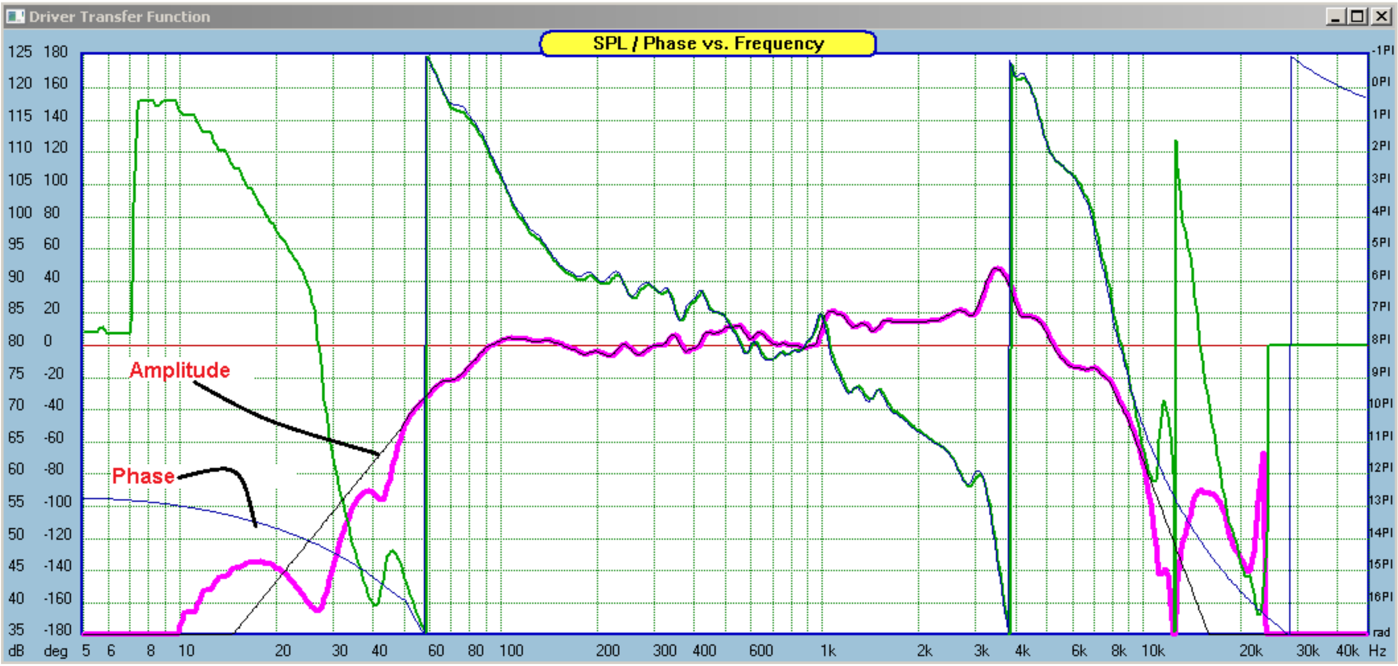


Now, we obtain **correctly extracted minimum-phase response** – see below.

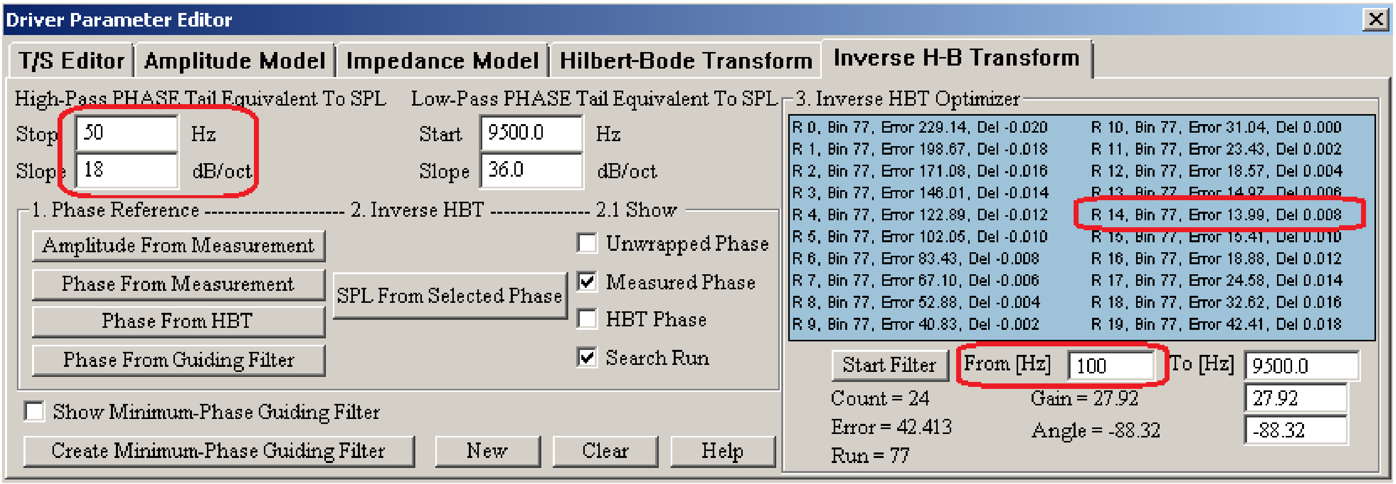


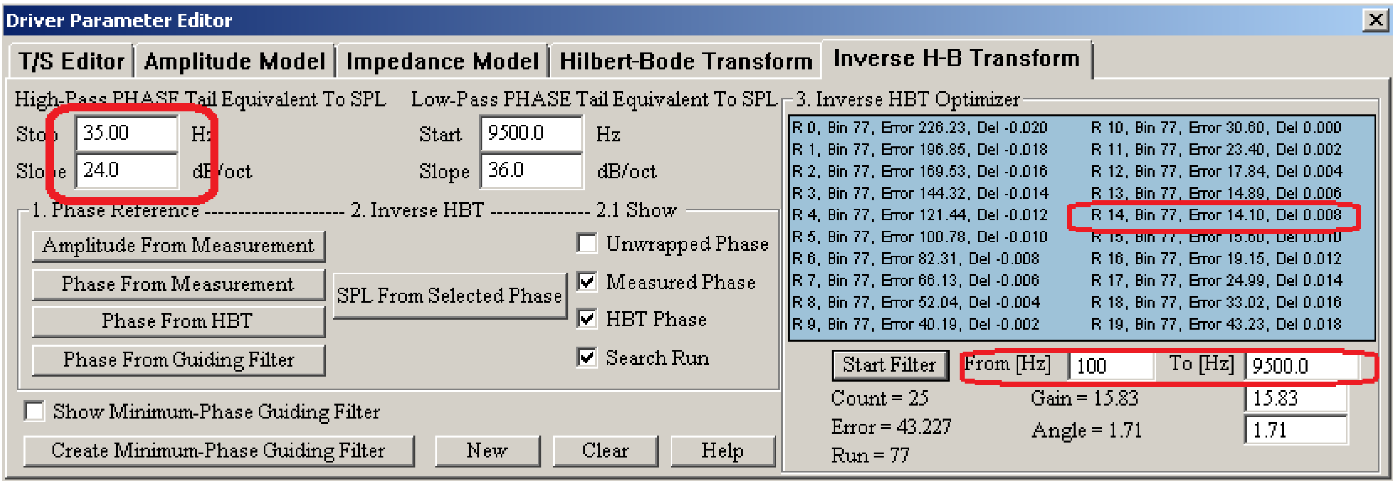
Having done that, the measured phase response becomes the PHASE TEMPLATE for correcting HBT settings. We can now go back to HBT screen and correct HBT settings to reflect correctly measured Minimum-Phase phase response.

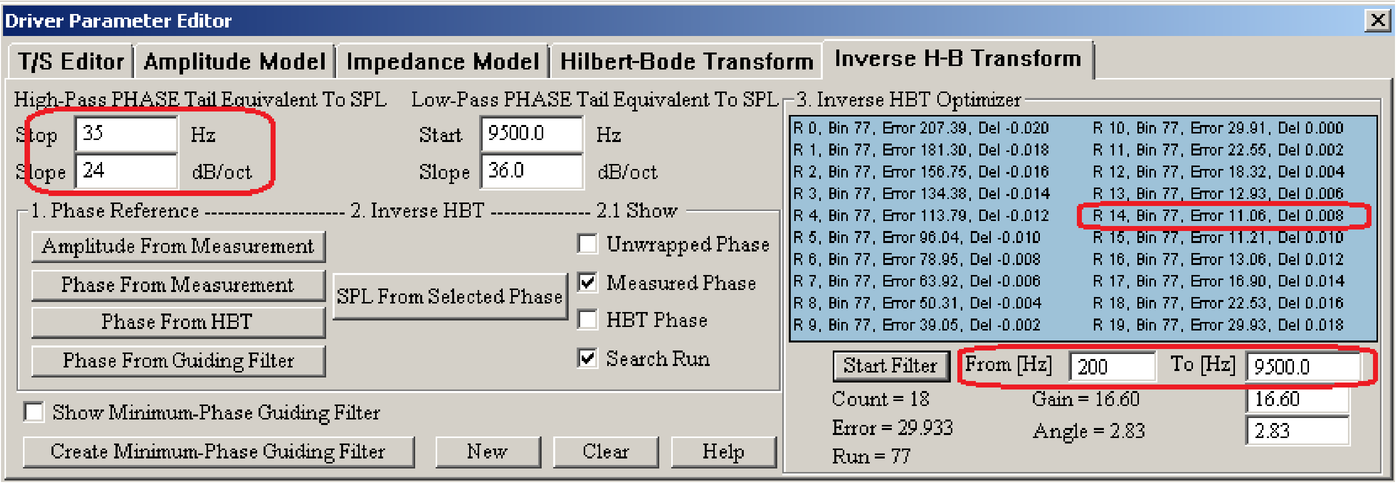




As mentioned before, the IHBT is not sensitive to the attached slopes. We can see this on the three examples shown below. Despite having low-frequency phase slopes (actually SPL slopes translated to phase slope) attached at different frequency points and with different slopes, the algorithm always selects Bin 77 with the minimum error.







Conclusions

The two examples shown above demonstrate the following advantages of the IHBT Automated Method.

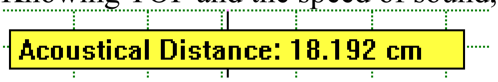
1. Loudspeaker drivers can be measured with accurately extracted minimum-phase phase response.

2. Final SPL and Phase responses are exceptionally clean and mathematically correct across the whole screen frequency range. They are locked together via HBT and IHBT transforms.

3. All undesirable background acoustical noises contaminating the measurements, are eliminated.

4. Having extracted minimum-phase phase response, it is easy to calculate corresponding **time-of-flight**. This would be TOF = "number_of_bins"/"sampling frequency" + Added Delay.

5. Knowing TOF and the speed of sound, we can calculate "Acoustical Distance"



6. Now, if you measure physical distance between the microphone and the front mounting baffle, you can correctly calculate the "Acoustic Centre Offset" for each driver.

Thank you for reading

Bohdan