

Automated Extraction of Minimum-Phase Phase response

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.
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.

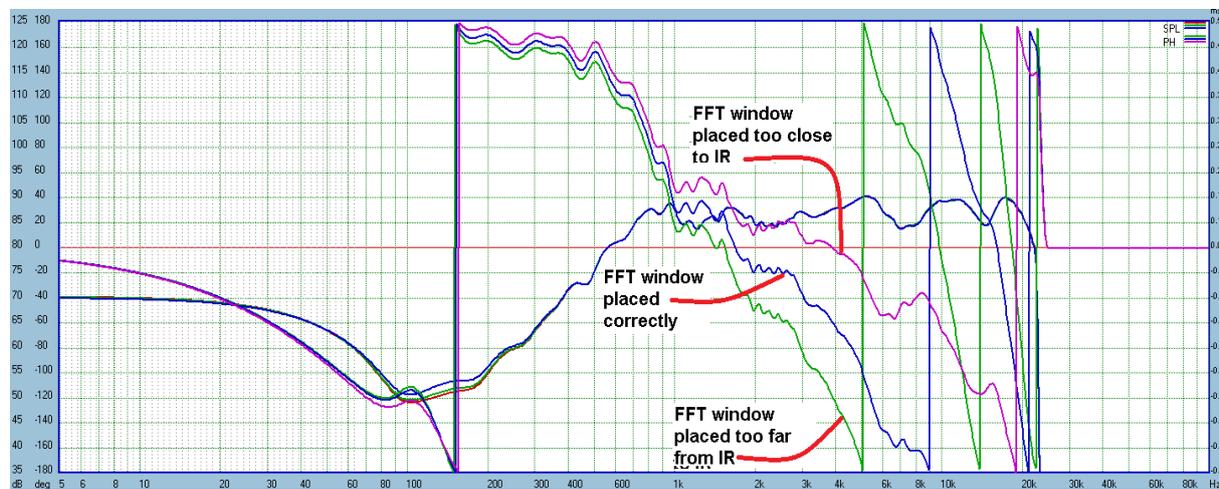


Figure 16.250. Three possibilities of placing FFT window.

The User Manual already 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. 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 chapter 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.

The measurement system can be 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 implemented in the program 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.

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”.

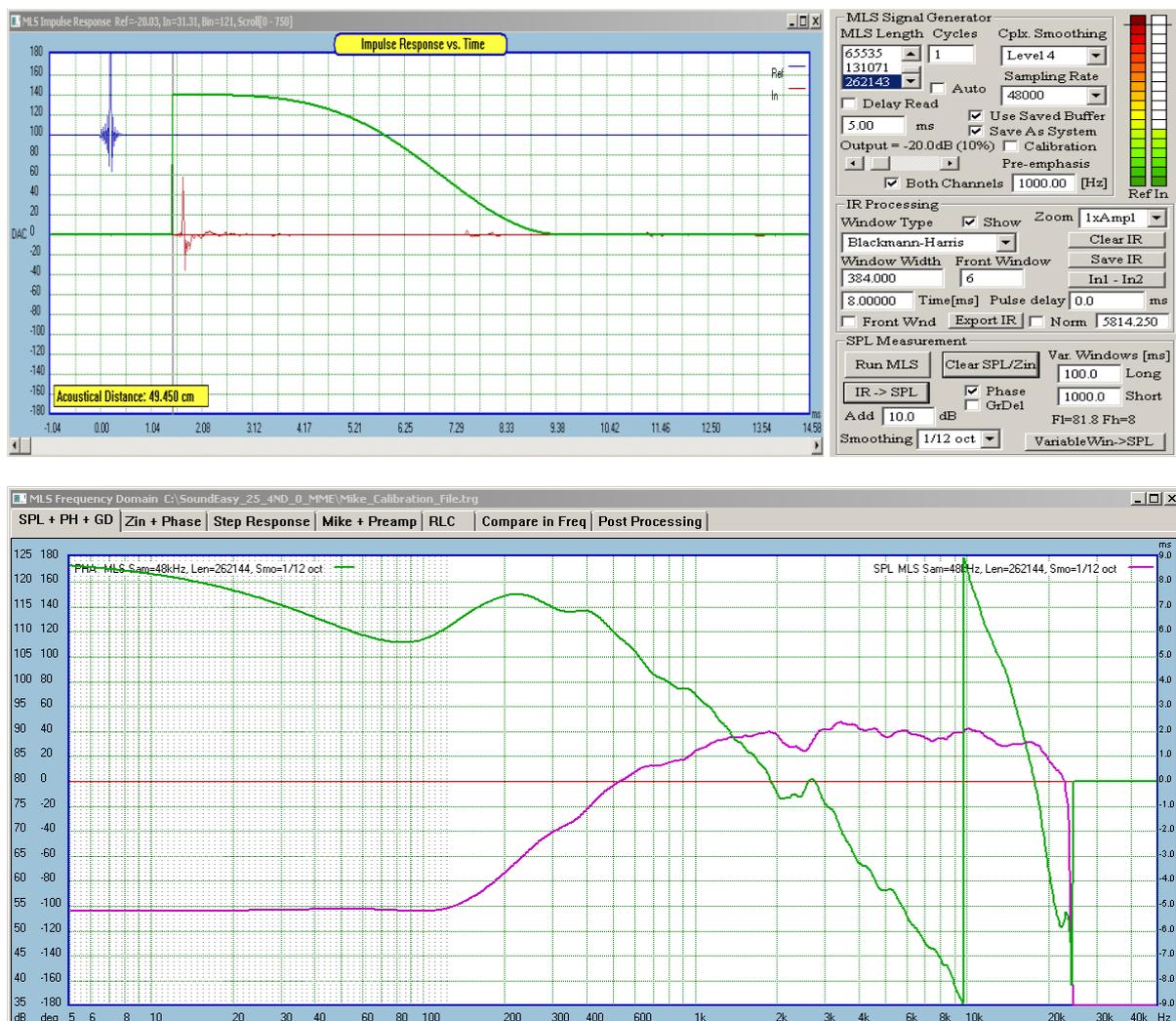


Figure 16.251. Impulse response and recovered SPL
16.191

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.

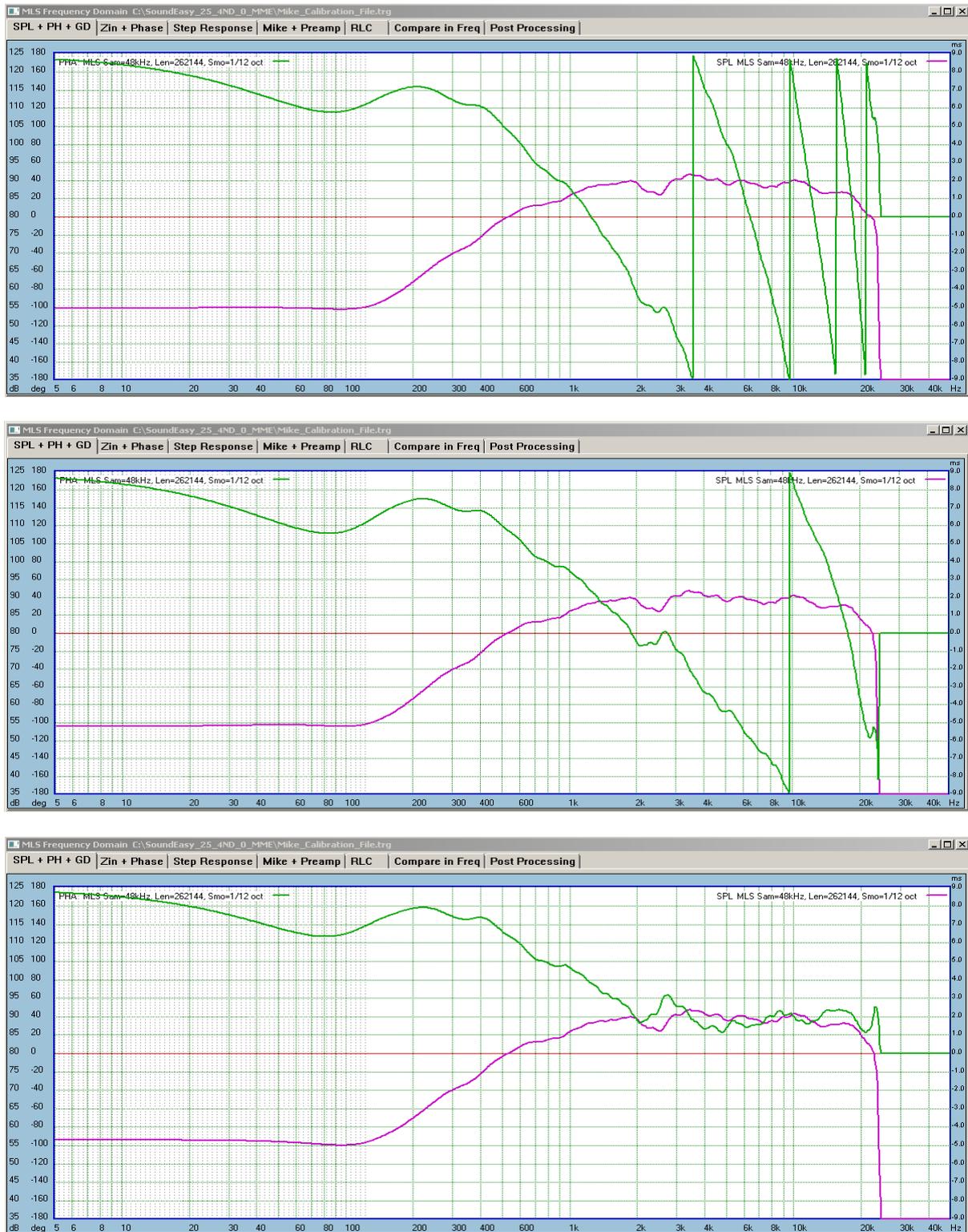


Figure 16.252. Three examples of recovered SPL/Phase – which one is correct?

As explained previously, the IHBT requires phase slopes to be attached to the phase response. There are several reasons for this. At high frequency range, the measured phase response is limited to $\frac{1}{2}$ sampling frequency and does not extend beyond, to the screen limit. At low frequency range, the SPL is heavily distorted by the effect of FFT windowing.

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 to be calculated over the whole frequency range of 5Hz-50kHz. As it was explained in previous chapter, 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.

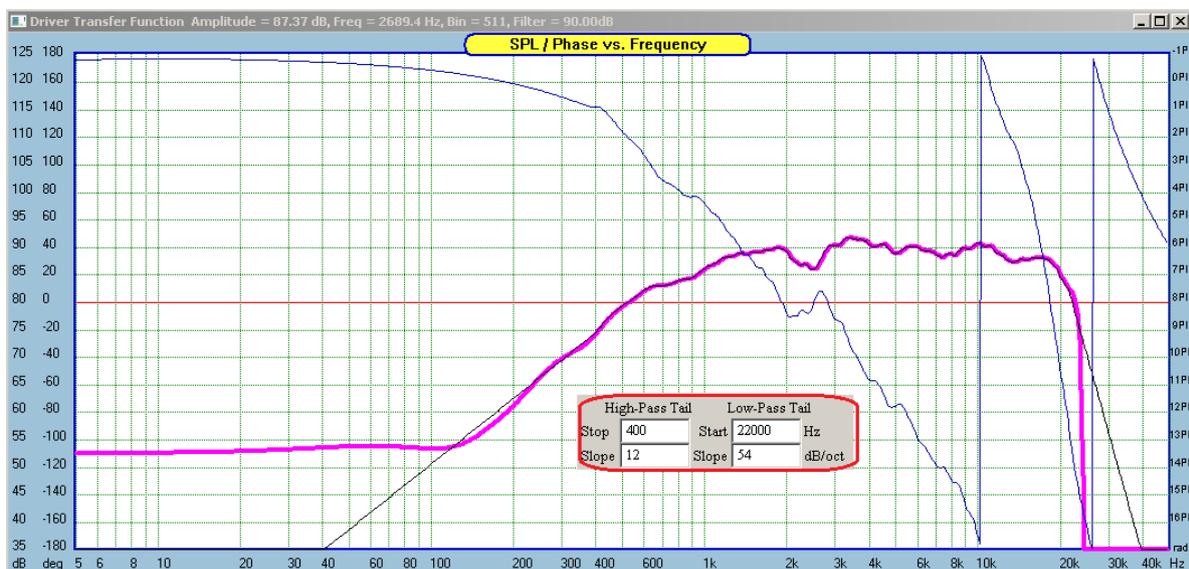


Figure 16.253. HBT used to nominate initial slopes.

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. Please select HBT parameters and checkboxes as shown below (**MUST check “Search Run” box to activate 2-stage HBT Optimizer**) and press “Start Filter” button.

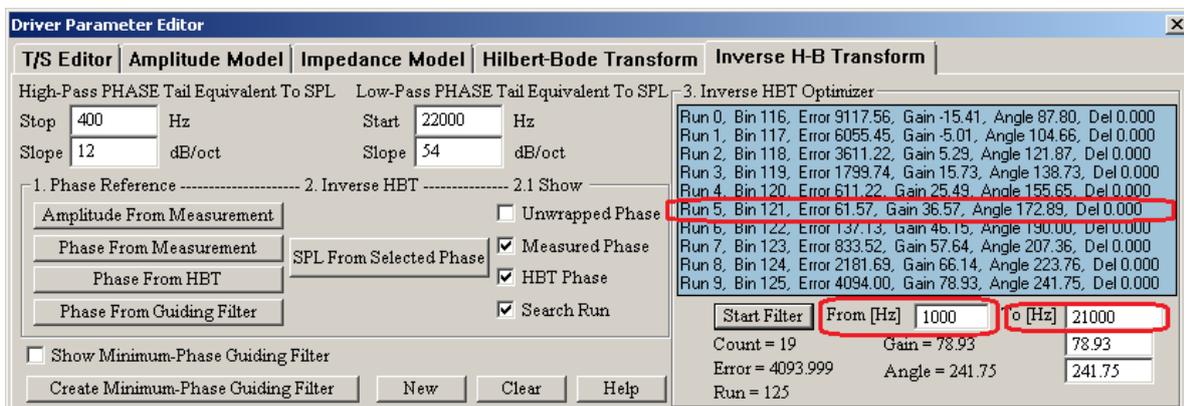


Figure 16.254. Error results in Stage 1.

Shown below, are the 10 SPL curves re-created by the IHBT algorithm. Please note, that the vertical scale resolution has been enhanced 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.

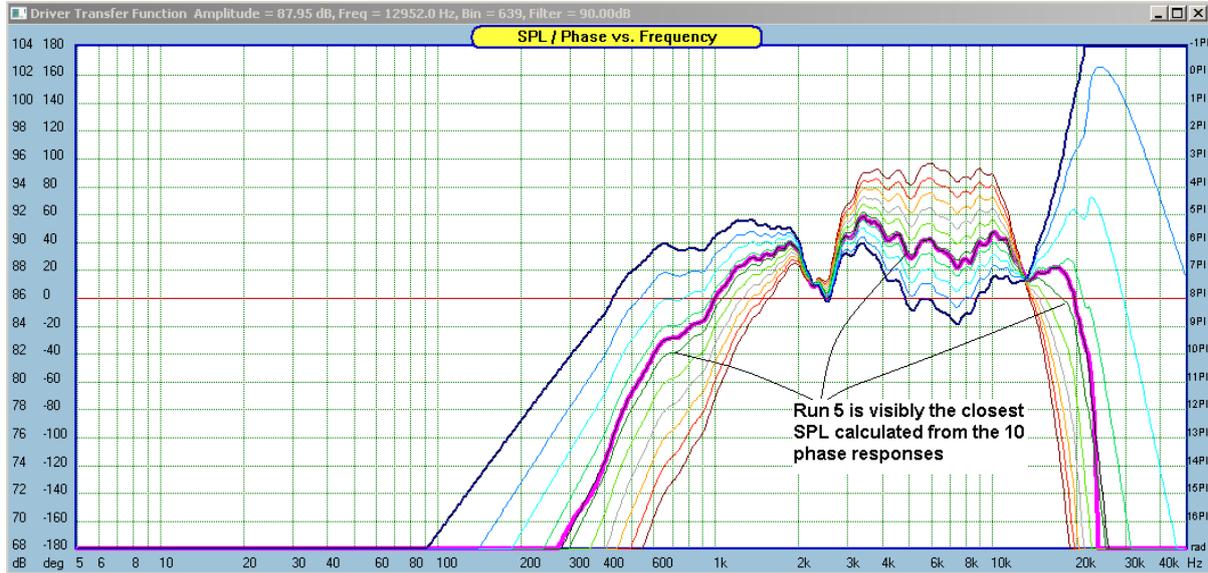


Figure 16.255. SPL error results in Stage I.

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.

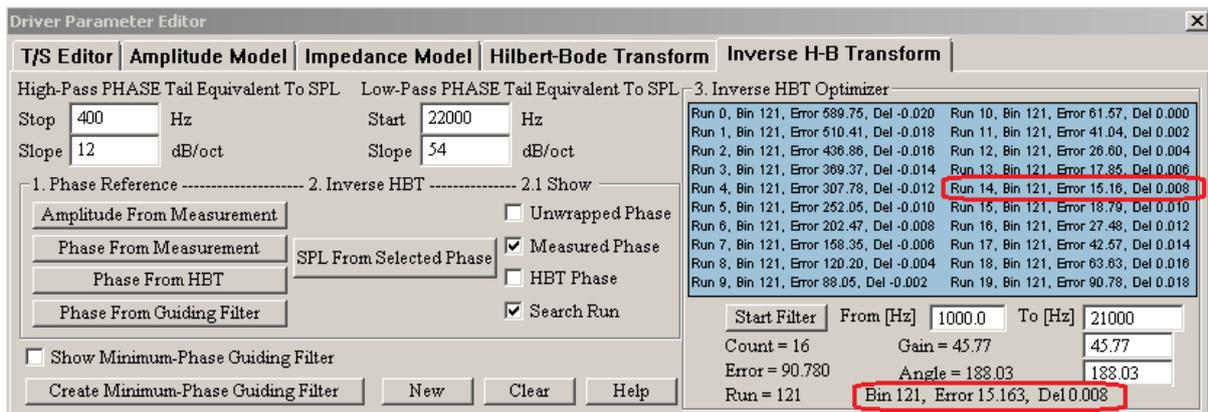


Figure 16.256. Error results in Stage II.

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

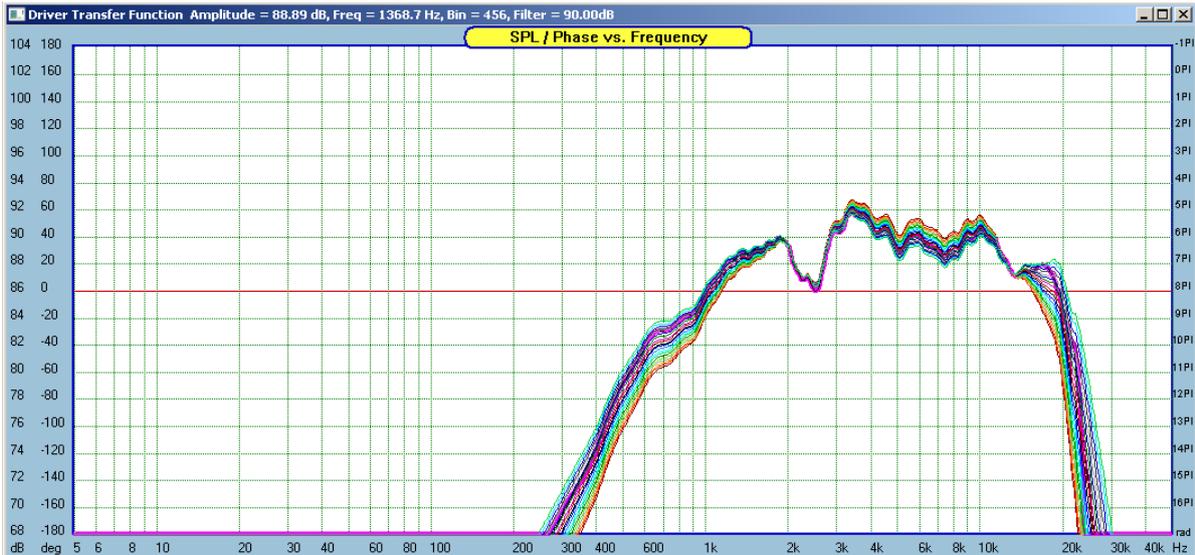


Figure 16.257. SPL error results in Stage II.

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.

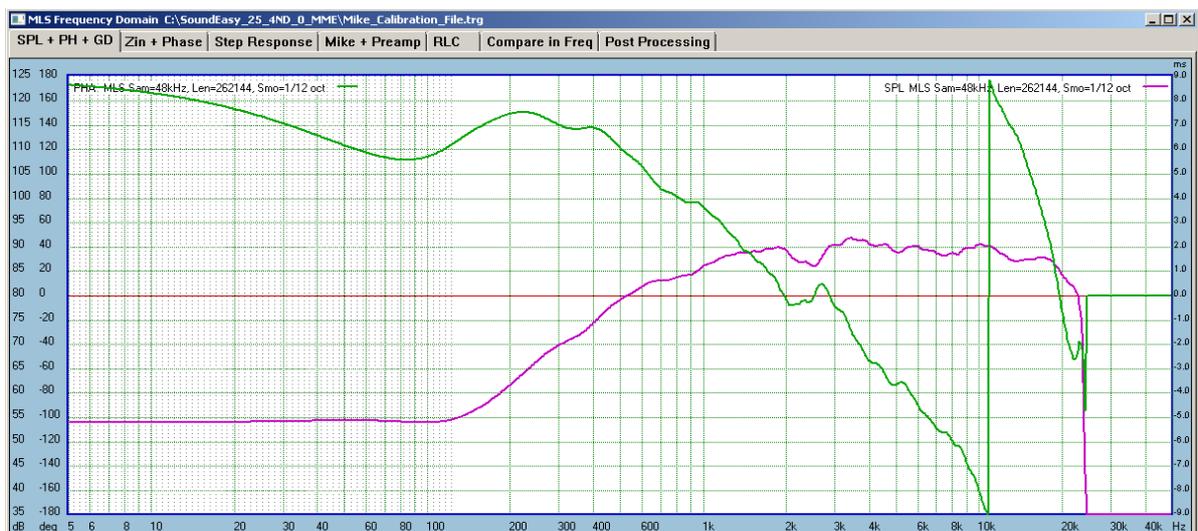


Figure 16.258. SPL/Phase after adding 0.008ms delay.

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.

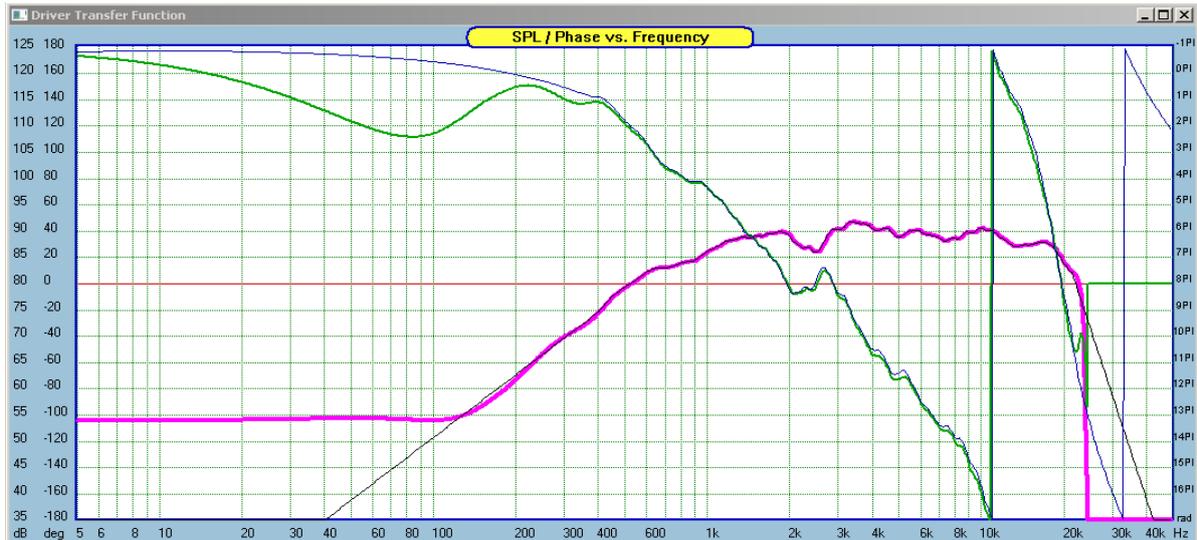
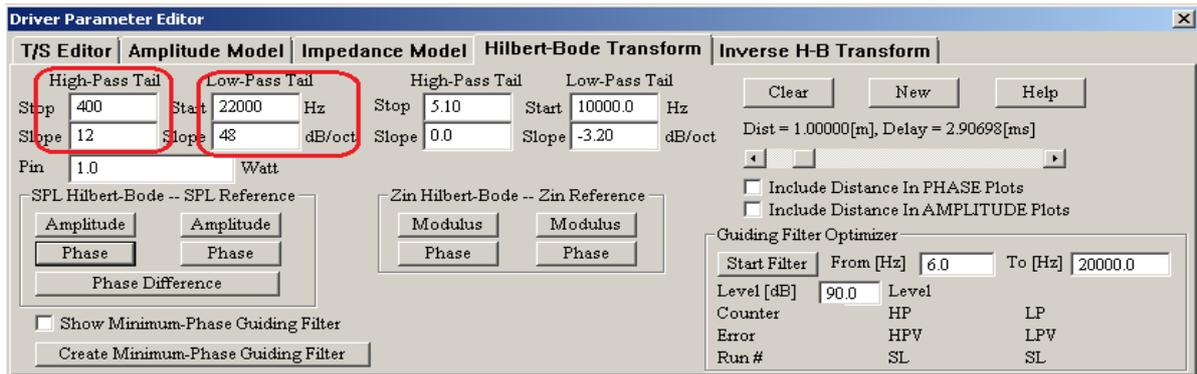


Figure 16.259. Final Minimum-Phase phase response obtained via HBT

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.

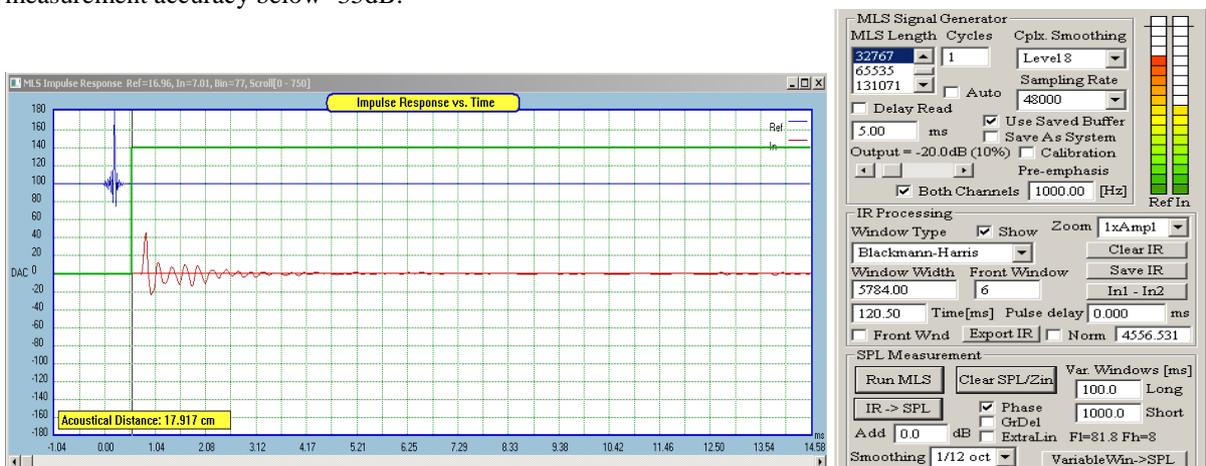


Figure 16.260. Impulse Response of a 12" woofer.

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.



Figure 16.261. Three examples of recovered SPL/Phase – which one is correct?
16.197

As explained in previous chapter, the IHBT requires phase slopes to be attached to the phase response. There are several reasons for this. At high frequency range, the measured phase response is limited to $5\text{Hz} - \frac{1}{2}$ sampling frequency and does not extend to the screen limit. At low frequency range, the SPL is heavily distorted by the effect of FFT windowing and reflections. 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 to be calculated over the whole frequency range of 5Hz-50kHz. As it was explained in previous chapter, 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 its 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.

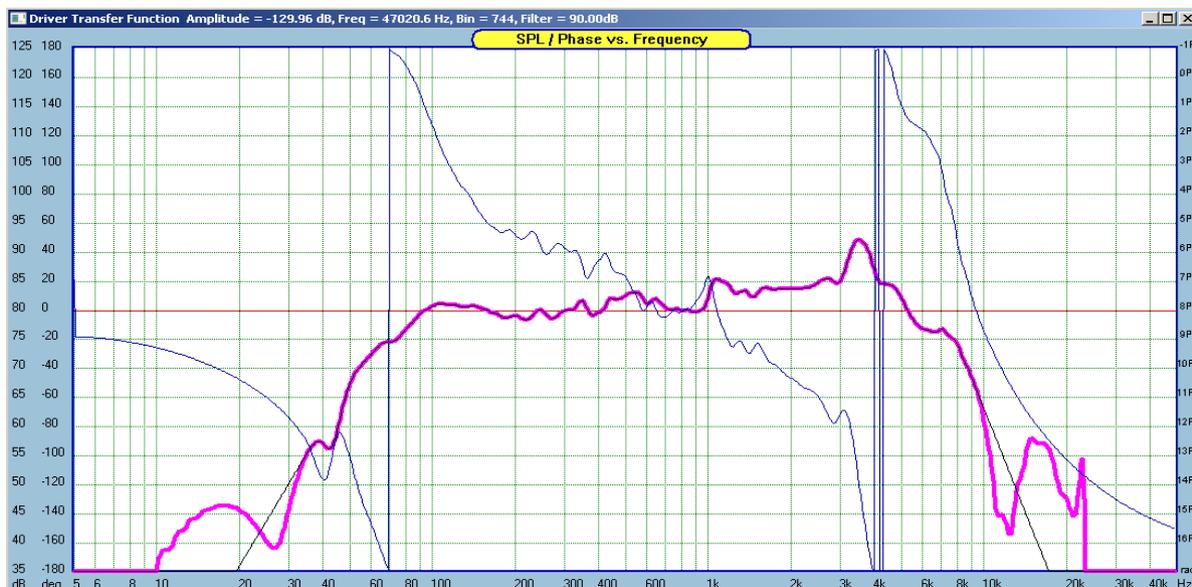
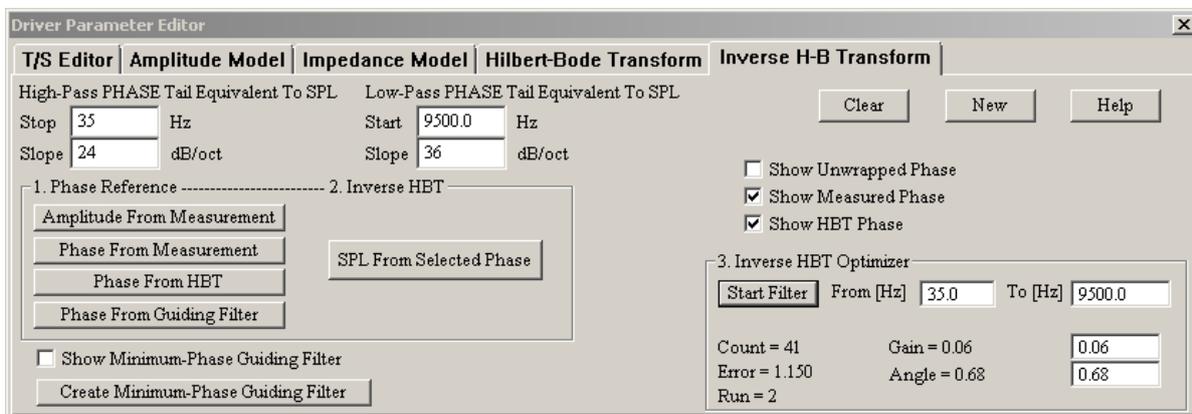


Figure 16.262. HBT used to nominate initial slopes.

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.

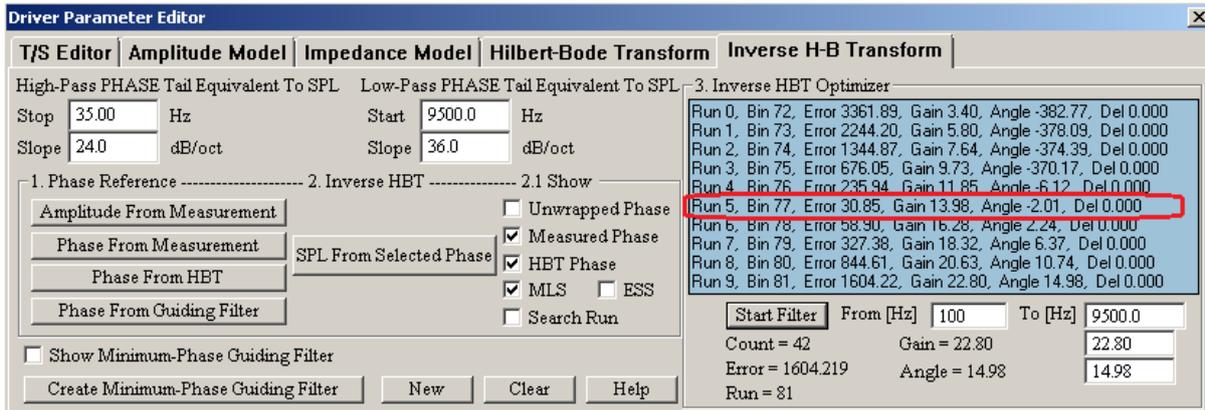


Figure 16.263. Error results in Stage I.

Shown below, are the 10 SPL curves re-created by the IHBT algorithm. Please note, that the vertical scale resolution has been enhanced 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.

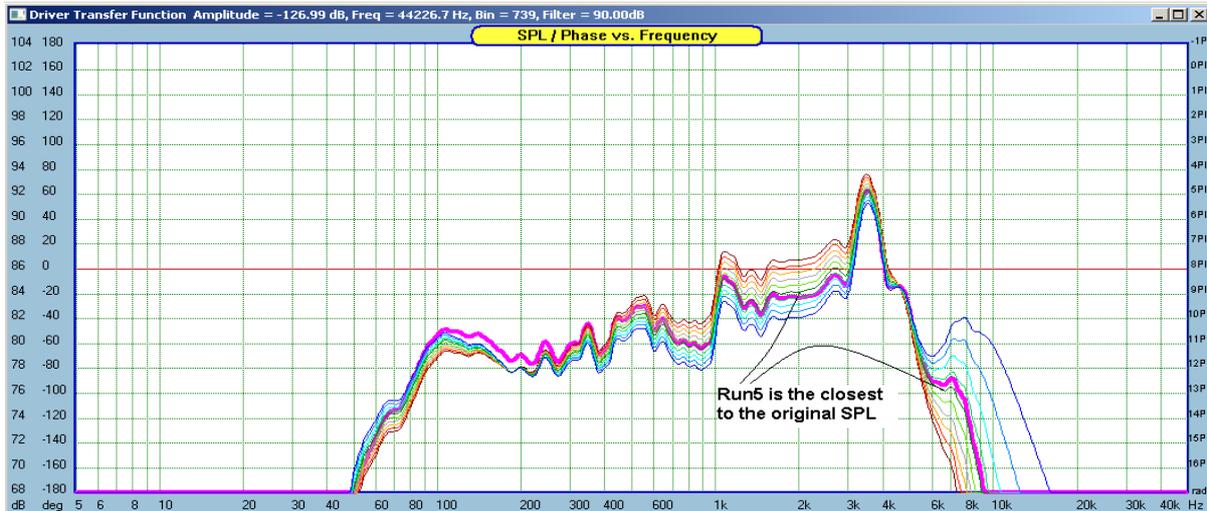


Figure 16.264. SPL Error results in Stage I.

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.

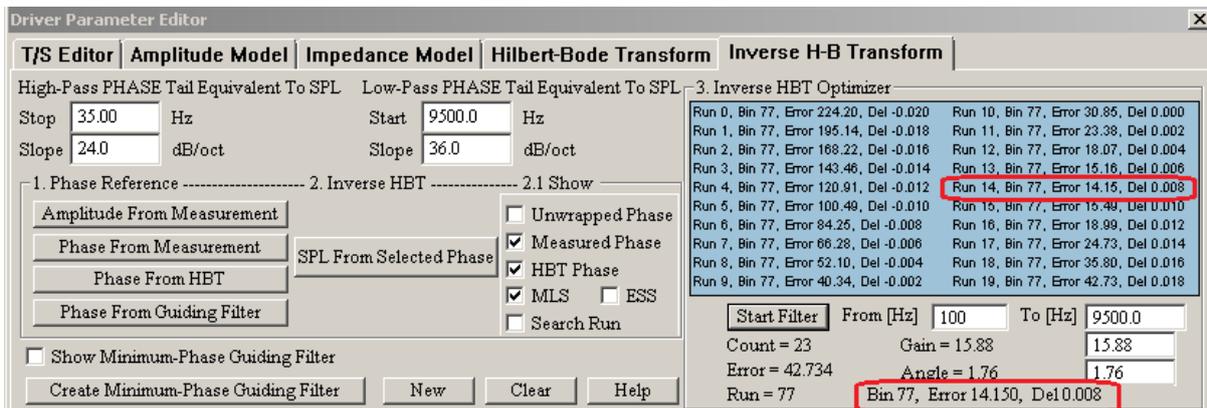


Figure 16.265. Error results in Stage I.

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

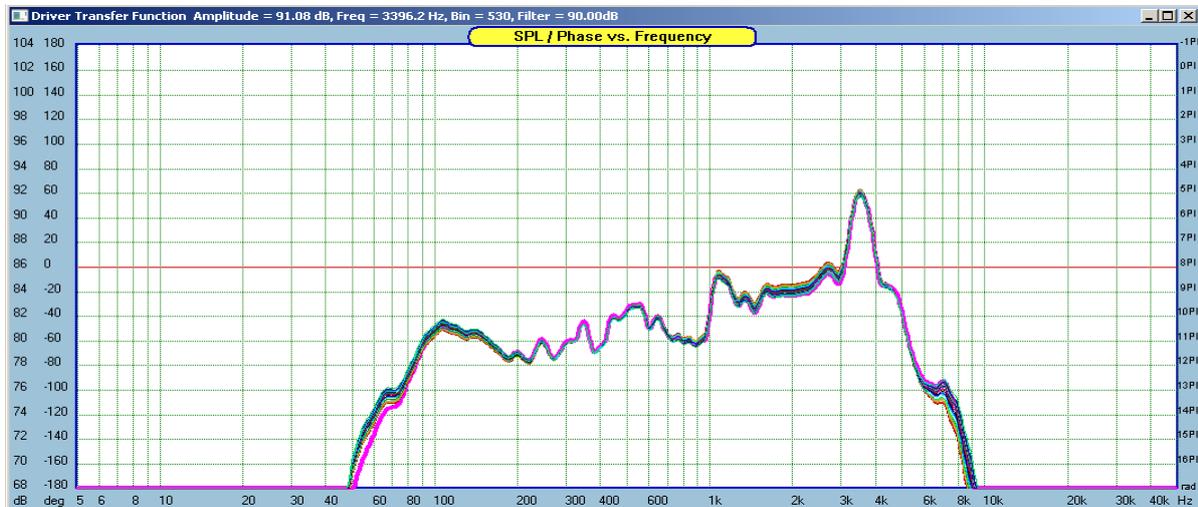
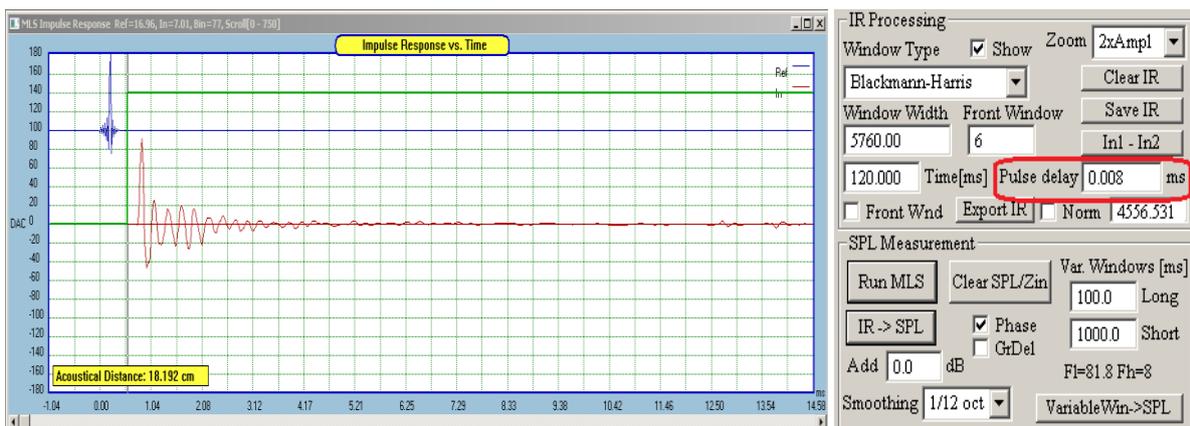


Figure 16.266. SPL Error results in Stage II.

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.

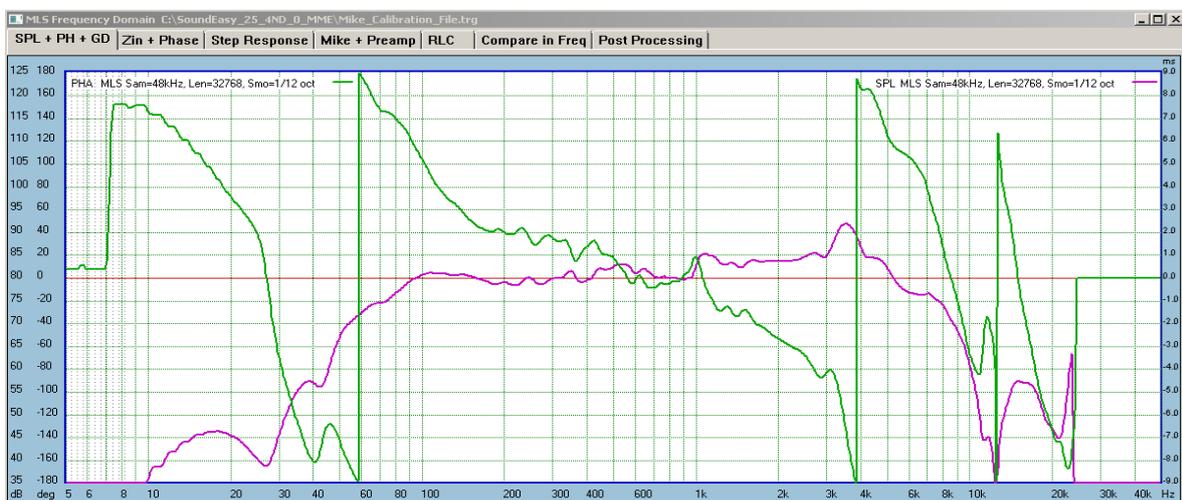


Figure 16.267. SPL/Phase after adding 0.008ms delay

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.

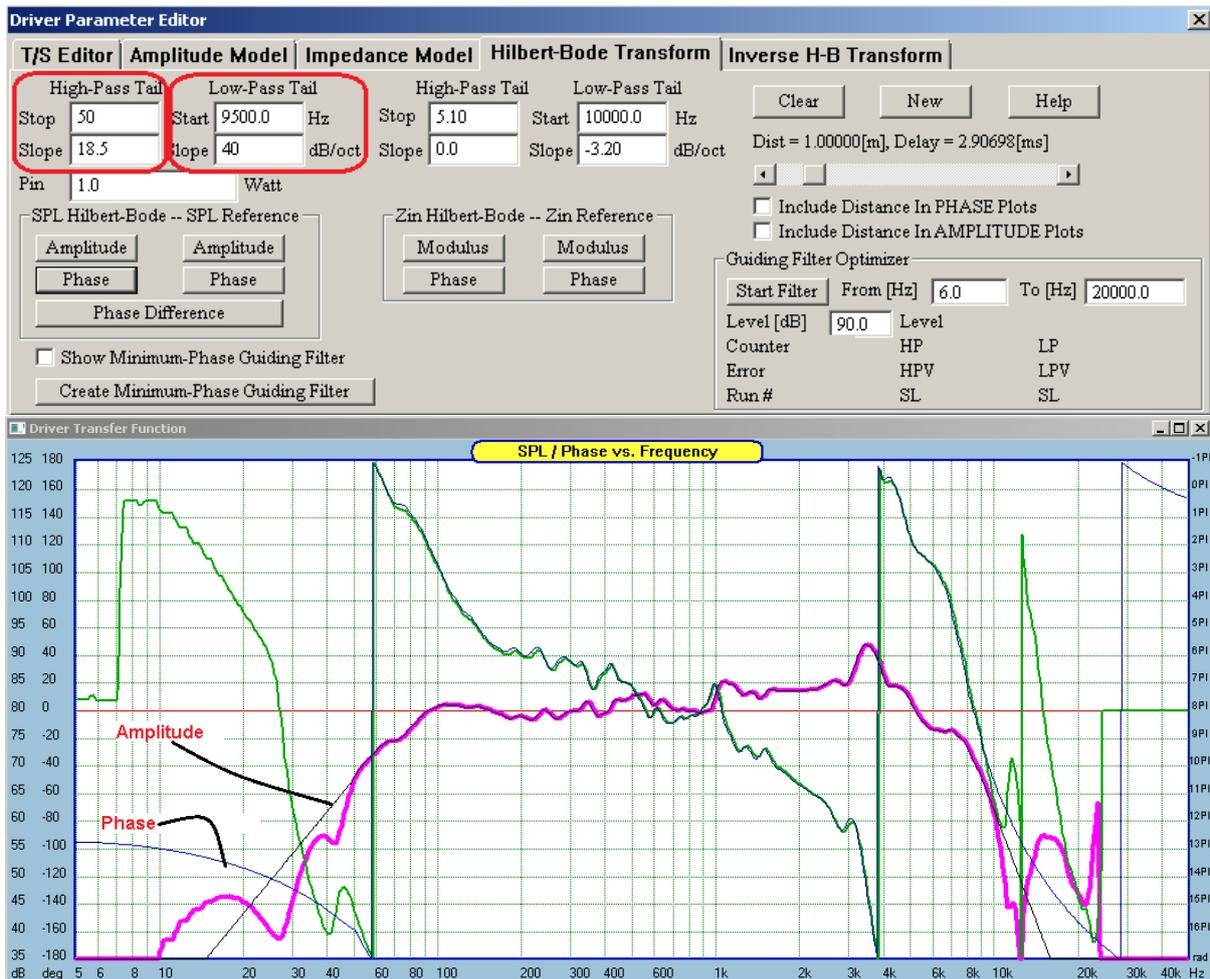
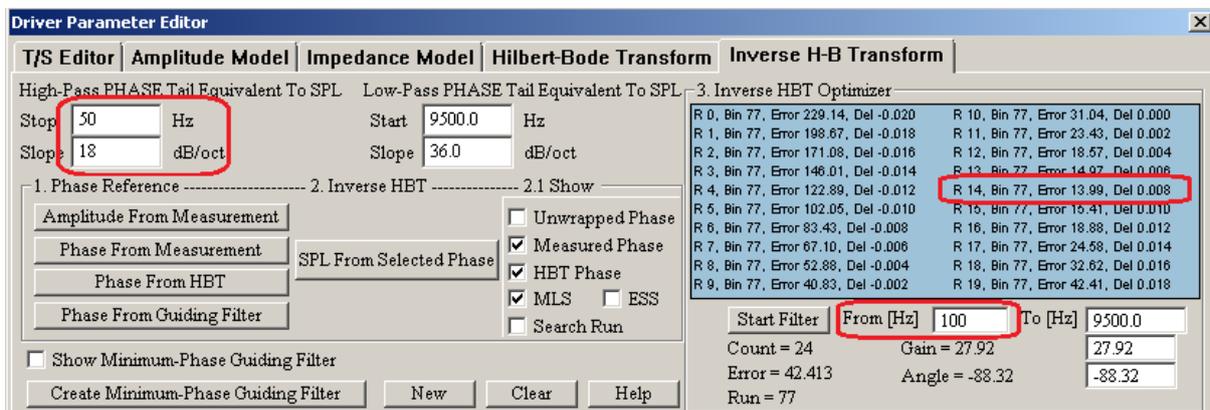


Figure 16.268. Final Minimum-Phase phase response obtained via HBT

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.



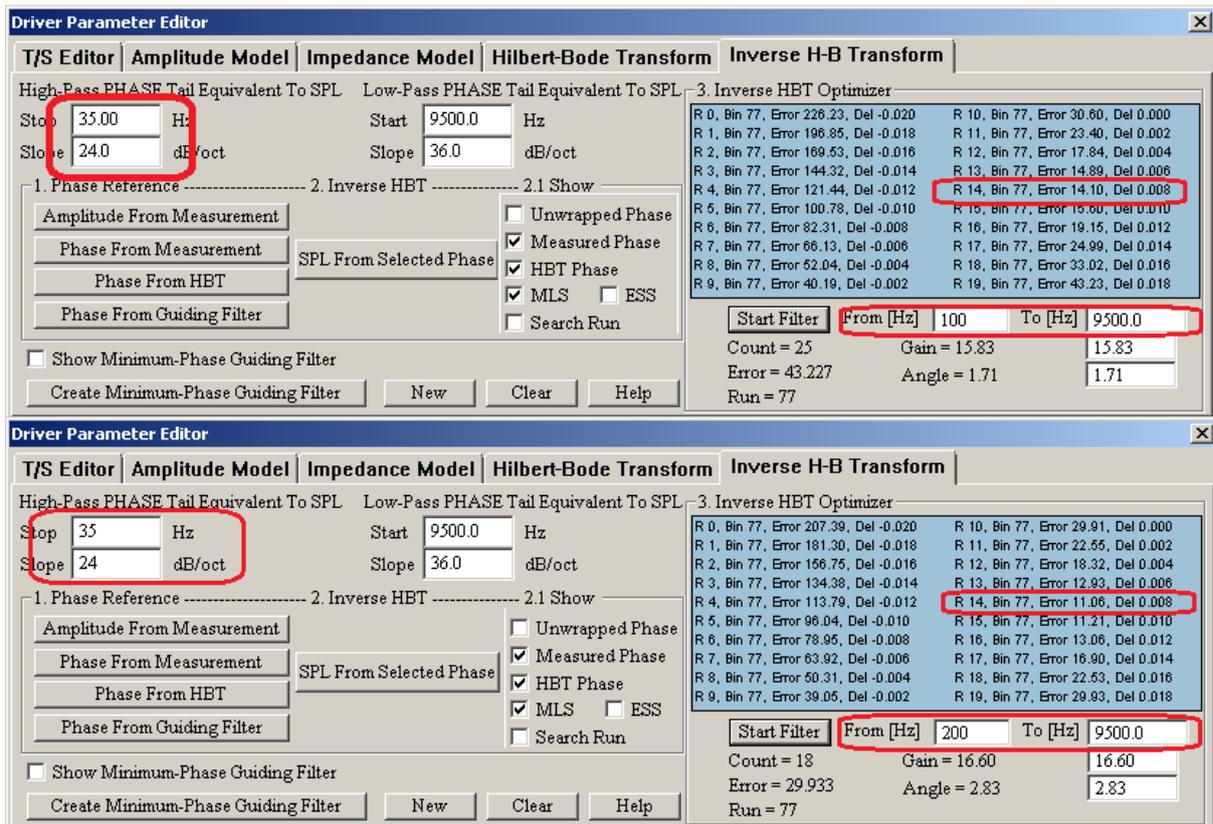


Figure 16.269. The algorithm always selects Bin 77 with the minimum error.

Summary

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

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

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.

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

Having extracted minimum-phase phase response, it is easy to calculate corresponding time-of-flight. This would be $TOF = \frac{\text{number_of_bins}}{\text{sampling frequency}} + \text{Added Delay}$.

Knowing TOF and the speed of sound, we can calculate “Acoustical Distance”

Acoustical Distance: 18.192 cm

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.

Loudspeaker Driver Characterization

Ideally, loudspeaker driver data associated with any particular driver, would be comprehensive enough to enable the system designer to enter all this information into a CAD program and create at least a “first cut” design of the intended system. If the first cut design passes initial criteria of the expected performance, then the design process becomes validated and established and a new loudspeaker system will hopefully see the light of the day. This would be my approach to loudspeaker design. It’s not the only one, and other methods have been used too. Loudspeaker data bases exist already. But to my knowledge, none of them supports driver “pick-and-place” design freedom.

So, what information would have to be associated with a loudspeaker driver to make it usable across a number of possible trials for the first cut design?.

Historically, we could start with simple electro-mechanical data with of SPL curves measured with drivers in boxes in anechoic chambers. Here is an example of Philips Components And Materials data, Part 3b, from October 1978 – Loudspeakers.

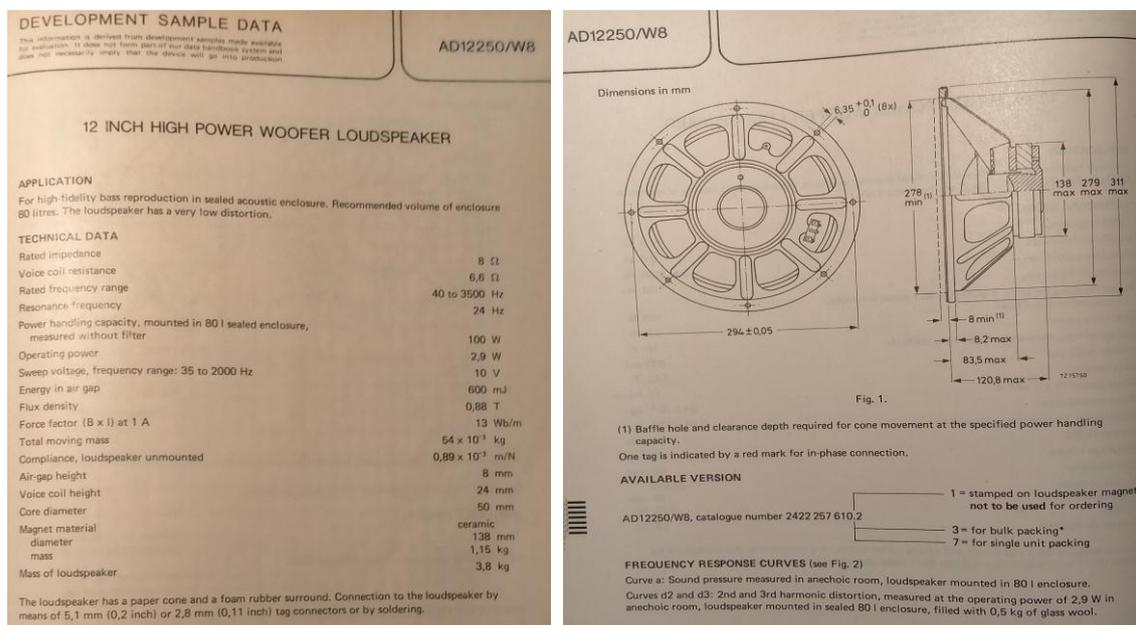


Figure 16.270. Loudspeaker specification in the 70'.

Please note, that “Curve a” on the picture below, is the driver’s SPL curve measured in anechoic chamber, with loudspeaker mounted in 80lt enclosure, filled with 0.5kg of glass wool. “Operating Power” was the sine wave power required the loudspeaker to operate at 96dB SPL level. It was 2.9 watts in this case. So far, no mention of Thiele/Small (small-signal) parameters.

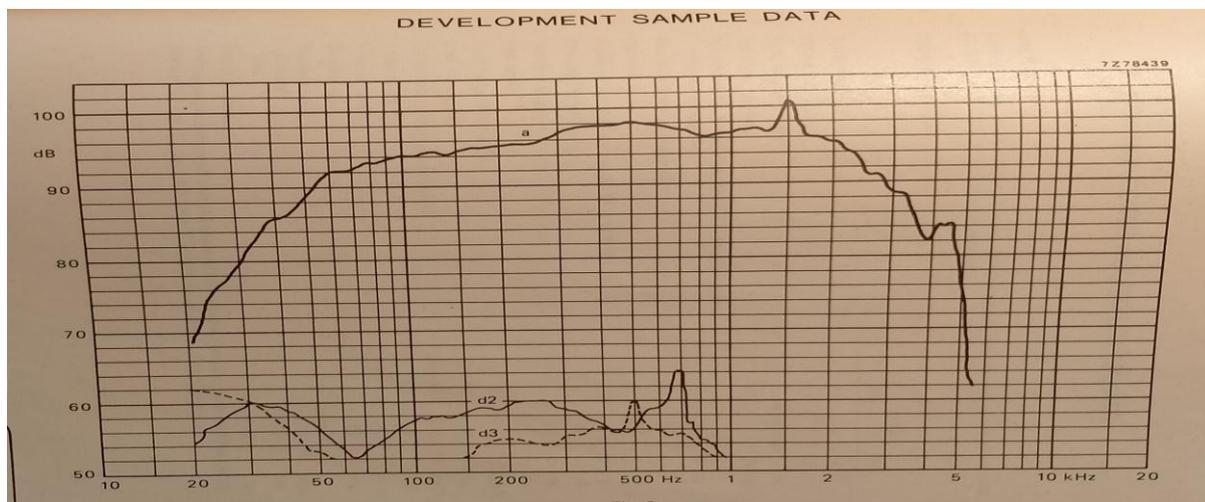


Figure 16.271. Loudspeaker specification in the 70'.

Well, that was year 1978, and we have made good progress from those years.

The Thiele/Small (small-signal) parameters were fundamental enablers of the birth of computer CAD programs, which would take T/S parameters and would allow the designer to model driver's behaviour in various enclosure sizes and types. Next improvement was inclusion of on-axis and off-axis SPL curves and impedance curves data. Voice coils were characterised more accurately, resulting in several models being used for modelling input impedance of a driver. We could go on....but this is not the purpose of this manual. Even today, loudspeaker manufacturers do not provide loudspeaker specifications/characterizations – ideally computer data files, which would allow the designer to load the file into the current loudspeaker project and predict (model) the proposed system performance characteristics.

What would it take to change current situation and characterize the loudspeaker driver to a degree, where you could plug-in any driver into the CAD project and predict at least a few basic performance curves: the acoustic on-axis/off-axis SPL/Phase, Zin/Phase, Cone Displacement and limits, and so on?.

What data is needed?

Before listing some basic requirements, we must stress, that the fundamental purpose of the existence of the database is to provide the designer with “minimum cost design path” to achieve the design goal. This requirement translates into model-before-buy principle. So that you would only start investing into the hardware, once you convinced yourself, that the design will fly.

Loudspeaker data collected in its file should allow the following design freedoms and more:

1. Measure the driver's SPL/Phase and Zin/Phase once only.
2. Allow for various baffle placements of the driver.
3. Support for unrestricted movement of all drivers on the front baffle.
4. Allow for unlimited variants of system design (2-way...5-way, D'Appollito, arrays, etc....).
5. Support measurements performed at one distance (e.g.; 1m), but design and optimization executed at different distance (e.g.; 2m).
6. Allow the design of different baffle sizes and shapes.
7. Support different enclosure sizes and types.
8. Support off-axis modelling, even with only on-axis measurement data in the driver file.
9. Provide support data for non-linear analysis of the cone excursion.
10. Provide support for vent performance analysis.
11. Provide SPL/Phase and Zin/Phase data over 750-1000 data point spaced logarithmically from 5Hz-50000Hz.
12. Facilitate linear-phase loudspeaker design.

Regarding Item 1

Just like T/S parameters, loudspeaker driver is uniquely characterised by its SPL/Phase and Zin/Phase performance curves. This data is tied to this particular driver and does not change in the presence or absence of other drivers in the system. Therefore, it makes sense to measure the driver once only and include this data permanently in the driver file. The critical issue here is the universal, standard measurement reference point for SPL/Phase measured data. Since the loudspeaker driver is a minimum-phase device it makes sense to define the “acoustic centre” as the point in space, where the acoustic radiator's (the driver) transfer function assumes minimum-phase characteristics. Up until now, the stumbling block was the difficulty with accurate extraction of the minimum-phase phase response. The method described in my previous papers removes this problem, so one can accurately extract the minimum-phase and then calculate the acoustic centre (as defined above) from there. As it will be demonstrated later, CAD modelling software should be able to predict all variants of SPL/Phase modelling, when given the SPL/Phase minimum-phase measurements and the “acoustic centre” offset, where “offset” is defined as positive or negative distance to the acoustic centre from the mounting baffle.

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

Regarding Item 2, 3 and 6

Obviously, one would have to record the original measurement conditions, such as the dimensions of the front baffle, and driver location on the front baffle. It goes without saying, that measurements are conducted on-axis in free-field. Why do we need this?.

Let's assume, that we have completed Item 1 above. So, that driver mounted in its enclosure, has been measured at, say 1m distance. The measurement result includes diffraction effects for this particular driver location on the front baffle, the size of the front baffle and measurement distance. Given all the above, we can calculate the diffraction effects and subtract them from the measured SPL/Phase curve. What's left is a "diffraction-less" SPL/Phase transfer function. That's exactly what we need, because now, we can design different baffle size and shape, place the driver in different location on the baffle, and re-calculate the new diffraction curve. And then add this new curve to the "diffraction-less" SPL/Phase curve.

In summary, we can take minimum-phase measurements performed on one baffle, and transfer those onto a different baffle. This is actually very valuable characteristics of the process. Because the driver can be measured by one user, and the data can be shared by all other users incorporating the same driver in their designs.

Regarding Item 4

Just as an example, one day, you may decide to tackle 5.1 HT system design. One possible option is to make Front-Right and Front-Left speakers as 3-way systems, the Centre Channel as 2-way D'Appolito configuration and Rear-Right and Rear-Left as smaller, 2-way systems.

The design would be less time consuming, if you incorporated the same, trusted tweeter in all boxes. The tweeter will interface with different drivers in each box. In the 3-way system, the tweeter interfaces with a midrange driver, in the D'Appolito system, that tweeter interfaces with two woofers, and in the 2-way system, the tweeter interfaces with single woofer. All enclosures have different sizes and different driver locations. So, this is just a simple example of possible driver and mechanical variations in the design.

However, none of those are problems from modelling/design point of view. As long as you have the minimum-phase SPL/Phase curves and acoustic offsets defined as per Item 1, your CAD software will be able to model all those configurations. Without re-measuring drivers or configurations.

Regarding Item 5

We have already mentioned, that the original measurement distance between the front baffle and microphone needs to be recorded. Why? – Because we need this distance to correctly calculate diffraction of the original measurement. Once we subtract the original diffraction (and make the driver diffraction-less) we can re-calculate new diffraction at any distance we desire. Some designers prefer to optimize their systems at listening distance of 2, 3, or even 4 meters. Modelling at different distances will be facilitated in your CAD software, as long as you followed Item 1.

Regarding Item 7

Sealed-back drivers, like midrange and tweeters are not affected by enclosure volume. So, these drivers can be moved from one baffle/enclosure to another, without considerations for enclosure volume or type. Woofers are affected by enclosure diffraction differences, volume and type (e.g.: sealed or vented). Diffraction was explained above, therefore, we can explore the issue of low-frequency driver loading. Fortunately, the lumped-element TS model works quite well in the frequency range below 100-150Hz. Therefore, a woofer driver properly measured in sealed box, can be modelled in a vented box by using "splicing method". Here, the low-frequency SPL/Phase response is modelled for a vented enclosure and spliced around 100-150Hz with the measured SPL/Phase minimum-phase curve. If there is any phase discontinuity, it can be fixed with HBT. This is possible, because if you followed Item 1, then you know what the minimum-phase phase response should look like. Please note, that you would apply similar rules when dealing with Zin/Phase curves.

Regarding Item 8

Off-axis modelling is typically accomplished by approximating the driver's radiating surface with a number of discrete points and calculating combined SPL from all radiating points at any desired point in the surrounding hemisphere, accounting for ever-changing diffraction. Calculations are based on more (or less) complicated geometry. But as long as you followed Item 1, your CAD should be able to handle these calculations. Impedance curves are not affected here.

Regarding Item 9 and 10

If the drivers' motor electro-magnetic, material and mechanical data is included in the driver's characterization, then your CAD software may be able to model suspension and BL-related performance for Large-Signal Analysis. This is outside the scope of this paper, and is only mentioned here for completeness.

Regarding Item 11

There is some value in modelling loudspeaker system performance somewhat outside the designated operating bandwidth. The bandwidth proposed in Item 11 is just one of the options.

Regarding Item 12

One of the most useful test signals in electronics is a humble square wave. The "ideal" square wave is a superposition of an infinite number of sine waves, each contributing its required amplitude and phase. It is due to this very feature, that when passed through an audio system, the square wave can reveal time domain performance issues of the system. This is because all of its sine wave components must be passed by the system without time distortion, or different delays, in order to recombine as a square wave at the output of the system under test – your loudspeaker. https://www.bodziosoftware.com.au/Square_Wave.pdf

The linear-phase loudspeaker system is the only system that will reproduce square wave. And not only that, it will correctly reproduce impulsive sounds, undistorted "punch", it will aid localization of the sound sources, removes "flobby" bass and more.

The loudspeaker will maintain its linear-phase characteristics over significant listening angle. <https://www.bodziosoftware.com.au/Linear%20Phase%20Loudspeakers.pdf>

A simple method of designing the linear-phase DSP loudspeaker system is to (1) equalize driver's SPL, then (2) invert driver's phase response. Each driver in the system is treated the same way, but separately. Each driver becomes an on-axis sound source with flat SPL and flat phase. Finally, acoustic offsets are equalized for the listening area by adding appropriate small delays. Done.

Given the above, once again, drivers characterized as per Item 1, are ready to go into the CAD modelling process right away, without any other "relative" considerations.

Example of a current driver characterization

Shown below is an example of another loudspeaker. This time, it's currently available JBL 18" Low Frequency Transducer, 2241H. BTW – it's one of the best 18" drivers I have ever seen. I use two of them in separate boxes in my bass guitar rig.



2241H 460 mm (18 in) Low Frequency Transducer

Professional Series

Key Features:

- ▶ 600 W continuous pink noise power capacity
- ▶ 100 mm (4 in) edgewound aluminum ribbon voice coil
- ▶ 30 Hz - 3 kHz response
- ▶ 98 dB sensitivity, 1 W, 1 m (3.3 ft)
- ▶ SFG magnet structure with patented Vented Gap Cooling™ technology¹



Specifications:

Nominal Diameter:	460 mm (18 in)
Rated Impedance:	8 ohms
Power Capacity ¹ :	600 W continuous pink noise
Sensitivity ² :	98 dB SPL, 1 W, 1 m (3.3 ft)
Frequency Range ³ :	30 Hz - 3 kHz
Power Compression ⁴	
at -10 dB rated power (60 W):	0.8 dB
at -3 dB rated power (300 W):	2.6 dB
at rated power (600 W):	4.3 dB
Distortion ⁵	
2nd harmonic:	≤ 1.0%
3rd harmonic:	≤ 1.0%
Highest Recommended Crossover:	800 Hz
Recommended Enclosure Volume:	140-340 l (5-12 ft ³)
Effective Piston Diameter:	397 mm (15.6 in)
Maximum Excursion Before Damage (p-p):	40 mm (1.6 in)
Minimum Impedance:	6.0 ohms ± 10% @ 25°C
Voice Coil Diameter:	100 mm (4 in)
Voice Coil Material:	Edgewound aluminum ribbon
Voice Coil Winding Depth:	19.05 mm (0.75 in)
Magnetic Gap Depth:	8.1 mm (0.32 in)
Magnetic Assembly Weight:	6.8 kg (15 lb)
Bl Factor:	19.2 N/A
Effective Moving Mass:	0.145 kg

Positive voltage on BLACK terminal gives forward diaphragm motion.

¹ AES standard (50-500 Hz).

² Sensitivity is based on a swept 100 Hz to 500 Hz signal for an input of 2.83 V @ 8 ohms or 2.0 V @ 4 ohms.

³ Frequency range is defined as the frequency extremes where the response is -10 dB from the rated sensitivity.

⁴ Power compression is the sensitivity loss at the specified power, measured from 50 to 500 Hz, after a 5 minute AES standard (50-500 Hz) pink noise preconditioning test at the specified power.

⁵ Distortion is measured at -10 dB rated power, from 100-500 Hz.

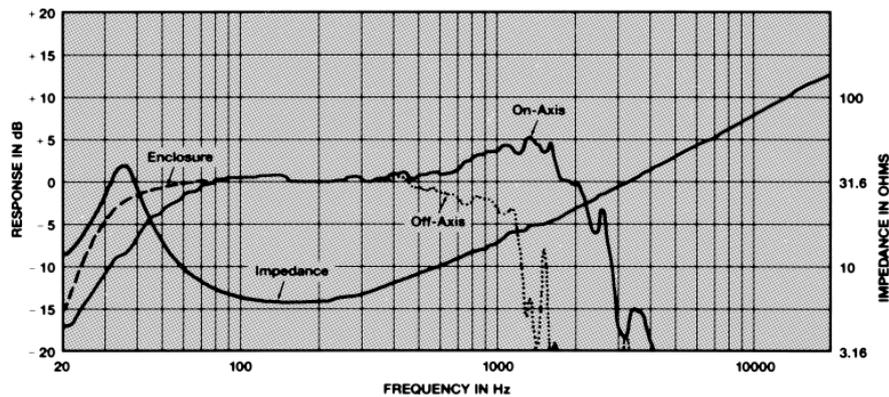
Figure 16.272. More contemporary loudspeaker specification.

THIELE/SMALL PARAMETERS ⁶ :	
f_s :	35 Hz
R_e :	5.0 ohms
Q_{ts} :	0.40
Q_{ms} :	5.7
Q_{es} :	0.43
V_{as} :	310 l (11.0 ft ³)
S_D :	0.124 m ² (192.4 in ²)
X_{max} :	7.6 mm (0.30 in)
V_D :	942 cm ³ (57.7 in ³)
L_e :	1.75 mH
η_o (Half space) ⁷ :	2.9%
P_e (Max) ¹ :	600 W continuous pink noise
MOUNTING INFORMATION:	
Overall Diameter:	464 mm (18¼ in)
Bolt Circle Diameter:	441 mm (17¾ in)
Baffle Cutout Diameter	
Front Mount:	427 mm (16 ³ / ₁₆ in)
Rear Mount:	428 mm (16 ⁵ / ₆₄ in)
Depth ⁷ :	191 mm (7½ in)
Volume Displaced by	
Driver:	8.51 (0.3 ft ³)
Net Weight:	10 kg (22 lb)
Shipping Weight:	10.9 kg (24 lb)

⁶Thiele/Small parameters are measured after a 2 hour exercise period using a 600 W AES power test and will reflect the expected long term parameter values once the driver has been installed and operated for a short period of time.

⁷Clearance of at least 76 mm (3 in) must be provided behind the magnet assembly and the gap vents to allow sufficient air circulation and proper cooling to take place.

Typical Response and Impedance Curves, Enclosure Volume and Port Tuning



Frequency response contour of the 2241H taken in a hemispherical free-field environment, a closed box of 280 l (10 ft³) internal volume enclosing the rear of the driver. Measured response of a typical production unit, including all peaks and dips, does not deviate more than 2 dB from the above curve. The dotted line represents measured 45 degree off-axis response. The dashed curve represents the response when the driver is mounted in a 280 l (10 ft³) vented enclosure tuned to 30 Hz using a port with an area of 320 cm² (50 in²) and a length of 20 cm (8 in). The impedance magnitude curve is measured in free-air.

Figure 16.273. More contemporary loudspeaker specification.

Please note, that the driver was measured in anechoic chamber in sealed and vented enclosures of 280lt volume.

Conclusions

For accurate design, one will need accurate measurements. For efficient design, you'll need a database of drivers with accurate measurements and pick-and-place approach to achieve first-cut design stage - fast.

There is a simple, low cost way of standardizing and publishing the essential on-axis measurement results of loudspeaker drivers. Such measurement results are referenced to a common and clearly defined minimum-phase response and the associated point in space – acoustic centre. If implemented, either by loudspeaker manufacturers, or as DIY community shared measurements, this process has the potential of reducing developmental costs, reducing development time, increase design freedom, and increasing accuracy of the modelling process.

The method described in the previously mentioned papers will correctly extract minimum-phase response, even from tweeter's SPL measurements restricted to Nyquist Frequency of 24kHz (half the sampling frequency of 48kHz). The bulk of the method is automated, and can be implemented on a typical MLS or ESS dual-channel measurement system. This is the essence of this method – minimum-phase response and the acoustic centre are measurable using a very unsophisticated test equipment. All you need is an old Windows 7/10 PC with a reasonable sound card to run the measurement system. It is really as simple as that.

As explained in the paragraphs above, loudspeaker driver measurements taken at one specific set of test conditions, can be “deconstructed” and then “reconstructed” for another set of test conditions, for the purpose of achieving the first-cut design. Therefore, it makes sense to measure the driver only once, and then adopt/modify these results to your individual circumstances.

As shown in the above example of the JBL brochure (and many others, not shown here), loudspeaker manufacturers already measure their drivers under some acoustically favourable test conditions. So, the SPL curve is already available. But, you are only being shown enough to entice you into purchasing the driver. What is needed, is the design data that would allow you to proceed with CAD modelling.

With a minimum cost and effort, and using the method described above, one could easily extend the SPL measurement into SPL/Phase measurement and store automatically extracted minimum-phase and SPL responses even in a TEXT file. This file would be published on the manufacturer's website for each driver.

Imagine that happening ☺.

Automated ANSI-CTA-2034 Measurements

Loudspeaker measurements have gone well beyond simple on-axis characterizations. The ANSI-CTA-2034 Standard “.....describes an improved method for measuring and reporting the performance of a loudspeaker.....”. The Standard does not however provide for a strict design rules, or numbers, that you would use to fine tune and optimize your design.

Publically available scientific publications provide some guidance as to how the on-axis SPL curve should behave, or “listening window” or sound power curves should perform. So, there is some way from the ANSI standard measurements to the design CAD software, that would use collected measurement data for the purpose of designing a loudspeaker system performing to acceptable level established by basically the designer himself/herself.

Upon opening “MLS Turntable” TAB, you will be confronted with 6 plotting screens and one dialogue box. Shown below is a suggested arrangement, but you may move them around and re-size at will.

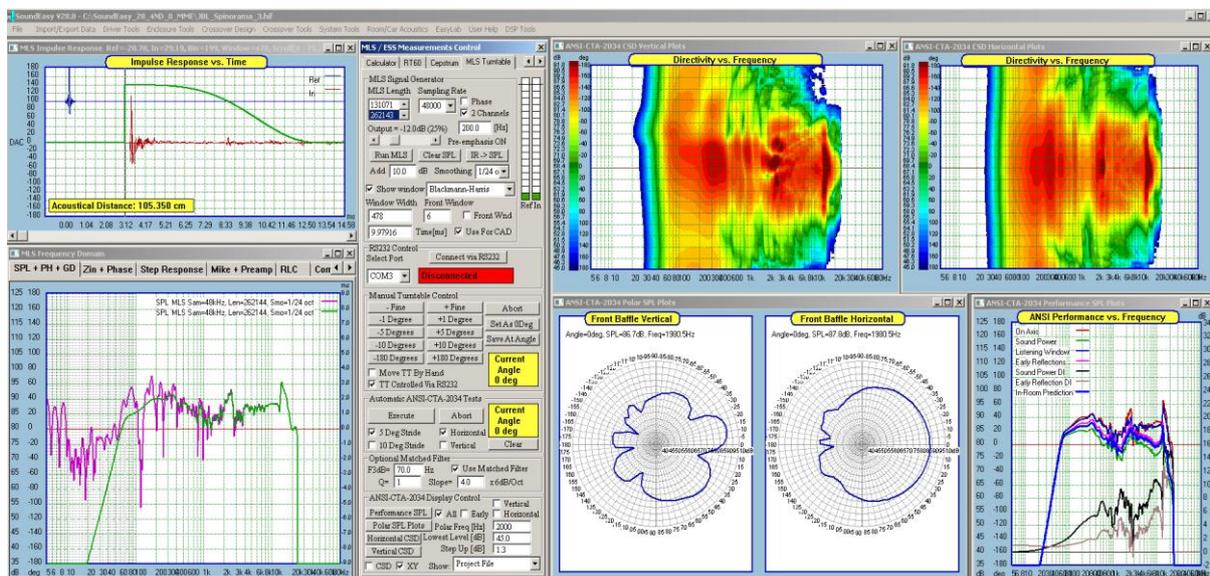


Figure 16.274. ANSI-CTS-2034 measurement system.

The main function of each screen can be described as follows:

1. The Impulse Response screen confirms the measurement is taking place and shows FFT windowing arrangement.
2. The SPL measurement plotting window shows frequency response derived from the captured impulse response.
3. As the polar response measurements is progressing, the horizontal and vertical XY/CSD windows show collected angular responses in both: horizontal and vertical planes.
4. There is also a standard 360deg polar plot for horizontal and vertical planes at the frequency selected by the user.
5. Next, a variety of ANSI-CTA-2034 performance curves are presented on a separate screen.
6. Finally the “MLS Turntable” TAB is the control dialogue box with a number of controls available on it.

MLS Turntable TAB

Shown below is the dialogue box controlling all functions necessary to perform either manual or automated ANSI-CTA-3024 tests.

The screenshot shows the 'MLS / ESS Measurements Control' dialog box with several sections highlighted by red boxes:

- MLS Signal Generator:** Includes controls for MLS Length (131071, 262143), Sampling Rate (48000), Phase (checked), 2 Channels (checked), Output (-20.0dB (10%), 0.0 Hz), Pre-emphasis (OFF), Run MLS, Clear SPL, IR -> SPL, Add (10.8 dB), Smoothing (1/24), Show window (checked), Blackmann-Harris, Window Width (1146), Front Window (6), Front Wnd (unchecked), 23.8750 Time[ms], and Use For CAD (checked).
- RS232 Control:** Select Port (COM3), Connect via RS232, Selected Port COM3.
- Manual Turntable Control:** Buttons for - Fine, + Fine, Abort, -1 Degree, +1 Degree, Set As 0Deg, -5 Degrees, +5 Degrees, Save At Angle, -10 Degrees, +10 Degrees, -180 Degrees, +180 Degrees, Move TT By Hand (unchecked), TT Cntrled Via RS232 (checked), and Current Angle 0 deg.
- Automatic ANSI-CTA-2034 Tests:** Execute, Abort, Current Angle 0 deg, 5 Deg Stride (checked), Horizontal (checked), 10 Deg Stride (unchecked), Vertical (unchecked), Clear.
- Optional Matched Filter:** F3dB= 80.0 Hz, Use Matched Filter (unchecked), Q= 0.70, Slope= 4.0 x 6dB/Oct.
- ANSI-CTA-2034 Display Control:** Performance SPL, All (checked), Early (unchecked), Vertical (unchecked), Horizontal (unchecked), Polar SPL Plots, Polar Freq [Hz] (1700.0), Horizontal CSD, Lowest Level [dB] (45.0), Vertical CSD, Step Up [dB] (1.3), CSD (checked), XY (unchecked), Show: Project File.

MLS Singal Generator
All essential controls are available here to perform MLS measurement

RS232 link control for connecting turntable

All test can be run manually. Even with a lazy-susan moved by hand

ANSI-CTA-2034 tests automatically run with turntable

Bass can be clean up

Close/Open all screens and control the display from here

Figure 16.275. ANSI-CTS-2034 measurement system control TAB.

MLS Signal Generator

This is a selection of the basic and familiar MLS control items. It is a subset of the controls available from the main MLS System TAB. It has been duplicated here on the “MLS Turntable” TAB for convenience and ease of operation.

Once the measurement system components, like measurement microphone, testing amplifier, cabling and so on, has been assembled and connected, you may want to start taking basic measurements to confirm the correct operation of the MLS measurement system. This includes positioning the FFT window, adjusting the measurement levels, positioning the turntable at the 0deg starting position. Basically, do not leave this section until you are satisfied, that quality measurements can be accomplished.

The “Use For CAD” checkbox enables saving the ANSI turntable test results between +90..-90deg angles in a Driver and Project Data Files. This data is treated as standard SoundEasy data. So, this may be useful for some future CAD work.

The MLS system has been described in details in this Manual and the operation of each controls will not be repeated here.

RS232 Control

The turntable is connected via USB/RS232 converter cable. The driver for the cable (the PL-2303 USB-to-Serial driver) together with very simple installation instructions is available from the turntable supplier. Once the driver is installed, your Windows Device Manager will indicate which port is available for communication with the turntable.

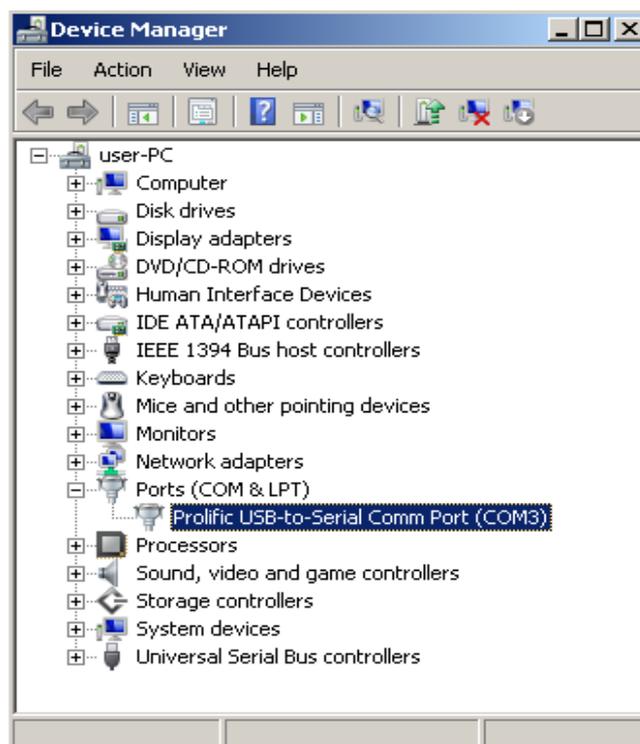


Figure 16.276. Serial Port control in Windows.

In case of my computer – it is Port COM3.

Please make sure, that the turntable is powered up and USB/RS232 cable is connected to the computer



Select the required com port from the list



and press “Connect via RS232” button.



Figure 16.277. Activating Serial Port control in ANSI TAB

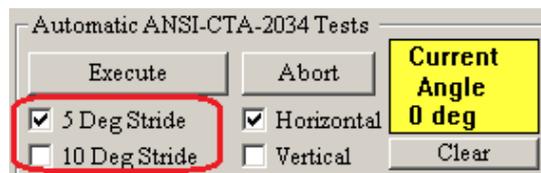
You should see the sequence of feedback coming from controls as shown above.

Manual Turntable Control

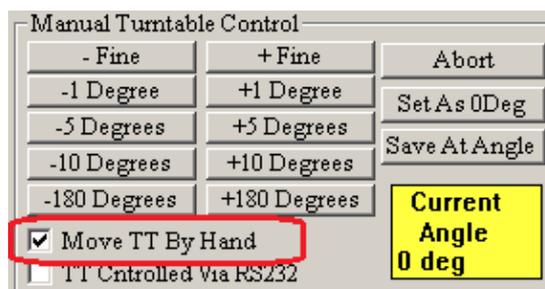
This section facilitates two modes of collecting polar performance data:

1. Manual by hand – this would be applicable if you built a simple “lazy-Susan” turntable arrangement and wish to collect the polar performance by manually moving your turntable in 5deg or 10deg increments and taking the measurements as you progress along the 360deg set of angles.

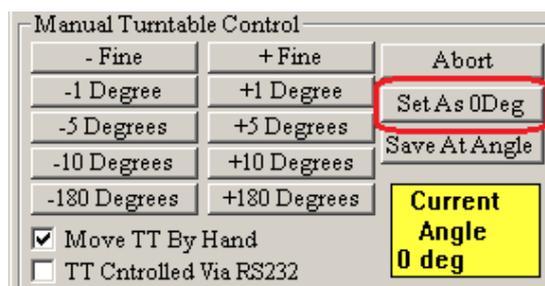
Start by nominating the stride (5 or 10 degrees). And start with horizontal measurements.



Select “Move TT By Hand” option

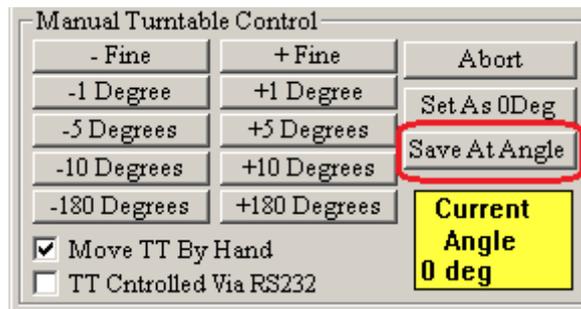


Move the turntable to the starting 0deg position by hand and press “Set As 0deg” button. This will be confirmed by the “Current Angle” yellow display field.

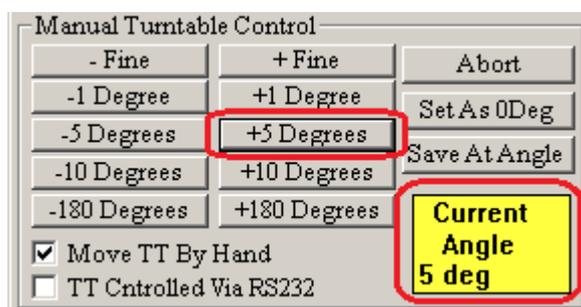


Now it's a time to take your first measurement at 0deg, so take the MLS measurement and make sure, you display the resulting frequency response SPL on the provided screen.

Finally, press “Save At Angle” button – **this will save the measured SPL curve in the program’s data field allocated for 0deg angle**. The curve will be displayed in CSD screen.

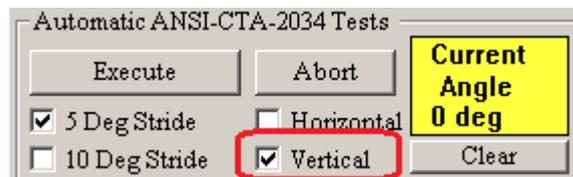


Advance the turntable to the next angle by hand and press “+5 Degrees” button. The current angle will be confirmed as +5 deg.



Take the next MLS measurement and press “Save At Angle” button – **this will save the measured SPL curve in the program’s data field allocated for 5deg angle**.

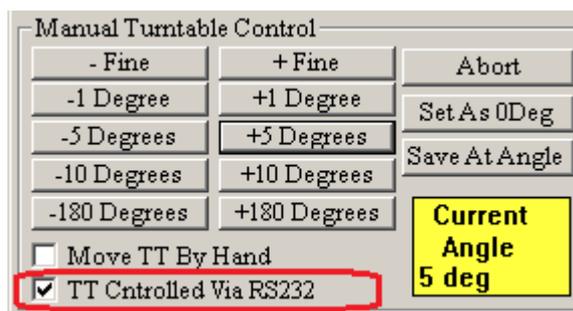
Continue till 360deg horizontal measurements are completed. Then select “Vertical” check box and repeat the whole process for vertical measurements.



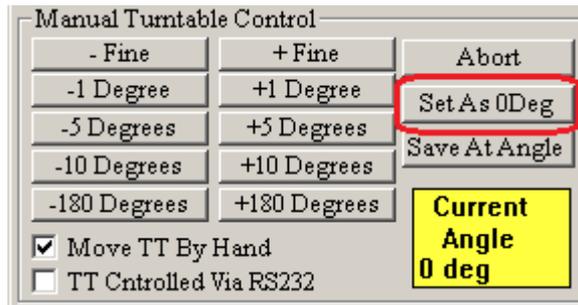
So, this is how the manual, “by-hand” mode works.

2. Manual Model using the RS232 turntable – this would be applicable if you have the actual RS232 controlled turntable.

Start by selecting the “TT Controlled Via RS232” checkbox.

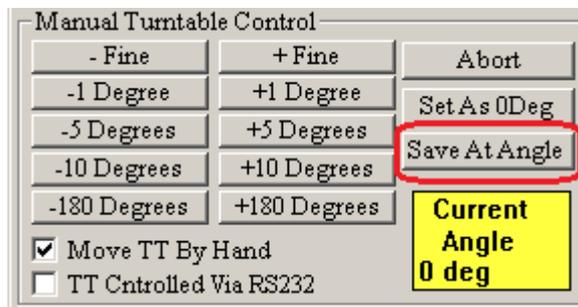


Move the turntable to the starting 0deg position by pressing “+1 Degree”, “-1 Degree”, “+Fine”, “-Fine” buttons and press “Set As 0deg” button. This will be confirmed by the “Current Angle” yellow display field.

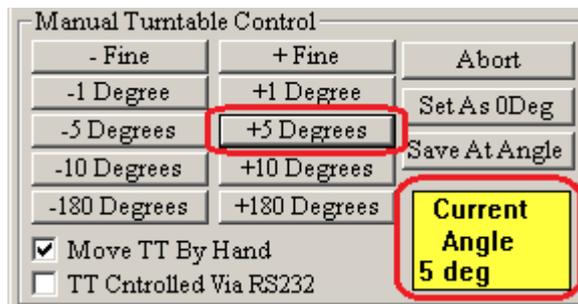


Now it's a time to take your first measurement at 0deg, so take the MLS measurement and make sure, you display the resulting frequency response SPL on the provided screen.

Finally, press “Save At Angle” button – **this will save the measured SPL curve in the program's data field allocated for 0deg angle.** The curve will be displayed in CSD screen

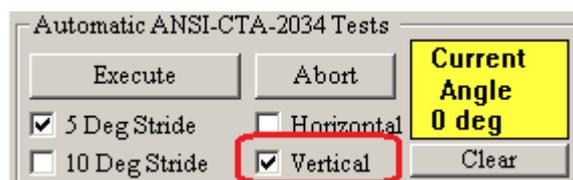


Advance the turntable to the next angle by pressing “+5 Degrees” button. The current angle will be confirmed as +5 deg.



Take the next MLS measurement and press “Save At Angle” button – **this will save the measured SPL curve in the program's data field allocated for 5deg angle.**

Continue till 360deg horizontal measurements are completed. Then select “Vertical” check box and repeat the whole process for vertical measurements.

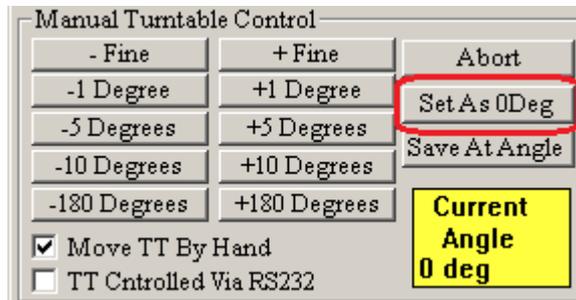


In all cases, the saved SPL curve will also appear in the corresponding “CSD Spectrum” horizontal or vertical windows. This way you can monitor the progress of your measurements. If you have other screens opened, they will present updated Polar and ANSI-CTS Performance curve information as you progress with the measurements.

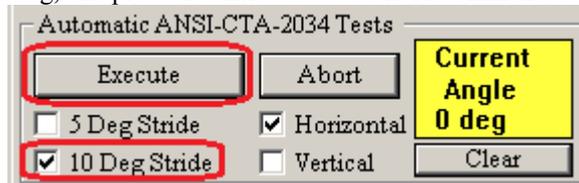
Fully Automated Turntable Measurements

This would be applicable if you have the actual RS232 controlled turntable and wish to use the fully automated ANSI-CTA-2034 measurements.

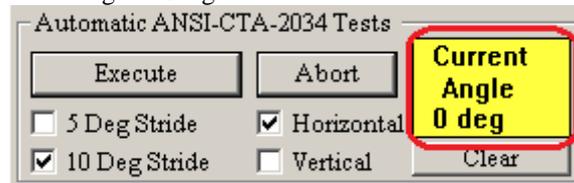
Move the turntable to the starting 0deg position by pressing “+1 Degree”, “-1 Degree”, “+Fine”, “-Fine” buttons and press “Set As 0deg” button. This will be confirmed by the “Current Angle” yellow display field.



Select the Stride 5deg or 10deg, and press the “Execute” button. Recommended is 5deg.



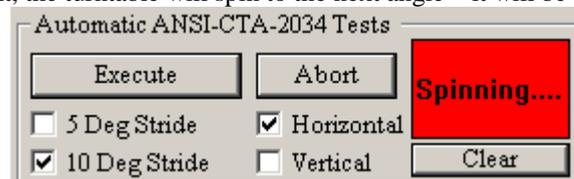
The process will display Current Angle = 0deg.



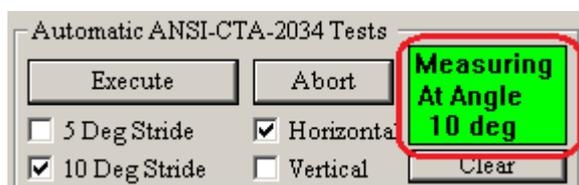
Next, The MLS measurement takes place at the destination angle and all screens are updated.



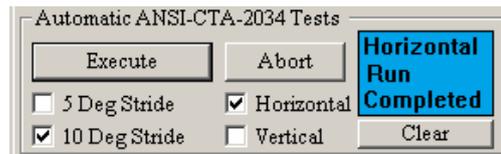
Following the measurement, the turntable will spin to the next angle – it will be 10deg, as per setting.



When the turntable stops, the next measurement will take place, and so on....



All measurements will commence and will run for all 360 degree angles. The display will confirm that.



The saved SPL curves will also appear in the corresponding “CSD Spectrum” horizontal or vertical windows. This way you can monitor the progress of your measurements. If you have the screens opened, they will present updated polar and ANSI-CTS Performance curve information as you progress with the measurements.

There are in total 144 measurements to be taken for 5deg angular resolution. It is highly recommended to run the ANSI-CTA-2034 measurements in fully automated mode. It will assure error-free, accurate data collection and will save you many unforeseen problems.

Cleaning Up Bass Response

Many SPL measurements are conducted in-room, which would constitute reflective environment. The inevitable wall/ceiling/floor reflections will contaminate SPL measurements. Shown below (green curve) is an SPL measurements with 500ms FFT window. The low-frequency response of a small vented system is clearly affected by room reflections. In order to clean-up the low-frequency response there is an **optional Matched Filter function** built-in. The operation of this technique has been extensively explained in this manual. The function is activated via the checkbox shown below, and will perform automatically.

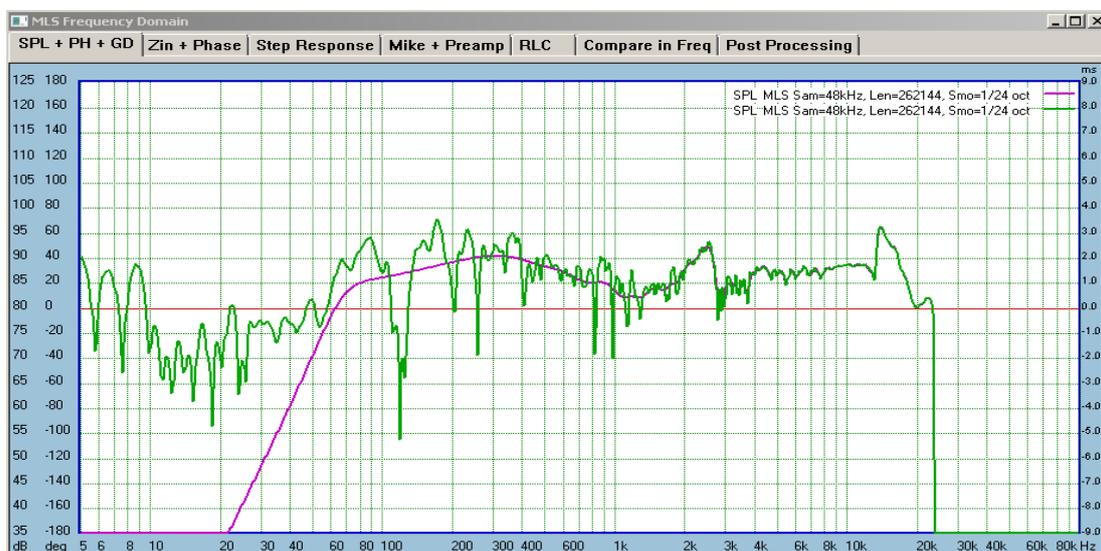
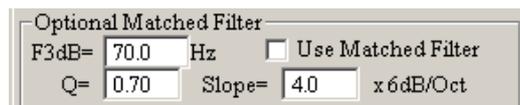


Figure 16.278. Green curve – Raw measurement in-room at 1m, Pink curve – Matched Filter activated.
16.217

You can try different (small) lengths of FFT window and different settings of the Matched Filter to bring the SPL within the expected, “anechoic” shape.

ANSI-CTA-2034 Display Control



This is the “all-important” control section of the ANSI measurements. When you open the “MLS Turntable” TAB, there are 4 ANSI- related display screen opened as well.

1. Performance SPL
2. Polar Plots
3. Horizontal CSD
4. Vertical CSD

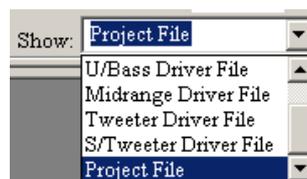
You may close any of the windows, and re-open them by pressing corresponding button with the window’s name on it. If the window is already opened, you can use these buttons to refresh the display in each window. For instance, you can change the frequency of interest for Polar Plots and the press “Polar SPL Plots” to refresh the display for the new frequency. In addition, the “Performance SPL” window can display 4-types of performance curves as defined in the ANSI Standard. These are selectable via checkboxes.

Polar Plots are displayed for any frequency of the display screen, and this chosen frequency can be edited on the “Polar Freq[Hz]” data field.

The CSD display can be changed to CD-style display by selecting appropriate check box – “CSD” or “XY”. When the CD-style display is selected, the “Lowest Level” and “Step Up” parameters can be edited on the corresponding data entry fields.

IMPORTANT – CSD screen refresh is much faster then CD-style screen refresh. Therefore, to save you total testing time, the Automatic Method will not start if the “XY” option is selected for screen style.

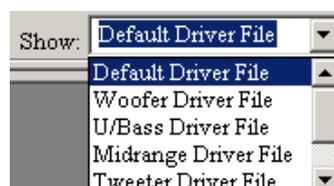
Finally, there is the “Show:” list box. This is actually very important control. It allows you to select where the displayed data will come from.



Project File – the displayed ANSI data will come from the loudspeaker system spinorama. This is when you placed the finished loudspeaker on the turntable and collected horizontal and vertical SPL curves for the whole system.

Woofers, U/Bass, Midrange, Tweeter and S/Tweeter options will display individual driver ANSI data when you load the project file and all associated driver files.

Default Driver File – the displayed data will come from a driver file currently loaded into the program.



It is essential to be able to display ANSI data for each individual driver in the system (or Project File). This is because crossover design and optimization is based on manipulating crossover components or filters assigned to individual drivers in the system.

This requirement makes the system design and optimization quite involving. Not only you may want to collect ANSI data for the whole system – to verify it's performance as per ANSI-CTA2034 Standard, but importantly, **you will need to collect ANSI data for each driver in the system.**

This would require the “Default Driver File” to be selected, the loudspeaker to be placed on the turntable, and only woofer driver (without crossover) to be tested horizontally and vertically. When finished, save the result of the spinorama as woofer file (xxxx.wfr file).

Repeat this process for all other drivers in the system.

So, for example, a 3-way system, intended to be designed and optimized for the ANSI Standard would require 144 x 3 measurements for the drivers and 144 measurements for the system. For 10degrees resolution the number of tests will be halved.

Once you have collected raw ANSI polar data for all drivers in the system, you are ready to use this data in the crossover/optimization design process.

In any case, the Automatic Method is the recommended way of collecting ANSI-CTA-2034 data.

ANSI-CTA-2034 Measurement Set Up

The loudspeaker is placed on a turntable. Assuming, that the operator is facing the loudspeaker as shown on the ANSI standard Figure 1, the right-side of the loudspeaker is facing exactly the 90deg rotational angle. For this to happen, the **turntable would have to move to the left** – as shown on the Figure 2 below.

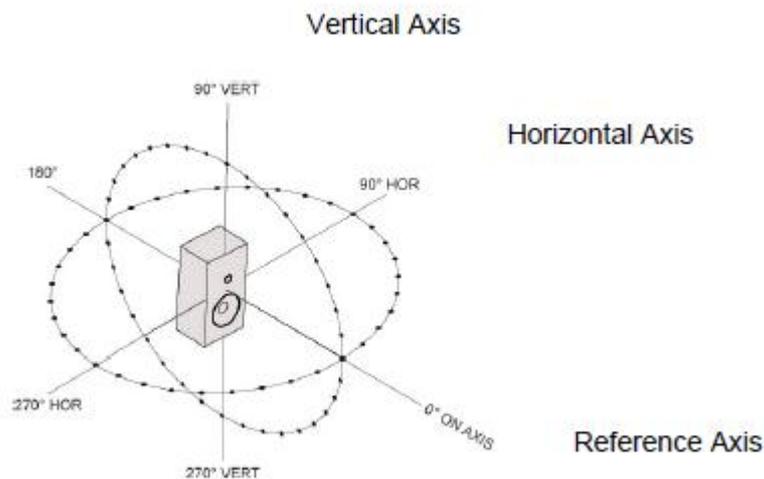


Figure 16.279. Description of ANSI-CTA-2034 angles.

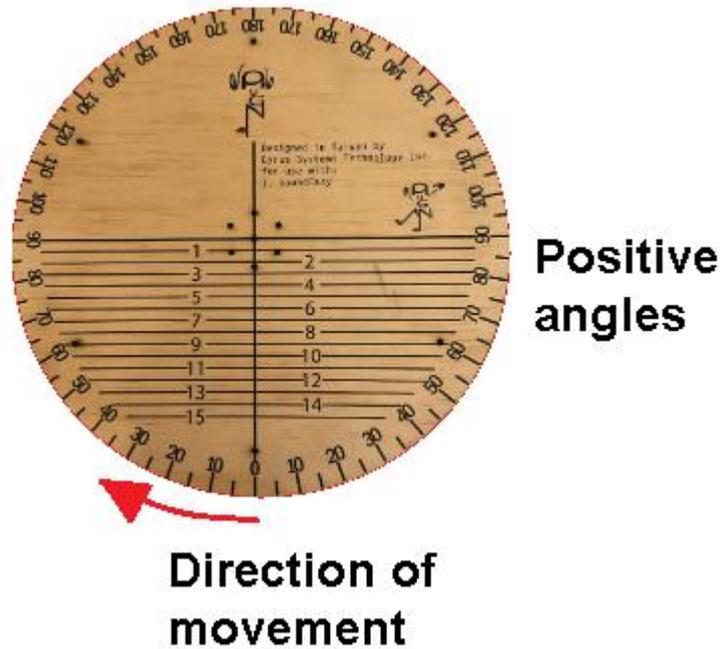


Figure 16.280. Top plate of a turntable with laser-made angle markings.

The above describes the setup for horizontal measurements.

As per ANSI requirements, vertical measurements require the top of the loudspeaker to be facing the 90deg angle. For this to happen, and the turntable also moving the left, the **loudspeaker needs to be placed on it's right-side wall**. So, that the tweeter is the most-right mounted driver.

Using the abovementioned approach, a relatively simple test setup can be arranged for both: horizontal and vertical measurements. There are three (3) ways, you can accomplish ANSI-CTA-2034 measurements:

1. **Using a hand-driven lazy-Susan table.**

This arrangement is possibly the most cost-effective option. However, if you require a full set of measurements, you will have to prepare yourself for taking of a total 144 SPL measurements for 5deg resolution in horizontal and vertical planes.

2. **Using RS232 driven turntable – option 1.**

In this case, you would be moving the loudspeaker to the exact measurement angles by pressing provided buttons on the TAB.

3. **Using RS232 - fully automated ANSI-CTA-2034 testing – option 2.**

This is the best option. The loudspeaker is being automatically moved to the required angle and the MLS system then takes the SPL measurement. The measurement is automatically processed and all ANSI performance screens are updated on-the-fly.

There is a large number of SPL curves to be collected, processed, combined and displayed. From ANSI standard:

1. **Listening Window.**

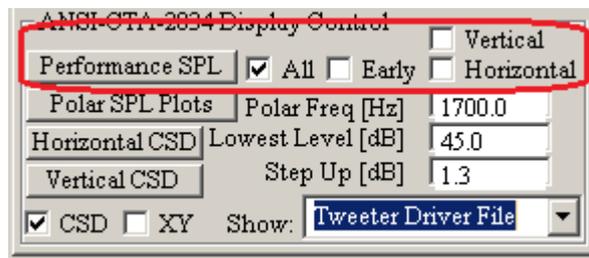
Spatial average of the nine magnitude response in the +/-10degree vertical and +/-30degree horizontal angular range.

2. Early Reflections
 Floor Bounce – 20,30,40 degrees down.
 Ceiling Bounce – 40,50,60 degrees up.
 Front Wall Bounce – 0, +/-10, +/-20, +/-30 degrees horizontal.
 Side Wall Bounces - +/-40,+/-50,+/-60,+/-70,+/-80 degrees horizontal.
 Rear Wall Bounces – 180, +/-90degrees horizontal.
3. Vertical Reflections
 Floor Reflection - -20,-30,-40 vertical.
 Ceiling Reflection – 40,50,60 degrees vertical
4. Horizontal Reflections
 Front – 0, +/-10,+/-20,+/-30 degrees horizontal
 Side - +/-40,+/-50,+/-60,+/-70,+/-80 degrees horizontal.
 Rear - +/-90.....+/-170, 180 degrees horizontal
5. Sound Power
6. Sound Power Directivity Index (SPDI).
7. Early Reflection Directivity Index (ERDI).

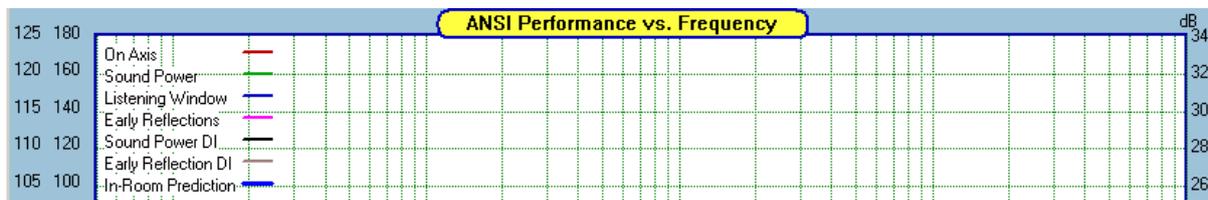
Given the above complexity, once again, we recommend the fully automated approach to the ANSI-CTA-2034 spinorama testing.

ANSI-CTA-2034 Performance Windows

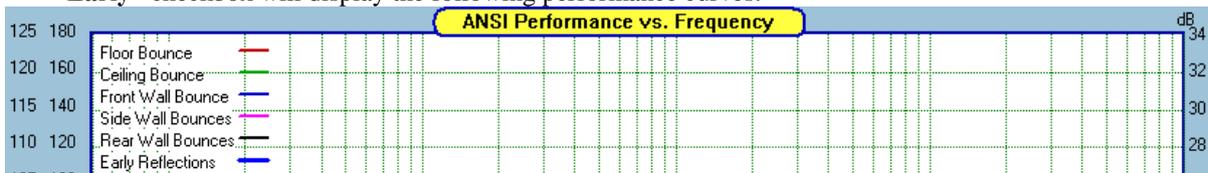
ANSI SPL performance windows are controlled via 4 check-boxes in the Display Control Group.



“All” checkbox will display the following performance curves:



“Early” checkbox will display the following performance curves:



“Vertical” checkbox will display the following performance curves:



“Horizontal” checkbox will display the following performance curves:



Figure 16.281. Example of ANSI Performance Screens.

ANSI-CTA-2034 In CAD and System Design

If you intend to modify crossover for performance evaluation using ANSI standard, it is essential to be able to collect ANSI data for each individual driver in the system (or Project File). This is because crossover design and optimization is based on manipulating crossover components or filters assigned to individual drivers in the system.

As mentioned before, this requirement makes the system design and optimization quite involving. Not only you may want to collect ANSI data for the whole system – to verify its performance as per ANSI-CTA2034 Standard, but importantly, **you will need to collect ANSI data for each driver in the system.**

This would require the “Default Driver File” to be selected, the loudspeaker to be placed on the turntable, and only woofer driver (without crossover) to be tested horizontally and vertically. When finished, save the result of the spinorama as woofer file (xxxx.wfr file).

Repeat this process for all other drivers in the system.

This way, data collected during ANSI-CTA2-34 tests for individual drivers is also available in CAD and System Design sections of the program. Two performance curves are available for plotting: **Sound Power and Listening Window** – as defined by ANSI Standard.

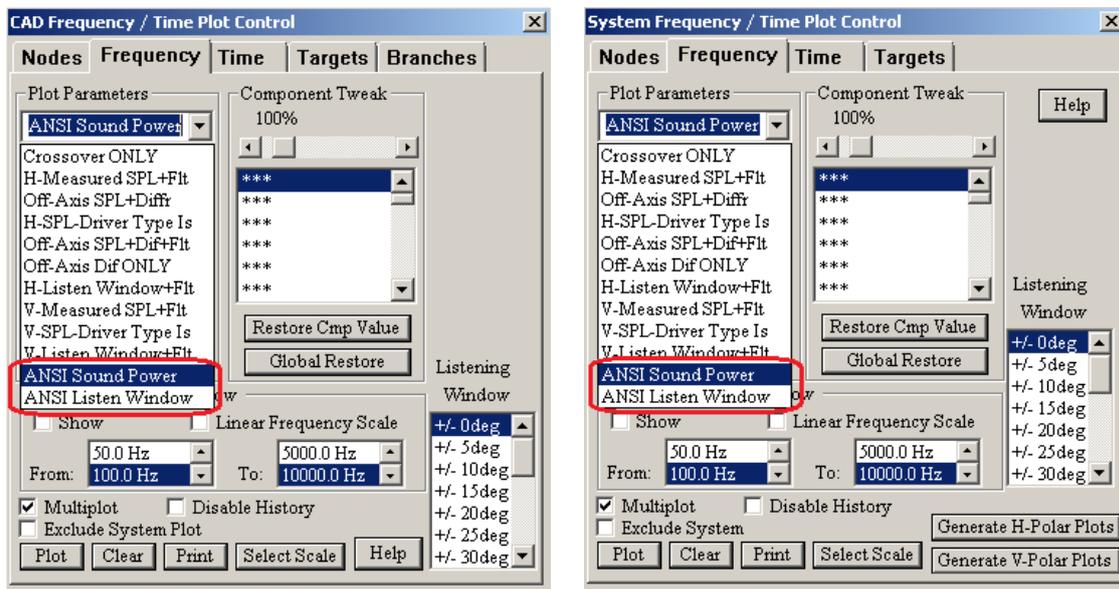
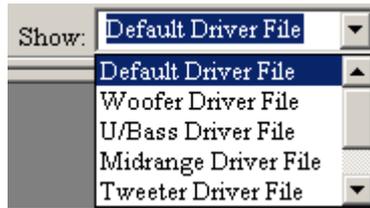


Figure 16.282. ANSI data available in CAD and System screens.

The data for these curves was collected for **raw drivers** (no crossover connected) for all 360deg angles. You will have to run several spinoramas to capture each driver separately in the system for horizontal and vertical orientation. Please note, that ANSI-CTA-2034 performance curves are defined for 10deg resolution, so this is helpful in reducing the total number of tests.

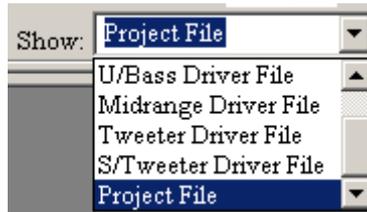
You can now modify CAD crossover components to observe changes in the two performance curves. In summary, if you intend to use ANSI-CTA2034 data for crossover design follow these steps:

1. Select “Default Driver File” from the list box. Data will be saved into Driver File ANSI section.



Place the loudspeaker on the turntable horizontally, and connect only woofer driver (without crossover). When finished, place the loudspeaker vertically and repeat the spinorama. Save the result of the spinorama as woofer file (xxxx.wfr file).

- Repeat the above for all drivers in the system.
- Select "Project File" from the list box. Data will be save into Project File ANSI section.



- Place the loudspeaker on the turntable horizontally, with the crossover connected. When finished, place the loudspeaker vertically and repeat the spinorama. Save the result of the spinorama as project file (xxxx.prj file).

Link the previously saved Driver Files into this new project file. Done.

ANSI Data in CAD And System Optimizers

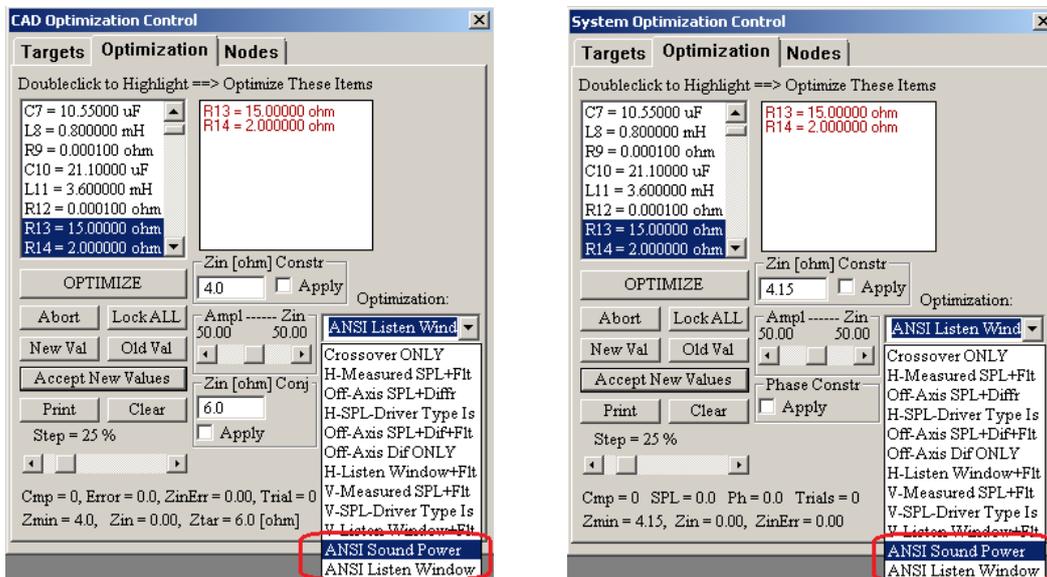


Figure 16.283. ANSI data available in CAD and System Optimizer screens.

Collecting and Displaying ANSI-CTA-2034 Test Results

Before you start the measurements, it is advisable, that you rotate the turntable around and make sure, that the FFT window is placed far enough to the left of the impulse response, so that if any of the drivers happens to rotate closer the microphone does not have it's impulse response chopped off. Examples of collected ANSI-CTA 2034 data are shown below.

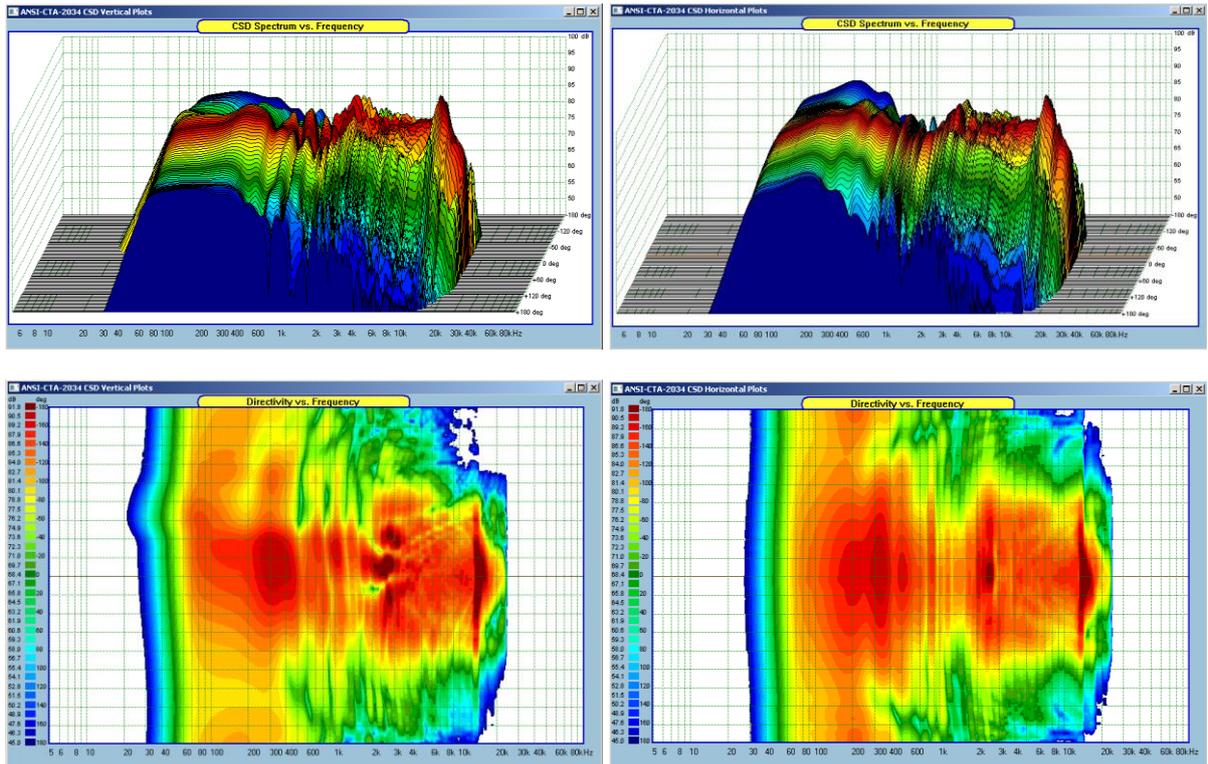


Figure 16.284. ANSI data available as CD-style and CSD-style display.

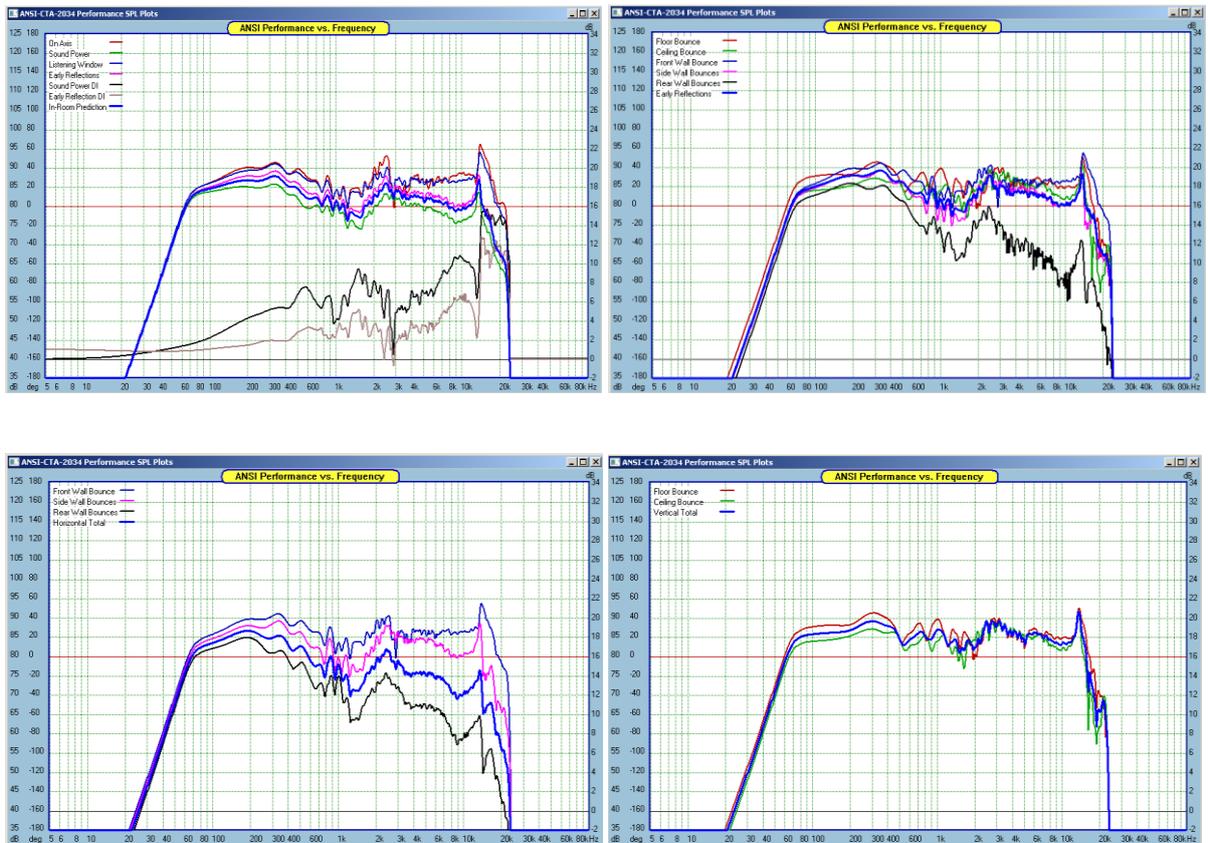


Figure 16.285. ANSI Performance Screens.

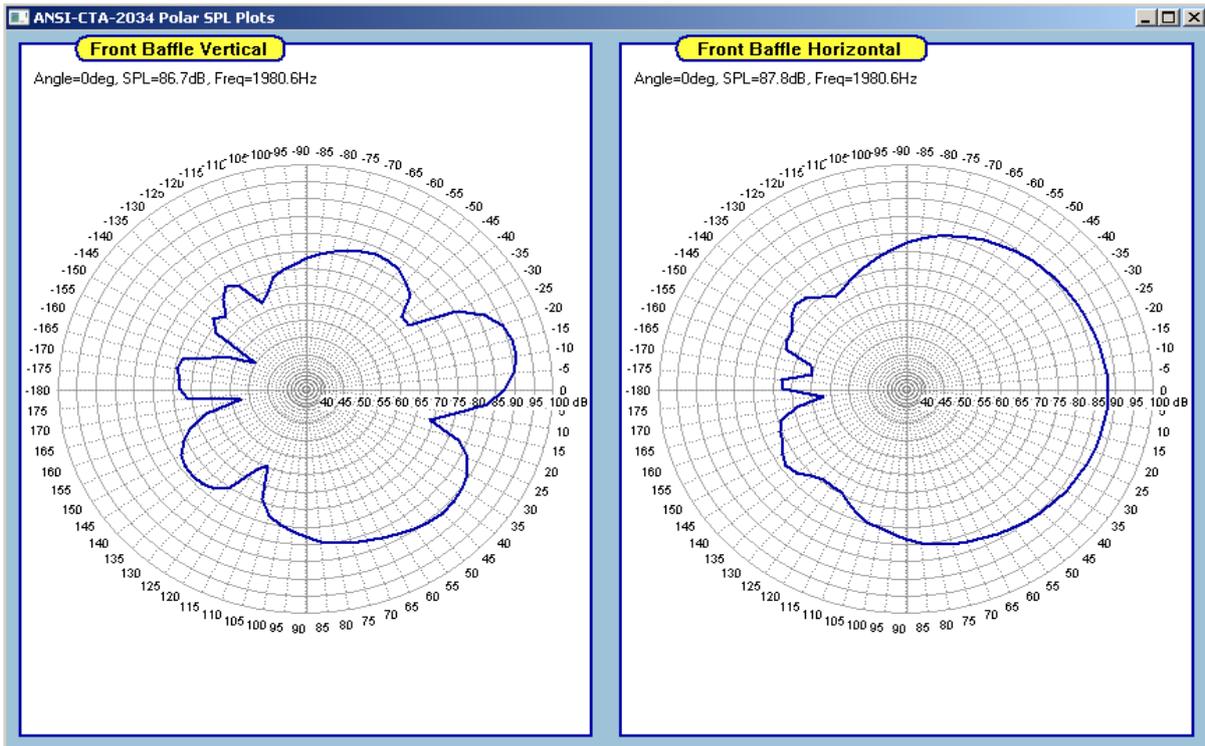


Figure 16.286. ANSI Polar Plots examples.

Secondly, if you intend to use measured data in the CAD section of the program, please select: **Use For CAD** checkbox. This will automatically deposit data collected for -90...+90degree angles in the CAD section of the program. Since the CAD/Polar/System sections of the program display data in 180deg frontal angles (rather than 360deg full circle of ANSI-CTA2034 data), this will also act as a **2x zoom-in function** for the collected data. There is a clear correspondence between Polar Plots and the CSD/XY/Circular data displayed – see below.

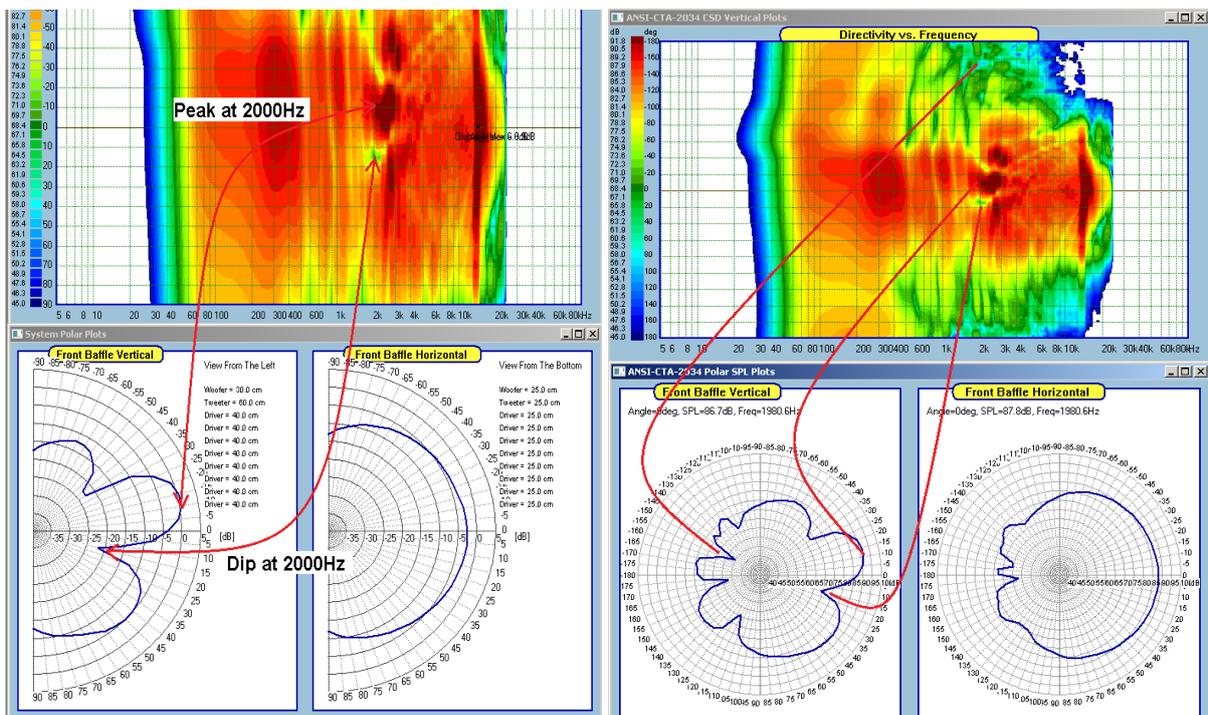
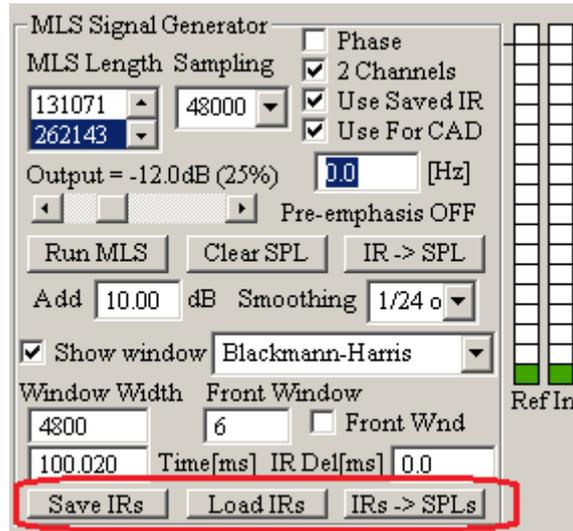


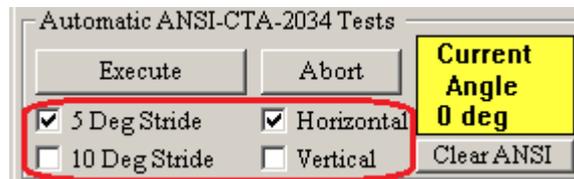
Figure 16.287. Correlating ANSI-collected data



The “**Save IRs**” and “**Load IRs**” buttons will open standard Windows dialogue boxes. Operation of those boxers is well-known, and will not be describer here.

Saved in the impulse responses file, the “*****.ans” file, are 72 impulse responses of a single test run, be it horizontal or vertical. Consequently, **for full representation of a driver in ANSI-CAT-2034 you will need two IR files.**

One will hold horizontal impulse responses and the other will hold vertical impulse responses. It is recommended, to embed driver’s name into the file-name for easier identification in the future. Data displayed below is also stored in the IR files.



The “**IRs -> SPLs**” button will activate batch-processing of uploaded impulse responses. Please note, that SPL/Phase data from re-processed impulse responses will be automatically stored in two locations: (1) Default Driver Files and (2) Project File.

For instance, you wish to change ANSI data stored in woofer Horizontal and Vertical sections of the file.

1. Load the woofer driver file.
2. Use “**Load IRs**” button to navigate to desired IR “***.ans” file, and load the file.
3. Select 5/10 Deg Stride, and check **Horizontal** checkbox to direct the algorithm where to store re-processed SPL/Phase data.
4. You can now change the location of the FFT window, window’s length and type, smoothing, Matched Filter parameters and frequency scale from Preferences window.
5. Press the “**IRs -> SPLs**” button. The re-processed SPL/Phase data will be stored accordingly to selected options.
6. Load the other IR file with vertical IRs data using “**Load IRs**” button.
7. Check **Vertical** checkbox to direct the algorithm where to store re-processed SPL/Phase data.
8. Press the “**IRs -> SPLs**” button. The SPL/Phase data will be stored accordingly to selected options.
9. Save driver file.

The most important reason for re-processing impulse responses would be the need to use different frequency scale for you design. So, if the original ANSI tests were run with frequency scale 5Hz-100kHz, you may now decide to use 10Hz-50kHz frequency scale. This is where the batch re-processing of impulse responses will come very handy. By using this feature, you will obtain re-scaled SPL/Phase data without any loss of accuracy.

The Turntable

Information about the turntable is available from: <https://augene.blogspot.com/?m=1>

The turntable connects to the computer via USB/RS232 cable. SoundEasy is set up to the COM port it is connected to using a baud rate of 9600 to allow for stable operation over a long RS232 cable.

The Turntable has a diameter of 42cm and a height of 20cm, and weights about 11Kg. Designed to carry a 30kG load with the centre of gravity on the 40cm diameter circle, but has been tested to drive loads up to twice the design criteria.



Figure 16.288. ANSI Turntable components.

The set consists of the turntable itself, a 10 meter RS232 cable, and a USB to RS232 converter (Prolific chipset). It is expected that the customer prepare their own power adapter with a 5021 DC connector that meets the safety requirements of their country of usage. What we recommend and currently use is the following power adapter:



Figure 16.289. ANSI Turntable components – front panel and power supply.

Preliminary Testing of Measured Phase Data Accuracy

Loudspeakers are typically minimum-phase devices, therefore, the measured phase response will exhibit minimum-phase characteristics. However, unfortunate microphone placement in reflective conditions may cause undesirable phase fluctuations.

Checking if the phase response from ESS/MLS measurement is close to “minimum-phase” response, or it contains regions of possible non-minimum-phase relationship with the recovered amplitude response, is recommended before executing Automatic Minimum-Phase Extraction Algorithm.

On the SPL/Phase plot below, there are two regions (**Region 1** and **Region 2**) with deep notches in the frequency response. These regions need to be tested. Please note, that the Smoothing selected for these plot was **1/48th Oct.** and there was Standard Complex Smoothing implemented in this measurement.

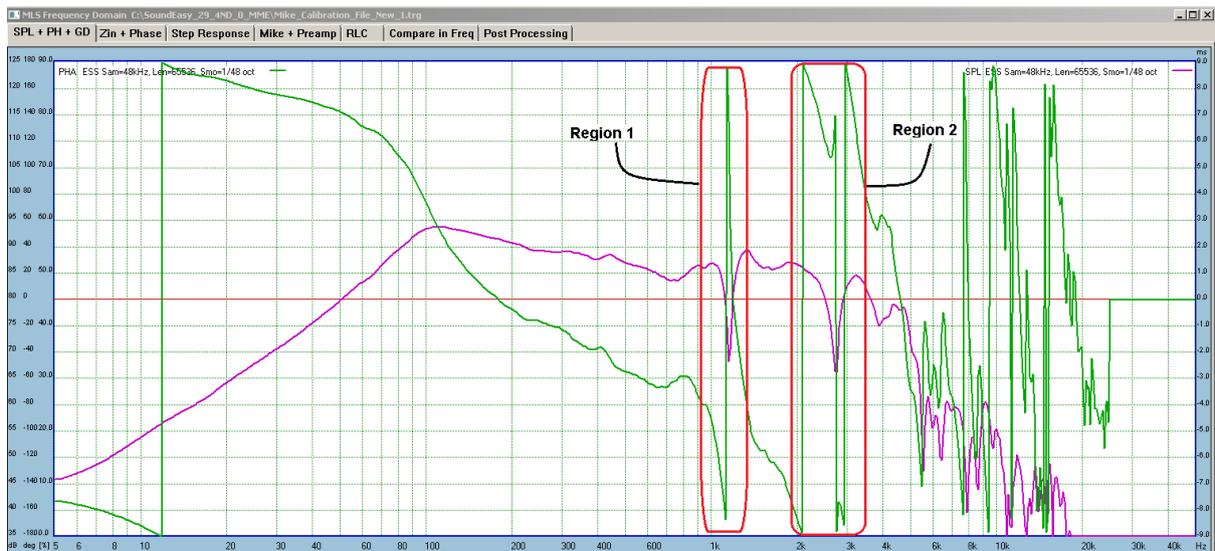


Figure 16.290. Potential non-minimum phase regions in the measured phase response.

Simple test for minimum-phase contamination can be performed using HBT function provided in the Post-Processing TAB

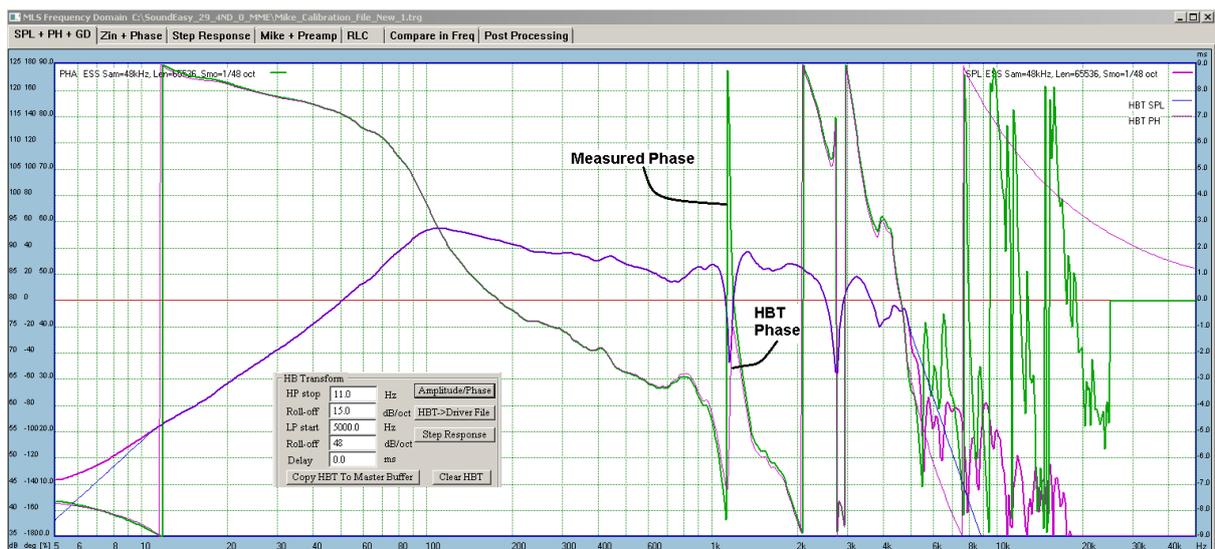


Figure 16.291. HBT shows different phase response in Region 1.

It is observable, that HBT phase is **significantly different** from measured phase in **Region 1**. **Region 2** is OK. Please note, that this technique was implemented in Ultimate Equalizer program and described in it's User Manual, Chapter 8.

If this measured phase response was supplied to the Automated Minimum-Phase Extraction Algorithm as the reference, the algorithm would fail, because the supplied phase is not minimum-phase type. This type of phase abnormality does not relate to time-of-flight, which the Algorithm is designed to eliminate. The abnormality is likely to be the consequence of unfortunate microphone placement in reflective environment.

What can be done?.

When Smoothing parameter is selected as 1/12th Oct for calculating the SPL/Phase, the measured phase response and the HBT response calculated in the Post-Processing TAB, almost perfectly overlap in both regions: Region 1 and Region 2 – see Figure below..

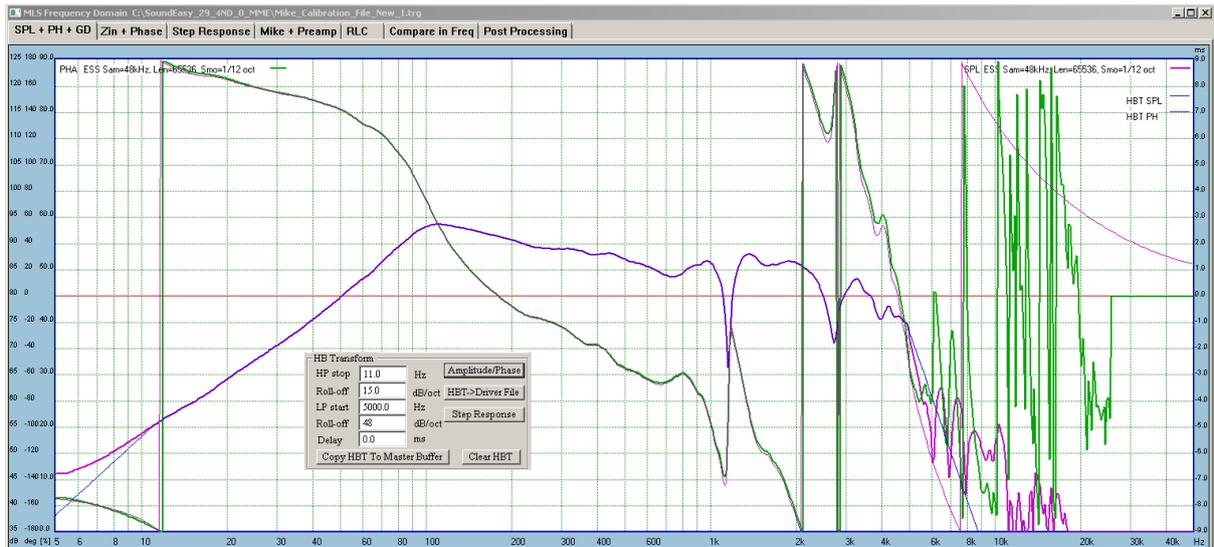


Figure 16.292. Improving Smoothing parameter from 1/48 to 1/12 Oct eliminates the phase abnormality in Region 1.

This is the correct setting for Smoothing parameters.

A prudent user may take this test even further. Using the 1/12 Oct. smoothing, and stepping the start of the FFT window from Bin 50 to Bin 60 does not reveal any other suspicious regions within the frequency of interest.

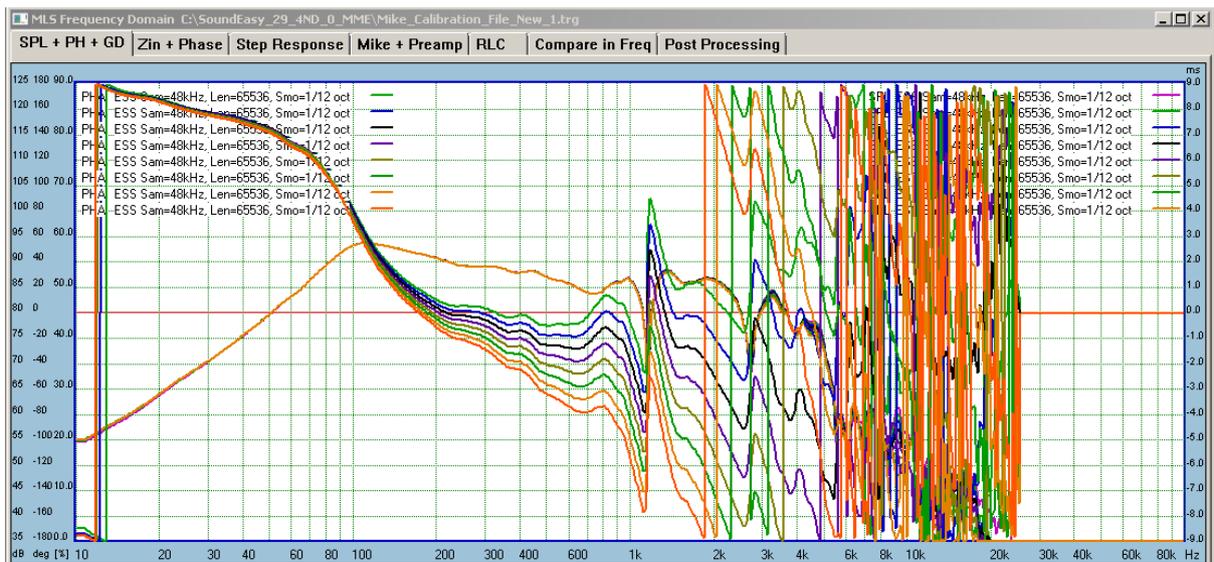
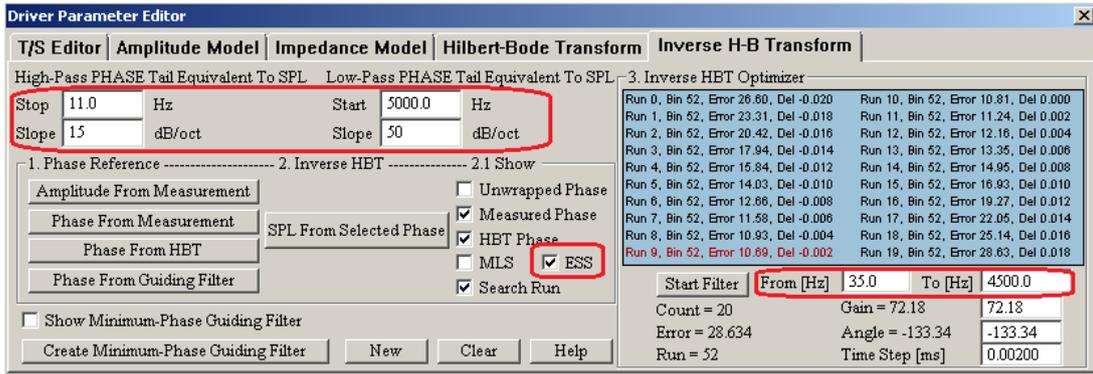
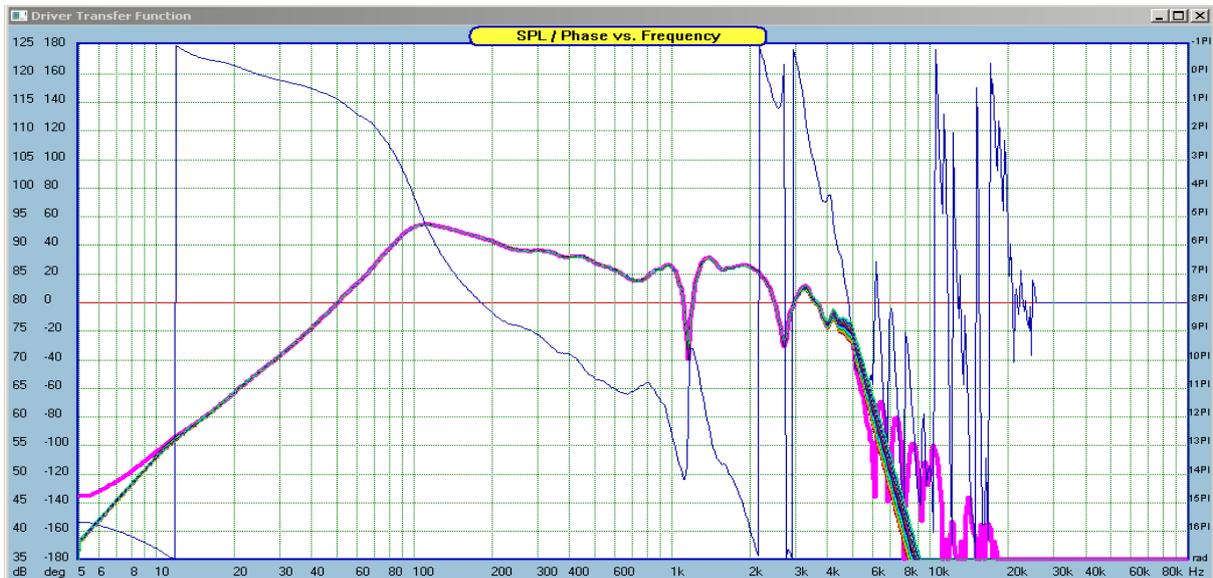


Figure 16.293. 1/12 Oct Smoothing eliminates the phase abnormality for FFT windows located at bins 50-60.

We are now in the position to calculate the true minimum-phase response using the automated process.



When finished, the process reports quite small error of 10.69, with the Delay parameter needed to be -0.002ms, with the FFT window located at Bin = 52.



We can now return to the ESS System and enter the Delay = -0.002ms, and position the FFT window at 52bin location. Then re-calculate the SPL/Phase. This is the true SPL and Measured Minimum-Phase response.

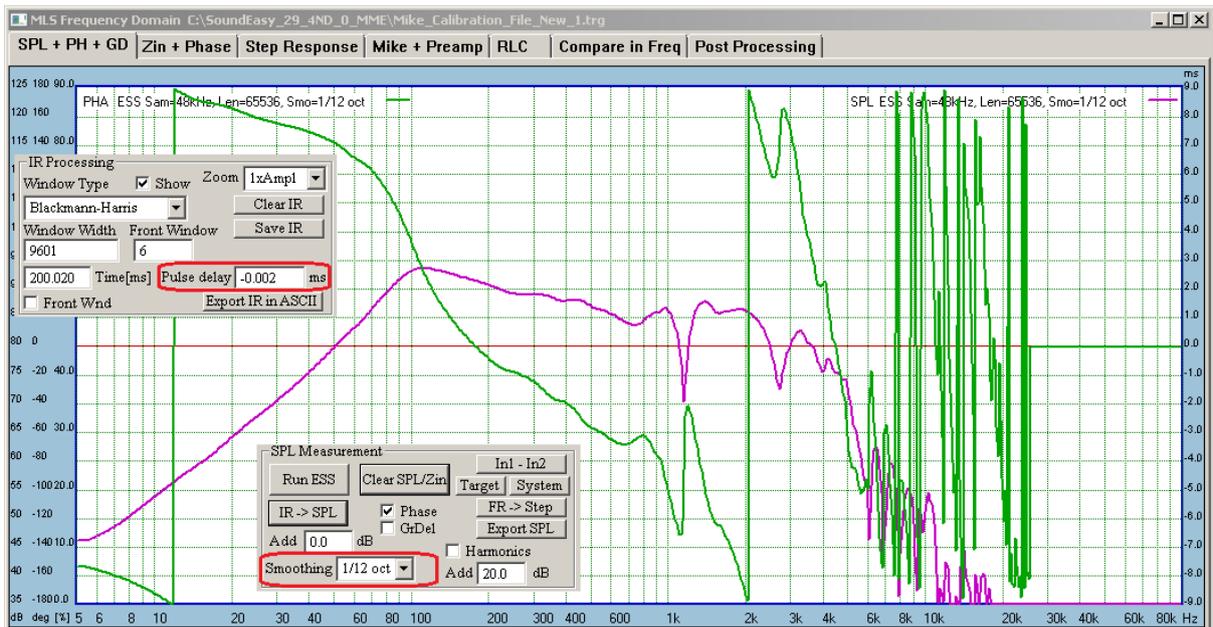


Figure 16.294. Measurement of phase response can now be completed, using calculated parameters.

Finally, please go to the **Driver Editor** -> **HBT TAB** and edit HBT parameters prior saving the data into the driver file.

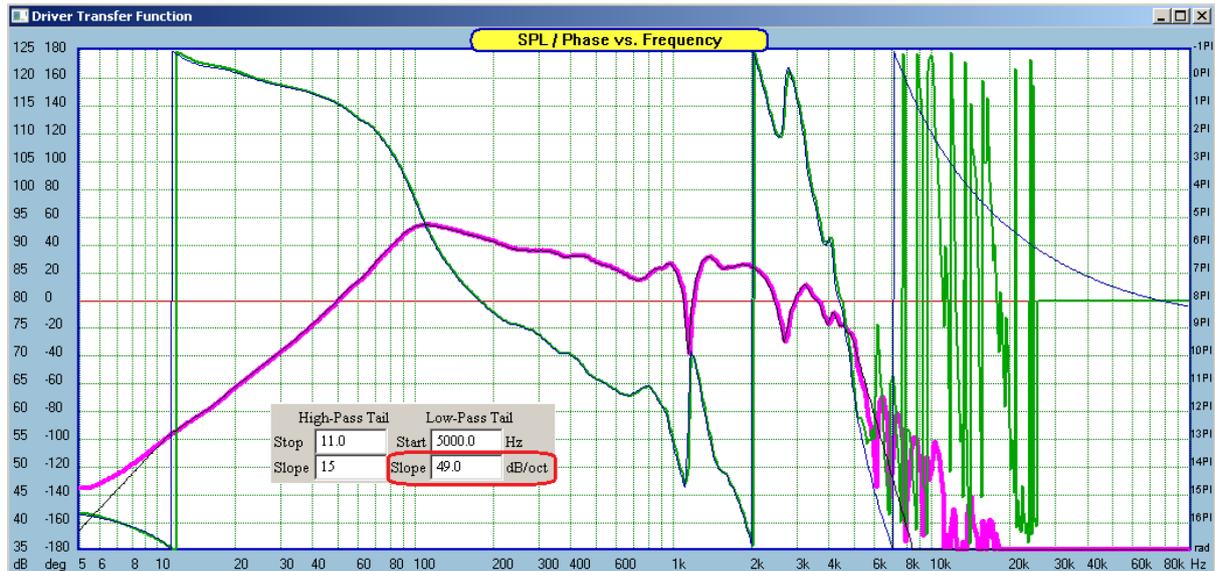


Figure 16.295. Final corrections to the HBT in Driver Editor system.

Another test has been performed, to check if selecting slightly different starting Bin for FFT window would affect the final Delay and FFT Bin parameters recommendations from the Automated Algorithm.

Starting FFT Bin = Bin 50. When finished, the Algorithm recommended **Bin 51 and delay = 0.010ms**

Starting FFT Bin = Bin 52. When finished, the Algorithm recommended **Bin 52 and delay = -0.010ms**

SPL and Phase responses for both recommended cases are shown below. Please note, that phase responses within operating range of the Algorithm, are **identical**.

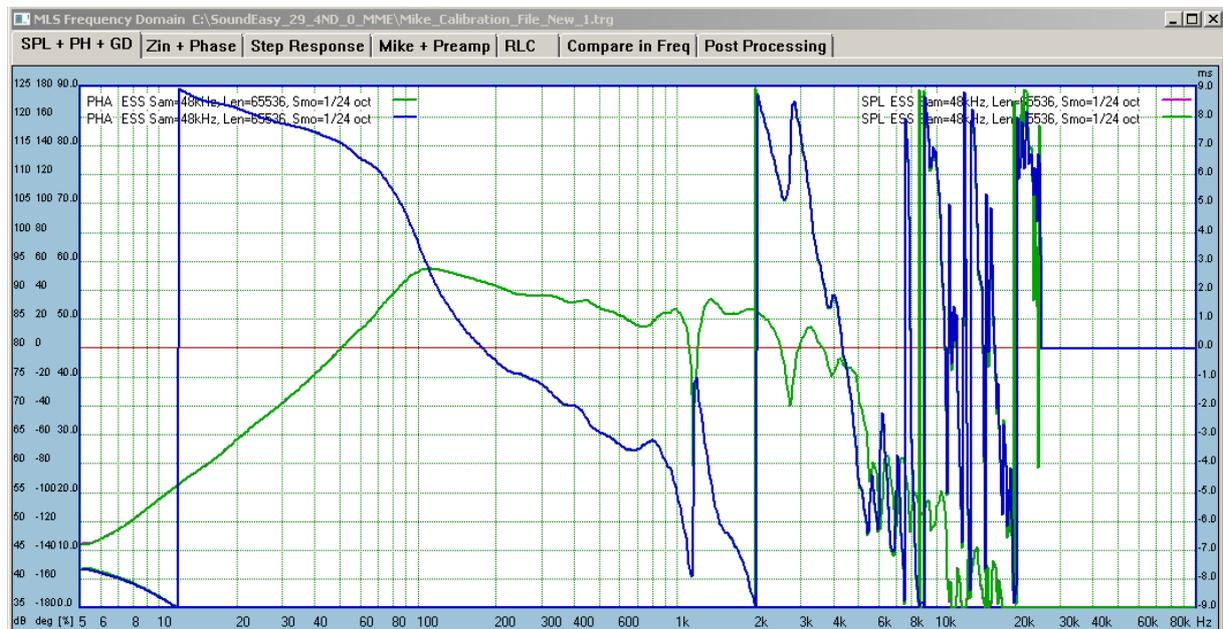


Figure 16.296. Identical phase responses calculated for different FFT window bins.

Harmonic Distortion Extracted with ESS Sweep

Logarithmic ESS frequency sweep has some very interesting properties. When measuring drivers in anechoic conditions, we obtain a train of harmonic impulse responses, arriving before the main impulse response. The de-convolution process will concentrate the energy of each individual harmonic to a fixed arrival time across the whole frequency span. Therefore, the frequency response and the frequency-dependant harmonic levels can be determined with one single sweep. In post-processing the individual harmonic impulse responses are separated by employing appropriate windows and FFT. The harmonic impulse responses will arrive before the fundamental impulse response at the time arrival, T_h , determined from the formula below.

$$T_h = -T * \ln(n) / \ln(f_2/f_1)$$

Where: T = sweep time, n = harmonic order, f2 and f1 are stop and start frequencies

In the first example, a simple “diode clipper” circuit was assembled and the time domain of the output waveform, as a response to sine wave input, is presented below. The clipping is symmetrical, therefore we expect even harmonics to be very small and odd harmonics to be significant.

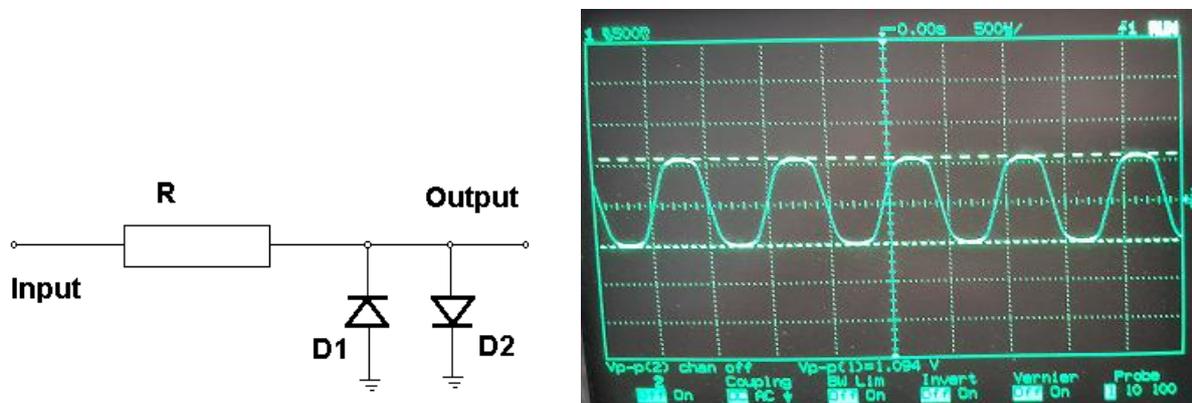


Figure 16.297. Diode clipper and the output wave form.

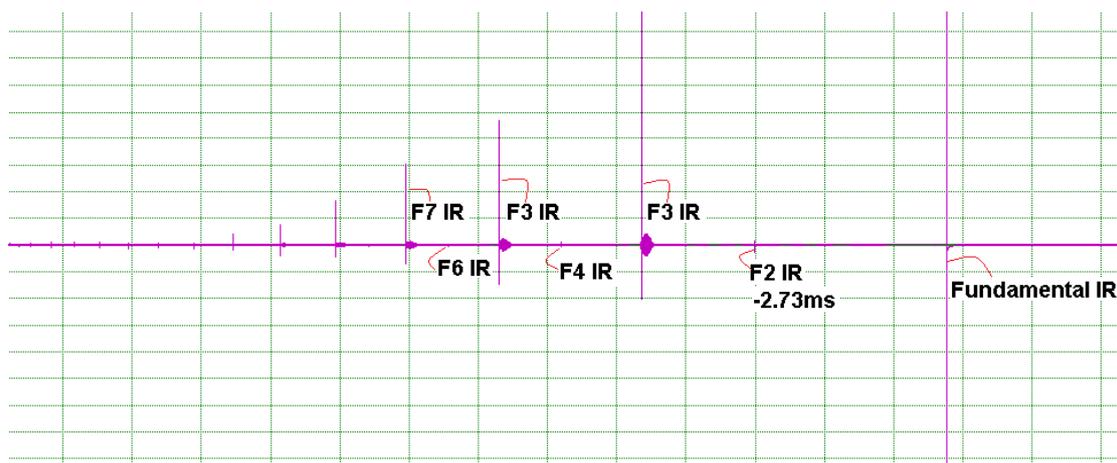


Figure 16.298. Spectrum of the time domain signal depicted on Figure 16.297

For 48kHz sampling frequency and 131072 ESS length, the F2 harmonic will appear 2.73ms BEFORE the main (fundamental) Impulse Response.

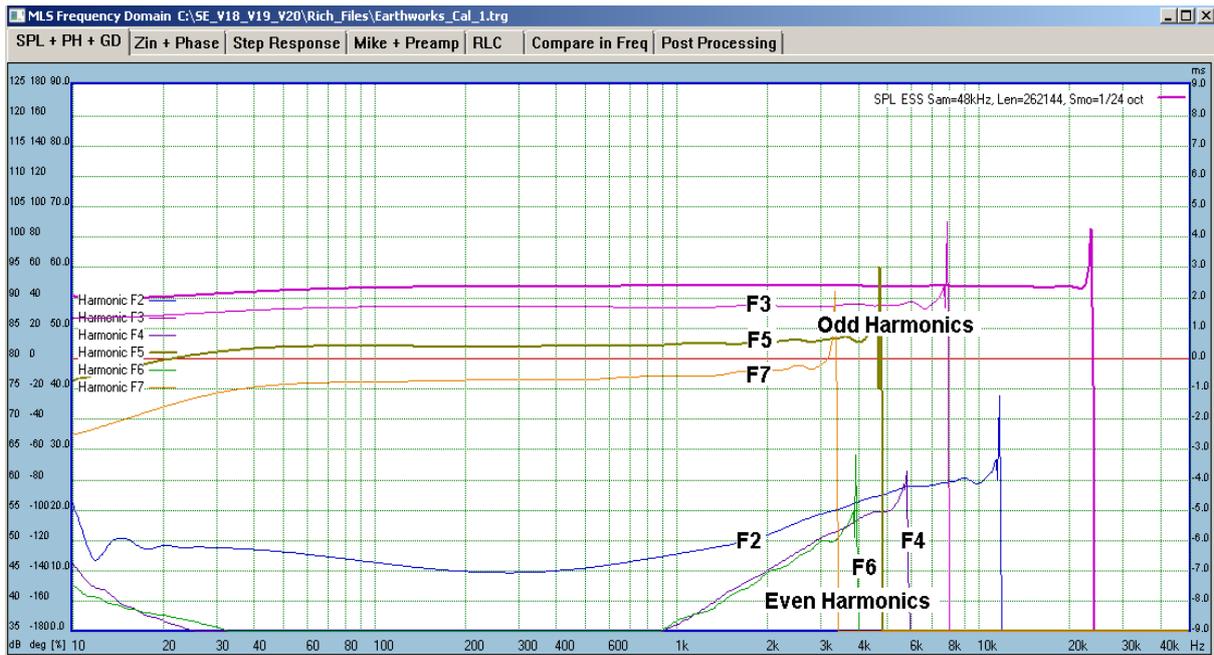
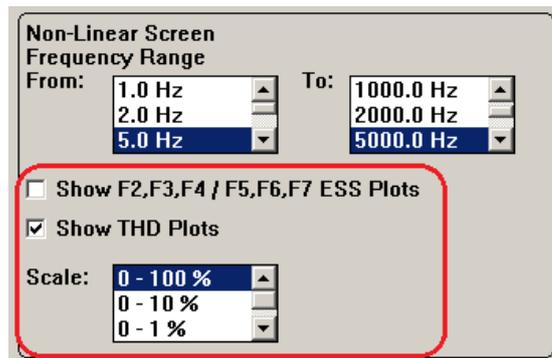


Figure 16.299. Level of harmonics in the output signal. **Harmonic are risen by 20dB.**

The ESS Sweep responds to the following control switches on the Preferences Screen.

1. **Show F2, F3, F4 / F5, F6, F7 ESS Plots** – this switch enables the display of F5-F7 harmonics.



2. **Show THD Plots** – this switch enables plotting of the THD curve (thick red line below).
3. **Scale** – this switch controls the THD scale display **ONLY** when using ESS Harmonics plotting.

Automatic extraction of harmonic distortions from ESS measurement follows typical measurement procedure for ESS measurements. Better results can be achieved with longer ESS lengths. The start of the FFT window should be placed in a position, that reduces unnecessary phase transitions $+180/-180\text{deg}$ due to time of flight delay incorporated into the phase response. Ideally, the phase response would be the minimum-phase response, but there is no need for absolute accuracy here – see Figure 16.300 below. Harmonics can be elevated to the desired level by adding the required number of decibels, as entered on Figure 16.301 in the **Add** field.

Often, the harmonic measurements are run at elevated signal levels (but not necessarily), therefore, you would run a few tests before the actual harmonic measurements are conducted.

Before commencing measurements, please check “Harmonics” box as shown on Figure 16.301. The harmonics IRs are calculated during the de-convolution process, which also calculates the main Impulse Response. The harmonics IRs arrive before the fundamental Impulse Response.

In the second example, an 8” woofer will be examined at 10W power level. After several tests aimed at establishing approximate location of the FFT window, the following data was collected. **Please note, that harmonics are automatically scaled down in frequency domain.**

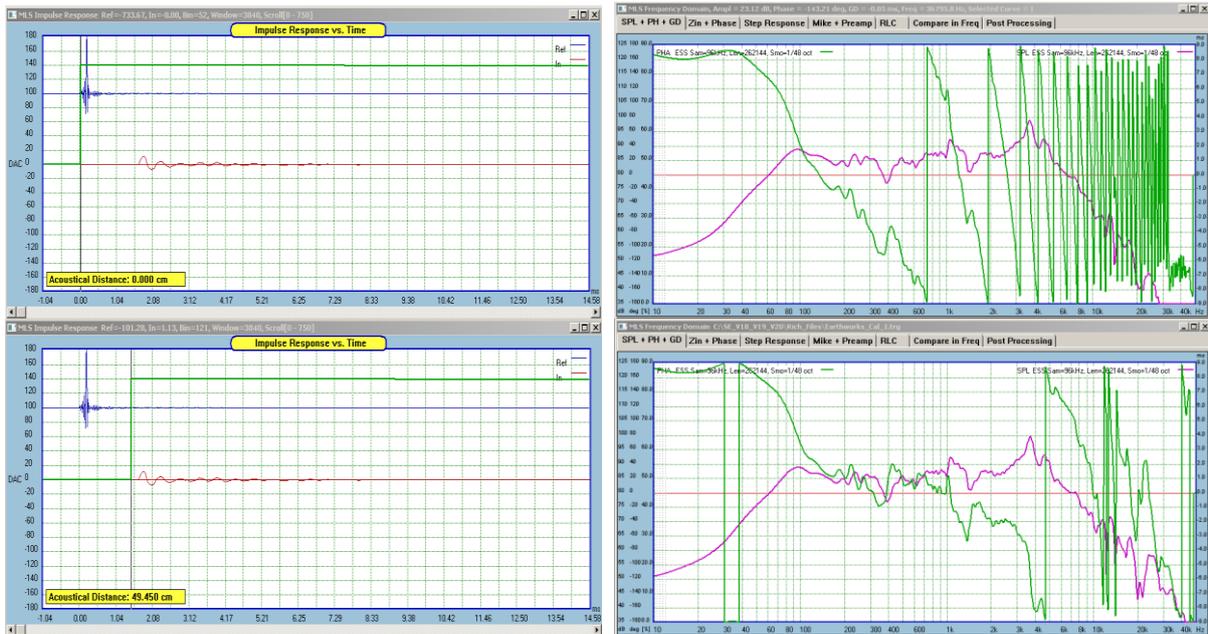


Figure 16.300. Place the start of FFT window to reduce (remove) time-of-flight effects.

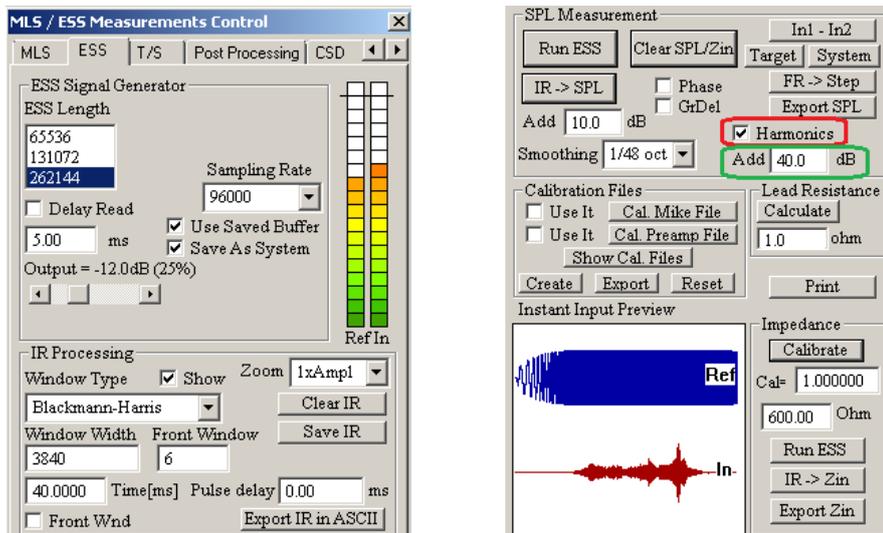


Figure 16.301. Typical settings of the ESS system for harmonic distortions measurements.

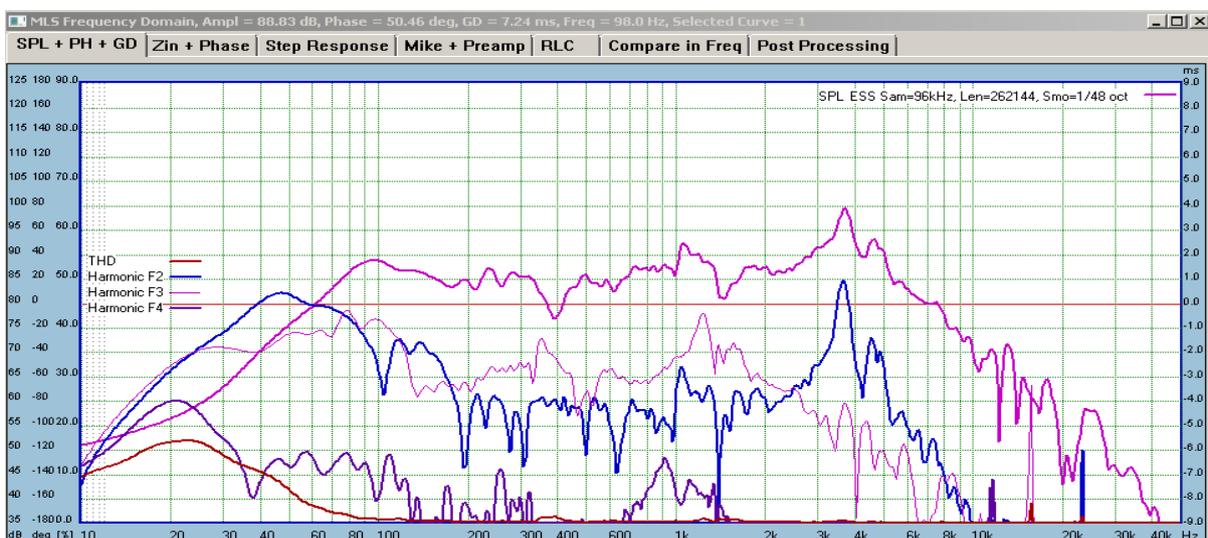


Figure 16.302. THD plots extracted from ESS sweep.