

## Centre Loudspeaker Transducer Measurements

Loudspeaker's frequency response was measured in-room, using windowed MLS technique. The speaker is presented on the picture below.



Tweeter frequency/phase response



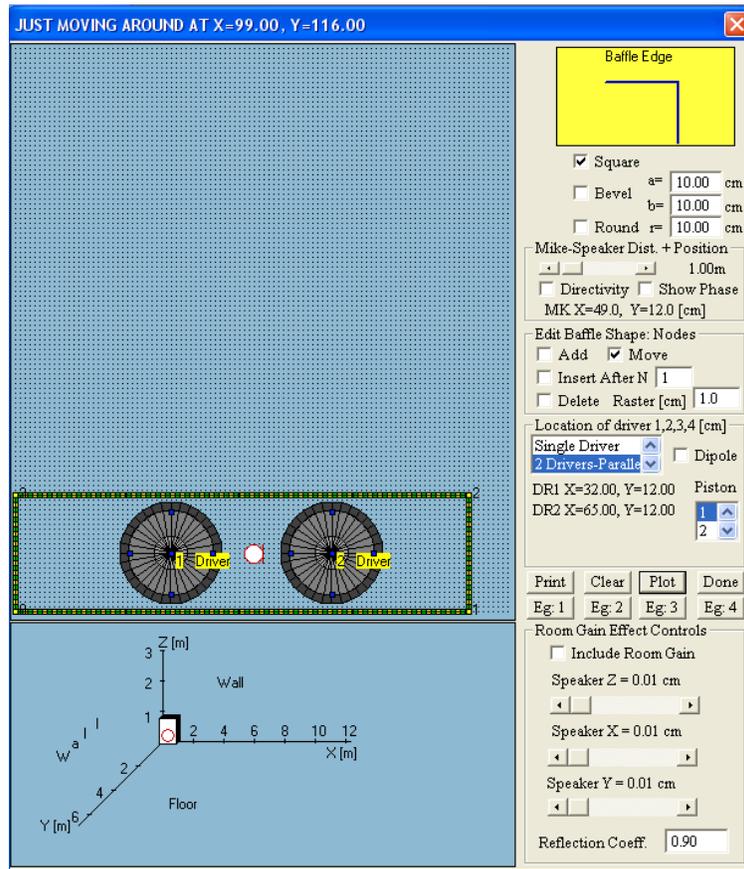
Woofers frequency/phase response

Tweeter's frequency response is fully usable for the design purposes, however, woofer's frequency response lacks information in the low-end and is really unsuitable for developing equalization. Consequently, woofer's frequency response was measured in-room, using close-mike technique. The way it works, is that you need to measure driver's frequency response, then port's frequency response, and add them together. Port response has to be scaled down by several decibels, due to the difference in effective diameter between port and driver:  $20 \cdot \log(\text{Driver\_Radius} / \text{Port\_Radius})$

In my case, the port SPL was shifted down by -10.0dB. On the top of this, you need to add pre-calculated diffraction for this box. The technique was also described in details in UE3 User's Manual,

<http://www.bodziosoftware.com.au/UE%20V3%20Manual.zip>

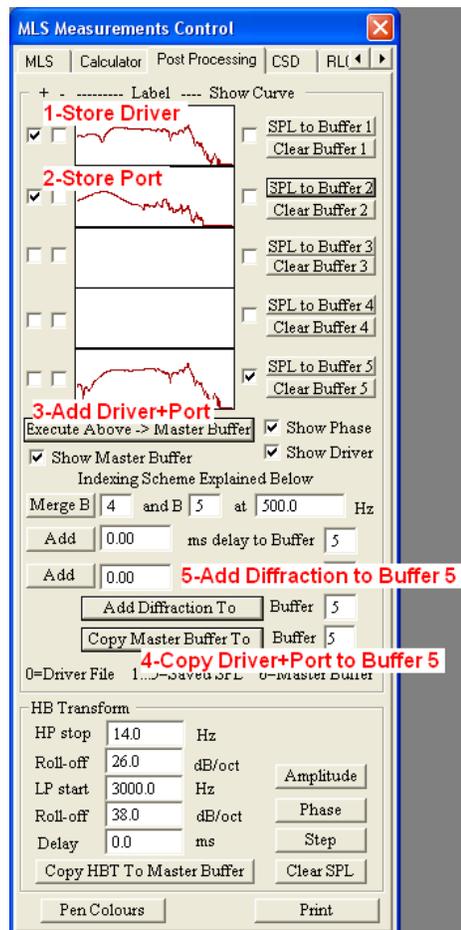
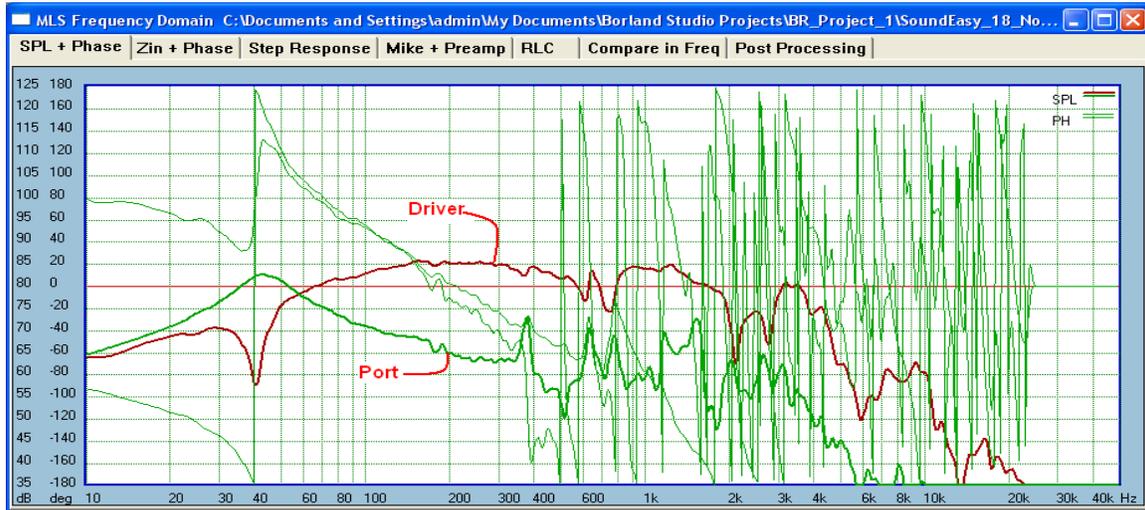
All operations and measurements were performed using SoundEasy V18. As a starting point, I calculated diffraction of the front panel. Dimensions are 93cm x 25cm.



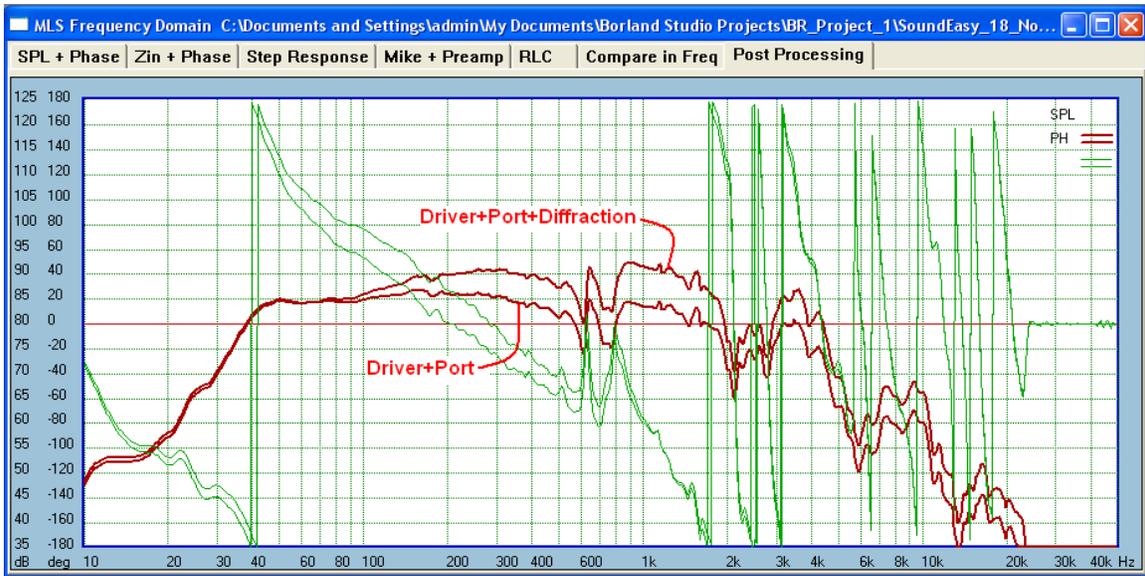
Diffraction plot, depicted by the red line, is shown below.



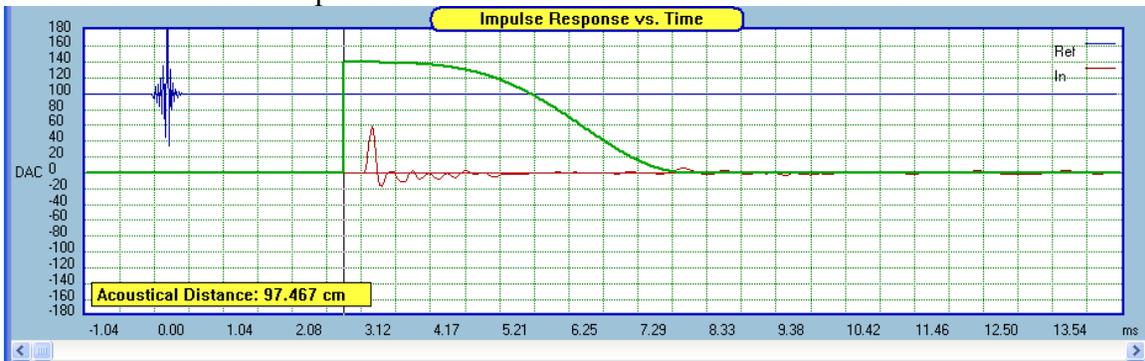
Next, close-mike measurements are performed on driver and port. Port is scaled down by -10.0dB. Driver's SPL is stored in Buffer 1 and port is stored in Buffer 2 - see below.



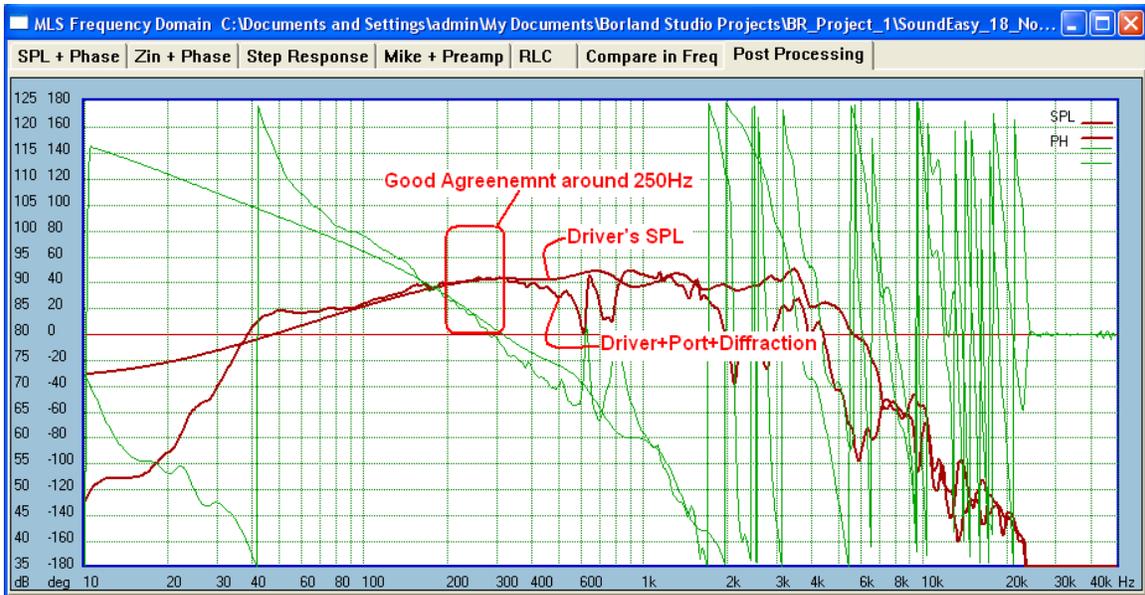
Next, port and driver SPL are summed in Master Buffer and the Master Buffer is copied to Buffer 5. Then, I added pre-calculated diffraction to Buffer 5 – see step 5 above.



Next, I moved the microphone to 1.0m distance and measured “far field” SPL. This is to make sure I capture diffraction effects. The result is shown below.

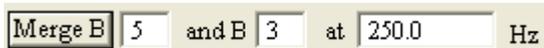


The “far field” SPL is stored in Buffer 3, and will be used to represent driver’s Transfer Function above 250Hz.



Driver's 1.0m SPL is stored in Buffer 3 and close-mike measurements (Driver+Port+Diffraction) are stored in Buffer 5. As Buffer 5 is the same as Buffer 3 at 250Hz (our merging frequency).

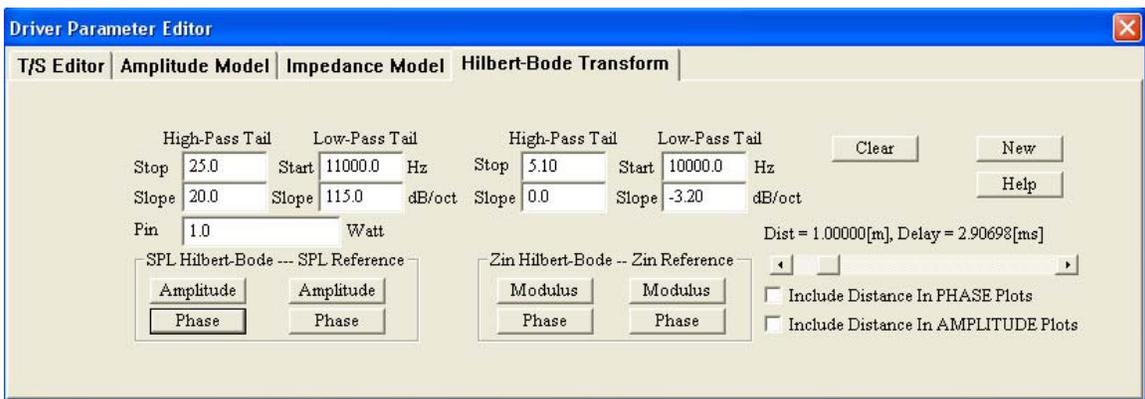
Next, Buffer 5 (lower-end of the SPL) is merged with Buffer 3 (higher end of the SPL) at 250Hz. The result will be automatically stored in Master Buffer (Buffer 6).



Finally, I transferred Master Buffer to Buffer 0 (Driver Editor Screen). And this completes curve arithmetic operations.



I can now move my activities to Driver Editor screen for developing proper Transfer Function of this driver.

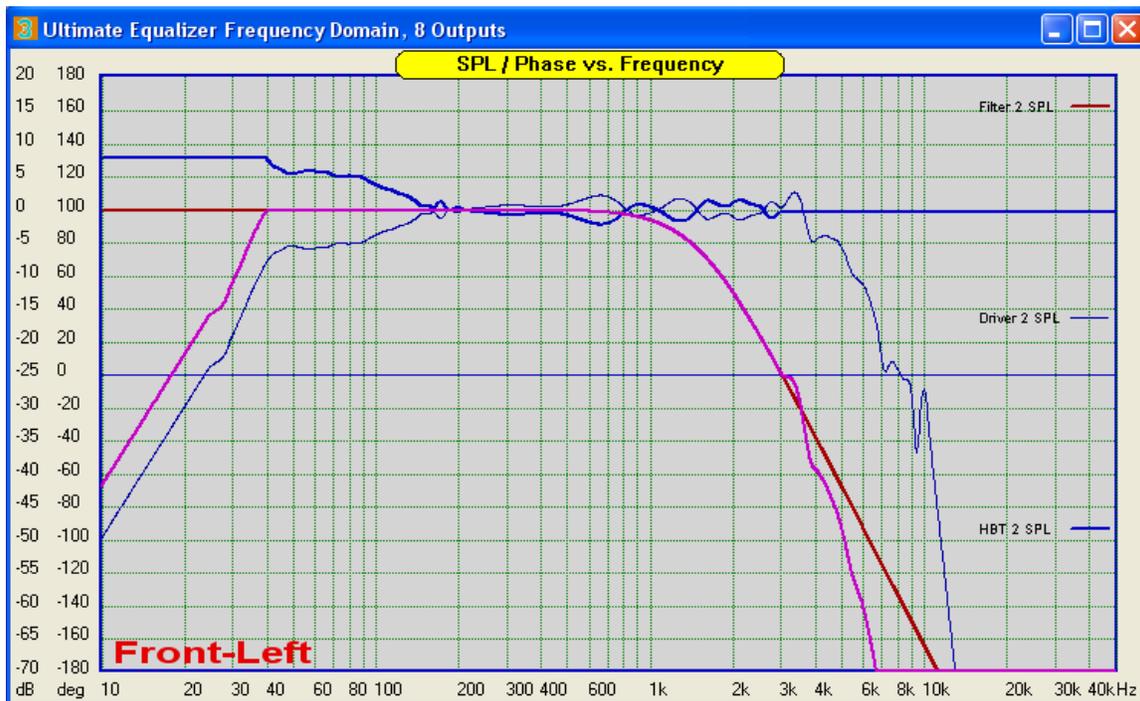


The above HBT parameters secure perfect agreement between measured SPL and HBT-generated Transfer Function from 25Hz to 11kHz. This is much better than we need.



I can now save woofer driver file for use with Ultimate Equalizer.

Here are the equalization curves developed in UE3 for the woofer driver. I aim at -24dB/oct Linkwitz filter at 1500Hz. HBT range selected in UE3 is 40Hz-3000Hz.



Thin blue curve – woofer measured SPL  
 Thick Blue curve – woofer's HBT equalization  
 Red curve – filter's template.  
 Pink curve – final woofer response

For the tweeter, I intended to capture diffraction effects, therefore for this small box, I measured the tweeter at 1.0m distance, and had to window room reflection – as shown below. I have also removed “flight time” from the impulse response, by shifting the starting point of the FFT window.



There are no curve arithmetic issues here, so I can now move to Driver Editor screen, and for the following HBT parameters.....

The screenshot shows the "Driver Parameter Editor" window with the "Hilbert-Bode Transform" tab selected. The parameters are as follows:

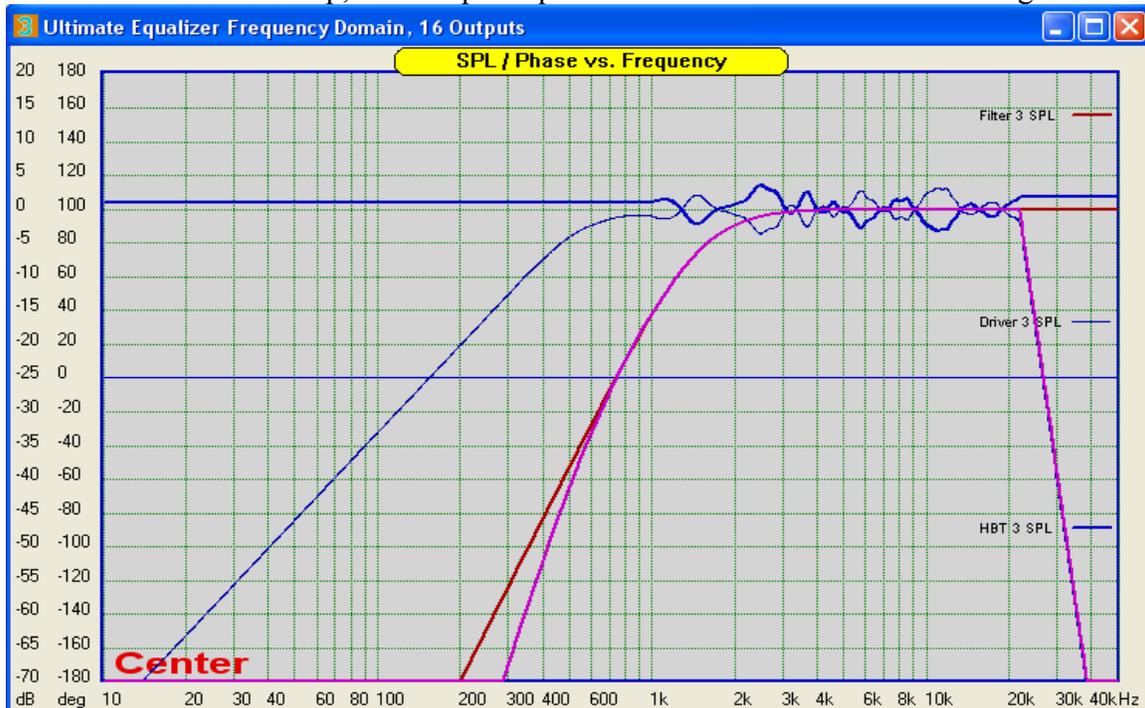
High-Pass Tail		Low-Pass Tail		High-Pass Tail		Low-Pass Tail	
Stop	330.00	Start	23000.0 Hz	Stop	5.10	Start	10000.0 Hz
Slope	13.0	Slope	36.0 dB/oct	Slope	0.0	Slope	-3.20 dB/oct

Other parameters shown include Pin = 1.0 Watt, Dist = 1.00000[m], and Delay = 2.90699[ms]. There are also buttons for "Amplitude" and "Phase" for both SPL and Zin Hilbert-Bode transforms, and checkboxes for "Include Distance In PHASE Plots" and "Include Distance In AMPLITUDE Plots".

I have obtained HBT-generated Transfer Function from 200Hz to 22kHz – see below.



I can now save tweeter driver file. In order to protect tweeter driver more effectively, I decided to use 24dB/oct LR crossover, at 1500Hz. If the system is run in linear-phase mode, the crossover slope does not matter, as the phase will always be flat in this mode. In the next step, I developed equalization curves for the tweeter using UE3.



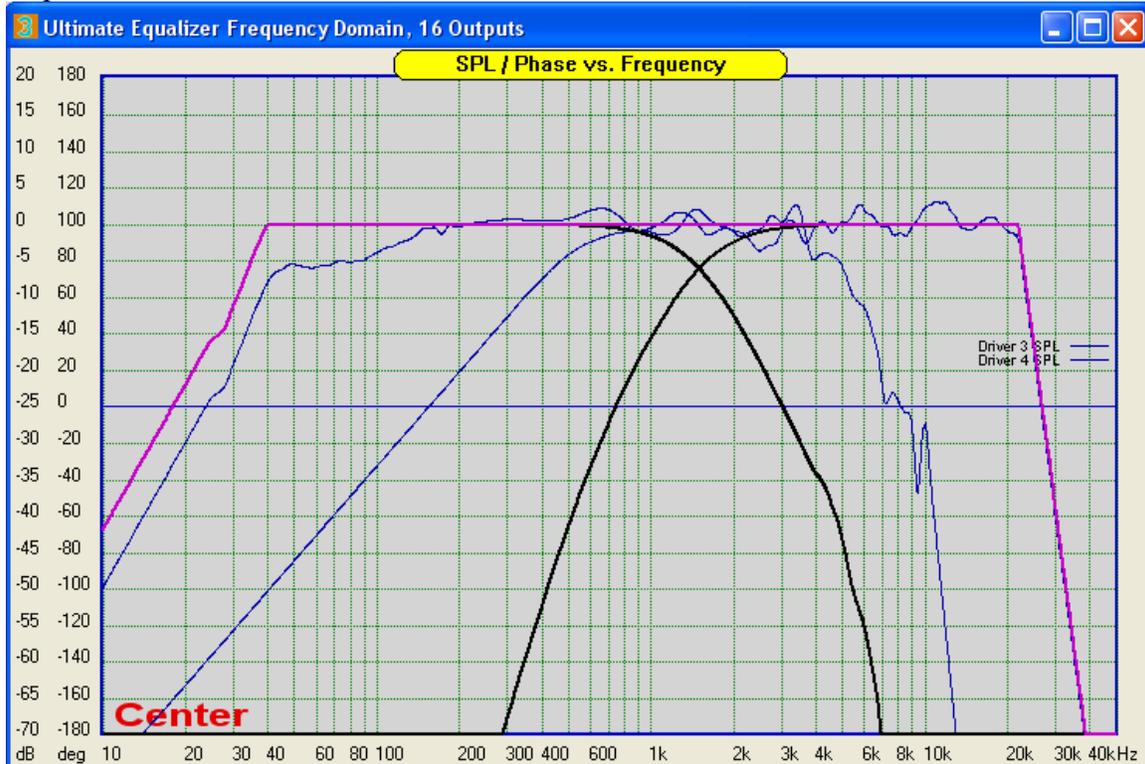
Thin blue curve – tweeter measured SPL

Thick blue curve – tweeter's HBT equalization

Red curve – filter's template.

Pink curve – final tweeter response

Putting the two drivers together, creates the following set of UE3 modeled responses:



Thin blue curves – woofer and tweeter measured SPL

Black curves – woofer and tweeter equalized SPL (they are partially overlapped by the pink curves)

Pink curve – Final System SPL response

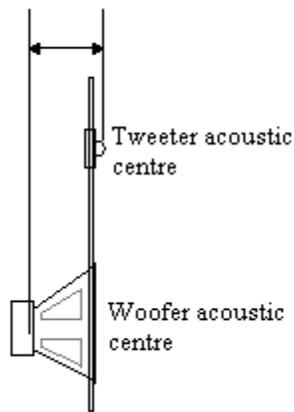
Blue curve – Final System Phase response (flat, linear curve in the middle of the screen).

It is observable, that final system amplitude response extends flat from 40Hz – 22000Hz. The 3dB low-frequency extension down to 35Hz is provided by 7dB HBT boost in this frequency range. If only 2.5dB of amplifier headroom is available, then the HBT boost must be reduced to 2.5dB, resulting in 45Hz cut-off frequency. This was the exact goal of this design. Un-equalized cut-off frequency was 41Hz, with a large sag below 150Hz due to diffraction. As I anticipated, diffraction effects (broad hump from 150Hz -800Hz) were correctly equalized and also tweeter SPL irregularities are gone.

### Time Alignment Of The Drivers

Due to quite simple mounting configuration on a flat, front baffle, acoustic centers of both drivers are likely to be offset against each other. This problem is explained on the diagram below, and will manifest itself during the MLS measurements as woofer phase response lagging behind tweeter's phase response.

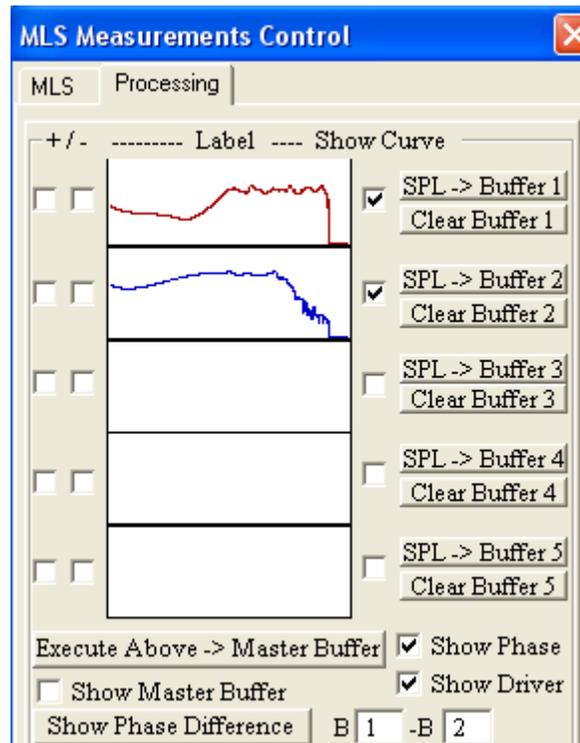
Acoustic Centre difference

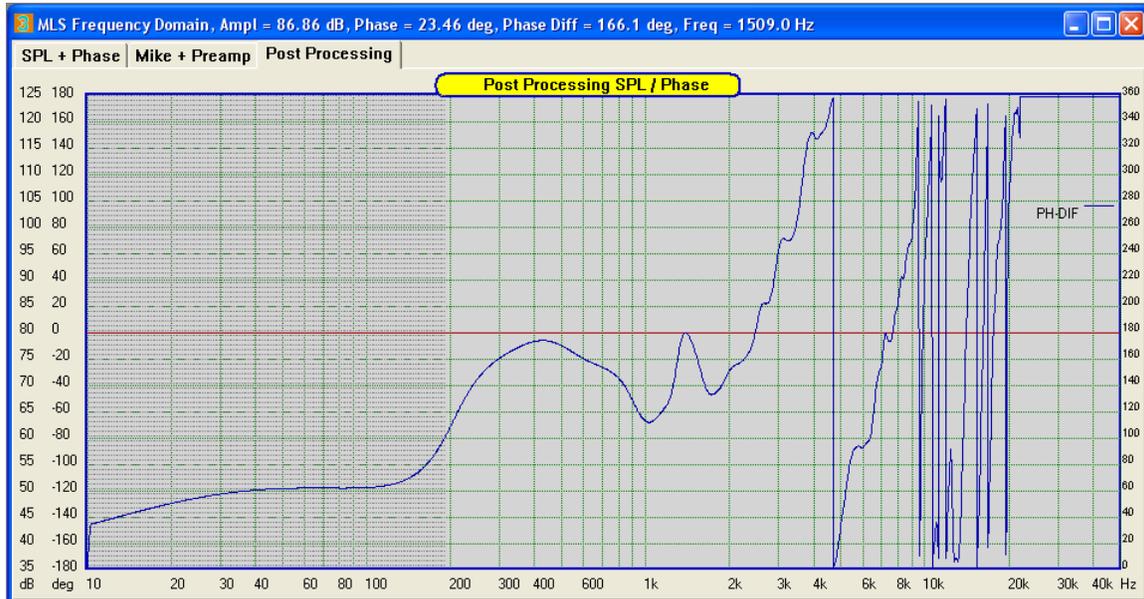


Fortunately, UE3 allows for easy manipulation of the “location” of the acoustic center. This is accomplished by introducing a small delay to the “forward” driver – in this case the tweeter.

The amount of delay can be calculated by comparing woofer and tweeter phase responses measured with the microphone located approximately half-way between woofer and tweeter center axis of rotation. I have located the microphone 1.0 m from the front baffle in such location, and measured both drivers without changing the measurement setup. The result is shown on the picture below.

Using a new UE3 functionality allowing straight visualization of phase difference between two data sets in buffers – see below – I have obtained a result of 166deg at 1500Hz.





Delay = (phase difference) \* 1000 / (360 x Fc) = 166 x 1000 / (360 x 2000) = 0.23msec.  
This value is entered in UE3 as the tweeter's "Delay" parameter.

The 230usec is equal to 79mm of acoustic centres offset. Given, that tweeter's dome is about 5mm in front of the baffle, this leaves us with the acoustic centre of the woofer, located about 75mm behind the front baffle. This would work, if the driver files represented minimum-phase measurement data.

There is perhaps a simpler method for assuring the driver's AC offsets are accounted for. I used 1.0m distance measurements for both drivers, and removed the same amount of time-of-flight from both drivers. What was left, the was phase response with the AC distance embedded in both cases. Now, if I use UE3 to phase-linearize driver's file created such way, the phase will be linearized, including the path differences.

Minimum-phase system delay adjustment is explained in the UE3 User's Manual, <http://www.bodziosoftware.com.au/UE%20V3%20Manual.zip> in the "Non-Linear Phase System" chapter and will not be repeated here.

In summary, the most involving and time consuming part of designing loudspeakers in this project seems to be the acoustical measurement itself. This issue is present in other loudspeaker design projects too, so no point procrastinating about it. Without anechoic chamber, the substitute techniques work reasonably, with an occasional hick-up. UE3 has reduced all other design and performance issues to a trivial button presses, or simple selections, with the exception of enclosure design – still performed using SoundEasy V18.

After all this hard work, now comes the enjoyable part - my loudspeaker is ready for listening tests, so that I can adjust voicing to my taste.

## Listening tests

I listened to this loudspeaker only, so I had no chance of evaluating any linear-phase improvements related to spatial sound reproduction. This will come later, when the whole system is auditioned.

For now, I only wanted to understand tonal balance, dynamic range and overall quality of the sound provided by the loudspeaker powered by UE3. I adjusted the UE3 gain of the tweeter amplifier to the same as woofer, plus 4dB. The 4dB rise is necessary due to higher efficiency of two woofers combined, accounting for diffraction compensation as well. So, now my 2-way, 3-driver loudspeaker is in full tonal balance over the whole volume range of UE3, with no possibility of overdriving or distortion.

I played CDs using an external CD-player (44.1kHz/16bit), so I had 3 A/D converters in the audio chain. This was possibly the worst case scenario for evaluating the dynamic range of the system. With the UE3 volume set to very loud, I listened to the loudspeaker when there were breaks between songs. I wanted to subjectively evaluate the electronic “noise floor” coming from the PC. The only faint noise I could hear, was when I stacked my ear right into the driver, and all this was at loud volume setting. Obviously, I could not play music with my ear so close to the cone. The faint noise becomes inaudible 10-20cm from the cone.

Conclusion – the published Dynamic Range of Delta1010LT sound card (Input = 99.6dB, Output = 101.5dB A-weighted) seems more than adequate for normal-to-loud listening levels.

As far as tonal balance is concerned, the 4dB boost in tweeter level flattens the overall SPL, and gives you a perfect starting point to voice the loudspeaker to your individual taste. HBT equalization extends the bass, removes diffraction distortions, and transforms the overall frequency response into a flat line. Now, the sound is balanced, and I was pleased with the amount of solid, low-end output that the set of 2x8” drivers produced. High frequencies could be described as “smooth”, pleasant and detailed.

Overall sonic quality was excellent, as I expected from this medium-sized, HBT-equalized loudspeaker. This system is intended as centre speaker.

I have also compared this loudspeaker to my Dali centre speaker – Concept Centre. The difference is quite dramatic, as the Dali loudspeaker has 72Hz low-end 3dB frequency. I am not sure why Dali insisted on such arrangement. When listening to dialogues on Dali loudspeaker, male voice seems quite affected and even female voice sounds unnatural.

## References

### Woofers' data

Rated Input Power: 60W RMS

Max Input Power: 100W

Impedance (300Hz):  $8\Omega$

Frequency Resp: 35Hz-4kHz

Resonant Frequency: 35Hz

Sensitivity 1W @ 1m: 91dB

Voice Coil Diameter: 35mm

QMS: 2.87

QES: 0.28

QTS: 0.25

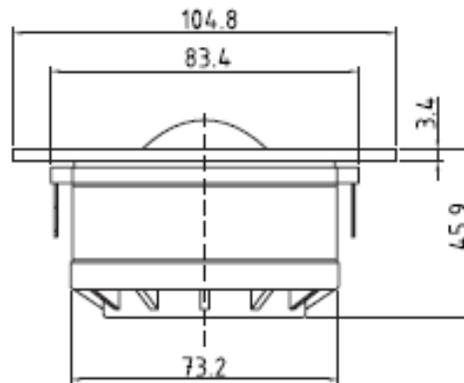
VAS: 48L

Nett Weight: 2.22kg

Magnet Weight: 848g

Overall Diameter: 205mm, Cutout Diameter: 183mm, Mounting Depth: 85mm

### Tweeter's data



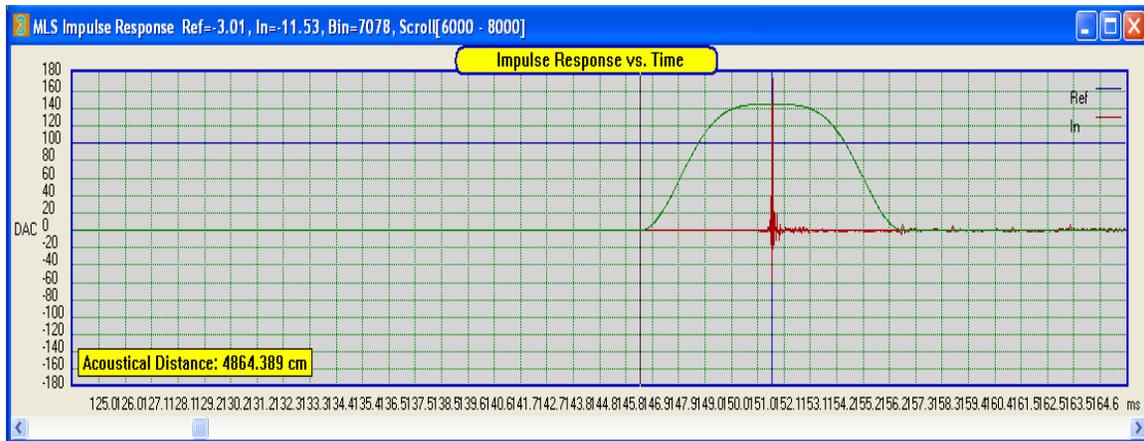
Impedance ( $\Omega$ )	4
Re ( $\Omega$ )	2.8
Le (mH) @ 10 kHz	0.04
Fs (Hz)	530.9
Qms	2.43
Qes	1.06
Qts	0.58
Mms (g)	
Cms (mm/N)	
Sd (cm <sup>2</sup> )	6.6
Vd (liters)	
BL (Tm)	
VAS (liters)	
XMAX (mm)	
VC Diameter (mm)	29
SPL (dB@1W/1m)	88.0
RMS Power Handling (w)	100
Usable Frequency Range (Hz)	1200 - 20000

## UE3 Acoustical Measurements

Acoustical measurements of UE3 system were performed using SoundEasy V18 in my AV room. Due to the size of FFT window (5.2ms on both sides of IR), the lowest reliable SPL/Phase frequency is 190Hz and the region below 190Hz is shaded on the figures below.



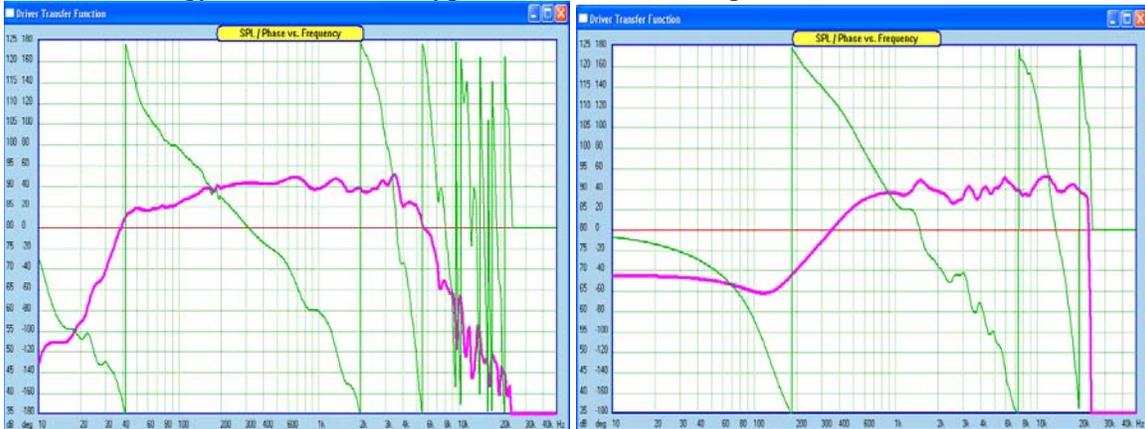
Acoustical measurements were performed with new, symmetrical FFT windows in MLS system – see figure below.



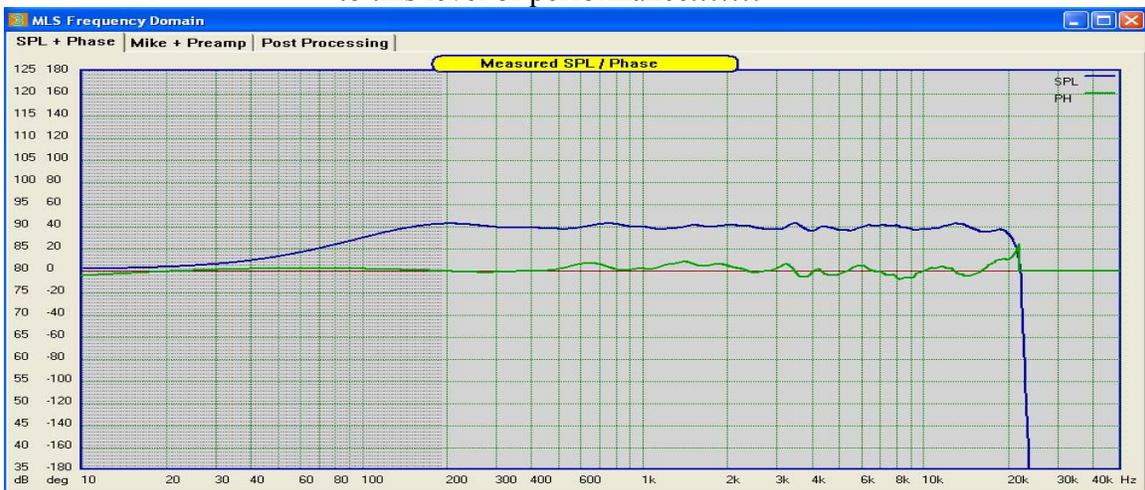
We are now in a position to review measured SPL at 1.0meter, on design axis, where the microphone is on tweeter's axis.

Pink/Blue = Magnitude response, Green = Phase response

UE Technology takes us from a typical level of driver's performance.....



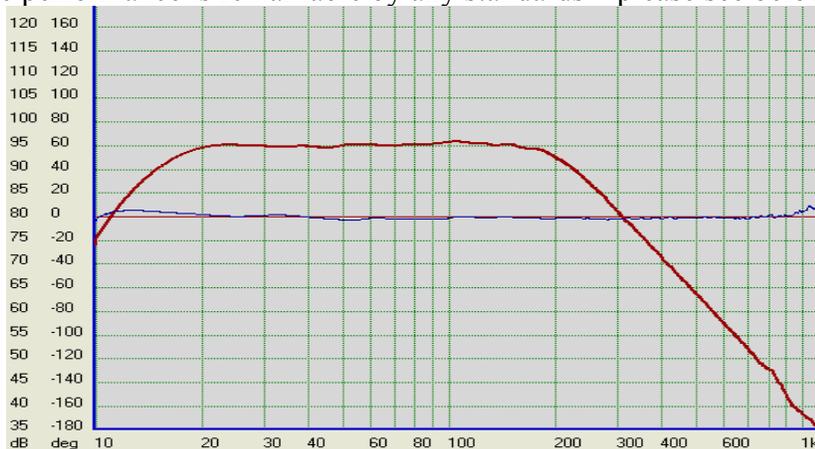
to this level of performance.....



It is clearly observable, that SPL is remarkably flat ( $\pm 0.8$ dB) below 20kHz. Even though the SPL/Phase measurements of the complete system are not presented here below 190Hz, the response continues to be flat below this frequency. In order to provide these plots for a full-range system, I would need an anechoic chamber. However, low-frequency performance of the HBT equalizer applied to a subwoofer is confirmed in:

[http://www.bodziosoftware.com.au/LP\\_MP\\_Subwoofer\\_Tests.pdf](http://www.bodziosoftware.com.au/LP_MP_Subwoofer_Tests.pdf)

Indeed, the performance is remarkable by any standards – please see below.



## Comments on Measurement Accuracy

Drivers' frequency response generation is not a simple process. Woofer's transfer function was measured and "glued" together, using four elements: (1) close-mike measurement of port's SPL, (2) close-mike measurement of driver's SPL, (3) modeled diffraction, (4) 1.0m distance SPL measurement – let's call it far-field. Then, tweeter's SPL response, measured at 1.0m was incorporated.

The above method is only a substitute for a proper 1m/1W anechoic chamber measurements, and as such, one would expect, that this less accurate method will result in some deterioration in flatness of both: SPL and phase. Then there is the issue of microphone's phase response – for which data is not available. Phase response above 15kHz is basically a wild guess. Calibration file for microphone was pretty much estimated from the available information.

The amount of SPL octave smoothing used when generating driver's SPL curves should be minimized. This is because overly smooth curves, with shallow dips and valleys will not be sufficient to equalize raw driver.

With 48kHz sampling frequency, any data above 22kHz is basically an artifact of digital processing. You may expect wild SPL/Phase irregularities in your measurements there, and these should be discarded.

Considering all the above, I regard the measured on-axis and performance very good indeed. Off-axis performance of UE3 system is briefly discussed in general in [http://www.bodziosoftware.com.au/Rear Loudspeaker Measurements.pdf](http://www.bodziosoftware.com.au/Rear_Loudspeaker_Measurements.pdf).

Anyway, it's just three drivers and some wires in a square box, but performance-wise (thanks to the UE3 Technology), the front speaker is a very well performing, medium-size and inexpensive loudspeaker.