Linear-Phase and Minimum-phase Subwoofers

By Bohdan Raczynski

Project statement

The goal of this project was to compare a standard minimum-phase and acoustically linear-phase subwoofers, with a 3dB bandwidth of 18-120Hz, and maximum SPL of 120dB across whole operating band. These requirements are basically aligned with a subwoofer requirements for 5.1HT (or 5.2HT) system.

Driver review suggested, that DSP-linearized, McCauley 6174 18" driver would meet these requirements in about 300litre vented enclosure. Possible corner placement combined with adequate amplifier (400W-800W) should secure the 120dB SPL level.

However, the critical part of this project was the requirement for the acoustically linear phase of the design. To meet this requirement, Ultimate Equalizer V3 was used in linear-phase mode.

The enclosure

A large, 300Lt vented enclosure (W=60cm, H=90cm, D=60cm) with internal bracing has been constructed. Enclosure resonance has been accomplished with two, 110mm in diameter PVC vents, tuned to 20Hz. The length of each vent is approximately 40cm. The driver was front-mounted, and ready for initial measurements.

Loudspeaker placement for measurements

While measuring subwoofers, the acoustic environment can be a major contributor to the accuracy of the measurements. With no access to an anechoic chamber, there are basically three options that can be contemplated for this task: "ground-plane measurements", "pit measurements", and "close mike" measurements.

The first choice was the ground-plane technique. I have evaluated noise level in front of my house for a couple of weeks to see if there are any "quiet periods", that I could slot into for making the measurements. No such luck. Between insect noise, distant (but frequent) car noise, birds chirping, household noises, kids playing, wind noise, distant aeroplane noise and occasional dogs barking, I stood little chance of completing the measurements without adverse noise contamination from the environment. However, I did drag out the measurement gear and perform some rudimentary measurements, to have a reference point for comparison with in-room measurements.

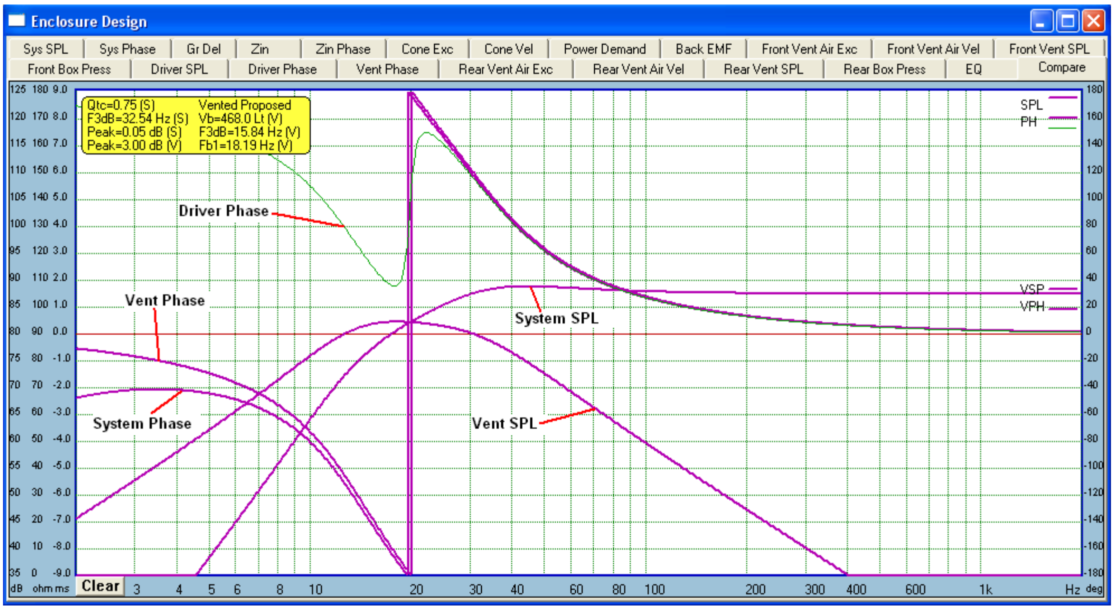
The main measurements were therefore conducted outdoor. Fore the indoor comparison, due to inherent background noise in a typical household, it is not expected, that dynamic range of the measurement, will be greater than 50dB. Therefore, once the SPL curve drops below 50dB, corresponding phase response will manifest itself just as noise.

The working assumption was, that every step in the measurement process would have to be examined and correlated with known theoretical aspects of loudspeaker operation in enclosed spaces, and if a discrepancy was found, it would have to be resolved before continuing with the goal of the project.

Since the close-mike technique was used, there was a good chance, that room resonances as such would not manifest themselves too visibly in the frequency response plot. This is the idea behind the close mike technique principle anyway. However, I did expect SPL taken during room measurements to be visibly more irregular, with small wiggles, though.

It’s worth noting, that measurement power amplifier and microphone pre-amplifier have been modified from their original commercial design to extend their frequency response quite far into the low-end of the frequency range. The power amplifier is based on LM3876, a 50Watt integrated design from National Semiconductor, and originally had 3dB cut off at 16Hz. Microphone pre-amplifier is based on low-noise, LM833 chip. This amplifier was also modified for the phase shift at 10Hz to be negligible, and also provide microphone DC bias and loading impedance. Microphone used was CLIO Mic01.

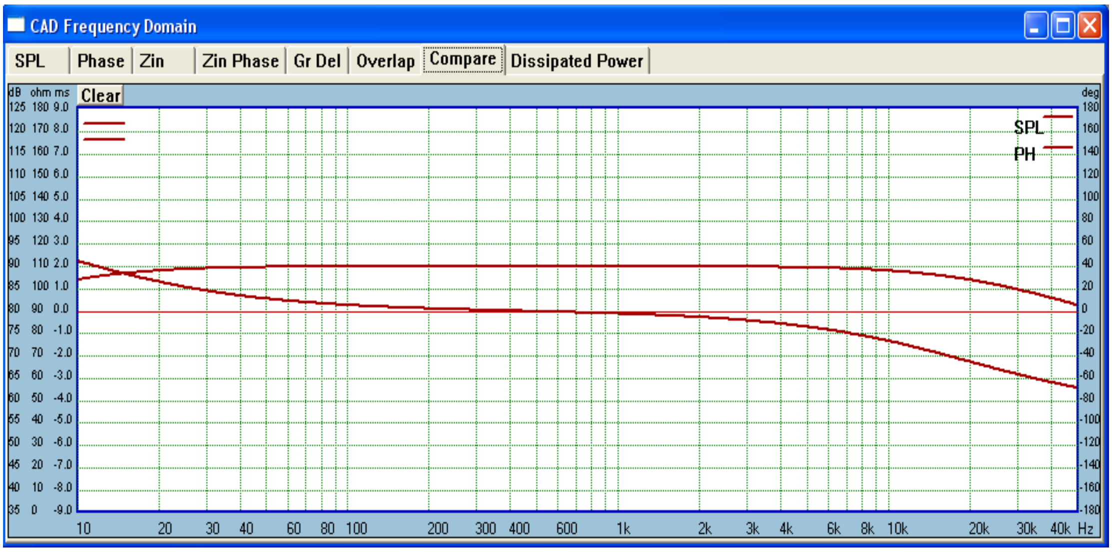
Prior starting close-mike measurements, I modelled the SPL and phase responses of a vented enclosure. This gave me a reference point for comparison with the actual measurements. I simply needed to see close agreement between theory and measurement results. For instance, driver’s phase stays in 0 to +180 deg region, and it has an N-shape ripple around box tuning frequency. Port phase looks distinctly different. It makes 360 deg revolution at box tuning frequency (from -180 to +180 deg). System phase follows the port phase very closely. These are the typical characteristics I would hope to see in real measurements.



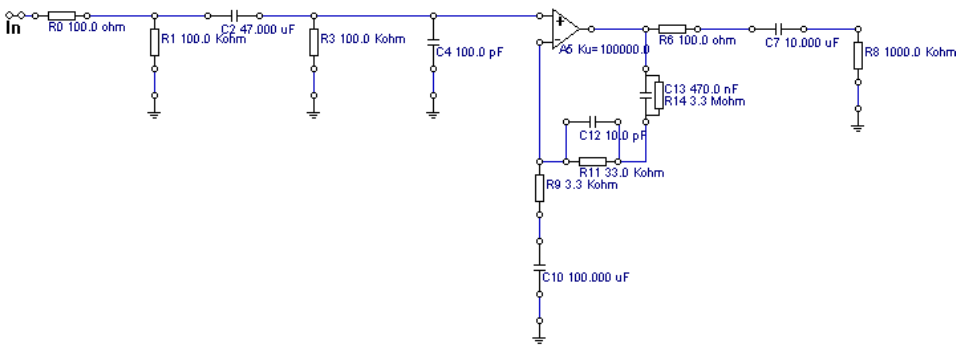
Turns out, that this precaution was well advised. In the later part of this paper, you will see the same phase characteristics during the actual in-room measurements.

Microphone pre-amplifier with low-frequency phase-correction circuit

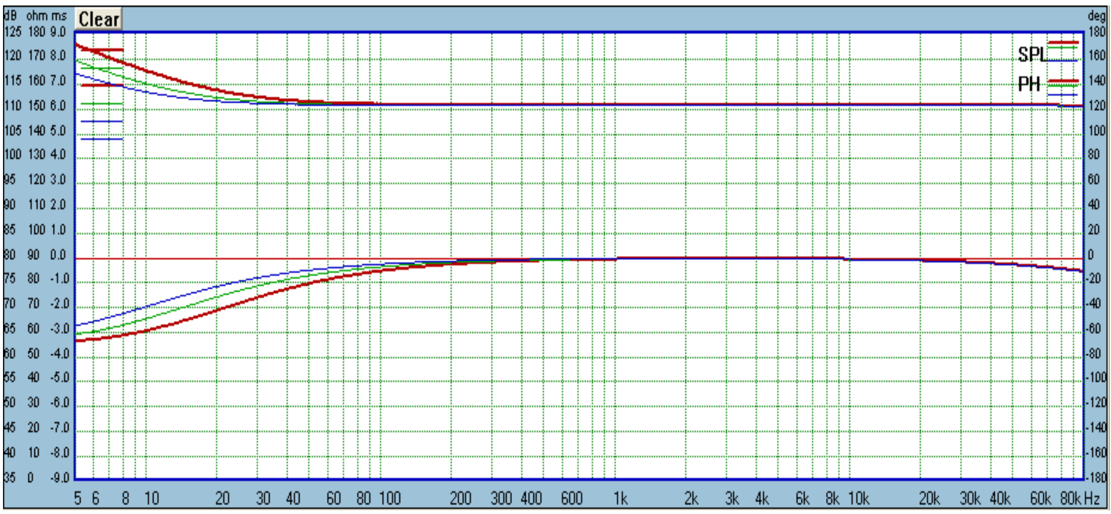
Having examined CLIO Mic01 specification, I have developed an approximation of microphone amplitude and phase responses. Please note the +45deg phase shift at 10Hz.



The above phase shift needs to be accounted for during the measurements or post processing. One option is to compensate for it in the microphone pre-amplifier.



C3 // R14 are the mike phase compensating components. C=220n, 330n and 470n



Frequency (top) and phase (bottom) responses of the mike pre-amplifier.

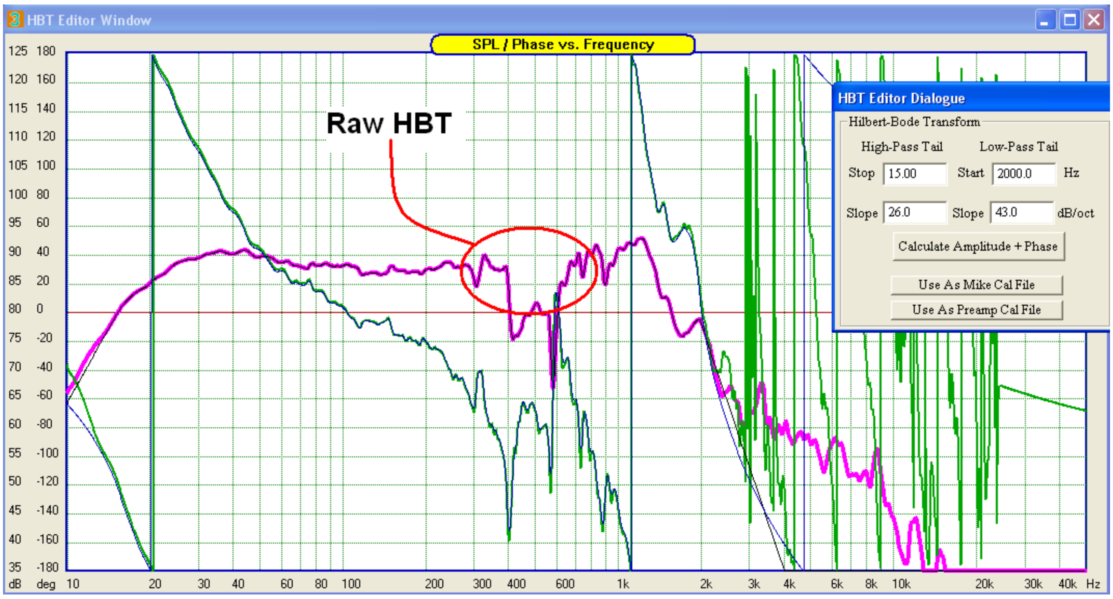
Outdoor Measurement Setup

The outdoor tests were extremely tedious and rather disappointing. This is due to unexpected amount of background noise, even on a quiet Saturday afternoon. Basically, the testing area was never completely free of background noise, and the most obtrusive was the wind noise and surprisingly, a very distant aeroplane noise. I have managed to take several measurements in the configuration as shown on the pictures below, at 1meter distance, and selected the best one for processing and comparison record.

In case you wander what are the two grey circles above the driver – this box used to be a 3-way system, with midrange and tweeter located above the woofer. I have since then pulled out crossover and these drivers, and bolted 3mm aluminium discs in place of the drivers. It is now a one of two subwoofers in my 5.2HT system.

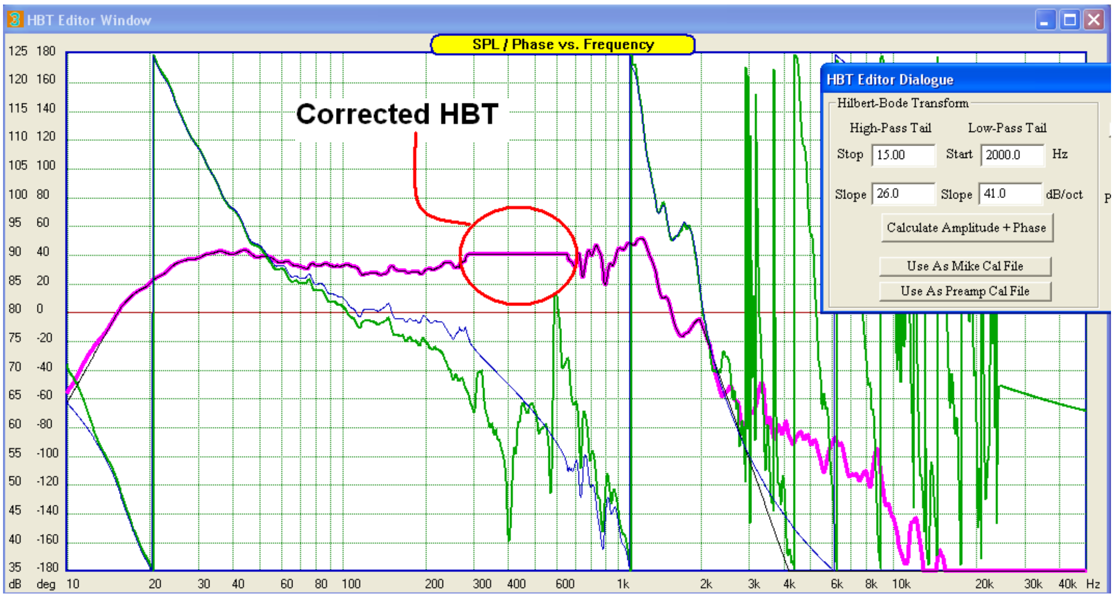
Outdoor measurement results are valid till about 300Hz.

Collected impulse response was post-processed using HBT and the excellent agreement between measured amplitude and phase and HBT-generated phase was obtained.



Measured SPL/phase and HBT derived amplitude and phase

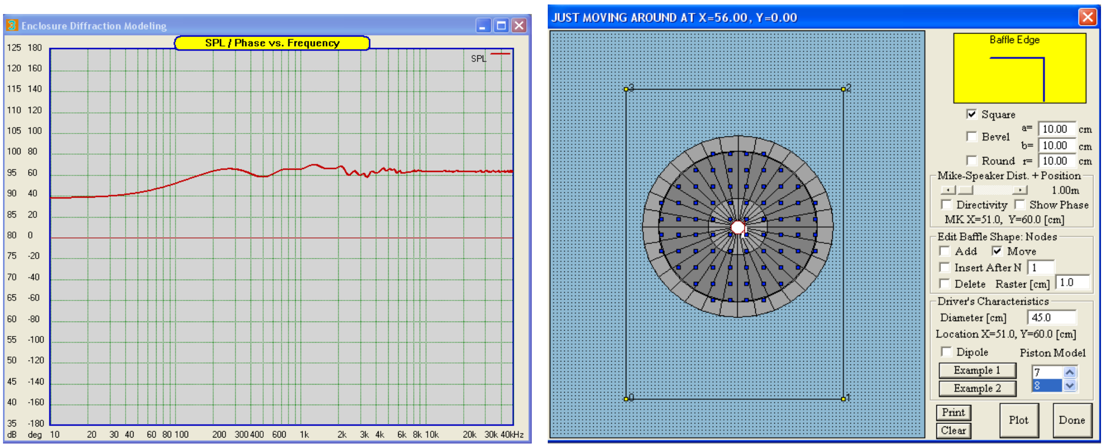
However, I had somewhat lower confidence in this measurement in the frequency range from 300Hz and above. This is the range where diffraction kicks-in, and my measurements were contaminated by ground reflections – therefore, not exactly following the anechoic diffraction model. The SPL drops by 12dB at 400Hz and exhibits another sharp notch at 600Hz before returning to average level – it all looked suspicious. I have therefore decided to switch off this frequency range from HBT equalization. This is done by using pixel editor in File Editor screen. I have inserted a flat section of SPL between 300-650Hz at 90dB level. As you can see on the picture below, after HBT, amplitude fluctuations and phase fluctuations have disappeared there.



Indoor Measurement Setup

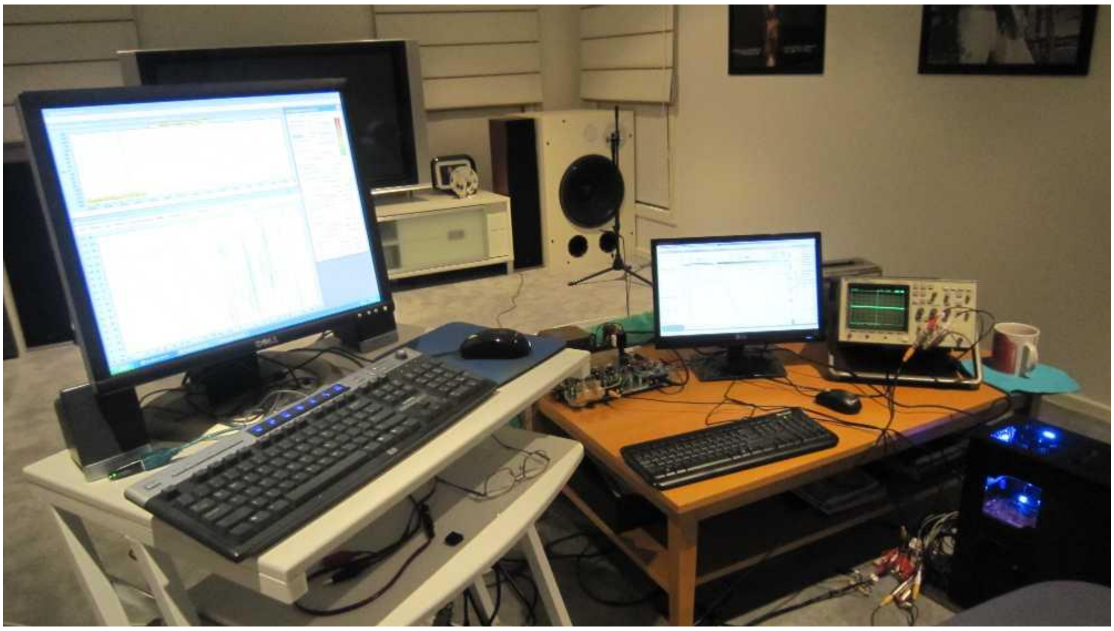
For the record, I have decided to try indoor measurements and compare the results with the outdoor measurements.

It was perhaps worth a try, as the goal of the project was to develop a subwoofer with flat response up to 150Hz, and the close-mike technique, coupled with diffraction modelling curve could yield satisfactory results. Ultimate Equalizer has diffraction modeller built-in, so this task was pretty simple. Here is the result.

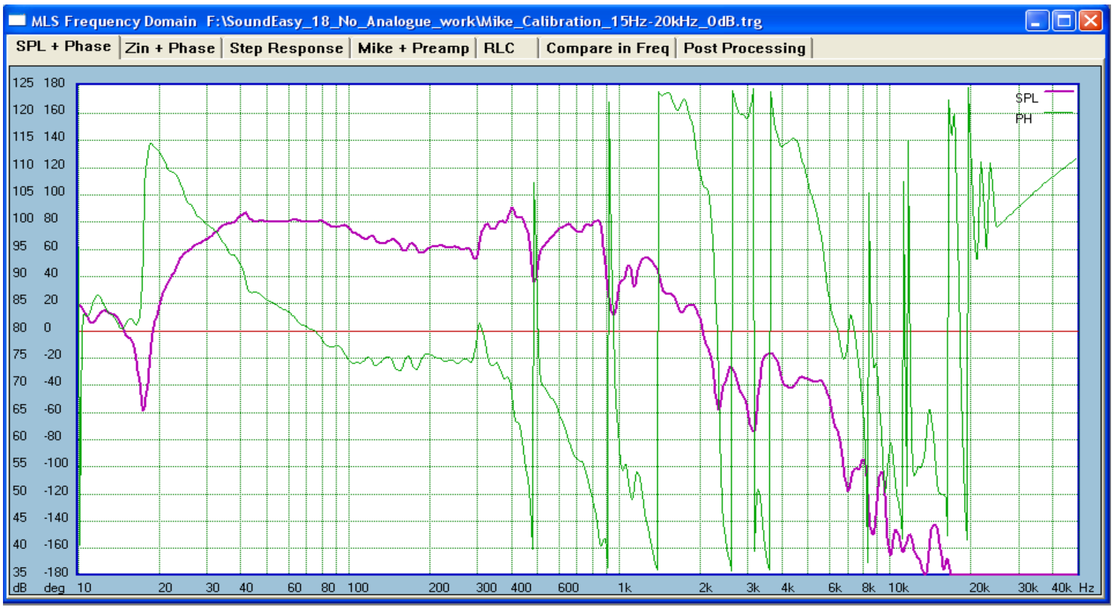


Diffraction calculated and included in the SPL plots

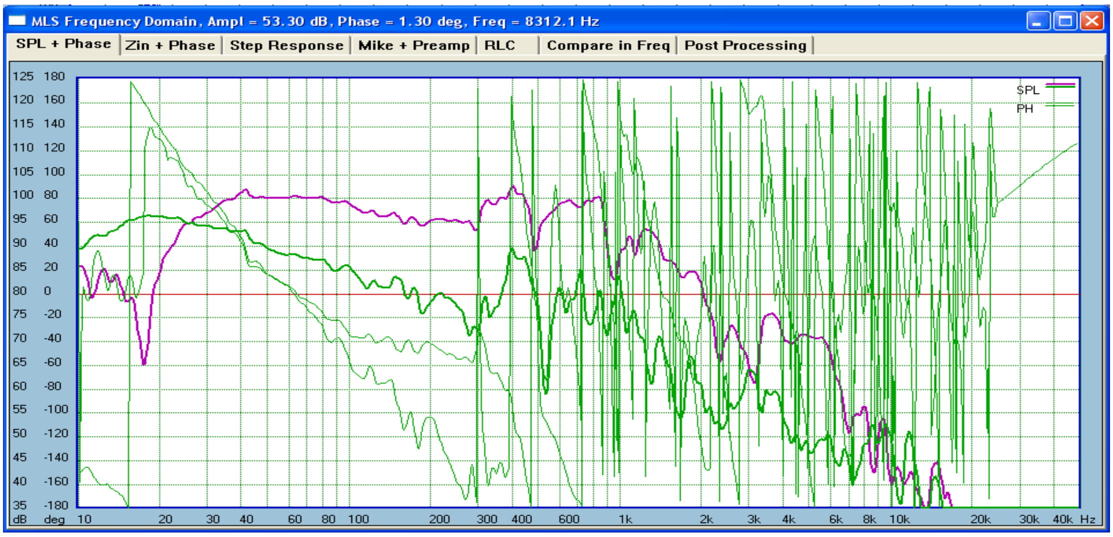
Listening room has the following dimensions: Length = 6.5meters, width = 4.5meters and Hight = 2.6meters. As you can see on the picture below, I used one computer to run MLS testing on UE3, and another computer to run the UE3. This way, I could also confirm operation of the UE3 equalization function.



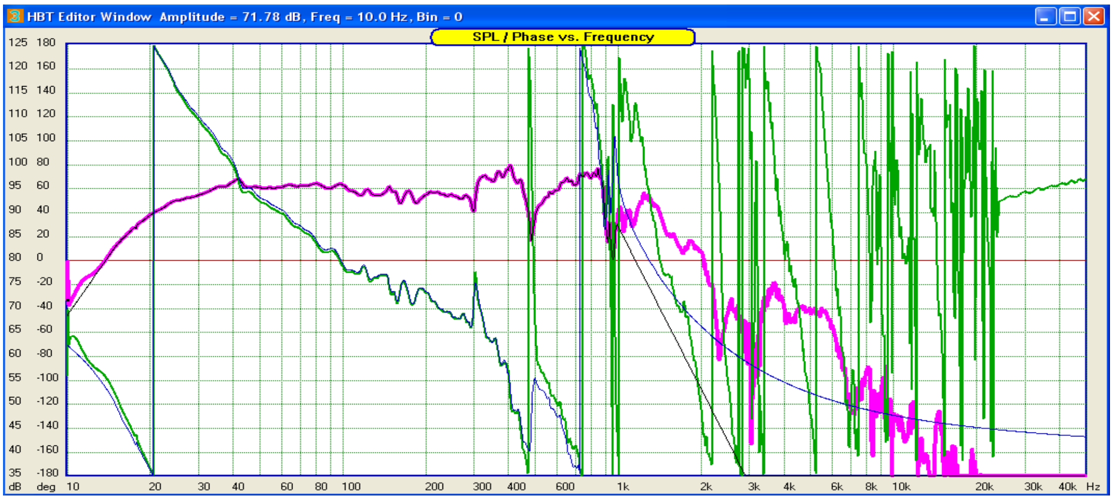
Next, the driver and port measurement results. Please note the phase response of the driver and port – it's in agreement with earlier theoretical modelling.



Driver close-mike SPL/phase measurement.

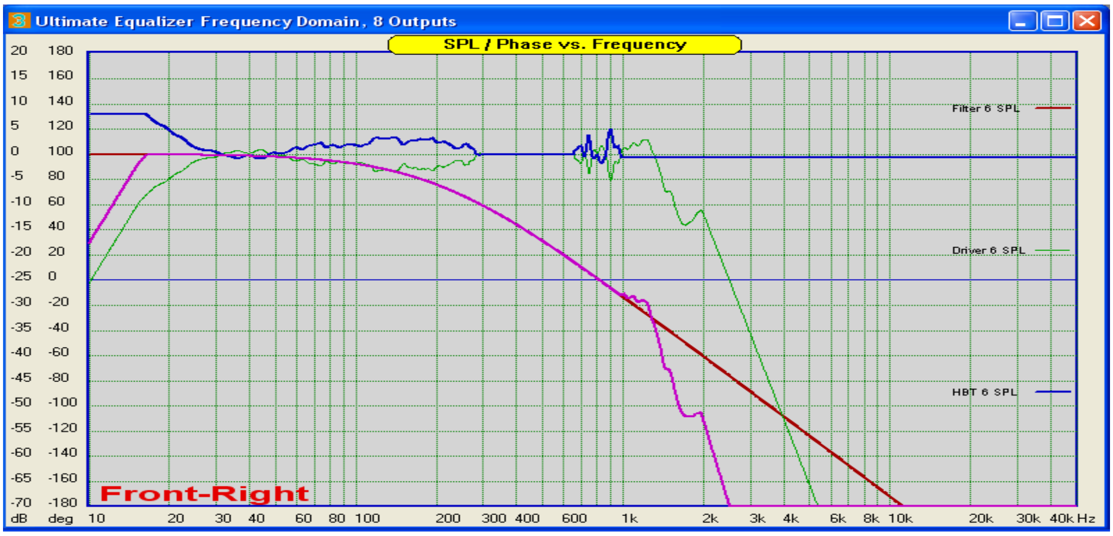


Driver + port close-mike SPL/phase measurement. Port shifted down by -8dB due to Sd differences.



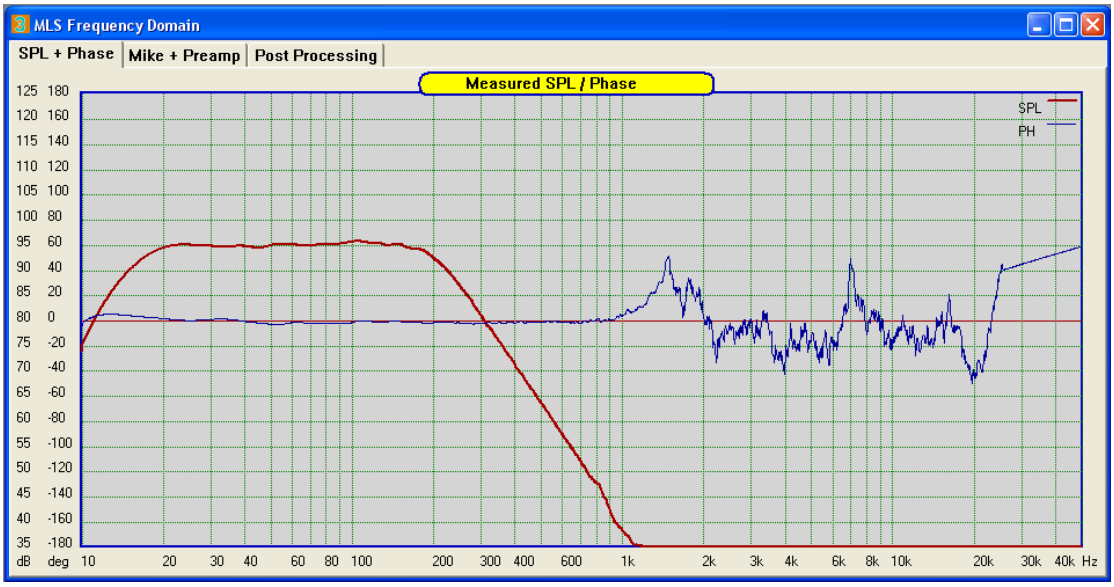
Driver + port + diffraction added together. Also shown HBT to 1kHz.

In the next stage, UE3 correction curves were constructed as shown on the picture below, and played by UE3.



Green curve – Loudspeaker measured SPL.
Red curve – Target Linkwitz filter: 200Hz/12dB/oct.
Blue curve – UE3 correction curve developed with HBT.
Pink curve – Loudspeaker's equalized response.

Temporarily, I have decided to try to extend the subwoofer bandwidth to 200Hz and switch to 24dB/oct Butterworth filter to see if it's possible to do this without major consequences. Running UE3 on one PC and using the other PC for measurements, I have obtained the following SPL/phase characteristics of the equalized subwoofer. Here is the result of the equalization as measured in-room.



In-room subwoofer frequency response in Linear-Phase Mode.

It is observable, that both SPL and phase are near-perfect examples of linear-phase subwoofer, operating between 18 – 200Hz.

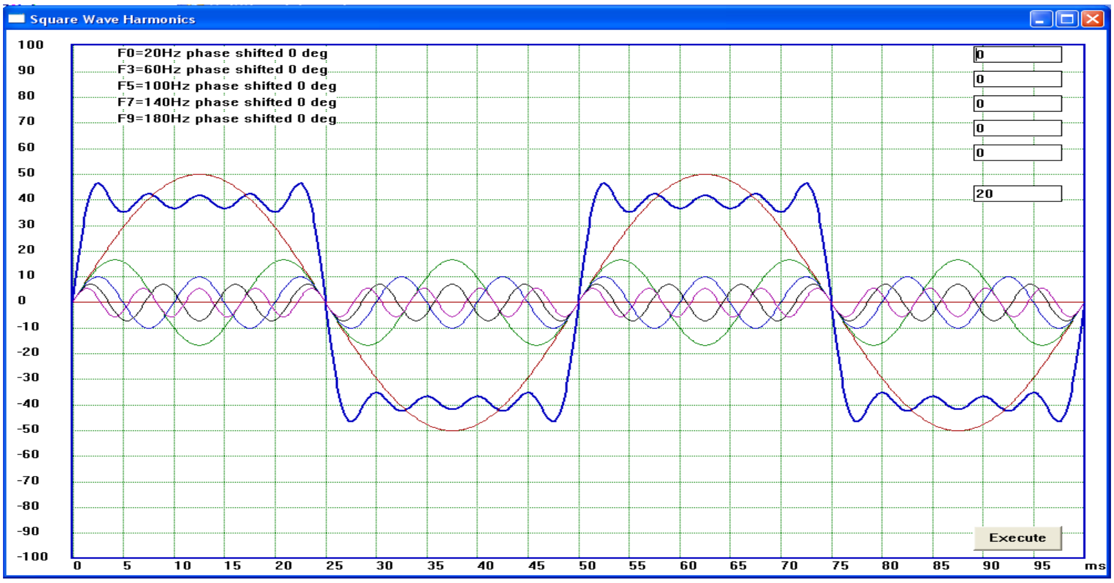
Flatness of the SPL and phase responses, achieved in this measurement, are attributable to HBT-style of equalization, and show near-perfect characteristics, even for in-room measurements, and with DSP pushed to it's limits and into the sub-audio frequency range. As predicted, phase response for the in-room measurements with SPL below -60dB, shows up as noise of no consequences.

It can not be stressed enough, that proper execution of this project should involve anechoic chamber SPL tests or at least more elaborate ground-plane or pit-type SPL measurements, so the resulting SLP and phase curves would be even more smoth.

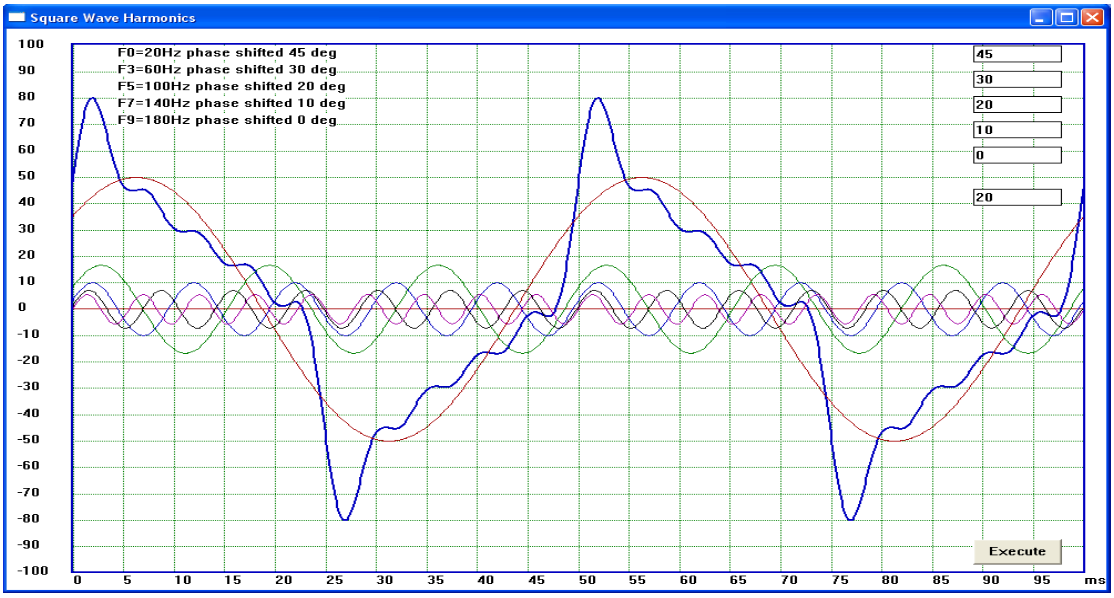
Outdoor measurements proved to be extremely difficult, and due to variability of the background noise conditions, repeatability was highly questionable. The in-room measurements introduce another set of problems, but at least, these are repeatable enough, so some countermeasures can be developed.

Square wave phase shifts considerations

A 45deg phase shift at the fundamental frequency does not seem like much, however, it will drastically alter the shape of a square wave recombined from it's shifted components. Here is a 20Hz square wave recombined from up to 9th harmonics, with all waveforms in-phase.



Here is the same process, except, that fundamental frequency of 20Hz is phase-shifted by 45deg, third harmonics is shifted by 30deg, fifth harmonics id shifted by 20deg, seventh harmonics is shifted by 10deg.



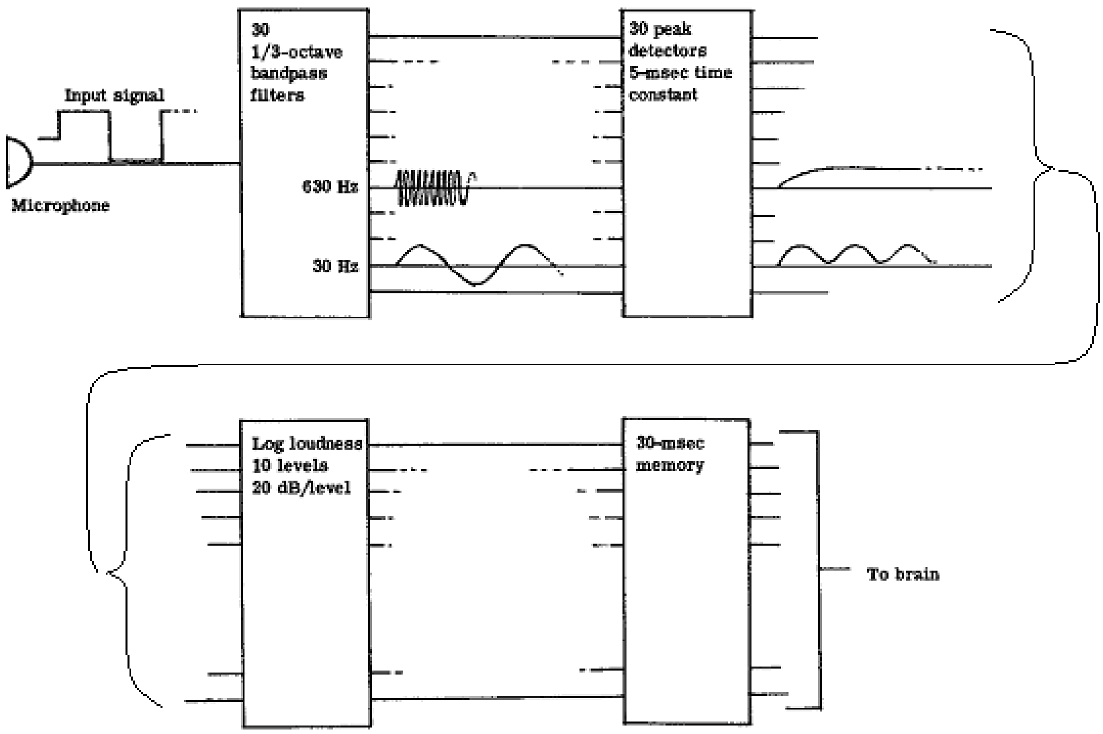
It is easy to observe, that the square wave almost become a triangular wave, even with no change in amplitudes of the harmonics took place. Clearly, in order to preserve the characteristics of source of the signal, phase linearity must be maintained.

Audibility of phase shifts – short scientific comments from BAS

In mid-70', Mr Mark Davies, was a doctoral candidate studying psychoacoustic phenomena at MIT, and was instrumental in experiments verifying a new model of the hearing process. The model of the ear that has been proposed by Professor Campbell L. Searle, formerly of MIT and later, at Queens University in Kingston, Ontario. Since then, the model has been widely used. Here is an excerpt from Boston Audio Society meeting:

"…The model attempts to account for all of the known psychoacoustic and physiological aspects of the human hearing process in such a way that an electrical analogue of the ear may be constructed that will simulate these effects. It is believed that the ear analyzes sounds in 1/3-octave bands spread uniformly through the audio spectrum. This behaviour is supported by measurements on cats' ears (which are similar to human ears), which showed individual nerve cells respond over 1/3-octave bands with band-edge response falling off at 96 dB/octave.

The ear model begins with a broadband microphone (representing the eardrum and bones connecting to the cochlea of the inner ear) feeding a bank of 30 1/3-octave filters (the individual frequency-sensitive nerve cells). This is followed by a parallel set of 30 peak detectors whose 11 outputs are proportional to the peak values of the signals from each of the 1/3-octave filters. The detectors have a time constant of 5 milliseconds, which means that for signals beyond a few hundred hertz, the detector can no longer follow instantaneous level fluctuations and responds only to the envelope of the signal.



This is more graphically explained with an example from Figure above.

Accordingly to this theory, phase shifts are much more readily apparent in transient signals with low repetition rates. According to the model, the reason for this can be seen by, again, looking at the output of the peak detector. For a transient signal, the peak detector output would also be a transient, its value following the energy content of that particular frequency band. When phase shift is introduced, the energy in the frequency band over which the phase shift occurs tends to be delayed (phase lag) with respect to the rest of the spectrum, delaying the output of the peak detector (s) in that band.

The delay clues the brain that a change has taken place….."

So, was the phase shift audible?

In some instances, yes it was. Before I elaborate on the listening test results, the following needs to be explained:

I was able to compare a subwoofer with no acoustical phase distortions (flat line phase response) to a subwoofer with minimum-phase phase characteristics (typical phase roll-off for driver + crossover systems). There is a lack of internet literature describing this exact type of tests performed on subwoofers. The only paper I was able to source, that used phase-equalized loudspeakers (but not subwoofers) was ***The Audibility of Loudspeaker Phase Distortion*** Preprint 2927, by Mr Richard Greenfield, Dr Malcolm Hawksford, Department of Electronic Systems Engineering, University of Essex, Colchester, England.

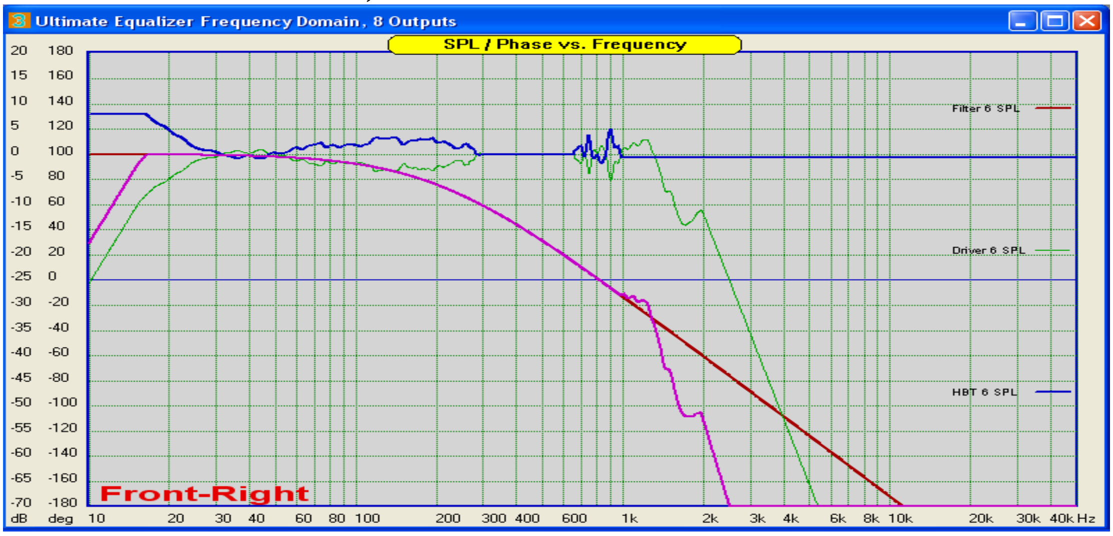
I decided to use artificially generated test signals: (1) square wave of various frequencies, (2) a pulse of various widths 1ms-100ms and repetition rate of 350ms, and (3) bi-polar pulse of various widths and repetition rate of 350ms. The reason for it was the ease of repeatability, and ease of differentiation between distorted and undistorted test result. Plainly speaking – it was obvious to see (particularly for you – the reader), which output waveform test result confirmed to the original (excitation) waveform. Out of the three signals above, I assumed, that the square wave and the bi-polar pulse, perhaps offered some resemblance to real-life encountered acoustical signals. The pulse signal was there to stress the subwoofer and bring out the worst of it.

This one is important. I have tested subwoofer alone, without complimenting it with a high-pass section. As the theory goes, linear-phase crossover is capable of reproducing impulse response perfectly, provided that low-pass section is complimented by high-pass section. Therefore, the pre-response of the low-pass impulse response is cancelled by the pre-response of the high-pass section, and the overall impulse response or step response is perfectly preserved (this is the feature, that minimum-phase systems can not do). So, without the complimenting high-pass section, the subwoofer was exposed to potential audibility of impulse response pre-response.

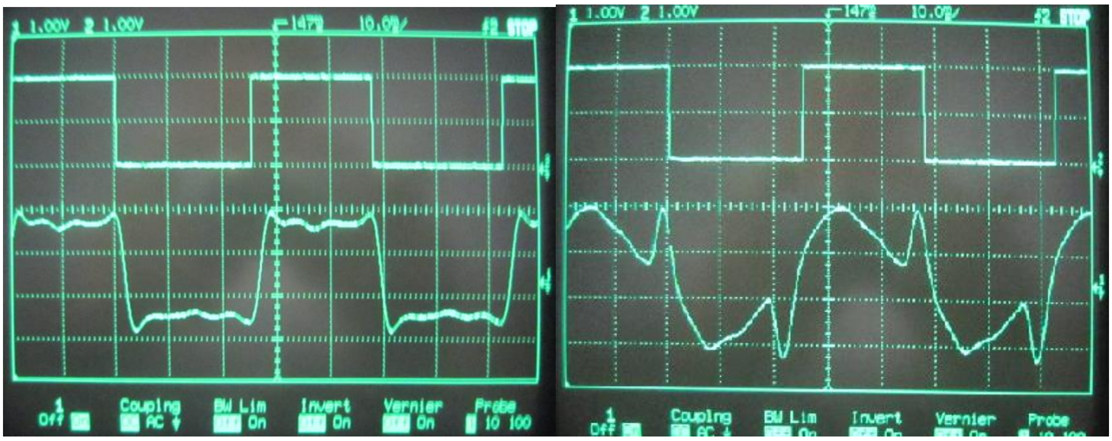
To make this situation even more complicated, the characteristic high-pass slope of the subwoofer, did not have a counterpart anyway, so there was a distinct possibility, that this could induce some form of pre-response effect.

Listening tests

Linkwitz 2nd order LP filter, with F3dB = 200Hz.



When listening to the minimum-phase and linear-phase versions of the subwoofer, with 20Hz square wave signal, the difference was audible. I expected the 20Hz, linear-phase output to have more “authority” in the bottom-end, but it was only slightly noticeable. However, that minimum-phase version had more audible “buzz” then the linear-phase version. Waveforms are shown below.



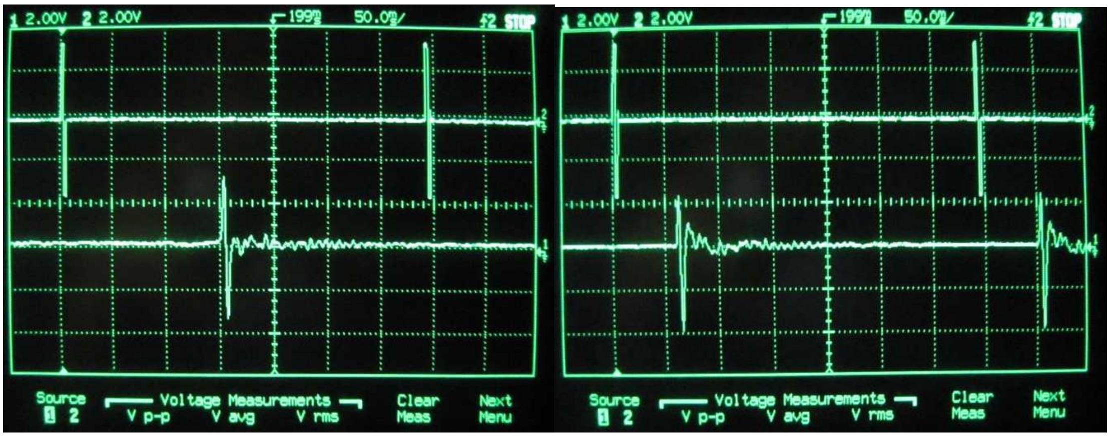
20Hz square wave: Linear-Phase Mode and Minimum-Phase Mode

Shown above, the time-domain comparison measurement results speak for themselves. It needs to be remembered, that we are dealing here with a very heavy-coned, 18” driver, low-pass filtered, in a vented (resonating) enclosure, and yet, the time domain performance is near-perfect accurate. It’s pretty amazing to see a vented loudspeaker, holding the acoustic pressure nearly constant for 25ms.

Next, I used 2ms-wide pulses separated by 350ms space as the source signal. On the 2ms pulse, the minimum-phase version delivered a more of a “thump” instead of a pop or a click. This is perhaps not surprising, as the post-ringing of the pulse extended to130ms and far exceeded the 30ms “memory effect” of the auditory system. Here, the driver, filter and vented enclosure added it’s own, combined signature. It is also observable, that the minimum-phase version of the subwoofer has converted the clearly asymmetrical pulse into a much more symmetrical bi-polar pulse with post-ringing. This is clearly visible on the screen shots below.

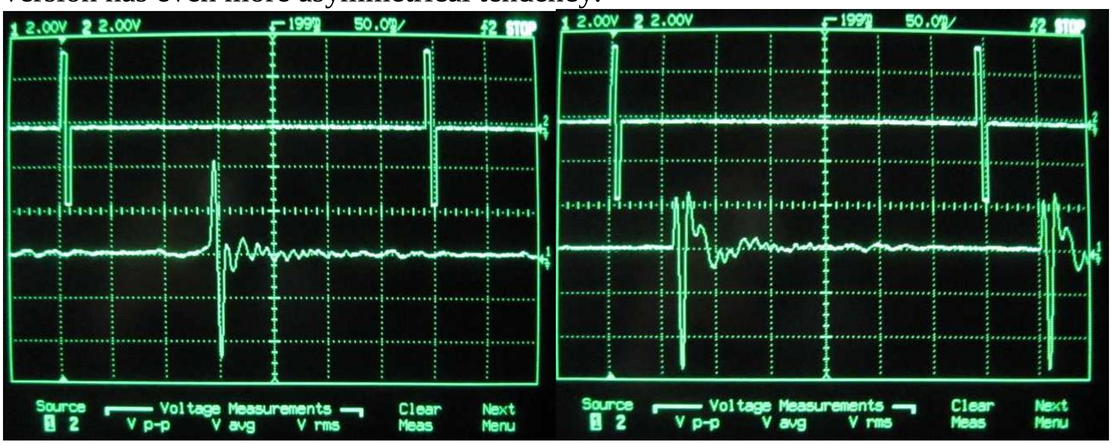
5ms Impulse in Linear-Phase Mode and Minimum-Phase Mode

When a 2ms bi-polar pulse was used for excitation, the minimum-phase version has done the opposite, and converted the symmetrical bi-polar pulse into a pulse with clear asymmetrical tendency. The ringing past the pulse is due to a more distant microphone placement, so now, the mike picks some of the room reflections.



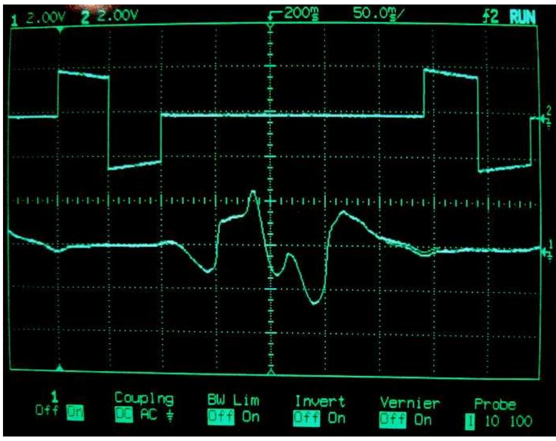
2ms Bi-polar pulse in Linear-Phase Mode and Minimum-Phase Mode

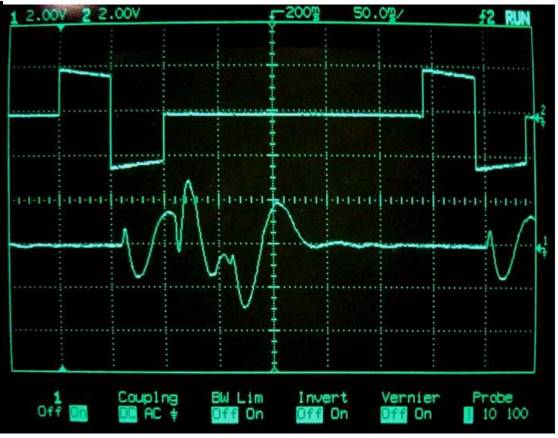
When a 10ms bi-polar pulse was used for excitation, the minimum-phase version has even more asymmetrical tendency.



10ms Bi-polar pulse in Linear-Phase Mode and Minimum-Phase Mode

In the final comparison, I used 100ms long bi-polar pulse. There was no chance, that either version of the subwoofer would reproduce this pulse, so I was only interested in the degree of distortion inflicted on the excitation signal by both versions of the subwoofer. The minimum-phase version has produced a sound that was dominated by **three** sharp pops and a very low-frequency vibration in the background. However, the linear-phase version delivered **two** distorted thumps. This would indicate, that minimum-phase version decided to emphasize and reproduce three slopes, rather then two thumps, still delivered by the linear-phase version. Pictures below depict this situation, but they do not really tell you what you hear. I included them for the record.





100ms Bi-polar pulse in Linear-Phase Mode and Minimum-Phase Mode

Regarding the pre-response. When I came close to the loudspeaker in linear-phase mode, with my ear stacked into the cone, I could hear a short, quite faint noise preceding the main sound. This noise was not there in minimum-phase mode. It is hard to describe this noise, as it was faint, and it did not have any ringing characteristics to it. It was just a short, faint noise. I would expect the pre-response to resemble a sound rather than a noise, because this is the way it manifests itself in the impulse response of the brick-wall filters – as a pre-ringing. I do not have an explanation for my observation, and since it was not audible at normal listening distance, I mention it here for the record only.

Listening tests conducted on phase-linear loudspeaker (Preprint 2927) by Mr Richard Greenfield, Dr Malcolm Hawksford and several other expert listeners also did not reveal any pre-response issues. Concluding from all the above, the pre-response of stand alone subwoofer (-12dB/oct, 120Hz Linkwitz filter), in normal HT application is not an issue.

The above tests were indeed extreme, and intended to reveal as much of the acoustical differences between minimum-phase and linear-phase implementation of the subwoofer as possible. The electrical results speak for themselves, and clearly show, that the linear-phase subwoofer is a far more accurate transducer. This was to be expected and followed the theory quite well.

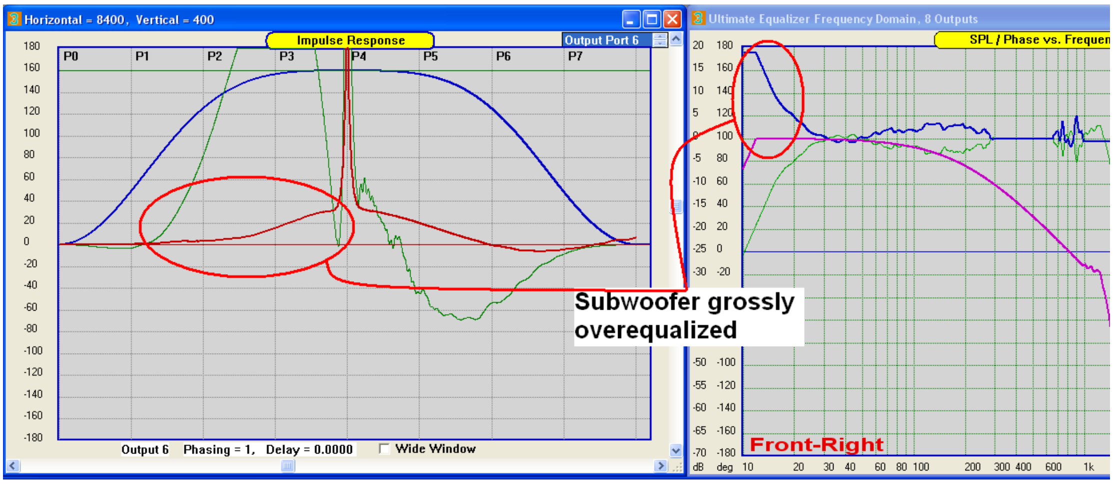
The acoustical results were actually quite surprising to me. It is hard to believe how much waveform destruction needs to take place before the auditory system is able to detect it. But eventually it does.

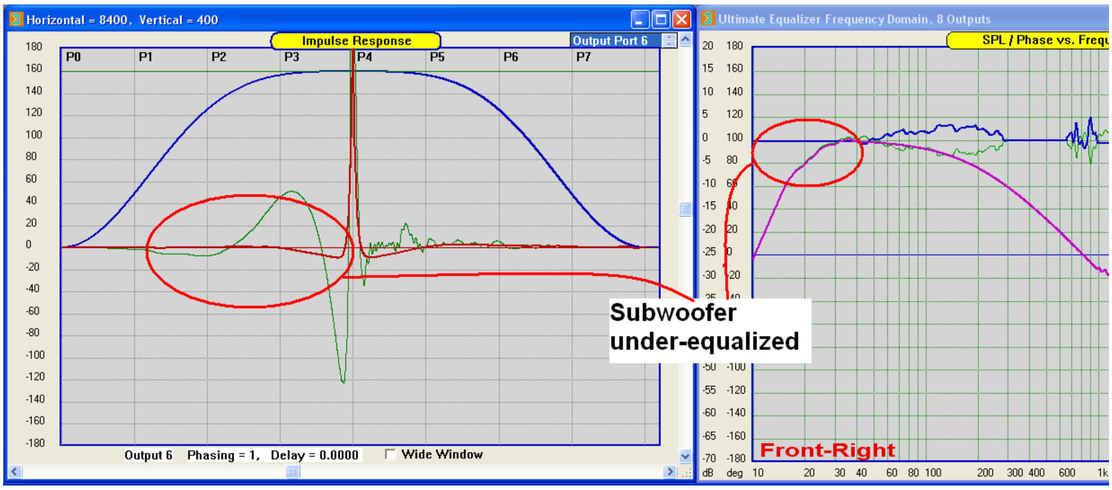
Impulse Response Tuning

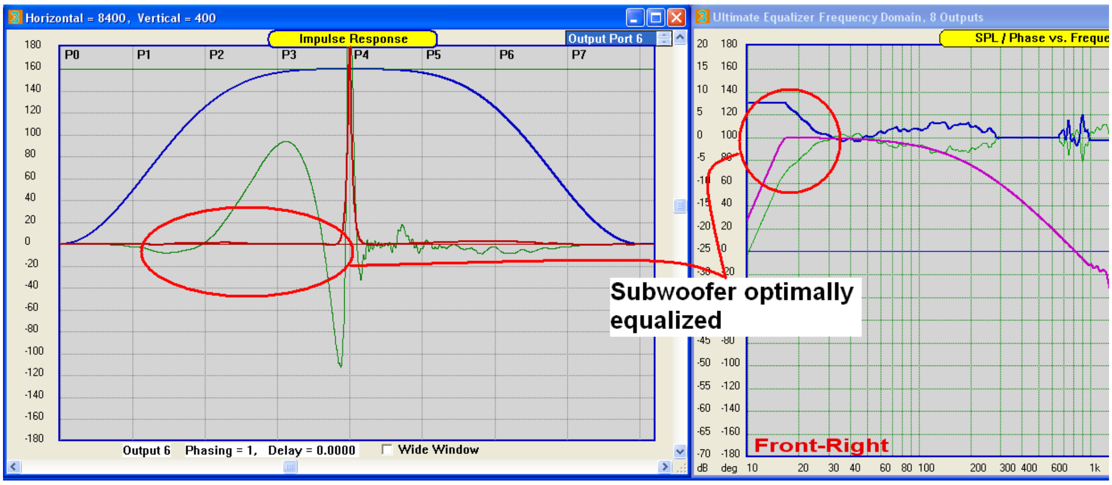
There seems to be a viable mechanism, by which the system (driver + UE3) impulse response can be optimized. “Optimization” is understood here as a process, that minimizes front-side deviations of the impulse response. Simply - we want the left-hand side of the impulse response curve to be as flat as possible.

The left-hand side of the screen shots below shows final FFT window (blue curve), UE3 correction impulse response (thin, green curve) and system (UE3 convolved with loudspeaker) impulse response (red curve).

The right-hand side of the screen shot shows corresponding curves in frequency domain: UE3 correction filter (blue curve), loudspeaker response (thin, green curve) and system (UE3 filter convolved with loudspeaker) response (pink curve).







The UE3 correction filter (blue curve on the left-hand side) is a direct outcome of the inverse HBT process, and the amount of equalizing boost depends on the HBT bandwidth extension into the low-end of the frequency range.

The “over-equalized” case corresponds to HBT bandwidth of 12Hz – 1000Hz.
The “under-equalized” case corresponds to HBT bandwidth of 40Hz – 1000Hz.
The “optimally-equalized” case corresponds to HBT bandwidth of 16.5Hz – 1000Hz.

The optimally equalized case would require about 7dB of equalization, or 5 times the nominal output power. For example, subwoofer designated as 100W device would require a 500W power amplifier to enable the above HBT equalization between 16Hz and 1000Hz, and would achieve 3dB bandwidth of 15Hz-100Hz. So it seems, that if you start with a “naked” subwoofer, capable of 3dB response down to 20-22Hz, and equalize it to flat response down to 16Hz, you can also improve the impulse response.

Conclusions

Stress testing of a stand-alone subwoofer, using signals other then sine-waves and music, appears to be quite useful process, but not a common practice. Perhaps because subwoofers are there to compliment and augment low-end of the frequency range and basically never operate alone, without the other, complimenting drivers. From this point of view, my tests were perhaps somewhat exaggerated.

Secondly, my listening tests were conducted in-room. Inevitably, following the “first arrival” waveform, the room has imprinted it’s own characteristics on the sound, and this only made the process of discriminating between the two versions more difficult. I run the tests for several days and repeated the same tests at different times to make sure, that I hear the same thing every time.

On the other hand, it was beneficial to uncover less known, and perhaps somewhat unexpected characteristics of linear-phase approach to subwoofer design, and confirm, that an excellent results can be achieved. The results tend to surpass minimum-phase design.

Electrical test results are best compared and examined by inspecting the CRO screen shots. Summarizing acoustical performance I would conclude the following:

20Hz square wave testing revealed more buzz on minimum-phase subwoofer, and I would attribute this to the "spike" at the end of the waveform transition. Linear-phase subwoofer exhibited tighter and subtly heavier bottom end.

2ms-wide pulse testing revealed that minimum-phase version delivered a thump with slightly more "boom" in it, possibly due to converting the excitation pulse into bipolar-pulse with audible post-ringing. I would prefer for this not to happen, and preferred somewhat tighter sound of the linear-phase device.

3-10ms bipolar pulse again sounded subtly different on both versions, with minimum-phase version sounding more "solid".

100ms bipolar pulse testing revealed, that the minimum-phase version produced a sound dominated by three audible pops and vibrations, while linear-phase version produced two distorted thumps.

The above differences were subtle and sometimes difficult to detect, with the 20Hz square wave buzz on the minimum-phase subwoofer the most obvious.

Overall, I was left with an impression, that for this unusual class of excitation signals, the minimum-phase version had "more of this and more of that" type of sound, like more buzz or more thump, or sharper pops. Since electrical tests show clearly, that linear phase version adhered far better to the original excitation signals, the only conclusion I can draw from this, is that the "acoustical extras" I was hearing, were simply phase-induced distortions added by the minimum-phase mode of operation.

Having gone through this whole, rather involving process, I have set UE3 to optimally equalize both of my 18" McCauley subwoofers in linear-phase mode, and achieved the 15Hz-120Hz bandwidth with 2nd order Linkwitz low-pass filter, and the required SPL levels with 2x400W amplifier. This would be my recommended option for 5.2 HT system.

Software & Hardware used

1. Excitation signal generator - SoundEasy V18, running on WinXP, with modifications to CRO section to include pulse generation and bi-polar pulse generation of adjustable width.
2. Equalization – Ultimate Equalizer V3, running on Windows7/64, used in linear-phase and minimum-phase modes.
3. Power amplifier - LM3876, a simple 50Watt integrated design from National Semiconductor, originally had 3dB cut off at 16Hz. Amplifier was modified to lower the 3dB down to 2Hz.
4. Microphone pre-amplifier – A commercial design based on low-noise, LM833 chip. Modifications done to equalize microphone's low-end roll off.
5. Microphone – CLIO Mic01. 8.2V DC bias provided by the pre-amplifier.
6. Listening room has the following dimensions: Length = 6.5meters, width = 4.5meters and Hight = 2.6meters.