Ultimate Equalizer
DSP Loudspeaker Management System

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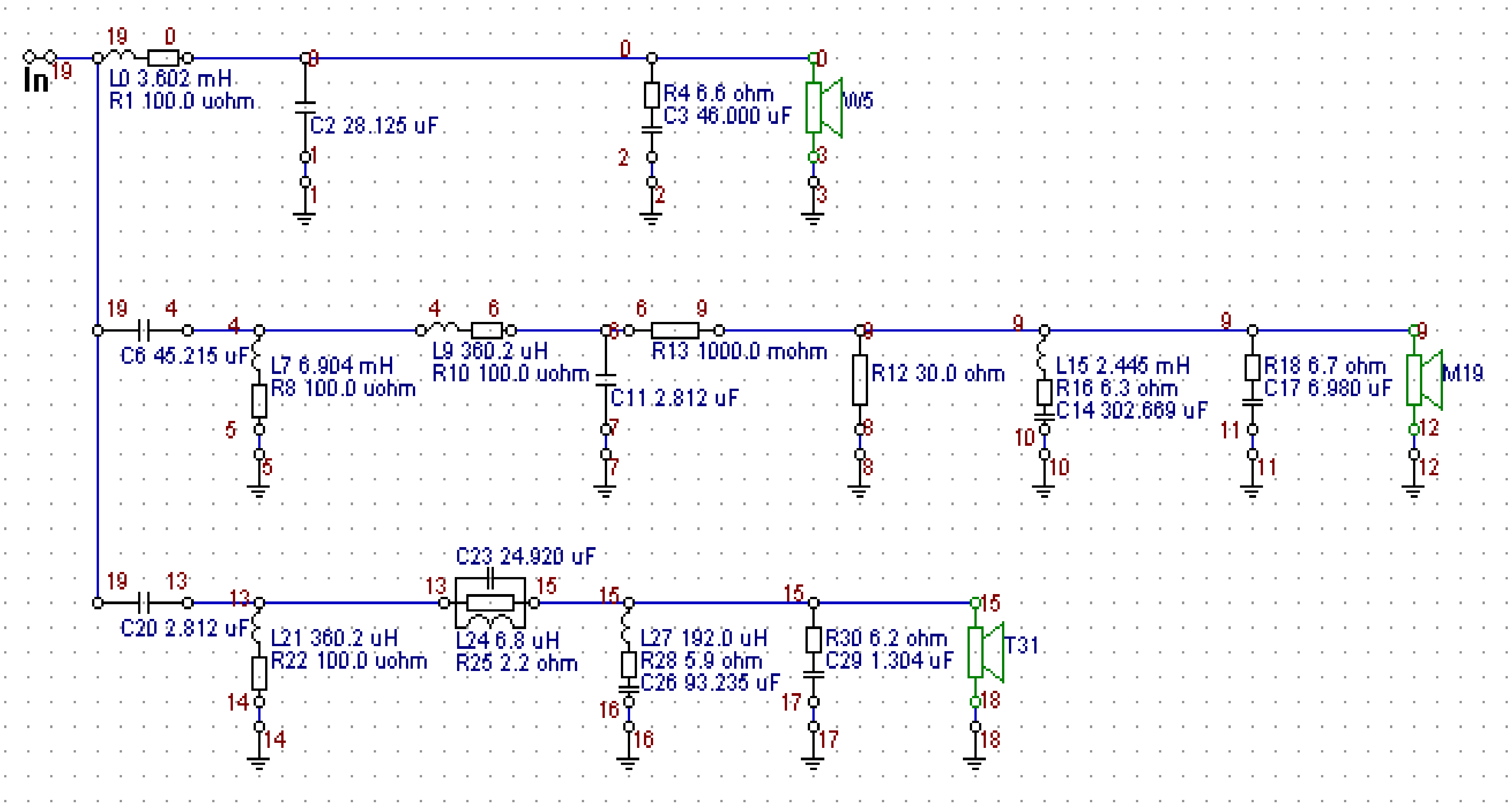
Contents at a Glance

Typical contemporary crossover with corrective circuits

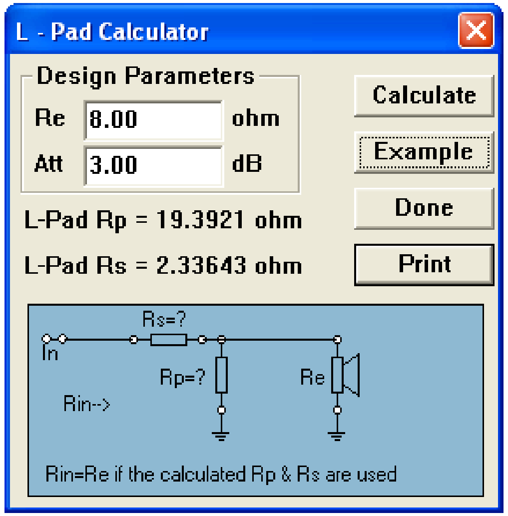
3-way, 12dB/oct + Zobel, L-pad, SPL Notch, Zin Notch



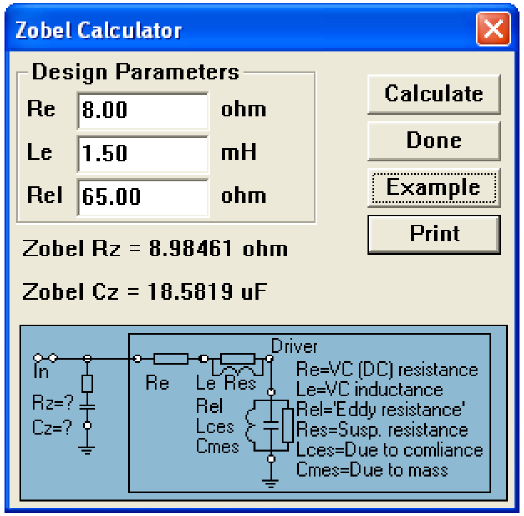
- Good quality inductors (low Rloss)
- High-power, low inductance resistors
- High-voltage bi-polar capacitors
- Inductors mutual orientation important (de-coupling)

Typical crossover with corrective circuits

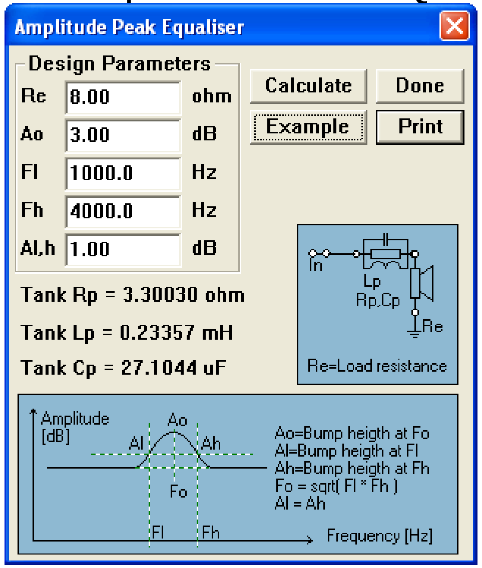
L-Pad



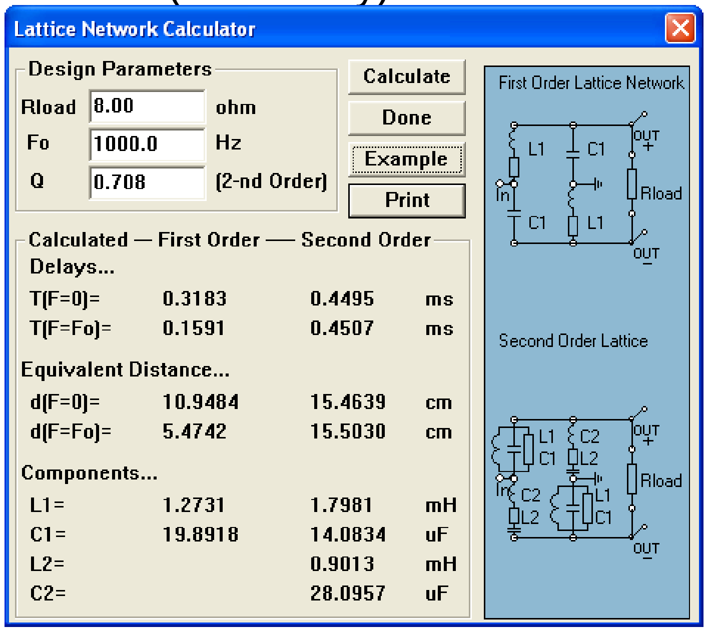
Zobel Network



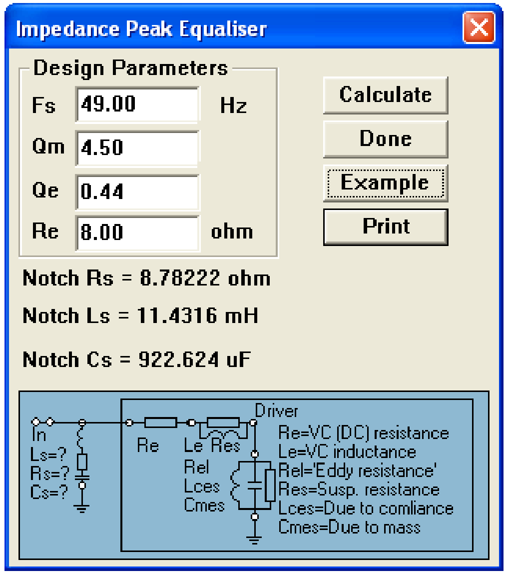
Amplitude Peak EQ



Lattice Network (time delay) needs stable load resistance



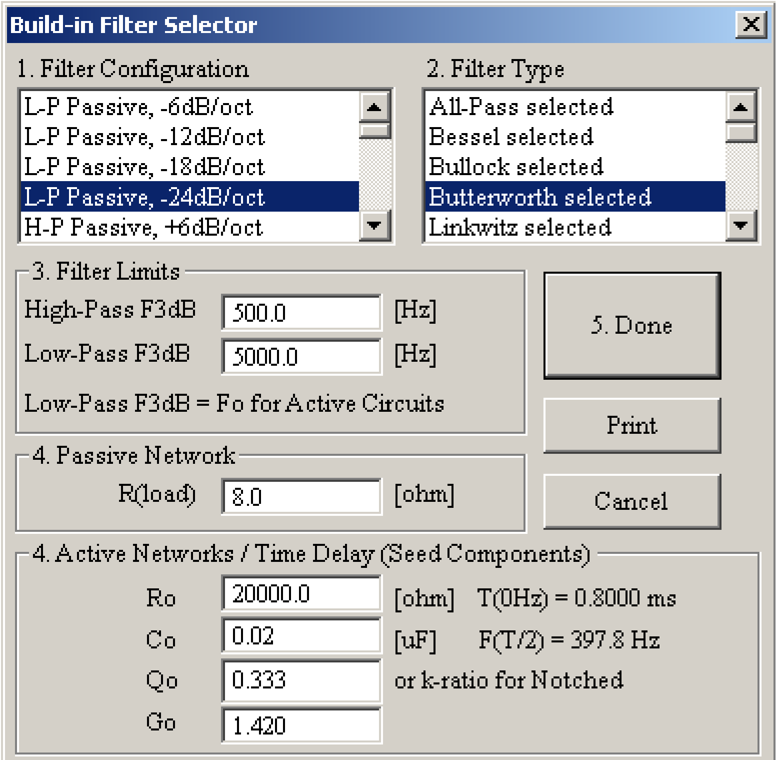
Impedance Peak EQ



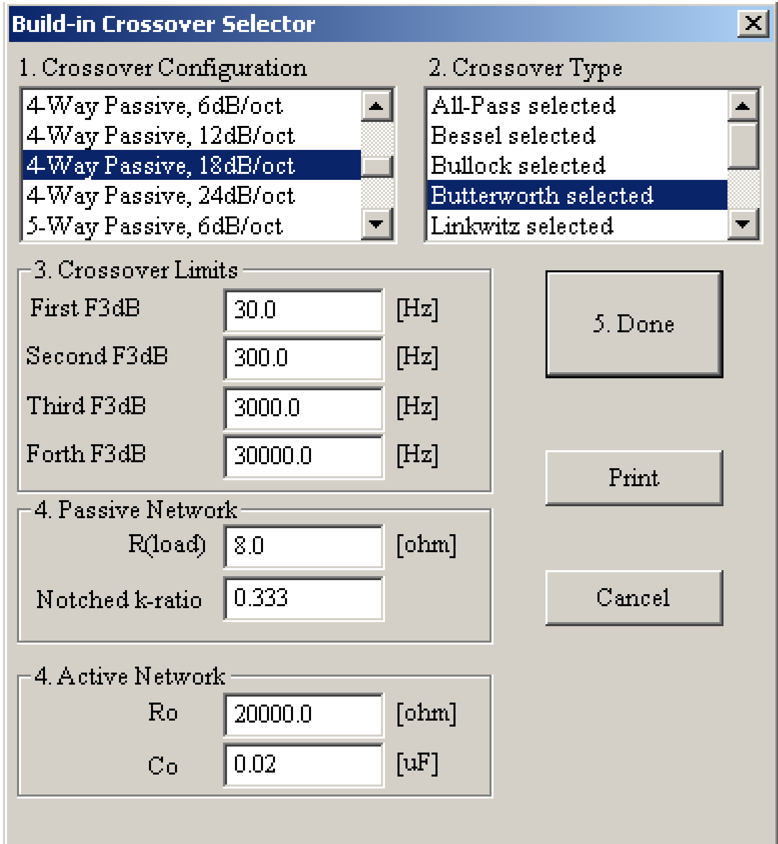
Typical crossover with corrective circuits

(Dedicated CAD for loudspeaker design should have these)

Filter Selector

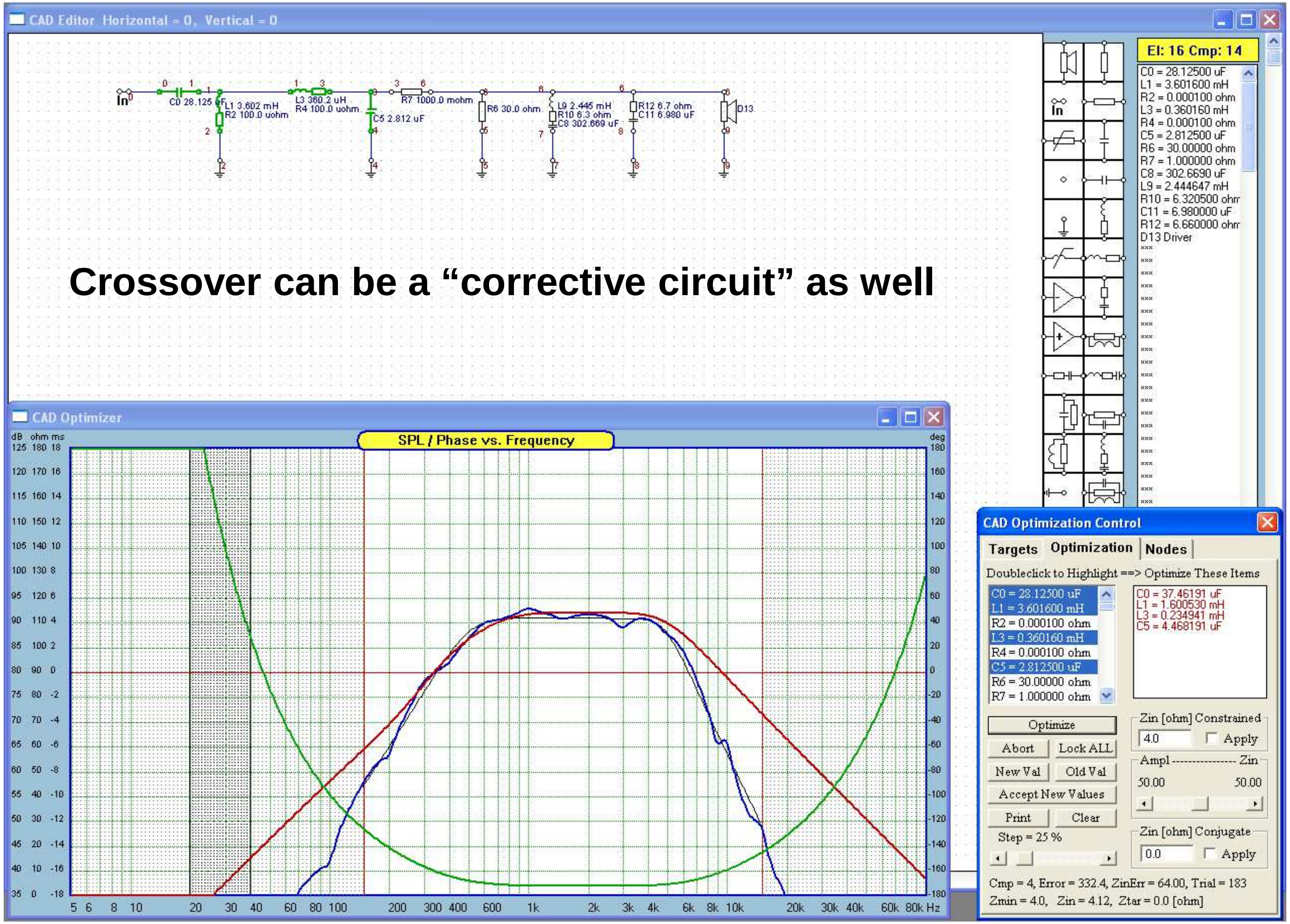


Crossover Selector



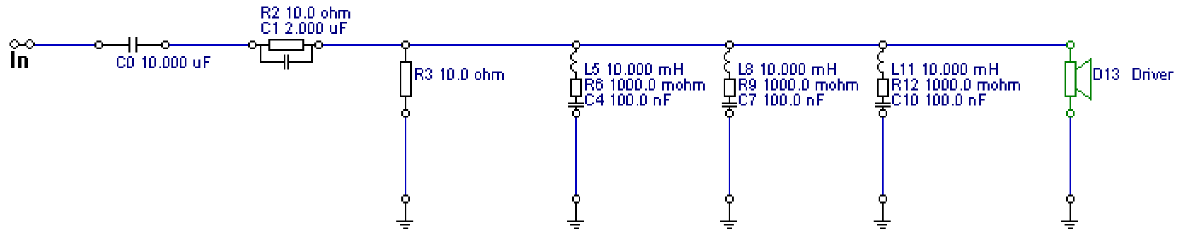
Typical crossover with corrective circuits

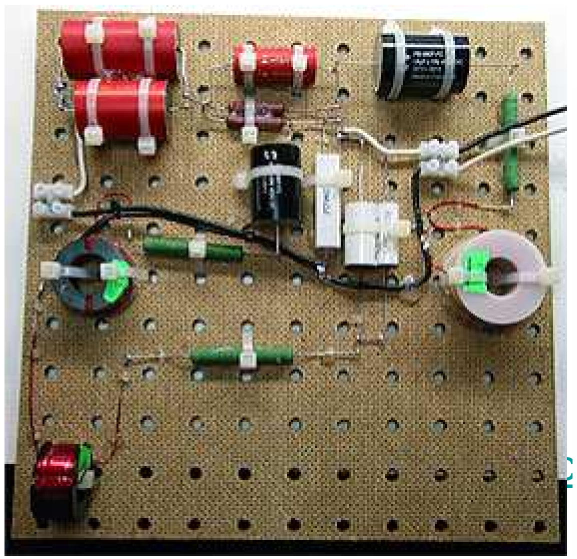
Crossover's frequency response (**green components**) optimization to selected target

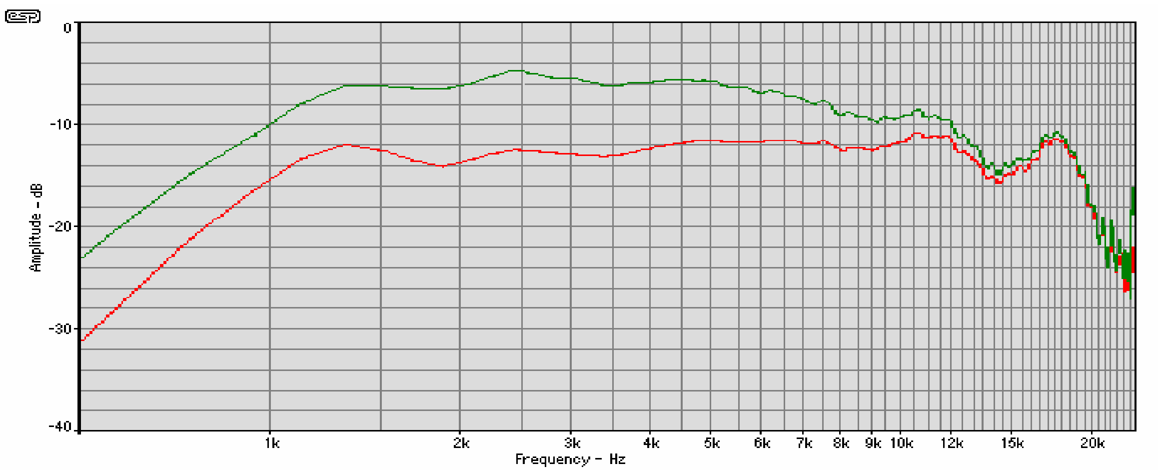


CD waveguide resonance corrections

- Dr. Geddes designs use a OS (Oblate Spheroid) waveguide mathematically designed to produce the fewest HOMs (High Order Modes) possible.
- http://www.enjoythemusic.com/diy/0309/gedlee_abbey.htm
- Type: 2 Way waveguide constant directivity loudspeaker
- Drivers: 12-inch B&C 12TBX100 woofer and B&C DE250-8 Polyimide compression driver
- Crossover: 2nd order passive, at approximately 1200Hz. Multiple LCR networks for the tweeter







- http://sound.westhost.com/articles/waveguides1.htm#intro
- Peaks around 11kHz and 16kHz can be reduced by series tank circuits

Problems with passive crossovers/corrective circuits

- Prevent the amplifier from taking full control of the loudspeaker. Crossover DC resistance introduces losses into the circuit and affects driver's Qt.
- Passive crossover requires ideal load resistance to work like an ideal electrical filter – driver impedance is not.
- Impedance measurements and equalization often necessary.
- Corrections to a bump in driver's SPL affect impedance and phase.
- Driver's parameters (heating, BL changes) affect crossover performance. Qes will affect Zin(w). Re depends on temp
- Practically, can only correct broad irregularities.
- Complexity of the passive circuitry – needs CAD to properly analyse.
- Unable to de-couple amplitude from phase.
- Inductors for subwoofers are large, heavy and expensive.
- Can we do better?

Foundations of Amplitude-Phase Relationship

Network Analysis and
Feedback Amplifier Design

By

HENDRIK W. BODE, Ph.D.,
Research Mathematician,
BELL TELEPHONE LABORATORIES, INC.

TENTH PRINTING

Dr Bode's book

CONTENTS

Dr Bode's book

$$B_c = \frac{2\omega_c}{\pi}\int_0^\infty \frac{A - A_c}{\omega^2 - \omega_c^2} d\omega \qquad \text{(14 - 9) page 307}$$

The above equation is suitable for developing a computer algorithm describing phase transfer function derived from amplitude frequency response. By expressing **F(jω)** in terms of amplitude and phase rather than real and imaginary parts $\mathbf{A + jB}$, the phase can actually be derived from amplitude response.

$$\mathbf{F(j\omega) = A(\omega)e^{j\Psi(\omega)}}$$

Where **A** is the amplitude response and **ψ** is the phase response. The above expression can be re-arranged as follows:

$$\mathbf{\ln[F(j\omega)] = \ln[A(\omega)] + j\Psi(\omega)}$$

F(jw) is analytic and has no zero in the right half-plane. Now, Bode's expression can be applied resulting in:

$$\Psi_c = \frac{2\omega_c}{\pi}\int_0^\infty \frac{\ln A(\omega) - \ln A_c(\omega)}{\omega^2 - \omega_c^2} d\omega$$

It's convenient to express amplitude in nepers, where **1 neper = 8.686 dB**.

- Integral is calculated from 0->Infinity.
- We need asymptotic slopes towards zero and towards infinity
- Passive circuits have easily determined asymptotes. Eg; +6dB/oct HP filter.
- Loudspeaker's asymptotic slopes in SPL are more difficult to determine.
- Integral calculated from 2 octaves below and up to 2 octaves above required bandwidth.

Hilbert - Bode Transform (HBT)

(name coined 15 years ago, first implemented in SoundEasy)

- The frequency range of interest is split into three ranges and contribution from each range added during final assembly of the phase response. “LF tail”, “HF tail” and “range of interest”

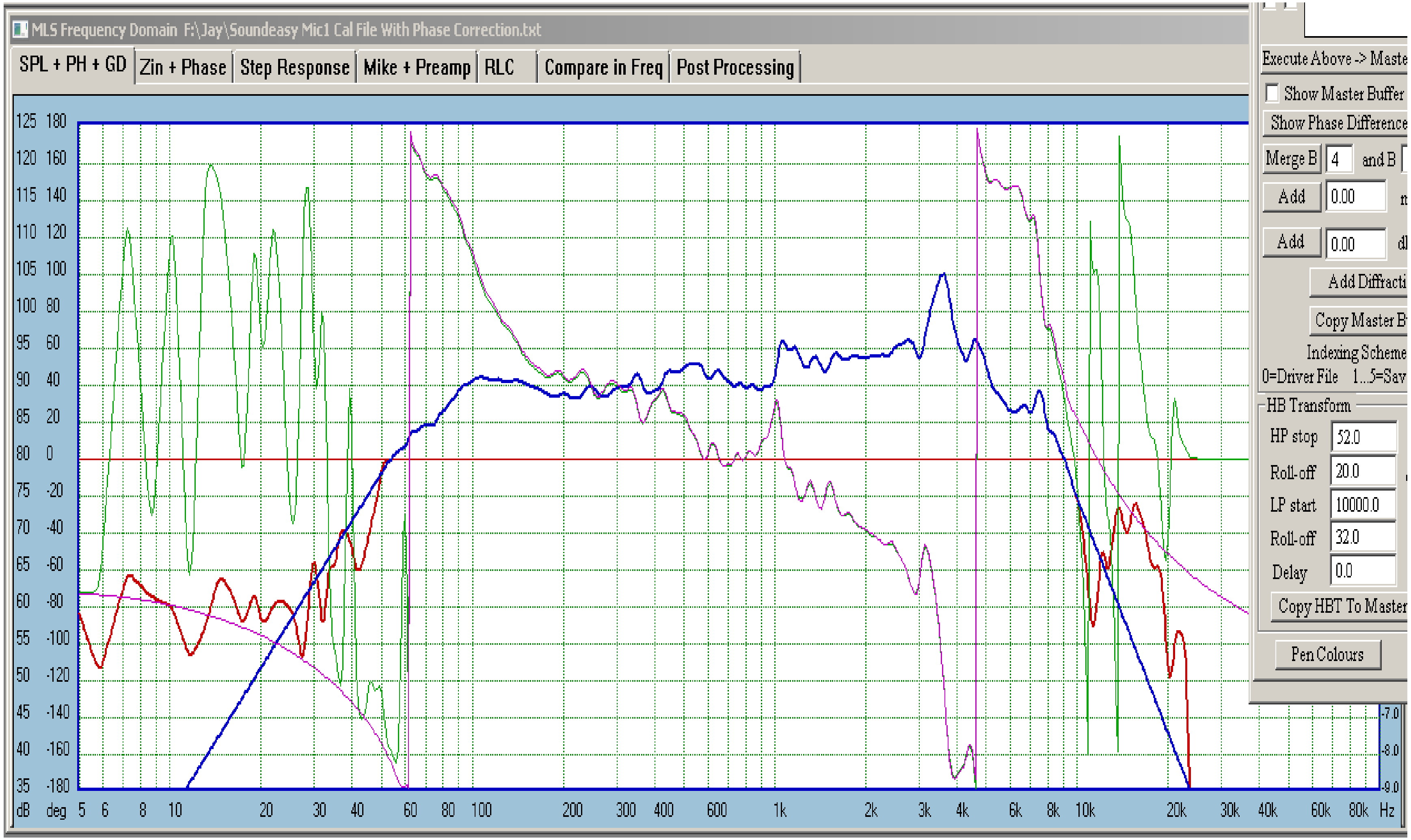
- User of the algorithm can visually inspect the loudspeaker frequency response and **determine the asymptotic roll-off order** on both frequency extremes.

- Frequently, the loudspeaker in question has the roll-off determined by design. For example, the final low-frequency roll-off of a sealed enclosure is -12dB per octave and -24dB per octave for vented enclosure. Another one is QB3 – 18dB/oct.

- In a typical implementation, the transform is driven by 4 editable parameters and they should be selected to obtain the **best match for phase and amplitude between measured signal and calculated transform over the widest frequency range**. Typically, good match can be obtained way beyond driver’s operating frequency range.

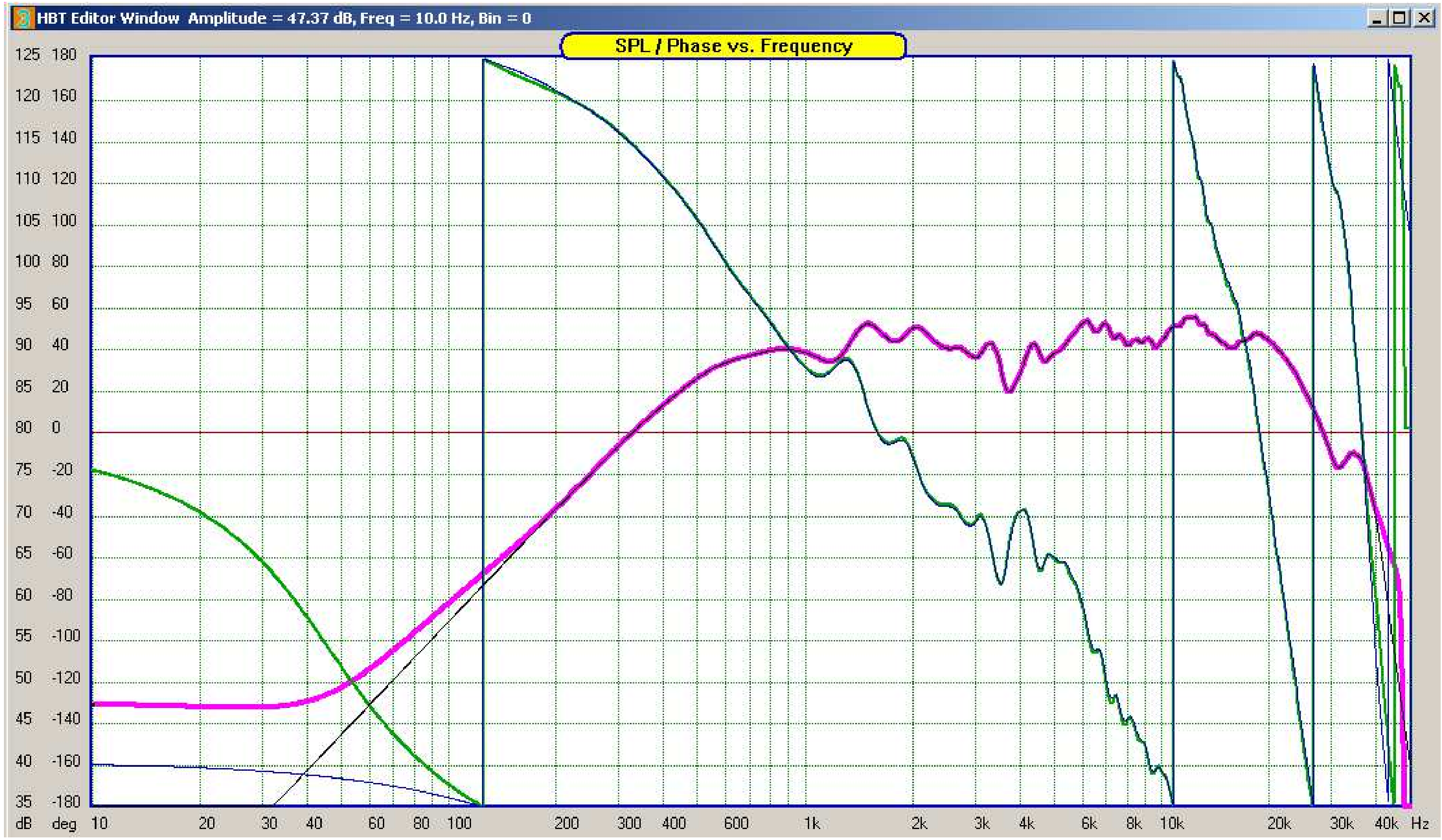
- “A minimum phase system is one which is able to transfer input energy to its output in the least amount of time for a given frequency response”.

Hilbert-Bode Transform: Phase from SPL on 12" guitar speaker
(appr. QB3 vented alignment)



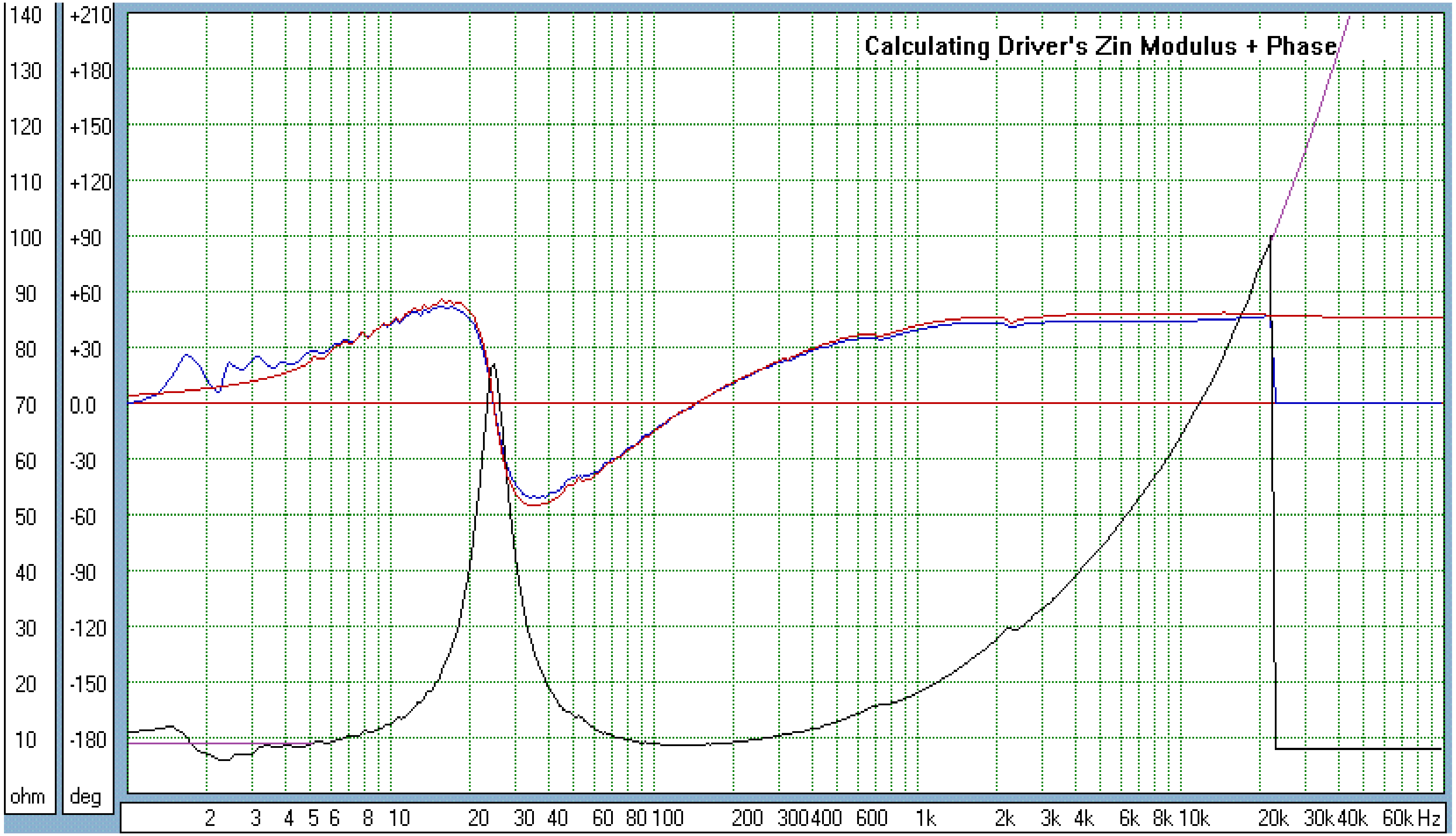
- Measurements conducted in noisy environment – SPL (**red**), Phase (**green**) VERY noisy.
- Noise more persistent in low-frequency range <50Hz.
- Cone break-up visible above 8kHz.

Hilbert-Bode Transform: Phase from SPL on RS28F-4 tweeter



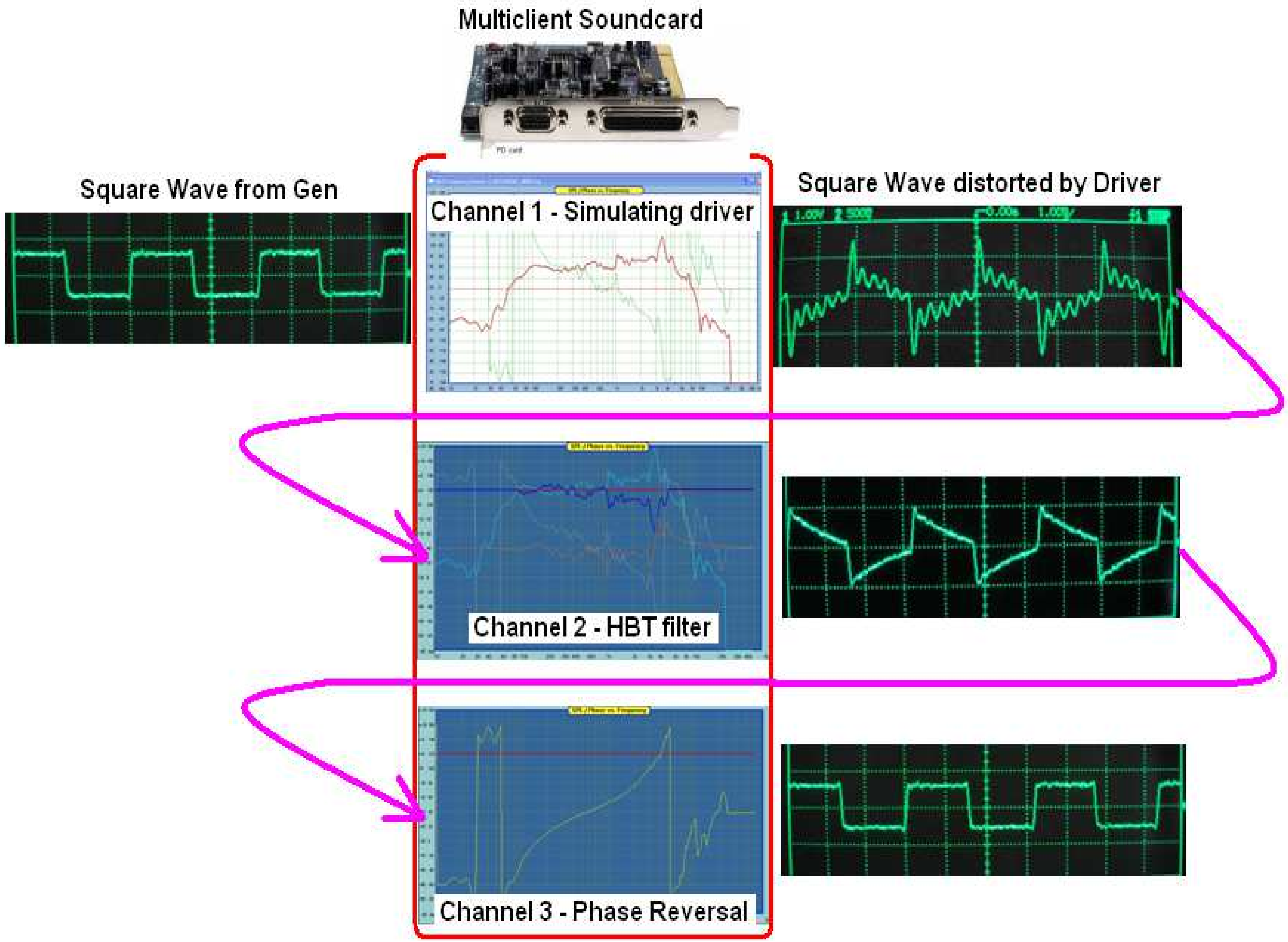
- Measurement is FFT-windowed to avoid room reflections.
- Low-frequency roll-off is -12dB/oct (sealed box).

Hilbert-Bode Transform: Phase from Zin



- MLS sampling frequency = 48kHz, so measurement data valid to ~23kHz.
- Sound card flat from ~22Hz up, so low-frequency noise evident below 5Hz.
- Measured Zin modulus = **black curve**, Phase = **blue curve**.
- Zin extended for HBT = **pink curve**, HBT calculated phase = **red curve**

Concept of the EQ Process

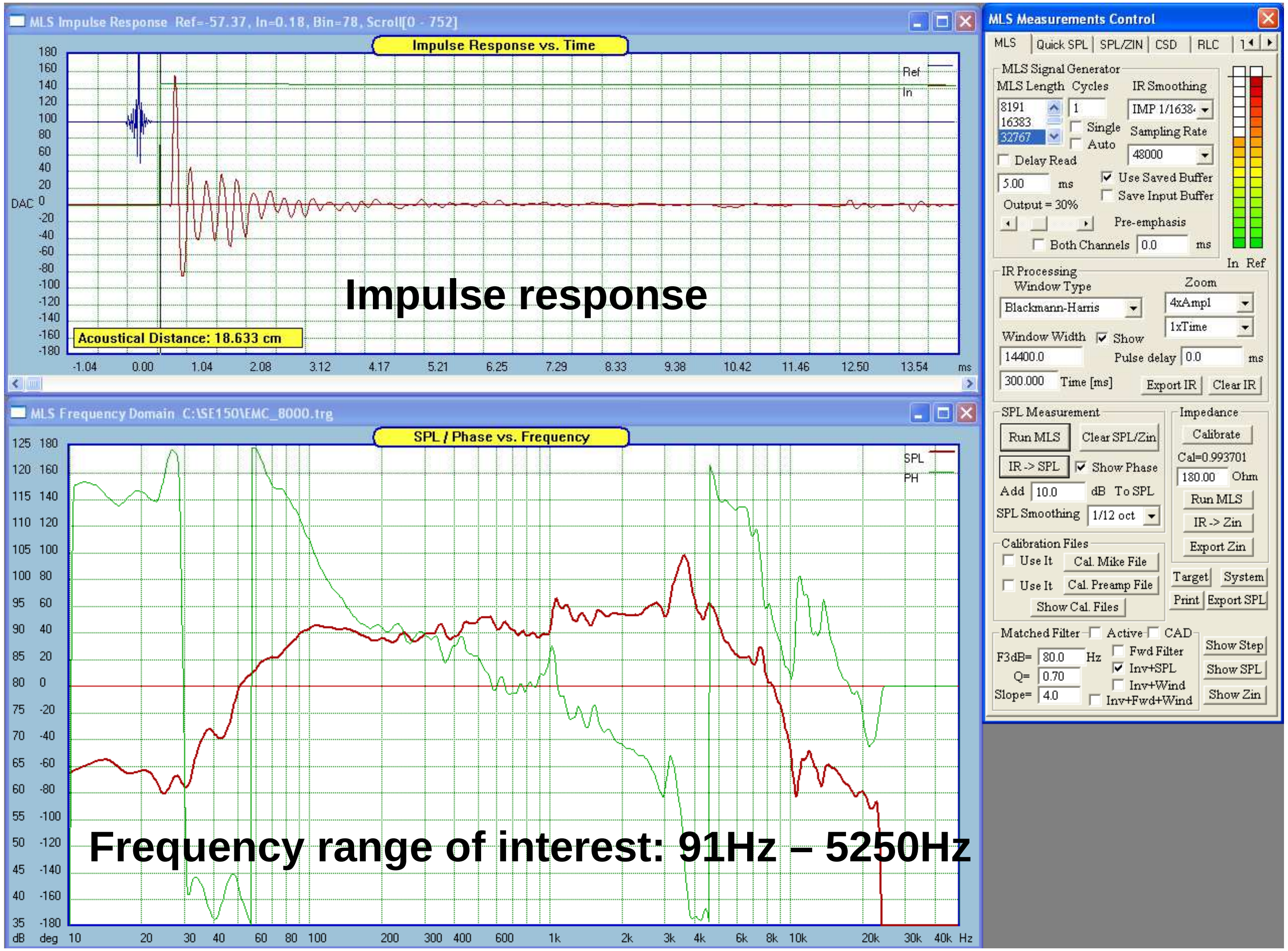


The Test Signal

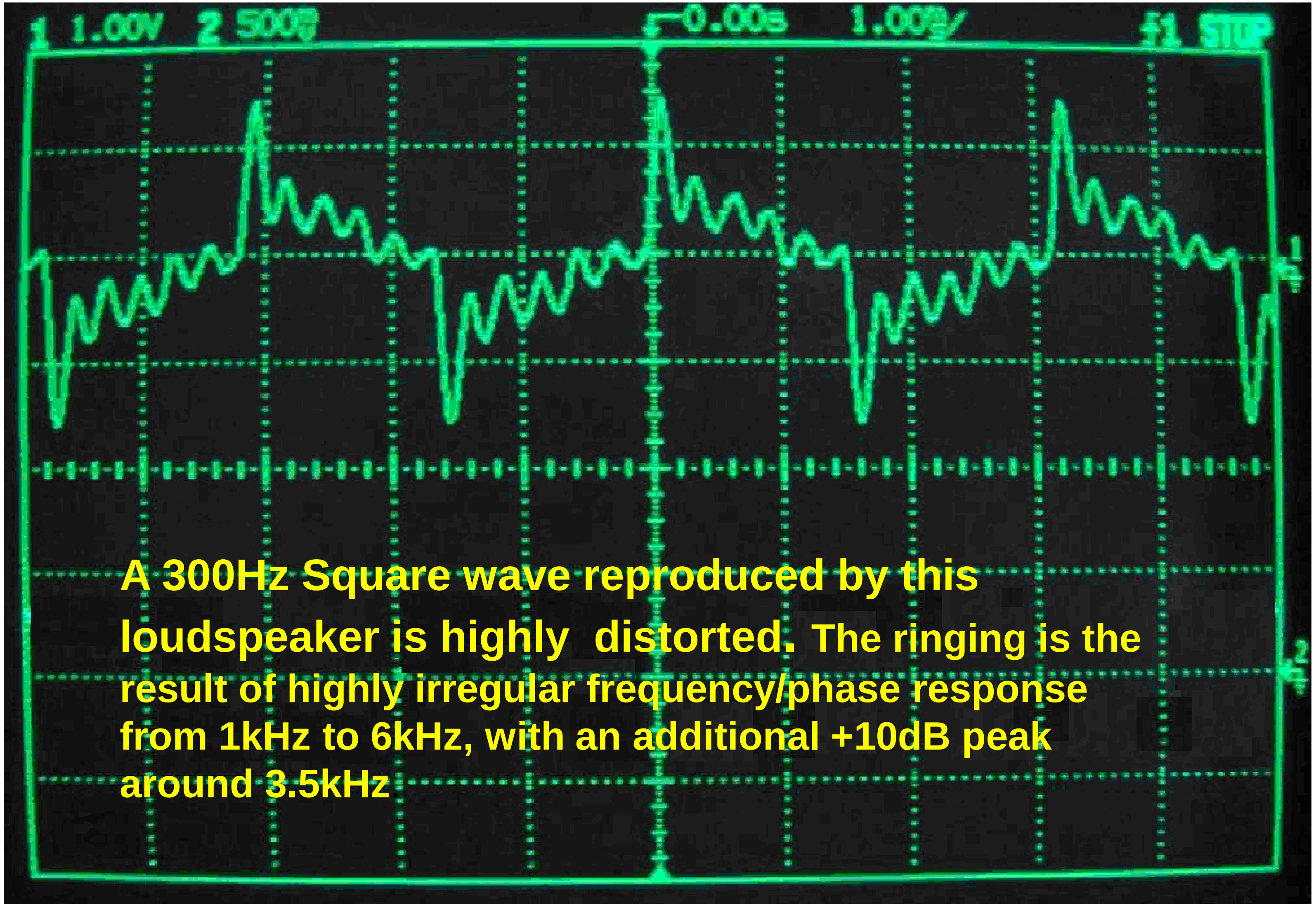
- One of the most useful test signals in electronics is a humble square wave.
- The “ideal” square wave is a superposition of an infinite number of sine waves, each contributing it’s required amplitude and phase.
- It is due to this very feature, that when passed through an audio system, the square wave can reveal time domain performance issues of the system.
- This is because all of it’s 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.

Real-life loudspeaker example

Measured system's magnitude (red) and phase (green).



Time-domain response to 300Hz square wave

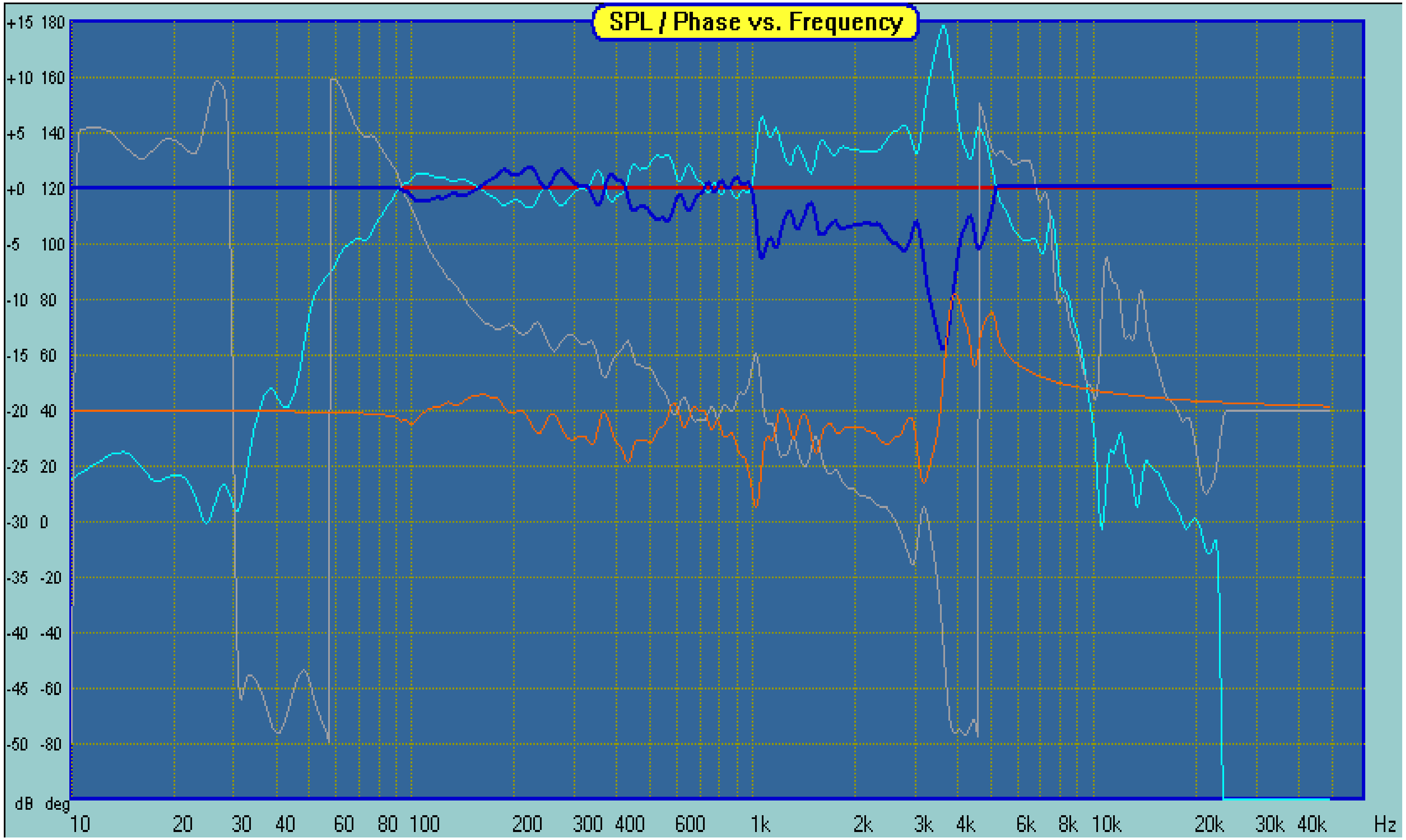


Amplitude Equalizer design

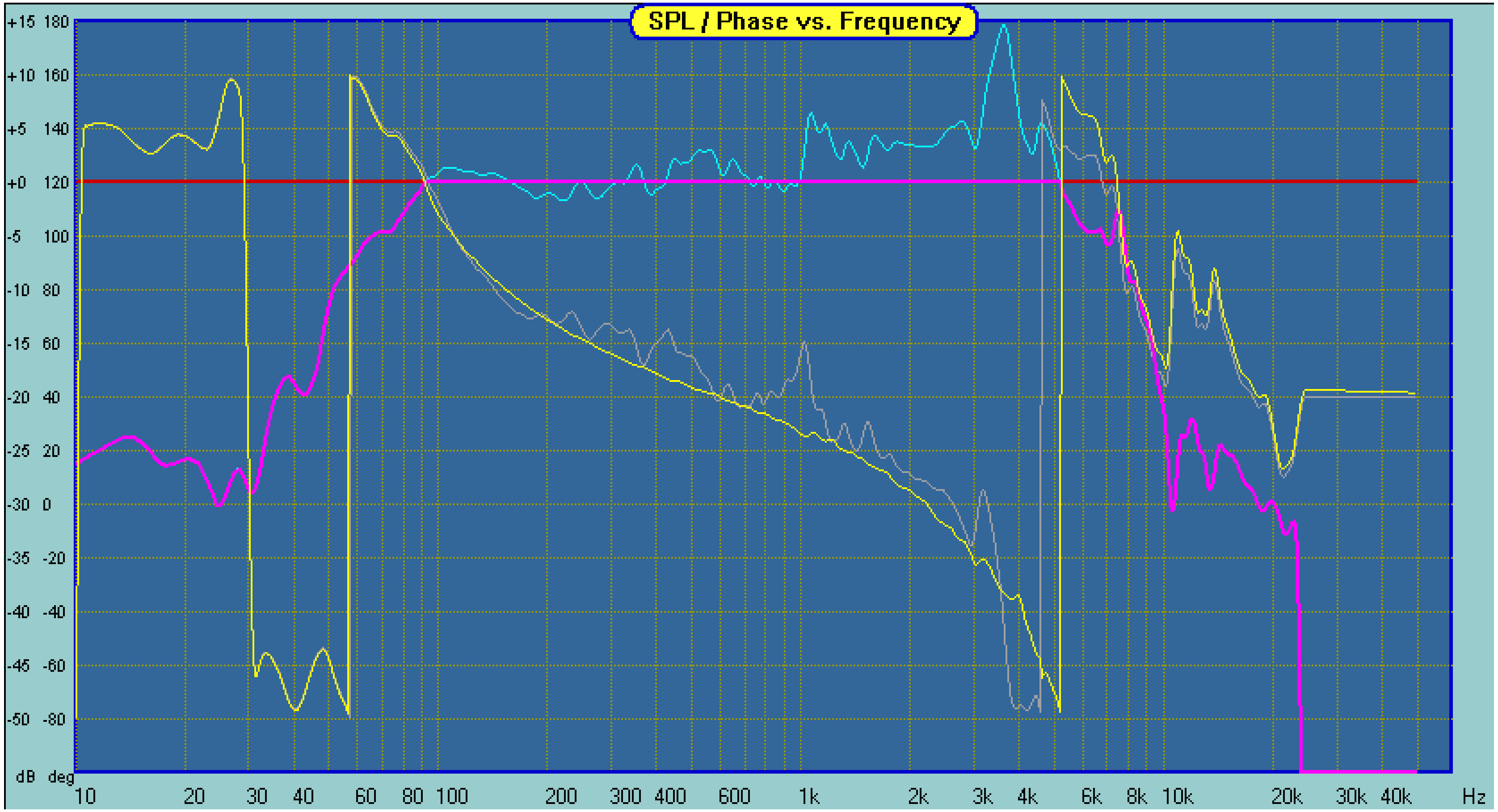
- An advanced tool used for linearizing a transfer function of an LTI (Linear Time-Invariant) system is an Inverted Hilbert-Bode Transform (HBT) technique.
- Just like Fourier Transform allows you to flip between time domain and frequency domains, the HBT allows you to move from magnitude response to phase response and vice-versa.
- I can therefore nominate a frequency range of interest within the loudspeaker's magnitude response, then **attach flat "tails" on the low and high-side of this frequency range** and apply this artificially created magnitude response to the HBT.
- As a result, I will get corresponding phase response, which in turn means, that I actually have **full complex transfer function** calculated via HBT.

HBT Equalizer design

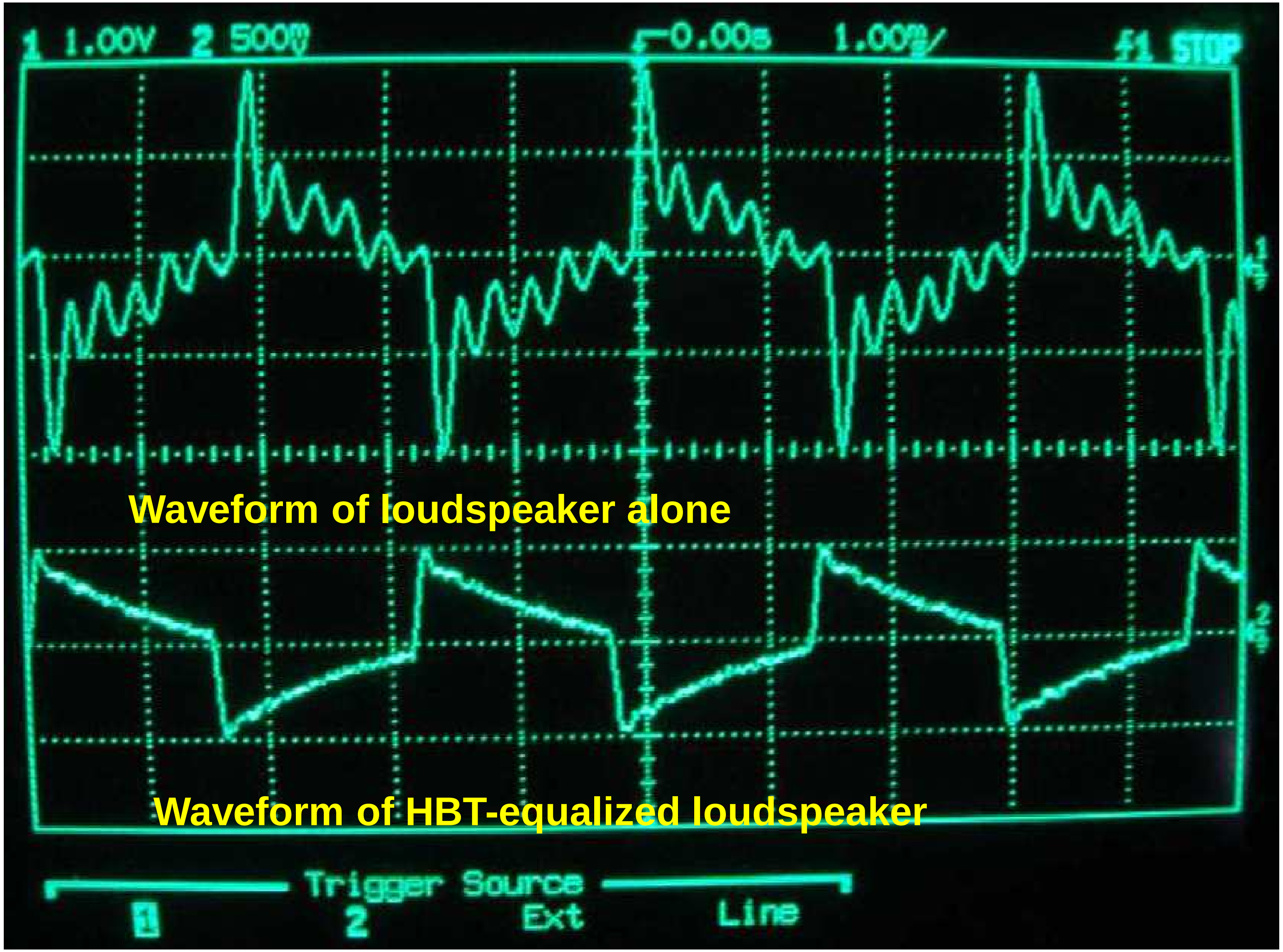
SPL of the Amplitude Error Function (thick blue line) - notice, it's inverted already
Phase of the Amplitude Error Function (orange line)
Please note mathematically correct phase response and it's transitions from irregular-to-flat sections. This is the HBT in-action.



Loudspeaker HBT-linearized: magnitude (pink), phase (yellow)
(Loudspeaker remains minimum-phase)



Square wave passed through HBT- equalizing system



Inverting System Phase

SMITH, S. W. (2003) Digital Signal Processing - A Practical Guide for Engineers and Scientists - Page 194.

represented by placing a star to the upper-right of the variable. For example, if $X[f]$ consists of $Mag\,X[f]$ and $Phase\,X[f]$, then $X^*[f]$ is called the *complex conjugate* and is composed of $Mag\,X[f]$ and $-Phase\,X[f]$. In rectangular notation, the complex conjugate is found by leaving the real part alone, and changing the sign of the imaginary part. In mathematical terms, if $X[f]$ is composed of $Re\,X[f]$ and $Im\,X[f]$, then $X^*[f]$ is made up of $Re\,X[f]$ and $-Im\,X[f]$.

Here are several examples of how the complex conjugate is used in DSP. If $x[n]$ has a Fourier transform of $X[f]$, then $x[-n]$ has a Fourier transform of $X^*[f]$. In words, flipping the time domain left-for-right corresponds to changing the sign of the phase. As another example, recall from Chapter 7 that correlation can be performed as a convolution. This is done by flipping one of the signals left-for-right. In mathematical form, $a[n] * b[n]$ is convolution, while $a[n] * b[-n]$ is correlation. In the frequency domain these operations correspond to $A[f] \times B[f]$ and $A[f] \times B^*[f]$, respectively. As the last example, consider an arbitrary signal, $x[n]$, and its frequency spectrum, $X[f]$. The frequency spectrum can be changed to *zero phase* by multiplying it by its complex conjugate, that is, $X[f] \times X^*[f]$. In words, whatever phase $X[f]$ happens to have will be canceled by adding its opposite (remember, when frequency spectra are multiplied, their phases are added). In the time domain, this means that $x[n] * x[-n]$ (a signal convolved with a left-right flipped version of itself) will have left-right symmetry around sample zero, regardless of what $x[n]$ is.

Inverting System Phase (Caution, use FIR!)

Original System

$$(a+jb)$$
$$\mathrm{Re}=a$$
$$\mathrm{Im}=b$$
$$Magnitude=\sqrt{\mathrm{Re}^2+\mathrm{Im}^2}=\sqrt{a^2+b^2}$$
$$Phase=-a\tan\frac{\mathrm{Im}}{\mathrm{Re}}$$

Equalized System Frequency Response

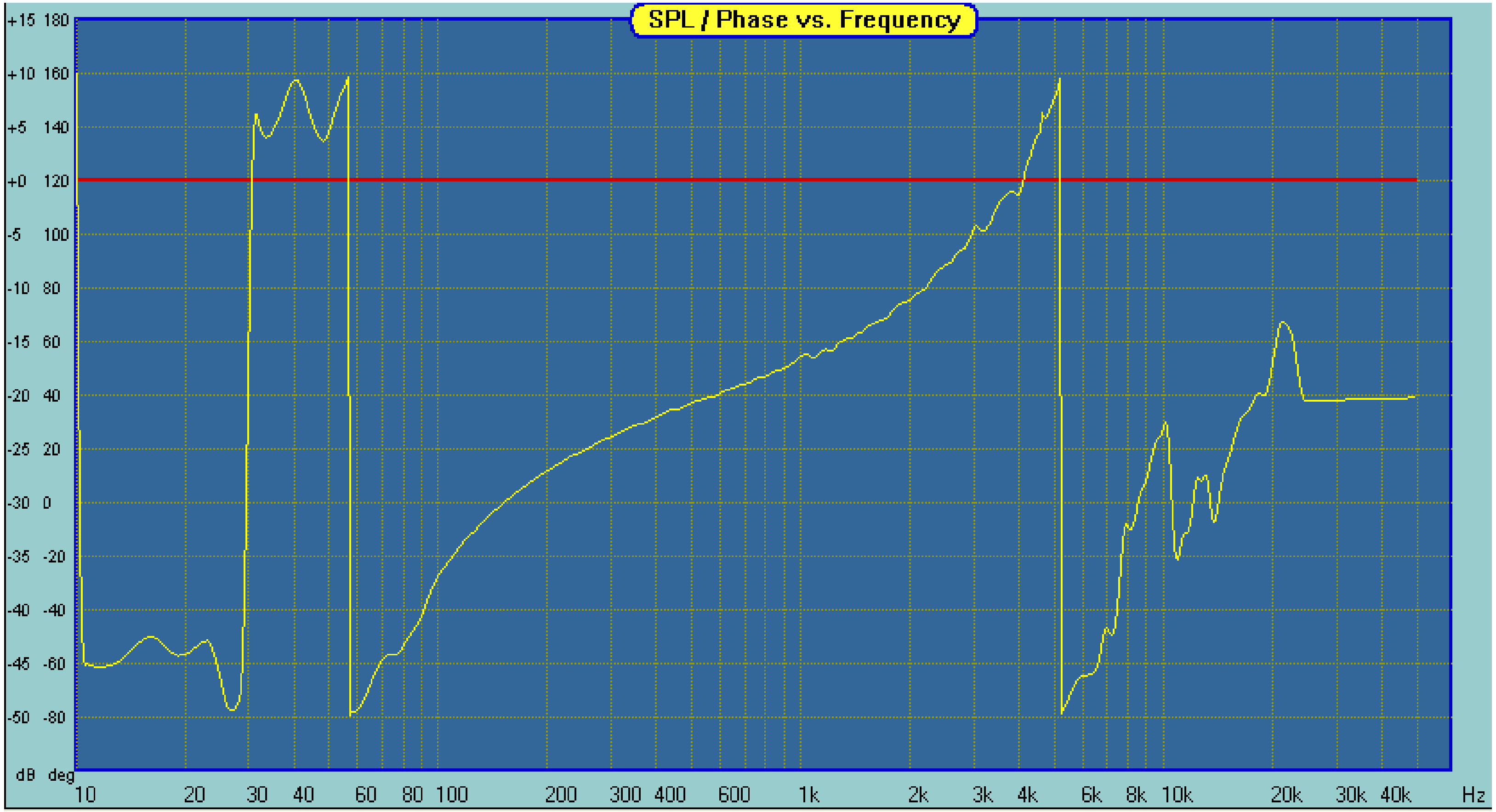
$$(a+jb)*(a-jb)=a^2-jab+jba+b^2=(a^2+b^2)+j0$$
$$\mathrm{Re}=(a^2+b^2)$$
$$\mathrm{Im}=j0=0$$
$$Magnitude=\sqrt{\mathrm{Re}^2+\mathrm{Im}^2}=\sqrt{(a^2+b^2)^2+0^2}=(a^2+b^2)=\left(\sqrt{(a^2+b^2)}\right)^2$$
$$Phase=0$$

Equalized System Frequency Response with Magnitude Forced to 1.

$$(a+jb)*(a-jb)=Me^{j\theta}*M_1e^{-j\theta}\quad if\quad M_1=1\quad then$$
$$Me^{j\theta}*M_1e^{-j\theta}=Me^{j(\theta-\theta)}=Me^{j0}=M$$
$$Magnitude=M$$
$$Phase=0$$

System Inverse Phase Function: magnitude (red), phase (yellow)

FIR filter can do this



Inverting System Phase

- We have now created a perfect phase-reversal device with flat amplitude response - System Inverse Phase Function – see figure above.

- Flat amplitude response requirement is important here, because at this stage, we do not want any more amplitude corrections. We have done this already in the previous stage, using our HBT-based, Amplitude Error Function.

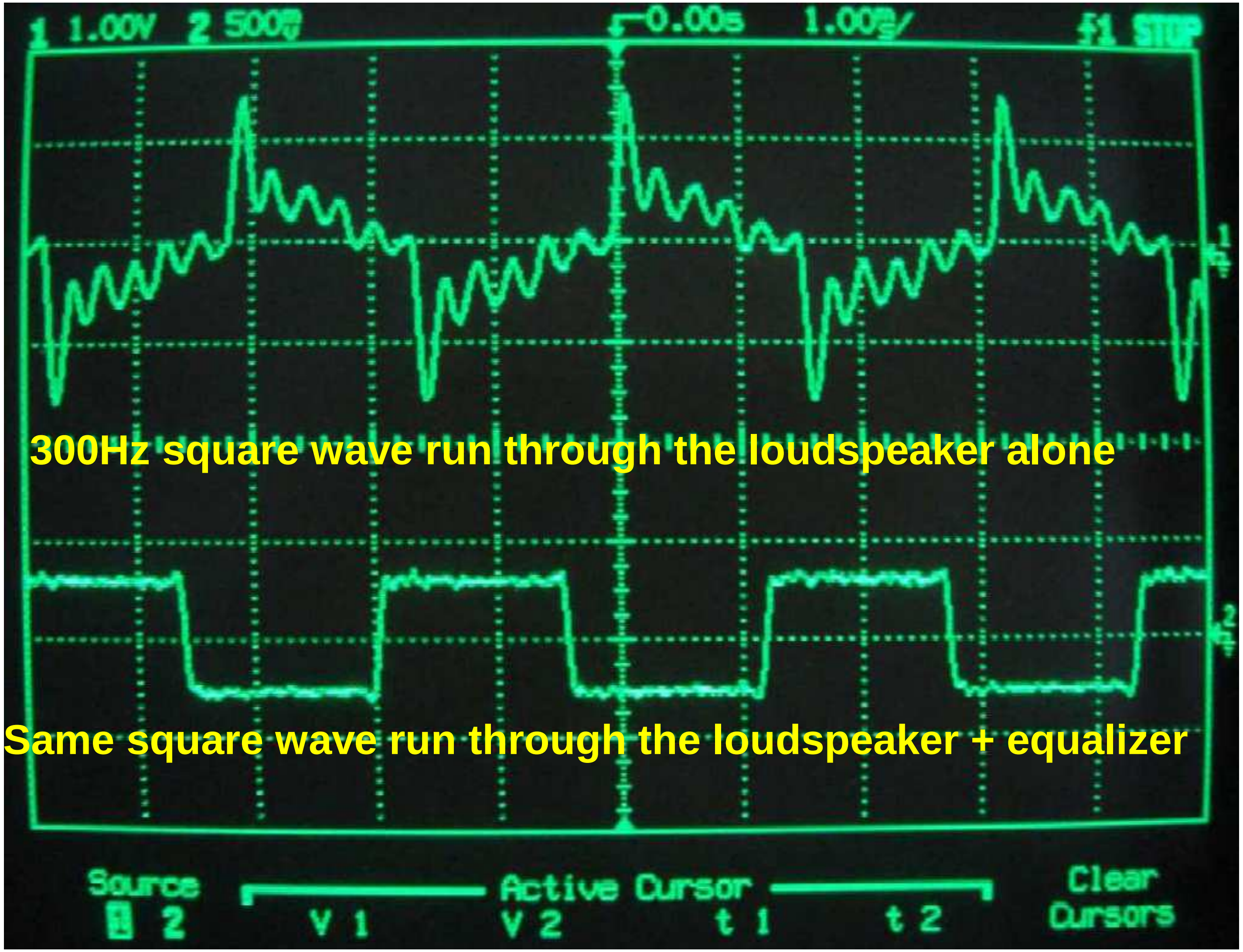
// Enter loop with HBT-equalized filter: Filt[i] = Filt[i].real + j * Filt[i].imag

```
•   for( i = 0; i < PARTITION_SIZE_USED; i ++)
•     {  // Calculate conjugate phase filter
•        c = atan2( Filt[i].imag,   Filt[i].real );
•        a = 1 * cos( c );          // real part, magnitude = 1
•        b = 1 * sin( c );          // imaginary part, magnitude = 1

•        // Substitute Filter variables before multiplication
•        A = Filt[i].real ;           // real part
•        B = Filt[i].imag ;          // imaginary part
•
•       // Perform multiplication with conjugate: (A + jB)*(a - jb)
•        Filt[i].real   = a*A + b*B;         // real part, phase-linear
•        Filt[i].imag = a*B - b*A;         // imaginary part, phase-linear
•     }
```

Conclusions – “two-step” equalization

Step 1: Amplitude Error Function (Inverse HBT)
Step 2: System Inverse Phase Function (Conjugate with magnitude=1)



Conclusions

- Implemented **two-stage equalization technique** removes driver-induced time and frequency domain distortions.

- The resulting outgoing square wave is almost perfectly recombined from individual sine waves constituting the input square wave.

- HBT-based, **Amplitude/Phase Error Function** can be equally applied to smooth the magnitude and phase response of non-minimum phase systems, such as multi-way loudspeaker system, complete with crossover.

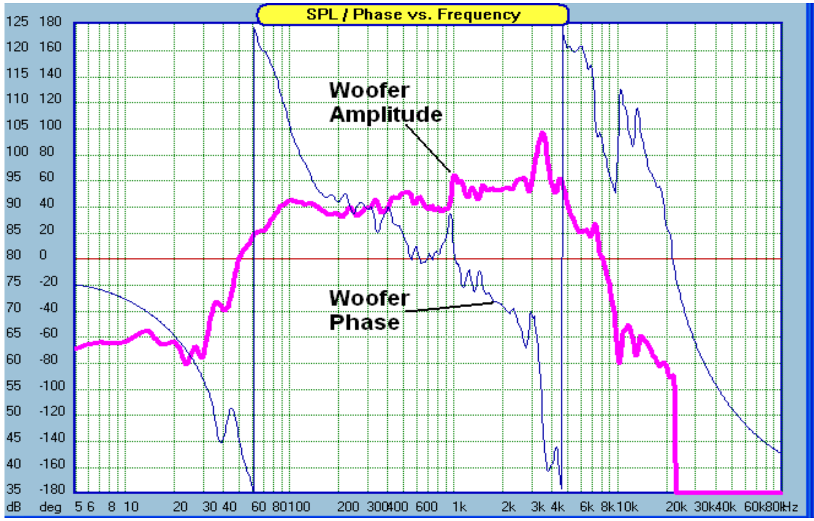
- Minimum-phase system (driver) remains minimum-phase and non-minimum-phase system (loudspeaker system) remains non-minimum phase.

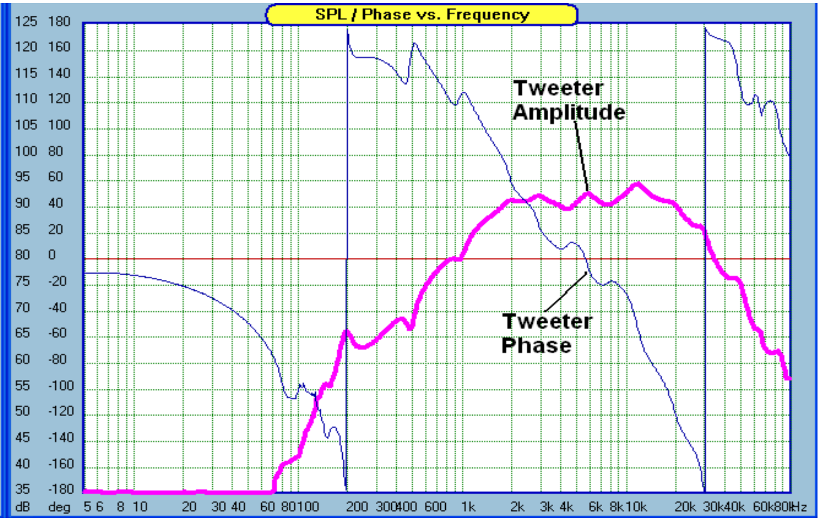
- Also, the **System Inverse Phase Function** inverts the phase of the complete system, as it was measured, and regardless of the trajectory of the phase response. Consequently, the whole two-stage equalization technique is fully applicable to multi-way loudspeaker systems.

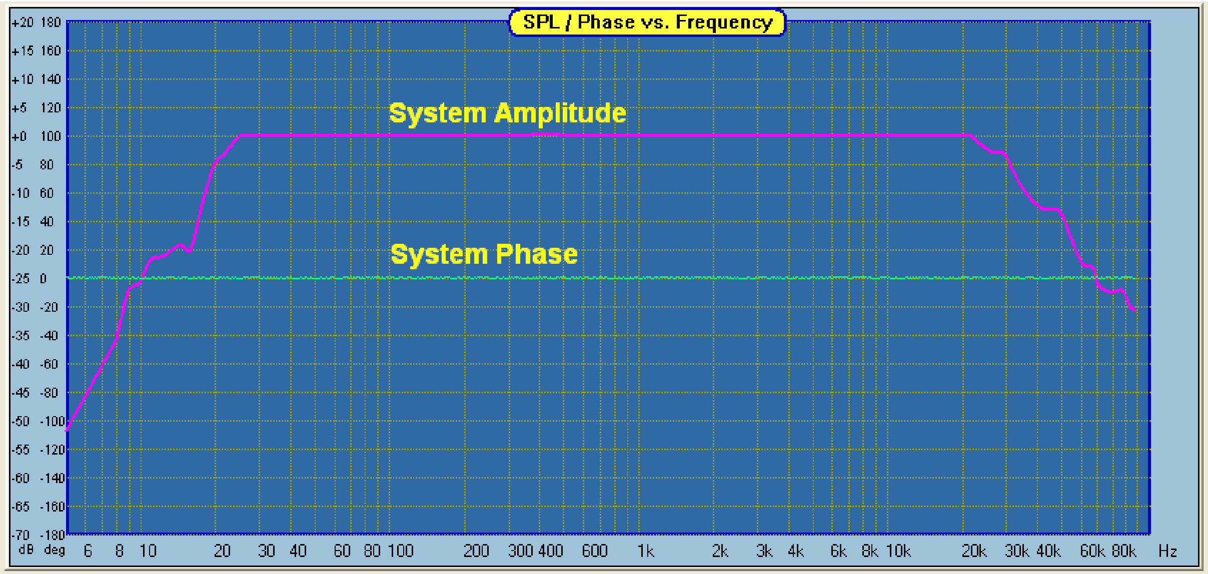
- Using two-stage approach allows us to **trade phase linearity for latency**. Max tolerable latency for AV lip-synch is ~180ms.

Conclusions

Example of a fully equalized SPL and phase of a 2-way loudspeaker







Loudspeaker Equalization Strategies

- Do not equalize frequency response at all – just use UE as an active crossover, and get full benefits of an active system.
- (1) and add alignment of acoustic centres by introducing correct delays to midrange and tweeter.
- Use built-in peak / notch / shelving filters to provide broad equalization. Up to 32 CAD elements can be used in each loudspeaker system.
- HBT Equalize at single point on the design axis, say 1meter or 2 meters. This will ensure ideal equalization at this point and very good EQ along the design axis.
- Perform multiple measurements at +/-15deg horizontal, and use the average to equalize. Horizontally symmetrical loudspeaker recommended.
- Perform minimum-phase equalization or linear-phase equalization. Linear-phase results in much larger latency.
- Use BBM (Binaural Bass Management - AES Preprint 6628) for enhanced bass management.

Room Equalization

(Complex problem with many issues involved)

- High performance loudspeakers = first-arrival flat within 100Hz - 10kHz in room. Need “room friendly” (constant directivity?) loudspeakers.

- Loudspeaker in room excite room modes - modes influence the character of the sound.

- At mid and high frequencies modal density is high, mods overlap, room response diffuse.

- At low frequencies modal density is low. The room imprints it’s own characteristics on the sound quite profoundly (<200Hz).

- Summation of direct and reflected sound will produce amplitude variations at the listening location that can span 30-40dB in magnitude – non minimum phase.

- The lower frequency of the sound wave, the more minimum-phase characteristics will be exhibited by the room.

Steps Towards Improvement

Follows Ph.D. Floyd E. Toole paper:

http://www.harmanaudio.com/all_about_audio/acoustical_design.pdf

- **First - start with a good room**
- **Secondly – use good speaker** (smooth, extended bass response).
- **Thirdly – employ a DSP to put the "icing on the cake**".

- Desirable to have some form of a "detector", that would indicate frequency range(s) where the room is definitely exhibiting minimum-phase characteristics.

- Application of a minimum-phase DSP process to control room modes, inevitably results in injecting less energy into the room within the correction frequency range. Resolution of 1/3oct is not sufficient. 1/10oct -1/20oct is recommended.

- Low frequency active absorber will reduce the SPL at modal frequencies, but for users favouring modal gain, it may also create a perception of lacking decay at those frequencies. It's like - well, where has the bass gone?.

- Some researchers suggest, that equalization, that results in notches deeper than -6dB should be applied with caution, as this would reduce node's original RT60 by half. Room reverberation below 0.3s results in unusually "dead" acoustics. As always, <u>extensive listening tests are the best criteria</u>

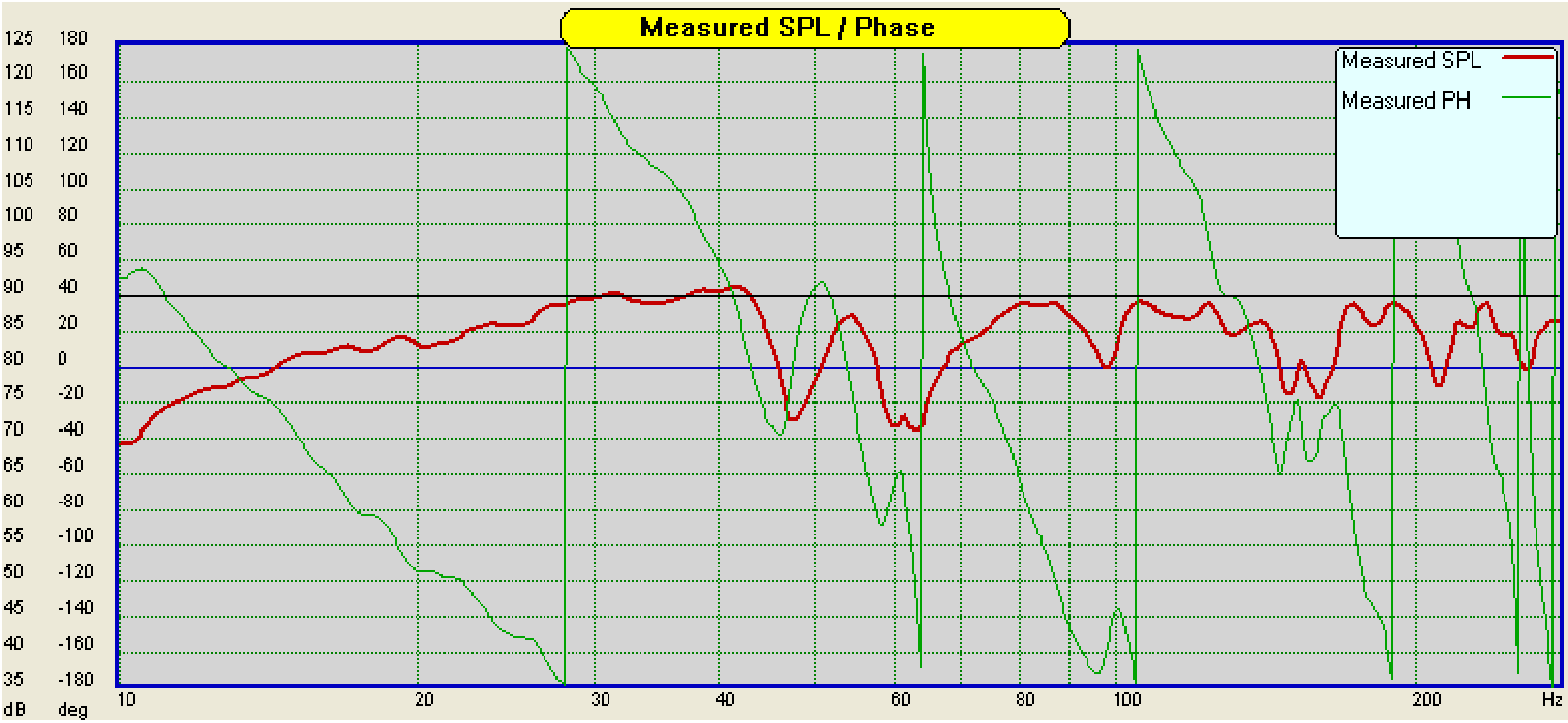
Implementation of the Room EQ

Room equalization strategy

- **What are the frequencies where we can deploy the equalizer**.
- **What are the locations, where the sound is expected to be improved and**
- **How much equalization we should provide.**
- The first step in approaching room equalization process is identification of the minimum-phase regions. A minimum phase system is one which is able to transfer input energy to its output in the least amount of time for a given frequency response. Then <u>system's response can be inverted by minimum-phase EQ</u>.
- If we have a system such as this, then we can create an "Inverse filter", which in combination with the system's transfer function, would produce a flat frequency response and correct the phase response as well. This is quite simplified view, but sufficient for our purpose. So the minimum-phase property of the room would qualify the usage of our room equalizer.

Identifying Minimum-phase regions

- Looking at the measured amplitude and phase responses of the loudspeaker in the room alone, it is not possible to determine the minimum-phase regions.



Identifying Minimum-phase regions

- Examining the **flatness of group delay** would bring us a step closer to determining the minimum-phase regions, but still no good.

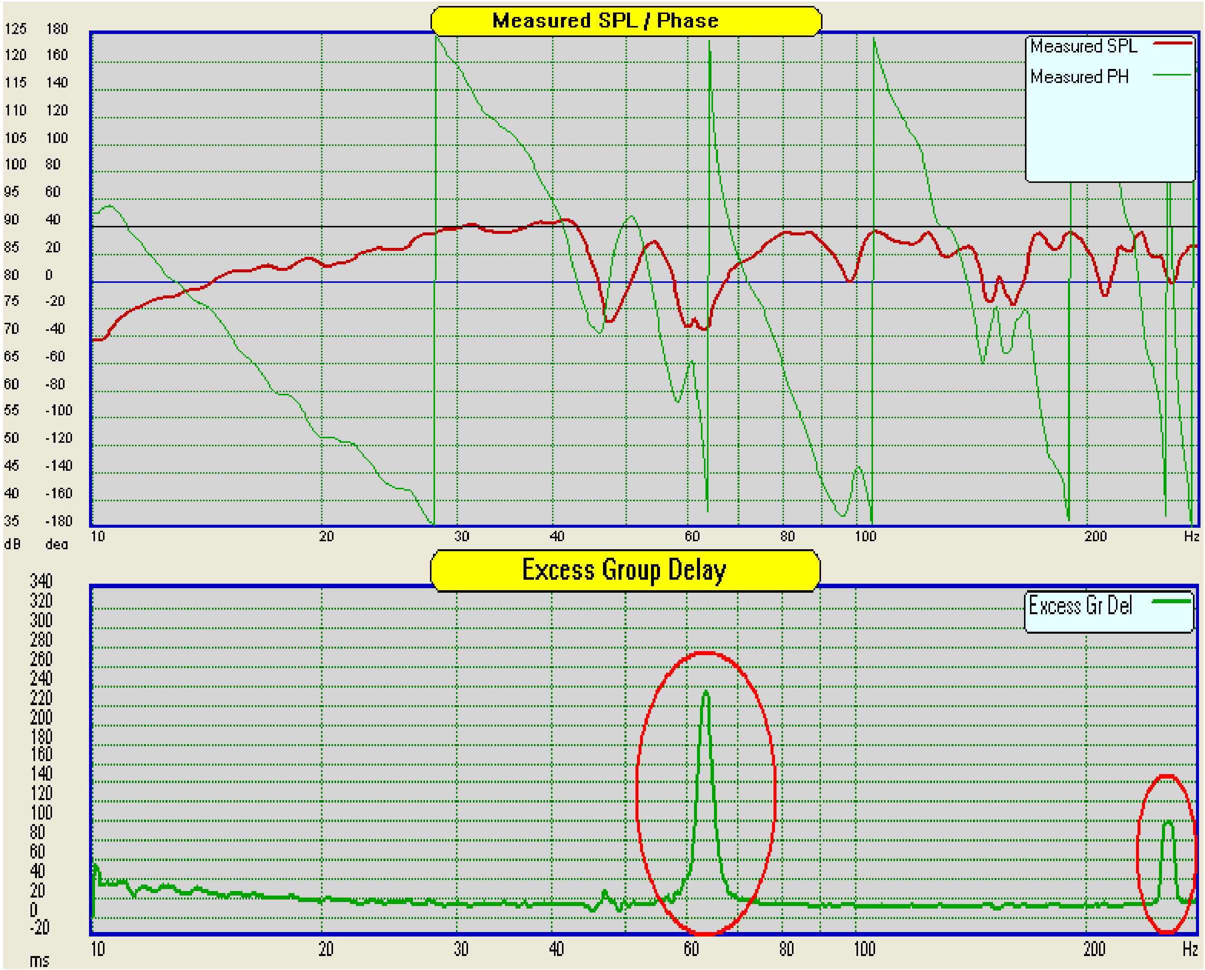
- **Excess group delay**. If we were able to create a system, that has the same frequency response as the measured one, but is definitely a minimum-phase type, we could then create a **differential phase response by subtracting phase response of such system from the measured phase response.**

- Now, if the measured system was a minimum-phase type, then the excess group delay, based on differential phase response would be a flat line.

- Conversely, any deviation of the excess group delay from a flat would indicate, that this frequency range is the non minimum-phase type. **Thus we have just laid down the principles of our "minimum-phase detector".**

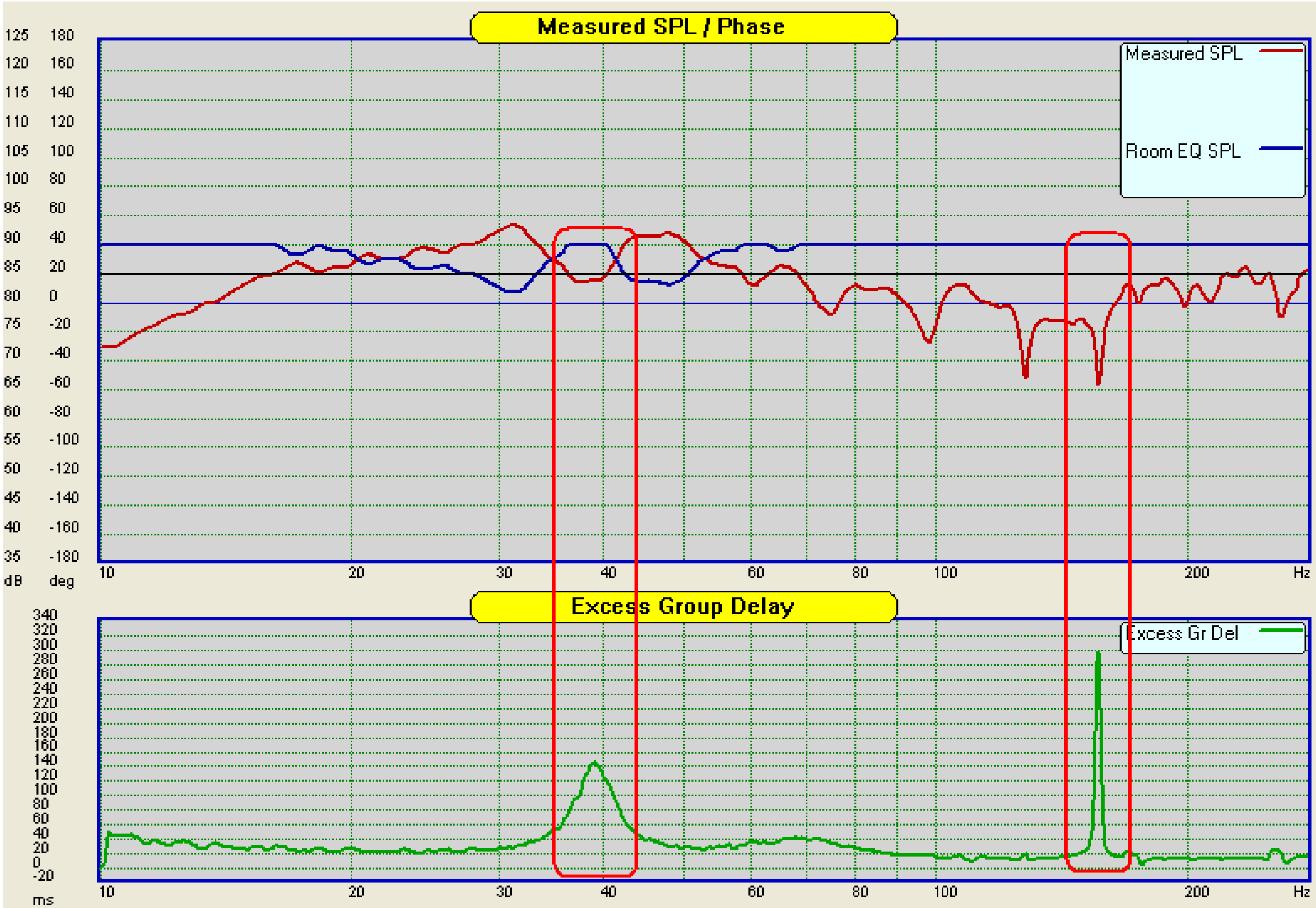
- **The minimum-phase system is created by taking a HBT of the measured frequency response. The HBT will output a minimum-phase phase of the measured system. We will then use this phase response in our calculations of the excess group delay.**

Identifying Minimum-phase regions



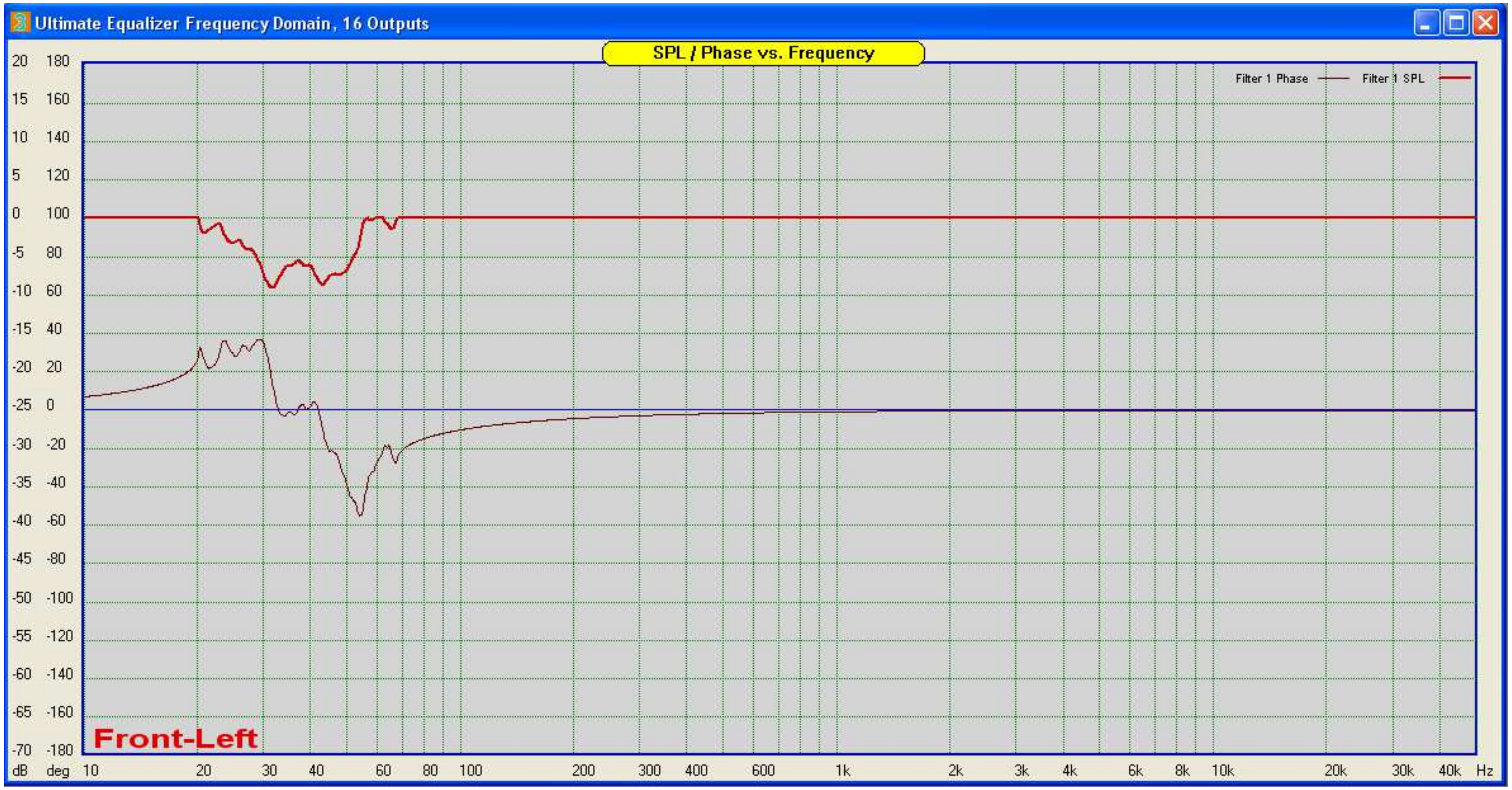
Calculated Excess Group Delay of loudspeaker in single location (room centre).

Spatial averaging + Equalization Threshold (black curve)



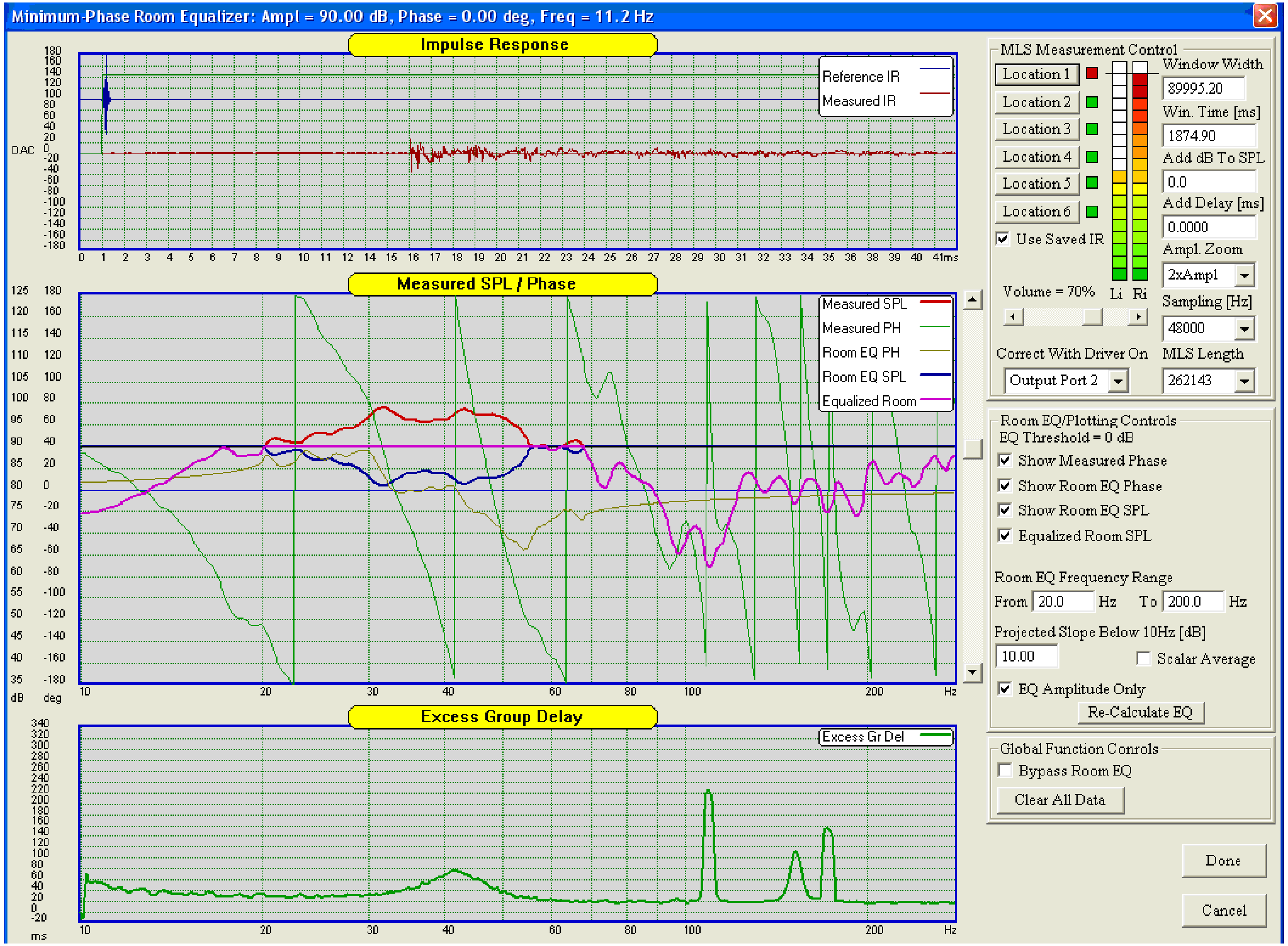
Being guided by the excess group delay graph, we should avoid equalizing the room response around 40Hz and 170Hz. **SPL below black curve will not be equalized**.

Room Equalization Transfer Function



Room Equalizer's complete Minimum-Phase (brown curve) and Linear-Phase (blue curve) transfer function.

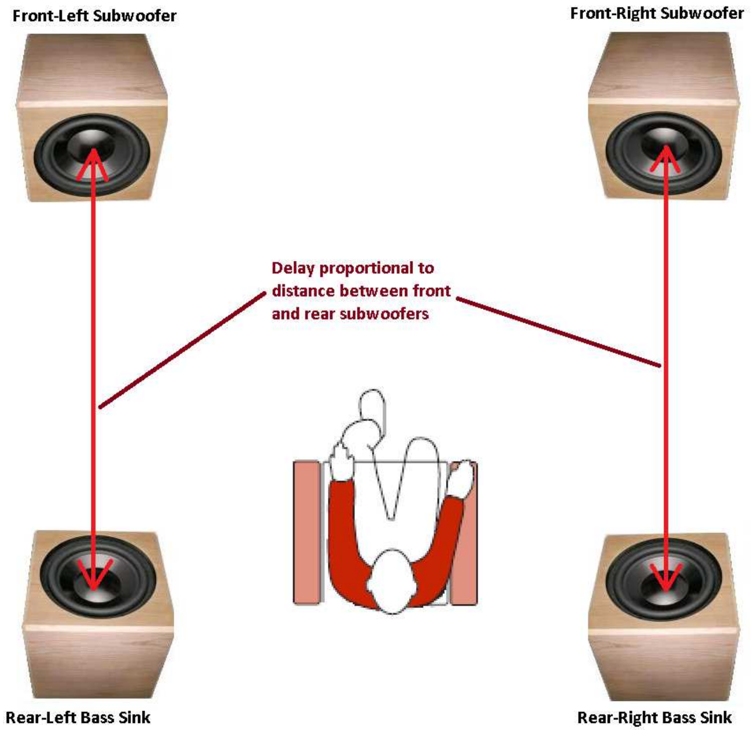
Summary of Room Equalizer function

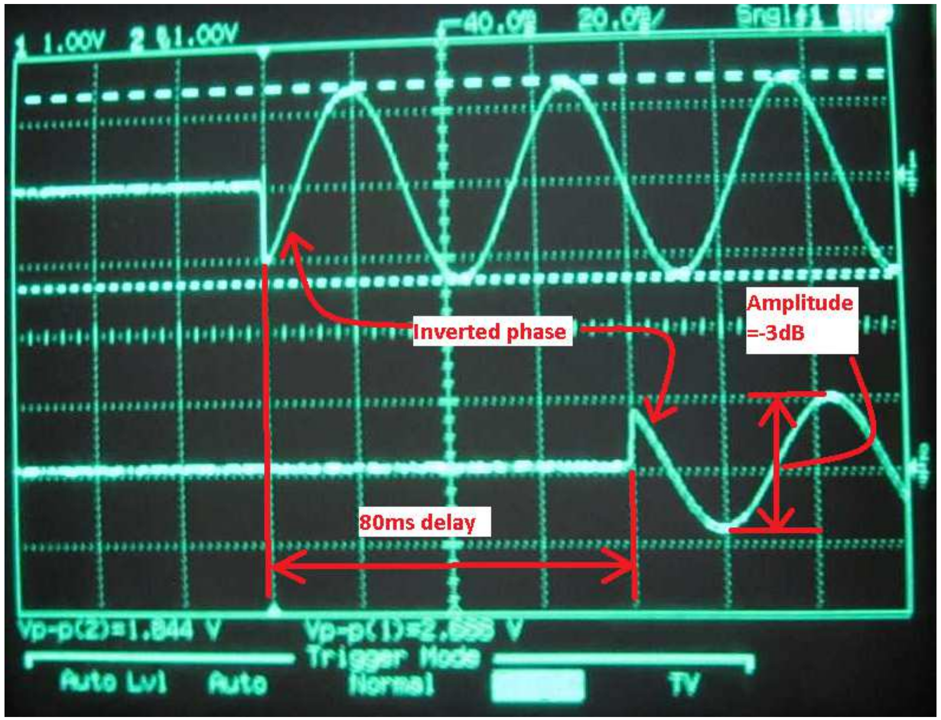


Controlled Acoustic Bass System (CABS)

(described by Nielsen in: http://vbn.aau.dk/files/62729248/LF_sound_field_control.pdf)

"…Create and maintain a plane wave propagating from front to rear. When the plane wave hits the rear wall another set of loudspeakers close to the wall will create a delayed version of the frontal signal but in opposite phase and with a proper gain so the reflection at the rear wall will be cancelled.."





"…Listening to music being played instead of a single frequency clearly shows that with CABS the **booming bass is removed in the source room and clearly reduced in the neighbour rooms**…"

Controlled Acoustic Bass System (CABS)

Nielsen's results

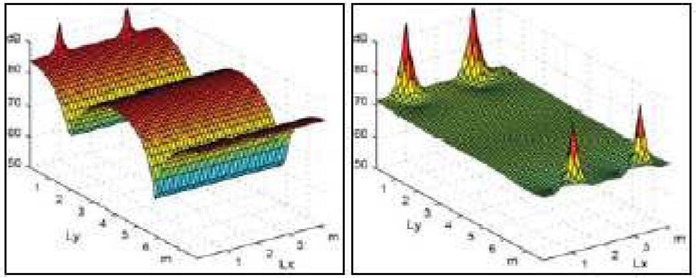


Figure 2. Simulation of 44Hz in room A .
Left: CABS off (0.2.0)   Right: CABS on (0.2.2)

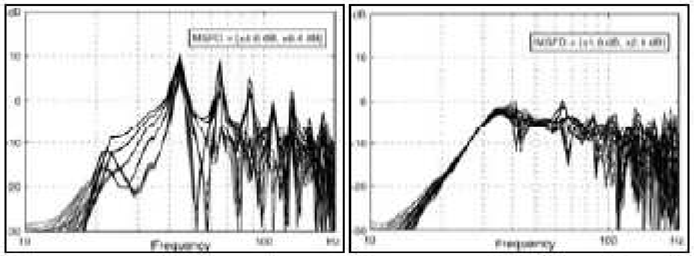


Figure 3. Measurements in Room A at 25 positions
Left: CABS off (0.2.0)   Right: CABS on (0.2.2)

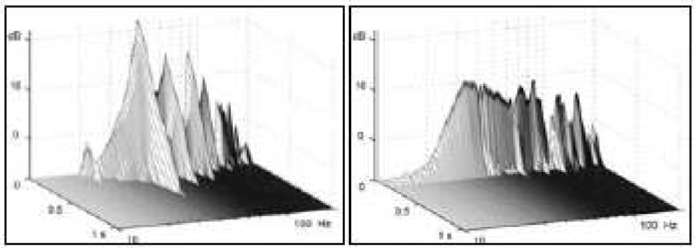
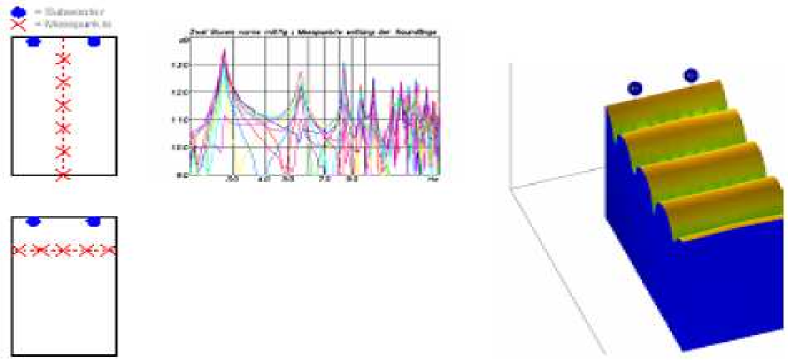


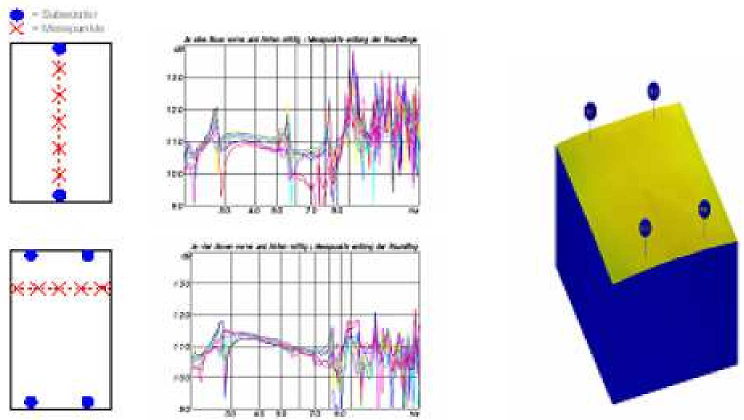
Figure 4. Measured cumulative spectral decay in room A in one point.
Left: CABS off (0.2.0)   Right: CABS on (0.2.2)

Kelin+Hummel in o800 Subwoofer User Manual

ARAM- Active Room Absorption Module



The bass coverage is much more even throughout the room with the addition of a second subwoofer in the front of the room. However, the troublesome resonance is not yet suppressed.



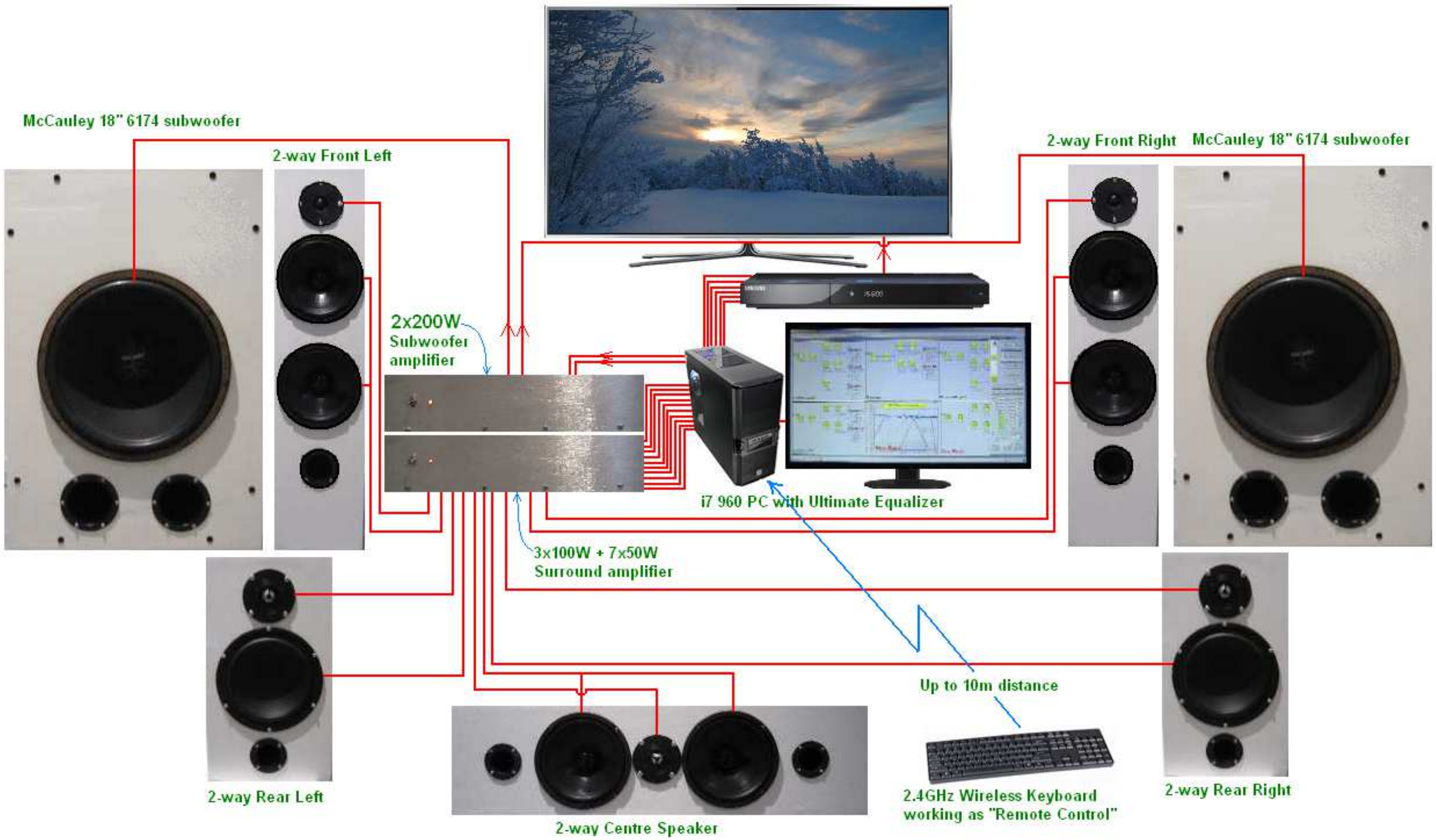
The addition of the ARAM subwoofer in the rear of the room produces an even distribution of bass across the room and over a wide frequency range

Room EQ Strategies

- Do not equalize frequency response at all – just use UE as an active DSP crossover with HBT equalizer, and get full benefits of an active system.
- Use built-in peak / notch / shelving filters to provide broad equalization. Up to 32 CAD elements can be used in each loudspeaker system. Even complex room EQ can be created this way.
- Equalize at single point.
- Perform multiple measurements at up to 6 locations, and use the average to equalize.
- Perform minimum-phase equalization or linear-phase equalization.
- Use CABS approach to “sink” bass energy at the back of room.
- Use both: RoomEQ + CABS.

Examples of UE Systems

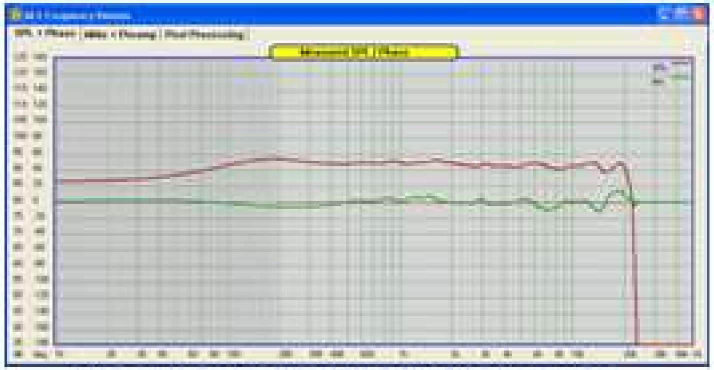
24bit/48kHz 5.2HT Audio Server with Analogue Amplifiers



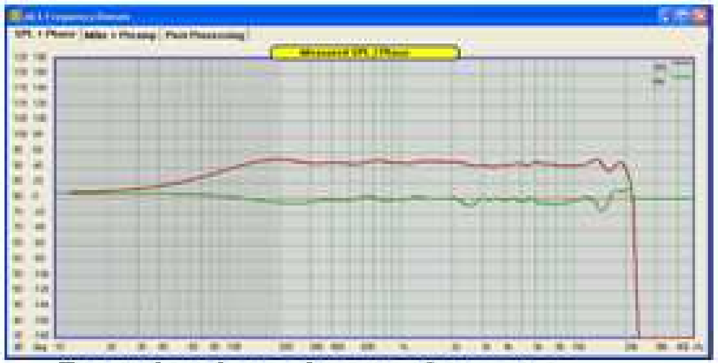
Examples of UE Systems

24bit/48kHz 5.2HT Audio Server with Analogue Amplifiers

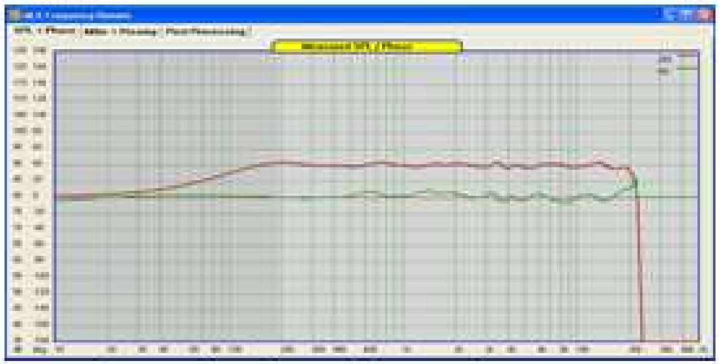
Subwoofer performance



Rear Loudspeaker performance



Front loudspeaker performance



Centre loudspeaker performance

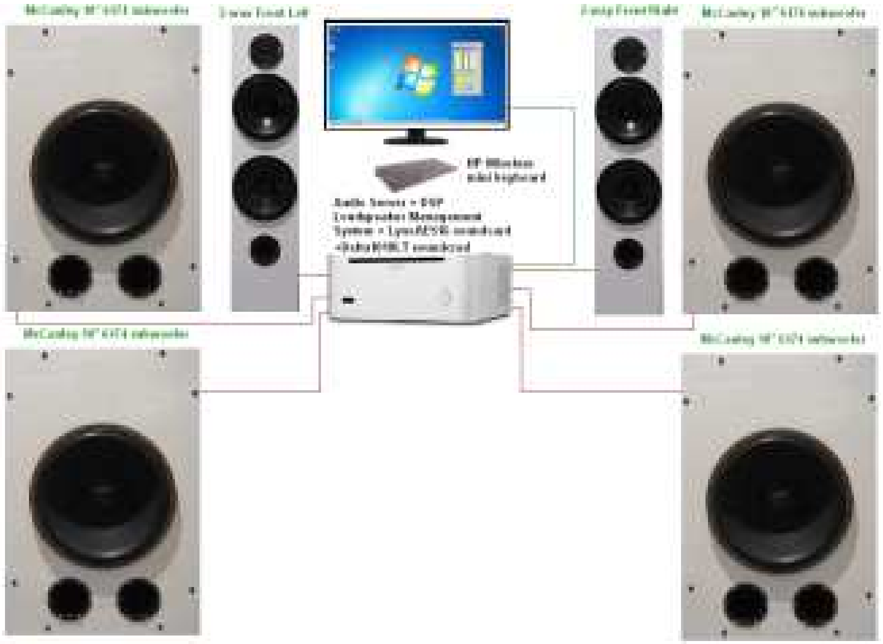
Evolution of the UE Systems

24bit/96kHz Digital Systems

2.0

2.1

2.2 BBM



2.4 BBM+CABS

Evolution of the UE Systems

24bit/96kHz Hybrid (Analogue/Digital) Systems

5.1

5.2 BBM

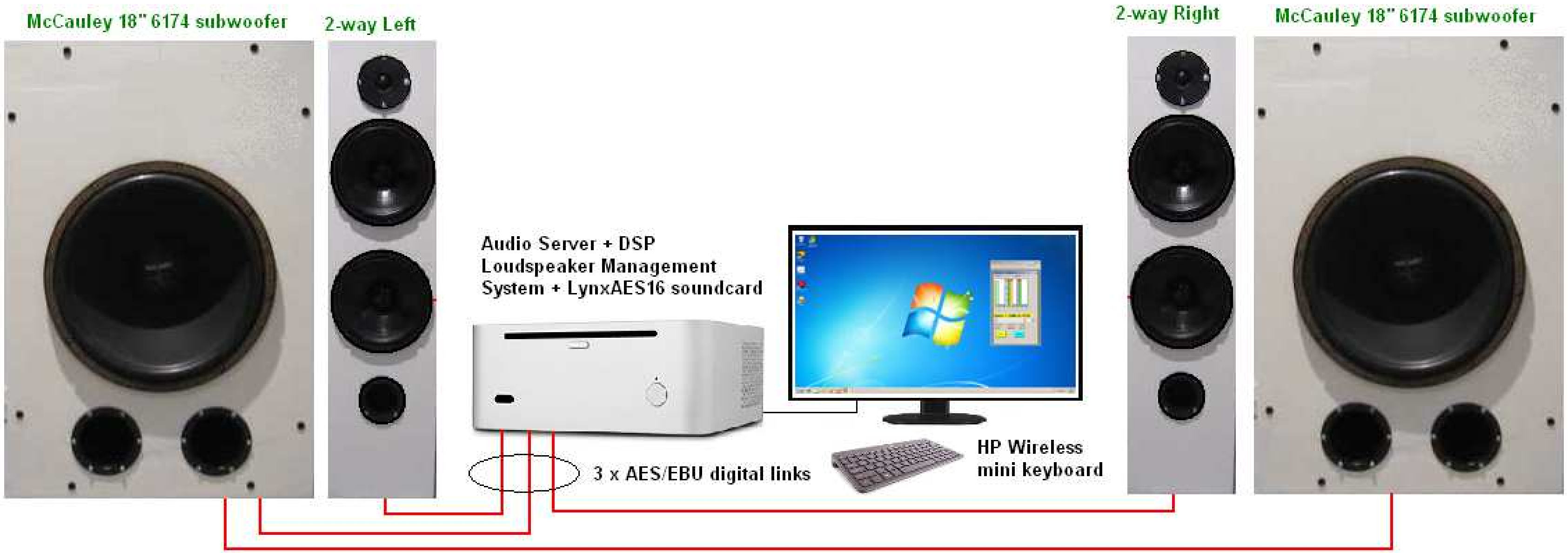
7.2 BBM

7.4 BBM+CABS

24bit/96kHz AES/EBU Audio Server with DSP Loudspeaker Management System

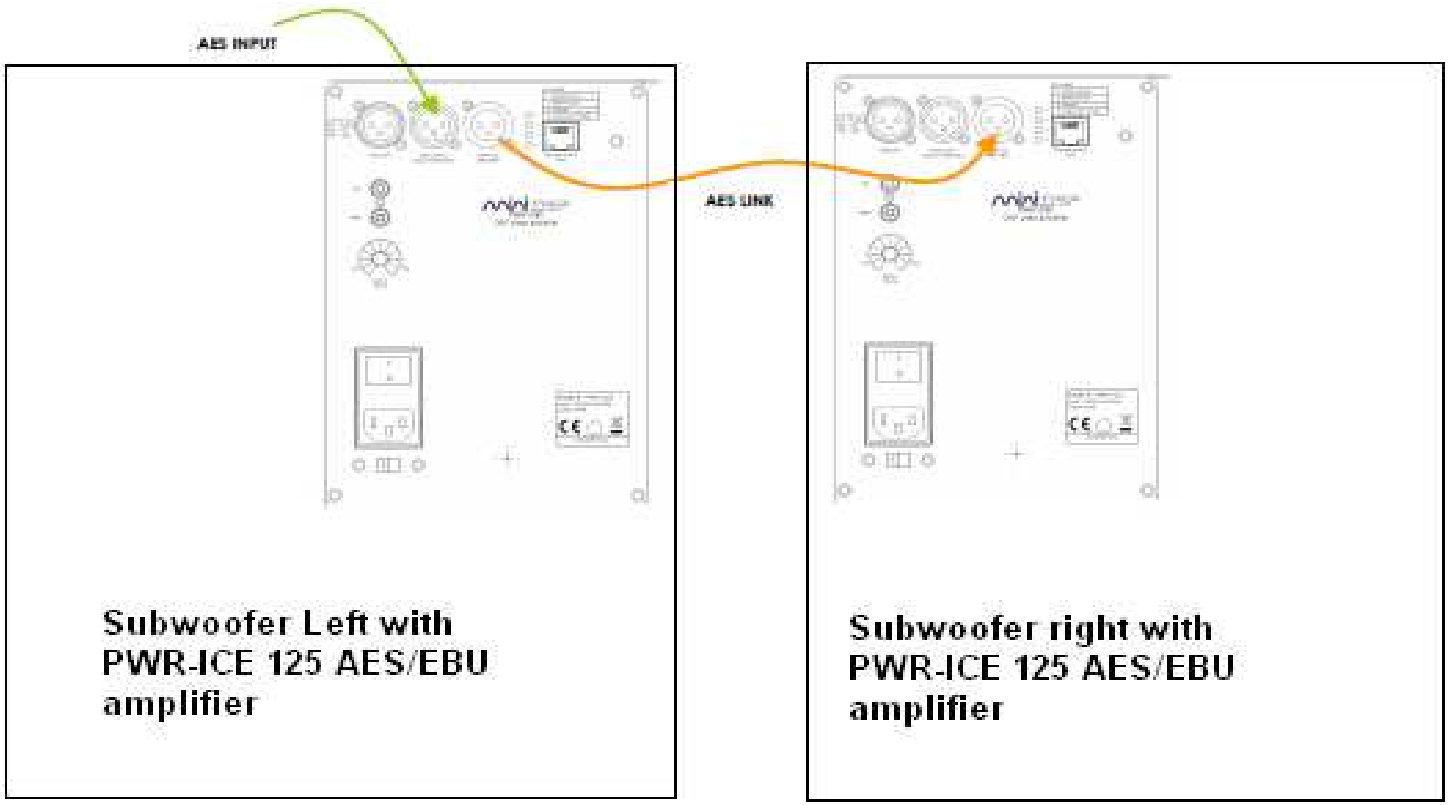
- System presented here is a complete ***audio playback system of studio quality.***
- ***Complies with*** http://www.aes.org/technical/documents/AESTD1001.pdf

- Realization involves only basic mechanical assembly with plug-and-play components, and can be easily accomplished by a DIY enthusiast.



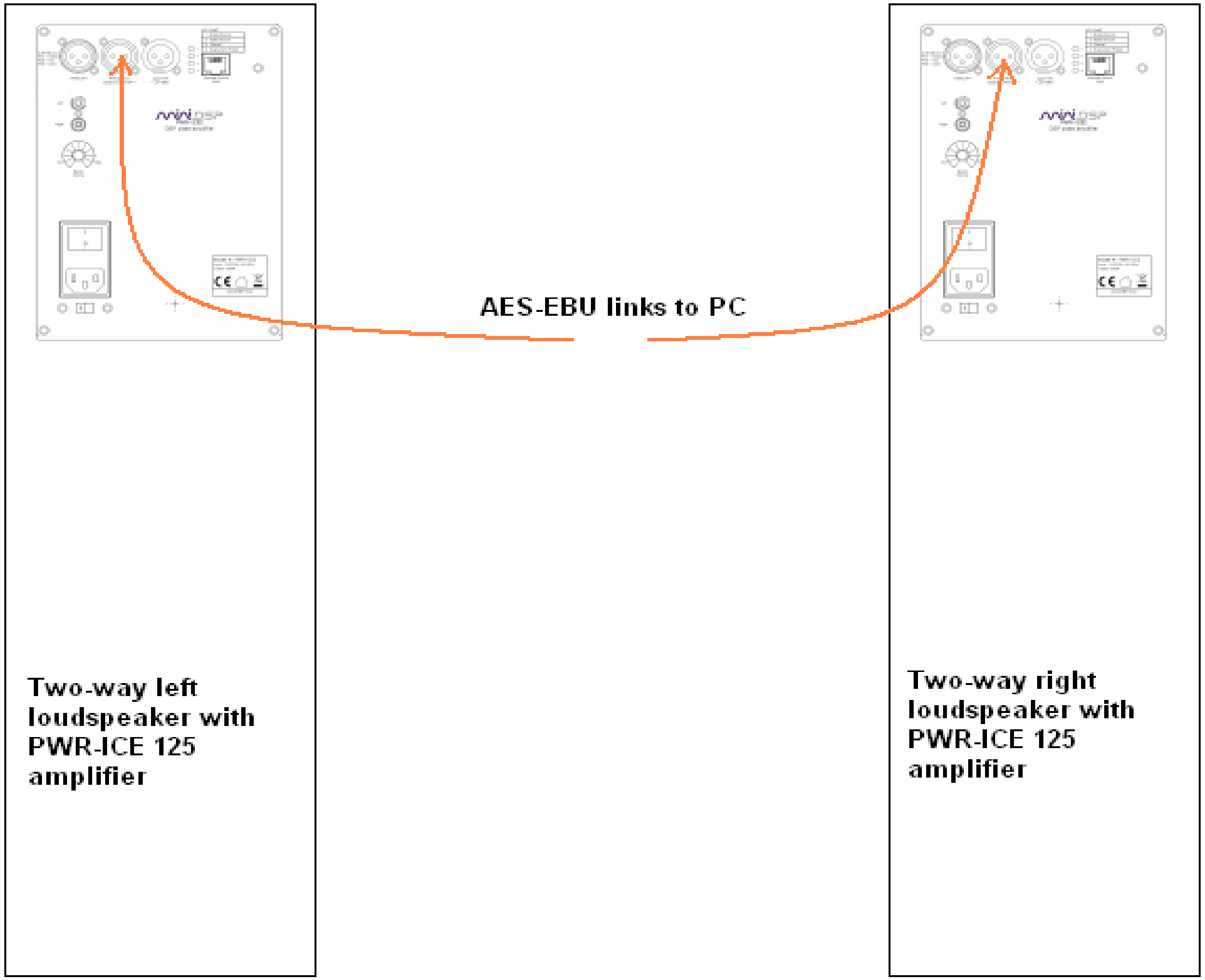
24bit/96kHz AES/EBU Audio Server with DSP Loudspeaker Management System

0.2 Subwoofer system



McCauley 6174 drivers in 300 litre vented (20Hz tuning) boxes with PWR-ICE 125 AES/EBU PWM amplifier in each box.

Left / Right Loudspeakers



2x8” woofer drivers + 1” tweeter driver with PWR-ICE 125 AES/EBU PWM amplifier in each 50 litre vented box, tuned to 45Hz.

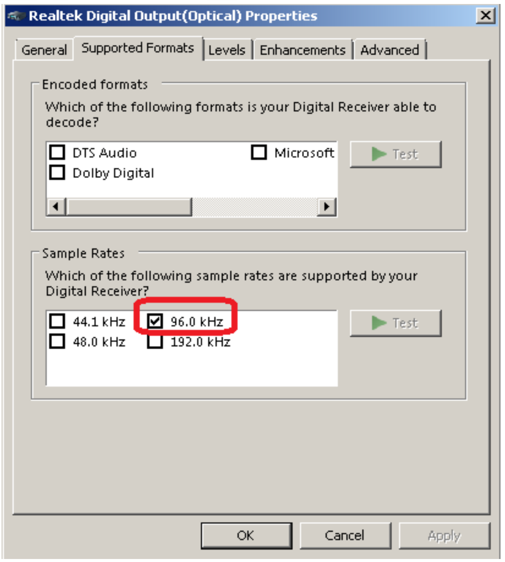
Some of the characteristics of the system

- 24bit/96kHz, studio quality processing system.
- Active system – allows amplifiers to exert maximum control over loudspeaker driver and makes crossover characteristics independent of driver loading.
- AES/EBU, or SPDIF links between all system components.
- HBT equalization of individual drivers to achieve flat frequency response.
- Linear acoustic phase for transient-perfect/image-perfect loudspeaker system.
- Precise time alignment of acoustic centres.
- Room EQ + CABS for sensible equalization/reduction of most offending room modes
- Practically unlimited loudspeaker voicing capabilities (all in linear-phase) executed with mathematical precision of a DSP software engine.
- Efficient PWM amplification system.

- Then, there is a very important, non-technical aspect of audio server. CD purchases are in massive continual decline these days – and for a good reason. The move to the internet-purchased music files started several years ago and is seen as the only way forward. Music files can be as popular as MP4 (good improvement from mp3) purchases from iTunes, right down to 24bit/96kHz high-end music files provided by a number of sources on-line. It’s convenient, but not only that. You can preview and purchase only the songs you like – rather than the whole CD. And this is a significant cost saving.

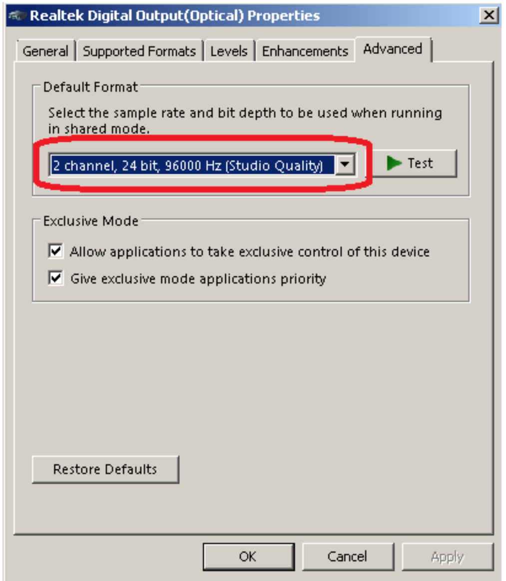
PC and Audio Codec

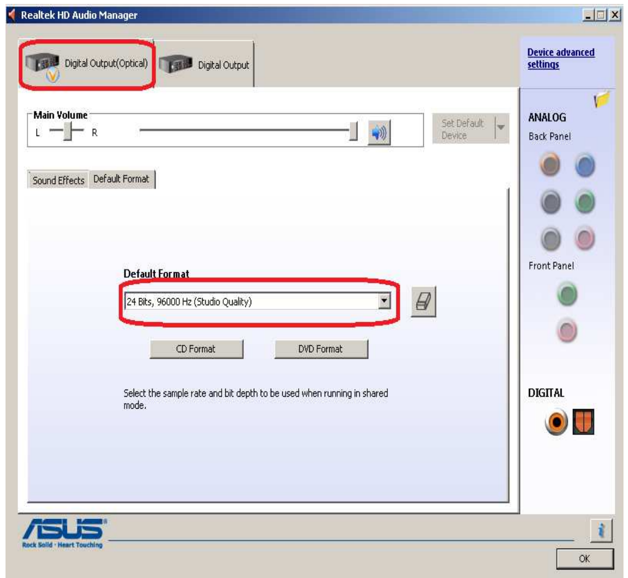
ASUS P6X58D-E motherboard, Socket 1366, which can accommodate Core™ i7/960 Extreme Edition/Core™ i7 Processors. http://www.asus.com/Motherboards/P6X58DE/



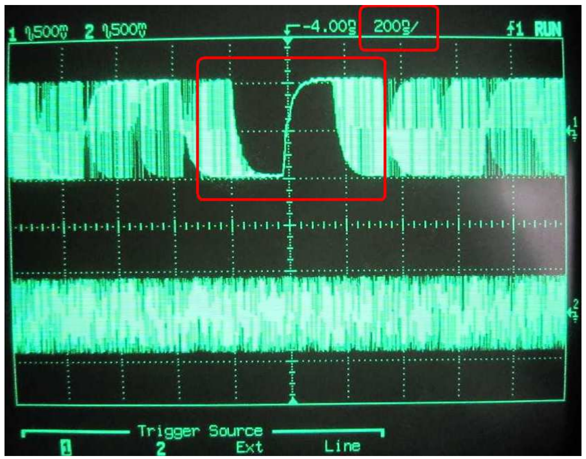
Motherboard SPDIF output is looped back to LynxAES16 Digital Input 1

PC and Audio Codec

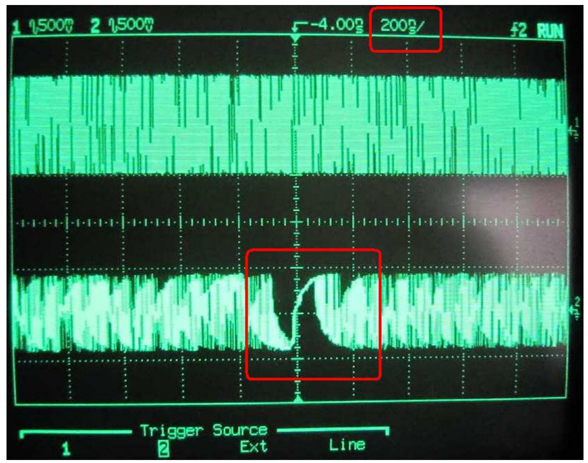




The ASUS Realtek Audio Manager is set to Digital Audio (Optical) and 24bit/96kHz sampling.

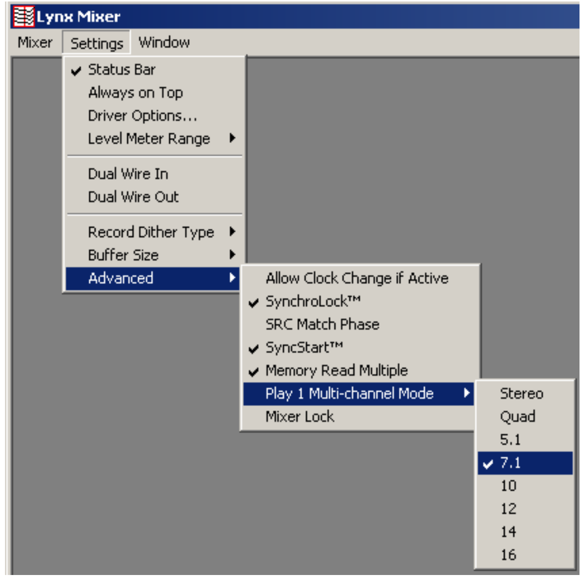


44.1kHz SPDIF with 200ns time-base

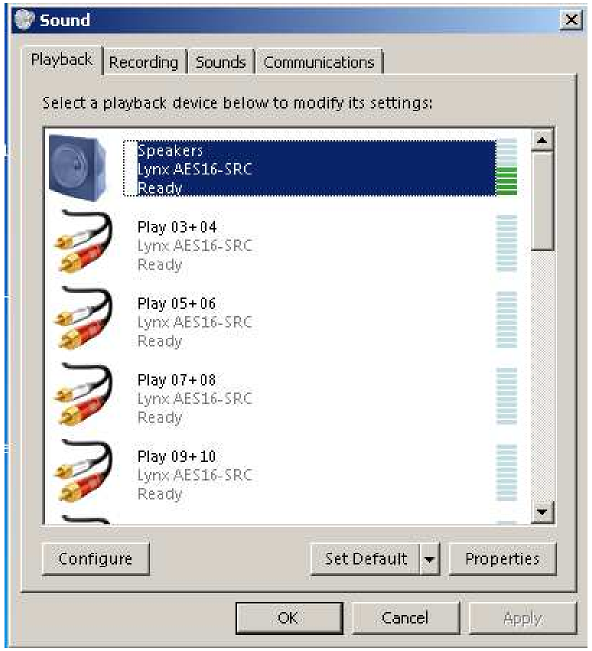


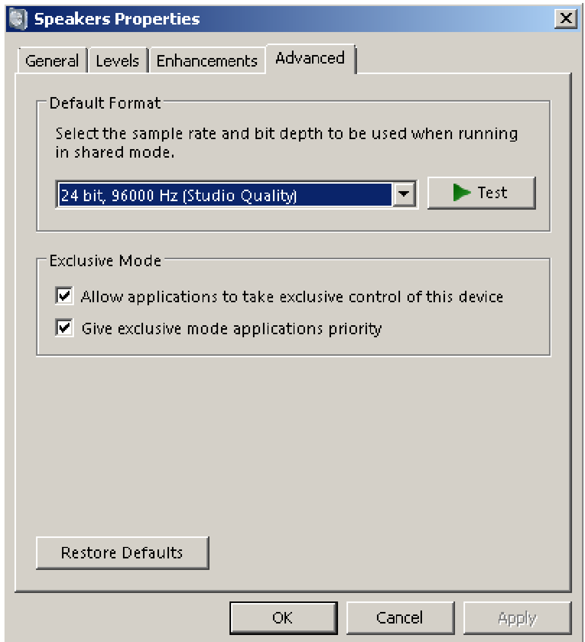
96kHz SPDIF with 200ns time-base.

The LynxAES16 PCI soundcard



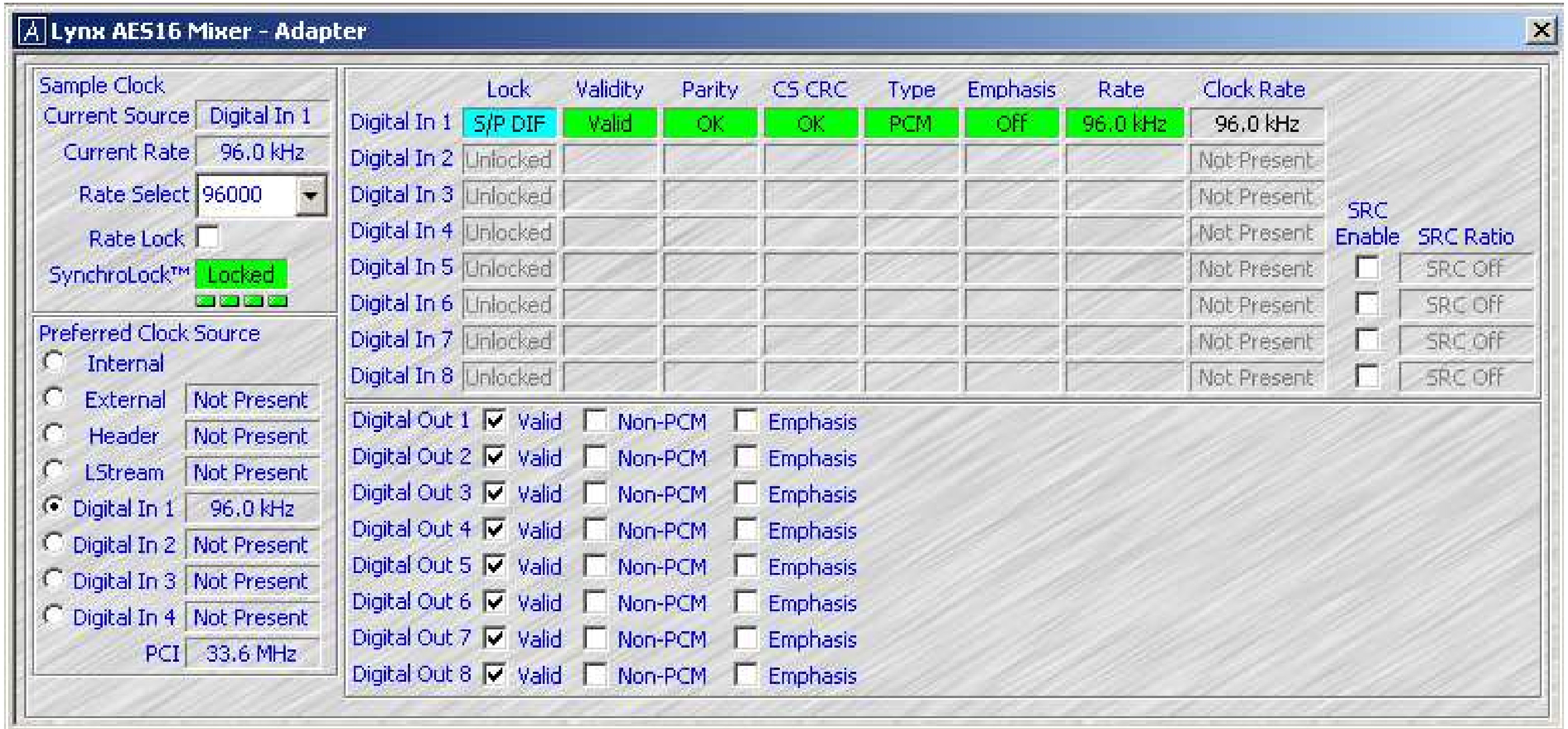
Lynx AES16 AES/EBU PCI digital soundcard, and sound card settings.





The LynxAES16 PCI soundcard

- Does it work OK when Windows Media Player is active? – hopefully yes.
- The following information should be displayed by the Lynx Mixer.
- Please note, that “Preferred Clock Source” is selected as “Digital In 1” – this is where we connected the motherboard audio link. The “Rate Select” is set to 96kHz.



Setting up PWR-ICE125 Amplifiers

3.1.4 Channel Mode

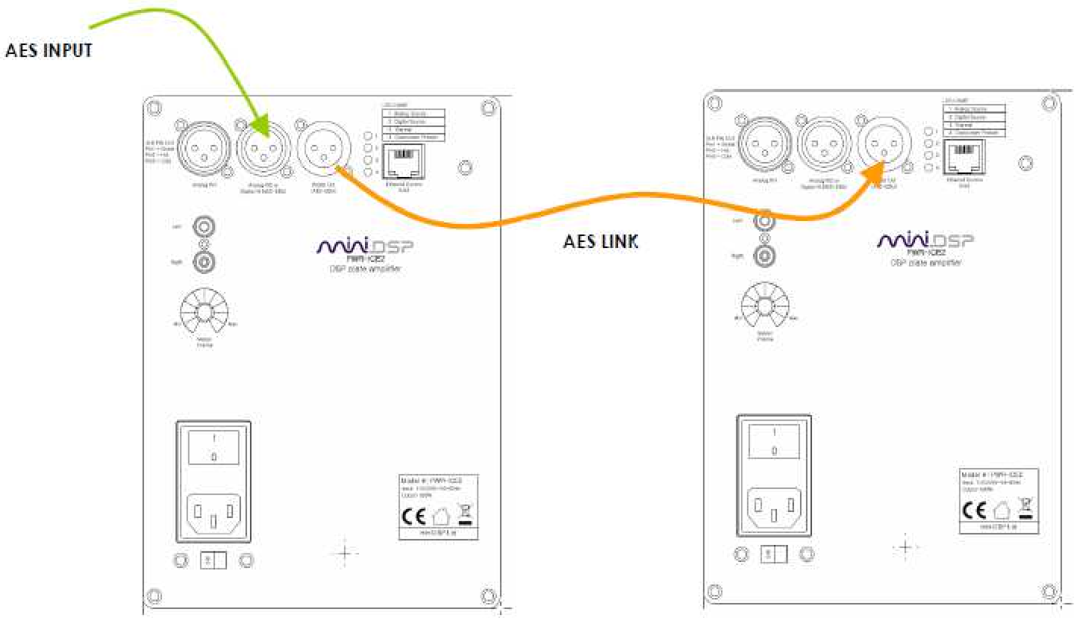
The PWR-ICE 125 runs a 1 x IN, 2 x OUT DSP configuration and will operate under 3 modes selectable from the GUI of the plug-in.

Channel 1 (L): Mono input from RCA unbalanced Left, analog XLR input 1 or Digital AES input Left
Channel 2 (R): Mono input from RCA unbalanced Right, analog XLR input 2 or Digital AES input Right
Mixed L&R (Mixed): Mixed input of RCA unbalanced Left & Right, analog XLR input 1& 2 or Digital AES input left&right.

2.3.7 Digital Link OUT

The digital Link OUT of the PWR-ICE is a "buffered" AES output of the Digital IN. It's an AES-Output allowing 2 x plate amplifiers to share Audio. It could be used for a stereo configuration or dual Subwoofer.

NOTE: The Digital Link OUT is only a buffered output of the digital Input. Unless you have a digital input (AES/SPDIF) on XLR#2, this output will not be enabled.



PWM Amplifier Phase Response

Frequency Response (SE-mode)

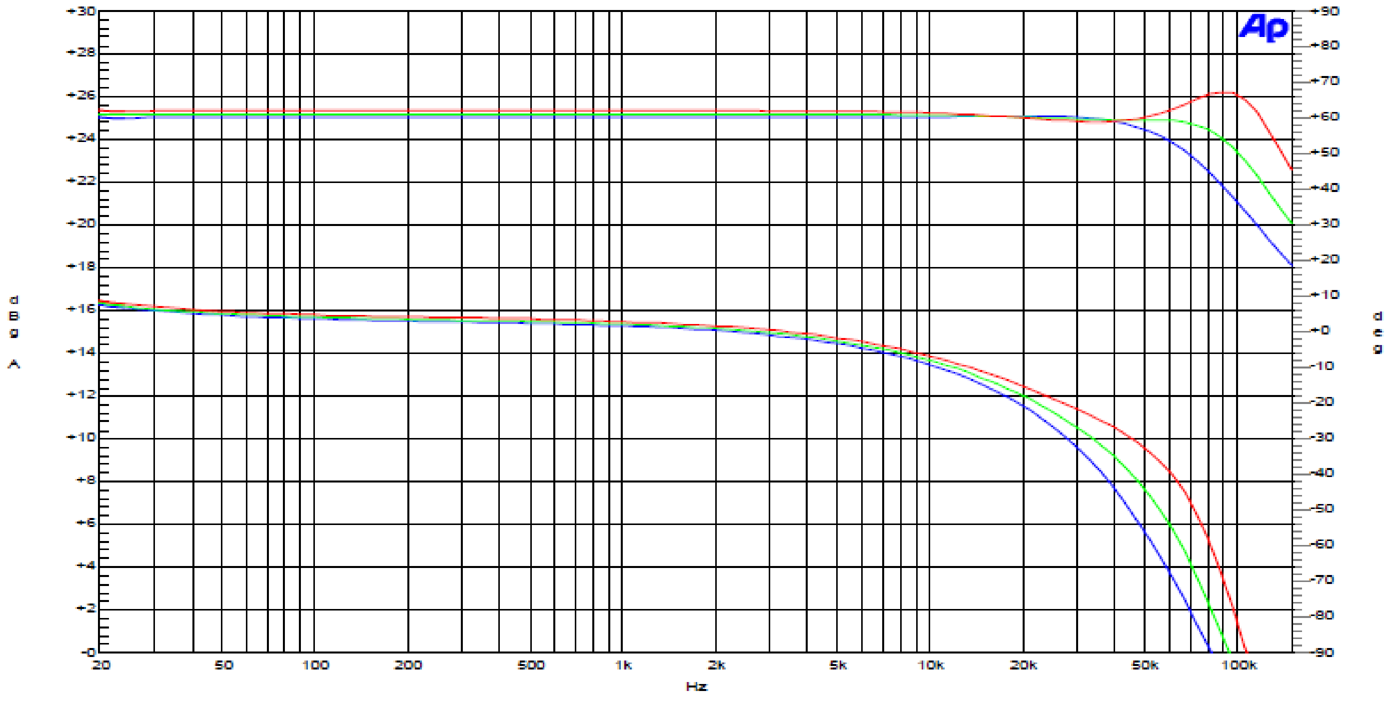


Figure 3: Frequency response in 4Ω (blue), 8Ω (green) and open load (red). Top – amplitude. Bottom – phase.

- If the design aim is a minimum-phase system, then the rolling phase response of the PWM amplifier can be disregarded.

- However, in a linear-phase system, the phase irregularity needs to be compensated for. The design strategy for accomplishing such compensation is as follows.

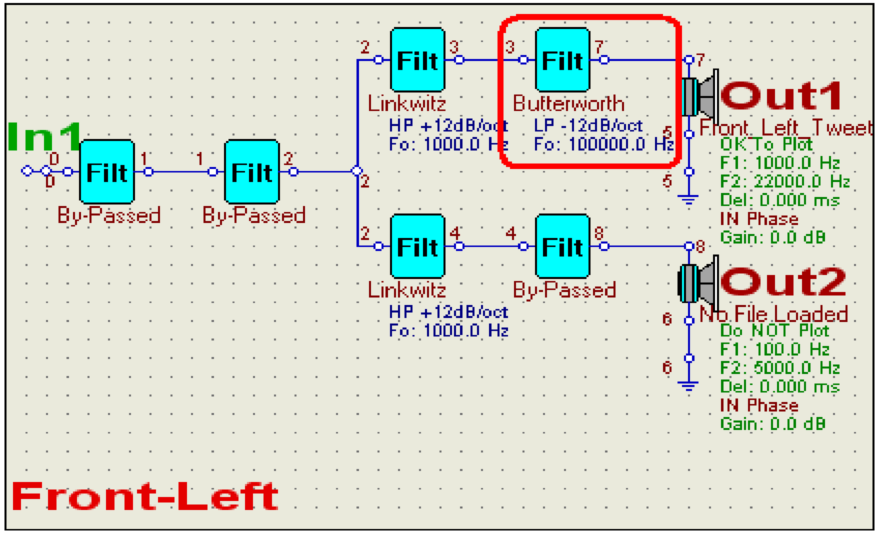
PWM Amplifier Phase Response

Introduce an extra phase roll-off, which mimics exactly the phase roll-off of the PWM amplifier.

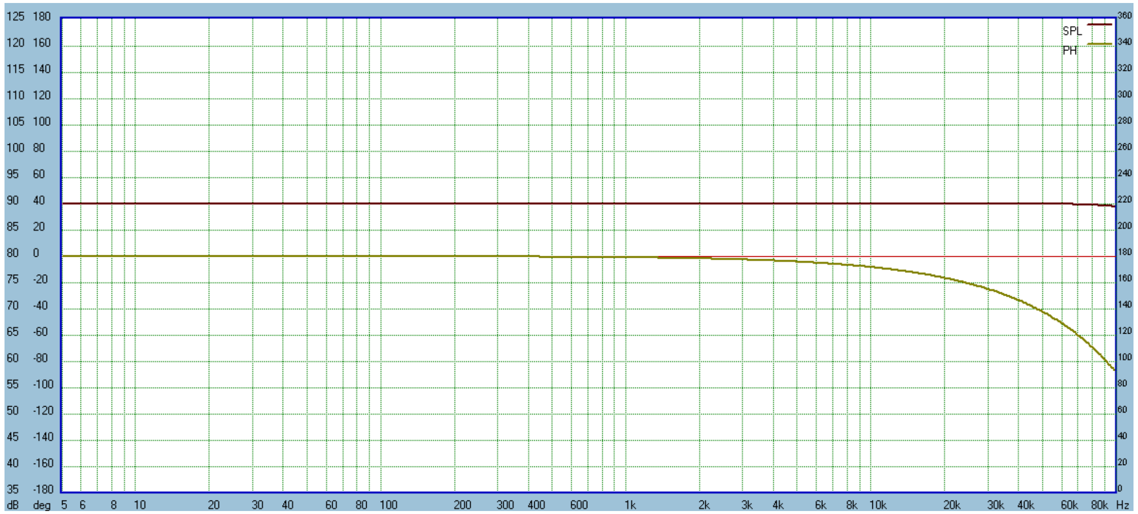
Therefore, the **inverted HBT method for phase linearization, will overcompensate** the phase by the exact amount of the extra phase roll-off.

Consequently, when the complete chain of devices: the loudspeaker + crossover + PWM amplifier + overcompensated inverted HBT phase response is played through, the final phase will be a flat line at 0deg.

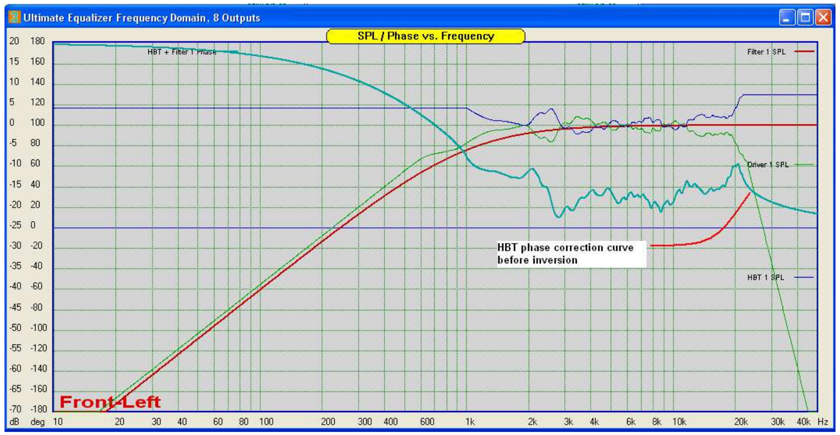
Here is an example of the extra device inserted in the tweeter DSP processing path.

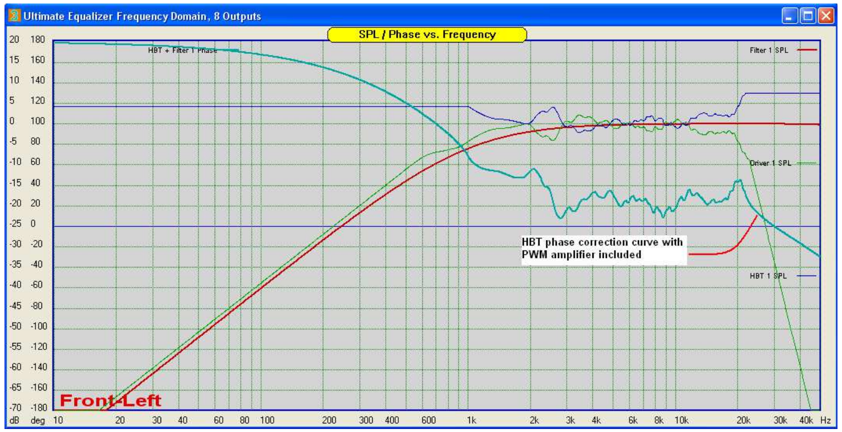


PWM Amplifier Phase Response



PWM Amplifier Phase ResponsePWM Amplifier Phase Response

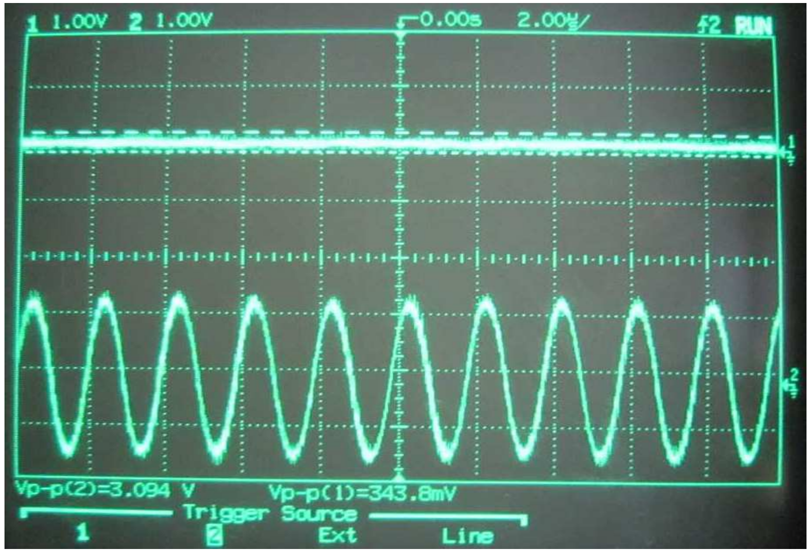




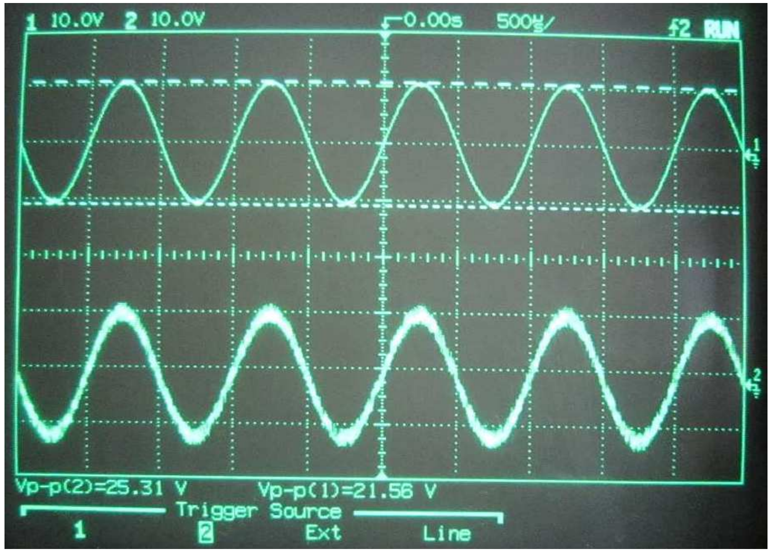
HBT phase response **without** and HBT phase response **with** the extra phase roll-off.

PWM Amplifier Phase Response

- 500kHz PWM amplifier switching component still being present on the output?.
- Without additional filtering there may be up to 3Vpp of 500kHz present in the output.
- Simple LC lo-pass filter with 25uH coil and 150nF capacitor, will improve suppression of the carrier significantly. This additional filter, will increase phase shift at 20kHz beyond the specified value, and again, may need to be taken into account for linear-phase designs. The effect of this additional filter also needs to be compensated in the Ultimate Equalizer. More information on PWM output filtering can be found in: http://www.ti.com/lit/an/sloa023/sloa023.pdf
- http://pdfserv.maximintegrated.com/en/an/AN624.pdf



Bottom: 500kHz component before filter, Top: after filter – 1V/div



Bottom: 1kHz tone + 500kHz component before filter, Top: after filter – 10V/div

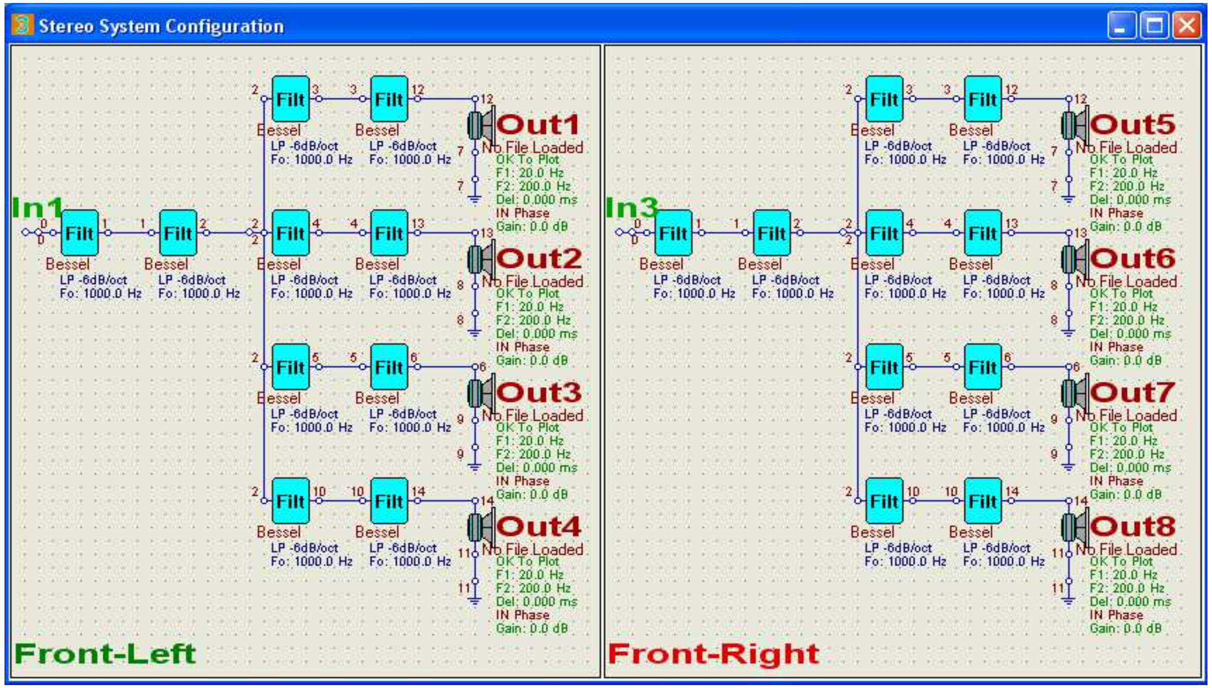
Loudspeaker Management System – UE6 DSP engine

Large selection of filter configurations and types, and the ability to cascade them any way you like.

Cascade other filtering elements, like notches, shelving and peaking elements with adjustable Q-factor.

Each one of these long chains can be applied as a filtering channel for individual driver in the enclosure.

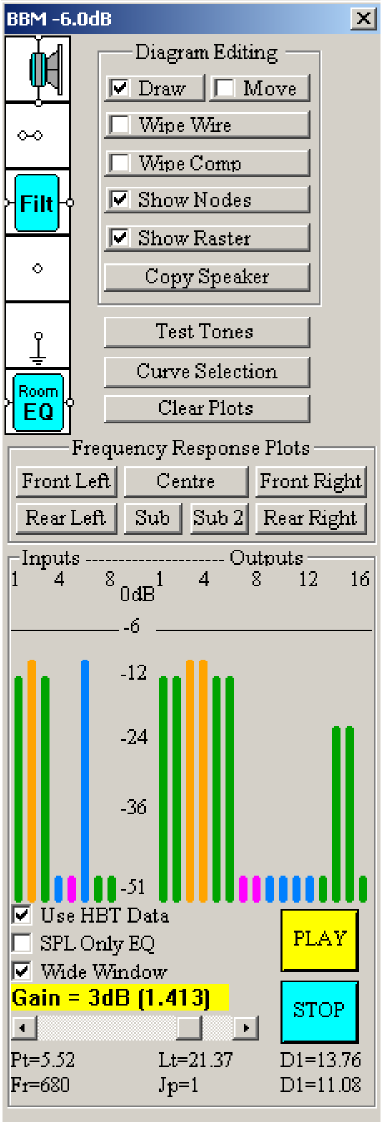
In order to visualize the whole crossover, you would simply pick filtering elements from the available tray of components, and then place and link them on the screen to effectively built the whole crossover as a block diagram with interconnected filtering elements.

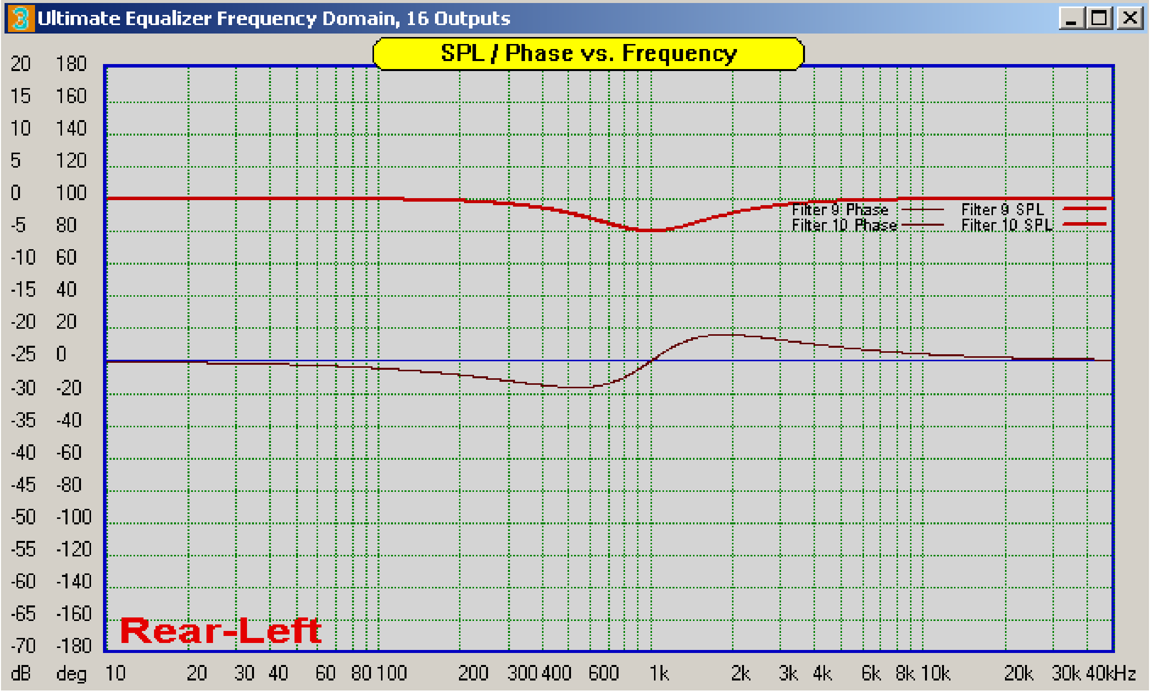


Loudspeaker Management System – UE6 DSP engine

The "tray" is shown to the right. To keep things simple, there are only three active elements, using which you can built the entire crossover and room EQ.

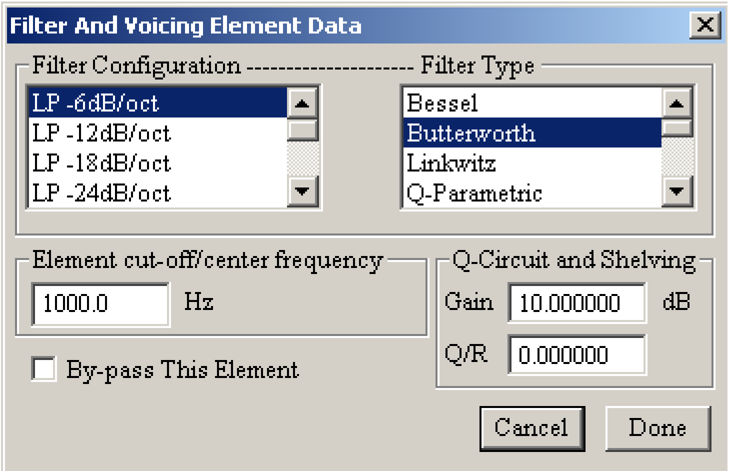
Schematic pick-and-place component tray or use one of 17 pre-set configurations



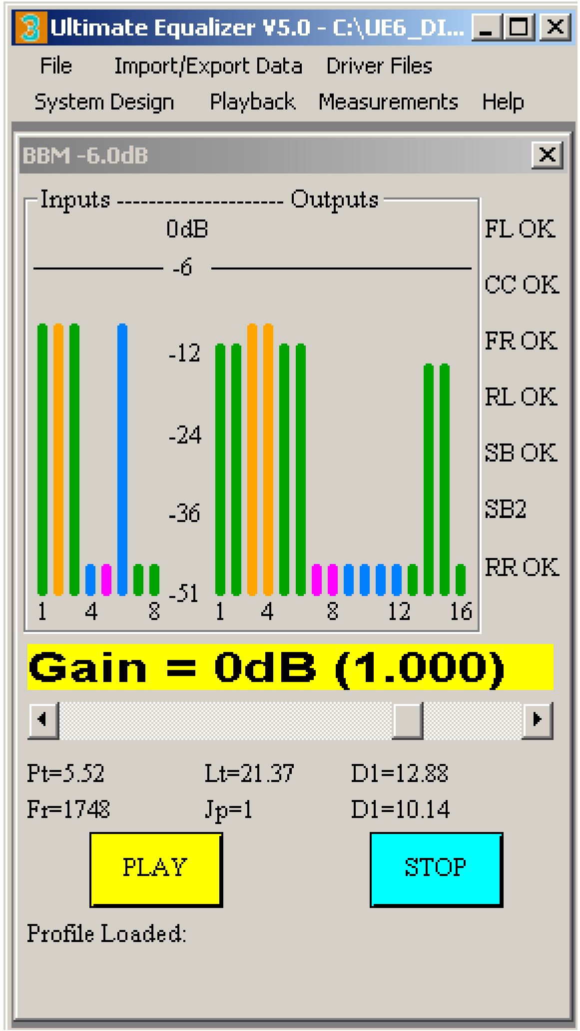


Frequency-domain curve plotting screen…

Loudspeaker Management System – UE6 DSP engine

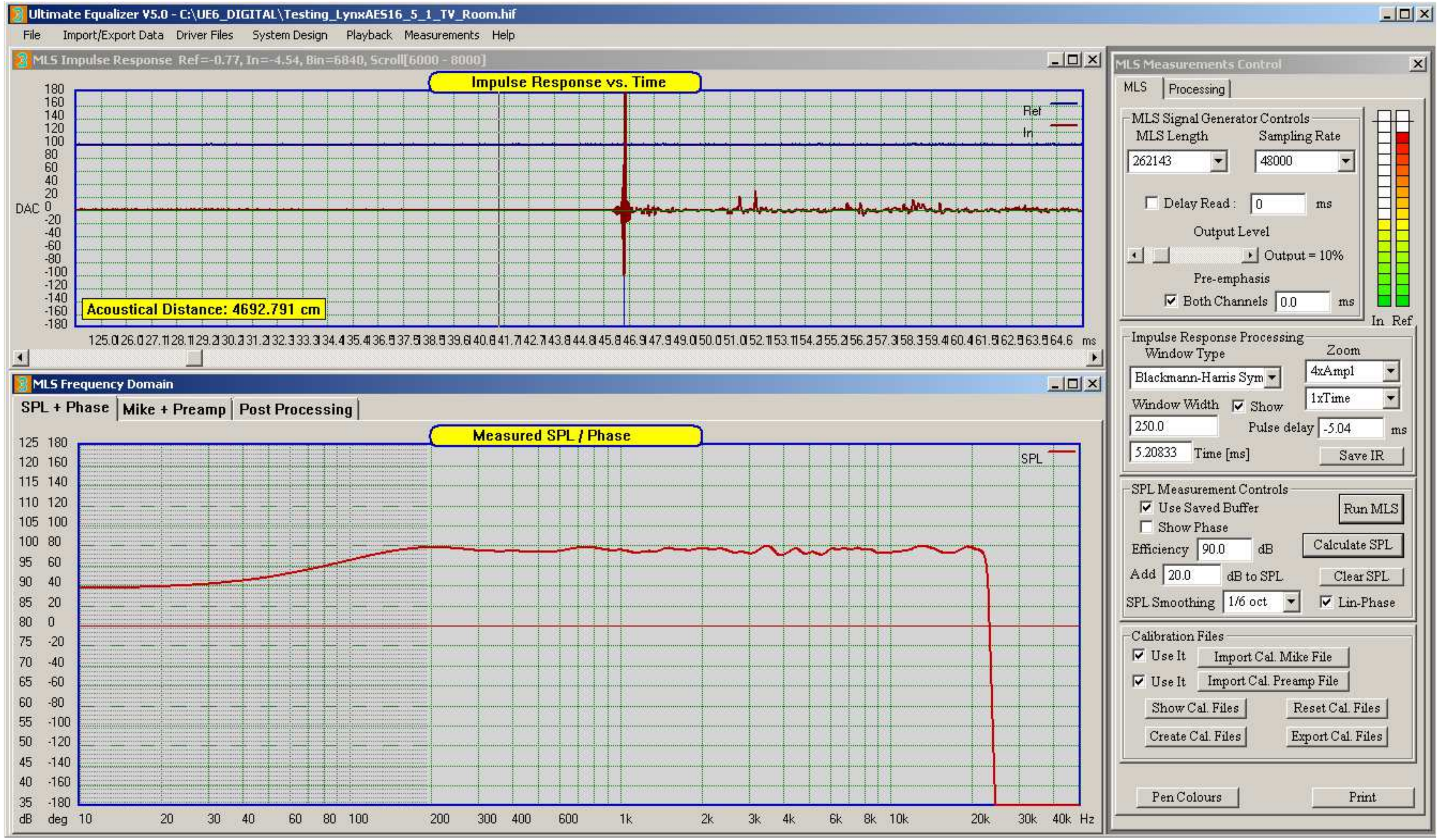


Large selection of built-in filters



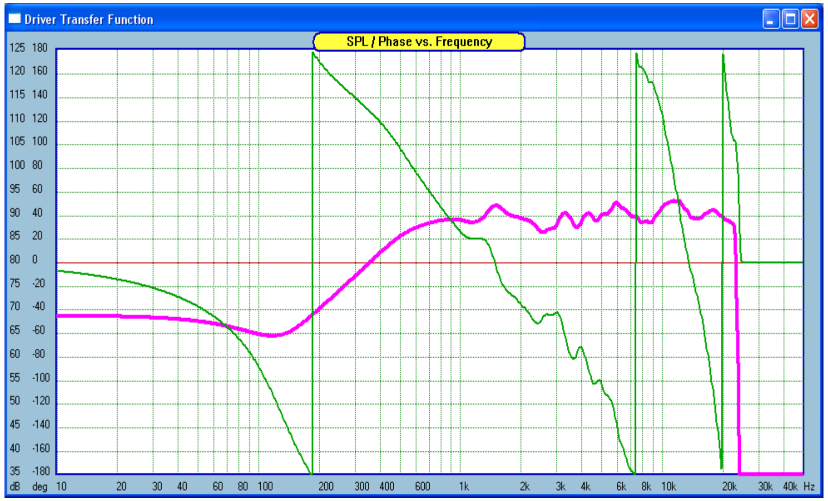
After the system has been designed, UE can be switched to “Playback” Mode.

UE MLS Measurement System



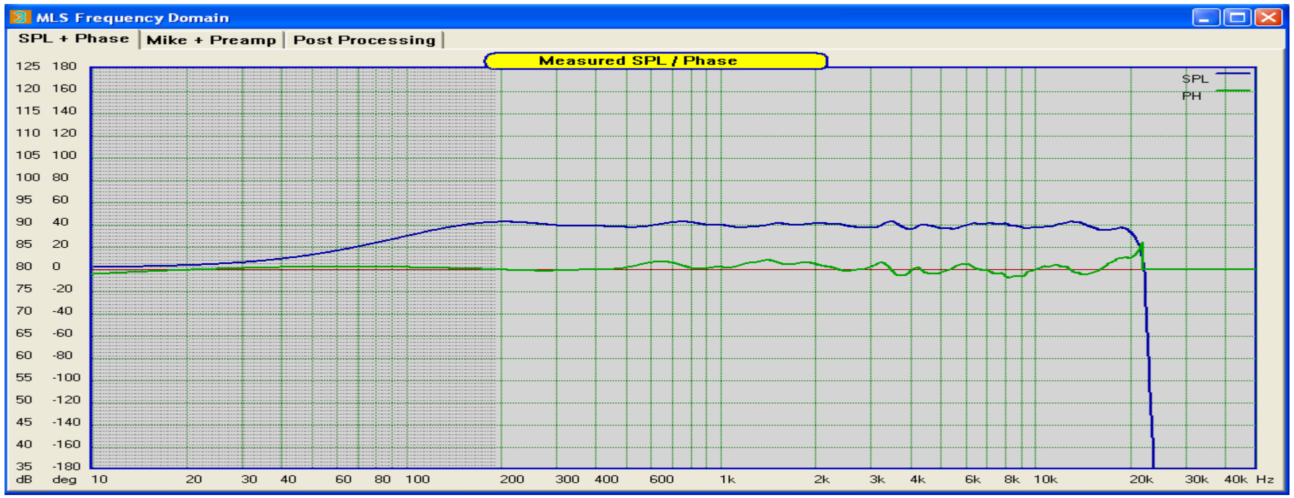
Typical Measurement Results

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

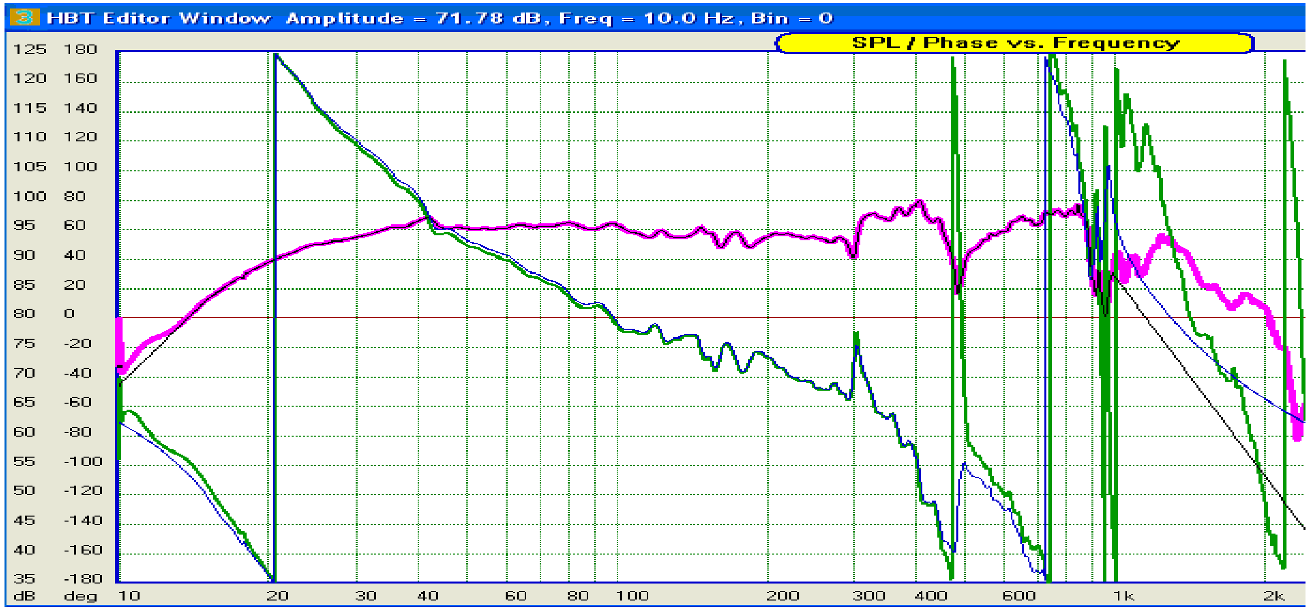


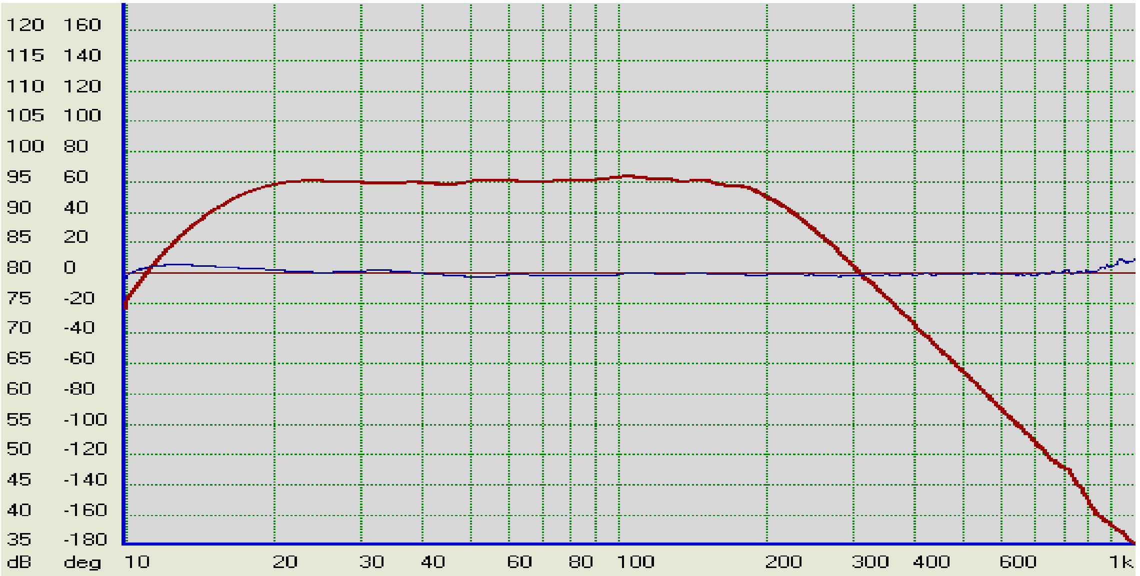
SPL/phase measurements of woofer and tweeter in a 2-way system.

to this level of performance…….

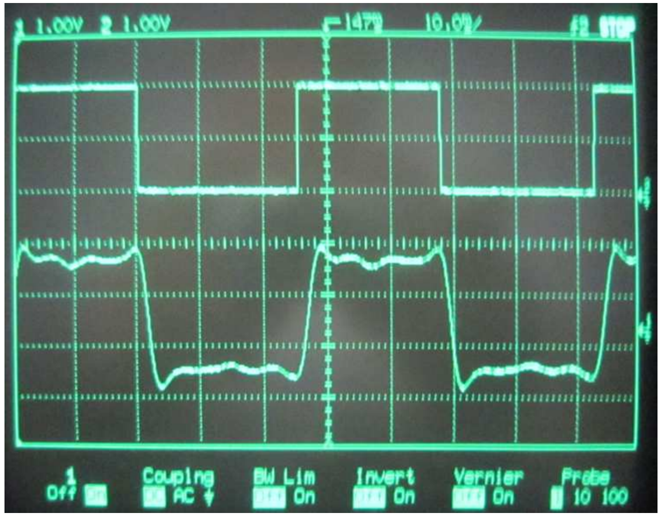


Typical Subwoofer Measurement Results

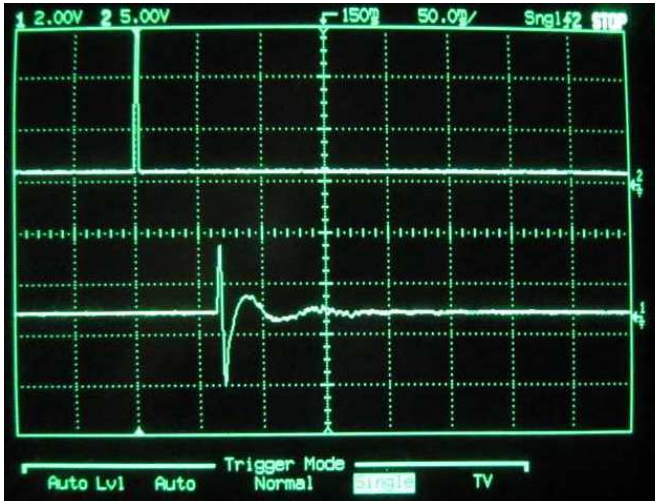


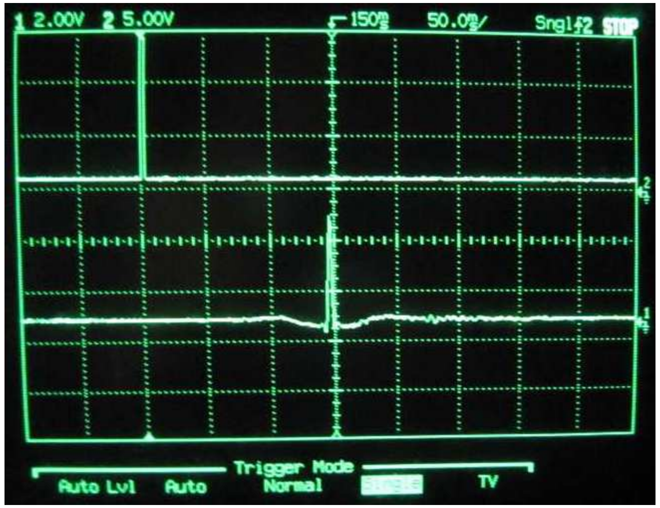


Typical Time-Domain Measurements



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





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

The minimum-phase version of the subwoofer has converted the clearly asymmetrical pulse into a much more symmetrical bi-polar pulse with post-ringing

Working UE5 Prototype

Summary Comments on the System

- Advanced, linear-phase audio system, which meets current and future requirements for handling digital music files of any type – Windows Media Player.

- Maximum DSP capabilities with LynxAES16 sound card are 2x8way system, and output power for each channel is determined by the PWM-ICE amplifier configuration from miniDSP.

- The prototype described here delivered frequency response between 45Hz – 21kHz (+/- 0.8dB), using quite average drivers in the 2-way stereo loudspeakers. And it delivered 16Hz – 200Hz (3dB) bandwidth for the subwoofers.

- Sensible room equalization may be required for your AV room. Just to neutralize the most offending room modes – that’s all you need there.

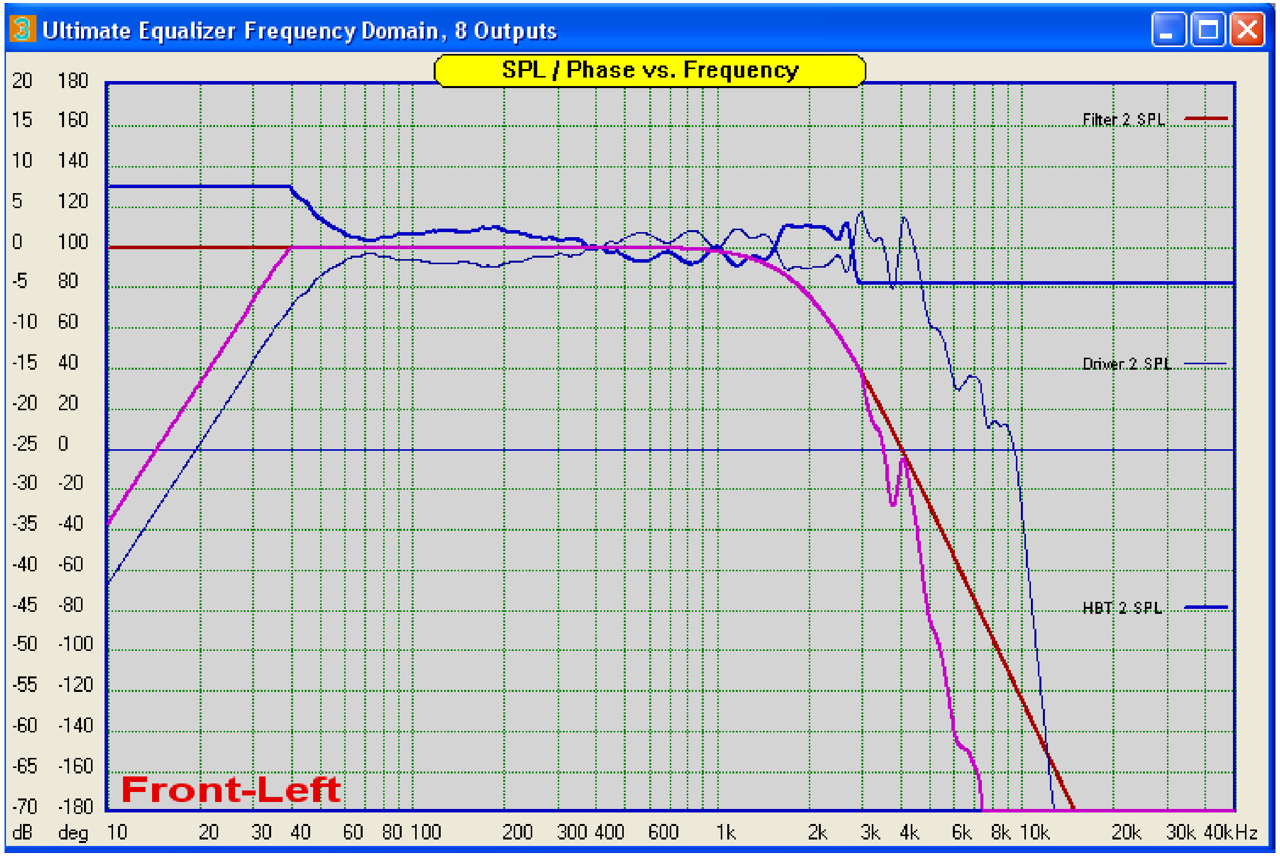
- The ease-of-use is guaranteed by the media player functionality. Downloading your favourite music files and grouping the files into play-lists, guarantees, that you’ll never pay more for your music than absolutely necessary.

ON / Off-axis Equalization

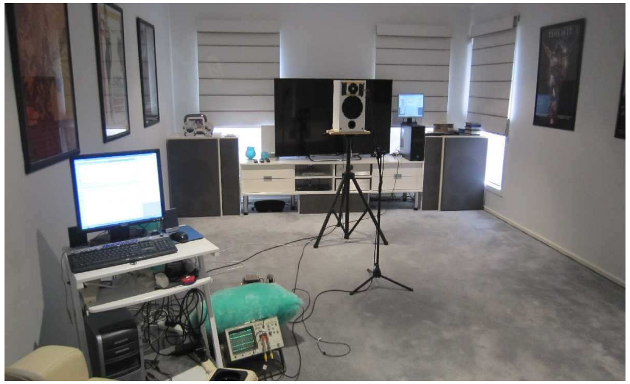
All subsequent measurements were conducted in-room.

Due to FFT windowing, the low-end of the frequency response is missing in all plots.

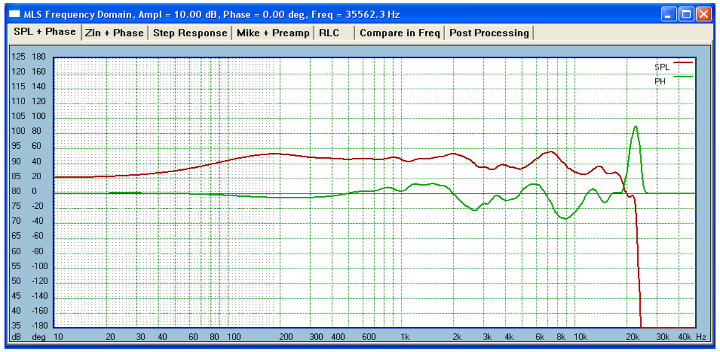
This is unfortunate, as the HBT equalization performs very well in the low-end for all polar angles but you'll not be able to see (and compare) these benefits on subsequent plots.



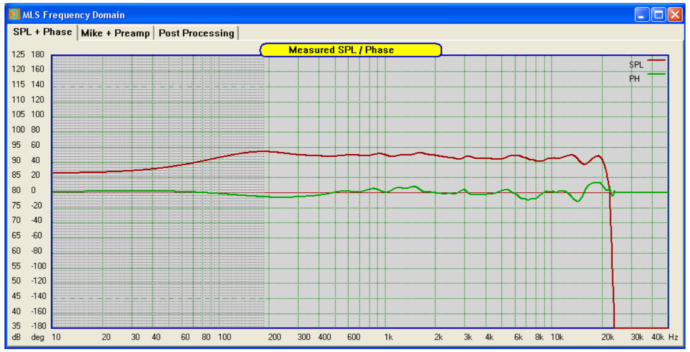
ON / Off-axis Equalization



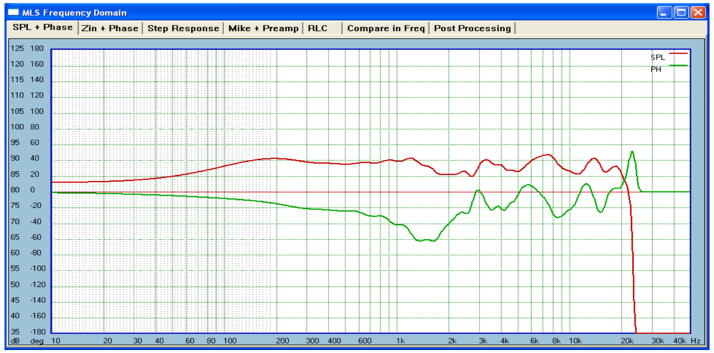
Measurement set-up



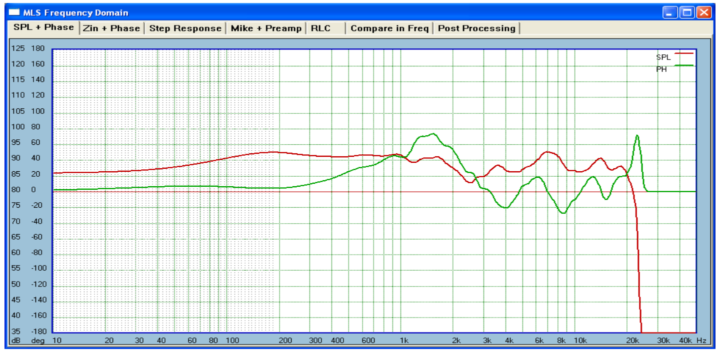
+/-30deg Horizontal



48kHz sampling. 1m, On axis



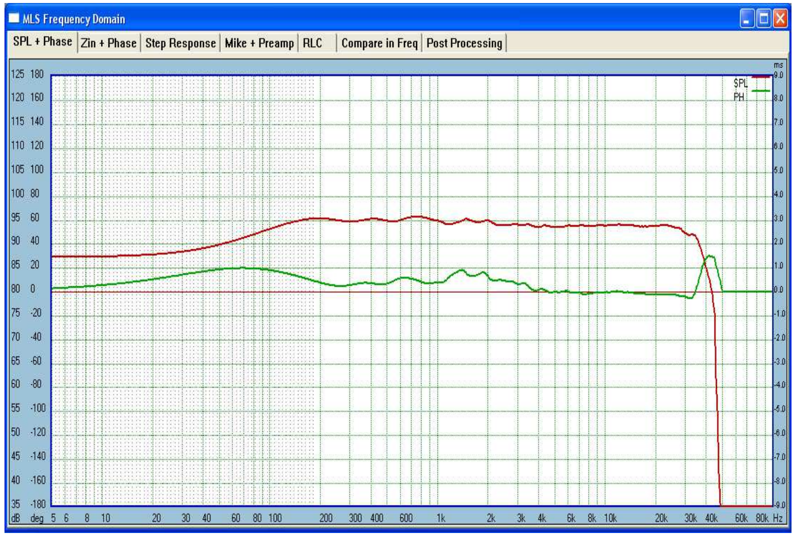
+15deg Vertical



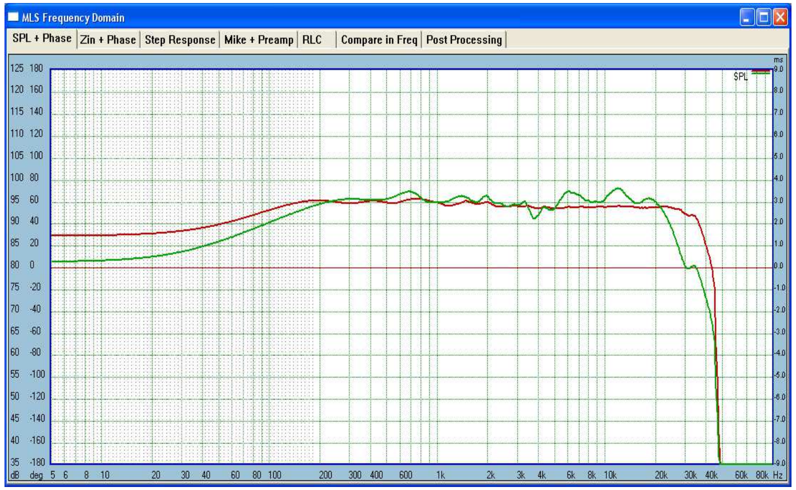
-15deg Vertical

WTW loudspeaker with HBT equalization to 30000Hz

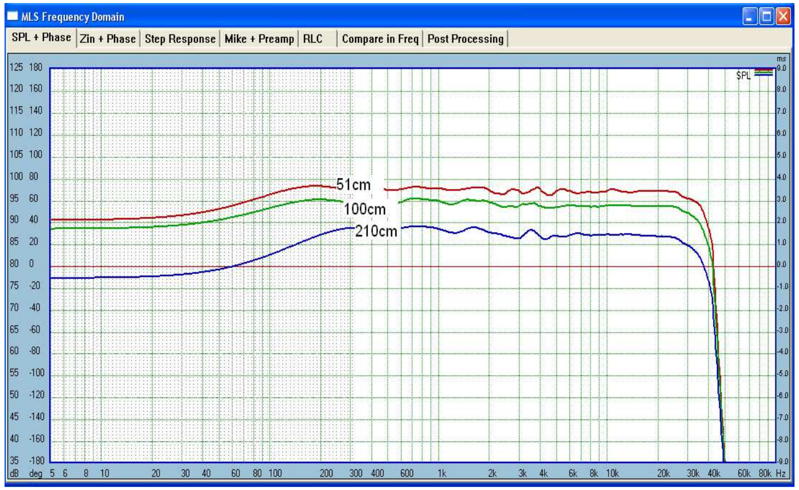
Measurement set-up



SPL with 96kHz sampling



0deg/1m

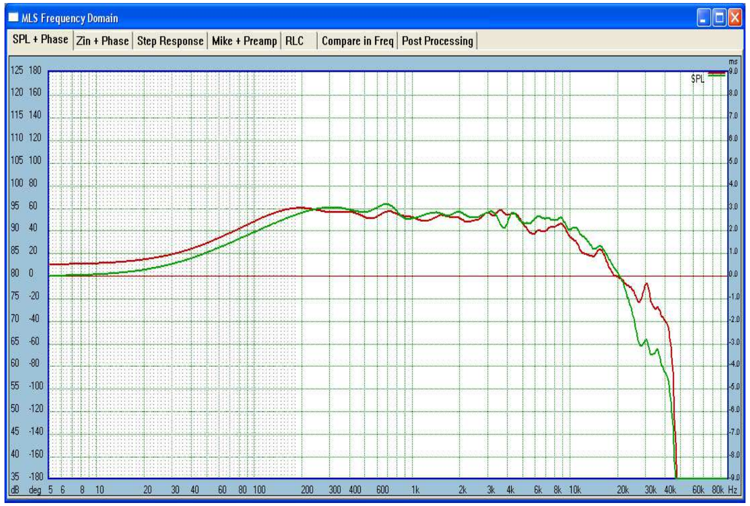


50cm, 100cm, 210cm

WTW loudspeaker with HBT equalization to 30000Hz

+/-15deg

+/-30deg



+/-45deg

+/-60deg

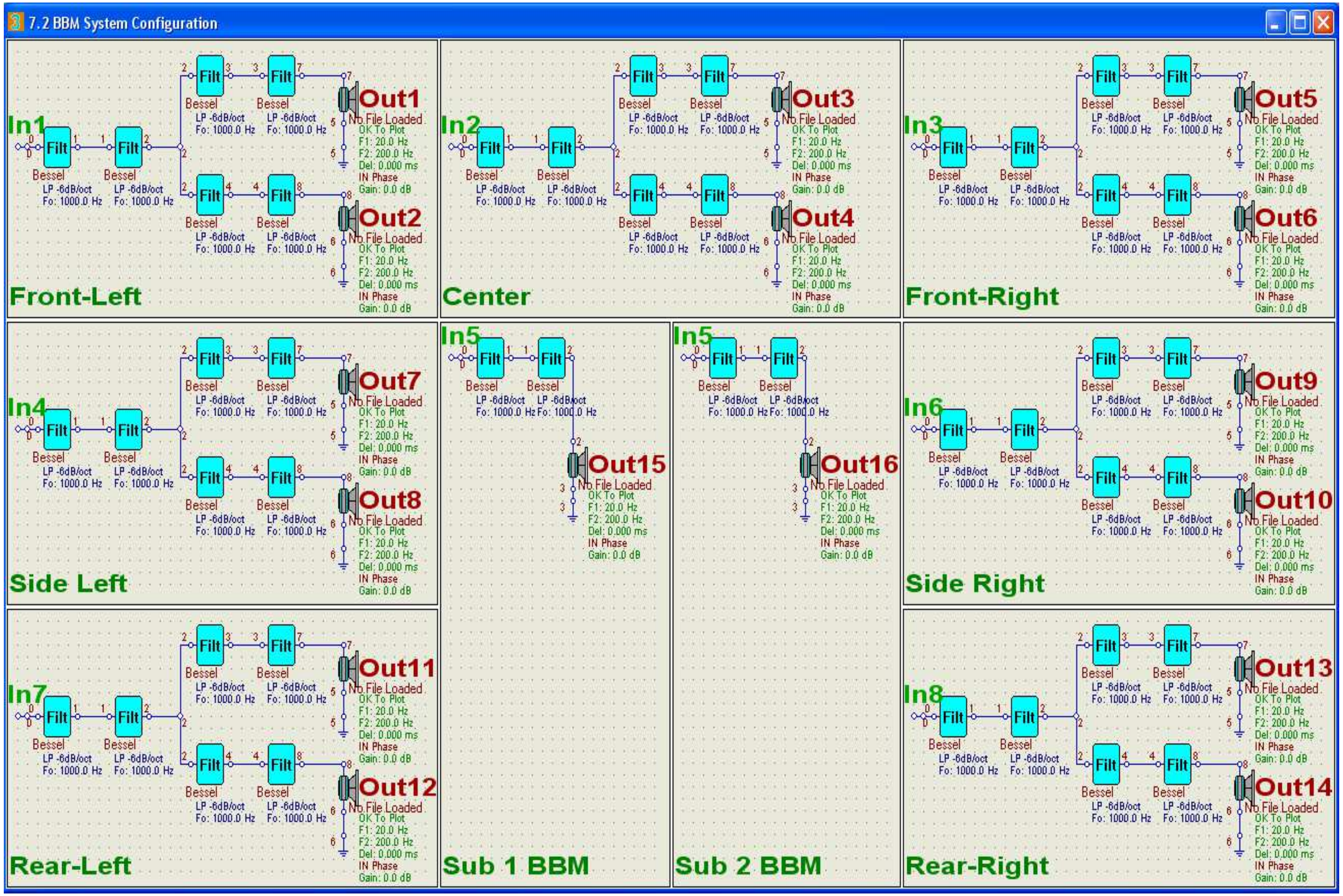
What's New in Ultimate Equalizer V6?

- Implemented CABS http://vbn.aau.dk/files/62729248/LF_sound_field_control.pdf

  Also known as ARAM – Active Room Absorption Module http://www.neumann-kh-line.com/klein-hummel/globals.nsf/resources/o800aram_bda_e_517277_rev_231106.pdf/$File/o800aram_bda_e_517277_rev_231106.pdf

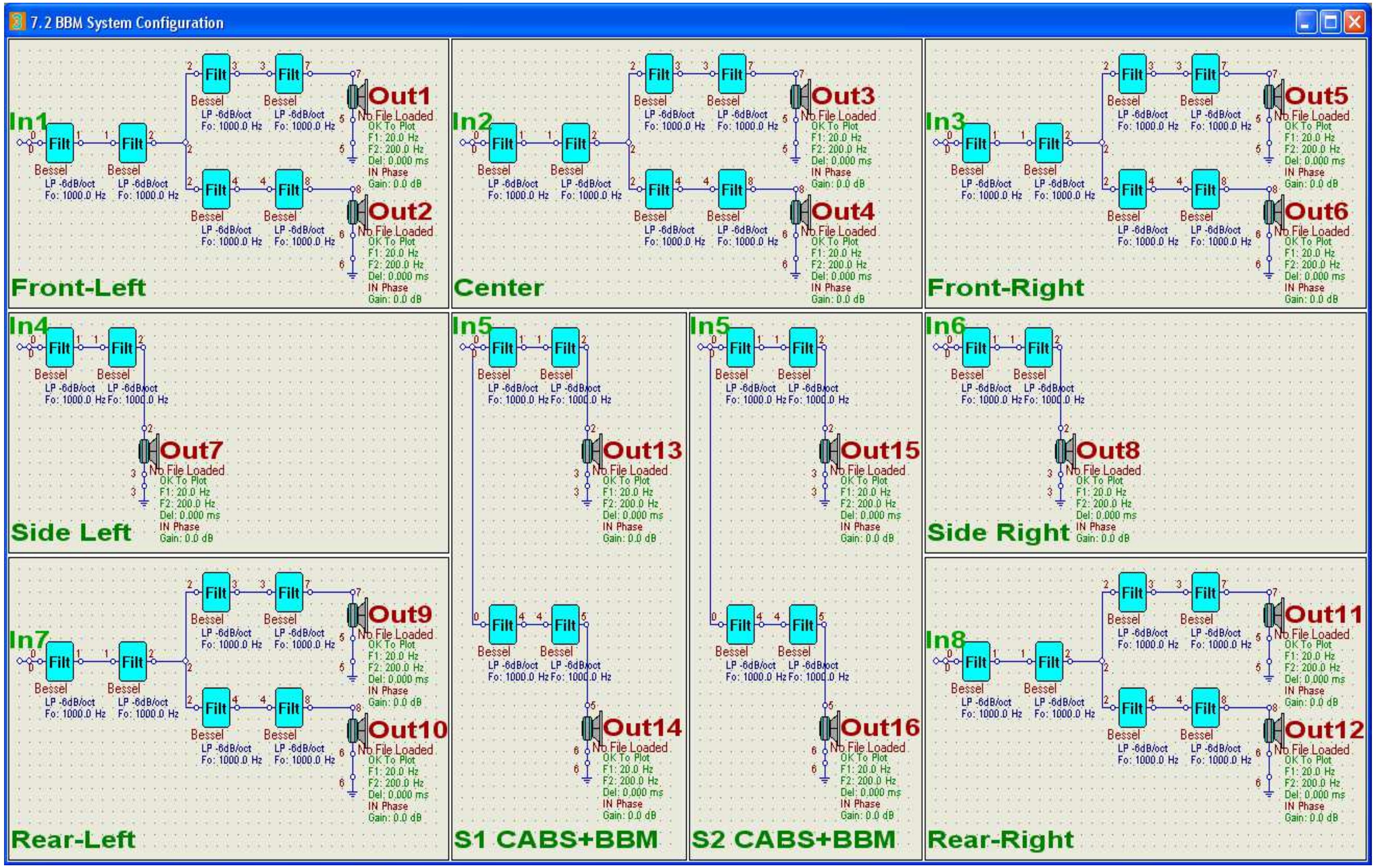
  There is also Convention Paper 7262, "**Multi-Source Room Equalization: Reducing Room Resonances",** John Vanderkooy
- Effectively, UE now offers **two methods of room equalization**, that can be used together in minimum-phase or linear-phase modes: CABS and FIR inverted filtering.
- **16 partition convolution engine** – for longer IRs, therefore better low-frequency resolution. Should reduce latency in Minimum-Phase Mode. The original 8-partition option still available for the lowest latency.
- **Long Channel Delays: 0-168ms delays.** This feature allows for adding long delays to each channel for creating special "echo" effects in QUADRO (or other) configuration.
- **7.1 HT system configuration available**. Also 7.2HT (BBM) and 7.4HT (BBM+CABS) systems available.
- Up to **1.47Hz low-frequency resolution**.
- Supports 16-channel **LynxAES16 digital sound** card and Delta1010LT analogue sound card.
- Runs Windows audio engine in **WASAPI Exclusive Mode.**

7.2 HT system with BBM

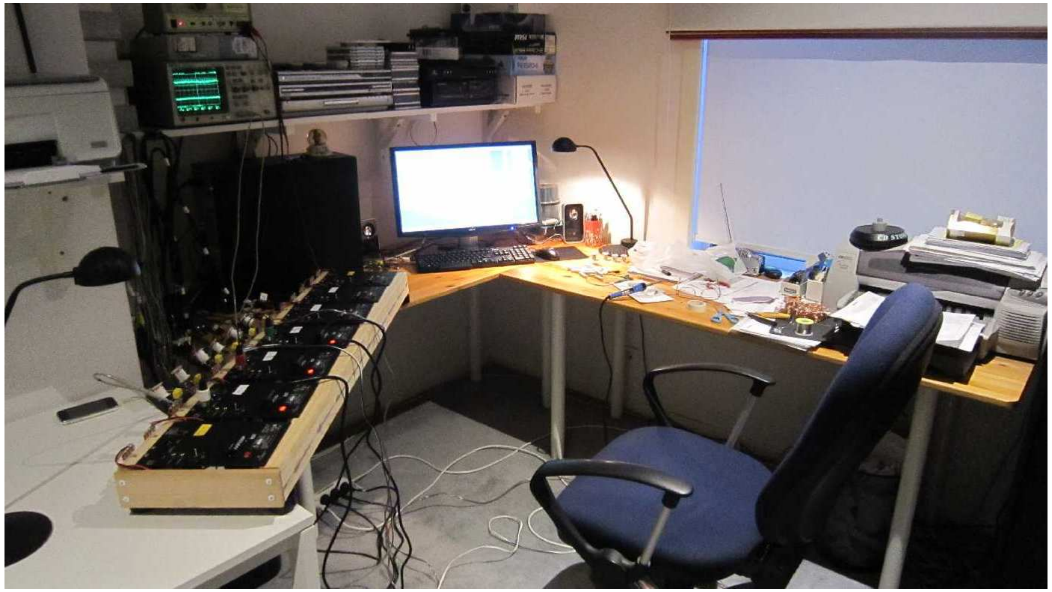


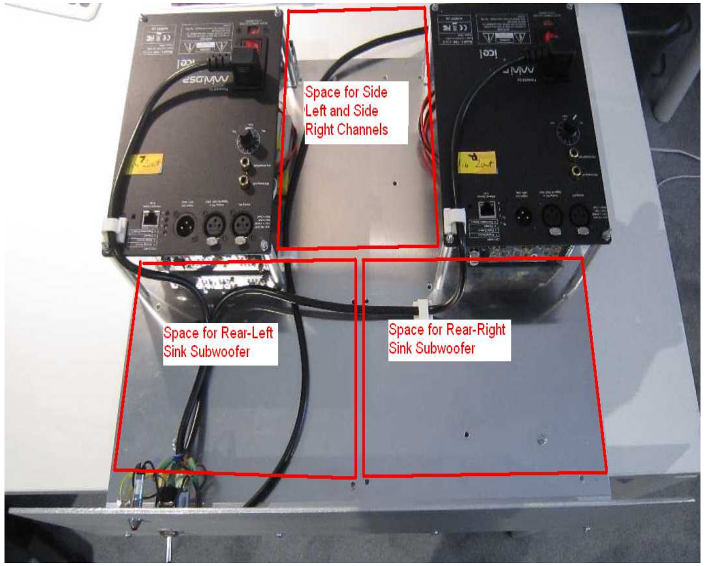
7.4 HT system with BBM + CABS

- Side-Left and Side-Right loudspeakers are wideband drivers eg: Dayton Audio PS220-8 8" Point Source Full-Range Neo Driver, 40Hz-20,000Hz. ~$130.
- http://www.parts-express.com/dayton-audio-ps220-8-8-point-source-full-range-neo-driver--295-346



Testing / assembly PWR-ICE Amplifiers in pictures.





Keele-Horbach Crossovers

http://www.linkwitzlab.com/Horbach-Keele%20Presentation%20Part%202%20V4.pdf

Application of Linear-Phase Digital Crossover Filters to Pair-Wise Symmetric Multi-Way Loudspeakers
Part 2: Control of Beamwidth and Polar Shape

D. B. (DON) KEELE, JR.

Harman/Becker Automotive Systems, Martinsville, Indiana, USA

ULRICH HORBACH

Harman Consumer Group, Northridge, California, USA

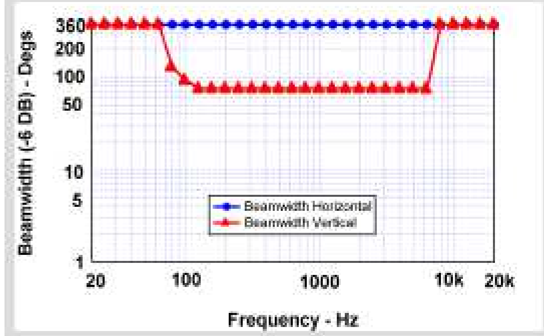
Keele-Horbach Crossovers

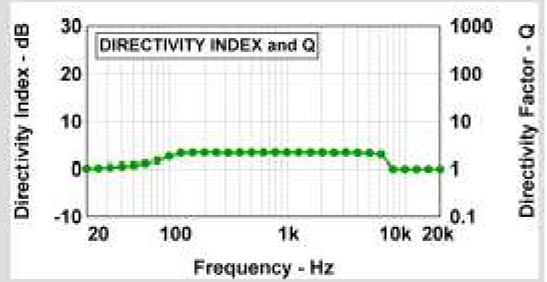
Five-Way Array Design Example

- Desired System Specifications:
  - Vertical beamwidth of 75º with side lobes down at least 17 dB.
  - Constant-beamwidth operating range of 100 Hz on up.
  - Height = 2 m (6.7 ft, 80 in) approximately.
  - Use two 15” sub woofers, two 8” woofers, two 4” lower midranges, two 2” upper midranges, and a single 1” dome tweeter.

Beamwidth and Directivity

(Note: Directivity is defined in a half space where an omni-directional source has directivity of 1 or a directivity index of 0 dB.)

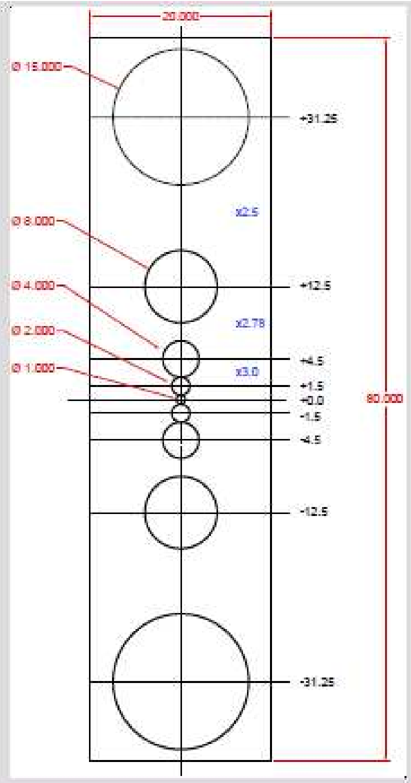




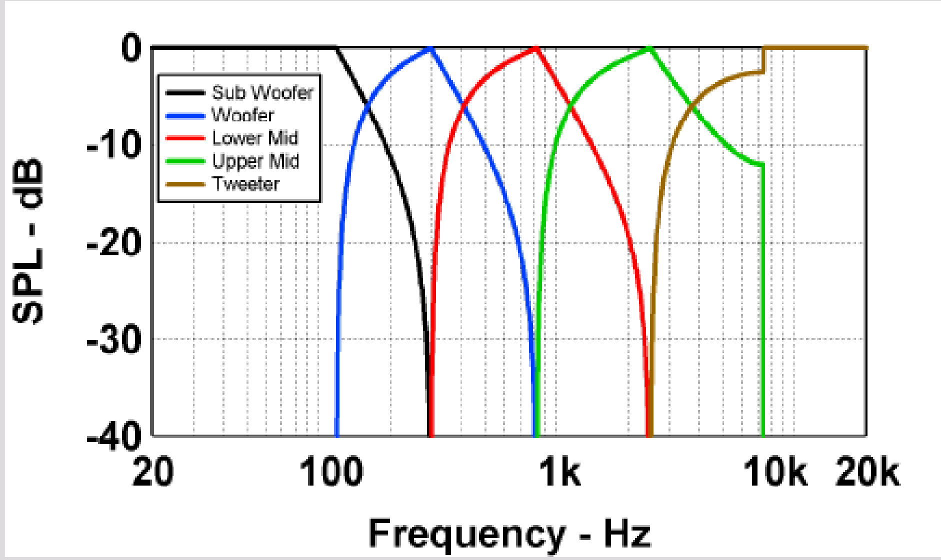
Keele-Horbach Crossovers

Front Panel Design

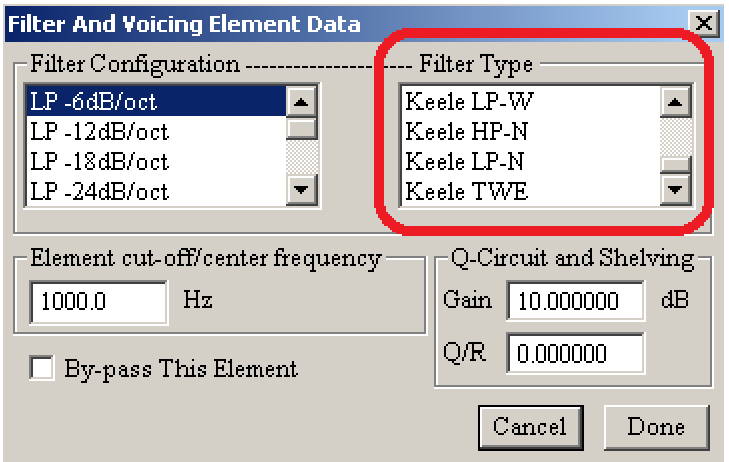
The chosen 75º vertical beamwidth and the lobe requirement dictates a critical driver spacing of about 0.55 wavelength.



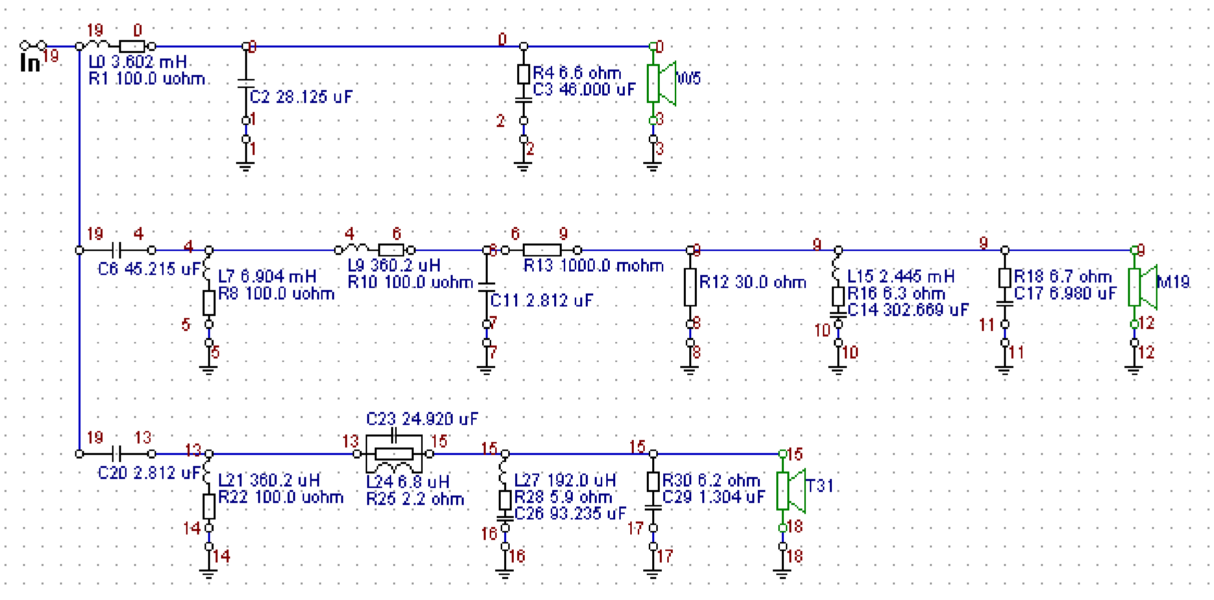
Crossover Frequency Responses

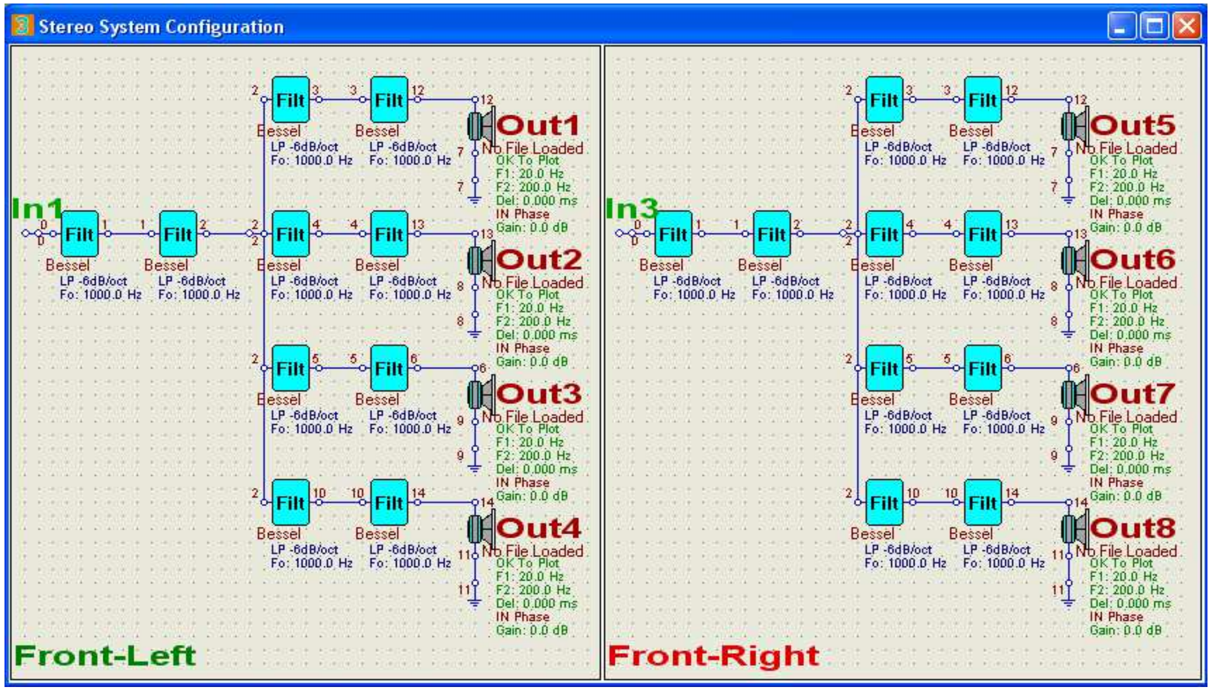


Complete Keele-Horbach Crossover in UE ->



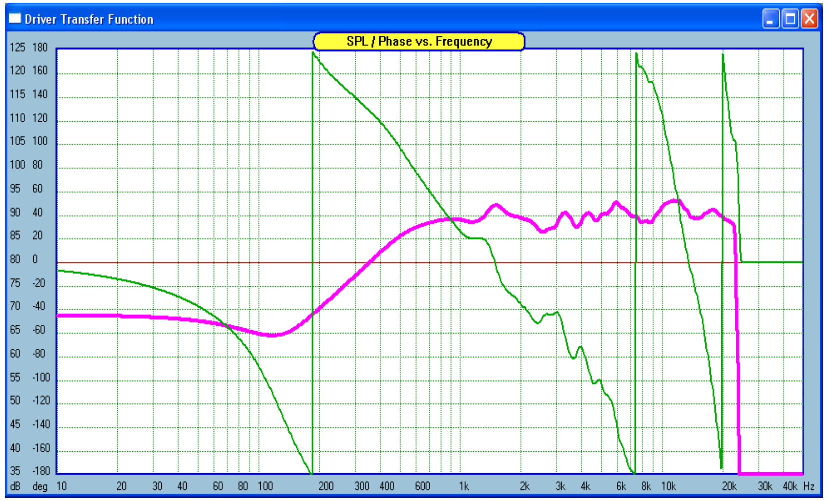
Summary


L0 3.602 mH
R1 100.0 uohm
C2 28.125 uF
R4 6.6 ohm
C3 46.000 uF
W5
C6 45.215 uF
L7 6.904 mH
R8 100.0 uohm
L9 360.2 uH
R10 100.0 uohm
C11 2.812 uF
R13 1000.0 mohm
R12 30.0 ohm
L15 2.445 mH
R16 6.3 ohm
C14 302.669 uF
R18 6.7 ohm
C17 6.980 uF
M19
C23 24.920 uF
C20 2.812 uF
L21 360.2 uH
R22 100.0 uohm
L24 6.8 uH
R25 2.2 ohm
L27 192.0 uH
R28 5.9 ohm
C26 93.235 uF
R30 6.2 ohm
C29 1.304 uF
T31


Stereo System Configuration
In1
Out1
Out2
Out3
Out4
Front-Left
In3
Out5
Out6
Out7
Out8
Front-Right

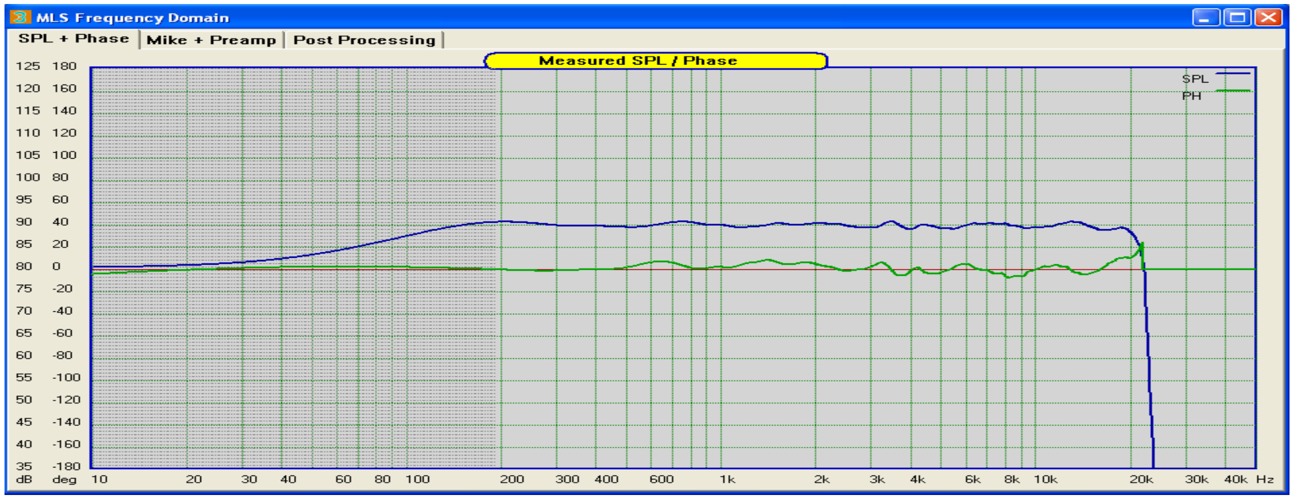
Summary

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

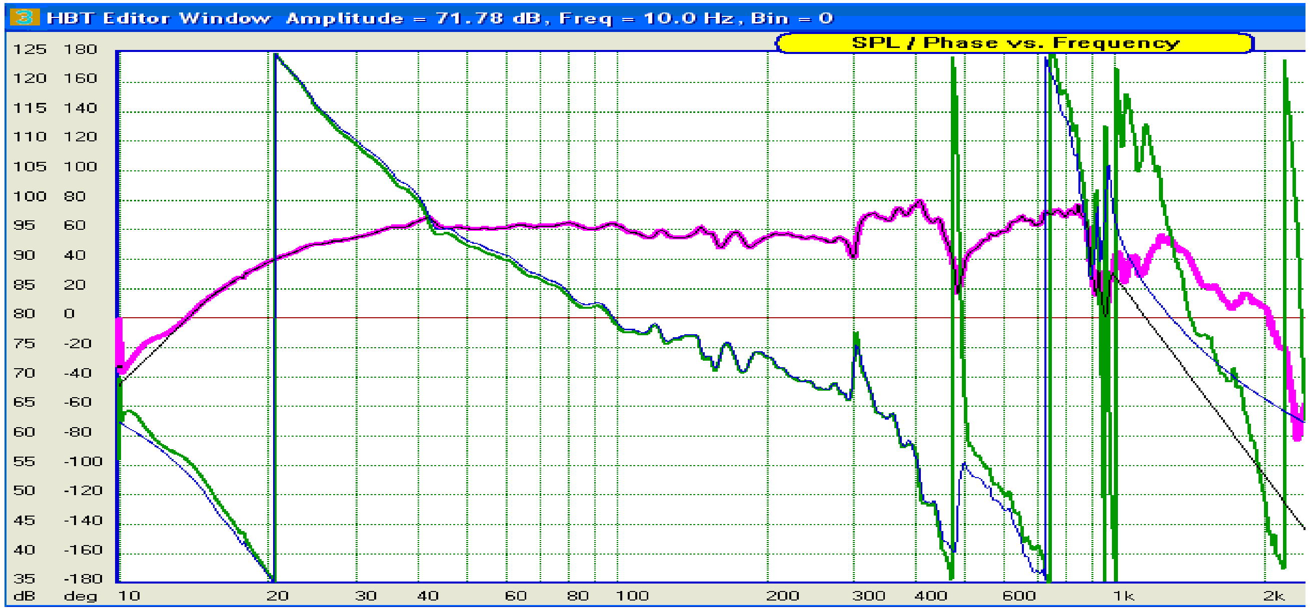


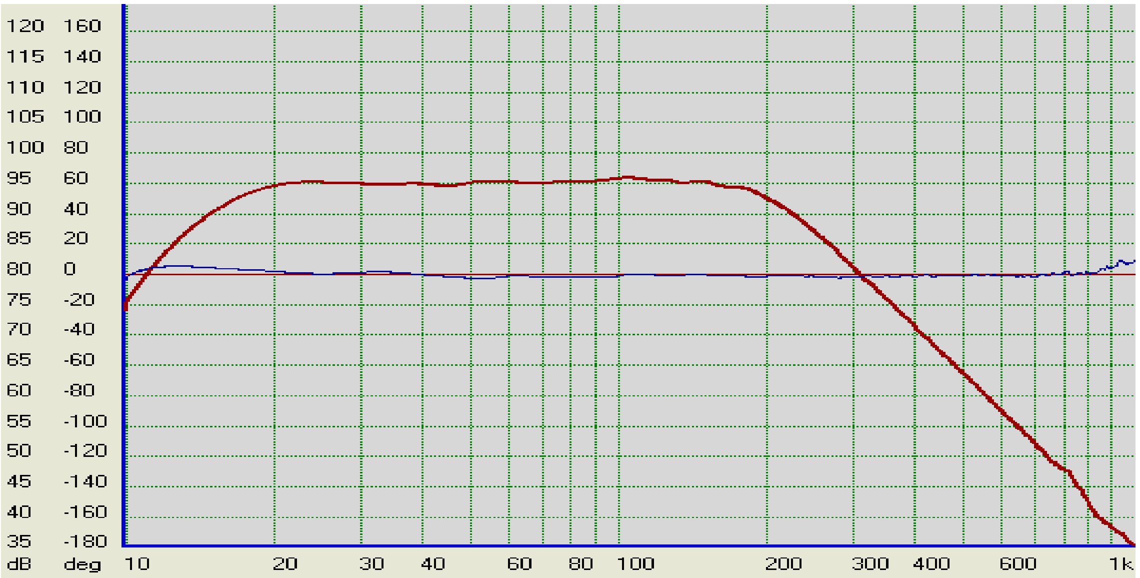
SPL/phase measurements of woofer and tweeter in a 2-way system.

to this level of performance…….

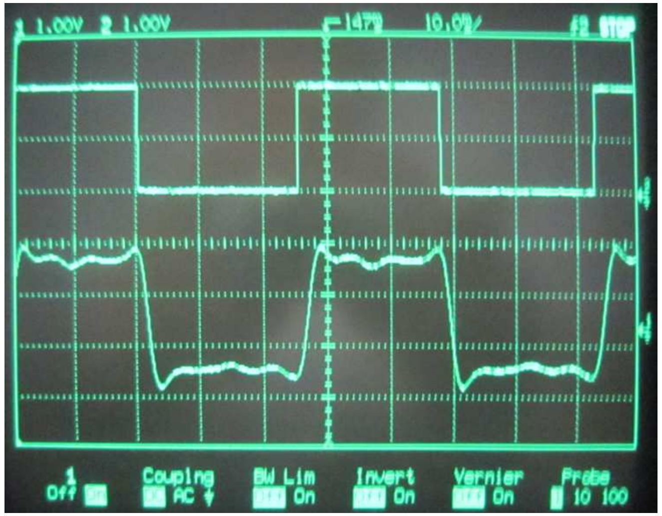


Summary

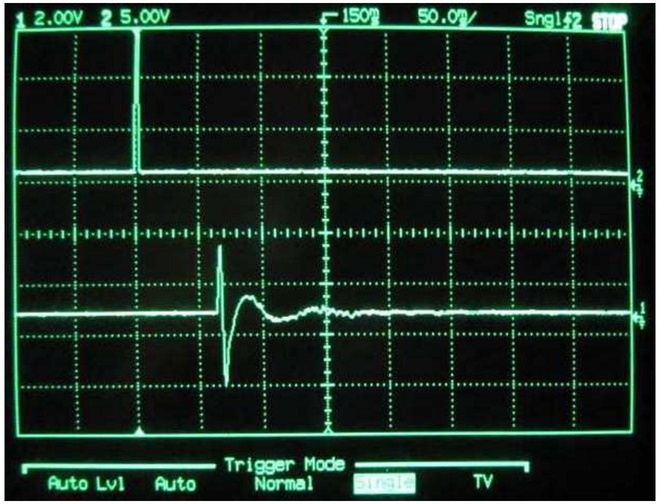

HBT Editor Window Amplitude = 71.78 dB, Freq = 10.0 Hz, Bin = 0
SPL / Phase vs. Frequency
125 180
120 160
115 140
110 120
105 100
100 80
95 60
90 40
85 20
80 0
75 -20
70 -40
65 -60
60 -80
55 -100
50 -120
45 -140
40 -160
35 -180
dB deg 10 20 30 40 60 80 100 200 300 400 600 1k 2k

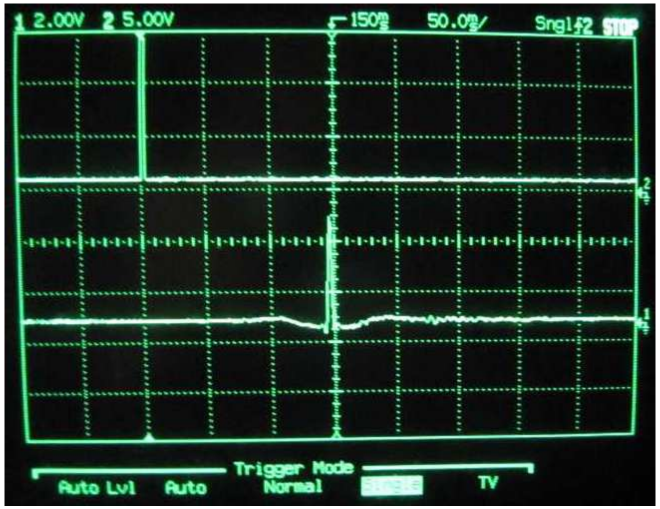

120 160
115 140
110 120
105 100
100 80
95 60
90 40
85 20
80 0
75 -20
70 -40
65 -60
60 -80
55 -100
50 -120
45 -140
40 -160
35 -180
dB deg 10 20 30 40 60 80 100 200 300 400 600 1k

Summary



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





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

The minimum-phase version of the subwoofer has converted the clearly asymmetrical pulse into a much more symmetrical bi-polar pulse with post-ringing

Thank You For Your Attention