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

By Bohdan Raczynski

A fully digital playback chain – from the original sound file, right to the loudspeaker power amplifier, is described in the pages that follow. Focus of this article is not on loudspeaker construction or design. Instead, it's a representative example of what can be easily accomplished with today's digital technology, to supplement and enhance any loudspeaker system of your design choice.

It is important to note, that the system presented here is not just a loudspeaker system. It's a complete ***audio playback system of studio quality***, to be enjoyed in your AV room. Surprisingly, it's realization involves only basic mechanical assembly with plug-and-play components, and can be easily accomplished by a DIY enthusiast.

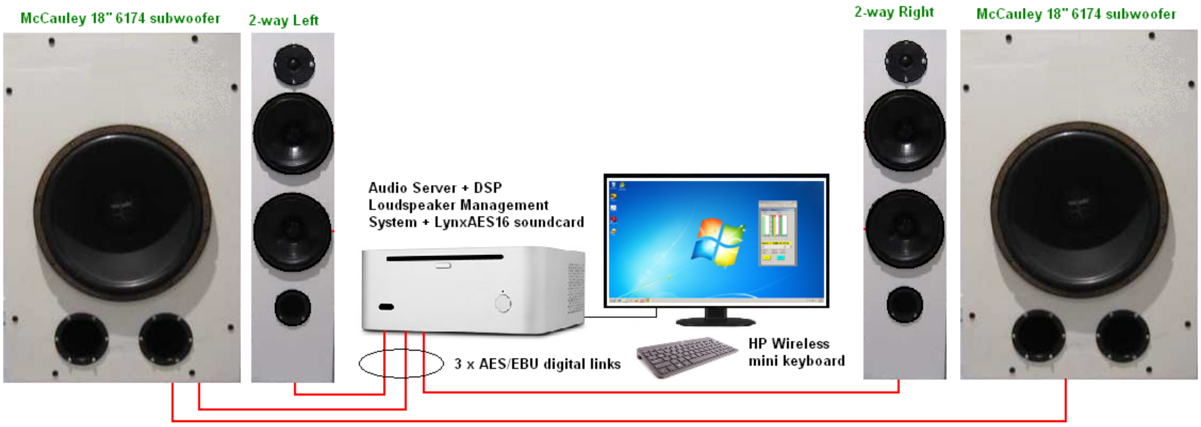


Figure 1. 24bit/96kHz AES/EBU Audio Server with DSP Loudspeaker Management System

You would be forgiven for being surprised at how simple it looks. So, what's at the back of the loudspeakers in this system?.

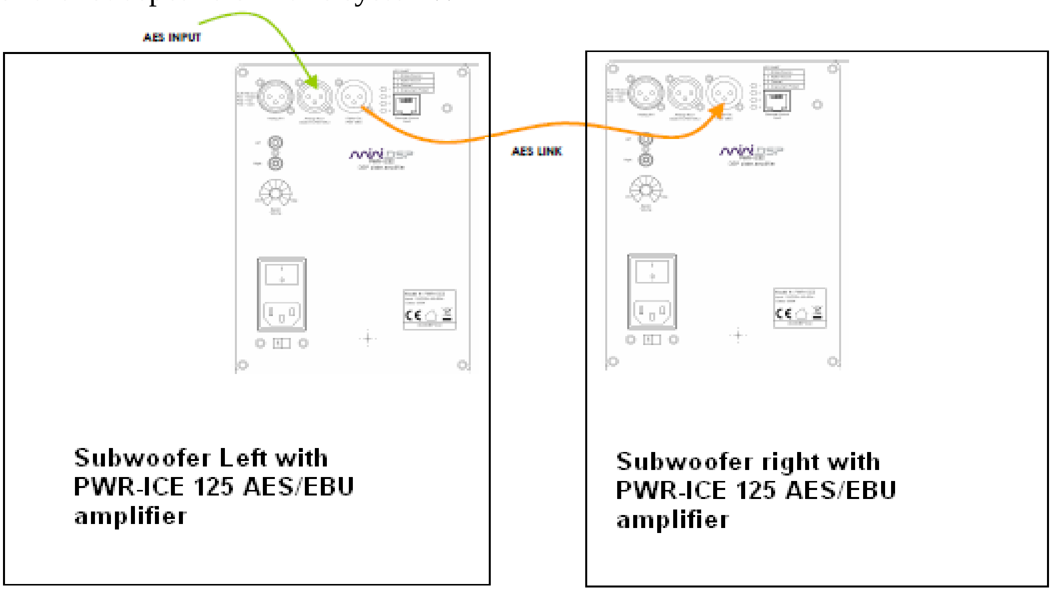


Figure 2. McCauley 6174 drivers in 300 litre vented (20Hz tuning) boxes with PWR-ICE 125 AES/EBU PWM amplifier in each box.

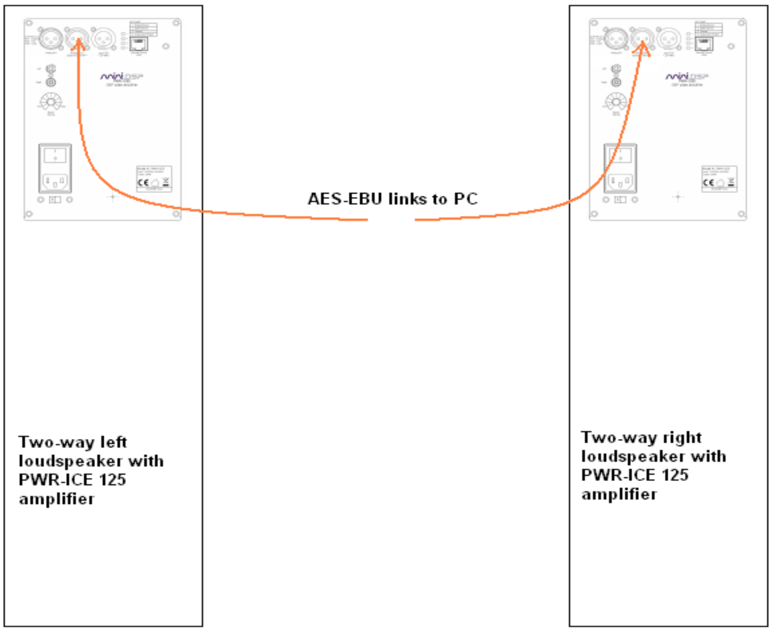


Figure 3. 2x8" woofer drivers + 1" tweeter driver with PWR-ICE 125 AES/EBU PWM amplifier in each 50 litre vented box, tuned to 45Hz.

Here are some of the characteristics of the system:

1. 24bit/96kHz, studio quality processing system.
2. Active system – allows amplifiers to exert maximum control over loudspeaker driver and makes crossover characteristics independent of driver loading.
3. AES/EBU, or SPDIF links between all system components.
4. HBT equalization of individual drivers to achieve flat frequency response.
5. Linear acoustic phase for transient-perfect/image-perfect loudspeaker system.
6. Precise time alignment of acoustic centres.
7. Room EQ for sensible equalization/reduction of most offending room modes
8. Practically unlimited loudspeaker voicing capabilities (all in linear-phase) executed with mathematical precision of a DSP software engine.
9. 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.

Whenever digital files are involved, there is a need for a device capable of handling and processing them – downloading, creating music libraries, playback, and if possible, complete management of your loudspeaker sound system and it's quality. Contemporary PC can do all of these things. And this is a significant benefit, as you can create an exceptionally good audio system in one convenient, easy to use package.

Now, let's take a closer look at all the individual system components.

PC and Audio Codec

There are several case and motherboard options, which can be used for powering this system. One of them is ASUS P6X58D-E motherboard, Socket 1366, which can accommodate Core™ i7 Extreme Edition/Core™ i7 Processors. This is actually about 3-year old motherboard, but is quite suitable for our purpose.

For more details, please visit http://www.asus.com/Motherboards/P6X58DE/

Figure 4. ASUS P6X58D-E motherboard with 2 PCI slots and Realtek® ALC889 8-Channel High Definition Audio CODEC, and coaxial SPDIF output.

The on-board audio codec is easily configurable for 24bit/96kHz processing. This will be accomplished in a few simple steps shown below.

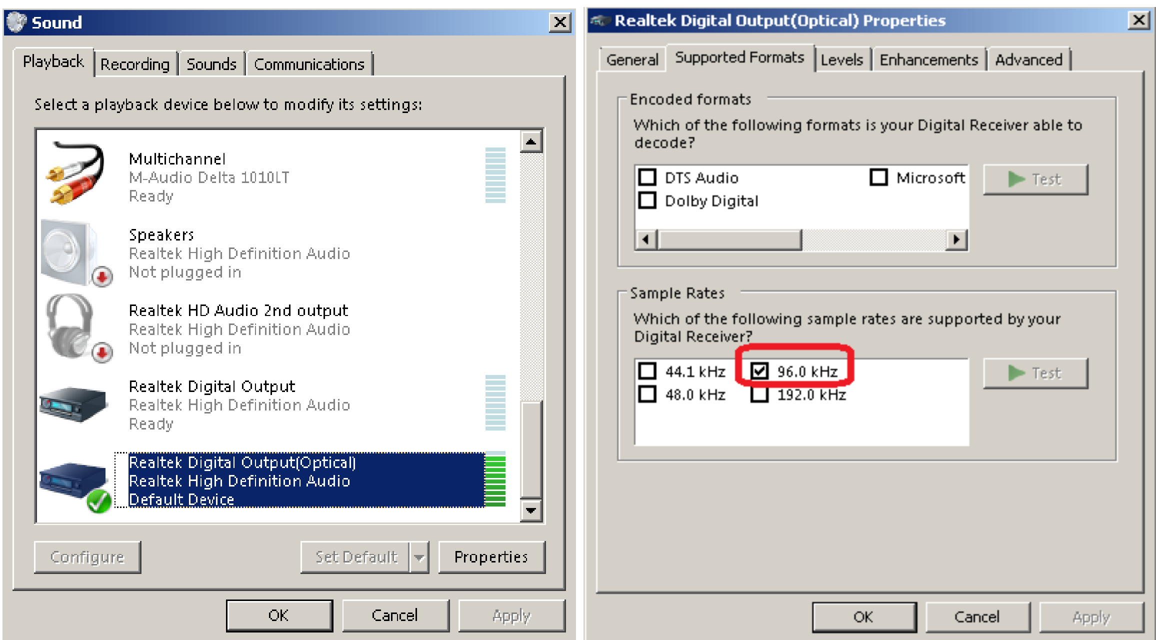


Figure 5. Windows 7 Sound Manager settings required for the ALC889 codec.

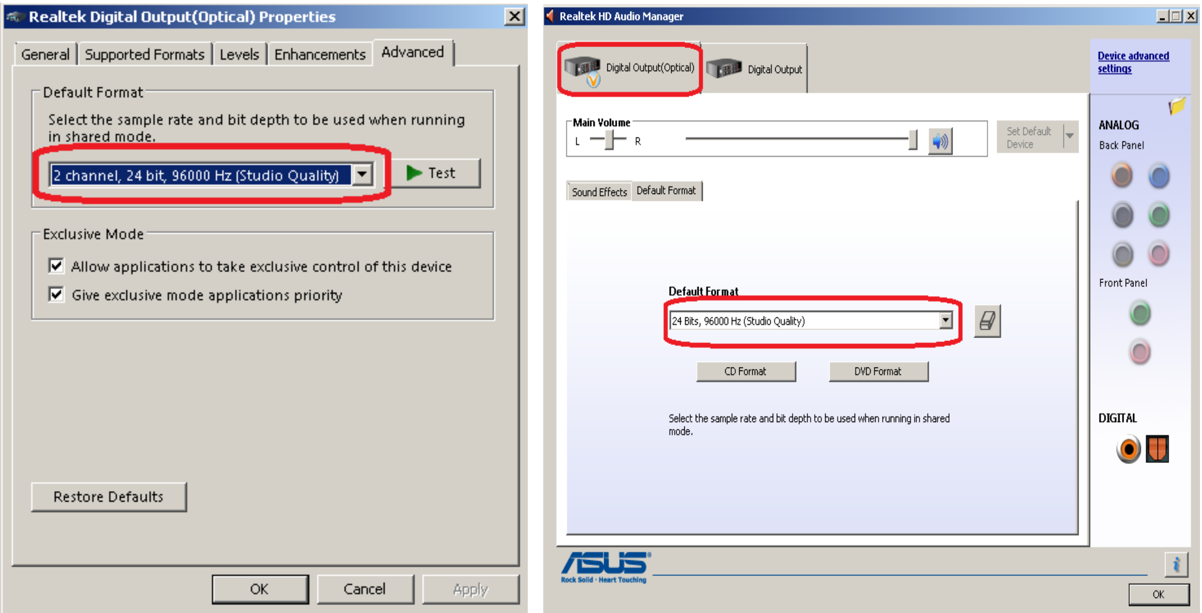


Figure 6. The ASUS Realtek Audio Manager is set to Digital Audio (Optical) and 24bit/96kHz sampling.

Here is a proof, that the Realtek ALC889 HD audio codec is operating at 96kHz and is also capable of up-sampling to 96kHz. I captured 44.1kHz SPDIF from a CD player, and 96kHz SPDIF played from the PC audio server.

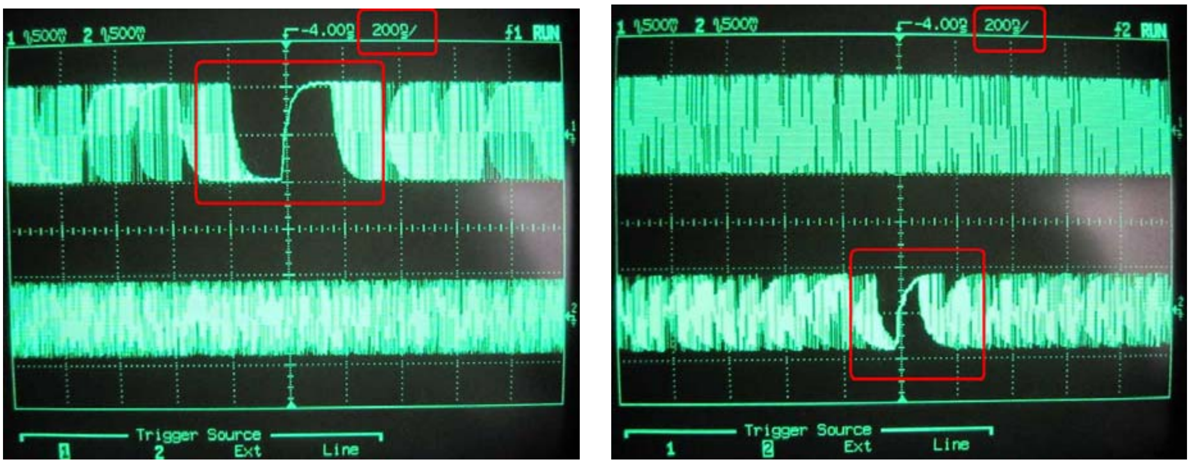


Figure 7. 44.1kHz SPDIF with 200ns time-base   96kHz SPDIF with 200ns time-base.

Later on, we need to connect the SPDIF coaxial RCA output from the motherboard to AES/EBU Input 1 connector of the Lynx AES16 sound card breakout cable. Here is the interconnecting cable, consisting of XLR-to-RCA adaptor barrel, and a 50cm of 75ohm coaxial cable with RCA plugs.

Figure 8. Connecting cable for SPDIF output from PC to LynxAES16 sound card.

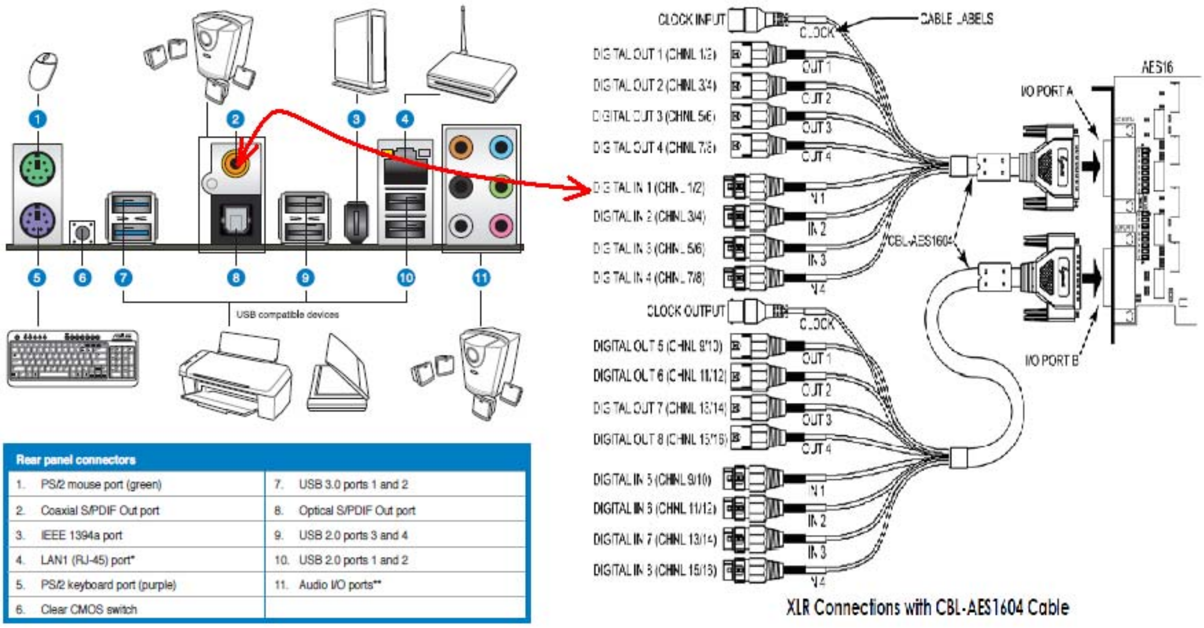


Figure 9. Cable on Figure 8 connects the motherboard and the soundcard as shown above.

At this point, the hardware of the PC is set-up and ready to accept the Lynx AES16 sound card. I would also recommend, that you select your media player (it could be the Windows Media Player incorporated in Windows7 operating system) and make sure, you have some music files to play. The Realtek Digital Output (Optical) has to be nominated In Windows Sound Manager as the “Default” player for the media player.

The LynxAES16 PCI soundcard

Figure 10. The Lynx AES16 AES/EBU PCI digital soundcard.

Having installed the Lynx AES16 sound **card as per instructions included with the card**, we can proceed to Windows Sound Manager and set the “Speakers” device to 24bit/96kHz.

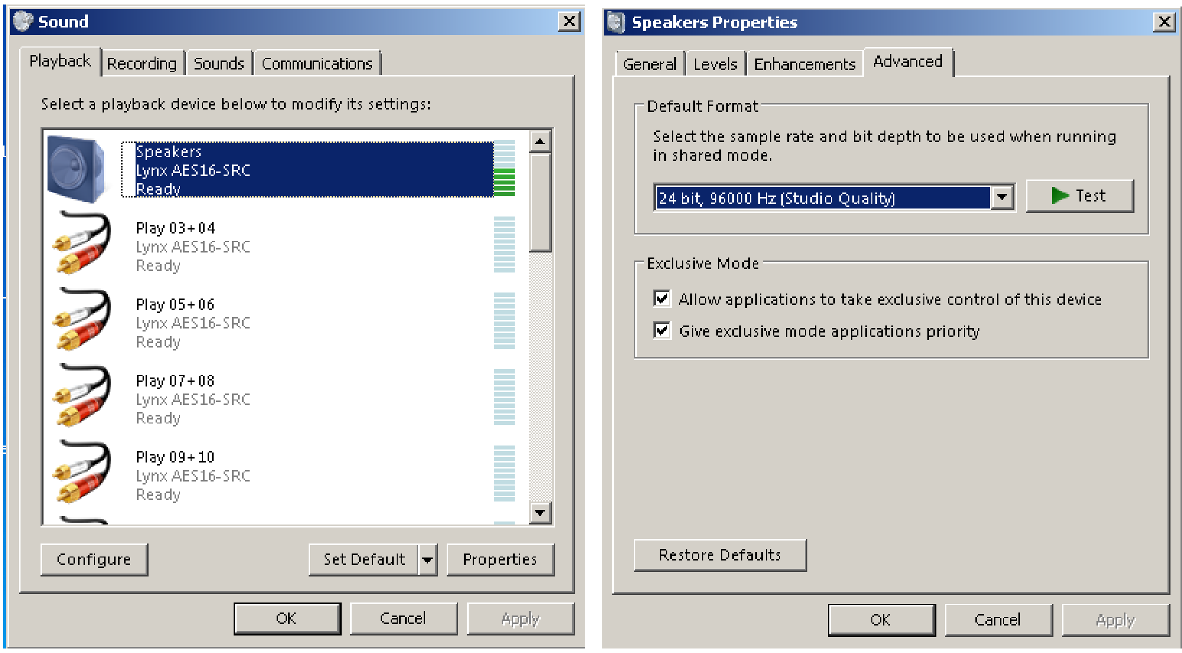


Figure 11. Setting the LynxAES16 “Speakers” device in Windows Sound Manager as 24bit/96kHz device.

In the next step, please open Lynx Mixer program, and navigate to “Play 1 Multi-channel Mode”, and set it to 7.1. This is an important step, as it commands the driver to time-align the first 8 AES/EBU channels of the sound card.

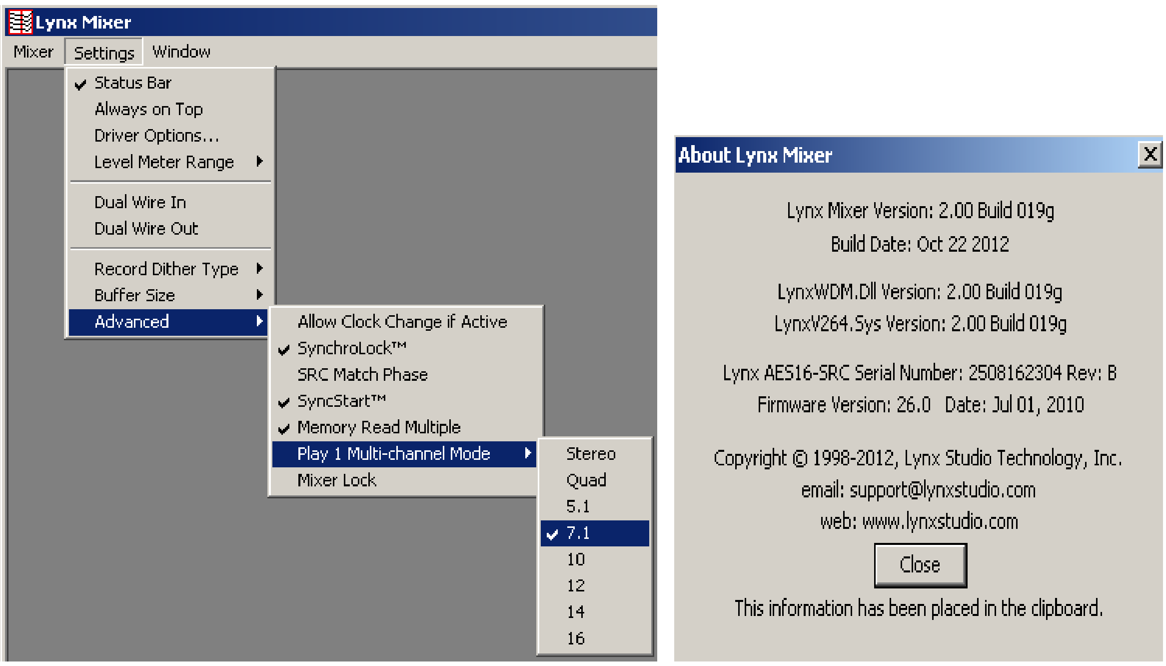


Figure 12. Setting the LynxAES16 for time-aligned, 8-channel output.

Important Note.

The LynxAES16 sound card arrived in the box with Firmware version 23. There is a newer Firmware (v26) available from Lynx website. You need to upgrade the Firmware before installing the latest driver, also available from the same website – see Figure 12.

Now, could be a good time to check if what we have done so far, actually works. Assuming your motherboard is connected to the Lynx sound card as described, you can start media player and if everything is OK, you should see the following information, 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.

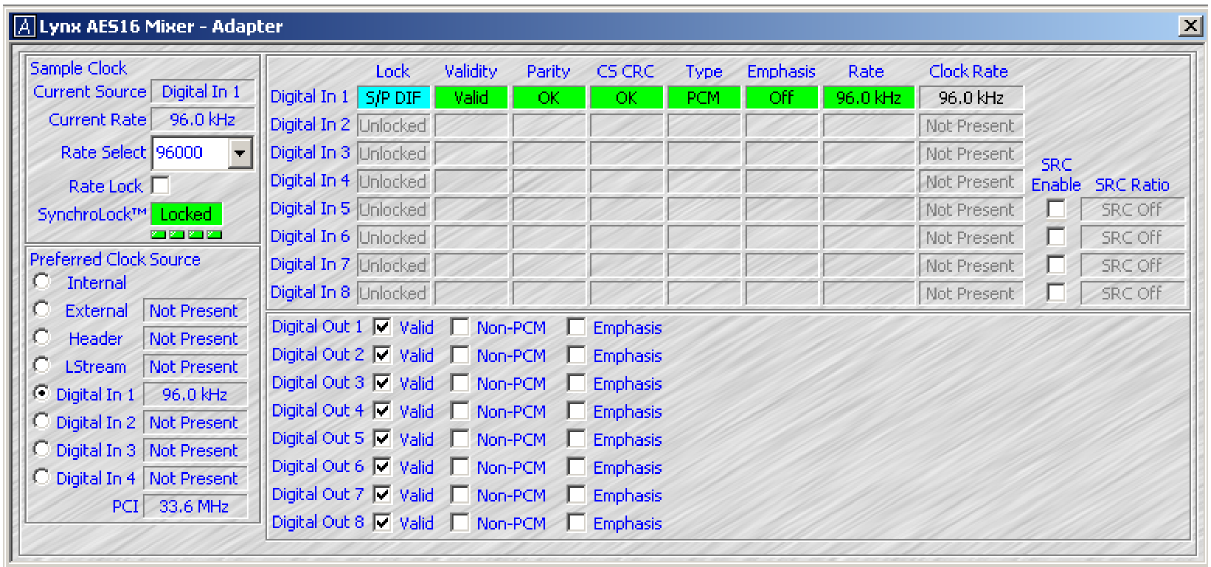


Figure 13. Information about input signal is displayed by the Lynx Mixer.

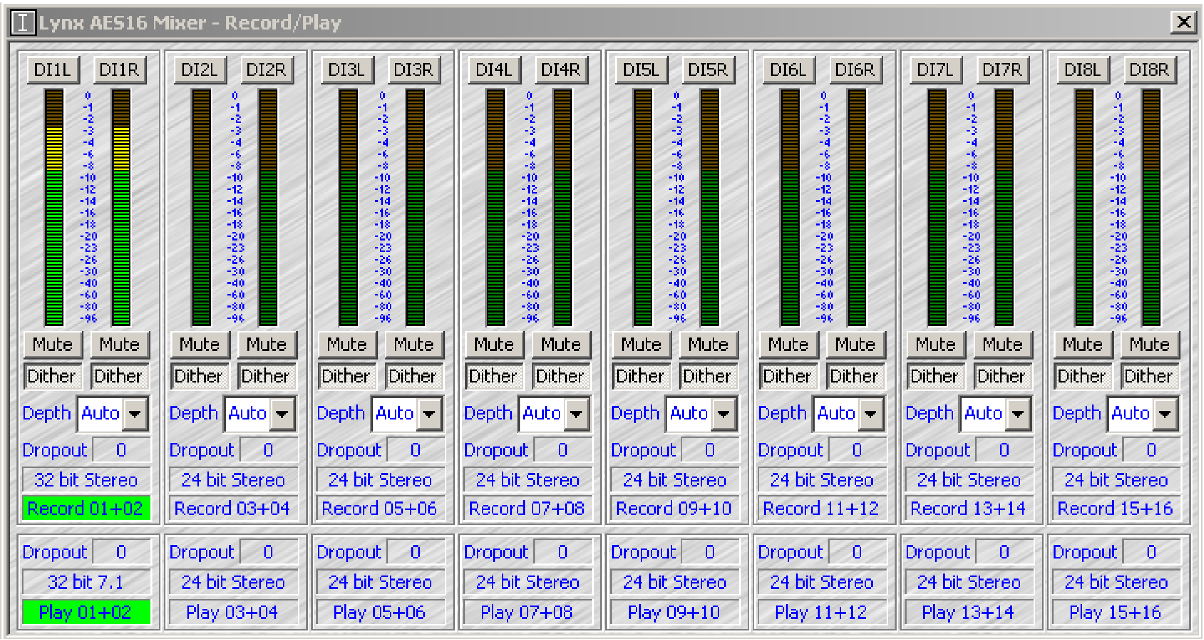


Figure 14. Presence of audio data is confirmed by the “Record/Play” screen of Lynx Mixer.

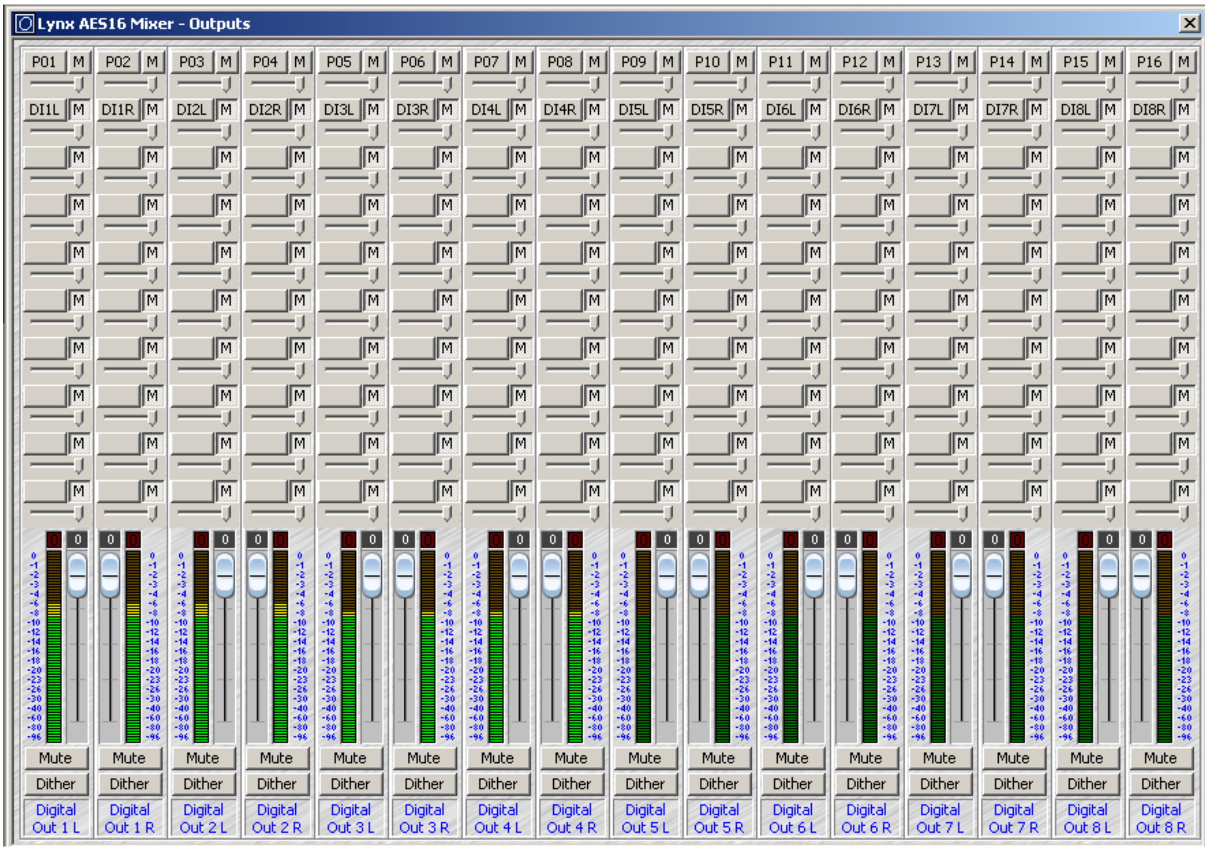
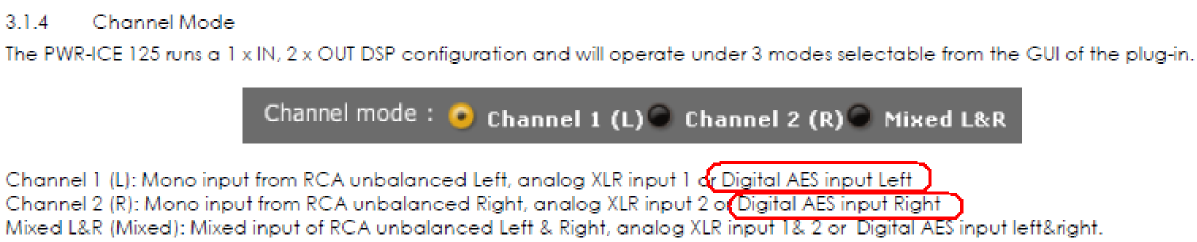


Figure 15. Presence of output audio data is confirmed by the "Outputs" screen of Lynx Mixer

Setting up PWR-ICE125 Amplifiers

There are three distinct modes of operation for the PWR-ICE125 amplifiers:

1. AES/EBU "Two-in, two-out" mode for the two-way loudspeakers. Amplifiers in single-ended mode.
2. AES/EBU "Left AES" mode for left subwoofer. Amplifier in bridged-mode.
3. AES/EBU "Right AES" mode for right subwoofer. Amplifier in bridged-mode.


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.

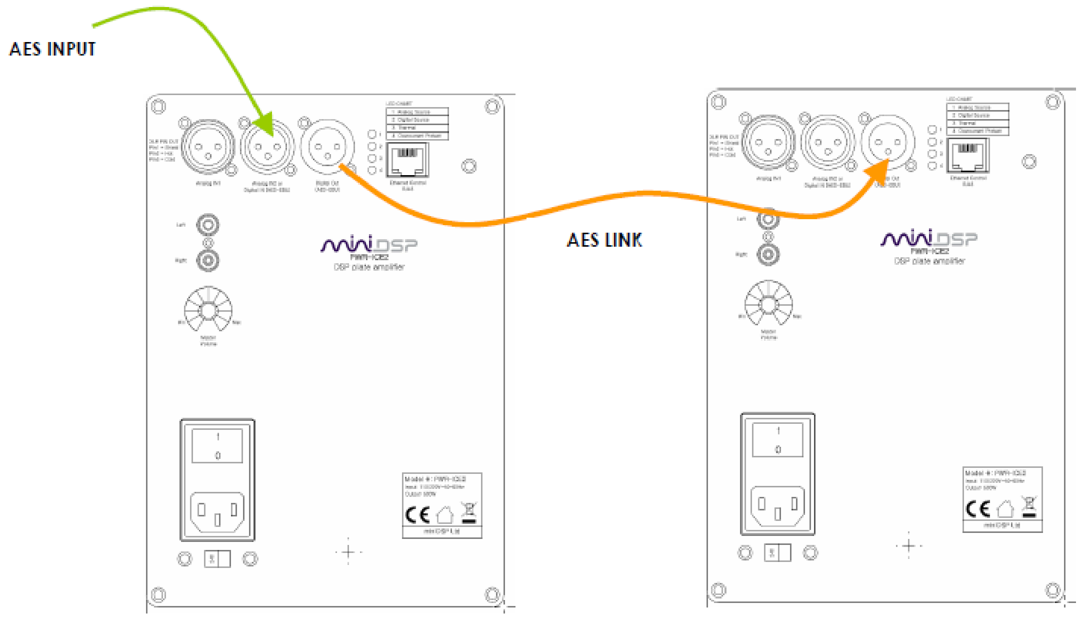


Figure 16. Interconnections between two subwoofers.

You may need XLR cables to interconnect the two subwoofers: 3 Pin XLR Type Plug to 3 Pin XLR Type Socket – 6mt

http://www.jaycar.com.au/productView.asp?ID=WA7042&form=CAT2&SUBCATID=1013#4

Figure 17. XLR extensions cable

PWM Amplifier Phase Response

Phase response of the PWM amplifier (as observable below in the module specification data), is not flat.

Frequency Response (SE-mode)

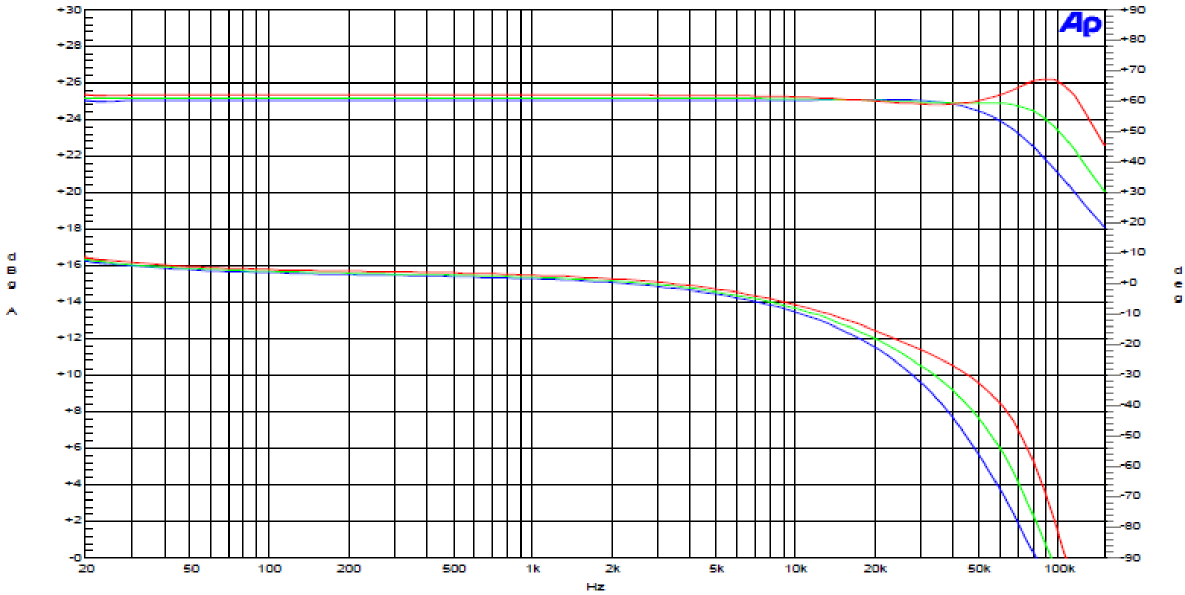


Figure 3: Frequency response in 4Ω (blue), 8Ω (green) and open load (red). Top – amplitude. Bottom – phase.

If your 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.

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.

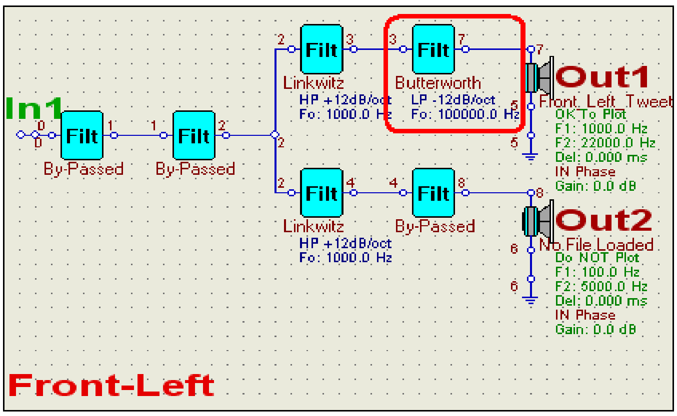


Figure 18. Compensating for PWM amplifier phase response.

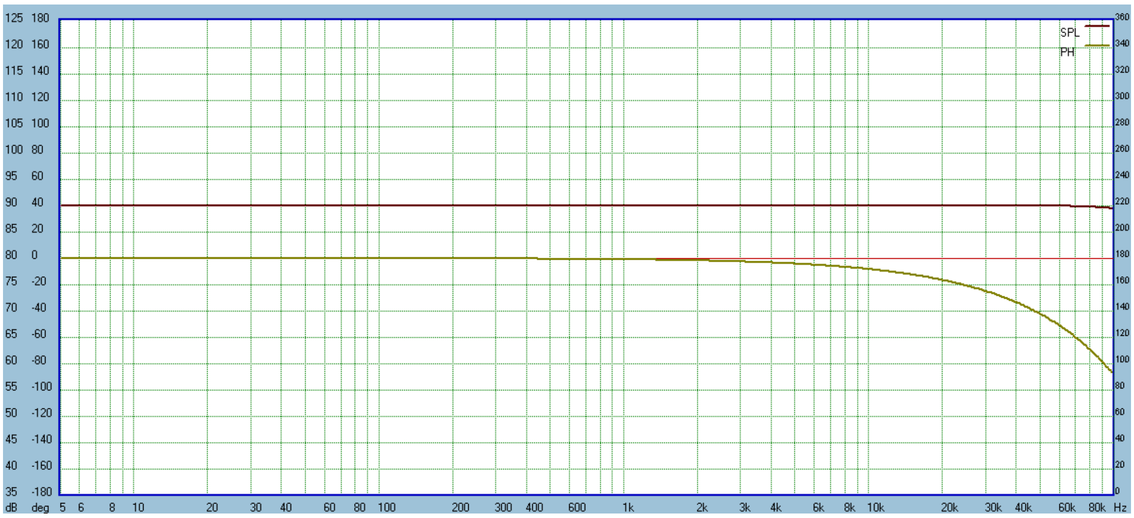


Figure 19. Model of the phase response for 8ohm load: -35deg@40kHz and -18deg@20kHz

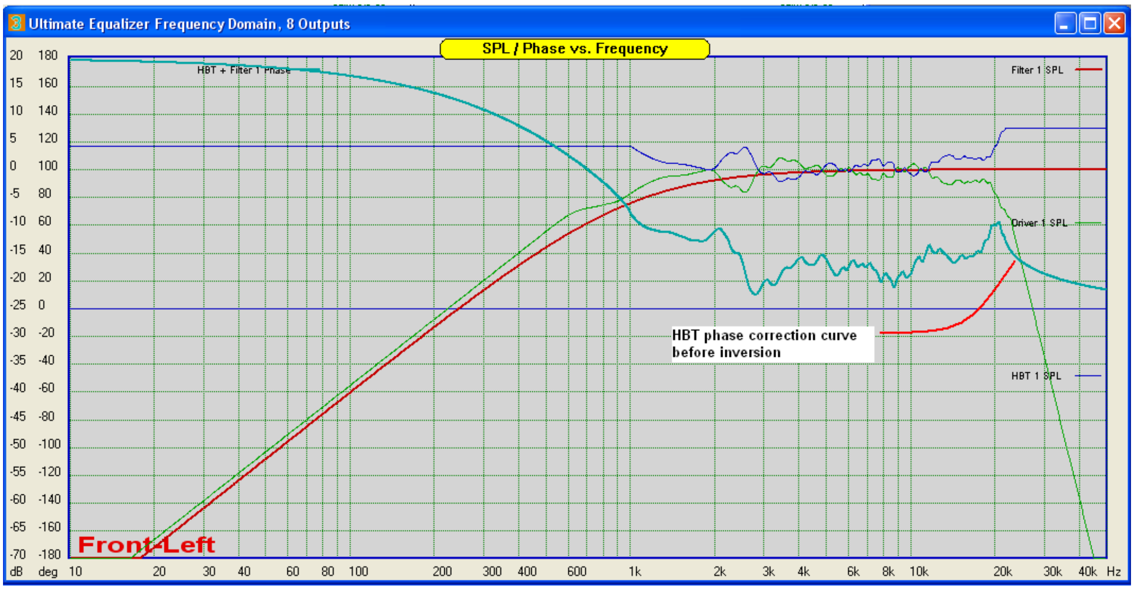


Figure 20. HBT phase response without the extra phase roll-off.

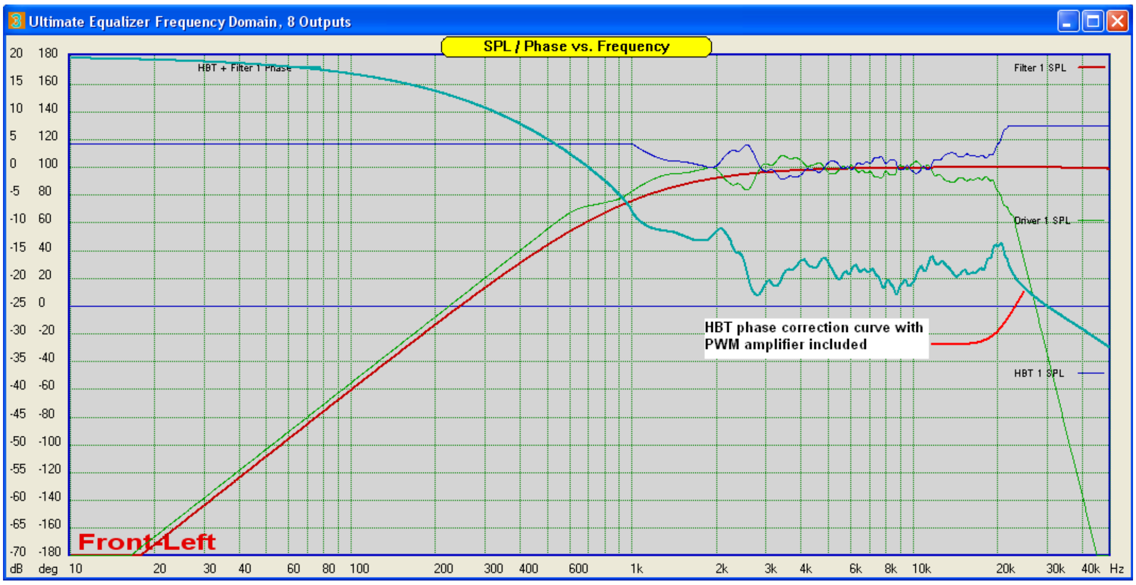


Figure 21. HBT phase response with the extra phase roll-off.

Next, is the issue of 500kHz PWM amplifier switching component still being present on the output. Without any additional filtering there will be up to 4Vpp of 500kHz present in the output signal coming from this amplifier. If this is of a concern to you, then simple LC lo-pass filter with 25uH coil and 150nF capacitor, shown below, 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 in PWM output filtering can be found in:
http://www.ti.com/lit/an/sloa023/sloa023.pdf
http://pdfserv.maximintegrated.com/en/an/AN624.pdf

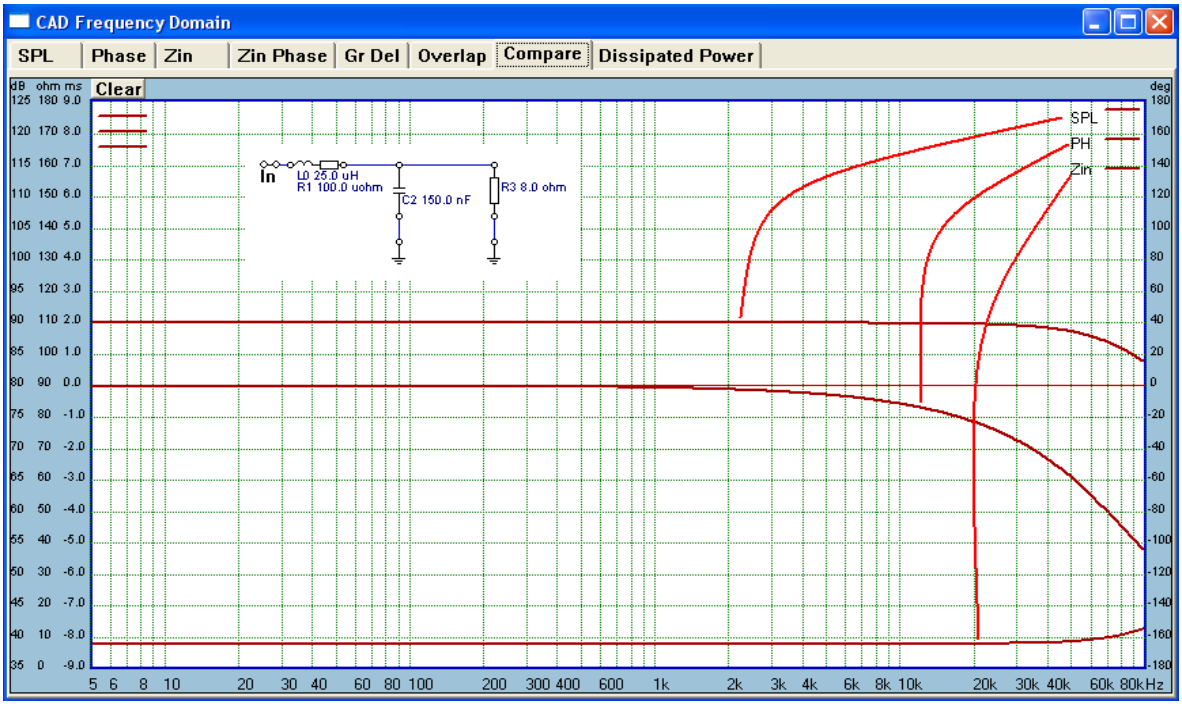


Figure 22. Low-pass LC filter to improve 500kHz suppression.

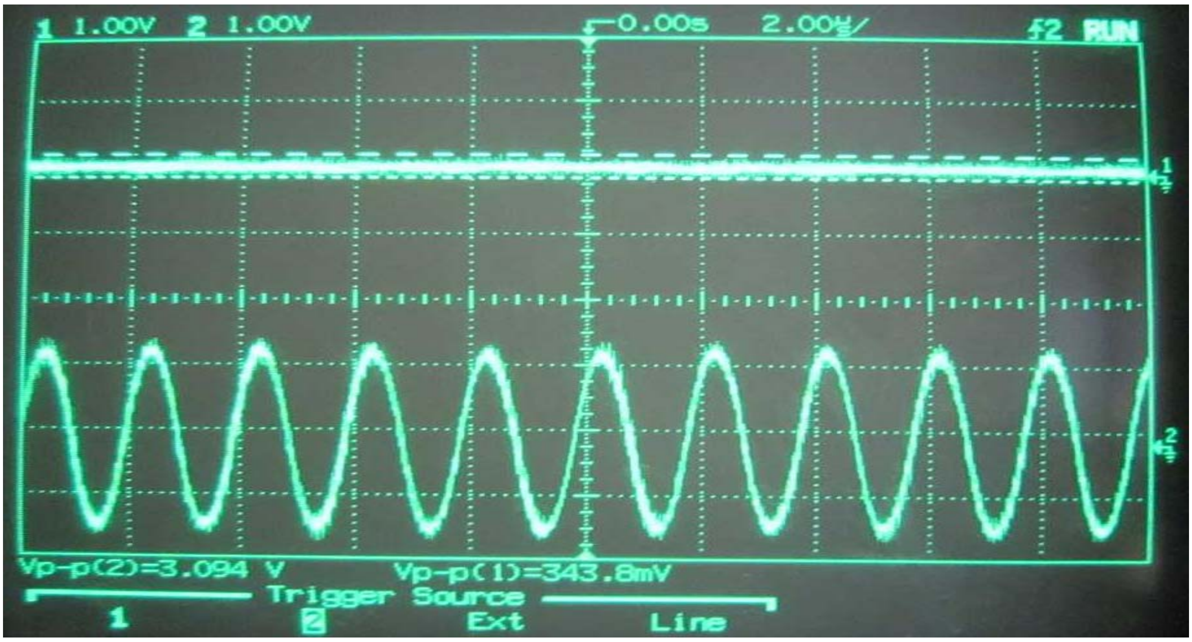


Figure 23. Bottom: 500kHz switching component before filter, Top: after filter – 1V/div

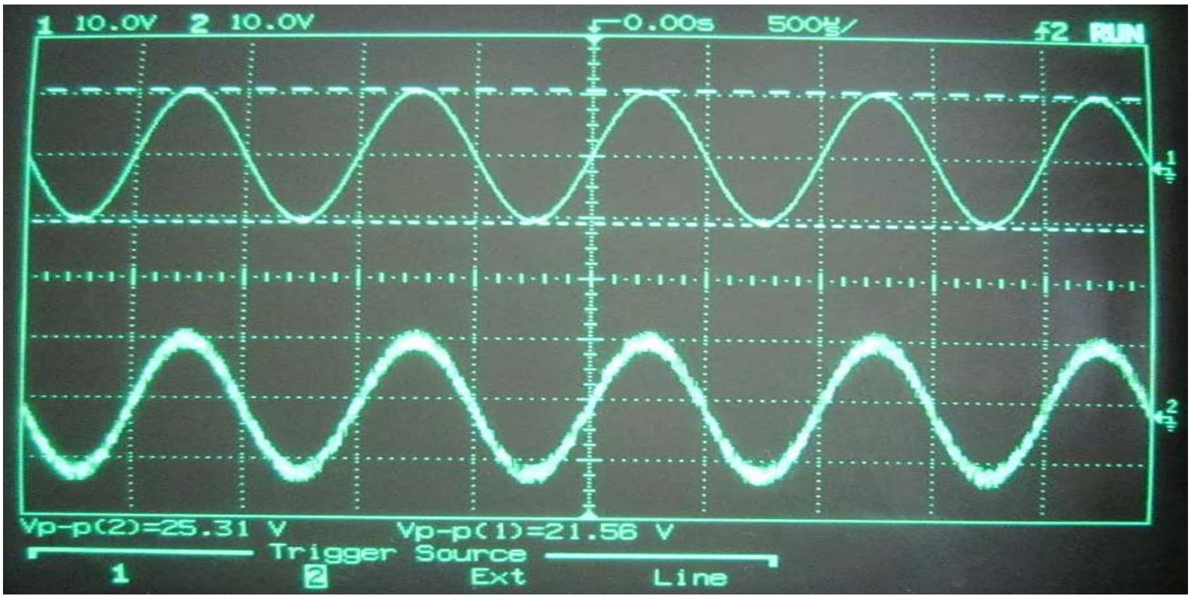


Figure 24. Bottom: 1kHz tone + 500kHz component before filter, Top: after filter – 10V/div

See also: http://pdf.datasheetarchive.com/indexerfiles/Datasheet-082/DASF0050292.pdf

Loudspeaker Management System – UE6 DSP engine

Imagine, that you can have a large selection of filter configurations and types, and the ability to cascade them any way you like. On the top of this, you are able to 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.

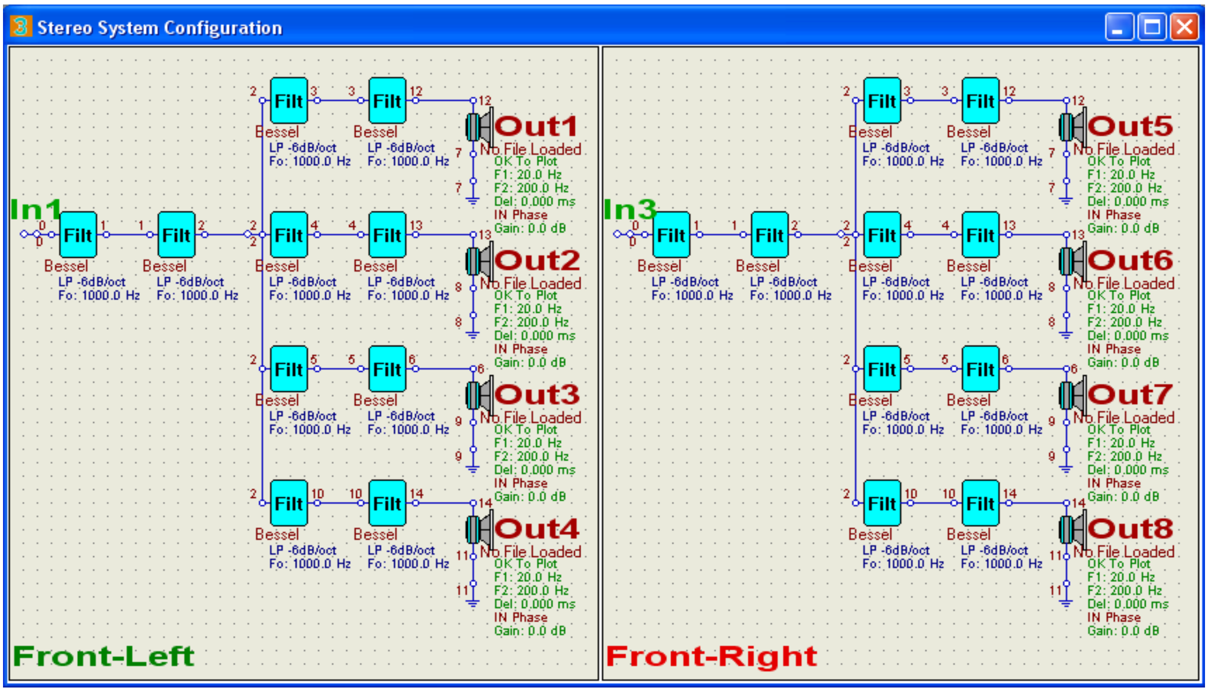


Figure 25. Ultimate Equalizer System Design screen.

The “tray” is shown below. To keep things simple, there are only three active elements, using which you can built the entire crossover.

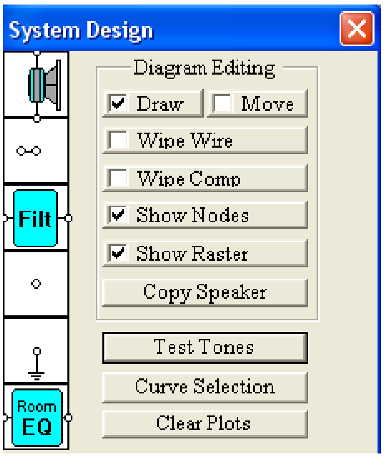


Figure 26. Schematic pick-and-place component tray.

First, the “Loudspeaker” element – incorporates measured driver file name, equalization frequency range, channel delay, channel phasing, channel gain and sound card output port assignment data. Equalization of the loudspeaker is implemented as inverted Hilbert-Bode Transfrom (HBT) function. It is very effective in equalizing the measured SPL to a flat line, and equalizing the measured phase to straight, line as well. Final phase equalization to a flat line is performed by running the DSP-PC in “Linear-Phase Mode”. This way the acoustical transfer functions of your loudspeaker (or frequency and phase responses if you like) are both flat lines. Such loudspeaker will reproduce square waves at the listening location.

The equalization process incorporated in Ultimate Equalizer was described in details in ***Audio Xpress June 2010,*** therefore, it will not be presented in details here. However, as a remainder, please consider Figure 27 below. Depicted here are the following curves:

Green curve – SPL of the measured driver
Dark Blue curve – SPL correction curve, calculated by inverted HBT
Red curve – filter transfer function – 2nd order, 1000Hz, Butterworth.
Pink curve – final equalized SPL. Equalization frequency range 60Hz-6000Hz.
Grey curve – final equalized phase. Equalization frequency range as above.

The final, flat phase response is accomplished by employing time-inverted filtering method. This example shows the crossover and equalization design for the woofer drivers in the Front-Left speaker system.

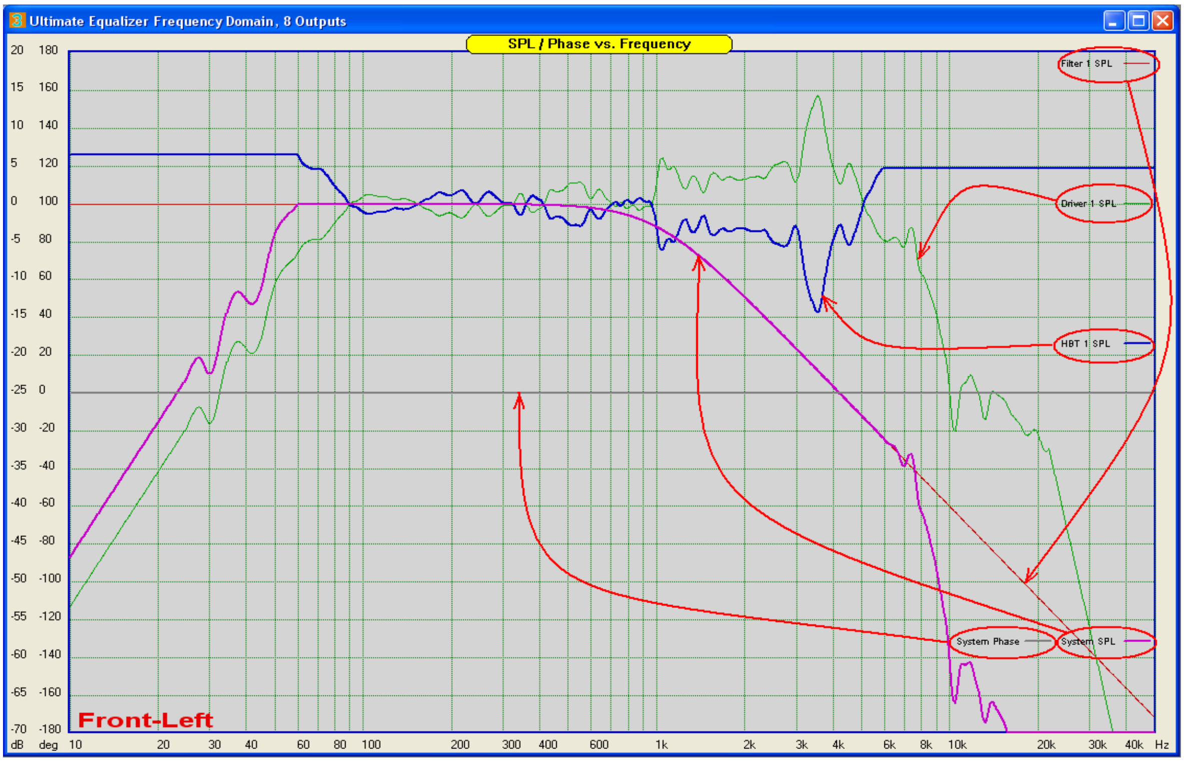


Figure 27. Example of woofer filtering and equalization design.

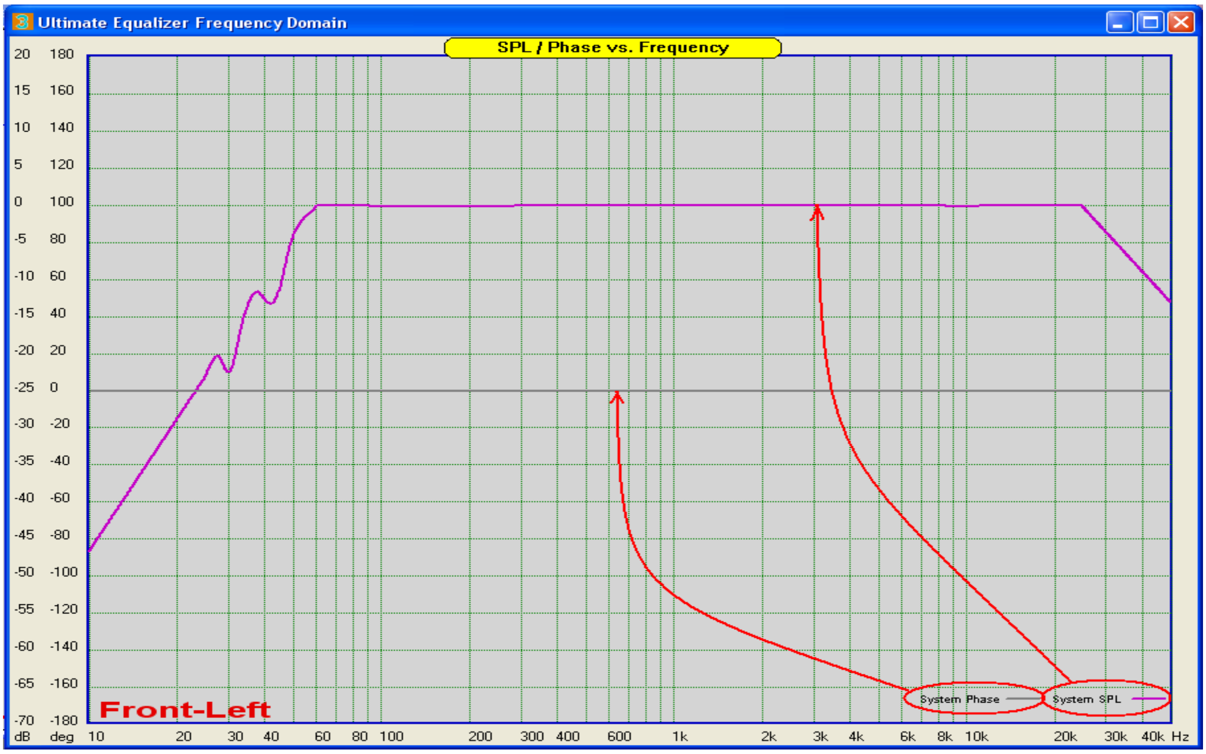


Figure 28. Front-Left loudspeaker system, fully equalized SPL (pink curve) and phase (grey curve) transfer functions.

The process described above, needs to be applied to the right loudspeaker as well. Having finished, you will be left with a superbly performing loudspeakers, having frequency and phase responses flat and possibly extended by an octave into low and high-ends of the old operating frequency range.

In the next step, you may want to audition each loudspeaker individually, in their designated location. While doing this, you may want to adjust “voicing” of each loudspeaker to your individual taste, by inserting “voicing” elements like: high-pass, low-pass shelving elements, or broadband parametric-Q elements to obtain BBC dip effect, and so on.

This is an important step in tweaking the complete system, because flat frequency response is one thing, but whole system also has to sound right for your own ears as well. Personally, I always enjoy this part of the system implementation, knowing, that I start from perfectly optimized system already, and this is just putting the final touches on the overall performance of he system.

For the 2-way speakers, I have employed Linkwitz, 2nd order filters with crossover frequency of 2000Hz. In this design, I have included RoomEQ component as well, and it is available for each of the loudspeaker boxes in the system, giving you four built-in room equalizers to work with. Also, one shelving element is included for in-room voicing tweaks on the bass side.

Typical Measurement Results

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

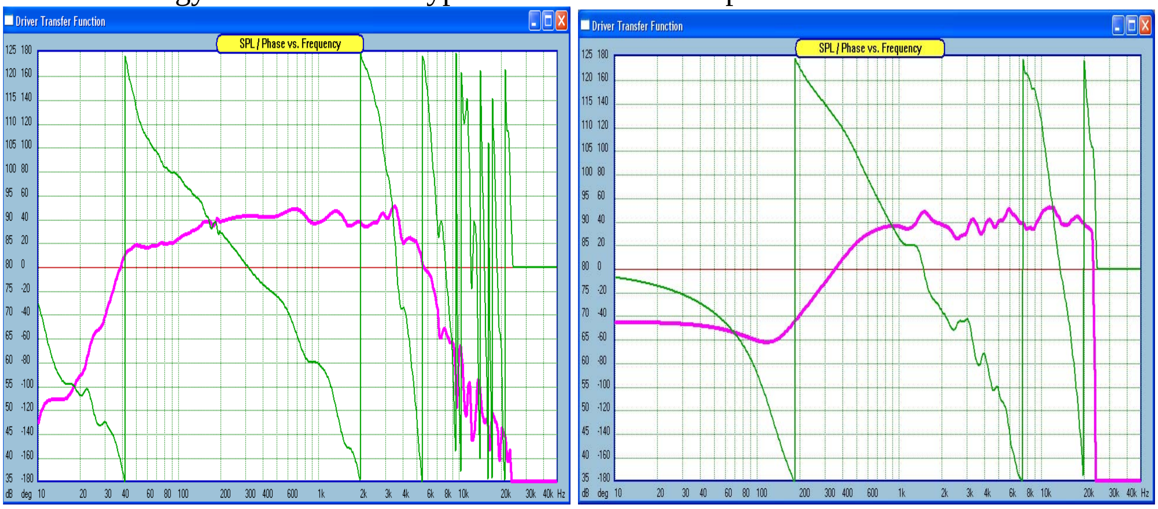


Figure 29. SPL/phase measurements of woofers ad tweeter in 2-way system.

to this level of performance……

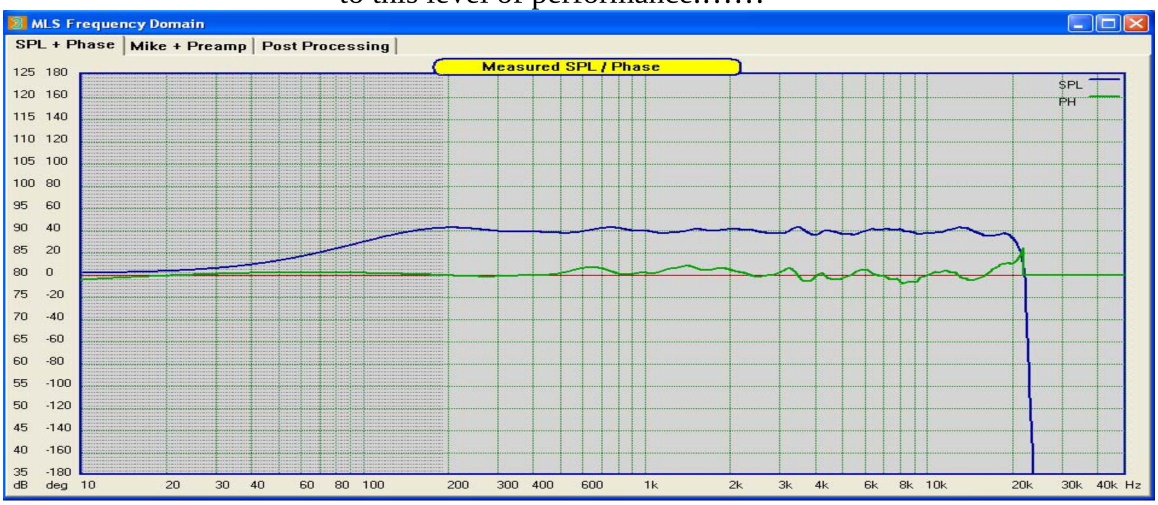


Figure 30. Equalized SPL and phase in 2-way system.

It is clearly observable, that SPL is remarkably flat (+/-0.8dB) 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 remarkable by any standards – please see below.

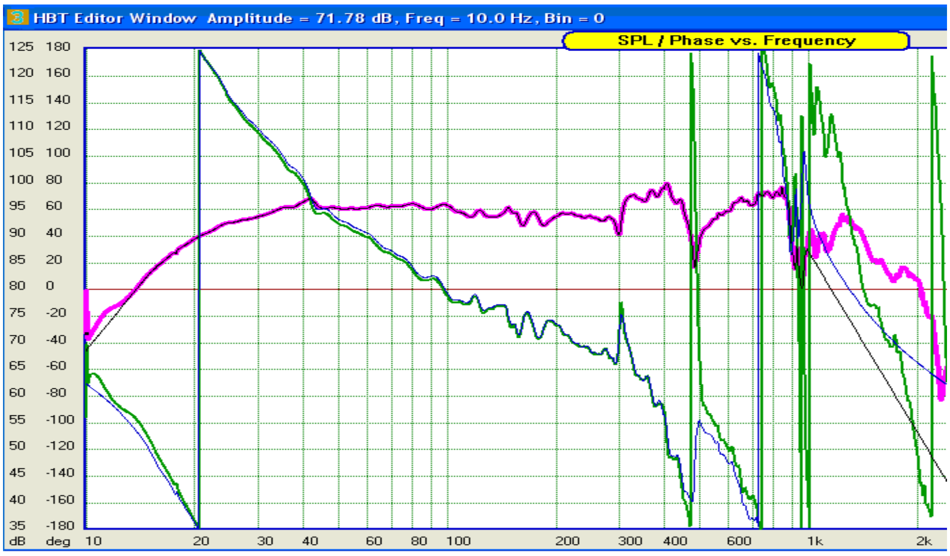


Figure 31. Raw 18” McCauley 6174 subwoofer SPL/phase measurements.

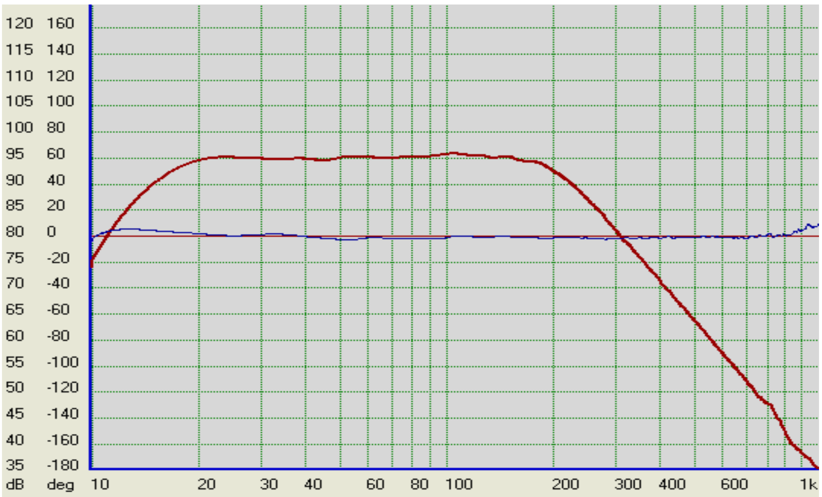


Figure 32. Equalized subwoofer SPL/phase measurements.

Linear-phase loudspeakers are accurate. You feed square wave into it, and square wave comes out acoustically. You feed a one-sided transient into it, and practically identical transient comes out acoustically too. Square-wave comparative measurements of the complete **band-limited (driver+LP filter) subwoofer from Figure 32**, are shown below.

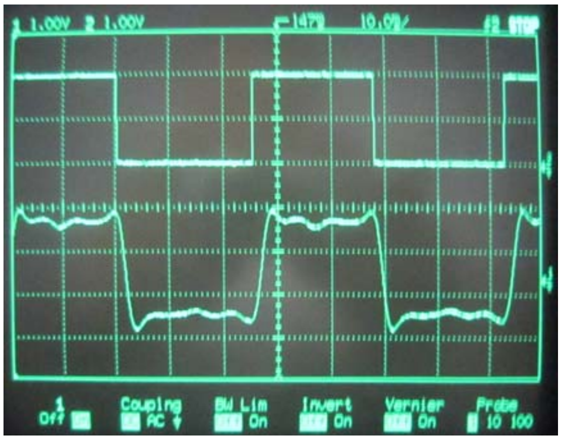


Figure 33. 20Hz square wave: Minimum-Phase Mode and Linear -Phase Mode

Conclusions

As indicated in the opening paragraph, I described main components of an advanced audio system, which meets current and future requirements for handling digital music files of any type. This aspect is solely controlled by the media player employed. There are many options, and apart from the obvious – Windows Media Player, you may consider JRivers media player, available from internet.

Maximum DSP capabilities with LynxAES16 sound card are 2x8way system, and output power for each channel is determined by the PWM-ICE amplifier configuration. The amplifiers can be run in single-ended mode, or bridged-mode if you need copious amount of power. Up to 500W / 4 ohm per channel can be extracted from each module in bridged-mode.

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 – 120Hz 3dB (+/-0.25dB flatness) 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. Improvements in SPL, phase and time domain transduction are very evident in the measurement results presented above.

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.

Construction of the system is extremely simple. All design issues related to impedance of the drivers are taken out of the equation. Crossover design is as simple as selecting filter options from provided menus. Optimization is ensured by HBT equalization, and done automatically, as is linear phase. Ultimate Equalizer comes with it's own MLS measurement system, so you have all you need at your fingertips. Once the system design is completed, you can switch UE to the self-start, "player mode" and just forget about it. Drivers and amplifiers are bolted to their respective enclosures – and that's all.

System components are interconnected via AES/EBU digital links, with the D/A converters right at the very end of the chain – in the loudspeaker enclosures. Therefore all processing of the music files read from the hard disc, right to the transducers is not affected by possible ground-loop hum, RCA cable capacitance, bandwidth of intermediate analogue components (they do not exist), potential PC noise and others unwanted sources. Importantly, as you would expect, there is single “master clock” in the whole system.

System’s exceptional frequency-domain and time-domain characteristics are documented by the measurement results.

Future expansion: “CABS” – Controlled Acoustic Bass System, as described in
http://vbn.aau.dk/files/62729248/LF_sound_field_control.pdf

Acknowledgements

Bodzio Software Pty. Ltd. would like to acknowledge professional contributions from two companies: Lynx Studio Technology Inc. and miniDSP Ltd.

David A. Hoatson from Lynx has customized driver for LynxAES16 AES/EBU digital sound card, to facilitate 16 time-aligned channels using MME and WASAPI Exclusive Mode driver options. Driver version is 19L.

Visit: http://www.lynxstudio.com/product_detail.asp?i=13

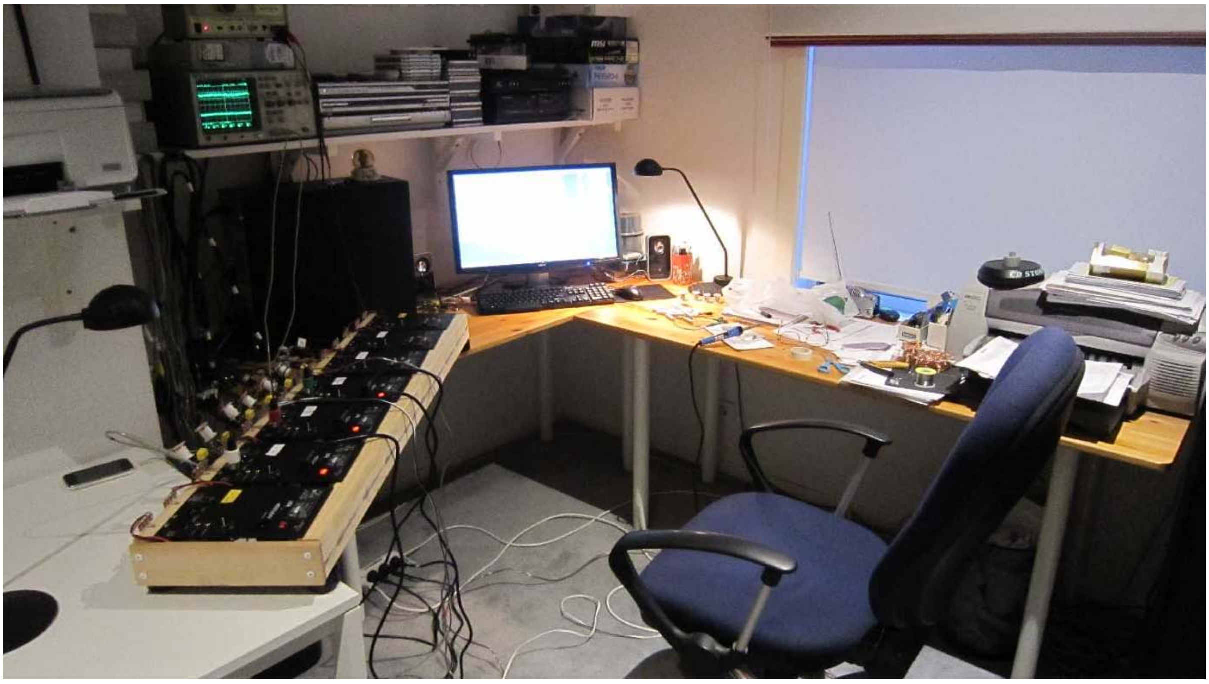
Tony Rouget from miniDSP customized firmware for the PWR-ICE class-D amplifier, to facilitate AES/EBU 2in/2out configuration, without DSP. Firmware for converting standard configuration to 2in/2out is available from miniDSP website.

Visit: http://www.minidsp.com/products/plate-amplifiers/pwr-ice125

Without these improvements, this project would not have happened. This degree of cooperation and professional integrity from these two companies is second to none.

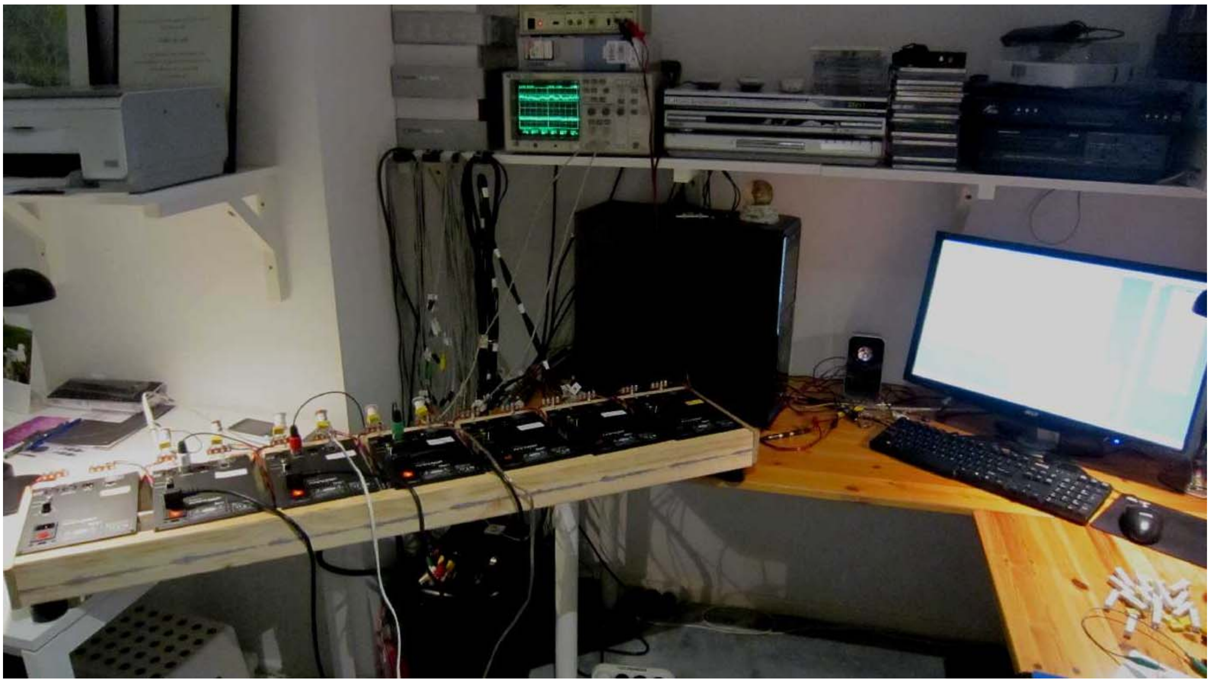
Thank you for reading.
Bohdan

Testing PWR-ICE Amplifiers in pictures.



Signal Generator, CRO, PC with Ultimate Equalizer.

7 PWR-ICE amplifiers are mounted on a frame for easy access.



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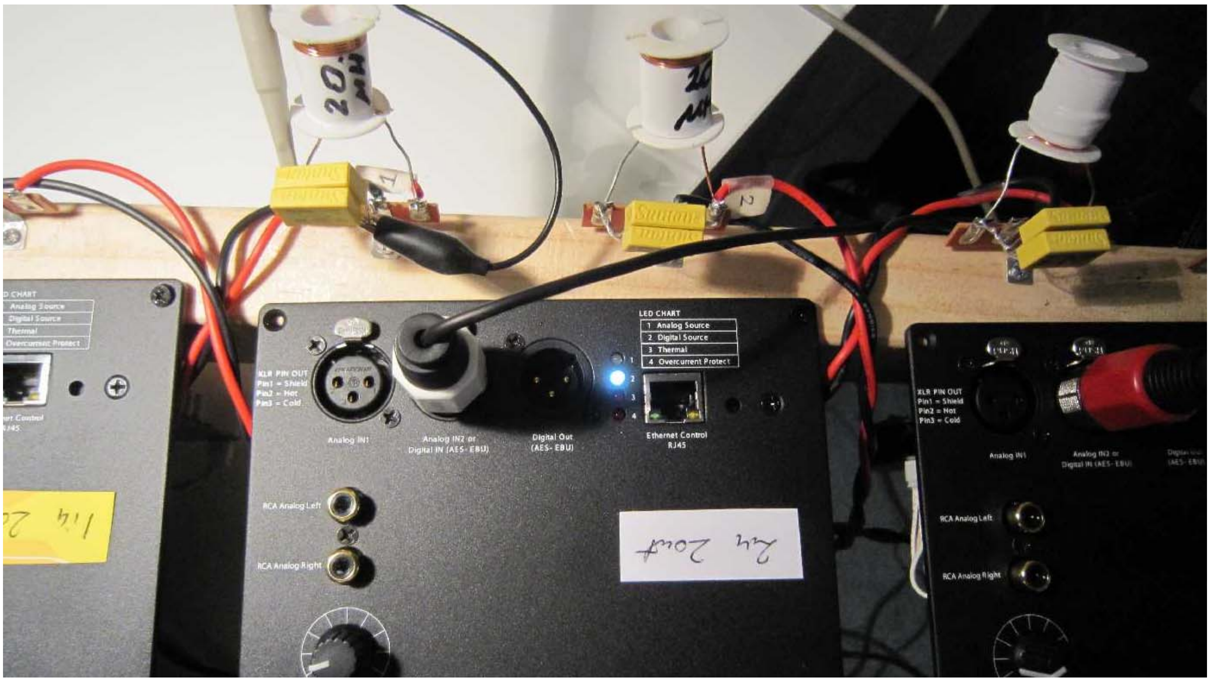

2in / 2out version of PWR-ICE amplifier. Additional filters shown.