Hybrid AES/EBU and Analogue 5.2 HT System

By Bohdan Raczynski

All Digital OR Hybrid?.

An AES/EBU stereo sound source can be connected directly to the input of LynxAES16 sound card, thus create a completely digital audio system. Sound sources with AES/EBU outputs do exist, for example:

http://www.canford.co.uk/Products/79-7204_DENON-DN-C620-CD-PLAYER-Analogue-and-AES-EBU-balanced-and-unbalanced-1U-rackmount

http://www.zzounds.com/item--TASCD500B

http://www.emmlabs.com/html/audio/xds1/xds1.html#

Another possible way of getting 5.1HT AES/EBU channels would be to use Multibridge Pro device, such as:
http://www.blackmagicdesign.com/products/multibridge/models

However, if you are looking for a less costly system implementation, and still capable of delivering up to 110dB SNR, then a hybrid system shown below will work perfectly. Input is provided by Delta1010LT 10-channel (8 analogue, time aligned inputs and 1 SPDIF input), input system, and the digital AES/EBU output is provided by 12 time-aligned, AES/EBU channels of LynxAES16 sound card. The actual card capability is 16-time aligned channels.

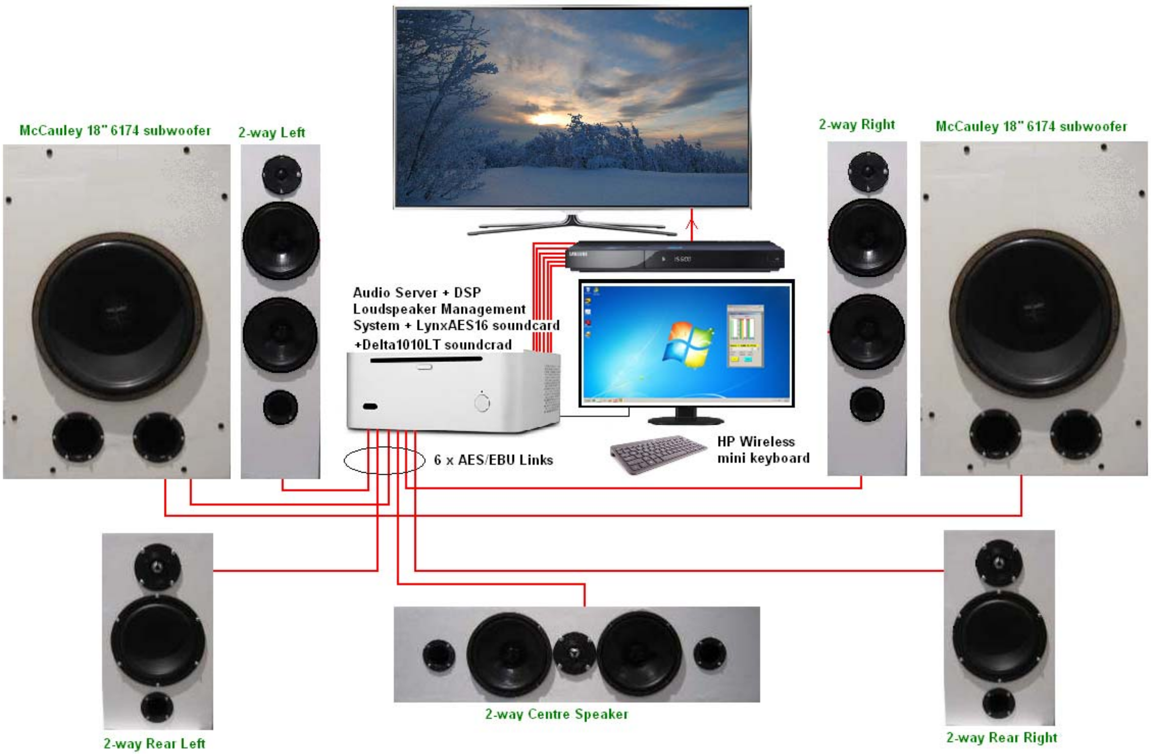

Figure 1. Hybrid 5.2HT Audio Processing system.

As shown in the example system above, each of the surround loudspeakers is implemented as a 2-way system, with one AES/EBU link connecting it with the PC. The AES/EBU digital link provides 2 channels and the digital amplifier is 2in / 2 out type.

System Design

All loudspeakers are powered by AES/EBU amplifiers amounted at the back of the loudspeaker boxes.

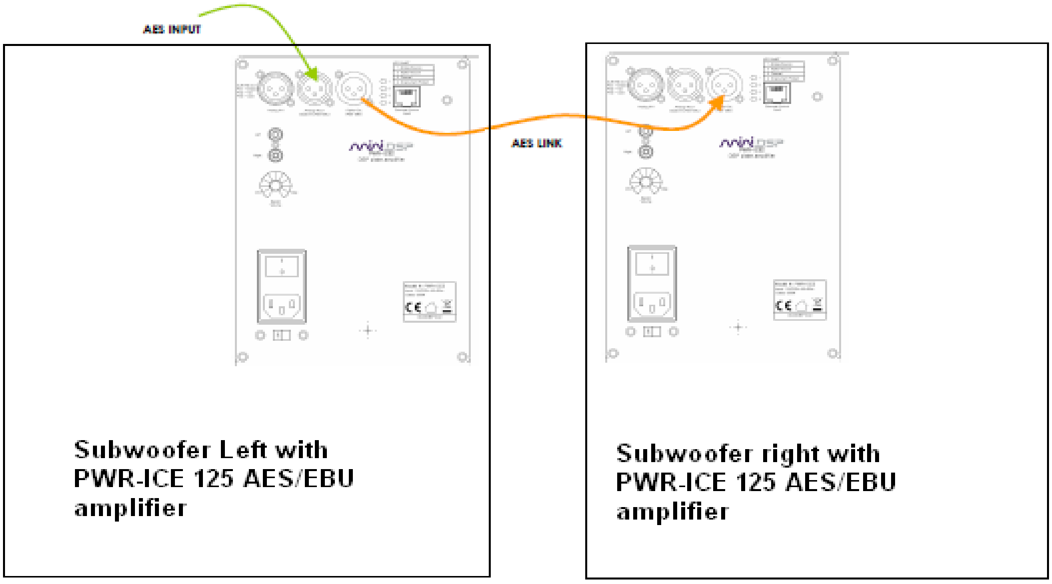

Figure 2. McCauley 6174 drivers in 300 litre vented (20Hz tuning) boxes with PWR-ICE 125 AES/EBU PWM amplifier in each box.

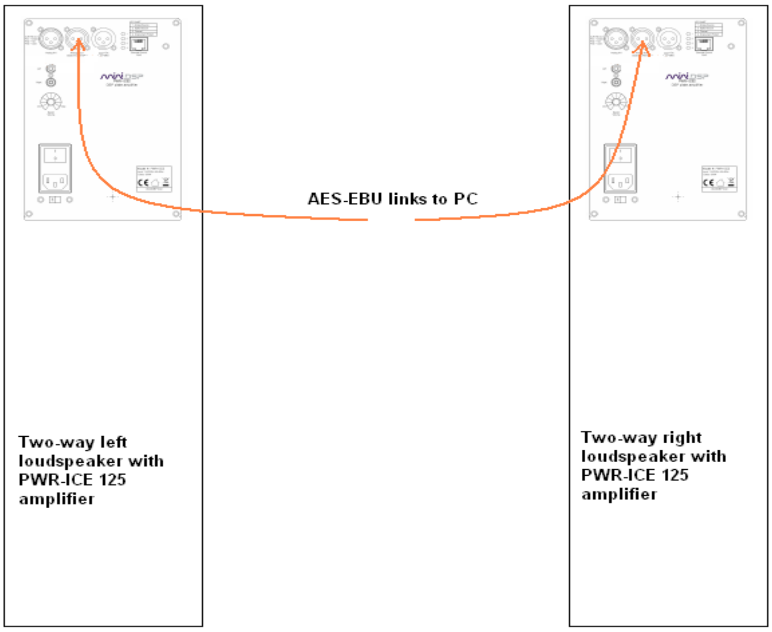

Figure 3. 2x8" woofer drivers + 1" tweeter driver with PWR-ICE 125 AES/EBU PWM amplifier in each 50 litre vented box, tuned to 45Hz.

Rear loudspeakers and the centre loudspeaker are also powered by the 2in / 2out ICE 125 AES/EBU PWM amplifiers. Here are some of the characteristics of the system, tested at 48kHz :

1. 24bit/48kHz, 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 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.

Contemporary PC can fulfil the above requirements. 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 or HTPC

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.

Input side - Delta1010LT Sound Card

For a proper operation of this system the Delta 1010LT sound card and PC Sound Manager setting must be selected correctly. The Delta 1010LT sound card has a built in microphone preamplifier associated with input channels 1 and 2. This microphone preamplifier must be disabled before installing the Delta 1010LT. This is accomplished by repositioning jumpers on the Delta 101 LT printed circuit board. Please refer the Delta 1010LT manual for instruction on disabling the microphone preamplifier. Input channels 1 and 2 also used XLR connectors. It will be necessary to either purchase or assemble XLR to RCA adapters to facilitate RCA input connections to channels 1 and 2. After installing the Delta 1010LT open the M-Audio control pane, and select the Hardware tad as shown in Figure 5. The setting should be as shown.

Figure 5. Delta 1010LT control panel.

When using the analogue inputs to the Delta 1010LT:

- Sample Rate: Set to 48000
- Buffer Size: Set to at least 128 / 256 (tested with 128).
- Mode: Set to consumer.
- Data Type: Set to audio.
- Emphasis: Not indicated.
- SCMS: None.
- Synch Source: internal

Delta1010LT also serves as the clock source. The Word Clock output of Delta1010LT must be connected to Word Clock input of the Lynx AES16 sound card.

Now, navigate to Windows Sound Manager and select the Recoding tab in the Sound dialog. The Dialog should appear as shown in Figure 6.

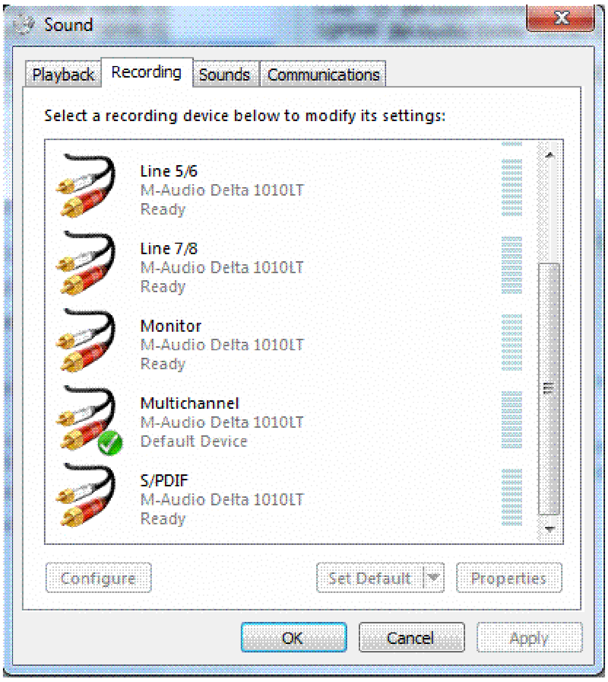

Figure 6. Sound dialog Recording tab.

· Highlight Multichannel, but do not select it as “Default Device”.
· Click Properties and select the Advanced tab. The display should appear as shown in Figure F5.

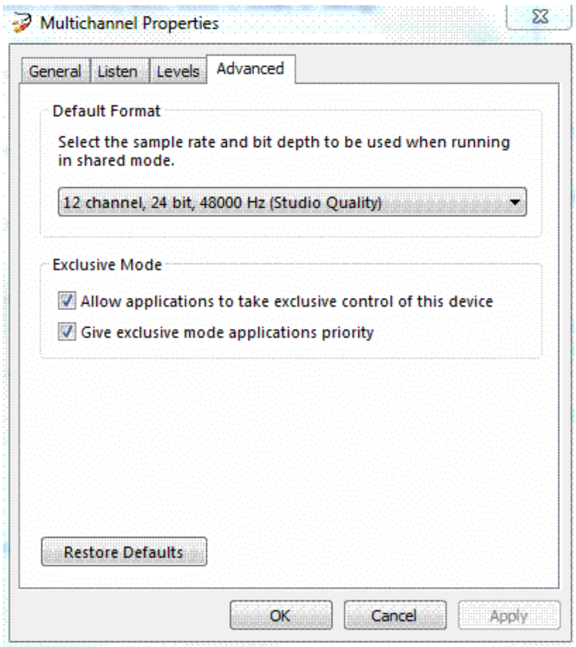

Figure 7. Multichannel Properties Advanced tab (recording).

· Set the sample rate and bit depth to 12 channel, 24 bit, 48000 Hz.
· Click the Levels tab and make sure the level is set to 100.
· Click Ok.
· Select the Sounds tab in the Sound dialog and set the Sound Scheme to No Sounds
· Click Ok to close the dialog.

Output side - LynxAES16 digital soundcard

Figure 8. 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 “Play 01+02” device to 16-channel, 24bit/48kHz.

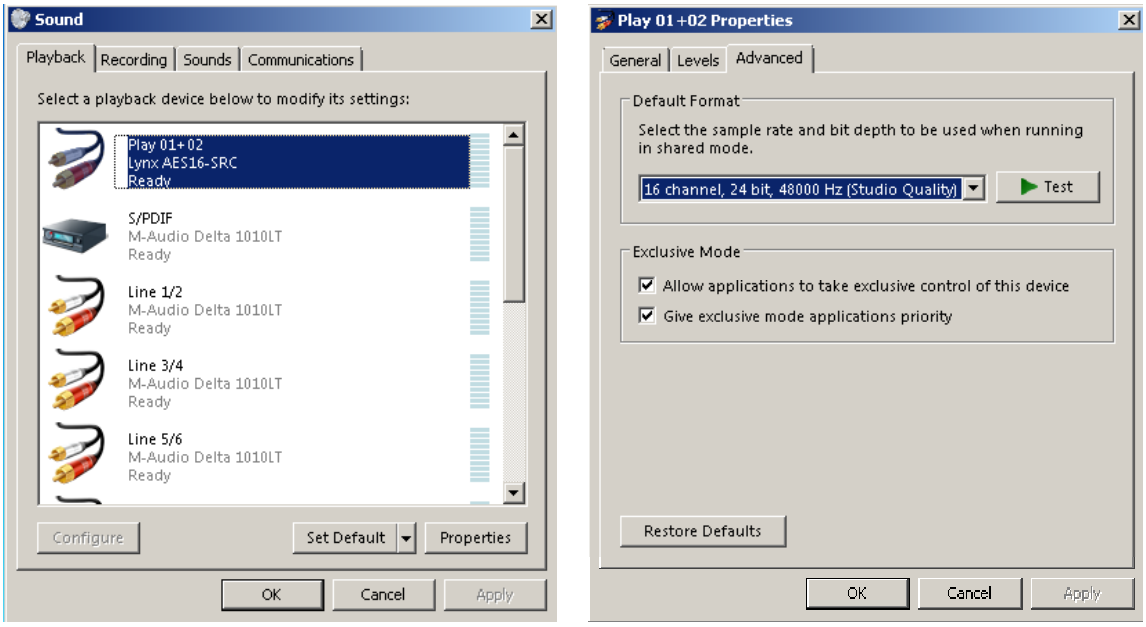

Figure 9. Setting the LynxAES16 “Play 01+02” device in Windows Sound Manager as 16-channel, 24bit/48kHz device.

In the next step, please open Lynx Mixer program, and navigate to “Play 1 Multi-channel Mode”, and set it to 16. This is an important step, as it commands the driver to time-align all 16 AES/EBU channels of the sound card.

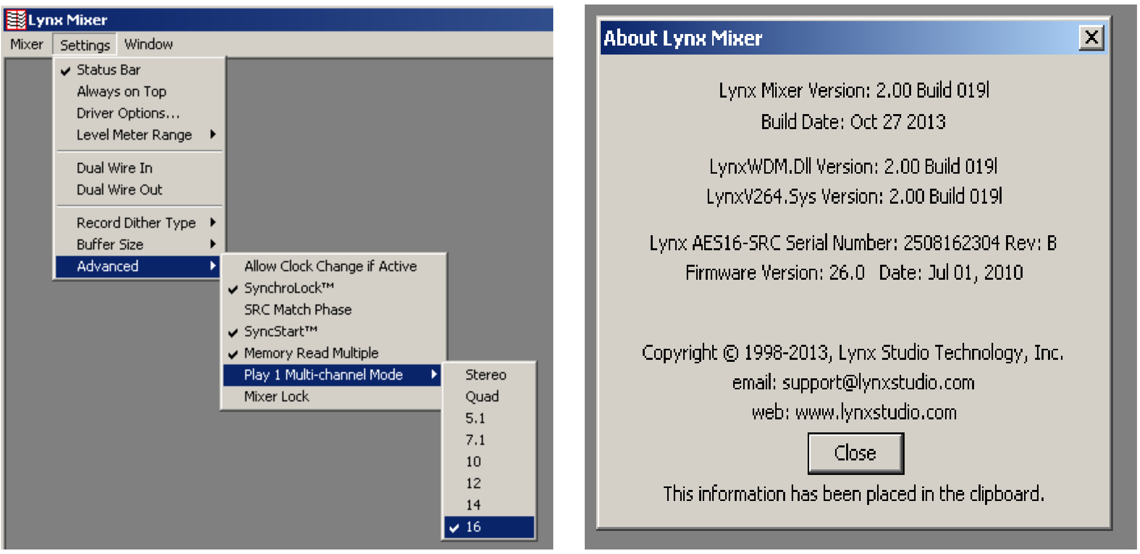

Figure 10. Setting the LynxAES16 for time-aligned, 16-channel output.

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 10.

Please note, that “Preferred Clock Source” is selected as “External” – this is where we connected the Delta1010LT word clock link. The “Rate Select” is set to 48kHz.

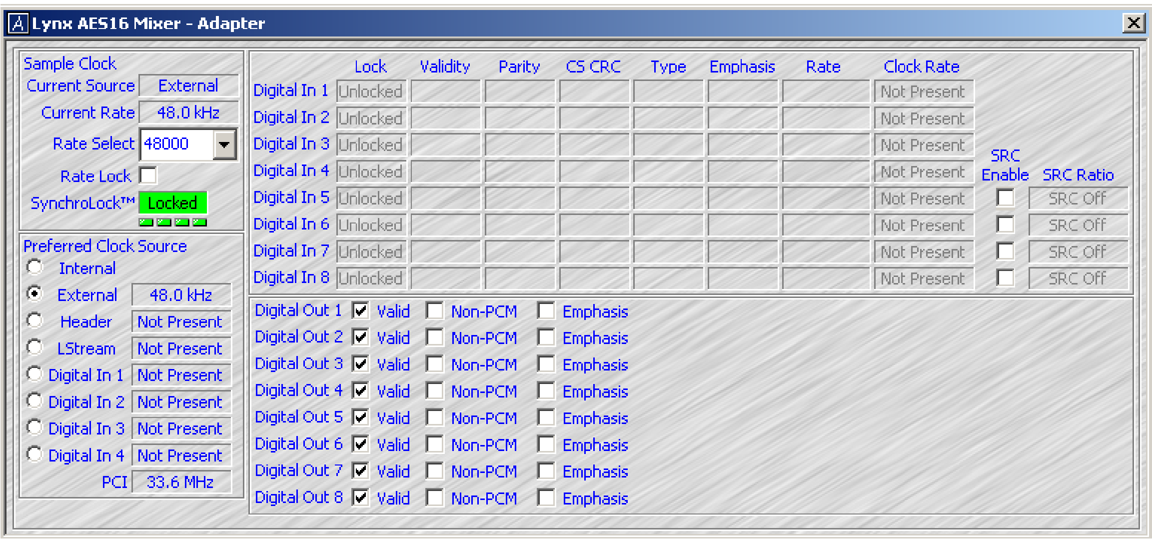

Figure 11. Information about input signal is displayed by the Lynx Mixer.

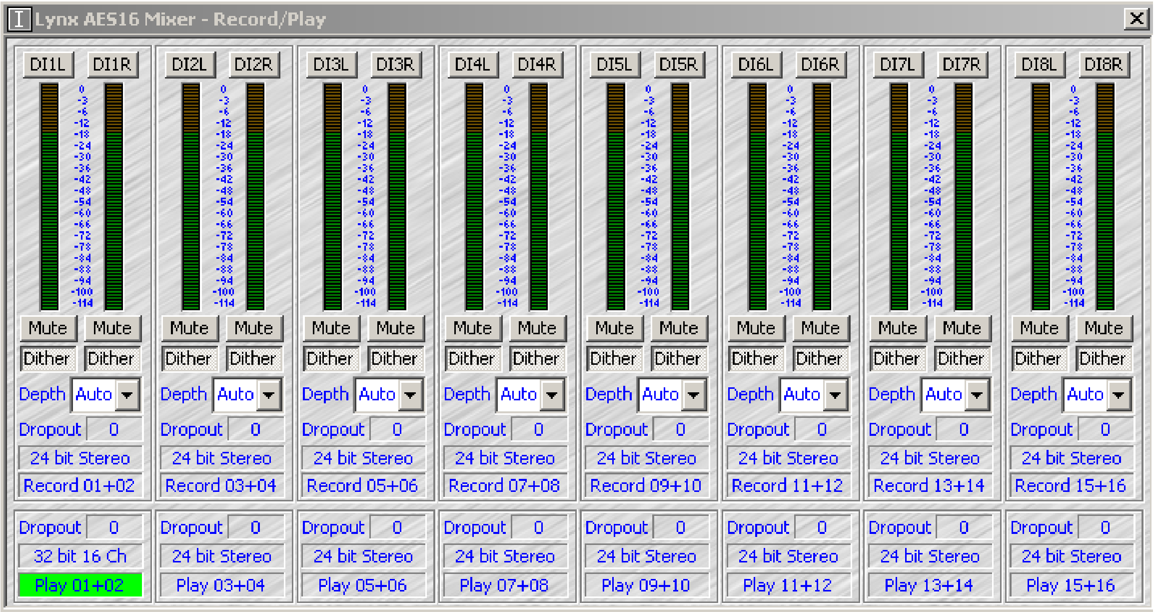

Figure 12. Input into the system is provided by Delta1010LT – so no activity on the "Record/Play" screen of Lynx Mixer.

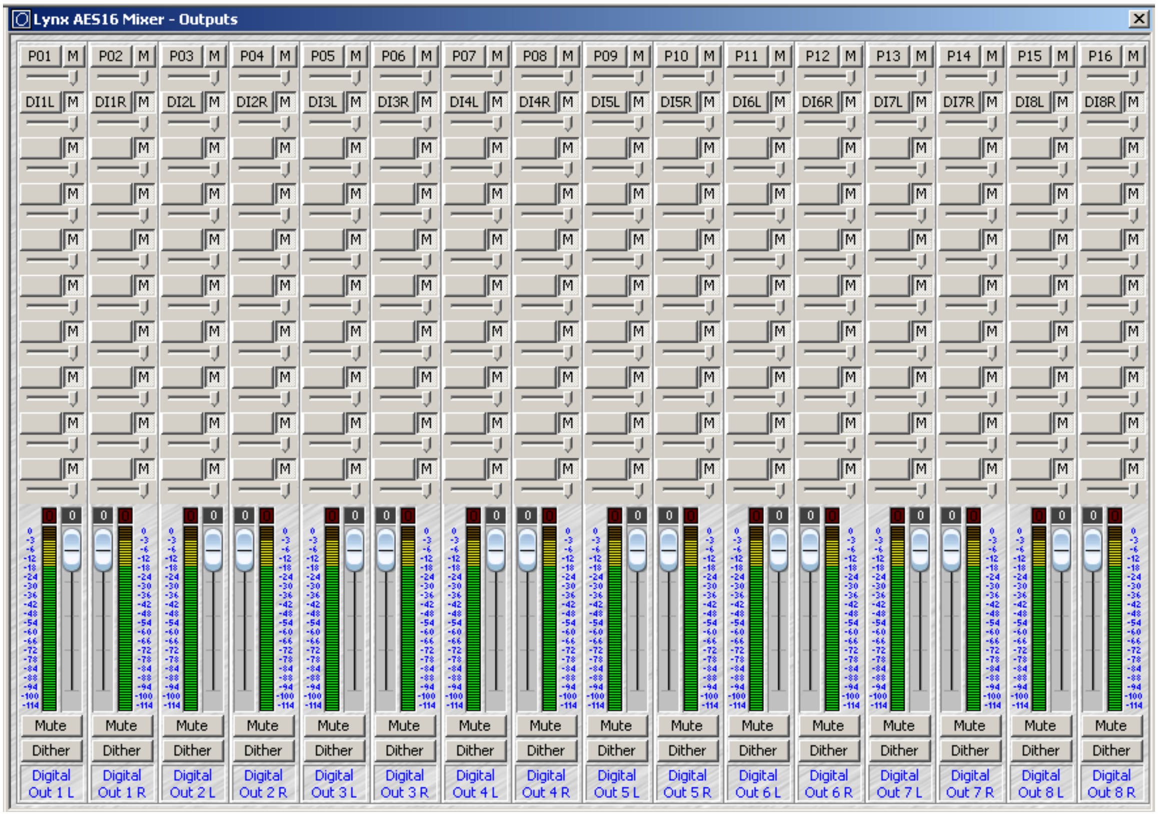

Figure 13. Presence of output audio data is confirmed by the "Outputs" screen of Lynx Mixer

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.

Subwoofers run in BLT Mode, and surround channels run in 2in / 2out 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 mode : Channel 1 (L) Channel 2 (R) Mixed L&R

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.

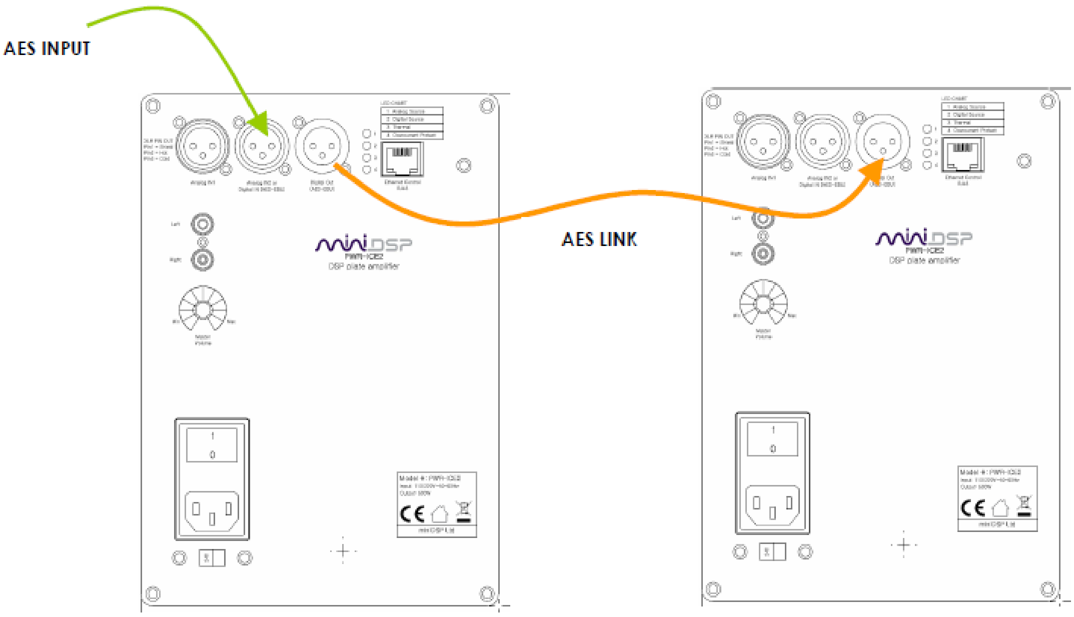

Figure 14. Interconnections between two subwoofers.

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 15. 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)

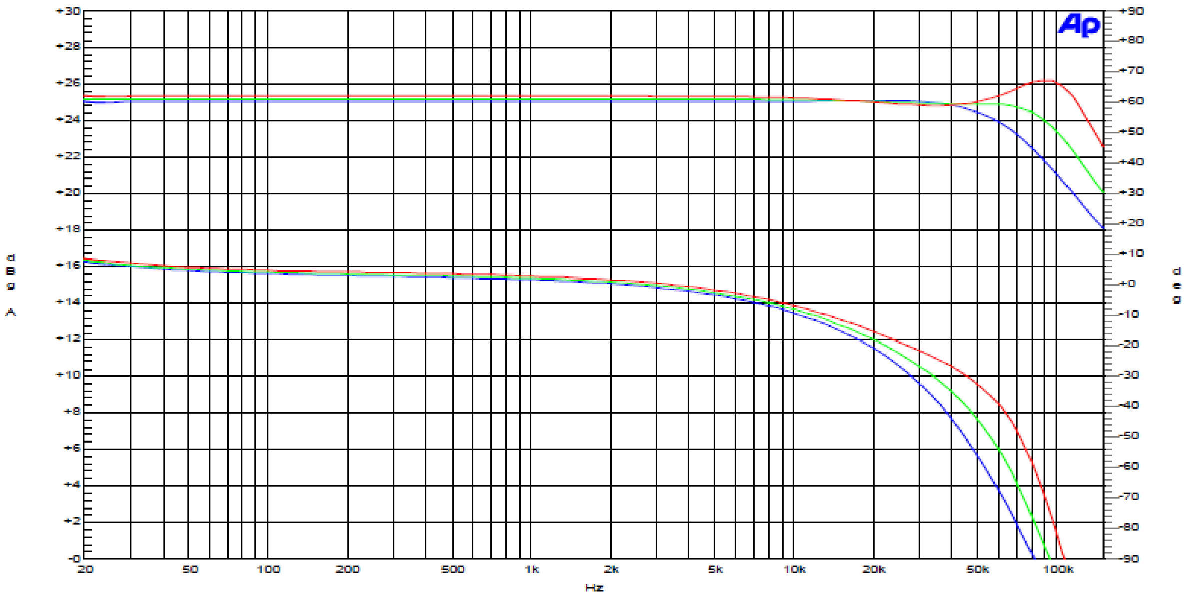

Figure 3: Frequency response in 4Ω (blue), 8Ω (green) and open load (red). Top – amplitude. Bottom – phase.

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.

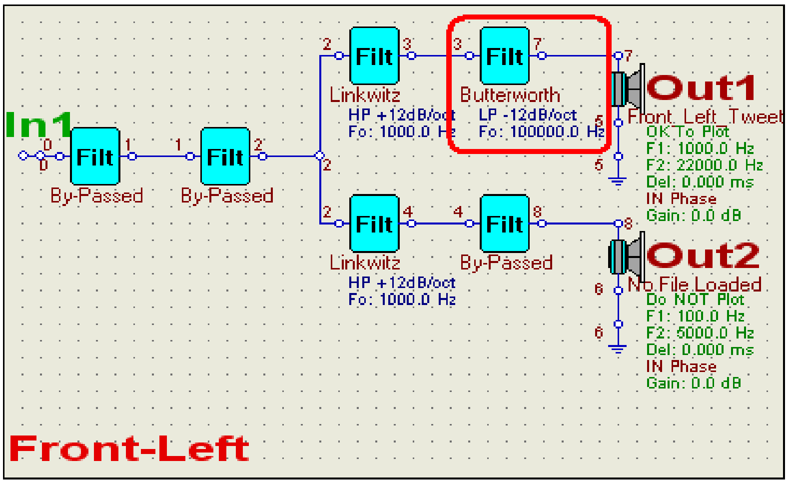

Figure 15. Compensating for PWM amplifier phase response.

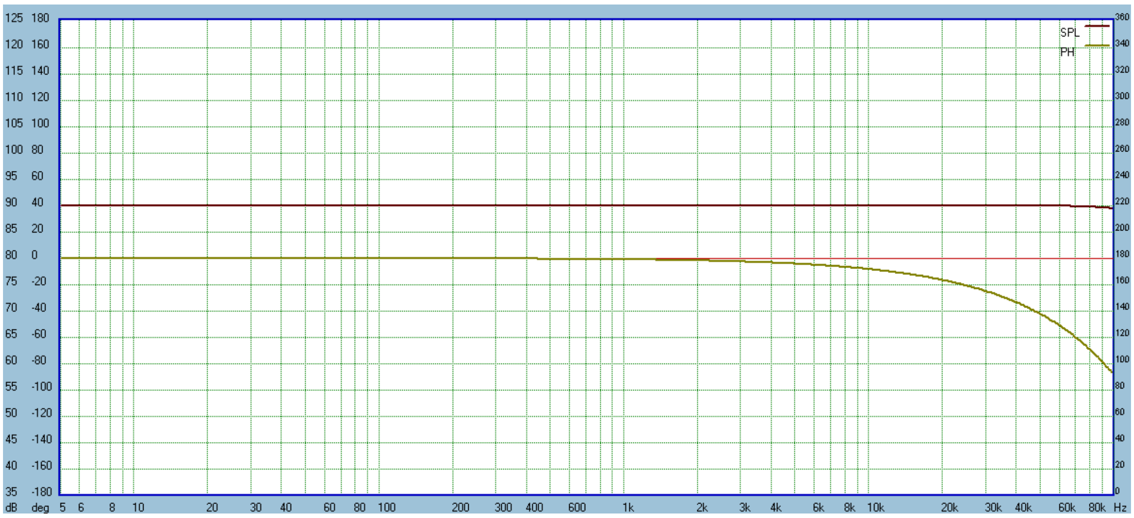

Figure 16. Model of the phase response for 8ohm load: -35deg@40kHz and -18deg@20kHz

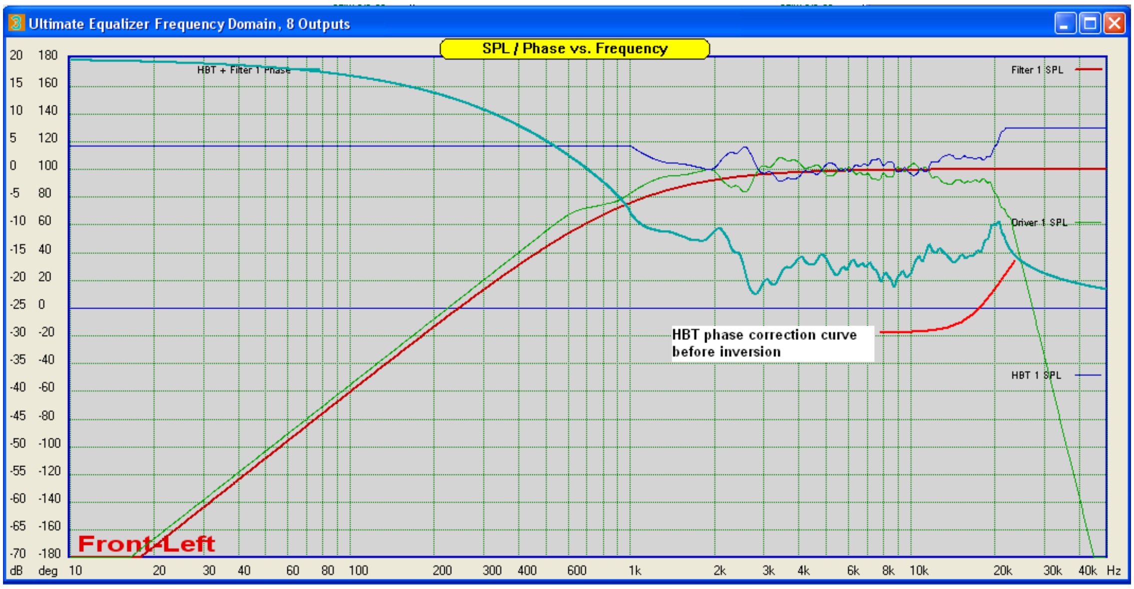

Figure 17. HBT phase response without the extra phase roll-off.

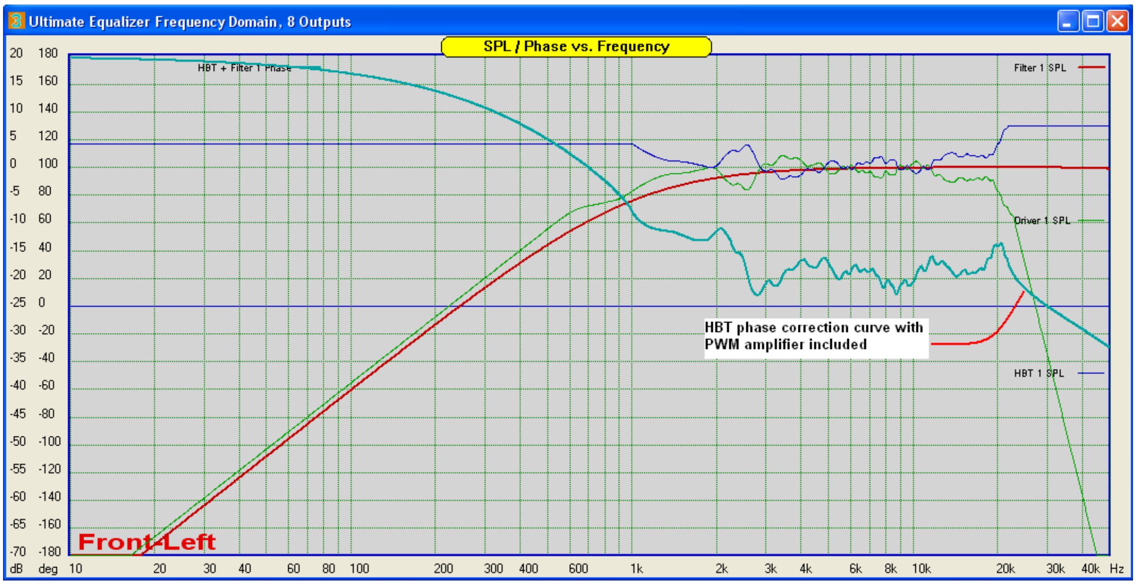

Figure 18. HBT phase response with the extra phase roll-off.

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

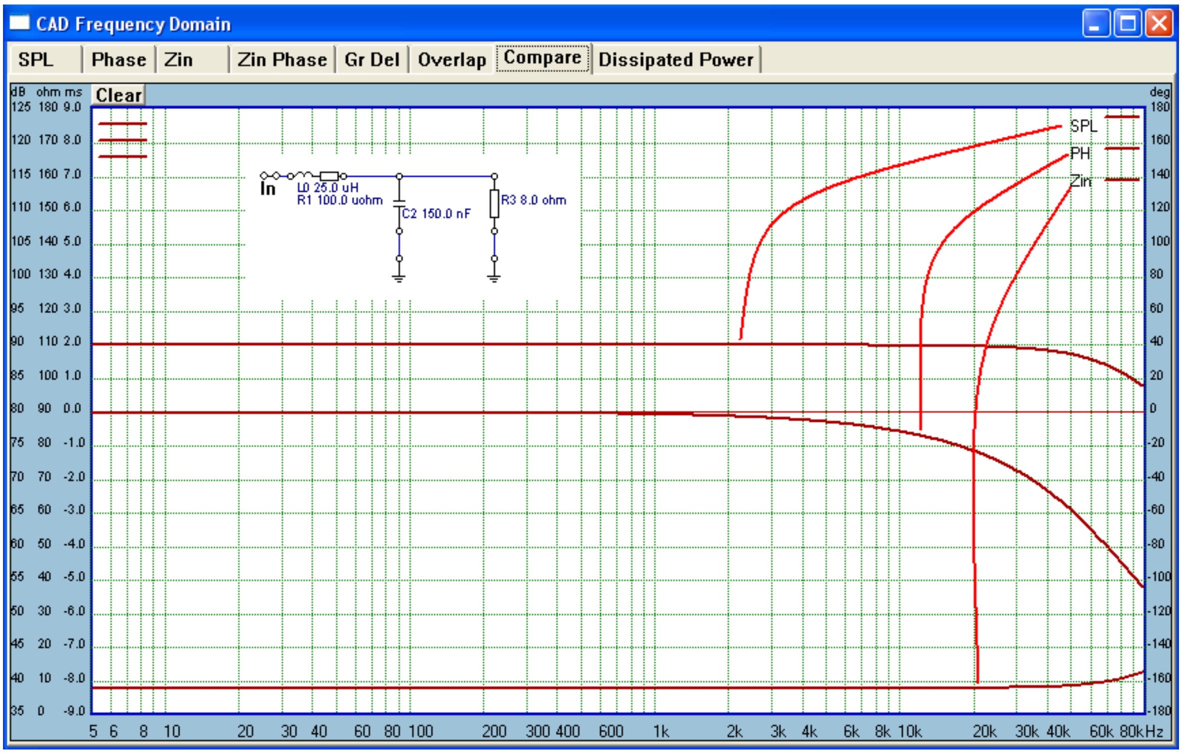

Figure 19. Low-pass LC filter to improve 500kHz suppression.

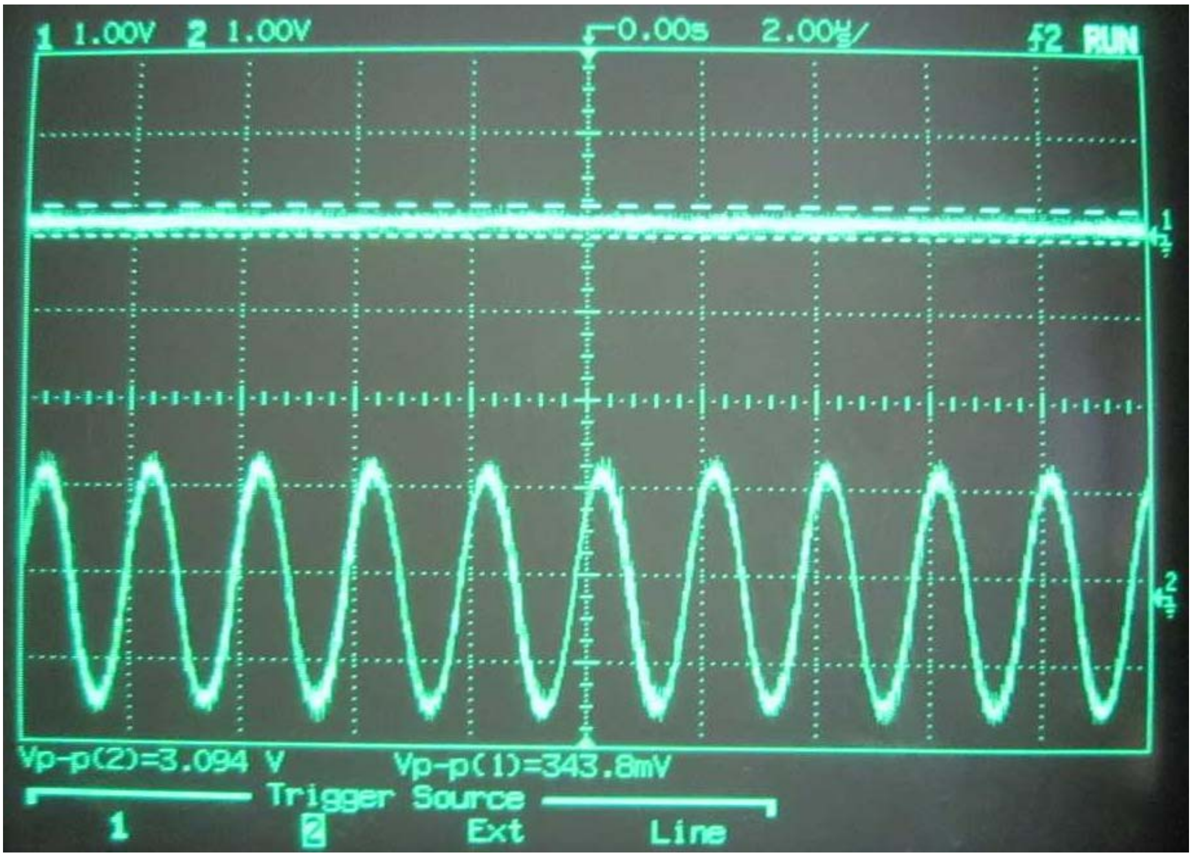

Figure 20. Bottom: 500kHz switching component before filter, Top: after filter – 1V/div

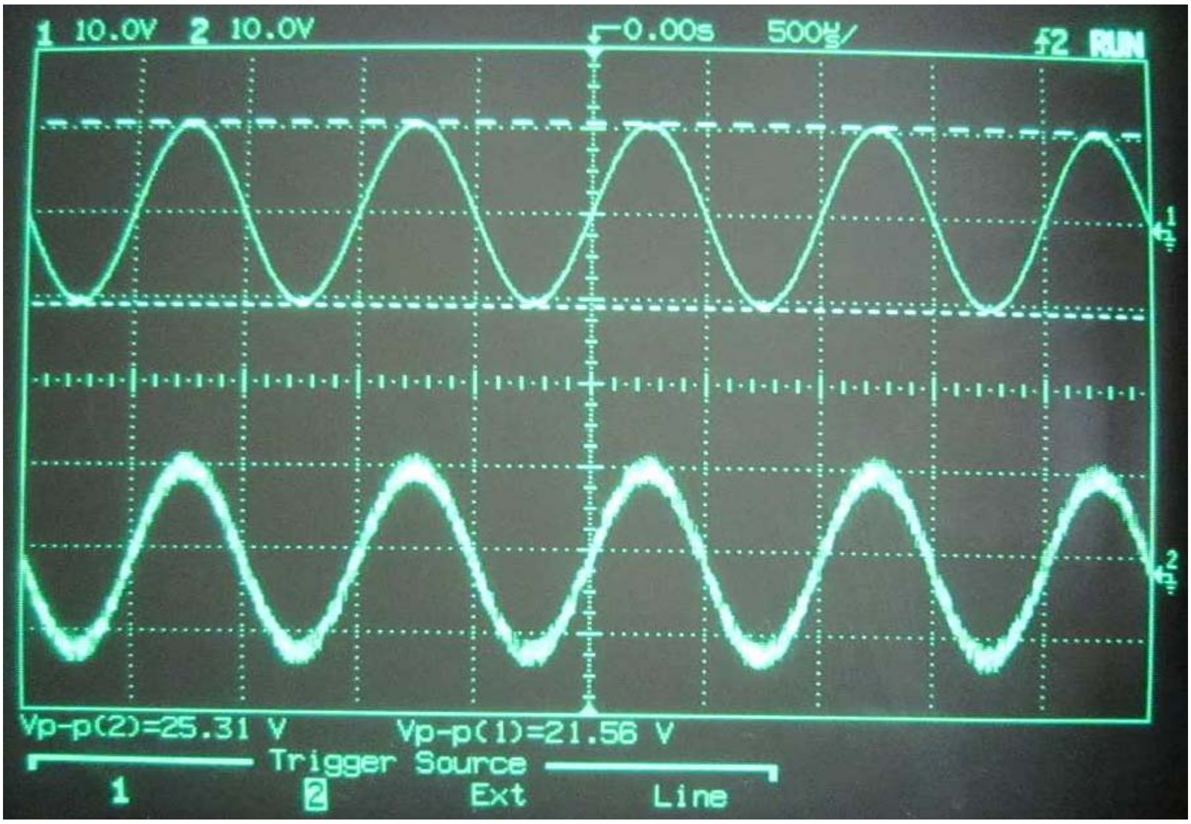

Figure 22. Bottom: 1kHz tone + 500kHz component before filter, Top: after filter – 10V/div

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

Loudspeaker Management System Development

An example of 5.2 HT system screen with BBM (Binaural Bass Management) is shown below. On the top of the traditional surround sound advantages, this system has several acoustical benefits and requires usage of two subwoofers. Theoretical foundations of this process are described in "Physiological and content considerations for a second low frequency channel for bass management, subwoofers, and LFE". AES Preprint 6628.

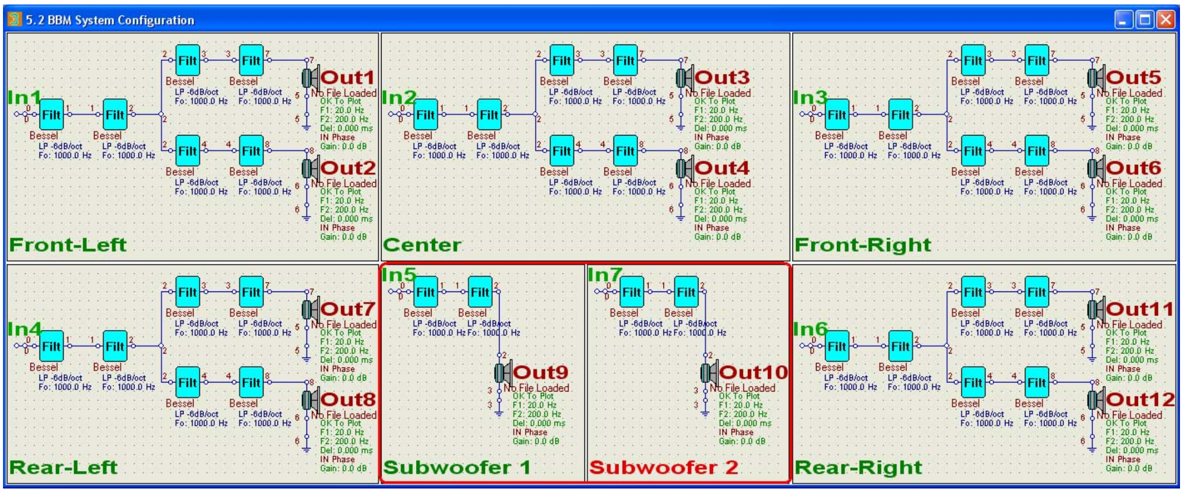

Figure 23. 5.2HT BBM version screen of Ultimate Equalizer.

Input to BBM: Left, Right, Surround Left, Surround Right, Centre, LFE. (this is the standard 5.1 configuration).

Output from BBM: Left, Right, Surround Left, Surround Right, Centre, Left SW, Right SW. Where:
Left SW = Left + Surround Left + Centre + LFE (all low-pass filtered <100Hz)
Right SW = Right + Surround Right + Centre + LFE (all low-pass filtered <100Hz)

Left subwoofer plays all left channels filtered down to below 100Hz, and the right subwoofer plays all right channels filtered down to below 100Hz. Filtering frequency, type of filter and it's slope is obviously selected by the user.

Typical Measurement Results

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

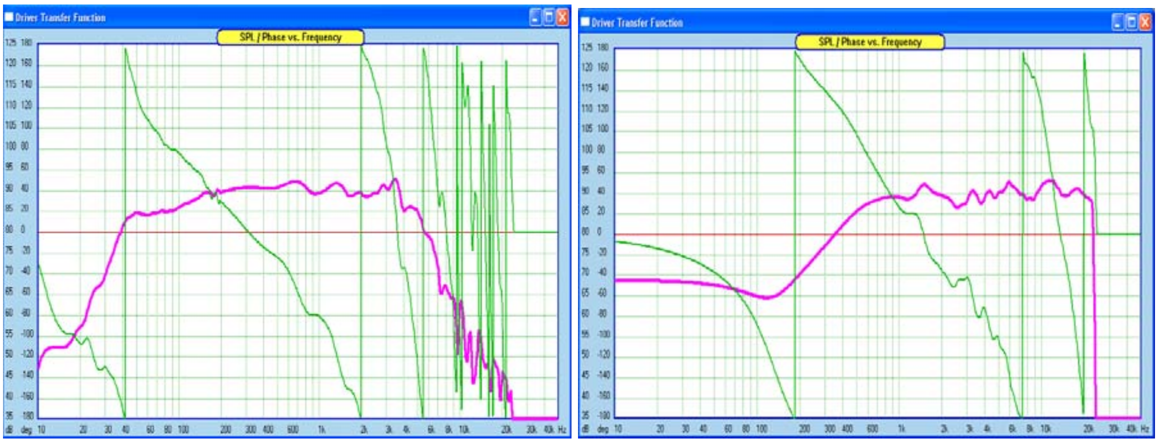

Figure 24. SPL/phase measurements of woofers ad tweeter in 2-way system.

to this level of performance……

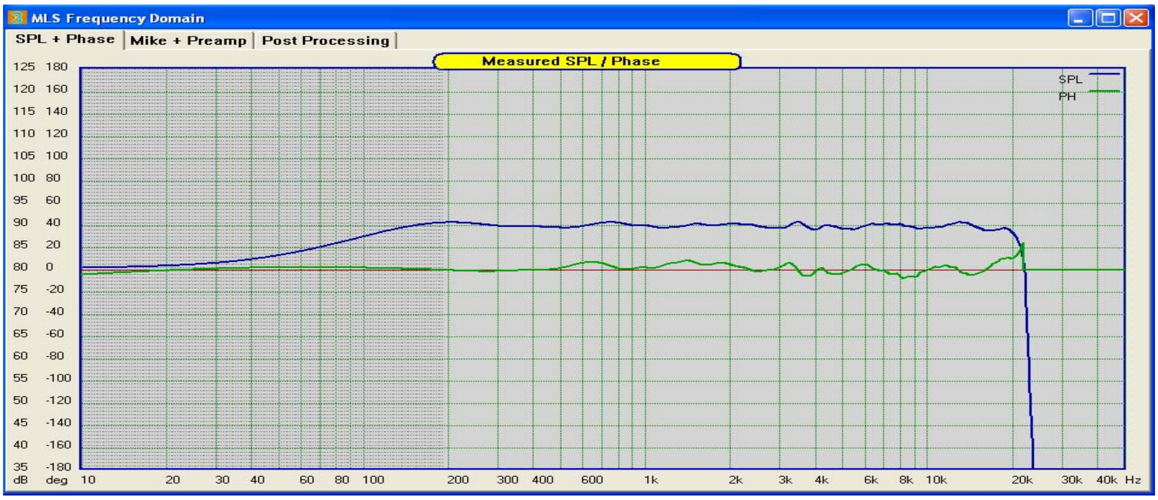

Figure 25. Equalized SPL and phase in 2-way system.

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.

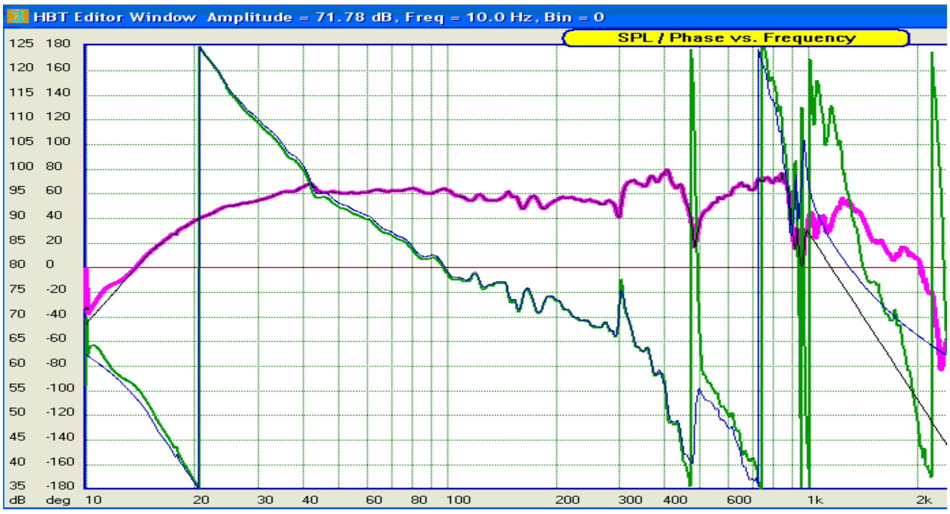

Figure 26. Raw 18” subwoofer SPL/phase measurements.

Figure 27. 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 band-limited subwoofer from Figure 27, are shown below.

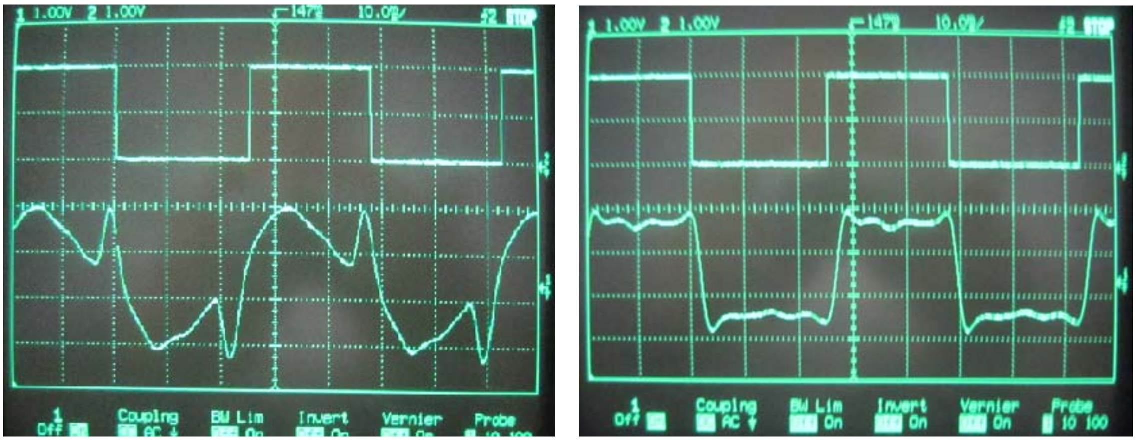

Figure 28. 20Hz square wave: Minimum-Phase Mode   and   Linear -Phase Mode

Conclusions

Maximum DSP capabilities are 8in/16out system, and output power for each channel is determined by the PWM amplifier configuration. The amplifiers can be operated 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.

Constructions 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 on the output side are interconnected via AES/EBU digital links, with the D/A converters right at the very end of the chain – in the loudspeaker enclosures. The only place you'll see an analogue signal is at the loudspeaker terminals.

As you would expect, there is single "master clock" in the whole system. The system's exceptional frequency-domain and time-domain characteristics are documented by the measurement results.

Thank you for reading.
Bohdan