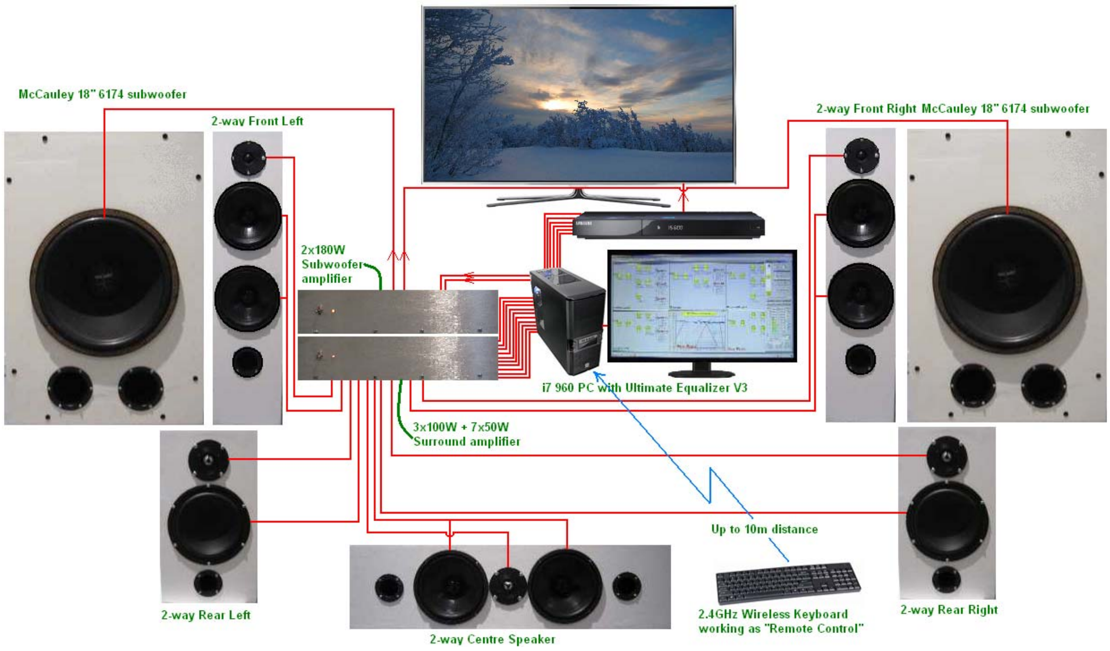
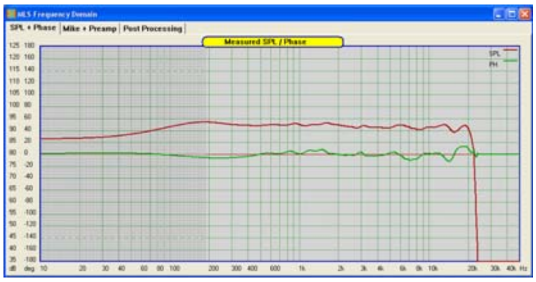
Conclusions And Observations

Here is the overall composition of the Linear-Phase 5.2HT System.




Subwoofer performance


Rear Speakers performance


Front Loudspeakers performance


Centre Speaker performance

It's design and implementation was executed by a non-professional (myself), using vary basic mechanical tools. I have no knowledge of timber and how to handle it, yet, I had to build seven enclosure boxes. I have no knowledge of metal works, yet, I have fashioned two chassis and front panels for the amplification system. As you may have noticed on some of my photographs published in this project's papers, I only have a little, portable work bench, which I would set up in my garage for each assembly working session. I do have basic hand-held power tools, but no fixed bench tools, therefore, cutting a straight line in a timber particle board was never 100% accurate. I had to find ways around it (belt sander – thank you), and I succeeded to a degree, because the finished product still looks acceptable.

Mechanically, all boxes are solidly built, but more experienced loudspeaker designer would have done much better design and construction job. For instance, it may be preferable to use soft dome tweeters for all loudspeakers, rather than metal dome for centre and rear channels. Some loudspeakers could have been implemented as 3-way systems. Drivers could have been recessed into the front panels, and enclosure edges could have been rounded. I could have experimented with some felt around tweeters. I could have simply used *much* better drivers….and so on.

Amplifier design and construction is very basic, and initially, I was contemplating publishing only a very limited documentation on it, and letting the potential follower sorting it out for his/her individual requirements. However, there was a significant learning factor involved in the design of the amplification system, and I felt, that it may be worth to share a complete picture of the "ups-and-downs" associated with this section of the project. Once again, your implementation may be completely different, and much better than mine.

Admittedly, I have always been interested in using computers in digital audio applications. I got my feet wet in this subject, with the release of SoundEasy's Digital Equalizer and Digital Filter capabilities, but this is where I would stop at that time. 12 years ago computers were not quite powerful to provide required processing capabilities and sound cards could not be synchronized. A number of years went by, during which computers and sound cards have made significant advances into quality digital audio. Finally, I made the decisive step in the digital domain direction by releasing UE1 two years ago. But it was not until UE3, where the full potential of employing a PC to enhance the audio playback system was realized. UE3 has a matured user interface and a host of DSP audio capabilities of high quality. Obviously, the development of UE Technology will continue, but this is where I put my stake in the ground and finally built this large 5.2HT system around it.

I'll never look back.

Overall, I must admit, that I have enjoyed the design, development and building of my new 5.2 HT system, even though I stumbled many times, and there were many disappointments too. It took several months to complete this project. Now, the system is fully installed and functional, and even though I am not an amplifier or speaker builder (as you could easily deduce from my photos), I am very glad that I put this system together - I got my reward.

If you are looking for an inspiration to progress beyond passive loudspeaker system designs, then project like UE 5.2HT system will keep you on your toes for quite a while (it was solid 6 months of assembly alone for me). This project is big, ***challenging***, advanced and future oriented, because at the same time, it's easy to upgrade. I have learned a lot during this project, and I had to sort out many things myself, as solutions were not available from the internet. I call this project "Linear-Phase HT5.2 Loudspeaker System Project", but linear-phase is just one of a number of improvements provided by this system, as it really inspires you to take a different, fresh look at the whole home audio reproduction issue.

On the technical side, each loudspeaker driver is individually filtered with DSP precision, and HBT equalized to achieve flat amplitude and corresponding phase response. The whole system runs in linear-phase mode or minimum-phase mode, depending on your preferences. In linear-phase mode the system will reproduce acoustically any signal (square wave etc…) without waveform distortion. All drivers are time-aligned and sonically balanced to provide 35Hz-22kHz for surround speakers and 16Hz-200Hz for subwoofers as flat, first arrival on the design axis.

It would be desirable to review the demonstrated major characteristics of the completed 5.2HT system based on UE Technology, right from the beginning.

New Design Approach

It's rewarding to discover how much the UE Technology changes the approach to loudspeaker design.

1. All impedance related issues, including measurements – gone.
2. All typical crossover-related issues, including CAD, design, compensation, optimization and actual crossover building – gone.
3. SPL and Phase have to be measured accurately – just as for any other design approach.
4. Design changes implemented instantly and auditioned instantly.
5. Amplification system can be purchased (say, Parts Express) or built.
6. Voicing of the system is just fun.
7. It's hard to believe how fast things happen using UE Technology approach, particularly from the most complex " *Oh – I just have an idea!* " stage to the " *Well, it sounds just as I hoped it would* " stage.

These are just some of the things I have come to discover and experience, working on this project. Life of a designer is a lot easier with UE. Even when using cost effective commercial amplification (MA1240 from Parts Express and other), you actually never have to deal with electronic components, or a network design issues. For a lot of users this may come as a relief.

Sound Quality

1. The all-important S/N (or dynamic range)

I had a real battle with ground-loop induced buzz, but I have managed to sort it out (http://www.bodziosoftware.com.au/Home_Theatre_Setbacks_Surprises.pdf). The second, equally important improvement came from adjusting power amplifiers' closed-loop gains to where they should be (see paper http://www.bodziosoftware.com.au/10_Channel_PA_Construction.pdf). Having done the above I was ready for an extensive sonic evaluation..

Sound quality exceeded my expectations. For starters, I was curious about the possibility of PC electronic noise being somewhat audible at THX volume settings. Instead, the sound is very loud, dynamic, perfectly balanced, and with quite satisfactory transparency. It seems, that there is some unfunded misconception and fear about integrating a computer into the audio chain. From my own listening tests, it was very clear, that Delta1010LT dynamic range, running 24bit/48kHz, (Input =

99.6dB, Output = 101.5dB A-weighted) seems more than adequate, even for very loud listening levels.

My overall conclusion regarding the dynamic range of the system is, that it is rather pointless to allocate and engage resources to extend the dynamic range beyond 100dB – because it will not be audible in your AV room, so no tangible benefits to your ears anyway. However, it makes more sense to redirect the resources to improve performance at the THX 0dB (or listening levels, like -10dB to -30dB) – because that's where you will hear the difference in sound quality.

It is obviously a surround system, as I will not go back to stereo. For a truly immersive and engaging experience, the "good old stereo" does not come close to surround sound experience. Many times during my music-DVD listening tests, I flipped from 2.0 stereo setting to 5.1 Dolby Digital setting. Each time the feeling was, that I was being plucked from the audience and put behind a glass wall to watch the concert from a distance. The 5.1 Dolby = enhanced experience, 2.0 stereo = diminished experience. I can't put it any simpler than that. Obviously, this has nothing to do with UE system's performance – it's just the surround sound experience. I also plan to start auditions of Audio-DVDs and SACDs.

2. Major characteristics of the sound.

The sound of the system in full flight (HBT equalization + time alignment + RoomEQ + acoustic linear-phase) is very alive and it's got quite a bit of punch. High-impact sounds like explosions, gun shots have authority and clarity to it at the same time. Low frequency rumblings like earthquakes actually shake the house, just as the movie director intended. The subwoofers are in the class of their own, and I use only around 3dB of RoomEQ at two broad room modes: 32Hz on 44Hz – it did help. Overall impression - smooth, extended and pleasant sound just fills the room. It's got power and authority when you want it too.

To my ear, most noticeable improvement in sound quality came from HBT equalization, and obviously – voicing profiles. I was able to run comparison tests of the same loudspeaker, using just the crossover component of UE vs. using crossover + HBT equalization. The difference in sound improvement is very noticeable, as the equalized loudspeakers sound smooth, extended and balanced with surprising amount of nicely controlled bass. The system as a whole, has very audible, broadband nature. Seismic bass (~16Hz) moves the furniture, and cymbals, high-hats and tingling triangles are delicate and have surprising clarity and detail in them. I have not heard these sounds like this before.

For standard stereo listening, I have added rear speakers at -3dB level. Just enough to increase the sense of spaciousness. It does not really open up stereo image, just fools you into thinking, that you are in much larger venue. The whole family likes it. Voicing is performed using custom-designed boost/cut curves with decibel-specified accuracy and precise location on the frequency scale. This is how it should be performed.

Anyway, it would be fair to say, that the more I listen to this system, the more I like it.

3. Linear-Phase Mode Tests.

I have conducted listening tests of Linear-Phase vs. Non-linear-Phase using variety of music, including country and symphonic music. The orchestral CD was "Wiener Philharmoniker on holiday", 2 CD set pressed by Deutsche Grammophon, and the pieces I auditioned were mostly on the second CD. Country music was represented by Alan Jackson "Good Time" CD, released in 2008. The UE set up consisted of Front-Left 2-way speaker augmented with it's own subwoofer and Front-Right 2-way speaker augmented with it's own subwoofer.

I could not detect much difference in sound between these two phase modes, except for a perceived distance from the musicians/orchestra. When using linear-phase playback, I felt closer to the musicians/vocalists, while the non-linear mode created an impression, that I am quite a bit further away. The effect was subtle, but I liked it because it creates more engaging feel and atmosphere during playback.

Some other interesting comments are also found here: http://redspade-audio.blogspot.com.au/2012/03/bathurst-2011-audio-event-of-year.html (scroll half-way down to DEQX tests).

Given the impressions outlined above, comments in the available internet literature documenting tests performed by other researchers, and most importantly to me - documented linear-phase measurement results presented in preceding papers of this project, I have developed a firm preference for linear-phase mode of operation.

4 Comparison with my reference Dali Surround Sound System.

A couple of years ago, while shopping for my first surround sound system, I have auditioned Dali system consisting of Concept 2 (for rear speakers), Concept 8 (for front speakers), and Concept Centre (for centre speaker). The audition included Concept SUB subwoofer as well, and all-together, it sounded pretty good. Unfortunately I would not be able to squeeze Concept 8 into my AV room arrangement, so I ended up with the following system – as highlighted in red:

	Concept 1	Concept 2	Concept 6	Concept 8	Concept 10	Concept Center
Frequency Response [Hz]	62.5-22500	43-25000	41-25000	41-25000	35-25000	72-22500
Crossover Frequency [Hz]	3000	3000	3000	610 / 3500	600 / 3200	2500
Sensitivity (2,83V/1m.) [dB]	87	87	91	90	91	91
Nominal Impedance [Ohm]	8	8	8	6	6	8
Maximum SPL [dB]	106	108	110	112	114	109
Recommended Amp. Power (8 Ohm) [Watt]	25-100	40-120	30-150	40-160	50-200	25-120
High Frequency Driver	1 x 1"	1 x 1"	1 x 1"	1 x 1"	1 x 1"	1 x 1"
Midrange Driver	-	-	-	1 x 5"	1 x 6½"	-
Low Frequency Driver	1 x 5"	1 x 6½"	2 x 6½"	2 x 8"	2 x 10"	2 x 5"
Bass Reflex System Resonanse [Hz]	58.5	42.5	38.0	36.5	29.5	55.0
Dimensions (H x W x D) [cm]	31x17x19	39x20x26	93x20x26	103x25x36	113x30x42	16x50x23
Dimensions (H x W x D) [inch]	12.2x6.7x7.5	15.4x7.9x10.2	36.5x7.9x10.1	40.5x9.8x14.0	44.4x11.8x16.3	6.3x19.7x8.9
Weight [kg]/[lb.]	4 / 8.8	6.3 / 13.9	14.2 / 31	25 / 55	34 / 75	6.6 / 14.5

Dali's subwoofer specification is as follows:

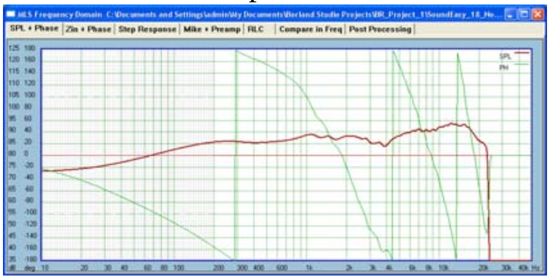
	Concept SUB
Design Principle	Active, built-in amplifier and crossover, sealed enclosure
Low Frequency Driver	1 x 12" air-dried pulp cone
Amplifier	160 Watt RMS
Inputs	Stereo line signal input RCA, input impedance 20 kΩ LFE Line signal input RCA, input impedance 35 kΩ
Controls	Volume level, lowpass crossover 50 - 170 Hz, phase (0 or 180°)
Frequency Response	29-170 Hz
Maximum SPL	110 dB
Power Consumption	115/230 VAC, 50 - 60 Hz, 330 Watt
Dimensions (H x W x D) [cm] Dimensions (H x W x D) [inch]	41x38x42 16.2x15.0x16.6
Weight [kg]/[lb.]	21.2/46.7

It is observable upon examining their specifications, that Concept 6 and Concept 8 are nearly identical loudspeakers, so I was comfortable with the decision to purchase smaller version. Concept Centre was the speaker I auditioned, and there was no other choice in Dali range, so I purchased it for my original HT system. For the rear speakers, I decided to purchase Concept 1 (instead of Concept 2), as two years ago, my understanding of surround sound requirements for rear channels was based on movies DVD, rather than SACD or Audio-DVD requirements. Needless to say, that today I would have made different decisions.

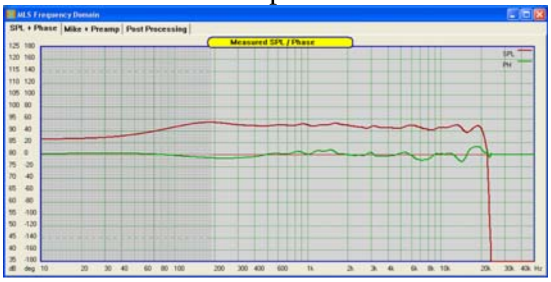
Anyway, I have been using my original Dali surround system for the past 2 years. I was accustomed to it's sound and therefore decided to use it as a reference system for comparison with the UE-based system. I switched off the McCauley subwoofers and used only surround speakers for this test. My DVD-player has 5.1 optical output, and this line was connected to JVC RX5032 surround sound receiver, which in turn, powered the original Dali speaker system. At the same time, the DVD-player has 7.2 analogue outputs, so I was able to connect these lines to UE3 system, and have the surround audio present simultaneously in two systems: Dali and UE3.

Mechanically and appearance-wise, Dali system is magnificently manufactured and looks perfect. In this department, the only good thing about my UE3 speaker system is it's colour. Yes, the white finish blends the boxes well into the room. Acoustically however, I was in for a surprise.

Dali Rear Speakers

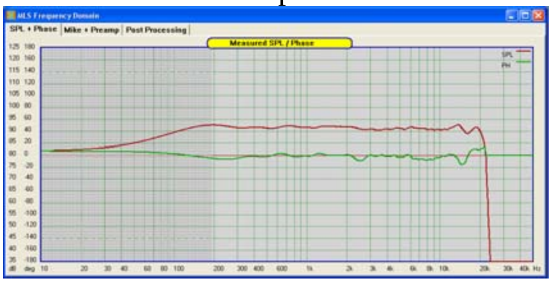


UE3 Rear Speakers

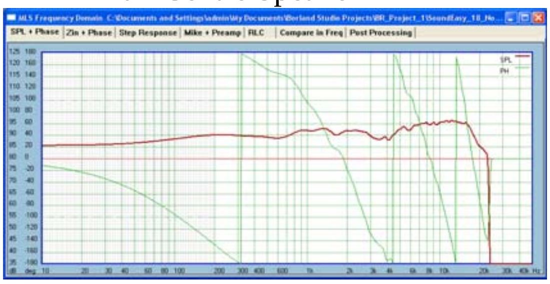


Dali Front Speakers

UE3 Front Speakers



Dali Centre Speaker



UE3 Centre Speaker

Interestingly, all UE loudspeakers are distinctly different designs. But they look pretty much the same on measurements (see above) – there is no magic here, this is HBT+Linear-Phase EQ in action.

In comparison with UE system.

- Dali system does not measure up to high-end 3dB cut-off frequency. Typically, all three types of loudspeaker have -3dB cut-off frequency of about 18kHz. This is not a big deal, but I would prefer, that loudspeakers specified up to 25kHz actually delivered this performance.

- Dali system has weak bass. I have set the RX5032 for bass boost of +10 and it still sounded thin (**Thin** - Fundamentals are weak relative to harmonics. Bass light). The biggest culprit here was the Centre Channel. It sounded Tinny and Nasal (**Tinny** - Narrowband, weak lows, peaky mids. The music sounds like it is coming through a telephone or tin can, **Nasal** - Honky, a bump in the response around 600 Hz). The Dali system I auditioned in the showroom had subwoofers operating at the same time, so perhaps this created an impression, that the reasonably good bass was there.

- Dali system lacks dynamics and definition. How could this be?. (**Dynamic** - The suggestion of energy and wide dynamic. Related to perceived speed as well as contrasts in volume both large and small. **Definition (or resolution)** - The ability of a component to reveal the subtle information that is fundamental to high fidelity sound).

- Dali system sounds dull. Again – how could this be?. I have listened to this system for two years and had no complaints against it, until UE3 came on line.

- Dali system sounds hard at higher volume levels. (**Hard** - Too much upper midrange, usually around 3 kHz. Or, good transient response, as if the sound is hitting you hard. Uncomfortable, forward, aggressive sound with a metallic tinge).

- Separate comments refers to Dali’s centre loudspeaker (Concept Centre – see specs above). I think, that limiting low-frequency response to 72Hz is a mistake. Dialogues just do not sound natural.

There was really no point going into comparison of finer sound details between these two systems. I have learned a lesson from this whole exercise and gifted the Dali system to a relative.

5 Loudspeaker Bandwidth and EQ headroom

Bandwidth and amplifier headroom are related via equalization scheme. This would typically manifest itself in the low-end of the frequency range. For instance, if you have a 3dB power headroom in your amplifier, you can use this 3dB for equalizing (boosting) the low-end of the woofer roll-off. I would suggest, that equalization should be kept to 4-7dB range. I have used 6dB equalization for subwoofers to get to 16Hz.

6 Vented boxes choice

This comment should be treated in conjunction with (3). I have specifically used vented enclosures in my project for two reasons: (1) they have better low frequency extension and (2) they have significantly reduced cone excursion around box frequency, which allows me to use equalization headroom to further extend the low-end frequency response without exceeding cone excursions too much. This mutual compatibility works very well, without adverse effects on the transient response, because linear-phase mode will maintain transient-perfect response anyway.

7 Flat vs. Voicing

A fundamental design goal of any respectable audio system is flat, on-axis, first-arrival frequency response from 100Hz-10kHz (there are other criteria, but let's keep things simple here). Apart from the obvious removal of random and unwanted SPL irregularities colouring the sound, such design proves, that you – the designer, is in full control of the design process, rather than some accidental activities or a sheer luck got involved.

But, then comes the influence of your acoustical environment (your room), your musical preferences and your hearing ability, and all these factors finally determine if the flat response is what you prefer. Personally, I see nothing wrong with voicing properly designed loudspeakers, particularly, that phase response in UE system stays linear in all circumstances.

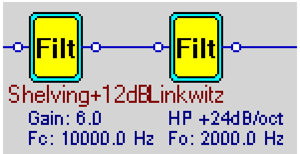
I voice my speakers all the time. More so, because I play a variety of recordings, starting with poorly balanced 60' and 70' and ending with contemporary cool guitar jazz and vocals, with impressive tonal balance.

Example: **UE3 voicing for vintage rock-and-roll music 60' and 70'**

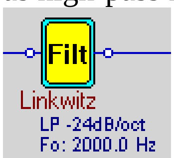
After HBT, the initial conditions for each stereo channel are:

- Subwoofer has flat frequency response 16Hz - 400Hz.
- Woofer has flat frequency response 40Hz – 3500Hz.
- Tweeter has flat frequency response 1000Hz – 22000Hz.

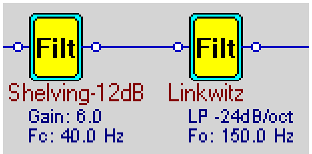
The idea behind this example of voicing parameters is: (1) to account for woofer and subwoofer SPL overlap between 40Hz and 150Hz, and NOT create a hump on the overall system SPL and (2) augment bass below 40Hz using separate subwoofer blended into each channel.
Tweeter crossover + voicing. Tweeter has a natural high-frequency roll-off above 22kHz -> NOT shown below.



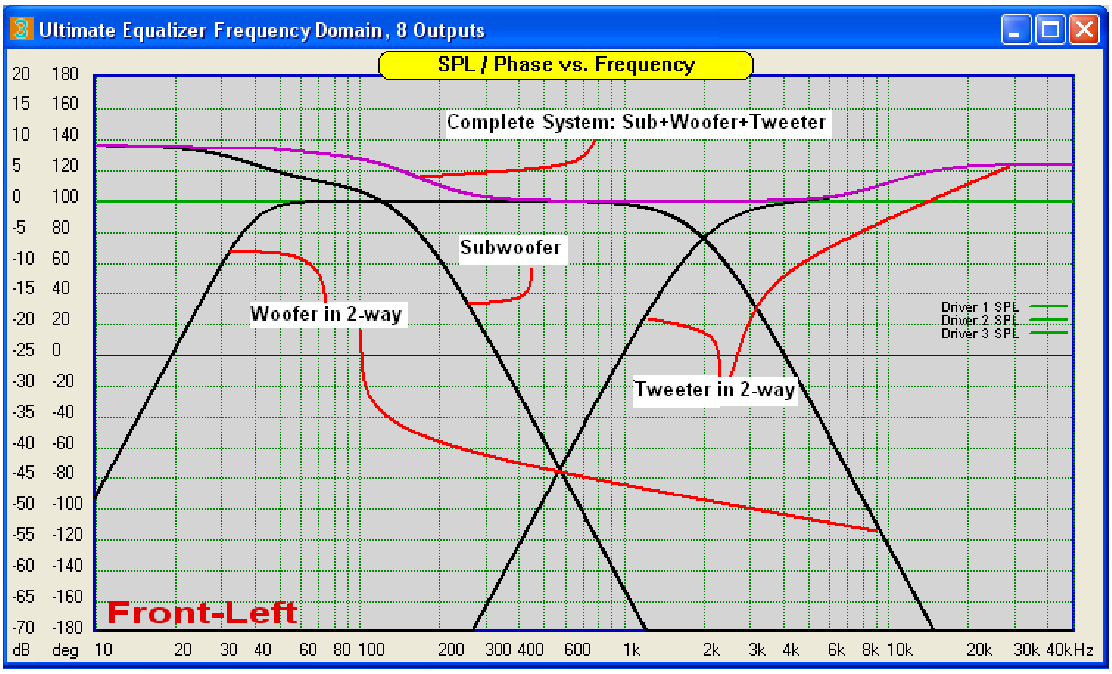
Woofer crossover. Woofer has a natural low-frequency roll-off below 40Hz -> shown as high-pass filter (black curve -3dB around 40Hz) on the diagram below.



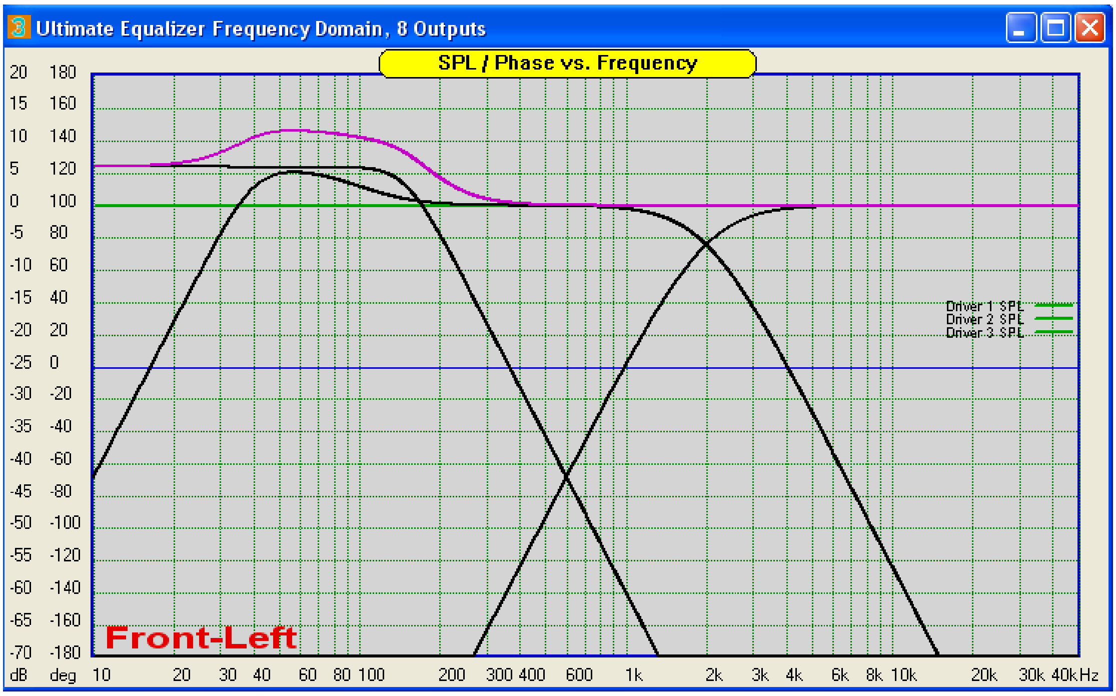
Subwoofer crossover + voicing. Subwoofer has natural low-frequency roll-off below 16Hz -> NOT shown below. Also, subwoofer channel gain = +3dB.



Pink = system SPL. Low- and high-transition frequencies can be easily adjusted.



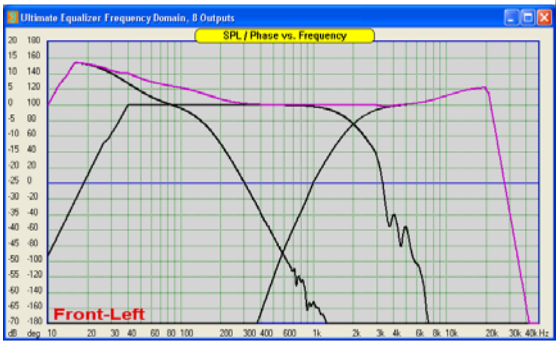
And this is what happens, when you allocate boost to the front speaker woofers and keep subwoofers flat.

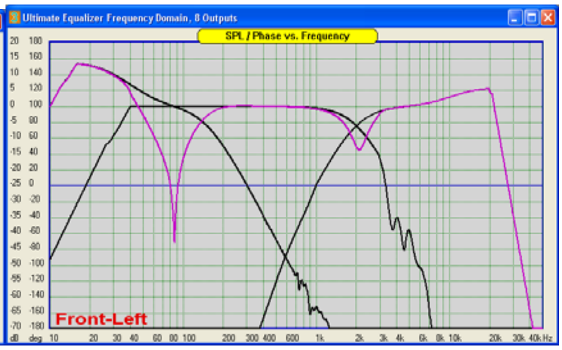


8 Integrating subwoofers

I prefer to engage the subwoofers for all audio configurations and listening sessions. Subwoofer integration, even with overlapping frequency band between the subwoofer and the woofer in the two-way system is very easy. There is no need for any phase adjustments or delays, because the phase is linear in every loudspeaker in the system. Pictures below show my attempt to integrate a subwoofer into a stereo front-left channel. My subwoofer is placed in the corner, and the two-way tower is right next to it (see page 1). The measurement microphone is placed 1m in front of the boxes and equidistant between the two enclosures.

In linear-phase mode, the resulting SPL is just a sum of the SPL from each loudspeaker – so the integration is perfect without any corrections. However, in minimum-phase mode I seem to have a 180deg phase difference between woofer and subwoofer at 80Hz (sharp dip), and this has to be accounted for. I have listened to both configurations, and the minimum-phase version without correction sounds quite bad.





Added Sub in Linear-Phase mode Added Sub in Minimum-Phase Mode

9 Latency

Linear-phase has more latency (145ms) than minimum-phase (65ms). But this is not necessarily bad. Here is what happens in my own system. I have 3D DVD Player from Samsung, BD-C6900, connected via HDMI to Samsung 3D 55" LCD screen (UA55C700). When I play a DVD with the sound through in-built TV speakers, the sound is in-synch with the picture. However, when I play a DVD and take the audio directly from the DVD player’s analogue 5.1 RCA outputs, the audio is advanced by 200-250ms. This is apparently due to the amount of video processing taking place in the LCD screen. In the first case, the LCD screen compensates internally for the advancing audio, but it will not compensate in the second case, because the audio is supplied directly by the DVD player.

It turns out, that I need more audio latency, therefore future UE releases will introduce option to go to 290ms latency. As I understand, this is quite common these days with 3D devices and HDMI. If you experience no picture delay in your setup, you have two options: (1) put up with 145ms latency, and this is quite OK, as you will only notice the latency sporadically. I have watched many hours of DVD movies and the 145ms latency is still quite acceptable, or (2) switch to minimum-phase mode, where you will not notice the latency.

Virtually all contemporary AV receivers have an audio delay capability up to 200-300ms. This is in response to the same issue as I experience in my DVD/TV system. If the latency introduced by UE3 is not sufficient for your application, there are delay units available from: http://audaud.com/2007/07/3-lip-sync-error-delay-units/

10 Volume level

It looks like my calculations for the required power output to secure THX loudness level had some serious margin in it. First of all, the UE system is ***seriously, loud***, and I can only stand full volume level for a few moments to complete the demo. Furniture is shaking, and picture frames are moving on the walls. There is no distortion, even at full volume level. It's just extremely loud, very powerful, clean sound.

I have re-designed volume control slider. It's now calibrated and displayed as "gain" from -51dB to +12dB. Steps are +/-3dB or +/-1dB. The lowest level is actually silence. It turns out, that my preferred listening level is -30dB. Now, I could perform much more rigorous analysis of the input and output levels, which lead to further ideas of improvement.

Another issue relates to recording level across CDs and DVDs. I have found this level to be significantly varied across my music and movies collection. In order to maintain S/N of the output A/D converters, and allow it to achieve full voltage swing on quieter recordings, the UE should allow for inserting greater gain. UE3 offers +6dB gain, but this is still not sufficient in some instances, therefore maximum UE gain is now +12dB.

During listening tests, I also have found, that LFE channel level varies quite significantly from one movie/music DVD to the next – by as much as 10-15dB. LFE setting, that seems perfect for one movie, were too much for another. I have therefore implemented a feature allowing the user to nominate two outputs (presumably two subwoofers) and control their level by additional +/-20dB (in 2dB steps) during the playback. This feature is also remotely controlled, and together with the remote master volume control, it is a major user's convenience feature of the UE system.

Functional Improvements

Functionally, the UE3 system changes the way you acquire, organize and play your music and movies. The system is also much **cheaper** to run and to add new music/movies, and is much **easier** to run, by using play-lists. It has actually changed my habits of dealing with music. Windows Media Player allows me to play a whole heap of different musical file formats in the same library.

I found myself using the new system much more often, as it is just fun to play a full library of songs, without ever changing a CD. I also found myself listening to the actual music again, rather than trying to find sonic imperfections in the old playback system. You can easily create "The Best Of….." style music libraries of all your favourite artists, and enjoy uninterrupted music for several hours.

I still use an external DVD player for 3D movies, but this will change in foreseeable future.

PC Integration

Windows7 PC with 128Gb SSD has become irreplaceable. It is my audio server and DSP processor in one box. I have several finely tuned system-wide, linear-phase “sound profiles”, (well, they are actually project files) that I load into UE3 depending on which “vintage” of my music libraries, movie-DVD or concert-DVD I play. As a result, the sound is much more balanced across my entire music/sound collection, with subwoofer channel being active at all times.

Ability to change the characteristics of the whole system (channel gains, delays, input re-routing, loudspeaker configurations re-routing, crossover frequencies and all voicing) almost instantly by using pre-defined “sound profiles” was something new, and very easy to implement with DSP PC. It is the presence of the PC, which is permanently inserted into the audio system chain, which enabled this irreplaceable feature. I’ve really come to appreciate it, as prior to this, many of my CDs were almost forgotten, because I did not care to listen to “less-than-perfect” tonally balanced recordings. You see, knowing that each of my loudspeakers has absolutely flat, first arrival, on-axis frequency and phase responses, gives me an opportunity to correct shortcomings of studio mastering process (all in linear phase), rather than endlessly and needlessly tweak my speakers. This is one of those “hidden” benefits, that I only discovered and came to appreciate after the system was being fully auditioned for a while.

In addition, the remotely controlled LFE channel has proven itself to be very handy indeed.

The PC has been very successfully integrated into the AV system, and has facilitated many additional functions in my system. It is clearly the heart and the brain of my system. The role of PC will be expanded in the future, as I have not implemented all extra features yet.

Role of SoundEasy

SoundEasy V18 has been extensively used in box design, modelling and measurements, including vent design and enclosure dimensioning. Final enclosure tuning (port length adjustment) was performed during the system measurements phase.

Secondly, SoundEasy was used to verify the performance of UE3 equalized loudspeakers. If you performed all design and measurement steps accurately, the verification will show flat frequency response and flat phase response on all drivers and loudspeakers. All design curves (SPL/Phase measurements, HBT, inverted HBT etc…) are available on the design screen, and if all checks out during the design process, there is nothing much that can go wrong.

Multi-channel amplifier

This device turned out to be a significant and important part of the UE3 project’s success. The idea of using a separate, dual-channel amplifier for subwoofers was accidental, but it was a right decision. Subwoofers in my system tend to work hard, and draw heavier current, so separating the amplification (and power supply) was the right thing to do.

Design and construction of the 10-channel amplifier was quite significant task. What made is much easier, was the usage of IC modules for 7 amplifiers. The 50W/8ohm modules worked straight away, and did not require any current adjustments, only gain adjustment and insertion of one extra capacitor in the feedback loop. The 100W/4ohm amplifiers took longer to assembly, they required a simple procedure to set quiescent current, and had to be slightly re-designed to maintain stability for low closed-loop gains. Putting it all together was possibly the most involving and intricate process. I found a number of issues, that I did not think about prior to assembly, like the need to drill heat-sinks to make threaded holes for bolting them down to the chassis. Or adding an extra angle bracket for this or for that. Chassis and front panel were made of 5mm aluminium, and was cut to size a local supplier. I tried to fit all parts into a 3U-size rack mounted off-the-shelf enclosure, but it would not work. My amplifier does not represent the pinnacle of modern amplifier technology – it's just class-AB analogue amplification. I am only familiar with this configuration, but I am quite sure, that professionally designed and built amplifier can be smaller and cheaper.

Associated issue is a number of speaker cables running around the room. This issue does require some planning and if you can hide some of the cables in walls or conduits, it will make some difference aesthetically. I used good quality, 4-core speaker cables in conduits colour-matched to skirting boards. There is also a number of RCA shielded audio cables between the PC, amplifiers and DVD player in my system. I used reasonably good quality cables.

Finally, I can't stress enough the importance of avoiding ground-loop problems. It may not only be the hum/buzz, that will disappear, there may also be some other noises, that will be removed from your system. During my experiments, I have tried one-sided "ground-lifted" RCA cables – that did not work. You may need to plan or experiment with AC power distribution in your system. For further information, please visit: http://www.epanorama.net/documents/groundloop/

Further Improving the PC System

In the PC world, things are changing rather fast. CPUs are being updated, PCI slots are being replaced with PCIe slots for faster, better card performance. Sound card manufacturers are sometimes left behind, but it's only a matter of time before they will catch up – or go out of business. I am not qualified to give any technical advice here, other than encouraging you to seek knowledge. For example, there are companies promoting PCs optimized for music reproduction in your AV system.
http://www.endpcnoise.com/cgi-bin/e/std/sku=mcubed_music_server.html

And companies promoting the PC music server approach.
http://www.goodwinshighend.com/music_servers/faq_slient_music_server.htm

Where To Go Next?

Having completed this project, down to the nuts-and-bolts, it's very unlikely, that I will ever built another sound system in my life. As presented on page 1 of this paper, this system does it all for me. I am more interested in making future incremental improvements to this system, as faster PCs with even better audio characteristics and new sound cards become available, and new, (and cheaper?) drivers come out with improved performance. I will also continue to progress with the Ultimate Equalize Technology.

And I will never have to design, re-design and re-built any crossover for new drivers - UE will do this for me. Amplifiers are also expected to last, but they may need new electrolytic caps in some years to come. Perhaps I will expand to 7.2 version too. Current system PC life-span is estimated to be around 4 years, which is when I intend to upgrade it. After that, I will be on the lookout for improved drivers, and this is when small incremental changes will start taking place. Well, I may also re-paint the boxes then.

From now onwards – it's just music, movies, concerts…..

So, that was my paradigm change. Thank you for reading.

Bohdan

AV room - front speakers with grilles

UE4 system – front speakers with grilles

UE4 – rear speaker

Appendix

Some interesting comments about surround sound from:
http://www2.grammy.com/PDFs/Recording_Academy/Producers_And_Engineers/5_1_Rec.pdf

"…….4.1 Music vs. Sound For Picture

Human beings tend to rely more on visual than aural cues; we focus more on what we are seeing than what we are hearing, which is why we often close our eyes when listening critically to music. Multi-channel audio can be used to accompany picture (i.e., film scoring and sound design or soundtrack accompaniment to concert video) or it can be used to present music on its own (i.e., DVD-Audio or SA-CD Multi-channel release). The production approach to each can be quite different.

When mixing multi-channel sound for picture, the audio follows the action on screen, and not vice-versa. The goal is to match the sound to what you are seeing; thus, in a concert video, if a musician is featured prominently onscreen, the tendency is to raise the level of that instrument. Conversely, if an instrumentalist is at the back of the stage or largely off screen, the tendency will be to tuck that sound in. In feature film production, wherever there is dialog (which most often will be anchored in the centre speaker), care must be taken to reduce the overall level of accompanying music and effects so that the spoken word can be clearly heard. In some cases, equalization has to be applied to midrange content of those tracks in order to carve out frequency notches which might be fighting the dialog. Separate mixes may be undertaken for cinematic exhibition and home theatre release. In the case of the former, the music mix is normally delivered as stems so that they can be rebalanced by the film mixer against the FX and dialogue stems; in the case of the latter, provision may be made for the fact that the centre and/or rear speakers may not be full range.

Perhaps most importantly, the audio content in sound for picture projects is determined ultimately by the director or producer of the film and not by an audio specialist. This can have both positive and negative implications. On the one hand, there is little fear of using radical panning or dynamic fly-overs from speaker to speaker; on the other, there may be less attention paid to musical content than there is to dialog and effects.

Because there are no visual cues to follow, mixing surround sound for music only can be a much more freeing experience. However, initial music-only multi-channel mixes were relatively conservative compared to today's mixes. The rear speakers were often used only to carry room ambience or effects returns, and the centre channel often carried only one or two select instruments, with other instrumentation placed statically in the front left-right speakers as if it were a standard stereo production. Recent years have seen a trend towards a more inventive approach, with more use of the rear speakers to carry significant musical content, greater flexibility in instrument placement and panning, an increased awareness of the importance of creating ambient spaces through de-correlated effects, and more willingness to pump up the LFE channel where musically appropriate……"