Ultimate Equalizer V10.0 Supplemental User Manual

`August 2026

New Features Implemented in Ultimate Equalizer 10.0.

1. The size of project data files is reduced to half of the V9 size. This is quite substantial change, and allows for faster loading of project files.

2. MLS system uses WASAPI 16bit/24bit drivers instead of the legacy WDM/MME drivers.

3. RoomEQ measurements use WASAPI 16bit/24bit drivers instead of the legacy WDM/MME drivers.

4. Added second version of the BBM system. The FL channel is independently filtered and summed with Left Subwoofer, and FR FL channel is independently filtered and summed with Right Subwoofer.

5. Current Bass resolution settings (buffer-size dependant) are displayed on MLS transfer function screen.

6. Bit depth on the input-side can be set to 16bit or 24bit. Many contemporary motherboards only allow for 16-Bit depth due to compatibility with CD recordings requirements.

7. MLS system uses the same impulse response files as SoundEasy V21. The UE10 and SoundEasy V21 programs are compatible for impulse response measurements.

8. WASAPI Debug Mode. Useful to debug sound card drivers implementation.

9. Automatically appends buffer size and sampling frequency to driver's filename. Also, appends sampling frequency to impulse response filename.

10. Improved Complex and Octave smoothing algorithms.

11. Ability to review logarithmic SPL/Phase in MSL system

This User Manual is intended to provide information about new features in version 10, and should be read in conjunction with the full User Manual for version 5, 6, 7, 8, and 9 available from Bodzio Software Pty. Ltd. website.

WASAPI Debug Mode

Wasapi Debug Mode is selected from Preferences screen as shown below.

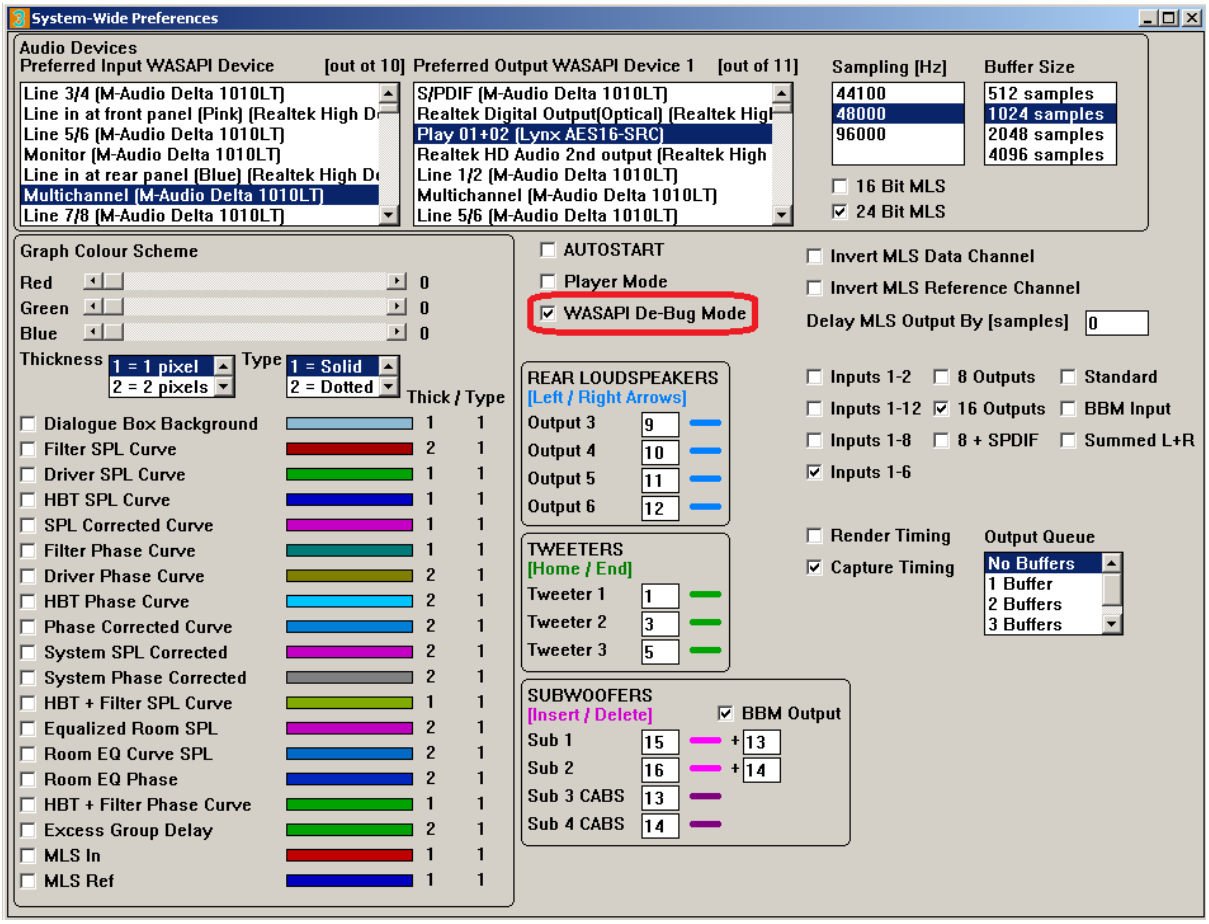


Figure 1. Selecting WASAPI Debug Mode.

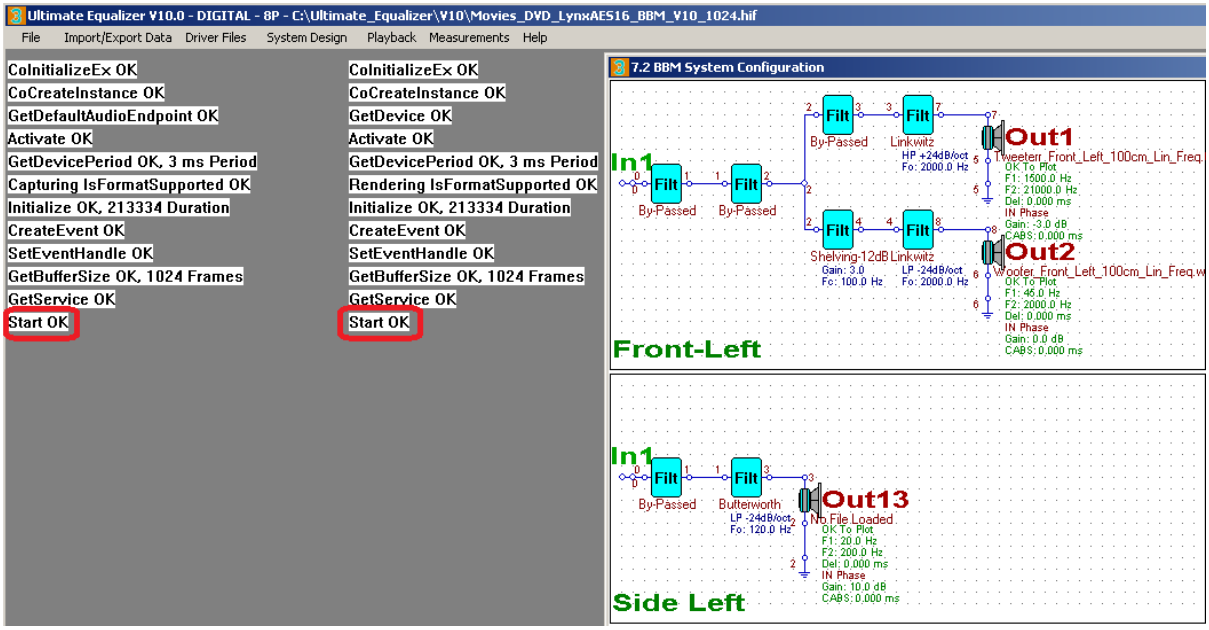


Figure 2. UE10 System Design screen displaying WASAPI information.

Then, proceed to the System Design Screen as usual, and plot all sub-systems responses. You will need to move the main System Configuration screen to the right, so a free section of the main screen is exposed to provide display of initialization progress of both: Capturing and Rendering WASAPI devices.

Important result to look for is the “Start OK” indication of both WASAPI devices: Capturing and Rendering – as shown on Figure 2.

Real-time SPL level adjustments

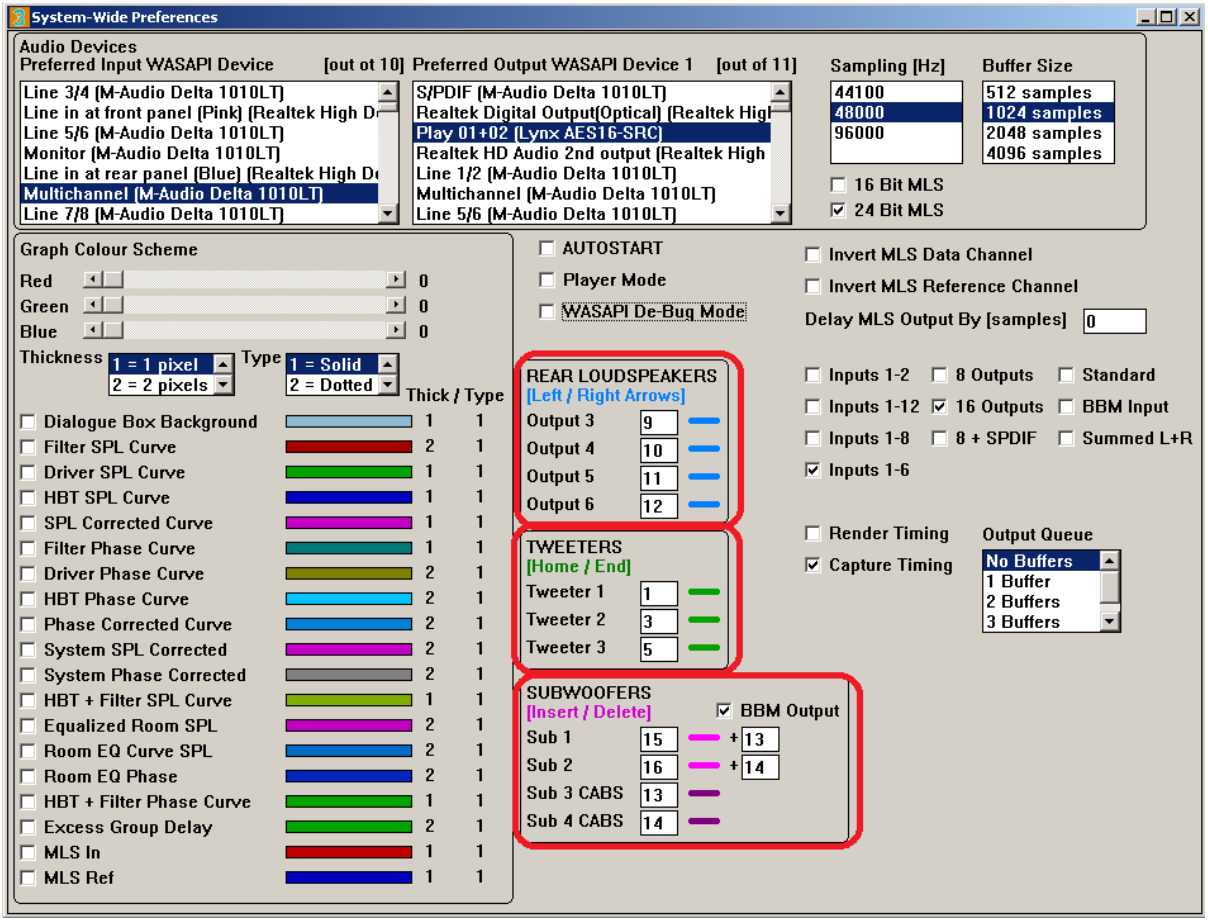


Figure 3. Preferences screen with outputs selectable for adjustments.

Important Note: When you first open the Preferences screen, the residual data displayed in the fields marked above may contain random value. Please assign your correct output ports to these data fields immediately.

Adjusting Master Volume

Master Volume control is performed remotely by pressing ARROW UP/DOWN keyboard keys. The SPL steps in 3dB increments/decrements.



Adjusting Volume of 4 Nominated Drivers (Rear Speakers)

This is a very handy and often used feature. It appears, that rear loudspeakers are not mastered in the recording studios at the same level. Thus some movies/concerts are lacking the sound coming from the rear speakers, and others are correctly edited. To remedy this problem, the program allows you to nominate 4 output ports (these will be your 4 drivers in the rear speakers) in the Preferences screen – see example below.



Adjusting Volume of 4 Nominated Drivers (Subwoofers)



Subwoofer levels are adjustable in +/-2dB steps. Up to 4 subwoofers can be nominated, so this includes the Left Subwoofer, Right Subwoofer and corresponding CABS sinks. INSERT (+2dB) and DELETE(-2dB) keys are used for this purpose.

Adjusting the BBM levels

Binaural Bass Management is a very useful feature allowing you to correct low-frequency content and balance between LFE channel and Front-Right and Front-Left channels.

Live concert reproductions can be enhanced by optimizing the usage of subwoofers in your systems. If the capabilities of LFE channel are not being fully utilized in the mixing process, the total listening experience can be diminished and simply lack the “bottom-end”. This can be particularly evident in “live rock” DVD recordings. Attending live concert impresses the patrons with a unique and memorable experience. Most people would comment about how “loud, solid and punchy” was the sound mix. Unfortunately, listening to the same concert on DVD often leaves quite different impression. For example, bottom-end balance in LFE channel features overpowering kick-drum and lack of bass guitar – this is almost opposed to what was experienced during the live performance.

BBM feature allows you to some extent re-balance the low-frequency mix, and feed the powerful low-frequency sound into the loudspeakers, that are built for it – the subwoofers.

Typical example here would be extracting bass guitar from the front channels and mixing it with the kick-drum in LFE channel to generate Left Subwoofer and Right Subwoofer signals.

BBM Input-Side

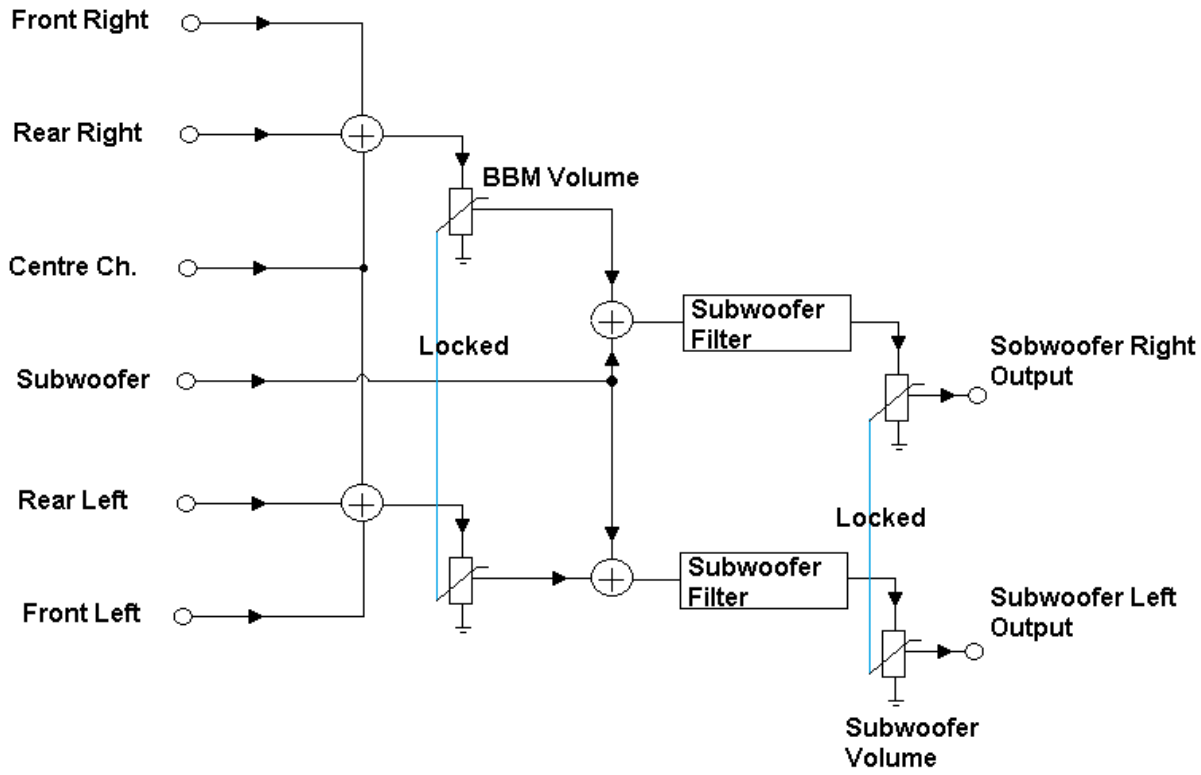


Figure 4. Processing diagram for BBM Input Side.

This configuration uses all input channels and controls the level of low-frequency (filtered from all-left channels and all-right channels) signal mixed into left and right subwoofers. FR + RR + Cntr channels have their own level control prior being mixed into the RIGHT subwoofer channel. Also, FL + RL + Cntr channels have their own level control prior being mixed into the LEFT subwoofer channel.

Selection of the BBM Input Side option is accomplished via Preferences screen, by selecting “BBM Input” check box – see Figure below.

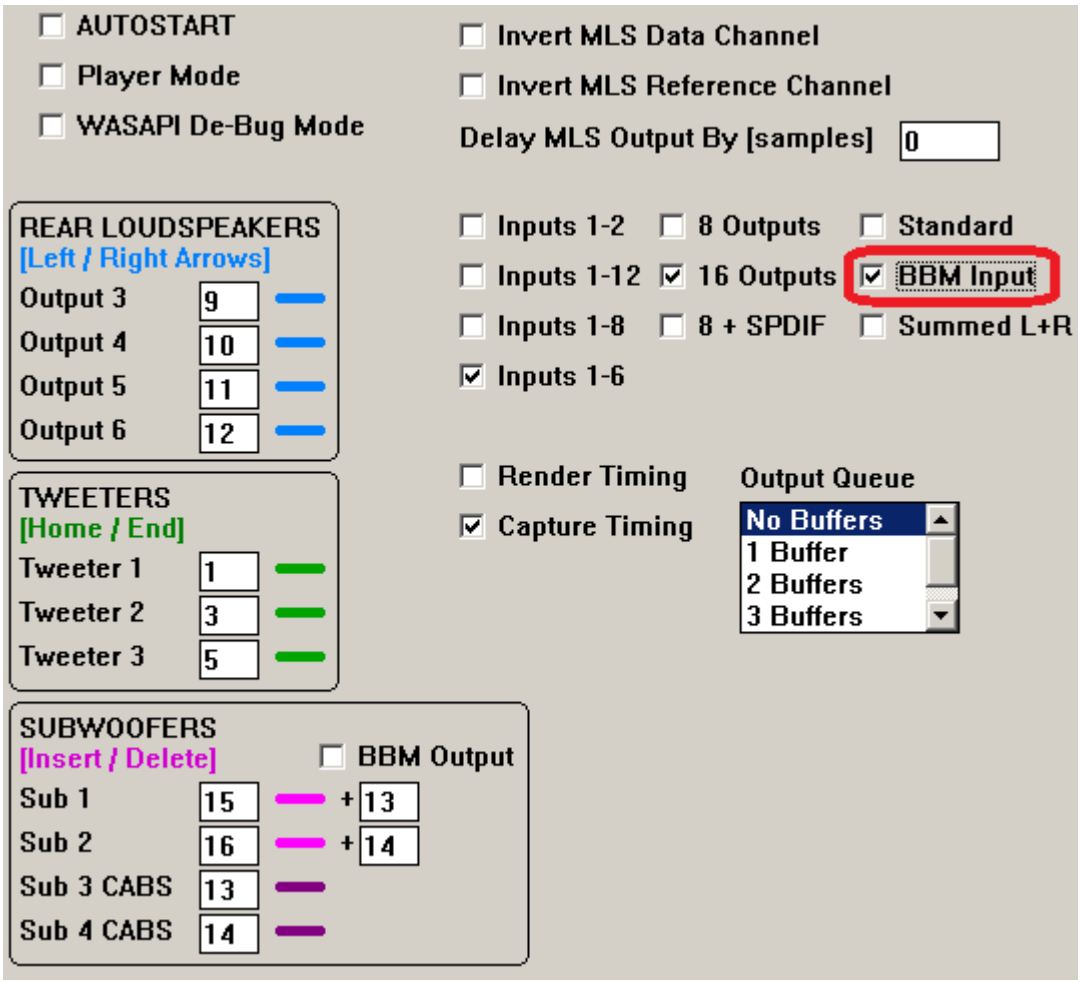


Figure 5. Selecting BBM Input Side.

Care needs to be taken when all input surround channels run at high level. There is a possibility of saturating (clipping) of the signal when adding all channels together.

BBM Output-Side

This configuration is also selected from the Preferences screen, and only extracts low-frequency information from Front Right channel and Front Left channels.

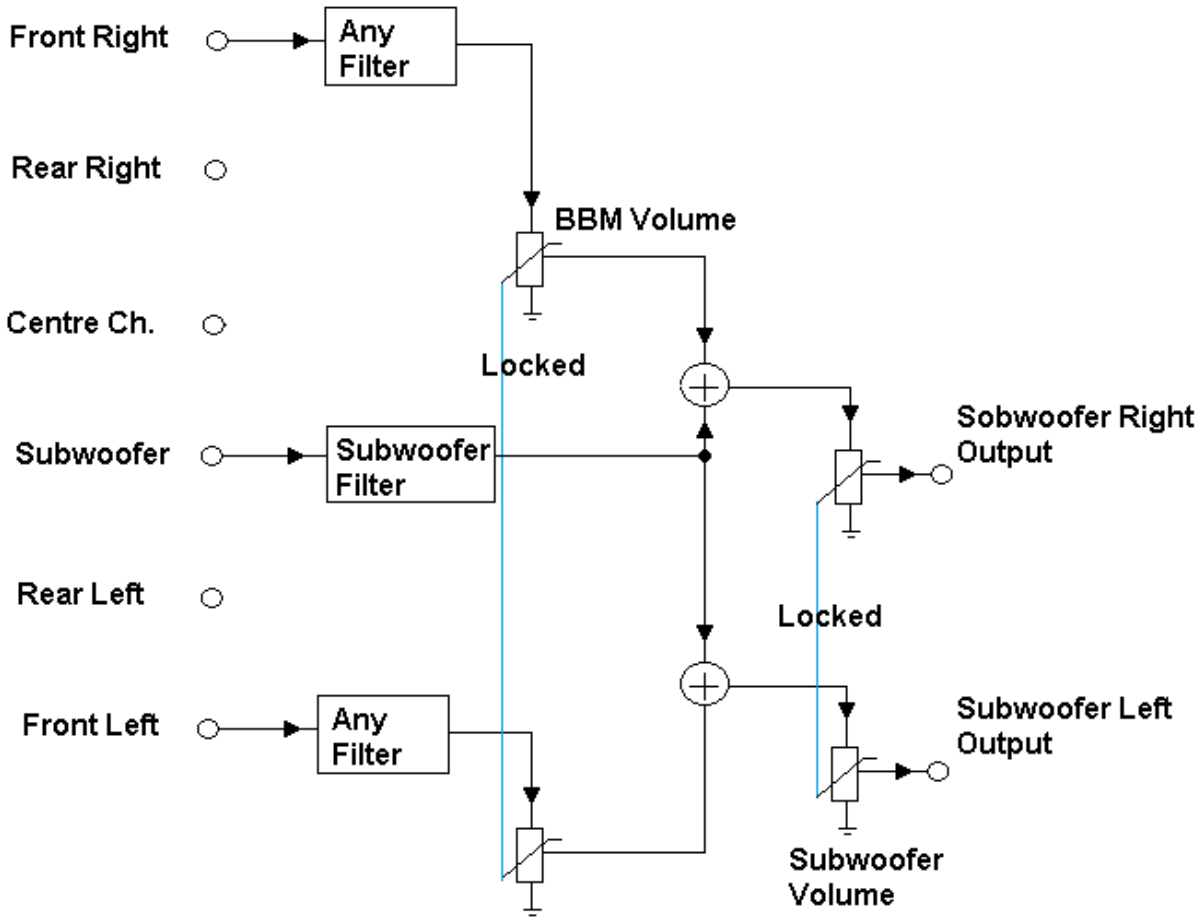


Figure 6. Processing diagram for BBM Output Side.

FR channel has its own filtering provided (it can be any filtering arrangement, even cascaded filters) and their own level control prior being mixed into the RIGHT subwoofer channel. Also, FL channel has its own filtering provided (it can be any filtering arrangement, even cascaded filters) and their own level control prior being mixed into the LEFT subwoofer channel.

This BBM processing option is much safer from the "signal overload" point of view, as all the input signals are filtered prior being summed together.

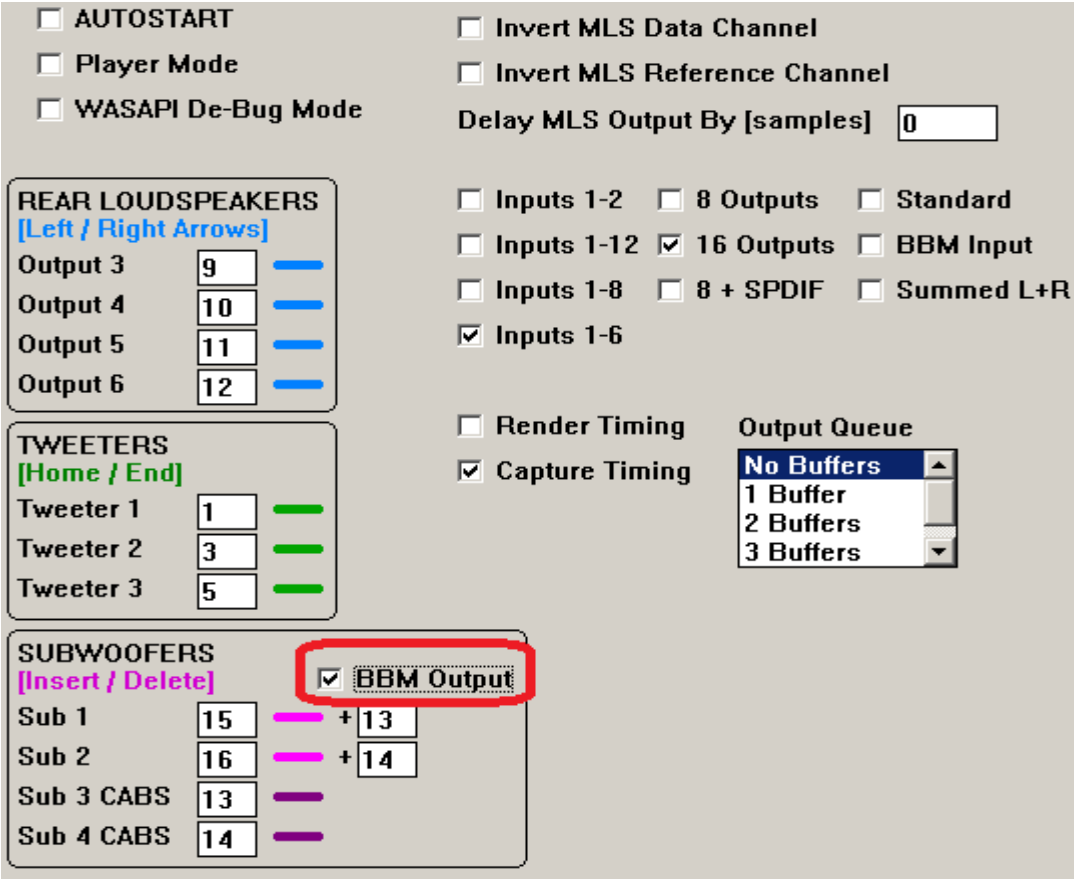


Figure 7. Selecting BBM Output Side.

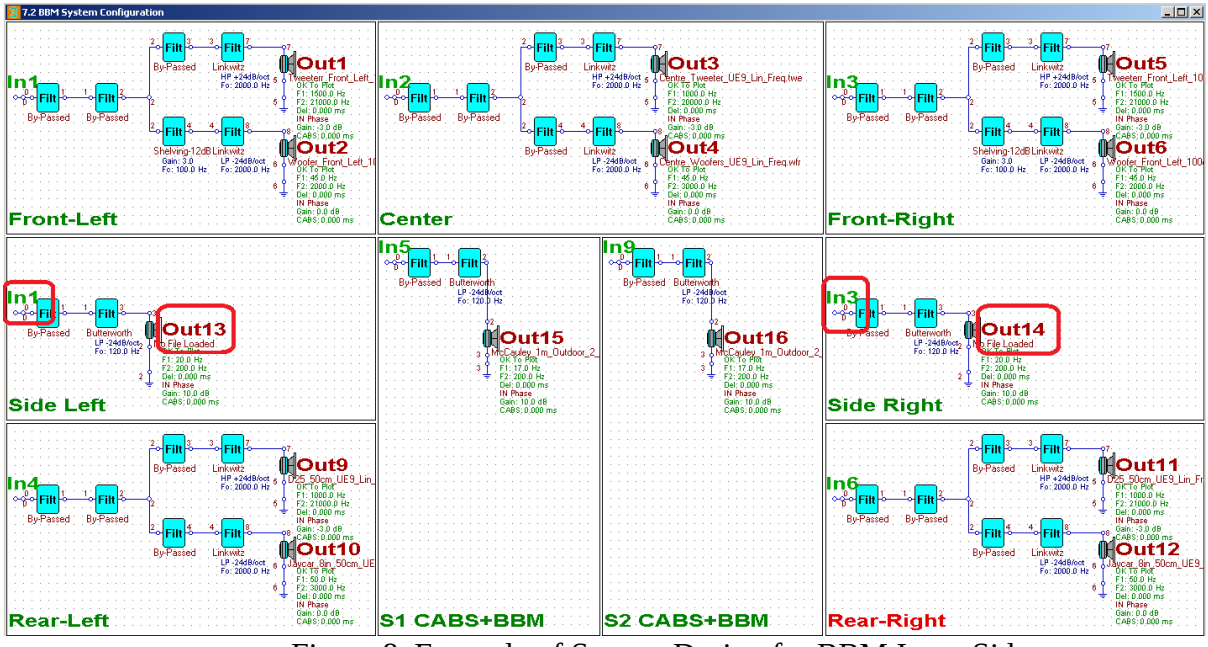


Figure 8. Example of System Design for BBM Input Side.

It is observable, that BBM-Left is the signal processed between Input 1 (Front-Left) and Output 13. BBM-Right is the signal processed between Input 3 (Front-Right) and Output 14. Figure 7 above shows, that Output 13 is summed with Output 15, and Output 14 is summed with Output 16.



Page Up = +2dB
Page Down = -2dB

Adjusting Volume of 4 Nominated Drivers (Front Tweeters)

This particular adjustment option came to being after auditioning many movies and concerts. There seems to be a subtle difference (well, sometimes quite pronounced) in the amount of treble recorded during the final studio mix. Some video material has excessive brilliance and sibilance, and these can be controlled to large extent by controlling the level of tweeters in front speakers.

Brilliance - The 6kHz to 16kHz range controls the brilliance and clarity of sounds. Too much emphasis in this range can produce sibilance on the vocals.

Sibilant - "Essy" Exaggerated "s" and "sh" sounds in singing, caused by a rise in the response around 6 to 10 kHz. Often heard on radio.

In order to be able to adjust those level during playback, you need to nominate the outputs to which the tweeters are connected. In the example below these are: Out 1, Out 3 and Out 5. Output 7 is a dummy one, and nothing is connected to it.

The levels are controlled via “Home” and “End” keys on the keyboard in +/- 1dB steps.



Home = Tweeters -1dB
End = Tweeters +1dB

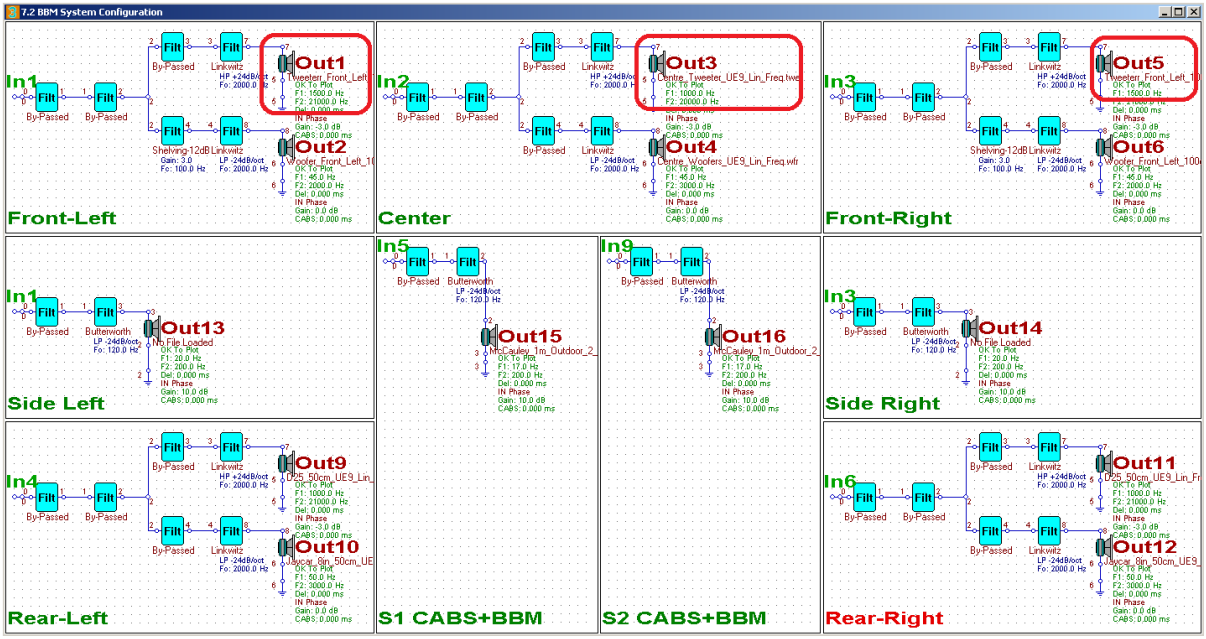


Figure 9. Example of a system with front tweeters assigned to outputs 1,3 and 5.

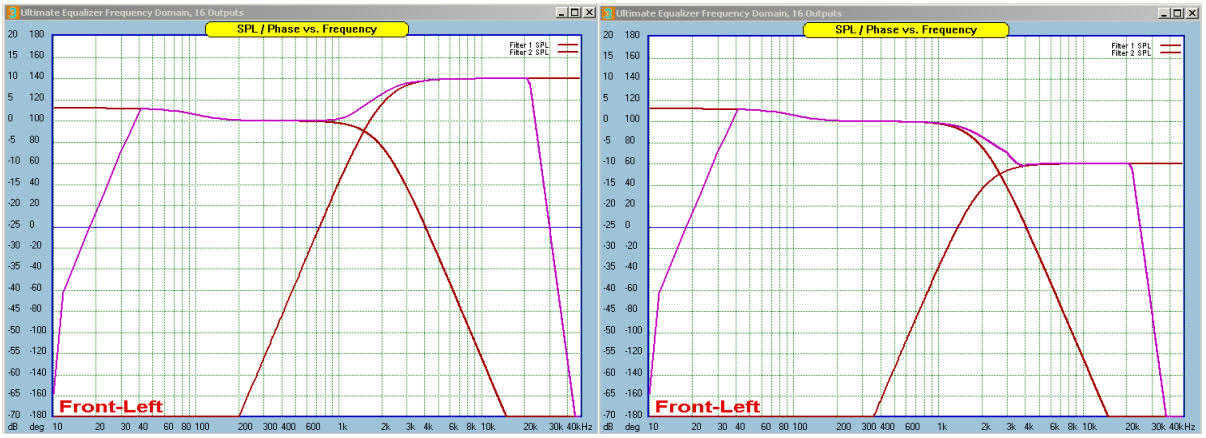


Figure 10. Tweeter at +10dB and tweeter at -10dB level.

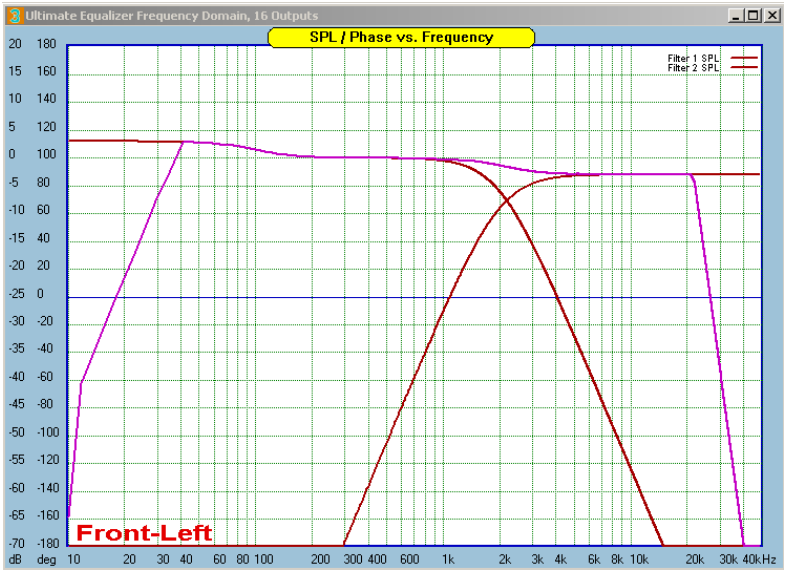


Figure 11. Commonly used voicing with 6dB difference between 50Hz and 10kHz.

Motherboard Audio Codec Issues

Contemporary PC measurement system should be able to deliver 24bit sampling resolution. This is however dependant on sound card quality or motherboard audio codec – if you decide to use motherboard audio.

Older sound cards (but still available on eBay) like Delta1010LT will provide 24bit sampling resolution on both: recording and playback side. Also good quality motherboard audio codec like ALC1150 Audio Codec supports 24 bit sampling for both: “Line In” input and outputs – see below. As an example, the ALC1150 codec is implemented in GIGABYTE Z97X-UD5H motherboard.

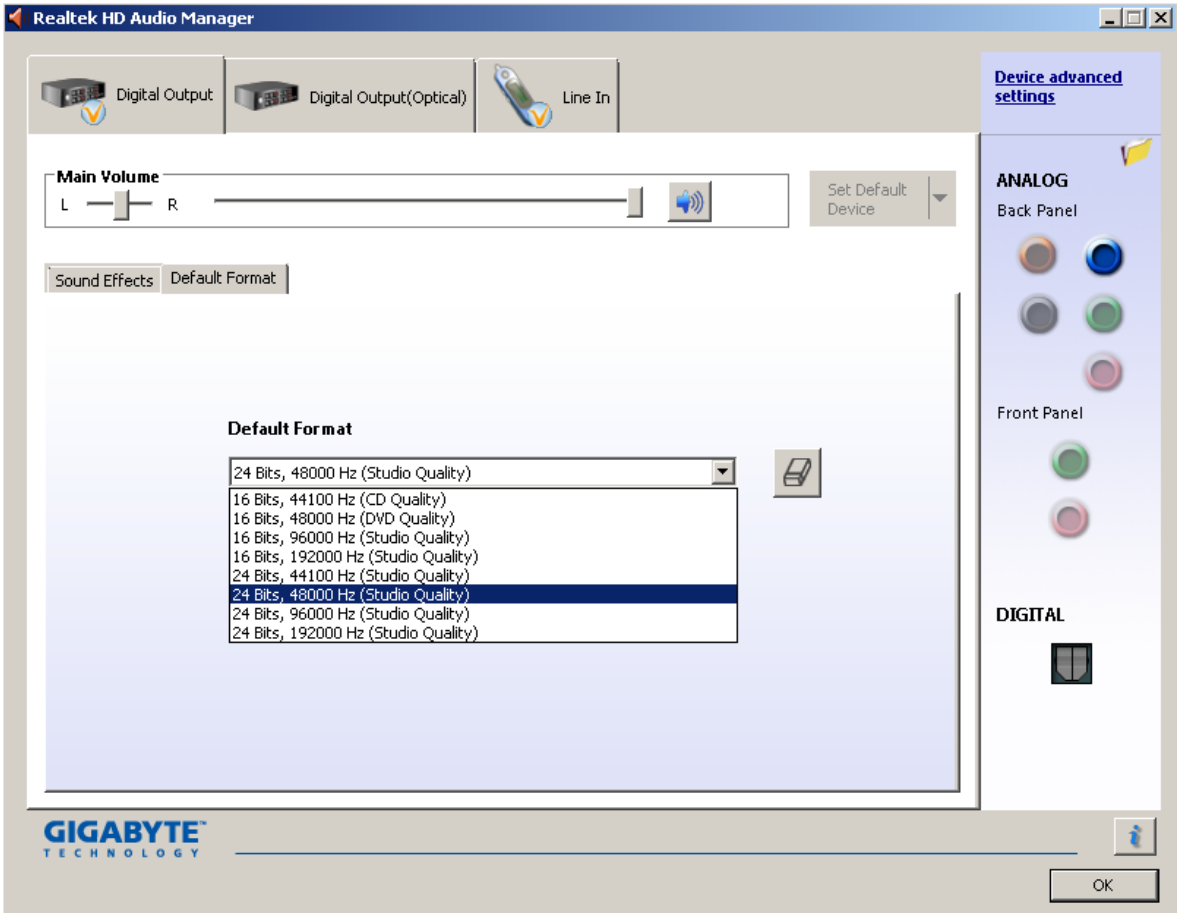


Figure 12. Realtek Audio Codec ALC1150 Line-In options.

Somewhat less fortunate implementation of audio codecs has been delivered in motherboards like ASUS P6X58D-E and GIGABYTE H170-H03. There is only 16bit sampling depth implemented on recording side, however, the full 24bits are available on playback side.

In order to help maximizing the overall performance of the MLS measurement system, the Preferences screen allows you to select 16bit/24bit option for sound cards. The selection is applicable to both: the recording AND playback of the measurement process. If your PC has only 16bit sampling depth available on the recording side, you MUST select “16 bit” option for the playback on your PC as well, and select “16-Bit Sound Card” from the Preferences screen.

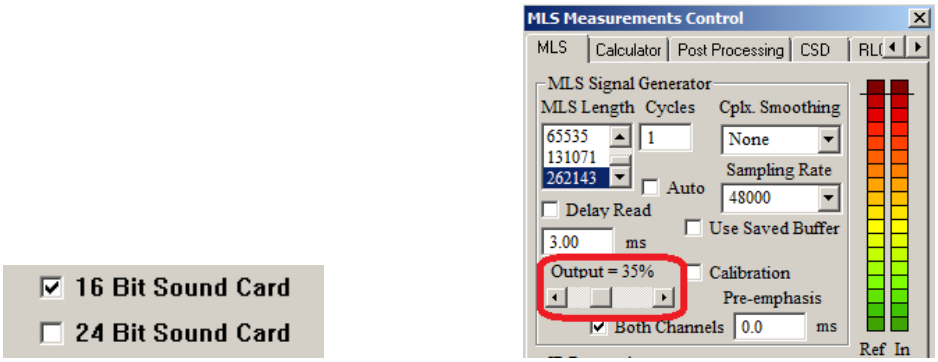
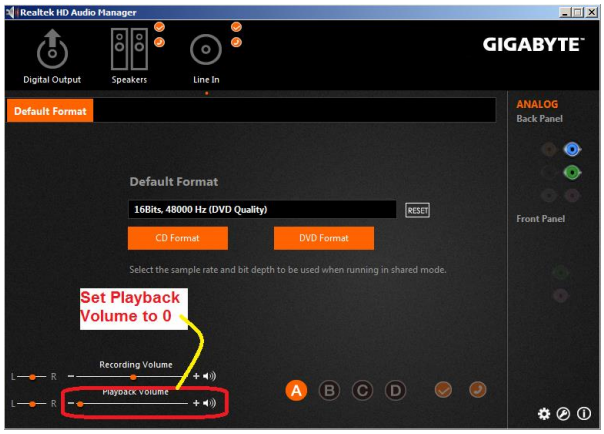
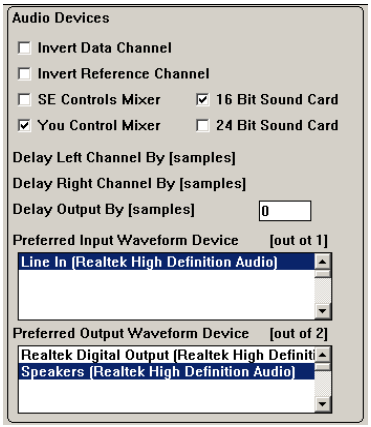


Figure 13. Preferences 16bit settings and MLS system audio level adjustment to 35%.

Please note, that due to MLS signal "crest factor", the output level should be less that 50% lower than maximum available output. It is recommended to set it to 34-35%. Shown below are two examples of motherboard audio codecs with 16bit sampling on the recording side and 24bit sampling on the playback side.

IMPORTANT - In the "Line Input" section of the sound card Manager, please set "Playback Volume" to 0 to prevent direct feed from the output to the input of the sound card.
GIGABYTE H170-H03 Recording Side – please note, Recording Volume set to 50%, Playback Volume = 0.





GIGABYTE H170-H03 Playback Side

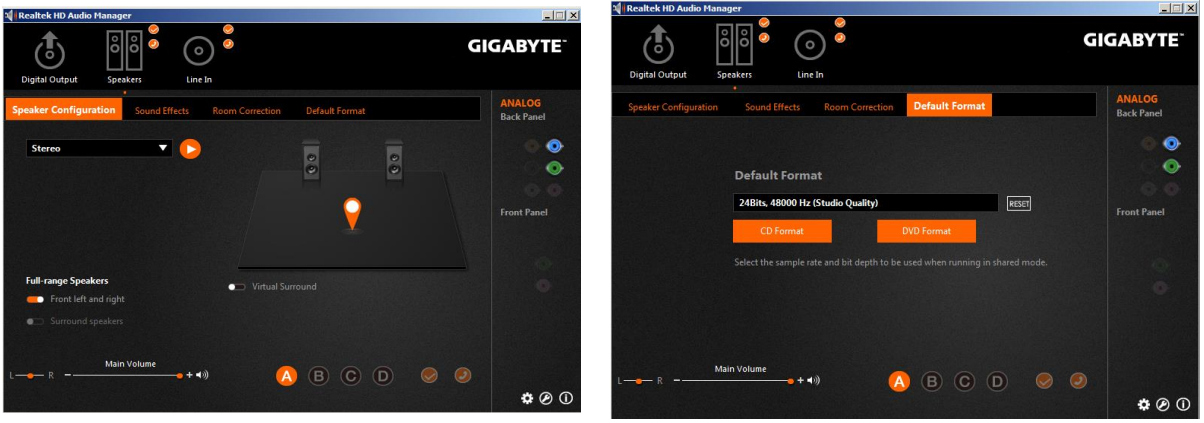


Figure 14. Audio Codec GIGABYTE H170-H03

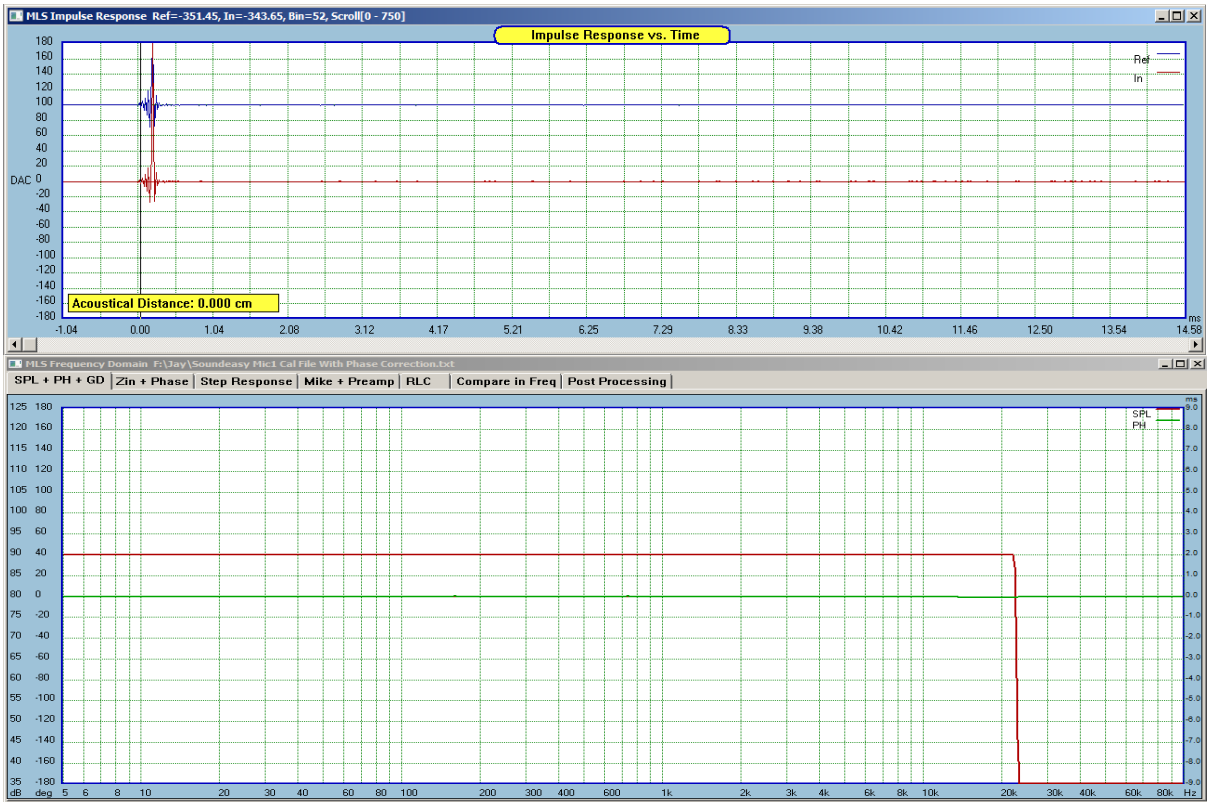
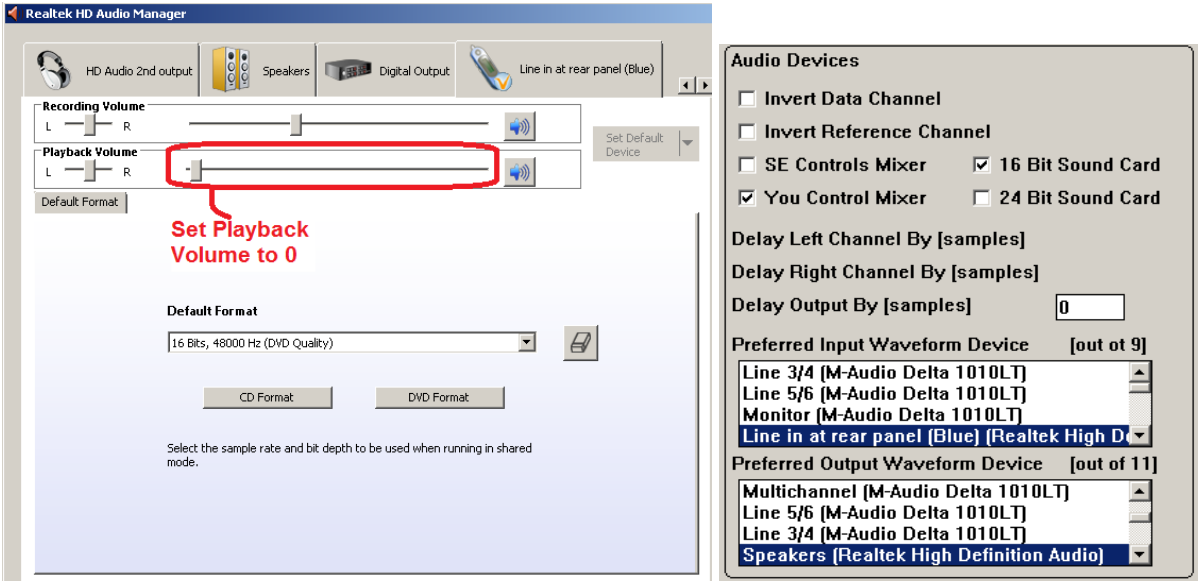


Figure 15. Resulting Loop-Test performance

ASUS P6X58D-E Recording Side - please note, Recording Volume set to 35%, Playback Volume = 0.



ASUS P6X58D-E Playback Side - please note, Main Volume set to 70%

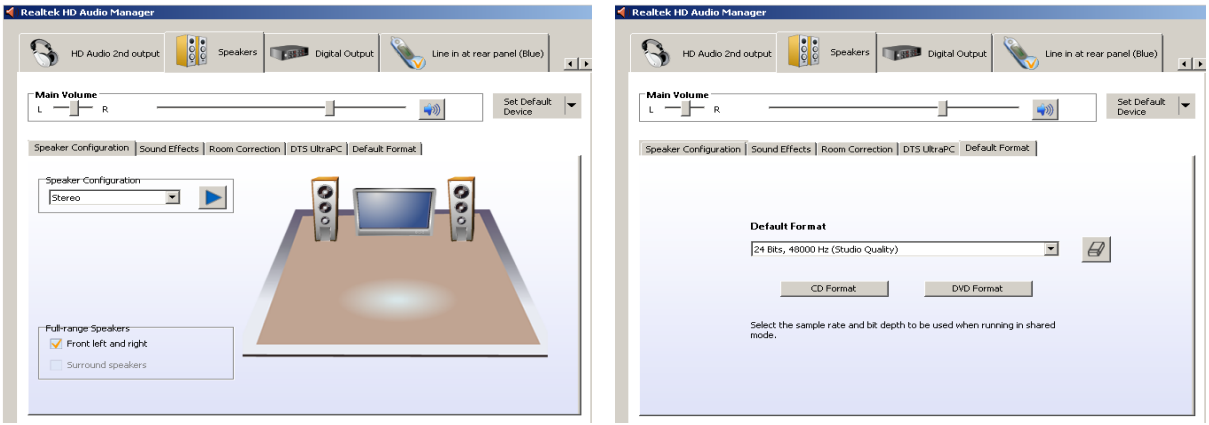


Figure 16. Audio Codec GIGABYTE H170-H03

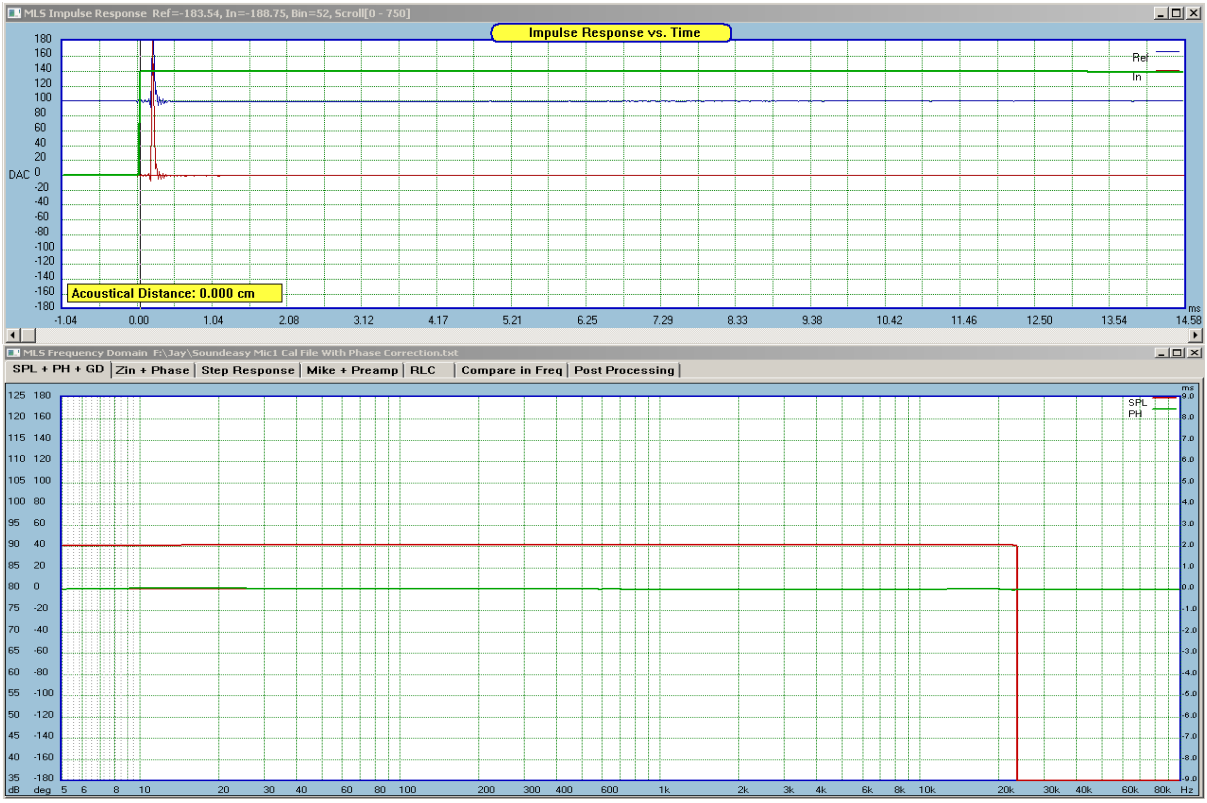
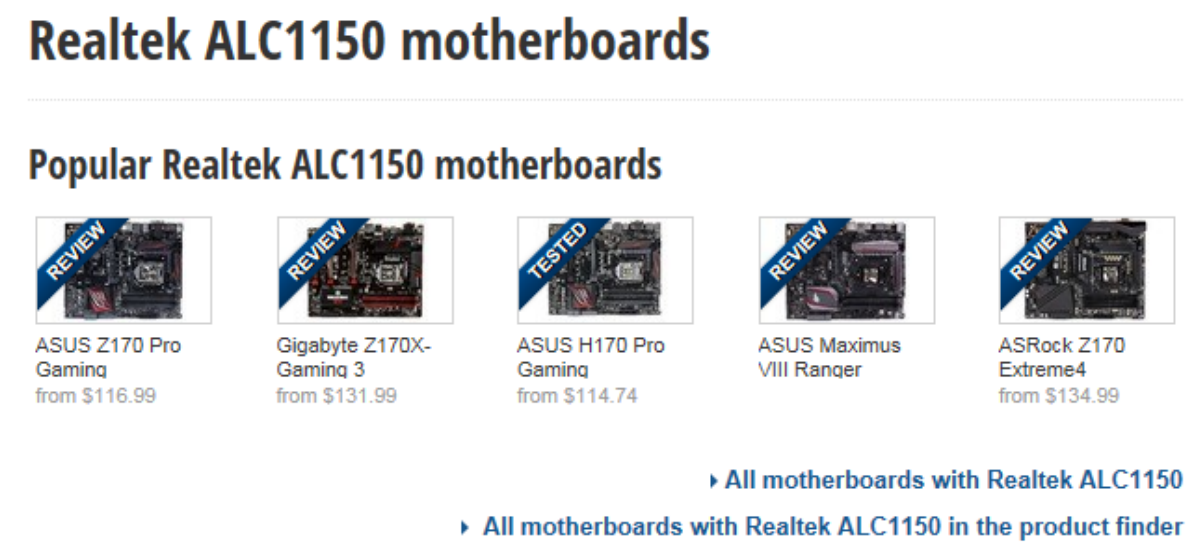


Figure 17. Resulting Loop-Test performance

It is strongly recommended, that prior to using Ultimate Equalizer software, you confirm the sampling rate and bit depth resolution settings with the sound card driver software or motherboard driver audio software for BOTH: playback and recording and set them to the same values.

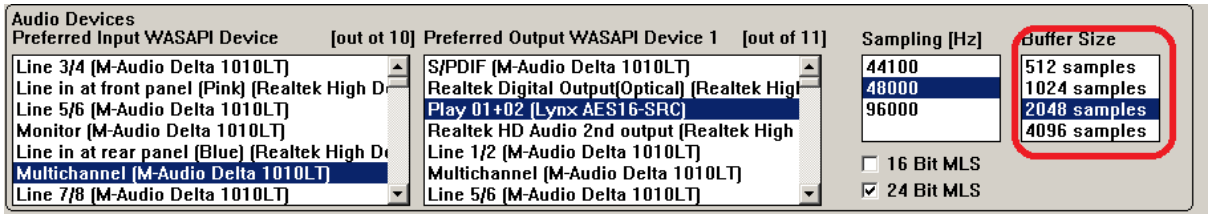
Interestingly, a website with a compilation of all available motherboards with ALC1150 coded lists 284 motherboards available. The ALC1150 codec is capable of outperforming typical sound card or CD-player for signal to noise ratio.

http://us.hardware.info/products/19/9372-realtek-alc1150-motherboards?page=2#allproducts



Bass Resolution vs. Buffer Size

Currently selected Bass Resolution settings (buffer-size dependant) are displayed on MLS transfer function screen.



For 48kHz sampling frequency and Buffer Size of 2048 samples, the bass resolution is 2.93Hz. The figure is displayed don the MLS SPL screen, together with vertical, dotted lines marking the exact location of the FFT frequency grid – see Figure 18.

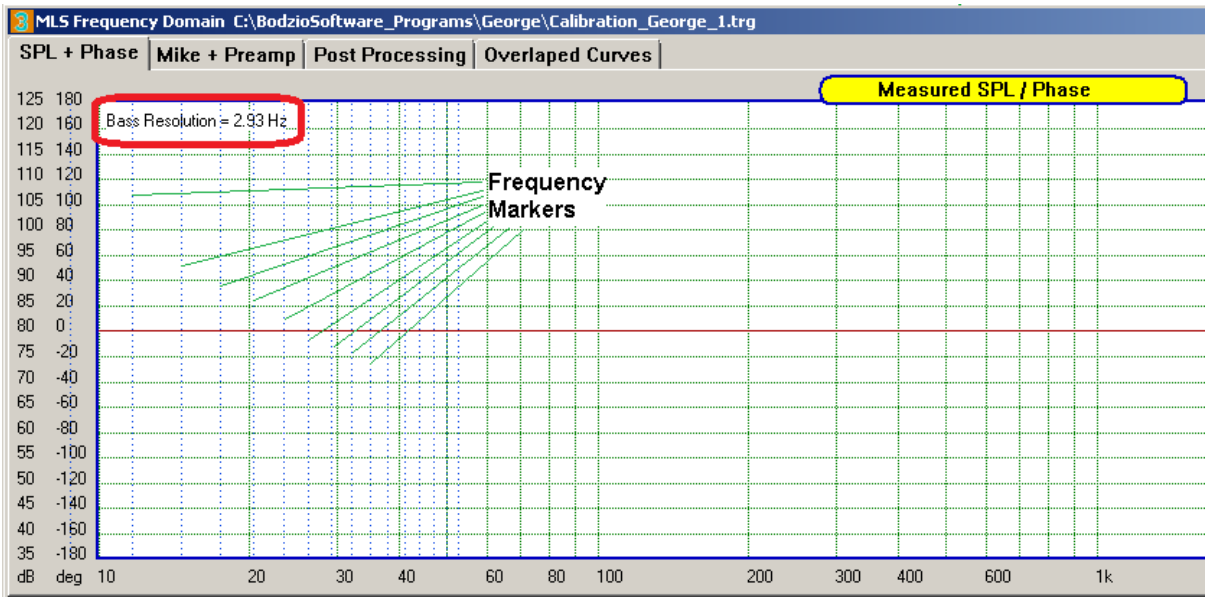


Figure 18. MLS SPL screen, with dotted lines marking the location of the FFT frequencies.

Automatically appends buffer size and sampling to filename

It has been recommended, that impulse response is measured at the same setup at three frequencies: 44.1kHz, 48kHz and 96kHz. This will secure usability of the impulse response file with future sampling options for the driver files.

In order to simplify saving of the impulse response file, the program will append information about the currently selected sampling rate in the MLS system to the driver's file name. The whole name is still available for editing, as per usual Windows File Saving mechanism – see Figure 18 below.



Figure 18. **"_48k**" appended if no file name has been entered.

Syntax of the portion appended to impulse response file is designed as follows:

_ single underscore character
XX two digits 44 – for 44.1kHz, 48 – for 49kHz, and 96 – for 96kHz, selected in Preferences screen
k single letter "k" for kilohertz.

Syntax of the portion appended to driver file is designed as follows:

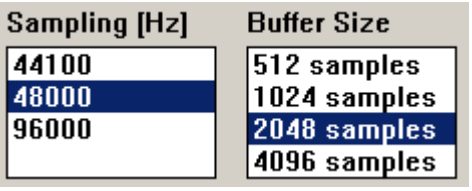


Figure 19. **"_1024b_48k**" appended if no driver file name has been entered.

_ single underscore character
XXXX four digits 0512 – for 512 UE Buffer, 1024 – for 1024 UE Buffer, 2048 – for 2048 UE Buffer and 4096 – for 4096 UE Buffer selected in Preferences screen.
b single letter "b" for buffer.
_ single underscore character
XX two digits 44 – for 44.1kHz, 48 – for 49kHz, and 96 – for 96kHz, selected in Preferences screen
k single letter "k" for kilohertz.

Complex Smoothing Part 2

Complex smoothing technique was described in UE9_Manual.pdf, starting from page 9. UE10 implements updated algorithm for complex smoothing, which works in conjunction with standard SPL Octave smoothing process. It is often beneficial to experiment with both smoothing algorithm's settings to obtain desired level of smoothness in both: SPL and Phase responses. In addition, the low-frequency response accuracy (bass resolution) will depend on the sampling frequency and buffer size settings activated on the Preferences screen.



Bass resolution = Sampling/(8*Buffer Size)

In the example below, Bass resolution = 48000/(8*2048) = 2.93Hz

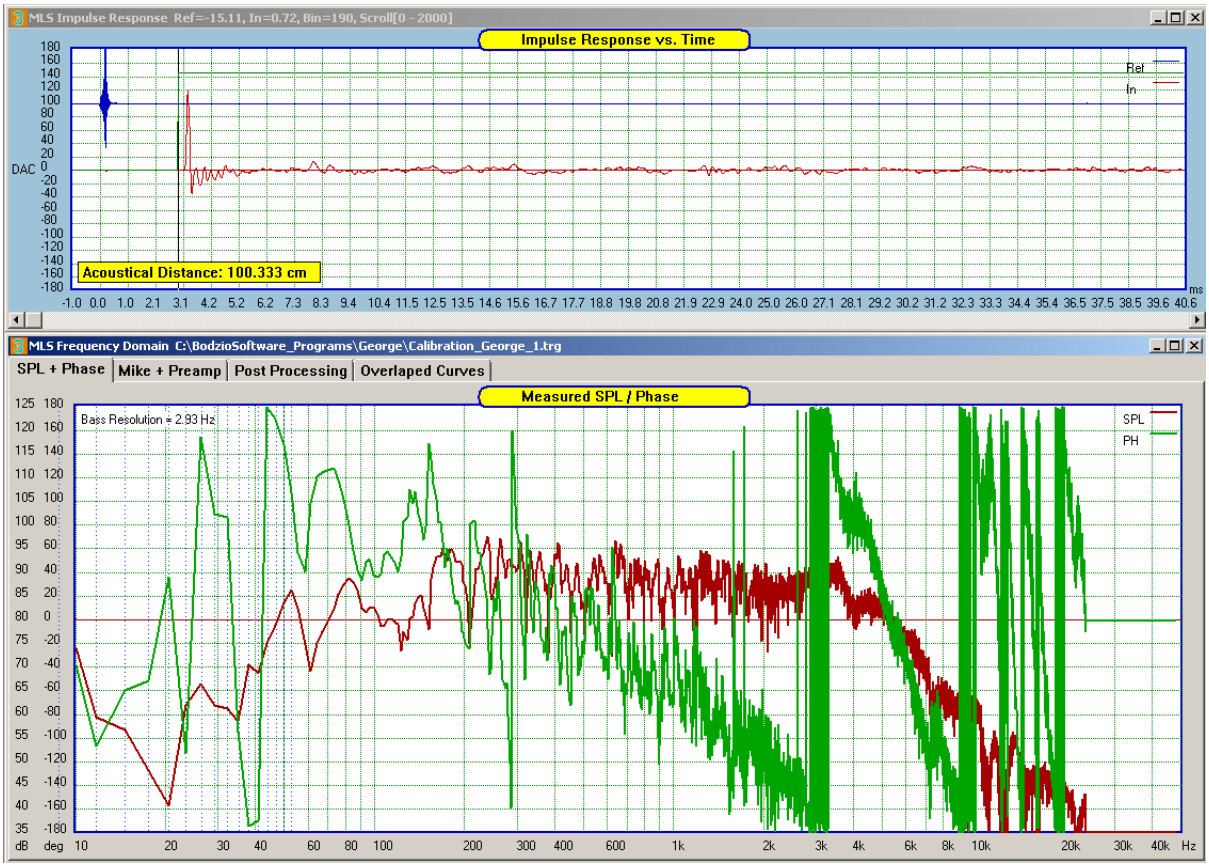


Figure 20. Raw SPL / Phase response with 2.93Hz bass resolution.

FFT window used in the example above is 500ms long. It is way too long and it captures large number of reflections from adjacent surfaces. You will very rarely see the SPL and phase curves with this amount of noise on them induced by the undesired surface reflections.

In the next step, the window was reduced to 15ms (still too long), and SPL and phase look significantly less noisy – see plots on Figure 21. These curves will be the basis for experimentation with Complex and Octave smoothing.



Figure 21. Raw SPL / Phase response with 15ms FFT window.

When SPL smoothing of 1/12dB/oct was applied to the curves above, we obtained quite reasonable SPL/Phase plots, without any noise on them. Also, NO complex smoothing was applied. This is clear indication, that complex smoothing was not necessary.

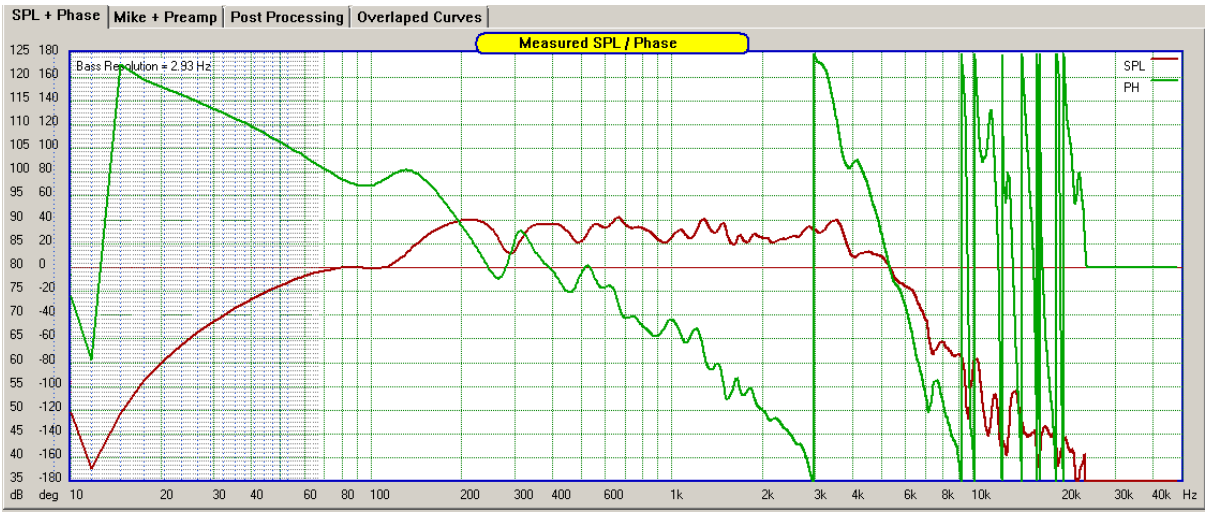


Figure 22. Raw SPL / Phase response with 15ms FFT window and 1/12dB Oct smoothing

In many instances this level of smoothness would by sufficient for SPL/Phase curves to be processed further. It is worth noticing, that the FFT window is still too long and still captures some reflections.

When Complex Smoothing is applied to the raw SPL/Phase curves from Figure 21. We obtain the following results.

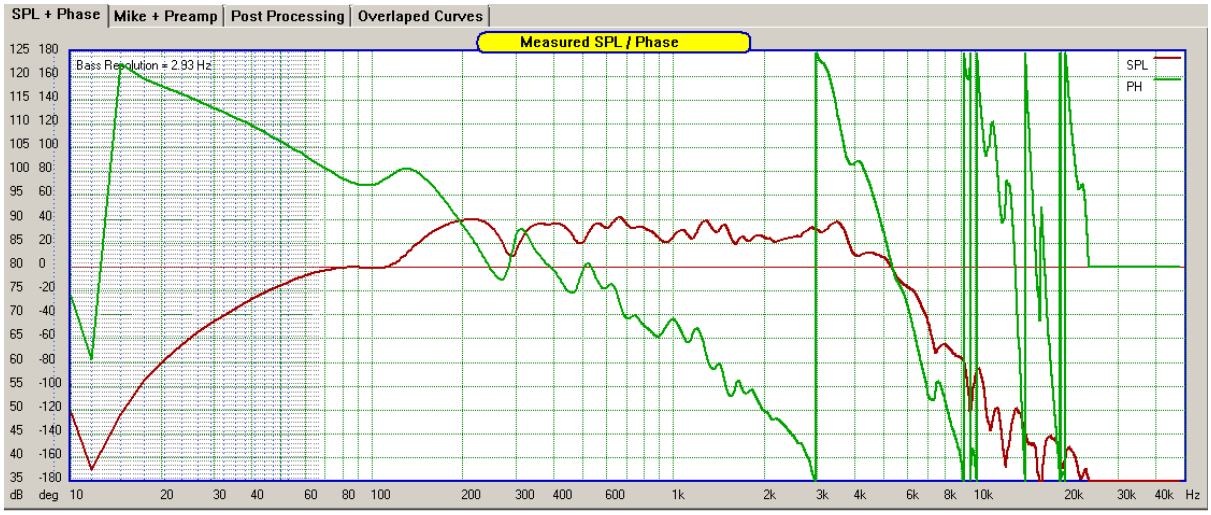


Figure 23. Raw SPL / Phase response with 15ms FFT window and Level8 Cplx smoothing.

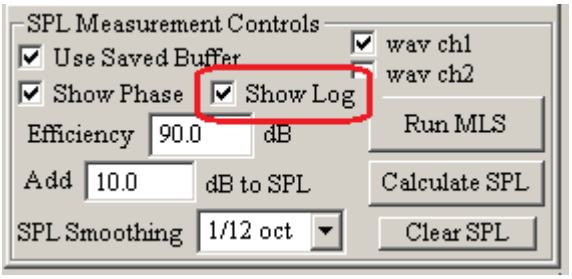
It is observable, that in this example, the SPL/Phase smoothness level is very similar for 1/12dB/ Oct and Level 8 Complex Smoothing.

There will be measurements, where you will need to apply both smoothing algorithms, on the top of necessary FFT windowing. The recommendation for this process would be as follows:

1. Apply correct FFT window to remove reflections.
2. Apply Octave Smoothing first to determine if the desired level of smoothness can be achieved.
3. If not, apply Complex Smoothing first (with no Octave Smoothing), then continue with Octave smoothing.

Generally, the level of smoothness shown on Figures 22 and 23 would be quite adequate for further processing by the DSP algorithms within the UE.

Displaying Logarithmic Curves in MLS system



The DSP engine runs with specific low-frequency resolution determined from Preferences settings of sampling and buffer size. The relationship was described above.

It is very likely, that you would be interested in equalization accuracy for the woofers and particularly subwoofers in your system. This is where the bass resolution comes into effect.

For instance: is the 5.86Hz bass resolution adequate for DSP engine to run on?.

To shed some light at this aspect of your design, the MLS system provides you with an optional switch “Show Log”, shown on the dialogue above. When activated, the logarithmic SPL (dark blue) and Phase (thin red) curves are superimposed onto the corresponding linear curves. You can now evaluate tracking of both curves. Please note, that smoothing will not be exactly the same for both types of curves, therefore the will always be some degree of mis-tracking of the curve’s peaks.

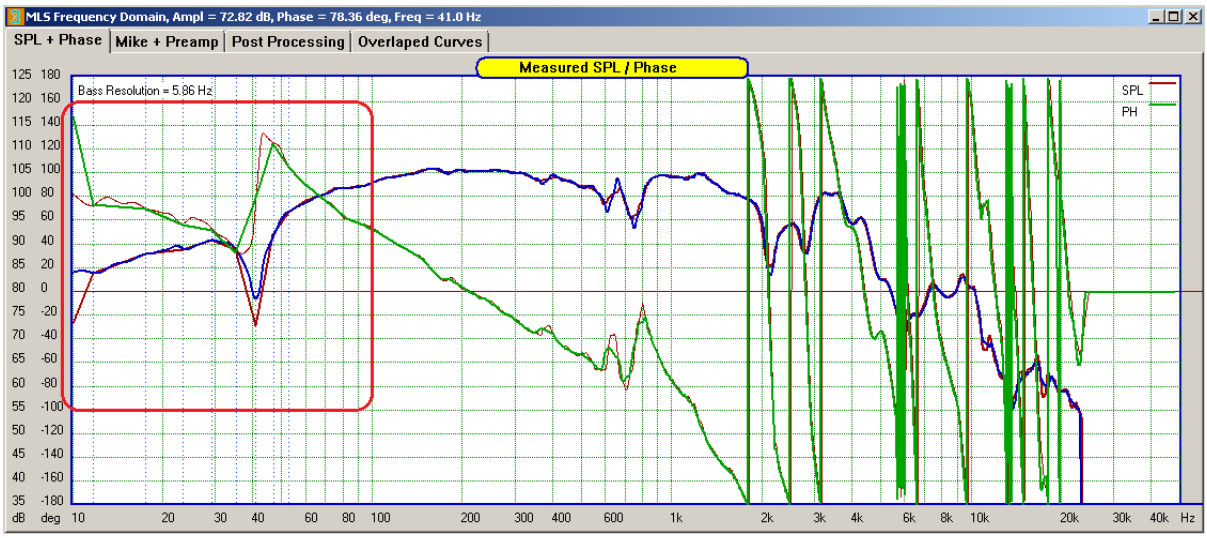


Figure 24. Low-frequency performance of Linear and Logarithmic SPL/Phase curves.

All signal processing algorithms within the UE software (MLS , DSP, Convolution etc.) run on linear frequency scale. This was specifically implemented to preserve integrity of Impulse Response data and prevent it from re-mapping onto multiple frequency scales.

As the consequence of this approach, the bass resolution is now dependant on the Preference settings for sampling frequency and buffer size. In our example, it is 5.86Hz – see below.



Figure 25. Low-frequency performance of bass resolution of 5.86Hz for subwoofer EQ .

As evident on Figure 25 above, the **measured equalized SPL and Phase** curves for a really big subwoofer (18") are excellent for the 5.86Hz bass resolution setting.

Time domain performance for linear-phase EQ is equally impressive as evident on the Figure 26 below.

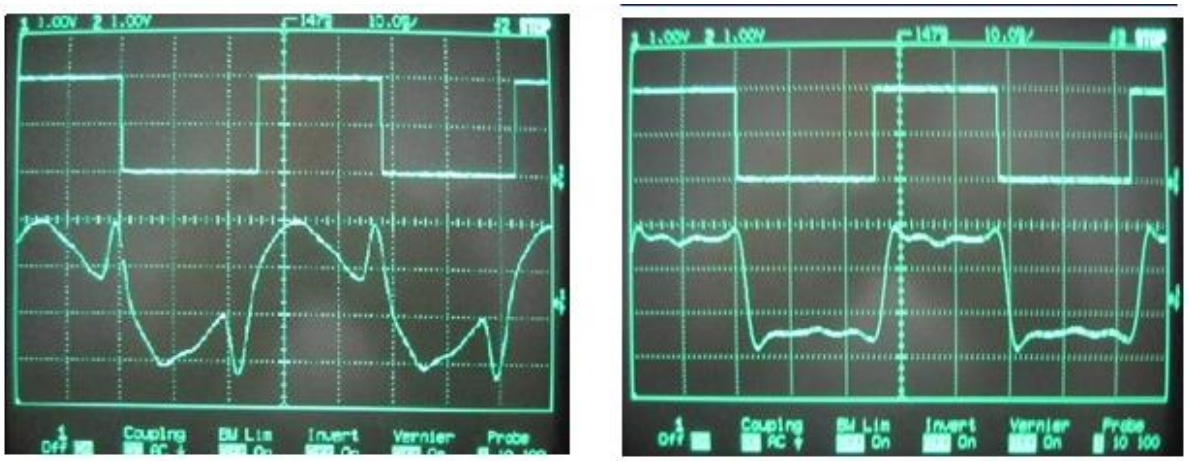


Figure 26. Time-domain performance for linear-phase EQ, and 5.86Hz bass resolution.

It is now evident, that the bass resolution discussed above if more than adequate for running large subwoofers in, HT system with the resulting excellent audio performance.