Ultimate Equalizer V8.0 Supplemental User Manual

June 2015

New Features Implemented in Ultimate Equalizer 8.0 – Digital

1. Implemented importing Impulse Response in ASCII files into MLS system.

2. New function for design and optimization of FIR impulse response.

3. Colour Cycling in MLS plots.

4. Export of Impulse Response in ASCII format and miniDSP format.

5. Performance improvements in HBT and MLS.

6. Imports UE V7 files.

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

Low-frequency accuracy

Low-frequency resolution

Low-frequency resolution considered in this manual is defined as:

LF-Res = (Sampling frequency) / (FFT length)

In DSP systems incorporating frequency-domain convolution, the IR length relates to the size of the FFT used in DSP process. Therefore, the above formula could also use IR length in the denominator.

Why is low-frequency resolution important?. Let's assume, that we have designed a FIR filter having four notches at low frequencies, just like the one below, where the first notch is located at 15Hz, second notch is located at 50Hz, third notch is located at 100Hz and the last notch is located at 200Hz. All Q-factors are equal to 20.
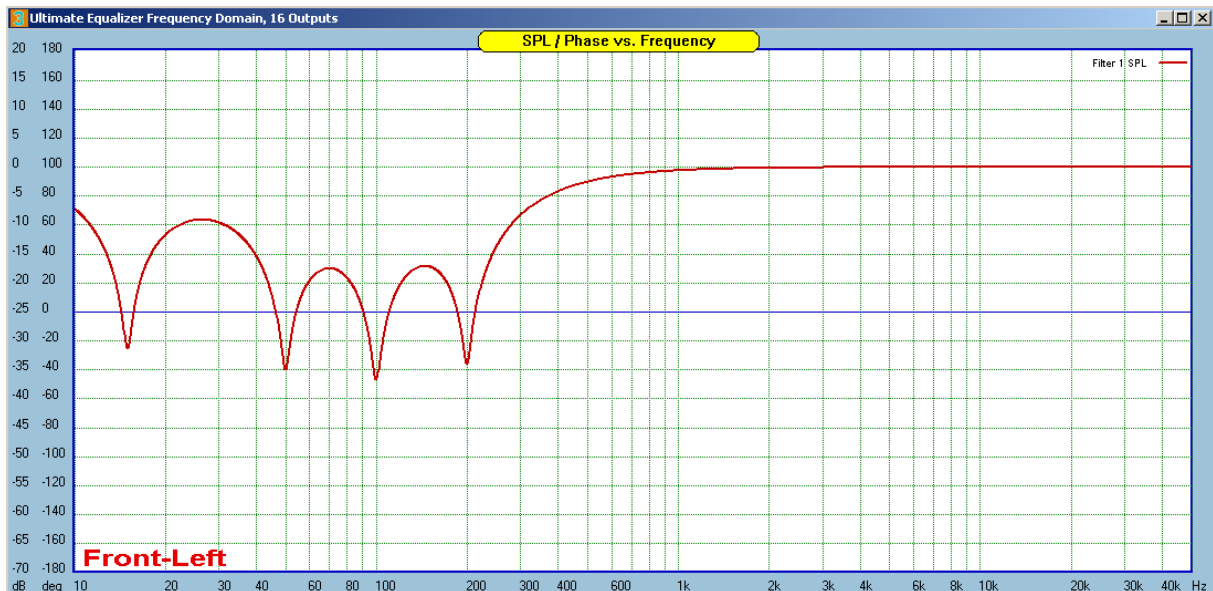

Figure 1. Experimental FIR filter with four notches.

The above filter is rather difficult proposition to manage by a DSP device. It has been specifically designed to stress the FIR filter and unmask any potential low-frequency deficiencies. It is quite unlikely, that you will ever need this type of DSP filter, as loudspeakers typically do not exhibit such high-Q peaks in the bass response.

We can now implement the filter from Figure 1 using a FIR filter with various low-frequency resolutions and examine the results.

First, a FIR filter with bass resolution of 11.72Hz was used and the result is shown on Figure 2 below.

Figure 2. Experimental FIR filter with four notches implemented in low bass resolution FIR.

Our FIR filter employs a sequence of frequency bins, starting like in the table below:

```
0.00  ,
11.72 ,
23.44 ,
35.16 ,
46.88 ,
58.59 ,
70.31 ,
82.03 ,
93.75 ,
105.47
```

It is observable, that none of the notch frequencies are close to the frequencies used by the FIR filter, therefore, the filter has great difficulties executing the requested filter curve.

Next, an FIR filter with bass resolution of 2.93Hz was used and the result is shown on Figure 3 below.
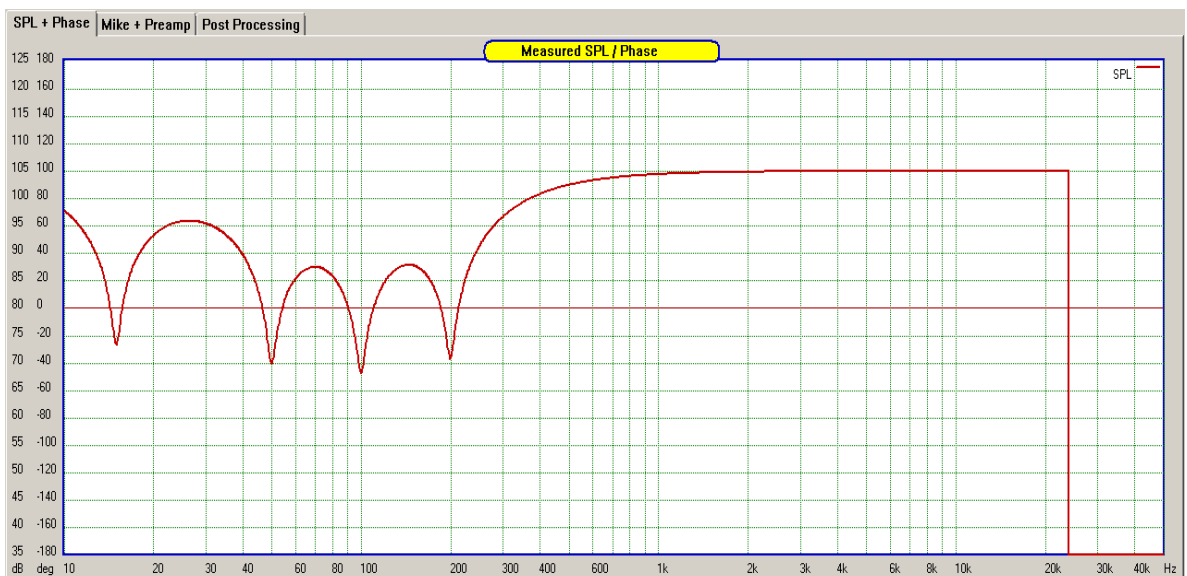

Figure 3. Experimental FIR filter with four notches implemented in high bass resolution FIR.

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The second FIR filter employs a sequence of frequency bins, starting like in the table below:

0.00	79.10	161.13
2.93	82.03	164.06
5.86	84.96	166.99
8.79	87.89	169.92
11.72	90.82	172.85
14.65	93.75	175.78
17.58	96.68	178.71
20.51	99.61	181.64
23.44	102.54	184.57
26.37	105.47	187.50
29.30	108.40	190.43
32.23	111.33	193.36
35.16	114.26	196.29
38.09	117.19	199.22
41.02	120.12	202.15
43.95	123.05	205.08
46.88		208.01
49.80		210.94

It is observable, that 15Hz notch will be approximated by 14.65Hz, 50Hz will be approximated by 49.80Hz, 100Hz will be approximated by 99.61Hz and 200Hz will be approximated by 199.22Hz. Now, all notch frequencies are close to the frequencies used by the FIR filter, therefore, the FIR filter has no difficulties executing the requested filter curve.

Generally, high-performance DSP filters would require 2.93Hz resolution or better. The next step up in resolution would be 1.47Hz and 0.735Hz. However, quite reasonable results in everyday loudspeaker work can be achieved with 5.86Hz and 11.72Hz can be used for crossovers upwards of 200Hzz and very basic loudspeaker equalization.

Speed and Performance

Ultimate Equalizer has been designed to run with equal performance level on 32-bit and 64-bit, Windows7+ computers. This has been mostly accomplished by using the following software approach:

1. **Multi-threaded coding.**

   The DSP engine within the UE opens several threads, and spreads the maths-intensive calculations over parallel CPU cores. The threads are synchronized, and the results collated when the slowest thread finishes. This approach can almost half the calculation time for each doubling the number of threads.

2. **Frequency-domain convolution.**

   Steven W. Smith in his landmark book "Digital Signal Processing A Practical Guide for Engineers and Scientists", page 237, concludes:
   " For example, a 32-point FFT is about *ten* times faster than the correlation method. However, **a** 4096-point FFT is *one-thousand* times faster. For small values of *N* (say, 32 to 12&), the FFT is important. For large values of *N* (1024 and above), the FFT **is** absolutely critical."

Secondly, the DSP engine uses Radix-8 FFTs, which offer speed improvement over standard FFT.

3. Single-precision DSP maths.

Single precision maths is known as 32-bit, floating-point arithmetic in computer language. The need for the program to run fast on 32-bit CPUs dictated the single-precision arithmetic. But, what are the consequences?.

Single-precision arithmetic may affect stop-band performance of the filter. Before we look at this issue in more details, let's first consider what level of performance we desire from our FIR filter. The filter can be assessed in pass-band, transition-band and stop-band.

Firstly, from the acoustical perspective, the stop-band of one driver is the band-pass of the complimenting driver. Acoustic summation of signals differing in level by 40dB is unlikely to cause the softer driver affecting the performance of louder driver. If the level difference is 60dB, the interference from the softer driver is inaudible.

Secondly, all drivers have a natural roll-off in the stop-band, which adds as much as **35-45dB of extra band-stop attenuation** – see tweeter and woofer examples below.
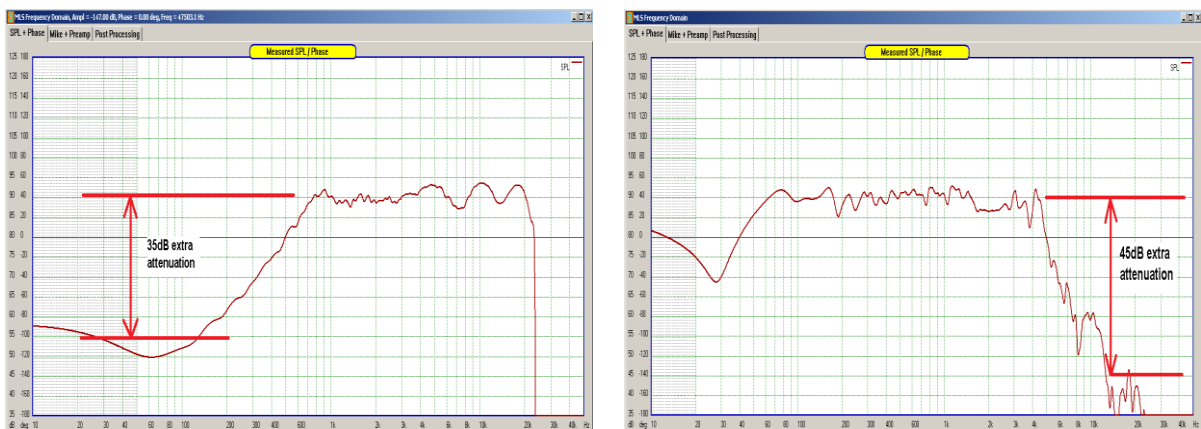
Figure 4. Examples of natural stop-band attenuation of loudspeaker drivers.

However, we certainly hope, that our FIR filter will run with better figures than those discussed above. Once again, Steven W. Smith in his book "Digital Signal Processing A Practical Guide for Engineers and Scientists", page 296, concludes:

"…If you really need more than -100dB of stop-band attenuation, you should use double precision. Single precision round-off noise on signals in the *passband* can erratically appear in the *stopband* with amplitudes in the -100dB to -120dB range…."

Realistically, if the single-precision FIR filter can achieve -110 to -120dB stop-band attenuation, it will be doing quite well. For more information please see "Design and optimization of FIR impulse response" chapter.

Performance Improvements in HBT and MLS

Typical DIY audio PC-based measurement system will measure and present the resulting frequency response as some SPL/phase curves between 20Hz and 20000Hz. Some systems will go beyond these limits and offer frequency range 10Hz – 50000Hz. An example of such system is shown below.
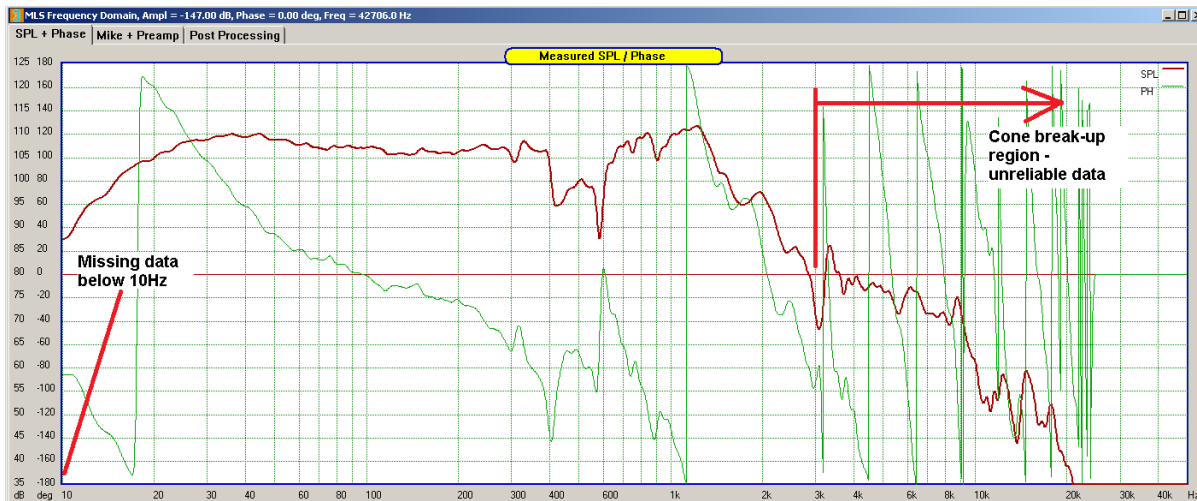

Figure 5. Measurement system showing results in 10Hz -50000Hz bandwidth.

One issue immediately observable from the Figure 5, is that, there is no visible frequency data below 10Hz and this may present a problem. Second issue is with the break-up region above 3kHz. The break-up region is about 35dB below driver's operating level and should be avoided and filtered out in the final design, but for now, it's not contributing anything to the design - just making the graphs less readable.

As discussed in previous chapters, good low-frequency resolution is important in any serious subwoofer and room equalization work. Recall, that high-performance FIR filters begin to use frequency bins well below 10Hz – see table below, showing several initial frequency steps of FFT in two example FIR filters.

```
 0.00 ,  0.00 Hz
 1.46 ,  0.73 Hz
 2.93 ,  1.46
 4.39 ,  2.20
 5.86 ,  2.93
 7.32 ,  3.66
 8.79 ,  4.39
10.25 ,  5.13
11.72 ,  5.86
13.18 ,  6.59
14.65 ,  7.32
16.11 ,  8.06
17.58 ,  8.79
19.04 ,  9.52
20.51 , 10.25
21.97 , 10.99
23.44 , 11.72
24.90 , 12.45
26.37 , 13.18
27.83 , 13.92
```

So, what are the options for improving the measured data quality?.

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1. Provide SPL/Phase measurements extending well below 10Hz and into the frequency range required by FIR low-frequency resolution.

Starting with V8 UE software release, all internal data processing is conducted from **0.51Hz to 272,000Hz**. This extended frequency range is available in the "File Editor" screen – as shown below.
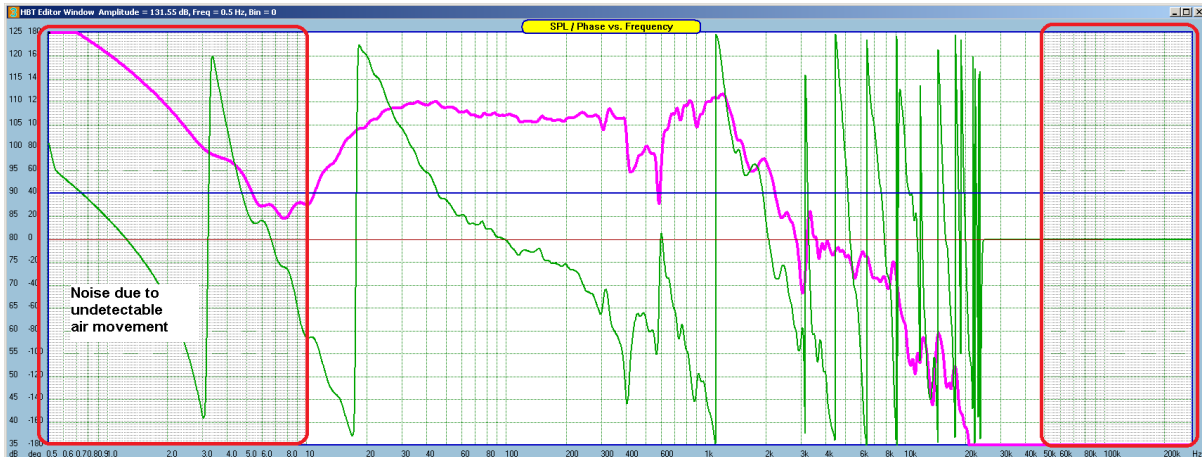

Figure 6. Measurement system showing results in 0.51Hz -272,000Hz bandwidth.

But, another problem is immediately visible – what is going on below 10Hz?.

This measurement was taken outdoor, and utmost care was taken to prevent even faint background noises from contaminating the measurement. However, an imperceptible air movement still managed to creep in, and resulted in erratic amplitude and phase fluctuations below 10Hz, and particularly below 1Hz. The same measurement taken short time later, had nearly identical frequency response above 10Hz, but was quite different below 10Hz again.

FFT windowing is not the solution, but fortunately, there is a simple processing tool designed to alleviate such problems – the HBT. The algorithm once properly applied between 12Hz and 2700Hz, will maintain the originally measured SPL and phase, but it will also clean up all wind noise below 10Hz, and remove the obnoxious break-up fluctuations below -30dB – see Figure 7 below..
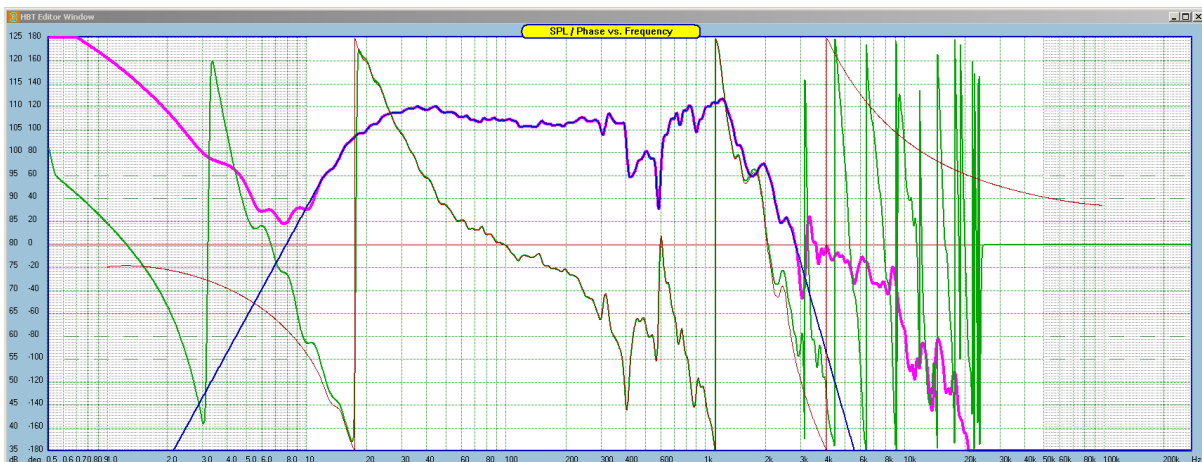
Figure 7. Using HBT to eliminate unwanted noise creeping into the measurements.

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We can now supply the improved measurement results to the DSP engine. This new data contains mathematically valid frequency data points down to 0.51Hz, and a simple parser can extract this data into the FFT frequency bins below 10Hz.
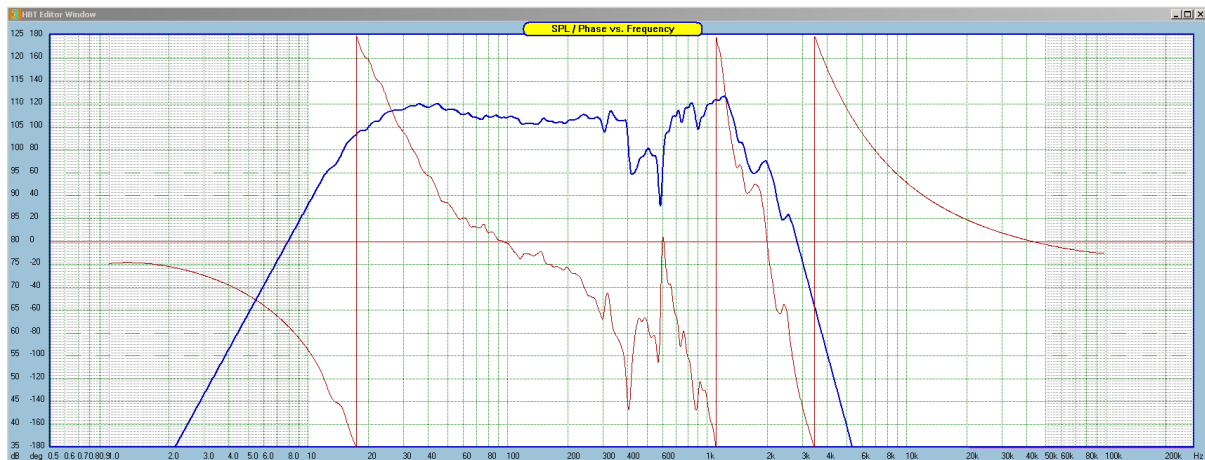

Figure 8. Measured amplitude and phase fully extended and perfectly cleaned by HBT.

In summary, extended MLS frequency range, extended frequency range of the HBT operation resulted in quality data points to match the requirements of high-performance FIR filter. In addition, please note, that the phase response can be made as close to true minimum-phase as they come.

Design and optimization of FIR impulse response

Note: Before you select "Optimize / Export Impulse response" please select "Position IR Windows" FIRST to position and re-size the two display windows, associated with the process – see menu option highlighted below..
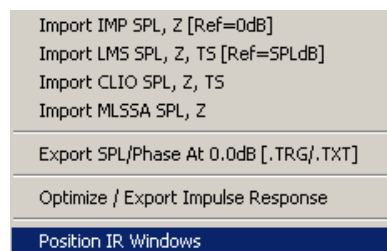
FIR impulse response can not be obtained by simply performing inverse FFT on the desired amplitude/phase response of your filter. Such "raw" impulse response would have to be *shifted, truncated and windowed* – in this order. The final outcome – the ability of the impulse response to deliver the required amplitude and phase responses – will depend on how the "raw" impulse response is processed. Design and optimization of the impulse response can be performed in both UE main operating modes: linear-phase and minimum-phase. The mode of operation is selected from a checkbox in the System Design dialogue box:
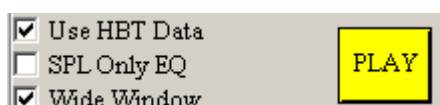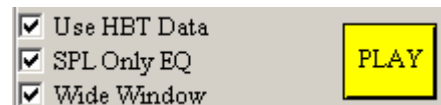

Figure 9. Linear-Phase Mode   Minimum-Phase Mode.

The design, optimization and exporting of the impulse response is available for all 16 DSP channels and can be performed using single dialogue box shown below:
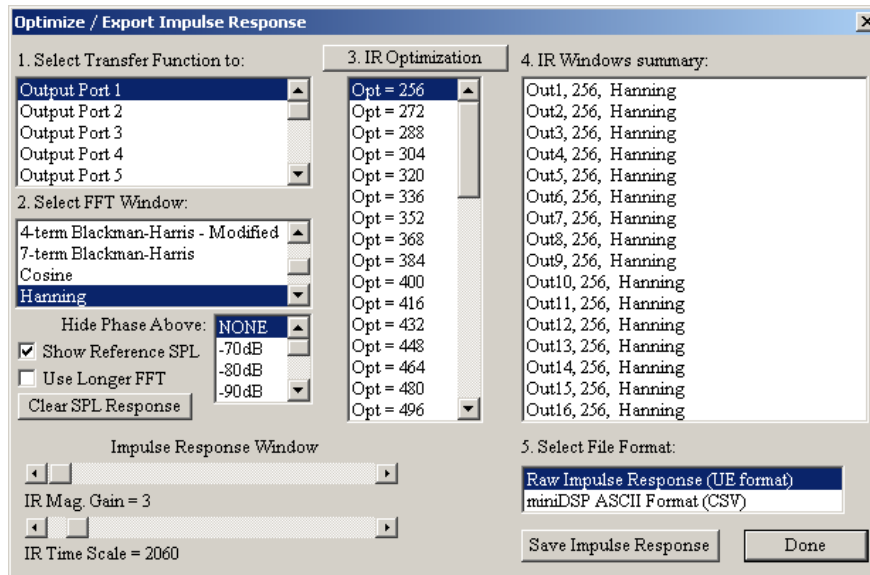

Figure 10. Controls for design, optimization of impulse responses for all 16 channels.

1. **Select Transfer Function to**: - This list box allows you to choose which transfer function will be converted to impulse response. It is essential to use this selection in accordance with the main system diagram. **This selection determines which impulse response will be exported**.

2. **Select FFT Window** – This list box contains a selection of many popular FFT windows. You may need to try a couple of different windows to see which one results in better stop-band performance

3. **IR Optimization** – This button, as step 3, activates calculation algorithm. The algorithm will provide 49 resulting options for your evaluation. All curves will be displayed in the "Impulse Response SPL Window" and colour-coded. You may need to pay some attention to the display, as the process runs uninterrupted till completion. The **vertical scale in the SPL window is 10dB per division**.
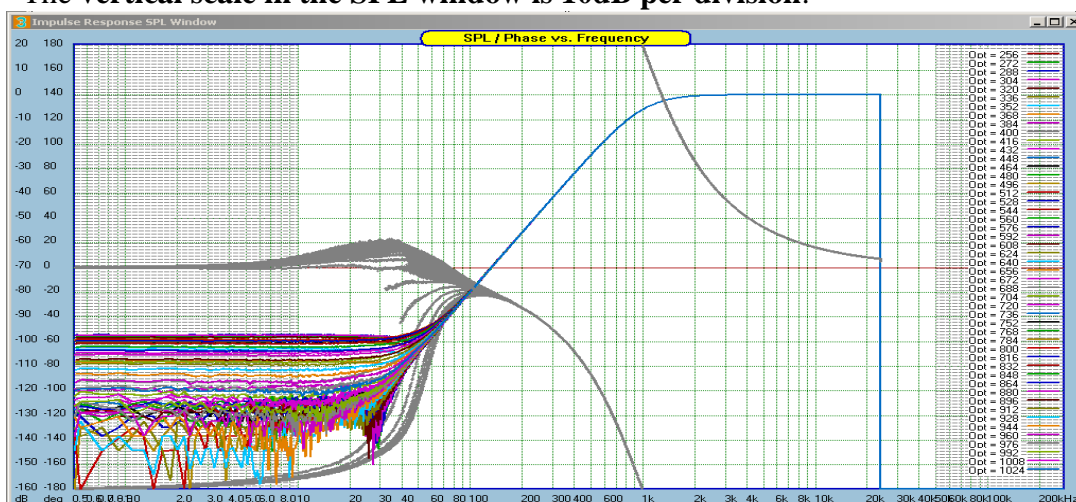
Figure 11. Colour-coded selection of SPL/phase curves resulting from IR optimization.

Once the algorithm has completed, you can display individual results by double-click on the items in the list box below the **"IR Optimization"** button. Your selection will be instantly transferred to **"4. IR Summary Window**" and will be used by the DSP engine from now on – see below.
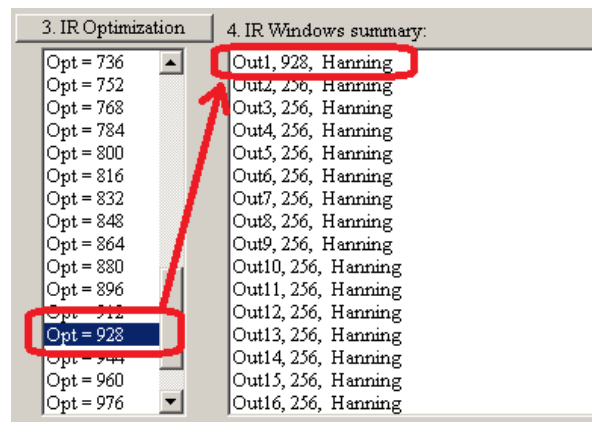
Visual inspection gives you the opportunity of selecting the lowest stop-band level, or somewhat higher level (say -120dB), but better defined phase response – if you prefer.
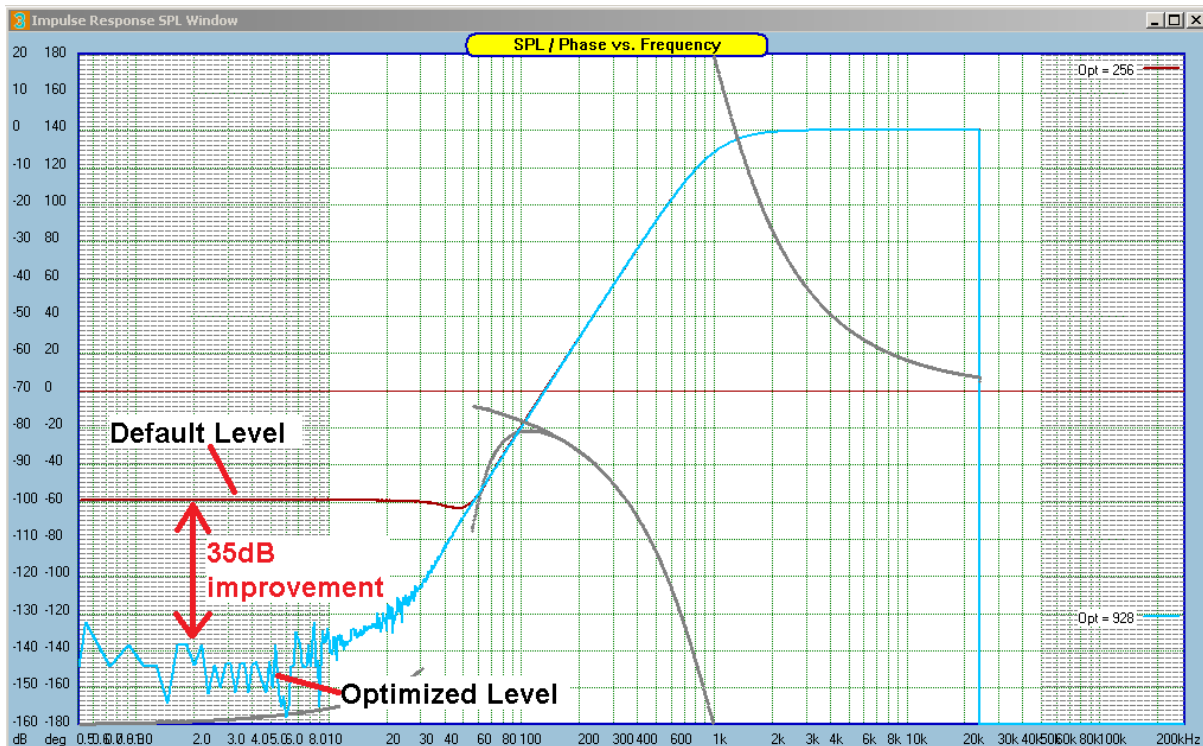

Figure 12. Selection "Opt-928" has resulted in 35dB improvement in stop-band attenuation.

If performed correctly, the above process can improve the stop-band attenuation of high-pass and band-pass filters as much as 40-50dB, in some instances. Average improvement figure would be in order of 30-35dB.

A couple of examples: a low-pass and a band-pass filter, with UE buffer of 512 bins and sampling frequency of 48kHz are shown below. These filters would be exported as 2048-long impulse responses and would comfortably fit into many commercially available DSP boards.
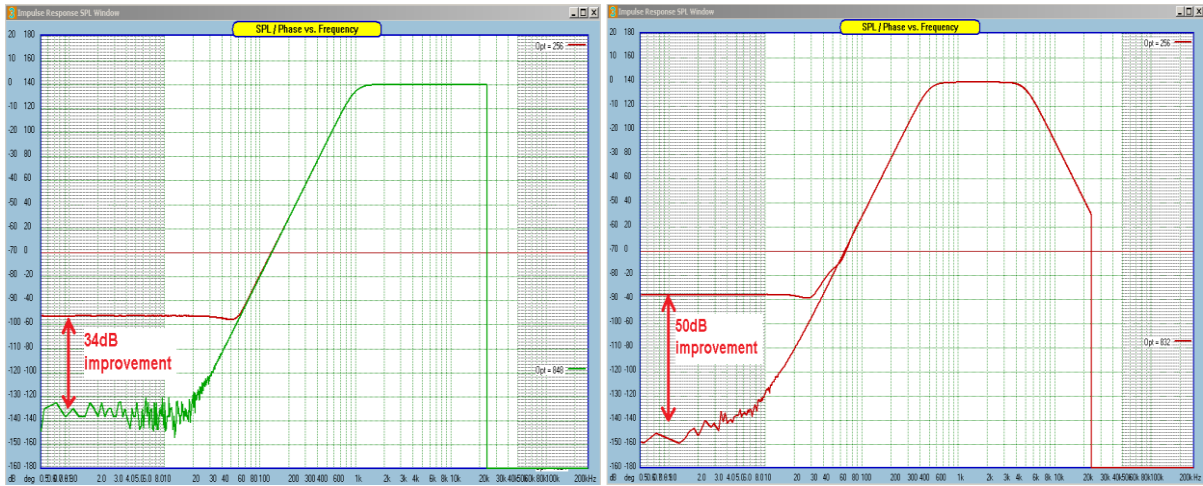
Figure 13. Example of optimization improvements

Generally, performance level of impulse responses will depend on their length (or in UE terms – buffer size), cut-off frequency, steepness of slopes, configuration, and optimization choices. The optimization algorithm is actually quite simple and gives you the final say. It is quite feasible to employ more advanced optimization scheme, one that would involve variable shift of the "raw" impulse response and perform optimization for every new position. However, it seems not necessary, as the figures obtained with simple algorithm are quite encouraging.

The average optimization improvements of 30dB will push the stop-band attenuation level into -120 to -130dB level, and this somewhat better than the expected level for single-precision arithmetic. If you prefer, you can couple this figure with natural driver attenuation in the stop-band of -30dB (conservative figure) would result in acoustical performance of stop-band attenuation in the vicinity of -150dB. Optimization results of low-pass filters, particularly with short IRs, looks differently, and yields somewhat different results.
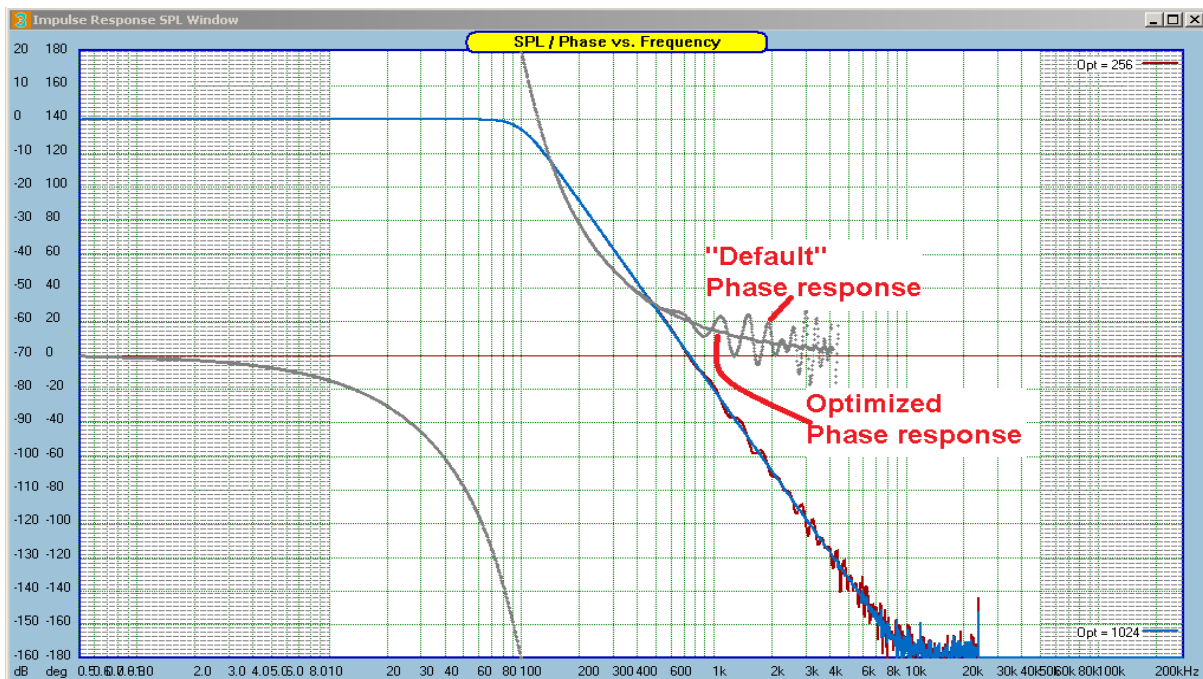
Figure 14. Optimization of short low-pass impulse response.

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This is particularly visible, as in the figure above, when original IR of the filter is quite short. In this example, the impulse response has 2048 coefficients, and the filter is asked to perform Butterworth -24dB/oct cut-off 100Hz. The zoom-in figure below shows, that the "default" filter has developed some irregularities below -70dB level, which translates into phase fluctuations. The "optimized" version is actually performing very well right down to -130dB level, with the noise floor at -155dB level.
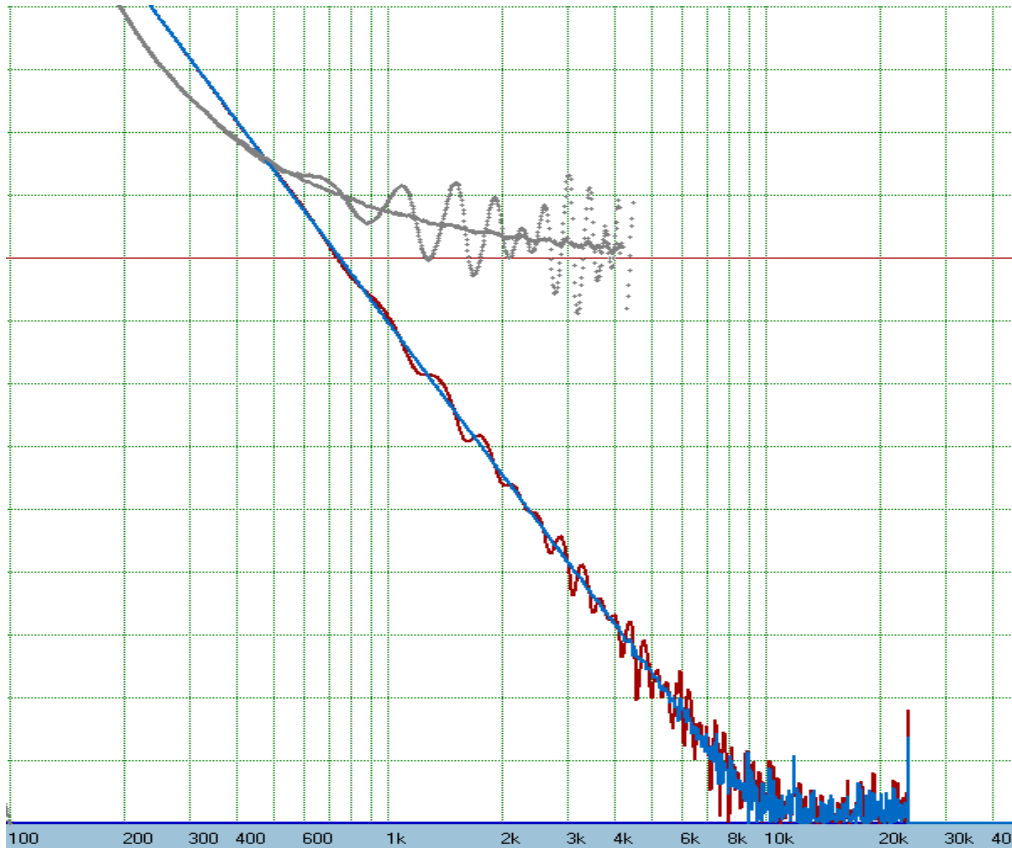

Figure 15. SPL and phase fluctuations disappear after optimization of short filter.

Optimization and export of the impulse response is the final stage in your project development. Optimization can only be performed using finalized SPL/phase plots, so please make sure, that you plot and review transfer functions of all channels, and you are happy with the results, before taking the final step of optimizing and exporting impulse responses. Every time the UE buffer size is changed (from Preferences screen), the SPL/phase needs to be re-plotted to obtain the most recent SPL/phase curves.

Optimization information is saved to Project File, so every time you load new project, new optimization information will be used.

Linear-Phase filters

This process is actually very simple. Because optimization is not really needed, but you may choose to check the impulse response performance with "Wide Window" option turned ON and OFF. Shown below is an example of the same filter as it was used above without any optimization. Please note the linear phase shown in grey colour.
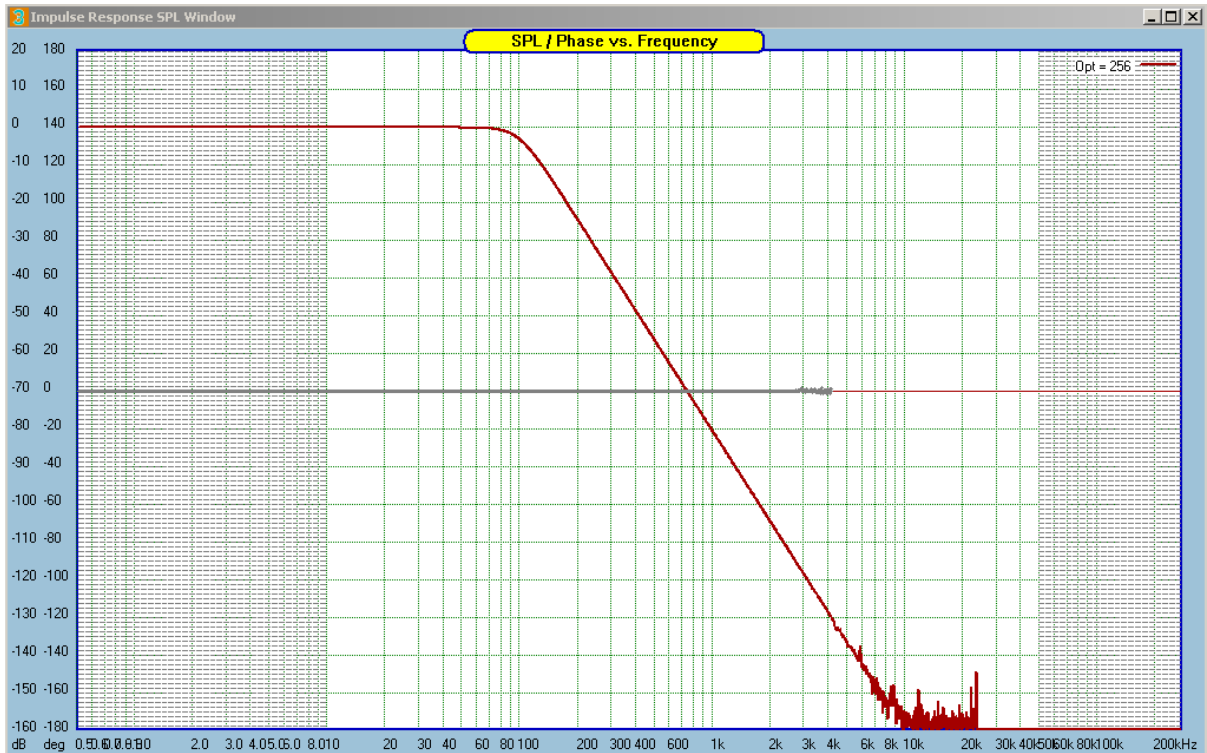
Figure 16. Linear-Phase impulse response is performing optimally.

The second example shows an L-R band-pass filter with 500/2000Hz cut-off frequencies and 4096 coefficients used for it's impulse response. It had it's minimum-phase version optimized, but linear-phase version performed with -140dB stop-band attenuation right away.
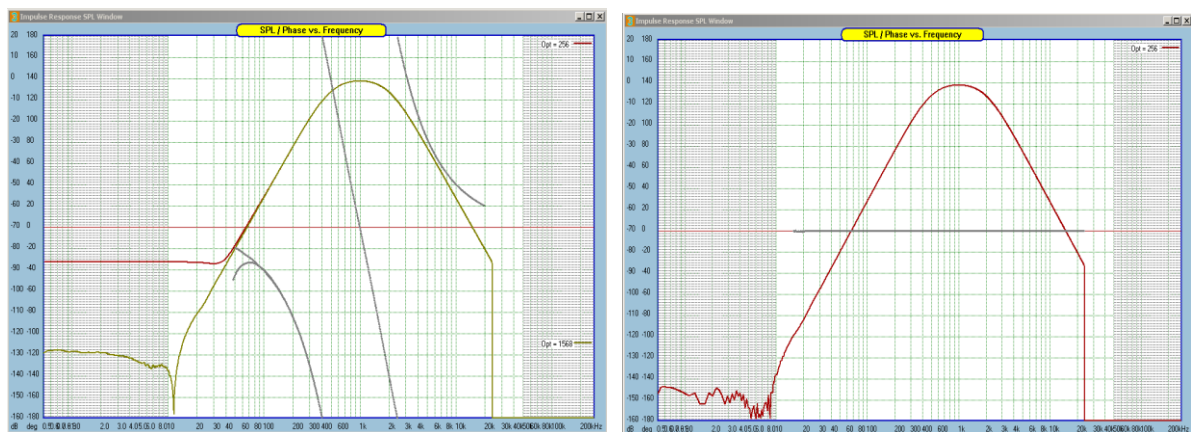

Figure 17. Examples of L-R band-pass filter, 500/2000Hz cut-off and 4096 coefficients.

You will need to perform the tree-step process of optimization outlined before, but the optimizer will throw the final answer right away. The only parameters, that may have visible effect on the stop-band performance is the selection of "Wide Window" ON/OFF in the System Design dialogue.
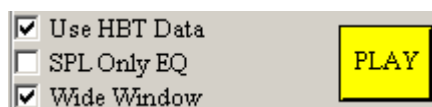
Optimizing Equalizing filters

So far we have been reviewing the design and optimization of impulse responses associated with crossover filters. As it happens, there is no difference in the sequence of this process when you design a filter intended to equalize your loudspeaker driver in both: linear-phase or minimum-phase.
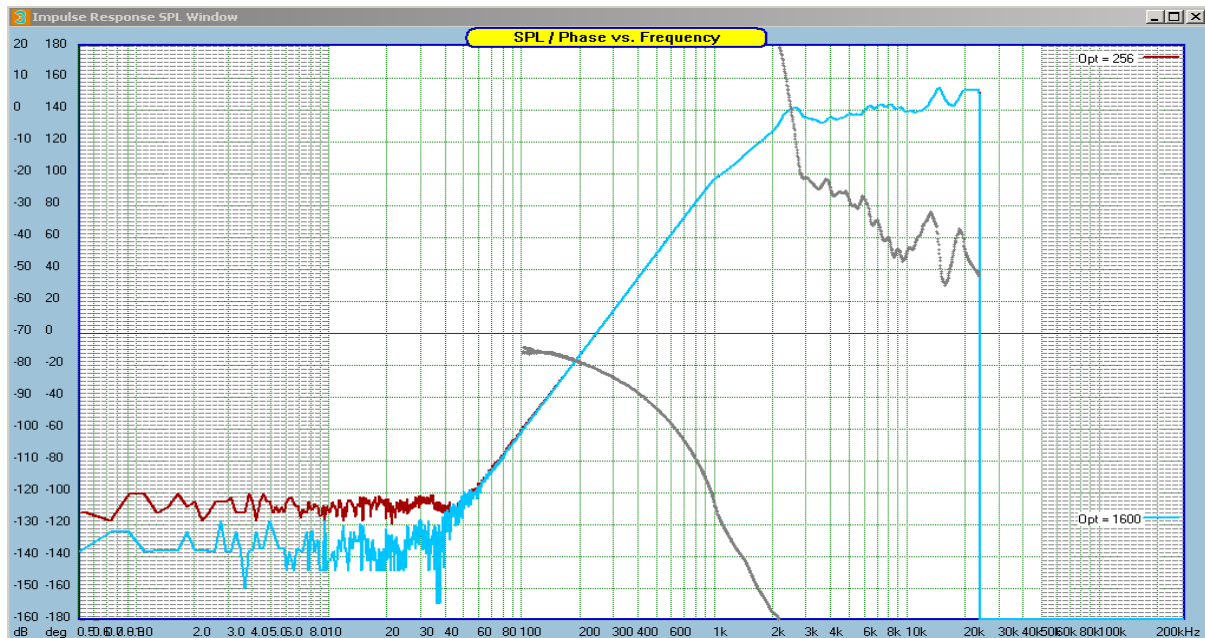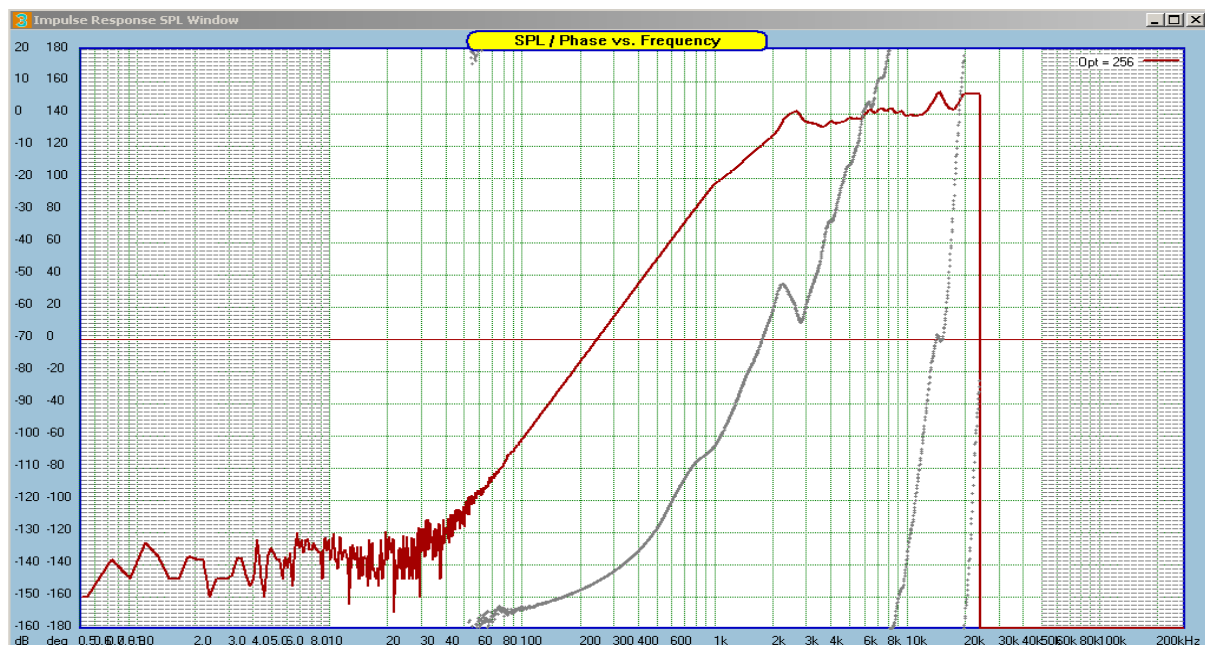

Figure 18. Minimum-phase equalizing filter

Figure 19. Linear-phase equalizing filter

As shown on the figures above, minimum-phase version after optimization and linear-phase version have very similar stop-band attenuation – better than -130dB. **Please observe phase response differences in both type of filters.**

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Monitoring Impulse Response

Impulse response can be viewed on a separate window, titled **"Impulse response Window"**.
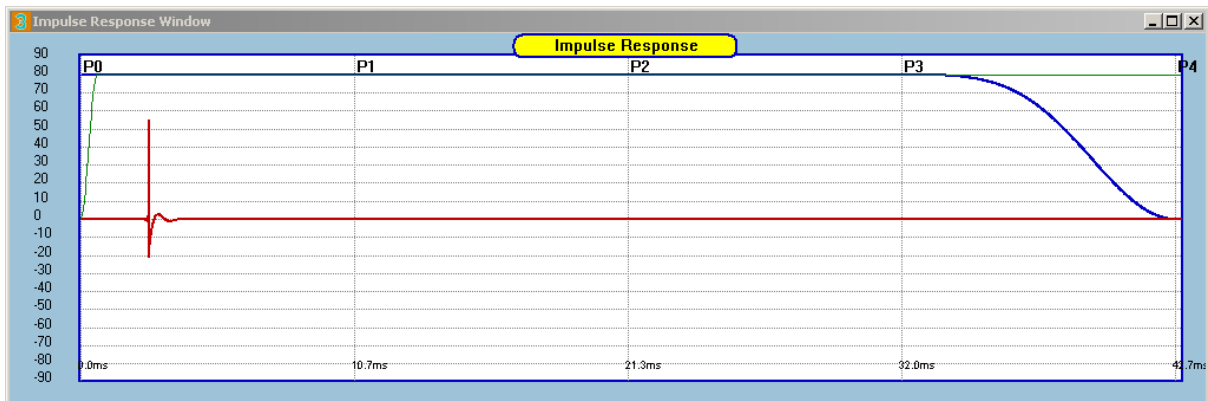
Figure 20. Impulse response monitoring window.

The impulse can be magnified in amplitude, and stretched in time domain using the controls provided on the control dialogue box – see below.
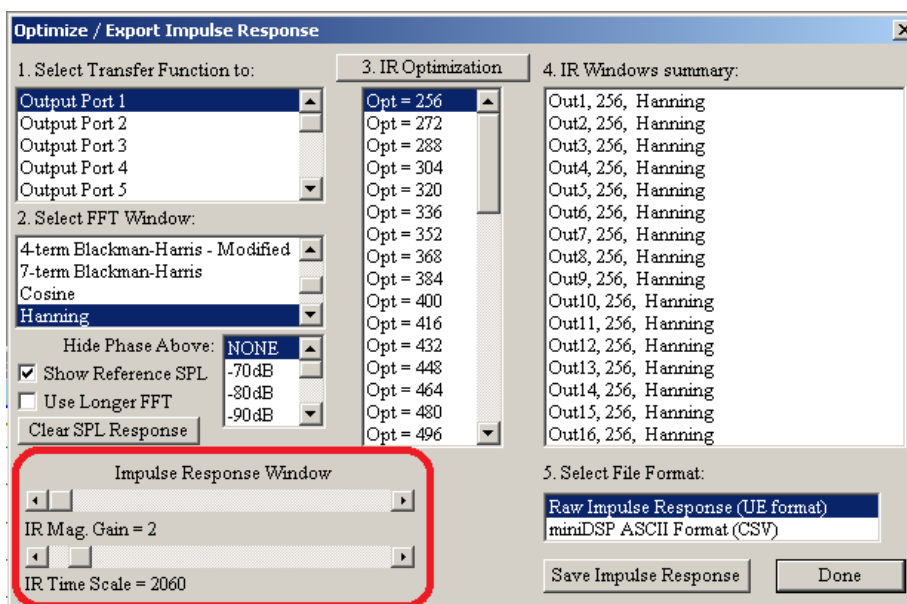

Figure 21. Controls for impulse response window.

It must be stressed, that the impulse response shown in this window is NOT the final impulse response of the audio channel assigned as Port1, Port2 ….Port16. The complete audio channel must obviously include the loudspeaker. Therefore the final audio channel impulse response would be obtained by convolving the impulse presented in the monitoring window with the impulse response of the loudspeaker.

Interestingly, when designing impulse response for the purpose of equalizing the loudspeaker, the final audio channel impulse response (EQ filter convolved with loudspeaker) will be that of a perfect textbook electrical filter. Well, this is the purpose of equalizing drivers with Ultimate Equalizer.

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Window shape may improve stop-band attenuation. In the next example, we have impulse response of an +48dB/oct, 20Hz Butterworth HP filter. Apart from the initial peak, there is nothing much that can be observed on this picture.
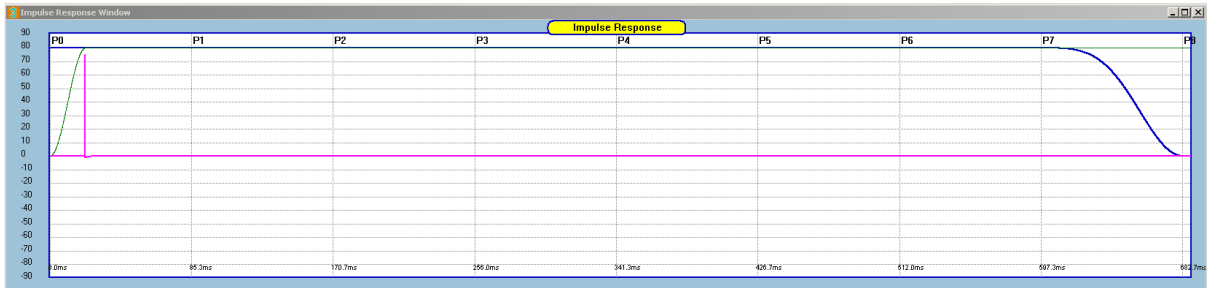

Figure 22. Full-view of impulse response

When the impulse response gain is set for 50000 magnification, we are able to, see what happens at the far-end of the impulse response. Minimum-Phase example is presented first..
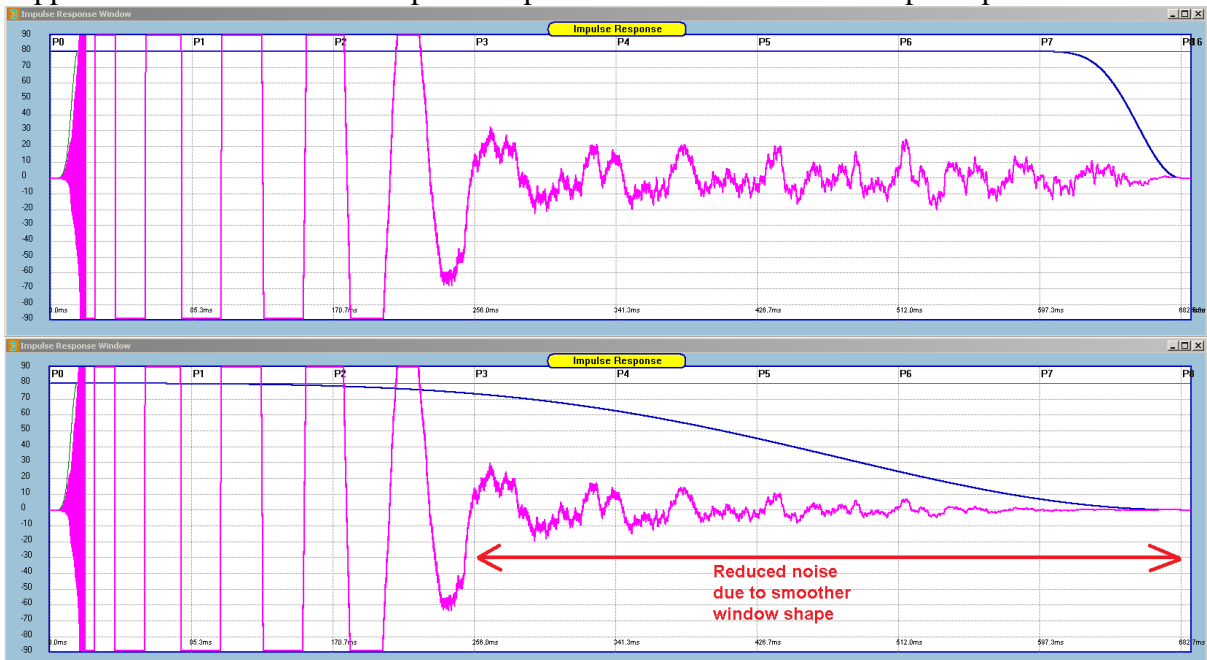

Figure 23. Applying smoother window.

By applying smoother window ("Wide Window" OFF), we can observe reduced noise on the left end of the impulse response. This reduction will result in improved stop-band attenuation – as can be seen on the picture on the right.
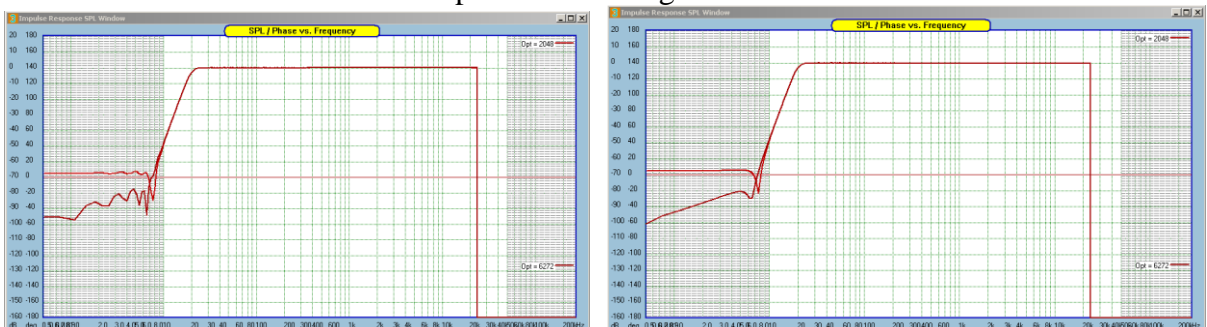
Figure 23. Example of MP stop-band attenuation difference with smoother window.

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The effect of the window's shape is even more pronounced in linear-phase mode. Here is the impulse response of the linear-phase version of the same filter.
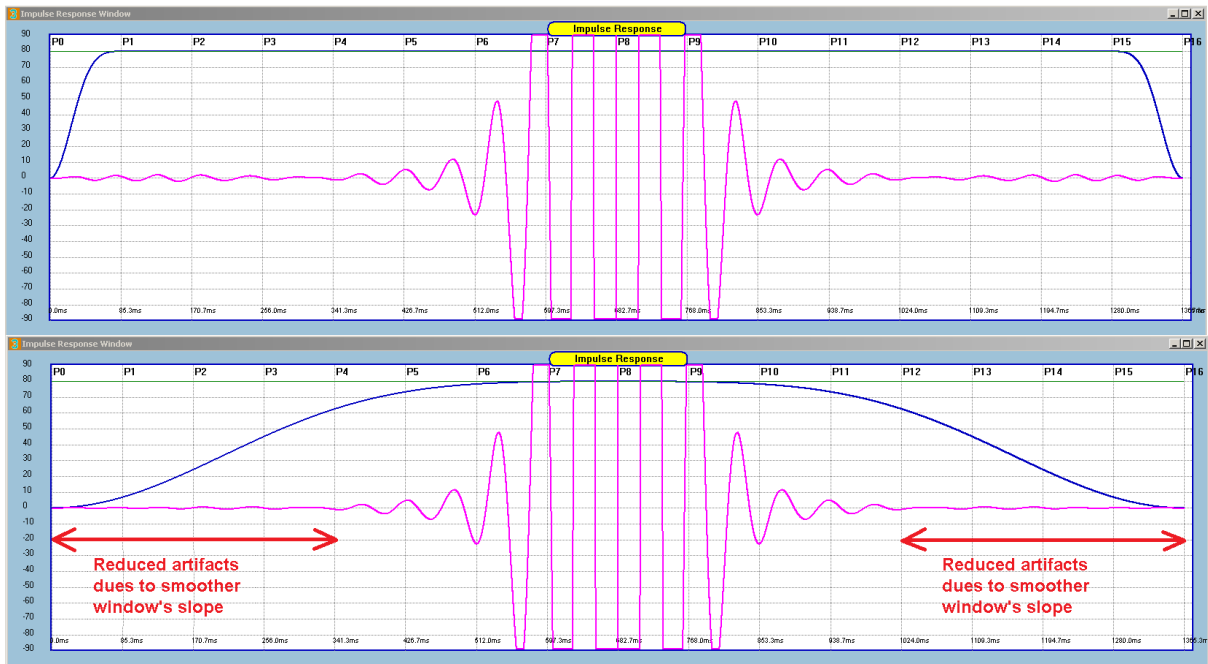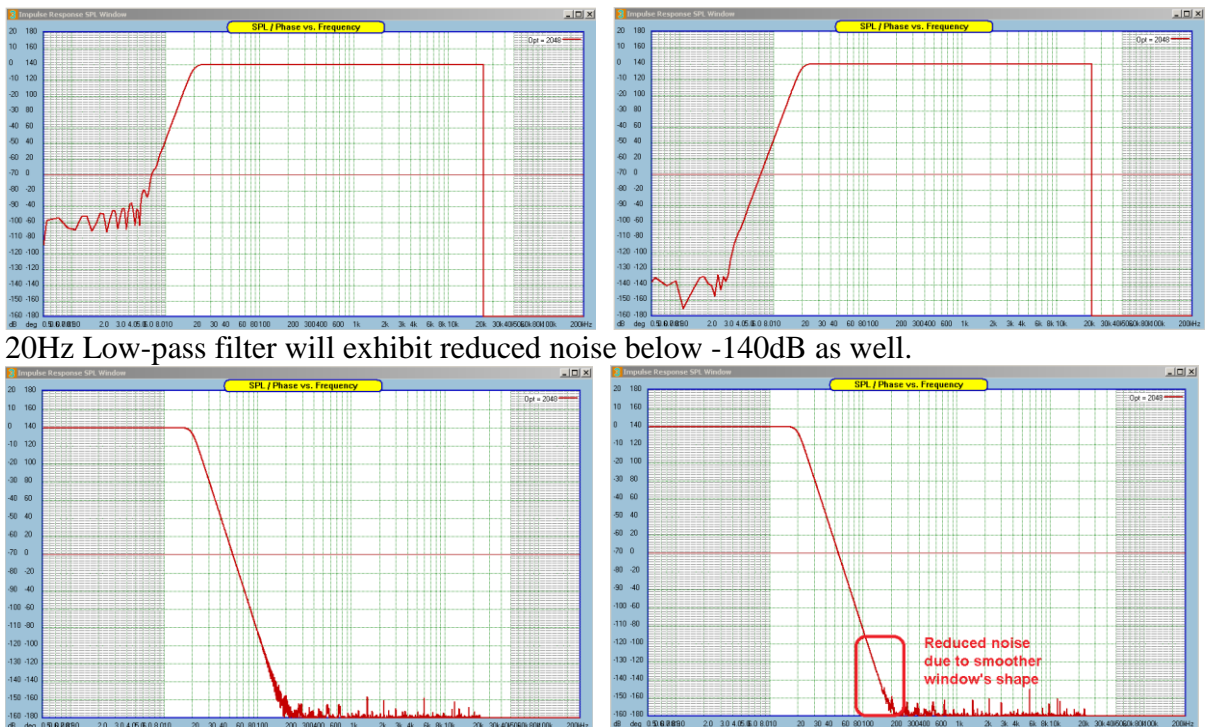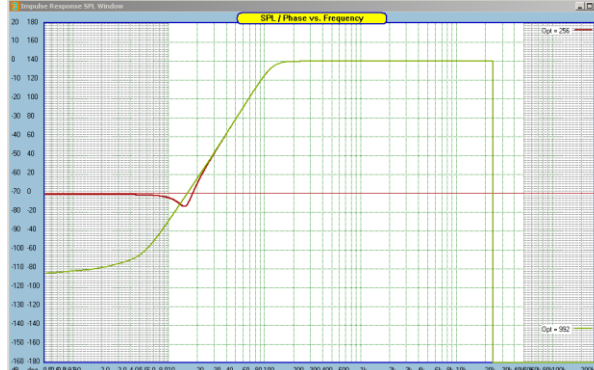

Figure 24. Applying smoother window.

By applying smoother window ("Wide Window" OFF), we can observe reduced artifacts towards both ends of the symmetrical impulse response. This reduction will result in improved stop-band attenuation – as can be seen on the picture on the right – in this case, by as much as 40dB.

20Hz Low-pass filter will exhibit reduced noise below -140dB as well.

Figure 25. Example of LP stop-band attenuation difference with smoother window.

When should you apply wide window?. Typically - for short filters, like **512/1024 buffer in 8-partition mode** – see comparison below with **121Hz, +24dB/oct, LR, 4T-Blackman window.**
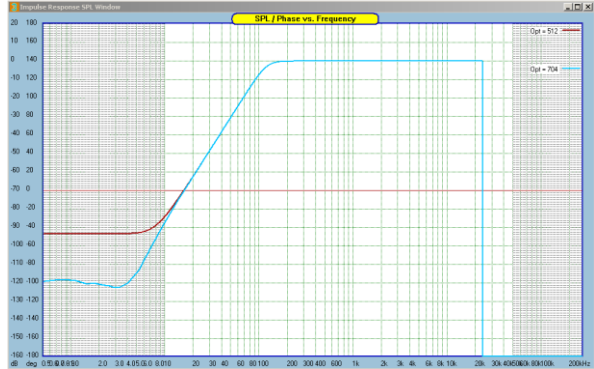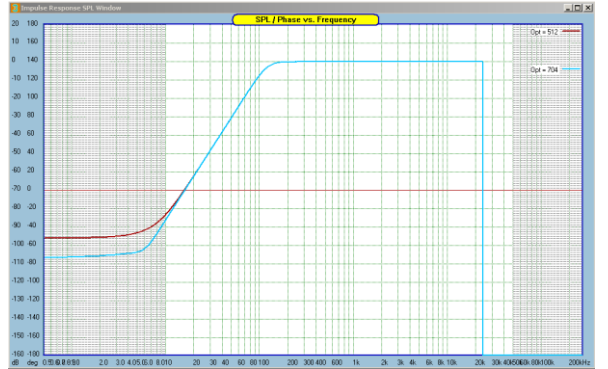
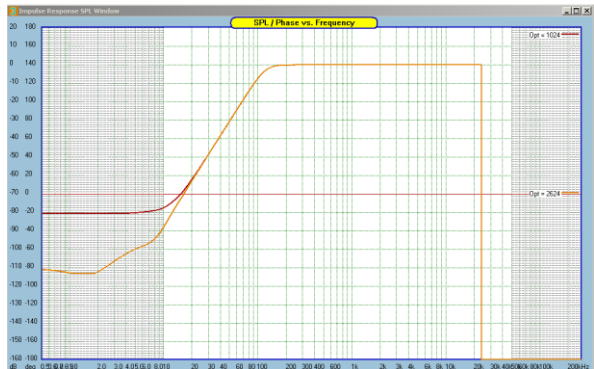**512 buffer, "Wide Window" - ON**   **512 buffer, "Wide Window" - OFF**
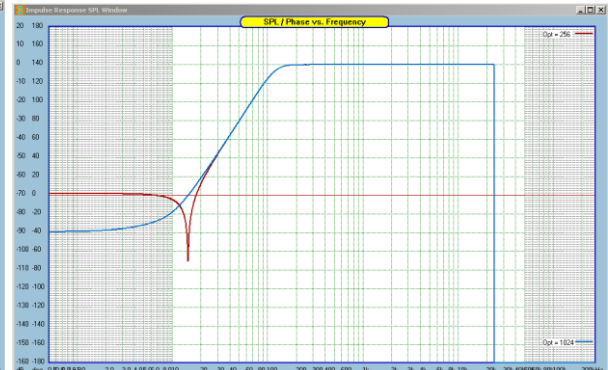
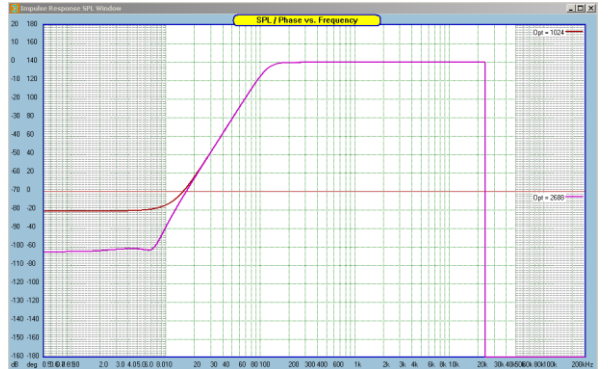**1024 buffer**   **1024 buffer**
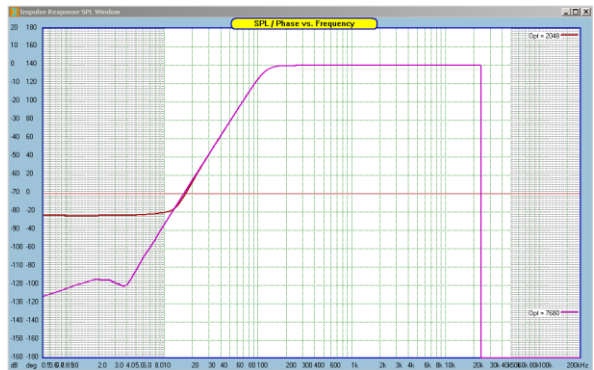

2048 buffer   **2048 buffer**

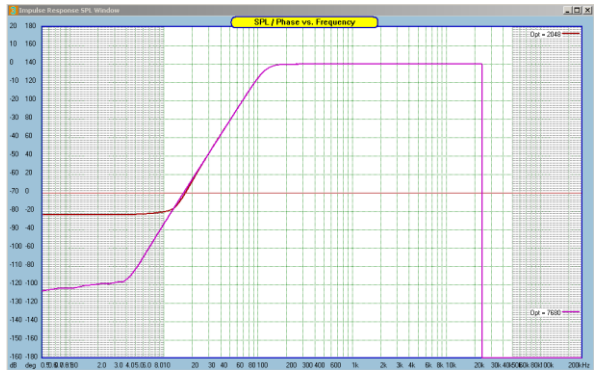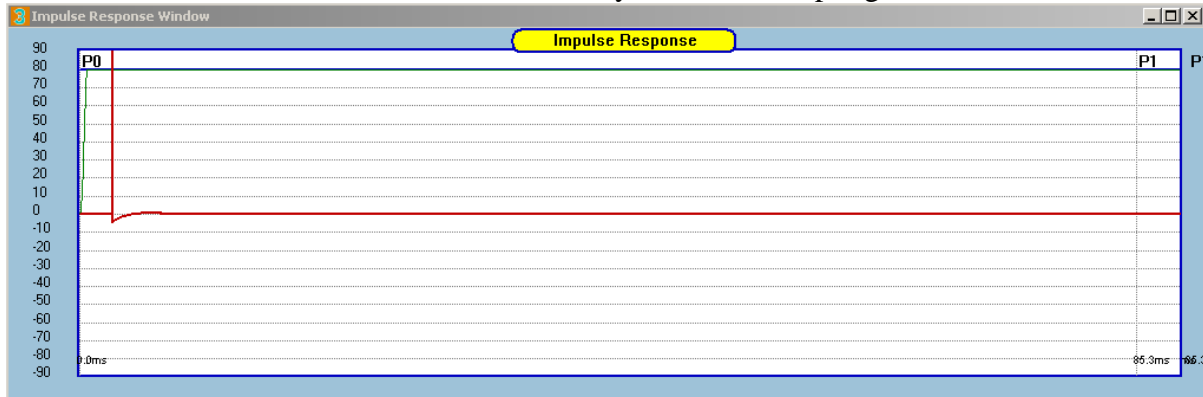**4096 buffer**   **4096 buffer**

Figure 26. Example of LP stop-band attenuation difference with smoother window.

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Method 1 and Method 2 Explained

Method 1 rotates impulse response by fixed amount = 128 bins.
Method 2 rotates impulse response by (UE partition size) / 4.

Location of IR – Method 1 – 2.667ms IR latency at 48kHz sampling

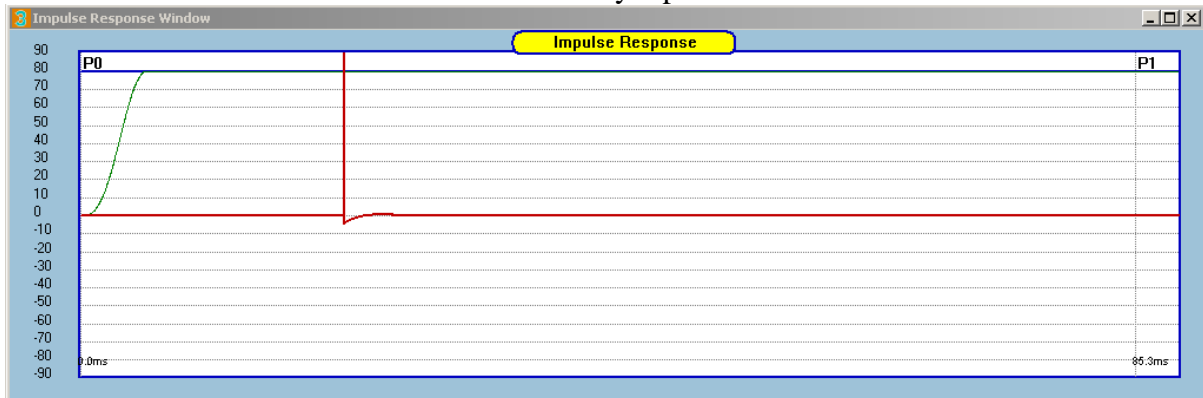Location of IR – Method 2 – 21.33ms IR latency – partition size 85.5ms.

Figure 27. Method 1 vs. Method 2 explained graphically.

Latency figures given in UE7 Manual on page 5 are given for Method 1. Method 2 will add additional 19ms latency.

There are some simple, but obvious rules when you deal with a set of impulse responses:

1. You can not mix minimum-phase and linear-phase impulse responses in one project file – this is obvious, as the latencies would be quite different for each option.
2. You can not mix **Wide Window ON and OFF** options in one project. This is a global, project-wide setting. If you wish to change this setting, you must exit **Optimize/Export Impulse Response** dialogue box, then change this setting in **System Response** dialogue box, and then re-deploy the **Optimize/Export Impulse Response** dialogue box – it will have now the correct settings.
3. You can not mix IR Optimizations **Method 1 and Method 2** – this is obvious, as the latencies would be quite different for each option.
4. You can choose and mix **FFT Windows** in one project. This is whole purpose of optimizations.
5. You can choose and mix **IR "Opt"** optimization parameter in one project. This is whole purpose of optimizations.

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6. Mixing Project Files between 8P and 16P versions of Ultimate Equalizer is allowed, but you may wish to re-optimize impulse responses to always get optimum stop-band attenuations.
7. Every time you change the UE buffer size, you must re-plot all frequency responses in the **System Design** screen **before** you re-optimize impulse responses.

An example of fully edited and optimized set of 16 impulse minimum-phase impulse responses (large 7.2 HT system) is shown below. All data is saved to the project file, so when the project is opened again, you should firstly, re-plot all port outputs on **System Response** screen, then go to the **Optimize/Export Impulse Response** dialogue box to see exactly what optimization options have been saved. Now, you can double-click on any of the listing in the **"4.IR Windows Summary"** box and you will se all saved options impulse response optimizations re-plotted on the **Impulse Response SPL Window**.
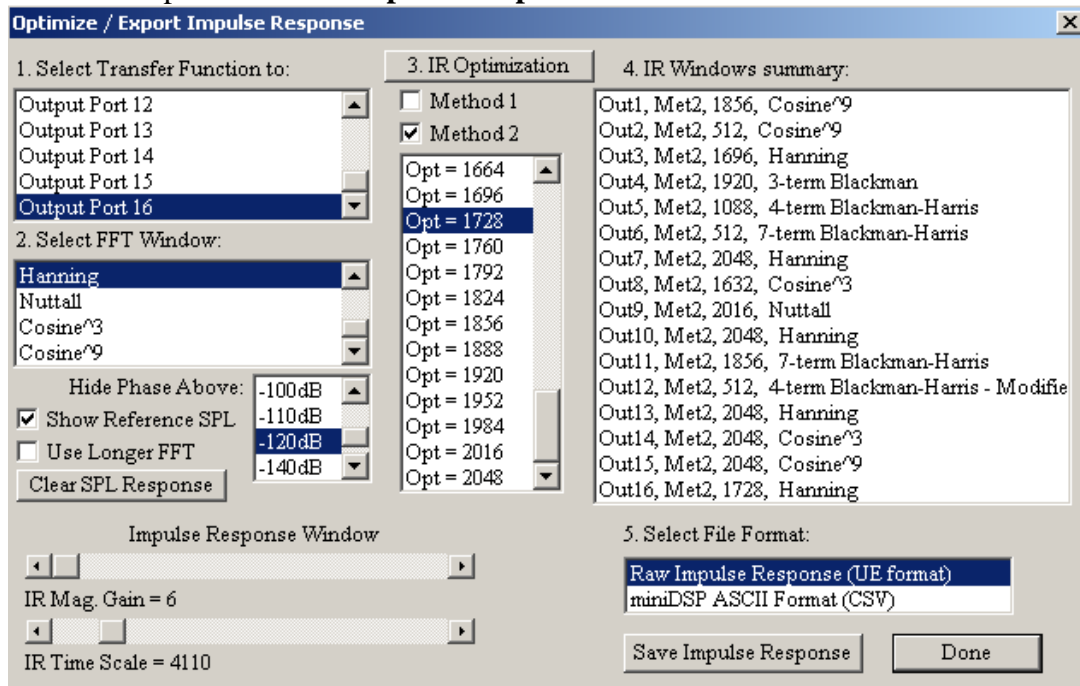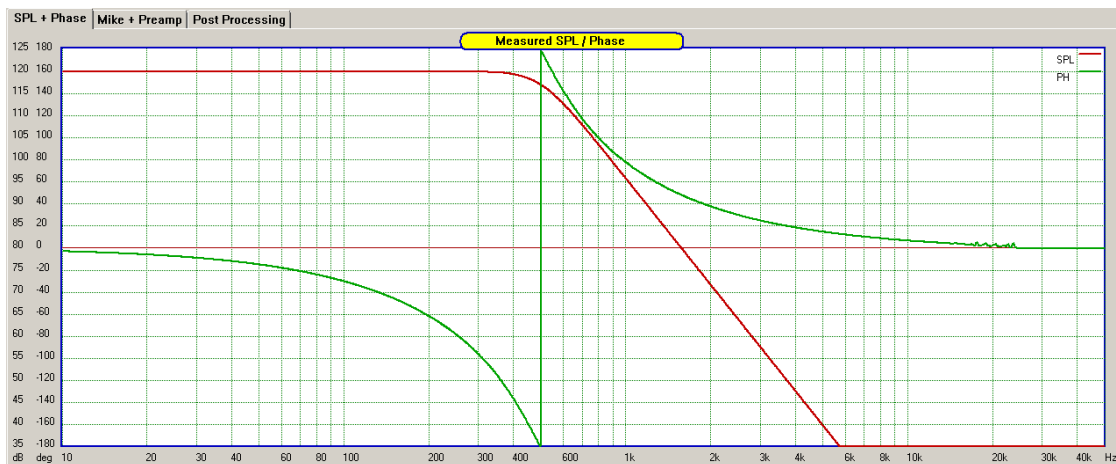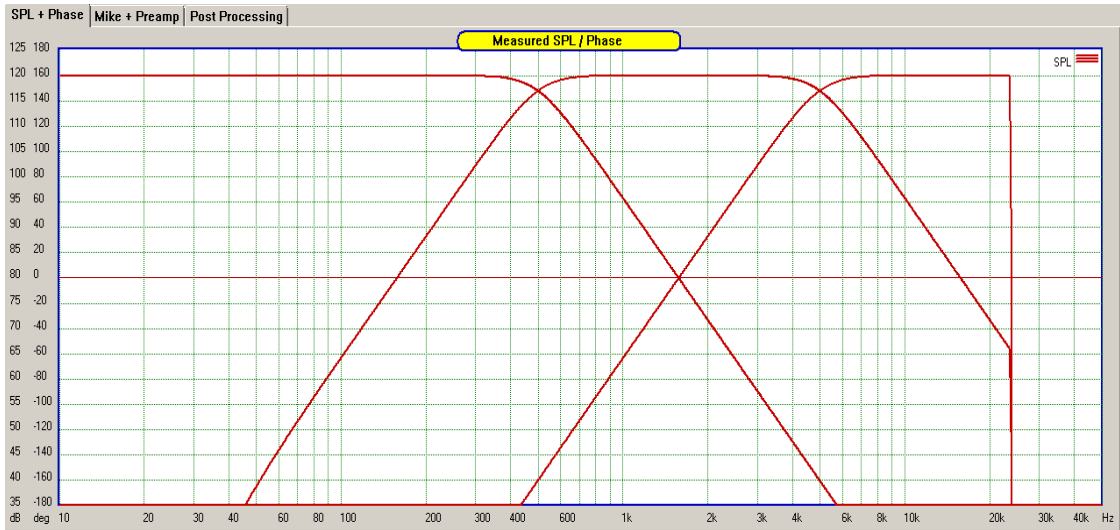

Figure 28. fully edited and optimized set of 16 impulse minimum-phase impulse responses

Typical crossover performance with 2048-long impulse response with 500Hz and 5000Hz, Butterworth, 24dB/oct, Sampling is 48kHz. Minimum-Phase first:
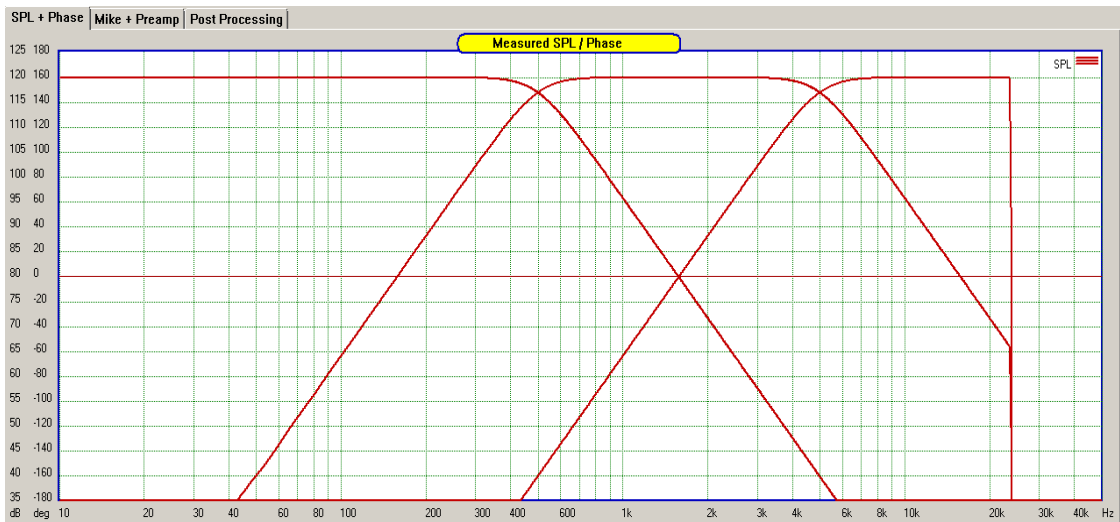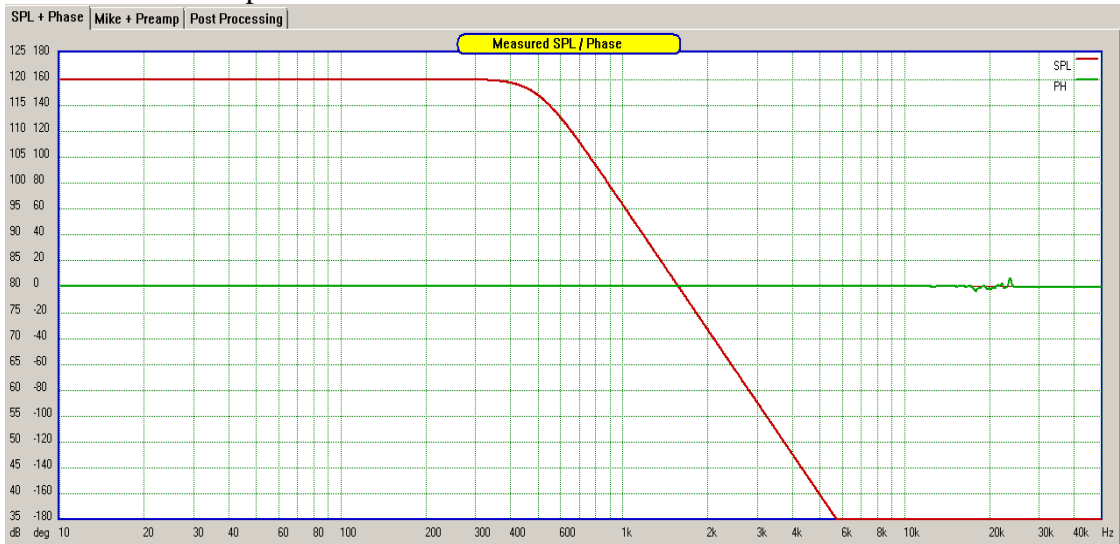
An now the linear-phase version

Figure 29. crossover performance with 2048-long impulse response with 500Hz and 5000Hz, Butterworth, 24dB/oct.

Exporting Impulse Response

After the optimization process, there is the final step in dealing with impulse response – the exporting process. There are two types of files available for exporting: standard ASCII files, and miniDSP formatted files. Data in both files is identical, the only difference is presentation of the data in the files. In order to export the impulse response file, please proceed to step **5. Select File Format** from the provided list box.
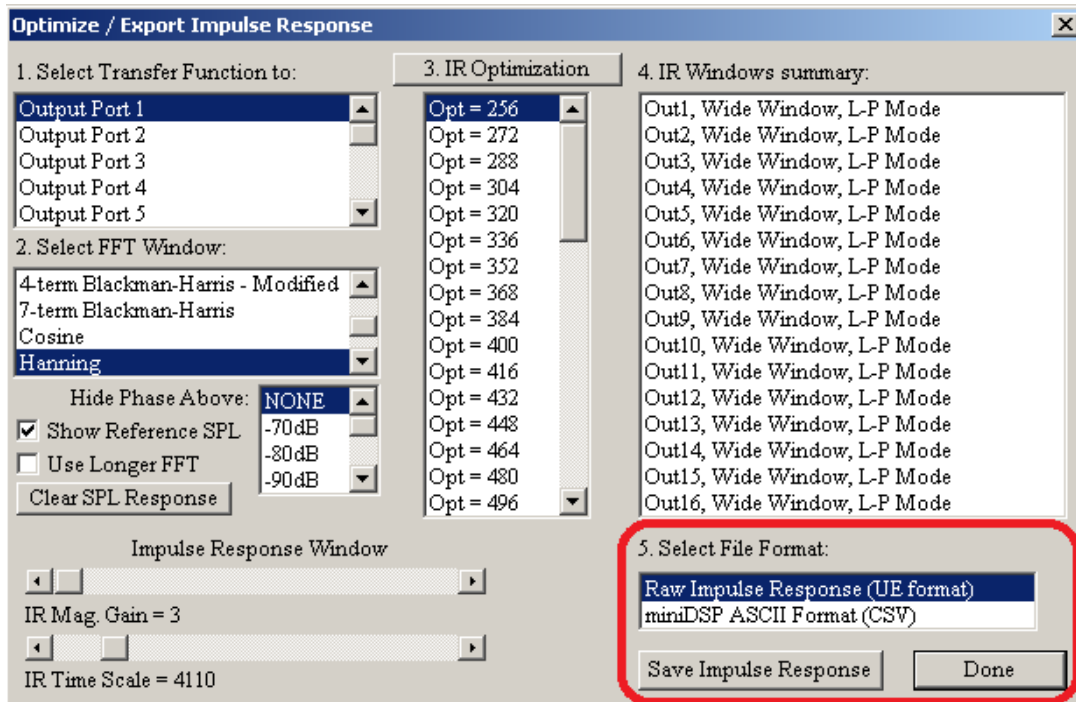

Figure 30. Controls for exporting impulse response.

And then press "**Save Impulse Response**" button. This will open standard Windows file saving dialogue box, where you can enter file name and finalise the export.
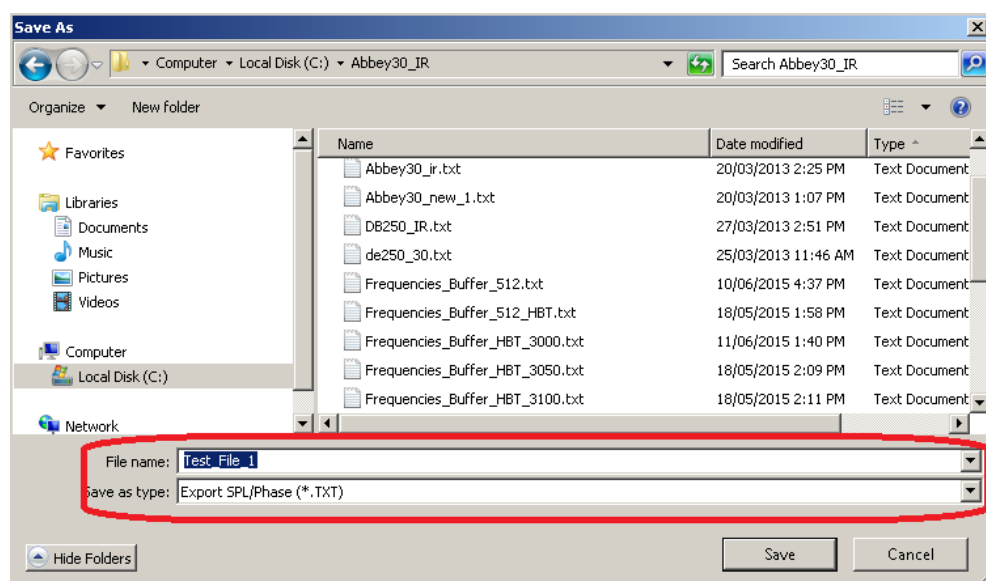

Figure 31. Saving impulse response.

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Standard ASCII files – example shown below

```
-0.000000000000e+00
5.888815834965e-08
-2.461291899181e-07
5.412740051725e-07
-9.798463906918e-07
1.567054596308e-06
-2.201860297646e-06
3.143334879496e-06
-3.998499323643e-06
5.163717105461e-06
-6.407774890249e-06
7.756128979963e-06
-9.573623174219e-06
1.081970822270e-05
-1.319972034253e-05
1.483026244387e-05
-1.729934592731e-05
1.907750629471e-05
-2.249994031445e-05
2.378372664680e-05
-2.795018372126e-05
3.031387859664e-05
-3.335949804750e-05
3.822875805781e-05
-3.838096017716e-05
4.690352943726e-05
-4.574963168125e-05
```

```
2.680297050972e-08
-2.088559192259e-08
4.637163542043e-08
-2.788452135860e-08
5.840448835670e-08
4.858725333179e-08
6.487110937314e-08
-7.634548104818e-09
6.242622418995e-08
7.280408720334e-09
1.043800823552e-09
-4.135016595797e-08
-2.819382594055e-08
-1.617480549498e-08
4.616143911562e-09
-6.656781792458e-10
1.276620498203e-08
1.108814817741e-08
1.231382285027e-08
-4.950949872295e-09
-4.058211988678e-09
-7.198097673466e-09
-3.302155882068e-09
-1.615749645190e-09
1.641004276909e-10
-2.116763535287e-11
```

Figure 32. Start of the ASCII file   end of the ASCII file

miniDSP ASCII Files – example shown below

```
b0  = -0.0000000000e+00,
b1  = 5.8888158350e-08,
b2  = -2.4612918992e-07,
b3  = 5.4127400517e-07,
b4  = -9.7984639069e-07,
b5  = 1.5670545963e-06,
b6  = -2.2018602976e-06,
b7  = 3.1433348795e-06,
b8  = -3.9984993236e-06,
b9  = 5.1637171055e-06,
b10 = -6.4077748902e-06,
b11 = 7.7561289800e-06,
b12 = -9.5736231742e-06,
b13 = 1.0819708223e-05,
b14 = -1.3199720343e-05,
b15 = 1.4830262444e-05,
b16 = -1.7299345927e-05,
b17 = 1.9077506295e-05,
b18 = -2.2499940314e-05,
b19 = 2.3783726647e-05,
b20 = -2.7950183721e-05,
b21 = 3.0313878597e-05,
b22 = -3.3359498048e-05,
b23 = 3.8228758058e-05,
b24 = -3.8380960177e-05,
b25 = 4.6903529437e-05,
b26 = -4.5749631681e-05,
b27 = 5.4350097344e-05,
b28 = -5.5478802096e-05,
b29 = 6.0109054175e-05,
```

```
b2019 = 3.8040393946e-08,
b2020 = 1.9065435897e-09,
b2021 = -7.0939378816e-08,
b2022 = 2.6802970510e-08,
b2023 = -2.0885591923e-08,
b2024 = 4.6371635420e-08,
b2025 = -2.7884521359e-08,
b2026 = 5.8404488357e-08,
b2027 = 4.8587253332e-08,
b2028 = 6.4871109373e-08,
b2029 = -7.6345481048e-09,
b2030 = 6.2426224190e-08,
b2031 = 7.2804087203e-09,
b2032 = 1.0438008236e-09,
b2033 = -4.1350165958e-08,
b2034 = -2.8193825941e-08,
b2035 = -1.6174805495e-08,
b2036 = 4.6161439116e-09,
b2037 = -6.6567817925e-10,
b2038 = 1.2766204982e-08,
b2039 = 1.1088148177e-08,
b2040 = 1.2313822850e-08,
b2041 = -4.9509498723e-09,
b2042 = -4.0582119887e-09,
b2043 = -7.1980976735e-09,
b2044 = -3.3021558821e-09,
b2045 = -1.6157496452e-09,
b2046 = 1.6410042769e-10,
b2047 = -2.1167635353e-11,
```

Figure 33. Start of the miniDSP ASCII file   end of the miniDSP ASCII file.

The number of impulse response coefficients in the exported file depends on the size of UE buffer, mode of operation and UE version (8 partitions or 16 partitions). Currently the 8-partition version is available as follows:

1. Minimum-Phase Mode

UE buffer size	Standards ASCII/miniDSP
512	2048 coefficients
1024	4096 coefficients
2048	8192 coefficients
4096	16384 coefficients

2. Liear-Phase Mode

UE buffer size	Standards ASCII/miniDSP
512	4096 coefficients
1024	8192 coefficients
2048	16384 coefficients
4096	32768 coefficients

You should be able to use miniSHARC DSP board with minimum-phase, 2048 coefficients impulse responses and openDRC with minimum-phase, 2048 coefficients, 4096 coefficients and linear-phase with 4096 coefficients. All sampled at 48kHz.

Other DSP products available on the market may be able to use standard ASCII files exported by program. Sample files of 1kHz LR high-pass and low-pass filters are available on Bodzio Software website.

Importing Impulse Response in ASCII files into MLS system

In addition to importing it's owne impulse response files (.RES files) the MLS system has now the ability to import ASCII impulse response files. The file has to be in the following format – just a sequence of floating point numbers:
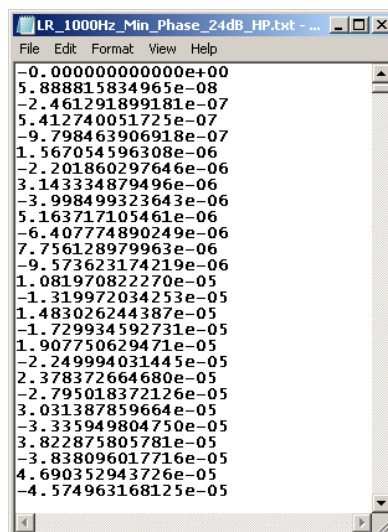
When the program detects the ASCII file import, it will display small information box confirming the number of data point:
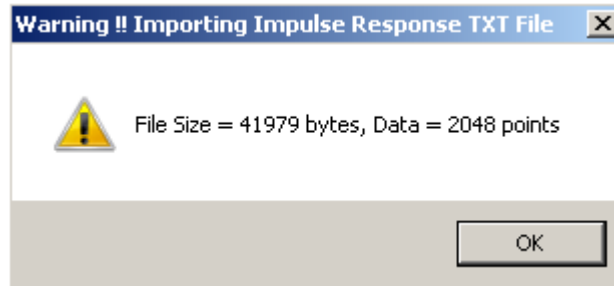
In this case, it is the same file as shown in the exporting example, with 2048 data points. And here is the content of this file:
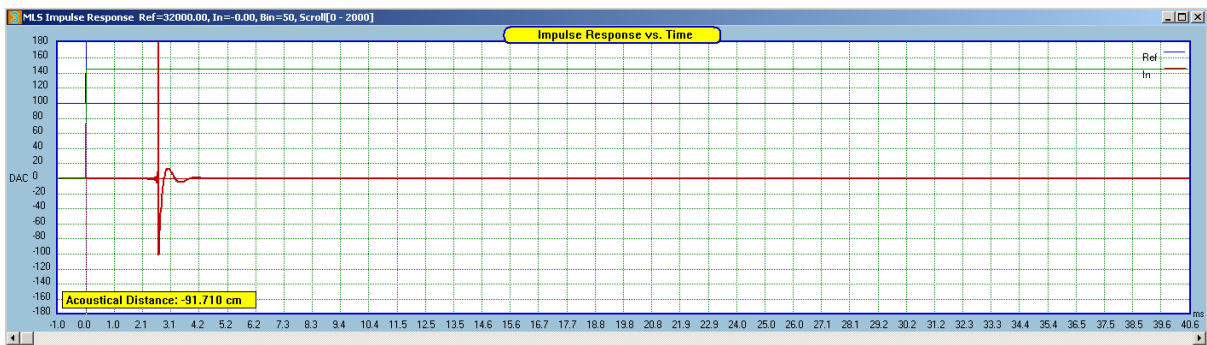

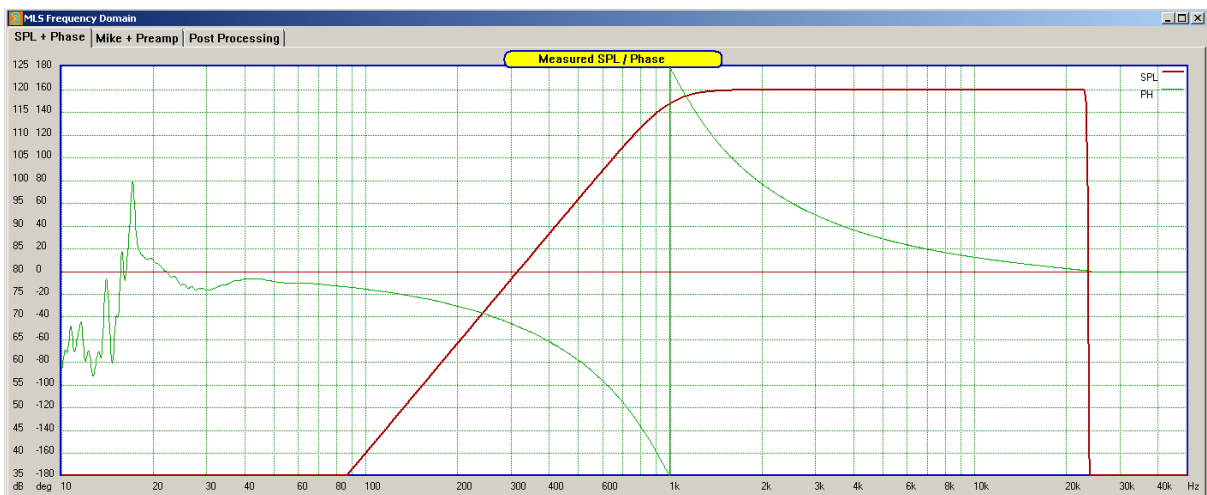Figure 34. Impulse response imported into MLS system.

Figure 35. Corresponding frequency response

File Compatibility

Driver files and project files generated with Version 8 are not compatible with Ultimate Equalizer's older files.

However, SPL/phase can be imported from V7 driver files, so measured frequency response can still be extracted from older files. If you intend to use older files, it is recommended to import those files into V8 and run Hilbert-Bode Transform in File Editor to extend the measured data into the new, wide frequency range of V8.

MLS measurement system in V8 will import impulse response files (.res files) generated by SoundEasy, so you can use driver measurements performed by SoundEasy.

Recommended Reading

"**Digital Signal Processing A Practical Guide for Engineers and Scientists**" by Steven K Smith. Newnes - an imprint of Elsevier Science. **ISBN 0-75067444-X**