RT60

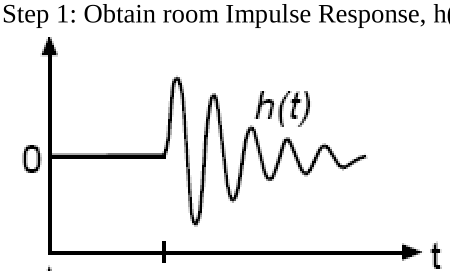
Reverberation Time (RT60) is recognized as the most important acoustical room parameter. It is defined as a time interval required for the sound energy to decay 60 dB after the excitation has stopped. The RT60 is estimated from the **Energy Decay Curve, L(t),** after the sound source is switched off. The energy decay curve is typically irregular and noisy curve, and some more mathematical processing needs to be done to enable a reliable calculations of the RT60.

The method of calculating the reverberation time (RT60) is due M. Schroeder, ("New Method of Measuring Reverberation Time". J. Acoust. Soc. Am., vol. 37 (1965) pp. 409-412). Schroeder has shown by statistical analysis that the room averaged energy decay r(t), can be obtained from the **backward integrated squared impulse response h(t)**. Here more complete formula for L(t):

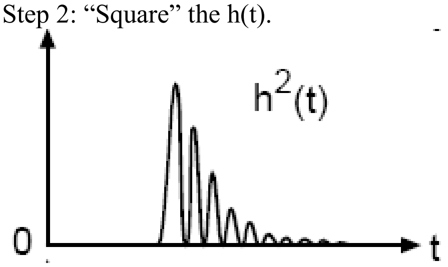
$$L(t) = 10\log(N\int_t^{\infty} h^2(\tau)d\tau)$$

Where: N is the noise power per unit bandwidth. The noise is considered to be a “stationary white noise”.

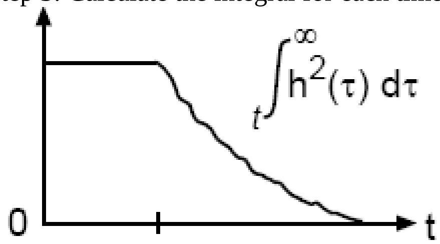
Step 1: Obtain room Impulse Response, h(t) using MLS technique.



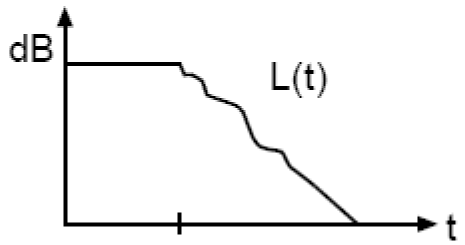
Step 2: “Square” the h(t).



Step 3: Calculate the integral for each time step from t = 0.



Step 4: Improve the RT curve by linear approximations (optional).



The **ISO3382** (ISO-3382, Acoustics – measurement of the reverberation time of rooms with reference to other acoustical parameters. 1997) also defines two additional parameters:

T20 - is the reverberation time determined from the average slope of the energy decay curve obtained from part of the decay curve between -5dB and -25dB.

T30 - is the reverberation time determined from the average slope of the energy decay curve obtained from part of the decay curve between -5dB and -35dB.

These can be useful, if the background noise level in the room makes it difficult to estimate RT60 from the L(t) curve.

Noise in RT60 measurements

Background noise is most likely to be present during your RT60 measurements and will manifest itself as a long, noisy tail after the RT60 has decayed to negligible values. In fact, many RT60 measurements do not even result in the decay curve shown over the full 60dB range. Simply because the noise level may be significantly higher than bottom end of this curve.

It would be beneficial to reduce or remove the noise from the measurements, so that a larger portion of the decay curve was available for estimation of RT60, thus making the estimations more reliable.

Two methods are being proposed:

1. **Truncation Method** – When selecting this method, the noise tail is simply chopped off after the selected truncation point. The essence of this method is to select the truncation point such a way, that the resulting RT60 curve is most linear, when measuring a single, decoupled room. The truncation point is selected by how much of the noise tail is to be removed from the calculations. It is simply the percentage of the Impulse Response (tail section) to be removed.

2. **Subtraction Method** – When selecting this method, the mean-squared value of the noise in the tail is being subtracted from the measurement before the Schroeder Backward Integration takes place.

The **Subtraction Method** was described by W.T Chu in “Comparison of reverberation measurements using Schroeder's impulse method and decay-curve averaging method” in J. Acoust. Soc. Am. 63(5), May 1978.

$$\left\langle s^2(t) \right\rangle = N\int_t^{\infty}[r(\tau)+n(\tau)]^2\,d\tau = N\int_t^{\infty}[r^2(\tau)+2r(\tau)n(\tau)+n^2(\tau)]d\tau$$

The $\mathbf{s^2(t)}$ is the ensemble average of the squared decaying sound pressure at a receiving point. The $\mathbf{r(\tau)}$ is the desired and measured signal, and $\mathbf{n(\tau)}$ is the unwanted noise. N is proportional to power spectral density.

As **n(τ)** be eider positive or negative, the second term will integrate to zero. The third term is the one, that needs attention. Mean-squared value of the noise tail after truncation point is calculated and subtracted from the **n^2(τ)**.

$$\bar{n}^2 = \frac{1}{L}\sum_{i=1}^{L} n_i(\tau) \qquad \text{therefore:}$$

$$\langle s^2(t)\rangle = N\int_t^{\infty}[r^2(\tau) + 2r(\tau)n(\tau) + (n^2(\tau) - \bar{n}^2)]d\tau$$

When using Subtraction Method for RT60 estimations, you should consider several factors affecting the process. Here is an example. Say, for instance that you have measured a medium-size room with **MLS of 262k** length and sampling rate of **48k.** This will result in **IR length = 262/48 = 5.45sec.** The Schroeder integral is calculated right to the end of the IR, so this is quite a long response to calculate. If your room has an RT60 of 300ms, for example, you will end up with rather short decay curve and a very long noise tail. The noise tail is **18 times longer than the useful recorded measurement**. The power of the noise tail will effectively compare with the power of the useful section of the recorded Impulse Response and will affect accuracy of the subtraction method. Simply speaking, too much power will be subtracted from the Schroeder curve.

In a situation like this, it is recommended, that you go back to MLS TAB and select MLS length as 32k. Do not repeat the measurement. This way, effective IR length taken into account for Schroeder integral RT60 calculations is 32k instead of 262k (or **IR length = 32/48 = 0.666sec)**. This way, the noise tail is only about **2 times longer then the useful recorded measurement**, so the balance is much better. Noise reduction methods greatly improve the estimation of RT60.

Making RT60 measurements

The process of measuring RT60 starts with collecting room Impulse Response using MLS signal, which is accomplished from the MLS TAB. All steps involved in setting up MLS generator, connecting your hardware are the same as for making standard SPL measurements, with some modifications. Here are some additional, simple recommendations for your measurement setup.

1. Microphone position must be at least 1 m from reflecting surfaces and not too close to any source position; the minimum distance from a source position can be calculated by the equation:

$$d_{\min} = 2\sqrt{\frac{V}{cT}[m]}$$

   where: *V* is the room volume, *c* is the speed of sound, *T* is an estimate of the expected reverberation time.

2. The sound source should be preferably omni-directional.
3. The microphone should be omni-directional.

Also, one needs to be sensible about selecting the MLS length. This will affect the processing time for RT60 (Schroeder Backward Integration of 262k IR is a very long process) and also the averaged noise power in the tail. If you encounter this kind of issue, you can “Abort” and recover following the hints in the last paragraph.

From the definition of RT60, we understand, that “the clock starts ticking” after the excitation signal stopped. Therefore, you need to discard the direct signal contribution from the collected IR. To accomplish this, you need to select the MLS IR screen and move the cursor past the direct response pulse. Where exactly you place the cursor?. This will depend on your measurement setup, and obviously the distance between your measurement microphone and the speaker. Figure 16.61 below shows an example of removing the direct sound signature from the IR.

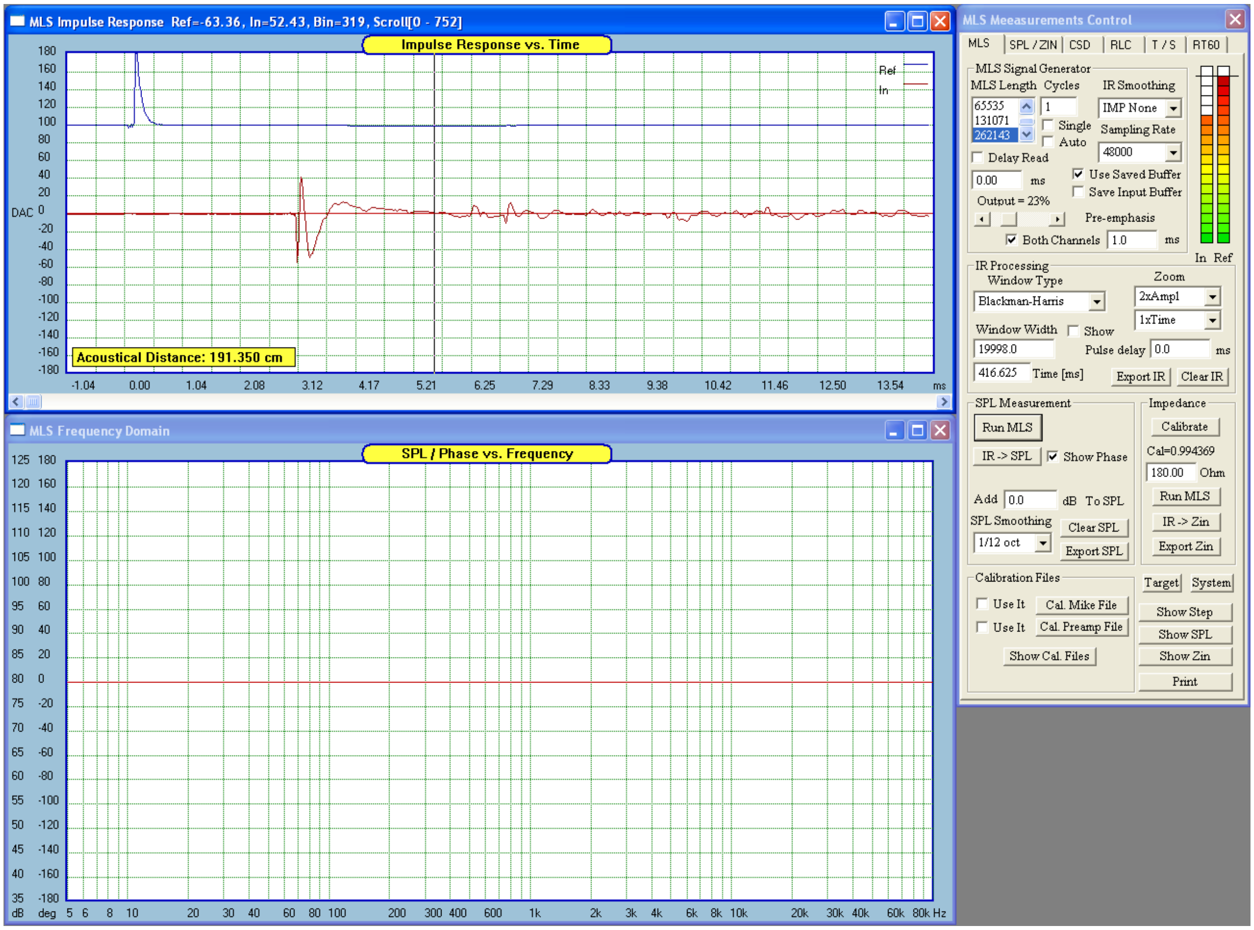
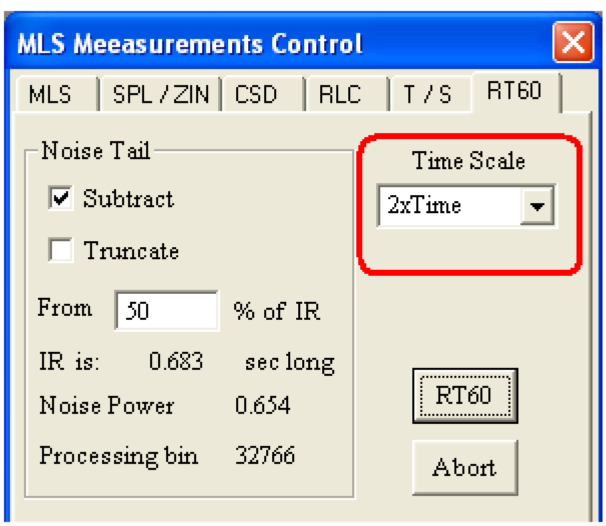


Figure 16.110 Removing the “direct sound” from collected IR.

When you open the RT60 display screen, you should consider selecting the time scale first. This is needed for proper display of long room IR, mapped onto the selected screen time scale. You can select time scale from **300ms to 16 * 300ms = 4.8 sec**. Obviously, you would select longer time scale for the expected longer RT60. Please note, that this is not a zoom function. The time scale is selected from the available list box.



Initially, you may select **Noise Tail + Truncate + 0% of IR settings**. Next, hit the **RT60** button and you should see that the program is processing measured IR bins and calculates RT60. Assuming everything went OK, you should see something similar to Figure 16.96 below.

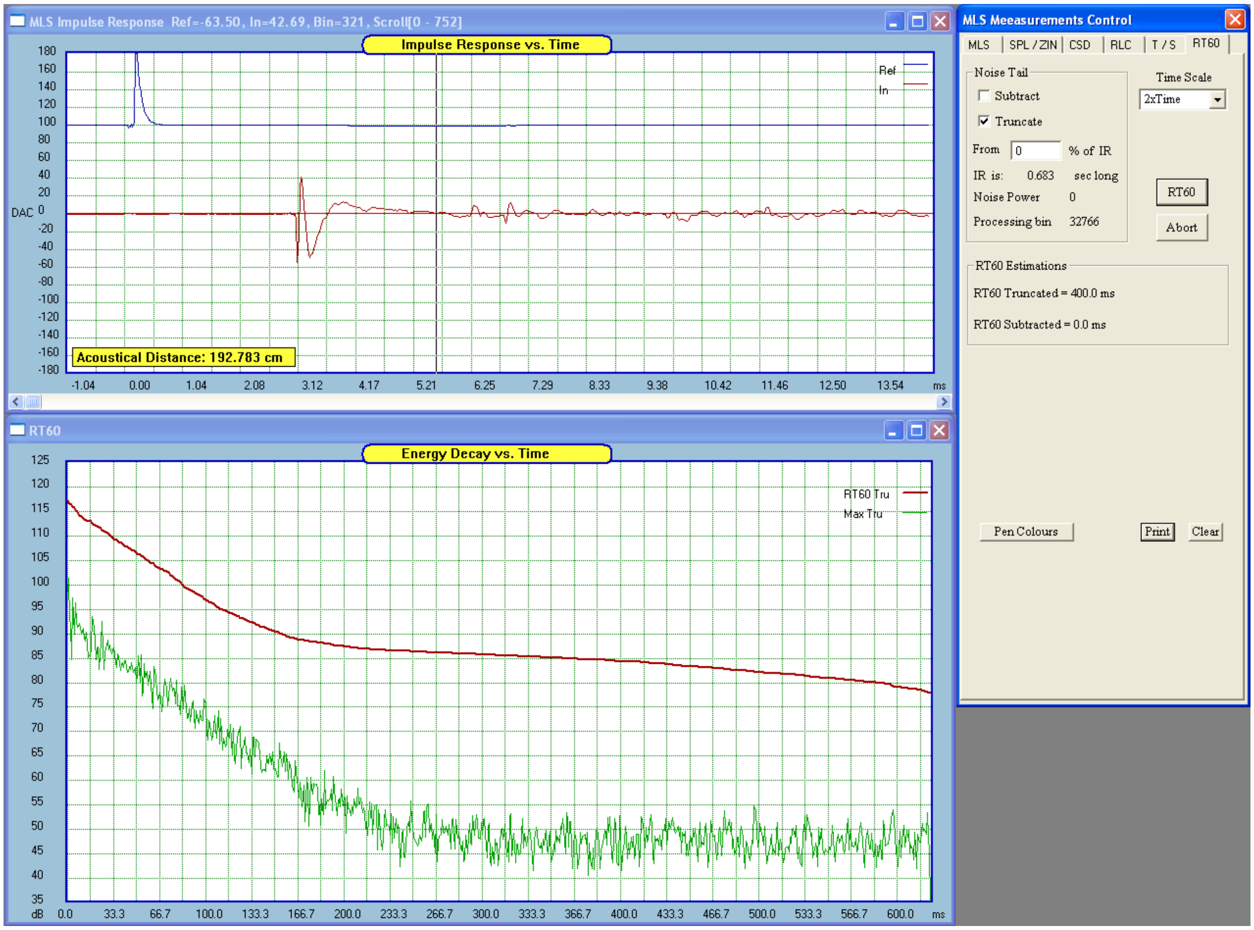


Figure 16.111. RT60 calculated without noise reduction techniques.

It is immediately observable, that the noise tail is contributing to the integration process, and rising the RT60 curve towards the right side of the plot. This prevents the RT60 curve to attain much better linearity.

In this test, the RT60 “Truncated” result is 400ms, and it is quite clear, that this result is not based on linear RT60 curve. In the next step, you can try to truncate say, 40% of the noise tail by entering “40” in the “From” data field in the “Noise Tail” group.

You can now try to calculate RT60 Truncated again, and visually inspect it’s linearity. The result is presented on the Figure 16.97 below. Now, the RT60 Truncated curve looks much better. By doing this simple exercise, you can easily observe, that RT60 curve has almost lost the flatter section, therefore, you are obviously moving in the right direction.

In addition, if you try to calculate RT60 using “Subtraction” method (check the “Subtract” check box and press the RT60 button again), you will notice, that the RT60 (green) curve is quite linear up to 233ms. At this point, the curve is -45dB below the maximum value, and can be used for calculating **RT30** (time interval required for the sound energy to decay 30 dB after the excitation has stopped). This is exactly what the algorithm does. It will calculate RT30 and double it’s value to get the RT60.

In the final step, RT60 Subtracted with 50% truncation (**RT60 = 305ms**) and RT60 Truncated with 65% truncation (**RT60 = 310ms**) were deemed to be the optimum. Please note, that RT60 values were 1.6% from each other – see Figure 16.98.

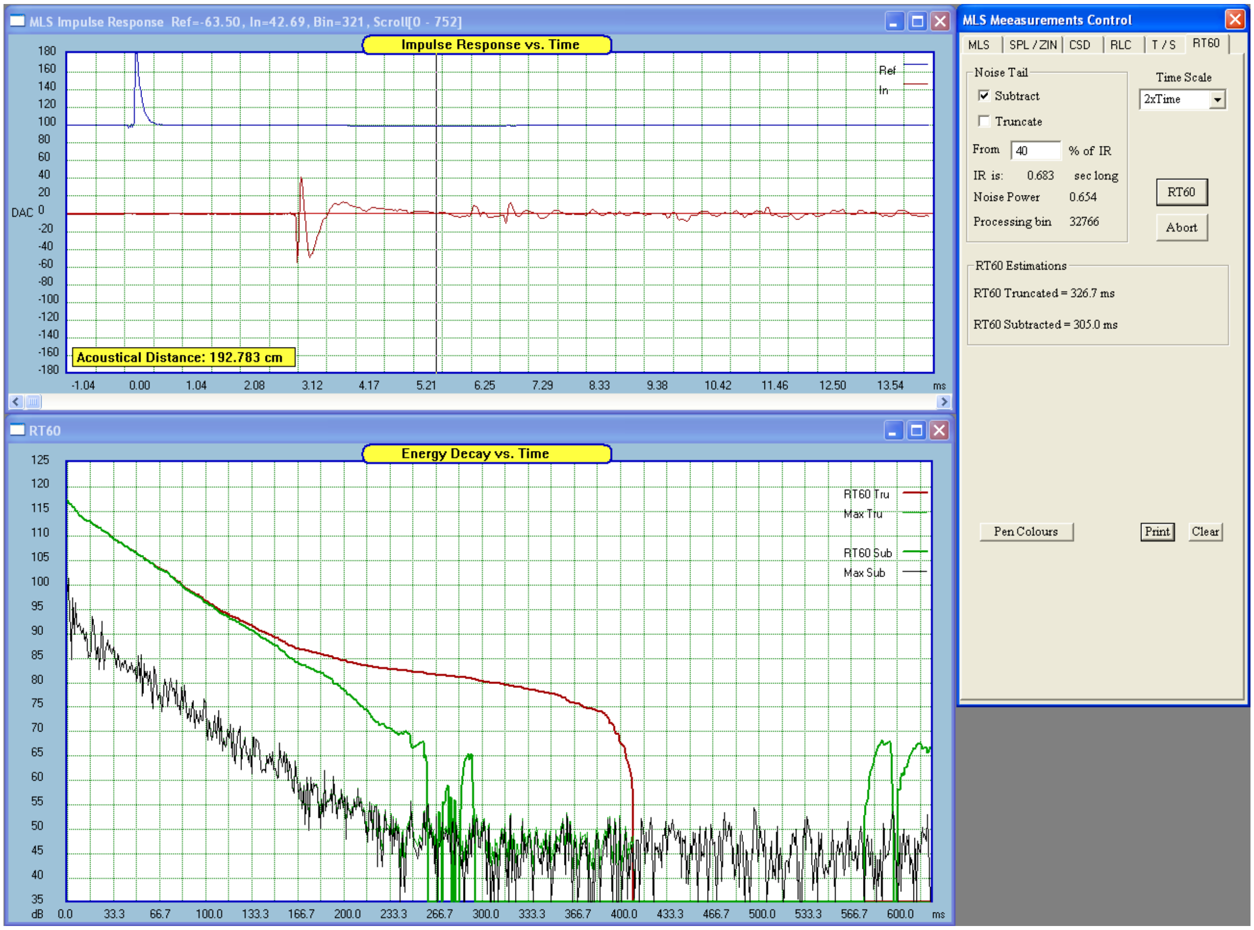


Figure 16.112. RT60 calculated with 40% of the noise tail truncated.

Description of the RT60 TAB Controls

1. **Noise Tail group**
   "Subtract" – check to calculate RT60 using subtraction method.
   "Truncate" – check to calculate RT60 using truncation method.
   "From" – enter the percentage of the tail you wish to be truncated.
   "IR is xxxxx sec long" – indication of how long is the collected room IR. Comes handy for estimating Time Scale setting.
   "Noise Power" – a figure of merit. If the noise tail is fairly flat, this figure will not change much for different truncation points.
   "Processing bin xxxxxx" – indication where you are with Schroeder integration process. You can "Abort" and change MLS length setting if the process drags forever.

2. **RT60 Estimations**
   "RT60 Truncated" = xxxxx ms – Result of Truncation method
   "RT60 Subtracted" = xxxxx ms – Result of Subtraction method

3. **Time Scale –** controls the time scale length
   "1x xTime" – Sets the time scale 0 - 300ms.
   "2x xTime" – Sets the time scale 0 - 600ms.
   "4x xTime" – Sets the time scale 0 - 1200ms.
   "8x xTime" – Sets the time scale 0 - 2400ms.
   "16 xTime" – Sets the time scale 0 – 4800ms.

4. **Abort** – Button to abort the RT60 process.
5. **Pen Colors** – Button to invoke pen colours dialogue.
6. **Print** – prints the screen.
7. **Clear** – Clears the screen

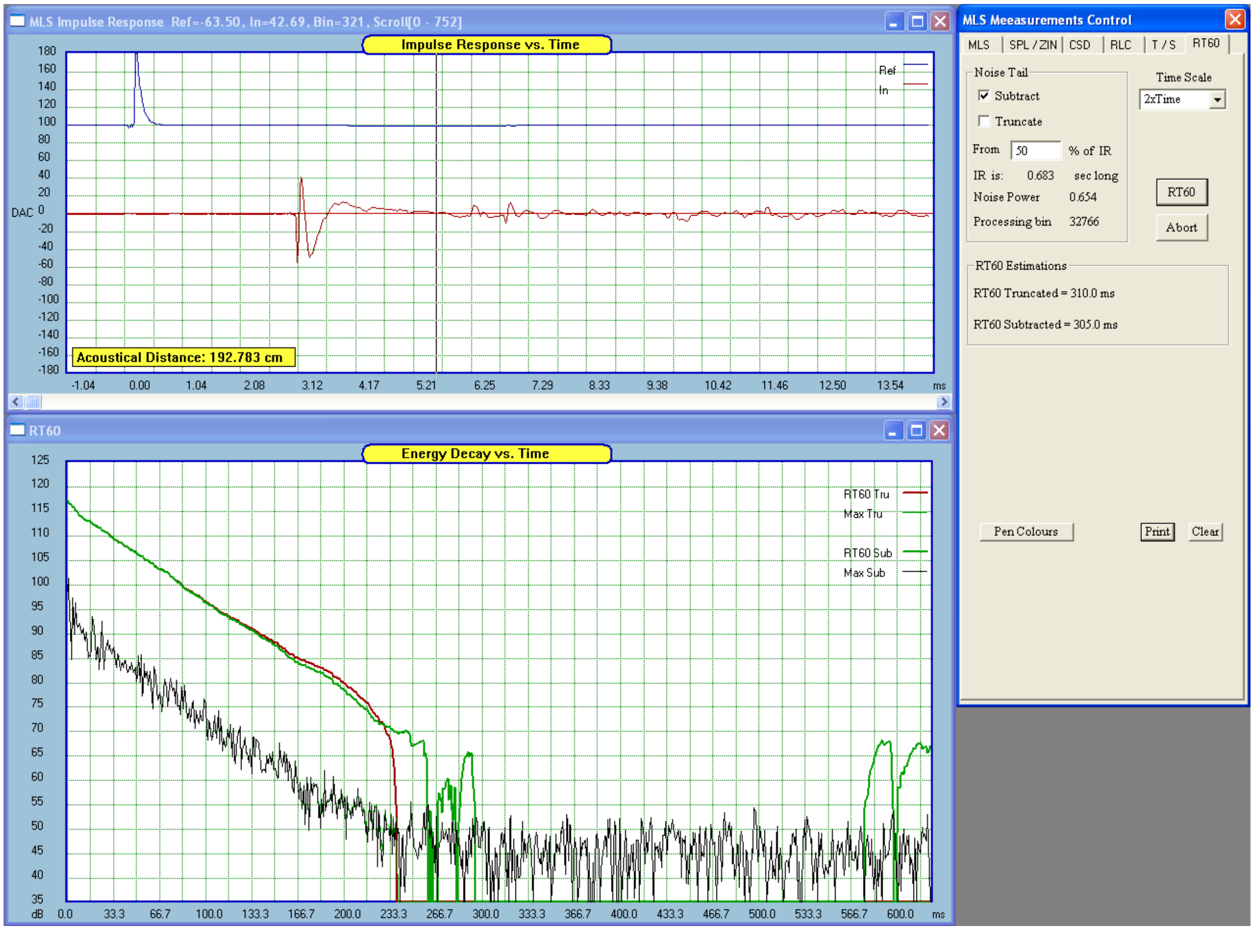


Figure 16.113. RT60 Subtracted with 50% truncation and RT60 Truncated with 65% truncation.

THD and IMD System

Non-linear effects in devices can be approximately modeled by Maclaurin series expansion. This assumes that the device under test can be considered memoryless.

$$V_0 = K_0 + K_1 V_i + K_2 V_i^2 + K_3 V_i^3 + ...$$

Harmonic Distortion

It turns out that the simplest and most practical way to separate out the individual distortion components from the linear response is to use a sine wave as the excitation signal. Since distortion is very level dependent, using a sine wave as the test signal makes interpreting input and output levels very straightforward.

For single tone input: $V_i = V_1 \cos(\omega_1 t)$

1st order (fundamental) $K_1 V_1 \cos(\omega_1 t)$

2nd order (including 2nd harmonic) $K_2 V_1^2 (\frac{1}{2} + \frac{1}{2}\cos(2\omega_1 t)$

3rd order (including 3rd harmonic) $K_3 V_1^3 [\frac{3}{4}\cos(\omega_1 t) + \frac{1}{4}\cos(3\omega_1 t)] \ldots\ldots\ldots$

If we develop the 2nd order and 3rd order terms in the Maclaurin series for a single tone input, we obtain distortion terms at the multiples of the fundamental frequency. These correspond to the 2nd order and 3rd order harmonics. Note that the level of the second harmonic is proportional to the square of the fundamental and that the level of the third harmonic is proportional to the cube of the fundamental. If the fundamental level changes by some number of dB, the level of 2nd and 3rd order harmonics change by two and three times that number of dB, respectively. For example, a 1 dB increase in the fundamental results in a 2 dB increase in the 2nd harmonic and a 3 dB increase in the 3rd harmonic.

However, harmonic distortion is typically specified *relative* to the fundamental level. This means that for the above example the *difference* between the 2nd harmonic and the fundamental will be 1 dB *less* than it was, and the *difference* between the 3rd harmonic and the fundamental will be 2 dB *less*. Therefore, when specifying the relative or absolute level of the 2nd harmonic distortion, for example, it is imperative to also specify the level of the fundamental at which the distortion was measured. Once this is provided, the 2nd harmonic distortion can theoretically be predicted for any power level at the fundamental. However, this prediction only holds true for the more linear section of the power transfer function of the device, so it can only model distortion in devices under small signal excitation.

Notice that the development of the 3rd order term in the Maclaurin equation also provides a distortion term at the fundamental frequency. This term may look larger than the 3rd order harmonic, but it is not significant relative to the fundamental. So, we are typically not worried about distortion at the fundamental frequency until the device gain or phase starts changing.

Total Harmonic Distortion can be calculated by measuring the levels of all harmonics and comparing them to the fundamental.

$$THD = 100\sqrt{\frac{V_1^2 + V_2^2 + V_3^3 + \ldots\ldots\ldots}{V_0^2}}$$

Where Vi are voltage levels of each harmonic, and Vo is the output voltage.

Asymmetrical system nonlinearities cause only even order distortion products. Signals, like the positive or negative peak limited sine wave, limited only on the upper or lower half-cycle, contain higher amplitude even order harmonics than odd order harmonics.

For illustration purpose, example below was created by clipping a sine wave with a diode, thus creating one-sided, soft clipping waveform.

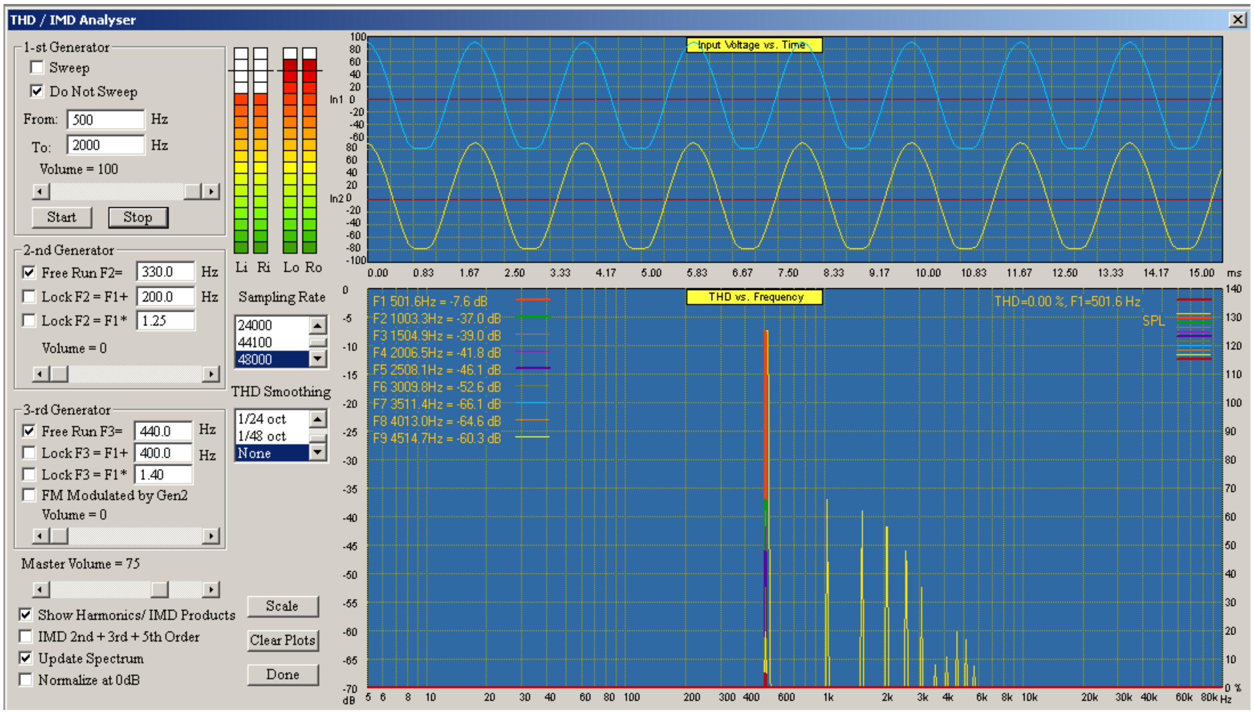


Figure 16.114. Asymmetrical clipping spectrum.

Symmetrical system nonlinearities cause only odd order distortion products. Signals, like the positive and negative limited sine wave, which will look like a square wave if limited enough, contain higher amplitude odd order harmonics than even order harmonics.

Figure below shows F3, F5, F7, F9…… at significantly higher levels than F2, F4, F6 and so on.

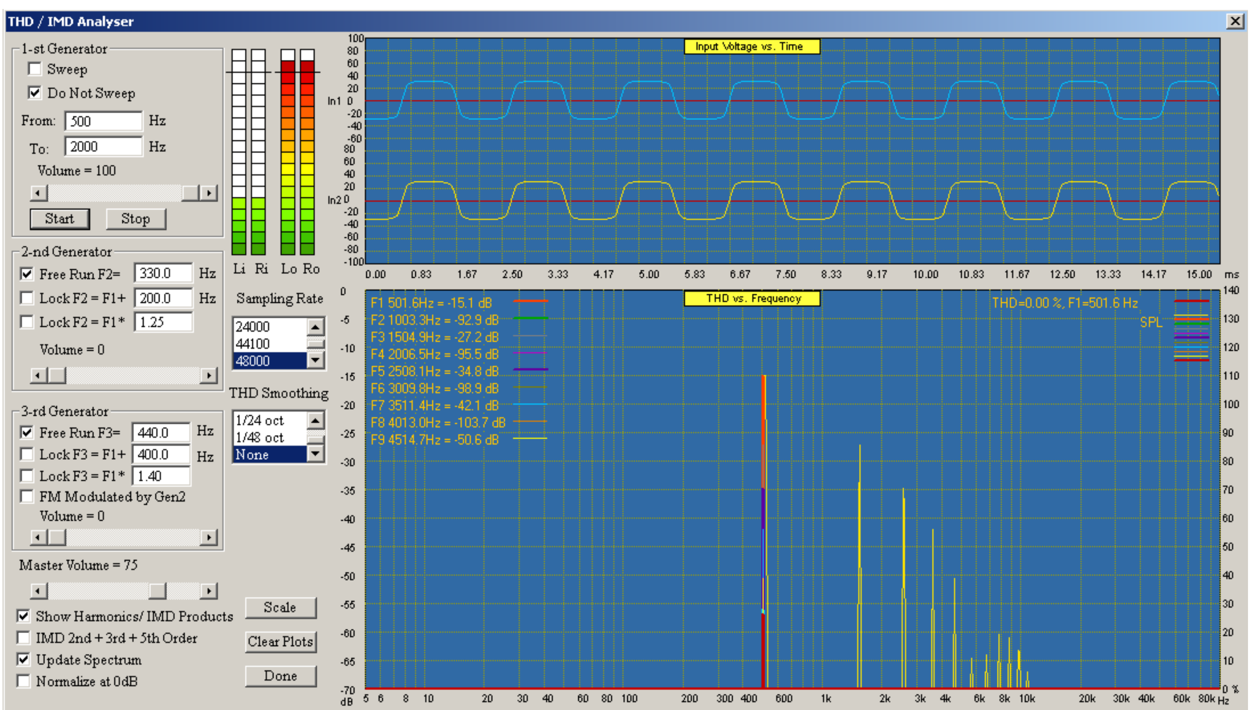


Figure 16. 115. Symmetrical clipping spectrum.

Intermodulation Distortion

For two-tone input: $$V_i = V_1[\cos(\omega_1 t) + \cos(\omega_2 t)]$$

3rd order: $$K_3 V_1^3[\frac{3}{4}\cos(2\omega_1 - \omega_2)t + \frac{3}{4}\cos(2\omega_2 - \omega_1)t + \frac{9}{4}\cos(\omega_1 t) + \frac{9}{4}\cos(\omega_2 t) + ...]$$

Intermodulation distortion is the result of two or more signals interacting in a non linear device to produce additional unwanted signals. These additional signals (intermodulation products) occur mainly in devices such as amplifiers and speakers.

Two interacting signals will produce intermodulation products at the sum and difference of integer multiples of the original frequencies. For two input signals, the output frequency components can be expressed as:

mF1 ± nF2 where, m and n are integers

The order of the intermodulation product is the sum of the integers m+n. The 'two tone' third order components, (2*F1-F2 and 2*F2-F1) are particularly important because unlike 2nd order distortion, i.e. harmonic distortion at 2*F1 or 2*F2, they can occur at frequencies close to the desired/interfering signals and so cannot be easily filtered. Higher order intermodulation products are generally less important because they have lower amplitudes and are more widely spaced. The remaining third order products, 2F1+F2 and 2F2+F1, do not generally present a problem. The distribution of harmonics and third order products are shown in figure below.

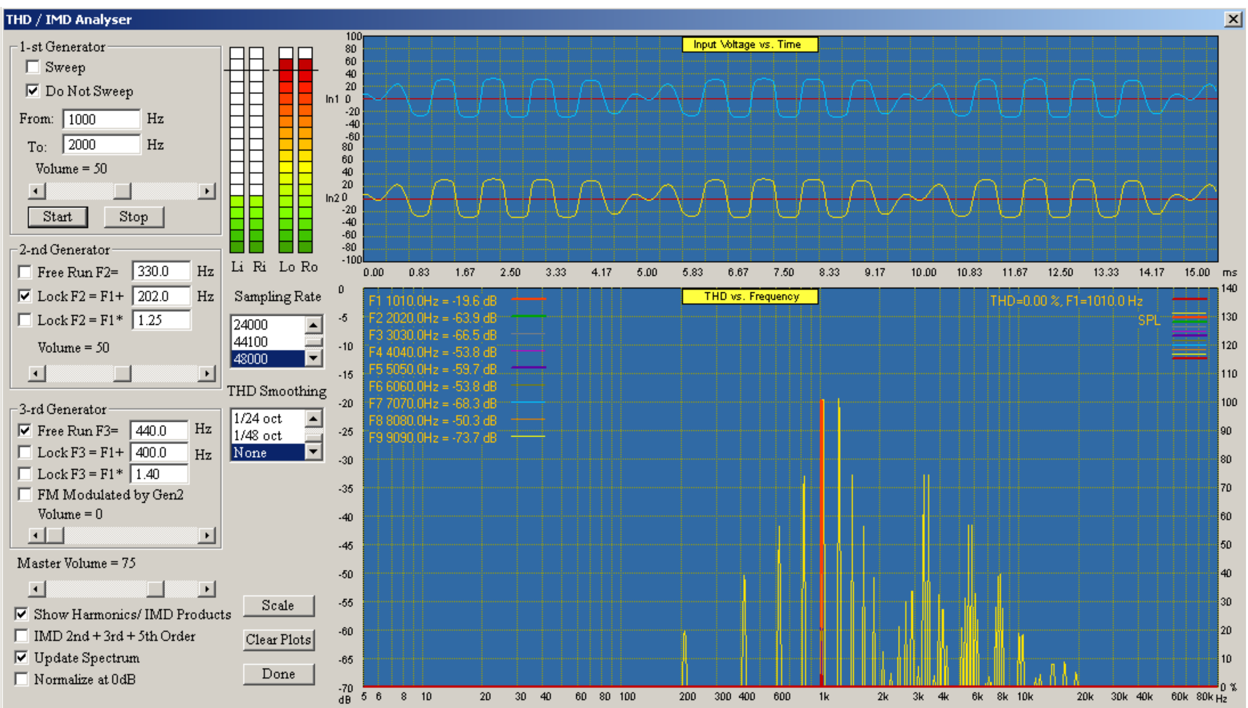


Figure 16. 116. Intermodulation distortion

Picture above shows distributions of 3rd, 5th and higher order products:

F1 = 1000Hz
F2 = 1200Hz
F2 – F1 = 1200 – 1000 = 200Hz (2nd order)
F2 + F1 = 12000 +1000 = 2200Hz (2nd order)
2*F1 – F2 = 2000 - 1200 = 800Hz (3rd order)
2*F2 – F1 = 2400 – 1000 = 1400Hz (3rd order)
2*F2 – 2*F1 = 2400 – 2000 = 400Hz (3rd order)
3*F1 – 2*F2 = 3000 – 2400 = 600Hz (5th order)
3*F2 – 2*F1 = 3600 – 2000 = 1600Hz (5th order)
4*F1 – 3*F2 = 4000 – 3600 = 400Hz (7th order)
4*F2 – 3*F1 = 4800 – 3000 = 1800Hz (7th order)
5*F1 – 4*F2 = 5000 – 4800 = 200Hz (9th order)
5*F2 – 4*F1 = 6000 – 4000 = 2000Hz (9th order) And so on…..

Another advantage of two-tone interaction distortion measurements is that they can be used over the entire frequency range of the system, whereas harmonic distortion measurements become meaningless when the distortion products approach the system's frequency limits. The two-tone CW distortion measurement is the most common test used to characterized the 3rd order IMD in a device. As with any 3rd order distortion, when the level of the fundamental increases by some number of dB, the level of the IMD will increase by three times that number in dB, but its level *relative* to the fundamental will *decrease* by *twice* that number in dB. Therefore, when specifying the relative or absolute level of 3rd order IMD, the level of the fundamental must also be specified. Once this is provided, the 3rd order IMD can also be theoretically predicted for any power level at the fundamental, assuming small signal excitation conditions. Note that the two fundamental tones are shown equal in amplitude. This is the usual case but is not necessary.

There are many different terms used to specify 3rd order IMD. IM3 or Intermodulation Ratio (IMR) are two different names for the same thing. As explained earlier, the distortion *relative* to the fundamental tones is typically specified along with the level of the fundamental tones (assumed here to be equal).

The Third Order Intercept (TOI) is the theoretical power level for the fundamental at which the 3rd order IMR is 0. In reality, the 3rd order distortion curve cannot be predicted for large signal levels, at which the power transfer function is no longer linear. However, the TOI is used as a single figure of merit from which the IMR can be calculated (or vice versa) given a power level for the fundamental. The 2nd order harmonic curve can also be calculated by knowing the 2nd order harmonic distortion at a certain power level at the fundamental, and by knowing that the slope is 1:1.

Intermodulation distortion are calculated accordingly the following formula:

$$IMD = 100\sqrt{\frac{V_{2F1-F2}^2 + V_{2F2-F1}^2 + V_{3F1-2F2}^2 + V_{3F2-2F1}^2 + V_{F1+F2}^2 + V_{2F1+2F2}^2 + V_{2F2-2F1}^2}{V_{F1}^2}}$$

It is observable, that the formula includes 3-rd and 5th order intermodulation products. Inspecting Figure 16.67, one can conclude, that contribution of 7th and higher order IMD are 12dB less than the 5th , and are 36dB below F1 and F2 at IMD=26%. Consequently the IMD figure will not suffer by disregarding higher products.

The THD/IMD system has a three-tone generator built in. The 3-tone measurement is more involving, mainly due to larger number of frequencies to examine.

For three-tone input: $V_i = V_1[\cos(\omega_1 t) + \cos(\omega_2 t) + \cos(\omega_3 t)]$

3rd order:
$$K_3V_1^3[\frac{15}{4}\cos(\omega_1 t)+\frac{15}{4}\cos(\omega_2 t)+\frac{15}{4}+\cos(\omega_3 t)......]$$ 1- tone terms

$$K_3V_1^3[\frac{3}{4}\cos(2\omega_1-\omega_2)t+\frac{3}{4}\cos(2\omega_2-\omega_1)t+\frac{3}{4}\cos(2\omega_1-\omega_3)t+$$
$$\frac{3}{4}\cos(2\omega_3-\omega_1)t+\frac{3}{4}\cos(2\omega_3-\omega_2)t+\frac{3}{4}\cos(2\omega_2-\omega_3)t]$$ 2-tone terms

$$[\frac{3}{2}\cos(\omega_1+\omega_2-\omega_3)t+\frac{3}{2}\cos(\omega_1-\omega_2+\omega_3)t+\frac{3}{2}+\cos(\omega_2+\omega_3-\omega_1)t......]$$ 3-tone terms

FM Modulated Signals

Subwoofer THD stress-testing can be accomplished by generating one large signal (V1 from Gen3) to move the voice coil into the nonlinear region of BL/Cas and then injecting desired sweep test (V2 from Gen1) signal to obtain harmonic content at large cone excursions. V1 signal can be frequency modulated with the help of Gen 2, which provides modulating frequency – up to 20Hz and deviation – up to 50%.

Such signal is quite complex and allows the large-amplitude signal to be moved around resonant frequency of the subwoofer driver.

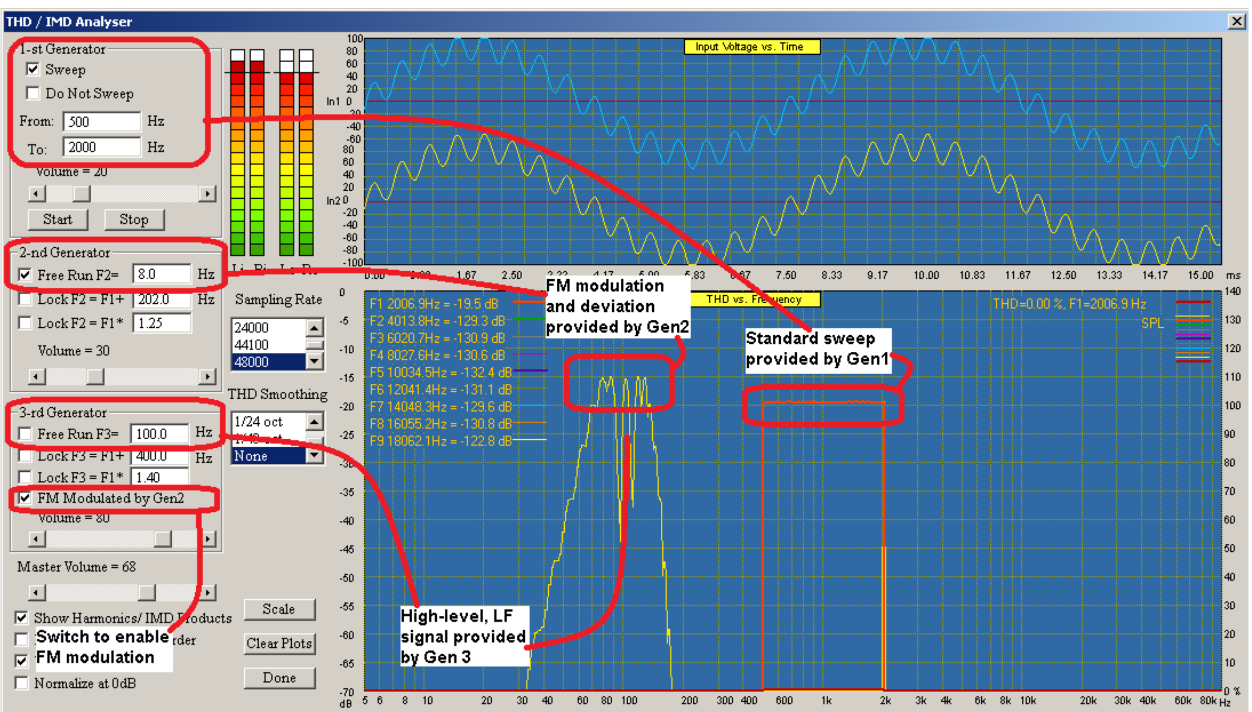
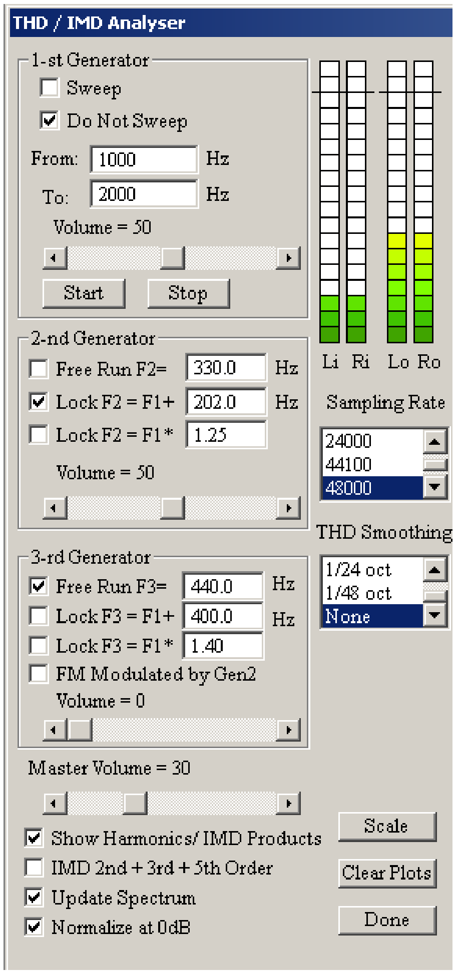


Figure 16.117. Settings for FM modulated signal from Gen 3.

THD System Controls

THD System can be invoked from the main menu “EasyLab” -> “Analog THD + IMD” option. The THD+IMD System consists of:

1. Time-domain window. This window is used to display the input voltage into the sound card.
2. Frequency-domain window. This window is used to display all captured signals, and transferred into frequency domain.
3. THD Control box – shown below.



THD + IMD Control Box

1. 1-st Generator – **“From”** starting frequency of the sweep.
2. 1-st Generator – **“To”** finishing frequency of the sweep.
3. 1-st Generator – **Volume slider** to adjust output voltage from 0-100%.
4. 1-st Generator – **“Start”** press this button to commence the sweep.
5. 1-st Generator – **“Stop”** press this button to terminate the sweep.
6. 1-st Generator – **“Do Not Sweep”** By selecting this option, the F1 Generator will remain on “From” frequency. This way, you can have continuous tone coming out of your sound card.
7. 1-st Generator – “**Sweep**” select this option to make the generator sweepable.
8. 2’nd Generator – **“Free Run”** By selecting this option, F2 Generator runs on the selected F2 = frequency.
9. 2’nd Generator – Lock F2 = **F1 + XXXX Hz** By selecting this option, you will lock F2 Generator to frequency F1, so the total output looks as follows F2 = F1+XXX and F1.

10. 2'nd Generator – Lock F2 = **F1 * X.XX** By selecting this option, you will lock F2 Generator to frequency F1, so the total output looks as follows F2 = F1*X.XX and F1.
11. 2-nd Generator – **Volume slider** to adjust output voltage from 0-100%.
12. 3-rd Generator – **"Free Run"** By selecting this option, F3 Generator runs on the selected F3 = frequency.
13. 3'rd Generator – Lock F3 = **F1 + XXXX Hz** By selecting this option, you will lock F3 Generator to frequency F1, so the total output looks as follows F3 = F1+XXX and F1.
14. 3'rd Generator – Lock F3 = **F1 * X.XX** By selecting this option, you will lock F3 Generator to frequency F1, so the total output looks as follows F3 = F1*X.XX and F1.
15. 3-rd Generator – **Volume slider** to adjust output voltage from 0-100%.
16. **FM Modulated by Gen 2** –This option enables Gen 3 to be frequency-modulated by Gen 2. Free Run F2 is limited to 20Hz, and deviation, adjusted by Volume slider in Gen 2 islimited to 50%.
17. **Master Volume** – Adjust total output from the generators.
18. **"Show Harmonics"** – Select this option to display plots of the harmonics in THD sweep.
19. **"IMD 2nd + 3rd + 5th Order"** – Select this option to perform IMD tests.
20. **"Update Spectrum"** – When checked, the screen will retain harmonics plot when generators are stopped.
21. **"Normalize at 0dB"** – The level of harmonics is normalized to 0dB for easier readout.
22. **"Sampling Rate"** – You can select sampling rate for your sound card from this list box.
23. **"THD Smoothing"** – Please select smoothing level of the THS plot from this list box. This function can be performed while the THD sweep is running.
24. **"Scale"** – Press this button to change vertical scale settings.
25. **"Clear Plots"** – This button clears plots when the sweep has finished.
26. **"Done"** – Closes the dialogue box.
27. **"Li Ri"** – Input signal level indicators
28. **"Lo Ro"** – Output level from signal generators.
29. **If you press right mouse button above the Frequency Domain plotting window, you will invoke Vertical Scales control box**. This function is active even during the sweep.

Steady-State THD

In the simplest form, the Total Harmonic Distortion Steady-State test can be run as follows (Example below relates to 48kHz sampling soundcard and screen setting from 5Hz to 20kHz).

1. Set the sweep generator from 1000Hz To 2000Hz.
2. Set the required volume for Gen1.
3. Set volume for Gen2 and Gen3 =0 (no output).
4. Check the "**Do Not Sweep**" box to disable sweeping test.
5. Check **"Show Harmonics**"
6. Uncheck "**IMD 2nd + 3rd + 5th Order**". <u>This will set the system to THD test</u>.
7. Finally press "**Start**" button to activate the test. The soundcard will output a steady sine wave until you press "**Stop**" button.

During the test, you can right-click the mouse button above the frequency domain plotting screen, to adjust DB scales and THD scales.

IMD Sweep

As mentioned above, IMD can be used over the entire frequency range of the system, Take for instance, a situation, where your soundcard is set to 48kHz sampling frequency. Then the maximum audio frequency captured by the soundcard will be less than 24kHz. Therefore, the second harmonic of the measured signal must be less than 24kHz. This implies fundamental of less than 12kHz. Situation is even worse for 3'rd harmonics. It must be less than 24kHz, so the fundamental must be less than 8kHz.

In order to overcome this difficulty, you can use IMD test. Example below relates to 48kHz sampling soundcard and screen setting from 5Hz to 20kHz.

1. Set the sweep generator from 3000Hz to 15kHz.
2. Set the required volume for Gen1 and Gen2, they should be equal.
3. Set volume for Gen3 =0 (no output).
4. Uncheck the “**Do Not Sweep**” box to enable sweeping test.
5. Check the “**Lock F2= 1.12**” check box to lock Gen2 to Gen1.
6. Set Gen2 to 1.25. This will be the “frequency ratio” between Gen1 and Gen2.
7. Check “**Show Harmonics”**
8. Check “**IMD 2nd + 3rd + 5th Order**”. <u>This will set the system to IMD test</u>.
9. Finally press “Start” button to activate the test.
10.

When finished, you should see something like the picture below.

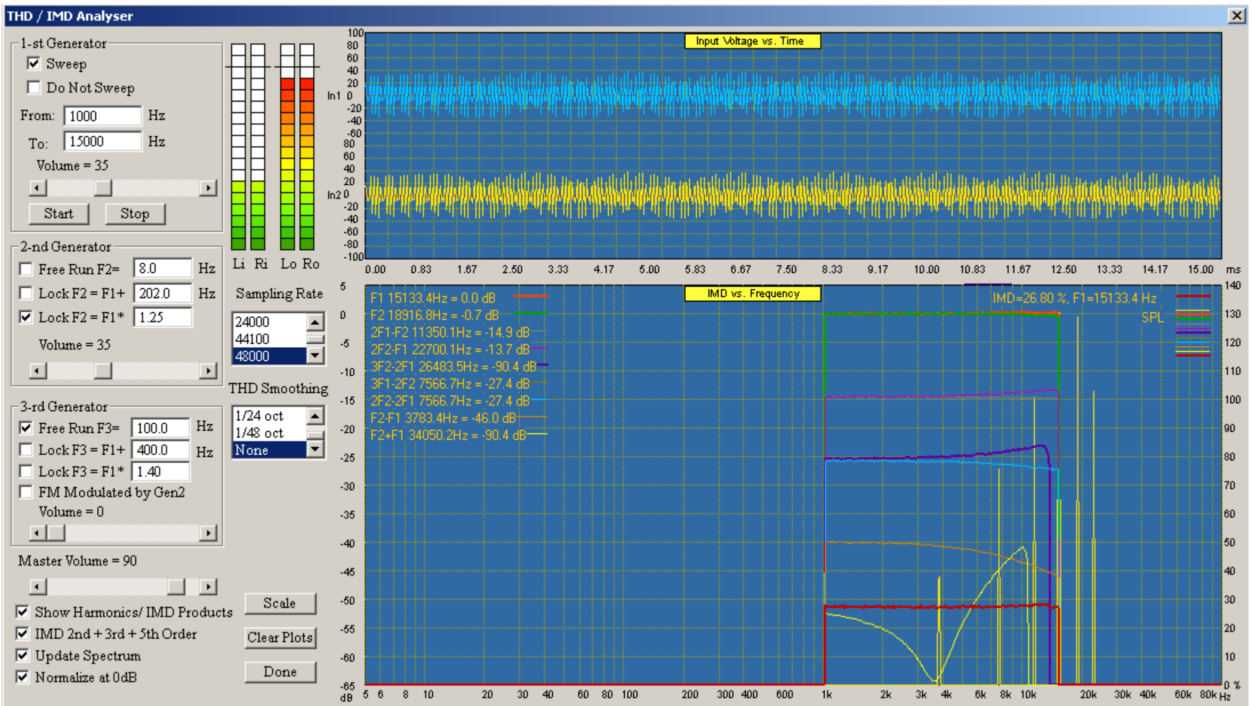


Figure 16.118. Example of IMD sweep

THD Sweep

In the simplest form, the Total Harmonic Distortion sweep test can be run as follows (Example below relates to 48kHz sampling soundcard and screen setting from 5Hz to 20kHz).

1. Set the sweep generator from 10Hz to 15kHz.
2. Set the required volume for Gen1.
3. Set volume for Gen2 and Gen3 =0 (no output).
4. Uncheck the “**Do Not Sweep**” box to enable sweeping test.
5. Check “**Show Harmonics**”
6. Uncheck “**IM 3rd + 5th Order**”. This will set the system to THD test.
7. Finally press “**Star**t” button to activate the test.

During the test, you can change “**THD Smoothing**” and right-click the mouse button above the frequency domain plotting screen, to adjust DB scales and THD scales.

Cepstral Deconvolution

Introduction

V15 of the software release introduced a new tool – Cepstral Analyser. You may be familiar with other names, that this process comes by, such as: Cepstral Deconvolution, or Homomorphic Filtering.

Basic functionality provided by this tool is a removal of room reflections, contaminating SPL obtained by MLS measurements. The tool is very powerful, up to the point of replacing much of the measured SPL with analytical curves. But this is not really recommended. Clearly, the goal of this application is a balancing act - retain as much information in the measured SPL curve, and remove as many reflections as possible. It will be fairly easy and tempting to get rid of all wiggles on the SPL curve, but once again – we recommend you apply common sense.

The tool is not a panacea for fixing every bad SPL measurement. And there will be loudspeaker measurements, that should not be corrected, due too much echo contamination. Still, the tool will allow you to proceed and clean the SPL curve, but in cases like this – you will be guessing, rather then being guided by the information contained in the original cepstrum curve.

In addition, before you start using this tool, please be aware, that close-mike measurement technique, coupled with diffraction analysis, will yield equally good, if not better results, simply because the reflections are attenuated by the virtue of the measurement technique. However, the close-mike technique is also complex, and requires utmost attention at every stage of summations and diffraction modelling. Therefore, once you become confident with Cepstral Deconvolution, you may try to use it for all quick, first-cut activities, and move to close-mike measurements for the final results.

Cepstral Deconvolution Algorithm

Now, a new vocabulary: the terms have been derived by paraphrasing the existing similar words.

Derived Terms	Original Terms
Cepstrum	Spectrum
Quefrency	Frequency
Rahmonics	Harmonics
Gamnitude	Magnitude
Saphe	Phase
Lifter	Filter
Short-pass Lifter	Low-pass Filter
Long-pass Lifter	High-pass Filter

Recommended Reading:

1. Cepstral Techniques for Transducer Measurement (AES Preprint 2172)
Paul D. Bauman, Stanley P. Lipshitz, Tony C. Scott, and John Vanderkooy, University of Waterloo, Waterloo, Ontario Canada

2. Cepstral Techniques for Transducer Measurement : Part II (AES Preprint 2302)
Paul D. Bauman, Stanley P. Llpshitz, and John Vanderkooy Audio Research Group, University of Waterloo, Waterloo, Ontario N2L 3Gl, Canada.

Cepstral Analysis, as applicable to **removing room reflections in loudspeaker measurements** (de-convolution), can be described as follows. Impulse response, as collected by MLS system consists of direct sound convolved with room reflections. Taking FFT of such signal, we accomplish two things: (1) we obtain system's frequency response, and (2) we map convolution in time domain into multiplication in frequency domain. Now, taking natural logarithm, maps the multiplication into addition in cepstral domain.

This enables simpler editing of the cepstrum by subtraction, effectively providing the desired de-convolution, so in the next step, we take IFFT to obtain cepstrum of the signal. Subsequently, we perform cepstral editing. The remaining steps need to be taken to “undo” the effects of IFFT and natural logarithm – thus we take FFT and finally the antilog of the cepstrum.

Vanderkooy (AES Preprint 2172) explains, “…cepstrum of a smoothly varying signal is concentrated about the quefrency origin and consequently, reflection effects are displaced to higher quefrencies and tend to be distinct from the cepstrum of the direct signal……. Reflection effects will be moved to higher quefrencies in the minimum-phase cepstrum but remain superimposed on the tail of the minimum-phase cepstrum as it decays to zero. If the reflection effects are minimum-phase, however, there will be no contribution until the first arrival and it is predicted that editing can be accomplished by smoothing the tail beyond the first-arrival quefrency.”

The complete process implemented for the purpose of Cepstral Analysis of a device under test can be summarized as follows:

1. Obtain Impulse Response, g(t) of a device using MLS techniques.
2. Obtain Transfer Function (frequency response, or SPL) from g(t).

   $G(w) = FFT\{ g(t) \}$
   $G(w) = Re\{ G(w) \} + jIm\{ G(w) \}$

   At this point, it is useful to perform Hilbert-Bode Transform on G(w) to obtain minimum-phase response and good quality phase response with minimum number of discontinuities.
3. Calculate natural logarithm of G(w) -> H(w).

   $H(w) = 1/2 \ln(Re(G(w))^2 + Im(G(w))^2) + j*atan(Im(G(w))/Re(G(w)))*n*PI$
   $A(w) = ½ \ln(Re(G(w))^2 + Im(G(w))^2)$
   $B(w) = atan(Im(G(w))/Re(G(w)))$
   $n = 1,2,3,4……$
4. Unwrap phase B(w).

   Typically, phase response would have only a small number (2-4 only) of transitions –PI/+PI. This is due to HBT performed earlier. Therefore, unwrapping phase can be simplified to detecting 180 deg discontinuities and adding 2PI after each transition.
5. Adjust Phase B(w)

   The unwrapped phase will typically result in a large phase discontinuity at the right-hand side of the spectrum. As a consequence of this (DFT assume periodicity) the cepstrum will exhibit a very “noisy” character. To counteract this artifacts, a provision is made for “tilting” the phase so as to bring the unwrapped phase back to zero at the right-hand end of the frequency band. After the cepstrum operation, the phase can be “un-tilted” back to it’s original shape.
6. Calculate Cepstrum C(t)

   $C(t) = IFFT\{ H(w) \} = IFFT\{ A(w) + j B(w) \}$
7. Filter the Cepstrum using “Lifters” with transfer function K(t).

   $C_{filtered}(t) = C(t) K(t)$

   In the current version of the program, this operation is **performed graphically using built-in function**. Vanderkooy (AES Preprint 2172) gives the following suggestions here: “The basic shape of the cepstral tail must be maintained during the course of editing, and simple nulling of data points would be quite unacceptable. Much of the low-frequency roll-off information which occurs in a localized frequency region is contained in the tail portion, while higher frequency information, which is usually smoother due to system resonances or reflections with moderate effective Q values, is contained closer to the quefrency origin. These properties suggest that smoothing the minimum-phase cepstrum will give acceptable room de-convolution without adversely affecting the basic system.”
8. Calculate Inverse Cepstrum. This will provide natural logarithm of magnitude and phase.

   $G_{filtered}(w) = FFT\{ C_{filtered}(t)) \}$
9. “Un-tilt” and “Un-adjust” phase response.

10. Calculate filtered Transfer Function (SPL) in frequency domain using anti-log function (* denotes multiplication)

$$\text{Mag} = \text{Re}(\ G_{\text{filtered}}(w)\ ), \qquad \text{Ph} = \text{Im}(\ G_{\text{filtered}}(w)\ )$$
$$\text{Re}\{\ Go(w)\ \} = \exp(\ \text{Mag}\ ) * \cos(\ \text{Ph}\ ), \quad \text{Im}\{\ Go(w)\ \} = \exp(\ \text{Mag}\ ) * \sin(\ \text{Ph}\ )$$
$$\text{SPL} = 10\log_{10}(\ \text{Re}(Go(w))^2 + \text{Im}(Go(w))^2)$$

Example of Cepstral Analysis

Now, please **carefully** review fully illustrated example of Cepstral Analysis, performed on a loudspeaker system. Cepstral Analysis is a very powerful, but rather complicated process. Good final results can be accomplished, if attention is paid to every step of the process, so that the results of each intermediate step are fully optimized. Vanderkooy advocates a technique called **minimum-phase edit of the complex cepstrum.** In this process, the complex cepstrum is calculated first, and is subsequently decomposed into minimum-phase cepstrum and all-pass cepstrum. Vanderkooy's (AES Preprint 2302) justification goes like this "…The decomposition of the complex cepstrum into **minimum-phase** and **all-pass cepstra** is particularly useful for editing minimum-phase interference effects. Reinserting the all-pass cepstrum after editing the minimum-phase cepstrum achieves, in essence a **minimum-phase edit of the complex cepstrum**. Oftentimes reflections are more visible in the minimum-phase cepstrum……than in the complex cepstrum where they can be obscured by the all-pass component. Minimum-phase editing of the complex cepstrum is also better in terms of phase integrity since the all-pass component is not erroneously edited….".It should be possible to determine the phase nature of the reflections (minimum-phase OR all-pass) and then use minimum-phase editing, remove room reflections to obtain anechoic frequency response. If room reflections are minimum-phase, they will show only on the minimum-phase cepstrum, and the all-pass cepstrum will be smooth and near zero.

<u>Start with MLS TAB</u>

1. Generate Impulse response

Typically, you would measure the loudspeaker in-room, collecting it's MLS response, so that the Impulse response can be automatically displayed. There is nothing new here – just your standard MLS process, for which you would use the MLS TAB. **IMPORTANT – please set sampling rate to 48000.**

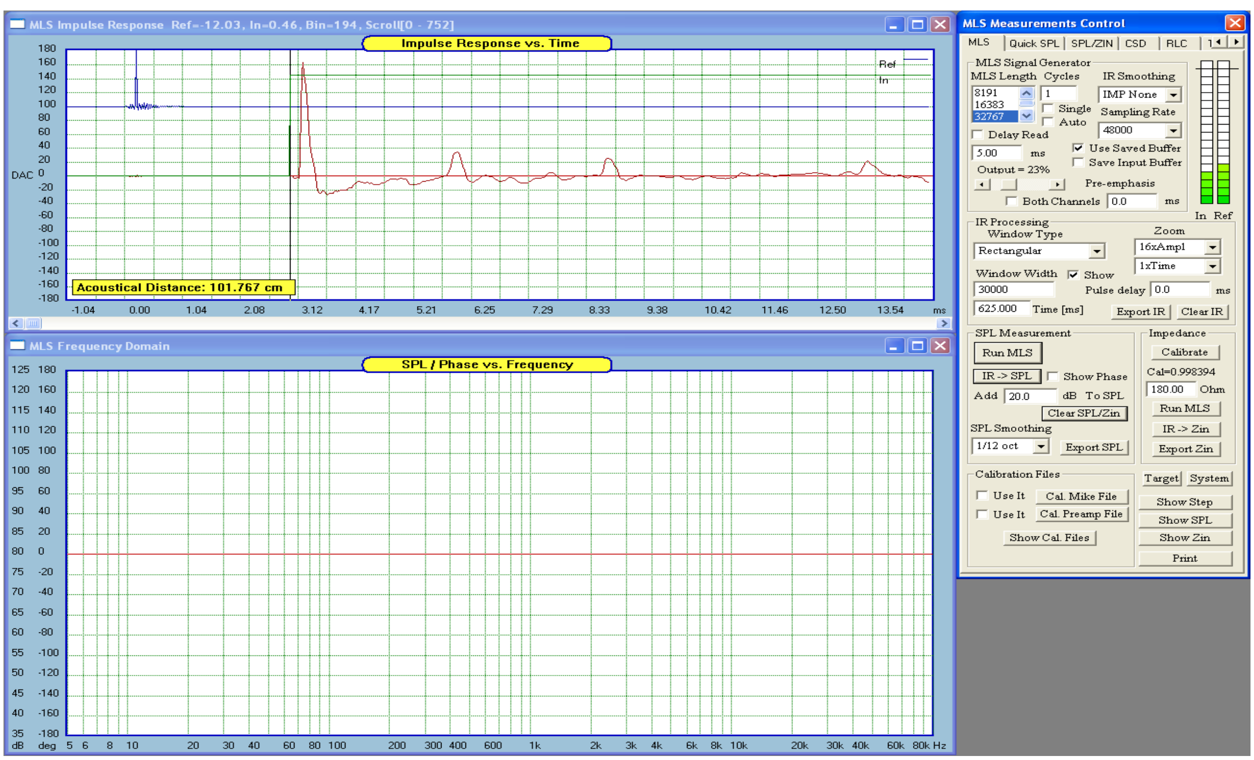


Figure 16.119 Impulse response of an example midrange driver.

2. Generate SPL curve. Simply press “IR->SPL” button to obtain SPL curve for your speaker.

Figure 16.120 SPL of the example midrange driver

3. HBT your SPL curve

Perform HBT on your SPL curve to obtain minimum-phase response and good quality phase response with minimum number of **2π discontinuities**. Good example is shown below.

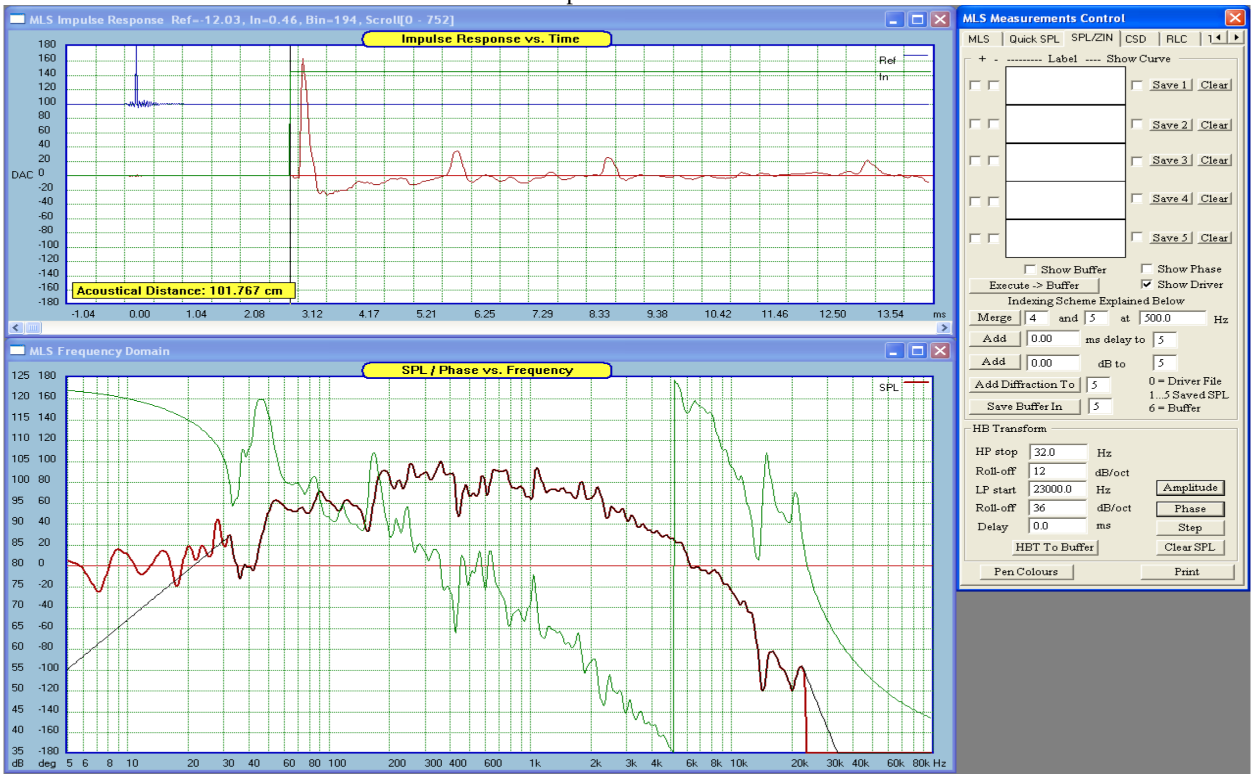


Figure 16.121 HBT (amplitude + phase) of the example midrange driver

For the remaining steps, please move to “CEPSTRUM” TAB

4. Show SPL+Phase

In the first step, you would simply confirm, that the correct SPL/Phase data has been generated for the cepstral analysis. Please press the “Show SPL + Phase” button.

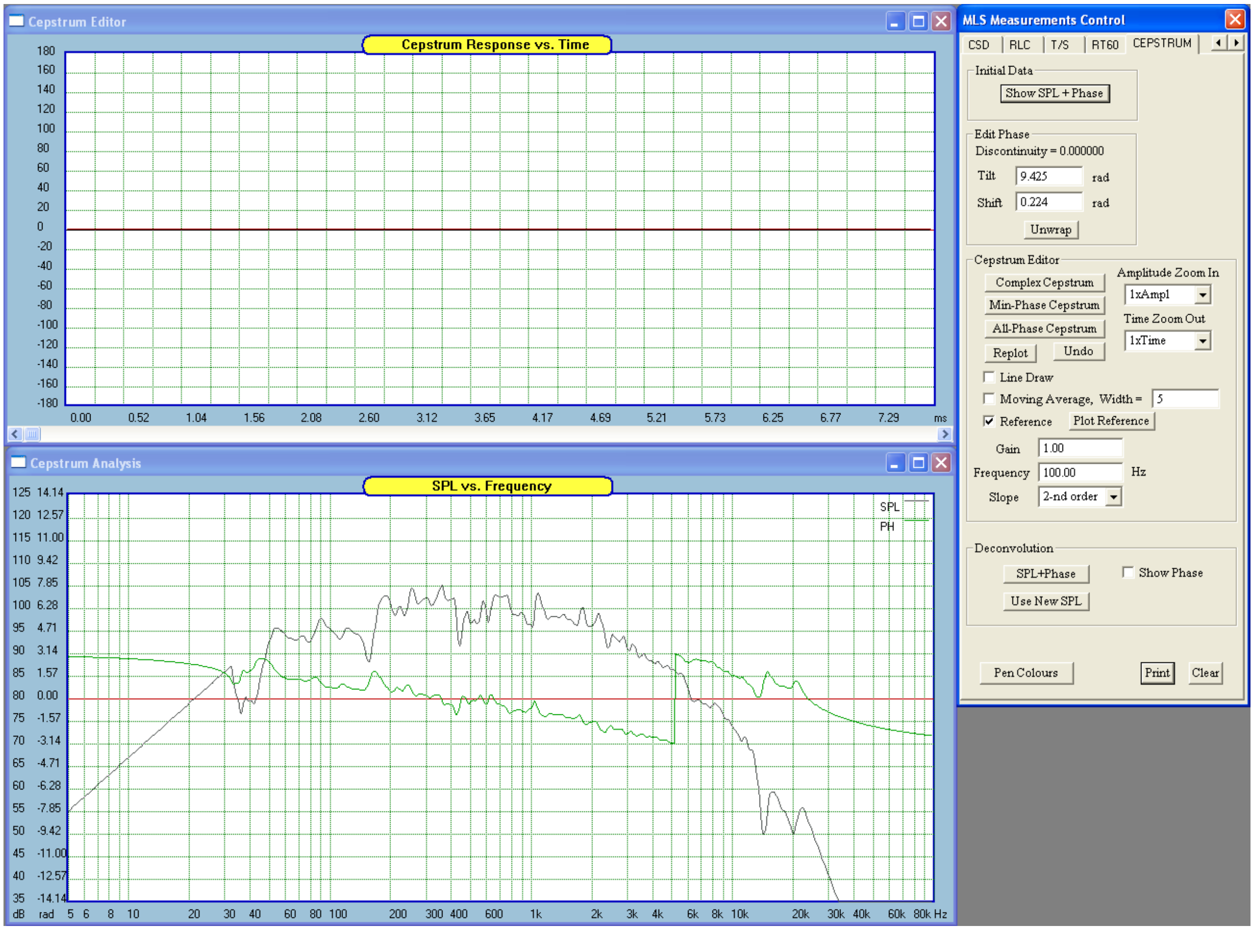


Figure 16.122 Show SPL and phase response

5. Un-wrap phase

If you decide to edit Minimum-Phase Cepstrum, please be aware of the following. Minimum-phase systems, such as loudspeakers, exhibit strict amplitude-phase relationship that is captured by HBT. In other words, phase response corresponds to amplitude response via well known Bode amplitude-phase relationships. When the phase is unwrapped, it becomes “absolute” phase response, so that Bode relationship manifests itself clearly.

For instance, when the asymptotic amplitude response (starting from the low-end of the frequency range) is +24dB/oct, then corresponding phase response must be $+(24/6)*\pi/2 = 4*1.571 = 6.284$rad, **when asymptotically approaching 0Hz**. As a consequence, one must assure, that the phase response is not only correctly unwrapped, but also starts on the left-hand-side from the correct value. The unwrapping algorithm calculates the proposed, starting left-hand phase value from the asymptotic left-hand slope of the HBT, and inserts into “Shift” data field. However, screen frequency response is limited, and typically starts at 5-10Hz, therefore, the calculated shift value is an approximation of the correct value in most instances. Similar problem will exist on the high-end of the frequency range. We can calculate asymptotic phase value for frequencies approaching infinity, but this will not equal the phase value at the screen high-end boundary. Given the above, the “Shift” and “Tilt” phase parameters are only close approximations of what needs to be inserted there.

To unwrap phase, please press the **“Unwrap”** button now.

Complex Cepstrum contains all-pass component, so that Hilbert-Bode relationship is not strictly applicable here. Therefore, shifting the left-hand origin of the phase response is not required.

Figure 16.123 “Unwrapped” and adjusted (shift = -1.5) phase response.

6. Calculate Minimum-Phase Cepstrum – press “Minimum-Phase Cepstrum” button

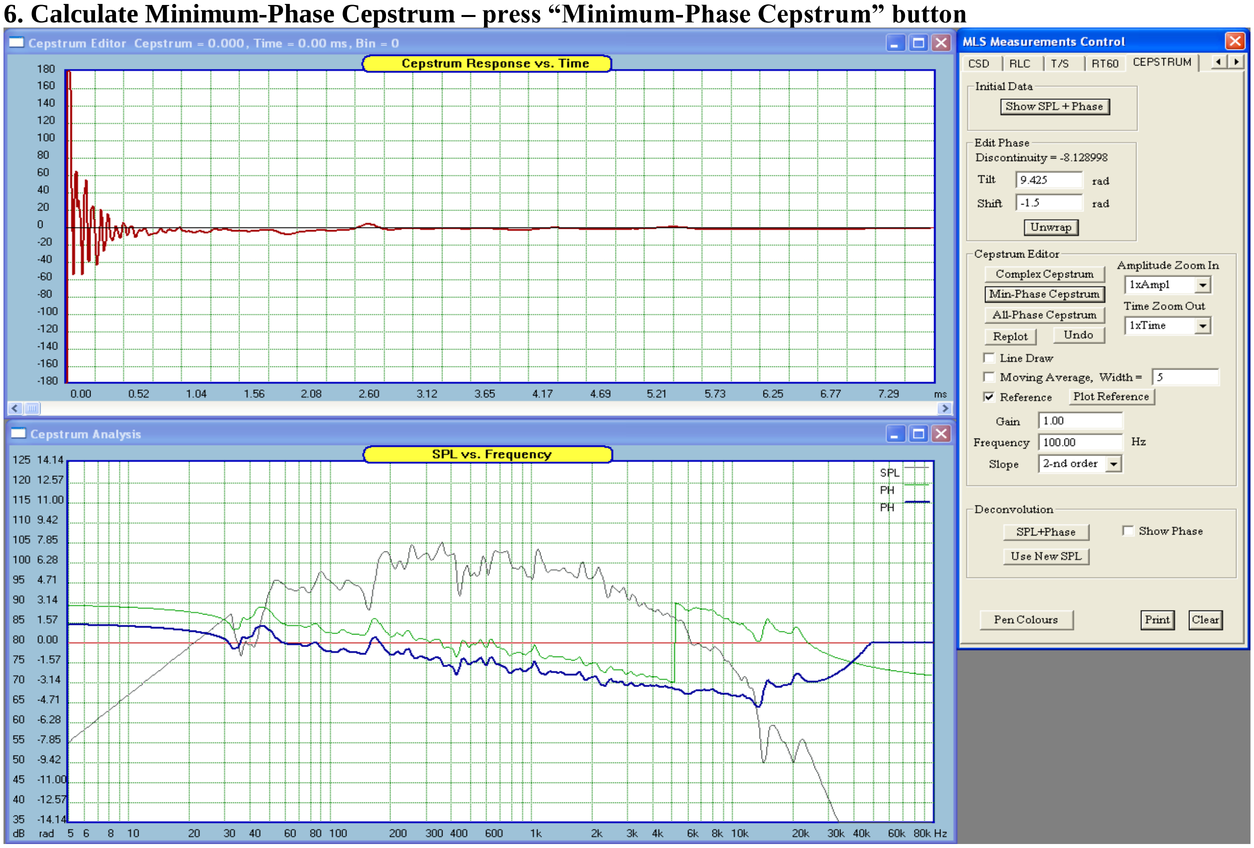


Figure 16.124 Calculated Minimum-Phase Cepstrum – red curve

1. Edit Minimum-Phase Cepstrum

Minimum-phase cepstrum editing is facilitated by taking into account two important properties described by Vanderkooy:

A. "Cepstrum of a minimum-phase echo is zero until the time of the first reflection arrival."

This offers us an immediate clue to the editing process. The time of the first arrival cam be estimated from the measurement setup. This is typically accomplished by measuring distances from reflecting surfaces to the transducers and then calculating arrival times of the reflected signals, thus making it possible pin-point where to look for signs of those reflections on the cepstrum's time scale. If the reflection is present, there should be a significant ripple on the cepstrum curve.

As a consequence of the above, the cepstrum should not be edited before the first reflection arrival.

B. **"Cepstrum of a smoothly varying signal is concentrated about the quefrency origin and consequently, reflection effects are displaced to higher quefrencies and tend to be distinct from the cepstrum of the direct signal. "**

Good quality, well designed loudspeaker system would exhibit rather smooth anechoic frequency response, consequently, we would expect that it's minimum-phase cepstrum to be concentrated around the origin. However, we would still expect, that the low-frequency tail of the cepstrum will extend into higher quefrency region. If the reflections are present, this tail would be contaminated by excessive ripples. The task at hand, would be to smooth these ripples without changing the basic shape of the low-frequency tail.

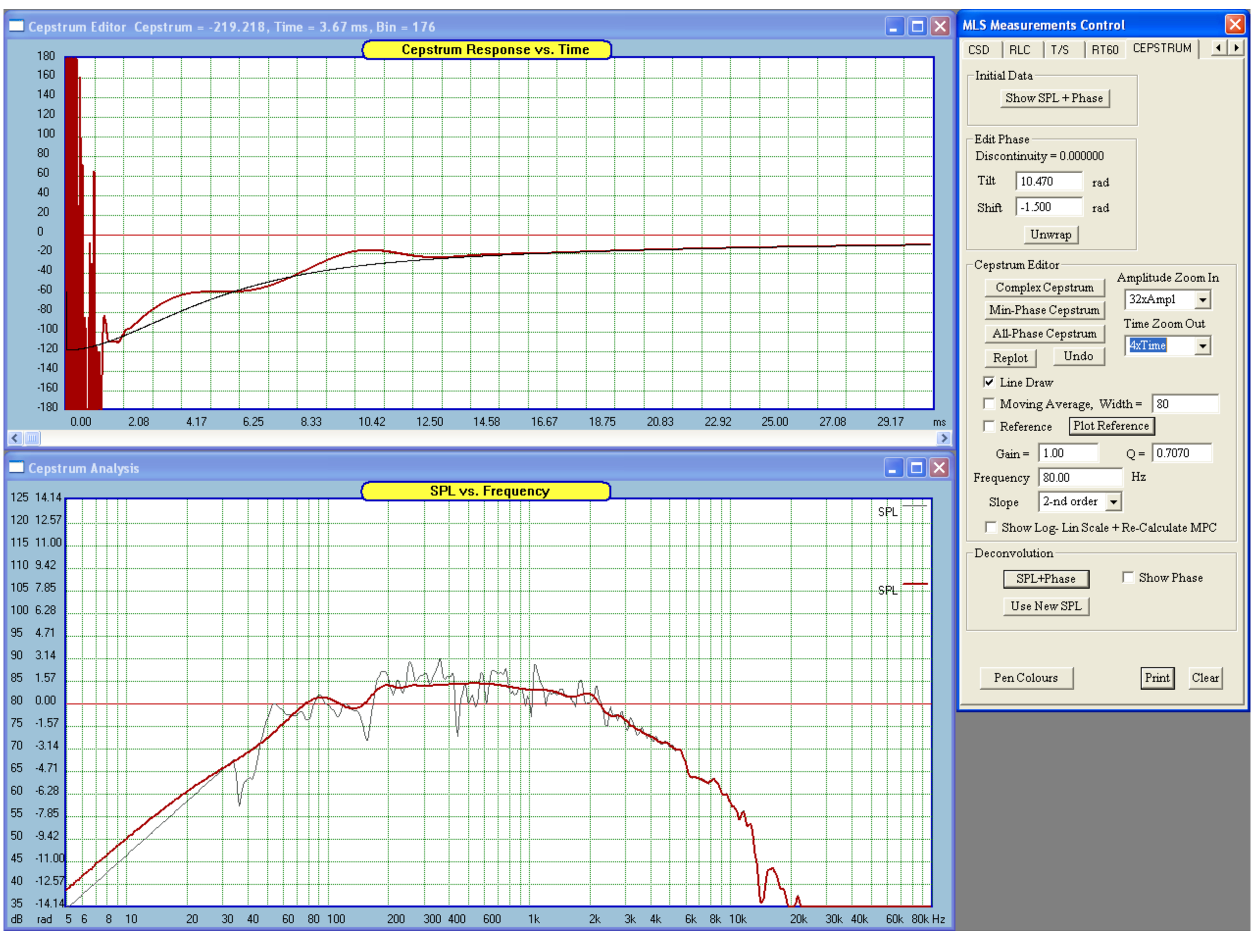


Figure 16.125. Edited cepstrum curve – using analytical curve + smoothing with parameters as shown

Cepstrum editor provided with the program is quite simple. You can use three options: (1) smotth the curve, (2) draw straight-lines on the screen, and (3) replace the tail-end of the cepstrum curve with analytical curve. It is estimated, that a combination of all three options may give satisfactory editing results. Please note, that the cepstrum can be scrolled using provided screen sliders.

A. Moving Average

To use this option, please first enter the width of the window used to calculate the average. Reasonable values are 2-20. Wider windows will obviously calculate the average over larger number of points.

1. Select the width of the ***window***, say 2-10 bins. 2 bins will give less smoothing then 10. It is not recommend to use wider windows than 50.
2. Check the "**Moving Average**" box.
3. Place the mouse pointer on the cepstrum plot, and press the Left Mouse Button. The cepstrum curve will be smoothed automatically all the way to the right-end, starting **from the current mouse pointer location**.

To be exact, the smoothing is actually performed over 2 x ***window***. This is because Symmetrical Moving Average filtering scheme has been adopted. The algorithm makes a spare copy of the whole cepstrum curve, and then calculates each smoothed curve value from (–***window***) to (+***window***) around current mouse pointer, always using the spare copy values. Then moves to the next cepstrum bin to the right and the process is repeated over and over again. This seems to work better than one-sided version of the same algorithm. If you select wide window (say 50) and place the mouse cursor too close to the origin (<50), the algorithm will not activate, because (–window) index would attempt to pick negative-index values from the spare copy curve.

B. Drawing Straight Lines

This option allows you to quickly edit large portions of the cepstrum curve. To activate this option, check the **"Line Draw"** box, and simply position mouse cursor above the plotting section of the cepstrum screen and press the left-mouse-key. When you move the mouse pointer, a straight-line curve will be drawn while you keep pressing the button and you move the mouse pointer. The screen will be re-drawn when you release the left-mouse-key, and the cepstrum curve will be replaced by the recently drawn straigt line.

C. Reference

To use this option, please select the closest combination of the slope and cut-off frequency for an imaginary filter that best approximates the low-end of the driver's SPL. To activate this option, check the **"Referene"** box , and simply position mouse cursor above the plotting section of the cepstrum screen and press the left-mouse-key. The cepstrum curve will be replaced by an analytical curve plotted accordingly to the parameters you selected.

If a mistake is made, you can press the "Undo" button, and this will return the cepstrum to the state just before the last change.

It is rather difficult to provide a step-by-step procedure to clean-up the SPL curve. Much of the technique used will depend on each particular driver – more specifically, how much guiding information is available from SPL curve, cepstrum curve, and knowledge about what the SPL should actually look like.

1. The bluntest option is to replace the cepstrum curve with analytical curve right before the first reflection manifests itself on the cepstrum curve. This should clean-up the SPL curve very well, perhaps too well.

2. Second strategy would be to replace only the tail-end of the cepstrum curve with analytical curve. This would aim only to clean-up the bottom-end of the SPL curve. Next, one would replace the most contaminated sections of the cepstrum curve with segments of straight lines. Finally, use the Moving Average option to progressively smooth the cepstrum curve closer and closer to the origin, but not closer then the first reflection.

3. Another strategy would involve Moving Average option with narrow window – perhaps 5 bins wide – and run the first-pass clean-up starting somewhere, perhaps from the first reflection. Next, one could increase the window to 10 – 20 bins and repeat the process.

More advanced users may attempt a following learning process. Perform a “close-mike” SPL measurement and keep it as a reference for both: SPL and cepstrum curves. Then increase microphone distance, so that reflections will start manifesting themselves on SPL curve. Then check the cepstrum curve, and try to note the difference between the original cepstrum curve and the newly contaminated one. Increase the mike distance again to see more deterioration in the cepstrum curve. Hopefully, the echos will manifest themselves more clear, so they will be easier to recognize and to edit out.

11. Plot resulting SPL

It is beneficial to keep checking the shape of the SPL curve frequently. As the cepstrum is progressively edited, you will notice corresponding changes in the frequency response. You may elect to check the SPL curve after each attempt at cepstrum smoothing. Seeing a mistake, you could cancel it immediately.

12. Keep looping (10) and (11)

It is anticipated, that you will be performing steps 10 and 11 many times. Cepstral editing is a learning process, and it may take time before you learn to see what-affects-what.

13. Save edited SPL

In the final step, once you are happy with the results of the cepstral editing and the look of the SPL curve, please save the resulting SPL into the program’s memory. **The new curve will replace the old frequency response**.

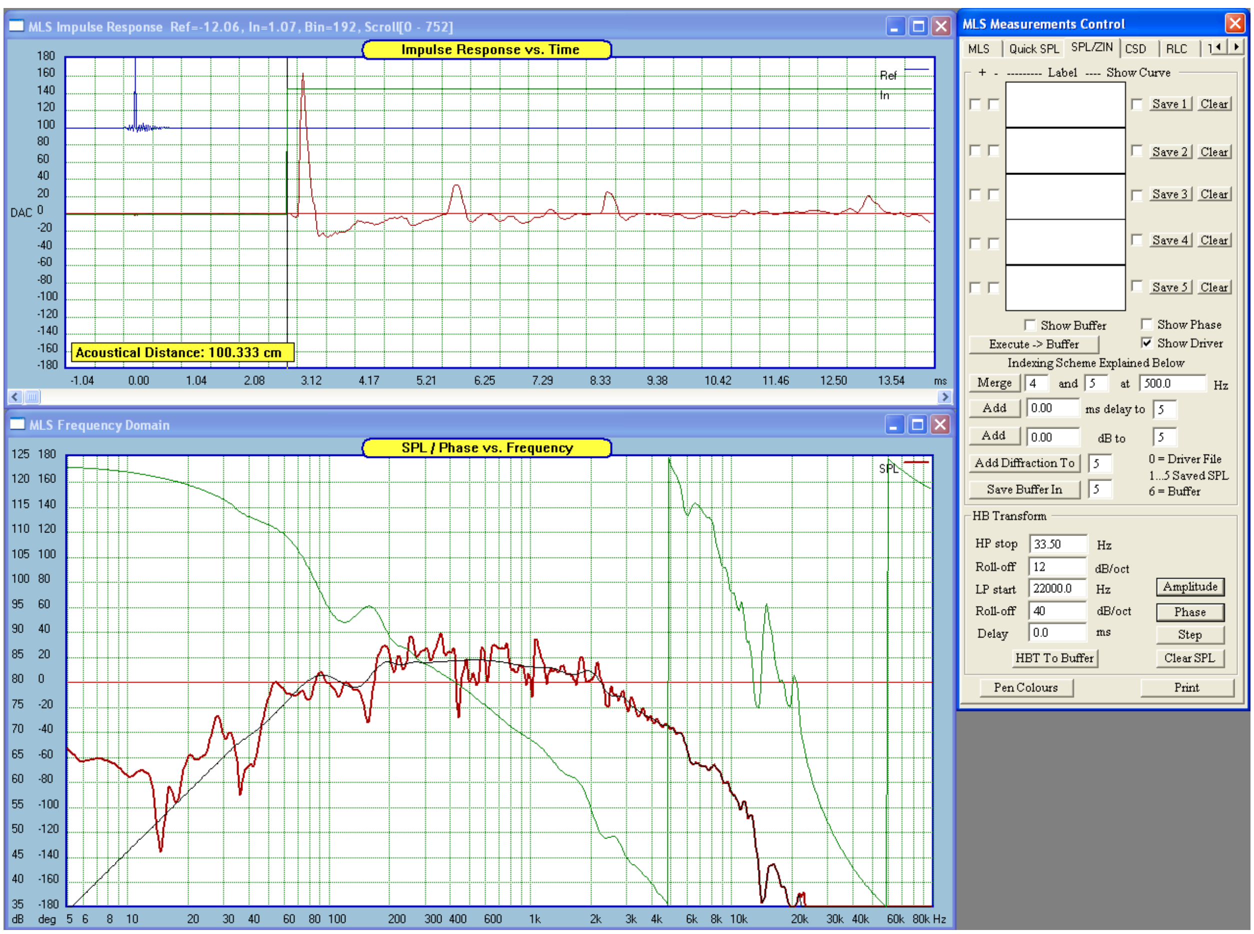


Figure 16.126. Comparison of “raw” SPL/Phase and edited SPL/Phase

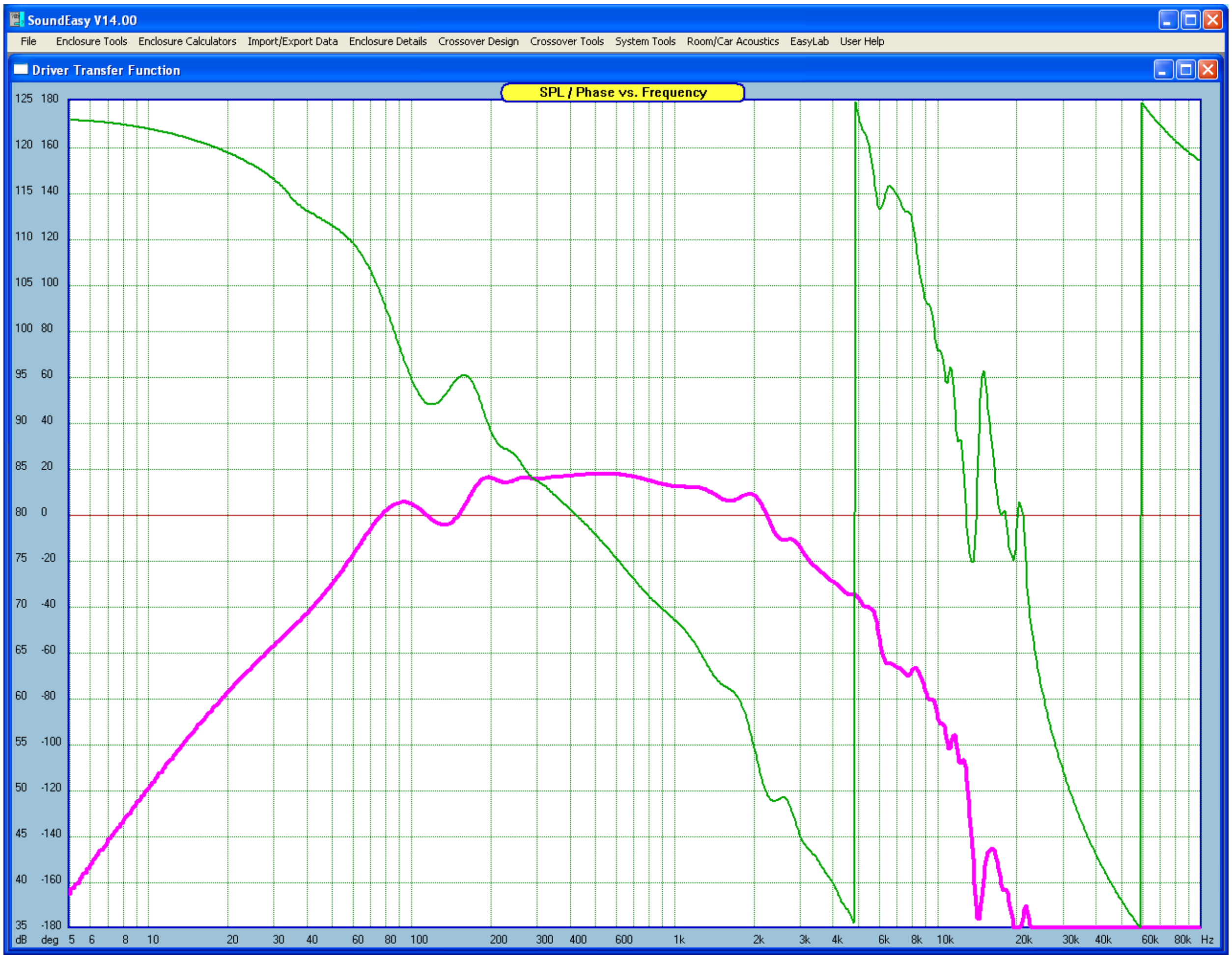


Figure 16.127. New SPL/Phase saved into he driver data space.

Learning About Cepstrum

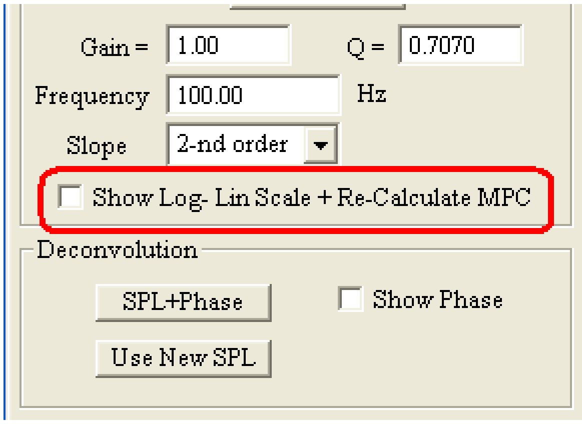


Figure 16.128. Changing scales.

The Cepstrum TAB has a checkbox, that changes cepstrum's linear time scale into logarithmic (natural logarithm). Therefore, you can display cepstrum in log-log scale, rather than log-lin scale. There are some benefits.

In traditional log-lin scale, the low-frequency wiggles on the cepstrum appear to be "smeared in time". This makes them less visible, by comparison to high-amplitude, sharp peaks closer to the origin. It makes them easier to disregard by the user. Unfortunately, those "smeared" wiggles are as important as the peaks. The log-log scale compresses the right-hand data into one screen, so you can see the whole tail on one screen. It also sharpens the "smeared" wiggles, so the whole cepstrum looks more balanced, and brings the important part of the cepstrum into the focus.

You can not edit the cepstrum, when you have this switch active. You can only look and learn from the display. The log-log time scale is displayed in (1/sampling rate) increments, so the first decade starts with large discontinuities and then gets smoother.

As far as the analytical curve is concerned, we are fully entitled to use all available knowledge about the system under test. The system was designed using TS parameters and available enclosure modelling process, so one would expect that the low-end response corresponds quite close to the predicted curves. We are am not suggesting here straight substitution, but only a guidance purpose for learning.

Using the log-log scale, coupled with guiding analytical curve, you can see the shape of the tail a lot better. This is where the HBT initial tailing will also manifest itself to some degree.

Initial HBT tailing is intended to remove all unwanted low-frequency noise, that inevitably enters the MLS (supposed to be LTI system) measurement process. The MLS system dynamic range is 45-50dB, so anything below this should be removed. And it also helps to get rid of all processing artefacts. So, if performed carefully, one could tail anything below -25 to-30dB SPL, using design intended parameters. Obviously, this tailing process will help improving the lowest end of the cepstrum curve, so one needs to be careful, not to overdo the HBT process.

So, in an attempt to learn about cepstrum shape, we can at look at a couple of available speakers.

1. Midrange driver at 1m. This speaker is in so much trouble, that even when tailing it from 100Hz down, you can still see large, asymmetrical peaks on the tail above 1.6ms, which should be almost flat.

2. A 12" guitar driver. (4^{th} order, F-3dB = 85Hz). Measurement taken with close mike, so one can see, that the cepstrum is relatively clean, and the cepstrum follows the analytical curve quite well.

3. Full-range system.

This is one way of learning about the cepstrum. It would be very beneficial to have your drivers measured with close-mike, so we could put the cepstrums side-by-side, and see the deterioration immediately. Be measuring several speakers, we can start seeing the trends a lot better.

From these cepstrums, you can see how far to the right should the cepstrum extend for a typical tweeter, midrange, woofer and a subwoofer. Given the design parameters are known, (visualized by the analytical curve) one could then learn what to look for, and how to edit the cepstrum better, and retain more of the speaker own characteristics.

From the examples below, it looks like low-frequency cepstrum is simple-shape curve, transitioning between two levels, and the echoes manifest themselves as various size wiggles on this curve, that otherwise should be smooth. Time location of the wiggles should correspond roughly to the frequency bin on the SPL curve.

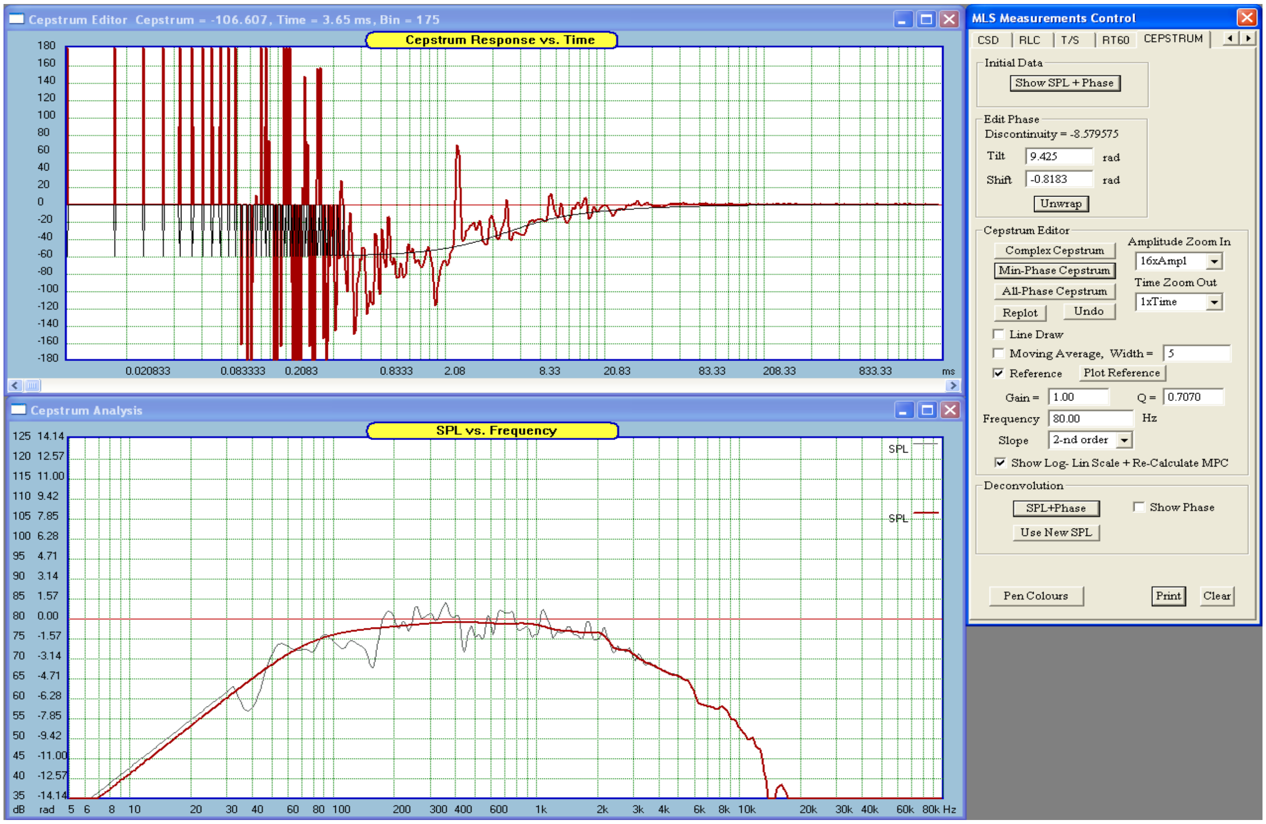


Figure 16.129. Example of a midrange driver.

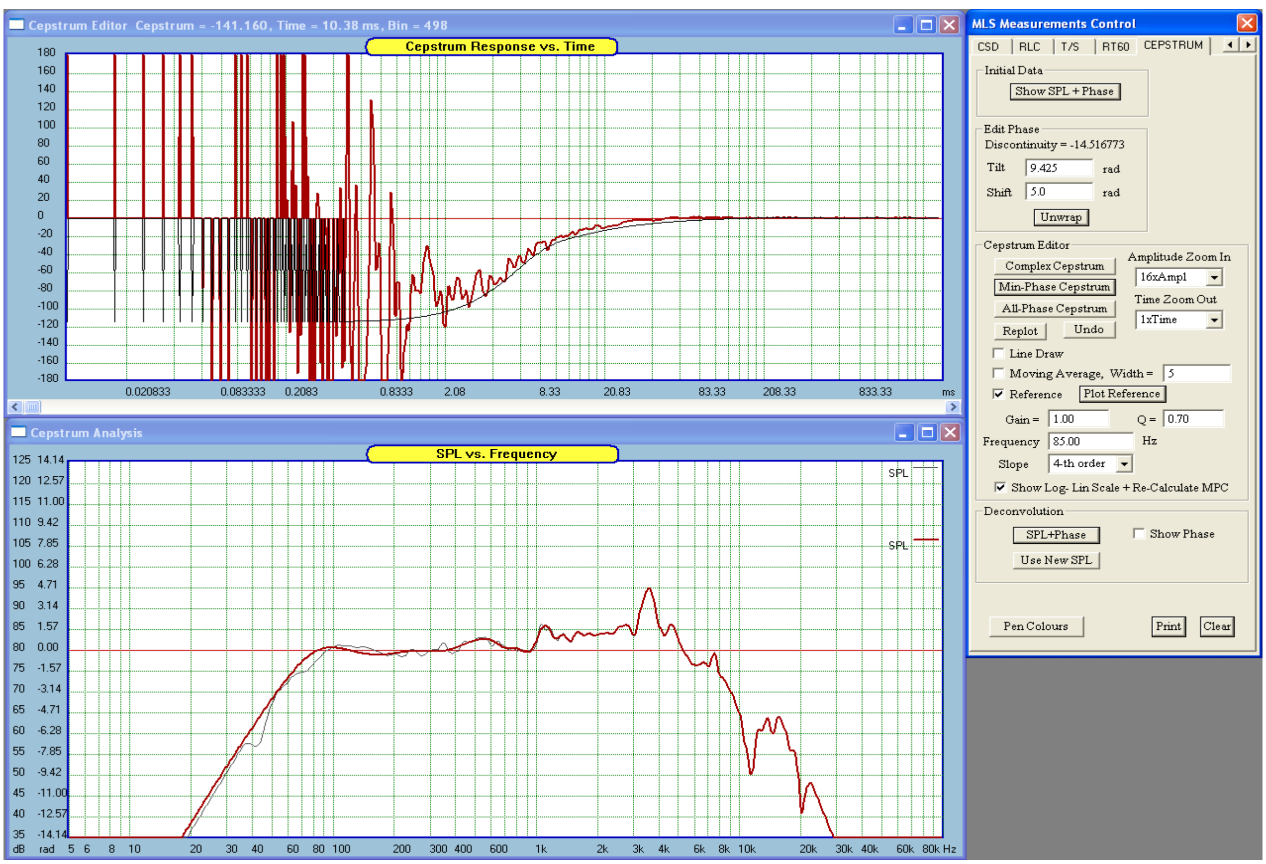


Figure 16.130. Example of a woofer driver.

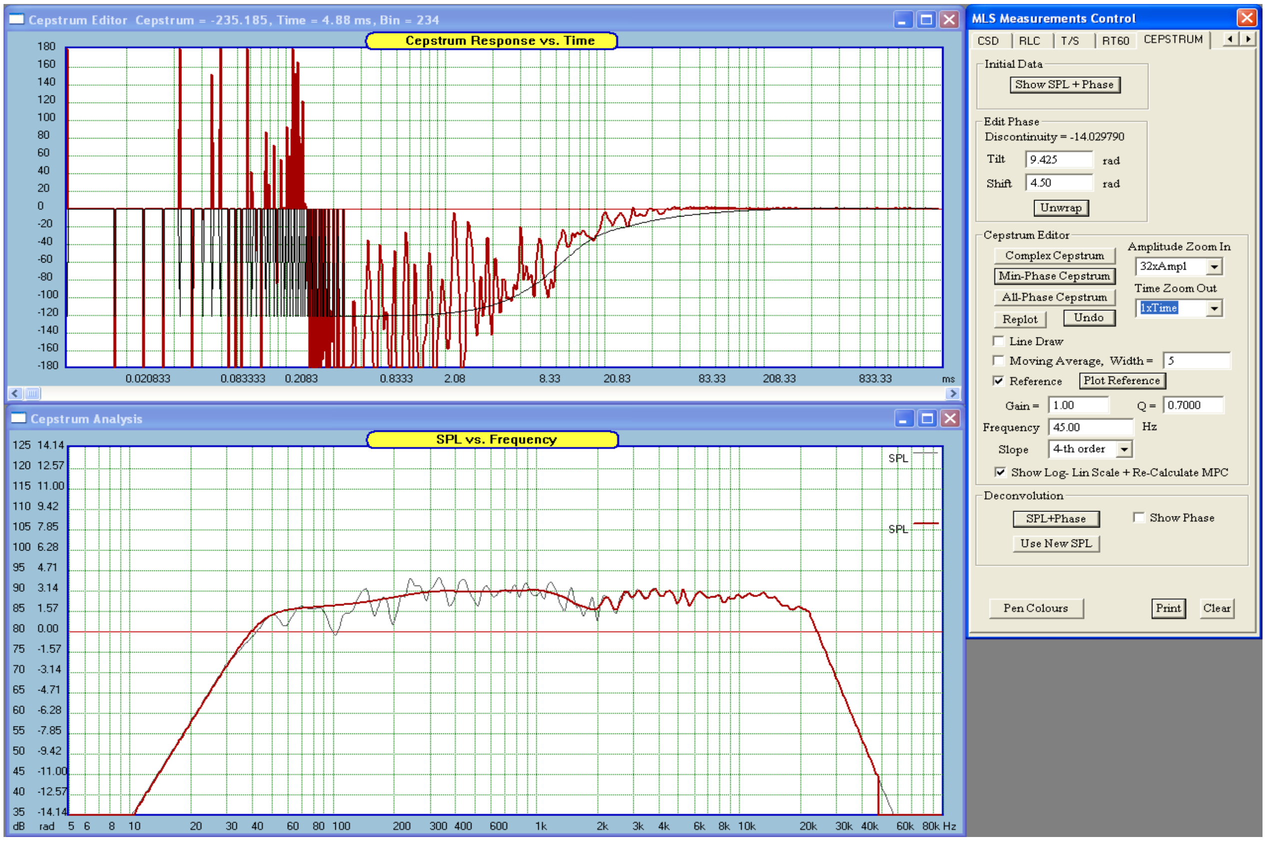


Figure 16.131. Example of a wide-band system

Example of a Woofer Driver

In the following example, it is evident, that the driver is well outside assumptions of smooth frequency response. As Vanderkooy predicted, "….cepstrum of a smoothly varying signal is concentrated about the quefrency origin…." This assumption is clearly violated here, and you will have difficulties applying Cepstral Deconvolution to this woofer.

The TS parameters of the driver + enclosure are: 2-nd order, F3dB = 55Hz, Q = 0.76

The SPL measured with **close-mike technique** is shown below. The SPL has been processed by HBT algorithm, with the parameters shown on the screen. It is immediately obvious, that the frequency response is nothing but smooth above 400Hz. We may expect, that the cepstrum of such driver will NOT be concentrated around the origin, and will extend further in time. Furthermore, we can anticipate, that the cepstral tail of this driver will be contaminated by turbulent behaviour of the SPL curve above 400Hz.

As explained before, initial HBT tailing is intended to remove all unwanted low-frequency noise, that inevitably enters the MLS (supposed to be LTI system) measurement process. The MLS system dynamic range is 45-50dB, so anything below this should be removed. And it also helps to get rid of all processing artefacts. So, if performed carefully, one could tail anything below -25 to-30dB SPL, using design intended parameters. Obviously, this tailing process will help improving the lowest end of the cepstrum curve, so one needs to be careful, not to overdo the HBT process.

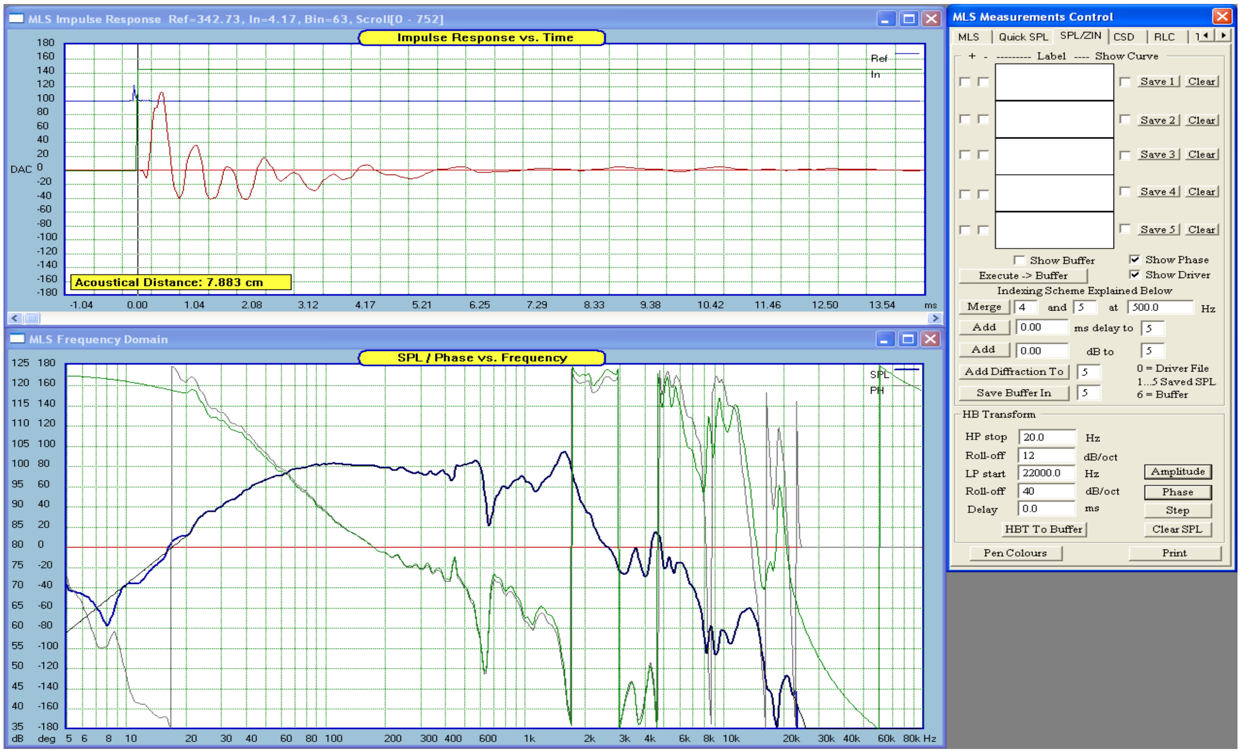


Figure 16.132. Close-mike measurements of a wooferdriver.

Indeed, once the phase was unwrapped, and the Minimum-Phase Cepstrum + analytical curve for the TS parameters was calculated, it is observable, that the cepstrum contains a number of symmetrical ripples, starting from about 2ms.

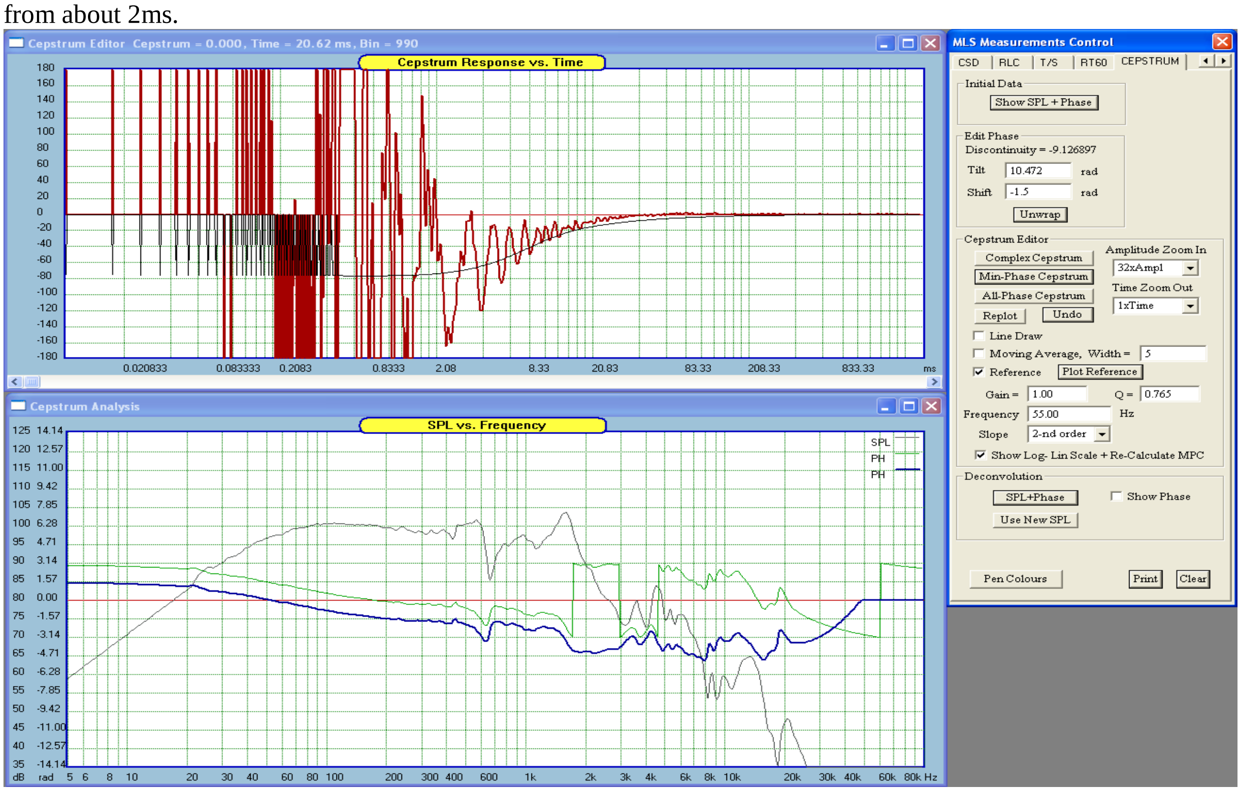


Figure 16.133 Using the log-log scale, coupled with guiding analytical curve, you can see the shape of the tail a lot better. This is where the HBT initial tailing will also manifest itself to some degree.

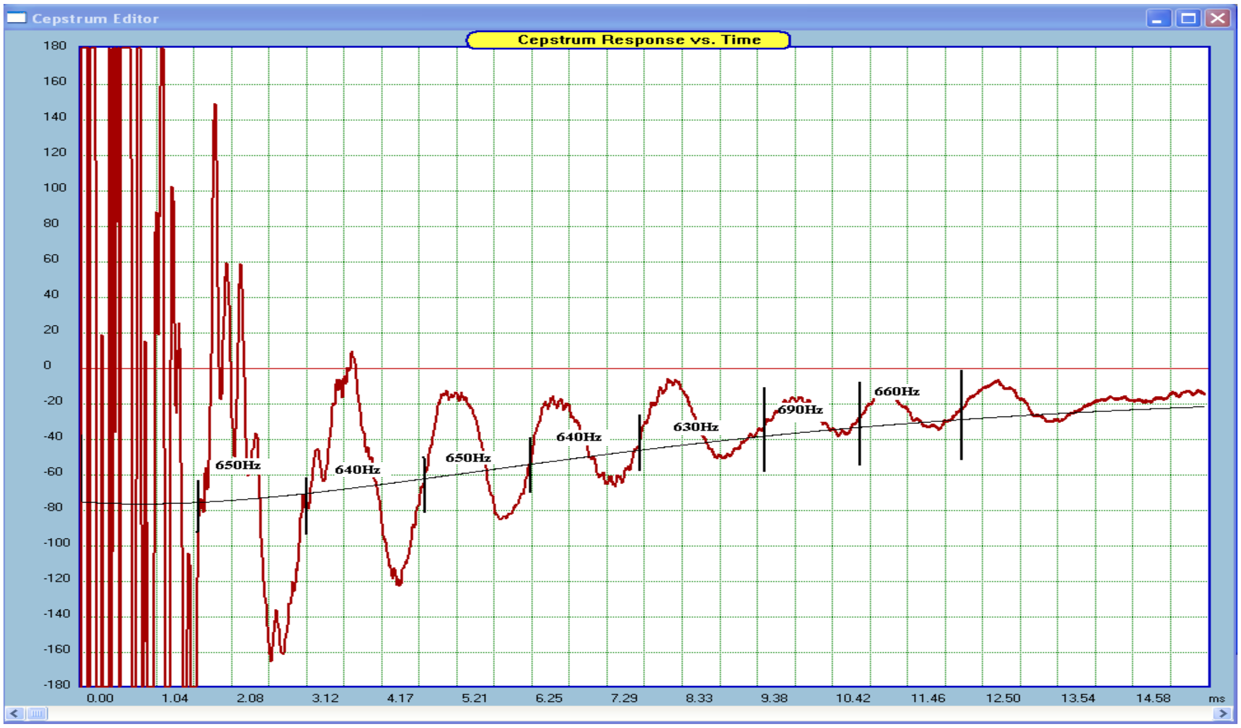


Figure 16.134. Closer analysis of these ripples confirm, that the frequency associated with those ripples is all **concentrated around 640Hz – which is where the deep notch is located on the SPL curve**.

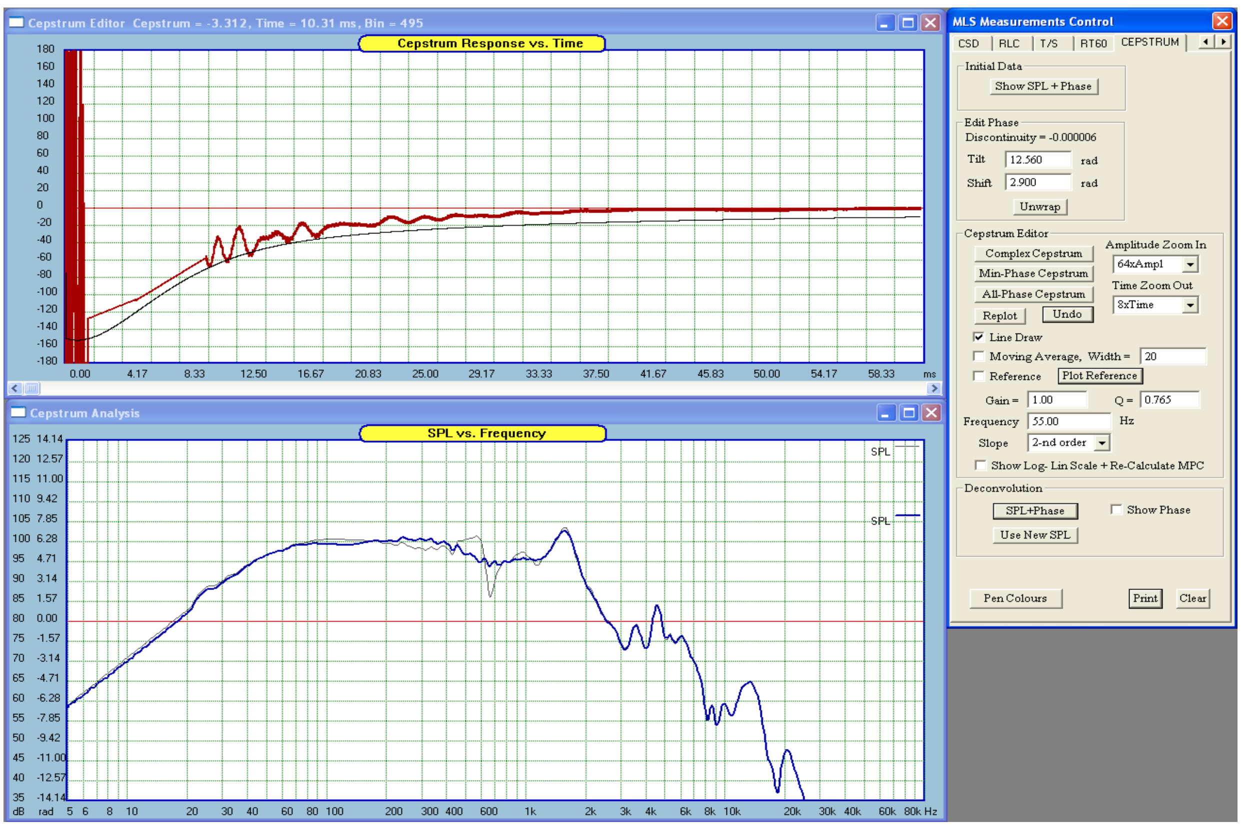


Figure 16.135. In order to confirm the problematic SPL issue, the Cepstrum was edited, but only from 2ms-10ms, to remove the largest oscillations from the cepstral tail, and the SPL was re-calculated. The result is shown above. Indeed, the notch around 630Hz has disappeared, and the SPL curve is much flatter from 400Hz-1.2kHz.

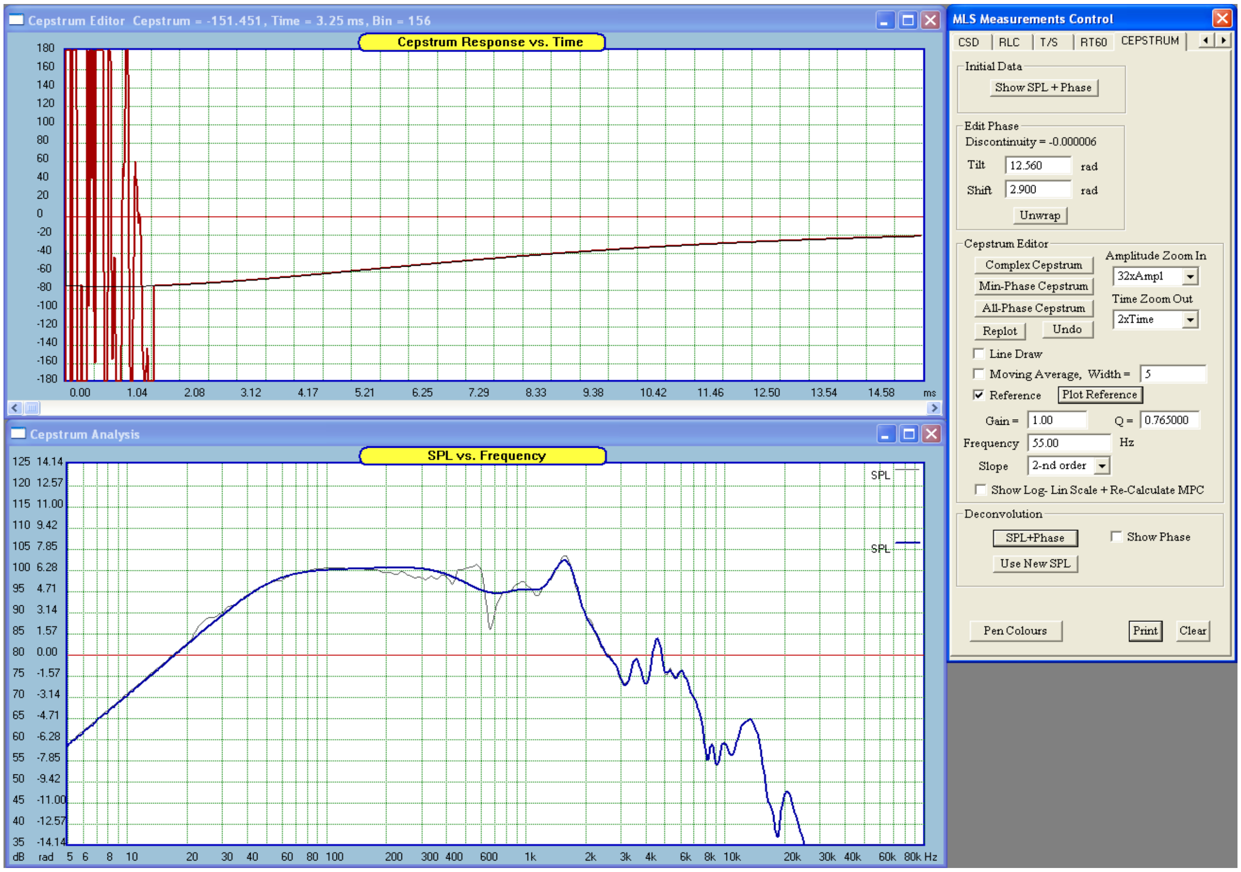


Figure 16.136. If you decide to replace the tail above 1.5ms with the analytical curve, the SPL will look as shown below – too good. **We have just replaced driver's OWN characteristics with something artificial.**

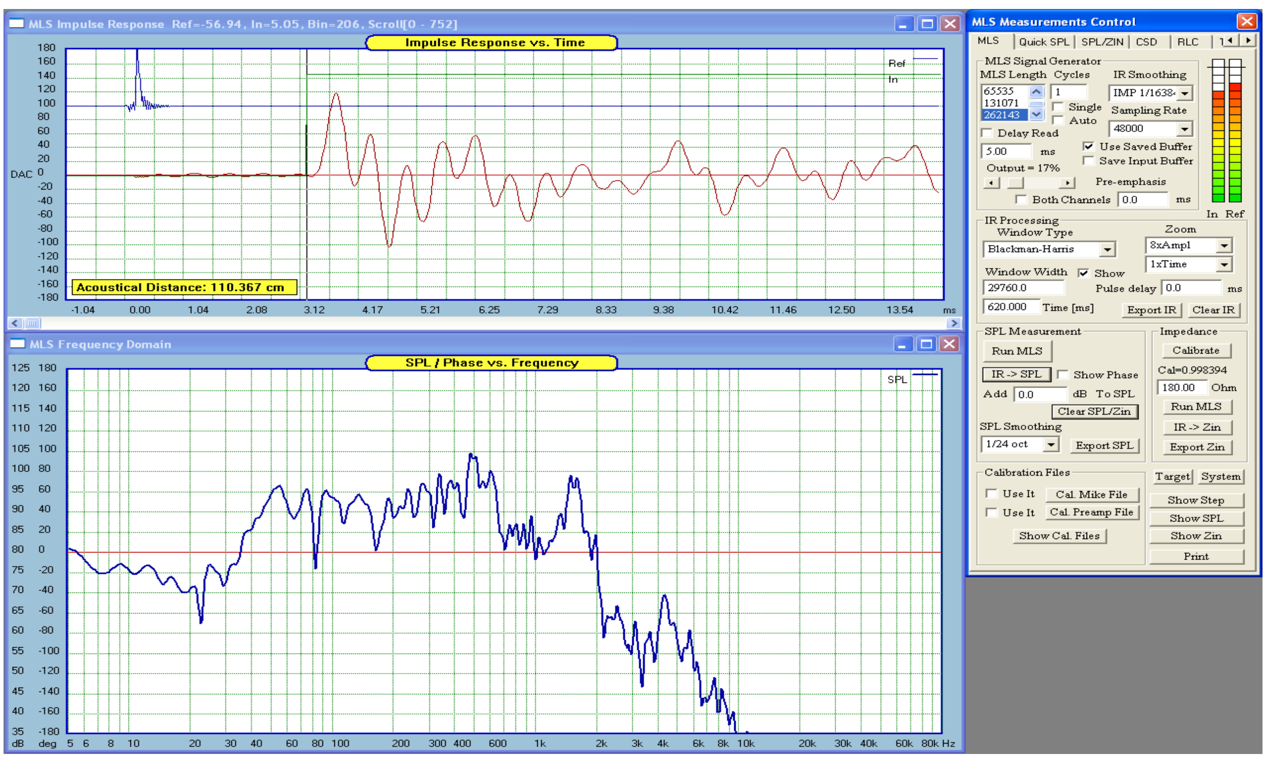


Figure 16.137. <u>The same loudspeaker measured in reverberant environment</u>, with the microphone at appr 1m distance is shown above. The deterioration in SPL curve is very dramatic.

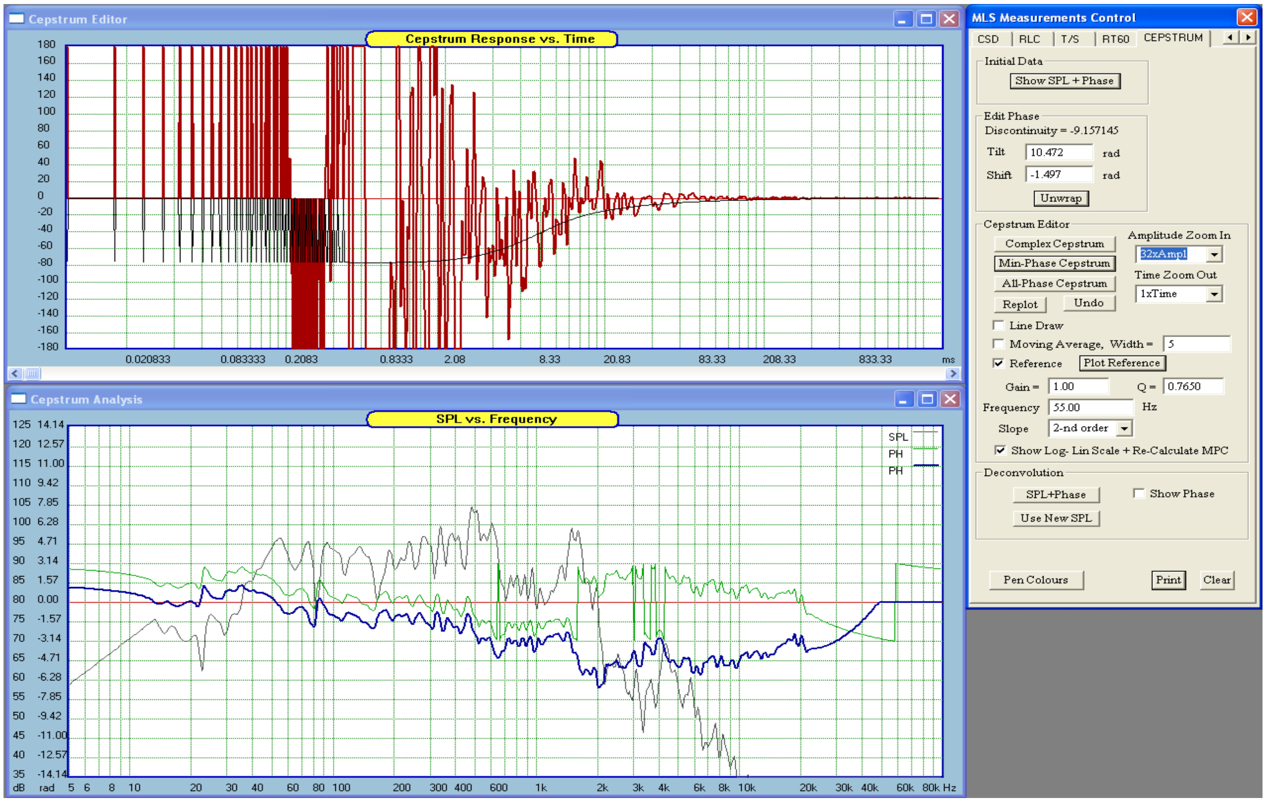


Figure 16.138. The cepstral curve no longer contains oscillatory-type ripples. The curve is now very asymmetrical and distorted.

How Can I Detect Problematic Drivers?

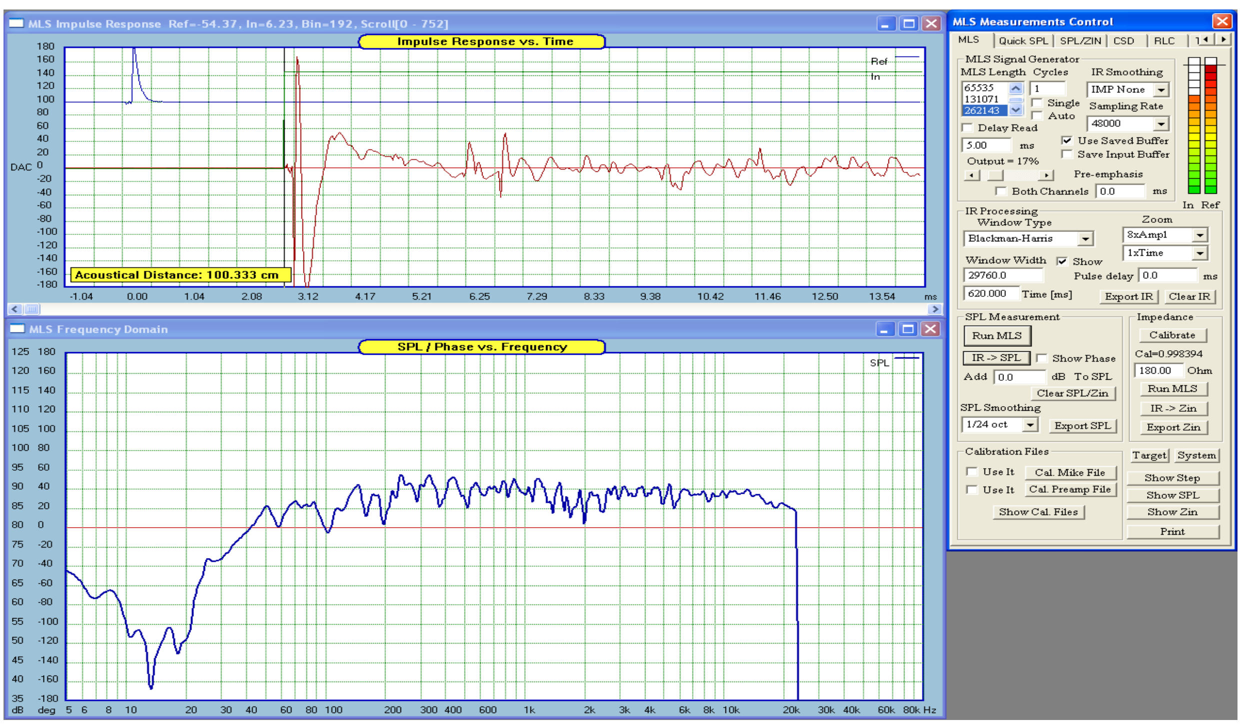


Figure 16.139. Loudspeaker shown above is a wide-band system exposed to a number of room reflections. In fact, this is the same driver, that was used to calculate RT60 – **"Figure 16.62. RT60 calculated without noise reduction techniques".** So the echoes are numerous.

Yet, taking the average, the overall frequency response could be approximated as a reasonably smooth curve. The same can not be said about the curve below. This is our problem woofer, with the SPL jumping wildly around 400-1500Hz.

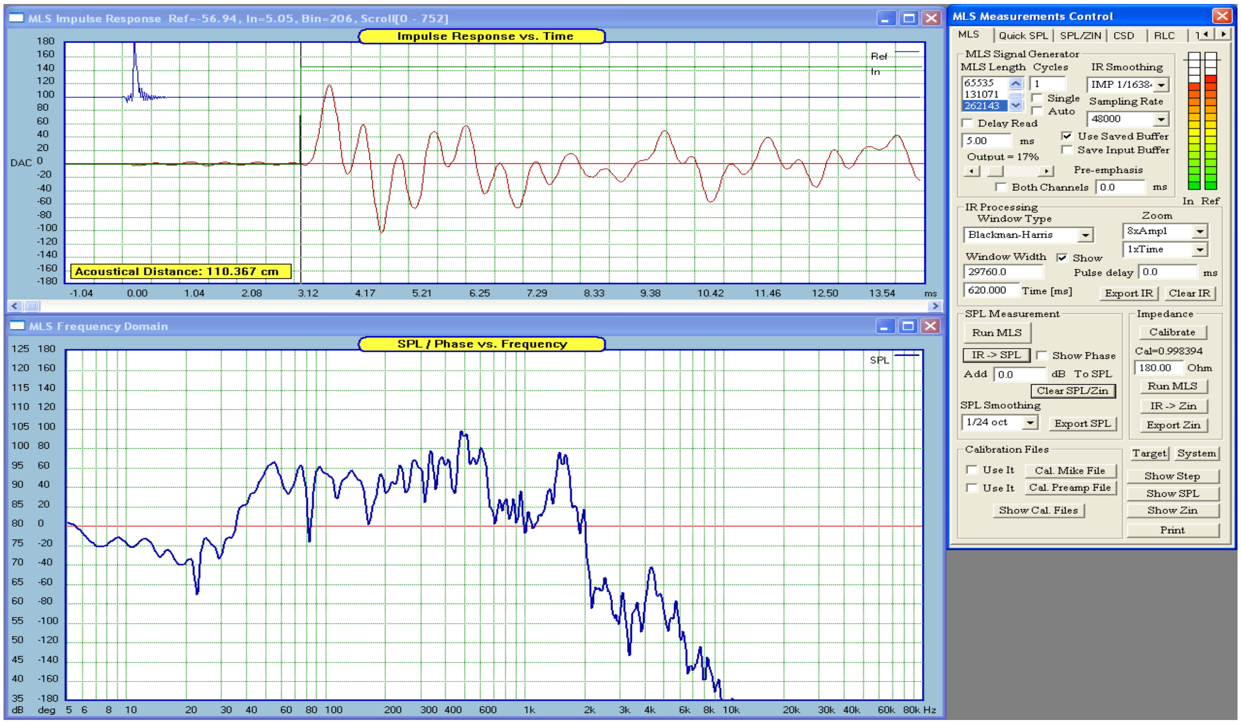


Figure 16.140. SPL of our problematic woofer.

Possible solution is presented below.

1. Tail HBT using parameters that would reflect the design specification (2-nd order, F3dB = 55Hz, Q = 0.765). See picture below, as it is very important. The thin red target curve is generated using **"Target**" button on the MLS TAB.

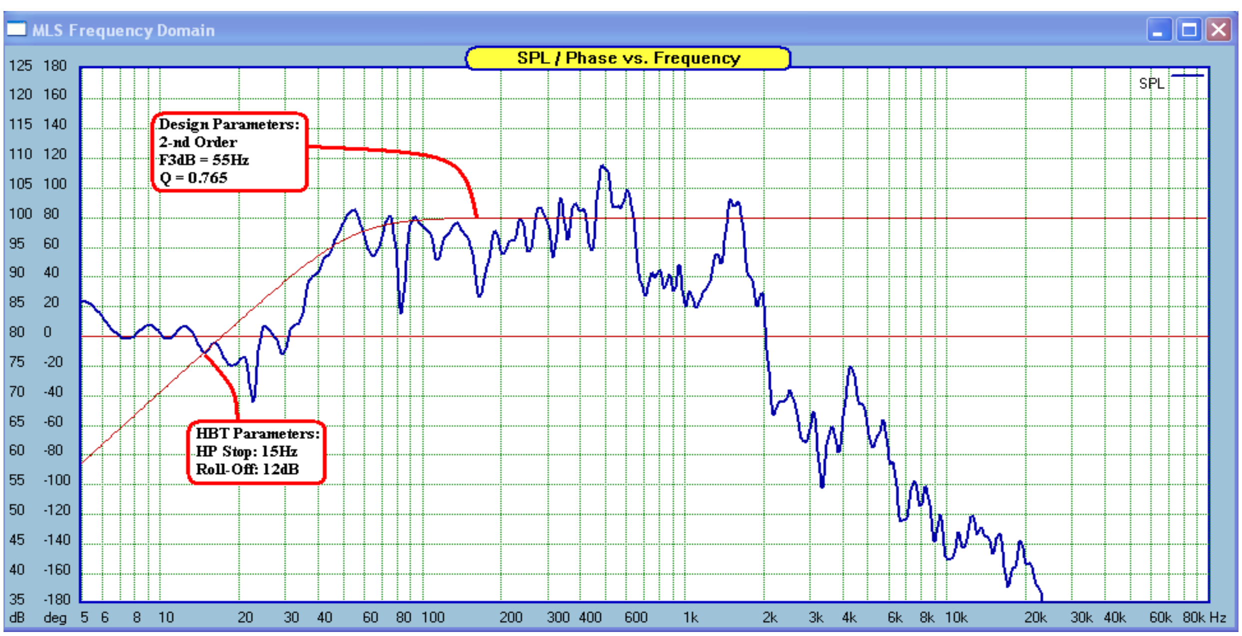


Figure 16.141. Evaluation of theoretical target curve vs. measured SPL.

2. Run Cepstrum Analysis with significant analytical curve substitution. The idea here a is to deliberately affect driver's SPL above 600Hz.

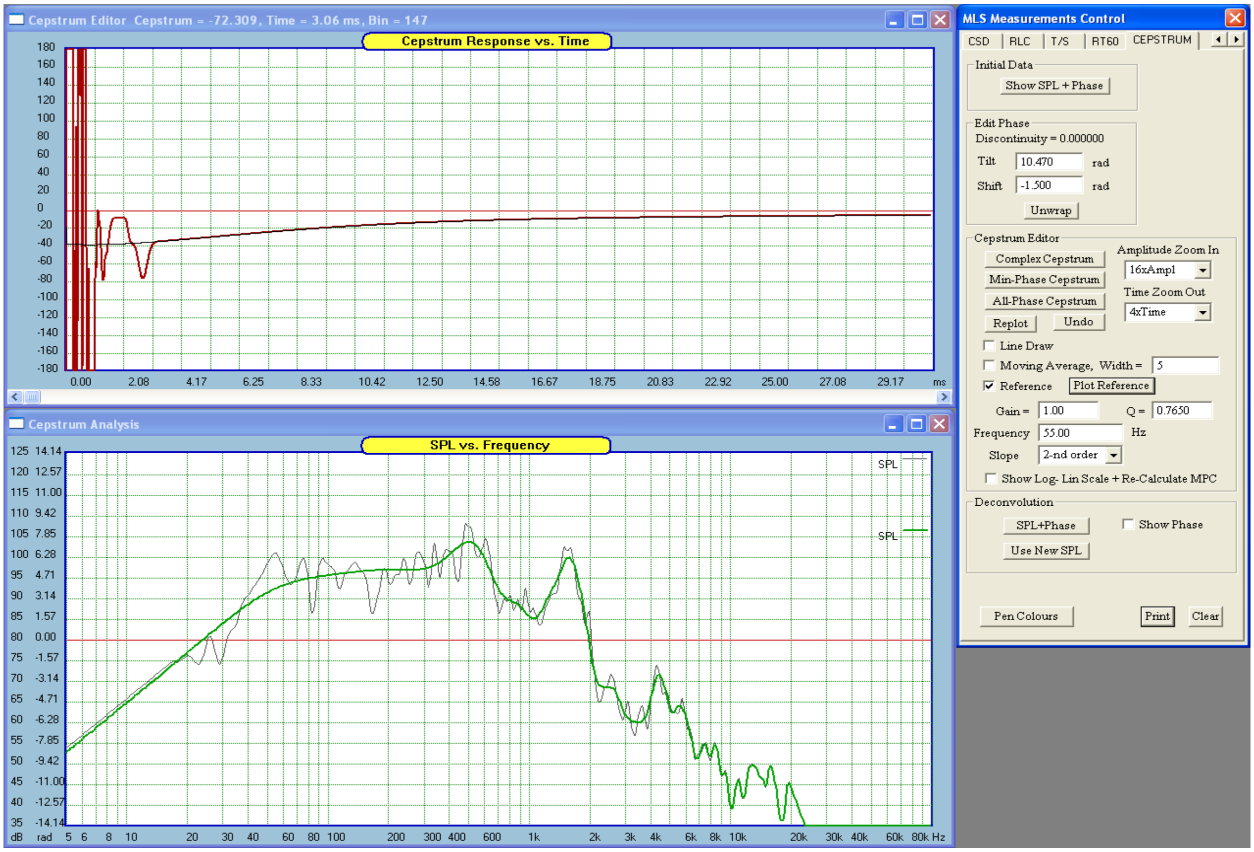


Figure 16.142. Cepstral Deconvolution using Analytical Tail Substitution+ Moving Average.

3. Restrict this driver for usage in your system to the frequency range below 600Hz. This is very important.

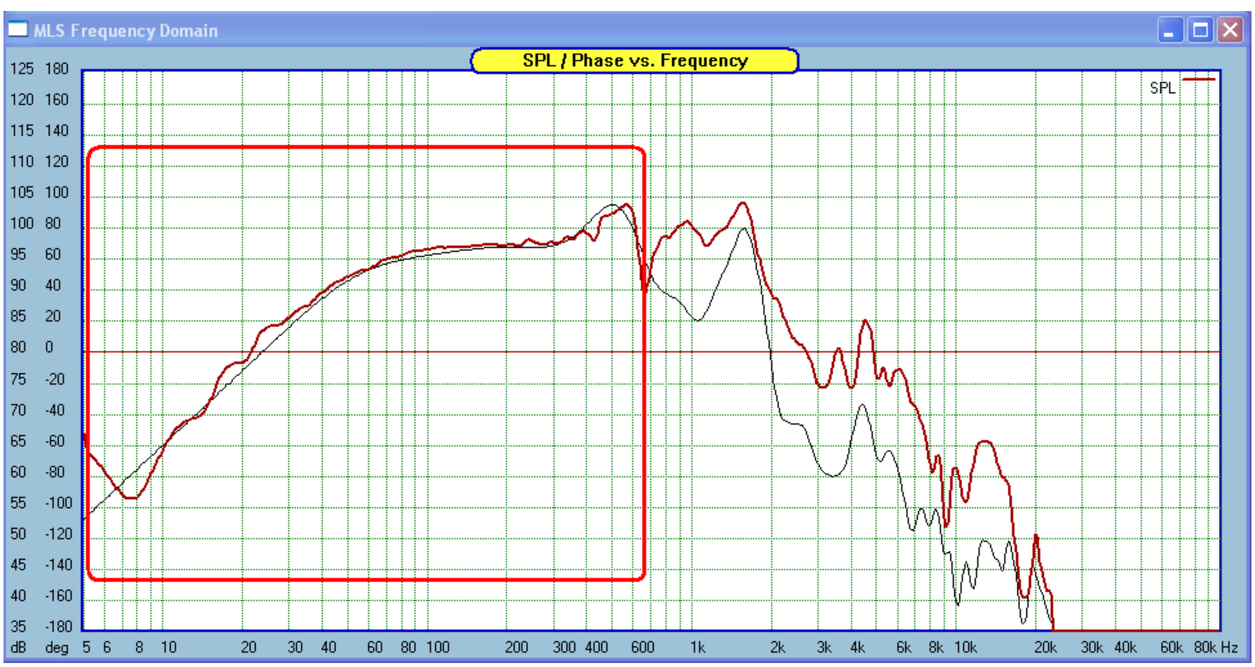


Figure 16.143. Example driver will be restricted to operate below 600Hs in the future system.

We have examined driver's frequency response taken in room conditions, and concluded, that there is a potential problem using this driver above 600Hz. As a consequence, we are bound to restrict this driver for usage in our system to the frequency range below 600Hz. This is very important, as we can disregard SPL above 600Hz, as long, as the SPL below 600 Hz has been recovered with the design parameters in mind.

Example presented above was rather difficult one to resolve. We started with a driver with extremely irregular frequency response. As if this was not bad enough, the driver was measured in a reverberant conditions, resulting in superposition of a multitude of reflections on the already irregular frequency response. To resolve the problem, we have employed:

1. "Target" function – to visualize what the theoretical design SPL would look like – this was helpful in selecting HBT parameters.
2. HBT – to tail and anchor the low end of the expected frequency response. In our case, these were the frequencies below 15Hz, with the anticipated SPL curve at the correct level. In addition, using the HBT, we performed a "cleaning" process on the SPL curve. As explained before, initial HBT tailing is intended to remove all unwanted low-frequency and high-frequency noise, that inevitably enters the MLS (supposed to be LTI system) measurement process.
3. Cepstral Deconvolution process, with a heavy Analytical Tail Substitution, to recover frequency response above 15 Hz.

Recovered SPL below 600 Hz s in good agreement with the close-mike measurement.

Final Comments

It can not be stressed strongly enough, that Cepstral Deconvolution Function is intended to help removing room reflections **while retaining the loudspeaker's own frequency response signature**. The process is aided by using the three editing tools: Moving Average Smoothing, Segment Substitution and Analytical Tail Substitution.

Wherever possible, it is recommended to use Segment Substitution supported by Moving Average Smoothing. The Analytical Tail Substitution is the most powerful method, therefore **most prone to overuse**.

Calculator TAB

At times, you may be interested in a simple distance/reflection time calculator. This TAB allows you to enter a couple of important distances: (1) acoustical distance between microphone and driver, and (2) acoustical distance between the driver and the nearest reflective surface. The built-in algorithm will then calculate the "reflection free" path and will use this values to se the width of the window to cut out the reflections.

The are following controls available on the "Quick SPL" TAB:

1. "**Mike-Speake**r" – Data field to enter acoustical distance between microphone and driver.
2. "**Driver-Reflection**"- Data field to enter acoustical distance between driver and the nearest reflective surface. Typically, it would be wall or floor etc.
3. "**Print**" – prints the screen.

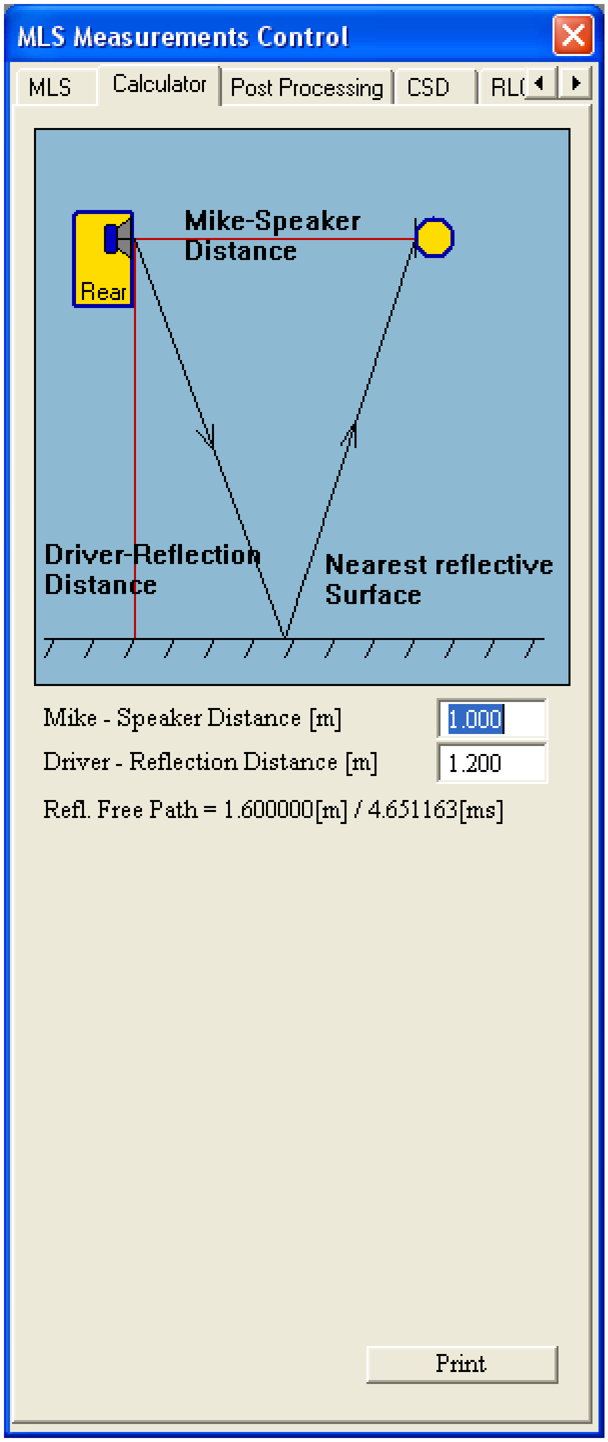


Figure 16.144. Quick SPL TAB

Matching Filter Technique

Loudspeaker measurements conducted in ordinary rooms are typically contaminated by multiple reflections from nearby surfaces. Such reflections usually manifest themselves as additional spikes and irregularities in the tail of the Impulse Response. This is unfortunate because low-frequency SPL information of the loudspeaker being measured is also contained in the tail of the impulse response. Several methods are being used to reduce or remove room reflections, but they typically come with a penalty of removing low-frequency SPL information from the measured response. Cepstral Deconvolution is one method, that attempts to reconstruct the low-frequency SPL information.

Matching Filter technique is another option to consider. Basically, this technique attempts to make the Impulse Response more compact (shorten the IR) such a way, that it will fit into the available, reflection-free time window.

The logic of this approach is clearly visible in the example below – during loop test, you can window the Impulse Response with a very short window (just 1.0ms) and you still get the expected, perfectly flat, undistorted, low-end frequency response.

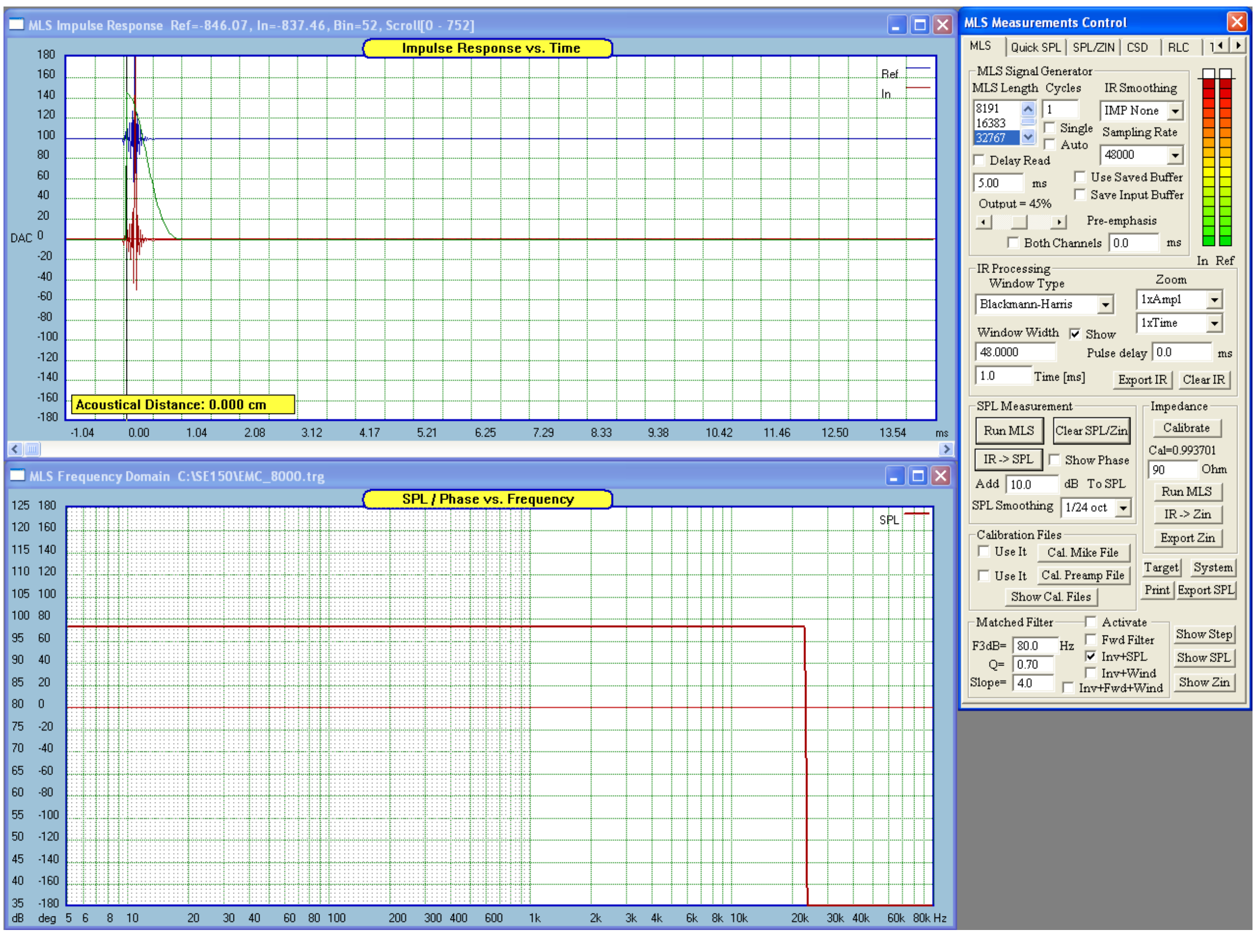


Figure 16.145. Impulse Response with a very short window.

Therefore, if we could somehow make the low-end of the frequency response flat, this would reduce the length of the tail in the Impulse Response, consequently, the shorter tail would fit into the available time window. Then, after windowing, we would simply “undo” the SPL flattening process. This is basically the concept of Matching Filters technique.

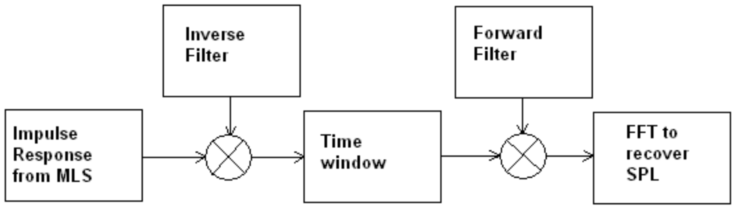


Figure 16.146. Concept of Matching Filters technique.

Detailed explanation of the process is presented below, and is based on measurements of a vented (4-th order) loudspeaker system in a living room. The resulting response is contaminated by room reflections and typically, also contains significant amount of low-frequency noise – see figure below.

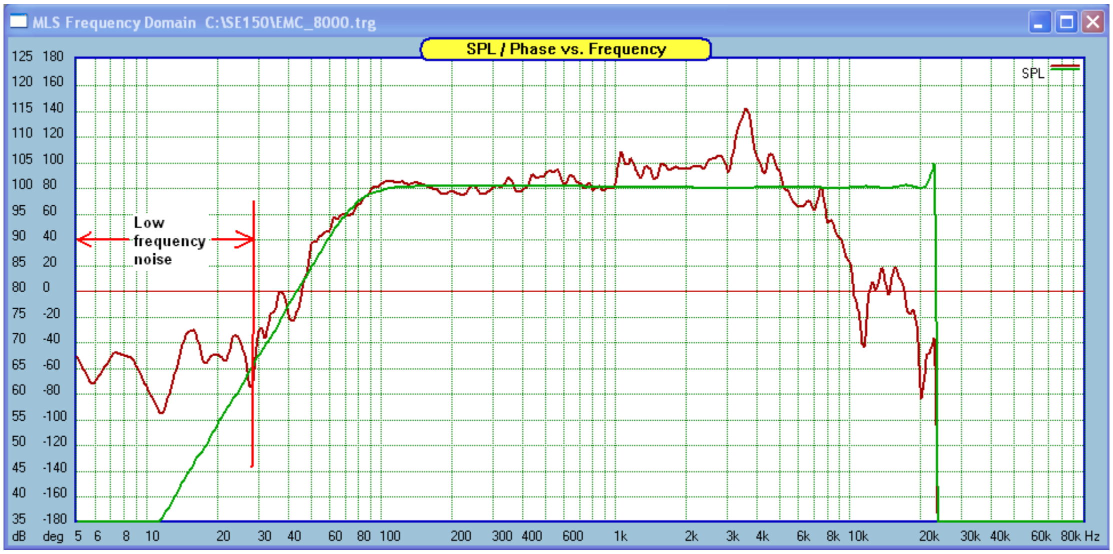


Figure 16.147. Vented (4-th order) loudspeaker system in a living room

Since our system is of 4-th order, it can be seen, that the measured in-room frequency response is quite irregular, and initially follows the 4-th order filter curve, but departs significantly from it towards the low-end of the frequency scale. This is due to in-room measurement conditions. “Fwd Filter” checkbox was selected and the “IR->SPL” function activated. The result of this operation is shown below

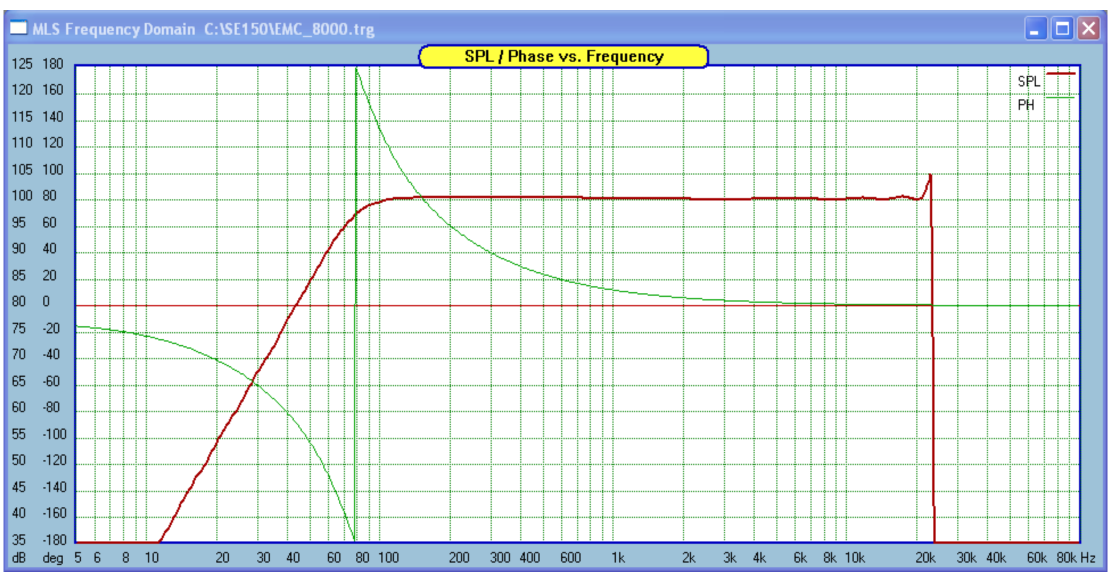


Figure 16.148. Vented (4-th order) loudspeaker system simulated SPL.

We would anticipate, that when the INVERSE filter (to the one shown above) is applied to the recovered frequency response, we may be able to flatten at least some portion of the resulting frequency response. This would translate into shorter Impulse response.

Inverse filter parameters for this example were selected as follows:

F3dB = 80.0Hz (allowed range is 5.0Hz – 1000Hz)
Q-factor = 0.70 (allowed range is 0.25 – 2.0)
Slope = 4.0 (allowed range is 1.0 – 6.0)

Now, "Inv + SPL" checkbox was selected and the "IR->SPL" function activated. The result of this operation is shown below. It is observable, that frequency response has been flattened between 35Hz and 100Hz. It is important to recognize, that if this measurement was conducted in free-field, our loudspeaker's SPL would be –35dB down at 30Hz. Therefore, it is quite acceptable to disregard the in-room SPL curve below 30-35Hz. It does not contribute meaningful information – it's just undesirable noise.

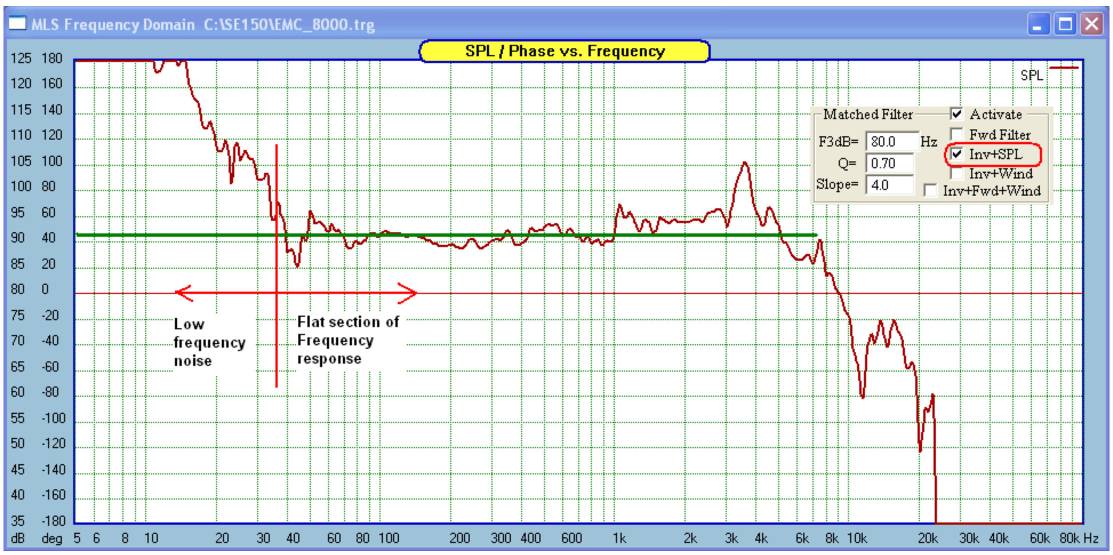


Figure 16.149. Disregard the in-room SPL curve below 30-35Hz.

In the next step, we must apply FFT window. There are 5 types of windows available for this process. The 5.0ms Blackman-Harris window was selected – see below. This window removes room reflections.

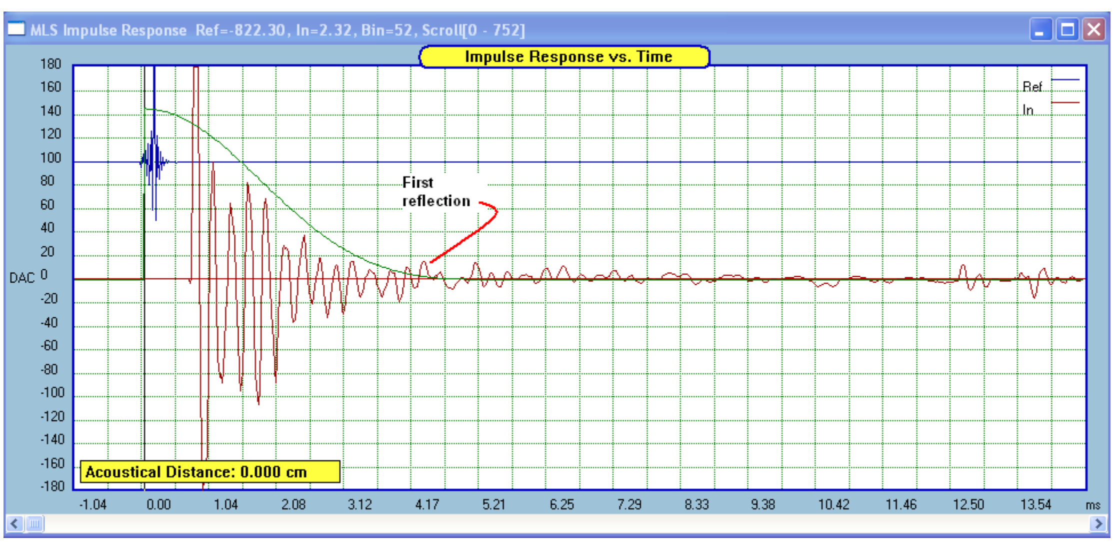


Figure 16.150. 5.0ms Blackman-Harris window.

Now, “Inv + Wind” checkbox was selected and the “IR->SPL” function activated. The result of this operation is shown below. It is observable, that frequency response has been practically flatten below 100Hz.

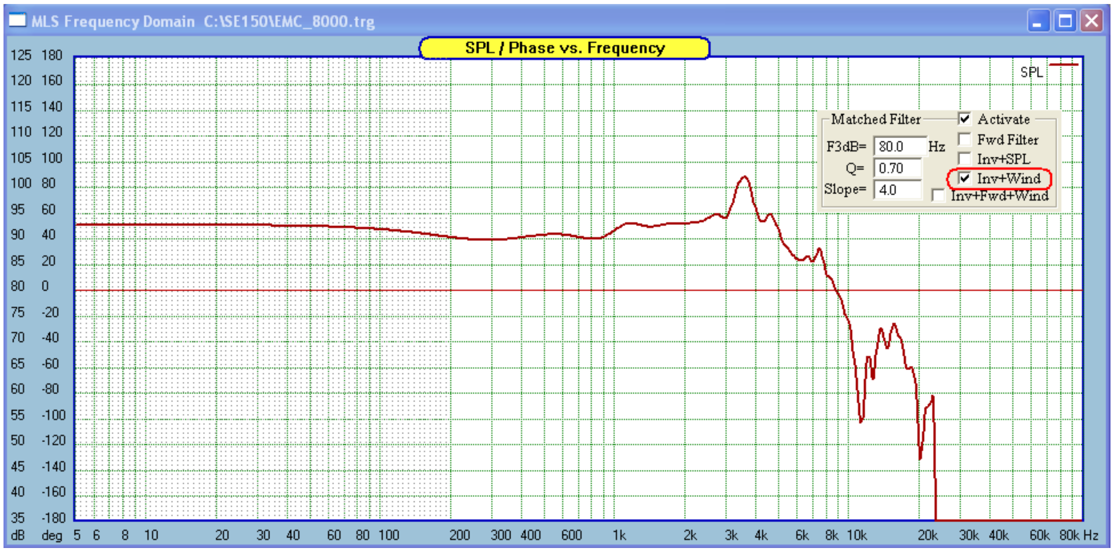


Figure 16.151. Frequency response has been practically flatten below 100Hz

In the final step, we must “undo“ the effect of the inverse filter. “Inv + Fwd + Wind” checkbox was selected and the “IR->SPL” function activated. The result of this operation is shown below. It is observable, that low-end of the frequency response has been practically cleared from room reflections and follows the expected 4-th order vented enclosure. This result was accomplish with 5.0ms Blackman-Harris FFT window.

It is beneficial to compare the Matched Filter technique with other options. This is illustrated on the figure below.

Green curve – raw SPL as measured in-room.
Brown curve – 5.0 ms Blackman-Harris FFT window was applied to raw SPL curve. The penalty is severe loss of accuracy below 500Hz.
Blue curve – Matched Filter approach. Room reflections practically removed.

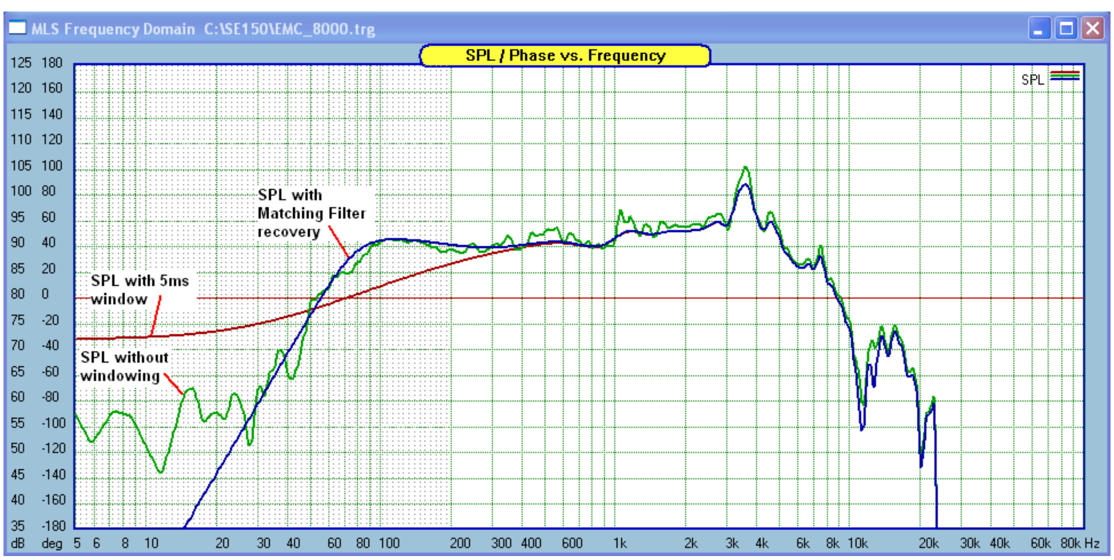


Figure 16.152. Final effect of Matched Filter approach on SPL.

Matching Filter Controls

The Matching Filter controls are grouped together as shown below.

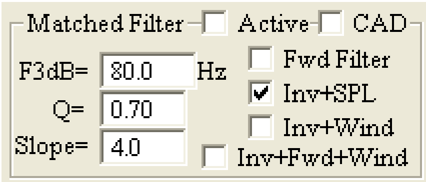


Figure 16.153. Matching Filter controls

Active – Activates Matching Filter method when the “IR->SPL” function activated.
CAD – Uses Filter topology developed, and SPL plotted in CAD system. For more information on this option please consult John Kreskovsky website – www.musicanddesign.com
Fwd Filter – Shows Forward Filter transfer function when the “IR->SPL” function activated.
Inv + SPL – Shows Inverse Filter + measured SPL when the “IR->SPL” function activated.
Inv + Wind - Shows Inverse Filter + FFT windowed SPL when the “IR->SPL” function activated.
Inv + Fwd + Wind - Shows Inverse Filter + Forward Filter + FFT windowed SPL when the “IR->SPL” function activated.
F3dB = – Inverse/Forward filter cut-off frequency (allowed range is 5.0Hz – 1000Hz)
Q-factor = Q-factor of the filter (allowed range is 0.25 – 2.0)
Slope = Roll-off speed, or filter order (allowed range is 1.0 – 6.0)