EasyLab Advanced

Important Note: Please consider PC and sound card related issues described in Chapter 18.18 to 18.28 prior commencing any work with the EasyLab system

Chapter 16. Linear Digital Measurements

Short Review of the Measurement Methods for SPL/Phase

When measuring the SPL of your loudspeaker, the utmost attention must be applied **to keeping the environmental effects out of the measurement**. The typical goal of an SPL measurement is to measure the transducer or system, without measuring the environment. To achieve this goal several methods of SPL measurement have been proposed. To measure SPL without environmental reflections, two methods can be employed:

1. Setup the environment so as not to produce any significant reflections, or
2. Use gating to remove reflections.

When making SPL measurements, any walls or other reflecting surfaces will produce multi-path signal reflections, which will be summed together into the response measurement. These reflections will result in significant irregularities (narrow peaks and dips) in the response curve that are not actually there in the transducer itself. As such, these measurements are environment dependant if the goal of the measurement is to analyze only the loudspeaker and not the room plus the loudspeaker. There are several types of environments which can be employed to eliminate, at least to a large degree, any multi-path reflections caused by walls or boundaries. In fact, these methods are recommended as the ones which can be used when measuring full range frequency response curves down to low frequencies. Gating method tends to be effectively used at mid and high frequencies, but still does not solve the problem of low frequency measurement. FFT-based SPL measurements can be successfully carried out in several types of environments:

1. Anechoic chamber,
2. Ground plane,
3. Half-space and
4. Near-field.

Anechoic Chamber Type Measurements

Anechoic measurements require either a specially built anechoic chamber, or suspending the speaker in mid-air, far from any boundaries. This method includes the enclosure diffraction effects. The purpose of an anechoic chamber is to absorb all sound striking the walls so that no reflection occurs. This is typically accomplished by covering the walls of a room with sound absorbing materials (pyramid-shape). However, even rigorously designed chambers still do not provide the ideal absorbing properties at low frequencies, where the total absorption is extremely difficult to achieve. For this reason most chambers exhibit a constant pressure behavior as frequency is decreased. At low frequencies the pressure field will cause a rise in level with decreasing frequency typically 12dB/Octave. In the mid and high frequencies the chamber will usually have good anechoic characteristics. True anechoic measurements are beyond the reach for most of us.

Ground Plane Type Measurements

Ground plane measurements are quite similar to anechoic measurements. Because the measurement includes both the direct sound from the enclosure, plus the reflection from the ground plane, the SPL for the same microphone placement distance will be 6dB higher than the anechoic SPL level. The only requirement for a ground-plane measurement is a large reflective surface, with no obstructions for a radius of 8-10 meters or more. The procedure is to place the loudspeaker on its side on the ground, and then place the microphone on the ground in front of the speaker. This method includes the enclosure diffraction effects. The increase in SPL is about 6dB, the frequency being determined by the total area of the baffle. The ground plane method is a very good (inexpensive) testing solution and all that is required is a flat surface with some open area around it.

Half-Space (Pit) Measurements

Half-space or pit measurements are made by placing the loudspeaker in the small pit ground with the drivers facing up, and the microphone placed right above the enclosure. Again, the microphone should be directly in line above the high frequency driver. This type of measurement does not include the enclosure diffraction effects. The radiation field is half of a full sphere so the level is 6dB higher than a free field measurement.

Near Field Measurements

The measurement methods discussed in the sections above are examples of far field measurements. The microphone is far enough away from the source that the sound radiation at the microphone is spherical. If the microphone is moved to a very close distance near the piston surface of the source, the radiation becomes a pressure field. This is known as the near field. By placing the microphone closer to the source, (5 mm) boundary reflections are eliminated since the reflection amplitude relative to the direct amplitude will be much smaller.

This type of measurement does not include the enclosure diffraction effects.

The recommended distance applicable to this method is dependent on the frequency range being measured. For measuring woofers, placing the microphone within 5mm from the cone will produce adequate near field curves. For ports, the test microphone should be placed in the port mouth. For midrange drivers placing the microphone as close as possible without touching the cone is necessary. For tweeters near field measurements may not be practical.

A good choice would be to take near field measurements on the woofer and port, and a gated measurement on the full enclosure to capture the mid and high frequency ranges. These curves could then be merged together to form a complete full range curve.

In far field measurements, the SPL output from the woofer and port are summed together acoustically. When taking near field measurements on the woofer, the port output does not appear in the pressure at the woofer. For this reason, ported enclosures require separate measurements to be taken at the woofer cone, and at the port. These must then be summed together mathematically.

The real goal here is to be able to measure the near field SPL, and then to predict what the SPL would be at a far field distance such as 1 Meter. This can be done if the piston area, Sd, for the source is known. It is simply a matter of scaling. The following formula gives the ratio of far field pressure to near field pressure at 1 Meter. Scaling Factor for Near Field Measurements (half-space):

$$P(far) = P(near)*20\log[\ 0.2821*\sqrt{(Sd)}\ ] \qquad [Sd \text{ in } m^2]$$

For a full-space environment, you would divide this ratio by 2, (or subtract 6dB).

Near-field loudspeaker measurements correctly predict the true full-space, far-field response only in the driver's piston range of operation below roughly f = 10897/d Hz where d is largest dimension of the largest diaphragm or port in centimeters.

Maximum-length Sequences

The program measures a linear, time invariant (LTI) system's impulse response using a pseudo-random excitation called a maximum-length sequence (MLS). ML sequences have some unique theoretical and practical features separating them from other test signals commonly used to measure linear systems such as narrow pulses, white noise and swept sine signals

Mathematical properties of the MLS excitation allow efficient computation of very long impulse responses. When a large FFT is applied to an extended wide-band impulse response the result is a wide-band transfer function with very fine frequency resolution. Therefore, there is no need for zoom processing with the MLS approach. SoundEasy uses FFT of the same length as MLS (well, plus 1 of course). Therefore, there is no FFT length selection.

When using ML sequences for LTI system's measurements you should **keep in mind, that the MLS method is accurate only for LTI systems** meaning their transfer function does not change over the measurement interval. Therefore, you should not attempt to make room measurements while the room acoustics are changing. This might occur if enough people are moving around a room during a measurement session. The opposite is also true – if your sound card (or PC) is unable to keep the In/Out timing accurate, the MLS measurement system will not work.

Basic MLS/FFT specification

The program will generate 6 different MLS sequences and use associated FFTs as shown below:

MLS order	MLS length	FFT length
13	8191 points	8192 bins
14	16383	16384
15	32767	32768
16	65535	65536
17	131071	131072
18	262143	262144

Smoothing: 1/3, 1/6, 1/12, 1/24 and 1/48 octave smoothing
Sampling: 11025, 12000, 22050, 24000, 44100, 48000, 96000 (192000 where applicable).

The MLS signal is periodic with period 2^N -1 where N is the order of the sequence. The MLS generator is clocked by the sample rate clock. Therefore, the period of a MLS, in seconds, is equal to the sampling interval times the sequence period in points. For example, for a 48 kHz sampling rate (20.8 microseconds sampling interval) a 262144-point MLS will have a period of 5.452 seconds.

The MLS can be regarded as a train of pulses of equal amplitude but appearing in positive or negative direction in pseudo-random manner. Such signal is fed into the loudspeaker, so that he microphone will pick up the response to these pulses. The application of Hadamard transform, causes the individual responses at different times to be re-arranged, so they appear simultaneous and in one direction. It is now easy to understand, why the LTI requirement for the system – it must not change during the measurement.

True Frequency Resolution

The system transfer function is computed by applying an FFT to the impulse response or a portion of it as in the case of windowed, anechoic loudspeaker measurements. The displayed frequency resolution is given as the sample rate divided by the FFT size (i.e. 48000/262144=0.183Hz). Displayed frequency resolution is just the FFT bin separation. However, the true frequency resolution of a measurement is **at best equal to 1/T** where T is the length of the impulse response segment used in the FFT calculation (i.e. the width of a rectangular time window). **Thus, it is possible to have a very fine displayed resolution but a rather coarse true resolution.** For example, you could apply a 32768-point FFT to a 512-point segment of impulse response. In that case, the large FFT only acts to interpolate the true frequency resolution to form a much finer displayed resolution.

Data-tapering window shapes would yield even worse resolution than a rectangular window. In the frequency domain, SoundEasy indicates the true frequency resolution by **a gray-shading the invalid section of the graphics display**. The data falling in the grayed frequency range should be regarded as invalid.

For example, if the time window T = 4 milliseconds the lowest accurate frequency is 250 Hz. The true frequency resolution at all higher frequencies will, in this case, also be 250 Hz. Therefore, the next correctly measured and calculated data point will be at 500Hz, and the next at 750Hz, and so on…..
Please note, that between 250Hz and 750Hz there is only ONE valid data point !.

To obtain better true resolution you must measure an extended impulse response and use an equally long time window and FFT size to compute the transfer function – go outdoor, close-mike or anechoic.

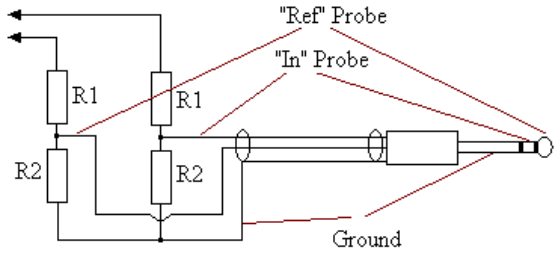
Noise In Presentation of MLS Measurement Results

This is significant issue. Virtually all MLS measurement systems include some kind of **averaging** of the results in time/frequency domains, necessary to produce better quality graphs. Typical figures are: average up to 16 MLS periods for a 12 dB SNR improvement. Further improvement can be obtained with 50-100 MLS runs. Please be aware, that if you use 262144-long MLS, clocked out at 48000Hz, you will generate 5.5 seconds of noise. The 100 MLS runs will result in 550 seconds (9.2 minutes) of loud noise. This timing becomes quite comparable with analog sweep measurement time. Averaging in frequency domain is typically performed for 1/3, 1/6, 1/12, 1/24 and 1/48 octave smoothing. Several other innovative, noise reduction schemes have been adopted in MLS software available on the market. This release of SoundEasy is designed to be a fast, high-resolution, single-shot, MLS tester, providing you with good quality presentation graphs. Therefore, we have implemented our propriety averaging scheme coupled with the Hilbert-Bode Transform to provide you with tools and choices capable of generating data over any desired frequency range**. The B-H Transform, on the top of it's usual task, essentially acts as a filter, generating and perfectly smoothing data outside the selected band – below "HP stop" and above "LP start".**

Additional Hardware Needed

No hardware devices are supplied with the program. You will need additional hardware as listed below:

1. **Audio power amplifier** capable of about 50W of power. The amplifier does not need to be of a "test equipment" grade. Any device with -80dB noise floor and the –3dB frequency range of 5Hz-30000Hz will do. The frequency/phase response of the amplifier will be automatically compensated for by the software. Input impedance of the amplifier need to be at least 10 times higher than output impedance of your sound card.
2. **Test microphone**. We have used CLIO "pencil type" electret microphone with a very simple, single-OPAMP, pre-amplifier. Bias voltage for the microphone was provided by the pre-amplifier. We live the choice of the microphone and pre-amplifier entirely to you. Just bear in mind, that the maximum non-clipped input voltage to your sound card is in the vicinity of 2V, so your test microphone + pre-amplifier should be able to provide this much voltage.



3. **Resistive probes**. For impedance measurements at 2.83V level, a simple voltage dividers of R1=47kΩ, R2=22kΩ may be sufficient. If you need to measure the impedance curve for different power levels, you may need to select different divider ratio. For SPL measurements, you may need to use different dividers again, depending of your particular voltage of interest.

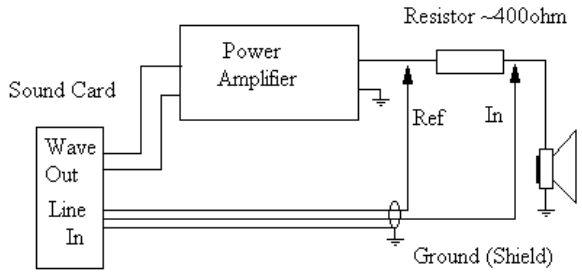
4. Additional audio cables as required.

As you can see, the above list is short and simple. You may already have your power amplifier, resistors and cables ready for use and the only piece of hardware you will need is good quality test microphone.

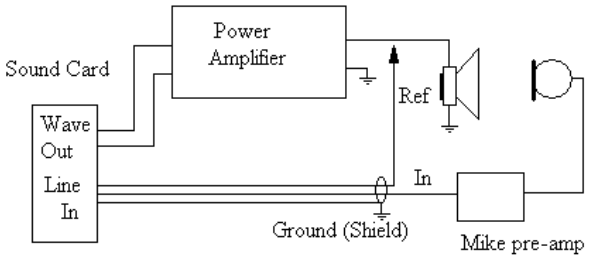
Operation of the MLS (Digital) Screen

The screen uses **two-port measurement technique**. One port ("Ref" probe) provides reference signal for amplitude and phase, and the other port ("In" probe) is actually used to measure the characteristics of the device under test – your loudspeaker. This simple arrangement allows the program to automatically calibrate out the amplitude/phase response of the power amplifier needed for the tests and also the amplitude/phase of the output circuitry of the sound card. Microphone calibration SPL/phase data needs to be imported into the screen.

When making impedance measurements, Zin, the "Ref" probe, sensing V1, is connected to the output of the power amplifier and the "In" probe, sensing V2, is connected right across the loudspeaker. The measured impedance is calculated from the formula: Zin = R*(V2/V1)/(1 – V2/V1). It is easy to observe, that V1 is a reference signal for voltage and phase. Phase of the input impedance is simply the difference between the phase of voltage V1 and the phase of voltage V2.



When making SPL measurements, the "Ref" probe, sensing V1, is connected to the output of the amplifier and the "In" probe, sensing V2, is connected to the microphone pre-amp output. The measured SPL (V2) is corrected in amplitude and phase accordingly to amplitude and phase measured by V1.



The **SPL measurements are relative**. They do not show the actual, calibrated results in dBspl. However, you can actually obtain very good results if you correlate MLS SPL screen curves with the indication of an inexpensive (but calibrated) SPL meter.

There is a range of SPL meters available on the market and you should get one for your home lab. For the best accuracy, correlation could be performed using "half-space" technique to avoid multi-paths or echoes picked up by the SPL meter.

Measurement Method for SPL

The MLS measurement technique involves three basic steps:

A. Exciting the loudspeaker under tests with the MLS signal.
B. Correlating the captured response using Hadamard Transform - impulse response of the loudspeaker.
C. Taking FFT of the impulse response to produce the frequency response curve.

Other parameters relate to FFT length, MLS length, sampling frequency and smoothing. Window type and length depends what type of measurement your are performing. You can easily get the job done selecting the highest sampling frequency, longest MLS and FFT and 1/12 octave smoothing. Follow the steps below to perform MLS measurements of the sound pressure (SPL/phase) of your loudspeaker.

1. **Connect the test setup**: power amplifier, test probes, microphone and the "loudspeaker under test".
2. **Generate MLS sequence using "MLS" TAB**
   - Select "MLS Length" from the provided list box – on SB Live! Sound card set to 131071.
   - Select sampling rate from the provided list box– on SB Live! Sound card set to 48000.
   - Select Generator level from 0% and rise it slowly to the required level.

- Press “Run MLS [SPL]” button. You should now hear the noise coming out from the measured speaker. The microphone will pick-up this signal and your “Input Level LED” indicator should lit-up for “In” bars and “Ref” bars. **The “In” indicator corresponds to the MEASURED signal from your microphone and the “Ref” indicator corresponds to the REFERENCE signal probe (connected to the output of the power amplifier).**

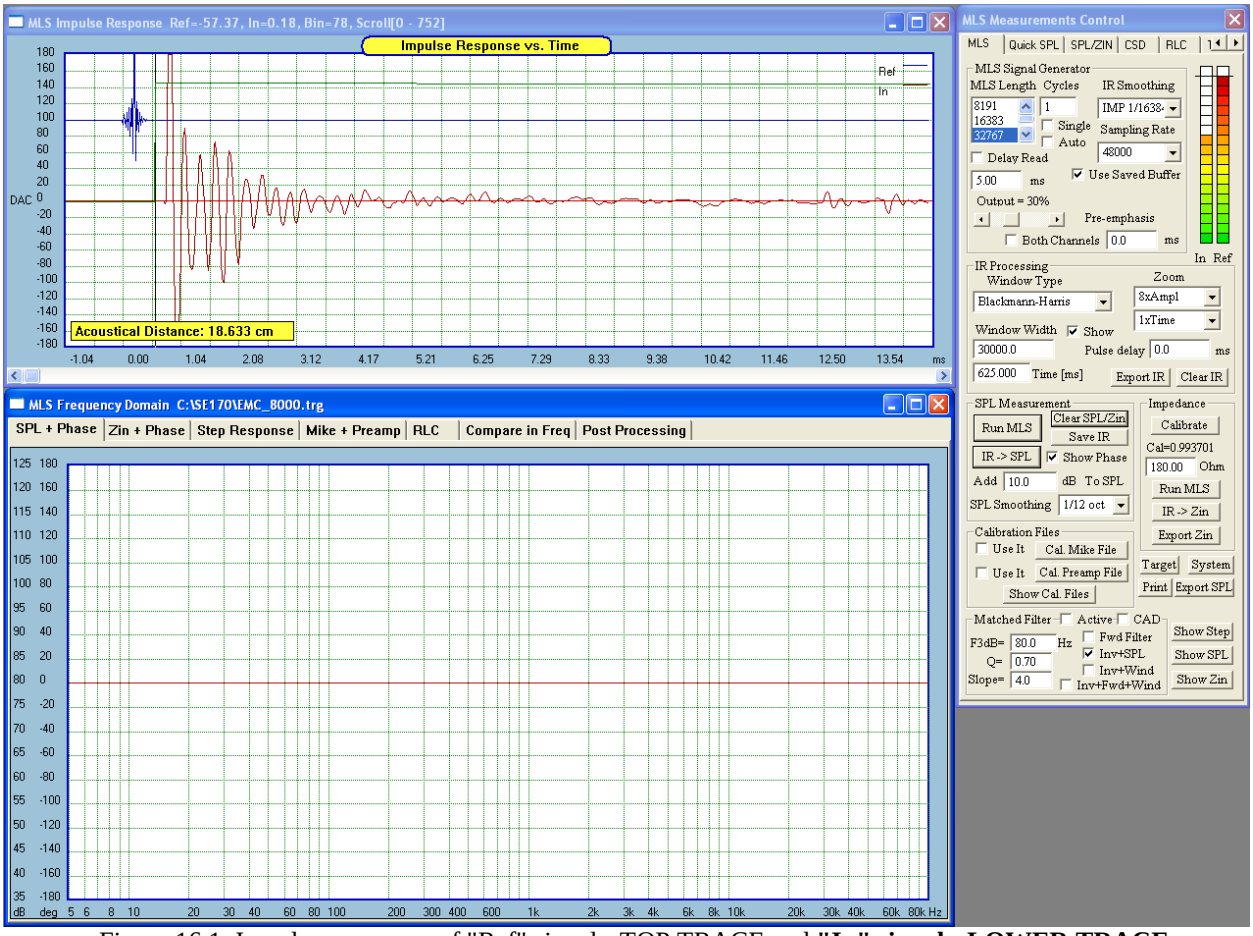


Figure 16.1. Impulse responses of "Ref" signal - TOP TRACE and **"In" signal - LOWER TRACE**

3. **Select windowing parameters**.
   When the MLS sequence has finished, you should see the impulse response of the measured signal (from the microphone) on the screen. The program will automatically place the time cursor 8 samples to the left of the peak value.
   - Position the time cursor on the screen now. This is accomplished by simply moving the mouse pointer onto the time domain screen and clicking the LEFT mouse button.
   - In order to remove the “time-of-flight” sound delay, position the time cursor right before the impulse starts to rise. You can do the fine cursor adjustments in 1 step increments/decrements by pressing the **“<-” and “->” Cursor buttons on your keyboard**.
   - Select the type and the width of the time window. For “close microphone” techniques you can select wider windows and for echoic environment, you should select narrow windows, aiming at cutting out the first echo.
   - Please DO NOT create wider windows than the length of the MLS.
   - **“Show”** – Check-box enabling the window to be displayed.
   - “**Type”** – Select window type using this list box
   - **“Width**” – Enter window’s width in **“samples”** into this data field.
   - **“Time”** – Enter window’s width in **“milliseconds”** into this data field.
4. **Select smoothing parameter**.
   This is a simple selection of one of the provided smoothing ratios: 1/3, 1/6, 1/12, 1/24 and 1/48 octave.

5. **Select additional delay.**
   - Adding a pure delay (enter **positive** number in the "Pulse Delay" field) causes the phase curve to tilt downward from left to right when displayed on a frequency scale. Conversely, removing a delay (enter **negative** number in the "Pulse Delay" field) causes the corresponding phase curve to tilt upwards.
   - A basic problem in loudspeaker phase measurements is that there is a large acoustical delay associated with the space between the loudspeaker and microphone. Placing the marker just before the Impulse response first arrival removes most of this delay in the time domain. Any residual delay can be removed through the "Pulse Delay" in the frequency domain. This field allows you to manually enter any desired phase delay.
6. **Generate SPL curve**.
   Simple press the **"IR->SPL"** button and wait a short while for the calculations to finish. Please note, that wider smoothing band will result in increased computation time.
   - "**Show Phase**" – check this button if you prefer phase response to be plotted as well.
   - "**Zoom**" – You can zoom in on the Impulse Response amplitude by selecting desired magnification from the provided list boxes.

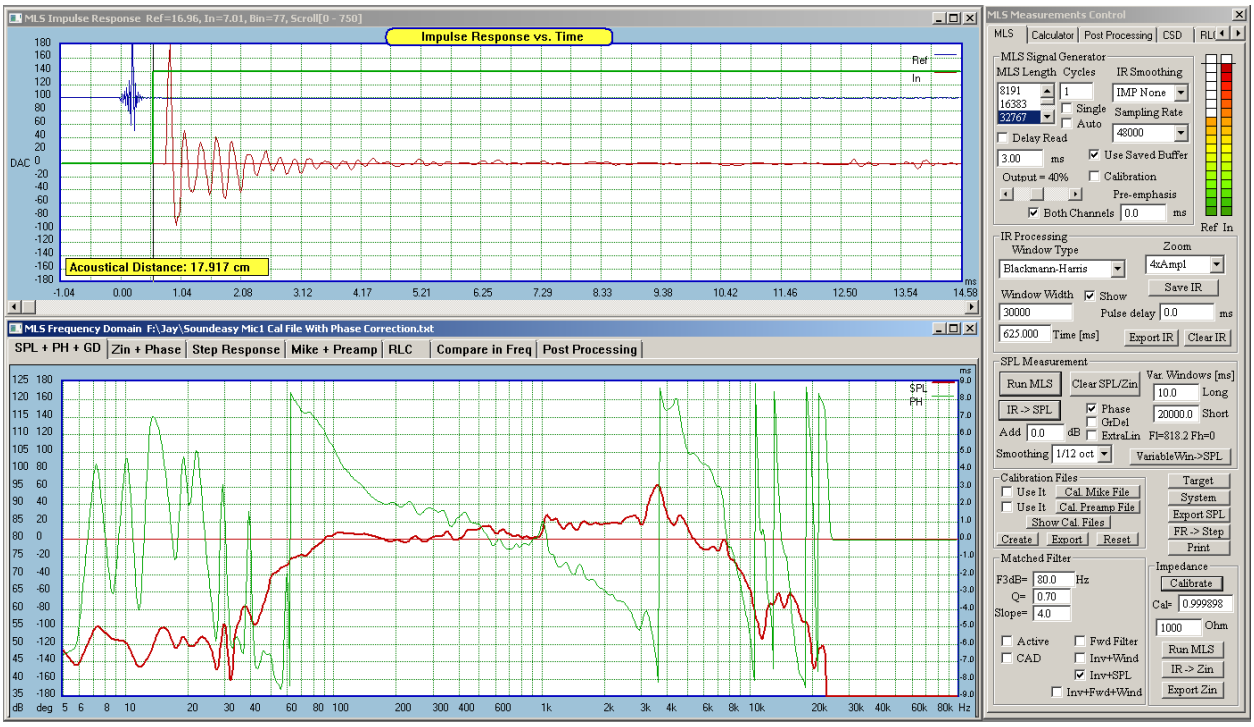


Figure 16.2 Example of SPL/phase response (1/12 oct smoothing) with echoes due to window of 30000 samples.

7. **If you need to extend the measured SPL/phase curves beyond the measurement frequency range**.
   This may happen if you selected screen frequency range wider then the sound card measurement frequency range. For example: reliable (with automatic equalization via Ref probe) frequency range of SB Live! card is approximately 10Hz – 20kHz. You have selected SoundEasy screens 1Hz 100kHz wide for your project work.
   The tool to use in this case, is the Hilbert-Bode Transform group-box provided at the right-bottom corner of the measurement screen. The transform is driven by 5 editable parameters and they should be selected to obtain the **best match for phase and amplitude between measured signal and calculated transform over the widest frequency range**. Typically, good match can be obtained between 20Hz and 20kHz. Generally, you would select the parameters to provide initial match for amplitude response and then fine tune the parameters to provide the match for phase.

Please note, that “Roll-off” parameters can be as high as +/- 100dB to +/- 120dB if the measured frequency response is highly irregular (resonant peaks) at the extremes of the measurement range. The program calculates phase using a Hilbert-Bode Transform from any magnitude curve. This is known as a Minimum Phase response and it does not contain any delay or positional information. It is a phase response produced solely from the magnitude data curve. This is very convenient for SPL measurements since the distance from the source to the microphone is completely unimportant. The phase curve will have no excess delay. This eliminates the usual problem of removing the arrival delay from the path of the source to microphone. “A minimum phase system is one which is able to transfer input energy to its output in the least amount of time for a given frequency response”. Mathematically, you would calculate the phase response of a minimum phase system by taking the Hilbert-Bode Transform of its log magnitude response (i.e. magnitude expressed in decibels). This relationship is also reciprocal: the log magnitude response of a minimum phase system equals the Hilbert-Bode transform of its phase response. Because the Hilbert-Bode Transform requires an infinite integral over all frequencies, however, any numerical computation of minimum phase is necessarily an approximation. Understanding the characteristics of the phase transform and how it works can be very important if maximum accuracy is to be obtained. **HBT is available from “Post Processing” TAB**.

For some data curves, the transform can be run directly and will provide excellent results with little attention to the original magnitude data. In other cases, a far better result can be obtained by correcting certain areas of the magnitude curve before running the minimum phase transform. Typically, the regions which can cause problems are **at the ends** of the measurement bandwidth. The transform must project the slope of the magnitude curve beyond the measured bandwidth. It does this by taking the last slope found as the curve reaches the low and high ends of the frequency range. It then assumes that: (1) the slope found at the low end will remain constant down to DC, and (2) the slope found at the high frequency limit will remain constant towards infinity. For this reason it is of fundamental importance that the slope of the magnitude curve at the high and low ends of the frequency range be approaching or already at their final asymptotic values.

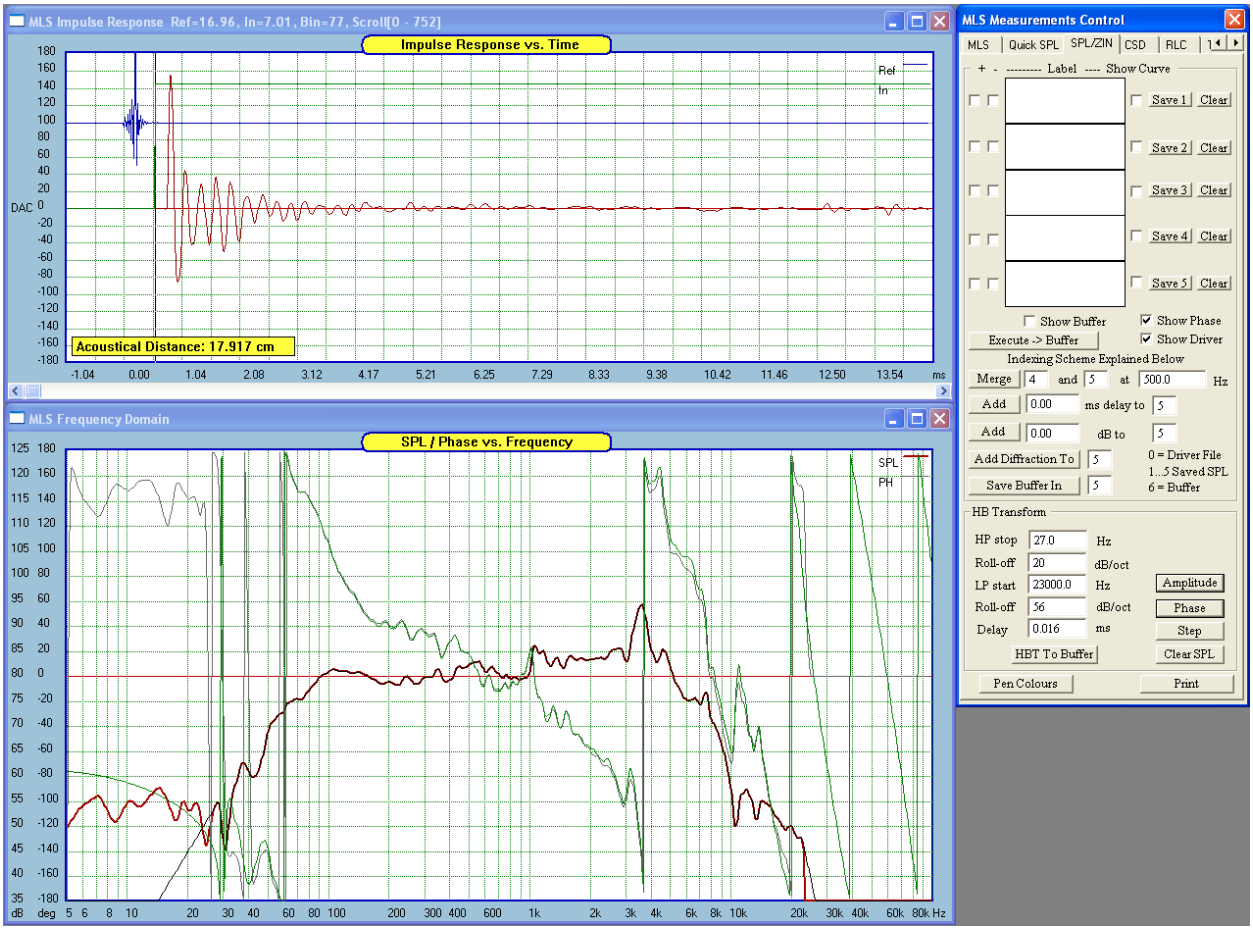


Figure 16.3 Hilbert-Bode Transform phase matching well with the measured phase from appr. 30Hz – 24kHz.

While most loudspeaker drivers usually exhibit nearly minimum phase behavior, this is not necessarily the case for complete loudspeaker systems due to possible non-minimum phase crossovers or, if the acoustic centers of the drivers do not lie in the same plane. Truly minimum phase loudspeaker system is a indeed an uncommon occurrence.

8. **Measured data (modulus of the amplitude and phase) is saved into the program data space as imported data. Therefore SoundEasy makes no distinction between imported measurements and its own measurement system data. However, if you decide to use Hilbert-Bode Transform on the measurement screen, the output of the transform will be saved directly into the Transfer Function data space.**

Measurement Method for Impedance

When making impedance measurements, Zin, the "Ref" probe, sensing V1, is connected to the output of the power amplifier and the "In" probe, sensing V2, is connected right across the loudspeaker. The measured impedance is calculated from the formula:

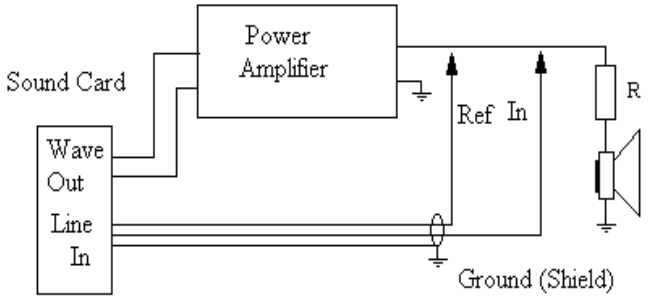
Zin = R*(V2/V1)/(1 – V2/V1), R > 400 Ω is recommended for constant current method

The method used for direct impedance measurements is close to constant current. This is because the source impedance (resistor R = 400 Ω, or higher inserted between the power amplifier output and the loudspeaker) is high so that the current following through the voice coil remains relatively constant**. Using lower resistance R is perfectly acceptable**. The formula above indicates clearly, that input impedance is heavily dependant on the voltage ratio, as measured by left and right channel of the sound card. There is a potential for significant error, as the voltage gains in both channels may not be exactly equal. More importantly, if you use probe resistive dividers, the voltage divider ratios may not be equal in both probes due to resistor tolerances. In order to ensure accuracy of the impedance measurements, you must CALIBRATE the system before a valid impedance measurement can be taken. During the calibration process, you will feed the same signal to both probes and the program will measure and memorize the difference in actual readings. This information will be use next for scaling the measured voltages.

CALIBRATION FOR IMPEDANCE MEASUREMENT

Follow the steps below to perform calibration for impedance .

1. **Connect the test setup**: power amplifier, test probes. Connect the test probes TOGETHER, at the output of the power amplifier – as shown below. Power-up the power amplifier.



2. **Select "Digital MLS**" from the main menu to invoke MLS measurement screen.
3. **Perform calibration**
   - Select Generator level from 0-100%.
   - Press "Calibrate" button in the Impedance group box. The setup will now generate 1000Hz tone needed for measurement of both channels. **The "In" indicator corresponds to the MEASURED signal from your loudspeaker terminal and the "Ref" indicator corresponds to the REFERENCE signal probe (connected to the output of the power amplifier).**

Having calibrated the system you can immediately follow the steps below to perform MLS measurements of the input impedance and phase of your loudspeaker.

1. **Connect the test setup**: power amplifier, test probes, test resistor and the "loudspeaker under test". Power-up the power amplifier.

2. **Select "Digital MLS**" from the main menu to invoke MLS measurement screen.

3. **Generate MLS sequence**
   - Select "MLS Length" to 262141. Select sampling rate to 48000. Select Generator level from 0-100%.
   - Press **"Run MLS**" button in the "Impedance" group of controls. You should now hear the noise coming out from the measured speaker. The microphone will pick-up this signal and your "Input Level LED" indicator should lit-up for "In" bars and "Ref" bars. **The "In" indicator corresponds to the MEASURED signal from your loudspeaker terminal and the "Ref" indicator corresponds to the REFERENCE signal probe (connected to the output of the power amplifier).**

4. **Select windowing parameters – this is done automatically**.
   - When the MLS sequence has finished, you should see the impulse response of the measured signal (from the microphone) on the screen. The program will automatically place the time cursor 10 samples to the left of the peak value. Select the type and the width of the time window. **Please DO NOT create wider windows than the length of the MLS.**

5. **Select smoothing parameter**.
   This is a simple selection of one of the provided smoothing ratios: 1/3, 1/6, 1/12, 1/24 and 1/48 octave.

6. **Select additional Zin scaling.**
   - The measured Zin curve can be shifted up/down by entering suitable data in the "Correction" data field.

7. **Generate Impedance curve**.
   Simple press the "Impulse->Zin" button and wait a short while for the calculations to finish. Please note, that wider smoothing band will result in increased computation time. If you need to see the impulse response again (to re-position the time marker), simply press "Show Impulse" button.

8. **If you need to extend the measured SPL/phase curves beyond the measurement frequency range**.
   This may happen if you selected screen frequency range wider then the sound card measurement frequency range. For example: reliable (with automatic equalization via Ref probe) frequency range of SB Live! card is approximately 10Hz – 20kHz. You have selected SoundEasy screens 1Hz 100kHz wide for your project work.

   1. The SPL/phase and impedance/phase data curves measured by the MLS screen are saved as "imported impedance " data. Therefore, you can view them in the "Editor" and "Transfer Function" screens the same way as you would deal with the imported measurements. For instance, you can use the "Transfer Function" screen to re-create Zin transfer function, if you wish. This step is however NOT NECESSARY. The MLS measurement screen will calculate transfer functions for SPL and impedance exactly the same way as the "Transfer Function" screen does. Please note the **correct behavior of the phase response**. The phase response decays to zero at low-end of the frequency response. This is because the modulus of the input impedance becomes purely resistive at low end of the audio frequency range. Also, the phase response approaches asymptotically 45^0, at the high-end of the audio frequency spectrum. This is due to "semi-inductance" element, rather than normal electrical inductance of the voice coil dominating at these high frequencies. This behavior can be clearly seen on full-range frequency screens, particularly, after HB Transform has been applied. This a very practical "sanity check" for your impedance measurements. If you can not see this pattern over an extended frequency range – something is wrong with the measurement. Finally, please note somewhat unusual parameters needed for Hilbert-Bode Transform to track the impedance curve correctly. The low-end slope has 0.0 dB/oct slope and the high-end slope of the impedance modulus is rising, therefore -3.4dB/oct (negative) value was entered. HBT is a unique and extremely useful tool in acoustics.

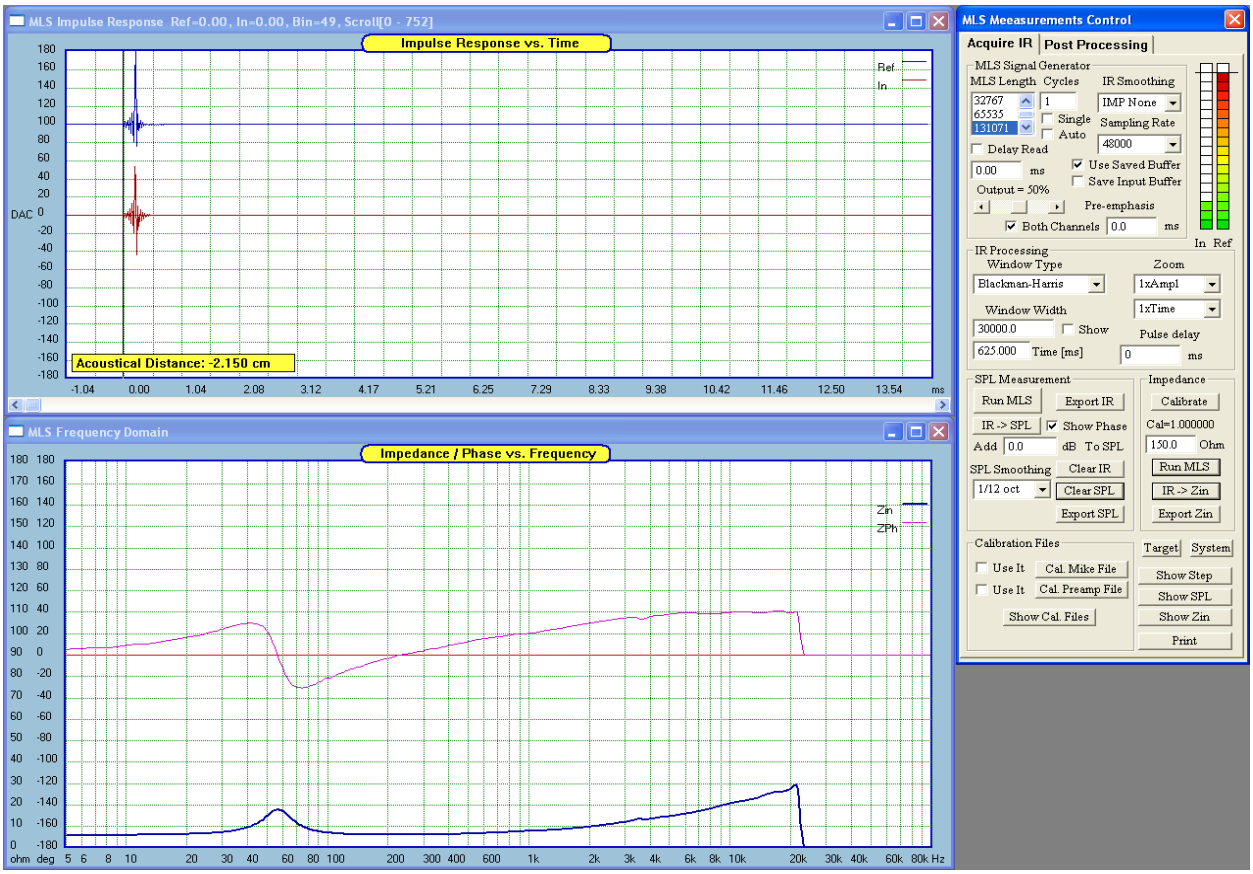


Figure 16.4 Input impedance Zin/phase over narrower frequency range of 5Hz – 20kHz.

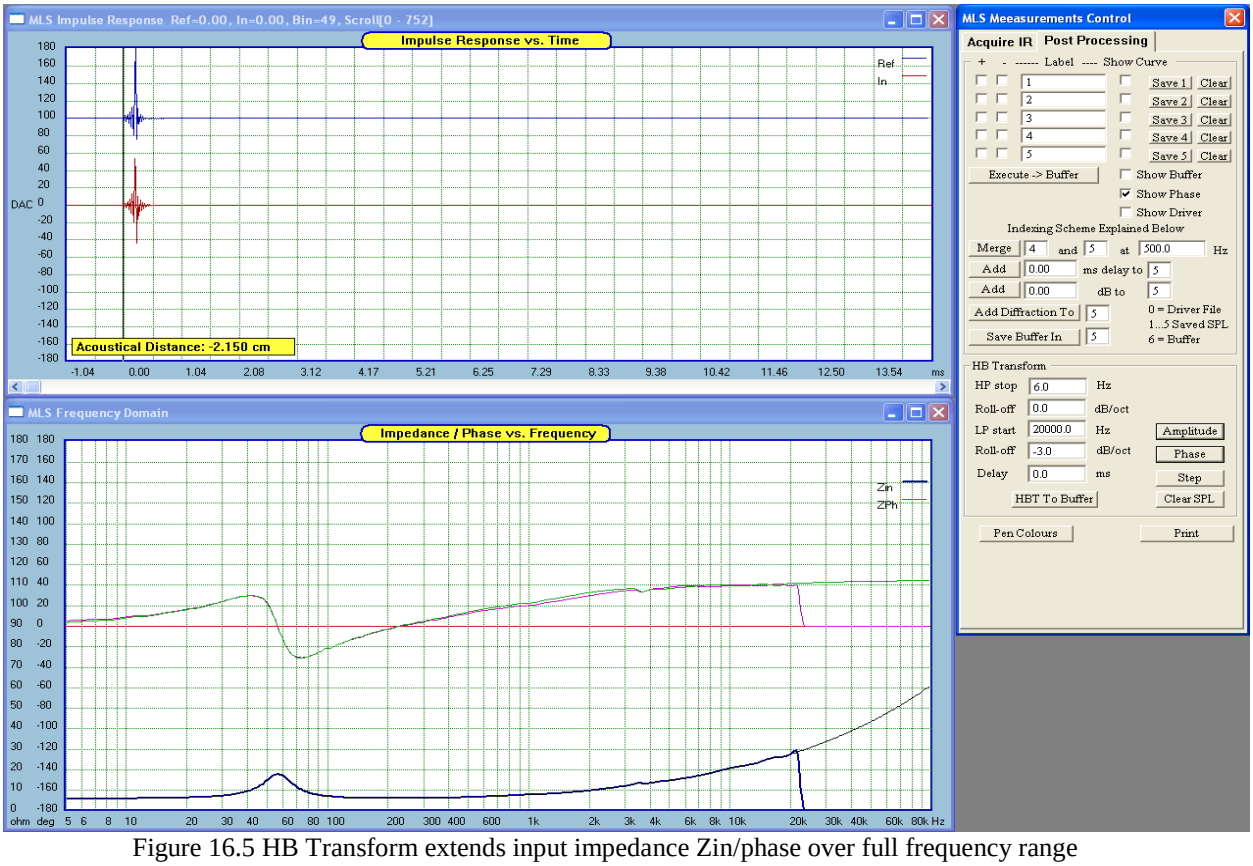


Figure 16.5 HB Transform extends input impedance Zin/phase over full frequency range

MLS Measurement Screen Utilities – Post-Processing

MLS Measurement System Curve Arithmetics.

As the measurement and processing capabilities of the MLS system continue to grow, so is the demand for more controls on the dialogue box. As the result of this, the MLS dialogue box has been re-designed and now incorporates 8 tabs: **"MLS", "Quick SPL", "SPL/Zin","CSD","RLC","T/S","RT60", and "Cepstrum"** TABs.. Various curve arithmetic can be performed on 5 curves. This includes:

- Adding / subtracting up to five curves at a time.
- Adding time delay to a curve.
- Adding / subtracting X dBs to a curve. This moves the SPL curve up / down.
- Performing Hilbert-Bode Transform on a curve.
- Erasing individual curve.
- Merging two curves at a specific frequency.
- Saving the results of each operation in a buffer. Buffer content can be moved to any of the 5 curves data space.
- Add box diffraction to the curve.
- You can also try Matching Filter Technique.

Theoretically, you can process unlimited number of curves. For instance, the results of the processing of the first five curves are saved into the buffer and 4 new curves can be processed. The fifth curve is the old result from the buffer. You can repeat this process indefinitely, and use the MLS screen to make all IR measurements.

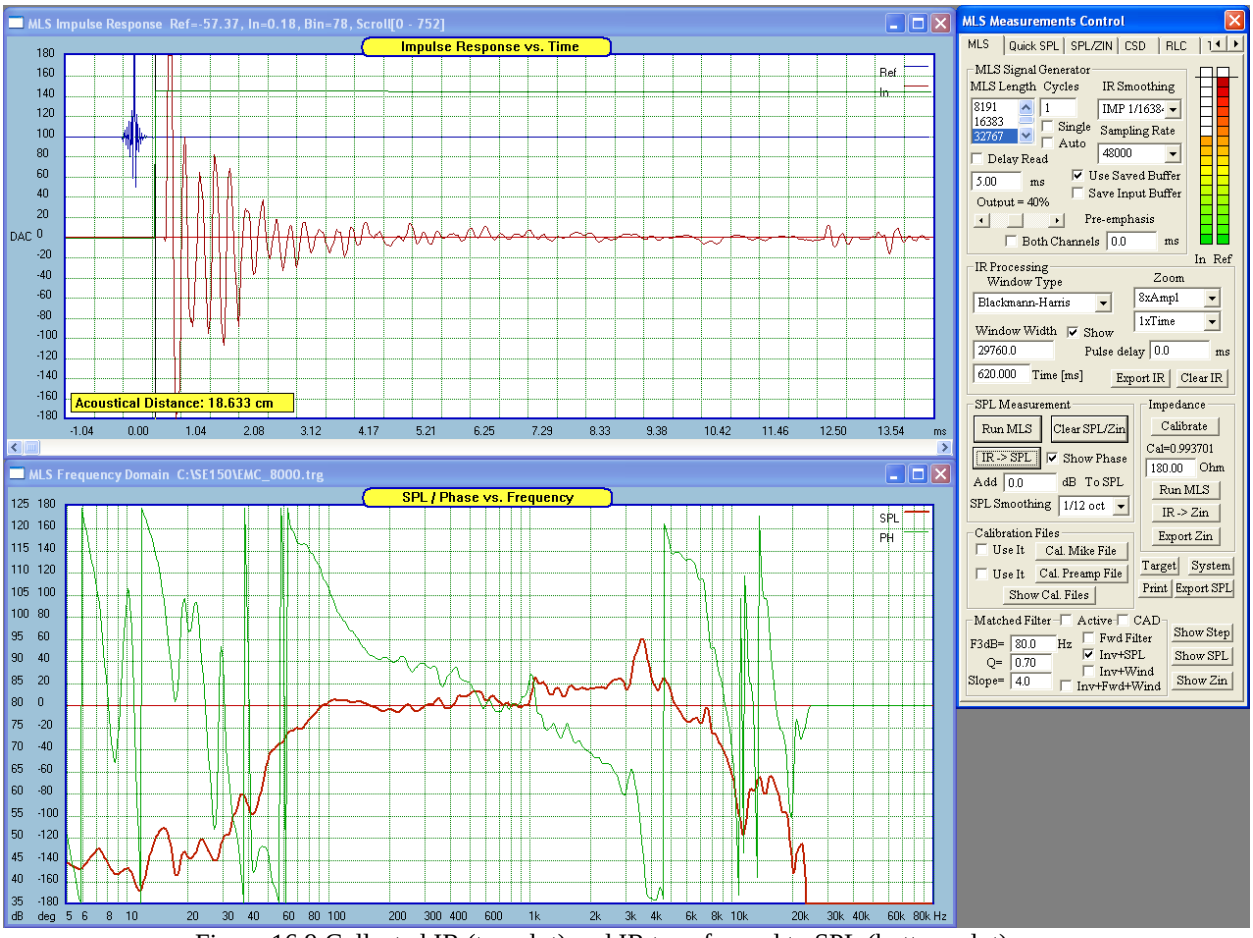


Figure 16.9 Collected IR (top plot) and IR transformed to SPL (bottom plot)

1. Measure SPL of the system under test using **"Acquire IR"** tab. It is not necessary to do this, but it would be recommended to save each individual measurement as an MLS buffer for future use. It is also useful, to label each measurement as a file name when saving the IR to a file. This will allow you to "mix-and-match" measurement results as you wish.

2. Load each MLS buffer file into MLS screen and perform **"IR->SPL"** transformation. Store each resulting SPL curve into the five "Post-processing" locations, using **"SPL/Zin"** TAB. When all required curves have been stored, perform required arithmetic. Up to five files can be included in the process at one time. You can engage more curves by performing the above process sequentially.

   Perform MLS measurement (or load saved MLS buffer into the program).
   Press **"IR -> SPL"** button to perform Inverse Fourier Transform on the Impulse response.
   Go to **"Post Processing"** tab.
   Press "Save 1" button. This will save SPL and phase data into curve 1 data space.
   Go to Step 2.1 and repeat the IFFT process, but this time press "Save 2" button on **"SPL/Zin"** TAB.
   Up to 5 curves can be stored this way.

Each curve can be labeled using a short name entered under the "Label" column on the **"SPL/Zin"** TAB. You will notice, that a miniature curve appears in the dialogue box. The curve can be displayed with different colors. You can select the color of each of the processed curves by pressing the **"Pen Color"** button on the **"SPL/Zin"** TAB BEFORE you press "Save X" button. A dialogue box will be opened, where you can select the desired curve color.

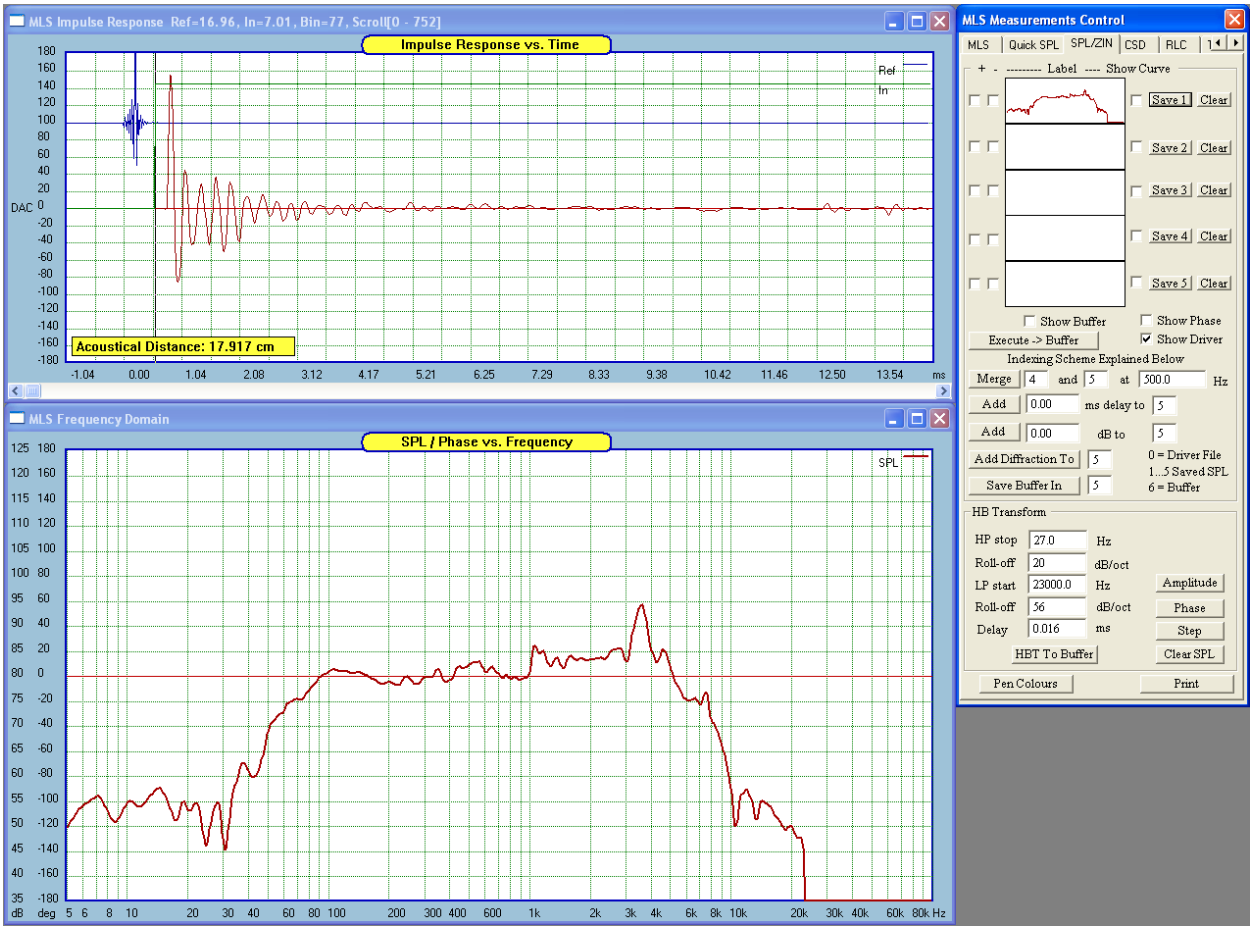


Figure 16.10 SPL (bottom plot) has the same color as the label in Post Processing TAB.

The "Post Processing" tab has several check boxes for you to select, to indicate what function is to be performed. There are **"+"** and **"-"** check boxes to indicate that the selected curves are to be added or subtracted.

There are also **"Show Curve"** check boxes to indicate which curve is to be displayed, and additional options for showing phase together with SPL for all curves.

All the above mathematical and graphical operations are executed when **"Execute -> Buffer"** button is pressed. Please note, that the results are always stored in the buffer data space. You can display the result of the operation, if the **"Show Buffer"** check box is selected.

Processing of the curves is based on a very simple indexing scheme:

0 - is the index of the original SPL/phase measurement result, which has also automatically been copied to the driver's data space, when pressing **"IR->SPL"** button on the "IR Acquire" tab. This way, it will appear in the Driver Editor screen and DRE screen.
1….5 - are the indexes of saved curves.
6 – is the index of data in the buffer.

Example 1: Adding driver and vent SPL measurement results.

- Measure driver's SPL using close-mike technique and save the result in Curve 1
- Measure vent's SPL using close-mike technique and save the result in Curve 2.
- Check "+" box and "Show Curve" box against Curve 1.
- Check "+" box and "Show Curve" box against Curve 2.
- Check "Show Buffer" box to show the result.
- Press **"Execute -> Buffer"** button.
- Enter '0' and Select **"Save Buffer In"** button to move the results to driver's file.

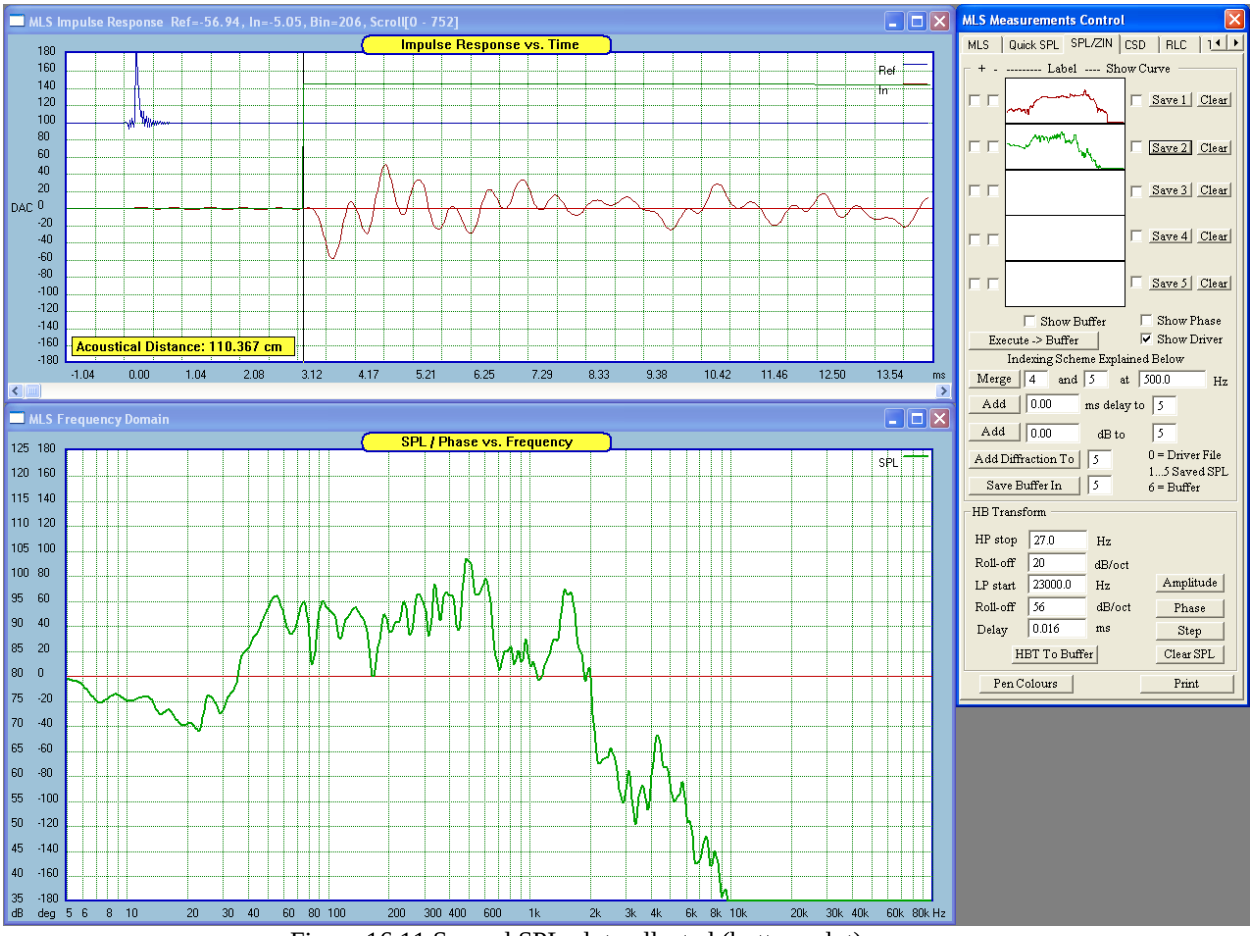


Figure 16.11 Second SPL plot collected (bottom plot).

Example 2: Merging curves.

- Measure SPL 1 and save the result in Curve 1
- Measure SPL 2 and save the result in Curve 2.
- Check “Show Curve” box against Curve 1.
- Check “Show Curve” box against Curve 2.
- Check “Show Buffer“ box to show the result.
- Enter “1” and “2” and desired merge frequency.
- Press **“Merge”** button.
- Enter ‘0’ and Select **“Save Buffer In”** button to move the results to driver’s file.

In the following simple graphical example, **red SPL curve** from Figure XXX was added to **blue SPL curve** from Figure YYY and this produced the **pink SPL curve** on Figure ZZZ. The pen color was changed from red -> blue -> pink.

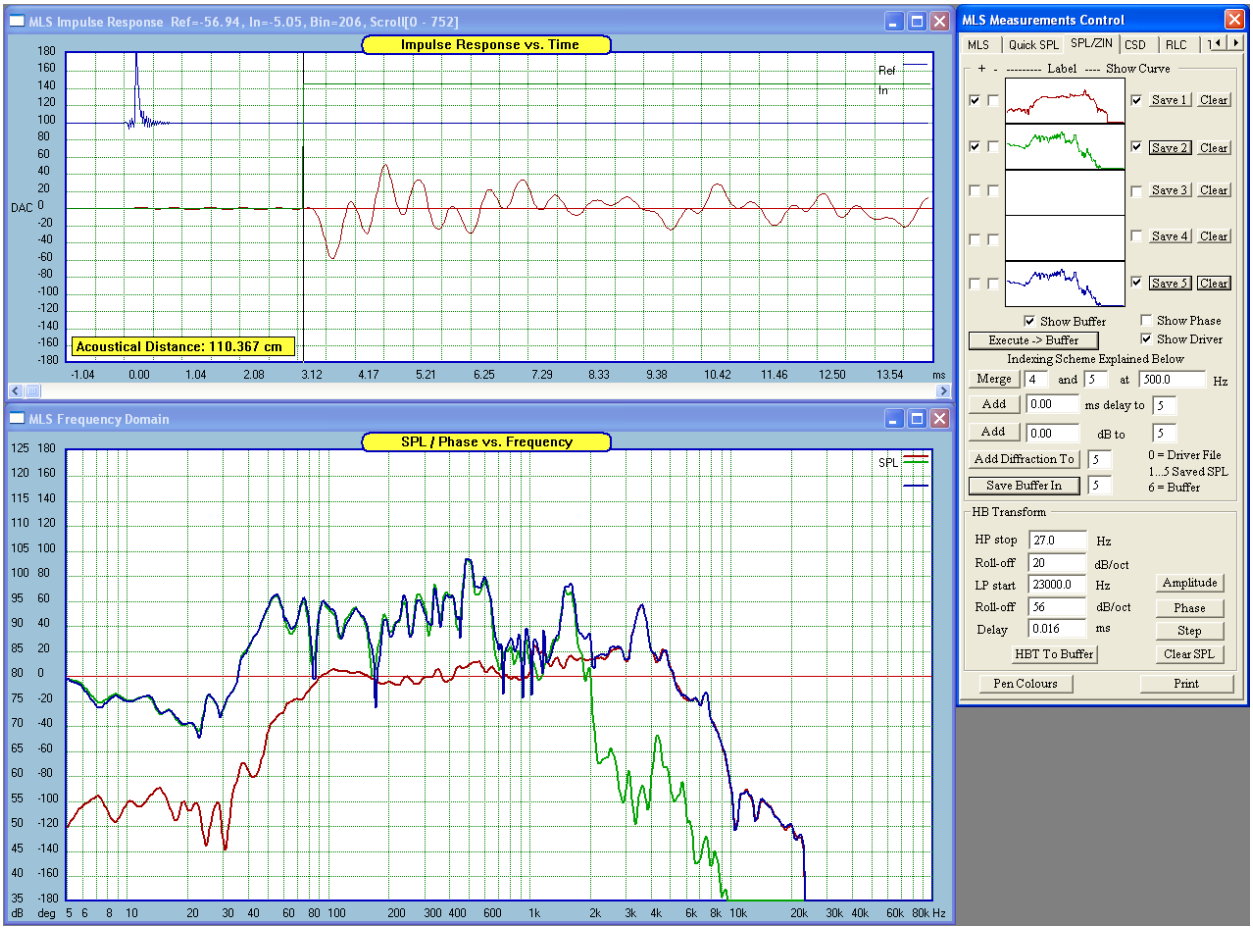


Figure 16.12. SPL (bottom plot) from Fig 10 + 11 added together (blue plot).

Please note, that you can move the results of Curve Arithmetic from the **Buffer to the Left / Right / Both channels of the Digital Room Equalizer**. First, please select where you want the results to go, by selecting appropriate check box on the **DRE** control dialogue box. Then select **“Save Buffer”** to **Curve 0**, on the MLS “Post-Processing” tab. This actions will save the Buffer content into driver data space (Curve 0) and also into the DRE data space, as requested from DRE dialogue box.

Curve Arithmetic Group

1. **"+" , "-"** – Column of check boxes to select arithmetical operation on the curve. "+" means, that the curve will be added, and "-" means, that the curve will be subtracted vectorally. Both check boxes blank indicate, that the curve will be disregarded.
2. **"Label"** – Column of 5 labels to use with the curves.
3. **"Show Curve"** – Column of 5 check boxes to select which curve will be displayed on the screen.
4. **"Save X"** – Column of save buttons to store the last SPL curve, generated on "Acquire IR" TAB.
5. **"Clear"** – Press to clear previously saved curve.
6. **"Execute -> Buffer" –** This button executes the +/- operations and saves the result in the buffer.
7. **"Show Buffer" –** Check this box to display current curve in the buffer.
8. **"Show Phase" –** Check this box to display phase on ALL plots.
9. **"Show Driver" –** Check this box to display currently available driver SPL/phase.
10. **"Merge" –** Button and data fields to merge two selected curves at the desired frequency.
11. **"Add" –** Button and data fields to add X.XX ms of delay to nominated curve.
12. **"Add" –** Button and data fields to add X.XX dBs to nominated curve.
13. **"Add Diffraction" –** Button and data field to add diffraction to the nominated curve.
14. **"Save Buffer In" –** Button and data field to move buffer content to the nominated curve data space.

HBT Group

1. **"HP stop"** – Enter the corner frequency for low frequency tail.
2. **"Roll-off" -** Enter the slope of the low-end tail.
3. **"LP Start"** – Enter the start frequency for the high end tail.
4. **"Roll-**off" – Enter the slope for the high-end tail.
5. **"Delay"** – enter additional delay in ms.
6. **"Amplitude"** – Button to display SPL of the HBT.
7. **"Phase" -** Button to display phase of the HBT.
8. **"Step"** – Button to display step response calculated from HBT.
9. **"Clear SPL"** – Clears SPL/Phase screen.
10. **"Pen Colors"** – Press to invoke Pen Color dialogue box. Please remember to select the colour of the plotted curves and labels BEFORE you plot or save the curve.
11. **"Print"** – Simply prints the screen.

Microphone Calibration File

Microphone calibration file could look as the example below:

```
Lsp-LAB:Compensation File v1.10
00128.MCP
318
BEGIN.
200.0000  1.5773  0.0000
202.9091  1.5830  0.0370
…
…
19740.2988 -3.5650 -134.3056
20000.0000 -3.7538 -136.2404
END
```

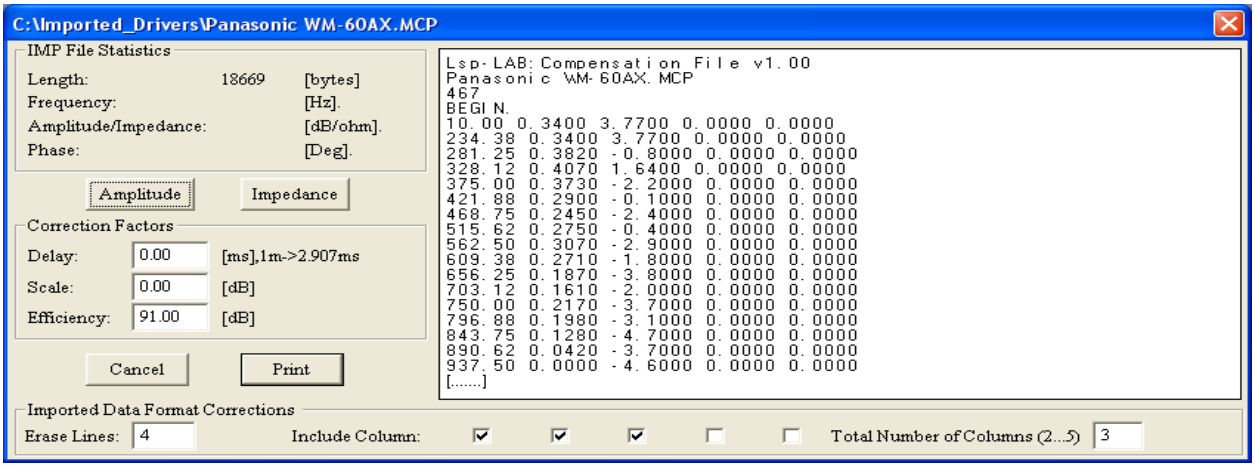


Figure 16.14 Importing microphone calibration file

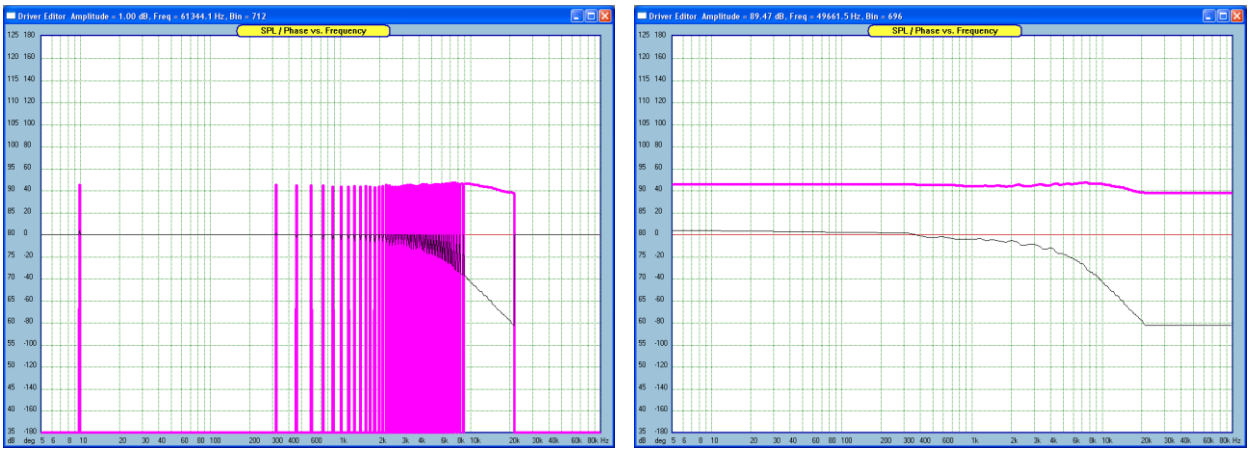


Figure 16.15 Imported and "Repaired" microphone calibration file.

Next, import the file into SoundEasy using "Import IMP" files as the option. The program will ask you to enter reference efficiency and you can enter say, 90.0dB to shift the relative SPL values in dB somewhere around 90dB display area of the screen. The 90.0 dB will be automatically subtracted when the fully edited microphone calibration file is imported into MLS or Analog measurement system screen. The file imported into SoundEasy may look like the one shown on the example below. As you can see, the file in its current state is not of much use and needs to be further processed. In the next step, you should press the **"Repair Amplitude"** button, resulting in display as shown above. This fractional data will be used by the Hilbert-Bode Transformer to re-create full SPL/phase information over the whole frequency spectrum. Please invoke the "Transfer Function" screen and select HB-Transform parameters for the best match with the "template" data. In the example below, the low-frequency slope has 0.0dB/oct roll-off and the high-frequency slope has 26.0dB/oct roll-off.

Now, Perform Hilber-Bode Transform and EXPORT this file as, say "XXXXXX_Corrected.txt" file. The SE exported file should look as below and **contain 750 lines – you can easily check for this using a text editor. It is important to use the microphone calibration file exported by SoundEasy. The MLS and Analog measurement screens are expecting this particular format and the 750 fully valid data points in the file.**

Example of Microphone calibration file exported by SoundEasy.

```
5.00     92.59     -0.04
5.07     92.59     -0.04
5.13     92.59     -0.04
5.20     92.59     -0.04
5.27     92.59     -0.04
….
….
93610.93     78.82     60.31
94855.23     78.74     52.35
96116.07     78.65     44.28
97393.66     78.57     36.11
98688.24     78.48     27.83
```

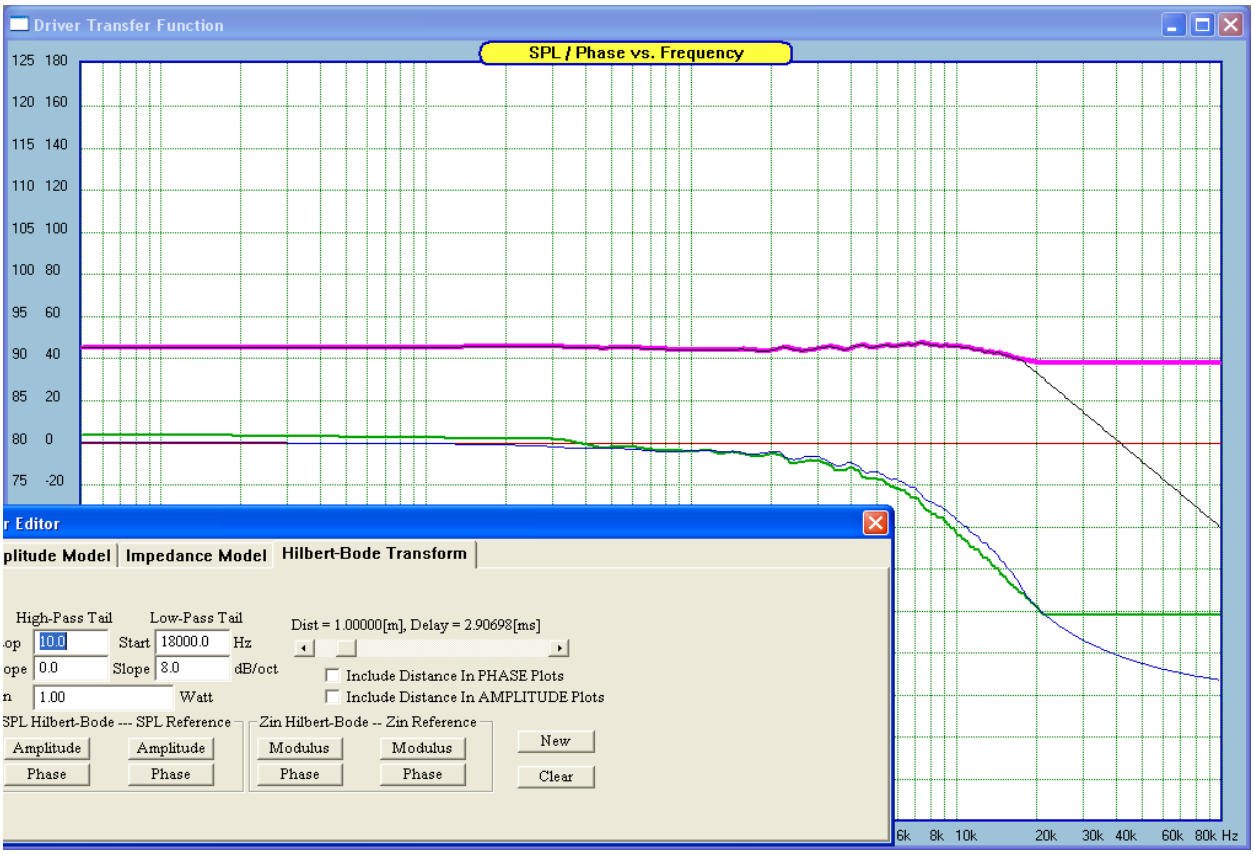


Figure 16.16 Fully edited microphone calibration file.

Select "Cal.Mike" button on the Analog or MLS screen to import the "XXXXXX_Corrected.txt" file. The actual curves will not be shown on the screen, but this amplitude response and phase response will be used for correcting ALL SPL measurements.

Exporting Data

Exporting measured SPL and impedance data from analog and digital measurement screens directly from the measurement screen can be accomplished by pressing "**Export SPL**" (creates .frd file) and "**Export Zin**" (creates .zma file) buttons. Data is exported "as you see" it on the screen.

Spectrum Analyzer/Signal Generator

Spectrum Analyzer screen is intended to show the frequency content of the audio data currently entering your sound card. Here are some of the uses for an audio spectrum analyzer:

1. Analyze the frequency characteristics of live/recorded music.
2. Analyze distortion (harmonic content) of audio equipment.
3. Analyze room acoustics using also the built-in pink noise generator.

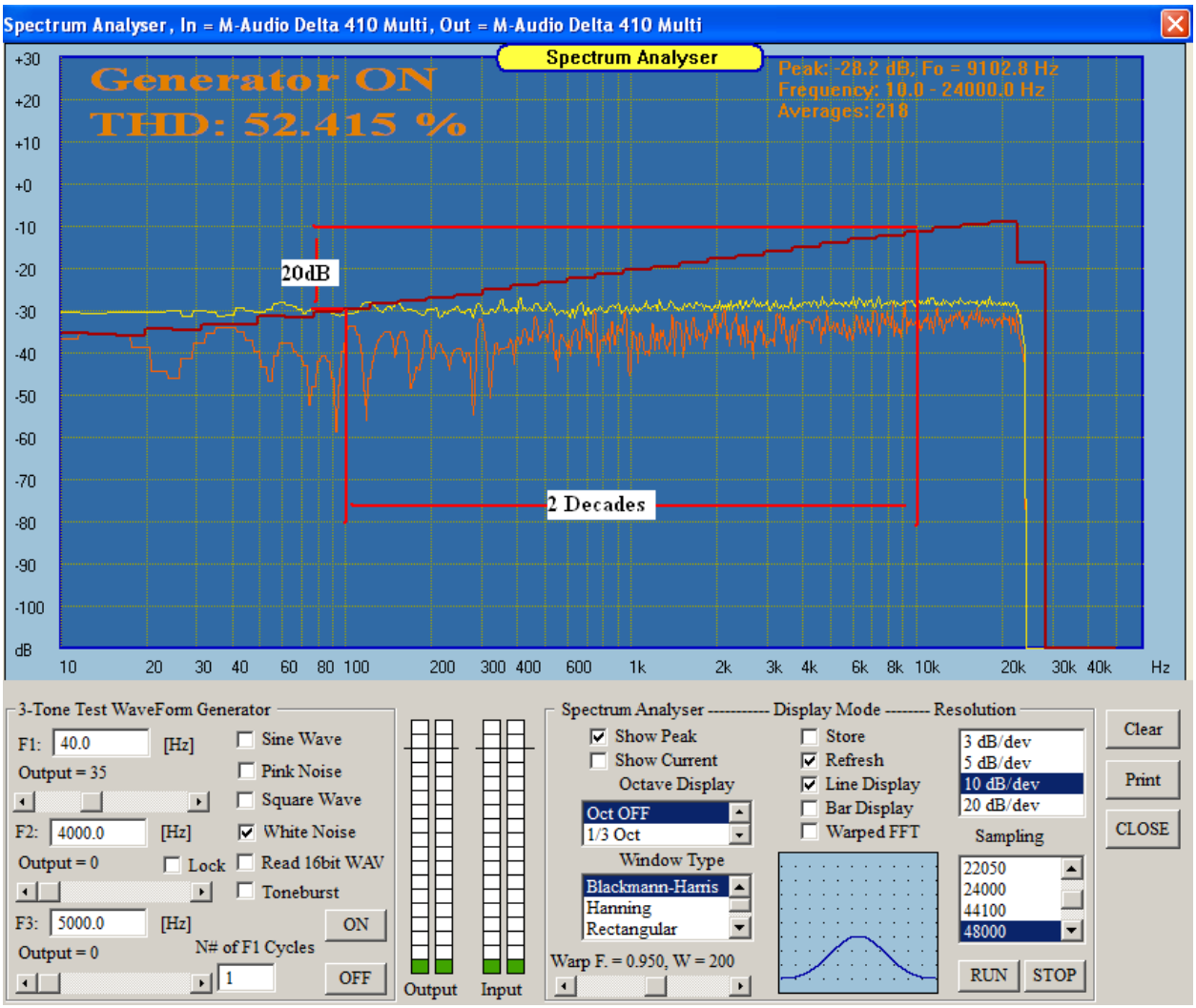


Figure 16.17 EasyLab spectrum analyzer screen showing characteristics of White Noise.

Another very useful feature is the built-in Pink Noise generator. The characteristics of the pink noise call for a spectrum drop towards higher end of audio range with a rate of 10dB/decade. The noise is a very useful tool in room acoustic analysis, showing room resonance and room transfer function for different speaker/microphone locations. The preferred use of this technique would be to click on “Show Average” check box to allow a number of samples (sweeps) to accumulate over time. Additional generators include White Noise, Square Wave, Tone Burst and Two-Tone Generator signals.

Total Harmonic Distortion, THD parameter, is calculated in reference to the fundamental frequency as entered in the Test Waveform Generator. If you use the internal generator for this measurement, the fundamental frequency of the signal is given automatically by generator. However, if you intend to measure THD on an externally generated signal, the program will track the fundamental frequency of your external signal generator’s frequency. The THD measurement is performed automatically on all input signals and the resolution is 0.01%.

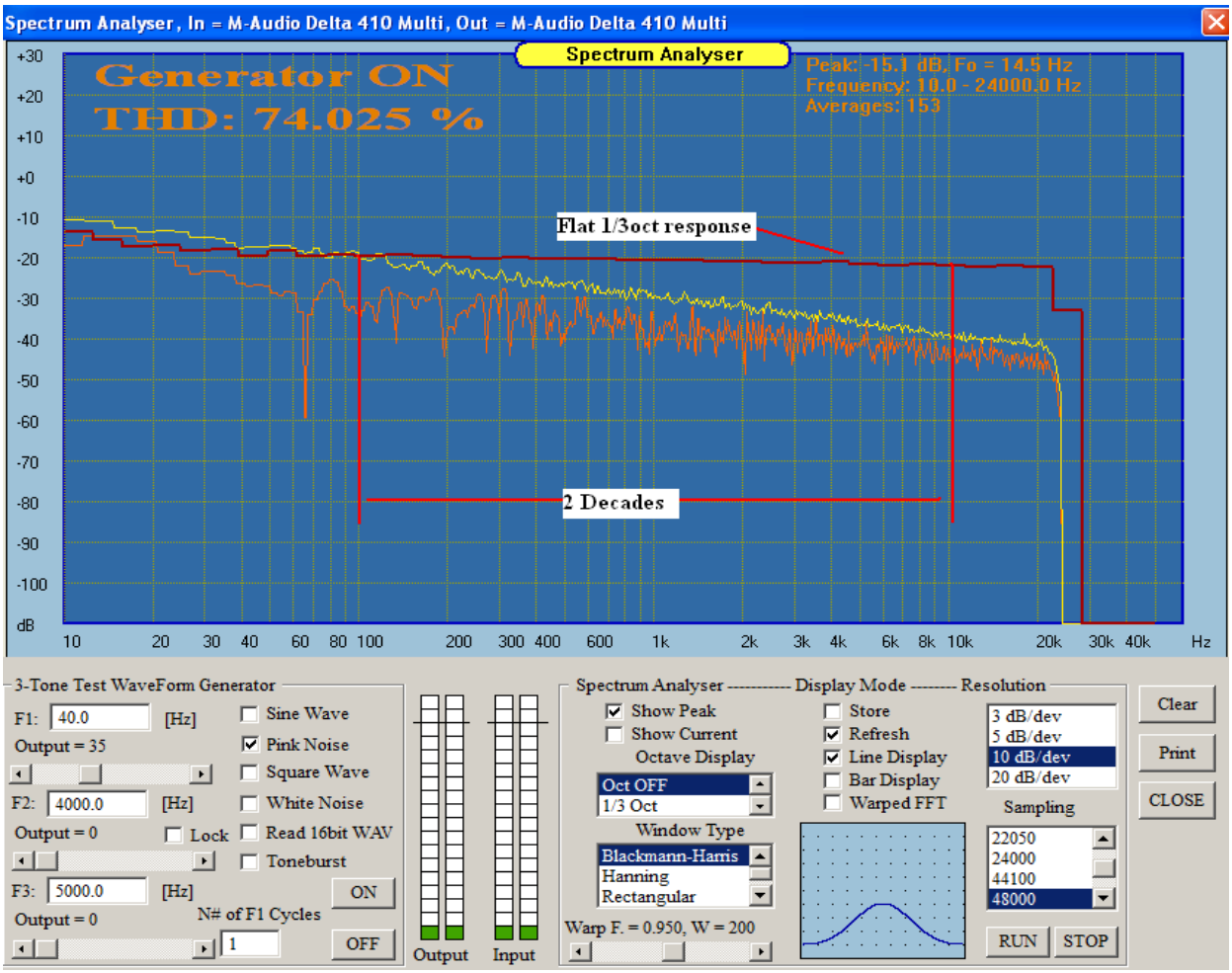


Figure 16.18. EasyLab spectrum analyzer with Pink Noise.

Test Waveform Generator Controls

The device has the following controls:

1. **Frequency** 1 data entry field – type-in the desired frequency of your test signal's sine wave.
2. **Output** level slider – use this control to adjust the voltage level out of your sound card.
3. **Frequency 2** data entry field - type-in the desired frequency of your second test signal's sine wave.
4. **Output** level slider - use this control to adjust the voltage level of the second signal out of your sound card
5. **Frequency 3** data entry field - type-in the desired frequency of your third test signal's sine wave.
6. **Output** level slider - use this control to adjust the voltage level of the third signal out of your sound card
7. 2'nd + 3'rd Generators – **"Lock To F1"** By selecting this option, you will lock F2 Generator and F3 Generator to frequency F1, so the total output looks as follows:
   F1 Generator runs on F1 frequency, F2 Generator runs on F1 + F2 frequency
   F3 Generator runs on F1 + F3 frequency
8. **"Number of Cycles"** in the burst for ToneBurst-type of signals.
9. **Sine Wave** – click on this check box to generate sine wave out of the sound card.
10. **Pink Noise** - click on this check box to generate pink noise out of the sound card.
11. **Square Wave** - click on this check box to generate square wave out of the sound card.
12. **White Noise** - click on this check box to generate white noise out of the sound card.
13. **Read WAV File – This check fiels takes priority over the other checkboxes.** Use it to red 16-bit CD WAV files.

14. **ToneBurst -** check box to generate burst of tone (F1) with number of cycles out of the sound card
15. **ON** button – click on this button to start signal generator.
16. **OFF** button – click on this button to turn the generator off.
17. **Input Level Indicator** – please observe the input level into the soundcard, so that **the signal does not get clipped.**

Spectrum Analyzer controls

1. **RUN** button – press this button to start spectrum analysis.
2. **Stop** button – press this button to stop spectrum analysis.
3. **Show Peak** - click on this check box to display averaged results of several sweeps.
4. **Show Current**. - click on this check box to display only current result of the sweep.
5. **Oct OFF list box** – select resolution 1/3, 1/6, 1/12, 1/24 or OFF of the sweep (black plot).
6. **Window Type** – You can select several FFT windowing options.
7. **Warped FFT"** – Selects warped FFT option – see "**Frequency Warped FFT" chapter.**
8. **Warp Factor** – Can be adjusted if "Warped FFT" was selected.
9. **Store** - click on this check box to display only current result of the sweep.
10. **Refresh** - click on this check box to display only current result of the sweep.
11. **Line Display** - click on this check box to display only current result of the sweep.
12. **Bar Display** - click on this check box to display only current result of the sweep.
13. **Resolution** (vertical) - click on this check box to display only current result of the sweep.

Dual Channels Oscilloscope/Signal Generator

Oscilloscope is a very handy device for analyzing signals in time domain. There is an array of controls provided for optimizing the quality of the display of the analyzed signal. You can use single channel mode or take advantage of the dual mode capabilities. Synchronization of the input signals is necessary for stable display and includes "slope" triggering and selection of synch channels.

The built-in signal generator provides you with ready-to-use test signals and all controls are on the same screen, so you do not need to browse the menu each time you need to change your test signal. **However, you do need to exercise caution when making live tests on your loudspeakers. You may accidentally use a test signal that will destroy the driver with too much power or a wrong spectrum.**

Test Waveform Generator Controls

1. **Frequency** 1 data entry field – type-in the desired frequency of your test signal's sine wave.
2. **Output** level slider – use this control to adjust the voltage level out of your sound card.
3. **Frequency 2** data entry field - type-in the desired frequency of your second test signal's sine wave.
4. **Output** level slider - use this control to adjust the voltage level of the second signal out of your sound card
5. **Frequency 3** data entry field - type-in the desired frequency of your third test signal's sine wave.
6. **Output** level slider - use this control to adjust the voltage level of the third signal out of your sound card
7. 2'nd + 3'rd Generators – **"Lock To F1"** By selecting this option, you will lock F2 Generator and F3 Generator to frequency F1, so the total output looks as follows:

F1 Generator runs on F1 frequency, F2 Generator runs on F1 + F2 frequency
F3 Generator runs on F1 + F3 frequency

8. **"Number of Cycles"** in the burst for ToneBurst-type of signals.
9. **Sine Wave** – click on this check box to generate sine wave out of the sound card.
10. **Pink Noise** - click on this check box to generate pink noise out of the sound card.
11. **Square Wave** - click on this check box to generate square wave out of the sound card.
12. **White Noise** - click on this check box to generate white noise out of the sound card.
13. **Read WAV File – This check field takes priority over the other checkboxes.** Use it to red 16-bit CD WAV files.
14. **ToneBurst -** click on this check box to generate burst of tone (F1) with number of cycles (F2) out of the sound card
15. **ON** button – click on this button to start signal generator.
16. **OFF** button – click on this button to turn the generator off. You should turn the generator OFF before closing the Spectrum Generator dialogue box.
17. **Input Level Indicator** – please observe the input level into the soundcard, so that **the signal does not get clipped.**

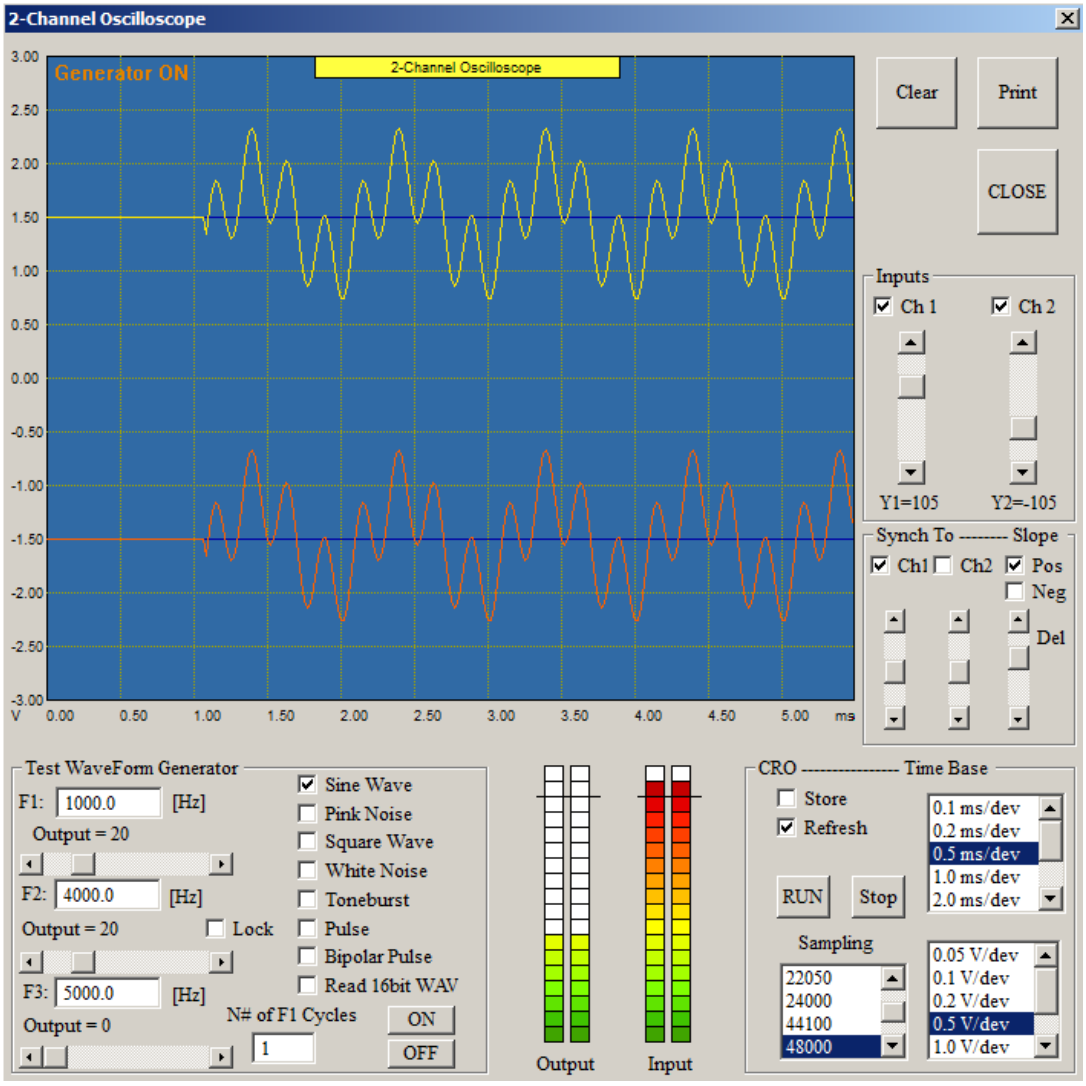


Figure 16.19. EasyLab Oscilloscope screen – Two-Tone signal 1000Hz + 40000Hz.

18. **Sampling list box –** use the selection provided to switch the sound card to the desired sampling frequency

Oscilloscope Screen Controls

1. **RUN** button – press this button to start display analysis.
2. **Stop** button – press this button to stop display analysis.
3. **Store** - click on this check box to display only current result of the sweep.
4. **Refresh** - click on this check box to display only current result of the sweep.
5. **Time Base** – select desired time scale.
6. **Vert. Resolution** (vertical) – Select appropriate vertical gain.

Inputs

1. **CH – 1** – click to turn ON channel 1.
2. **CH – 2 –** click to turn ON channel 2.
3. **Vertical slider for CH –** 1 – use it to position/move Ground Line for CH 1 on the display screen.
4. **Vertical slider for CH –** 2 – use it to position/move Ground Line for CH 2 on the display screen.

Synchronization

1. **Synch To CH – 1** – click to synchronize display to channel 1. Synch Level is adjustable
2. **Synch To CH – 2** – click to synchronize display to channel 2. Synch Level is adjustable

Slope

1. **Pos** – click to synchronize display to positive (going through "0" from negative to positive voltage).
2. **Neg** - click to synchronize display to negative (going through "0" from positive to negative voltage).
3. **Delay** – Moves the displayed signal right, as per set delay

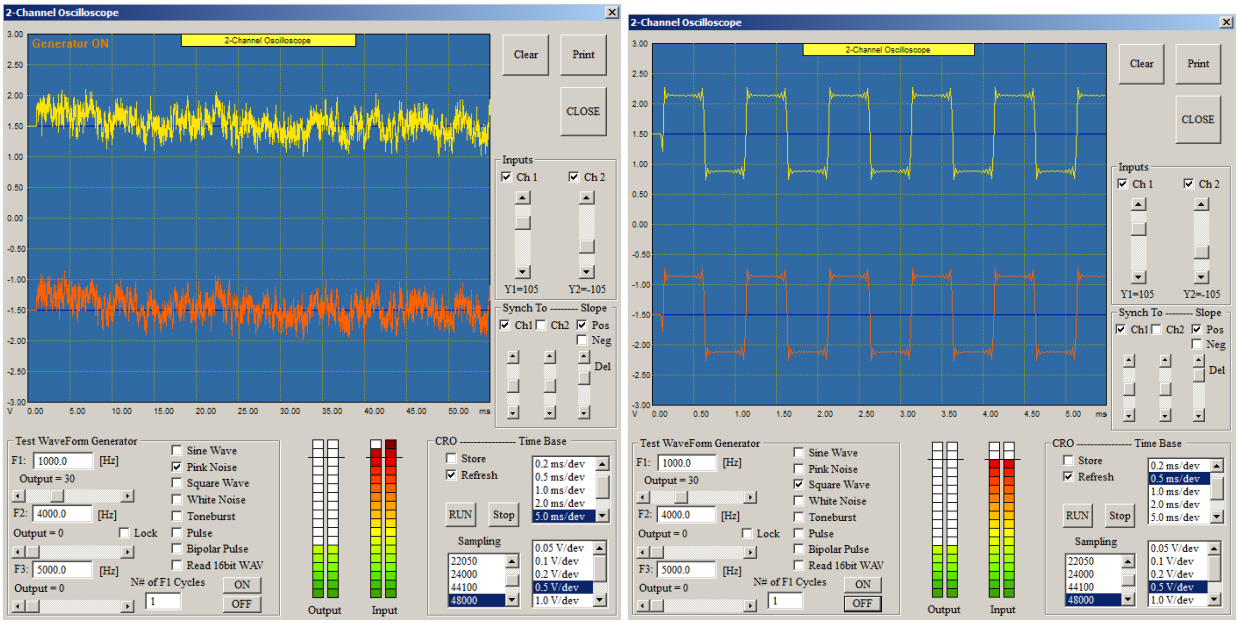


Figure 16.20. Pink Noise Generator Square Wave Generator

Additional Controls

1. **"Clear**" – clears the display.
2. **"Print"** – use this button to obtain a hard copy of the screen.
3. **"Done"** – closes the Oscilloscope screen.

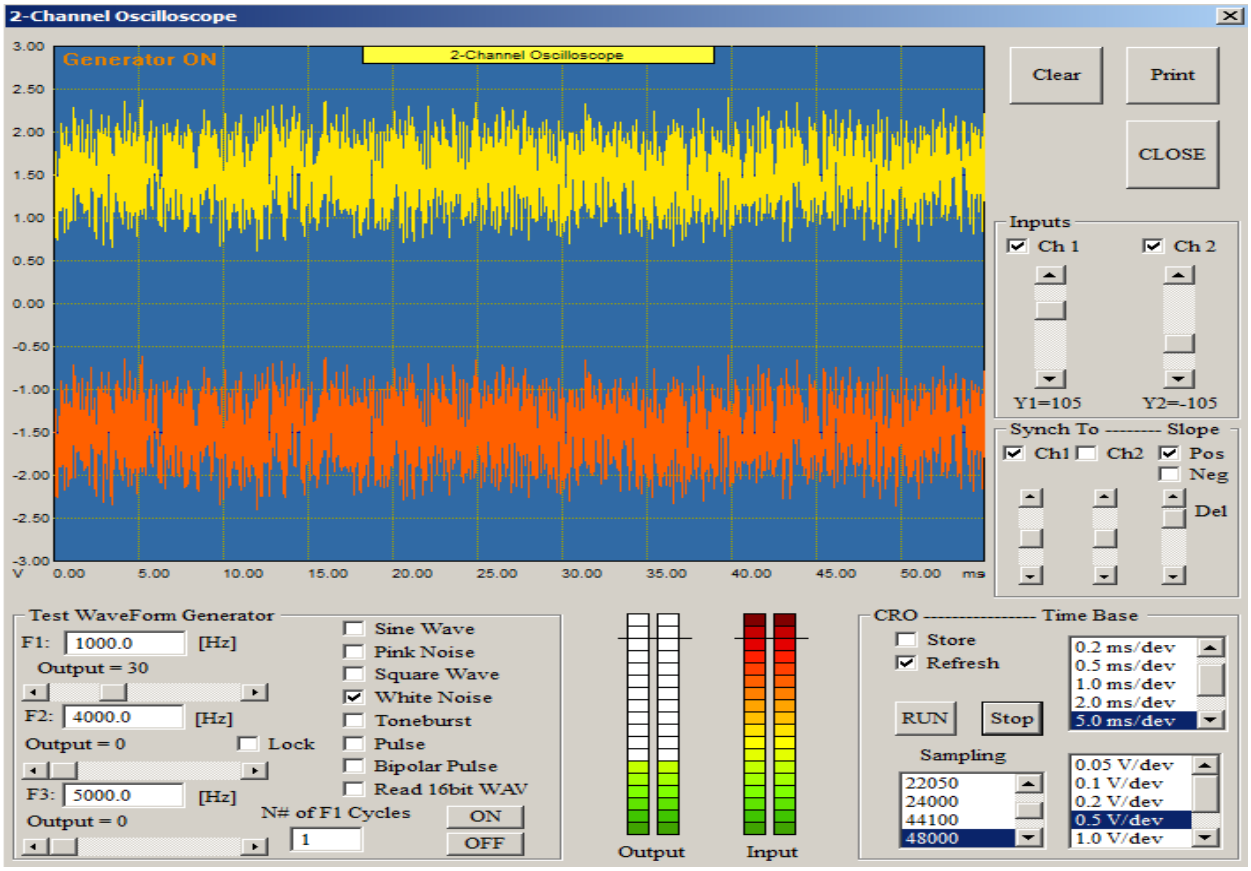


Figure 16.21 White Noise Generator

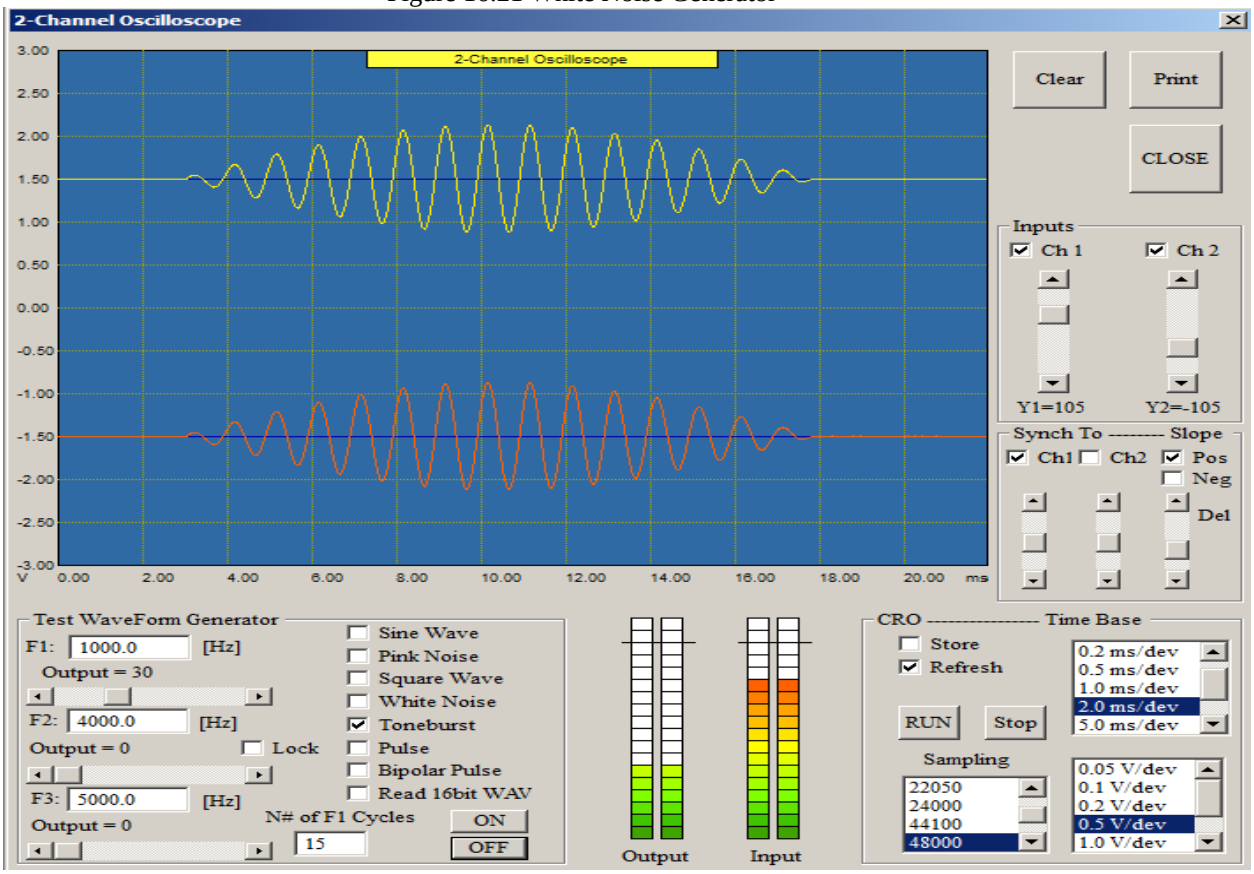


Fig 16.22. ToneBurst Generator.

Cumulative Spectral Decay and Tone Burst Waterfalls

Cumulative Spectral Decay plots are derived from impulse response. Therefore, the first step is to perform the classical MLS-type measurement on your loudspeaker and obtain the Impulse Response – see **Operation of the MLS Screen** section. Now you are ready to start decimating the impulse response to see what it is composed of. Each slice is then fed into the FFT to obtain corresponding frequency response. The first slice of the CSD plot (at zero delay) is the same as the standard frequency response. Successive slices are computed by shifting the windowed impulse response segment (as determined by the marker and cursor positions) right by a specified number of samples. This process is repeated many times generating the successive slices, which form the CSD plot. You can select the number of time points to shift between slices through the provided list box "**CSD Time Step**".

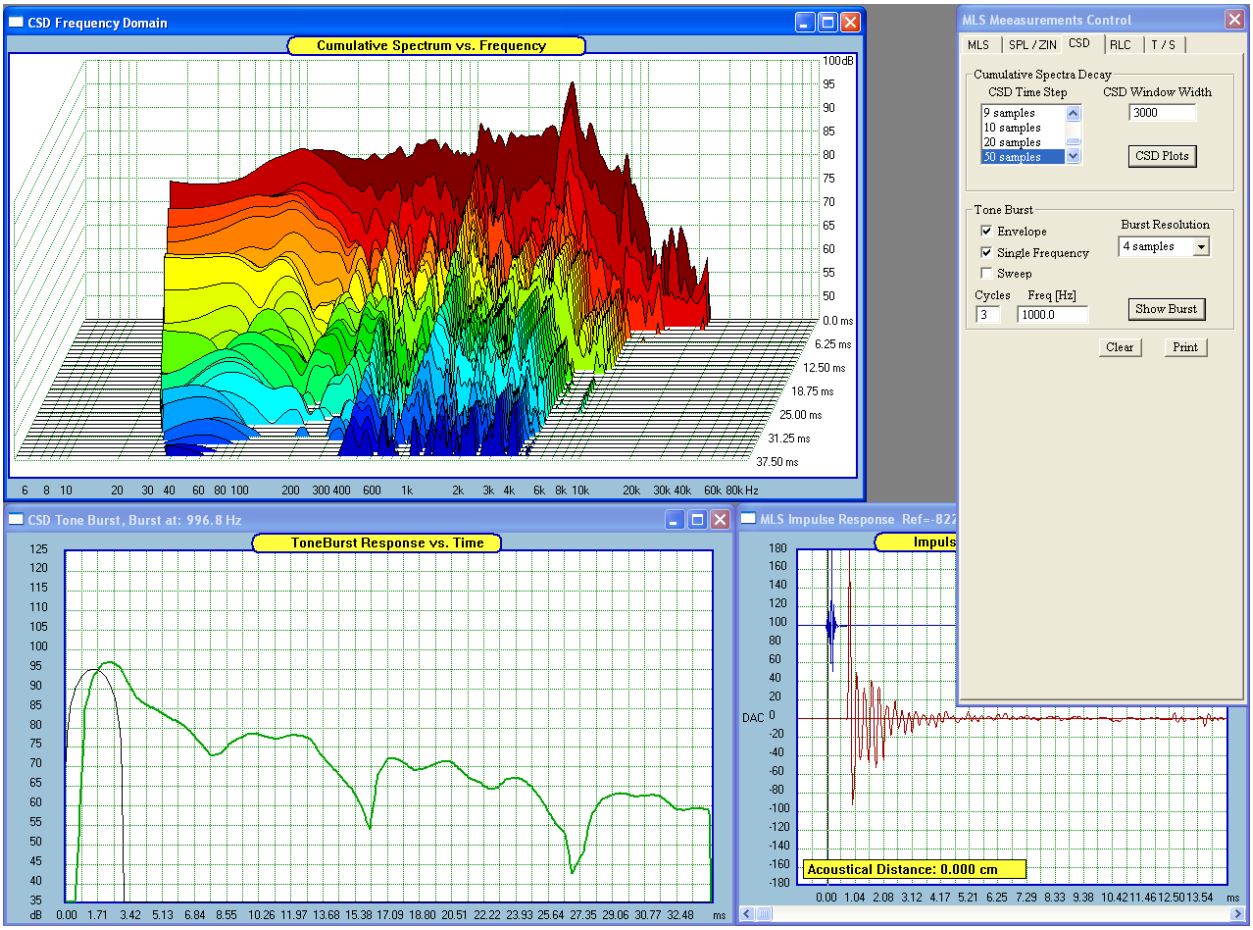


Figure 16.22. Impulse response ready to be decimated into CSD. 1000Hz ToneBurst response.

Impulse response of the loudspeaker under test is obtained the same way as you would perform MLS measurements. You need to get the hardware set-up ready and functional. Next, you would select the gating time window and smoothing option. If you need to remove the "time-of-flight" for the sound to reach the test microphone, please position the time cursor accordingly. Please also note, that SPL plot will not change when the time domain cursor is repositioned.

Finally, press the " CSD Plots" button to start the waterfall plots. CSD waterfall plots are frequently used to detect and visualize mechanical resonant characteristics in speaker cones, enclosures or horns. A resonance will manifest itself as a slowly decaying ridge along the time axis, due to its slow energy release over time. In other words, CSD plot shows you the frequency-decomposed decay. The basic idea behind the waterfalls involves using ever shorter time window to FFT process shorter and shorter section of the impulse response. This has profound consequences in the frequency resolution of each "slice".

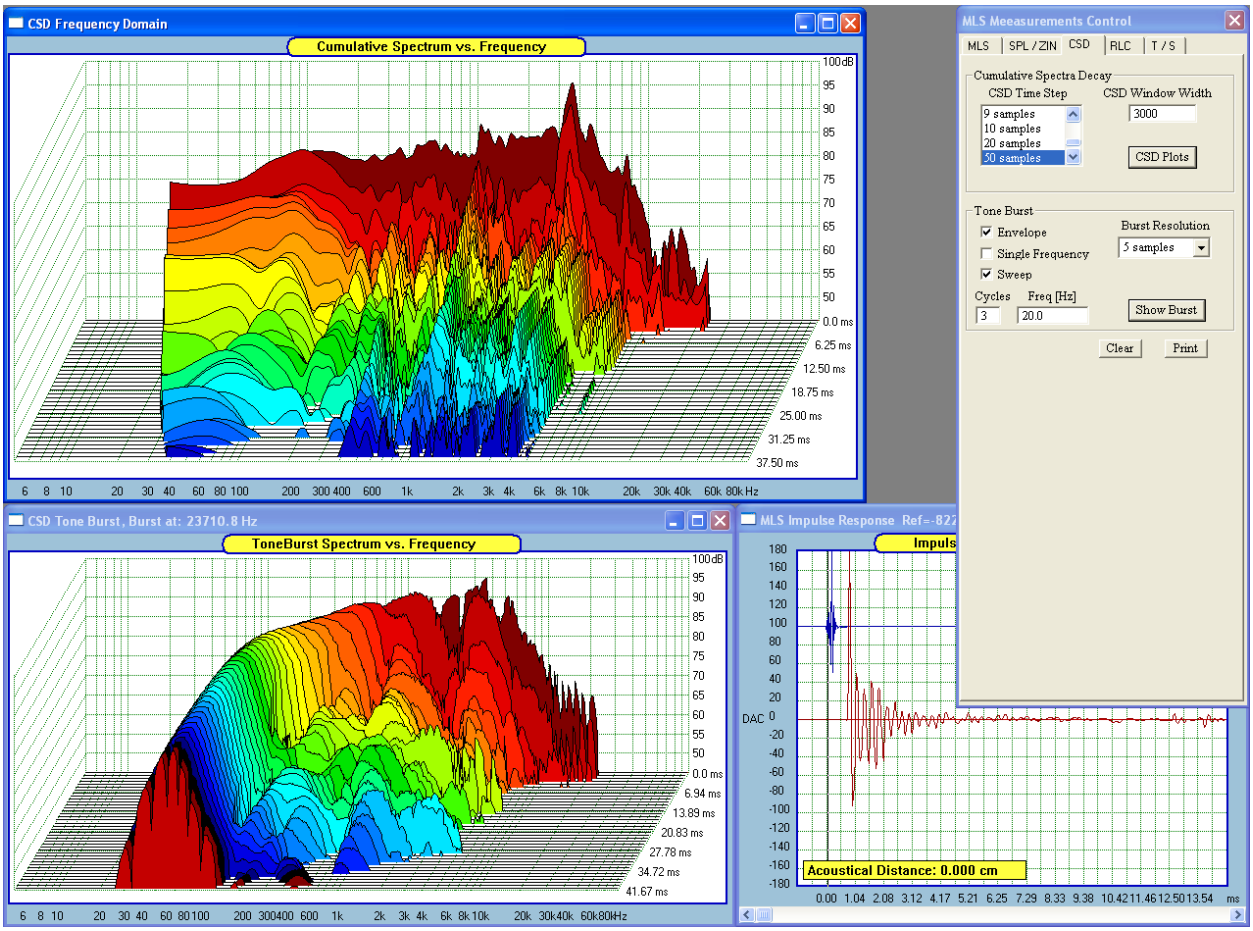


Figure 16.23. Example of CSD waterfalls AND ToneBurst waterfalls.

As the time window, T, gets shorter, the Fc = (sampling rate)/(window width) [Hz] gets larger and larger. Let's assume, that there are 25 "slices" to be computed and the "Time step" is selected as 20 samples. With the initial time window of 1000 samples and sampling rate of 48000 Hz, we have: **Fc1 = 48000/1000 = 48Hz resolution.** When the CSD process has finished, the time window has been reduced by 25 slices x 20 sample = 500 samples. **Fc2 = 48000/(1000-500) = 96Hz resolution.**

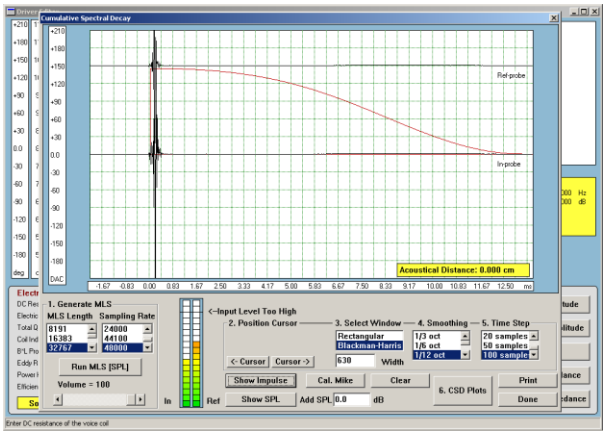
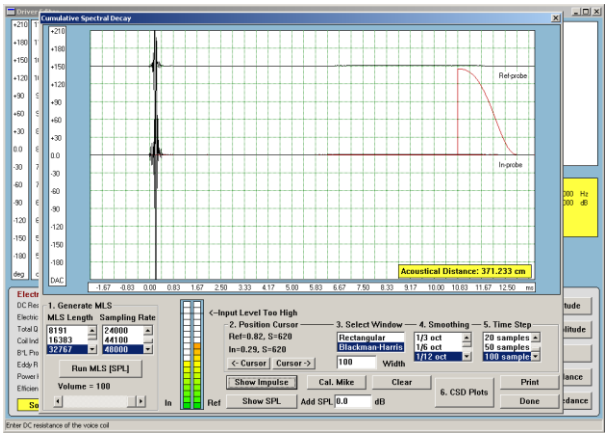


Figure 16.24 Time window at the start of the analysis



Time window at the end of the analysis

The resulting loss of resolution AND the lowest cut-off frequency is automatically accounted for in the program. In another example below, you can observe the effect of 3500 samples long initial time window progressively reduced by 36 slices and 100 samples/per step settings – the invalid portion of the CSD plots are simply not displayed.

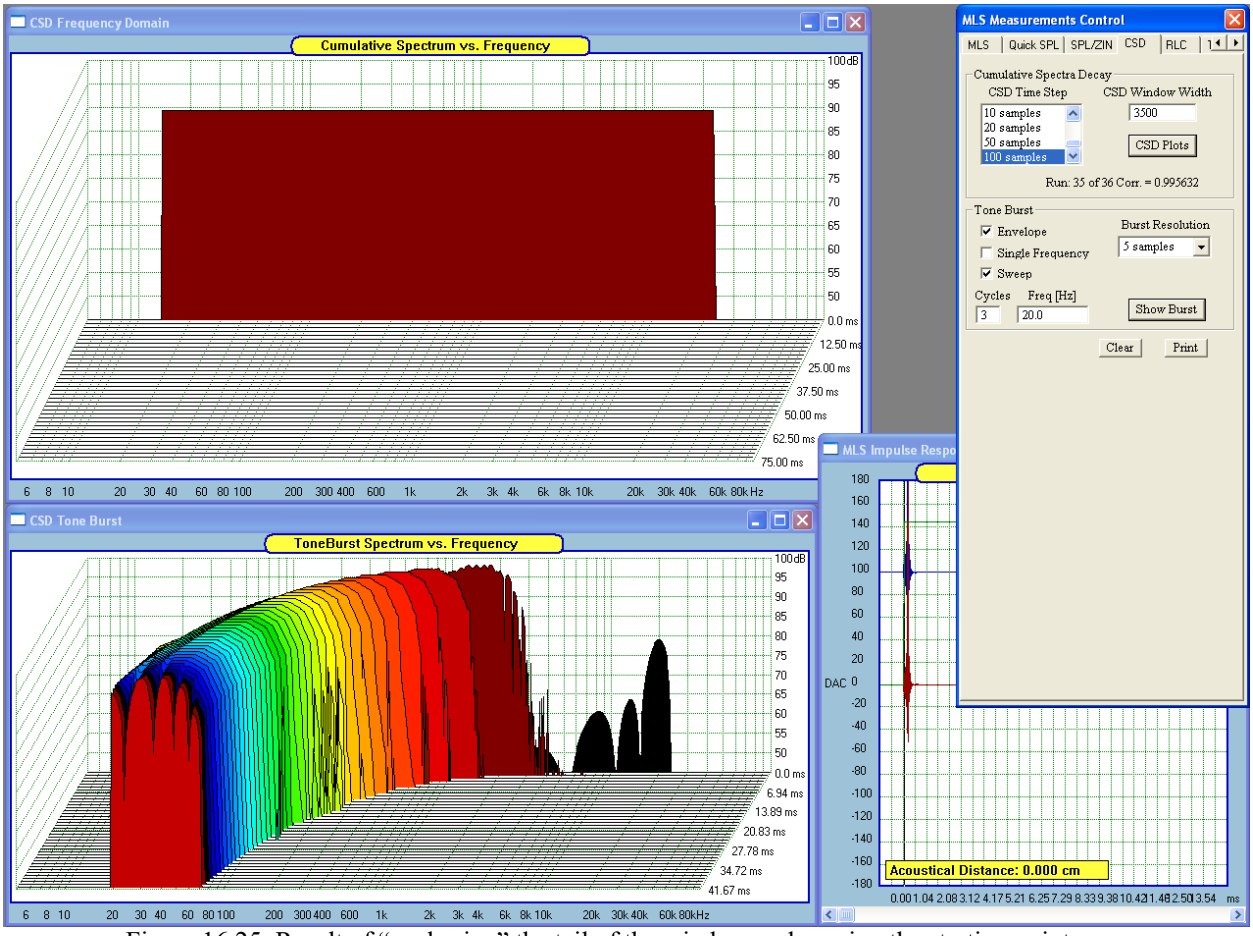


Figure 16.25. Result of "anchoring" the tail of the window and moving the starting point

Measurement results can be also saved into the nominated files and recall at the later time for post-processing. "Save Input Buffer" check box activated saving mechanism and "Use Saved Input Buffer" opens loading dialogue box.

When the "Cumulative Spectra" dialogue box is opened, you will notice a section in the middle of the box – see Figure 16.23. It's called "Tone Burst" and contains all controls necessary to perform tone-burst excitation analysis of the collected Impulse Response. You can generate tone-burst response at individual frequency or you can also generate waterfall-like family of responses, where the excitation was a tone burst of several cycles of each frequency. The responses were then assembled to resemble a waterfall-like display.

The measurement procedure is very similar to the standard CSD techniques. Firstly, you must collect the Impulse Response of your loudspeaker. The top section of the control box includes all standard controls for generating MLS signal and collecting the IR. The exact procedure was described already in this manual. If you need to use mike calibration file, please do so, using provided controls as well. Collected Impulse Response will be shown in the CSD Impulse response window.

We can now proceeded to the "Tone Burst" section:

1. **"Single Frequency"** – check this box to generate tone burst response for a selected frequency
2. **"Sweep"** – check this box to generate waterfall-like 3D display with a defined starting frequency.
3. **"Cycles"** – data field to enter number of cycles of in the tone burst.
4. **"Freq"** – enter frequency response of the tone burst, OR starting frequency for the Sweep mode.
5. **"Burst Res"** – Burst Resolution. Select the time scale using this control. For instance, if you select "5 samples", the program will display every 5-th sample of the convolution result. This way, you can zoom-in or –out on the time scale.
6. **"Burst"** – pressing this button activates calculation and display.

You will notice, that convolution calculations take much longer time at lower frequencies, therefore please select the starting frequency for the "Sweep" mode sensibly. In a way of an example, please collect the IR of your midrange driver and set the controls to:

1. **"Single Frequency"** – check this box.
2. **"Sweep"** – un-check this box
3. **"Cycles"** – enter number 3
4. **"Freq"** – enter 1000.
5. **"Burst Res"** – Select "5 samples"
6. **Press "Burst" button**. Now, see Figure 16.23 for example.

T/S (Thiele/Small) Parameters Extraction

Characterization of loudspeaker involves many parameters and can be quite involving task. There are small-signal parameters, large-signal parameters (temperature dependence, BL(x), Cas(x), Le(x), Xmax), frequency response, polar characteristics and so on.

In this section, we focus on a sub-section of this array of data that has become known as small-signal parameters or Thiele/Small parameters. Both researchers provided us with a good insight of the process of analysis and synthesis of the loudspeaker system based on those parameters.

It is perhaps worth pointing out, that Thiele's working model was a significant simplification. Consequently, following his published papers, more work went into expanding the model and making it more accurate. For instance, some research activities focused on more accurate representation of the voice coil impedance at higher frequencies. As a result, SoundEasy implements two impedance models. One model is an expanded and much more accurate, 4-stage LR version of the simple "single segment" representation of the "semi-inductor" in the voice coil. The second model represents somewhat different approach and involves "variable resistor" and variable inductor" proposal from J. Wright.

Basic small-signal parameters are:

Bl	-	Product of magnetic flux density, B and length of the voice coil wire in the magnetic gap
Cas	-	Acoustical compliance of the loudspeaker's suspension system
Cms	-	Mechanical compliance of the loudspeaker's suspension system
Fs	-	Resonant frequency of the driver in "free-air", including air load
Le	-	Inductance of the voice coil
Mms	-	Mechanical mass of the driver's vibrating system, including air load
Qms	-	Driver's Q-factor at resonance, considering only mechanical losses
Qes	-	Driver's Q-factor at resonance, considering only electrical losses
Qts	-	Driver's Q-factor at resonance, considering ALL losses
Re	-	DC resistance of the voice coil

Sd	-	Effective cone area (piston area)
Vas	-	Volume of air exhibiting the same compliance as the driver's suspension
η_0	-	Reference efficiency
SPL	-	Sound Pressure Level, usually measured for 1watt input power at 1 meter distance

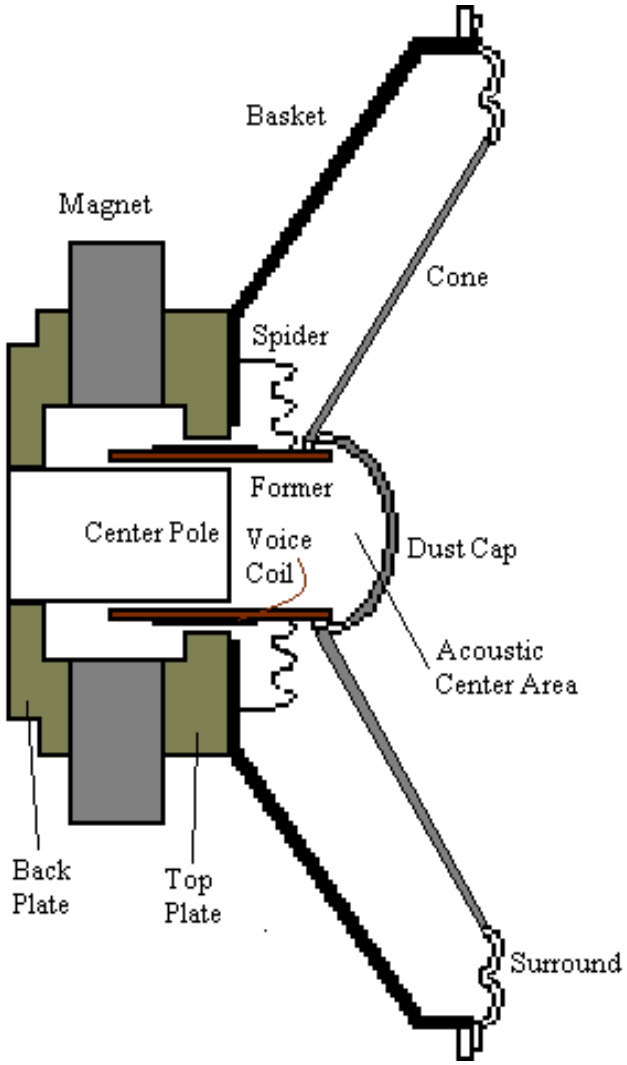


Figure 16.27. Cross-section view of the electro-dynamic loudspeaker

The surround + spider are commonly referred to as the "suspension system" and the magnet + the voice coil are referred to as the "motor system" of the loudspeaker. The "acoustic center" of the loudspeaker is the area most often located somewhere behind the dust cap. The exact acoustic center of radiation needs to be determined by measurement and is most often conveniently referenced to the mounting plane of the loudspeaker basket. In this case, you would refer to the "acoustic center offset".

The small-signal, electro-mechanical parameters described in this section are derived from input impedance of the driver. The only other data you will need to supply is the "effective" cone diameter and properly measured DC resistance, Re, of the voice coil. The Re value could be extracted from the impedance curve as well, but **better accuracy can be expected from a digital ohmmeter with 3 digit accuracy**.

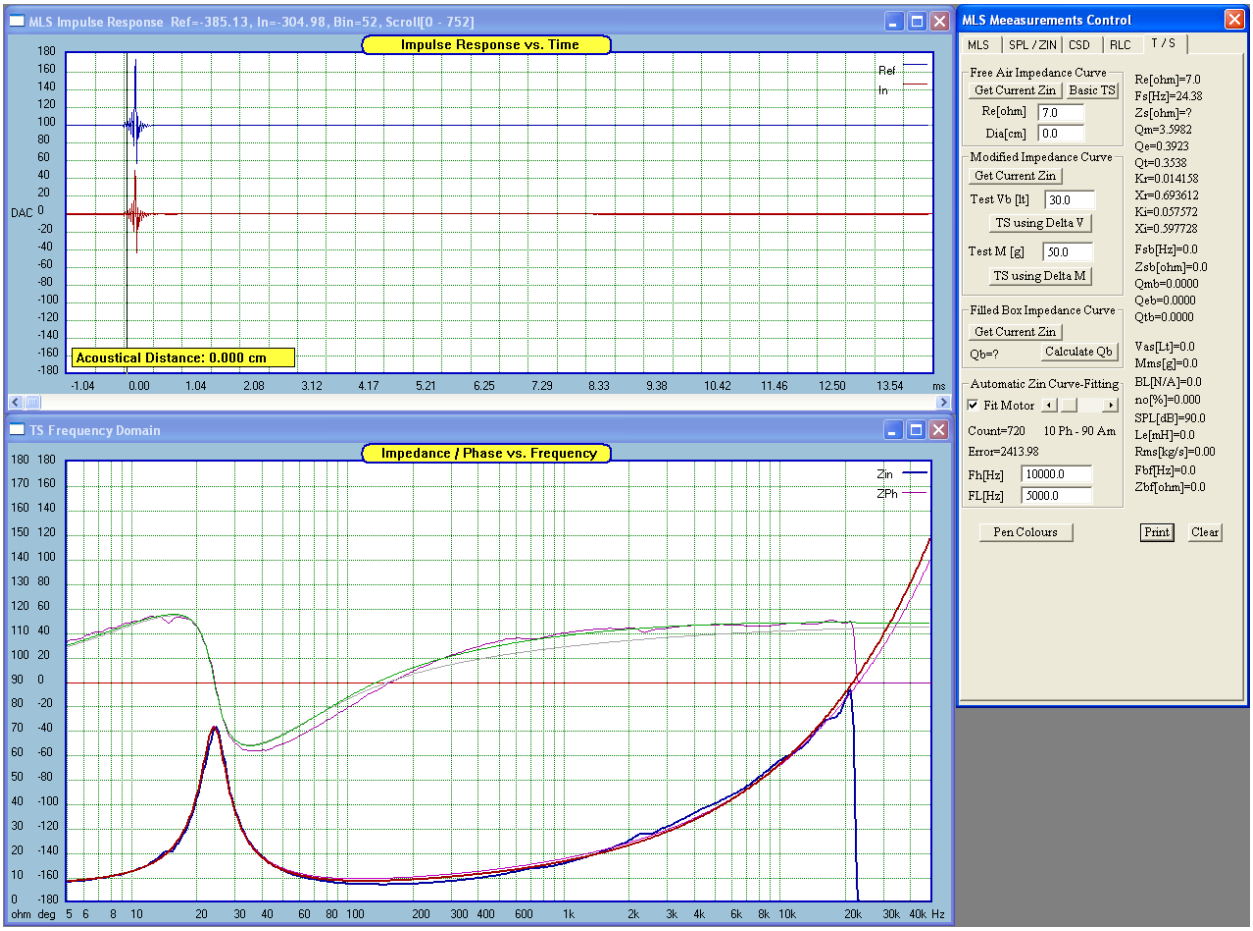


Figure 16.28 Free-air input impedance curve with basic TS parameters extracted

There are two simple methods for extracting Vas and other T/S small-signal parameters:

Added Mass method

The Added Mass technique requires additional mass, M_{Add} to be attached to the driver's cone and the new, "free-air" resonant frequency determined again. The extra mass should be large enough to cause resonant, Fs, shift by at least 25%.

$$M_{ms} = M_{Add}\left(\frac{F_s * F_s}{F_{sb} * F_{sb}} - 1\right)$$

Where:
Madd is the added test mass,
Fs is the loudspeaker free-air resonance,
Fsb is the loudspeaker resonant frequency with the test mass, Madd, attached.

With the Mms value calculated, we can proceed to establish the Vas as follows:

Cms = 0.0253/(Fs*Fs*Mms) and finally **Vas = ρ*c*c*Sd*Sd*Cms**

Where c, is the speed of sound, Sd, is the driver's piston area and ρ=1.18kg/cubic meter

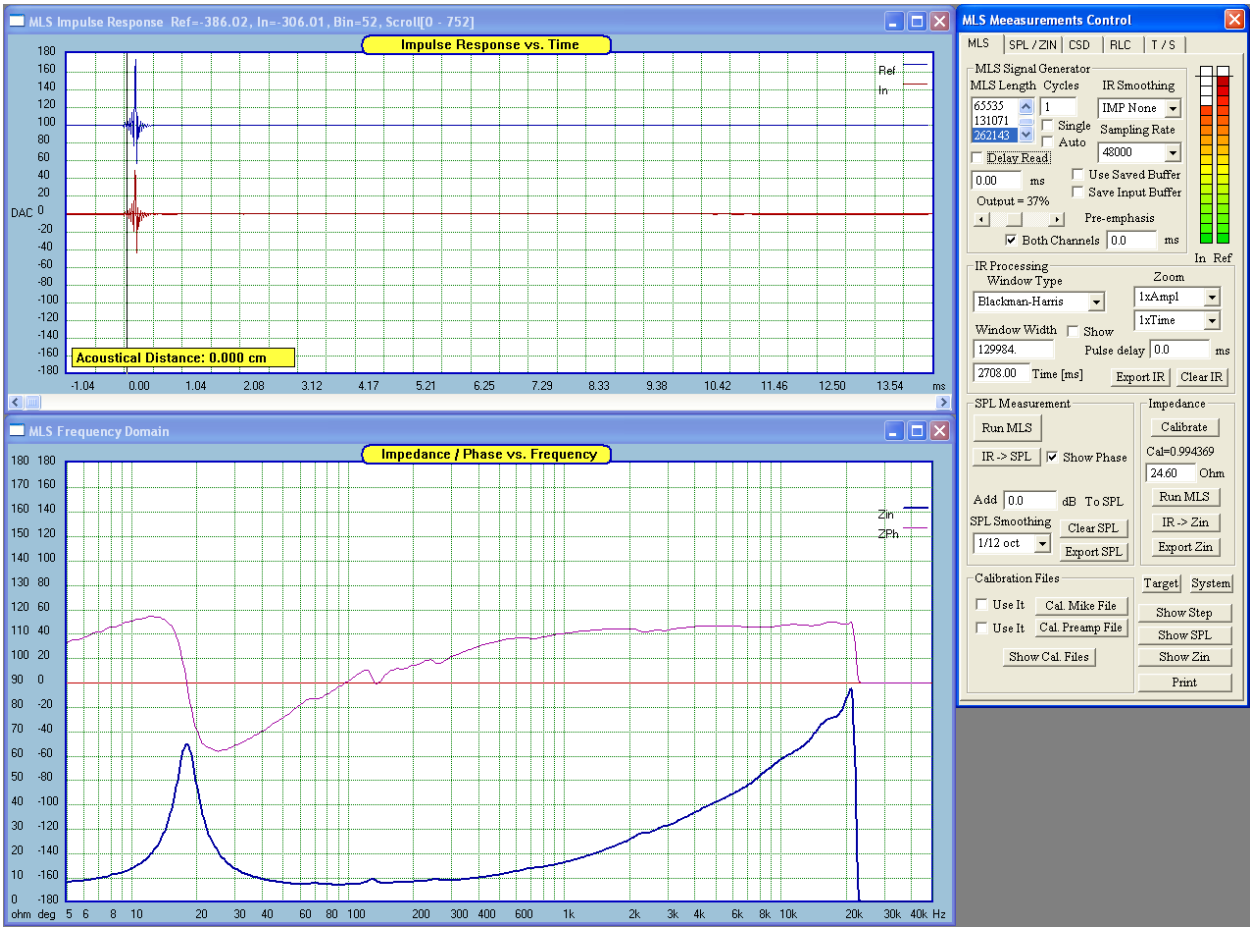


Figure 16.29. Shifted resonance due to added extra mass.

Delta Vas method

The Delta Vas method uses a well designed and well built test enclosure for the purpose of mounting the driver in it and measuring the new Fsb and new Qesb "in-the-box". This arrangement will shift the resonant frequency of the driver in the box upwards. You should aim at the enclosure volume, Vb, that would cause the resonant frequency to shift by at least 25%.

$$V_{as} = V_b\left(\frac{F_{sb} * Q_{esb}}{F_s * Q_{es}} - 1\right)$$

Where:

Vb is the test enclosure volume,
Fs is the loudspeaker free-air resonance,
Qes is the electrical Q-factor of the driver in free-air
Fsb is the loudspeaker resonant frequency when mounted in the test enclosure with Vb volume
Qsb is the loudspeaker electrical Q-factor when mounted in the test enclosure with Vb volume

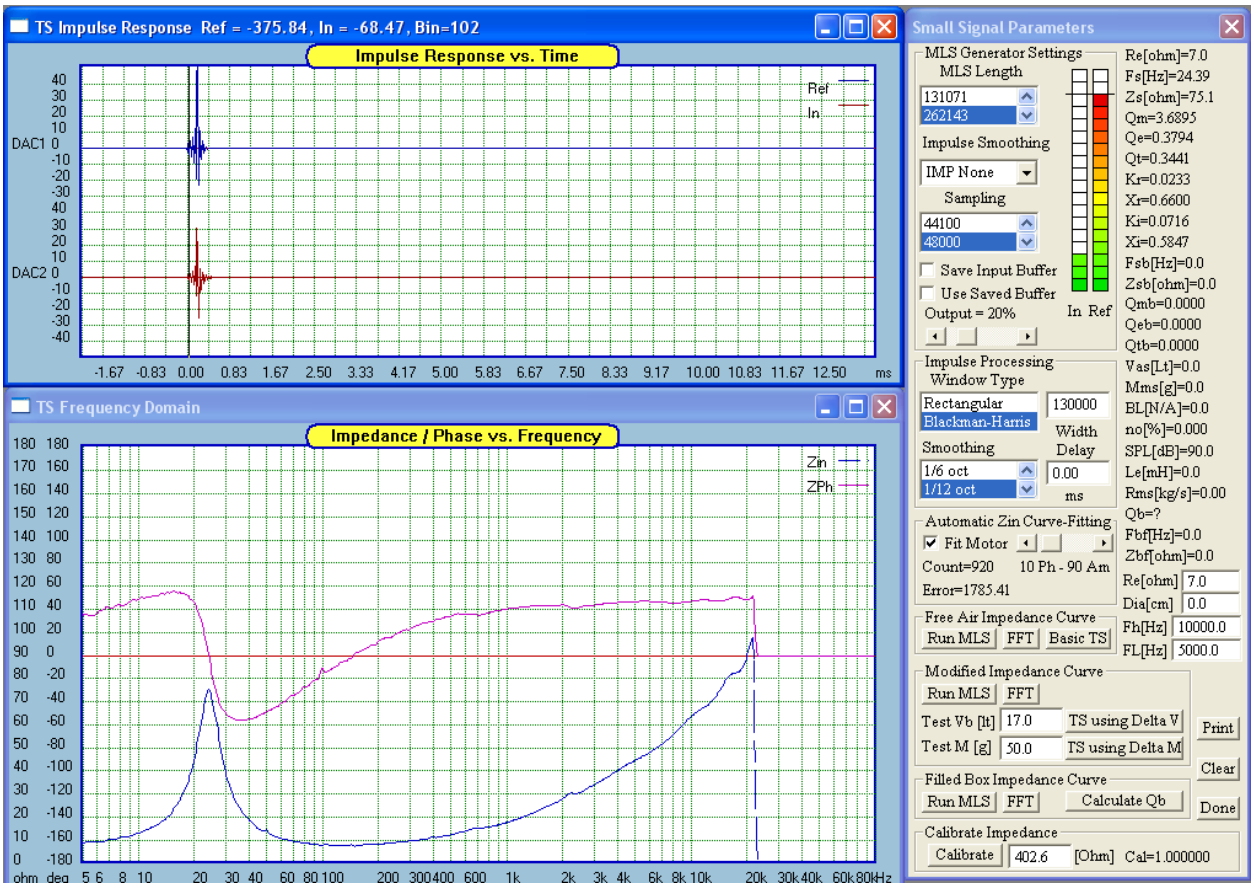


Figure 16.30. Free Air impedance curve

Extracting Voice Coil Parameters

This method of extracting parameters of the voice coil is attributed to J.Wright. Basically, the "free-air" input impedance of the driver, Zin(ω) is evaluated at two frequencies selected past the midrange frequency range. SounEasy uses f1 = 5kHz and f2 = 10kHz frequencies. Extracted parameters are used to calculate "variable inductor" and "variable resistor", which together, approximate the voice coil "semi-inductor" quite well. The method has the potential to provide much more accurate representation of the loudspeaker impedance at higher audio frequencies. Here are the formulas used by the program.

$$\omega_1 = 2\pi f_1$$

$$\omega_2 = 2\pi f_2$$

$$X_r = \frac{\log \frac{\mathrm{Re}(Z_{in}(\omega_1))}{\mathrm{Re}(Z_{in}(\omega_2))}}{\log \frac{\omega_2}{\omega_1}} \qquad K_r = \frac{\mathrm{Re}(Z_{in}(\omega_1))}{\omega_1^{X_r}}$$

$$X_i = \frac{\log \frac{\mathrm{Im}(Z_{in}(\omega_1))}{\mathrm{Im}(Z_{in}(\omega_2))}}{\log \frac{\omega_2}{\omega_1}} \qquad K_i = \frac{\mathrm{Im}(Z_{in}(\omega_1))}{\omega_1^{Xi}}$$

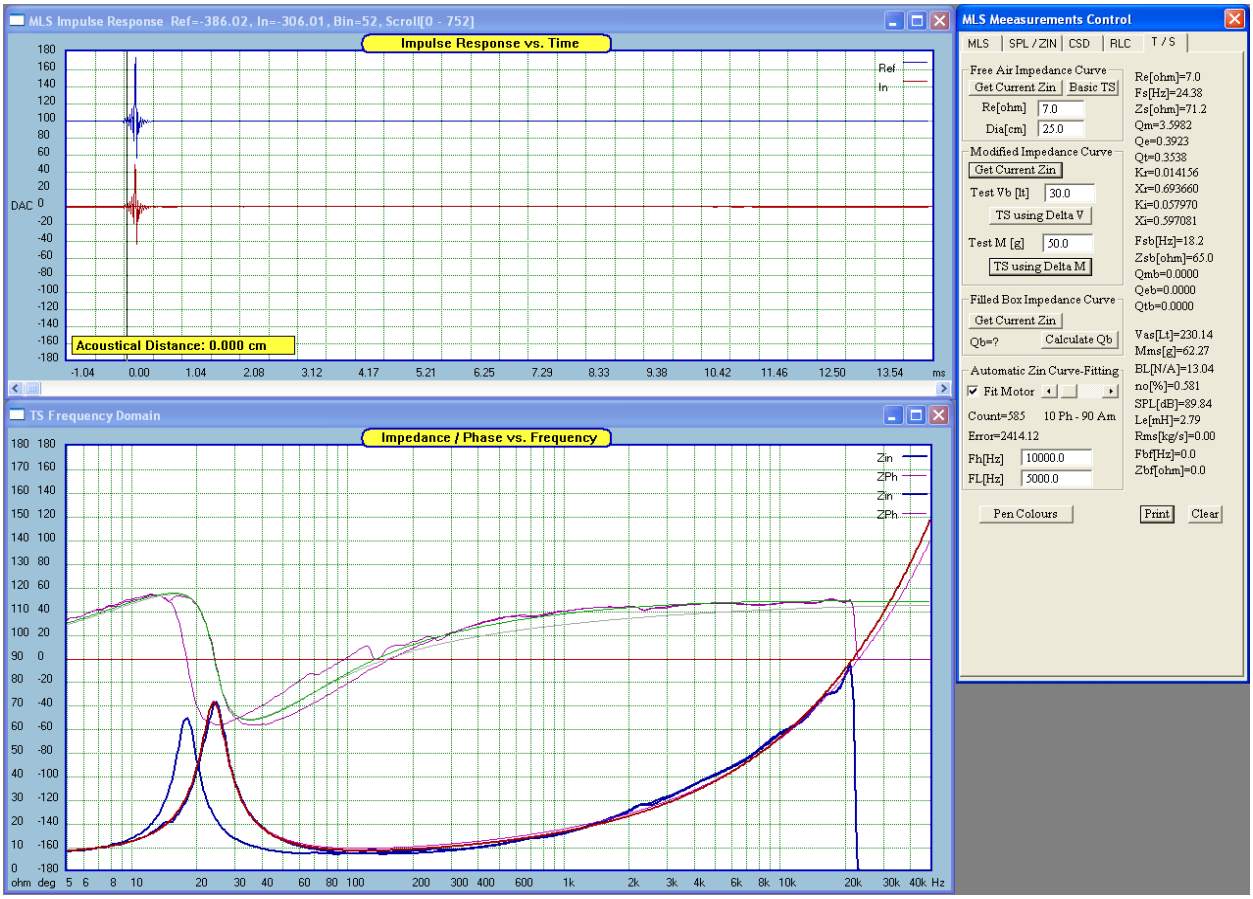


Figure 16.31. "Delta M" method.

Example of measured T/S set of parameters for a 12" woofer, using Delta Mass Method (added 50gram) are:

1. Fs = 24.38 Hz
2. Zmax = 71.2 ohm at Fs
3. Qm = 3.5982
4. Qe = 0.3923
5. Qt = 0.3538
6. Le at 152Hz = 2.79 mH
7. Mms = 62.27 gram
8. Vas = 230.13 Lt
9. BL = 13.04 N/A
10. Efficiency = 0.581%
11. SPL = 89.84 dB/1W/1m
12. Kr = 0.014156, Xr = 0.69366
13. Ki = 0.05797, Xi = 0.597081
14. Re = 7.0 ohm (measured separately with ohmmeter)
15. Diameter = 25.0 cm (measured separately)

Measurement Method for T/S Parameters Extraction

In general, you can use any MLS length and sampling frequency combination. However, more often than not, you will find, that certain sampling frequencies do not produce good results. This is to do with soundcard internal frequency conversions and clocking issues. Also, you can use test resistor of any reasonable value (10-500ohms), however we recommend, that you use the highest resistor value you can work with and also the longest MLS sequence. The long MLS sequence will assure good frequency resolution required at the low end of the frequency band (48000/262000 = 0.18Hz) and then you can use Blackman-Harris smoothing window of 131000 length, giving you smoother curves and still good resolution of 0.36Hz.

Higher resistor value will assure, that your test setup is working indeed in "small-signal" area. This is often forgotten issue, that we are trying to extract small-signal (no BL(x) or Cas(x) type of non-linearity involved).

When making impedance measurements, Zin, the "Ref" probe, sensing V1, is connected to the output of the power amplifier and the "In" probe, sensing V2, is connected right across the loudspeaker. The measured impedance is calculated from the formula:

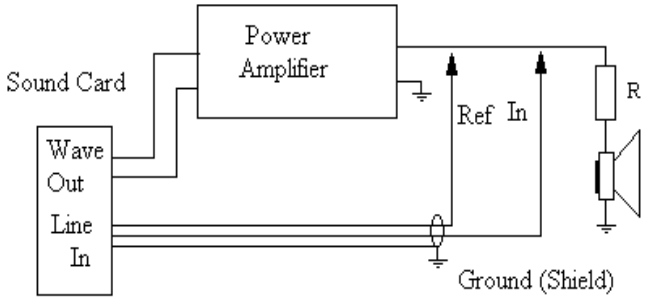
Zin = R*(V2/V1)/(1 – V2/V1), R > 400 Ω is recommended for constant current method

The method used for direct impedance measurements is close to constant current. This is because the source impedance (resistor R = 400 Ω, or higher inserted between the power amplifier output and the loudspeaker) is high so that the current following through the voice coil remains relatively constant**. Using lower resistance R is possible**. The formula above indicates clearly, that input impedance is heavily dependant on the voltage ratio, as measured by left and right channel of the sound card. There is a potential for significant error, as the voltage gains in both channels may not be exactly equal. More importantly, if you use probe resistive dividers, the voltage divider ratios may not be equal in both probes due to resistor tolerances. In order to ensure accuracy of the impedance measurements, you must CALIBRATE the system before a valid impedance measurement can be taken. During the calibration process, you will feed the same signal to both probes and the program will measure and memorize the difference in actual readings. This information will be use next for scaling the measured voltages.

CALIBRATION FOR IMPEDANCE MEASUREMENT

Follow the steps below to perform calibration for impedance measurements.

4. **Connect the test setup**: power amplifier, test probes. Connect the test probes TOGETHER, at the output of the power amplifier – as shown below. Power-up the power amplifier.

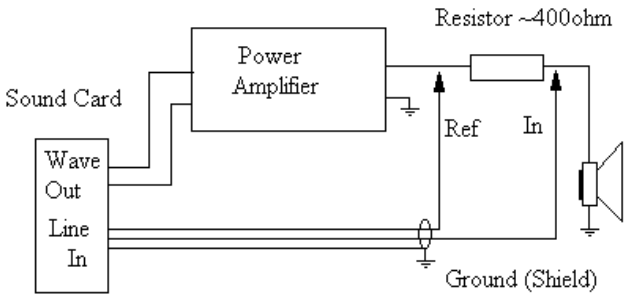


16.32 Calibration for impedance measurements.

5. **Select "T/S Parameters**" from the main menu to invoke MLS measurement screen.
6. **Perform calibration**
   - Select Generator level from 0-100%.
   - Press "Calibrate" button in the Impedance group box. The setup will now generate 1000Hz tone needed for measurement of both channels. **The "In" indicator corresponds to the MEASURED signal from your loudspeaker terminal and the "Ref" indicator corresponds to the REFERENCE signal probe (connected to the output of the power amplifier).**

Having calibrated the system you can immediately follow the steps below to perform MLS measurements of the input impedance and phase of your loudspeaker.

9. **Connect the test setup**: power amplifier, test probes, test resistor and the "loudspeaker under test". Power-up the power amplifier.



16.33 Test setup for small-signal parameters extraction. Resistor can be used from 10-500ohm.

10. **Select "T/S Parameters**" from the main menu to invoke corresponding MLS measurement screen.

11. **Set up MLS sequence**
    - Select "MLS Length" from the provided list box – on SB Live! Sound card set to 262141.
    - Select sampling rate from the provided list box– on SB Live! Sound card set to 48000.
    - Select Generator level from 0-100%.

12. **Select windowing parameters – this is done automatically**.
    - When the MLS sequence has finished, you should see the impulse response of the measured signal on the screen. The program will automatically place the time cursor 10 samples to the left of the peak value.
    - Select the type and the width of the time window. **Please DO NOT create wider windows than the length of the MLS.**

13. **Select smoothing parameter**.
    This is a simple selection of one of the provided smoothing ratios: 1/3, 1/6, 1/12, 1/24 and 1/48 octave.

14. **Select additional Impulse smoothing.**
    - The calculated impulse response will be smoothed before it is processed with FFT. However, calculation time will be longer.

15. **Enter "Re" AND "Effective Diameter" data - IMPORTANT**.

16. **Generate Impedance curve – this is two-step process**

    - 7.1. Simple press the **"Run LMS**" button in the **"Free Air Impedance Curve"** group and wait a short while for the calculations to finish. You should now hear the noise coming out from the measured speaker. Your "Input Level LED" indicator should lit-up for "In" bars and "Ref" bars. **The "In" indicator corresponds to the MEASURED signal from your loudspeaker terminal and the "Ref" indicator corresponds to the REFERENCE signal probe (connected to the output of the power amplifier).** Please note, that wider smoothing band will result in increased computation time.

      Now, press the **"FFT"** button to obtain the Zin curve for free-air loudspeaker.
      At this point, you can press "**Basic TS"** button to obtain a subset of the loudspeaker parameters.

    - 7.2 Finally press the **"Run LMS**" button in the **"Modified Impedance Curve"** group and wait a short while for the calculations to finish. You should hear again the noise coming out from the measured speaker. Your "Input Level LED" indicator should lit-up for "In" bars and "Ref" bars.

      Now, press the **"FFT"** button to obtain the Zin curve for the added mass technique OR the added Vb technique.

17. **You are now ready to extract the small-signal parameters. Depending on the chosen method, enter "Test Vb" – for the Delta V method OR enter "Test M" for Delta M method and press corresponding button to extract the T/S parameters.**

Advanced MLS Measurement Features

Improving Accuracy of TS Parameters Extraction

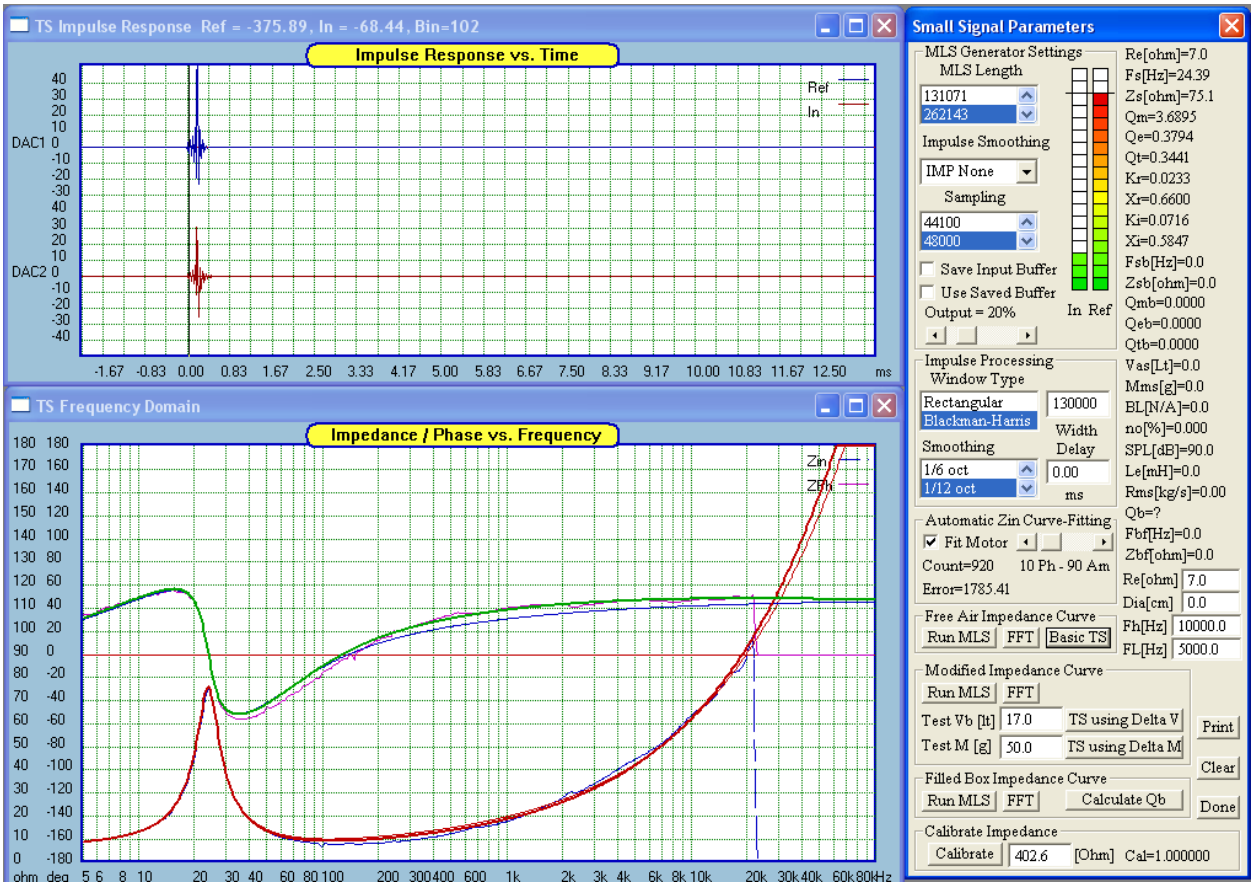
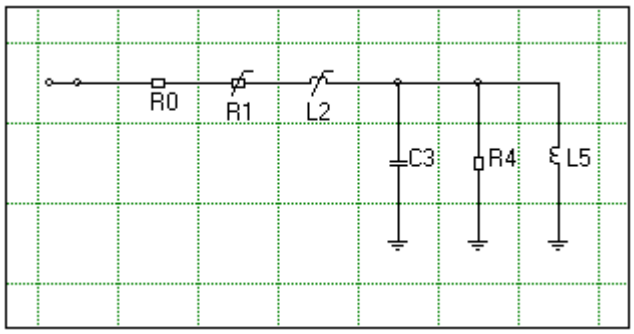


Figure 16.34. Kr, Ki, Xr, Xi. parameters calculated via "curve fitting" from the impedance curve.

Loudspeaker input impedance model developed by J.Wright is proving to be quite popular. The most obvious reason for this is the need for only 4 elements to characterize the high frequency section of the impedance curve. Taking a closer look at the results of calculated Kr, Ki, Xr, Xi reveals, that indeed a good match between calculated and measured curves can be accomplished for modulus of the input impedance. The phase of the input impedance is somewhat less accurate on some loudspeakers. To overcome this problem, we have implemented additional operation of "curve fitting" the modeled Zin onto the measured modulus and phase of the input impedance. Now, the modeled curve has to fit the measured curve across the whole measurement frequency range – not only at two selected frequencies, as calculated before. The values of Kr, Ki, Xr, Xi as calculated by previously given equations are used as starting parameters in the "curve fitting" process.



R0 = Re
Ts = 1/(2*π*Fs)
R = Re*Qm/Qe
C = (Ts*Qe)/Re
L = (Ts*Re)/Qe
R1 = Kr *ω^{Xr}
L2 = Ki*$\omega^{(Xi-1.0)}$

Simple impedance model for curve fitting of Kr,Ki, Xr,Xi.

The results of the optimization are shown on Figure 16.27. It is easily observable, that both: modulus and phase of the input impedance are now much better matched to the measured results. The new Kr, Ki, Xr, Xi are highlighted in red. The "curve fitting" mechanism will be automatically activated if you check the **"Fit Motor"** check box and is implemented as optimization (error minimization) algorithm. The algorithm accounts for both: (1) modulus and (2) phase of the input impedance assigning weighting in accordance with the position of the slider provided below. For instance, **10Ph – 90Am** indicates, that Phase contributes 10% and Amplitude (modulus) contributes 90% to the total error. This way, you can guide the optimization algorithm for the best fit between phases and amplitudes for each measured speaker individually.

Extracting Enclosure Qb Parameter

The method for extracting enclosure Qb, is based on Dr. D'Appolito's concept described in his Loudspeaker Design Cook book. You will need to perform three runs of MLS measurements on your loudspeaker under different conditions and the program will extract the Qb from collected data. Extracting Qb parameter is provided from **"T / S" TAB**. Please follow the steps outlined below:

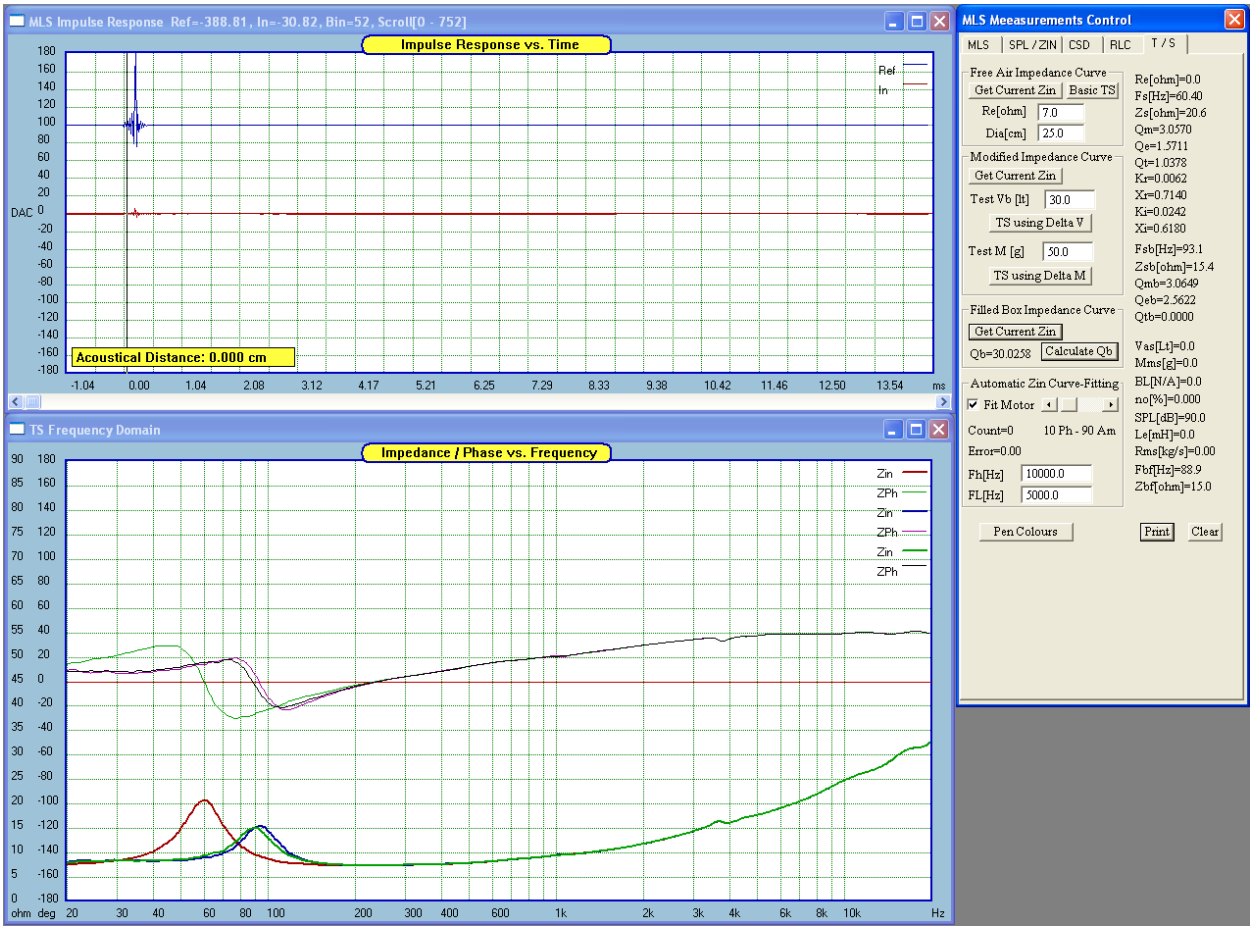


Figure 16.35 Extracting Qb process.

1. **Setup MLS** – Select MLS length (262143), Sampling Rate (48000), Volume (say 30), and your test resistor value (360ohm in our example).
2. **Calibrate system** – follow impedance calibration procedure described before.
3. **Select window parameters** – Blackman-Harris and 130000-width.
4. **Select smoothing parameters –** 1/6 oct and perhaps you may want extra smooth curves with Impulse Averaging – thus select one of the options from the "**IR Smoothing"** list box**.**
5. **Run 1 (Free Air Impedance Curve) –** Place the loudspeaker in the free-air conditions. Press **"Run MLS"** and then **"IR->Zin"** buttons on the MSL screen to capture the first impedance curve**.**

6. Go to **"T/S"** TAB and press "Get Current Zin" in the "Free Air Impedance Curve" group.
7. **Run 2 (Modified Impedance Curve) –** Mount the loudspeaker in **unfilled** enclosure you wish to measure Qb. Press **"Run MLS"** and then **"IR->Zin"** buttons on the MSL screen to capture the second impedance curve**.**
8. Go to **"T/S"** TAB and press "Get Current Zin" in the "Modified Impedance Curve" group.
9. **Run 3 (Filled Box Impedance Curve)** – Mount the loudspeaker in **filled** box you wish to measure Qb. Press **"Run MLS"** and then **"IR->Zin"** buttons on the MSL screen to capture the third impedance curve**.**
10. Go to **"T/S"** TAB and press "Get Current Zin" in the "Filled Box Impedance Curve" group.
11. Finally, enter value for Re in the T / S TAB, and **press "Calculate Qb" button** – if all previously measured parameters check out, you will se Qb calculated and displayed on the screen in the results area.

Figure 16.35 shows and example measurements taken on a small 8" woofer. Green impedance curve is the "**free-air**" plot. The red impedance curve is the "**unfilled box**" plot and the black impedance curve is the "**filled enclosure**" plot.

Zoom-In and Scroll Impulse

Having captured the Impulse Response of the loudspeaker, you may wish to review the amplitude inn greater details. Or perhaps you wish to review the impulse past the current window display. There are several controls allowing you to zoom-in and scroll the impulse. **Scroll is accomplished by moving the slider or clicking the arrows at the bottom of the "MLS Impulse response" window.** Zoom-in feature is implemented in amplitude domain. There is a list box located in the "Zoom" area. Amplitude zoom can be selected from x1 to x64.

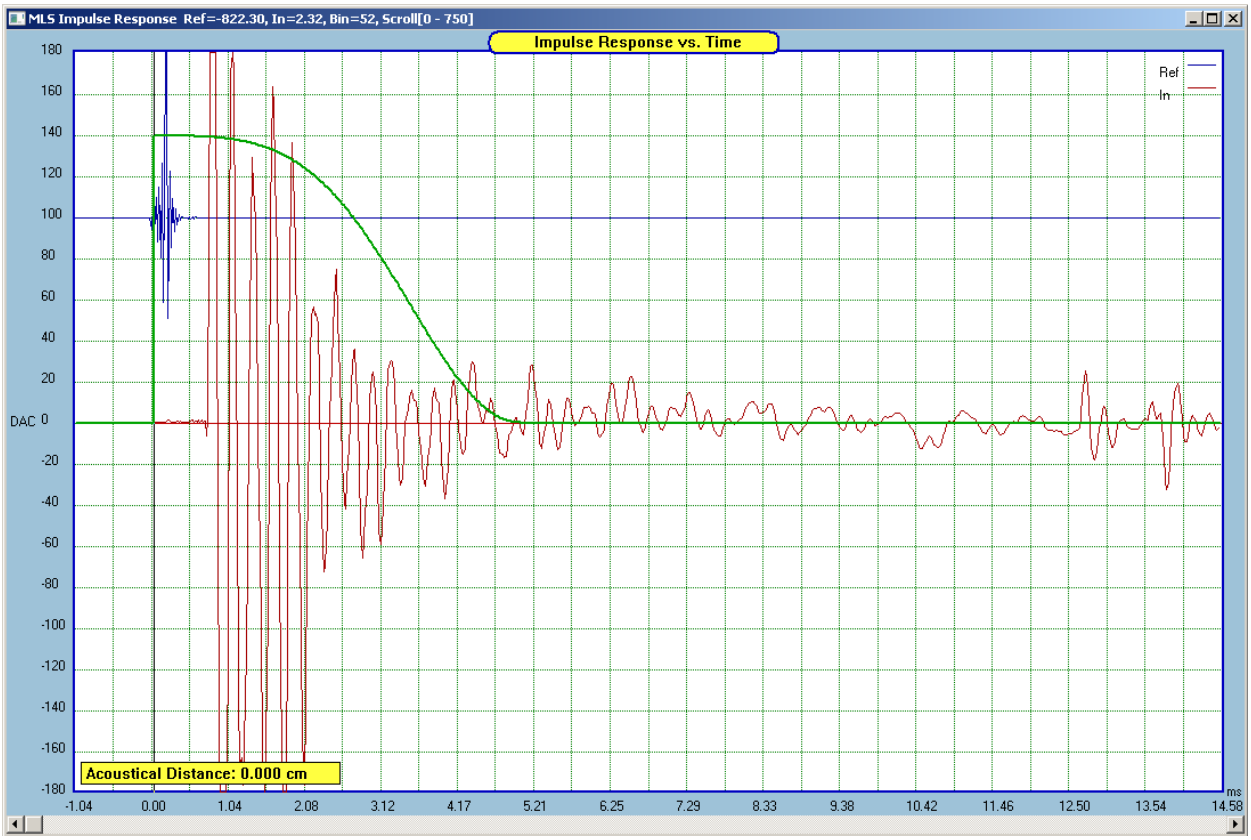


Figure 16.36. Example of "16 x Zoomed-in" amplitude of the impulse.

Pre-emphasized MLS signal

Standard MLS signal is basically a broadband spectrum noise. When testing higher frequency driver using such signal, you could unknowingly drive the loudspeaker into non-linear area of operation. In cases like this, it may be beneficial to use MLS signal with reduced low-frequency spectrum – the pre-emphasized MLS signal. **The pre-emphasis feature is activated automatically, by entering greater than zero value into "Pre-emphasis" data field**. Pre-emphasis is typically employed as a first order, high-pass circuit and the cut-off frequency is expressed as a time constant, "**τ**". Example on Figure 16.37, shows the spectrum of pre-emphasized MLS signal with the time constant of 1.0ms. Performing normal measurements, you would use the pre-emphasized MSL signal to excite the DUT and connect the "Ref" probe to the amplifier's output and the "In" probe to the mike pre-amp output (typical SPL measurements configuration). If you perform loop-test with both probes connected to the same signal, the frequency response will show as flat. Pre-emphasis it traditionally expressed as a time constant "**τ**", and it relates to 3dB cut-off frequency as follows:

$$\tau = 1/(2\pi f) \quad ---> \quad f = 1/(2\pi\tau) \quad \text{eg: 1.0ms -> 159Hz.}$$

You can also output the normal MLS signal on one sound card output and the pre-emphasized MLS signal on the other output. This will give you the opportunity to review the actual spectrum of the pre-emphasized MLS signal. Simply clear the "Both Channels" check box and connect the broadband MSL output to "Ref" probe and the pre-emphasized MLS to "In" probe – see below. Please note, that MLS length is 131071 and Blackman-Harris window is 30000.

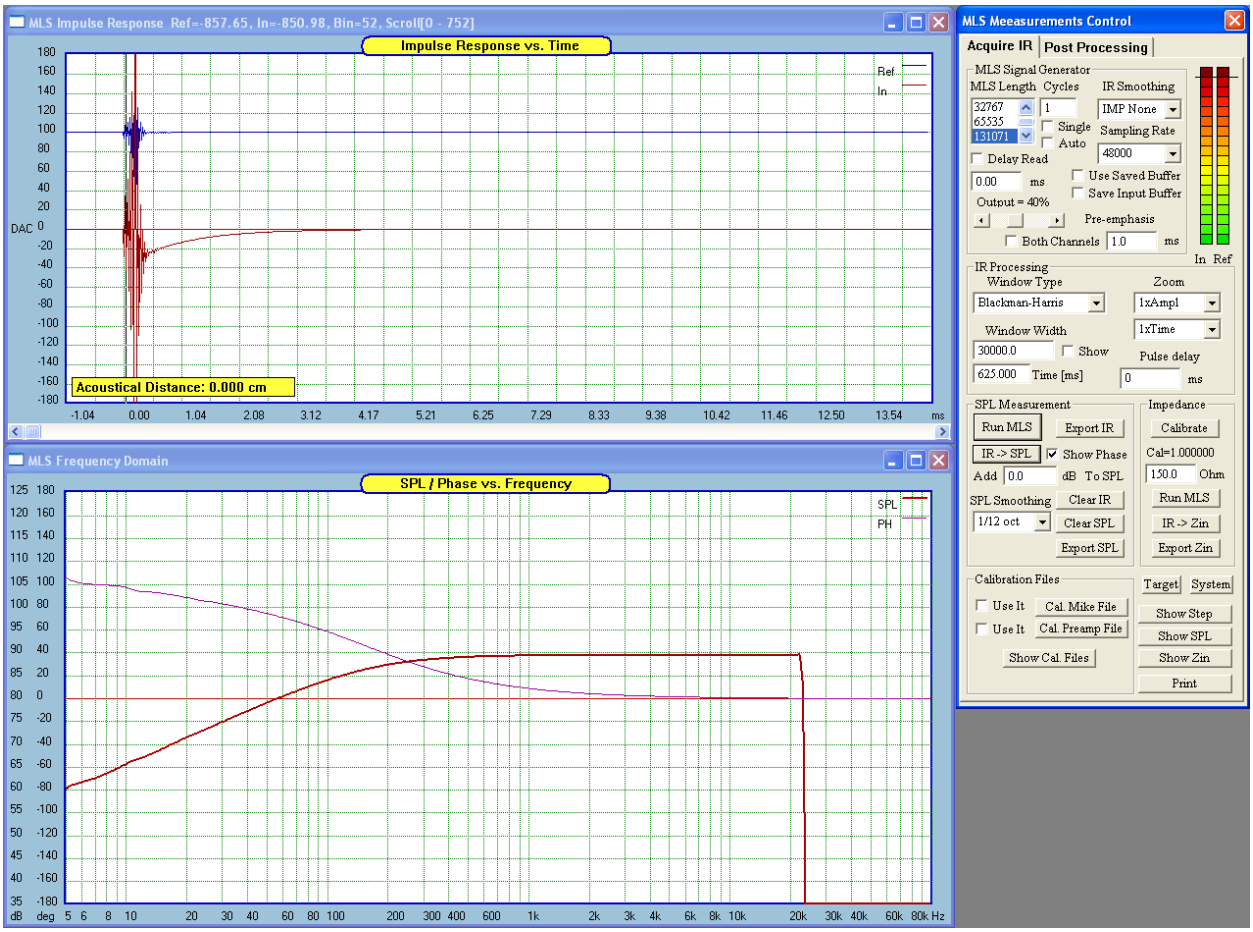


Figure 16.37. Example of pre-emphasized MLS signal

Using Input Impulse Buffer

There maybe some of the post-processing activities, that may require re-taking the measurement several times. Should you require to do this, SoundEasy now has the ability to save the captured data into a buffer and reading it from the buffer repeatedly. Because of this, you only need to make single MLS measurement and then you can switch off the test equipment. Saving the buffer is performed via a standard file saving dialogue box, that allows you to assign a file name to the saved buffer, when the **"Save IR"** button is pressed. After that, you can check the **"Use Saved Input Buffer**" box and then read the saved buffer rather then make repetitive measurements. **You can save and read unlimited number of Impulse Responses for later post-processing.**

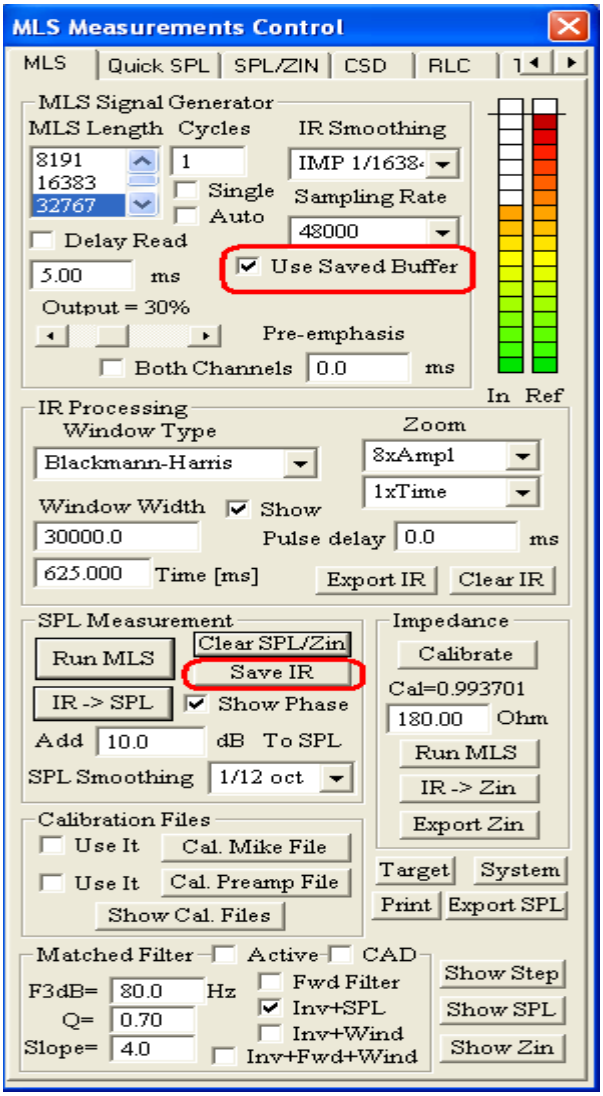


Figure 16.38. "Save Input Buffer" and "Used Saved Input Buffer" check boxes.

Impulse smoothing

Impulse smoothing may be useful when you require better phase clarity at higher frequencies. Several smoothing options are available and selectable from "pull-down" list box located in the "Generate MLS " group of controls. If you wish to switch off the impulse smoothing, please select **"IMP None"** from this list box. The **"IMP 1/32"** provides the strongest smoothing. An example of non-smoothed impulse phase response is shown on Figure 16.39. It is easy to observe, the phase response around 10kHz undergoes several jumps. When the impulse smoothing with "IMP 1/32" option was selected, the phase response around 10kHz and also HBT response are much smoother.

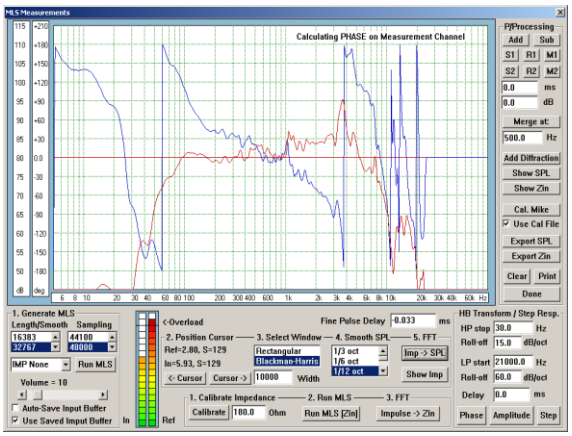


Figure 16.39 No impulse smoothing

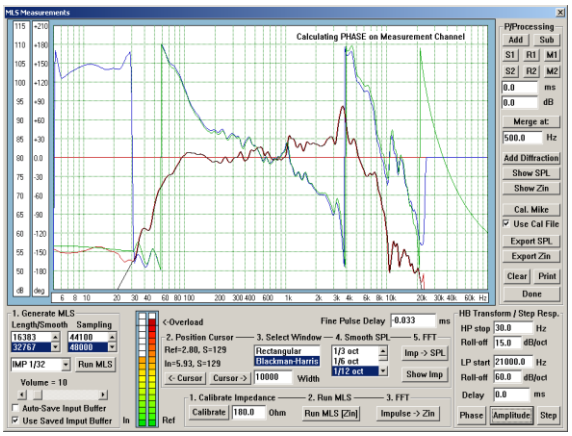


Figure 16.40. "IMP 1/32" impulse smoothing.

Step Response

Step response is calculated from captured frequency response in MLS TAB (for non-minimum phase systems) OR from Hilbert-Bode Transform in Post-Processing TAB (for minimum-phase systems). This approach allows you to basically use HBT to "clean" high and low-end of the spectrum before the step response is calculated.

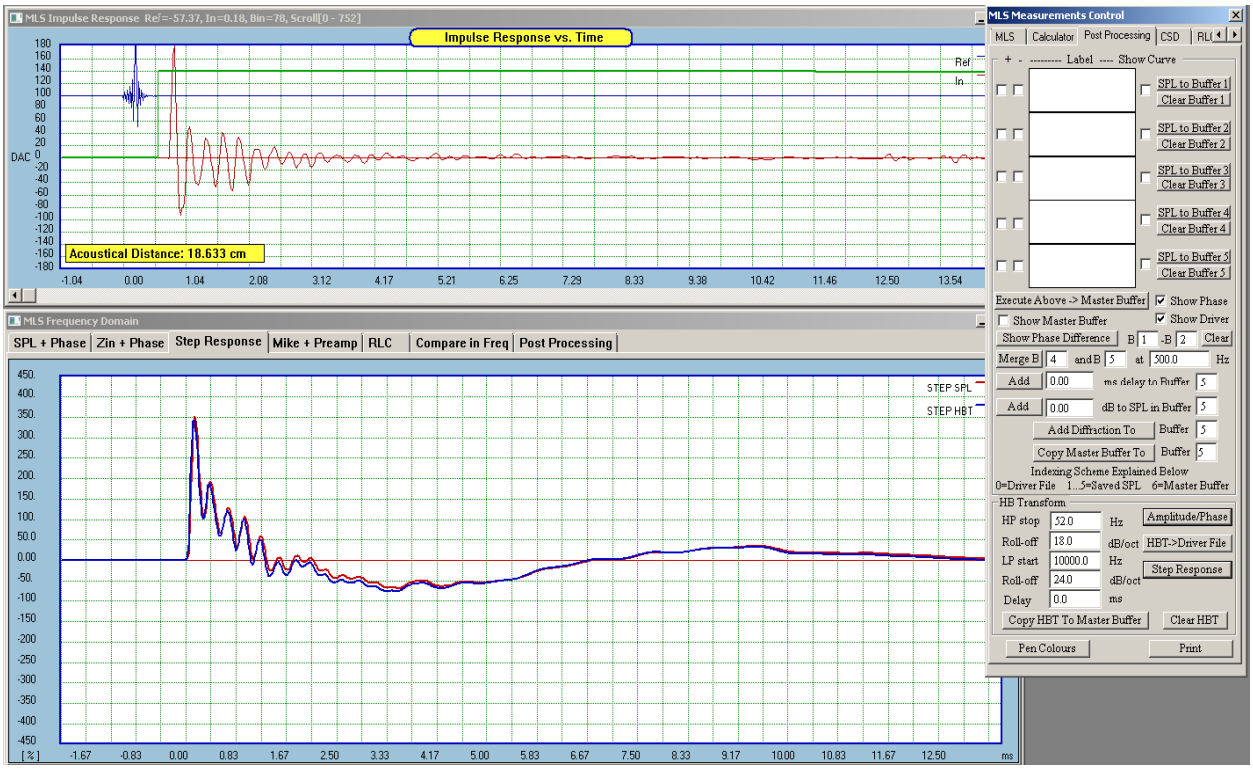


Figure 16.41 Step Response.

Step response calculated for signal captured around 80dB level (see Figure 16.39) may be a little to low in amplitude when the Step response is calculated. This my be prevented by adding, say 20dB to the amplitude just before the HBT is calculated – see Figure 16.42. Simply enter 20dB into the [dB] **"Add"** on the "Acquire IR" TAB,

Press **"IR->SPL"** button to actually add the 20dB to the curve – see Figure 16.42.

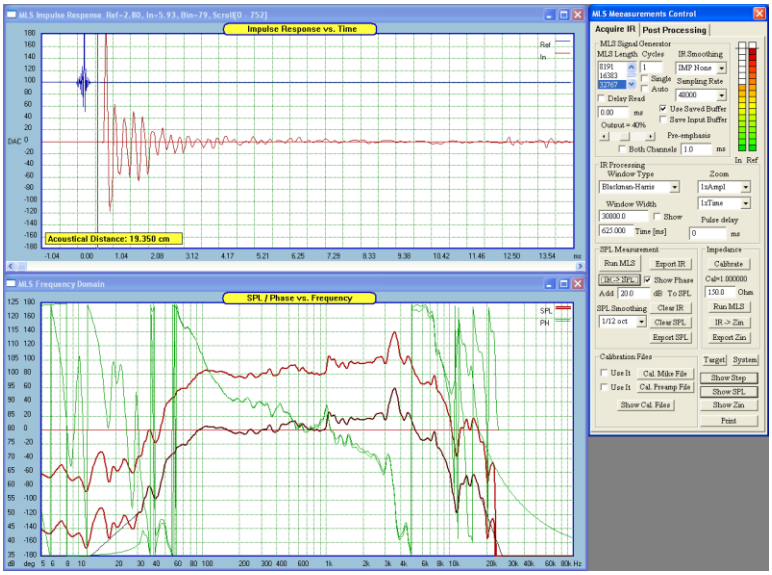


Figure 16.42. 20dB was added to the example curve.

Microphone And Pre-Amplifier Calibration Files

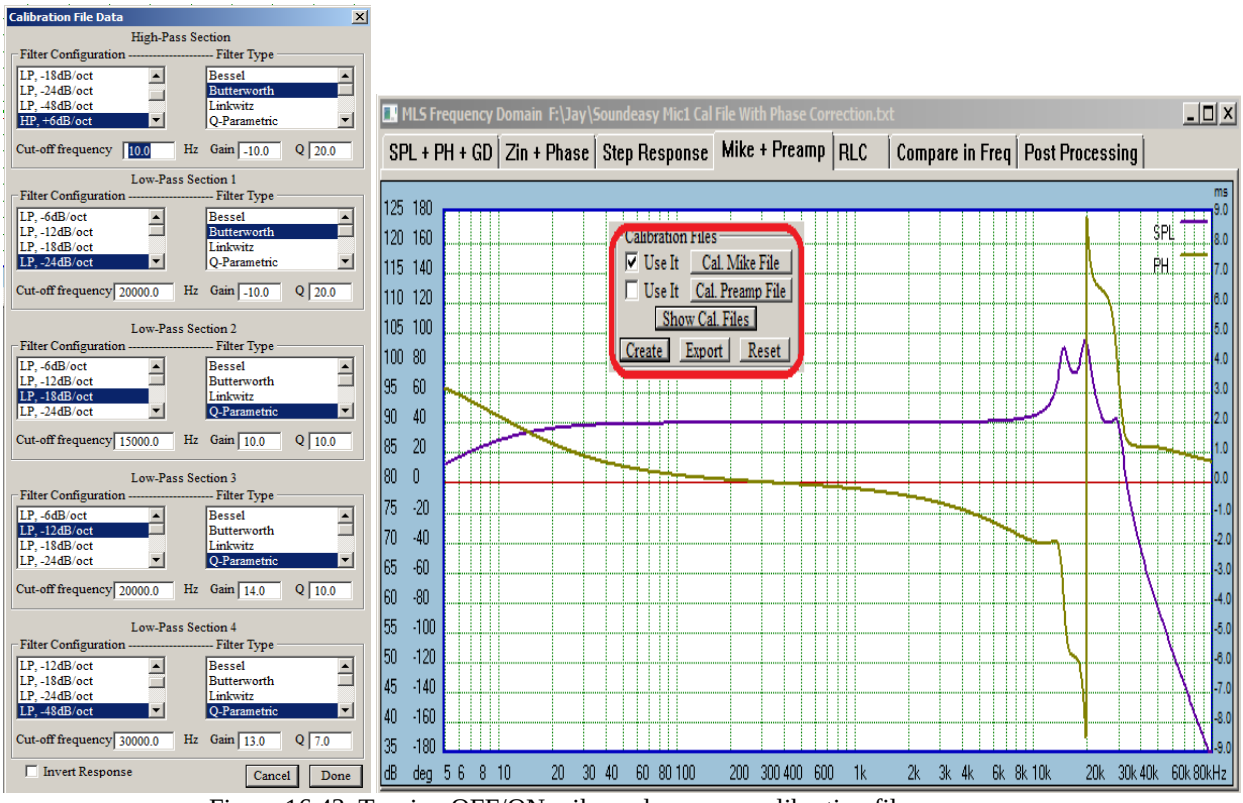


Figure 16.43. Turning OFF/ON mike and pre-amp calibration files.

A very elaborate microphone calibration file can be created using the ”Calibration Files” editor. The editor generates accurate amplitude **and correct phase responses** by approximating the microphone SPL/Phase with 5 built-in filter components – see Figure 16.43. The data entry is clearly labeled and the buttons are also self-explanatory.

The MLS and analog screens can be easily used with or without the microphone/pre-amp calibration files. Switching function is accomplished by checking/un-checking the box left to the “Cal. Mike File” and “Cal. Preamp File” buttons.

Adjusting Vertical Impedance Scale

To adjust impedance scale in MLS and Analog screens please use "**Vertical Resolution Control**" dialogue box, available from "**Enclosure Calculators**" menu.

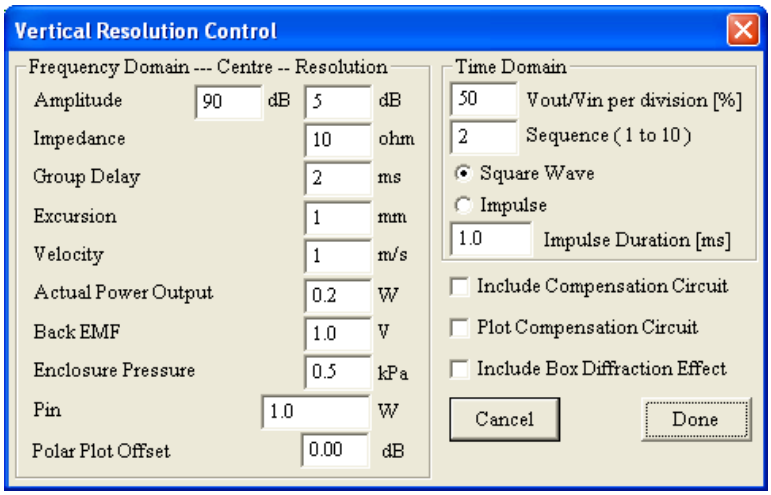


Figure 16.44 "Vertical Resolution Control" dialogue box – controls MLS impedance scale.

Measuring R, L, C Components

The RLC measurement system is intended to provide you with evaluating components, that are typically used **in Hi-If passive crossover networks**. This function is provided by the "RLC" TAB in the Easy Lab section.

Sound Easy determines the reactive components by measuring the impedance over a user-defined frequency range and then curve-fits simple component model to the measured data. Using broad frequency range rather than single frequency ensures more accurate results. Impedance measurements are performed using MLS-type of signals, as this method is quite fast. All MLS-related signal generating and capturing controls are on the **MLS TAB,** and the process of generating and capturing impedance-related measurements has been described before. Intuitively, one would select the measurement frequency range in accordance with the designated frequency range of operation of the component. For instance, 3.0mH coil could be used in low-pass filter below 1-2kHz. Therefore, you would set the screen resolution to 5Hz – 10kHz to obtain good representation of the coil's reactance. Inversely, a 0.47uF capacitor could be used in higher frequency range of filters, so more prudent choice of the frequency range would be 200Hz – 50kHz and sampling rate of 96kHz.

Component's models: **$Xc = 1/(2\pi fC)$ and $Xl = 2\pi fL$**

Output impedance of the soundcard is typically in the order of kilo-ohms, which does not match the need for providing higher drive for low-impedance components such as small inductors. We would therefore highly recommend, that you use your trustworthy power amplifier, as you did for other impedance measurements. The recommended test setup is shown on Figure 16.45. Please remember, that you need to perform calibration procedure as described in "CALIBRATION FOR IMPEDANCE MEASUREMENT" section before.

The value of the Reference Resistor (~400ohm on the schematic) can be varied depending on the component value. For instance, small capacitors will measure more accurately with higher resistors and vice-versa.

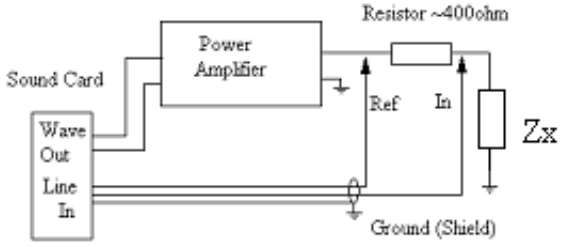


Figure 16.45. Test setup for measuring **Zx**

Typical measurement sequence looks as follows:

1. Connect the test setup as for calibration
2. Enter the Reference resistor value and perform calibration.
3. Generate Impedance Curve
   Select MLS length – 131071 recommended, sampling rate – 48K or 96K
   Set Volume to obtain full LED indication right up to the "Overload" level
   Select Window – Blackman-Harris, 130000 recommended
   Select Smoothing – 1/12 oct
   Press "**Run MLS**" in the Impedance MLS section – this will capture impulse response of the component under test.
   Press "**IR -> Zin**" button – this will plot the component's reactance.

Now is a good time to re-evaluate if the Reference Resistor and the frequency range were selected sensibly. You should aim at the plots looking like on 16.46 to 16.50 depending on the component type. Once you are happy with the reactance curve, representing the component value over broad frequency range, you are ready to "curve-fit" the built-in component's model to the measured data. There are three buttons provided in the **"RLC" TAB**.

1. Press the **"Find R" -** button to activate resistor fitting function
2. Press the **"Find R+L"** button to activate inductor fitting function
3. Press the **"Find C"** button to activate capacitor fitting function

Please focus now on Figure 16.49 (measurements of 3.0mH inductor). It is observable, that screen frequency range was selected to show a **flat reactance section from 5Hz to 50Hz**. This was done purposefully, as the program is able to determine the DC resistance from the flat section of the curve, located towards the left side of the screen. Larger inductors would require reduction in upper frequency range to 2-5kHz to cut out artifacts error of MLS processing at the frequency extremes.

Resistors

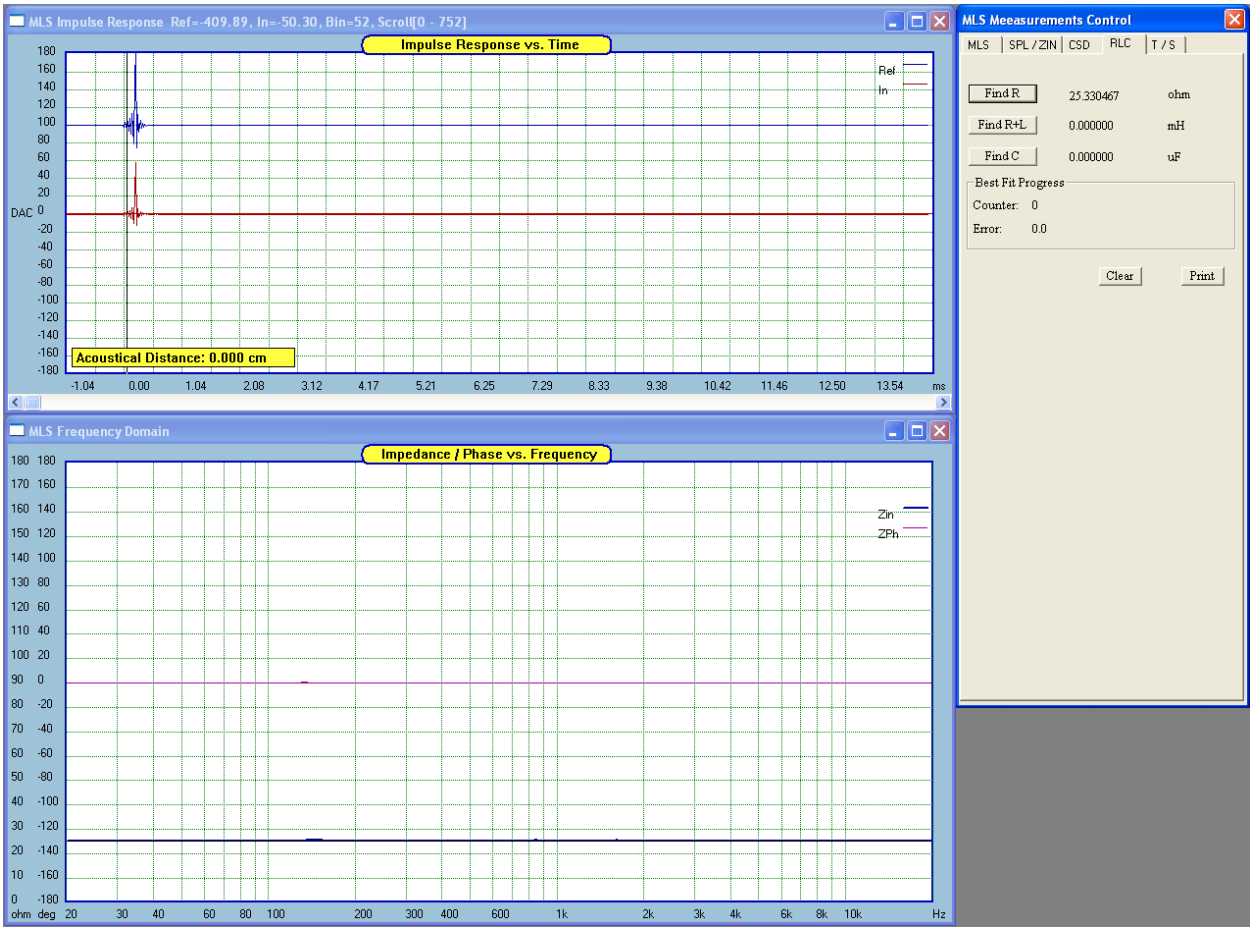


Figure 16.46. R = 25.0 ohm resistor

Capacitors

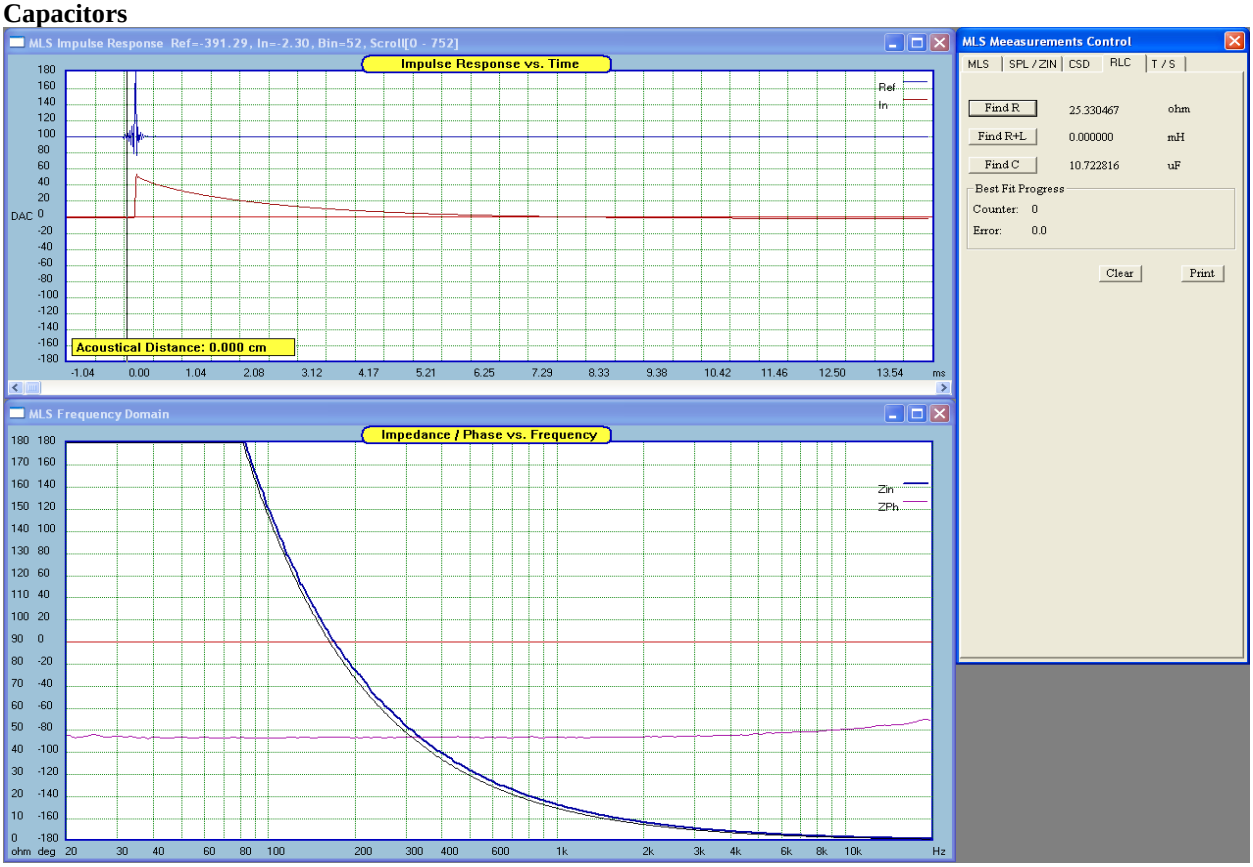


Figure 16.47. **Label: C = 10.0uF, 5%, Measured C = 10.72uF**

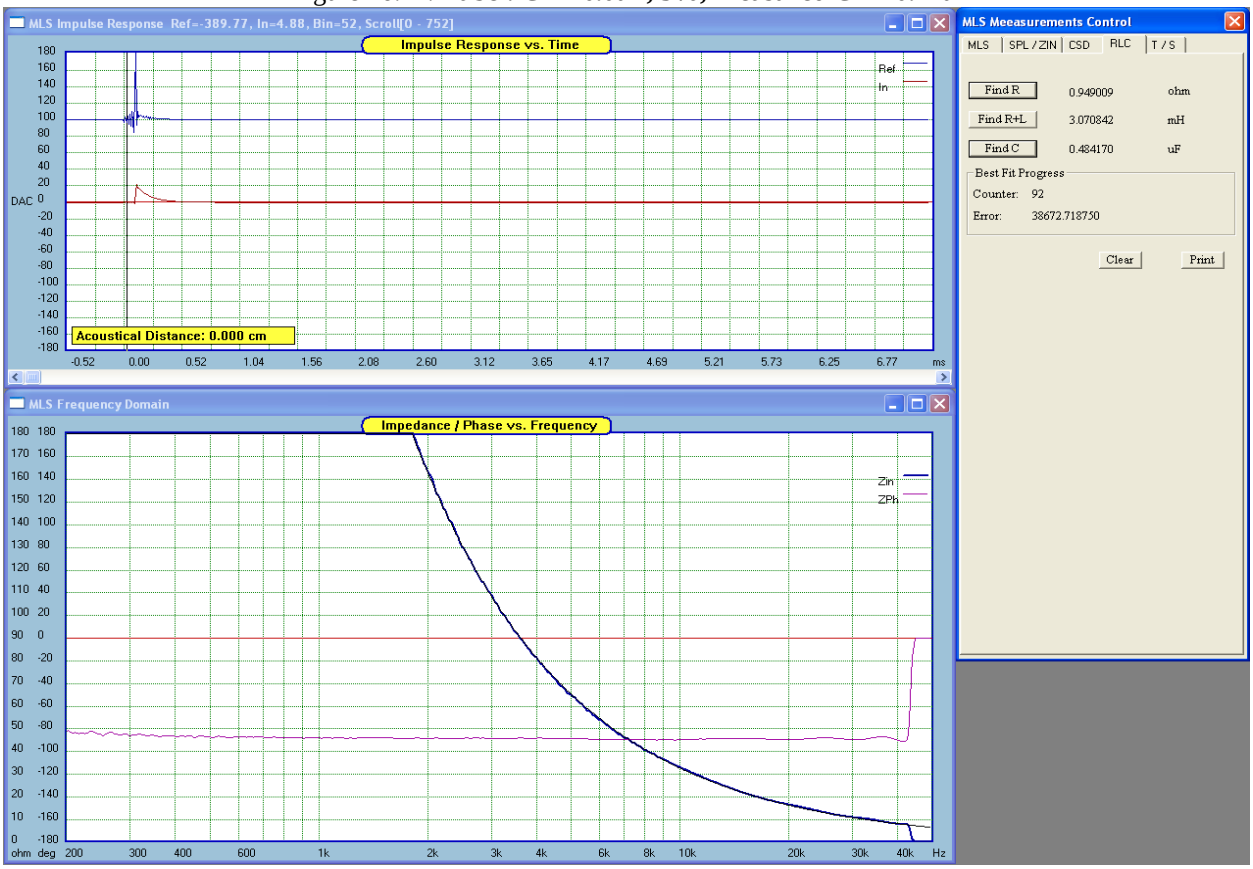


Figure 16.48. **Label: C = 0.470uF, 5%, Measured C = 0.484uF**

Inductors

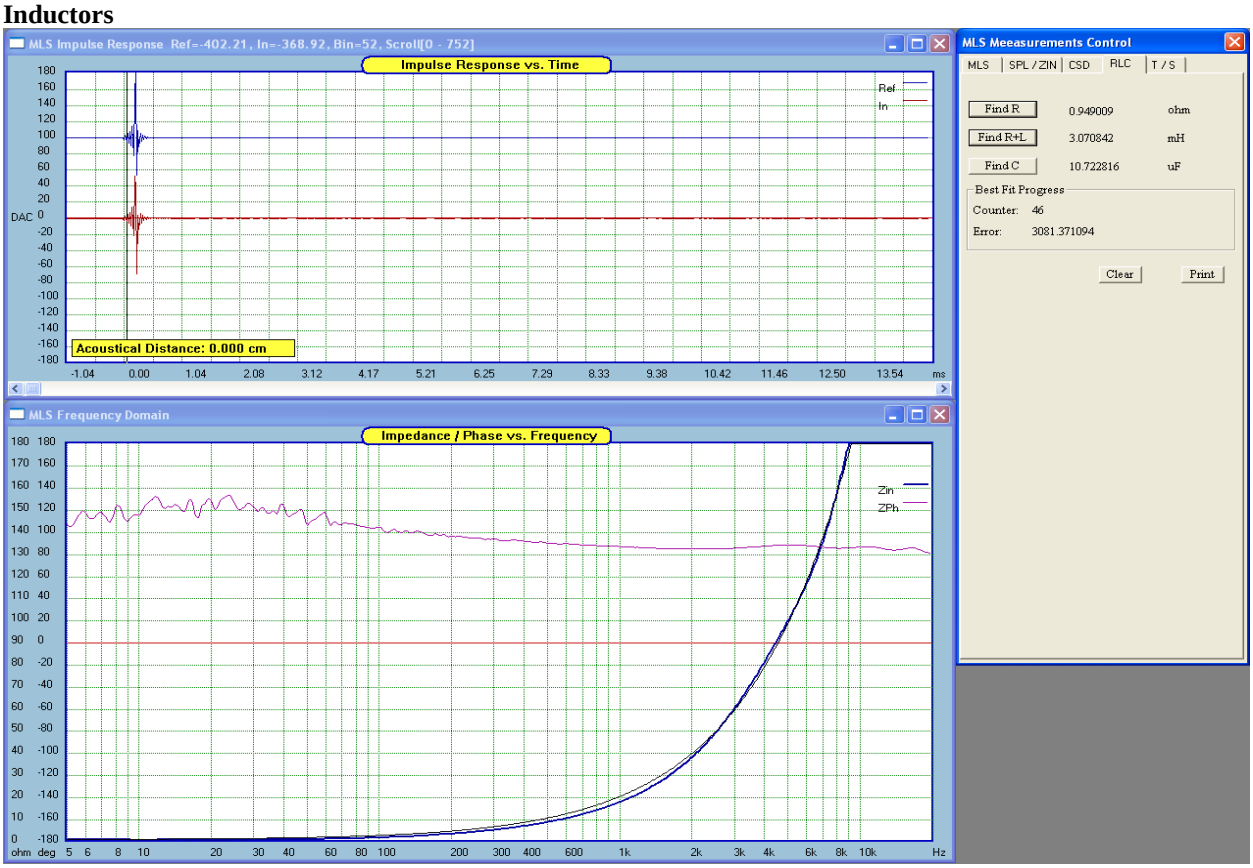


Figure 16.49. **Label: L = 3.0mH, RDC=1.0ohm, Measured L = 3.07mH, RDC= 0.94ohm**

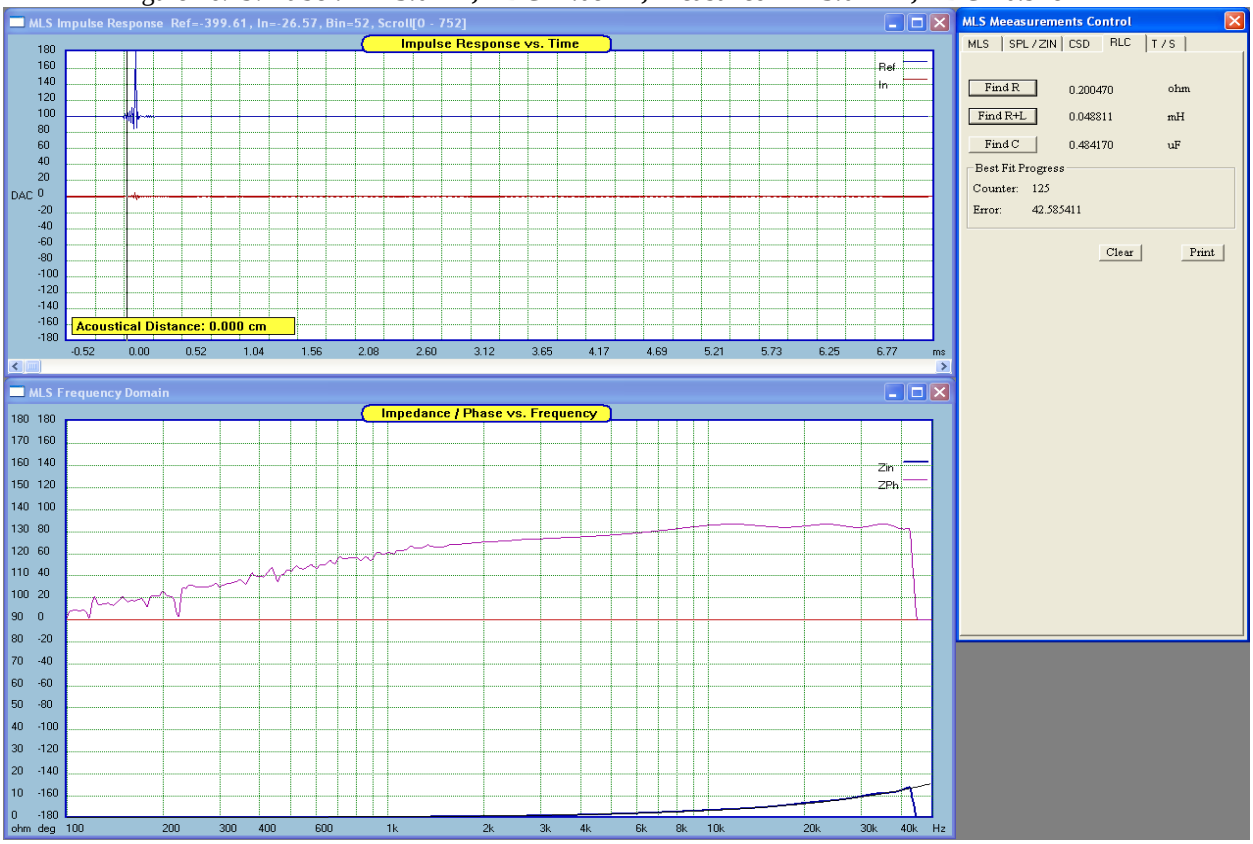


Figure 16.50. **Label: L = 0.05mH, RDC=0.10ohm, Measured L = 0.04881mH, RDC= 0.200ohm**

Single MLS Channel Mode

If you need to determine a frequency response of one of your correction circuits in your amplifier, you may be interested in the "Single MLS" mode of operation of the MLS screen. An example of the usability of this mode would be the frequency response measurements performed on parametric, shelving or other EQ circuits.

In a normal MLS measurement mode, both measurement inputs of the soundcard (Ref +In) are connected somewhere to the output of the audio amplifier. Therefore, no matter what are the internal parametric settings on your amplifier, the "Ref" probe will receive already equalized and corrected signal.

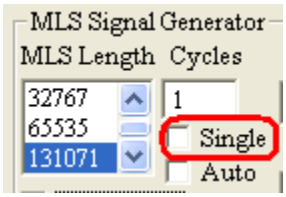


Figure 16.51 Single Mode

The "Single MLS" mode will help in this situation. The concept here is to obtain a reference SPL of your amplifier with flat EQ settings. This signal will be picked by the "Ref" probe in normal MLS mode of operation. Then you would switch the MLS system to "Single MLS" mode by checking the box. Now, the measurements are ONLY taken via the "In" probe and referenced to the previously captured "Ref" signal. As you can see, the "Ref" and In" are now decoupled. Here is the typical usage step-by-step:

1. Run normal, dual-channel MLS measurement with "Single MLS" checked OFF. During this run, the "Ref" impulse is saved in a buffer.
2. Check the "Single MLS" box to indicate, that the next measurement will be using Impulse Response from "In" probe and the last saved "Ref" IR for reference.
3. Run FFT to convert IR -> SPL.
4. "Clear Imp" button re-sets the "Ref" input buffer to a default IR, which is close to the loop-back IR.

There one thing worth mentioning here. When you control your soundcard using a computer program, the PC will signal the current status of the soundcard using Interrupt Requests (IRQs). These are time-dependent events, and it may happen, that subsequent IRQ response from your CPU will be delayed by one sample in comparison to the previous measurements. This simply means, that your stored "Ref" IR will not peak at the same distance to subsequent "In" measurements, and as a consequence, a phase shift occurs between the stored "Ref" and current "In" IRs. If your PC happens to create this event occasionally, there is nothing much you can do about it, except for trying to re-capture the "Ref" IR and hoping, that next "In" impulses will be more stable. Obviously, dual-probe measurements are always correct, because you "In" probe is always referenced to the current "Ref" IR.

Auto Mode

Another very handy function of the MLS measurement screen is the "Auto Mode". Selecting this mode switches the MSL system into continuous measurement/FFT operation. The process runs as follows:

1. MLS system generates the measurement pulse.
2. DUT response pulse is collected in dual-channel measurement mode.
3. Collected IR if converted to SPL using FFT.
4. SPL is displayed.
5. If the "Auto" box is still checked, the process goes to Step 1 to continue the cycle.

Setting up your MLS system to the "Auto Mode" enables you to have both hands free to make your crossover adjustments. You could even position your PC screen close to where you have your crossover hardware and just let the MSL screen run in "Auto Mode". Here is how to run the mode:

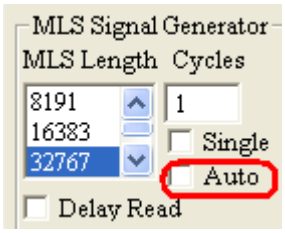


Figure 16.52 Auto Mode

1. Set ALL necessary MLS parameters first.
2. Uncheck "Single" box to indicate, that dual-channel option will be used.
3. Set number of "Cycles" to 1.
4. Check "Auto" box.
5. Press "Run MLS" button. The program will run IR/FFT cycles automatically.
6. To stop the process, **uncheck the "Auto" box**.

Cyclic Mode

In this mode, EasyLab SPL and Zin sections will generate 1 – 10 MLS impulses and will average the results for FFT calculations. This mode promises an improvement in noise immunity. To activate:

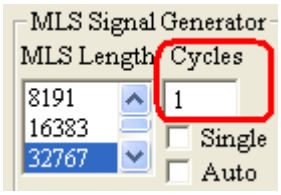


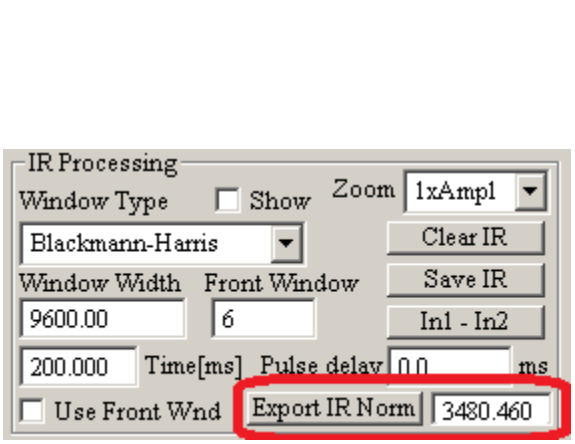
Figure 16.53 Cyclic Mode

1. Enter the desired number of MLS impulses in the "Cycles" data field.
   You must select a number from 1 to 10.
2. Press "Run MLS" button.

Export/Import Impulse Response as ASCII data

If you need to make use of the information embedded in the collected Impulse Response, you can also export the IR into an ASCII file. The file can be read by any text editor, EXCEL and so on. The file is a **single column file** and has the following example format:

-2.59092321182e-006
-1.94087079308e-006
-1.93158432996e-006
and so on …….



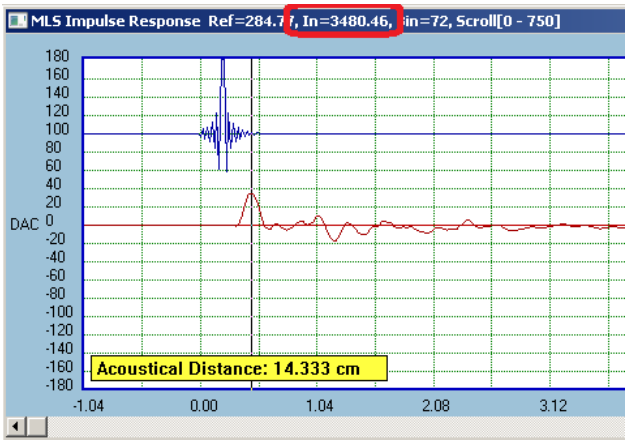


Figure 16.54 "Export IR Normalized" button and normalization coefficient data field.

The exported Impulse Response can be normalized to it's peak value. This is illustrated on Figure 16.54. When the program finishes calculating the IR, it's peak value will be displayed in the editable field, next to the

"**Export IR Norm**" button. This is the normalization coefficient, that will be used to divide all FFT bins by. If you leave it without changing, the IR will be exported as "normalized" vector of numbers in ASCII file. If you change the coefficient to "1.0" the IR will be saved as the measured values. To export the IR simply press the "**Export IR Norm**" button. A standard dialogue box will let you save your file. Importing the previously saved IR in ASCII file is a reverse process. Select "**Use Saved Buffer**" check box and press "**Run MLS**" button. A standard file loading dialogue box will pop up, where you select "TXT" file type. Next dialogue box will allow you to enter multiplication coefficients for both "Input" and "Reference" channels. These will allow you to "de-normalize" the imported IR to the value which will result in LEDs displaying signal level close to the maximum values. Please note, that Reference channel impulse is artificially inserted at FFT Bin 60, to make the imported IR (single-channel) appear as dual-channel measurement system to the rest of the program. Finally, the above process works with ASCII files only.

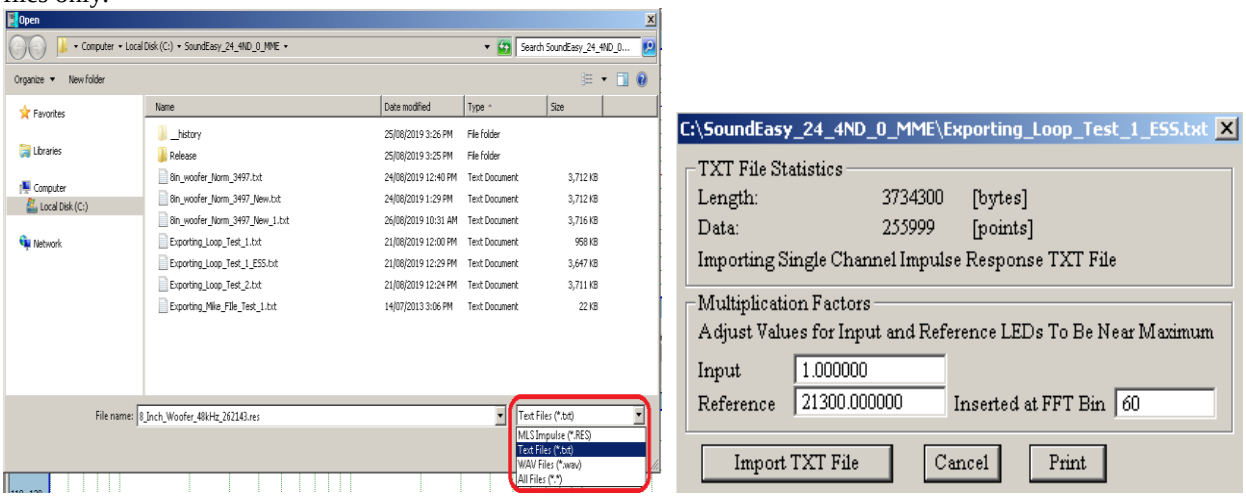


Figure 16.54a. Importing Impulse Response from ASCII files.

"**Delay Read" MLS Measurement mode.**

Allows you to make delayed MLS measurements referenced to stored "Ref" channel.

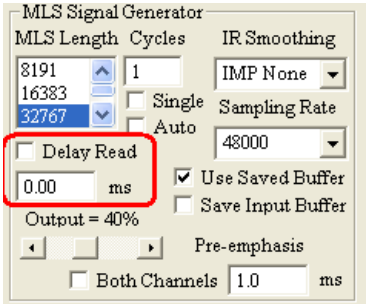


Figure 16.55 Delay Read function

In some measurements, you may experience a significant time delay introduced by the system under test. Situation like this would develop when measuring a loudspeaker at larger distances (1-5ms), when measuring room transfer function (3-20ms) or perhaps when measuring a transfer function of a DSP processor with longer processing delay (1ms -1sec). The **"Delay Read"** function allows you to define the length of time, that the MLS system will wait before processing the incoming system under test response. The delay is defined in milliseconds and is recalculated to sampling intervals. For instance, a delay of 1.0 ms is recalculated to be 48 sampling intervals for 48kHz sampling. The Delay Read function is applicable to "single shot" MLS test signals. Please avoid entering the delay time longer than the actual delay introduced by the system under test. The Impulse Response of the system under test still needs to arrive after (or at best, at the same time as) the reference Impulse Response.

EasyLab settings stored to Preferences file.

Almost all EasyLab settings are saved in the Preferences file. This way, you can close and then re-open the program with the last-saved measurement settings automatically available.

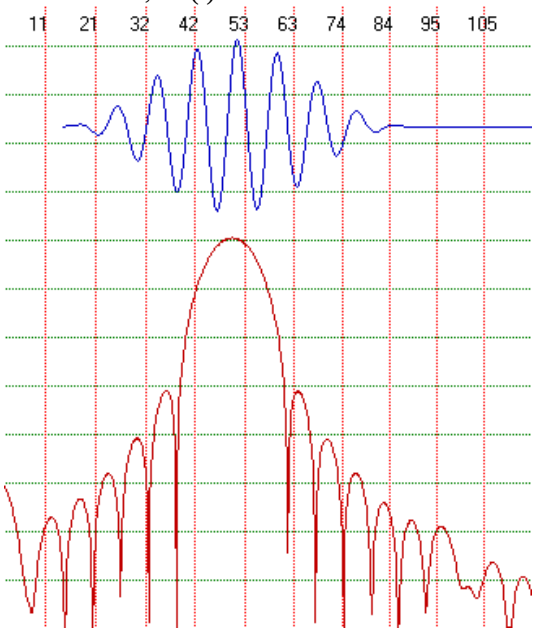
Frequency Warped FFT

FFT Windows

Hanning Window

w (t) = 1 - cos 2 π t /T = 2 sin2 2 π t/T

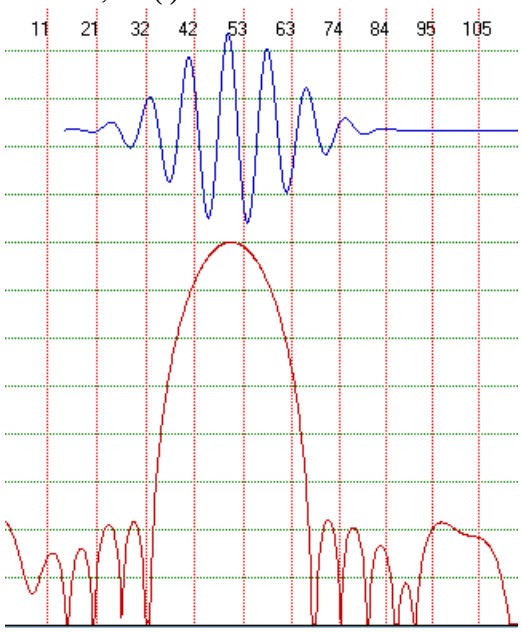
for 0 <= t < T, w (t) = 0 elsewhere



Blackmann Window

w(t) = 0.42 - 0.5*cos2 π t /T + 0.08*cos4 π t /T

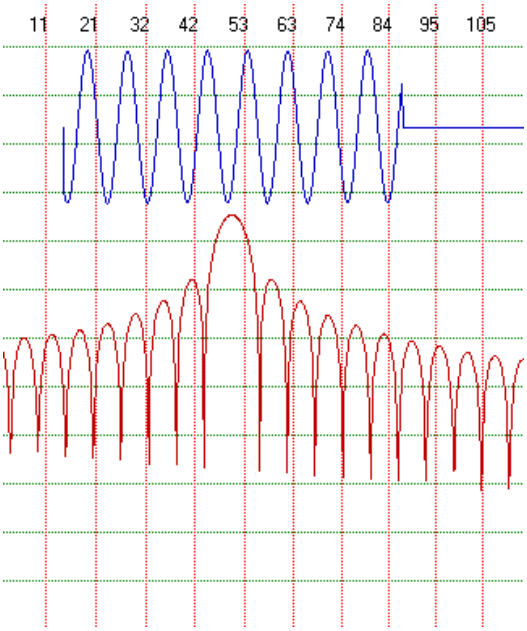
for 0 <= t < T, w (t) = 0 elsewhere



Rectangular Window

w (t) = 1 for 0 t < T

w (t) = 0 elsewhere

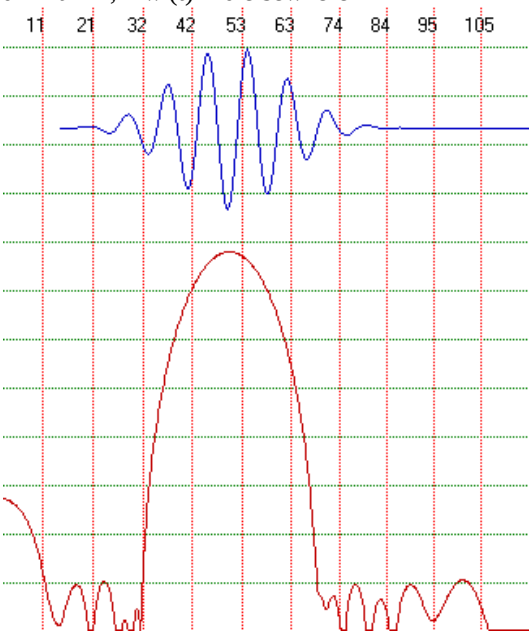


Kaiser-Bessel Window

w (t) = 1 - 1,24 cos 2 π t/T +

0,244 cos 4 π t/T- 0,00305 cos 6 π t/T

for 0 =< t < T, w (t) = 0 elsewhere



Window	Noise BW	3dB BW	Ripple	Highest Sidelobe	Sidelobe fall-off/Dec	60dB BW	Shape Factor	
Rectangular	Δf	0.89Δf	3.92dB	-13.3dB	20dB	665 Δf		750
Hanning	1.5Δf	1.44Δf	1.42dB	-31.5dB	60dB	13.3 Δf		9.2
Kaiser-Bessel	1.8Δf	1.71Δf	1.02dB	-66.6dB	20dB	6.1 Δf		3.6

The choice of windows will result in somehow different looking spectrum. **All windows are simply a compromise between the undesired level of side-lobes and 3dB bandwidth.**

Frequency Warping Technique

Standard FFT is computed with uniform frequency increments (or time intervals).

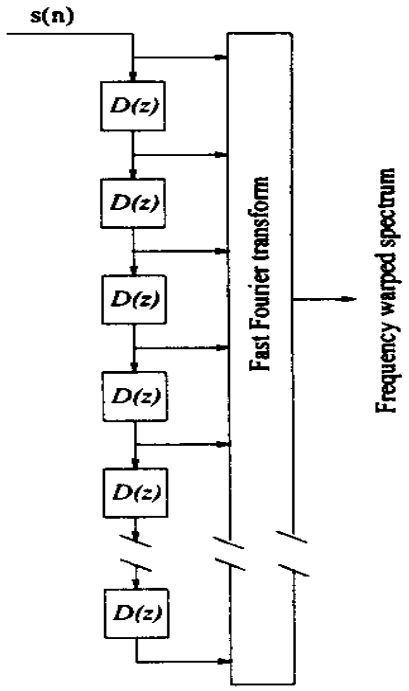
Δf = Sampling frequency/ FFT length

For instance, sampling with 48000Hz and using FFT length of 16384, Δf = 2.929Hz. Incidentally, Δf also determines how well two, close-frequency spaced spectral components can be separated in the resulting spectrum. In our example, two signals of 20Hz and 22 Hz would not be clearly separated by the FFT.

The technique for computing nonuniform-resolution FFT was developed by Oppenheim at el. (A.V. Oppenheim, D.H. Johnson and K. Steigitz, "Computation of Spectra with Unequal resolution Using the Fast Fourier Transform", Proc IEEE, vol. 59, pp299-301 (1971 Feb). Here, the signal is fed into a chain of all-pass filters, simulating the delay line, and then processed directly by the FFT.

Transfer function of the first order all-pass filter is given as:

$$D(z) = \frac{z^{-1} - \lambda}{1 - \lambda z^{-1}}$$



The λ in the above equation is called "warping parameter" and allows for different frequency transformations. The mapping from natural frequency domain to a warped frequency domain is described by the phase response function of the all-pass filter.

$$\omega' = \arctan\frac{(1-\lambda^2)\sin(\omega)}{(1+\lambda^2)\cos(\omega)-2\lambda}$$

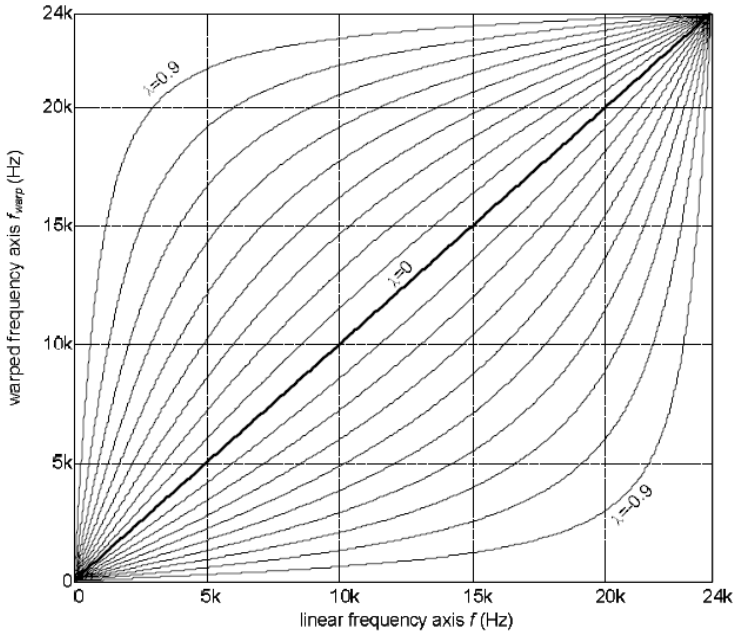


Figure 16.61. Warping effect for several values of λ

When the collected sound sample x[n] is fed into an all-pass filter chain with positive λ, the signal propagates through the chain with the delay determined by the filter's phase response. This delay is longer for lower frequencies, and makes the low frequency components proceed slower than higher frequency components. This process is effectively a frequency-dependent re-sampling of the input signal.

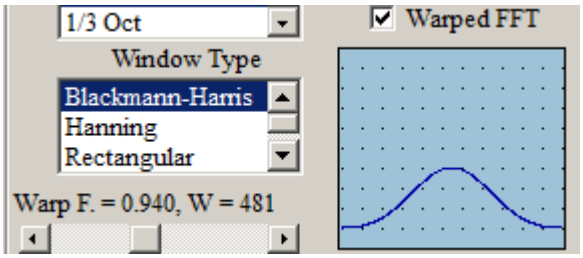
In other words, by changing the warping parameter λ, we can zoom-in on a given frequency range – typically, we would appreciate better resolution at lower frequency range.

When sampling with 48000Hz, and using 16384-bin long FFT, one can collect 16834/48000 = 341.3ms long sound sample for processing by FFT. This way, when trying to capture 20Hz waveform (wavelength = 50ms), one can collect 6.826 cycles of the 20Hz signal in one input buffer. These whole 6.826 cycles will be mapped onto another frequency, determined by the warping parameter. However, ONLY the 6.826 cycles will be mapped. Say for instance, the choice of warping parameter resulted in mapping 20Hz onto 1000Hz out of the delay chain. Corresponding 6.826 cycles of 1000Hz (wavelength = 1ms) have only 6.826 ms duration. Consequently, we must window out ALL remaining data outside 6.826ms before fetching the collected sample into the FFT process.

Warped FFT (WFFT) is not a panacea for significantly increasing FFT resolution at the low frequency end. This is because the FFT can only resolve Δf = (Sampling frequency/ FFT length) anyway. Certainly, we can greatly improve resolution of the WFFT by increasing the FFT length. But the same is also true for the standard FFT.

However, in many applications WFFT allows for improved spectrum presentation by reducing baseline noise level and somewhat better discrimination between closely spaced frequencies.

Warped FFT Controls



Warped FFT – Activates Warped FFT Mode.
Window Type – This list box allows selection of 4 types of FFT windows.
Warp Factor – To select this parameter, you must turn OFF Signal Generator and Spectrum Analyser. Warp Factor can be adjusted between 0.90 and 0.995.

IMPORTANT NOTE: WFFT is extremely computationally intensive. The lower the Warp Factor, the more your CPU will be engaged. The program defaults to Warp Factor = 0.950. If your PC stops operating for some selections of Warp Factors, you may be left with no choice, but to re-boot the PC.

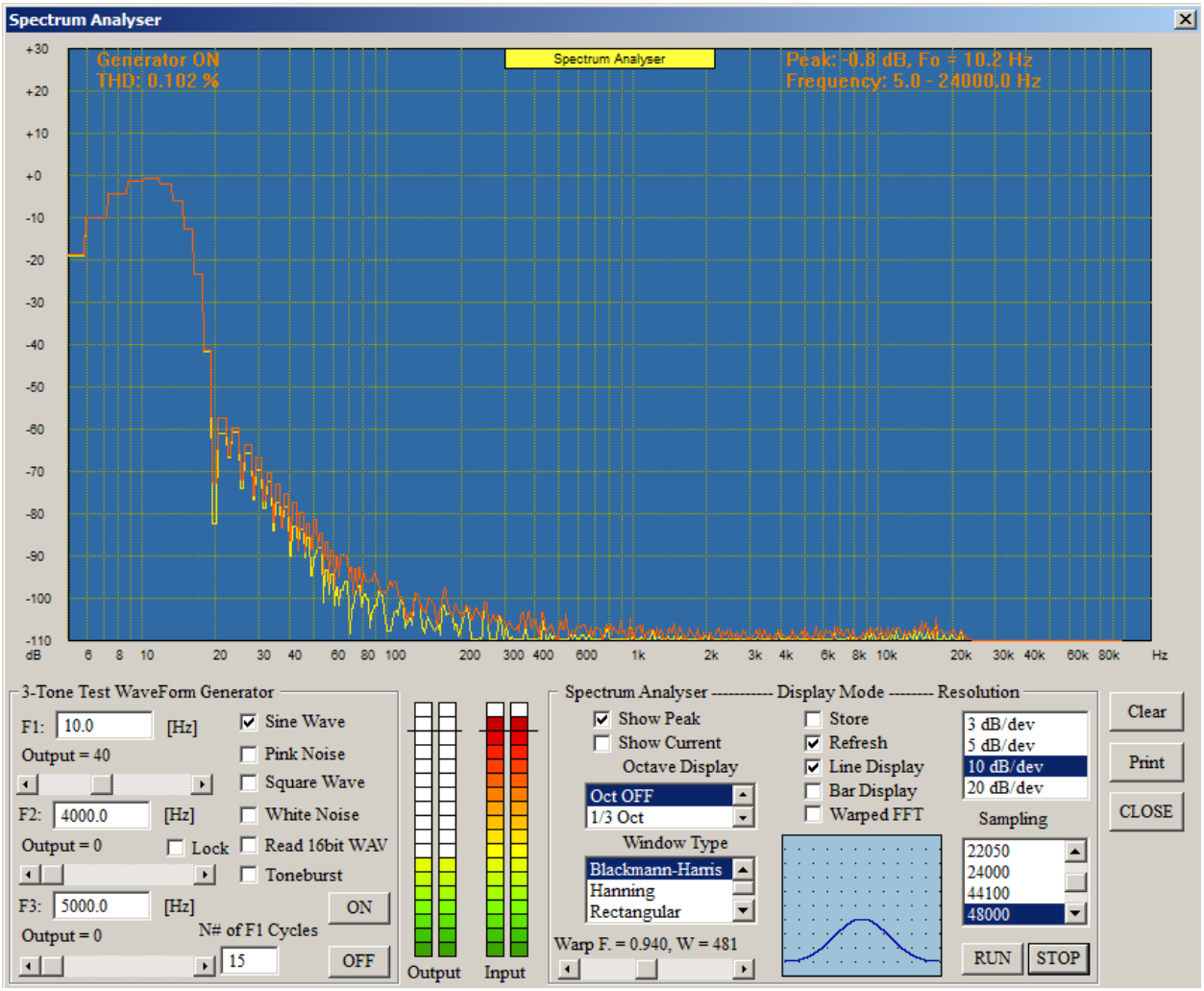


Figure 16.62. 10Hz sine wave displayed in Standard FFT Mode.

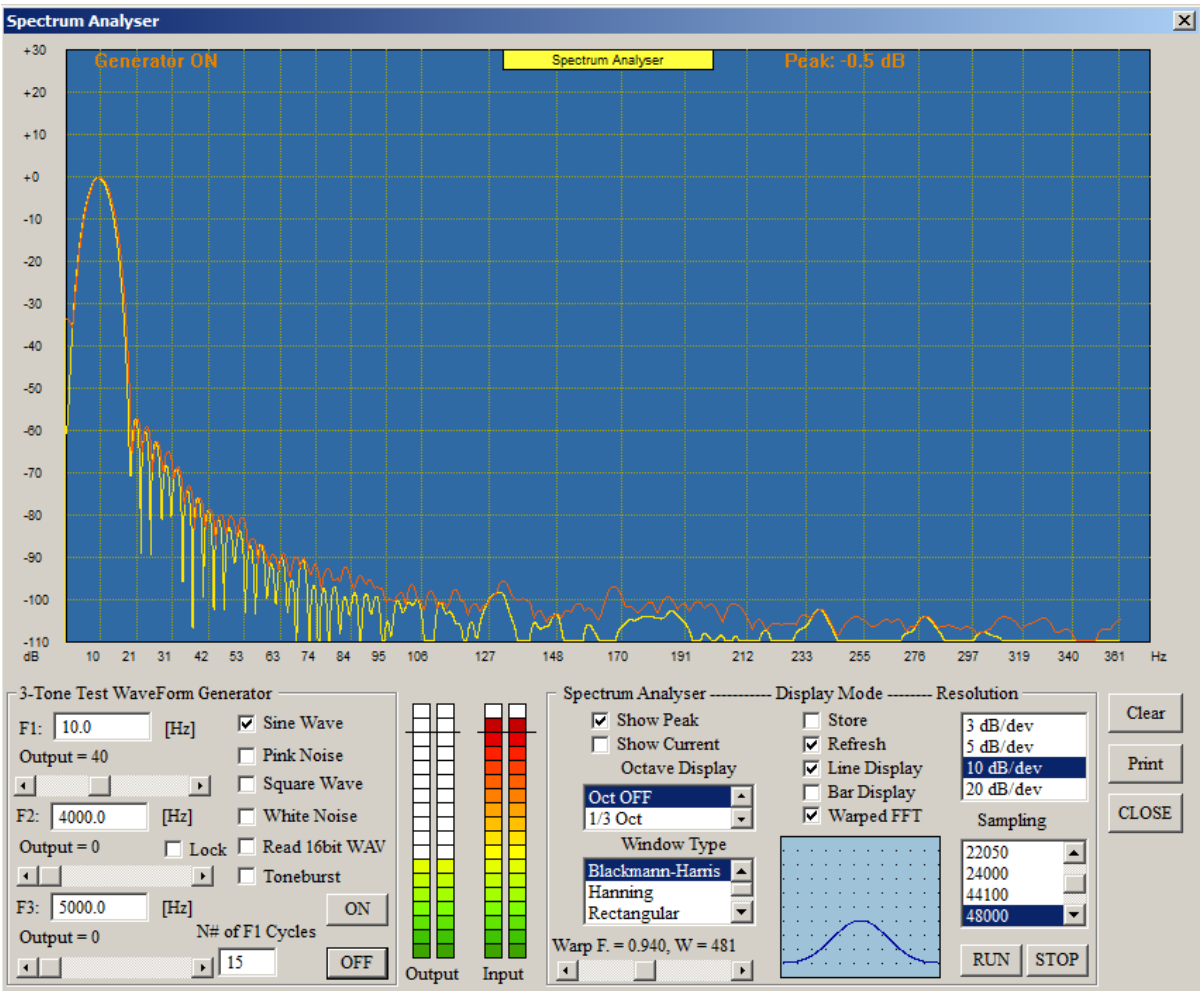
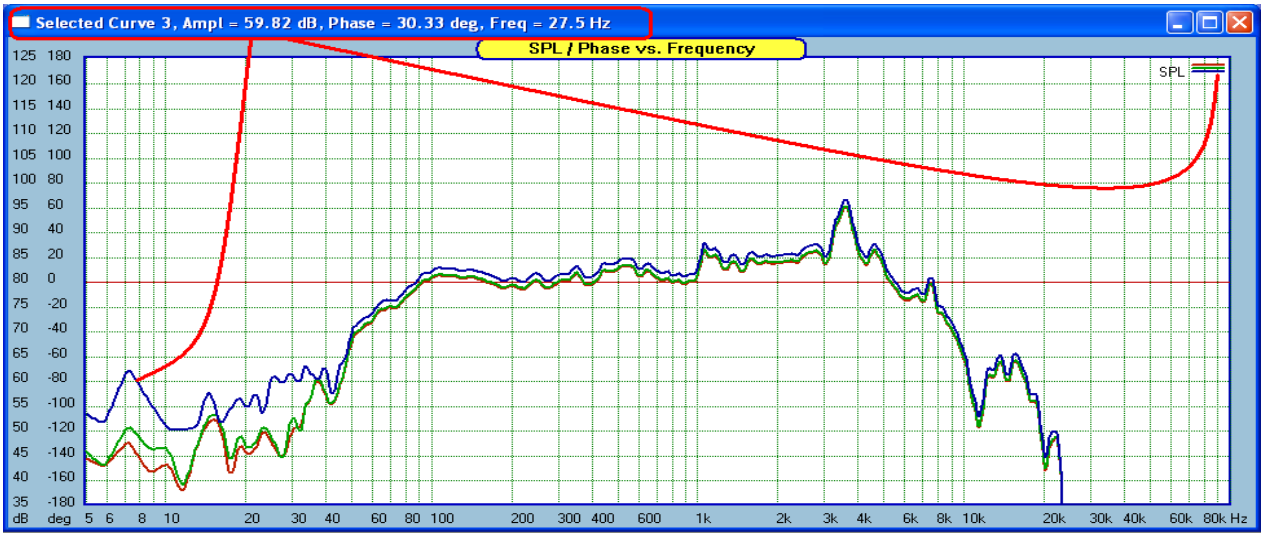


Figure 16.63. 10Hz sine wave displayed in Warped Mode.

Attaching Cursor to SPL Curves in MLS System

SPL curves generated in MLS measurements can be closely examined by attaching a cursor readout to any of the SPL curves. Simply click on the desired SPL curve, and detailed SPL, Phase and Frequency data will be displayed in the window's frame – see below.



MLS measurements TAB screens

There are 7 TABs, each one allocated to display individual measurement results. TABs can be activated manually by simply selecting the TAB with the mouse. However, TABs are also activated automatically, when you change processing function. For instance, “Step Response” TAB will be activated automatically and focused when you attempt to display driver’s step response (press Step Response button on the post processing screen).

1. **SPL + Phase** – this TAB will show measured SPL and phase. It is also used to display HBT results, because SPL and HBT are best viewed on the same screen.
2. **Zin + Phase** - this TAB will show measured Zin and phase. It is also used to display HBT results, because Zin and HBT are best viewed on the same screen.
3. **Step Response** - this TAB will show measured step response.
4. **Mike + Preamp** - this TAB will show SPL and phase of the microphone and pre-amplifier.
5. **RLC** – this TAB shows curves associated with RLC component measurements.
6. **Compare in Freq** – this TAB accumulates curves transferred from other screens for comparison. Curves on this TAB can be erased and labeled for improved presentation and documentation.
7. **Post Processing** – this TAB shows all post processing curves’ details.

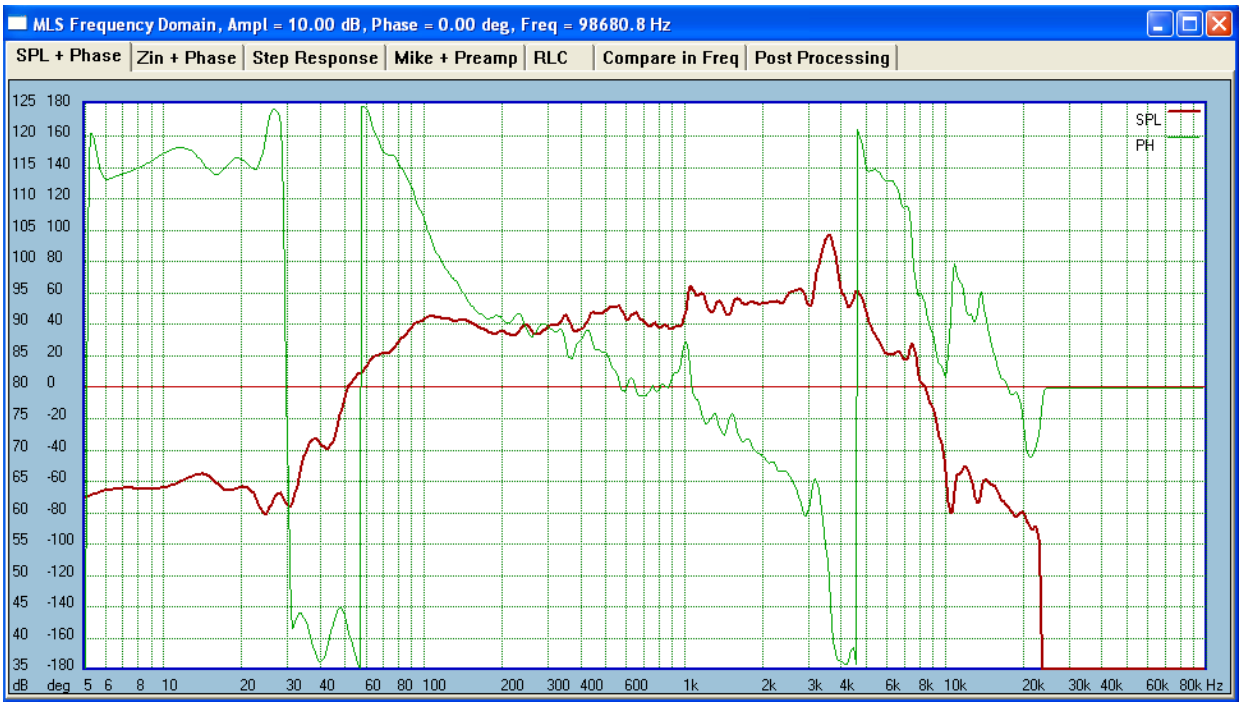


Figure 16.64 TAB’ed screens in MLS system.

Spectrograms

Several methods exist for calculating spectrograms, including the windowed FFT method (WFT), and the Wavelet Transform.

SoundEasy employs the windowed FFT method and produces a spectrogram by applying a data window to the impulse response and computing the FFT of the windowed data. The window is progressively shifted in time to obtain a frequency response for each discrete time.

In the example below, there are two "ideal" impulse responses, separated by 5ms, fed into the Spectrogram function.

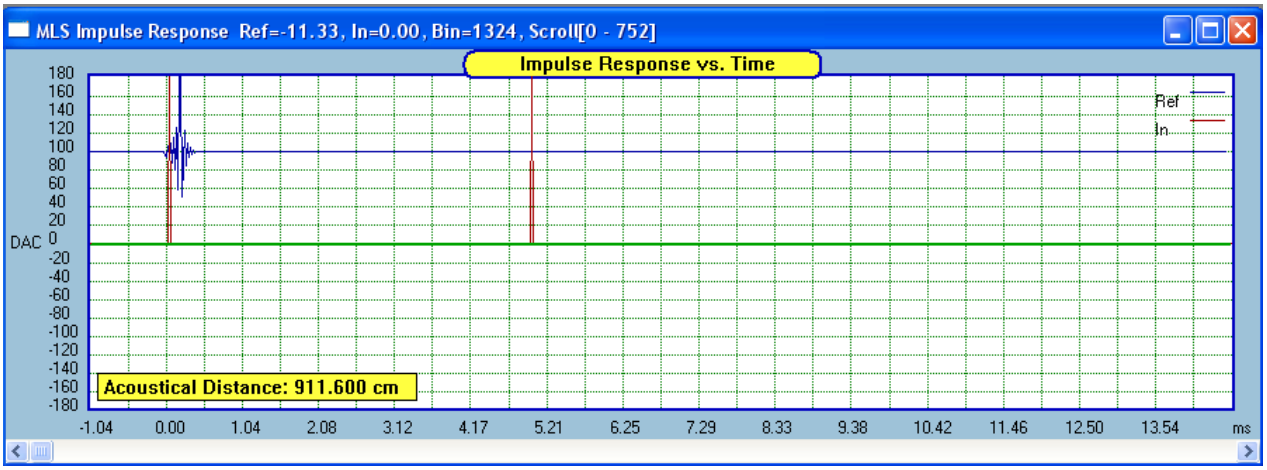


Figure 16.65. Two ideal impulse responses.

A user-selected window is then positioned at the front of the first impulse response, and progressively shifted, while computing FFT for each window location. Figure below illustrates this concept.

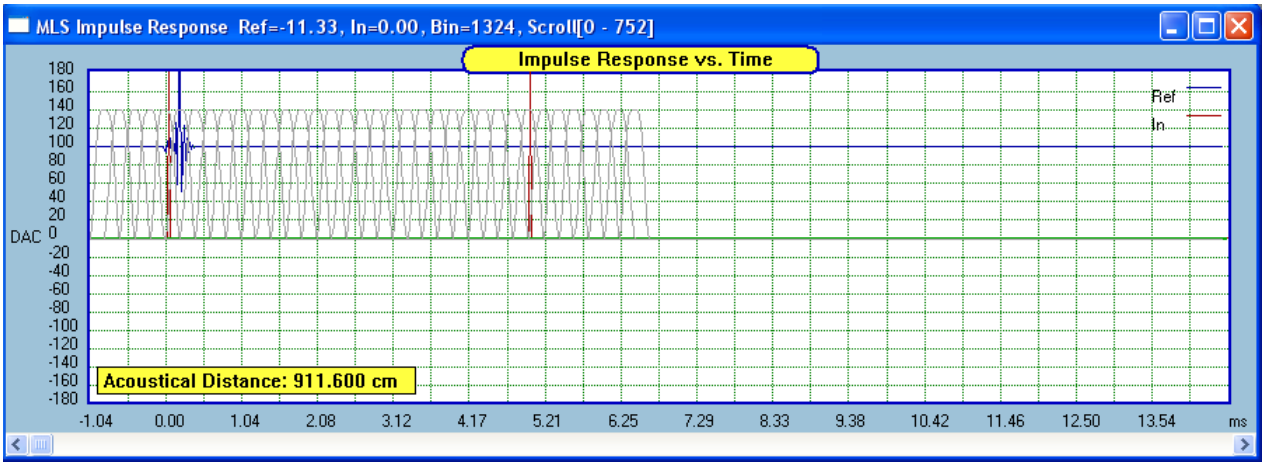


Figure 16.66. FFT window progressively shifted for each FFT.

Finally, collected frequency domain data is collated on time-frequency-amplitude screen shown below. As expected, there are two lines corresponding to the location of the two impulse responses on time axis.

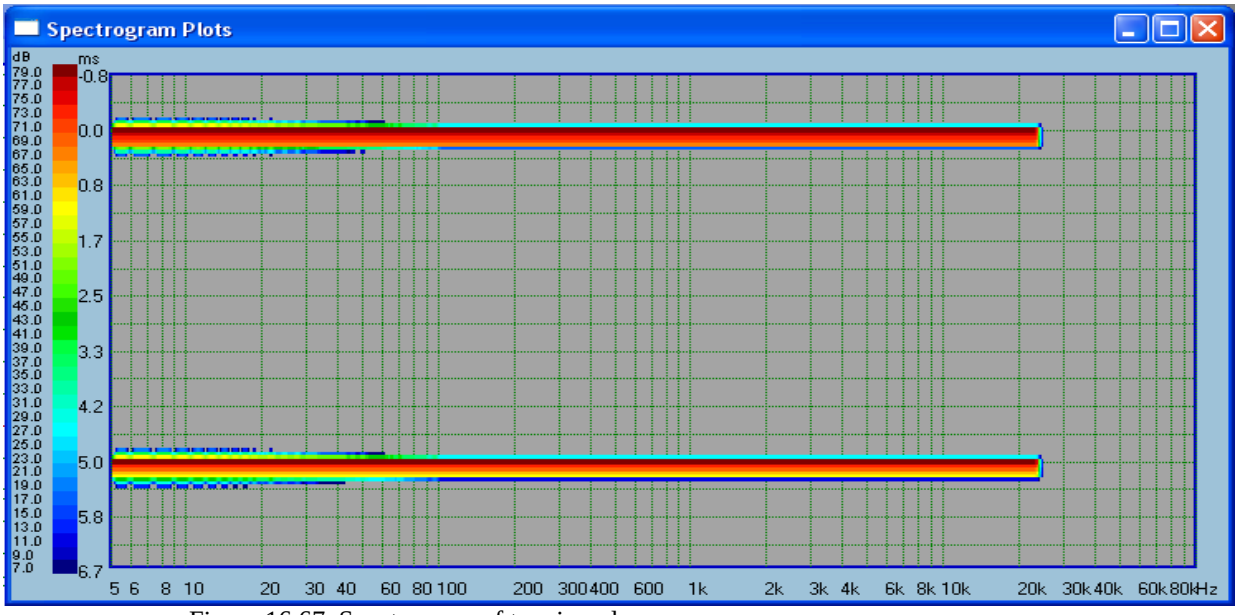


Figure 16.67. Spectrogram of two impulse responses.

Next example is a 12” guitar speaker, with it’s impulse response and corresponding frequency response shown on figures below.

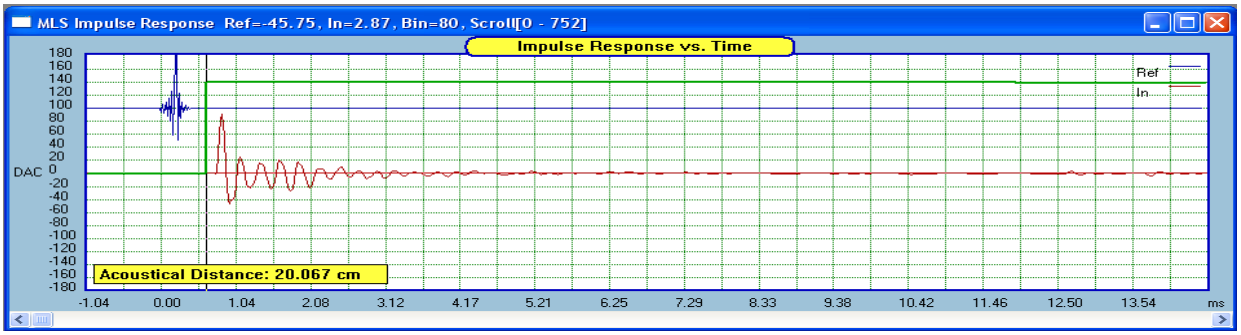


Figure 16.68. A 12” guitar speaker impulse response.

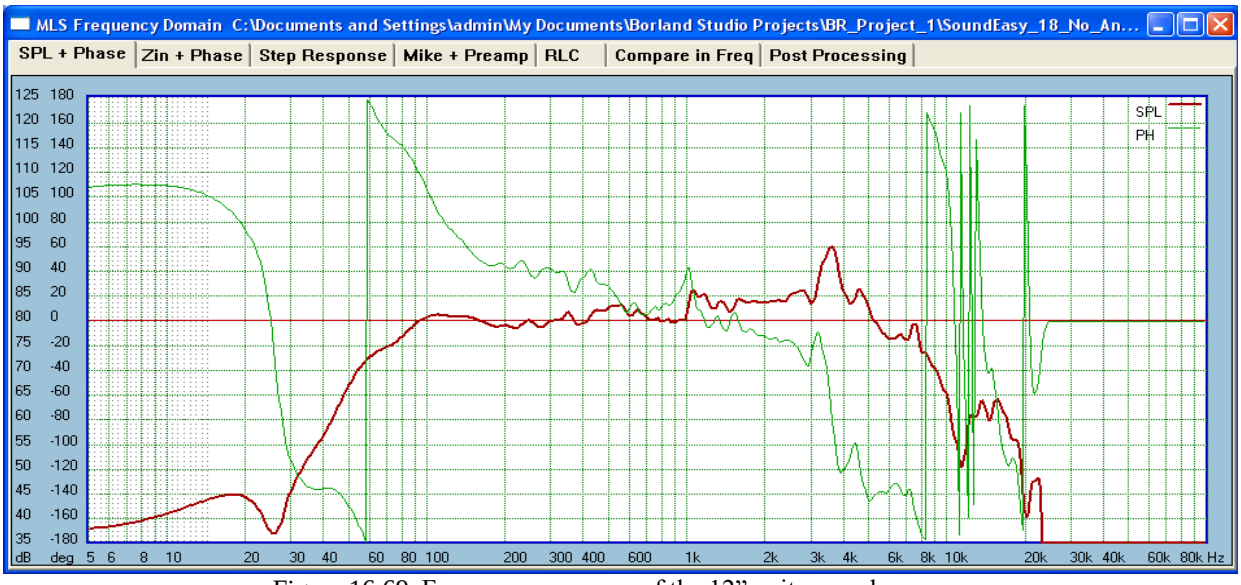


Figure 16.69. Frequency response of the 12” guitar speaker.

The impulse response is now converted to spectrogram, by the same method.

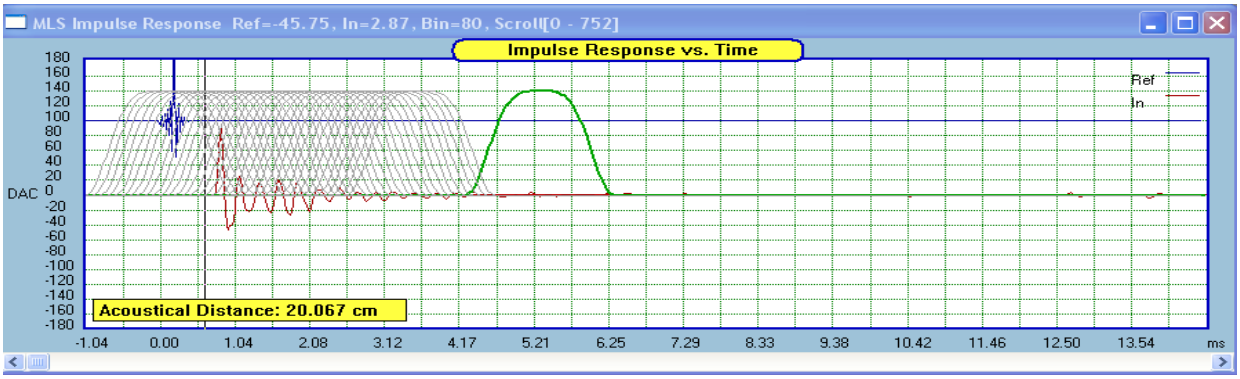


Figure 16.70 Shifted FFT window

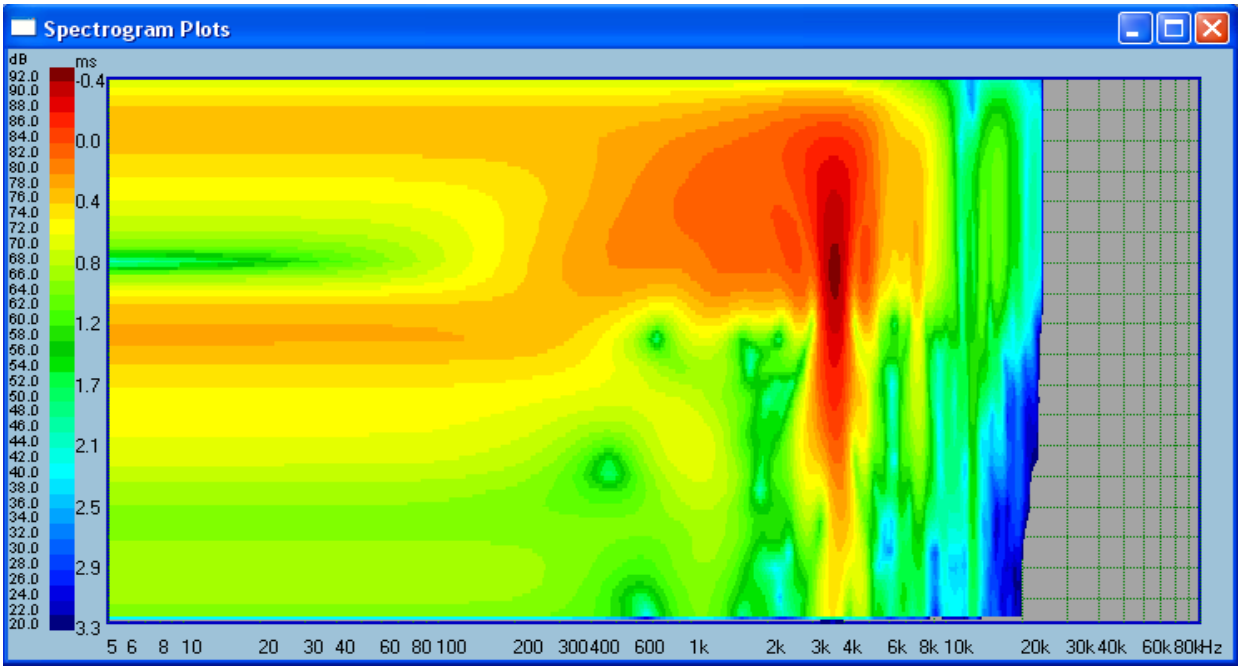


Figure 16.71. Spectrogram of the 12” guitar speaker.

It is observable on Figure 16.71, that this loudspeaker has a problem around 3.5kHz. This is exactly where the peak on SPL curve is located. The spectrogram above shows prolonged and sustained energy around the 3.5kHz.

In the next example, a waveguide loudspeaker is examined.

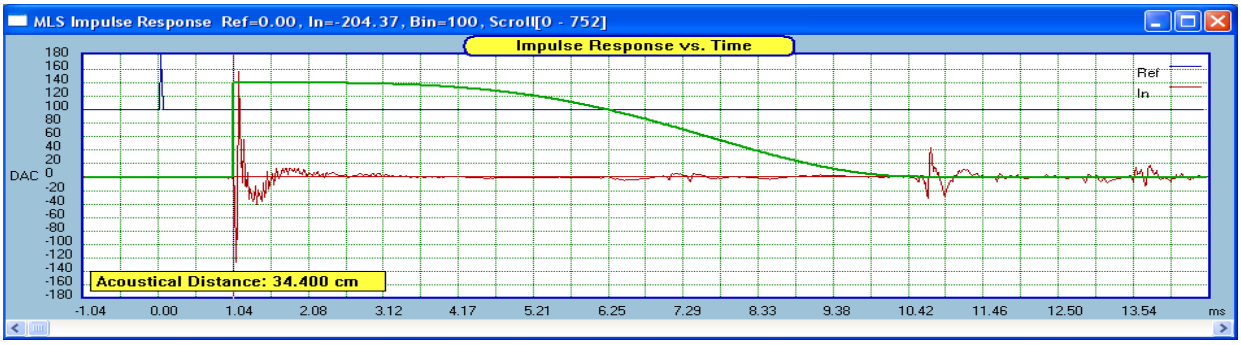


Figure 16.72. Impulse response of a waveguide loudspeaker.

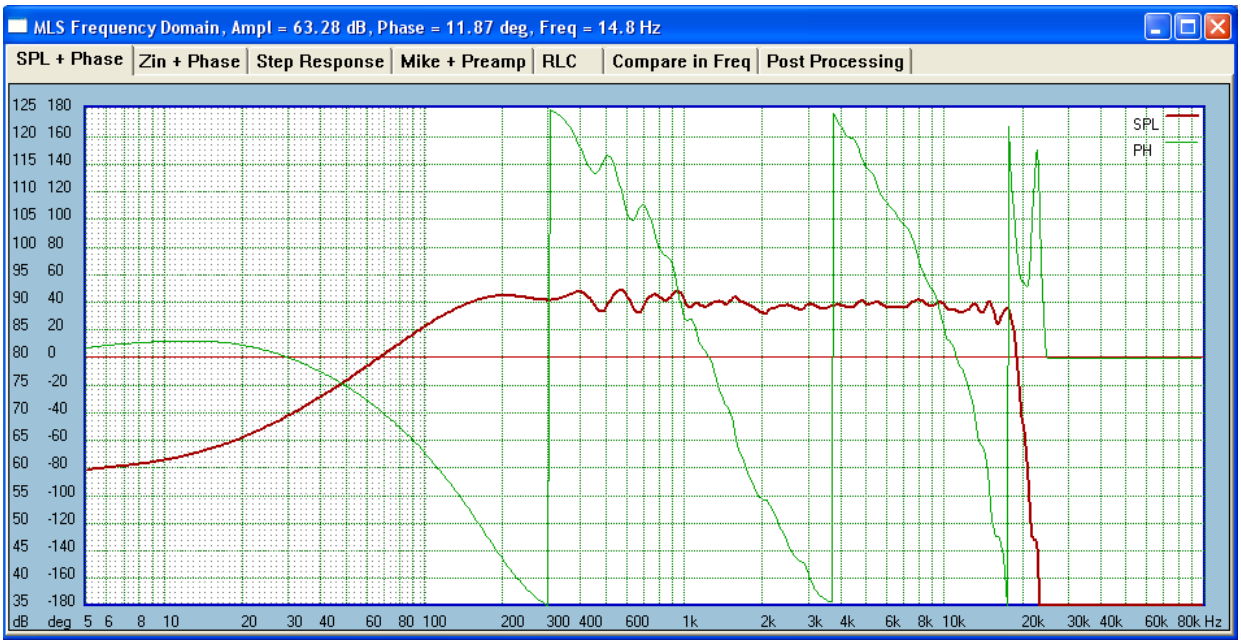
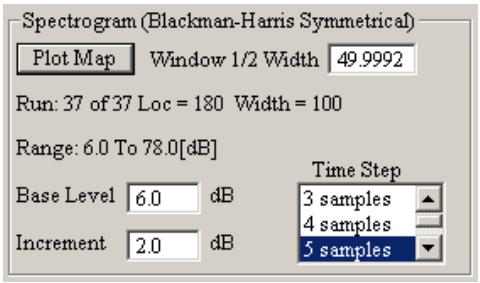


Figure 16.73 calculated SPL

The Spectrogram Function will display 36 discrete levels, starting from the **Base Level**, with **Increments** entered in "Spectrogram" section – see below. The SPL increments are also user-editable. The type and width of the FFT window used in Spectrogram process is also user-defined. Finally, the Time Step (or time shift) is expressed as number of samples.

Only **symmetrical Blackman-Harris windows** are used. Size of the window is entered from the "**Window ½ Width**" data entry field (the same as entered from MLS screen). The program automatically adds the other half, and makes it into a double-sided window.



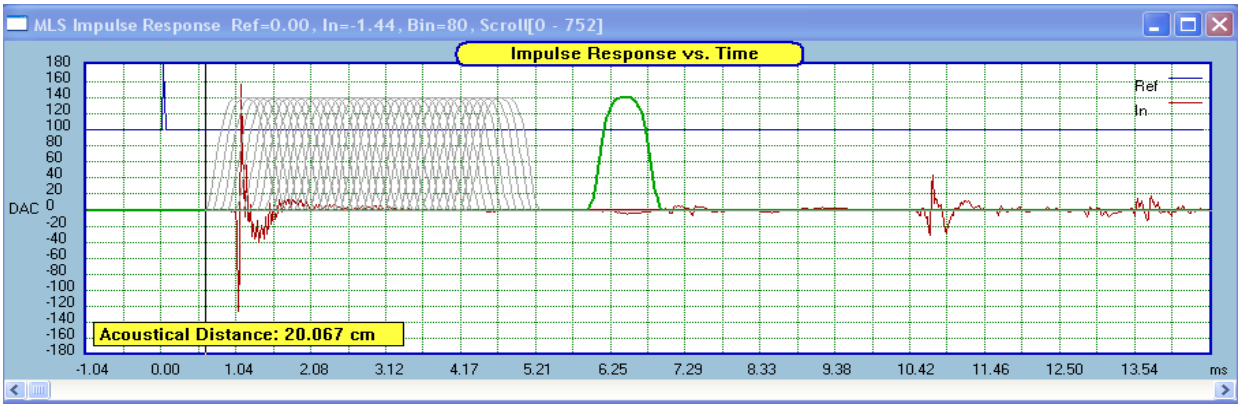


Figure 16.74. Spectrogram mechanics for settings entered above.

The window needs to be centered on the peak of the impulse response. You can click on the “MLS Impulse Response” window and use mouse or keyboard Arrow Keys to move the window into the desired location.

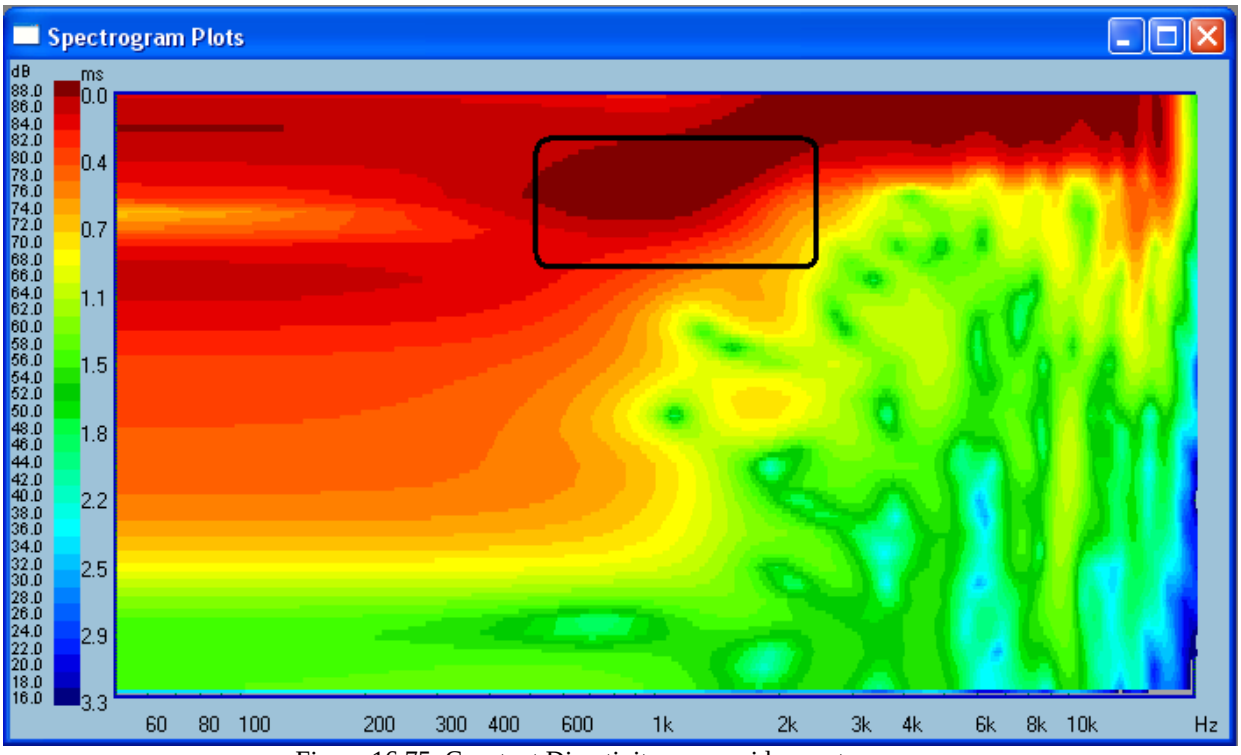


Figure 16.75. Constant Directivity waveguide spectrogram.

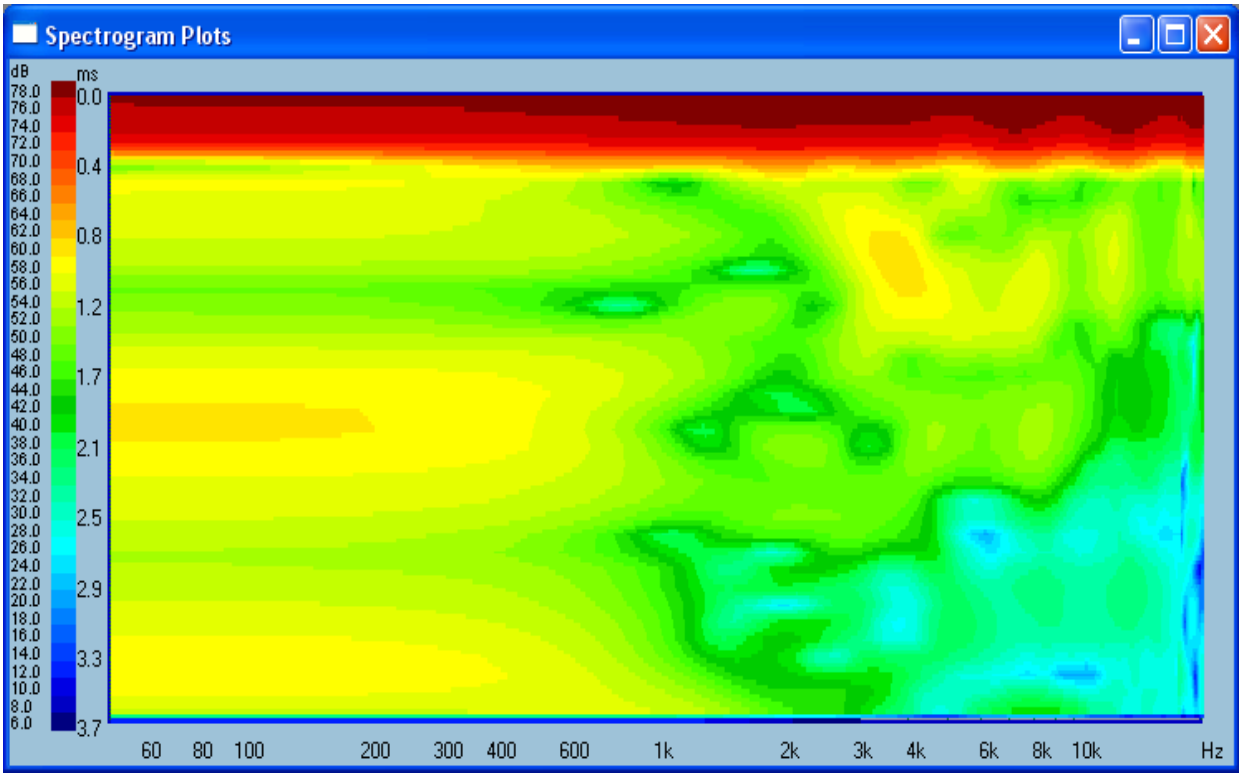


Figure 16.76. UE4 equalized loudspeaker spectrogram

Setting up CSD Plots

There is a subtle difference in the way the FFT half-window is positioned for SPL plotting and for CSD plotting.

Correct position of the FFT window for plotting SPL is shown on the figure below. You will notice, that the window is shifted 8 bins to the left, to capture the same, left-hand portion of the Measured Impulse Response as it is automatically arranged for the Reference Impulse Response collected via reference channel. This way, all digital artefacts are included in the measurement process, and more importantly the phase response is referenced correctly.

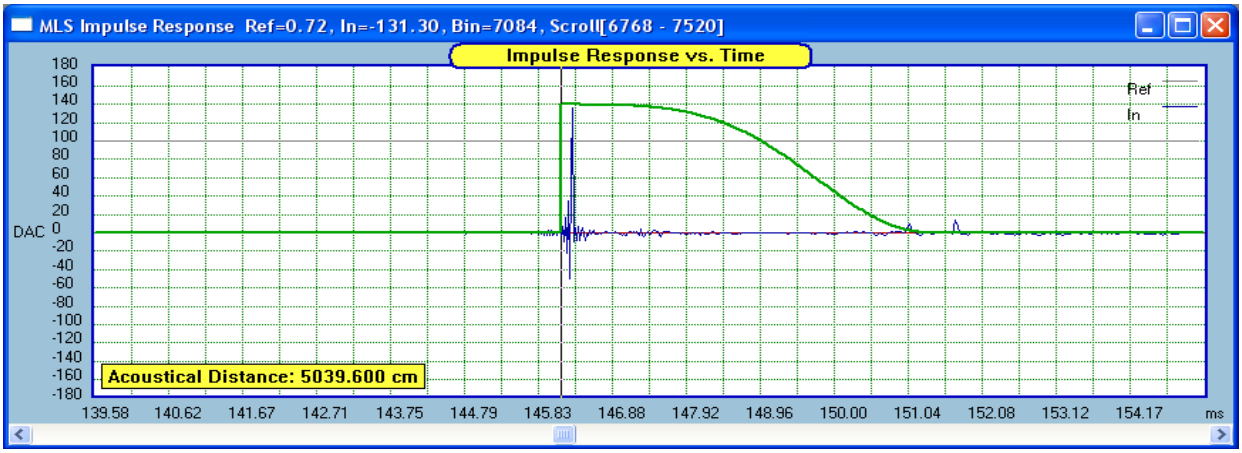
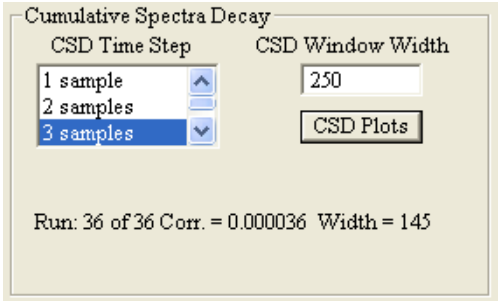


Figure 16.77. Positioning FFT window

Positioning of the FFT window for the CSD process is slightly different. In each pass, the CSD algorithm shifts the FFT window ever slightly more to the right by the CSD Time Stamp and also shortens the FFT window by the same amount, so that room reflections are not being accidentally captured.

For example, if the CSD Time Stamp is selected as 3 steps (see below), and the FFT window is positioned 8 steps (bins) to the left, the impulse response will be re-captured the first 3 passes of the algorithm, before the FFT window finally moves to the right of the IR peak and continues further this way. This is quite unnecessary, and possibly confusing, as we are accustomed to see only one full frequency response plot on the CSD plots – the rest of them are "waterfalls" derived without the IR peak.



It is therefore recommended to locate the peak of the IR first. This can be accomplished by moving the start of the FFT window via cursor keys and observing the "In = xxxx.xx" displayed in the window's top blue border frame. The "Bin=xxxx" displays the location of the peak.

Having found the IR peak, move the start of the FFT window **one step to the left.** This way, the very first pass of the CSD algorithm will capture the full frequency response (this is hard coded in the algorithm) and in the next pass, 3 steps are added – thus moving the FFT window straight to the right-side of the IR peak, creating the first "waterfall". The initial position of the FFT half-window and the subsequent shifts are shown on the Figure below.

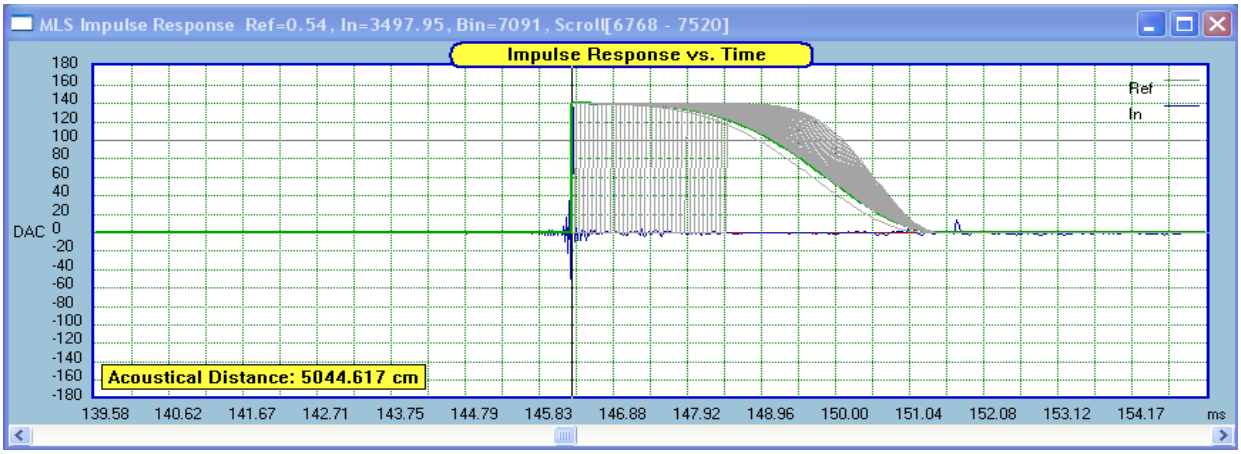


Figure 16.78. CSD algorithm shifts the FFT window

And this is the complete CSD diagram, with the frequency response plots as the first (reference) plot.

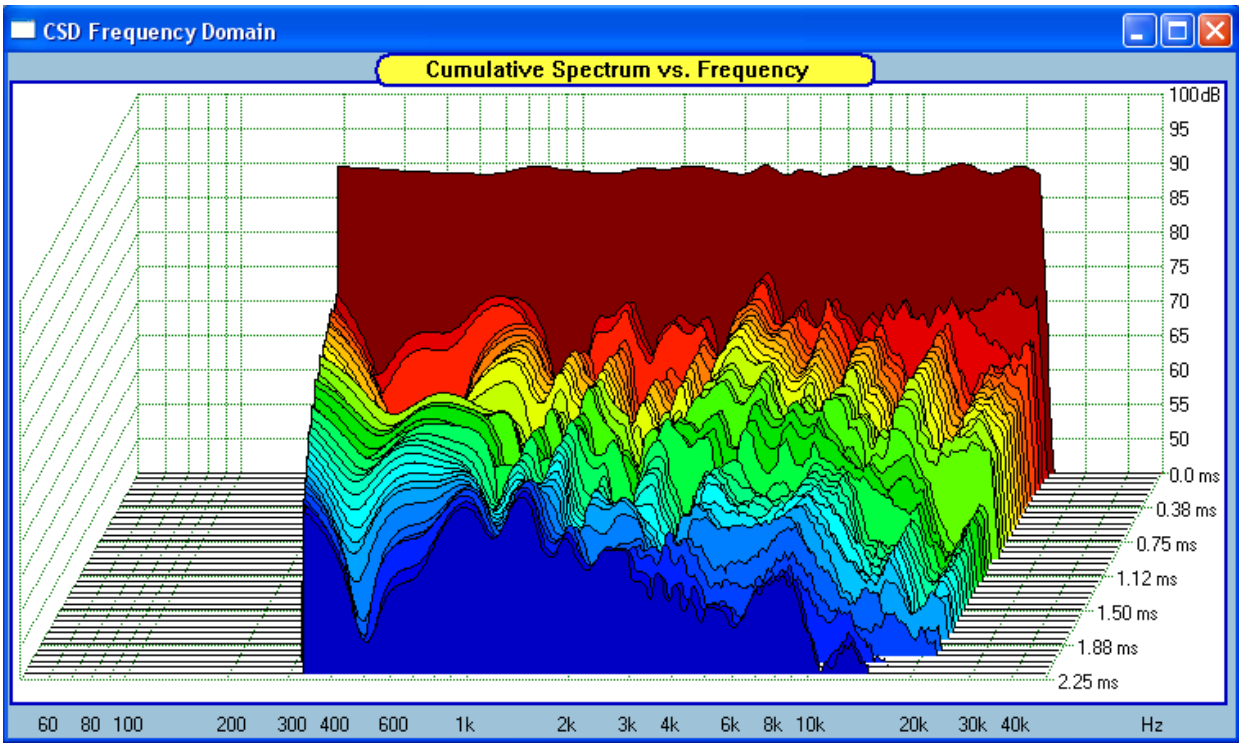
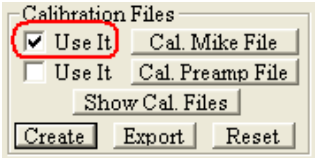


Figure 16.79. Resulting CSD plots

MLS Calibration Files

Acoustical measurements of loudspeaker drivers involve usage of microphones and their pre-amplifiers. A good quality measurement microphone will include SPL/phase response of the microphone, so these can be accounted for during the actual measurements. Apart from importing calibration data directly into the program, you can also re-create calibration file data and save them to you hard disk for future usage.



For creating microphone calibration data, please check the "Use It" box as shown above and then press "**Create**" button to open the following dialogue box:

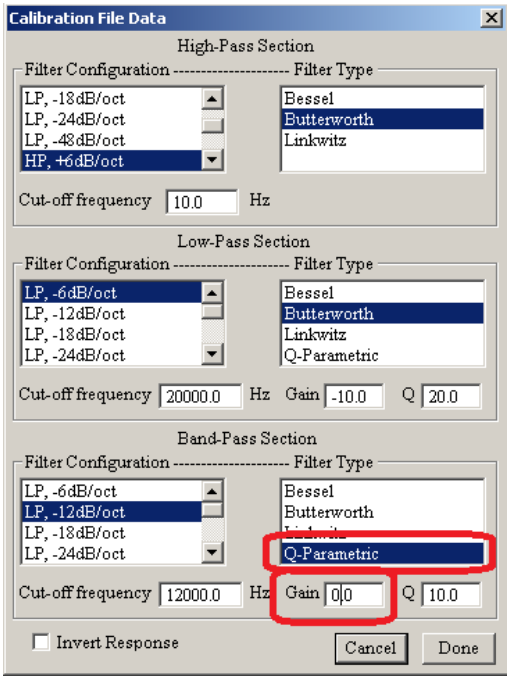


Figure 16.80. Calibration Data dialogue box.

Typically, to construct your microphone calibration file, you would use High-Pass and Low-Pass sections of the dialogue box. In this case, please select Q-Parameter and enter "Gain" parameter in the Band-Pass section as 0.0. This will effectively disable the Band-Pass section (see above). If your microphone requires some more elaborate treatment in the higher frequency range, you can use Band-Pass section to add some peak or dip in the frequency response. Data-entry fields are self-explanatory.

Here is an example of Frequency/Phase data as created by the dialogue box data shown above.



Figure 16.80. Microphone Calibration Data created by the above dialogue box

To save the newly generated microphone calibration data, press "**Export**" button to open the following dialogue:

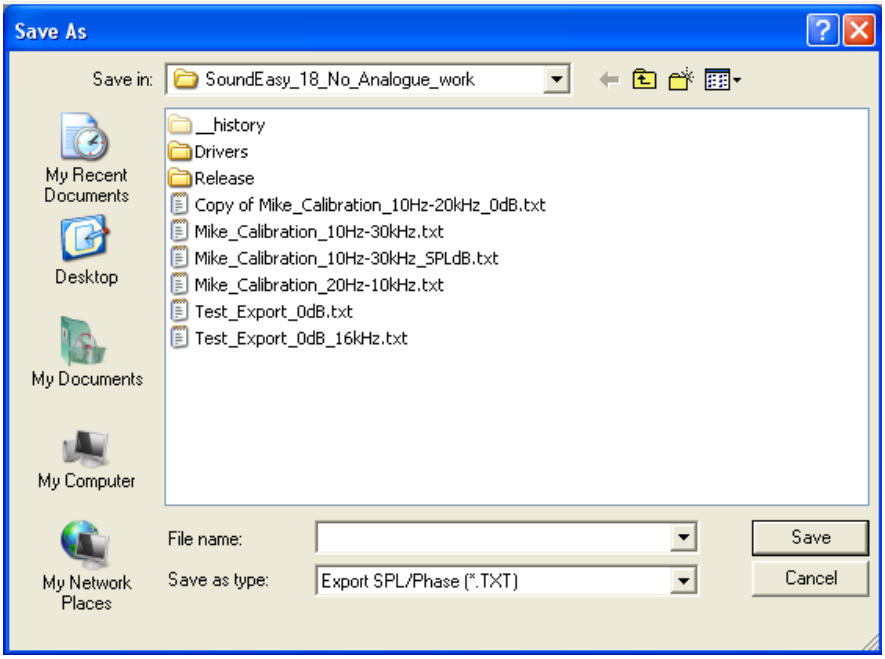


Figure 16.81. Saving the newly created calibration data.

The "Reset" button resets all calibration files to 0dB/0deg Phase.

Import of Impulse Response as TXT File

Impulse response from other measurement systems can be imported into SoundEasy for processing. The example below, shows impulse response collected by HOLMImpulse measurement software, and converted to a TXT file using EXCEL. The resulting file will have just one column representing the actual impulse response coefficients. This can be visualized using Windows Notepad program as shown on Figure 16.82 below.

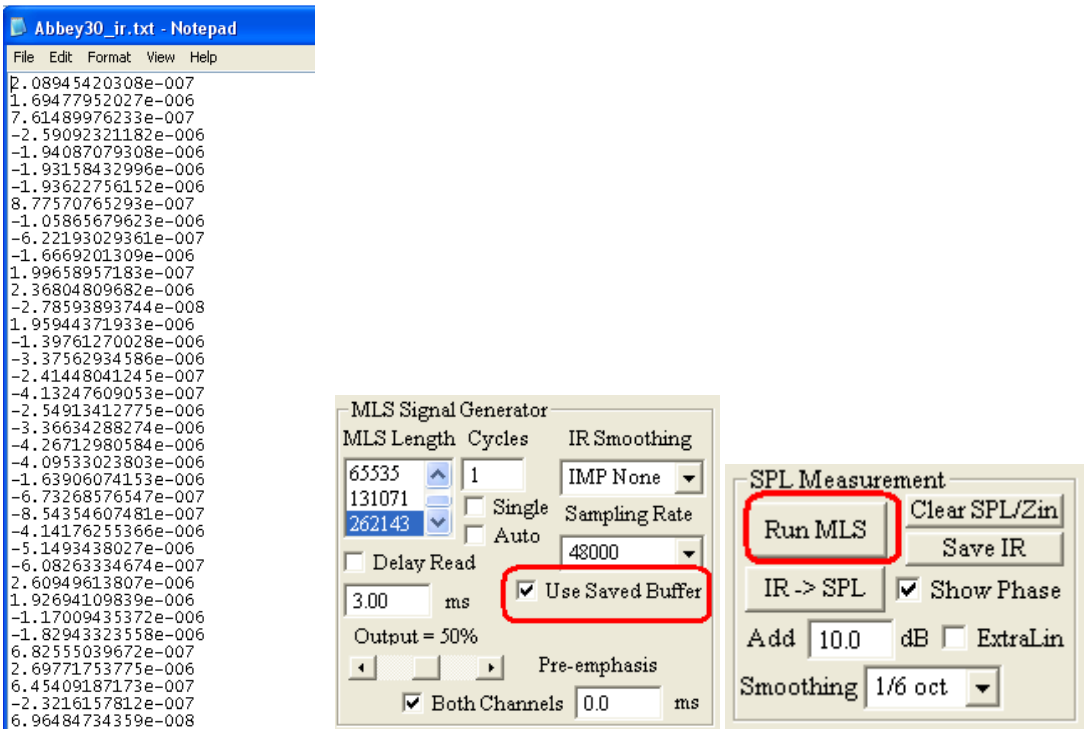


Figure 16.82 – Impulse response converted to TXT file.

To start importing process, please select “Use Saved Buffer” option and the press “Run MLS”. Now, you can select the desired file from the dialogue box shown below.

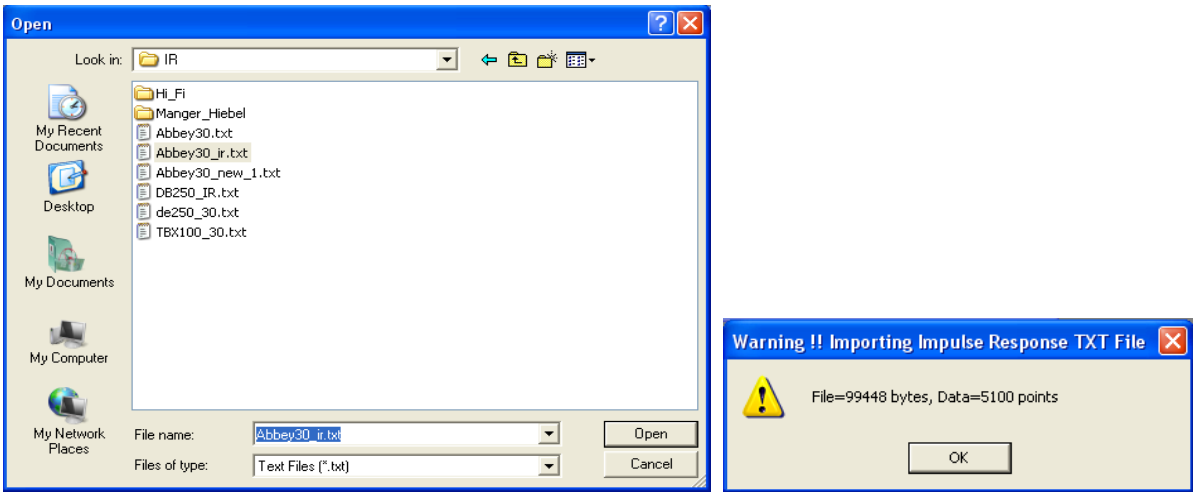


Figure 16.83 – Selecting TXT file to import.

Once the file is selected, the program will display the length of the file and the number of data point in this file – see below.

Pressing the “OK” will allow the process on importing to continue and you should see the impulse response displayed on the “MLS Impulse Response” screen. Please note, that the reference “Ref” pulse is also visible, and it was inserted there as if a dual-channel measurement had taken place.

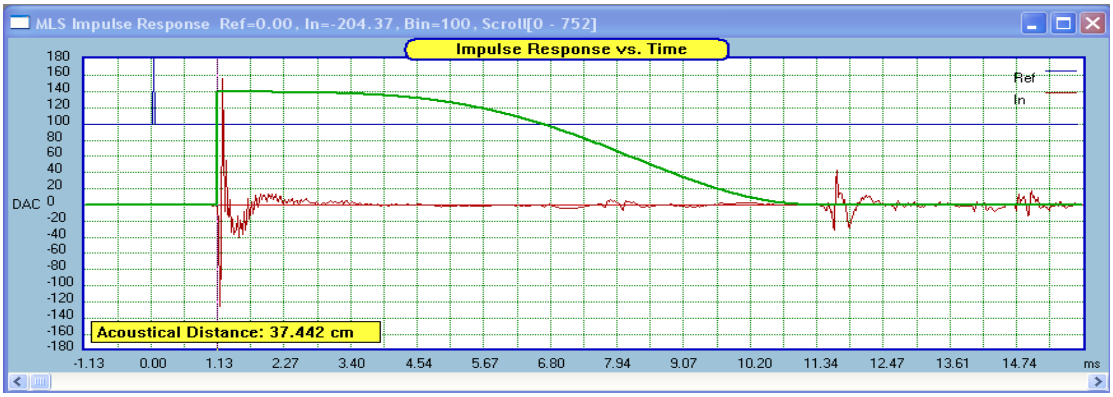
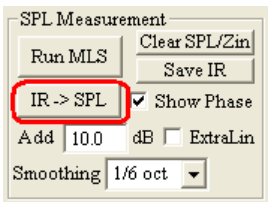


Figure 16.84. Imported impulse response from TXT file

You can now press “IR->SPL” button to obtain frequency/phase responses.



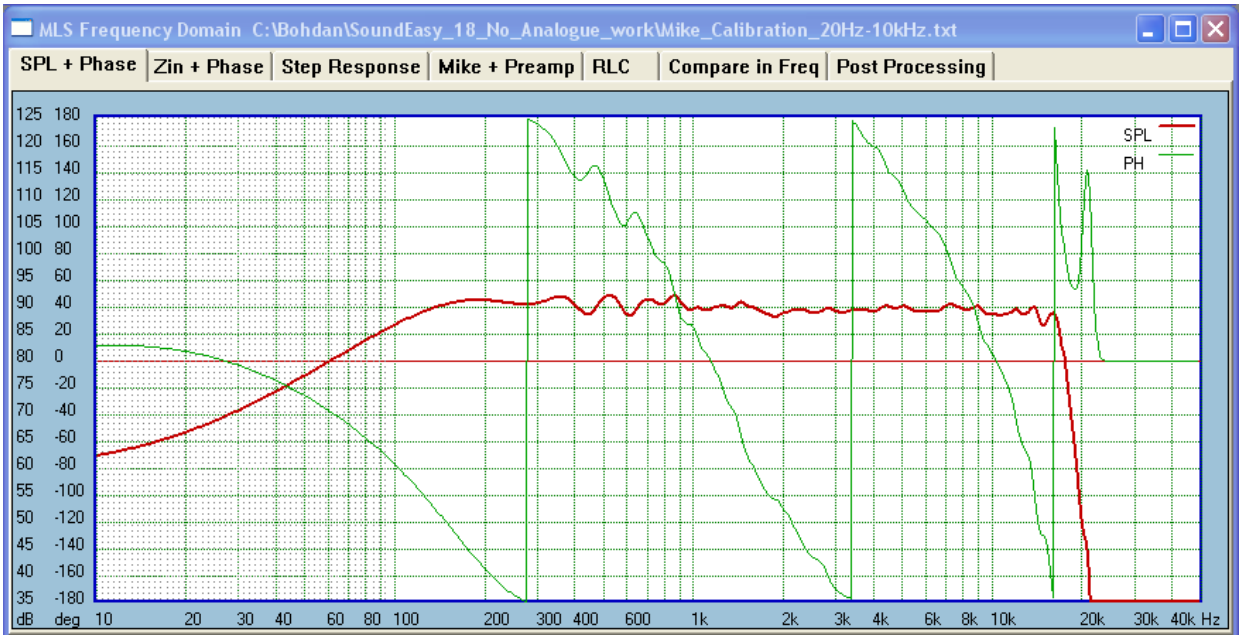


Figure 16.85. Calculated SPL/Phase.

Symmetrical FFT windows

Linear-phase systems are likely to have quite symmetrical impulse responses. Therefore, it is not advisable to use standard, one-sided FFT windows prior to FFT process. Instead, fully symmetrical, double-sided windows are necessary. There are five symmetrical FFT windows available in MLS measurement system. The windows are selectable from the same list box as all other windows.

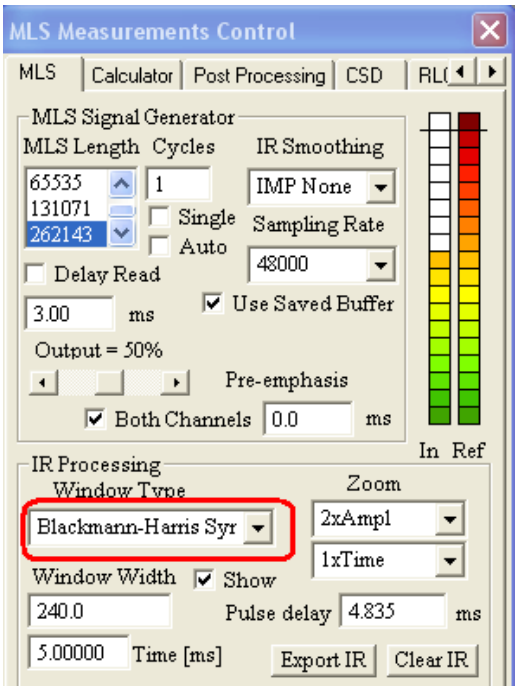


Figure 16.86. Selecting Symmetrical FFT windows.

An example of Blackmann-Harris symmetrical windows is shown below.

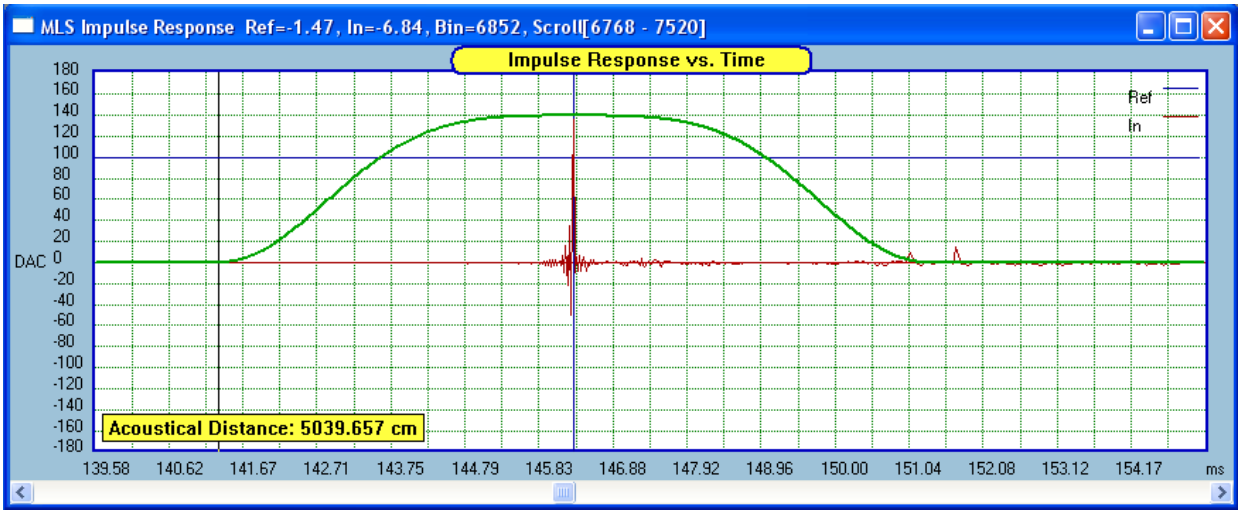


Figure 16.87. Selected FFT window

Extra-Linear Phase Filtering

Phase response is typically calculated and displayed with 360deg rotations. Averaging such jugged phase response leads to tilting the phase response around the 360deg revolutions – and this is undesirable. A better smoothing algorithm was added for phase response, which is calculated and displayed as a flat line. This would be the case for linear-phase loudspeakers.



Figure 16.88 (top) standard filtering (bottom) Phase linear filtering

Showing Differential Phase

Using a new functionality allowing straight visualization of phase difference between two data sets in buffers – see below. The result obtained in this example was 75deg at 2000Hz.

We start with tweeter amplitude and phase responses.

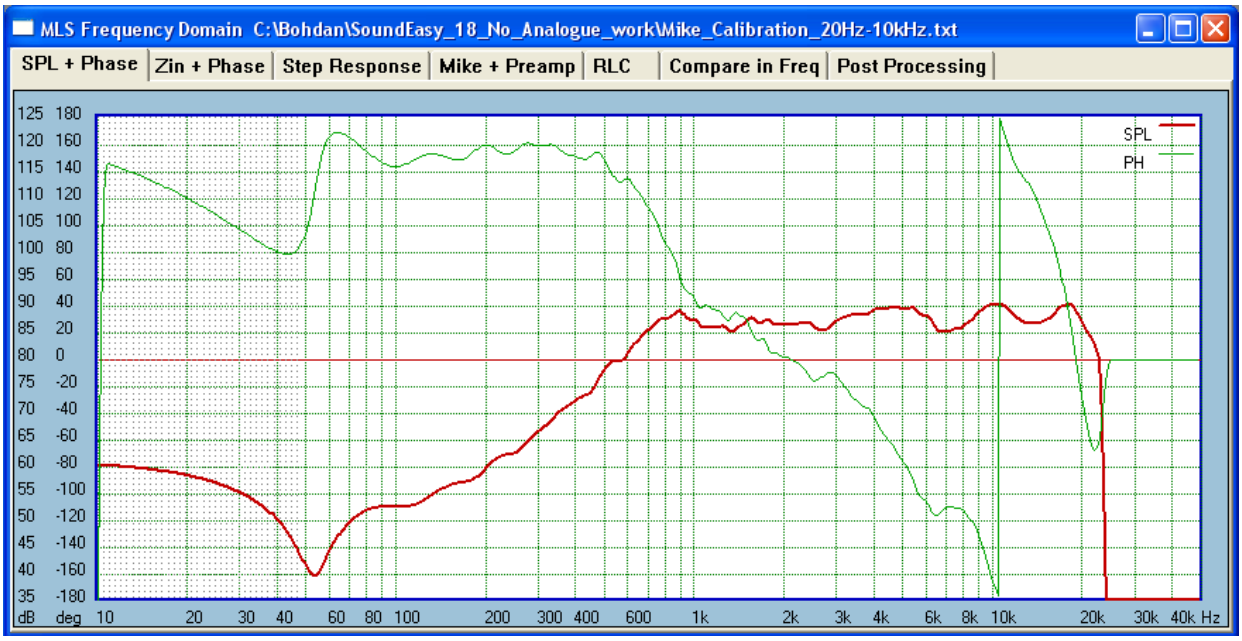


Figure 16.89 This result is saved into "Buffer 1" using "SPL->Buffer 1" button on the Post Processing TAB.

Next, the woofer is measured.

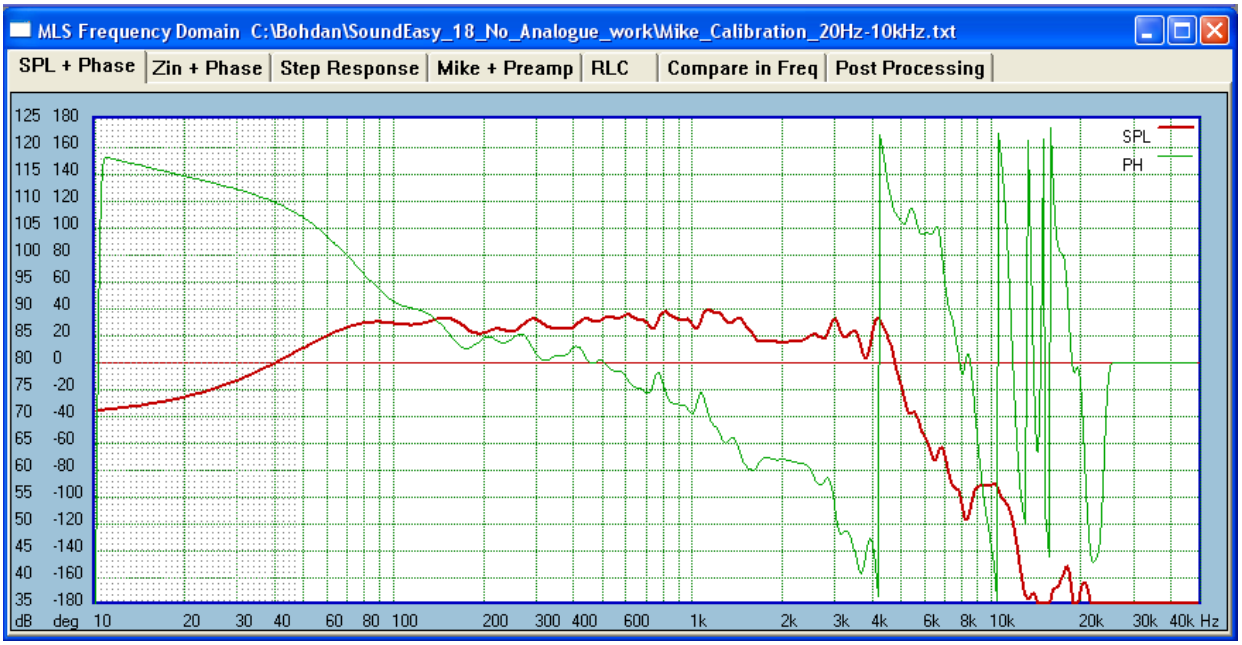


Figure 16.90 This result is saved into "Buffer 2" using "SPL->Buffer 2" button on the Post Processing TAB.

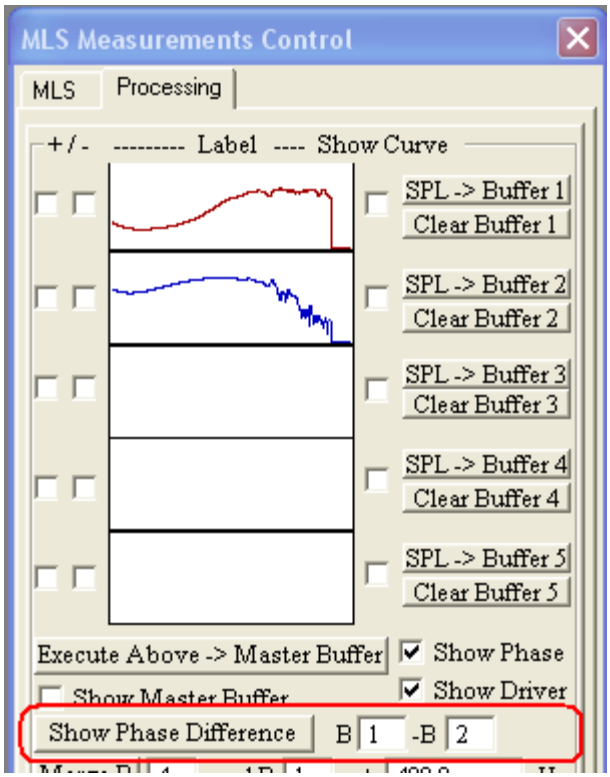


Figure 16.91. Selecting buffers for phase differential plotting

Finally, the B1 – B2 data fields are specified as shown on the Figure above. The differential phase is obtained as shown below.

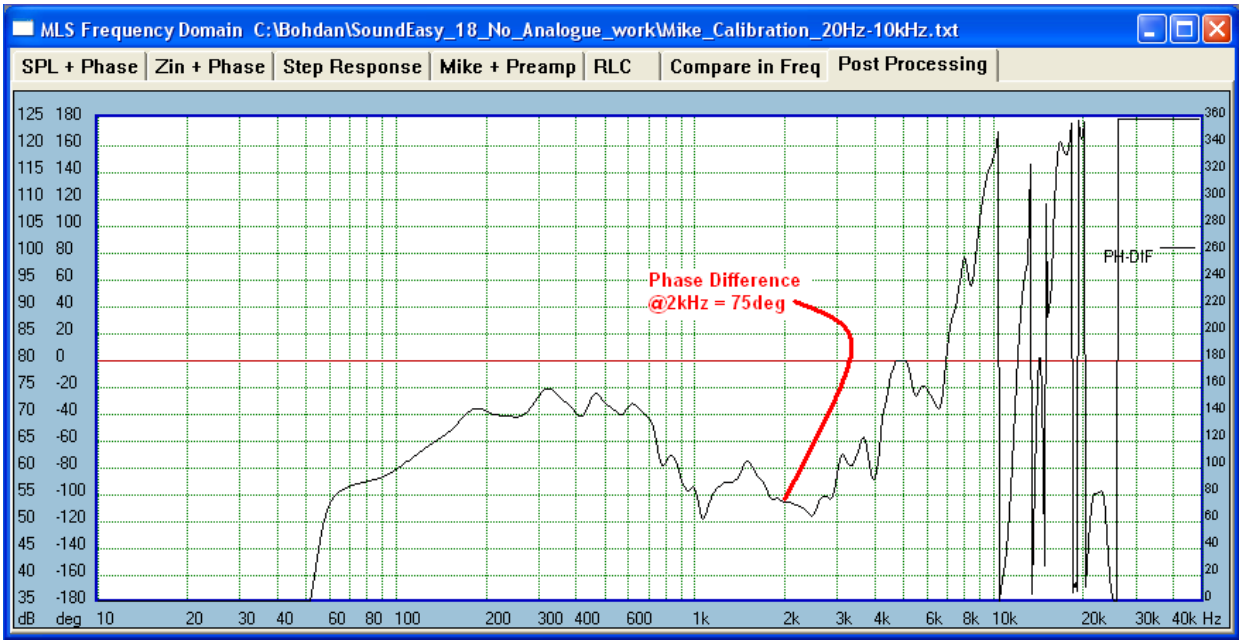


Figure 16.92. Delay = (phase difference) *1000 /(360 x Fc) = 75 x 1000 / (360 x 2000) = 0.104msec.

This value can be used as the tweeter's "Delay" parameter.

Pulse and Bipolar Pulse in Oscilloscope

Time-domain loudspeaker testing often reveals issues, that would not manifest themselves in frequency domain. For this reason, the Oscilloscope function implements pulse and bipolar pulse of various lengths.

The length of the **bipolar pulse** is determined by it's corresponding frequency. Therefore for instance:

10Hz = 100ms
100Hz = 10ms
1000Hz = 1ms
And so on….

The length of the **a pulse** is determined by it's corresponding frequency. Therefore for instance:

10Hz = 50ms
100Hz = 5ms
1000Hz = 0.5ms
And so on….

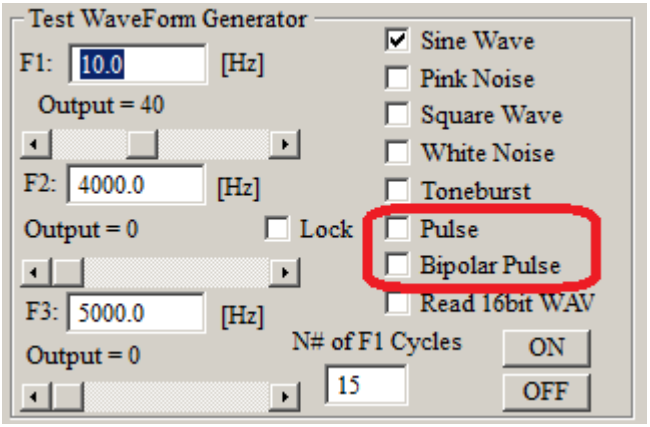
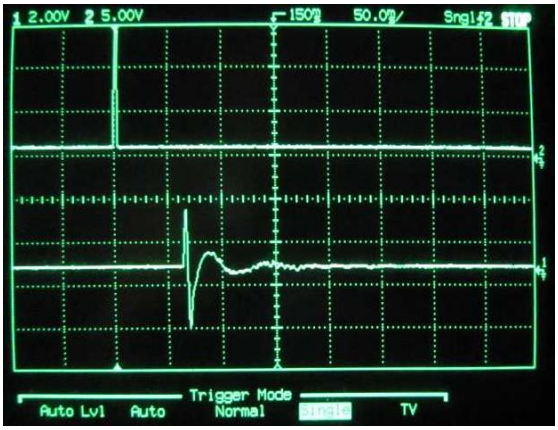


Figure 16.93. Selecting pulse and bipolar pulse in Oscilloscope



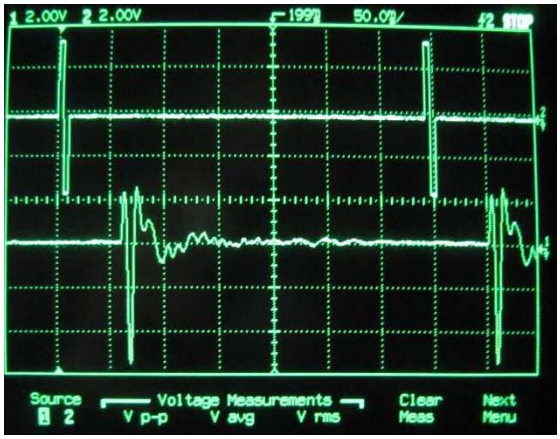


Figure 16.94. Pulse and Bipolar pulse on top traces.

Exponential Sine Sweep - ESS

Exponential Sine Sweep method (ESS), also known as "Logarithmic Sweep" has been in practical use since 2007 and was originated by Prof. Angelo Farina.

Mathematical definition of the generated test signal is as follows:

$$x(t) = \sin\left[\frac{\omega_1 T}{\ln\left(\frac{\omega_2}{\omega_1}\right)}\left(e^{\frac{t}{T}\ln\left(\frac{\omega_2}{\omega_1}\right)} - 1\right)\right]$$

Formula 1.

Figure 16.95. Analogue test signal in time domain.

Where the sweep frequency starts at ω1, ends at ω2, taking T seconds to complete. Time taken to sweep from 50Hz to 100Hz is the same as time taken to sweep from 100Hz to 200Hz and so on…

The ESS method implemented in this software is a "dual-channel" version of the ESS method. There are: Reference Channel and the Signal Channel. Signal feeding the Reference Channel is typically extracted from right across the loudspeaker terminals, so the amplitude, phase and timing distortions of the sound card and power amplifier are included in the Reference Channel. Signal feeding the Signal Channel is typically obtained from the microphone pre-amplifier. The dual-channel method is therefore self-calibrating to a large degree, as it calibrates out the sound card and power amplifier issues, and provides exact timing reference for absolute phase measurements.

Picture below depicts example waveforms involved in ESS system, where the test signal (blue) is time-varying sine wave entering system under test, and the collected signal (red) is the system's response.

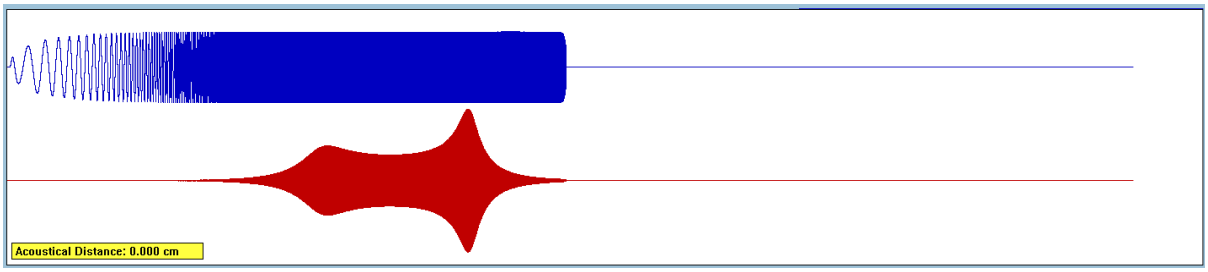


Figure 16.96. ESS test signal (blue) and collected signal (red).

A block diagram representing the dual-channel ESS method is shown below.

Reference Channel Processing

In the first step, the reference signal in normalized in amplitude to unity. Next, the FFT transfers the reference signal from time-domain into the frequency-domain, and here we can observe, that spectrum of the sine-wave with it's frequency changing as per Formula 1, decays at -10dB/dec. The spectrum is now inverted to create signal suitable for convolving with the signal received from the microphone.

In the meantime, convolving the inverted and non-inverted spectrum of the reference signal and passign it through IFFT process, creates impulse response of the reference channel. This impulse response has correct timing attributes embedded in it's location on the time scale. In further processing, the impulse response is windowed by the same window as the signal impulse response and then passed through FFT process, which finally produces SPL and phase of the Reference Signal.

Signal Channel Processing

In the first step, the signal collected from the test microphone is fed into FFT process, which transfers the reference signal from time-domain into the frequency-domain

Convolving the reference signal with microphone signal generates impulse response of the system under test. In further processing, the impulse response is converted into time domain with the help of IFFT process, so it can be windowed and then passed through FFT process, which finally produces SPL and phase responses of the Signal Channel.

The whole process sis graphically depicted on figure below.

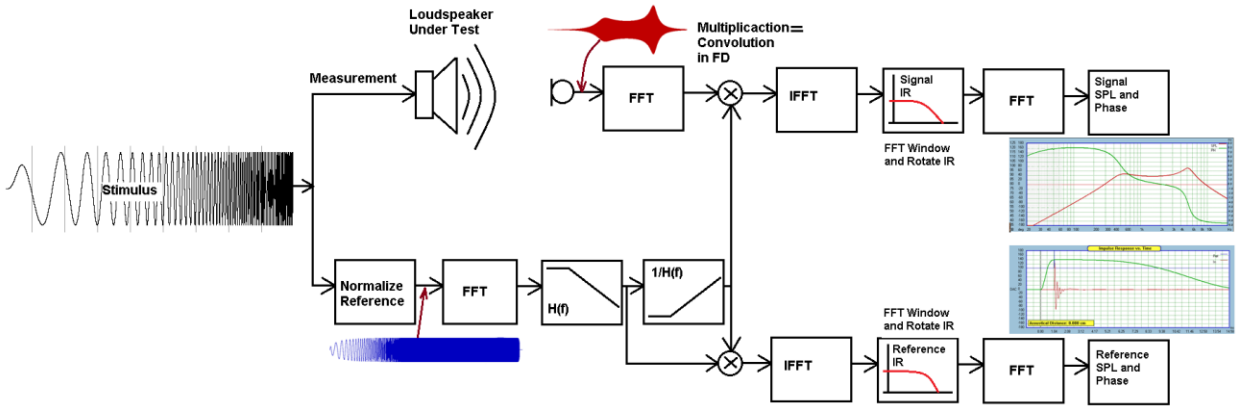


Figure 16.97. Dual-channel ESS measurement system.

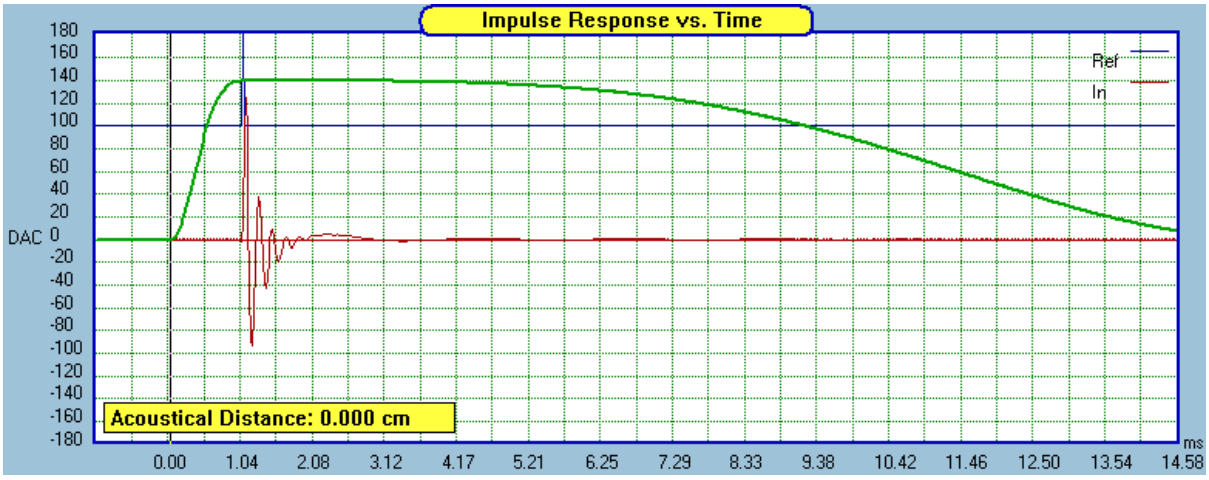


Figure16.98. Example of Signal Channel impulse response (red) and FFT window (green).

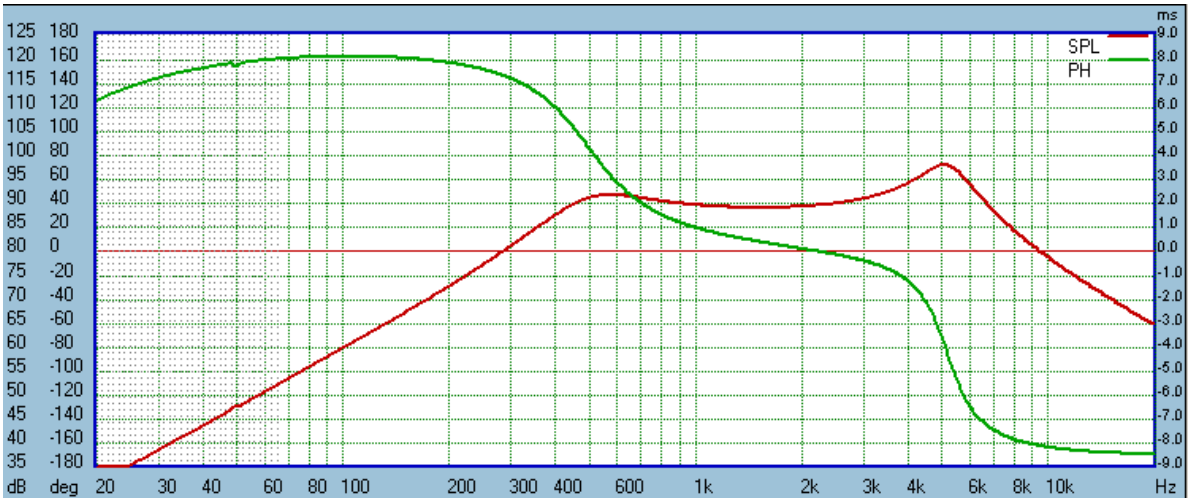
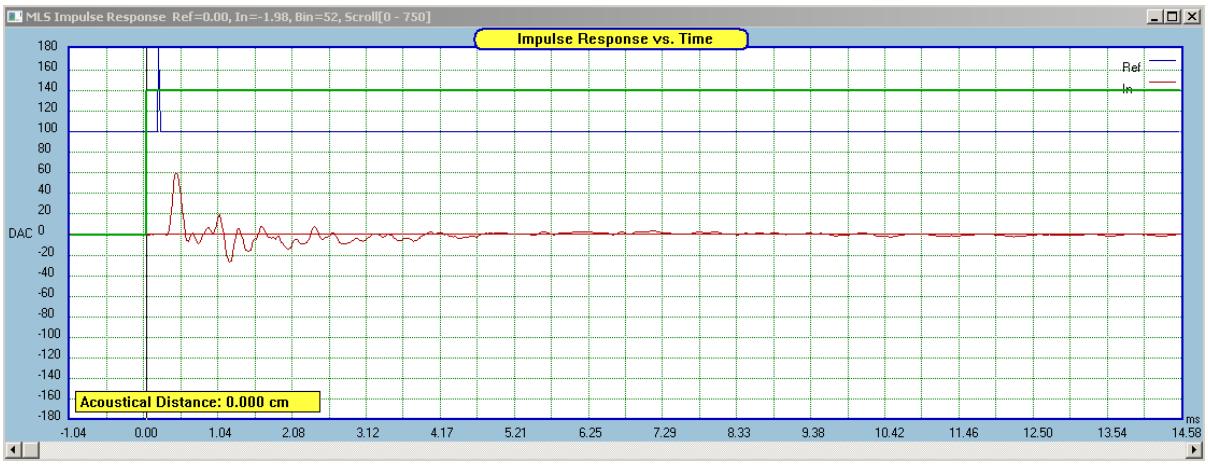


Figure 16.99. SPL (red) and phase (green) corresponding to impulse response from Figure 16.96.

Comparison between MLS and ESS measurements

Impulse response of ESS System



Impulse response of MLS System

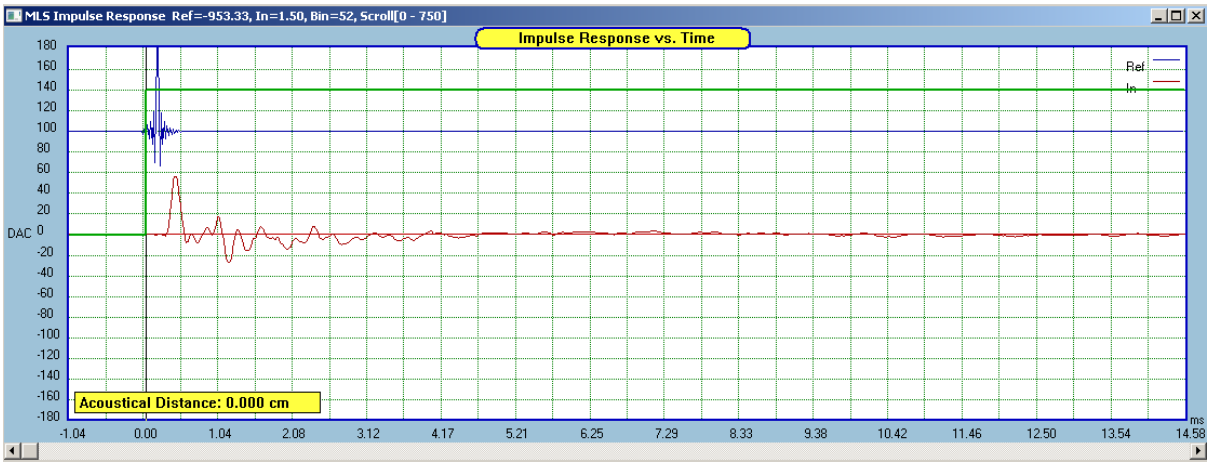


Figure 16.100. ESS and MLS impulse responses.

Impulse responses are virtually indistinguishable from each other.

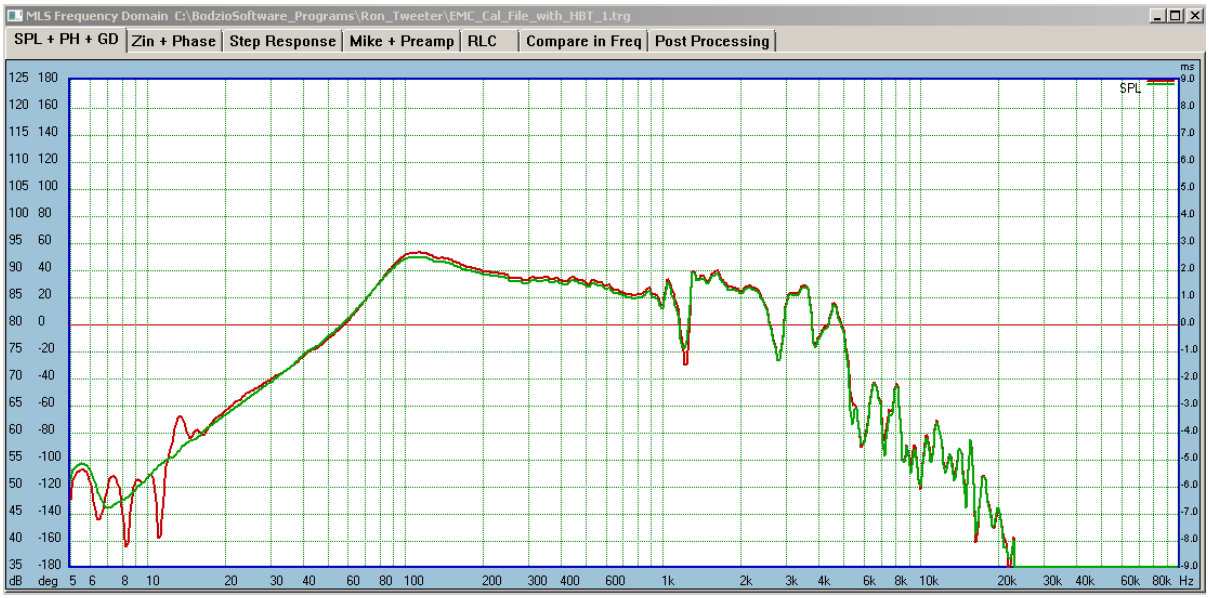


Figure 16.101. FFT-calculated SPL: MLS System – red curve, ESS System – green curve

SPL curves are extremely close (virtually overlapping most of the time) the same above 15Hz.

Loop Test of ESS System

A simple Loop-Test, with sound card outputs connected directly to the inputs (Delta1010LT was used), reveals perfect linearity of the amplitude and phase responses.

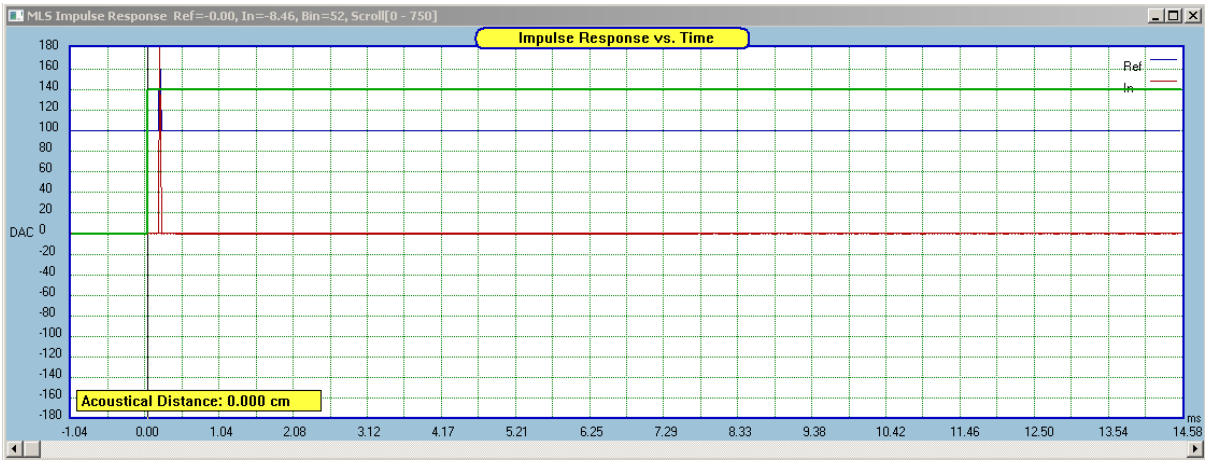


Figure 16.102. Impulse response of the Reference Channel and Signal Channel

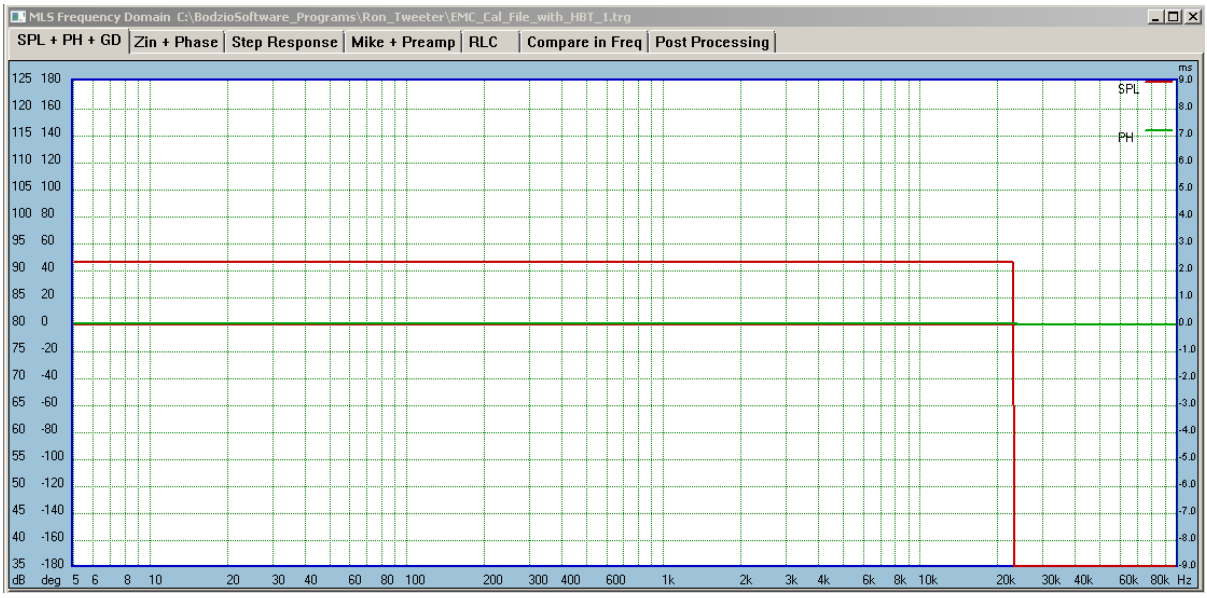


Figure 16.103. Resulting SPL (red) and Phase (green) responses collected at 48kHz sampling frequency.

Use of Small Front Window

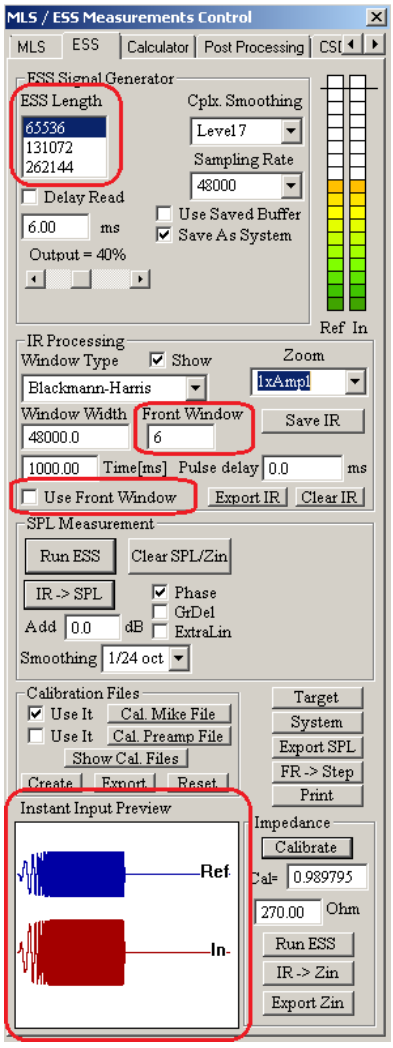


Figure 16.104. Small Front Window

First, we illustrate the case of impedance measurement with no Front Window. The “Use Front Window” box is unchecked – see below

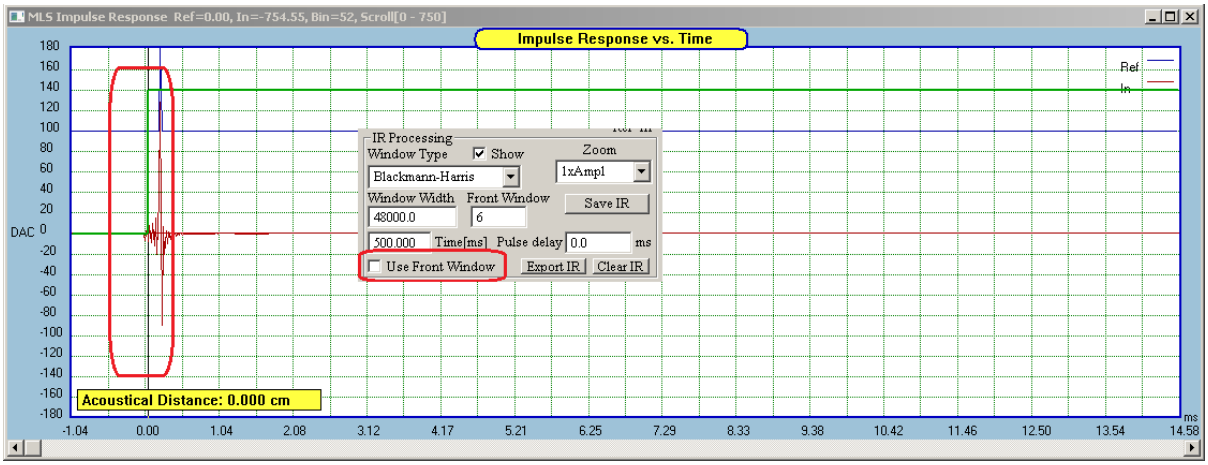


Figure 16.105. Small Front Window NOT used.

Corresponding input impedance, Zin curve is shown on figure below.

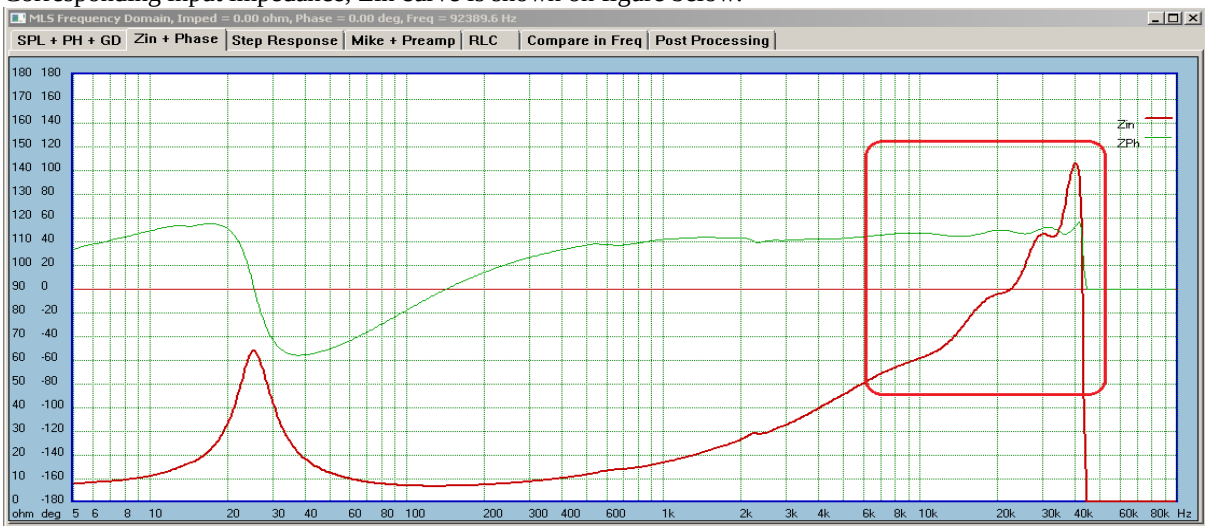


Figure 16.106. Small Front Window NOT used.

Please note small amount of ripple on the SPL/Phase responses above 10kHz.In the next example, the Front Window of 20 sample-length is being used. This example runs at 96kHz sampling frequency.

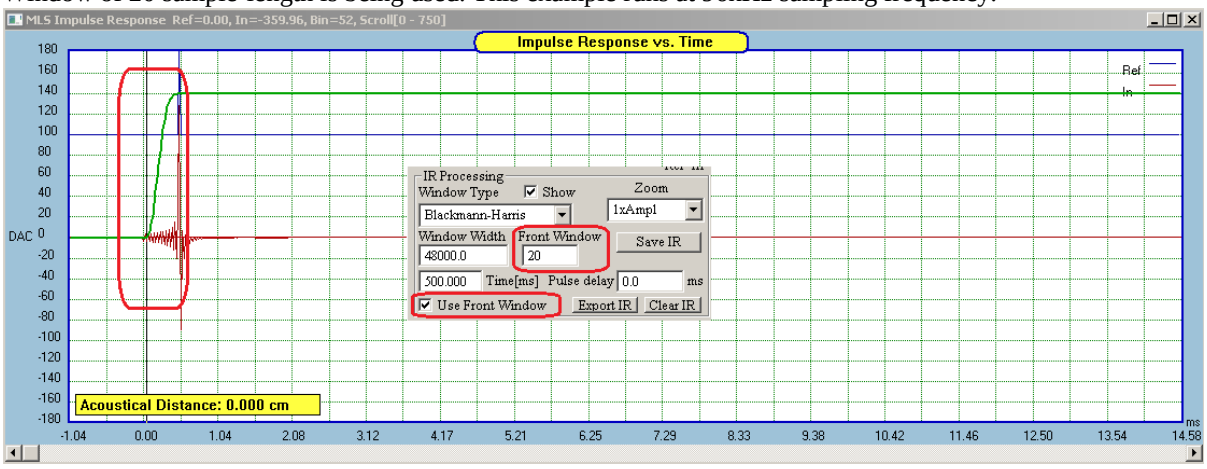


Figure 16.107. 20-sample, Small Front Window used.

The resulting input impedance curve, Zin, and phase curve are completely free from ripples.

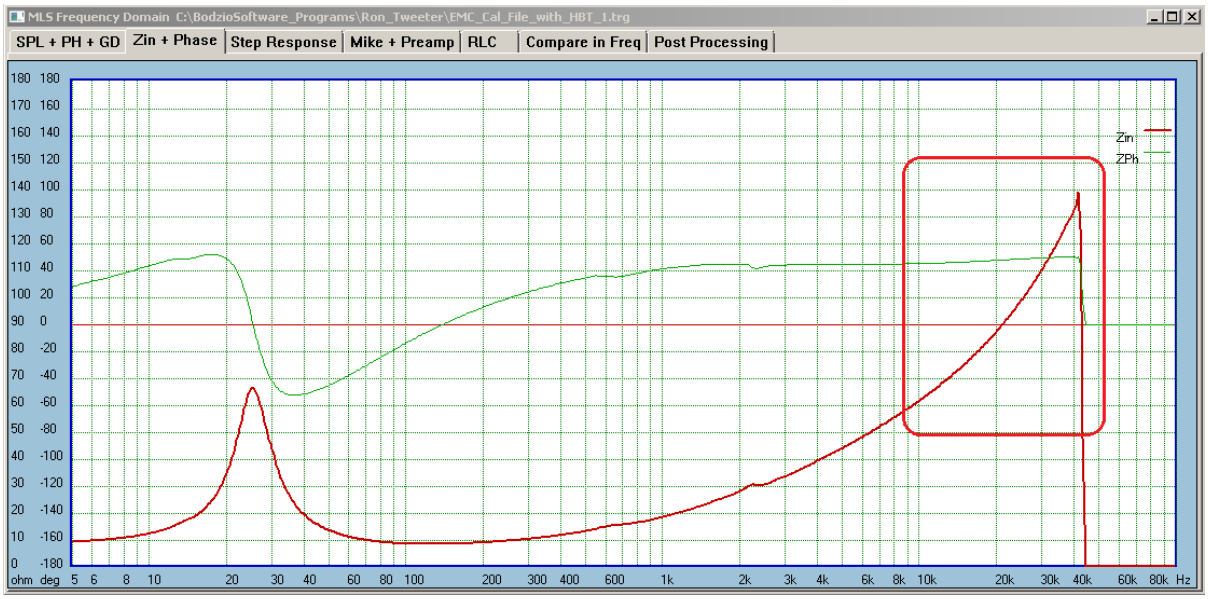


Figure 16.108. 20-sample, Small Front Window used.

The need for the Front Window usage needs to be determined experimentally. In some instances (impedance measurements) it may be beneficial to use it. Often, it makes no difference in measurements. **Please note, that ESS TAB controls are performing the same functions as in the MLS system. Therefore, for full description of their operation please refer to the MLS section of this manual.** Furthermore, Signal and Reference probe channel assignment in ESS system is the same as in MLS system. It is therefore very easy to use one system or the other for measurements. It is advisable to adjust MLS signal to about 50% (-6dB) of the ESS signal output from the sound card. This is due to MLS signal crest factor created by the sound card low-pass filtering of square waves of the MLS signal.

SPL Level Display Of the Merged Curves

In order to facilitate merging SPL curves, the SPL level of both curves is displayed in the caption of the plotting window. In the example below, “Buffer 4” and “Buffer 5” were filled with measured curves, and are being merged, so the caption would display Ampl[4] = XX.XdB, Ampl[5] = XX.XdB, at the frequency pointed to by the mouse pointer.

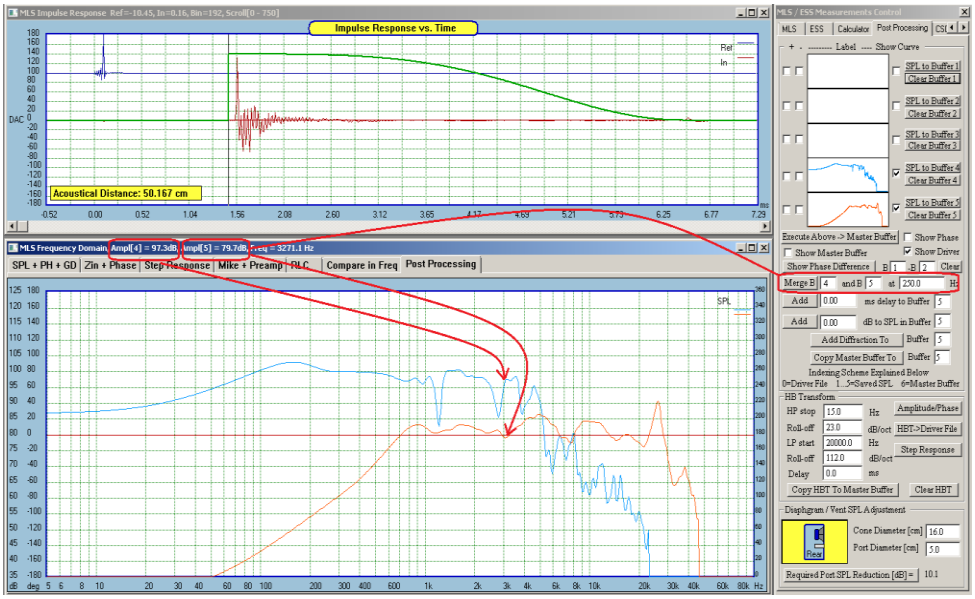


Figure 16.109. SPL of merged curves displayed in the caption.

Calculator For Vent/Cone SPL Ratio

Woofer's frequency response can be correctly measured in-room, using close-mike technique, rather than single 1m distance measurement. The way the process works, is that you need to (1) measure driver's frequency response, then (2) port's frequency response, and (3) add them together with simulated diffraction. Port response has to be scaled down by several decibels, due to the difference in effective diameter between port and driver:

SPL reduction = 20*log(Driver_Radius / Port_Radius)

In the case below, the port SPL was shifted down by -10.1dB. Simply enter cone and port diameters in [cm] and press the "Required Port SPL Reduction" button, to obtain the required scaling in "dB".

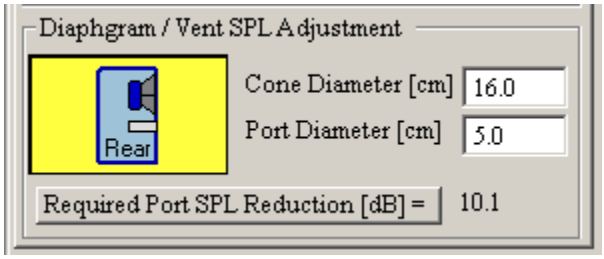


Figure 16.110. Calculator For Vent/Cone SPL Ratio

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Remember, on the top of this, you need to add pre-calculated diffraction for this box.

Calibrating Long Lead Resistance Out

Lead resistance, Rlead, is calculated as average resistance of the long leads between 50Hz and 5kHz. This value is permanently subtracted from the real part of the input impedance of the driver. It acts as a scaling factor. So, if your leads are short, or you do not want the Rlead to be included in the input impedance curve, **then you must enter value of 0.0 as the Lead Resistance** – see Figure below.

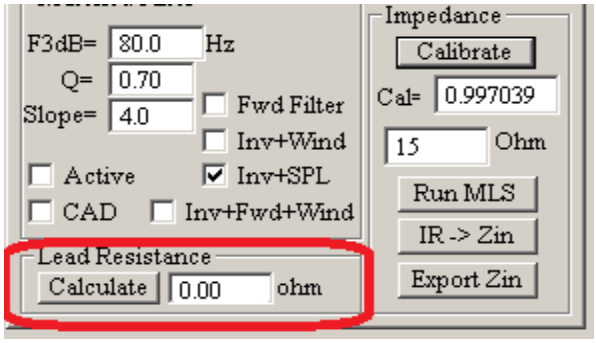


Figure 16.111. Calculator For Vent/Cone SPL Ratio

If your leads are long enough to exhibit non-negligible resistance in comparison with the loudspeaker impedance, then it would be advisable to calibrate them out of the impedance curve. This is a two-step process.

1. Use small (10-15ohm) test resistor and perform standard loudspeaker impedance measurement, but with the loudspeaker terminals SHORTED. This way, the only resistance that is measured will be the lead resistance. You can adjust the vertical scale of the impedance screen to 1ohm, and after pressing "IR->Zin" button, you should see a flat line, similar to the 1.15ohm line shown below.

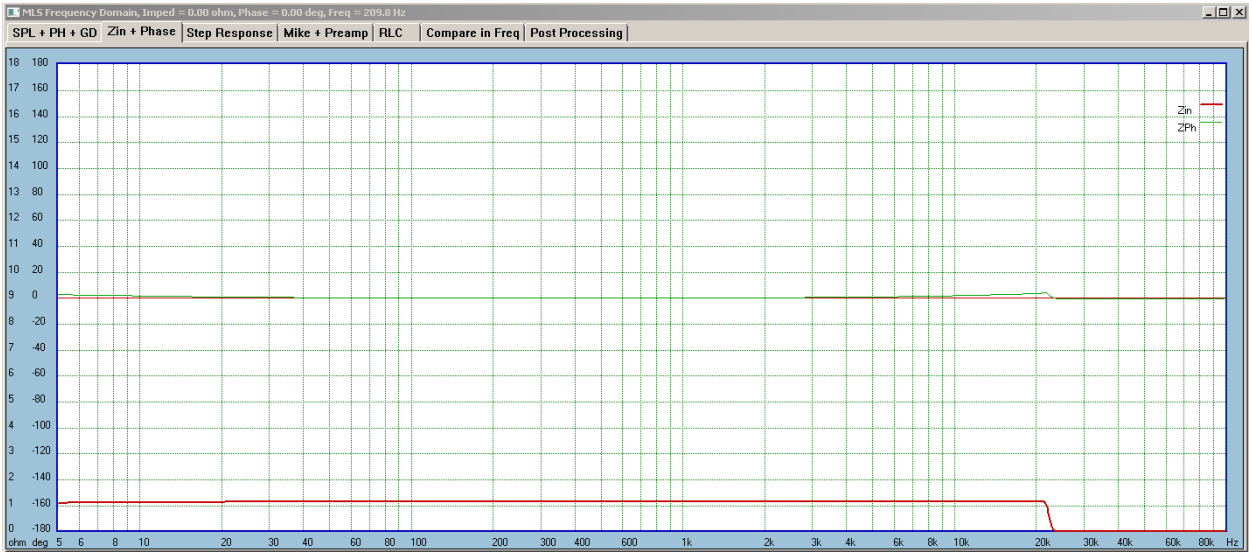


Figure 16.112. 1.15ohm line shown for shorted leads.

2. Now press the “Calculate” button in the “Load resistance” box, and the program will calculate the lead resistance and will display it in the editable box next to the button. This value is now permanently subtracted from the real part of the input impedance of the driver.

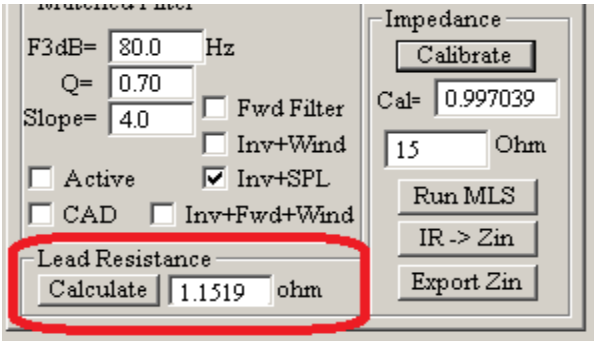


Figure 16.113. Calculated 1.15ohm for shorted leads.