FINE R+D))™

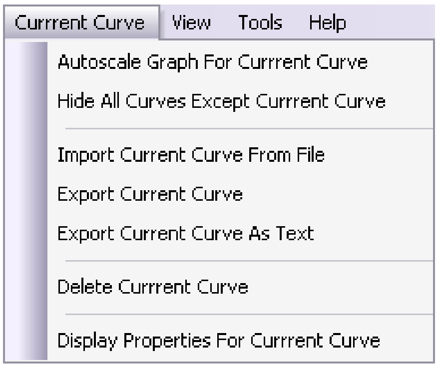
Acoustic Measuring System

- Up-to-date Replacement for LMS and MLSSA
- Multiple curves 16 + 16 + ? (depending on memory)
- Same calibrated sine wave level for both SPL and Impedance
- THD and 2nd to 9th harmonic distortion
- Precise Thiele/Small Parameters at correct drive level
- SPL + Phase / Imp + Phase in one measurement file
- Windows 7 – 64 bit compatible
- Preview & Drag & drop files from/to Klippel / LMS / VACS / MLSSA etc.
- FINE Hardware (USB) with built-in 25W Power Amplifier
- Fast measurements < 1 sec
- Measures also USB devices:
  - USB Headphones
  - USB Radios
  - USB Speakers
  - USB output level in % FSD
- Works with FINE QC™ End of Line Testing:
  - SPL & Sensitivity with individual tolerances
  - Automatic "Golden Average"
  - Advanced Rub & Buzz : FINEBuzz
  - Running Average and Statistics
  - File-sharing with FINE QC / R+D and other LOUDSOFT programs
- FINE R+D and FINE QC are both developed by experienced engineers

LOUDSOFT
www.loudsoft.com

Contents

Overview



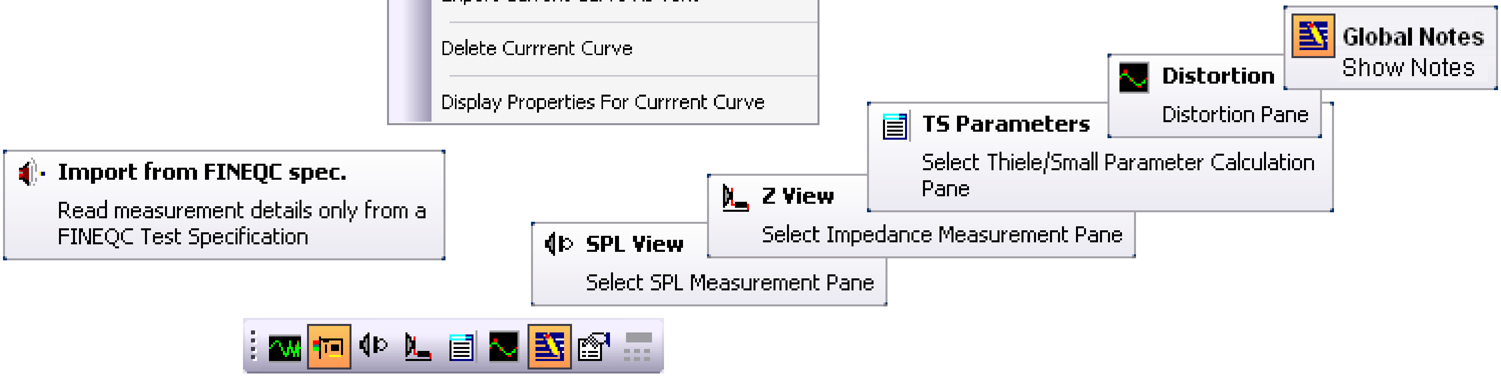
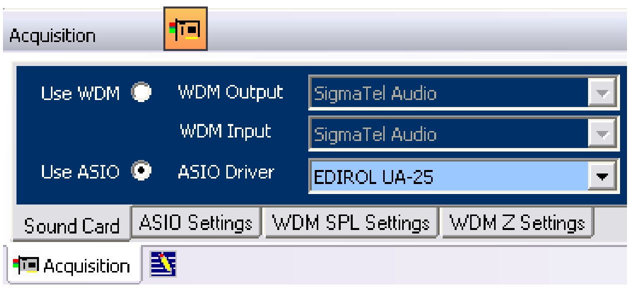
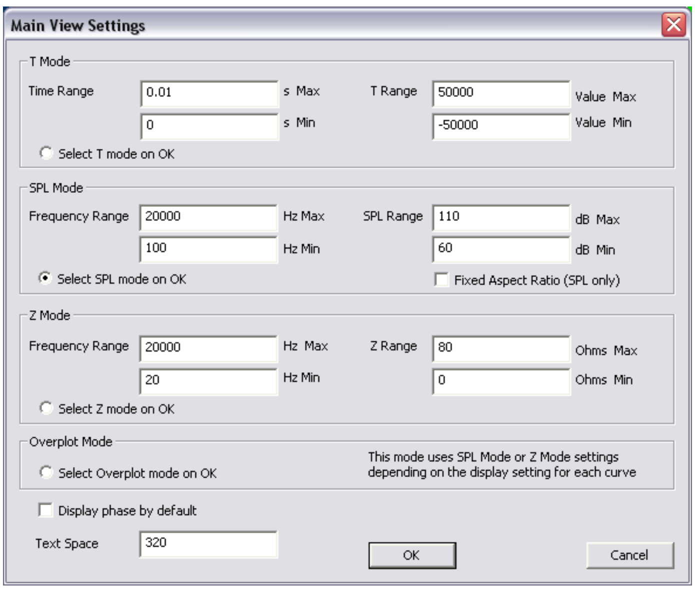


Figure 1 - FINE R+D Main Screen

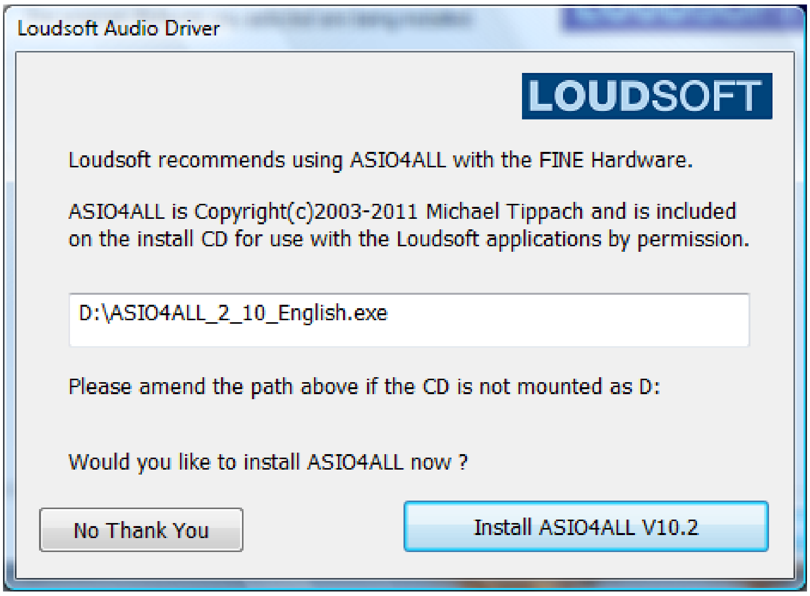




HOW TO GET STARTED

1. First install the FINE R+D software as ADMINISTRATOR from the CD without hardware connected. You must install for ALL users.

2. Then you will be asked to install the ASIO4ALL driver, which is needed to work with the FINE hardware. (The driver Is also available as download from http://www.asio4all.com/)



3. Then connect the power supply to the hardware box

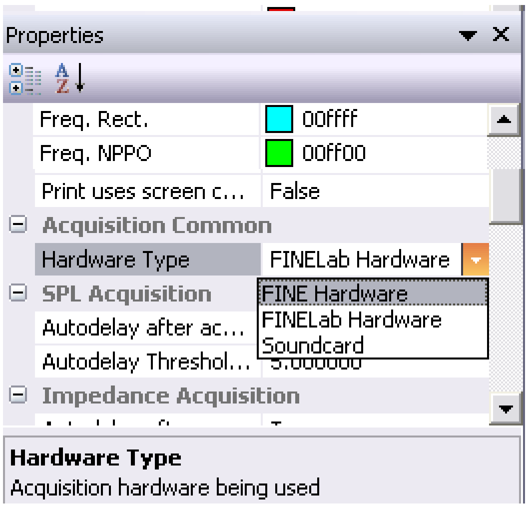
4. Connect the hardware box to the computer with the USB cable

5. Start FINE R+D

Select your hardware first!

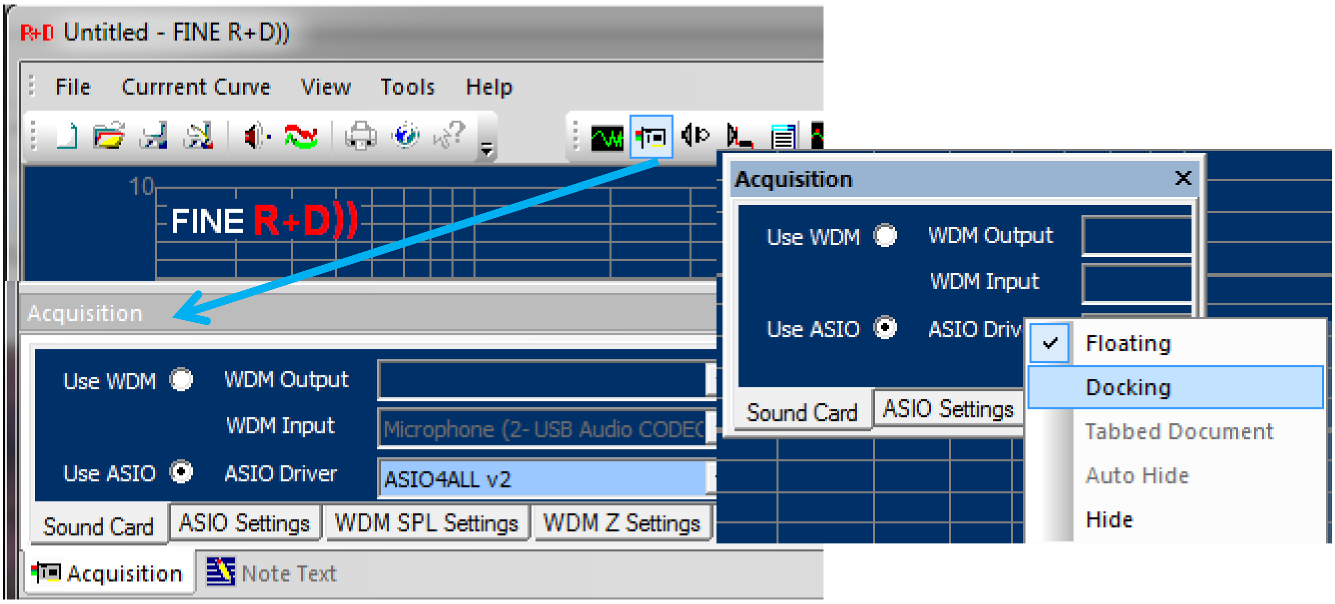
First you need to specify which hardware you are using:

Go to properties (View/Toolbars and Docking Windows/Properties) and select the hardware you are using:



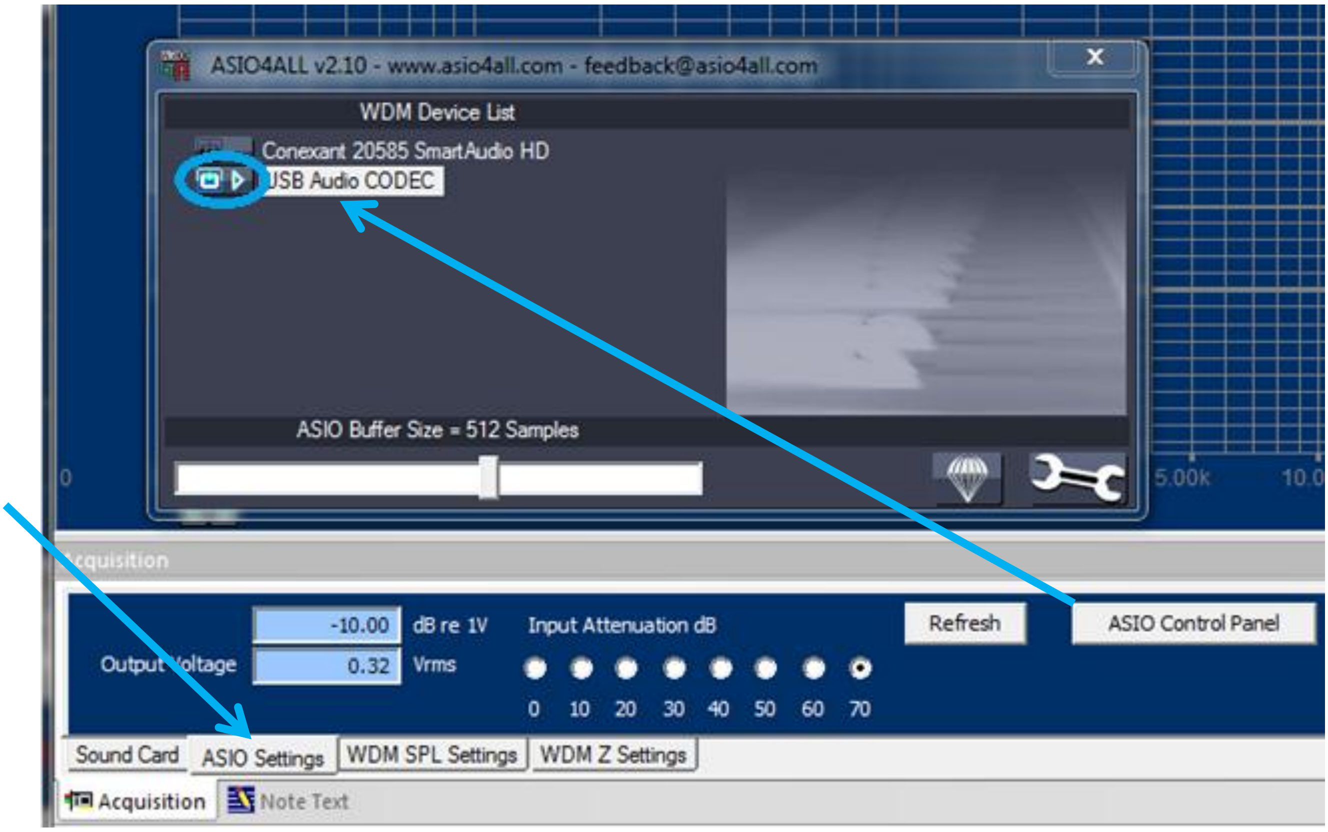
Verify Driver Settings

Open the acquisition window and verify that ASIO driver is selected. Then make sure that the ASIO4ALL driver is selected in the drop-down box. Right click the acquisition window to select either docking or floating. Docking will show always as a part of the screen.



Then press the [ASIO Settings] tab and the [ASIO Control Panel] button to open the ASIO4ALL window. Here are listed the current sound drivers.

Click on the left buttons until you get ONLY the USB Audio Codec highlighted as shown:

Then Do Calibration (ASIO recommended)

Before you can measure with high accuracy we strongly recommend performing the ASIO system calibration below. This is valid for the FINE Hardware and FINELab (hardware) as well as soundcards with ASIO drivers.

Open the acquisition window  and select “Use ASIO”. Check that the chosen ASIO driver is active.

Select Tools/Calibration – ASIO to perform accurate system calibration using the precise ASIO drivers (When you use the WDM Windows drivers, select that calibration instead)

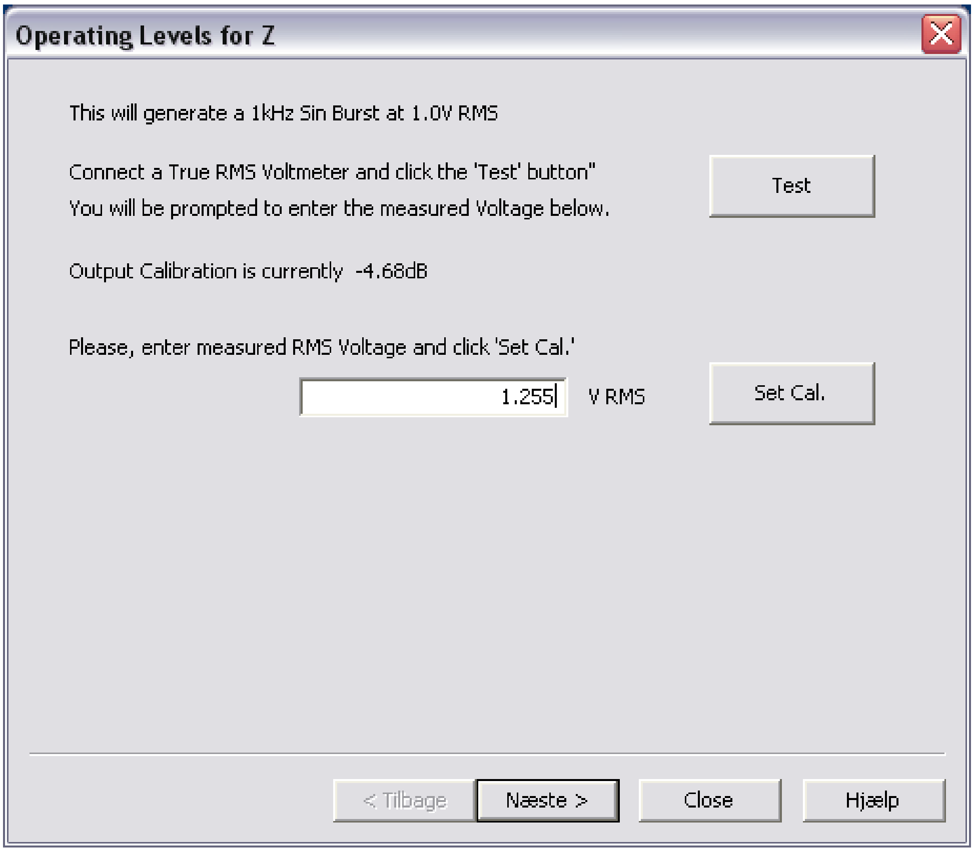


Figure 2 - Output calibration

Connect a true RMS voltmeter to the output and click the [Test] button. A short 1 kHz sine wave is played at the speaker output. Enter the measured value (here 1.255Vrms) in the blank field and click [Set Cal.]

This will calibrate the output to 1.00 Vrms. (press Test again if you want to verify).

Press [Next]

Connect a special loopback cable from output to microphone input. Press [Measure] and measure the total loopback response as a green response curve. When you press [Apply] this curve will be used to calibrate and equalize the total system.

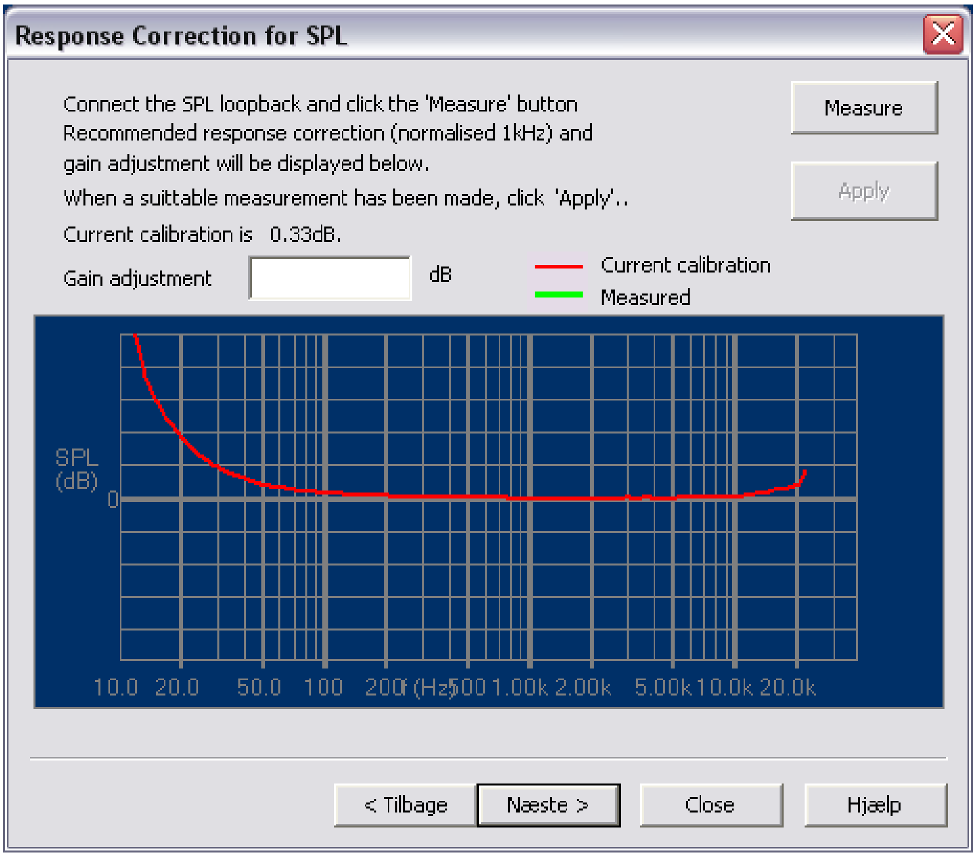


Figure 3 - Loopback response calibration

Press [Next]. Connect a precise known power resistor to the output. The value should be around 4 ohms. If you normally measure high impedance devices you can choose a higher value.

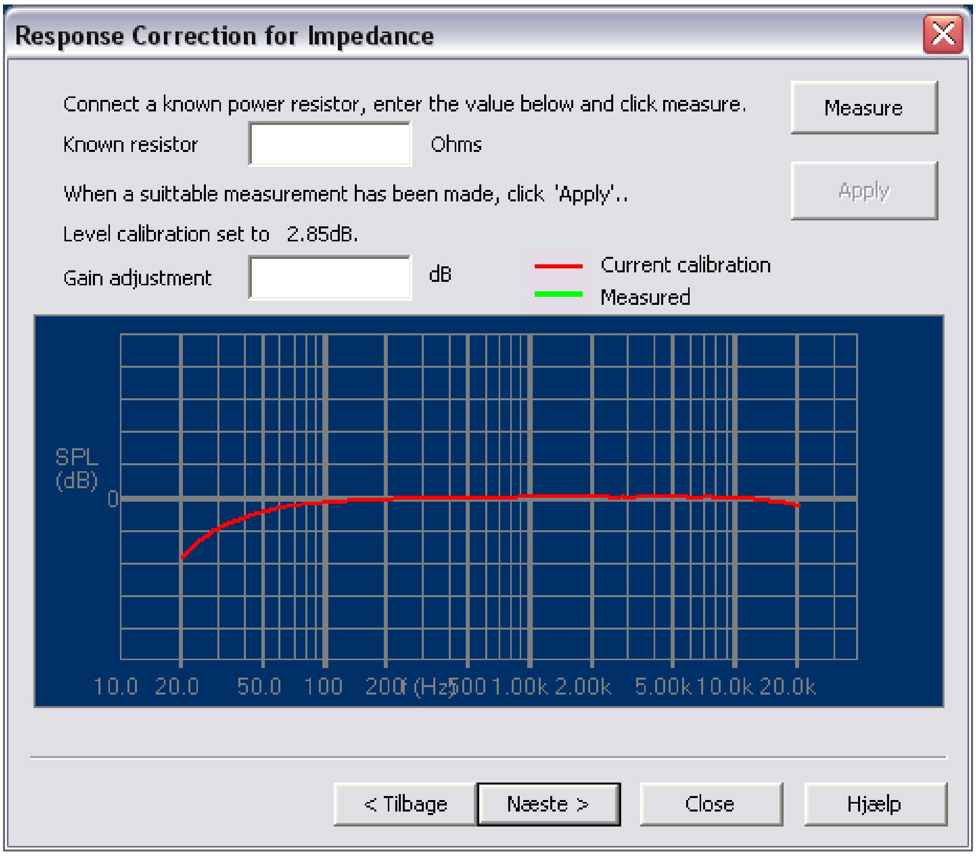


Figure 4 - Impedance Loopback calibration

Enter the resistor value in the upper field [   ] and press [Measure]. (Do not enter anything in the lower field [   ], that is done automatically). When you press [Apply] this curve will be used to calibrate and equalize the system for impedance

Press [Next]

Connect a microphone calibrator if available and press [Measure]. The measured microphone calibration is shown as dB re 1V/Pa. Press [Apply] to calibrate the system for correct SPL measurements using this microphone.

Alternatively you can enter the microphone calibration data from the calibration sheet, which for example is supplied with the LOUDSOFT FL1 microphone.

The full system calibration is now done and the settings have been saved for future use. You may perform this calibration again when needed.

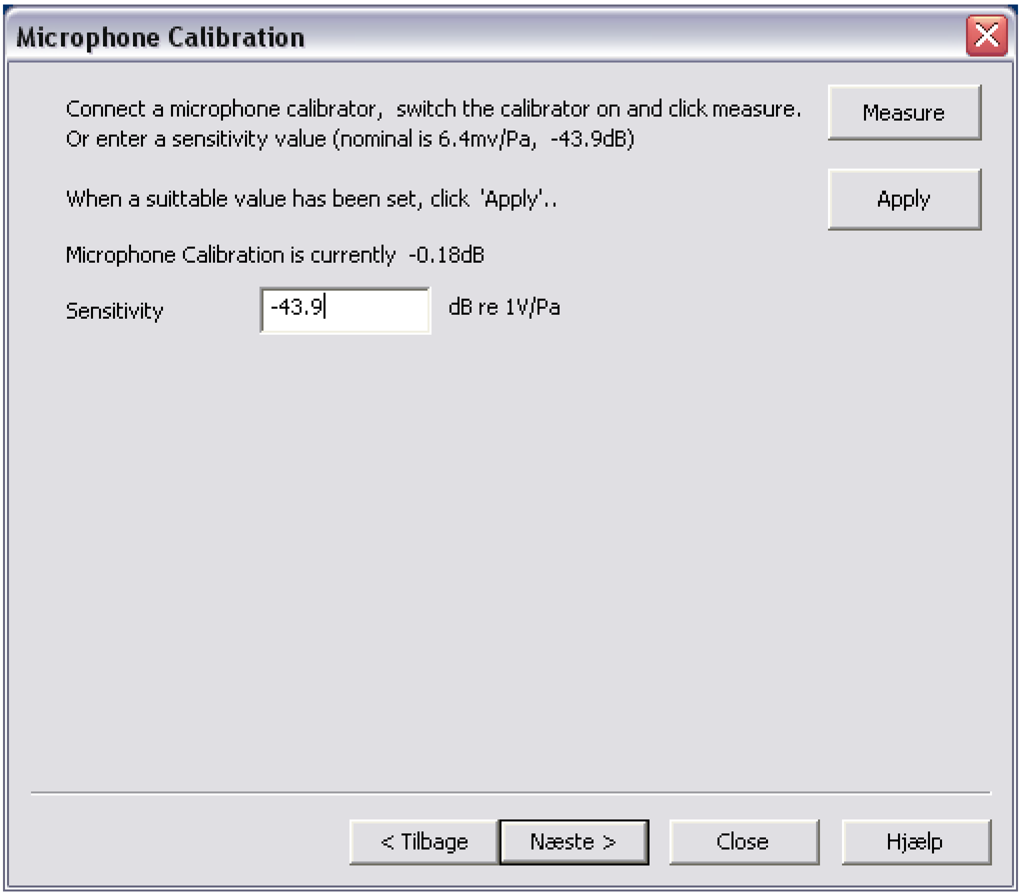


Figure 5 - Microphone calibration

Frequency Response / SPL Measurement

This chapter is a tutorial how to measure a normal loudspeaker with FINE R+D. If you have not yet done calibration, please do this first (go to page 4).

Select "New" from the menu, or open another worksheet to add measurements.

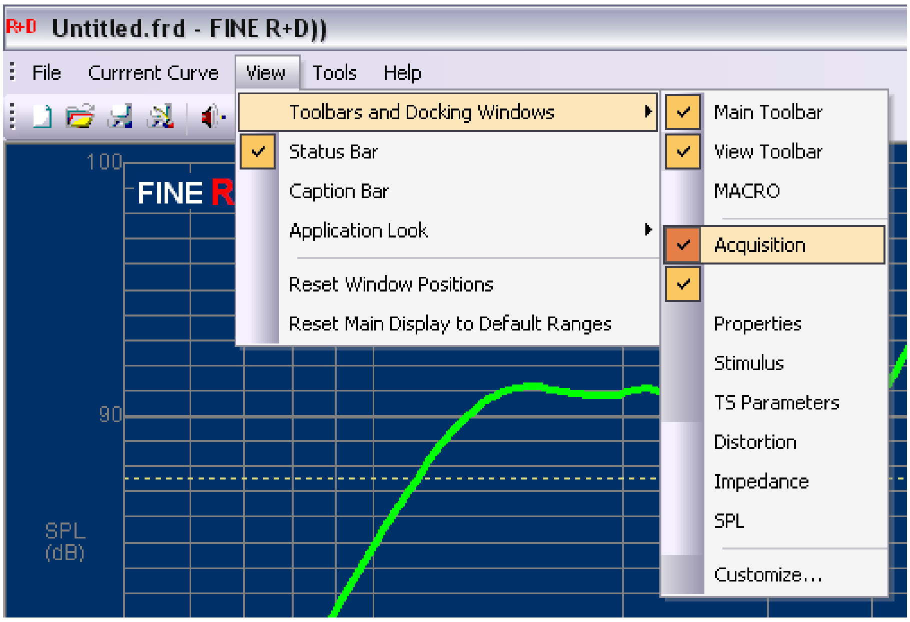


Figure 6 - View/Toolbars/Acquisition

Acquisition is where you define output and input levels. See View/Toolbars and Docking Windows to set which windows are active. Under the tab "ASIO Settings" you may set the output voltage to 2.83V (industry standard), or set the voltage you need.

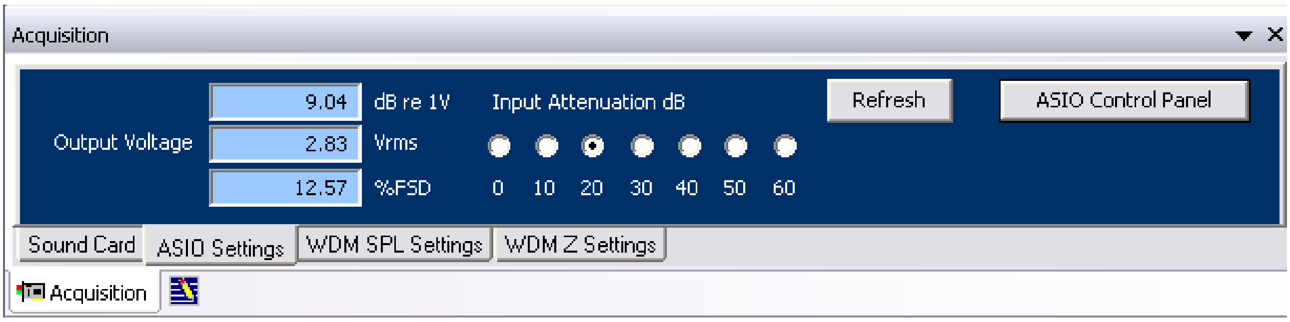


Figure 7 - Acquisition window: Set Output

When you measure you may get a warning that the input level is too high. In that case you need to bring down the input level using the attenuator in 10 dB steps.

Figure 8 - Open SPL window

For the first measurement you should open the SPL window to set the FFT window to exclude any reflections and smoothing etc: (You can then use the button **Measure SPL** to measure all following curves with this setting)

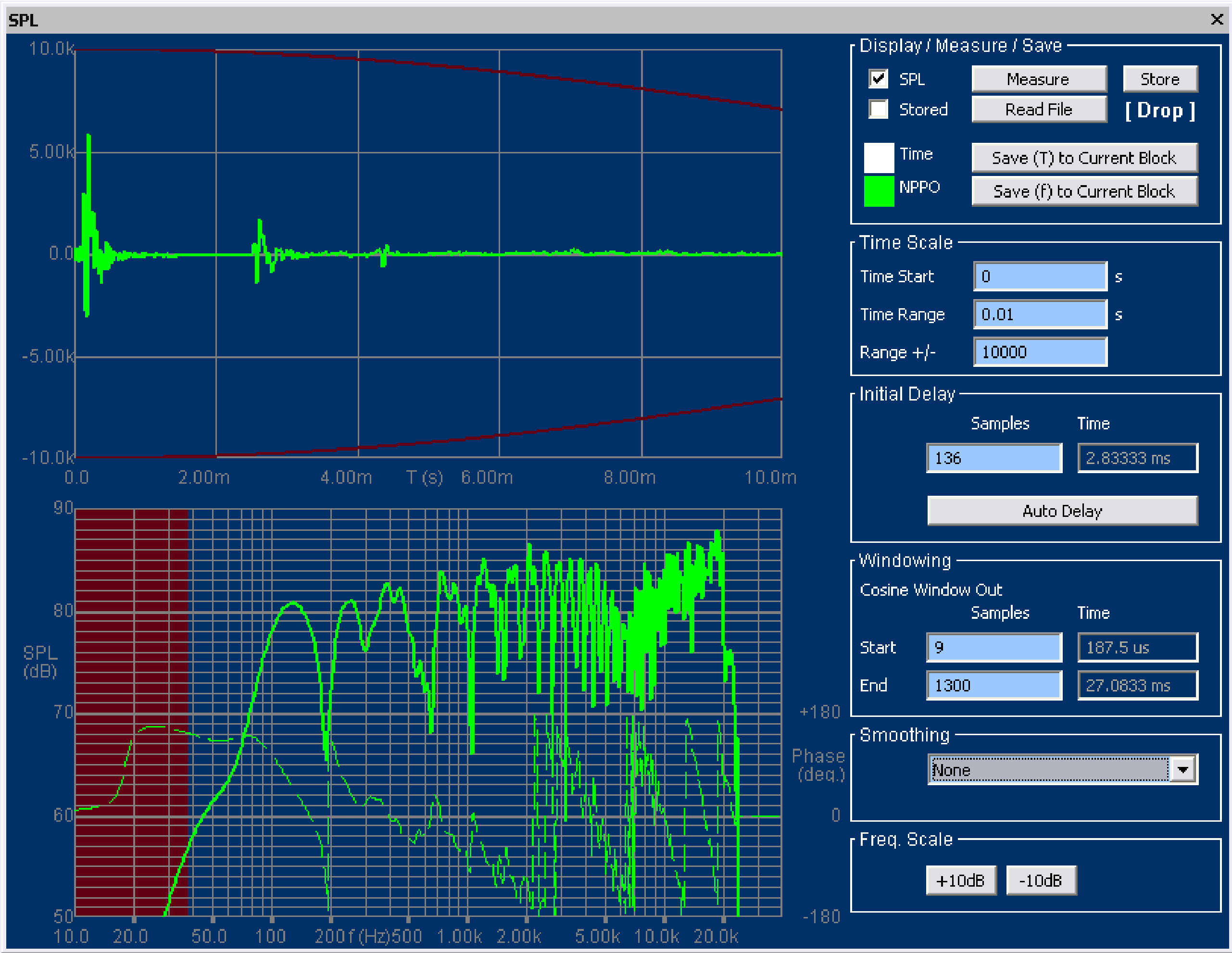


Figure 9 - Initial SPL Window - Set FFT time window for later measurements

Now measure the speaker by clicking the (upper) [Measure] button in the SPL window. The upper part will normally display the large pulse at left thanks to the Auto Delay.

The Time domain impulse response measured 1m from a typical satellite speaker is shown in Fig. 11. The time axis is automatically set to 0 at the start of the main impulse. (This arrived after ~3ms, corresponding to 1m (The speed of sound is ~343m/s or 34.3cm/ms ~1ft/ms).

All you need here is to set a value for the “Cosine Window Out / End”, ideally to exclude all reflections in the time domain.

When using 1300 samples / 27ms the response is valid down to 40 Hz (indicated by the red field). However that gives a very ragged frequency response because we have included all the reflections in the room (Fig. 11). So we need a much smaller window to exclude also the first reflection at ~2.5ms.

Next is shown (Fig. 12) the response using a 2.7ms window, which this time excludes all the reflections. The curve is now nice and smooth, but the red field indicates that the curve is only valid above ~400 Hz.

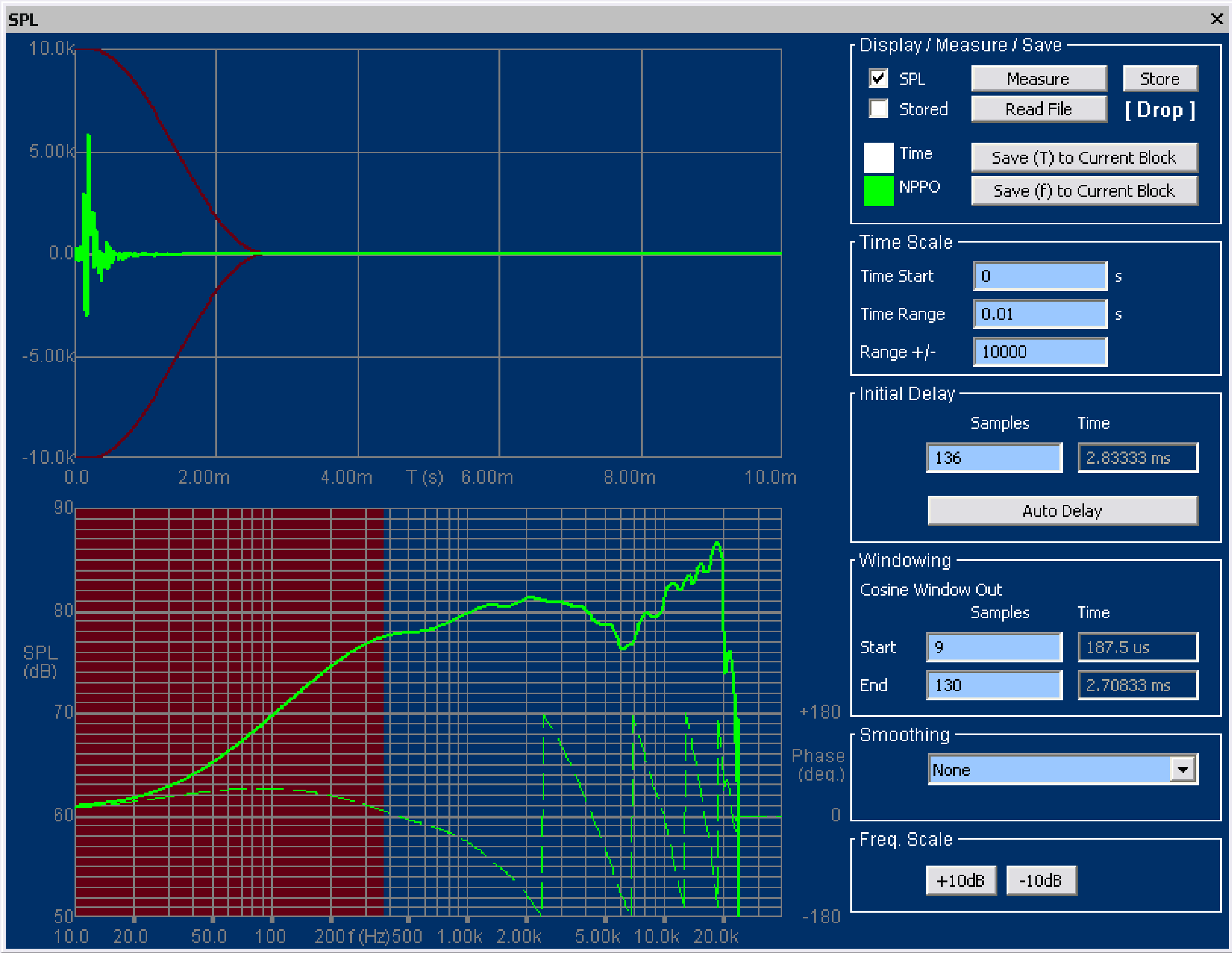


Figure 10 - Satellite response using short 2.7ms window. Only valid from ~400 Hz

Using this short (2.7ms) time window until the first reflection unfortunately gives poor low frequency response, because the 1/f ratio using 2.7ms will only allow ~400 Hz as the lowest frequency (indicated by the red field). The impulse at 2.7ms is the reflection from the floor, which in this example was only 82cm below the speaker and microphone, see Fig. 13.

Click [Save (F) to Current Block)] to transfer the response to the current response main window using the chosen window settings and smoothing if selected.

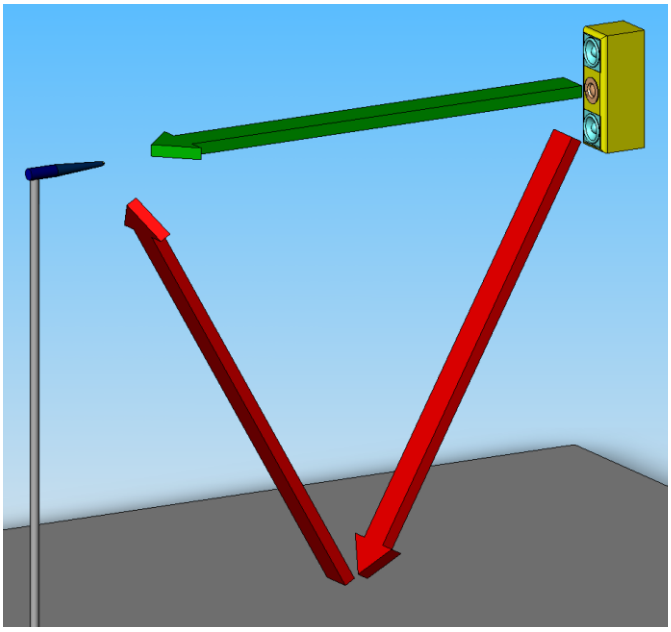


Figure 11 - Satellite close to floor, Red is Reflection

Fig. 13 is illustrating the problem where the reflection from the floor is too close to the main signal, because there is little difference between the direct distance (green arrow) and the reflection path (red arrows). We can do two things to improve this: Move the microphone closer to the speaker and/or move both speaker and microphone further away from the floor (and other surfaces).

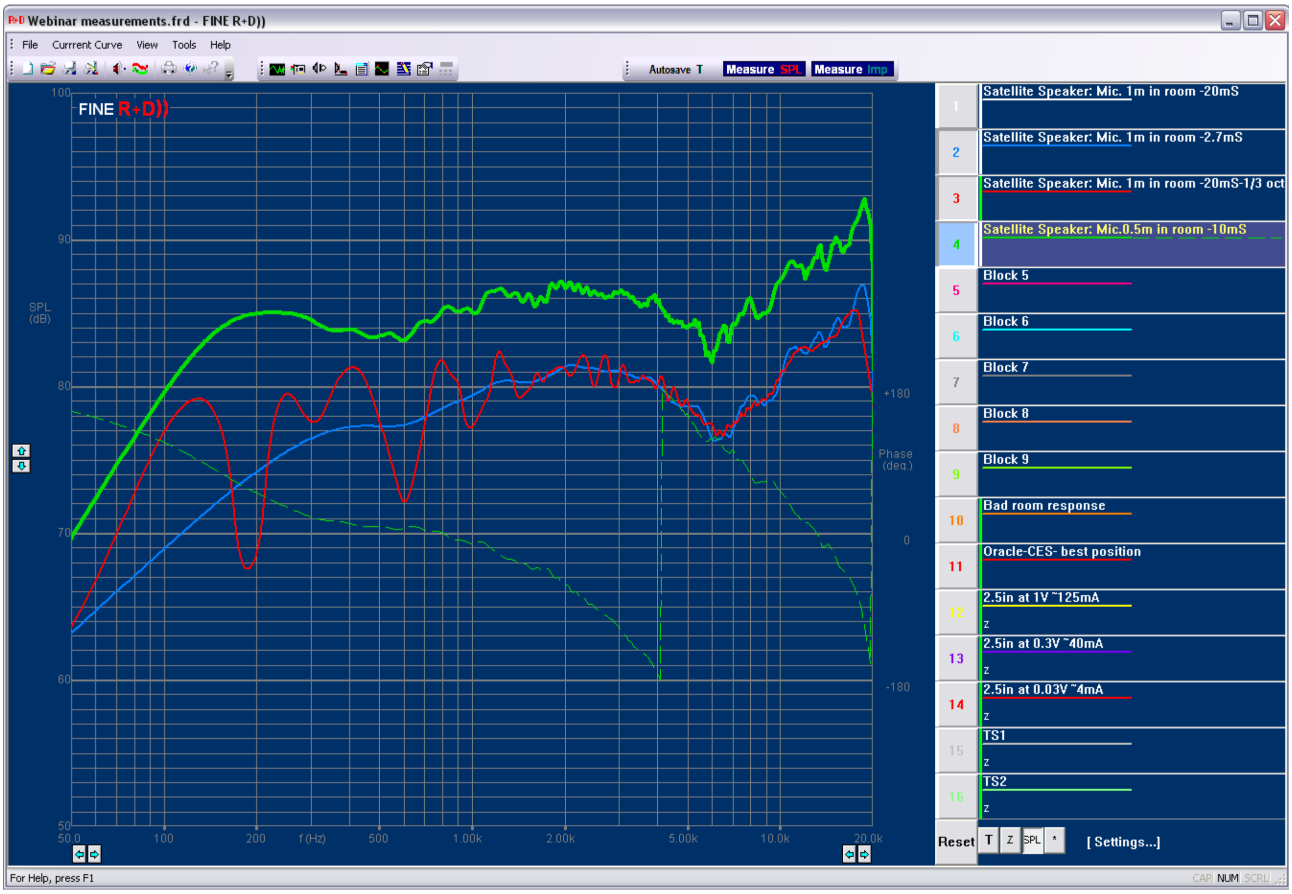


Figure 12 - Satellite Speaker measured at 1m and 0.5m/away from walls (Green curve #4)

Both have been done for the green curve no. 4 (Fig. 15), by moving both the microphone and speaker up to 154cm from the floor and adjusting the microphone distance to 0.5m. This time we get the reflections much later and can use a window of 10.4ms. Therefore we can measure the real low frequency response of the satellite, starting from approximately 150 Hz.

Note the dashed line in the SPL window is the acoustical phase response of the corresponding SPL/ frequency responses. All FINE R+D measurements are full resolution responses with phase, but the phase is here only selected for curve no.4

As soon as you have found a satisfying SPL window and suitable microphone distance plus speaker placement in the room you can continue to measure only by clicking the  button, and you will every time get a new curve in another color. You will be prompted for inserting a comment for each curve.

This way of saving measurements is called frequency NPPO (green)

Note: You can also save the raw (Time-) measurements in the SPL window using the white button [Save (T) to the current Block] Fig. 14. This will save the raw (un-windowed) response and show it in the main display. Despite it looks ragged it may later be used for calculation of distortion and waterfall.

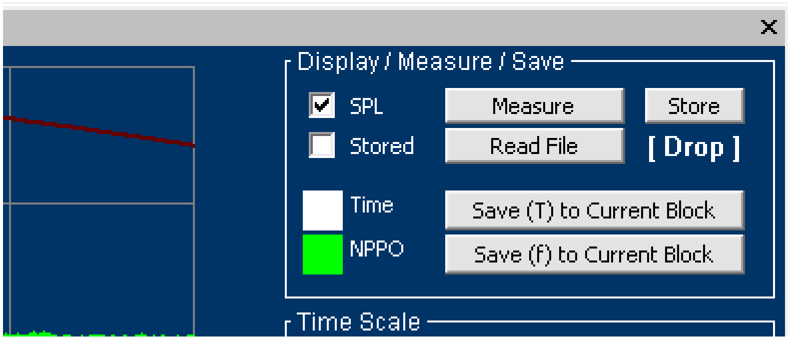


Figure 13 - Time / frequency (NPPO) Save in SPL window

Room Response Measurements

In case you want to measure how the loud speaker actually sounds in your room, you should include all the reflections, by using the maximum Window/End.

The next example is a measurement of a very good Hi-Fi loudspeaker measured in a good listening room. The first (brown) curve was measured in the listening position on axis with the tweeter. Note that the curve was 1/3 octave smoothed by selecting 1/3Oct in the SPL window.

This response is well behaved above 1000 Hz, while the low frequency peaks and dips are caused by the standing waves in the room, Fig. 16.

In stead of the 1/3 octave smoothing you can also select ½ octave or even 1/1 octave smoothing. 1/1 octave smoothing will not show the room modes, but only indicate the general sound balance in the room.

The other (red) curve was measured after the loudspeaker was moved to another position in the room. Some of the low frequency peaks are now less dominating, indicating that these standing waves are less excited in the room.

These two examples are well behaved room responses; however in many cases you will see much more variation in the low frequency range. FINE R+D is extremely useful for adjusting your (room-) EQ and/or finding the best loudspeaker position in the room. Also furniture positions and damping material can be optimised this way.

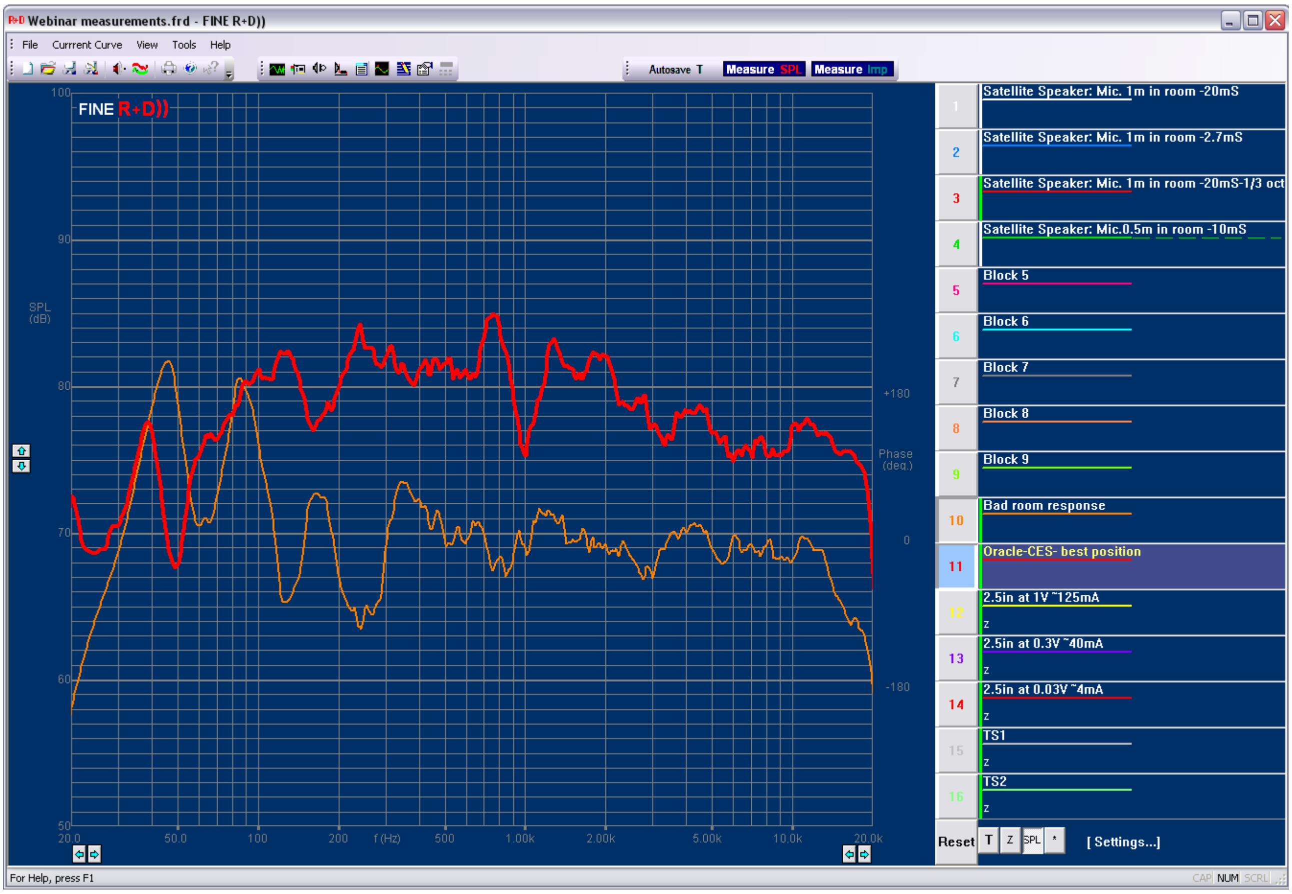


Figure 14 - Room responses measured far from loudspeaker

Impedance Measurement

Measuring loudspeaker impedance is done simply by clicking the right button 

Since the impedance measurement is purely electrical, the range and time window is already defined and set with the Auto delay. There is normally no need to open the impedance window (Z view) since the “Cosine Window Out/End” should be using the max (9600) always.

Impedances down to 2 ohms can be measured well, if the max power of 25W is not exceeded. Lower impedances are also possible at low power.

The measured impedance may later be used to calculate the TS Parameters (Thiele/Small Parameters). If the Added Mass or Added Volume option is used you need to measure an additional impedance curve with the added mass/Volume. These two curves should then be transferred to the TS window. (See later page 16).

Please see Fig.18, showing two impedance curves in the lower part of the main screen. Here is used the combined (SPL +Imp) display with the *button depressed Fig. 17.
The impedance curves can be scaled independently of the SPL curves with the up/down arrows on the right side.

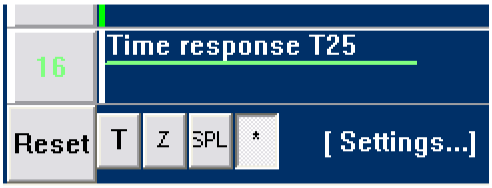


Figure 15 – Combined SPL+ Imp (Z) display selected [*]

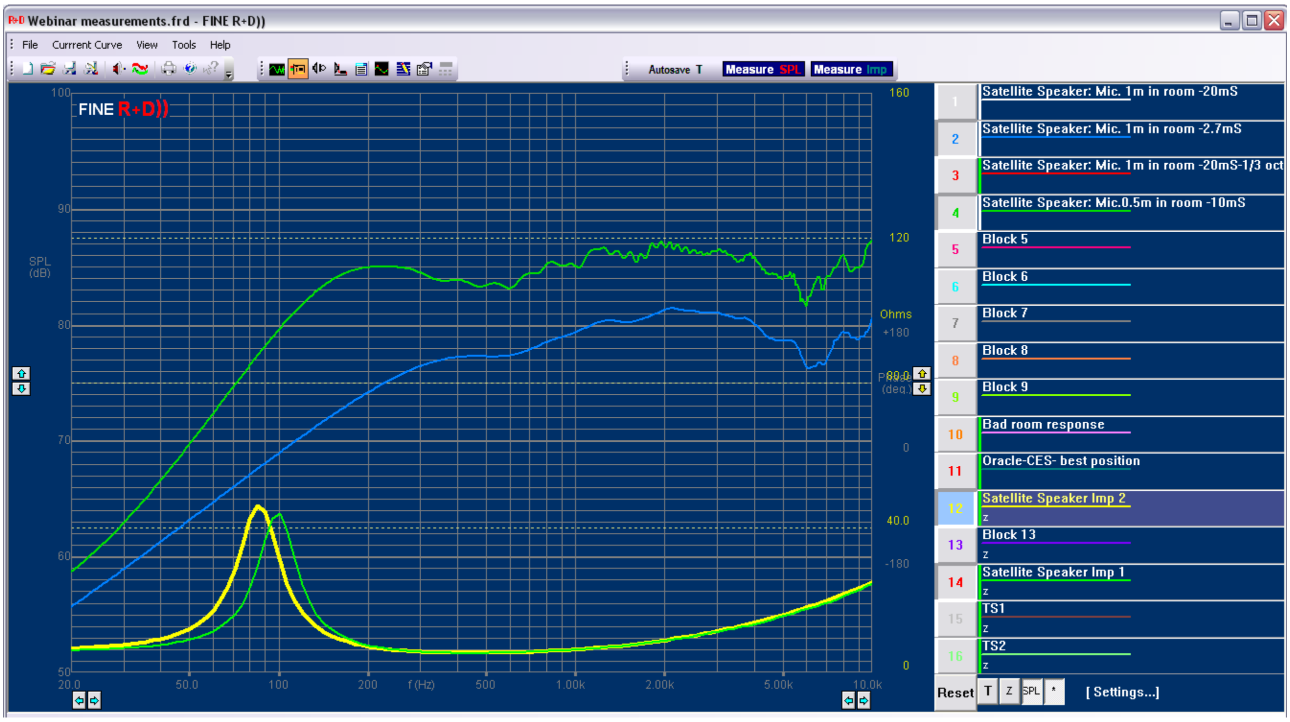


Figure 16 - Main Display with both SPL and Impedance

Thiele/Small Parameter Measurements

In this chapter I want to measure the TS parameters of an 8 inch subwoofer. Press the TS Parameter button

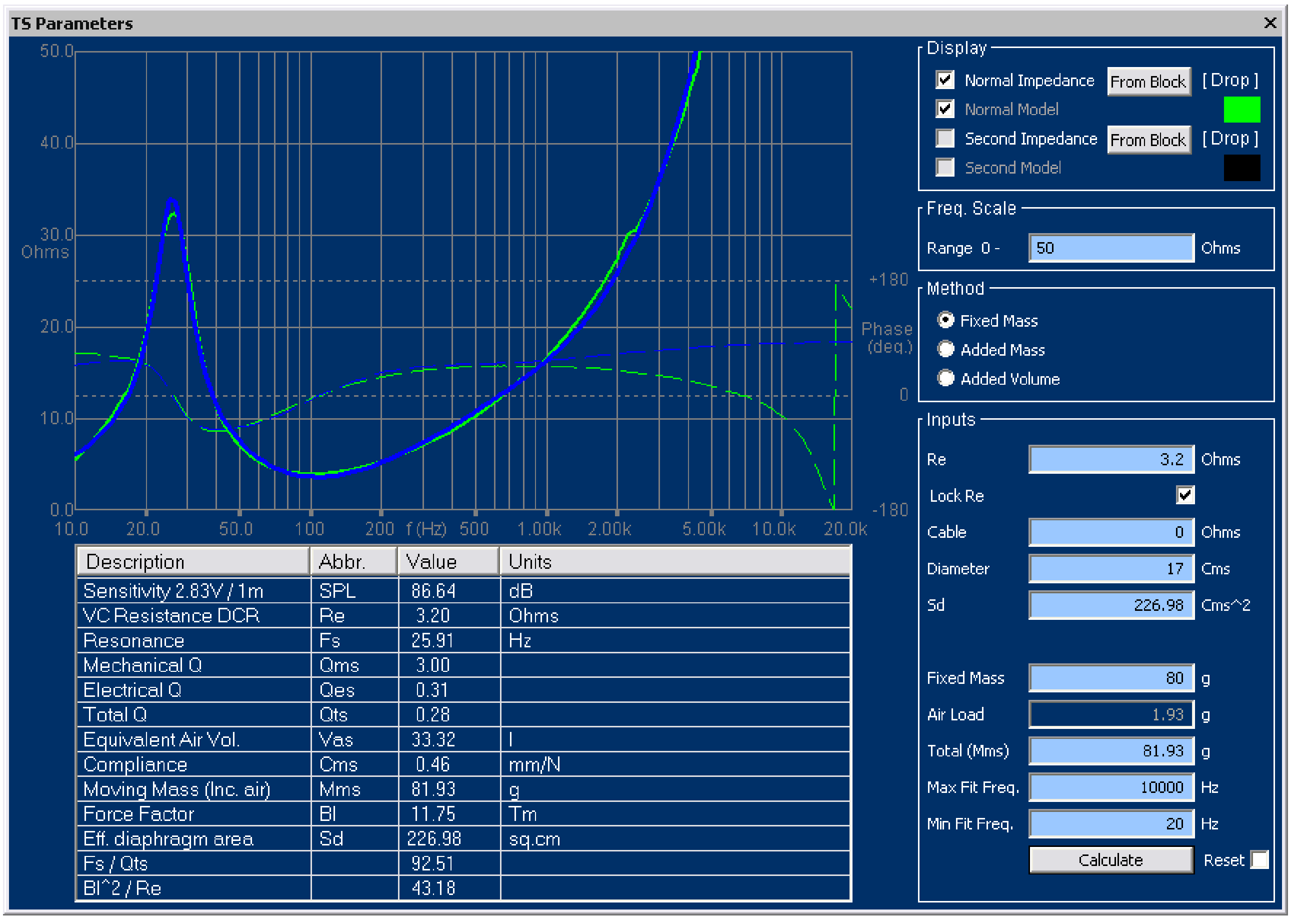
First I need to input the cone area Sd and Re Fig. 19. I choose to input the effective diameter of 17cm (half of surround) and Sd will automatically be calculated. FINE R+D can estimate Re (DCR) from the impedance curve, but in order to get the best accuracy I have measured Re=3.2 ohms with a precise Multimeter (DVM). That value is fixed by lock [v].

Now you can choose a measured impedance curve if not already present from the last measurement. There are two ways to select an already measured impedance curve:
(1) Either press [block] and find the curve in the table, or (2) drag the curve from the main display by holding down the left mouse button and drop it over the [Drop] field.
The Fixed Mass option is the most accurate method. However this requires that you know the total moving mass of cone + Voice Coil + half surround + half spider (including dust cap and glue etc.) = Fixed Mass Md. If you know this mass (from the datasheet) you should enter that, which causes the air load mass (Mair) to be calculated, Md + Mair=Mms.

When the “Calculate” button is pressed, FINE R+D will calculate all the TS parameters by fitting a simulated impedance curve (blue) in the chosen frequency range. In this case we get a very good fit around Fs, which is important for getting accurate TS parameters.

If none of the masses are known you can choose the standard Added Mass or Added Box methods, which both require you to make a second impedance measurement. In those cases we end up with a total of 4 curves, 2 measured and 2 fitted.

If the curve fitting is bad, you should set [x] in the Reset box, causing a restart of fitting.



Description	Abbr.	Value	Units
Sensitivity 2.83V / 1m	SPL	86.64	dB
VC Resistance DCR	Re	3.20	Ohms
Resonance	Fs	25.91	Hz
Mechanical Q	Qms	3.00	
Electrical Q	Qes	0.31	
Total Q	Qts	0.28	
Equivalent Air Vol.	Vas	33.32	l
Compliance	Cms	0.46	mm/N
Moving Mass (Inc. air)	Mms	81.93	g
Force Factor	Bl	11.75	Tm
Eff. diaphragm area	Sd	226.98	sq.cm
Fs / Qts		92.51	
Bl^2 / Re		43.18	

Figure 17 – TS Parameter window with accurate curve fitting

THD and Harmonic Distortion

In order to calculate distortion you need the raw time (T) data. If you have not yet measured that go to the SPL window and make a new measurement, and then save it using the white button [Save (T) to the current Block] or drag it to the distortion window.

Now open the distortion window and select the measurement(s) having the raw time (T-) response saved. Press [From block] or drag the Time response to the [drop] field. The saved time blocks have a white vertical stripe after the curve number. (Frequency NPPO files have a green stripe)

Figure 18 - Select raw (T) time data for distortion calculation

Click [Calculate from SPL Chirp] to get the THD and 2nd-9th harmonics calculated and displayed in % (x1-x10) for a 2.5in full range, see Fig. 21. The red fill at left is invalid data.

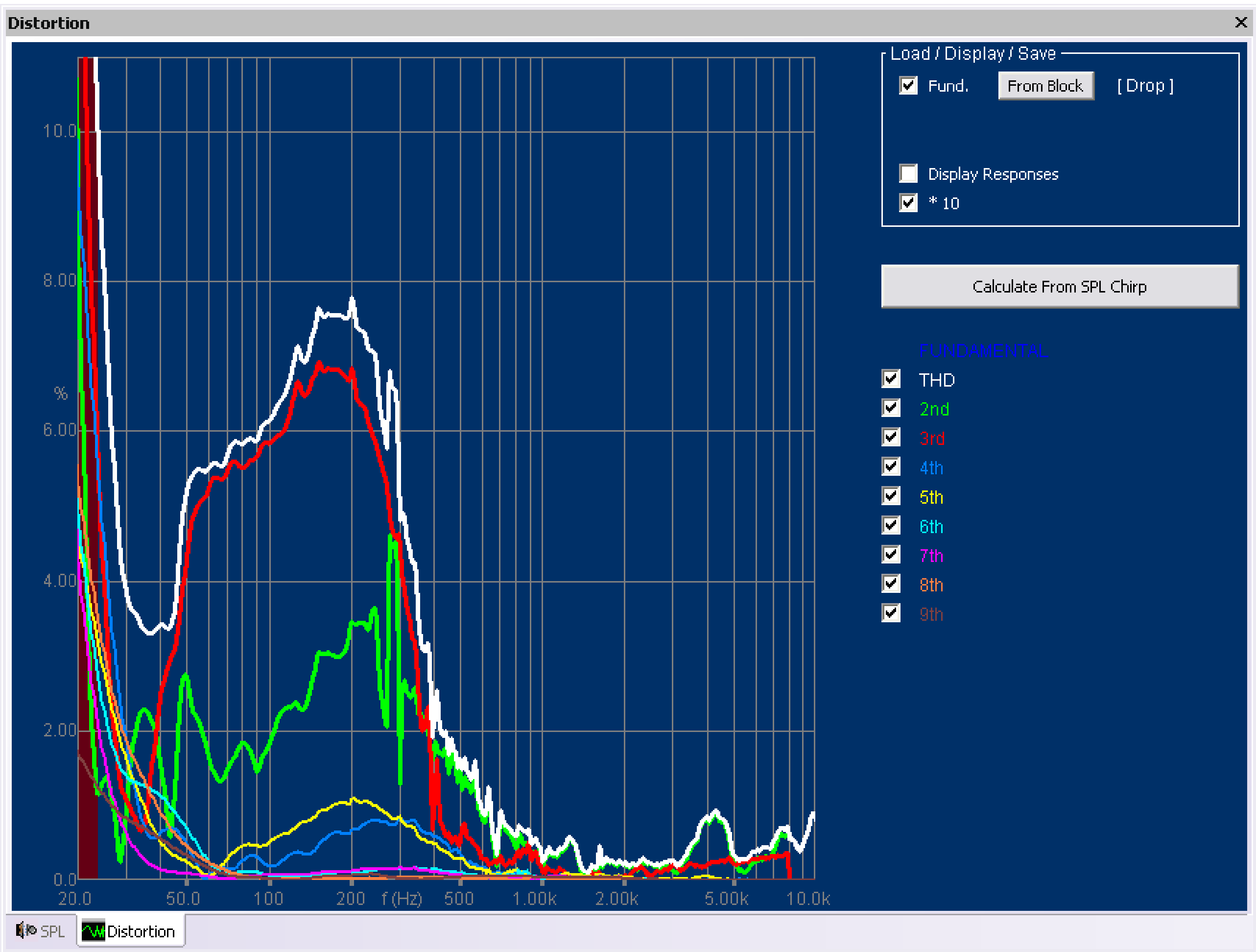


Figure 19 - THD and 2nd-9th harmonic Distortion x10 for 2.5in full range

Fig 22 gives the THD and 2nd-9th harmonics for the same 2.5, but now as responses in dB. The fundamental is also shown in blue.

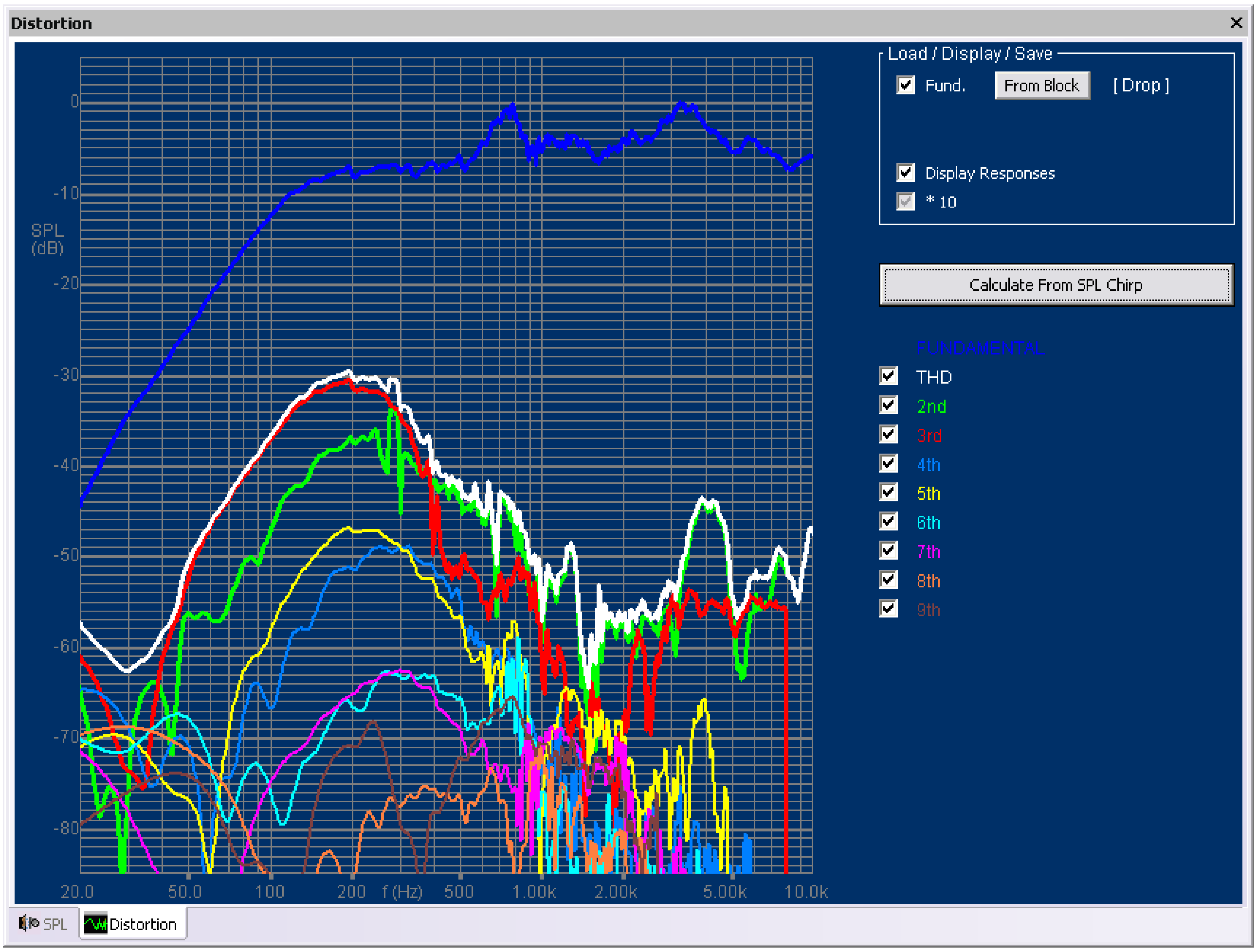


Figure 20 - THD and 2nd-9th harmonics for the same 2.5, but now as responses in dB

Waterfall (Cumulative Decay Spectrum)

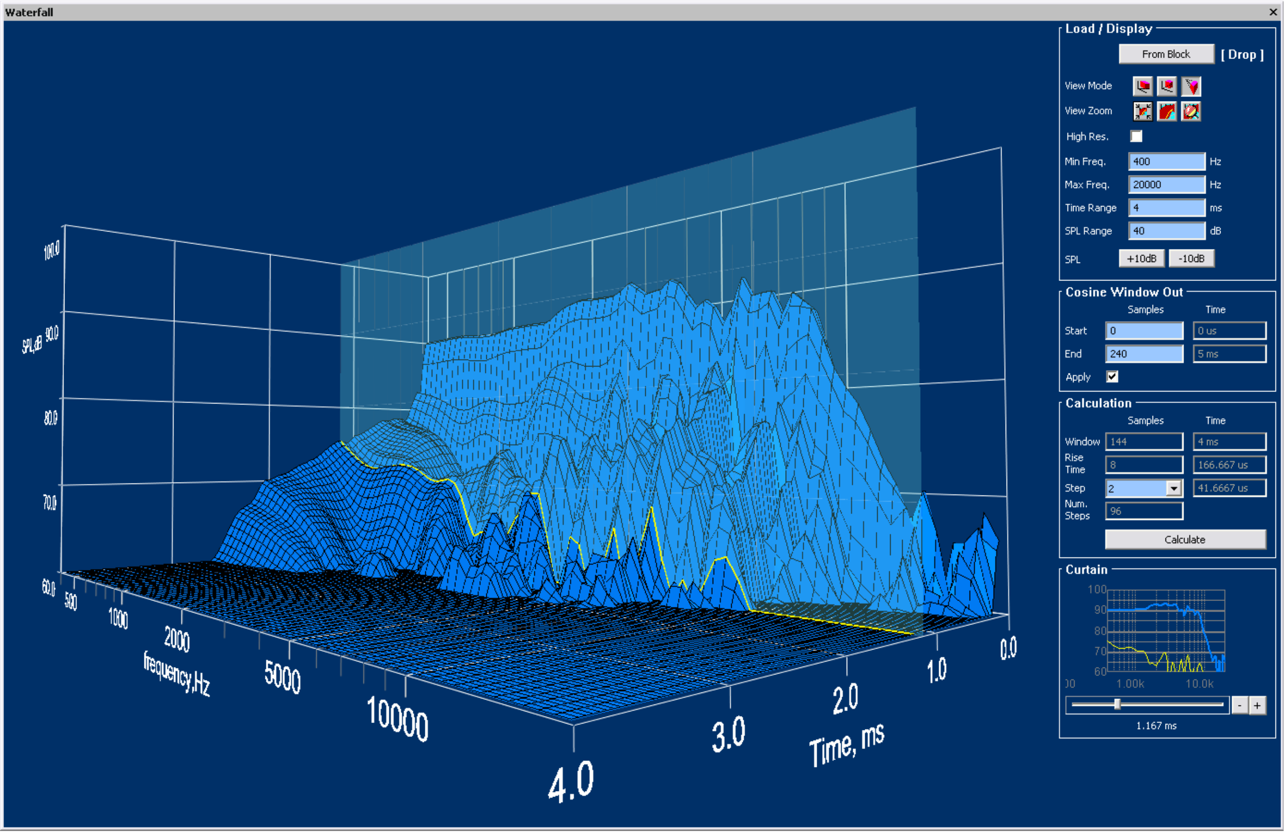


Figure 21 - Example of Waterfall from a woofer with Time Curtain @ 1.167 mS

The raw time data further gives the possibility to view the decay of a measured response as a waterfall called Cumulative Decay spectrum. Thereby resonances will show clearly as ridges parallel to the time axis, while reflections will show as dips and peaks in the time direction in the 3D display.

The special Time Curtain may be used to see the individual time slices as shown in Fig. 21, where the Time Curtain is set at 1.167 mS. Use the lower right screen to control the Time Curtain, however you can also just roll the mouse wheel to change the position.

Press [From block] or drag the Time response you want to see as waterfall to the [drop] field like in the distortion window. The saved time blocks have a white vertical stripe after the curve number. (Frequency NPPO files have a green stripe).

Press [Calculate] and the waterfall will then be calculated using default settings. When you change settings the waterfall curve will turn grey to indicate that a recalculation is needed. There are several settings for the display like Isometric + shear View and Perspective View. In all cases you can rotate and move the curve by holding and dragging with the left mouse button

The default resolution is 1/12 octave. Press High Res. [ ] to show the Waterfall in the full 1/48 octave resolution.

Figure 22 - Select Raw Time Data from Block

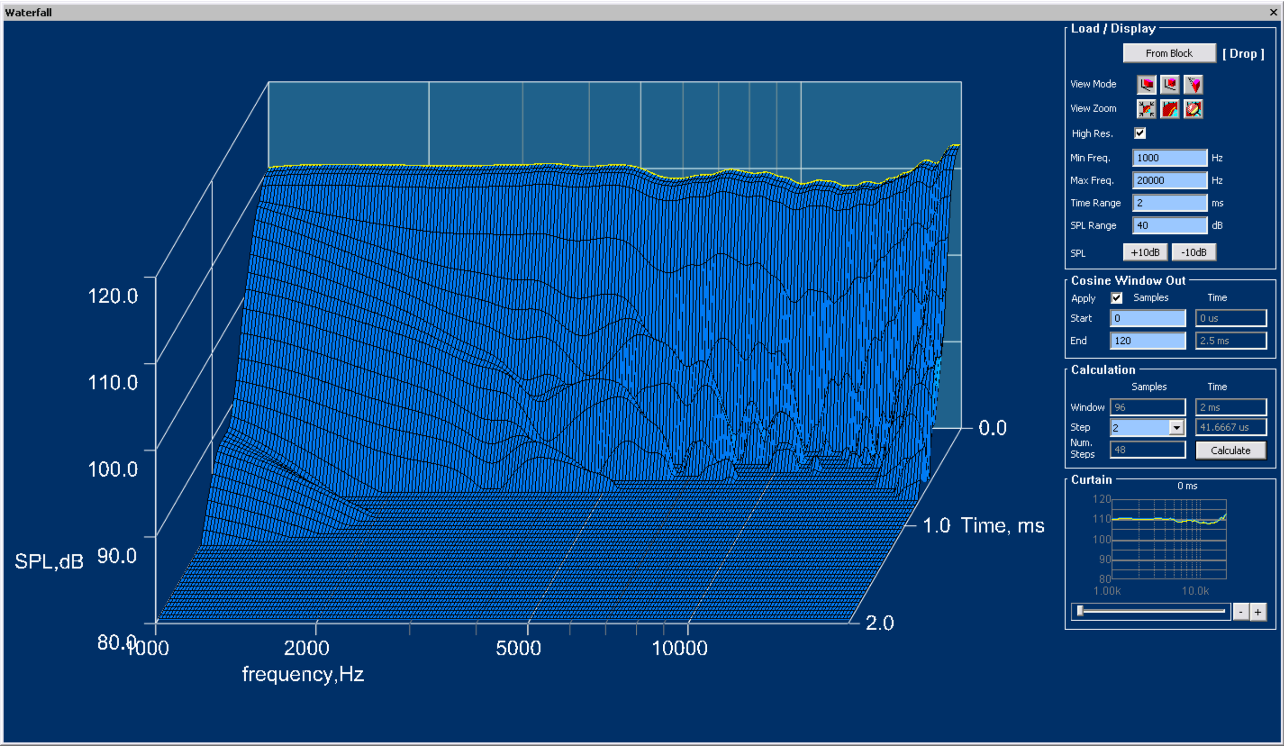


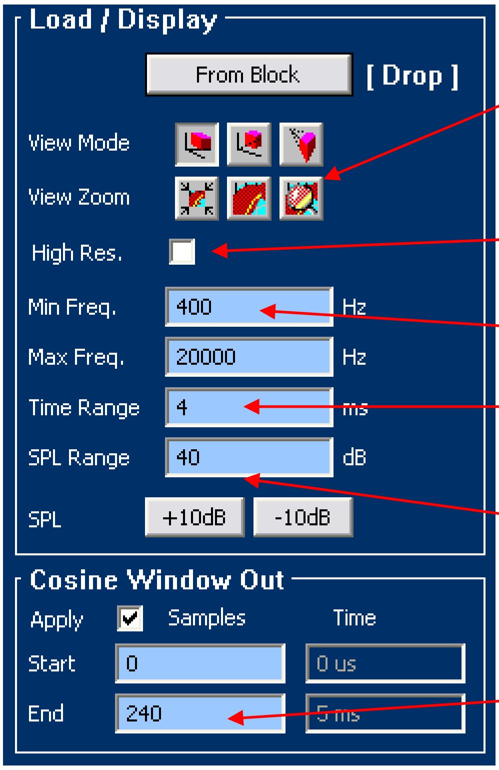
Figure 23 - Waterfall of well-behaved Tweeter in High Resolution (1/48 Oct) View Mode

Fig. 23 shows the waterfall of a good tweeter using the default Isometric display in High Resolution 1/48 Oct. mode. The time Range is this time set to max 2mS, and the displayed waterfall shows that the response has dropped below the floor after approximately 1mS. The step size between slices is set to 2, so the automatic settings indicate there will be 48 slices in the waterfall.

Fig. 24 explains how to set the Display settings. The high resolution 1/48 Oct. mode is very detailed and may take some CPU time to calculate. Be sure to set the (Cosine) Window End to exclude the unwanted reflections like you have done when measuring the SPL response.

Fig. 25 shows the setup of the waterfall slices and Time Curtain. The max Time is automatically taken from the Time Range (see Fig. 24) so you only have to specify the step (slice) size, which has to be an integer number. The step size is calculated in the right column, and the number of steps is automatically calculated to fill the available display.

Below you find the setting of the special Time Curtain. Moving the lower slider with the left mouse button will move the Time Curtain in the main display, while highlighting the current response, which is also shown in the small display. Alternatively you may move the Time Curtain with the + / - buttons.



Select View Mode and Zoom with these buttons

Select 1/48 Oct. High resolution Display (Slower)

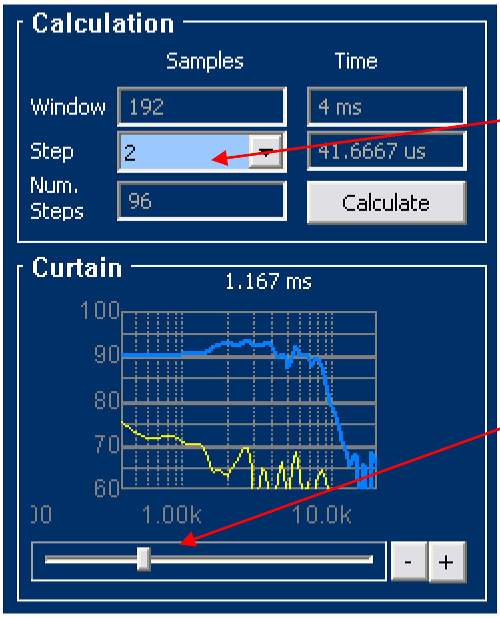
Set min. and max display frequency

Set Time Range

Choose SPL range

Set (Cosine-) Window End. Normally use same setting as for frequency measurement (To avoid reflections from room)

Figure 24 - Waterfall Display settings

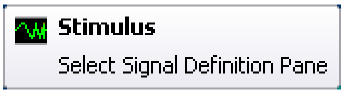


Most of these settings are automatic. You only have to select the step size (size of Time slice)

The slider below the small picture is used to set the Time Curtain, which will highlight a certain Time slice in the main display. The slice is also shown here in the small picture

Figure 25 - Slices and Curtain settings

Stimulus



Here you can set the used sweep range. It is recommended to start min ½ octave lower than needed. For ASIO use max. =23kHz with 48kHz sampling. For WDM max freq is 21 kHz. The sweep time can be set from 0.1 to 2.5 sec, but below 0.5 sec is not recommended.

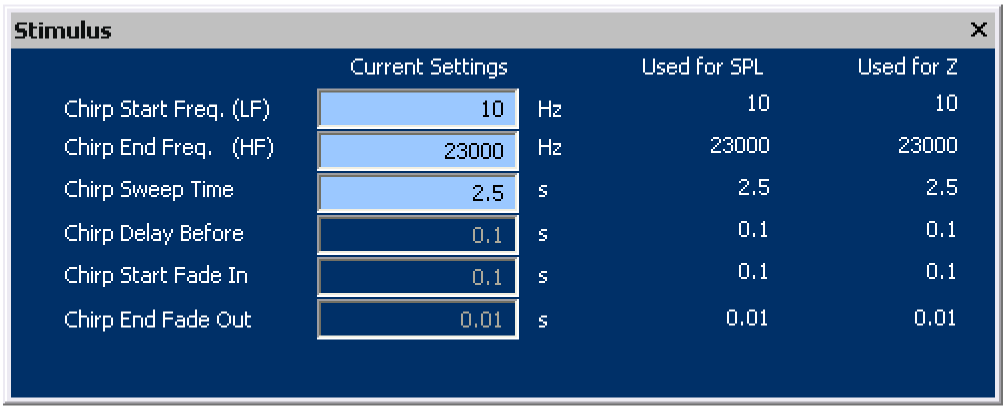


Figure 26 - Stimulus settings

Import/Export of SPL and Imp to FINE X- over and other

FINE R+D is a unique tool for measuring loudspeaker responses, which may be exported in the standard Loudsoft format *.lab or text format, for example to the acclaimed software FINE X-over (www.loudsoft.com). FINE R+D will export the responses *with phase*, which is necessary for getting precise cross-over designs, that measure exactly as simulated. The SPL and Impedance can also be imported or exported in standard *.txt formats for most other software, like Klippel, VACS, LMS and MLSSA, see Fig 25.

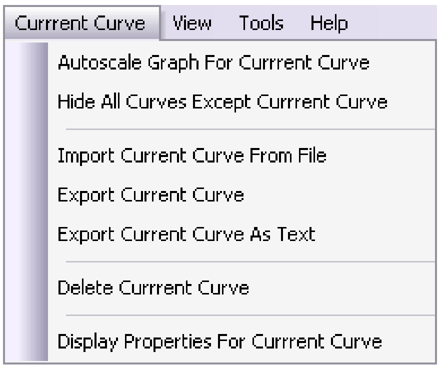


Figure 27 - SPL and Impedance export from FINE R+D

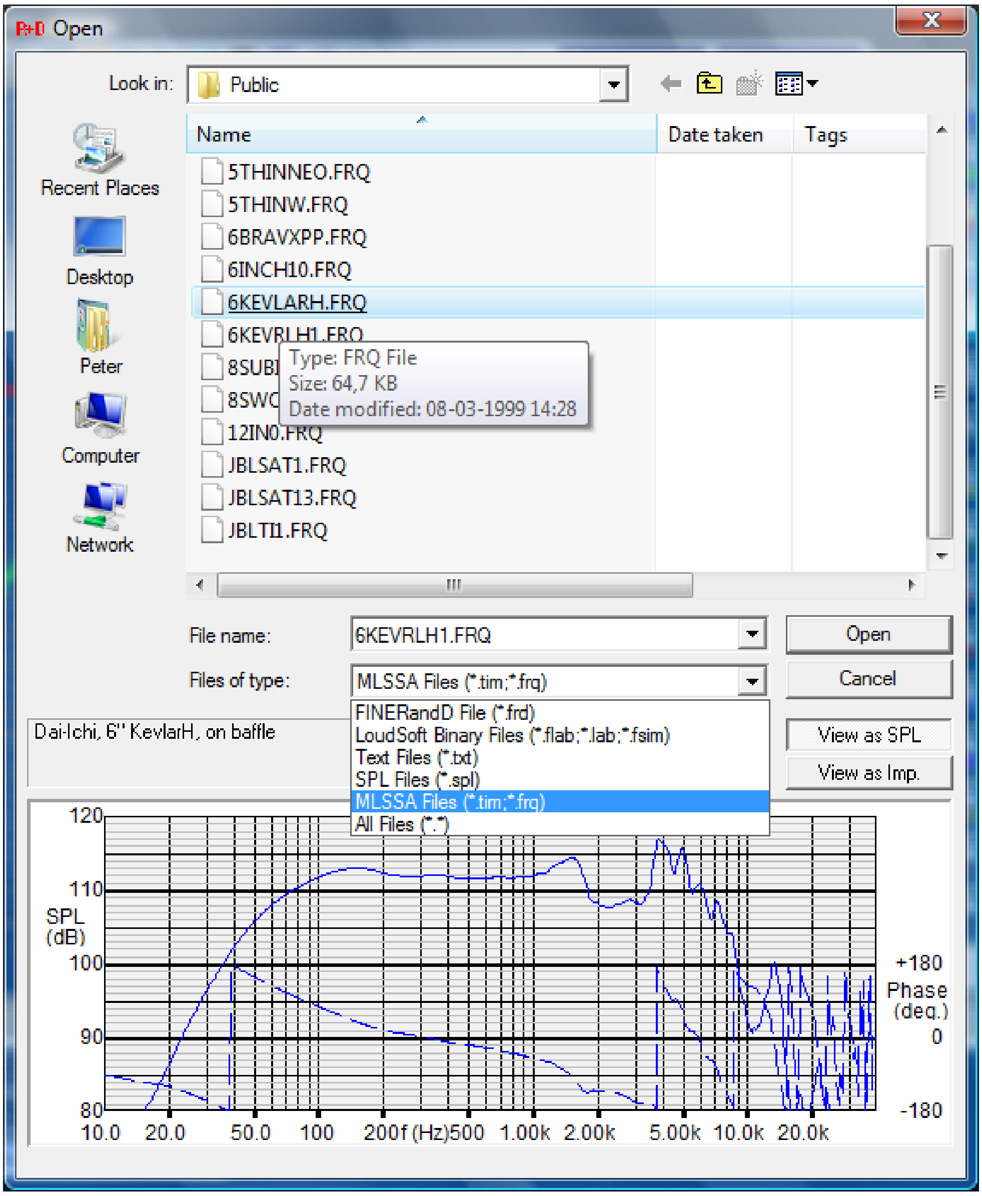


Figure 28 - File (Import) Preview of FRQ, TXT, Lab FSIM etc. responses

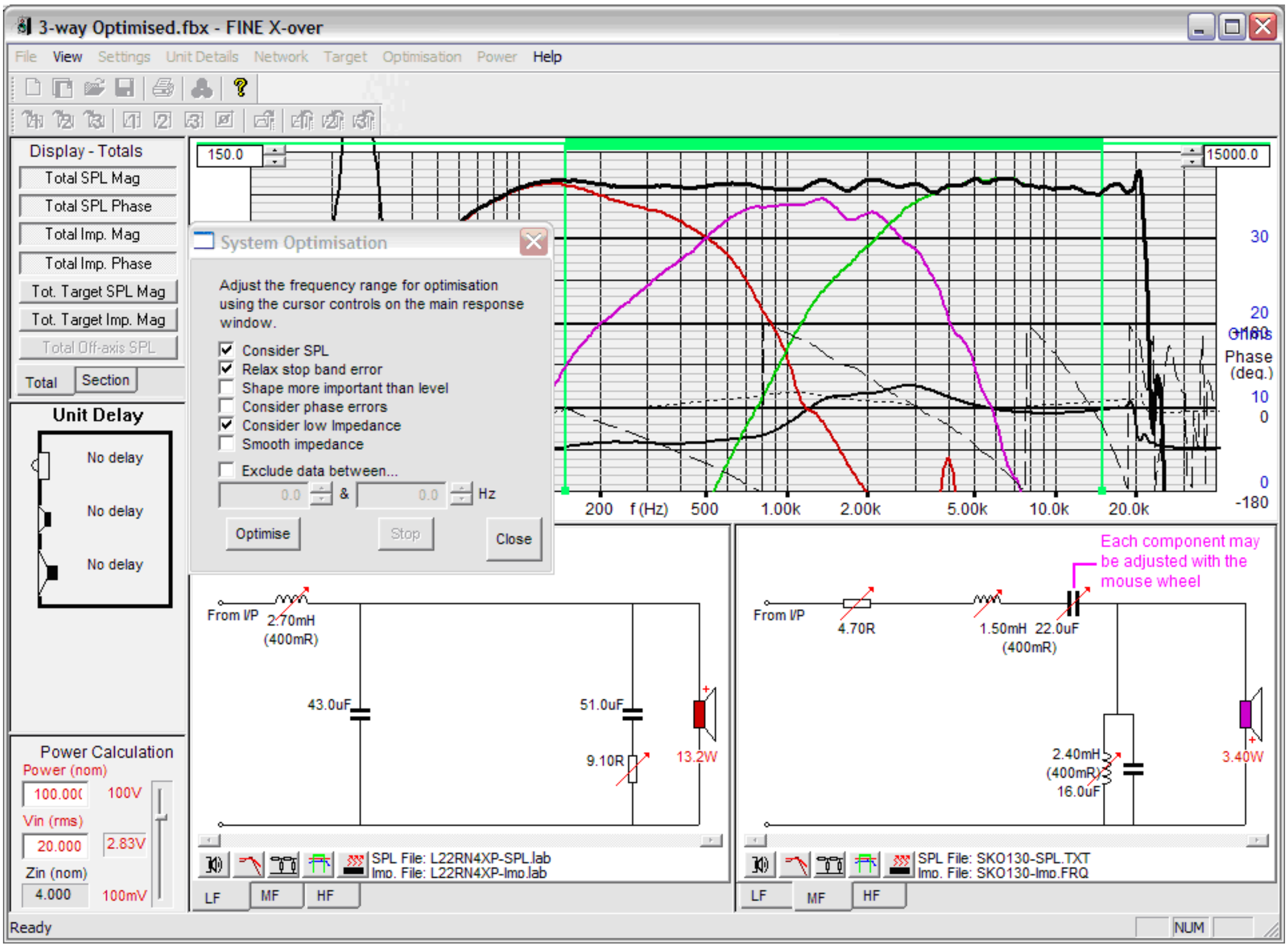


Figure 29 - 3-way crossover optimised in FINE X-over using SPL and Imp from FINE R+D

The example above (Fig.26) shows a 3-way cross-over designed in FINE X-over. The SPL responses and impedance were imported with phase directly from FINE R+D. The cross-over was optimized while keeping the minimum impedance (here 3.2 ohms for 4 ohms nominal Imp.).

Printing and Reports (PDF)

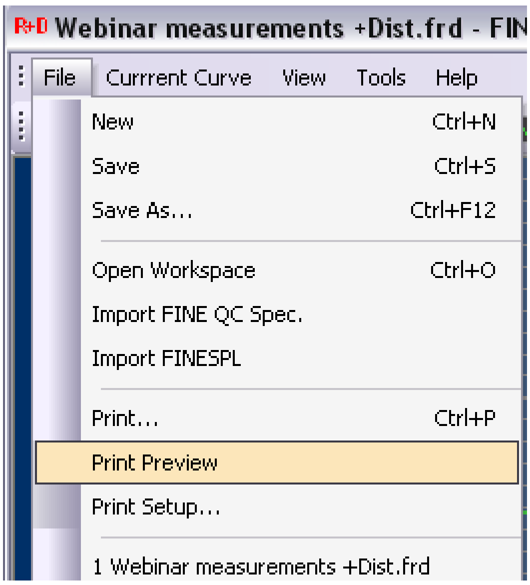


Figure 30 - Print Preview

The print command will print up to 5 pages including Main Overview, SPL and Imp windows, TS Parameters and Distortion. Select Print Preview (Expand down if not visible) to see what you print in advance, Fig. 28.

PDF Reports can be made by downloading for example the free PDF writer 995 http://www.pdf995.com/ . Then you can print 1-5 pages to this and thereby create a fine PDF-report (See example in documents)

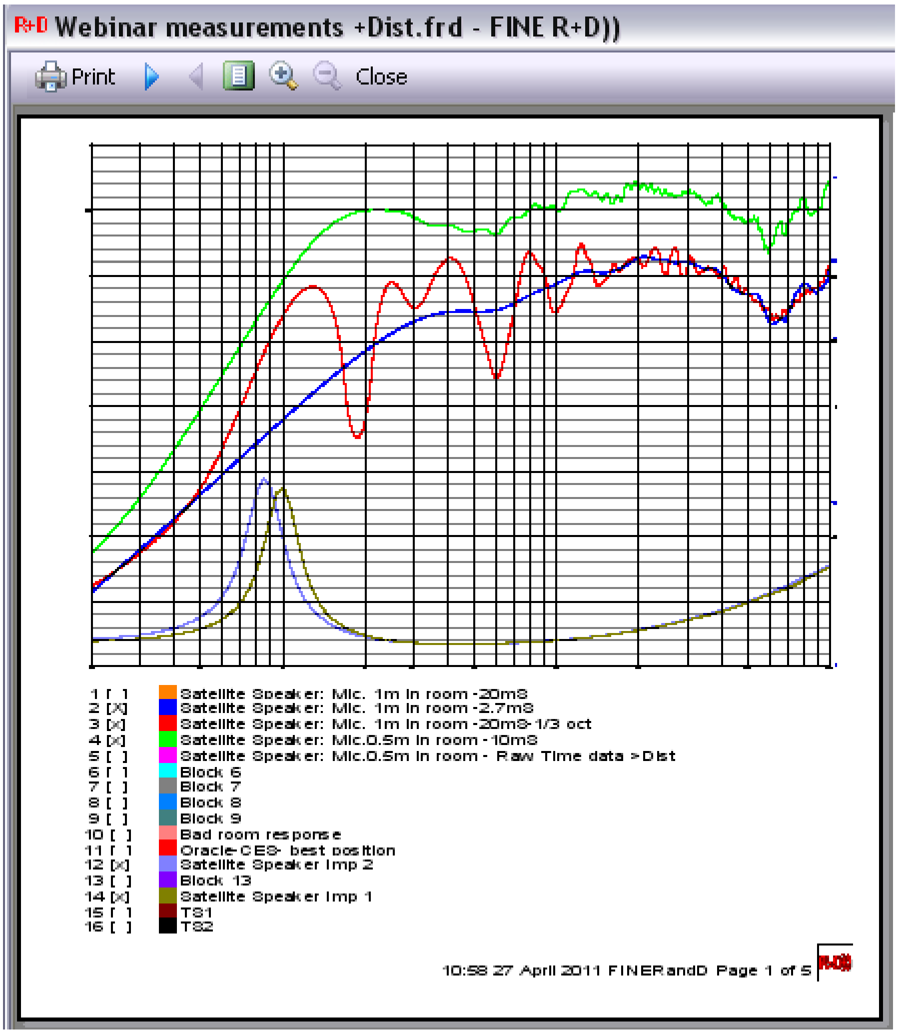


Figure 31 - Print Preview Main, SPL and Imp, TS Pars and Distortion

Properties

Go to properties  (View/Toolbars and Docking Windows/Properties)

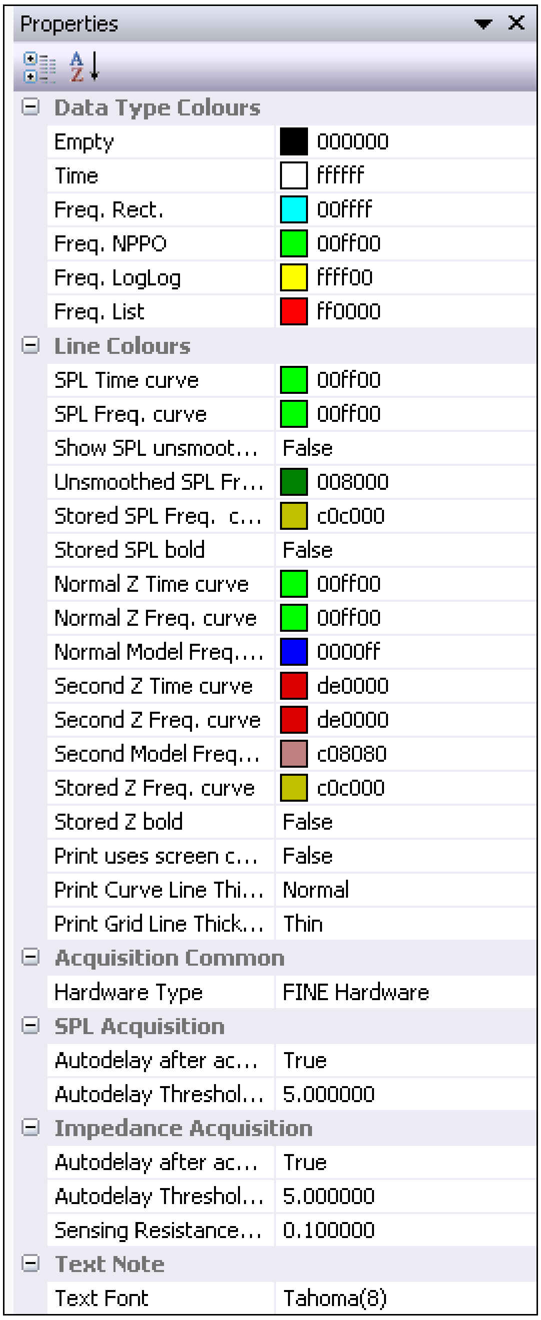


Figure 32 – Properties

(The Auto Delay has a default amplitude threshold of 5% of max. If the program is triggered on unwanted noise spikes, you can change this setting in Properties, but only if you are really sure).

Select the hardware you are using in Acquisition Common: (Default is FINE Hardware).

Windows Sound Settings / WDM drivers

Figure 33 - Windows Sound Input

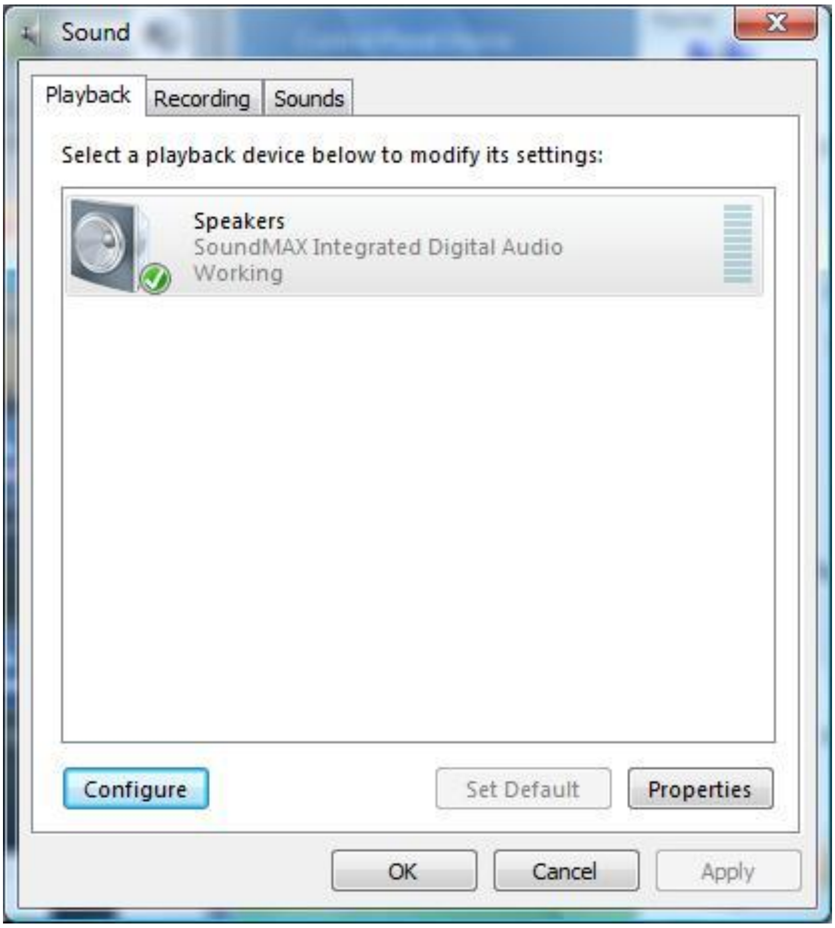


Figure 34 – Windows Sound Output

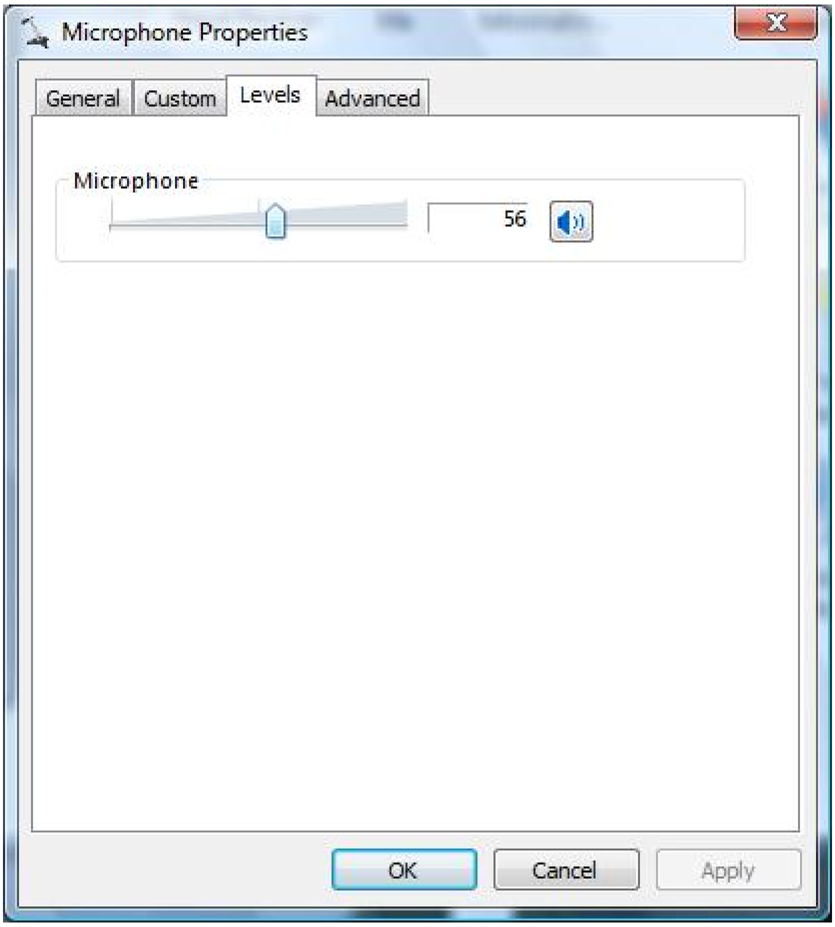


Figure 35 – Windows Microphone Level

FAQ

FINE Hardware USB problems

If you have USB Problems with the FINE Hardware here are some tips:

- If the USB cable is plugged out, DO remember to switch off the power to the FINE hardware, and switch it on again, before you plug the USB cable in the computer again. Otherwise Windows might not be able to detect the ASIO USB hardware.
- If a similar problem happens when restarting the computer, you can go through the following procedure to solve the problem:
  - Unplug the USB cable from the computer
  - Switch off the power to the FINE Hardware
  - Switch it on again
  - Plug in the USB cable

Verify Driver Settings

If you get error messages or wrong levels during measurements, you should verify that the correct drivers are being used and selected. Go to page 5 and perform the listed checks.

You must have installed the ASIO4ALL drivers (page 4), and selected the correct hardware (page4).

Peter Larsen

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