

MANUAL



FINE **R+D)))**™

R & D Measurement system



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Overview

Measure SPL Step, Measure SPL save as Time (F3)
Step to next curve, measure SPL with the last used settings and save the data.
Will prompt for description text
Press Ctrl to suppress the step

Measure Imp Step, Measure Imp. save as NPPO (F4)
Step to next curve, measure Impedance with the last used settings and save the Freq. data
Will prompt for description text
Press Ctrl to suppress the step

Real Time Analyser
RTA Pane

Block Averaging
Block Average Pane

Block Calculator
Block Math Pane

Waterfall (F8)
Waterfall Pane

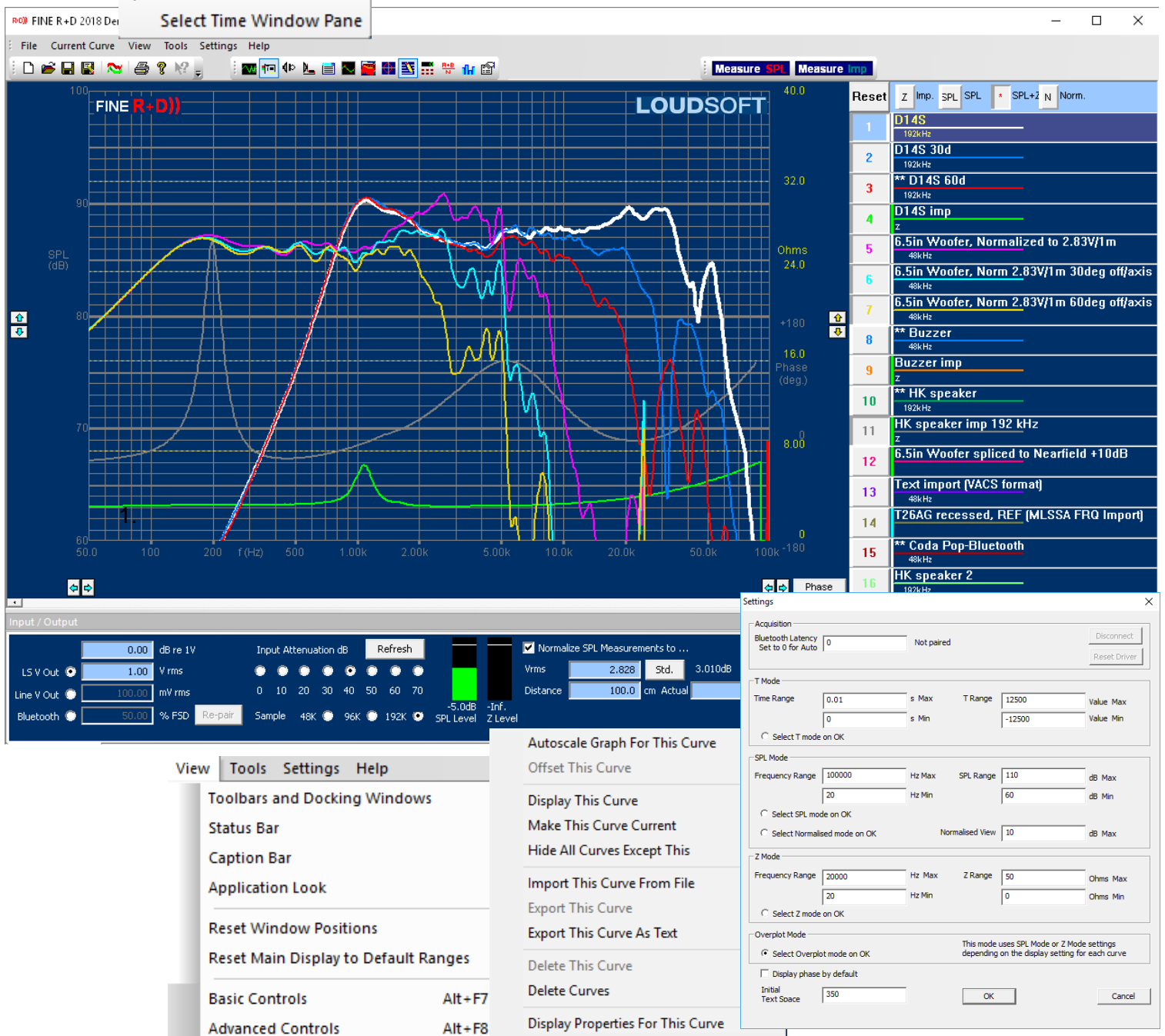
Distortion
Distortion Pane

TS Parameters
Select Thiele/Small Parameter Calculation Pane

Z View
Select Impedance Measurement Pane

Global Notes
Show Notes
Properties
Select Properties Pane

SPL
Select Time Window Pane



1. Installation / How to get started

1. First install the FINE R+D software as from the CD or run setup.exe from the download without the hardware connected. If you choose to start the software later you must Run as ADMINISTRATOR. You should install for ALL users.
2. Now connect the power supply to the hardware and connect the hardware to the computer using the USB cable.
3. Start FINE R+D
4. The first time you run FINE R+D the software will ask for your microphone calibration data (Sensitivity), see Figure 1.
 - a. Enter the microphone sensitivity from your Microphone Calibration Chart (see Figure 2). Note this is a negative number, i.e. -47.33 dB
 - b. If you have no calibration data, you can calibrate your microphone using a calibrator.

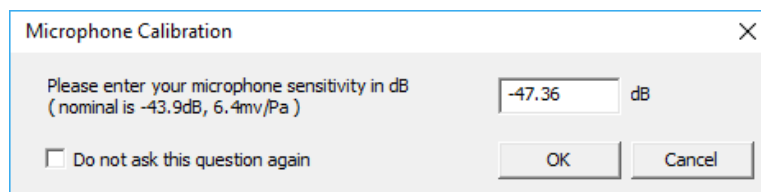


Figure 1 - Initial microphone calibration

5. The FINE R+D system is factory calibrated. However, you may perform an individual calibration if needed (Sec. 22). This can improve the accuracy slightly at high

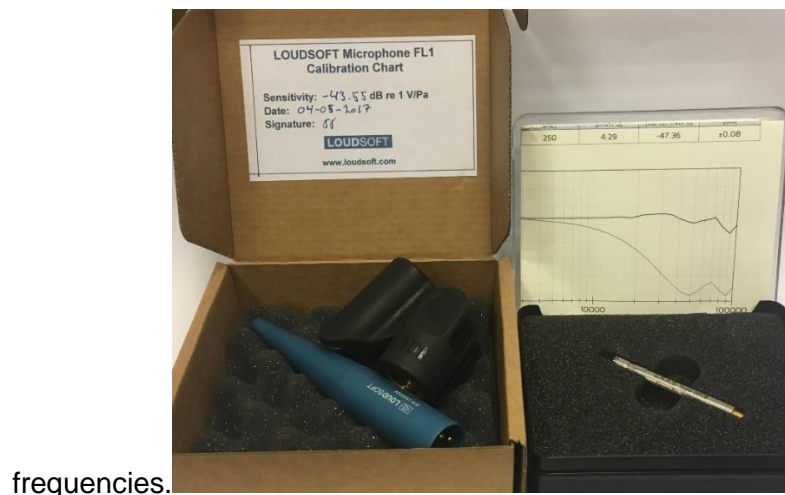


Figure 2 - Microphone sensitivity charts. Left is the LOUDSOFT FL1 microphone and right is the GRAS 46BE microphone.

6. Activation: You must be on the Internet

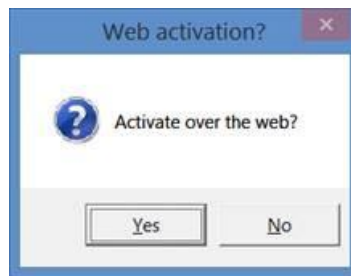


Figure 3 - Activate the software on the internet

7. Insert ID and Password are received from LOUDSOFT



Figure 4 - Insert License ID and Password

8. FINE R+D is ready

You may open the 3 demo files in

C:\Program Files (x86)\Loudsoft\FINERandD\Examples

Use the FINE R+D Viewer for viewing and organizing data without hardware connected.

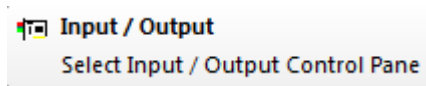



Figure 5 - FINE Hardware 3

2. Frequency response / SPL measurement

This chapter is a tutorial on how to measure a normal loudspeaker with FINE R+D.

Select “New” from the menu or open another worksheet to add measurements.



Make sure the Input / Output pane is open: 

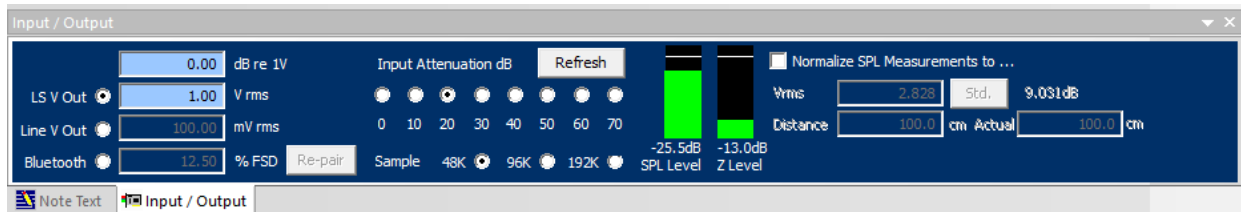
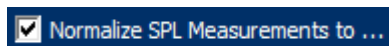


Figure 6 - Input / output pane

Here you define output and input levels. You can specify the output in V_{RMS} or dB (rel. to 1V).

Connect your microphone to Mic1 and select the lowest input attenuator setting which will get you a good measurement without overload. Note the level on the SPL and Z (impedance) level meters. The best level is between -3 to -10 dB ensuring the optimal S/N ratio.

You will get an overload warning when the input level is too high. In that case you need to bring down the input level using the attenuator in 10 dB steps.



You can set the output to any voltage and measure at any distance, and then use the [Normalize SPL Measurements] to normalize the following measurements to the industry standard 2.828V at 1m, or any other voltage and/or distance.

For example, using 50cm (=0.5m) actual microphone distance in most normal rooms is usually better for avoiding reflections than 1m (illustrated in Figure 9). A rule of thumb is to use a distance no less than 2 x the woofer diameter for single drivers.

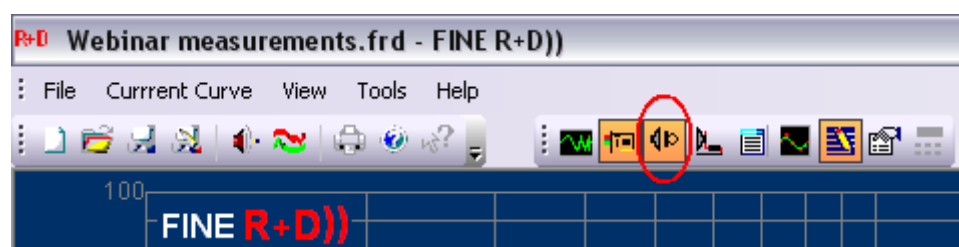


Figure 7 - Open SPL View (FFT Window settings)

For the first measurement (Mic 1 input) you should open the SPL View to set the FFT window to exclude any reflections and maybe add smoothing etc.

Thereafter you may use the **Measure SPL** button (or F3) to measure all following curves with this setting. Each new curve is shown in a new color in the next line at right in the main window. Here you can add comments about each individual curve.

Measure the SPL by clicking the (upper) [Measure] button in the SPL View. The top curve is the Time Response with the large impulse to the left thanks to the Auto Delay.

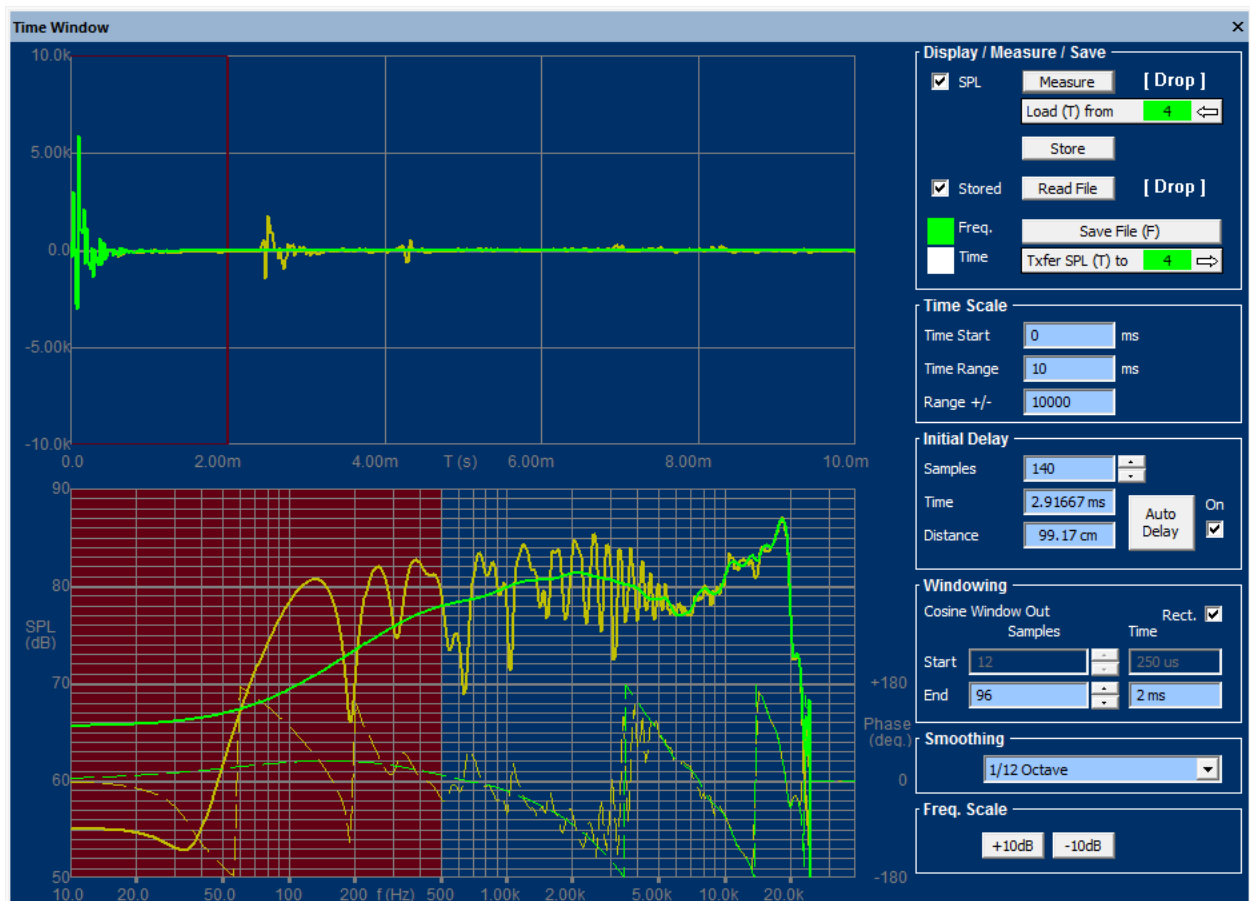


Figure 8 - SPL window - Set FFT time window to exclude reflections for later measurements

Figure 8 shows the impulse response measured in a room at 1m from a typical satellite speaker. The time axis was 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 approx. 1ft/ms).

To obtain the actual speaker response we only need to include the part of the impulse (Time) response before the reflections arrive.

Fortunately, we can do that by applying a window which is set to only pass the part we need, see the upper part of Figure 8 where we have inserted a (rectangular-) window at 2ms (red vertical line).

The green SPL response (lower in Figure 8) is the result of using this short window. The red area at left indicates the invalid part of the curve. So in order to measure lower frequencies we need a longer window. This means that we should push the reflections further away to the right.

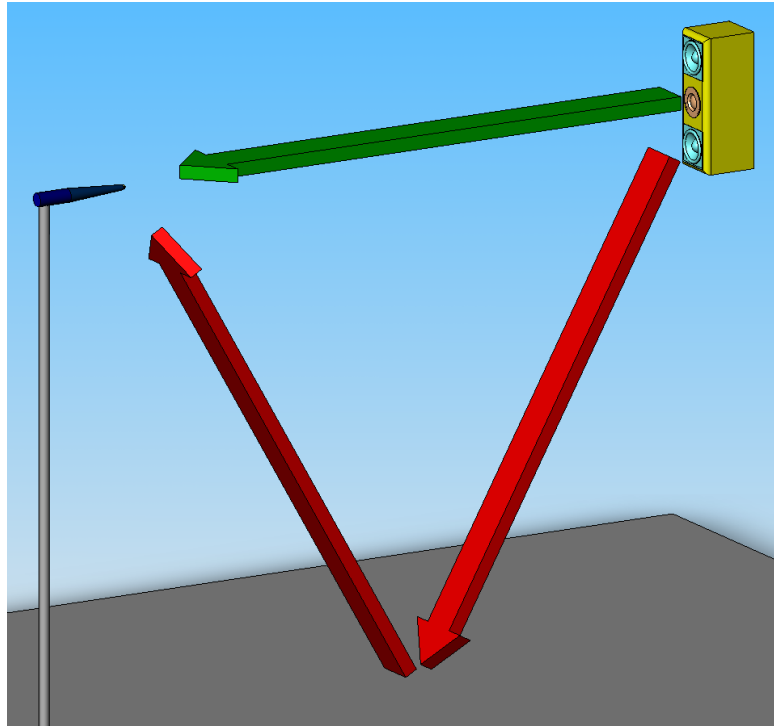


Figure 9 - Direct (green) and reflection (red) from floor

Figure 9 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). For systems the absolute minimum microphone distance is equal to the distance between tweeter and woofer.

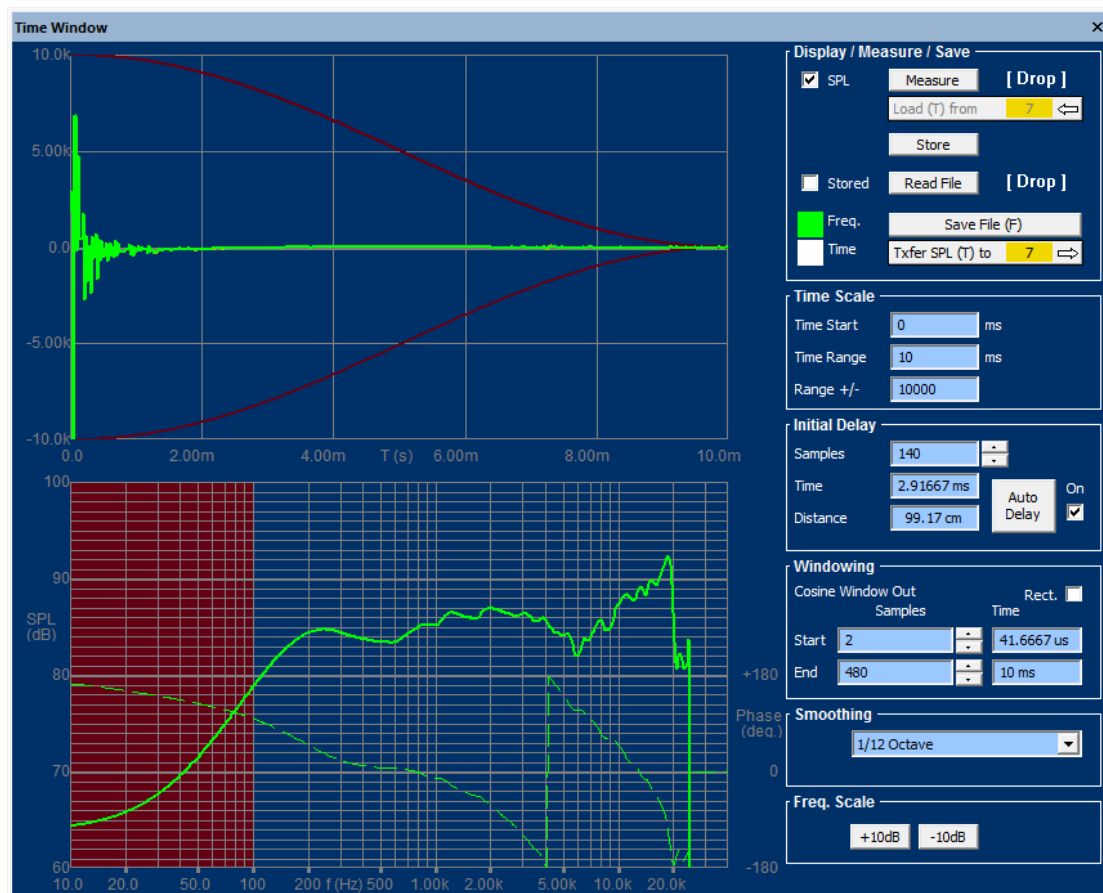


Figure 10 - Satellite measured at 0.5m with reflections happening further away

Figure 10 shows the same speaker measured where both the microphone and speaker were moved up to 154cm from the floor and the microphone distance adjusted to 0.5m. This time we get the reflections much later (just after 7-8ms, shown in yellow) and can use a Cosine window of 10ms, enabling us to measure the real low frequency response of the satellite starting from approximately 150 Hz.

By choosing the Rectangular window you get an exact time/frequency limit whereas the Cosine (Hann) window is much more gradual. The rectangular window may be best for well-defined reflections and the Cosine window for other more spread out reflections. We generally recommend using the Cosine window.

Click **Transfer SPL to [1]** to transfer the response to the next available curve in the main window using the chosen window settings.

Note: Alt + F8 will show the Advanced SPL View. This will also enable the Time display. (Alt + F7 normal view)

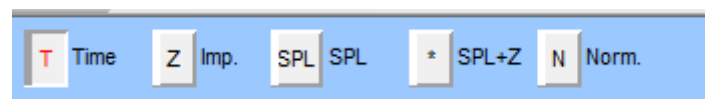


Figure 11 - Time Display shown with Alt + F8

Advanced users: You can specify the start-up screen with your settings. Save your settings as _default.frd in C:\Program Files (x86)\Loudsoft\FINERandD.

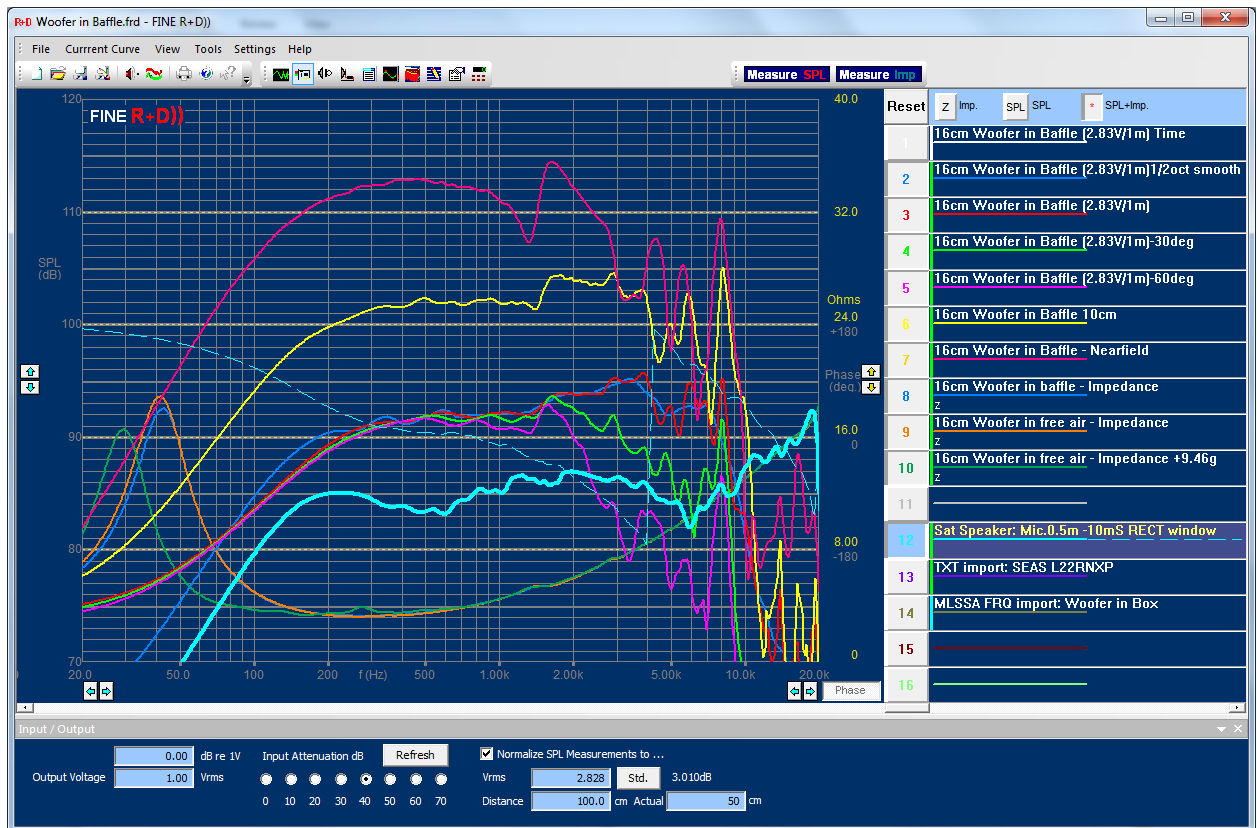


Figure 12 - Satellite speaker measured at 0.5m and normalized to 2.83V/1m (cyan curve #12)

The SPL was transferred as the cyan curve no.12 (Figure 12). Note the dashed line in the SPL window is the acoustical phase response of the SPL/ frequency responses. All FINE R+D measurements are full resolution responses with phase, but here only shown for curve no. 4.

As soon as you have found a suitable window setting in the SPL view, you can continue to measure only by clicking the **Measure SPL** (or F3) button, and you will get a new curve in another color. You will be prompted for inserting a comment for each new curve.

The SPL measurements contain 48 points/oct. and are shown with a white vertical line in the comment field. A green dashed line indicates a frequency response only, which may also be saved, see curve no. 12.

Note: FINE R+D will automatically save responses (blocks) as a Time file (T), indicated by a white vertical stripe. This raw Time format contains much more information and is used for calculation of distortion and waterfall.

You can anytime load a saved time response and view / change the FFT window settings, or apply smoothing.

Display / Measure / Save

☒ SPL

Measure [Drop]

Load (T) from 10

Store

☒ Stored

Read File [Drop]

☒ Freq.

Save File (F)

☐ Time

Txfer SPL (T) to 10

Figure 13 - Load time (T) and transfer SPL in SPL view

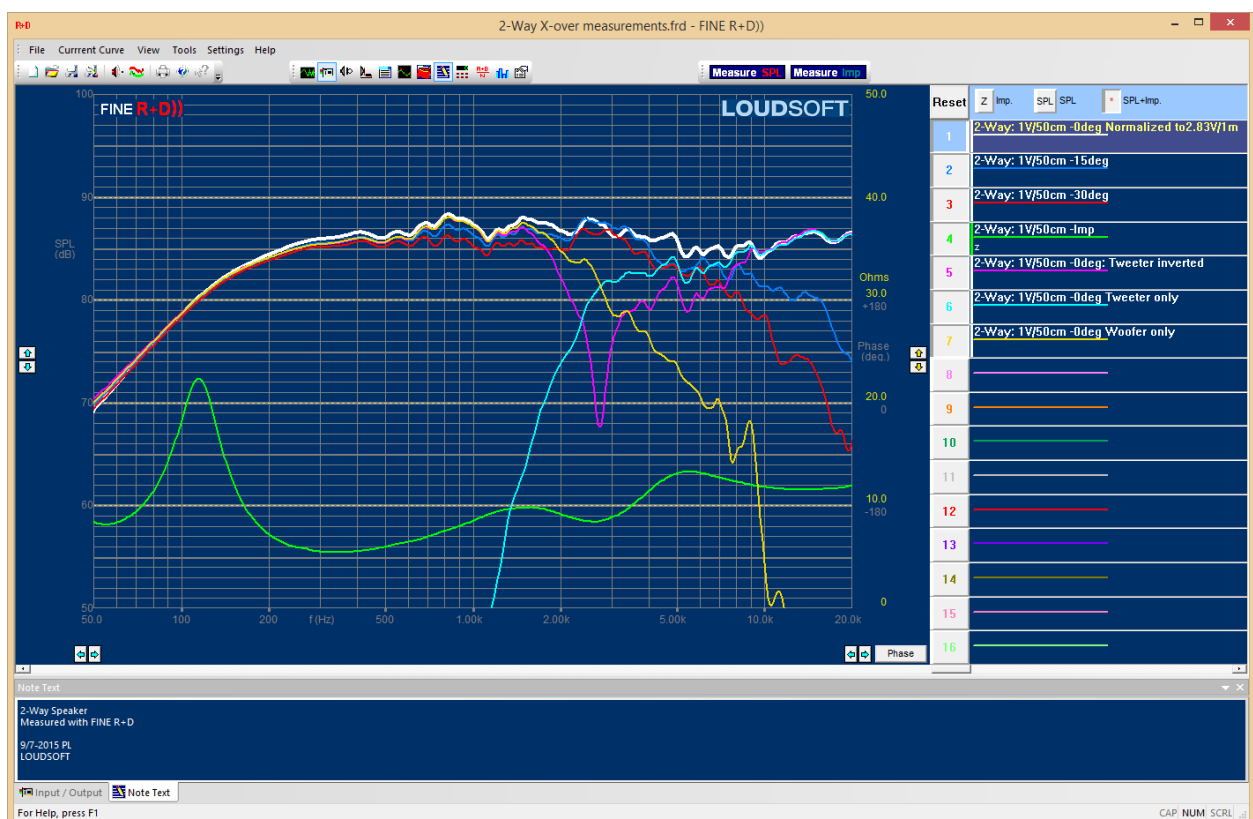


Figure 14 - Typical 2-way system. Note: curve #5 (violet) has the tweeter polarity reversed

3. Curve operations

3.1. Cursor

Figure 15 shows the cursor with frequency and level readout, which is activated simply by clicking the left mouse button. You can move the text up/down with the mouse.

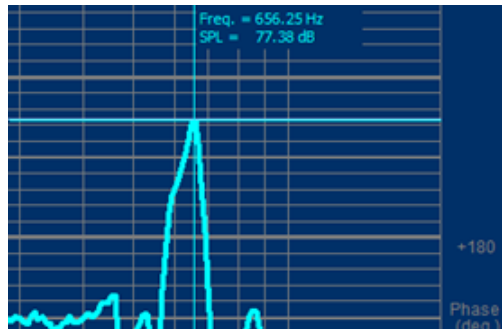


Figure 15 - Cursor readout (hold left mouse button)

3.2. Offset

The right-click menu includes an option for offsetting the chosen curve, see Figure 16. You can input any factor directly or in dB.

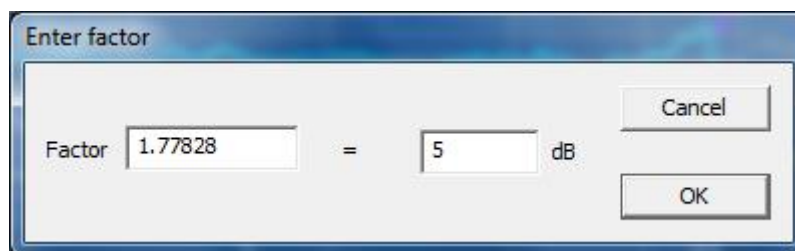
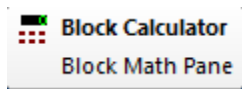


Figure 16 - Offset menu

3.3. Combine A & B



The Block (Curve) calculator will perform practical curve combinations like combining bass reflex port and woofer near-field responses or splicing two curves into one.

Start by loading the A and B curves [From Block] or simply dragging A and B responses from the main curve field onto the [Drop] fields.

Combine A + B by activating that option and input the inner diameter of the port and the effective diameter of the woofer. This will compensate for the difference in pressures thus providing a true amplitude and phase summation of A+B.

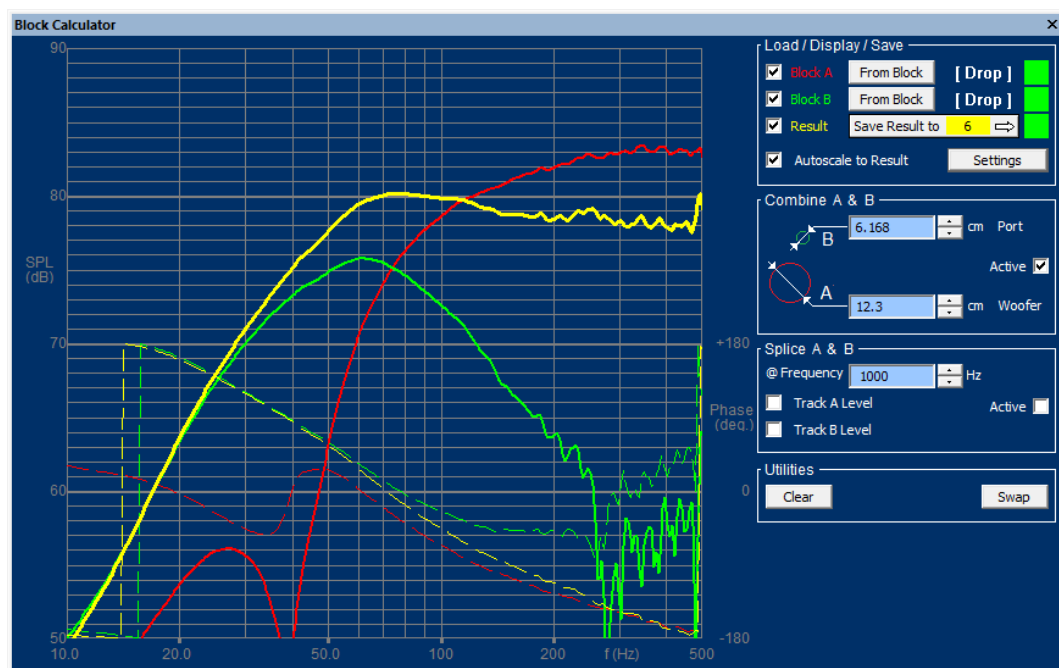


Figure 17 - Combine bass reflex woofer SPL and port SPL

3.4. Splice A & B

Likewise, you can splice two responses A and B together. Here are A and B spliced at 472.6 Hz and the result (yellow) is moved to the level of curve A.

The shown frequency response range can be set in Settings.

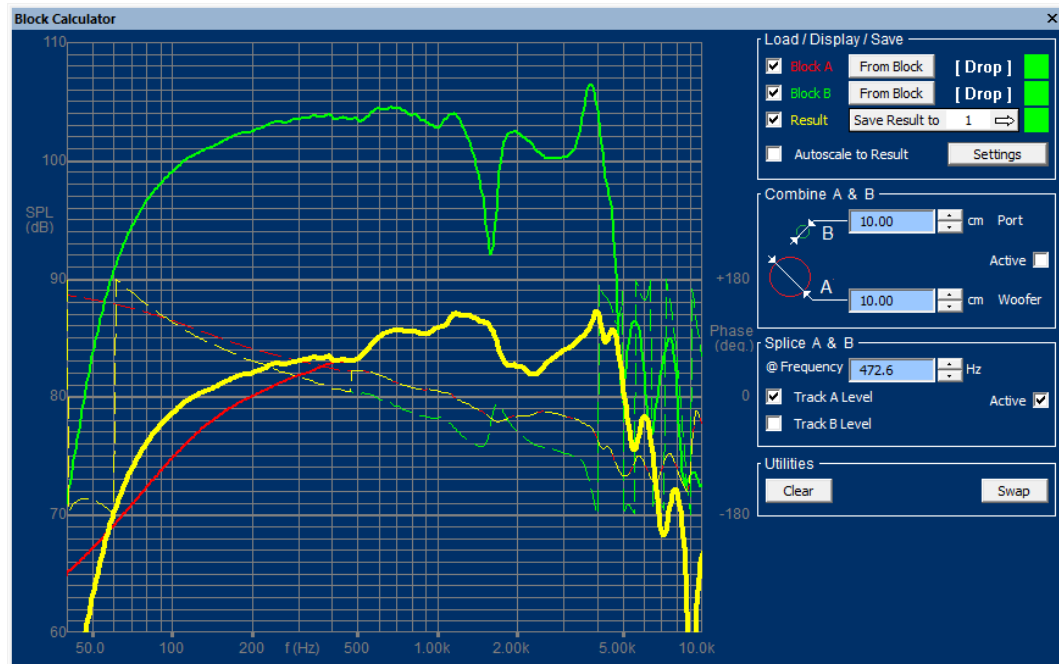


Figure 18 - Far and near-field curves spliced at 472.6 Hz

4. Room response measurements

In case you want to measure how the loudspeaker sounds in your room you should include all the reflections by using the maximum window.

The next example is a measurement of a very good hi-fi loudspeaker measured in a good listening room. The first (brown) curve in Figure 19 was measured in the listening position on axis with the tweeter. Note that the curve was 1/3 octave smoothed by selecting 1/3 octave in the SPL window. (See also RTA in Sec. 7).

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

Instead of the 1/3 octave smoothing you can also select 1/2 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.

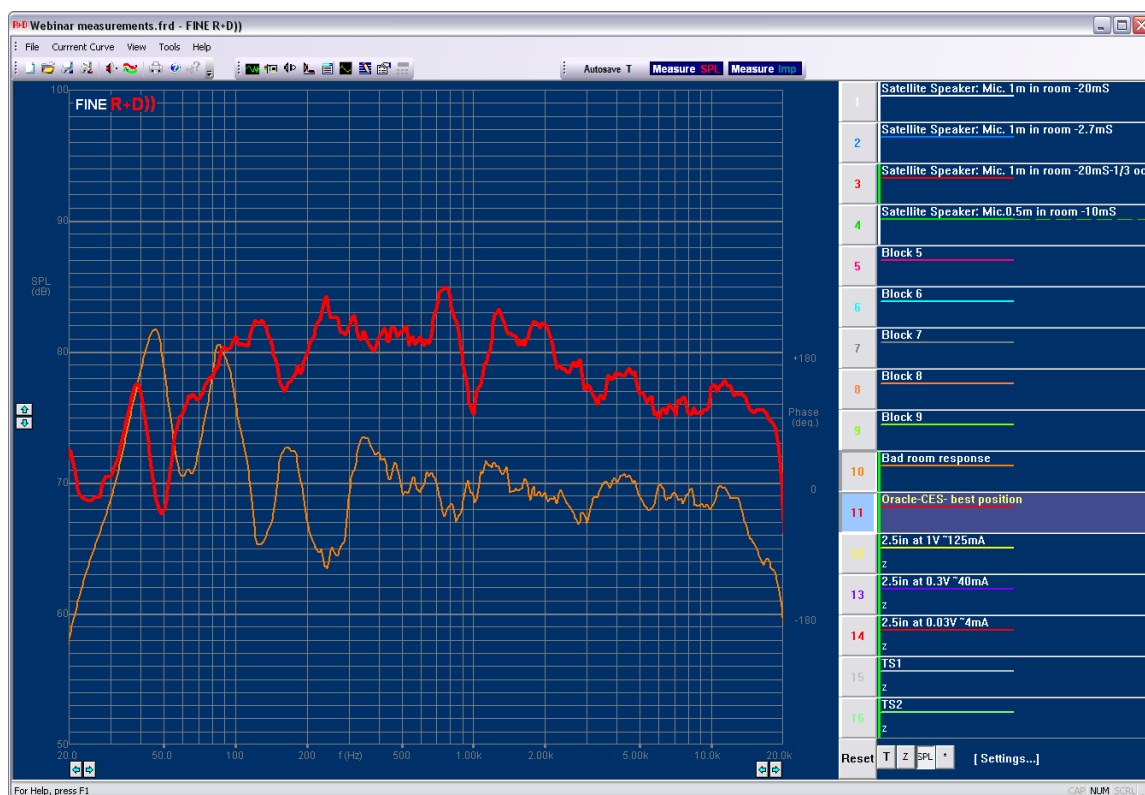
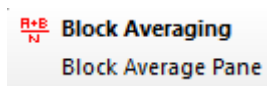


Figure 19 - Room responses measured far from loudspeaker

5. Block (curve) averaging / power response



The block or curve feature may be used to find the RMS average of up to 8 individual responses. Select the curves from blocks or just drag responses from the main screen.

If you for instance have measured a speaker system in increasing angles finding the RMS average will be close to the acoustical power response of the speaker system.



Figure 20 - Example of averaged responses (yellow curve)

6. Normalization / relative curves – microphone measurements

Relative measurements are very useful for measuring microphones and other devices. An artificial mouth or simply a well behaved full range speaker is often used for this purpose.

First you should measure the speaker using a reference (class 1) microphone like the G.R.A.S. ¼-inch precision condenser microphone Type 46BE, which is sold by Loudsoft.

Following you can select Normalizing [N] indicated by the red arrow in Figure 21. You will be asked to set the Normalization as shown in Figure 22. This can also be set in View/Normalization.

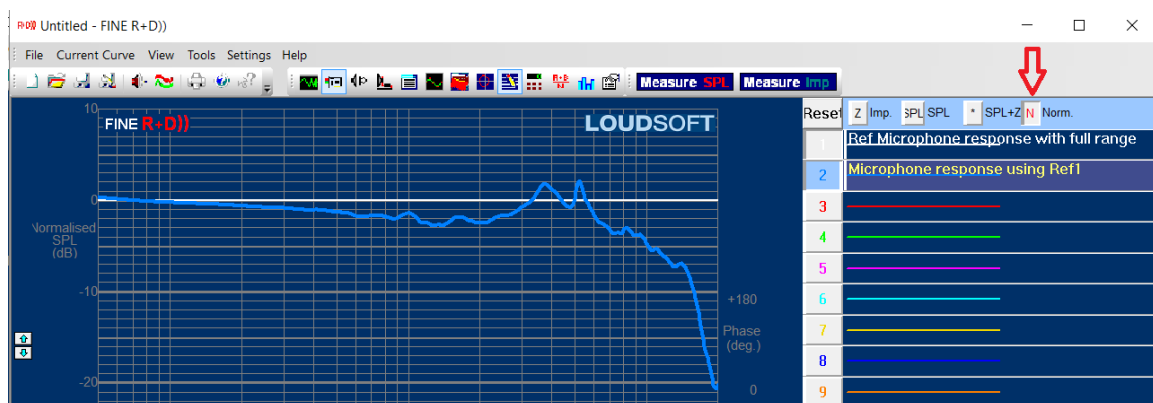


Figure 21 - Measured microphone response using Normalizing Mode indicated by the red arrow

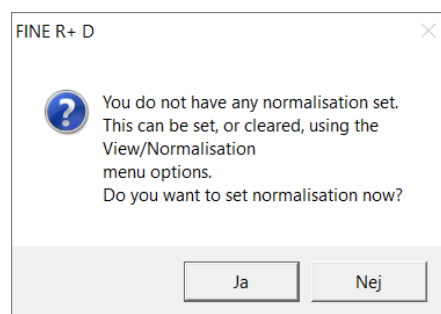


Figure 22 - Message window when selecting Normalization [N]

Then select the reference response, here curve No. 10, from the drop-down menu as shown in Figure 23.

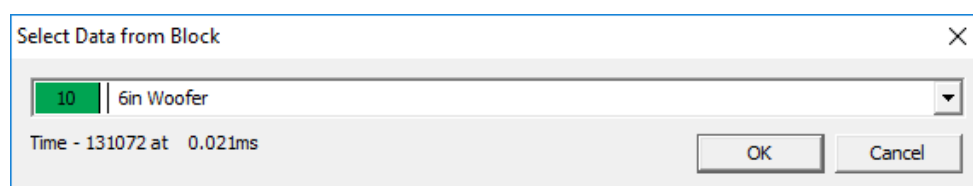
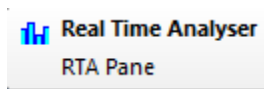


Figure 23 -Select response (block) for use as reference

7. Real time analyser (RTA)



Another useful feature is the Real Time Analyser. This is using the same sweep to show the measured response in columns in ~real time. The default frequency resolution is 1/3 octave as shown above. 1/3 octave is close to the human hearing resolution. However, you can select 1/12-1/6-1/3-1/2-1/1 octave smoothing, see also the 1/12 octave curve in Figure 25.

The RTA is especial useful for tracking changes, i.e. when moving a speaker in a listening room.

Press RUN RTA to let the RTA keep measuring. (The default is 2.5sec, which may be lowered to 1s in Stimulus). The maximum will be kept if "Show Maximum" is checked.

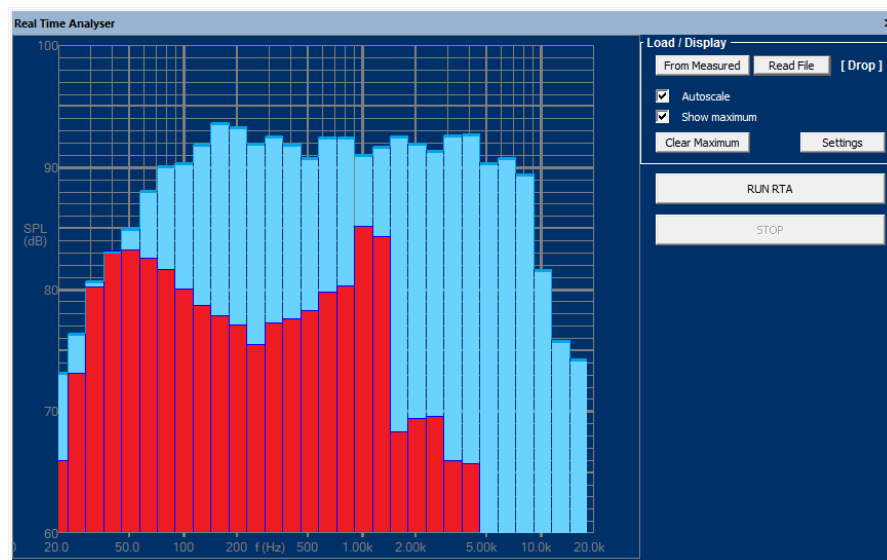


Figure 24 - 1/3 Oct. Real Time Analyser (RTA). Blue is stored maximum

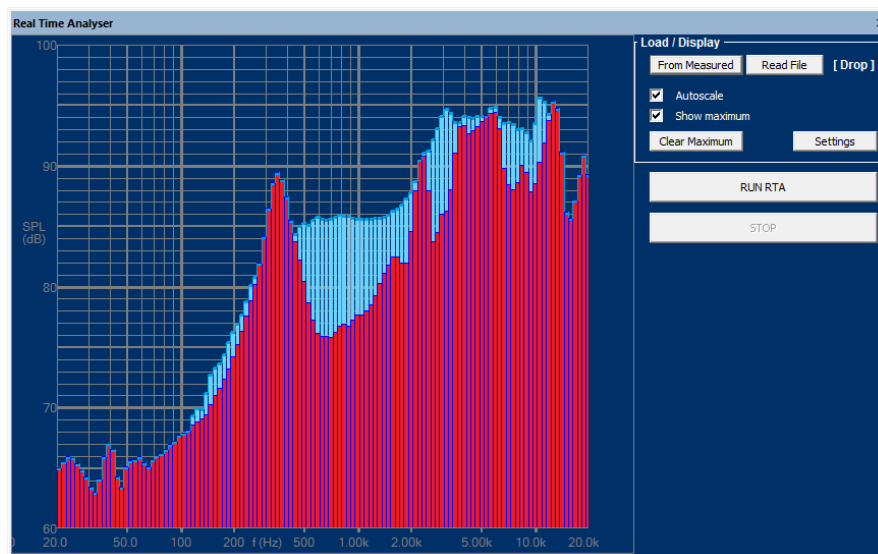


Figure 25 - 1/12 Oct. Real Time Analyser (RTA). Blue is stored maximum

8. Impedance measurements

Measuring loudspeaker impedance is done automatically by clicking the “Measure impedance” button (or F4). The default impedance mode is LO (Low current) which should be used for TS parameters and is accurate up to ~90 kHz. Open Z View (Imp Pane) for HI/LO ranges Figure 26.

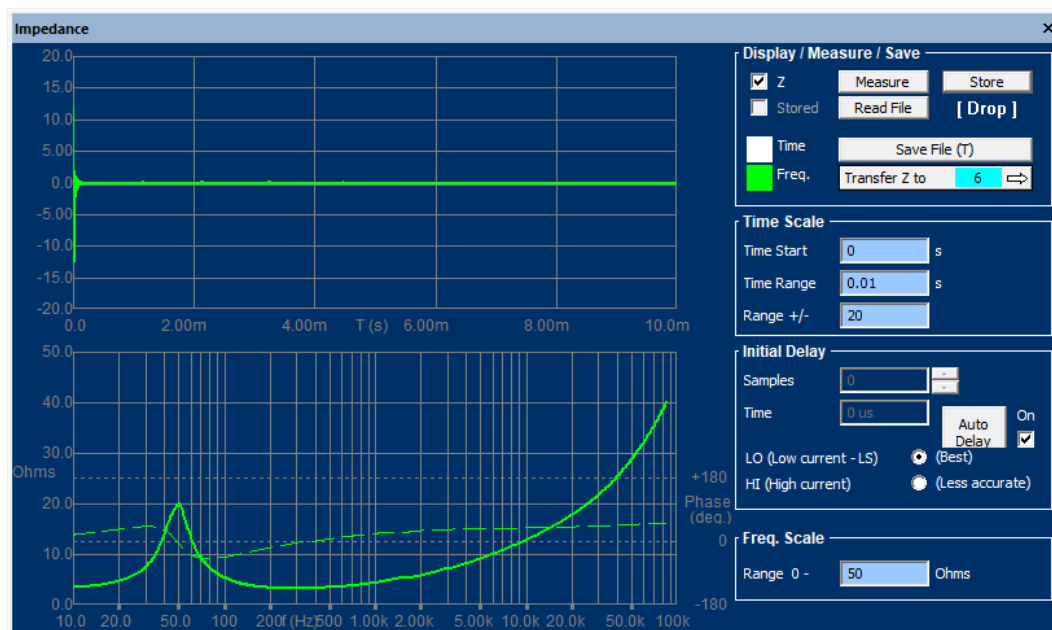


Figure 26 - Impedance Window. Select LO current (Best) or HI current (Less accurate)

You can select the HI (High current) mode which in general is less accurate. It generally gives a lower F_s . The high current measurement is closer to the normal music input at normal levels and is the same as used in FINE QC

Since the impedance measurement is purely electrical the range and time window are pre-defined and set max with the auto delay.

Impedances down to 1 ohms can be measured well if the max power is not exceeded.

The measured impedance may later be used to calculate the T/S 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 Sec. 0.

Figure 28 shows two impedance curves in the lower part of the main screen. The combined (SPL +Imp) display is used here with the *button depressed, see Figure 21.

The impedance curves can be scaled independently of the SPL curves with the up/down arrows on the right side.

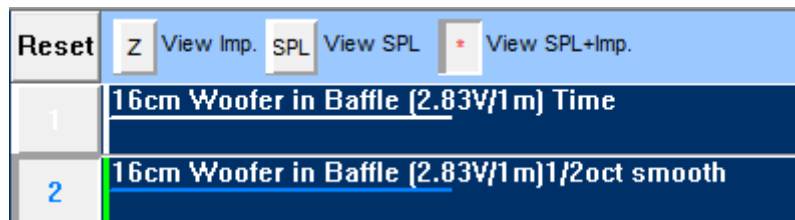


Figure 27 - Combined SPL + Imp (Z) display selected [*]

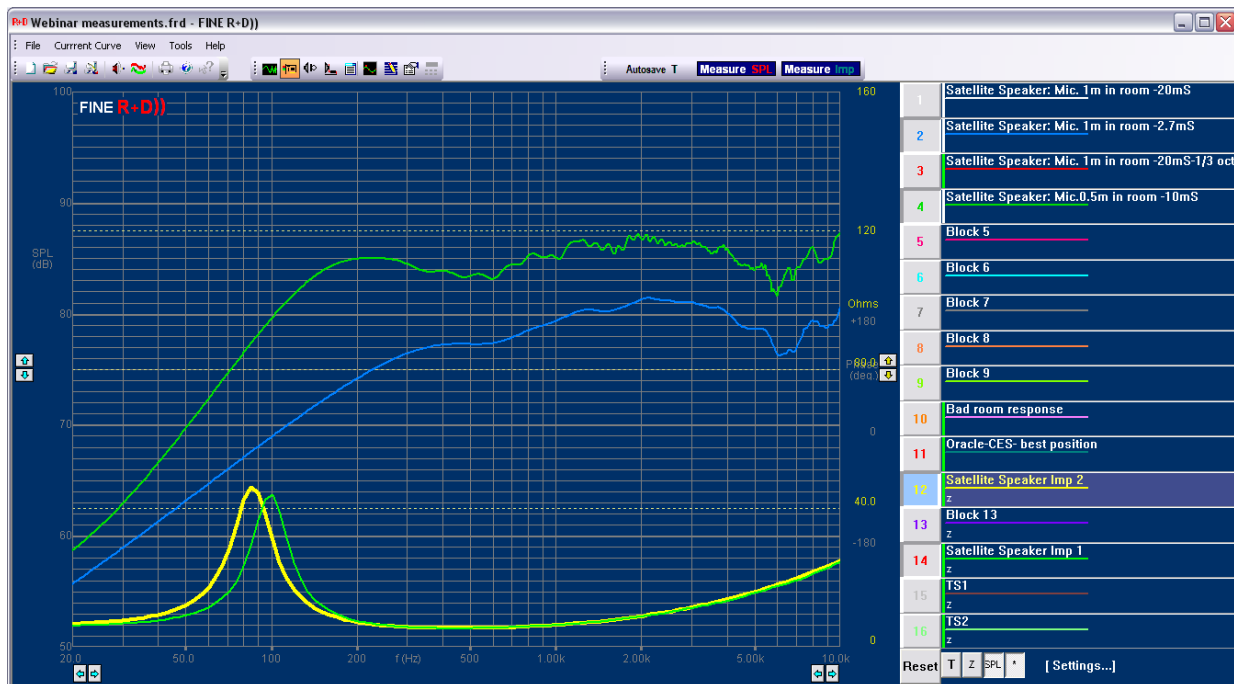


Figure 28 - Main display with both SPL and impedance

9. Thiele-Small parameter measurements

FINE R+D allows for Thiele-Small parameter calculations using the measured impedance of the loudspeaker unit as shown in Figure 29.

First enter the cone area (S_d) or the cone diameter (measured from half the surround across the cone to half the surround on the other side). FINE R+D can estimate R_e (DCR) from the impedance curve, but in order to get the best accuracy measure R_e with a precise voltage multi-meter (DVM) and enter the measured value into FINE R+D. The value can then be fixed by checking Lock R_e .

Now choose a measured impedance curve (LO). The impedance curve should be measured in LO mode for highest accuracy. 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. Then you must know the mass of cone + Voice Coil + half surround + half spider (incl. dust cap and glue etc.) = Fixed Mass M_d . Enter this mass if known and the air load (M_{air}) is calculated, $M_d + M_{air} = M_{ms}$.

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. The most important is a very good fit around F_s , 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, you 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 the fitting.

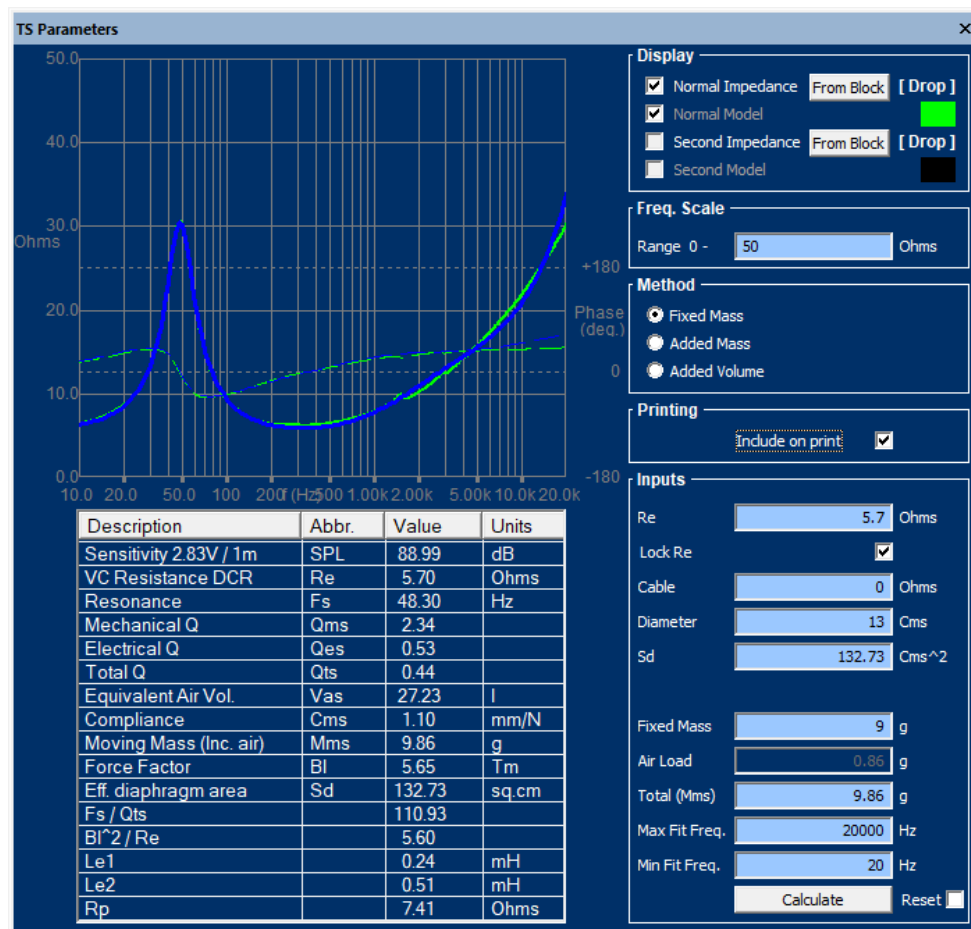


Figure 29 - Thiele-Small parameter window with accurate curve fitting

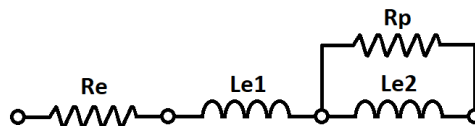


Figure 30 - Impedance Fit model using Le1, Le2 and Rp

The result of the impedance curve-fitting also produces the 3 parameters Le1, le2 and Rp, see Figure 30. This LR2 model simulates the rising impedance of the driver voice coil very well. The same model is used in FINECone. You can set the Max Fit freq. to 20kHz for FINECone use, but the TS parameters actually focus on the impedance peak at resonance Fs, so the Max Fit freq. may be set for example where the min. impedance has doubled for this purpose.

10. Harmonic distortion



To calculate distortion, you need the raw time (T) data. FINE R+D automatically saves all measurements as time data, so you can just select the corresponding block.

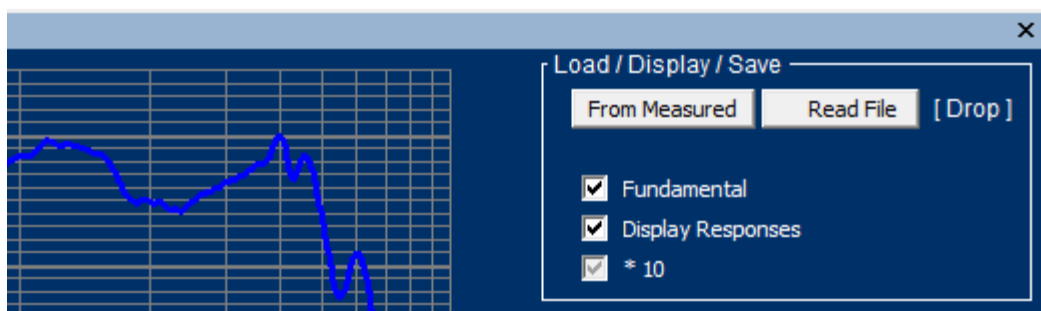



Figure 31 - Select raw (T) time data for distortion calculations

The last measurement (T) is automatically saved so click "From Measured" to use that. You can also open a saved time file "Read File" or drag the Time response (White square in SPL View) to the [drop] field.  Time

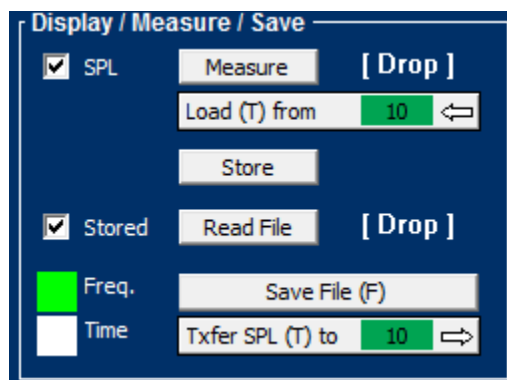


Figure 32 - White square [] Time data in SPL view

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 Figure 33. The red fill at left is invalid data.

(If you have not yet measured that go to the SPL window and make a new measurement, and then select the block current (T) block or drag it to the distortion window.)

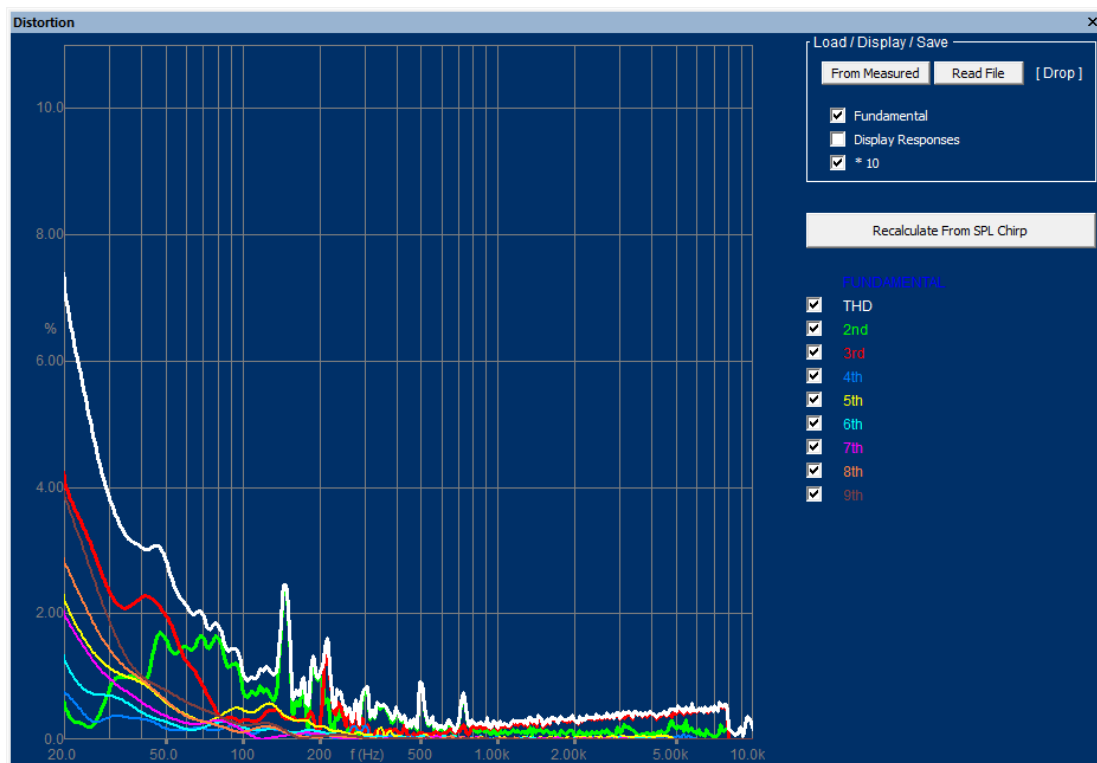


Figure 33 - THD and 2nd-9th harmonic distortion x10 for 2.5in full range

Figure 34 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.

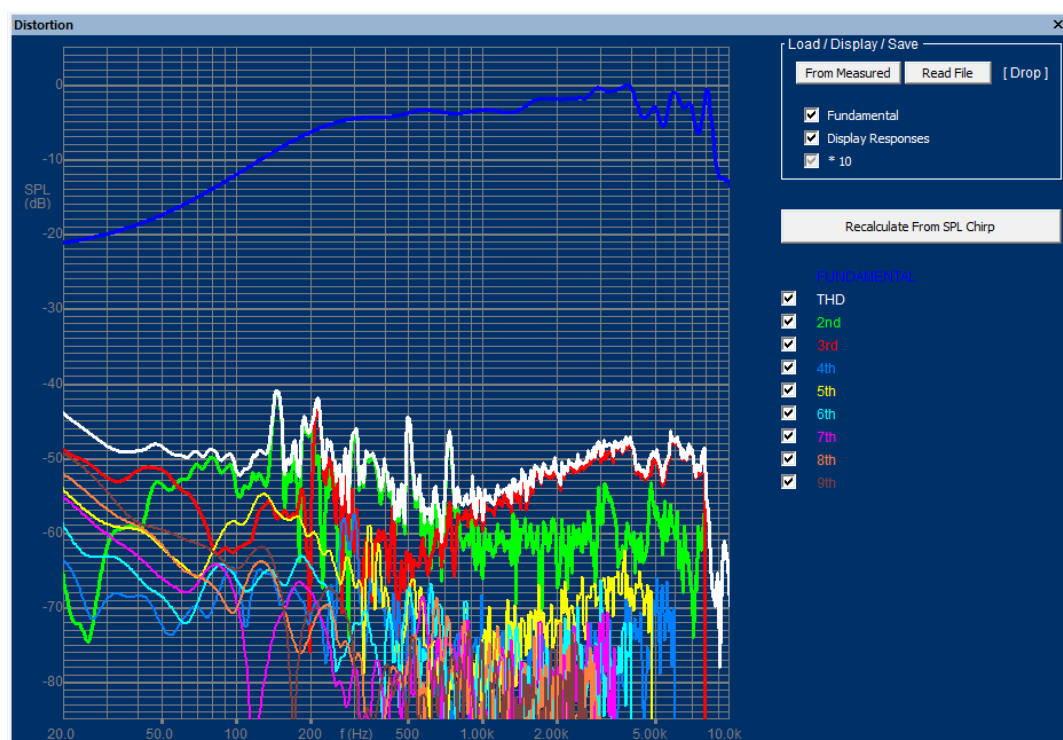


Figure 34 - THD and 2nd-9th harmonics for the same 2.5in full range, but now shown in dB

11. Waterfall (cumulative decay spectrum)



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 Figure 35, 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.

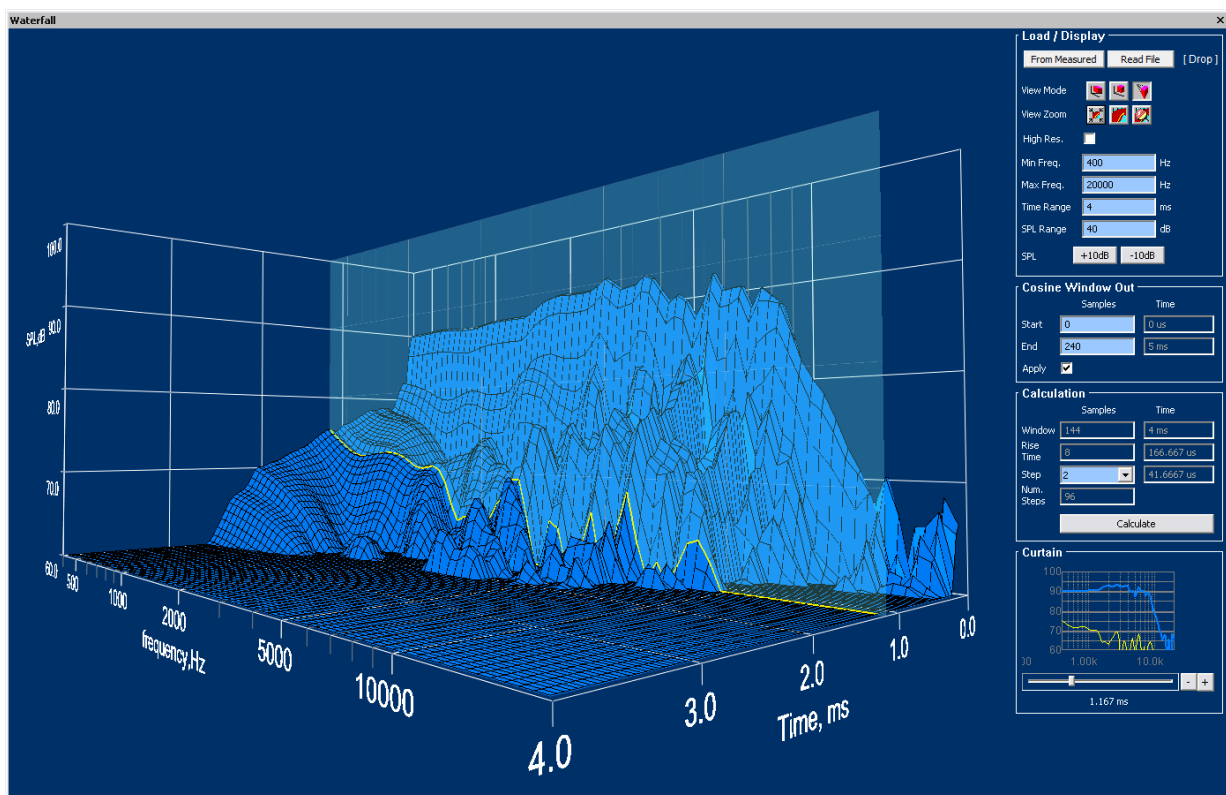


Figure 35 - Example of waterfall plot from a woofer with time curtain set to 1.167 ms

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.

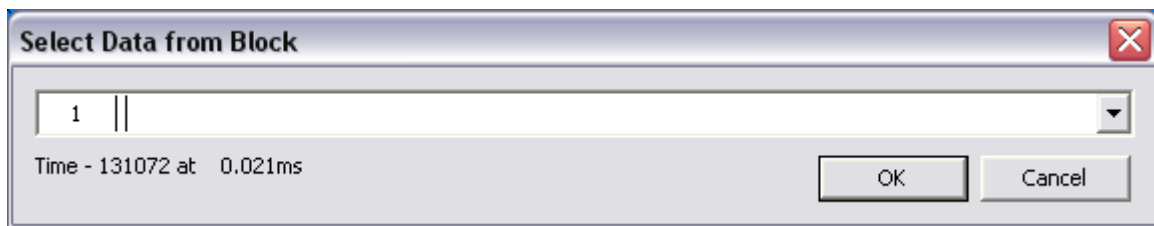


Figure 36 - Select raw Time data from block

Figure 37 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. 2 ms, and the displayed waterfall shows that the response has dropped below the floor after approximately 1 ms. The step size between slices is set to 2, so the automatic settings indicate there will be 48 slices in the waterfall.

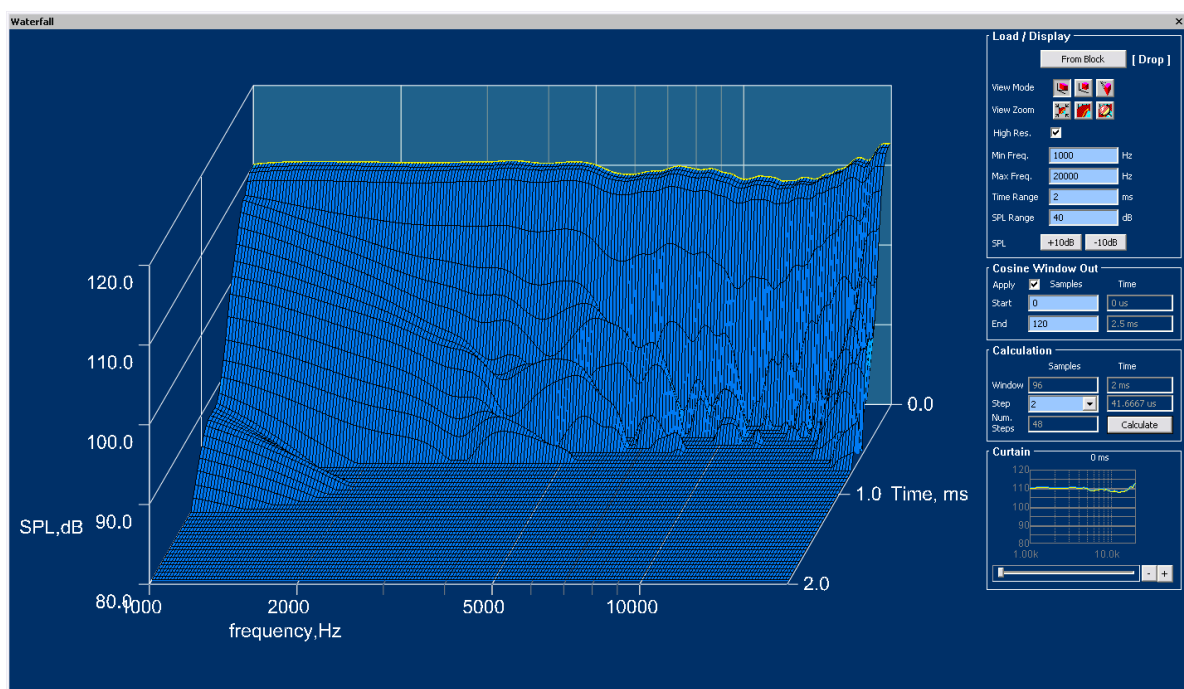


Figure 37 - Waterfall of well-behaved tweeter in high resolution (1/48 Oct.) view mode


Figure 38 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.


Figure 39 shows the setup of the waterfall slices and Time Curtain. The max Time is automatically taken from the Time Range (see Figure 38) 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.

Load / Display

From Block [Drop]

View Mode 

View Zoom 

High Res. ☐

Min Freq. Hz

Max Freq. Hz

Time Range ms

SPL Range dB

SPL

Cosine Window Out

Apply ☒ Samples Time

Start

End

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 38 - Waterfall display settings

Calculation

Samples Time

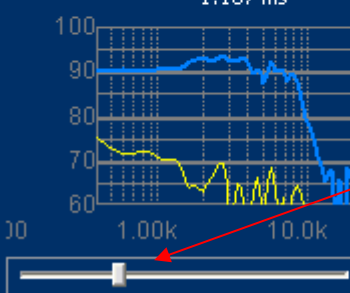
Window

Step

Num. Steps

Curtain

1.167 ms



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 39 - Slices and curtain settings

12. Polar plots



It is sometimes preferable to view the sound dispersion around a speaker as polar curves rather than off-axis responses. This would normally require a turntable and plenty of time for measuring.

Fortunately FINE R+D has a smarter feature. The Polar Plot in Figure 40 and the Contour Plot in Figure 41 are done by dragging many off-axis responses directly into the Polar Plot.

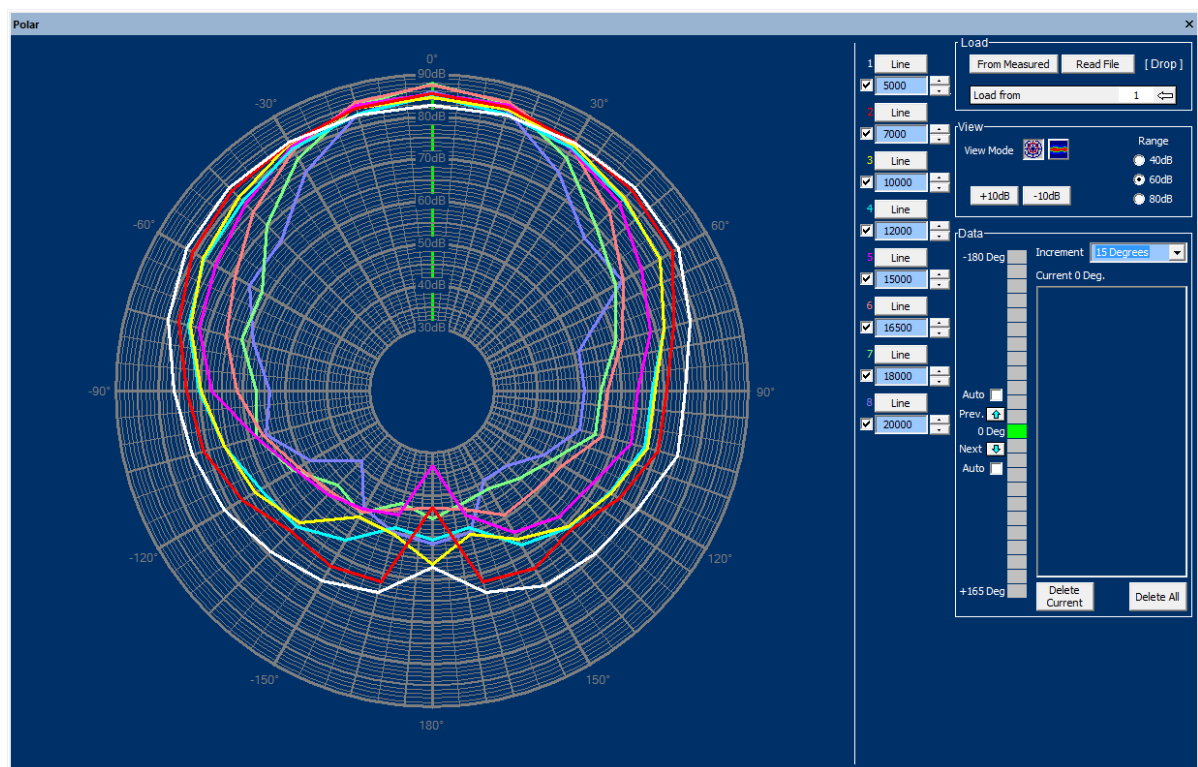


Figure 40 - Polar plot calculated by using off-axis responses

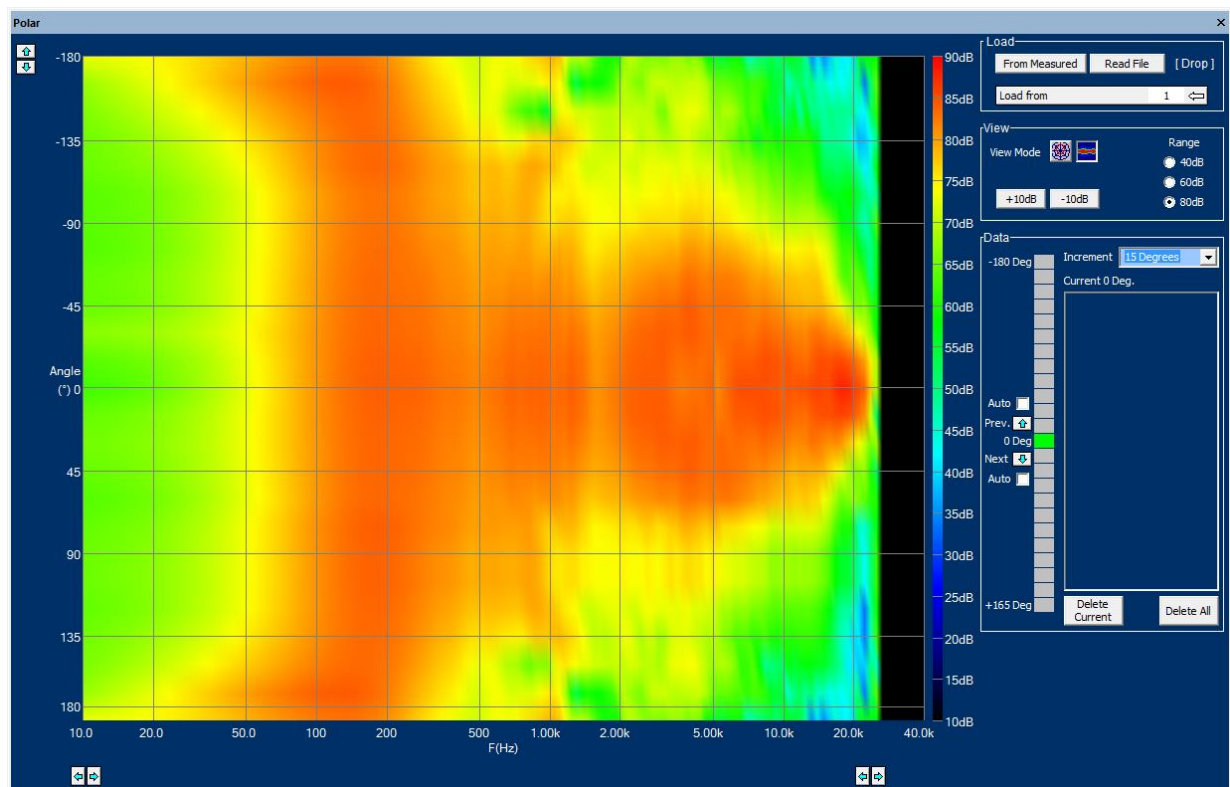


Figure 41 - Contour plot using polar data

12.1. Creating polar plot from SPL curves

Click on 0 deg. (or any other) in the vertical data to indicate the angle you want to start from. Then press Auto [v] up or down. This will show a white arrow indicating the actual angle. Select the increment between curves in the dropdown box (here 15 degrees).

Hold down the left mouse button over the curve you want to drag. After 1-2 seconds a small arrow + will indicate that the curve is ready. Then drop that in the [Drop] field in the polar screen shown in Figure 42. Or you can load from a block by selecting each.

Then drag all the curves in the same way and the display will automatically move and show the polar plots. Select up to 8 frequencies to be displayed as the lines 1-8. You can set the frequencies and line colours individually for all 8. Choose the appropriate range 40 – 60- 80 dB. You may also move the curves up and down by + / - 10 dB buttons.

Select Polar or Contour display in View Mode.

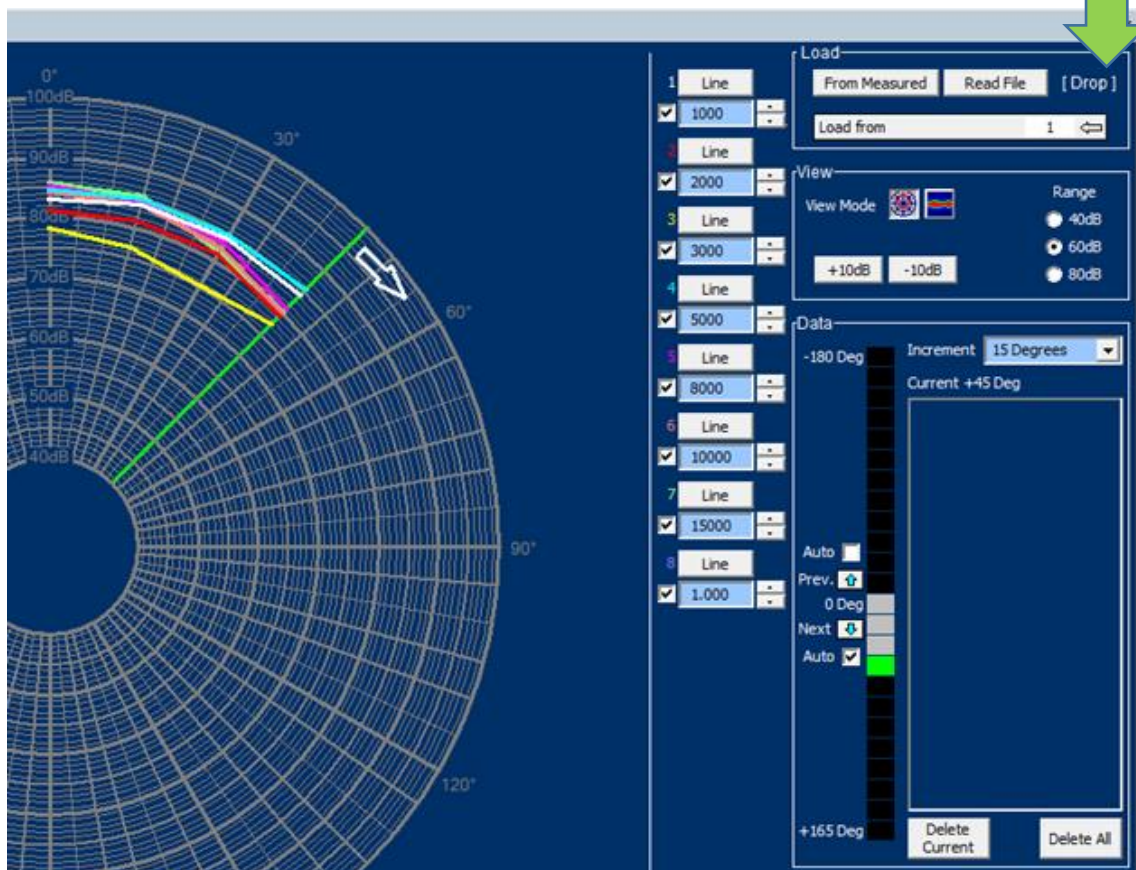
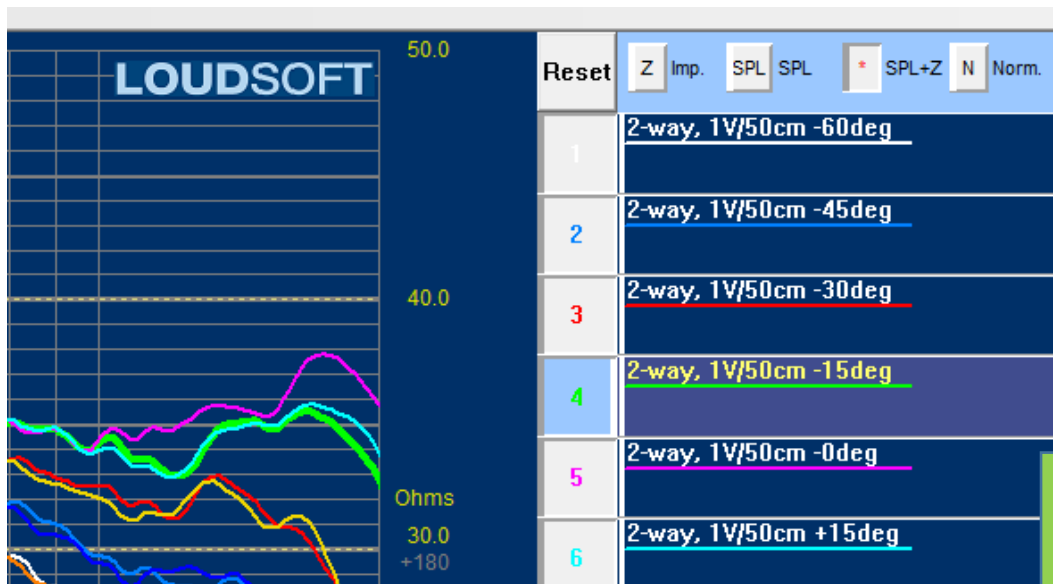


Figure 42 – Polar input: Drag curves directly from Blocks in steps

13. Bluetooth testing

Wireless Bluetooth Speakers are difficult to test at the Production Line because of the latency problem.

First you need to add the Bluetooth device to your PC and then connect it, see Figure 43.

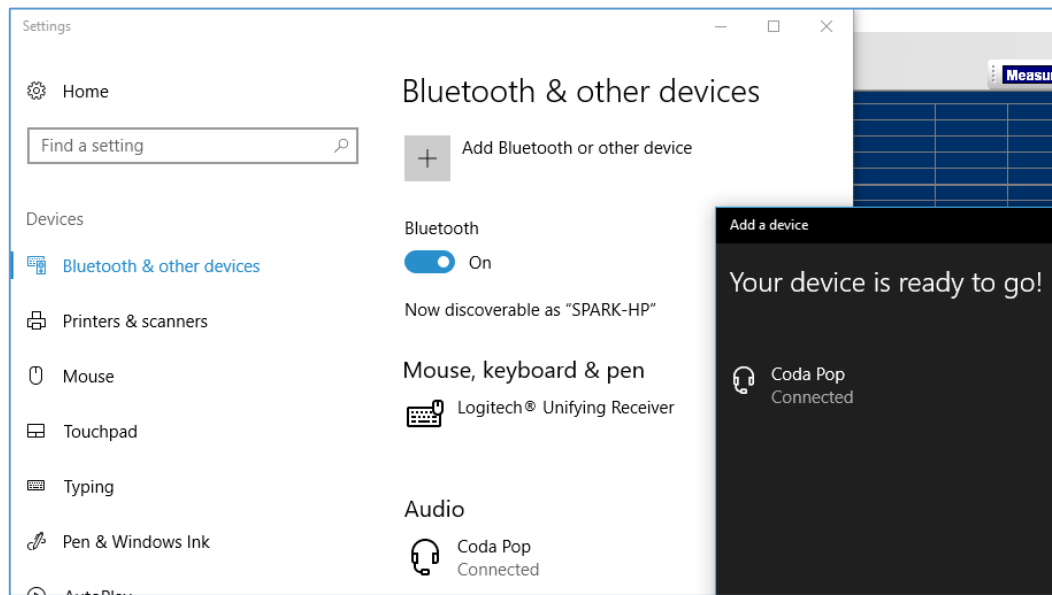


Figure 43 - Add Bluetooth Device to your PC and connect

Select Bluetooth (BT) Mode in the input/output pane shown in Figure 6. Now you are asked to pair your BT device as shown in Figure 44.

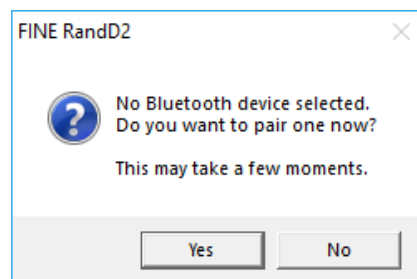


Figure 44 - Select BT pairing

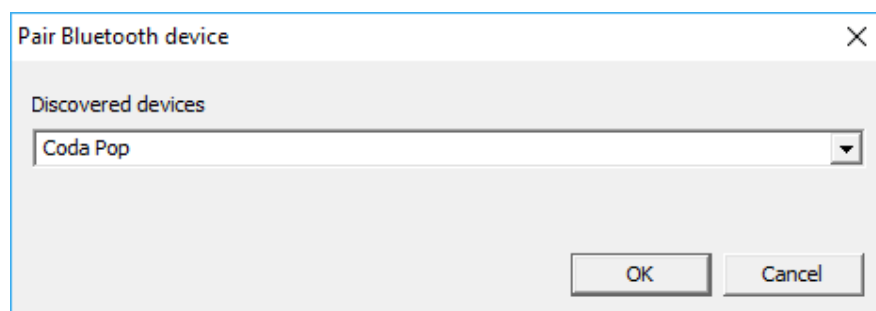


Figure 45 - Bluetooth device found

Next you must select an output device. This can be very tricky as Windows tries to “help” which is not always the best. In general, you should select a headphone rather than a stereo device, see Figure 46.

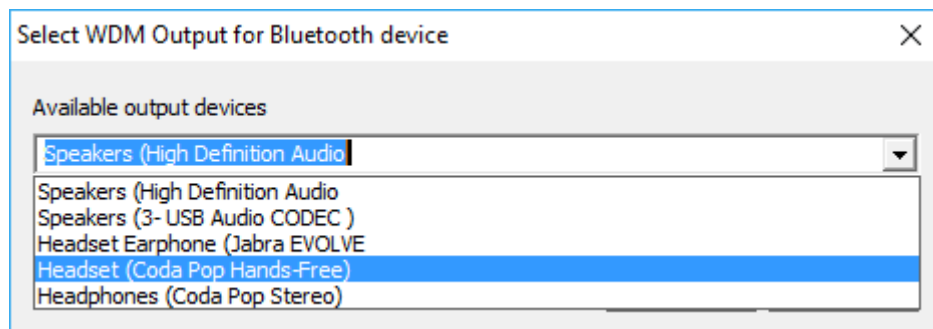


Figure 46 - Select output device. Be careful selecting the best output device

13.1. Alternative Bluetooth Device

Using a separate Bluetooth transmitter you may avoid many of the problems with the windows device/drivers.

Figure 47 shows a simple Bluetooth module. The Bluetooth Transmitter should be connected to the headphone output (or Line Out XLR) which is in parallel with the Line Out.

You must first select pairing and wait for that to happen. Thereafter you can test as normal. The accuracy of the Bluetooth test is close to the normal high accuracy of FINE R+D.



Figure 47 - Bluetooth transmitter connected to Headphone output, which is in parallel with the Line Out.

14. Stimulus



Here you can set the used sweep range. The main range 10 to Max Freq. for sample rate is default and can be used for everything. Using 192 kHz sample rate the range is from 10-88kHz.

The 200 Hz to Max Freq. for sample rate is mainly for tweeters, so the measurement is focused on the High frequency range, and possibly avoiding overloading tweeters at very low frequencies, where these cannot accept much power.

Use the (default) 10Hz to Max Freq for subwoofers and woofers in general.

The Stimulus window is shown in Figure 48.

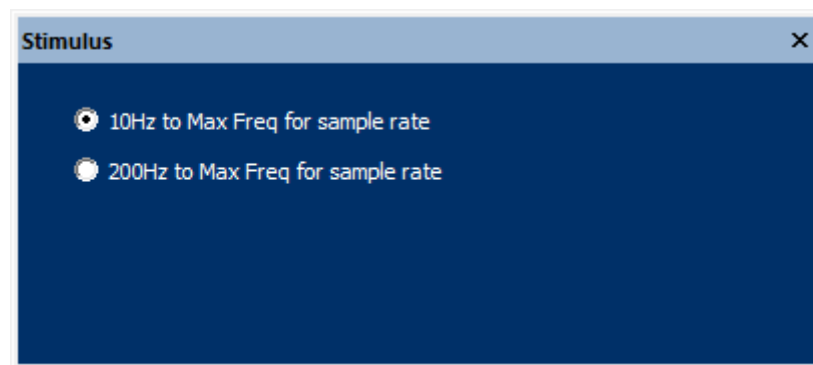


Figure 48 - Stimulus settings

15. Import/export of SPL and impedance to FINE X-over and other software

FINE R+D is a unique tool for measuring loudspeaker responses that can be exported in the standard LOUDSOFT format *.lab or text format, for example to the acclaimed loudspeaker cross-over 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 as shown in Figure 49.

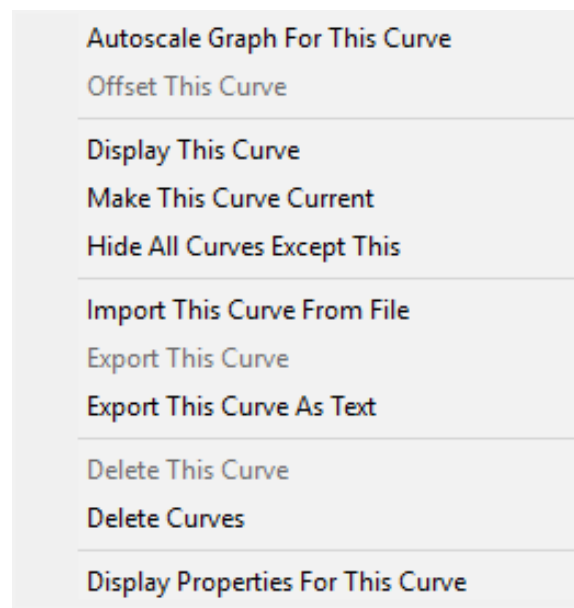


Figure 49 - SPL and impedance export from FINE R+D

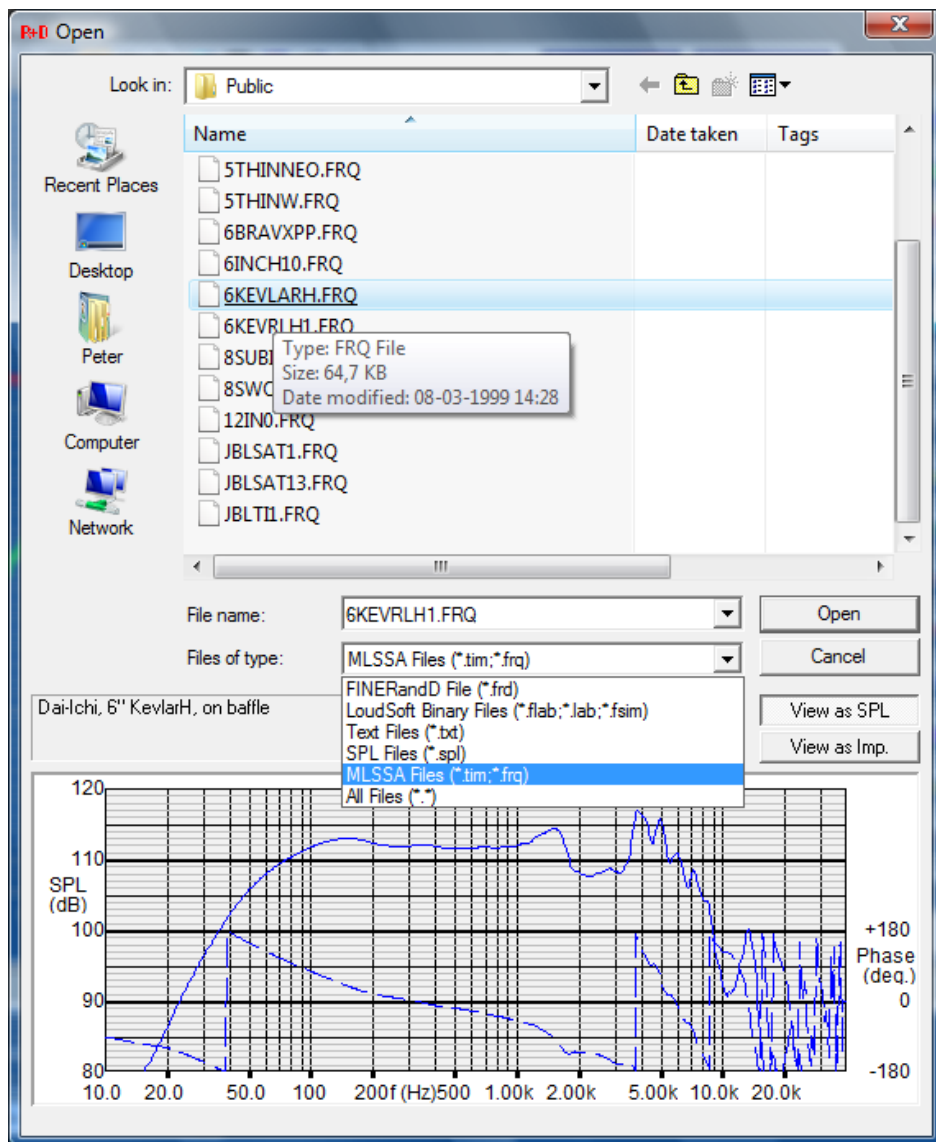
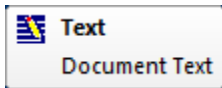


Figure 50 – Frequency response preview of FRQ, .TXT, .LAB and .FSIM as well as for other formats

16. Text (global)



Add general text here. This could contain basic information about the general measuring conditions etc. The text will appear on the main print.

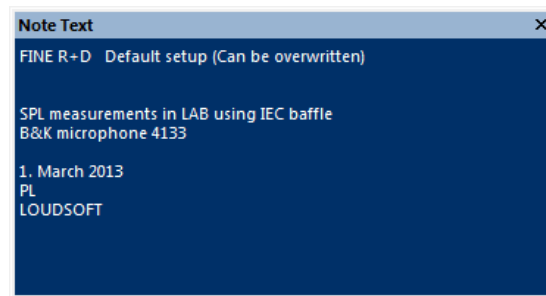


Figure 51 - Add text about the measuring setup, the loudspeaker unit or other important information

17. Measuring X-over circuits

17.1. Acoustical

You can measure both the total response as well as the individual driver responses and impedances. These are the most important and can then be exported to FINE X-over or FINE DSP for the final system design.

17.2. Electrical

You can directly measure the signal across the drivers or any component in the cross-over circuit. Be sure to connect the drivers for correct responses. Make a connection to the Microphone1 input using terminal 2 (Hot) and terminal3 (cold) only! (Figure 52). DO NOT CONNECT TO GROUND!

You can also connect resistors for QC check. (Best use FINE QC)



Figure 52 - Measure directly on crossover circuits

18. Measuring Amplifiers and Electronics

Performing electronic measurements of amplifiers and other electronics is quite easy with the advanced FINE Hardware3. Just connect the amplifier input to the balanced Line Out XLR. Then make a connection from the amplifier output to the Microphone1 input using **only** terminal 2 (Hot) and terminal 3 (cold), see the diagram in Figure 53. **DO NOT CONNECT TO GROUND!**

BE SURE TO KEEP THE OUTPUT LEVEL LOW in order not to overload the amplifier. Remember to set the output as Line V out, see the blue arrow in Figure 55.

Fortunately the line out signal is 1/10 (-20dB) of the main output from the FINE Hardware 3 (2).

You can also measure unbalanced using the Hot (2) for signal and the Cold (3) for return (ground) terminals in the XLR connector.

Note that the microphone inputs accept up to 12V_{peak}!

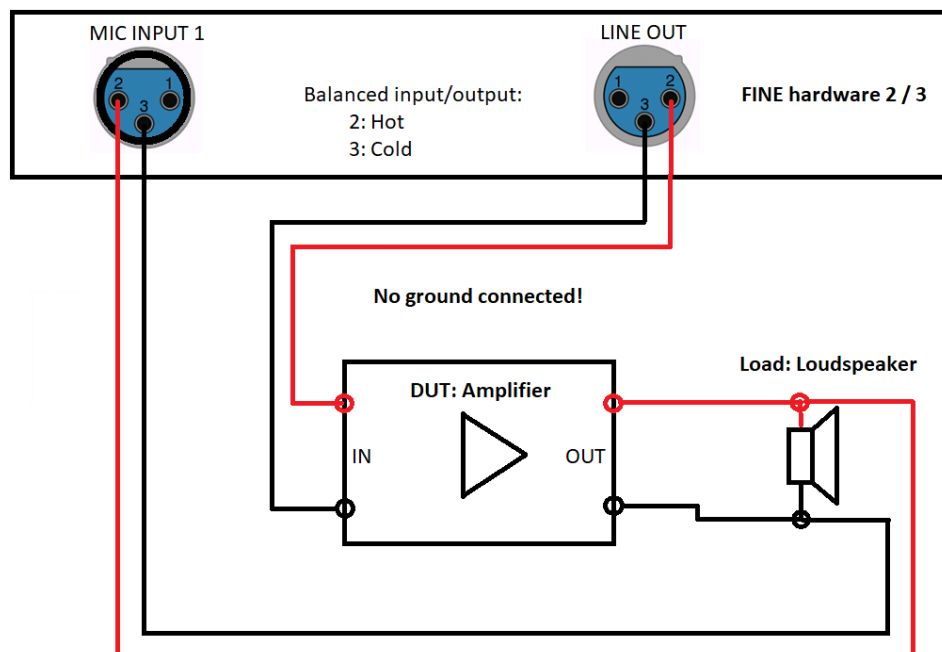


Figure 53 - Electronics Testing: Amplifier connection diagram

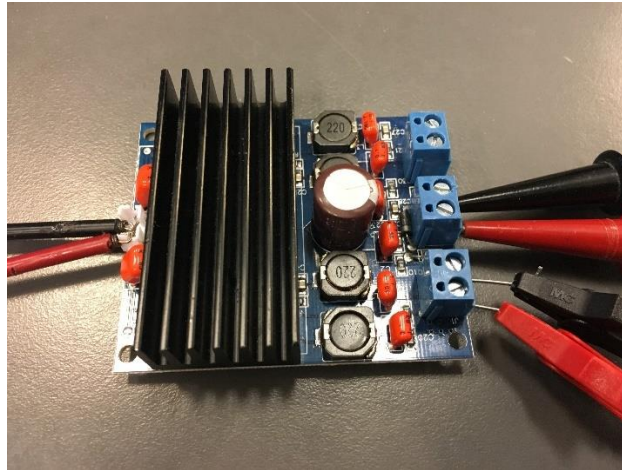


Figure 54 - Small (Class D) amplifier measured with FINE R+D

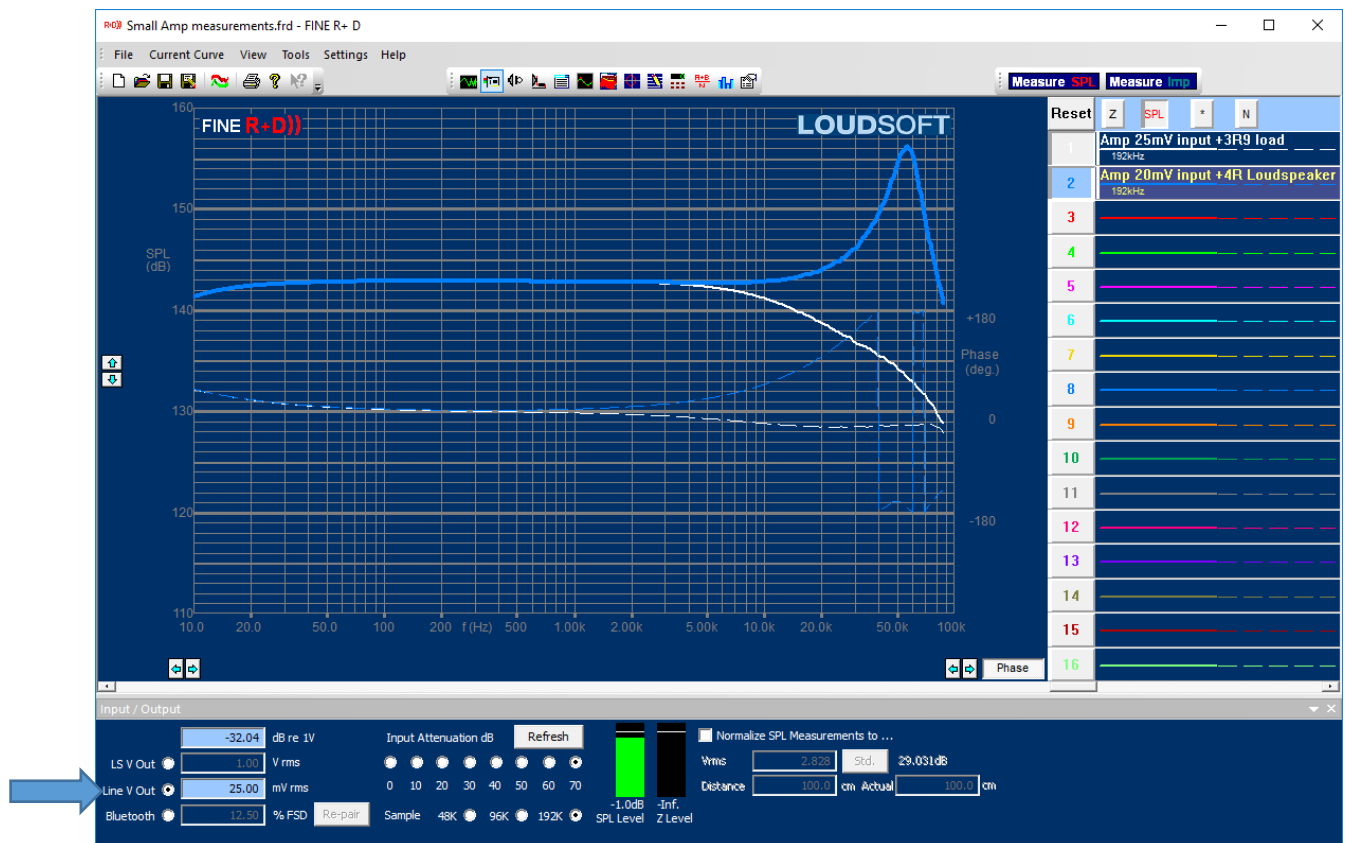


Figure 55 - Small amplifier response with 4R load (White) and Loudspeaker (Blue), 10 Hz-90 kHz

Figure 55 shows the testing of the small class D amplifier in Figure 54. The first #1 curve is tested at 192 kHz sampling with a 4R resistor as load. However, curve #2 (blue) is measured with a loudspeaker connected! So the amplifier response is very dependent of the actual load.

18. FINEDSP

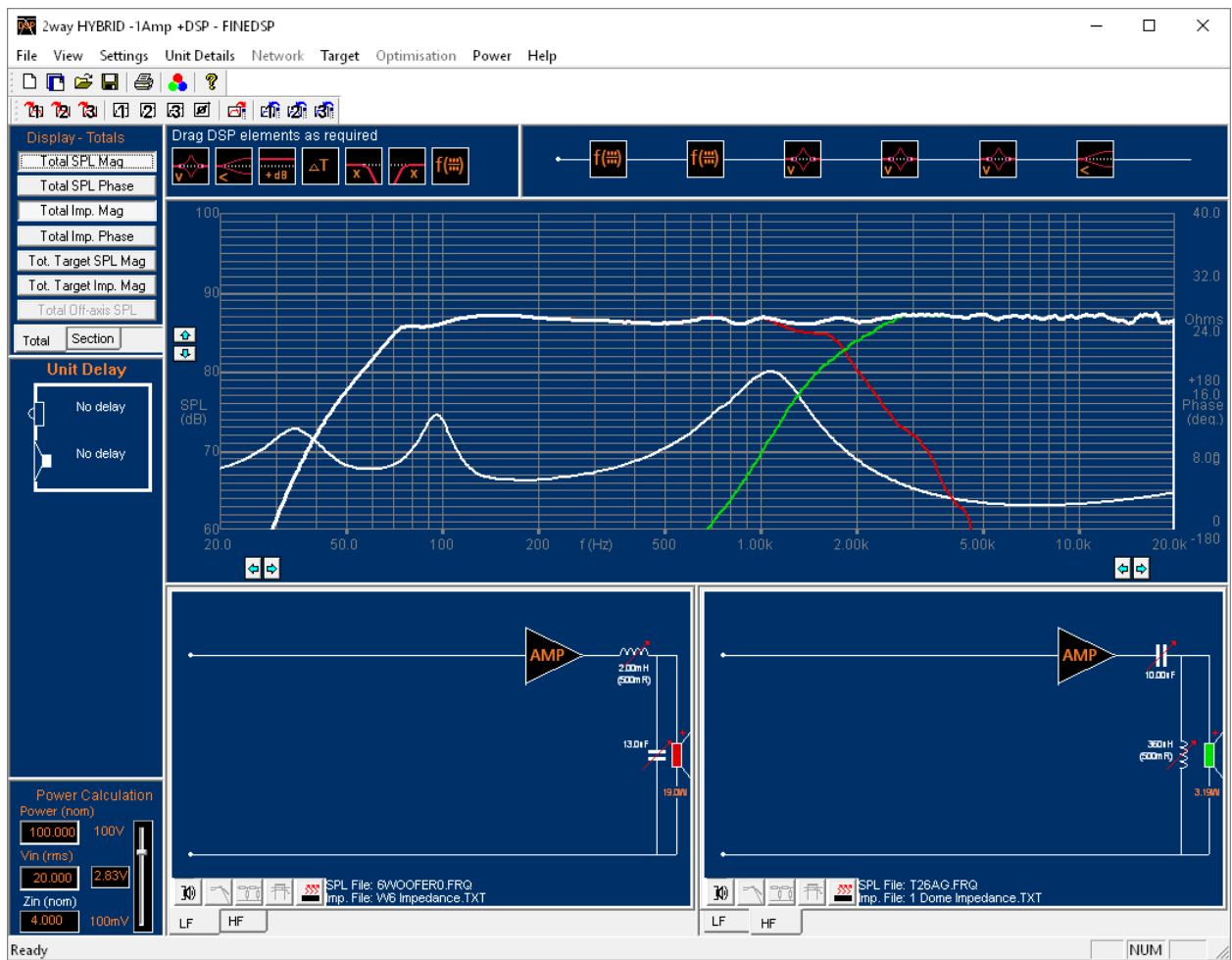


Figure 56 – Hybrid cross-over optimised in FINE DSP with SPL and impedance imported from FINE R+D

The example in Figure 56 shows a 2-way hybrid cross-over designed in FINE DSP. The hybrid has both some passive components + DSP. 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 impedance).

19. G.R.A.S microphone 40BE

LOUDSOFT can now supply the world-renowned G.R.A.S. microphones. The free field $\frac{1}{4}$ -inch microphone set 46BE / 26CB is supplied in a special configuration FL4 which directly connects to FINE R+D (and FINE QC) without using a separate microphone pre-amplifier.

The G.R.A.S. Microphone Type 46BE (40BE) is a $\frac{1}{4}$ -inch class 1 precision condenser microphone for general purpose measurements in open acoustic fields. It is a pre-polarized free-field microphone with a large dynamic range and a wide frequency response.

As a free-field microphone the Type 46BE is for measuring the sound pressure which existed before it was placed in the sound field pointing towards the sound source.

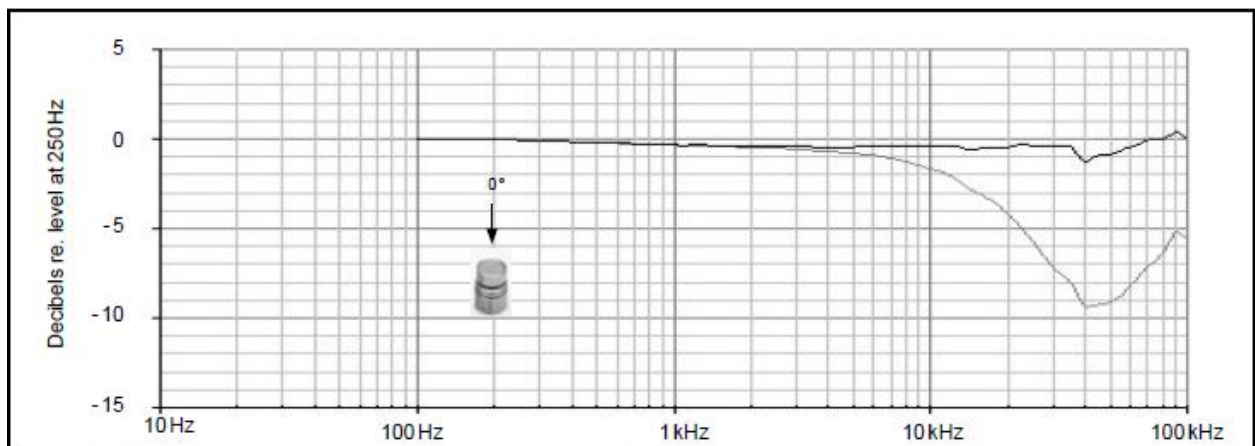


Fig. 2 Typical frequency response of Type 40BE (without protection grid). Upper curve shows free-field response for 0°, lower curve shows pressure response

20. Printing and reports (PDF)

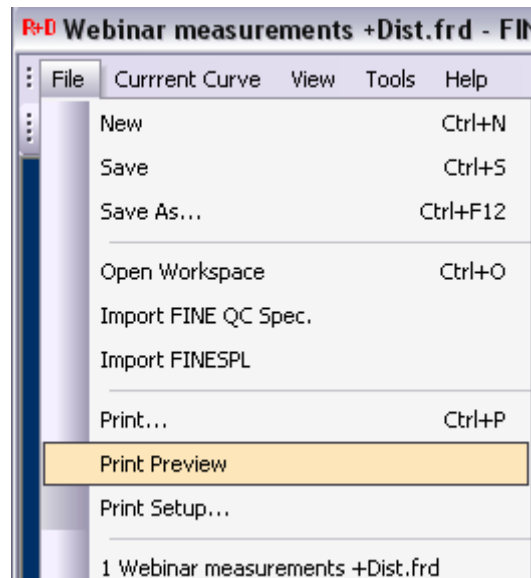


Figure 57 - Print preview

The print command will print the main page (Default) including the notes in the general text. Also, TS parameters will be included if specified as shown in Figure 58.

You can print up to 8 pages including Main Overview (= old style), SPL and Imp windows, TS Parameters, Distortion, Waterfall and Block Calculations. To activate this Print Mode, go to Properties and set "Print as Summary" to "False". ("True" is default printing only the main page).

Setting "Print uses screen colours" as True will print 8 pages with graphs in print colours. Alternatively press the "Print Screen" button to copy the actual screen to the clipboard, which may be inserted into a document and printed.

PDF Reports can be made by selecting the Microsoft Print to PDF or downloading for example the free PDF writer 995 <http://www.pdf995.com/>. Then you can print 1-8 pages to this and thereby create a fine PDF-report.

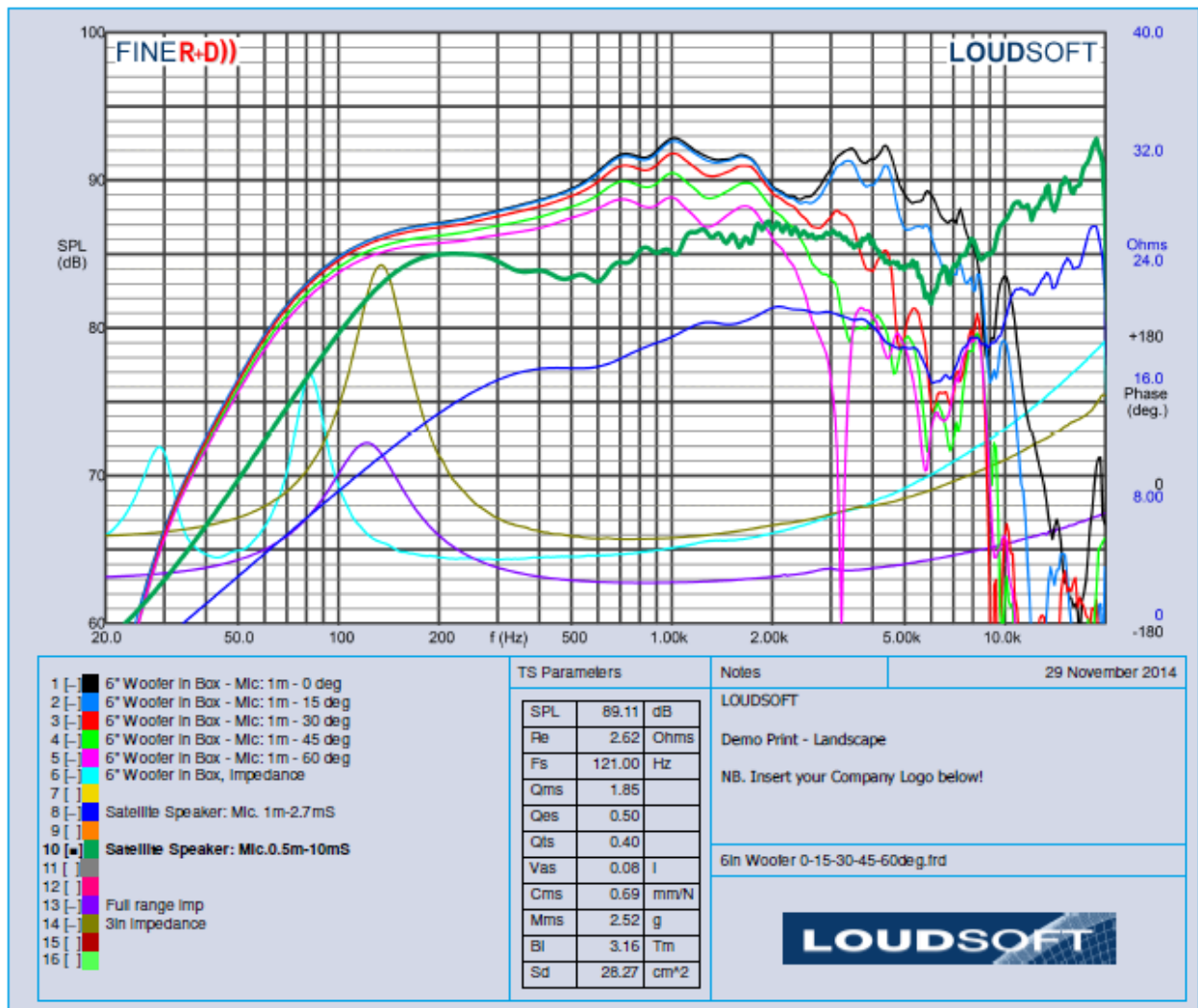


Figure 58 - Print main SPL

The default print has the LOUDSOFT logo at the bottom right. You can however insert your own logo as shown in Figure 59. The logo must be placed in the FINERandD directory: "C:\Program Files (x86)\Loudsoft\FINERandD" and must be named printlogo.bmp.

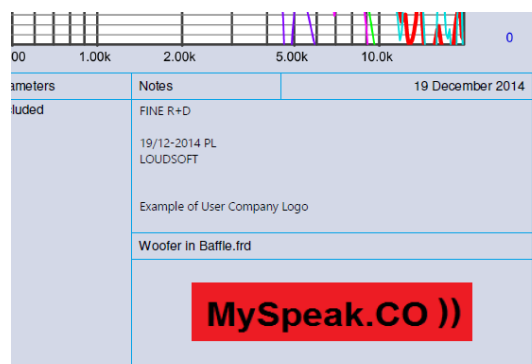


Figure 59 - Print with customized company logo inserted

21. Properties

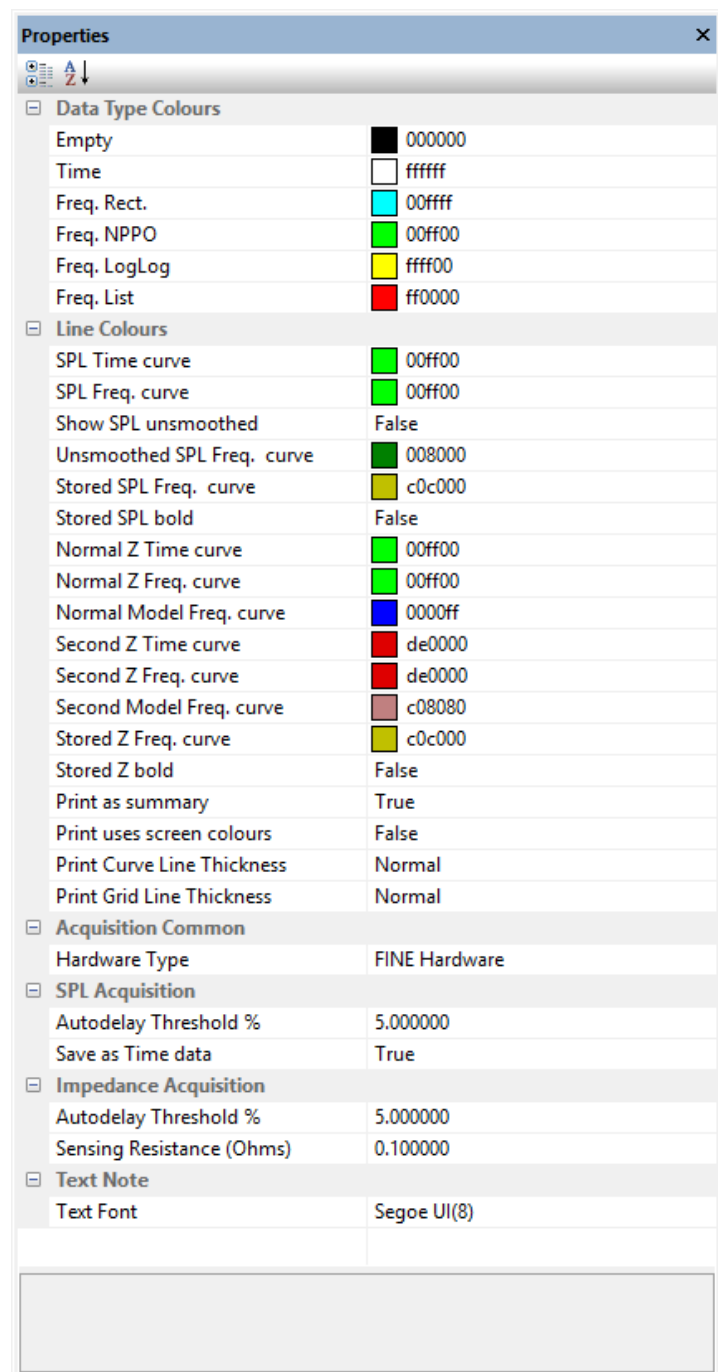


Go to properties



(View/Toolbars and Docking
Windows/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.



22. Calibration (including microphone)

Should you need to re-calibrate the system please follow the Tools/Calibration Wizard. This is valid for the FINE Hardware (3).

If you only need to insert calibration for a new microphone go to Figure 65.

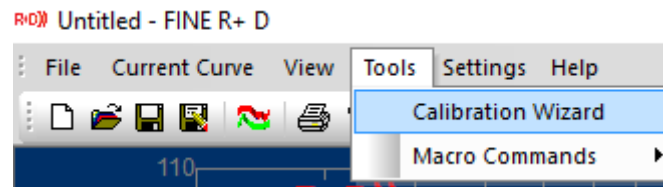


Figure 60 - Tools / Calibration Wizard

Select Tools/Calibration Wizard to perform accurate system calibration.

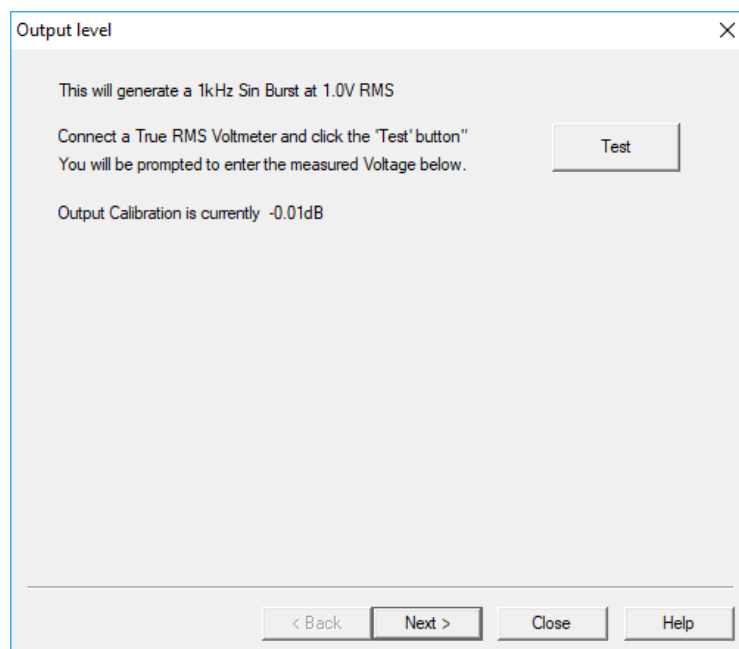


Figure 61 - Output calibration

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 V_{RMS}. (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. Repeat this for all 3 sample rates: 48kHz, 96kHz and 192kHz. Be sure to press [Apply] after each measurement as this curve will be used to calibrate and equalize the total system.

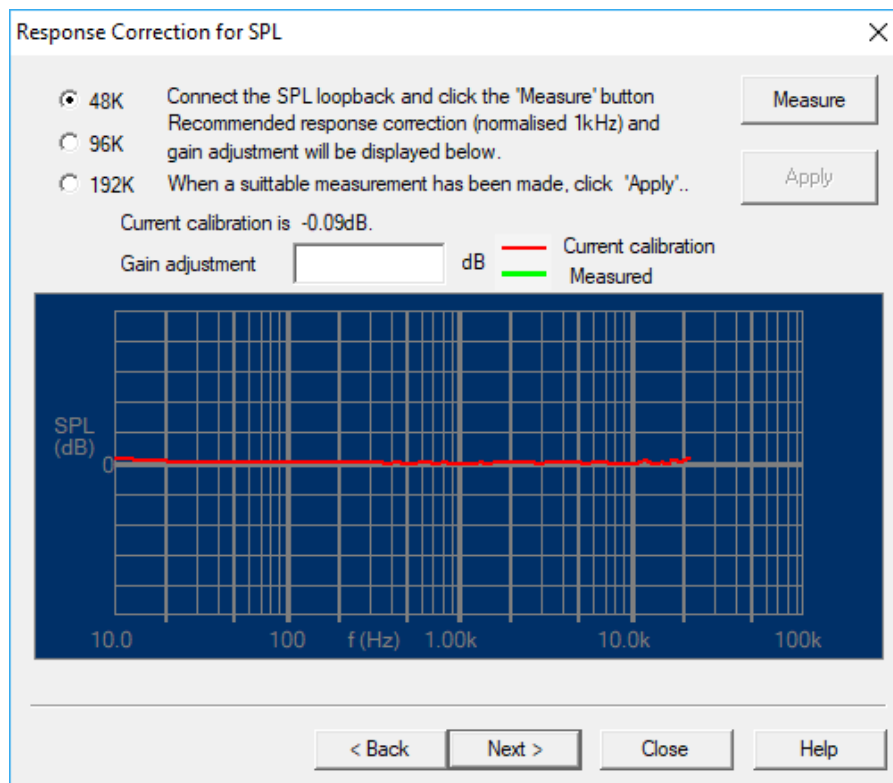


Figure 62 - Loopback response calibration

Press [Next]

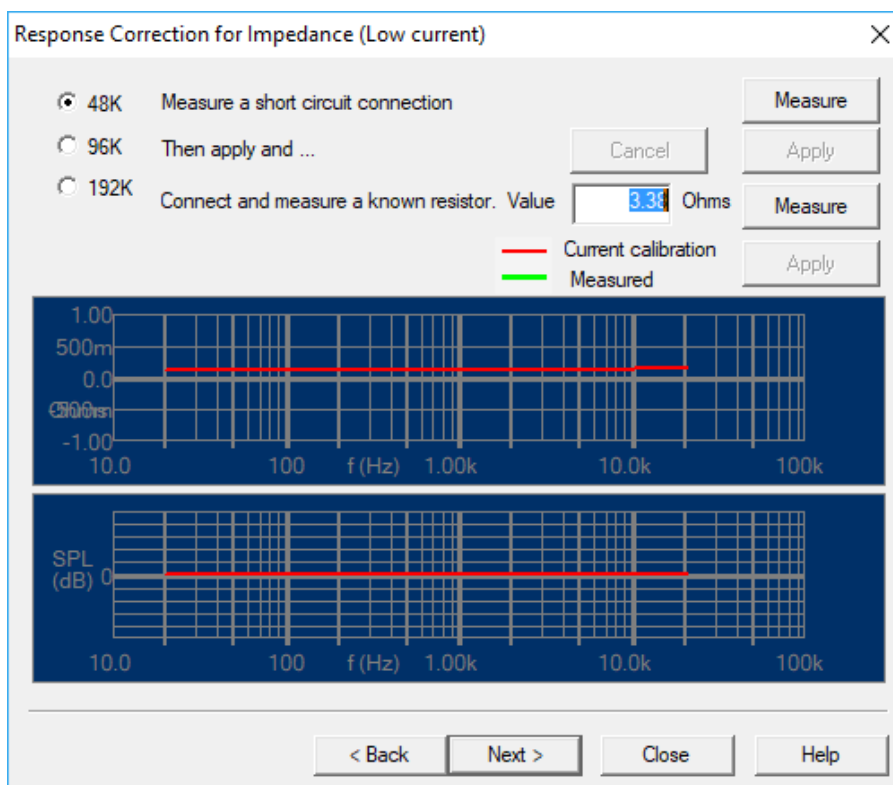


Figure 63 - Impedance loopback calibration LO

Make a short circuit across the output banana terminals, and measure using the **Upper** [Measure] button for all 3 sample rates: 48k, 96k and 192k Hz.

Then attach an accurate resistor across the banana terminals and repeat the measurements for all 3 sample rates. Be sure to press [Apply] each time as the short circuit plus resistor measurements fully calibrate each range.

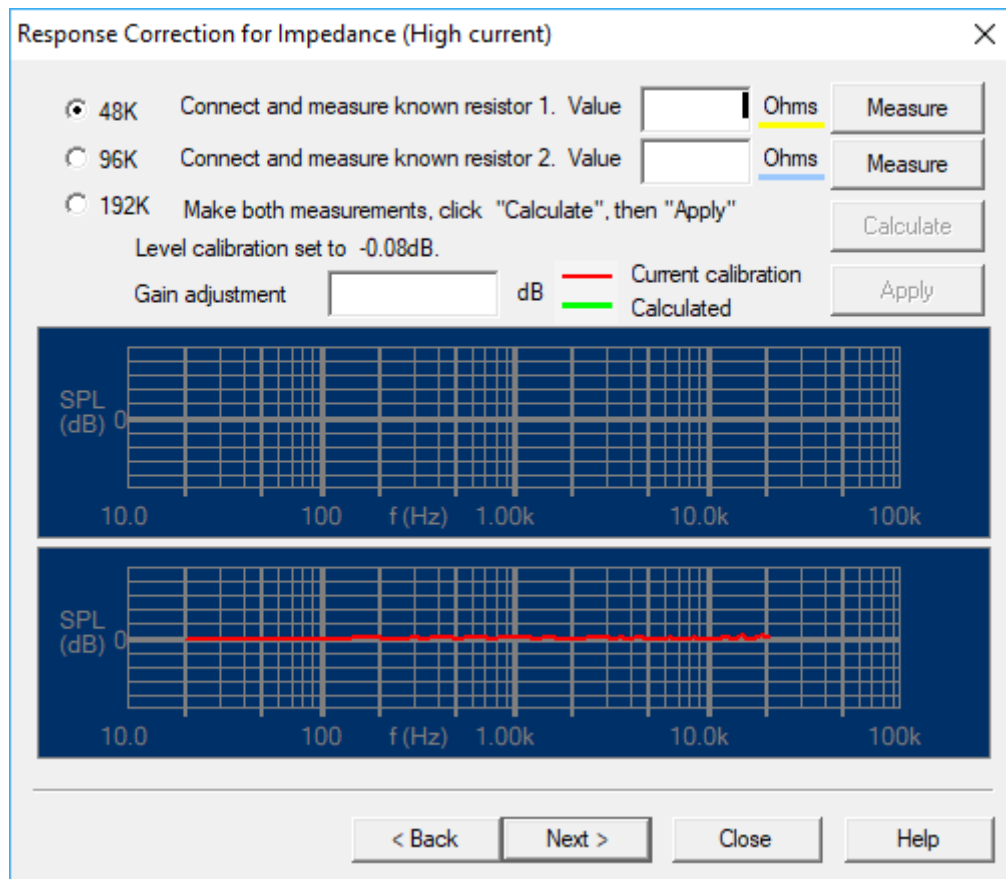


Figure 64 - Impedance Calibration HI

Now start the HI current impedance calibration by choosing two resistors with different values (around 1 ohm and around 20 ohms). Input BOTH values first and connect the first resistor, then click the **Upper** [Measure] button for all 3 sample rates: 48k, 96k and 192k Hz.

Now connect the second resistor (around 20 ohms) and click the **lower** [Measure] button for all 3 sample rates. Remember to press [Calculate] and [Apply] to calculate the output impedance in each case.

This procedure will measure and compensate for the output impedance of the FINE hardware.

Press [Next]

Connect a microphone calibrator (1Pa or 94 dB) if available and press [Measure]. The measured microphone calibration will then be 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 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.

Press [Next]

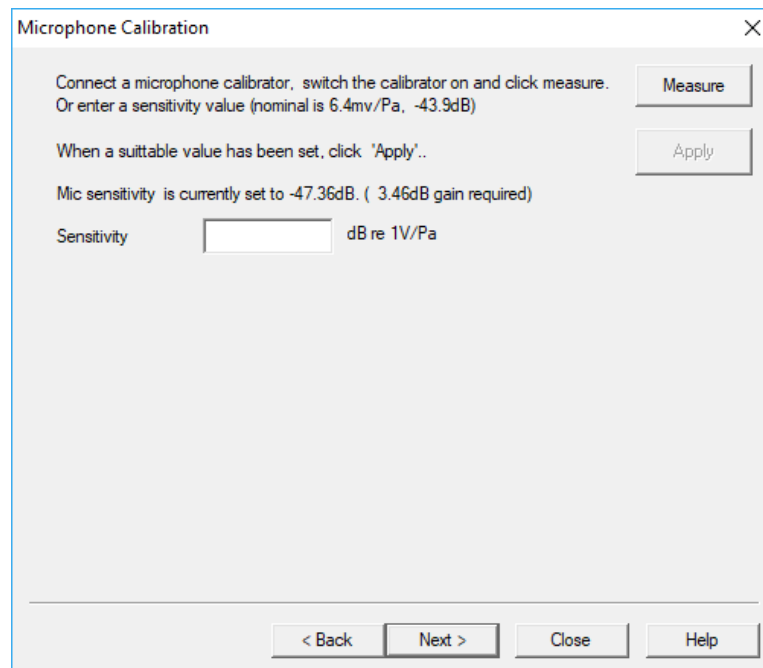


Figure 65 - Microphone calibration

Connect a microphone calibrator (1Pa or 94 dB) if available and press [Measure]. The measured microphone calibration will then be 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 is supplied with microphones from LOUDSOFT, see Figure 2.

23. Troubleshooting

23.1. Default setup

You can save your preferred start up settings as a default file, which is automatically used by FINE R+D at start-up and New:

1. Save a start.frd-file with your settings to a **different** directory first.
2. Then copy your start.frd-file to the folder "C:\Program Files (x86)\Loudsoft\FINERandD"
3. Rename the existing _default.frd to _default.frd.old
4. You can now rename your start.frd-file to _default.frd
5. Now FINE R+D will always start with this file including your settings
6. NB! If you begin having content in curve No. 1 you can just right-click and delete this curve to get a blank sheet while still keeping your settings in the SPL View/Time window

23.2. Problems when saving calibration data

You may experience an error as shown below. The path may not exist, or the program cannot save calibration data:

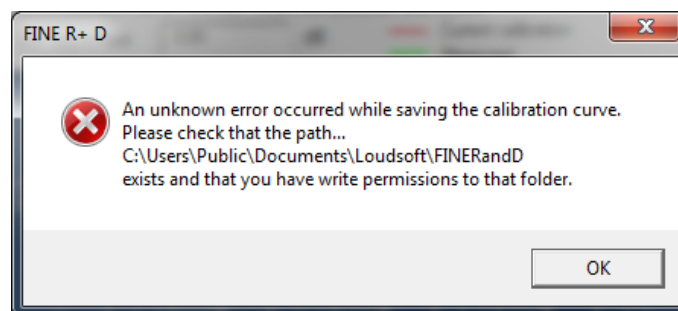


Figure 66 - Missing folder error message

In that case verify that this folder exists: C:\Users\Public\Documents\Loudsoft\FINERandD

If it does not exist you could make this new folder yourself as Administrator. Remember you must have permission to write in this folder

23.3. Standby mode, hibernate mode or sleep mode

If your computer enters standby (suspend), hibernate, or sleep mode while the FINE R+D is connected, the FINE R+D may not operate correctly when the computer returns to normal operation. Close all software that uses the FINE R+D and then reconnect it.

23.4. Unstable computer or failed start up

On some computers, starting up the computer with a USB device already connected may cause the computer to operate unstably or fail to start up. If so, start up your computer with FINE R+D disconnected.

24. FINE Hardware – Line Out

In case the built-in 25W amplifier is too small for testing your system, you can connect a larger power amplifier from the Line out XLR (balanced) output on the FINE Hardware. Remember to select the Line V out in the input / Output pane (See Figure 6)

Note that the **Line Out level must be set low**, not to overload your speaker connected to the large amplifier.

If for example your power amplifier has a gain of 30dB, then 100mV nominal Line output will produce $100\text{mV} + 30\text{dB} = 3.16\text{V}$ from your power amplifier.

The XLR Line Out on the FINE hardware 3 can output enough level ensuring that very large speakers systems, for example a large subwoofer system can be tested at full output.

Note: Do not use high output for a full range system with Tweeters/HF units as these units cannot withstand high power!

25. Specifications

FINE Hardware:

Sampling Frequency

AD/DA Conversion: 48 / 96 / 192 kHz - 24Bit

Frequency Response

SPL: 10 Hz to 22 / 44 / 88 kHz (± 0.5 dB)

Impedance: 10 Hz to 22 / 44 / 88 kHz ($\pm 10\%$)

Mic Nominal Input Level

XLR 0-1000 mV RMS. Max input is 12V peak or +14dbu (24v p-p)

Line Output:

0 – 1200 mV

Interface

USB 2.0

Amplifier

Output Power: 35 W (1 ohms load, Max THD<0.1%, 1 kHz)

10 W (4 ohms load, Max THD<0.1%, 1 kHz)

Protection: Short circuit and over temperature shutdown.



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