CONTENTS

HOW TO GET STARTED

IMPORTANT

1. First install FINELab QC software from the CD without hardware connected.

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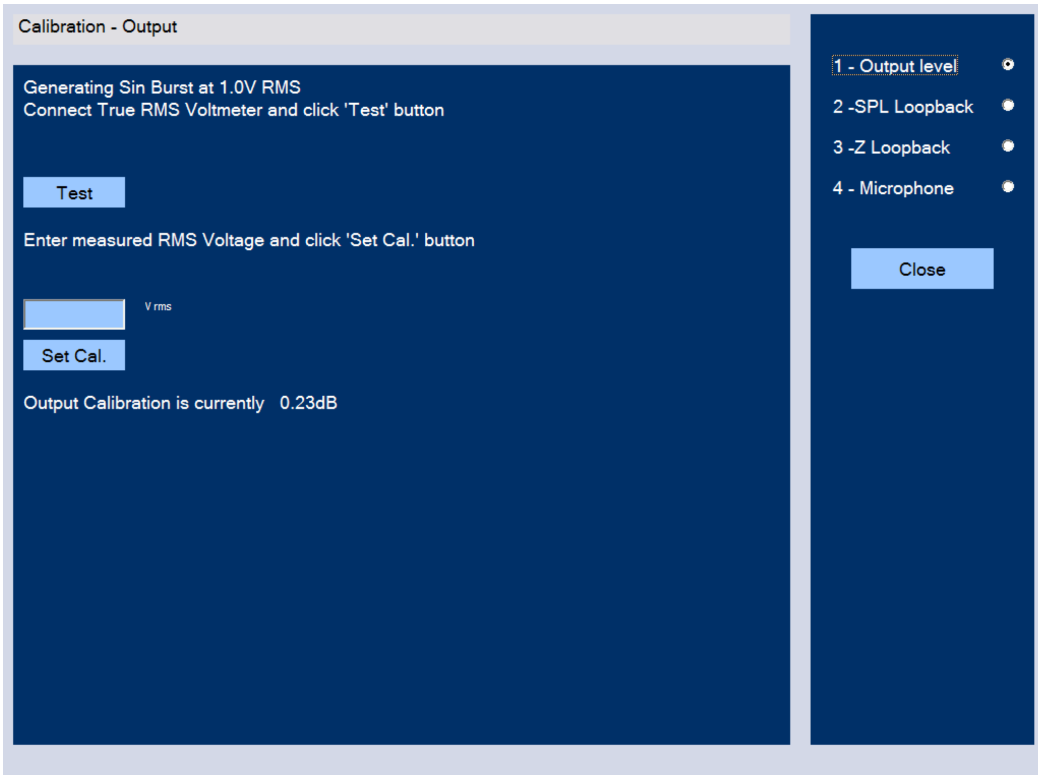
2. Then connect the power supply to the hardware box
3. Connect the hardware box to the computer with the USB cable
4. Install sound driver UA-25 from the CD
5. Start FINEQC and log in as engineer (no password needed now)
6. **Calibrate FINEQC according to page 3, 4, 5 and 6**

Now the system is ready to run.

- Open some example files, which are shown in this manual at page 19
- Start your own measurements using either: New “based on” existing file or New
- When defining your test with “Edit QC test” use “select Bands” to load standard limits like Tweeter, Subwoofer etc.
- When you have used FINEQC for some time, please tell us what you like and dislike and give us your proposal to improve the system. We will then try to make the system better and give you a **free update for your help**.

1.FINELab CALIBRATION PROCEDURE

1.1 Output level



Soundcard only:
Connect UA-25xx output (Phone/Jack) to the input of your amplifier.

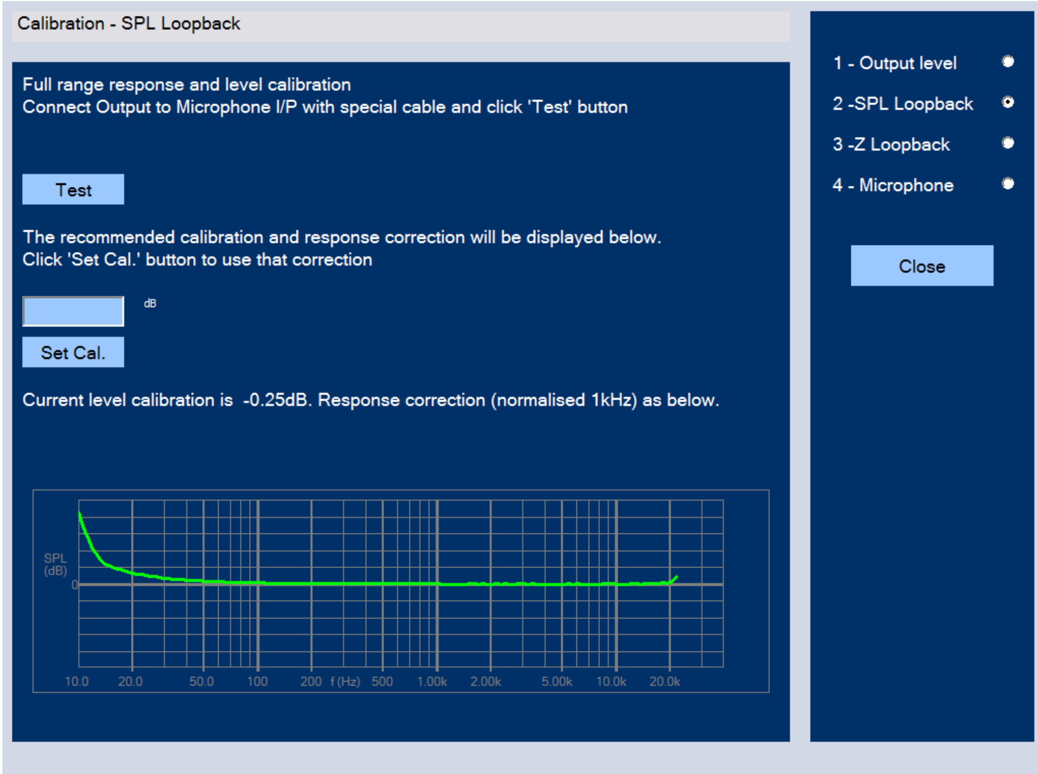
Make sure the output (upper right on the UA-25xx) is turned fully up. Then turn the volume on your amplifier almost half way up, until reading appr. 0.7-1.4Vrms, when clicking "Test".

Connect a True RMS Voltmeter to the output of FINELab/Power amplifier.
Click "Test" button.

An RMS Voltage around 1V RMS should be measured on the Voltmeter. Enter the measured RMS Voltage in the empty field "[ ] V rms" and click "Set Cal." button. The calibration of the Output level is now done.

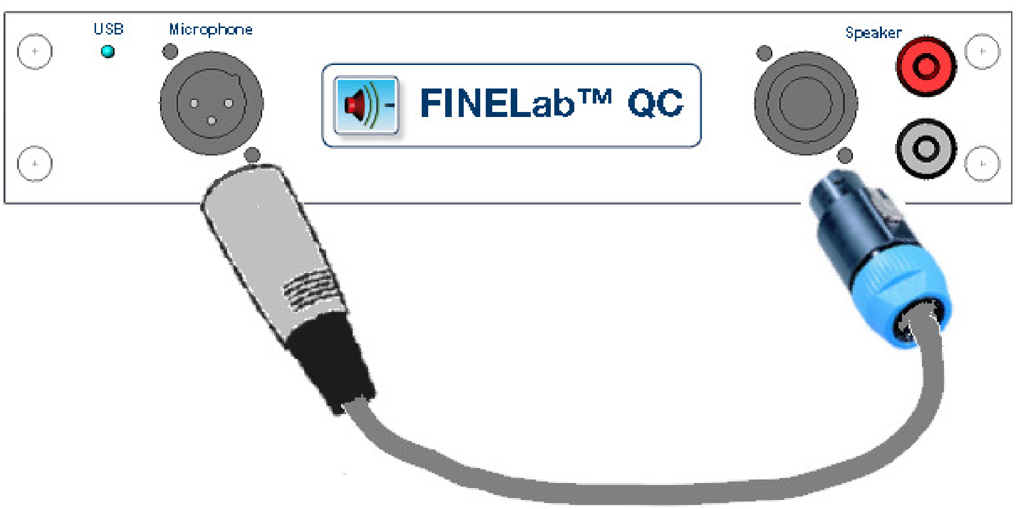
Note (soundcard): Keep the volume setting on your amplifier, and use the Output Level setting in "Edit that" to set the output voltage (Vrms).

1.2 SPL Loopback calibration



Select “2 – SPL Loopback” at the upper right of the window. This is a full range response and level calibration of the SPL.

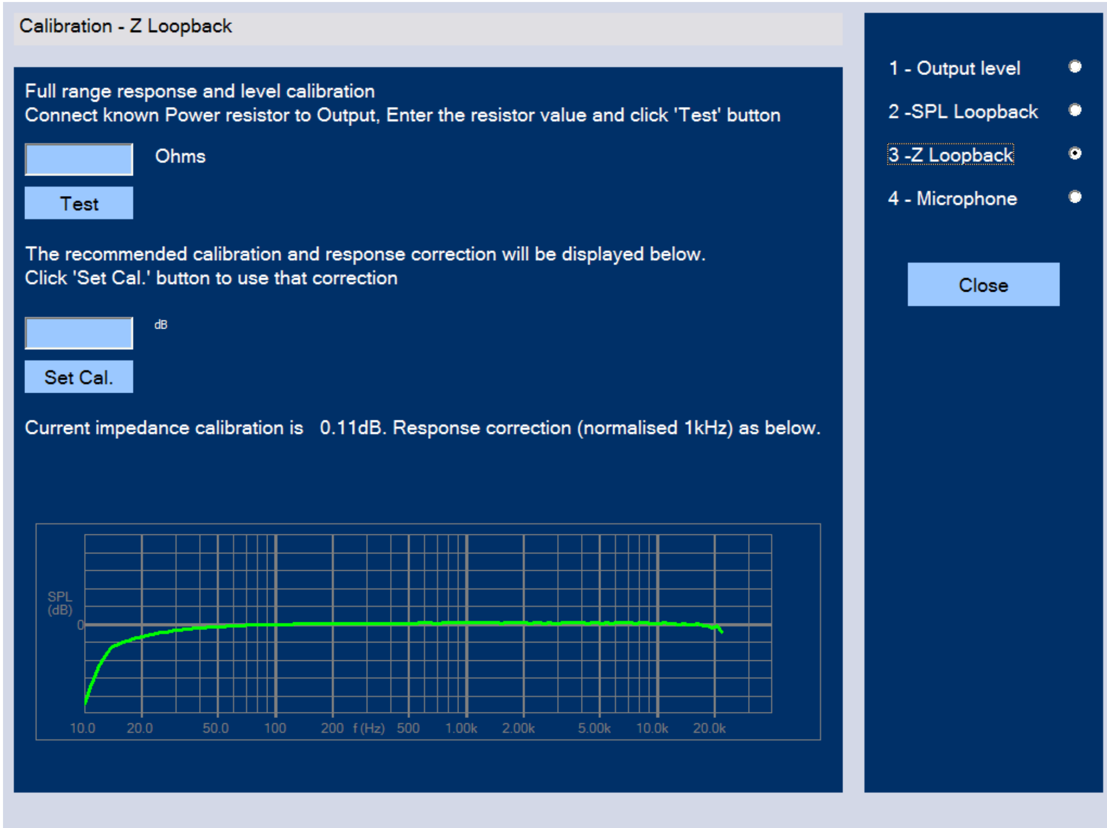
Connect Output to Microphone Input with the special loopback cable (see the figure below) and click ‘Test’ button. Soundcard: use output from amplifier to Input1/L. Keep the input sensitivity about horizontal (lower left on UA-25xx) and avoid overload. Increase microphone/input sensitivity later if needed.



After the recommended calibration and response correction is displayed, click “Set Cal.” button. Now the SPL loopback calibration is finished.

Soundcard: Keep this volume setting on your amplifier, if you want to keep the SPL calibration. Otherwise just repeat the SPL Loopback calibration.

1.3 Impedance (Z) calibration using FINELab hardware



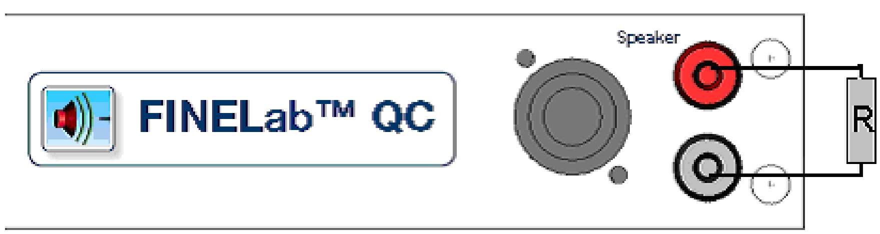
Select "3 – Z Loopback" at the upper right of the window. This is the full range response and level calibration of the impedance.

FINE(Lab) - Do the following steps to calibrate the impedance: (See Soundcard version on next page)

1. Take a 4 ohms 5W resistor (or closest available)

2. Measure the resistor with an accurate Low Ohms Digital Multimeter. The value should have two decimals, e.g. 4.31 ohms.

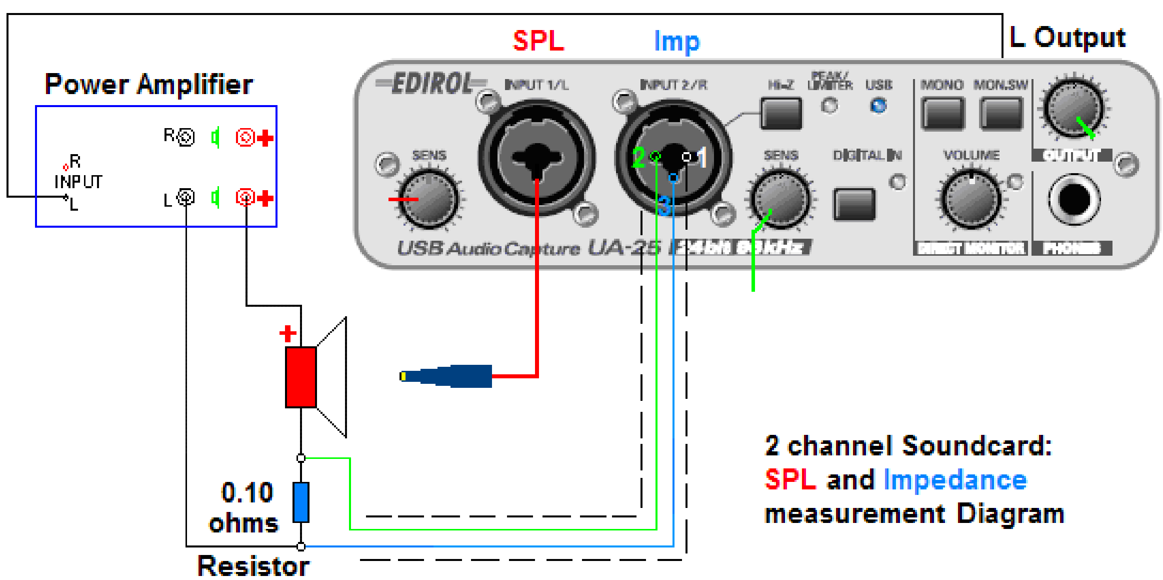
3. Connect the resistor to the Output; shown in the figure below (You may include your cables to make a more accurate calibration). Enter the measured resistor value in the empty field "[ ] Ohms" and click "Test" button.



4. After the recommended calibration and response correction are displayed, click "Set Cal." button. The calibration is finished.

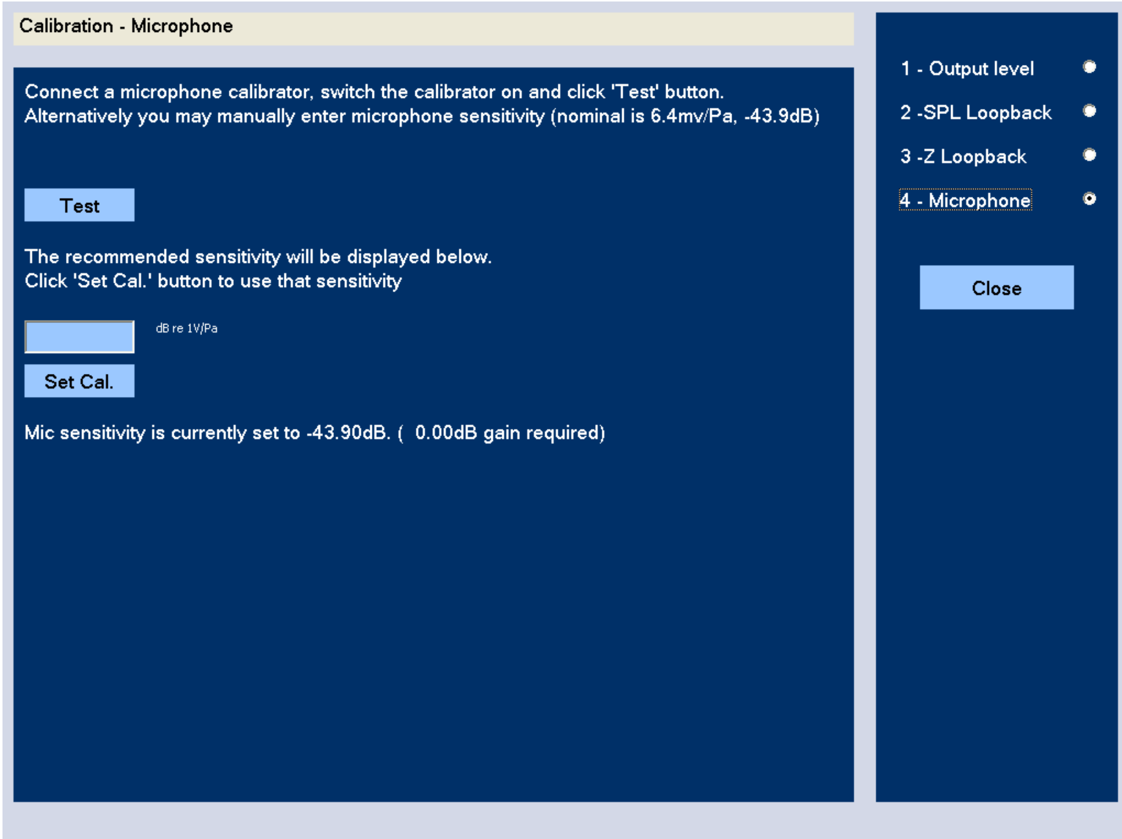
1.4 Impedance (Z) Calibration using soundcard

1. Take a 4 ohms 5W resistor (or closest available)
2. Measure the resistor with an accurate Low Ohms Digital Multimeter, preferably with two decimals, e.g. 4.31 ohms. Or use the given value.
3. Replace the loudspeaker, shown in the figure below, with the (4 ohms) resistor. Connect the two cable leads across the (0.10 ohms) series resistor as shown in the diagram. This cable should be connected to the right input (2/R) using XLR connector.
4. Turn the right channel input Attenuator (Sens.) fully down as indicated ___. Enter the measured resistor value from 2) in the empty field "[ ] Ohms" and click "Test" button. (Mon SW should NOT be active – no red light).
5. After the recommended calibration and response correction are displayed, click "Set Cal." button. The calibration is finished. *Note: You can measure both SPL and Impedance simultaneously with a two channel soundcard (UA-25xx)*



Note: You can measure both SPL and Impedance simultaneously with a two-channel soundcard (UA-25xx)

1.5 Microphone calibration



A). Calibration using microphone data:

The microphone sensitivity can be found from the LOUDSOFT microphone Calibration Sheet supplied with the microphone.

Enter the microphone sensitivity value in the empty field “[ ] dB re 1V/Pa”, and click “Set Cal.” button. Now the calibration is done.

Note using Soundcard: This calibration (A and B) is only valid if the SPL Loopback was done with the current volume (gain) setting of the amplifier.

B). Calibration using a microphone calibrator:

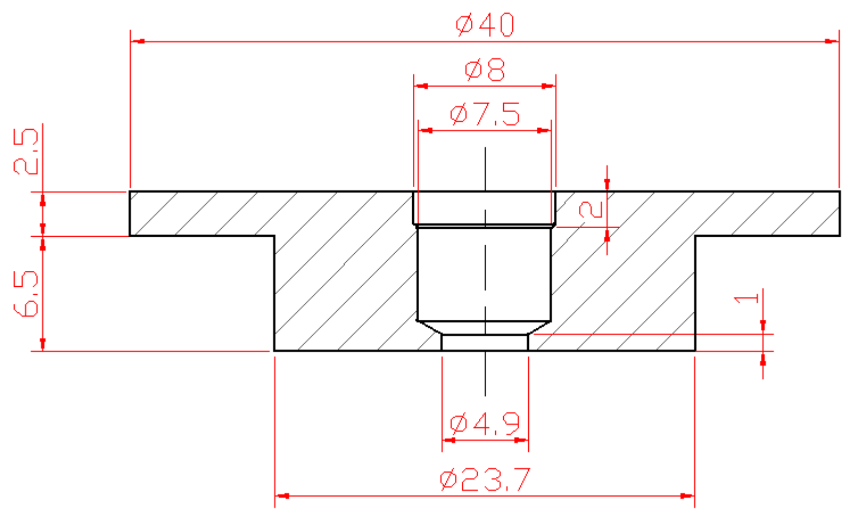
This is the most accurate method. Place the microphone on the calibrator using an adaptor. (See the attached sketch showing an adaptor for the LOUDSOFT FL1 microphone.)

Switch on the calibrator and hold the microphone firmly (the sound from the calibrator is then barely heard).

Press the test button and the measured microphone sensitivity is displayed.

Press the “Set Cal.” button to use this. The microphone calibrator must give 1 Pa (94 dB SPL).

1.6 Adapter for LOUDSOFT FL1 microphone



Adapter for LOUDSOFT FL1 Microphone

For use with B&K Type 4230 Calibrator

Material: ABS or similar

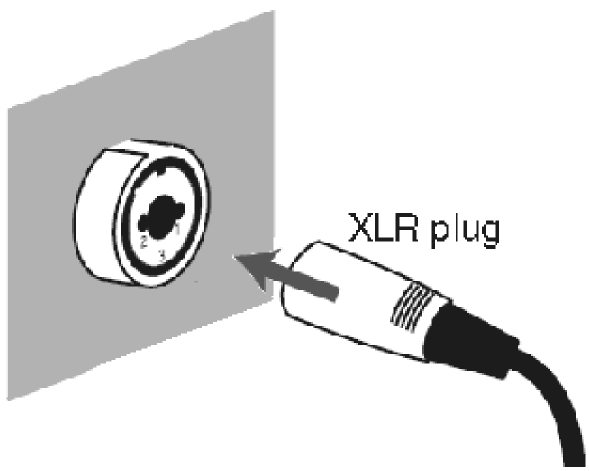
2. FINELab HARDWARE (Box)

2.1 Front panel

1. USB Blue LED

When FINELab is connected to the computer with a USB cable, the blue USB light LED is on.

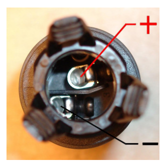
2. Microphone input



Phantom power (48 V) is provided for XLR type connections, allowing you to connect a condenser microphone that requires phantom power.
NB: REMEMBER TO SET THE REAR PHANTOM 48V SWITCH TO ON! (SEE PAGE 9)

3. Amplifier output - Speakon connector

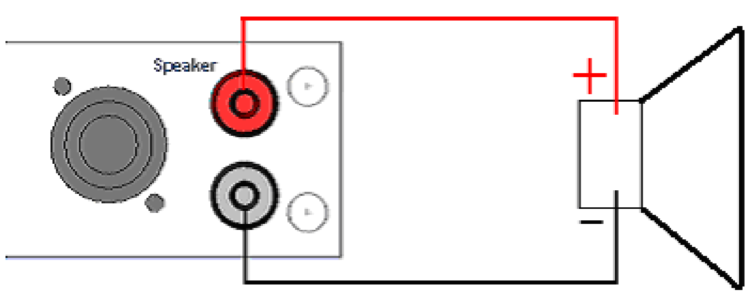
You can use the Speakon connector to connect a loudspeaker.



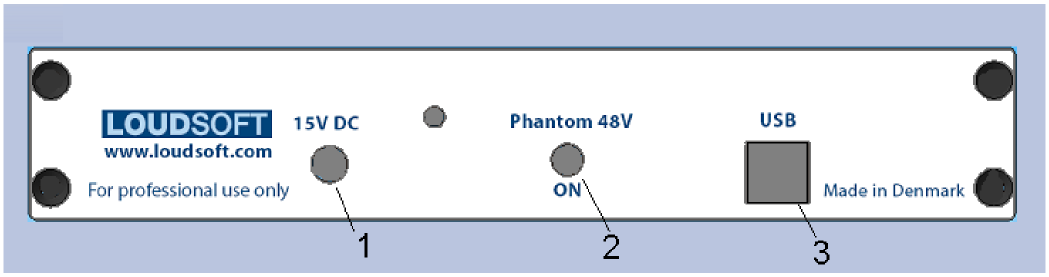
Speakon rear connections

4. Amplifier output - Banana connector

Or, you can use the banana connectors to connect a loudspeaker.



2.2 Rear panel



1. **Power Input jack:** 15 V DC

2. **48 V Phantom Power Switch on/off button:** Must be on with the LOUDSOFT FL1 Condenser Microphone.

3. **USB connector:** Use a USB cable to connect the FINELab hardware (Box) to your computer.

2.3 Main Specifications

Signal Processing
PC interface: 24 bits
AD/DA Conversion: 24 bits (linear)

Sampling Frequency
AD/DA Conversion: 48 kHz

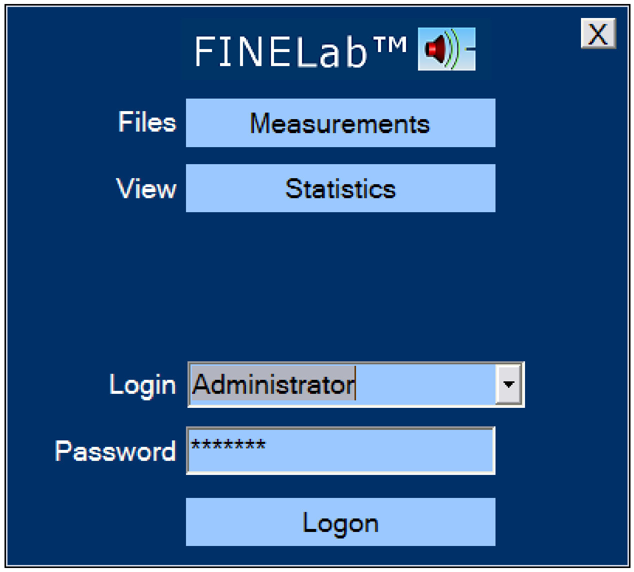
Frequency Response
SPL: 20 Hz to 20 kHz (±0.5 dB)
Impedance: 10 Hz to 20 kHz (±10%)

Nominal Input Level
XLR 0-100 mV RMS

Interface
USB 1.0

Amplifier
Output Power: 35 W (2 ohms load, Max THD<1%, 1 kHz)
20 W (4 ohms load, Max THD<1%, 1 kHz)
Protection: Short circuit.
Over temperature shutdown.

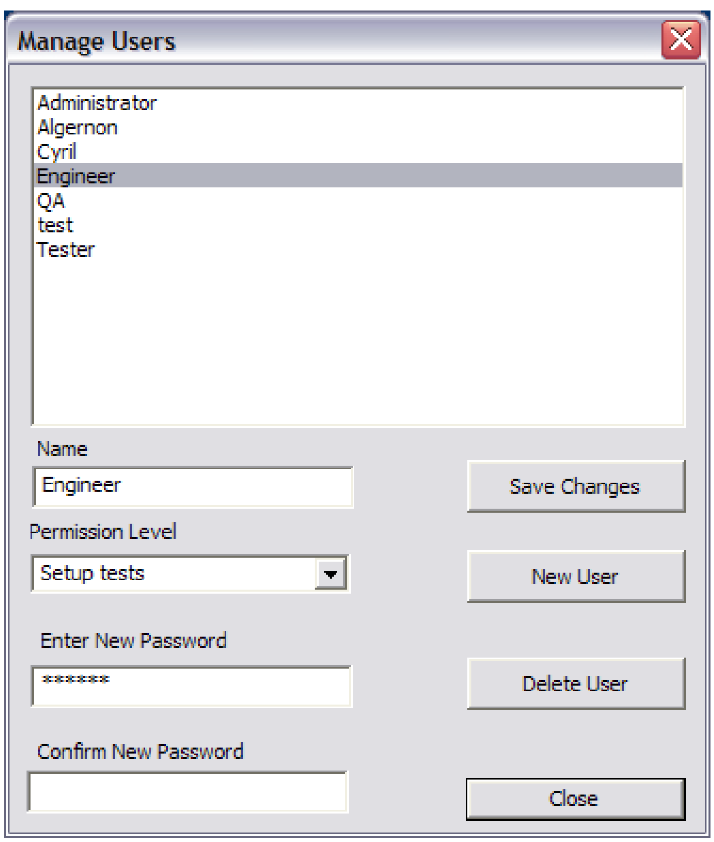
3.LOGIN - Users



FINELab start screen

You can login the first time as Engineer without a password. To setup Users you need to login as “Administrator”. The default password is “FINELAB” (Capital letters!).

There are 4 default user levels: Administrator/Engineer/QA/Tester. Each has limited access, except the Administrator. You can setup User Passwords and change all as Administrator: [Admin Options/Users].



4. T U T O R I A L

4.1 Statistics

The FINELab Start Screen is shown on page 13. Without logging in all users can directly load Measurement Files and view Statistics from automatically saved QC test series. An example is shown in the next picture which is the result of a previous test series:

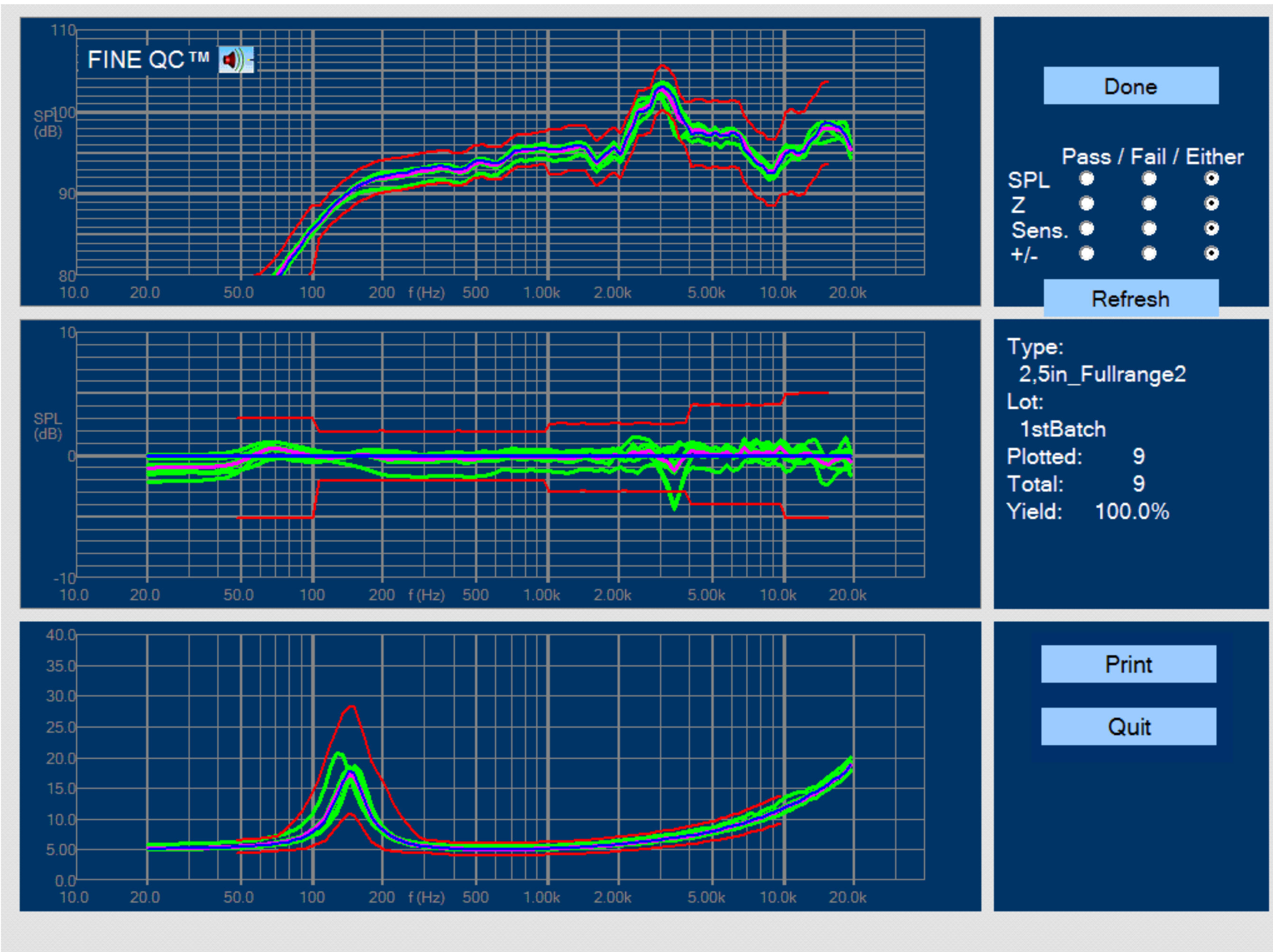


Figure 1 - Statistical Result of previous QC Test Series

The frequency responses of all units tested are shown in the upper window as green lines, with the tolerance limits shown in red. The average of all responses is shown in violet, however you can select to view all Pass or Fail for SPL, Impedance, Sensitivity or Polarity as you wish plus getting the test yield for each.

The middle window shows the same responses as above, but now plotted relative to the reference (blue). For example a change in sensitivity is very easy to see this way.

The Impedance with limits is plotted in the lower window. The average of all responses is shown as the violet curve and the reference as the blue response.

4.2 QC Testing

This time I log in as an engineer with my own password and select “2,5in_Fullrange2” and “Run that Test”. The following screen appears:

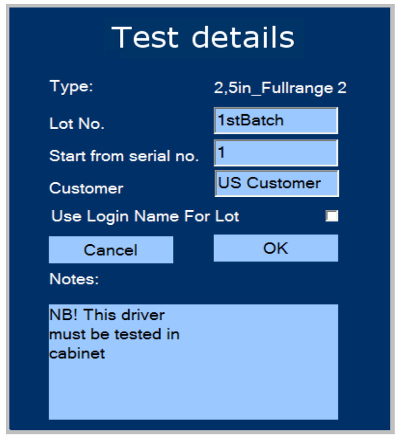


Figure 2 – FINE QC Test Details

After filling in the Batch number and customer name this series will start with #1 and automatically count up as you test. If you enter the last tested number instead, FINELab will count up from that.

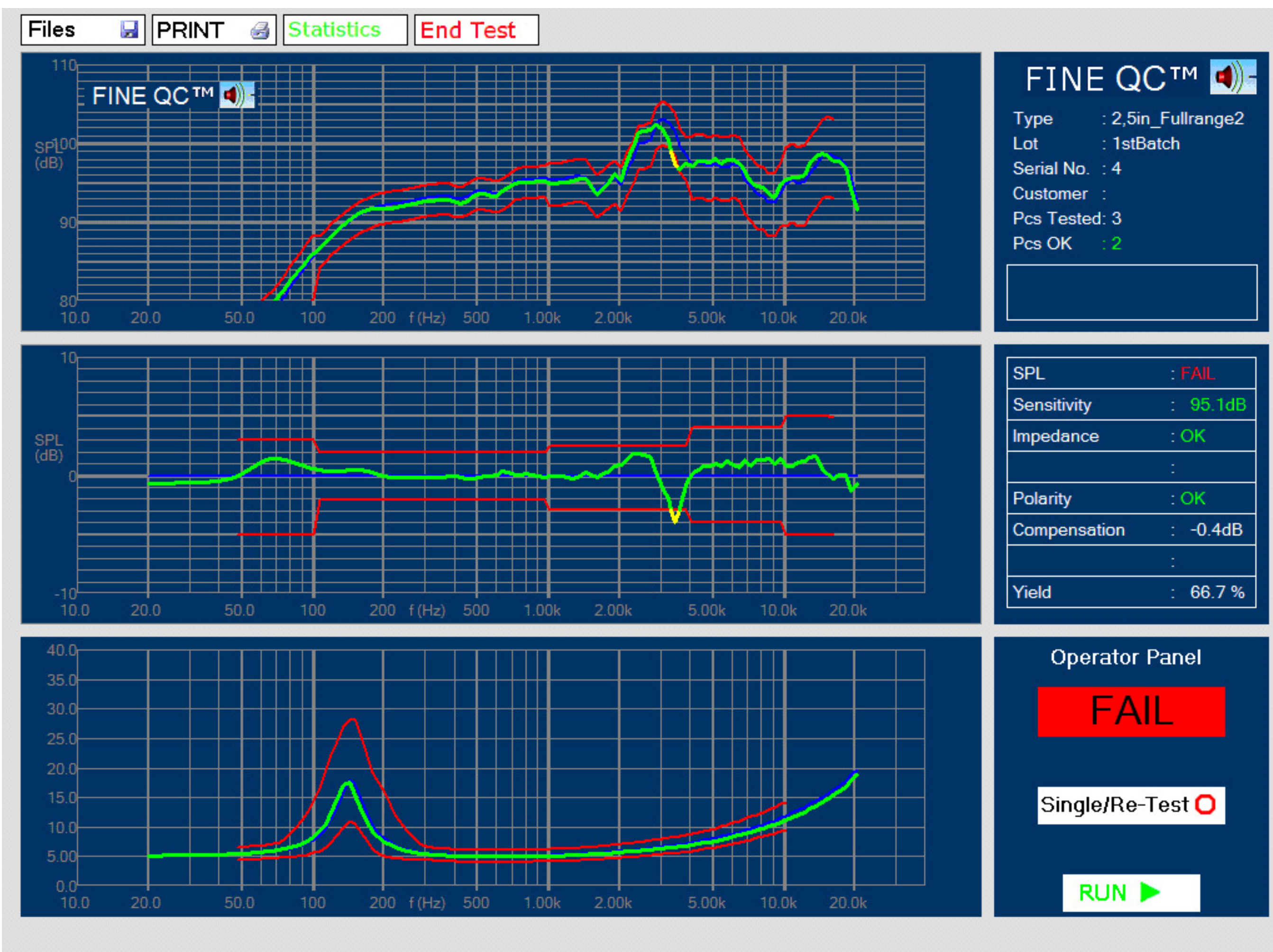


Figure 3 – FINE QC Test Display: SPL FAIL at 3.5 kHz

As a good rule I will start by testing my reference driver by pressing “Single/Re-Test”, which ensures that this initial reference test will not be saved in the test series. (I can accept small sensitivity changes of the measured reference driver response if caused by a change in environment temperature).

I start the test by pressing RUN ▶

Now the first driver is tested with a fast sine sweep. In this case the speed was chosen to be 2.5 seconds, so the tester can listen for distortion and Rub & Buzz while testing. The measured response is shown in the upper window as a green curve, with the response limits in red.

If the measured response is exceeding the limits, the colour changes to yellow and SPL: FAIL is reported in the right centre window. The large centre window is showing the measured response compared to the reference driver. Here it is much easier to see when the response is outside the limits, see the example in Fig. 3.

The sensitivity can be defined as an average over a frequency range, in this case 95.1dB (700-1200Hz). The impedance is measured with the *same sweep.* That saves time and ensures that the level and resonance Fs is the same as in the frequency response. (A too low current may show Fs much higher). The polarity is also checked and reported OK.

The next line is (Limits-) Compensation: This is a kind of Sensitivity Controlled Floating Limits. When the response is measured it is allowed to move within the limits as determined by the sensitivity tolerance. Finally the actual yield of the test is calculated.

As soon as “End Test” is pressed the Statistics of the series is displayed and the user can view rejects etc. (see page 14).

4.3 Golden Average Driver / Preproduction

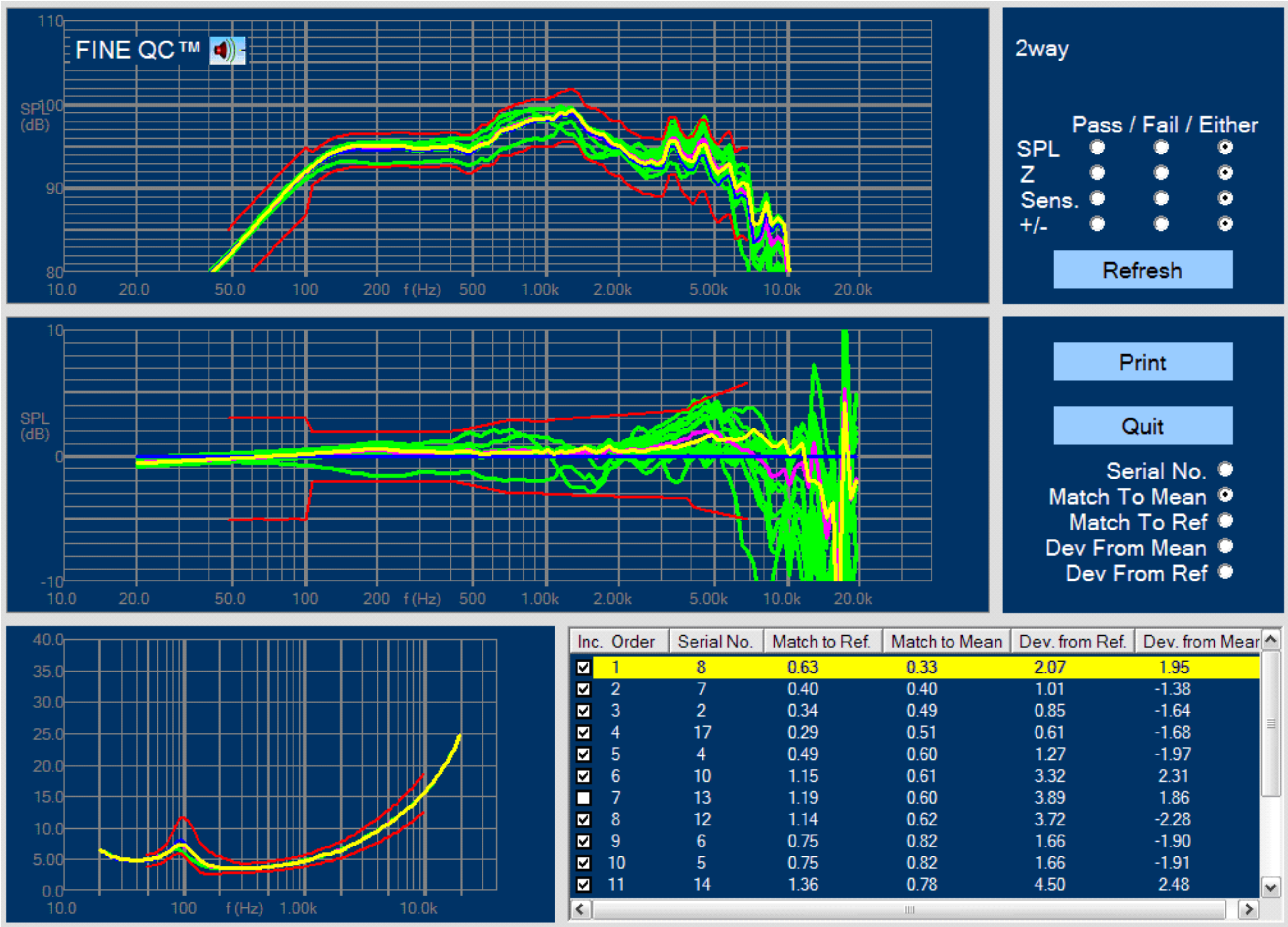


Figure 4 -Golden Average Driver Auto Finder

When starting a new production, the most important is to find the unit which is closest to the average of the good units, so it can be used as reference.

Our pilot run consists of 17 woofers for a 2-way system, which are sorted using the Preproduction feature, see Fig. 4. The highlighted driver response (yellow) is Serial No.8 in the table and is the best match to the average i.e. the *Golden Average Unit.*

Note that driver No. 7 is deselected in the table because that response was considered non-typical and should not disturb the average.

Should it be necessary to find a similar reference driver later, that can be found by selecting “Best Match to Reference”.

Note: You can also use this feature to match drivers or speaker systems!

The matching is using the frequency range of the red limits. You may choose to see the max deviation instead.

All the data including phase, can be exported as a *.csv file to Excel, see Fig. 5. The default is export of SPL without phase. When you want to export with phase you need to go to ADMIN and set this option (see Login in the Reference Manual).

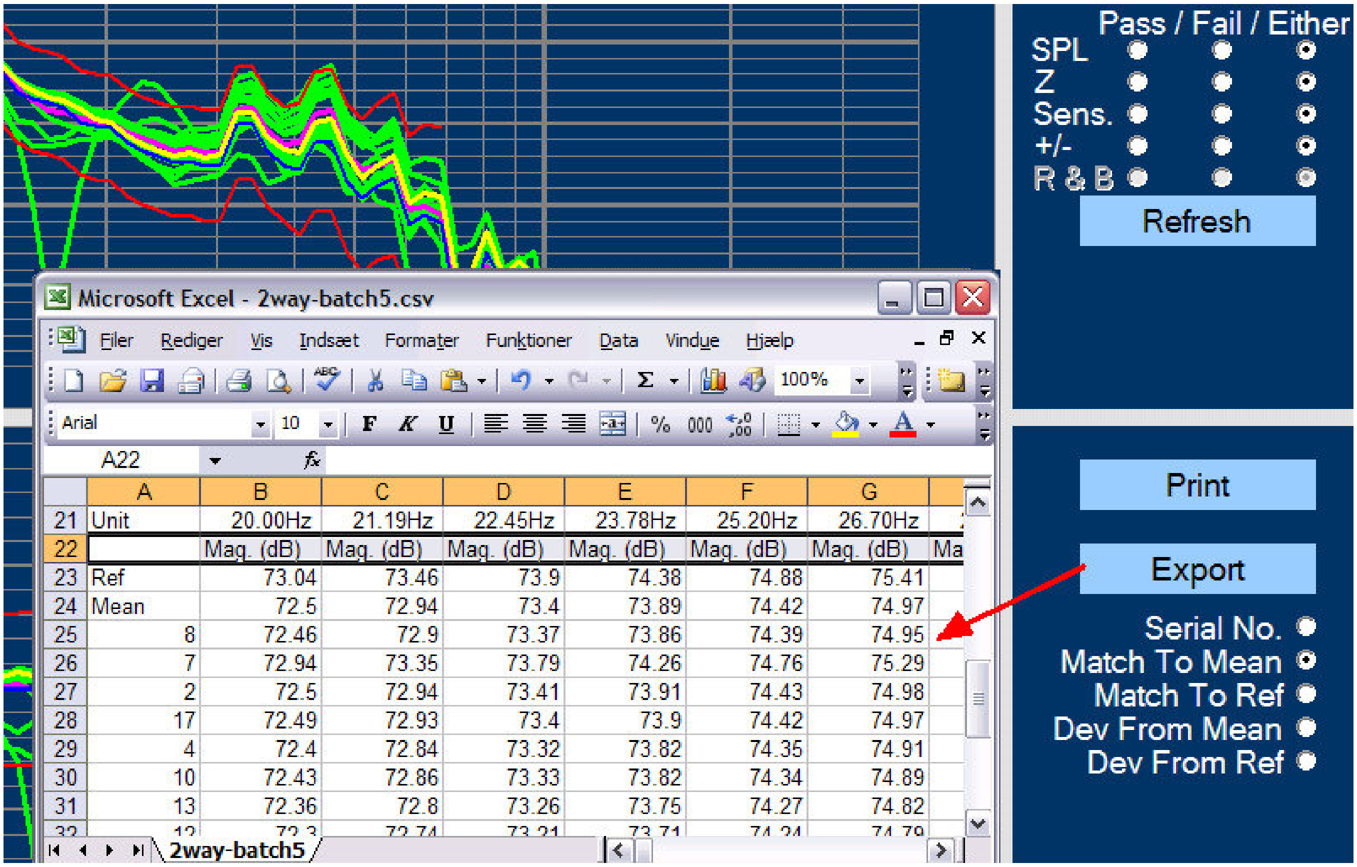


Figure 5 – Export of all data to Excel, including phase (if set in Admin)

4.4 Edit Limits

From the statistics in Fig. 1, I see that the rejected responses have a dip at 3.5 kHz, but actually it is the slope of the peak which has changed. Therefore it makes sense to adjust the limits to allow that. Select “Edit that test” from the menu (Fig. 6):

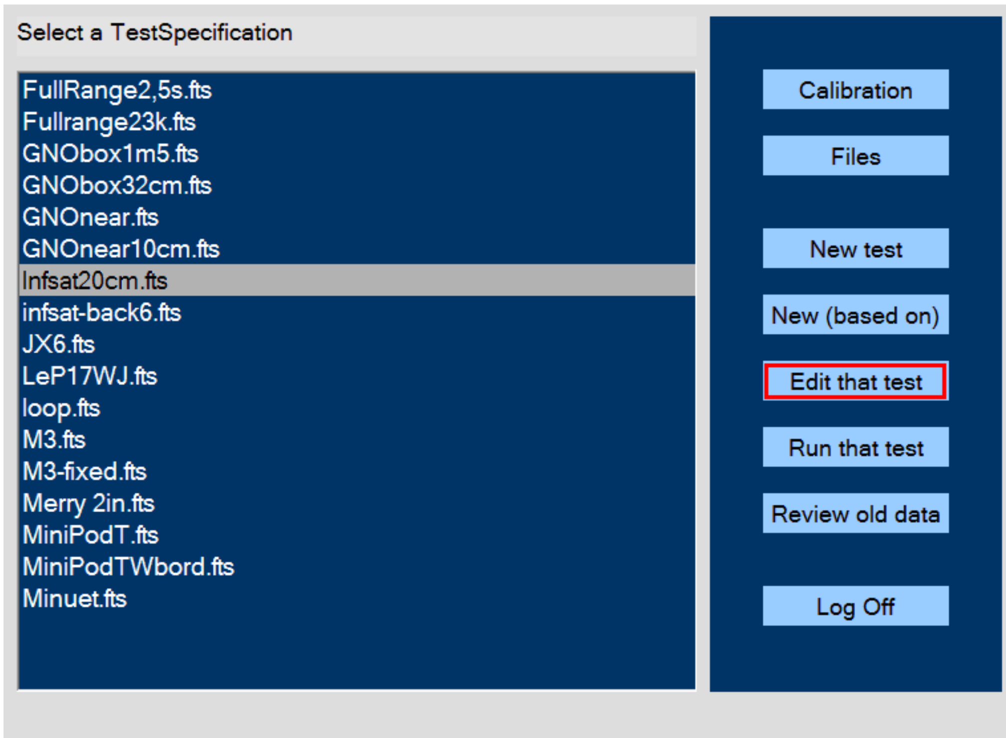


Figure 6 – Menu

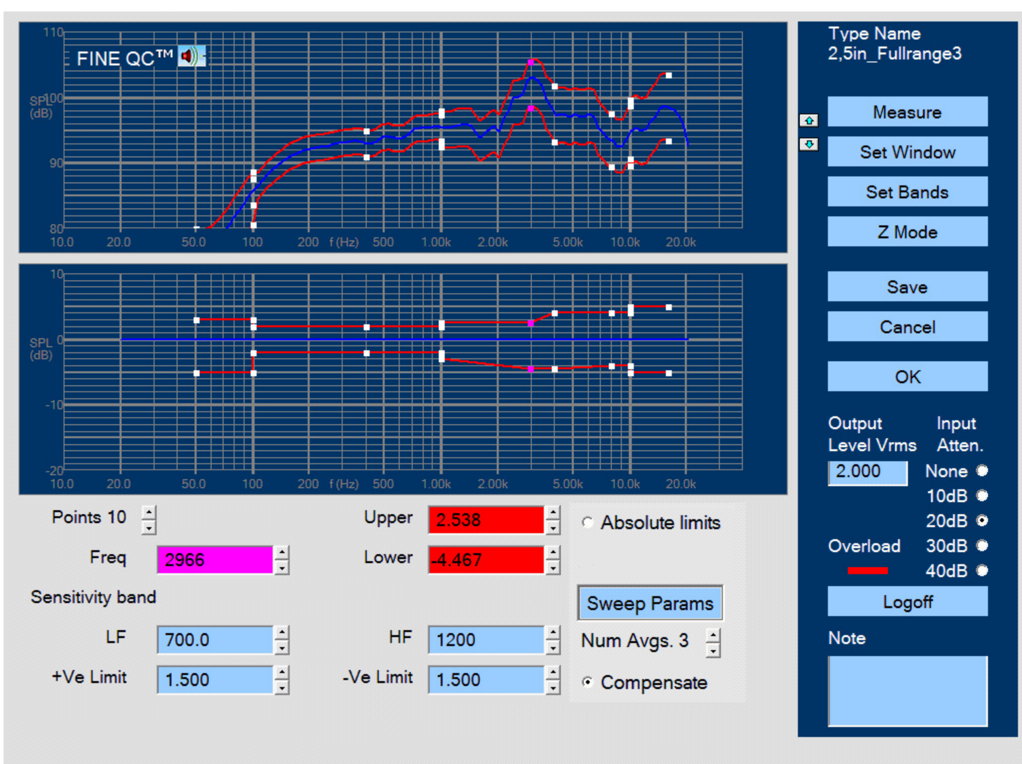


Figure 7 - Edit Limits & Sensitivity Band

The limits are broken up in ranges separated by white square points (Fig. 7). These can be selected and changed individually by clicking with the mouse or by using the up/down arrows for the points. I have selected point #10 (shown in magenta when selected) and I can either just drag and move the point, or use the upper and lower fields (Red) to change the frequency and dB deviation from the reference.

In order to allow for the moved peak I have lowered the frequency of point #10 to 3 kHz (2966 Hz magenta) and increased the lower limits to 4.5dB (red). After saving the setup file as 2,5in_Fullrange3.fts I want to test the batch again and view the statistics, Fig. 8:

This time all responses are within the limits and all units passed. While the difference in the lower window seems large, the upper window shows the lower peak being close to the frequency of the peak of the reference.

Note that the screen Fig. 7 is also where I have specified the sensitivity range from 700-1200 Hz (Blue fields), where the sensitivity is calculated as the average with a +/- 1.5dB tolerance as specified in the two lower fields (Blue). The radio button “Compensate” is also activated, meaning that the tolerance limits are allowed to move up and down with the same tolerance. This is the most realistic way to test the frequency response limits; for example a driver with +1dB sensitivity should be tested with the limits offset by +1dB.

The output drive level from the built-in amplifier is here set to 2Vrms, which is producing a reasonably high SPL without overloading this 2.5 inch full range speaker. Fig. 7 shows overload (Red) and the attenuator should be set one step lower.

If there is casual noise close to the test, you can specify a number of averages (Fig. 7). This way the test will automatically repeat the specified number of times, and use the average for testing.

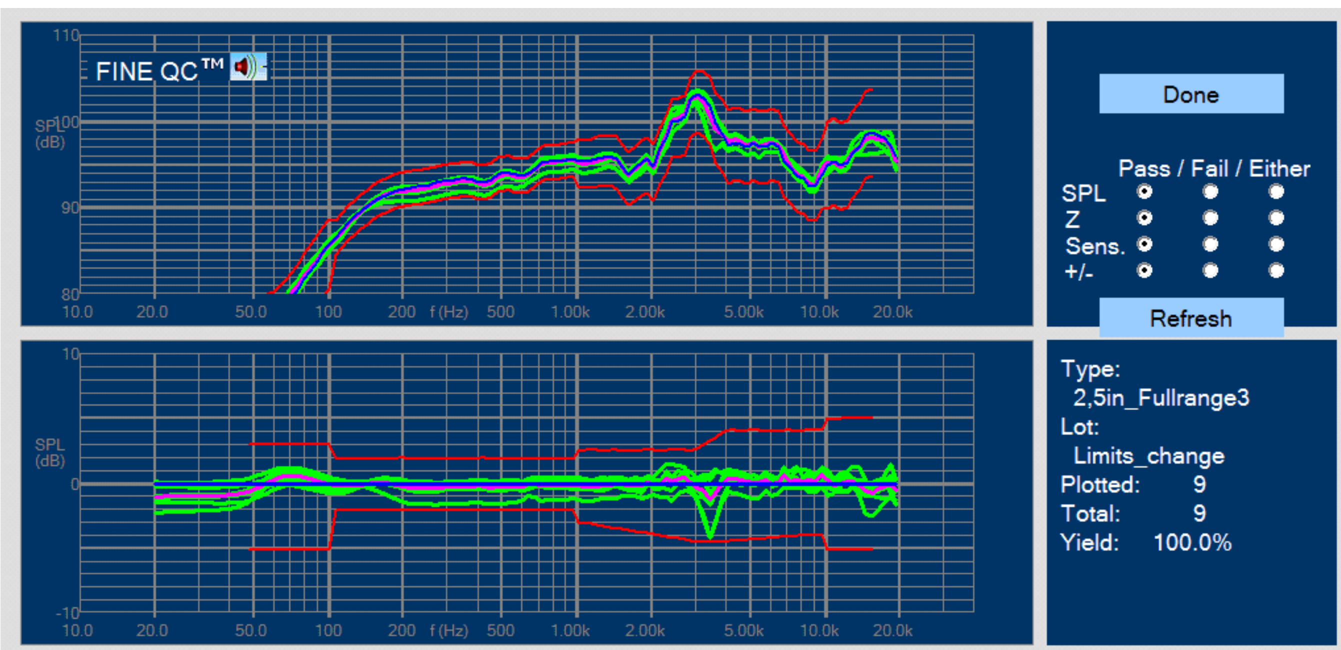


Figure 8 – SPL Statistics with adjusted limits

Note: When using soundcard - Keep the volume setting on your amplifier, and use the output Level (Fig.7) to set the output voltage (Vrms).

4.5 Measurement

In this chapter I will show how an entire test setup is created. I select “New” from the menu and get the following window (Fig. 9), where I first specify the File name:

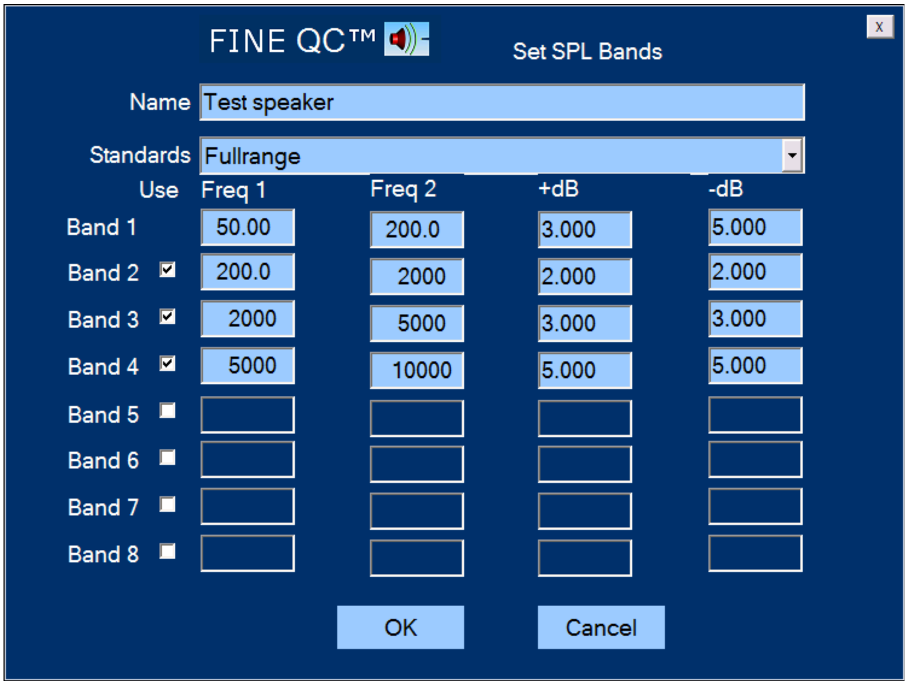


Figure 9 – Full-range SPL Template

I have selected the “Full-range” standard (Template) from the drop-down menu, which contains generic standard templates for the most used speakers. The 2.5second sweep time is slow enough for the tester to listen for bad sounding drivers during normal testing. However the best is to find the Rub & Buzz using the new FINEBuzz feature, which can be set in “Edit QC Test”.

The limits can be specified in up to 8 bands; in this case the standard template is using 7 bands with +/-2dB from 100-1000 Hz, which is the stable mid-band region before break-up. The standard limits “Window” or “Mask” is opening up towards low and high frequencies, where we expect more deviation due to shifts in resonance frequency Fs and break-up at high frequencies. The user can change the limits any time if needed.

Figure 10 – Full-range Impedance Template

After accepting the SPL limits the Impedance Template appears Fig. 10, with just one range specified from 50 to 10,000 Hz. The deviation is here defined as a ratio because we are measuring impedance. The max and min ratios are 1.2 (20%) and 0.8 (80%) which I choose to use for now. I may later need to open the limits to allow for variation of Fs.

Now I measure the speaker by clicking the “Measure” button. After the sine sweep is played the next button down is “Set Window”, which brings up the Time Domain window Fig. 11:

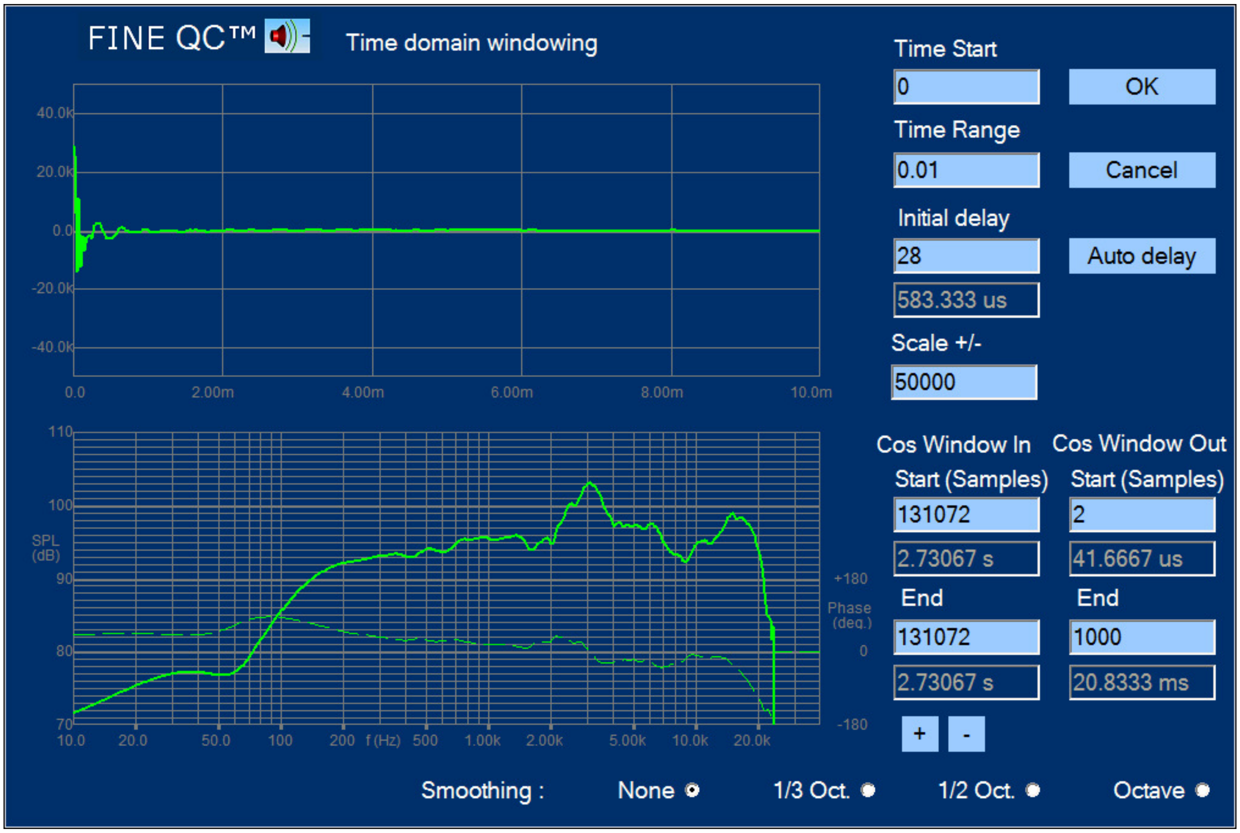


Figure 11 - Time Window settings

The impulse response is shown in the upper half and the windowed frequency response below. Most of the buttons in the upper half are automatically providing scaling of the impulse response. The time before the impulse is arriving is called the “Flying Time”, which is the travel time (in air) from the speaker cone and until it reaches the microphone. The “Auto delay” button will find this time automatically and is on by default.

Note:
If you move the microphone closer to the speaker after saving the Setup file (.fts), you may miss the initial impulse and the frequency response will look “strange”. If you know that the microphone could move say 1cm closer, you should reduce the Initial Delay setting by 2 (1 gives appr.0.7cm).*

Therefore I only need to care about the end of the impulse, which is indicated in the lower right field: I have chosen 1000 samples or 20.8mS corresponding to 50Hz (using the 1/f ratio) using a cosine/Hann window (Cos window Out). 20.8mS is a long time, but is useable in this case because a large well damped test box was used with the microphone at ~20cm distance, which can be calculated from the initial delay of 562.5uS, giving the actual distance of 19.3cm.

The final windowed frequency response is useable from about 50Hz, showing good response from 150 Hz and almost up to 20 kHz. The acoustic phase response is shown as the dashed line and is well behaved and close to minimum phase.

After OK the Z Mode button is pressed to enter the impedance limits screen Fig. 12. First I pressed “Measure” to make sure the measured curve is the actual impedance.

I have chosen to modify the pre-defined limits of +/-20% around the Fs impedance peak to allow for a natural variation in production. That was done the easy way by simply clicking the white squares and dragging the limits with the mouse. Note that the limits are automatically updated in both windows when dragging.

Since the impedance measurement is purely electrical, the range and time window is already defined when the “Auto delay” is active. So there is no need to open the “Set Window”. When the limits are OK click Save.

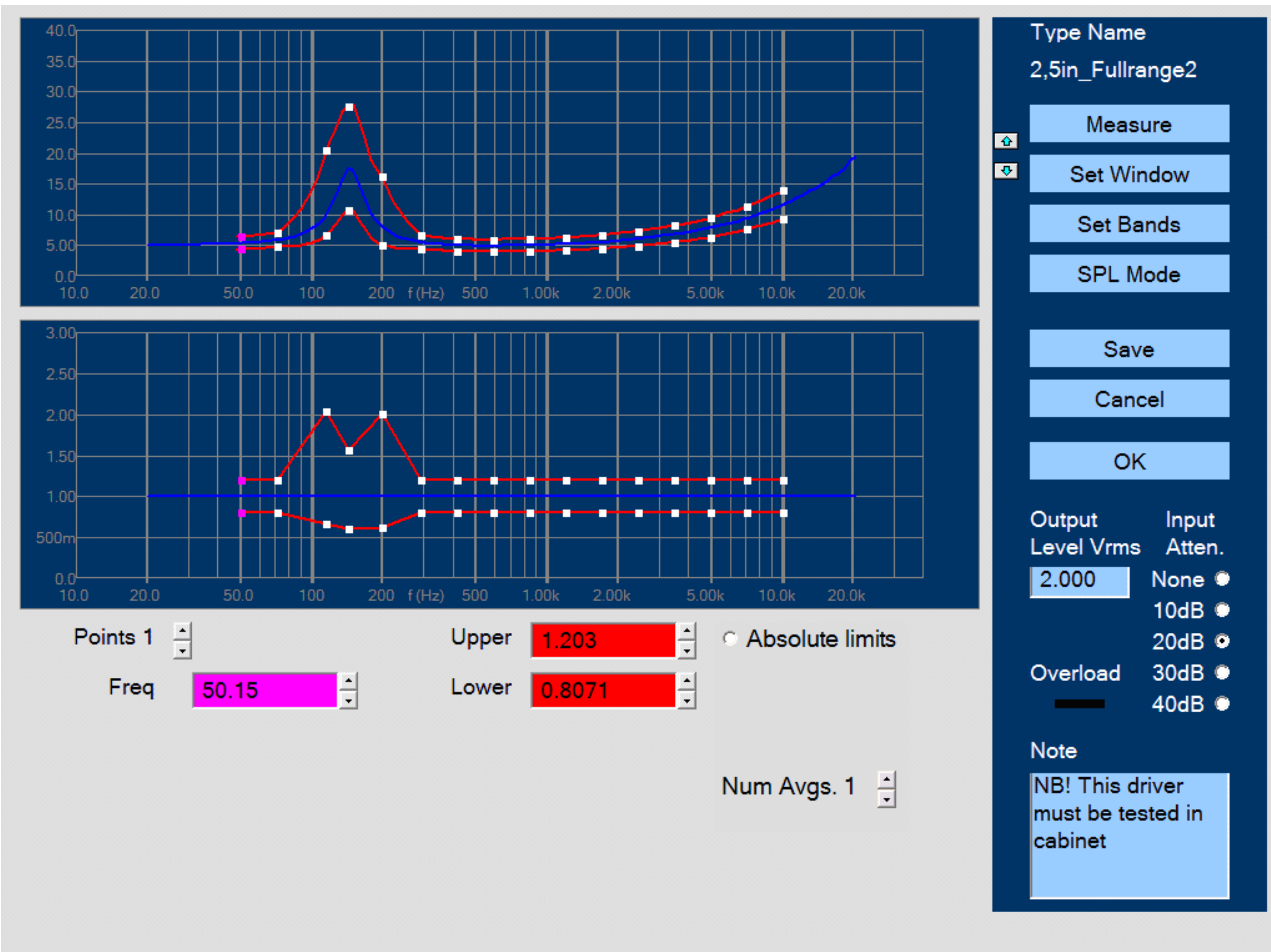


Figure 12 – Impedance Limits, adjusted

In this window the engineer can also add a note, which will be displayed for the tester. A typical note is shown in Fig.12:

[NB! This driver must be tested in cabinet)

Set Sweep Parameters

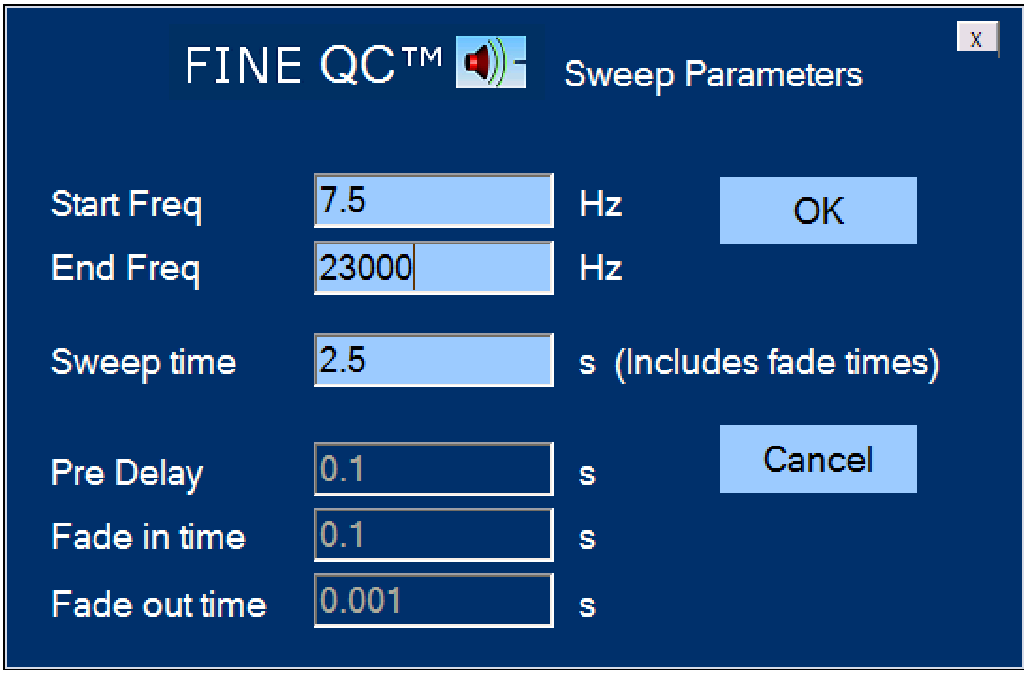


Figure 13 - Sweep Settings

The sweep can be set as shown. The sweep should be about ½-1 octave longer than the frequency range to be measured, but the End frequency should not exceed 23 kHz to avoid false triggering. The max sweep time is 2.5 seconds. A sweep time of 0.1s is possible, but minimum 0.5 second is recommended, because short sweeps are more noise sensitive.

Note: the sweep changes are not fully active until save and Edit again or Run

4.6 Test in a normal room

Fig. 14 shows the response of a satellite speaker tested at 1m in a normal room with the microphone in line with the tweeter, which is the normal listening axis. The tweeter of the speaker was about 82cm above the floor. Note that the low end response is limited around 300Hz. This is unfortunately not the true response, but the result of a poor measurement.

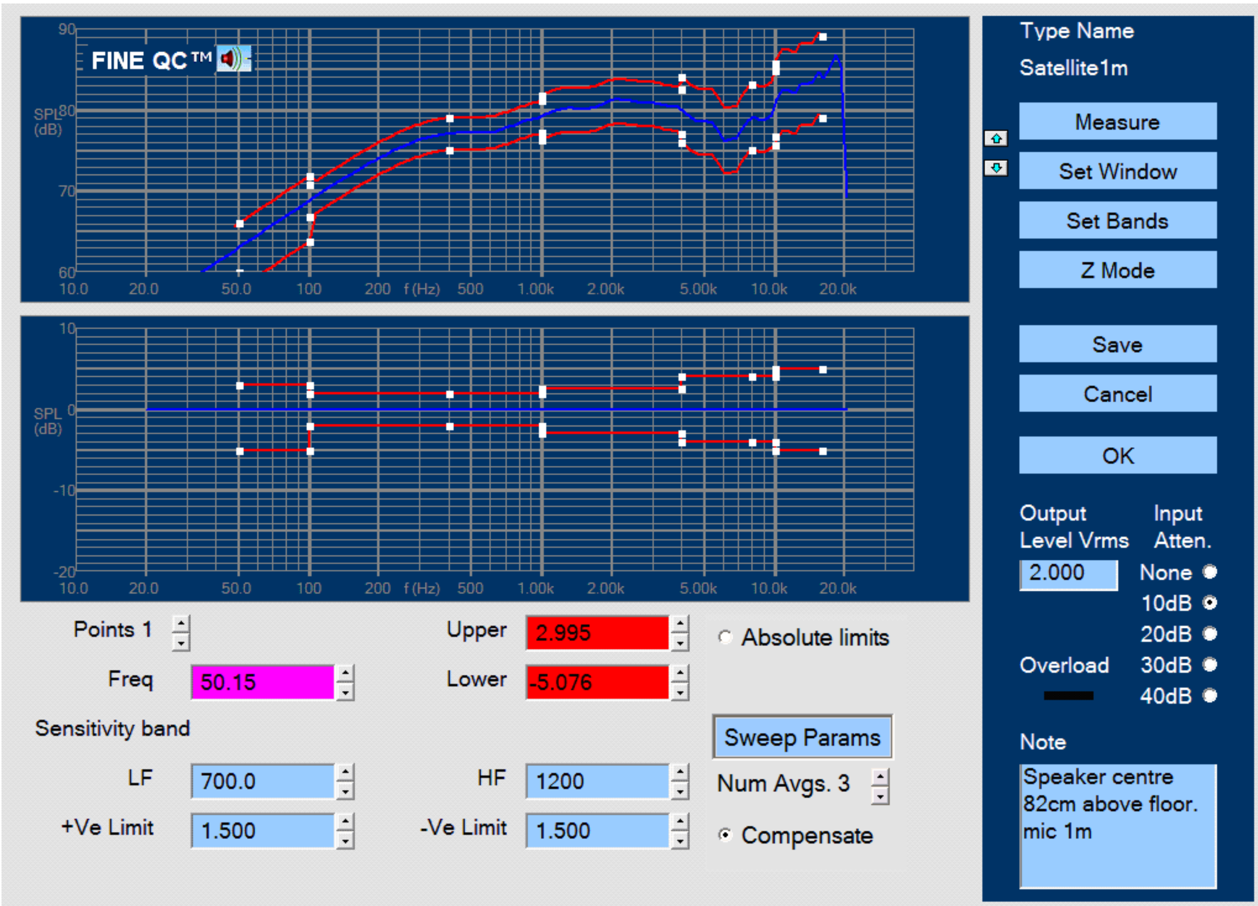


Figure 14 - Satellite Speaker tested in Normal Room at 1m

The Time domain impulse response of the satellite is shown in Fig. 14. The main impulse is arriving after approximately 3mS corresponding to 1m (the speed of sound is ~343m/s or 0.343m/mS).

However you can see another strong impulse arriving already about 2.5ms after the main impulse. That is the reflection from the floor, which is only 82cm below the speaker and microphone, Fig. 15.

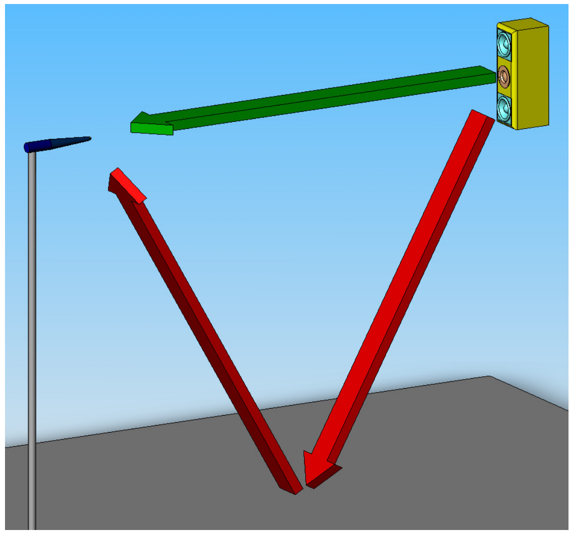


Figure 15 - Satellite close to floor, Red is Reflection

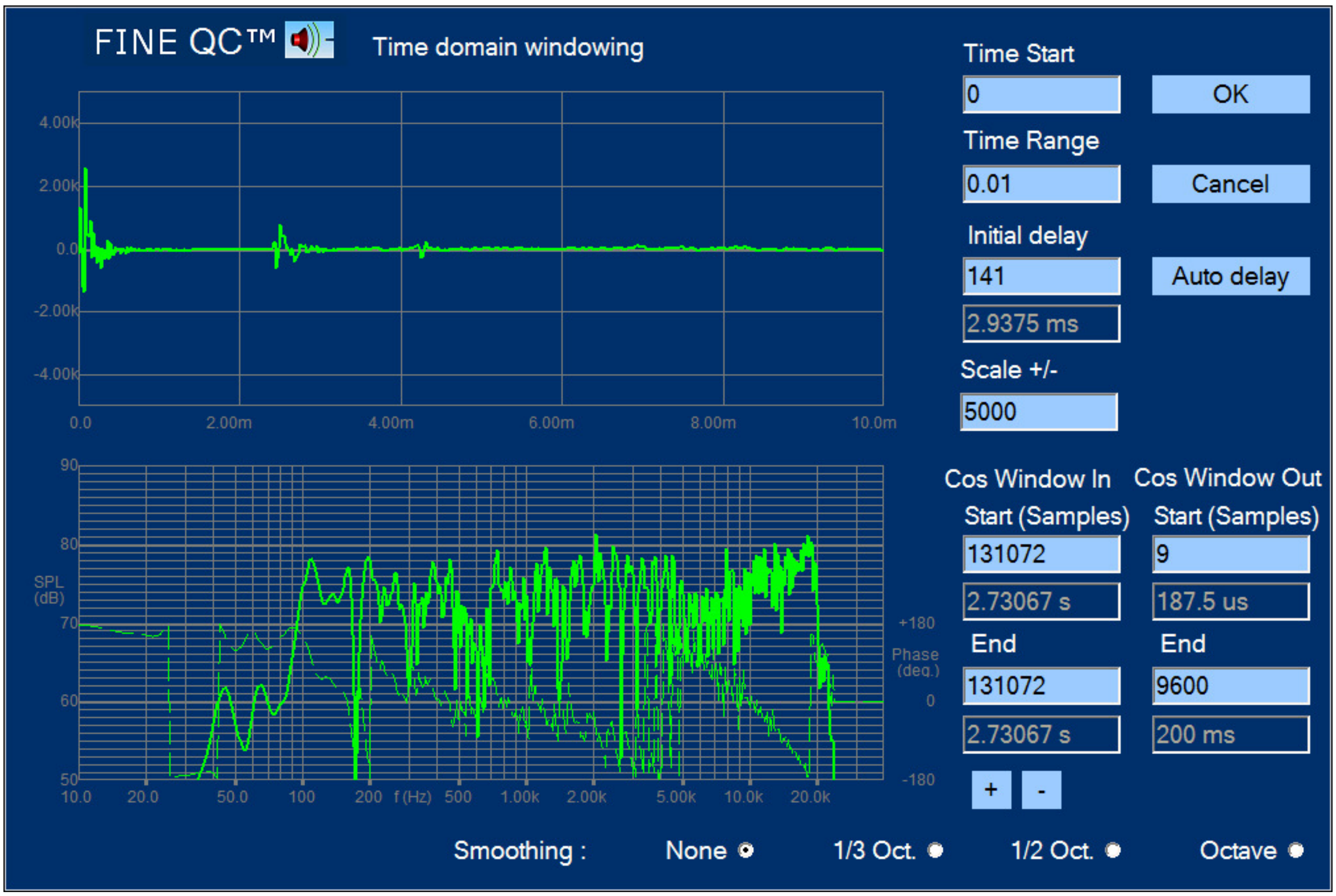


Figure 16 - Time Response of Satellite at 1m microphone distance and 82cm above floor

The short time between the two impulses is the reason for the poor low frequency response (Fig. 14) Using the 1/f ratio the 2.5ms will only allow 400 Hz as the lowest frequency. Since we are using a cosine window we may extend that to 2.7-3mS, but that does not really help.

Fig. 15 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 that: Move the microphone closer to the speaker and/or move both speaker and microphone further away from the floor (or other surfaces).

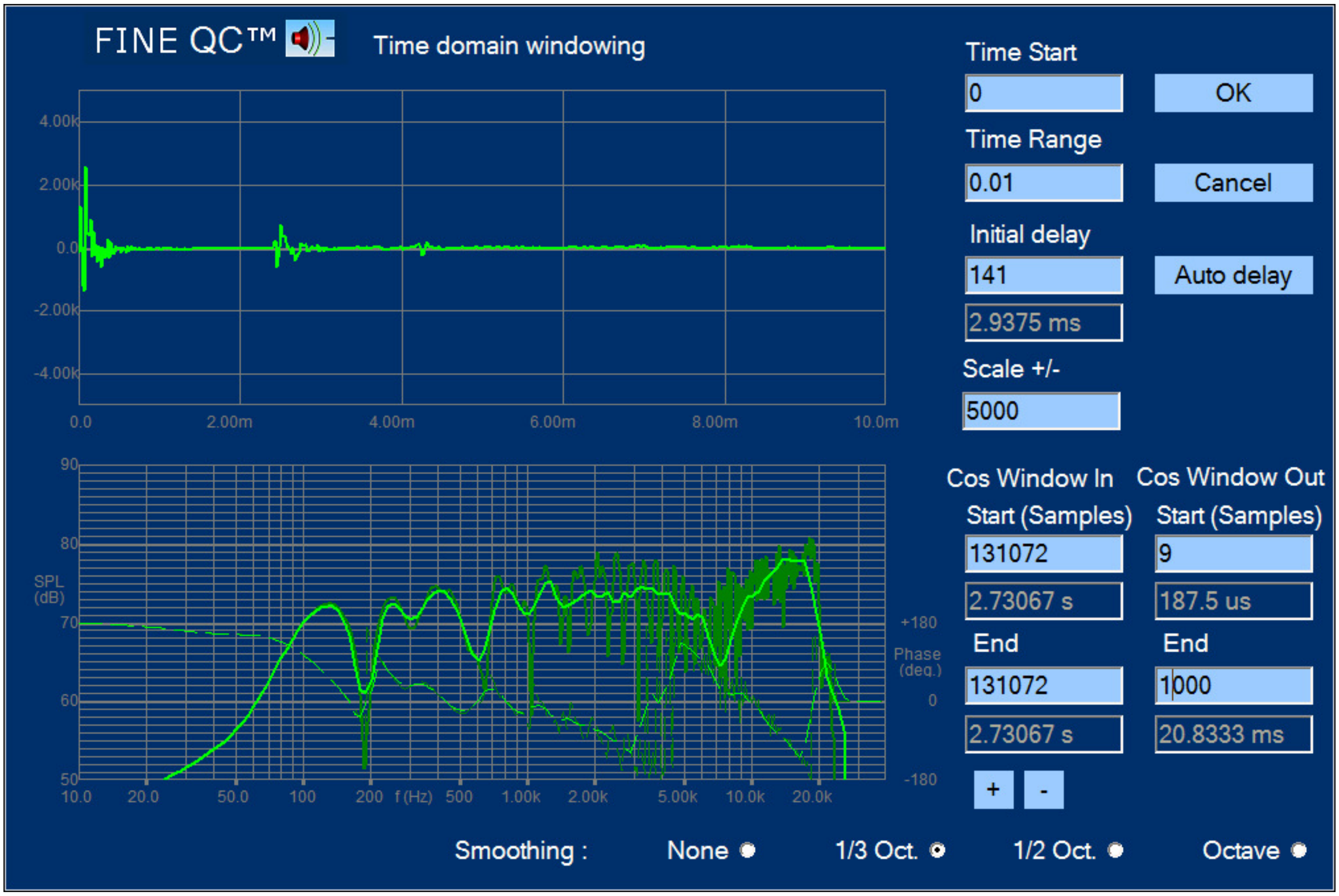


Figure 17 – 1/3 Octave Smoothing of Frequency Response (Unsmoothed in dark green)

The frequency response in Fig. 16 is quite ragged due to the reflections. You can apply smoothing to view the response anyway as in Fig. 17, where 1/3 Octave smoothing was chosen. Comparing with the unsmoothed response which is shown behind (dark green), you can still see the dips in the response at low frequencies, but it is possible to use such a response for QC-testing.

The dips and peaks can be leveled out by further smoothing up to 1/1 Octave, but that is not recommended, because you also hide most of the response problems you want to measure in the QC-test.

Note:
You can use the smoothing feature for room measurements, by including all reflections. Set the Cos End to max (~9600) and select for example 1/3 Octave smoothing. This way you can measure the actual response at the listening position or anywhere in the room, to estimate the room modes. This is particularly useful for positioning of loudspeakers and/or applying damping or room equalization (EQ)

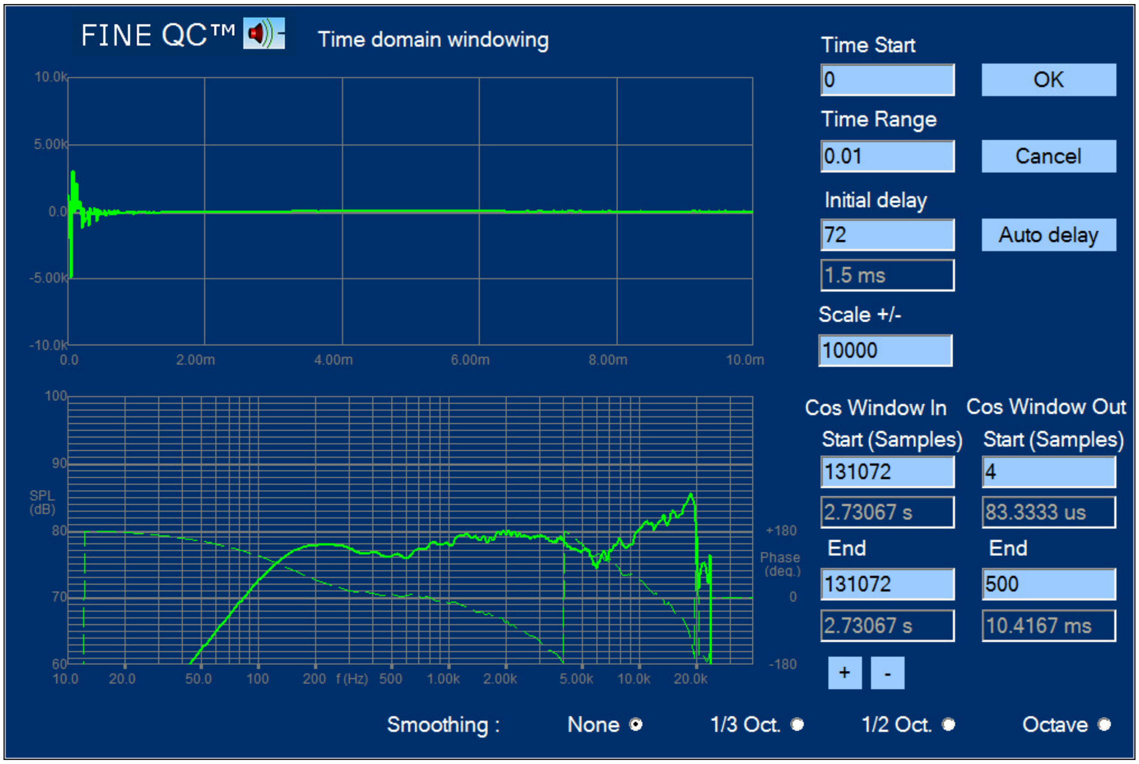


Figure 18 - Satellite and microphone moved up to 154cm above floor, microphone distance 0.5m

The optimal solution however, is shown in Fig. 18, where both the microphone and speaker has been moved up to 154cm from the floor. Because the speaker is quite small it was safe to adjust the microphone distance to 0.5m (the distance to the other walls and ceiling is equal to or larger than 154cm). Due to these precautions, this time we get the reflections much later and can use a window of 10.4ms. Therefore we get the real low frequency response of the satellite, starting from approximately 150 Hz.

4.7 Subwoofer in Near Field

The final example is an 8 inch subwoofer which I choose to measure without any baffle or cabinet, using the Near Field Measurement method with the microphone very close to the cone. This method is quite powerful and will show the full low end response as if the driver was placed in a very large baffle (~infinite baffle). The only drawback is that the response is only valid at low frequencies (below break-up). The -1dB limit is around 500 Hz for an 8 inch woofer so the LF response and sensitivity can be measured well, and the subwoofer roll off can be estimated.

The time domain response is shown as Fig. 19, and no reflections are observed. In fact I have used the default 200ms to enable measurement down below 20Hz. The final test screen for the 8 inch subwoofer is shown in Fig. 20. The limits are tight from 100-500 Hz which is the piston range before break-up. The sensitivity is measured as an average from 100-400Hz.

FINE QC 2 has a Subwoofer2.5s-5k.fts setup file, which is limited to 5k and has FINEBuzz enabled. This setup is recommended for subwoofer near-field testing.

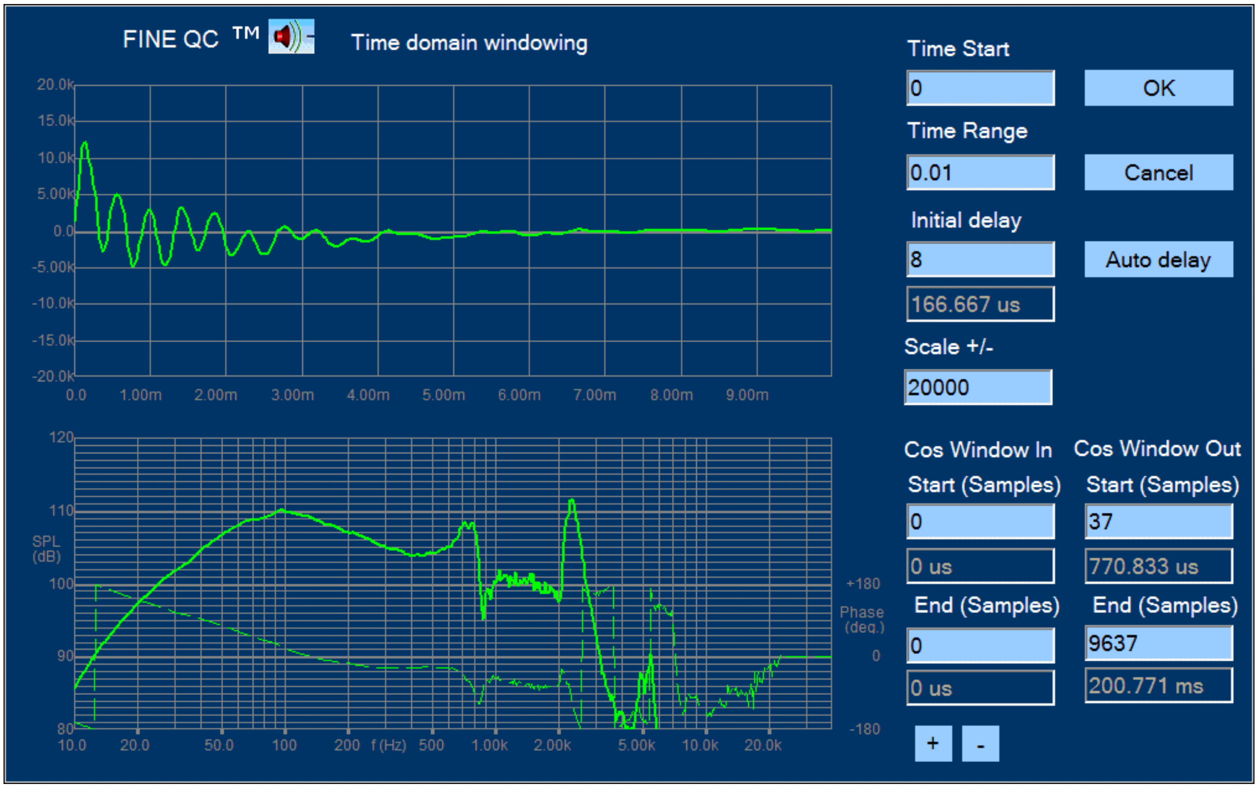


Figure 19 - 8 inch Woofer measured in NearField

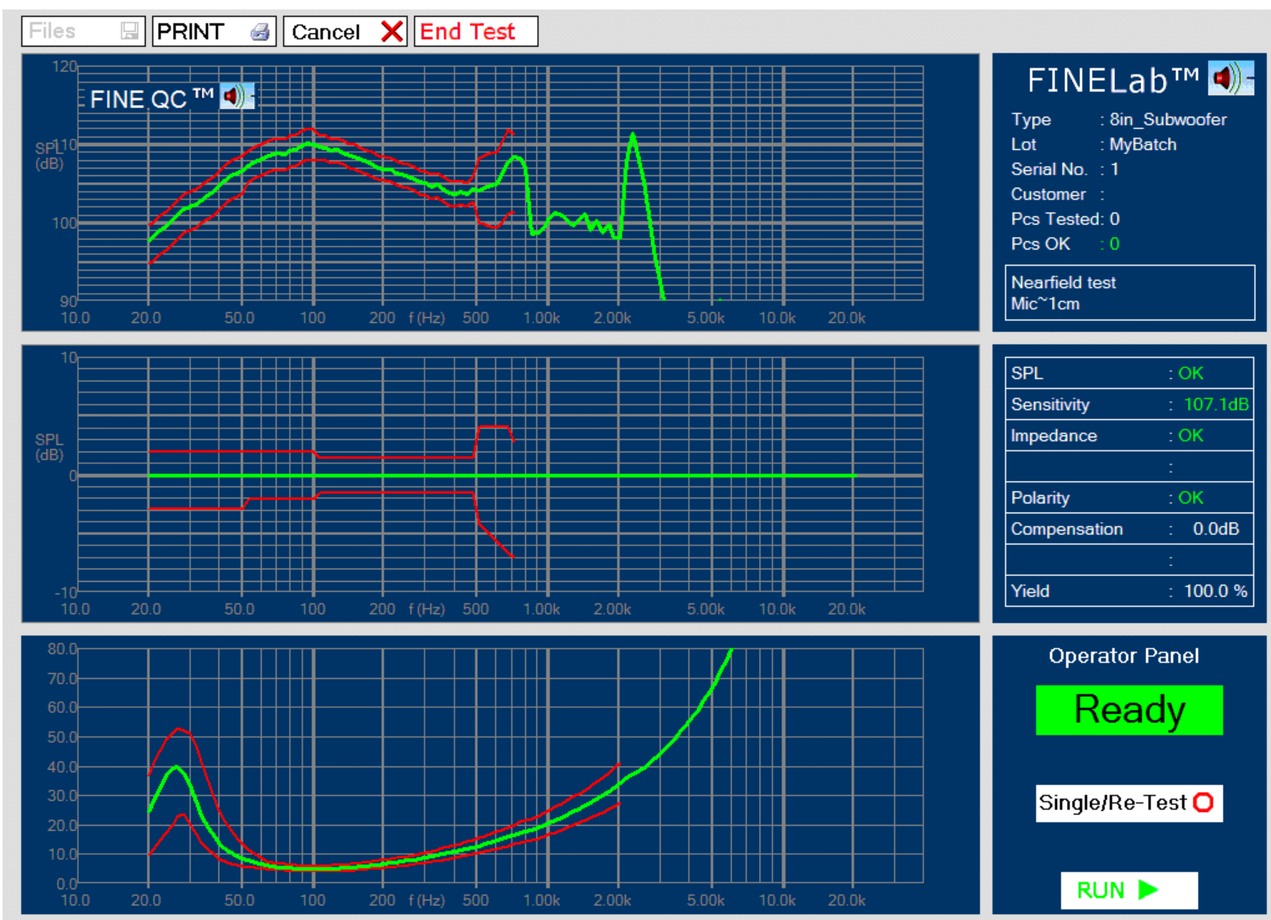


Figure 20 - 8 inch Subwoofer Test in NearField

4.8 FINEBuzz – Rub & Buzz Detection

On the production line it is necessary to check all units for bad sound. The frequency response and impedance of a driver or system may well be within limits but can unfortunately still sound bad, for example due to a rubbing voice coil or a rattle from the cabinet.

The new FINEBuzz detection method is based on the latest Danish research on hearing mechanisms, and uses a completely new algorithm to find the annoying sounds, which cannot be detected with conventional methods like THD, high harmonics or IM distortion. The new method is extremely sensitive and can detect even the smallest buzzing tinsel in a tweeter.

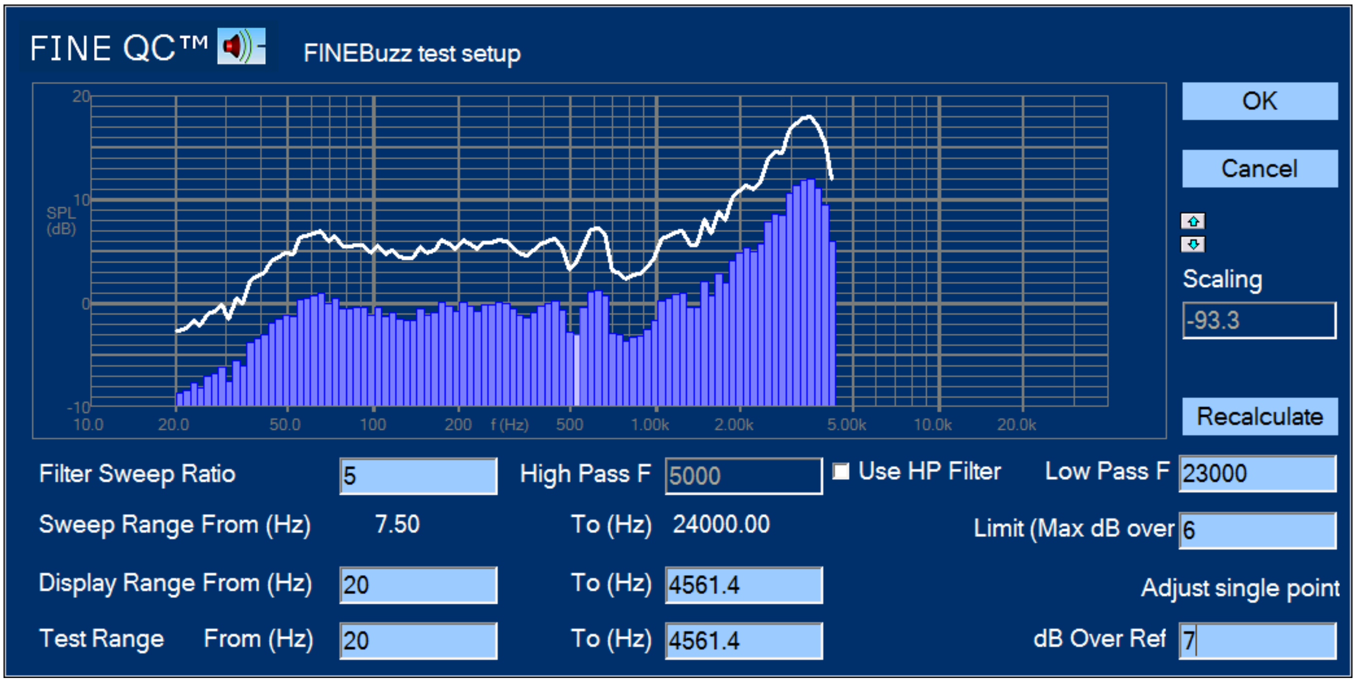


Figure 21 - FINEBuzz Setup Screen

Press “Setup R&B Test” to get Fig. 21. The rub and buzz is normally concentrated at low frequencies where the driver excursion is high. These annoying sounds, contains high harmonics where the ear is most sensitive, especially around 1- 3 kHz. FINEBuzz has a sweeping filter to pick up the rub and buzz, which is normally set to a ratio of 5 (5x test frequency) or higher. Ratio 5 is generally recommended for most drivers and systems and 5-8 for tweeters.

Assuming silent conditions (See note below) the white limit line is set only 6 dB over the acceptable rub and buzz +noise level (blue columns). Most display settings are automatically set, so you only need to set the sweep ratio (here 5) and the Limit (Max dB over) and then press “Recalculate”.

You can adjust the white limit line for each blue column individually by clicking it and adjusting the level with the mouse wheel, or entering the number in the field at lower right. In Fig. 21 the column at 5.3 kHz was selected (light blue) and adjusted up to 7 dB.

Note: Setting the FINEBuzz limit less than the default 10 dB requires very silent test conditions! Use a separate test chamber and avoid noise sources like air guns, fans and bumping carts and pallets. In this case you can also select “Use HP Filter” normally set at 5000 Hz. This will enhance the high harmonics above 5000 Hz, to make the Rub & Buzz detection most sensitive. Use this function with care and use a closed test box and avoid high frequency noises!

Fig. 22 shows a woofer which failed due to a rubbing voice coil, indicated by the red columns where the rub and buzz is above the white maximum line.

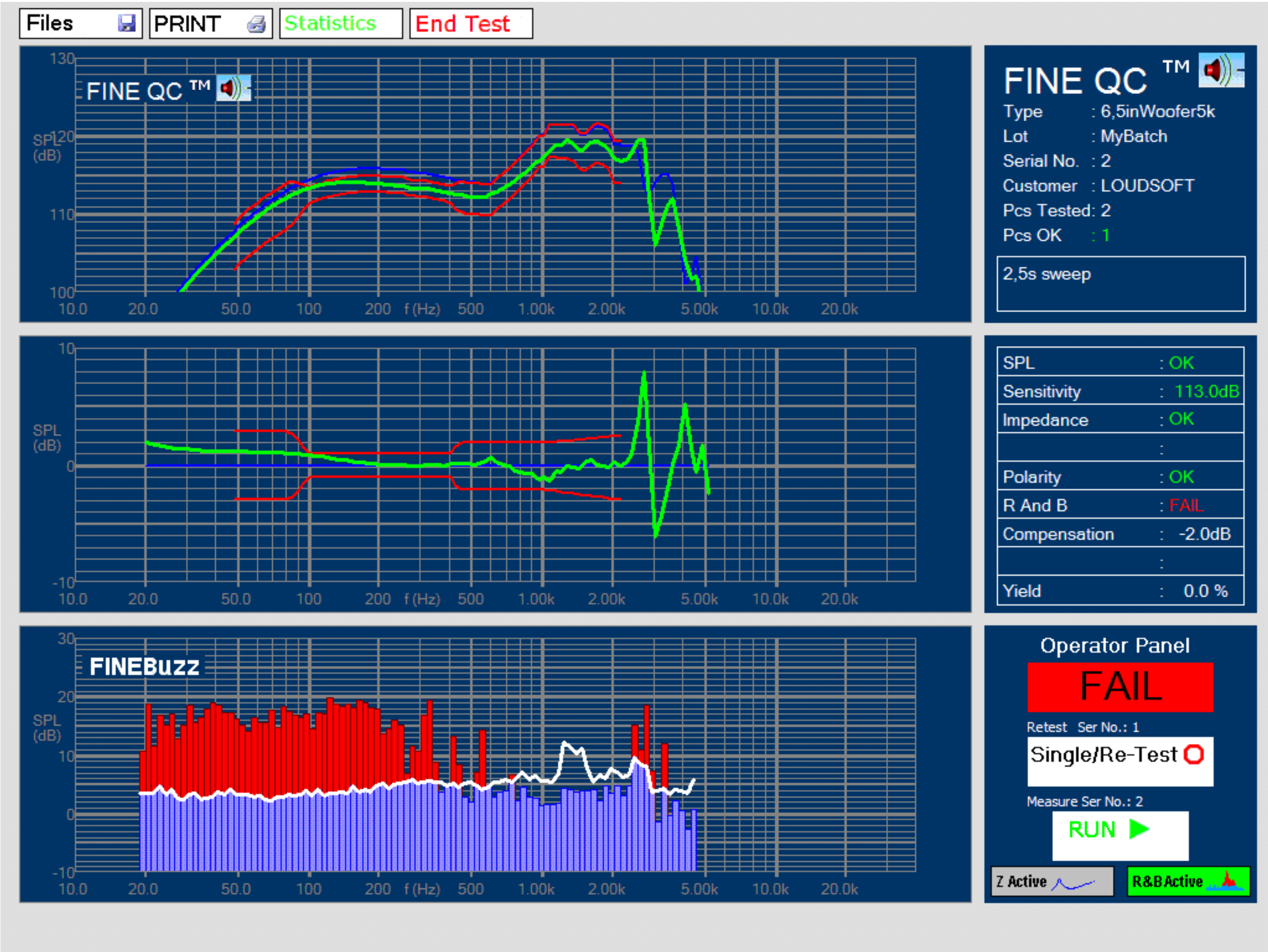


Figure 22 – 6.5inch Woofer with rubbing Voice Coil found with FINEBuzz

It is possible to test Rub & Buzz in 1 second, but I recommend using the longer 2.5s setup, because a fast sweep may not contain enough energy to find very small resonances. In the above example, I used the 20-5kHz sweep, which further concentrates the energy in this band.

Likewise the tweeter2.5s_200-20k setup is ideal for finding small resonances like buzzing tinsels in tweeters. Fig. 23 shows a tweeter with very subtle buzzing tinsel at 900Hz.

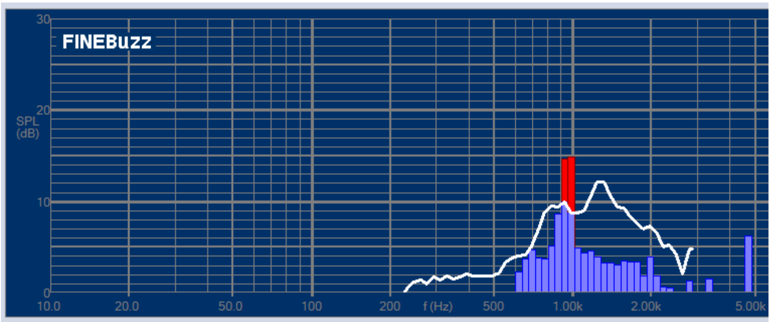


Figure 23 - Tweeter with very subtle buzzing tinsel at 900Hz

4.9 Thiele / Small (TS) Parameters

Finally I want to measure the TS parameters of a 10 inch subwoofer. Pressing “Edit TS Test” from a previously defined QC Test will show the screen in Fig 24:

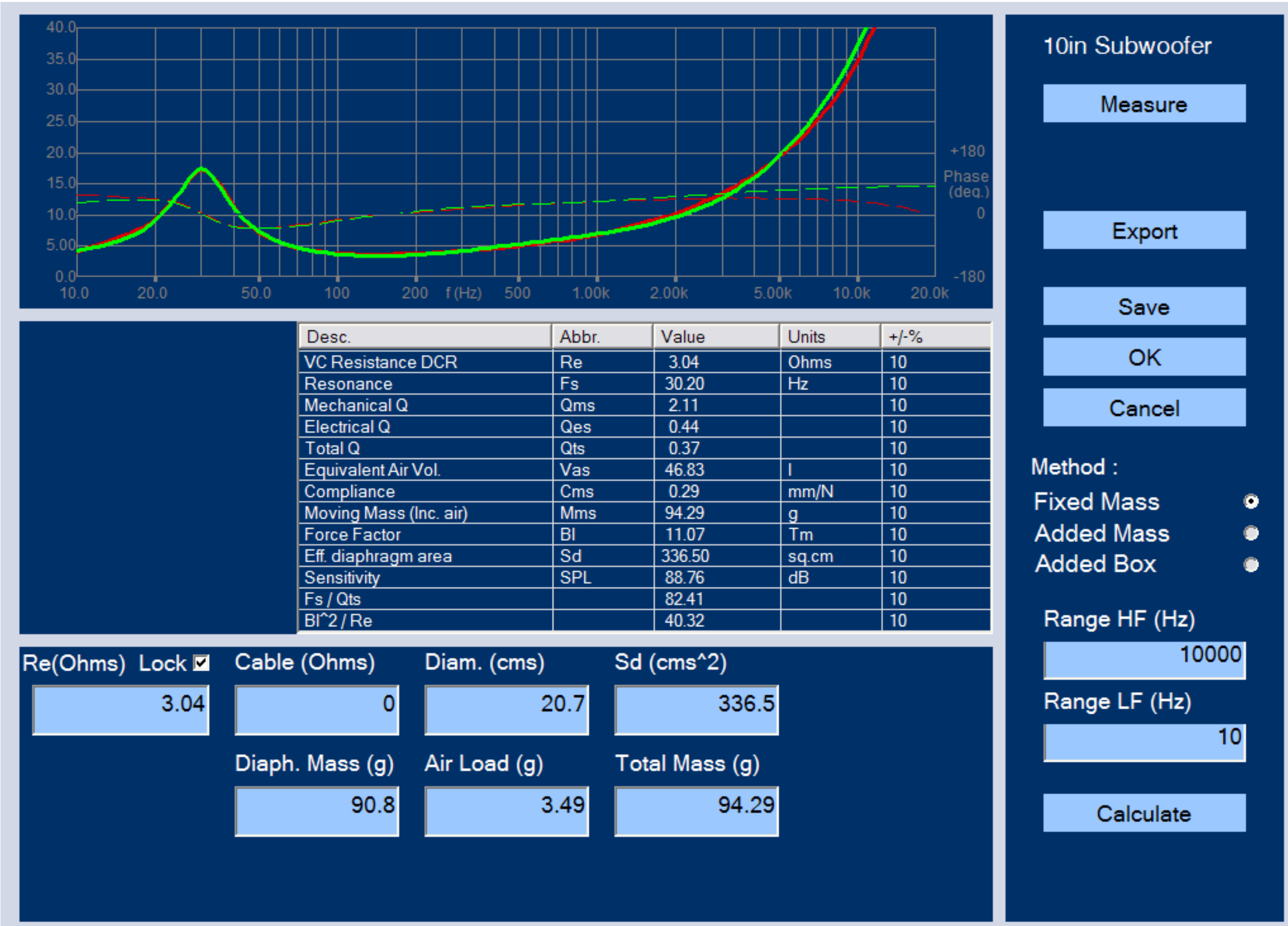


Figure 24 - TS Parameters of 10inch Subwoofer

First I need to input the cone area Sd and Re. I choose to input the effective diameter of 20.7cm (centre of surround) and Sd will automatically be calculated. FINELab QC can estimate Re from the impedance curve, but in order to get the best accuracy I have measured Re=3.04 ohms with a precise Multimeter (DVM). That value is fixed by lock [v]. Now I press “Measure” to get the impedance curve (green).

I could choose the standard Added Mass or Added Box method, but the Fixed Mass option is much more accurate. However I must cut a typical woofer, so I can weigh the cone + Voice Coil + half surround + half spider (including dust cap and glue etc.). This mass (Md) is entered as Diaphragm Mass which causes the air load mass (Mair) to be calculated, Md + Mair=Mms.

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

Qts is calculated as 0.37 with Fs= 30.2Hz, but we also get the sensitivity SPL= 88.76 dB/2.83V. We must accept a large variation in Qts, because it depends strongly on Fs. Therefore FINELab QC also calculates the ratios Fs/Qts=82.41 and BL^2/Re=40.32. These ratios are more important for controlling the bass response than Qts and Fs and other parameters.

4.10 Typical Test Setup Procedure

In this chapter I will summarise the necessary steps in a typical test setup:

1. Log in as Engineer
2. Select a test specification as close as possible to the kind of speaker you want to test.
3. Select “New Based On” and input a name for the test
4. Specify a suitable amplifier Output Level (Vrms). Choose a level so the driver will move close to half of Xmax for woofers and around 1W or less for tweeters.
5. Press “Measure” to do the first sweep test with Input Attenuation set to: None
6. If you see the red “Overload” light, then select one step lower Input Attenuation and “Measure” again. Repeat until there is no overload
7. Press “Set Test Window” to enter the Time Domain Window
    a. Check that the large pulse is close to 0mS. That is normally done by the “Auto Delay”. If not adjust the “Initial Delay”. (Auto Delay is by default 5%. This can be changed in Admin. A higher number will prevent FINELab from trigging on noise)
    b. The dashed curve is the acoustic phase response which will show less variation when the large pulse is close to 0mS
    c. Input a suitable number of samples in: (Cos Window Out / End (samples))
        i. If you are using an anechoic room or really well damped box use ~10-20mS
        ii. If you are using a normal room or standard test box use ~3-10mS
        iii. If you are measuring in the near field you may use the full 200mS.
    d. The idea is to choose a window which will pass the decaying pulse, but avoid the reflections which are arriving later (see for example Fig. 13)
    e. The window may extend to include some reflections in the end, because the cosine window will attenuate much towards the end
8. Select “Set Bands”
9. Choose a suitable SPL tolerance limit standard
    a. Modify the tolerances and bands if you know how much you need
    b. If you do not know how much to change the limits, then use the standard one and check the response statistics before considering changes to the limits
10. Press “Z mode” to enter the Impedance window
11. Press “Measure” to do a first impedance sweep test
12. Use the up/down arrow buttons to scale the impedance curve as necessary
13. You do not need to press “Set Window” to enter the Time Domain Window, but you can.
14. Select “Set Bands” if you want to choose standard limits, OR
15. Click and drag the white squares to suitable production limits.
16. Press “Save” when you are satisfied with the limits
17. You should now run the test file you have created with a small number of units (Test Batch) to verify your settings and limits. Press “End Test” when done and the statistics Display will automatically appear showing all the responses with your limits. You can choose to display good/rejected SPL, Impedance, Sensitivity or Polarity.
18. Select “Review Old Data” and view the Pre-Production of your Test Batch “Pre-Prod”
19. The Pre-Prod window highlights the Golden Reference (You can de-select non-typical curves).
20. Press “Edit that test” if you want to change the limits according to the statistical results

5. LOUDSOFT MICROPHONE FL1 DATA SHEET

The LOUDSOFT FL1 is a high quality microphone made in Denmark. The gold plated diaphragm and a double-vent protection system ensures the highest durability in production environments.

Directional characteristics:
Omni directional

Principle of operation:
Pressure

Cartridge type:
Pre-polarized condenser

Power supply:
48 V Phantom power

Frequency range, ±2 dB:
20 Hz – 20 kHz

Sensitivity, nominal, ± 3 dB:
6 mV/Pa; -44.5 dB re. 1 V/Pa

Equivalent noise level, A-weighted:
Typ. 26 dB(A) re. 20 µPa (max. 28 dB(A))

S/N ratio, re. 1 kHz at 1 Pa (94 dB SPL):
68 dB(A)

Total Harmonic Distortion (THD):
<1 % up to 123 dB SPL peak

Polarity:
Inward movement of diaphragm produces positive going voltage on pin 2

Cable drive capability:
Up to 300 m (984 ft)

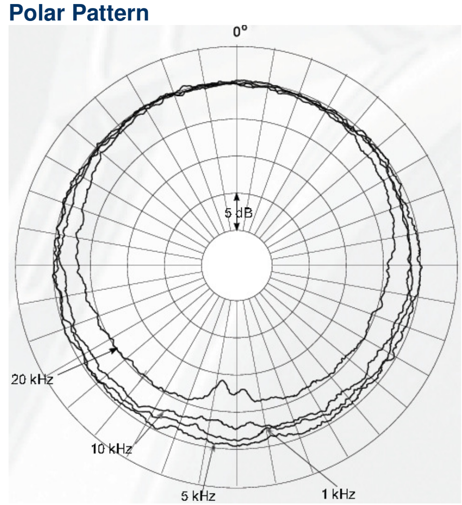
Connector:
3-pin XLR (Standard P48)

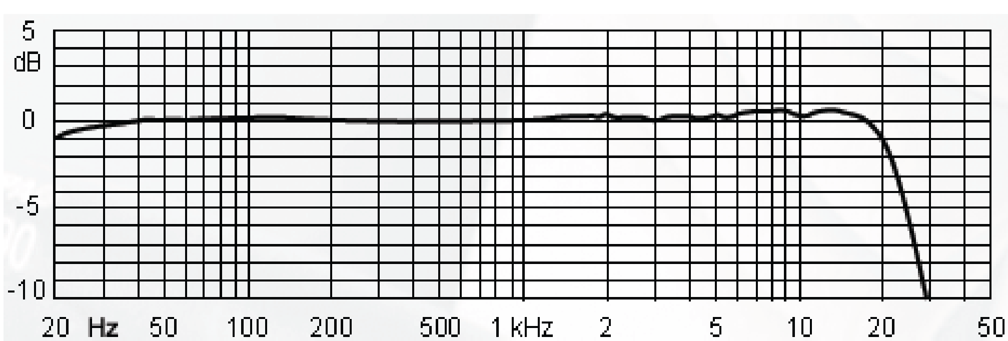
Dynamic range:
Typ. 97 dB

Max SPL, peak before clipping:
144 dB

Output impedance:
<40 Ohm

Polar Pattern





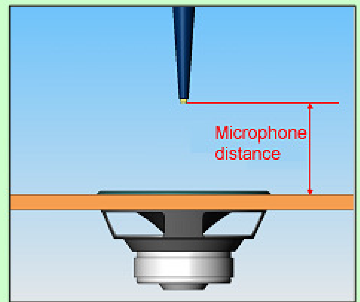
On-axis Frequency Response of LOUDSOFT microphone FL1 in free field.

6. Guide to QC Measuring Method

Guide to QC measuring method

FINELab Application Note

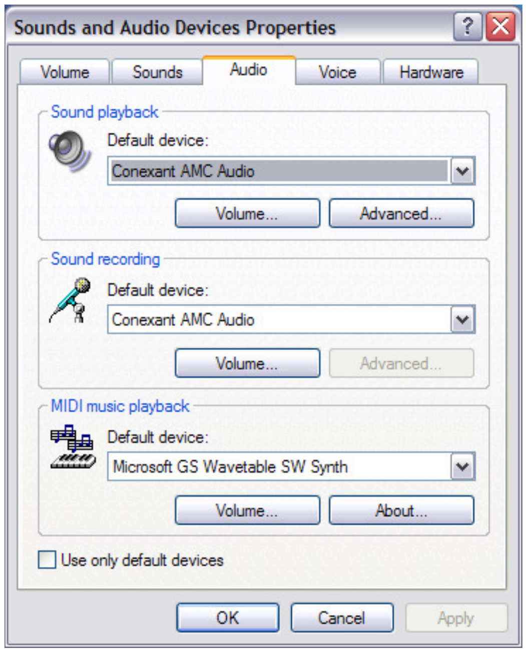
Choose the type in the first column and select a measuring method, and then the recommended microphone distance can be found from the table.



Recommended Microphone Distance				
Type	In Baffle	In Test Box	In Free air	
			Normal room	Anechoic chamber/Hall
Tweeter	10 cm	10 cm	N/A	N/A
Midrange/Woofer	10 cm	10 cm	N/A	N/A
Subwoofer	< 1 cm from center of dust cap	< 1 cm from center of dust cap	< 1 cm from center of dust cap	< 1 cm from center of dust cap
Small Loudspeaker System	N/A	20 cm (mic. in line with tweeter)	0.5 m (mic. in line with tweeter) or 1 m (if room is large)	2 m (mic. in line with tweeter)
Large Loudspeaker System	N/A	N/A	0.5 m (mic. in line with tweeter) or 1 m (if room is large)	2 m (mic. in line with tweeter)

7. Keep built-in Soundcard for Windows Sounds while using FINE QC

Go to Control Panel and select “Sounds and Audio Devices”. Open the “Audio” tab and select your standard built-in soundcard both for Sound Playback and Recording. This way all the standard Windows sounds will be played by the built-in soundcard using the PC- speakers and NOT through the tested speakers connected to FINELab.



Finally you should check that FINELab is still correctly connected using UA-25. Open the “Voice” tab. Check that UA-25 is selected both for Voice Playback and Recording as seen in the next picture.