# justMLS

**users guide**

rel 1.00

# 1    Introduction

justMLS is a measurement system especially aimed for those who wish to become more familiar with loudspeaker measurements. Loudspeaker measurement systems are generally quite expensive. justMLS does basic measurement of:

- Transfer function with phase, a popular rewriting is perhaps sound pressure level. It is however possible to measure e.g the transfer function of an amplifier also.
- Impedance with phase measure.

**Please note that justMLS is NOT freeware or shareware or any other no cost alternative. It is however free for all registered users of LspCAD standard or professional versions.**

For info about LspCAD box and crossover modelling software go to
http://www.ijdata.com

# 2    Technical data

| | |
|---|---|
| MLS size: | 32764,16384,8192 and 4096 |
| FFT size: | 8192, 1024 and 256 point, used in different frequency bands |
| Sample rates: | 96000, 88200, 64000, 48000, 44100, 32000, 22050, 16000, 11025, 8000 Hz and user selectable (Custom). |
| Smoothing: | None, $1/24^{th}$ ,$1/12^{th}$ , $1/6^{th}$ , $1/3^{rd}$,1/2 and 1 octave. |
| Window: | Half hanning |
| Internal representation: | 5Hz to 50000Hz, 1000 frequency points with logarithmic spacing. |

# 3 Necessary hardware

justMLS is designed to be as hardware free as possible, yet there are a few things that may be needed.

## 3.1 Soundcard

A decent soundcard is needed, key factors for success are low distorsion and low noise, especially for the mic input. There is however no need for an extreme soundcard, an SB16 card may be enough. One warning about portable PC's though, they tend to have very lousy soundcards !.

## 3.2 Input probes

Input probes may be needed to protect the input from e.g. the high output voltage of a power amplifier. Input probes are easy to put together. What is needed is the below items.

4        Alligator clips

2        47kΩ ¼W
         1% resistors

2        10kΩ ¼W
         1% resistors

+ some wire
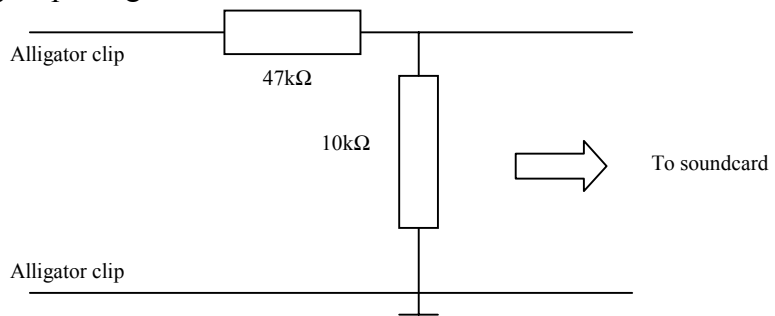
The simple schematic shows how a probe is built.

In cases with higher voltages it may be necessary to replace the 47kΩ resistor with a higher value (e.g 470kΩ). The attenuation ratio is not critical as justMLS anyway cannot measure absolute values.

## 3.3 Microphone

A decent microphone is needed for this purpose. Depending on the accuracy that one wants one can either put together a microphone capsule or buy a measurement microphone. A purchased microphone comes with a calibration data file, something that is necessary if you wish to measure frequency and phase accurately.

See below a list of companies that sell calibrated microphones (only one name for the moment):

```
Kim Girardin
Wadenhome Sound
1400 Homer Rd. Suite 2
Winona, MN, 55987, USA
507-454-8844
kmgrdn@luminet.net
```

## 3.4 Power amplifier

Pretty obvious, there is no need for a high tech power amplifier though. Any 20W power amp will do.

## 3.5 Mic preamplifier

Needed for two channel measurements unless you are among the lucky persons who own a soundcard with a stereo mic input..

# 4    The user interface

## 4.1    A first glance at the user interface

The justMLS user interface is designed with simplicity as primary goal. Still it should be advanced to provide with the measurement capability of professional systems at a fraction of the cost. The justMLS window contains two tabbed controls, one that allows the user to select between time (impulse) or frequency response. The other tabbed control contains the "main menu" and it allows adjustment of sampling rate, and merge/add operations among other things.
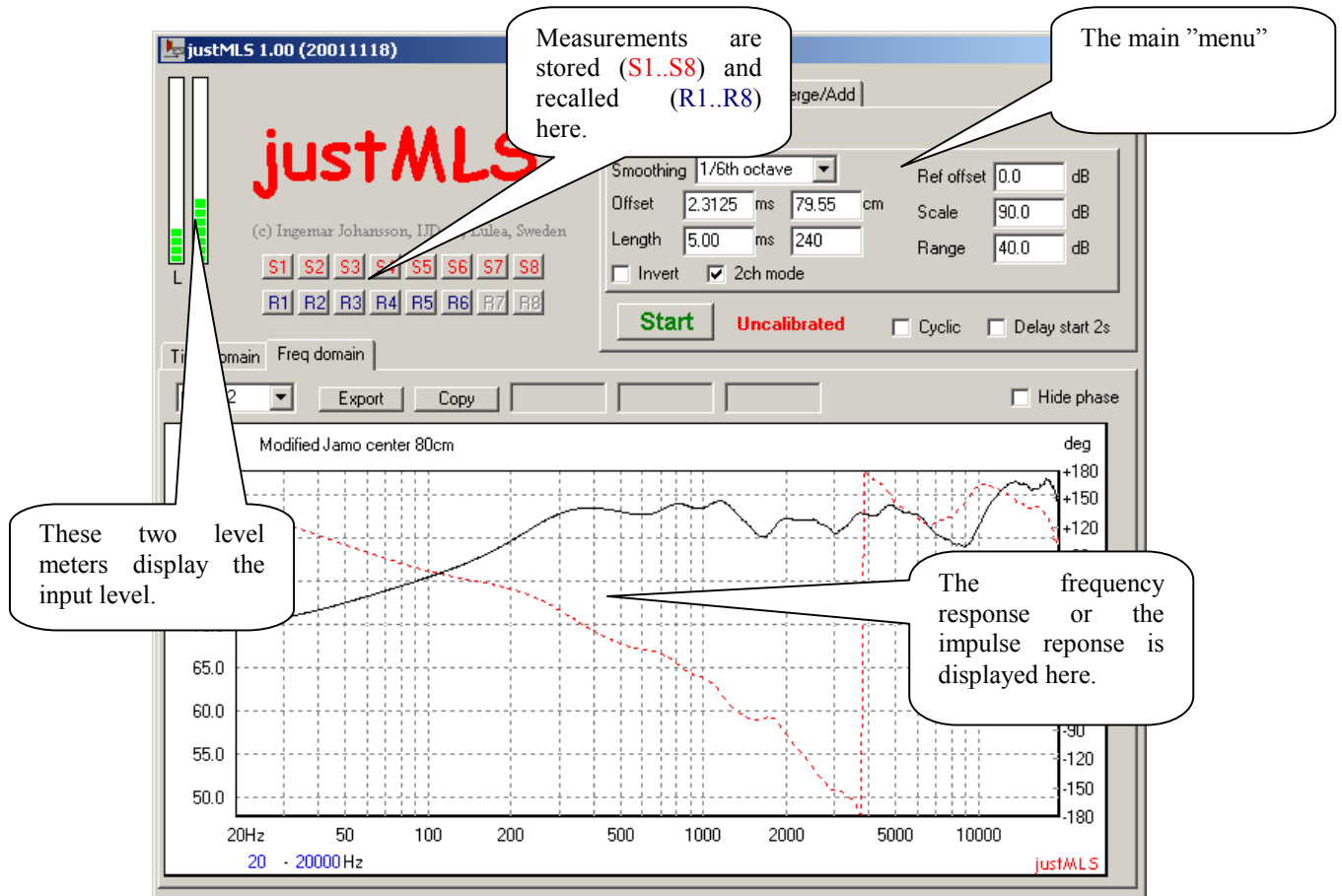


Up to eight different measurements can be stored in memory. The latest done measurement visible in the graphs is referred to as the current measurement. Two vertical bars display the input level.

The controls contain numerical values can be modified either as normal edit controls. It is also possible to modify the value by means of the Up/Down arrow keys on the keyboard. Holding down the Ctrl key while pressing down the Up/Down arrow keys will increase the step size.

### 4.2 Time / frequency domain graph

The time frequency domain displays the impulse response (**Time domain** tab) or frequency response (**Freq domain** tab). The figure below shows the frequency domain graph. In this case the frequency response of a small two-way centerchannel loudspeaker is displayed.
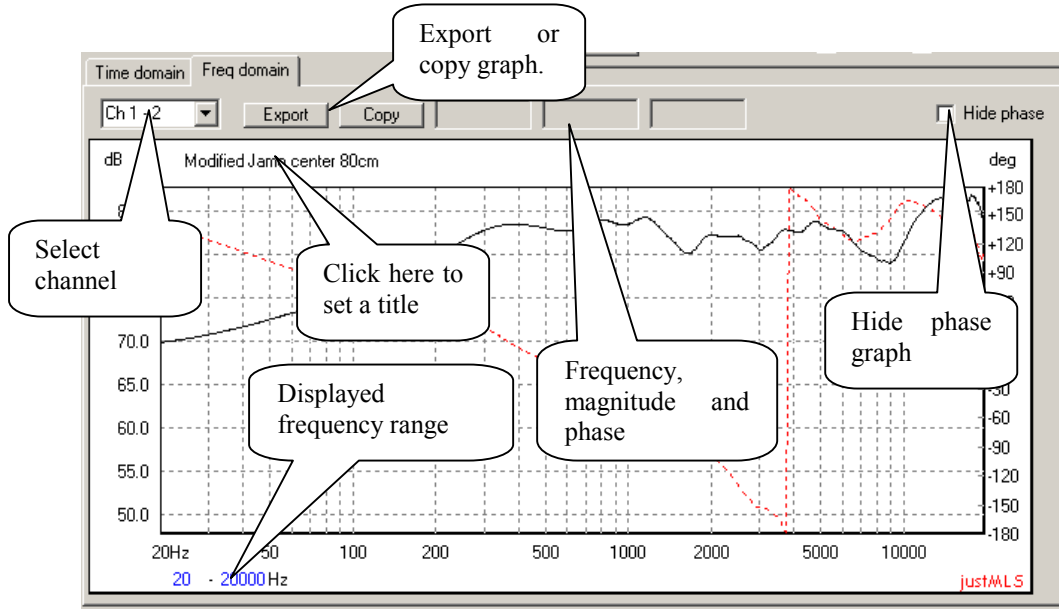


The display frequency range can be set by clicking on either the lower or the upper limit in the graph and typing in the new value, as an alternative one can also use the KEY-UP and KEY-DOWN keys on the keyboard to modify the values. It is possible to hide the reading of the phase by checking the **Hide phase** checkbox. Graphs are **Export**ed in ASCII text file format with one frequency point per line. The first column is the frequency reading, the second column is the magnitude (ohms or dB) and the third column is the phase (degrees). A **Copy** button makes is possible to copy the graph to the windows clipboard. For two channel measurements it is possible to select to view the frequency response of **Ch 1** (left), **Ch 2** (right) or **Ch 1-2**, in the latter case the difference between Ch1 and Ch2 is shown. The Time domain tab displays the impulse response, for two channel or impedance measurements the impulse response of both the channel 1 and 2 is displayed.

### *4.3     The main menu*

The main menu contains three tabs.

- Settings
  The soundcard settings as well as level adjustment and calibration is done here.
- Measure!
  The measurement menu, lets you measure transfer functions (**SPL**) or impedance (**Z**).
- Merge/Add
  A few simple functions that let you perform merge/add operations with your measurements.
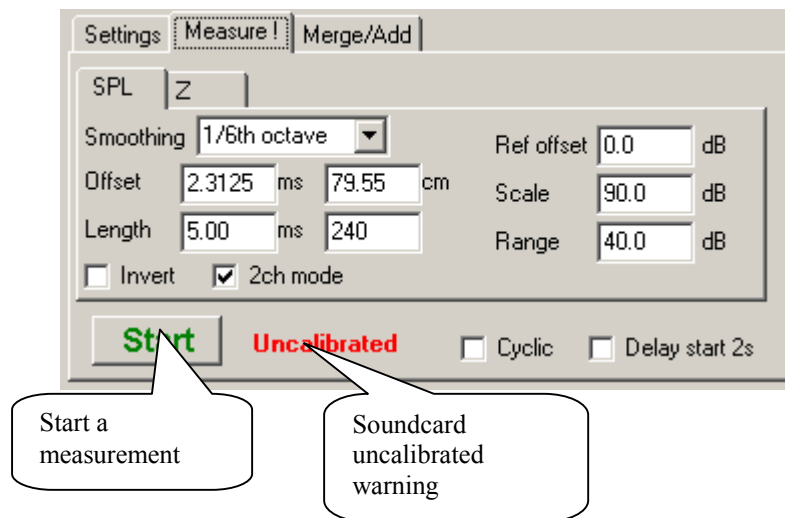


### 4.3.1  Measure!

The measure menu contains two submenus, one for SPL and one for Z measurements. A measurement is done when the **Start** button is pressed.
A warning flag **Uncalibrated** lights up if the soundcard is not calibrated.
For repeated measurements the **Cyclic** checkbox is checked, if the **Delay start 2s** checkbox is checked the start of the measurement is delayed by app. 2s.

**SPL measurements**
The SPL submenu contains the controls that are used for measurements of transfer function, or as one would say in the area of loudspeaker building, sound pressure level (SPL).
**Invert**: If the invert checkbox is checked the phase of the measurement is shifted 180 degrees. If may sometimes be necessary to invert the phase, a good rule of thumb is that (for a single loudspeaker driver, and with the analysis window offset correctly) the phase should pass through a zero value somewhere in the passband (the operating frequency range) of the loudspeaker driver.
**2ch mode**: SPL measurements in justMLS can be done either in single channel mode or in two channel mode. In single channel mode only the left channel (Ch 1) input is used in the analysis. This is simple to use, just hook up the min to the mic input of your soundcard and measure. In two-channel mode the mic is connected via a mic preamplifier to the left line input (Ch 1), the right channel (Ch 2) is connected via an input probe to e.g. the electrical terminals of the loudspeaker under test. The neat idea here is that one can subtract the analysis of Ch 2 from the analysis of Ch 1 (select **Ch 1-2** in the Time/frequency domain graph) and thus get a measurement that in it self compensates for the frequency response anomalies in soundcard and power amplifier.

**Smoothing**: Smoothing of measured data is often desired as it makes reading of the graphs a lot easier. The smoothing can be selected from no smoothing (None) to 1 octave smoothing. A good alternative is to use $1/6^{th}$ or $1/12^{th}$ octave smoothing.

**Offset**: The offset of the analysis window is given both in milliseconds (ms) and in centimetres (cm). In 2ch mode the selection of the offset only affects Ch 1. The meaning of this is of course to compensate for the time of flight between the loudspeaker unit and the mic. It may be a good idea too have a look at the time domain response and see that the impulse response starts close to the left of the window, it is however important that the beginning of the impulse response is not chopped off.

Note that the internal representation of the offset is on samples, this you will notice if you enter the value 30cm, the value will then be changed to 29.64cm.

**Length**: Sets the maximum length of the analysis window, this value is given either in samples or in milliseconds.

**Ref. offset**: A non-zero value here will cause the graph to move up or down, this may be useful when doing measurements of e.g. very resonant drivers.

**Scale**: The graph is scaled up or down, good to use if you want to set the reference SPL to e.g. 90dB.

**Range**: The display range in dB.

**Impedance measurements**:

For impedance measurements both Ch 1 and Ch2 are used. The resistor value that is used in the measurements is entered in the **Resistor** control. The range of visible impedance values can be select from two combo boxes, the maximum range is 0.1 to 1000 ohm. $1/24^{th}$ octave smoothing can be selected to smooth out noisy measurements (check the **Smooth 1/24 octave** checkbox for this function).
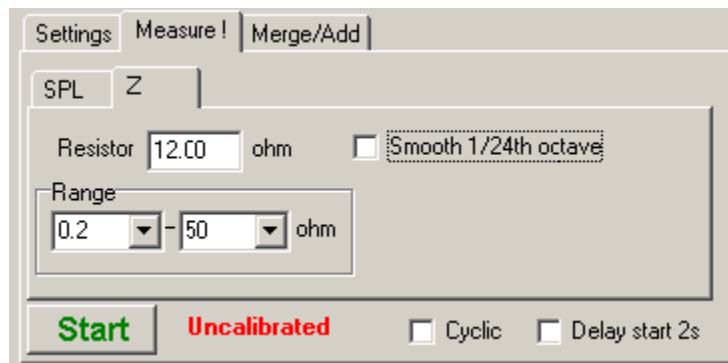
### 4.3.2  Settings

The settings menu is pretty easy and intuitive to use. Here one set the e.g the desired input and output device and calibrate the soundcard.

With the **Test levels** button one can verify that the input level is not to high or too low.

The soundcard is calibrated when the **Soundcard cal.** button is pressed. See separate section about calibration for detailed information.
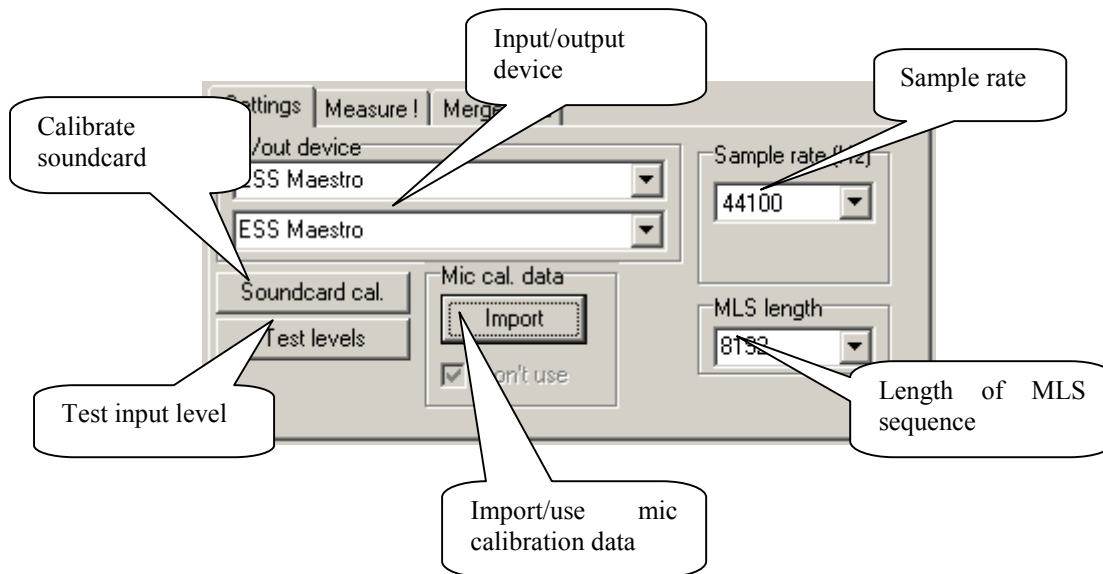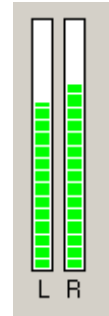
If you have a mic with calibration data you can load the calibration data file with click on the **Import** button, mic calibration data is disabled if the **Don't use** checkbox is checked.

### 4.3.3  Merge/Add

The Merge/Add menu functions help you to merge e.g. near field low frequency measurement with a far field measurement. The looks of this menu differs a little depending on whether the current measurement (the measurement that is visible in the graphs) is an SPL or a Z measurement. Merge/Add functions require that you have performed at least two measurements. When the **Apply** button is clicked current measurement will be overwritten and the menu will switch back to the **Measurement** menu.
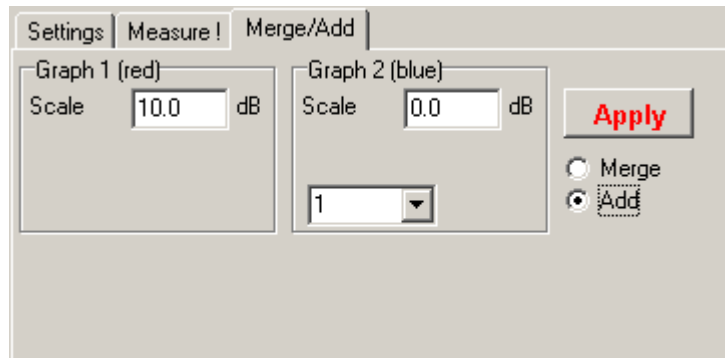
**Merging SPL measurements**
When SPL measurements are merged the menu looks like the figure below, the lower part of the resulting measurement is taken from the current measurement and the upper part is taken from the stored measurement (in this case measurement 1). For each part one can scale the measurement (**Scale** in dB) and also modify the phase so that the phase at the merge frequency (**Freq**. control) does not jump.
The benefit of using near field measurements for the low frequency part is that one does not get problems with room reflections that disturb the measurements. The drawback is that one does not see the diffraction effect caused by the baffle then. The simple solution is to compensate for this by adding a baffle step. If a value other than zero is entered to the Baffle step control a 6dB baffle step function is applied to the current measurements. If a value of 200Hz is entered then the -3dB point will be 200Hz, which corresponds to a baffle width of app. 20cm.
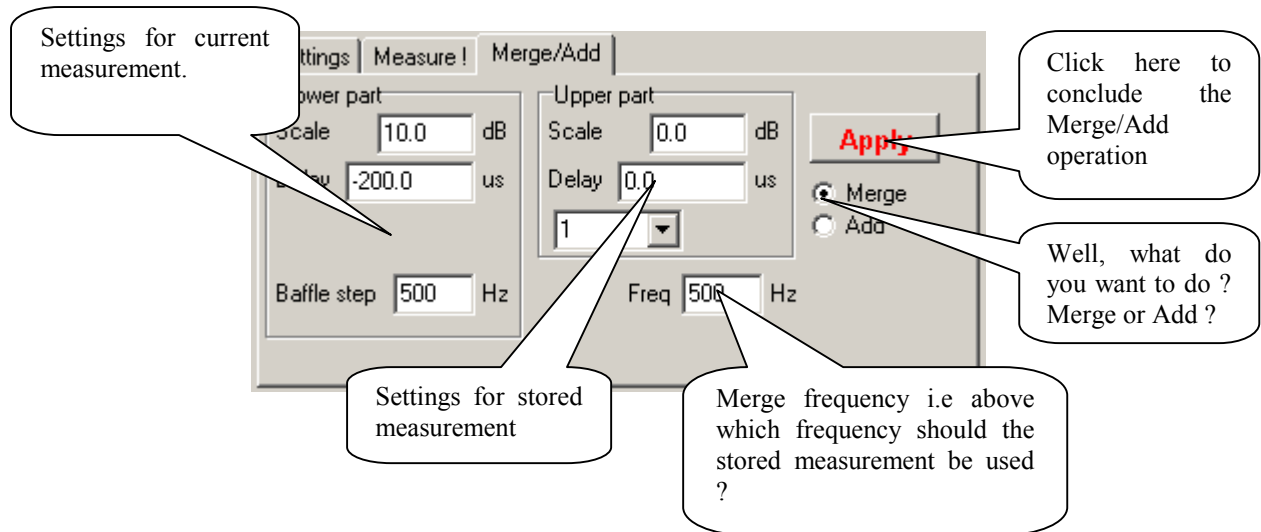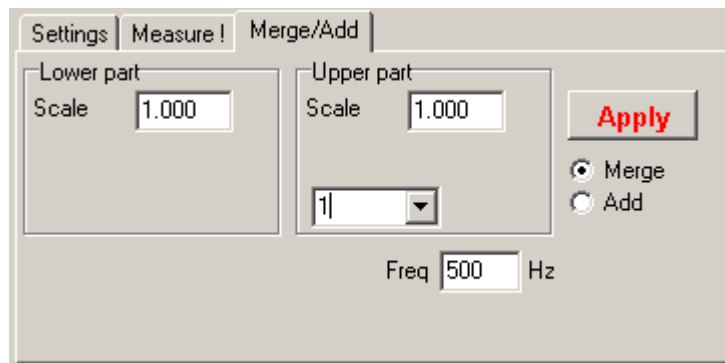
**Adding SPL measurements**

When SPL measurements are added one select the appropriate stored graph as indicated earlier and set the appropriate scaling. The resulting graph is depicted with a thick line. Adding of SPL measurements is used when calculating the total response from a bass reflex

enclosure from near field measurements of the loudspeaker driver unit and the port.

**Merging impedance measurements**

When Z measurements are merged the menu looks like below, the scaling is in times the original value i.e a **Scale** value of 0.5 will half all the impedance values.

**Add impedance measurements**

This feature does not have much of a practical use, it is however available for use.
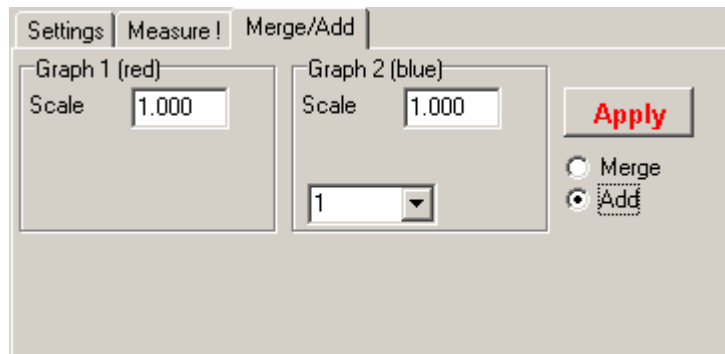
### *4.4 Store/Recall measurements*

Up to eight measurements can be stored, when one wish to store a measurement e.g in memory position 1 one click on the S1 button. Important to know is that it is the current measurement (i.e. the measurement that is visible in the graph) that is stored. It is however possible to alter the scaling and smoothing of the measurement after it has been stored.
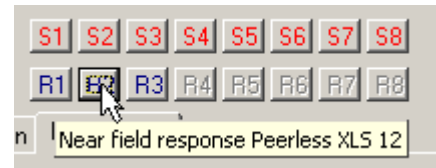
# 5 Configuration

### *5.1 Input configuration*

First of all, before doing anything else, one must make sure that the mixer is set so that the correct input is selected. Open the mixer device (double click on the small loudspeaker icon on the taskbar).

In the mixer device, select menu ***Options|Properties*** and click on **Recording** + **OK**. Here you will (most likely) see a track bar volume control for the microphone and the line input. Select the appropriate input.
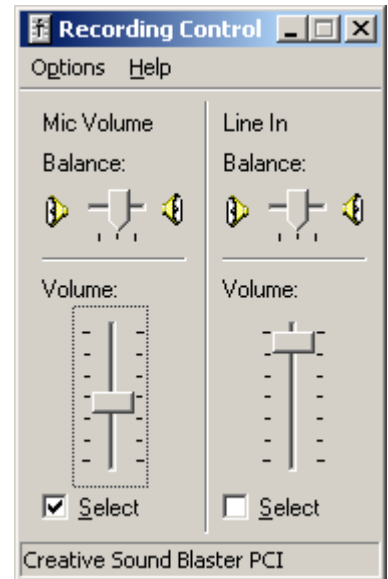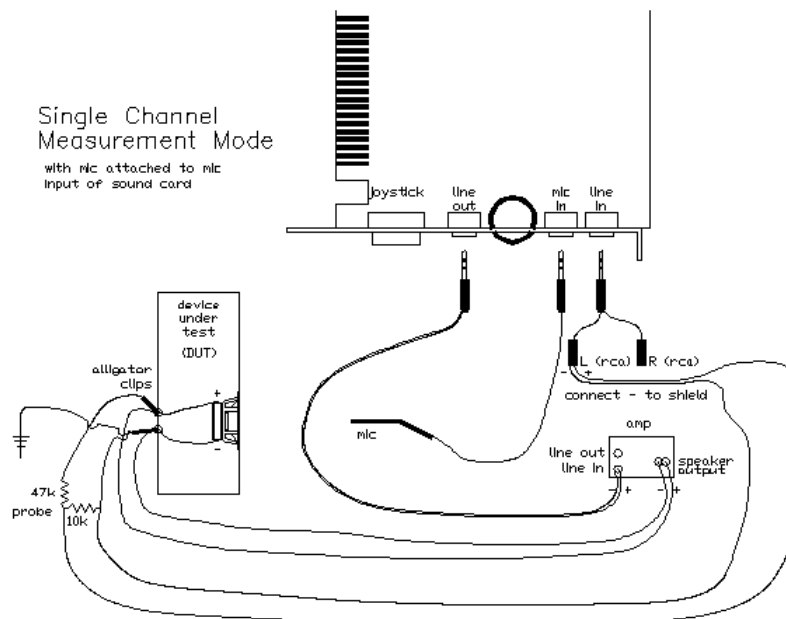
### 5.1.1 SPL measurements

SPL measurements can be performed either in single channel or in two-channel mode.

**Single channel mode**

Single channel mode is the simplest way to measure the acoustic loudspeaker response. The microphone is connected to the microphone input and mic input is selected in the mixer control.

**Two channel mode**
For two channel mode SPL measurements the line input is preferred as it is guaranteed to be a true stereo input, it is important that the mic preamplifier is connected to the left channel (also referred to as Ch 1 in this manual). The right channel (Ch 2) is connected to the terminals of the loudspeaker that is measured (remember to use the probe!).
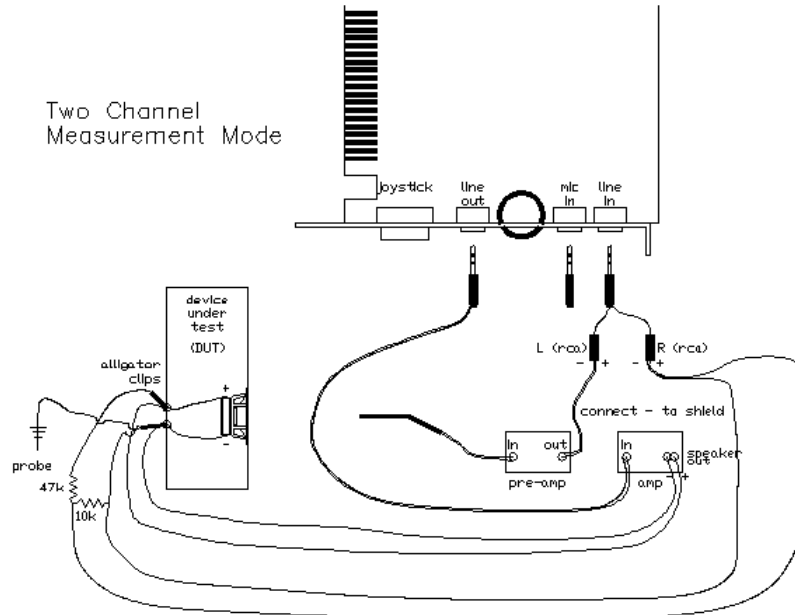
### 5.1.2  Z measurements

For Z or impedance measurements a resistor is needed. The resistor and the loudspeaker unit should be connected according to the figure below.

It is recommended to use a power amplifier to drive the setup, this also means that it IS HIGHLY RECOMMENDED to use the probes also!
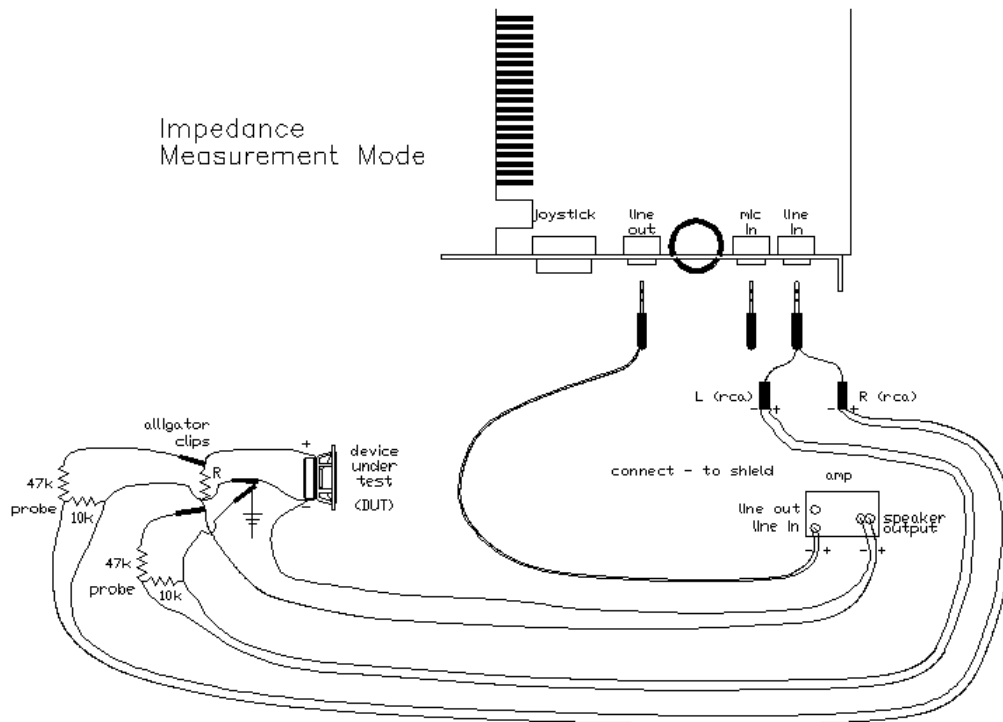


In some cases it is possible to use the line output from the soundcard, then an extra "optional resistor" of 1kΩ is also recommended. The connection of the left and right channels is not critical as justMLS will figure out the connection anyway.
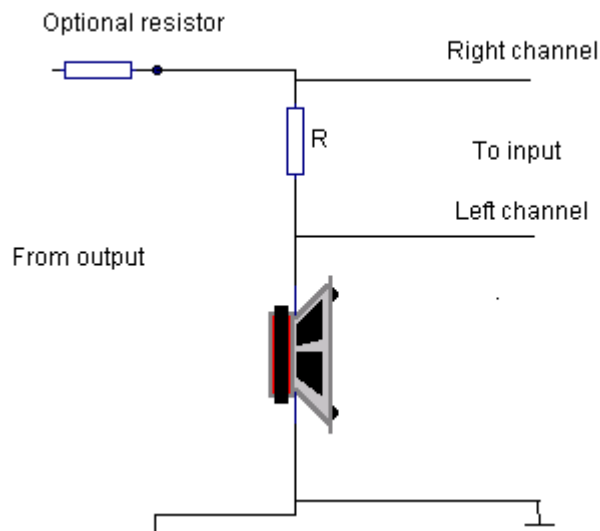
The recommended procedure is to select the input (preferably line input), calibrate the soundcard, set the levels and press the **Start** button. It is important to do a calibration as a very small difference between the channels may reflect largely on the impedance measurement.

## 5.2    *Setting the input level*

The input level is tested by clicking on the **Test levels** button. The level should be set so that the yellow area of the level bars is lighted, the red area should not be lit. Avoid overloading (clipping) of the inputs as it will degrade the measurements severely. The red part of the level meters should not be lit.

Do this procedure and adjust the level in the mixer so that the level is correct. Keep in mind that different loudspeaker units will cause different levels on the input, it is therefore good practice to set the level according to the strongest loudspeaker unit and keep it throughout all measurements (not necessary for impedance measurements). A general rule of thumb is that treble units will produce higher peak levels as the impulse response is much more concentrated in time for these, in other words hook up the treble unit first and adjust the level so that the input will not be overloaded
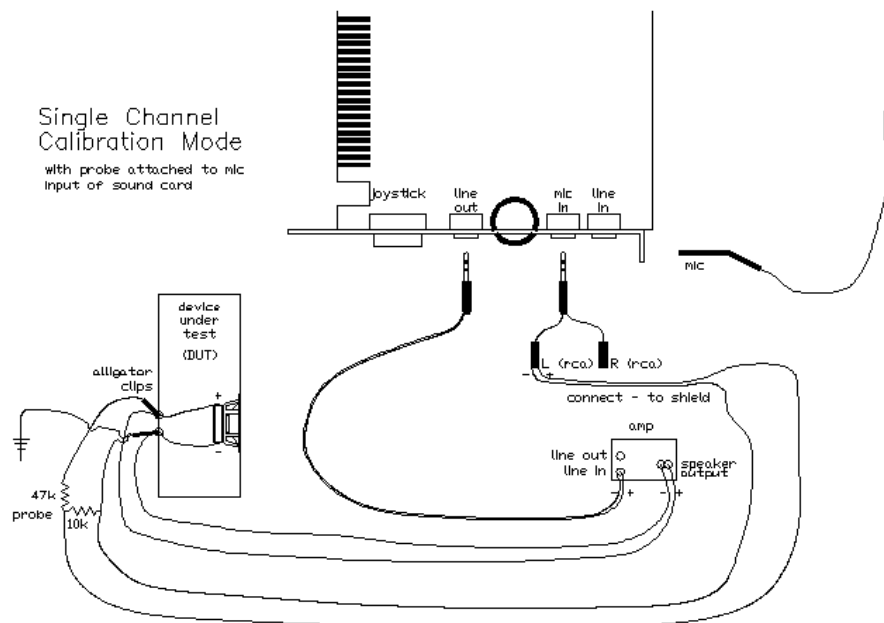
### *5.3*    *Calibrating soundcards*

The soundcard needs calibration for every sampling rate and for every MLS length. One would object against the need for calibration for every MLS length as the frequency response should not be affected by the MLS length, fact of the matter is however that the anti-aliasing filters in the soundcards are not entirely perfect and that this may give rise to ripples in the frequency response. By calibrating the soundcard for every MLS length that is used, this problem is alleviated. Calibration for SPL and Z measurement is separated, reason here is that for Z measurements one will most likely use different gear than for SPL measurements.

### 5.3.1  Calibration for SPL measurements

Calibration for SPL measurements is quite uncomplicated. The method differs a little between single channel and two channel measurements.

**Single channel measurements**
- Connect the mic input to the loudspeaker terminal of the loudspeaker that is to be measured, note that it may be necessary to attenuate the level.
- Test and adjust input levels.
- Run a calibration for the selected sampling rate/MLS length
- Connect the mic to the mic input.
- Test and adjust input levels.
- Measure.



**Two channel measurements**
  If you trust your soundcard to have the same response (both phase and magnitude) throughout the measurement range, calibration is not necessary. If not the simple calibration step is to connect the output of the soundcard to the line input.
  A more complex (but more accurate) way to calibrate is to connect the soundcard output to the right channel via a probe. The mic preamp is also connected to the soundcard output, it must however be attenuated before. The calibration this time will account for the transfer function of the mic preamplifier.

### 5.3.2 Calibration for Z measurements

It is very important that the calibration is correct for the Z measurements as small errors in balance between the channels may give rise to large errors in measured values. The simplest way to calibrate the soundcard for Z measurements is to connect the left channel in the figure in ch 5.1.2 to the same place that the right channel is. This way response differences are compensated for and the resulting measurements will be accurate.

# 6      Exporting measurements

Measurements are exported in ASCII format by clicking on the **Export** button. Below is an example of an exported file.

| Freq | Mag | Phase |
|------|------|--------|
| 20.12 | 67.8 | 66.1 |
| 20.30 | 68.2 | 64.1 |
| 20.49 | 68.6 | 62.3 |
| 20.68 | 69.1 | 60.6 |
| 20.87 | 69.5 | 59.1 |
| 21.07 | 69.9 | 57.7 |
| 21.26 | 70.3 | 56.5 |
| 21.46 | 70.7 | 55.4 |
| . | | |
| . | | |
| . | | |
| 18991.08 | 63.2 | -129.6 |
| 19166.98 | 83.9 | -4.7 |
| 19344.50 | 77.7 | -102.0 |
| 19523.68 | 78.8 | -10.8 |
| 19704.51 | 105.8 | -160.4 |
| 19887.02 | 83.8 | -121.8 |

The number of frequency points depends on the selected range if the selected display range is 20-20000Hz then only the frequency points within that interval are exported.

# 7      Analysis

As the name implies justMLS makes use of MLS sequences. MLS sequences give measurements that are very immune to noise. There must be some problem though… The problem is that the analysis assumes that the system under test is linear, measuring on a non-linear system such as a loudspeaker under heavy power loads will give meaningless results. Another requirement is that the MLS sequence is longer than the reverberation time in the room, this means that if your measurement room has a long decay time then the MLS sequence must be long. It is a general recommendation to use the longest MLS sequences as the measurements become cleaner then.

The frequency analysis in justMLS is done using Fast Fourier Transforms of different lengths on the impulse response that is visible in the Time domain graph.
For maximum frequency resolution, which is important in the low frequency region, a long input sequence is needed. For this purpose a 8192 point FFT is used. Problem with using a long sequence is that the noise immunity is not very good especially at high frequencies.
To alleviate this problem shorter FFTs are used for the higher frequency regions. In justMLS three different FFT's are computed.
8192pt FFT:        Used up to 1.25% of the sampling rate.
1024pt FFT:        Used from 1.25% to 12.5% of the sampling rate.
256pt FFT:        Used from 12.5% to half the sampling rate.
This means that if one use a sampling rate of 44100Hz then the 8192pt FFT is used up to 550Hz, the 1024pt FFT is used from 550Hz to 5500Hz and the upper part is taken care of by the 256pt FFT.
This method works quite nicely, it demands however that the analysis windows are located so that the beginning of the impulse response is located in the beginning of the FFT analysis window. The analysis window is tapered with a half-hanning window. To make subsequent processing easier the frequency points are converted from the linear frequency spacing of the FFT's to a logarithmic spacing using 1000 frequency points between 5Hz and 50000Hz by means of linear interpolation.

# 8      Doing far field measurements.

Far field measurements are done with the microphone placed at some distance from the loudspeaker. The distance is typically 50 to 100cm depending of the baffle size of the loudspeaker, the larger the baffle, the larger the measurement distance in order to reach far field conditions. The measurement methodology differs a little depending on whether one channel or two channel mode is in use.

**One channel mode:**
The **Offset** setting can be can most often be left as is as justMLS will search for the most likely starting point of the impulse response. Here it is possible to move that starting point of the FFT window forward and backward too, and also to fine adjust it to the beginning of the impulse response.

**Two channel mode:**
In two channel mode justMLS will find the start of the reference input (Ch 2). The mic input will be delayed some amount depending on the distance between mic and loudspeaker, the simple way is probably measure the distance and type the value as an analysis window offset into justMLS.

When doing measurements in one channel mode it is possible that the phase response might differ between measurements, what justMLS does is that it searches for the largest peak in the impulse response, then rewinds back and searches forward until the level is at least 1/10th of the peak level. The staring point of the FFT window is set 10 samples behind that point. The starting point may thus differ between measurements and that will affect the measured phase response simply because different amounts of delay is added. This is a sort of side effect of the hardware free approach of justMLS one channel mode.

The above methodology means that the user must find out the acoustical centers of the loudspeaker units. The simplest way to do this is to measure all loudspeaker units individually, then measure them all connected in parallel and model that in a loudspeaker crossover CAD such as LspCAD. In LspCAD one will then have to tweak with the dZ values until a match is found. Below is a more detailed description how this is done with the help of LspCAD (extracted from the tutorial section of the LspCAD manual).

"The measured frequency response must be verified in some way. A method commonly used is to do a final measurement with all driver units connected together and the microphone placed      in      front      of      the      treble      unit      (at      some      distance). The measured response will look awful but don't let that scare you off.  Create an advanced passive crossover and import both the measured frequency response and impedance data. Remember also to set the positions of all the drivers to the positions relative to the location of the treble unit, let dZ=0 for all driver units. This of course means that the position of the treble unit is dX=dY=dZ=0. Moreover set the measurement distance (in the General menu) to be the same as the distance between the microphone and the baffle.
Now take a look at the *Summed frequency response* diagram, does it look similar to the real measurement ?. If yes then we can continue with the crossover construction. If no, then we have to set the dZ values for some of the driver units. Often it is possible to get dZ by measuring the depth of the driver cone i.e how recessed the dust cap is relative to the rim of the cone.

For two channel mode measurements the scenario is a bit simpler, it also gives a little more possibilities, the procedure below is recommended then. For the verification the method described earlier can be used.

1. Start with measuring the impedance of the individual driver units, note that both magnitude and phase is needed in LspCAD.
2. When measuring the frequency response of the driver units. The microphone must be placed at a distance from the driver unit, directly on axis. This is especially important for midrange and treble units as the near field and far field response are very different due to cabinet edge diffraction effects. It is recommended to use a distance of about 1m, in some cases one can use a distance of 60-70cm.  It is recommended to use a microphone stand so that the microphone can be raised and lowered so that it is always in front of the driver unit being tested. Remember to keep the distance between the microphone and the **<u>baffle plane</u>** constant throughout the measurements.

# 9    A small tutorial

This section describes a frequency response measurement session where the various features of justMLS are used.

The frequency response in this example will be measured both in near and far field conditions. First of all the soundcards should be calibrated, this is not critical for response measurements with justMLS in two channel mode, but is nevertheless recommended. As an example a two way small center channel bass reflex loudspeaker is used.

First of all we would like to determine the low frequency response as accurately as we can. To achieve this we need to do near field measurements i.e. we measure with the mic very close to the loudspeaker unit and the port.

**Low frequency response**

The bass unit is measured with 8000Hz sample rate and 32768 samples MLS length, the resulting near field response is shown below. The measurement is done with the mic directly in front of the cone only 5mm from the cone surface. For the analysis the Ch 1-2 is selected to
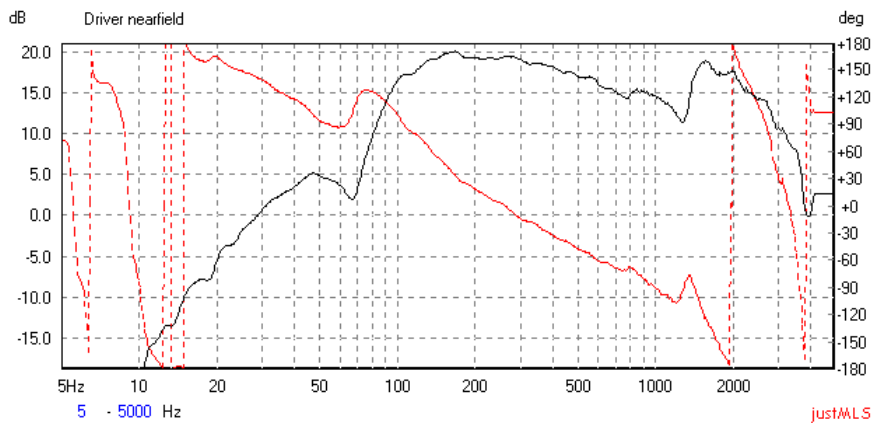
be viewed in the frequency response graph as we are only interested in the frequency response of the loudspeaker, not the combination power amplifier and loudspeaker.

The port is measured with the mic almost in the port, the frequency response is shown below.
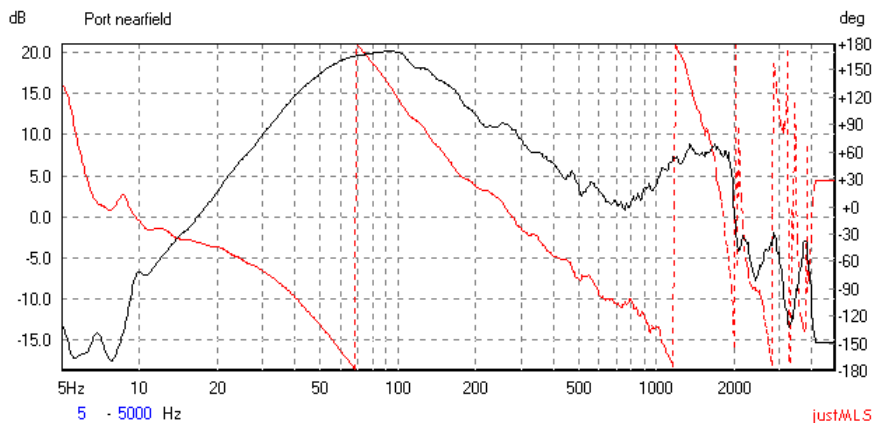
The driver and port measurements are stored in memory locations 1 and 2. The driver and port measurements are added by means of the Merge/Add menu (see fig below). The port measurement (red) is scaled down 8.6dB o that it matches the level of the loudspeaker unit (blue) below the port resonance. The thick black graph is the summed response. The **Apply**
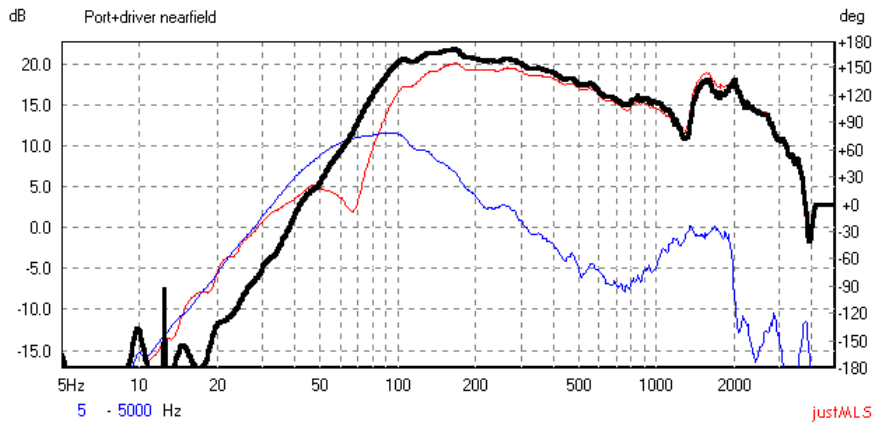


button is pressed and the result is stored in memory location 3.

**Midrange and above response**
The sample rate is set to 48000kHz, this sample rate fits the SB Live card very well and it will assure good measurements.
As we do a two-channel measurement the reference signal will be recorded on the right channel and the mic signal will be recorded on the left channel. As the mic needs to be located at some distance to ensure that it is really the far field response that is measured we will have a delay in the mic measurement relative to the reference. In this case the distance between the mic and the loudspeaker is a little more than 50cm, therefore we need to set an offset of the analysis window of 50cm.
The figure below shows the impulse responses of the loudspeaker (black) and the loudspeaker input signal (blue). The length of the impulse response window is 7ms and the offset is 50.88cm or 1.4792ms.
It is important that the impulse response of the loudspeaker in it entirety is located in the analysis window, it is also important that the first part is located to the left.
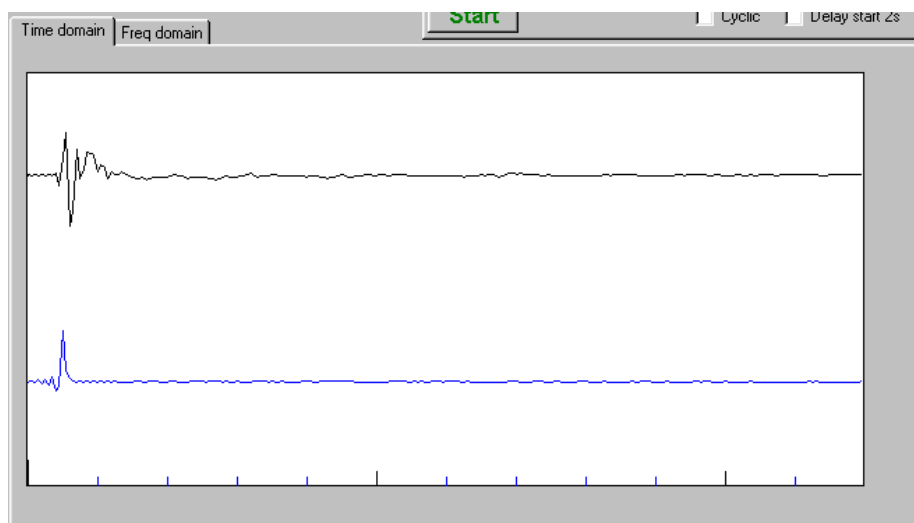
When the offset of the window is set appropriately it is high time to have a look at the frequency response. The length of the analysis is set such that as much of the room reflections are avoided still with the goal of at least some resolution in the lower frequency region. The
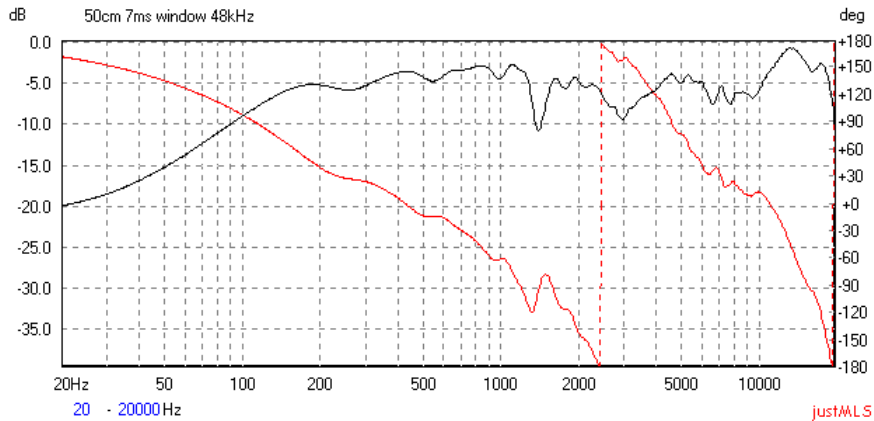


best choice this time is a 7ms window, see figure above. A 1/12$^{th}$ octave smoothing is applied to the measurement to make it look less noisy.

The measurement is stored in memory location 4.

**Total response**

With measurements available we are now ready to compose a full range frequency response. We recall the measurement stored in memory location 3 (the port + loudspeaker near field measurement) and open the Merge/Add menu again, this time we select the Merge function. For the lower frequency range we have the near field measurement available as the current measurement. For the upper range we select the measurement stored in memory location 4. The upper range measurement is scaled so that the levels match at the merge frequency
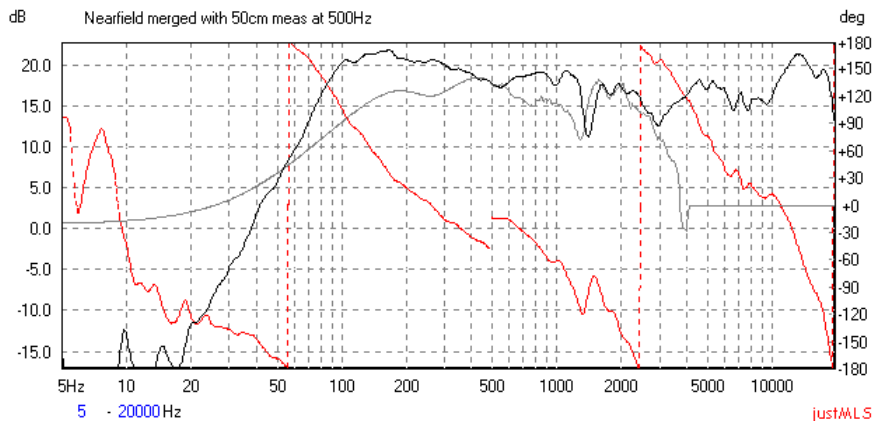


(500Hz). One may as well scale down the measurement of the lower frequency range, it is a matter of taste. The black graph is the merged response that one would get when the **Apply** button is pressed.

The result is not totally satisfactory yet, one problem with near field measurements is that they don't take into account the baffle step that one automatically get when the loudspeakers are mounted in a baffle. Therefore we see a very pronounced peak at 150Hz that would not show if we do a free field measurement.

The baffle width is 20cm. If we set the baffle step control to 200Hz we will get a frequency response where the baffle step is incorporated into the near field response with a more realistic response as a result (see below).
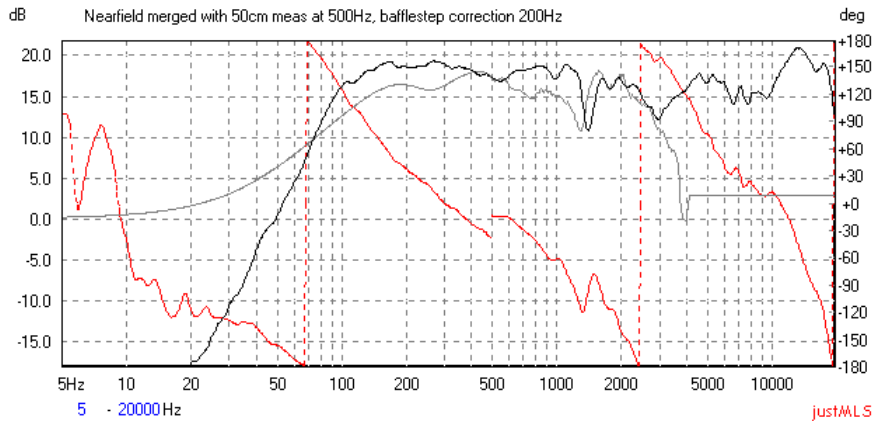


The phase response (red) shows a slight jump at the merge frequency (500Hz) that is not natural, we set the delay of the lower part to –138us and thus we get rid of the phase jump.
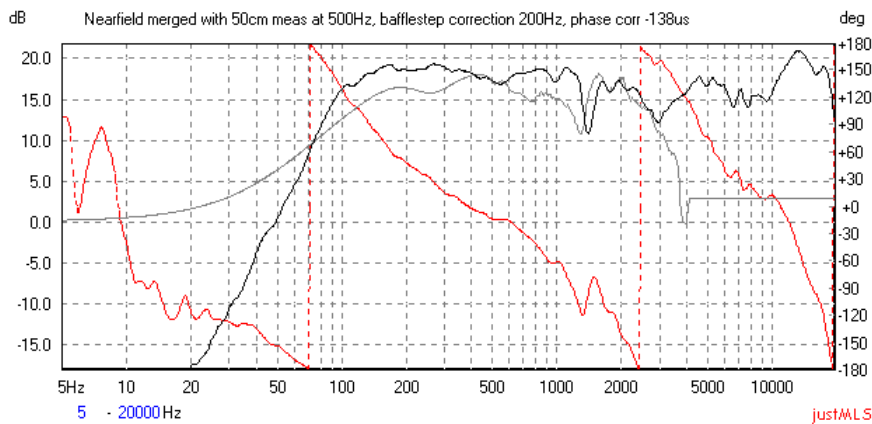


The final response is shown below.