# LIMP

Program for Loudspeaker Impedance Measurement

# User Manual

Version 1.7.0.

**Ivo Mateljan**

Electroacoustics Laboratory
Faculty of electrical engineering
R. Boskovica bb.
21000 Split, Croatia

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# Content

# 1 What is *LIMP*?

The *LIMP* is a program for the measurement of the loudspeaker impedance and estimation of loudspeaker physical and dynamical parameters (also called Thiele-Small parameters). It is also a general purpose program for measuring impedance in the range from 1 to 200 ohms.

Requirements to use the *LIMP* are:

- o Operating systems: Windows 2000 / XP / Vista / 7 – 32 bit
- o Processor class Pentium, clock frequency 600 MHz or higher, memory 256M for Windows 2000/XP or 2MB for Vista/Windows 7
- o Full duplex soundcard with synchronous clock for AD and DA converters
- o WDM or ASIO soundcard driver (ASIO is trademark and software of Steinberg Media Technologies GmbH).

The *LIMP* has been used successfully with the following soundcards:

- o RME Fireface 800, RME Fireface 400, RME DIGI96, RME HDSP
- o Duran Audio D-Audio, EMU 1616m, EMU 0404 USB
- o Terratec EWX 24/96, Firewire FW X24, YAMAHA GO46,
- o M-audio Audiophile 2496, USB Transit, Delta 44,
- o Digigram VxPocket 440 - a notebook PCMCIA card
- o Echo Layla 24
- o TASCAM US-122 - USB audio
- o ESI Quatafire 610, U24 USB and Waveterminal,
- o Soundblaster Live 24 and Extigy-USB (but only at 48kHz sampling frequency),
- o Turtle Beach Pinnacle and Fuji cards,

The *LIMP* may be used with slightly degraded performance with following soundcards:

- o Soundblaster MP3+ USB (notice: don't install SB driver, use a Windows XP default driver),
- o Soundcards and on-board audio with AC97 codecs (problem with high noise in FTT mode).

The installation of program is through common setup program for the ARTA Software.

All Windows registry data for *LIMP* will be automatically saved at first program execution, and files with extension ".LIM" will be registered to be opened with a program *LIMP*.

Results of measurement can also be saved as ASCII formatted files in .ZMA format.

The *LIMP* does not dump graphs to the printer, instead of this all graphs can be copied to the Windows Clipboard and pasted to other Windows applications.

# 2 Impedance measurement theory

## 2.1 Basic circuit for impedance measurement

The measurement of the loudspeaker impedance is based on the system shown in Fig. 2.1. The referential resistor *R* is connected between the signal generator and a loudspeaker impedance Z.
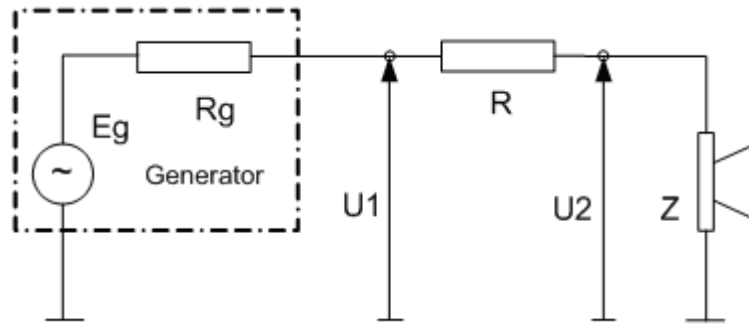


**Figure 2.1** A circuit for loudspeaker impedance measurement

Impedance is defined in the frequency domain - $Z(f)$. If we measure voltages $U_1(f)$ and $U_2(f)$, at both end of the referential resistor, we estimate the loudspeaker impedance as:

$$Z(f) = \frac{U_2(f)}{U_1(f) - U_2(f)} R \tag{1}$$

## 2.2 Stepped sine and Pink periodic noise generator

*T*he *LIMP* has two modes of impedance measurement:

*1*) Stepped sine mode
*2*) FFT mode with pink periodic noise excitation (Pink PN)

In the stepped sine mode *LIMP* generate burst of pure sinusoidal signal, frequency by frequency with 1/6, 1/12, 1/24 or 1/48 octave increment, and measures response to sinusoidal signal by filtering out noise and distortion components in sinusoidal response.

FFT mode is faster method for impedance measurement, but with much lower measurement S/N ratio than stepped sine mode. In this mode the *LIMP* simultaneously measures values of the impedance in the whole audio range of frequencies by using the signal generator with a wideband periodic pink noise excitation (pink PN).
The wideband pink PN excitation is realized as periodic random phase multisine signals. It is a zero dc, periodic signal that contains *M* sine components, each with a random phase:

$$g(t) = \sum_{k=1}^{M} A_k \cos(2\pi k f_0 t + \varphi_k), \quad \varphi_k = random \in [0, 2\pi] \tag{2}$$

This pink multisine has $A_k = \sqrt{2}A_{2k}$ and its spectral magnitudes roll-off 3dB/oct (after some cut-off frequency). In the *LIMP*, a variable low frequency cut-off frequency can be changed in the '**Signal Generator Setup**' dialog box. The pink noise is usually used with a cut-off frequency set close to the frequency where loudspeaker has an impedance maximum (20-100Hz). The pink periodic noise has the statistical distribution close to the normal distribution, and in *LIMP* it is generated with a crest factor lower than 12 dB.

## *2.3 Simple measurement procedure*

In **Stepped sine mode**, a simple measurement procedure is as follows:

1. The *LIMP* generates burst of sine signal with frequency *f*. We assume that this signal drives the generator shown in Fig. 2.1.

2. After an arbitrary *transient time*, that is necessary to reach the steady state, voltages at both end of the referential resistor are measured as discrete time series $u_1$ and $u_2$ of length *N*.

3. The magnitude and phase of signal $u_1$ and $u_2$ at frequency *f* are estimated by finding fundamental sinusoidal components $U_1$ and $U_2$ at generator frequency f. It is done by directly solving Fourier integral in *integration time* T. These fundamental harmonics are used in the equation (1) to calculate the impedance *Z*.

4. After some arbitrary time, we call it *intra burst pause*, that is necessary for measured system to release reactive energy, a frequency is incremented for 1/24 octave (or 1/48 octave) and the process is being repeated form step 1) until the predefined stop frequency is reached.

In **FFT mode**, a simple measurement procedure is as follows:

1. The *LIMP* generates the Pink PN as a discrete periodic sequence with a period equal to *N*. We assume that this sequence drives the generator shown in Fig. 2.1.

2. After one preaveraging cycle, which is necessary to reach the steady state, voltages at both end of the referential resistor are measured as discrete time series $u_1$ and $u_2$ of length *N*.

3. The DFT is applied to time series $u_1$ and $u_2$ to get spectral components $U_1$ (*f*) and $U_2$(*f*). They are used in the equation (1) to calculate the impedance *Z*(*f*).

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**Important note:**
The signal generation in the stepped sine mode gives sinusoidal components with at least 30dB higher level than it is possible in FFT mode. That is why, in FFT mode the measurement results can be greatly affected by the noise that can be generated by the measured loudspeaker.

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## 2.4 Measurement in a noisy environment

The main source of the measurement noise is a loudspeaker that acts as a microphone for the environmental noise and vibrations. Fig. 2.2 shows modified circuit for the loudspeaker impedance measurement, with included the noise generator $E_n$.
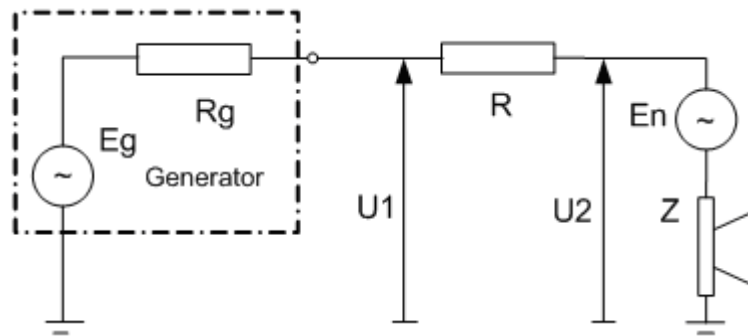


**Figure 2.2** A circuit for the loudspeaker impedance measurement, with the noise generator $E_n$

If we apply equations for impedance measurement (1) to this circuit, we get the estimated impedance value:

$$Z_{estimated} = R\frac{U_2}{U_1 - U_2} = Z + \frac{R + Z + R_g}{1 + E_g / E_n} \qquad (3)$$

What this equation shows is that estimated impedance differs from the true impedance Z by the term that is dependant on the S/N ratio ($E_g/E_n$) and values of resistors R, $R_g$ and the impedance Z. We can conclude:

1. The signal generator must supply a high voltage, to assure high S/N. In practice, we need a generator with at least 1V of a peak output voltage. This can easily be achieved with stepped sine excitation.
2. When we use FFT method the measurement results are highly affected with noise. The loudspeaker act as a microphone with a highest sensitivity in the region of the membrane resonance. It means the highest level of the noise at low frequencies, so we must generate the signal with highest level at low frequencies, i.e. the pink noise.
3. Values of resistors R and $R_g$ must be small, an optimum being a value close to the magnitude of the measured impedance. Practically, we can use R = 10-27 ohm to get a very good impedance estimation, but then we need a power amplifier to supply large current. If we use the soundcard phone output as a signal generator, then we can use R = 47-100 ohms. If we use the soundcard line output as a signal generator, due to the limited current capability, we must use R>600 ohms. In that case we can't get a good estimation with FFT method, but still we can use stepped sine method.

## 2.4.1 Lowering measurement noise in stepped sine mode

In stepped sine mode *LIMP* uses "heterodyned" principle to filter all spectral components that are out of the passband which is centered at the measured frequency. The bandwidth of the filter is equal to 1/T where T is *integration time* of the Fourier integral. For example, if we use integration time 200ms, then the width of the passband of the "heterodyne" filter is 5Hz.

## 2.4.2 Lowering measurement noise in FFT mode with periodic noise excitation

The noise can be partly lowered by using the averaging technique in the estimation of $U_1$ and $U_2$. *The* LIMP averages the auto-spectrum of U1 and the cross-spectrum of $U_1$ and $U_2$, giving the following estimated impedance:

$$Z_{esitimated} = \frac{R}{\dfrac{\langle U_1 U_1^* \rangle}{\langle U_2 U_1^* \rangle} - 1}$$

(4)

**Note:** brackets <> denotes averaged values and star denotes the conjugate complex value.

Two type of the averaging can be applied in the *LIMP*. In a "time domain synchronous averaging" the LIMP generates multiple periods of one noise sequence (Fig. 2.3). In a "frequency domain asynchronous averaging" the *LIMP* generates a different noise sequence for every acquired FFT block (Fig. 2.4). The Frequency Domain Asynchronous Averaging can give slightly better results than the synchronous averaging in systems with nonlinear distortion, but it needs longer time for measurements.
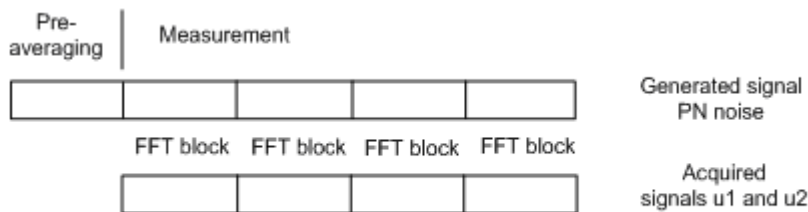


**Figure 2.3** Signal generation and acquisition during the synchronous time domain averaging process
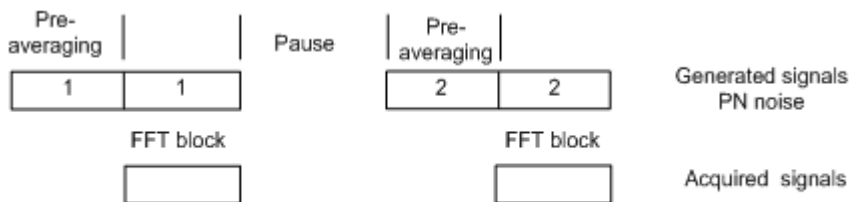


**Figure 2.4** Signal generation and acquisition during the asynchronous frequency domain averaging

# 3. Hardware Setup

The simplest measurement configuration is shown in Fig. 3.1. The soundcard phone-out (or loudspeaker out) is used as a signal generator output. The soundcard left line-input is used for recording the voltage $U_1$ and the soundcard right line-input is used for recording the voltage $U_2$.

**Note:** If the soundcard has no phone-out or loudspeaker-out, then we have to use the line-out and a external power amplifier, as shown in Fig. 3.2.

**Figure 3.1** Measurement setup for loudspeaker impedance measurements, using the phone output
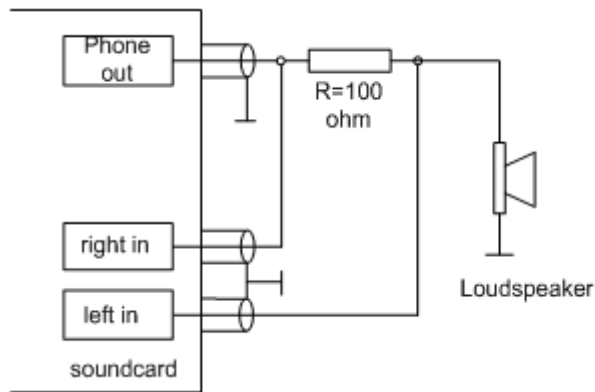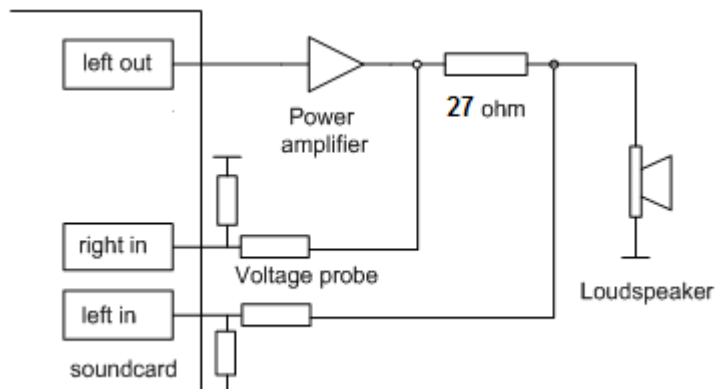
**Figure 3.2** General measurement setup for loudspeaker impedance measurements

To protect the soundcard input form large voltage that is generated at a power amplifier output, it is recommended to use the voltage probe circuit with Zener diodes, as shown on Fig. 1.3. Values of resistors R1 and R2 have to be chosen for arbitrary attenuation (i.e. R1=8200 and R2=910 ohms gives probe with -20.7dB (0.0923) attenuation if the soundcard has usual input impedance - 10kΩ). In a single channel mode this probe is not connected.

**Figure 3.3** Voltage probe with the soundcard input channel overload protection
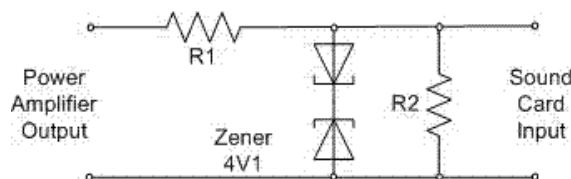
# 4 Working with *LIMP*

When you start the *LIMP* you get the program window shown in Fig. 4.1. There are: menu bar, toolbar and a dialog bar at a top of the window and a status bar on the bottom of the window.  The central part of the window will show the magnitude and the phase plot.
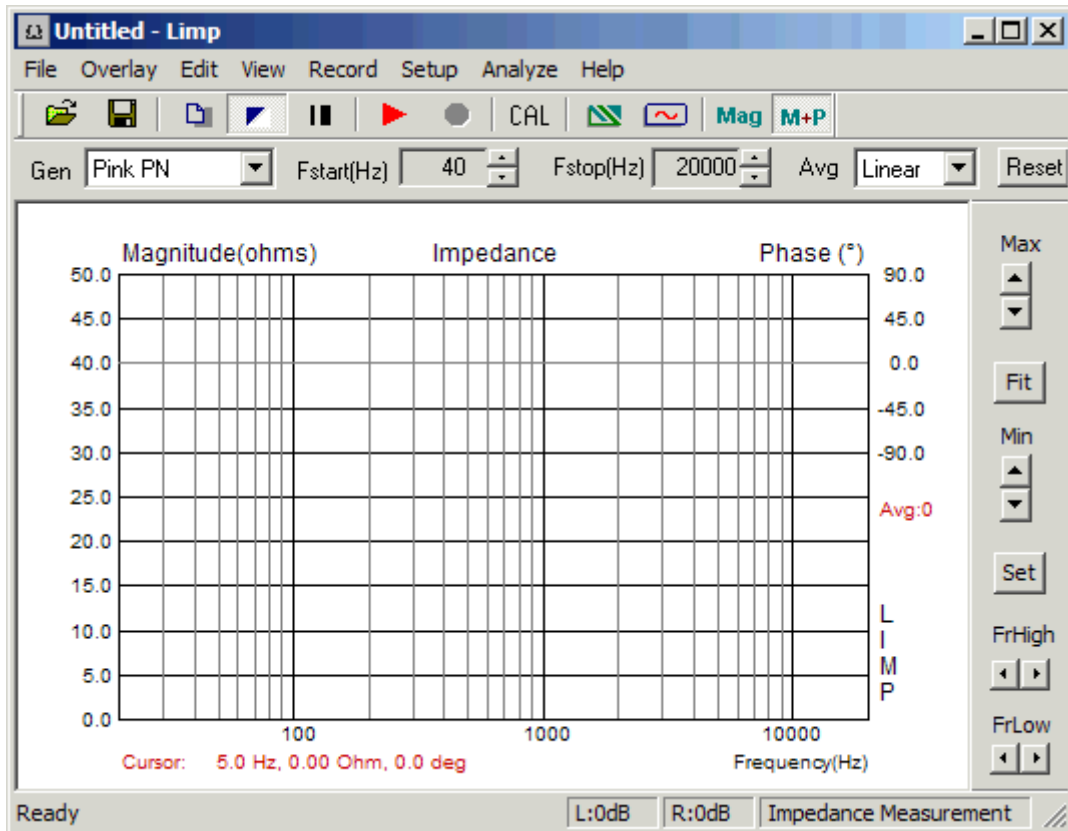

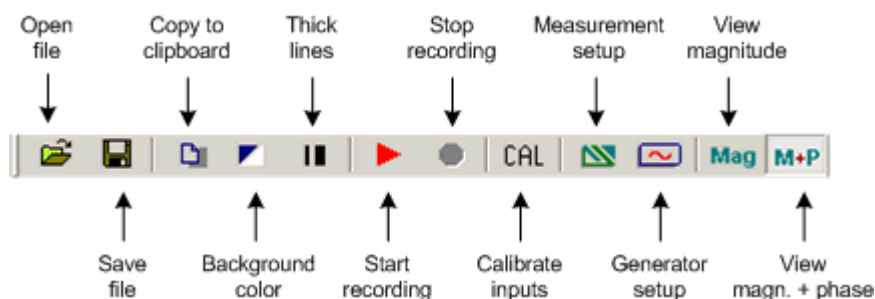
**Figure 4.1** Main program window



**Figure 4.2** Toolbar icons

**Figure 4.3** Status bar shows the peak level (ref. full scale) of left and right line inputs
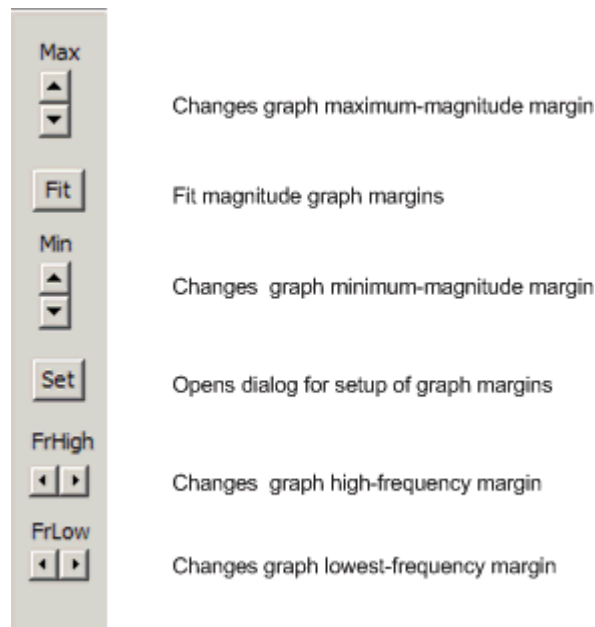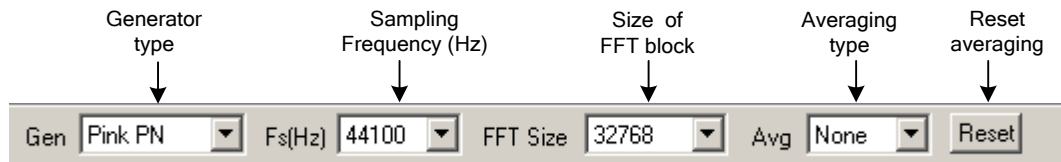


**Figure 4.4** Top and right dialog bars

Normally, we work with graph windows and dialog boxes. We also need to get the copy of the graph or the graph window picture.

Copying of the full window picture is simple. User needs to simultaneously press keys **Ctrl+P**. After that command the window picture will be saved in the System Clipboard, from were the user can paste it in other opened Windows applications (MS Word, MS Paint and Adobe Paint Shop).

To obtain the copy of the graph picture, that is shown inside the window, user needs to simultaneously press keys **Ctrl+C** or activate the menu command '**Edit->Copy**', or press appropriate '**Copy**' button. In main window toolbar, the 'Copy' button is shown as toolbar icon  .

This command opens the dialog box '**Copy to Clipboard with Extended Information**', shown in Figure 1.16. Here user has to set three copying options:

1) In the **Edit box** user optionally enters the text that will be appended at the bottom of the graph.

2) Check box '**Add filename and date**' enables adding text to the graph that shows file name, date and time.

3) Check box ' **Save text**' enables saving entered text for the next copy operation.

4) **Bitmap size** is chosen by selecting one of following combo box items:
   - Current screen size      - variable width and height option
   - Smallest  (400 pts)      - fixed graph width 400 points
   - Small  (512 pts)          - fixed graph width 512 points
   - Medium  (600 pts)      - fixed graph width 600 points
   - Large  (800 pts)          - fixed graph width 800 points
   - Largest  (1024 pts)      - fixed graph width 1024 points

   The options with fixed width gives graph copy with the aspect ratio 3:2.

   Pressing the button 'OK'  copies the graph to the system clipboard.
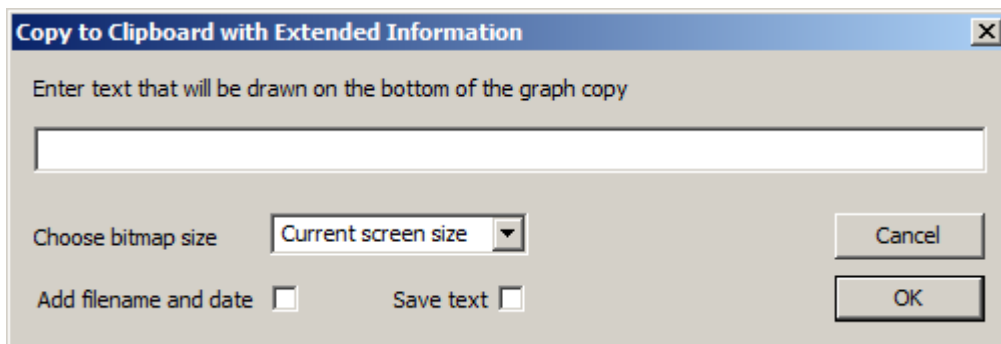   Pressing the button 'Cancel'  cancels the copy operation.



**Figure 4.5** Dialog box '**Copy to Clipboard with Extended Information**',

# *4.1 LIMP Menus*

Here is a brief explanation of *LIMP* menus:

## File

**New** - initializes a new file content
**Open...** - opens an existing impedance file (*.LIM)
**Save** - saves the file with a current name
**Save As...** - saves the file under a new name
**Info.** - shows information about current file
**Export ASCII** - exports the file content in a textual - ASCII file
        **Commented .TXT file** - exports textual file with comment and rows containing frequency, impedance magnitude and phase
        **Plain .ZMA file** - exports textual file with rows containing values of frequency, impedance magnitude and phase
        **CSV (Excel) file** - exports CSV file containing frequency, impedance magnitude and phase

**Recent File** - opens a recently opened file
**Exit** - exits the program

## Overlay

**Set as overlay** - sets the current impedance curve as an overlay
**Delete** - deletes overlay
**Load** - loads an overlay from an impedance file

## Edit

**Copy** - copies the plot as a bitmap to the clipboard
**Setup colors and grid style** - sets graph colors and grid style
**B/W Color** - changes the background color to black or white
**Use thick pen** - uses thick pen for plotting curves

## View

**Toolbar** - shows or hides the Toolbar
**Status Bar** - shows or hides the Status bar
**Fit graph range** - changes graph top margin to show full impedance curve
**Magnitude** - shows only impedance magnitude curve
**Magnitude+phase** - shows impedance magnitude and phase curves

## Record

**Start** - starts recording (measurement)
**Stop** - stops recording (measurement)
**Calibrate** - opens the dialog box for calibration of line input channels

## Setup

**Audio devices**  -  sets current input and output devices
**Generator** -  sets generator parameters
**Measurement**  -  sets measurement parameters
**Graph**  -  sets graph margins
**CSV format**  -  opens dialog box for setup of  decimal separator character in CSV files

**Analyze**

**Loudspeaker parameters - Added mass method**  - shows the dialog box  for the estimation of loudspeaker parameters using Added mass method
**Loudspeaker parameters - Closed box method**  - shows the dialog box for the estimation of loudspeaker parameters using Closed box method
**RLC impedance values at cursor position** - estimate values of impedance serial RLC elements

**Help**

**About**  -  gets information about the *LIMP*
**Registration**  -  shows the user license registration
**User Manual**  -  shows the help file

**Useful shorcut keys are:**

**Up** and **Down** keys  - change the top graph margin
**Left** and **Right** keys  - move the cursor left and right
**Ctrl+S** key  -  saves the file
**Ctrl+N** key  -   makes a new file
**Ctrl+O** key  -  opens the file
**Ctrl+C** key  -  copies a graph bitmap to the clipboard (with user defined options )
**Ctrl+P** key  -  copies a whole window bitmap to the clipboard
**Ctrl+B** key  -  changes background color

## *4.2 Soundcard Setup*

Activate the menu **Setup->AudioDevices**. You will get the '**Soundcard Setup'** dialog box shown in Fig. 4.6. In this dialog box you choose which soundcard will be used as an input or output device. Generally, choose the same card as an input and output device.
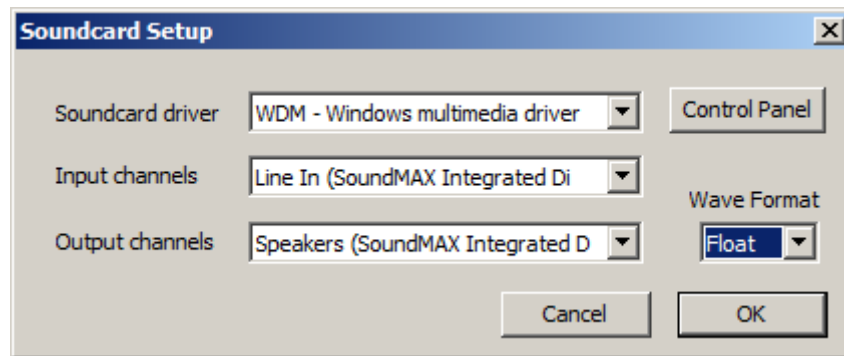


**Figure 4.6** Soundcard setup

The '**Audio Device Setup'** dialog box has following controls:

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**Soundcard driver** - chooses the type of soundcard driver (WDM – windows multimedia driver or one of installed ASIO drivers).
**Input channels** - chooses the soundcard input stereo channels. ASIO driver can have large number of channels.
**Output Device** - chooses the soundcard output stereo channels.
Generally, user chooses input and output channels of the same soundcard (mandatory in ASIO driver mode).
**Control panel** button – if WDM driver is chosen, it opens sound mixer on Windows 2000/XP or Sound control panel in Vista/Win7. If ASIO driver is chosen, it opens ASIO control panel.
**Wave format** – on Windows 2000/XP chooses Windows wave format: 16 bit, 24 bit, 32 bit or Float. Float means IEEE floating point single precision 32-bit format. It is recommended to use 24-bit or 32-bit modes when using a high quality soundcard (many soundcards are declared as 24-bit, but their real bit-resolution is less than 16-bits). For Windows Vista / Windows 7 it is recommended to choose resolution type Float. This control has no effect in the ASIO mode, where a bit resolution has to be setup in the ASIO control panel.

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> **Important notice: Please mute the line and microphone channels at the output mixer of the soundcard; otherwise you might have a positive feedback during measurements. If you use a professional audio soundcard, switch off the direct or zero-latency monitoring of the line inputs.**

## 4.2.1 Windows 2000 / XP WDM driver setup

After selection of the soundcard user has to disable (mute) line-in and microphone inputs in output mixer. Also user has to select which input will be used for recording: Line In or Microphone (Mic). For a standard PC soundcards the procedure is as follows:

5) In '**Audio device setup**' dialog click the button '**Control panel'** to open the Windows '**Master Volume**' dialog box, which is shown on Fig. 4.8.
6) Click on menu '**Options->Property**' and select soundcard channel that will be used for output (playback), as shown in Fig. 4.7.
7) Mute Line In and Mic channels in dialog '**Master Volume**' (Fig. 4.8)
8) Set Master Volume and Wave Out volume to maximum.
9) Click on menu '**Option->Property**' and select soundcard channel that will be used for input and enable Line In and Mic channels in recording mixer.
10) Choose Line In or Mic Input. Normally, Limp uses Line In input on which external microphone amplifier should be connected.
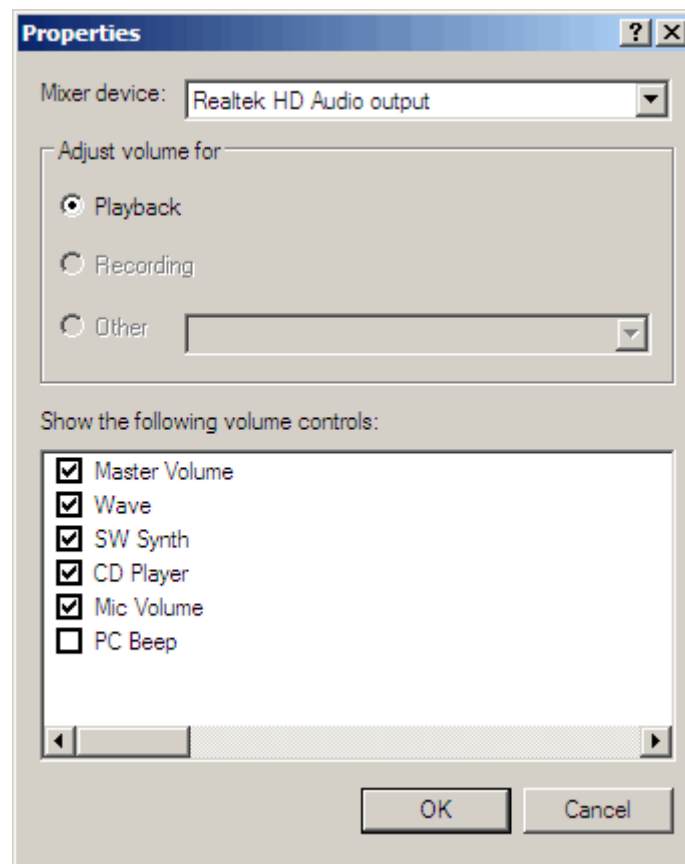11) Set volume control of Line In to some lower position. Later it will be set more precisely.



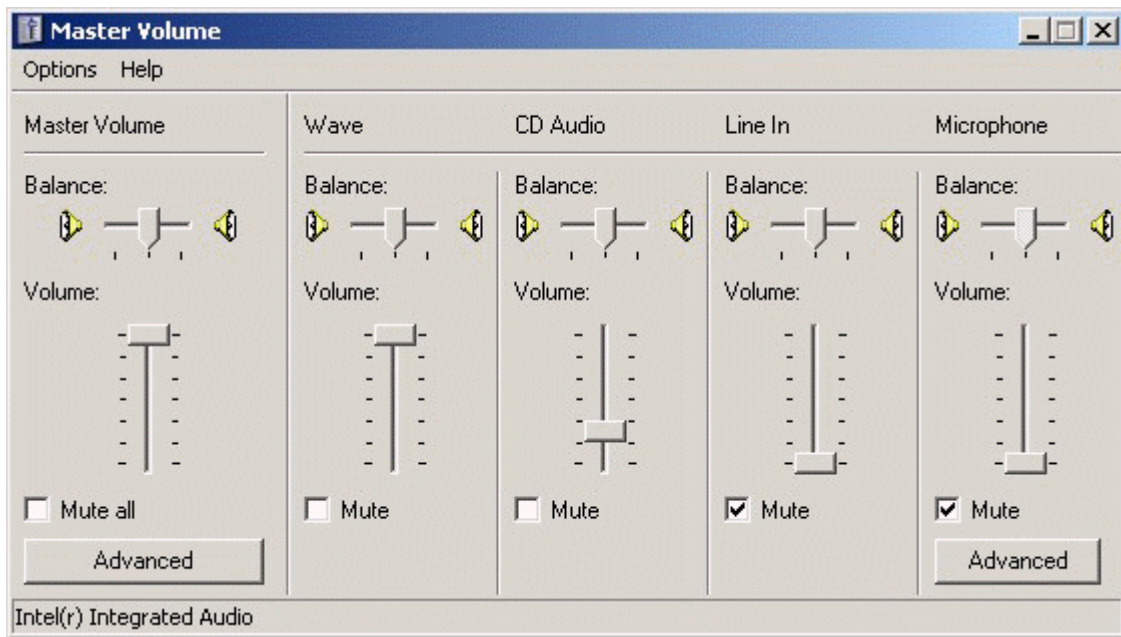**Figure 4.7** Dialog for choosing soundcard and input/output channels

**Figure 4.8** Typical setup of a soundcard output mixer in Windows XP

**Note: Most professional audio soundcards have their own program for adjustment of input and output channel, or have hardware control of input monitoring, and input and output volume controls.**

## 4.2.2 Vista / Windows 7 WDM driver setup

Microsoft has changed their approach in control of sound devices in Vista / Win7. Now operating system (also, sometimes in conjunction with control programs of professional soundcards) is responsible for setting soundcard native sampling rate and bit resolution.  Operating system changes native resolution to floating point format for high quality mixing and eventually for the sample rate conversion.
For LIMP, this means that it is strongly recommended to use '**Float**' resolution and sets the sampling rate to the native format. Access to these values is in '**Windows sound control panel**', which user gets by clicking on the button '**Control Panel**' in '**Audio Device Setup'** dialog**.**

Fig. 4.9 shows Vista/7 control panel, that has four property pages.

As first step, user has to adjust '**Playback page'** and later repeat the same procedure for '**Recording page'**. Adjustment steps are:

1)  Click on channel info to choose a playback channel. It is not recommended to use the measurement channel as a default audio channel.
2)  Click on button '**Properties**' to opens channel '**Sound properties**' dialog.
3)  Click on the tab '**Levels**' to open the output mixer (as in Fig. 4.10).  Then mute Line In and Mic channels, if exist.
4)  Click on the tab '**Advanced'** to set the channel resolution and a sample rate (as in Fig. 4.11)
5)  Repeat previous procedure 1) to 4) for recoding channel, and choose the same sampling rate as in the playback channel.
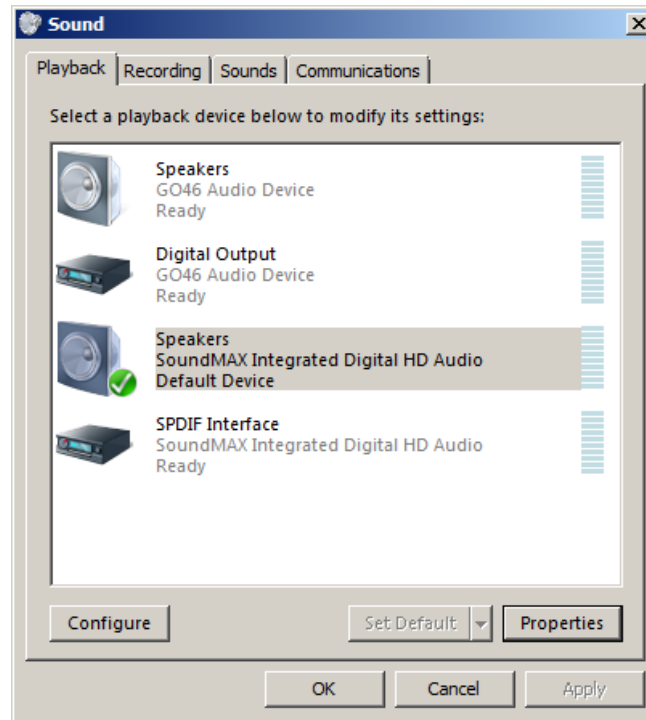
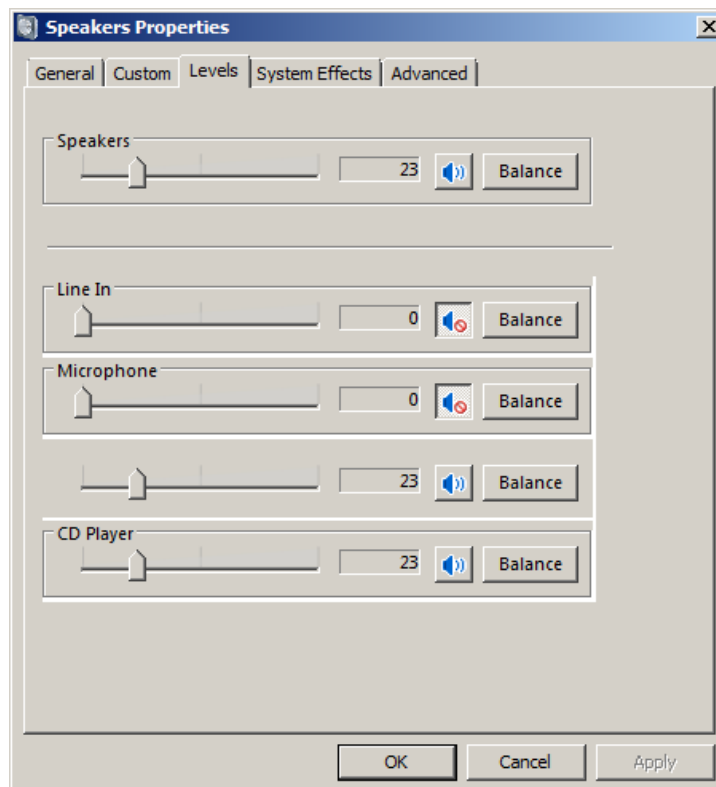**Figure 4.9** Vista Sound Control panel



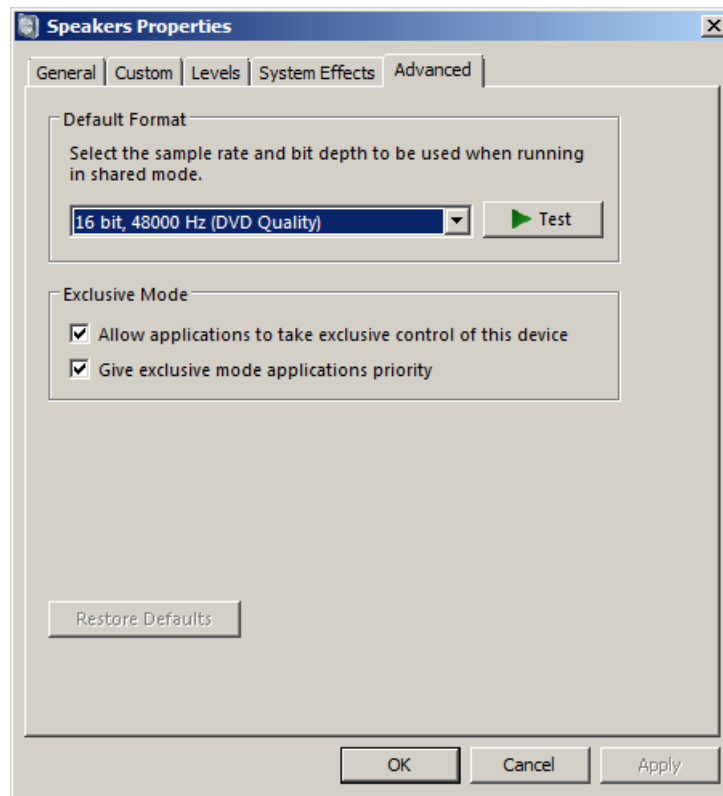**Figure 4.10** Playback channel properties – Output levels

**Figure 4.11** Setting the native bit resolution and sampling rate in Vista

**Note**: There are a lot of drivers that do not work stable under Windows 7. In that case please use ASIO driver if it is available for your soundcard.

## 4.2.3 ASIO driver setup

ASIO drivers are decoupled from the operating system control. They have their own control panel to adjust native resolution and memory buffer size. The buffer is used for the transfer sampled data from the driver to the user program. User opens the ASIO control panel by clicking button '**Control Panel**' in the '**Audio Device Setup**' dialog. Fig. 4.12 shows an example of ASIO control panel.
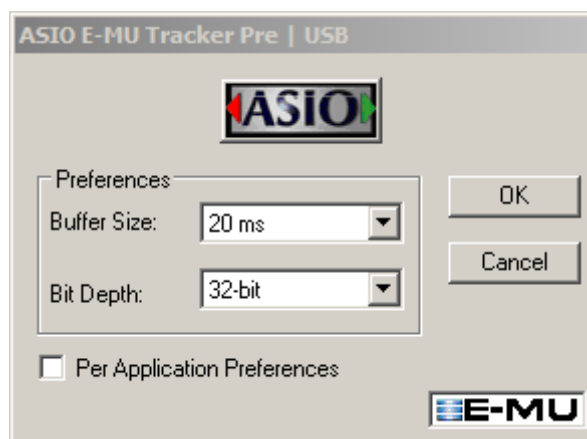


**Figure 4.12** E-MU Tracker Pre ASIO Control panel for setting bit-resolution and buffer size

In music applications user usually sets buffer size as small as it is possible for the stable work. That gives the lowest input/output latency (system introduced delay).

In Limp, the latency is not problem, as it is encountered in software, but it is not recommended to use buffer with size larger than 2048 samples, or smaller than 256 samples. Some ASIO control panels express the buffer size in samples, while other express the buffer size in time [ms]. In that case we can calculate the size in samples using following expression:

buffer_size [samples]  = buffer_size[ms] *samplerate[kHz] / number_of_channels.

Some ASIO drivers allow setup of buffer size (in samples) that is a power of number 2 (256, 512, 1024, …). In that case Limp adjusts buffer size automatically.

Limp always work with two input channels, and two output channels, treating them as a stereo left and right channels. As ASIO support multichannel devices, user has to choose in a dialog box '**Audio Device Setup**'' which pair of channels will be used (1/2, 3/4, …).

## 4.3 Generator Setup

Two types of excitation signals are possible in the *LIMP*:

- Sine
- Pink periodic noise (Pink PN).

Choose the signal type in the top dialog bar or in the **'Generator Setup'** dialog box shown in Fig. 4.13. You get it by clicking the menu **Setup -> Generator** or by clicking the toolbar icon .
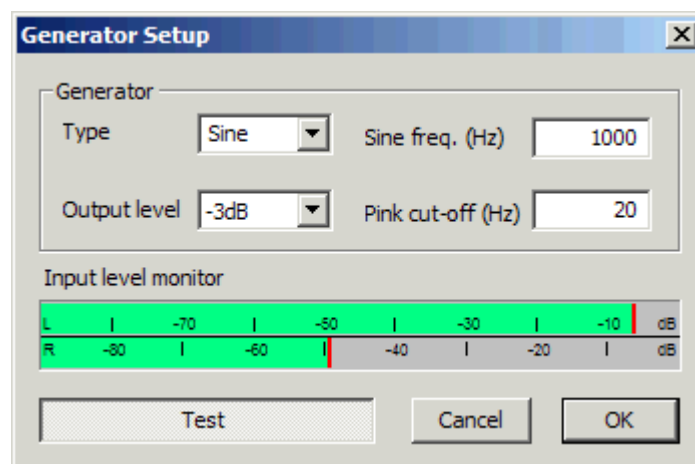
**Figure 4.13** Signal generator setup

The dialog **Generator Setup** has following controls:

> **Type -** chooses the excitation signal: **Sine** or **Pink PN**.
> **Sine freq. (Hz)**  -  enter test frequency of the sine generator.
> **Pink cut-off (Hz)**  -  enters the low-frequency cut-off of the pink noise generator.
> **Output level (dB)**   -  chooses the output level (0-15dB).
> **Test  -**  starts/stops generator with current settings.
> **Input level monitor**  -  the input volume peak meter.  Meter bars are shown in following colors:
>> **green** (for levels bellow -3dB),
>> **yellow** (for levels between -3dB and 0dB) and
>> **red** (for input overloaded).

**Recommendation:**

- For most reliable results use sine generator, but don't push loudspeaker into large displacement (the largest displacement is at frequencies below the loudspeaker resonance frequency).
- For measurements of bass or mid-bass loudspeakers set the Pink cut-off frequency close to the loudspeaker resonance frequency (20-100 Hz).
- Press the button **Test** to monitor input/output levels. If peak meter bars are in red color input channels are overloaded, then lower the output volume until bars go to yellow or green color.

## *4.4 Measurement Setup*

For setup of measurement parameters use the **'Measurement setup'** dialog box, shown in Fig. 4.14. You get it by clicking the menu **Setup->Measurement** or by clicking the toolbar icon  .
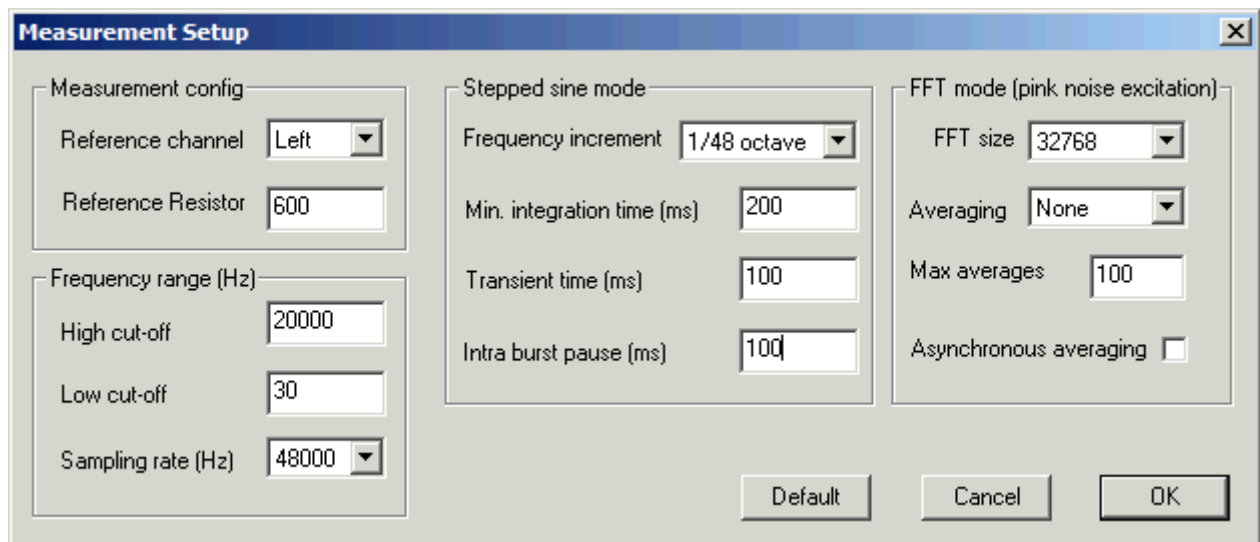


**Figure 4.14** Measurement setup

The dialog ' **Measurement setup** ' has following controls:

In section **Measurement config:**
  **Reference channel** - sets the reference channel ($U_1$) to: Left or Right.
  **Reference resistor** - enters the value for the referential resistor.

In section **Frequency range:**
  **High cut-off** - enters measurement upper frequency margin.
  **Low cut-off** - enters measurement lower frequency margin.
  **Sampling rate (Hz)** - chooses the sampling rate (from 8000 to 96000Hz).

In section **Stepped sine mode:**
  **Frequency increment** - sets to 1/24 or 1/48 octave
  **Min. integration time (ms)** - enters minimal integration time (the higher value gives higher noise reduction).

**Transient time (ms)** - enters transient time to allow the system to get to the steady state (before starting the integration).
**Intra burst pause (ms)** - enters time needed for system to release the energy from reactive components.

In section **FFT mode (pink noise excitation):**
**FFT size** - chooses the length of the FFT block ( 32768 or 65536).
**Type** - chooses: None, Linear or Exponential averaging.
**Max. Averages** - enters the maximum number of averaging for the 'Linear' mode.
**Asynchronous averaging** - check box to use the asynchronous averaging.

## *4.5 Measurement Procedures*

After you have done the audio device setup, the generator setup and the measurement setup you are ready for measurements.

**a) Procedure in FFT mode**
Connect loudspeaker in test fixture (Fig. 3.1 or 3.2) and click the menu **Record->Start** or click the toolbar icon ![play]. Measurements will be periodically repeated and results shown as an impedance plot. You will get the graph like one shown in Fig. 4.15. You can stop the measurements by clicking the menu **Record->Stop** or by clicking the toolbar icon ![stop].

You can copy the graph bitmap by clicking the menu **Edit->Copy** or by clicking the toolbar icon ![copy].
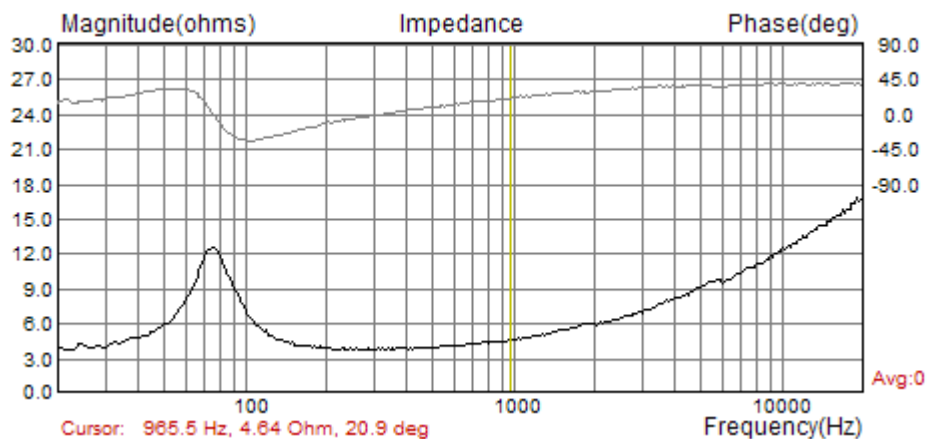


**Figure 4.15** Impedance measurement without averaging – a noisy measurement

If you set averaging to **Linear**, measurements will be repeated and averaged until the number of averaging reach the predefined value for **Max averages** in the '**Measurement setup'** dialog box. You can stop the averaging at any time by clicking the menu **Record->Stop** or by clicking the toolbar icon ![stop]. You will get the graph like one shown in Fig. 4.16.

You can also choose the exponential averaging. It differs from linear averaging in a way that it gives more weight to results from last five measurements.
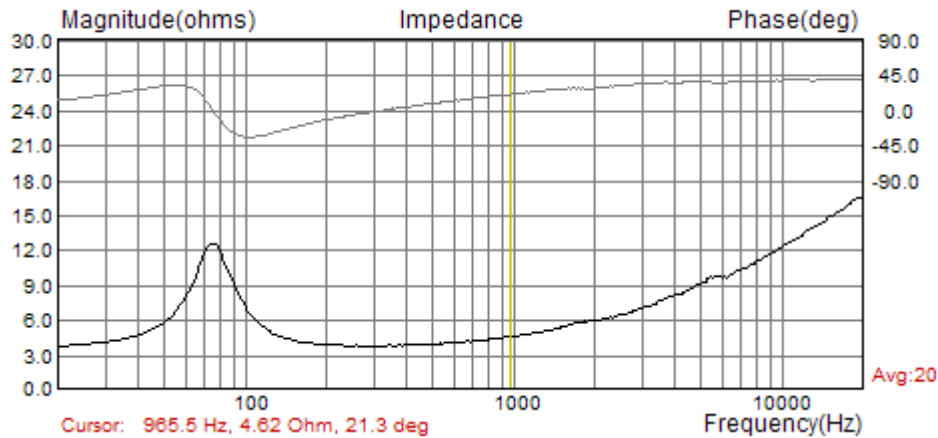
**Figure 4.16** Impedance measurements with averaging

**a) Procedure in Stepped sine mode**

Measurement procedures for stepped sine mode are almost the same as in FFT mode. The only difference is that in FFT mode user will see the impedance plot for full measured frequency range almost instantly, while in stepped sine mode procedure will be repeated for many frequencies. Cursor will show current progress in measurements, and that process will be very slow.

## 4.6 Graph Setup and Browsing

The menu command **Setup->Graph Setup** (or by *right-clicking mouse in the plot area*), opens the **'Graph Setup'** dialog box, shown in Fig. 4.17. Use this dialog box to adjust the impedance magnitude range shown and the frequency range shown.



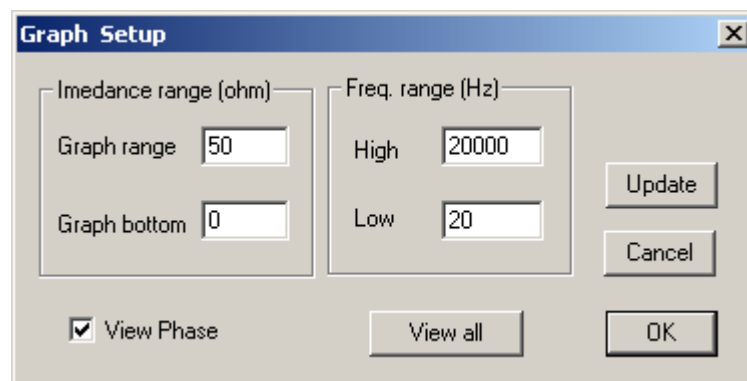**Figure 4.17** Graph setup

The '**Graph Setup**' dialog box has following controls:

**Impedance range (ohm)** section:
  **Graph range** – enters the impedance magnitude range.
  **Graph bottom** – enters the impedance magnitude for graph bottom margin.

**Freq. range (Hz)** section:
  **High** – enters the highest frequency shown (in Hz).
  **Low** – enters the lowest frequency shown (in Hz).

**View All** – enables the view of all DFTspectrum components that are used in impedance estimation.
**View Phase** – enables a phase plot.
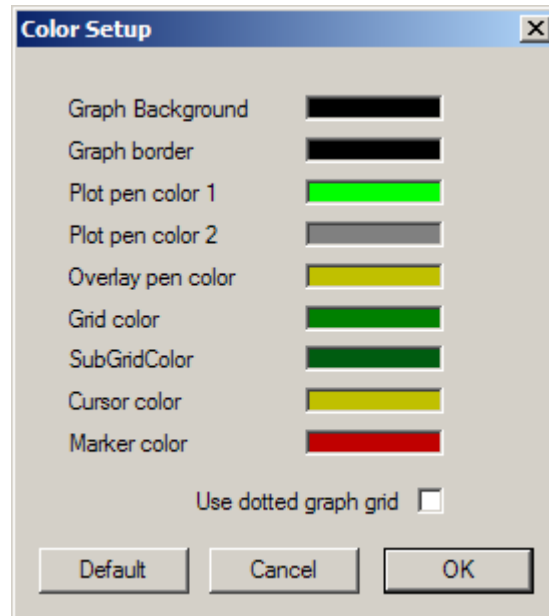**Update** – updates the graph with a new setup.



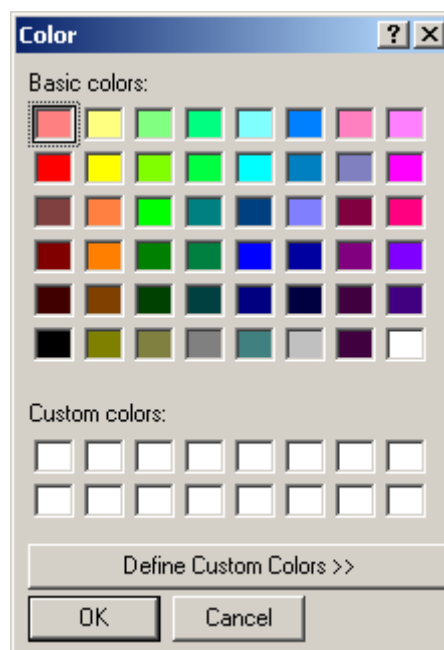**Figure 4.18** Dialog boxes for a graph colors setup



**Figure 4.19** Standard Windows dialog for color setup

Graph colors can be changed in two ways. The first one is to change the background color from
"Black" to "White" by clicking the menu command **View->B/W color** or by clicking the toolbar icon
.

The second way to change graph colors is a "user mode". User sets an arbitrary color for every graph element using the **'Color Setup'** dialog box, shown in Fig. 4.18. This dialog can be activated by clicking the menu **Edit->Colors**.

Clicking the left mouse button on colored rectangle opens the standard Windows dialog box '**Color**' (Fig. 4.19). Button '**Default**' restores default colors.

A check box '**Use dotted graph grid**' enables drawing of grid with dotted line style.

We can browse the graph by moving the cursor. At the bottom of the graph the label '**Cursor:**' denotes values of the magnitude and the phase, at the cursor position.

## *4.9 Calibrated Measurements*

In previously described measurements we have assumed that both soundcard input channels have the same sensitivity, but usually it is not the true. If the difference in sensitivity of the input channels is larger than 0.2 dB, we have to make a calibration.

The *LIMP* has a built in calibration procedure. It has to be done for a particular generator sequence length and a sampling rate.

**In the calibration procedure, both input channels have to be connected to the generator output.**

By clicking the menu **Record->Calibrate**, or by clicking the toolbar icon CAL, we get the **'Calibrate Input Channels'** dialog box, shown in Fig. 4.20.
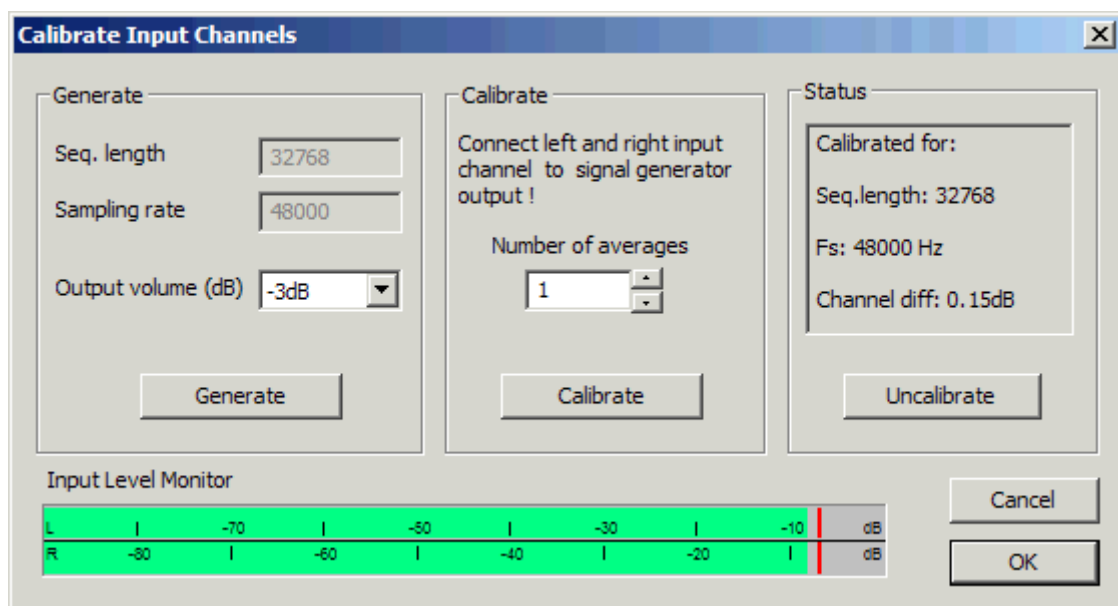


**Figure 4.20** Dialog for calibration of input channel sensitivity

In section **Generate** we set the output volume for Pink PN signal of known sampling rate and sequence length. Pressing the button **'Generate'** activates the signal generation and monitoring of input levels on the Input Level Meter. If necessary, we can adjust the output volume by using the combo box '**Output volume**'.

Finally, in section **Calibrate** we enter the **Number of averages** (recommended value is 1, as we measure high level signals) and press the button **'Calibrate'**. After a second, we have the calibrated system.

**Attention:** If we change the sequence length or a sampling rate, we have to repeat the calibration procedure.

The last section shows a report of the calibration procedure. The most important is information is **Channel difference** in dB. If the Channel difference is larger than 2 dB LIMP gives a warning as in figure 4.21, and calibration is rejected.
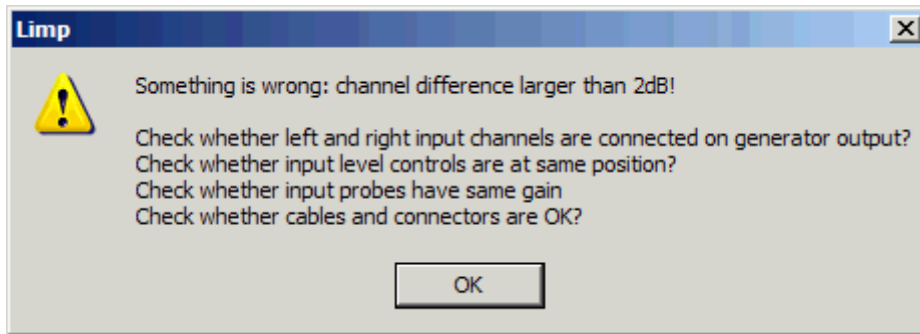


**Figure 4.21** Report when channel difference is larger than 2dB.

## 4.8 File Manipulations

Impedance data can be saved in *LIMP* binary format (.LIM files) and in ASCII format (.TXT or .ZMA files).

Beside impedance data .LIM files can contain user defined text of arbitrary length. The user can enter any text in edit box of the **File Info** dialog box (see Fig. 4.22). This dialog box can be opened by clicking menu command '**File->Info**'.



**Figure 4.22** Dialog for file information

Textual files have extension .txt or .zma. They contain lines of text with three numeric literals: frequency (in Hz), magnitude (in ohms) and phase (in degrees). The line ends with new line character.

The only difference between .txt and .zma files is that .txt files can contain comment lines. Comment lines start, after arbitrary number of space or tab characters, with nonnumeric characters, while data lines start with digit or dot character.
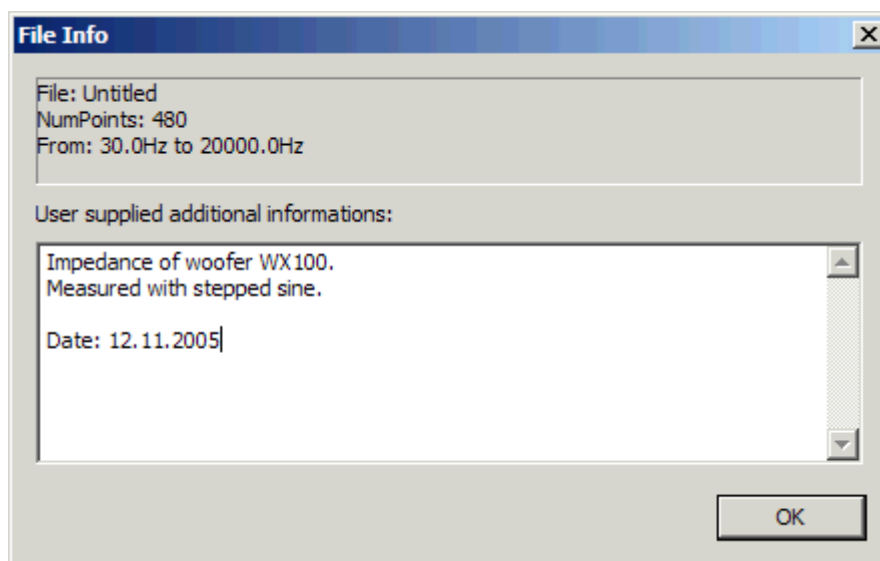
The file can be loaded as regular impedance data or as data for the overlay plot . Only one overlay is allowed.

.LIM binary file format is as follows:

```
char id[4];              // four characters id {'L', 'I', 'M', '\0' };
unsigned version;        // version number - started from 0x0101
unsigned reserved;       // 0
int numdata;             // number of data points
int cursorpos;           // last position of cursor
int fftlen;              // length of FFT in FFT mode
float  fs;               // sampling frequency
float data[3*numdata]    // contains  frequency, magnitude and phase
int infolength;          // info string length
char string[infolength]; // info string data
```

# 5 Loudspeaker parameters

This chapter gives some definitions and measurement procedures for the estimation of loudspeaker parameters.

## *5.1 Definition of physical and dynamical loudspeaker parameters*

An electrodynamic loudspeaker that is mounted in the infinite baffle is usually ([10], [11]) characterized by the following physical parameters:

*Electromagnetic parameters:*
$R_E$       - voice coil DC resistance ($\Omega$)
$L_E$       - voice coil self-inductance (H)
$L_2$       - inductance due to inductive coupling of eddy currents (H)
$R_2$       - resistance due to eddy currents ($\Omega$)
$Bl$       - force factor (Tm)

*Mechanical parameters:*
$S$       - effective area of membrane (m$^2$)
$C_{MS}$       - membrane mechanical compliance (m/N)
$M_{MS}$       - mechanical mass of membrane plus mass of air load on membrane (kg)
$R_{MS}$       - mechanical resistance plus membrane radiation resistance (kg/s)

(**Note:** The piston area $S$ is normally obtained from a cone diameter measurement that includes the 1/3 of the surround.)

Fig. 5.1 shows wideband and low-frequency equivalent circuits of an electrodynamic loudspeaker that is mounted in a infinite baffle. The circuit for definition of low-frequency input impedance uses following elements:

$$L_{CES} = \left(Bl\right)^2 C_{MS}, \quad R_{ES} = \left(Bl\right)^2 / R_{MS}, \quad C_{MES} = M_{MS} / \left(Bl\right)^2$$
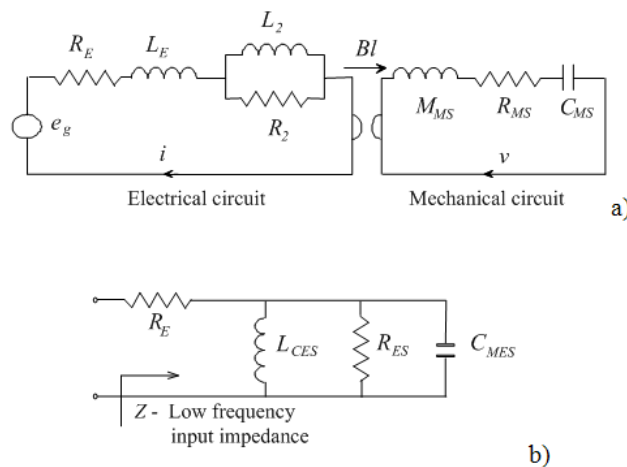


**Figure 5.1** a) Wideband analogous electrical circuit of an electrodynamic loudspeaker that is mounted in the infinite baffle, and b) circuit for the estimation of the low- frequency input impedance

Using these analogous circuit elements, Thiele and Small [6, 7] introduced *dynamical* loudspeaker parameters. They are defined in the Table 5.1.

| | |
|---|---|
| Resonant frequency in free air (Hz) | $f_s = \dfrac{1}{2\pi \sqrt{M_{MS} C_{MS}}}$ ,   $\omega_s = \dfrac{1}{\sqrt{M_{MS} C_{MS}}}$ |
| Mechanical Q-factor | $Q_{MS} = \dfrac{\omega_S M_{MS}}{R_{MS}} = \dfrac{1}{\omega_S C_{MS} R_{MS}}$ |
| Electrical Q-factor | $Q_{ES} = \dfrac{\omega_S M_{MS} R_E}{(Bl)^2}$ |
| Total Q-factor | $Q_{TS} = \omega_S C_{MES} \dfrac{R_E R_{ES}}{R_E + R_{ES}} = \dfrac{Q_{MS} Q_{ES}}{Q_{MS} + Q_{ES}}$ |
| Power available efficiency  (%) | $\eta_0 = \dfrac{\rho_0}{2\pi c} \dfrac{S^2 (Bl)^2}{R_E M_{MS}^2}$ |
| Sensitivity  (1W/1m)  in  dB | $L_p(1W/1m) = 10\log\left(\dfrac{\rho_0 c}{2\pi} \eta_0\right) - 20\log(p_{ref}) = 112{,}1 + 10\log(\eta_0)$ |
| Equivalent acoustical volume (m$^3$) | $V_{AS} = \rho_0 c^2 S^2 C_{MS}$ |

**Table 5.1**  Thiele-Small dynamical loudspeaker parameters
( $\rho_0$ = 1.18kg/m$^3$, $c$ = 345m/s, $p_{ref}$ = 20μPa)

Thiele and Small have shown that by using these parameters it is easy to express the response of closed box as 2$^{nd}$ order high-pass filter and response of bas-reflex box as 4$^{th}$ order high-pass filter. Today, almost every loudspeaker manufacturer gives physical and Thiele-Small parameters in their loudspeaker data sheets.

**Note on electrical inductance:**
An industry standard is to measure voice coil inductance at 1 kHz. In the *LIMP*, an inductive impedance component (as well capacitive one) can be measured at any frequency, as shown in chapter 4.6.
The truth is that we cannot specify the exact value of the voice coil inductance $L_E$, because it is a circuit element that depends on frequency. The voice coil inductance can be better described as a primary coil of the "transformer" whose secondary winding is the pole piece. Eddy currents in the pole piece give rise to primary circuit resistance. The simplest model [8] for a primary input impedance is K√ω + jK√ω, where K is some constant. The impedance increases with √ω rather than ω.

This model is not practical as engineers in many numerical simulations use  some form of an analogous electrical circuit that closely matches measurement data. The most commonly used circuit for the electrical voice coil impedance is serial connection of resistor $R_E$, inductor $L_E$ and parallel connection of resistor $R_2$ and inductor $L_2$, as shown in Fig. 5.1. It has been proven as useful model in many simulations. LIMP estimate value of $L_E$, $R_2$ and $L_2$ as values that give the least  sum of squared differences of the model impedance  and a measured impedance over all frequencies.

**The voice coil resistance $R_E$ should be measured with a DC ohmmeter.**

## *5.2 Estimation of Thiele-Small parameters*

It is easy to estimate Thiele-Small parameters if we have measured data for the loudspeaker impedance.

At low frequencies the influence of a voice coil inductance and eddy currents is small and an expression for the loudspeaker input impedance has the following form:

$$Z \cong R_E + \frac{(Bl)^2}{R_{MS}} \frac{\dfrac{s\omega_S}{Q_{MS}}}{\omega_S^2 + \dfrac{s\omega_S}{Q_{MS}} + s^2} = R_E \frac{1 + sT_S/Q_T + s^2T_s^2}{1 + sT_S/Q_{MS} + s^2T_s^2}$$

where $s = j\omega$, $T_s = 1/\omega_s$.

The impedance has maximum value at the resonance frequency:

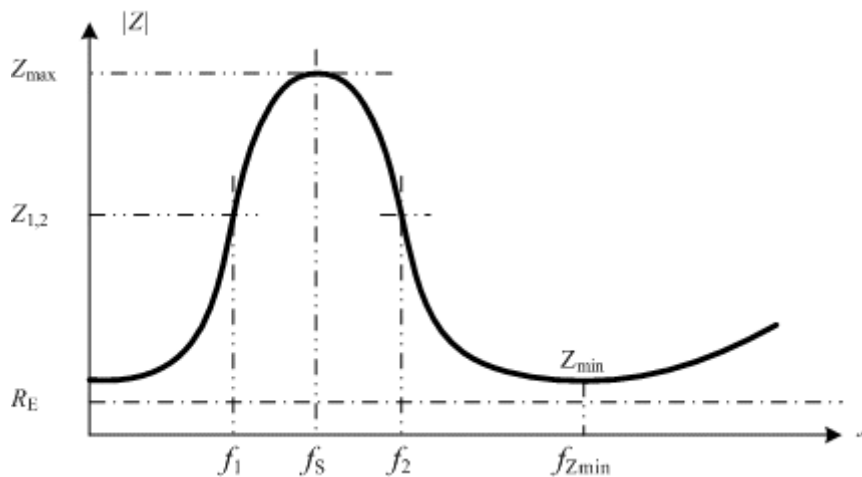$$Z_{max} = Z(\omega = \omega_S) = R_E Q_{ES}/Q_T = R_E + R_{ES}$$

because $Q_{ES} > Q_T$.



**Figure 5.2** Typical impedance curve of a loudspeaker that is mounted in a free air

At frequencies $f_1$ and $f_2$, (where $f_1 < f_S < f_2$, $f_1 f_2 = f_S^2$) impedance values are of equal magnitude;

$$|Z(j\omega_1)| = |Z(j\omega_2)| = r_1 R_E, \quad \text{if} \quad \omega_1 \omega_2 = \omega_S^2$$

If we substitute this expression in the impedance equation, we get

$$|Z(j\omega_{1,2})|^2 = r_1^2 R_E^2 = R_E^2 \frac{r_0^2 + (Q_{MS}(\omega_1 - \omega_2)/\omega_S)^2}{1 + (Q_{MS}(\omega_1 - \omega_2)/\omega_S)^2}.$$

From this equation we get the mechanical Q-factor:

29

$$Q_{MS} = \frac{f_s}{f_2 - f_1} \sqrt{\frac{r_0^2 - r_1^2}{r_1^2 - 1}}$$

Now we can define a step-by-step procedure for the measurement of $Q$ factors:

1. Measure a voice coil resistance $R_E$ with a dc ohm-meter

2. From impedance curve find $f_S$ and $Z_{max}$. Define $r_o = Z_{max}/R_E$.

3. Choose some impedance magnitude $R_E < |Z_1| < Z_{max}$ and find both frequencies ($f_1$ and $f_2$) where $Z = Z_1$. Define $r_1 = Z_1/R_E$.

4. Calculate $Q_{MS}$ (from the above equation),

5. Calculate $Q_{ES}$ using the equation $Q_{ES} = Q_{MS} / (r_o - 1)$.

6. Calculate $Q_T$ using the equation $Q_T = Q_{ES} Q_{MS} / (Q_{ES} + Q_{MS})$.


## *5.3 Estimation of physical loudspeaker parameters*

Two methods are used [7] for the estimation of physical loudspeaker parameters ($M_{MS}$, $C_{MS}$ and $R_{MS}$);

1. Added mass method,
2. Closed box method.

Both methods are implemented in the *LIMP*.


### 5.3.1 Added Mass Method

In this method, we first measure the impedance curve and estimate Thiele-Small parameters $f_S$, $Q_{MS}$ and $Q_{ES}$, for the loudspeaker mounted in a free air. Then we put an additional mass ($M_{added}$) on the membrane, measure impedance curve and estimate the shifted resonance frequency $f_M$ and electrical $Q$-factor $Q_{EM}$. From equations for $Q_{EM}$ and $Q_{ES}$ we get:

$$M_{MS} = \frac{M_{added}}{\dfrac{f_S Q_{EM}}{f_M Q_{ES}} - 1}$$

When we know $M_{MS}$ and $f_S$, it is easy to get the mechanical compliance $C_{MS}$, resistance $R_{MS}$ and force factor *Bl*. By using equations that are defined in Table 5.1 we get

$$C_{MS} = \frac{1}{M_{MS}\omega_S^2}, \quad V_{AS} = \rho_0 c^2 S^2 C_{MS}, \quad Bl = \sqrt{\frac{\omega_S M_{MS} R_E}{Q_{ES}}}, \quad R_{MS} = \frac{\omega_S M_{MS}}{Q_{MS}}$$

### 5.3.2 Closed Box Method

In this method we first measure the impedance curve and estimate Thiele-Small parameters $f_S$, $Q_{MS}$ and $Q_{ES}$, for the loudspeaker mounted in a free air. Then we mount the loudspeaker in a closed box, of known volume $V_B$, measure impedance curve and estimate Thiele-Small parameters $f_C$, $Q_{MC}$, $Q_{EC}$. From these we find

$$V_{AS} = V_B(\frac{f_C Q_{EC}}{f_S Q_{ES}} - 1)$$

Then, by using equations that are defined in Table 5.1, we get

$$C_{MS} = \frac{V_{AS}}{\rho_0 c^2 S^2}, \quad M_{MS} = \frac{1}{C_{MS}\omega_S{}^2}, \quad Bl = \sqrt{\frac{\omega_S M_{MS} R_E}{Q_{ES}}}, \quad R_{MS} = \frac{\omega_S M_{MS}}{Q_{MS}}$$

## 5.4 Automatic estimation of physical and dynamical loudspeaker parameters

*LIMP* procedures for the estimation of loudspeaker parameters are as follows.

To estimate Q-factors of a loudspeaker we need:

1. measured impedance data,
2. measured voice coil dc resistance (in ohms),
3. estimated membrane diameter in cm (cone diameter measurement that includes 1/3 of the surround).

By clicking the menu **Analyze->Loudspeaker parameters – Added mass method**, we get the **'Loudspeaker Parameters - Added Mass Method '** dialog box, shown in Fig. 5.2.

In this dialog box, we enter values for **Voice coil resistance** and **Membrane diameter**. Press on button **'Calculate'** gets the report shown in the left edit box. This report can be pasted to the clipboard by pressing the button **'Copy'**.
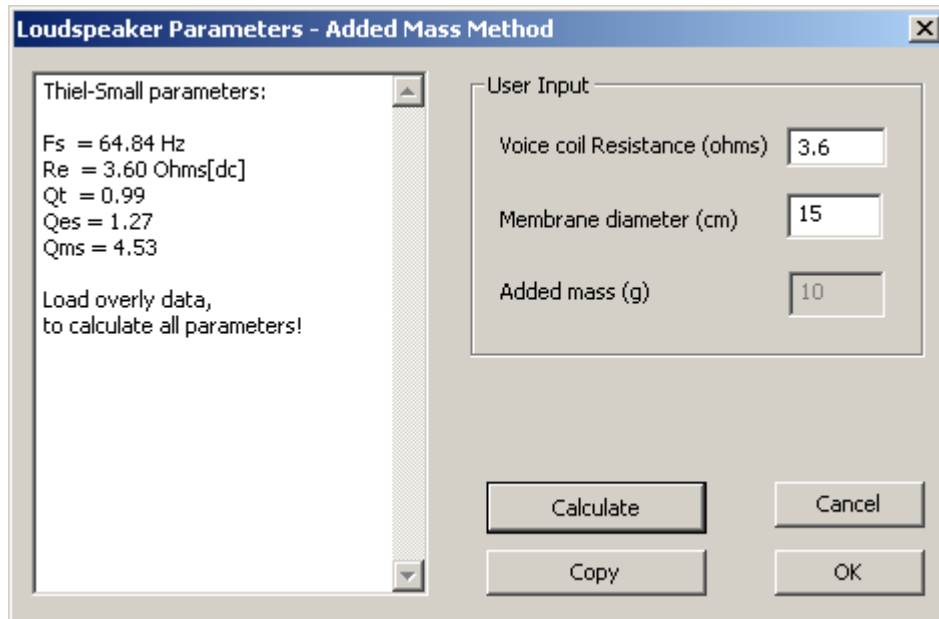
**Figure 5.2** Dialog box for the estimation of loudspeaker Q-factors

## 5.4.1 Added Mass Method

To estimate all physical and dynamical loudspeaker parameters we must make two impedance measurements of the loudspeaker in a free air. In one of the two measurements, the membrane must be loaded with an additional mass. That is what we need:

1. measured impedance data,
2. impedance data measured with an additional mass on the membrane (as in Fig. 5.3),
3. one of curves (1 or 2) must be set as an overlay graph,
4. measured additional mass - in grams,
5. measured the voice coil dc resistance - in ohms,
6. diameter of the membrane - in cm (cone diameter that includes 1/3 of the surround),
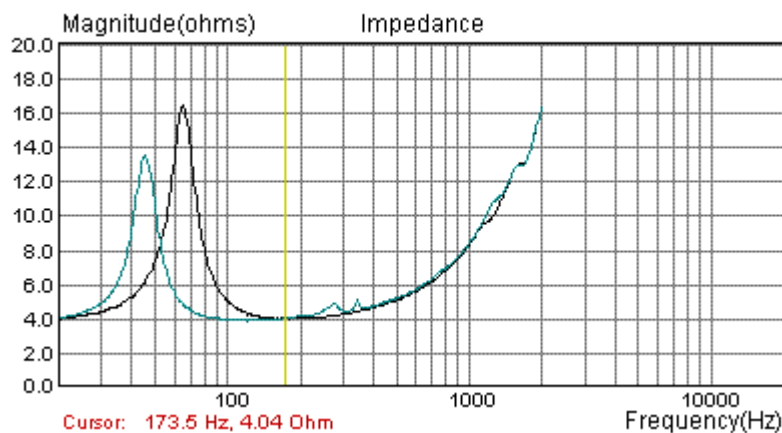


**Figure 5.3** Impedance curves (the one with lower resonance frequency is obtained by adding mass to the membrane)

Then we activate the menu **Analyze->Loudspeaker parameters – Added mass method** to get the **'Loudspeaker Parameters'** dialog box, shown in Fig. 5.4.
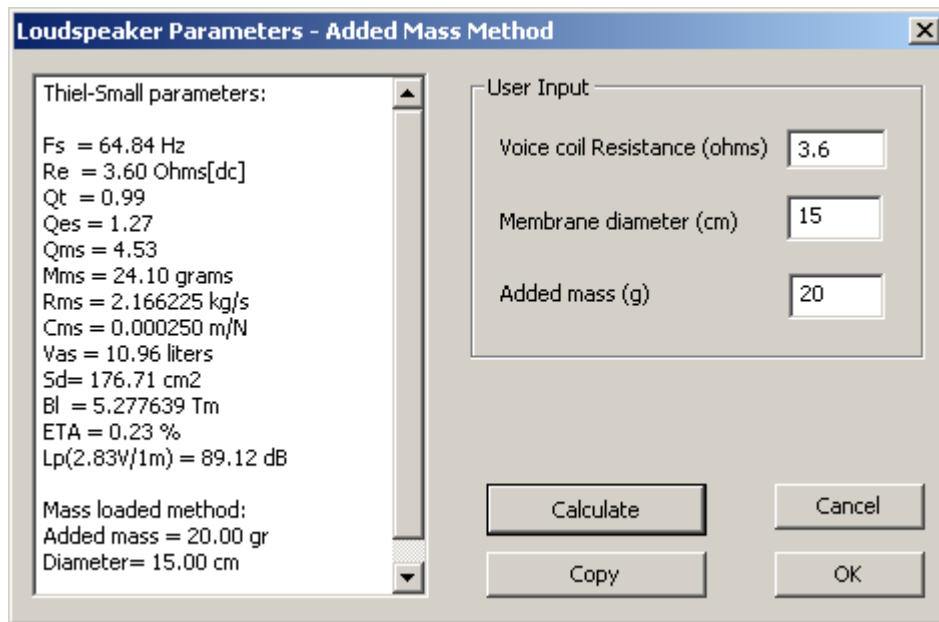


**Figure 5.4** Dialog box for the estimation of loudspeaker parameters

Finally, we enter values for **Voice coil resistance, Membrane diameter** and **Added mass**. Press on the button **'Calculate'** gets the report shown in the left edit box. This report can be pasted to the clipboard by pressing the button **'Copy'**.

## 5.4.2 Closed Box Method

To estimate all loudspeaker parameters we must have two impedance measurements. In one of measurements the loudspeaker must be mounted in a closed box of known volume. That is what we need:

1. measured impedance data,
2. impedance data measured with loudspeaker mounted in a closed box,
3. one of curves (1 or 2) must be set as an overlay graph (as in Fig. 5.5),
4. estimated value of box volume - in liters,
5. measured the voice coil dc resistance - in ohms,
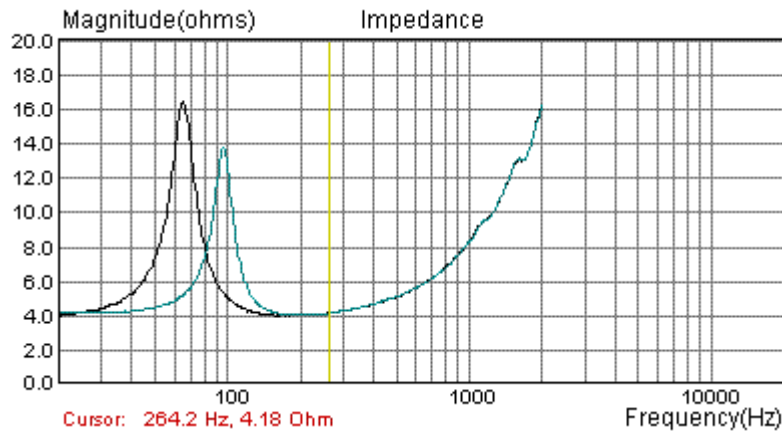6. diameter of the membrane - in cm (cone diameter that includes 1/3 of the surround),

**Figure 5.5** Impedance curve (the one with higher resonance frequency is obtained by mounting the loudspeaker in a closed box)

Then we activate the menu **Analyze->Loudspeaker parameters – Closed box method** to get the **'Loudspeaker parameters – Closed Box Method'** dialog box, shown in Fig. 5.6.
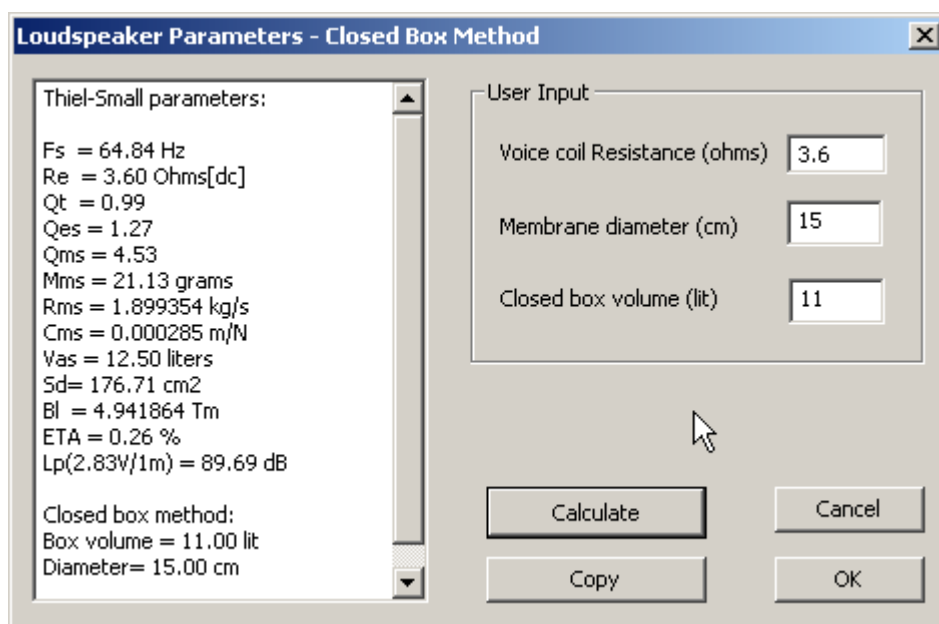


**Figure 5.6** Dialog for the estimation of loudspeaker parameters using Closed Box Method

Finally, we enter values for **Voice coil resistance, Membrane diameter** and **Closed box volume**. Press on the button **'Calculate'** gets the report shown in the left edit box. This report can be pasted to the clipboard by pressing the button **'Copy'**.

# 6 RLC measurement

The *LIMP* can be used to measure value of resistors, capacitors and inductors, simply by calculating resistive, inductive or capacitive parts of the measured impedance. For example, Fig. 6.1 shows impedance curves of an inductor with nominal value of 1.5mH.
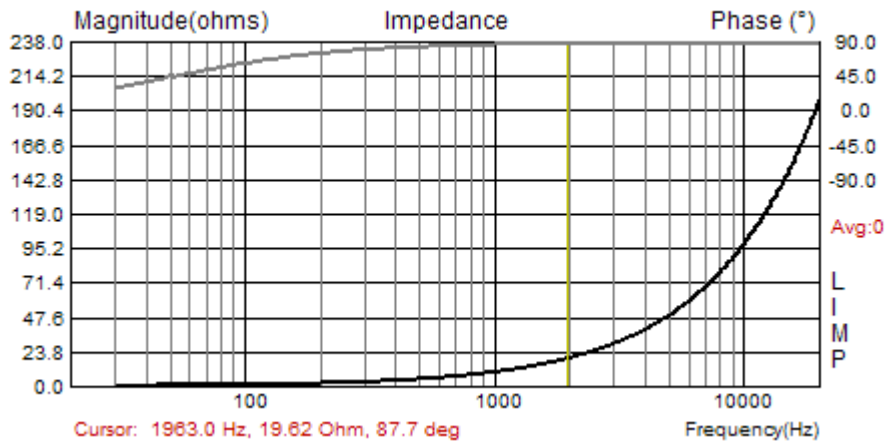


**Figure 6.1** The impedance graph of a 1.5mH inductor

By clicking menu command **Analysis->RLC Impedance value at cursor position** we get the dialog box with report as shown on Figure 6.2. LIMP reports that measured impedance has resistive part of 0.776 ohms and imaginary part is inductive with value of 1.589mH.
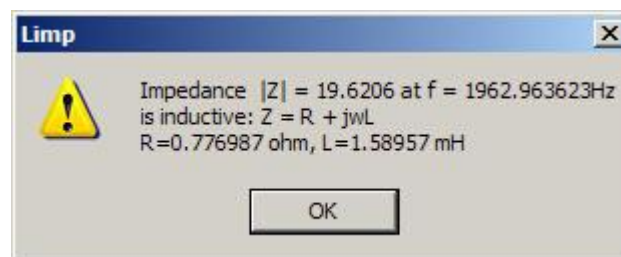


**Figure 6.2** Impedance of an air core inductor

The same way LIMP measures the capacitance.

## *6.1 Importance of calibration*

**When measuring impedance and capacitance it is very important to calibrate the system before the measurement, and it is best to make the calibration with the impedance connected (DUT).**

**Why?**

Even if there is a very small difference between channel sensitivities (i.e. 0.1dB) the LIMP can give very erroneous result, because inductor impedance has phase close to 90 degree, and capacitor impedance has phase close to minus 90 degree. In that case if there is a difference in sensitivity of measured generator voltage V1 and impedance voltage V2 (if sensitivity of probe V2 is larger than sensitivity of probe V1) estimated impedance gives phase values that are larger than 90 degree and graph shows warped jump in the phase for 180 degree. Figure 6.3 shows the case of measuring capacitor without the calibration. In half the range the phase is close to 90 degree. It is very erroneous results as it suggest that we deal with an inductance. Figure 6.4 shows capacitor impedance after the calibration. We see correct values for phase in the whole frequency range.
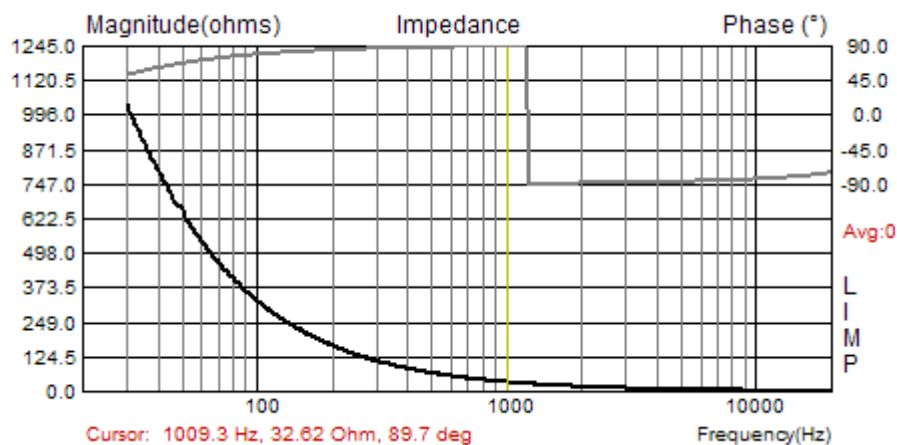


**Figure 6.3** Wrongly estimated impedance of plastic capacitor 4.7uF/250V (measured without the calibration)
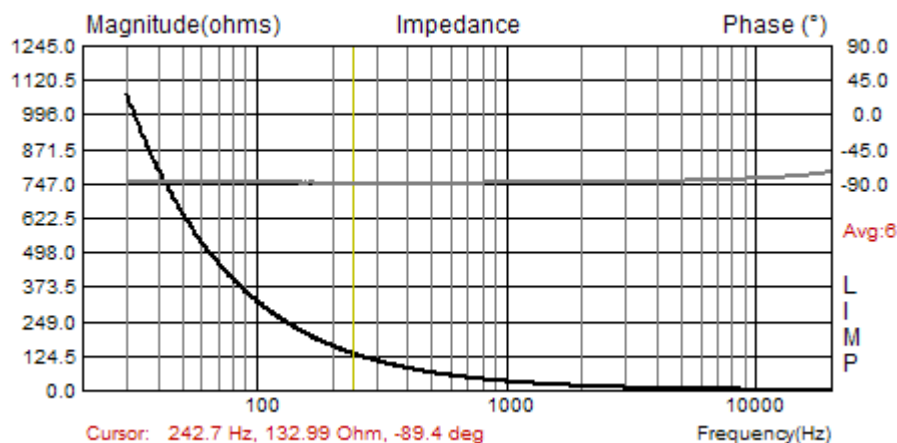


**Figure 6.4** Correctly estimated impedance of plastic capacitor 4.7uF/250V (measured after the calibration)

Not all LIMP users will have this problem. As explained before, the problem exists when probe for impedance voltage V2 has higher sensitivity than it is the sensitivity of probe for generator voltage V1. To circumvent this condition we can change probes sensitivity or we can simply exchange input channels and also change the reference channel in the LIMP measurement setup.

**Note:**
To get good estimation of capacitance or inductance it is recommended to put cursor on frequency where impedance magnitude is lower than 100 ohms. It assures measurement of impedance with 1% tolerance. Why? The answer lies in fact that measured impedance is bypassed with probe or soundcard input impedance that usually has value of 10k ohms.
A lot of professional audio sound systems (like RME Fireface, EMU 0404USB,  Yamaha G046) have unbalanced instrumentation inputs with input impedance in the range 470-1000kΩ. Using these systems, a much better accuracy is possible.

# Literature

[1]   Mateljan, I., "Signal Selection for the Room Acoustics Measurement", *Proc. 1999 IEEE Workshop on Applications of Signal Processing to Audio and Acoustics*, New Paltz, New York, 1999.

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