LIMP

Program for Loudspeaker Impedance Measurement

User Manual

Version 1.9.8

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Content

1 What is LIMP?

LIMP is a program for the measurement of loudspeaker impedance and estimation of the loudspeaker's physical and dynamical parameters (also called Thiele-Small parameters). It is also a general-purpose program for measuring impedance in the audio range.

Requirements to use LIMP are:

- o Operating systems: Windows Vista / 7 / 8 / 10 /11
- o Processor class Pentium, clock frequency 1 GHz or higher, 2GB memory for 32bit or 4GB for 64bit Windows
- 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 installation of LIMP is through a 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.

LIMP does not dump graphs to the printer, instead, all graphs can be copied to the Windows Clipboard and pasted to other Windows applications or saved to files in .BMP, .PNG or .JPG formats.

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 reference resistor *R* is connected between the signal generator and a loudspeaker impedance Z.
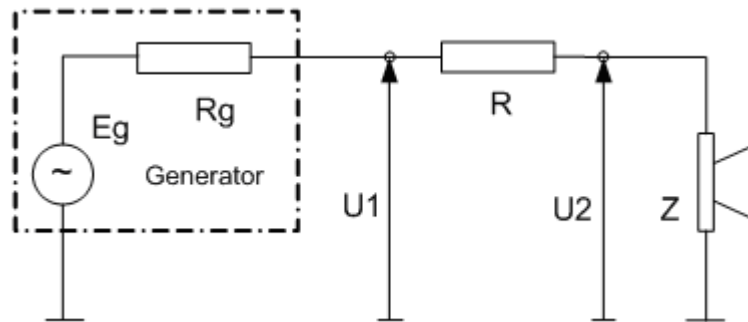

Figure 2.1 A circuit for loudspeaker impedance measurement

Impedance is defined in the frequency domain as *Z* (*f*). In Fig. 2.1 we denote U_1 (*f*) as *reference voltage on generator output* and U_2 (*f*) as *voltage on measured impedance*. These voltages define current that flow through reference resistor *R* as $(U_1 (f) – U2 (f)) / R$, and we can estimate the impedance using expression:

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

2.2 Stepped Sine and FFT mode of measurements

LIMP has two modes of impedance measurement:

   1) **Stepped sine mode**

   2) **FFT mode** with multitone or pink periodic noise (Pink PN) excitation

In the stepped sine mode LIMP generates a pure sinusoidal signal, frequency by frequency with 1/24 or 1/48 octave increment, and measures response to sinusoidal signal by filtering out noise and distortion components in sinusoidal response. It is the preferred mode for impedance measurement, but the measurement time is very long.

FFT mode is a faster method for impedance measurement, but it gives a much lower measurement S/N ratio than in stepped sine mode. In FFT mode LIMP simultaneously measures the values of the impedance in the whole audio range of frequencies. It allows for two kind of wideband excitation signals: multitone or wideband pink PN.

The wideband pink PN and multitone signals are realized as periodic random phase multi-sine signals. Both are a zero dc, periodic signals that contain *N* sine components, each with a random phase:

$$g(t) = \sum_{k=1}^{N-1} A_k \cos(2\pi f_k t + \phi_k), \quad \phi_k = random \in [0, 2\pi]$$

The pink PN has frequency components on all the FFT frequency bins ($f_k = k \, f_s/N$) and its spectral magnitudes roll-off at -3 dB/octave ($A_k = \sqrt{2}A_{2k}$) after specified cut-off frequency.

In 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 the loudspeaker has an impedance maximum (20-100 Hz). The pink PN signal has the statistical distribution close to the normal distribution.

Multitone signal, as defined in the IEC 60268-21 standard, is the sum of random phase spectral components that have equal magnitude (A_k =const.), but they are sparsely generated only on FFT frequency bins that are approximately 1/n-octave apart ($f_{k+1} = f_k \, 2^{1/n}$). In Limp the resolution of 1/24 octave is used if FFT size is less than 64K, otherwise the resolution is 1/48 octave.

Although multitone and pink PN are mathematically similar signal, they sound very different and are used for different purposes. Pink PN is always generated as a wideband signal while the multitone spectrum can be user defined in a restricted frequency range [*Fstart*, *Fstop*].

In LIMP multitone and pink PN signals are generated with a crest factor slightly larger than 12 dB. The sine signal has crest factor 3 dB. Signals generated using D/A converters have limited peak signal value, which means that maximal total RMS value of all multitone components is about three time smaller (9 dB) than maximal RMS of sine signal (for the same peak value). If we generate multitone with resolution of 1/48-octave it means that in audio range there will be 480 multitone components each with RMS value that is sqrt(480) times lower than multitone total RMS, or 3*sqrt(480) lower than sine rms value. **This reasoning leads us to the conclusion that measurement in stepped sine mode theoretically has 36 dB better S/N ratio**.

2.3 Simple Measurement Procedure

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

1. LIMP generates a pure sine signal starting with *f* = *Fstart*. We assume that this is output signal of the generator shown in Fig. 2.1.

2. After an arbitrary *transient time*, that is necessary to reach the steady state, voltages at both ends of the reference 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 from step 1) until the predefined stop frequency is reached.

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

1. LIMP generates the Pink PN or multitone 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 pre-averaging cycle, which is necessary to reach the steady state, voltages at both ends of the reference 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 $Z(f)$.

2.4 Measurement in a Noisy Environment

The main source of measurement noise is a loudspeaker that acts as a microphone for the environmental noise and vibrations.

Loudspeaker generated noise depends on three sources:

- First source is airborne noise that activate loudspeaker to work as microphone.

- Second source of noise is external mechanical vibration that moves loudspeaker and modulate membrane position – again we have microphonic effect.

- Third source of noise is self-vibration of loudspeaker that exists when loudspeaker is driven by large signal. This kind of noise is very strong if loudspeaker is loosely mounted, then vibration of basket and magnet nonlinearly interact with mounting body.

To get reliable results, particularly in FFT mode, measurement should be done in low noise environment, with loudspeaker firmly decoupled from vibrational sources. Also, it is beneficial to orientate membrane vertically, as it reduces gravitational force influence on membrane displacement.

Fig. 2.2 shows modified circuit for the loudspeaker impedance measurement, with the noise generator E_n included.
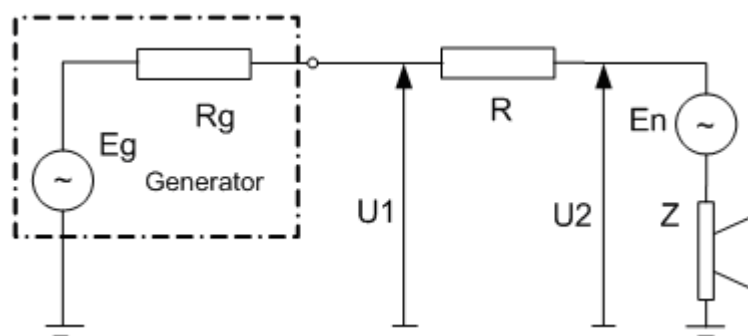

Figure 2.2 A circuit for the loudspeaker impedance measurement, with the noise generator E_n

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

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

What this equation shows is that estimated impedance differs from the true impedance Z by the term that is dependent on the S/N ratio (E_g/E_n) and values of resistors R, R_g and the impedance Z. By using additional knowledge from measurement process we can conclude:

1. The simple theory shows that the signal generator must supply a high voltage, to assure high S/N, but practical measurements have shown that it is better to use smaller output voltage. Two facts prove that. First, a larger voltage drives the loudspeaker to highly nonlinear region. Second, a larger voltage drive can enlarge driver self-vibration that can produce discontinuous noise in interaction with mounting. AES recommends voltages in the range 0.1 – 0.2 V, but even lower values are acceptable. This is especially important if we measure impedance to get data for driver parameters estimation.

2. When we use the FFT method the measurement results are highly affected with noise. The loudspeaker acts as a microphone with a highest sensitivity in the region of the membrane resonance. It means the highest level of the noise is at low frequencies, so we must generate the signal with high level at low frequencies, i.e. the pink noise, or multitone with spectral components equally distributed on log-frequency scale.

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 headphone output as a signal generator, then we can use R = 40-100 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 we still can successfully use the stepped sine method.

2.4.1 Lowering Measurement Noise in Stepped Sine Mode

In stepped sine mode LIMP uses the "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 an integration time of 200ms, then the width of the passband of the "heterodyne" filter is 5 Hz.

2.4.2 Lowering Measurement Noise in FFT Mode

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

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

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

Two types of averaging can be applied in LIMP.

- time domain synchronous averaging

- frequency domain asynchronous averaging

In a "time domain synchronous averaging" LIMP generates multiple periods of one noise sequence (Fig. 2.3).

In a "frequency domain asynchronous averaging" 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.
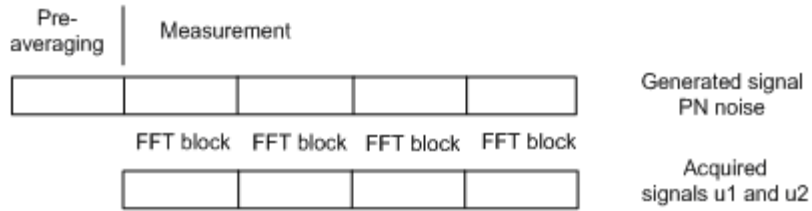

Figure 2.3 Signal generation and acquisition during the synchronous time domain averaging process

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 (headphone output) is used as a signal generator output. The soundcard right line-input is used for recording the reference voltage U_1 and the soundcard left line-input is used for recording the voltage U_2.

If the soundcard has no headphone output, then we have to connect an external power amplifier to the soundcard line-out output, as shown in Fig. 3.2.

Figure 3.1 Measurement setup for loudspeaker impedance measurements, using the headphone output
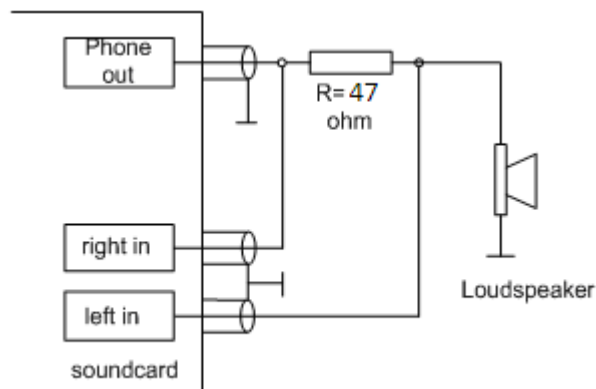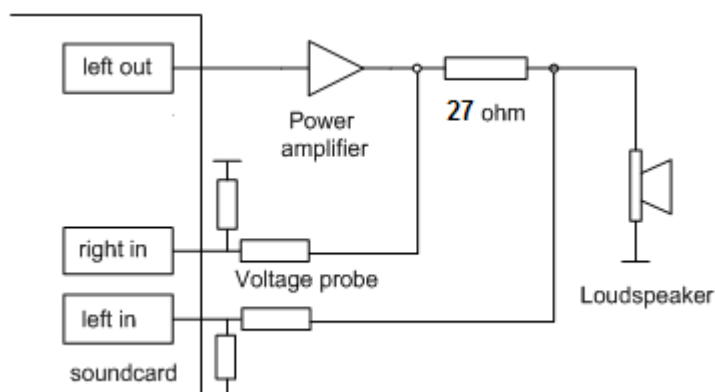

Figure 3.2 General measurement setup for loudspeaker impedance measurements

To protect the soundcard input from a large voltage that may be generated at a power amplifier output, it is recommended to use the voltage probe circuit with Zener diodes, as shown on Fig. 3.3. Values of resistors R1 and R2 have to be chosen for arbitrary attenuation (i.e. R1=8200 and R2=910 ohms give probe with -20.7 dB (0.0923) attenuation if the soundcard has usual input impedance – 10 kΩ).

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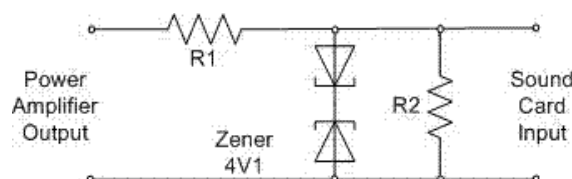
4 Working with LIMP

When you start LIMP you get the program window shown in Fig. 4.1. There are: menu bar, toolbar and a dialog bar at the 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.
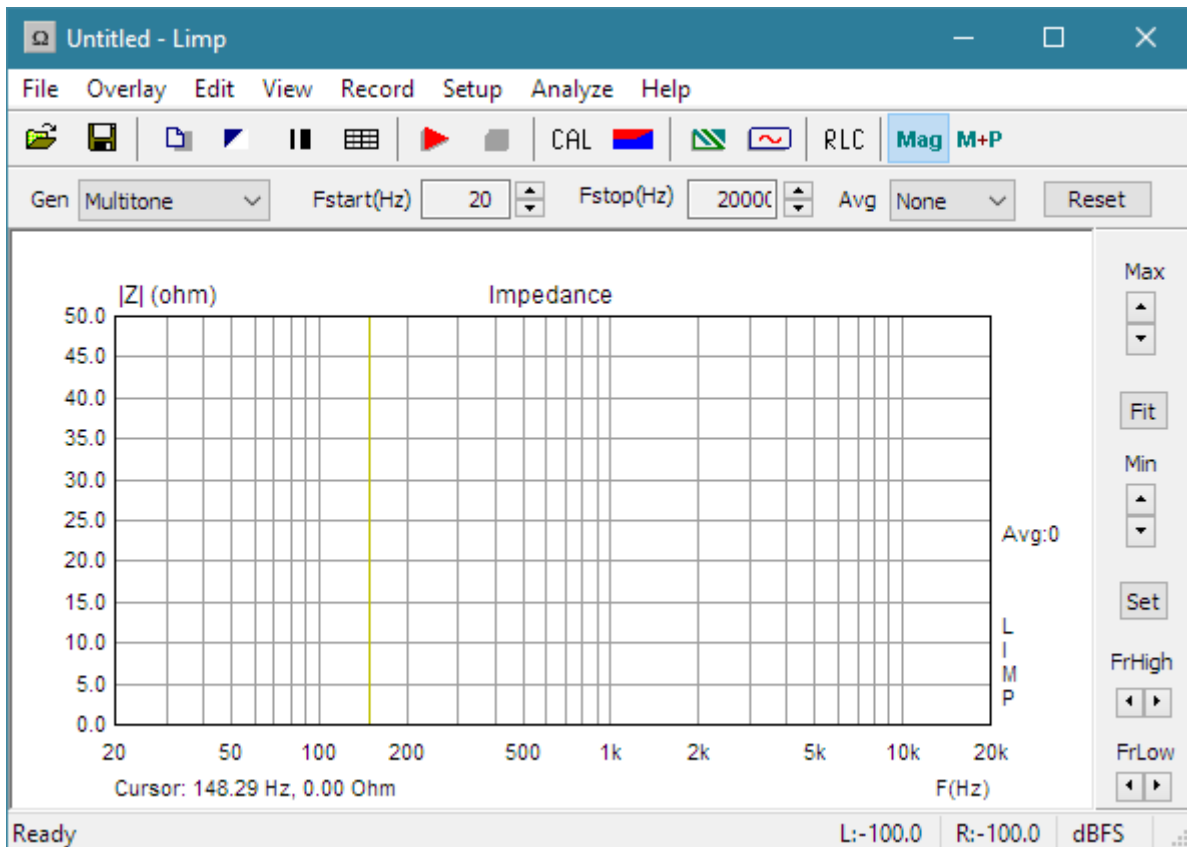

Figure 4.1 *Main program window*
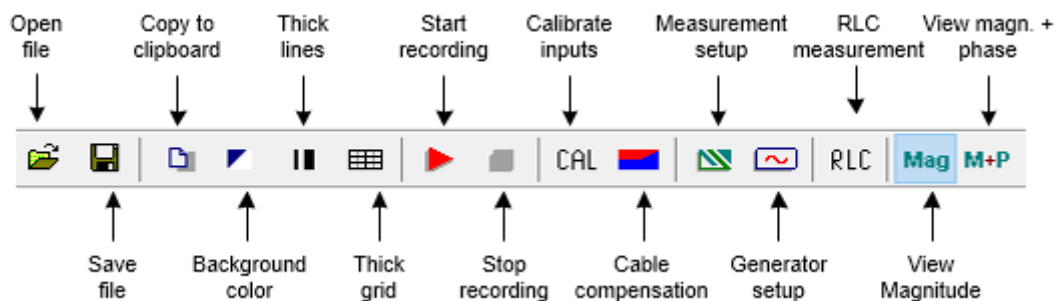
Figure 4.2 Toolbar icons
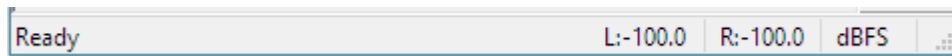
Figure 4.3 Status bar shows the peak level (ref. full scale) of left and right line inputs
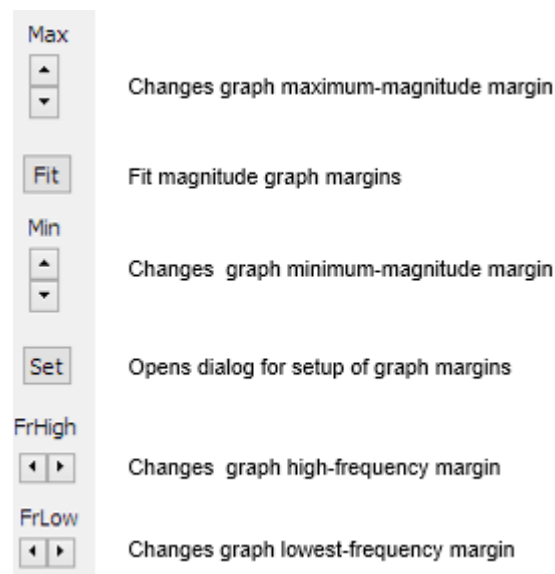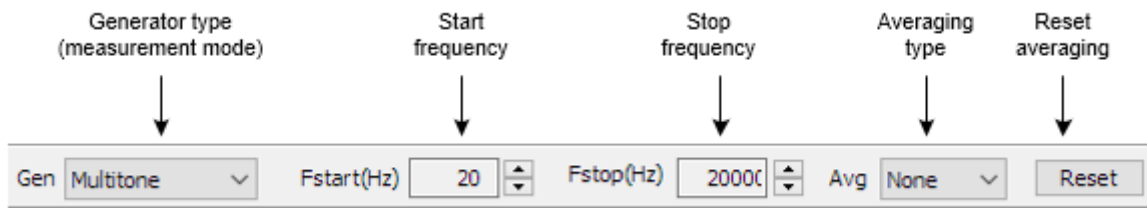
Figure 4.4 Top and right dialog bars

Images of graphs and windows can be copied to Windows clipboard or saved to the file in a three image formats: **png, bmp** and **jpg**. It is recommended to use **png** format.

Obtaining a copy 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 there the user can paste it in other open Windows applications (MS Word, MS Paint). Keys **Ctrl+Alt+P** activate command to save that image in the file.

To copy or save 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 the main window toolbar, the '**Copy**' button is shown as toolbar icon 🗐 .

The **Copy** command opens the dialog box '**Copy/Save Image with Extended Information**', shown in Figure 4.5. Here user has to setup following options:

1) By using the combo box above 'OK' button, user chooses one of three modes of saving the image: **Copy to Clipboard**, **Save to File** and **Save to File** + **Copy to Clipboard**.

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

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

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

5) Combo box '**Aspect ratio**' – sets aspect ratio for copying graphs with fixed width.

6) **Bitmap size** is chosen by selecting one of following combo box items:

   - Current screen size - variable width and height option
   - Smallest - fixed graph width 500 points
   - Small - fixed graph width 600 points
   - Medium - fixed graph width 800 points
   - Large - fixed graph width 1000 points
   - X Large - fixed graph width 1200 points
   - XX Large - fixed graph width 1500 points
   - XXX Large - fixed graph width 2000 points

The button '**OK**' copies the graph to the system clipboard or opens dialog to enter name of file in which picture will be saved.
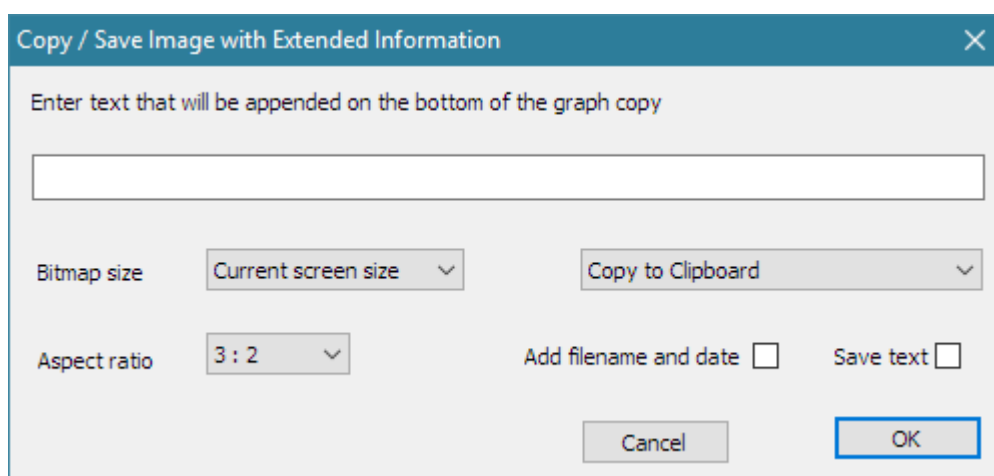
The button '**Cancel**' cancels the copy operation.

Figure 4.5 Dialog box '**Copy / Save 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
    Import from .txt or .zma file – imports impedance data from a textual file whose rows contain values of 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
    Set as target curve - sets current curve as target curve
    Load target curve - loads an target curve from an impedance .lim or .zma file
    Delete target curves - deletes all target curves
    Make target curve from TSP – makes and shows target impedance curve calculated from measured loudspeaker parameters

Edit

    Copy - copies the graph and user defined text to the clipboard or saves that image to the file
    Setup colors and grid style - sets graph colors and grid style
    B/W background color - changes the background color to black or white
    Thick pen - uses thick pen for plotting curves
    Thick grid - uses thick pen for plotting grids
    Subtract overlay - subtracts overlay impedance from current impedance
    Add overlay - add overlay impedance to current impedance

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 and Input Monitor - sets generator parameters and measures input voltages
Measurement - sets measurement parameters
Cable compensation - sets resistance and inductance for cable impedance compensation
Graph - sets graph margins
CSV format - opens dialog box for setup of decimal separator character in CSV files

Analyze

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

Help

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

Useful keyboard shortcut 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 the graph and user defined text to the clipboard or saves that image to the file
Ctrl+P key - copies a whole window bitmap to the clipboard
Ctrl+Alt+P key - saves a whole window bitmap to the file
Ctrl+B key - changes background color
F3 key - make target curve by calculating loudspeaker impedance using current TSP parameters

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.
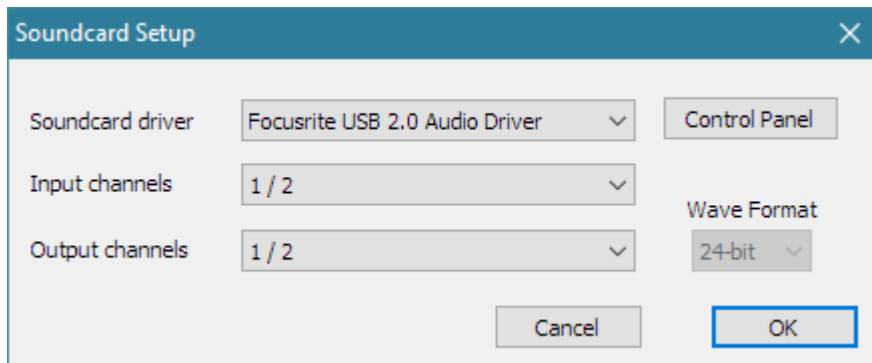

Figure 4.6 Soundcard setup

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

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 control panel. If ASIO driver is chosen, it opens ASIO control panel.

Wave format – 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). 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.

Important notice: **Please mute the line and microphone channels at the output mixer of the soundcard; otherwise you might have 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 WDM Audio Driver Setup

Microsoft has changed their approach in control of sound devices in Vista / Win7. Now, the 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 to set 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. 4.7 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. 4.8). 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. 4.9)

5) Repeat previous procedure 1) to 4) for recording channel, and choose the same sampling rate as in the playback channel.
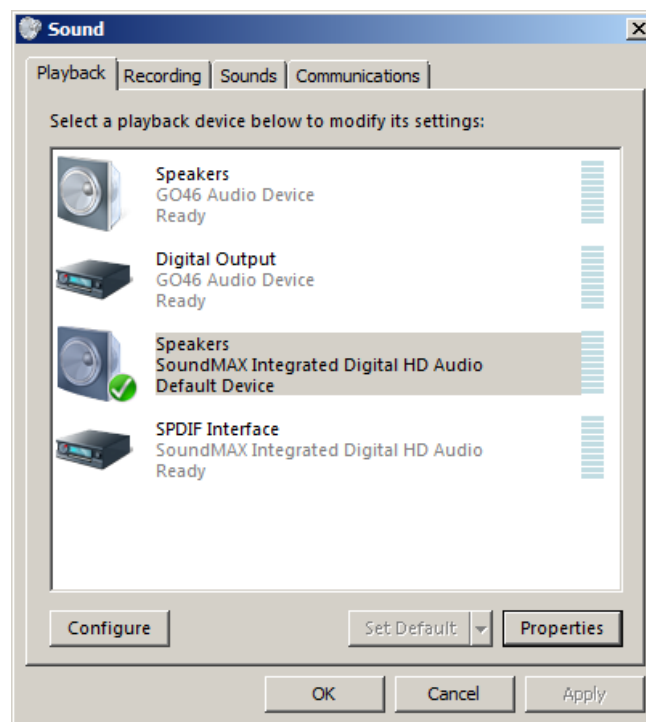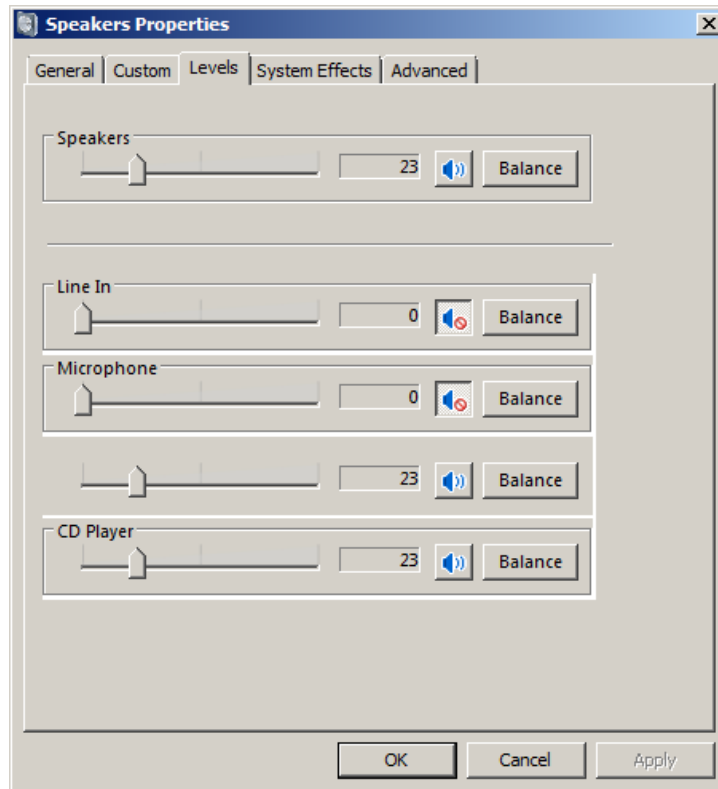

Figure 4.7 Vista Sound Control panel

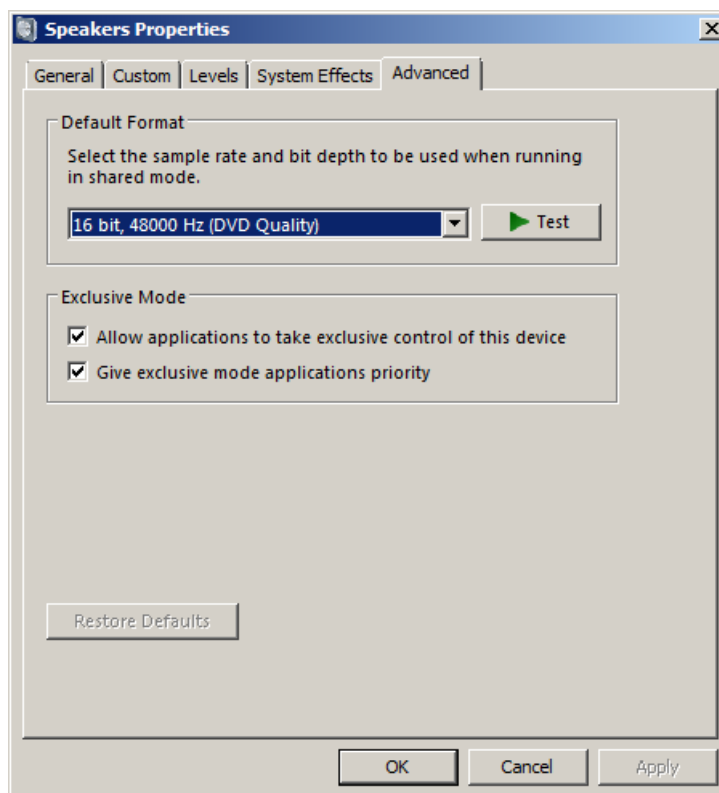**Figure 4.8** Playback channel properties – Output levels

Figure 4.9 Setting the native bit resolution and sampling rate in Vista

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

4.2.2 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 transferring 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.
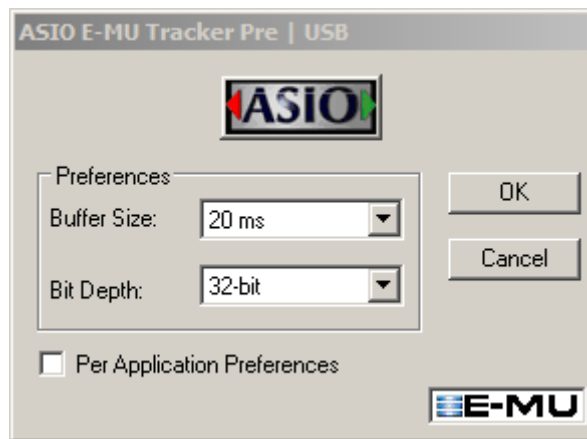

Figure 4.10 E-MU Tracker Pre ASIO Control panel for setting bit-resolution and buffer size

In music applications user usually sets buffer size as small as possible with stable operation. 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 the following expression:

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

ARTA automatically sets the buffer size for signal duration of 10 ms (i.e. 512 samples for sample rate 48kHz, 1024 samples for sample rate 96kHz and 2048 samples for sample rate 192kHz)..

LIMP always works 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 'Soundcard **Setup**'' which pair of channels will be used (1/2, 3/4, …).

4.3 Generator and Input Monitor Setup

Three types of excitation signals are possible in LIMP:

- **Sine**
- **Multitone**
- **Pink PN.**

User chooses signal type in the top dialog bar or in the dialog box **'Generator and Input Monitor Setup'** shown in Fig. 4.13. It opens by clicking the menu **Setup -> Generator and Input Monitor** or by clicking the toolbar icon [icon]. Besides signal type, the user can choose other signal characteristics (output level relative to maximum level and pink PN cut-off frequency).

The dialog box also contains a two-channel virtual voltmeter for monitoring voltages on the generator output (usually left input channel) and on measured impedance (usually right input channel). Input monitoring has no influence on the measurement of impedance; it just serves to set measurement signals to levels that will be treated as small signal or large signal excitation.

The virtual voltmeter should be calibrated. The calibration steps described below sets the input channel sensitivities. We define sensitivity as follows:

The digital measurement system records sequence of discrete measured values $D[i]$ in a restricted dynamic range. In ARTA software, discrete values $D[i]$ are floating point values in the range from -1 to 1. Virtual measurement system maps internal presentation $D[i]$ to external discrete voltage values $V[i]$, that has been recorded, using sensitivity factor:

$Sensitivity = V[i] / D[i]$,   where $|D[i]| <= 1$, $i = 0,1,2,\ldots$

The unit of *Sensitivity* is the Volt as $D[i]$ is dimensionless. The same hold for RMS values of periodic signals;

$Sensitivity = Vrms / Drms.$

Maximum possible value of discrete sequence $D[i]$ is 1. That gives us an alternative definition of sensitivity as the maximum peak value (or full scale) voltage that can be recorded by digital instrumentation;
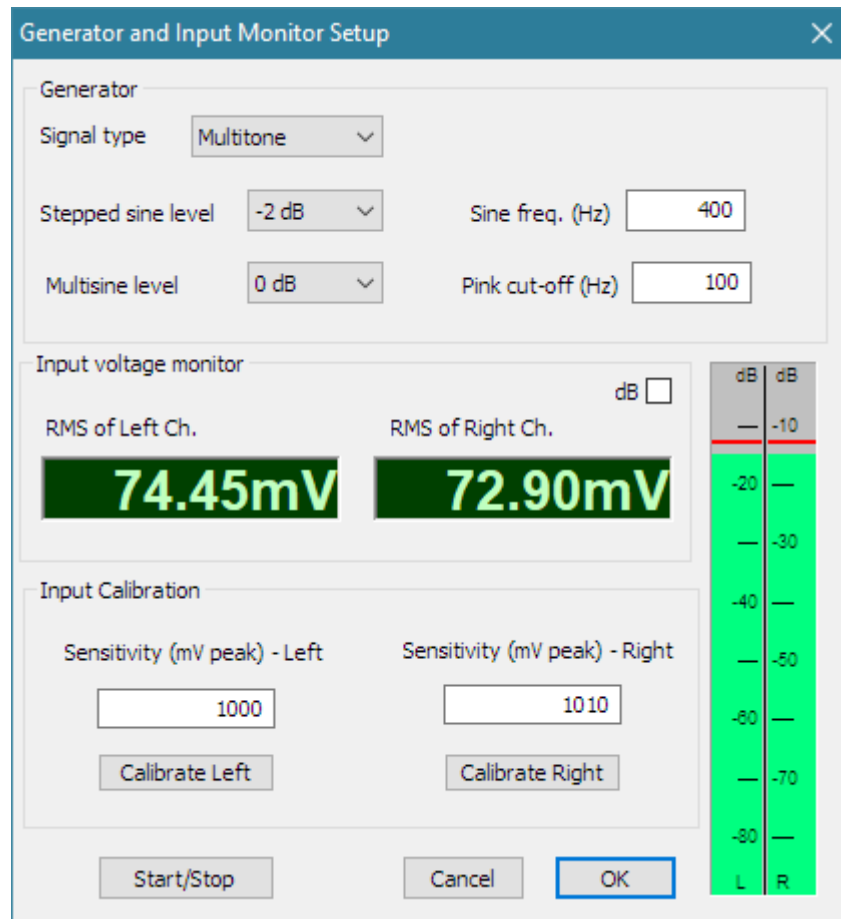
Sensitivity = Vpeak-max (Volts).

Figure 4.13 Generator and Input Monitor dialog box

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

Generator section
   Type - chooses the excitation signal: **Sine, Multitone** or **Pink PN**
   Sine freq. (Hz) - enters test frequency of the sine generator
   Pink cut-off (Hz) - enters the low-frequency cut-off of the pink noise generator
   Stepped sine level (dB) - chooses the output level (0 to -30 dB)
   Multisine level (dB) - chooses the output level (0 to -20 dB) for multitone or Pink PN

Input level monitor section
   Two voltmeter controls - show show either RMS voltages or levels in dBV for left and right input channels
   dB check box - if checked "LED" controls shows rms voltage in dBV units

Input calibration section
   Sensitivity (mV peak) - Left - sets sensitivity for Left input channel.
   Sensitivity (mV peak) - Right - sets sensitivity for Right input channel.
   Calibrate Left and **Right** buttons - starts calibration procedure for Left and Right channel.

Start/ Stop button - starts/stops the generator with current settings, also capture signals on both channels
Ok button closes dialog and saves dialog state.
Cancel button closes dialog and saves dialog state.
Peak meter - shows peak value ref. full scale. Meter bars are shown in following colors:
   Green - for levels below -3 dB,
   Yellow - for levels between -3 dB and 0 dB,
   Red - for overloaded input.

With LIMP the calibration is simple. It requires that user chooses sine signal and sine frequency in the range 300-500 Hz. The circuit for impedance measurement should be connected to the sound card, as shown in Fig. 3.1 or 3.2. The process starts by clicking button **Start/stop**. If recording starts successfully we can do calibration by clicking buttons **Calibrate Left** and **Calibrate Right.** That opens dialog box shown in Fig. 4.14.
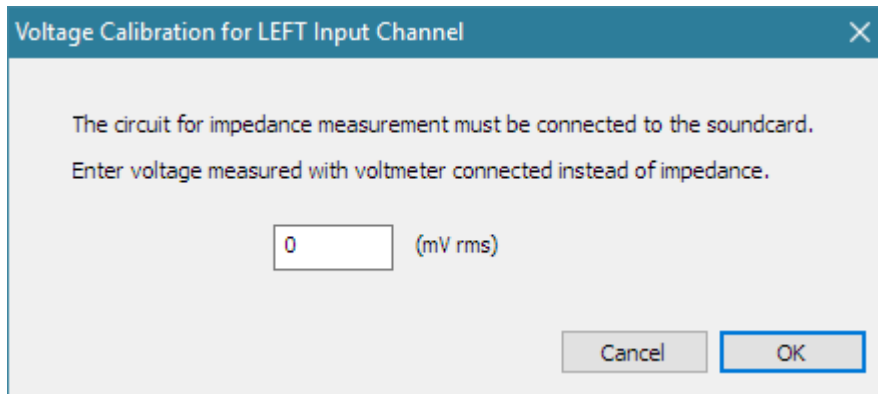

Figure 4.14 Dialog for calibration of impedance measurement system

Then, user should measure voltage using voltmeter connected instead of measured impedance and enter that value in the edit box. The calibration finishes by clicking button '**OK**'. If user click button '**Cancel**' the calibration is rejected.

For usual measurement setup shown in Fig. 3.1 or Fig. 3.2 this procedure will give voltage measurement accuracy better than 1%, which is more than enough for monitoring purpose.

Note: If system uses voltage probe, as in Fig. 3.2, calibration encounters sensitivity scaling automatically. Input impedance of voltage probe should be 100 times larger than reference resistor to retain accuracy below 1%.

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). Small signal regime requires sine voltages on impedance smaller than 100mV rms.

- For measurements of bass or mid-bass loudspeakers set the Pink cut-off frequency close to the loudspeaker resonance frequency (20-100 Hz).

- If measuring impedance for TSP estimation use low level signals. If you want that estimation of TSP using stepped sine mode is comparable to FFT mode lower the sine level 13 -15 dB.

- Standard AES2-2012 recommends measurement of impedance with 0.1 V rms of sine voltage. If we apply multitone or pink with 0.2 V rms we will get comparable results.

4.4 Measurement Setup

For setup of measurement parameters use the **'Measurement setup'** dialog box, shown in Fig. 4.15. You get it by clicking the menu **Setup->Measurement** or by clicking the toolbar icon [icon].
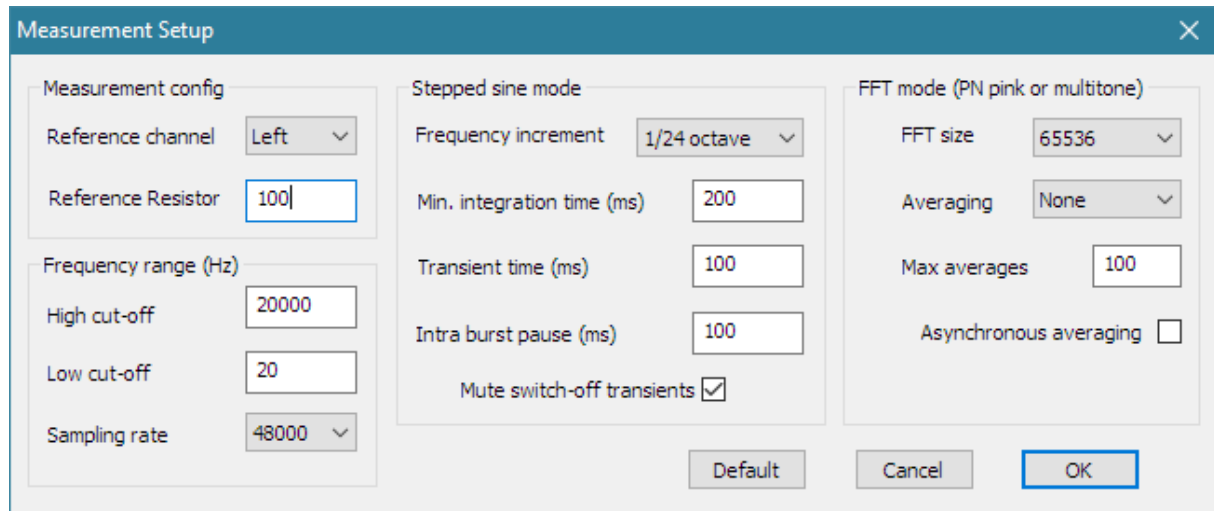

Figure 4.15 Measurement setup

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

In section Measurement config:
> **Reference channel** - sets the reference channel for measuring U_1 to Left or Right.
> **Reference resistor** - enters the value for the reference 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 96000 Hz).

In section Stepped sine mode:
> **Frequency increment** - sets to 1/24 or 1/48 octave
> **Min. integration time (ms)** - enters minimum integration time (higher values give 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.
> **Mute switch-off transients -** check box to remove annoying sound due to sine burst abrupt switch-off

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 ▶.   Measurements will be periodically repeated and results shown as an impedance plot. You will get the graph like one shown in Fig. 4.16. You can stop the measurements by clicking the menu **Record->Stop** or by clicking the toolbar icon ●.

You can copy the graph bitmap by clicking the menu **Edit->Copy** or by clicking the toolbar icon ▭.
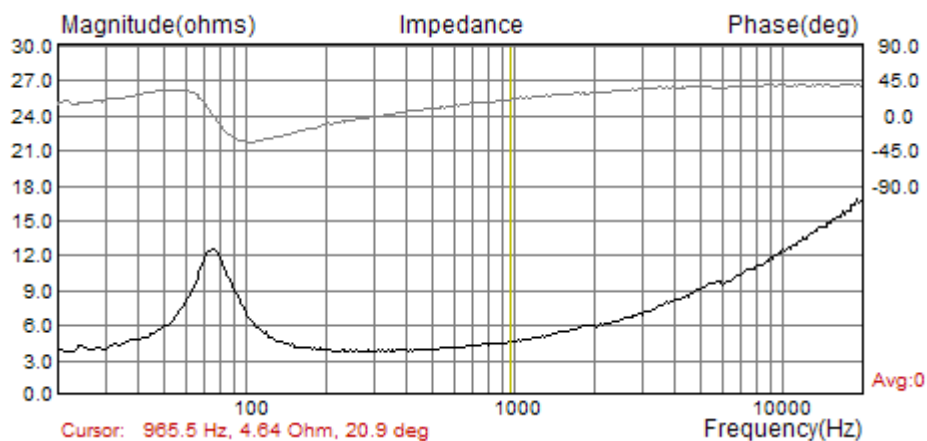
Figure 4.16 Impedance measurement without averaging – a noisy measurement

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 ●. You will get the graph like one shown in Fig. 4.17.

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.
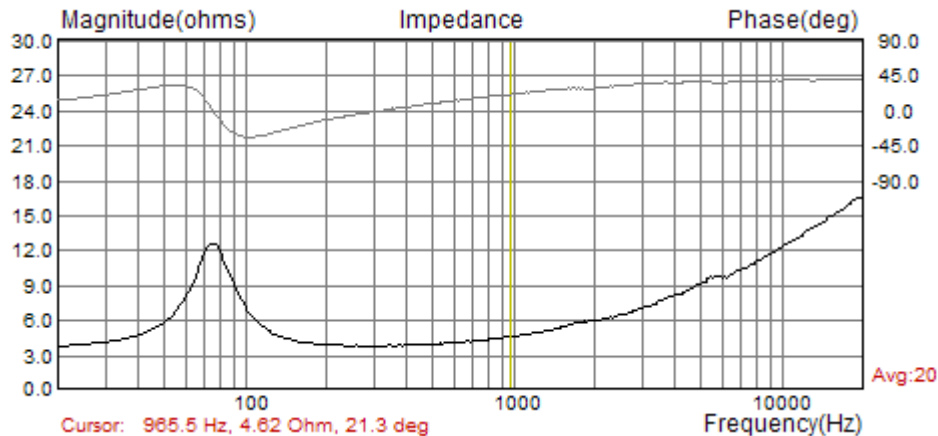
Figure 4.17 Impedance measurements with averaging

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, step by step. The cursor will show current progress in measurements, and that process will be very slow.

During a measurement, the cursor legend, shown below graph, shows frequency, impedance and THD of voltage signal measured on loudspeaker impedance. When the measurement finishes, THD will not be shown, but the frequency and maximal value of THD will be saved in a file. You can always see that information by clicking menu item '**File->Info**'.

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. 4.18. Use this dialog box to adjust the impedance magnitude range shown and the frequency range shown.
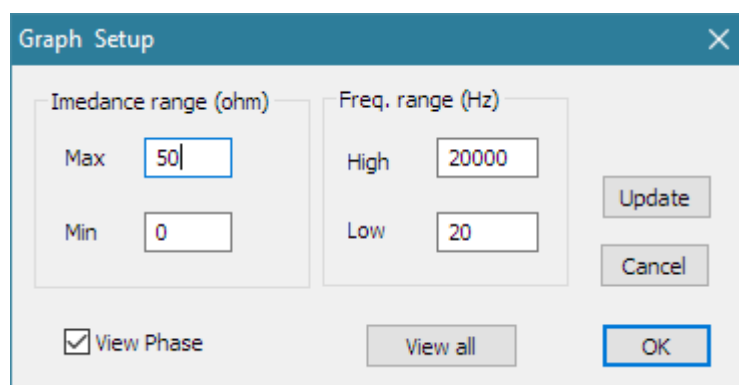
Figure 4.18 Graph setup

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 spectrum components that are used in impedance estimation.
View Phase – enables a phase plot.
Update – updates the graph with a new setup.
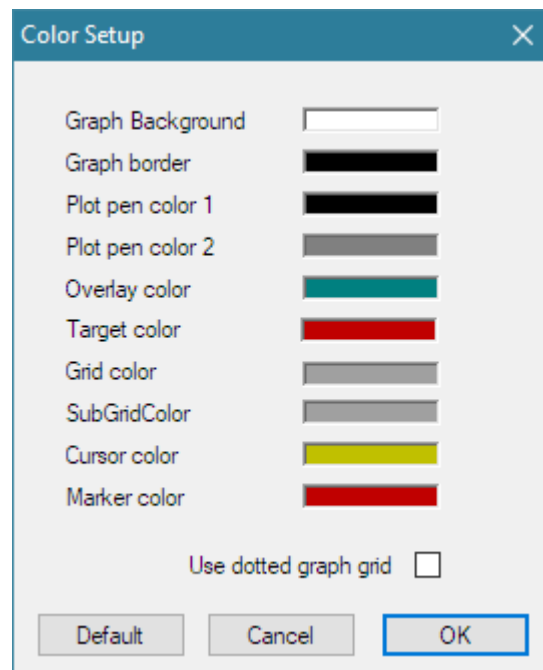
Figure 4.19 Dialog boxes for a graph colors setup
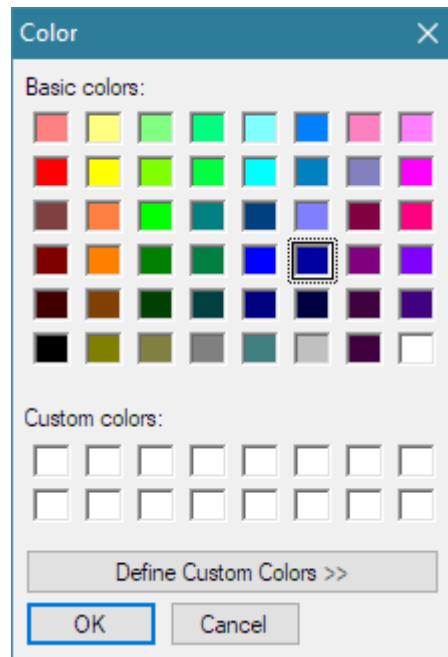
Figure 4.20 Standard Windows dialog for color setup

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.


4.7 Calibrated Measurements

In previously described measurements we have assumed that both soundcard input channels have the same sensitivity, but usually it is not true. If the difference in sensitivity of the input channels is larger than 0.2 dB, or if soundcard input impedance is small, we need a measurement system calibration.

4.7.1 Calibration of Input Channels

LIMP has a built-in calibration procedure. In FFT mode it has to be done for a particular generator sequence length and a sampling rate. Stepped sine mode accepts calibration for any sequence length.

By clicking the menu **Record->Calibrate**, or by clicking the toolbar icon CAL, we get the **'Calibrate Input Channels'** dialog box, shown in Fig. 4.21.

During the calibration procedure measured impedance should be disconnected.

Calibration will be done for full frequency range using Pink PN signal.
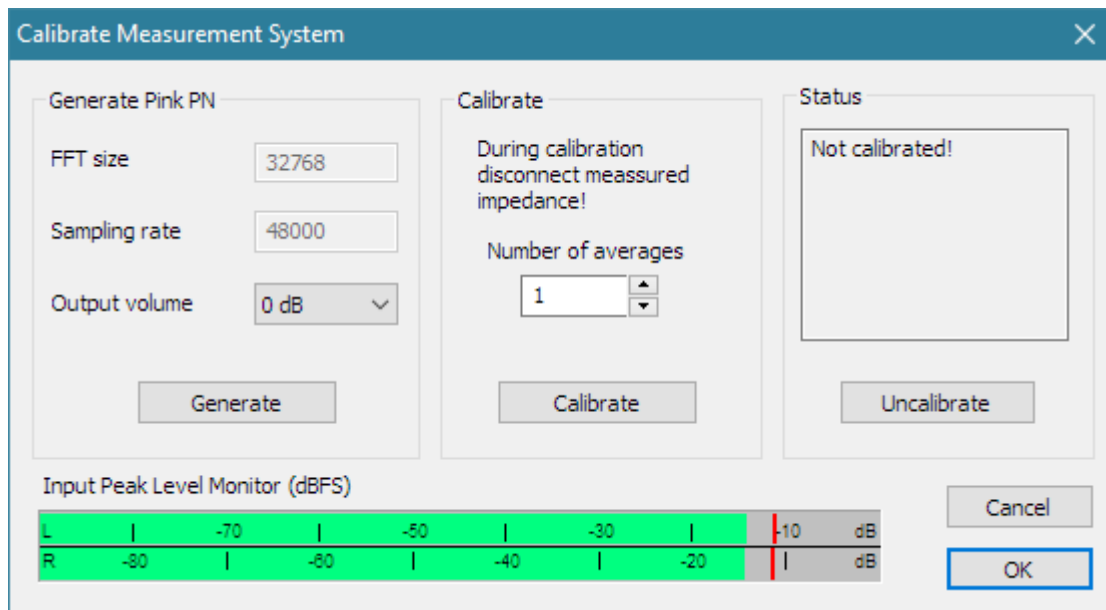
Figure 4.21 Dialog for measurement system calibration

In section **Generate Pink PN** we set the output volume for Pink PN signal which sampling rate and sequence length was set in Measurement setup dialog. Clicking 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 click the button **'Calibrate'**. After a second, we have the calibrated system.

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

The last section shows a report of the calibration procedure. If the Channel difference is larger than 3 dB LIMP gives a warning, and calibration is rejected. This precaution is required because some soundcards change input impedance if gain changes.

When we click the **OK** button, the dialog will be closed but the calibration process to be fully effective needs one more thing:

It is recommended that instead of the current value of reference resistor R in measurement setup we enter instead the value R || Rin, where Rin is the soundcard passband input impedance, which is a real value. This is necessary if we need accuracy better than 1%.

Note: calibration procedure is changed from LIMP version 1.9.2. In older versions only channel gains differences were compensated and measurements were restricted to impedances values up to 200 ohms.

To explain how calibration is implemented let's look at Fig. 4.22. which shows real measurement condition.

Inputs to "ideal" A/D converters
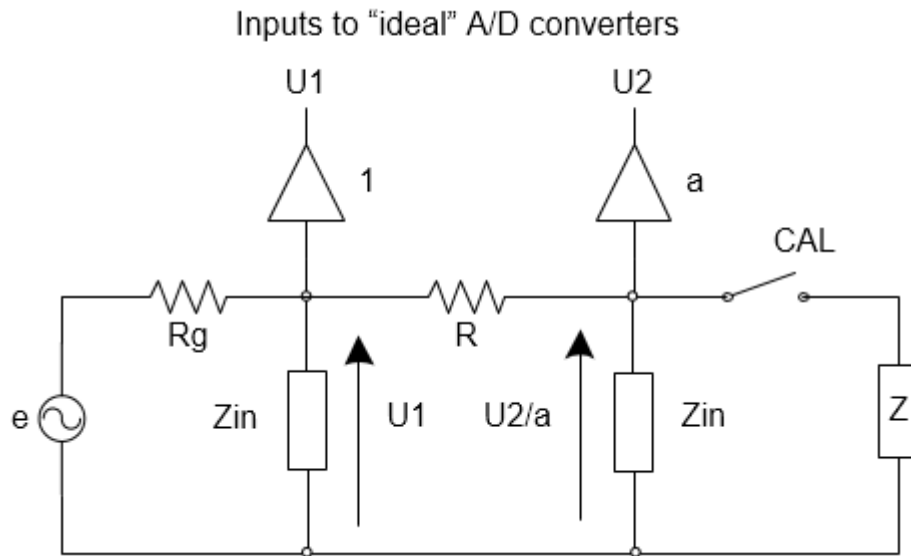
Figure 4.22 Electrical circuit for impedance measurement and system calibration

We assume that voltages are measured using soundcard input channels which have input impedances equal to Zin. Reference channel (usually left channel) measures voltage $U1$. The other channel is supposed to measure voltage $U2$ on measured impedance Z, but value of voltage on Z is $a\,U2$, where a is the ratio of possible different input channels sensitivities. The generator has output voltage e and output impedance Rg, but that values are not important for current discussion.

The circuit has switch CAL that should be closed during measurement of impedance Z and open during the calibration.

The current through reference resistor R is $I = (U1-U2/a)/R$. During impedance measurement following equations hold:

$$Z \| Zin = \frac{U2/a}{I} = R\frac{U2}{aU1-U2}$$

here $Z \| Zin$ denotes parallel circuit of Z and Zin. It is obvious that even if we have equal gains in both channels ($a = 1$) we measure $Z \| Zin$ not the impedance Z alone.

After some algebraic operation we get expression for impedance Z:

$$Z = \frac{RZin}{R+Zin}\frac{1}{\dfrac{U1}{U2}\dfrac{a\,Zin}{R+Zin}-1}$$

This expression is different from basic impedance measurement expression from chapter 1. It is biased with values of gain a and input impedance Zin.

Now we will define calibration as process which give us the value of factor: $a\,\mathrm{Zin}\,/\,(R+Zin)$.

In calibration procedure we measure voltages $U1$ and $U2$, but with measured impedance disconnected. From circuit on Fig. 4.21 we get:

$$\frac{U2cal}{U1cal} = \frac{a\,Zin}{R+Zin}$$

This value can be memorized and applied in expression for impedance, which now gets form:

$$Z = \frac{RZin}{R+Zin} \ \frac{1}{\dfrac{U1}{U2}\dfrac{U2cal}{U1cal} - 1}$$

Most soundcards input impedance is a real value in the whole audio range (*Zin* = *Rin*). If we know the value of *Rin* we can enter in LIMP measurement setup value *R* || *Rin* instead of reference resistor *R*, and we will get very accurate measurements.

Let's se what will be relative error without accounting for *Rin*.

$$\frac{\Delta R}{R} = (R - \frac{R\,Rin}{R+Rin})/R = 1 - \frac{Rin}{R+Rin}$$

Let *Rin* = 10k and *R* = 47 ohms, then relative error is 0.46%. If input impedance is 100k error is 0.05%. Practically, even without correction for *Rin* we will measure with error lower than 1%.

Note: The measurement impedance range is also limited by input channel noise when we measure small and large impedances. If we measure a small impedance, the voltage *U2* becomes small. If we measure a large impedance, voltage difference *U1-U2*, that determines current, becomes small. As a rule of thumb, we can define measurement range from *R*/100 to *R**100 for most soundcards and signal types. If we use high quality soundcards with low noise inputs and stepped sine measurements that range can be much larger.

4.7.2 Cable Compensation

Sometimes we use long cables to connect the loudspeaker to the reference resistance. If we know the cable resistance and inductance, then we can enter them in LIMP and substract from measured impedance. That procedure, called cable compensation, can be done by menu command **Setup->Cable Compensation**. It opens the dialog box shown in Fig. 4.23.
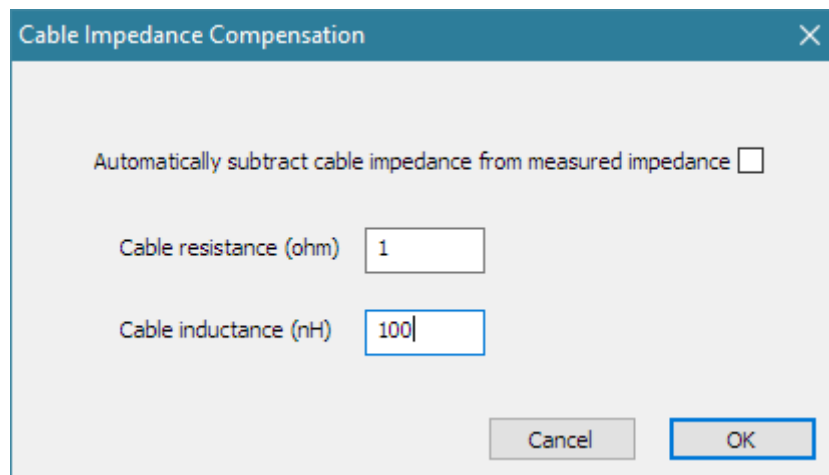

Figure 4.23 Dialog for cable impedance compensation setup

Two edit boxes are used for entering cable resistance (in ohms) and inductance (in nH). The check box "**Automatically subtract cable impedance from measured impedance**" must be checked if user wants that cable compensation be applied during the measurements.

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. 4.24). This dialog box can be opened by clicking menu command '**File->Info**'.
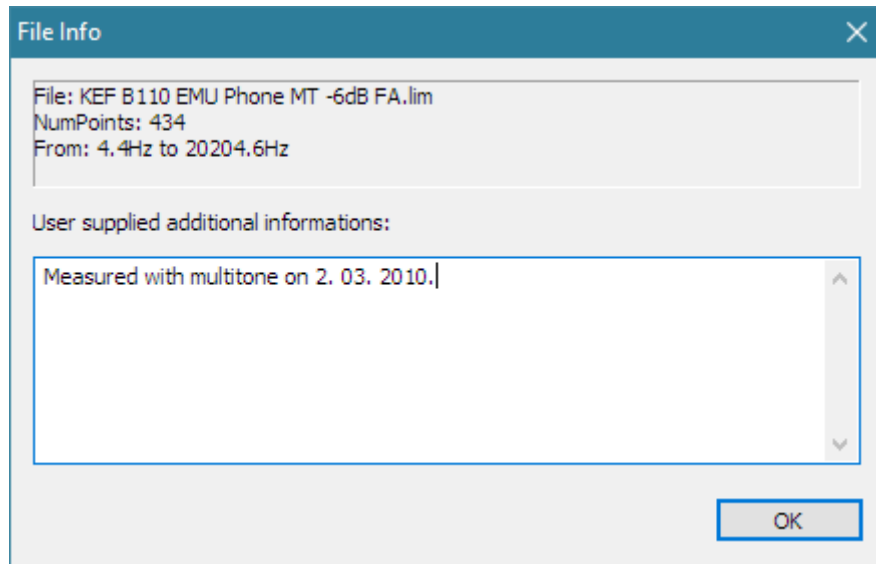
Figure 4.24 Dialog for file information

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 followed.

Binary file .LIM 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 unbaffled in free air or mounted in an infinite baffle is characterized by the following physical parameters:

Electromagnetic parameters:

R_E - voice coil DC resistance (Ω)

Z_{LE} - impedance of voice coil inductance and coupling to the pole piece (Ω)

Bl - force factor (Tm)

Equivalent circuits used to characterize the impedance Z_{LE} are shown in Fig. 5.1c, and denoted as: L2R [10], [11], L3R [7], and L2RK [12], [13].

Mechanical parameters:

S - effective area of membrane (m^2)

C_{MS} - membrane mechanical compliance (m/N)

M_{MS} - mechanical mass of membrane M_{MD} plus mass of air load on membrane M_{MA} (kg)

R_{MS} - mechanical resistance plus membrane radiation resistance (kg/s)

For the unbaffled driver the membrane air load mass is $M_{MA} = 0.5658\ S^{3/2}$, which is half of the air load mass in an infinite baffle.

The membrane effective area S, also called piston area, is normally obtained from a cone diameter measurement that includes 1/2 of the surround width [6], [11]. This value is good estimation for S if motion of surround is linear, but in practice it can vary. Program MLSSA [10] suggests diameter value that include only 1/3 of the surround width.

Fig. 5.1 shows wideband and low-frequency equivalent circuits of an electro-dynamic loudspeaker that is mounted in a free air unbaffled or in an infinite baffle. The circuit for definition of low-frequency input impedance uses the following elements:

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

a) Wideband analogous circuit

b) Low-frequency analogous circuit
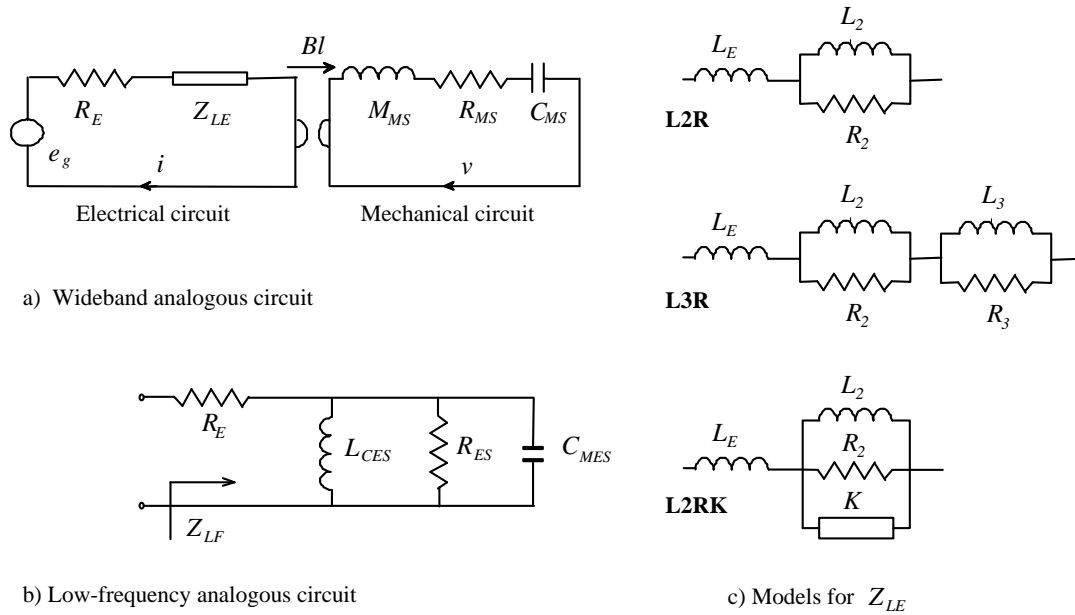
c) Models for Z_{LE}

Figure 5.1 a) Wideband analogous circuit of an electro-dynamic loudspeaker that is unbaffled or mounted in an infinite baffle, b) circuit for the estimation of the low- frequency input impedance, and c) three circuit models for lossy inductor Z_{LE} (L2R, L3R and L2RK model)

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	$Q_{TS} = \omega_S C_{MES} \dfrac{R_E R_{ES}}{R_E + R_{ES}} = \dfrac{Q_{MS}Q_{ES}}{Q_{MS} + Q_{ES}}$
Equivalent acoustical volume (m³)	$V_{AS} = \rho_0 c^2 S^2 C_{MS}$
Power available efficiency (%)	$\eta_0 = \dfrac{\rho_0}{2\pi c} \dfrac{S^2 (Bl)^2}{R_E M_{MS}^2} = \dfrac{4\pi^2}{c^3} \dfrac{f_S^3 V_{AS}}{Q_{ES}}$
Sensitivity (for 1W/1m or 2.83V/1m) (dB re 20 uPa)	$L_p(1W/1m) = 10\log\left(\dfrac{\rho_0 c}{2\pi}\eta_0\right) - 20\log(p_{ref}) = 112{,}1 + 10\log(\eta_0)$ $L_p(2.83V/1m) = L_p(1W/1m) + 20\log(2.83/\sqrt{R_E})$

Table 5.1 Thiele-Small dynamical loudspeaker parameters ($\rho_0 = 1.18$kg/m³, $c = 345$m/s, $p_{ref} = 20\mu$Pa)

Thiele and Small have shown that by using these parameters it is easy to express the low-frequency response of a closed loudspeaker box as 2nd order high-pass filter and response of bass-reflex box as 4th order high-pass filter. Today, almost every loudspeaker manufacturer provides physical and Thiele-Small parameters in their loudspeaker data sheets.

An old industry standard assumed that the voice coil electrical impedance contains resistive (R_E) and pure inductive ($Z_{LE} = j\omega L_E$) components. Voice coil inductance (L_E) was usually specified as inductive component of driver impedance at 1 kHz. In modern instrumentation that model is abandoned, and the voice coil is treated as "lossy" inductor whose impedance Z_{LE} depends on frequency. Physically, Z_{LE} represents the input impedance 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 $K\sqrt{j\omega} = K\,(1+j)\,\sqrt{\omega}/2$, where K is constant. It is expressed in unit called semiHenry (sH). The impedance of such semi-inductance increases with $\sqrt{\omega}$ rather than ω. This model is not practical as engineers in many numerical simulations use some form of an equivalent electrical circuit that closely matches measurement data.

The most commonly used equivalent circuit for the electrical voice coil impedance in modern instrumentation ([10], [11]) is serial connection of resistor R_E, inductor L_E and parallel connection of resistor R_2 and inductor L_2. It is denoted in Fig. 5.1 as L2R model. For many years it has been proven as useful model in many simulations.

LIMP also enables estimation of two additional and more accurate equivalent circuits for lossy inductor: L3R and L2RK models. They are also shown in Fig. 5.1. The L3R model extends L2R model with one more parallel circuit $L_3||R_3$, while in L2RK model, as defined by Thorborg [12], a semi inductance K is added parallel to circuit $L_2||R_2$.

Originally, Small [6] defined parameters for loudspeaker mounted in an infinite baffle, but he allowed the same expression to be used for unbaffled loudspeaker. The same is allowed by the IEC 60268-5 standard [15], with requirement to report which type of mounting is used in measurements. Instead od using an infinite baffle, the IEC standard requires the use of a baffle of dimension 135x165 cm with driver mounting hole offsets from center 15x22.5cm. Standard AES2-2012 [16] exclusively advocates measurements with unbaffled driver. LIMP procedures support both type of driver mounting.

Thiele-Small parameters can be used to measure and analyze the response of a loudspeaker mounted in a closed box. The analogous circuit have the same elements as circuit on fig. 5.1., but values of elements change: C_{MS} is replaced with parallel connection of C_{MS} and equivalent box volume compliance, M_{MS} changes in air load mass term, and R_M changes due to added dissipation inside the box. The air load mass of loudspeaker mounted in closed box is between values for unbaffled driver and driver mounted in infinity baffle. This is probably the main reason why both types of mounting are accepted in practice - both are equally accurate and valuable to predict box response sensitivity and efficiency.

5.2 Theory for the Estimation of Loudspeaker Parameters

For the estimation of Thiele-Small parameters (TSP), LIMP offers two methods:

1) Thiele-Small method and

2) Nonlinear Least Square Error (LSE) minimization method.

LIMP also uses LSE minimization for estimation of lossy inductor equivalent circuit elements. It is a first step in TSP estimation because it helps to remove influence of inductance element from analog circuit before using it in TSP estimation.

5.2.1 Estimation of Lossy Inductor Equivalent Circuit Elements

LIMP estimates the values of lossy inductor equivalent circuit elements as values that give the least sum of squared differences between lossy inductor impedance $Z_{LE}(f)$ and a measured input impedance $Z_M(f)$ over all frequencies that are above the frequency of minimum of impedance. LSE error function is defined as:

$$\varepsilon = \sum_f \left[|Z_M(f)| - |Z_{LF}(f)| - |Z_{LE}(f)| \right]^2$$

where $Z_{LF}(f)$ is a low frequency loudspeaker impedance estimated by Thiele-Small method (which is described in the next Section).

By default, LIMP estimates elements of L2R model by stable, *linearized LSE* error minimization method – sometimes also called a linear regression method. In addition, user can choose more accurate *nonlinear LSE* minimization methods for L2R, L3R and L2RK model circuit elements. For full discussion of these and TSP estimation methods it is recommended to read the paper [13].

5.2.2 Thiele-Small Method for the Estimation of TSP

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

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

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:

$$Z(f) \cong Z_{LF} = R_E \frac{(1 - f^2/f_S^2) + j\, f\, /(f_S Q_T)}{(1 - f^2/f_S^2) + j\, f\, /(f_S Q_{MS})}$$

The impedance has maximum value at the resonance frequency:

$$Z_{\max} = Z(f = f_S) = R_E Q_{ES}/Q_T = R_E + R_{ES} \qquad \text{as } Q_{ES} > Q_T.$$
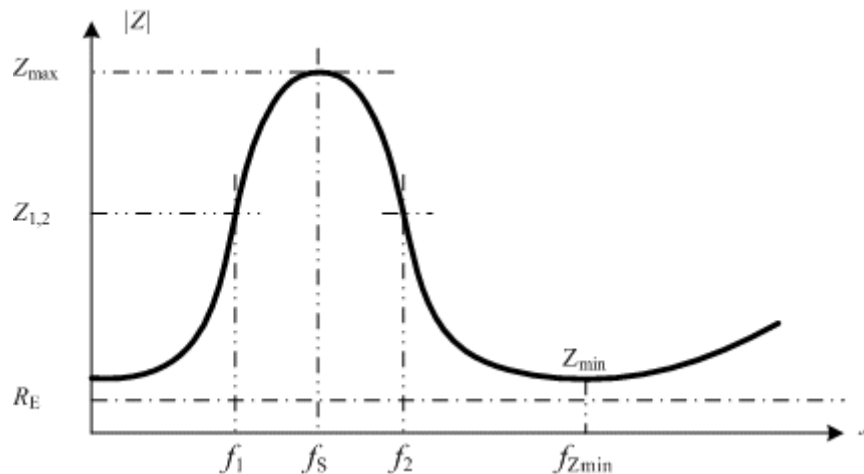

Figure 5.2 Typical impedance curve of a loudspeaker that is mounted in free air

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

$$|Z(f_1)| = |Z(f_2)| = r_1 R_E, \quad \text{if} \quad f_1 f_2 = f_S^2$$

If we substitute this expression in the expression for the input impedance, we get:

$$\left|Z(f_{1,2})\right|^2 = r_1^2 R_E^2 = R_E^2 \frac{r_0^2 + (Q_{MS}(f_1 - f_2)/f_S)^2}{1 + (Q_{MS}(f_1 - f_2)/f_S)^2} \ .$$

From this equation we get the mechanical Q-factor:

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

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 $Z = Z_1$. Then 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 $Q_T = Q_{ES}Q_{MS} / (Q_{ES} + Q_{MS})$.

This procedure gives good results under two conditions:

1. if we have low noise level in measurement of impedance, and

2. if we first subtract magnitude of lossy inductor impedance $|Z_{LE}|$ from the magnitude curve. In first iteration $|Z_{LE}|$ is unknown. In that case TSP estimation procedure uses values of Z_1 and f_1 from measured magnitude curve that are below the resonance frequency f_S and sets $f_2 = f_s^2 / f_1$.

If we have noisy measurements, we can reduce the estimation error by repeating Thiele-Small procedure for different frequencies and average value of Q_{MS}. Good choices are frequencies where magnitude is 20% lower than maximum magnitude and 20% larger than minimum magnitude.

These two simple modifications of standard Thiele-Small procedure give accurate and reliable estimation of TSP.

5.2.3 Nonlinear LSE Minimization Method for the Estimation of TSP

The Thiele Small procedure is realizable only if the value of R_E is known, but sometimes that value is not known. In that case we can use the nonlinear LSE minimization procedure where we minimize squared difference between measured impedance Z_M and model impedance Z_{LF}, i.e. we define LSE function as:

$$\varepsilon = \sum_f \left[|Z_M(f)| - |Z_{LE}(f)| - |Z_{LF}(f)| \right]^2$$

where optimization variables are R_E, f_S, Q_{MS} and Q_T. $Z_{LE}(f)$ is impedance of lossy inductor that is also estimated by LSE minimization.

If the value of R_E is known, optimization variables are: f_S, Q_{MS} and Q_T.

In some rare cases a convergent LSE minimization cannot be realized. In that case the modified standard Thiele-Small procedure would be used.

5.3 Estimation of Physical Loudspeaker Parameters

Three methods will be used for the estimation of physical parameters M_{MS}, C_{MS}, R_{MS} and Bl;

1. **Added mass method,**

2. **Closed box method,**

3. **Fixed mass method.**

All these methods can be used for measurement of a loudspeaker mounted in an infinite baffle or in free air. The results for two mountings will differ slightly as air load mass is not the same. The air load in a large baffle is two times larger than in free air. Although results will differ slightly we argue that they can be successfully used in box design.

There are two standards that define measurement of physical loudspeaker parameters. IEC 60268-5 [15] defines closed box method. The other standard AES2-2012 [16] extends IEC standard with added mass method. AES2-2012 also specifies that mounting should be done with firm clamp in vertical position so that gravitational force does not offset membrane.

For estimation of physical parameters LIMP uses and improves methods promoted by Small [6] and measurement system MLSSA. They are described in next sections.

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 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 Bl^2. By equating both expression for Bl^2, as we assume it to be constant in both measurements, we get:

$$M_{MS} = \frac{M_{added}}{\dfrac{f_S Q_{EM}}{f_M Q_{ES}} - 1} \qquad \text{"constant force factor Bl"}$$

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

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

This method is used in LIMP by default. It is called "constant Bl" method. Standard AES2-2012 uses "constant compliance" method, where expression for C_{MS} with and without added mass are equated to get:

$$M_{MS} = \frac{M_{added}}{\dfrac{f_S^2}{f_M^2} - 1} \qquad \text{"constant compliance"}$$

In most cases assumption of constant compliance gives worse results than "constant Bl" solution. Why? In low frequency region membrane has large displacement which generates nonlinear behavior of compliance and force factor Bl, but compliance also has rise toward low frequencies due to viscoelastic creep effect. In both cases it is important to measure impedance with small signal excitation.

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 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}, and Q_{EC}. From these we find [6]:

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

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

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

The problem of measurement piston area S

The accuracy of estimation of physical parameters depends on the value of membrane area S. The value is normally obtained from a cone diameter measurement that includes 1/3 - 1/2 of the surround width. MLSSA [10] suggests a method to estimate area S from measurement data. The method is based on applying two measurement methods; added mass and closed box, successively to the same driver. This technique makes use of the fact that V_{AS} is independent of S for the closed-box method while M_{MS} is independent of S for the added-mass method. Then the value of area S in square centimeters can be computed as,

$$S(cm^2) = 10 \sqrt{\frac{M_{MS} V_{AS} \omega_S^2}{\rho_o c^2}} = 0.1677 f_S \sqrt{M_{MS} V_{AS}}$$

where M_{MS} in grams is measured using the added-mass method while V_{AS} in liters is measured using the closed-box method. Frequency f_S is driver resonant frequency that is measured without box.

5.3.3 Fixed Mass Method

The fixed mass method assumes that we have unassembled driver membrane of area S. The static weight of membrane M_M (including voice coil and half of suspension and spider) is measured statically.

Total dynamical mass M_{MS} is obtained by adding air loading mass. If driver is unbaffled the total mass is calculated by the expression:

$$M_{MS} = M_M + 0.5658 \, S^{3/2}$$

If loudspeaker is mounted in IEC or larger baffle the air load mass should be doubled.

In this method we first measure the impedance curve and estimate Thiel-Small parameters f_S, Q_{MS} and Q_{ES}, for the loudspeaker mounted in free air or in IB, then by using known value of M_{MS}, we get:

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

This method is most accurate, and simplest as we need only one impedance measurement.

5.4 Automatic Estimation of Physical and Dynamical Loudspeaker Parameters

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

To estimate Q-factors of a loudspeaker we need:

1. measured impedance data (without any overlay curve),
2. measured voice coil dc resistance (in ohms),
3. estimated membrane diameter in cm.

By clicking the menu '**Analyze->Loudspeaker parameters – Added mass method**', we get the '**Loudspeaker Parameters - Added Mass Method** ' dialog box, shown in Fig. 5.3.

Figure 5.3 Dialog box for the estimation of loudspeaker Q-factors

In this dialog box, user enters values for **Voice coil resistance** and **Membrane diameter** in appropriate edit boxes. Click 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'**. The click on button **Export** saves data in file using the Excel .csv format.

In the section "**Nonlinear LSE Optimization**" user sets three check boxes to allow more accurate nonlinear LSE optimizations:

> **Estimate TSP by LSE minimization**
> **Estimate voice coil resistance** (if voice coil resistance is unavailable)
> **Estimate lossy inductor model**

To apply LSE method, impedance measurement should be done in full frequency range (20 kHz), or at least to frequency that is a hundred times larger than loudspeaker resonance frequency.

5.4.1 Added Mass Method

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

1. measured impedance data,

2. impedance data measured with an additional mass set on the membrane (as in Fig. 5.4),

3. one of the curves (1 or 2) must be set as an overlay graph,

4. measured additional mass - in grams,

5. measured voice coil dc resistance - in ohms,
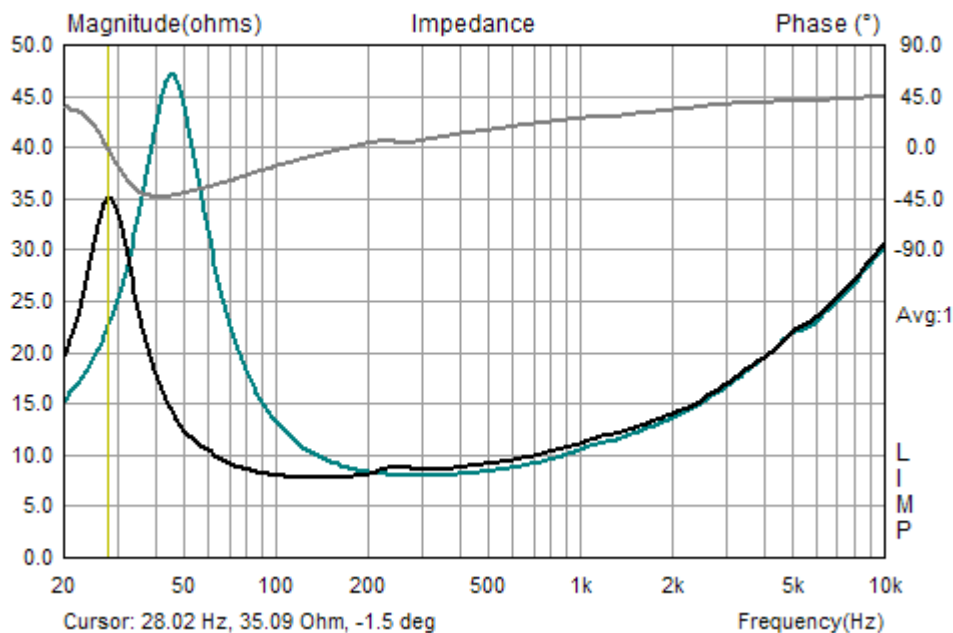
6. diameter of the membrane - in cm,

Figure 5.4 Impedance curves (the one with lower resonance frequency is obtained by adding mass to the membrane)

Then we click the menu item **Analyze->Loudspeaker parameters – Added mass method** to get the '**Loudspeaker Parameters – Added Mass Method**' dialog box, shown in Fig. 5.5.
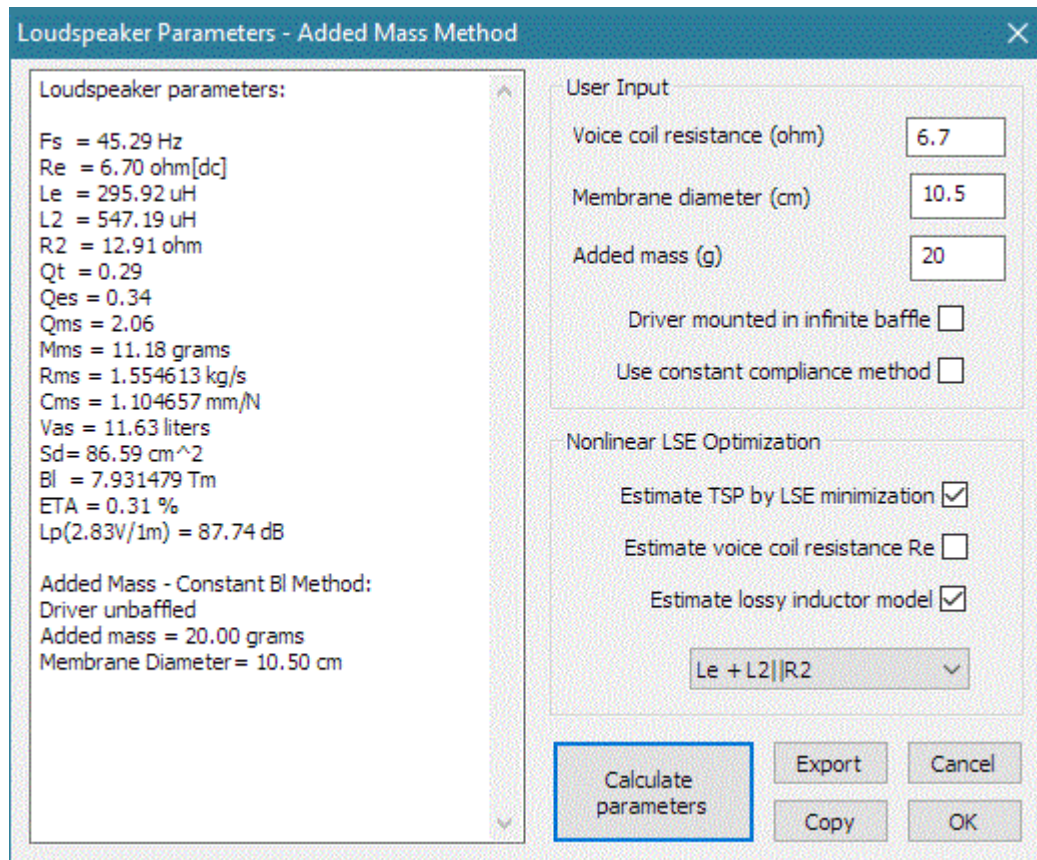

Figure 5.5 Dialog box for the estimation of loudspeaker parameters using Added Mass Method

User enters values for **Voice coil resistance, Membrane diameter** and **Added mass** in appropriate edit boxes. Click on the button **'Calculate parameters'** gets the report shown in the left edit box. This report can be pasted to the clipboard by pressing the button **'Copy'**. The click on button **Export** saves data in file using the Excel .csv format.

Check box '**Driver mounted in infinite baffle**' sets condition of measurement. Here it is empty which means that driver is unbaffled.
Check box '**Use constant compliance metho**d' enables alternate equation to estimate physical parameters. The default is "constant Bl" method.

In the section "**Nonlinear LSE Optimization**" user sets three check boxes to allow more accurate nonlinear LSE minimization.

> **Estimate TSP by LSE minimization**
> **Estimate voice coil resistance** (if voice coil resistance is unavailable)
> **Estimate lossy inductor model**

Three lossy inductor circuits:

> Le+L2||R2
> Le+L2||R2+ L3||R2
> Le+L2||R2||K

are selectable by using the list box at the bottom of this section (+ denotes serial connection, || denotes parallel connection).

5.4.2 Closed Box Method

To estimate all loudspeaker parameters, we must have two impedance measurements. In one of the 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 the curves (1 or 2) must be set as an overlay graph (as in Fig. 5.7),

4. measured value of the box volume - in liters,

5. measured voice coil dc resistance - in ohms,

6. diameter of the membrane - in cm,

Then we click the menu item **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** in appropriate edit boxes. 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'**. The click on button **Export** saves data in file using the Excel .csv format.

In the section "**Nonlinear LSE Optimization**" the user sets three check boxes to allow more accurate nonlinear LSE minimization:

> **Estimate TSP by LSE minimization**
> **Estimate voice coil resistance** (if voice coil resistance is unavailable)
> **Estimate lossy inductor model**

Three lossy inductor circuits:

> Le+L2||R2
> Le+L2||R2+ L3||R2
> Le+L2||R2||K

are selectable by using the list box at the bottom of this section (+ denotes serial connection, || denotes parallel connection).

5.4.3 Fixed Mass Method

In this method we use single impedance measurement. The loudspeaker is unbaffled or in infinite baffle

This is what we need:

1. measured impedance data,
2. measured static mass of membrane with voice coil and half of suspension and spider.
3. measured voice coil dc resistance in ohms,
4. estimated membrane diameter - in cm

Then we click the menu **Analyze->Loudspeaker parameters - Fixed mass method** to get the '**Loudspeaker parameters - Fixed Mass Method**' dialog box, shown in Fig. 5.7.

Figure 5.7 Dialog for the estimation of loudspeaker parameters using Fixed Mass Method

User enters values for **Voice coil resistance, Membrane diameter** and **Static membrane mass** in appropriate edit boxes. Click on the button **'Calculate parameters'** gets the report shown in the left edit box. This report can be pasted to the clipboard by pressing the button **'Copy'**. The click on button **Export** saves data in file using the Excel .csv format.

In this dialog the check box '**Driver mounted in infinite baffle**' is not for informative purpose only, like in previous cases. Here it determines the proper value of air load mass shall be used in calculations of M_{MS}.

5.4.3 Monitoring the Quality of Loudspeaker Parameters Estimation

LIMP doesn't show any "single number" indicator of TSP estimation quality, but the user can visually check the quality of estimation – by pressing the F3 key in measurement window, or by menu command **Overlay->Make target curve from TSP**. It will generate impedance curve from currently calculated loudspeaker parameters and show it as target curve.

The quality of Loudspeaker parameter estimation is not high. The estimation of V_{as} and M_{MS} can show differences, depending on method used in the range 20-30%. Here is partial explanation of these differences, and where they came from.

The Thiele-Small model is a simplified linear model, while the loudspeaker is a highly nonlinear device. The nonlinearity is largest below resonance frequency where the loudspeaker membrane has largest displacement. It causes nonlinearity in compliance and force factor Bl. The compliance is also changing as a function of frequency – that is called a creep effect, and it makes larger compliance variation in the added mass method than in the closed box method. The closed box measurement method is very often disturbed by large air leakage. Compliance is also a function of temperature. Loudspeaker that are used in cars are exposed to large temperature variations, and in extreme cases compliance can change more than 50%.

The Fixed mass method can give best results, as it is less dependent on compliance shift, but in many cases, it is impossible to have unassembled driver membrane.

For proper measurement it is extremely important that a loudspeaker passes a long burn-in period, as it helps stabilize suspension dynamics. Burn-in could be done with music signals with high low-frequency content or with ARTA test signals (pink noise or multitone).

6 RLC Measurement

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 capacitor with nominal value of 6.9 uF.
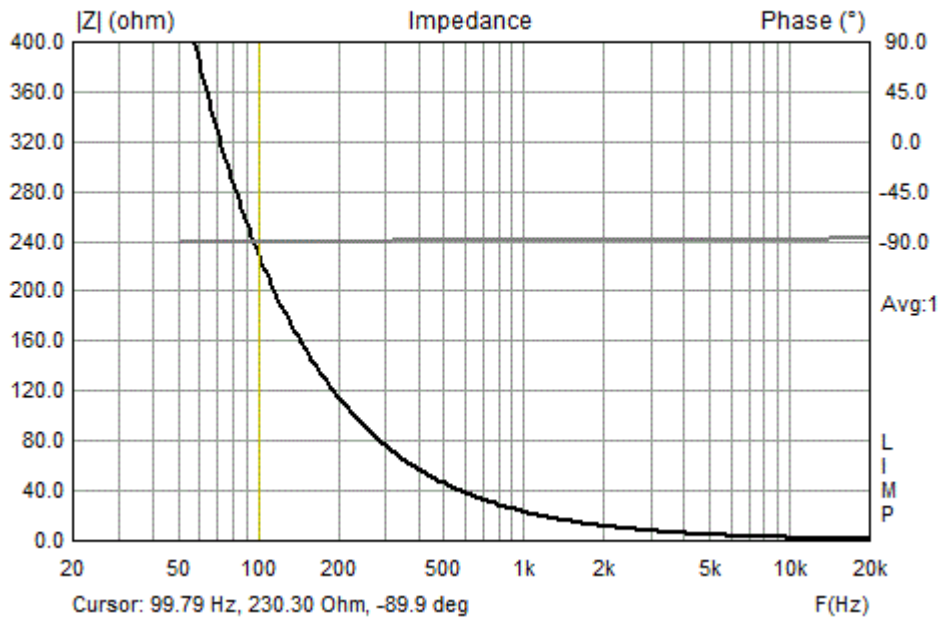

Figure 6.1 The impedance graph of a 6.9 uF capacitor

By clicking menu command **Analysis->RLC Impedance value at cursor position** we get the dialog box with report as shown on Figure 6.2. Here LIMP reports that measured impedance has resistive part of 0.587 ohms and imaginary part is capacitive with value of 6.93 uF.
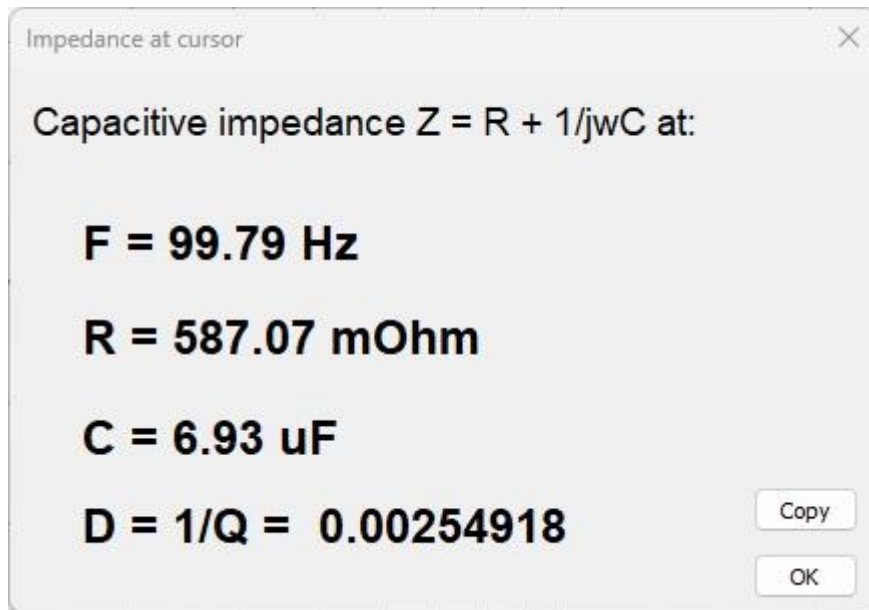
Figure 6.2 Impedance of 6.9 uF capacitor

LIMP measures the inductance in the same way. The report of measured impedance is shown in Fig. 6.2. The last line shows the value of dissipation factor D. It is equal to the ratio of impedance real and imaginary parts at specified frequency. The reciprocal value of D is called quality factor Q. It is normally used to show the quality of inductors.

Note: These measurements can be confusing if impedance phase is close to zero degree or very close to -90 or +90 degrees. A very small error in measurement can results with false capacitive or inductive values.

6.1 Importance of Calibration

When measuring impedance and capacitance it is very important to calibrate the system before the measurement. Why?

Even if there is a very small difference between channel sensitivities (i.e. 0.1 dB), LIMP can give erroneous result, because inductor impedance has phase close to plus 90 degrees, and capacitor impedance has phase close to minus 90 degrees. 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 degrees and the graph shows warped jump in the phase for 180 degrees. Figure 6.3 shows the case of measuring a capacitor without the calibration. In half the range the phase is close to plus 90 degrees, and suggests that we are dealing with an inductance.

Figure 6.4 shows capacitor impedance after the calibration. We see correct values for phase in the whole frequency range.
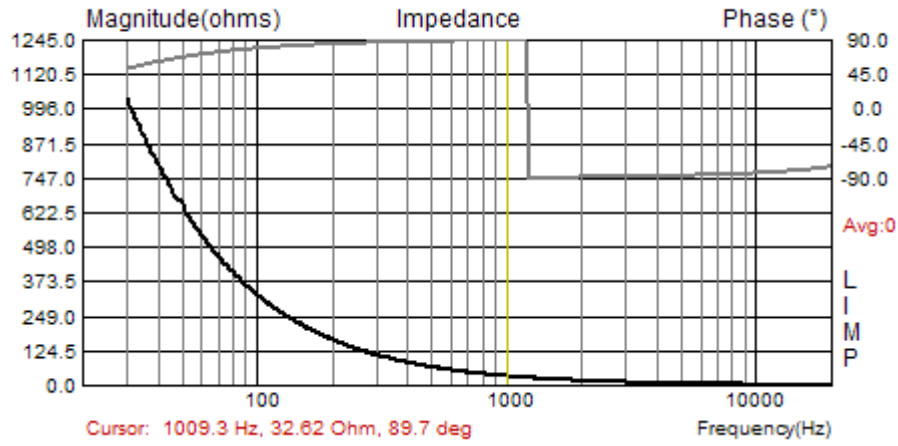
Figure 6.3 Wrongly estimated impedance of plastic capacitor 4.7 uF/250V (measured without the calibration)
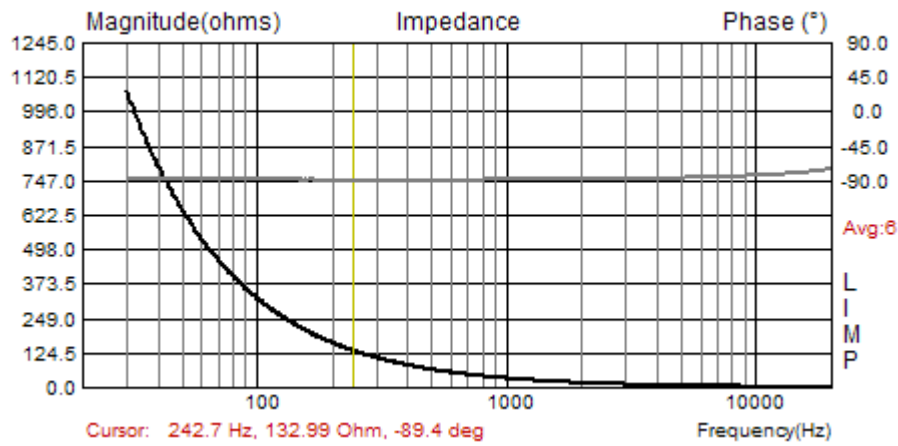

Figure 6.4 Correctly estimated impedance of plastic capacitor 4.7 uF/250V (measured after the calibration)

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