ELectro Acoustic COMPensation

A solution for

quality improvement of

electro acoustic systems

and resultant audio recordings

Content

Abbreviations

ASIO	Audio Stream Input Output - for multi-channel audio streaming
cof	File extension for LRZ filter coefficients
CTR	Cochlea Transformation
ELACOMP	Electro acoustic compensation
emf	File extension for the standard picture format enhanced meta files
dBov	level below maximum undistorted excitation
DSP	Digital Signal Processing
GUI	Graphical User Interface
IR	Impulse Response
LMB	Left Mouse Button
Lng	File extension for language files
LRZ	Loudspeaker Room Listener
mic	File extension for microphone response files
RIFF	Resource Interchange File Format
RMB	Right Mouse Button
SNR	Signal to Noise Ratio
THD	Total Harmonic Distortion (in %)
tim	File extension for sound coefficients
wav	File extension for PCM wave files with RIFF header for standard IO
WINMM	Windows Multimedia

Introduction

The impulse response of a loudspeaker room listener (LRZ) system is not ideal in most cases. The playback sounds unarticulated, dull or shrill leading to considerable decreased intelligibility particularly for speech and so leads frequently to an increased poor sounding echo problem. Reasons for such problems are the properties of the LRZ - transfer characteristics. An impulse at the loudspeaker input will be received at the listener's ears as a series of superimposed impulses with different phase, polarity and magnitude by the convolution with the transfer function of the LRZ-system (Figure 1).
ELACOMP was developed at Sound acoustics research as principle for the compensation of the impulse response of electro acoustic systems.

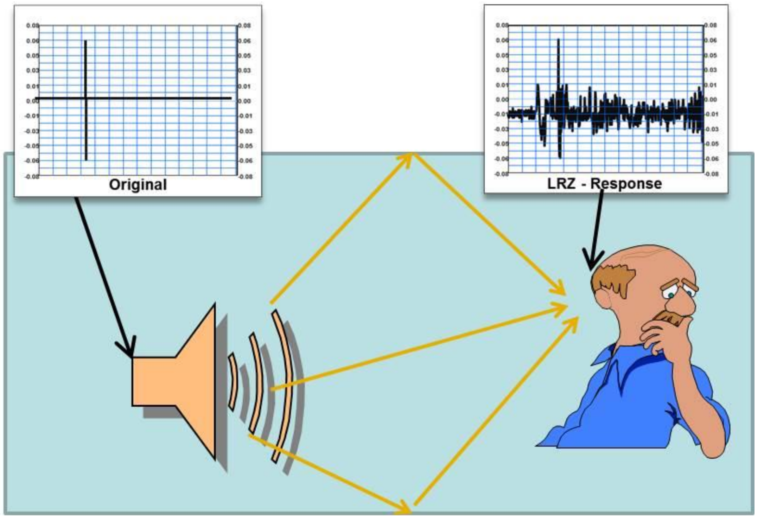


Figure 1: Impulse response of a LRZ system

ELACOMP explores an unknown LRZ system and compensates the impulse response error with nearly no latency with brilliant results. Basses sound dry and music can be played back crystal clear with drastically improved speech recognition.
The correction of the LRZ system can be carried out both individually for each involved channel and combined as for a group of loudspeakers.

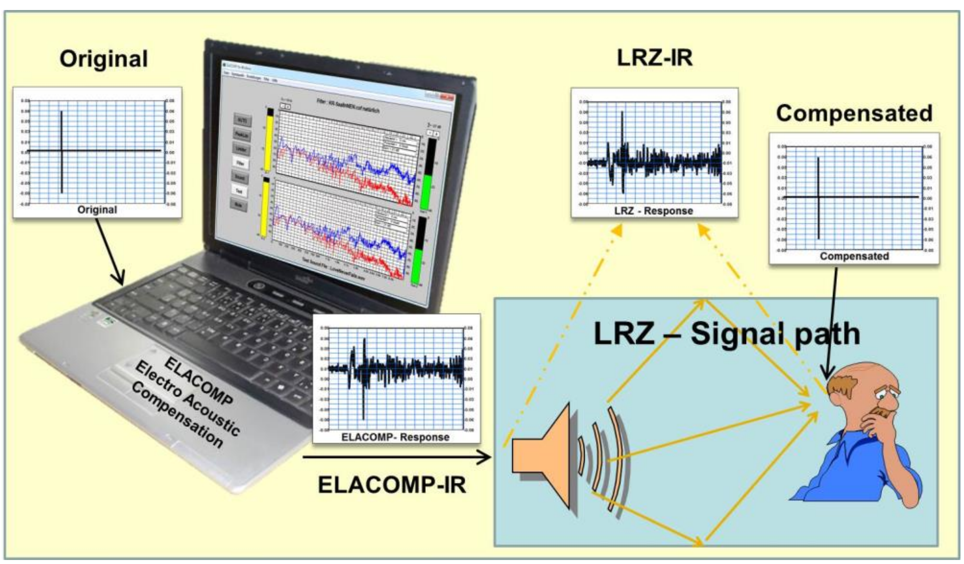


Figure 2: Compensation of LRZ- systems

The ELACOMP - principle is suitable for the implementation in existing audio systems (embedded, ASIC, DSP) and is available as software solution (ELACOMP_W) for all current Windows operating systems.

ELACOMP_W

ELACOMP_W operates with onboard and USB sound cards and supports standard Windows multimedia (WINMM) and ASIO drivers. The software recognizes automatically valid ASIO drivers and makes the compensation of the LRZ- system possible with imperceptible latency and a nearly perfect impulse response at the location of the listener. Different hardware systems can be realized with this universal adaptive solution. This software includes online help, which can be accessed by the known short key F1. The following description covers only a part of the available ELACOMP_W- functions.

User interface

The program control is supported by a program menu with classical menu items and so by a graphically user interface.

Classical Menu

File		
Open Record	Ctrl+O	Open a *.wav file (test signal)
Correct Record		Corrects a record by post processing
Save Record as	Ctrl+S	Save a *.wav file
Load filter coefficients	Ctrl+L	opens the dialog for the selection of a *.cof file (filter)
Save Filter as	Ctrl+F	save the current filter with new name
Save diagram as (.emv)	Ctrl+G	save the current indicated graphics as EMF-picture
Exit		finish ELACOMP program
Signal		
Sine	Shift+F4	creates sine wave and switches to phase and polarity test
Noise signal	Shift+F5	creates white noise
Test Sound File	Shift+F6	playback of the *.wav file for subjective tests and demo
Original Input	Shift+F7	original signal of the connected input interface
Control		
Softlimiter	F9	on/off
Outputs	F6	on/off (muting)
Adapt Microphone	F10	adaptation of the microphone sensitivity to the current acoustic
Mic. Frequency response		Compensation of the measurement microphone
Timbre frequency response		determination of the sound color
View control buttons		hide / view the control buttons in the GUI
Options		
Colors		individual coloring of the GUI
Language		Selection of the language - Deutsch, English - further languages on request
Filters		
Filter	F8	on/off if on the current filter is indicated
Sound	F7	switch between natural (flat) and colored sound
Process New		-
All Channels difference	F4	individual compensation each loud speaker separate (AUTO)
All Channels - sum		single compensation with all loudspeakers (sum)
Channel 1		single compensation of channel 1
Channel 2		single compensation of channel 2
Channel 3		single measurement and compensation of channel 2
Channel 4		single measurement and compensation of channel 4
All Channels - Group		compensation of combined groups 1+2 and 3+4 remaining the stereo base
Help		
About ...	-	opens the dialog box indicating the Software properties
Content	F1	opens the online help books for this software

GUI Controls

The GUI uses keys, buttons, mouse controls and touchpads for the control of the ELACOMP_W software with clear relation to the system and quick access to the desired functions.

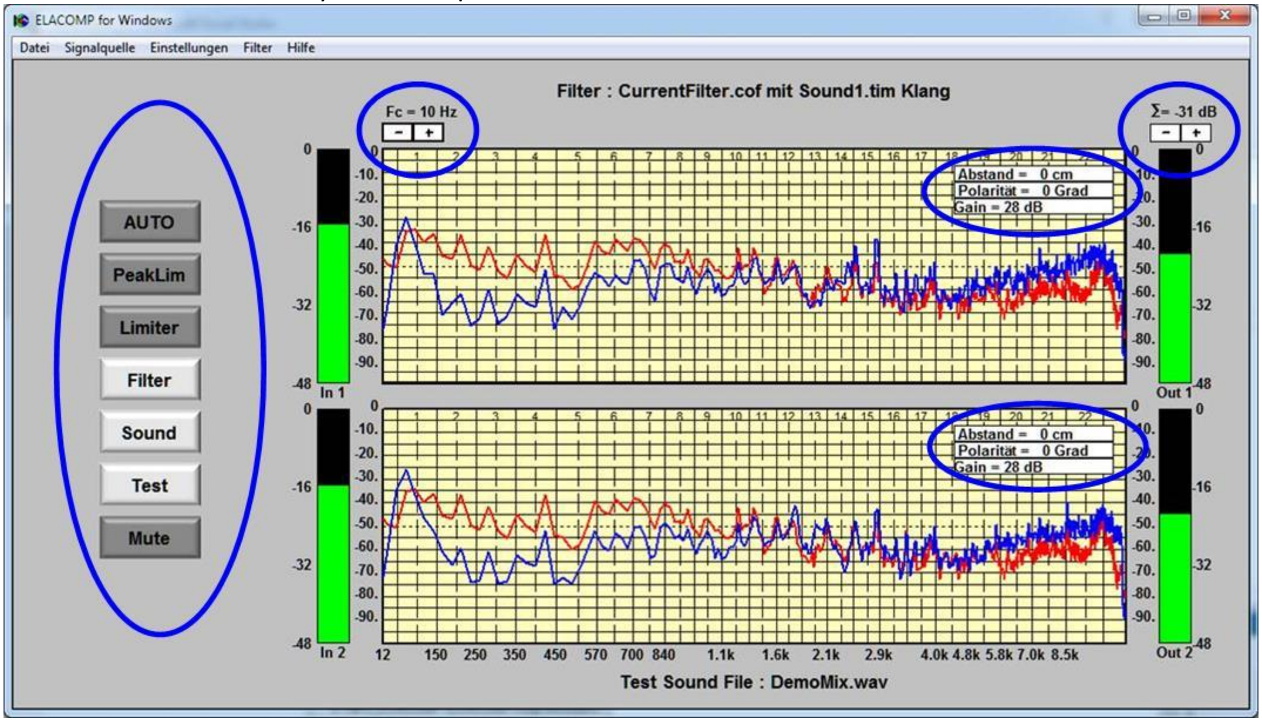


Figure 3: Control units of the graphical users interface

Available Functions

AUTO

The AUTO button is used to start individual and separate measurements and LRZ compensations for each connected loud speaker system. After pressing the AUTO button the filter computation is carried out for each channel one after the next. The measurement procedure is indicated with glowing AUTO button, a blue background colored measurement fields and by the acoustics noise signal taking some seconds for the procedure. It is strongly recommended to be quiet and not to disturb the measurement procedure by strange acoustic noise sources.

After this measurement procedure the Filter button glows and the measurement fields are white colored, indicating that the filter computation is ready and the LRZ compensation is active. The output gain of all channels (Σ) is set automatically to a filter dependent value which should prevent signal overriding at nominal input signal level during active filter operation.

Remark: The AUTO button corresponds with the menu item “All Channels Difference” (F4). Other procedures (under the menu item “Process New”) differ only in the usage of different LRZ paths. All procedures check the presence of loudspeakers at each output channel and switch unconnected interfaces through as all pass filters.

PeakLim

If PeakLim is activated, the button glows and any rare occurring overriding signal peaks are detected and applied for the reduction of the fixed output gain (Σ) in order to make a distortion free playback possible. This function is preferably applied after each new filter computation as some rare excitations at the filter input might lead to very high signal amplitudes as it e.g. occurs if a certain frequency range must be increased strongly in order to become audible. This function sets fixed remaining gains in such manner, that the output of the ELACOMP filter is never overridden even so with maximum excited signal at critical compensated frequencies and hence introduces no dynamic compression as it is known from soft limiters.

Limiter

If the limiter is activated, the corresponding button glows and the level of the output signals are limited at
-3dBov below maximum undistorted excitation level. Rare overriding peaks might lead to audible "breathing" due to dynamic compression.

Filter

The Filter button is used to switch between unfiltered operation with grey measurement window and dark Filter button and filtered operation with white measurement window and glowing Filter button.

Sound

The Sound button is used to extend the flat and clean filter with individual sound color. The background color of the measurement window is changed to yellow and the Sound button is glowing with extended sound. See also Soundfilter.

Test

The Test button makes playback of former loaded *.wav files possible. The playback of known acoustic signals offers the evaluation by individual listening tests and is useful for the representation of the expected results during real time operation.

MUTE

The Mute button is used to switch the outputs silent. Mute is indicated with glowing mute button.

Aids - and add on functions

Compensation of the measurement microphones

A small frame size of an elektret - condenser microphones might result in a flat and clean frequency response which doesn't need to be compensated. Many other microphones however show more or less strong deviations from a flat ideal response and should be compensated according to the indicated frequency response of the belonging data sheet.

The compensation of the microphone frequency response for correct LRZ compensation is aided by the function behind the menu item "Control/Mic. Frequency Response".

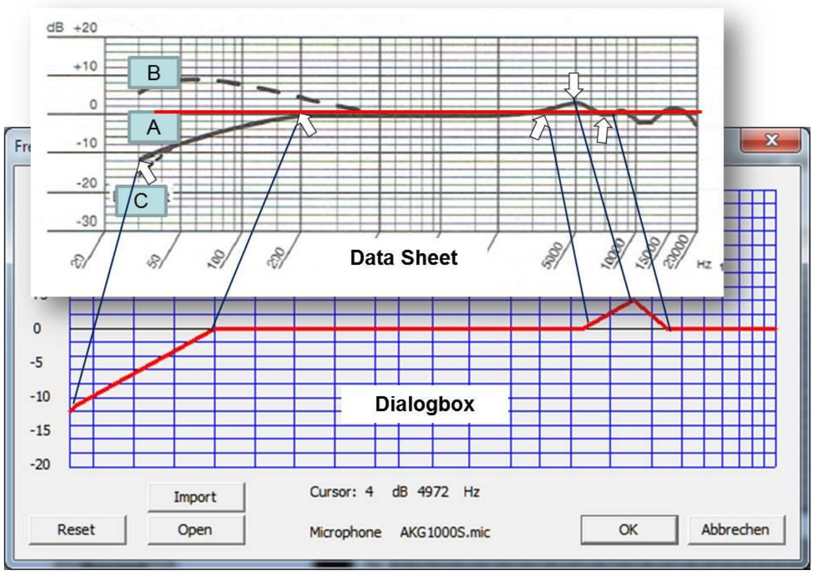


Figure 4: Correction of the measurement microphone according to the data sheet

The determination of the frequency response can be carried out manually and direct with the aid of the GUI or can be imported as table. The direct input using the GUI is mouse controlled. If the curser is moved to a desired frequency point, it can be set with the left mouse button. By this way between 2 and up to 2000 points can be determined. It is recommended to apply the [import function](#).

Sound filter

The LRZ compensation provides a clean playback at the location of the listener without any sound influence (ideal flat filter). Records sound with this ideal filter natural and exactly like recorded.

Figure 5: Determination of an individual sound

A special or individual sound might be very comfortable in order to increase the timbre of a voice, to optimize the sound of musical instruments (string, guitar, piano…) or due to the style (pop, classic). For this purpose a programmable Filter is provided additionally to the LRZ Filter, which can be activated with the aid of the Sound button.

The main advantage is that a clean LRZ compensated signal can be colored precisely without any distortions. The input of the frequency points can be done manually and direct with the aid of the GUI or can be imported as table.

AWAV

The AWAV function can be accessed with menu item "Control/Timbre Frequency response" and aids indirect sound optimization by the current selected recording.

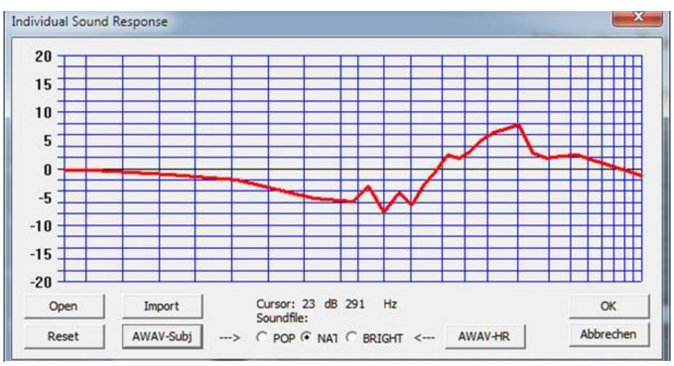


Figure 6: Estimated sound correction by AWAV

The sound design process is determined by the selected AWAV button. AWAV Subj. starts a natural Bark and HR a high resolution process with the desired sound color POP, NAT or BRIGHT.

An indirect sound enhancement can be achieved even so at live concerts, if recordings about live recordings are used for the AWAV function.

The different sound designs make individual adaptations to the desired taste and conditions possible. A Bark based design is in most cases suitable, whereas a Natural correction leads to the simulation of high quality recordings.

The correction with high resolution (HR) is especially for poor recordings with strong undesired resonance behavior suitable, due to the strong cancellation of such distortions, but not suitable for all kind of recordings.

Post Processing Of Recordings

The quality of recordings depends on the applied microphone and the recording room which are in many cases not ideal. Room resonances and partial frequency cancellations might lead to remarkable quality degradation.

After a record is selected and opened, a record can be renewed with the menu item " Correct Record" and might be saved under a new name which is derived and proposed automatically from the original record and the desired design method.

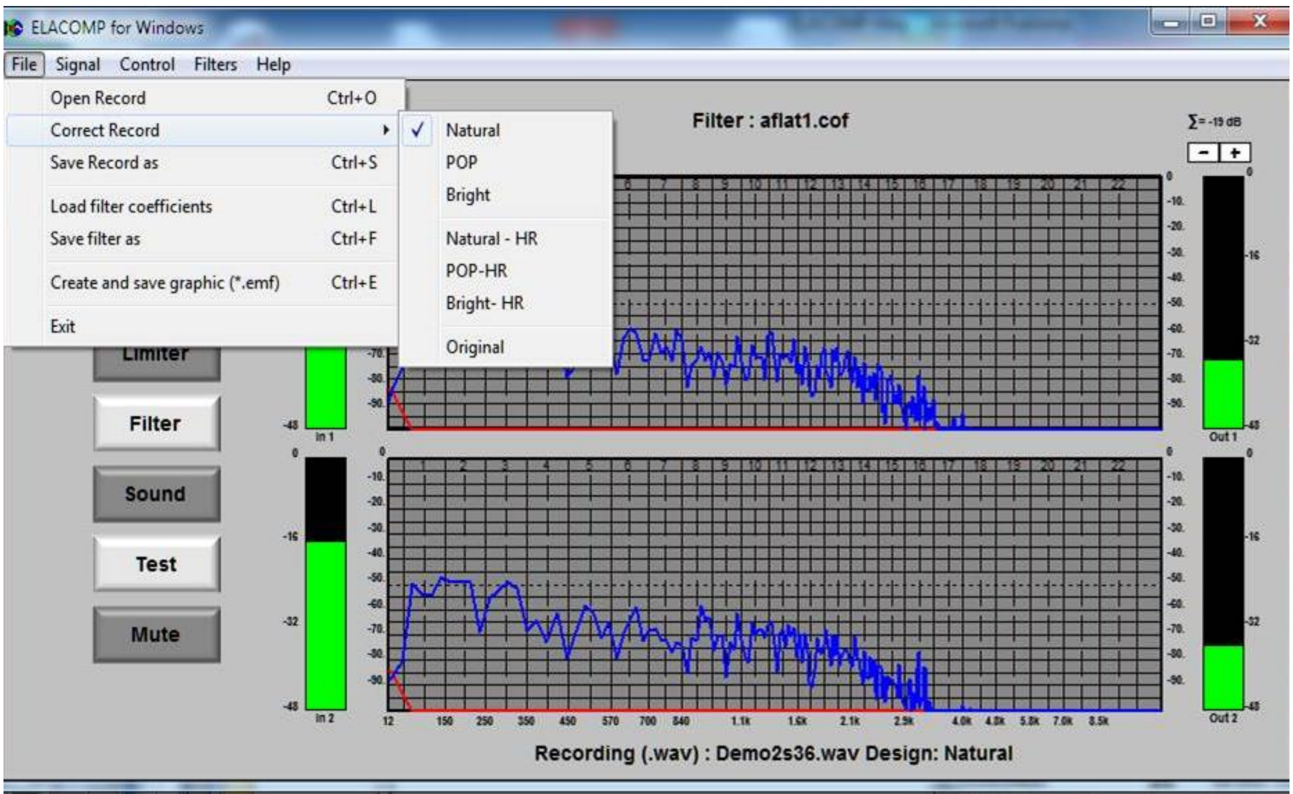


Figure 7: Menu items for post processing of recordings

Following different sound designs are made available for the correction of recordings.

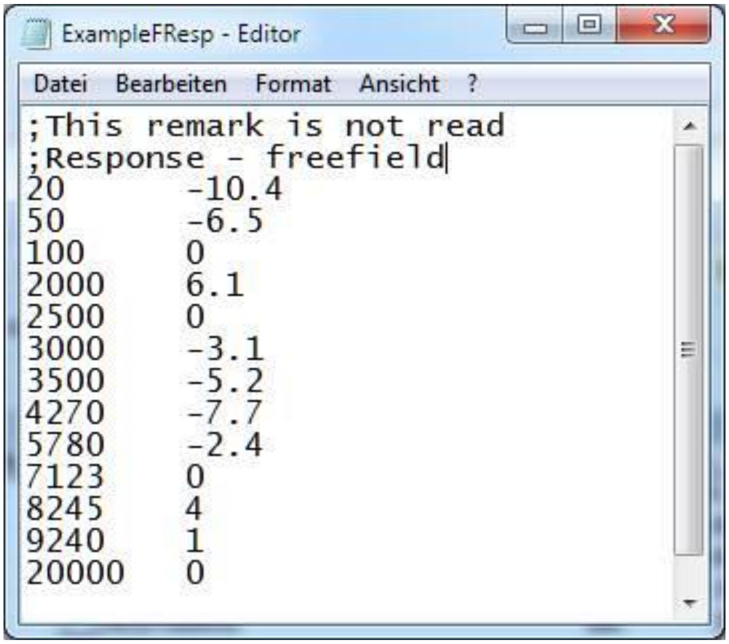
Natural --> Correction with clean uncoloured sound Bark scale based
POP --> Correction with bass and treble enhancement Bark scale based
Bright --> Correction with treble enhancement Bark scale based
Natural-HR --> Correction with clean un coloured sound with High Resolution
POP-HR --> Correction with bass and treble enhancement with High Resolution
Bright-HR --> Correction with treble enhancement with High Resolution

The recording is recomputed after click onto the desired menu item and so indicated with the belonging sound design as here shown with Natural. The different sound designs make individual adaptations to the desired taste and conditions possible. A Bark based design is in most cases suitable, whereas the Natural correction leads to the simulation of high quality recordings.

The correction with high resolution (HR) is especially for poor recordings with strong undesired resonance behaviour suitable, due to the strong cancellation of such distortions, but not suitable for all kind of recordings.

Import function

Both frequency responses of the microphone compensation and the sound color can be loaded with the aid of the import function from a simple ANSI text file. The determination of the desired frequency response is carried out as tabular frequency - level list to be saved as ANSI text file as shown below.




```
;This remark is not read
;Response - freefield
20	-10.4
50	-6.5
100	0
2000	6.1
2500	0
3000	-3.1
3500	-5.2
4270	-7.7
5780	-2.4
7123	0
8245	4
9240	1
20000	0
```

Figure 8: Simple text file for the determination of a frequency response

The example in Figure 8 shows how a table must be built in order to determine a desired frequency response. Each line is consisting of the coordinates X with the indication of the frequency in Hz and Y with the indication of the corresponding level in dB. X and Y are separated with a TAB and Y ends with a new line.

The input of the frequency takes place with integer numbers and the Y with floating point numbers with decimal dot.

Lines starting with semicolon are interpreted as remarks and are skipped during import.

The import is carried adaptively to the number of determined frequency points. The list must indicate at least 2 points but not more than the maximum of 2000 points. Points between points are linear interpolated.

Delay compensation

A phase and delay compensation of a loudspeaker path difference can be achieved by the introduction of an artificial delay of the nearest loud speaker. In this case the loudspeaker signal with the lowest distance to the listener must be delayed exactly with the same time as the acoustic sound needs to travel the path difference.

A correct compensation can be achieved with the aid of the measurement program which is started automatically with the selection of Signal / Sine in the program menu as follows.

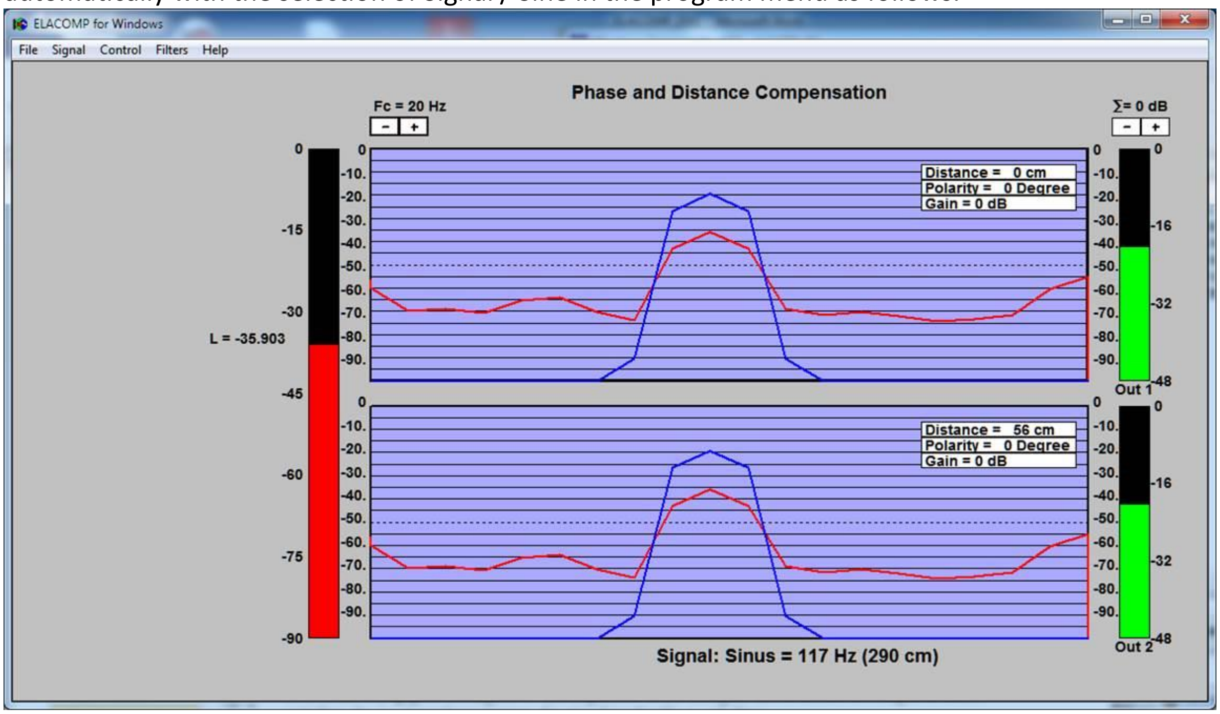


Figure 9: Phase- and delay compensation by maximum level adjustment

Distance

A difference in the distance between loudspeakers and the listener location might lead to audible frequency dependent sound degradations.

A compensation of this difference can be achieved if the signal of the nearest loudspeaker to the listener is delayed by increasing the distance. The level measurement L as indicated in Figure 9 aids a precise phase correction by maximum level adjustment dependent on the controlled distance.

The distance can be changed with the mouse wheel or the + and - buttons of the keyboard if the cursor is moved to the indicated field “Distance = xx cm” of the corresponding channel and the cursor has changed to NS arrow. The step size of the changings can be changed between 1, 7 and 70 cm with the right mouse button.

Please notice: The sine wave frequency is adjustable and amounts 117 Hz in this example which corresponds to a wavelength of 290 cm covering possible estimation errors. See also Signal.

Polarity

The polarity of the signal can be switched between 0 and 180 degree with the aid of the left mouse button if the cursor is located onto the field polarity.

Gain

The Gain can be changed with the mouse wheel or the + and - buttons of the keyboard if the cursor is moved to the indicated field “Gain = xx dB” of the corresponding channel and the cursor has changed to NS arrow.
The step size amounts 1 dB. The settings are saved automatically and are available with every new program start. The individual level adjustment of each channel makes a correction of the perceived loudspeaker location possible.

With the left mouse button and the same cursor position it is possible to mute a single channel for certain examinations as e.g. the influence of the superposition of the muted loudspeaker or if the loudspeaker fits to the assumed channel.

Signal

The frequency of the sine wave can be adjusted with the aid of the mouse wheel or the + and - buttons of the keyboard if the cursor is moved onto the indicated frequency (Figure 9: Sinus = 117 Hz).
The step size of can be changed between 1, 10 and 100 Hz with the right mouse button onto the indicated frequency.

Miscellaneous SW - functions

Further functions as e.g. a programmable dialog box for the selection and determination of the sound card operation or alternatively a recognition function for ASIO drivers are described in the online help books of the ELACOMP software.

ELACOMP_W is subject to a continuous software maintenance including the upgrades of the corresponding help books.

Software installation and first operation

This ELACOMP software is made available at our download page www.sound-acoustics.eu as password protected zipped file ELACDesign.7z.

Software installation:

It is necessary to be authorized to install following program as administrator onto the destination computer as follows.

- Download and save ELACDesign.7z onto your computer
- Unpack setupELACOMP.exe with the aid of the pass word from Sound acoustics
- Start setupELACOMP.exe and follow the program instructions

Hardware connection:

- Connect your sound card with the USB interface of your computer if you intend to run this software not with the onboard sound card
- Install the corresponding sound card drivers if necessary
- Connect channel 1 or the left channel with a microphone
- Connect all signal interfaces (in- and output) of your sound card with your audio devices according to one of the indicated examples of the online help

Taking ELACOMP into operation:

Start ELACOMP.exe . Each available channel will be indicated in the display on top of each other. This software supports up to four channels and can be extended on request.

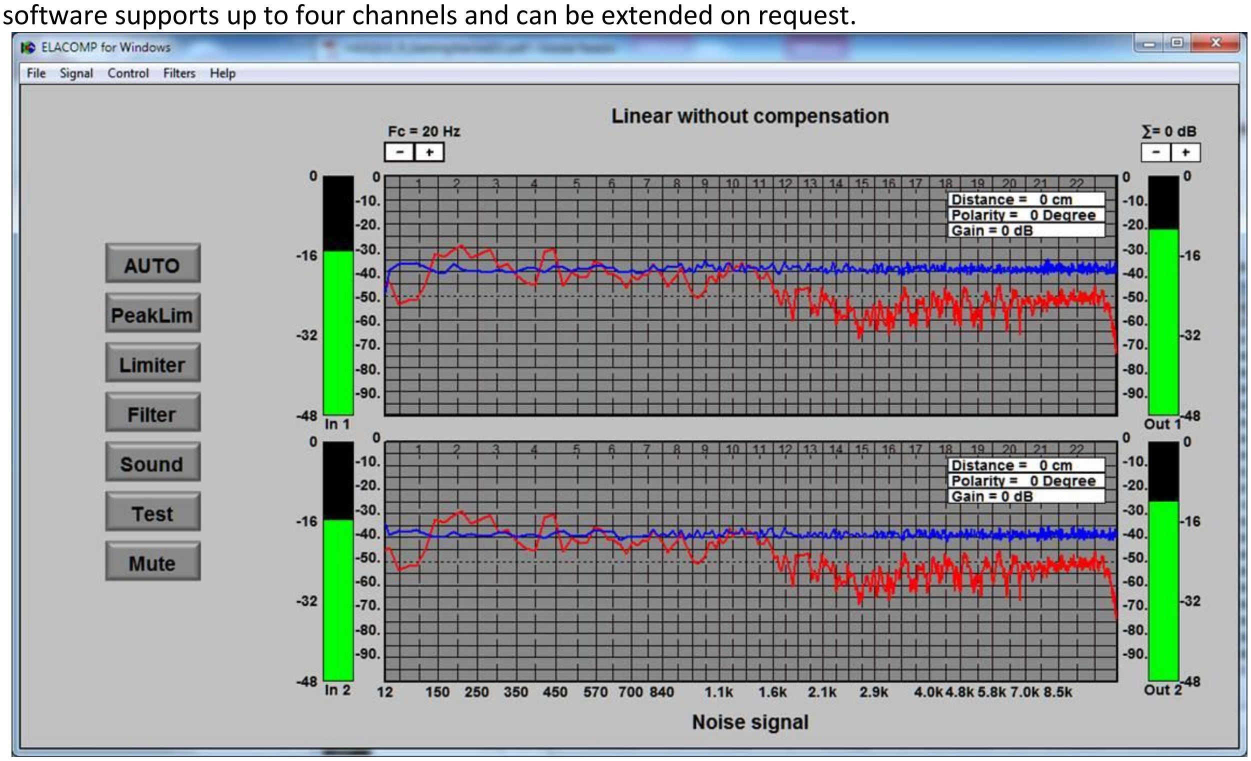


Figure 10: Two channel operation with selected noise signal

Select Signal / Noise signal (Shift + F4) in the program menu, switch the microphone on and adjust the necessary microphone sensitivity in this way that the output level (right green level bars) of the microphone spectra (red) amounts about -16 dB like the input level (left green level bars) of the reference spectra (blue). The microphone should be placed at the listener's location during this test. If the adjustment is not possible e.g. due to missing or to weak control range you might apply the automatic gain control by pressing the F10 key. Switch the noise signal off (Shift + F4).

It is possible to check and correct phase and polarity of the connected loudspeaker systems before impulse response compensation is carried out if desired. For this purpose you apply the corresponding test program by the selection of the menu item Signal / Sine (Shift +F4). The corresponding test procedure is described in the online help books.

Start the filter computation by pressing the AUTO button and wait until the Auto button goes out.

Test operation:
Open an audio file (e.g. ELAC4.wav or your own test file) with the menu item File/open (or short key CTRL+O). Press the Test button and adjust the volume to the desired playback loudness. You might switch between filtered and unfiltered operation with the Filter and Sound button in order to make the impulse response compensation and sound extension audible.

Normal operation:
Normal operation can be selected with the menu item Signal / Original Input (Shift + F7), whereas the analog input interfaces are applied as signals to be compensated and filtered.
In this operation it is also possible to switch between filtered and unfiltered operation with the Filter and Sound button in order to make the impulse response compensation and sound extension audible or to select the desired filter for the real time operation.

It is recommended to adapt any external volume and gain or level control of the peripheral hardware (e.g. mixer, amplifier…) in order to prevent overriding of the signal interfaces.

Hardware solutions

ELACOMP_W is designed for all Windows operating systems and supports different sound cards. It works with ASIO or WINMM drivers and operates with onboard and USB audio devices with different sampling rate and a bit resolution of 16 and 24 Bit. ELACOMP_W recognizes Steinberg ASIO drivers making the LRZ compensation at the listener's location without perceptible latency possible.

ELACOMP_W checks the availability of valid ASIO drivers. If valid ASIO drivers are installed and the corresponding sound card is connected, the system starts automatically with the settings of the ASIO driver. If no ASIO drive is found, ELACOMP_W searches for other standard audio devices, and opens a dialog box with all available sound cards for the selection and programmable settings of the desired audio device.

Operation with ASIO drivers:

This Software is supporting ASIO drivers from the Steinberg UR series.

Figure 11: UR22 with connected microphone

Sound cards with ASIO drivers provide as a rule remarkable lower latency (<10 ms) than standard audio devices for PC operating systems (typ. >>10ms) and hence are qualified for real time sound systems.

Other applications as e.g. the playback of recordings don't require extremely low latency giving more freedom for the selection of a sound device.

Following operating conditions result from sound cards with ASIO drivers:

Operation with UR22

The UR 22 (Figure 12) is consisting of two symmetrical input and output interfaces. Input interfaces are fitted with combi jacks, which might be used optionally for microphones with and without phantom voltage as well as line interface to individual signal sources. The input sensitivity can be adjusted for each channel with separate potentiometers.

The software expects at Input1 a microphone for the LRZ measurement. An extension with two microphones is intended but not implemented yet. Connected microphones might be removed after filter computation, if the interface is needed as line in for the compensation of other signal sources.

The output interfaces (1/L and 2/R) should be connected with interfaces to loud speakers (e.g. amplifier input or the input of an active box).

The LRZ measurement and computation can be started if above connections are carried out and the connected devices are playing without distortions. The correctness of the LRZ compensation can be checked with the available test function, which does not need the input interfaces.

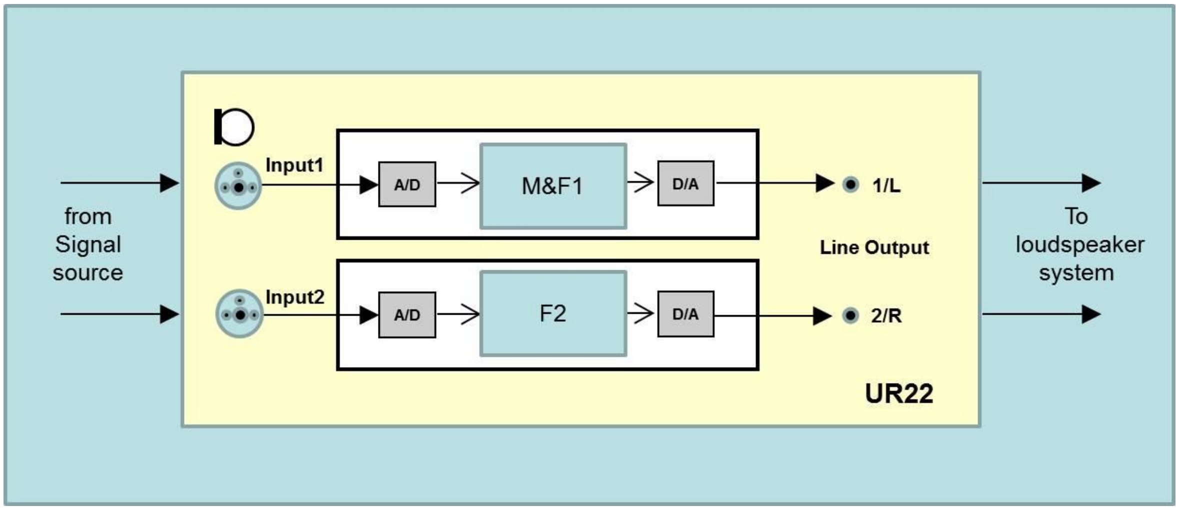


Figure 12: Operation with UR22

A standard real time operation over input1 and input2 can be selected per software with the selection of the program menu item "Signal / Original Input" (Shift + F7). As the microphone is only needed for the impulse response measurement during LRZ filter computation, input1 and input2 might be used for the compensation of any signal source. Hence this input interfaces might be connected with a stereo mixer output or other audio equipment. It is also possible to apply this interface for the compensation of microphone signals to reach best possible sound quality for musicians.

Important notice: The audio interfaces of the UR22 should be connected properly with screened cable and clean contacts and should operate with level adjustments.

Operation with UR242

The UR 242 (Figure 13) is consisting of two symmetrical input (line 3 and 4) and output interfaces (1L and 2R) and additionally of two input interfaces (input 1 and 2) which are fitted with combi jacks for microphones with and without phantom voltage. The input sensitivity of input 1 and 2 can be adjusted for each channel with separate potentiometers. The software expects at input1 a microphone for the LRZ measurement. An extension for two microphones is intended but not implemented yet.

The output interfaces (1/L and 2/R) should be connected with interfaces to loud speakers (e.g. amplifier input or the input of an active box).

The LRZ measurement and computation can be started if above connections are properly carried out and the connected devices are playing without distortions. The correctness of the LRZ compensation can be checked with the available test function, which does not need the input interfaces (Line 3 und 4).

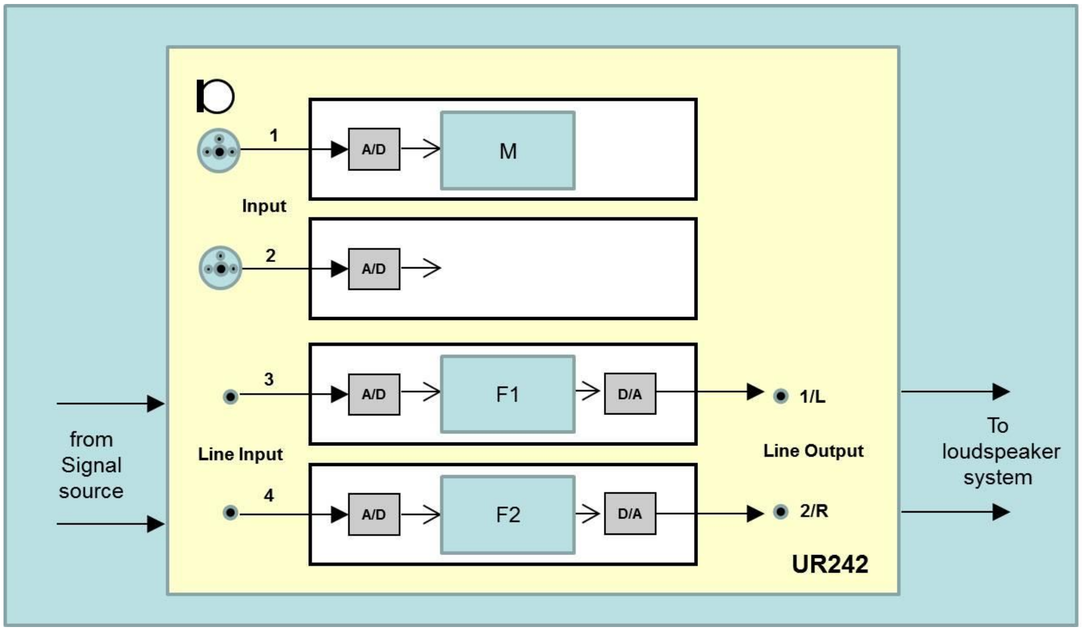


Figure 13: Operation with UR242

A standard real time operation over Line input 3 and 4 can be selected per software with the selection of the program menu item “Signal / Original Input” (Shift + F7).

Line input 3 and 4 might be used for the compensation of any signal source. Hence this input interfaces might be connected with a stereo mixer output or other audio equipment. It is also possible to apply this interface for the compensation of microphone signals to succeed in best possible sound quality for musicians.

Important notice: The audio interfaces of the UR242 should be connected properly with screened cable and clean contacts and should operate with correct level adjustments. Furthermore it must be ensured, that the adjustments of the software mixer dspFxMix, which belongs to the UR242, must not introduce any effects nor signal mixing and that the settings which are indicated in the online help are met.

Other sound cards with ASIO drivers

The UR series with ASIO drivers from Steinberg are automatically recognized and applied if available and connected. The program start might be blocked, if another audio device is connected. Please contact Sound acoustics in this case, as we will try to support all others if possible.
Please note that only one card sound with ASIO driver can be applied and hence only one sound card with ASIO driver should be connected.

Operation with WINMM drivers:

ELACOMP_W searches for available audio devices if no audio device with ASIO driver is available and opens following dialog box (Figure 14).

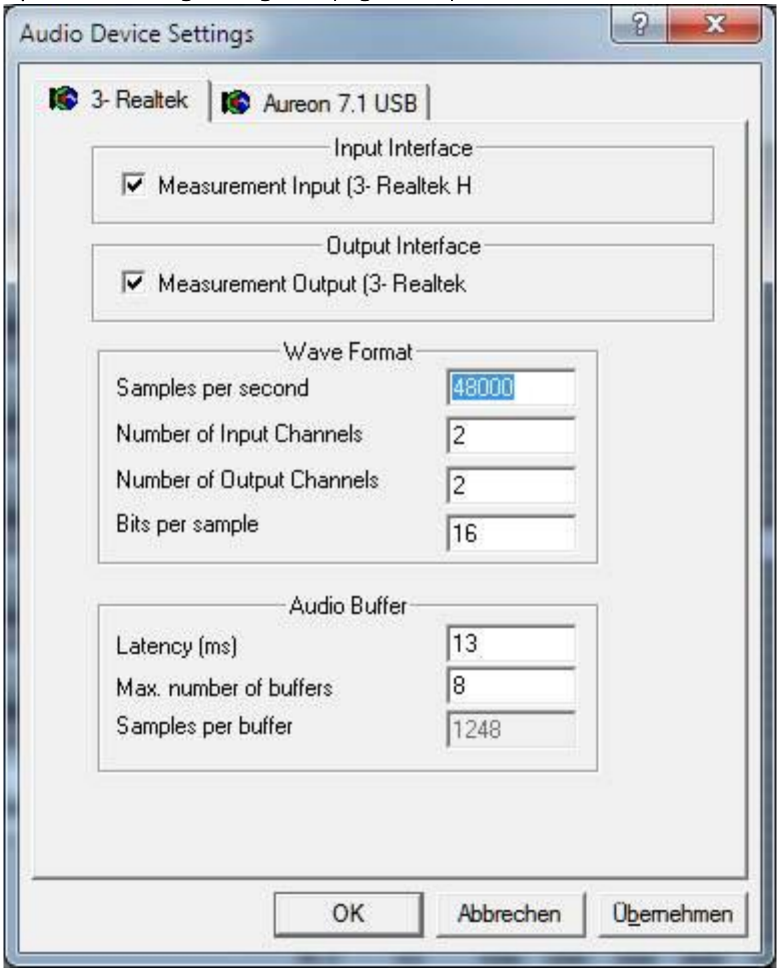


Figure 14: Dialog box for the selection of sound cards

The PC is fitted in this example with two audio devices, an onboard - sound card (Realtek) and an Aureon7.1 USB.

The selection of the desired sound card is carried out with the corresponding check boxes to be marked.

The setting of the wave format must equal the Windows settings to be accessed in System control\Hardware and Sound\Sound.

The Audio buffers are determined with the desired latency to be typed into the corresponding edit box, whereas the WMM-driver might need higher latency dependent on the operating system and the available computational power.

The minimum possible latency is hardware dependent and must be examined iteratively.

Hardware selections are as a rule rare and might be carried out only once after software installation, or if a new hardware is connected. In these cases the dialog box in Figure 14 opens automatically.
In other cases, the dialog box might be opened with the hidden key combination CTRL+ALT+A, making other audio device settings or selections possible.

Product implementation and DSP solution:

The ELACOMP - principle is modular structured, designed for optimum computational power and is suitable for the implementation onto DSP, ASIC or other PC - platforms.

Each channel needs about 20 MIPS at a sample rate of 48 kHz. The data memory effort amounts about 30 Kbyte plus about 30 Kbyte program memory. Precise specifications are only possible with clear product and hardware specifications.

Technical Data

This software is suitable for applications with different sampling rate, operates with 16 and 24 Bit resolution and supports ASIO drivers and WMM streaming for the operation of USB and onboard sound cards. Technical data result from the applied sound cards and are application dependent. Following specifications are relevant for the ELACOMP_W software:

Operating system:	Windows Vista … Windows 10
Signal to Noise Ratio:	typ. >120 dB
Sample rate:	all frequently used sample rates
Latency:	<1ms
Frequency response:	typ. 20Hz - 20 kHz (dependent on the sampling rate)

Figures