CONEQ™ Workshop

Acoustic Power Frequency Response
Measurement and Equalization System

REFERENCE MANUAL

(for software version 3.2.0)

real sound lab

real sound lab

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Part I
OVERVIEW

Real Sound Lab's CONEQ™ is a technology to measure and equalize the acoustic power frequency response (APFR) of electro-acoustic transducer devices, such as loudspeakers. With CONEQ™ every loudspeaker sounds better – clear, natural, and without attenuations – within its whole frequency range. The CONEQ™ technology is applied in two steps. First, a precise measurement of how well the loudspeaker transmits energy at each frequency is done, and a compensation filter is automatically produced. Second, the filter is applied to the audio signal by any of the software or hardware tools supporting CONEQ™.

This manual describes all the features of the CONEQ™ Workshop application – the tool for measuring and filter synthesis. For a quick introduction to CONEQ™ Workshop we suggest reading the CONEQ™ Workshop Quickstart Guide first. Return to this manual after having done the first measurements to learn about all the features in detail.

The screenshots in this manual are taken on the Windows operating system. The corresponding screens on Mac OS will look similar and the functionality on both operating systems is the same.

APPLICATION REQUIREMENTS

SUPPORTED OPERATING SYSTEMS

CONEQ™ Workshop has been tested to run on the following operating systems:

- Windows XP, Service Pack 3
- Windows Vista
- Windows 7
- Mac OS X 10.5.8
- Mac OS X 10.6.8
- Mac OS X 10.7.2

The CONEQ™ Workshop for Windows requires Microsoft .NET framework 2.0 and the set-up program will install the framework if necessary.

AUDIO REQUIREMENTS

CONEQ™ measurement is done by sending a series of test signals to a device-to-be-measured (typically, a loudspeaker) and recording the output of the device (typically, using a microphone). Proper measurement is crucial to achieve the best possible results with the CONEQ™ technology. That is why several points about the system set-up must be taken into account before using CONEQ™ technology:

- The sound card or audio interface (audio device) and, if necessary, drivers must be properly installed and configured;
- An omnidirectional measurement microphone with flat free-field frequency response must be used. An acceptable alternative is to use an omnidirectional microphone with a known, electronically available individual frequency response curve in a format that is compatible with CONEQ™ Workshop;
- The audio device must be working properly and be compatible with the CONEQ™ Workshop application. The audio device must be capable of simultaneous (full duplex) playback and recording, and both the audio device and the measurement microphone, along with the connecting cables, must be in perfect working order;
- A loop-back test should be performed to check for any undesired disturbances in the playback/recording path.

Failure to take the steps mentioned above will increase the probability of incorrect measurements, which may cause undeserved disappointment about the achieved results, CONEQ™ Workshop software, and CONEQ™ technology in general.

DIFFERENCES BETWEEN CONEQ™ WORKSHOP EDITIONS

LICENSES

There are three editions of the CONEQ™ Workshop application:

- CONEQ™ Workshop Unlimited
- CONEQ™ Workshop Annual
- CONEQ™ Workshop Limited

All editions are functionally equal but the Annual and Limited editions have limitations on the usage time. The Annual edition will work for a year (53 weeks) from its activation. To continue using CONEQ™ Workshop, a new license must be acquired. The Limited edition is similar to Annual but will work for 30 days from its activation.

FEATURES

Table 1: List of major features of CONEQ™ Workshop

Filter sampling frequency (kHz)	0.5 / 1 / 2 / 8 / 16 / 32 / 44.1 / 48 / 88.2 / 96
Filter resolution (number of taps)	1024 and 4096
Support for APEQ™ hardware equalizers	Yes
Support for CONEQ™ P2/P8/P2pro/P8pro plug-ins	Yes
Support for CONEQ™ COMPATIBLE third-party products	Yes
Adjustable limits for maximum gain/attenuation	Yes
Parametric equalizer for defining target curve	Yes
Graphic equalizer for defining target curve	Yes
Custom target curve files	Yes
Microphone calibration file support	Yes
Composite (averaged) measurement	Yes
Measurement from pre-recorded file	Yes
Phase Filters	Linear / Minimum
Changeable smoothing (octave)	1/6 / 1/3 / 1/2 / 1
Changeable test signal periods	4
Changeable time window	8 + custom

USING COPY-PROTECTION DEVICE

CONEQ™ Workshop is protected against illegal copying using eLicenser technology. It is a hardware based copy-protection system and the eLicenser copy-protection USB device is

Figure 1: The eLicenser copy-protection USB device.

delivered together with the software. It must be connected to the computer before starting CONEQ™ Workshop.

The eLicenser devices delivered with the Unlimited edition of CONEQ™ Workshop are ready to use right out of the box. The eLicenser of the Annual and Limited editions contains a non-active time-limited license. It must be activated to start the countdown and enable use of the software.

To activate a time-limited license, connect the eLicenser and open the eLicenser Control Center application (***Start⇒All Programs⇒eLicenser⇒eLicenser Control Center***). The connected eLicensers will appear on the list. Select the CONEQ™ Workshop Annual/Limited license and choose ***Actions⇒Start License Usage Period*** from the menu (see figure 2). Follow the instructions and after successfully completing the activation process, the CONEQ™ Workshop software is ready to run.

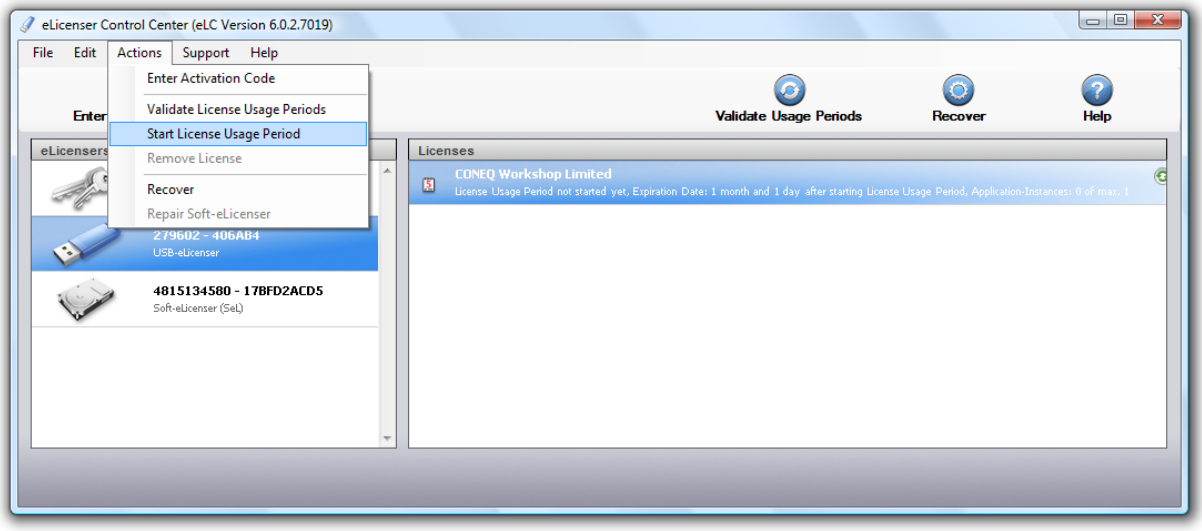

Figure 2: Activation of the time-limited license for CONEQ™ Workshop Annual/Limited.

The activation process requires an Internet connection. If the computer where the CONEQ™ Workshop is to be used is not on-line, then activation can be done on any other computer after installing the eLicenser Control Center application there. The latest version of the eLicenser Control Center application can be downloaded from http://www.elicenser.net/en/latest_downloads.html

STARTING CONEQ™ WORKSHOP

To start the application, use ***Start⇒Real Sound Lab⇒CONEQ Workshop***

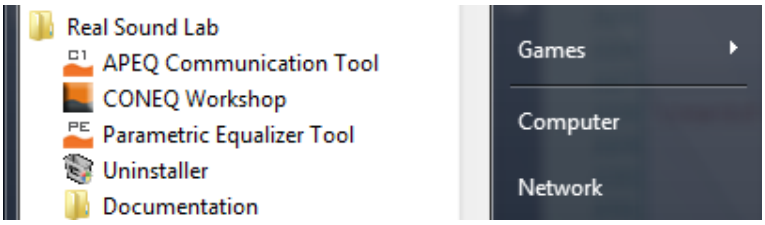

Figure 3: CONEQ™ Workshop in the Start menu.

CREATING/OPENING A PROJECT

To start working with CONEQ™ Workshop a project must be created. A project in CONEQ™ Workshop is a collection of measurements and filters. To create a new project select ***Project⇒New*** from the menu.

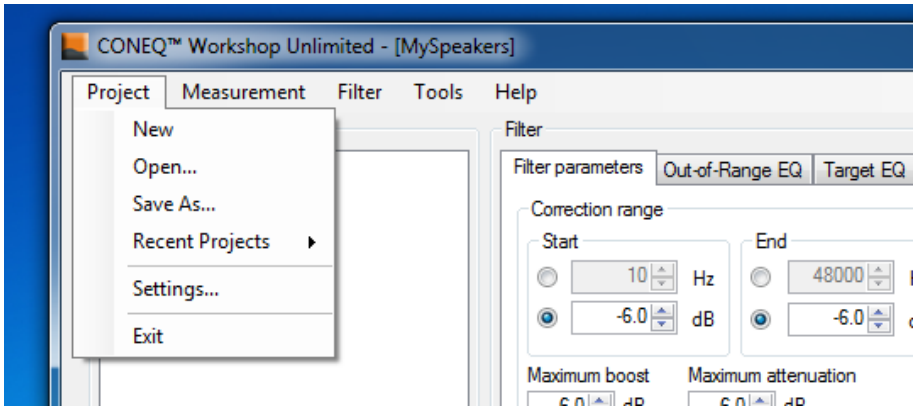

*Figure 4: The **Project** menu.*

A folder selection dialogue (figure 5) will open. CONEQ™ Workshop projects are folders on the hard disk. To create a new project, select the parent folder where the project folder is to be stored. Then write the name of the project in the corresponding field. The full project path, including the project folder will be shown at the bottom of the dialogue.

For instance, to save a project in the folder `Measurements/MySpeakers`, use the folder selection dialogue to select the folder `Measurements` and enter `MySpeakers` as the project name.

To open an existing project, choose ***Project⇒Open...*** from the menu and point CONEQ™ Workshop to the project folder.

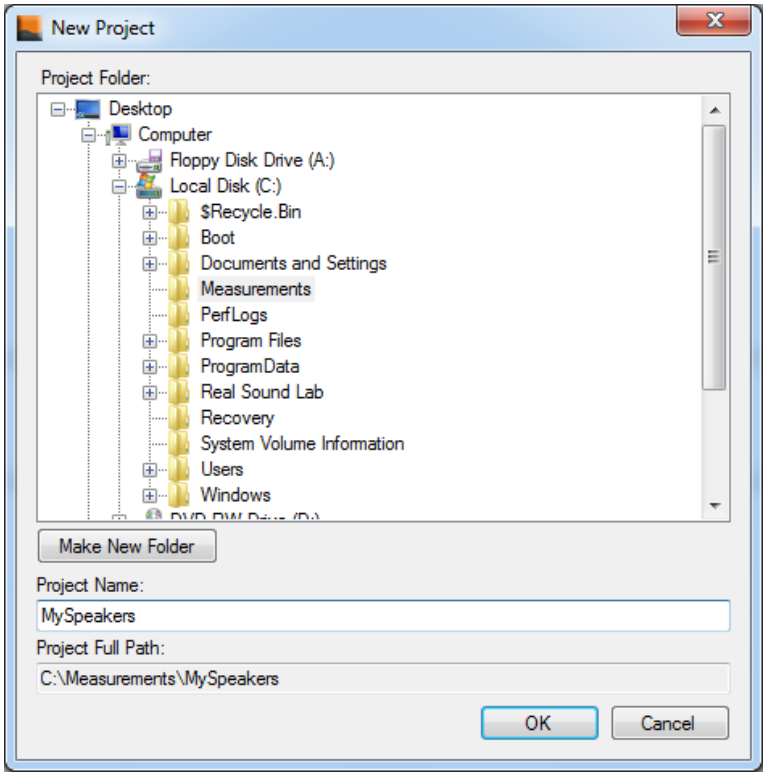

Figure 5: The new/open project dialogue.

> ***IMPORTANT!*** *If an existing project was created by CONEQ™ Workshop 2.x then opening it will require a conversion. The original project will be left unmodified and CONEQ™ Workshop will ask for the location where to put the converted project. The converted projects cannot be opened with earlier versions of CONEQ™ Workshop.*

MAIN WINDOW

The main application window has the following areas:

- Measurements
- Parameters
- Graph

The measurements list at the top left is where all measurements of the currently open project are shown. For each measurement, all filters derived from it are shown.

The parameters area at the top right is where the parameters of the currently selected filter and measurement are shown. For newly created projects with no measurements, this area is empty. Parameters (such as correction range, filter resolution, low/high frequency cut, target curve, etc.) can be changed and either applied to the current filter or used to create a new filter.

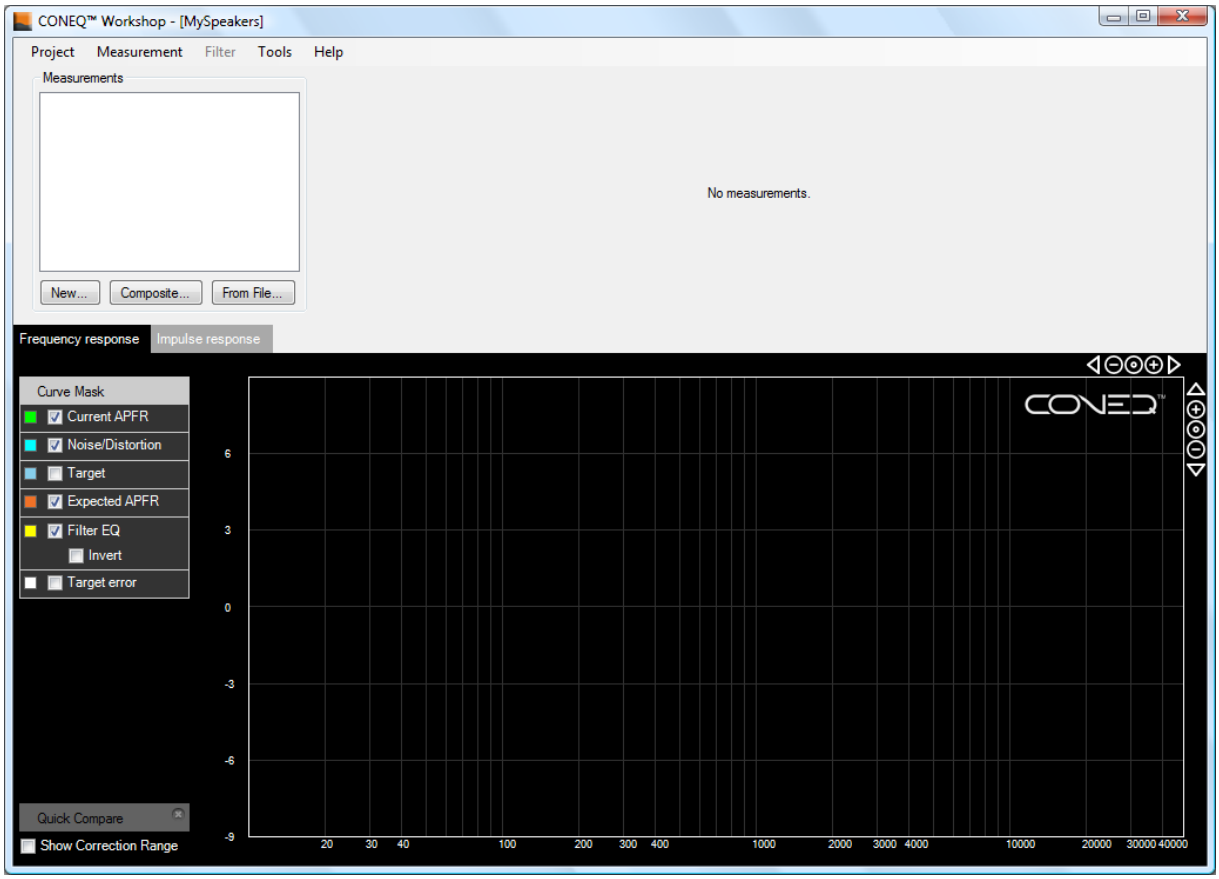

Figure 6: The main application window with a new project just created.

The Graph area at the lower part is where the data of the currently selected measurement and filter are visualized. It features horizontal and vertical zoom. Some curve parameters (e.g. zero level or a target curve EQ) can be edited with the mouse directly in the graph.

The next part of this document describes all the features in detail.

Part II
REFERENCE MANUAL

Here comes the description of every function of CONEQ™ Workshop.

PROJECTS

PROJECT FOLDER

A project in CONEQ™ Workshop is a collection of measurements and filters. It is stored in a folder with a specific file and sub-folder structure. There is a sub-folder for each measurement and a further sub-folder for each equalization filter. Figure 7 shows an example project folder structure.

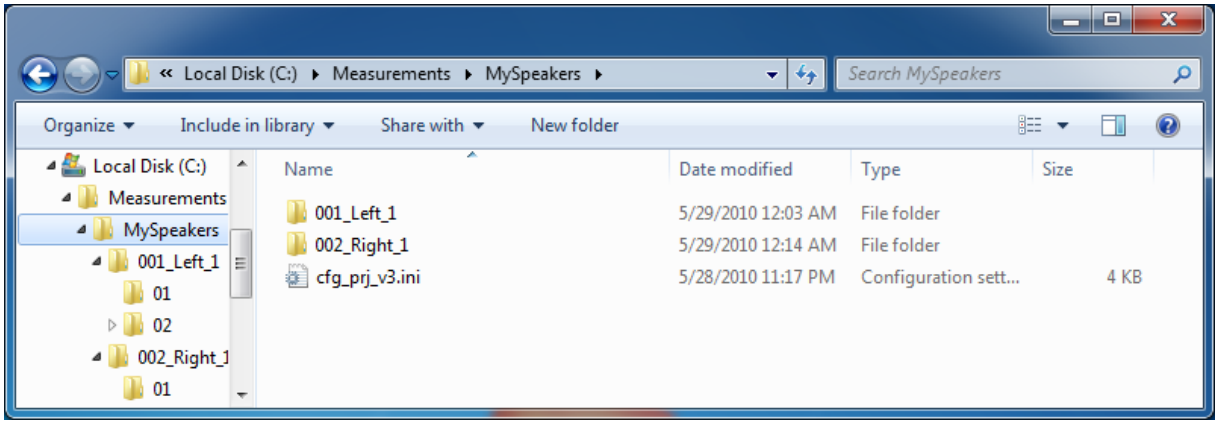

Figure 7: The structure of a CONEQ™ Workshop project folder.

The project and its folder are called `MySpeakers` and all project-related data are stored in this folder. The project contains two measurements – **Left_1** and **Right_1**. Each measurement is stored in an own folder within the project folder. In this case, the corresponding folders are `001_Left_1` and `002_Right_1`. Each version of the equalization filter of a measurement is stored in an own folder within the folder of that measurement. In this case, the **Left_1** measurement has two versions of the equalization filter stored in folders `01` and `02`, whereas the **Right_1** measurement has only one filter in the folder `01`.

MEASUREMENT FOLDER

Each measurement folder contains the recorded test signal WAV file, the calculated impulse response data file (e.g. `MySpeakers_001_Left_1_IR.IR`), and the measurement information file `cfg_msr.ini`. The measurement folder contains sub-folders for each equalization filter version. The filter sub-folder names are sequential two-digit numbers (01-99).

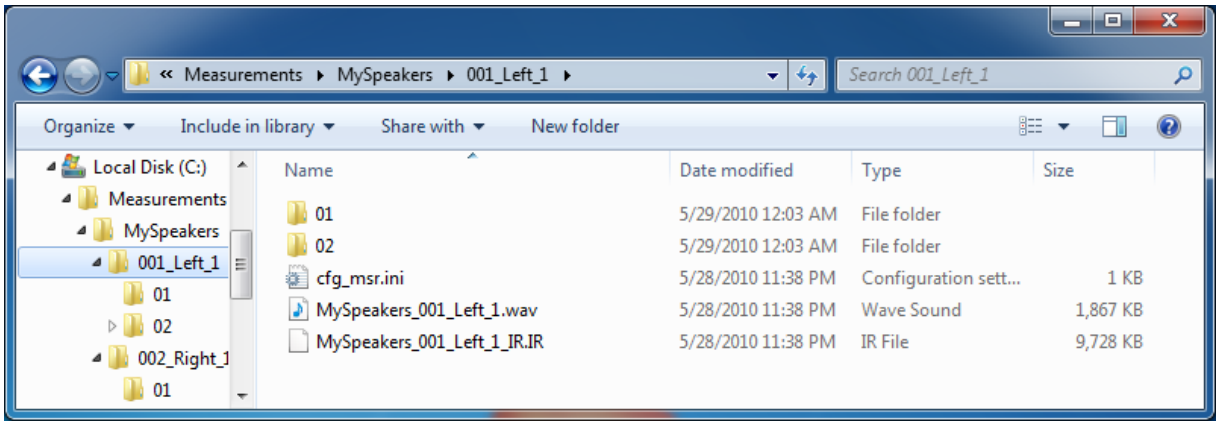

Figure 8: Contents of a measurement folder.

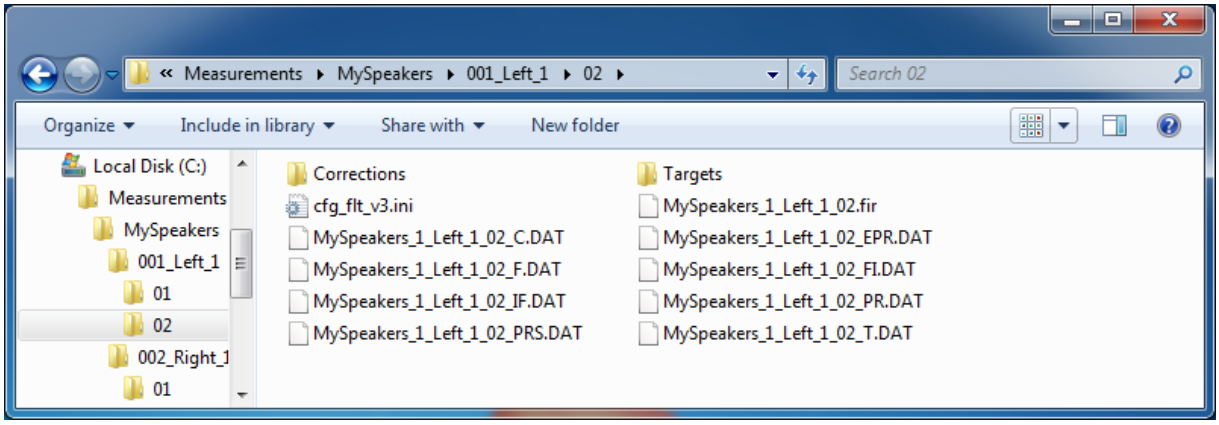

Figure 9: Contents of an equalization filter folder.

EQUALIZATION FILTER FOLDER

Each equalization filter folder contains data files for the equalization filter. The most important file is the CONEQ™ equalization filter file with the extension `.FIR` (FIR file). This is the file that can be used by the software and hardware tools that perform CONEQ™ equalization, e.g. APEQ™ hardware equalizer and CONEQ™ software plug-ins. The FIR file is created automatically when the filter is created or modified.

The table 2 gives the summary of the file names of various data files in the filter folder. Most data files of a measurement share a common part. This part consists of the project name, an underscore, a measurement number, another underscore, the measurement name, and the filter index. An example of a common part for the second filter of a measurement named **Left_1** in a project named **MySpeakers** is `MySpeakers_1_Left_1_01`. The table 2 lists all possible file names in the filter folder and describes the contents of the corresponding files.

All `.DAT`, `.GEQ`, `.PEQ`, and `.FIR` files are usable as target or correction curves for other filters and can be displayed as custom curves for comparison purposes.

Table 2: The data files in the filter folder.

File description	File name
Filter parameters	`cfg_flt_v3.ini`
CONEQ™ equalization filter	`*.FIR`
Combined APFR correction	`*_C.DAT`
Expected APFR after CONEQ™ equalization	`*_EPR.DAT`
Filter graph	`*_F.DAT`
Inverted filter graph	`*_FI.DAT`
Noise/Distortion	`*_IF.DAT`
Acoustic power frequency response	`*_PR.DAT`
Smoothed APFR with shelves	`*_PRS.DAT`
Combined target EQ	`*_T.DAT`
Used target EQ curves	Folder `Targets`
Used correction curves	Folder `Corrections`

PROJECT MENU

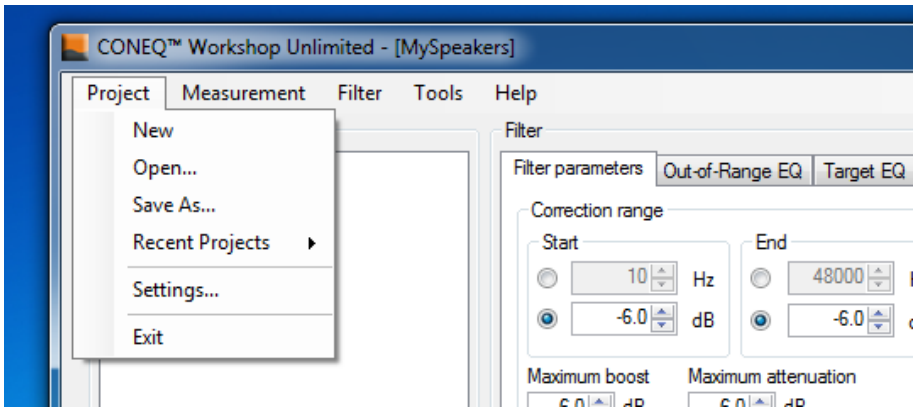

*Figure 10: The **Project** menu.*

New...

Create a new project. It was described on page 8.

Open...

Open an existing project. It was described on page 8.

Save as...

Save the current project with all the measurements and filters to another location. This is also the supported way of renaming projects.

> ***IMPORTANT!*** *Renaming only the project folder using the means of the operating system will break the project and CONEQ™ Workshop will not be able to open it. To rename a project, open it in CONEQ™ Workshop and use* ***Project⇒Save as...*** *and give it the new name.*

Recent projects

Open one of the most recently used projects.

Settings...

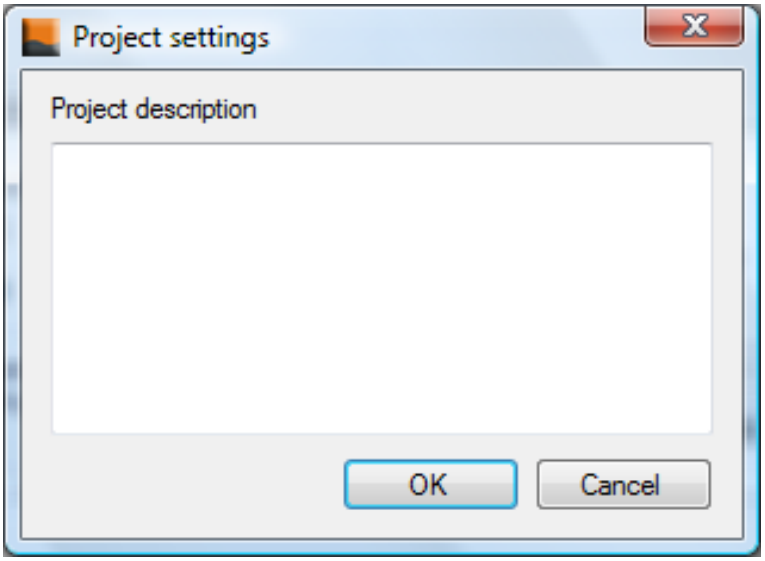

Figure 11: The project settings window.

Open a window as in figure 11 to edit the description of the current project.

Exit

Close the CONEQ™ Workshop application.

MEASUREMENTS

The measurement list at the top left of the main window (figure 12) displays the measurements in the current project. For each measurement, a list of filters that are derived from the measurement is displayed. If a filter has a description (see page 23) then that is shown in parenthesis. In the measurements list only the filters can be selected. The currently selected filter is called the current filter in this document. The measurement from which the current filter is calculated is called the current measurement.

Click the **New...** button to initiate a new measurement (see page 15). Click the **Composite...** button to create an average of several previous measurements (see page 19). Click the

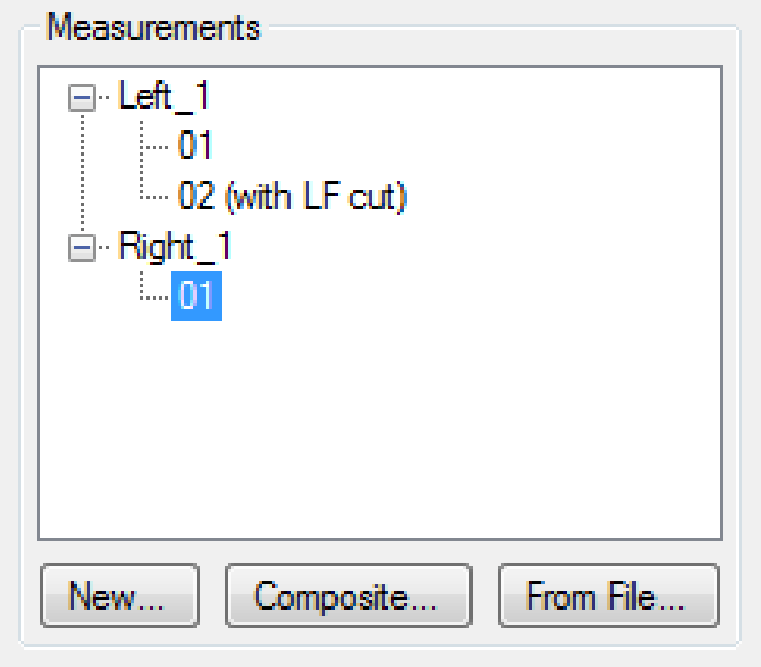

Figure 12: The measurement area of the main window.

From file... button to create a measurement from a pre-recorded WAV file (see page 19). This is useful for importing a single measurement from another CONEQ™ Workshop (2.x and 3.x) project.

NEW MEASUREMENT

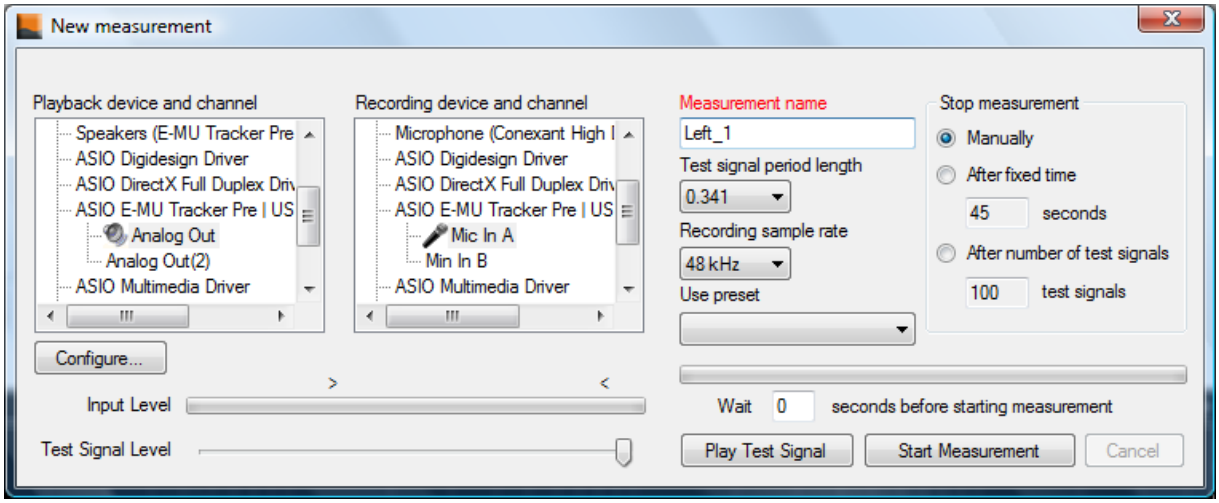

Figure 13: The new measurement window.

The new measurement window (figure 13) allows taking a new measurement according to given parameters.

Playback device

A playback device and a channel must be selected. That is the device and channel to which the test signal will be sent. Typically it will be the sound card connected to speakers that are to be measured. The output volume should be adjusted to normal listening level. The preferred way of adjusting the volume is by using the means of the audio interface or amplifier but it can be done also using the slider control below the list of playback devices.

The default test signal output level slider setting is –30dBFS to prevent very high output signal levels. However, this control is purely digital and affects the resolution of the test signal. Therefore, if the used sound output devices (e.g. audio interfaces, amplifiers) have their own hardware level controls then use them to attenuate the output signal and set the slider in CONEQ™ Workshop to the maximum level.

The output signal level setting is remembered for the current session but is reset to the safe –30dBFS level on the next start of the CONEQ™ Workshop application.

> ***IMPORTANT!*** *Be careful to set the playback volume to a safe level to avoid distortion, prevent damaging the equipment, and exclude possible damage to hearing.*

Initially, the primary sound playback device is selected by default. Later, the same device and channel as used in the last new measurement is selected by default. The device choice is saved across sessions.

If the selected playback device is an ASIO device, then the **Configure...** button is enabled and brings up the ASIO control window, provided by the driver manufacturer. Selecting an ASIO device for playback will select the same device also for recording.

Recording device

A recording device and a channel for receiving the test signal must be selected. Typically this is a sound card and channel where the measurement microphone is connected. The sensitivity must be adjusted so that the incoming signal is loud enough but does not clip. This must be done by using the means provided by the audio interface or microphone preamplifier.

Initially, the primary sound recording device is selected by default. Later, the same device and channel as used in the last new measurement is selected by default. The device setting is saved across sessions.

If the selected recording device is an ASIO device, then the **Configure...** button is enabled and brings up the ASIO control window, provided by the driver manufacturer. Selecting an ASIO device for recording will select the same device also for playback.

If the selected recording device is APEQ–8pro DIO, then clicking the **Configure...** button brings up the APEQ–8pro DIO recording configuration dialogue (figure 14). This allows switching on the phantom power and adjust the sensitivity level of the microphone input.

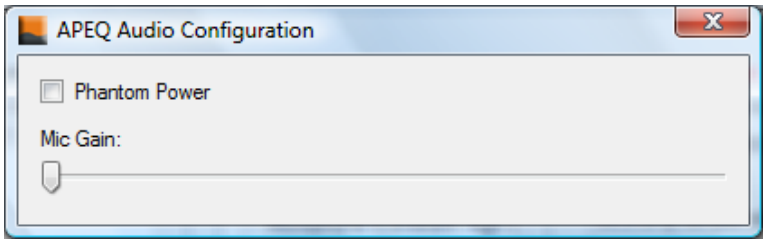

Figure 14: The APEQ-8pro DIO recording configuration dialogue.

Measurement name

Each measurement needs a name by which it will be shown in the measurements list. The name is entered in the **Measurement name** field. It is not recommended to give several measurements the same name. Measurements can be renamed later.

Test signal period length

The test signal period length must be selected. There are four different lengths provided: 0.341, 0.683, 1.365, and 2.731 seconds. The default period of 0.341 seconds is suitable for small to medium size rooms. Longer sweep periods are required for larger and very reverberant or very noisy venues, or when measuring low frequency drivers (subwoofers).

Recording sample rate

The measurement sample rate must be selected. 48 and 96 kHz sample rates are supported. It is recommended that the measurement has the same or higher sample rate than the filters that will be generated from it. Otherwise, the measurement will provide no useful information for equalization of the highest frequencies. It is recommended to always use 96 kHz recording sample rate if the audio interface supports it.

Using presets

Presets are user-defined sets of parameters for new measurements and filters. Presets can be edited by using the Preset Editor (see page 37). The new measurement uses parameter values from the preset that is chosen in the **Use preset** field. The default selection in the **Use preset** field is **<Default for this project>**.

Ways to stop the measurement

There are three choices for how to stop the measurement once it is started:

- Manually, by pressing the **Start/Stop measurement** button;
- After a fixed time in seconds;
- After a fixed number of test signals.

Taking the measurement

Click the **Start test signal** button to start playing the test signal on the selected playback device and channel. This also initiates monitoring of signal level on the selected recording device and channel. The button text will change to **Stop test signal**.

While the test signal is playing, the output signal level and input sensitivity should be adjusted so that the peak level indicator is predominantly between –30 and –5 dB levels (the ">" and "<" marks, respectively). The level meter should never fill completely because that means signal clipping (overload). If that happens then the volume or sensitivity should be reduced.

When the input level is adjusted appropriately, click the **Start Measurement** button to start the measurement. There are situations when some time is needed to prepare for the measurement (e.g. the speakers are far from the computer). For such cases it is possible to specify the time to wait until the actual recording starts after the **Start Measurement** button has been pressed.

When the measurement is started, the progress bar will show the measurement progress. Just above the progress bar, elapsed time in seconds and the number of recorded test signals is displayed. The **Cancel** button can be used at any time during the measurement to stop the measurement process. The new measurement window will not be closed, however. To close it, use the window close button in the top right corner.

After a successful measurement is made, the new measurement window will close. The first filter for the new measurement will be calculated based on the preset that was used for the measurement. The new filter will be selected in the measurements list.

MEASUREMENT MENU

The figure 15 shows the ***Measurement*** menu which provides means to create, import, delete, and rename measurements as well as produce a composite (average) measurement.

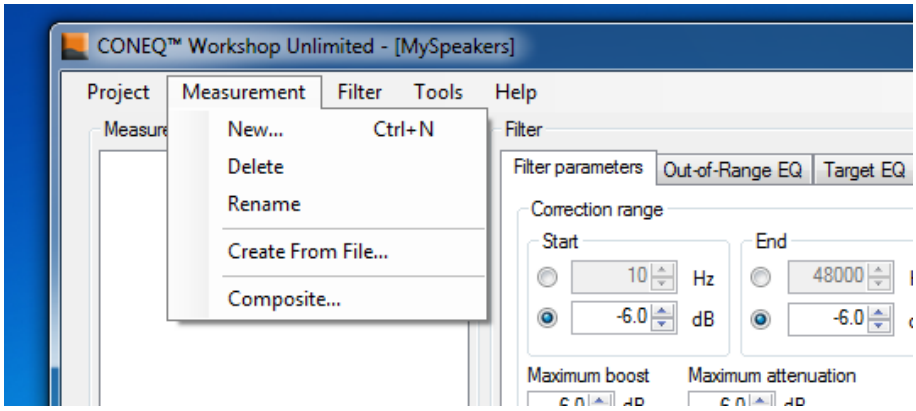

*Figure 15: The **Measurement** menu.*

New...

Create a new measurement. It was described on page 15.

Delete

Delete the current measurement from the project. The measurement data along with all the corresponding filters will be removed from the project. The corresponding measurement folder from the project folder will be moved to the Recycle Bin (Windows) or Trash (Mac). This is done to allow recovery of measurement data if it was deleted by accident.

Rename

Rename the current measurement.

Create from file...

Create a measurement from a file that contains a recorded sequence of test signals. This is useful for cases when it is impossible to record the test signal with the same computer that is generating the test signal or as a way to import measurements from other projects (including previous versions of CONEQ™ Workshop).

The file must be a 48 or 96 kHz, 16 or 24-bit, signed little-endian integer PCM mono file in WAV format.

Composite...

CONEQ™ Workshop allows producing a "measurement" that is an average of the acoustic power frequency responses of several measurements. The composite result is an average of the components calculated in the acoustic power domain. Composite measurement is used, for example, to exclude any measurement error by repeating the same measurement several times, or make a measurement of the full measurement surface that is impossible with a single measurement because of technical or physical limitations.

Any number of measurements in the current project can be selected. It is possible to adjust the level of each measurement relatively to others. Increasing the level means that the measurement will be more dominant in the composed result.

A particular filter version from each measurement must be selected for composition. Information about the filter version is needed because filters define APFR zero level adjustment, time window and correction curves – parameters that influence the acoustic power frequency response.

The figure 16 shows the measurement composition window and table 3 describes the meaning of controls.

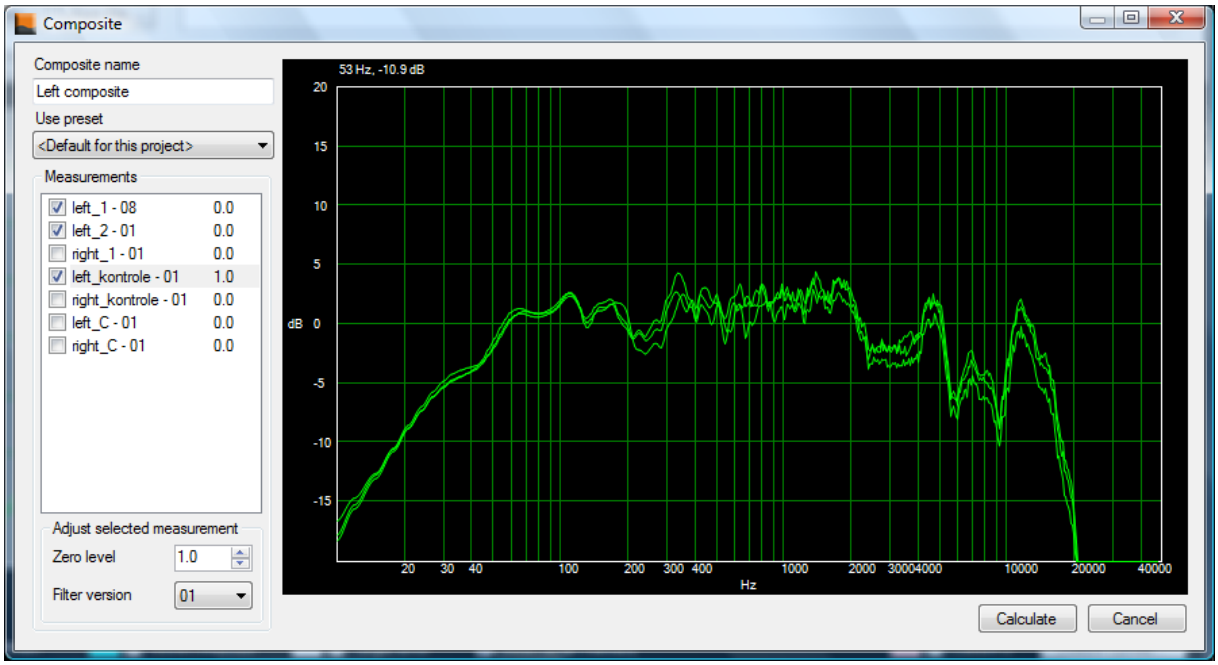

Figure 16: The measurement composition window.

Table 3: The composite measurement window controls.

Field	Description
Composite name	The name of the resulting composite "measurement".
Use preset	Use the settings from the selected preset for the first equalization filter calculated from the composite.
Measurements	All measurements of the current project are listed. Use checkboxes on the left of the measurement name to include the measurement in the composition. The measurement name is followed by the index of the currently selected filter that will be used for composition. The APFR level adjustment value is shown in the second column.
Zero level	Adjust the APFR level of the selected measurement. This way the "importance" of each included measurement can be adjusted.
Filter version	Select the filter version of the currently selected measurement which will be used for the composition.
Graph area	The graph shows the APFRs of the checked measurements. The graph of the selected measurement is highlighted. The graph can be zoomed in or out on both axis. Use the mouse wheel to zoom in and out vertically. Hold the **Ctrl** key (**Shift** on Mac) to zoom horizontally.
Calculate button	Start the composition process.
Cancel button	Close the window.

FILTERS

The parameter area (figure 17) at the top right of the main window shows the parameters of the currently selected filter. All the parameters that determine how a CONEQ™ equalization filter is generated can be adjusted in the multiple tabs of the parameter area.

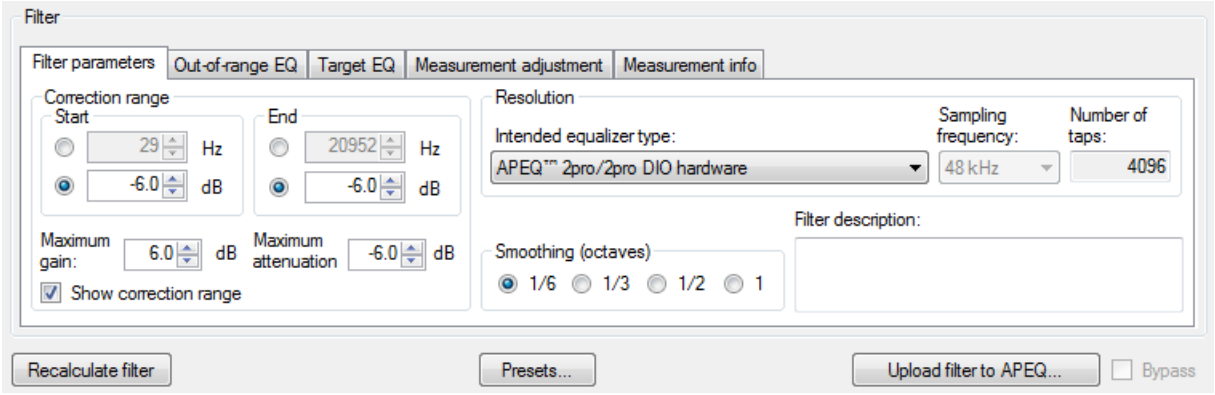

Figure 17: The filter parameter area.

At the bottom left of the parameter area, there is a button **Recalculate filter** to create a new filter based on the chosen parameter values. On the bottom right there are controls to upload the currently selected filter to an APEQ™ hardware equalizer, if one is connected. At the bottom centre there is a button **Presets...** to apply predefined sets of parameter values and to open Preset Editor (see page 37).

FILTER PARAMETERS

Correction range

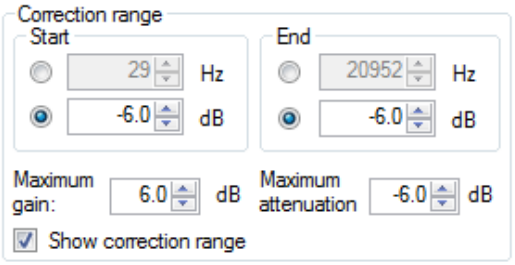

Figure 18: The correction range parameters.

The CONEQ™ equalization will only be applied in a certain frequency range (typically the frequency range of the loudspeakers). There are two ways to specify the start and end frequencies of the CONEQ™ correction range.

One way is to specify the level of the APFR at which the equalization should start/end. In this case the start (end, respectively) frequency will be the lowest (highest) frequency at which the APFR level is at least the specified level and is at least 10 dB over the noise/distortion level. This is the default and suggested way for typical loudspeakers.

The other way is to specify explicitly the frequencies at which the equalization should start/end.

The maximum boost and maximum attenuation of the equalization filter will be limited to levels in the respective fields. This is to prevent too much gain leading to distortion or even damaging the speakers and too much attenuation reducing the maximum power of the speakers. These parameters limit the filter to be in the specified bounds regardless of the APFR zero level adjustment, target curves, and correction curves. The produced equalization filter will never exceed these maximum boost and attenuation limits. The low/high frequency cut settings (see **Out-of-range EQ** settings tab on page 23), however, override these limits.

The **Show correction range** check-box controls the visibility of the correction range on the graph.

Resolution

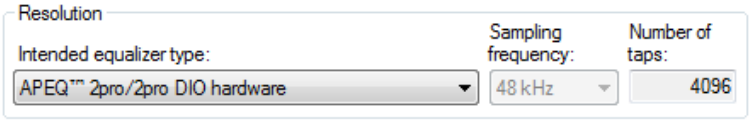

Figure 19: The filter resolution parameters.

CONEQ™ Workshop allows generation of filters for the following products supporting CONEQ™ equalization filters:

- APEQ-2pro/2pro DIO hardware
- APEQ-8pro DIO hardware
- CONEQ P2/P8 plug-in
- CONEQ P2pro/P8pro plug-in
- CONEQ™ COMPATIBLE product (1024 taps)
- CONEQ™ COMPATIBLE product (4096 taps)

Choosing a tool will select the maximum filter resolution (precision of the filter) supported by the corresponding tool. The **Number of taps** field shows the precision of the filter. The more taps a filter has the better the precision.

For CONEQ™ equalization the sample rate of the filter must match the sample rate of the audio signal. Therefore, for tools that support several sample rates, it is possible to select the filter sample rate from the **Sampling rate** selection list. The list adapts to the chosen equalizer type. For the APEQ-2pro/2pro DIO hardware it is always set to 48 kHz, for the APEQ-8pro DIO hardware the supported sample rates are 44.1, 48, 88.2, and 96 kHz. For CONEQ™ plug-ins and CONEQ™ COMPATIBLE products the allowed sample rates are 250, 500 Hz, 1, 8, 16, 32, 44.1, 48, 88.2, and 96 kHz.

Smoothing

The CONEQ™ equalization filters are generated from a smoothed APFR and not the originally measured one. This is done to avoid compensating for small APFR irregularities that may not be real defects of the speakers but rather are measurement errors. The APFR is smoothed using a raised cosine impulse filter with a resolution from 1/6th (more precise) to a full octave (more smooth).

Filter description

The **Filter description** field can be used to describe the particular filter version. If used, the filter description will appear behind the corresponding filter entry in the measurement list.

OUT-OF-RANGE EQ

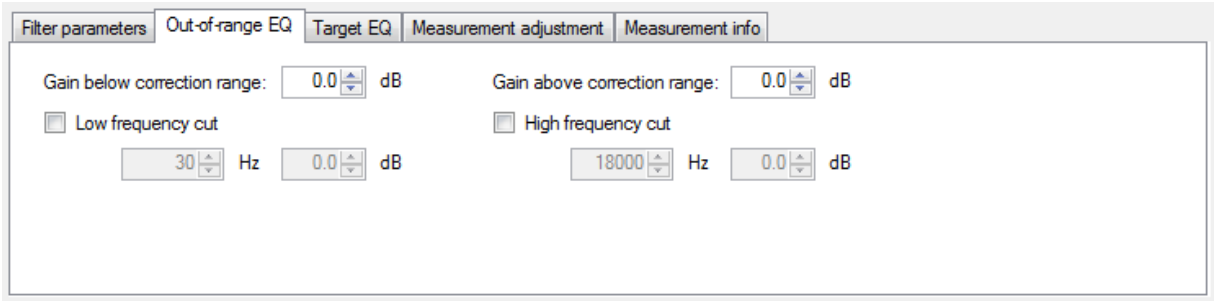

Figure 20: Out-of-range gain and low/high frequency cut parameters.

Frequencies below the correction range will be boosted/attenuated by the value specified in **Gain below correction range**. Frequencies above the range will be boosted or attenuated by the value specified in **Gain above correction range**. The maximum value of both shelves is 20 dB and the minimum is -60 dB.

The high and low frequency cuts are useful for equalizing speakers with limited frequency range. They allow to remove frequencies from the audio stream that the speakers are anyway not able to reproduce.

The values in the **Low/High frequency cut** fields override other limits such as maximum boost/attenuation or target EQ curves.

TARGET EQ

When no target EQ curves are used, CONEQ™ Workshop creates a CONEQ™ equalization filter that achieves flat frequency response. Sometimes flat frequency response is not desired (relative HF loss over the distance, "club-sound", correction for hearing defects, etc.). In such cases, the **Target EQ** tab (see figure 21) allows specifying the desired frequency response of the system after the CONEQ™ equalization filter is applied.

One or more target EQ curves can be added together to produce the final target EQ that will be applied on top of the CONEQ™ equalization and made into a single equalization filter.

IMPORTANT! *Target EQ is not a fine-tuning tool but an advanced feature that can help in the more complex situations when it is impossible to measure in a way that a default equalization to flat frequency response can be applied or for specific cases when a non-flat frequency response is needed.*

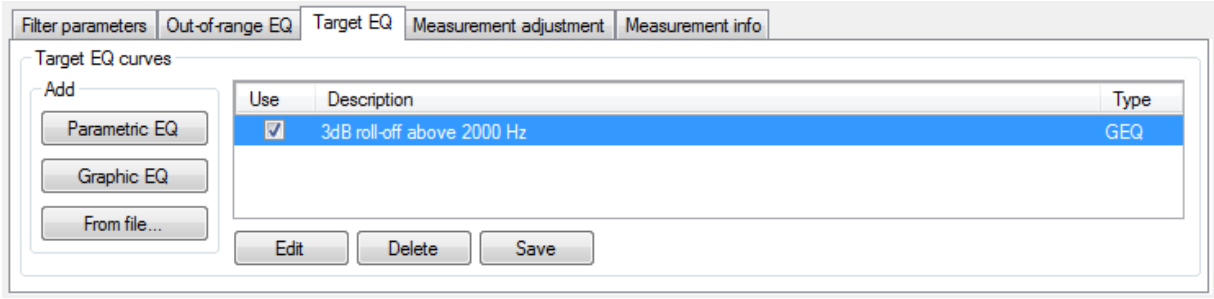

Figure 21: The **Target EQ** *tab.*

Clicking the **Parametric EQ** button adds a curve that can be specified using familiar controls of a parametric EQ. Clicking the **Graphic EQ** button adds a curve that runs through freely placed nodes on the frequency/amplitude plane. Clicking the **From File...** button adds either a previously saved Parametric or Graphic EQ curve, or a curve from a text file containing frequency-gain value pairs, or any `.DAT` or `.FIR` file (see table 13). The final target EQ curve will be the sum of all the target EQ curves from the list.

Selecting a curve in the target EQ curve list will display the curve in the Graph area. Clicking the **Edit** button or clicking directly on the target curve in the Graph area will enter editing mode (for graphic and parametric EQ curves only).

Clicking the **Delete** button will remove the currently selected target EQ curve from the list.

Clicking the **Save** button will save the currently selected target EQ curve (for graphic and parametric EQ curves only) to a file. This way a target EQ curve can be reused for other filters and/or projects.

Double-clicking on the name of the target EQ curve allows giving it a descriptive name.

Using Parametric EQ

Parametric EQ allows defining a target curve using the familiar concept of parametric equalizer. There can be up to twelve bands of equalization in one curve. If more bands are needed, more curves can be added. Each band can work in either Low Shelf, Low Pass, Band Pass, Hi Pass, or Hi Shelf mode. All modes have adjustable centre frequency. Low Shelf and High Shelf additionally have adjustable amplitude. The Band Pass filter additionally has an adjustable bandwidth (Q). Low Pass and High Pass filters have a fixed 24 dB/octave slope.

Selecting the curve in the target curve list will display the curve in the Graph area. Clicking the **Edit** button or directly on the target curve in the Graph area will enter editing mode. In this mode (see figure 22), a window with band parameters will open.

To use a band, switch it on, select a mode and adjust parameters. All parameters can be specified by writing a number in the corresponding field. Amplitude and bandwidth can be

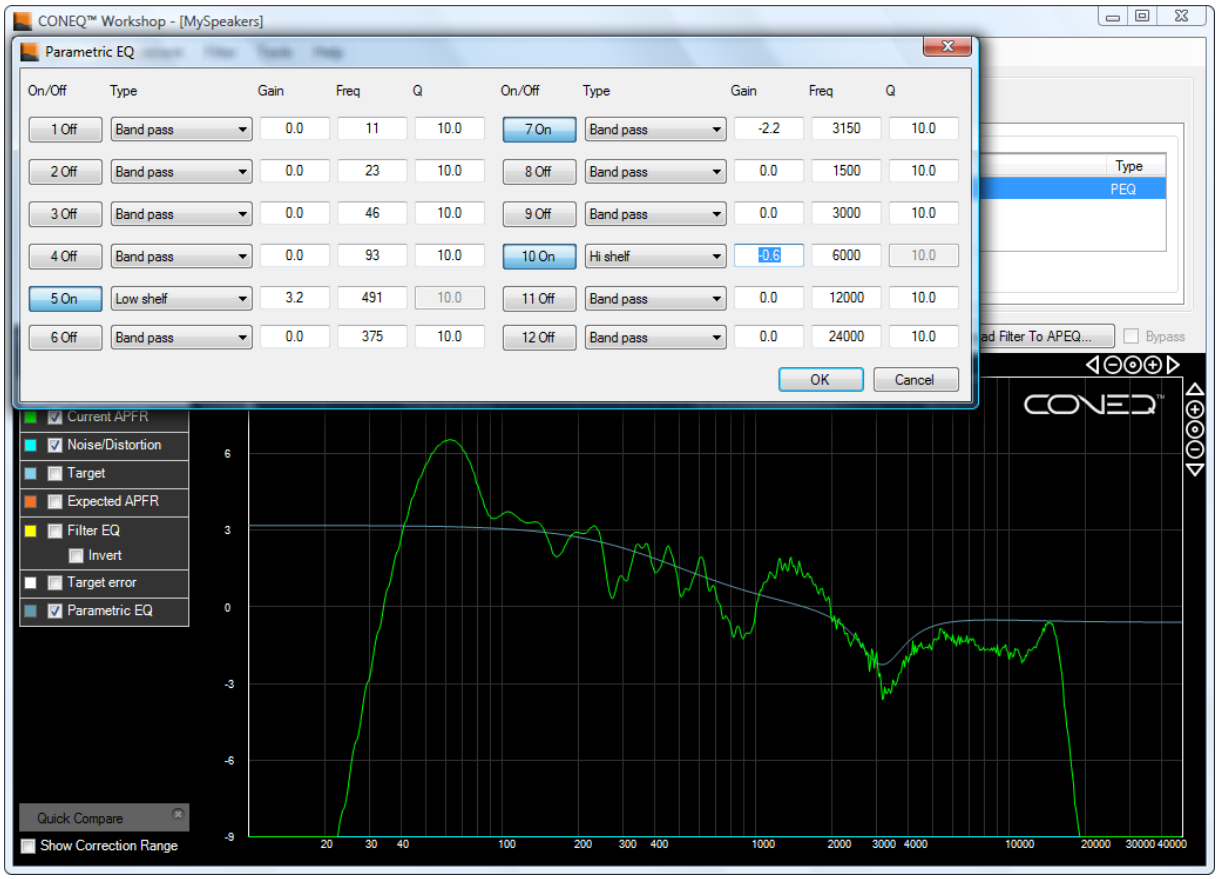

Figure 22: The parametric EQ in edit mode.

adjusted by pressing the mouse button while the cursor is on the corresponding field and moving mouse up or down. The frequency adjustment is the same except that the mouse is to be moved to the left or right.

Pressing the **Enter** key or clicking the **OK** button ends the editing mode and accepts the changes made. Pressing the **Esc** key or clicking the **Cancel** button will discard the changes.

Using Graphic EQ

Graphic EQ allows defining arbitrary target curve that runs through freely placed nodes on the frequency/amplitude plane. The EQ curve will be formed by connecting the nodes with straight lines.

Selecting the curve in the target curve list will display the curve in the Graph area. Clicking the **Edit** button or directly on the target curve in the Graph area will enter editing mode. In this mode (see figure 23), small rectangles will be displayed for the nodes that define the curve. Dragging a node will change its location. Dragging a node and dropping it outside the graph area will remove the node. Clicking on a location on the graph area where there are no nodes, will create a new node.

Pressing the **Enter** key ends the editing mode and accepts the changes made. Pressing **Esc** will discard the changes.

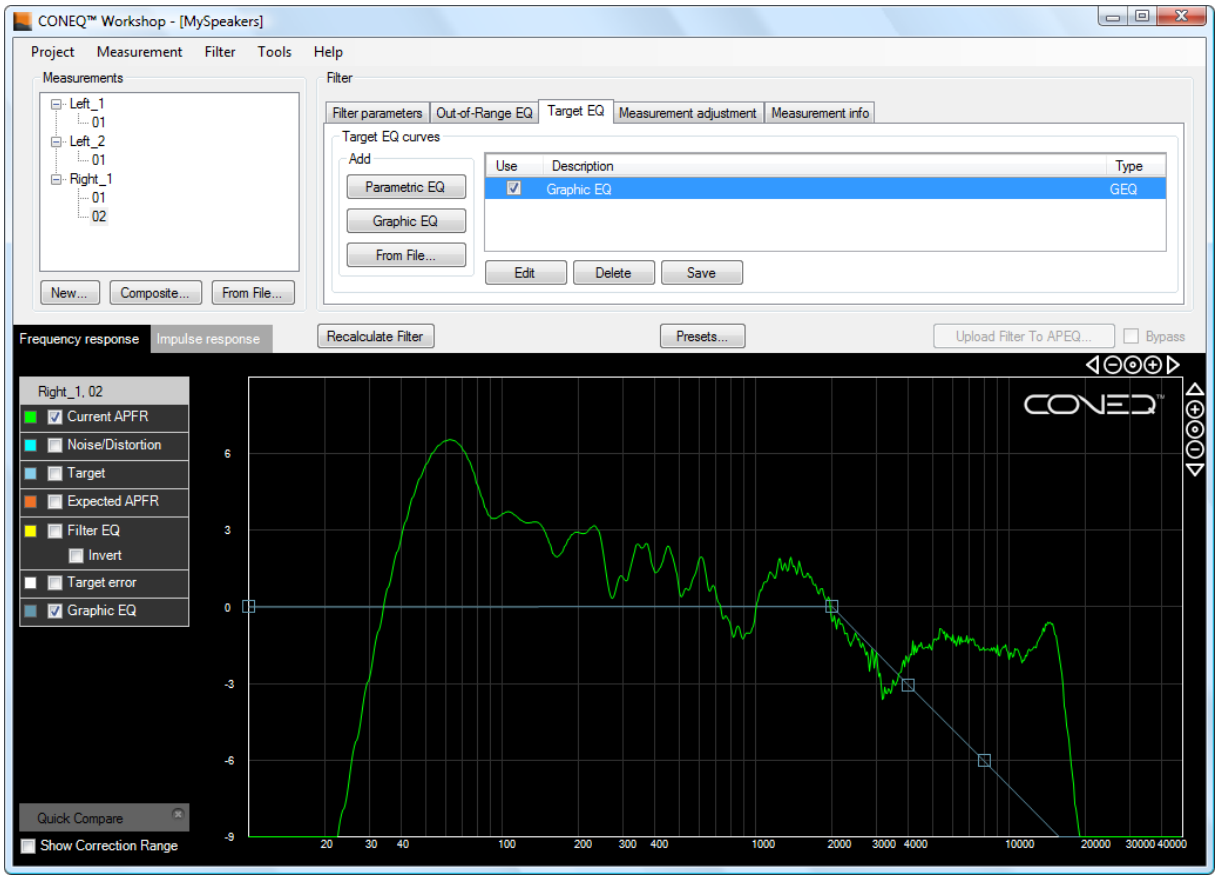

Figure 23: The graphic EQ in edit mode.

Using target EQ files

The target curve can be specified as a text file with one or more lines where each line contains frequency value in Hz, a **Tab** or **Space** character and the gain/attenuation value in dB. Lines starting with a semicolon (;) character are comments and can contain arbitrary text. An example of a target curve file is shown in the table 4

Any `.DAT` or `.FIR` file created by CONEQ™Workshop (see table 13) can also be used as a target EQ file. This feature can be used, for instance, to make one loudspeaker sound like another loudspeaker by specifying the measured APFR curve (`..._PRS.DAT`) of the other loudspeaker as the target EQ.

MEASUREMENT ADJUSTMENT

The **Measurement adjustment** tab (figure 24) allows adjusting parameters that determine how the APFR of the measurement looks like, thus, affecting the generation of the equalization filter. There are three ways how the measurement can be adjusted – adjusting the zero dB level of the APFR, applying a time window to the measured impulse response, and applying microphone calibration and audio interface frequency response correction curves.

Table 4: The format of a target EQ file.

```
; This is an example
; of a target EQ file
698.46   0
783.99   0
880      0
987.77   0.83
1108.73  1.67
1244.51  2.5
1396.91  3.33
1567.98  4.17
1760     5
1975.53  4.17
2217.46  3.33
2489.02  2.5
2793.83  1.67
3135.96  0.83
3520     0
3951.07  0
4434.92  0
```

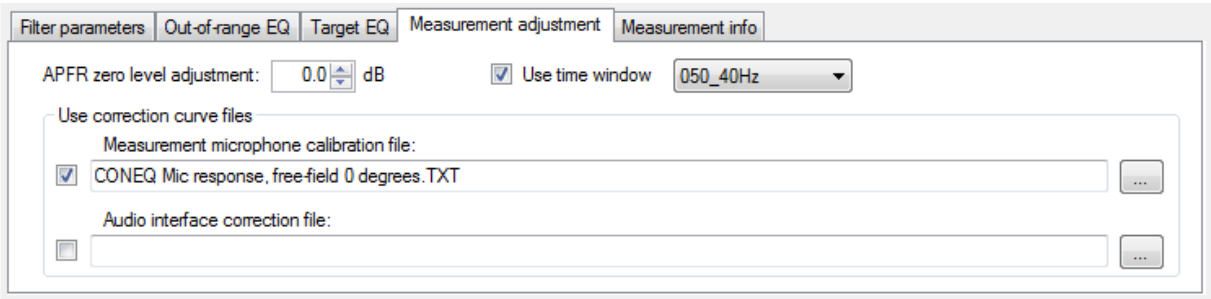

Figure 24: The **Measurement adjustment** *tab.*

APFR zero level adjustment

The APFR zero level can be adjusted if the default automatic adjustment is not optimal. One reason to do this is to align the APFR and the desired target curve so that the generated equalization filter fits better in the defined limits for maximum gain and attenuation. Another reason is when measuring speakers with very limited frequency range (high frequency drivers or subwoofers in particular) then CONEQ™ Workshop tends to place the APFR so that it is above the 0dB level across the whole correction range. This is not desired as the equalization filter would attenuate all frequencies in the correction range effectively reducing the sound volume.

The APFR zero level can be adjusted also in the Graph by clicking on the APFR curve and

dragging it up or down.

Time window

The time window determines what part of the measured impulse response is used to calculate the APFR. The default is a 50ms window (050_40Hz) that is suitable for most typical cases. Using a longer time window (100_20Hz or even 200_10Hz) is useful to measure low frequency drivers more precisely. For limited bandwidth speakers having negligible response in low frequencies (e.g. small ceiling speakers) use a shorter time window (025_80Hz, 015_130Hz, etc.). In general, the shorter the time window the less of the reflected sound is taken into account when calculating the APFR but also the less low frequencies can be measured.

Correction curve files

To compensate for the deviations from flat in the frequency response of the used microphone or audio interface, correction curve files can be used. Microphone calibration files are typically available for high quality measurement microphones. The correction curve file format is the same as for the target EQ files (see page 26).

There are other factors that can be compensated by using the correction curve files, for instance, uneven frequency response of an audio interface. To correct the frequency response of an audio interface, a loopback measurement should be done first. Then, the filter folder of the created filter must be browsed to find the frequency response file (`*._PR.DAT`, see page 13). This is the file that can be used subsequently as a correction file for other measurements with this audio interface.

MEASUREMENT INFO

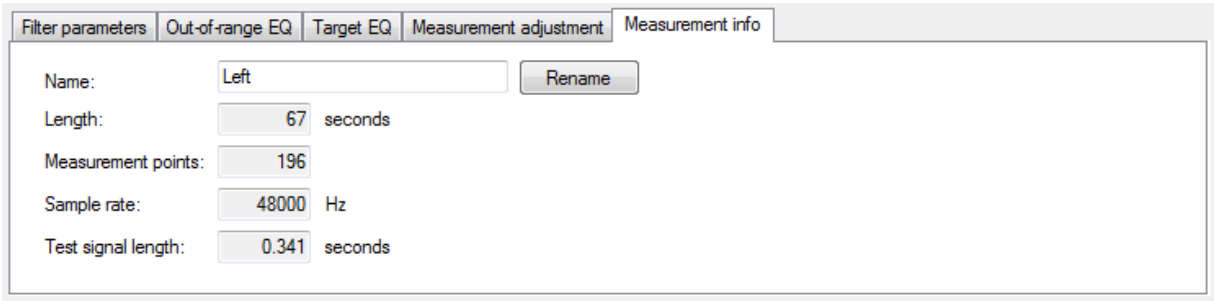

Figure 25: The measurement details window.

The figure 25 shows the **Measurement info** tab that displays the parameters of the current measurement. The table 5 lists all the fields and their meaning.

For composite measurements, only the **Name** field is meaningful and other fields will display zero.

Table 5: Fields on the **Measurement info** tab.

Field name	Field description
Name	The name of the measurement. The name is assigned before the recording of the measurement. It can be changed by editing it here and clicking the **Rename** button.
Length	The length of the measurement in seconds.
Measurement points	The number of recorded test signals that are used to calculate the acoustic power frequency response.
Sample rate	The sample rate used to record the measurement.
Test signal length	The test signal period in seconds.

APEQ™ HARDWARE CONTROL PANEL

On the lower right of the Parameters area there is a small set of controls (see figure 26) for uploading filters directly to the APEQ™ hardware equaliser, if one is connected.

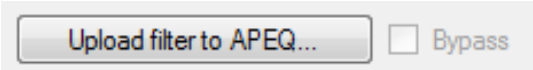

Figure 26: The APEQ™ hardware control panel.

When an APEQ™ device is connected to the computer, and a compatible filter is selected in the measurement list, the controls get enabled. Clicking the **Upload filter to APEQ...** button will open a menu as in figure 27. Selecting a channel will upload the current filter to the selected channel of the hardware. Selecting ***Upload to all channels*** will upload the filter to all channels of the hardware. If more than one APEQ™ device is connected to the computer then the ***Select APEQ device*** menu item allows selecting the one which will be used for filter upload. For more sophisticated filter management it is possible to launch the **APEQ™ Communication Tool (C2)**.

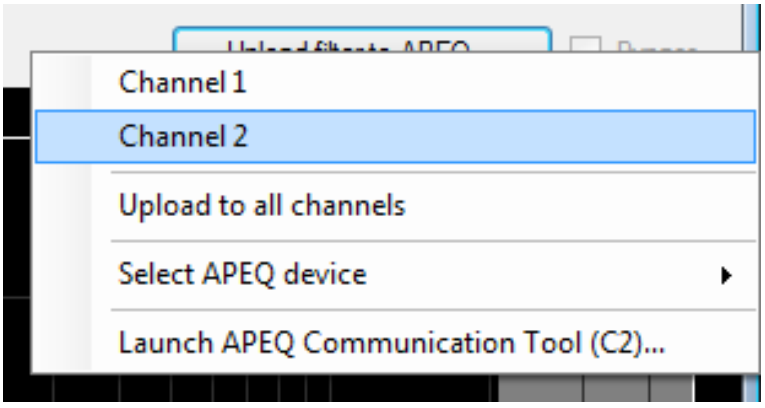

Figure 27: The channel selection menu of the APEQ™ hardware control panel.

Only filters with sample rate of 48 kHz can be uploaded to the APEQ-2pro/2pro DIO

hardware units. The APEQ-8pro DIO hardware units support filters with 44.1, 48, 88.2, and 96 kHz sample rates.

The **Bypass** check-box is enabled for the APEQ-8pro DIO devices and allows switching CONEQ™ filtering temporarily off. This way, a quick comparison can be made between filtered and non-filtered sound.

FILTER MENU

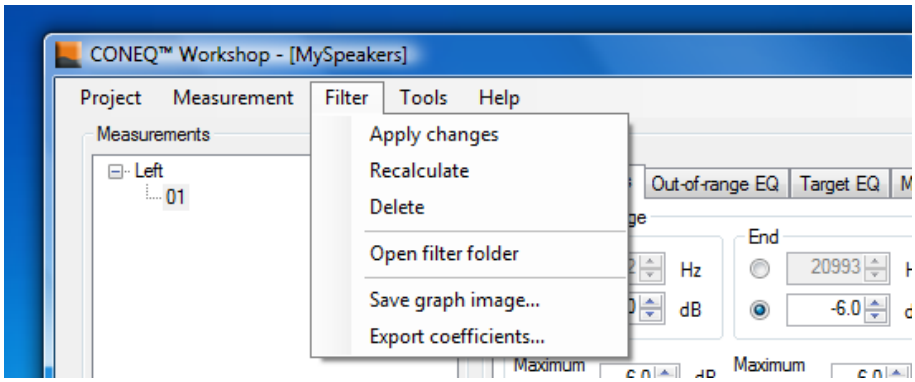

*Figure 28: The **Filter** menu.*

Recalculate

Creates a new filter with the current set of parameters. The current filter will not be affected. This is the same functionality as pressing the **Recalculate filter** button on the parameters area.

Apply changes

Applies the current set of parameters to the current filter.

> ***IMPORTANT!*** *Any changes to filter parameters have to be confirmed by either applying changes to the current filter or recalculating a new filter. Unconfirmed changes to any parameter (levels, limits, EQ curves, filter description, etc.) will be lost when another filter is selected or the project is closed.*

Delete

Deletes the current filter.

Open filter folder

Opens the folder of the filter by means of the operating system. This is useful for accessing the filter file and various curve files that were described on page 13.

Save graph image...

Save the currently displayed graph, including a legend as a PNG image.

Export coefficients...

Export the CONEQ™ filter as a `.csv` file containing coefficients of a finite impulse response (FIR) filter for use in CONEQ™ COMPATIBLE third-party products. A special license is required to be able to export filter coefficients for CONEQ™ COMPATIBLE products. The CONEQ™ COMPATIBLE program is described in greater detail in the **CONEQ™ COMPATIBLE Program User's Guide** document.

GRAPH

The Graph is the area in the lower part of the main window that visualizes data either in frequency domain (APFR, filter EQ, etc.) or time domain (impulse response, time window).

The current Graph image with a legend can be saved by choosing ***Filter⇒Save graph image...*** from the menu or by right-clicking on the graph area and choosing ***Save graph image...*** from the context menu.

The Graph supports zooming (see page 35) and display of data from several measurements and filters (see page 34)

It is posible to change the colours of the curves by clicking on their colour patches in the legend or by adjusting program options (see page 38)

FREQUENCY DOMAIN VIEW

When the frequency domain view is selected (figure 29) the graph shows frequency domain data (such as APFR, filter EQ, etc.) with Hz on horizontal axis and dB on vertical axis.

The frequency domain view is selected by clicking on the **Frequency response** tab at the top left corner of the Graph.

The following frequency domain data of the currently selected filters can be visualized:

- Acoustic Power Frequency Response (APFR)
- Noise/Distortion
- Total Target EQ
- Expected APFR after equalization
- Filter EQ
- Deviation of filter EQ from desired equalization
- Each individual Target EQ
- Microphone and audio interface correction curves

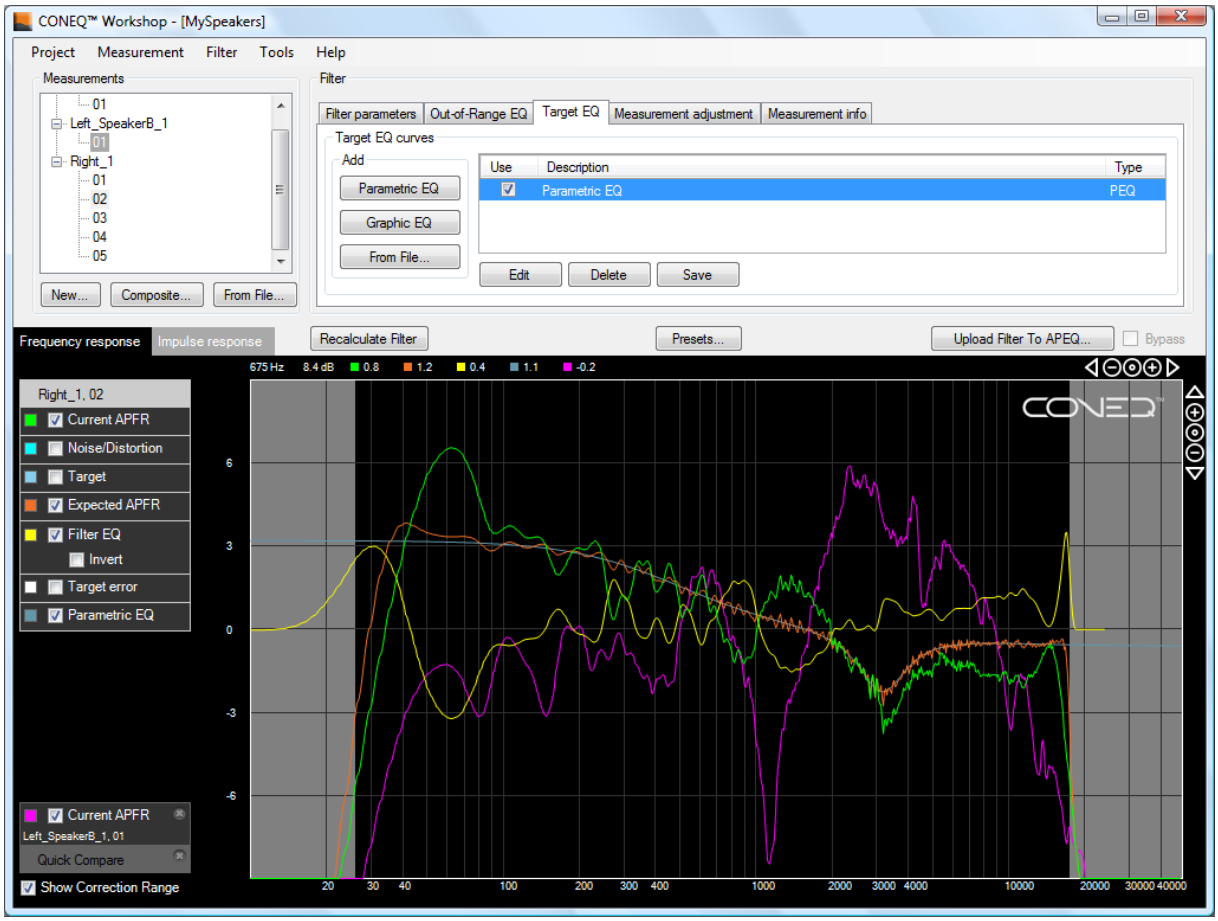

Figure 29: The frequency domain view of the Graph.

To show a curve, check the box in front of the corresponding curve name on the legend at the lop left part of the Graph area.

Additionally to that, frequency domain curves from other measurements/filters and from external files can be shown (see page 34).

Current APFR

The current APFR curve (green) shows the acoustic power frequency response of the currently selected measurement after APFR zero level adjustment, time window, and correction curves had been applied. By clicking on the APFR curve of the current filter and dragging it up or down, it is possible to adjust the APFR zero level. The changes, as usual, must be confirmed by either creating a new filter or applying the changes to the current filter.

Noise/Distortion

The noise/distortion curve (cyan) can be used to determine how much noise was present during the measurement. It also shows the effect of the non-linear distortions. If the noise/distortion curve is less than 10 dB below the APFR within the correction range then

the measurement and the resulting equalization filters are not reliable.

Technically, noise/distortion curve is calculated from the end of the impulse response in the same way as the APFR is calculated from the beginning.

Target

The target curve (blue) shows the desired frequency response of the speakers after applying the equalization filter. It shows the sum of individual target curves defined on the **Target EQ** tab. See page 23 for more information on editing the individual target EQ curves. When no target EQ curves are defined, the total target EQ curve is a line at 0dB, meaning that the resulting equalization will aim to produce a flat frequency response.

Expected APFR

The expected APFR shows a good approximation of what the APFR will look like when actually measuring the speakers after the corresponding CONEQ™ equalization filter is applied. Ideally, this curve should match the target curve exactly. There are several reasons why there can be deviations:

- Maximum gain/attenuation settings
- Correction range settings
- Shelf and cut-off settings
- Difference of the measured APFR and smoothed APFR
- Difference of the generated filter and the smoothed APFR due to insufficient filter resolution (number of taps)

Filter EQ

The filter EQ curve (yellow) shows what the CONEQ™ equalization filter looks like as an equalizer curve. The **Invert** check-box can be used to invert the filter EQ curve around the 0dB level. This view allows seeing how closely the equalization filter matches the measured APFR.

Target error

The target error curve (white) shows how much the generated filter differs from the smoothed APFR used to generate it. Use this curve to evaluate the quality of the filter – the more it looks like a line at 0dB the better. The bigger the filter resolution (the number of taps) the more flat is the target error curve.

Individual target EQ curves

When an individual target curve is selected on the **Target EQ** tab, it is also displayed on the Graph. For parametric EQ and graphic EQ curves, clicking on the displayed curve will enter the edit mode. See page 23 for more information on editing the individual target EQ curves.

When the current filter selection changes, the individual target curve is removed from the Graph.

Correction curves

If the microphone calibration curve or audio interface correction curve is specified on the **Measurement adjustment** tab (see page 27) then the corresponding curve can be displayed on the Graph.

Range markers

This check-box triggers greying out of the frequencies below and above the correction range as defined on the **Filter parameters** tab (see page 21).

TIME DOMAIN VIEW

When the time domain view is selected (figure 30) the graph shows time domain data – the measured impulse response of the currently selected measurement and the time windows used for the currently selected filter. The horizontal axis shows milliseconds and the vertical axis shows dB.

The time domain view is selected by clicking on the **Impulse response** tab at the top left corner of the Graph.

QUICK COMPARE FEATURE

There are cases when several measurements need to be compared (e.g. left and right channels, frequency response before and after CONEQ™ equalization, two different speakers, etc.). For such cases there is a possibility to add up to six curves to the **Quick compare** list. The list is shown on the bottom left part of the Graph area. Each entry in the list has a unique colour. Each **Quick compare** curve can be adjusted vertically by dragging it up or down on the Graph area. The **Quick compare** list is saved for each project.

Adding curves to the Quick compare list

There are two ways how to add curves to the **Quick compare** list. The simplest is to press the **Ctrl** key and click on a filter in the measurement list. This marks the corresponding filter as used for comparison and adds a curve from that filter to the **Quick compare** list. **Ctrl**-clicking on the marked filter again removes all curves of that filter from the **Quick compare** list and unmarks the filter.

The second way is to right-click on the **Quick compare** header area (the gray rectangle with the words **Quick compare** in it) and selecting the desired curve from the menu that

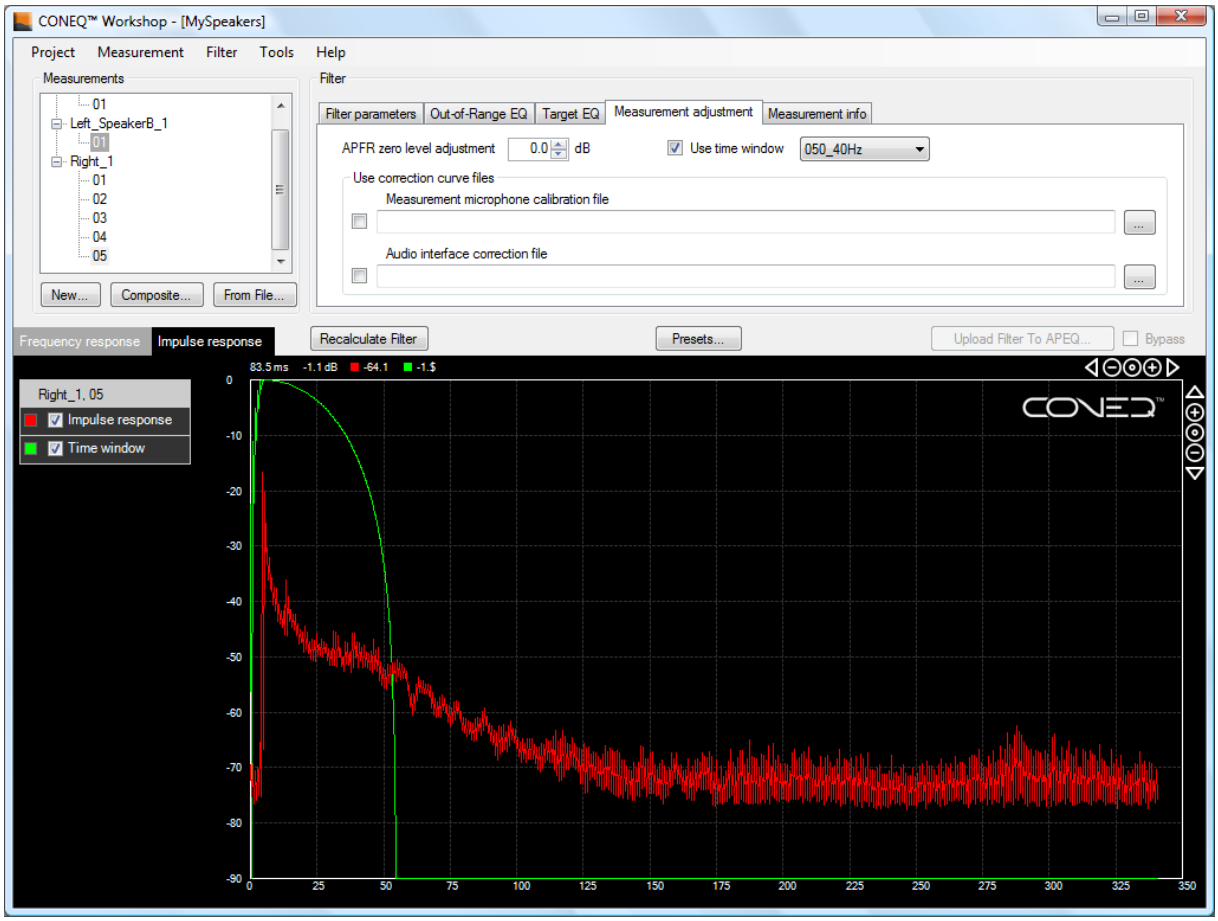

Figure 30: The time domain view of the Graph.

opens. This way must be used to add curves from external files (such as curves from other projects or curves generated by other applications).

Visually, marked filters are similar to the selected filter but have a dark gray background. On Mac, marked filters have an icon of an eye shown on the right side.

The figure 31 shows the **Quick compare** list with two curves added. One of the curves is from another filter in this project but the other curve is loaded from an external file (a microphone calibration file in this case).

ZOOM

The Graph supports changing the vertical and horizontal scale by either clicking on appro-priate zoom control buttons (see figure 32) or by operating the mouse wheel.

Using mouse, the vertical scale of the graph can be changed by positioning the mouse cursor over the Graph and turning the scroll wheel on the mouse. The scale is changed in 6dB increments. The horizontal scale can be changed by holding the **Ctrl** key (**Shift** key on the Mac) and turning the scroll wheel on the mouse. The horizontal zoom takes place around the horizontal position (frequency or time) on which the mouse pointer is located. The vertical zoom on time domain data acts similarly but on frequency domain data is centred

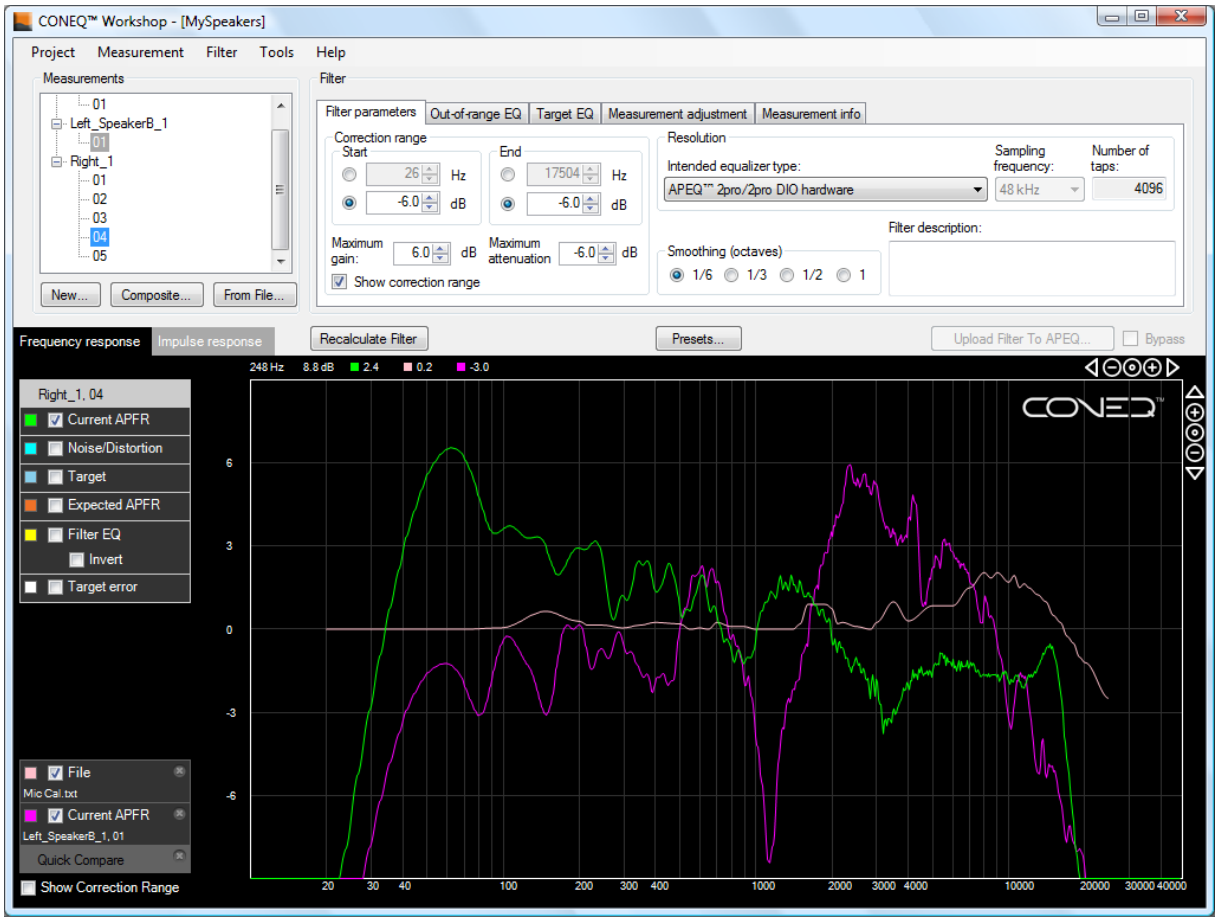

Figure 31: Two curves added to the **Quick compare** *list*

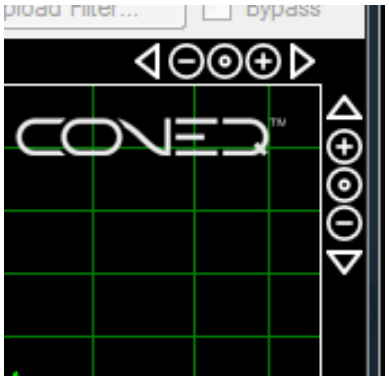

Figure 32: The zoom controls.

around the middle line (typically, 0dB) in the graph.

It is possible to move the visible range in any direction by using arrow buttons on the zoom controls.

PRESETS

Presets in CONEQ™ Workshop are named sets of parameters that can be used when creating or importing measurements to set many measurement and filter parameters at once. Presets can be edited using the preset editor (see figure 33) which can be invoked by selecting the ***Tools⇒Presets...*** menu or by clicking the **Presets...** button on the main window and selecting ***Preset editor...*** from the pop-up menu.

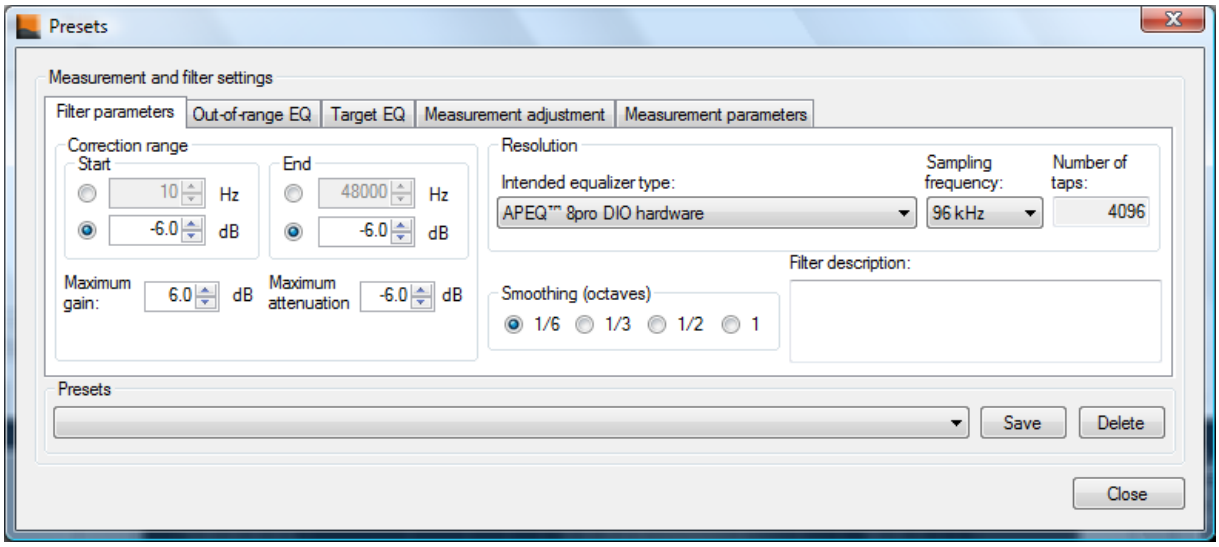

Figure 33: The preset editor.

There are two special presets named **<Default for new projects>** and **<Default for this project>**. The **<Default for this project>** is a set of parameters that will be used by default for all new measurements made in this project. The **<Default for new projects>** will become the **<Default for this project>** for every new project. These two presets can be modified the same way as other presets. The **<Default for this project>** preset is saved for each project while all the other presets are saved globally and are the same for all projects.

CREATING A NEW PRESET

To create a new preset, open the preset editor and edit the parameters. Then click the **Save** button. In the dialogue that opens, enter the name of the new preset. If an existing preset is selected from the list then it will be overwritten with the new values.

To create a preset that uses the settings of the current filter, simply open the preset editor (it will open initialized with settings from the filter parameters area) and save the preset.

APPLYING A PRESET TO CURRENT FILTER

To apply an existing preset to the current filter, click on the **Presets...** button on the main window and select the desired preset from the pop-up menu. Then click the **Recalculate filter** button (or choose ***Filter⇒Apply changes*** from the menu).

DELETING A PRESET

Press the **Delete** button in the preset editor. In the dialogue that opens select one or more presets to delete.

OPTIONS

There are a few settings of the CONEQ™ Workshop application itself that can be adjusted by opening the options window (figure 34). The options window can be opened by choosing ***Tools⇒Options*** from the menu.

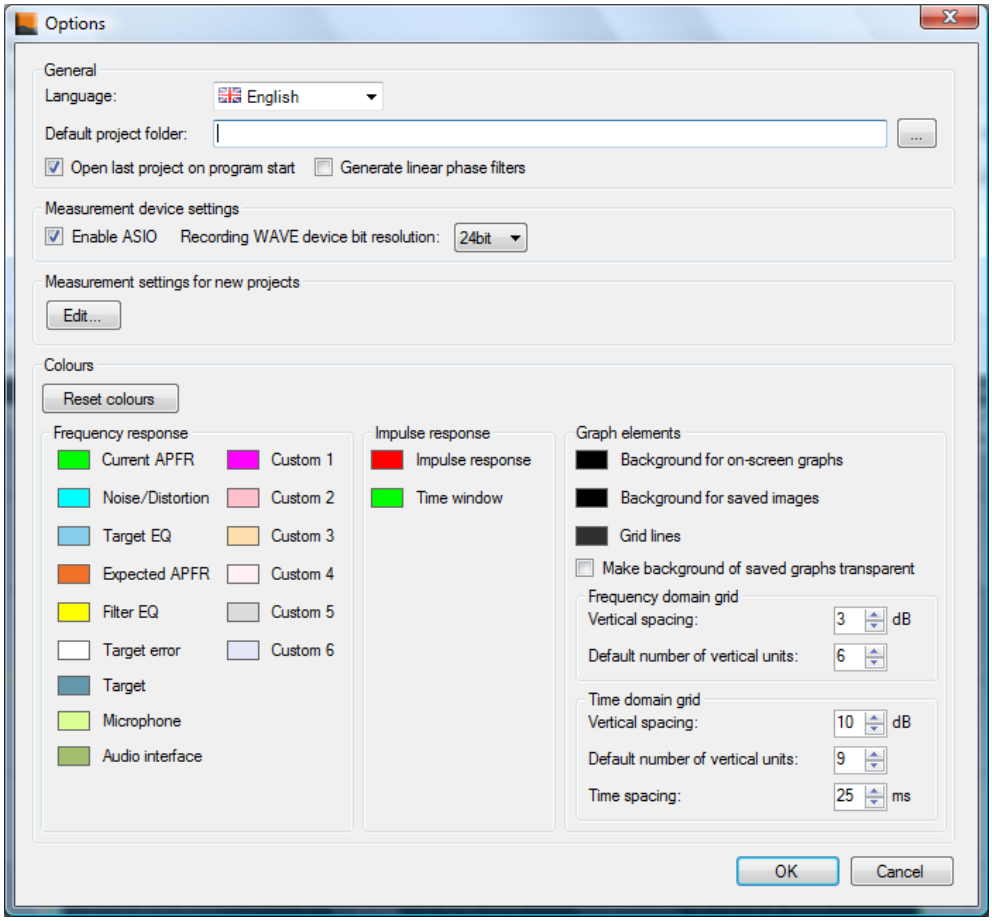

Figure 34: The options window.

The table 6 lists the available options and describes their effect.

TECHNICAL SUPPORT

Please feel free to contact us at any time about any difficulties using CONEQ™ Workshop software (our contact details are listed below). We will do our best to respond to you as quickly as possible. Our mission is to make your experience with CONEQ™ as simple and

Table 6: The CONEQ™ Workshop options.

Option	Description
Language	The language of the user interface. Currently supported are English (default) and Chinese. Changing the language requires a restart of the CONEQ™ Workshop application.
Default project folder	Use the location by default on opening and creating new projects.
Open last project on program start	Open the last used project when CONEQ™ Workshop starts.
Generate linear phase filters	Besides the standard minimum phase filter files, create also the linear phase version for each filter.
Enable ASIO	Enable using ASIO devices for taking measurements.
Recording WAVE device bit resolution	Set bit resolution for WAVE devices to 16 bits to support some audio interfaces that have problems recording in 24 bits using WAVE drivers (e.g. Terratec Phase 22).
Measurement settings for new projects	Click the **Edit...** button to open preset editor and edit the default measurement and filter settings for all new projects.
Colours	Specify colours for the various graphs.
Make background of the saved images transparent	Allows putting saved images on some custom backgrounds for maximum flexibility. Note: If the saved image is to be put on light background, select the background for saved images to be light as well. Even though the specified background colour will not be present in the saved image, the setting will influence the colours of the axis labels and colour of the legend texts.
Frequency domain grid	Specify the default vertical scale and number of grid lines for the frequency response graphs.
Time domain grid	Specify the default vertical scale as well as the number of vertical and horizontal grid lines for the impulse response graphs.
Reset colours button	Revert to the default colours.
OK button	Accept the changes and closes the options windows.
Cancel button	Close the options window without saving changes.

rewarding as possible. To help us resolve issues quickly, we might need one or more of the following data:

- CONEQ™ Workshop software edition and version (It can be found in ***Help⇒About***).
- The operating system: Windows XP, Windows Vista or Windows 7, U.S. or international version, OEM or stand-alone software package.
- Computer information: CPU type and speed, installed memory.

- Description of the problem (as much information for us to understand the problem as possible).

CONTACTING REAL SOUND LAB

Should you have questions or comments, please contact us using one of the following ways:

- E-mail: support@realsoundlab.com
- WWW: www.realsoundlab.com
- Phone: +371 6788 9828 (Real Sound Lab SIA Headquarters, Riga, Latvia)

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