

A Quick Introduction to Acourate

0. Preparation: Measurements with AcourateLSR

It is assumed that already some measurements have been carried out by AcourateLSR (of course also other measurement programs are possible). AcourateLSR is saving the pulse responses in a subfolder \Pulses in the protected .tri format.

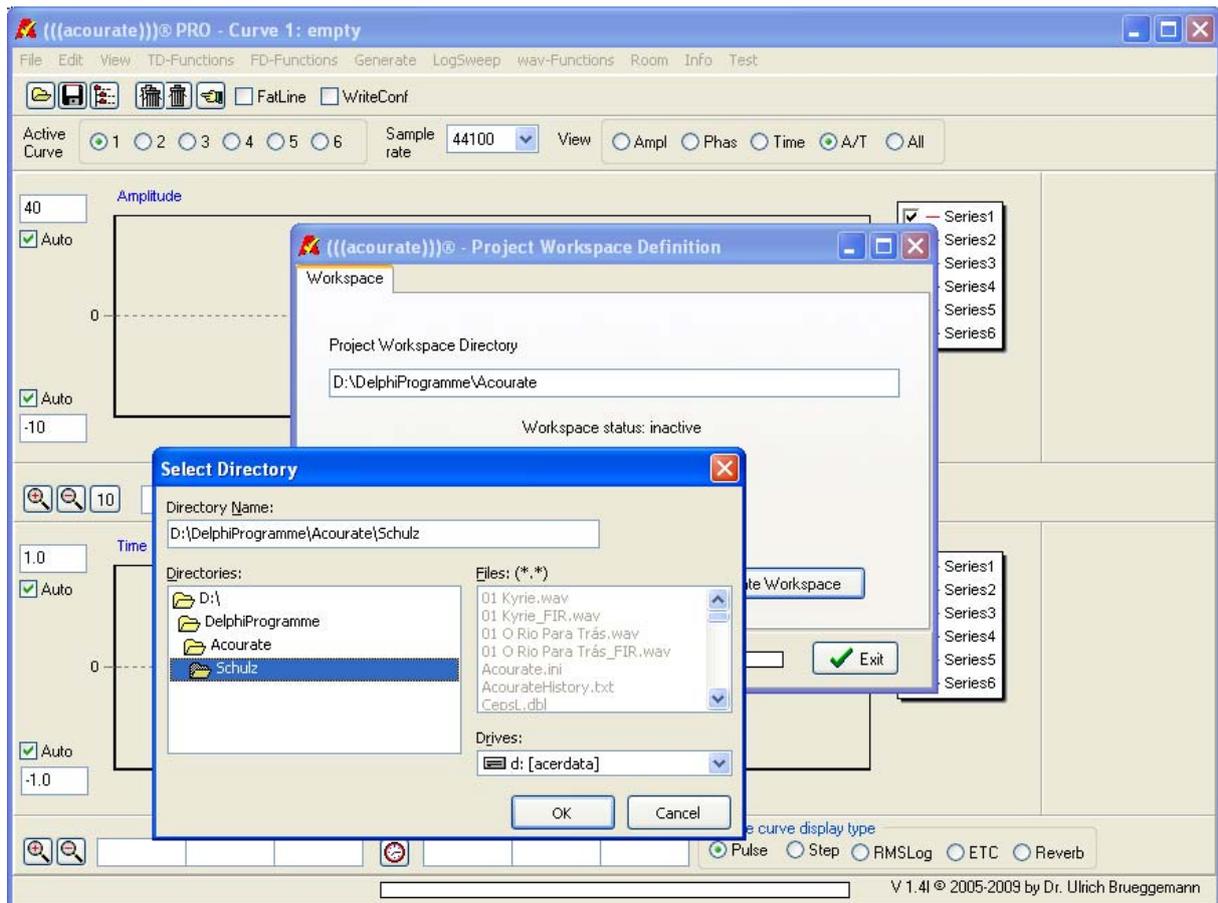
1. Definition of a workspace directory

For example create a subdirectory \quick and copy the pulses Pulse48L.tri and PulseR48.tri into this folder.

2. Start Acourate

Select the workspace by the menu
File – Project Workspace Definition

Select the project folder defined in 1. by a double click and confirm



The folder will be displayed with blue fonts in the upper panel of the screen.

3. Select Samplerate

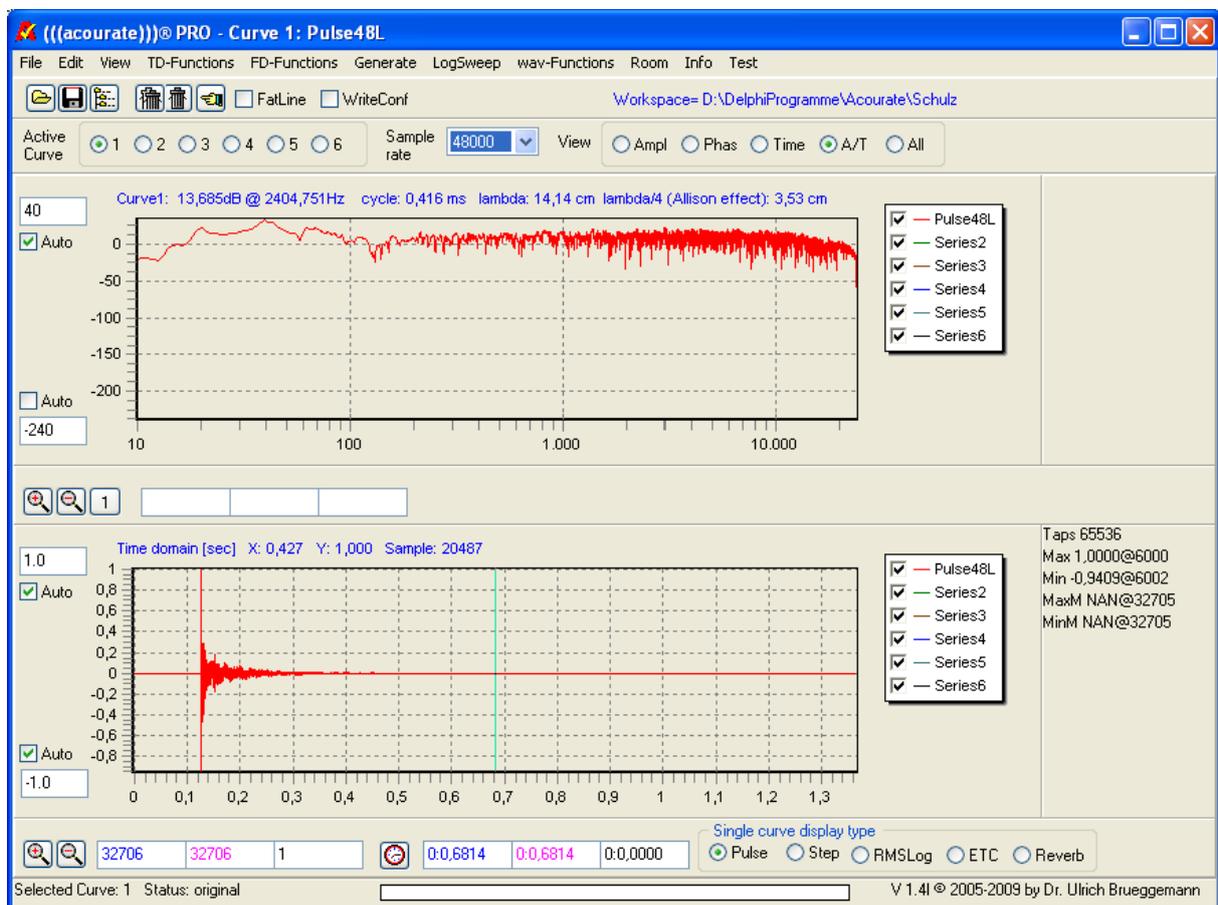
It is important to select the correct samplerate. This is the samplerate selected during the measurement. In case of AcourateLSR pulses the filename helps to get the correct rate (Pulse44L = 44.1 kHz, Pulse48L = 48 kHz)



4. Open the measurement pulses

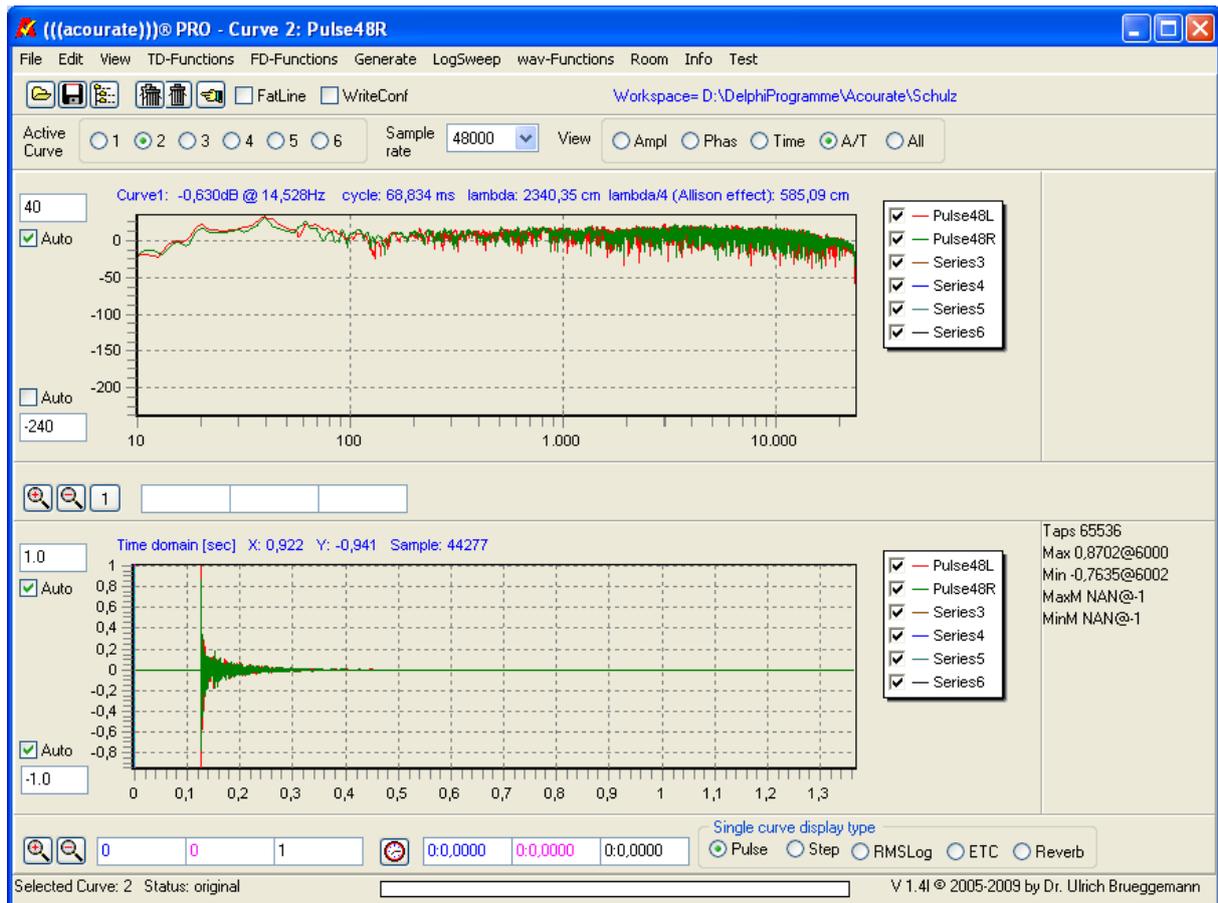
Before opening of a file simply select the active curve by the radio buttons where the data have to be loaded into (default is curve 1)

Then open the left pulse Pulse48L.tri



Please save immediately the pulse. This way the licensed Acourate version will save the file in the standard format as Pulse48L.dbl. Acourate uses the double floating point format by default.

Then load the right side pulse. Select curve 2 before and then open the file. Save the file again.



The picture now shows the pulse responses as well as the according unsmoothed frequency responses.

Before starting further calculations it makes sense to play with the scaling of the diagrams. A left and right mouse click will show up some markers and a selected area and it is possible to zoom into the display with the zoom buttons.

It is also possible to switch the type of display with the radio buttons in the upper screen panel:

- amplitude
- phase
- time
- amplitude and time (default)
- amplitude, phase and time

In the time domain chart the display for the active curve can be switched between:

- pulse display (default)
- step response
- logarithmic display of mean square value
- energy-time-curve
- reverberation according to Schroeder

Please remember in general:

It is necessary always to select a curve as active curve before any further computation. You can do this with the Active Curve radio buttons 1 to 6

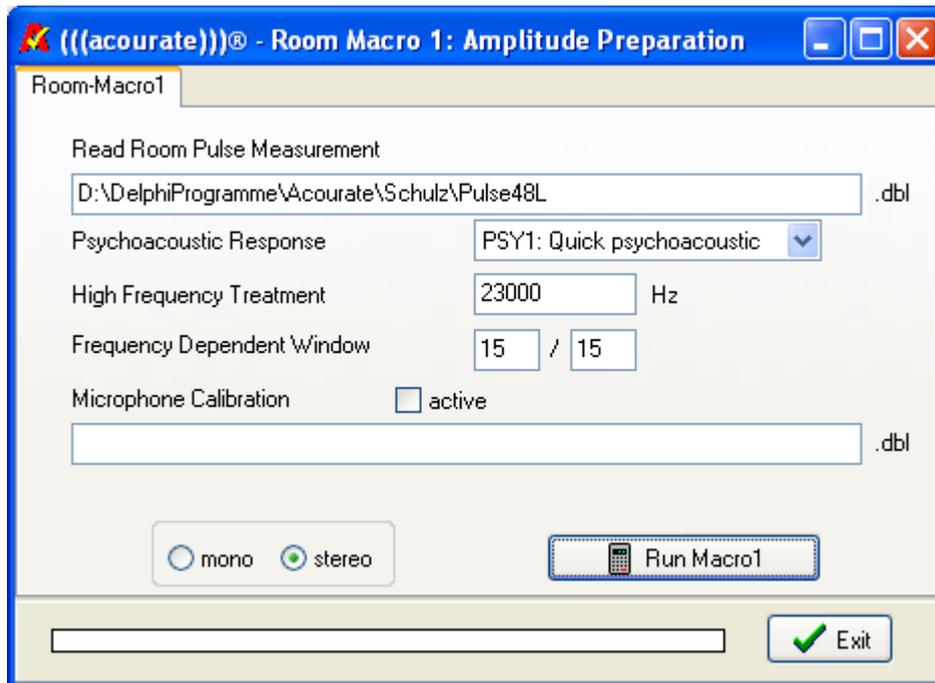
5. Room correction macro 1

Acourate in principle is like a scientific pocket calculator with many functions. For a room correction there are some macros defined which simplify the operations. Different single sub-functions are combined in a macro.

So first run menu

Room – Room Macro 1 – Amplitude Preparation

Select the left side pulse response in the edit field (click at the edit field) if not already shown. (Remark: Acourate will memorize the selections for later re-calculations)



Psychoacoustic Response: Acourate calculates a frequency response which takes care about the transient behaviour of music signals (music is not a steady-state signal) in combination with the given pulse responses. PSY1 quick is a quicker function but also gives a good accuracy already. PSY1 long may take much more computation time.

High frequency behaviour: in the higher frequency range the curve mostly has a steep fallign slope. This is quite normal (it depends on speaker, room, microphone and frequencies close to samplerate/2). With the given example frequencies above 23 kHz will be more smoothed.

FDW – frequency dependen windowing: for a correction only a part of the pulse response can be used, a window has to be calculated. The width of the window is dependent on the frequency and defined as number of cycles. A value of 15 e.g. means a window width of 15 ms @ 1 kHz frequency. There is a parameter for the lowest frequency and another one for the highest frequency. In between an interpolation takes place.

In our given example 15 cycles constant over the full frequency range are selected.

Smaller values will result in more smoothed frequency responses, larger value will follow the frequenc response with more accuracy and details but als a more dependant position of the listening place.

Microphone calibration: if a calibration file is available it can be also taken into account

After starting the macro the calculation will be carried out, the results will be shown in curve 1 and 2. Remark: overwriting or deleting of a curve in the display does not destroy saved files, here e.g. the files Pulse48L.dbf and Pulse48R.dbf will still exist.

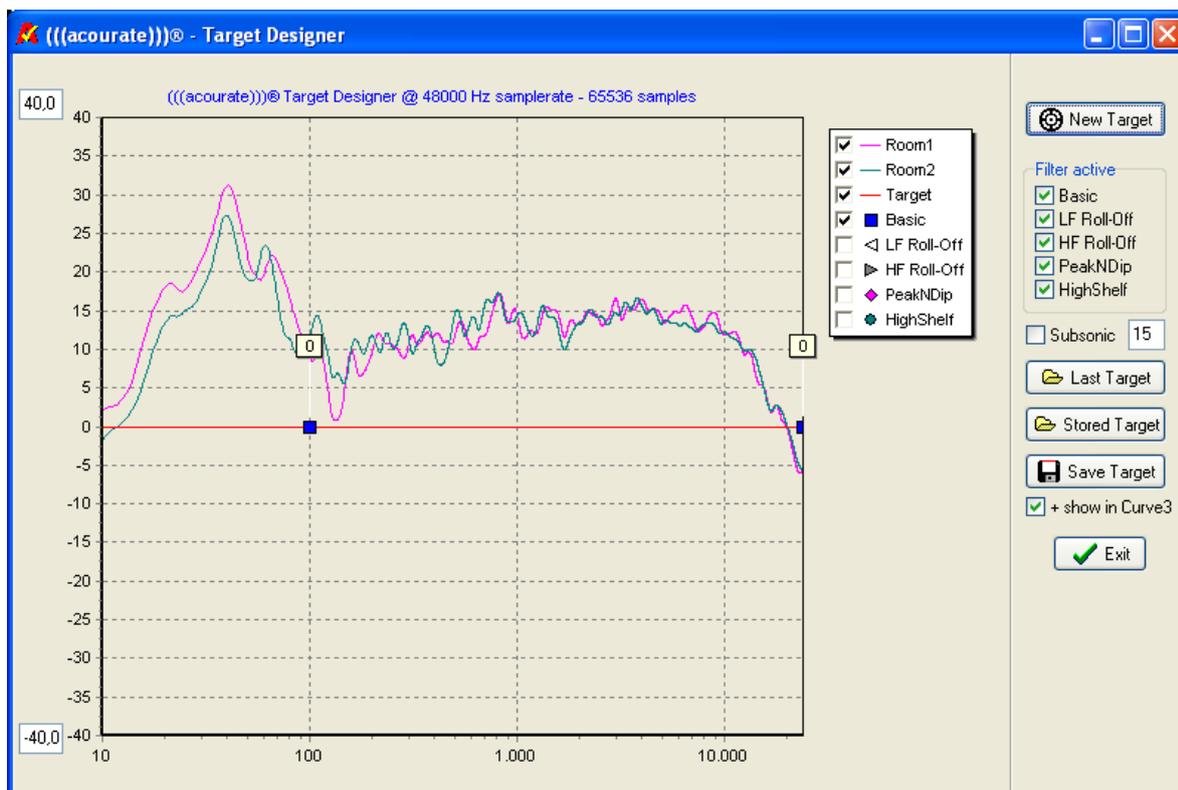
The results will be saved as Pulse48Lmp.dbf and Pulse48Rmp.dbf in the project workspace automatically. Existing files will be overwritten without any query.



6. Target curve design with macro 2

Menu

Room – Room macro 2 – Target curve designer



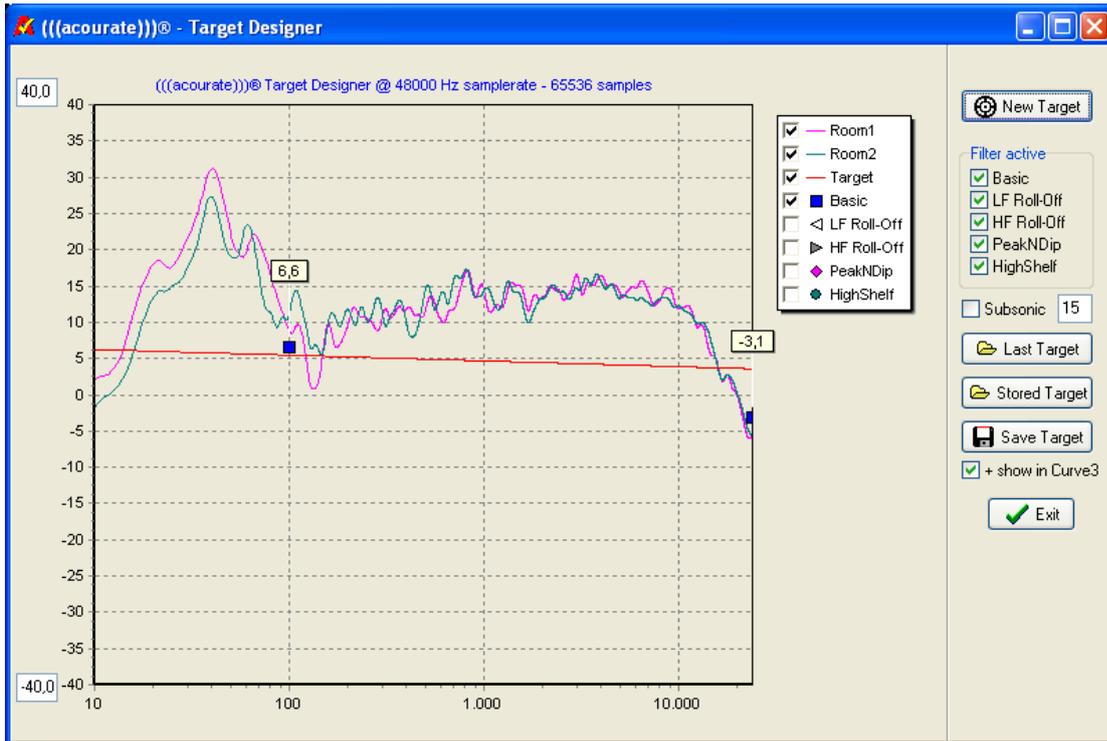
New Target starts with a horizontal flat target curve.

The blue point markers can be dragged with the mouse

a) Curve slope. It is recommended to start with a slope of about -3 dB. Drag the right point down.

b) Move target curve up and down. Drag the curve with the left point marker.

The target curve shall be placed BELOW the measurement curves but as close as possible. All curve parts above the target will be corrected, all parts below the target are not taken into the correction calculation. It is allowed that the target may cross some single dips. Thus dips are not boosted in the correction and the overall correction get a better resolution and less attenuation.



The example shows that around 125 Hz the dip is crossed by the target. Otherwise it would be necessary to lower the target by -6 dB. Thus there would be more attenuation by the correction filter just because of this single dip.



Further on the subsonic filter is switched on @ 11 Hz. This allows to efficiently cancel low frequencies like rumbling.



In the next step the HF Roll-Off function has been selected in the chart legend. (a new small triangle point marker appears). Drag the marker to get a parallel target curve compared to the measurement curves. This will result in a flat correction at high frequencies.



With the HighShelf function you can easily create a lifted target curve in the low frequency area.

There are other functions in the chart legend which can be used to modify the target curve. But for the first steps the example already gives a good start point.

Save the target (default name: Target.dbl). Exit will switch back to the Acourate window, the target is shown as curve 3.

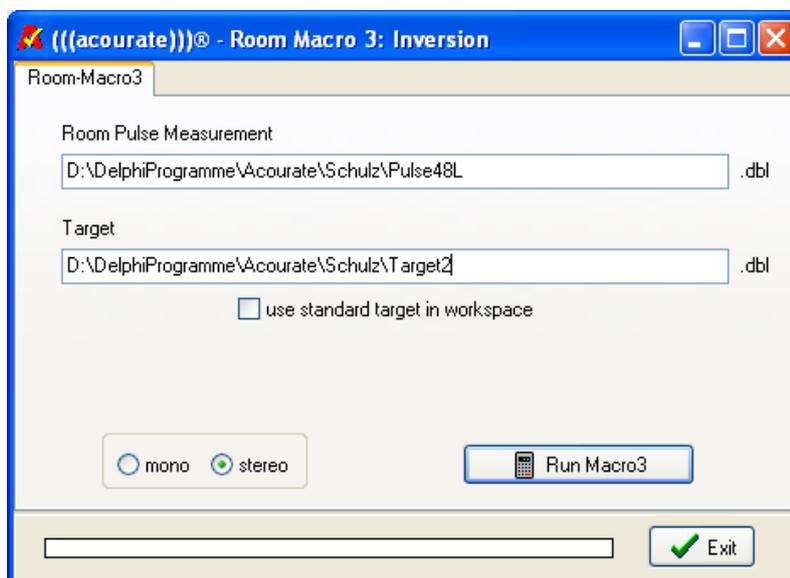


Remark: the curve display in the target designer has less resolution compared to the Acourate target display. This is intended to allow a quick manipulation of the target curve in the designer.

Now a target curve has been created..

7. Macro 3 – Target curve minus Measurement curve = Correction

The next step is menu Room – Room macro 3: Inversion



Therefore click at the edit fields and select the desired pulse responses. Acourate will make a proposal automatically based on the previous macros.

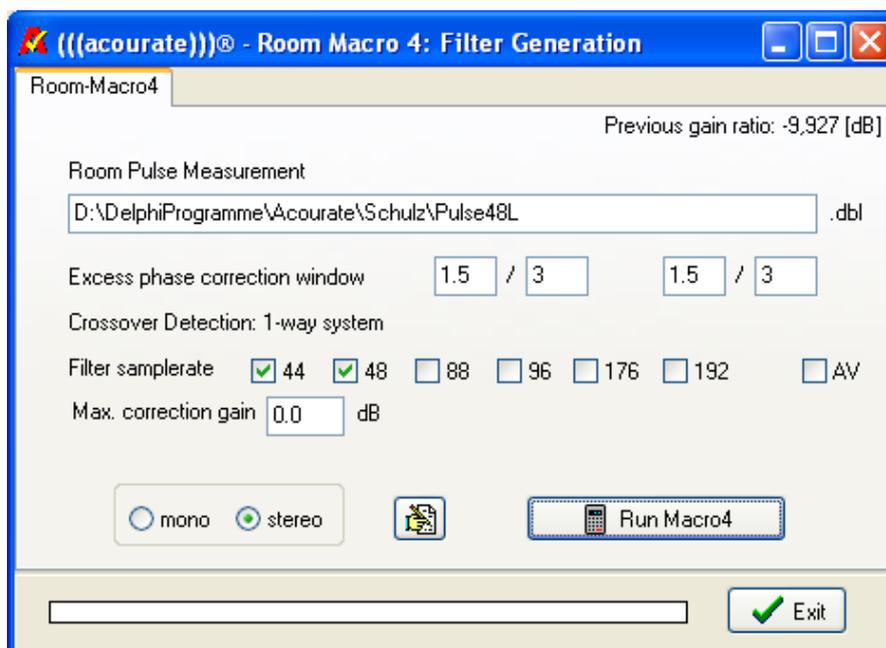


The computed corrections will be displayed as curve 4 and 5. They are saved as Pulse48Linv.dbl und Pulse48Rinv.dbl.

8. Macro 4 – Computation of the final Correction Filters incl. Excessphase Correction

Menü

Room- Room macro 4: Filter Generation:



Excessphase correction: similar to the frequency dependent windowing parameters have to be defined for the excessphase correction. Start values are 1.5/3. For left and right channels you can define different values, but they should not be too different. A bigger value will result in a bigger phase correction. But it is necessary to watch out for instable results with big values, see 9. Test Convolution

In the most simple case there are no crossovers defined and Acourate will detect a 1-way-system.

Filter samplerate: it is possible to define filters with several samplerates. Start point in our example is the samplerate 48 kHz. As e.g. the filter shall be used with wav-files also 44.1 kHz are checked.

Max. Gain: by default the correction will be normalized in a way that no frequency will be boosted above 0 dB. It is still possible to modify the gain but clipping can happen in case of a higher gain. Please note the dependency on the maximal gain in the wav-signal.

After start of macro 4 the correction filters will be computed and saved under the filenames Cor1L44.dbl, Cor1R44.dbl, Cor1L48.dbl und Cor1R48.dbl.

9. Menu Room Special: Test Convolution

It is recommended to run a test convolution after finishing macro 4. This step convolves the original pulse responses with the new correction filters and shows the result as curves 1 and 2.

Select the step response display in the time domain chart.



Typically you can notice some oscillation before the step. A low frequency oscillation is ok. But high frequency oscillations may be experienced as a nasty pre-ringing. A bad example:



Macro 4 has been calculated with 10/10. The step responses now get closer (they are even better) but the right channel now shows an oscillation. If you notice such a pre-ringing during listening you need to reduce the excessphase correction parameter in macro 4.

So there may be some iteration Steps between macro 4 and test convolution to find a good step response without a noticeable pre-ringing. This is dependent of the given system and room.

General rule: a pre-ringing can be accepted or tolerated as long it is not experienced during music playback.

10. Using the filter results

With the steps described before the correction filters have been calculated and are ready for use.

Enjoy the first listening!