Notes On AcourateLSR2 – LogSweep Recorder

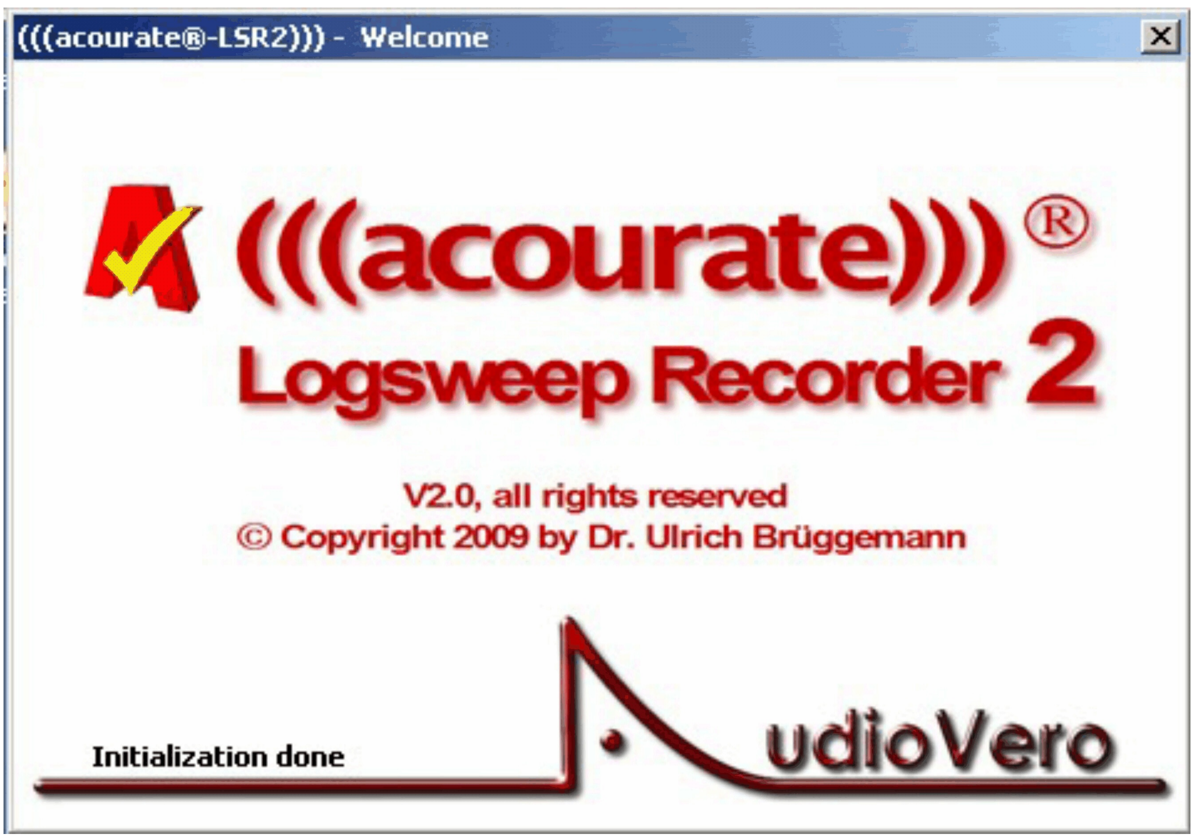
Ron Tipton – June 2016

This is a **free** stand-alone Windows (7 and above) program from www.acourate.com. It generates and records a log frequency sweep and generates and saves to a disc file the impulse response as a double-precision (64-bit) wav file – more on this later.

It doesn't have a User Guide, as such, but I've included a copy of the 2-page **readme.txt** file that gets installed when AcourateLSR2Setup.exe is run

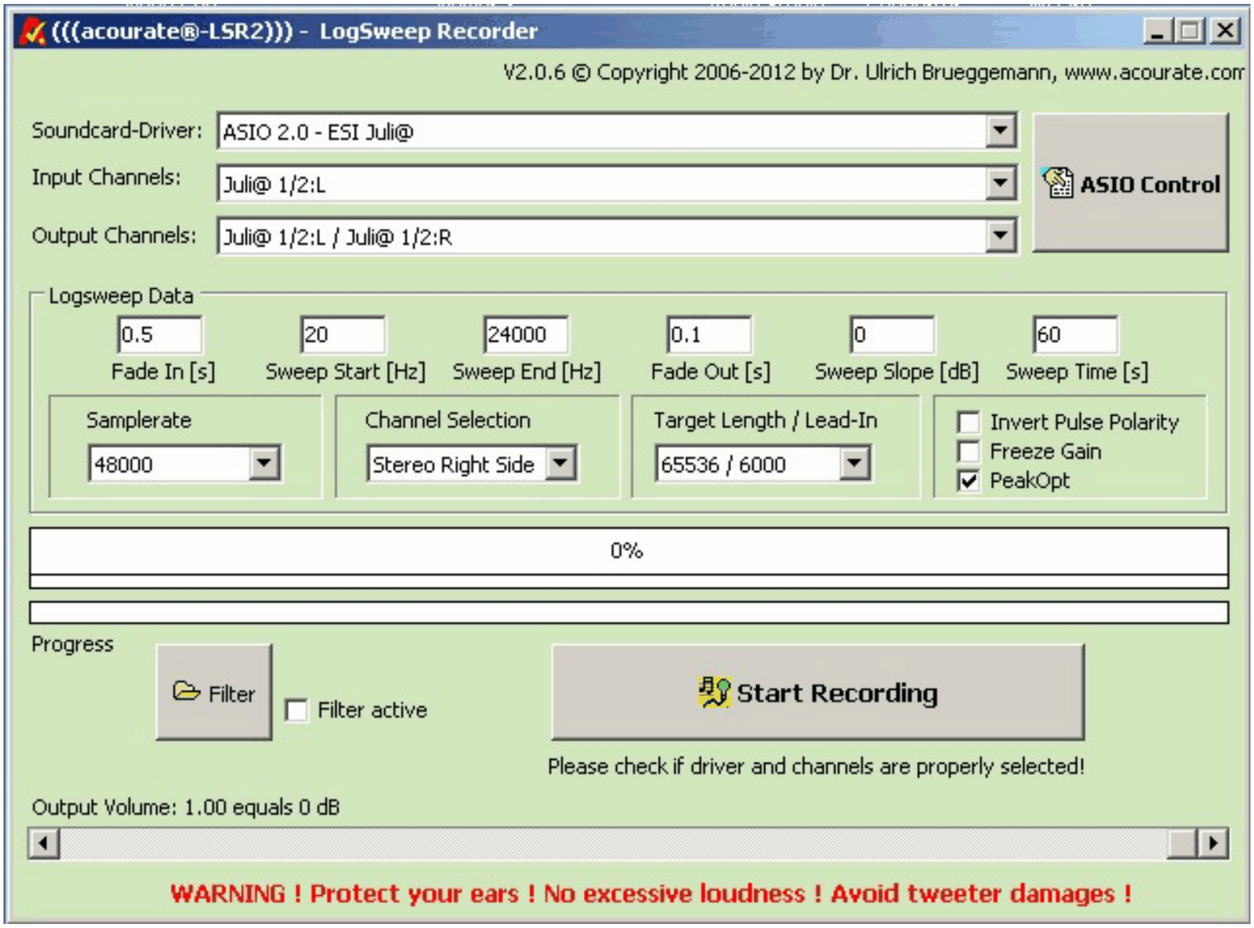
I have also included an online review from the http://digitalroomcorrection.hk/logsweep-recording website. The author's name was not mentioned..

Also, I've added a few notes of my own where, I think, some clarification is needed. I get this screen when the program is launched.



The **"Initialization done"** is important because it won't appear unless a sound card with an ASIO driver is found on your system.

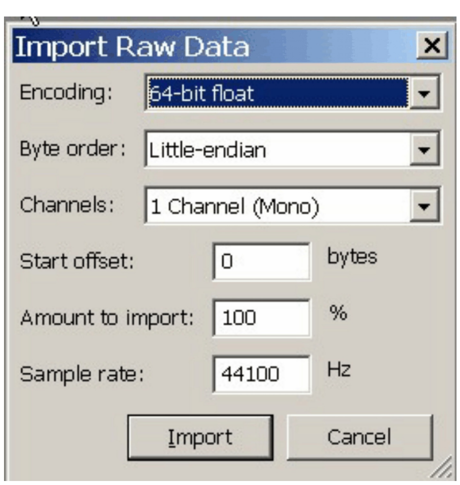
Click anywhere on the screen to continue



This is the next screen and all of its features are well described in the other included documents. But note that an ASIO sound card was located.

In Windows 7 (x64), the files are stored as follows:
C:\Users\(name)\My Documents\AcourateLSR2\LSRSweeps for the logsweep files and
C:\Users\(name)\My Documents\AcourateLSR2\Pulses for the impulse files.

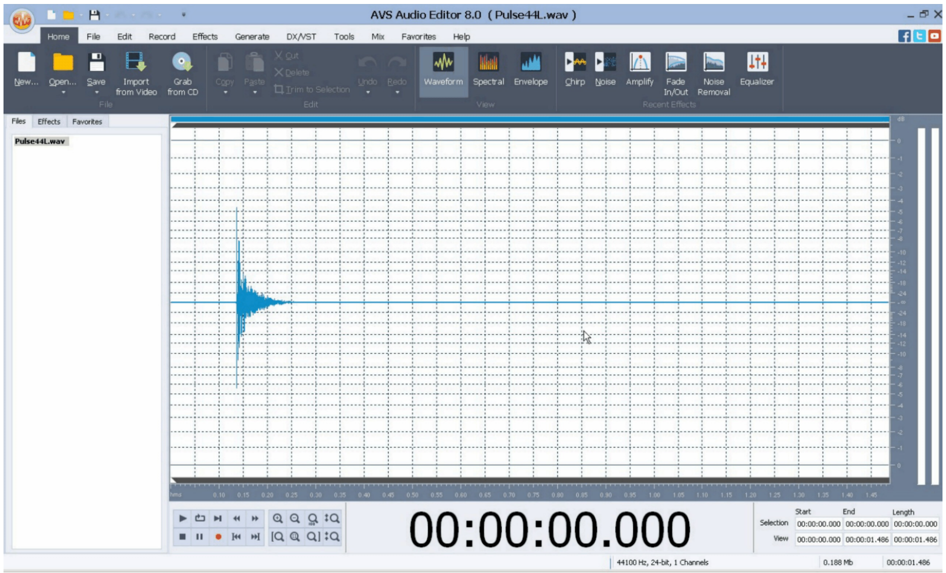
As I mentioned, the impulse files are 64-bit wav but the free **Audacity Audio Editor** will load them by clicking **Files | Import | Raw data**.



Audacity filled in all this automatically, just click **Import**.

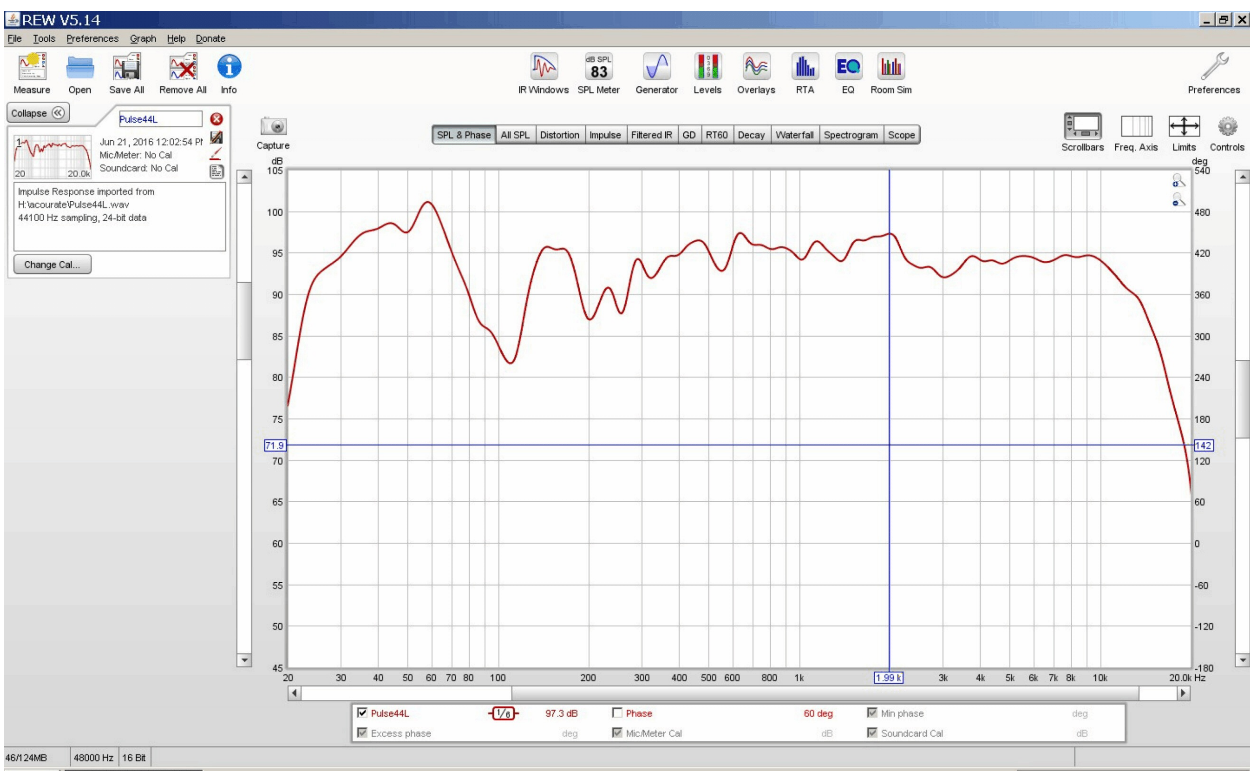
Some non-Acourate software will not recognize or load a double-precision wav file. You can easily convert to signed 24-bit format by clicking **Export** and then saving it with a new name and a wav extension.

The **AVS Audio Editor** loaded the converted file and recognized it as signed 24-bits.



Room EQ Wizard (REW), among others, does not recognize a floating point wav format. I launched **REW**, clicked **File** and **Import Impulse Response**. It did so and displayed the frequency response with 1/6-octave smoothing, shown below. Virtually the same left-front speaker response as I previously measured with **REW**.

You can see the windowing **REW** applied by clicking **Impulse** in the display bar.



(((acourate®-LSR2)))
LOGSWEEP RECORDER
README

The logsweep recorder is intended to be used as supplement to (((acourate)))®. It allows an easy playback and recording of a logsweep file for room measurements.

(((acourate®-LSR2))) fulfills the following tasks:

*** Generation of a logsweep audio test signal. Beside the logsweep the signal also contains short pulses of 1 sample length (Dirac clicks) to identify the logsweep sequences. It is possible to set the Dirac amplitude to 0.0 in the ini-file if you like to suppress it. The logsweep is stored generally as a 64-bit float stereo wav-file.
You can define the logsweep signal to be played:
+ first left, then right = stereo
+ mono = left + right together
+ left side only
+ right side only
Also further logsweep parameters can be set:
+ Start Frequency
+ End Frequency
+ Logsweep length in seconds (max. 60 seconds)
+ Slope in dB (default 0 dB). This parameter defines the attentuation of the highest frequency (22050 Hz/24000 Hz). The sweep gain is reduced along a slope to protect a tweeter against an overload.
+ FadeIn time (default 0.5 sec)
+ FadeOut time (default 0.1 sec)

*** Payback of the logsweep through a selected soundcard and output channel. The output volume can be influenced by a volume slider.

*** Automatic recording during playback. The level meter indicates the input level by green, yellow and red color. The red colour indicates the danger of clipping which should be avoided. The recording stops automatically at the end of the playback.

*** Generation of the logsweep inverse and computation of the pulse responses from the measured logsweep. The resulting pulse/pulses are stored in a raw 64-bit double precision floating point format (file extension .dbl). The pulse lengths can be 65536 samples with 6000 samples before the peak or 131072 samples with 12000 samples before the peak.

*** Filter definition: It is possible to apply a filter in combination with the generated playback. This allows to apply e.g. a correction filter for the frequency response of the microphone or the soundcard. It is also possible to check a room correction by using the correction filter during the logsweep recording. For a proper filter function a 32-bit wav-file, either mono or stereo has to be prepared, max. length is 262144 taps.

*** Samplerate: The available samplerates are reported by the soundcard. USB soundcards may have limited samplerates due to the full-duplex operation of the USB bus

*** Pulse Polarity: If an undesired polarity change is caused by the recording equipment it is possible to change the result polarity

*** Freeze Gain: The computed pulses typically are normalized for a maximum amplitude of 1.0. It is possible to freeze the gain for a constant normalization factor. Simply check the according checkbox for the first measurement. This will keep the amplitude relationship between several measurements

*** PeakOpt: A special function will adjust the pulse peak to a sample position by sub-sample shifting. The resulting pulse peak will automatically get its maximum peak.

*** IACC: For stereo recordings finally the IACC values for durations of 10 ms, 20 ms and 80 ms will be displayed. IACC is defined as interaural coherence coefficient and it is a measure for the interchannel matching. The max. theoretical result is 1.0. A high value indicates a good stereo focusing.

PLEASE NOTE:
The result can be used directly by (((acourate)))®.
You also can send the result to the author for a further preparation of a demo application (treatment of your favourite music track with a correction filter)

*** ASIO interface
The communication to the soundcard is carried out by an ASIO interface. Tyically good soundcards are equipped with proper ASIO drivers. If (((acourate®-LSR2))) does not find a driver you can try to use www.asio4all.com with your soundcard. It may happen that the driver fails. Check the supporting website of the soundcard manufacturer for a newer driver.

*** Installation
Simply run AcourateLSR2setup.exe and follow the instructions. (((acourate®-LSR2))) itself is not using the registry.

Logsweep Recording

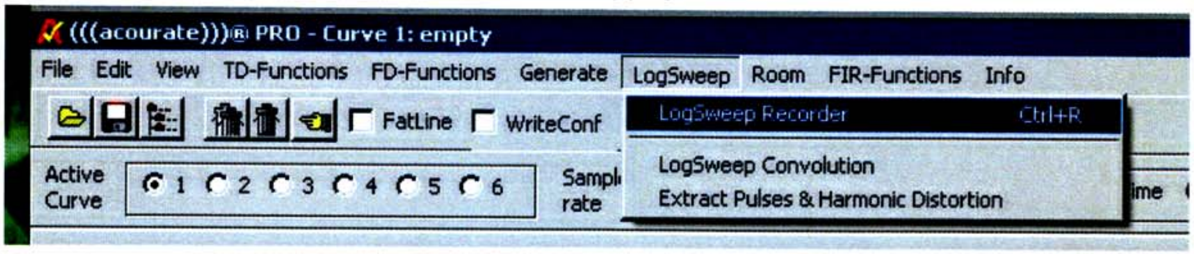
Acourate Software

Recording Logsweep

Logsweep recording is done by Uli's program LogsweepLSR2. Here is some info from the readme file. It is also now available as part of the Acourate software and everything is integrated and automatic making measurement a lot easier.

I used to use MacBook Pro with bootcamp for the recording purpose. But it turns out that there is a lot of latency problem with Bootcamp during recording and in the end, it just dropped the idea and use a Lenovo laptop running Windows 7 for recording. Life is a lot easier then!

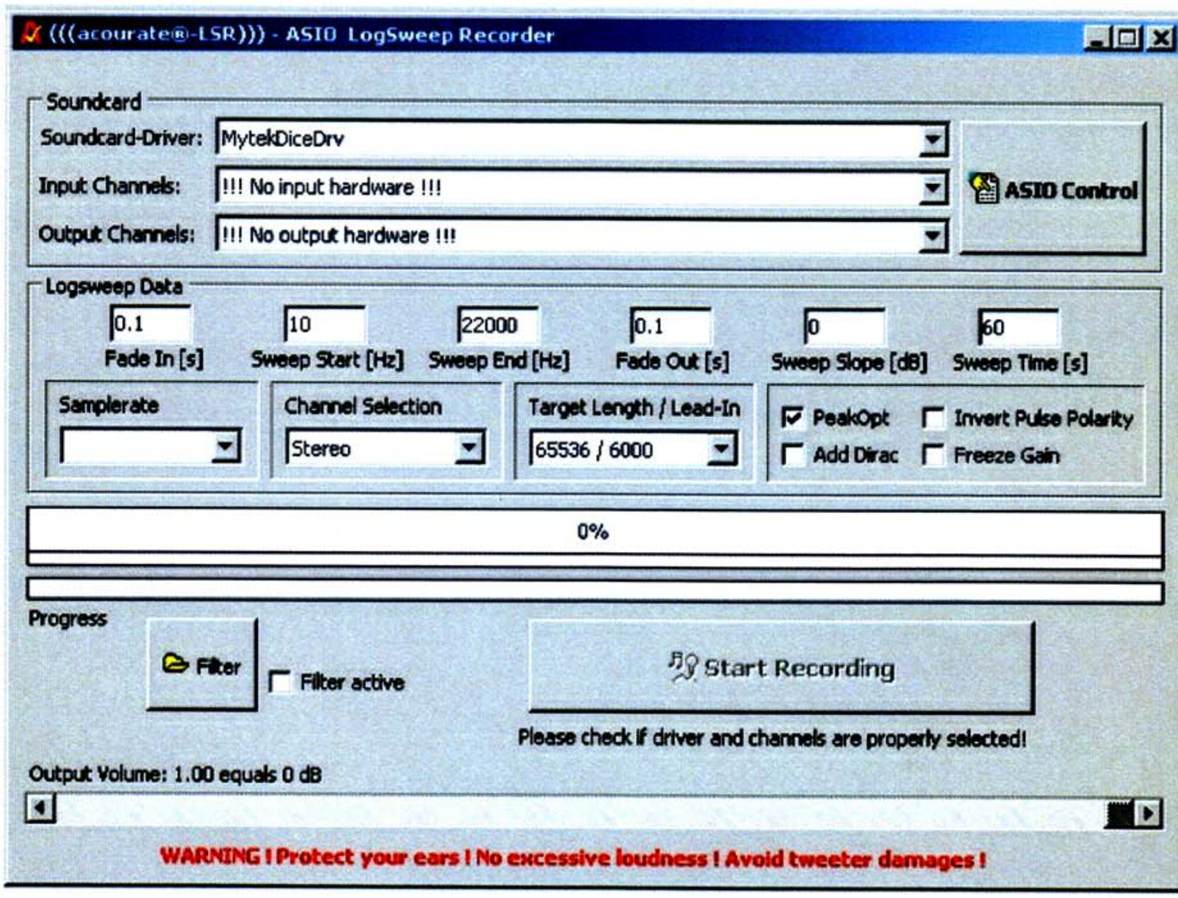
When you open the logsweep recorder as shown



you will get the following screen. This is the whole recording center.

The new version actually comes with a setting when you can use the mic calibration file. I use earthworks mic and I got the calibration file from earthworks directly. Here is the steps involved:

import the text file.
Then apply FD-Functions – Amplitude Inversion – minphase and save the result as mono wav. Use the wav file in the logsweep recorder.



Recording level: The best case is when the meter goes up to the red level. A red color means that you are just before clipping of the input signal.

Beside the logsweep the signal also contains short pulses of 1 sample length (Dirac clicks) to identify the logsweep sequences when the "Add Dirac" box is checked. The logsweep is stored generally as a 64-bit float stereo wav-file.

You can define the logsweep signal to be played with the following options.
+ stereo = first left, then right
+ mono = left + right together
+ left side only
+ right side only

Also further logsweep parameters can be set:
+ Start Frequency
+ End Frequency
+ Logsweep length in seconds (max. 60 seconds): recommended sweep time 60 seconds, for 192 kHz max 40 seconds
+ Slope in dB (default 0 dB). This parameter defines the attenuation of the highest frequency (22050 Hz/24000 Hz). The sweep gain is reduced along a slope to protect a tweeter against an overload.
+ FadeIn time (default 0.5 sec)
+ FadeOut time (default 0.1 sec)

Payback of the logsweep through a selected soundcard and output channel. The output volume can be influenced by a volume slider.

Automatic recording during playback. The level meter indicates the input level by green, yellow and red color. The red colour indicates the danger of clipping which should be avoided. The recording stops automatically at the end of the playback.

Generation of the logsweep inverse and computation of the pulse responses from the measured logsweep. The resulting pulse/pulses are stored in a raw 64-bit double precision floating point format (file extension .dbl). The pulse lengths can be 65536 samples with 6000 samples before the peak or 131072 samples with 12000 samples before the peak. Use 65536/6000 for 44.1KHz, 48KHz, 88.2KHz and 96KHz. Use 131072/12000 with 192 kHz.

Filter definition: It is possible to apply a filter in combination with the generated playback. This allows to apply e.g. a correction filter for the frequency response of the microphone or the soundcard. It is also possible to check a room correction by using the correction filter during the logsweep recording. For a proper filter function a 32-bit wav-file, either mono or stereo has to be prepared, max. length is 262144 taps.

Samplerate: The available samplerates are reported by the soundcard. USB soundcards may have limited samplerates due to the full-duplex operation of the USB bus

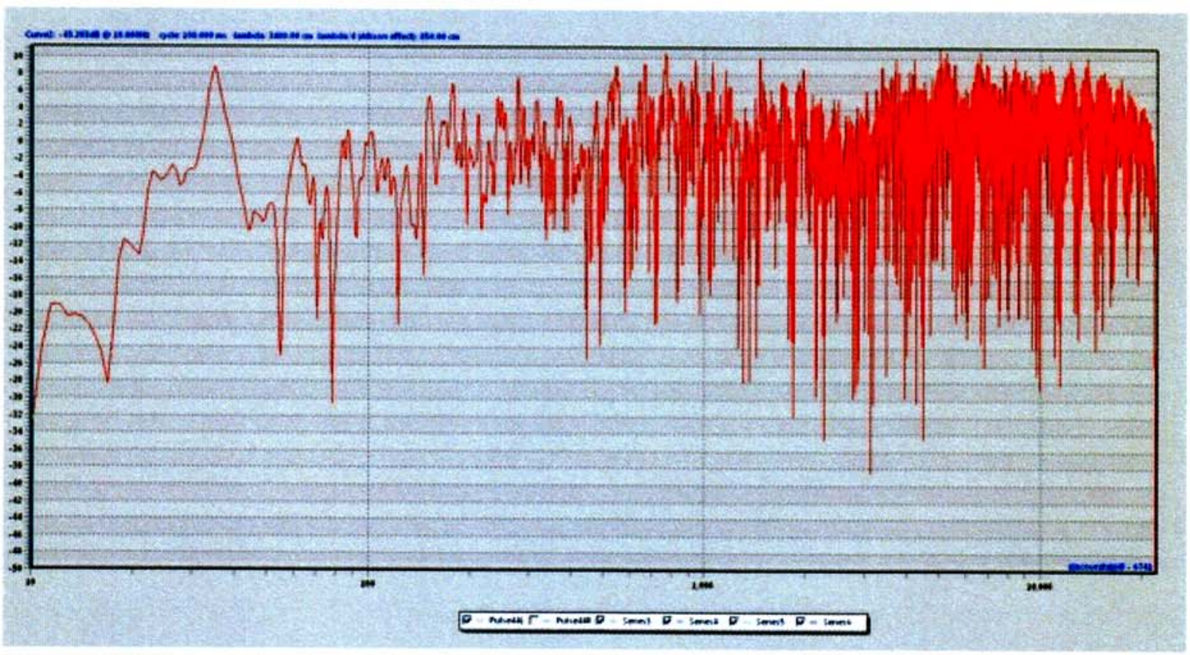
Pulse Polarity: If an undesired polarity change is caused by the recording equipment it is possible to change the result polarity

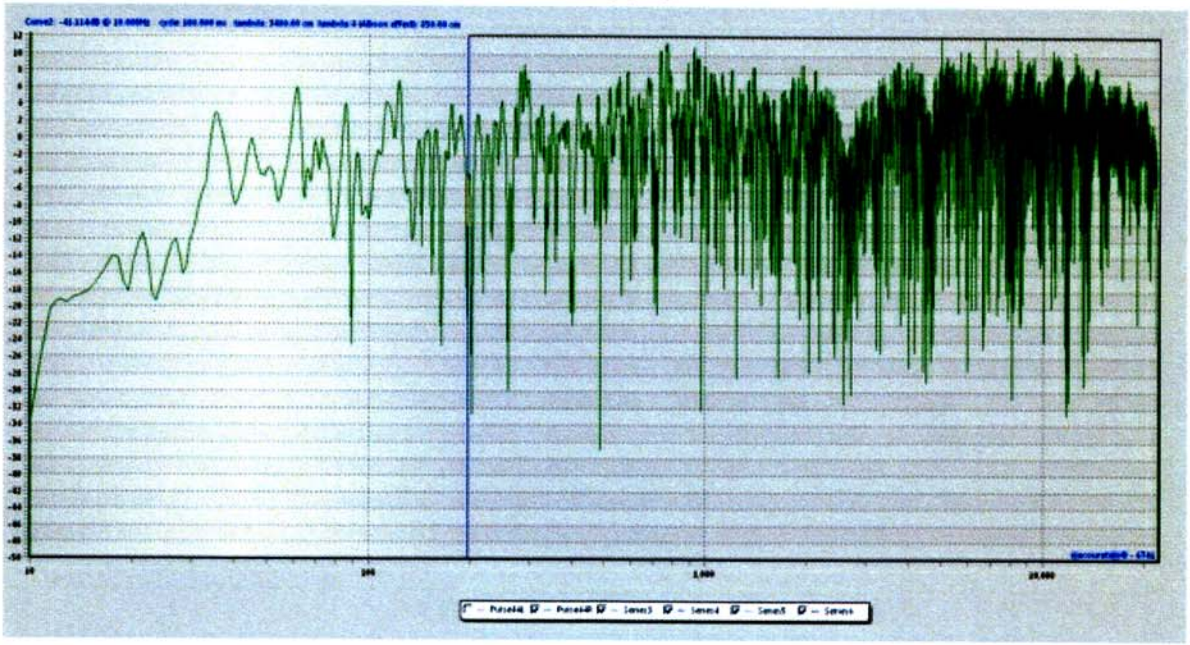
Freeze Gain: The computed pulses typically are normalized for a maximum amplitude of 1.0. It is possible to freeze the gain for a constant normalization factor. Simply check the according checkbox for the first measurement. This will keep the amplitude relationship between several measurements. Tips: use freeze gain is only necessary if you measure single drivers, then start with tweeter and freeze gain. This ensures that all drivers have the same level in the result.

PeakOpt: A special function will adjust the pulse peak to a sample position by sub-sample shifting. The resulting pulse peak will automatically get its maximum peak. Leave it checked by default.

IACC: For stereo recordings finally the IACC values for durations of 10 ms, 20 ms and 80 ms will be displayed. IACC is defined as interaural coherence coefficient and it is a measure for the interchannel matching. The max. theoretical result is 1.0. A high value indicates a good stereo focusing.

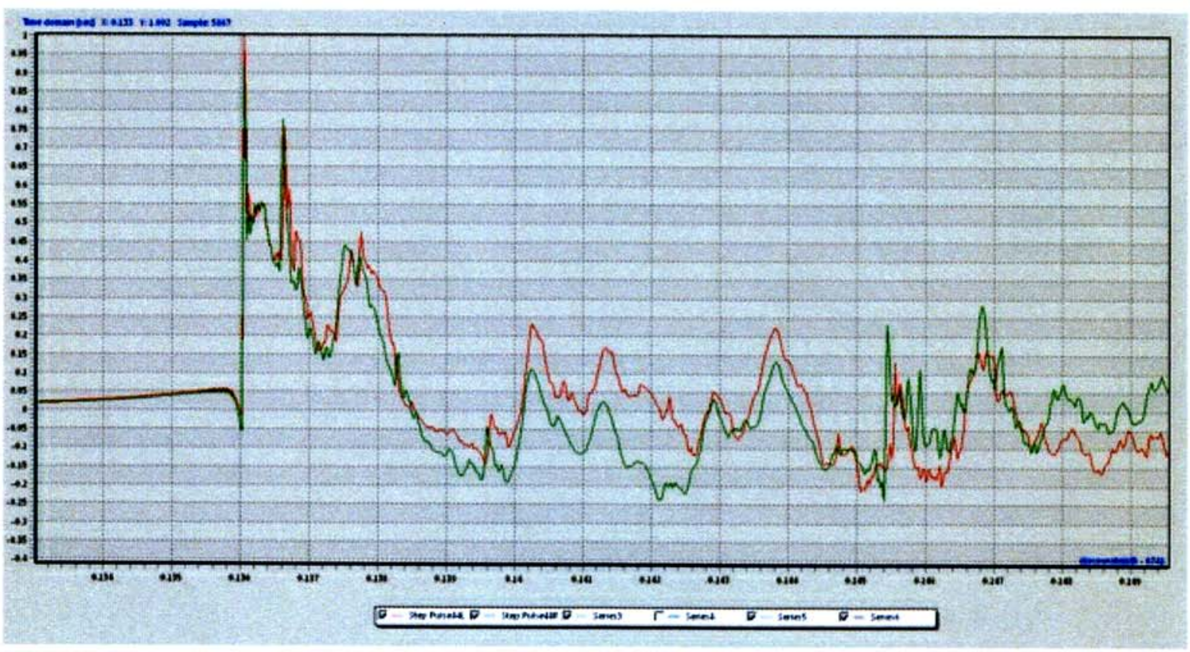
I started with measuring the existing setup straight out of the XO generator from Acourate. With my setup, my ULN-8 gain is set at 50dB. Let's have a look at the recorded signal with the XO generated with the previous chapter



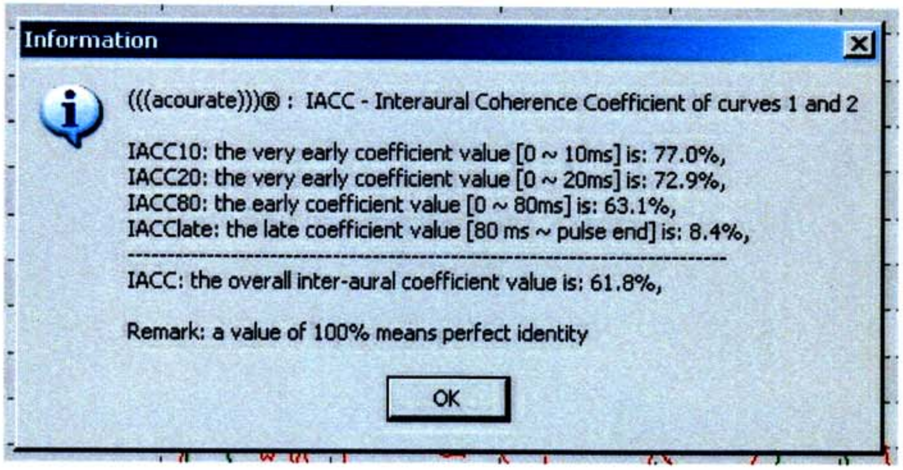


Both the pulseL and pulseR is shown. It seems that the frequency response is quite similar in both side with a peak at about 34Hz and quickly drop down to -14dB at 20Hz. There is also a dip at 2600Hz in both sides indicating that could be a crossing over or driver alignment issue. If you look at the graph on previous page, this is the exact crossover point.

The step response in both side is really not bad at all. But I think I can still improve on this.

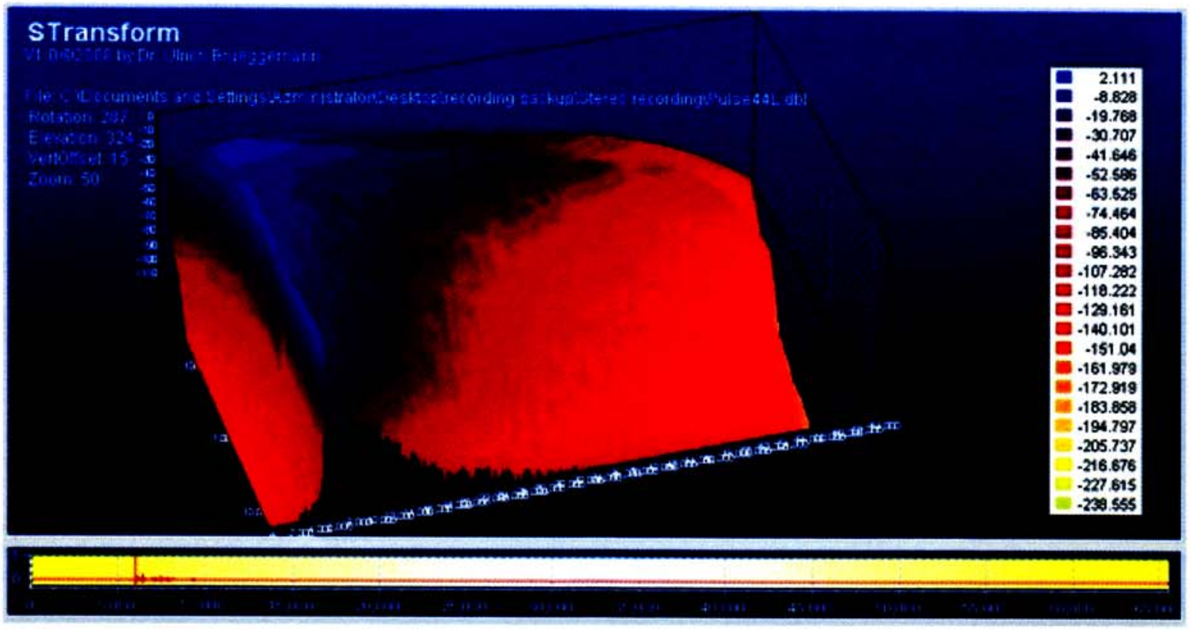


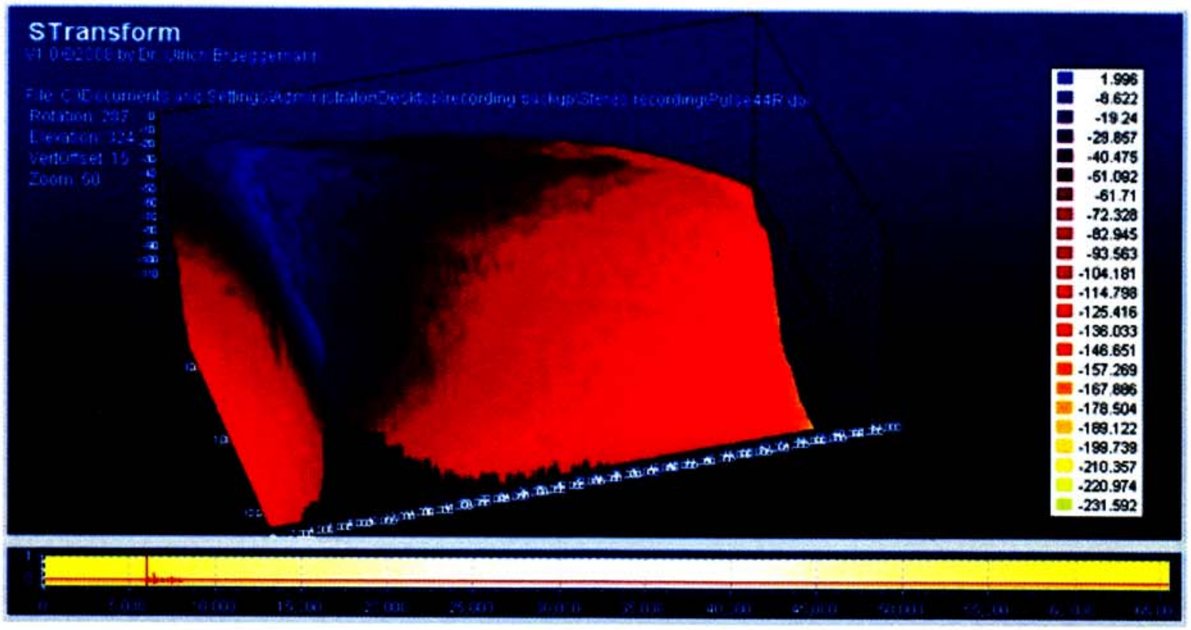
IACC in this set up is



Not bad for a setup without driver alignment and correction. I remember Uli once mentioned that the best he has seen is 80%, so 61.8% is not too bad!

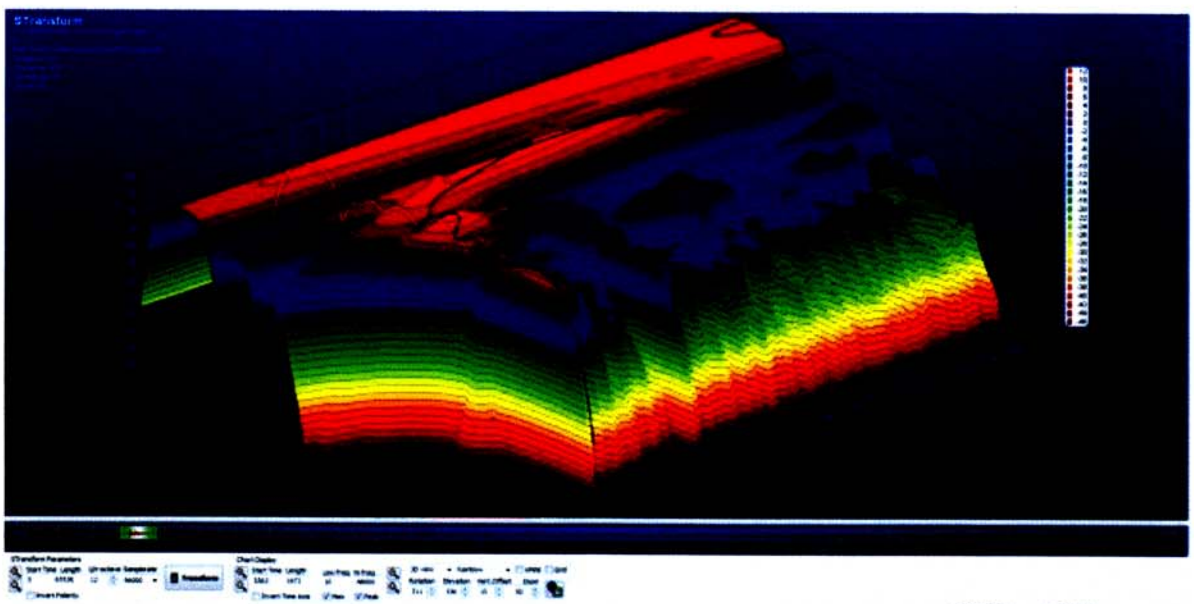
Using STransform provided by Uli, here is the pulseL and pulseR





In these maps, you can see that there is a single peak for each frequency indicating that there is no strong reflections from the walls. The bass region also has a nice slow run off with no pressure building in the room.

The following picture is coming from another room



You can see that there are many peak for the frequency above 100Hz, indicating there are are reflections. The bass is also built up with no signs of decay. Good or not, is subject to the listerner.