waveFLEX-caraudio.fr

Setting up your caraudio system

DSP A8 and Room Equalizer Wizard V5

Part 1

1) Foreword

The first DSP units arrived on the market more than 15 years ago and today most users understand the overall functions and take advantage of the possible settings.

The leap forward in technology of the latest DSP systems on the other hand also means that:

- The human hearing is not up to the task anymore
- We are unable to clearly analyze the situation

Only the use of appropriate measuring tools, correct methods of analysis and properly implemented correction algorithms allows using the new DSP units up to their full potential.

We have put a lot of time and effort into creating this MOOC, it first should help you to understand what correction is all about and in the end to master the corrections yourself.

2) Goals

This reference guide will help you getting familiar with your DSP unit and the measuring/simulation tools that ROOMEQWIZARD offers.

And of course you will learn one method of sound correction to get the most out of your car install.

We'll talk about the following things amongst others:

* Setting up the necessary equipment
* Measuring the speakers
* Analyze the measurements
* Simulating filters and level equalization
* Phase correction
* ... and much more

Each chapter will contain a little theoretic background. The goal is not to spoon-feed you with theoretical stuff, just an overview of what settings to change and for what reason.

3) Requirements

- 1 PC or Mac computer
- Room Equalizer Wizard (REW V5.0 beta 18 minimum) software
- ***DSP A8*** software interface
- For DSP NX users: also the ***DSP NX***, ***DSP NX*** software + WaveFlex FIR Tool (WFT)
- 2 stereo RCA to 3.5 mm jack adapter cable
- 1 ***DSP A8*** (+ 1 ***DSP NX***)
- 1 UMIK-1 microphone and supplied calibration file
- 1 microphone stand

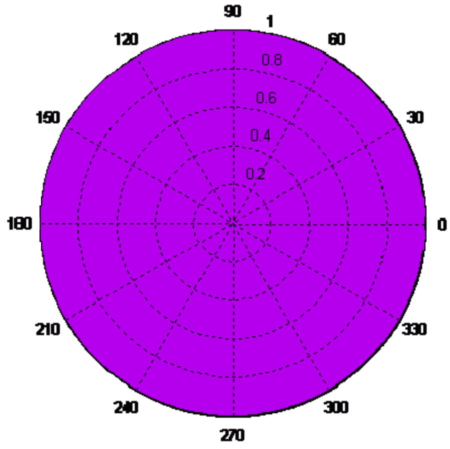
If you measurement equipment is different (for instance XLR microphone + Phantom power supply and/or digital output soundcard) you will have to adapt the settings in REW and your soundcards settings accordingly. This document cannot cover all the possible set-ups.

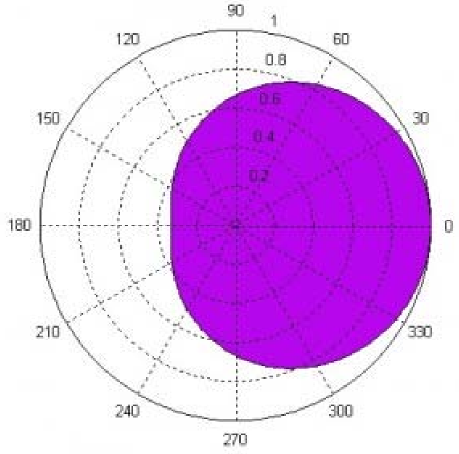
4) Setting up your measurement gear

a) Reminder on microphones

The choice of the physical location of the microphone mainly depends on the number of listening positions you want to improve. In this document we will only refer to driver position.

Generally speaking « capsule » microphones (i.e. UMIK-1) are considered being omnidirectional: The frequency response is independent from the source of the sound signal and the frequency. Well that's the theory at least, and more or less true at low frequencies only. At higher frequencies the masking effect of the microphone body and the physical size of the capsule will induce a directional behavior; we then talk about a large cardioid. The frequency where the behavior changes is around 5 KHz.





To avoid this phenomenon, we strongly advise you to use your microphone horizontally. That way you can also take advantage of the supplied microphone calibration file as it's given for a 0° angle (i.e. horizontal).

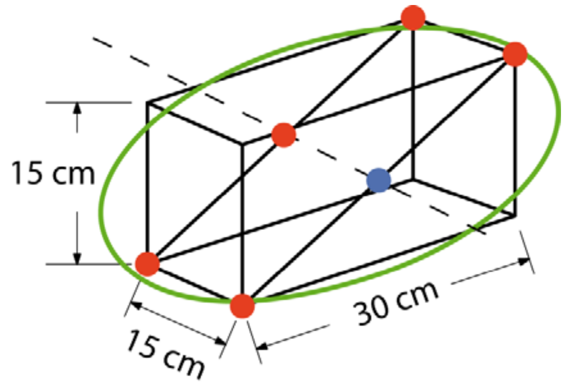
If you own another microphone having a 90° calibration file you might as well use it vertically.

b) « The sound power method » by Earl Geddes

The physical location of the microphone in the car is crucial and requires your attention. In fact, to optimize the correlation of the result of our measurements and our listening impressions we need to use a specific measuring² method. The frequency response highly depends on the position of the microphone (positioned in the listening area of course). A measurement at one single position only will not allow us to properly equalize our system. In order to have a more representative measurement of the perceived sound in our ears we will use multiple measurement points in space (virtually) where our head is.

To obtain a sufficiently precise measurement of the frequency response at the listening position a total of six measurement points, as demonstrated by the works of Earl GEDDES, will give us a precision of more than 90%, enough for us to equalize the sound levels within 0.5 dB

These 6 measurement point are situated in an elliptic area that represents best the area our ears perceive. The 6 measuring point are enough distance from each other to be considered independent. The drawing hereunder shows the 6 measuring points. The central position (blue dot) is the virtual center between our ears.



To obtain the global frequency response of our system we will average the 6 measuring points.

We will also have to create differential measurements curves between every measuring point and a chosen reference point. We will use the central position as reference as usually it's the easiest to set up, but any position will work.

Now, in case of future equalization or even to control the results of the actual equalization it is enough to measure the frequency response at the reference point and apply our correction curves to the other measuring points to have 6 measurements without having to measure them all and keeping the 0.5 dB precision we want. This method saves a lot of time to obtain the global frequency response.

We will see how to handle the obtained results later in this document.
Now let's have a closer look at the needed tools and how to make all the needed measurements.

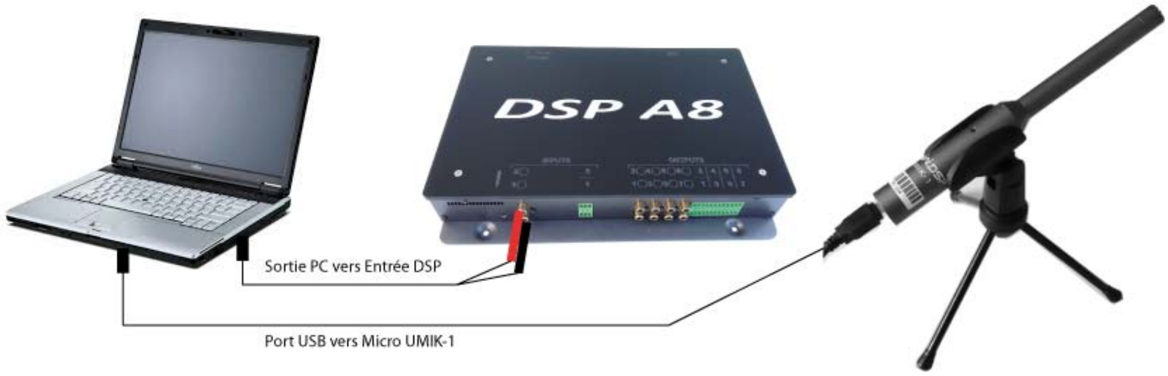
5) SPL Calibration

To obtain a representative measurement of your sound levels, you need to calibrate the sound level first. We will use our calibrated microphone and the corresponding calibration file in REW to calibrate the entire measurement chain.

c) Measurement Setup:

Connections to make

* UMIK-1 to PC/MAC via USB
* Headphone output to *DSP A8* inputs using the 3.5mm Mini-Jack-RCA cable

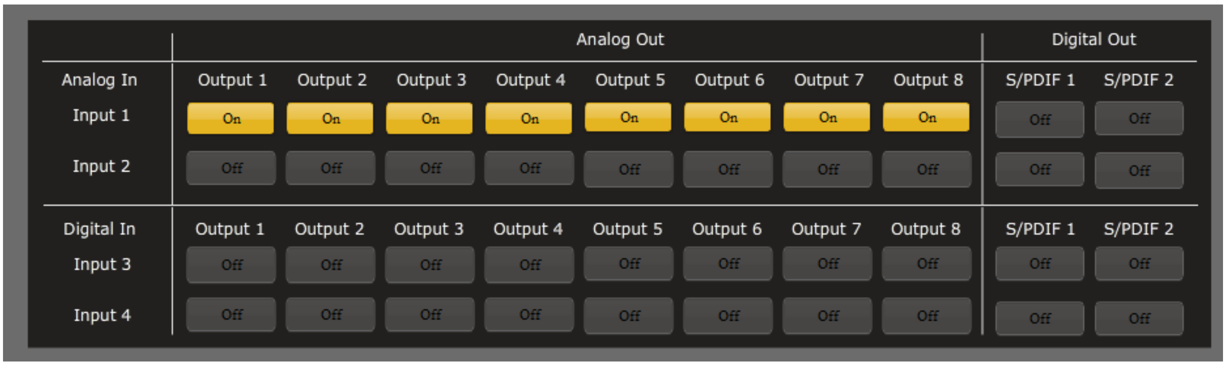


d) SOFTWARE CONFIGURATION

Disable all sound enhancing options as ambiances, effects etc. in the sound settings of WINDOWS and/or your soundcard. In the sound console set both the WAVE and the MASTER VOLUME tab to maximum.

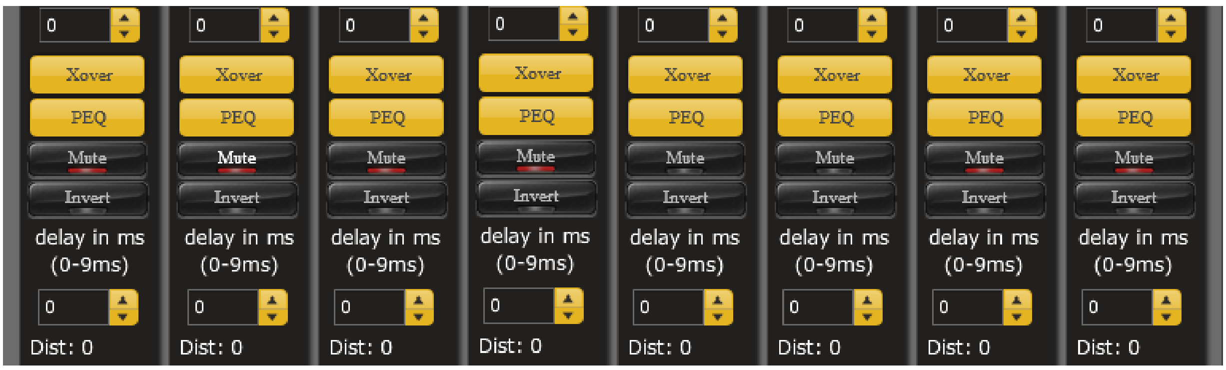
DSP A8 Configuration

Open the **DSP A8** software and configure the routing table as shown (Only input 1 is used and routed to all the output



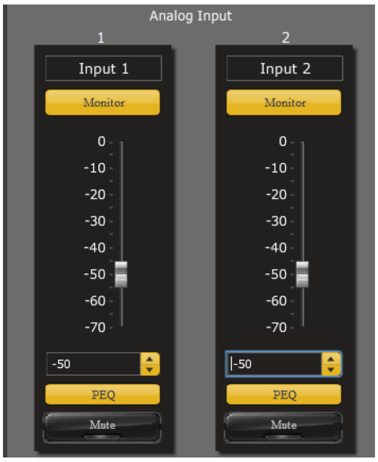
	Analog Out								Digital Out	
Analog In	Output 1	Output 2	Output 3	Output 4	Output 5	Output 6	Output 7	Output 8	S/PDIF 1	S/PDIF 2
Input 1	On	On	On	On	On	On	On	On	Off	Off
Input 2	Off	Off	Off	Off	Off	Off	Off	Off	Off	Off
Digital In	Output 1	Output 2	Output 3	Output 4	Output 5	Output 6	Output 7	Output 8	S/PDIF 1	S/PDIF 2
Input 3	Off	Off	Off	Off	Off	Off	Off	Off	Off	Off
Input 4	Off	Off	Off	Off	Off	Off	Off	Off	Off	Off

Configure all the 8 outputs channels without any X-over/EQ/Delay/gain filter. Mute all speakers but the 2 woofers (channels 5 & 6 in the example hereunder).



Then synchronize the software with your *DSP A8*.

The master volume needs to be attenuated during the measurement process.

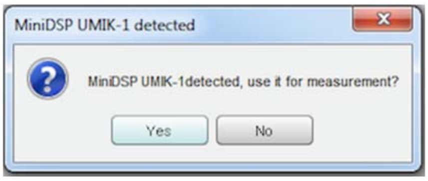


Turn down the master volume either using the (optional) ***MCR remote*** or attenuate the analog inputs.

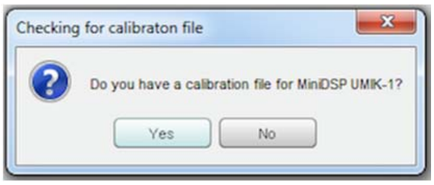
<u>Be careful: without attenuation the sound level might damage your installation</u>

Setting *REW* parameters

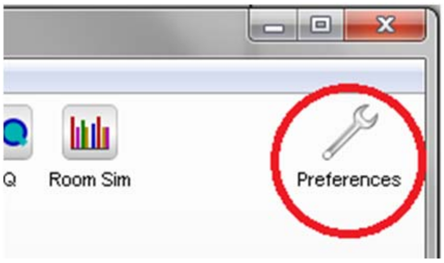
On start-up *REW* will detect the connected USB microphone and ask you to confirm the selection.



When asked if you have a calibration file for your *UMIK-1* load the *.txt file supplied with your microphone.



Open the PREFERENCES menu and configure the different tabs:

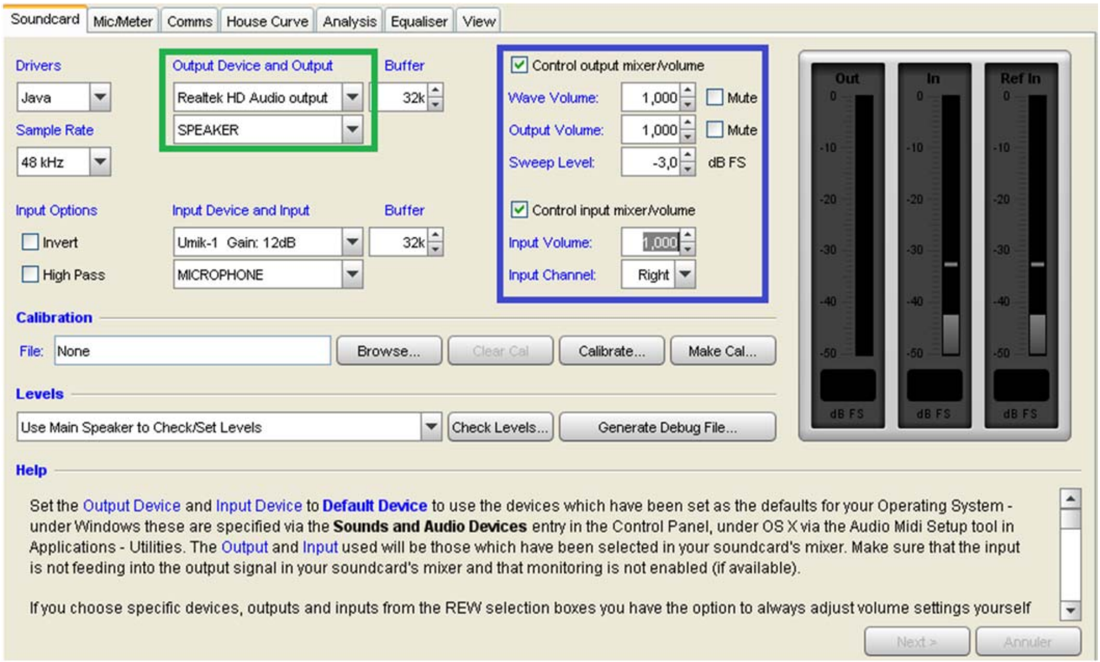


SOUNDCARD Tab

Check that *UMIK-1* is selected as *INPUT DEVICE.*

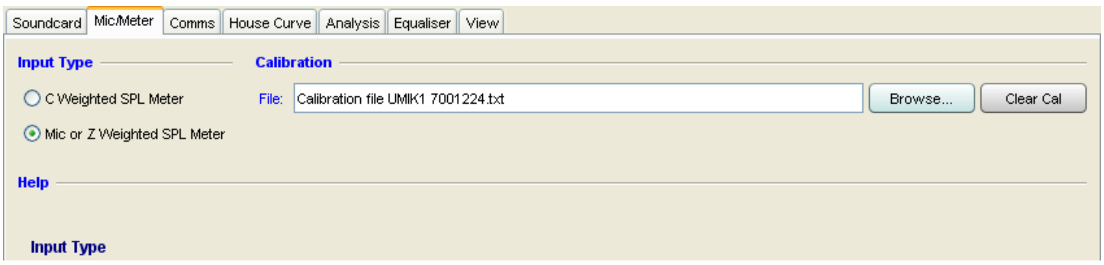
Select the headphone output in the *OUTPUT DEVICE* (green box)

Set the different parameters as in the blue box hereunder.



Mic/Meter tab

Check that the microphone calibration file has been loaded and that the Z-weighted input is selected.

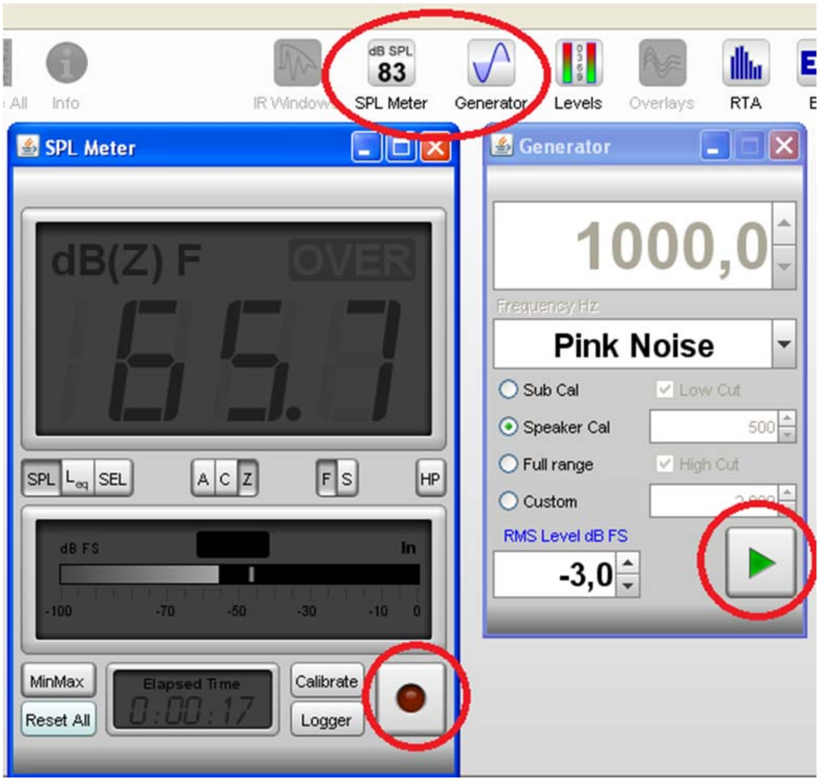


e) SPL Calibration

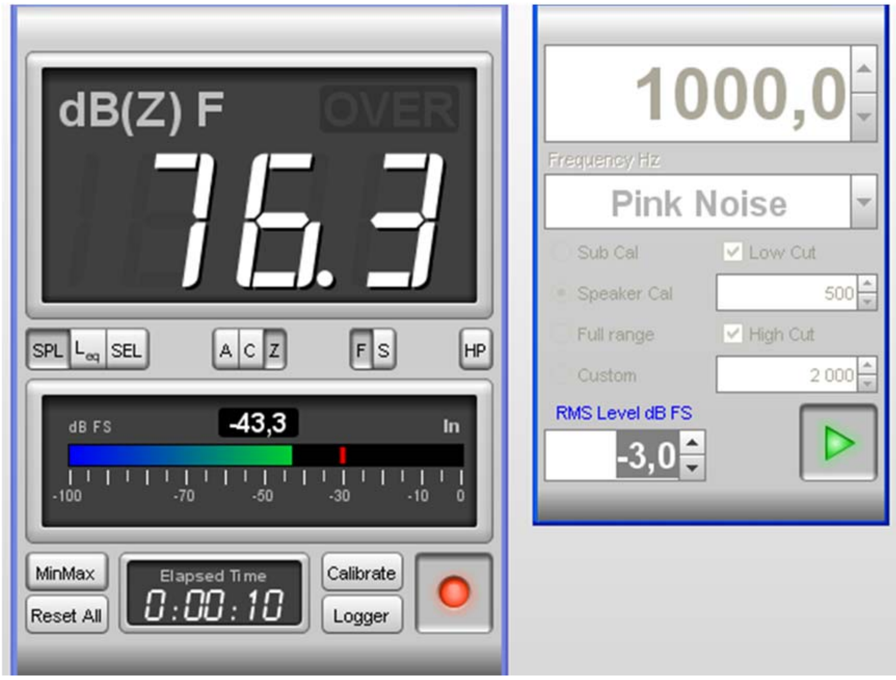
Open both the *GENERATOR* and *SPL METER* tabs and configure it as shown hereunder.

SPL METER tab: *SPL* mode selected, *z-weighted* microphone is used, and analysis is set to *fast*

GENERATOR tab: calibration signal set to pink noise from 500 Hz – 2 KHz at -3db.



Start both the *recorder* and the *generator* (play button/record button) and slowly increase the volume either on the (optional) *MCR remote* or increase the input level of the used the analog inputs up to something like 75 db.



Once you have reached that level turn off of both the *SPL METER* and the *GENERATOR.* Do not touch the settings of the analog inputs or the volume of the *MRC remote control* from now on. We need it to be at the same level for all our measurements.

We are now almost done configuring *REW*, we can leave the *SOUNDCARD* tab now.

I suggest you save your *DSP A8* configuration at that point, with the set input level and all outputs set to flat under « *PRESETMESURE* » for instance.

