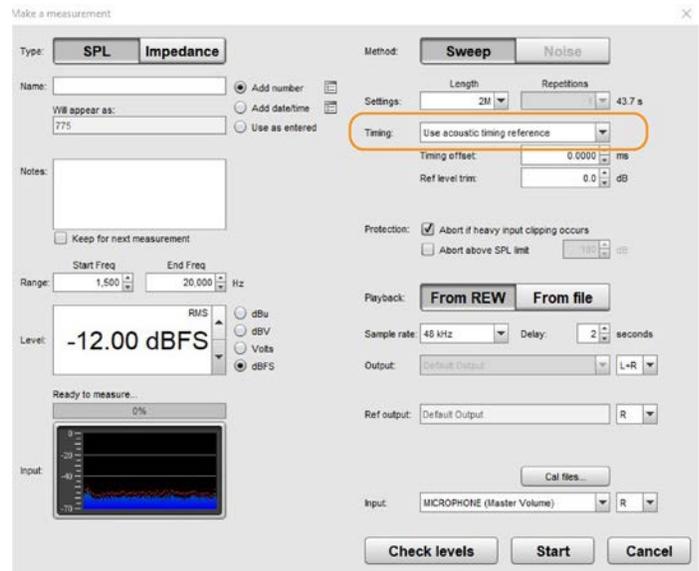
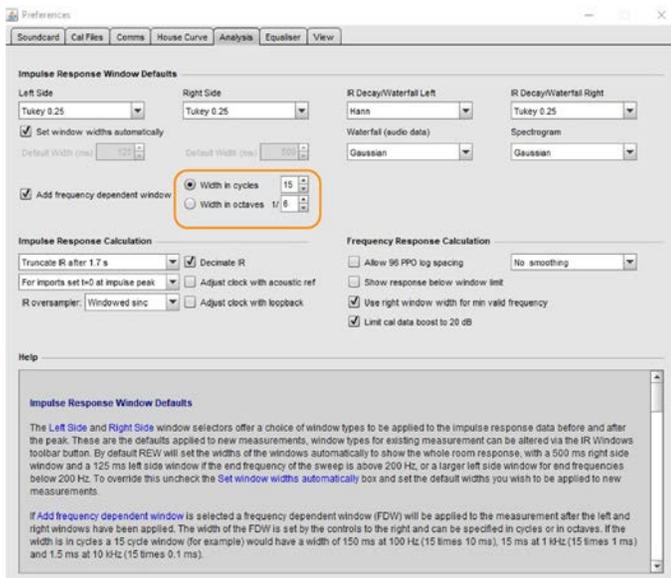


## REW average measurements and impulse correction rePhase

- Tutorial is meant to be used for speakers/ drivers implementation into your listening area. Integrating impulse corrected filters for your DSP device with or without "room curve". "Room curve" bases is Bruel and Kjer work in room acoustics: <https://www.bksv.com/media/doc/17-197.pdf> Goal is to produce the best possibly sounding speakers /drivers for your room. Take a measurement point microfon`s tip vertically ,towards sealing and use a 90° calibration file for your mikrofon, resulting in much better measuring results for REW.

### 1.REW Preferences



1/6th octave smoothing and 15 cycles FDW to generate the correction filters and avoid 'micro-managing' the amplitude and phase corrections.

Timing reference activated.

### 2. Room curve

Data used to make a "room curve" can be imported to REW as a text file or using "target settings" in REW.

"EQ" -> "target settings" -> activate "Add room curve".

It could take to do the same tuning to reach the desired result and this task is up to you. Settings in a picture is just a guideline/starting point.

Assuming you follow this tutorial with a goal is to implement a "room curve" with impulse corrected filters for your DSP device.

Then, "Equalizer" chooses "rePhase", because we will end up with producing FIR filters in rePhase software for your device.



Importing text file into the REW software.

First you must make a text file. Copy digits and paste in to your text editor and save it as txt file with a name ( free choice ) :

25.198 0.000  
31.748 -0.001  
40.000 -0.005  
50.397 -0.016  
63.496 -0.039  
80.000 -0.079  
100.794 -0.134  
126.992 -0.203  
160.000 -0.290  
201.587 -0.397  
253.984 -0.528  
320.000 -0.683  
403.175 -0.866  
507.968 -1.087  
640.000 -1.351  
806.349 -1.651  
1015.937 -1.972  
1280.000 -2.302  
1612.699 -2.634  
2031.873 -2.967  
2560.000 -3.300  
3225.398 -3.634  
4063.747 -3.967  
5120.000 -4.300  
6450.796 -4.634  
8127.493 -4.967  
10240.000 -5.301  
12901.592 -5.634  
16254.986 -5.967  
20480.000 -6.301  
22050.000 -6.301

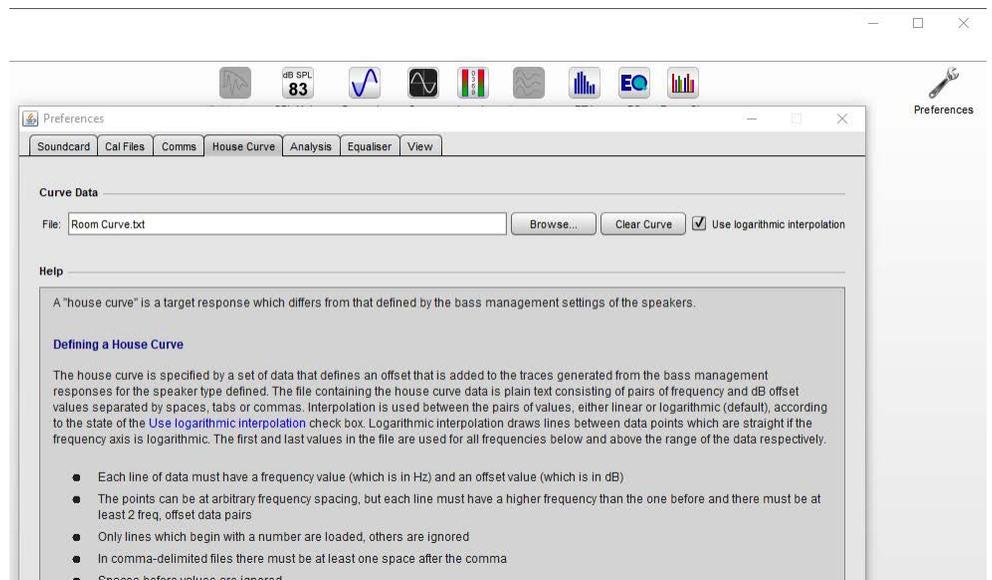
**This room curve based on Bob Katz' recommendations, diyAudio forum.**

0 -90.0  
5 -6  
6 -3  
7.5 -1  
10 -0.1  
20 0.0  
50 0.0  
100 -1  
200 -2  
400 -3.0  
800 -4.0  
1000 -4.2  
2000 -4.7  
3200 -5.3  
6400 -6.5  
12800 -7.5  
19200 -8  
22050 -10

**This room curve based on "fluid" recommendations, diyAudio forum.**

Then saved file import to REW.

"Preferences" ->"House Curve"->"Browser" find your saved file and import it.



How to Integrate impulse corrected filters and "room curve" for your DSP device ,will be explained in detail in a step 7. Combined filter

Each and every person on this earth is equipped with a singular hearing organ belonging only to him and no one else has it the same sound interception experience. "Room curve" is how most people would love sound to be in their listening room for music. It is up to you to use " room curve" or don't.

If you are using it:

"room curve" should be activated for all times in REW software. From step one of this tutorial up to the end.

If you are not using it:

Just don't activate .

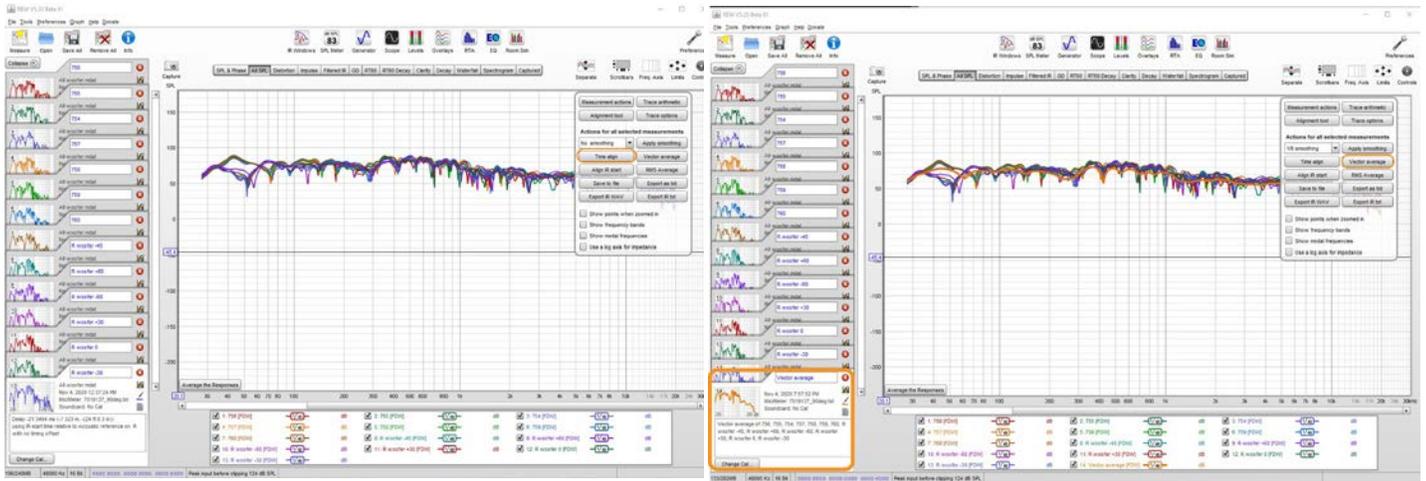
How you will use the tutorial depends on your DSP hardware/software set up. Following up to the end of tutorial step 7. Combined filter,we will end up with having needed information for rePhase software to produce file for FIR filter for your hardware/software.

Theoretically it would be only one filter that you need,minimizing use of PEQ filters.

If your set up doesn't have FIR in it,tutorial for you it is usefully up to step 4.PEQ generated filter.

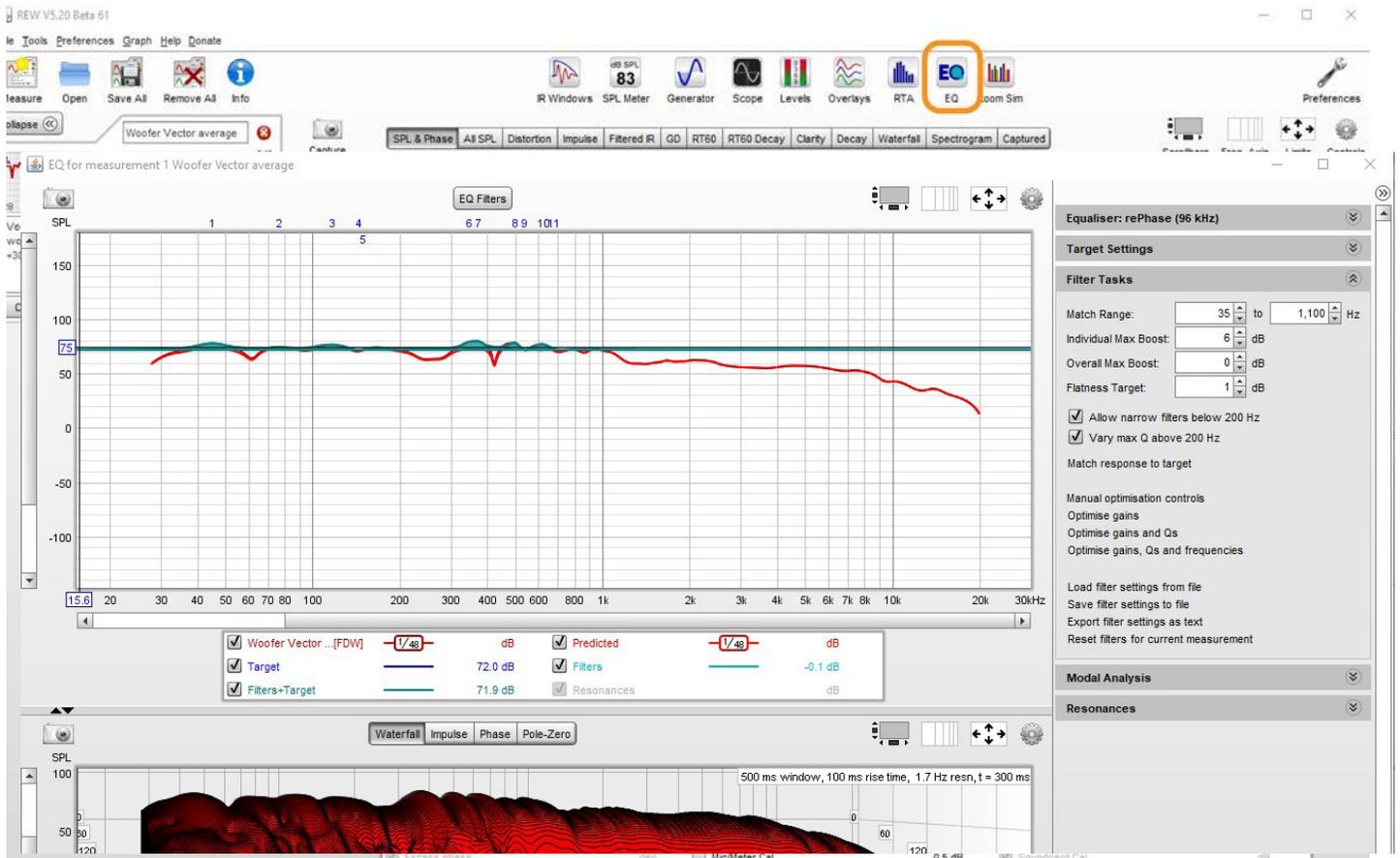
### 3. Averaged measurements.

Take several meaningful measurements representing your listening area . Import all of them in to the REW .  
"All SPL" -> "Control" -> "Time Align"->"Vector average".

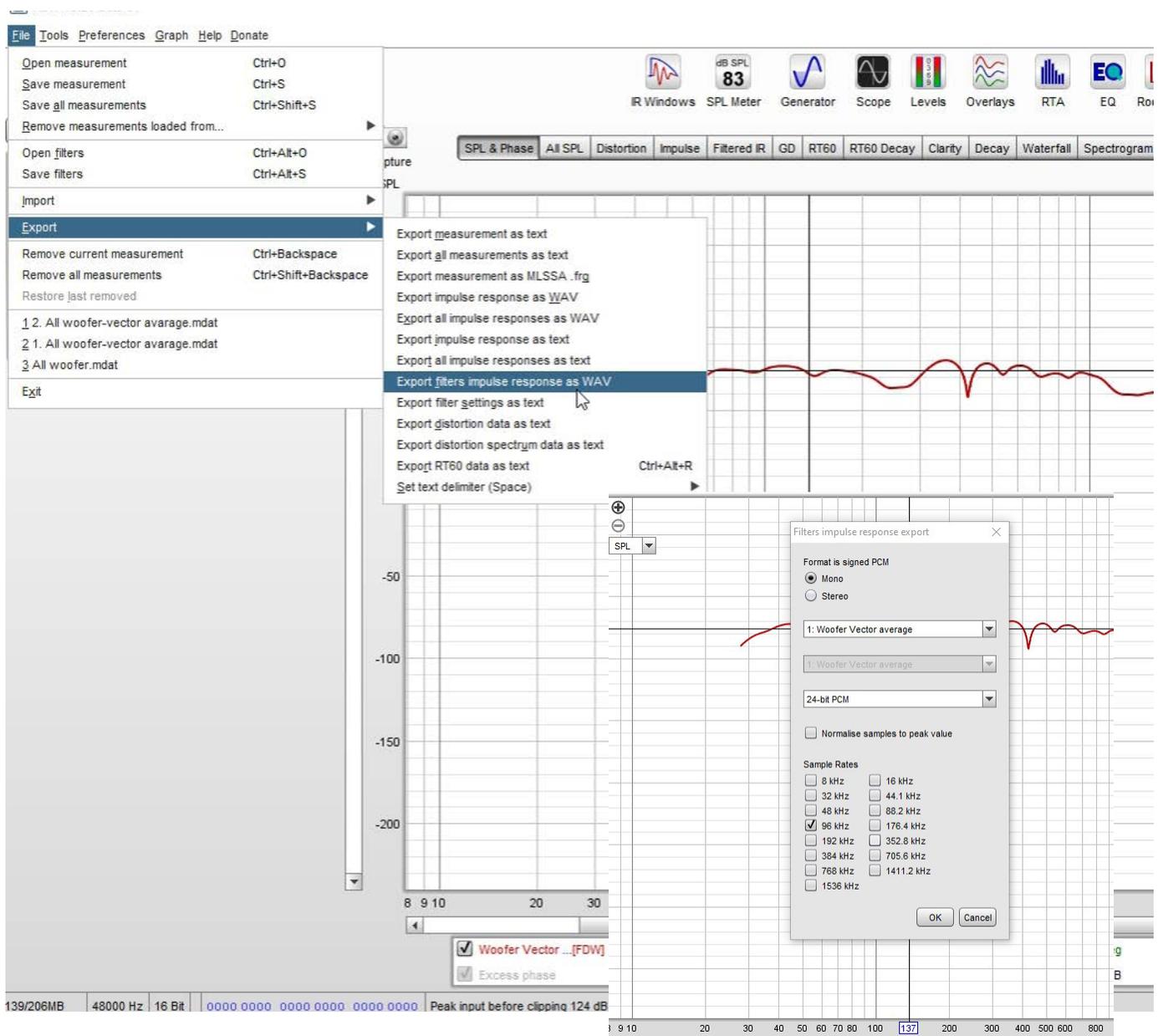


### 4. PEQ generated filter

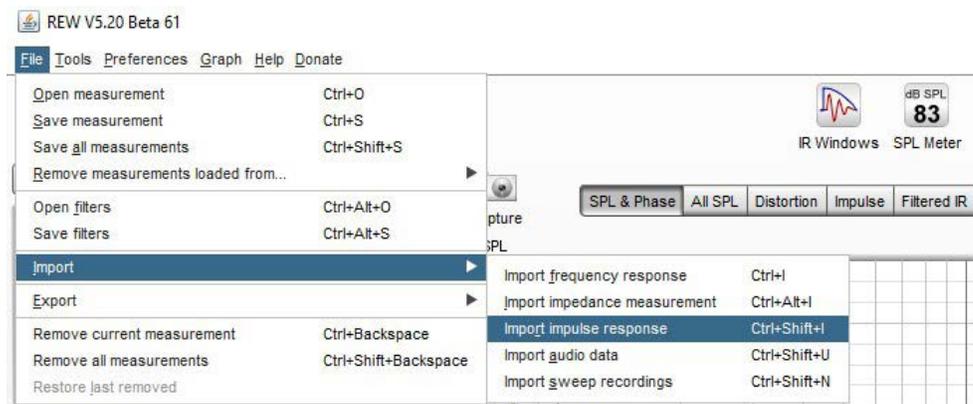
We have ended up with a generated "Vector average " in REW measurement. Next is to make a PEQ filter for "Vector average " measurement. At this point you would like to save the PEQ filter to be used in your DSP.  
Or exported PEQ filter before as an xml for RePhase to create a combined filter -step 6. Combined filter



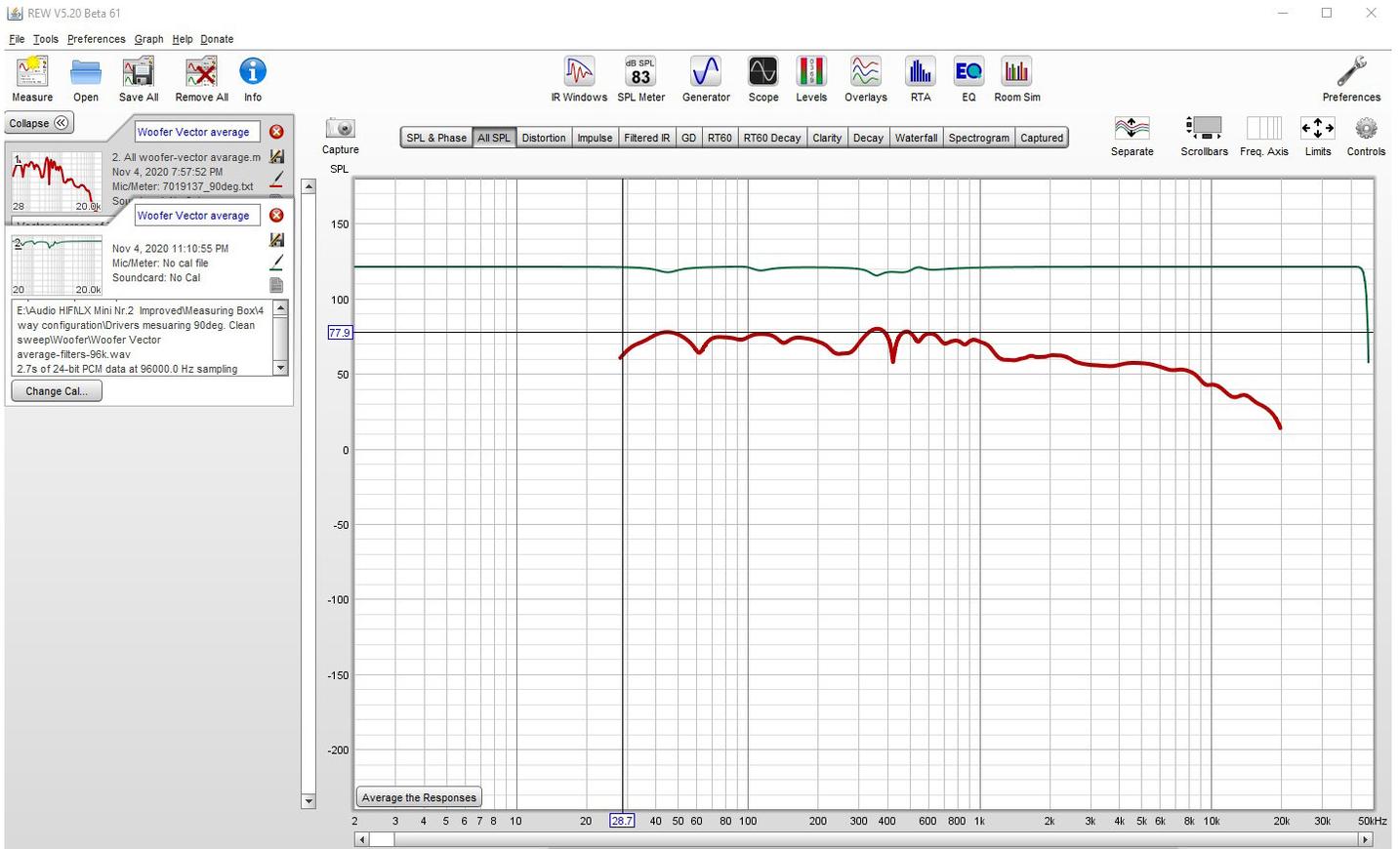
Go back to main REW window and File-> Export -> Export filters impulse response as a wav file and save it .



Import wav file. Main REW window " All SPL". Controls--> File--> Import --> Import impulse response.



Result should be “ Vector average ” and “ Vector average .wav ” measurements with in a main REW window “All SPS” tab activated.



### 5.Trace Arithmetic.

Controls -> Trace arithmetic -> Choose both measurements in windows A and B -> Choose A\*B -> Generate

Decay Clarity Decay Waterfall Spectrogram Captured

Separate Scrollbars Freq. Axis Limits Controls

Trace arithmetic

A: 1: Woofer Vector average

B: 2: W. Vector average wav

A \* B Generate

Measurement actions Trace arithmetic

Alignment tool Trace options

Actions for all selected measurements

No smoothing Apply smoothing

Time align Vector average

Align IR start RMS Average

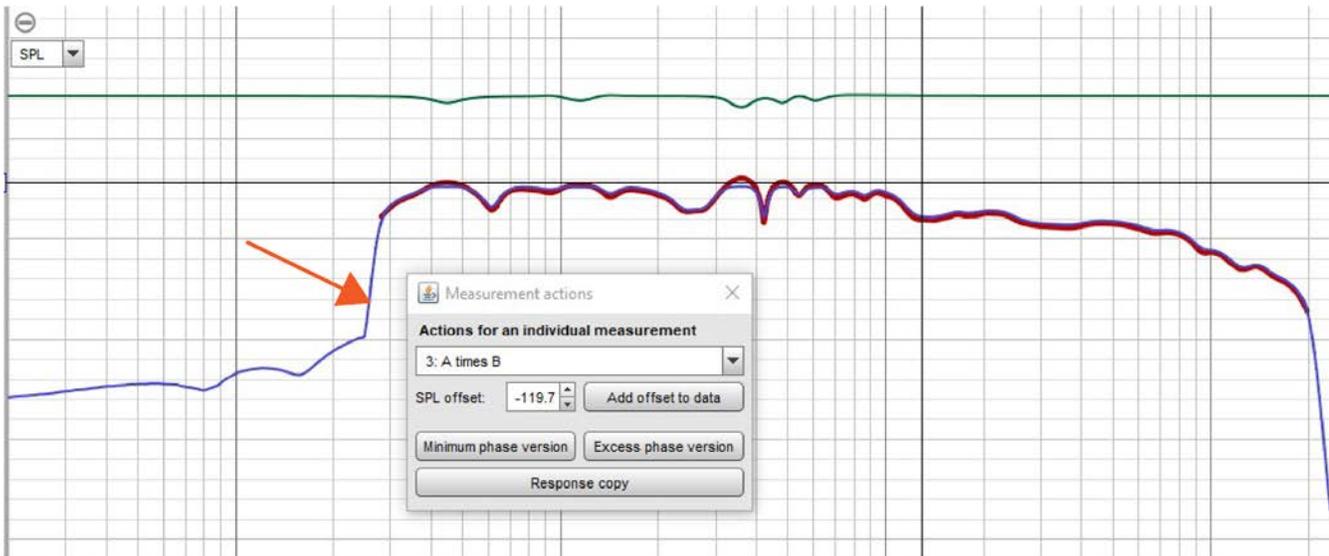
Save to file Export as txt

Export IR WAV Export IR txt

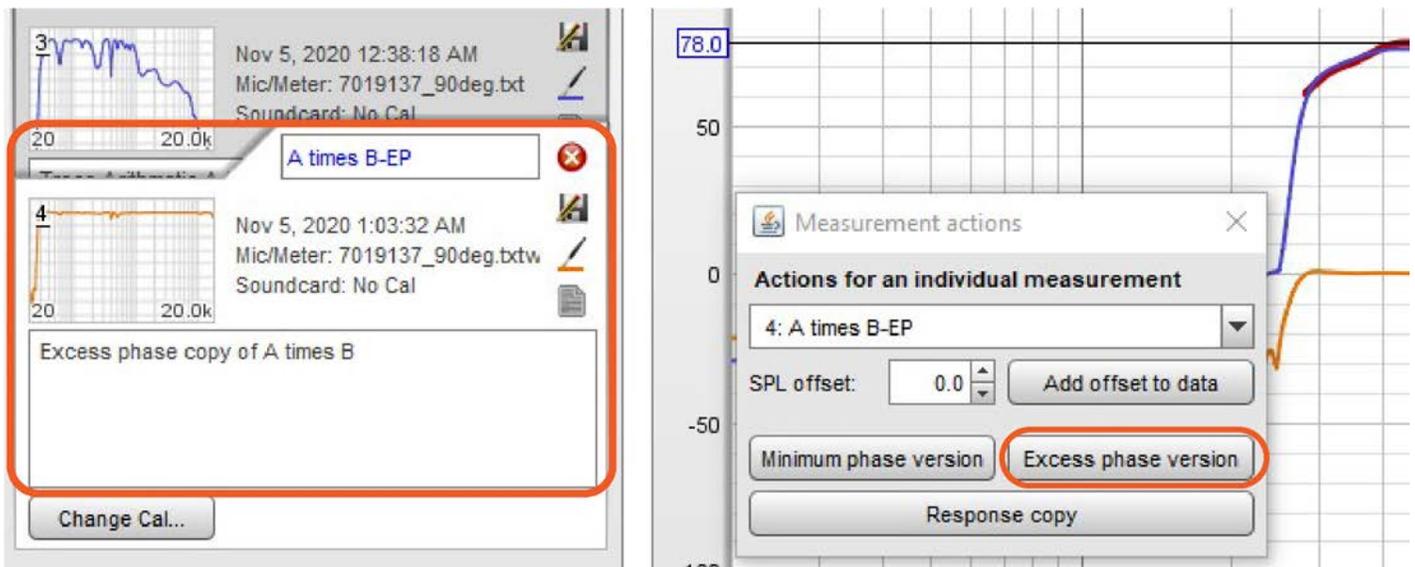
Show points when zoomed in

Controls -> Measurement actions -> A times B -> then enter a negative "SPL offset" to match "Vector average" value -> when you are happy press "Add offset to data".

The ultimate level does not matter for this only relative level so use the same amount for each channel. If the level not be reduced the measurement will end up at 150dB or more.



When press " Excess phase version". That will result to "A times B-EP" measurement .



## 6. Finalization

Main REW window " All SPL"File>Export>Export measurement as text.

Import saved txt file to rePhase. Then from rePhase generate \*.bin file be used in MiniDsp FIR filter.

## 7. Combined filter

Everything the same as in the first part of step **“3.PEQ generated filter”**.

“Vector average ” measurement. Next is to make PEQ filter for “Vector average ”. Choose “Equalizer Rephase” -Filter Tasks-Save filter settings to file. Filtar will be saved as \*.xml file

The screenshot shows the rePhase software interface. On the left, a frequency response graph displays a red curve representing the measurement. A dialog box titled "Select speaker" is open, with "Left Subwoofer" selected in the dropdown menu and "OK" button highlighted. On the right, the "Target Settings" panel is visible, with the "Filter Tasks" section expanded. The "Save filter settings to file" option is highlighted with an orange circle. Other settings in the Filter Tasks panel include Match Range (30 to 800 Hz), Individual Max Boost (6 dB), Overall Max Boost (0 dB), Flatness Target (1 dB), and checkboxes for "Allow narrow filters below 200 Hz" and "Vary max Q above 200 Hz".

Open Rephase import \*.txt file made in step **6.Finalization**. Then : Paragraphic Gain EQ -> Tools-> Import REW filter settings and import saved \*.xml file

The screenshot shows the rePhase software interface. On the left, the "File" menu is open, with "Import Measurement..." highlighted. On the right, the "Paragraphic Gain EQ" settings panel is visible, with the "Tools" dropdown menu open. The "Import REW filter settings..." option is highlighted in the Tools menu. Other options in the Tools menu include "load EQ settings...", "load EQ settings from clipboard", "save EQ settings as...", "save EQ settings to clipboard", "toggle constant Q / proportional Q", "invert", "bypass all", and "activate all".

Finally "Generated" file in RePhase for your device.

The screenshot shows the rePhase 1.4.3 software interface. At the top, there is a phase plot with frequency on the x-axis (10Hz to 20kHz) and phase on the y-axis (-180° to 180°). Below the plot is a control panel with several tabs: General, Filters Linearization, Linear-Phase Filters, Minimum-Phase Filters, Paragraphic Phase EQ, and Paragraphic Gain EQ. The Paragraphic Phase EQ tab is active, showing a grid of 16 filter banks. The Impulse Settings panel is highlighted with a green box, containing the following parameters:

- taps: 1024 samples
- FFT length: 16384 samples
- centering: 0%
- windwing: blackman
- optimization: none to -100 dB
- rate: 96000 Hz
- format: 32 bits IEEE-754 (.bin)
- filename: woofercombinedfilter
- directory: E:\Audio HFiVlX Mini Nr.2 Improv

The Ranges and Measurement panels are also visible on the right side of the interface.

Special thanks "fluid" from diyAudio without this involvement this tutorial was not to be born in to existence.  
Thank you "fluid"