

This tutorial is supplemented by a YouTube video: <https://youtu.be/5YcH7j2-L1Y>

Preface:

- The following procedure is quite different from typical correction methods used in REW. It makes use of the new REW trace arithmetic options some of which did not exist in previous versions. It's been tested and optimized for accuracy, speed, and maximum compatibility.
- The whole procedure is completed in the first 10 pages (the second part is optional) and it's possible to generate system correction impulses with great results in under 30 minutes including the time to take the measurements. However, the whole process is quite prone to human errors. Throughout the guide, double check every option and every produced result for each REW operation before you move on to the next step. Try and follow the instructions in full, you are advised to not attempt to short-cut or change them unless you really know what you are doing. This includes, for instance not changing the order of impulses in trace arithmetic vector operations (fast Fourier transforms are not necessarily commutative).
- Although all steps are explained in detail and no previous REW experience is necessary, some familiarity with REW menus would be helpful. There are many well written manuals on REW including the author, John Mulcahy's excellent own tutorial and it's not an objective of this guide to explain how to use REW.

Best practices before making measurements:

- Move the mic and the stand near each speaker and adjust the mic with stand height so that the top of the mic is exactly at each speaker's tweeter midpoints. Then move the mic and the stand as is to the listening position:



- Make sure the mic is pointing perfectly upwards at the LP and you have the 90 degrees calibration file for the mic:



- Verify to your best visual capacity that the mic is positioned at exactly the middle of and the same distance from both speakers at the LP. I find standing as far back as possible from directly behind the microphone and checking that the tip of the mic and a fixed midpoint precisely between the left and the right speakers to be on the same straight line, to work quite well. Alternatively, if you're obsessive, you can download Acourate software free trial version and use its microphone alignment tool!
<https://www.audiovero.de/AcourateTrial/AcourateTrialSetup.exe>
- All measurements will be taken at a SINGLE perfectly centred position (LP). Latest research confirms that's better than using multiple measurement points even for the wider listening area as the filters created will work by the same gain ratios to SPL throughout the frequency band for all locations surrounding LP. Spatial averaging using multiple measurement points will only work if based on impulse alignment using cross-correlation ~~which is not possible with REW~~ (*the latest versions of REW can do cross correlation alignment, so feel free to take multiple measurements and vector average them after alignment if you so like – see updates at the end of this document for other recent revisions in REW). Please note that despite using multiple point measurements, neither Audyssey nor Dirac Live uses a cross correlation method for impulse alignment.
- Playback and recording device should both be set to 48kHz, 16 bits if possible as REW internally works with these settings. Some devices come with either 24-bit or 32-bit settings lately (like Minidsp Umik-2 microphone) and you will have to use them at either of these sample depths but sample rate synchronization at 48kHz among all devices is essential for accurate impulse calculations. Using 48kHz sampling rate also necessitates measurements taken to be in the full range of 0 Hz - 24,000 Hz (Nyquist limit is one-half the sampling rate) for accurate vector calculations.

Required REW preferences:

- Download latest REW version (V5.20.11 - **early access version** as of Aug 2022 is used in this guide):
https://roomeqwizard.com/installers/REW_windows-x64_5_20_11.exe
 If you already have this or a later version installed in your computer, from **Preferences** tab in main menu "Delete preferences and shut down REW" first and reopen REW to change all settings to default. Afterwards, you will ONLY need to change the settings mentioned in this tutorial.
- **Soundcard tab:** Sample Rate (48 kHz), Drivers: ASIO, ASIO Device: ASIO4ALL (you will need to install ASIO4ALL if you don't already have it installed in your computer):
https://www.asio4all.org/downloads_11/ASIO4ALL_2_15_English.exe

Click "ASIO Control Panel" button in REW, check that only the mic and your actual playback device show ticks to their left, untick and tick devices as necessary in Asio control panel window and then click REW "Reload" button. Repeat this process until correct devices show up on REW Soundcard Output and Input boxes.

Optional: If you fancy instantly and continuously testing your correction results with "correction convolved" new REW measurements, you may install a free PC convolve engine: "Equalizer APO"

(<https://equalizerapo.com/download.html>). However, this program is for PC only and you will then need to use "Java" drivers instead of "Asio4all" in REW. Java drivers will work just fine unless your computer is connected to your amplifier through HDMI but the timing accuracy will be a bit compromised especially while EQ APO is active in parallel. It tends to randomly change system delay value for the same configuration thus you are advised to do timing alignments without it and use it for testing the final frequency responses of your correction files.

- When asked for mic calibration file, untick "Separate cal file for each input" and upload 90 degrees calibration file for your mic.
- Output: your left channel, Timing Reference Output: your right channel
- Change Levels to "Use main speaker test signal to check/set levels"
- **House Curve tab:** Download "Harman Target Curve.txt" from the link below and upload in REW:

https://drive.google.com/file/d/1kfTm89YaYik7kNujyY3bM_LU9UXJzsO/view?usp=sharing

- **Analysis tab:** Untick "Decimate IR", tick "Adjust clock with acoustic ref" (very important especially if you are using a USB mic as they ALL have clocking/timing issues and its severity is proportional to the length of the sweep)

Measurement:

- Settings: Length: 4M - 87.4 s (the longer the sweep, the higher the measurement's "signal to noise" ratio), Timing: "Use acoustic timing reference", End Freq: 24,000 Hz
- Optional: Ref level trim: 2.0 dB (we need to give an offset to the chirp to be able check your left and right speaker distances to the LP in the impulse graph)
- Click "Check levels" button. **The volume of your amplifier should be at the level which you most frequently listen to your music.** This is important for proper bass response adjustment. We tend to perceive relative bass volume differently for different overall volume levels.
- Be silent and measure left and right speaker separately with right speaker kept as acoustic reference for both measurements. This will take around 3 minutes in total.

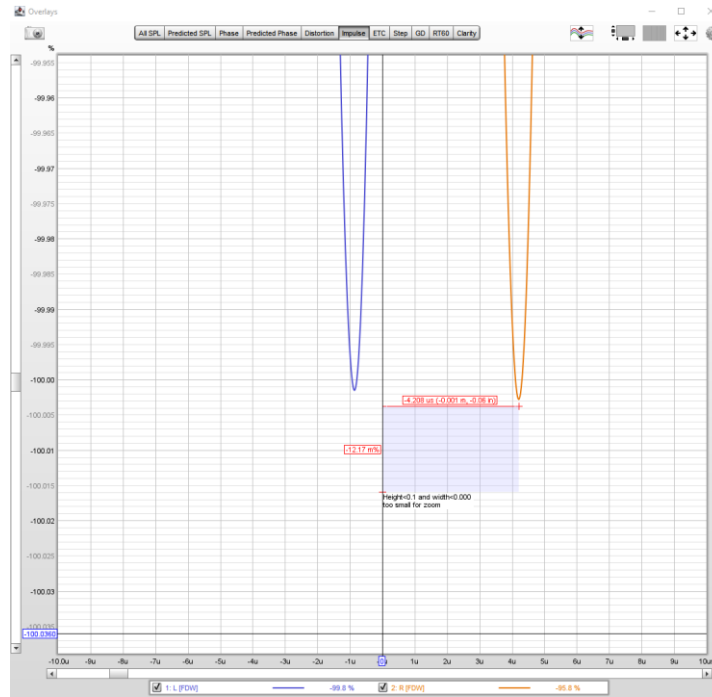
Correction Process:

1. Name the left speaker measurement "L" and the right speaker measurement "R"
2. All SPL / Limits / Fit to data
3. "Save All" to a folder you dedicated to this correction and name it "Step 0"
4. Open "IR Windows" and move it to the side of REW window on your screen. Do the same for Controls / "Trace arithmetic" and "Measurement actions" windows. This is only for convenience as we will use these windows frequently. Note that "IR Windows" will always stay on screen unlike the other two windows which will only show up when you switch to "All SPL" tab. Tick "Add frequency dependent window", Width in cycles:35, click "Apply Windows to All, Keep ref Time"
5. L / MA / Response copy, repeat for R (make copies of each measurement)
6. With only the copies ticked in ALL SPL / Controls / Align SPL / "Use average SPL of measurements" ticked, Alignment centre: 315 Hz, Alignment span: 4 octaves (this will equalize the volumes of L & R speaker with each other from 78.75Hz Hz to 1260 Hz covering all locatable bass and the entire male/female vocals)
7. With only "L" & "L-copy" ticked, select "L" in MA, click SPL offset arrows until "L" sits perfectly on "L-copy", take a note of the SPL offset level for this speaker, "Add offset to data", repeat for R
8. Delete L-copy & R-copy.
9. As it's possible that correction filters might change overall volumes of speakers differently, you will need to repeat step 6 to 8 one more time at the very end with your final responses. You will need to adjust speaker gains in your DSP engine by the sum of these values. For example, if you had to decrease the volume of your right speaker 0.2 dB at this step and another 0.1dB at the very end to align L & R speaker SPLs, then you should add a total of +0.3 dB gain to your right speaker in your DSP set up independent of your convolution. In Roon, you can do this under "Speaker Set-Up" DSP sub menu.
10. Overlays / Impulse, change vertical axis from dBFS to %, magnify impulse response until you can only just see the acoustic timing signal and the speaker response peaks on same screen (you can select a region with CTRL + mouse right click and then click inside the rectangle to zoom in a region or you can use axis "+" and "-" buttons

or both). Starting from the tip of the acoustic ref peak on the left, use CTRL + right mouse click method to draw a rectangle to the peak of the speakers:

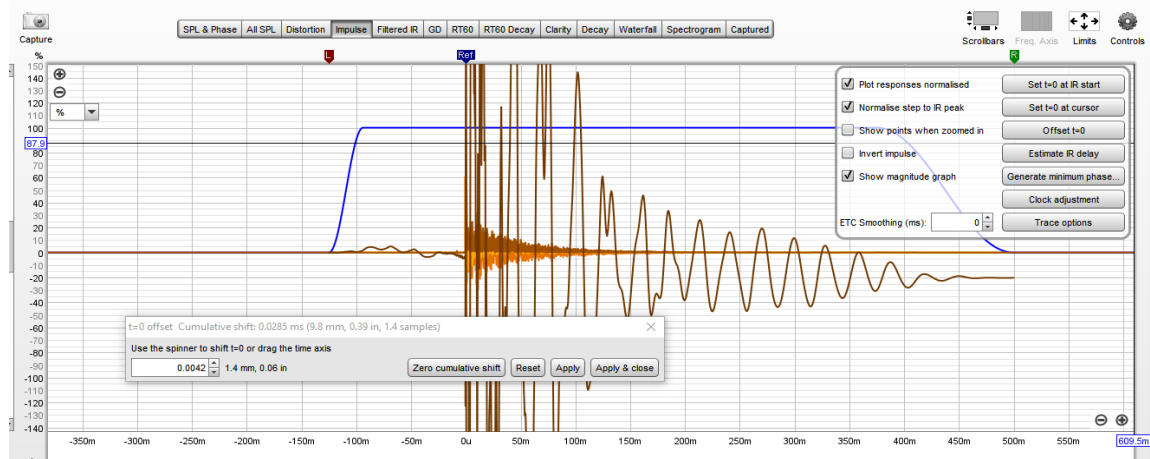


11. You can now read the distance of your speakers to the LP. They are both around 280 cm away from the LP in my case above - my impulse peaks are inverted and point down, yours may well be upward. This is just a ballpark figure. Left and right speaker impulses will almost never be exactly on top of each other and finding out the precise difference is important. So, we boldly continue magnifying the impulse peaks using both axis buttons until we no longer can and both tips are still on the same screen and then measure the distance of each tip to time 0:

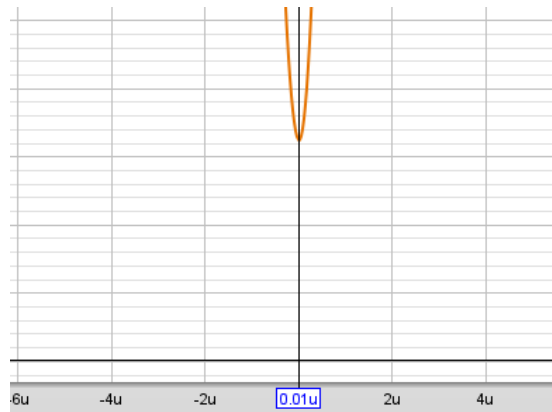


The impulse tip on the right belongs to the speaker which is too close to LP by the amount you read on the screen. My right speaker (orange) needs to be moved back by 1 mm and my left speaker (blue) needs to move towards the LP a little bit as well. The distances you will find will probably be much higher as my speakers were already precisely adjusted before. Going forward, we will compensate for these differences in our calculations, and we will not need to retake measurements after moving the speakers. However, at some point, you will need to move your speakers according to these findings (while keeping its toe-in angle relative to LP unchanged) for your digital corrections to work as intended. You can also adjust your DSP engine delay settings instead of moving your speakers.

12. Adjust Overlays and REW windows so that they are simultaneously visible to you on your monitor(s)
13. REW window / Measurement R selected / Impulse / Controls / Offset t=0, use the spinner buttons (or type values in milliseconds inside the box and press keyboard tab key to see its effect and then click apply to make extremely accurate adjustments - this tool also saves the total cumulative shift applied thus it's possible to revert them back later if necessary) until the impulse peak tip is exactly on time 0 in Overlays window. For my right speaker, I entered the time difference I measured above in "us" (1/1000th of a millisecond) directly in the time offset box:



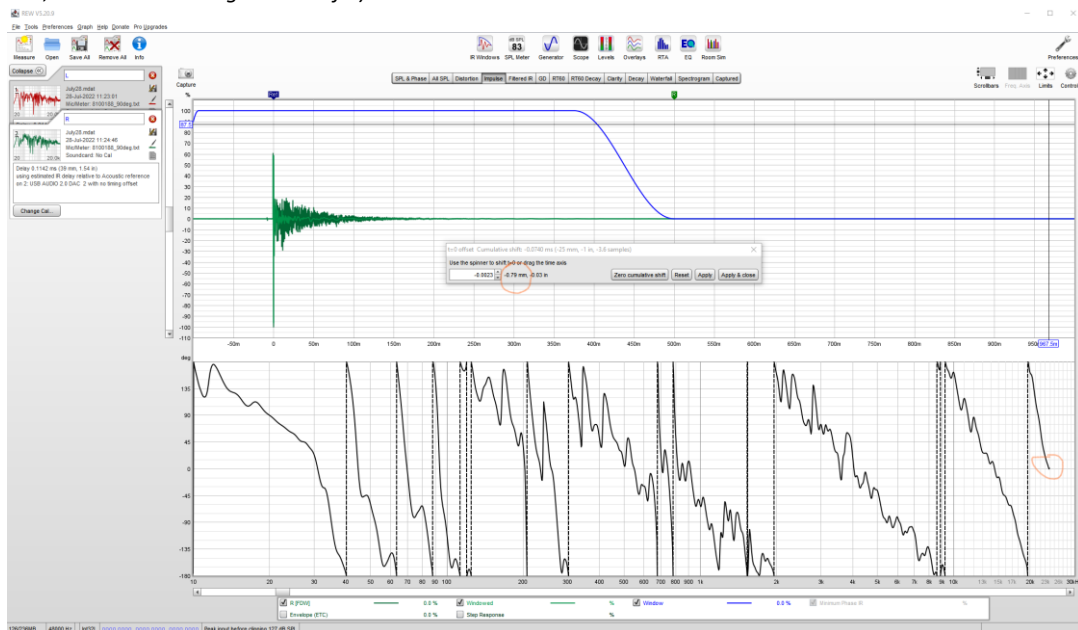
and when applied:



Repeat the same procedure for both speakers until their impulse peak tips are both on time 0. The total distances you entered in this process are the distances you should move your speakers (I need to move my right speaker 1.4mm away from LP).

Optional:

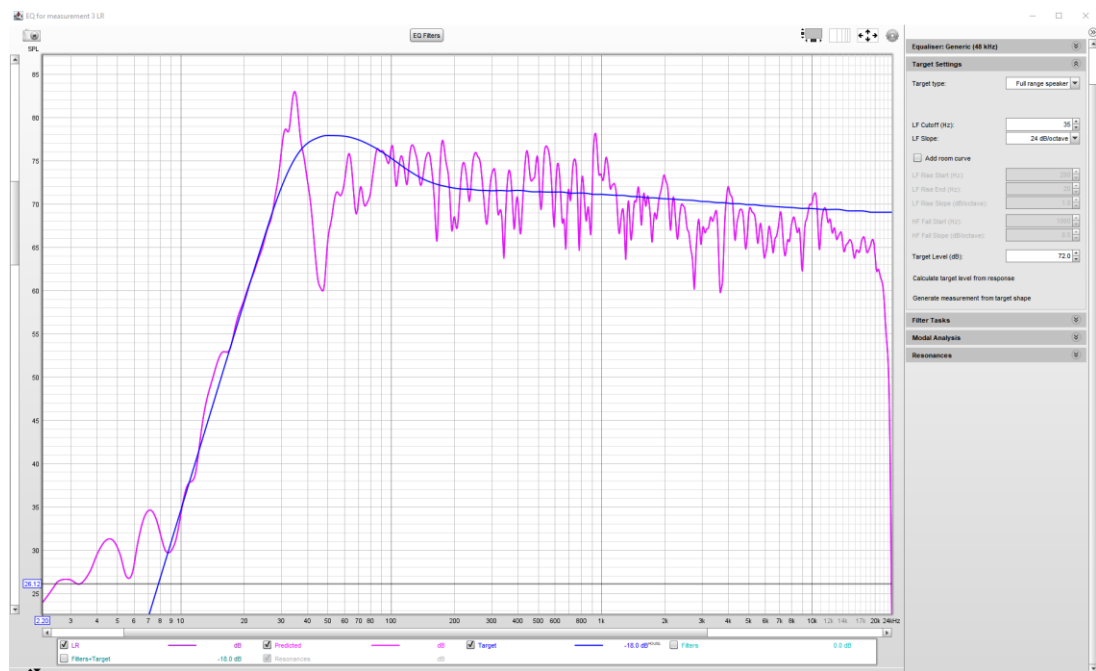
While still in the "Impulse" tab, for each speaker, adjust the right end tip of the phase graph (the graph that automatically opens in the lower window) with spinner buttons until the tip lands on 0°. Do NOT click "apply". Keep a note of the extra change you needed to dial in distance (mm/in) terms for each speaker. As seen below, my right speaker turns out to be -0.79 mm off axis, meaning it needs to be **rotated** on its acoustic centre clockwise so that the centre of its tweeter will move 0.79 mm towards the LP (for the right speaker with a negative offset value, this means moving "to the left"):



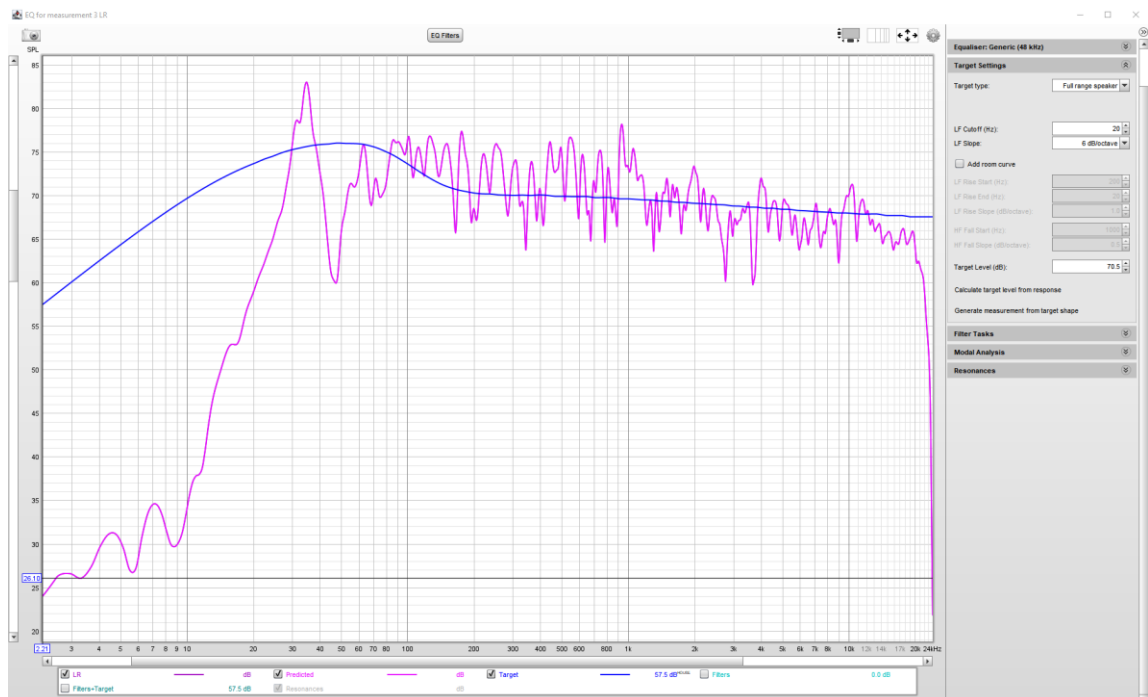
You can equalize toe-in degrees of both speakers very accurately with this method. It does not intend to determine the ideal amount of toe-in for your speakers, but it will ensure both speakers are toed-in equally towards the LP. Again, we will ignore these differences during this correction process but at some point, you should adjust your speakers' toe-ins according to these findings. Remember that rotating a speaker on its acoustic centre is different to moving a speaker back and forth on an axis towards the LP. Also be advised that a speaker's acoustic centre is NOT the woofer centre as commonly known but rather half a woofer diameter in front of it and this is the axis centre the speaker should be rotated around to keep the distance from LP unchanged (you should also use this acoustic centre for measuring distances from the front wall while you are initially placing your speakers)

14. Now that we have time and volume aligned L & R speakers, we can start to produce the correction filters. Save All / name: Step 1
15. All SPL, both speakers ticked, Controls / "RMS Average" / name: LR
16. With "LR" selected, EQ, close "Waterfall" window for better visuals, Limits / "Fit to data", Target Type: Full range speaker, LF Cutoff: "your speakers expected bass cutoff", LF Slope: "your speakers expected LF slope", click

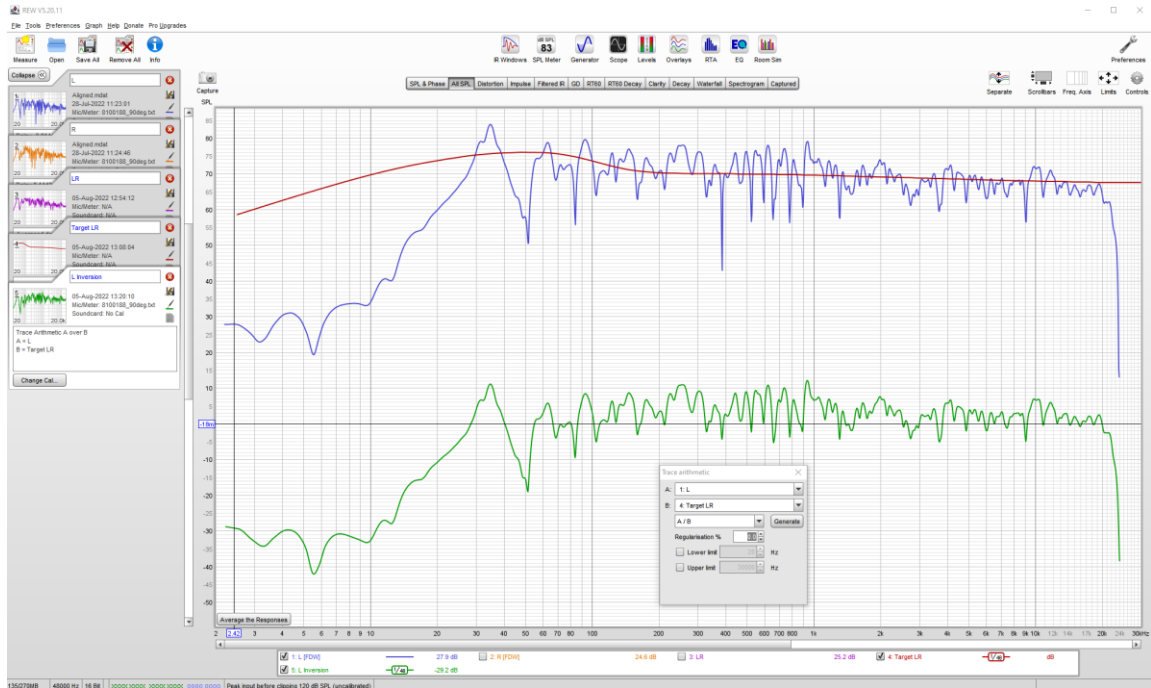
“Calculate target level from response”, change LF Cutoff and LF slope with spinner buttons until your response matches with the house curve like so:



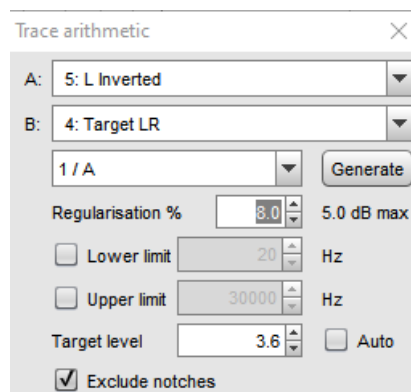
At this stage, you can keep the target level as calculated by REW or you can lower the target level a bit if you want to decrease the size of the dips. Our method corrects for the dips to some extent, and you should first go with REW’s calculated target level and see how you like the results but since I have quite massive dips (the speakers are kind of too large for my room), I want to utilize every bit of bass my speakers can produce for this once and mostly to illustrate an alternative, I have amended my target curve as below:



17. Click “Generate measurement from target shape”, close EQ (believe it or not, we are all done with EQ)
18. TA / A: “L” / B: “Target LR” / “A/B” / Regularisation %: 0.0 / all unticked, name it “L inversion”, Repeat for R:



19. TA / A: "L Inversion" / "1/A" / "Regularisation %: 8.0" / "Target level: 3.6 (Auto – see updates at the end)", "Exclude notches" ticked, name: "L Inverted", repeat for R

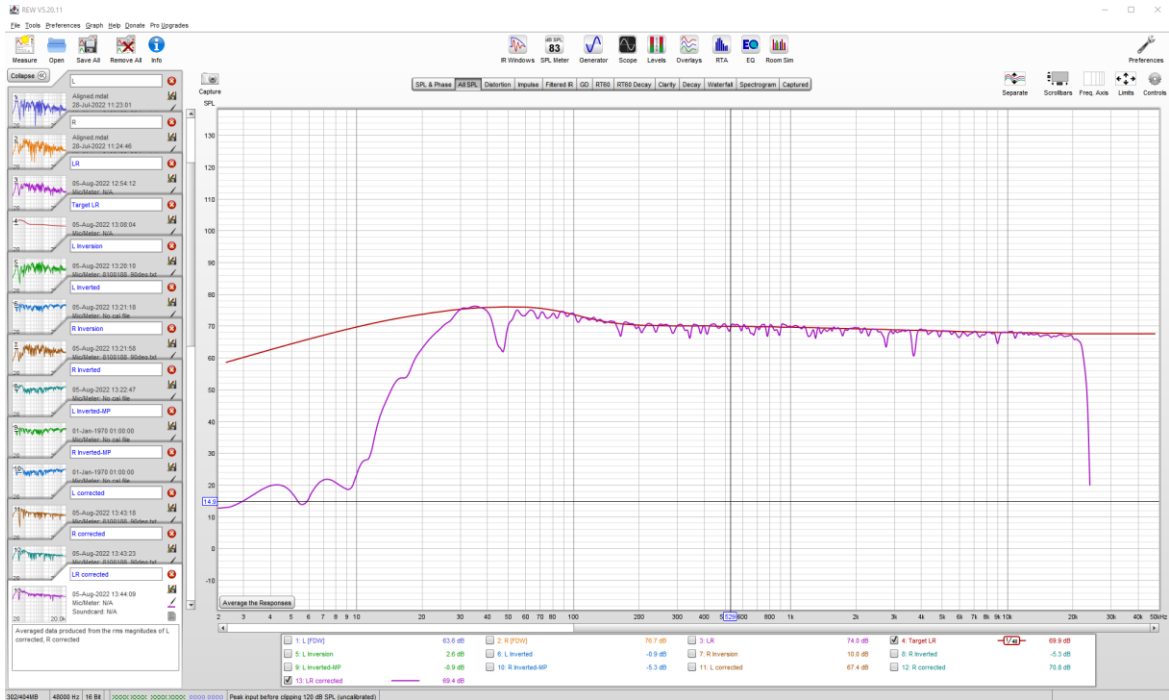


Feel free to lower regularisation percentage if you think you need larger than 5 dB corrections throughout the frequency band but 8% should be sufficient for most rooms.

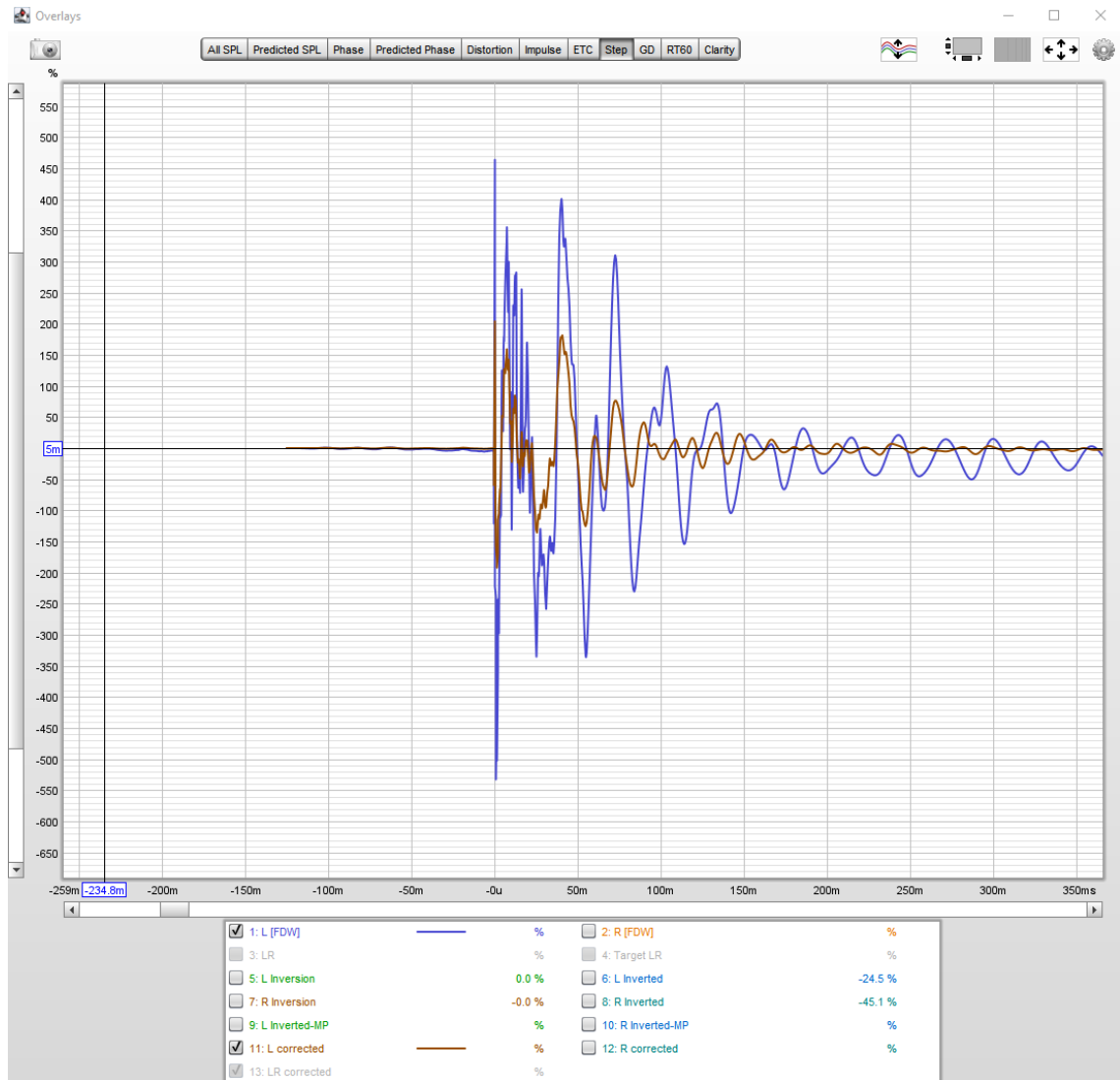
20. MA / Select "L Inverted" / "Minimum phase version" / Everything unticked / "Make min phase copy" / "IR Windows" / "Left Window: 125 ms" / "Right Window: 1,000 ms", change Left window type to "Tukey 0.25" from "Rectangle", no other change, "Apply Windows", repeat for R
21. Finished!

"L Inverted-MP" and "R Inverted-MP" are your correction impulses for each speaker and are now ready to be exported and convolved in your DSP engine. If you want to see the results:

- TA / A: "L" / B: "L Inverted-MP", "A x B", name: "L corrected", repeat for R
- ALL SPL / With only "L corrected" & "R corrected" ticked, "RMS Average", name it "LR corrected"
- Untick "L corrected" & "R corrected", tick "Target LR" and compare your overall system response with the target curve:



- d. Check under Overlays for: ETC, Step, Group Delay and Clarity. You will see improvements in all of them. For instance, despite quite major bass response correction all the way down to 20 Hz, there is zero pre-ringing in the Step response:



22. Finally, select “L Inverted-MP” / File / Export / “Export impulse response as WAV” and save the convolution “.wav” file to be used in your DSP engine. I suggest using the settings below and create a single stereo impulse containing correction for both left and right speakers, but you may use separate impulses for each speaker as well depending on your DSP engine options:

Measurement impulse response export

Format: 32-bit Float

Mono
 Stereo

Left channel: 7: L Inverted-MP

Right channel: 8: R Inverted-MP

Export measured IR
 Export EQ filtered IR
 Export Min Phase Version of IR

Normalise samples to peak value
 Apply IR Window before Export

Sample rates (native rate: 48 kHz)

8 kHz 16 kHz
 32 kHz 44.1 kHz
 48 kHz 88.2 kHz
 96 kHz 176.4 kHz
 192 kHz 352.8 kHz
 384 kHz 705.6 kHz
 768 kHz 1411.2 kHz
 1536 kHz

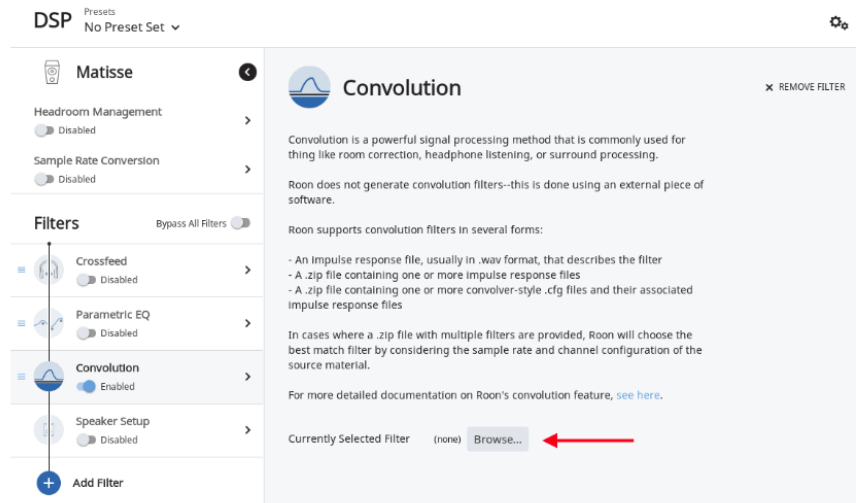
OK Cancel

For Roon, I zip the two (one for 44100 Hz and one for 48000 Hz) stereo “.wav” files together and direct Roon’s convolution engine to that zip file:

Name	Date modified	Type	Size
L Inverted-MP_R Inverted-MP-44k.wav	05/08/2022 14:01	Media Center file	1,882 KB
L Inverted-MP_R Inverted-MP-48k.wav	05/08/2022 14:01	Media Center file	2,049 KB
Roon.zip	05/08/2022 14:01	Compressed (zipp...	1,760 KB

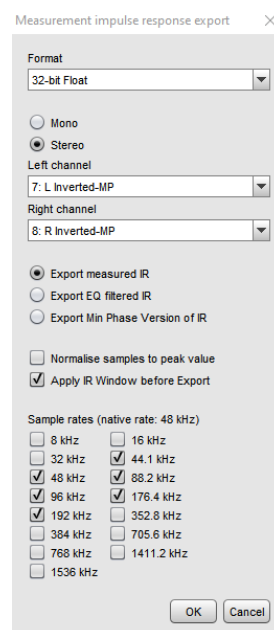
Loading a filter into Roon

Convolution filters are loaded using the Browse button:



You can use the distance and volume differences you have calculated during this process in the “Speaker Setup” sub menu (seen disabled in the above picture right below Convolution) in Roon if you don’t want to move your speakers and/or change the amplifier balance settings.

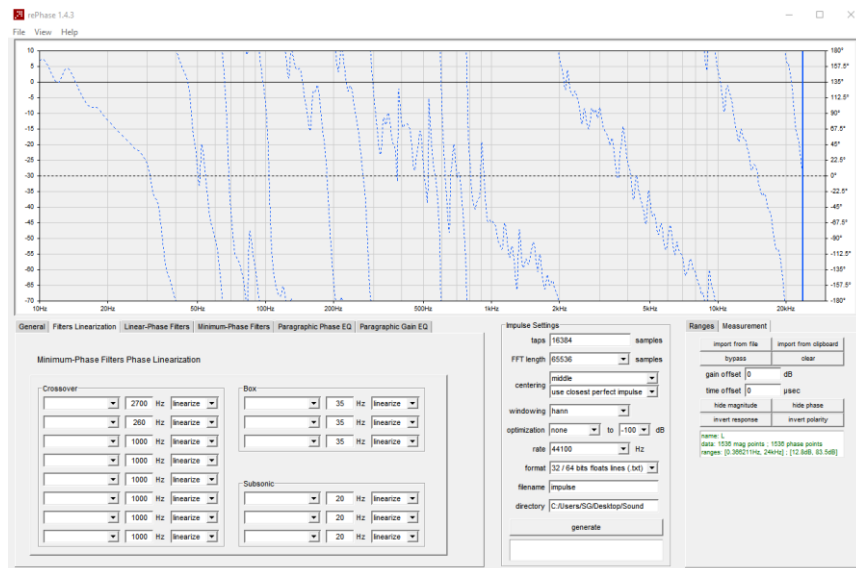
I prefer to leave the sample rate upscaling of these filters to Roon (“Sample Rate Conversion” sub menu above) as its upscaling engine is state of the art, but the latest REW also does quite a decent job if you want to create higher sample rate filters:



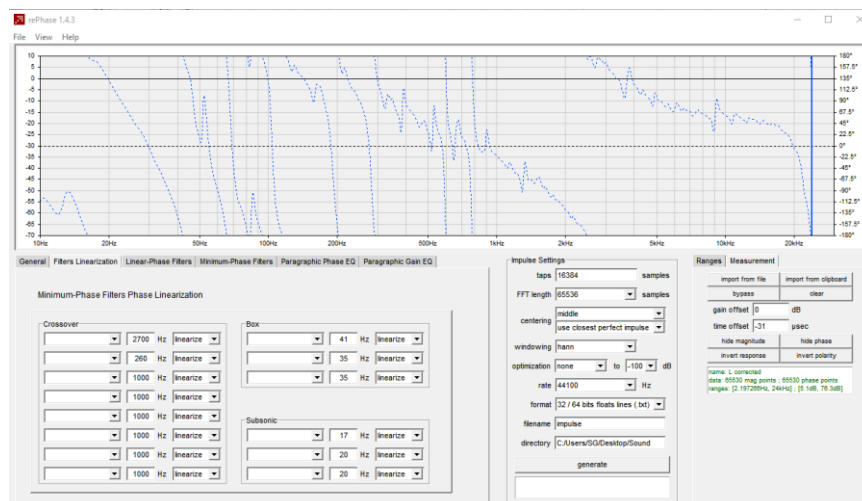
23. Save ALL, name: “Step 2”

Phase correction (optional):

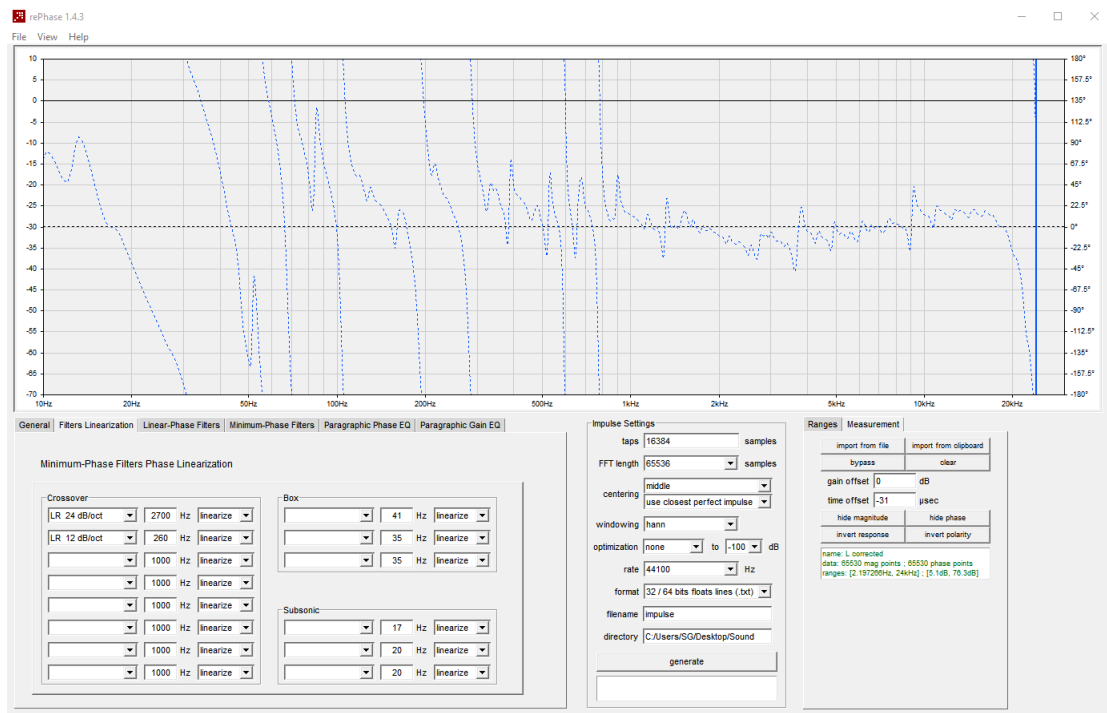
24. You may have already observed that the correction filters we produced, despite being minimum phase, have smoothed out the phase fluctuations in a positive way. But almost all speakers cause unintended phase shifts at their crossover frequencies or due to their enclosures in a room. Despite all the debate going on, it's now widely accepted that phase shifts outside 100Hz - 800 Hz range are undetectable to our ears and the phase value on its own will not affect the sound pressure (SPL) of a speaker at a certain frequency. However, phase differences between speakers will affect total SPL of the combined (stereo) sound as they can cause cancellations between speakers if phases are opposite of each other. Plus, as mentioned, there's the "100Hz - 800Hz" area which we can detect phase shifts as delays. Thus, phase response needs to be dealt with for optimum room correction and we will use a great, free tool; rePhase (<https://rephase.org/download/rePhase%201.4.3.zip>) for this purpose.
25. Firstly, try to find out your speakers' bass response lower limit, crossover frequencies, port frequencies (if they are ported speakers) and optionally filter order levels. You can use their owner's manuals or search online for detailed tests of your speakers.
26. Select "L corrected", File / Export / Export measurement as text, File Name: "L", save it to your correction folder
27. Open rePhase, "Measurement" tab / "Import from file" / browse to and double click "L.txt". You should see a phase graph.
28. Click into "time offset" box and move the right tip of the phase with keyboard up/down keys until it ends on 0-degree axis at 20 kHz and 5kHz-10kHz region is relatively flat. You may need to roll over the phase tip (keep rotating it beyond -180 degrees to cycle back on the top of the screen at 180 degrees or vice versa) a couple of times over the 0 axis until that happens. In my own case, phase tip before:



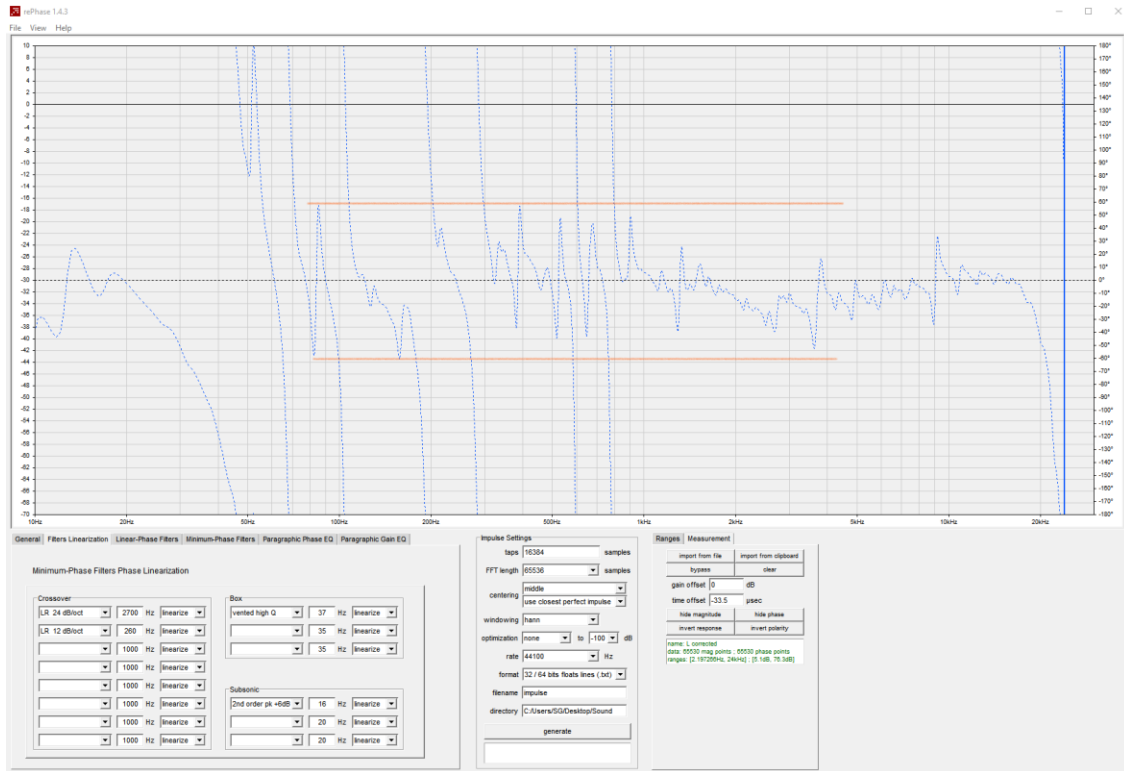
and after:



- 27 Select “Filters Linearization” tab / in the top-middle box of the “Crossover” area, enter your speaker’s highest crossover frequency / if you know your crossover filter order select the correct slope from the drop down menu on its left (6 dB/octave per filter order – i.e. “LR 24 dB/octave for 4th order XO filter), if you don’t know the filter order, then select “LR 12 dB/octave” from the drop down menu, check if the phase graph is intersecting 0-degree axis at that frequency. If not, try “LR 24 dB/octave” and then try “LR 36db/octave” and so on. It will start moving further away from 0 axis at some point and it’s extremely rare having to go beyond “LR 48 dB/octave” unless you have some sort of DIY speakers. If none of the options bring phase at that XO frequency precisely on the zero axis, then select the slope that brings phase the closest to 0 degrees at the XO frequency. Tip: Clicking inside the frequency box and moving it up and down once with the keyboard arrow keys will bring back the “regularly disappearing” yellow frequency line back to the screen so that you can see where exactly you are correcting.
- 28 If you have a two-way speaker, you are done. If you have 3-way or 4-way speakers and thus more than one crossover, continue in the box below the first one with the second highest XO frequency and repeat the same procedure. Here is how I end up with my 3-way speakers with two crossovers at 2700 Hz and 260 Hz:



- 29 You can see that the phase curve is now reasonably flat but the important area in the 100Hz – 800 Hz range is still in need of improvement, and we’ll use other rePhase “Filters linearization” tools for that. These filters are quite special minimum phase types (LR stands for Linkwitz–Riley) and they will not cause pre or post ringing. Any other phase correction you do below 500 Hz i.e., using the “Paraphrastic Phase EQ”, will cause inversely proportionally increasing pre-ringing with the frequency unless it has a very low Q (below 1.0) and a maximum of 45 degrees rotation. A very low Q, by definition, will shift the whole phase curve so it cannot be used to adjust regional phase problems. Additionally, phase anomalies in the deep bass region are usually far worse than 45 degrees. Apparently, we need to limit our phase correction within this tab. The second linearization filter we can use is the “Box” filter. Enter your port frequency here and select low, standard, or high Q depending on where your speaker port is located (at the back or front of the speaker). For sealed box enclosures, experiment with the “closed Q” varieties. The third type of filter is the “Subsonic” filter and it’s originally intended only for Vinyl playback (as this state-of-the-art, analog system produces very loud, deep low end bass frequencies while the sharp cartridge tip chafes on the rotating vinyl’s surface to read the audio). But sometimes it can help improve phase response in the bass region in some way so it should be tinkered with (common use is between 10-20 Hz). Check my result achieved through ONLY “Filters Linearization” tools. Minimizing the distances of the most extreme fluctuations from axis zero was the priority:



Note that the “time offset” value had to be readjusted at the end after all other linearization tools were completed. Also note that in the impulse response it produces, **rePhase does not take into consideration ANY adjustment done on right most window** and that includes the timing offset so, this is only for visual purposes

- 30 Now we need to export our aggregate phase adjustments as one impulse file to REW. For “Impulse Settings”, you can use as many taps as you want but too many taps will induce a delay. I use the default 16384 taps with the slightly more efficient rectangular windowing, energy centring and minimal optimization. It’s short and accurate enough for most purposes. Here are the settings to use for generating our phase linearization impulse:

Impulse Settings

taps samples

FFT length samples

centering

windowing

optimization to dB

rate Hz

format

filename

directory

impulse delay: 10039.89 samples, 209.164 ms
max response: 0 dB, max impulse: -4.34 dB

- 31 Select your correction directory and “generate”, save rePhase settings also to the same folder in case you wish to return and change some settings later, close rePhase

- 32 Pick with your mouse "PhaseFix.wav" from your correction directory and drop "onto" REW (or use File / "Import impulse response" or keyboard "CTRL + Shift + I")
- 33 Select "PhaseFix.wav", MA, SPL offset: -117 dB, "Add offset to data". You should now see an almost flat line on 0 dB in "All SPL" tab. If it's not exactly at 0 dB, try adding another -7 dB offset (rePhase generated impulses are offset by either 117 dB or 124 dB and it does not affect the calculations)
- 34 Rename "Phasefix" to "L phase fix", MA / "Response copy", name: "R phase fix"
- 35 TA / A: "L corrected" / B: "L phase fix" / "A x B", name: "L final"
- 36 Impulse tab / "Offset t=0", flatten right end of the phase response (the graph in the automatically opened window below) using spinner (illustrated in step 13 above), highlight and copy the value in the spinner box (keyboard CTRL + c), no need to "Apply"
- 37 Now select "L phase fix" while impulse tab is still open and paste in the spinner box the same value you found in step 36 (keyboard CTRL + v) / "Apply" / delete "L final"
- 38 Repeat steps 35-37 with "R corrected" and "R phase fix"
- 39 All SPL tab / TA / A: "L Inverted-MP" / B: "L phase fix" / A x B, name: "L pulse", repeat for "R Inverted-MP"
- 40 Save All, name: "Completed"
- 41 We are finished. "L pulse" and "R pulse" are final convolution files ready to be used in the DSP engine. Export them as explained in step 22.
- 42 You can produce your end results with: TA / A: "L" / B: "L Pulse" / "A x B", name: "L final", offset its SPL by -117 dB, do the same for R and compare their phase graphs ("L pulse" vs "L corrected" for example) in Overlays. You can also "Vector Average" "L final" and "R final" to see the final system response including the combined phase effects or you can simply start listening to your system with these corrections convolved in!

In sum, we have:

- determined time and volume alignment offsets for each speaker and either moved the speakers and/or adjusted gains accordingly in our DSP engine
- inverted the average room response over a target curve and created a minimum phase version of that for each speaker (think of unlimited number of perfectly shaped IIR filters)
- corrected all through the audible frequency band, way above Schroeder's frequency (room transition frequency usually around 150 Hz) and achieved this with no degradation in sound as is the case with majority of room correction attempts
- have taken care of the crossover phase shifts, box enclosure related phase shifts and timing related phase shifts by using FIR filters and embedded them into our correction.

This is enough to see substantial improvements in your system's response at the "listening position" in terms of spectral balance, smoothness, clarity, and attack.

Finally, here is a list of the test songs I use to check my corrections:

Overall low-bass response and pre-ringing:

Sweet Jezebel – Turboweekend: You will not need to wait for long with this song if there's any kind of pre-ringing in the bass region and you will not doubt it because it will repeat until you stop the song and go back to your correction files. Your system might have pre-ringing and it's possible that you don't even know about it because you have never heard it. That's until you test it with this song. One of the hardest songs to get to sound right I know of and it's my go to song for the ultimate bass calibration. And don't worry too much if you cannot get it right, many bass deep songs might still play fine.

Post-ringing:

Bubbles – Yosi Horikawa: Great system show-off piece, excellent panning effects throughout and post-ringing will come out the moment the marbles start bouncing.

Speaker toe-in:

Her Majesty – Beatles: An only 23 seconds long song. It helps to adjust the initial toe-in as well as the distance between the speakers. The vocals and guitar start off in one channel and slowly pan all the way to the other channel. If the vocalist gets closer towards the LP as it moves to the middle, there is too much toe in. If the vocalist moves away from the LP during the panning, there is too little toe in. It should sound as if McCartney's vocals roll smoothly and continuously across the sound stage. If it sounds like he jumps from Right - to - Center - to - Left, the speakers are too far apart.

Amused to Death – Roger Waters: Once speakers are roughly toe-in, I use this one for optimization. The song starts with a conversation coming from a TV to your rear left! You will literally hear the whole conversation coming from behind you and this ONLY happens with the most accurate of toe-ins. Later in the song, there are parts for center stage adjustment as well.

Centre sound stage:

Little Room – Norah Jones: She sings all of the song from dead center. Good to optimize SPL level differences between your speakers around 1-2 kHz

Sound stage depth and width:

Pink Panther Theme – Henry Mancini: You can locate several instruments in the orchestra (and not exactly on the far left or right) if all is fine with your system.

Black Sabbath – Black Sabbath: One of the best mixed records however only perfectly calibrated systems will lift the thunderstorm at the beginning of the song high up the front wall where it belongs.

Enjoy your music!

Serkan Gür
Aug 2022

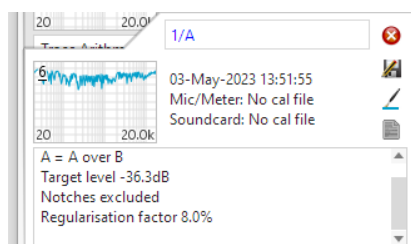
Update:

As of May 2023, REW early access version has evolved from version 11 to version 61 and is still work in progress. Many new tools were added and there were some revisions to some operations. The important changes relevant to this tutorial are as follows:

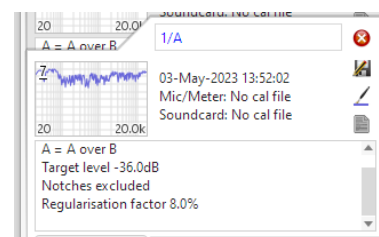
- You can now take multiple measurements at different microphone positions and cross correlation align them to the central position and then vector average and use this average as your speaker response to be corrected. It will help eliminate some local reflections and might result in more uniform bass response around the central listening area.
- Additionally, you can “cross corr align” left and right speaker measurements between each other so you don’t have to deal with the manual alignment process explained in this tutorial. You can simply continue the correction process after that. Precise central mic position is not important unless you’re trying to eliminate comb filtering between left and right speakers and measuring them together.
- “A over B” trace arithmetic operation has been revised slightly. It now produces the resulting impulse at around -40dB. The process is still the same. The only difference needed is to **tick Auto for target level** when you’re calculating 1/A. 1/A division results will have a “Target Level” written by REW in the notes section of the results. Make sure you increase/decrease the SPL level of one of the 1/A calculations (from Measurement Actions) such that left and right 1/A Target levels will be equal:

Example:

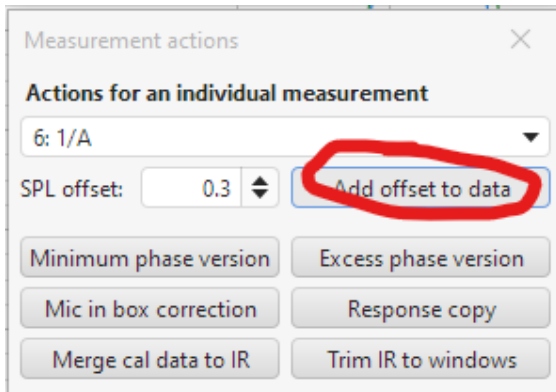
Left speaker 1/A operation result:



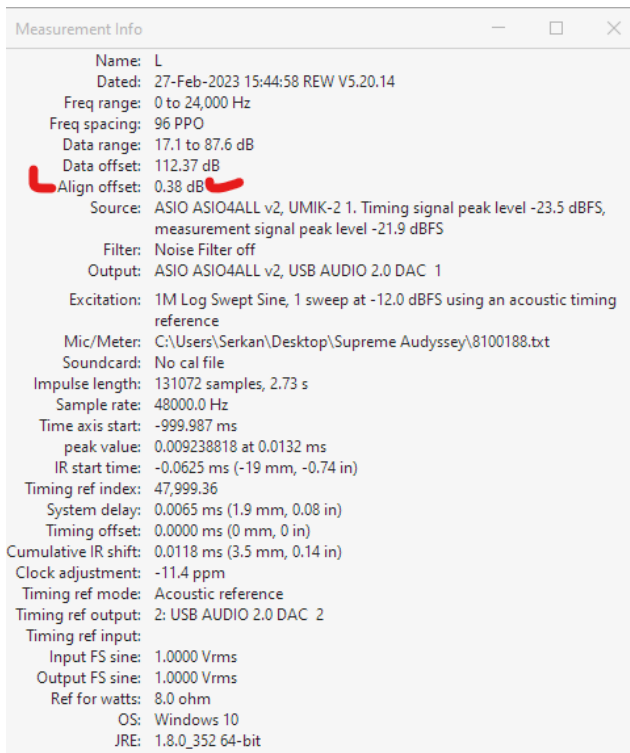
Right speaker 1/A operation result:



You will need to add $36.3 - 36.0 = +0.3\text{dB}$ to Left speaker 1/A division before you generate their minimum phase versions:



- For SPL Alignment, you do not need to make copies of measurements. You can simply align left and right speaker measurements with “Align SPL” command and then check for “Information- “I” window of each measurement. REW will show you how many dB volume alignment was done to that response:



- You can use the new “Trim IR Windows” feature under “Measurement Actions” to decrease the number of taps of the final convolution filters you generated if you want faster filters.