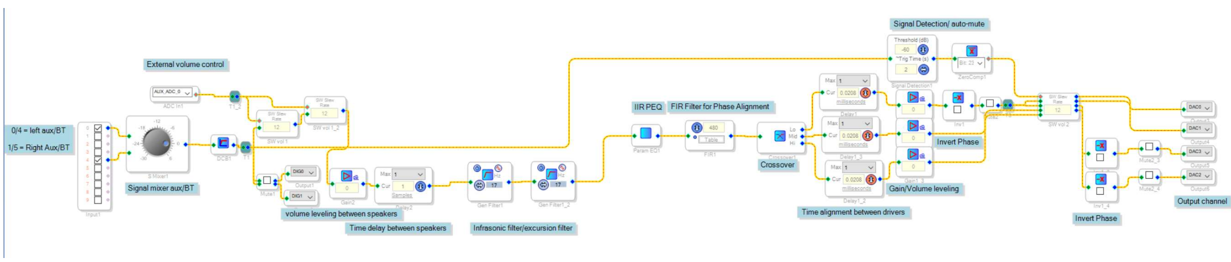
I. Introduction[1]

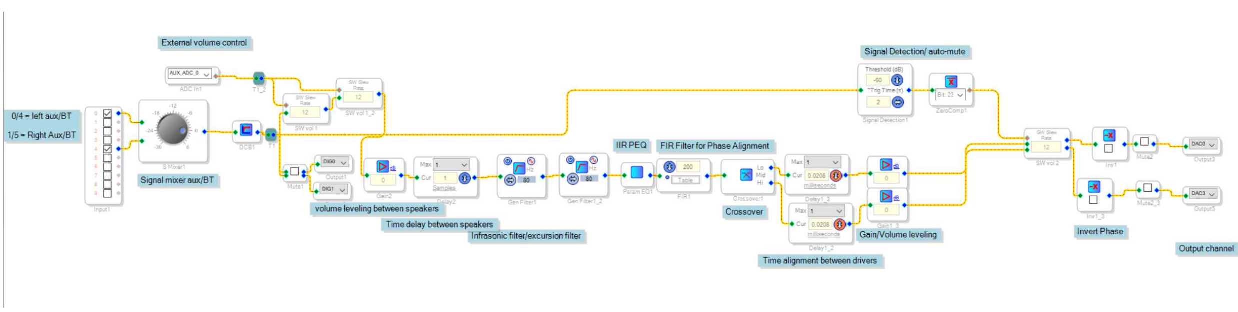
When people get started playing with active speakers or digital signal processors, it can seem intimidating to know where to start or what one should add to a schematic. Many like to tweak around to bring things to where they sound right, and they may even get testing equipment to help, but they can sometimes get confused on what all should be done to help the sound of their speakers to maximize the computational power of their processor.

Here, I provide SigmaStudio schematics made for a 3-way and a 2-way system. I will be focusing on how to implement the 3-way speaker, but the process is the same for the 2-way design. This was made using the Wondom JAB5 DSP Amplifier, made by Sure Electronics. This should be substantially similar to the Dayton Audio KABD-4100, if not being a direct rebrand. With that being said, each company has guidance on how to set the Hardware Configuration register tab for flashing to the device. FOLLOW THOSE INSTRUCTIONS TO THE LETTER! If you do not, you can blow up the amplifier you just bought, and no one wants that. When in doubt, read the manual (“RTFM”).

3-Way Speaker



2-Way Speaker



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[1] **THIS INFORMATION IS PRESENTED AS-IS AND WITH ALL FAULTS. TO THE MAXIMUM ALLOWED BY LAW, ALL EXPLICIT AND IMPLICIT REPRESENTATIONS AND WARRANTIES, INCLUDING WARRANTY OF USE FOR INTENDED PURPOSE, ARE HEREBY DISCLAIMED. THE AUTHOR AND ANYONE THAT PUBLISHES OR REPRODUCES ANY OF THIS DOCUMENT, IN WHOLE OR IN PART, SHALL BE HELD HARMLESS IN THE EVENT OF ANY DESTRUCTION OF PROPERTY THAT MAY ARISE FROM ANY USE OF ANY PART OF THIS DOCUMENT. BY USING ANY INFORMATION CONTAINED IN THIS DOCUMENT, IN WHOLE OR IN PART, THE READER EXPRESSLY AGREES THAT THEY ARE LIABLE FOR ANY DAMAGE TO THEIR OWN PROPERTY AND AGREES TO HOLD HARMLESS THE AUTHOR AND ANY PUBLISHERS OF THIS INFORMATION.**

This guide will not show exactly how to design the schematics. It will not show the nuances of SigmaStudio[2], Room Equalization Wizard ("REW")[3], or rePhase[4], although these three free software are used in the process and it is always recommended to, when possible, support those that create programs and content which you enjoy and that help you in your hobbies or your daily life. Always be aware of the End User License Agreements and restrictions on how some software is allowed to be used. Either way, that is not my business and not the point of this guide. Instead, let's dig into the meat and see what is in the schematics and why.

II. Components of the schematic

I designed the schematics, for the most part, to be used from right to left. Why right to left? Just with how you draw out the diagram, most of what is closest to the reproduction device, in this case a driver, will have the most impact, and much of the correction comes after the active crossover point to address each driver and its individual nuances for how that adds up to the whole.

A. Mute and Invert Phase

For convenience, I added a mute per channel of the speaker as you will be turning each channel off and on throughout the testing used to populate the schematic with values. Also, I have included switches to invert the phase of the signal by 180 degrees. This is in case you wired up the speaker incorrectly and you don't want to go in to fix it. This can be seen in testing if you have "suck out," a rapid declination or depression in the signal around the area of the crossover caused by the wave cancellation between the two drivers due to their out of phase signals. These are quality of life additions to the schematic, but are not essential. Since you are not going to be able to use these after you are done setting up the speaker, they can be removed near the end if desired.

B. Signal Detection Circuit

Aside from these, there is the signal detection circuit. This does not need messed with, unless you want to adjust the threshold and the time it takes to trigger. The purpose of this circuit is to auto-mute the speaker when music isn't playing. This will allow a quieter speaker with less hiss that may be present on some configurations. Currently, the threshold is set at -60dB, which should be sufficient to not be triggered by noise.

C. Bias and Time Alignment

Next is the volume leveling and the time alignment per driver (as a reminder, we are working right to left through the above schematic; as such, if following along, we are at the labeled blocks Gain/Volume Leveling and Time Alignment between drivers). Volume leveling is used to adjust the sensitivity differences between the drivers, while time alignment is used to make sure that the drivers are producing sound that arrives at the same time to the listener. Normally, you will only need two of these

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[2] Analog Devices, Inc., *SigmaStudio®*, *available at* https://www.analog.com/en/design-center/evaluation-hardware-and-software/software/ss_sigst_02.html#software-overview (last visited May 22, 2022).
[3] John Mulcahy, Room Equalization Wizard, *available at* https://www.roomeqwizard.com/ (last visited May 22, 2022).
[4] Thomas Drugeon, *rePhase*, *available at* https://rephase.org/ (last visited May 22, 2022).

three, but there are occasions where you may have to look for the lowest common denominator to find the best solution in which all three drivers align regarding time.

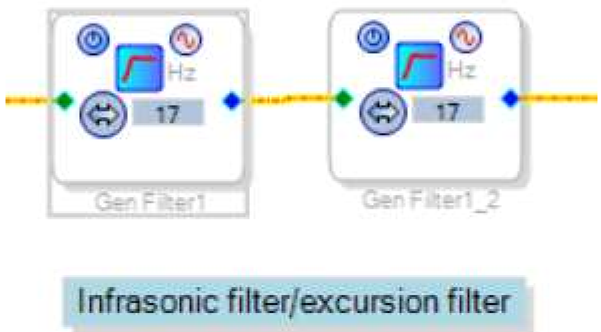
Time alignment is extremely important. Normally, a person will be listening at the level of the tweeter. But, if you measure the distance from your ear or a point in space to each driver, with your ear or point directly in plane with the tweeter, you will find that the mid-woofer, woofer, or subwoofer are further away from you, if in the same enclosure, that the tweeter due to the angle toward the floor. As such, the sound from the woofers will take longer to get to your ear because they have to travel further. How do you correct for this? Impulse response testing and time alignment. How to achieve this is below in the testing and application section.

D. Crossover

After the drivers have been aligned for sensitivity and time, it is time to setup the crossover.

SigmaStudio has an excellent graphic interface for adjusting the crossover points. You are able to grab and drag the points in the chart or you can manually enter the numbers into the fields below. You can link the crossover points between drivers or, as is sometimes the case, you will want to leave them separate because the point of crossover needs set to slightly different frequencies for the drivers to match up perfectly. They have the options for Linkwitz-Riley ("LR") stated as the dB per octave drop, not by the bend or the -6dB down at the stated value. Butterworth and Bessel filters are also available. Also, less drop per octave is available, but there are trade-offs and that is beyond the scope of this guide.

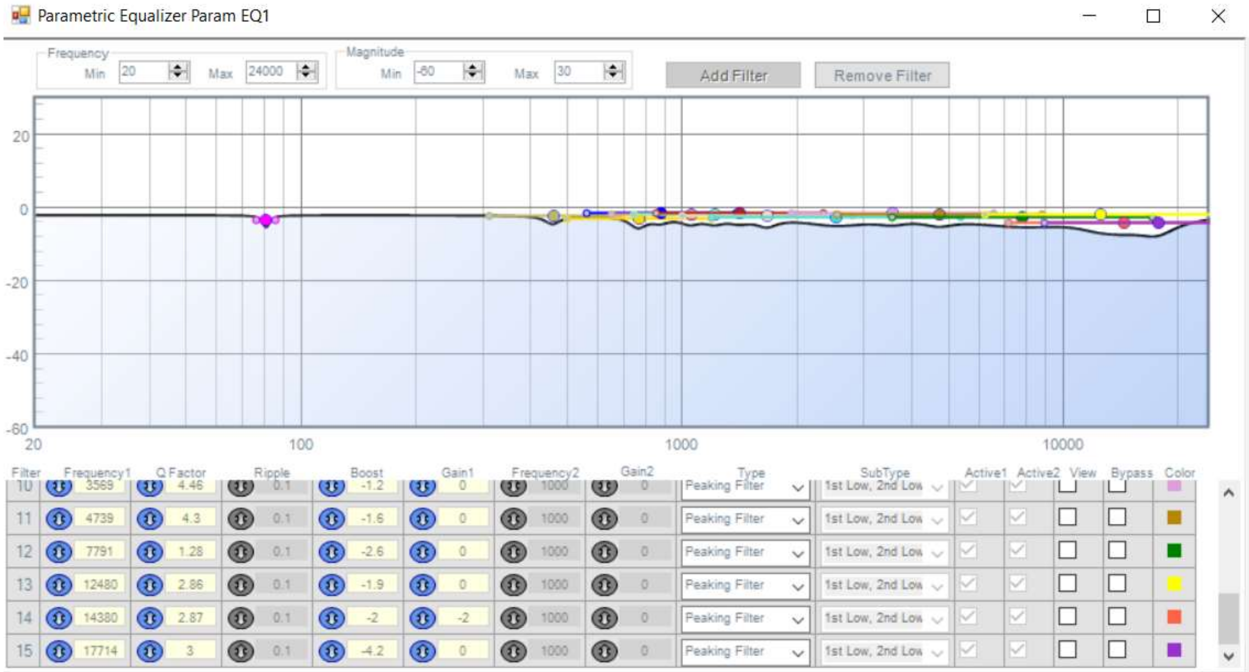
E. Infrasonic/Excursion Filter



Next up is the infrasonic/excursion filter. This is especially important if using a vented enclosure. When using a vented enclosure, there exists two frequencies at which the driver will reach excursion. One is the same approximate frequency area of the sealed enclosure. The other is often below the resonant frequency of the driver. If the speaker tries to play those frequencies, even if, as is the case with a subwoofer, the driver is playing inaudibly to you, it could be pushing the driver past X-max. But there is a second benefit to setting this to prevent that X-max excursion—linearity. By reducing how close the driver is playing to the excursion limit of the driver, the driver will remain positioned within the magnetic field more often, thereby it is moving more linearly than drivers do toward or beyond X-max. As such, but preventing the driver from getting too close to the X-max value, one can increase linearity and decrease distortion, to a degree, thereby getting cleaner sound. Obviously, there is a trade-off. Here, the speaker will not be able to play as low, thereby compromising on bass sounds. As such, you should prevent excursion, but how much more to reduce, or rather how high to set the lowest extension of the lowest frequency woofer in the enclosure, is up to your judgment. Here, I cascade two second order butterworth high pass filters, thereby making this a 24dB/oct LR filter. Just change the values on both to match.

F. Parametric Equalization Filters

After all this fun, there is still more to do. Next is the Parametric Equalization filters.



As seen here, this schematic is able to use 15 parametric EQ points. These points can be derived through the use of the program REW. By entering the frequency, the Q factor, and the gain (labelled “Boost” in SigmaStudio’s PEQ menu), you will be able to correct for certain issues in frequency response. It should be remembered that even though using an active processor, you still want to bring the gain down, not up, generally. Why? Because the driver can only play so loudly at any given frequency. If you push that higher at a given volume, the driver may start having compression artifacts or similar issues earlier than otherwise by going up in sensitivity at a given frequency instead of coming down to match the low points, which gives more headroom, but limits the overall max volume possible with the speaker. These are trade-offs and it is important to understand what is being gained and what is being lost.

G. Limited FIR Filter for Phase Correction at Crossover

Either way, after setting up the parametric equalizer, it is time to adjust the phase using the very limited FIR filter. This cannot correct all phase issues throughout the frequency response. This cannot and is not meant to incorporate the above PEQ filters. This is solely to correct aberrations in phase around the crossover points. There are not enough taps to do more than this basic task. But, it is worth the old college try to correct this as much as possible.

H. Speaker Pairing and Leveling

Finally, we get to the point of speaker pairing. There exists both a gain and time alignment module at the left near the start of the schematic. These are used to volume level both speakers to each other, followed by then using impulse response testing to align your left and right speakers together at the listening area. For this, you will have to set the speakers up in the room they are to be used in, test, and then adjust so that the alignment matches the environment for use. Unlike the first time these were used earlier, which only was to align drivers in their enclosure, we are now aligning two loudspeakers within the system that is your listening room. As such, you have to consider where everyone will be seated in the room, if you will have one sweet-spot or have to measure and average to get the best compromise in alignment for all seats in a given area. Either way, by time aligning the drivers and volume aligning, it will help with the staging in the listening area. This is because the sounds played by each speaker will be arriving to the listener at the right time to give an image or sound stage. If speakers are not aligned with each other, then it can throw off an immersive experience. Many receivers ask for the distance of the speakers from the user. In part, one aspect of that is to figure out the delay needed for the sound from each speaker to reach the listener. There are other factors involved, such as signal propagation, but considering those are in less than a millisecond over coper wires in the environment we are talking about, it really depends exactly how pedantic one wants to be. If all wires are the same length, then that removes that consideration as the signals, sent at the same time, should arrive at each speaker at relatively the same time, making the distance the most important factor to then estimate delay. But why estimate when one can measure both speakers, then align them on the basis of measurements? That is why the settings are there.

So, with the explanation of the parts of the schematic that need explaining, ignoring the signal mixer and the cascaded external volume control to make turning the knob more exponential than linear, it is time to explain how to fill in what you need to use it.

III. Application and Testing

Testing is a complex topic. As such, I am only going to give a basic description of the software, the settings that can be used, and how to populate the data in the schematic. But, just because I will not cover all nuances, I can provide resources to help one learn more on the topic.[5]

A. Gating: When Should One Use It and How to Do It in REW

One important aspect of testing that is cursory to all testing mentioned in this section is how to gate and positioning needed in order to accurately test the speaker. This is a complicated topic. As such, I will give a brief description and provide links to manufacturers that have released helpful documents on these topics. No reason to reinvent the wheel.

First is deciding if one is going to use smoothing or gating, if one is going to measure nearfield, or use ground measurements, or measure outside to allow longer periods of time before any reflection will be seen in the measurements.[6] Now, gating allows for the removal of the reflection data from the measurement. This will allow for accurate measurements above a certain point. Because you are gating off reflections, this also gates off low-frequency waves. As such, when you gate, you only have useable down to 200-400Hz, depending on the environment in which you are testing. Testing outside, away from buildings and roads, while lifting the loudspeaker up high to give the longest possible amount of time

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[5] There are two standards that will interest most Do It Yourself (“DIY”) individuals that are using these DSP amplifier boards in their designs. First is the *Standard Method of Measurement for In-Home Loudspeakers*, Consumer Technology Association, CTA-2034-A-R2020 (Feb. 2015), *available at* https://shop.cta.tech/products/standard-method-of-measurement-for-in-home-loudspeakers (last visited May 22, 2022). This will give information on how to test loudspeakers generally according to one rulemaking body, helping to standardize testing. The second is the *Standard Method of Measurement for Subwoofers*, Consumer Technology Association, CTA-2010-B R-2020 (Nov. 2014), *available at* https://shop.cta.tech/products/standard-method-of-measurement-for-subwoofers (last visited May 22, 2022).

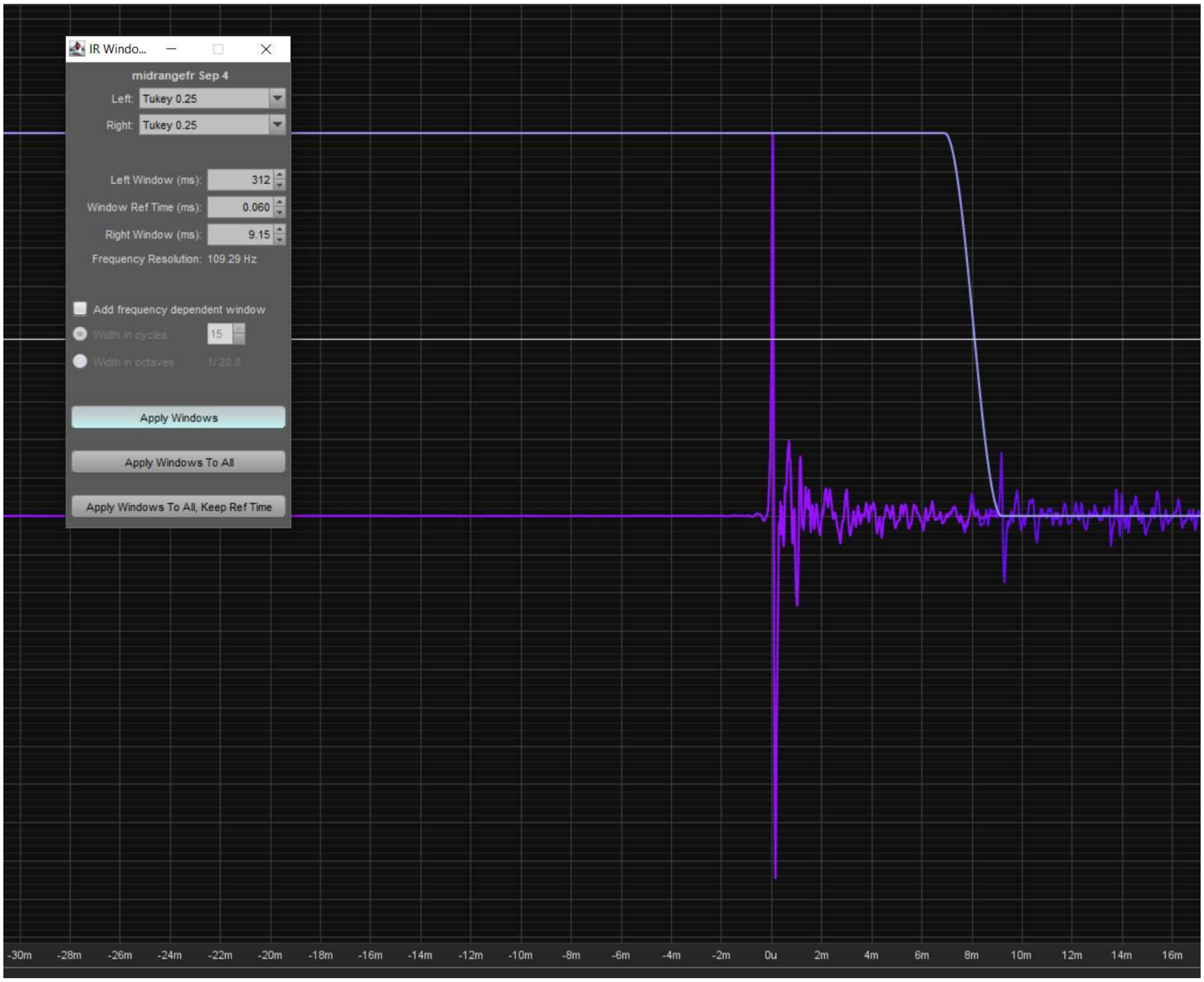
[6] MiniDSP, *Loudspeaker measurement with UMIK-1 and REW*, https://www.minidsp.com/applications/acoustic-measurements/loudspeaker-measurements#gated-measurements.

before a reflection could appear in the measurement is preferred if not using an anechoic chamber or a Klippel nearfield scanner. Most home enthusiasts have neither the chamber nor a Klippel. Many do not live in the boondocks which would allow for the best outside setup. As such, work with what you have, whether it be a backyard, front yard, or a large room or small room. Just be aware, the sooner the reflections arrive, the higher the usable Hertz, the less adequate information you have to use to adjust low-end frequencies. But don't be dismayed, as there are other solutions or you can use multiple ways in conjunction.

So how does one determine the gating one must use? To determine this, one must look at impulse response results.[7] Well, let's head over to the REW software and take a look. First thing you want to do is take a sine-sweep. I will assume that the person knows how to setup the microphone with a calibration file for REW already. Once a sweep has been done, go to the impulse tab.

As seen in the miniDSP document, there will be an obvious point at which reflections will show themselves as a ripple with extra amplitude compared to before that point. You can then enter that into the program by going to tools>IR Windows, then put the time value for where that ripple occurs into the Right Window input. This will gate the measurement to exclude everything that is picked up by the microphone after that time period.

[7] For a paper comparing and contrasting methods for taking impulse results, see Guy-Bart Stan et al., *Comparison of Different Impulse Response Measurement Techniques*, 50 J. Audio Eng. Soc. 249 (Apr. 2002), *available at* https://gstan.bg-research.cc.ic.ac.uk/JAES_Online_version.pdf (last visited May 22, 2022). *See also* Farina, *Simultaneous Measurement of Impulse Response and Distortion with a swept-sine technique*, 48 J. Audio Eng. Soc. 350 (Apr. 2000), *preprint available at* https://www.researchgate.net/profile/Angelo-Farina/publication/277293870_Simultaneous_measurement_of_impulse_response_and_distortion_with_a_swept-sine_technique_Audio_Engineering_Society_Convention_108_Audio_Engineering_Society/links/5a6f6e6f458515015e615ffe/Simultaneous-measurement-of-impulse-response-and-distortion-with-a-swept-sine-technique-Audio-Engineering-Society-Convention-108-Audio-Engineering-Society.pdf?origin=publication_detail (last visited May 22, 2022); Farina, *Impulse Response Measurements*, 23rd Nordic Sound Symposium: Training and Information Seminar for Audio People (Sept. 27-30, 2007) (describing advances and changes to the swept-sine technique to overcome shortcomings), *available at* http://pcfarina.eng.unipr.it/Public/Papers/238-NordicSound2007.pdf (last visited May 22, 2022).



This is given as an example, but the measurement itself was taken in less than optimal conditions, to say the least. As such, the example given in the miniDSP source is much cleaner and should be referenced if one cares. Now, an issue, as previously explained, is that by gating the input, one cannot trust the low-end of the readings for the frequency response.[8]

For that, one may want to take ground measurements.[9] When conducting ground measurements, the microphone is on the ground. The benefit to this is that you can read very low frequencies. But a drawback is that it is only accurate to a specific frequency in the kHz range. This is because the microphone has a small amount of elevation between where the microphone capsule and the ground. As such, above a certain frequency, it can have reflection data.

[8] To convert this to frequency, divide 1 second by the time period of the window. In my example above, you convert 9.15ms to 0.00915s. Then, you divide 1 by that value to get your frequency (1/0.00915=109.29Hz). *See* miniDSP, supra note 6.

[9] See miniDSP, supra note 6; *see also* CTA-2010-B R-2020 Subwoofer, *supra* note 5.

So, when used together, in theory, the gating for the tweeter referenced measurement and the ground measurement should give useable data that can be roughly overlapped to design the parametric equalizer filters we will make at a later point.

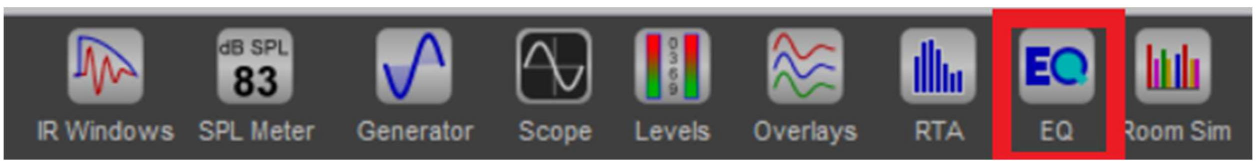
The other alternative is to use smoothing. Smoothing allows for using the data taken fully from the tweeter reference point, but does have drawbacks, such as losing detail. See the miniDSP document for a brief overview of those drawbacks.

B. Testing and Application

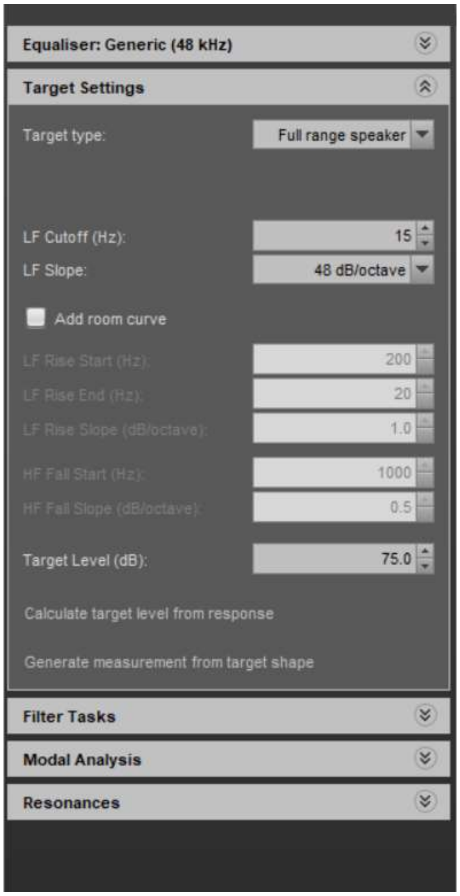
Now that you know your method for generating impulse responses, and method for measuring the data, we will push forward with how to apply that to the schematic presented above.

i. Gain Leveling and Time Alignment

First, conduct frequency response sweeps of each driver individually, using the mute button in SigmaStudio for each channel not currently being used. With this data, then select the EQ button.



Once there, open the drop down on the right hand side that says “Target Settings.”



In this menu, click on the "Calculate target level from response." This will give a target level for the response of the driver measured. Repeat this with each driver. Take the lowest of the three values, then do the basic math to reduce the other two drivers to then be about equal with the lowest dB target level. It should be noted that this estimated level will not necessarily be optimal and should be considered as a ballpark. So in addition to this, one should check with one's eyes that the sensitivity of each driver is relatively matching past the points around which the crossovers are roughly planned, looking for a continuity of the frequency sensitivity. Adjust until one is satisfied that the values are roughly equal. Once volume leveling has been achieved for the drivers, move on to the time alignment.

To do time alignment, one must look to the alignment of the impulse response. After taking the measurement of each driver individually once volume leveled, compare the three readings to each other in the impulse menu found by clicking the button above the chart. Whichever of the three drivers has the longest time period between t=0 and the largest spike point (re-adjusting window size may be necessary) will be used as the anchor, the point to which you will delay the other two drivers to in order to align their output.[10] Test and change these values until the time for the peak amplitude of all drivers matches each other. At this point, you should have roughed out volume and time alignment. As such, it is time to start designing the crossover.

ii. Crossover and Infrasonic/Excursion Filter

[10] *See* Scott Evans, *Impulse Measurement and Time Alignment using Systune, Smaart7, and REW*, YouTube at timestamp 16:35 (Sep. 30, 2021), https://www.youtube.com/watch?v=1jPtvZg_6RU (last visited May 22, 2022); *see also* Nathan Lively, *What is "estimate IR delay" in REW*, Youtube (Nov. 22, 2021), https://www.youtube.com/watch?v=QqiJsJk6rjc (last visited May 22, 2022). As discussed, the estimated impulse response delay in REW may not be accurate, so expect to change these values recursively, checking between each round until alignment has been achieved.

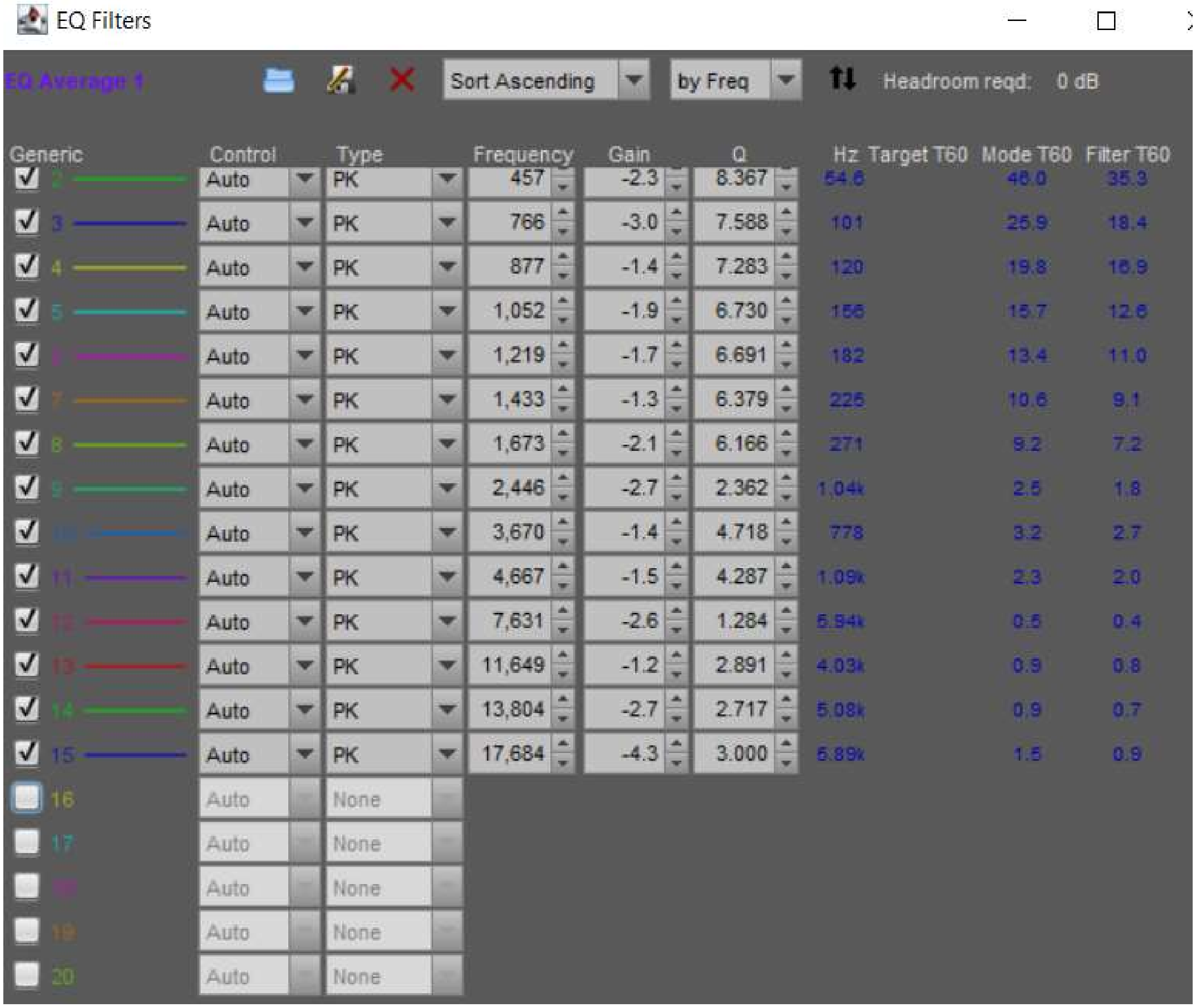
Whittling it down to the perfect crossover between the drivers is going to be recursive. So dive in close to where you believe the drivers should crossover, then measure two drivers at the same time and those same two separately. Once the high pass and low pass is set, then repeat the steps with the other two drivers. I recommend starting with the lowest frequency value driver to the driver right above it in frequency response. After these are set, repeat that with the values for the infrasonic/excursion filters (which are high pass filters) for the lowest playing driver. This will establish the band pass for each driver.

After this, go back and verify both the gain levels per driver match and that the time alignment matches. Why? Because now that you have changed the bandpass frequency range for each driver, there is less frequency range to draw from, which may impact the average sensitivity per driver, even if a tenth of a dB. Same is true with the impulse response alignment. Without the full range of frequencies, this may shift slightly, although less likely to be effected compared to the gain levels. Tuning these recursively helps to correct this before creating the parametric equalization filters. And after the time and sensitivity alignment has been corrected, just verify that you do not need any tweaks to the crossover points by playing a sweep with all three drivers active. If this looks good, it is time to move on to the parametric equalizer module.

iii. Parametric Equalization

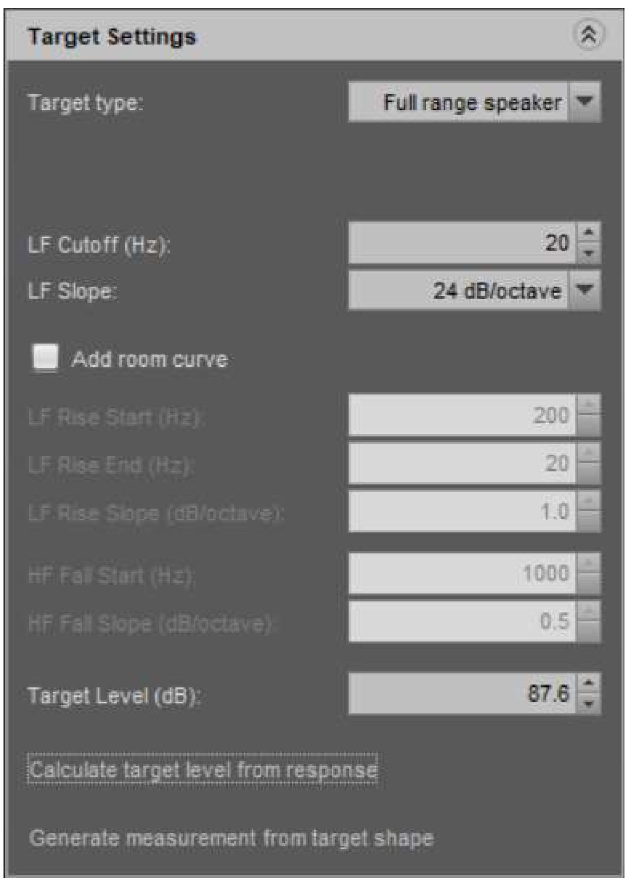
To generate these values, take the verification sweep in REW and click on the EQ button at the top of the screen. It is now time to use this menu in earnest.

First step is to select your equalizer. Depending on what you select, it will have different values, such as the sampling rate. You do not want a sampling mismatch. Because the ADAU1701 only has so much computational power, we would need the 48kHz or 44.1kHz sampling rate. As such, I have used 48kHz for the SigmaStudio file and the equalizer selection in REW. The reason I chose generic is very few allow for up to 15 slots to be used for the parametric EQ bi-quad filters. As such, after selecting generic, click on the EQ filters button above the graph.

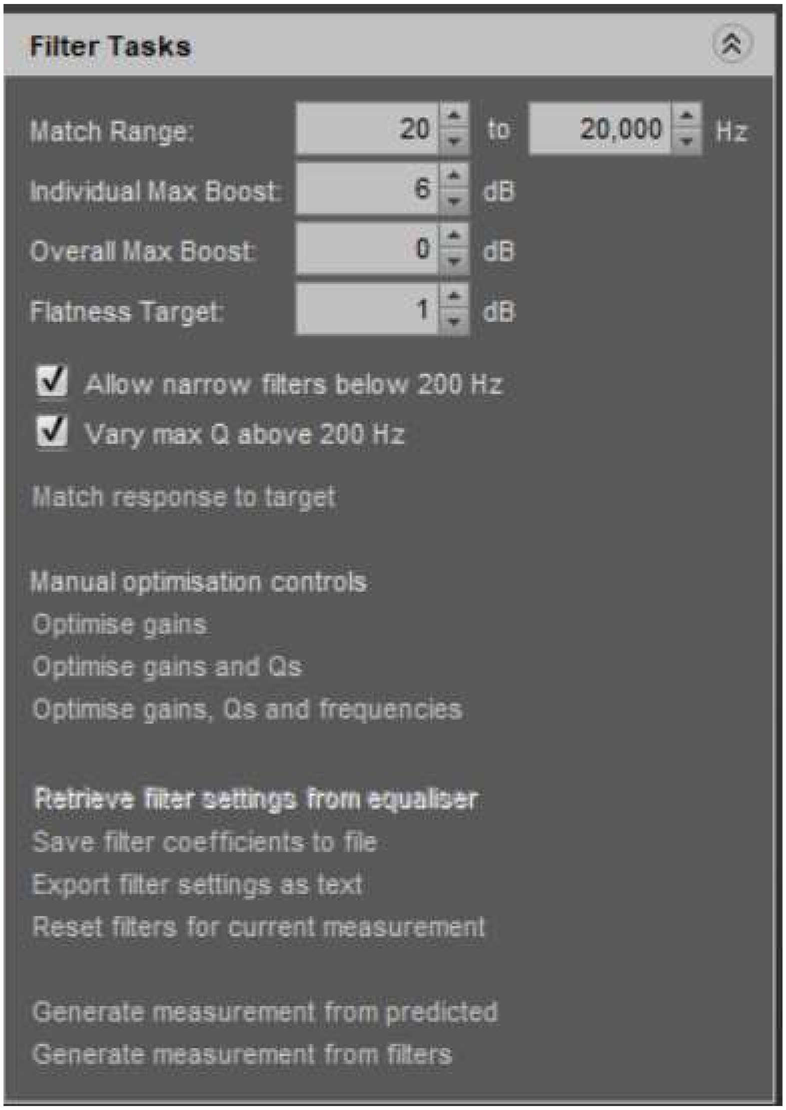


Once there, deselect all above the number 15 slot. This will ensure that 15 slots are the maximum number of bi-quads that can be created. This is also the menu that will have the general values that you need to move over after they are created to the PEQ filter segment of the schematic.

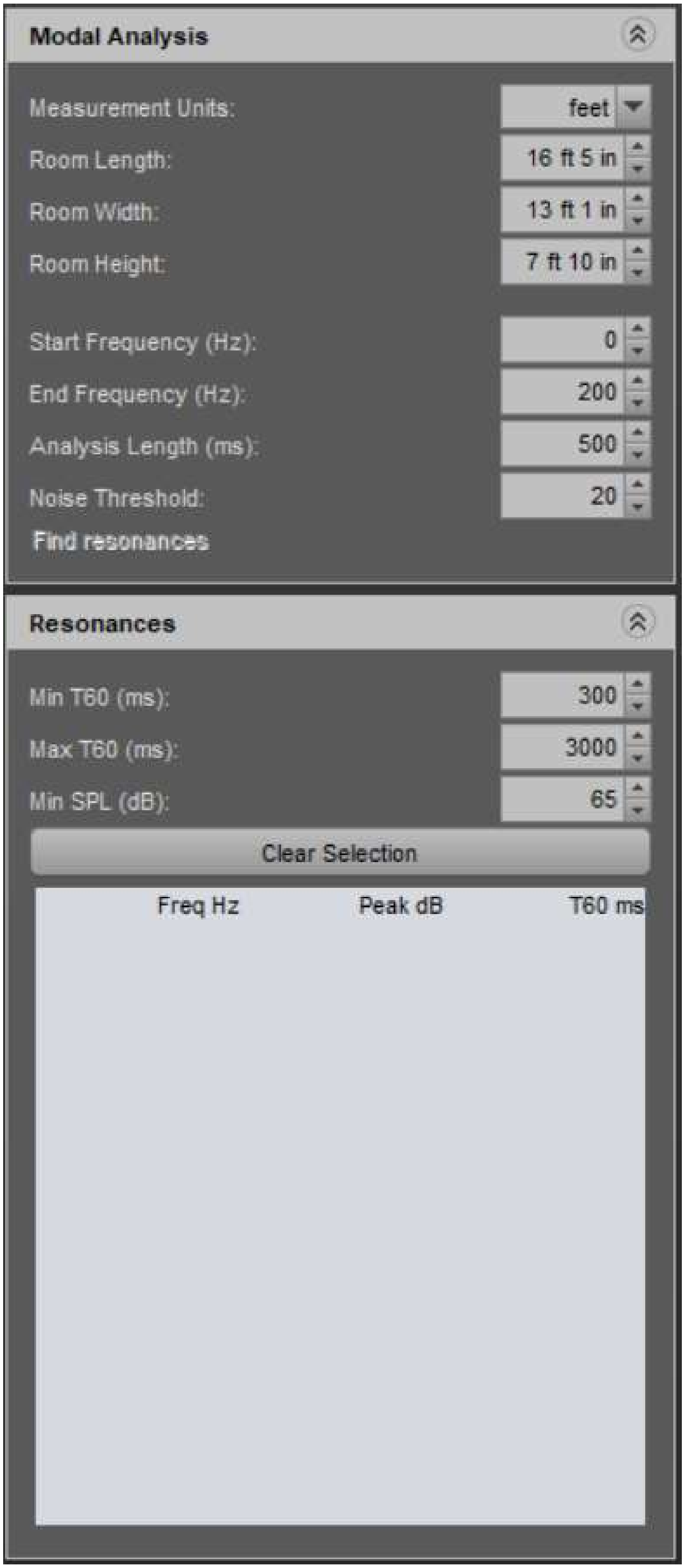
Then we move to the next drop down menu.



Here, you would need to set the target type, in our case a full range speaker, the low frequency cutoff, and the low frequency slope. With the cascading butterworth filters, the 24dB/octave should be correct. You can add a room curve here, but that is beyond what will be discussed here. If you would like, you can click on "calculate target level from response" once again. This value is now the frequency sensitivity the filter generator will be trying to achieve.



Next, set the range needed for the filters to be generated. Generally, you don't want to allow a boosting of gain to occur due to compression possibilities. If from cutting gain on the drivers earlier you know the specific spot that needs boosted was reduced due to generally reducing the gain of that specific driver, then a positive value for overall max boost can be set higher than zero, just be aware that compression can become an issue. Generally, you will be using negative gain values ("boost" value in SigmaStudio menu) for the filters. It is up to you if you would like to have narrow filters below 200Hz or variable max Q above 200Hz. That is beyond the scope of this document. Instead, click on "match response to target." If you didn't see all of the optimizations, you can manually click to optimize gains, optimize gains and Qs, or to optimize all. After clicking this, click on EQ Filters once again to make sure all 15 filters are populated. If not, reduce the flatness target and click to generate the filters again. I started with the flatness target of 3dB. I reduced it to 2dB, but still had one or two filters not calculated. But by reducing the target to 1dB, all filters were populated. These can now be moved over to SigmaStudio.

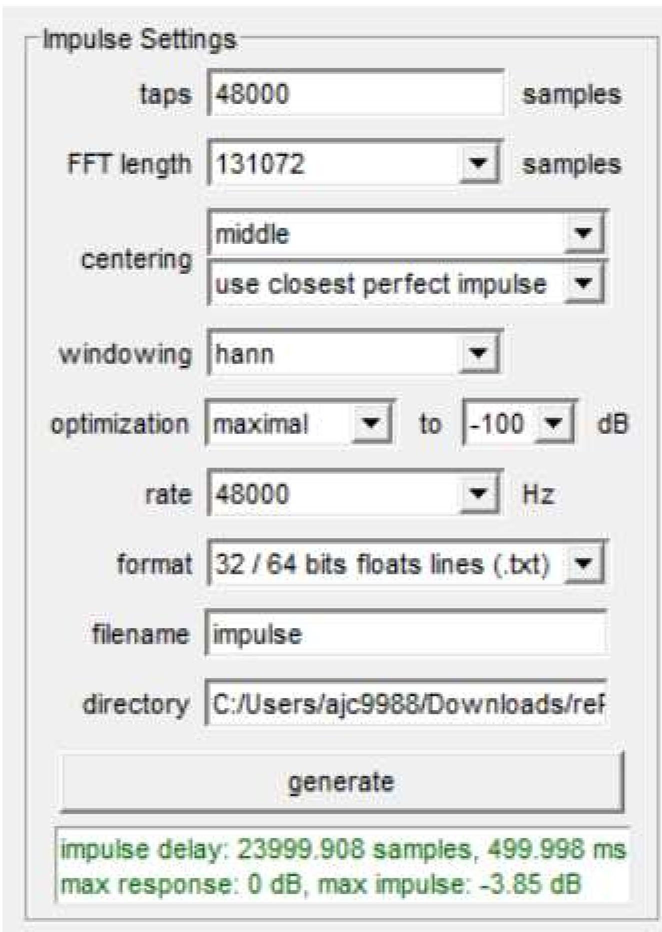


One final note, if one wants, one can enter the information into modal analysis. This gives boundary information to REW to help generate the filters. I will not be covering that nor resonances at the moment, but it is worth mentioning that these are factors that can be plugged in and considered by REW.

iv. Limited FIR Filter

At this point, you should have completed the bulk of tuning for the speaker. You have the drivers time aligned and bias aligned. You have set the crossover values and the high pass filter for the low end. And now you have finished the PEQ filters. What is left? Well, you now get to remove any instructions you are not using, such as inert phase or mute. You then can increase the number of coefficients until when you click on Link Compile Connect in Sigma Studio, it finally returns an error which states it is out of instructions. At that point, back it down until the error disappears. That is the number of coefficients for phase correction that you are able to work with.

At this point, you will need to export a frequency response and phase graph from REW to use in rePhase. This needs exported to be used.[11] Once imported into rePhase, it is a matter of changing the number of taps to the number found in sigmastudio. The sampling rate must also match both sigmastudio and REW EQ settings. The level of optimization and windowing can be adjusted to fit your needs and preferences.



Once these settings are to your liking, click generate. Take those values generated and saved in a text format and load that text file into the table for the FIR filter. That will correct the phase as much as it will be able to be corrected.

v. Speaker Pairing and Time Alignment

Finally, we get to in room testing. Even though speakers with the same drivers will be similar when given the same stimulus, you are also correcting now for room gain and seating distance from both speakers, left and right channel. As such, make sure the volume is set equally between both loudspeakers. Then take measurements in the seating area and adjust the gain value to the appropriate level so that both speakers have the same gain sensitivity at the seating area.

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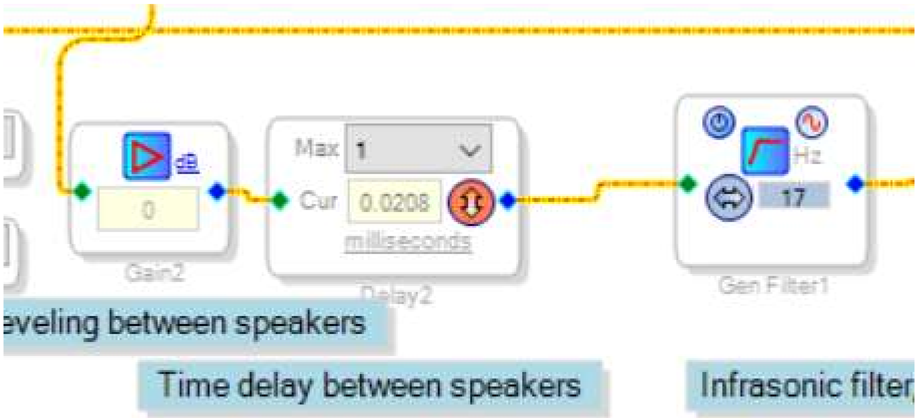
[11] See *REW average measurements and impulse correction rePhase*, *available at* https://rephase.org/tutorials/REW%20average%20measurements%20and%20impulse%20correction%20rePhase.pdf (last visited May 22, 2022). You do not need to export the calculated corrections from REW as those will not be used for the FIR filter. It is limited and can only correct aspects of phase. Otherwise you are wasting computation power for frequency response which is already addressed with the PEQ filters.



Once the speakers are volume leveled, it is back to the impulse response menu of REW to then time align the arrival of the left and right channels at the seated area. Once that is finished, congratulations! You should have significantly improved the listening quality of the active speakers now in your room. Take a victory lap. Throw on your favorite album. Soak it in. You deserve it after all of that work!

As a final note, this schematic, although in a modified form, should be able to be used with most adau1701 devices, so long as you are able to load custom schematics from sigmastudio on your DSP device. Make sure if you use this on another device that you change the registers in the hardware configuration tab to match what is needed for your device.