Deadline: Before 4 .00 pm on 22/09/2023 (Friday)

Project: Passive Filter Design

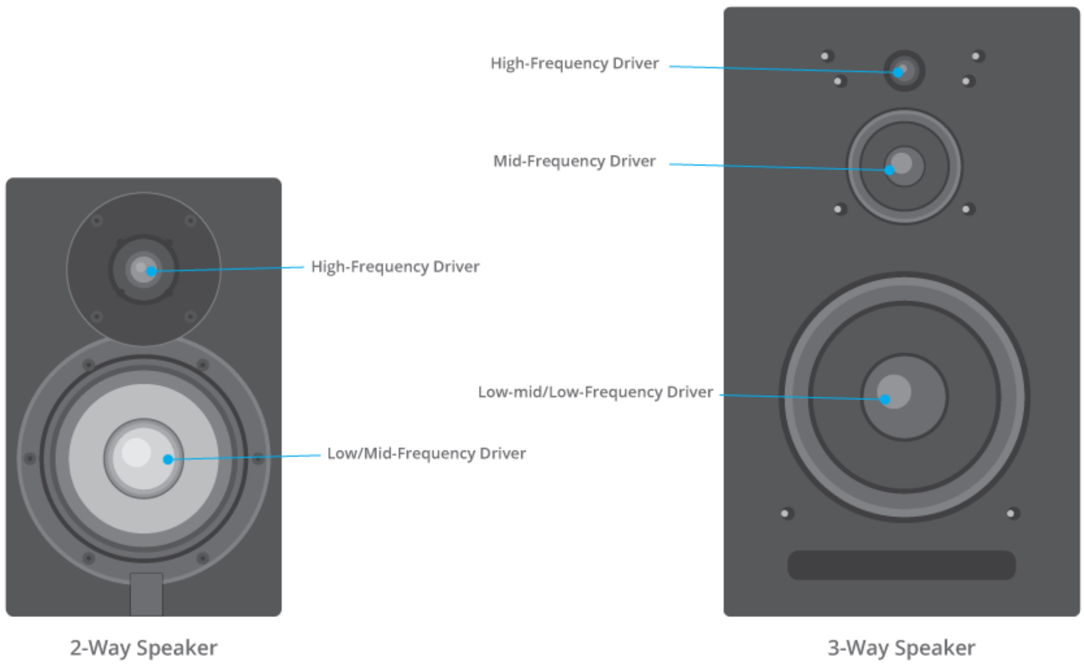
Speakers are wonderful things. You connect them to a piece of electronic equipment and they produce sound. It doesn't matter if you're talking about a computer, a home entertainment center or a smart phone; as long as there is sound to produce, they will do it. The humble speaker of today actually started out in the 1920s, almost a century ago. While efforts to replace electromagnetic speakers with some newer technology have been made, nothing has touched it for faithfulness in sound production and price. Other than minor changes in materials and designs, the speaker of today is essentially the same as the first one developed so long ago.

Speakers convert a complex AC (alternating current) signal into mechanical energy in the form of sound waves. Sound waves are nothing more than moving air. This air hits our eardrums and is converted back into neurological signals that our brain interprets as sounds. To create sound waves, a speaker has to move. The electrical connection is attached to a coil, making it an electromagnet. This electromagnet is housed in a fixed magnetic field, created by a permanent magnet. Since the signal to the electromagnetic coil is an AC signal, the positive and negative poles of the electromagnet are constantly changing. Obeying the laws of physics, this causes the coil to move within the fixed magnetic field. The speaker cone, attached to the end of the coil, moves back and forth, creating the sound waves.

If a perfect sine wave was put into the speaker, it would produce a humming sound. The tone of that sound would depend upon the frequency of the sine wave. A higher frequency would produce a higher tone. The volume of that hum would depend upon the amplitude of the sine wave, or how high the sine wave's voltage was. However, most sound is much more complex than a simple sine wave. If you consider an orchestra playing, there will be several frequencies being played, all at the same time. Several different volume levels will be present. Not only that, but each type of instrument interjects its own unique characteristics into the sound wave, modifying it so that we can identify that instrument. The same speaker has to be able to play all of that at the same time.

The more complex a sound wave, the harder it is for a speaker to faithfully reproduce the sound. Even so, they normally do a remarkable job of doing so. The true challenge comes when the speaker is required to produce frequencies or tones that are drastically different at the same time. That's like asking it to vibrate 100 times a second and 10,000 times a second at the same time. The human ear can hear a range of tones, generally stated as being from 20 Hz to 20,000 Hz. To give you an idea of what that means, the lowest note on a piano resonates at 27.5 HZ, while the highest note on a piano resonates at 4186.01 Hz. So, we can hear a range of notes that goes from lower than the lowest note of a piano up to much higher than any musical instrument can reach.

To faithfully reproduce such a wide range of tones, it is normal to use more than one speaker at a time, creating a "speaker system." Such a system can produce more accurate sound by splitting the full frequency spectrum into different parts, with each speaker only receiving the signal for a certain frequency range. That reduces the workload for each speaker, helping it to produce the tones more accurately, with less total distortion. The job of splitting the frequencies and sending them off to each individual speaker in the system is done by a Crossover Network. This is essentially a series of filters, which filter out the frequencies that should not go to each speaker. The most common sort of crossover network is what is known as a passive 2-way crossover. This type of crossover is normally built into the speaker cabinet, with the outputs going to each of the speakers mounted in the cabinet. In a 2-way crossover, there is a low-pass filter, and a high-pass filter is used. When designing and building a speaker system, it is important that the crossover selected not only match the number and type speakers which are installed in the cabinet, but that it can handle the total amount of power (measured in watts) that is to be sent to the speaker system.

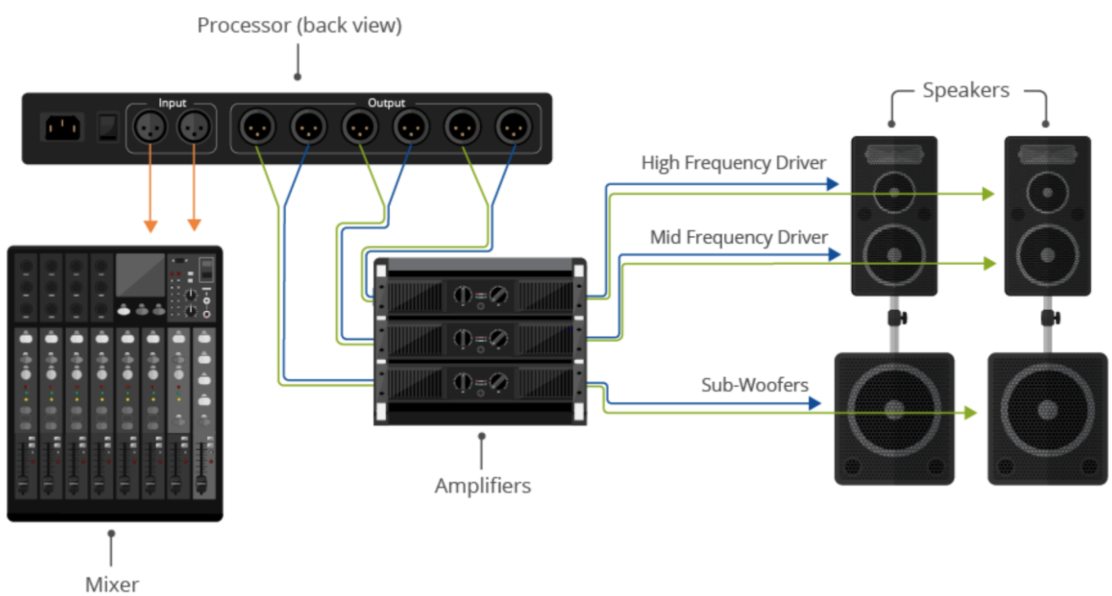


Since the audio signal is broken up into different frequency ranges to be sent to different speakers, it only makes sense that the speakers be designed to handle those frequency ranges. That's where woofers and tweeters come in. A woofer is a speaker designed for low-frequency sounds and a tweeter is a speaker designed for high-frequency sounds. At a glance, the main difference between woofers and tweeters is that the woofers are a lot larger than the tweeters. A good woofer might be 12 inches in diameter or more. There are a couple of reasons for that. First of all, the speaker has to move slower and the diaphragm (the speaker cone) has to move farther to create the sound wave. Secondly, the speaker must produce a higher volume of sound, as low frequency sound waves don't travel as well as high frequency ones do and are much more likely to dissipate and be absorbed by surfaces they come into contact with. The speaker enclosure and the woofer interact with each other; so the speaker enclosure is usually designed specifically to match the woofer. There are several types of designs, but the two basic categories are a sealed enclosure and a ported enclosure. Sealed enclosures try to trap the sound coming off the back side of the speaker, providing the cleanest, crispest bass sound. However, the sound volume is lower. Ported speakers are designed to allow that sound to escape, adding to the volume. However, the sound coming off the back of the speaker is 180 degrees out of phase with that coming off the front of the speaker. That can cause the sound waves to cancel each other out. However, the extra distance that the sound waves coming off the back of the speaker have to travel prevents that. Instead, the sound becomes less distinct and "muddy" due to the phase shift between the two sets of sound waves. Tuned ports are used on some speaker enclosures. These ports are created to a specific size, so that they will cause the sound to reach the area in front of the speaker exactly one cycle later than the sound coming off the front of the speaker. While this still creates distortion, it is less than that caused by an unturned port.

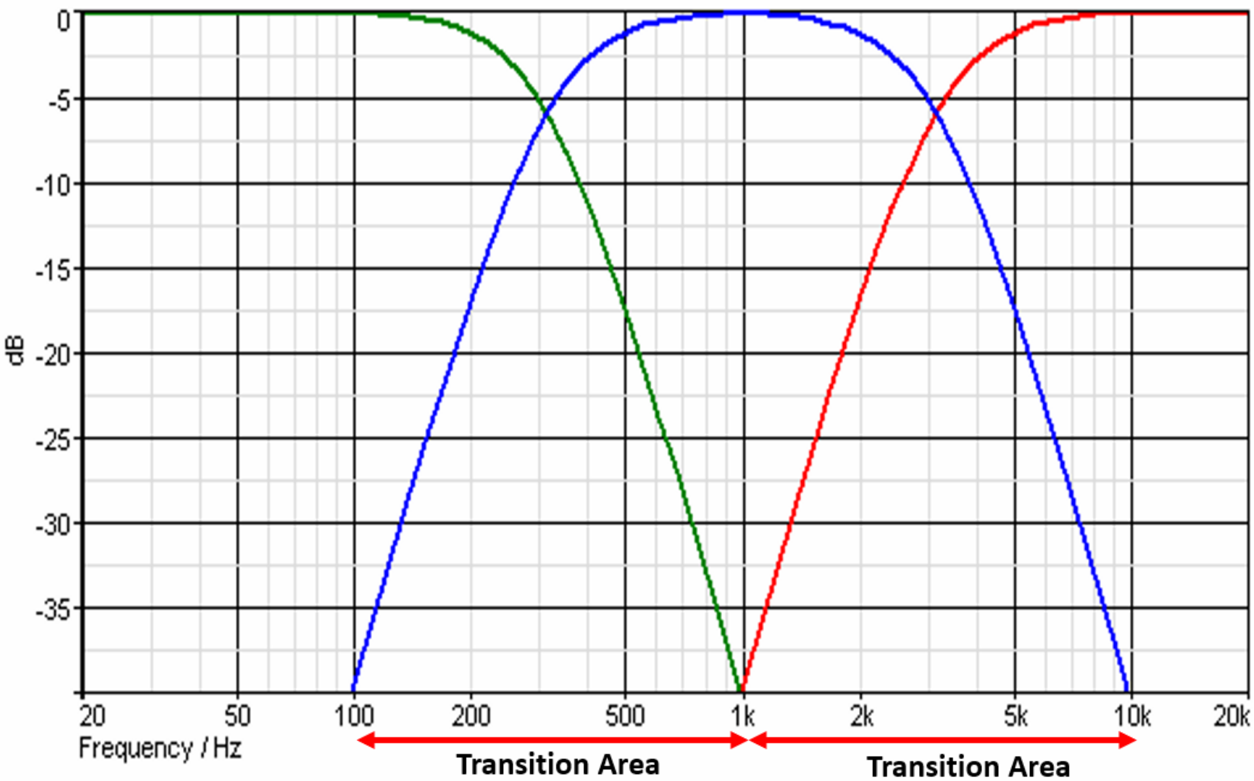
Tweeters (high frequency drivers) do not interact with their cabinets at all, and at times are used without a cabinet. While the construction is similar to a standard electromagnetic speaker, they usually use a dome-shaped diaphragm in place of a speaker cone. These are referred to as "dome tweeters." This diaphragm can either be made of plastic, plastic impregnated silk, aluminum or titanium. Each material type produces its own unique sound characteristics. Since tweeters are extremely small, they don't produce a lot of volume. To help this, many are attached to a horn. This horn resonates or vibrates with the tweeter, mechanically amplifying the sound that it produces, in much the same way that a trumpet or other brass instrument amplifies the buzzing of the musician's lips.

As previously mentioned, some speaker systems use three or more speakers. In those cases, midrange speakers are attached to each of the band-filters. A midrange speaker is essentially the same in appearance as a full-range speaker or woofer. The major difference is that midrange speaker will not be as big as a woofer, but only about 5 to 8 inches in diameter.

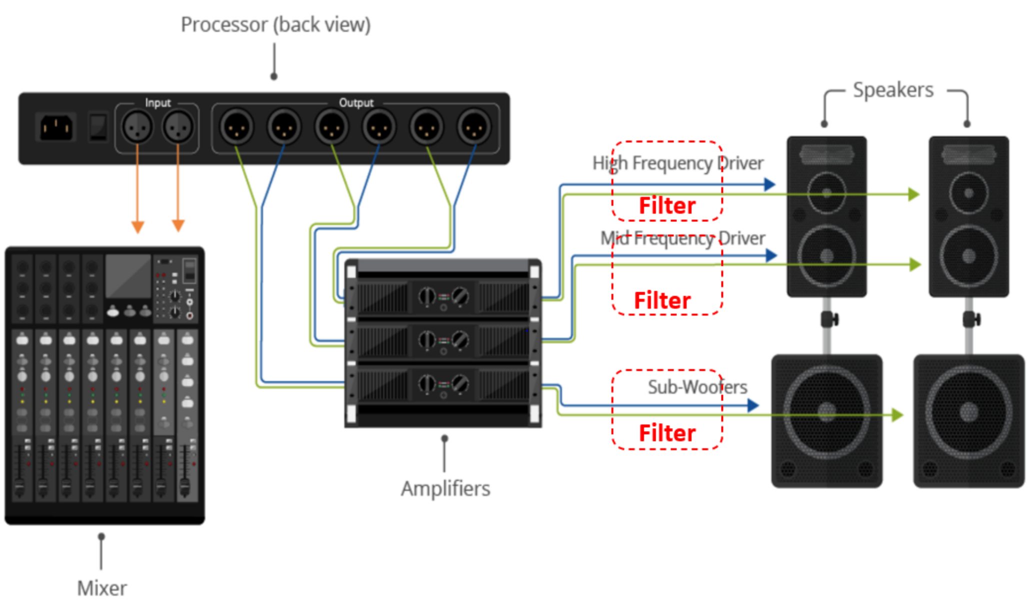
One of your friends is having a famous music band. He conducts music programs in various stages every month. Once you went to his music program and noticed the sound quality from the speakers are not in a good quality. As an Electrical Engineer you have decided to investigate and troubleshoot the issues with the speaker system. The way your friend connects the music equipment and the speaker system is given bellow. The Processor, Mixer and the Amplifiers are purchased recently from a highly reputed company and there are no issues with them.



You decided to check the speaker system and figured out he is using an old speaker for this purpose. You obtained the filter response of the speaker system from the datasheet, which is provided below.



From the datasheet, you have realized that the transition area of the filter response is larger. You anticipated that might be the reason for the low-quality audio output. Therefore, you decided to reduce the transition area by removing the old speaker's passive filter and connecting a newly designed passive filter, as shown below.



You decided to use the same cutoff frequency specifications used in the previous filter design.

Q1. What is the order of the filter that they have used in the old filter design?

Q2. What are the cutoff frequencies of each filter that they have used in the old filter design? Approximate your answer to a one-decibel point.

Q3. Design and synthesize maximally flat **filters** (all three) with the passband tolerance of $\boldsymbol{3dB}$ and frequency attenuation of $\boldsymbol{90dB}$ at $(\boldsymbol{5 \times cuttoff\ frequency})$. (a) Determine the transfer function of the filter (*no need to simplify it*). (b) Using LTspice software, determine the bode plot of the filter. (c) Compare the bode plot with the old filter bode plot and determine the transition area reduction in Hz. Assume the $source\ resistance = load\ resistance = 50\Omega$.

Q4. Design and synthesize a Chebyshev **filter** (Type 1) with a passband tolerance of $\boldsymbol{0.5dB}$ and frequency attenuation of $\boldsymbol{90dB}$ at $(\boldsymbol{5 \times cuttoff\ frequency})$ **only for the sub-woofer**. (a) Determine the transfer function of the filter(*no need to simplify it*). (b) Using LTspice software, determine the bode plot of the filter. Assume the $source\ resistance = load\ resistance = 50\Omega$

Q5. Based on the results from both type of sub-woofer filters, which filter you will select for the application mentioned above. Explain why you have chosen this filter.

Note: You are requested to submit a document that consists of

1. Your answer sheets. (show all the necessary steps)
2. LTSpice drawing for each filter (indicate the values of the components in the figure).
3. Bode plots for each filter (should be clear enough to see the X and Y labels).

You are allowed to use the supplementary materials that I have uploaded to Moodle.

Copying is strictly prohibited. If I identified identical answer sheets, then the total marks will be reduced for all of them.