Final Project: Systems with Analog Circuits

ENGR 212 - Design of Dynamical Systems

Section #LA

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Group #2

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I pledge my honor that I have abided by the Stevens Honor System. - Melanie Maksoud, Alix Perez-Dorado, Kylee Scherer, Mina Vargas-Gines

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Introduction

The purpose of this final project is to design a butterworth lowpass and highpass filter circuit that can split an audio signal into three frequencies. These frequencies include one low "bass" frequency, one high "treble" frequency, and one middle "mids" frequency. The goal was to completely filter out the middle frequency so that the low and high frequencies were split to separate parts of the speaker, specifically the woofer and midrange components. To complete this task we utilized the Fast Fourier Transform (FFT) in MATLAB to determine the frequencies of the signal, the Analog Filter Wizard website to design our filters, and finally the lab equipment to physically design our filters. Ultimately, this project required the use of digital platforms and physical circuit-building to create a filter that can split the audio signal into specific frequencies.

Materials & Methods

To begin the implementation of our circuit we first had to prepare by determining the frequency signals, planning the filter design and inputting it in Simscape. We first had to utilize the FFT in MATLAB to determine the number values of the frequencies. We used these numerical values derived from the peaks of the MATLAB graphs to find the cutoff frequency of both our high and low pass filters. For our low pass filter we used a cutoff frequency of 180 Hz, and for our high pass filter we used a cutoff frequency of 500 Hz.

To create the design of the filters we used the *Analog Devices| Analog Filter Wizard* website to automate a filter that corresponded to our desired outcome. More specifically, we determined that our low-pass filter would have a Passband gain of 10 dB, -3dB (30 Hz), a stopband of -16 dB (180 Hz) and a +- 10V voltage supply. As for the High-pass filter we determined that the gain values would be 10dB, -3dB (1.9 Hz) the stopband would be -12 dB (500 Hz) and that the voltage supply would be +- 10V. We created two 2nd order Butterworth and repeated this process twice, one to design our high pass filter and another to design our low pass filter.

We chose to create a Butterworth filter because we believed that its positive attributes would better serve our system rather than the pros of implementing a chebyshev or bessel filter. This was because they have a maximally flat response meaning that butterworth filters are designed to have the flattest frequency response in the passband (no ripples). This would result in minimal distortion of the signal's amplitude within the desired range which would be beneficial towards our desired outcomes of this project. It also has a smooth transition from passband to stopband, which helps avoid sudden changes that might introduce artifacts. The Butterworth filter also has a predictable phase response so while not perfectly linear like Bessel filters, Butterworth filters have a relatively well-behaved phase response compared to Chebyshev filters. We did not choose a Chebyshev filter because there are ripples in the stopband that can distort the signal. Unlike the Butterworth filter, it does not have a linear phase response which can lead to further signal distortion, especially in audio processing. Bessel filters also have a slower transition from the passband to stopband when compared to our filter.

Below you can see our specifications when creating our filter design in the *Analog Devices| Analog Filter Wizard* website.

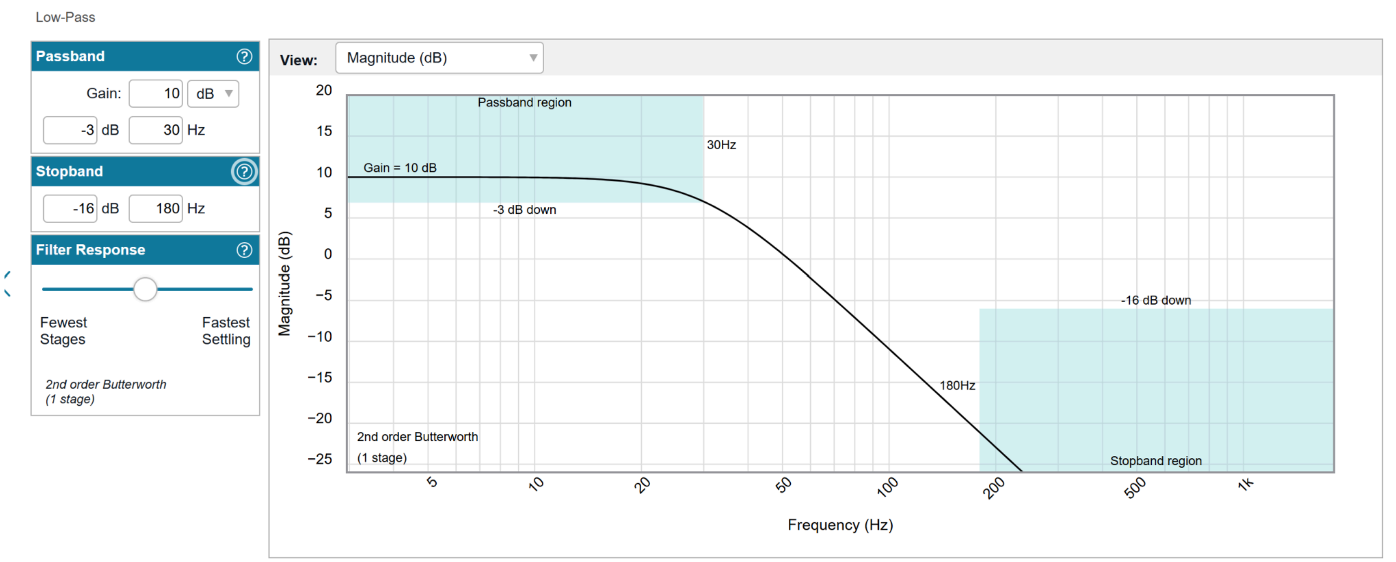

Figure 1: Low-Pass Filter Graph

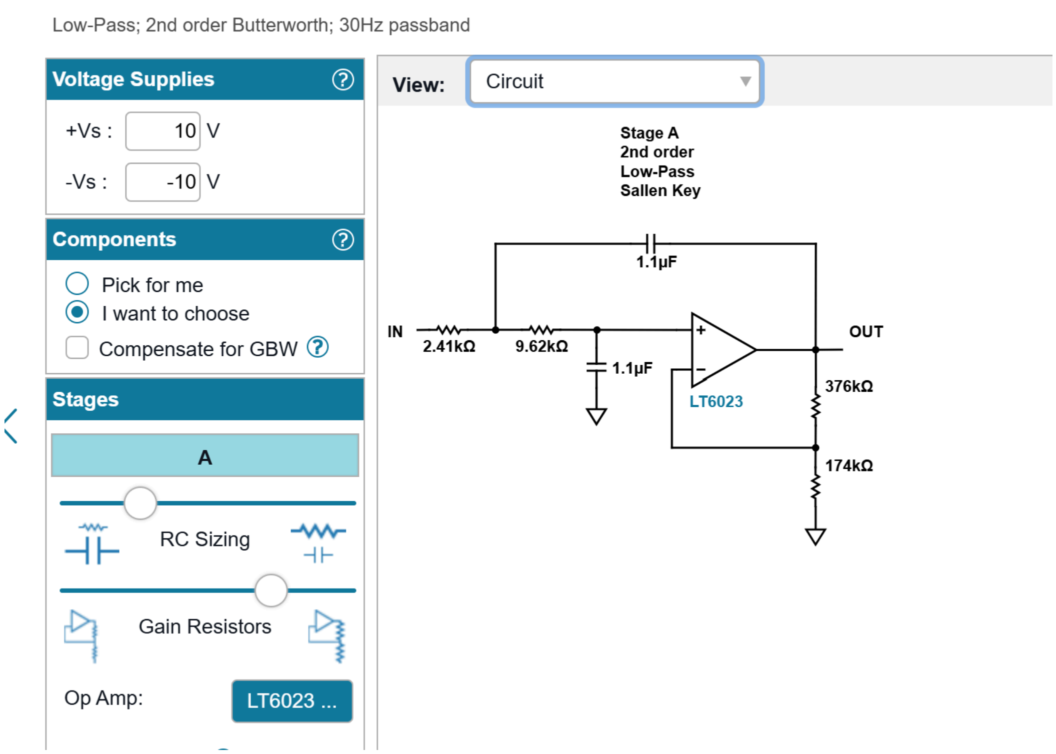

Figure 2: Low-Pass Filter Design in Filter Wizard

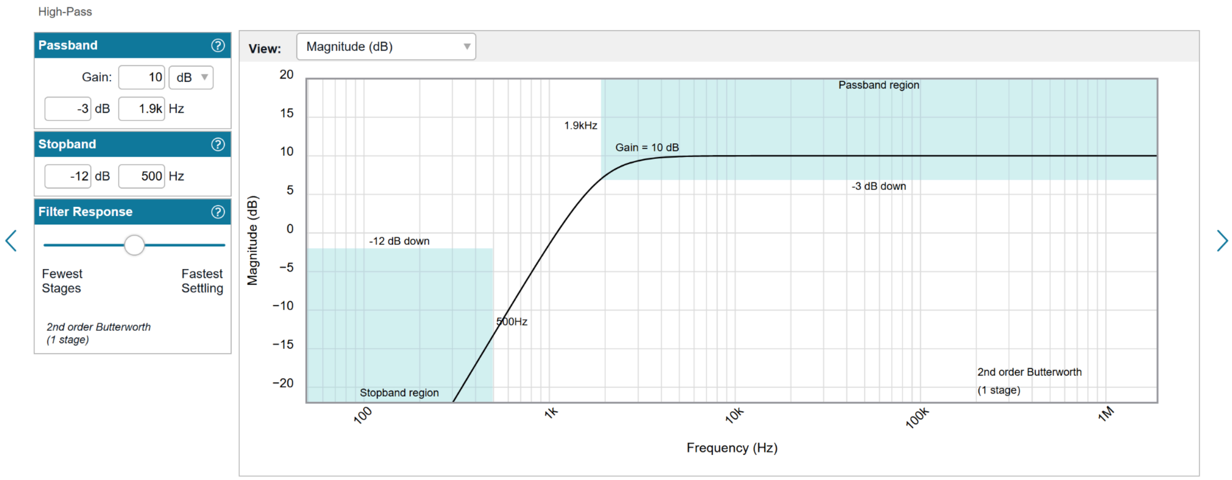

Figure 3: High-Pass Filter Design in Filter Wizard.

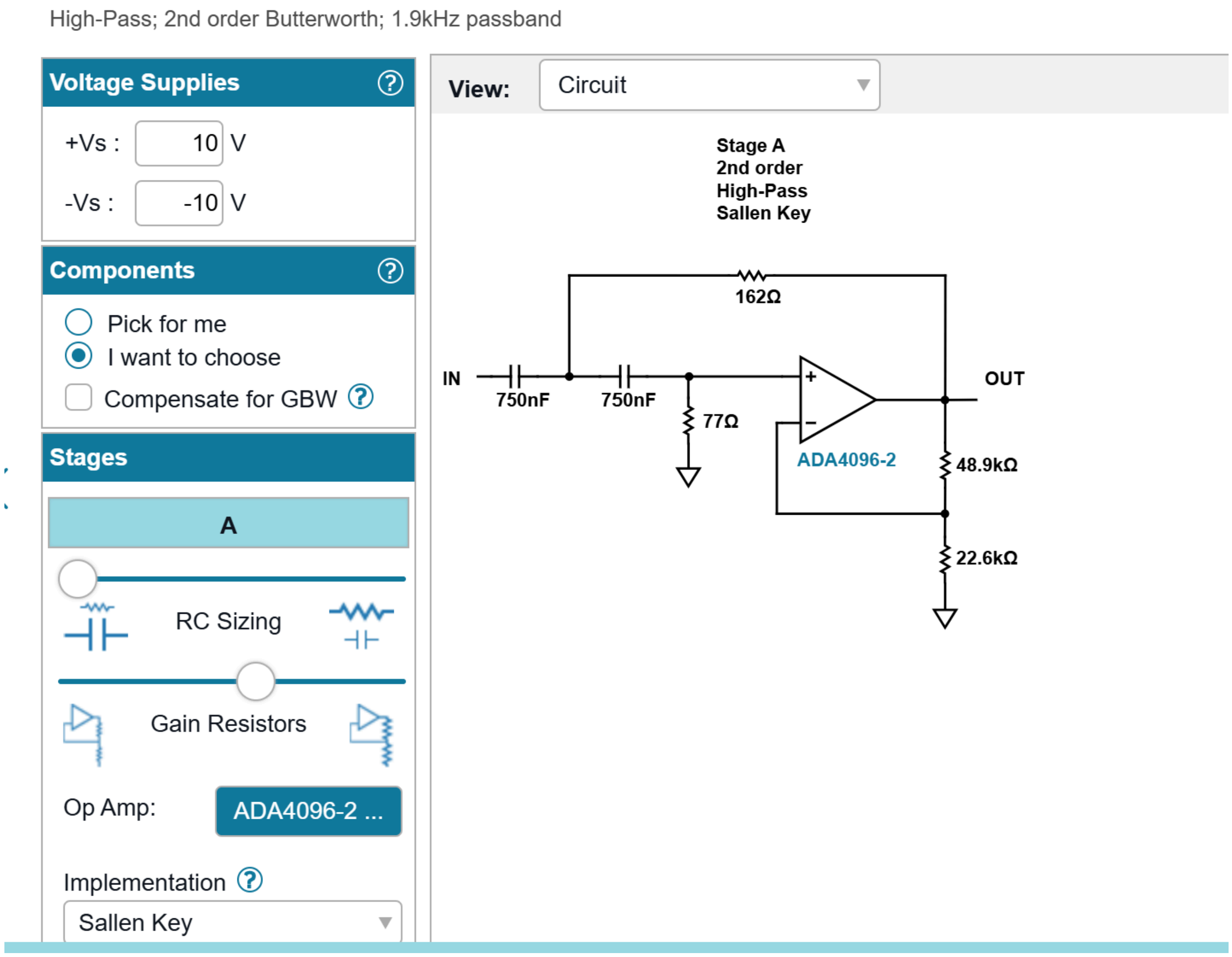

Figure 4: High-Pass Filter Graph

As shown in the images above for our Low-Pass 2nd order Butterworth filter with a 30 Hz passband we used a LT6023 Op Amp. For our High-pass 2nd order Butterworth filter with a 1.9kHz passband we used an ADA4096-2 Op Amp. We then matched up our resistors to the components available in the lab. As shown, we used two 1.1μF capacitors, a 2.4kΩ, 9.62kΩ, 376kΩ, and a 174kΩ resistor. As for our high pass filter we used 77Ω, 162Ω, 48.9kΩ, 22.6kΩ resistors and two 750nF capacitors.

After successfully generating the system using the *Analog Filter Wizard* website we then implemented it in MATLAB using the Simscape Modeling application. To begin we sent the audio file into the Simscape model as the input voltage and then exported the voltages across the speakers. This process allowed us to play them as audio signals. When building our model in Simscape we first included our amplifier circuit, then our filter designs. This was done by feeding the audio signals in the input of the amplifier circuit and then putting out output into the input of our filters. This process did result in a loading error; a low-impedance load (speakers) draws too much current from the preceding stage, altering the voltage levels and affecting signal integrity. To mitigate the loading error caused by the low-impedance speakers (3.35 Ω and 4 Ω) we implemented a buffer circuit. This is shown in the figure below depicting our completed simscape circuit.

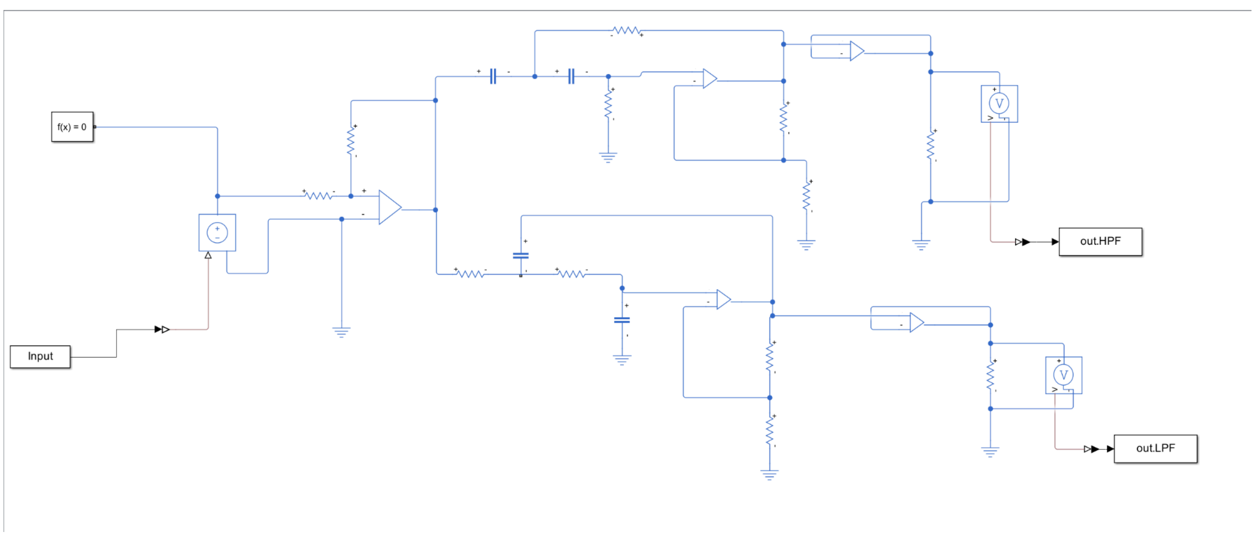

Figure 5: Circuit implemented in Simscape

When building the circuits we used the MATLAB functions to send our signals through a digital filter. The digital filter affected the output signal by modifying its frequency. This is because digital filters allow certain frequencies to pass and attenuate others. For example low-pass filters pass low frequencies blocking high ones, high-pass filters pass high frequencies blocking low ones, Band-pass filters pass a range of frequencies, and Band-stop filters block a specific range. In our case we used a woofer for our low-pass filter to let through the bass and a bandpass filter to handle the midrange speaker and frequencies.

After implementing our circuit designs in MATLAB we began the physical build. We first built our low pass filter which included using a breadboard, a LT6023 Op Amp, two 1.1μF capacitors, a 2.4kΩ, 9.62kΩ, 376kΩ, and a 174kΩ resistor. Then when building the high pass filter we used 77Ω, 162Ω, 48.9kΩ, 22.6kΩ resistors and two 750nF capacitors. To connect our computer audio to the speaker, we first plugged a male-to-male aux cable into the headphone jack of the computer. We then used grabber clips to connect to the different parts of the aux plug. The sleeve was connected to the ground of our circuit. To make the connections, we created two banana plug cables by joining banana plug ends together.

Next, we connected the output of the high-pass filter to the "HI" input on the back of the speaker and the output of the low-pass filter to the "LO" input. We also connected the ground terminals of both "HI" and "LO" to the ground of the circuit using jumper wires. If we only wanted to hear the high-frequency content, we made sure the "HI" wires were plugged in and the "LO" wires were not, and vice versa for the low frequencies.

To power our op-amps, we used a ±10 V DC power supply. Since only one banana plug could be inserted into each terminal of the power supply, we used the breadboard buses to distribute power. One bus was used for +10 V, another for -10 V, and a third for ground, allowing us to power all components using jumper wires. Before adjusting anything on the breadboard, we always made sure the DC power supply was turned off to prevent damaging components or causing injury. We only turned the power on after carefully verifying all connections were correct.

Our results met our expectations as we were able to split the audio signal into two separate signals, filtering out one to pass through the low-end portion of the speaker and the other to pass through the midrange portion of the speaker. However, if we used a different filter design our results would have been different. For example if we were to use a chebyshev, elliptic, or bessel filter, our frequency response would vary as each filter shapes the signal and even cutoff frequency differently. This is due to each of the filters unique attributes:

- **Butterworth filters** - smooth, flat passband with no ripples and a moderate roll-off. The transition between passband and stopband is gentle.
- **Chebyshev filters**- faster roll-off but introduce ripples in either the passband. Can have a nonlinear phase shift which can cause distortion in audio transients.
- **Elliptic filters**- have the sharpest roll-off but include ripples in both passband and stopband. Have nonlinear phase shifts that can cause audio distortion
- **Bessel filters-** maintain phase linearity well but have a slower roll-off. This preserves phase relationships which keeps waveform shapes more enact in audio applications.

Changing the filter could also require a change in filter order. We used a 2nd order Butterworth filter which allowed us to retrieve our desired audio output while also keeping our circuits simple. However, if we were to change our design to require a bigger order then it would increase the difficulty of the build, and introduce phase shift.

Below you will find a **Table of Filter and Amplifier Parts:**

Component Type	**Value**	**Quantity**	**Used In**
Capacitor	1.1 µF	2	Low-pass Filter
Capacitor	750 nF	1	High-pass Filter
Resistor	2.4 kΩ	1	Low-pass Filter
Resistor	9.62 kΩ	1	Low-pass Filter
Resistor	376 kΩ	1	Low-pass Filter
Resistor	174 kΩ	1	Low-pass Filter
Resistor	77 Ω	1	High-pass Filter
Resistor	162 Ω	1	High-pass Filter
Resistor	48.9 kΩ	1	High-pass Filter
Resistor	22.6 kΩ	1	High-pass Filter
Op-Amp	LT6023	1	Low-pass Filter
Op-Amp	ADA4096-2	1	High-pass Filter

Table 1: Table of Filter and Amplifier Parts

Results & Discussion

Our main objective was to design and simulate a physical system that could isolate and process audio signals using low-pass and high-pass filters. As stated above, we used Analog Filter Wizard to implement Butterworth filter designs that targets metrics such as >= 10dB amplification and <= -12 dB attenuators and used that to create Simscape simulations and a physical circuit prototype.

Our design process began by analyzing the given audio clip using MATLAB's Fast Fourier Transform (FFT). The FFT plot revealed three frequency peaks of 32.7994 Hz (bass), 184.997 Hz (mid), and 1975.56 Hz (treble). Based on this, we selected a low-pass filter cutoff of 32.8 Hz and a high-pass filter cutoff of 1975.56 Hz which ensured that the mid-frequency would be completely filtered out in both channels.

We successfully processed three audio frequencies (32.8 Hz, 184.997 Hz, and 1975.56 hz) through the filter circuits. The use of MatLab and Simscape also allowed us to accurately simulate the audio output for both low-pass (cutoff at 180 Hz) and high-pass (cutoff at 500 Hz) filters. Although we retrieved our required audio outputs we did have some errors. Some of the primary sources included component tolerances, as the standard resistor and capacitor values did not perfectly replicate the simulation specifications. This made us have to go back and try different values to get our needed output.

As for our reasoning behind our decisions we chose to use the Butterworth filter because there was no ripple in the passband. This means that the signal passing through the filter remains constant across different frequencies and that all frequencies are attenuated equally. We also chose it due to its simplicity as it has a fast frequency response, minimized variation in magnitude response, and less unnecessary distortion. In the section above you can find further reasoning behind our design choices and our specified frequency values.

Overall, this project was effective as it required us to utilize the knowledge gained during lecture and lab to design an amplifier/filter circuit. Collectively, as a group we enjoyed using the Simscape modeling software as it allowed us to prototype our design and easily test different design processes. It also helped with our circuit visualization, as there were many more electrical components in comparison to our weekly labs. Another enjoyable aspect of this project was the physical implementation and real-world audio outputs. It was satisfying to see and hear the physical outputs rather than a circuit that produces some intangible. Some more difficult parts of this lab was difficulty in using new electrical systems like the speakers. We never used something like that in our previous labs, so we were very novice in using it.

We did notice some similarities and differences in both applications. When implementing the physical circuit vs in Simscape we noticed that some resistors and capacitors were not exact but within a reasonable range and that the output noise was not exactly the same despite implementing the same circuit design. In addition when implementing in simscape we found it difficult to create a path for the audio file, while when doing the physical circuit we found problems in varying values in resistors and capacitors.

Appendices & References

This section includes our citations as well as a picture of our amplifier and filter in Simscape, our physical circuit build, and our audios on YouTube.

MathWorks. Simscape. MATLAB, The MathWorks, Inc., www.mathworks.com/products/simscape.html. Accessed Week 9 April 2025.

Texas Instruments. Analog Filter Designer. Texas Instruments Incorporated, https://webench.ti.com/filter-designer/. Accessed Week 9 April 2025.

Picture of built amplifier and filter circuit (physical)

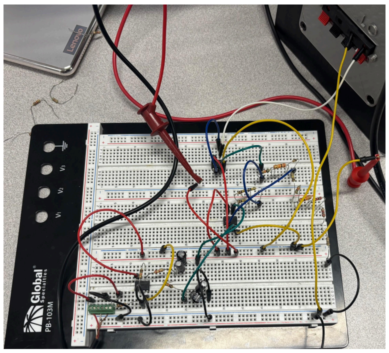
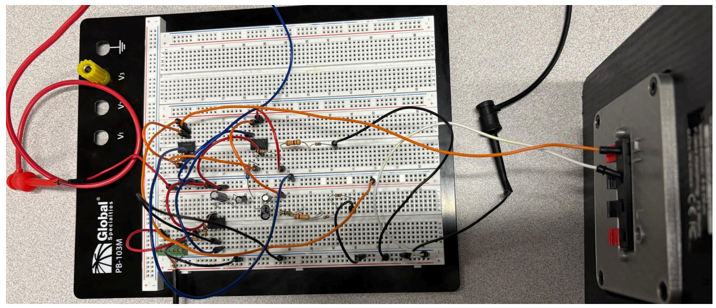

Figure 6 and 7- Low-pass and High-pass physical circuit

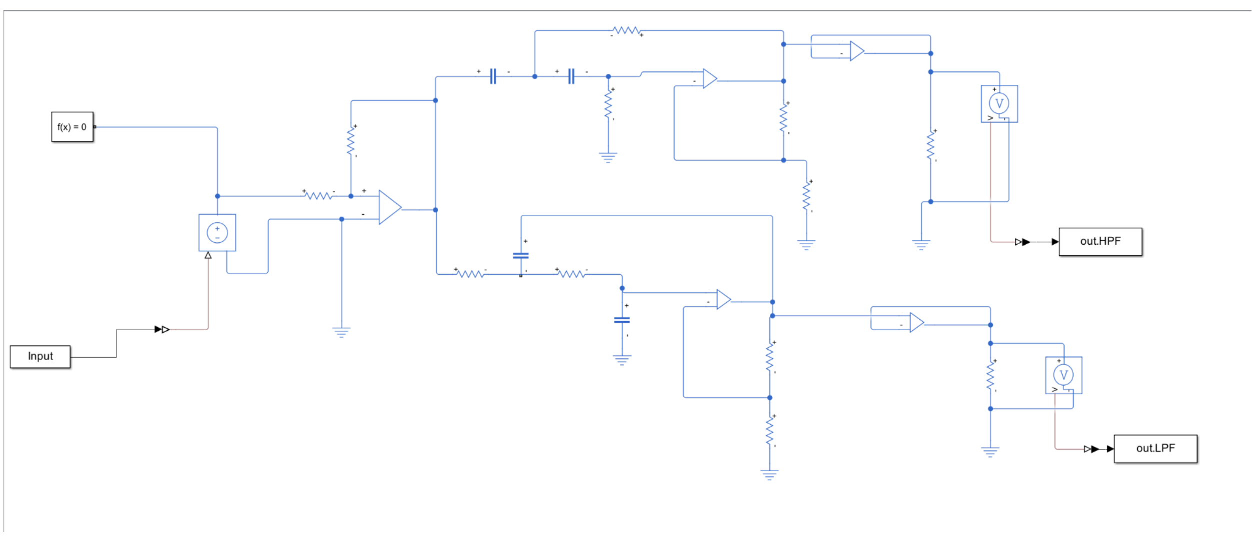

Figure 8- Simscape Model

https://youtu.be/rbggVIy9WNM?si=6fBf8jzP3ttSPSMv
First audio: LPF second audio: HFP
Above you can find the link to our YouTube video showing the physical circuit working with the speaker.

Team breakdown:
Physical and Simscape Modeling: Alix and Mina
Presentation and Final Report: Melanie and Kylee