Portland State University
Maseeh College of Engineering and Computer Science
Electrical and Computer Engineering Department
EE223 - Circuits Analysis III
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Final Project

Music-Driven Light Display

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Abstract

This report outlines the comprehensive design, simulation, construction, and testing of a frequency-selective musical light show system. The primary objective of this project was to create an innovative display that dynamically synchronizes with music by illuminating various color lights based on the frequency content of the audio input. The system utilizes standard audio devices as input sources and offers user-adjustable light sensitivity for each channel. Through the utilization of basic components such as switches, potentiometers, operational amplifiers, resistors, capacitors, and LEDs, the project aims to achieve an engaging visual representation of music without relying on microcontrollers. This report details the project's objectives, design process, simulation results, experimental findings, teamwork dynamics, and conclusions drawn from the implementation process. The project's success is evaluated based on the design's accuracy, simulation outcomes, experimental outcomes, and its potential for further development. The report concludes with reflections on lessons learned and areas for future enhancements. The project encourages creativity, teamwork, and an interdisciplinary approach, combining electrical engineering principles with music to create an aesthetically pleasing and interactive experience.

1. Introduction

the Music-Driven Light Display project set out on a journey that resulted in an intriguing combination of auditory and visual aspects. The core of this project is to investigate sound frequencies inside a chosen musical composition, which is illustrated in *Figure 1*. This graphic demonstrated how sound frequencies were divided into three primary groups: low frequencies (0–1000 Hz), medium frequencies (1100–8000 Hz), and high frequencies (9000 Hz) for a selected song of analysis.

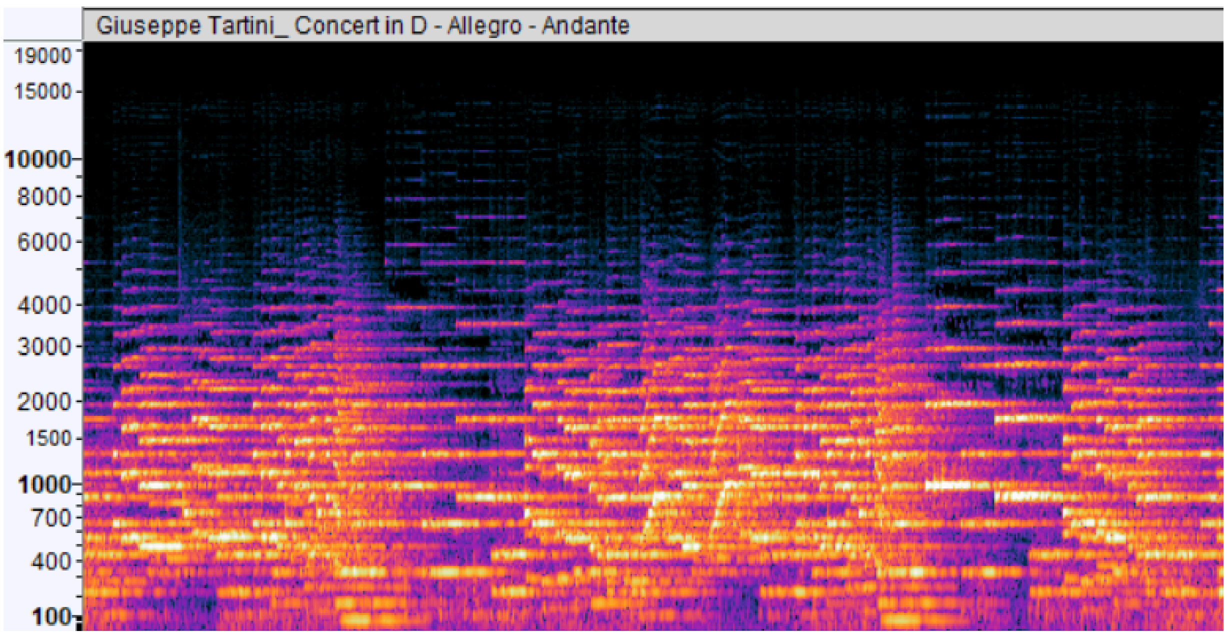


Figure 1: Testing song frequency spectrogram analysis

1.1. Block diagram

As shown in *Figure 2*, this was guided by a highly detailed design that was graphically depicted using LUCID drawing. A methodical series of steps was drawn out in this architectural map, providing a seamless conversion of auditory signals into dynamic visual displays. The project began by utilizing the LM386 operational amplifier, which it used to amplify sound and broadcast it via speakers. A 10k potentiometer, which functions as a volume knob, was interestingly integrated into this system. This development made it easier to precisely adjust the speaker's output volume. After that, modified audio waves are separated using specialized filters resembling selective gates. Three types of Butterworth 2nd Order filters were used: low-pass, band-pass, and high-pass. Each category attenuated some frequency ranges of audio transmissions while allowing others to pass. The synchronization of the auditory and visual components was noteworthy. The filters converted audio frequencies into dynamic visual cues by causing Light Emitting Diodes (LEDs) to glow in different colors. The lack of additional amplification steps for this translation, stressing design effectiveness, was an intriguing characteristic.

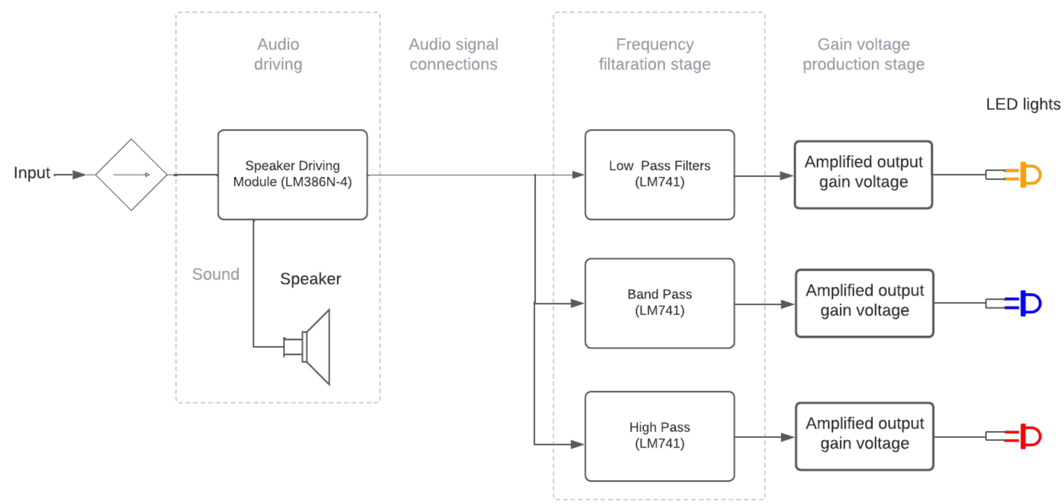


Figure 2: Project Design Block Diagram

1.2. Speaker module design

A remarkable aspect of the project is the addition of a 10k potentiometer into the circuitry, using the LM386 operational amplifier once more as showing in *Figure 3*. This integration, when placed carefully before connecting to certain amplifier pins, enabled loudspeaker volume manipulation without changing the signals intended for further filtration. The project's goal to smoothly integrate technology innovation and creative expression was demonstrated by this considerate inclusion.

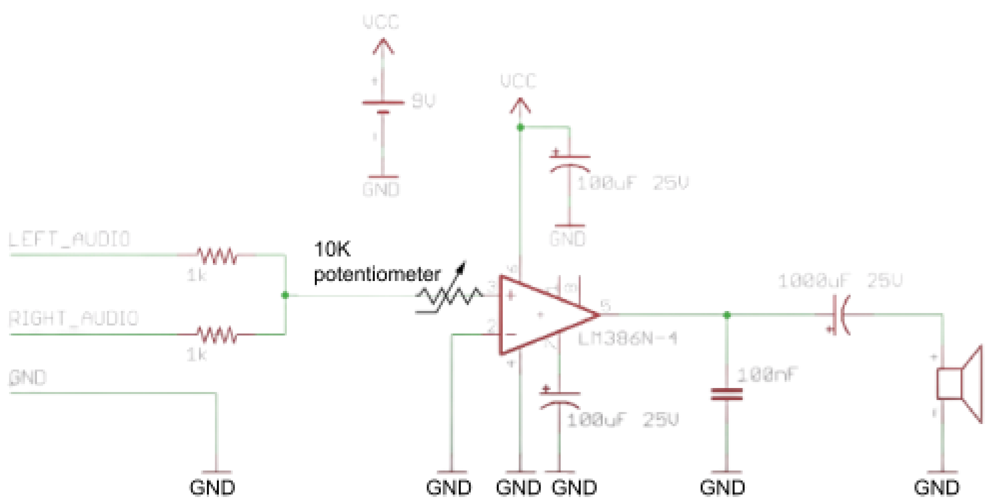
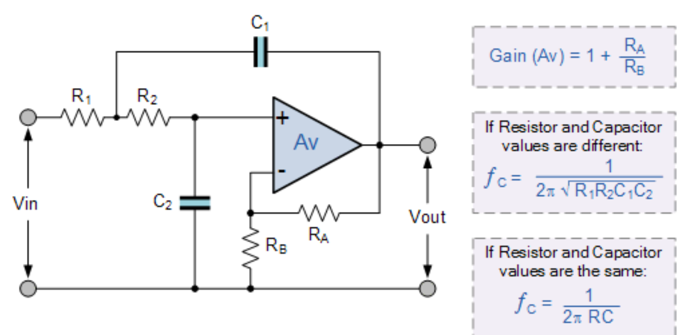


Figure 3: Sound speaker module circuit with a variable resistor

1.3. *Filters design*

1.3.1. *Low Pass*

Introducing the concept of filter design, particularly the Low Pass filter, which plays a vital role in signal processing and communication systems. A Non-inverting Butterworth Second-Order Low Pass filter allows signals with frequencies lower than a certain cutoff frequency to pass through while attenuating higher frequencies. The transfer function for this design is given by:



$$H(s) = k \cdot \frac{1}{s^2 + p + S + 1}$$

Where 's' is the complex frequency variable, 'k' and 'p' are constants, and the filter is designed using the principles of the Butterworth filter family. The component scaling prototype employs specific resistor and capacitor values: R1 27= k ohms, R2 = 3.8 k ohms, C1 = 22 nF, and C2 = 10 nF. The cutoff frequency, denoted as 'fc,' is a critical parameter that defines the point at which the filter starts attenuating frequencies. For this design, the cutoff frequency is calculated using the formula:

$$f_c = \frac{1}{2\pi R_1 R_2 C_1 C_2}$$

Resulting in a value of approximately 1059.34 Hz. This filter configuration showcases the fundamental principles underlying filter design and its applications in signal manipulation, with the added characteristic of a non-inverting Butterworth second-order response.

Component Scaling prototype:

$$R_1 = 27\,k\Omega,\, R_2 = 3.8\,k\Omega,\, C_1 = 22\,nF,\ C_2 = 10\,nF$$

$$= \frac{1}{2\pi(27\times10^{3})(3.8\times10^{3})(22\times10^{-9})(22\times10^{-9})}$$

$$= 1059.34\,Hz$$

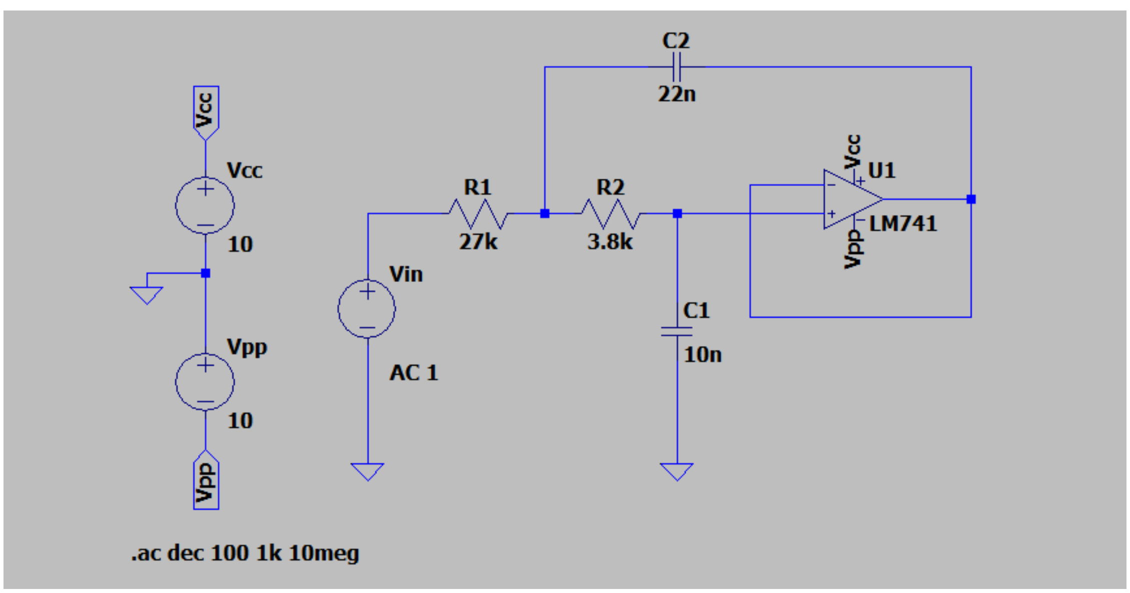


Figure 4: Non-inverting Butterworth Second-Order Low Pass filter

1.3.2. Band Pass

A Band Pass filter permits a specific range of frequencies to pass through while attenuating others. The transfer function for this design is characterized by its selective frequency response. Its component scaling prototype includes resistor and capacitor values. The filter's center frequency, denoted as 'fc,' determines the range of frequencies it allows. With a central calculated cutoff frequency of approximately 5578.24 Hz, lower frequency of 1066 Hz, and higher of 9748.24 Hz

$$H(s) = k \, . \, \frac{s^2}{s^2 + p + S + 1}$$

Component Scaling prototype:

$$R_1 = 3.8 \, k\Omega, \, R_2 = 20 \, k\Omega, \, R_3 = 15 \, C_1 = 22 \, nF, \, C_2 = 10 \, nF$$

Cut-off frequency:

$$\omega_c = \frac{1}{\sqrt{\frac{1}{R_3 C_2 C_1} \times (\frac{1}{R_1} + \frac{1}{R_2})}}$$

$$= \frac{1}{\sqrt{\frac{1}{(15\times10^3)(22\times10^{-9})(10\times10^{-9})} \times (\frac{1}{3.7\times10^3} + \frac{1}{20\times10^3})}}$$

$$= 35049.11 \, rad/s$$

$$f_c = \frac{371134.80}{2\pi}$$

$$= 5578.24 \, Hz$$

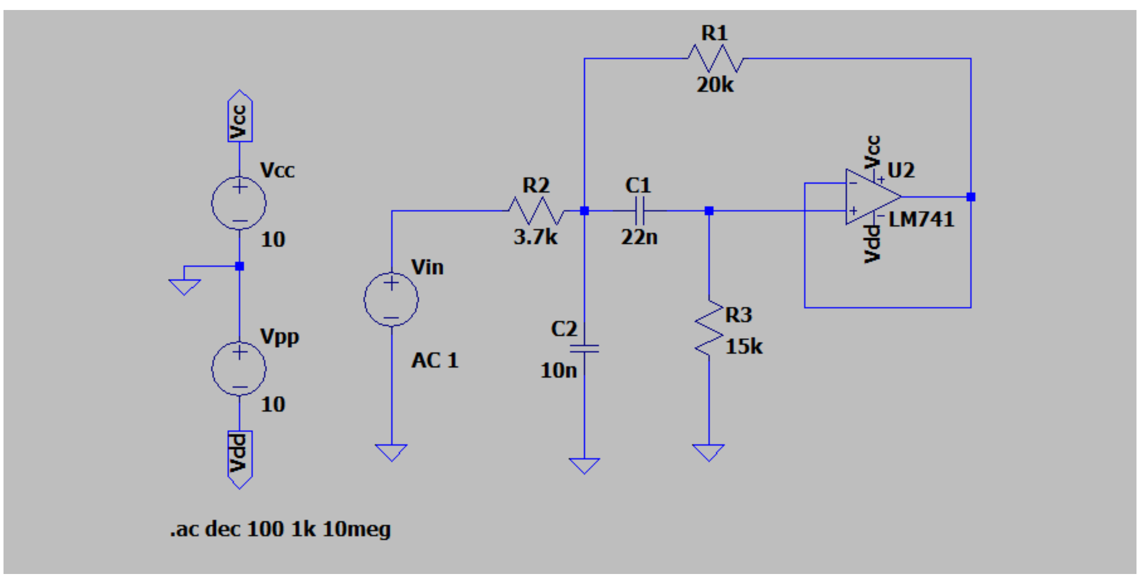
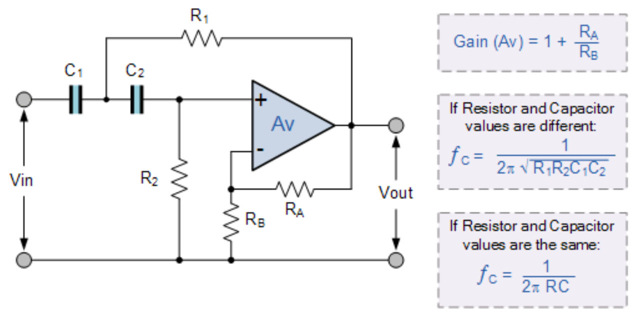


Figure 5: Non-inverting Butterworth Second-Order Band Pass filter

1.3.3. High Pass

Introducing the High Pass filter design—a key tool in signal processing. The High Pass filter allows frequencies above a specific cutoff while reducing lower frequencies. The transfer function is:

$$H(s) \;=\; k\,.\;\frac{s^2}{s^2 + p + S + 1}$$



Component Scaling prototype:
$R_1 \;=\; 1\,k\Omega,\, R_1 \;=\; 33\;\Omega,$
$C_1 \;=\; 22\,nF,\, C_2 \;=\; 1\,nF$

Cut-off frequency:

$$\omega_c \;=\; \frac{1}{\sqrt{R_1R_2C_1C_2}}$$
$$=\; \frac{1}{\sqrt{(1\times10^3)(33)(22\times10^{-9})(1\times10^{-9})}}$$
$$=\; 9851.47\; rad/s$$

$$f_c \;=\; \frac{371134.80}{2\pi}$$
$$=\; 9400.95\; Hz$$

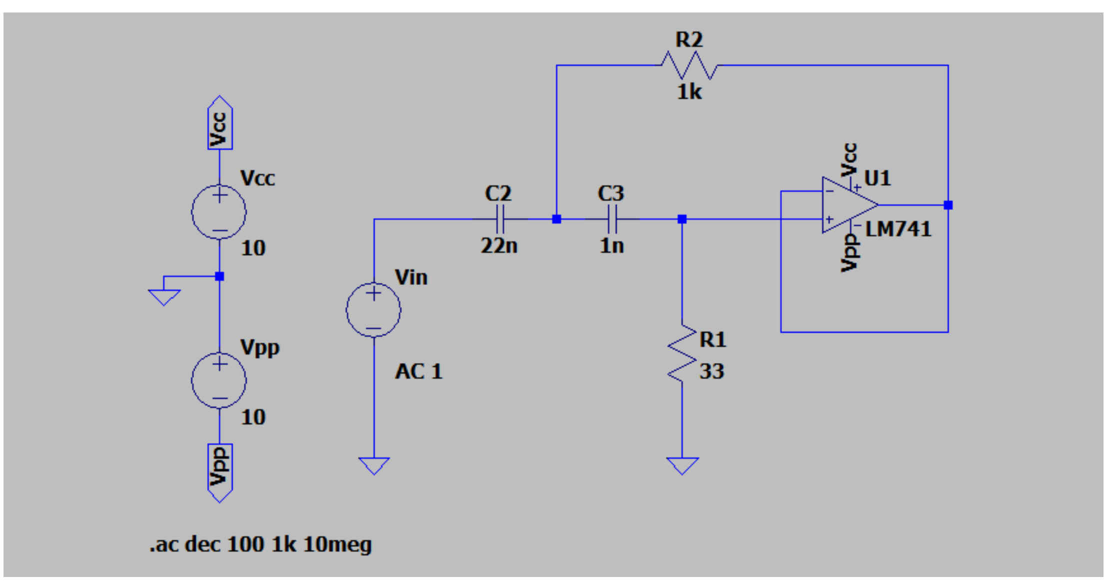


Figure 6: Non-inverting Butterworth Second-Order High Pass filter

2. Simulation results

Below MATLAB code sets generate Bode plots showcasing the frequency responses of three Butterworth 2nd Order filters. These filters—Low-pass, Band-pass, and High-pass—have varying component values and cutoff frequencies. The Bode plots illustrate the filters' behavior, clearly depicting their respective frequency passbands and responses. The MATLAB simulations effectively visualize how these filters manipulate signal frequencies for specific applications. Moreover all the filters were simulated on LTspice software to illustrate the frequency response of each filters

2.1. Bode Plots

```
% Given component values
R1 = 27e3; % Resistance R1 in ohms
R2 = 3.8e3; % Resistance R2 in ohms
C1 = 2.2e-9; % Capacitance C1 in farads
C2 = 2.2e-9; % Capacitance C2 in farads
% Calculate parameters for the transfer function
R = R1; % Use R1 for the resistor value (equal resistors)
C = C1; % Use C1 for the capacitor value (equal capacitors)
k = 1 + R2 / R1; % Gain factor
p = 1 / (R * C); % Pole frequency
% Transfer function in Laplace domain
s = tf('s'); % Create the Laplace variable 's'
h = k * (1 / (s^2 + p * s + 1)); % Transfer function G(s)
% Cut-off frequency and quality factor
fc = 589.46275219221; % Cut-off frequency in Hz
Q = 0.5; % Quality factor
% Bode plot
figure; % Create a new figure for the Bode plot
bode(h); % Plot the Bode plot of the transfer function
hold on; % Hold the plot for adding additional lines
plot([fc fc], [-100 20*log10(k)], '--r'); % Plot vertical line for cut-off frequency
plot([fc/Q fc/Q], [-100 20*log10(k) - 3], '--g'); % Plot vertical line for -3dB point
legend('Bode Plot', 'Cut-off Frequency', '-3dB Point');
grid on; % Add a grid to the plot for better visualization
title('Bode Plot of Butterworth 2nd Order Low-pass Filter'); % Set the title of the plot
xlabel('Frequency (Hz)'); % Label the x-axis as Frequency
ylabel('Magnitude (dB)'); % Label the y-axis as Magnitude (in dB)
```

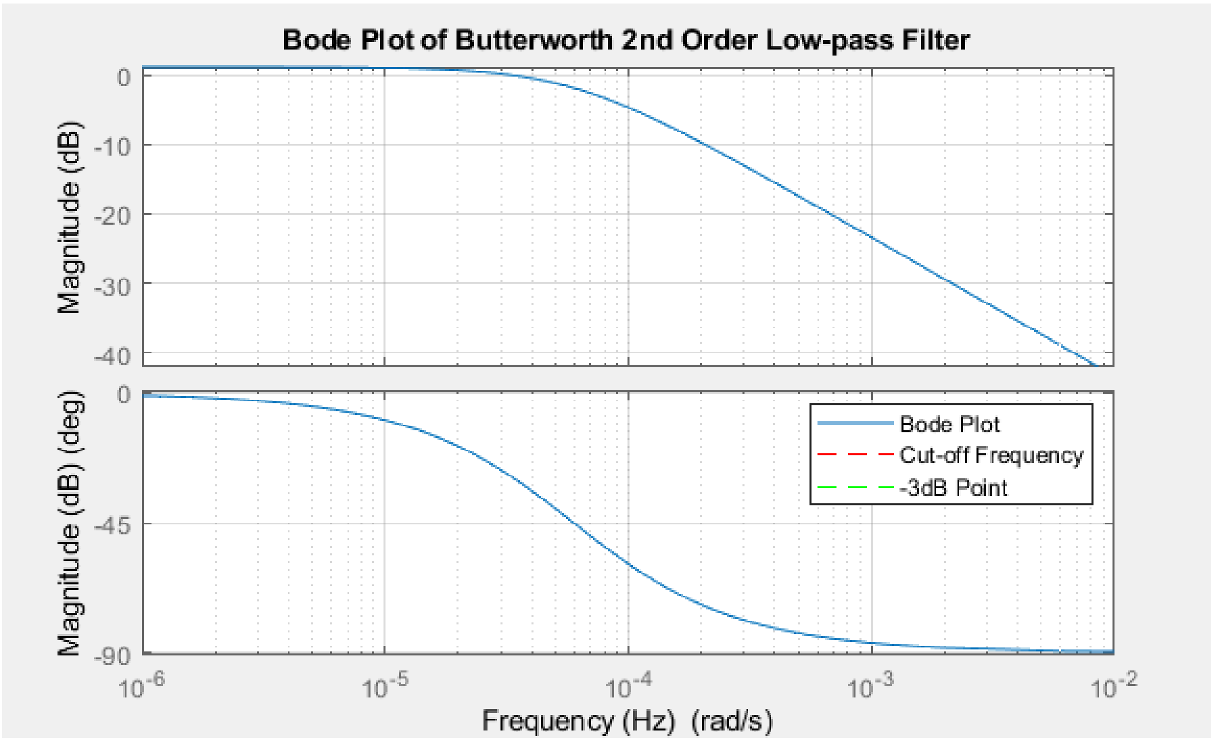


Figure 7: Low pass filter bode plot using MatLab

```
% Given component values
R1 = 3.7e3; % Resistance R1 in ohms
R2 = 20e3; % Resistance R2 in ohms
C1 = 2.2e-9; % Capacitance C1 in farads
C2 = 10e-9; % Capacitance C2 in farads
% Calculate parameters for the transfer function
R = R1; % Use R1 for the resistor value (equal resistors)
C = C1; % Use C1 for the capacitor value (equal capacitors)
k = R2 / R1; % Gain factor
p = 1 / (R * C); % Pole frequency
% Transfer function in Laplace domain
s = tf('s'); % Create the Laplace variable 's'
h = k * (s / (s^2 + p * s + 1)); % Transfer function G(s)
% Cut-off frequency and quality factor
fc = 12242.687930146; % Cut-off frequency in Hz
Q = 0.5; % Quality factor
% Bode plot
figure; % Create a new figure for the Bode plot
bode(h); % Plot the Bode plot of the transfer function
hold on; % Hold the plot for adding additional lines
plot([fc fc], [-100 20*log10(k)], '--r'); % Plot vertical line for cut-off frequency
plot([fc/Q fc/Q], [-100 20*log10(k) - 3], '--g'); % Plot vertical line for -3dB point
legend('Bode Plot', 'Cut-off Frequency', '-3dB Point');
grid on; % Add a grid to the plot for better visualization
title('Bode Plot of Butterworth 2nd Order Band-pass Filter'); % Set the title of the plot
xlabel('Frequency (Hz)'); % Label the x-axis as Frequency
ylabel('Magnitude (dB)'); % Label the y-axis as Magnitude (in dB)
```

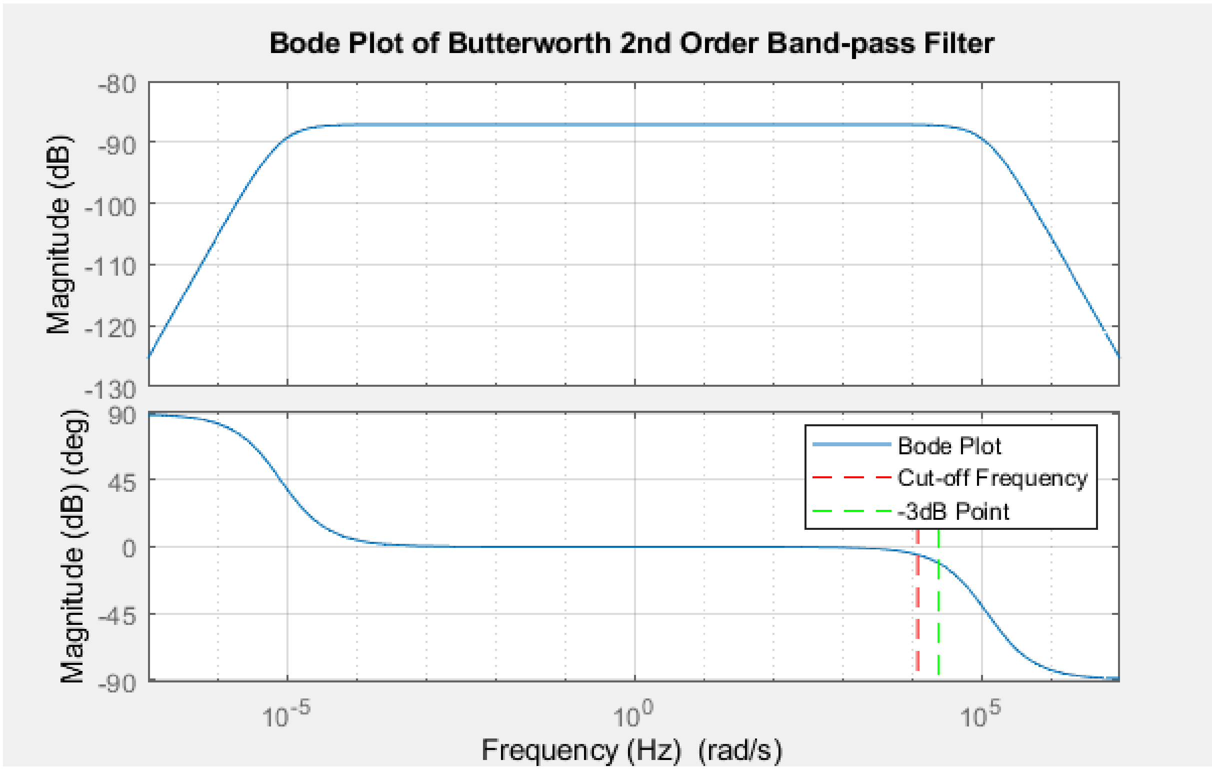


Figure 8: Band pass filter bode plot using MatLab

```
% Given component values
R1 = 120e3; % Resistance R1 in ohms
R2 = 55e3; % Resistance R2 in ohms
C1 = 47e-9; % Capacitance C1 in farads
C2 = 1e-9; % Capacitance C2 in farads
% Calculate parameters for the transfer function
R = R1; % Use R1 for the resistor value (equal resistors)
C = C1; % Use C1 for the capacitor value (equal capacitors)
k = 1 + R2 / R1; % Gain factor
p = 1 / (R * C); % Pole frequency
% Transfer function in Laplace domain
s = tf('s'); % Create the Laplace variable 's'
h = k * (s^2 / (s^2 + p * s + 1)); % Transfer function G(s)
% Bode plot
figure; % Create a new figure for the Bode plot
bode(h); % Plot the Bode plot of the transfer function
hold on; % Hold the plot for adding additional lines
plot([fc fc], [-100 20*log10(k)], '--r'); % Plot vertical line for cut-off frequency
plot([fc/Q fc/Q], [-100 20*log10(k) - 3], '--g'); % Plot vertical line for -3dB point
legend('Bode Plot', 'Cut-off Frequency', '-3dB Point');
grid on; % Add a grid to the plot for better visualization
title('Bode Plot of Butterworth 2nd Order High-pass Filter'); % Set the title of the plot
xlabel('Frequency (Hz)'); % Label the x-axis as Frequency
ylabel('Magnitude (dB)'); % Label the y-axis as Magnitude (in dB)
```

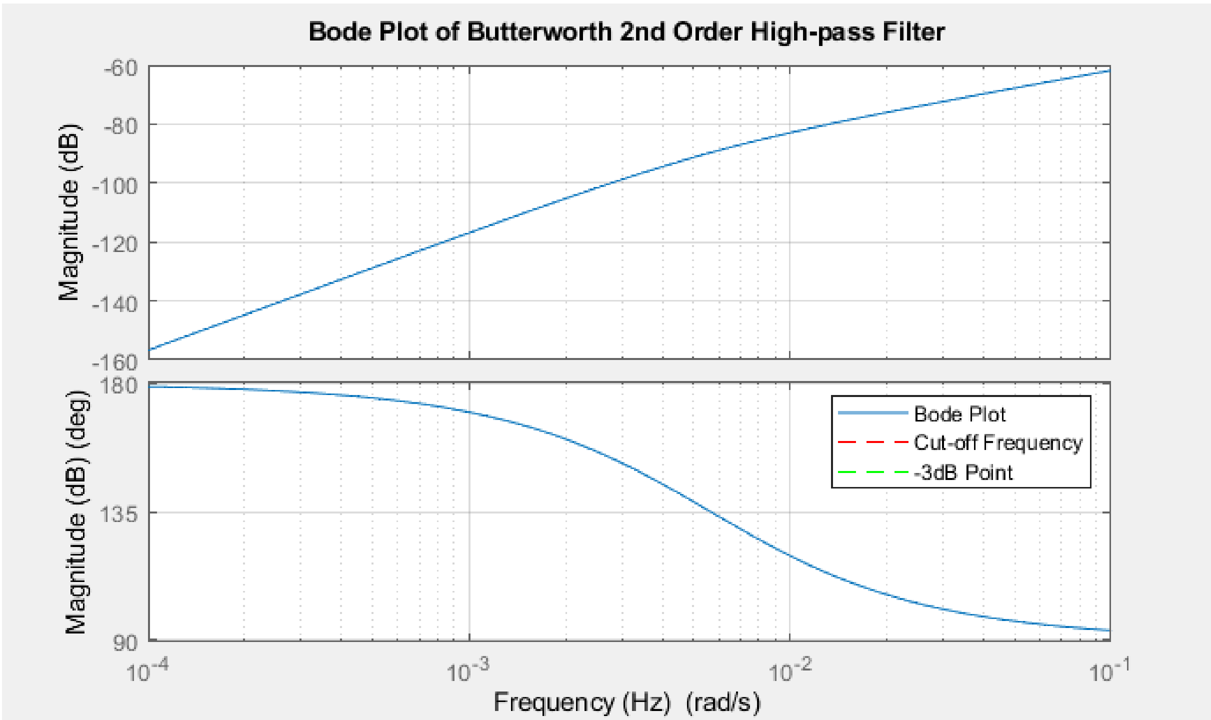


Figure 9: *High pass filter bode plot using MatLab*

2.2. Frequency response

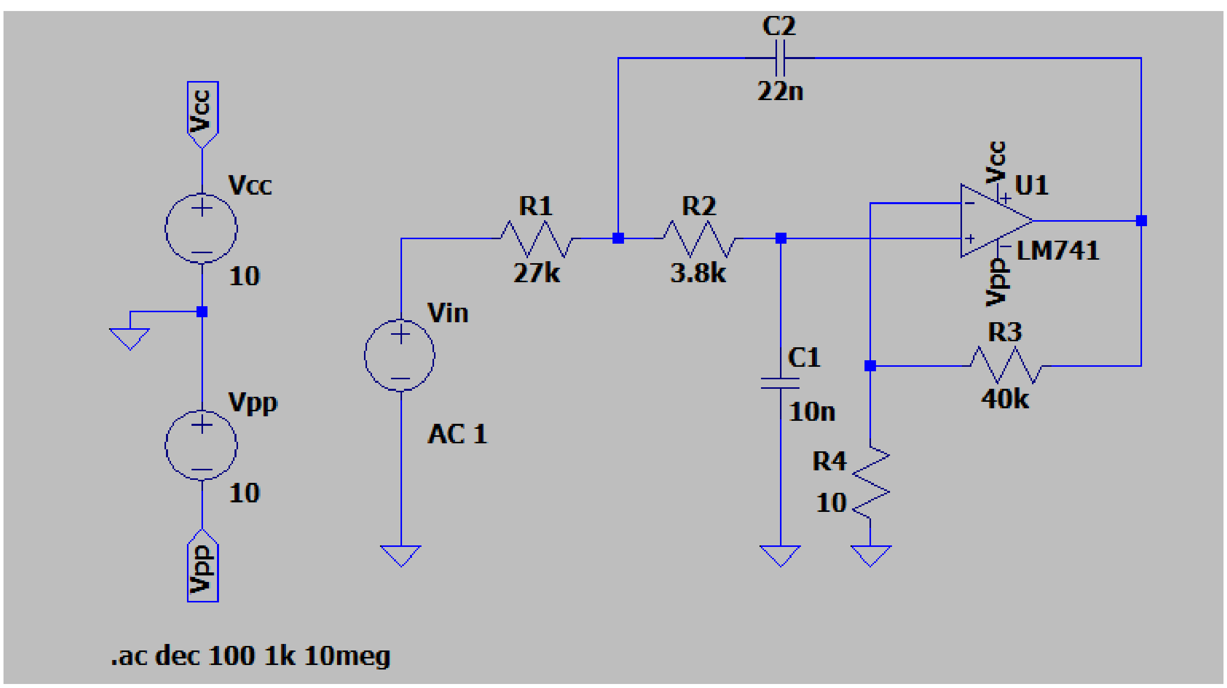


Figure 10: Non-inverting Butterworth Second-Order Low Pass filter with voltage gain

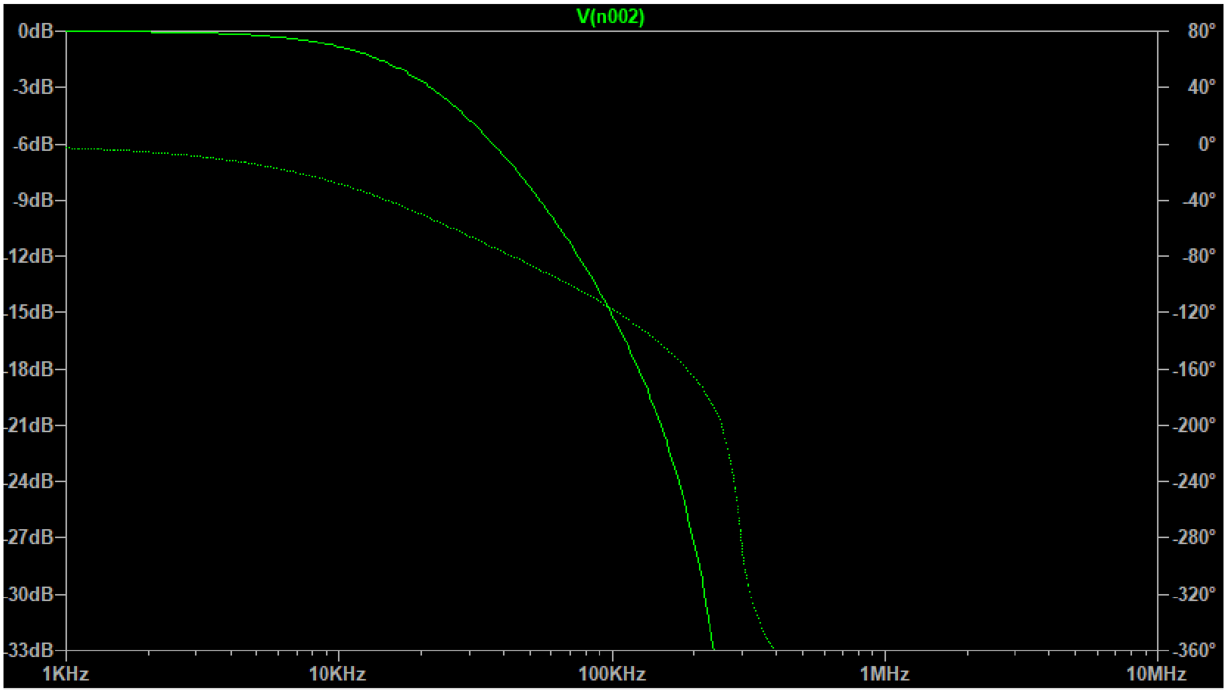


Figure 11: Frequency response output for Low Pass filter

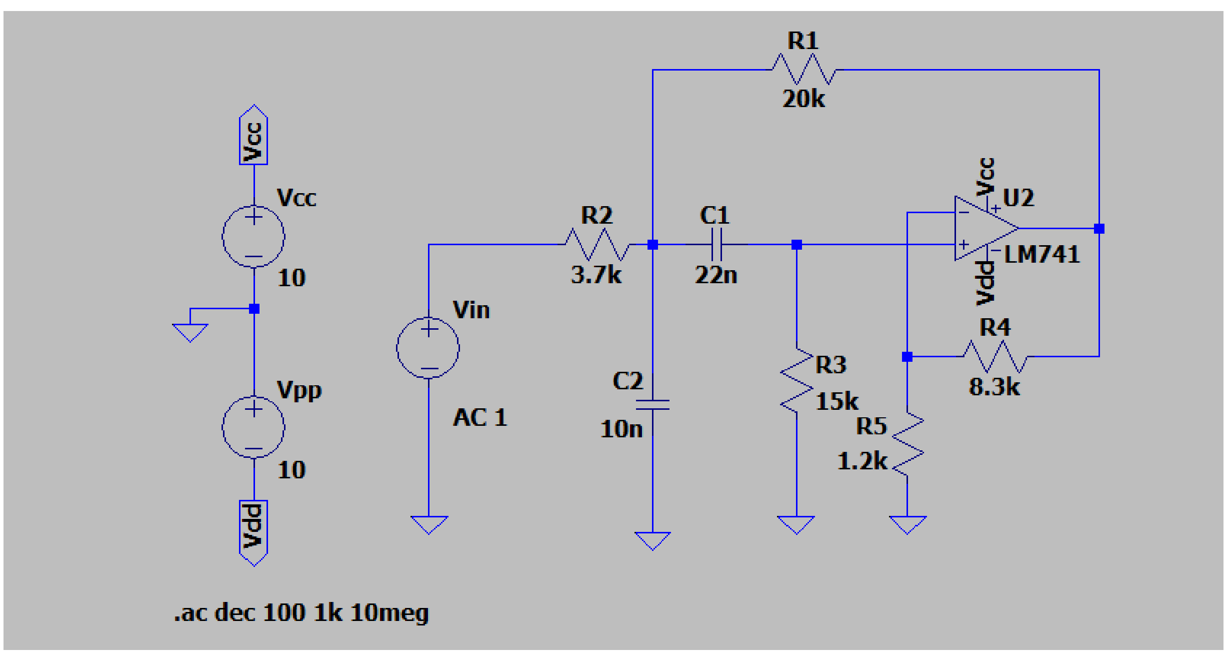


Figure 12: Non-inverting Butterworth Second-Order Band Pass filter with voltage gain

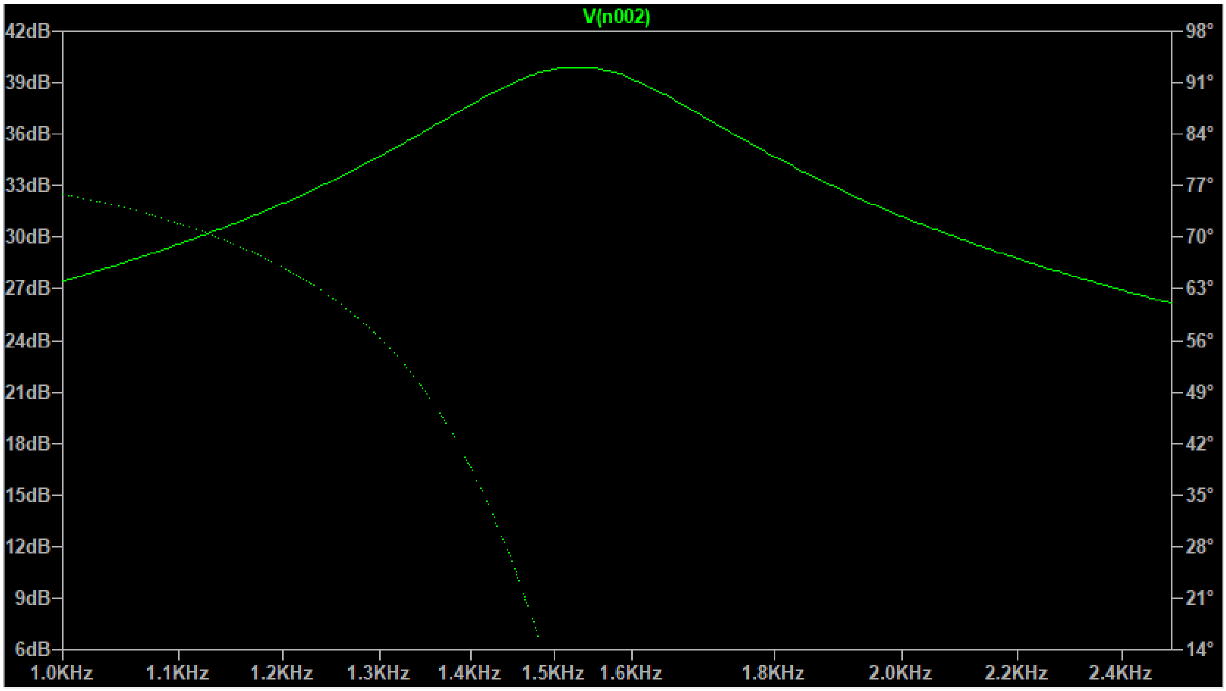


Figure 13: Frequency response output for Band Pass filter

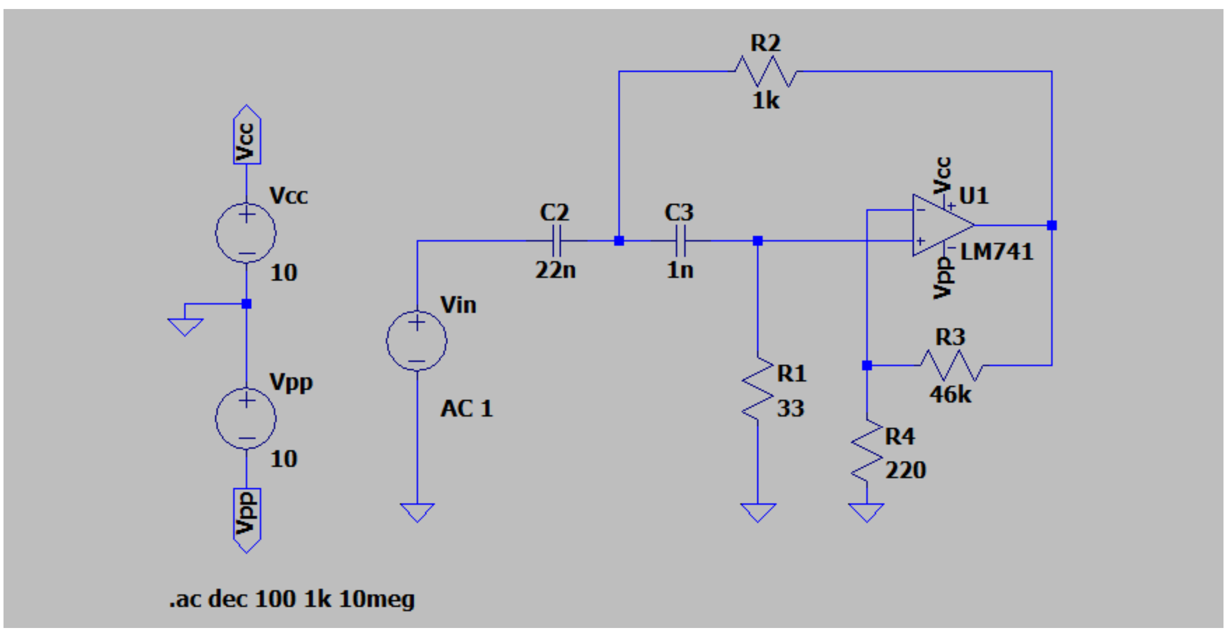


Figure 14: Non-inverting Butterworth Second-Order High Pass filter with voltage gain

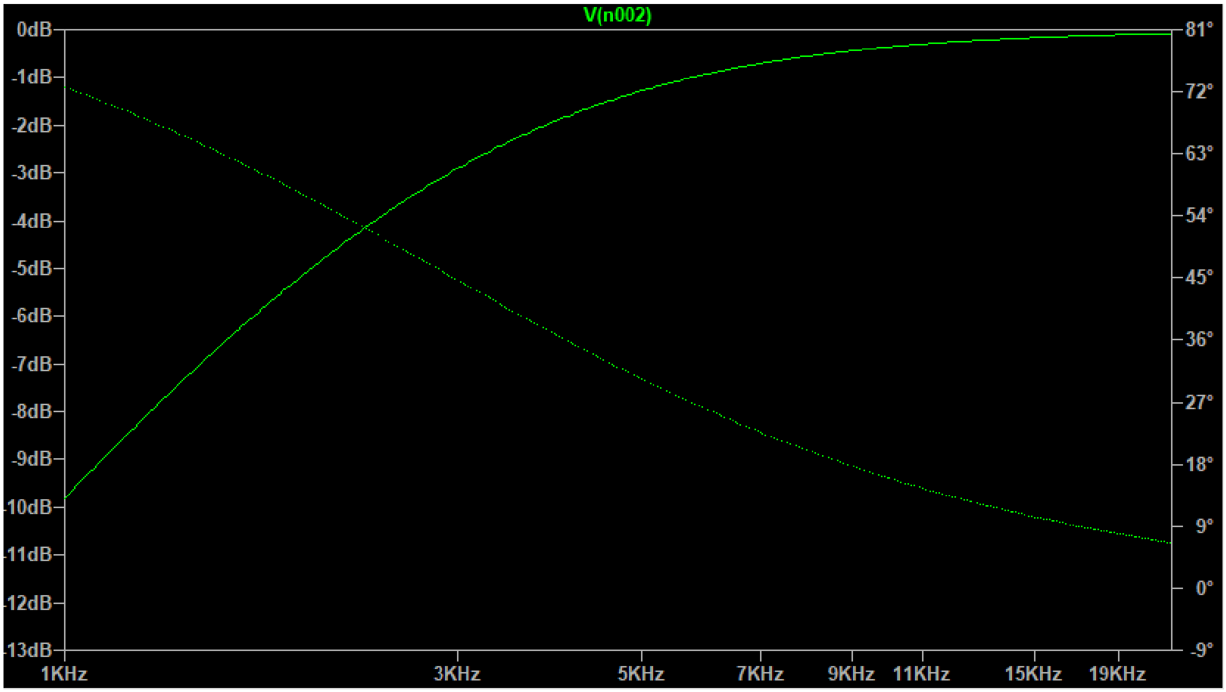


Figure 15: Frequency response output for High Pass filter

3. Test results

3.1. Breadboard circuit

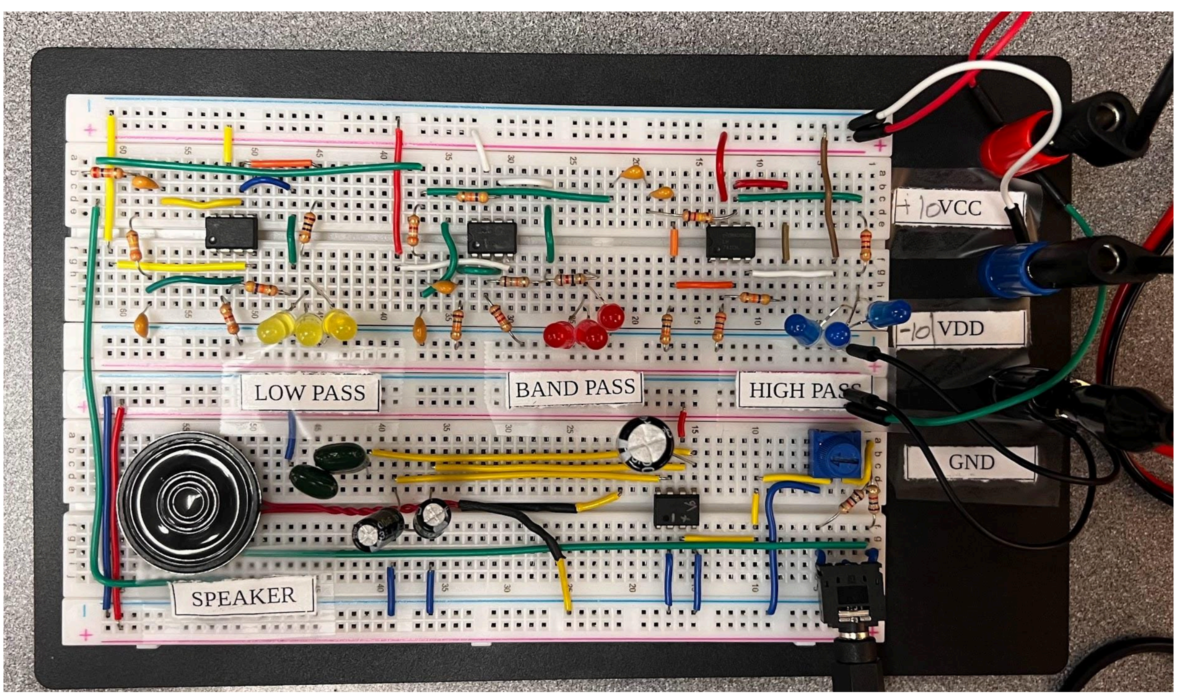


Figure 16: Complete design circuitry

3.2. Demonstration

The project's circuit design underwent several iterations. In the initial version, each filter had an amplification stage to tailor the sine wave and yield gain voltage for the corresponding LED. However, after extensive research and discussions with fellow groups, I realized the benefits of overhauling all three filter circuits. Opting to utilize a non-inverting op-amp to generate the gain voltage proved advantageous. This choice streamlined the design, sidestepped complexities, and mitigated potential noise issues between connections on the breadboard.

3.2.1. Frequency sweep

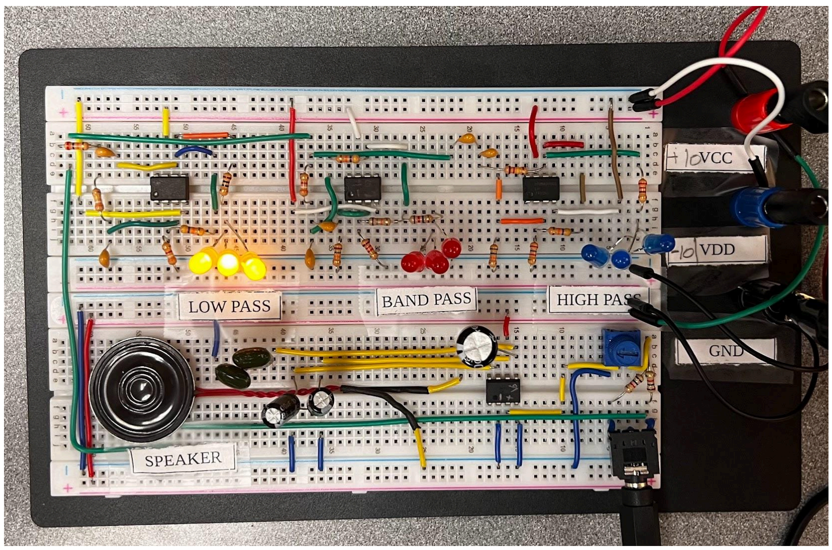


Figure 17: Circuit with resulted Low Pass frequency sweep

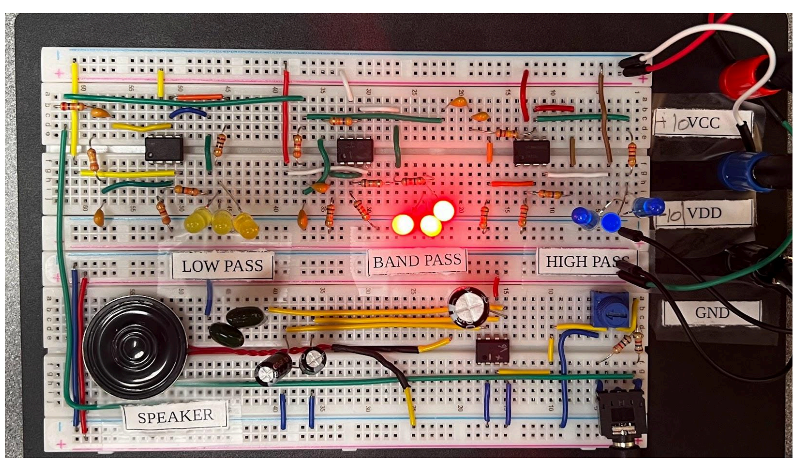


Figure 18: Circuit with resulted Band Pass frequency sweep

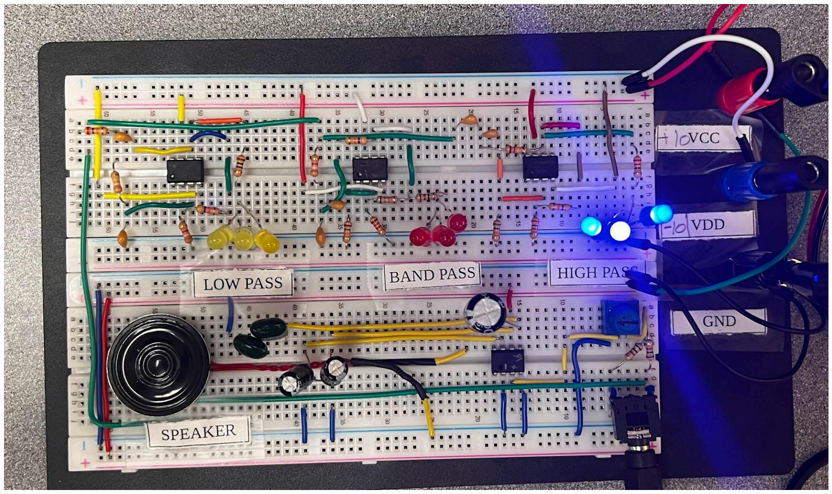


Figure 19: Circuit with resulted High Pass frequency sweep

4. Discussion

The decision to redesign all three filter circuitries to solely employ non-inverting op-amps to generate the gain voltage proved to be a more optimal solution. This adjustment not only improved the overall design's efficiency but also alleviated potential design complexities. Furthermore, it offered the advantage of reducing the chances of encountering noise interference between connections on the breadboard. In terms of potential errors, the initial design with individual amplification stages introduced complexities that might lead to signal distortion or incorrect gain levels. Additionally, these separate amplification stages could potentially introduce noise and signal coupling issues between components. This configuration might also demand more components and consume additional power, making the circuit more intricate. By transitioning to a uniform non-inverting op-amp setup, these potential errors were mitigated. The streamlined design reduced the possibilities of component mismatch and enhanced the overall signal integrity. Moreover, by avoiding excessive amplification stages, the risk of signal saturation or distortion was curtailed.

5. Teamwork

I worked on the project primarily, covering various tasks. Such as designing the sound speaker circuit, using LTSpice for simulations, doing calculations with MATLAB, and building the circuit on a breadboard. Initially, I was paired with another student engineer, but they dropped the course early on. This meant I had to put in double the effort to handle all the work on my own.

6. Conclusion

In conclusion, the Music-Driven Light Display project stands as a successful fusion of art and technology, culminating in captivating auditory and visual synergy. The integration of a 10k potentiometer, and implementing non-inverting 2nd order filters showcased a balanced approach to innovation and artistic expression, while the adept design of specialized filters demonstrated a newfound skill in manipulating sound frequencies for different effects. The harmonious synchronization of audio and visual elements through LEDs revealed an efficient and precise design strategy. This project not only achieved its creative goals but also equipped me with the ability to construct filters and comprehend the core functions of operational amplifiers. It exemplifies the transformative possibilities that arise when technology and creativity coalesce in interdisciplinary endeavors.