EE 233 Circuit Theory
Lab 4: Second-Order Filters

Table of Contents

Table of Figures

1 Introduction

This lab is designed to teach students how to design filters from a given topology and specifications, analyze the characteristics of the designed filters, measure the characteristics of the designed filter, and complete the entire audio mixing console.

Filter design is the process of creating a signal processing filter that satisfies a set of design specifications. The purpose of the design is to develop a type of filter that meets each of the requirements to a sufficient degree in order to make it useful.

The filters that are built in this lab are part of an equalizer system. Connect these filters to the buffer and output summing amplifier and you will have a three-channel equalizer. The equalizer is used to alter the frequency response of the audio system and adjust the amplitude of audio signals at particular frequencies. Read **Overview of Audio Mixer.pdf** for more information of the entire project.

This lab is split into a prelab exercise and hardware implementation. Submit one prelab report and one lab report per group, with the members' names are clearly written on the front page. There is no template for the prelab report, and the lab report template is available on the class webpage. These reports must be in pdf format. There are multiple apps, including CamScanner, for Apple and Android phones that turn photos into pdf's.

2 Precautions

Op-amps (and all other IC chips) can be easily damaged by static electricity, so make sure that you always store them with their prongs attached to plastic foam, which you can acquire (for free) at the EE store. In addition, make sure to ground yourself by touching a metallic surface before handling an op-amp.

Over the course of the lab, your op-amp might burn out due to improper handling and use:

Static Discharge Damage: Your finger might carry a high static voltage (up to hundreds of volts) due to a combination of the clothing you wear (synthetic or wool is worse), the environmental humidity (dry is worse), or other factors. Picking up an IC package could burn out the circuit inside due to this static voltage. Remember to touch a grounded piece of metal (usually a wrist-strap attached to test benches) to discharge the static voltage before handling the IC.

Applying Out-of-Range Input Values: The input signals must be in the range set by the power supplies (see the specifications). If the input signal exceeds the power supplies, either more negative or more positive, the circuit might get burned out.

Burned-out chips look the same as good ones, and you can waste a lot of time trouble-shooting your circuit. Two signs of a burned-out op-amp are excessive current drawn from the power supply (greater than about 10mA with no load) and/or an op-amp hot to the touch. Of course, a blown-out op-amp may exhibit none of these symptoms. If you suspect that your op-amp is faulty, replace it.

You should also pay close attention to the circuit connections and the polarity of the power supplies, function generator, and oscilloscope inputs.

3 Prelab Exercises

The goal of this prelab exercise is to calculate the transfer function for a given filter topology and compare it with a SPICE simulation. The calculation is relatively complex, however, so the transfer function will be derived through intermediate steps.

3.1 Generic Equalizer Filter

Consider the circuit shown in Figure 3.1. This is the circuit schematic for the filters inside the equalizer, with the circuit elements shown as generic impedances.

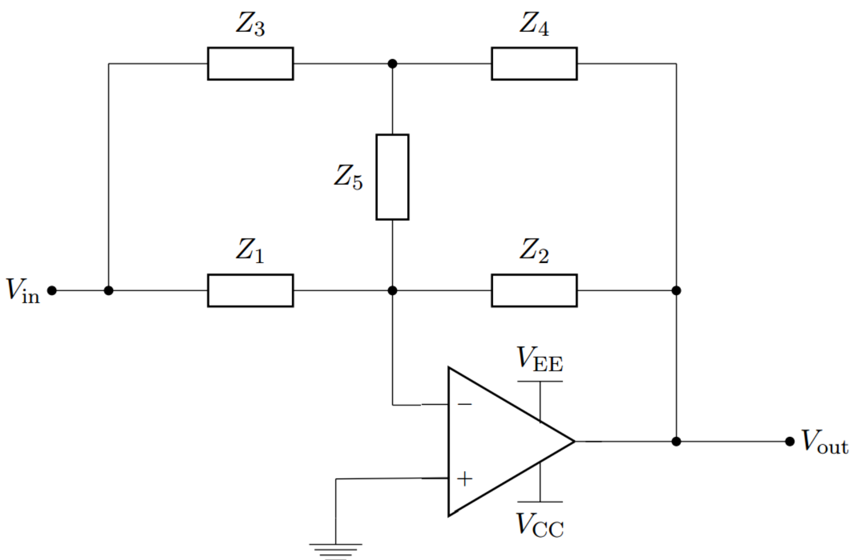

Figure 3.1: Audio equalizer with generic impedances

Prelab #1:
Show that the transfer function for the filter in Figure 3.1 is

$$H(s) = \frac{V_{\text{out}}(s)}{V_{\text{in}}(s)} = -\frac{\frac{1}{Z_3} + \frac{Z_5}{Z_1}\left(\frac{1}{Z_3} + \frac{1}{Z_4} + \frac{1}{Z_5}\right)}{\frac{1}{Z_4} + \frac{Z_5}{Z_2}\left(\frac{1}{Z_3} + \frac{1}{Z_4} + \frac{1}{Z_5}\right)}$$

Now, suppose $Z_1 = Z_2 = Z$.

Prelab #2:
Rewrite the transfer function in terms of Z, then design the other impedances so that the magnitude of the transfer function is always $|H(s)| = 1$.

This circuit topology is quite flexible because it allows the filter to be a low-pass, high-pass, or even a band-pass or band-stop filter by using a single potentiometer. The following questions will help you discover some characteristics of this filter.

Now, instead of generic impedances, let all the impedances except Z_5 be resistors, and Z_5 a capacitor.

Prelab #3:
Assume $Z_1 = R_1,\ Z_2 = R_2,\ Z_3 = R_3,\ Z_3 = R_3,\ Z_4 = R_4$, and $Z_5 = \frac{1}{j\omega C}$:

a) Derive the frequency response with R_1, R_2, R_3, R_4, and C.

Using $R_1 = R_2 = 2.4\text{k}\Omega$, $R_3 = 100\text{k}\Omega$, $R_4 = 50\text{k}\Omega$, and $C = 0.01\mu\text{F}$:

b) Draw the gain of the filter from 10Hz to 5kHz and prove that the filter is a low-pass filter.

Now switch R_3 and R_4:

c) Draw the gain of the filter from 10Hz to 5kHz and prove that the filter is a high-pass filter.

Suppose we then choose the impedances Z_1, Z_2, Z_3, Z_4, and Z_5 carefully so that the circuit becomes a band-pass filter.

Prelab #4:
Assume $Z_1 = Z_2$. Under these circumstances, if Z_3 and Z_4 are switched what type of filter does the circuit become? Justify your answer.

Hint: Switch Z_3 and Z_4 in the transfer function and compare the new one with the original transfer function.

3.2 Equalizer Filter for Audio Mixer

The topology of the filter that is going to be implemented in the mixing console is similar to, but more complex than, the filter that has been calculated. The goal of this circuit is to make three band-pass filters with different central frequencies and a tunable gain, so that we can change the gain of the equalizer at separate frequency ranges. Once the band-pass filter is made, the band-reject filter can be built by changing the potentiometer in each filter, which means that the same filter topography can be changed between band-pass and band-reject by only tuning the potentiometer.

A Y-Δ transformation will be needed to analyze the filter. The Y-Δ transform, also written wye-delta, is a mathematical technique to simplify the analysis of an electrical network. The name derives from the shapes of the circuit diagrams, which look like the letter Y and the Greek capital letter Δ, respectively. These shapes are shown in Figure 3.2.

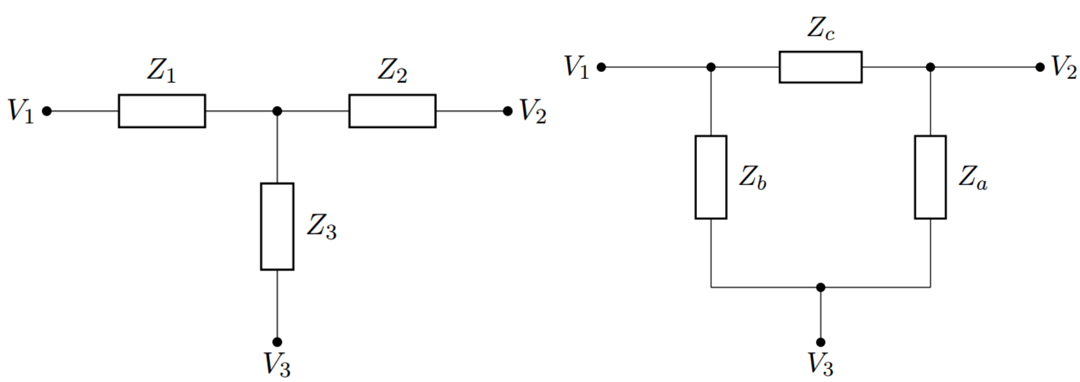

Figure 3.2: Y-Δ transform circuits

To transform a Y-load to a Δ-load:

$$Z_a = \frac{Z_1Z_2 + Z_2Z_3 + Z_3Z_1}{Z_1} \qquad Z_b = \frac{Z_1Z_2 + Z_2Z_3 + Z_3Z_1}{Z_2} \qquad Z_c = \frac{Z_1Z_2 + Z_2Z_3 + Z_3Z_1}{Z_3}$$

To transform a Δ-load to a Y-load:

$$Z_1 = \frac{Z_bZ_c}{Z_a + Z_b + Z_c} \qquad Z_2 = \frac{Z_aZ_c}{Z_a + Z_b + Z_c} \qquad Z_3 = \frac{Z_aZ_b}{Z_a + Z_b + Z_c}$$

Figure 3.3 shows the circuit diagram for the equalizer you will build for your audio mixer.

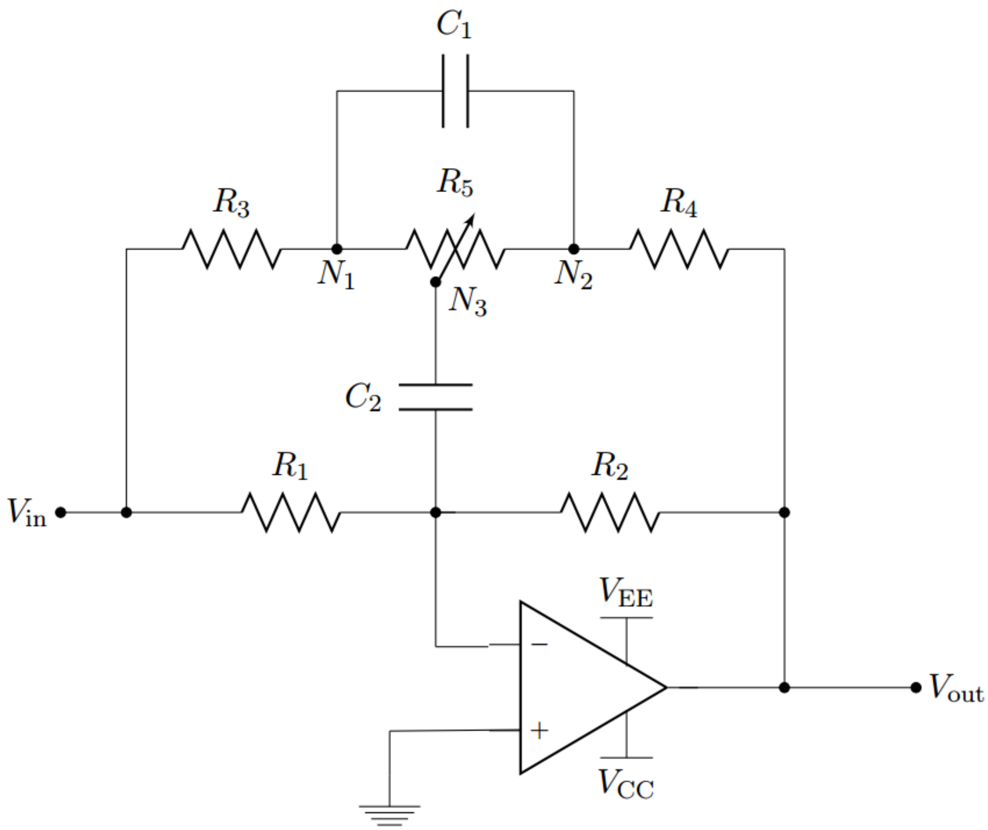

Figure 3.3: Audio equalizer

Prelab #5:

a) Use the transform equations to change the Δ topology among N_1, N_2, and N_3 in Figure 3.3 to a Y topology, and label the new impedances as Z_a, Z_b, and Z_c.

b) Derive expressions for Z_a, Z_b, and Z_c.

Hint: Node N_3 is on the potentiometer R_5, which means the movable terminal of the potentiometer is at the node N_3. You can assume that the resistance on the left of the movable terminal is γR_5, where $0 \leq \gamma \leq 1$, while the resistance on the right is $(1-\gamma)R_5$.

c) Draw the new circuit schematic and turn it in with your completed prelab.

Prelab #6:
Derive the transfer function for the filter in terms of R_1, R_2, R_3, R_4, Z_a, Z_b, Z_c, C_1, and C_2.
Hint: Use the equation derived in Prelab #1.

Prelab #7:
Using **Lab4_Prelab.m** and the capacitors in the lab kit, make three adjustable band-pass or band-reject filters whose center frequencies are centered around 250Hz, 1kHz, and 4kHz.

Prelab #8:
Use AC analysis in Multisim to simulate the gain of these three filters. Record the center frequencies, cutoff frequencies, and gain. Turn these simulated plots in with your completed prelab.

4 Experimental Procedure and Data Analysis

4.1 Audio Equalizer Filters

Build the circuit in Figure 3.3 using power supplies of ± 12V and the components from your design in the Prelab section. The center frequencies are designed for approximately 1kHz. Also build the other two filters whose center frequencies are designed for 250Hz and 4kHz.

Analysis #1:
What resistor and capacitor values did you use in your filters? Justify your answer.

Check your circuit with function generator and oscilloscope and make sure that the circuits are working well as band-pass or band-reject filters. To do this, you'll need to plot the gain of your filter.

There are two ways to plot the filter's gain. The first is to set the function generator to output a sine wave input with a small amplitude (100mV) so that the output is not affected by the slew rate and saturation of the op-amp during this part. Starting with an input frequency of 10Hz, then varying it using the 1-2-5 sequence up to 5kHz (i.e., set input frequency to 10Hz, 20Hz, 50Hz, 100Hz, 200Hz, …, up to 5kHz). Display the input and output waveforms (2-3 complete cycles) on the scope. For each frequency setting above, measure the gain of the three circuits.

Analysis #2:
a) Set the potentiometer to 25%. Plot the gain of each filter between 10Hz and 5kHz. **Use a table to store the center frequency, 3-dB frequencies, and maximum gain for the filter**
b) Set the potentiometer to 50% and repeat part (a) (there will be no 3-dB points for this one).
c) Set the potentiometer to 75% and repeat part (a).

4.2 Audio Mixer

Now you should be ready to build the whole mixing console and play music with it. Combine all of the parts that you have built in the correct order (see **Overview of Audio Mixer.pdf**) and use the following steps to test your circuit system:

1.) Use a sine wave with a frequency of 250Hz and amplitude of 100mV as an input to the system. Make sure that there is no saturation among the signal processing in the system. Display the input and output waveforms in 2-3 complete cycles on the oscilloscope. **Turn this oscilloscope waveform in as part of your lab report.**
2.) Change the potentiometers in the equalizers and display the output on the oscilloscope.

> **Analysis #3:**
> Which filter controls the amplitude of the output filter? Use the oscilloscope waveforms to support your conclusion.

3.) Repeat items 1 and 2 using sine waves with frequencies of 1kHz and 4kHz.

> **Analysis #4:**
> Which filter controls the amplitude of the output filter at 1kHz and 4kHz? Use the oscilloscope waveforms to support your conclusion.

Now that you've analyzed all three of the equalizer circuits it's time to analyze your audio mixer as a whole.

> **Analysis #5:**
> Plot the gain of the entire audio mixer between 10Hz and 5kHz with a single input by using the 1-2-5 sequence method. Turn in the plot of the gain in your lab report and comment on how many bands you see in the plot.

Build the whole audio mixer system in Multisim and use the AC analysis function to see the transfer function of the whole system. **Turn this simulation plot in as part of your lab report.**

> **Analysis #6:**
> What does the transfer function look like with only one input track? Describe the roles of each part in the system.

4.3 Lab Test Preparation

If everything is working well and appears as you expected, use music or instruments as input signals and play them with the audio mixer system. Let your TA check your circuits to make sure it is working well, because it will be used for the demo in your lab test.