EE 233 Circuit Theory

Lab 4: Filters

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1. Introduction

The objectives of this lab are to design filters from a given topology and specifications, analyze the characteristics of the designed filters, measure the characteristics of the designed filter, and build up the entire mixing console.

Filter design is the process of designing a signal processing filter that satisfies a set of requirements. 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. The audio mixing will be completed when the equalizer is built. Please read Supplemental Material Audio Mixer for more information of the whole project.

2. Precautions

None of the devices used in this set of experiment are particularly static sensitive; nevertheless, you should pay close attention to the circuit connections and to the polarity of the power supplies, operational amplifier and oscilloscope inputs.

3. Prelab

In this procedure, you are going to calculate the transfer function for a given filter topology and compare it with a SPICE simulation. This process will help you understand how this filter works. The calculation is relatively complex, and this procedure will help you to get the transfer function progressively.

3.1 Simple Filter

1. Show the transfer function

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

in s-domain for the filter in Figure 1.

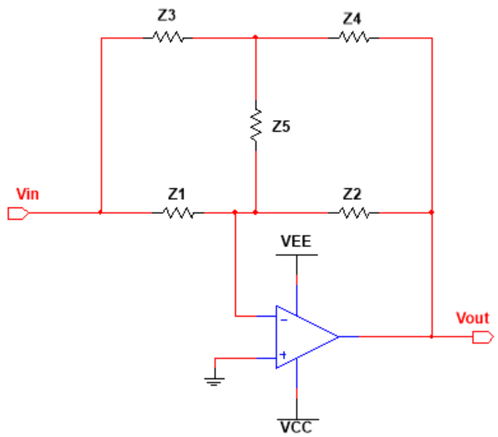

Figure 1 Simple filter

2. Assume $Z_1 = Z_2 = Z$. Rewrite the transfer function and design the other impedances so that the magnitude of the transfer function is always $|H(s)| = 1$.

This topology is quite flexible because it allows the filter to be a low-pass, high-pass, or even band-pass or band-stop filter with a single potentiometer. Now consider this filter in the frequency domain, and the following questions will help you to figure out some characteristics of this filter.

3. Assume all the impedances except Z_5 are resistors, and that Z_5 is a capacitor. Assume $Z_1 = R_1,\ Z_2 = R_2,\ Z_3 = R_3,\ Z_4 = R_4$ and $Z_5 = \frac{1}{j\omega C}$. Derive the frequency response with $R_1,\ R_2,\ R_3,\ R_4$ and C. Then using $R_1 = R_2 = 2.4\ \text{k}\Omega, C = 0.01\ \mu\text{F}, R_3 = 100\ \text{k}\Omega, R_4 = 50\ \text{k}\Omega$, draw the gain of the filter from 20 Hz to 20 kHz and prove that the filter is a low-pass filter. After that, switch R_3 and R_4 and draw the gain from 20 Hz to 20 kHz and prove that the filter is a high-pass filter.

4. Now suppose we choose the impedances Z_1, Z_2, Z_3, Z_4 and Z_5 carefully and it becomes a band-pass filter. Assume $Z_1 = Z_2$, and if Z_3 and Z_4 are switched is it a band-pass or band-reject filter?

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

3.2 Band-pass and Band-reject Filters

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. Now we are going to analyze the filter that is going to be implemented in the audio mixer. The goal 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-stop filter can be achieved by changing the potentiometer in each filter, which means that the same filter can be changed between band-pass and band-reject by only tuning the potentiometer.

1. Read Extra Credit Y- Δ Transform at the end of the prelab. Use the transform equations and change the Δ topology among N_1, N_2 and N_3 in Figure 2 to Y topology and label the new impedance as Z_a, Z_b and Z_c. Draw the new circuit and turn it in with your completed prelab. Derive the expressions for new impedances in your topology with C_1 and R_5 in the s-domain.

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 could assume that the resistance on the left of the movable terminal is γR_5, where $0 \le \gamma \le 1$, while the resistance on the right is $(1-\gamma)R_5$.

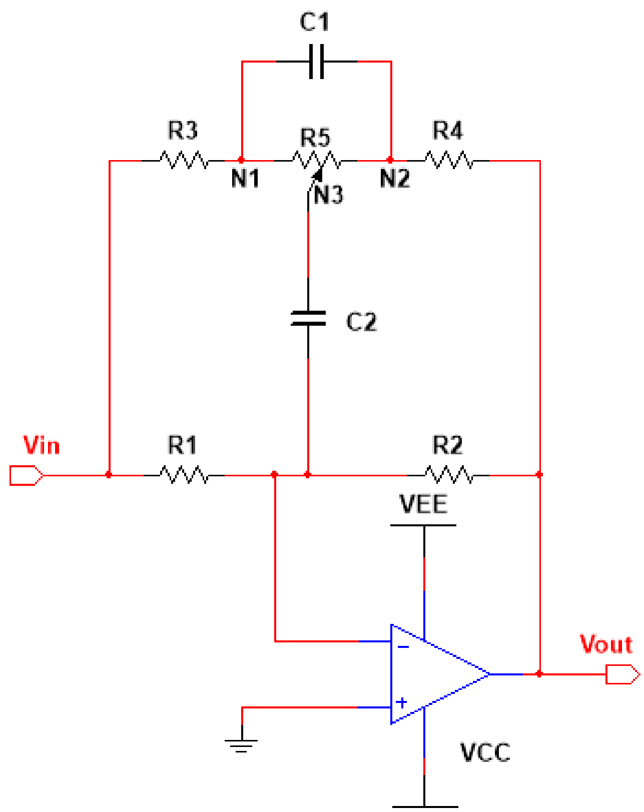

Figure 2 Filter Circuit

2. Derive the transfer function for the filter with R_1, R_2, R_3, R_4, Z_a, Z_b, Z_c, C_1 and C_2.

   Hint: You could use the equation derived in 3.1 item 1.

3. Assume $R_1 = R_2 = 100$ kΩ, $R_3 = R_4 = 1.0$ kΩ and $R_5 = 10$ kΩ. Derive the transfer function, then choose capacitors in the lab kit to make three adjustable band-pass or band-reject filters whose center frequencies are centered around 500 Hz, 2.5 kHz, and 10 kHz.

   Hint: you could use the introduction of graphic equalizer in Reference 5.1 to find the center frequencies and the values of capacitors.

4. Use AC analysis in MultiSim to simulate the gain of these three filters. Record the center frequencies, cutoff frequencies, and gain and turn them in with your completed prelab.

(Optional) Y-Δ Transform

A Y-Δ transformation should be used 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.

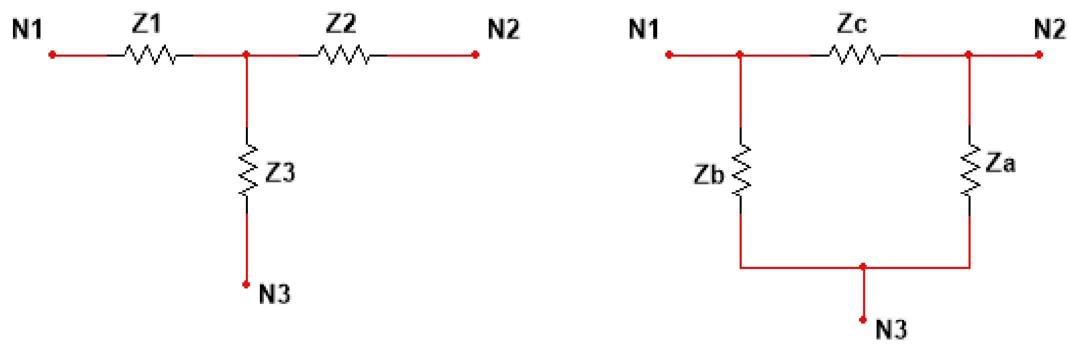

Figure 3 Y-Δ transform

1. Show that the following equations for transformation from a Y-load to a Δ-load circuit are correct.

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

Hint: The Y-load and Δ-load can be transformed into each other because they are assumed to be equivalent. This equivalence means that for any external voltages (V_1, V_2 and V_3) applying at the three nodes (N_1, N_2 and N_3), the corresponding currents (I_1, I_2 and I_3) are exactly the same for both the Y and Δ circuit, and vice versa.

2. Show that the following equations for transformation from Δ-load to Y-load circuit are correct.

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

4. Experimental Procedure and Data Analysis

NOTE: A lab report summarizing the design, results, and conclusions is NOT REQUIRED. Instead, the mixing console built as part of this lab will be demonstrated to the TA. This demonstration will include presenting a working mixing console, presenting requested test results, and addressing questions regarding the design and functionality. Prior to demonstration, please make sure that the mixing console is functioning properly and all of the requested results are assembled. Results, including figures and tables, may be printed out for presentation or displayed on a computer screen. Be prepared to answer any of the questions presented below during demonstration.

4.1 Band-pass and Band-reject Filters

1. Build the circuit in Figure 2 using power supplies of ± 12 V and the components from your design in the Prelab section. The center frequencies are designed for approximately 2.5 kHz.

Question 1: What are the resistors and capacitors that you used in this filter?

Before doing the following experiments with spectrum analyzer, it is better to check your circuit with function generator and oscilloscope and make sure that the circuits are working well as band-pass or band-reject filters.

2. Plot the gain of the filter between 20 Hz and 20 kHz by setting the potentiometers to 25%, 50% and 75%. There are two ways to plot the gain. The first is to set the function generator to output a sine wave input with a small amplitude 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 20 Hz, then varying it using the 1-2-5 sequence up to 20 kHz (i.e. set input frequency to 20 Hz, 50 Hz, 100 Hz, 200 Hz, …, up to 50 kHz). 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. Alternatively, use of the spectrum analyzer could automatically measure the gain. Please read Reference 5.2: Spectrum Analyzer to learn the operations of the spectrum analyzer.

Question 2: Prepare plots of the filter frequency response. Prepare a table containing the center frequency, 3-dB frequencies, and maximum gain for the filter by setting the potentiometers to 25%, 50% and 75%.

3. Build the other two filters whose center frequencies are designed at 500 Hz and 10 kHz. Repeat step 4.1 item 2 and 3 for both of these two filters.

Question 3: What are the resistors and capacitors that you used in these filters? Prepare plots of the filter frequency responses. Prepared a table containing the center frequency, 3-dB frequencies, and maximum gain for the filter by setting the potentiometers to 25%, 50% and 75%.

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 and use the following steps to test your circuit system.

1. Use a sine wave with 500 Hz with a small amplitude as an input to the system. Make sure that there is no saturation among the signal processing in the system.

Question 4: Save the input and output waveforms in 2-3 complete cycles on the scope

2. Change the potentiometers in the equalizers and display the output on the oscilloscope.

Question 5: Which filter controls the amplitude of the output filter? Save waveforms demonstrating your conclusion.

3. Use sine waves with 1 kHz and 4 kHz and repeat 4.2 item 1 and 2.

Question 6: Which filter controls the amplitude of the output filter at 2.5 kHz and 10 kHz? Save waveforms demonstrating your conclusion.

4. Plot the gain of the audio mixer between 20 Hz and 20 kHz with a single input. There are two options to do this task. The first is to use function generator and a sine wave input with small amplitude so that

the output is not affected by the slew rate and saturation in this part. Then start with an input frequency of 20 Hz and vary it using 1-2-5 sequence up to 20 kHz (i.e. set input frequency to 20 Hz, 50 Hz, 100 Hz, 200 Hz, … up to 50 kHz). 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. The second option is to use the spectrum analyzer to automatically measure the gain. Please read Reference 5.2 Spectrum Analyzer to learn the operations of spectrum analyzer.

Question 7: Plot the audio mixer frequency response. How many bands do you see in the plot?

Extra Credit

1. If you have not completed the microphone circuit in Lab 3, you could still work on it. You will still get the extra credit when you show it to your TA and answer the question in your lab report.

2. Build the whole audio mixer system in SPICE and use the AC analysis to see the transfer function of the whole system. Show your simulation to your TA and keep it for the lab test.

Question E: What does the transfer function look like with only one input track? Describe the roles of each part in the system.

5. Reference

5.1 Graphic Equalizers (ref.: Sergio Franco, Design with op-amp and analog ICs)

The function of a graphic equalizer is to provide gain control within intermediate frequency bands. Figure 4 shows a simple design of one of the equalizer section.

The circuit is designed so that over a specific frequency band (bass-band or reject band), C_1 acts as an open circuit and C_2 acts as a short, thus allowing for gain control, depending on the wiper position. Outside the specific band, the filter provides gain equals to one as shown in Figure 4 (b).

If we choose the component values that

$$R_1 = R_2 \gg R_3 = R_4, \qquad R_1 = R_2 = 10R_5, \qquad C_1 = 10C_2$$

Then the center frequency of the specific band is

$$f_0 = \frac{\sqrt{2 + R_5/R_3}}{20\pi R_5 C_2}$$

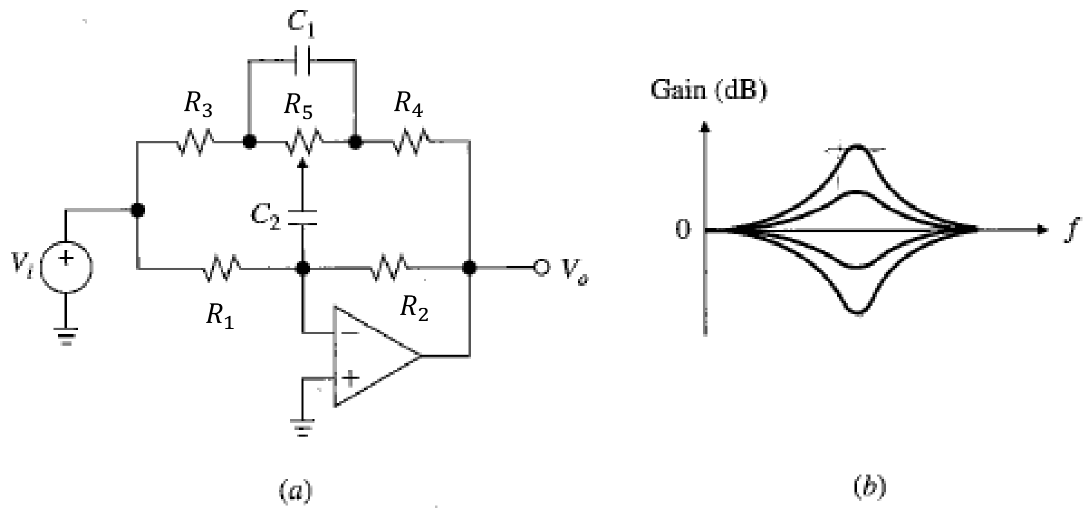

Figure 4 section of a graphic equalizer

5.2 Spectrum Analyzer

The spectrum analyzer located in the lab is an HP 3562A Dynamic Signal Analyzer. It is used to measure the magnitude of an input signal versus the frequency within the full frequency range of the instrument. Its primary use is to measure the power of the spectrum of known and unknown signals. Use the following steps to measure the frequency response of your circuit system:

1. Connect **SOURCE** to the input of your circuit and **CHANNEL 1** to the output of the circuit.

2. Press **MEAS MODE** button in the **MEASUREMENT** panel.

3. There are several selections to choose from on the right column of the screen. Choose **SWEPT SINE** by pressing the button on the right of this item.

4. Choose **LOG SWEEP** on the right column of the screen.

5. Press **FREQ** button in the **MEASUREMENT** panel.

6. Choose **START FREQ** and set it to 20 Hz using the digital buttons in the **ENTRY** panel. The value will be shown at the bottom line of the screen.

7. Choose **STOP FREQ** and set it to 20 kHz.

8. Choose **SWEEP RATE**. The sweeping is more accurate if the sweep rate is larger, but it will take longer to run. Change the sweep rate depending on whether a faster or more accurate sweeping is desired.

9. Press **SOURCE** button in the **MEASUREMENT** panel.

10. Choose **SOURCE LEVEL** and set it to 1 Vpk.

11. Choose **SOURCE ON** and **SWEEP UP**.

12. Press **MEAS DISP** in the **SELECT DATA** panel and choose **POWER SPEC** on the right column of the screen. If you wish to go back to the state menu, you could press **STATE/TRACE** in the **SELECT DATA** panel.

13. Press the **SCALE** button in the **DEFINE TRACE** panel and choose **Y AUTO SCALE**.

14. Press the **START** button in the **CONTROL** panel and observe the output on the screen.