EE233 Circuit Theory Lab3

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EE233 Circuit Theory

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Abstract—The purpose of lab3 is to understand the Bode plots of electronic circuits and the audio mixer system. the preamplifier and summing amplifier for equalizer can implement the functions of the integrator and the differentiator. The mixing console can mix the multiple signals up. The value of potentiometer can adjust the whole volume and the ratio of volumes. The microphone circuit built on the breadboard has a characteristic of low-pass filtering.

Keywords:Bode plot, Preamplifier for the Equalizer, Summing Amplifier for the Equalizer, Mixer and Microphone

I. INTRODUCTION

The aim of this lab are to understand the Bode plots of electronic circuits, and in this lab experiment we mainly focus on integrators circuit and differentiators circuit.Another aim is to learn multisim simulation to design electronic circuits, and analyze and measure characteristics of the circuits we built. The next circuits that we built in this lab are also part of the audio mixer system. To be more specific, the preamplifier and output summing amplifier in the equalizer is going to be built in this lab. And we use the circuit to mix three input signals to see how it works.

II. LAB PROCEDURE

A. Preamplifier for the Equalizer

In part 1, we use two resistors R_1=1kΩ, R_2=47kΩ, a capacitor C=47pF, and V_{CC}=12V, V_{EE}=-12V as power supplies to construct a circuit as Figure 1 shows.

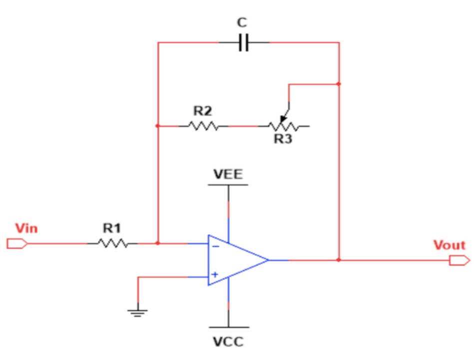

Fig. 1: Preamplifier Circuit

Initially, we choose a 50 kΩ potentiometer and set it to zero in the circuit. We will adjust the frequency of the input signal and record the amplitude and phase change of the output signal. Specifically, we need to sweep the frequency of Vin starting at 10 Hz, then vary it using the 1-2-5 sequence up to 1 MHz while keeping the amplitude at 100 mV. We will record the required data in this case.Secondly, we will set the potentiometer to 50 kΩ and sweep the frequency of Vin starting at 10 Hz, then vary it using the 1-2-5 sequence up to 1 MHz while keeping the amplitude the same. What we need to do is to record the amplitude and phase change of the output signal.Lastly, we will use a function generator which provides a sine wave with an amplitude of 300 mV and a frequency of 300 Hz as input signal. In order to display both waveforms to confirm that the circuit is an integrator , we use the oscilloscope.

B. Summing Amplifier for the Equalizer

In part2, initially, we build the circuit in Figure 2 with power supplies V_{CC} = 12 V, V_{EE}= −12 V.

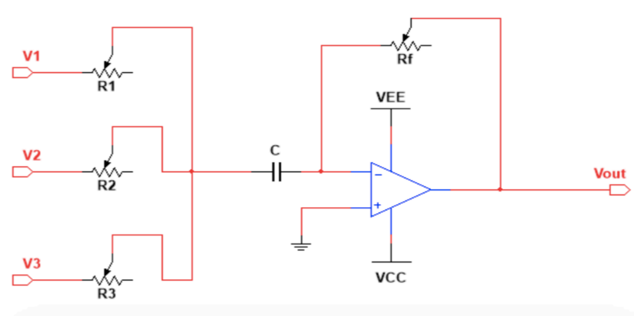

Fig. 2: Summing amplifier circuit

We set the capacitor to 0.1 μF, all resistors and potentiometers to 100 kΩ. Then, we will apply the function generator to V_1 (V_2 and V_3 are disconnected) with an amplitude of 500 mV. Then, we will sweep the frequency of Vin starting at 10 Hz, then vary it using the 1-2-5 sequence up to 1 MHz while keeping the amplitude the same. What we need to do is to record the amplitude and phase change of the output signal.Secondly, we set R_1 = 0 Ω and Rf = 50 kΩ. Then, we will apply the function generator which provides a sine wave with an amplitude of 300 mV and a frequency of 300 Hz as input signal to V_1. In order to display both waveforms to confirm that the circuit is a differentiator, we use the oscilloscope. Thirdly, we will apply the function generator which provides a sine wave with an amplitude of 1V and a frequency of 1kHz as input signal to V_1. Then we set R_1=1kΩ and increase R_f until the output waveform becomes distorted. When the waveform becomes flat on the top and bottom, we need to record the maximum and minimum value of the distorted waveform and compare them with V_{CC} and V_{EE}. Lastly, we are going to figure out the function of the capacitor in the output summing amplifier.

C. Mixer and Microphone

In part3, we play with the mixing console to mix signals up. Initially, we connect the speaker to the output terminal of the output summing amplifier and use headphone jacks to provide three channels of signals from our laptops, PCs or cell phones. After listening to the sound of the speaker, we need to tell whether the three signals are mixed. Secondly, we use the potentiometers in the output summing amplifier to change the whole volume and the ratio of volumes between three channels. We need to figure out what happens in audio studios.

III. EXPERIMENTAL PROCEDURE AND ANALYSIS

A. Preamplifier for the Equalizer

1) Testing procedure:

Firstly, we build the circuit in Figure 1 with V_{CC}=12V and V_{EE}=-12V. We implement these two voltages with two 12V power supply. The anode of one voltage source and the cathode of the other one are both connected to the ground. Then we choose R_1=1kΩ R_2=47kΩ, C = 47pF and a 50kΩ potentiometer which is set to zero in the circuit. Similar to the method we used in previous labs, we sweep the frequency by using the 1-2-5 sequence from 10Hz to 1MHz. In the meantime, we record the amplitude and phase change of the output signal.

Secondly, we set the potentiometer to 50kΩ and again sweep the frequency by 1-2-5 sequence while keeping the amplitude of the input signal constant. Meanwhile, we record the amplitude and phase change of the output signal.

Finally, we change the input to a sine wave with the amplitude of 300mV and a frequency of 300Hz. Then we display the input signal and output signal on the oscilloscope.

2) Analysis of the results: 1.As we sweep the frequency from low frequency to relatively high frequency, we find that the amplitude of the output signal obviously decreases when frequency reaches 1kHz and the downtrend becomes increasingly obvious when frequency continues increasing. In the meantime, the phase angle of the output voltage decreases as frequency increases.

In our theoretical calculation, the amplitude and the phase angle of the output can be expressed as the equations below:

$$|H(j\omega)| = \frac{R_2}{R_1\sqrt{1+\omega^2C^2R_2^2}}$$

$$\angle H(j\omega) = \pi - \arctan(\omega C R_2)(in\ degree)$$

The original data of the experiment is as follows.

f/Hz	V_{in}/V	V_{out}/V	$\Delta\varphi/\circ$	f/Hz	V_{in}/V	V_{out}/V	$\Delta\varphi/\circ$
10	0.19	8.75	-178.2	5k	0.19	8.25	-160.4
20	0.19	8.75	-178.2	10k	0.19	7.25	-144.2
50	0.19	8.75	-178.5	20k	0.195	5.50	-124.9
100	0.19	8.75	-179	50k	0.19	2.75	-104.8
200	0.19	8.75	-177.6	100k	0.19	1.17	-92.23
500	0.19	8.75	-178.5	200k	0.19	0.60	-84.75
1k	0.19	8.75	-176.1	500k	0.19	0.24	-72.24
2k	0.19	8.75	-173	1M	0.19	0.145	-56.8

The Bode Plots from 20Hz to 20kHz are shown below.

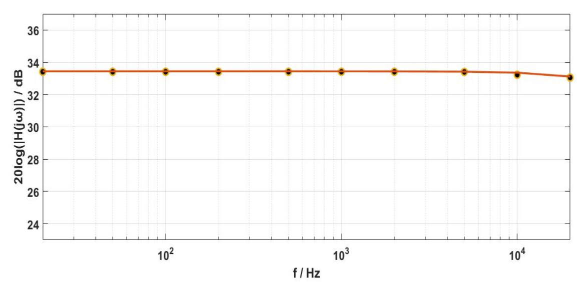

Fig. 3: The magnitude of output signal in terms of frequency

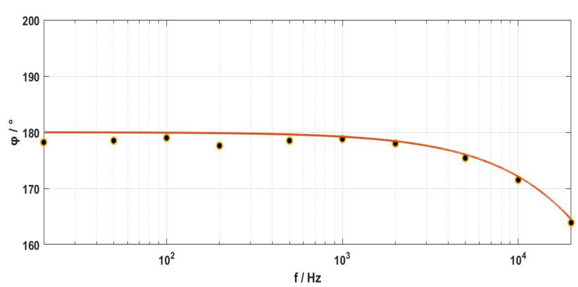

Fig. 4: The phase of out put signal in terms of frequency

We compare the results to those in Prelab and then conclude that the experimental results are approximately the same as the theoretical ones. As we know in the previous lab, the preamplifier can be regarded as a low-pass filter. This is exactly the same as the first Bode plot above. These means our experiments are successful. But there is still error in our experiment.

This can be attributed to the noise generated by the parts in the circuit. The other probable result is that the coefficients of the resistors, capacitor and the op-amp can be not very accurate, which will cause error in theoretical calculation.

2. We record the amplitude and frequency of the output voltage and utilize the data to plot two Bode plots for the amplitude and phase angle versus frequency. The Bode plots are shown below.

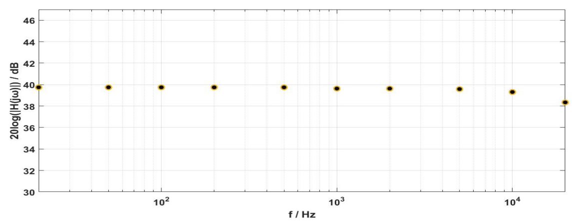

Fig. 5: The magnitude of output signal in terms of frequency

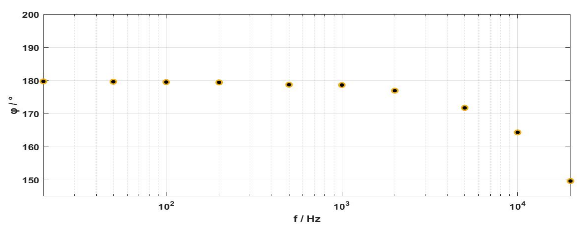

Fig. 6: The phase of out put signal in terms of frequency

Firstly, we try to transform the form of the theoretical expressions of two coefficients of the output voltage. We figure out that when the value of R_2 increases, the amplitude of the output will increase and the phase angle of the output will decrease. Secondly, we compare two pairs of Bode plots in item 1 and 2. We find that the variation trend of two parameters of the output perfectly matches the theoretical results. In a word, the change of the potentiometer increases the value of R_2 in the equation below, thus changes the amplitude and frequency of the output voltage.

3.The figure is shown below.

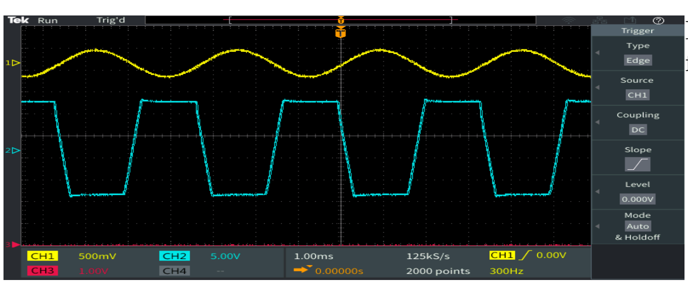

Fig. 7: The waveforms of both input and output signal

It verifies that the circuit is an integrator, because the form of the output signal is the integration of the input signal, and meanwhile the specific parameters matches perfectly.

B. Summing Amplifier for the Equalizer

1) Testing procedure:

Firstly, we build the circuit in Figure 2 with the same power supply as part A. What needs to specify is that we only use one port for input. So, we disconnect V_2 and V_3. Then we set C=0.1 F and all resistors to 100kΩ potentiometers which are set to their maximum resistance. After we finish setting the coefficients of the circuit, we sweep the frequency using the 1-2-5 sequence form 10Hz to 1MHz while keeping the amplitude of the input the same. Simultaneously, we record the amplitude and the frequency of the output voltage.

In the second part of the experiment, we set R_1=0Ω and R_f=50kΩ and apply a sinusoidal waveform with an amplitude of 300mV and a frequency of 300Hz. Then we display the input and output voltage on the oscilloscope.

Then we set R_1=1kΩ and change the amplitude of the signal to 1V and the frequency to 1kHz. To find the value of R_f when obvious distortion just occurs to the output wave, we increase the value of R_f step by step.

In the final assignment, we apply a low frequency to the input and connect the output to the speaker. Then we listen to the sounds in conditions that the capacitor C is in the circuit and the capacitor C is removed form the original circuit.

2) Analysis of the results: 1. According to circuit in Figure 2, we sweep the frequency of V_{in} starting at 10 Hz, then vary it using the 1-2-5 sequence up to 1 MHz while keeping the amplitude the same. Then, we record the amplitude and phase change of the output signal.

Therefore, we plot the amplitude and phase change of the output signal in terms of frequency in Figure 8 and 9.

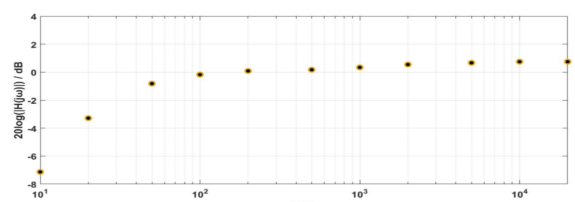

Fig. 8: The amplitude of the output signal in terms of frequency

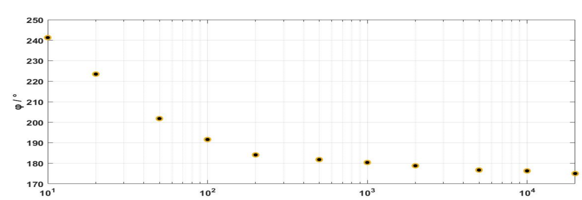

Fig. 9: The phase of the output signal in terms of frequency

According to Figure 8, we can find that as the frequency of the input signal increases, the amplitude of the output signal also increases gradually. To figure out whether it is a low-pass,high-pass or band-pass filter, we plot the amplitude of the output signal in terms of frequency with larger range.

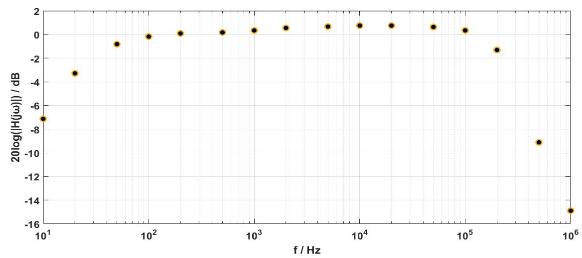

Fig. 10: The frequency response of Summing Amplifier

According to Figure 10, we can conclude that the circuit that we built is a band-pass filter.

2. We apply a sine wave input signal to V_1 with an amplitude of 300 mV and a frequency of 300 Hz. Then, we display the input signal on Channel 1 of the oscilloscope and the output signal on Channel 2. The Figure 11 shows the waveforms of both input signal and output signal.

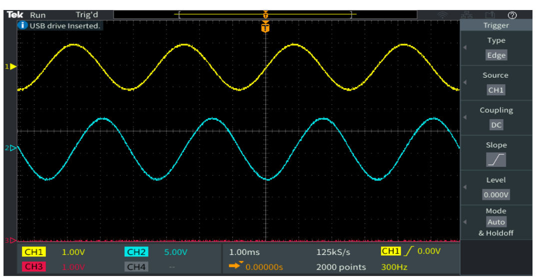

Fig. 11: The waveforms of both input and output signal

According to the phase difference between the waveforms of the input signal and the output signal, we can conclude that the circuit that we built is a differentiator.

3.In this part, we set R_1= 1 kΩ and increase R_f until the output waveform becomes distorted. Then, we get the distorted waveform as Figure 12 shows:

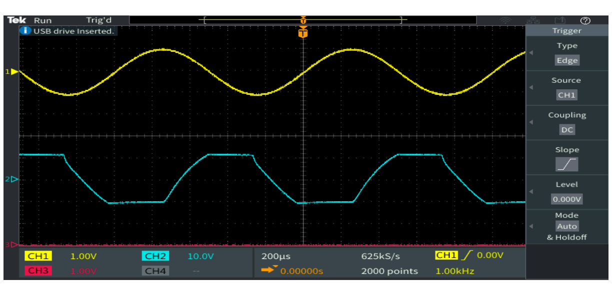

Fig. 12: The plot of distorted waveform

After obtaining the distorted waveform, we measured the maximum and minimum value of the distorted waveform. Then ,we find that the maximum value is about +12V, which has the same value with V_{CC}. The minimum value is about -12V, which has the same value with V_{EE}.

4. In this part, we apply a sine wave with a frequency of 50Hz to V_1 and connect V_{out} to speaker.We found that when there is a capacitor, the sound of the speaker is much lower. When the capacitor is removed, the sound becomes sharper. Therefore, we can conclude that the capacitor can filter the components of the high-frequency signal. In other words, the capacitor acts as a high-pass filter.

C. Mixer and Microphone

1) Testing procedure: In this part, we connect the other two input in the circuit used in part B. Then we connect three peripheral devices to the input and a speaker to the output and listen to the sound of speaker. Then we try to change the value of the four potentiometers respectively to control the ratio of the volumes between three inputs. After confirming our circuit and test results with TA, we start the extra experiment.

2) Analysis of the results: In this part, we set up a 'mixing console' to mix three channels of signals from our laptops, PCs or cell phones up. We can listen the sound through the speaker. Then, we find that the output sound is a mixture of three input signals. Figure shows the waveform of output signal.

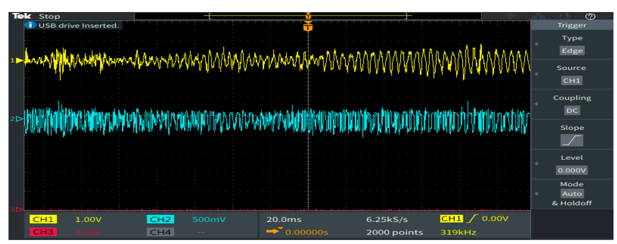

Fig. 13: The waveform of output signal

In addition, we try to figure out the function of the potentiometers. We adjust the value of the potentiometers and listen to changes in the sound of the speaker. Then, we find that the value of the potentiometers can make a difference in the whole volume and the ratio of volumes between three channels. Same as the audio studios in lab2, the value of $R_i(i = 1, 2, 3)$ has an effect on the ratio of volumes between three channels. The larger $R_i(i = 1, 2, 3)$ is, the greater the proportion of the input signal to $V_i(i = 1, 2, 3)$. The larger R_f is, the greater the whole volume is. We also build a microphone circuit. When we speak to microphone, we can hear our own sounds from the speaker clearly.

D. Extra Assignment: Implementation of Microphone Circuit

1) Testing procedure: As we have enough time, we decide to complete the extra assignment. In the end we find it interesting and worthwhile to do this experiment. Generally, the circuit consists of a microphone and a two-stage amplifier. Referring to the basic principal of the circuit, we build the circuit and choose power source V_s=12V and C=1μF. Then we connect the circuit in Figure 14 to the preamplifier and then to the speaker. The final step is to input some kinds of sound, listen to the output sound and try to find out the characteristics difference between the input and the output.

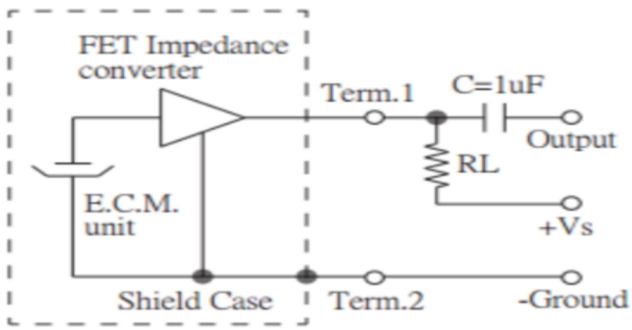

Fig. 14: The diagram of Microphone circuit

2) Analysis of the results: After building the circuit in the instructions PDF, we blow to the speaker and hear the amplified voice of wind we 'generated'. Then we play a sound from a cellphone. Then we hear the sound which comes from the speaker. In the meantime, we hear noise in the speaker. Then we are curious about the reason of generation of this noise and try to find it.

As reference 5.2 says, thermal voltage is approximately white noise. So, we don't take thermal noise into account. After reading the reference in Part 5, we find one possible reason. Though the microphone has great frequency response, its output voltage has a DC bias. The existing bias causes a loud noise to come from the speaker. So we conclude that the DC output generates a bias, thus causes the noise. An approach to reduce the noise is to add a capacitor before the output to remove the DC noise.

In Appendix, we present the extra work we have done. We try to gain more knowledge about the thermal noise and read some after-class references. Then we summarize some of the pointts out to learn more from the lab.

IV. CONCLUSIONS

A preamplifier is bulit in the first part of this lab. As the analysis above said, this circuit acts as a loss-pass filter by applying integral of the input signal, which means the amplitude of the output as well as the phase angle will decrease as frequency increases. In addition, if the resistor changes its value, both the amplitude and phase angle will change.

In the second part, we bulit a summing amplifier by applying differentiating to the circuit. After the analysis in a larger range of frequency, we can identify that this is a band-pass filter: when the frequency is very low or very high, the amplitude gets low but the situation becomes the opposite when the frequency is not too low or too high. Also, the capacitor acts a role of a high-pass filter.

In the final part, we built a mixer and a microphone. When we apply three inputs to the circuit, we can get the mixing signal successfully and we can change the whole volume as well as the ratio of volumes between three channels by changing the value of the resistor. A microphone was bulit successfully according to the Extra Assignment and the reference. Obviously, however, there is a noise in the output of the microphone and we solved this problem by adding a capacitor before the output.

V. APPENDIX

Noise and Preamplifier

As we all know, noise is an unavoidable phenomenon in electronic experiments. The existence of noise will affect our experimental results if the magnitude of noise is close to that of the original data. To make our results more accurate and reduce the error, the preamplifier is used to block the noise in the frequency we don't want to use.

After completing the extra experiment, we want to learn more about the noise. So, we find some reference about Johnson-Nyquist Noise, which all conductors generate.

Johnson-Nyquist Noise is also regarded as Thermal Noise. The generation of it is mainly attributed to the thermal agitation which causes internal charge carrier in the conductor to keep in balance. The thermal noise of an ideal resistor is close to white noise. When the band width is restricted finite, the plot of thermal noise approximately matches Gaussian Distribution.

The power of noise in decibel at room temperature can be described as the equation below:

$$P_{dBm} = -174 + 10log_{10}(\Delta f)$$

We can use this equation to calculate the power under specific frequency.

Finally, we try to derive the equation of the noise generated by a single capacitor. The equation is shown below.

$$Q_n \quad = \sqrt{k_B T C}$$

As the equation shows, the noise generated by the capacitor is usually known as 'KTC noise'.