EE313 Analog Electronic Laboratory 2017-2018 Fall Term Project FMCW Based Distance Measuring System

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***Abstract*—Design of a Frequency Modulated Continuous Wave (FMCW) Based Distance Measuring System**

***Index Terms*—oscillator, amplifier, fmcw, mixer, filter.**

I. Introduction

Design of a Frequency Modulated Continuous Wave (FMCW) Based Distance Measuring System Aim: In this project, you are going to design a distance measurement system as depicted in Figure 1.

II. Transmitter

A. Voltage Controlled Oscillator

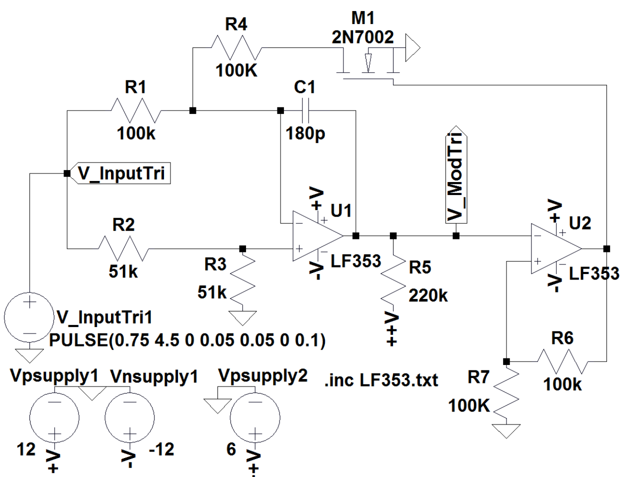
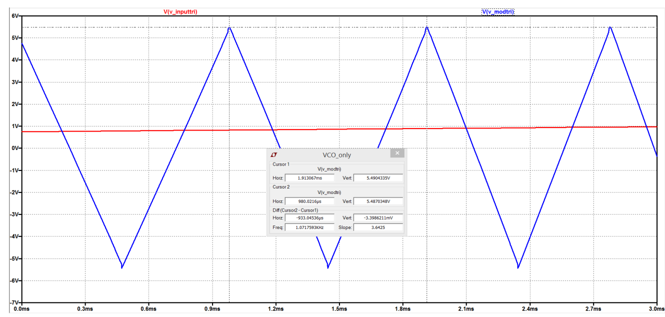
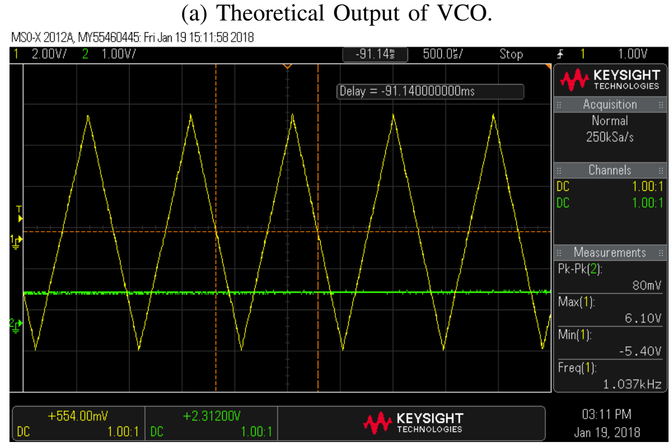


Figure 1: The Voltage Controlled Oscillator Circuit

Voltage controlled oscillator (VCO) is a voltage-to-frequency mapper. VCO outputs variable frequency voltage as the input voltage changes. The used VCO circuit in this project can be seen in *Figure 1*. To be .able to generate frequency modulated signal that is the basic function of VCO, the main input to the circuit is a triangular wave. The triangular wave provides varying input voltage so that output frequency changes gradually. The modulated triangular wave is observed at the V_{ModTri} node. Charging and discharging phenomena of capacitor C_1 is the main reason of the oscillation at the V_{ModTri} node. While capacitor charges, modulated wave climbs down and as capacitor discharges through R_4, modulated wave climbs up. Capacitor charging-discharging cycles do effect the output frequency, however, are not the real reason of modulated triangular wave at the V_{ModTri} node. If there were a constant DC input, the output would be just an oscillating triangular wave with a certain frequency. The main reason for modulated frequency is the changing input voltage. The output frequency depends on three factors. The first one is the capacitance. As capacitance of C_1 increases, the ability of holding charge of the capacitor increases. This results in longer cycles meaning lower frequency at the V_{ModTri} node. Secondly, the resistance R_1 sets a barrier for current flow through C_1. In other words, as R_1 increases, the time for C_1 to charge up becomes longer. Hence longer wave cycles are observed again. Third and last factor is the input voltage value. A higher input voltage implies higher current through C_1. Thus, the time for C_1 to charge up becomes shorter. As a result, higher frequencied cycles are observed at the V_{ModTri}

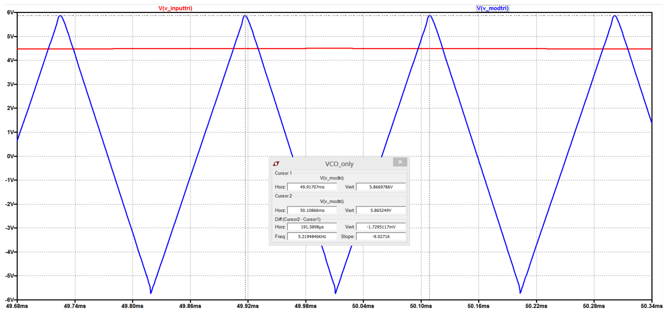


(a) Theoretical Output of VCO.

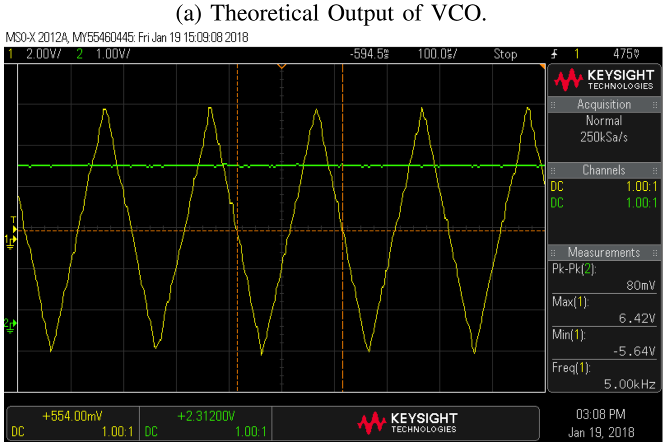


(b) Practical Output of VCO.

Figure 2: VCO Outputs for 1kHz.



(a) Theoretical Output of VCO.



(b) Practical Output of VCO.

Figure 3: VCO Outputs for 5kHz.

node.

To handle these frequent voltage changes, high slew rated LF353 [1] opamps are used. Also for biasing purposes, a N-MOS with low open voltage provides lower DC offset at the input $V_{InputTri}$. 2N7000 [2] model N-MOS suits for this application. Simulation and theoretical results for two corner frequencies are presented in *Figure 2* and *Figure 3*. VCO outputs are revealed as expected in theoretical results.

B. Power Amplifier and Speaker

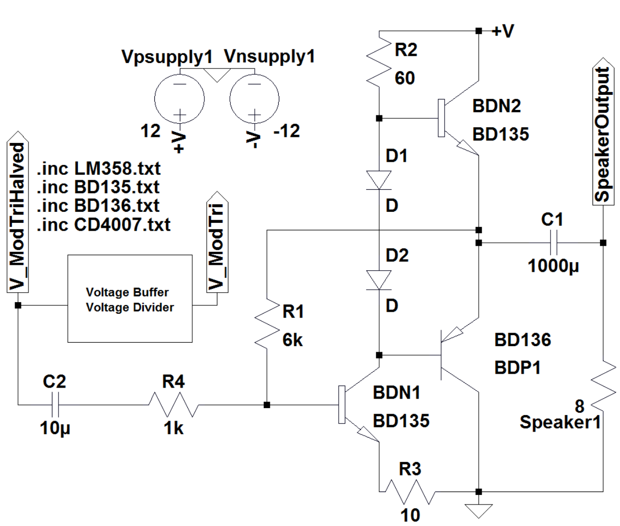


Figure 4: Power Amplifier Circuit Driving the Speaker

This part is aimed to transmit the frequency modulated signal, which is the output of the VCO, to a medium by using a power amplifier. A Class AB amplifier is utilized for this purpose. The amplifier is basically composed of two stages that are common emitter driver and AB amplifier.

The main transistor of common emitter driver is BDN_1. This transistor sets the DC biasing of BDN_2 and BDP_1. It may be considered as a current source. Also, the input signal is output as inversely polarized at the collector of the BDN_1.

The class AB amplifier mainly consists of two transistors one of which is a NPN and the other is PNP. The diodes D_1 and D_2 are to provide Q point stability of the transistors. Also they provide constant voltage difference between the bases of BDN_2 and BDP_1. When the voltage in positive cycle, BDN_2 amplifies the signal whereas BDP_1 amplifies the signal in negative cycle. These transistors operate cooperatively according to the cycle of the signal. Lastly, feedback resistor R_1 reduces the distortion of the output signal by introducing a negative feedback to the input signal in shung-shunt topology. On the other hand, the DC biasing of BDN_1 is improved with the use of R_1.

III. Receiver

A. Microphone & Microphone Driver

To receive the sound signal coming from the speaker and turn it into a electrical signal, some kind of speaker should be used. For this purpose, an electret microphone will be used.

To drive the electret microphone, a driver circuit will be designed. The design should include feeding for the transistor inside the microphone and amplify the output signal since the output signal would be too small to be used. But before going amplifying the signal, we shall remove possible dc offsets and noises by a passive high pass filter. The designed driver can be seen at *Figure 5*.

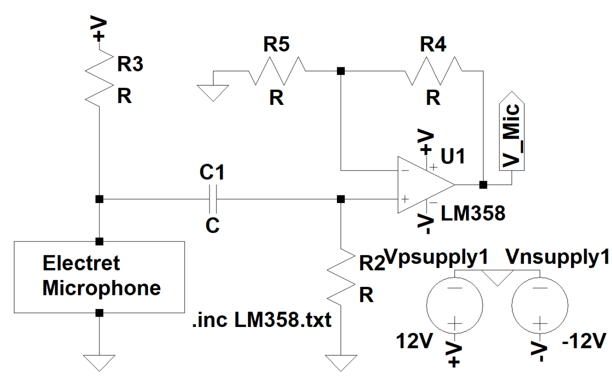


Figure 5: Driver Circuit for Electret Microphone

By node analysis, the lower cutoff frequency of the highpass filter can be found as

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

If we take $R_2 = 10k\Omega$ and $C_1 = 0.1\mu F$, cutoff frequency f_c can be found as

$$f_c = \frac{1}{2\pi * 10\ k\Omega * 0.1\ \mu F} \approx 159Hz$$

The resulting f_c will be enough to remove noise and DC offsets.

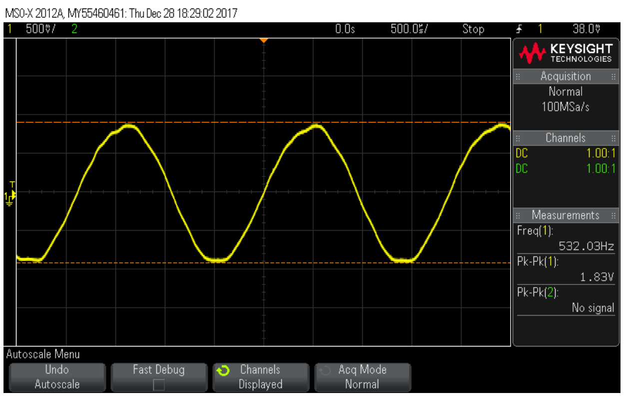


Figure 6: Output Signal of Microphone Circuit at Laboratory

B. Mixer

Mixer is the device that accepts two input signals and gives a output signal that consists of two distinct signals with different frequencies. While, one of these signals has a frequency that is equal to the difference between the frequencies of first signal and second signal, other signal has a frequency that is summation of the frequencies of first and second signal. In other words, if we assume input signal 1 has a frequency f_1 and input signal 2 has a frequency f_2. The output would look like

$$O/P\ \ Signal\ = \ A(f_1 + f_2) + B(\mid f_1 - f_2 \mid)$$

where A and B are the signals having frequencies

$$f_A\ =\ f_1 + f_2\ \ \&\ \ f_B\ =\ \mid f_1 - f_2 \mid$$

respectively.

We know that the distance between source and receiver causes a time delay proportional to the distance for the received signal in comparison to the received signal. Thanks to triangular wave used at the very beginning of the project, the distance between the source and receiver is also proportional to the the frequency difference between original source signal and received signal by receiver. This can be understood from the basic principle of VCO. Assume that the first signal left the VCO and entered the mixer at Time 0 and has a frequency f_1 proportional to the magnitude of triangular wave at Time 0. Similarly second signal entered the Mixer at Time 1 and has a frequency f_2 proportional to the magnitude of triangular wave at Time 1. One period of the input triangular wave and "Time 0" & "Time 1" on top of it can be seen at *Figure 7*.

Therefore, the time shift "$Time\ 1\ -\ Time\ 0$" is proportional to the "$\mid\ f_2\ f_1 \mid$". Thus, if we can find the frequency difference between this signal, we can easily find the desired distance since the distance can be found by

$$d\ =\ (Time\ 1\ -\ Time\ 0) * v_s\ =\ K * \mid f_1\ -\ f_2 \mid$$

where K is a constant and v_s is the speed of sound.

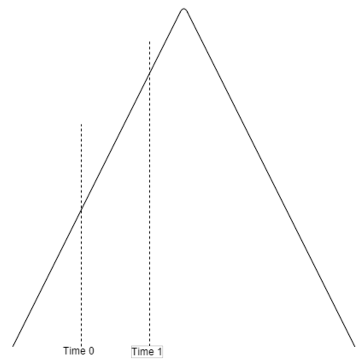


Figure 7: One Period of Triangular Wave

For that purpose, a mixer circuit is what we need and a basic mixer circuit we are planning to use can be seen at *Figure 8*.

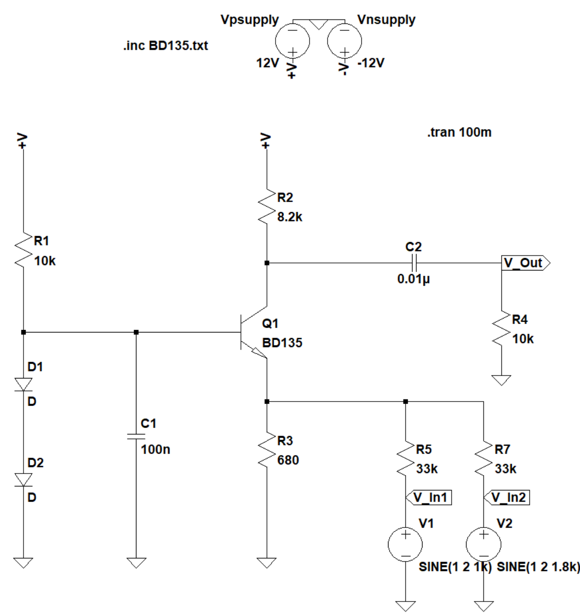


Figure 8: A Mixer Circuit

C. Low Pass Filter

Unfortunately, the output signal of mixer does not only carry the frequency difference info but also frequency summation info. Since we are interested with the difference between the frequencies only, a low pass filter should be used to extract the wanted signal.

Low Pass Filters are the type of filters that passes and amplifies the signals having the frequencies lower that the desired frequencies also known as the cut-off frequencies of the filter. Other signals having higher frequencies would be eliminated at the output of the low pass filter.

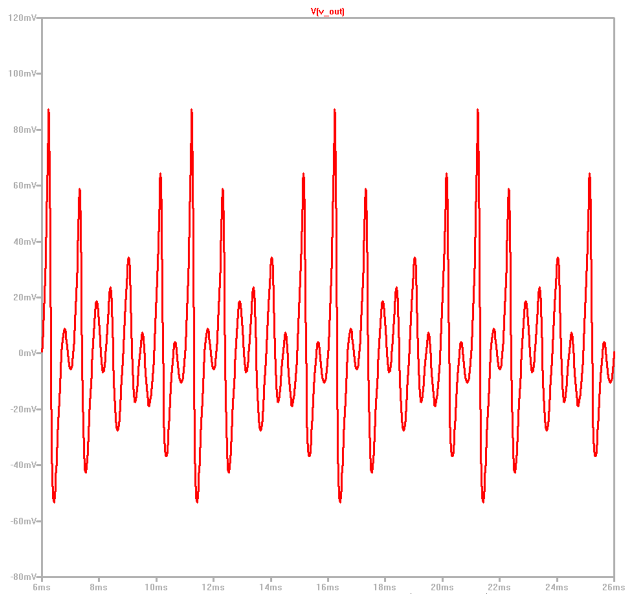


Figure 9: The Output Waveform of the Mixer Circuit

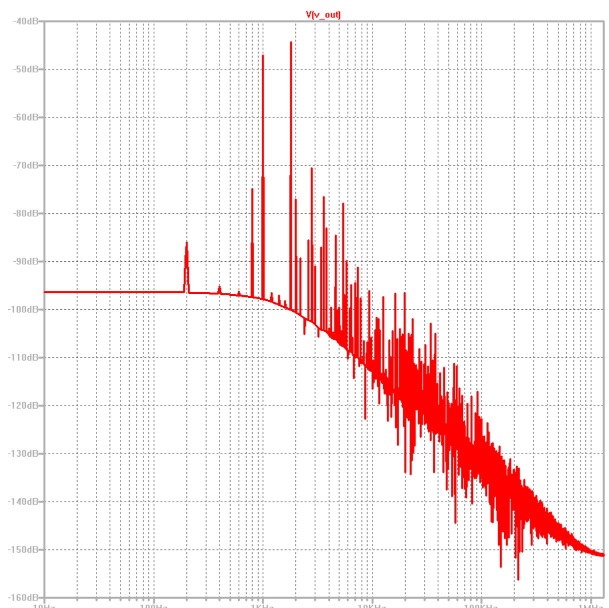


Figure 10: The Output FFT Waveform of the Mixer Circuit

A basic second order low-pass filter that we are going to use can be seen at *Figure 11*.

Doing some KCL at the circuit ,at *Figure 11*, at S-Domain and setting the expressions in order, the gain and cut-off frequency can be found;

$$\frac{V_a - V_{in}}{R} + \frac{V_a - V_o}{1/sC} + \frac{V_a - Vo/2}{R} = 0$$

$$\frac{V_o/2 - V_a}{R} + \frac{V0/2}{1/sC} = 0$$

If the expressions are set in order, the gain and cut-off frequency can be found to be;

$$A_V = 1 + \frac{R_A}{R_B}$$

$$f_c = \frac{1}{2\pi RC}$$

Assuming same resistance and capacitor values for simplicity. The resistance values can be found from there assuming the capacitor values for desired frequency. For

$$f = 1\ kHz = 10^3\ Hz$$

$$\&$$

$$C = 10\ nF = 10^{-8}\ F$$

Resistance R can be found as

$$R = \frac{1}{2\pi * 10^3 * 10^{-8}} \approx 16\ k\Omega$$

R_A & R_B can be chosen equal to each other and $1\ k\Omega$ for the simplicity.

Two test the low pass filter in the LTspice, two input signals in *Figure* **??** with two different frequency can be applied to the filter. The distorted input & filtered output waveforms can be seen from *Figure* **??**.

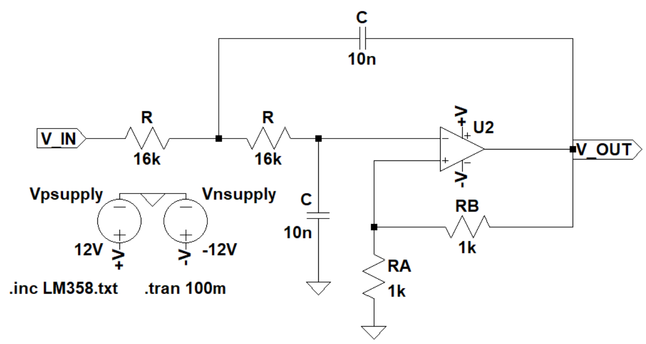


Figure 11: A Second Order Low-Pass Filter

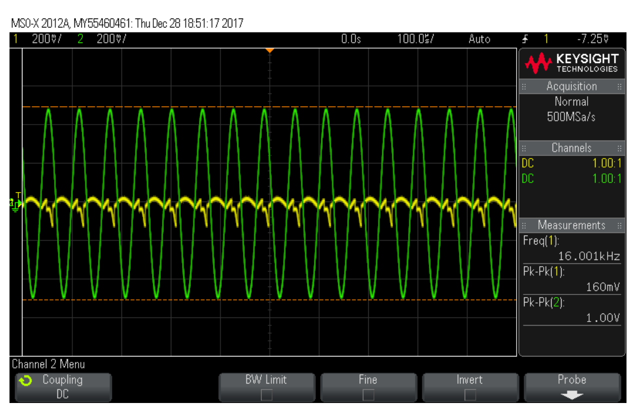


Figure 12: Output Signal for 16 kHz

IV. General Diagram

General diagram of the project can be seen at *Figure 15*.

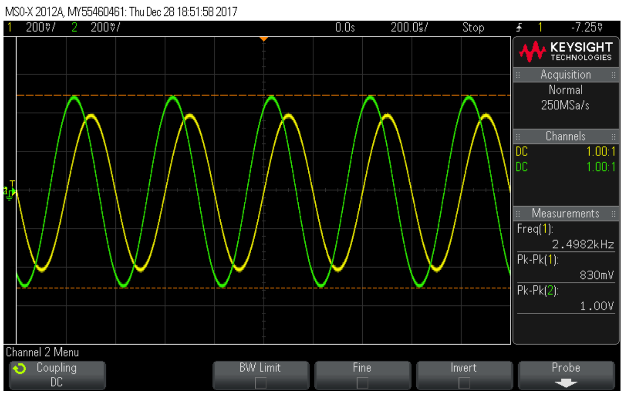


Figure 13: Output Signal for 2.5 kHz

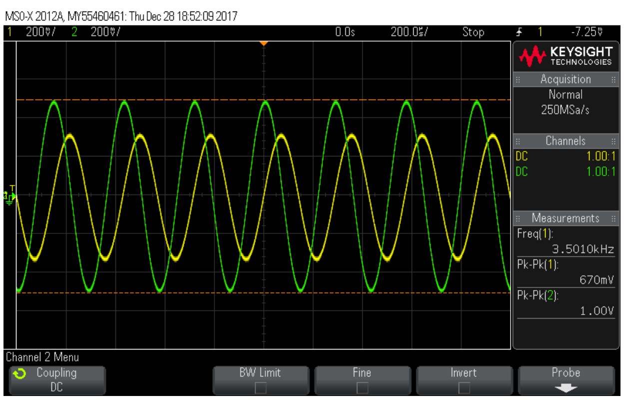


Figure 14: Output Signal for 3.5 kHz

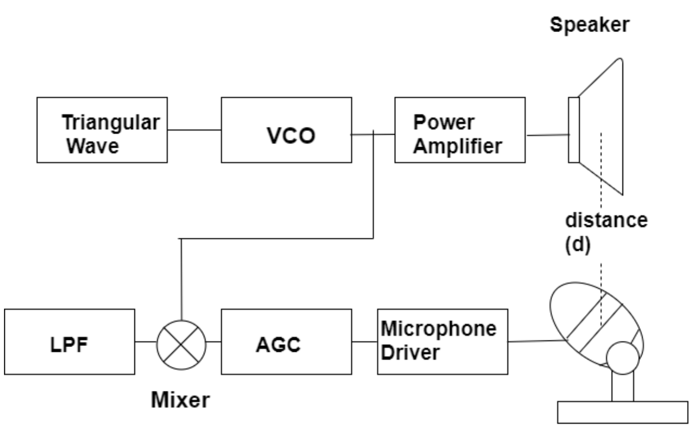


Figure 15: General Diagram of the Project

V. Conclusion

First, a triangular voltage is generated with varying frequencies by a VCO circuit and 10 Hz triangular wave input. Then modulated triangular wave is sent to microphone by a speaker and to mixer at the same time. The microphone transmits the captured signal to the mixer. In the mixer, there are two equivalent inputs but with a phase shift. Output of the mixer is basically sum of two waves having frequencies $(f_1 + f_2)$ and $(\mid f_1 - f_2 \mid)$ that are frequencies of two inputs of the mixer. If this signal is low-passed, the filtered signal have $(\mid f_1 - f_2 \mid)$ frequency. By measuring $(\mid f_1 - f_2 \mid)$, the distance between the speaker and the microphone can be found by a simple algebra, $K * (\mid f_1 - f_2 \mid)$ where K is a constant to be determined and having unit of $\frac{Meter}{Hertz}$.

References

[1] "LF353 Datasheet." [Online]. Available: http://www.ti.com/lit/ds/symlink/lf353-n.pdf. [Accessed: 20-Jan-2018].

[2] "2N7000 Datasheet." [Online]. Available: https://www.onsemi.com/pub/Collateral/2N7000-D.PDF. [Accessed: 20-Jan-2018].