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

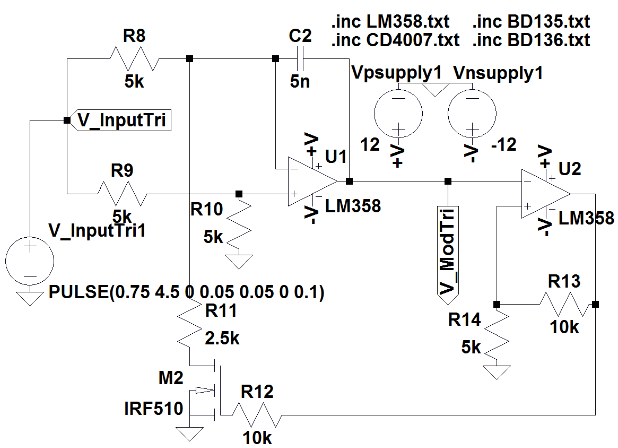

Figure 1: The Voltage Controlled Oscillator Circuit

Voltage controlled oscillator (VCO) is a circuit that basically maps certain voltage values at the input to a frequency at the output. In other words, VCO outputs different frequencies as the voltage value of the input changes. The circuit to be used as VCO in this project can be seen in *Figure 1*. To utilize from this mapping property of the VCO in a radar application, the input to the circuit is a triangular wave such that input voltage changes continuously. Essentially, R_1 and C_1 in *Figure 1* are the determinative components of the voltage-to-frequency mapping. The M_1 acts like a switch that is ruling the current through the C_1. The V_{ModTri} is the frequency modulated output of the circuit. This output will be used in the rest of the project. However, V_{ModTri} goes one more step in the VCO circuit which is a Schmitt Trigger represented with LM2 opamp. With the positive feedback, LM2 outputs either $+V$ or $-V$. The output of the LM2 effects the state of M_1 with the input triangular voltage.

B. Power Amplifier and Speaker

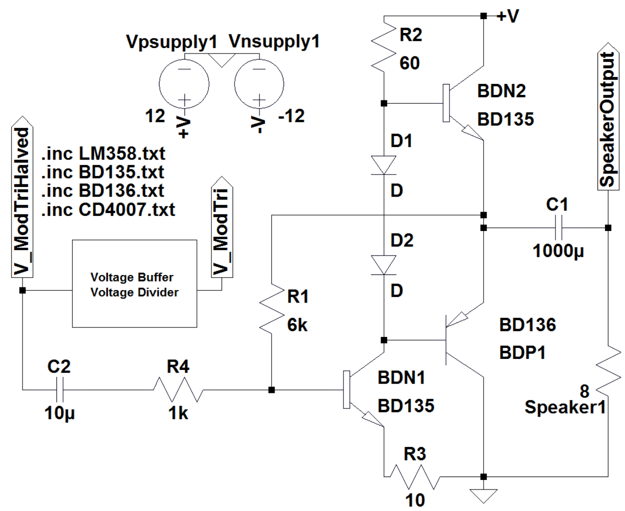

Figure 2: Power Amplifier Circuit Driving the Speaker

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 improvd 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 3*.

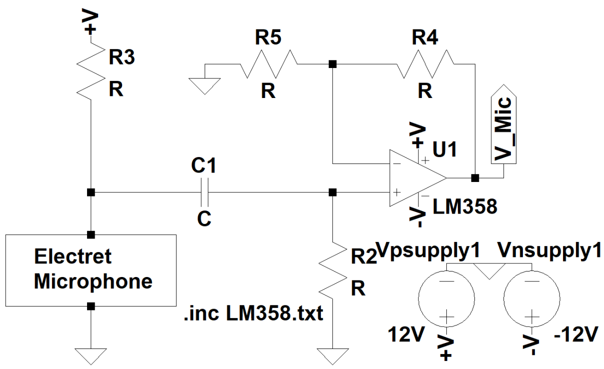

Figure 3: Driver Circuit for Electret Microphone

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.

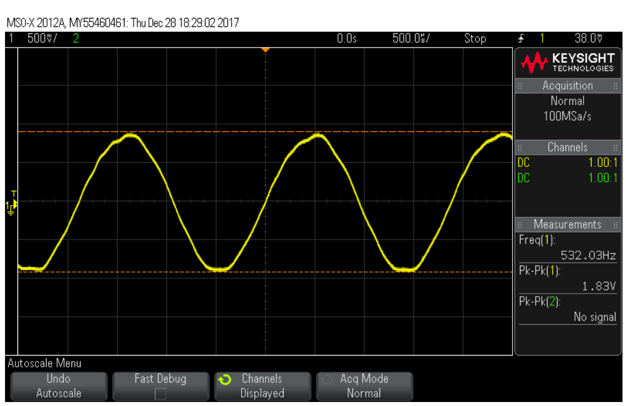

Figure 4: Output Signal of Microphone Circuit at Laboratory

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 5*.

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.

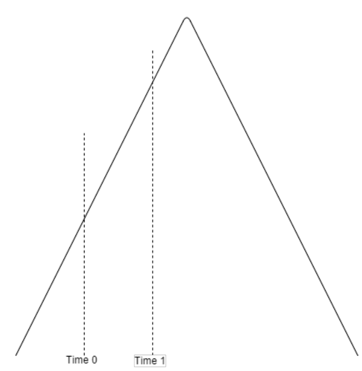

Figure 5: One Period of Triangular Wave

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 6*.

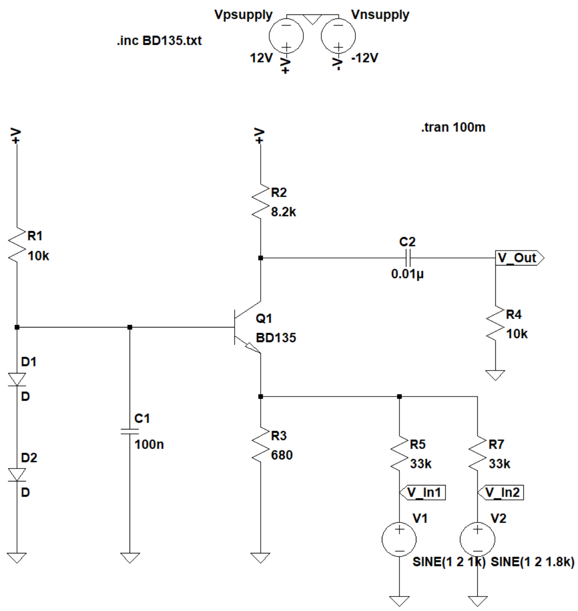

Figure 6: A Mixer Circuit

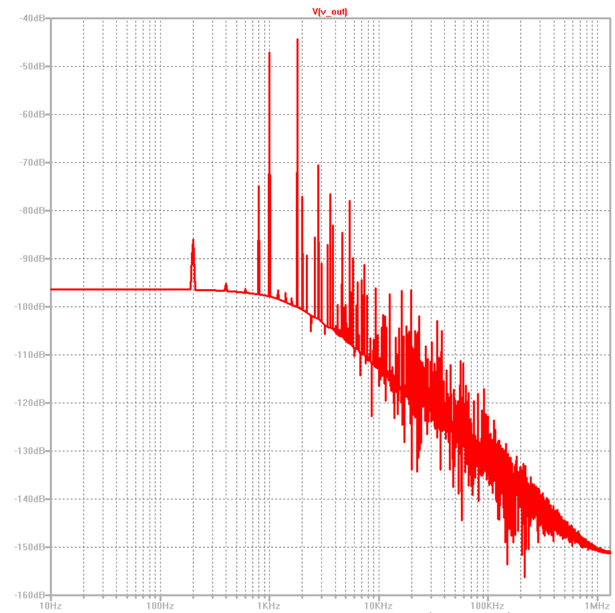

Figure 8: The Output FFT Waveform of the Mixer Circuit

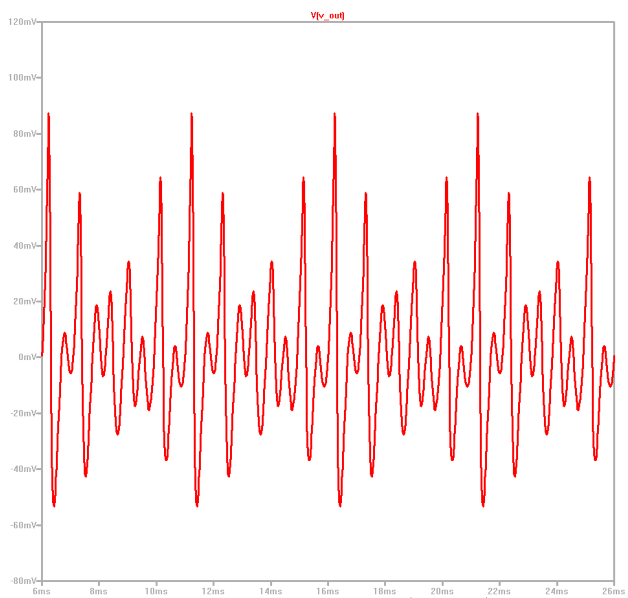

Figure 7: The Output Waveform of the Mixer Circuit

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.

A basic second order low-pass filter that we are going to use can be seen at *Figure 9*.

Doing some KCL at the circuit ,at *Figure 9*, 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* **??**.

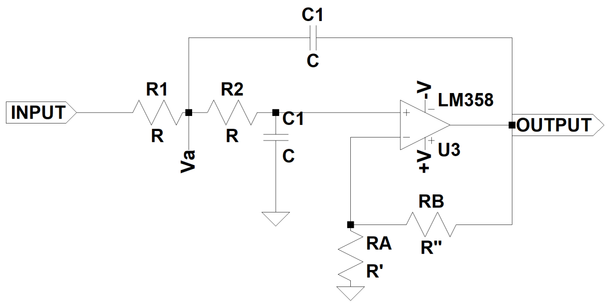

Figure 9: A Second Order Low-Pass Filter

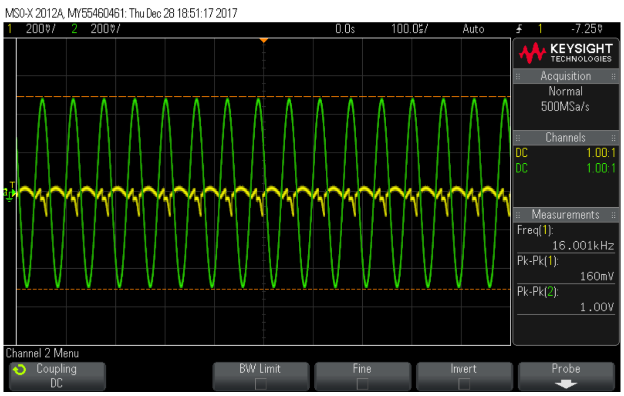

Figure 10: Output Signal for 16 kHz

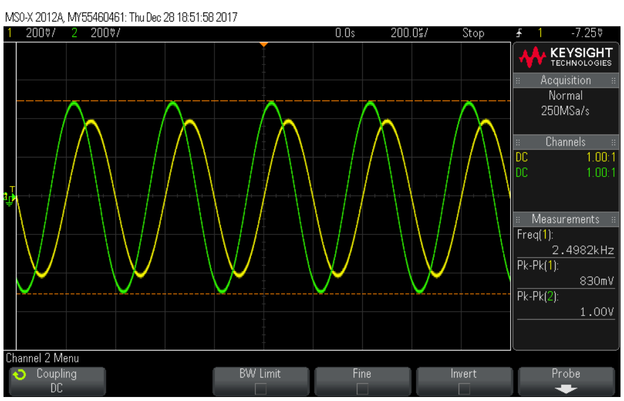

Figure 11: Output Signal for 2.5 kHz

IV. General Diagram

General diagram of the project can be seen at *Figure 13*.

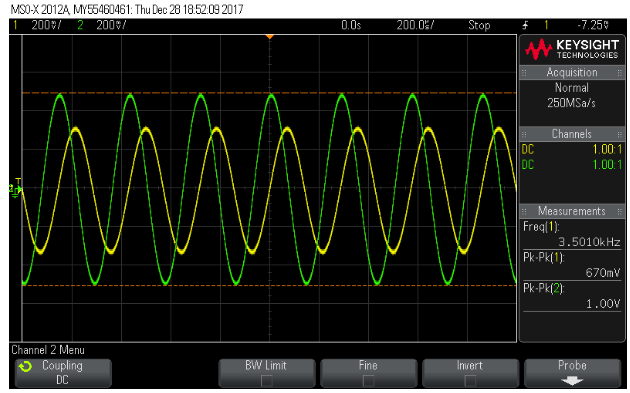

Figure 12: Output Signal for 3.5 kHz

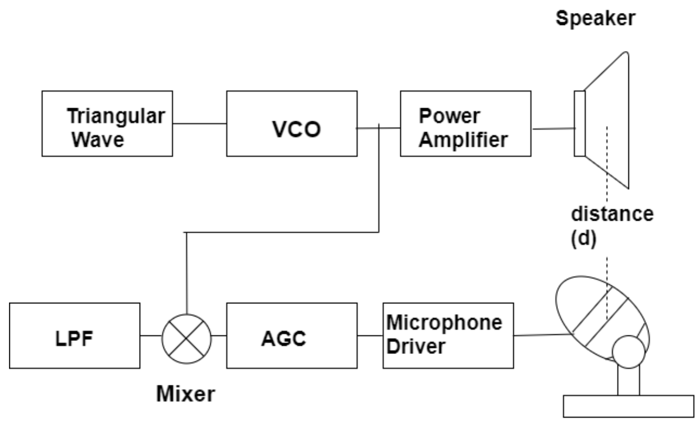

Figure 13: General Diagram of the Project

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 $\frac{Meter}{Hertz}$.