EE 313 Analog Laboratory Final Project Report

Frequency Modulated Continuous Wave (FMCW) Based Distance Measuring System

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***Abstract*—This report presents project which is conducted in EE313 Analog Electronics Laboratory course. Content of this document gives detailed information about our work.**

***Index Terms*—FMCW; Voltage Controlled Oscillator; Power Amplifier; Microphone Driver; Mixer; Filtering; Distance Measuring**

I. Introduction

In this project, our aim is to design a distance measurement system based on FMCW (Frequency Modulated Continuous Wave) principle. For this purpose, we will design a VCO (Voltage Controlled Oscillator) circuit whose output is a square wave with varying frequency controlled by a triangular wave input. Then, we will feed this signal to a speaker through a power amplifier and read this signal using a microphone. Then, we will convert the sound waves to electrical signals by a microphone driver circuit so that we can multiply the VCO output and received signal using a mixer circuit. The mixer circuit multiplies the two signals generates an output consisting of two signals with different frequencies which are the addition and subtraction of the two input signals' frequencies. This system works due to the difference of the frequencies of the transmitted and received signals because when the sound waves travel through air, a time delay in the signal occurs and it causes a frequency difference since the VCO's output has a continuously changing frequency. Finally, we will design a low pass filter and take the mixer's output as input and the low pass filter passes only the difference signal so that we can measure the distance between the speaker and the microphone by observing frequency of the final output signal on the oscilloscope.

II. Specifications and Components

A. Specifications

We are required to generate a signal whose frequency is changing between 1kHz and 5kHz linearly. We are required to dissipate at least 1W of power at the speaker. We are required to convert the sound waves into meaningful electrical signal using a microphone driver circuit. We are required to multiply the two signals and measure the distance by observing a single frequency sinusoidal signal at the output which is the difference of the two signals' frequencies after feeding the output of the mixer to a low pass filter. The difference between the pass band and the stop band of the low pass filter should be at least 10dB.

B. Components

- Various Resistors (1kΩ, 50kΩ, 100kΩ, 390kΩ)
- 8 Ω & 6W Speaker
- Electret Microphone
- Various Capacitors (47nF,100nF, 1µF)
- Diodes (1N4007)
- Op-Amp (LM358)
- BJTs (BD135, BD136)
- Mosfets (IRFZ44N)

III. Circuit Subparts

A. Voltage Controlled Oscillator

In this part, we design a VCO circuit that generates a square wave output whose frequency is controlled by a 10Hz triangular wave so that we can generate a triangularly modulated output. This type of wave is useful for distance measurement system since the difference in the frequencies of the transmitted and the received signals remains constant in the measurement interval. For this purpose, we use the charging and discharging of a capacitor through a transistor and compare the voltage across the capacitor using an op-amp.

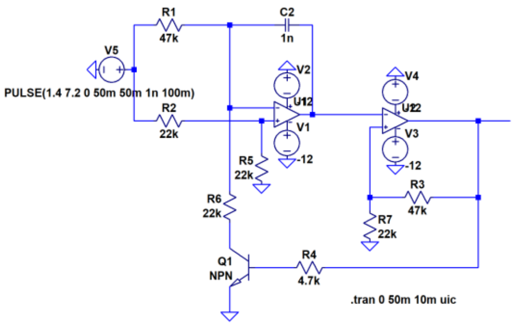

Fig.1. The VCO circuit schematic.

In Fig.1. the VCO circuit schematic is shown. The first op-amp is an integrator circuit and it integrates the triangular input voltage and the second one is used as a Schmitt trigger to give positive saturation voltage or negative saturation voltage above and below a threshold value. This threshold value is determined by the voltage division at the positive input of the second op-amp. The transistor is used as a switch which starts to allow current flow when the capacitor voltage is reached at the mentioned threshold voltage. After this point, the capacitor is discharging through the transistor until it reaches the negative threshold value. Then, the transistor switches back to open and the capacitor starts to charge again and this procedure continues to generate a square wave output signal at the second op-amp[1]. The triangular input voltage enabled us to obtain frequency-changing property of the circuit. When the input voltage is high, the current though the capacitor is high and the capacitor charges and discharges quickly resulting in a high frequency and vice versa when the input voltage is low.

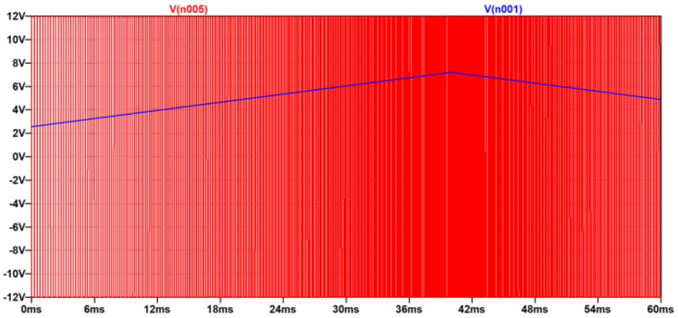

Fig.2. The output(red) and the input(blue) signals of the VCO circuit.

In Fig.2. it can be seen that the frequency of the output signal is increasing with increasing input voltage and vice versa when the input voltage is decreasing. The maximum frequency of the output signal is 5kHz which is obtained when the input voltage is 7.2V and the minimum frequency is 1kHz which is obtained when the input voltage is 1.4V and this oscillation interval is what is required from us. Also, the rate of change of the frequency of the output signal is 10Hz which is the frequency of the input signal.

In Fig.2. the square waveform is seen to be a perfect square wave with sharp rising and falling times. However, in practice, we used LM358 as op-amp and its slew rate is considerably low. In practice, we observed a square wave output that has longer rising and falling times. Therefore, the square wave output did not have sharp change of negative to positive and positive to negative voltages.

B. *Power Amplifier*

In this part, we design a power amplifier to dissipate at least 1W of power on the speaker. However, since the impedance of the speaker is very low, 8 Ω, we need to design an amplifier circuit which has a high output impedance and low distortion characteistics so that most of the output current will flow thorugh the speaker. For this purpose, we choose a class AB output stage which has high output resistance and very low distortion characteristics.

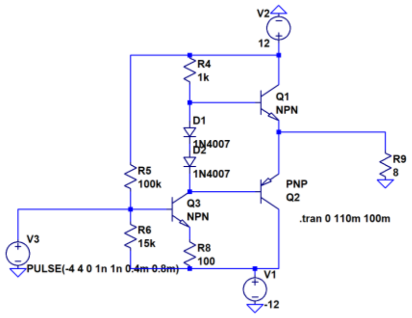

Fig.3. The class AB output stage circuit.

In Fig.3. the class AB output stage is shown and it can be seen that we biased the npn and pnp transistor by using two diodes. This is because we want to ensure that the transistors are always in forward active region. In this region, the transistor have around 0.7V across base emitter junctions and it corresponds to a voltage difference of 1.4V between the two bases. By connecting two diodes which have 0.7V of opening voltage, we bias the two transistors so that they will always be in forward active region. This property of the class AB amplifier differs from the class B amplifier. In a class B amplifier, the two transistor do not enter the active region until they reach a certain base emitter voltage and it causes a distortion at the output. By using a class AB output stage, we prevent distortion at the output[2]. Another advantage of the class AB output stage amplifier is that the transistor are not always working. That is when the input signal is on the positive cycle the npn transistor is active and the pnp transistor is off and vice versa when the input signal is on the negative cycle. This property, prevents overheating of the components. Normally, the collector of the pnp transistor is connected to ground in class AB circuits. However, we connected a negative 12V supply to ensure that negative parts of the input signals are not clipped.

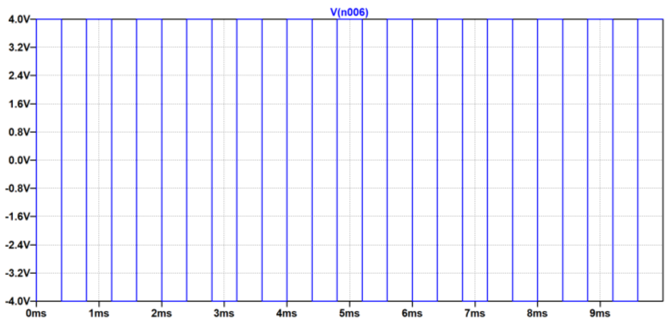

Fig.4. The output waveform of the class AB output stage.

As seen in Fig.4. the voltage across the speaker has 8 Volts peak to peak which gives a power of more than 1 Watt.

$$P = V_{rms}^2/R \tag{1}$$

The power dissipated on the speaker can be calculated by inserting the rms value of the speaker voltage and the impedance of the speaker into equation (1). Therefore, the power dissipated on the speaker is 1 Watt. Also as seen in

Fig.4. the voltage gain of the class AB circuit is 1. However, this circuit amplifies the current thus, it amplifies the power.

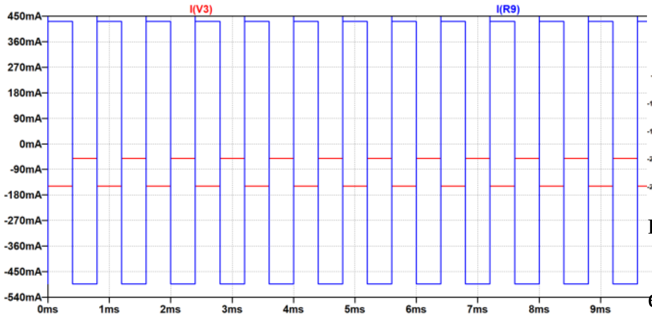

Fig.5. The input(red) and output(blue) current waveforms.

As seen in Fig.5. the input current is amplified. The current gain of the circuit can be calculated as 9A/A by observing the current waveforms in Fig.5.

In practice, the amplification of the signal is not perfect. The output of the circuit has some nonlinearities at the shift of the square wave from positive to negative and negative to positive.

C. *Microphone Driver*

In this part, it is needed to obtain meaningful signal waveform in order to compare the output of microphone with the transmitted waveform. However, the microphone does not generate meaningful electrical signals alone. Hence, the microphone driver circuit is integrated to convert the sound waves into meaningful electrical signals to compare the output of the driver circuit with the transmitted signal.

As far as working principle of microphone is concerned, the impedance of microphone changes accordingly with the input signal[3]. Therefore, the output waveform can be obtained basically by voltage division circuit. Having the signal obtained, the output is isolated with a capacitor to prevent DC offset. The circuit is illustrated in Fig.6.

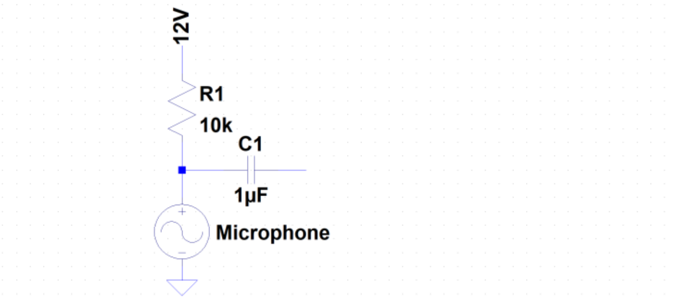

Fig.6. Microphone driver circuit.

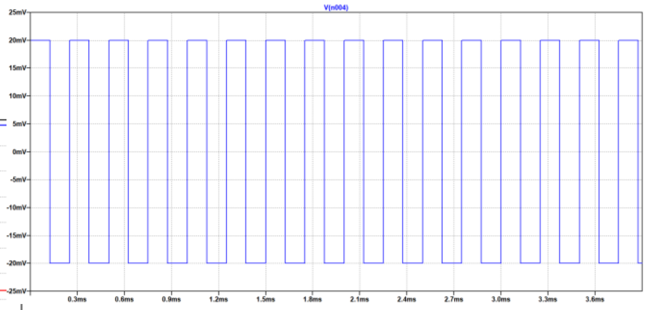

Fig.7. Output waveform of microphone driver circuit.

As we see in Fig.7. the output voltage waveform is not high enough to compare with the transmitted signal. Hence, an amplifier circuit is implemented for this purpose as shown in Fig.8.

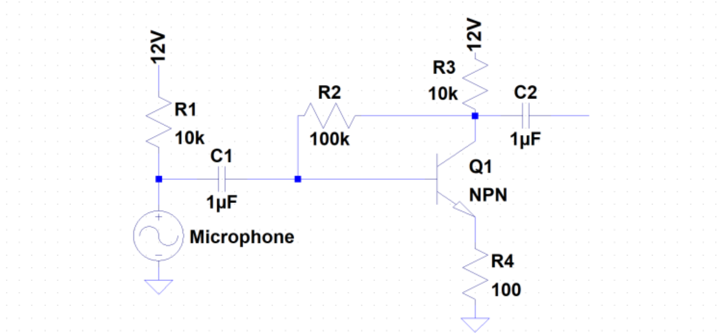

Fig.8. Microphone driver circuit with amplifier circuit.

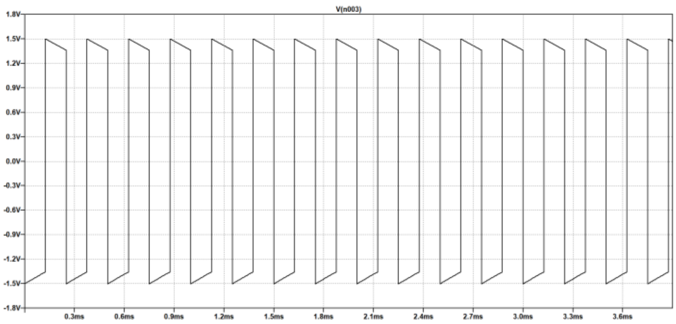

Fig.9. Output waveform of microphone driver circuit with amplifier circuit.

The desired output waveform is illustrated in Fig.9. As seen in Fig.9. the output of the microphone is amplified. The gain of the overall driver circuit can be calculated as 75V/V by observing the results of Fig.7. and Fig.9. Now two signals can be input to the mixer properly.

D. *Mixer*

In this part, two signals, namely the VCO output and microphone driver output, are multiplied with the help of the mixer circuit as shown in Fig.10.

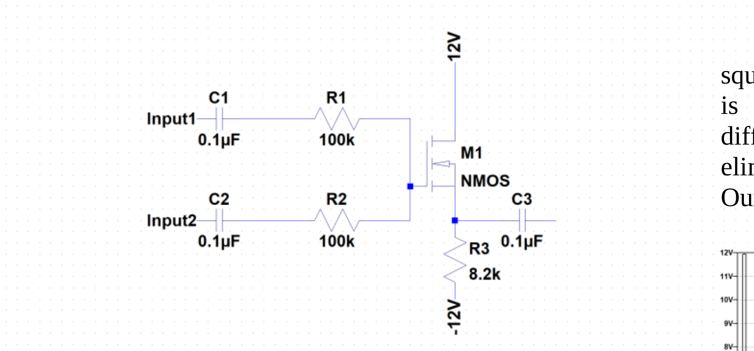

Fig.10. Mixer circuit.

The main purpose of this subcircuit is to determine the delay between transmitter and receiver signals. In other words, in order to get instantaneous frequency difference between these signals since this delay causes a difference in the frequency as well. The output waveform is composed of two signals with different frequencies which are the summation and difference of these two input signals. The multiplication of the two signals in the mixer circuit is achieved by using the nonlinearity of the transistor. The drain current of the MOSFET is given as follows[4]:

$$i_D=K_n/2*(v_{GS}-v_T)^2 \quad (2)$$

It can be inferred from (2) that the output voltage of the mixer circuit is proportional with the square of the summation of the two input signals. Therefore, they are multiplied and it can be seen at the output.

There will be two different frequency components in the output waveform as we can understand from the Fourier analysis i.e. multiplying in time domain means convolution in Fourier domain. The output waveform is illustrated in Fig.11.

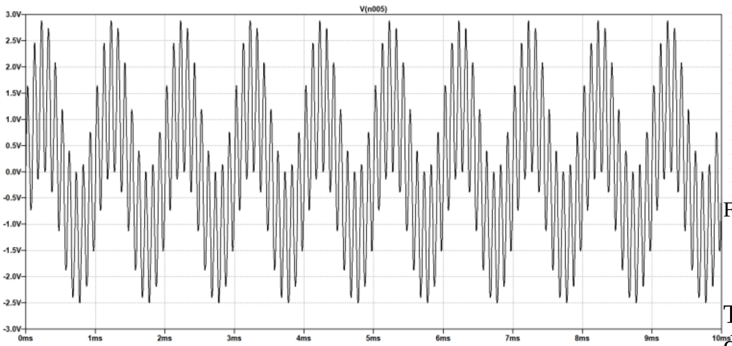

Fig.11. The Output waveform of the mixer circuit.

As seen in Fig.11. the output of the mixer is composed of the high frequency signal modulated by the low frequency signal i.e. the high frequency signal carries the low frequency signal.

Unfortunately, the mixer circuit did not work properly not because the topology is wrong, but the inputs are not in sinusoidal waveform. Our microphone output signal is in square waveform as shown in Fig.7. Also, the output of VCO is in square waveform. Thus, our mixer circuit output is different than it should be. This is because we could not eliminate the harmonics inside the square waveform properly. Our mixer output waveform is illustrated in Fig.12.

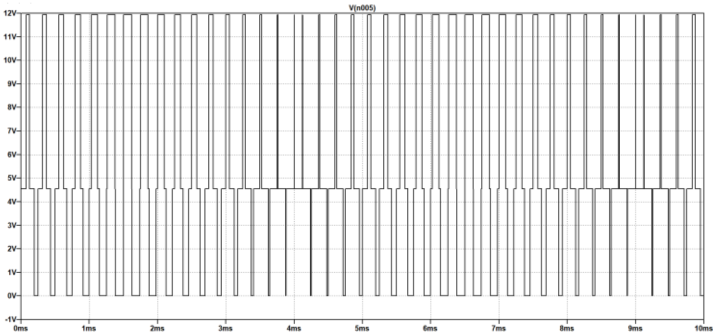

Fig.12. Output waveform of mixer circuit with square wave inputs.

E. Low Pass Filter

The output of the mixer circuit is composed of both difference and summation of instantaneous frequencies as mentioned above. However, we solely need the difference output since we are interested in distance. Therefore, two-stage passive RC low pass filter is implemented for this purpose. The circuitry is show in Fig.13.

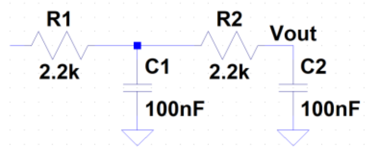

Fig.13. Two stage passive RC low pass filter.

The upper bound for the filter is arranged as 750Hz since the difference in the frequencies of the received and the transmitted signals will not go beyond 1kHz since we measure small distances in the lab. The resistance and capacitance values are chosen to arrange the cut-off frequency[5] according to the equation (3).

$$f_c = 1/2\pi RC \quad (3)$$

Finally, we can calculate the distance between the speaker and the microphone by taking into account the speed of sound, the bandwidth of the transmitted signal, the output frequency and the frequency of the input triangular signal as follows[6]:

$$D=v\Delta fT/BW \tag{4}$$

In equation (4), the v corresponds to the speed of sound waves i.e. 343m/s, Δf to the difference of the received and transmitted signals' frequencies, T to the period of the input triangular wave and BW to the bandwidth of the output of the VCO which is4kHz.

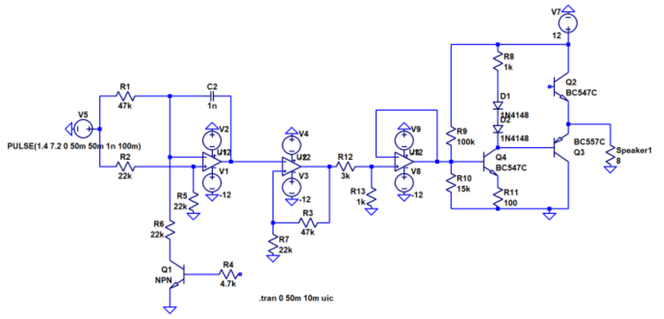

Fig.14. The transmitter part of the circuit

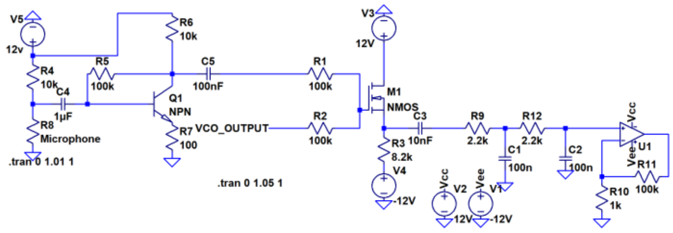

Fig.15. The receiver part of the circuit

In Fig.14. and Fig.15. the transmitter and receiver part of the circuit is shown.

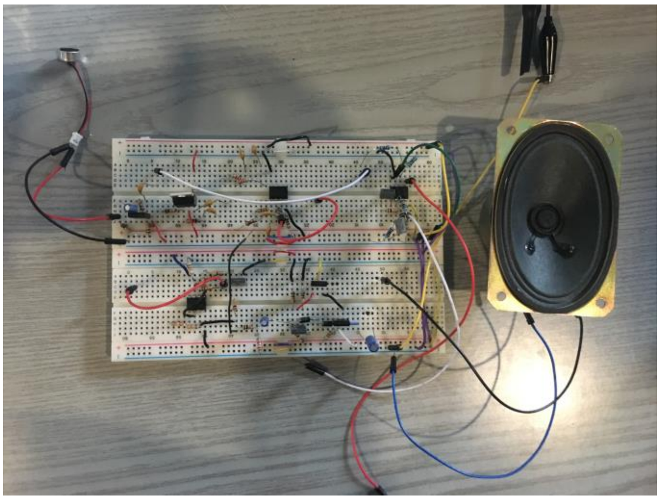

Fig.16. The overall circuit implemented on breadboard

In Fig.16. the overall circuit implemented practically on breadboard is shown.

IV. Conclusion

In conclusion, we designed a distance measurement system based on FMCW principle. For this purpose, we generated a varying frequency signal whose frequency is controlled by a triangular input signal and transmitted this signal via a speaker. We used a class AB output stage to load the speaker since it has low impedance and we were required to dissipate at least 1Watt. We received this signal via a microphone and multiplied the transmitted and received signals. The mixer gives an output whose frequency components are the difference and the summation of the two input signals' frequencies. We constructed a low pass filter at the output of the mixer to obtain the difference component of the signal. By observing this final signal's frequency, we calculated the distance between the speaker and the microphone.

In practice, we produced a square wave from the VCO. All parts of the circuit worked until the mixer. The mixer did not give a reasonable output since the square wave consists of infinite harmonics and the mixer multiplies all those harmonics. Therefore, the output of the mixer is the multiplication of all the harmonics and it does not have any meaning to infer the distance information.

In this project, we learned how to design a VCO and adjust its output frequency range. We learned how to design power amplifiers to load small resistances and meet a certain power condition. We learned how to design microphone driver circuits and convert sound waves into meaningful electrical signals. We learned how to design mixer circuits and multiply two input signals by using the nonlinearity of the components. We learned about FMCW principle and where and how it is used.

Overall, this project was really helpful to learn new concepts and apply what we have learned in practice and it helped us gain a lot of experience in analog electronics circuits and their uses.

References

[1] Texas Instruments, "LM358 Low Power Dual Operational Amplifiers", FMCW Radar Sensors(2011). Retrieved from https://siversima.com/wp-content/uploads/FMCW-Radar-App-Note.pdf

[2] Electronic Tutorials, "Class AB Amplifier", https://www.electronics-tutorials.ws/amplifier/class-ab-amplifier.html

[3] Best Sound Electronics, "Spesifications of Electret Condenser Microphone", https://www.endrich.com/fm/2/SOB-413S42-EM.pdf

[4] Cengiz Beşikçi, "EE212 Lecture Notes"

[5] All About Circuits, "Low Pass Filters", https://www.allaboutcircuits.com/textbook/alternating-current/chpt-8/low-pass-filters/

[6] FMCW Radar Sensors(2011). Retrieved from https://siversima.com/wp-content/uploads/FMCW-Radar-App-Note.pdf