

III. RECEIVER

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.

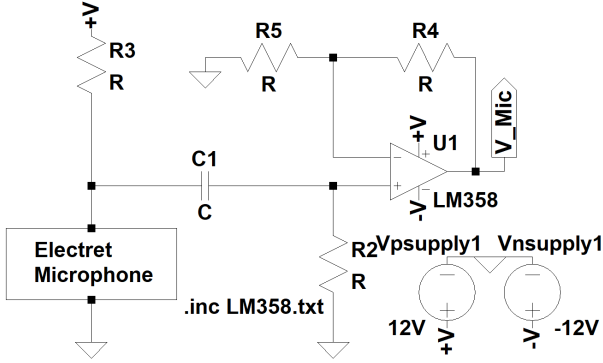


Figure 3: Driver Circuit for Electret Microphone

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.

B. Mixer

Mixer is the device that accepts two input signals and gives an 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

$$O/P \text{ Signal} = A(f_1 + f_2) + B(|f_1 - f_2|)$$

where A and B are the signals having frequencies

$$f_A = f_1 + f_2 \text{ \& } f_B = |f_1 - f_2|$$

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 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 4.

Therefore, the time shift "Time 1 - Time 0" is proportional to the " $|f_2 - f_1|$ ". 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

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

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

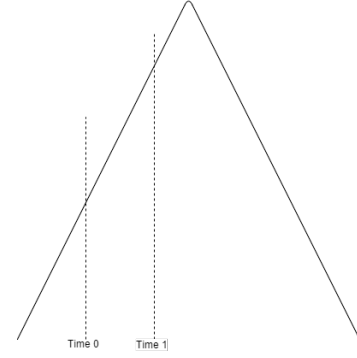


Figure 4: One Period of Triangular Wave

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 5.

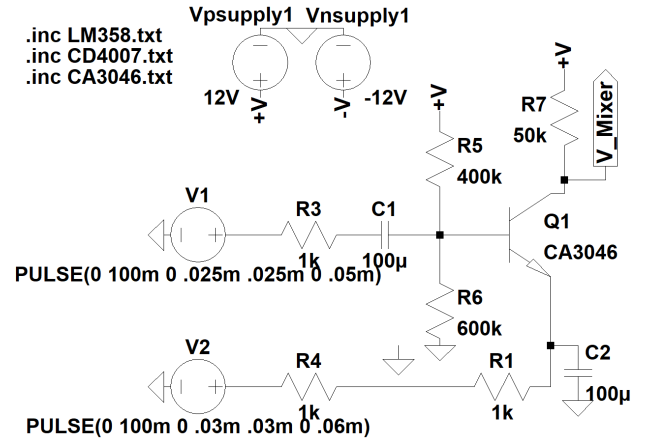


Figure 5: A Mixer Circuit

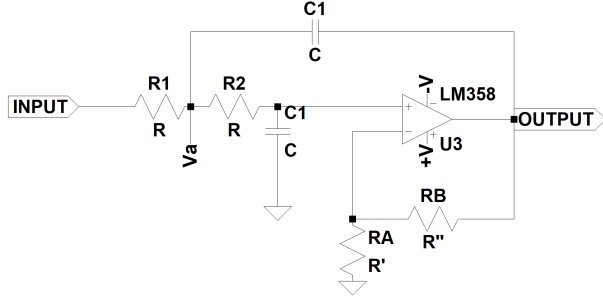


Figure 6: A Second Order Low-Pass Filter

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 than 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 6.

IV. GENERAL DIAGRAM

General diagram of the project can be seen at Figure 7.

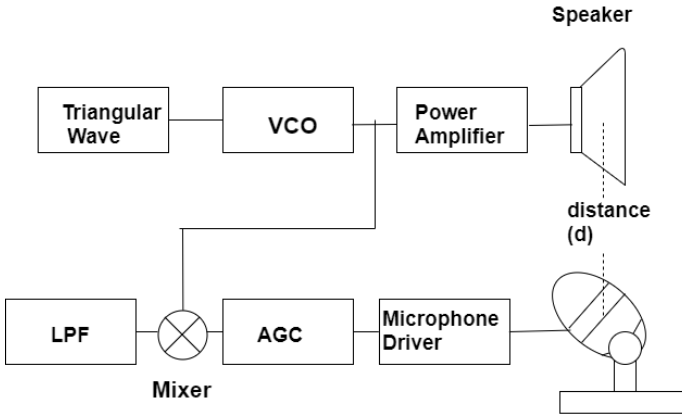


Figure 7: General Diagram of the Project

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 $(|f_1 - f_2|)$ that are frequencies of two inputs of the mixer. If this signal is low-passed, the filtered signal have $(|f_1 - f_2|)$ frequency. By measuring $(|f_1 - f_2|)$, the distance between the speaker and the microphone can be found by a simple algebra, $K * (|f_1 - f_2|)$ where K is a constant to be determined and having unit of $\frac{\text{Meter}}{\text{Hertz}}$.