Bangladesh University of Engineering and Technology

DEPERTMENT OF ELECTRICAL AND ELECTRONIC ENGINEERING

Course-EEE-310 **Group no-6**

Project Report

Course Name: Communication Systems I Laboratory

Project Title: Sampling & Reconstruction of A Signal Using Electronic ICs

EEE 19 - A1

Project submitted by: 1906029-Chinmoy Biswas

1906030-Abrar Al Shadid

1906031-Vivek Chowdhury

1906032-Sabir Mahmud

1906033-Nafis Faisal

Supervised By:

Dr. Md. Saifur Rahman

Barproda Halder

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Objectives :

The main objectives of our project are:

- To learn the importance of sampling and reconstruction in digital communication system.
- Getting an idea about how sampling and reconstruction of an analog signal is done.
- Knowing how to apply the sampling and reconstruction method in practical life.
- Learning the problems or limitations we may face in real life while implementing this project.

Theory :

The signals we use in the real world, such as our voices, are called 'analog' signals. Meaning that these are actually continuous time signal. But when it comes to machines and computers, they can work with only digital signals. So, in order to process these signals in computers, we need to convert the analog signals to 'digital' form. While an analog signal is continuous in both time and amplitude, a digital signal is discrete in both time and amplitude. The process we use to convert a signal from continuous time to discrete time, is called sampling. Sampling is also very helpful in transmission process of a signal. As sampling allows only certain portions of an analog signal to pass, it reduces the transmitting power significantly.

Once we send the discretized version of a signal through communication media, we need to rebuilt the sampled signal at the receiving end. The process used to retrieve the original signal from its digital form is called ‘Reconstruction’. In order to reconstruct a digital signal, we need to simply interpolate the discrete portions of it. And thus, we get our analog signal reconstructed. In real life, we usually use Lowpass Filters for interpolation.

Various types of sampling:

Theoretically we find three different methods of sampling.

a) Ideal

b) Natural &

c)Flat Top sampling.

Three methods are briefly described below:

1) Ideal Sampling:

In this method, the sampling signal is a periodic impulse train. The area of each impulse in the sampled signal is equal to the instantaneous value of the input signal x(t). Ideal sampling is also known as instantaneous or impulse train sampling.

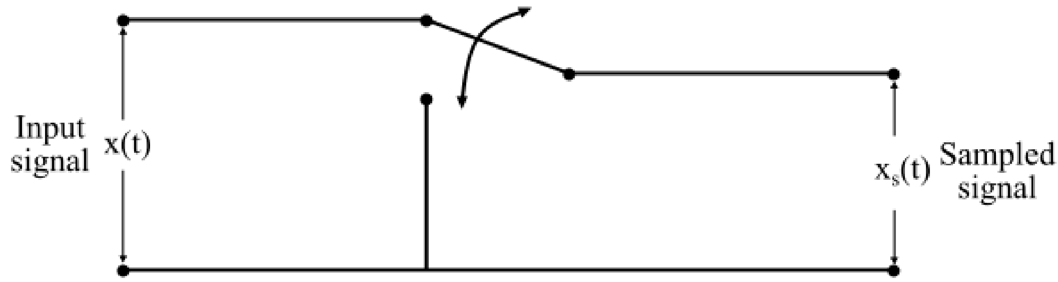


Fig: Ideal Sampling – Sampler Circuit

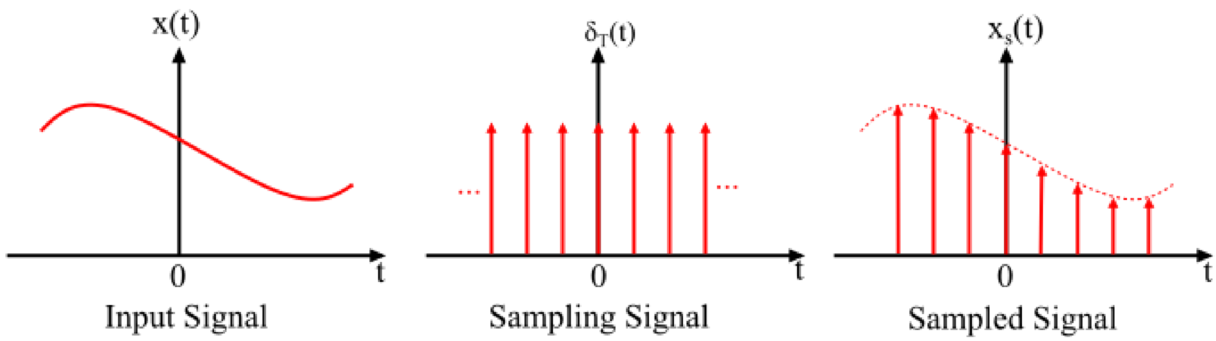


Fig: Ideal Sampling – Process of Sampling

2) Natural Sampling:

In this sampling technique, the sampling signal is a pulse train. In natural sampling method, the top of each pulse in the sampled signal retains the shape of the input signal x(t) during pulse interval.

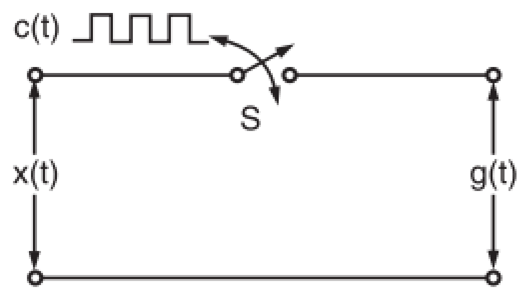


Fig: Natural Sampling – Sampler Circuit

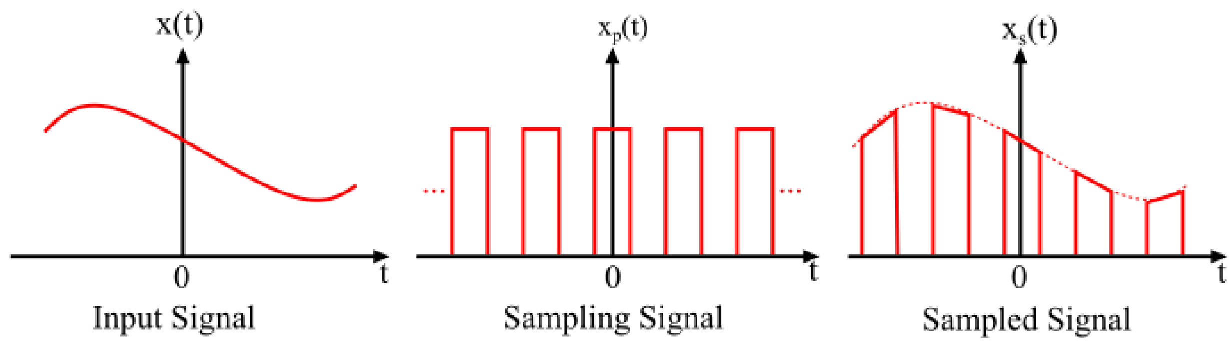


Fig: Natural Sampling – Process of Sampling

3) Flat Top Sampling:

In the flat top sampling, the sampling signal is also a pulse train. The top of each pulse in the sampled signal remain constant and is equal to the instantaneous value of the input signal $x(n)$ at the start of the samples.

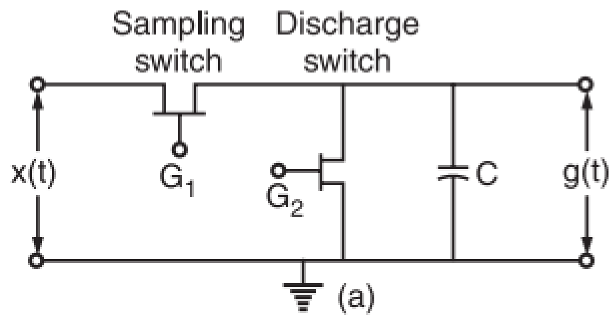


Fig: Flat Top Sampling – Sampler Circuit

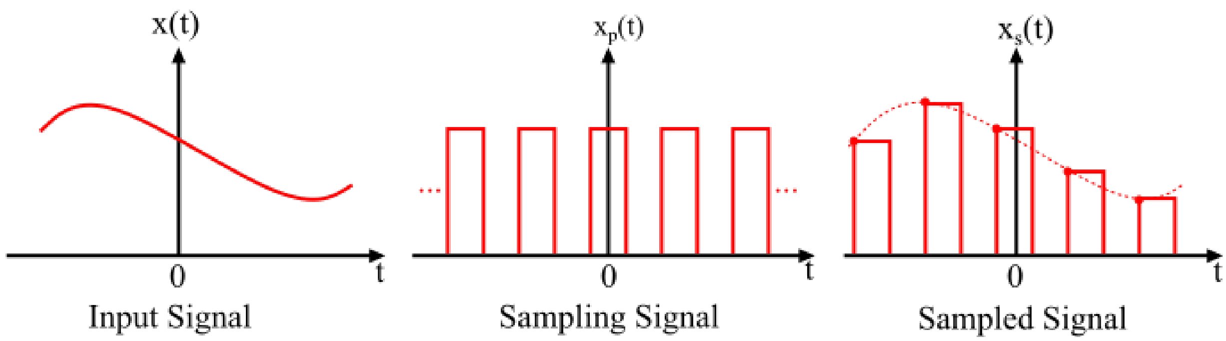


Fig: Flat Top Sampling – Sampling Process

We have seen, three types of sampling process above. The portions of the original signal we keep after sampling are known as samples.

One thing needed to be mentioned here that in real life we cannot use the 'Ideal Sampling' method. Since it is impossible to generate zero width pulses practically because it is not possible to perform any kind of switching operation for infinitesimally small period. So, we use only natural or flat-top sampling method in real life.

Reconstruction Process:

To get the original version of a signal from its sampled version, we need to reconstruct it. To do this we have to simply connect the missing portions between the samples. This connecting method is known as interpolation, through which we can get a smooth curve as the original signal.

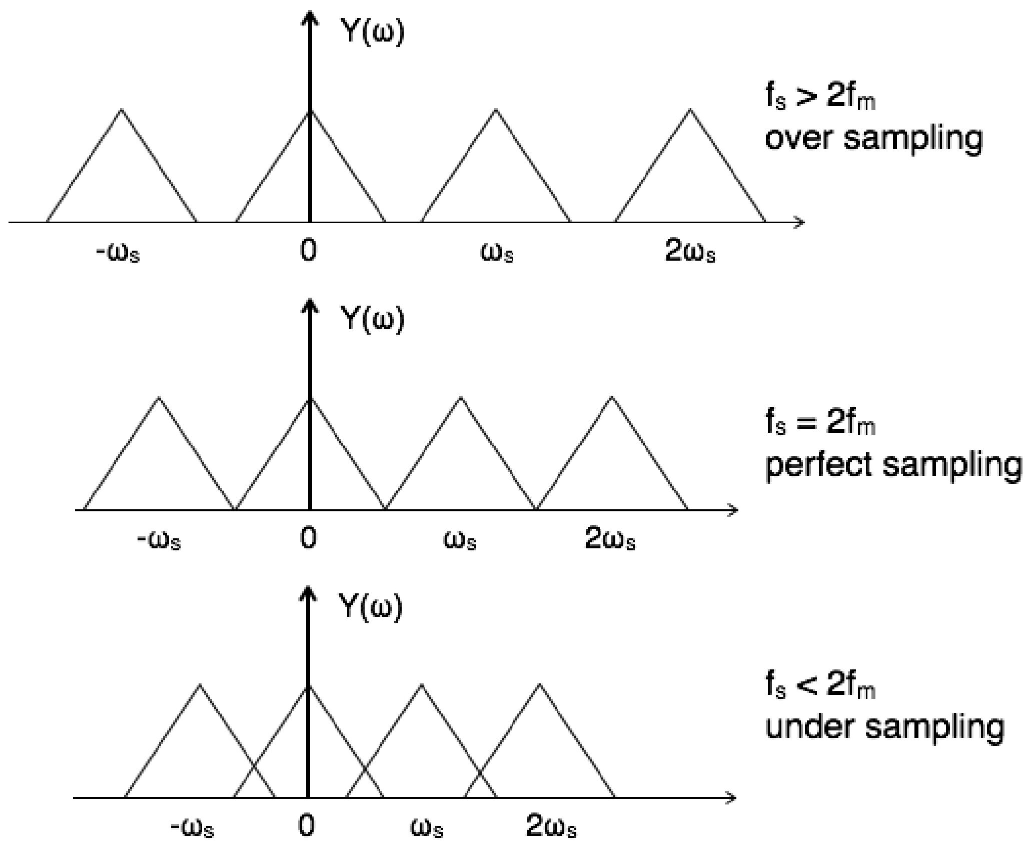
There are some criteria that needed to be maintained if proper reconstruction is desired. We can not simply connect samples at random distances and get back the original signal. For proper reconstruction the sampling rate has to be above certain value. This criterion is known as **'Nyquist Criterion'.** Which states that - to get back the original signal by interpolation, the sampling rate (or sampling frequency) should be atleast twice times higher than the original signal frequency.

$$F_s \geq 2F.$$

where, F_s= Sampling Rate

F = Original Signal Frequency

To get nearly perfect reconstruction, we usually use much higher sampling frequency than just twice of the original frequency in real life.



Practically, Lowpass Filters work as very good interpolators. When we sample a continuous signal, the resulting discrete signal includes more frequency components than did the analog signal. Actually, these extra frequencies that are included later are high frequency components. If we block these high frequency components, we can get back the original signal only. That's why we use appropriate Lowpass filters to retrieve the main signal from its sampled version.

The higher the order of a low pass filer, the better the reconstruction is. As higher order LPFs can block the higher frequency components and allow the desired frequency component to pass more precisely.

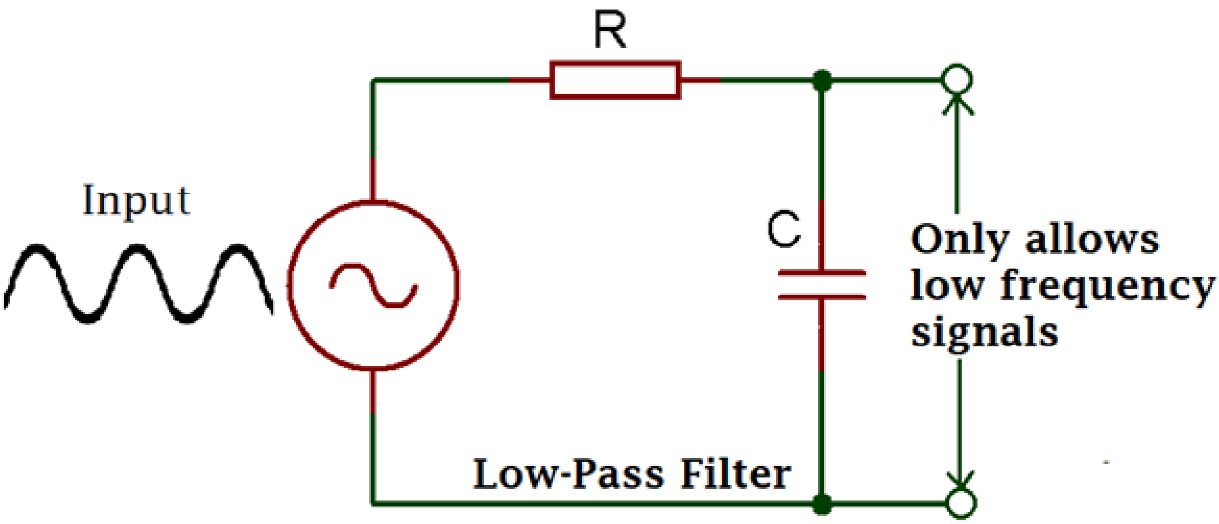


Fig: Construction of a first order passive low-pass filter

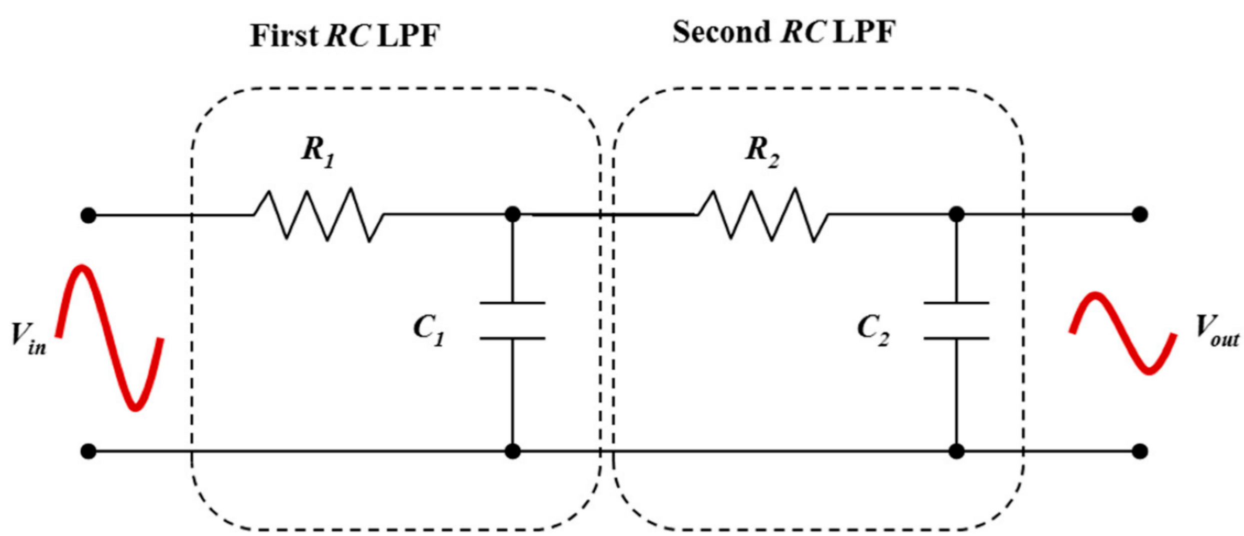


Fig: Second Order Low Pass Filter

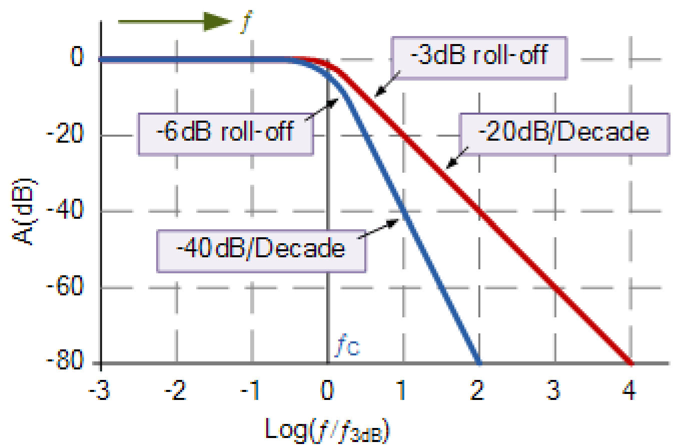


Fig: Normalized Low Pass Frequency Response plot

From frequency, response we can clearly visualize that higher order filters have steeper curves. That's why higher order Lowpass filters are always better choices for signal reconstruction.

Equipment:

For this project, the following equipment were used:

1. NE555p, a 555 timer
2. LF398, a sampler IC
3. Potentiometer of 10k, 50k and 100k
4. Resistors of values 1k, 47k
5. Capacitors of values 0.001 μF, 1 μF, 10 μF
6. Breadboard
7. PCB
8. Jumper cables
9. Function generator
10. Oscilloscope

Circuit diagram:

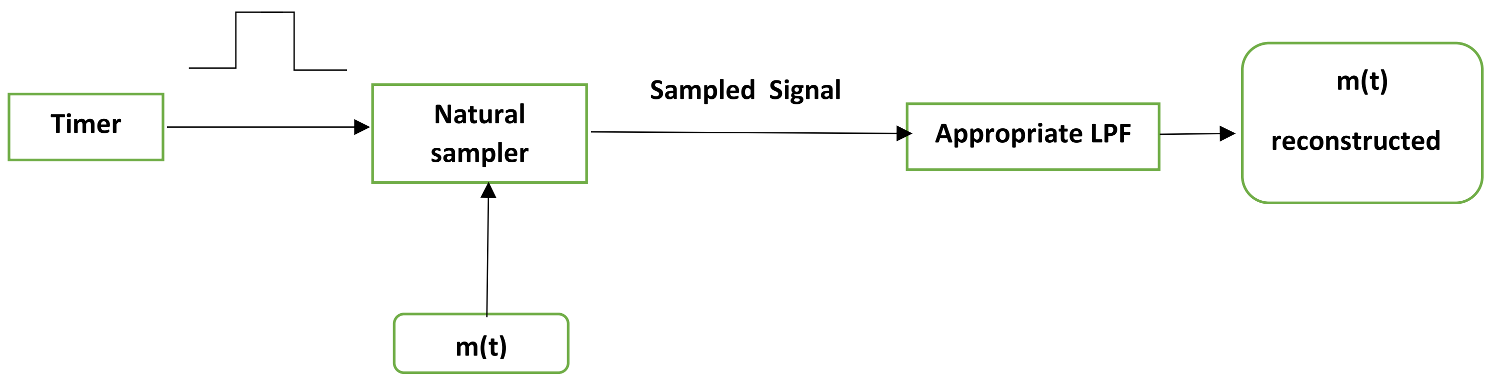


Fig: Circuit for natural sampling

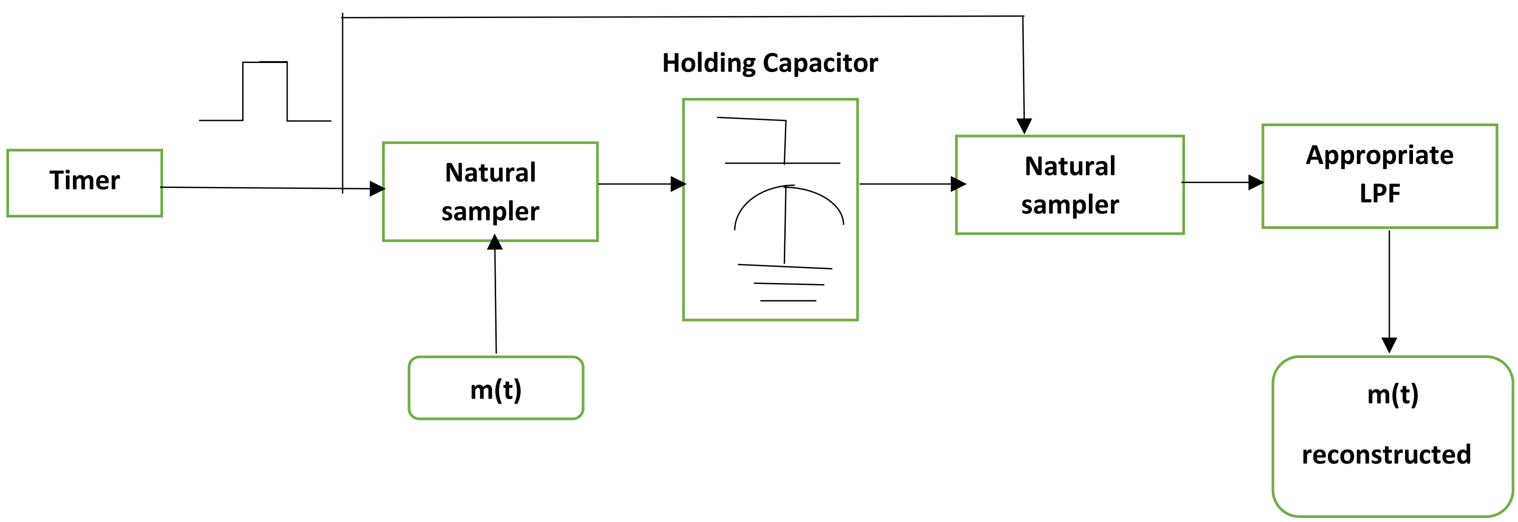


Fig: Circuit for Flat-top sampling

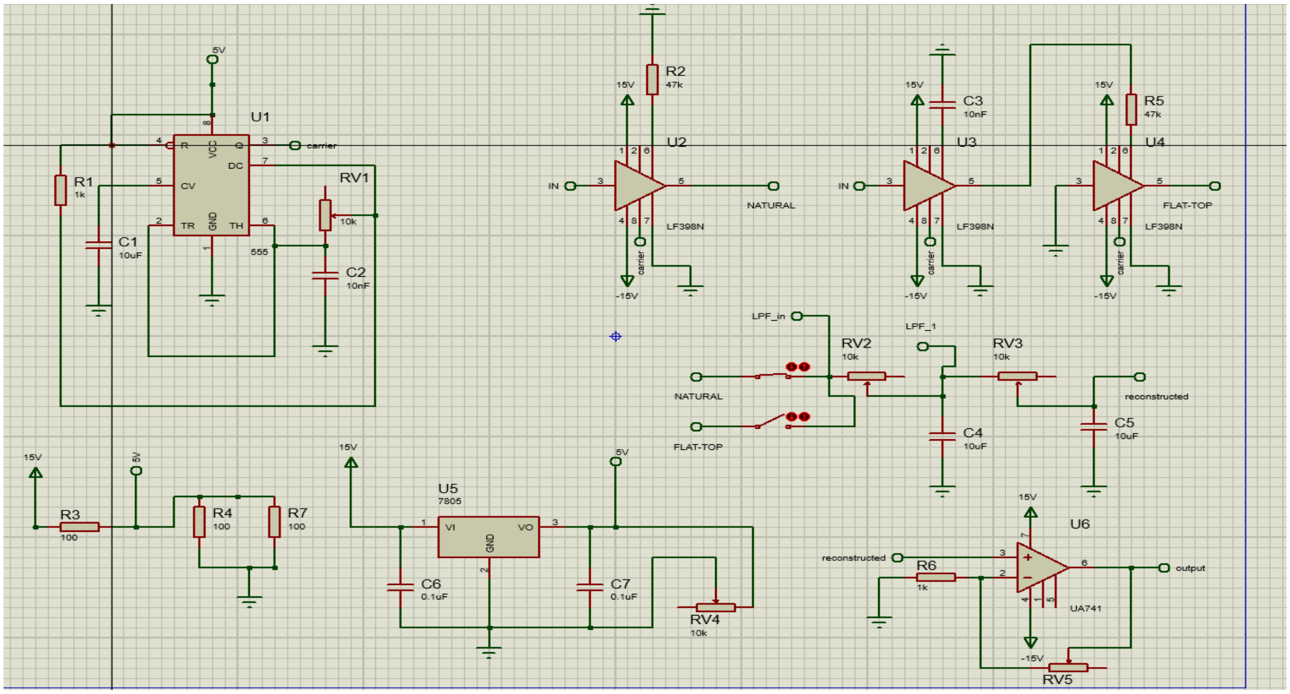


Fig. Proteus Layout for PCB board

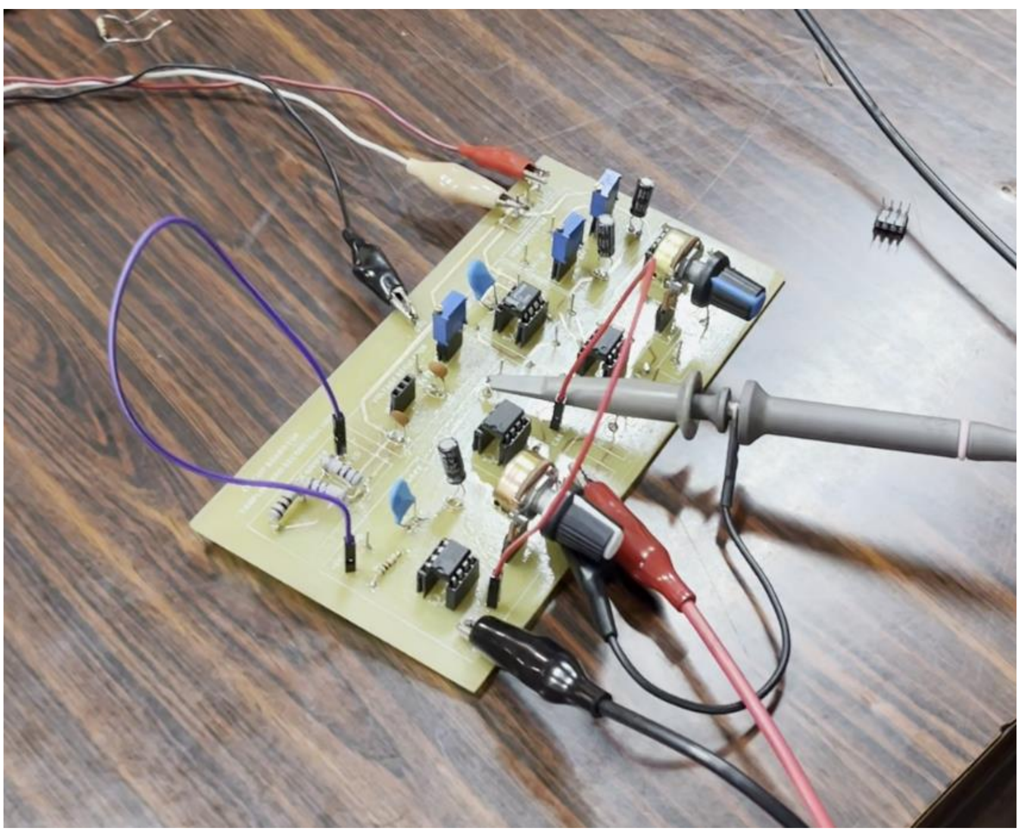


Fig. PCB board of the sampler circuit

Working principle:

In the project, both natural sampling and flat top sampling were done following the circuits as shown in the diagram above.

In case of natural sampling, at first a clock pulse is generated by the NE555p, a 555 timer. The clock pulse is just a rectangular pulse train with pulse width equal to the width of the samples and time period equal to the sampling time interval. The time period and pulse width of the generated clock pulse can be adjusted by rotating the knob of the 10k potentiometer connected to it.

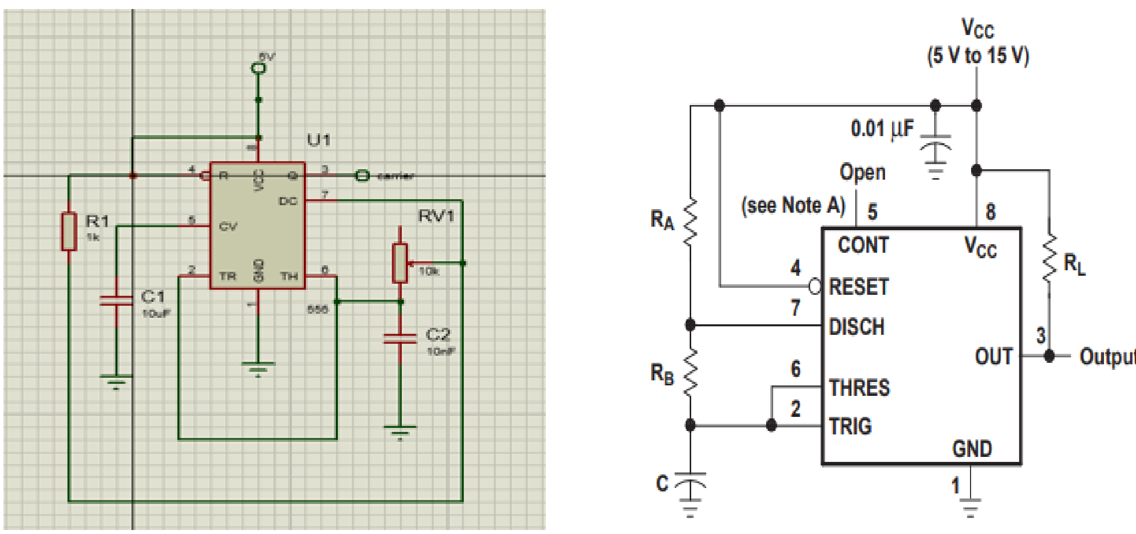


Fig: NC555p Timer used to create clock pulse

The created clock pulse was kept at a frequency between 8.6 kHz and 10 kHz by adjusting the potentiometer to sample a sinusoidal message signal of frequency 1 kHz. The clock pulse is shown below:

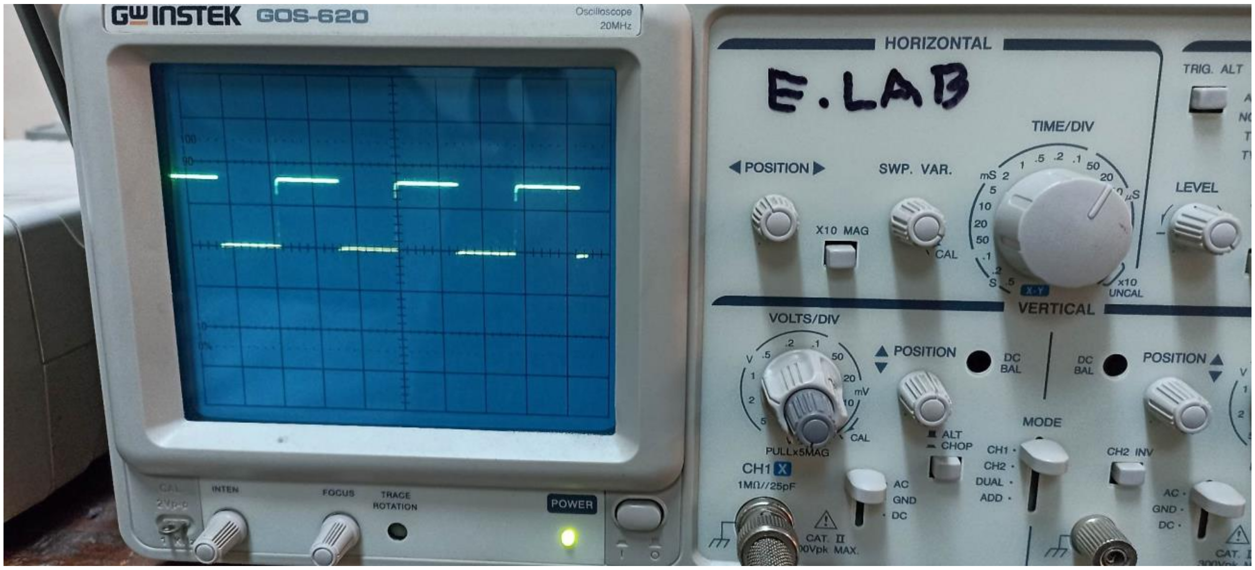
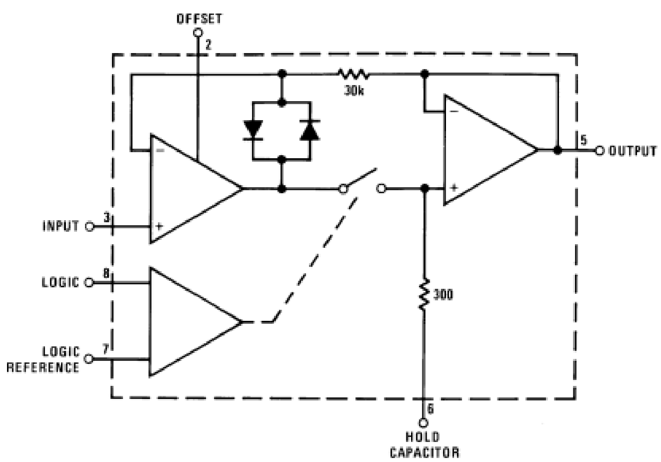


Fig. Clock pulse found as NC555 timer output



LF398

Fig. LF398 Sampler IC

This clock pulse is fed into a sampler circuit along with the sinusoidal message signal. The sampler used in this project is the LF398 IC whose functional block diagram is shown above. The IC samples signals by turning a switch on and off according to the clock pulse received by it. When it receives a clock pulse at ON state that is near 5V, it switches the circuit on and the message signal passes through. When the clock pulse is at OFF state that is near 0V, the circuit gets turned off and no message passes. This switching operation is done by a MOSFET inside the IC. The IC also contains a hold circuit that will be used for flat top sampling later. The message signal gets sampled naturally when passed through the sampler IC.

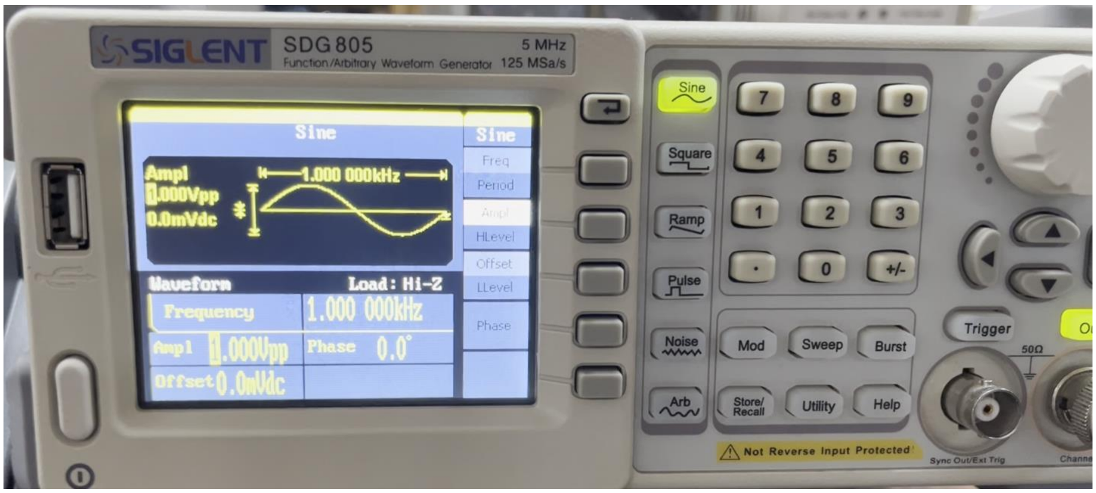


Fig. Message signal generated at function generator

The output of the naturally sampled message signal is as shown below:

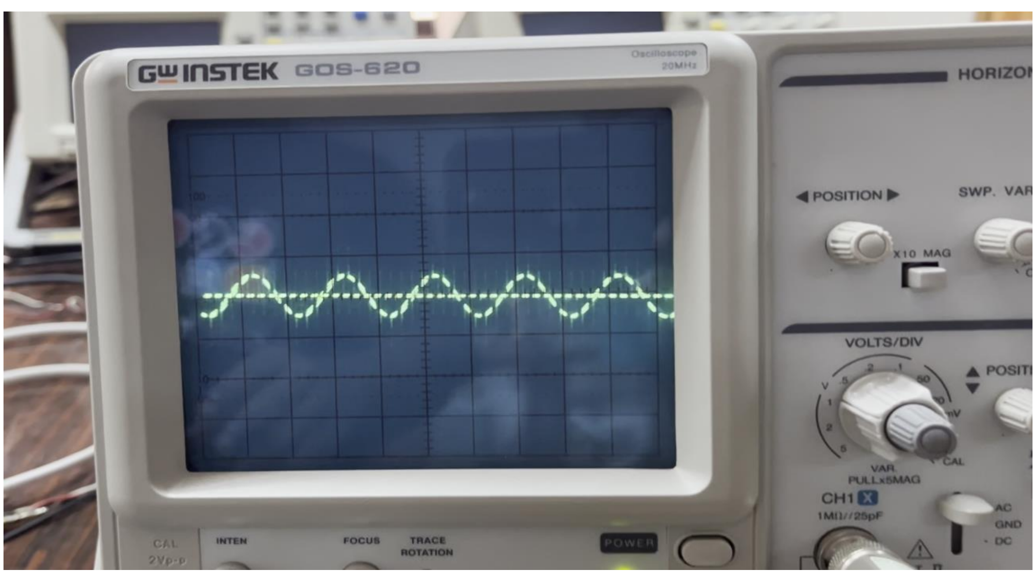


Fig. Naturally sampled message signal

This naturally sampled signal is then passed through a low pass filter for reconstruction. The output of the reconstruction is shown below:

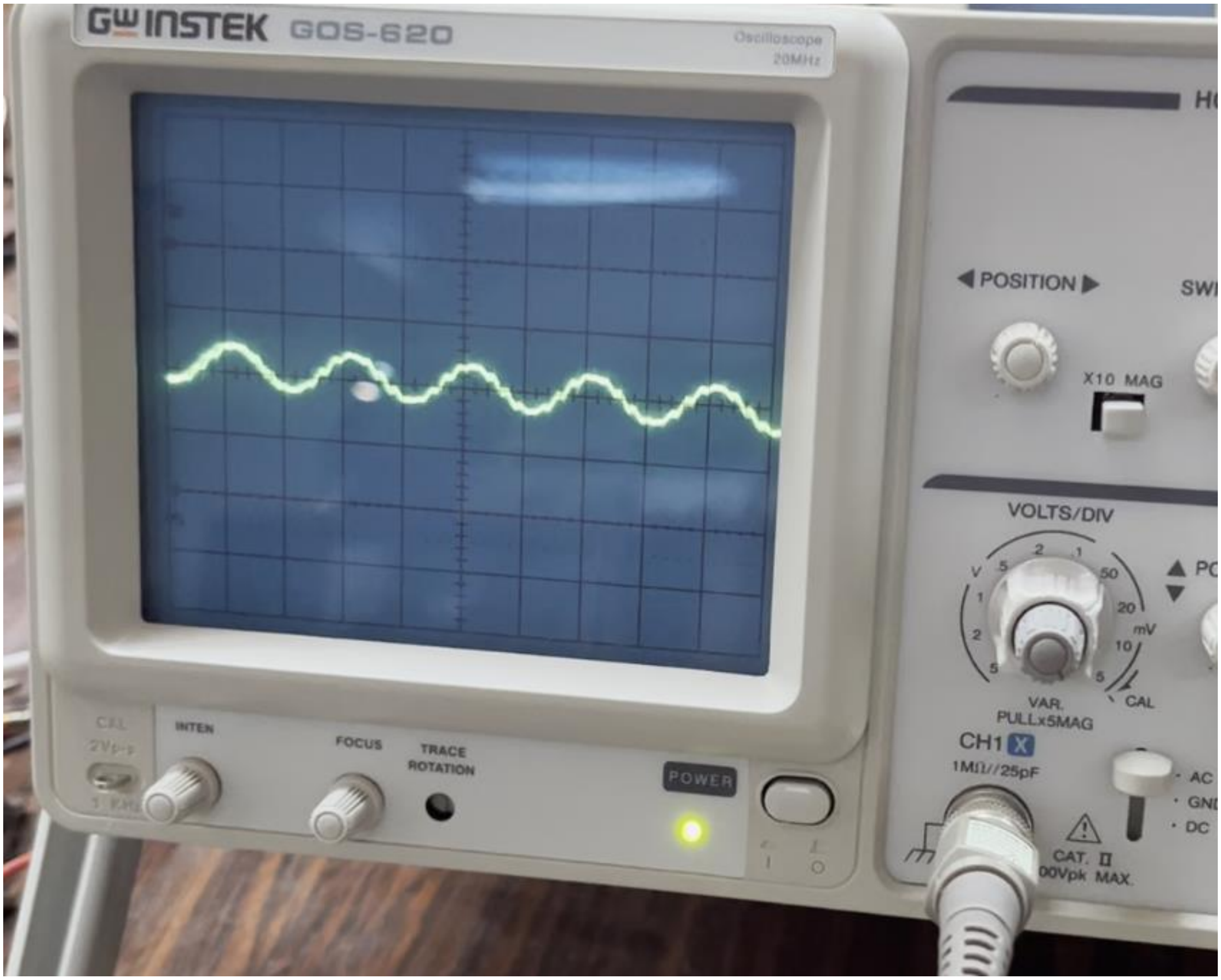


Fig. Reconstructed signal after 1st LPF filter

As can be observed, there is presence of ripple in the reconstructed signal. This is due to the presence of some high frequency components that could not be filtered out as the filter is non ideal. To remove this, another filter is cascaded and the output of the 2nd filter, that is the final reconstructed signal is as shown below:

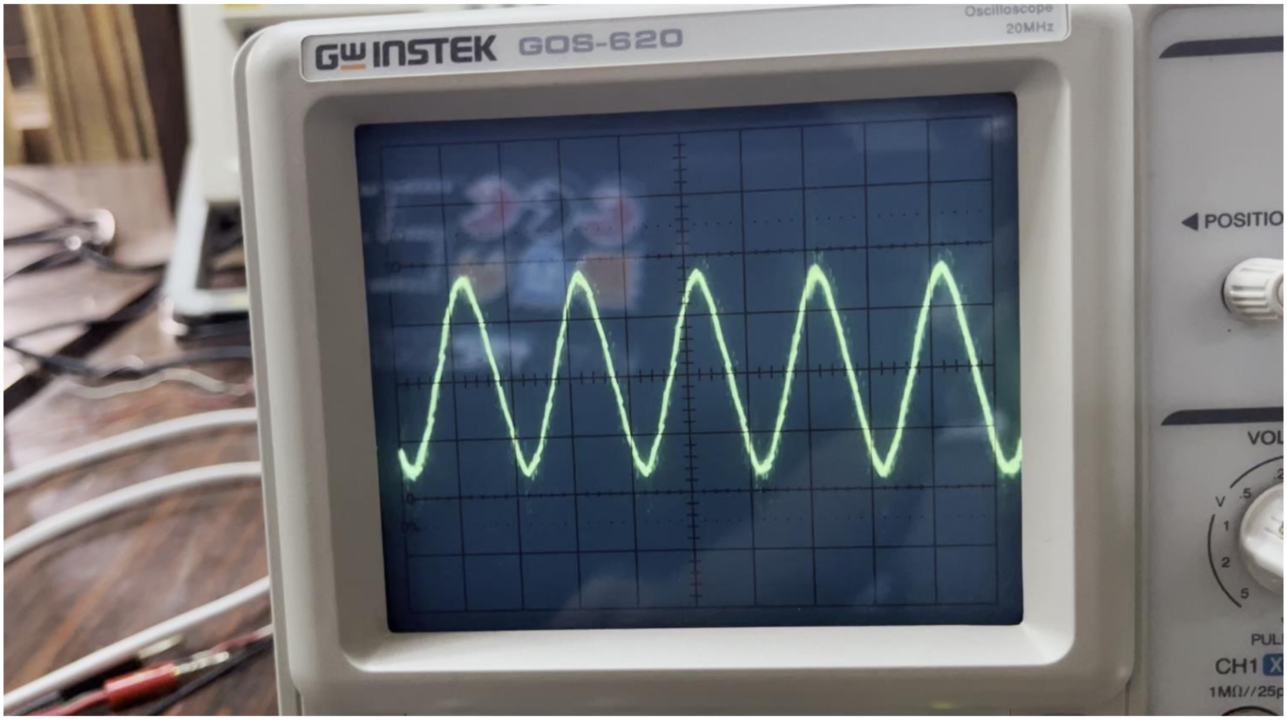


Fig: Reconstructed signal after 2nd order filtering

This time, the output is much better. The higher the order of the filter the better will be the reconstruction. This can be understood better from the graph of frequency response of various order filters. Higher order filters have better drop off after threshold frequency and so they allow lesser higher frequency components. Lower order filters do not have such sharp roll off and so much higher order component are left over. This can be easily observed as ripple as can be seen in our reconstruction after 1st order filtering. These ripples were eliminated after 2nd order filtering done by cascading another filter with the first one. The results can be seen above and show that the higher order components have been satisfactorily removed.

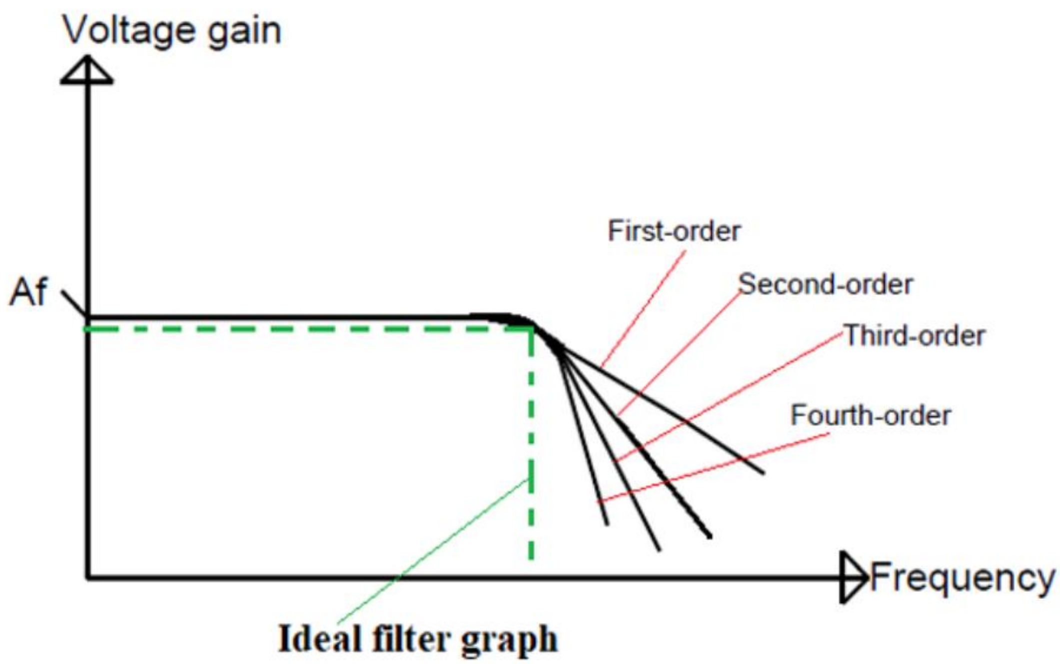


Fig. Frequency Response of Filters

The above graph shows how the roll off becomes steeper with filter order. The 2nd order filter has a roll off of about -40dB/decade while the 1st order has only a roll off of -20dB/decade. That's why 2nd order filtering can eliminate higher order components while only 1st order can't.

Thus, the original signal has been reconstructed well enough, concluding the natural sampling part.

Now comes flat top sampling.

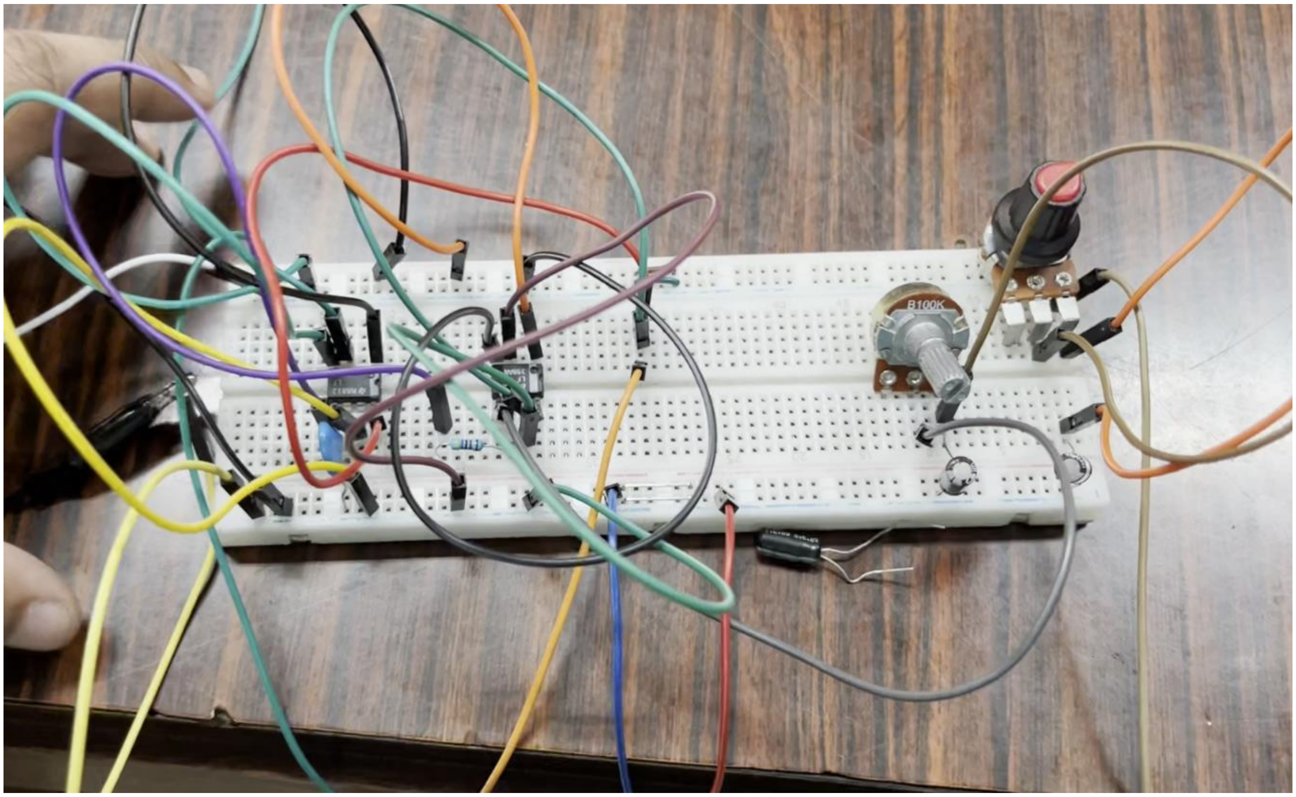


Fig. Breadboard circuit for sample-and-hold circuit

For flat top sampling, the clock pulse from the 555 timer is fed into the LF398 sampler IC. The circuit after this is similar but contains another sampler inside with a port for placing a capacitor in between. All of this is contained in a single LF398 IC. The hold circuitry utilizes a capacitor to hold the value of the signal that is the voltage received at the very beginning of switching on and keeps it stable for the whole on phase. Finally, the flat top is found by sampling the output of the capacitor. Therefore, a flat top sampling circuit is just two natural samplers with a capacitor connecting them. Graphically, the effect is flattening the heads of the samples in natural sampling.

The output of the flat top sampled message signal is shown below:

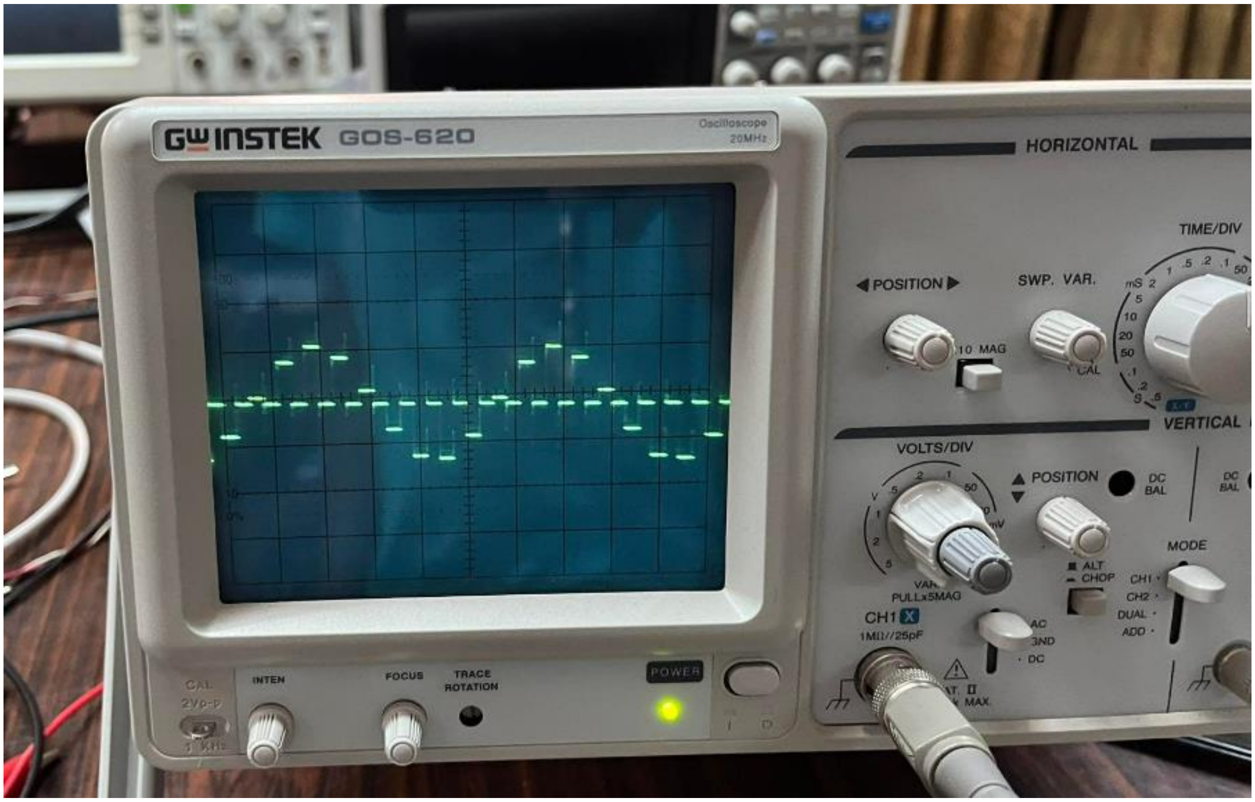


Fig. Flat top sampled signal

This signal is sent to the 1st order low pass filter for reconstruction. After this the signal is again filtered, thus going through 2nd order low pass filtering. The 2nd order filtering removes any ripple that might have remained due to the presence of higher frequency components after 1st order filtering. The output of the 2nd order filter is given below:

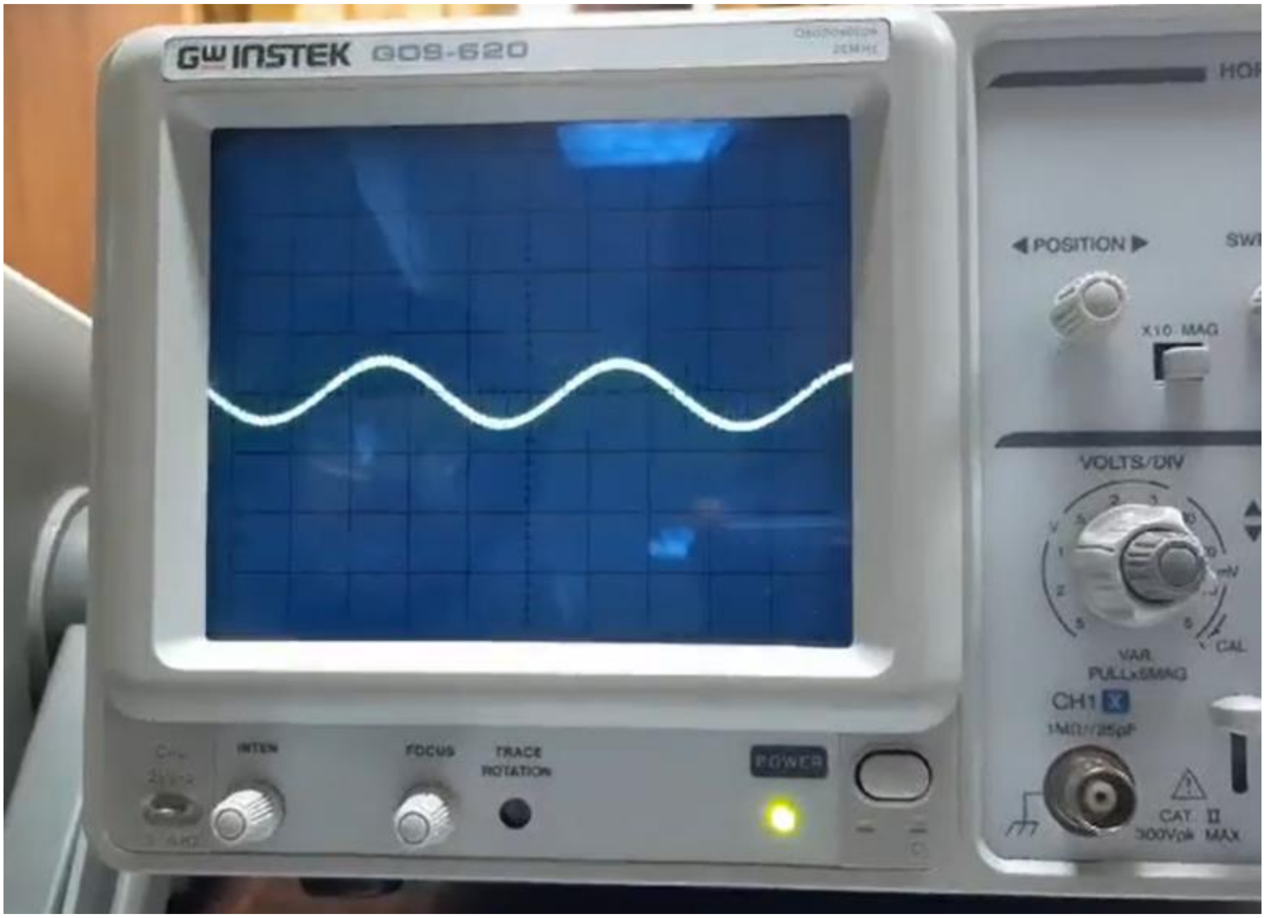


Fig. Output of 2nd order filter

This concludes the flat top sampling function of the PCB.

Problems faced during the project:

i. <u>PCB soldering:</u> The soldering done would sometimes come loose or not have proper connection with the circuit and thus the expected output was not found at times

ii. <u>PCB shorting:</u> The first PCB did not have shorting between two layers and thus the required output was not found between them. Realizing this error in the board took some time and set us back by a bit. A 2nd PCB was ordered afterwards.

iii. PCB construction delay: The PCB manufacturer printed the 2^{nd} PCB but made mistakes while printing it. As such the PCB delivery was delayed and progress was halted.

Conclusion:

Sampling and reconstruction are very important functions for converting any analog signal to digital signal, reducing the signal power in the process so that equipment can handle that power and also to convert the digital signal to analog. This project implements the two types of sampling most commonly used- uniform natural and flat top sampling. Sampling is used before performing any kind of digital operation on any analog signal and also before transmitting because this reduces the transmission power. The outputs got from the project were almost optimum and so, the sampling operations were performed perfectly by the circuit created during the project. The reconstructions were also quite good but required 2^{nd} order filtering for better output. Various problems were faced during the construction of the project due to which the project got delayed. But the final result found from the project was satisfactory. A better understanding of the theory underlying sampling was achieved along with a gain in practical knowledge involving circuit implementation, PCB designing and troubleshooting. This project is very useful in all kinds of applications involving digital signals as it includes both analog to digital conversion and also digital to analog conversion using low pass filters.