Aim: To examine sampling and reconstruction of signal, verify Nyquist criteria by varying sampling frequency. Draw the sampled version of waveform for the conditions. (i) $F_s < 2F_m$ (ii) $F_s > 2F_m$ and (iii) $F_s = 2F_m$ where F_s = sampling frequency, F_m = maximum base band frequency and represents the output response for different order LPF. Use virtual mode with appropriate software.

Apparatus
Nyquist applet.

Theory A continious time (or analog) signal can be stored in a digital computer, in the form of equidistant discrete points or samples. The higher the sampling rate (or sampling frequency, f_s), the more accurate would be stored information and the signal reconstruction from its samples. However high sampling rate produces a large volume of data to be stored and make neccessary the use of very fast analog-to-digital convertor.

Analog signal - continious time varying feature

Digital signal - It represents data as a sequence of discrete values, at any given time it can take one of finite number of values.

Techniques that can be used for analog to digital conversion:

PULSE CODE MODULATION (PCM):

- Sampling
- Quantization
- Encoding

Sampling:

- It is the process of measuring the instantaneous values of continious time signal in discrete form.
- Sample is a piece of data taken from the whole data which is continious in time domain
- when a source generates an analog signal and if that has to be digitized, having 1's and 0's i.e. high or low, the signal has to be discretized in time and this discritization of analog signal is called sampling.

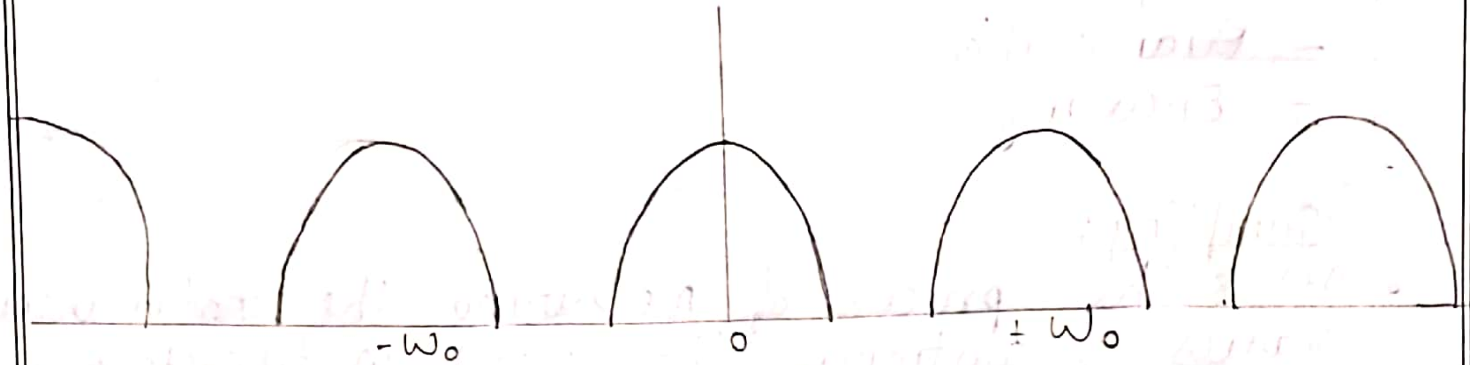
It is obvious that much higher sampling rate is required for sampling a signal which is rich in high frequency.

The minimum sampling frequency of a signal that it will not distort its underlying information should be double the frequency of its highest frequency component. This is the Nyquist sampling Theorem.

Condition -1 ($f_s > 2W$)

- Oversampling If sampled at higher rate then $2W$ in the frequency domain ($f_s > 2W$)

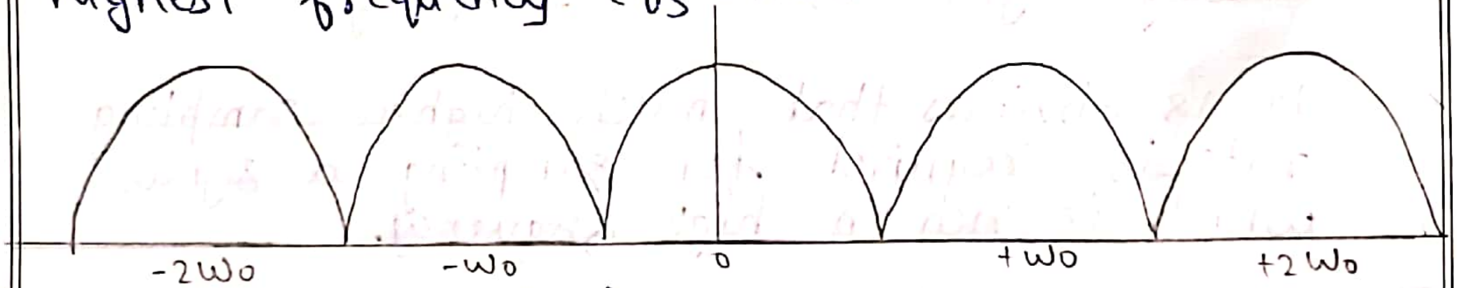
$$X_s(\omega) = \frac{1}{T_s} \sum_{n=-\infty} X(\omega - n\omega_0)$$



Here the information is reproduced without any loss. There is no mixing up and hence recovery is possible.

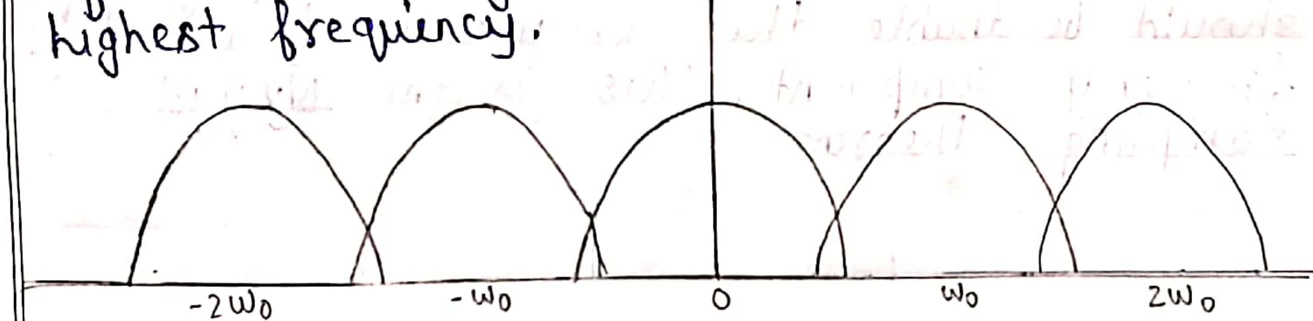
Condition -2 ($f_s = 2W$)

- If the sampling rate is equal to twice the highest frequency ($f_s = 2W$)



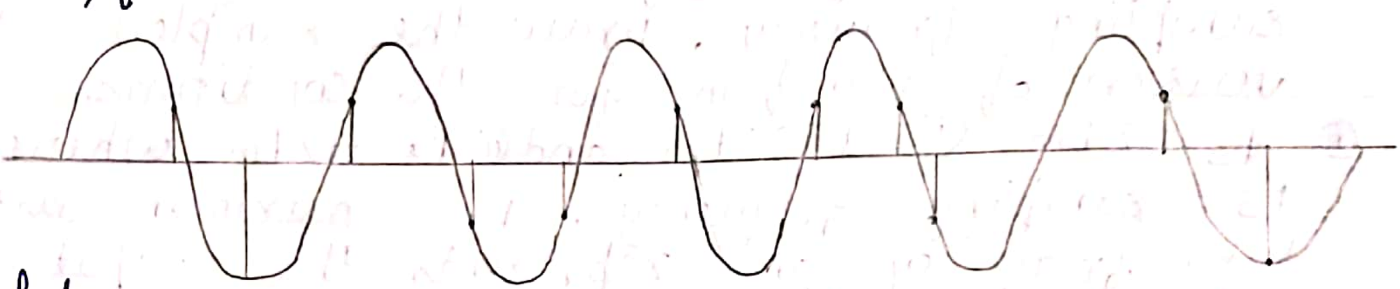
Condition -3 ($f_s < 2W$)

If the sampling rate is less than twice the highest frequency.

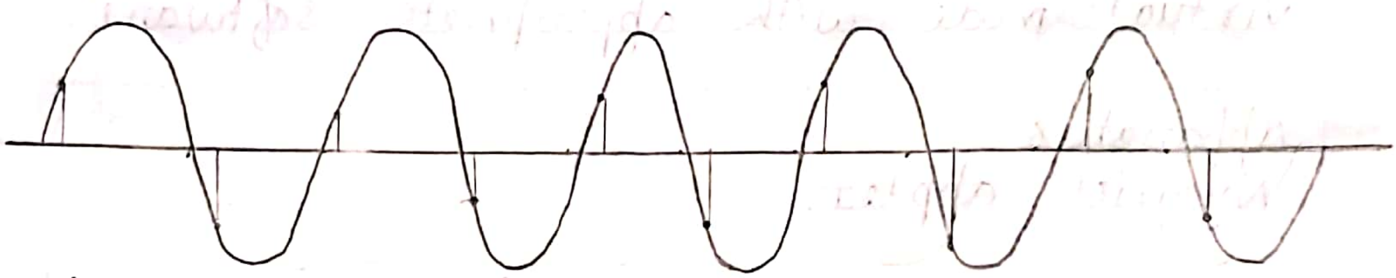


Sampling of a sinusoidal signal of frequency f_s at different sampling rates f_s with dashed lines are shown the alias frequency occuring when $\frac{f_s}{f} < 2$

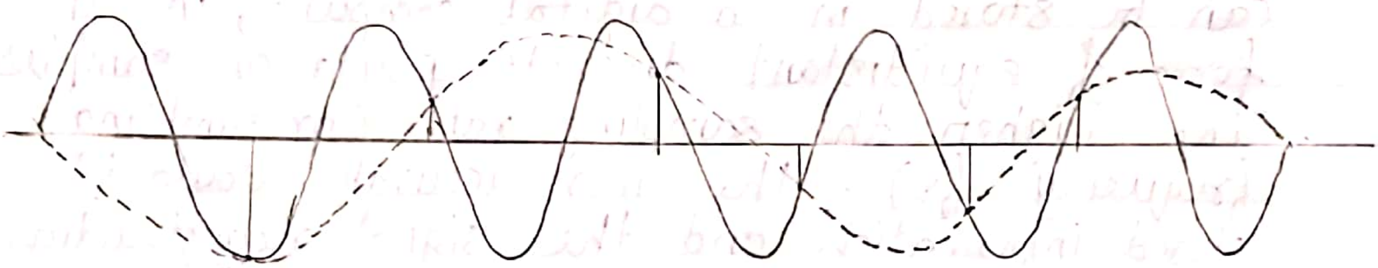
→ $f_s/f = 2.6$



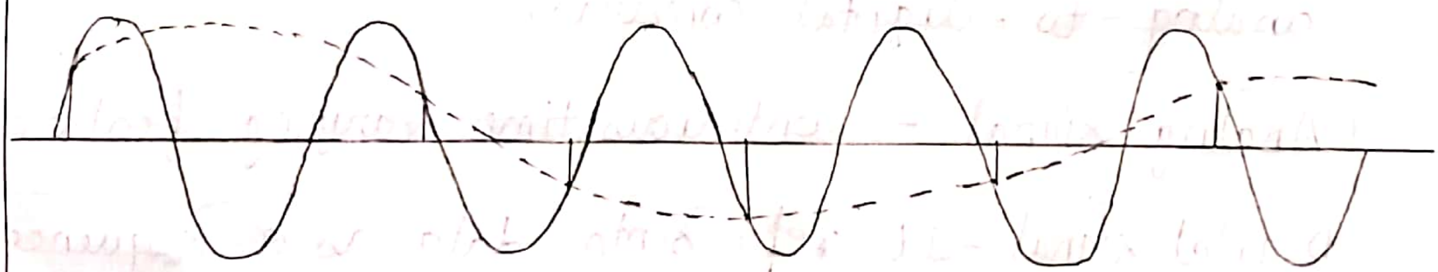
$f_s/f = 2.0$



$f_s/f = 1.4$



$f_s/f = 0.8$



Nyquist Rate -

- Suppose that a signal is band-limited and 'w' is highest frequency. Therefor the effective reproduction of the original signal the sampling rate should be twice the highest frequency.

$$\boxed{f_s = 2w}$$

$f_s \rightarrow$ sampling rate
$w \rightarrow$ highest frequency.

This rate of sampling is called Nyquist rate, and theorem called sampling theorem

(4) Quantization: This method of sampling chooses few points on the analog signal and then these points are joined by round off the value of near stablized value is called quantization.

(5) Encoding

→ The digitization of analog signal is done by encoding.

→ After each sample is quantized, the number of bits per sample is decided.

→ Each sample is changed to an n bit code

→ Encoding is also used to minimize the bandwidth

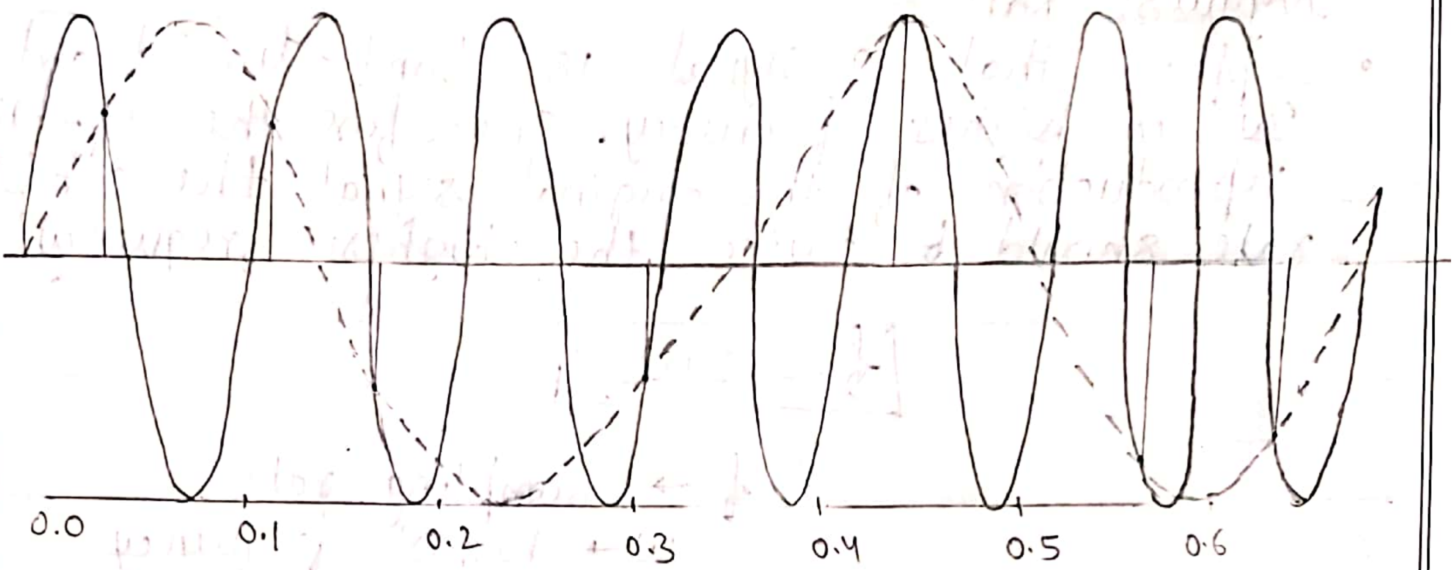
Observation

Signal Frequency = 10 Hz	
SAMPLING FREQUENCY	ALIAS FREQUENCY
7	3
10	0
15	5
20	–
22	–

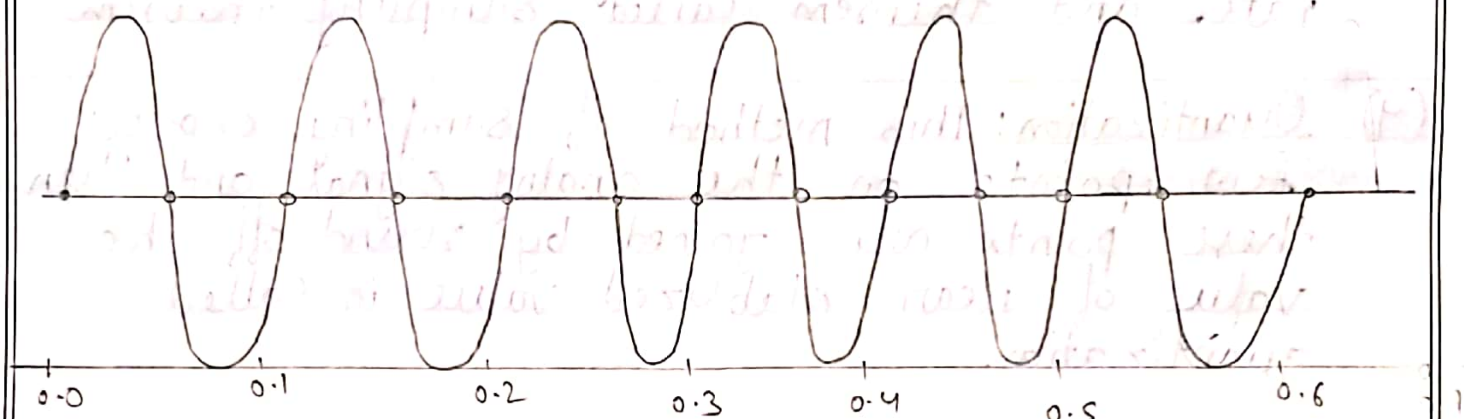
Signal Frequency = 20 Hz	
Sampling freq	Alias Freq
19	0
20.1	0.1
30	10
40	–
42	

SIGNAL FREQUENCY: 10Hz

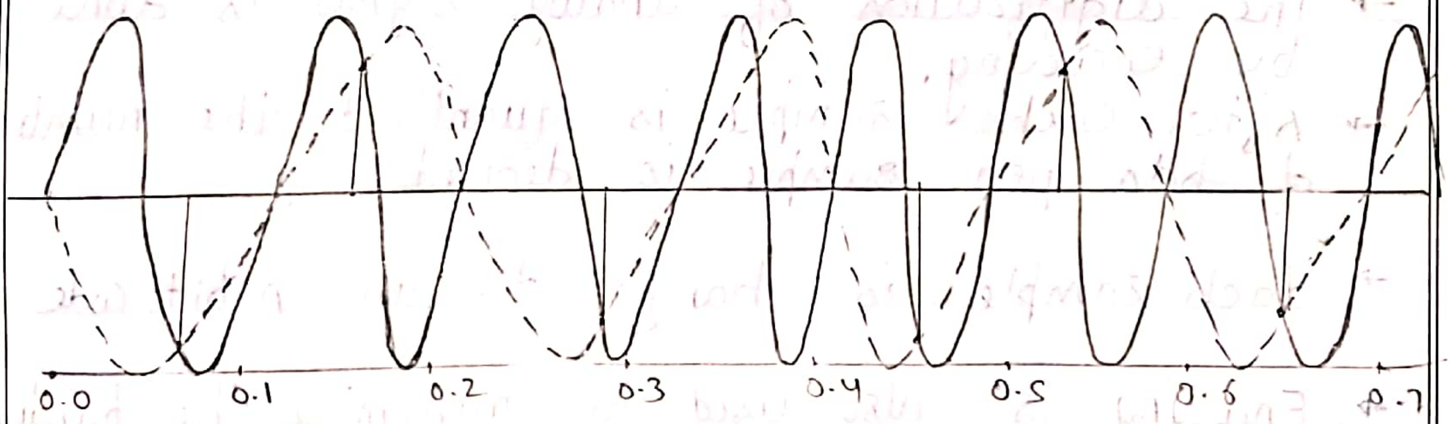
1) Sampling frequency = 7 Hz



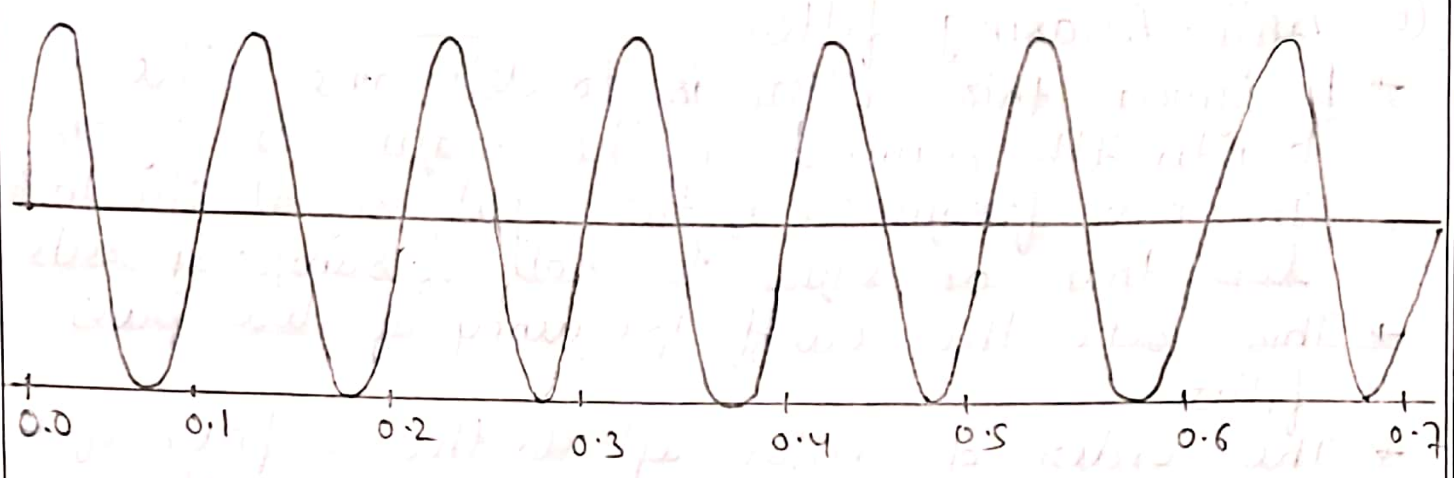
2) Sampling frequency = 10 Hz



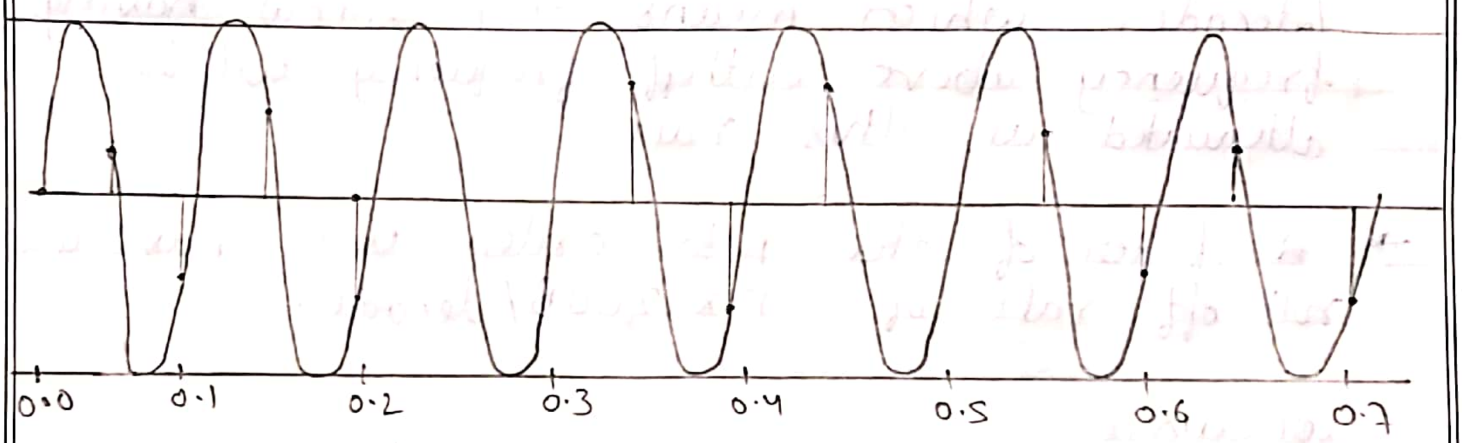
3) Sampling frequency = 15 Hz



4) Sampling Frequency = 20Hz



5) Sampling frequency = 22 Hz



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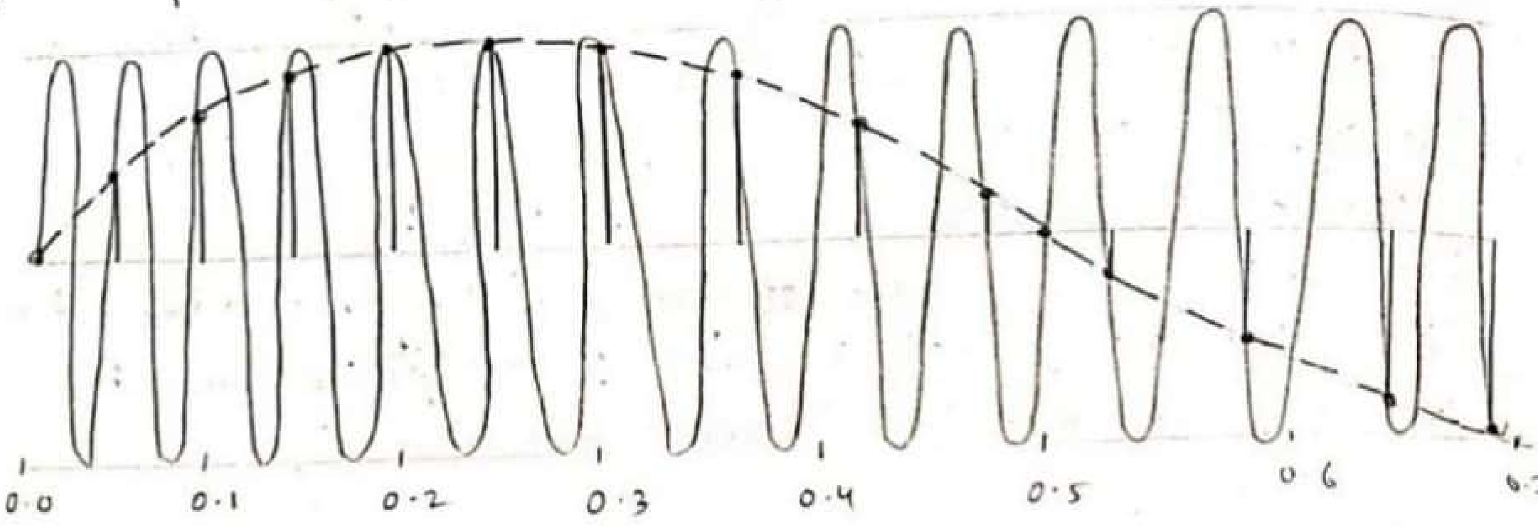
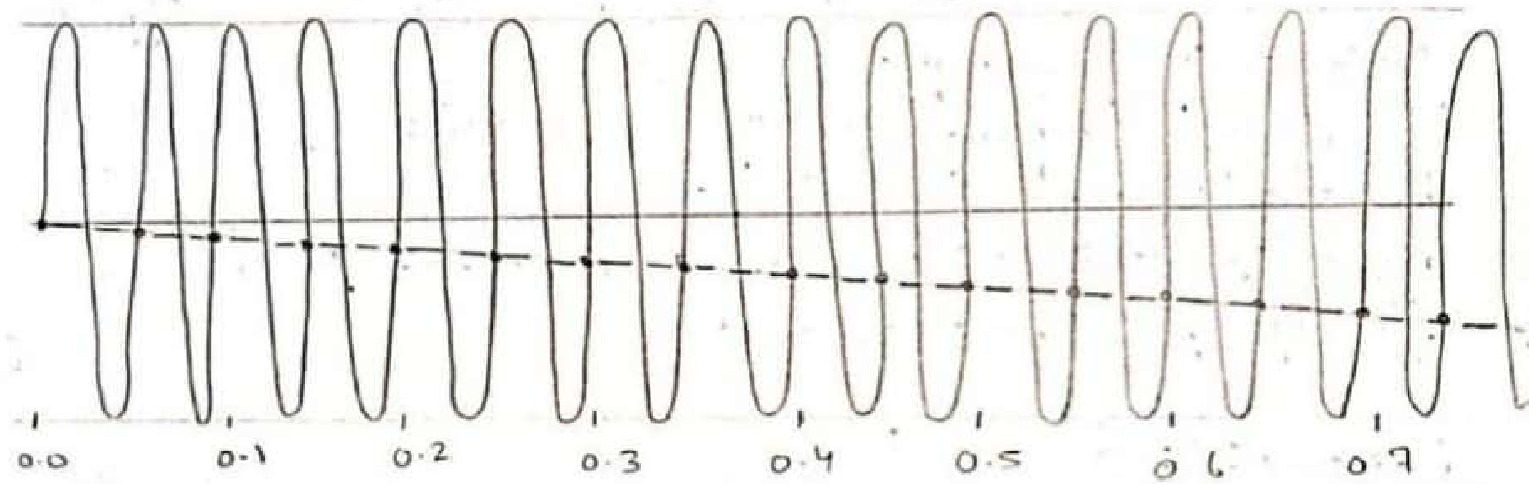
Signal Frequency = 20Hz

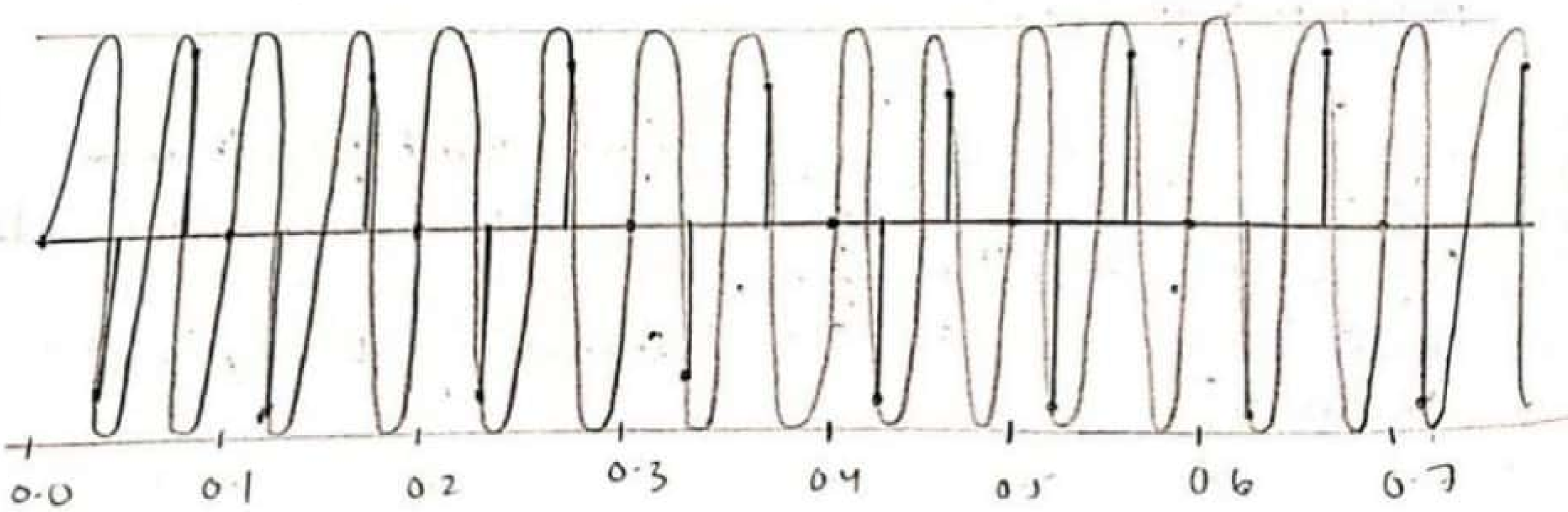
1) Sampling Frequency = 19 Hz



2) Sampling Frequency = 20.1 Hz



3) Sampling Frequency = 30Hz



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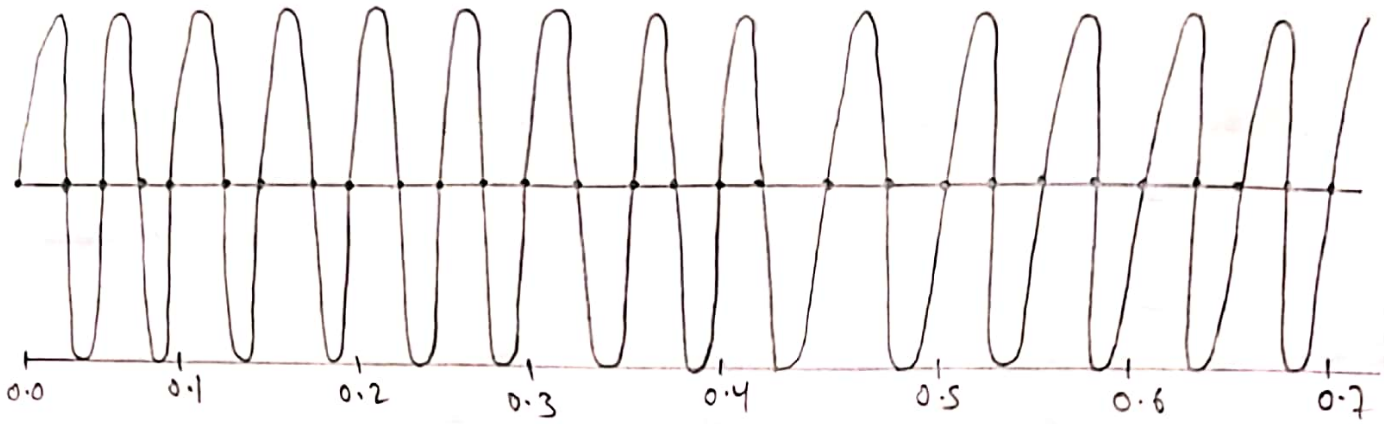
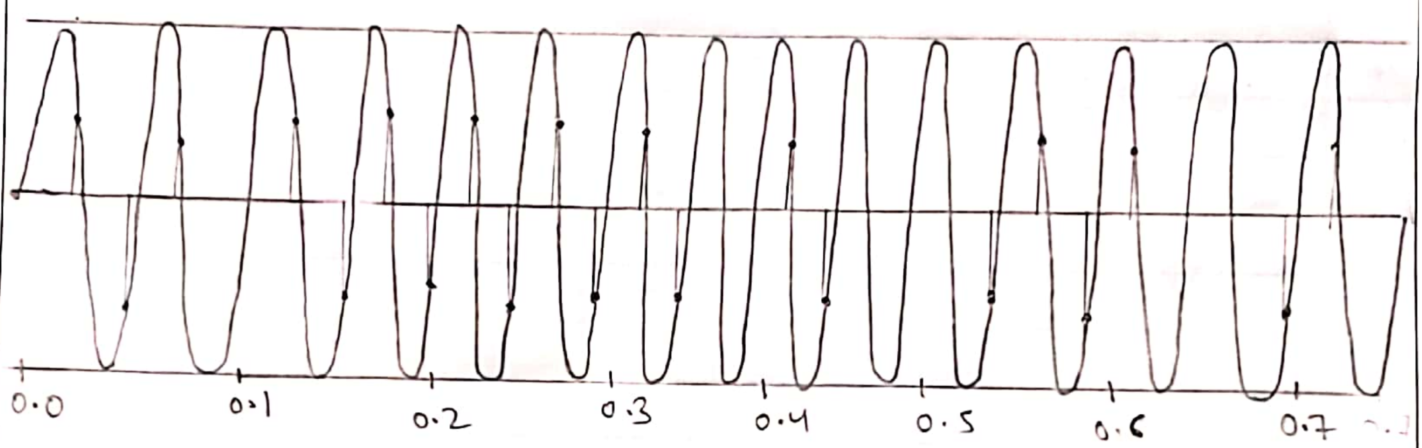
4) Sampling frequency = 40Hz



5) Sampling frequency = 42 Hz



(6) Anti-Aliasing filter

→ Designing this filter is to determine the bandwidth required in the acquisitian system The max frequency of the input signal should be less than or equal to half of sampling rate
→ This sets the cutoff frequency of low pass filter
→ The order of filter affects the stepness of the transmitter region roll-off and hence the width of the transmittor region.

→ A first order filter has a roll-off of 20dB 1 decade, which means any signal having frequency above cuttoff frequency will be attenuated at this rate

→ A filter of the nth order will have a roll off rate of n x 20dB/decade.

Conclusion
Therefore, sampling and reconstruction of the signal has been performed successfully on Nyquist Applet and Nyquist criteria has been verified