[U19CS012]

EXPERIMENT 2:

SAMPLING AND RECONSTRUCTION OF SIGNAL NYQUIST CRITERIA

> **AIM:** To perform sampling and reconstruction of signal and obtain its waveforms. Also verify the nyquist criteria.

> **APPARATUS:** Nyquist Applet (Software)

> **THEORY:**

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

> Analog Signal: It is continous time varying feature of the signal.

> Digital Signal: It represents data as sequence of discrete values at any given time, it can only take any one of the finite number of values.

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> The technique that can be used for Analog to Digital conversion is Pulse Code Modulation

> It has three processes
- Sampling
- Quantization
- Encoding

(2) Sampling

→ Sampling is the process of measuring the instantanous values of continous - time signal in discrete form. Sample is a piece of data taken from the whole date which is continous in the time domain.

→ When a source generate an analog signal and if that has to be digitized, having 1's and 0's, the signal has to be discretized in time. This discretization of analog signal is called Sampling.

→ It is obvious that a much higher sampling rate is needed for sampling a signal which is rich in high frequency components such as sound of music compared to the sampling frequency needed for sampling a slowly varying signal, such as the output of a gas - chromatograph detector or the potential of a glass - electrode during acid - base titration.

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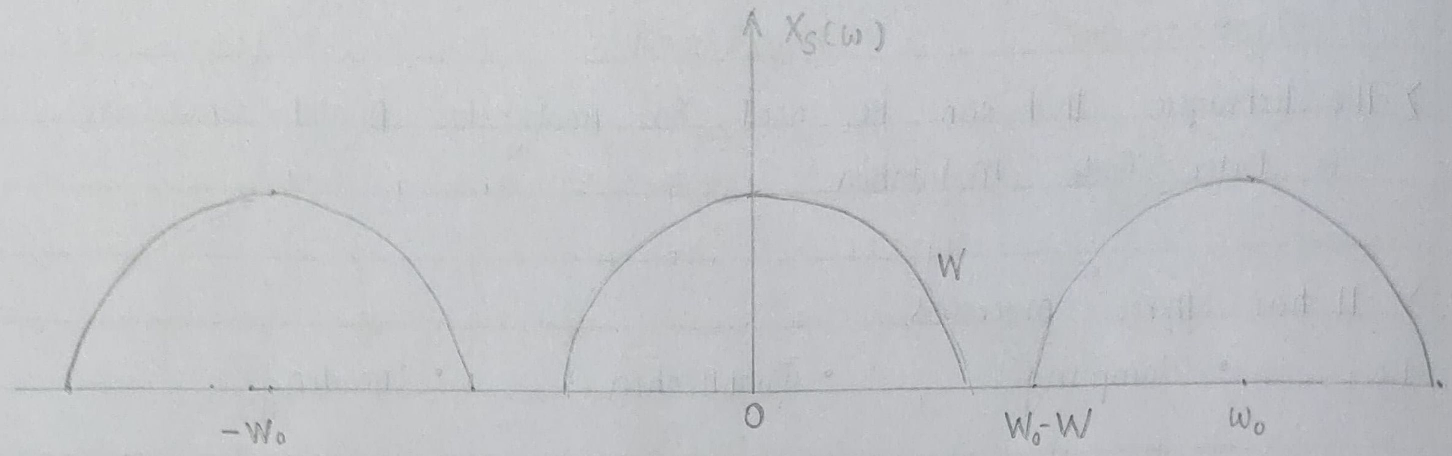
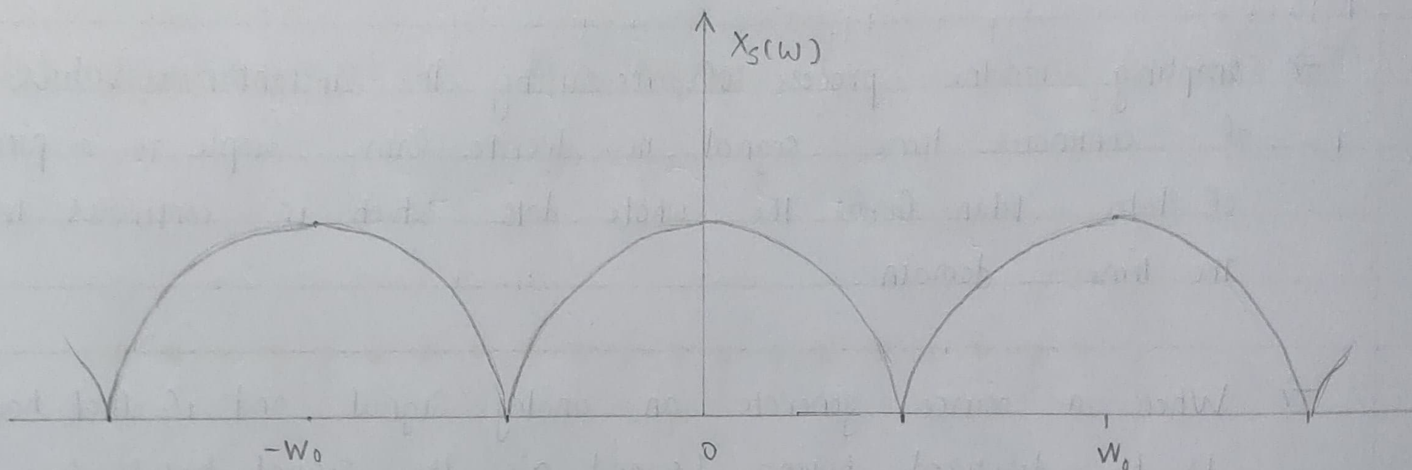
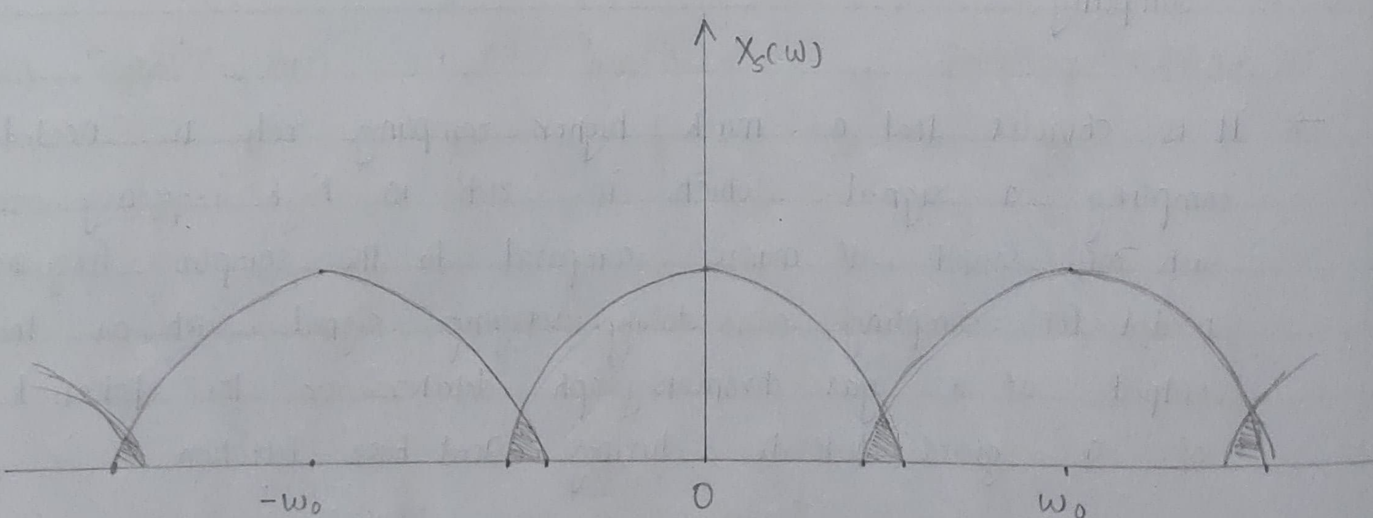


(a) Oversampling ($f_s > 2W$)



(b) Nysquist ($f_s = 2W$)



(c) Undersampling ($f_s < 2W$)

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

(3) Nyquist Rate

→ suppose that a signal is band-limited and W is the highest frequency.

→ Therefore, for effective reproduction of the original signal the sampling rate should be twice the highest frequency

$$\boxed{\therefore \quad F_S = 2W}$$

F_S: Sampling Rate $\quad$ W = Highest frequency

This is Nyquist Rate and theorem is called Sampling theorem.

(A) Condition 1: OVERSAMPLING ($F_S > 2W$)

If sampled at higher rate the 2W in the frequency domain

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

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

(B) Condition 2 :

→ If the Sampling rate is equal to twice the frequency

$F_S = 2W$

→ The information is retrived without any loss. Hence, this is also a good sampling rate.

(C) Condition 3 : UNDER SAMPLING

$F_S < 2W$

→ The below pattern shows overlapping of information which leads to mixing up and loss of information. This unwanted phenomena of over-lapping is called Aliasing.

(a) Aliasing : A high frequency component is taking on the identity of a low-frequency component in the spectrum of sampled version.

The effect of aliasing is reduced by :

1) The signal needs to be sampled at a rate slightly higher than the Nyquist rate.

2) In the Transmitter section of PCM, a low pass anti-Aliasing filter is employed to eliminate the unwanted high frequency components.

(4) Quantization : The method of sampling chooses few points on the analog signal and then these points are joined to round off the value of a near stabilized value is called Quantization.

(5) Encoding :

> The digitization of analog signal is done by encoder.

> 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.

(6) Anti- Aliasing filter

> Designing this filter is to determine the bandwidth required in the acquisition system. The maximum frequency of the input signal should be less than or equal to half of sampling rate.

> This sets the cutoff frequency of the low- pass filter

> The order of a filter affects the steepness of the transition region roll-off and hence the width of the transition region.

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

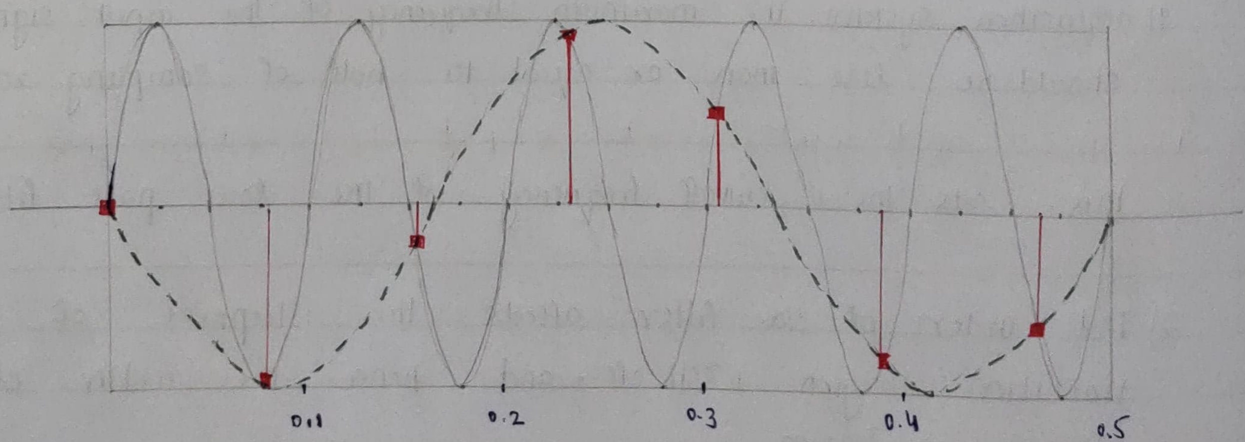
A filter of the n^{th} order will be have a roll-off rate of $n \times 20$ dB/ decade.

Conclusion: Therefore, sampling and reconstruction of the signal has been performed successfully on Nyquist Applet (software) and Nyquist criteria has been verified.

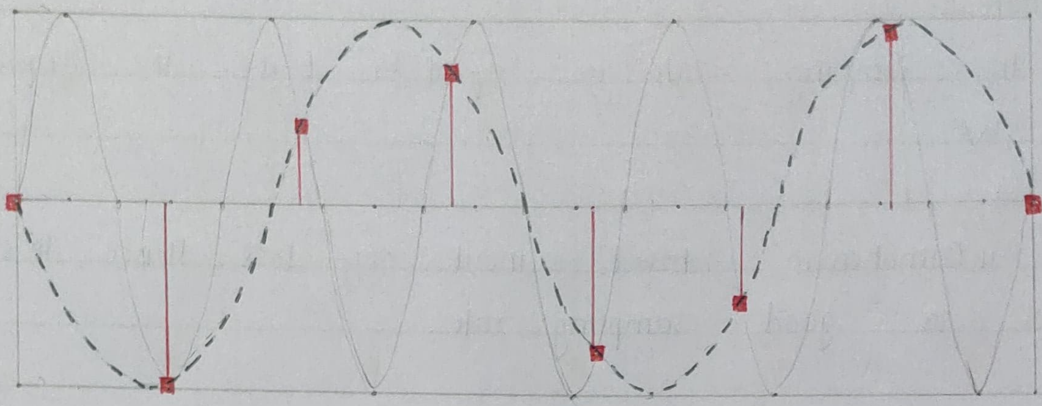
Observation Table

Signal Frequency (Hz)	Sampling Frequency (Hz)	Alias Frequency (Hz)
10	13	3
	14	4
	15	5
	20	-
	25	-
20	19	1
	22	2
	30	10
	40	-
	50	-

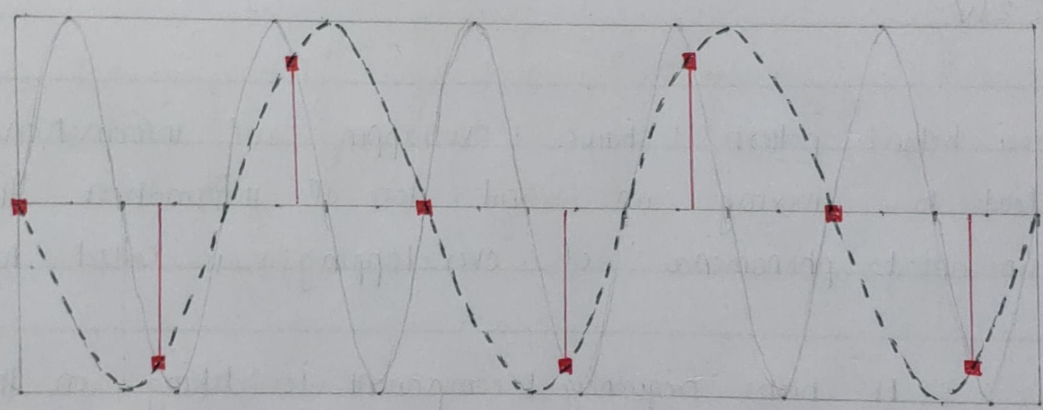
(1) Signal freq. (Hz) = 10.0 Alias freq. (Hz) 3.0 Sampling freq. (Hz) = 13.0



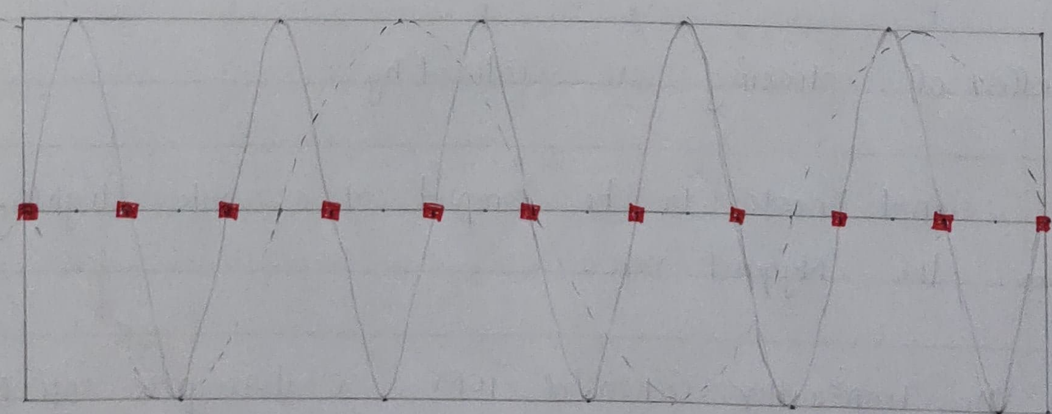
(2) Signal freq. (Hz) = 10.0 Alias freq (Hz) = 4.0 Sampling freq (Hz) = 14.0



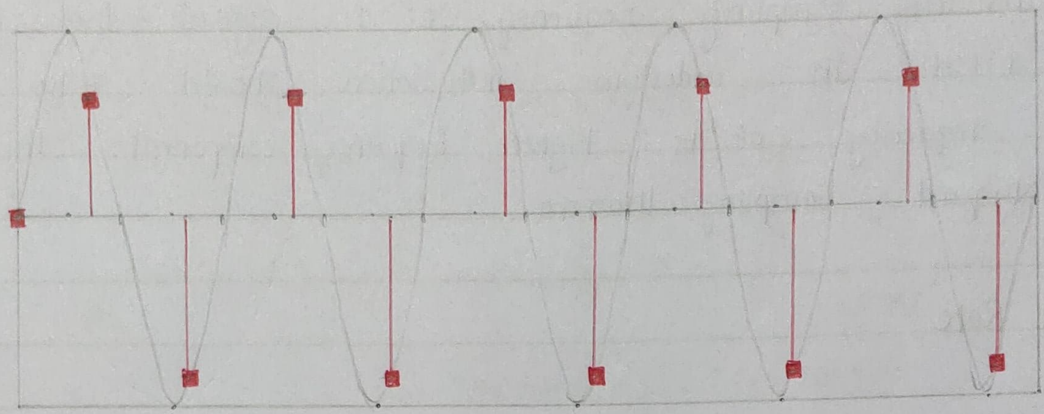
(3) Signal freq (Hz) = 10.0 Alias freq (Hz) = 5.0 Sampling freq (Hz) = 15.0



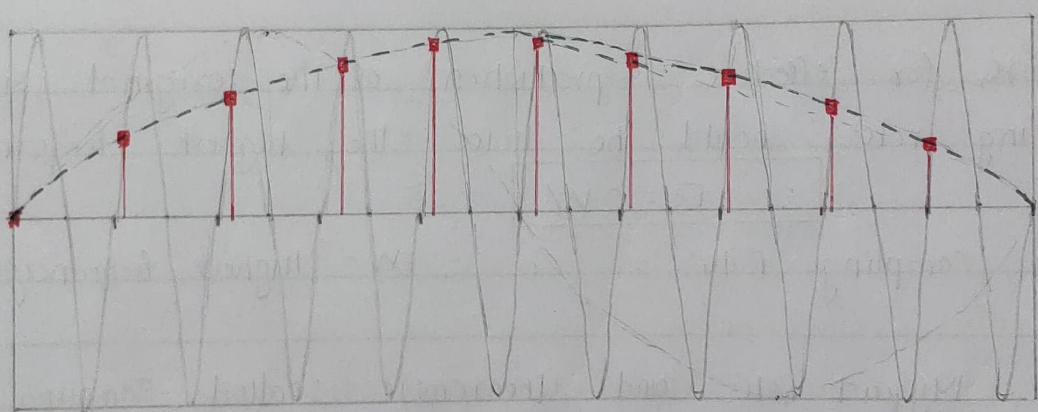
(4) Signal freq (Hz) = 10.0 Alias freq (Hz) = - Sampling freq (Hz) = 20



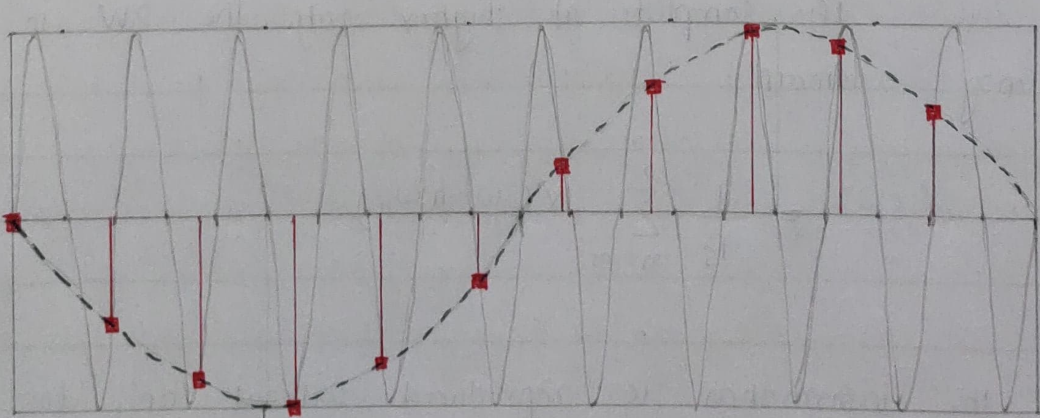
(5) Signal freq (Hz) = 10.0 Alias freq (Hz) = - Sampling freq (Hz) = 25



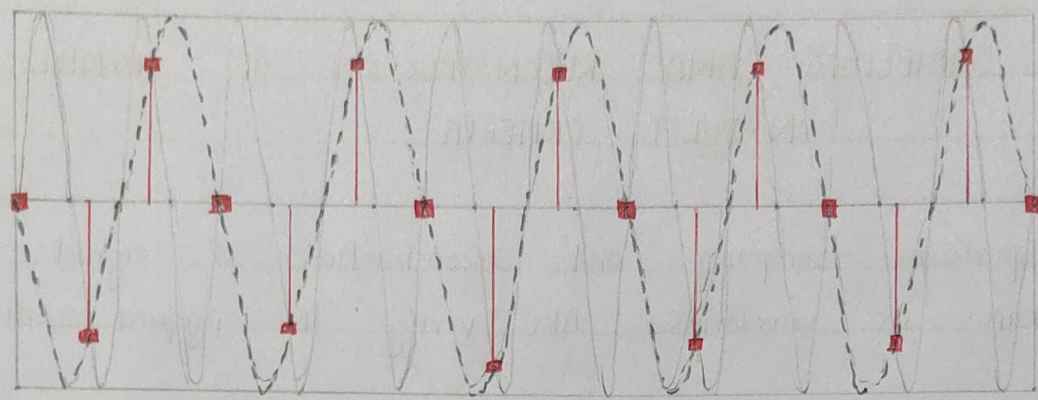
(6) Signal freq. (Hz) = 20.0 Alias freq. (Hz) = 1.0 Sampling freq. (Hz) = 19



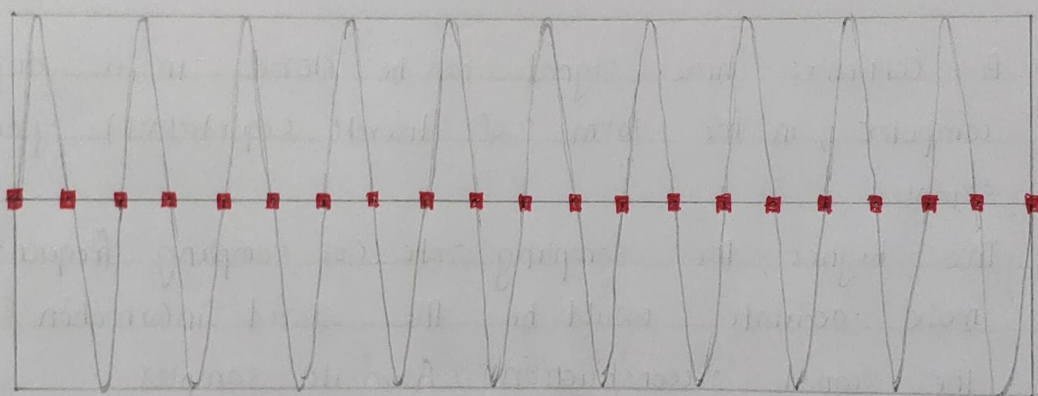
(7) Signal freq. (Hz) = 20.0 Alias freq. (Hz) = 2.0 Sampling freq. (Hz) = 22.0



(8) Signal freq. (Hz) = 20.0 Alias freq. (Hz) = 10 Sampling freq (Hz) = 30



(9) Signal freq. (Hz) = 20.0 Alias freq (Hz) = - Sampling freq (Hz) = 40



(10) Signal freq. (Hz) = 20.0 Alias freq (Hz) = - Sampling freq (Hz) = 50