

# Experiment No : 01

Expt. No.

01

## Spectrum Analyser And Observe Spectrum

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AIM :

To study Spectrum Analyzer and observe the spectrum of sinusoidal signal and square wave.

APPARATUS: Spectrum Analyzer (9 khz - 30 hz) Function generator.

THEORY: A spectrum analyser is a laboratory instrument that displays signal amplitude (strength) as it varies by signal frequency. The frequency appears on the horizontal axis and the amplitude is displayed on the vertical axis. To the casual observer, a spectrum analyzer looks like an oscilloscope and in fact, some lab instruments can function either as oscilloscopes or spectrum analyzers.

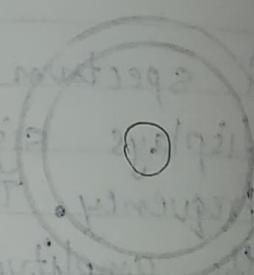
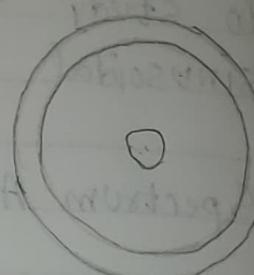
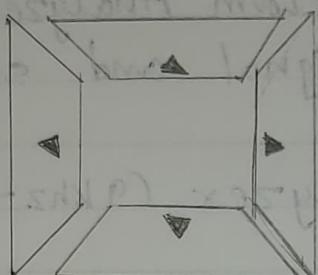
- A spectrum analyzer can be used to determine whether or not a wireless transmitter is working according to federally defined standards for purity of emissions. Output signals at frequencies other than the intended communications frequency appear as vertical lines (pips) on the display. A spectrum analyzer can also be used to determine, by direct observation, the bandwidth of a digital or analog signal.
- A spectrum analyzer interface is a device that can be connected to a wireless receiver or a personal computer to allow visual detection and analysis of electromagnetic signals over a defined band of frequencies.

Function Keys					
Freq.	Span	Ampl	Marker	Mkr	
BW	Sweep	Trace	Lines		
Means	Mode	Setup	Save Recall	Reset	

7	8	9	6Hz
4	5	6	MHz
1	2	3	KHz
0	.	+/	Hz
ESC	X		

ON/OFF

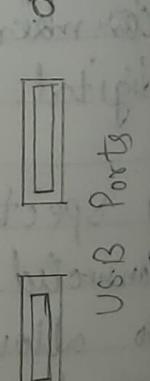
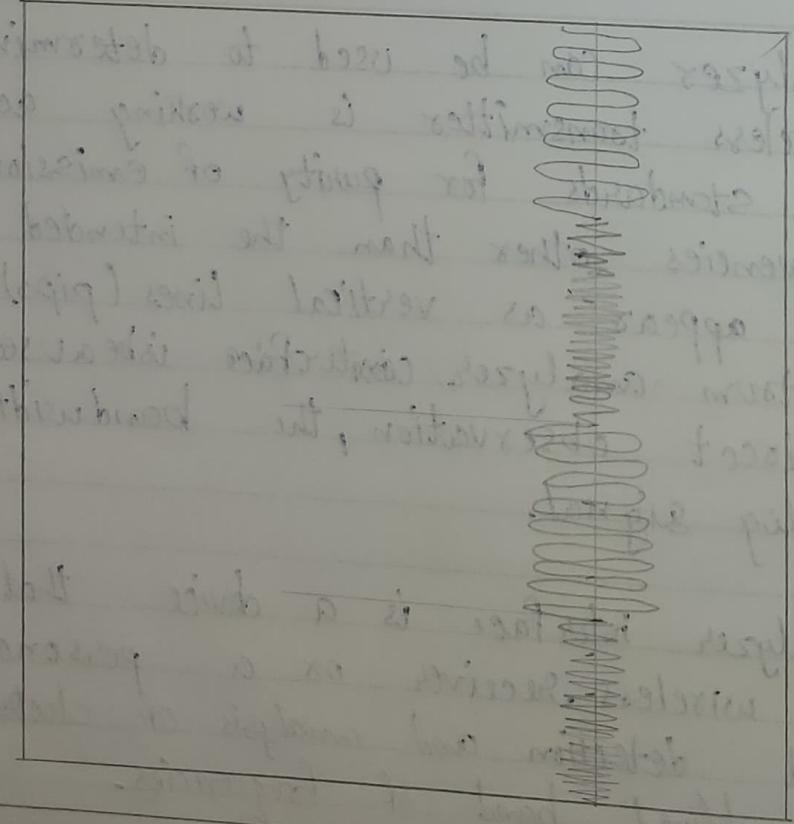
Control



RF Input

F1	F2	F3	F4	F5	F6
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LCD Display



Spectrum Analyser

6830

AIM :

## \* Features of LAB Instrument GSP-830 (OWINSTEK)

- 5 markers with delta marker and peak functions.
- 3 traces
- Split windows with separate settings.
- 6.4" TFT color LCD, 640 x 480 resolution
- AC / DC / battery multi-mode power operation
- Autoset
- 9 kHz - 3 GHz Frequency range

## \* Frequency selection and their selection methods

### 1) Frequency:

- Frequency / Span : The frequency key, together with span key sets the frequency scale.
- View Signal (Center & Span) : Center and span method defines the center frequency & the left/right bandwidth (span) to locate the signal.

AIM :

- Setting frequency adjustment step: Frequency adjustment step defines the arrow keys resolution for center, start and stop frequency.

### Panel Operation :

- Press Frequency Key
- press F4 (step)
- Enter the value using numerical and unit keys, arrow keys and scroll nops.

### 2) Range

- 9 KHz to 30Hz.

### 3) Set Center frequency :

#### Panel Operation :

- Press Frequency Key
- Press F1 (center)
- Enter the value using numerical and unit keys, arrow keys and scroll nops.

### 4) Set Frequency span

#### Panel Operation :

AIM :

- Press span key
- press F1 (span)
- Enter the value using numerical and unit keys, arrow keys and scroll nops.

### 5) View Signal (Start & Stop):

- Start and stop method defines the beginning and the end of the frequency range.
- Arrow keys and scroll knob resolution: 1/10 of span

### 6) Set start frequency:

Panel Operation:

- Press frequency key
- Press F2 (start)
- Enter the value using numerical and unit keys, arrow keys and scroll nops.

### 7) Set stop frequency:

Panel Operation:

- Press frequency key
- press F3 (stop)
- Enter the value using numerical and unit keys and scroll nops.

AIM :

## \* Amplitude Selection and settings methods

### 1) Amplitude:

- Amplitude key sets vertical attribute of the display, including the upper limit (reference level), vertical range (amplitude scale), vertical unit and compensation for external gain or loss (external offset).

### 2) Set Vertical Scale:

- Vertical display scale is defined by reference amplitude, amplitude range, measurement unit and external gain/loss.

### 3) Set reference amplitude

- The reference level defines the amplitude at the top of the displayed range.

### Panel Operation:

- Press amplitude key
- press F1 (reference level)
- Enter the value using numerical and unit keys, arrow keys and scroll knob. Arrow keys and scroll knob, scroll knob resolution; vertical scale.

AIM :

Range :

- dBm -110 to +20 dBm, 0.1 dB resolution
- dBmV -63.1 to 66.99 dBmV, 0.01 dB resolution
- dBuV -3.01 to 126.99 dBuV, 0.01 dB resolution

## 4) Select amplitude scale

Panel Operation:

- Press amplitude key
- Press F2 (Scale dB/Div)
- Repeatedly to select the scale

Range : 10, 5, 2, 1 dB/Div.

Panel Operation:

- Press amplitude key
- Press F3 (Units)
- Select and press the unit from F1(dBm), F2(dBmV) and F3 (dBuV)
- Press F6 (return) to go back to previous menu.
- dBm -110 to +20 dBm, 0.1 dB resolution
- dBmV -63.1 to 66.99 dBmV, 0.01 dB resolution
- dBuV -3.01 to 126.99 dBuV, 0.01 dB resolution
- Set external offset level.

AIM :

## 5) Background:

- External offset compensates the amplitude gain or loss caused by an external network or device.

## Panel Operation:

- Press amplitude key
- Press  $\times$  F4 (external gain)
- Enter the value using numerical and unit keys, arrow keys and scroll knob.

## Range :

- -20 dB to +20 dB, 0.1 dB resolution.

## ICON:

- The amplitude icon appears at the bottom of the display when the external offset changes.
- To check whether Spectrum analyzer working properly.
- Generate Auxiliary signal: Press system key, press auxiliary signal, select an option from side given menu, following signal will generate. It generates 10 MHz signal with 10dB amplitude.

## Observation Table.

Waveform : SINE

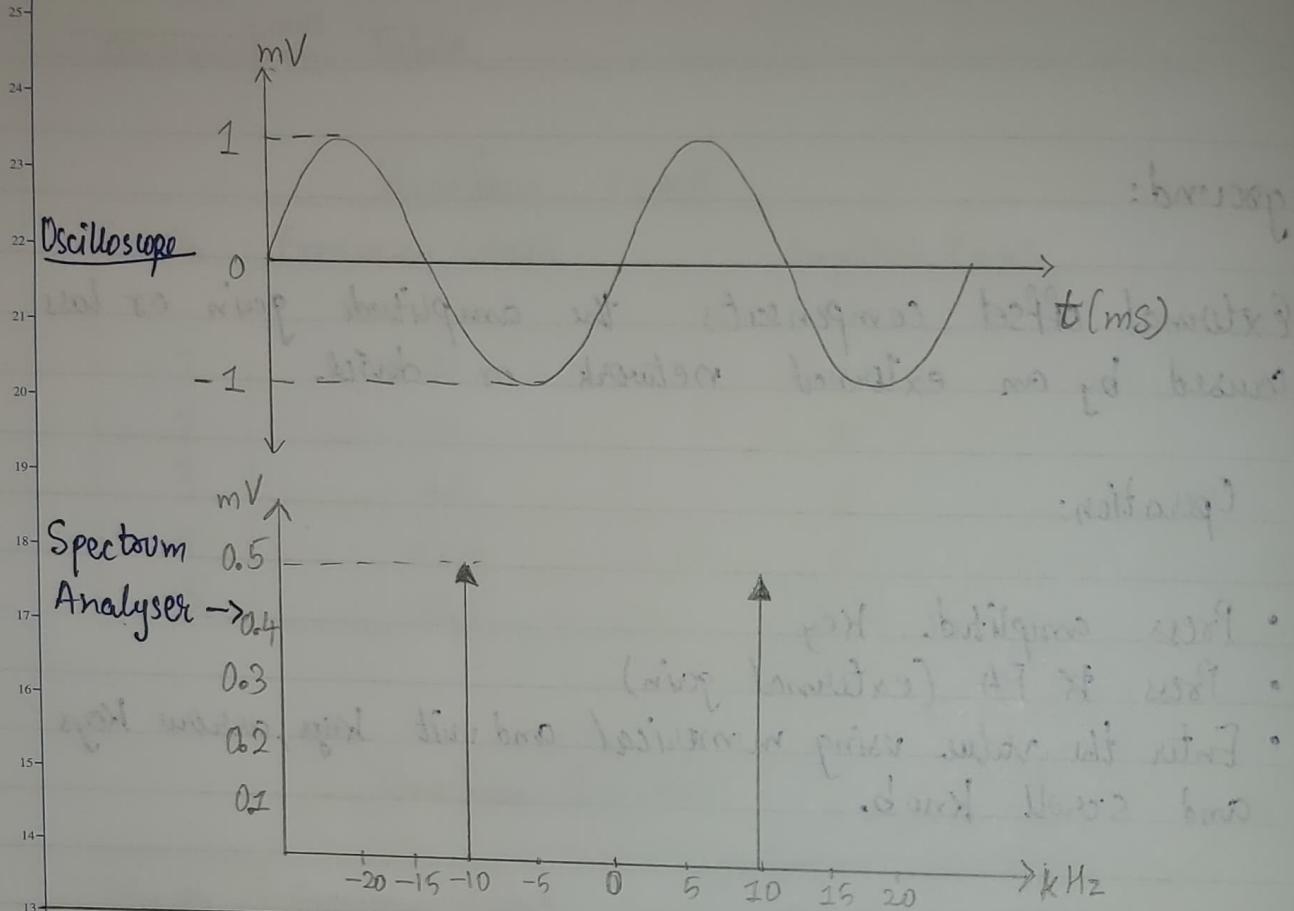
Sr. No.	Frequency (KHz)	Amplitude (mV)
1	10	1
2	15	1
3	15	2
4	12.5	2
5	12.5	0.5

Waveform: SQUARE

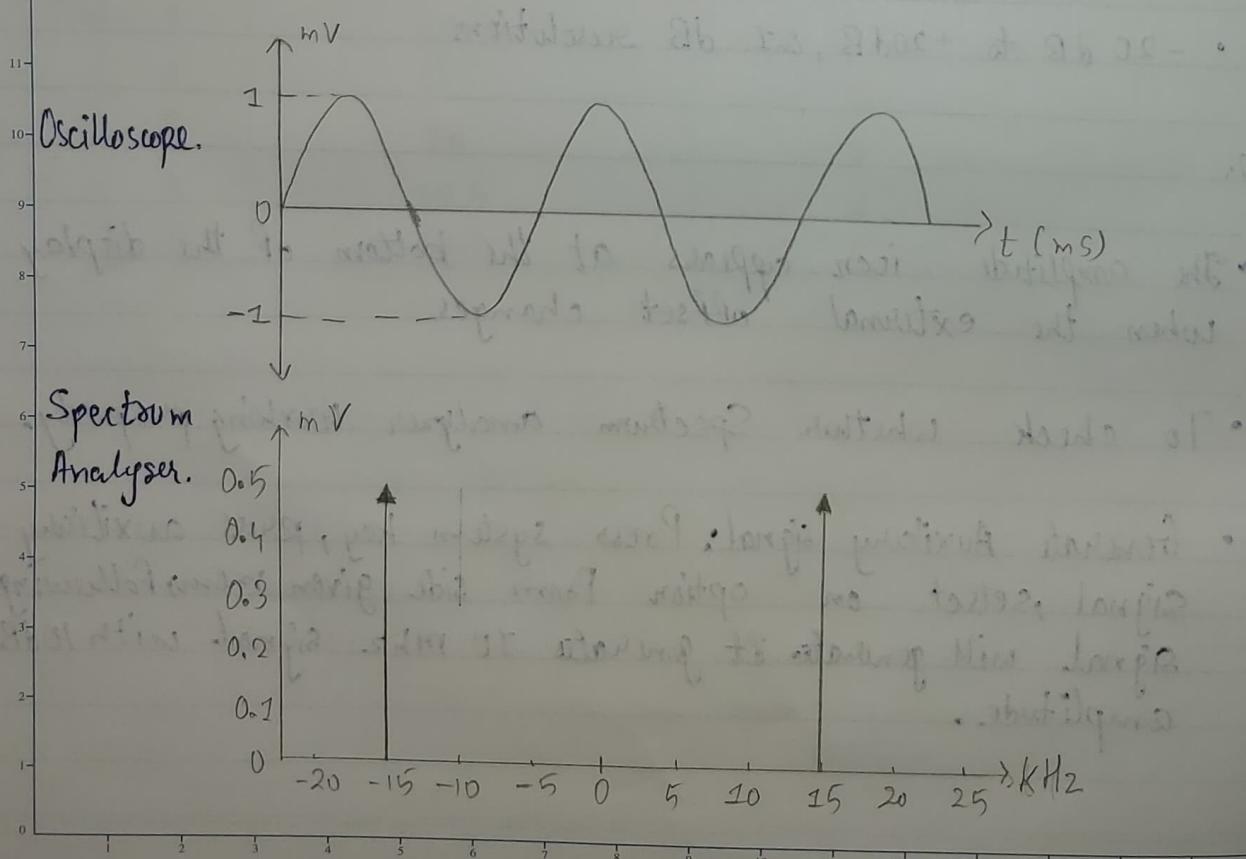
Sr. No.	Frequency (KHz)	Amplitude (mV)
1	10	1
2	5	1
3	10	2
4	12.5	1
5	12.5	2

# SINE Wave.

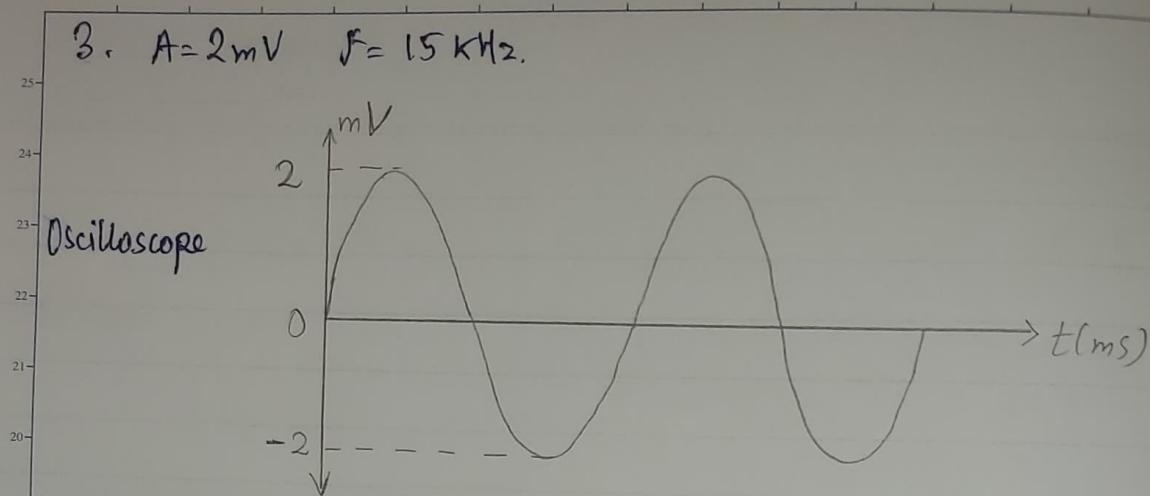
$$1. A = 1 \text{ mV} \quad f = 10 \text{ kHz.}$$



$$2. A = 1 \text{ mV} \quad f = 15 \text{ kHz.}$$

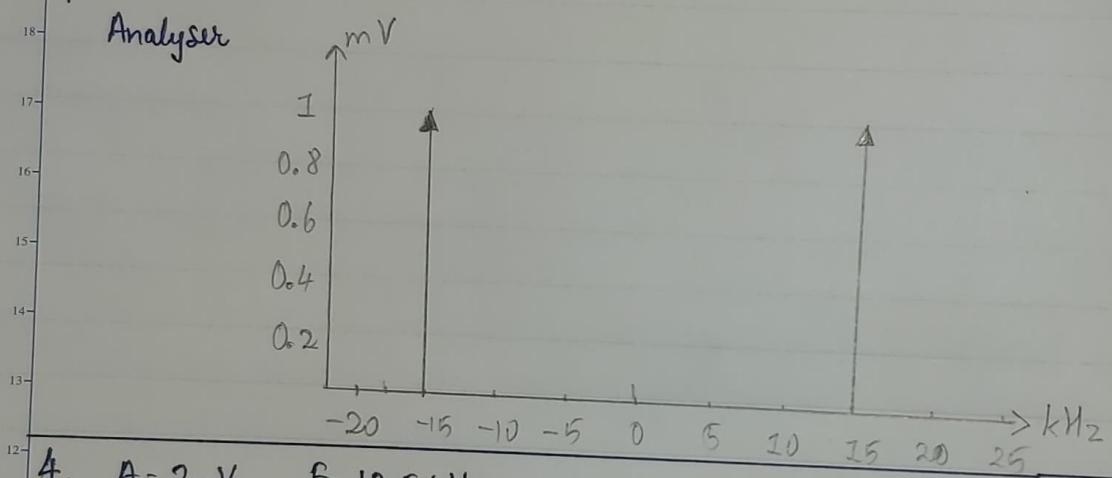


$$3. A = 2 \text{ mV} \quad f = 15 \text{ kHz.}$$



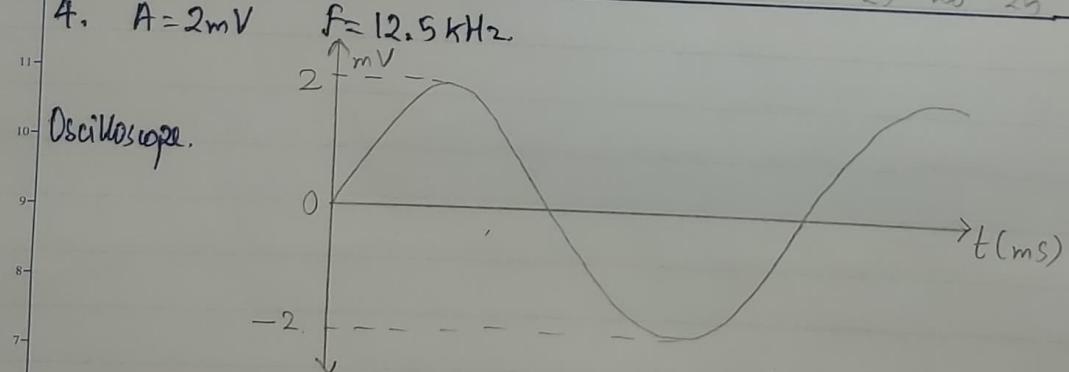
Spectrum

Analyser



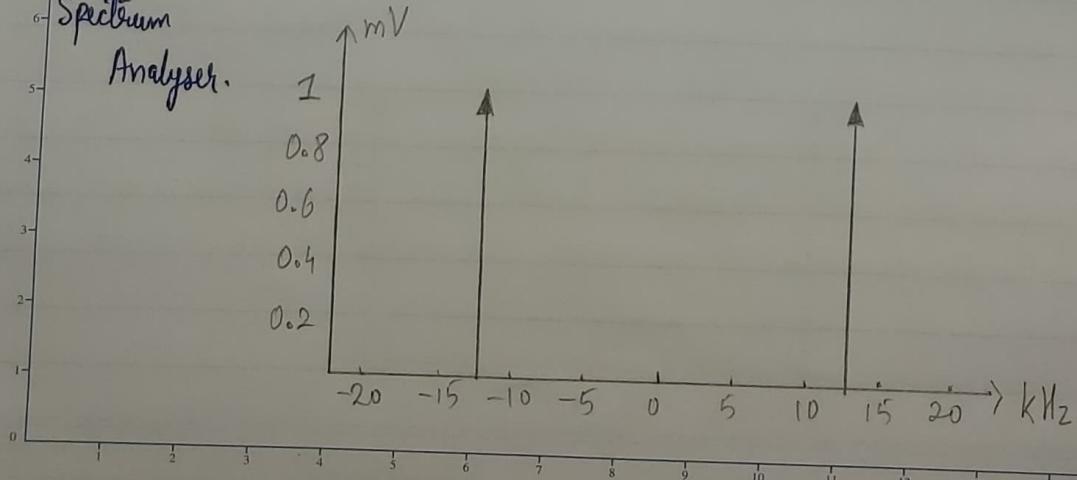
$$4. A = 2 \text{ mV} \quad f = 12.5 \text{ kHz.}$$

Oscilloscope.



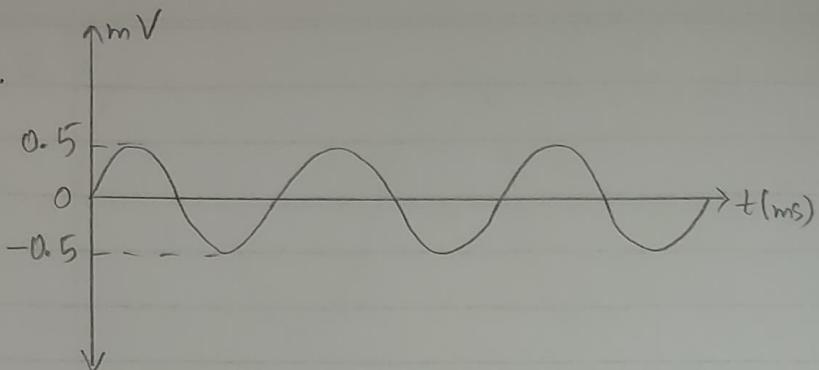
Spectrum

Analyser.



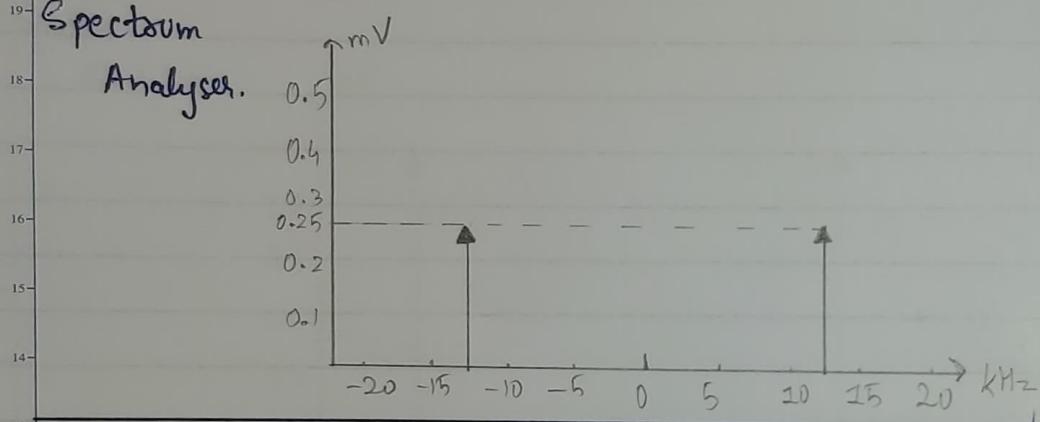
5.  $A = 0.5 \text{ mV}$   $f = 12.5 \text{ kHz}$ .

Oscilloscope.



Spectrum

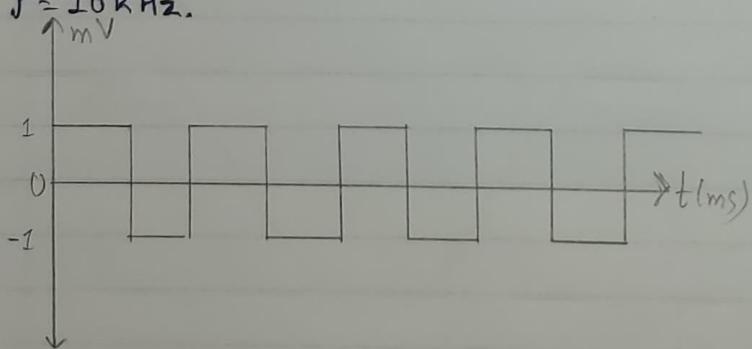
Analyser.



SQUARE Wave.

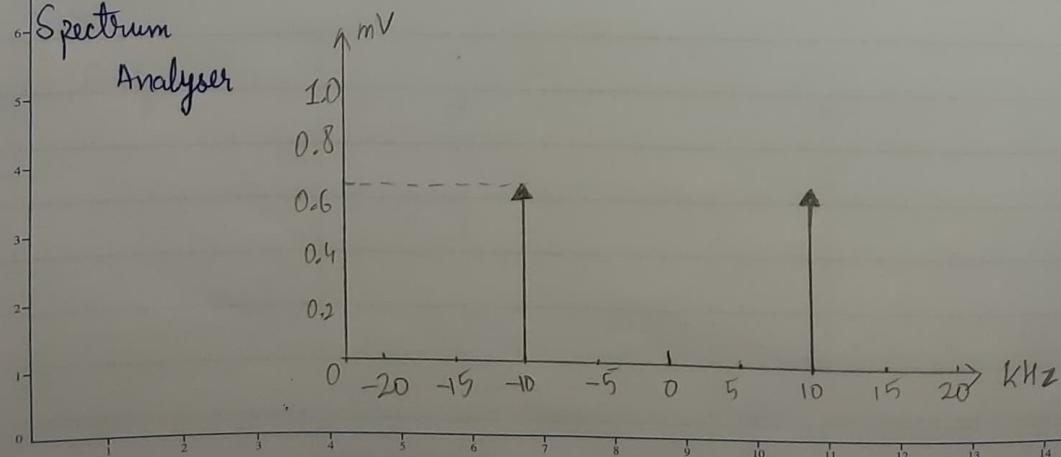
1.  $A = 1 \text{ mV}$   $f = 10 \text{ kHz}$ .

Oscilloscope

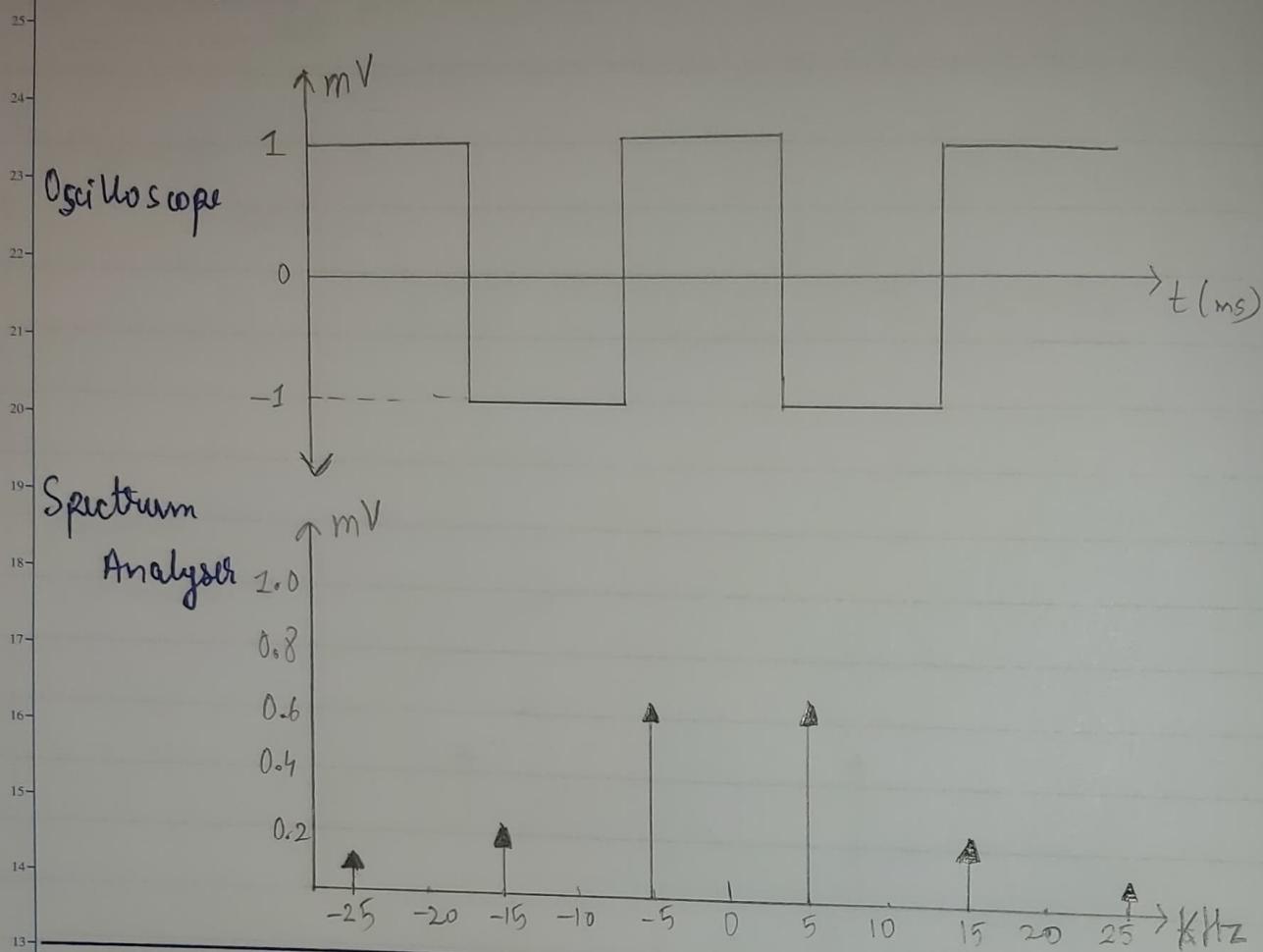


Spectrum

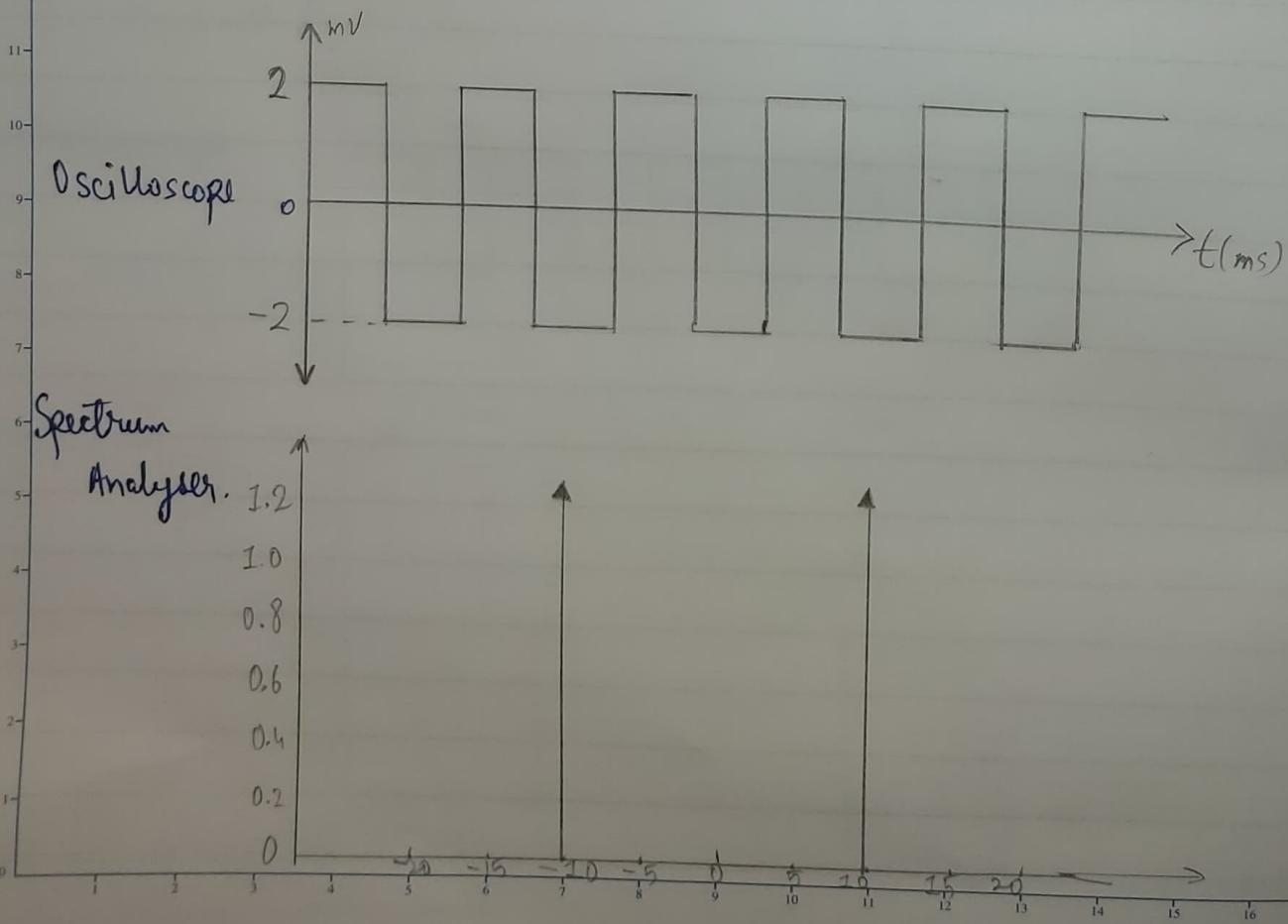
Analyser



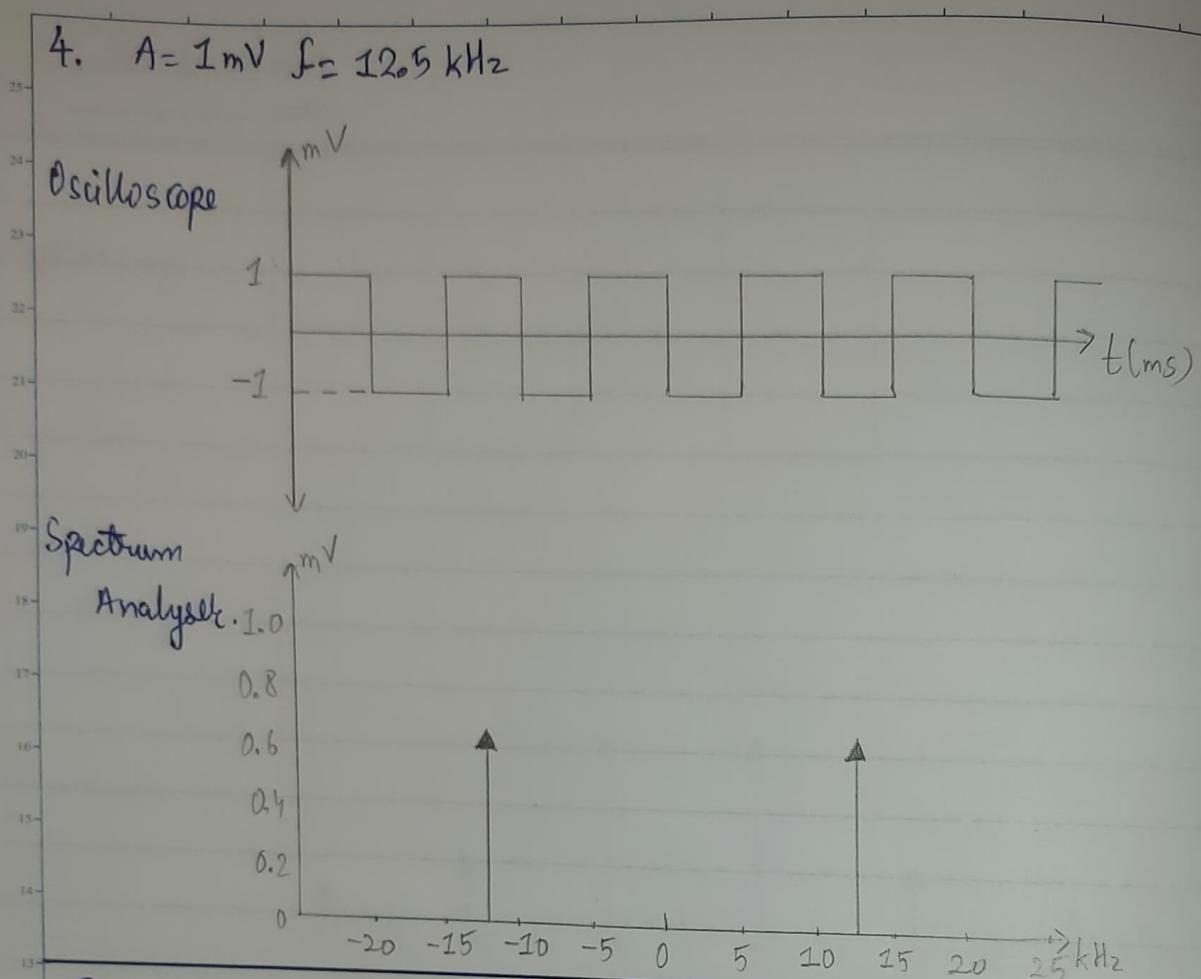
$$2. A = 1 \text{ mV} \quad f = 5 \text{ kHz}$$



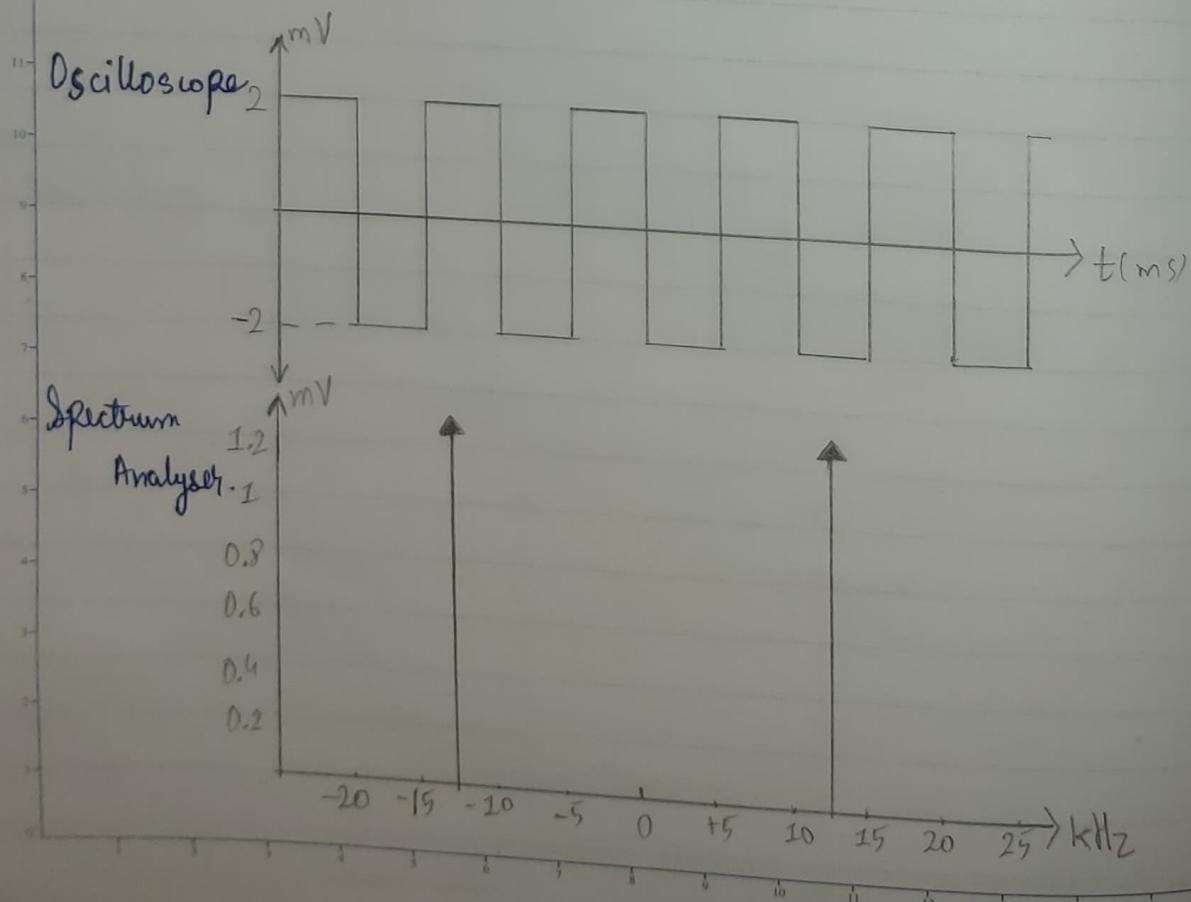
$$3. A = 2 \text{ mV} \quad f = 10 \text{ kHz.}$$



$$4. A = 1 \text{ mV} \quad f = 12.5 \text{ kHz}$$



$$5. A = 2 \text{ mV} \quad f = 12.5 \text{ kHz}$$



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AIM :

**CONCLUSION:** Hence, successfully verified and analysed the spectrum of sinusoidal signal and square wave. For different frequency and amplitude.

**AIM :** To perform Sampling and reconstruction of signal and obtain its waveforms. Also verify the Nyquist Criteria.

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

**THEORY:**

- A continuous - time 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), 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 continuous time varying feature of the signal.
- Digital Signal: It represents data as sequence of discrete values at any given time, it can only take on one of the finite number of values.
- The technique that can be used for Analog to Digital conversion is Pulse Code Modulation.
- It has three processes.
  - Sampling
  - Quantization
  - Encoding

AIM :

## \* Sampling

- Sampling is the process of measuring the instantaneous values of continuous-time signal in discrete form. Sample is a piece of data taken from the whole data which is continuous in the time domain.
- When a source generates an analog signal and if that has to be digitized, having Is & Os, 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.
- 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.

## \* Nyquist Rate:

- Suppose that a signal is band-limited and  $W$  is the highest frequency.

AIM :

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

$$\therefore f_s = 2W$$

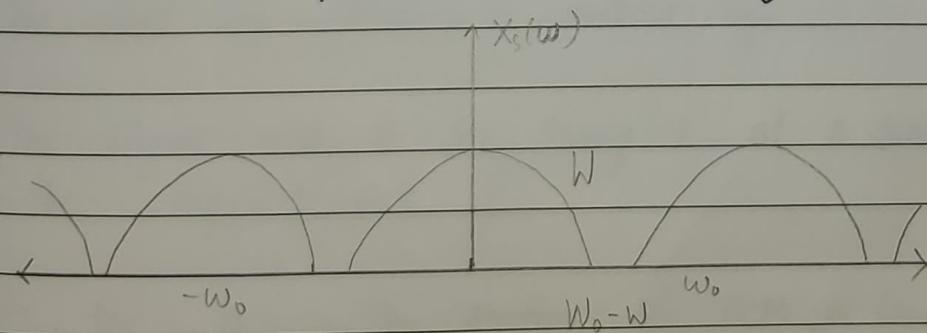
$f_s$ : Sampling rate ;  $W$ : highest freq.

This is Nyquist rate and a theorem is called sampling theorem.

⇒ Condition 1 ⇒ Over Sampling → If sampled at higher rate than  $2W$  in the frequency domain ( $f_s > 2W$ )

$$X_s(w) = \frac{1}{T_s} \sum_{n=-\infty}^{\infty} X(w - nw_0)$$

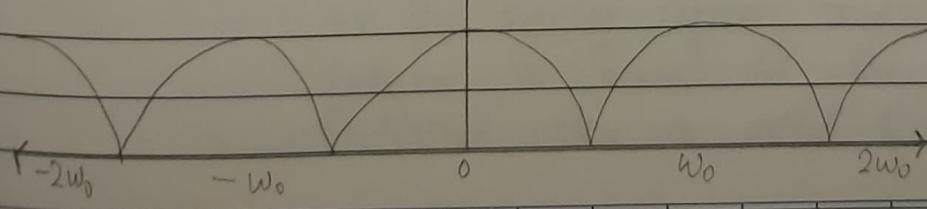
- How the information is reproduced without any loss. There is no mixing up and hence recovery is possible.



⇒ Condition 2

If the sampling rate is equal to twice the frequency.

$$f_s = 2W$$



AIM :

- The information is retrieved without any loss. Hence, this is also a good sampling rate.

$\Rightarrow$  Condition 3. (Undersampling)

$$f_s < 2W$$

- The below pattern shows over-lapping of information which leads to mixing up and loss of information. This unwanted phenomenon of over-lapping is called Aliasing.
- \* 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.

\* Quantization : The method of sampling chooses a few points on the analog signal and then these points are joined to sound off the value of a near stabilized value is called Quantization.

AIM :

## ★ Encoding

- The digitization of analog signal is done by the encoder.
- After each sample is quantized, the number of bits per second sample is decided.
- Each sample is changed to an  $n$  bit code.
- Encoding also used to minimizes the bandwidth.

## ★ Anti-aliasing Filter

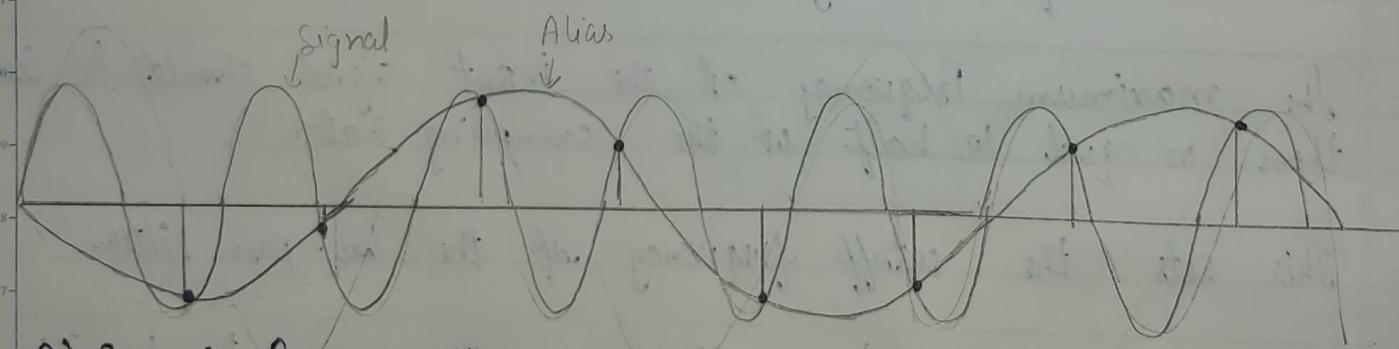
- Designing this filter is to determine the bandwidth required in the acquisition system.
- The maximum frequency of the input signal should be less than or equal to half of the sampling rate.
- This sets the cutoff frequency of the low-pass filter.
- The order of a filter affects the steepness of the transition region roll-off and hence the width of the transition region. 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  $n^{\text{th}}$  order will be have a roll-off rate of  $n \times 20 \text{ dB/decade}$ .

### Observation Table.

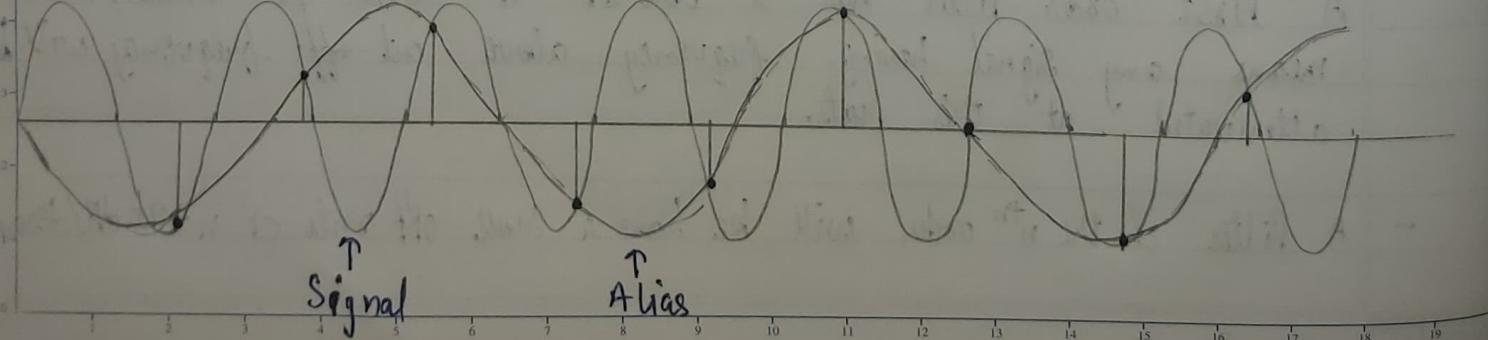
Signal Frequency (Hz)	Sampling Frequency (Hz)	Alias Frequency (Hz)
10	13	3
	14	4
	15	5
	20	-
	25	-
20	19	1
	22	2
	30	10
	40	-
	50	-

Signal frequency = 10 Hz.

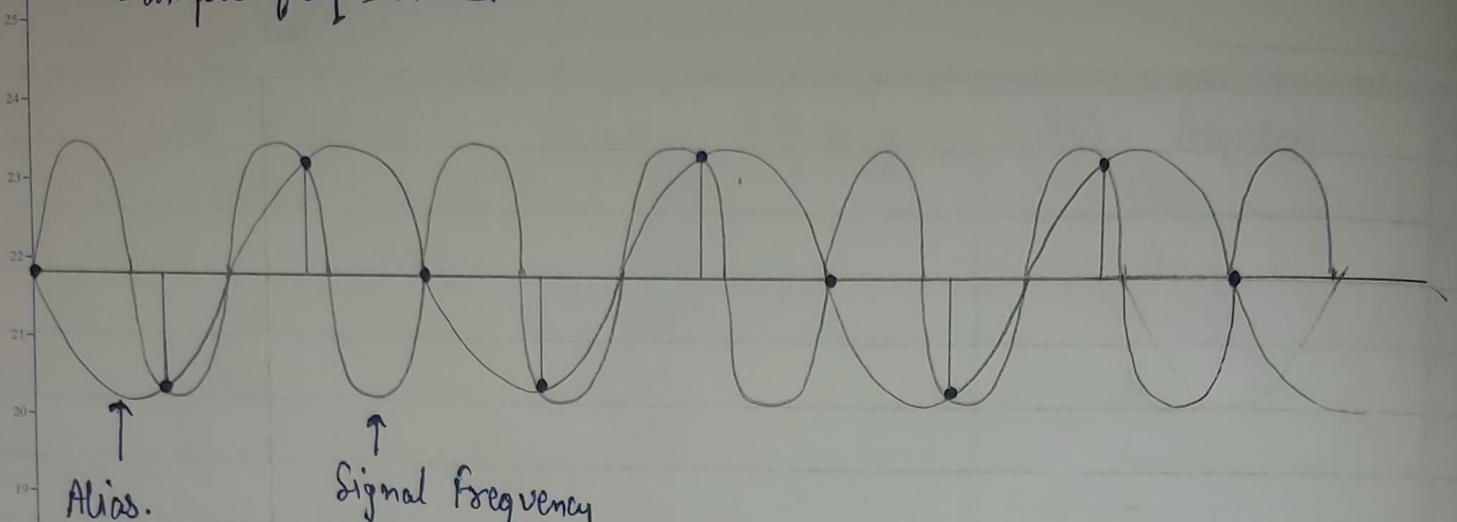
1) Sample freq. = 13 Hz



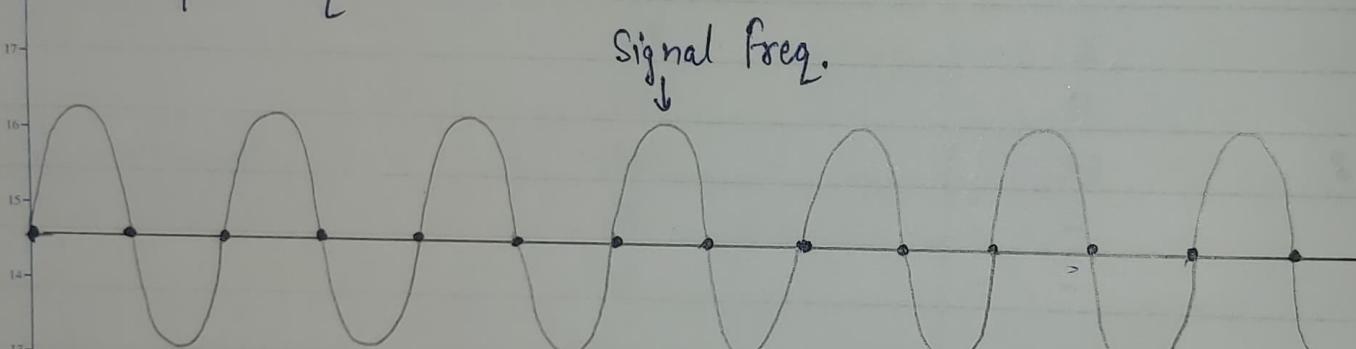
2) Sample freq. = 14 Hz



3) Sample freq = 15 Hz.



4) Sample Freq = 20 Hz.



5) Sampling Freq = 25 Hz.

