Aug 19, 2020

[U19CS012]

# EXPERIMENT No: 01

## SPECTRUM ANALYSER AND OBSERVE SPECTRUM

(*) **AIM:** To study Spectrum Analyzer and observe the spectrum of sinusoidal signal and square wave.

(*) **APPARATUS:** Spectrum Analyzer (9 kHz - 3 GHz) Function Generator

(*) **THEORY:** (1) A Spectrum Analyzer 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.

(2) 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

(3) 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.

(4) 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.

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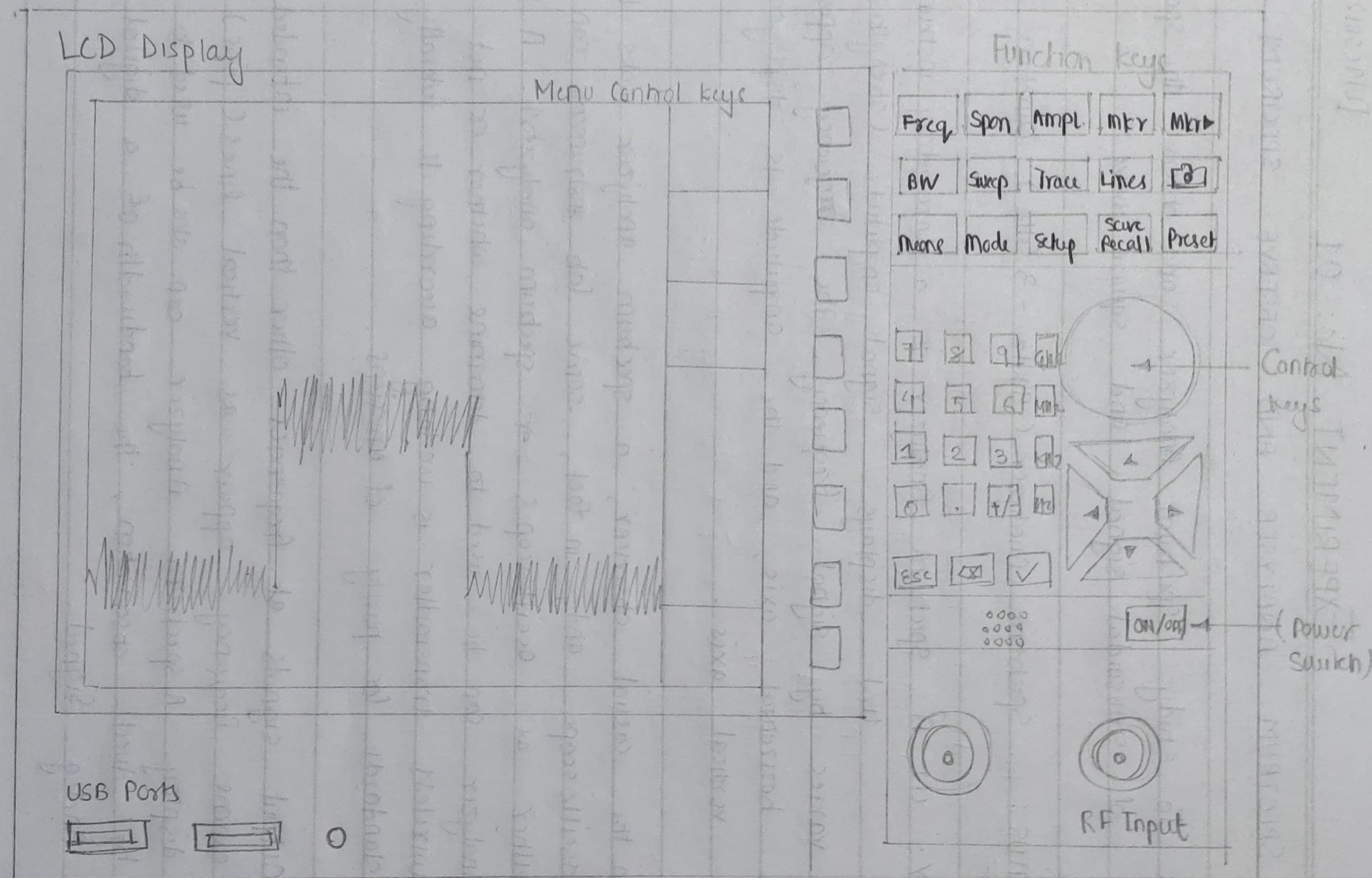


Spectrum Analyzer

## FEATURES OF LAB INSTRUMENT GSP-830 (GWINSTEK):

- ✓ 5 markers with delta marker & peak functions
- ✓ 3 traces
- ✓ Split windows with separate settings
- ✓ 6.4" TFT color LCD, 640 X 480 resolution
- ✓ AC/DC/ battery - multi-mode power operator
- ✓ Autoset
- ✓ 9 kHz - 3 Ghz frequency range

## FREQUENCY SELECTION AND THEIR SELECTION METHODS

### (1) FREQUENCY:

Frequency/ Span: The frequency key, toghether 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.

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 & scroll nobe.

(2) RANGE: 9 KHz to 3 GHz

(3) Set Center Frequency:

Panel Operation:

✓ Press frequency key
✓ press F1 (center)
✓ Enter the value using numerical and unit keys, arrow keys and scroll nobe.

(4) Set Frequency Span:

Panel operation:

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

(5) View Signal (Start & Stop)

✓ Start and stop method defines the beginning and end of the frequency range.
✓ Arrow keys and scroll knobe 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 nope.

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(7) Set Stop Frequency:

Panel operation:

✓ Press Frequency key

✓ press F3 (stop)

✓ Enter the value using numerical and unit keys, arrow keys and scroll nope.

(8) Full or zero span:

Full or zero span setting set the spans to extreme values : 3 GHz (full) or 0 KHz (zero) They provide faster ways to view signals in certain signals such as in time domains (0 span) for viewing modulation or in full span for viewing signals with unknown frequencies.

(9) Display full frequency span

Panel operation

✓ Press the span key

✓ Press F2 (full span)

✓ Range : 3 GHz (fixed)

✓ Full span also sets these parameters to fixed values

✓ Center frequency : 1.5 GHz

✓ Start frequency : 0 KHz

✓ Stop frequency : 3 GHz

(10) Zero Span Display ~~Panel operations~~

✓ Zero span display can be obtained by pressing F3 Key

✓ Start frequency & Stop frequency remains Same as center frequ-ency

✓ Note: Last span setting can be recalled by F4 key

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# AMPLITUDE SELECTION AND SETTING 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 operations:

✓ 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.

Range:

dBm : -110 to +20 dBm, 0.1 dBm resolution
dBmV : -63.1 to 66.99 dBmV, 0.01 dB resolution
dBμV : -3.01 to 126.99 dBμV, 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 ( dBm )
and F3 ( dBuV )
Press F6 (return) to go back to previous menu.
dBm = -110 to +20 dBm, 0.1 dBm 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

(5) Background
External offset compensates the amplitude gain or loss caused by an external network or devices.
Panel operations:
1. Press Amplitude key
2. Press F4 (external gain)
3. 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 Auxilary Signal : Press System key, press auxilary signal, select on option from side given menu, following signal will generate. It generate 10 MHz signal with 10 dB amplitude.

**Observation Table :** Next Pages →

Spectrum Diagrams ( waveforms)

(*) **Conclusion :** Hence, we have successfully verified and analysed the spectrum of sinusoidal signal and square wave for different frequency and amplitude

Observation Table

Waveform : SINE

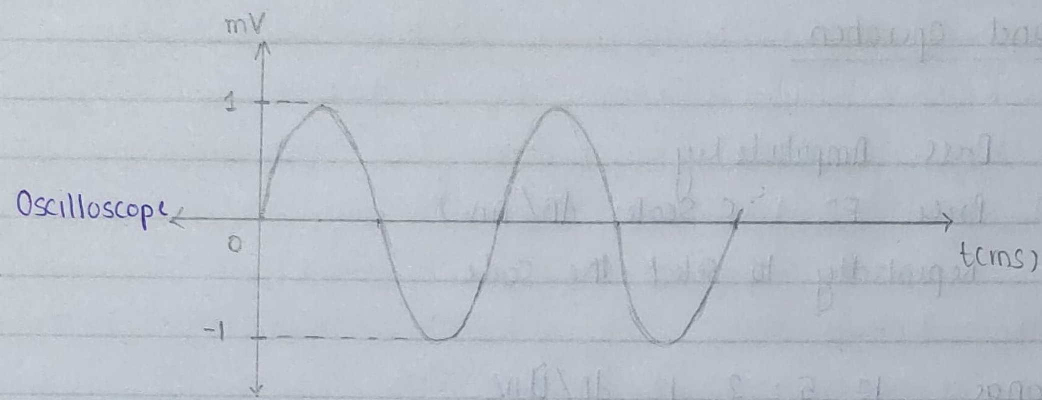
| Sr. No. | Frequency (KHz) | Amplitude (mV) |
|---|---|---|
| 1 | 10 | 1 |
| 2 | 15 | 1 |
| 3 | 15 | 2 |
| 4 | 12.5 | 2 |
| 5 | 12.5 | 0.5 |

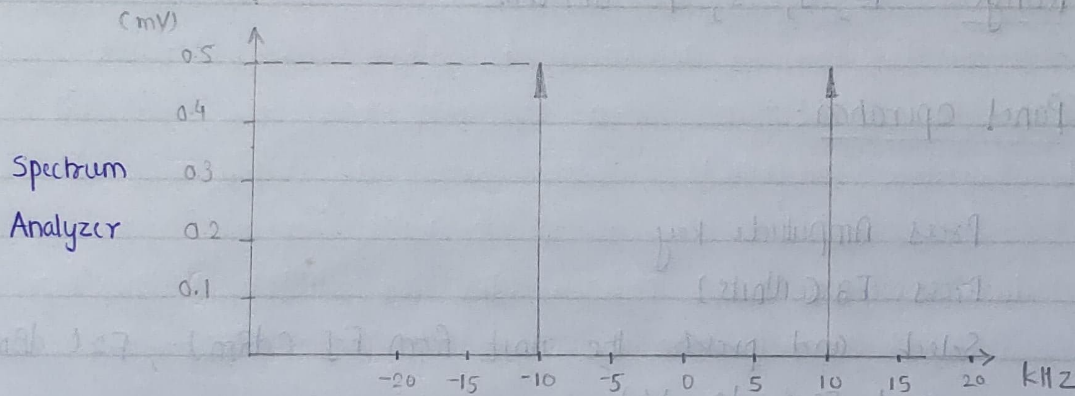
Waveform : SQUARE

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

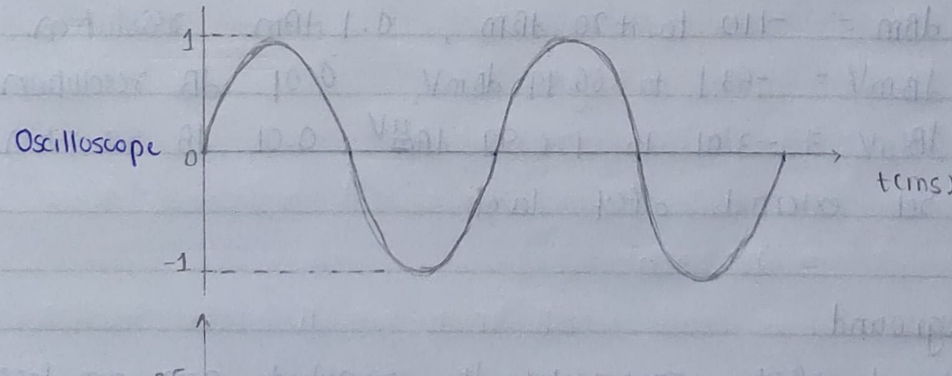
# SINE WAVE

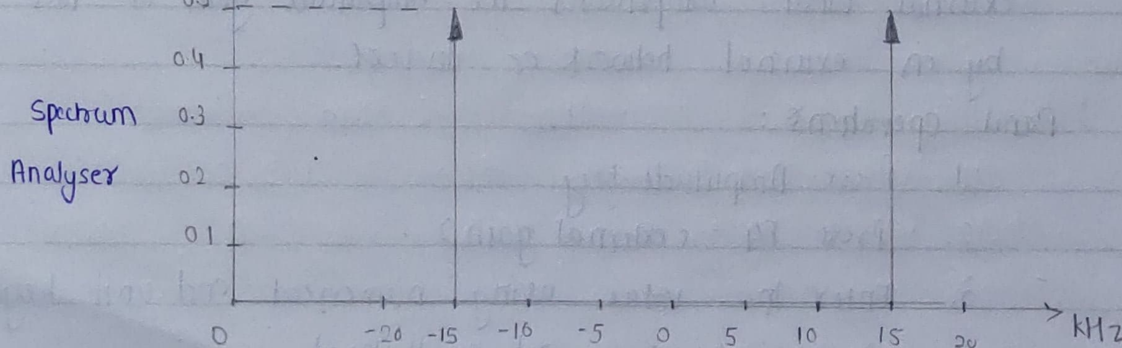
1.> A = 1 mV    f = 10 KHz



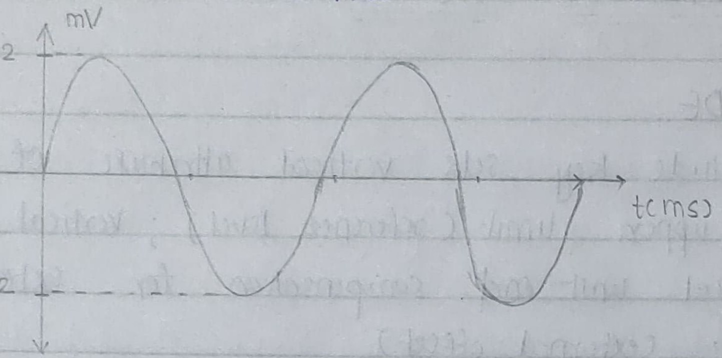

2.> A = 1 mV    f = 15 KHz



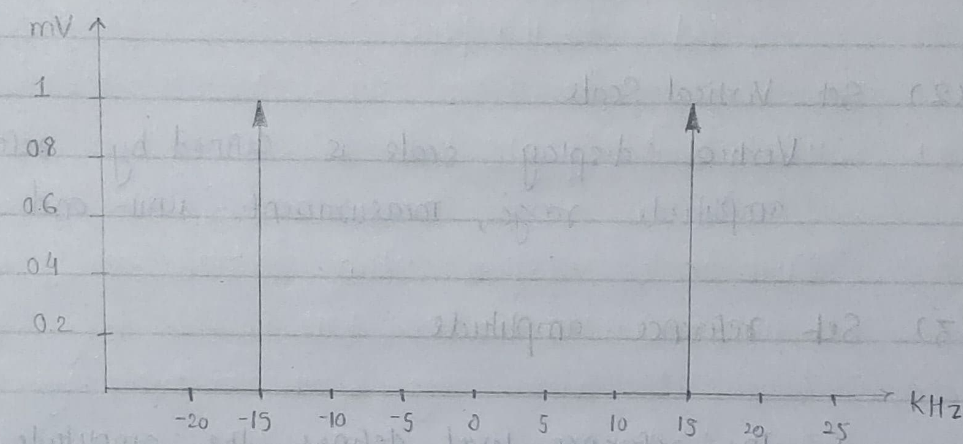



3.> $A = 2\,mV$   $f = 15\ kHz$

Oscilloscope

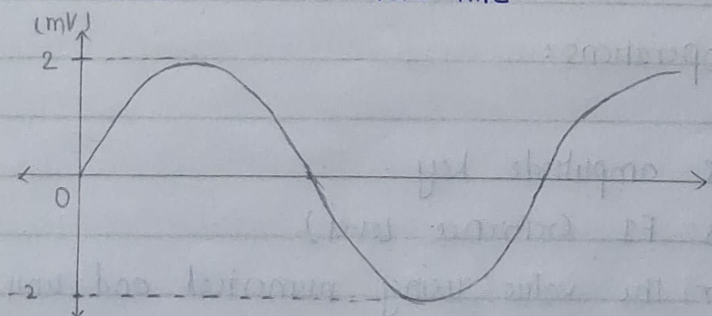


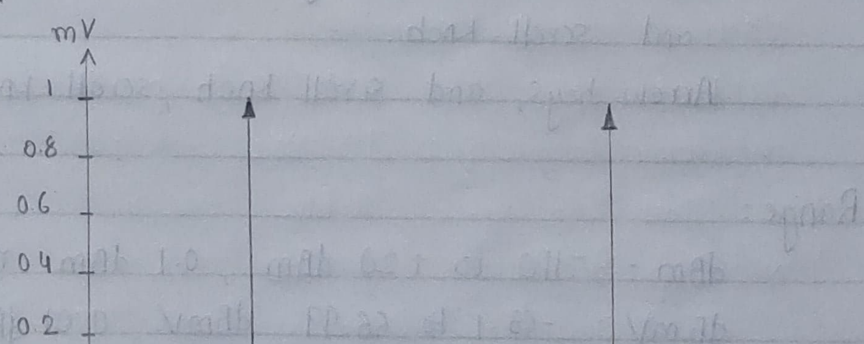

Spectrum Analyser




4.> $A = 2\,mV$   $f = 12.5\ kHz$

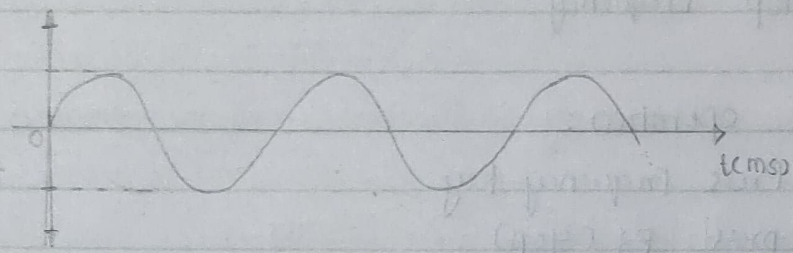
Oscilloscope



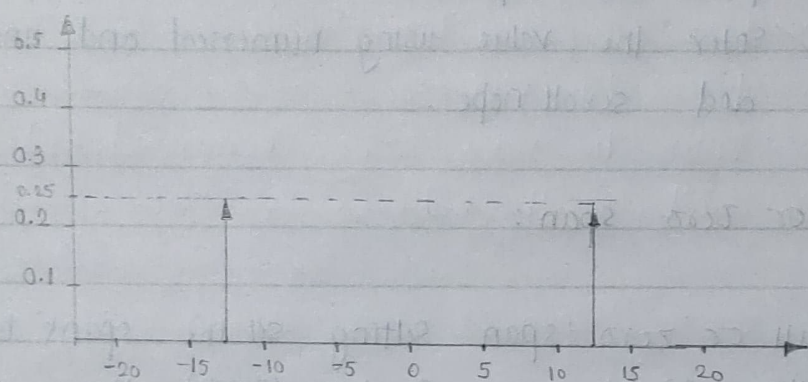



5.> A = 0.5 mV  f = 12.5 KHz

Oscilloscope

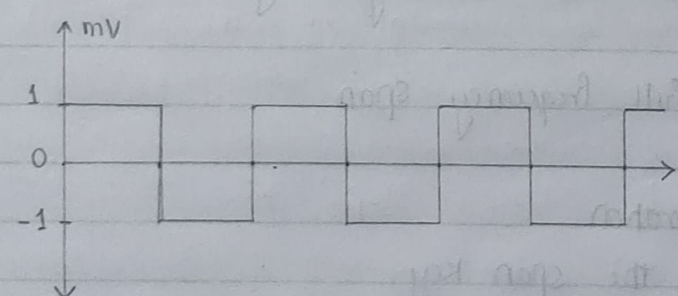


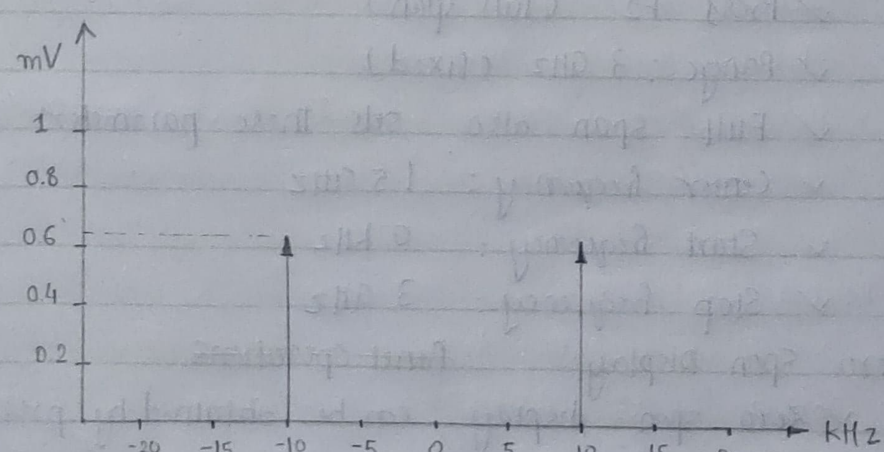

Spectrum Analyser




## SQUARE WAVE

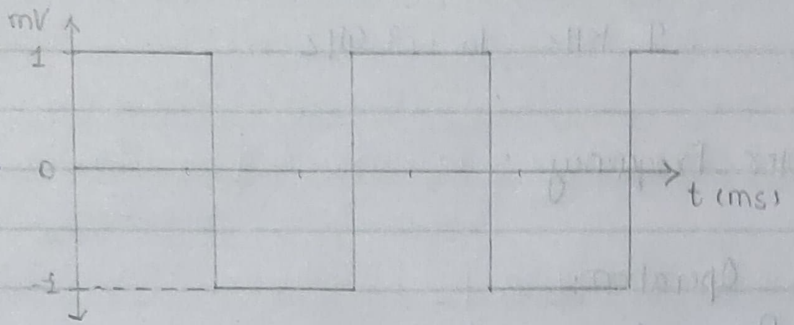
1.> A = 1mV  f = 10 kHz



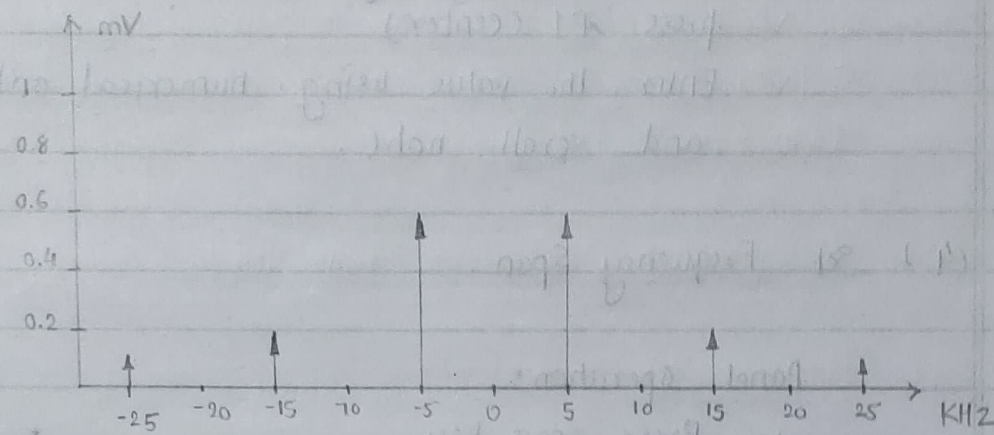



2.> A = 1 mV  f = 5 KHz

Oscilloscope

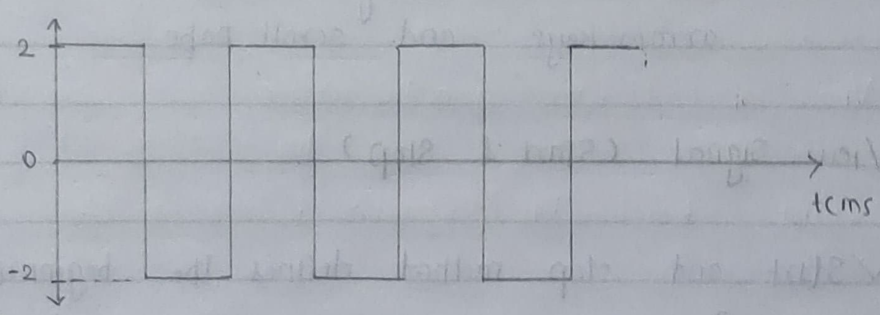



Spectrum Analyser

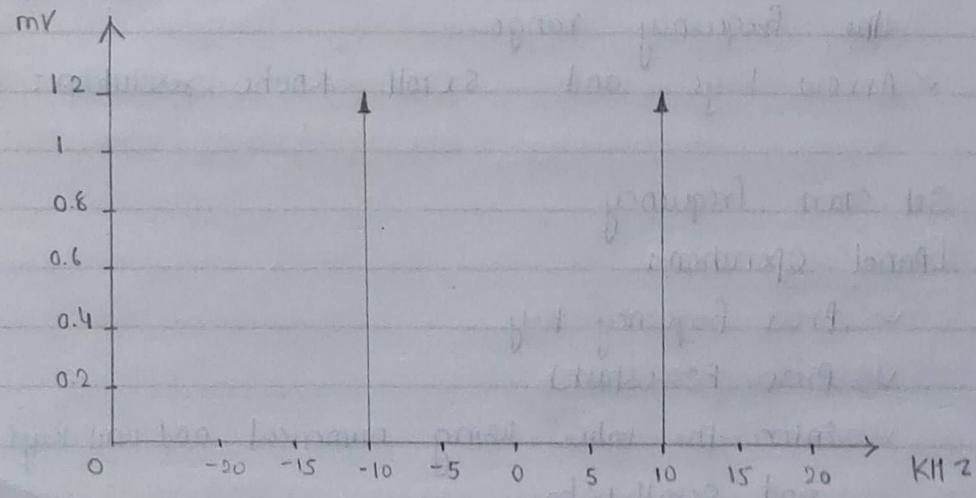



3.> A = 2 mV  f = 10 KHz

Oscilloscope




Spectrum Analyser



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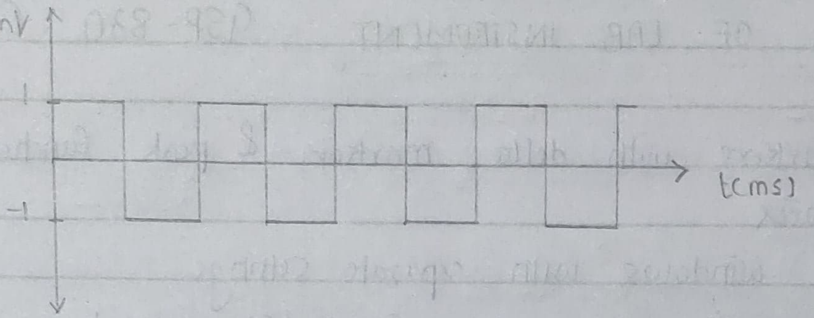
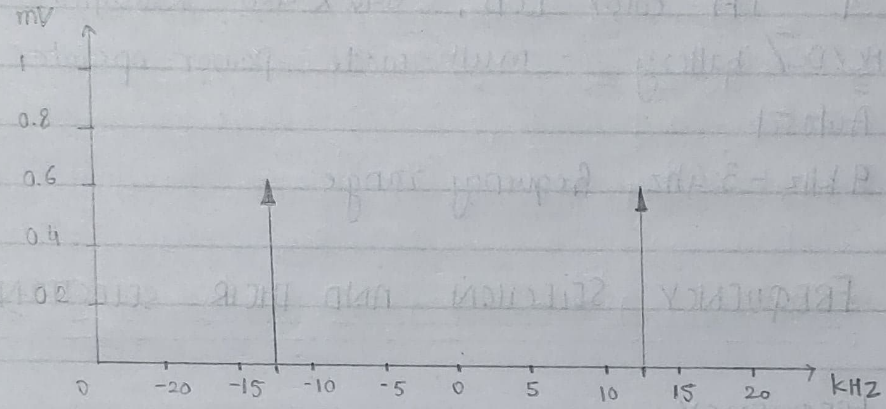
4.> A = 1 mV    f = 12.5 KHZ

Oscilloscope

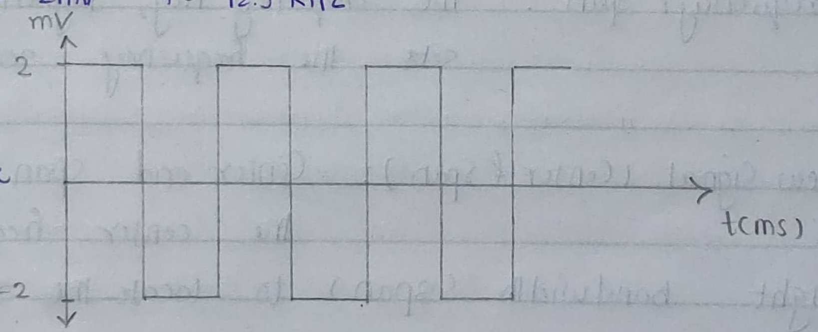


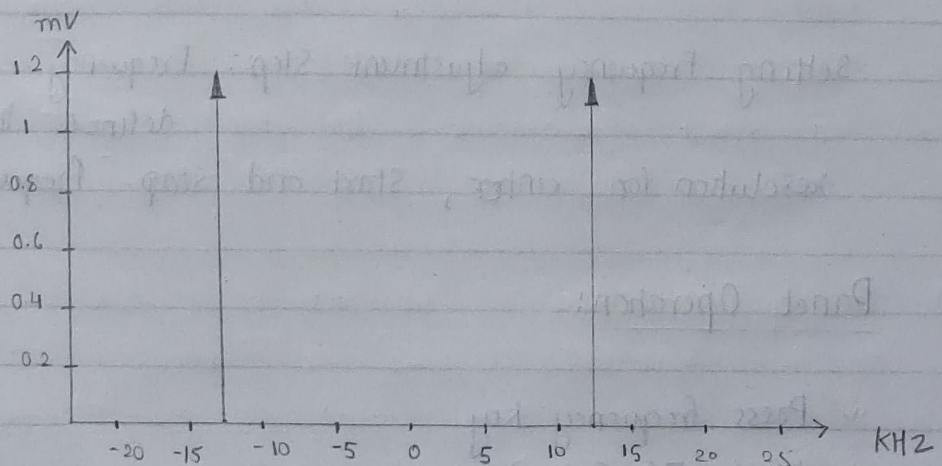

Spectrum Analyzer




5.> A = 2mV    f = 12.5 KHZ

Oscilloscope



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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.

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

> It has three processes

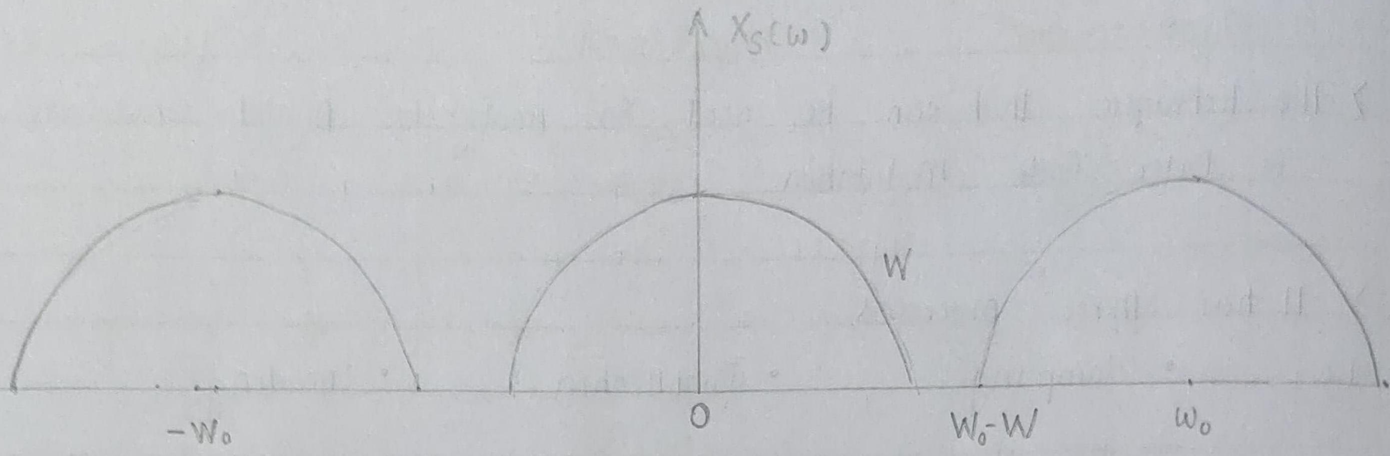
- Sampling
- Quantization
- Encoding

(2) Sampling

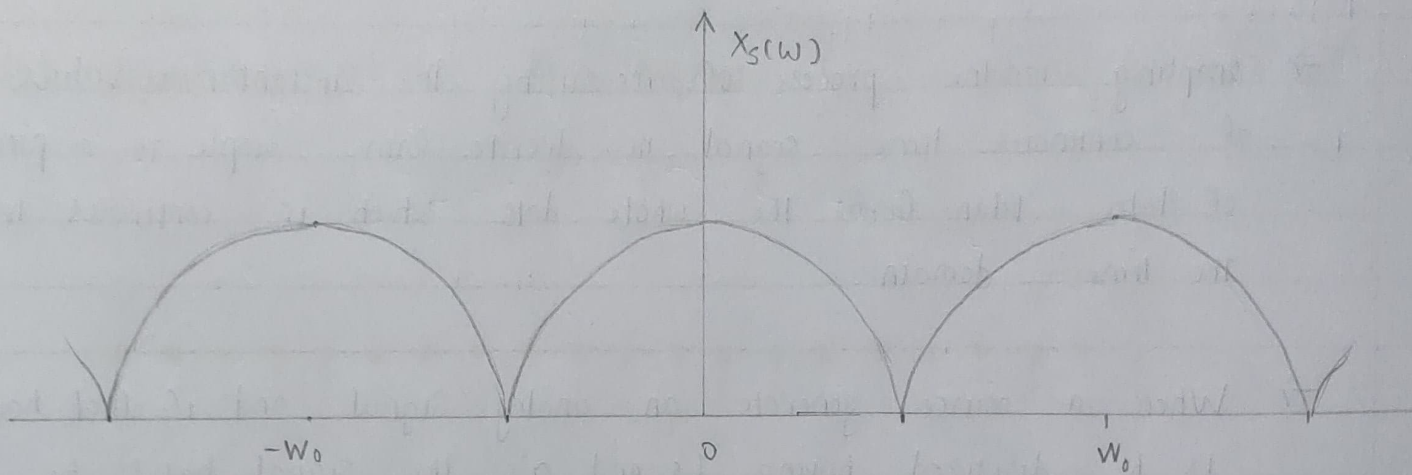
→ Sampling is the process of measuring the instantanous values of continous - time signal in discrete form. Sample is a piece of data taken from the whole data which is continous in the time domain.

→ 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.

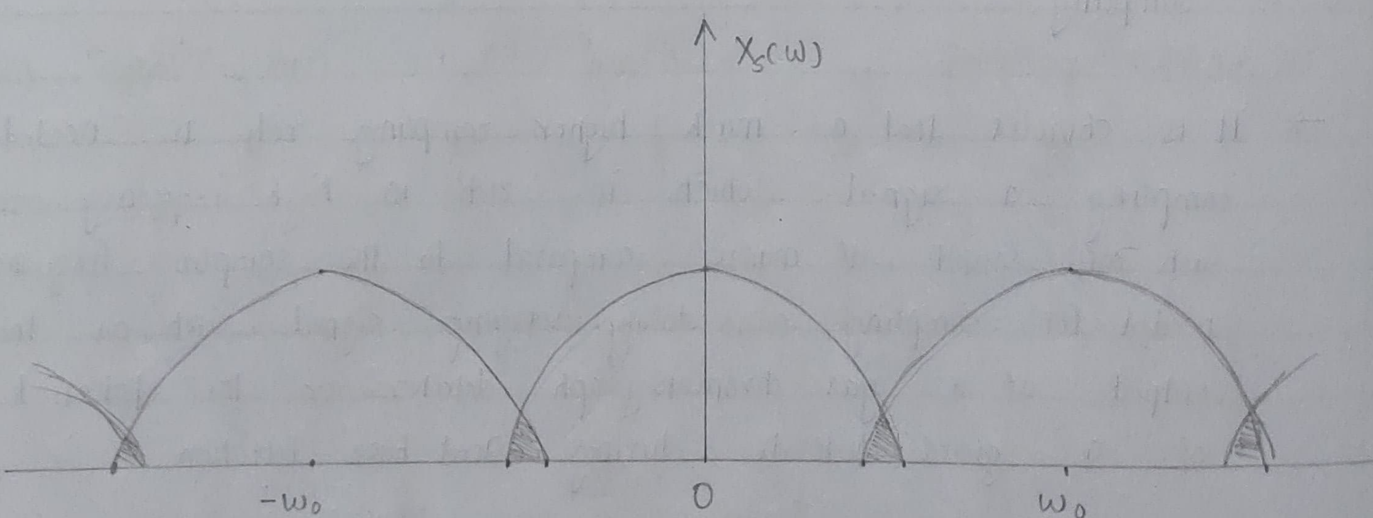
→ It is obvious that a much higher sampling rate is needed for sampling a signal which is rich in high frequency components such as sound of music compared to the sampling frequency needed for sampling a slowly varying signal, such as the output of a gas - chromatograph detector or the potential of a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




(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 $\qquad$ $W$ = 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

$$X_s(\omega) = \frac{1}{T_S} \sum_{n=-\infty}^{0} X(\omega - n\omega_0)$$

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.

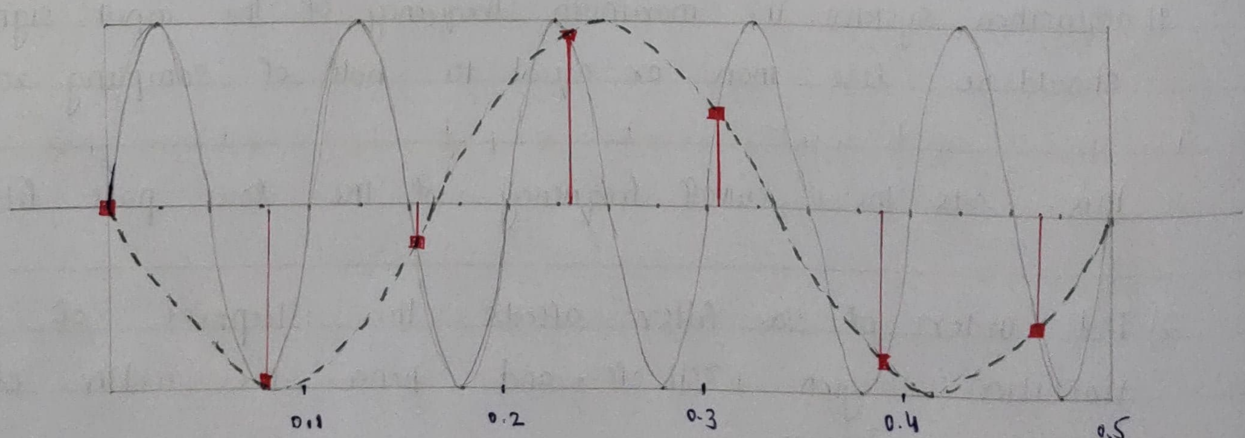
A filter of the $n^{th}$ order will be have a roll-off rate of $n \times 20$ dB/ decade.

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

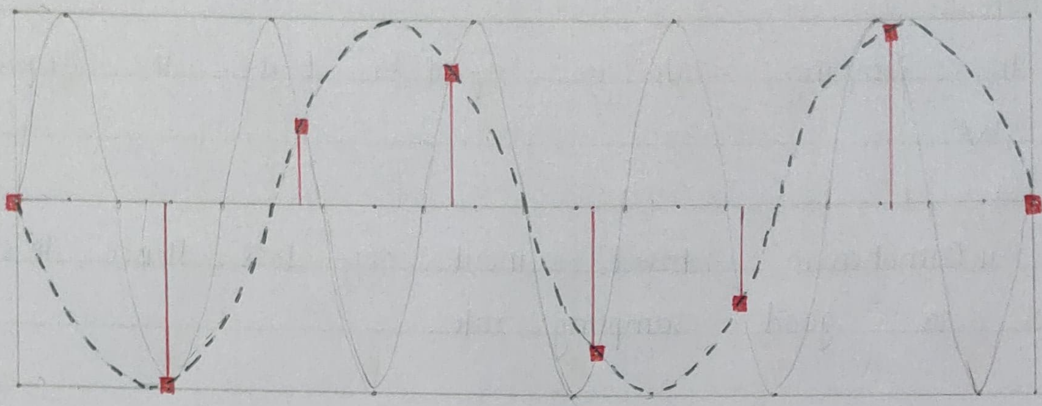
## Observation Table

| 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 | - |

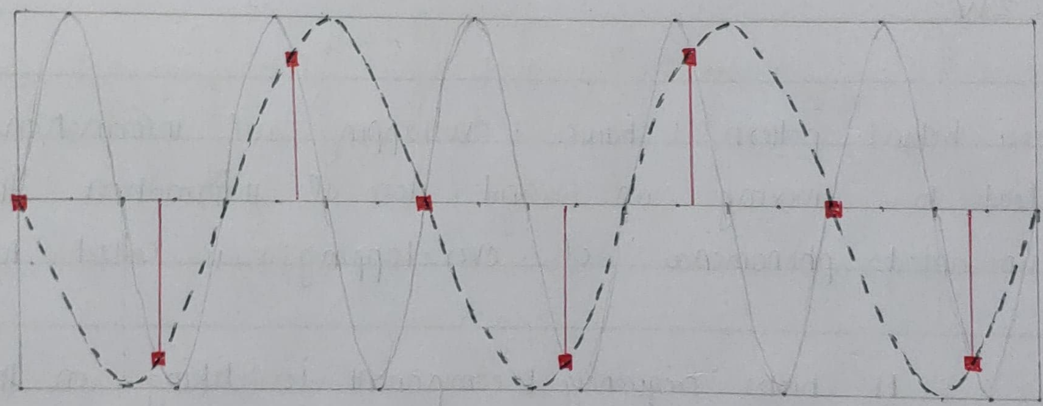
(1) Signal freq. (Hz) = 10.0 Alias freq. (Hz) 3.0 Sampling freq. (Hz) = 13.0



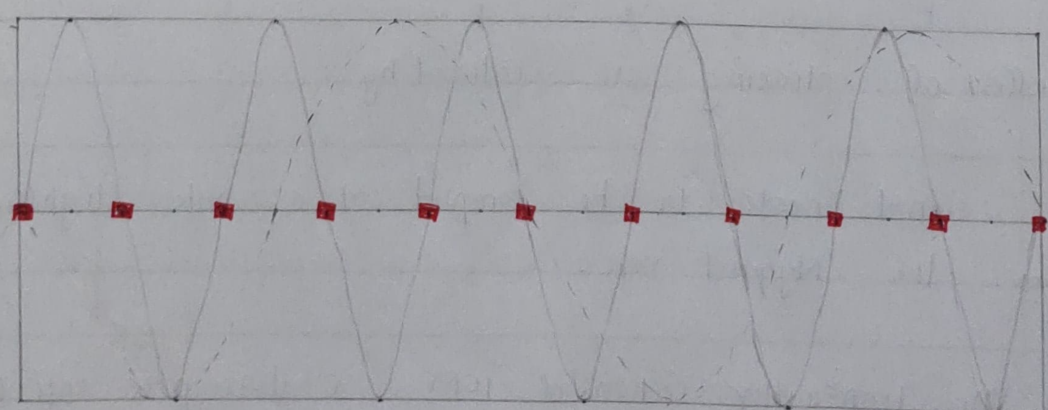
(2) Signal freq. (Hz) = 10.0 Alias freq (Hz) = 4.0 Sampling freq (Hz) = 14.0



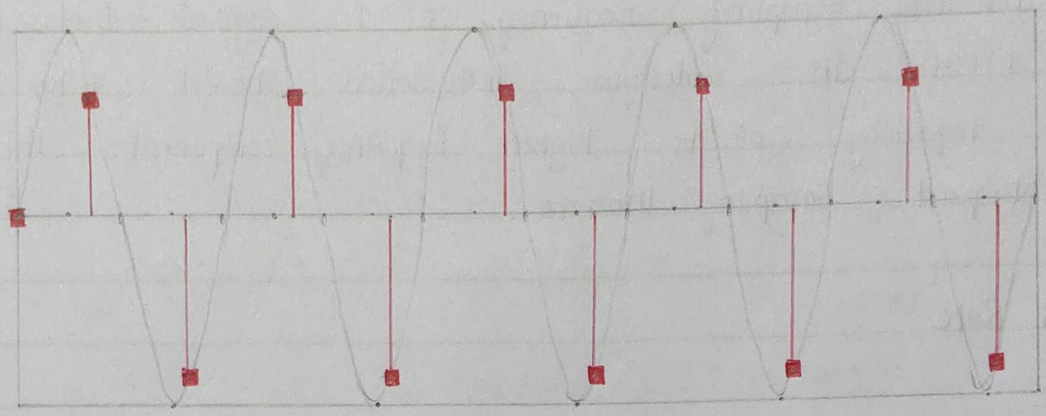
(3) Signal freq (Hz) = 10.0 Alias freq (Hz) = 5.0 Sampling freq (Hz) = 15.0



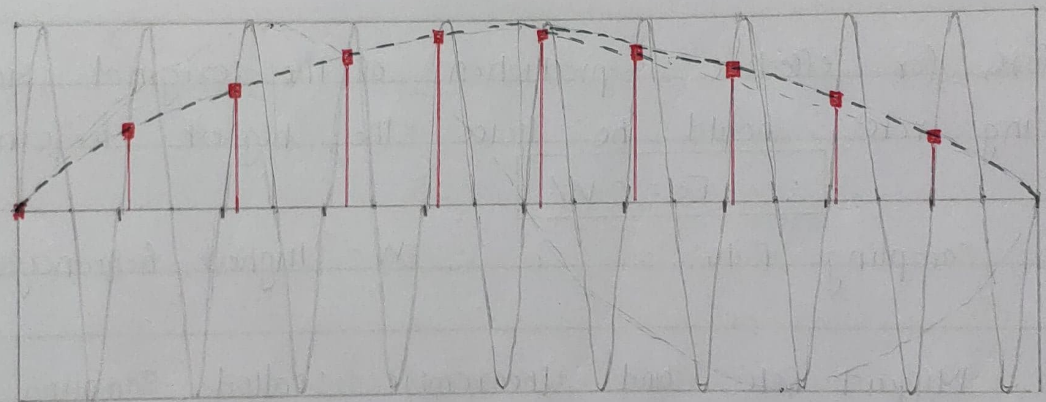
(4) Signal freq (Hz) = 10.0 Alias freq (Hz) = - Sampling freq (Hz) = 20



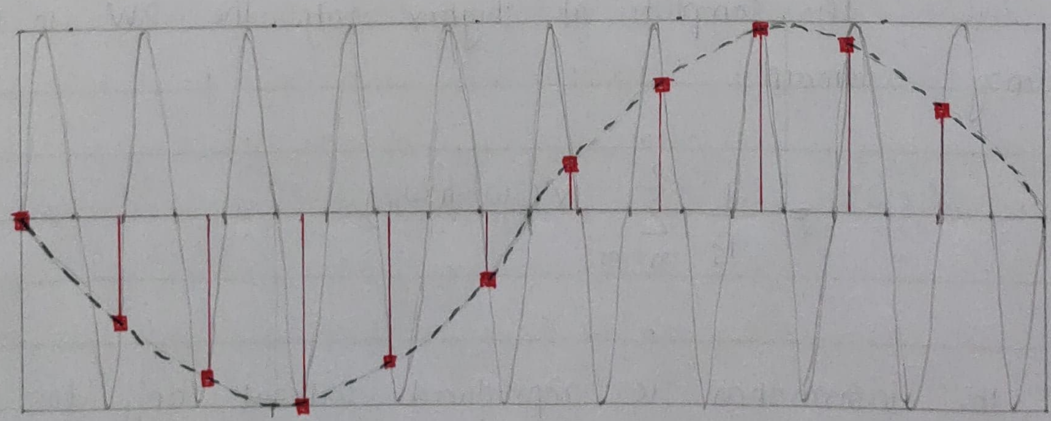
(5) Signal freq (Hz) = 10.0 Alias freq (Hz) = - Sampling freq (Hz) = 25



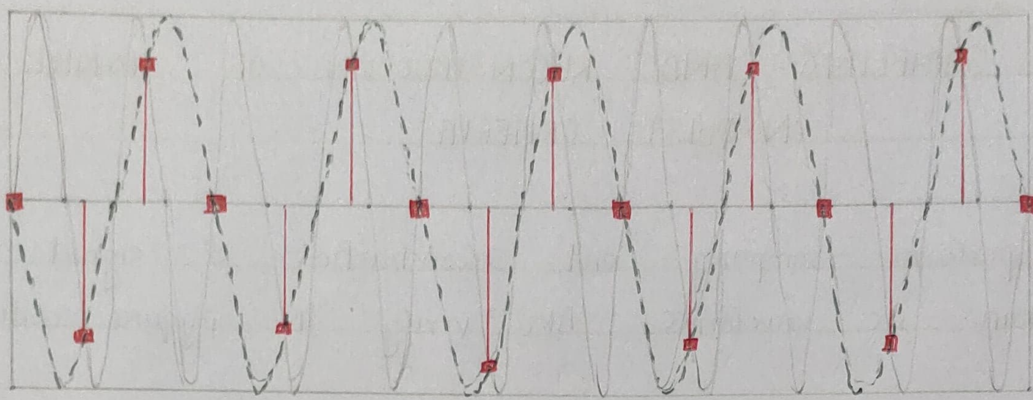
(6) Signal freq. (Hz) = 20.0 Alias freq. (Hz) = 1.0 Sampling freq. (Hz) = 19



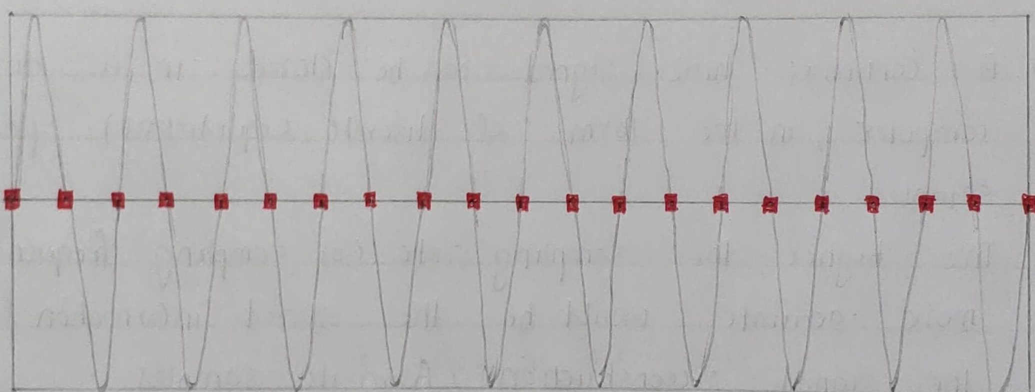
(7) Signal freq. (Hz) = 20.0 Alias freq. (Hz) = 2.0 Sampling freq. (Hz) = 22.0



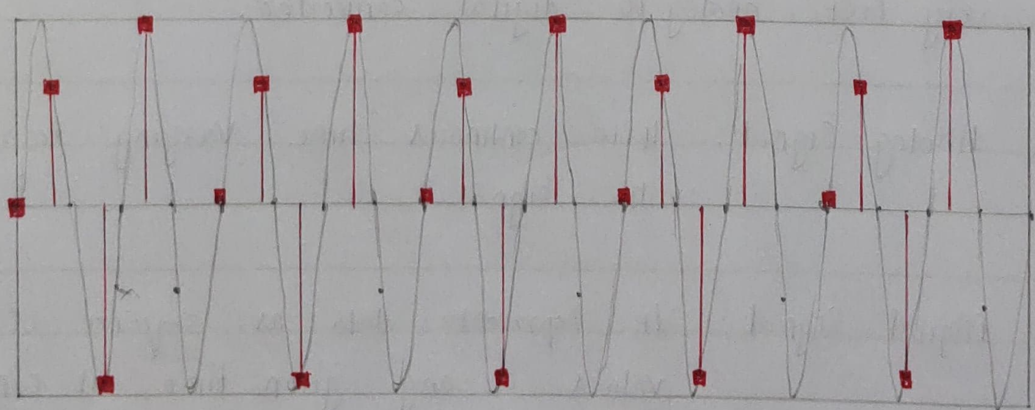
(8) Signal freq. (Hz) = 20.0 Alias freq. (Hz) = 10 sampling freq (Hz) = 30



(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



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EXPERIMENT 3 :

# AMPLITUDE MODULATION

> AIM : Study of an Amplitude modulated (A.M.) scheme, depth of modulation, waveforms, spectra and trapezoidal display.

> APPARATUS : Lab Alive Software

> THEORY : 1.> Classification of AM modulation

- Double Side Band Suppressed Carrier (DSB-SC)
- Double Side Band with carrier (AM)
- Single side Band (SSB)
- Vestigal side Band (VSB)

2.> AM

Let modulating signal be $m(t) = A_m \cos(2\pi f_m t)$, carrier signal be $c(t) = A_c \cos(2\pi f_c t)$

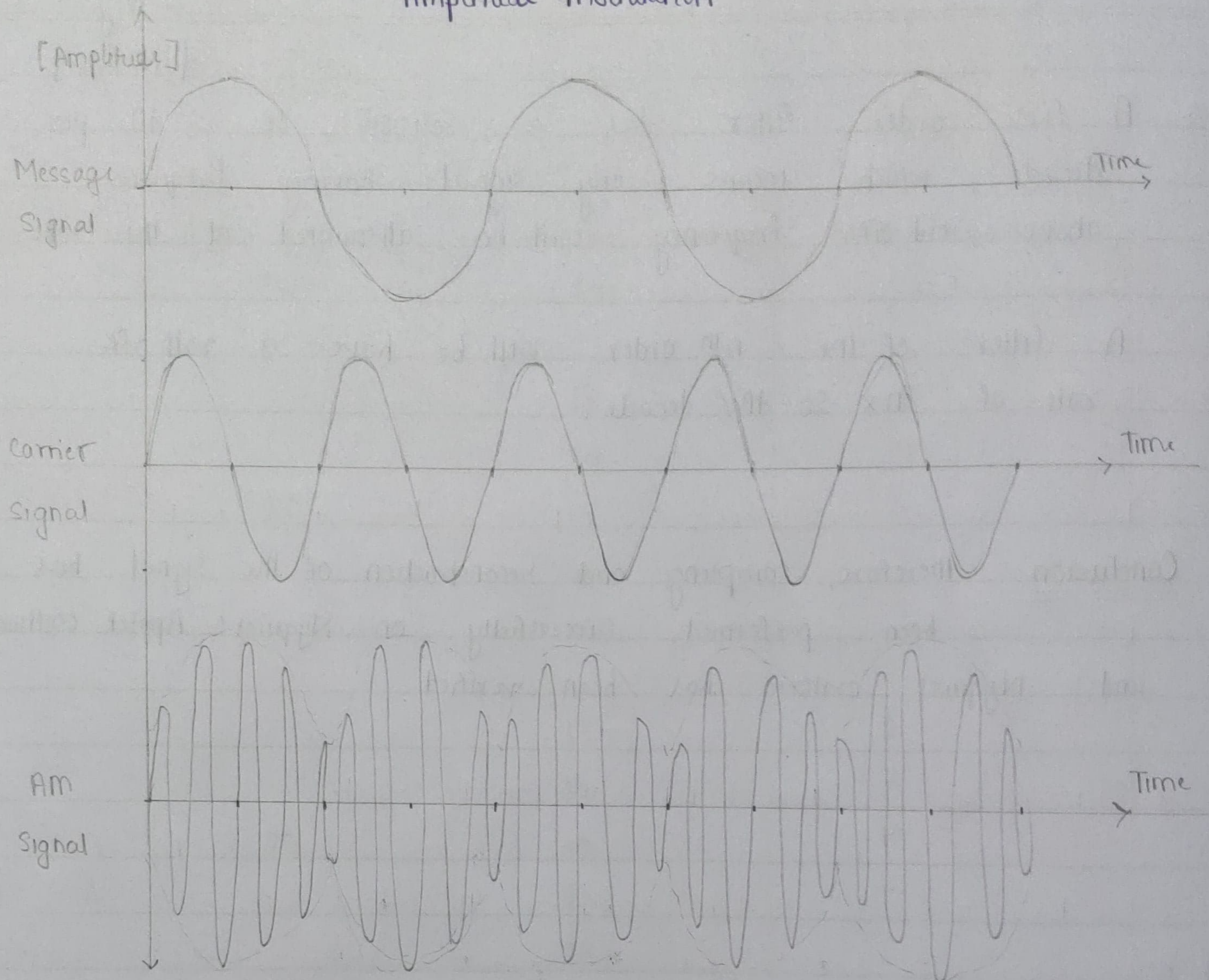
$\therefore$ AM wave be $s(t) = [A_c + A_m \cos(2\pi f_m t)] \cos(2\pi f_c t)$

$$s(t) = A_c \left[ 1 + \frac{A_m}{A_c} \cos(2\pi f_m t) \right] \cos(2\pi f_c t)$$

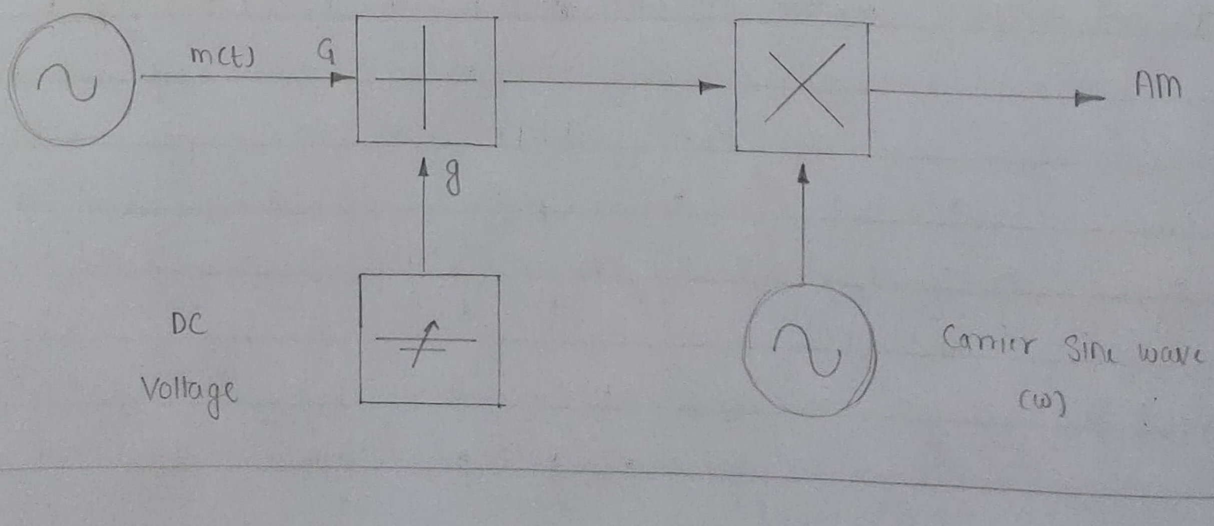
Modulation index = $m = \frac{A_m}{A_c}$

$$s(t) = A_c \cos(2\pi f_c t) + \frac{m}{2} A_c \cos(2\pi)(f_c - f_m)t + \frac{m}{2} A_c \cos(2\pi(f_c + f_m)t)$$

# Amplitude Modulation




## Schematic Block diagram for AM, $T_x$ and $R_x$



## 3.) Measurement of 'm'

- The magnitude of 'm' can be measured directly from the Am display itself.
- maximum and minimum amplitudes of the Transmission signals ~~ener~~ envelope, determine the modulation depth:

$$m = \frac{Am}{Ac}$$

Max. Amplitude of modulated wave, $a = Am + Ac$
Min. Amplitude of modulated wave, $b = A_c - Am$

$$\therefore \quad Ac = \frac{(a+b)}{2}, \quad Am = \frac{(a-b)}{2}$$

$$\therefore \quad m = \left(\frac{Ac}{Am}\right)^{-1} = \left(\frac{(a+b)}{(a-b)}\right)^{-1} = \frac{(a-b)}{(a+b)}$$

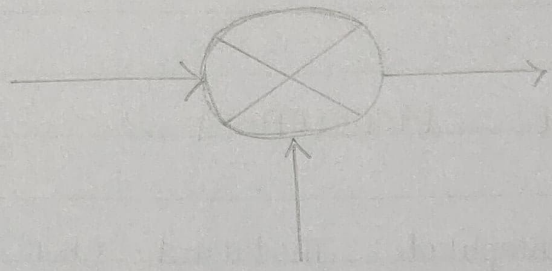
## 4.) Envelope Detector

- This non-coherent detection doesn't require a carrier recovery circuit. In its simplified form, it consist of a rectifier diode and a low pass filter.

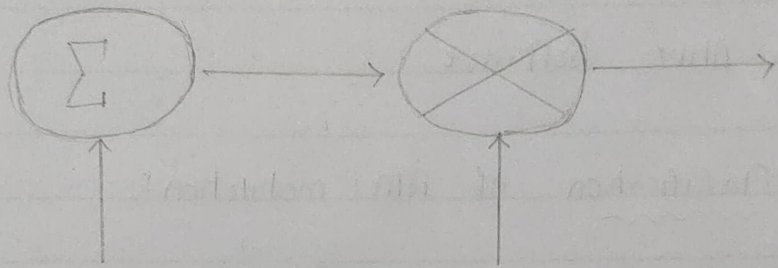
## 5.) Synchronous detector

- Am without a carrier. Envelope detection can't be deployed because the transmitted signal's envelope changes sign. Transmit spectrum of DSB-SC.

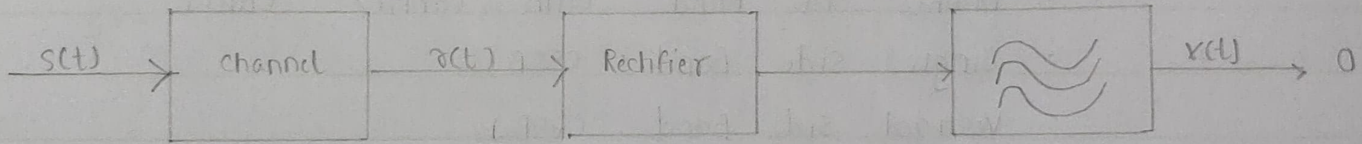
## DSB - SC

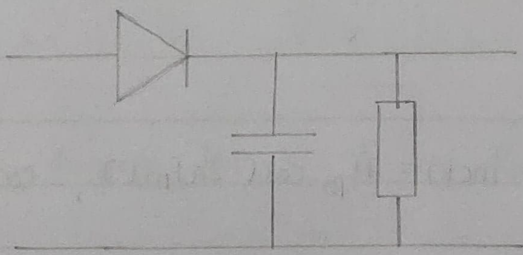


## AM



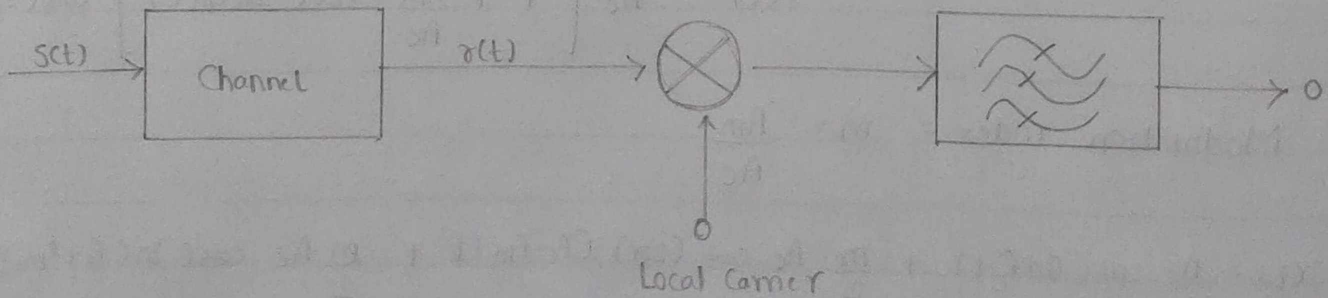
## Envelope Detector



Rectifier
(Diode, Capacitor)

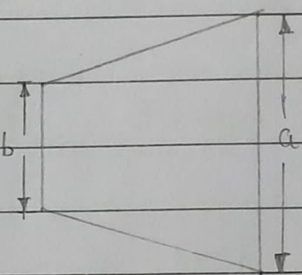
## Synchronous Detector



## 6.) Trapezoid Method

- We can calculate 'm' in the time domain using an oscilloscope and the trapezoid method.

- The slope is placed in XY mode
  - X : modulating signal
  - Y : modulating signal




- The modulation index is then calculated from the vertical edge lengths using

$$m = \frac{(a-b)}{(a+b)}$$

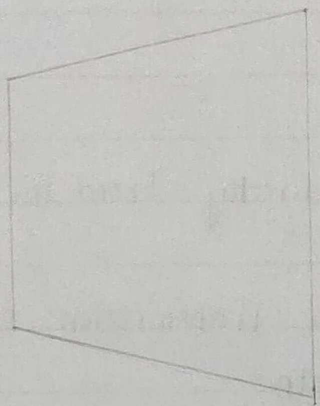
> **PROCEDURE:** In this online mode of practical, we perform the experiment on Labaliva application.

1.) We will first execute the Am analyser simulator

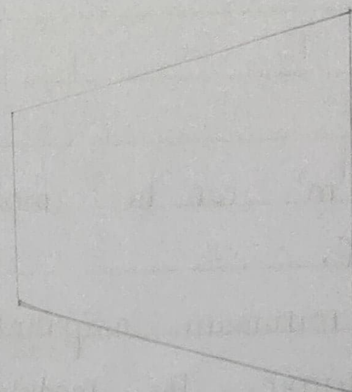
2.) After executing the Am analyser simulator, click on the $\hat{S}$ in the Am modulation window.

3.) For D.S.B. with carrier click on the D.C. and for D.S.B. with suppressed carrier off the D.C output

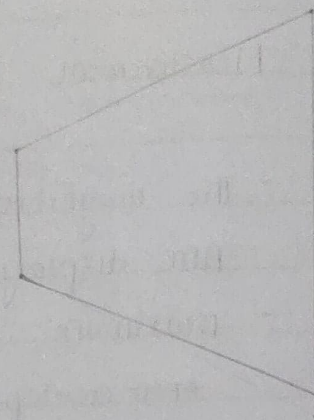
Trapezoid width is unaffected by modulation depth.



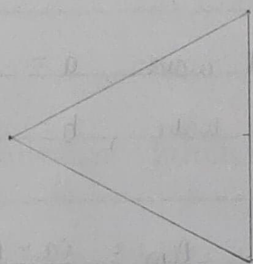
m = 0.1



m = 0.3

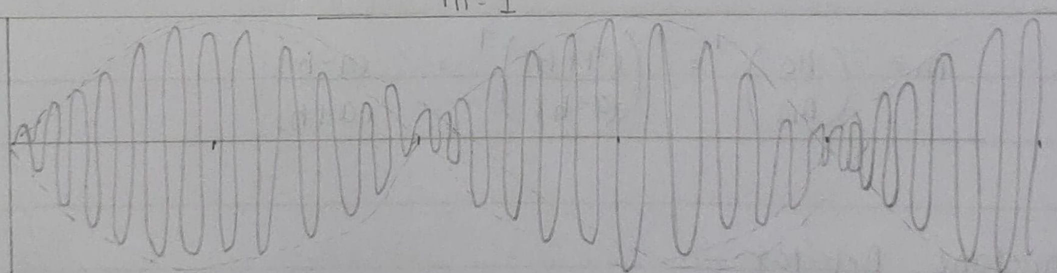


m = 0.5

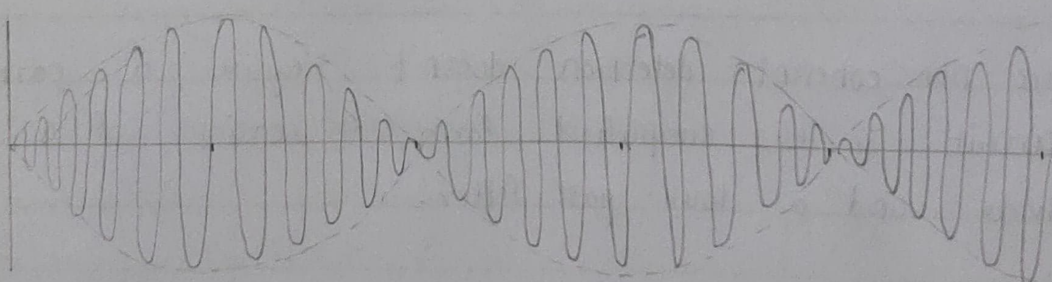


m = 1

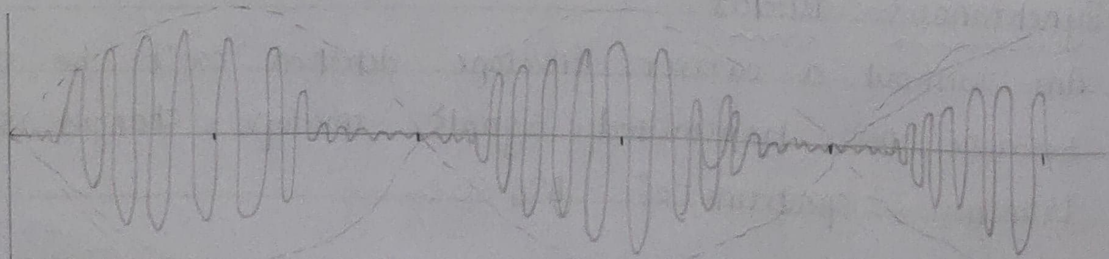
Modulation Index = 0.5 (m<1)



Modulation Index = 1 (m = 1)



Modulation Index = 1.5 (m > 1)



## Observation

## DOUBLE SIDE BAND WITH CARRIER

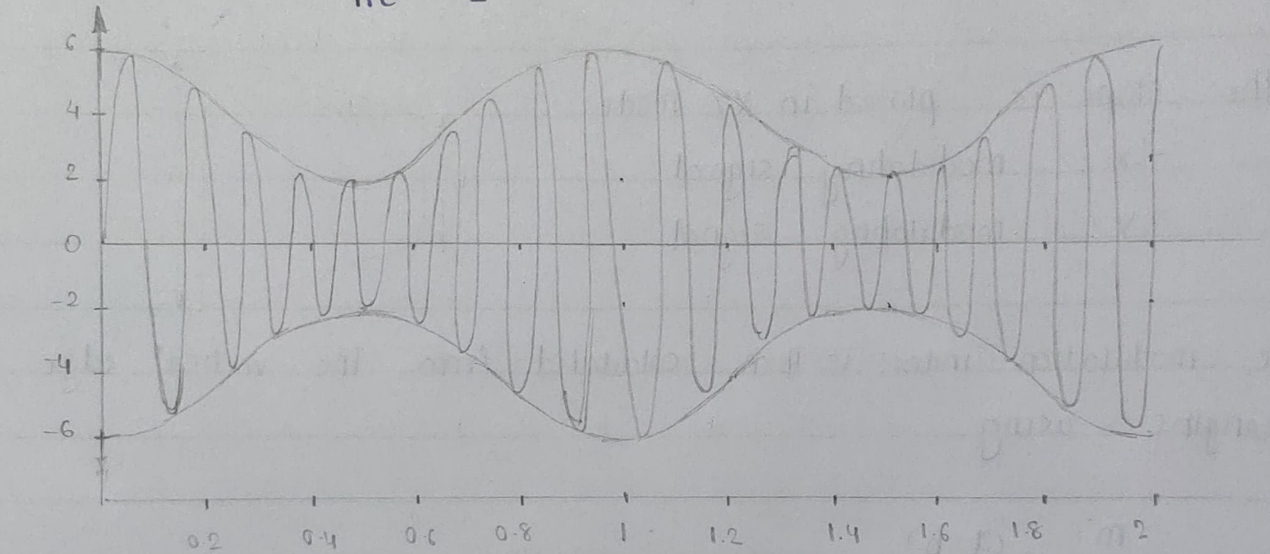
a) $m<1$

Message = $A_m = 1V$ $f_m = 1\,MHz$ (Cosine)

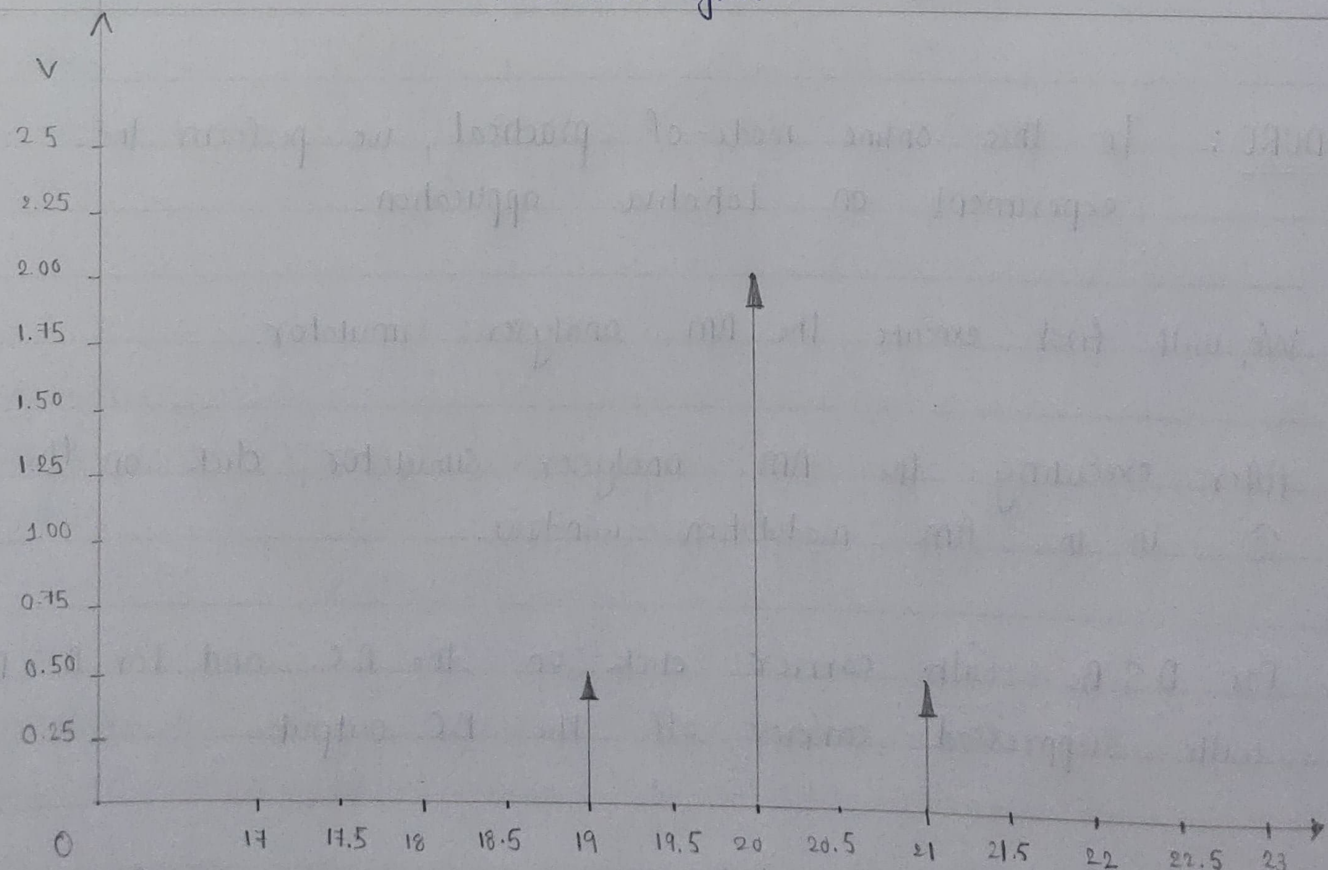
Carrier = $A_c = 2V$ $f_c = 20\,MHz$ (cosine)

$\hat{S}$ = $A = 2V$ $f = 1\,MHz$ (DC on)

$$\mu = \frac{A_m}{A_c} = \frac{1}{2} = 0.5$$




Transmitted signal

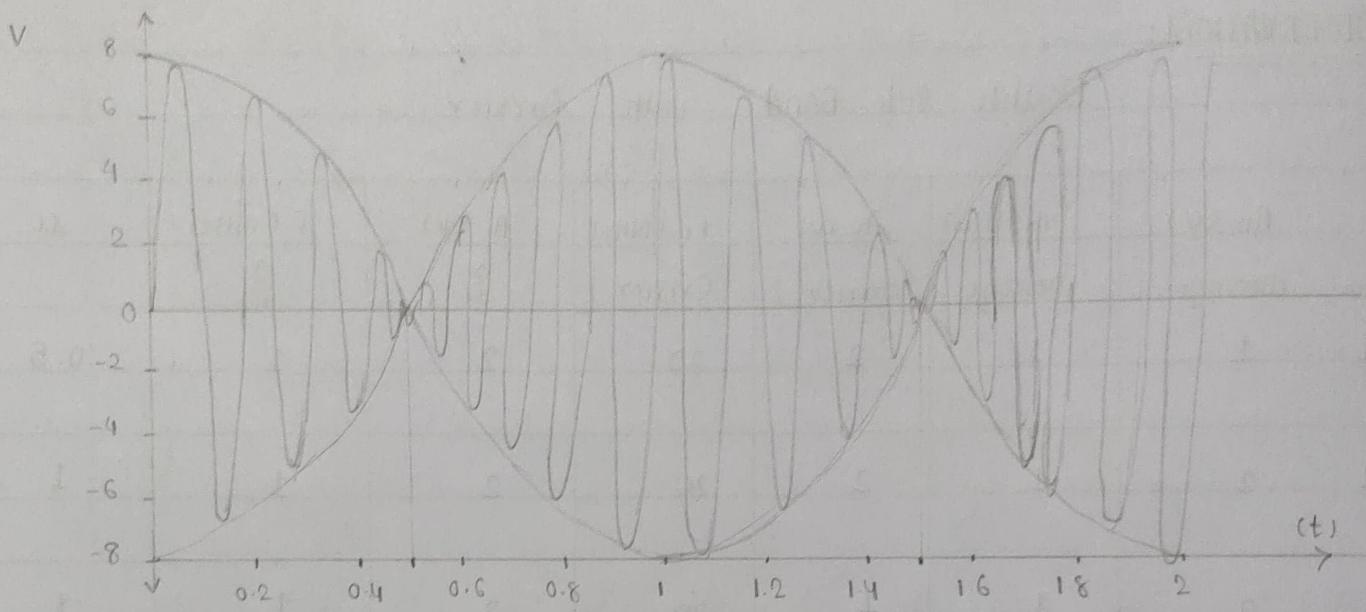



Spectrum

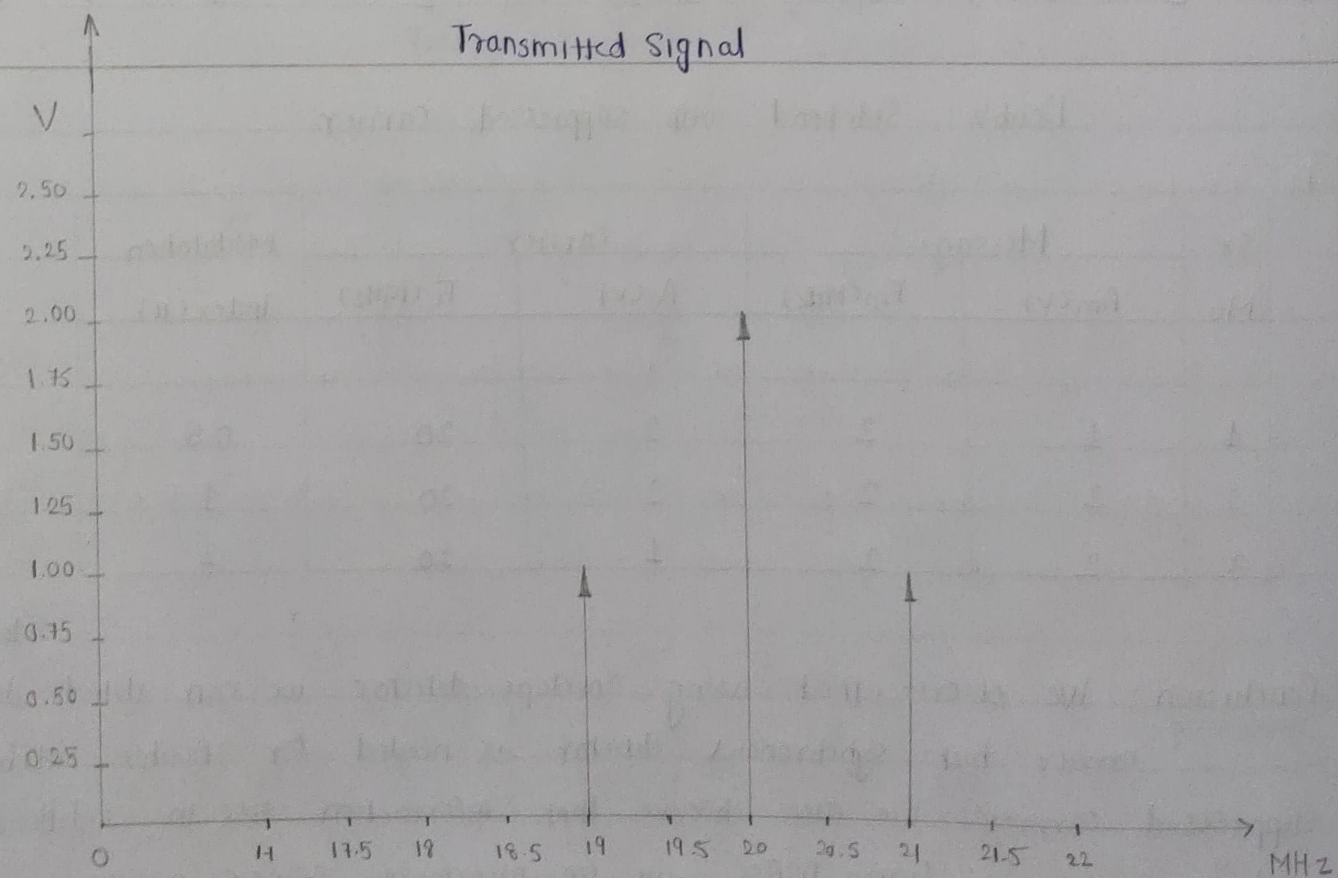
(b) m = 1

| | | |
|---|---|---|
| Message | $A_m = 2V$ | $f_m = 1 MHz$ (cosine) |
| Carrier | $A_c = 2V$ | $f_c = 20 MHz$ (cosine) |
| $\hat{\delta}$ | $A = 2V$ | $F = 1 MHz$ (DC on) |

$$\mu = \frac{A_m}{A_c} = 1$$




Transmitted Signal

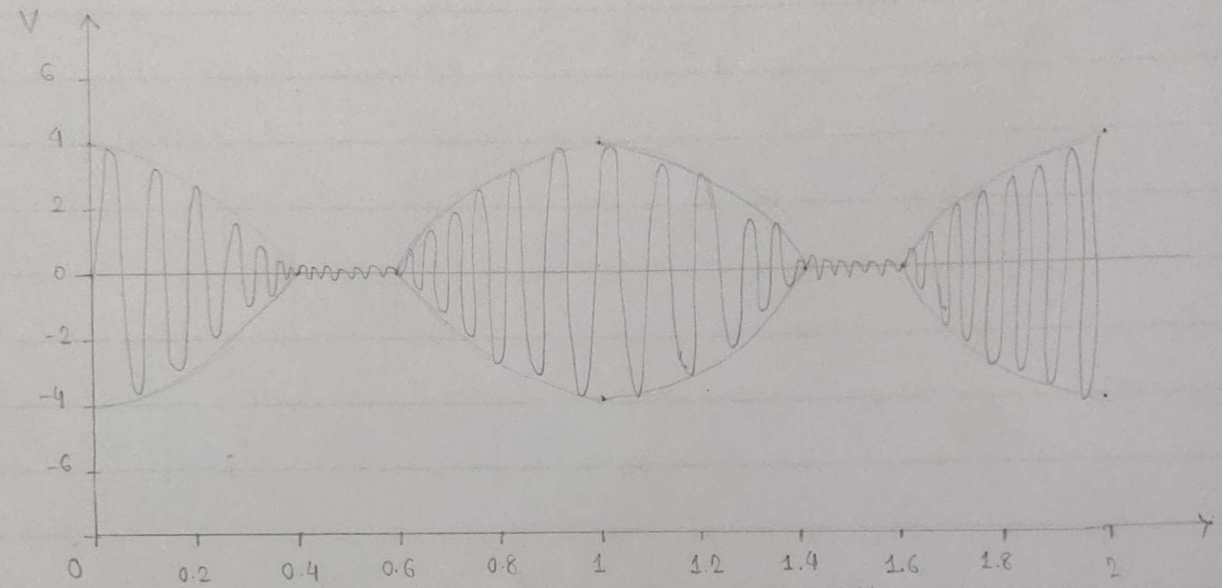



Spectrum

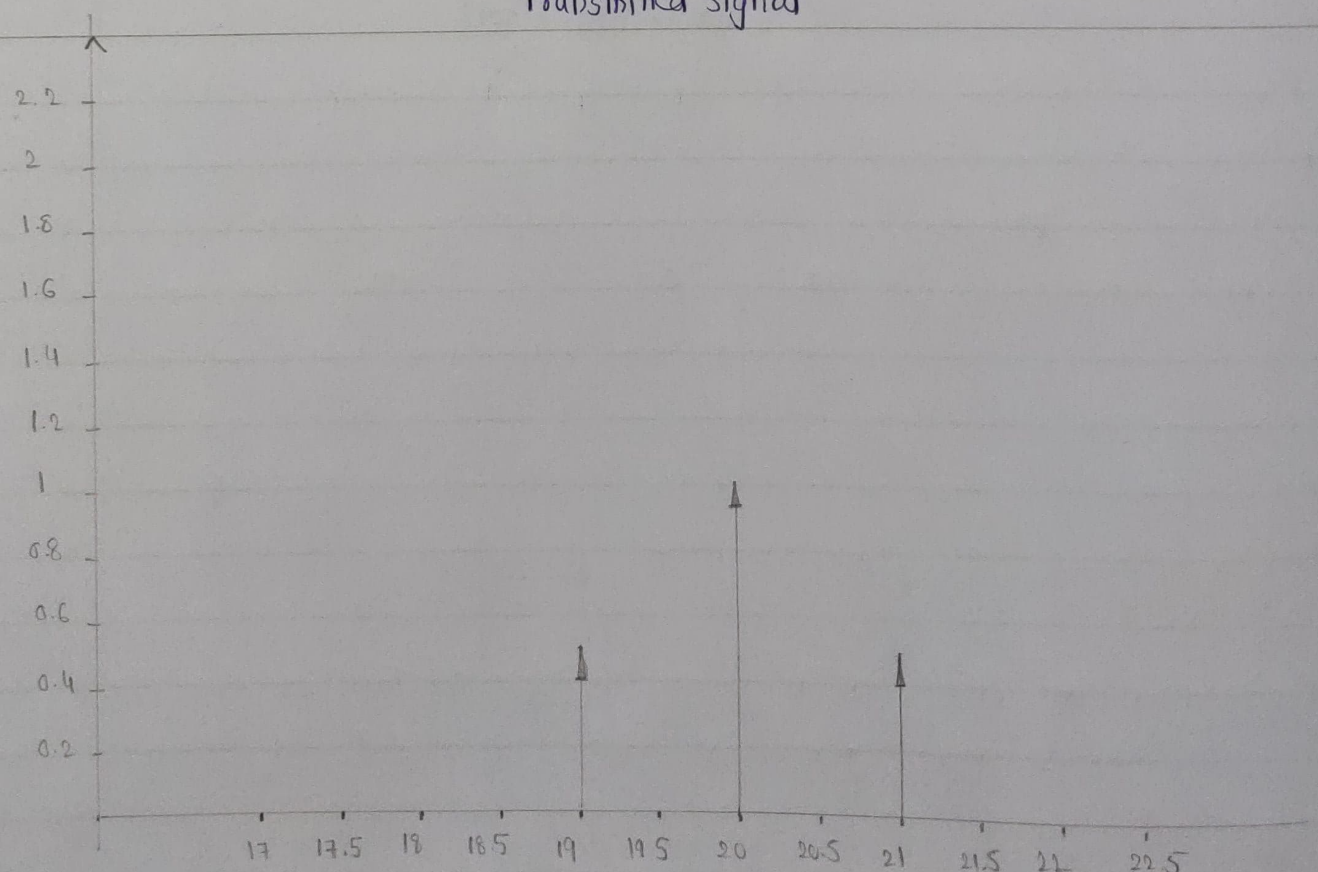
c) $m > 1$

| | | | |
|---|---|---|---|
| message | $A_m = 2V$ | $F_m = 1$ MHz | (cosine) |
| carrier | $A_c = 1V$ | $F_c = 20$ MHz | (cosine) |
| $\hat{S}$ | $A = 2V$ | $F = 1$ MHz | (DC on ~~cosine~~) |

$$\mu = \frac{A_m}{A_c} = \frac{2}{1} = 2$$




Transmitted Signal




Spectrum

# DOUBLE SIDEBAND SUPPRESSED CARRIER [DC OFFSET OFF]

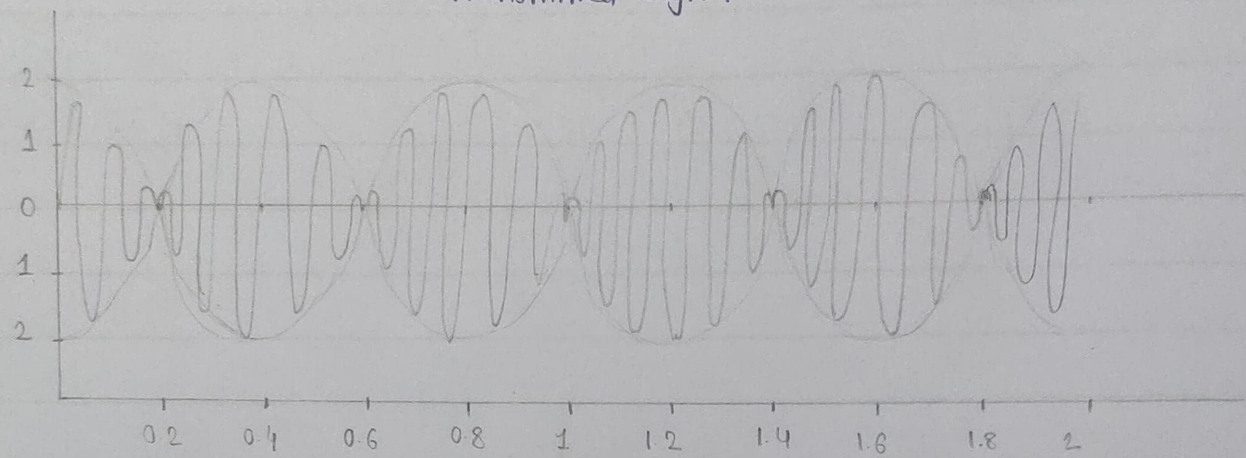
a) m<1

Message $A_m = 1V$ $F_m = 2MHz$ (cosine)

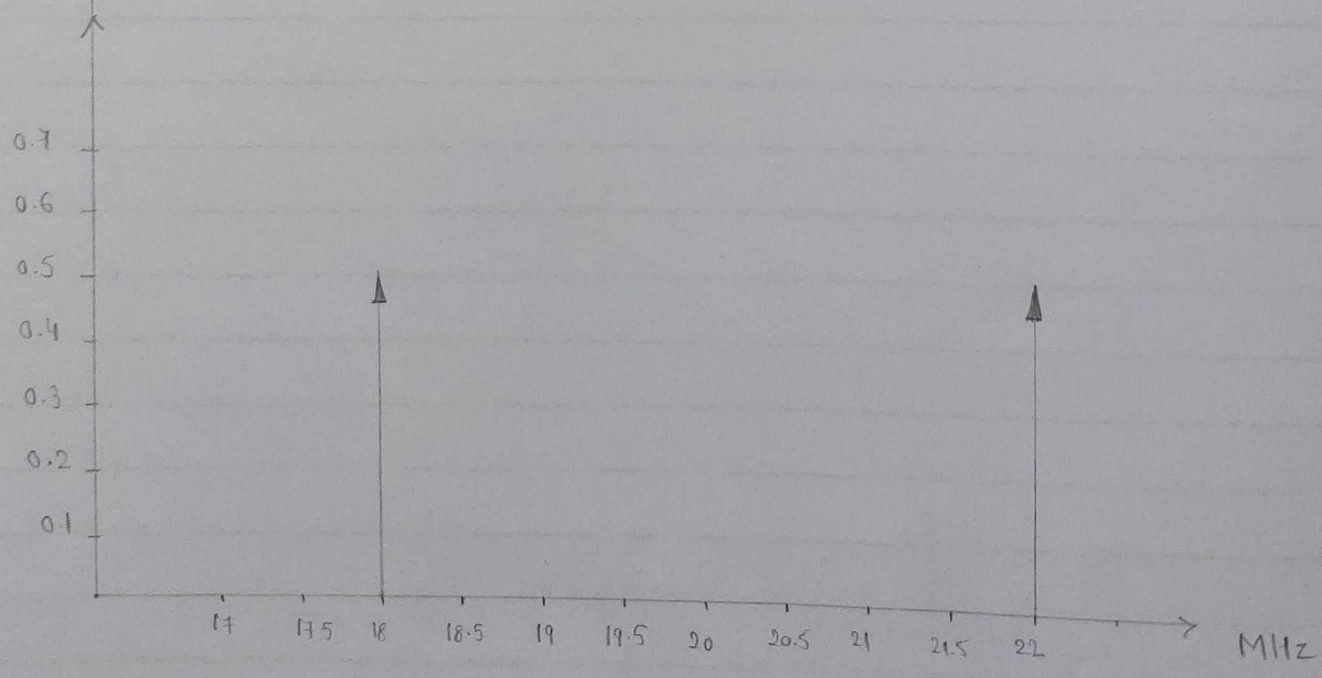
Carrier $A_c = 2V$ $F_c = 20 MHz$ (cosine)

$\mu = \frac{A_m}{A_c} = 0.5$

Transmitted signal




Spectrum

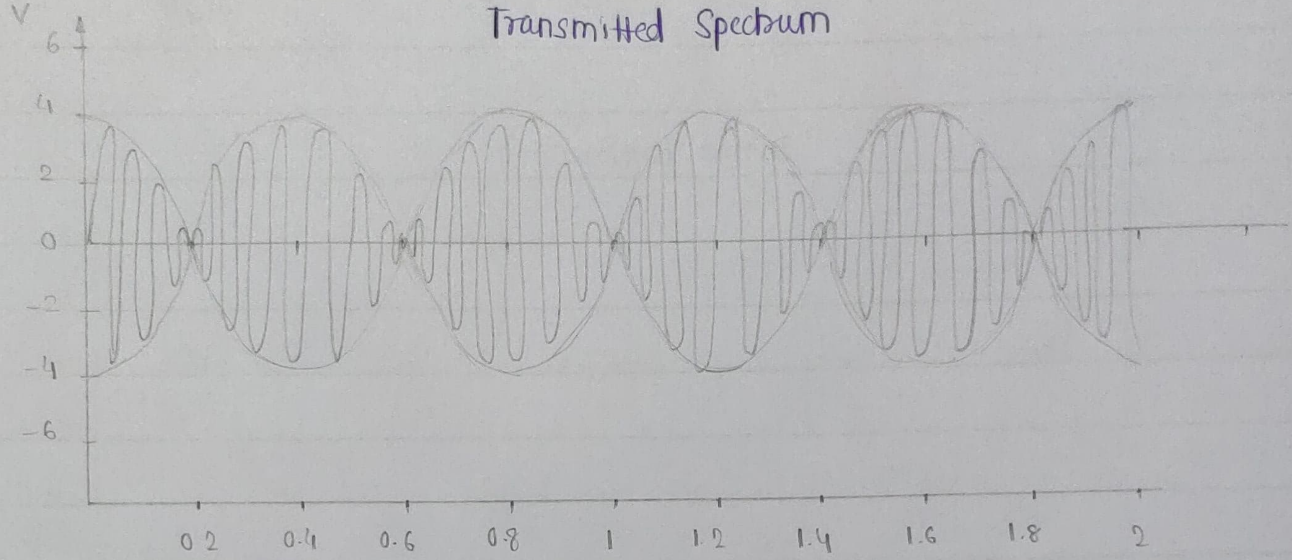


b) m = 1 Message $A_m = 2V$ $F_m = 2\,MHz$ (cosine)

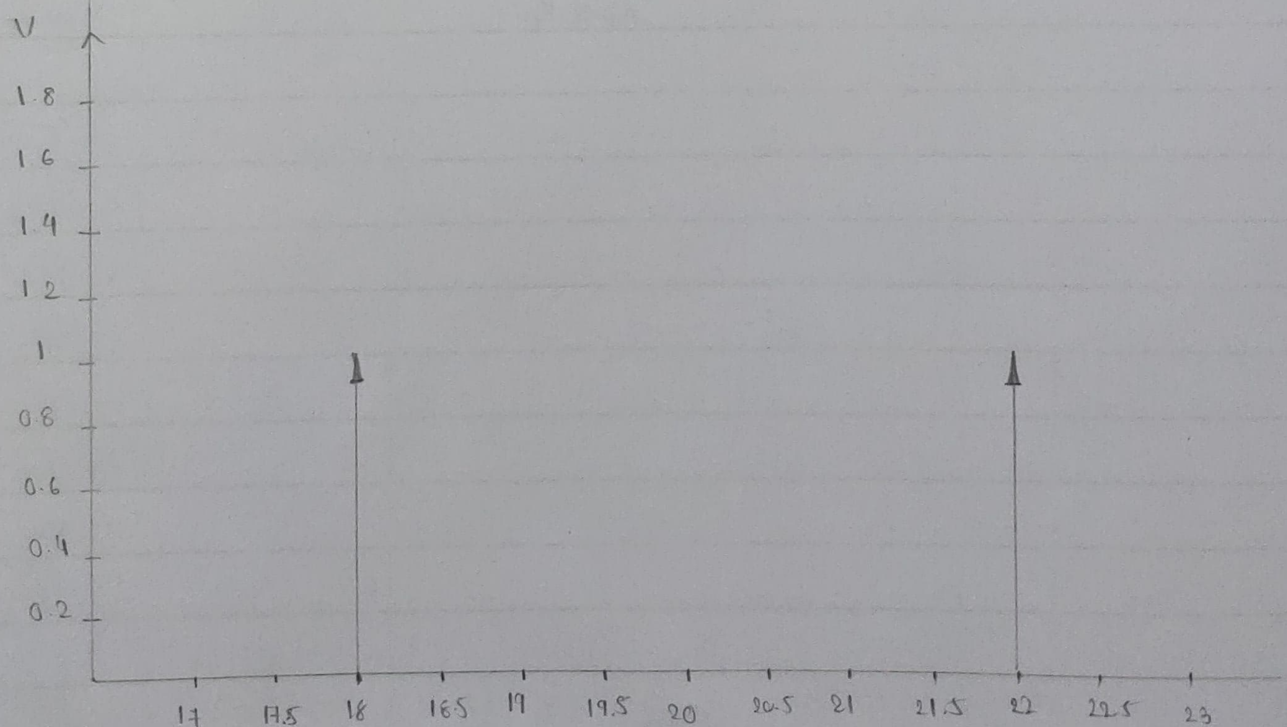
Carrier $A_c = 2V$ $F_c = 20\,MHz$ (cosine)

$$\mu = \frac{A_m}{A_c} = \frac{2}{2} = 1$$

Transmitted Spectrum



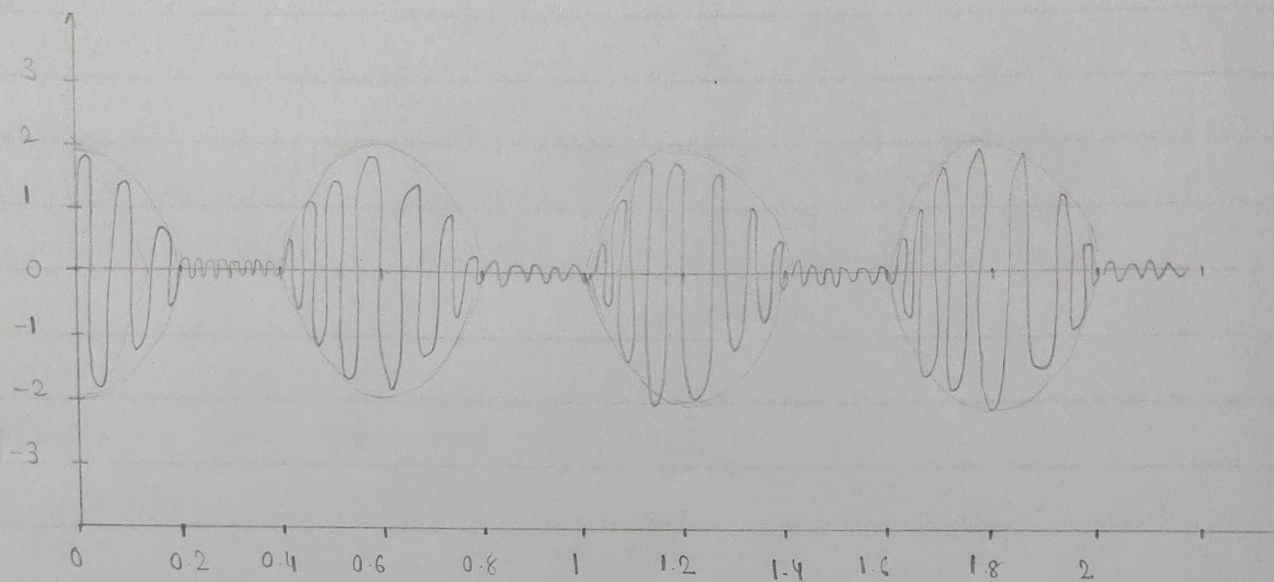

Spectrum



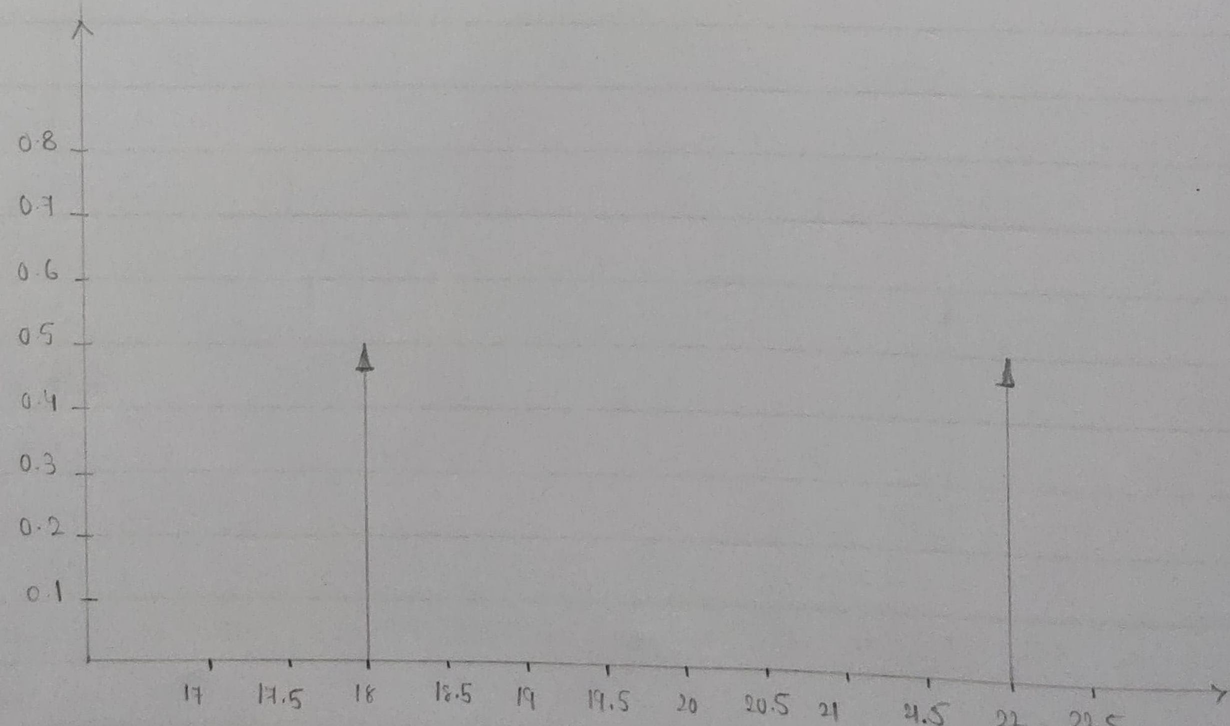
c) $m > 1$   Message   $A_m = 2V$   $F_m = 2\ kHz$ (cosine)

Carrier   $A_c = 1V$   $F_c = 20\ kHz$ (cosine)

$$\mu = \frac{A_m}{A_c} = \frac{2}{1} = 2$$




Transmitted signal




Spectrum

4.) For the different value of m observe the transmitted signal using oscilloscope and spectrum analyzer.

## OBSERVATION:

Double Side Band with Carrier

| Sr. No. | Am (V) Message | Fm (MHz) Message | Ac (V) carrier | Fc (MHz) Carrier | A (V) $\hat{\delta}$ | F (MHz) $\hat{\delta}$ | $\mu$ |
|---|---|---|---|---|---|---|---|
| 1 | 1 | 1 | 2 | 20 | 2 | 1 | 0.5 |
| 2 | 2 | 1 | 2 | 20 | 2 | 1 | 1 |
| 3 | 2 | 1 | 1 | 20 | 2 | 1 | 1 |

Double sideband with suppressed carrier

| Sr No. | Message | | Carrier | | Modulation Index ($\mu$) |
|---|---|---|---|---|---|
| | Am (V) | Fm (MHz) | Ac (V) | Fc (MHz) | |
| 1. | 1 | 2 | 2 | 20 | 0.5 |
| 2. | 2 | 2 | 2 | 20 | 1 |
| 3. | 2 | 2 | 1 | 20 | 2 |

Conclusion: We observe that using envelope detector we can detect double sideband with carrier but synchronous detector is needed for Double sideband suppressed carrier. We also observe that information lies in sidebands and in carrier. ∴ Using DSBSC, we can minimize power usage.

Teacher's Signature:

EXPERIMENT 4: [U19CS012]

# FREQUENCY MODULATION AND DEMODULATION

**AIM:** To study Frequency Modulation (F.M.) and Frequency demodulation with its Application

**APPARATUS REQUIRED:** LabAlive Software, MATLAB Software (Online Mode)

**THEORY:**

(1) Angle modulation is the process in which the frequency or phase of the carrier varies according to message signal.

(2) The standard equation of the Angle modulated wave is

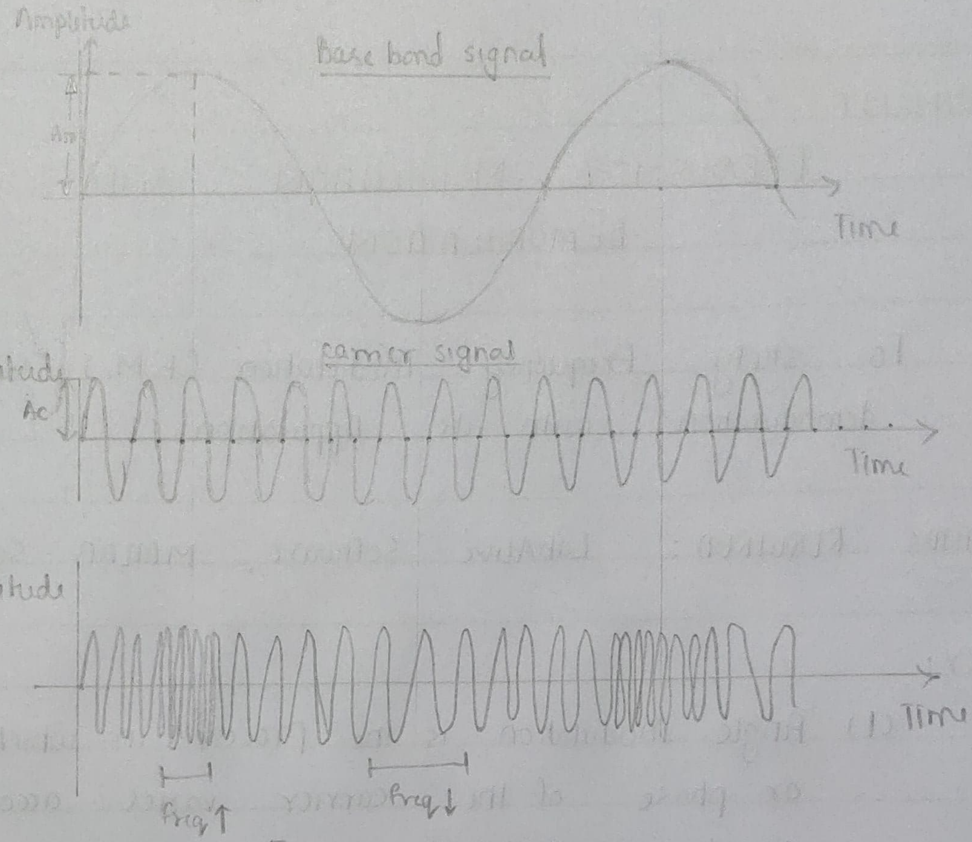
$$s(t) = A_c \cos(\theta_i(t))$$

where $A_c$ = Amplitude of the modulated wave / carrier signal
$\theta_i(t)$ = Angle of modulated wave.

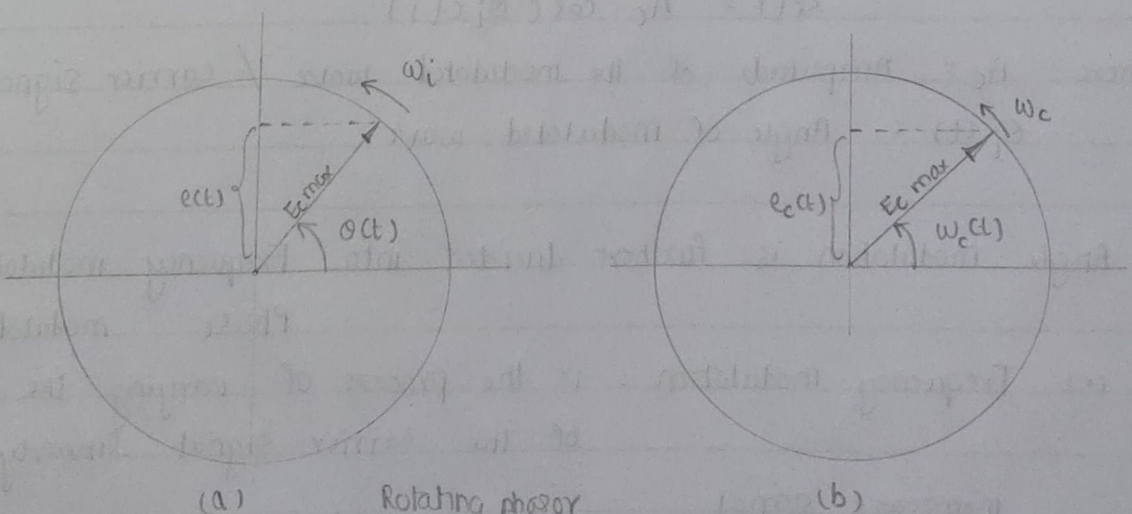
(3) Angle modulation is further divided into Frequency modulation and Phase modulation

(i) Frequency modulation: is the process of varying the frequency of the carrier signal linearly with message signal.

(ii) Phase modulation: is the process of varying the phase of carrier signal linearly with message signal.




Frequency Modulated wave




(a) Rotating phasor representation of carrier of amplitude $E_c$ max (b)

(a) Instantaneous angular velocity $\omega_i(t)$

(b) at constant angular velocity ($\omega_c$)

(4) As The frequency of modulated wave increases, when the Amplitude of the modulating or message signal increases.
Similarly, the frequency of modulated wave decreases, when the amplitude of the modulating signal decreases.

Note: The frequency of modulated (carrier) wave remains constant and is equal to frequency of carrier signal, when Amplitude of modulating signal is zero.

(5) Mathematically,
The equation for instantaneous frequency $f_i$ in FM modulation

$$\boxed{f_i = f_c + (K_f)\, m(t)} \quad \text{--- (1)}$$

$m(t)$ → message signal
$K_f$ → frequency sensitivity
$f_c$ → carrier frequency

(6) We know relationship between $w_i$ and $\theta_i(t)$.

$$\left[ w_i = \frac{d(\theta_i)}{dt} \right] \quad \text{--- (2)}$$

$$2\pi f_i = \frac{d(\theta_i)}{dt}$$

$$\theta_i(t) = 2\pi \int (f_i)\, dt$$

Substitute $f_i$ from eqn (1)

$$\theta_i(t) = 2\pi \int (f_c + k_f\, m(t))\, dt$$

$$\boxed{\theta_i(t) = 2\pi f_c t + 2\pi k_f \int m(t)\, dt} \quad \text{--- (3)}$$

Substitute $\theta_i(t)$ value in standard eqn of Angle Modulated wave;

$$\boxed{s(t) = A_c \cos\left(2\pi f_c t + 2\pi k_f \int m(t)\, dt\right)} \quad \text{(Eqn of FM wave) (4)}$$

(7) Finally, Equation of F.M. wave

$$s(t) = A_c \cos\left(2\pi f_c t + 2\pi k_f \int m(t)\,dt\right) \quad \text{--- (4)}$$

If modulating signal $m(t) = A_m \cos(2\pi f_m t)$, then eq$^n$ of F.M.

$$\boxed{s(t) = A_c \cos\left(2\pi f_c t + \beta \sin(2\pi f_m t)\right)} \quad \text{--- (5)}$$

$$\boxed{\beta = \frac{k_f A_m}{f_m} = \frac{\Delta f}{f_m} = \text{modulation index}}$$

(8) The difference between FM modulated frequency (instantaneous frequency) and normal carrier frequency is termed as Frequency Deviation. It is denoted by $[\Delta f = f_i - f_c = k_f A_m]$ and is equal to product of $k_f$ and $A_m$.

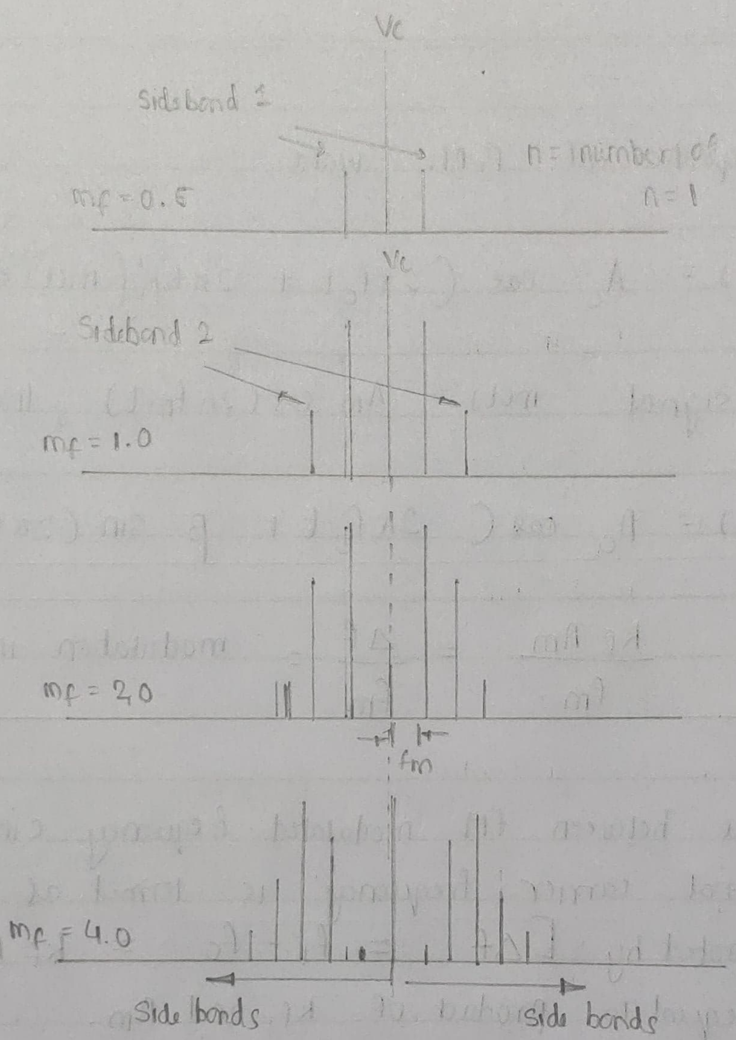
(9) FM can be divided into Narrowband F.M. and Wideband F.M. based on values of modulating index. ($\beta$)

(10) The amount of change in carrier frequency produced, by the amplitude of input modulating signal, is called frequency deviation.
carrier frequency swings between $f_{max}$ and $f_{min}$ as input varies.

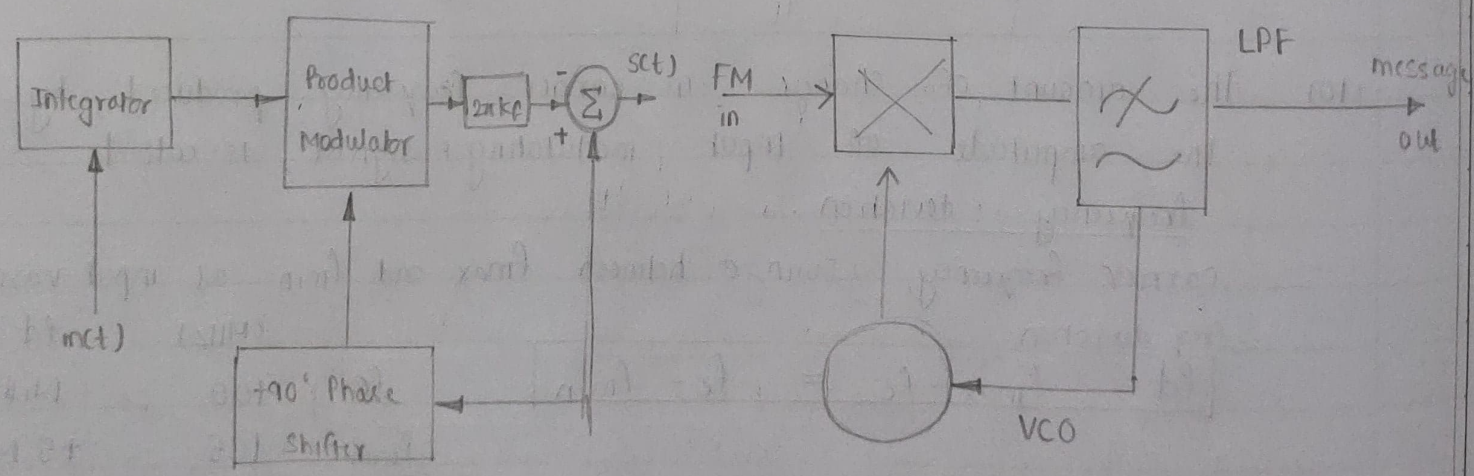
Freq. deviation

$$\boxed{f_d = f_{max} - f_c = f_c - f_{min}}$$

| | (MHz) | Carrier $f_d$ from |
|---|---|---|
| $f_c$ | 100 | Nil |
| $f_{max}$ | 105 | +5 MHz |
| $f_{min}$ | 95 | - 5 MHz |




How spectrum F.M. varies with mf



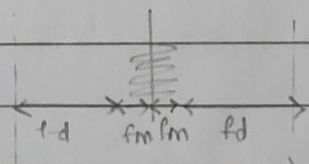

Block diagram of FM modulator and demodulator

(11) F.M. signal spectrum is quite complex and will have infinite number of side band as shown in figure.

This figure gives an idea, how the spectrum expands as the modulation index increases.

Side bands are seperated from carrier by $f_c \pm f_m$, $f_c \pm 2f_m$, $f_c \pm 3f_m$, and so on...

$$\boxed{\text{Bandwidth} = 2 * (f_m + \Delta f)}$$




(12) In F.M., carrier Amplitude is constant,
∴ Transmitted Power is constant.
& Transmitted Power does not depend on modulation index.

(13) F.M. has better noise immunity. FM is rugged / robust against noise ∴ The quality of FM will be good even in presence of noise.

(14) Applications & Advantages of F.M.

(A) FM is resilient to noise and interference. ∴ It is used for high quality broadcast transmission.

(B) FM is ideal for mobile radio communication application including more general two way radio communication or portable applications where signal levels are likely to vary considerably.

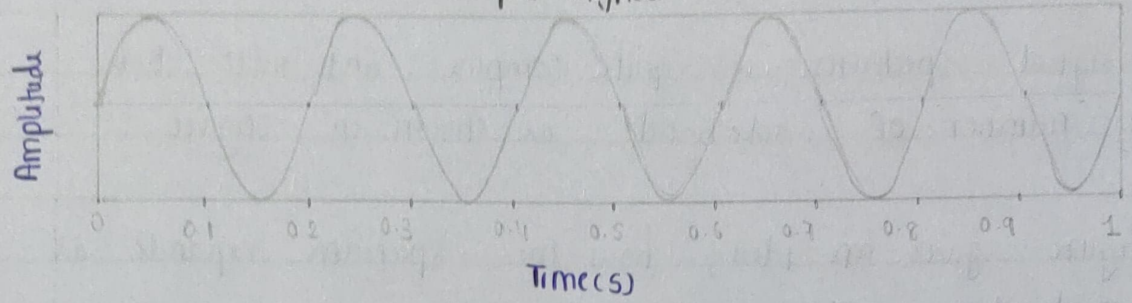
(C) Radar, Telemetry, observing infants for seizure through EEG, music synthesis (magnetic tape record system)

## MATLAB Code:

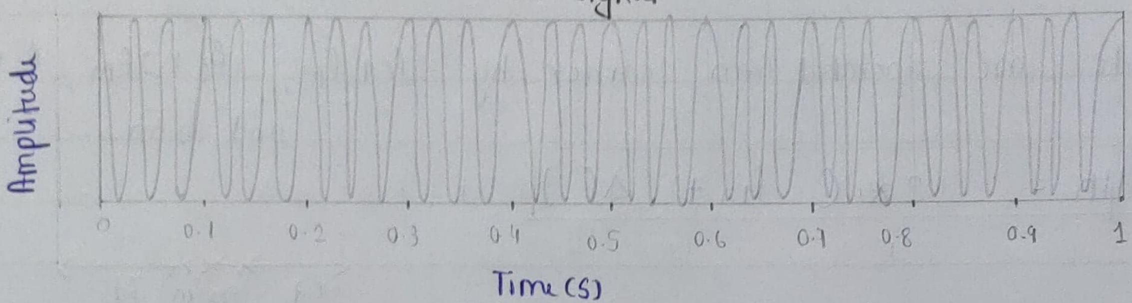
```
% Plot the frequency modulated signal
fc = 30 ;
fm = 5 ;
ts = 1/ (10 * fc) ;
fs = (1/ts);
fdev = 10;            % Frequency deviation
t = 0 : ts : 1;
m = sin( 2 * pi * fm * t) ;
c = cos ( 2 * pi * fc * t) ;
%  y = cos ( wc * t + (Kf * 2 * pi * cumsum(m) ).* ts ) ;
y = fmmod (m, fc, fs, fdev) ;
figure ;
subplot (5,1,1)
plot ( t, m)
title ('Input Signal')
xlabel ('Time (s)')
ylabel ('Ampitude')
subplot (5,1,2)
plot( t, c )
title ('Carrier signal')
xlabel (' Time(s)')
ylabel ('Ampitude')
subplot ( 5,1,3)
plot (t, y)
title('FM Modulation of input signal')
xlabel ('Time(s)')
yLabel ('Ampitude')
```

# MATLAB OUTPUT

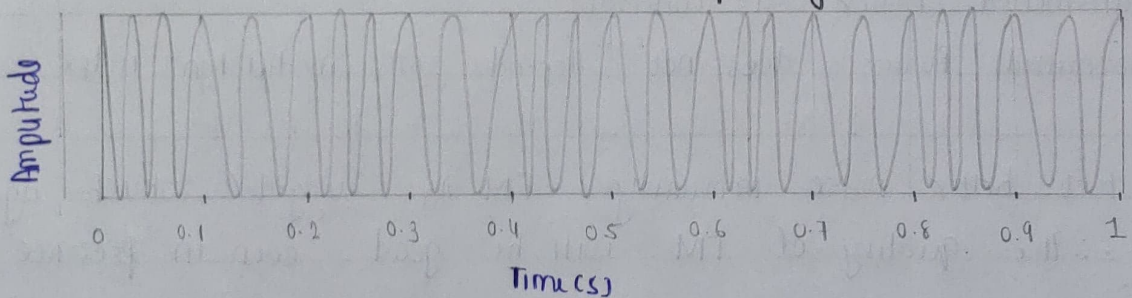
Input Signal



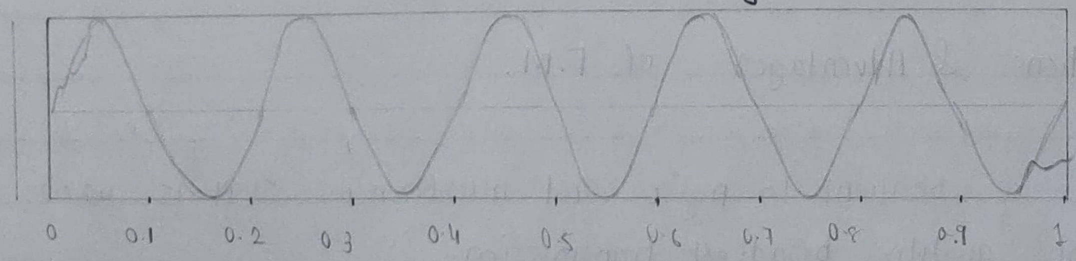

Carrier signal



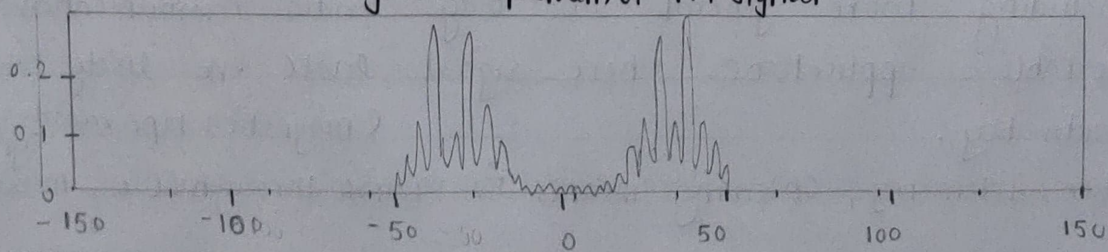

FM Modulation of Input signal




Demodulation of FM signal




Magnitude spectrum of FM Signal



[U195012]

```
% Demodulation
   z = fmdemod (y, fc, fs, fdev);
   subplot (5,1,4)
   plot (t,z)
   title ('Demodulated FM signal');

% Plot the frequency spectra
   a = fftshift (fft (y)) *ts;
   delta = fs/ length (a);
   f = -fs/2 : delta : fs/2 - delta;
   subplot (5,1,15)
   plot (f, abs (a))
   title ('Magnitude spectrum of FM signal');
```

CONCLUSION: We have successfully verified and understood the concept of Frequency modulation and demodulation using MATLAB and also learnt various applications of FM.

—x—

Teacher's Signature:

[U19CS012]

EXPERIMENT 5 :

PULSE AMPLITUDE MODULATION (P.A.M.)
PULSE POSITION MODULATION (P.P.M.)
PULSE WIDTH MODULATION (P.W.M.)

AIM : To examine pulse Amplitude modulation (PAM), Pulse Position modulation (PPM) and Pulse width modulation (PWM) and verify and draw the resultant waveform.

APPARATUS : MATLAB software online

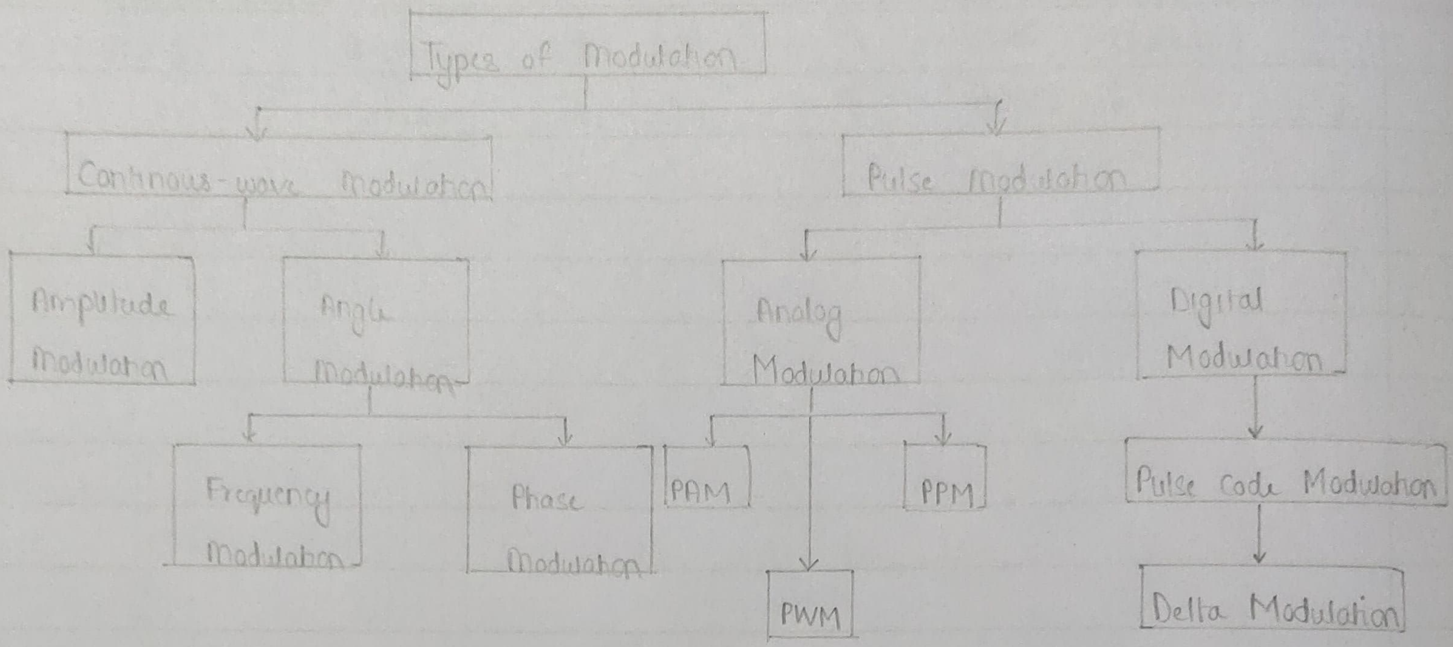
THEORY :

1.) Pulse modulation is a type of modulation in which the signal is transmitted in the form of pulses.

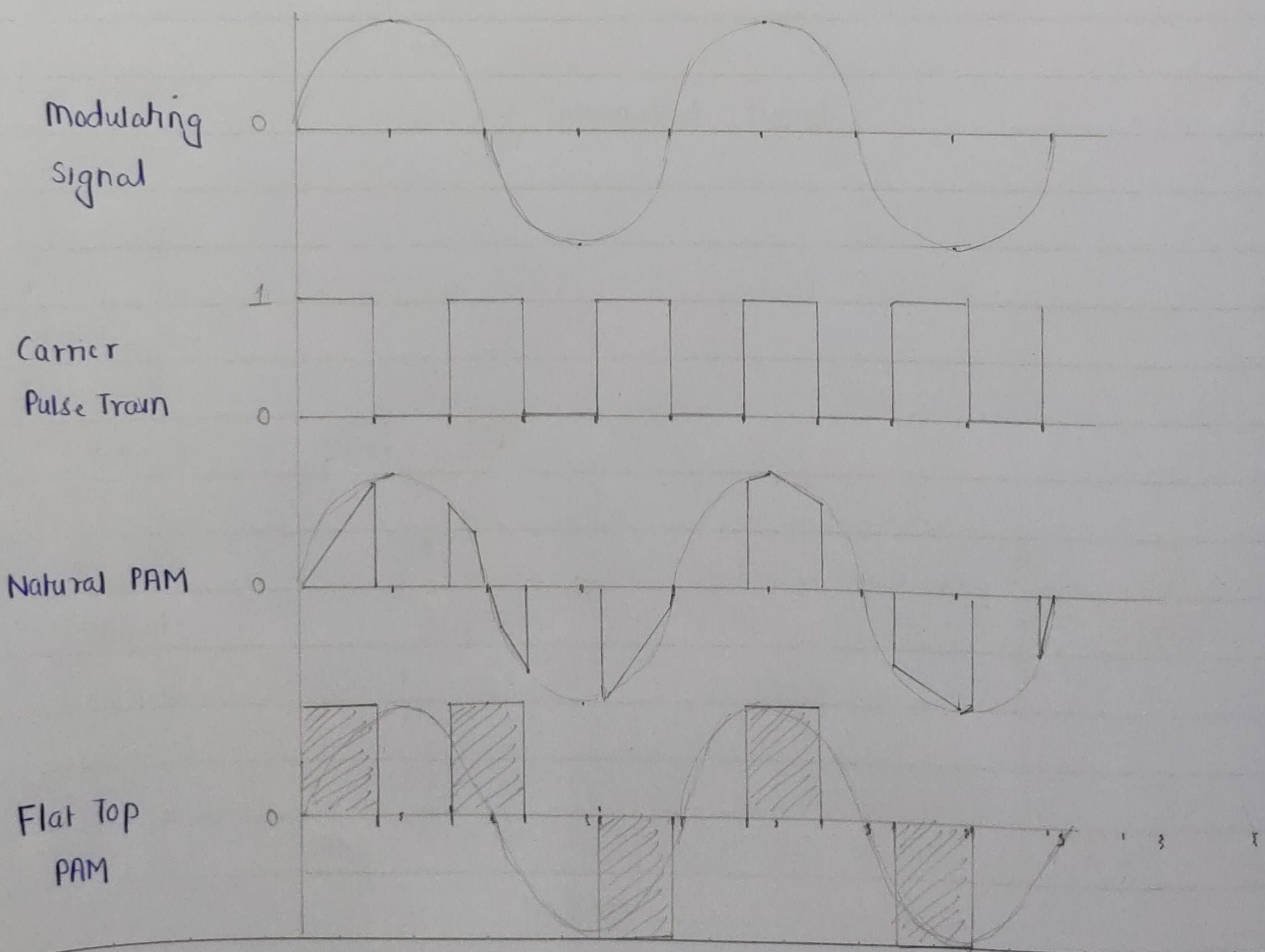
In Pulse modulation, continuos signals are sampled at regular intervals. Pulse modulation is further divided into Analog and Digital communication and further analog and digital modulation is subdivided in PAM, PWM, PPM (analog) and PCM, DM (digital).

2.) Pulse Amplitude Modulation (PAM) :

→ In PAM, a pulse signal is used to sample an analog signal. The result is a train of constant-width pulses. The amplitude of each pulse is proportional to the amplitude of the message signal at the time of sampling. The PAM signal follows the amplitude of the original signal, as the signal traces on the path of the whole wave.




## Natural Sampling and Flat Top Sampling



→ PAM signal generation: We can generate PAM signal by two types of sampling process.

Natural Sampling: For a PAM signal produced with natural sampling, the sampled signal follows the wave-form of the input signal during the time that each sample is taken.

Flat-top Sampling: In this type of sampling, a sample and hold circuit is used to hold the amplitude of each pulse at a constant level.

3.> Pulse Width Modulation (PWM)

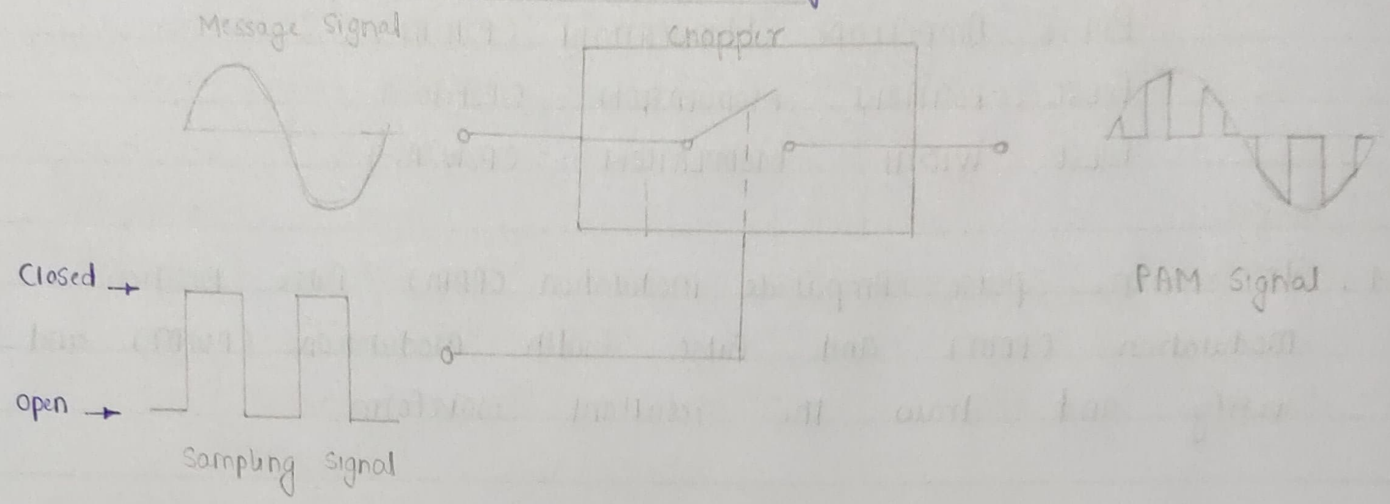
→ In this type, the amplitude is maintained constant but the duration or length or width of each pulse is varied in accordance with instantaneous value of analog signal.

4.> Pulse Position Modulation (PPM)

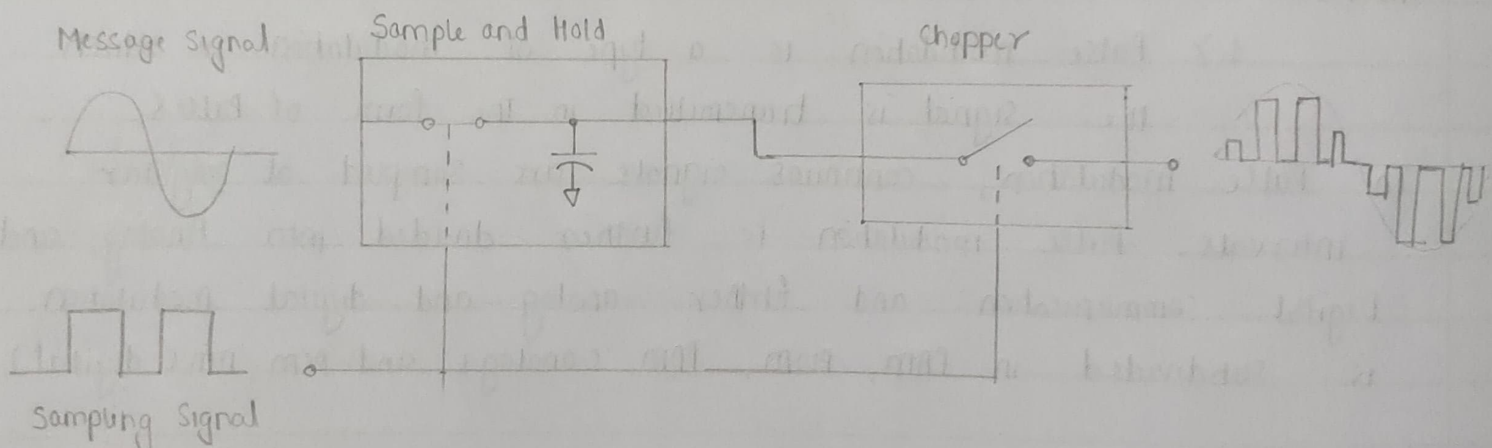
→ In this types of modulation, both the amplitude and width of the pulse are kept constant. we vary the position of each pulse according to the instantaneous sampled value of the message signal.

→ PPM is further modification of PWM

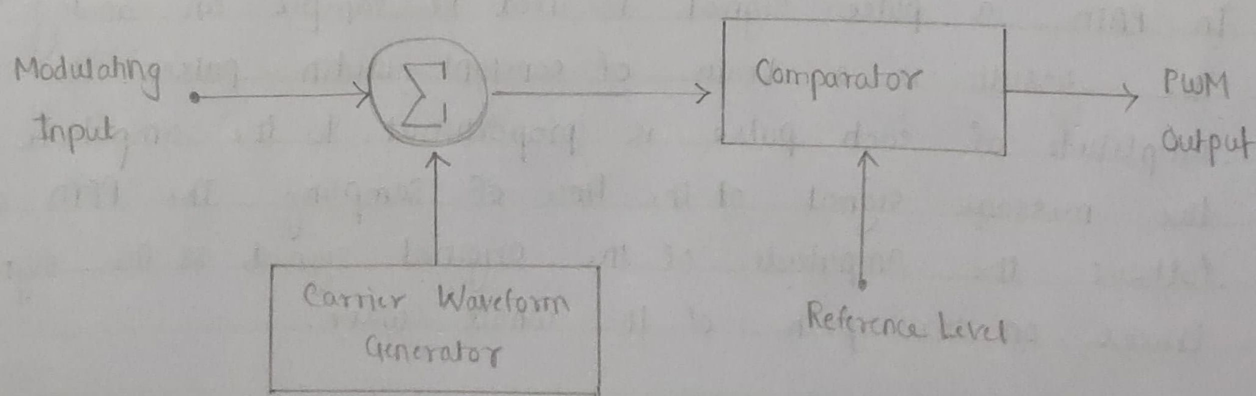
## Generation of PAM by Natural Sampling :




## Generation of PAM signal by Flat-top Sampung




## Generation of PWM signal

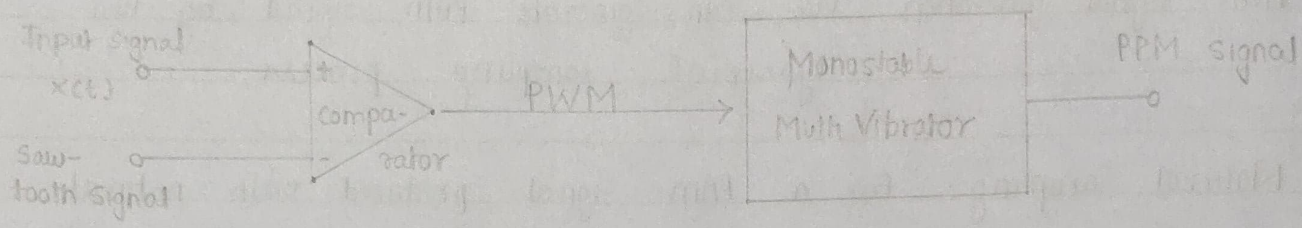


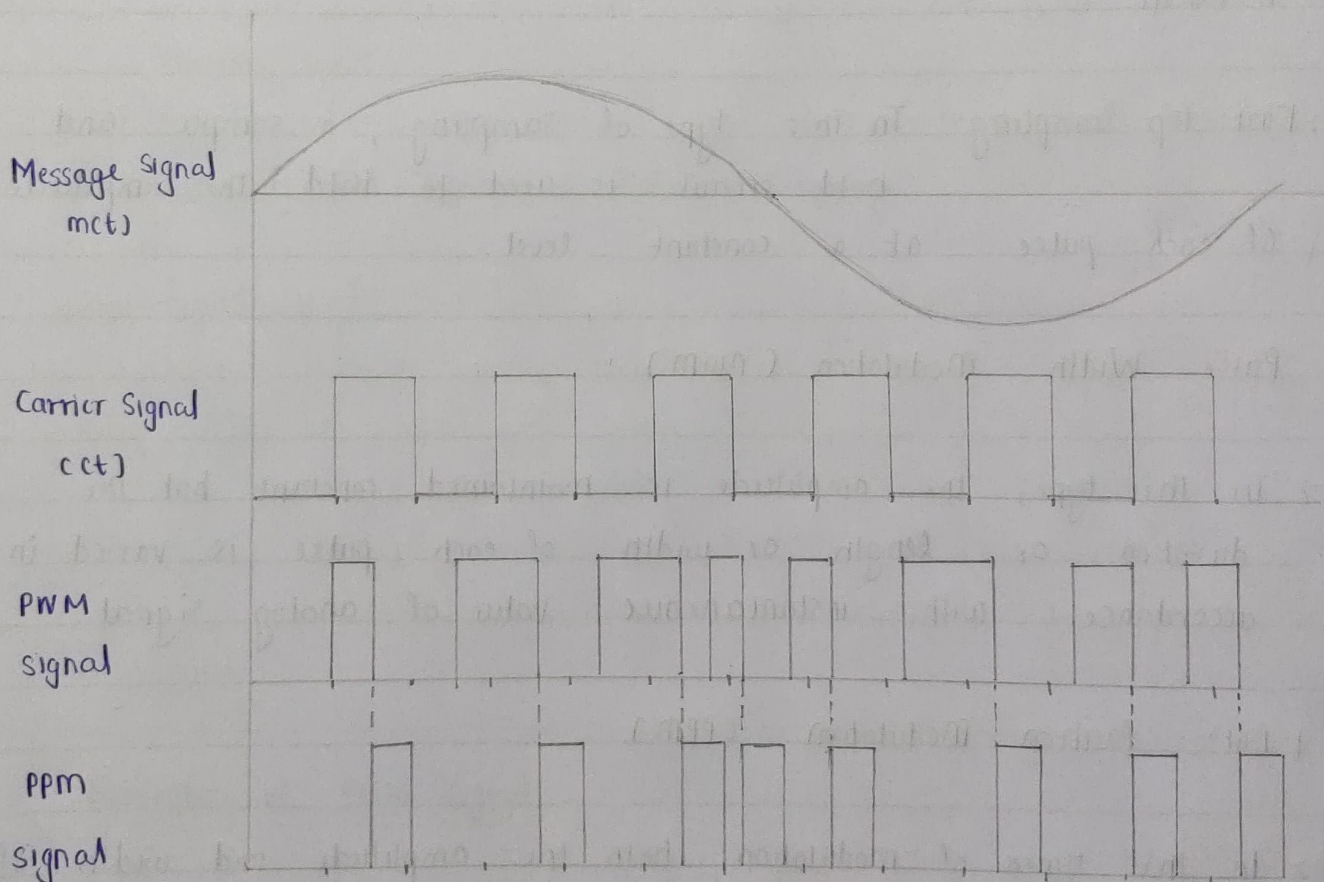
5.) Comparision of PAM, PWM and PPM :

| Sr No | Pulse Amplitude modulation (PAM) | Pulse Width modulation (PWM) | Pulse Position modulation (PPM) |
|---|---|---|---|
| 1.) | Amplitude of the pulse proportional to amplitude of modulating signal. | Width of the pulse is proportional to amplitude of modulating signal. | The relative position of the pulse is proportional to amplitude of modulating signal. |
| 2.) | Bandwidth of the transmission channel depends on the pulse width. | Here, it depends on the rise time of the pulse. | Here, it depends on Rising time of the Pulse. |
| 3.) | Instantaneous power of transmitter varies. | Instantaneous power of transmitter varies. | Instantaneous power of the transmitter is constant. |
| 4.) | Noise interference is high. | Noise interference is minimum. | Noise interference is minimum. |
| 5.) | System is complex to implement | System is simple to implement | System is simple to implement. |
| 6.) | Similar to Amplitude Mod. | Similar to Frequency mod. | Similar to Phase Mod. |

> CONCLUSION: We successfully examined Pulse Amplitude modulation, Pulse Position modulation, Pulse width modulation and also verified their waveforms. We also illustrated circuits for PAM and PWM. We performed our experiment successfully using MATLAB.

## Generation of PPM Signal



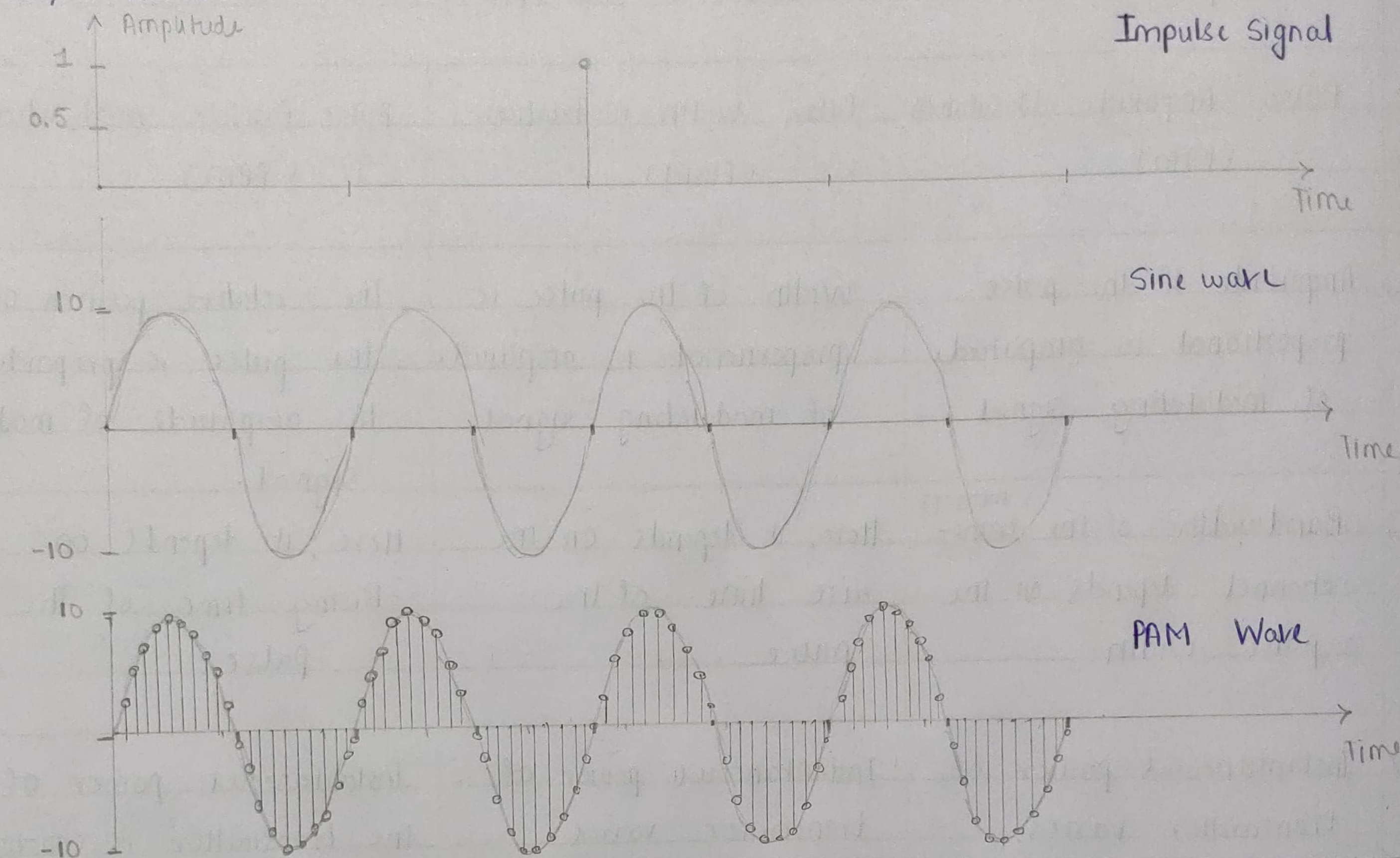

Waveform representation of PPM signal generation

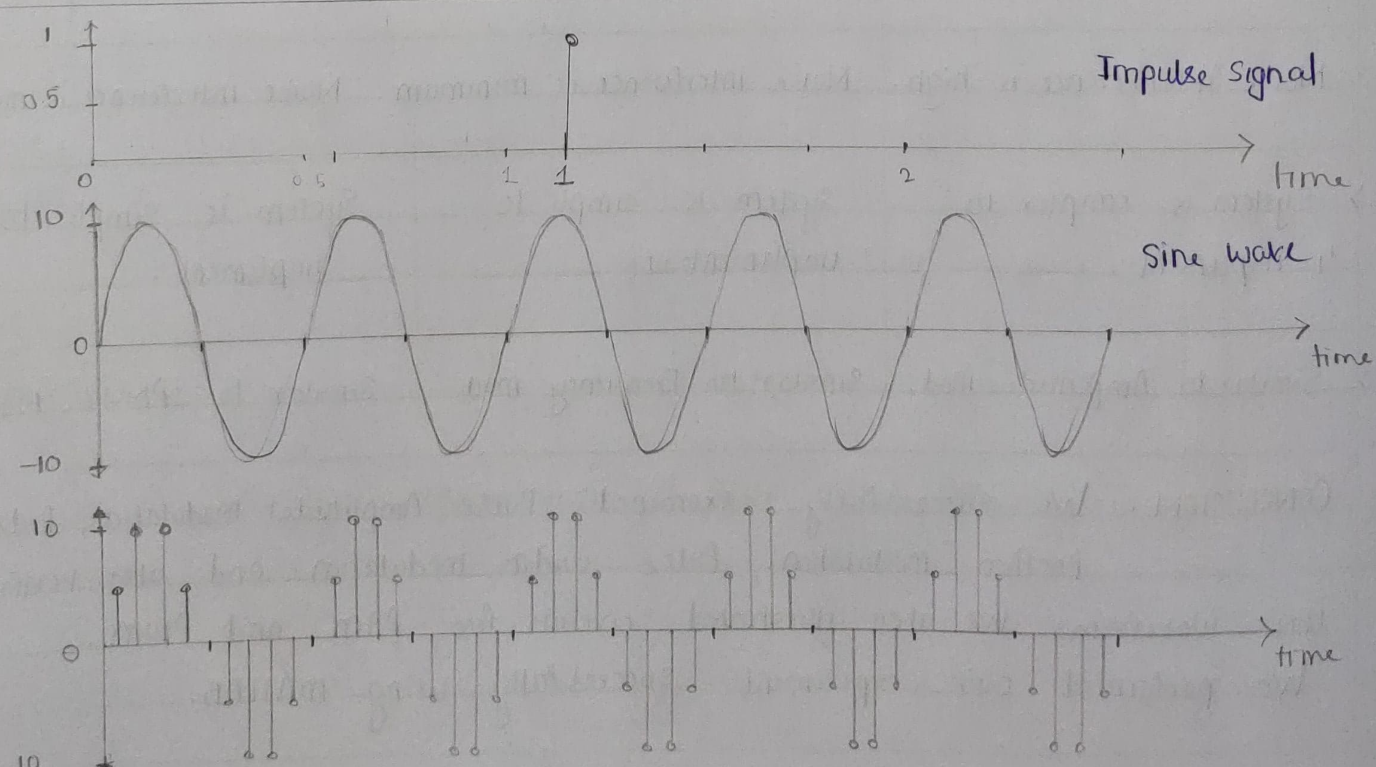
MATLAB CODE :

```
% PAM using Ideal Sampling
clc;
close all;
clear all;
a = input('Enter the amplitude = ');
f = input(' Enter the frequency = ');
t = 0 : 0.02 : 2 ;
x1 = 1;                          % generation of Impulse signal
x2 = a * sin( 2*pi*f*t) ;        % generation of sine wave
y = x1.*x2 ;                     % modulation step
subplot (3,1,1) ;                % for impulse signal plot
stem (x1) ;
title (' Impulse Signal');
xlabel (' Time');
ylabel ('Amplitude');
subplot (3,1,2);                 % for sine wave plot
plot (t, x2);
title ('Sine Wave');
xlabel (' Time ');
ylabel ('Amplitude');
subplot (3,1,3);                 % for PAM waveplot
stem (t,y);
title(' PAM wave');
xlabel ('Time');
ylabel (' Amplitude');
```

PAM with Ideal Sampling 

1.7 A = 1V f = 2 Hz




PAM

% PAM using Natural sampling

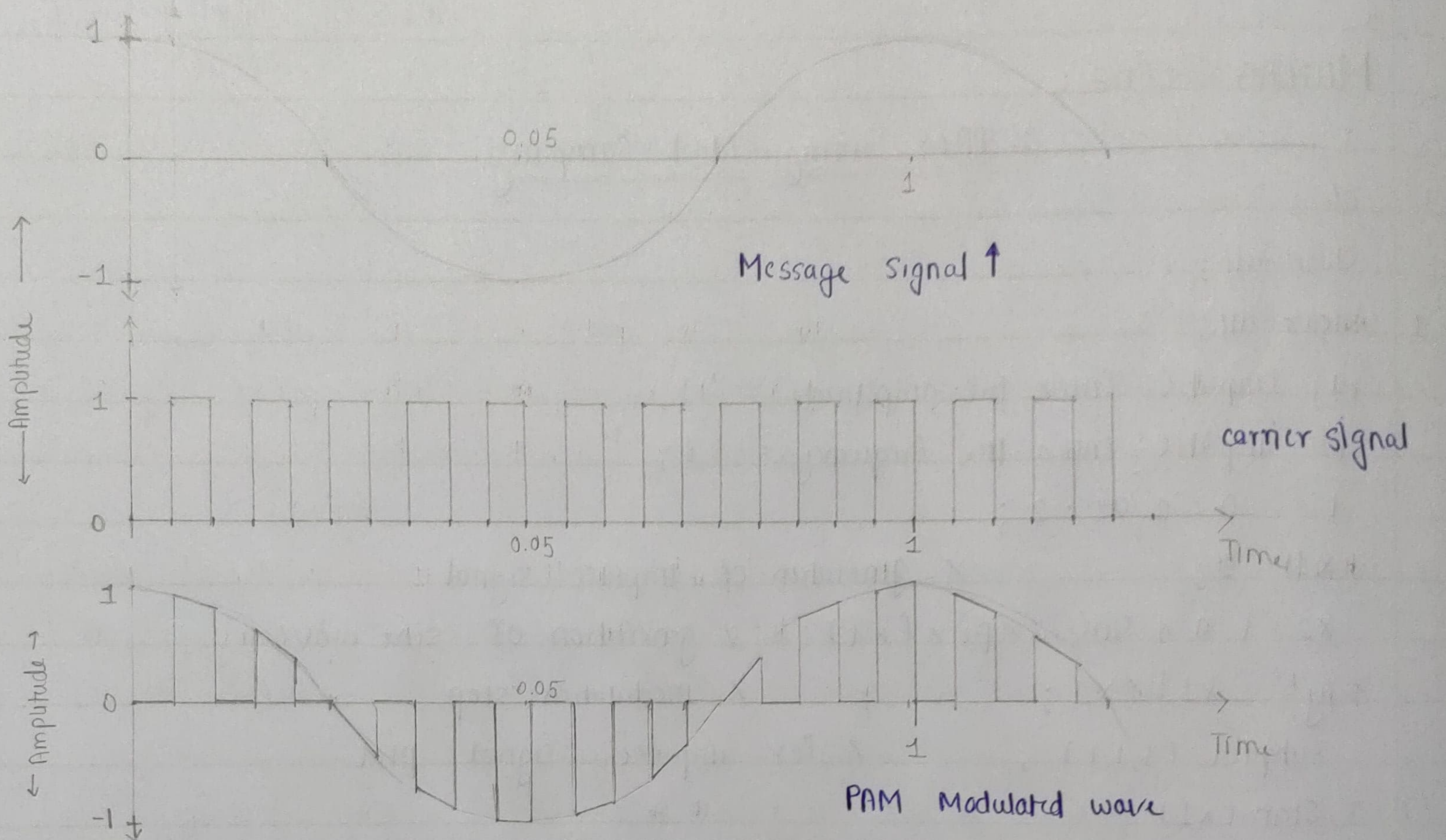
```
clc; clear all; close all;
fc = 100
fm = fc/10
fs = 100 * fc
t = 0: 1/fs : 4/fm ;
Msg_sgl = cos( 2*pi * fm *t);
Carr_sgl = 0.5 * square (2*pi* fc *t) + 0.5
Mod_sgl = Msg_sgl.* Carr_sgl ;
tt = [];
for i = 1: length( Mod_sgl) ;
    if Mod_sgl(i) == 0;
        tt = [tt, Mod_sgl(i)] ;
    else
        tt = [tt, Mod_sgl(i) + 2] ;
    end
end
figure(1)
```

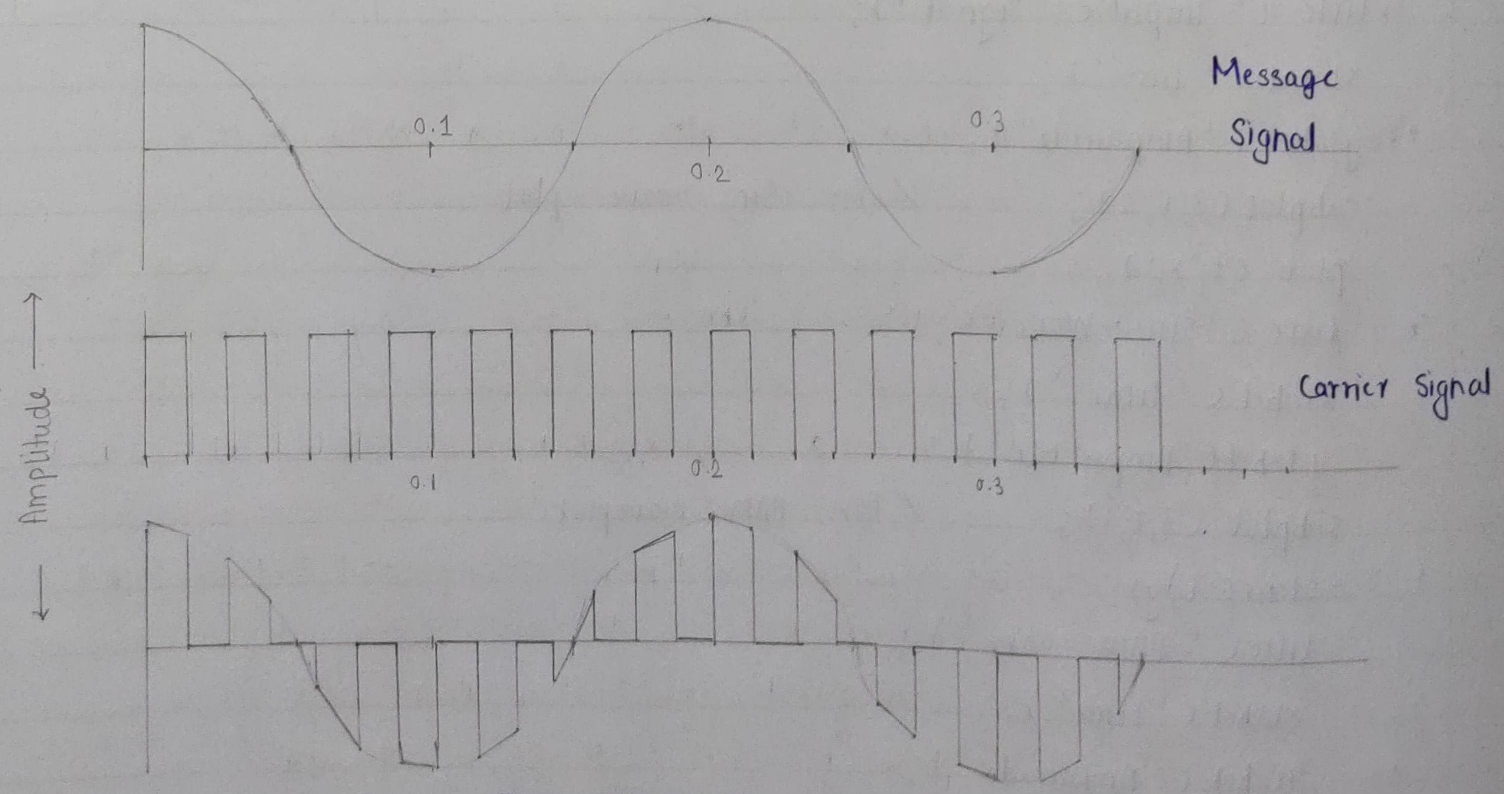
| Message | Carrier | PAM Modulated |
|---|---|---|
| subplot(4,1,1); | subplot (4,1,2); | subplot(4,1,3); |
| plot(t, Msg_sgl); | plot (t, Carr_sgl); | plot ( t, Mod_sgl ); |
| title('message Signal'); | title('Carrier Signal'); | title('PAM modulated signal') |
| xlabel('Time period'); | xlabel ('Time period'); | xlabel ('Time Period') |
| ylabel('Amplitude'); | ylabel ('Amplitude'); | ylabel ('Amplitude') |

# PAM Using Square Wave

(Natural Sampling) 

1> $F_c = 100$ $F_m = 10$ $F_s = 10,000$



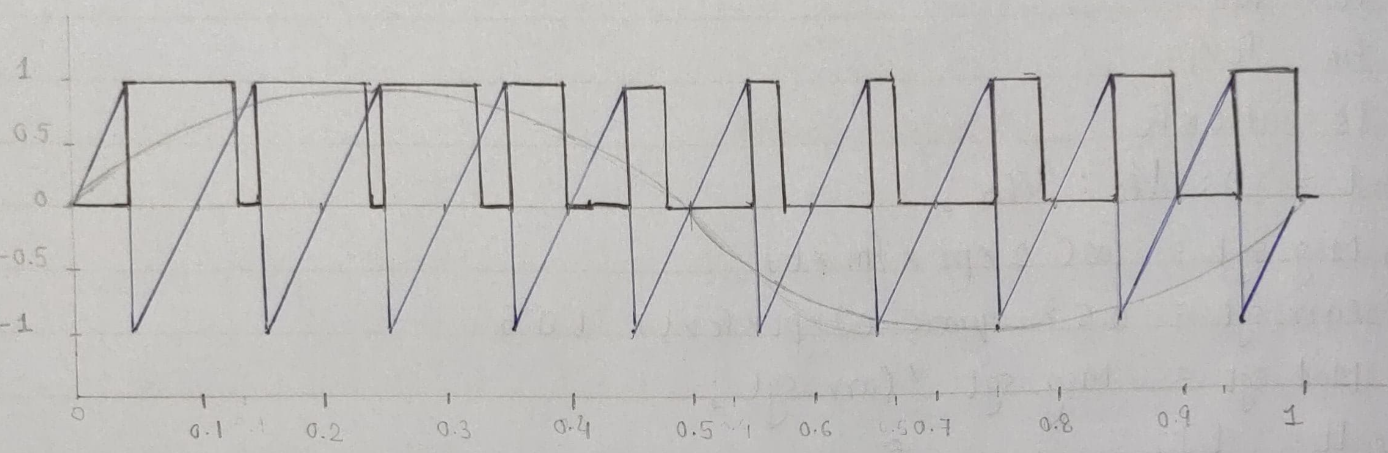

PAM Modulated wave

% PWM Signal

```
clc;
close all;
clear all;
t= 0: 0.0001 : 1;
s= sawtooth (2*pi* 10*t +pi);
m= 0.75 * sin(2*pi* 1*t);
n= length(s)
for i=1:n
    if ( m(i)>=s(i))
        pwm(i) =1;
    else if ( m(i)<=s(i))
        pwm(i) = 0;
    end
end
plot(t, pwm,'g', t,m,'r', t,s,'b');
ylabel('Amplitude');
axis ([0 1 -1.5 1.5]);
xlabel ('Time index');
title ('PWM wave');
grid on;
```

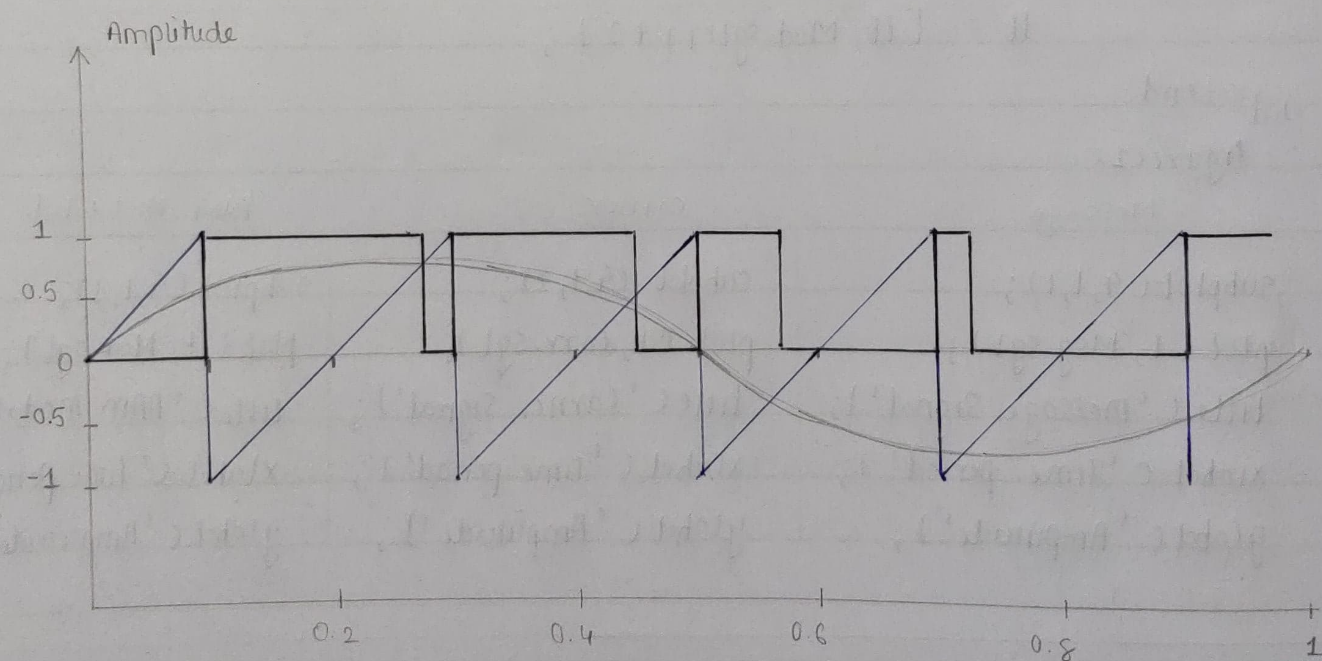
# PWM

1.) S = sawtooth (2*pi*10*t + pi);




— : Saw tooth — : PWM signal — : Message signal

2.) S = sawtooth (2*pi*5*t + pi);




— : saw tooth — : PWM signal — : Message signal

[U19CS012]

% PPM Signal

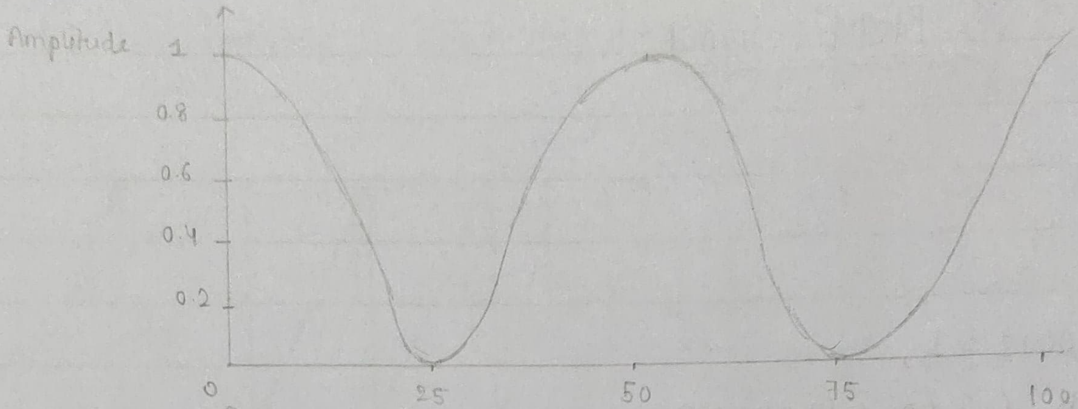
```
clc;
clear all;
close all;
fc = 10;
fs = 100;
fm = 2;
t = 0: 1/fs : ((2/fm) - (1/fs));
X = 0.5 * cos(2*pi*fm*t) + 0.5
Y = modulate(X, fc, fs, 'PPM');
subplot( 2,2,1);
plot(x);
title('msg signal')

subplot(2,2,2)
plot(Y);
axis([0 20 -0.2 1.2]);
title('PPM');
```

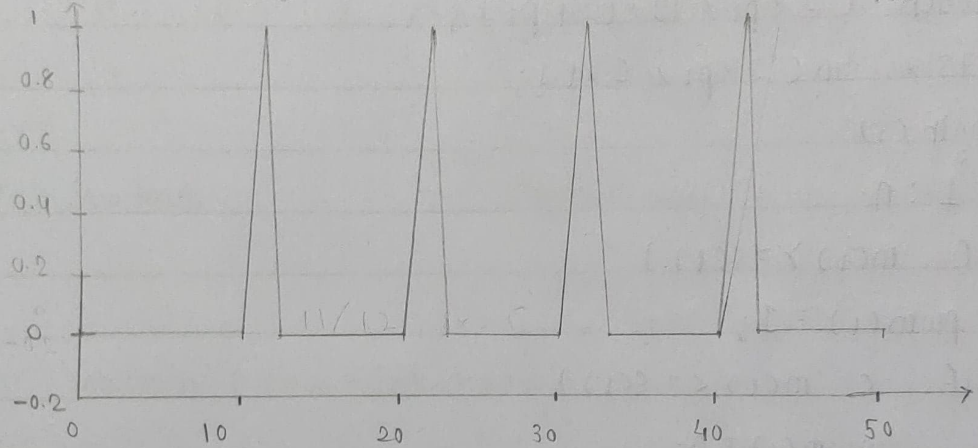
CONCLUSION: We have successfully examined Pulse Amplitude Modulation Pulse Position modulation, Pulse width Modulation and also verified their waveforms using MATLAB.

# PPM Signal

1.) $F_c = 10$ $F_s = 100$ $F_m = 2$

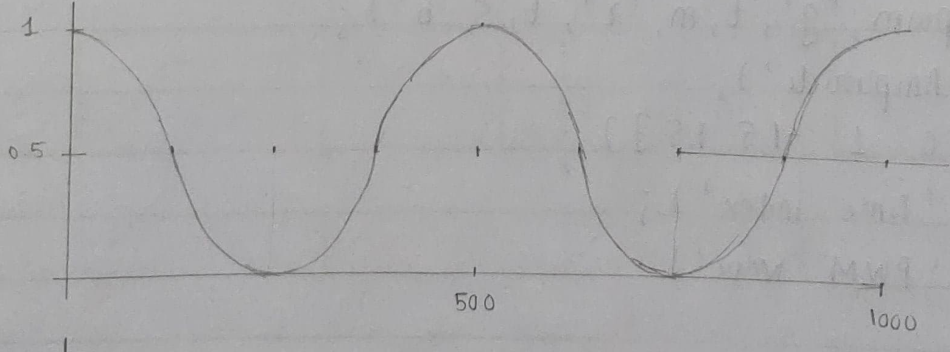



Msg Signal

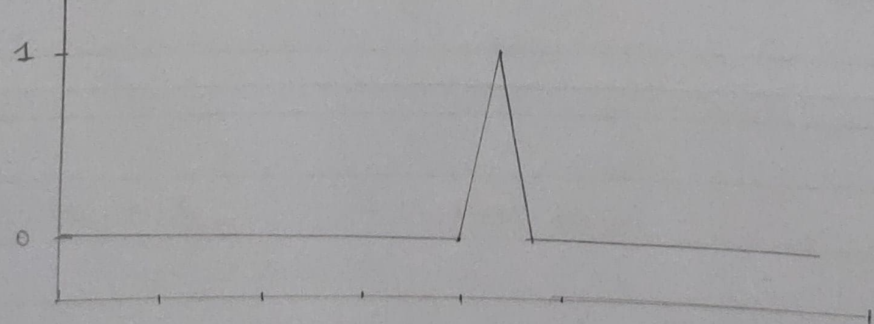



PPM

2.) $F_c = 40$ $F_m = 1000$ $F_m = 2$




Msg Signal




PPM