HAND WRITTEN
LABORATORY JOURNAL

Submitted in partial fulfillment of the requirement
For the Subject

"DIGITAL COMMUNICATION"
(EC 209)

: **Prepared & Submitted By** :

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B. TECH. II (CSE) 3rd Semester
(Academic Year : 2021-22)
ONLINE MODE

(July to Dec - 2021)

DEPARTMENT OF ELECTRONICS ENGINEERING
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Surat-395 007, Gujarat, INDIA.

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SUB : DIGITAL COMMUNICATION (EC209)

CERTIFICATE

This is to certify that the **Laboratory Journal** is prepared & submitted by **B. Tech. II (CSE-3rd Semester)** student **Mr. Aditya Raj** bearing **Admission No.U20CS0100** in the partial fulfillment of the requirement for the **Subject Digital Communication (EC209)** through ONLINE MODE.

The evaluation is done for the journal submitted herewith.

Laboratory Teachers :

Name	**Signature with date**

July -Dec. 2021.

DIGITAL COMMUNICATION (EC209)
(Academic Year 2021-22)

LIST OF EXPERIMENTS
ONLINE MODE

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EXPERIMENT NO - 01

SPECTRUM ANALYSER AND OBSERVE SPECTRUM

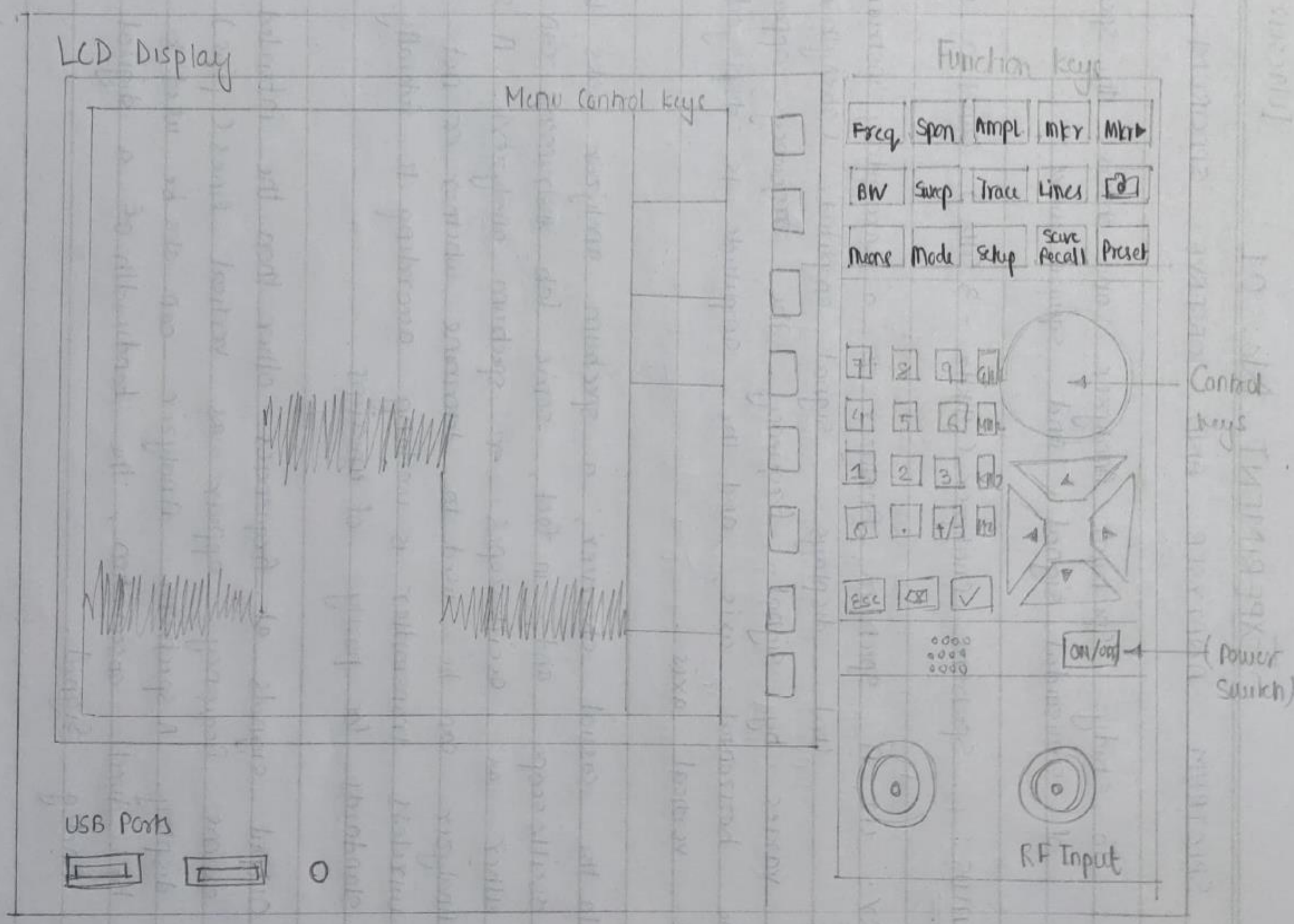
AIM : To study spectrum Analyser and observe the spectrum of sinusoidal signal and Square wave.

APPARATUS : Spectrum Analyser (9KHZ - 3GHZ)
Function Generator.

Theory : (i) A spectrum Analyser is a ~~temp~~ 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.

(ii) To the ~~Caste~~ Casual observer, a spectum analyzer looks like an oscilloscope and, in fact, same lab instruments can function. either as Oscilloscopes or spectrum analyzer. A spectrum Analyzer can be used to determine whatever or not a wireless transmitter is working according to federally defined standards for purity of emissions.

(iii) 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 anlog Signal.



Spectrum Analyzer

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(iv) 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 frequency.

FEATURES of LAB INSTRUMENT GSP-30 (GWINSTEK):

- 5 markers with delta markers & peak functions.
- 3 traces
- Sput windows with seperate settings.
- 6.9" TFT color LCD, 640 x 480 resolution.
- AC/DC/battery - multi-made power operator.
- Autoset
- 9KHZ - 3GHZ frequency range.

FREQUENCY SELECTION AND THEIR SELECTION METHOD

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.

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

Panel operation :

- Press frequency key.
- Press f4 (step)
- Enter the value using numerical and unit keys, arrow keys & scroll nobe..

2) RANGE : 9KHz 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 key & Scroll nobe.

5) View Signal (start & stop)

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

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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.

7) Set stop frequency:
Panel operation:

- Press frequency Key
- Press f3 (stop)
- Enter the value using numerical & unit keys, arrow keys & scroll nope.

8) Full or zero span :

Full or zero span setting set the span to extreme values: 3GHZ (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 : 3GHZ (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

- Zero span display can be obtained by pressing F3 Key.
- start frequency & stop frequency remains same as Center frequency.
- Note : last span setting can be recalled by. f4 Key.

AMPLITUDE SELECTION AND SETTING METHOD.

1) AMPLITUDE

Amplitude Key sets vertical attribote of the display, including the upper limit (reference level), vertical range (amplitude vertical unit and compensation for external gain for 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 f_1 (reference level)
- Enter the value using numerical and unit keys, arrow key 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.
dB μV : -3.01 to 126.99 dB μV, 0.01 dB resolution.

4) Select amplitude scale.

Panel operation :

- Press Amplitude key
- Press f_2 (scale dB/DIV)
- Repeatedly to select the selet

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 (dB μV)
- Press f6 (return) to go back to previous ~~m~~ menu .
- 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

Set external offset level.

5) Background

External offset componentes the amplitude gain or loss caused by an external network or devices.

Panel operation :

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 + dB , 0.1 dB resolution.

ICON : • They amplitude icon appears at the bottom of the display when the external offset changes.

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- To check whether spectrum analyzer working properly
- Generate Auxilary signal: Press System Key, press Auxilary signal, select an option from side given menu, following signal will generate. It generate 10 MHZ signal with 10 dB amplitude

Observation Table:
Spectrum Diagrams (wave forms)

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

Observation

Waveform: sine

S.NO	Frequency (KHz)	Amplitude (mV)
1	2	1
2	2.5	1.1
3	3	1.5
4	4	2
5	5	2.4

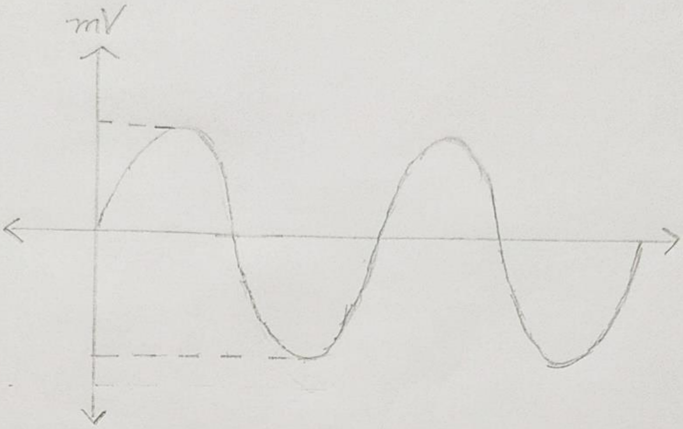
Waveform : square.

S.NO	frequency (KHz)	Amplitude (mV)
1	2	1
2	2.5	1.2
3	3	1.5
4	4	1.6
5	5	2

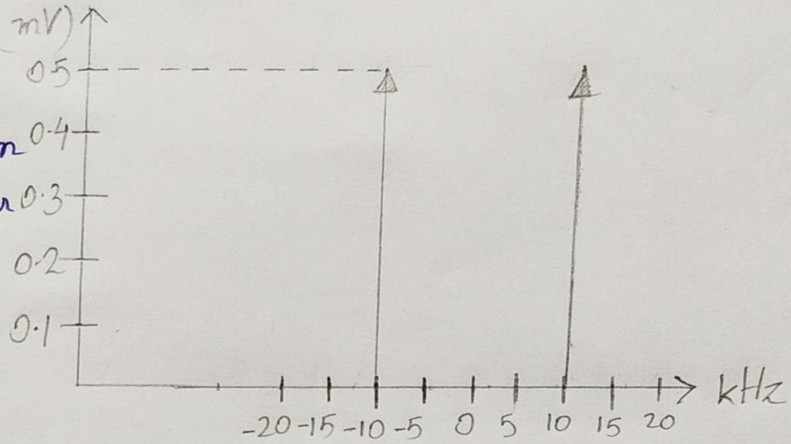
SINE WAVE

1.) $A = 1\,mV$ $f = 10\,kHz$

Oscilloscope

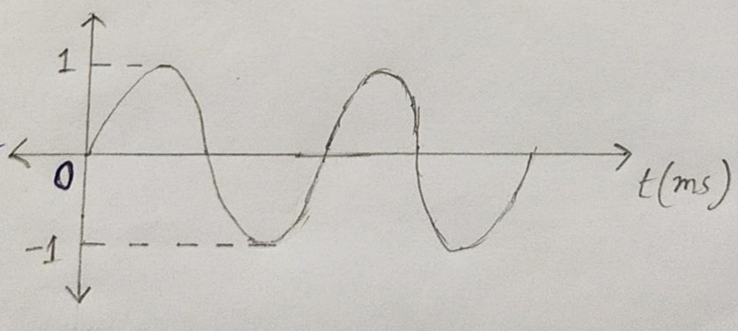


Spectrum Analyzer

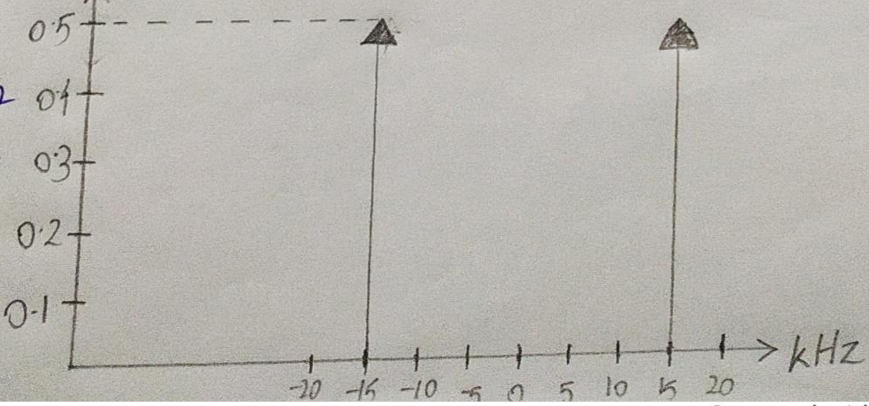


2.) $A = 1\,mV$ $f = 15\,kHz$

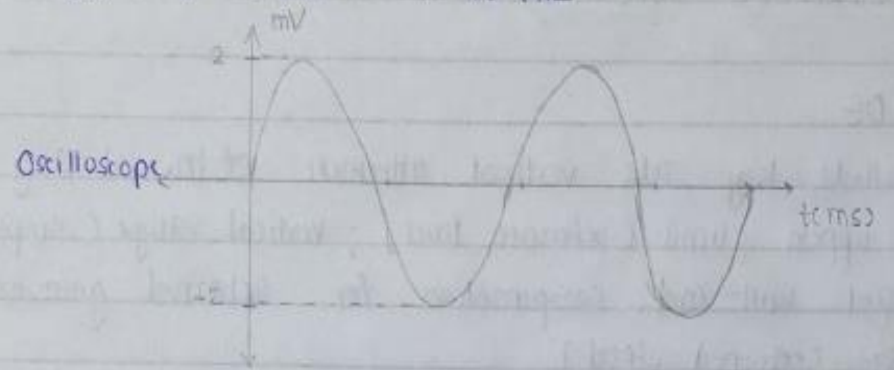
Oscilloscope

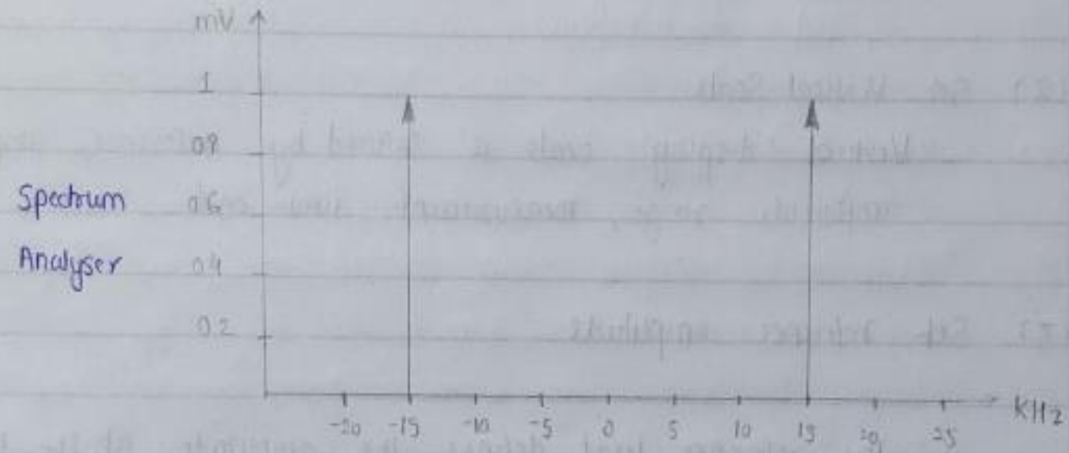


Spectrum Analyzer

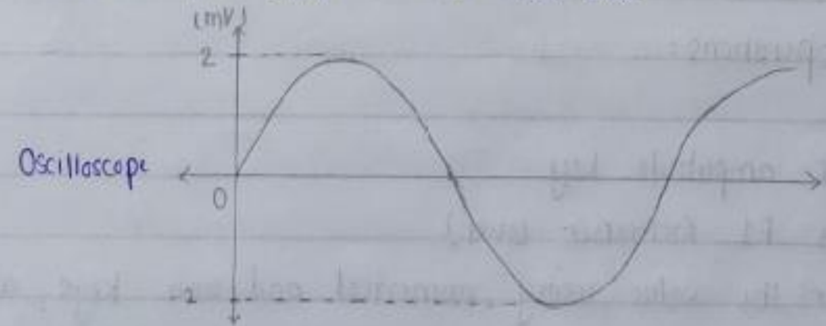


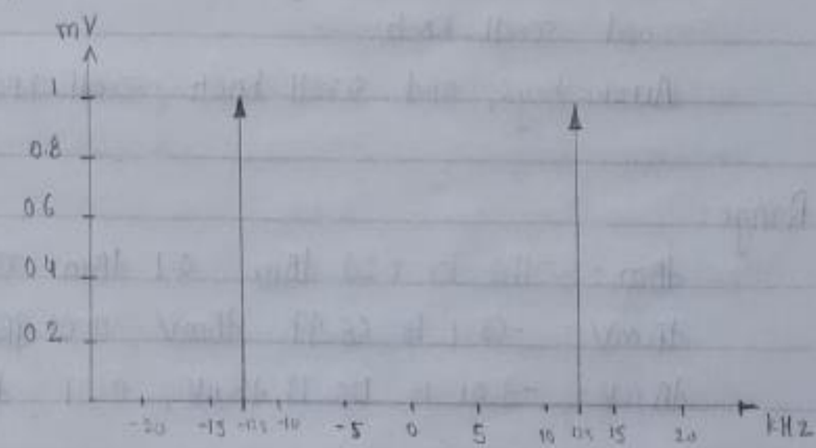
3.> A = 2mV f = 15 kHz





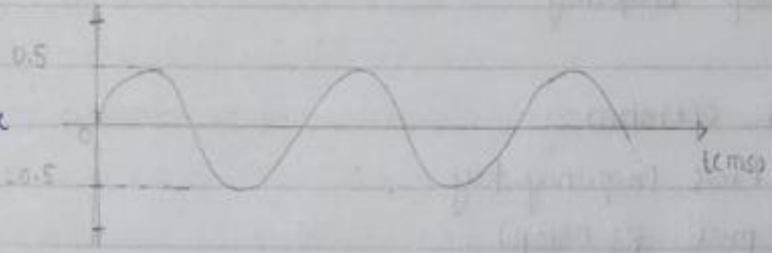
4.> A = 2mV f = 12.5 KHz



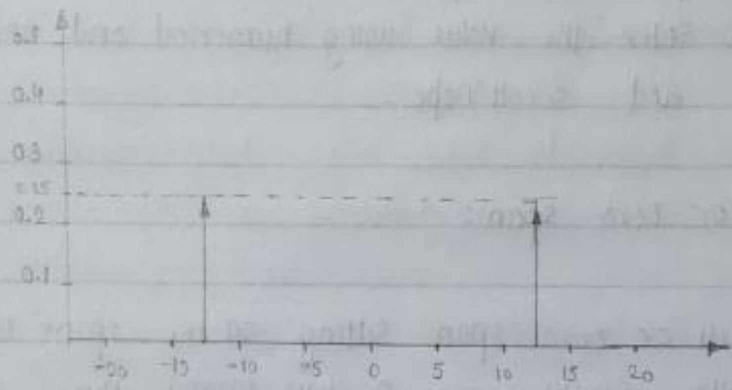


5.) $A = 0.5$ mV $f = 12.5$ kHz

Oscilloscope

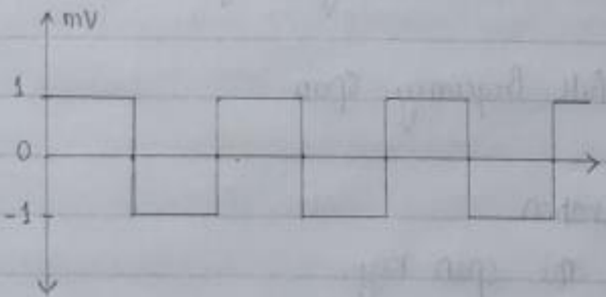


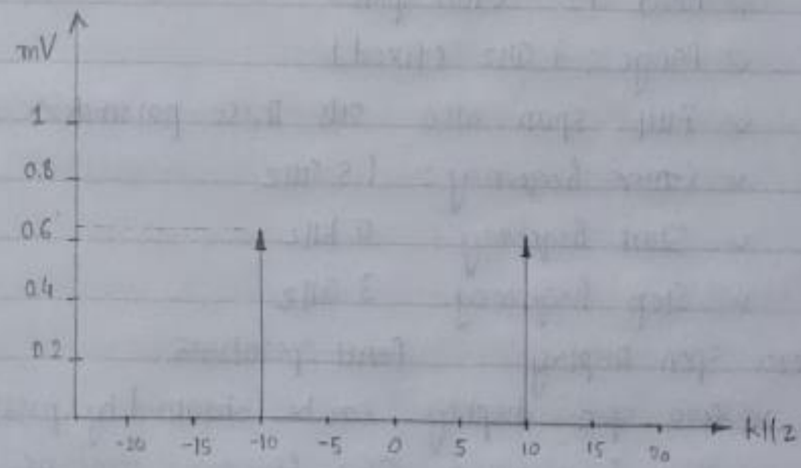
Spectrum Analyser



SQUARE WAVE

1.) $A = 1$ mV $f = 10$ kHz





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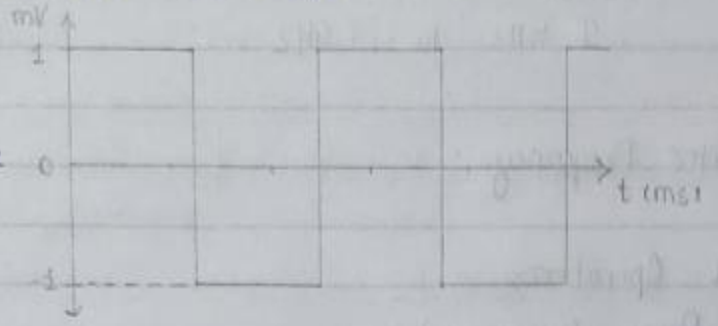
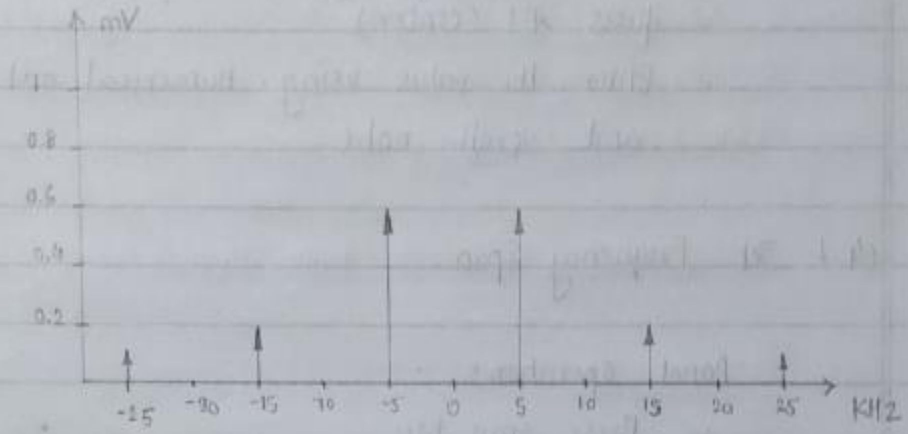
2.) A = 1 mV f = 5 kHz

Oscilloscope

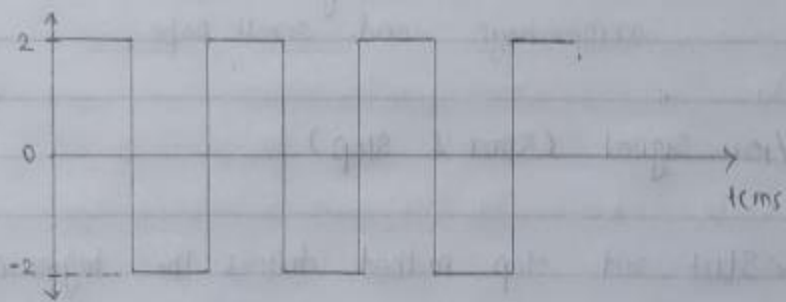


Spectrum Analyser

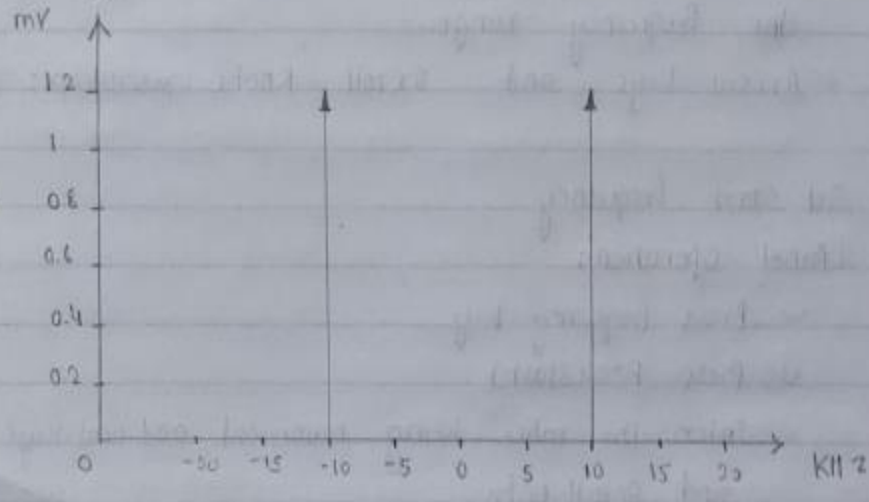


3.) A = 2 mV f = 10 KHz

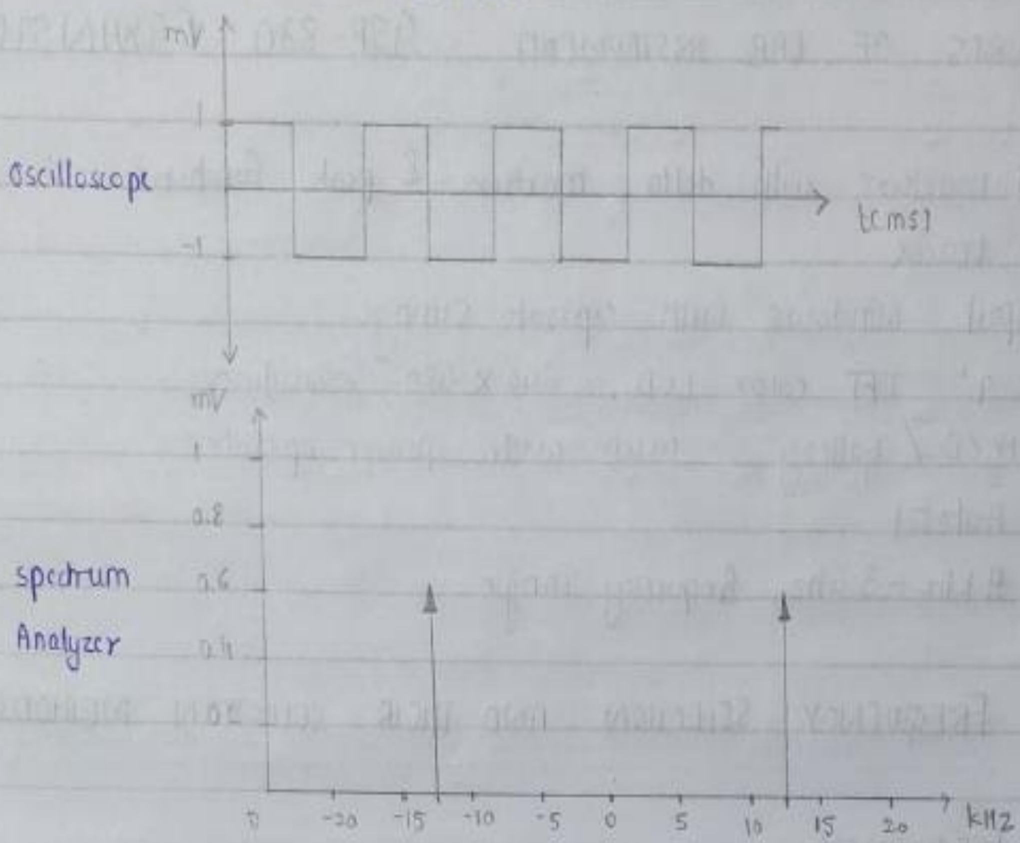
Oscilloscope



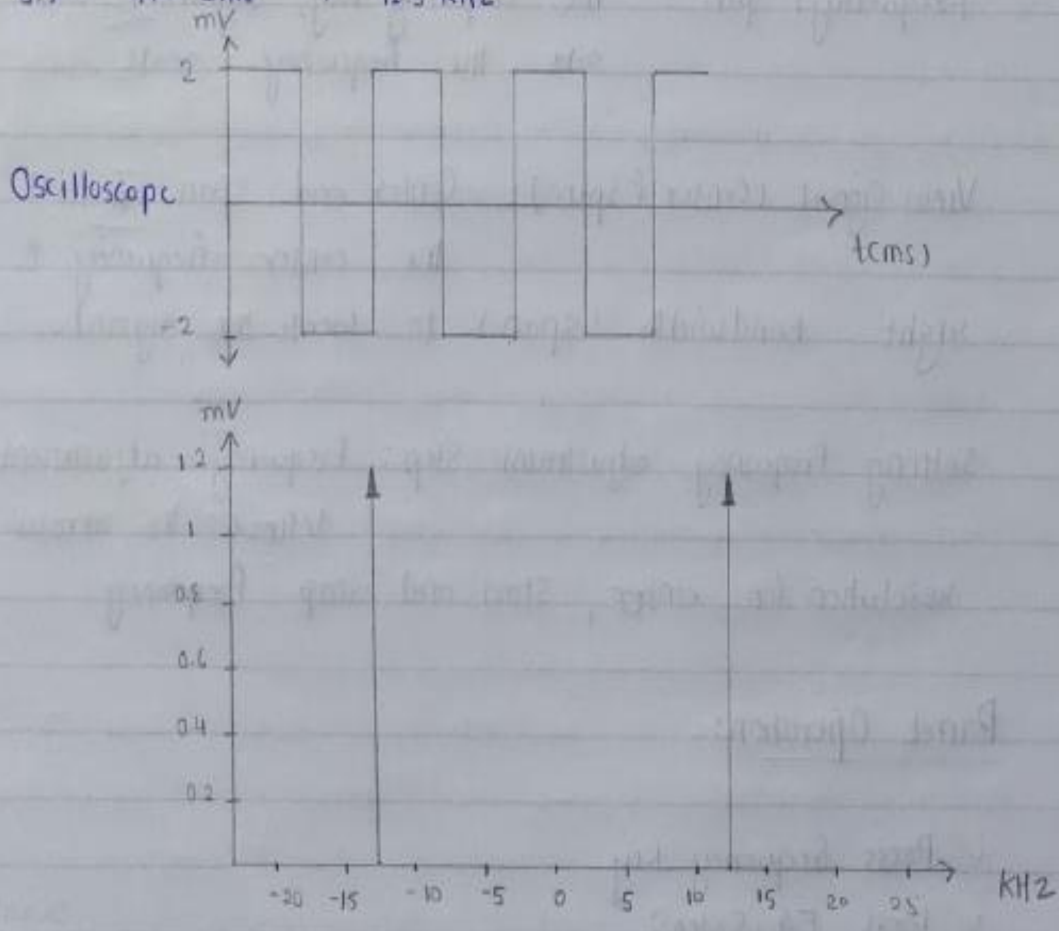
Spectrum Analyser



4.> A = 1mV   f = 12.5 KHz



5.> A = 2mV   f = 12.5 KHz



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EXPERIMENT - 2

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SAMPLING AND NYQUIST CRITERIA

~~Experim~~ SAMPLING AND..

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

Apparatus : Nyquist Applet (software)

Theory : (1) A Continous - 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 of the feature of the Signal.

→ Digital Signal : It represents data as sequence of discrete values at any given time, it can only takes any of one of the finite no. of values.

→ The technique that can be used for analog to Digital Conversion is pulse Code modulation.

→ It has three process
- Sampling • Quantization • Encoding

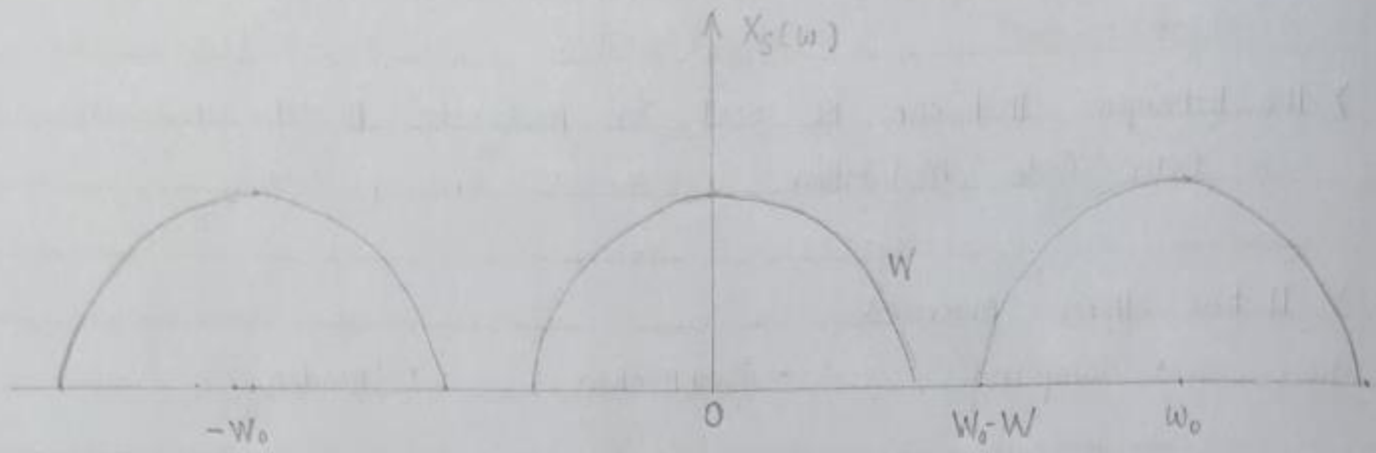
a) Sampling

→ Sampling is the process of measuring the instantaneous values of Cont.- time Signal in discrete form sample is a piece of data taken from the whole date which is Continous in the time domain.

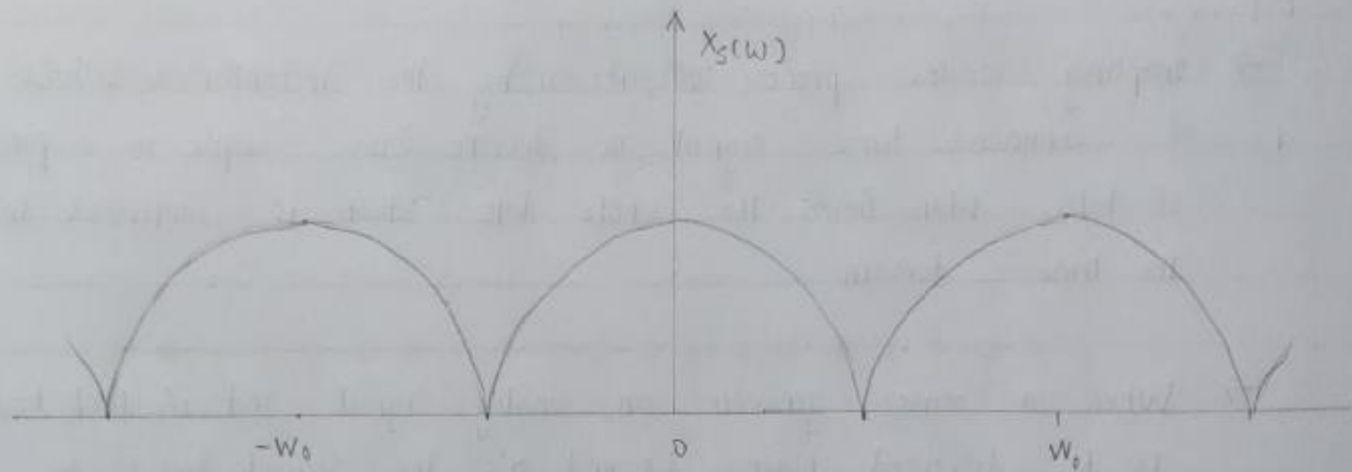
→ When a source generate an analog signal & if that has to be digitized, having 1's and 0's. The Signal has to be discretized in time. This discretization of analog signal is called Sampling.

→ 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 freq. needed for Sampling a slowly varying signal. Such as the Output at a gas- chromatograph detector or the potential of a glass - electrode. during acid base titration.

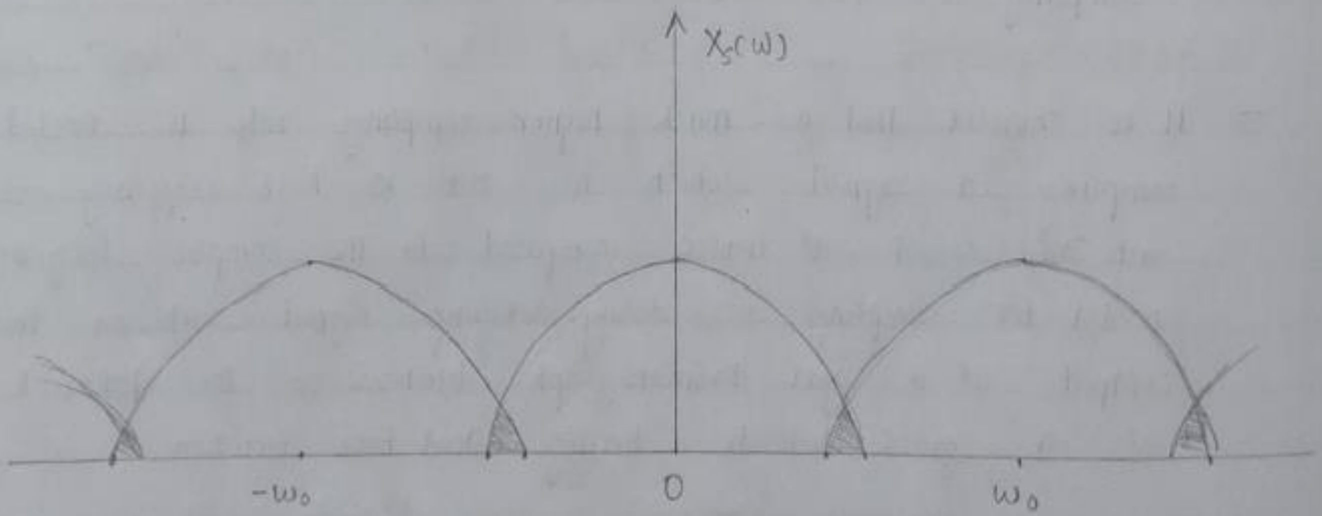
→ The minimum sampling frequency of a signal that it will not distort its underlying info. Should be double the frequency of its highest frequency component. This is the Nyquist Sampling theorem.



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



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



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

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3) Nyquist Rate

→ Suppose that a Signal is band-limited and ω is the highest frequency.

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

$$\therefore \boxed{F_s = 2W}$$

F_s = Sampling Rate W = Heighest frequency

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

→ If sampled at higher rate the $2W$ in the frequency domain.

$$X_s(\omega) = \frac{1}{T_s} \sum_{n=-\infty}^{0} x(\omega - n\omega_n)$$

Hence, the information is reproduced with .

B) Condition 2 :

→ If the Sampling rate is equal to twice the frequency $\boxed{F_s = 2W}$

→ 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 lose of information. This unwanted phenomena 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.

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 stablized. value is called Quantization.

5> Encoding

→ The digitization of analog signal is done by encoders.

→ 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 acquisation system. The max^m. frequency of the input signal should be less than or equal to half of sampling rate.

→ This sets the cut-off frequency of the low-pass filter.

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

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

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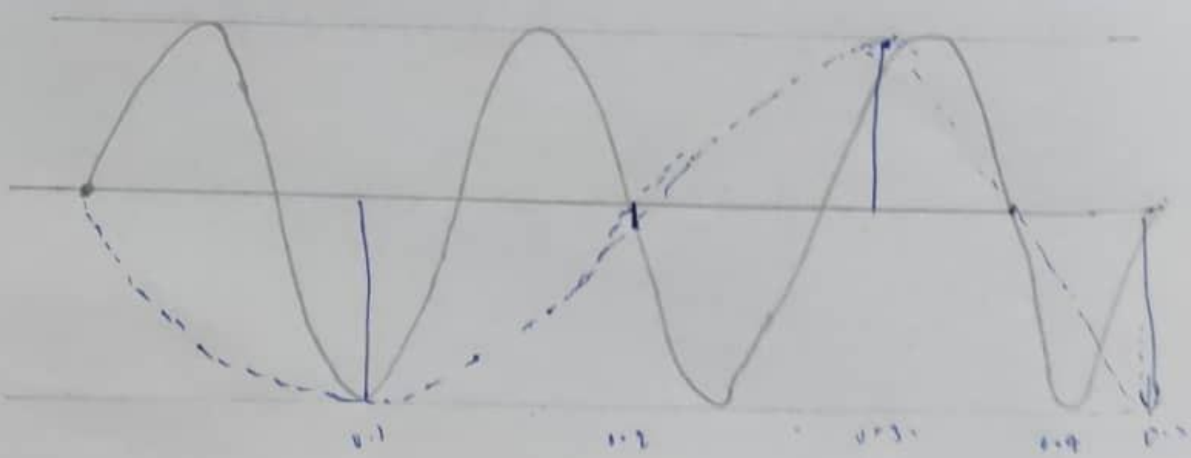
Conclusion : Therefore, sampling and reconstruction of the signal has been performed successfully on Nyquist Applet (software) & Nyquist Criteria has been verified.

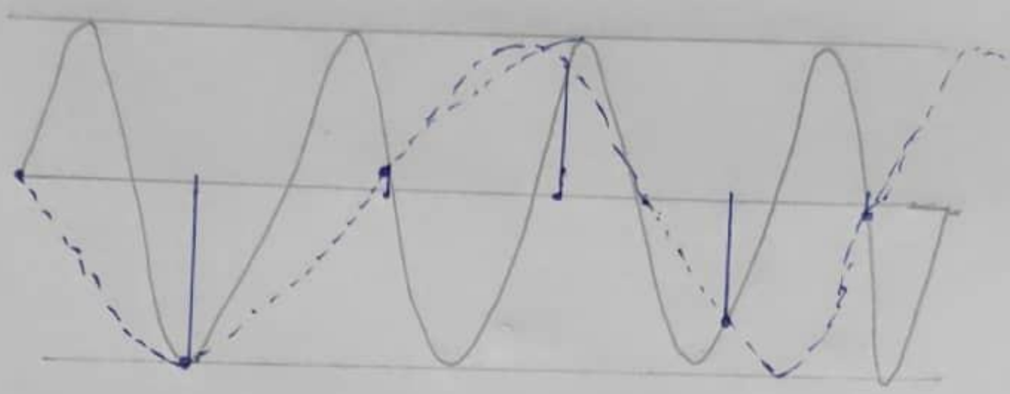
observation table

Signal frequency (Hz)	Sampling frequency (Hz)	Alias frequency (Hz)
10	13	3
	10	0
	15	5
	20	-
	25	-
20	19	1
	20.1	0.1
	30	10
	40	-
	42	-

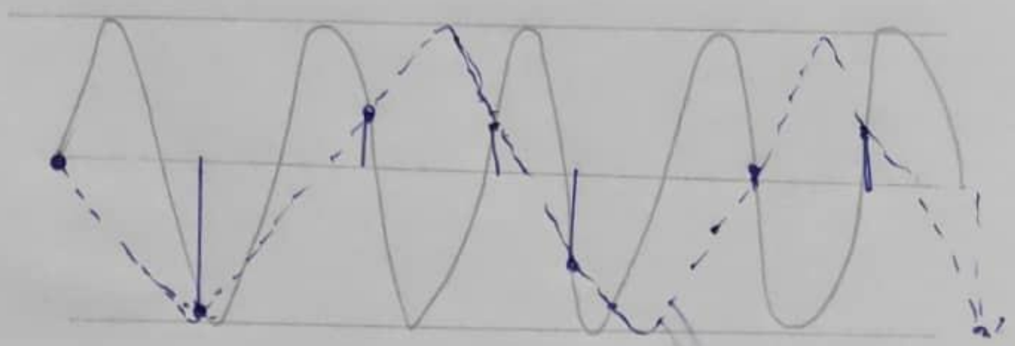
① signal freq (Hz) = 10.0 Alias freq (Hz) 3.0 sampling freq (Hz) = 13.0



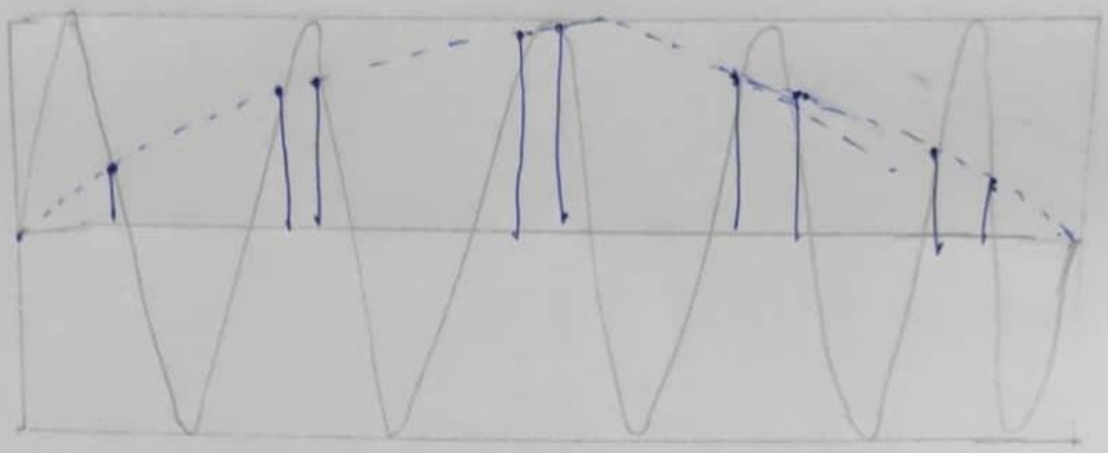
③ signal freq (Hz) = 10.0 Alias (Hz) = 4.0 . sampling = 14.0
freq (Hz) .



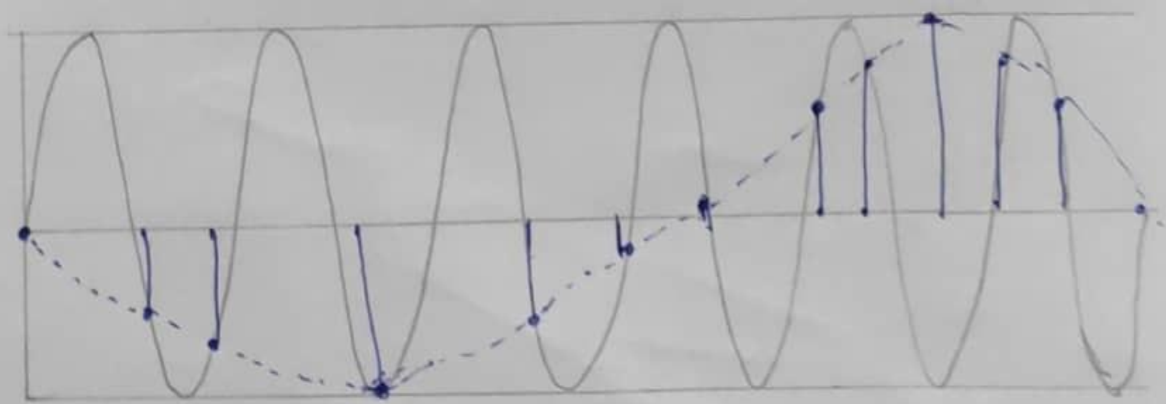
④ signal freq (Hz) = 10.0 Alias freq (Hz) = 5.0 sampling (Hz) = 15.0 .



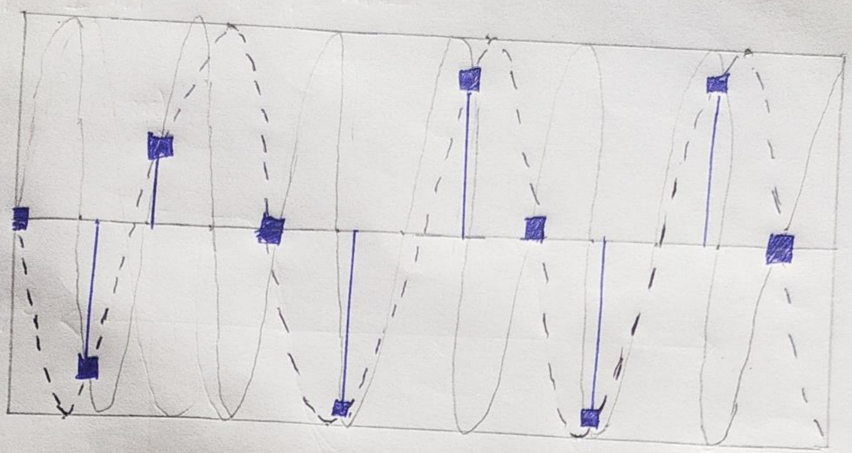
6) signal frequency (Hz) = 20 Alias freq Hz = 10 sampling freq Hz = 19



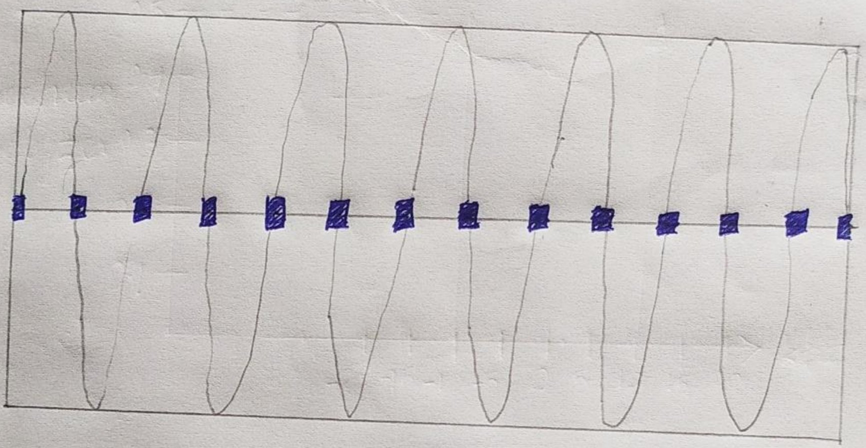
① Signal freq (Hz) = 20.0 Alias freq (Hz) = 2.0 sampling freq (Hz) = 22.0



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



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



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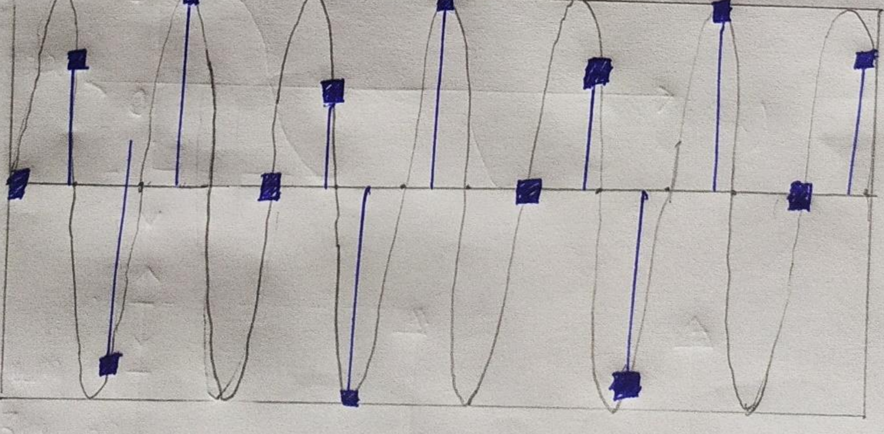
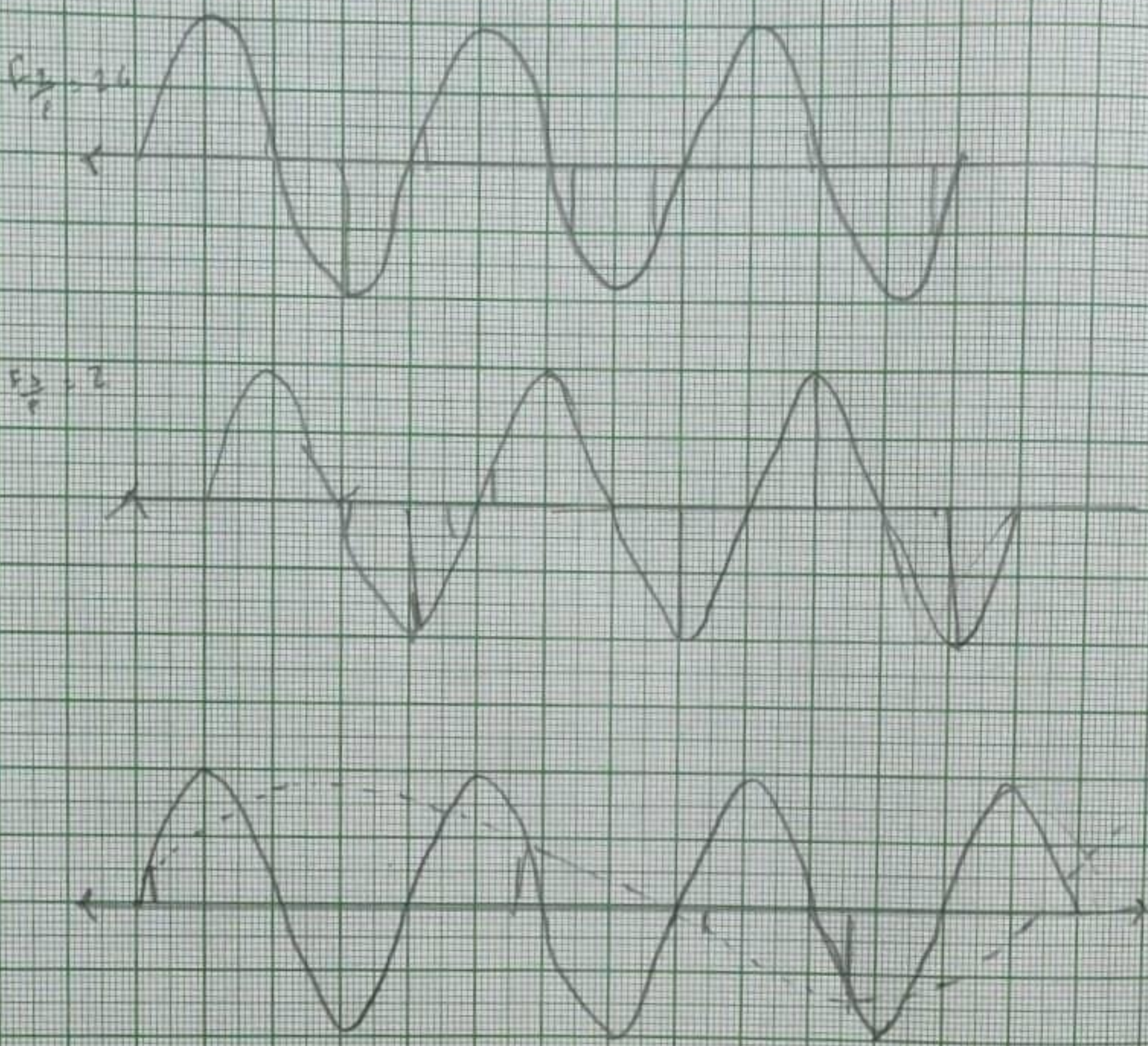




Sampling of a sinussoidal signal of freq f at diff sampling rates with dashed lines are shown the alias frequency when $\frac{f_s}{f} < 2$.

EXPERIMENT - 3

AMPLITUDE MODULATION

AIM : Study of an Amplitude modulated (A.m) scheme, depth of modulation, waveforms, spectra & 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. (VSB).
- 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 $s(t) = [A_{(t)} A_m \cos(2\pi f_m t)] \cos(2\pi f_c t)$

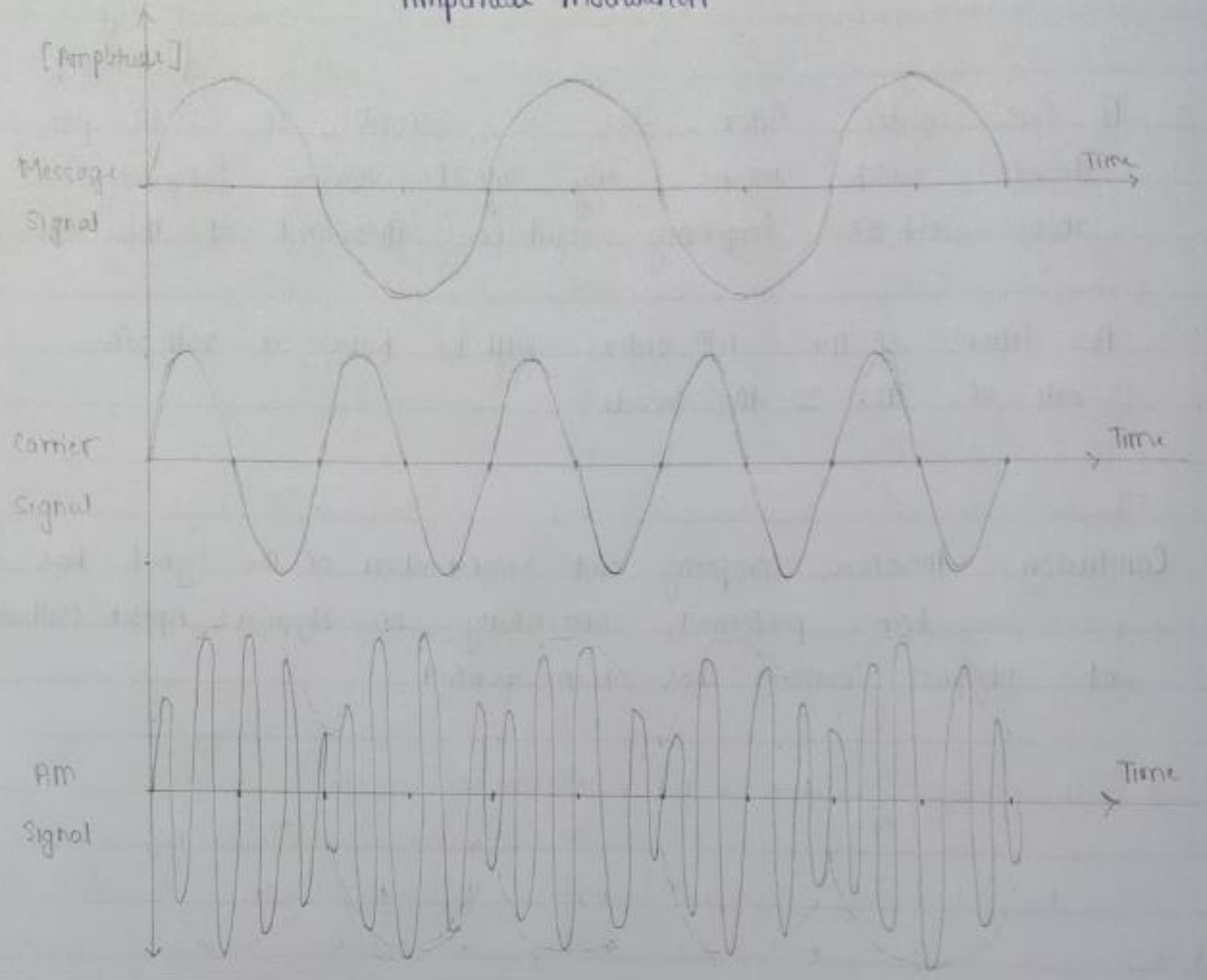
$$s(t) = A_c\left[1 + \frac{A_m}{A_c} \cos(2\pi f_m t)\right] \cos(2\pi f_c t)$$

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

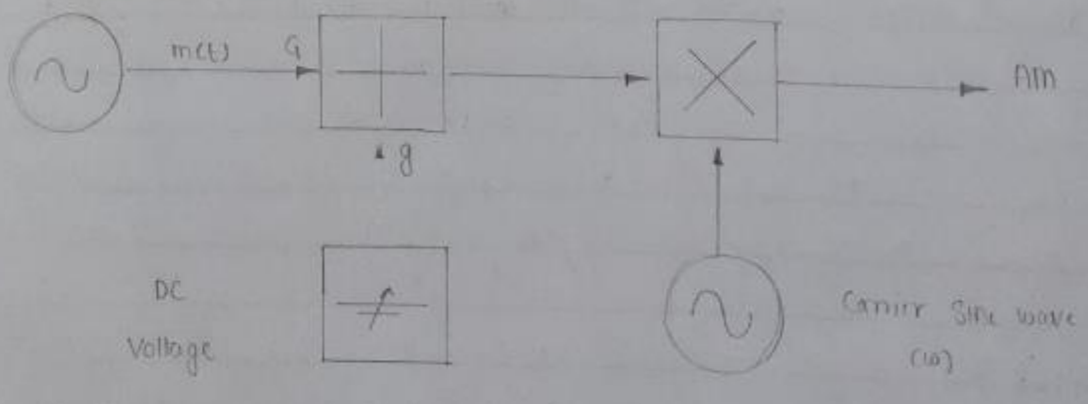
3) Measurement of 'M'

- The magnitude of 'm' can be measured directly from the AM display itself.

Amplitude Modulation



Schematic Block diagram for AM, T_x and R_x



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→ Maximum & minimum amplitude of the transmission signal envelope, determine modulation depth:

$$M = Am / Ac$$

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

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

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

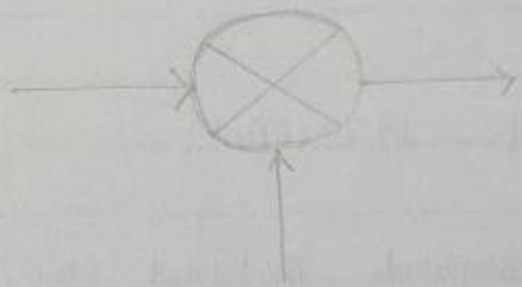
4) Envelope Detector.

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

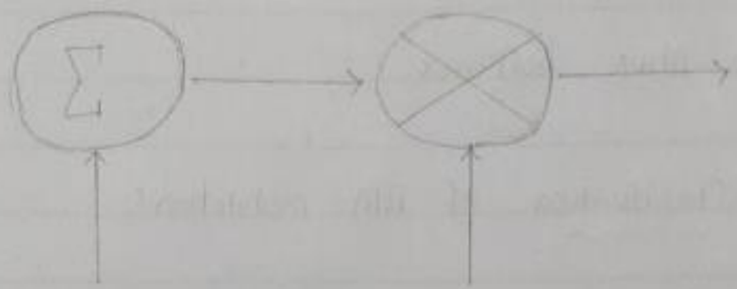
5) Synchronous detector

→ Am without a carrier, envelope detection Can't be deployed because the transmitted signal's envelope changes sign Transmit spectrum of DSB - SC

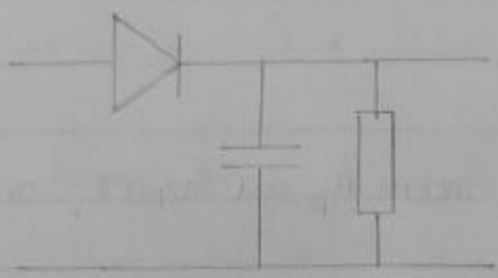
DSB - SC



AM

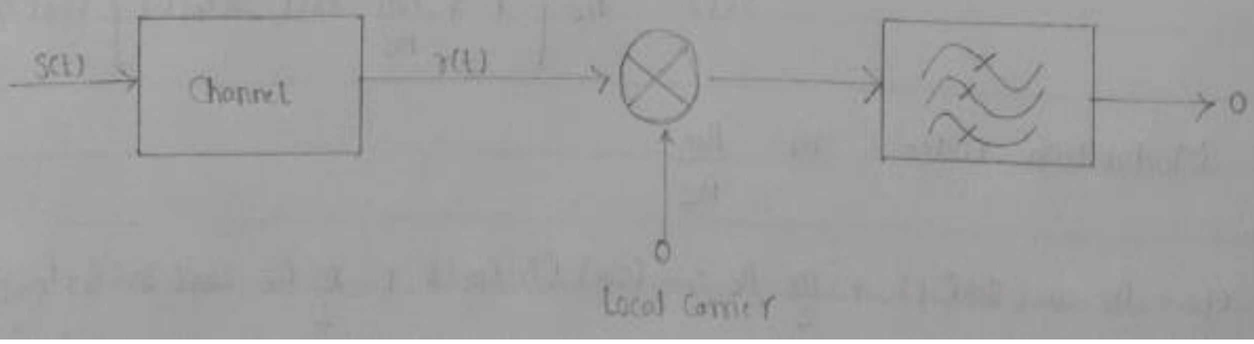


Envelope Detector



Rectifier
(Diode, capacitor)

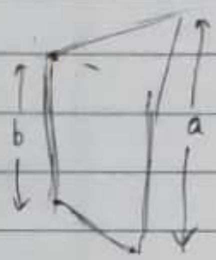
Synchronous Detector



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 practical, we perform the exp. on labalie applicat

1) We will first execute the AM analyser Simulator.

2) After executing the AM analyser simulator, click on the S in the AM modulation window

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

Obs. Table

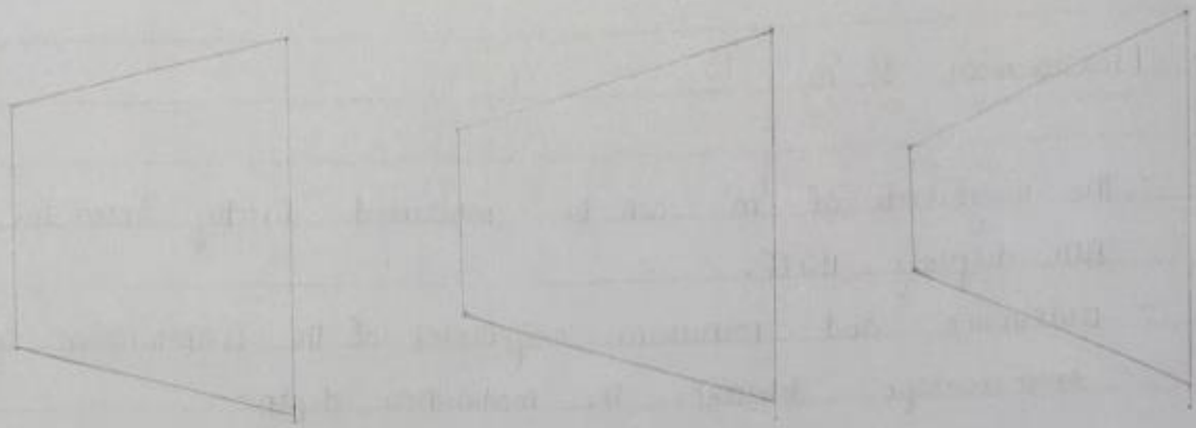
	m(modulation index)
m < 1	0.5
m = 1	1
m > 1	1

m(modulation index)	
m <	0.5
m = 1	1
m > 1	2

Conclusion

we observe that using envelope detector we can detect double side band with carrier but synchronoes detector is needed for Double Sideband suppressed Carrier. we also observe that information lies in Sidebands and in carrier : using DSBCC, we can minimize power usage.

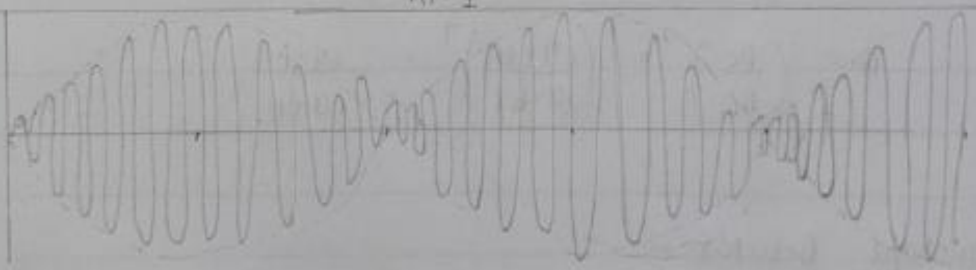
Trapezoid width is unaffected by modulation depth.



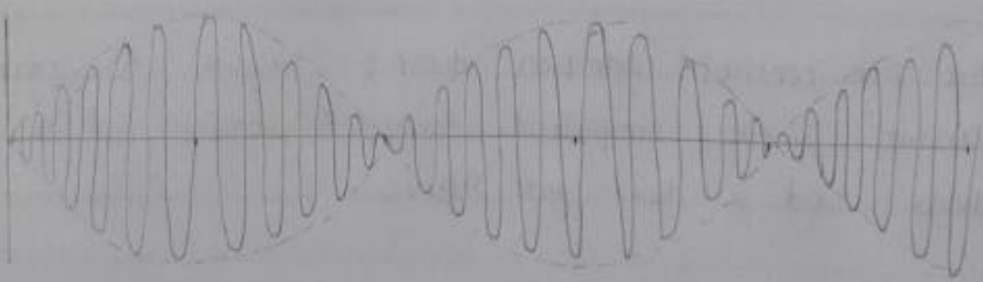
m = 0.1 m = 0.3 m = 0.5

m = 1

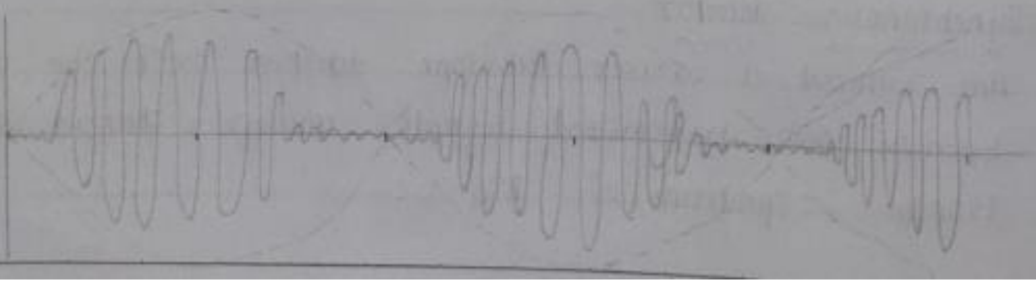
Modulation Index = 0.5 (m<1)



Modulation Index = 1 (m=1)

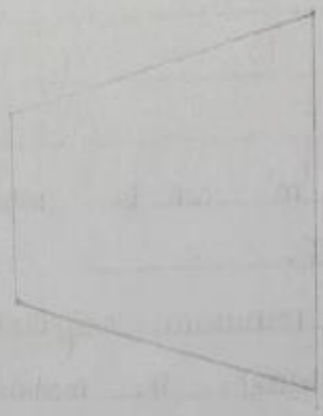


Modulation Index = 1.5 (m>1)

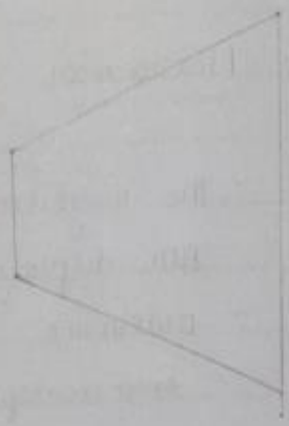


Trapezoid width is unaffected by modulation depth.

m = 0.1



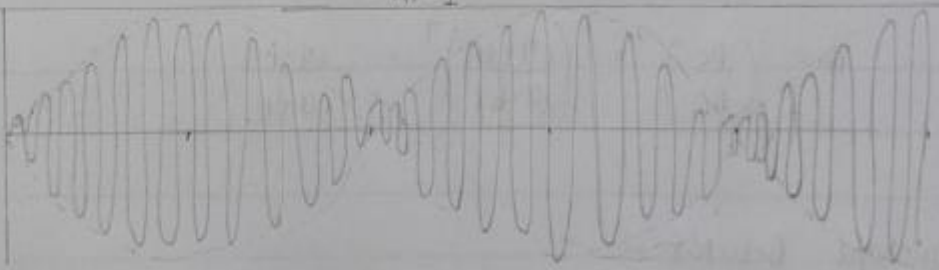
m = 0.3



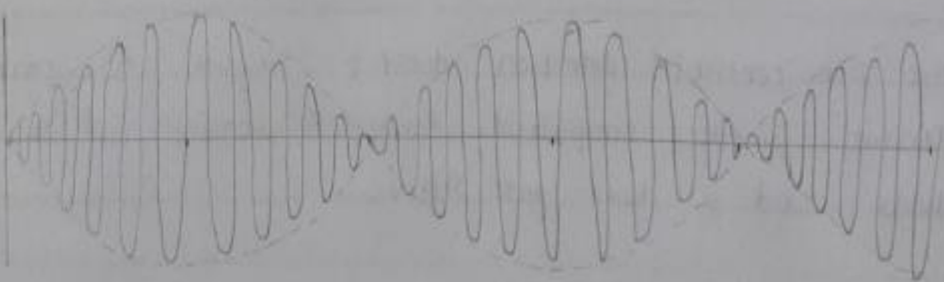
m = 0.5

m = 1

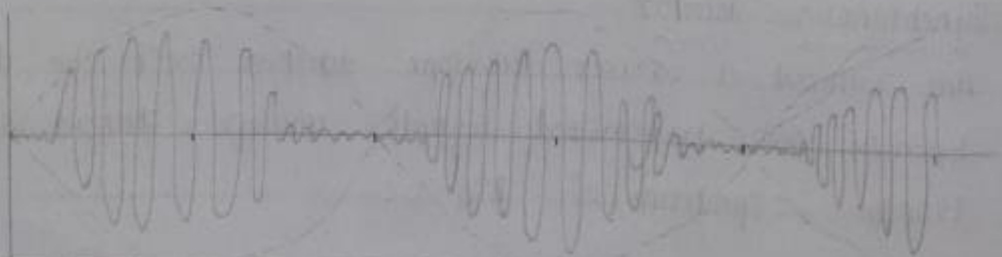
Modulation Index = 0.5 (m<1)



Modulation Index = 1 (m=1)



Modulation Index = 1.5 (m>1)



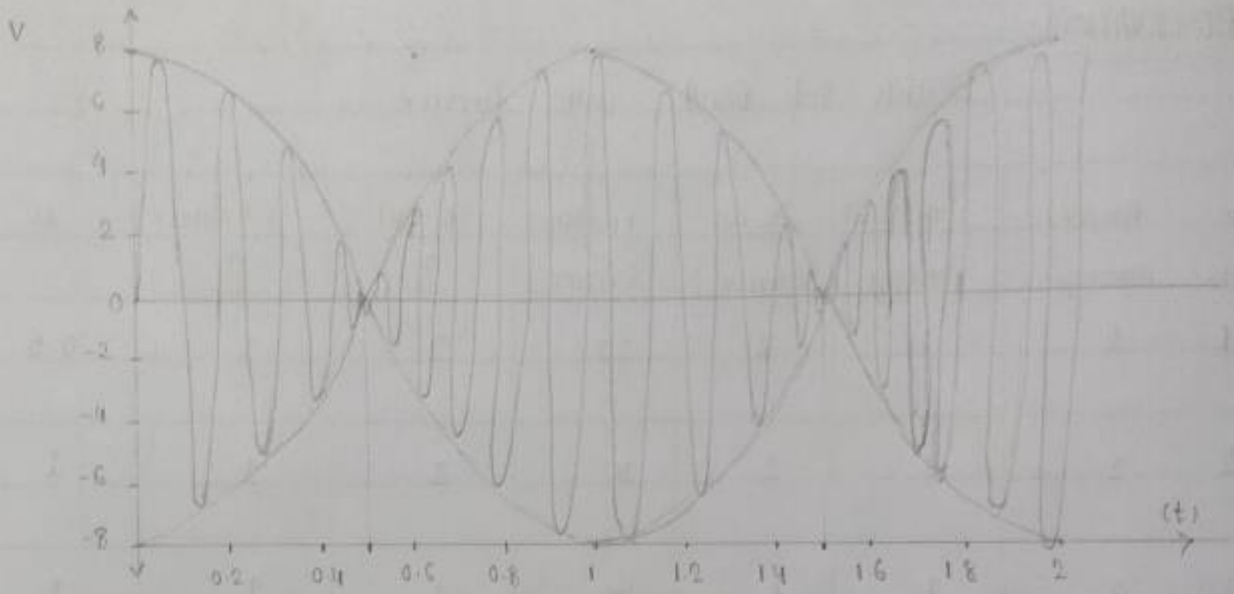
(b) m = 1

Message $A_m = 2V$ $f_m = 1\,MHz$ (cosine)

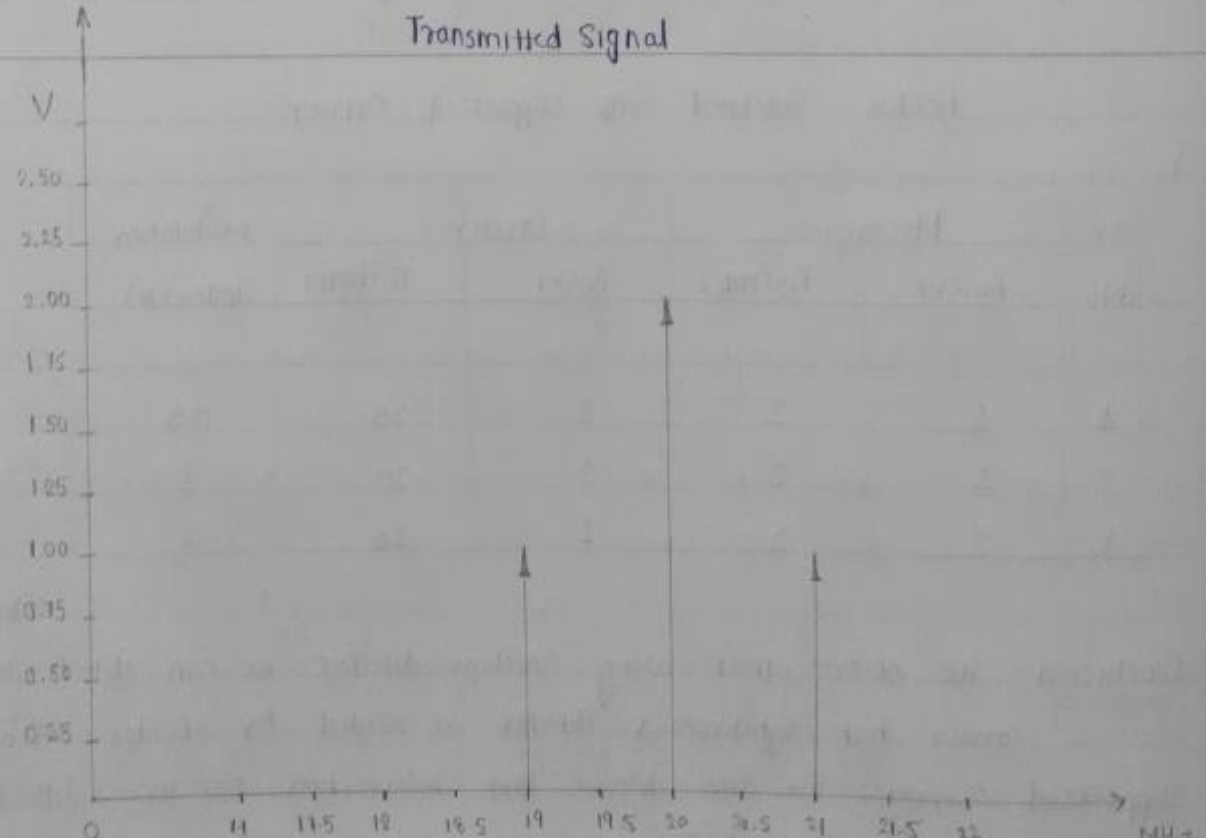
Carrier $A_c = 2V$ $f_c = 20\,MHz$ (cosine)

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

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



Transmitted Signal



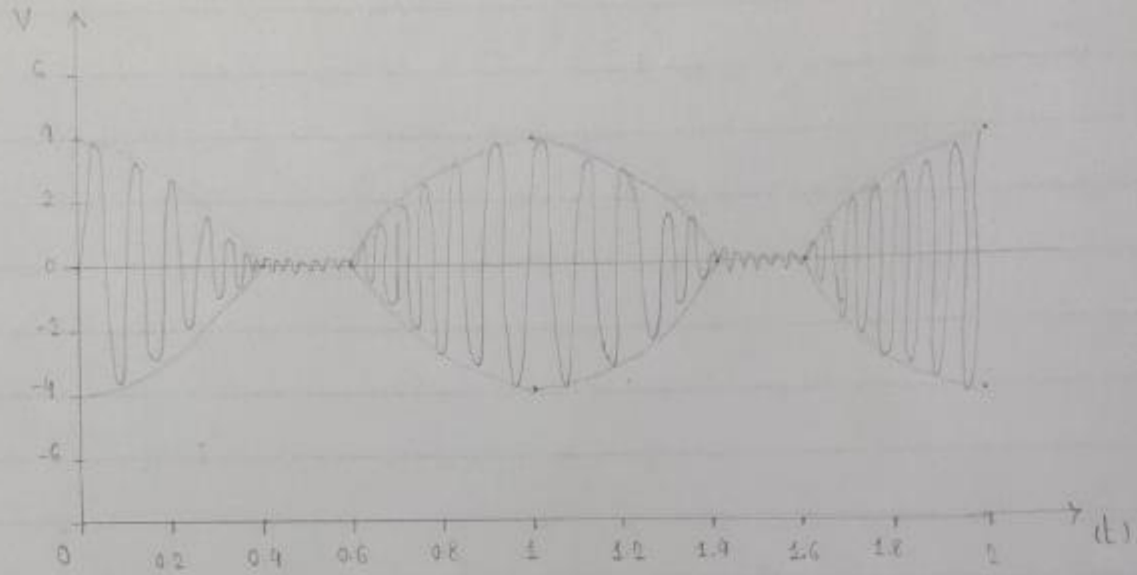
Spectrum

c) m > 1 message $A_m = 2V$ $F_m = 1$ MHz (cosine)

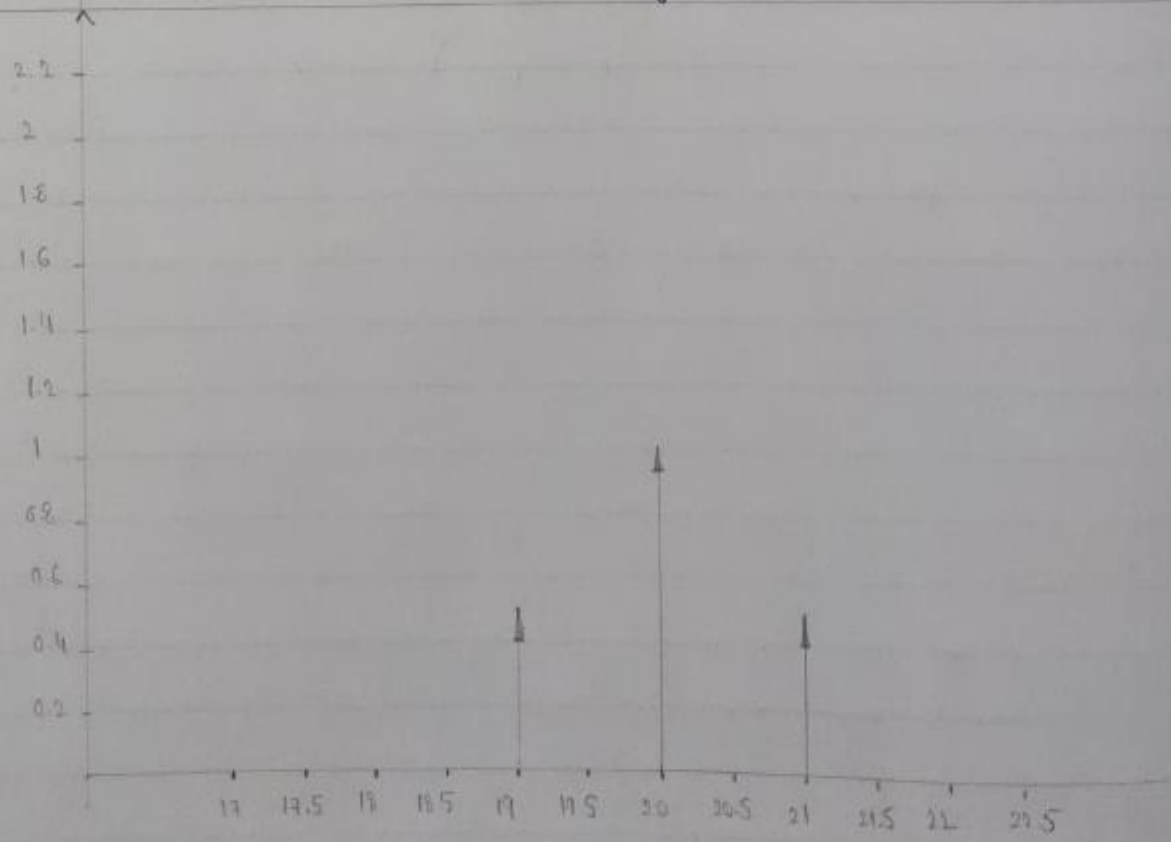
carrier $A_c = 1V$ $F_c = 20$ MHz (cosine)

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

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



Transmitted Signal



Spectrum

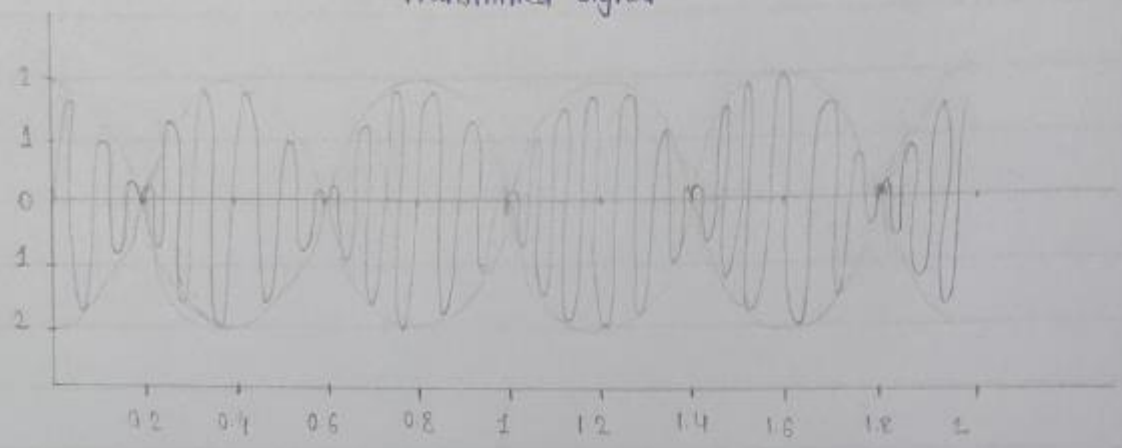
DOUBLE SIDEBAND SUPPRESSED CARRIER [DC OFFSET OFF]

a) m<1

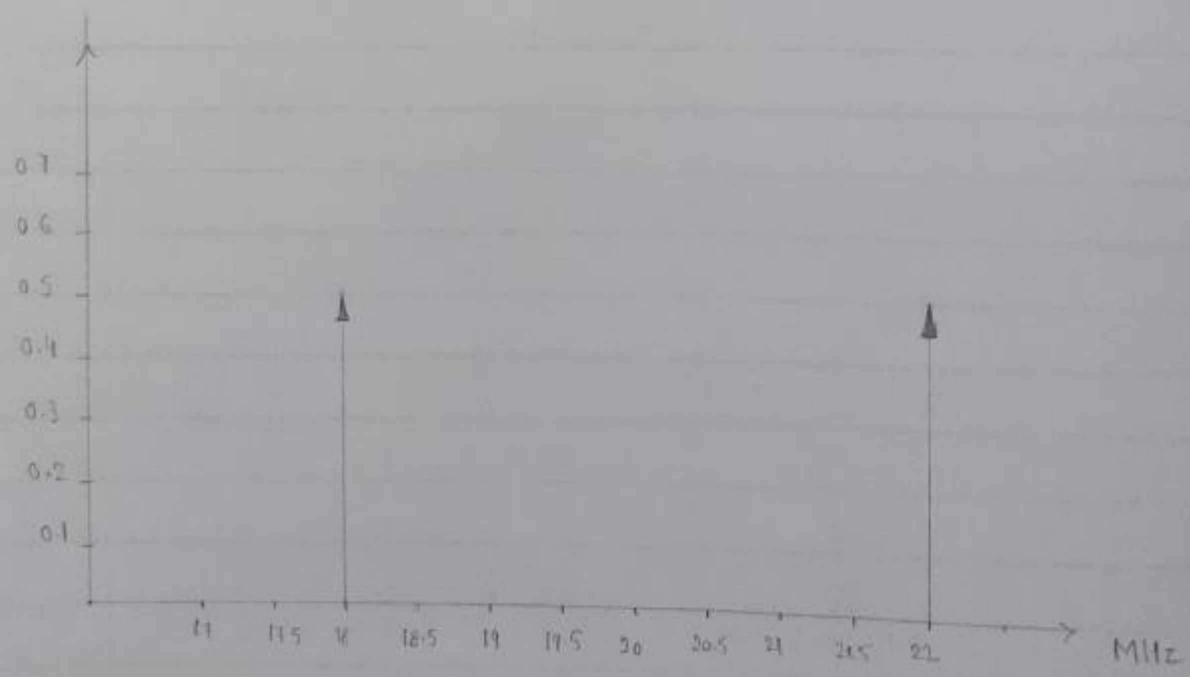
Message	$A_m = 1V$	$F_m = 2MHz$	(cosine)
Carrier	$A_c = 2V$	$F_c = 20 MHz$	(cosine)

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

Transmitted signal



Spectrum

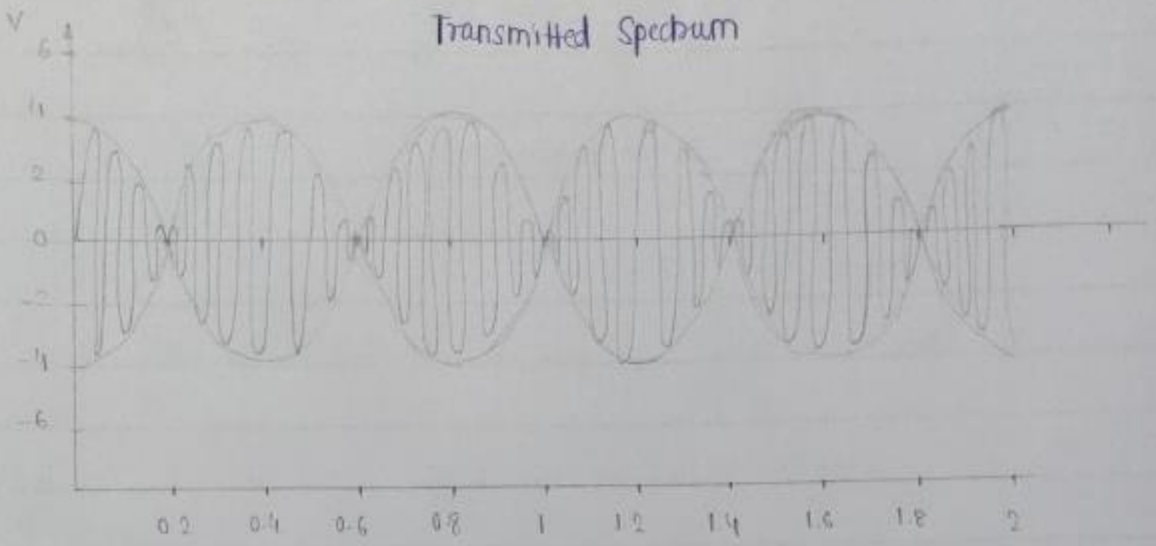


b) m = 1 Message $A_m = 2V$ $F_m =$ ~~10~~ 2 MHz (cosine)

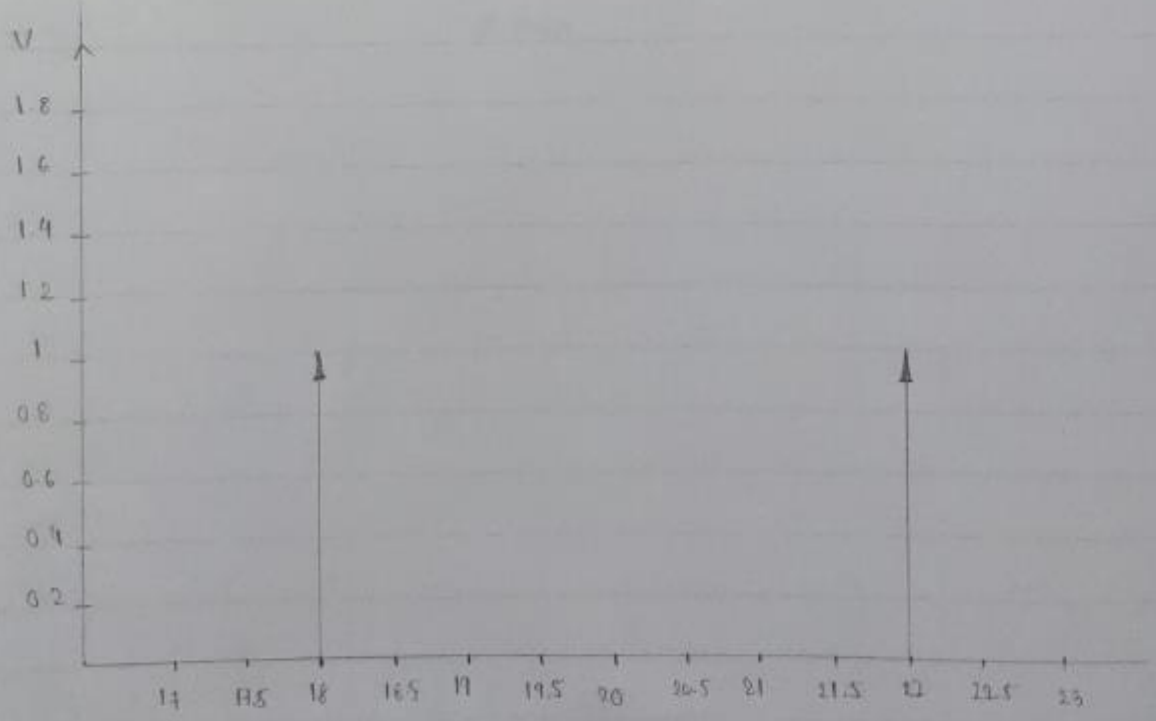
Carrier $A_c = 2V$ $F_c =$ 20 MHz (cosine)

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

Transmitted Spectrum



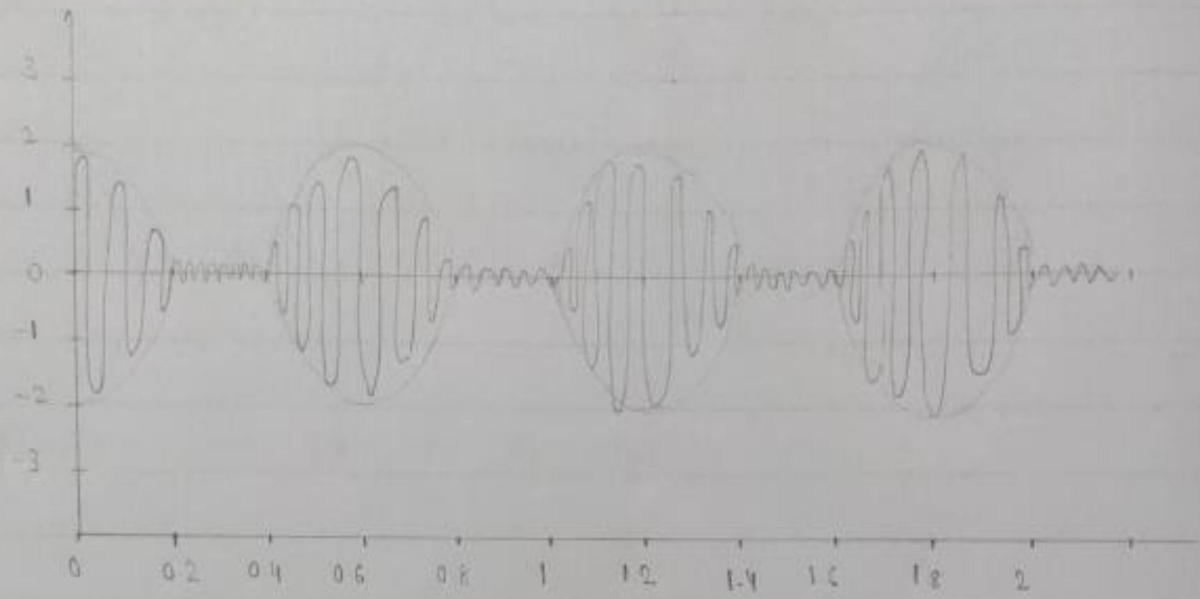
Spectrum



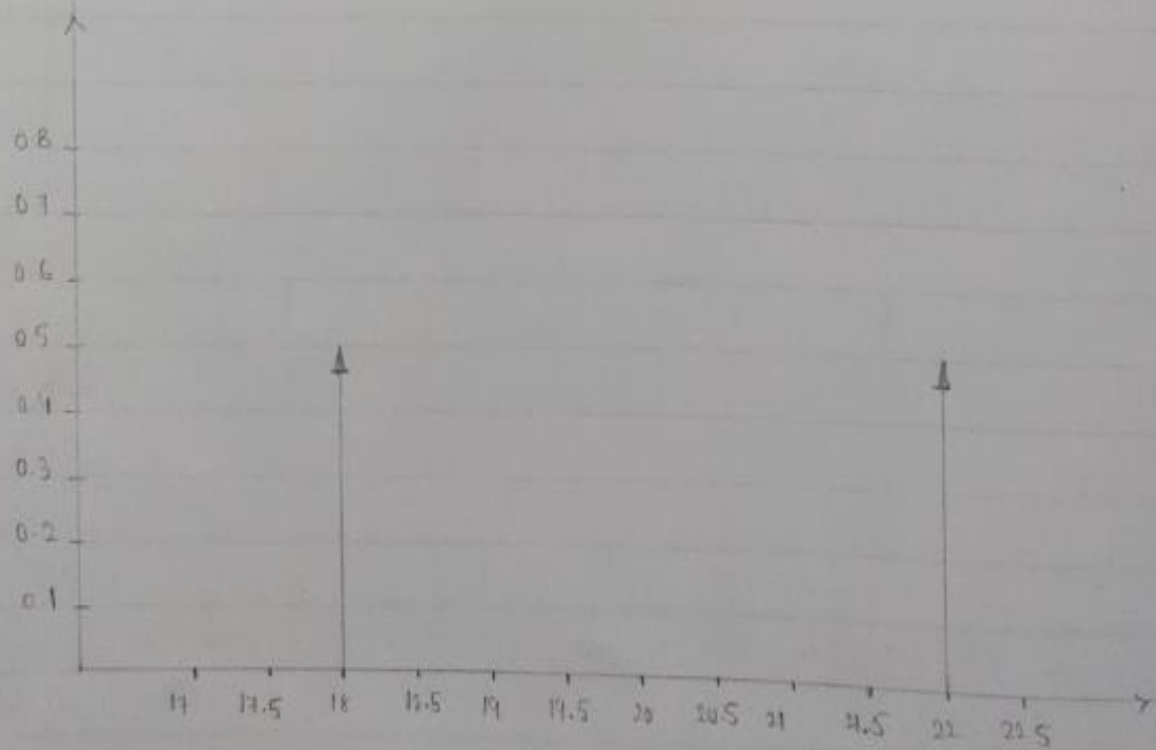
c) $m > 1$ Message $A_m = 2V$ $F_m = 2\,kHz$ (cosine)

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

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



Transmitted signal



Spectrum

EXPERIMENT - 4:

FREQUENCY MODULATION AND DEMODULATION

AIM: To study frequency modulation (F.M) & frequency modulation with its Application.

Apparatus Required : Lab Alive 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.

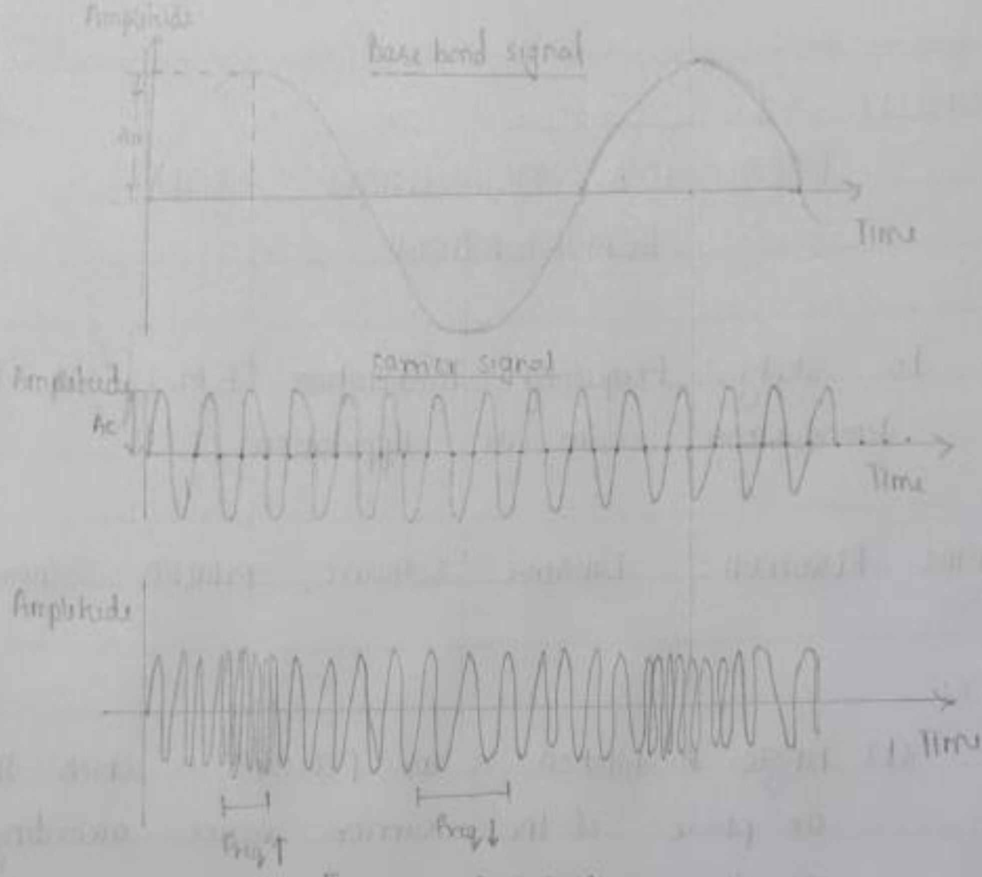
② The standard equation of the Angle modulated wave is

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

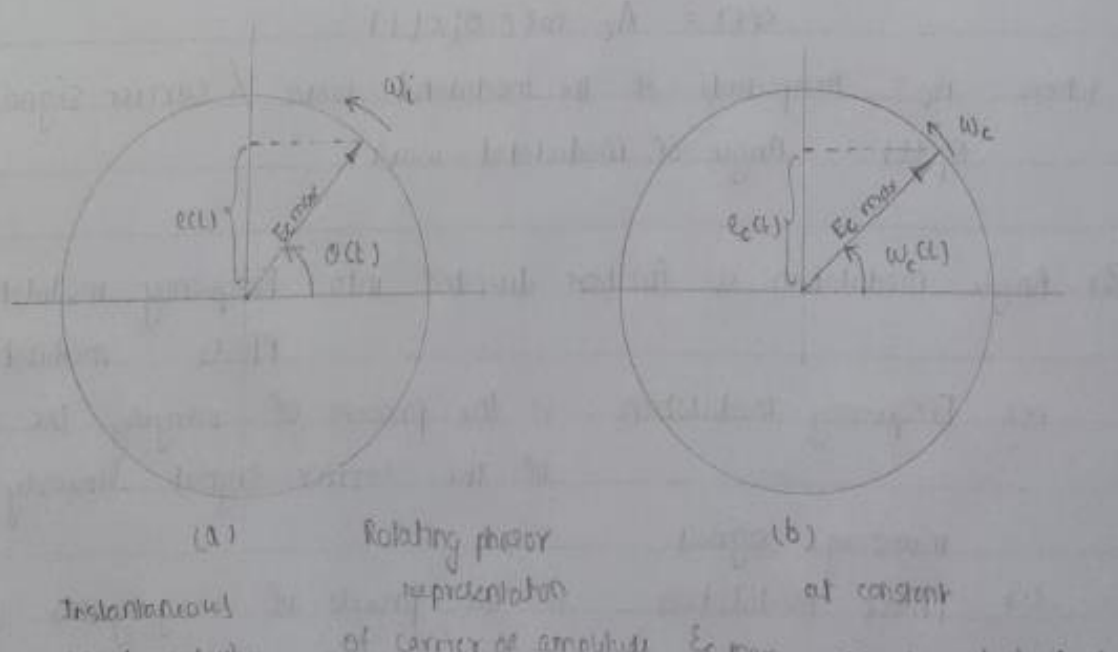
where, A_c = Amplitude of the modulated wave / Carrier signal. $\theta_i(t)$ = Angle of modulated wave.

③ Angle modulation is further divided into frequency modulation & Phase modulation.

① Frequency modulation : It is the process of varying the frequency of the carrier signal linearly with message signal.



Frequency Modulated wave



(a) Rotating phasor (b)

representation

Instantaneous angular velocity $\omega_i(t)$ of carrier of amplitude E_c max at constant angular velocity (ω_c)

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

4) The frequency of the modulated wave increses, when the Amplitude of the modulation 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.

$$\boxed{f_i = \underbrace{f_c}_{\text{carrier frequency}} + \underbrace{(K_f)}_{\text{frequency sensitivity}} \underbrace{m(t)}_{\text{message signal}}} \quad -①$$

6) we know, relationship between ω_i & $Q_i(t)$.

$$\left[\omega_i = \frac{d(Q_i)}{dt}\right] \quad -(ii).$$

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

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

Substitute f_i from eq^n ①

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

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

Substitute $\theta_i(t)$ value in standard eq^n of Angle modulated wave :

$$\boxed{s(t) = A_c \cos(2\pi f_c t + 2\pi K_f \int m(t)\, dt)} \text{ — ④}$$

(Eq^n of fm wave)

7) finally, equation of FM, wave

$$s(t) = A_c \cos(2\pi f_c t + 2\pi K_f \int m(t)\, dt) \text{ ④}$$

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

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

$$\boxed{\beta = \frac{K_f A_M}{f_m} = \frac{\Delta f}{f_m} = \text{modulation index}}$$

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8) The difference between FM modulated frequency (instantaneous frequency) and normal carrier frequency is termds 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 & A_m.

9) FM can be divided into Narrowband FM & wideband FM. based on values of modulating index (β).

10) The amount of change in carrier frequency produced, by the amplitude of input modulating signal, is called frequency deviation.

Carrier frequency swings b/w fmax and fmin as input varies. freq deviation.

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

f.	(MHZ)	fd from carrier
fc	100	Nill
fmax	105	+5 MHZ
fmin	95	-5 MHZ

11) F.M signal spectrum is quite complex & 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.

Sidebands are seperated from Carrier by $f_c \pm f_m$, $f_c \pm 2f_m$, $f_c \pm 3f_m$, and so on....

$$\boxed{\text{Band width} = 2(f_m + \Delta f)}$$

12) In F.M, carrier Amplitude is constant,
∴ Transmitted Power is constant. & Transmitted power doesn't depend on modulation index.

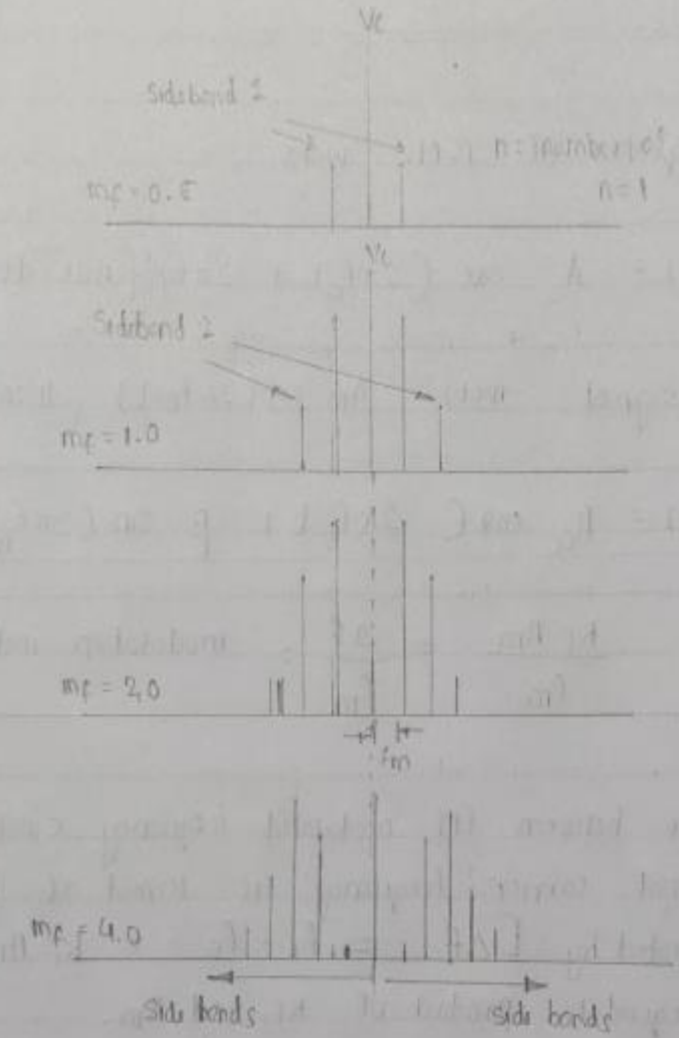
13) F.M has better noise immunity. FM is rugged/. robust against noise. The quality of FM will be good even in presence of noise.

14) Applications & Advantages of FM.

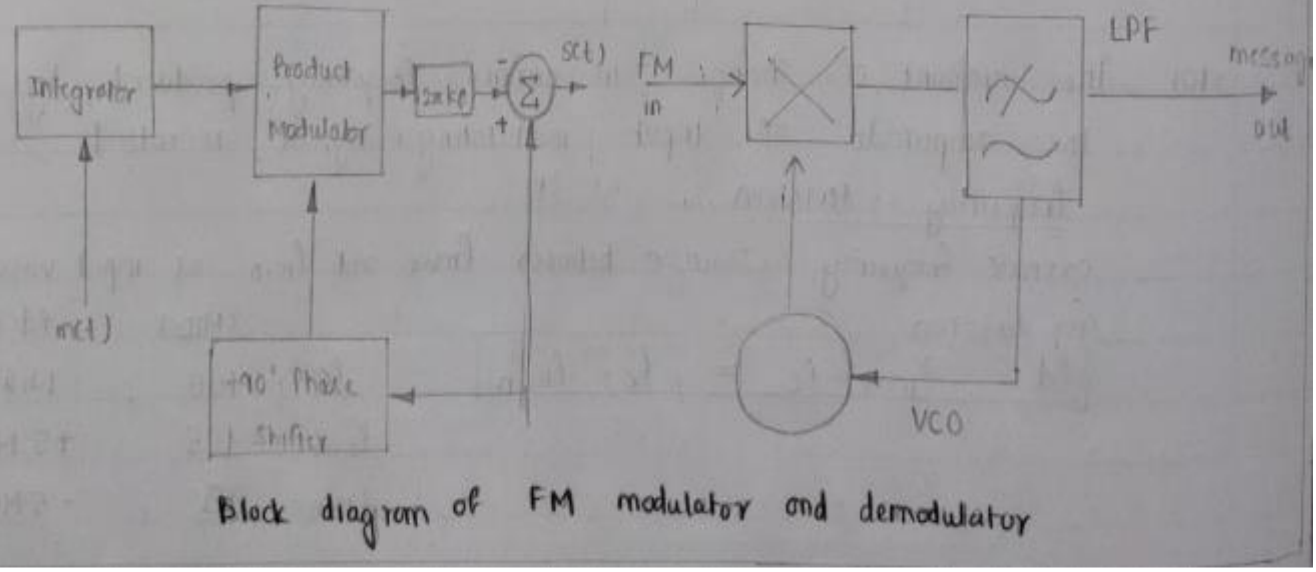
(A) FM is resilent to noise and interference.
∴ It is used for high quality broadcast transmission.

(B) FM is ideal for mobile radio communication application including more general two way radio Communication or portable application where signal levels are likely to vary ~~[illegible]~~ considerbly.

(C) Radar, Telemetry, observing infants for squrure through EEG.



How spectrum F.M. varies with mf



Block diagram of FM modulator and demodulator

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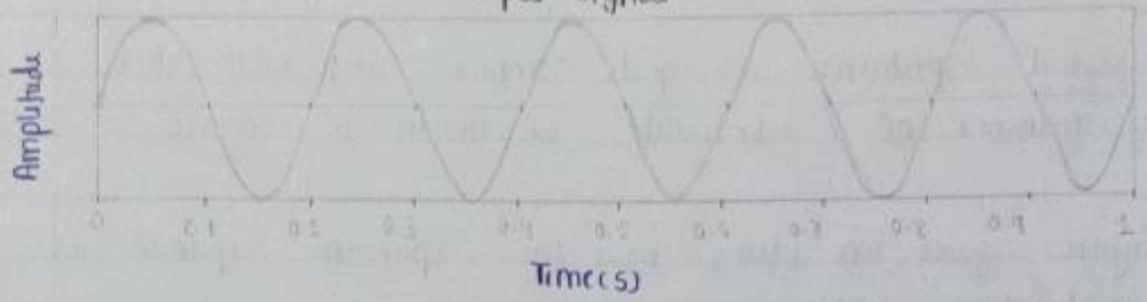
MATLAB code :

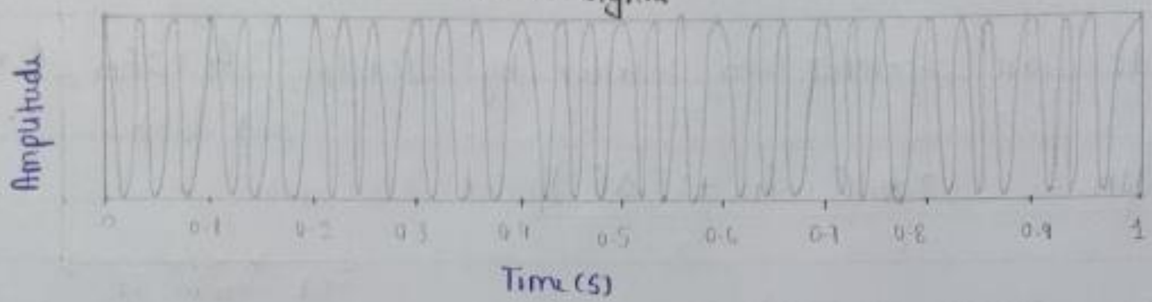
```
% Plot the frequency modulated signal.

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 ('Amplitude')
subplot (5,1,2)
Plot (t,c)
title ('Carrier signal')
xlabel ('Time (s)')
y label ('Amplitude')
subplot (5,1,3)
Plot (t,y)
title ('FM modulation of input Signal')
x label ('Time (s)')
y label ('Amplitude')
```

MATLAB OUTPUT

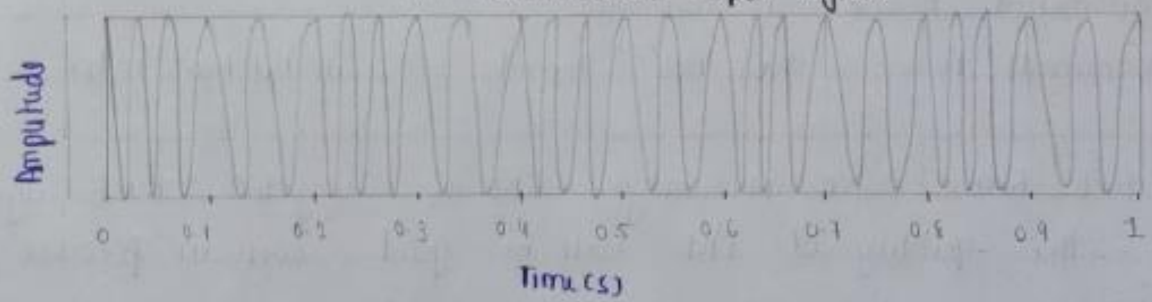
Input Signal



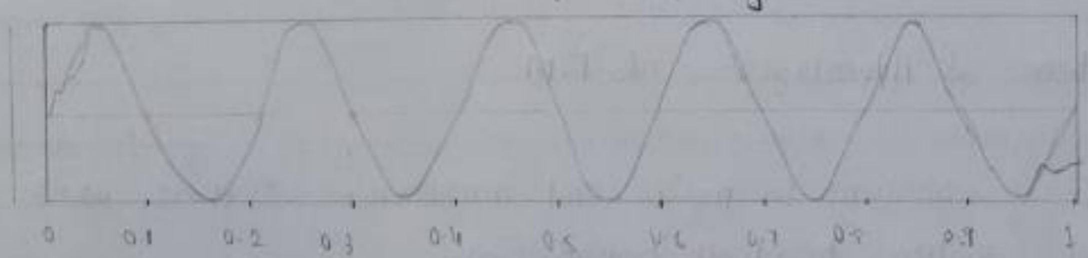
Carrier signal



FM Modulation of Input signal



Demodulation of FM signal



Magnitude spectrum of FM Signal

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% 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) * fs;
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 successfuly verified and understood the concept of frequency modulation and demodulation using MATLAB and also learnt various application of fm.

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EXPERIMENT-5 :

PAM , PPW & PWM

AIM : To examine pulse Amplitude (PAM), pulse Position modulation (ppm) & pulse width modulation (Pwm) & verify & draw the resultant waveforms.

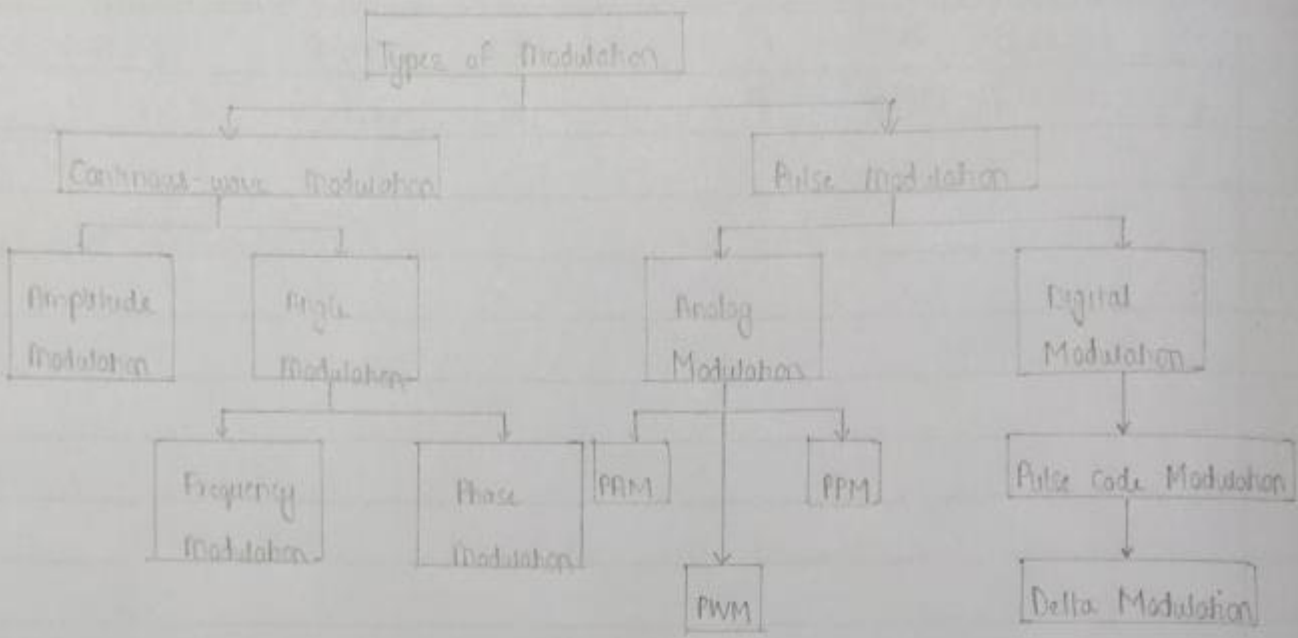
Apparatus : MATLAB software online

Theory :

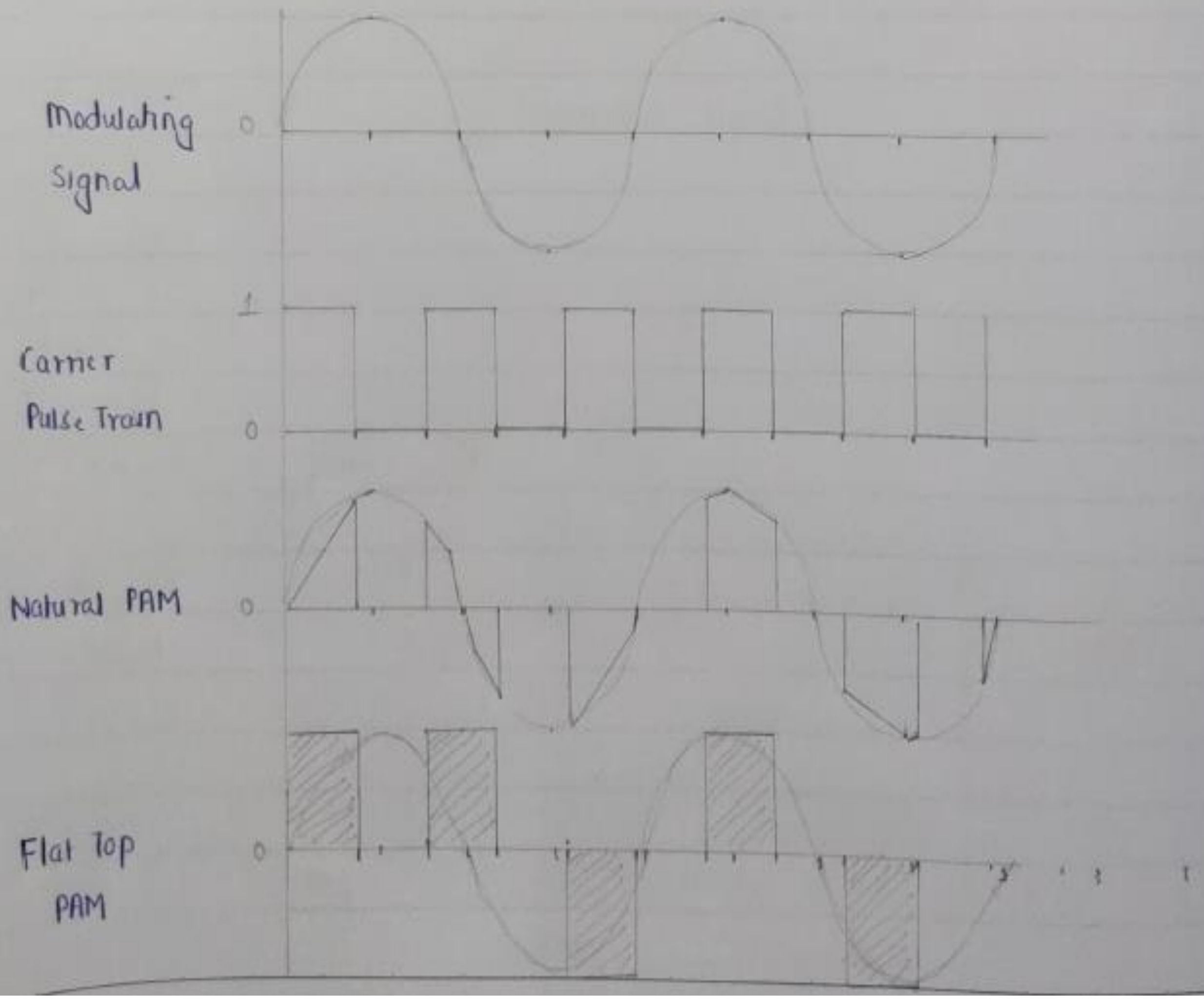
1> Pulse modulation is a type of modulation in which the signal is transmitted in the form of pulses.
In Pulse modulation, continues signals are Sampled at regular intervals. pulse modulation is further divided into Analog and Digital Communication & further analog & digital modulation is subdivided in PAM, PWM, PPM (analog) & 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 orignal signal as the singnal traces on the path of the whole wave.



Natural Sampling and Flat Top Sampling



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→ 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).

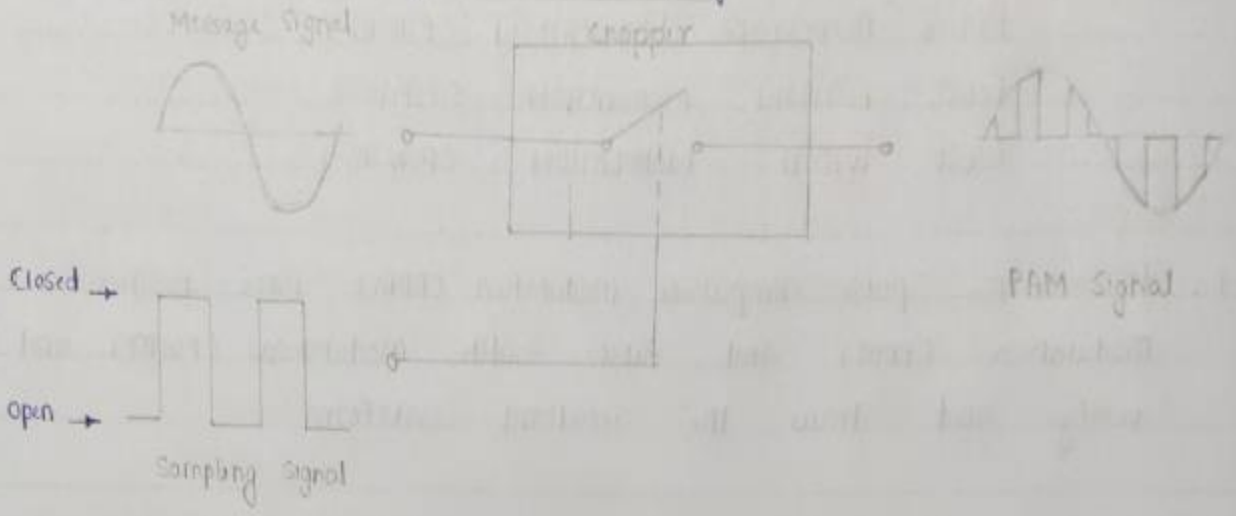
→ In this type, the amplitude is maintained constant but the duration or length or width of each pulse is varied accordance with instaneous value of analog signal.

4) Pulse Position modulation (PPM)

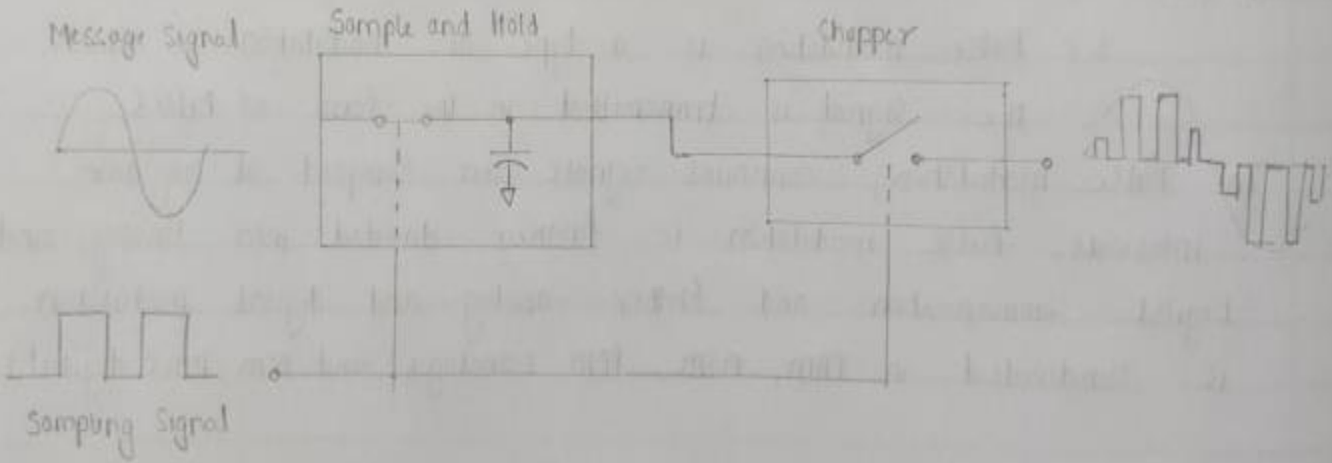
→ In this type of modulation, both the amplitude and width of the pulse are kept const. we vary the position of each pulse according to the instantaneous sampled value of the message signal.

→ PPM is further modification of PWM.

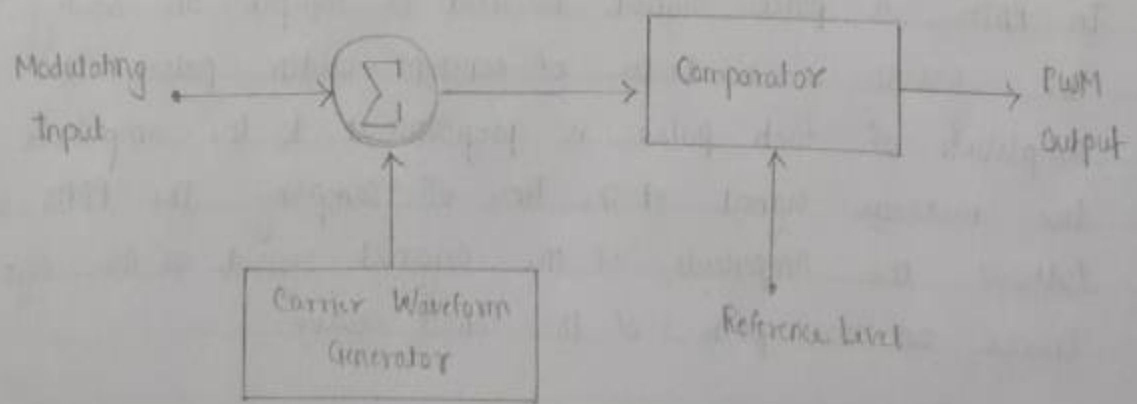
Generation of PAM by Natural Sampling :



Generation of PAM Signal by Flat - top Sampling



Generation of PWM Signal



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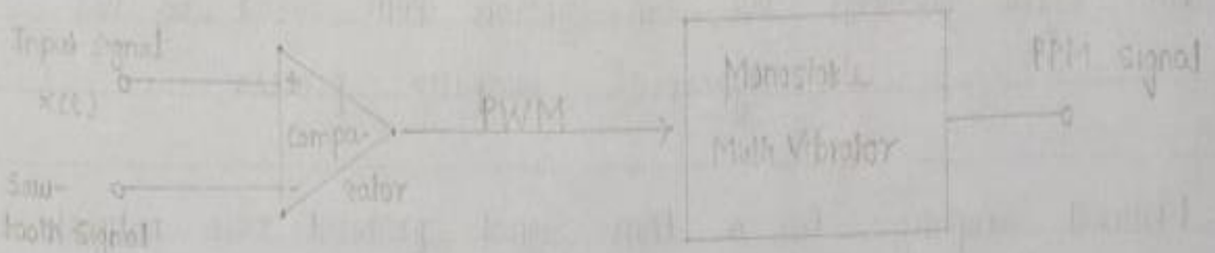
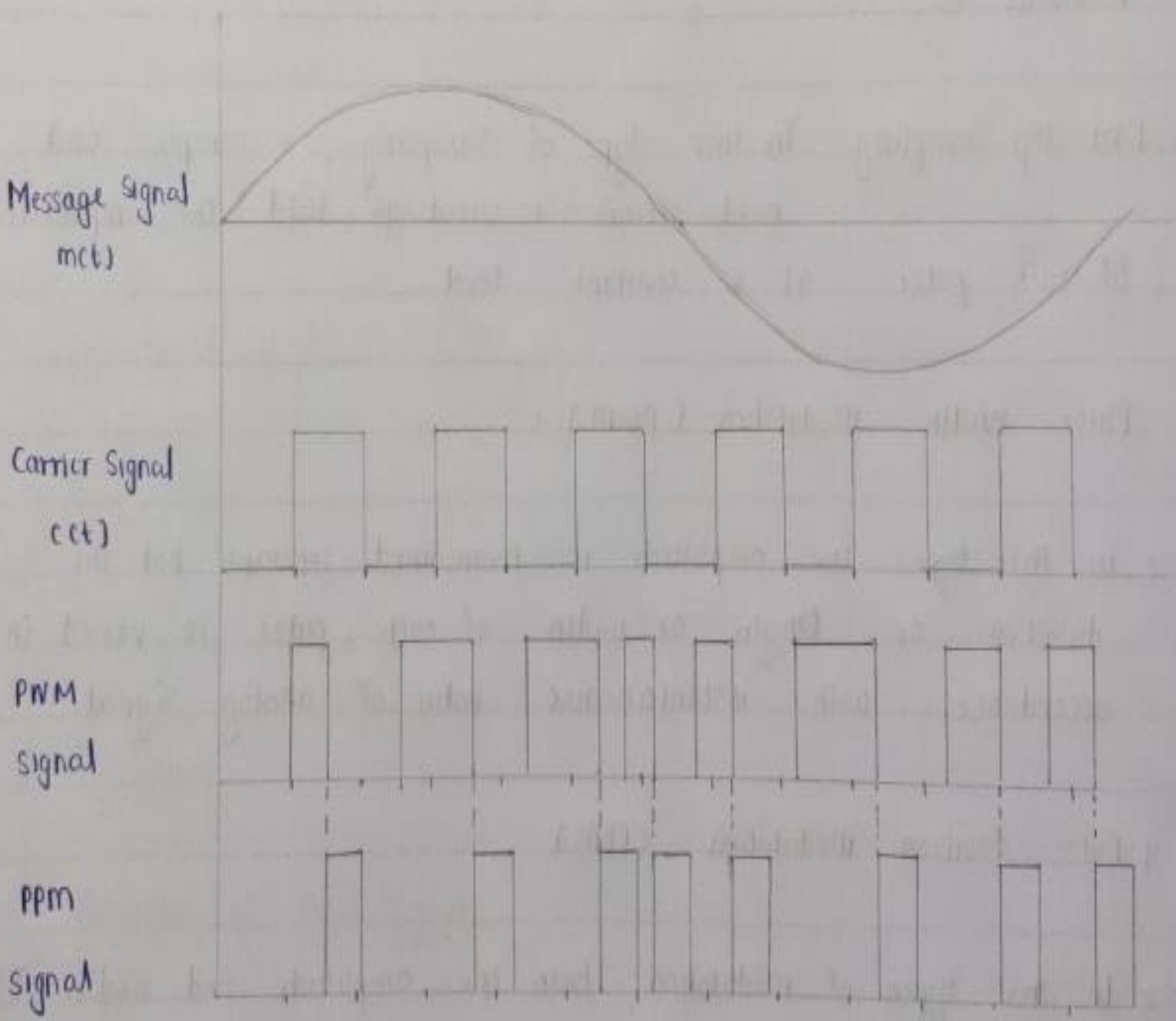
5) Comparison of PAM, PWM and PPM:

Sr.	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 ampli. of modulating signal.	The relative position of the Pulse is proportional to amplitude of mod. signal.
ii.	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.
iii.	Instantaneous power of transmitter varies	Instanteous power of transmitter varies	Instantaneous power of transmitter is const.
iv.	Noise interference is high interfere is high	Noise interference is minimum	Noise interference is minimum.
v.	System is complex to impliment.	System is simple to impliment.	System is simple to impliment.
vi.	Similar to Amplitude modulation.	Similar to frequency modulation	Similar to phase modulation.

Generation of PPM Signal





Waveform representation of PPM signal generation

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Conclusion : we successfully examined pulse Amplitude modulation, Pulse ~~position~~ position modulation and also verified their waveforms. we also illustrated Circuits for PAM & PWM.
We performed our experiment successfully using MATLAB.

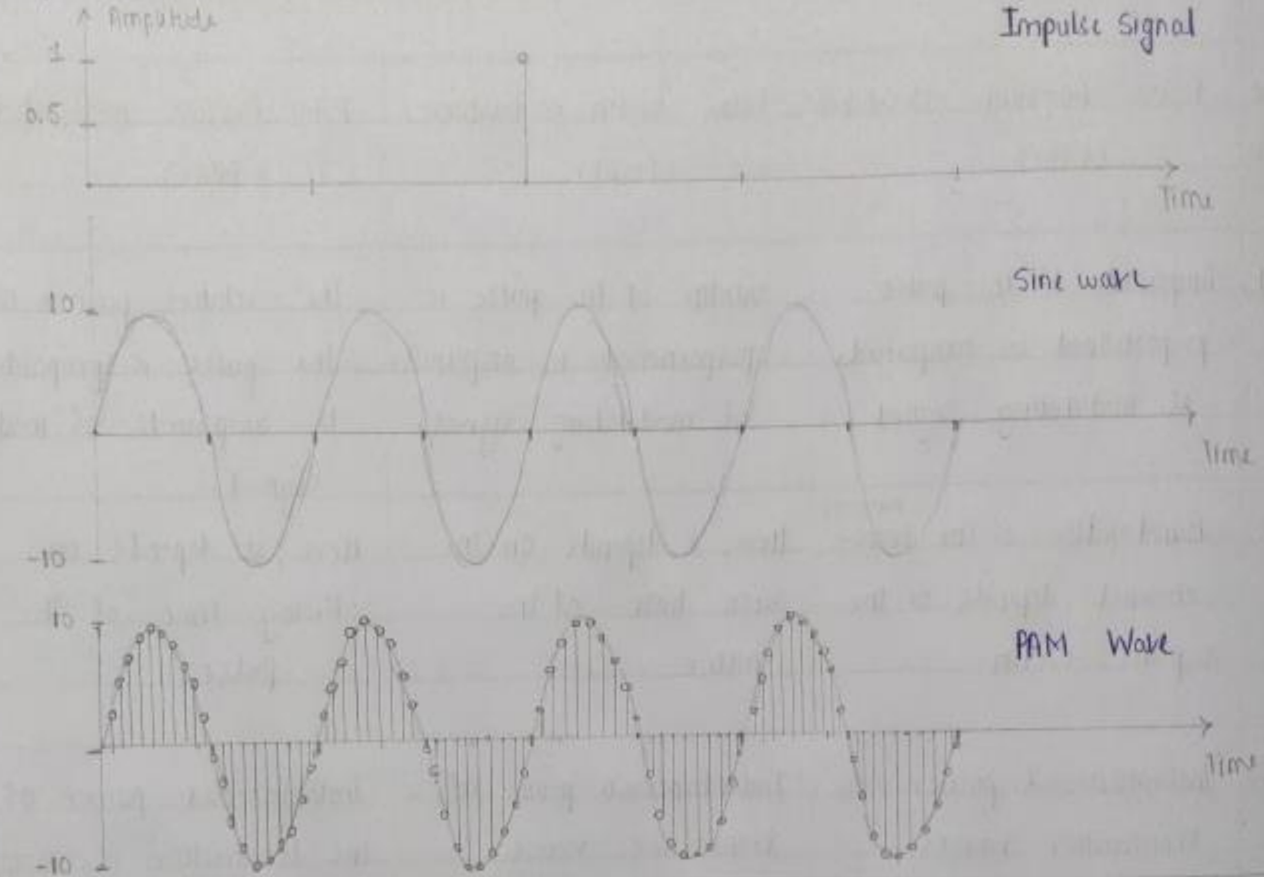
MATLAB CODE

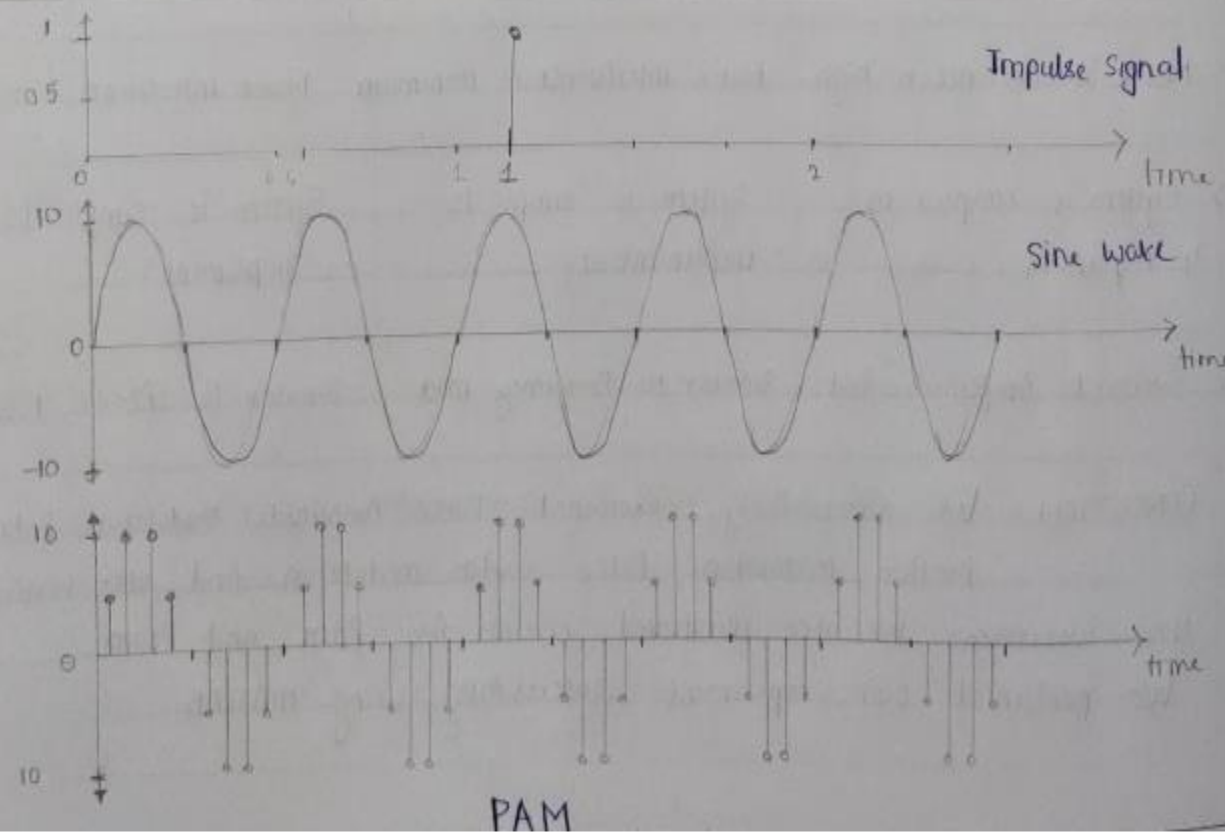
```
% PAM using Ideal Sampling

clc;
close all;
Clear all;
a = input ('Enter the amplitude = ');
f = input ('Enter the amplitude = ');
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');
x label ('Time');
y label ('Amplitude');
subplot (3,1,2);         % for sine wave plot
Plot (t, x2);
title ('sine wave')
x label ('time');
y label ('Amplitude');
subplot (3,1,3);         % for PAM waveplot
stem (t,y);
title ('PAM wave');
```

PAM with Ideal Sampling

1.) A = 1V   f = 2Hz





PAM

% PAM using Natural sampling.

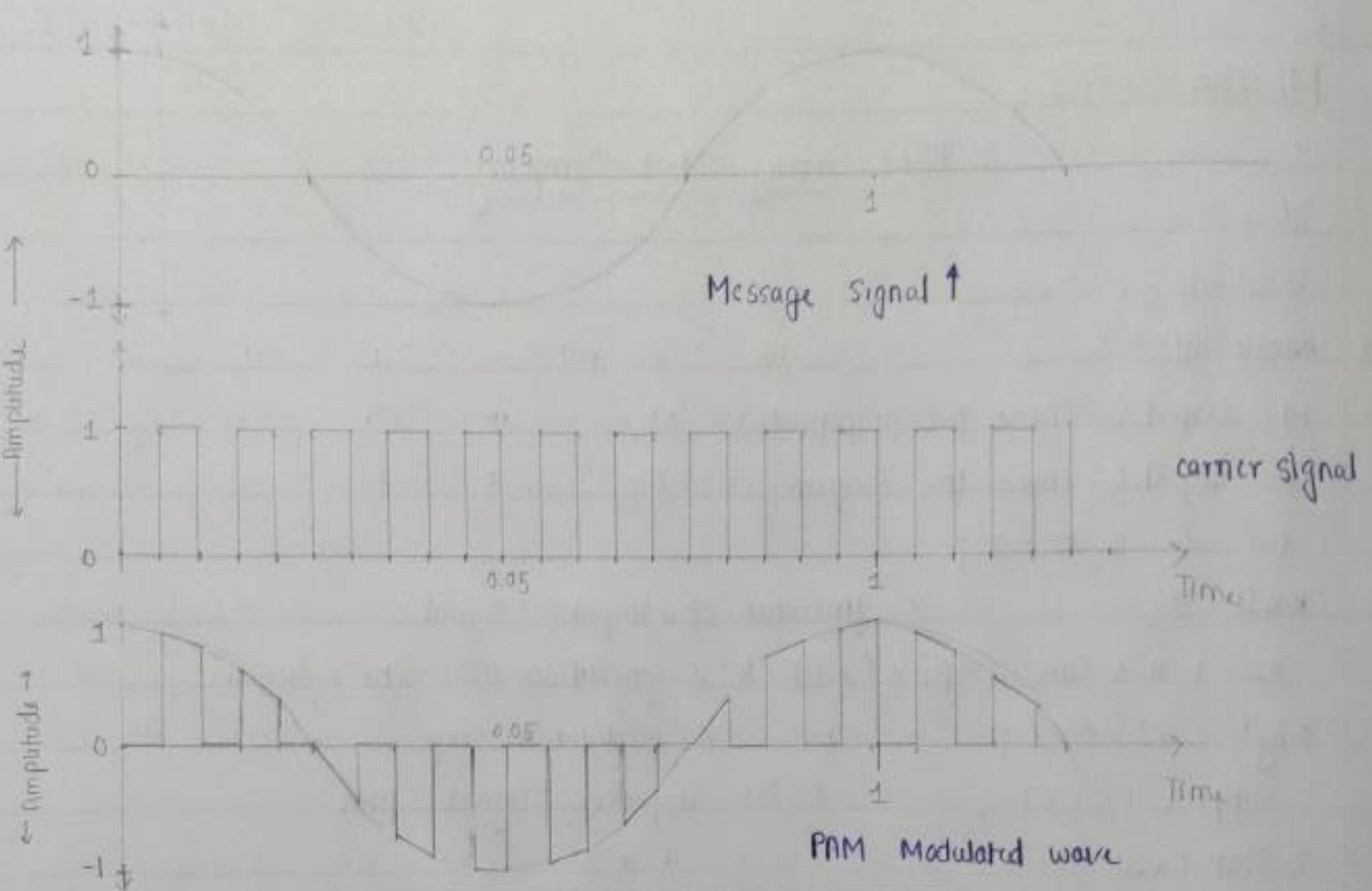
```
clc;   clear all ;  close all ;
fc  =100
fm  = fc/10
4 = 0 : 1/fs  : 4/sm ;
m sg sgl = cos(2 * pi * fm * t);
Carr sgl  =  0.5 x square (2 * pi * fc*t) + 0.5
mod sgl =  msg . sgl * carr. sgl
tt  = [] ;
for i=1  : length (mod. sgl):
        if  mod . sgl (1) == 0 ;
         tt  = (tt, mod - sg(i)) ;
    else
         tt = [tt , mod - sgl(i) + 2];
         end,
      fig (1)
```

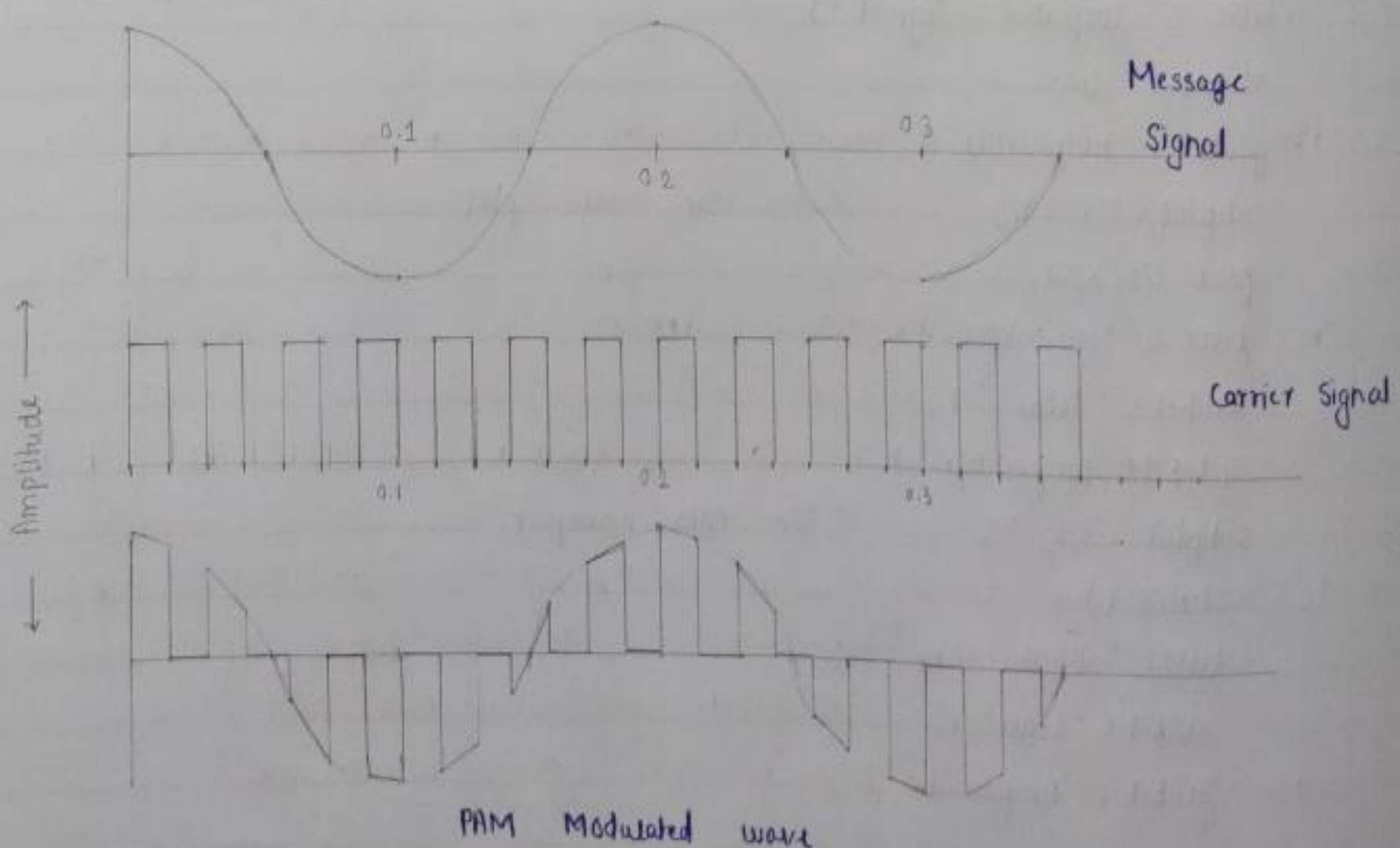
message	carrier	PAM modulated
Subplot (4,1,1);	Subplot (4,1,2);	Subplot (4,1,3);
Plot (t, msg. sgl);	Plot (t, carr.sgl);	Plot (t, mod.sgl);
title ('msg. signal);	title ('carrier signal')	title ('PAM modulated signal'
Xlabel ('Time Period');	X label ('Time Period');	X label ('Time Period')
Ylabel ('Amplitude');	Ylabel ('Amplitude');	Y label ('Amplitude').

PAM Using Square Wave

(Natural Sampling)

1) $F_c = 100$ $F_m = 10$ $F_s = 10{,}000$



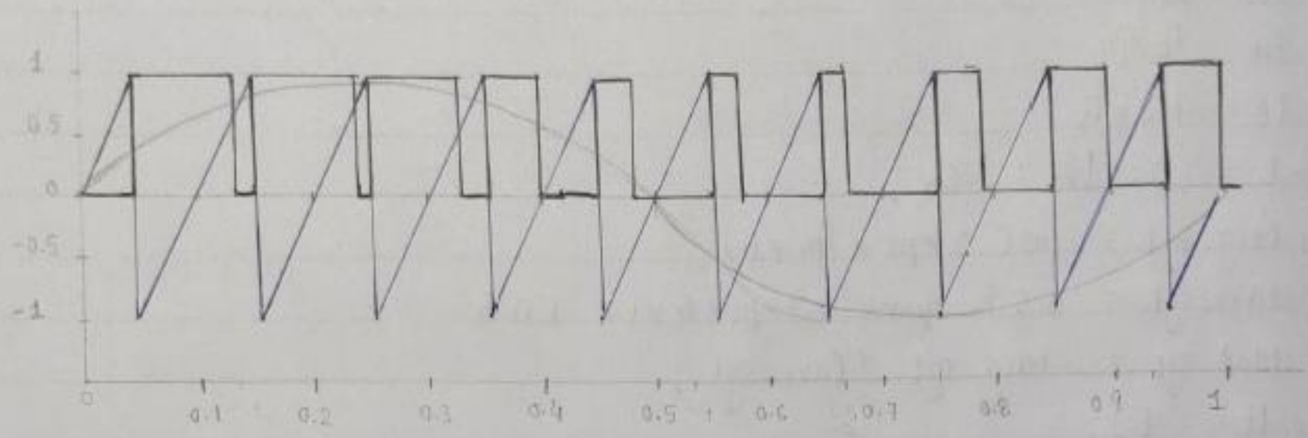


% PWM signal

```
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');
y label (' Amplitude');
axis ([0 1 -15 ,15]);
xlabel ('Time index');
title ('PWM wave');
grid on;
```

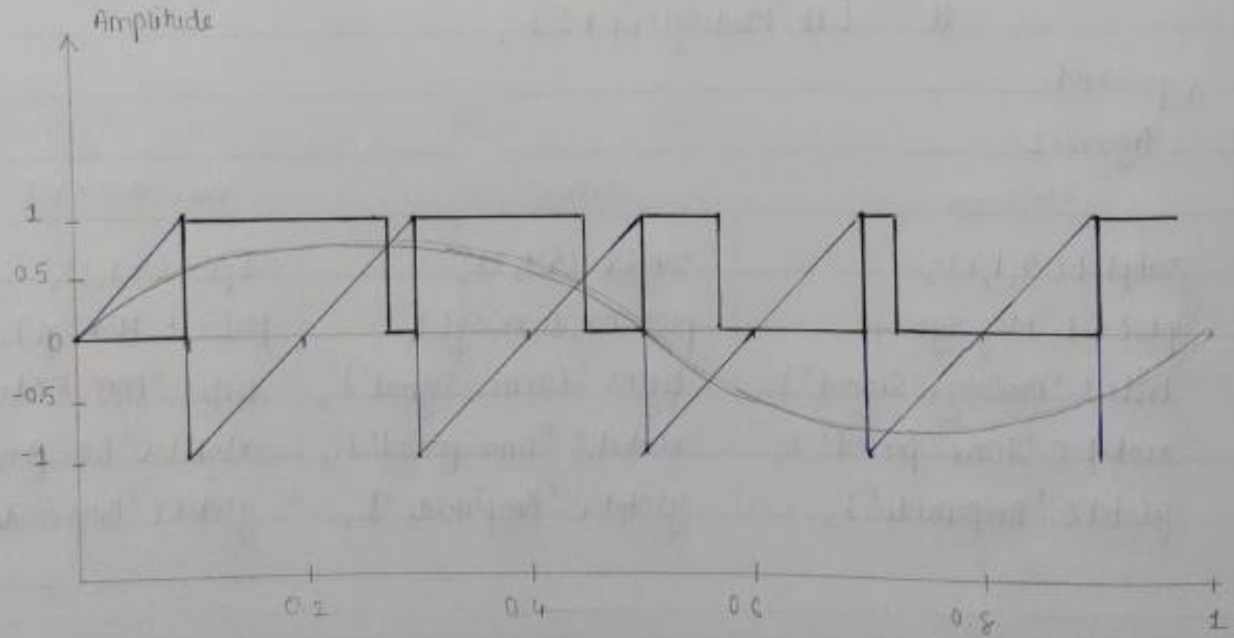
PWM

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

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% PPM signal

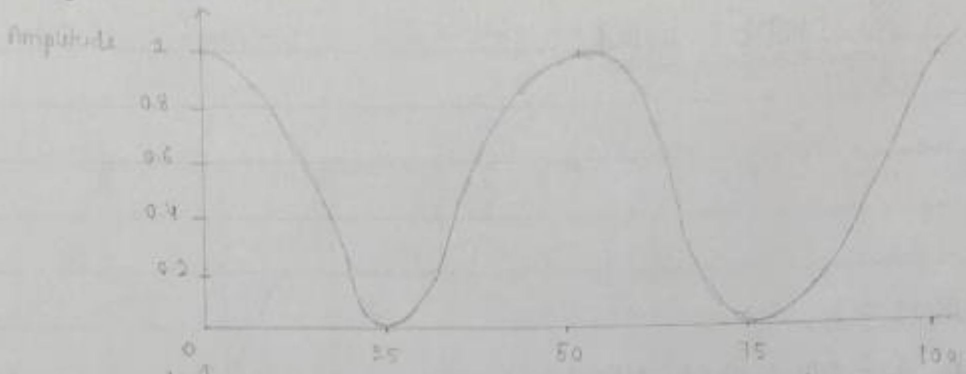
```
clc;
clear all;
Close all;
fc = 10;
fs =100;
fm= 2
t = 0: 1/fs :((2/fm) - (1/fs));
x = 0.5 * cos (2* pi * fm * t) +0.5
Y = modulate (x, fc, fs, 'ppm');
subplot { 2,2, 1};
plot (x);
title ('msg signal')

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

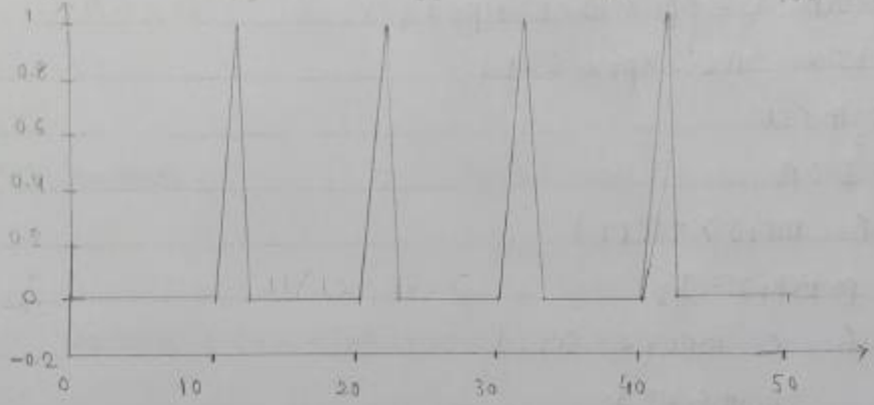
Conclusion : we have sucessfully examined pulse Amplitude Modulation, pulse position modulation & also verified their waveforms using MATLAB.

PPM Signal

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

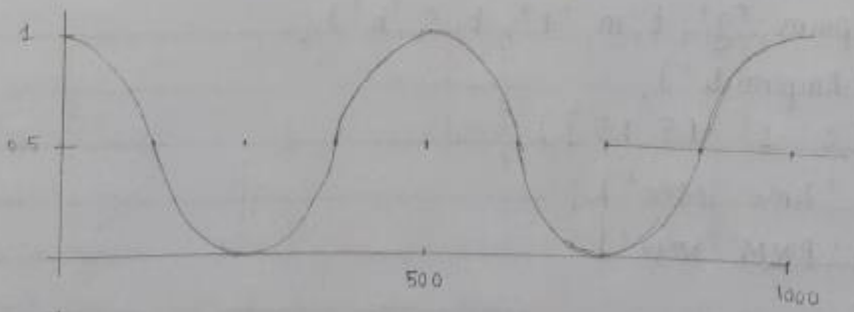


Msg Signal

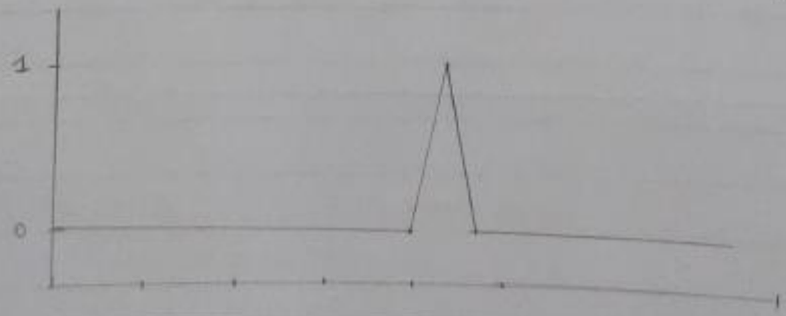


PPM

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



Msg Signal



PPM

Date :
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EXPERIMENT - 6

ASK, FSK and PSK

AIM : To study Amplitude shift keying (A.S.K), frequency shift keying (F.S.K) and phase shift keying (PSK) modulation Technique & verify uniforms.

Apparatus : MATLAB.

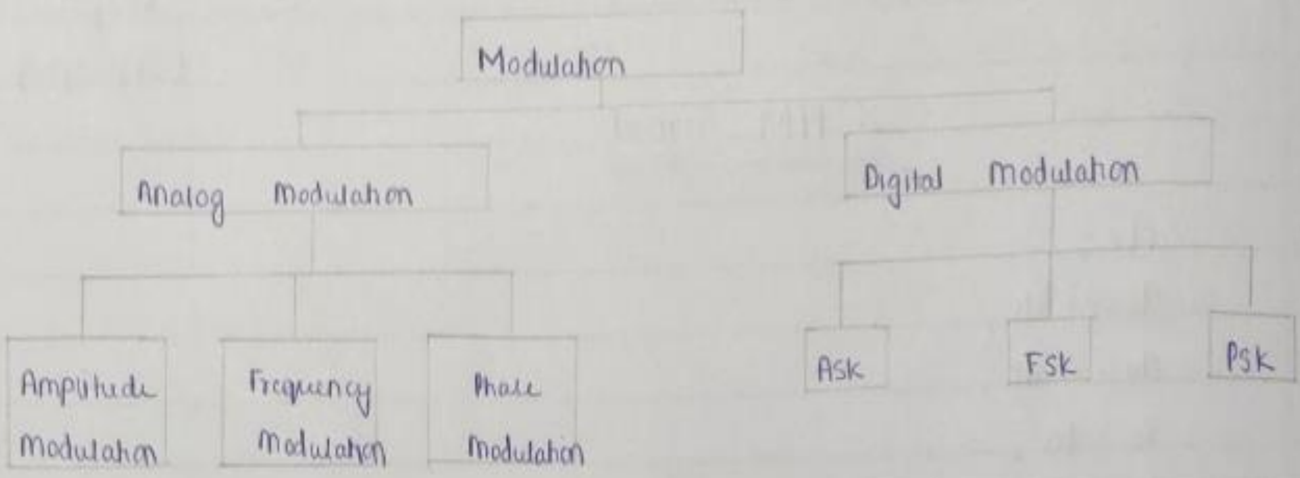
THEORY : 1) modulation : Modulation is a process by which some characteristics of a carrier wave is varied in accordance with a modulating (message) signal.

Digital modulation : It is a special kind of modulation where the message signal is digital in nature and the carrier wave is analog (sinusodial) in nature.

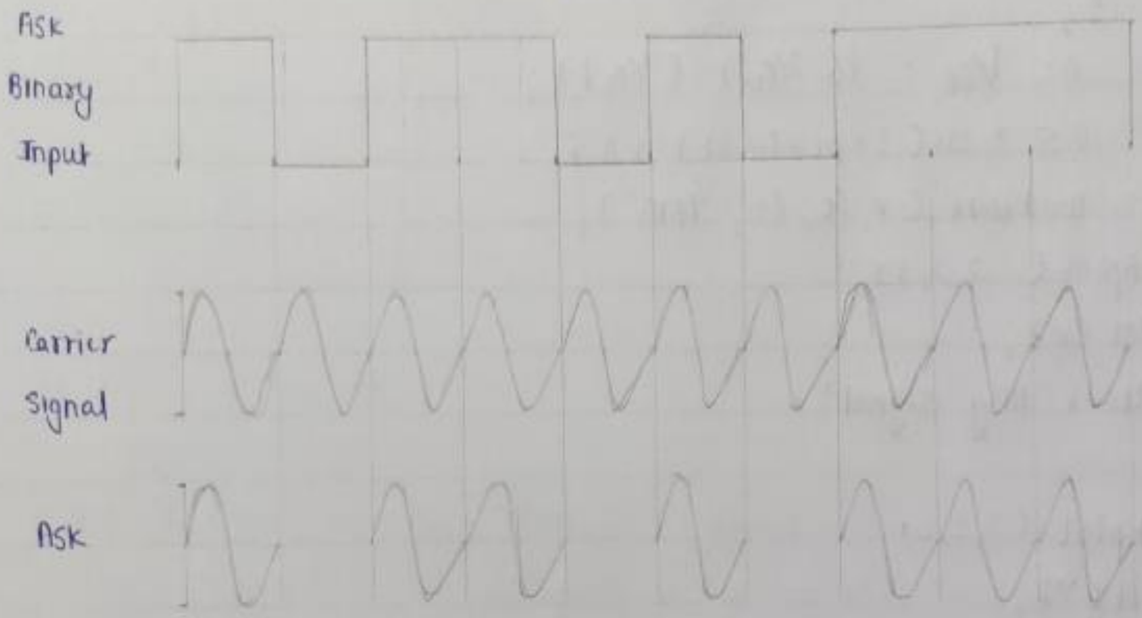
The ASK, FSK and PSK are analogues to Am, fm and pm respectively. The difference is that in digital modulation techniques (ASK, FSK & PSK) the modulation signal is digital in nature. while in Am, fm & Pm modulating signal is analog in nature.

2) ASK (Amplitude shift Keying)

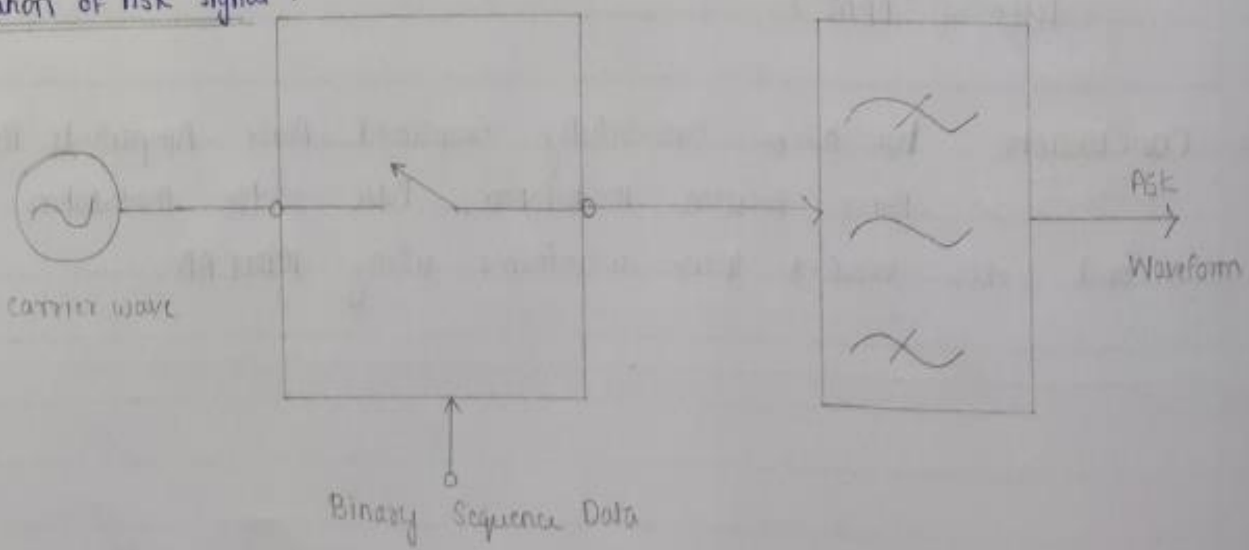
→ In ASK, the amplitude of the Carrier wave is changed (switched) according to the digital input Signal (modulating signal).



ASK



Generation of ASK signal :



Application of ASK : i) wireless Base stating
ii) low frequency RF Application
iii) Industrial Network Devices.

3) FSK (Frequency shift keying)

→ If the frequency of sinosoidal carrier wave is varied (switched) depending on the digital input signal, then it is known as the frequency shift keying.

→ Application of FSK :-

1. High Frequency Radio Transmission.

4) PSK (Phase shift keying)

→ In PSK, phase of the carrier wave (analog in nature) is switched as per the input digital signal.

Application of PSK:

1. It is widely used for wireless LANs, RFID & Bluetooth Communication.

ii. It is widely

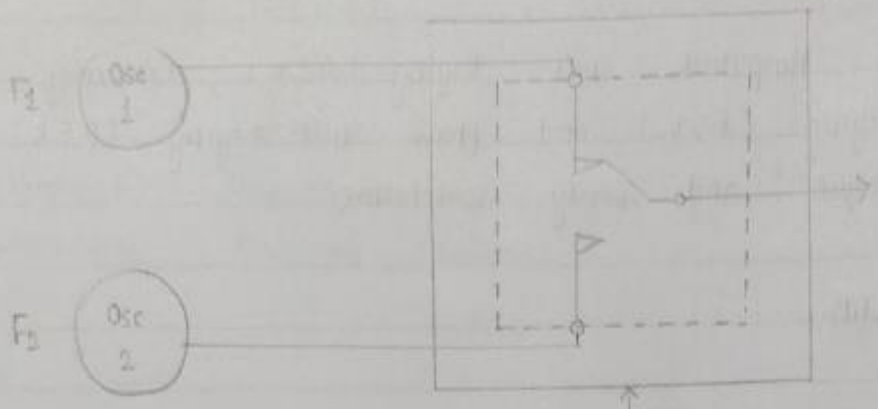
MATLAB CODE

```
%% ASK

clc ; close all;   clear all;  % for deleting all variable from memory
fc = input ('enter the freq of sine wave carrier');
```

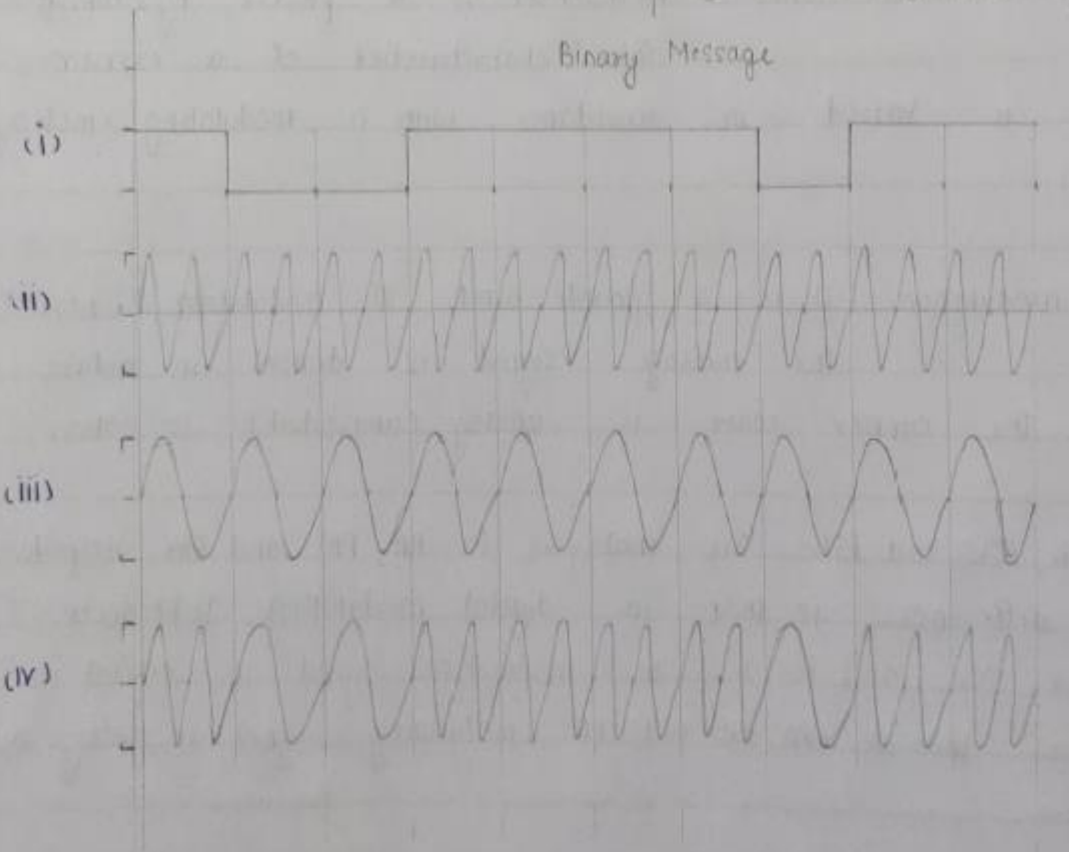
FSK

Generation of FSK



$F_1 \cong F_2$

Bit rate $F_s \ll F_1$



(i) Digital Bitstream

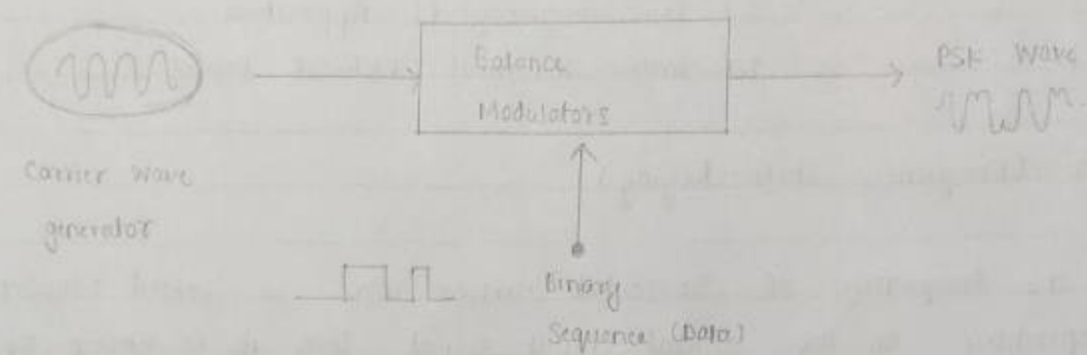
(ii) High frequency carrier wave

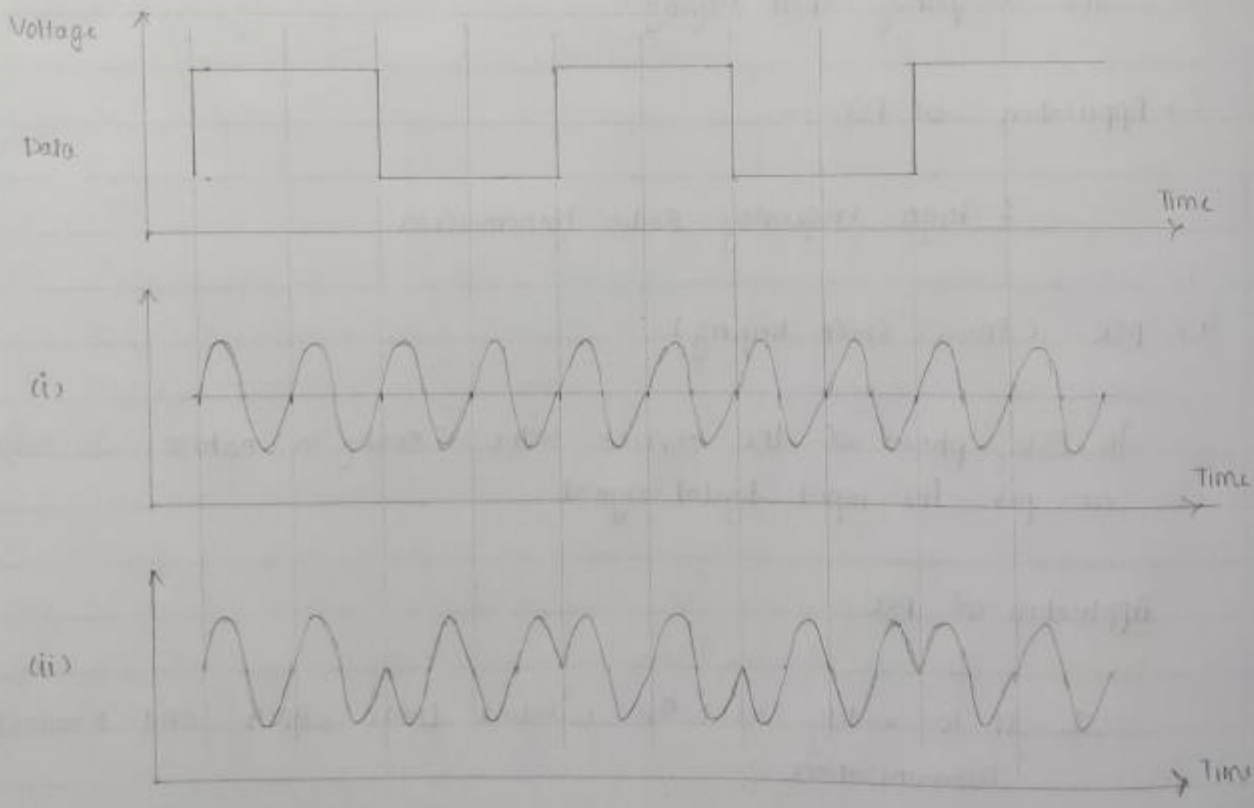
(iii) Low frequency carrier wave

(iv) FSK modulated wave

PSK

Generation of PSK





(i) Carrier Frequency Before modulation
(ii) Carrier Frequency After modulation

```
fp = input ('Enter the freq of Periodic Binary Pulse (message):');
amp = input ('Enter the amplitude (for carrier & Binary Pulse message):');

t = 0: 0.001 : 1;      % for setting the sampling interval
c = amp .* sin (2* pi * fc * t);  % for Generating carrier
                                    sine wave.

subplot (3,1,1) ; % for plotting carrier wave.
plot (t,c)
xlabel ('Time')
ylabel ('Amplitude')
title ('carrier wave')
m = amp/2 * sq. (2*pi*fp*t ) + (amp/2); % sq.wave msg
subplot (3,1,2)  % Plotting sq. Binary pulse.
plot (t,m)
xlabel ('Amplitude')
title ('Binary message pulses')
w = c*m    % The shift keyed wave
subplot (3,1,3)
plot (t,w)
xlabel ('Time')
ylabel ('Amplitude')
title ('Amplitude shift keyed signal')
```

%% FSK

```
clc; closeall; clear all;
fc1 = input ('Enter the freq of 1st sine wave carrier');
fc2 = input ('Enter the freq of 2nd sine wave carrier:');
fp = input ('Enter the freq of Periodic Binary pulse (message):');
amp = amp/2
```

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```
t = 0: 0.001 : 1;
C1 = amp * sin (2*pi * fc1 *t);
C2 = amp * sin (2*pi * fc2 * t);
```

```
(i) subplot (4,1,1);
    Plot (t, c1)
    x label ('Time')
    y label ('Amplitude')
    title ('carrier 1 wave')
```

```
(ii) subplot (4,1,2)
     Plot (t, c2)
     xlabel ('Time')
     ylabel ('Amplitude')
     title ('carrier 2 wave')
```

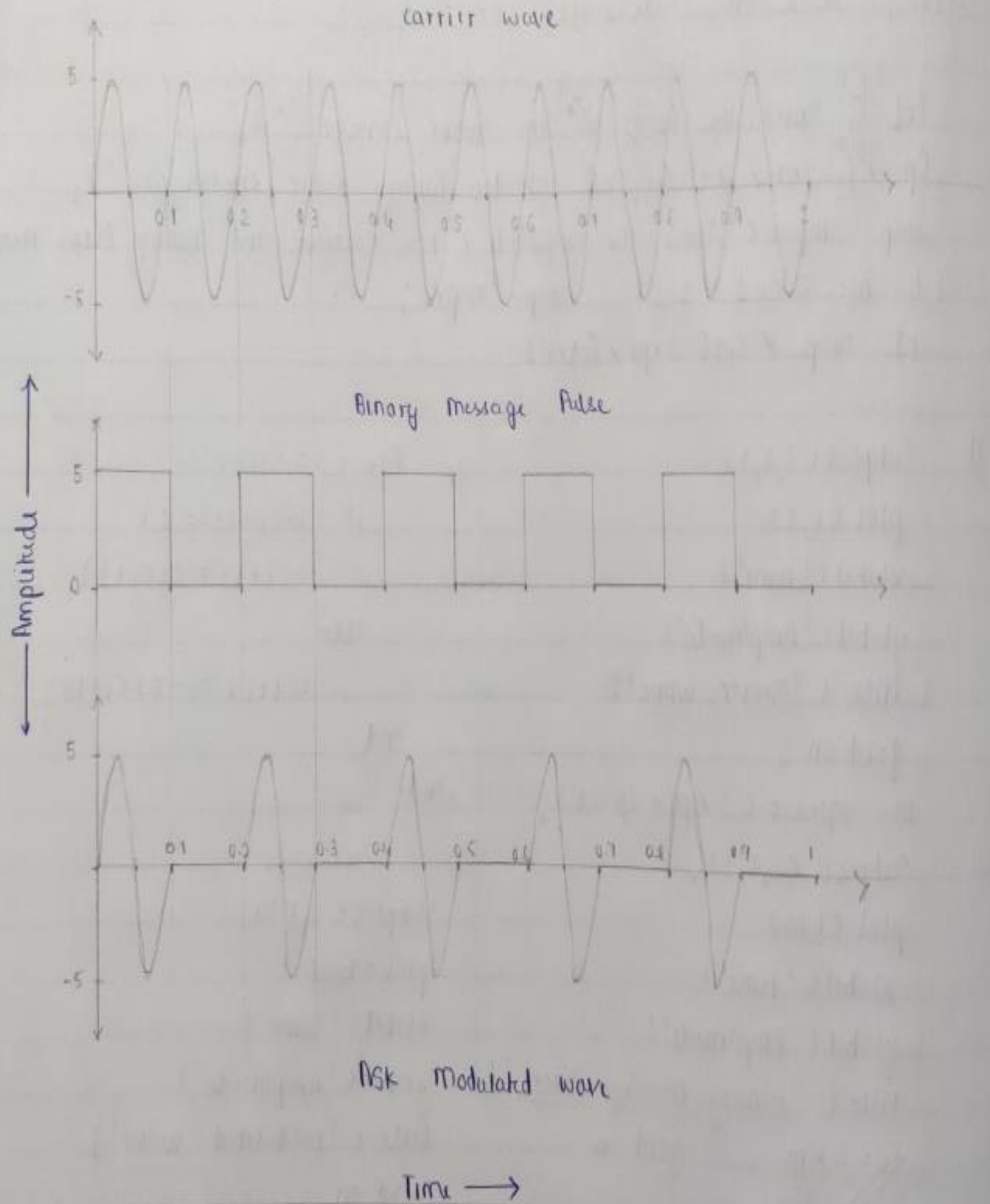
```
(iii) m = amp * square (2*pi * fp*t + amp % sq wavefo
      Subplot (4,1,3)
      Plot (t,m)
      x label ('Time')
      y label ('Amplitude')
      title ('Binary msg pulses')
```

```
(iv) For i=0; 1000
       if (m(i+1) == 0
          mm (i+1) = c2 (i+1);
       else
          mm(i+1) = c1 (i+1);
       end
     end.
```

```
(v) subplot (4, 1, 4)
    Plot (t, mm)
    xlabel ('Time')
    y label ('Amplitude')
    title ('modulated wave')
```

(A) ASk:

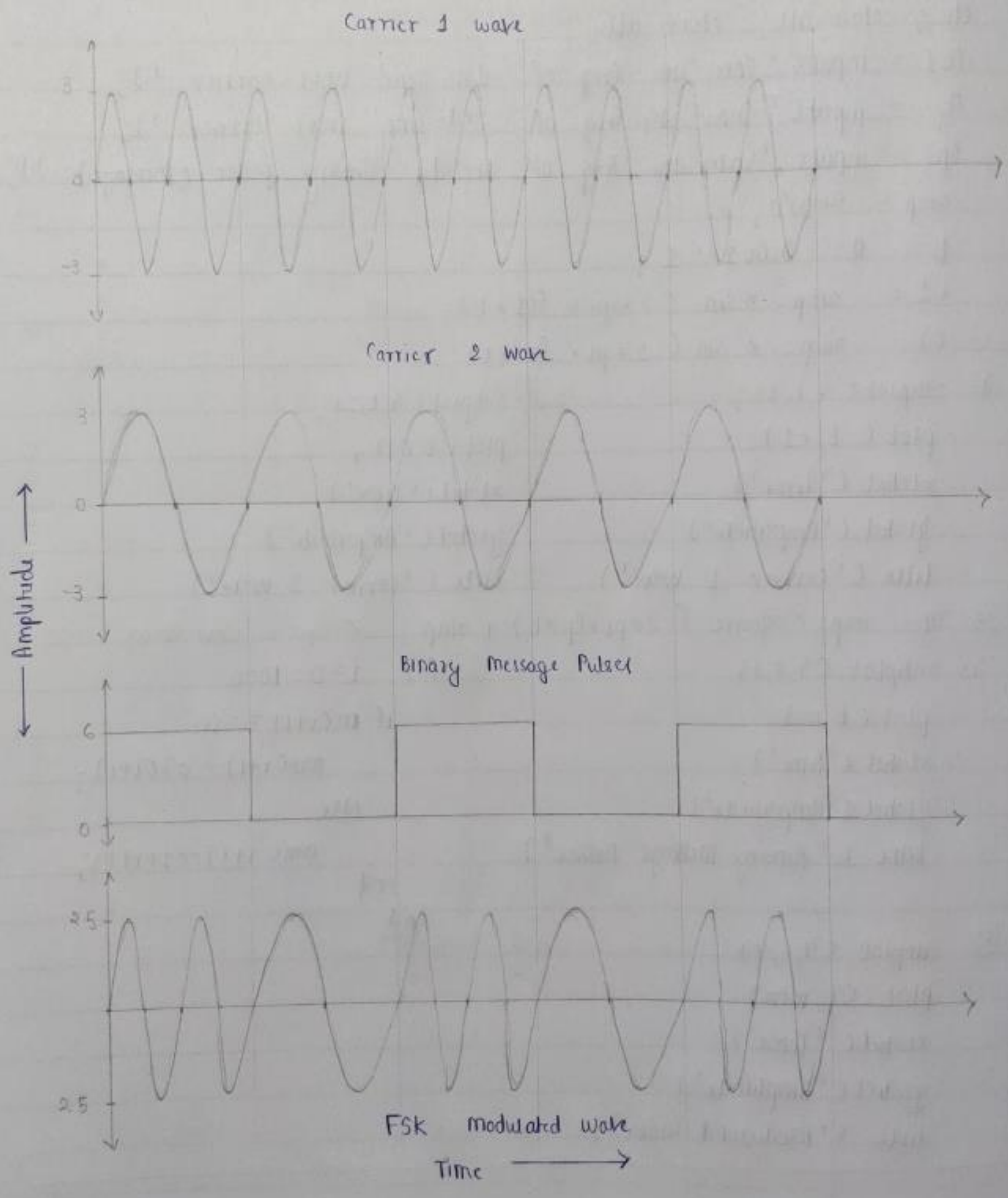
3.) $F_c = 10\text{ Hz}$ $F_p = 5\text{ Hz}$ Amp : 5



(8)

$\boxed{\text{FSK :}}$

2) $F_{c1} = 30$ $F_{c_2} = 10$ $F_p = 5$ amp = 6



%% PSK

```
clc; close all; clear all;
fc1 = input ('Enter the freq. of 1st sine wave carrier:');
fp = input ('Enter the freq. of Freq of periodic Binary pulse (msg):');
amp = input ('Enter the amplitude (for carrier and binary pulse msg
t = 0 : 0.001 : 1;          amp = amp/2
c1 = amp * sin (2*pi*fc1*t);
```

```
(i) subplot (3,1,1)
    plot (t,c)
    xlabel ('Time')
    title ('Carrier wave')
    grid on;
    m = sq.(2*pi*fp*t);

(ii) subplot (3,1,2);
    plot (t,m)
    xlabel ('Time')
    ylabel ('Amplitude')
    title ('Binary msg Pulses')

(iii) for i = 0 : 1000
      if (m(i+1) == 1)
          s(i+1) = c1(i+1)
      else
          s(i+1) = -c1(i+1);
      end
      end

(iv) subplot (3,1,3)
    plot (t,s)
    xlabel ('Time')
    ylabel ('Amplitude')
    title ('modulated wave')
    grid on
```

Conclusion

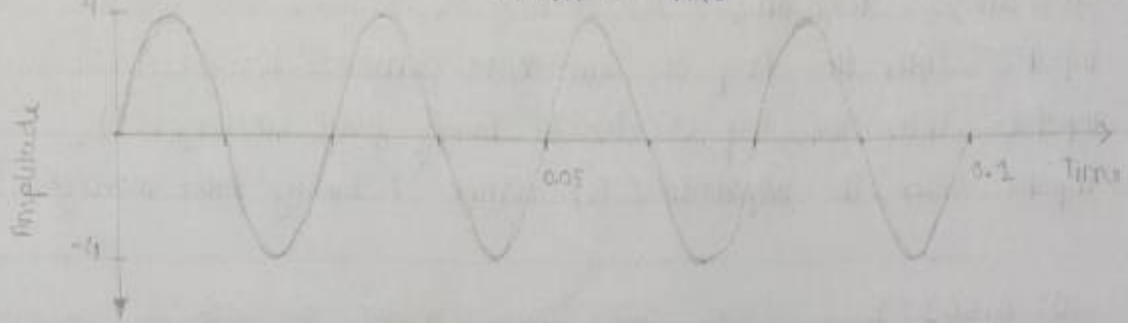
1) We have successfully studied ASK, PSK & FSK. modulation technique and verified their Waveforms using MATLAB. we also observed the schematic diagrams for ASK, FSK & PSK.

Waveforms using MATLAB

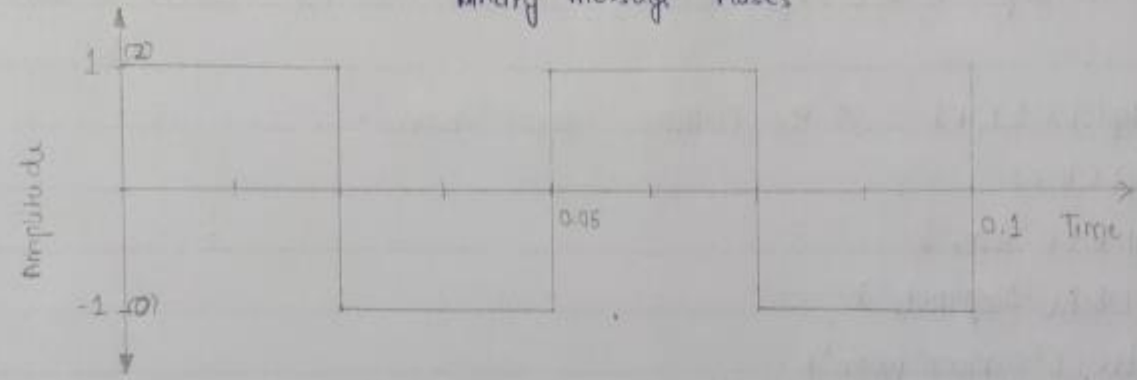
PSK :

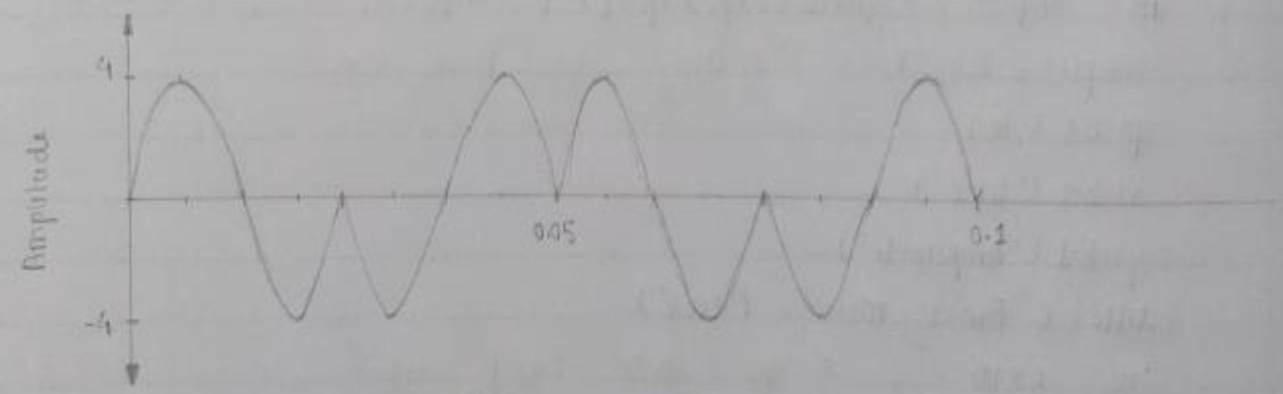
1.) $F_{c1} = 40\,Hz$ $F_p = 20\,Hz$ $Amp = 8\,V$

Carrier 1 Wave



Binary message Pulses





Modulated Wave