LABORATORY JOURNAL

Submitted in partial fulfillment of the requirement
For the Subject

"DIGITAL COMMUNICATION"

(EC 209)

: Prepared & Submitted By :

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(Admission No. U19CS012)

B. TECH. II (CSE) 3rd Semester
(Academic Year : 2020-21)
ONLINE MODE

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(Aug to Dec - 2020)

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ELECTRONICS ENGINEERING DEPARTMENT

Academic Year : 2020-21

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. BHAGYA VINOD RANA** bearing **Admission No.U19CS012** in the partial fulfillment of the requirement for the **Subject Digital Communication (EC209)** through ONLINE MODE.

We, certify that the work is comprehensive, complete and fit for evaluation.

Laboratory Teachers :

Name	**Signature with date**
1. Prof. N. B. Kanirkar	
2. Prof. M. C. Patel	
3. Mr. Abhishek Tripathi	

Aug-Dec. 2020.

DIGITAL COMMUNICATION (EC209)
(Academic Year 2020-21)

LIST OF EXPERIMENTS
ONLINE MODE

- Submitted By

Admission Number : U19CS012
B.Tech. II (CSE) 3rd Semester

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.

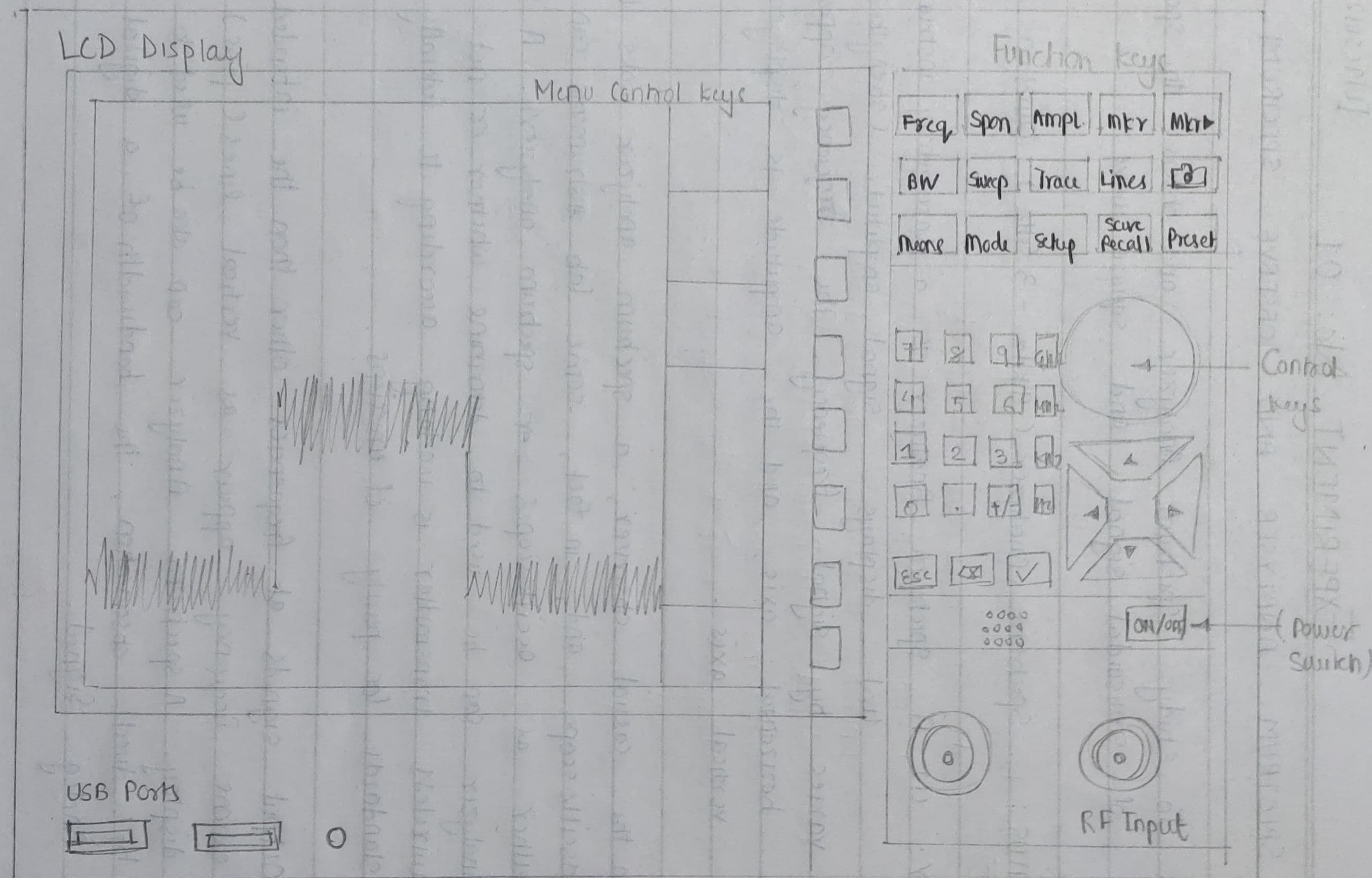
(*) **APPARATUS:** Spectrum Analyzer (9 kHz - 3 GHz) Function Generator

(*) **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.



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.

(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

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 (dBμV)
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
dBμV = -3.01 to 126.99 dBμV, 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

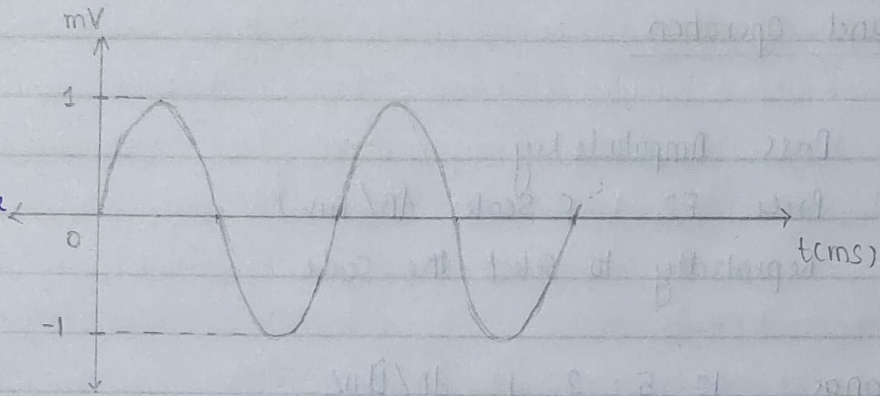
Waveform : SQUARE

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

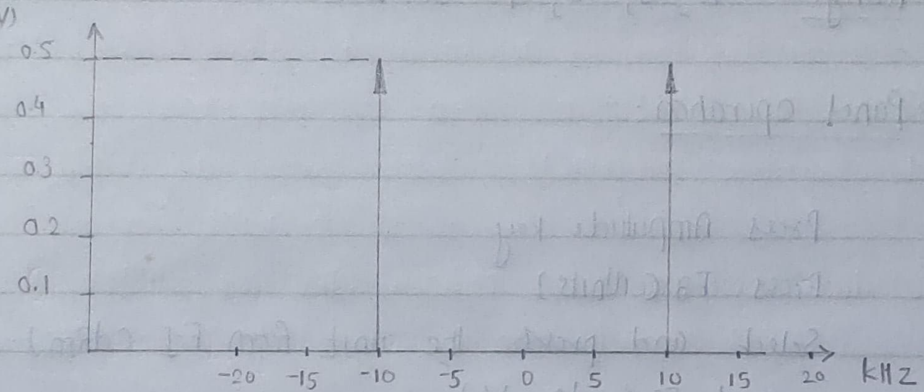
SINE WAVE

1.) A = 1 mV   f = 10 KHz

Oscilloscope

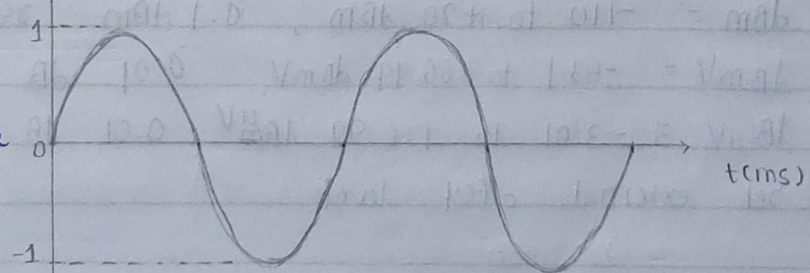


Spectrum Analyzer

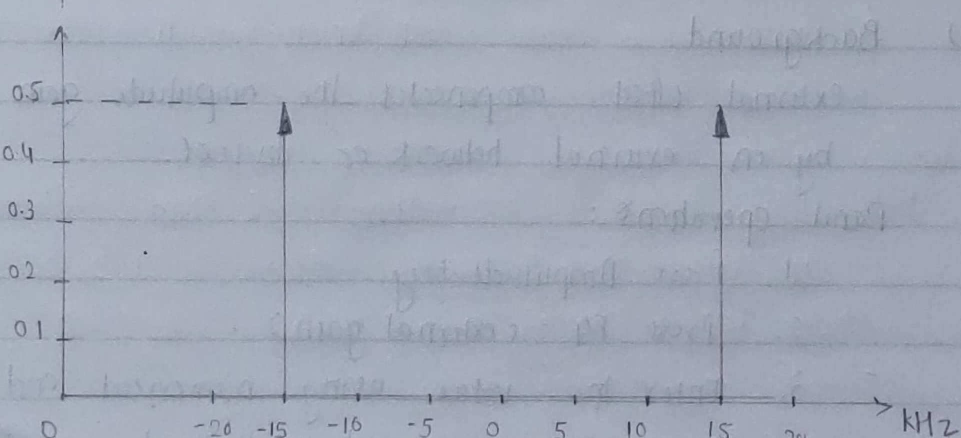


2.) A = 1 mV   f = 15 KHz

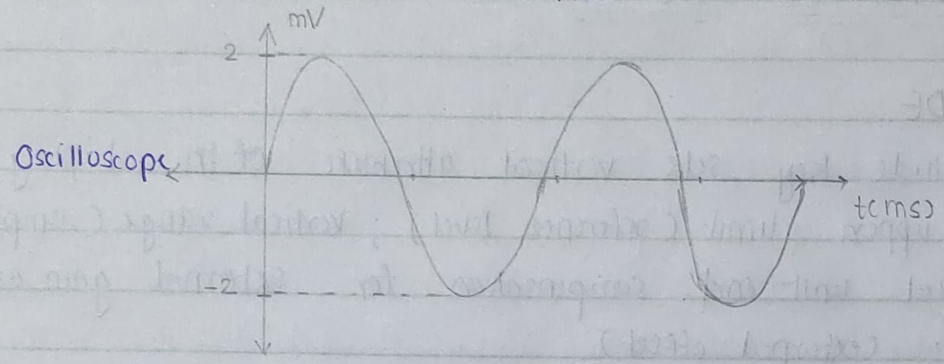
Oscilloscope

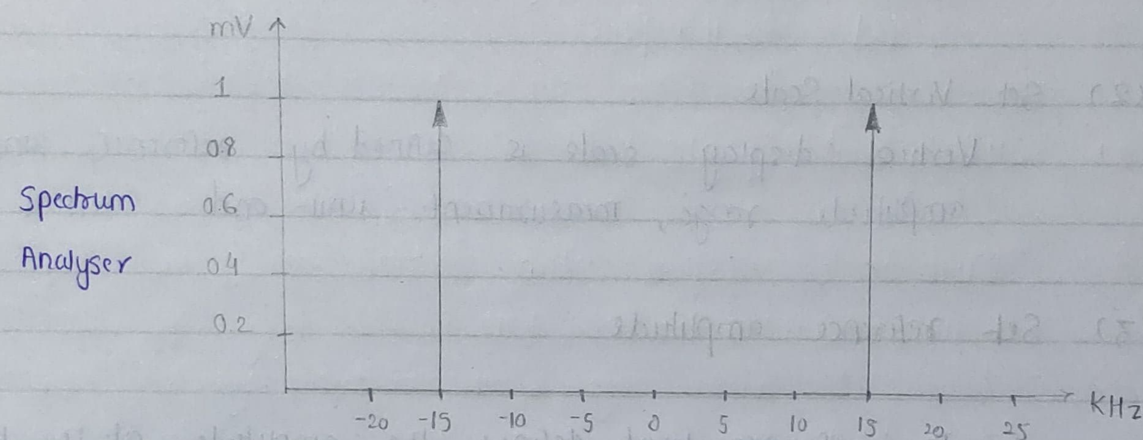


Spectrum Analyser

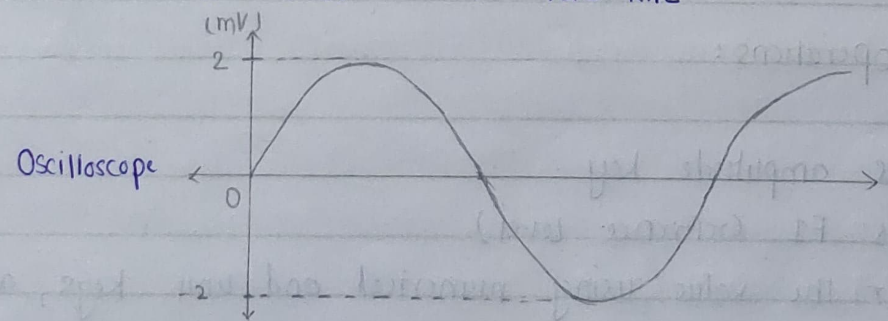


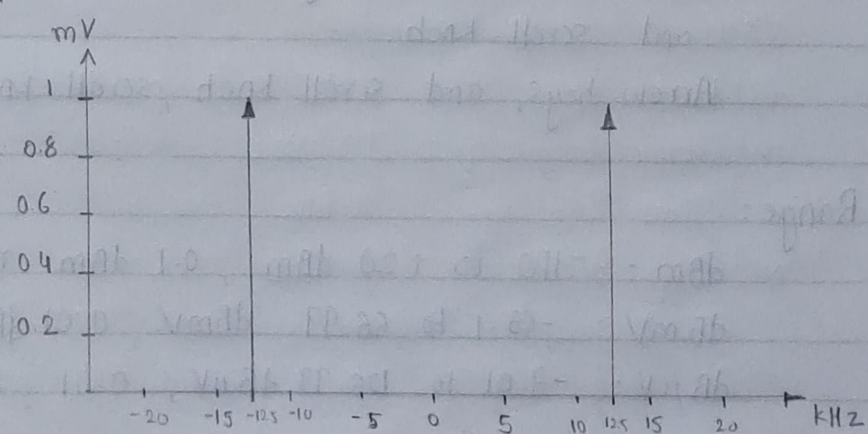
3.> A = 2mV f = 15 KHZ





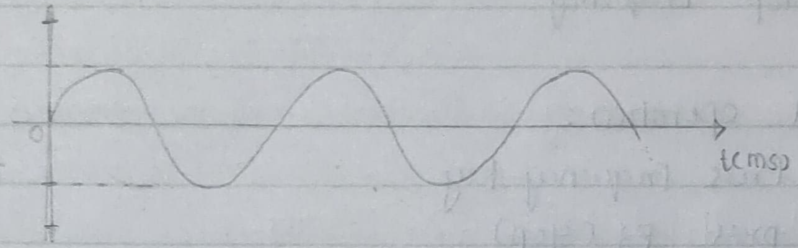
4.> A = 2 mV f = 12.5 KHZ



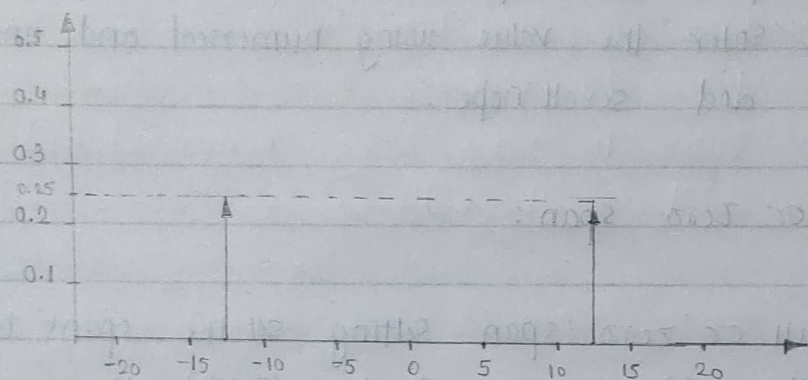


5.) A = 0.5 mV f = 12.5 KHz

Oscilloscope

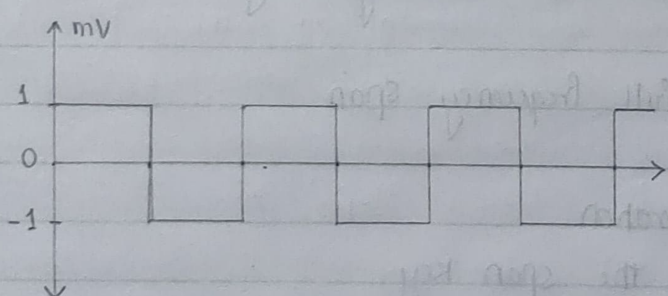


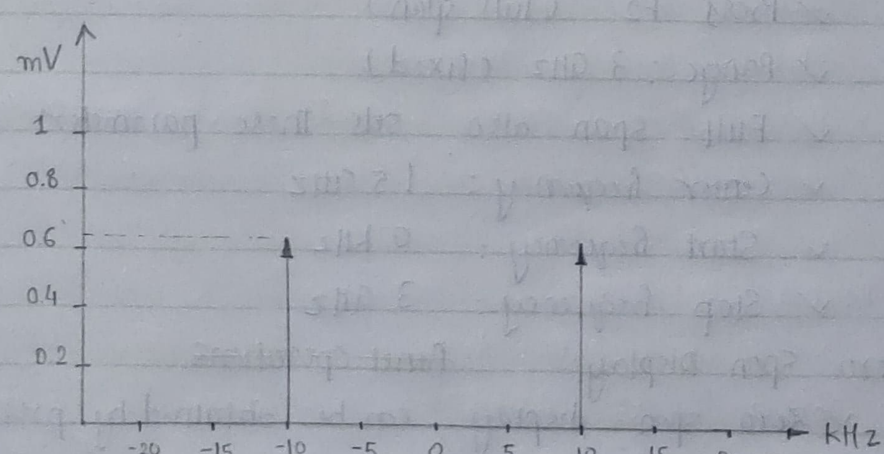
Spectrum Analyser



SQUARE WAVE

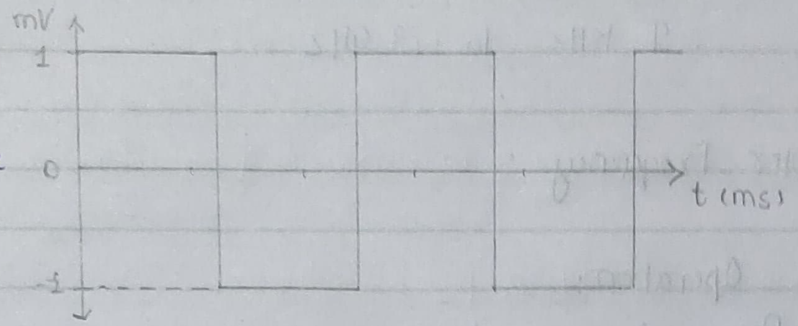
1.) A = 1 mV f = 10 kHz



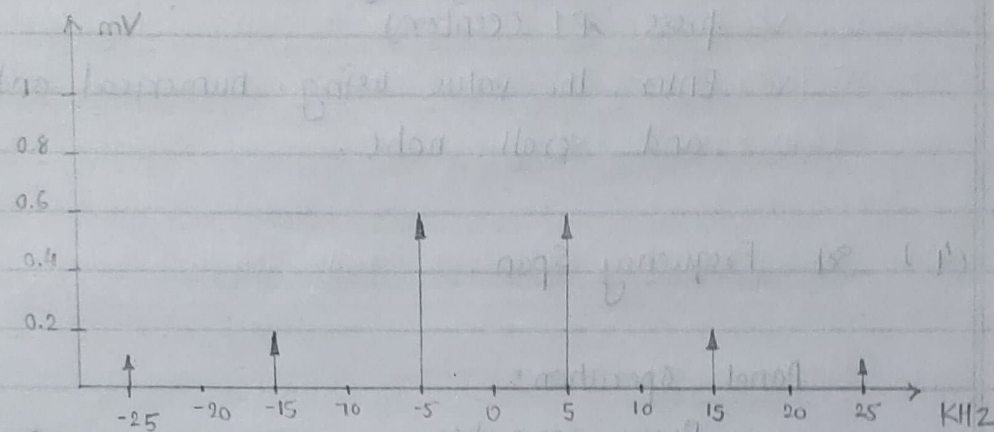


2.> A = 1 mV f = 5 KHz

Oscilloscope

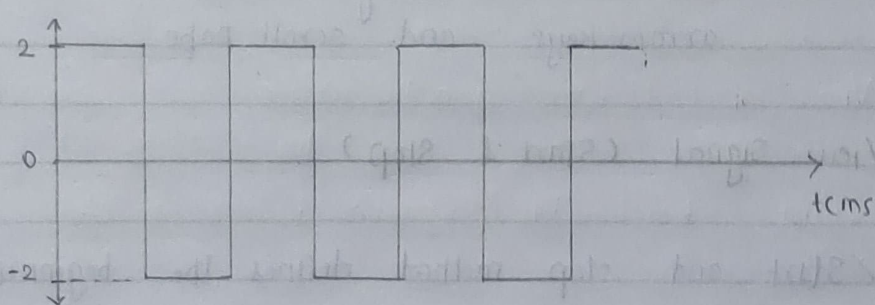


Spectrum Analyser

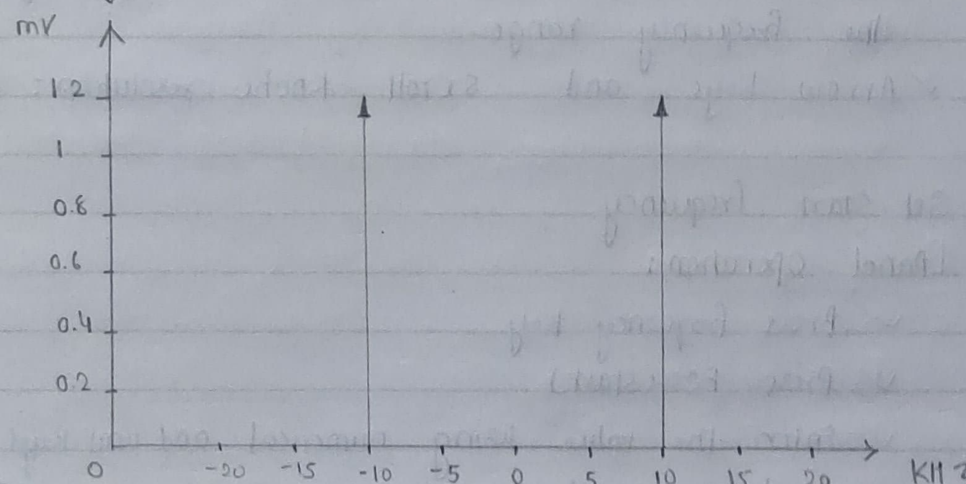


3.> A = 2 mV f = 10 KHZ

Oscilloscope



spectrum Analyser



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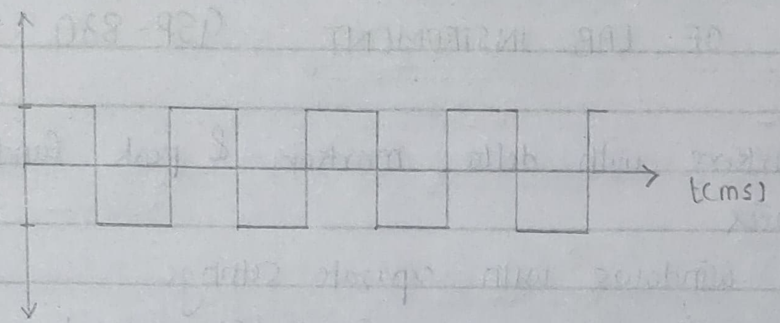
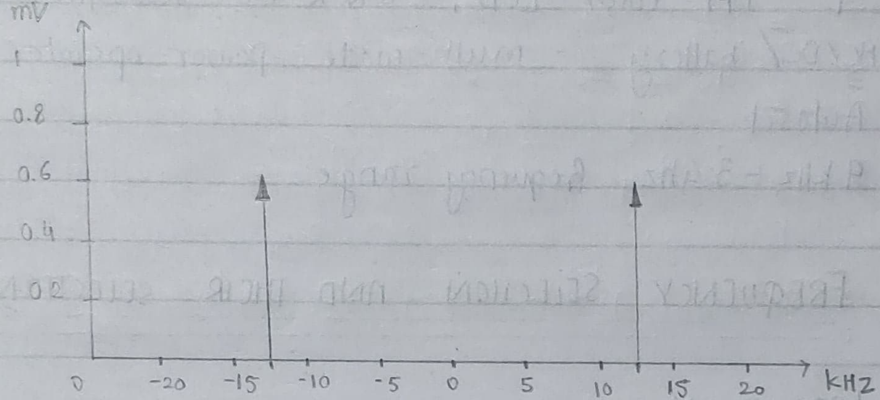
4.> A = 1 mV f = 12.5 KHZ

Oscilloscope

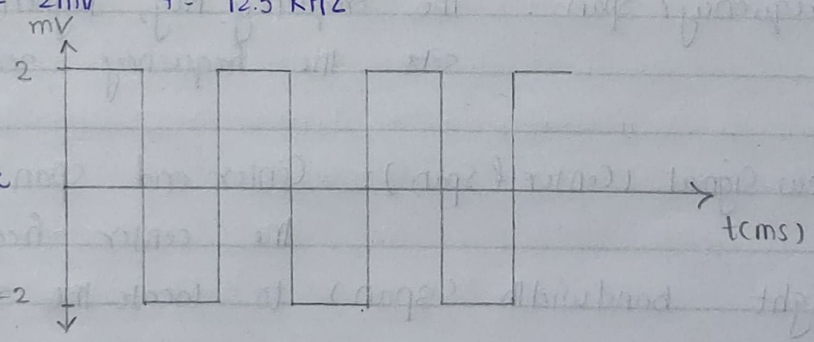


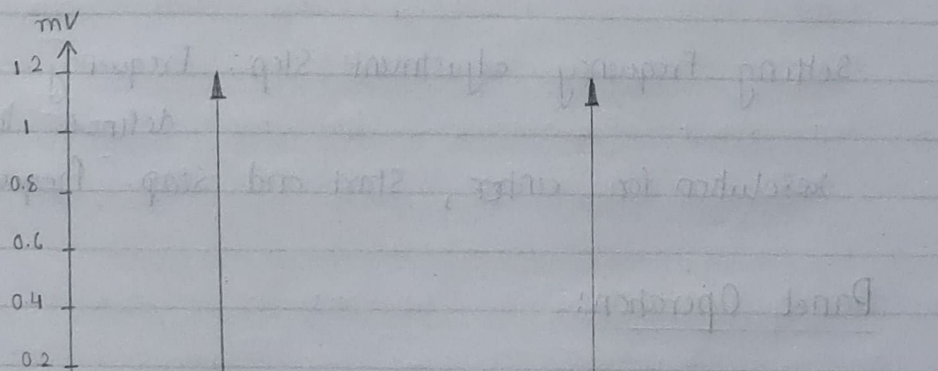
Spectrum Analyzer



5.> A = 2mV f = 12.5 KHZ

Oscilloscope





EXPT. NO. 2 NAME: SAMPLING AND NYQUIST CRITERIA

Page No.: 1 Date: 26 Aug

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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.

Teacher's Signature:

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

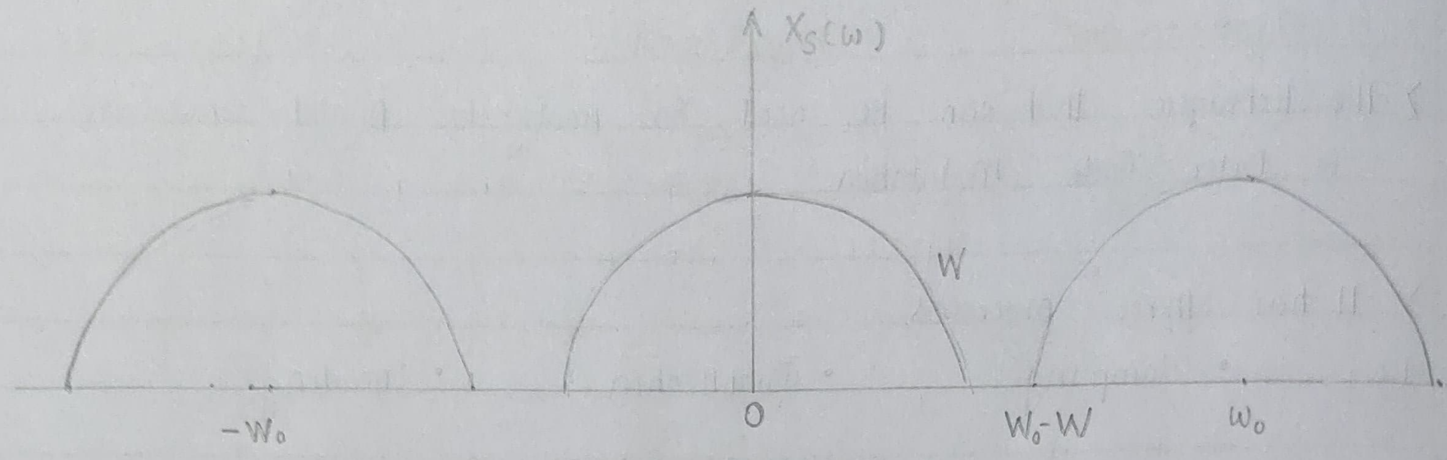
> It has three processes
- Sampling
- Quantization
- Encoding

(2) Sampling

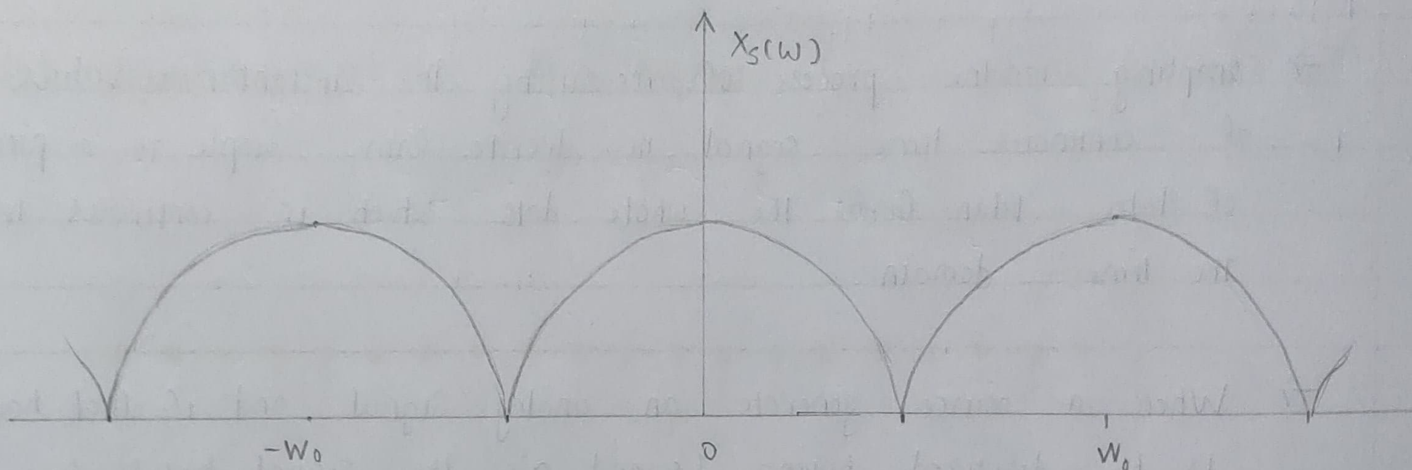
-> Sampling is the process of measuring the instantanous values of continous - time signal in discrete form. Sample is a piece of data taken from the whole date which is continous in the time domain.

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

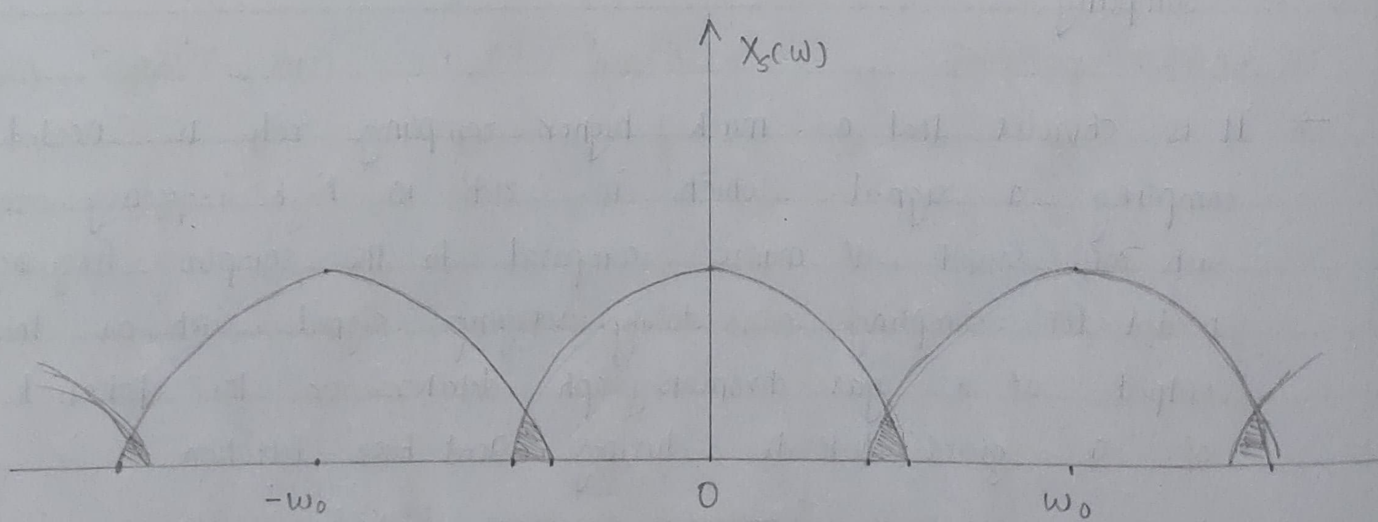
-> It is obvious that a much higher sampling rate is needed for sampling a signal which is rich in high frequency components such as sound of music compared to the sampling frequency needed for sampling a slowly varying signal, such as the output of a gas - chromatograph detector or the potential of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



(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 Theoram.

(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 theoram is called Sampling theorom.

(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.

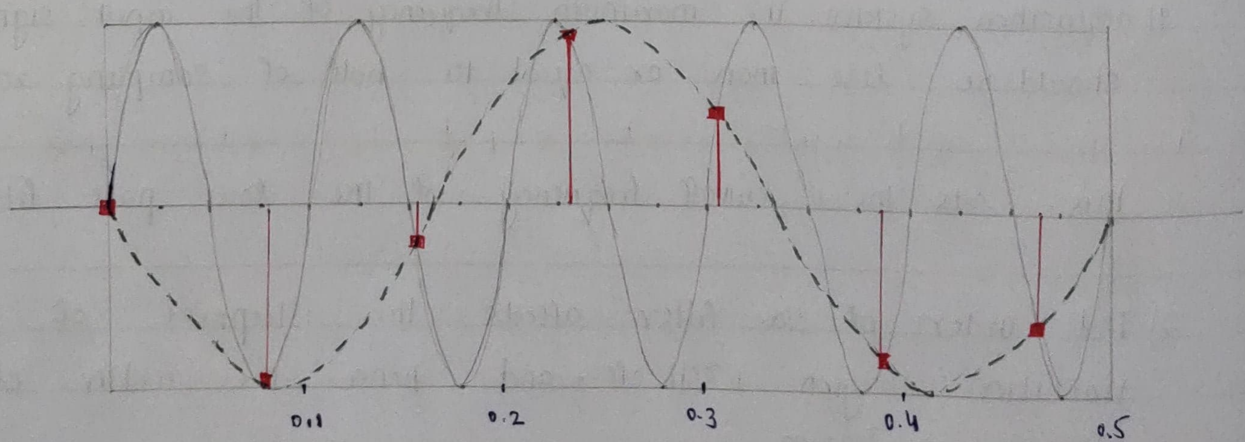
A filter of the n^{th} order will be have a roll-off rate of $n \times 20$ dB/ decade.

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

Observation Table

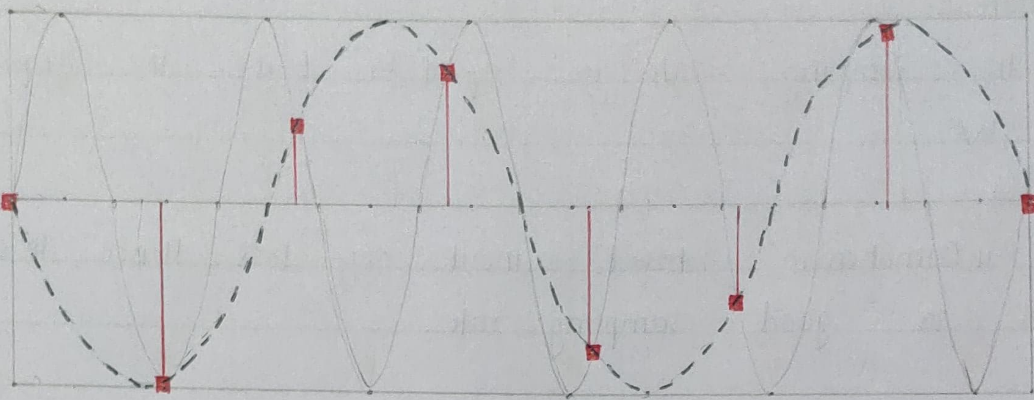
Signal Frequency (Hz)	Sampling Frequency (Hz)	Alias Frequency (Hz)
10	13	3
	14	4
	15	5
	20	-
	25	-
20	19	1
	22	2
	30	10
	40	-
	50	-

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

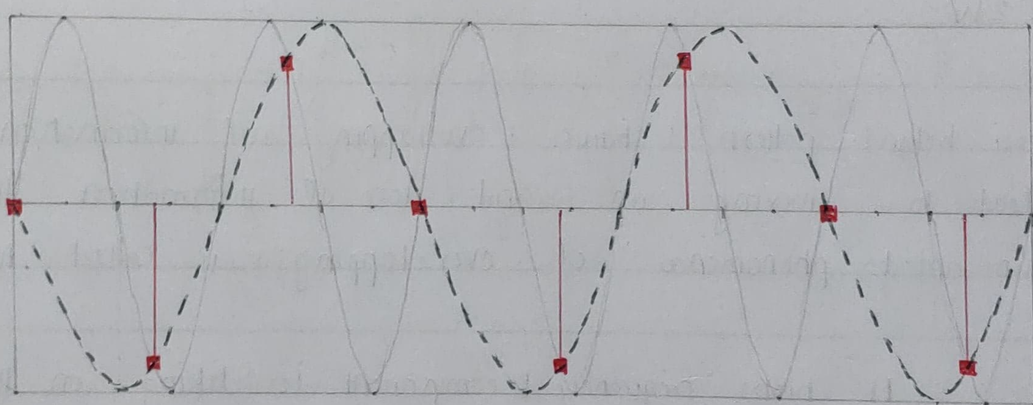


(9)

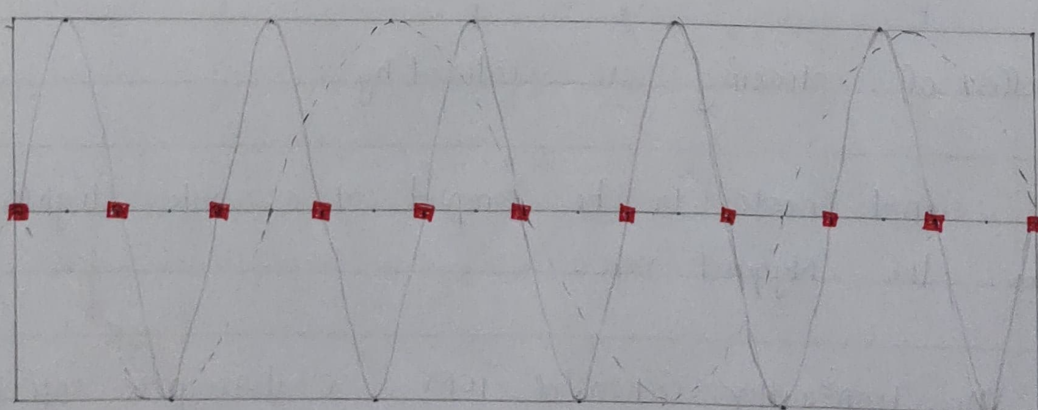
(2) Signal freq. (Hz) = 10.0 Alias freq (Hz) = 4.0 Sampling freq (Hz) = 14.0



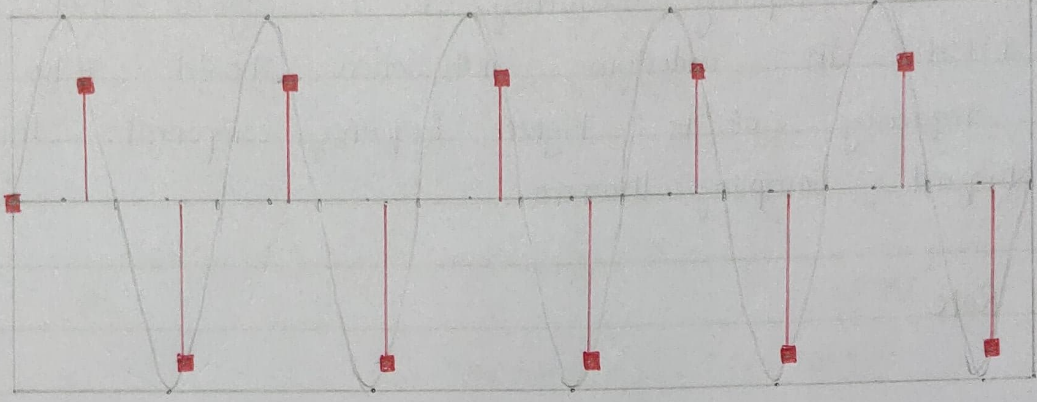
(3) Signal freq (Hz) = 10.0 Alias freq (Hz) = 5.0 Sampling freq (Hz) = 15.0



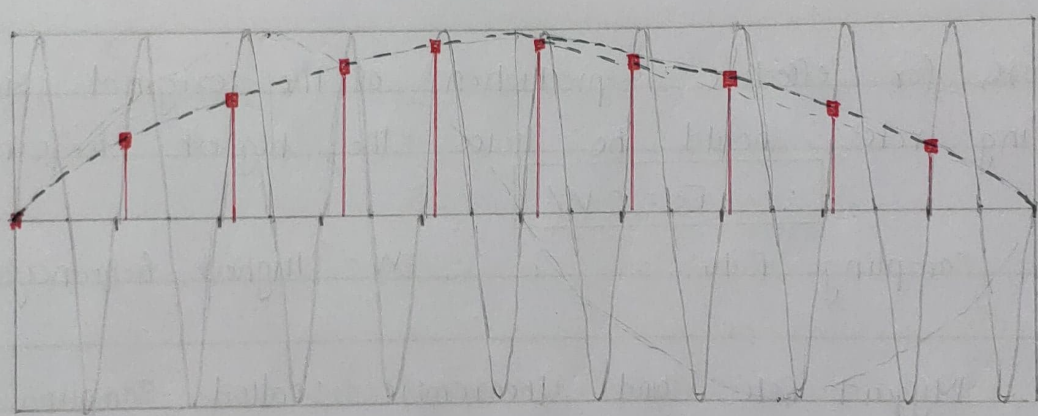
(4) Signal freq (Hz) = 10.0 Alias freq (Hz) = - Sampling freq (Hz) = 20



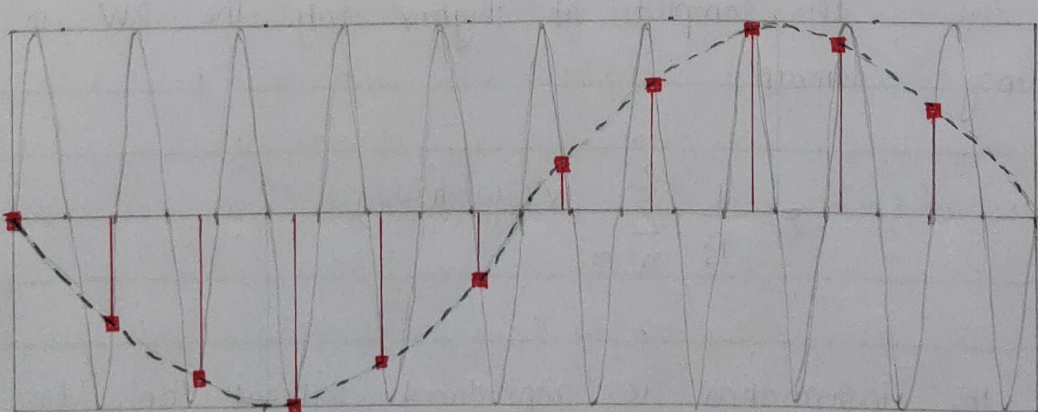
(5) Signal freq (Hz) = 10.0 Alias freq (Hz) = - Sampling freq (Hz) = 25



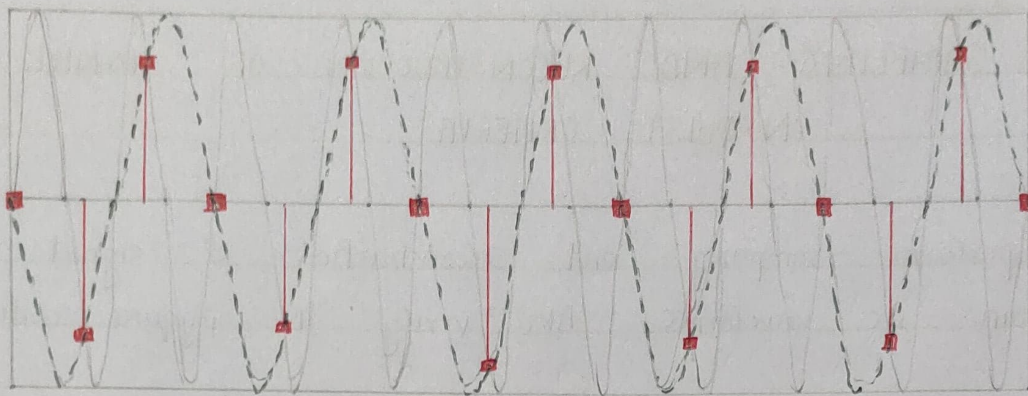
(6) Signal freq. (Hz) = 20.0 Alias freq. (Hz) = 1.0 Sampling freq. (Hz) = 19



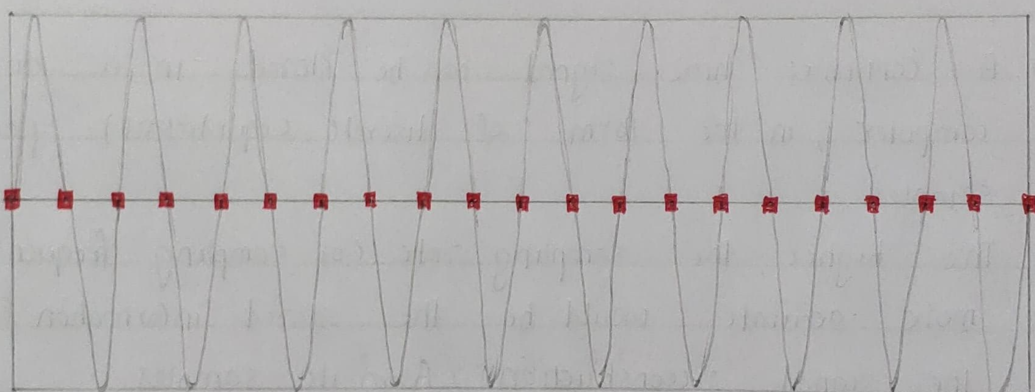
(7) Signal freq. (Hz) = 20.0 Alias freq. (Hz) = 2.0 Sampling freq. (Hz) = 22.0



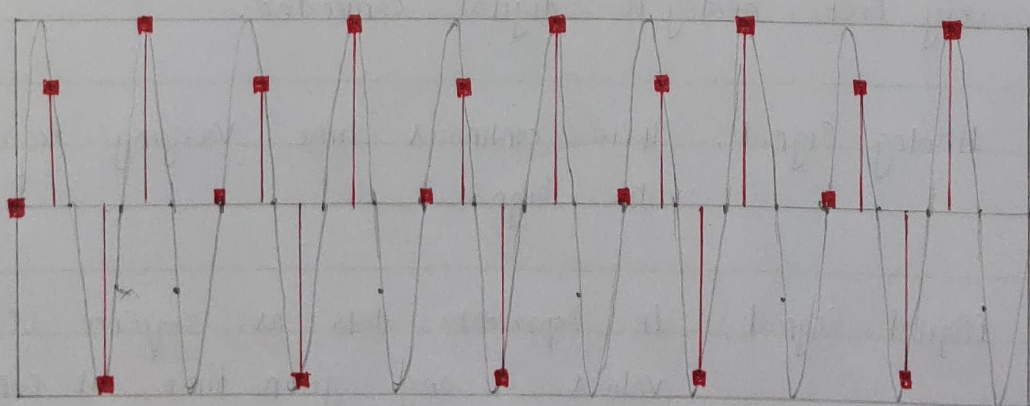
(8) Signal freq. (Hz) = 20.0 Alias freq. (Hz) = 10 Sampling freq (Hz) = 30



(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

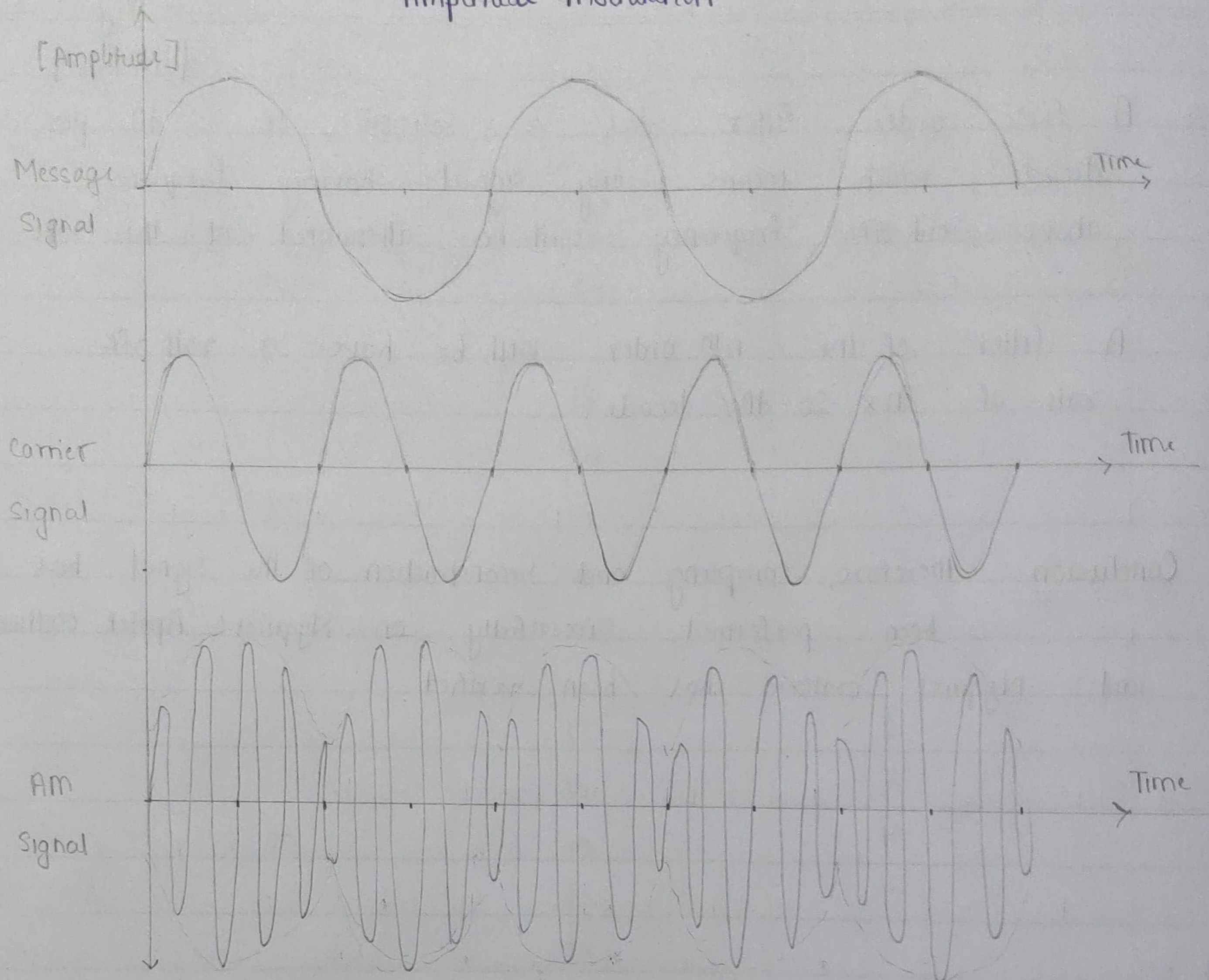
$$s(t) = [A_c + A_m \cos(2\pi f_m t)] \cos(2\pi f_c t)$$

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

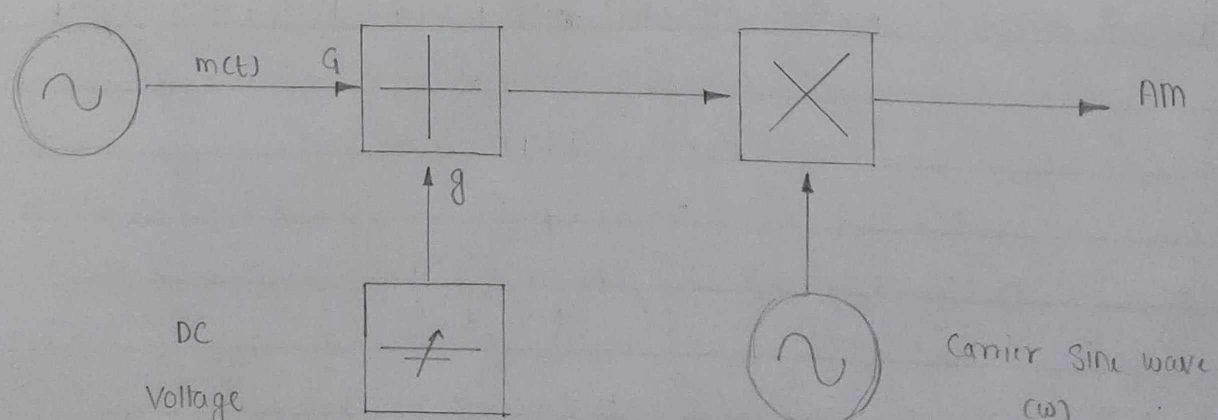
Modulation index = $m = \frac{A_m}{A_c}$

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

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, $b = A_c - Am$

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

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

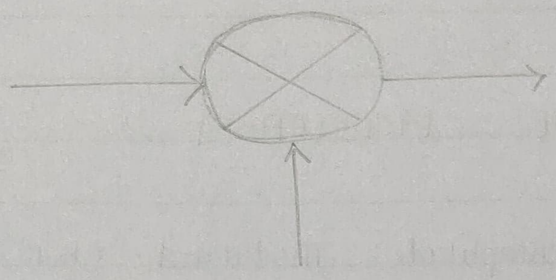
4.) Envelope Detector

- 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.

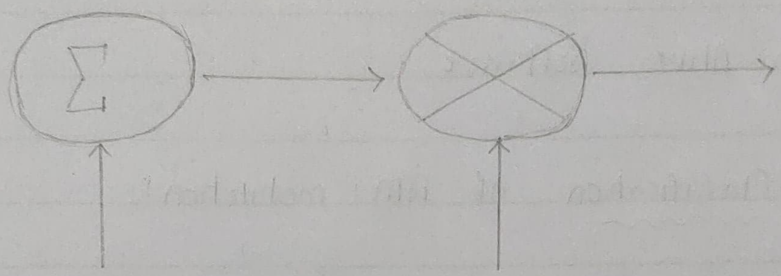
5.) Synchronous detector

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

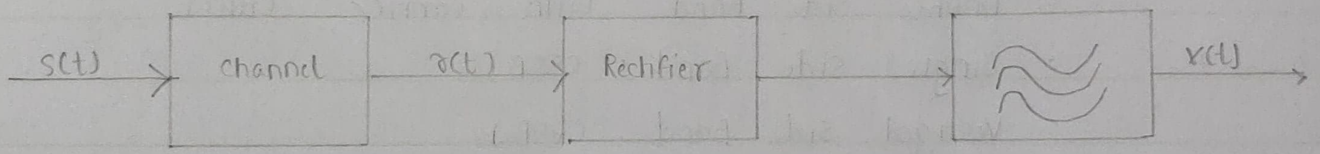
DSB - SC



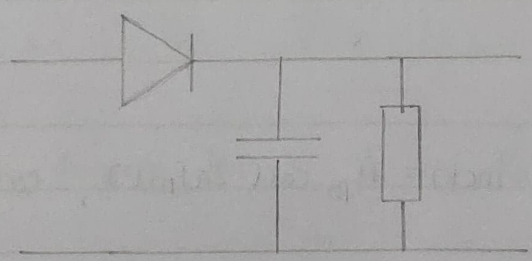
AM



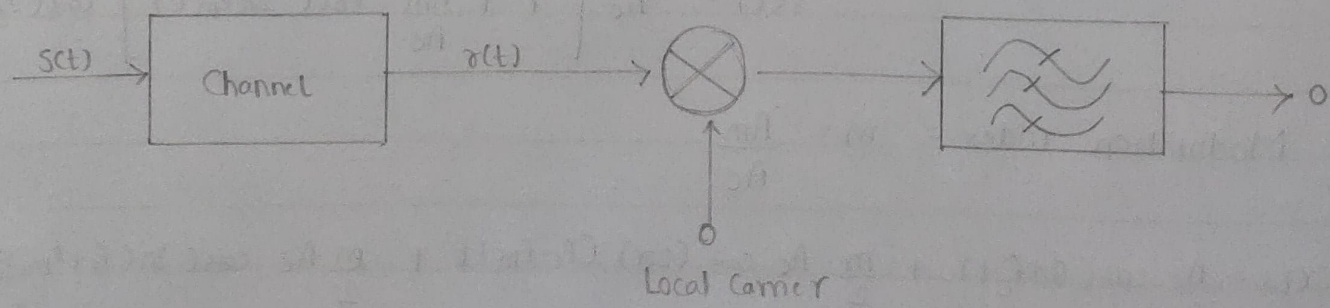
Envelope Detector



Rectifier
(Diode, capacitor)



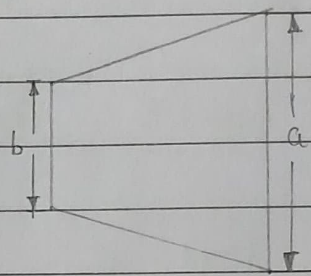
Synchronous Detector



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

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

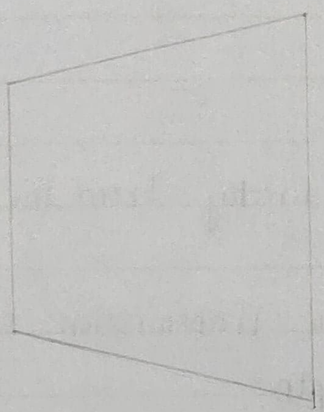
> PROCEDURE : In this online mode of practical, we perform the experiment on Labalive application.

1.) We will first execute the AM analyser simulator

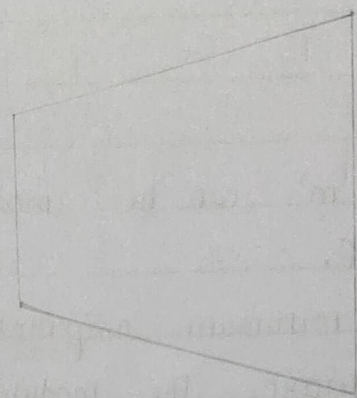
2.) After executing the AM analyser simulator, click on the $\hat{S}$ in the AM modulation window.

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

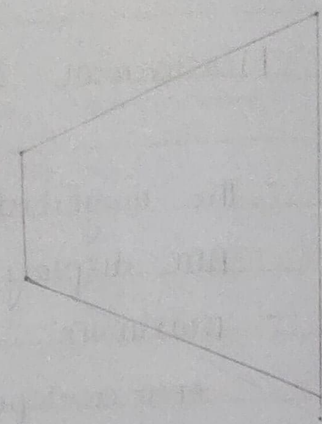
Trapezoid width is unaffected by modulation depth.



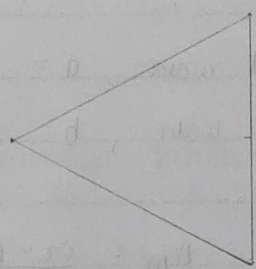
m = 0.1



m = 0.3

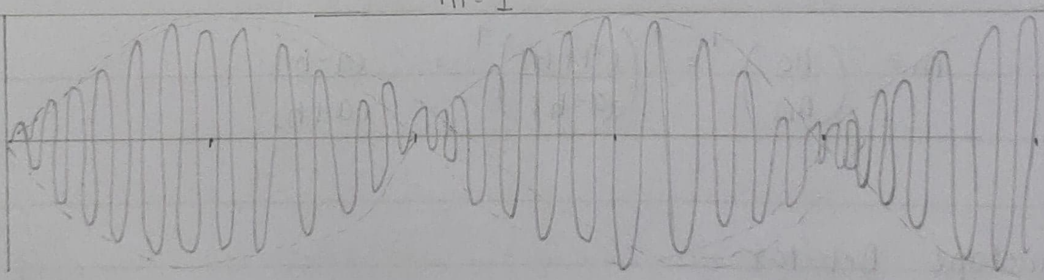


m = 0.5

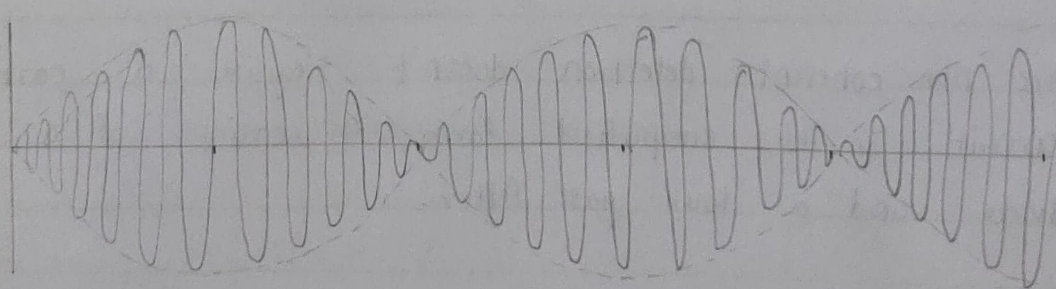


m = 1

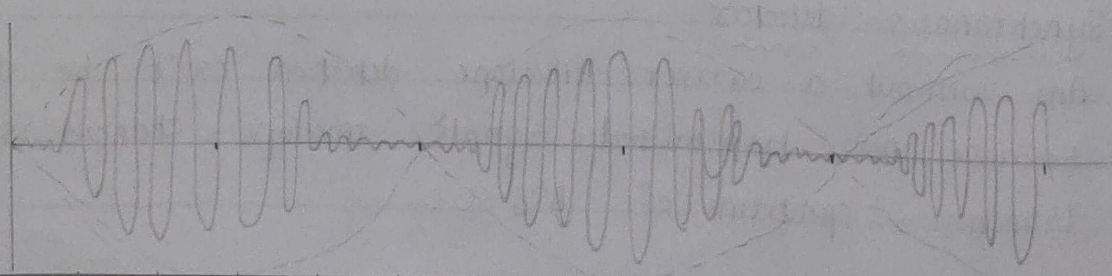
Modulation Index = 0.5 (m<1)



Modulation Index = 1 (m = 1)



Modulation Index = 1.5 (m > 1)



Observation

DOUBLE SIDE BAND WITH CARRIER

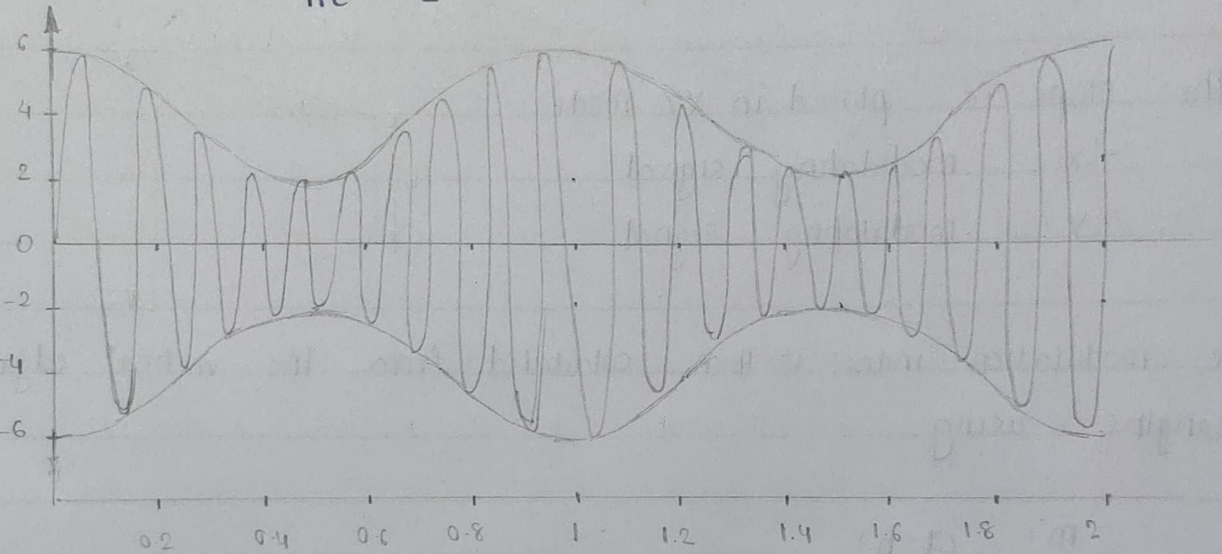
a) m<1

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

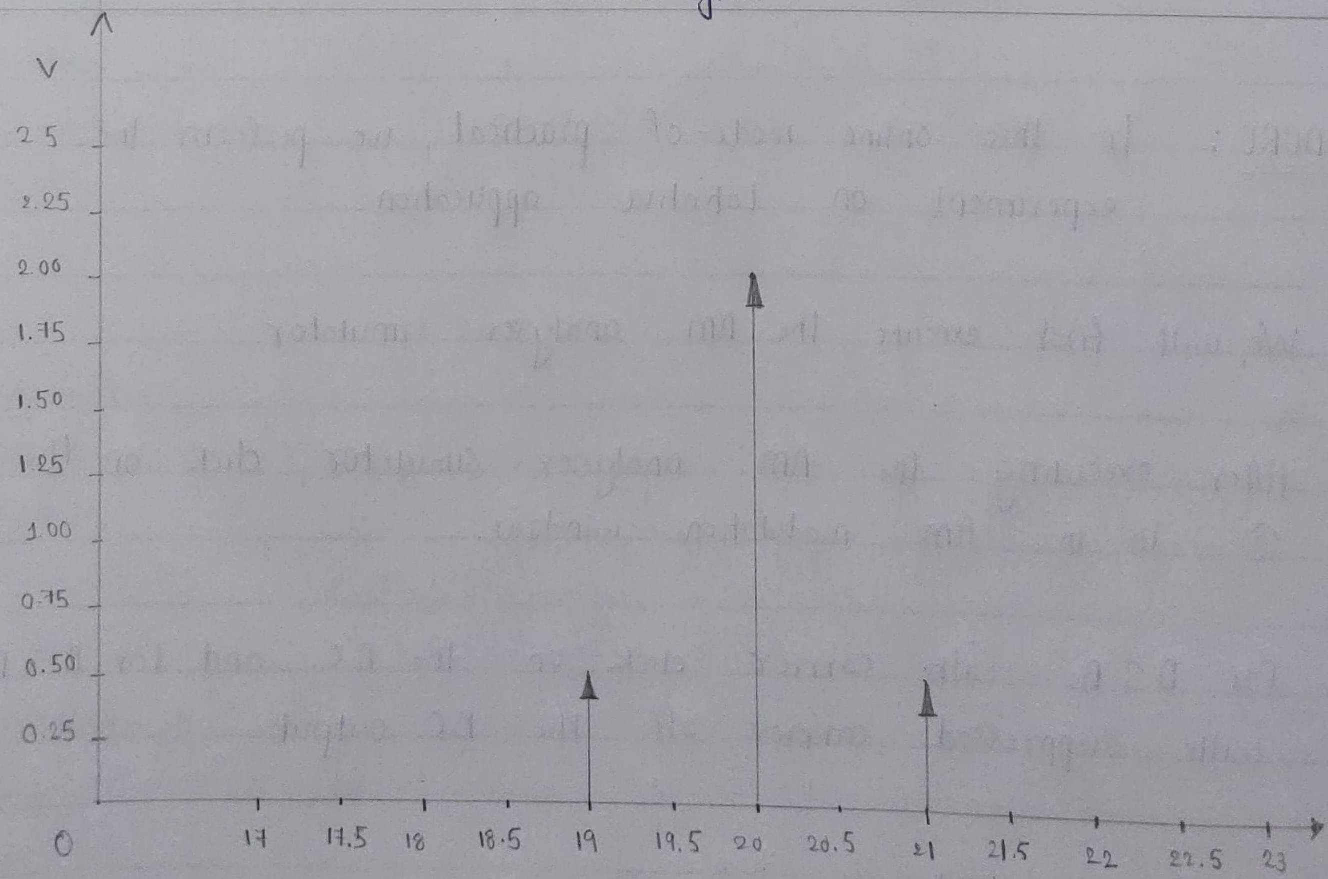
Carrier = $A_c = 2V$ $f_c = 20 MHz$ (cosine)

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

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



Transmitted signal

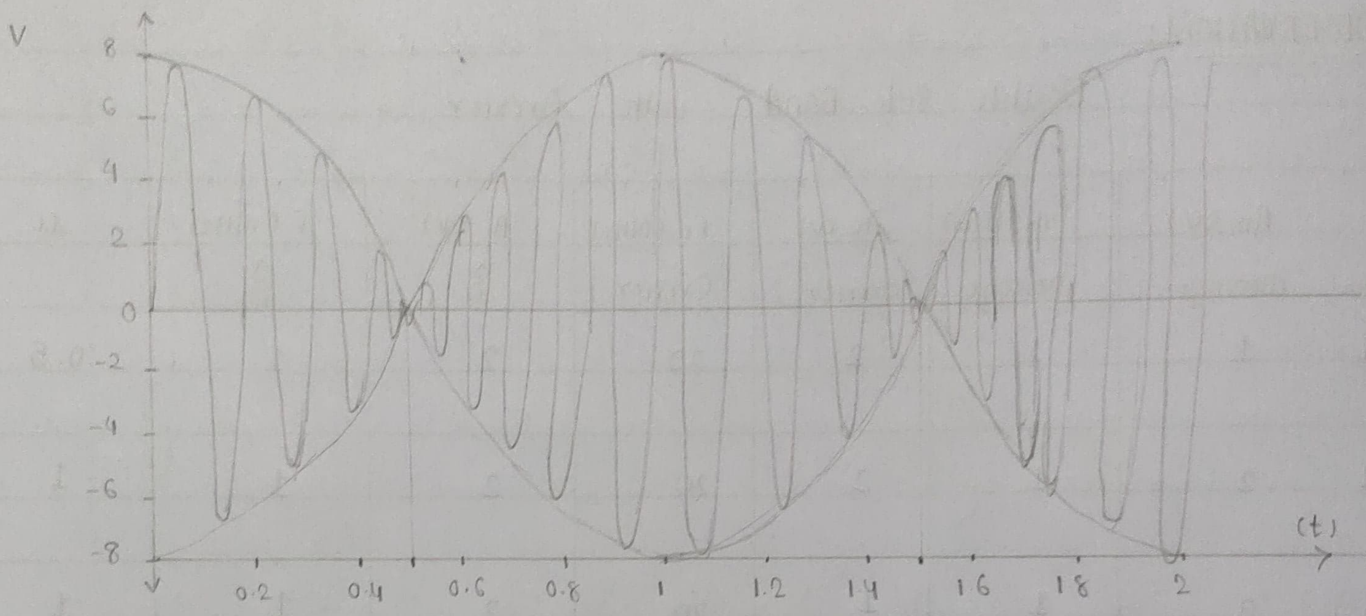


Spectrum

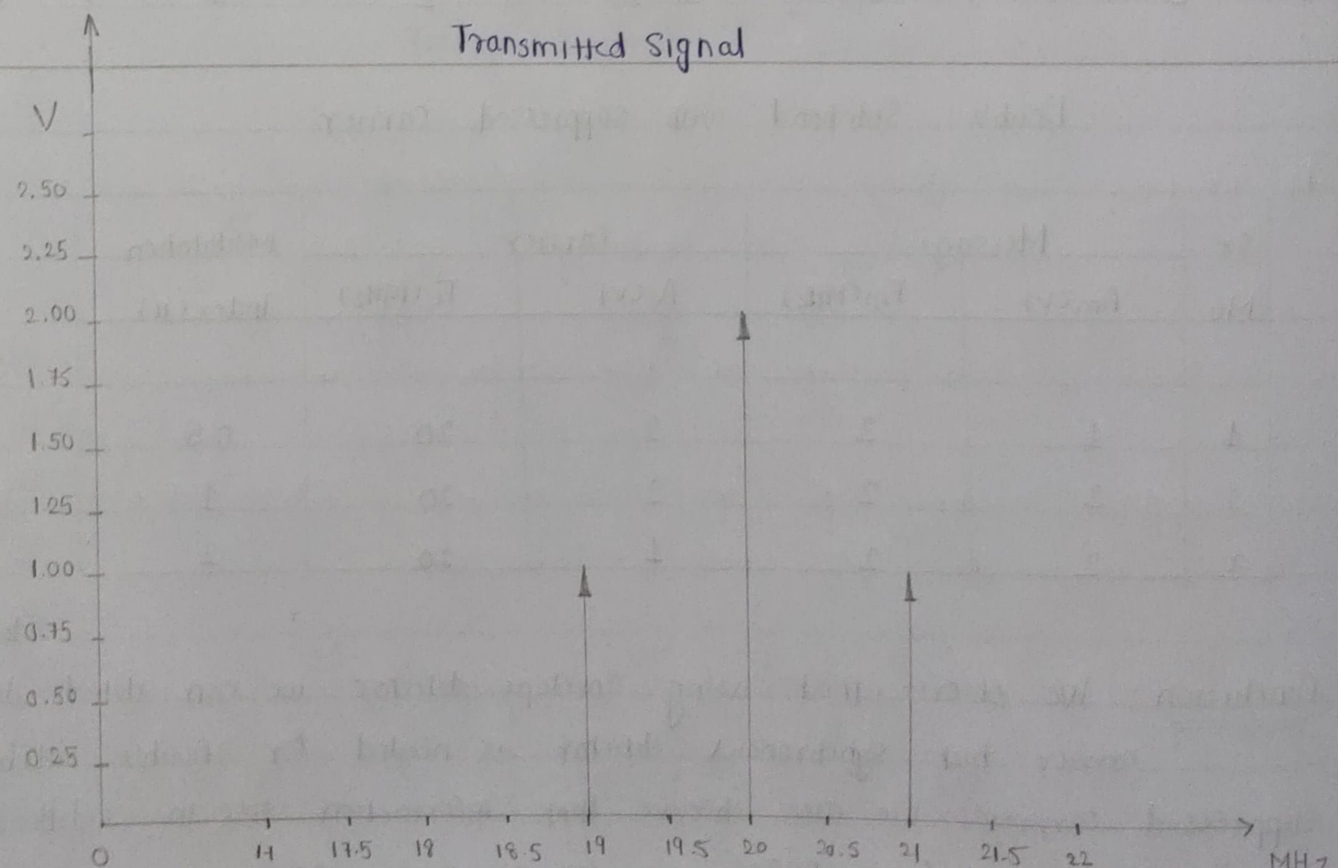
(b) m = 1

Message	$A_m = 2V$	$f_m = 1MHz$ (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

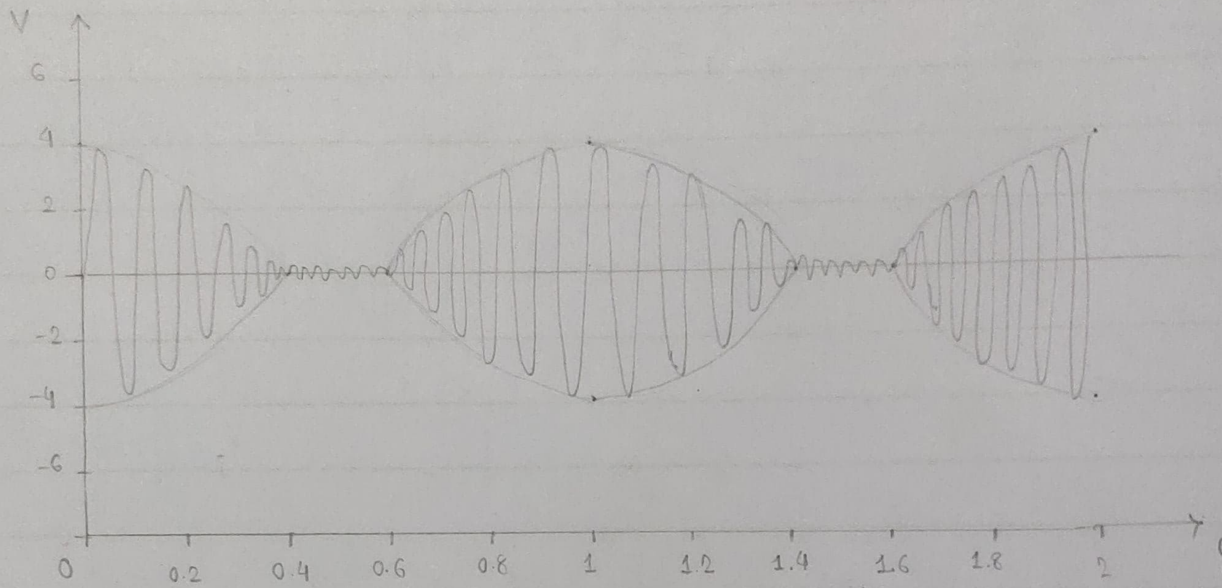


Spectrum

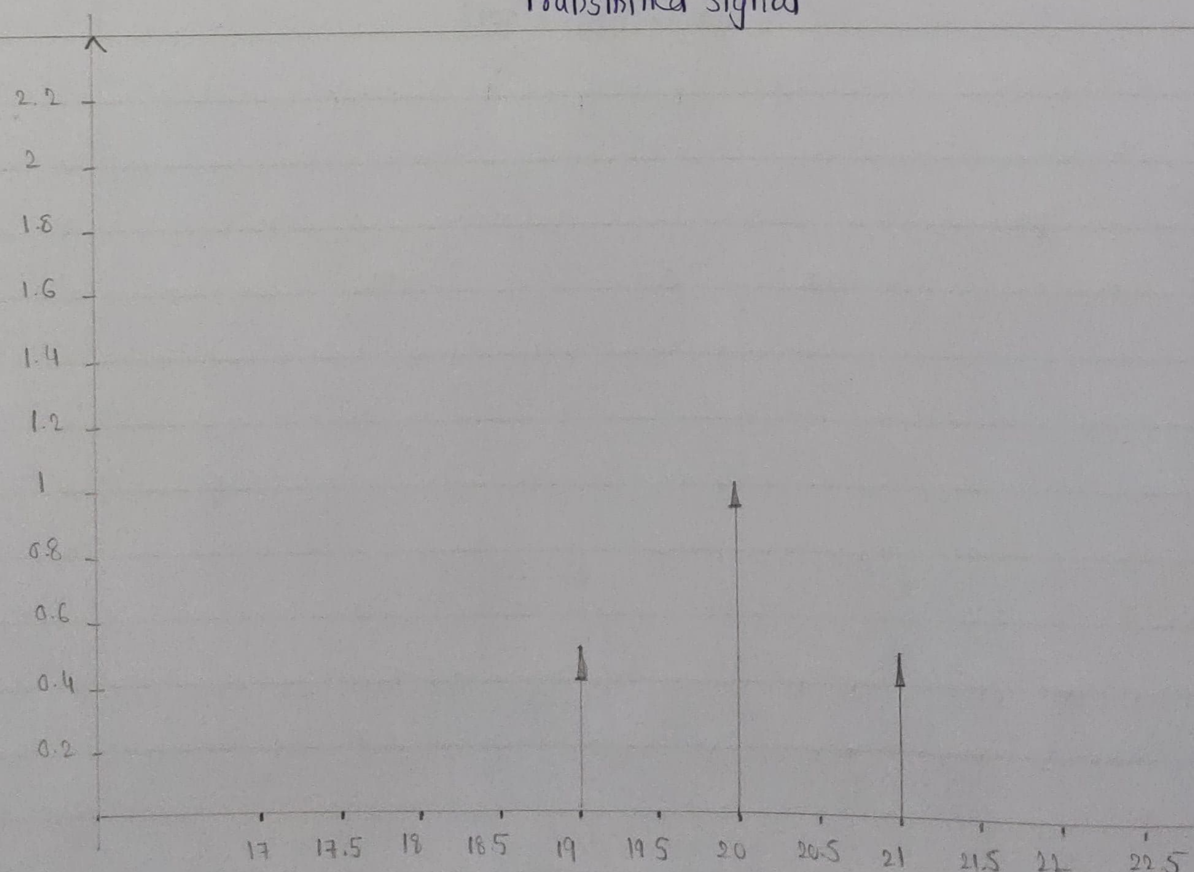
c) $m > 1$

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

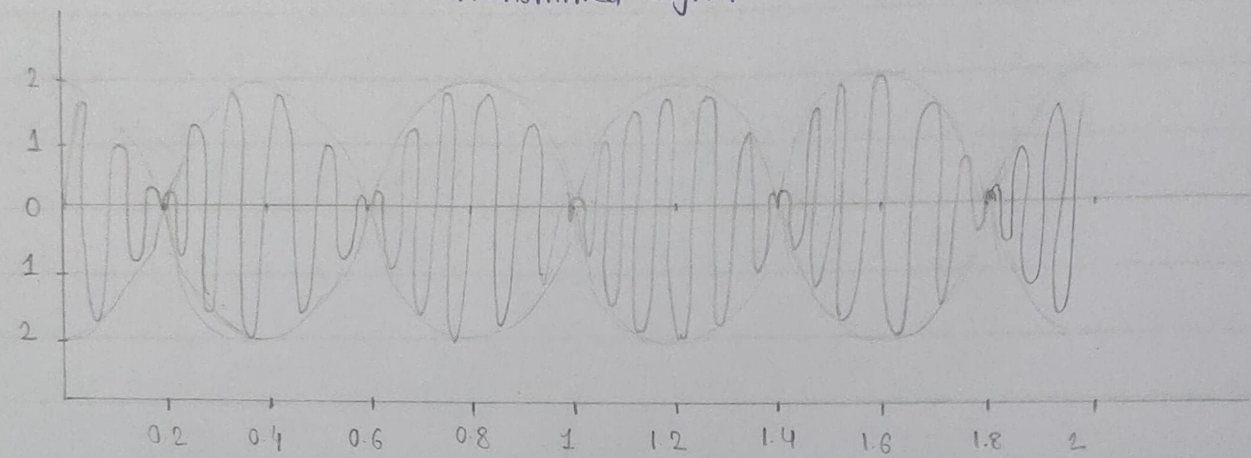
DOUBLE SIDEBAND SUPPRESSED CARRIER [DC OFFSET OFF]

a) m<1

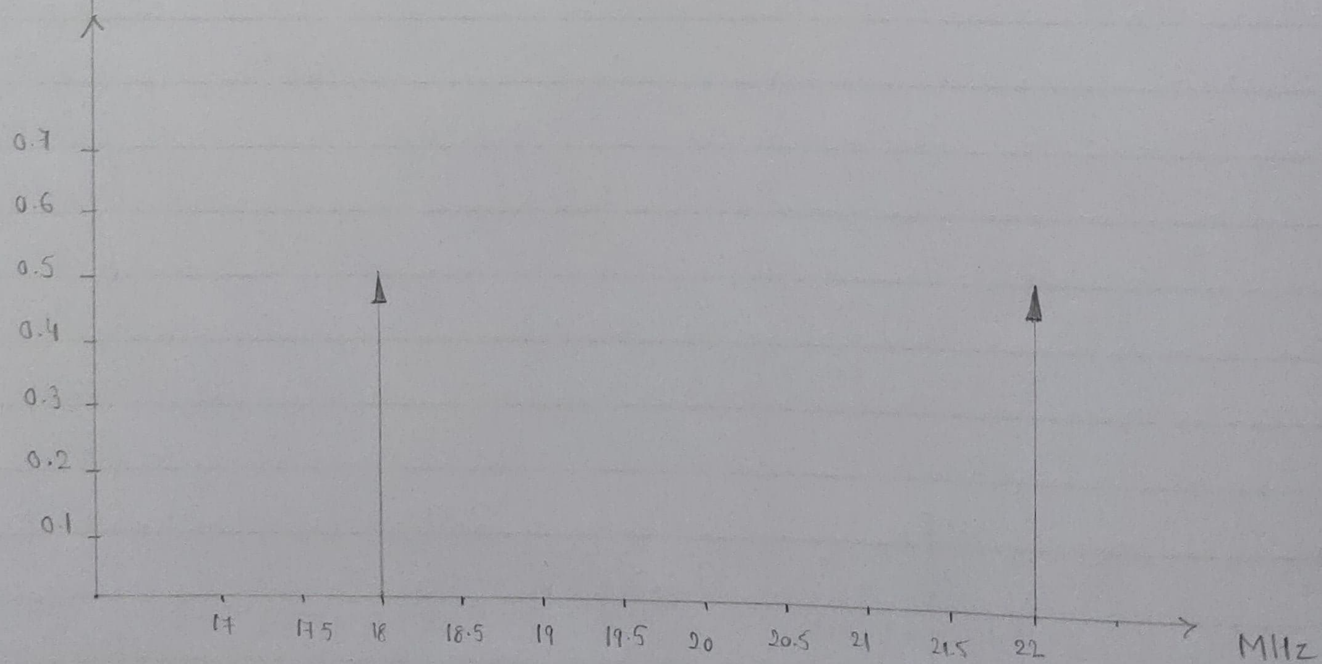
Message	$A_m = 1V$	$F_m = 2 MHz$	(cosine)
Carrier	$A_c = 2V$	$F_c = 20 MHz$	(cosine)

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

Transmitted signal



Spectrum

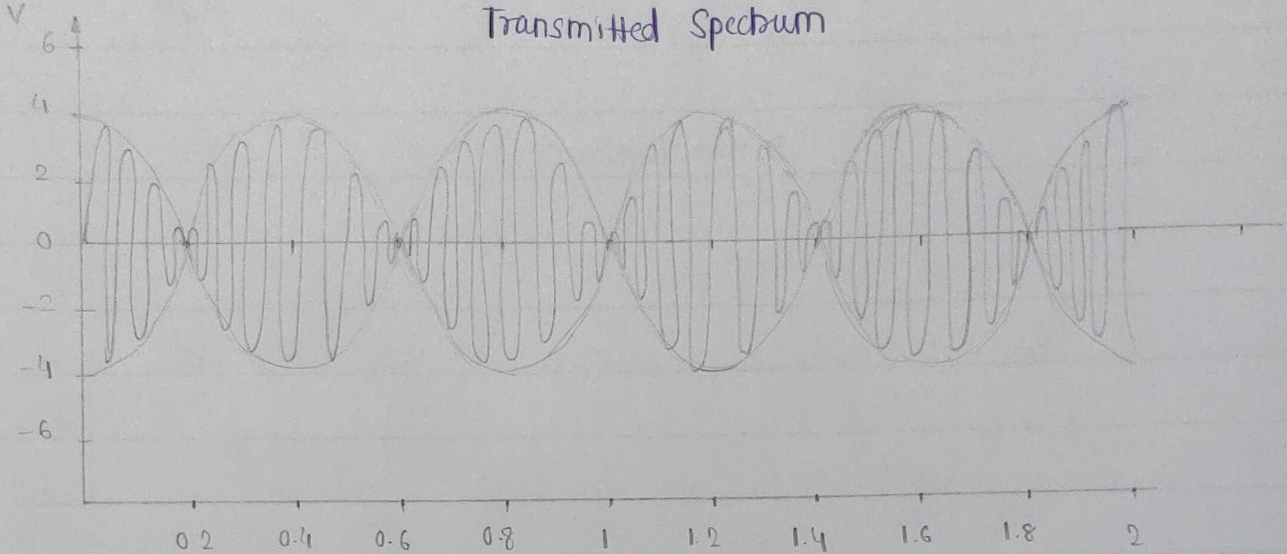


b) m = 1   Message $A_m = 2V$ $F_m = 2\,MHz$ (cosine)

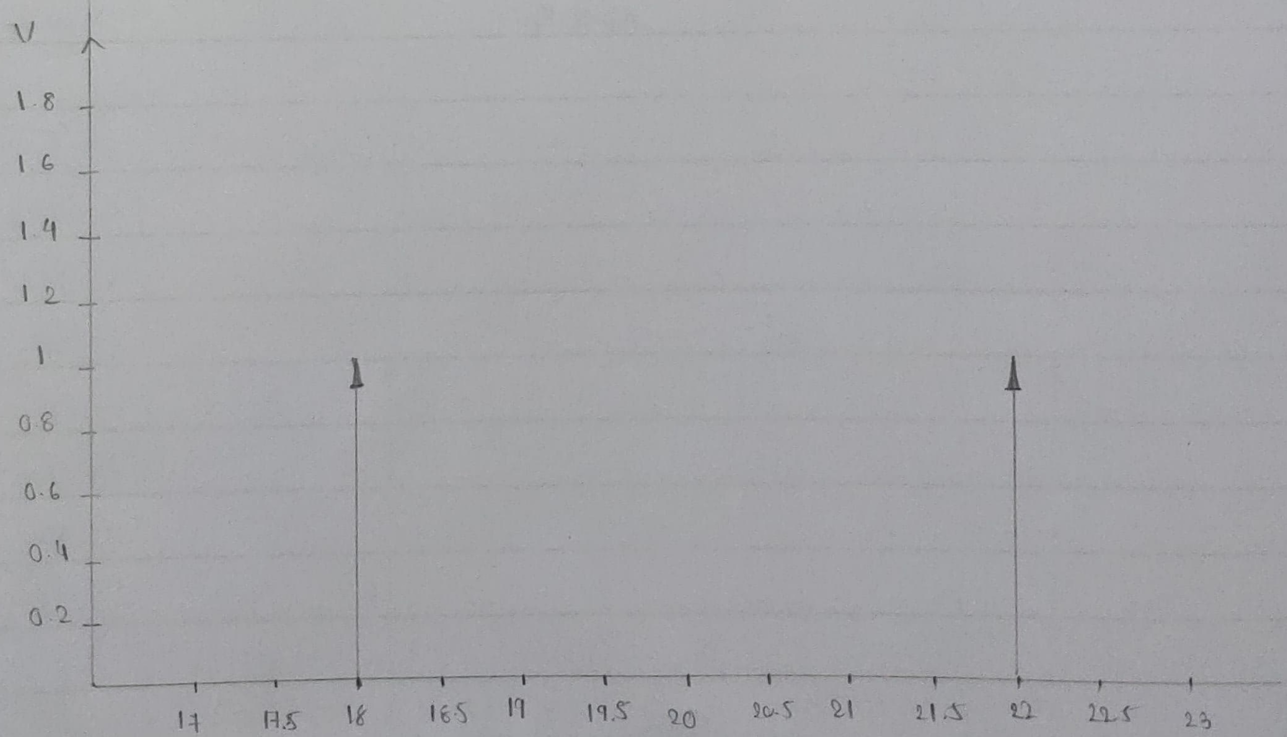
Carrier $A_c = 2V$ $F_c = 20\,MHz$ (cosine)

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

Transmitted Spectrum



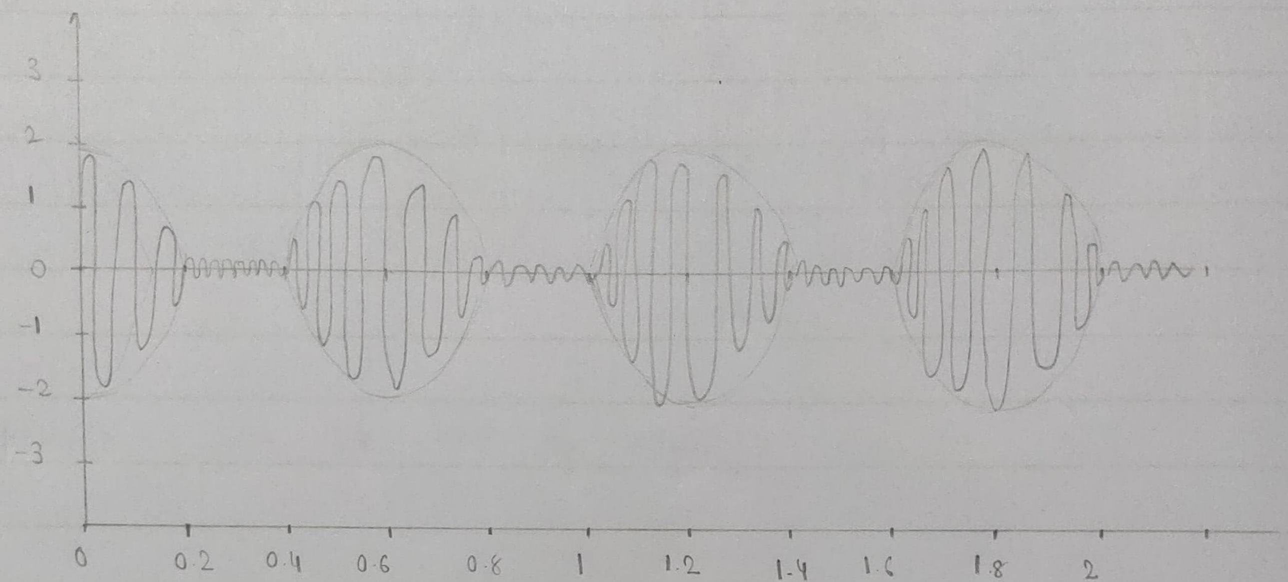
Spectrum



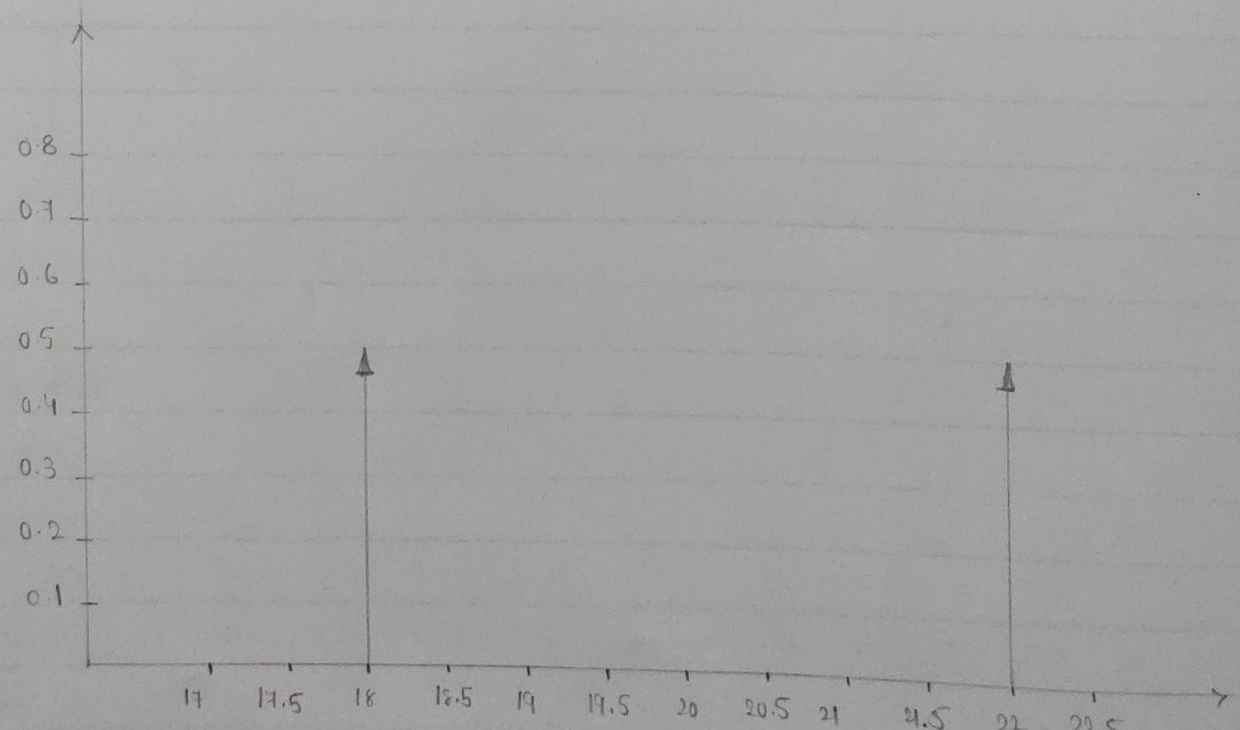
c) $m > 1$ Message $A_m = 2V$ $F_m = 2\,kHz$ (cosine)

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) $\hat{\delta}$	μ
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
	Am (V)	Fm (MHz)	Ac (V)	Fc (MHz)	Index (μ)
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

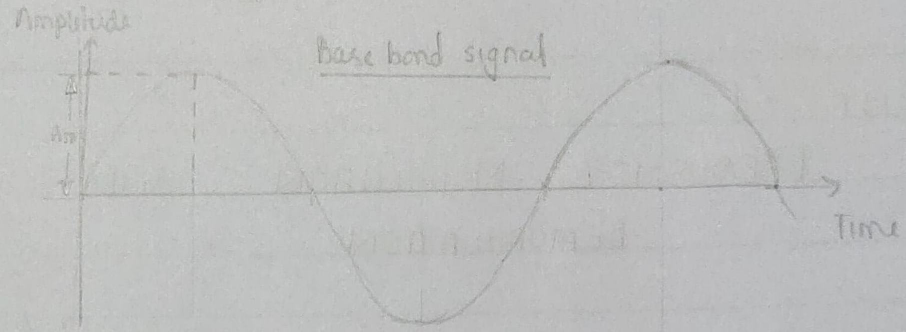
$$s(t) = A_c \cos(\theta_i(t))$$

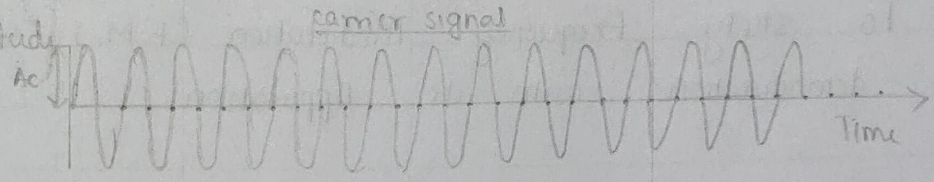
where A_c = Amplitude of the modulated wave / carrier signal
$\theta_i(t)$ = Angle of modulated wave.

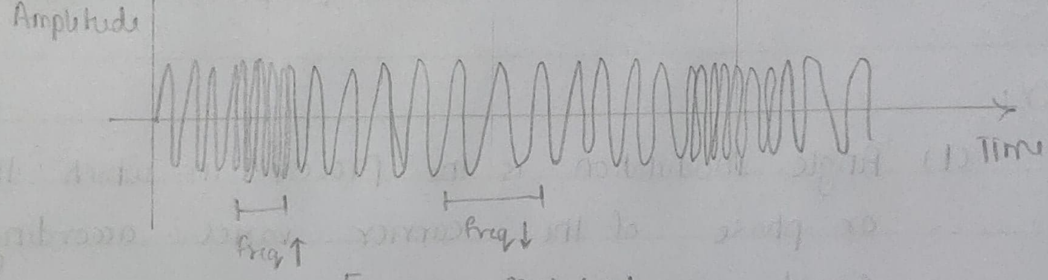
(3) Angle modulation is further divided into Frequency modulation and Phase modulation

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

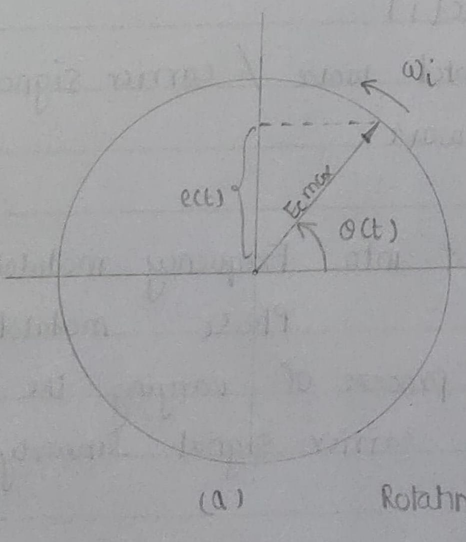
(ii) Phase modulation: is the process of varying the phase of carrier signal linearly with message signal.

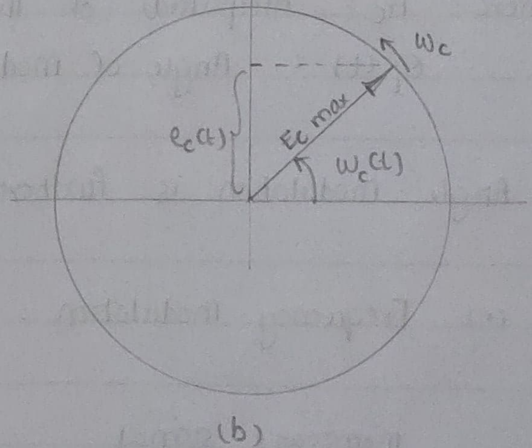






Frequency Modulated wave





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

Rotating phasor representation of carrier of amplitude E_c max

(b) at constant angular velocity (ω_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

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

(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 eqn 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. (β)

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

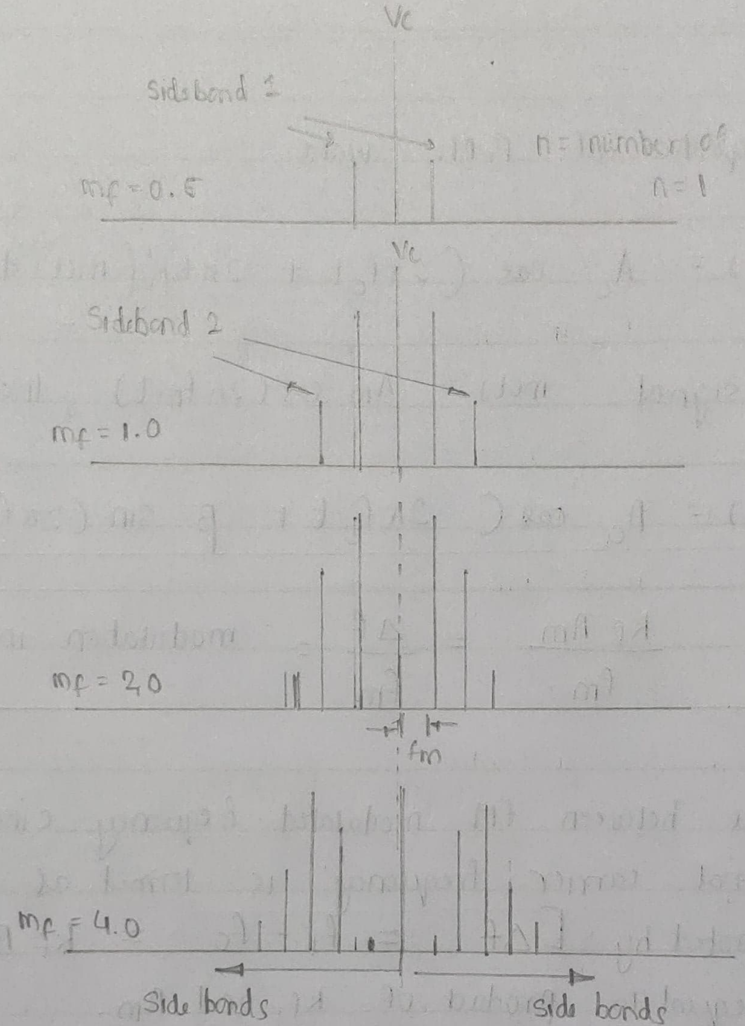
Carrier frequency swings between f_{max} and f_{min} as input varies.

Freq. deviation

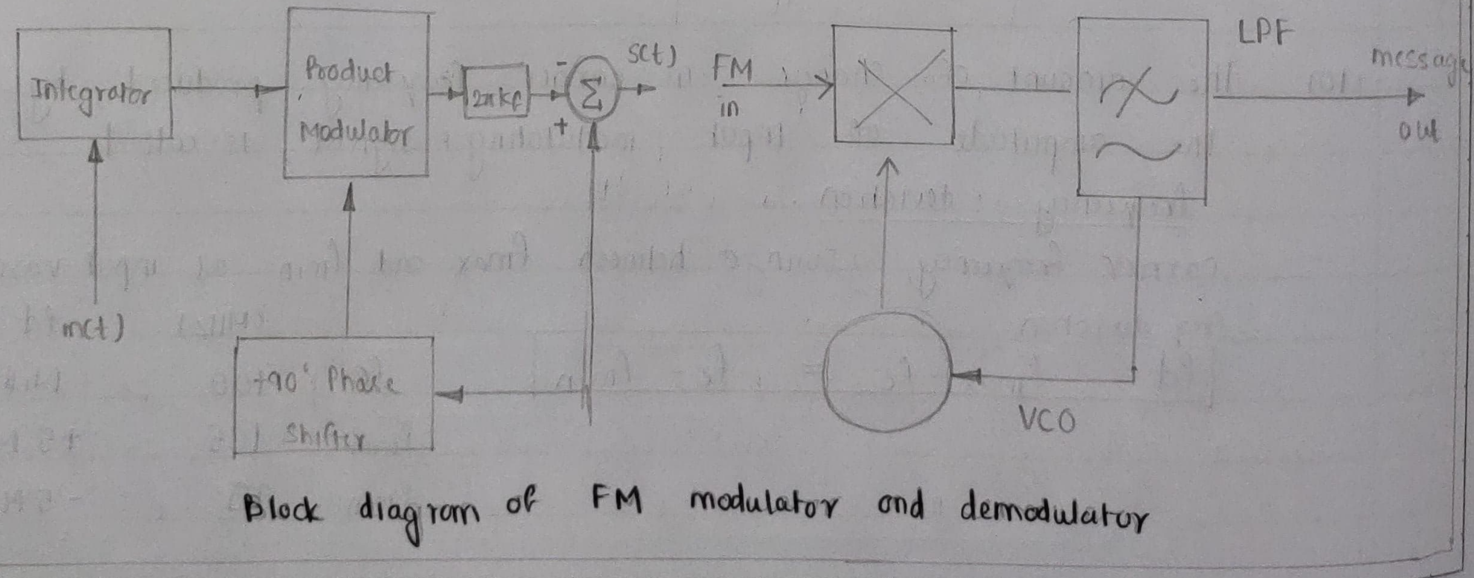
$$\boxed{f_d = f_{max} - f_c = f_c - f_{min}}$$

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

Teacher's Signature:



How spectrum F.M. varies with mf



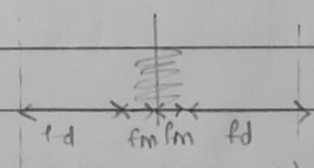
Block diagram of FM modulator and demodulator

(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)

EXPT. NO. NAME: Page No.: 1 Date:

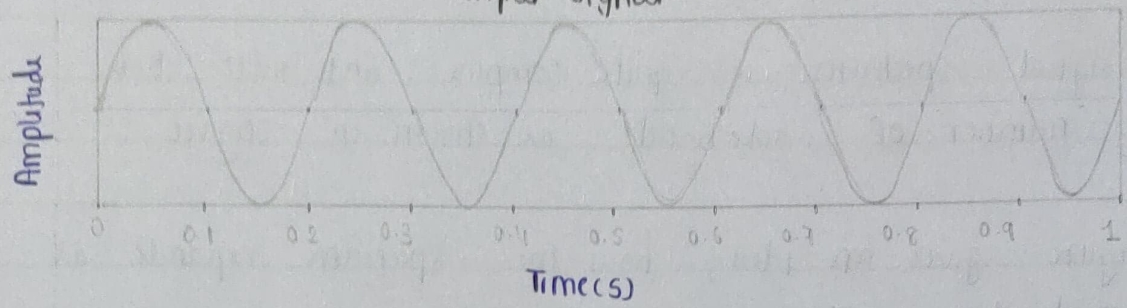
MATLAB Code:

```
% 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)')
ylabel ('Amplitude')
subplot ( 5,1,3)
plot (t, y)
title('FM Modulation of input signal')
xlabel ('Time(s)')
ylabel ('Amplitude')
```

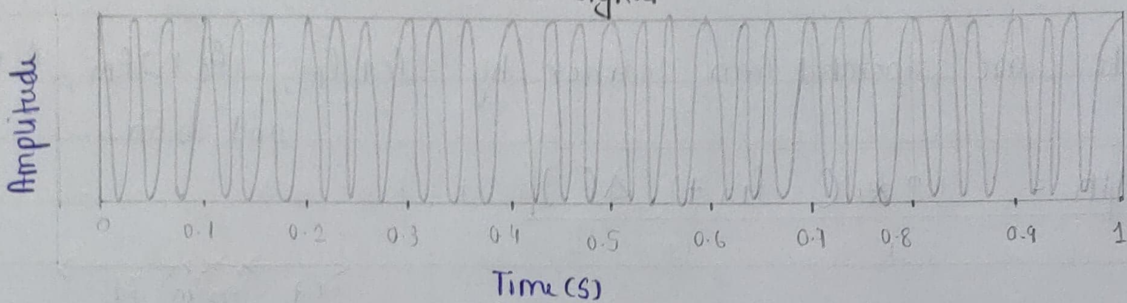
Teacher's Signature:

MATLAB OUTPUT

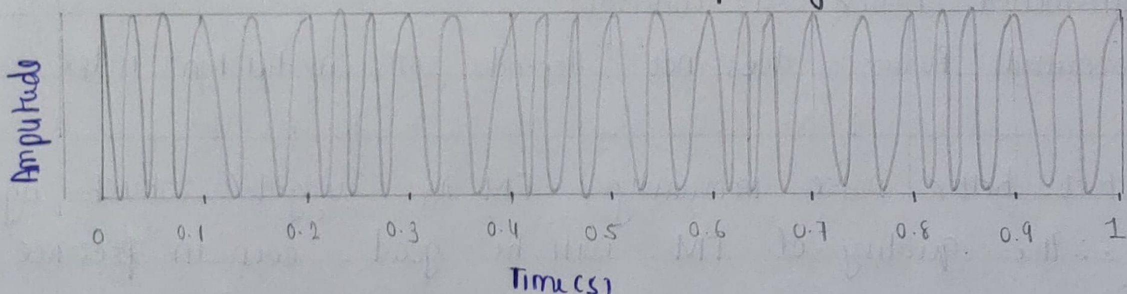
Input Signal



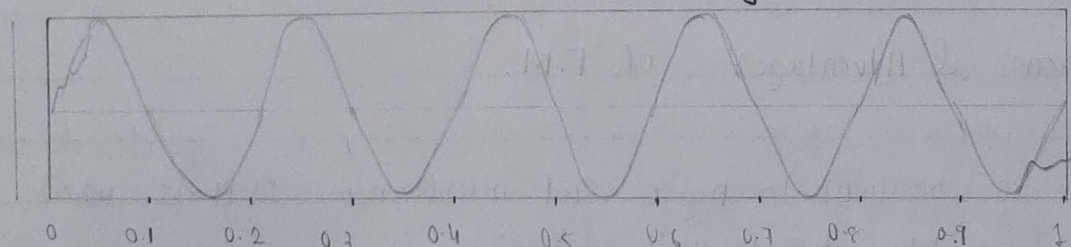
Carrier signal



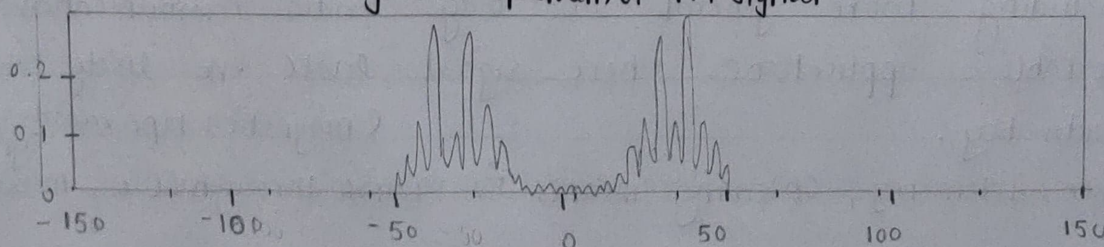
FM Modulation of Input signal



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—

EXPT. NO. 5 NAME: PAM, PPM, & PWM

Page No.: 1 Date: 16 Sept

EXPERIMENT 5 : [U19CS012]

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 :

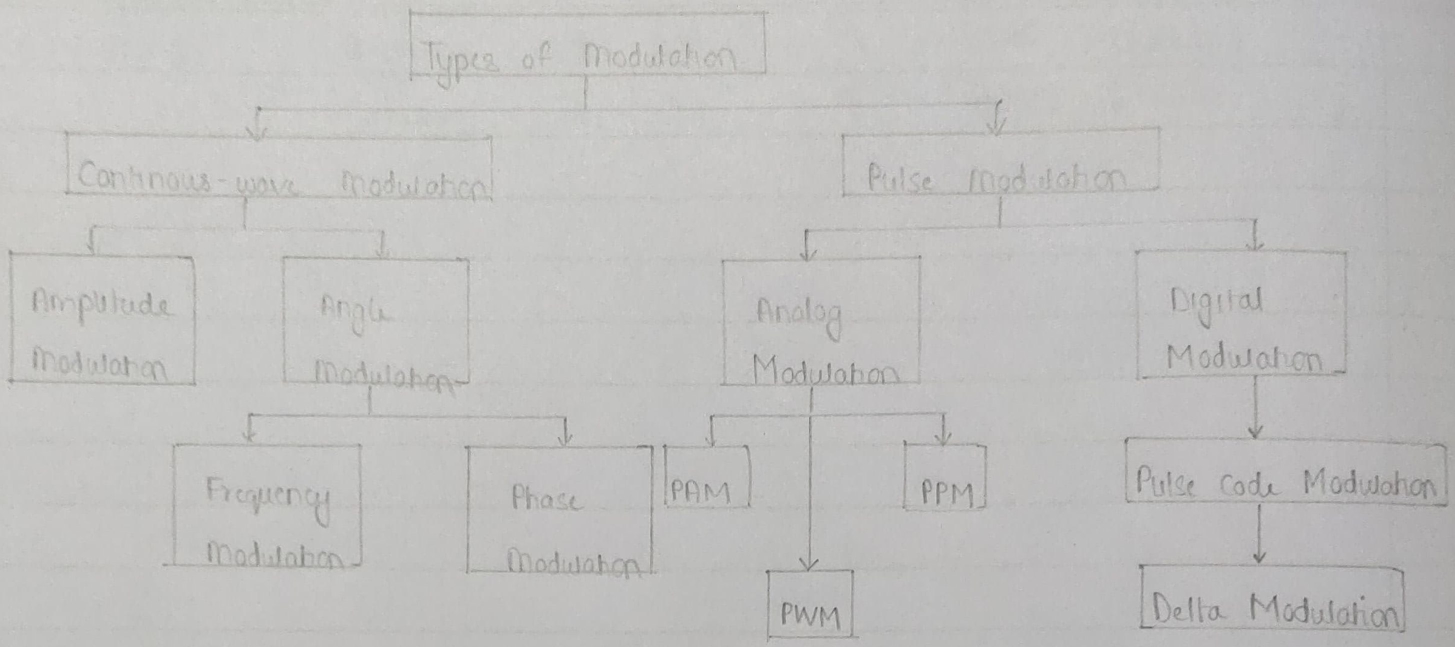
1.) Pulse modulation is a type of modulation in which the signal is transmitted in the form of Pulses.

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).

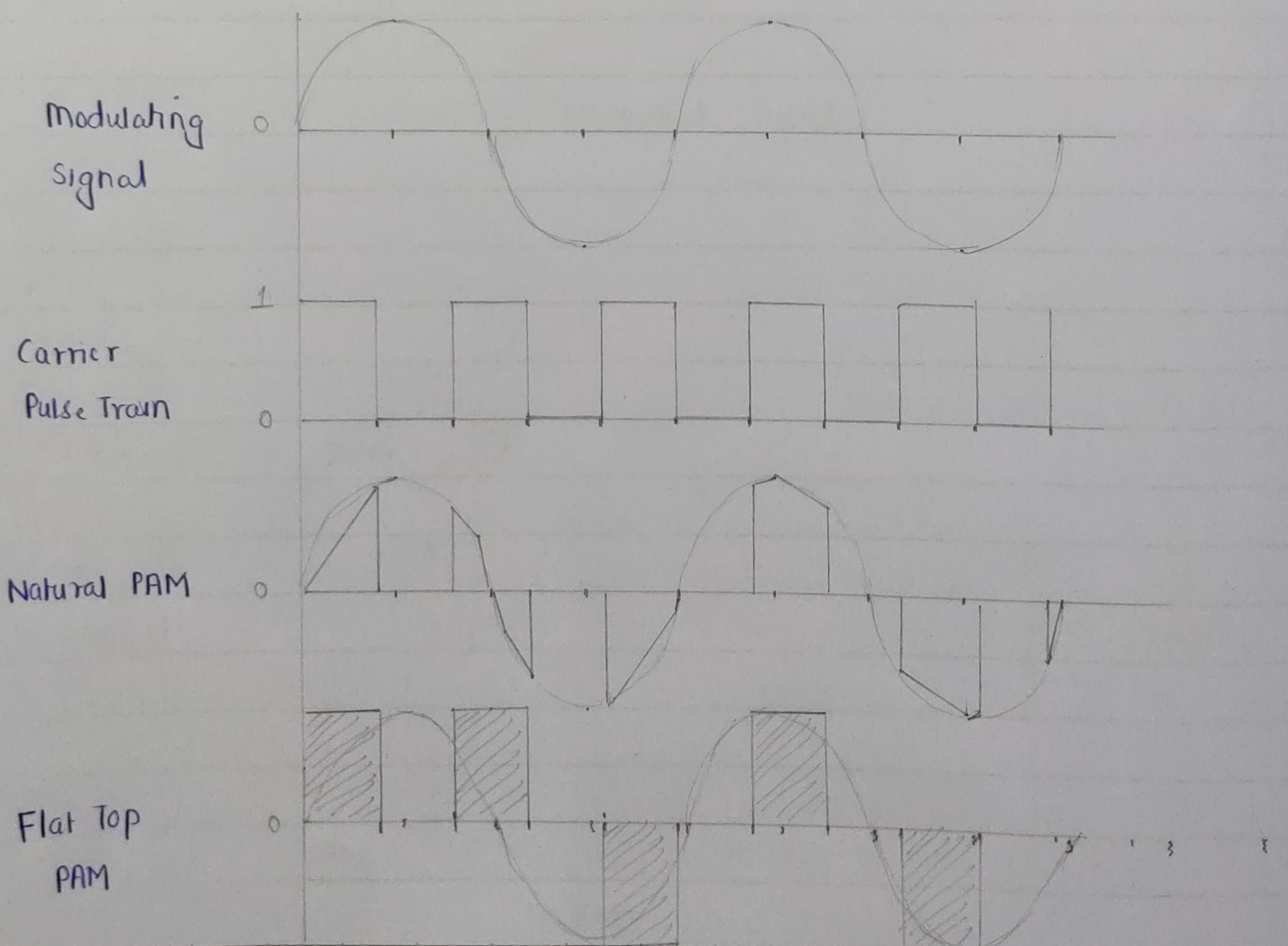
2.) Pulse Amplitude Modulation (PAM) :

→ 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.

Teacher's Signature:



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.

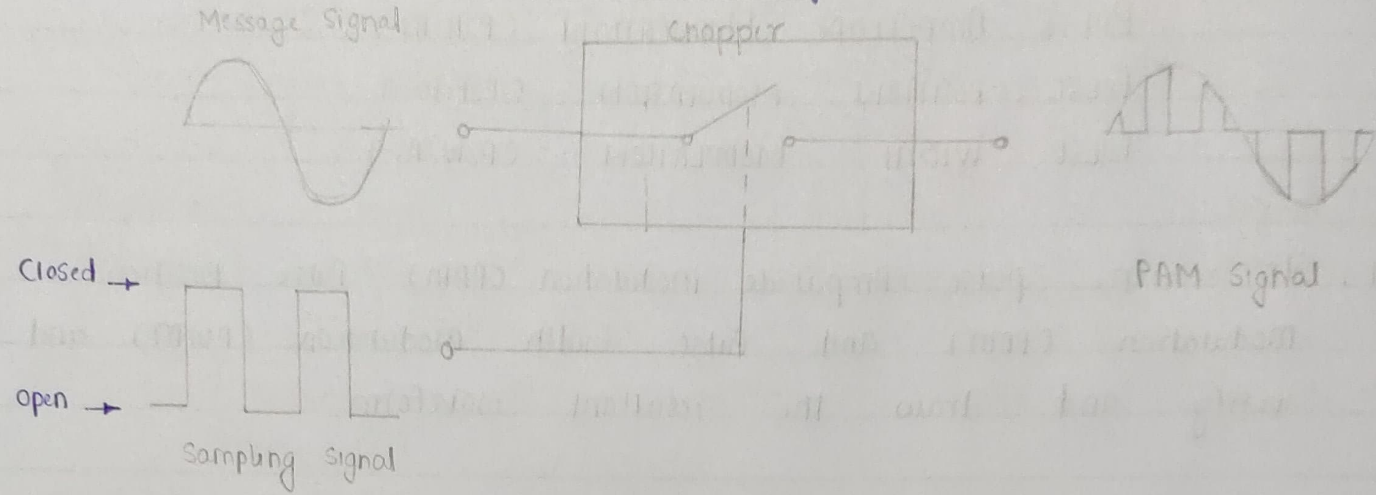
4.) Pulse Position Modulation (PPM)

→ 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.

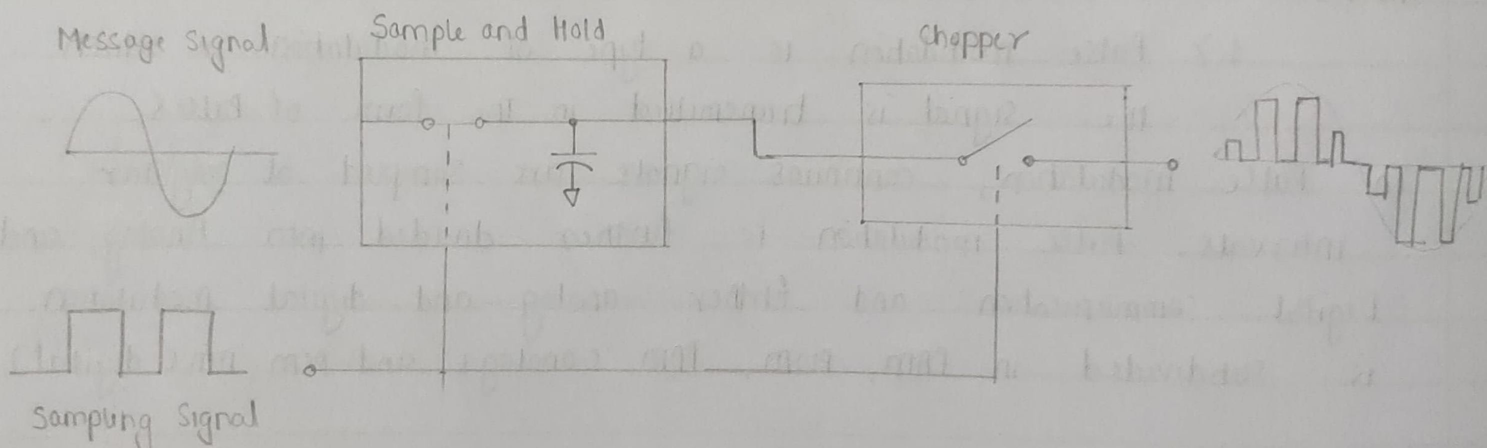
→ PPM is further modification of PWM.

4

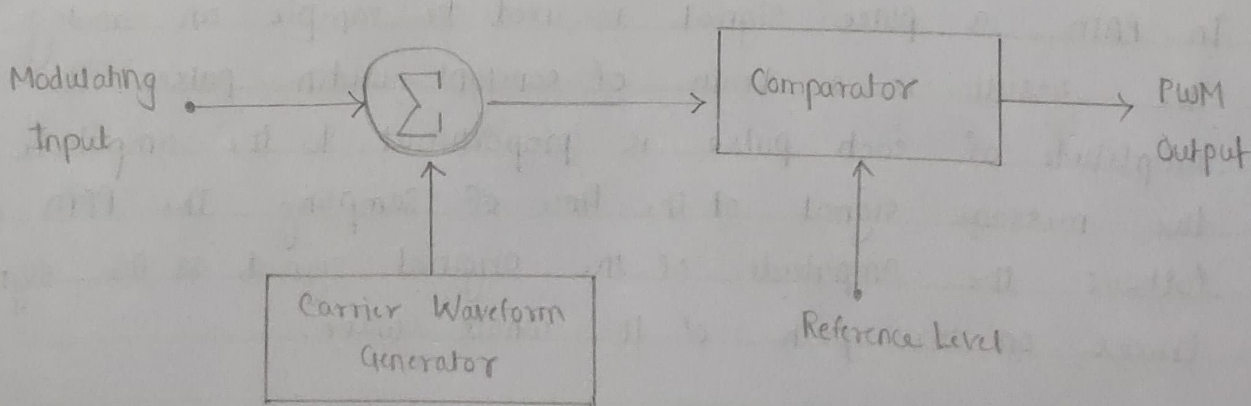
Generation of PAM by Natural Sampling :



Generation of PAM signal by Flat-top Sampling



Generation of PWM Signal

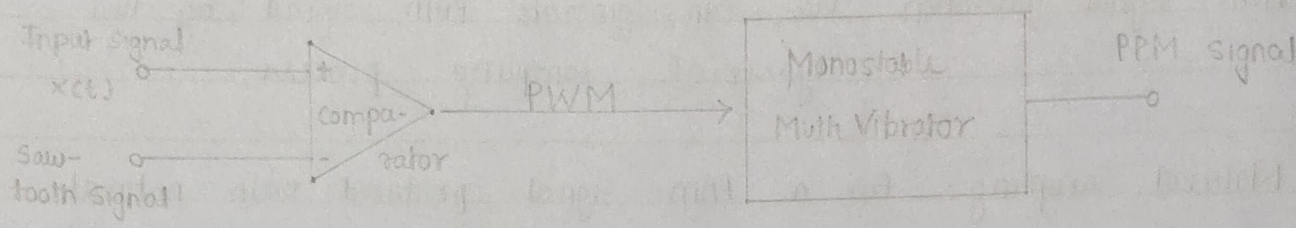


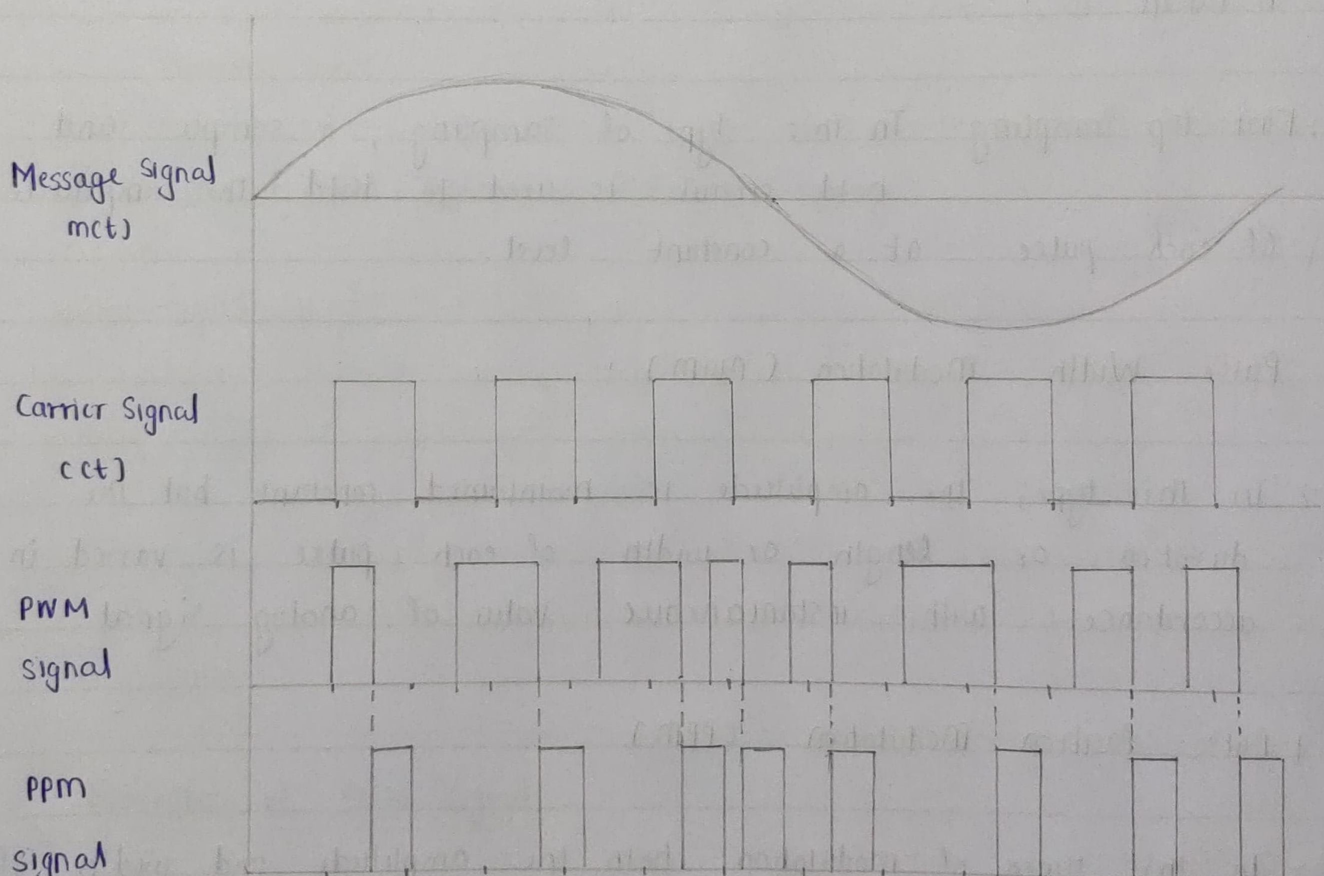
5.) Comparision of PAM, PWM and PPM :

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

EXPT. NO. NAME: Page No.: 7 Date:

MATLAB CODE :

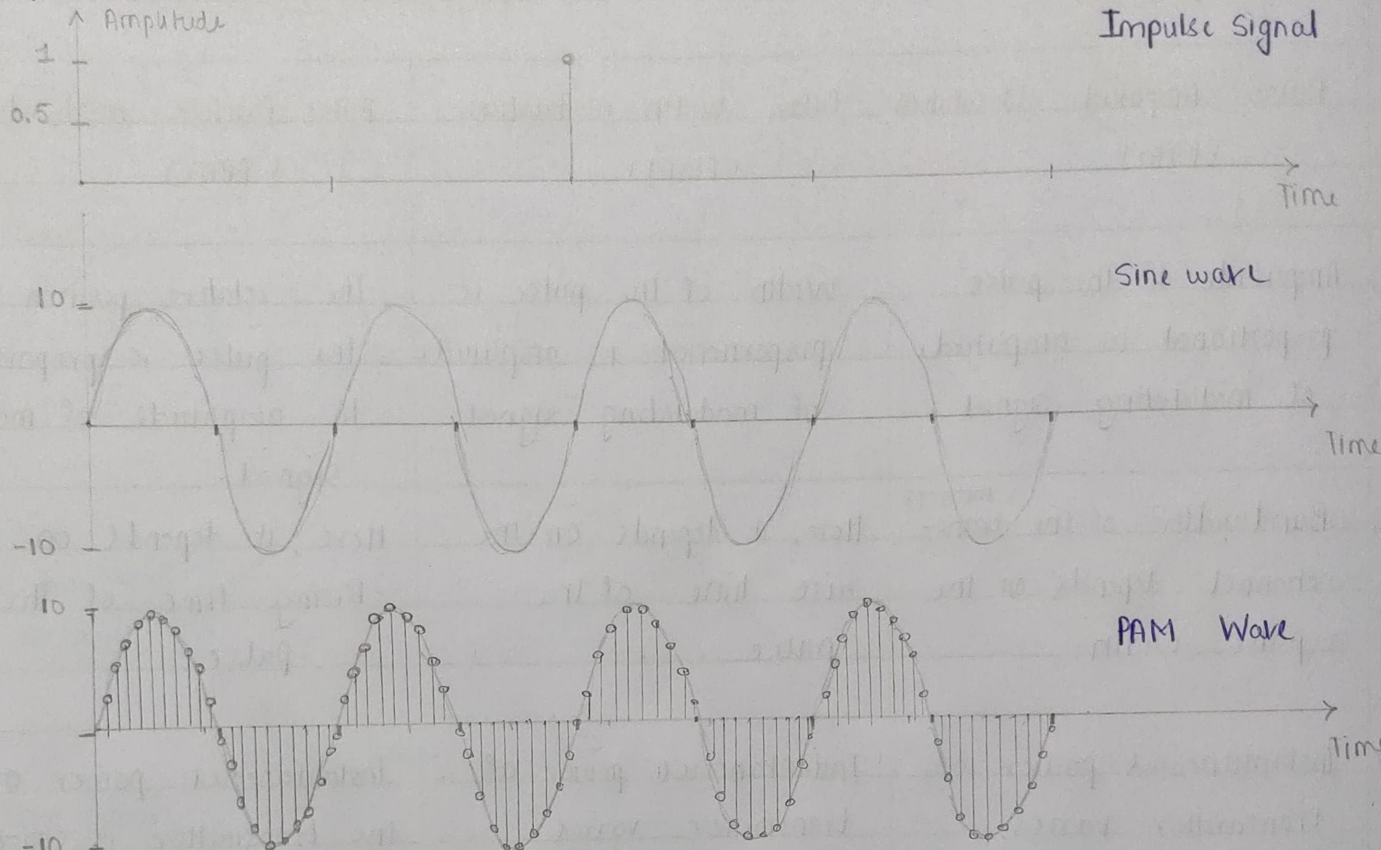
% PAM using Ideal Sampling

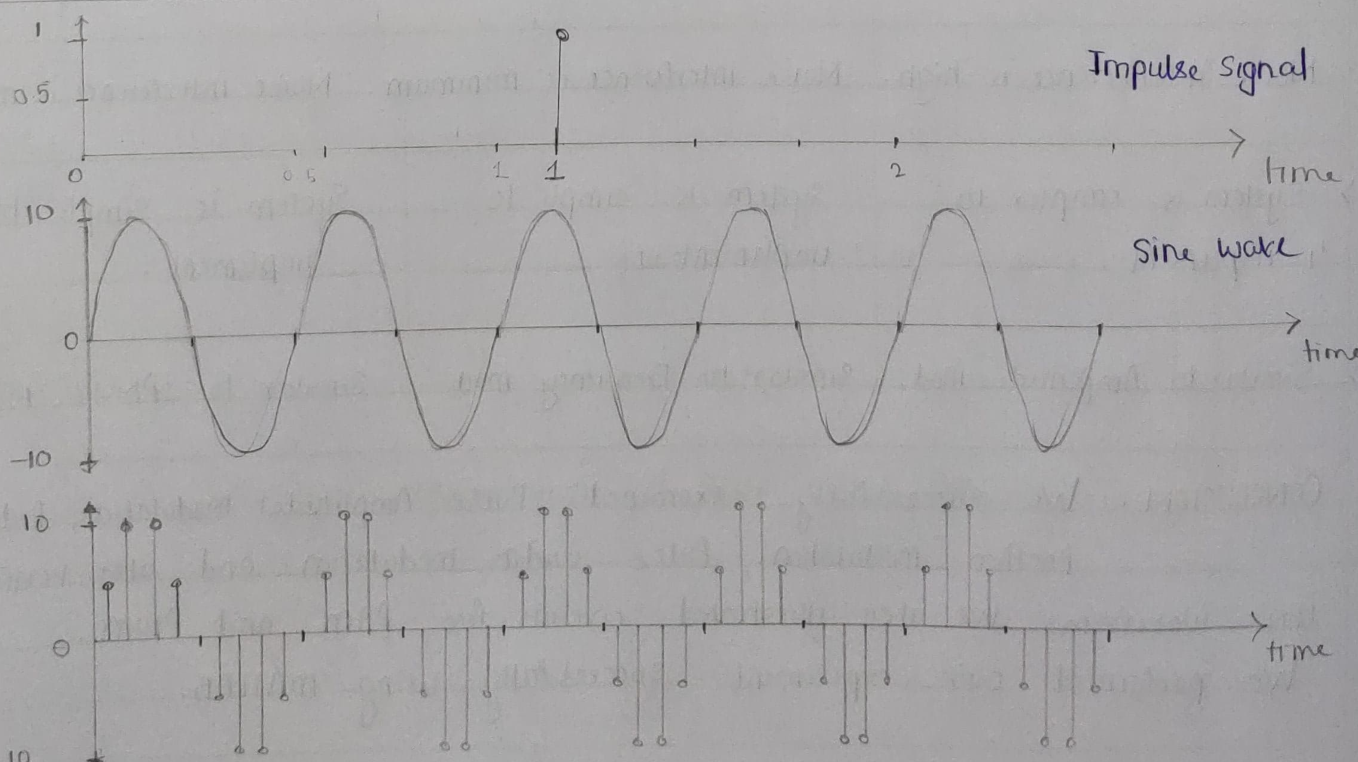
```
clc;
close all;
clear all;
a = input('Enter the amplitude = ');
f = input('Enter the frequency = ');
t = 0:0.02:2;
x1 = 1;                          % generation of Impulse signal
x2 = a*sin(2*pi*f*t);            % generation of sine wave
y = x1.*x2;                      % modulation step
subplot(3,1,1);                  % 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');
```

Teacher's Signature:

PAM with Ideal Sampling

1.7 A = 1V f = 2Hz





PAM

% PAM using Natural sampling

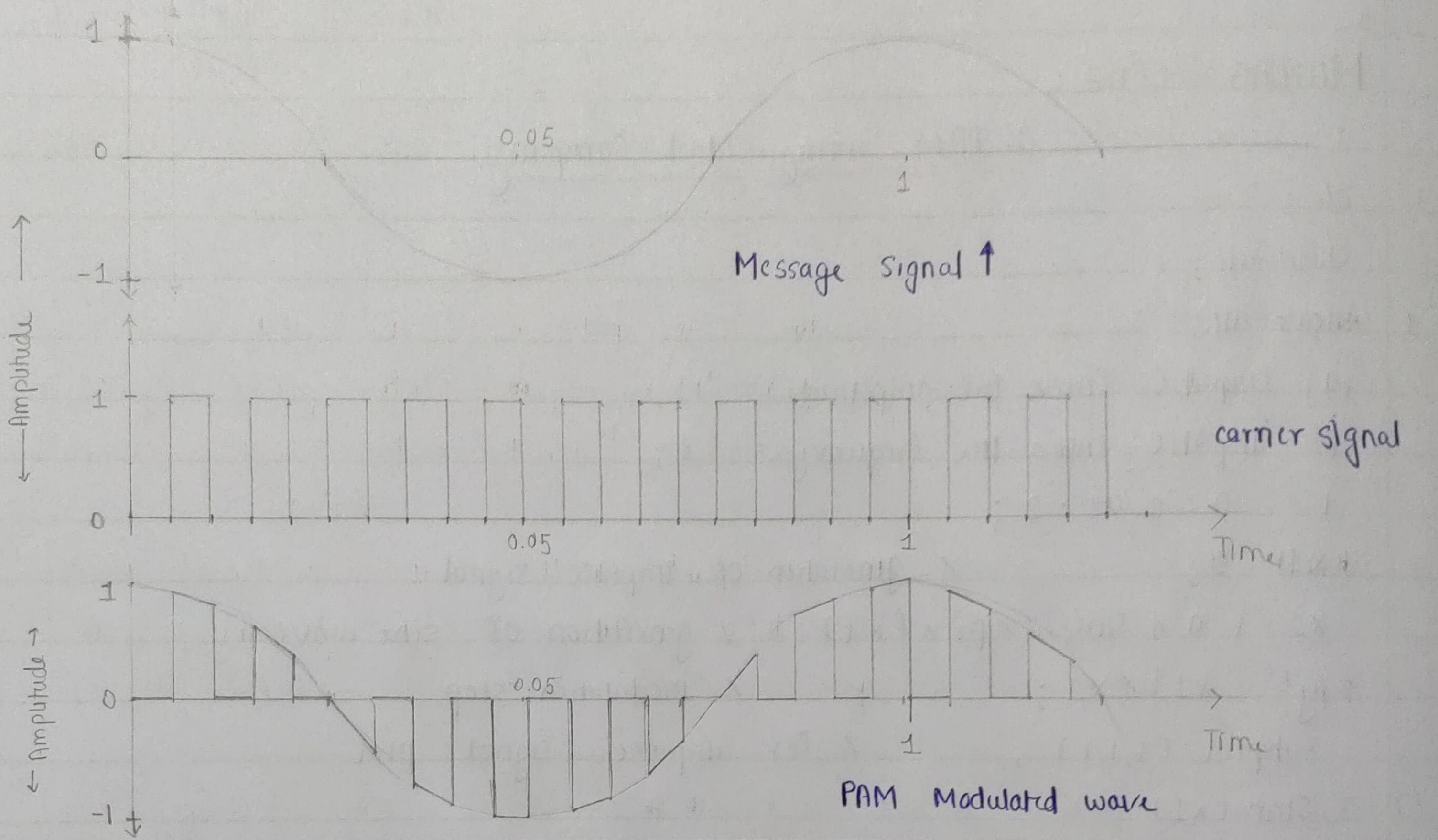
```
clc;  clear all;  close all;
fc = 100
fm = fc/10
fs = 100 * fc
t = 0: 1/fs : 4/fm ;
Msg_sgl = cos( 2*pi* fm *t);
Carr_sgl = 0.5 * square (2*pi* fc *t) + 0.5
Mod_sgl = Msg_sgl.* Carr_sgl ;
tt = [];
for i=1: length( Mod_sgl) ;
    if Mod_sgl(i) ==0 ;
        tt = [tt, Mod_sgl(i)] ;
    else
      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

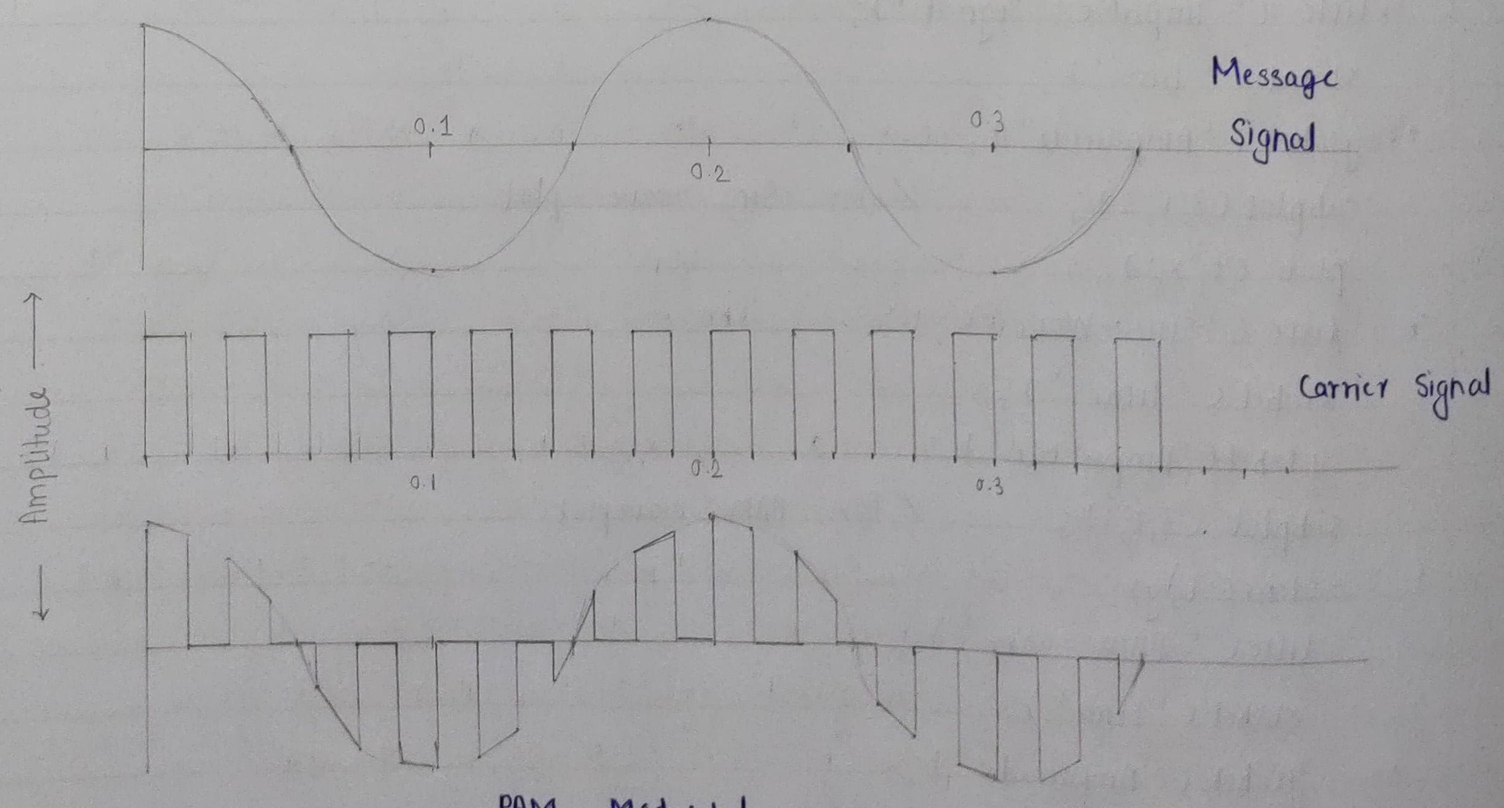
Message	Carrier	PAM Modulated
subplot(4,1,1);	subplot (4,1,2);	subplot(4,1,3);
plot(t, Msg_sgl);	plot (t, Carr_sgl);	plot (t, Mod_sgl);
title('message Signal');	title('Carrier Signal');	title('PAM modulated signal')
xlabel ('Time period');	xlabel ('Time period');	xlabel ('Time Period')
ylabel('Amplitude');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

PAM Using Square wave

(Natural Sampling)

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





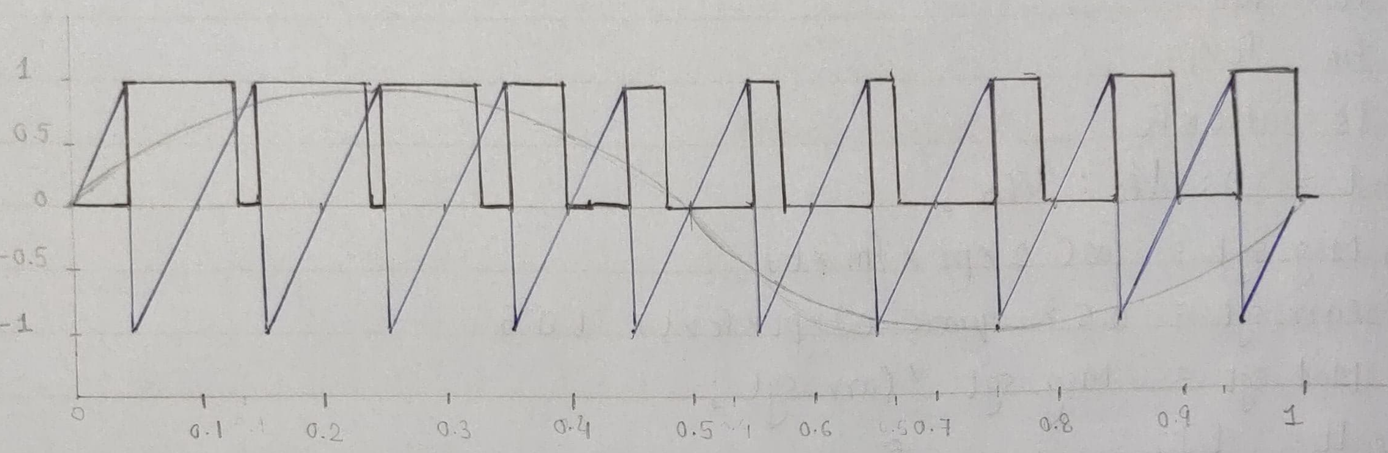
PAM Modulated wave

% 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;
    elseif ( 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;
```

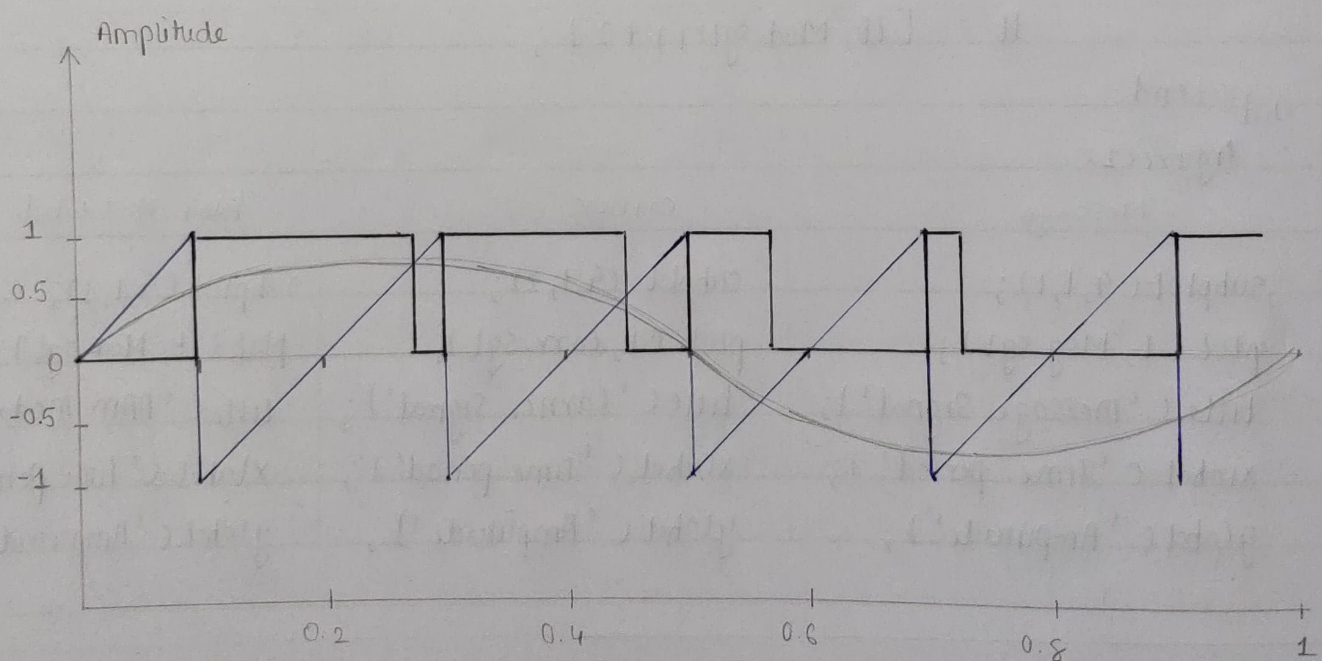
PWM

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

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[U19CS012]

% PPM Signal

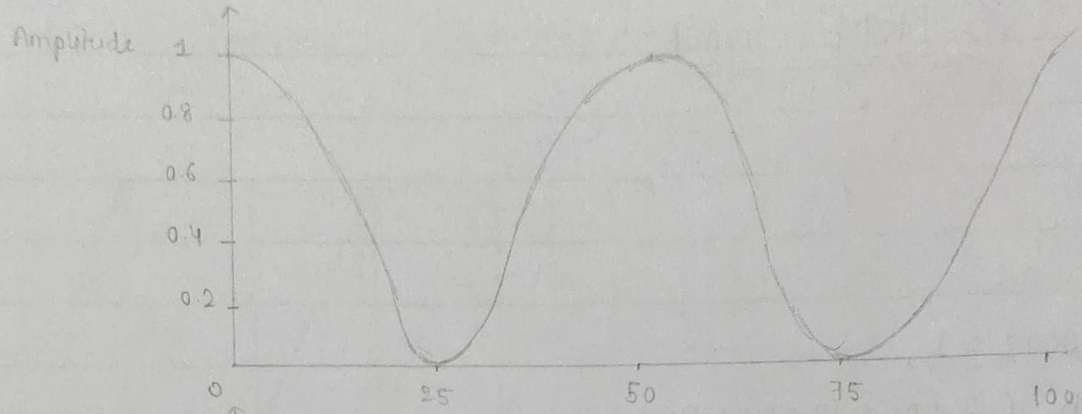
```
clc;
clear all;
close all;
fc = 10;
fs = 100;
fm = 2;
t = 0: 1/fs : ((2/fm) - (1/fs));
X = 0.5 * cos(2*pi*fm*t) + 0.5
Y = modulate(X, fc, fs, 'PPM');
subplot( 2,2,1);
plot(X);
title('msg signal')

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.

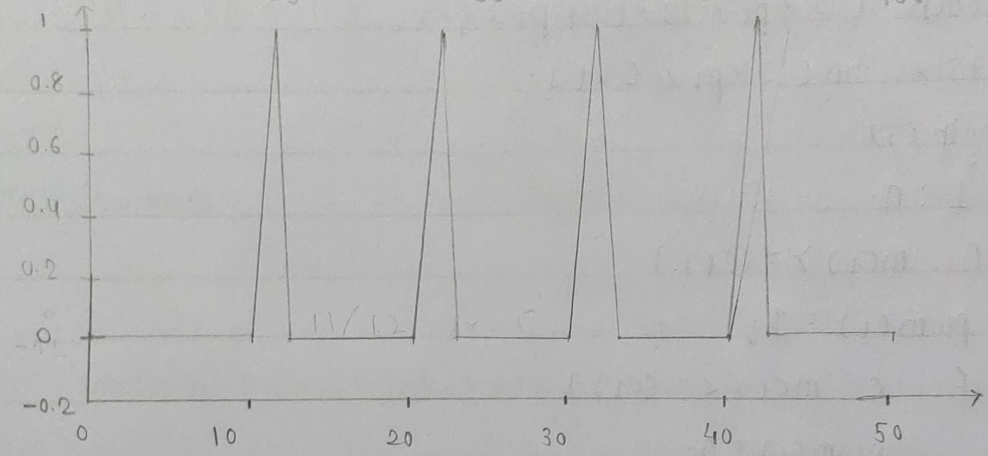
Teacher's Signature:

PPM Signal

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

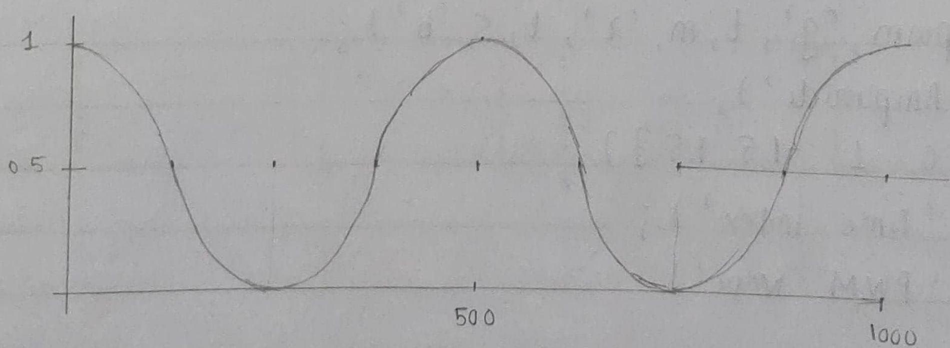


Msg Signal

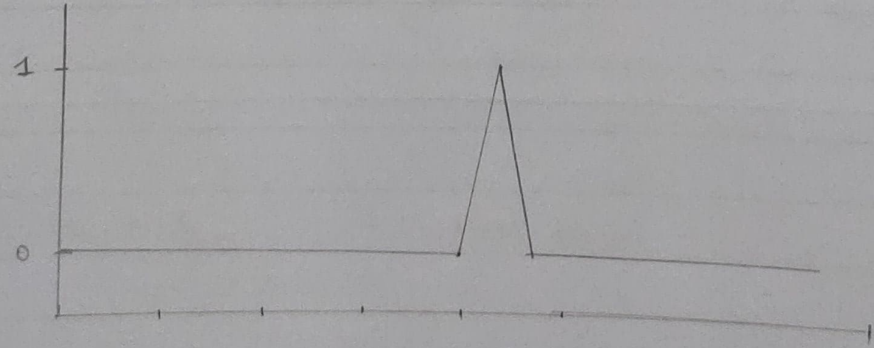


PPM

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



Msg Signal



PPM

EXPERIMENT - 6 [U19CS012]

ASK, FSK and PSK

AIM : To study Amplitude Shift keying (A.S.K.), Frequency Shift keying (F.S.K.) and phase shift keying (P.S.K) modulation Technique and verify waveforms.

APPARATUS : MATLAB

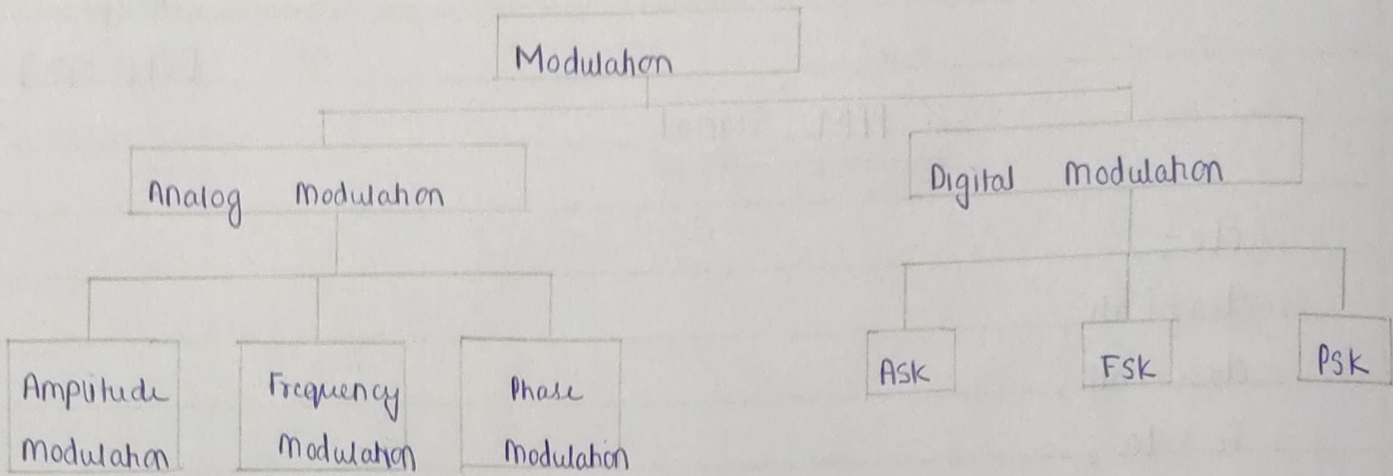
THEORY: 1.) Modulation : Modulation is a process, by which some characteristics of a carrier wave is varied in accordance with a modulating (message) signal.

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.

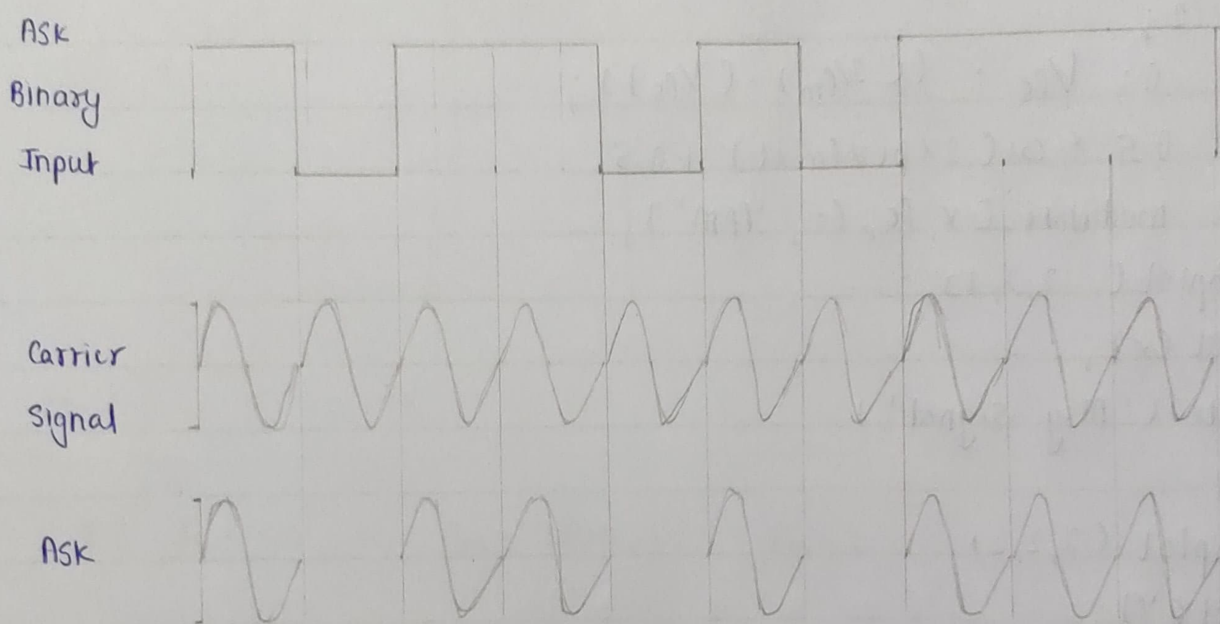
The ASK, FSK and PSK are analogous to AM, FM and PM respectively. The difference is that in digital modulation techniques (ASK, FSK and PSK) the modulation signal is digital in nature. while in AM, FM and PM modulating signal is analog in nature.

2.) ASK (Amplitude Shift keying)

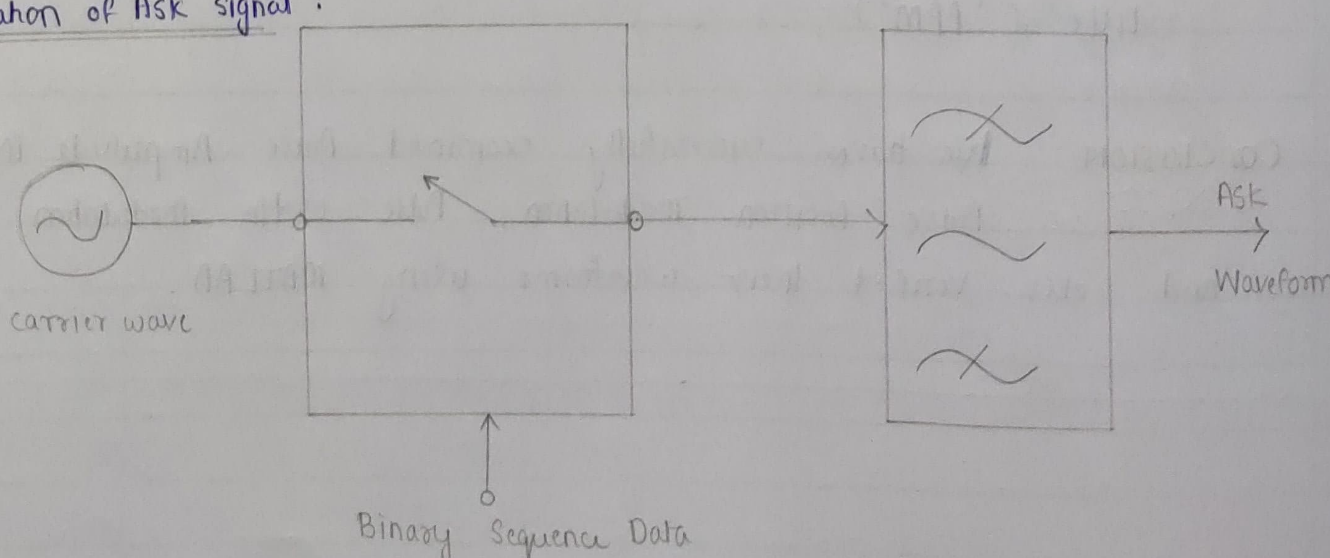
In ASK, the amplitude of the carrier wave is changed (switched) according to the digital input signal (modulating signal)



ASK



Generation of ASK signal :



Application of ASK : 1.) Wireless Base station
2.) Low Frequency RF Application
3.) ~~Instru~~ 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)

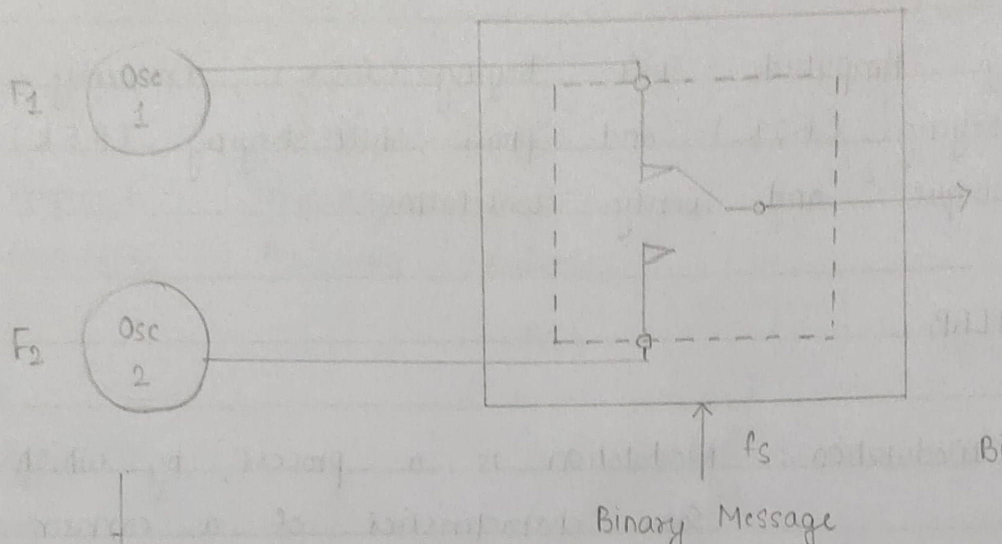
In PSK, phase of the carrier wave (analog in nature) is switched as per the input digital signal.

Application of PSK :

1. It is widely used for wireless LANs, RFID and Bluetooth communication.

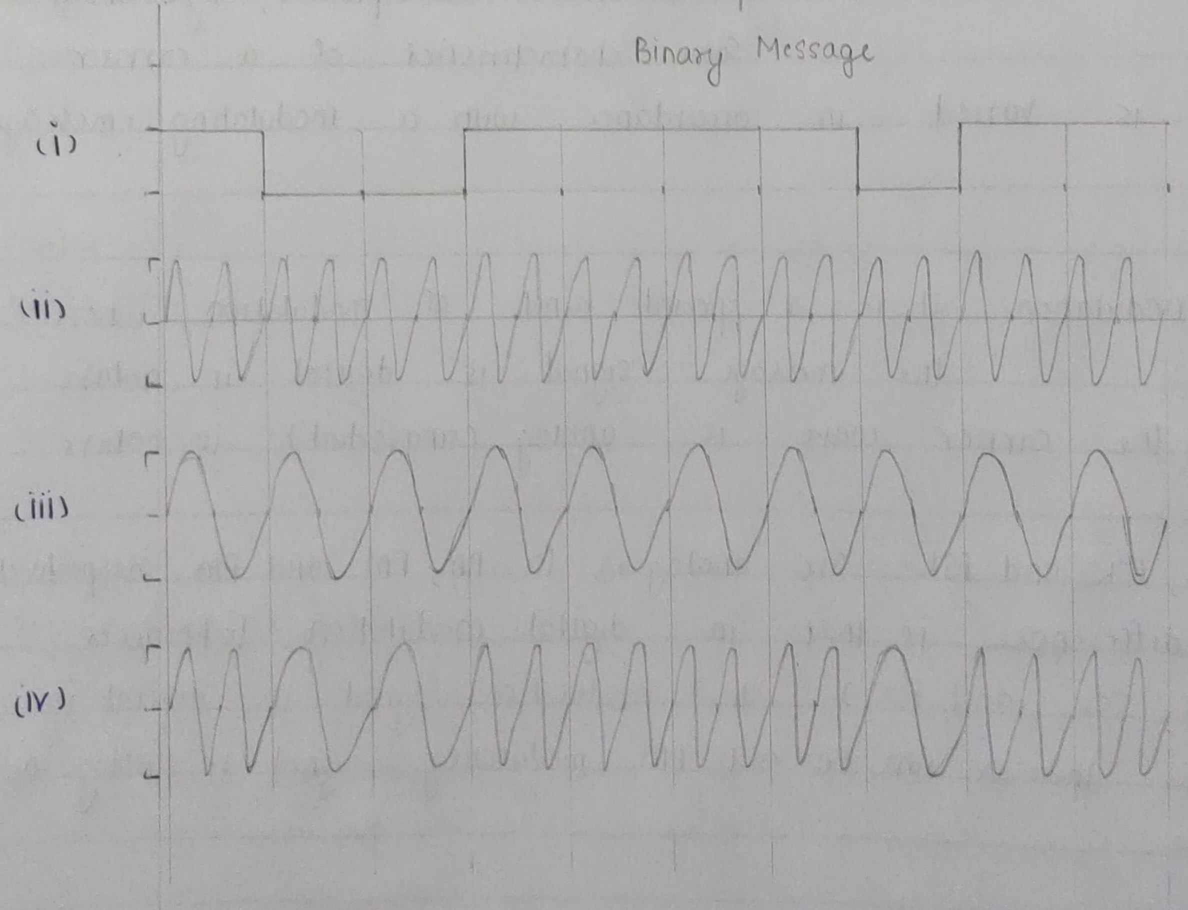
FSK

Generation of FSK



$F_1 \cong F_2$

Bit rate $F_s \ll F_1$



(i) Digital Bitstream

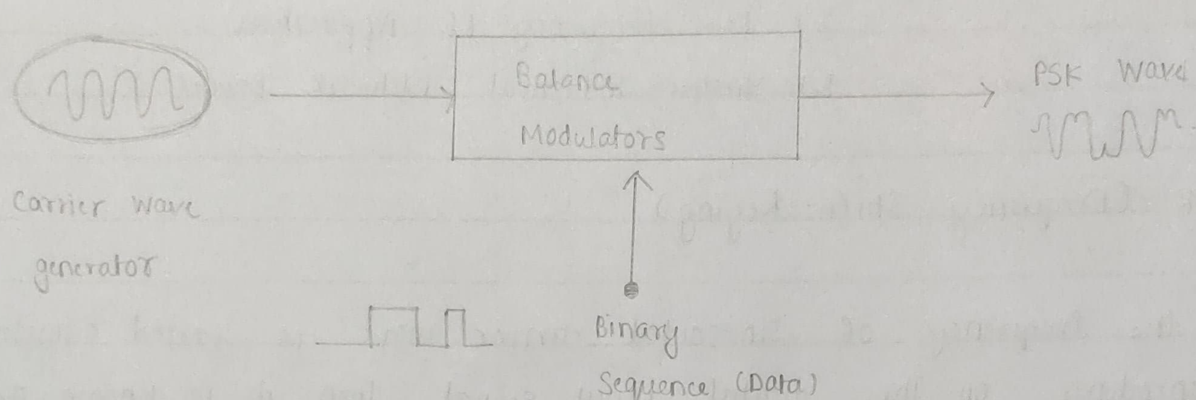
(ii) High frequency carrier wave

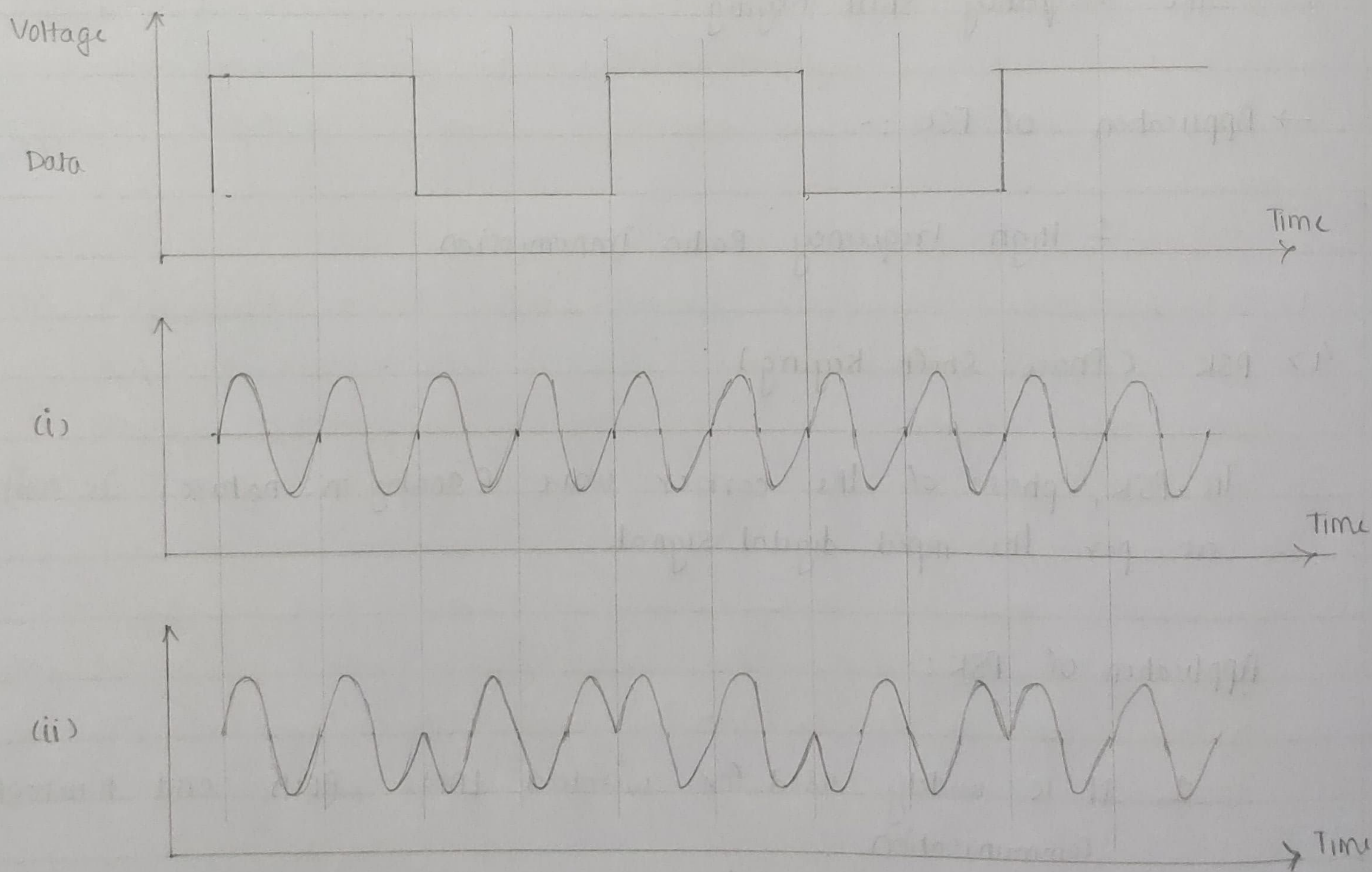
(iii) Low frequency carrier wave

(IV) FSK modulated wave

PSK

Generation of PSK





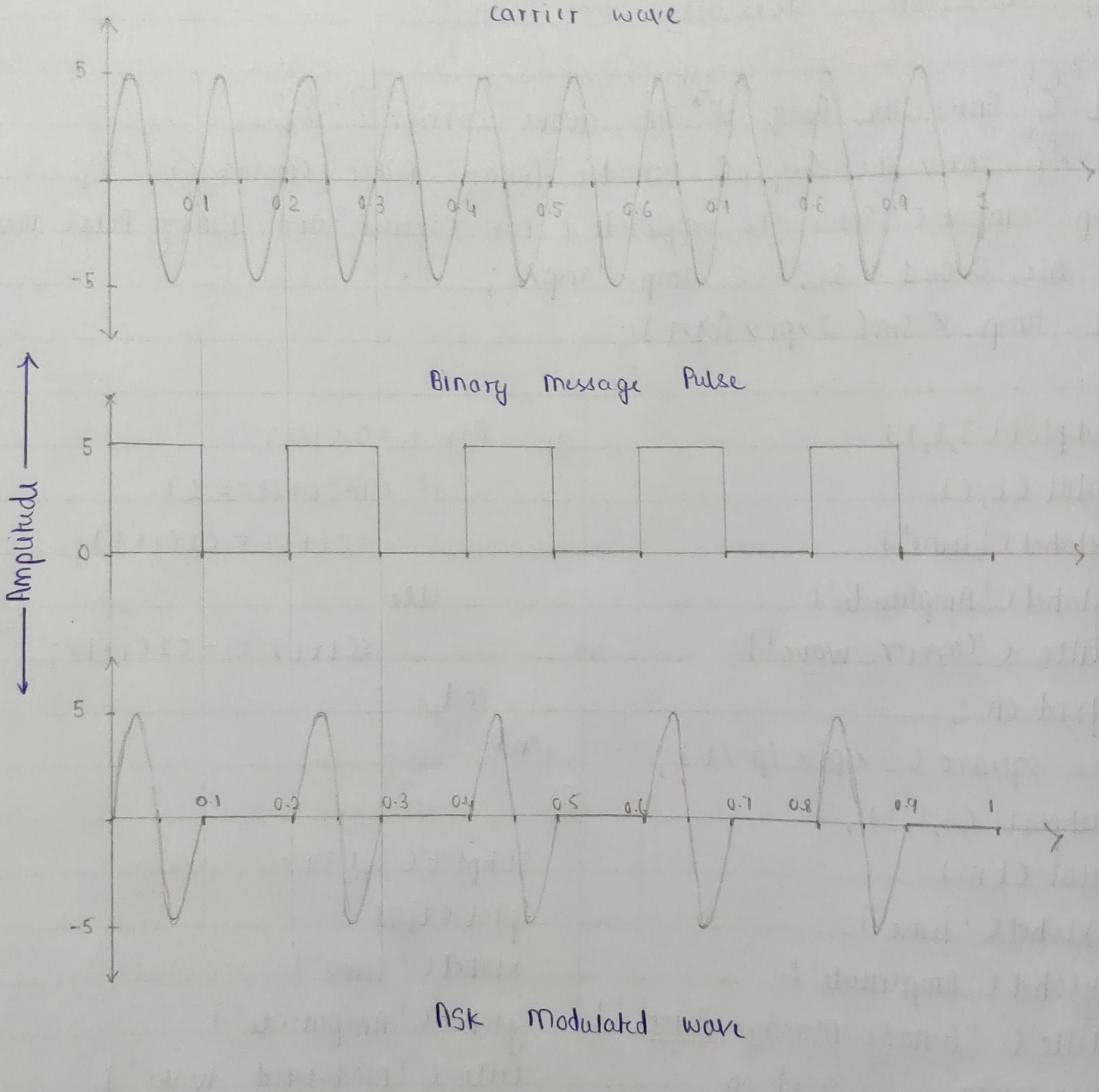
(i) Carrier Frequency Before modulation
(ii) Carrier Frequency After modulation

> MATLAB CODE : %% ASK

```
clc; close all; clear all; % for deleting all variables from memory
fc = input ('Enter the freq of Sine wave carrier:');
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 . * square (2*pi* fp *t) + (amp/2);  % square wave msg
subplot (3,1,2)   % Plotting Square Binary Pulse
plot (t,m)
xlabel ('Time')
ylabel ('Amplitude')
title ('Binary Message Pulses')
w = c*m    % The Shift Keyed wave
subplot (3,1,3)   % for plotting ASK wave
plot (t,w)
xlabel ('Time')
ylabel ('Amplitude')
title ('Amplitude Shift keyed Signal')
```

(A) ASK :

3.) $F_c = 10\,Hz$ $F_p = 5\,Hz$ Amp = 5



%% FSK

```
clc; close all; 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;
t = 0: 0.001 : 1;
c1 = amp.*sin(2*pi*fc1*t);
c2 = amp.*sin(2*pi*fc2*t);
```

①
```
subplot(4,1,1);
plot(t,c1)
xlabel('Time')
ylabel('Amplitude')
title('carrier 1 wave')
```

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

③
```
m = amp.*square(2*pi*fp*t) + amp;   % square wave form
subplot(4,1,3)
plot(t,m)
xlabel('Time')
ylabel('Amplitude')
title('Binary message Pulses')
```

④
```
for i=0:1000
    if m(i+1)==0
        mm(i+1)= c2(i+1);
    else
        mm(i+1)=c1(i+1);
    end
end
```

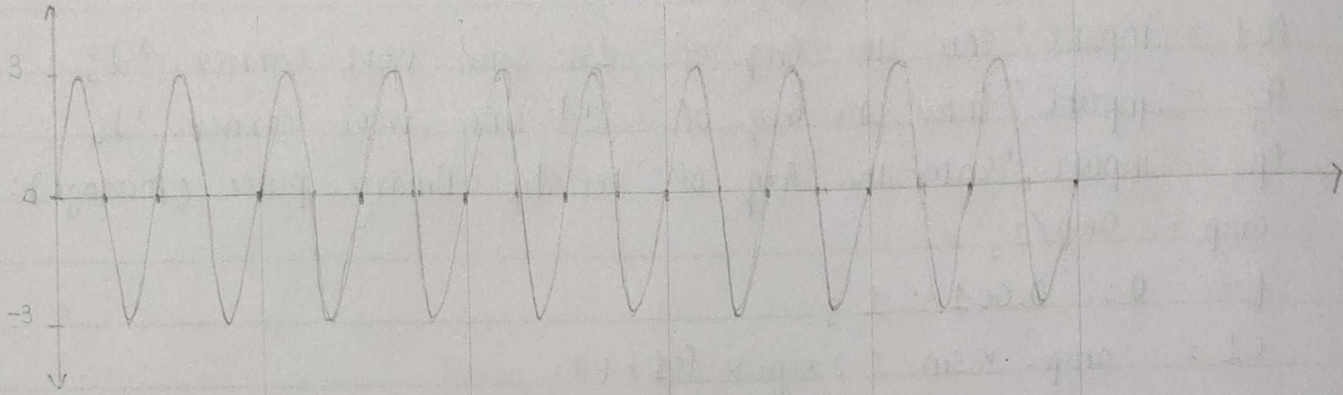
⑤
```
subplot(4,1,4)
plot(t,mm)
xlabel('Time')
ylabel('Amplitude')
title('Modulated wave')
```

(B)

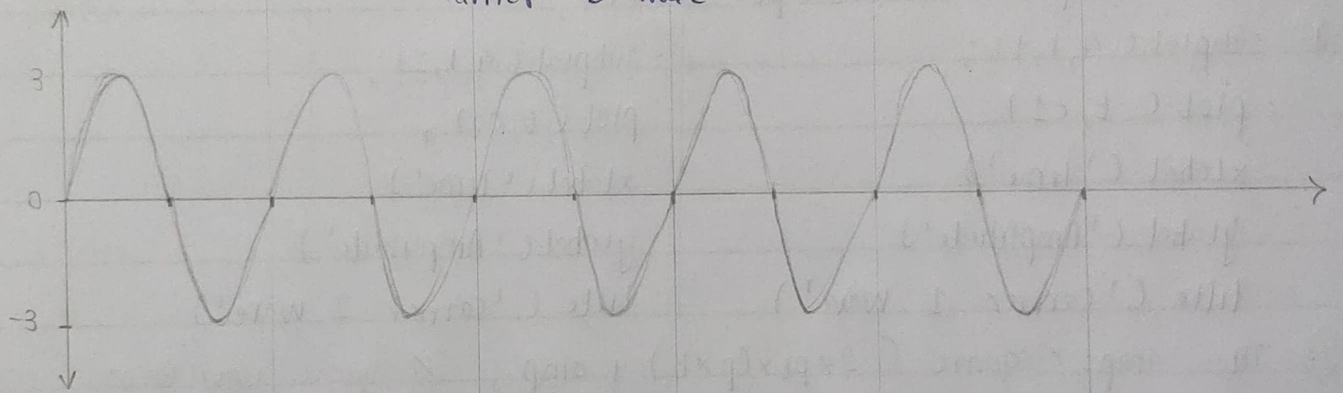
FSK :

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

Carrier 1 wave

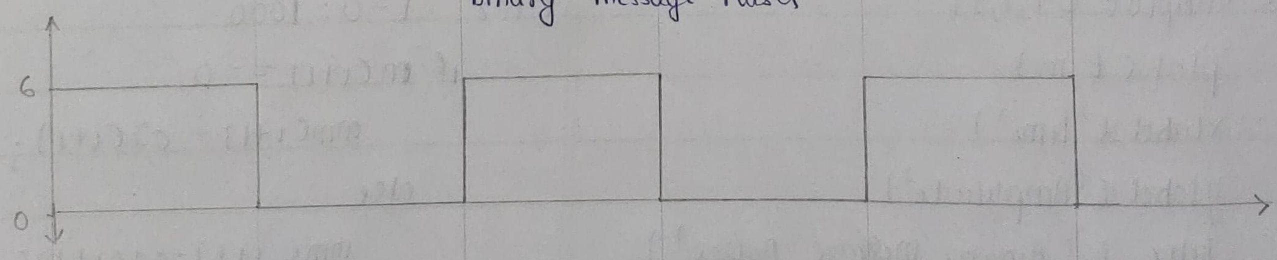


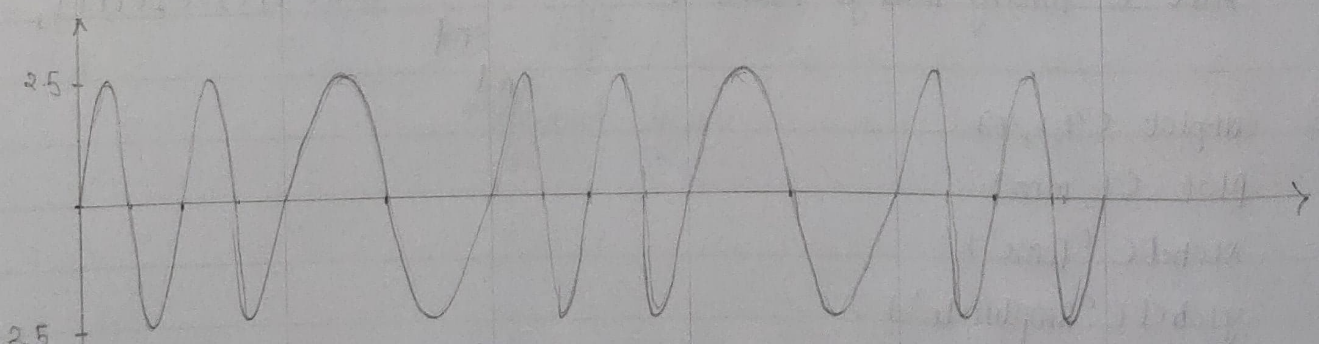
Carrier 2 wave



Amplitude

Binary Message Pulses





FSK modulated wave

Time

%% PSK

```
clc; close all; clear all;

fc1= input(' Enter the freq of 1st sine wave carrier :');
fp = input(' Enter the freq of Periodic Binary Pulse (message) :');
amp = input(' Enter the amplitude ( For Carrier and Binary Pulse Message)');
t = 0: 0.001 : 1;        amp = amp/2;
c1= amp. * sin( 2*pi * fc1*t);
```

①

```
Subplot(3,1,1)
plot(t,c)
xlabel('Time')
ylabel('Amplitude')
title ('Carrier wave')
grid on;
m= square (2*pi* fp *t);
```

②

```
subplot (3,1,2),
plot (t,m)
xlabel('Time')
ylabel('Amplitude')
title ( 'Binary Message Pulses')
w = c x m        grid on
subplot(3,1,3)
```

③

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

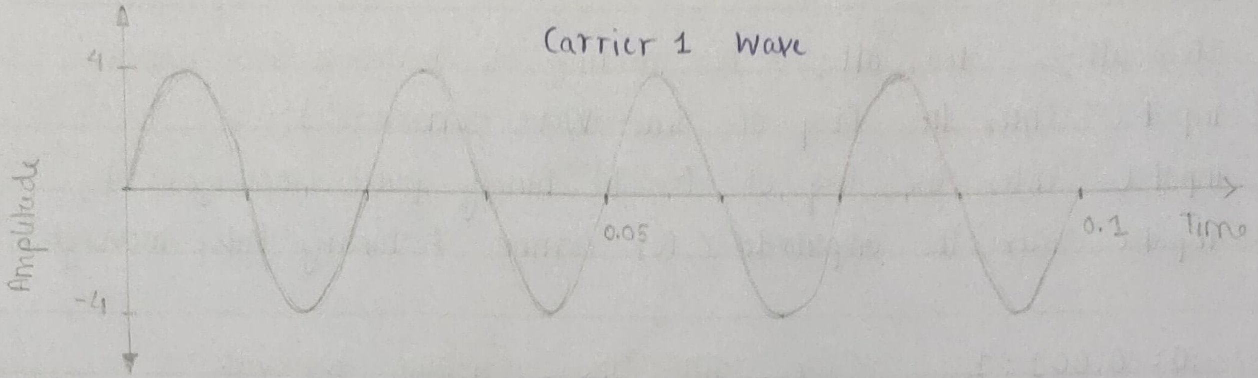
④

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

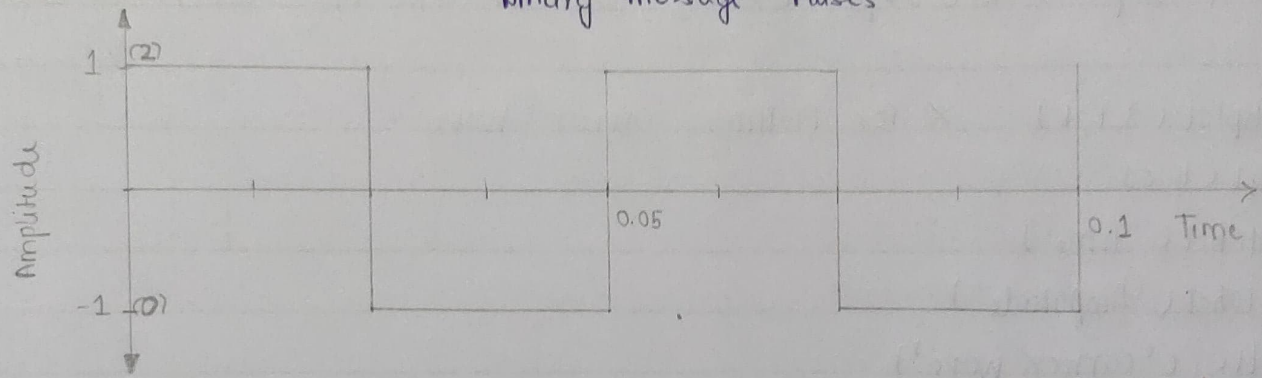
Waveforms using MATLAB

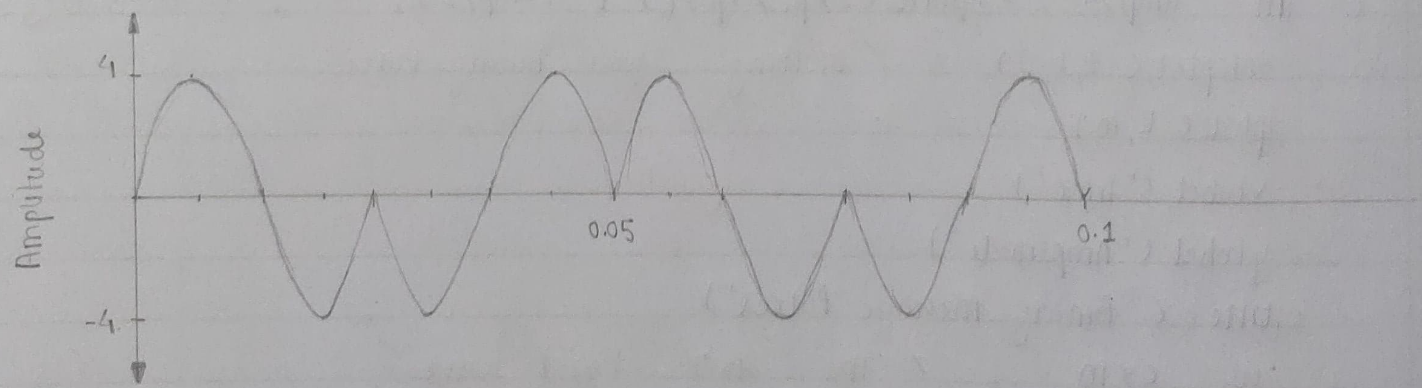
$\boxed{\text{PSK :}}$ 1.) $F_{c1} = 40\,Hz$ $F_p = 20\,Hz$ $Amp = 8\,V$

Carrier 1 Wave



Binary message Pulses





Modulated Wave

[U19CS012]

CONCLUSION :

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

EXPERIMENT 7 :

[U19CS012]

EFFECT OF AWGN on AM and FM

AWGN : Additive white Gaussian noise

AIM: To study the transmission of Amplitude modulated (AM) and Frequency modulated (FM) signal under the Additive Gaussian Noise channel.

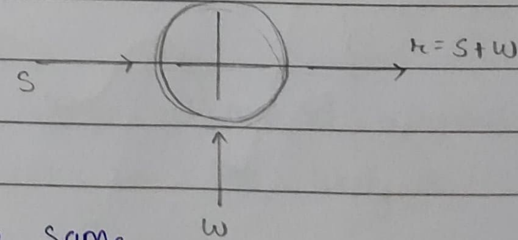
APPARATUS:- MATLAB

THEORY: 1.> Additive White Gaussian Noise (AWGN)

A Basic Noise model used to mimic the effect of many random processes that occur in nature. Channel produces Additive white Gaussian Noise. (AWGN)

(a) Additive - The received signal equals the transmit signal plus some noise, where the noise is statistically independent of the signal.

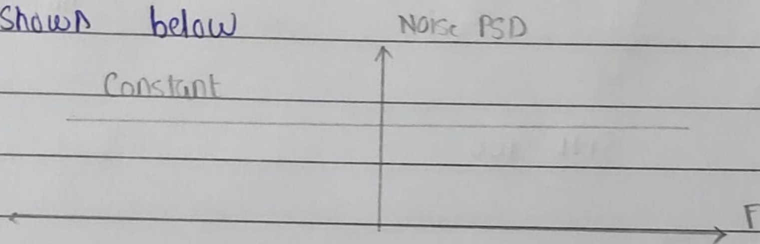
$$r(t) = s(t) + w(t)$$



(b) White : It refers that the noise has the same power distribution at every frequency OR it has uniform power across the frequency band for the information system.

- It is an analogy to the color white which has uniform emissions at all frequencies in the visible spectrum. If we focussed a beam of light for each color on the visible spectrum onto a single spot, that combination would result in a beam of white light.

- As a consequence, the Power spectral Density (PSD) of white noise is constant for all frequencies ranging from $-\infty$ to $+\infty$ as shown below



ⓒ Gaussian - Gaussian distribution, or a normal distribution, has an average of zero in the time domain, and is represented as a bell shaped curve.

- The probability distribution of the noise samples is Gaussian with a zero mean.

- The values close to zero have a higher chance of occurance while the values far away from zero are less likely to appear.

- In reality, the ideal flat spectrum from $-\infty$ to $+\infty$ is true for frequencies of interest in wireless communication (few KHz - Hundred GHz) but not for higher frequencies.

2.) Signal to Noise Ratio

- The SNR or S/N is a measure used in science and engineering that compares the level of a desired signal to the level of background noise.

- It is defined as the ratio of signal power to the noise power, often expressed in decibels. A ratio higher than 1:1 (greater than 0 dB) indicates more signal to that noise.

- SNR, bandwidth and channel capacity of a communication channel are connected by the Shannon-Hartley theorem

$$SNR_{dB} = 10 \log_{10}\left(\frac{P_{signal}}{P_{noise}}\right)$$

Shannon-Hartley Theorem

It states that the channel capacity (bits per second) OR information rate of data that can be communicated at low error data using an average received signal power through communication channel subject to additive white Gaussian noise (AWGN) of power.

$$C = B \log_2\left(1 + \frac{S}{N}\right)$$, B = Bandwidth of channel in Hz

- It is related to signal to noise (SNR) or carrier to noise (CNR) [Linear Power ratio]

- 5 dB - 10 dB ⇒ It is below the minimum level to establish a connection due to the noise level being nearly indistguishable from the desired signal (useful information)

- 25 dB - 40 dB = deemed to be good.

- 41 dB or higher = considered to be Excellent

3.) Mathematics of AM

- let modulating signal be $e_m = E_m \sin(\omega_m t)$
  carrier signal be $e_c = E_c \sin(\omega_c t)$

$$\therefore E_{AM} = E_c + e_m$$
$$= E_c + E_m \sin(\omega_m t)$$

The instantaneous value of the amplitude modulated wave can be given as

$$e_{AM} = E_{AM} \sin(\theta)$$
$$= E_{AM} \sin(\omega_c t)$$
$$e_{AM} = (E_c + E_m \sin(\omega_m t)) \sin(\omega_c t)$$

This is an equation of AM wave.

4.) Mathematics of FM

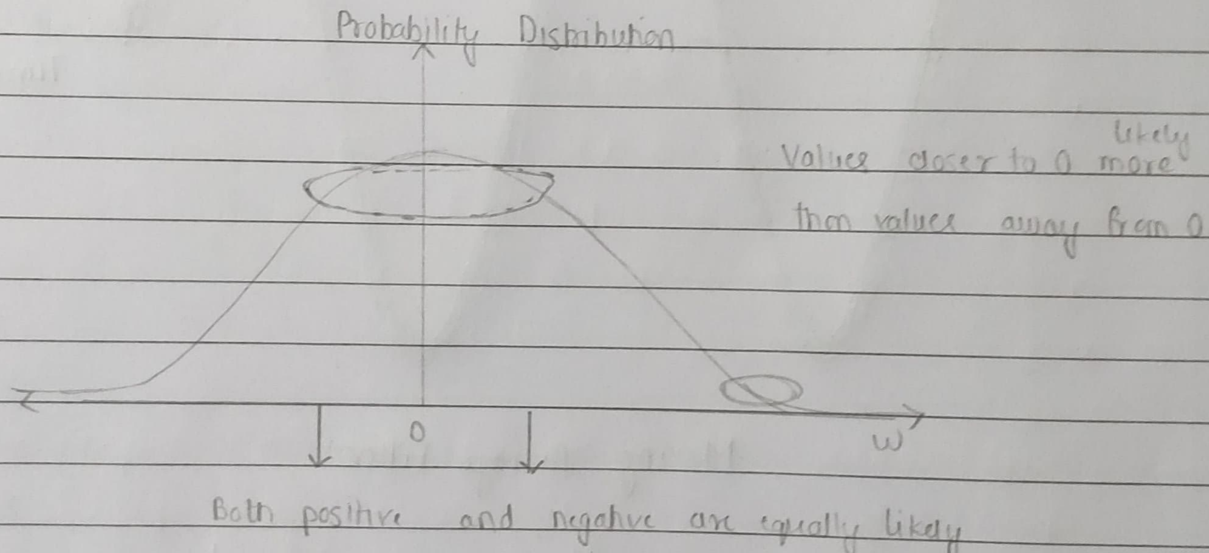
$$s(t) = E_s \sin(\omega_c t + m_s \sin(\omega_m t))$$

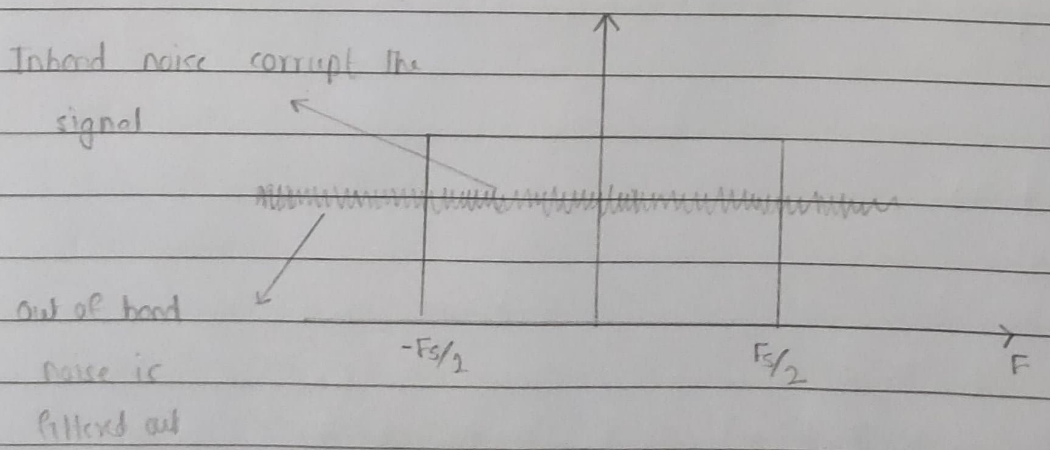
This is the expression for FM wave, m_f = modulation index

$$m_F = \frac{\text{Frequency deviation}}{\text{Modulating Frequency}} = \frac{\Delta f}{f_m}$$

- Frequency deviation Δf = represents the maximum departure of the instantaneous frequency $F_i(t)$ of the FM wave from the carrier frequency F_c.

⇒ Gaussion





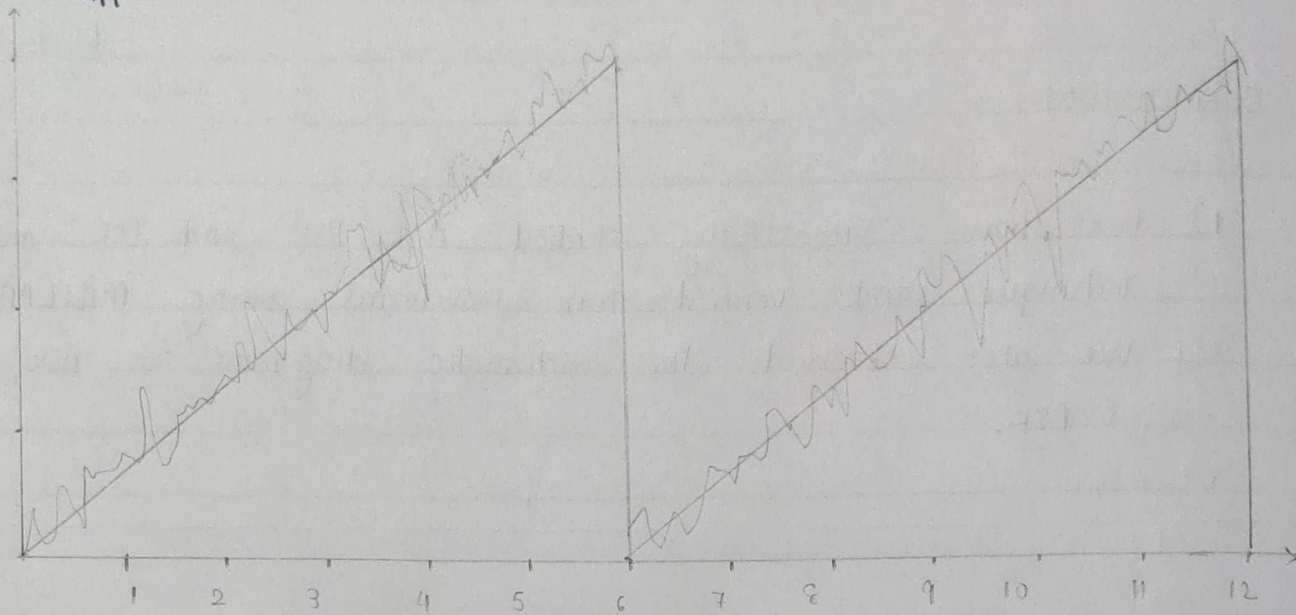
MATLAB CODE :

AWGN in different function

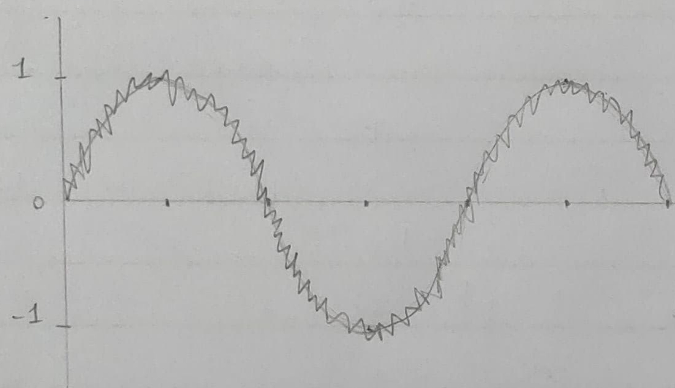
```
clc;
clear all;
t= 0:0.1 :10);
x= sawtooth (t);
y= awgn (x, 10, 'measured');
plot( t, [x,y]);
legend ('original signal', 'signal with AWGN');

clc;
clear all;
t= (0 : 0.1 : 10);
x = sin( t);
y = awgn( x, 10, 'measured' );
plot ( t, [x,y] );
legend ( 'original signal', 'signal with AWGN')

clc;
clear all;
t= (0: 0.1 : 10);
x = cos (t);
y= awgn (x, 10, 'measured');
plot ( t, [x,y]);
legend ('original signal', 'signal with AWGN');
```

AWGN Effect on Different functions

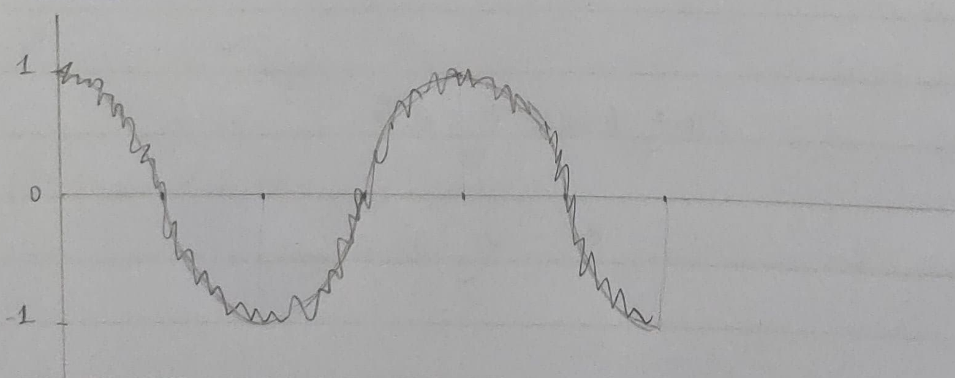
1) Sawtooth



2) Sine



3) Cosine



MATLAB CODE :

AWGN in AM

```
clc;
clear all;
t= 0:0.001:1;
vm= 5;
vc= 10;
fm= 2;
fc= 25;

m = vm * sin( 2*pi* fm*t);
c = vc * sin( 2* pi* fc*t);
amp = vc + vm * sin( 2*pi* fm*t);
am = amp.* sin (2*pi* fc*t);
y = awgn (am, 10, 'measured');
```

①

```
subplot (4,1,1);
plot (t, m);
xlabel ('time')
ylabel (' amplitude)
```

②

```
subplot (4,1,2);
plot( t,c);
xlabel ('time');
ylabel ('amplitude');
```

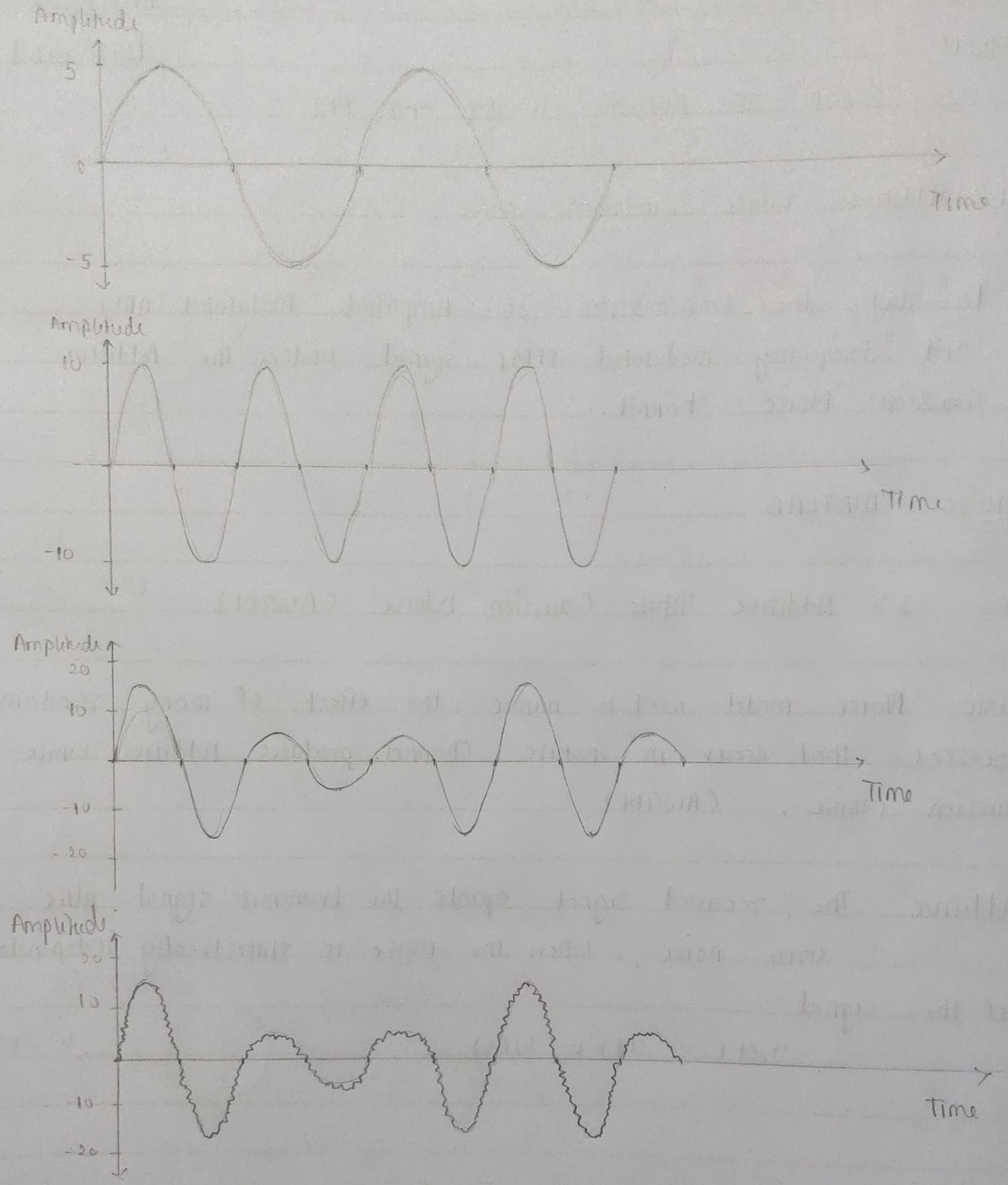
③

```
subplot (4,1,3);
plot (t, am);
xlabel ('time')
ylabel ('amplitude')
```

④

```
subplot (4,4,4);
plot (t, y);
x label ('time')
y label ('amplitude')
title ( 'amplitude modulated signal with AWGN');
```

AM signal under AWGN



AWGN in different SNR

```
clc; clear all;
t = 0 : 0.001 : 1;
vm = 5;      fm = 2;
vc = 10;     fc = 25;
m = vm * sin( 2*pi* (fm*t) );
c = vc * sin( 2*pi* fc*t );
amp = vc + vm* sin( 2*pi* fm*t );
am  = amp. * sin( 2*pi* fc*t );
y1 = awgn ( am, 10 , 'measured' );
y2 = awgn ( am, 100 , 'measured' );
y3 = awgn ( am, 1000 , 'measured' );
```

(1)

```
subplot (4,1,1);
plot (t, am)
xlabel ('time');
ylabel ('amplitude');
title ('amplitude modulated signal');
```

(2)

```
subplot( 4,1,2);
plot( t, y1);
xlabel ( 'time' )
ylabel ( 'amplitude')
title (' AM signal with AWGN [snr 10]');
```

(3)

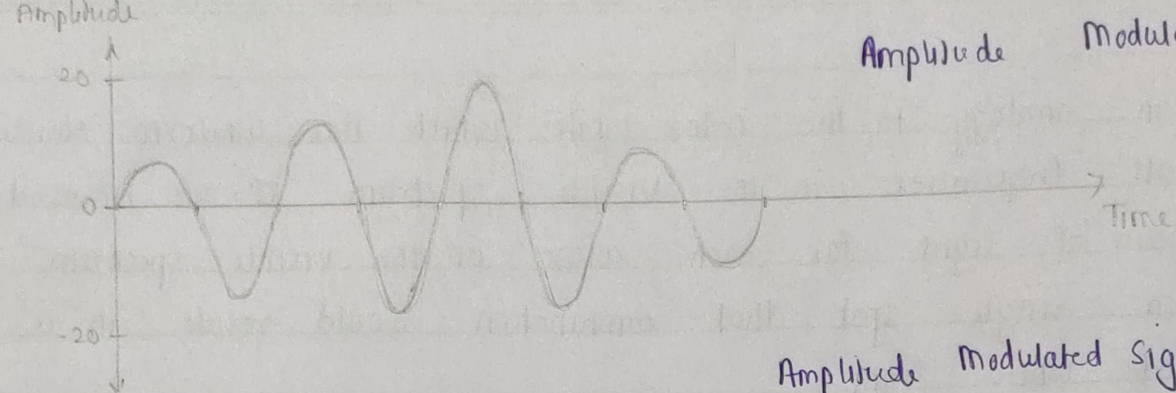
```
subplot (4,1,3);
plot (t, y2);
xlabel ( 'time');
ylabel ('amplitude');
title (' AM signal with AWGN [snr 100]');
```

(4)

```
subplot (4,1,4);
plot (t, y3);
xlabel ('time')
ylabel ('amplitude')
title (' AM signal with AWGN
      [snr 1000] ');
```

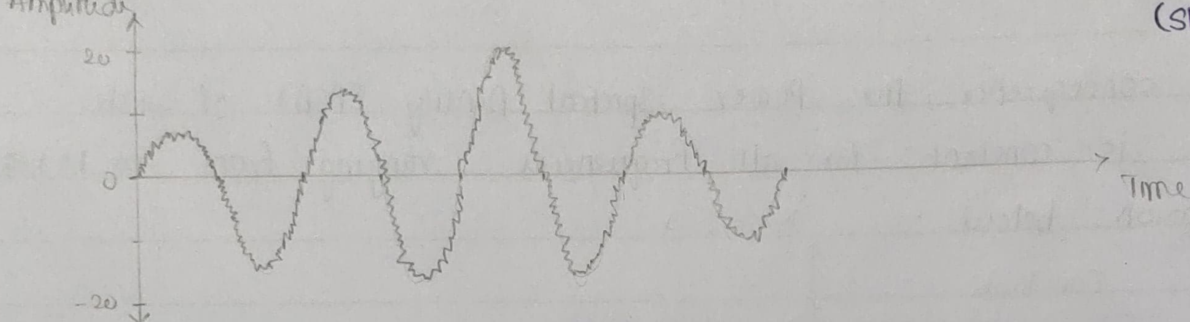
AM Signal with different SNR values

Amplitude Modulated signal

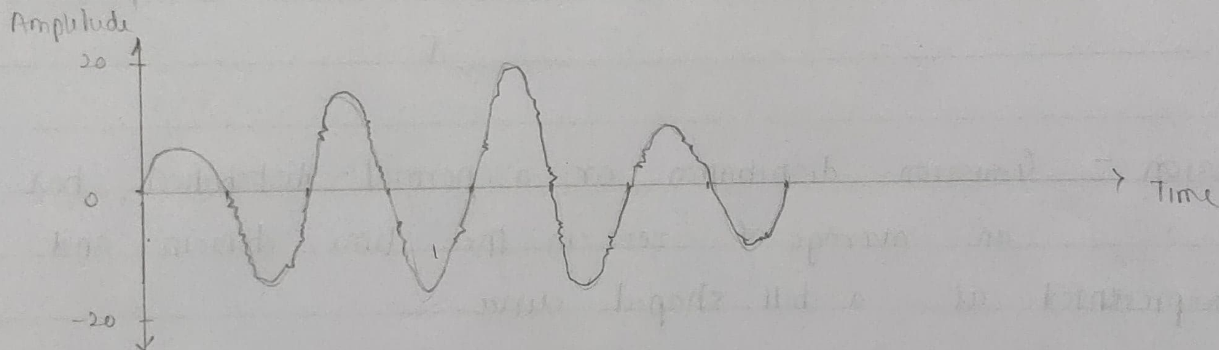


Amplitude Modulated Signal with (AWGN)

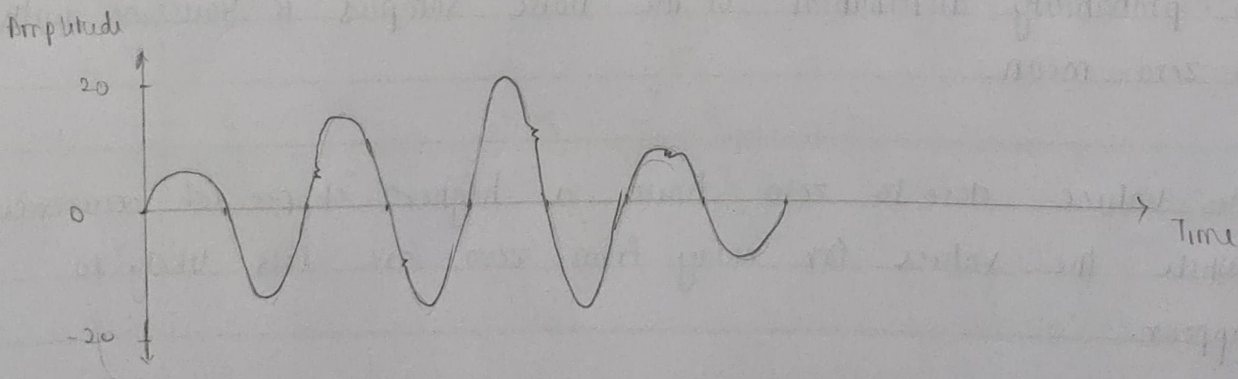
(SNR = 10)



SNR = 100



SNR = 1000



AWGN in FM

```
clc;
clear all;
t = 0 : 0.001 : 1;
Vm = 5;
Vc = 5;
fm = 2;
fc = 25;
fd = 5;
msg = Vm*sin(2*pi*fm*t);
c = Vc*sin(2*pi*fc*t);
y = Vc*sin(2*pi*fc*t + fd.*cos(2*pi*fm*t));
z = awgn(y,5,'measured');
```

(1)

```
subplot(4,1,1);
plot(t,msg);
xlabel('time');
ylabel('Amplitude');
title('message signal');
```

(2)

```
subplot(4,1,2);
plot(t,c);
xlabel('time');
ylabel('amplitude');
title('carrier signal');
```

(3)

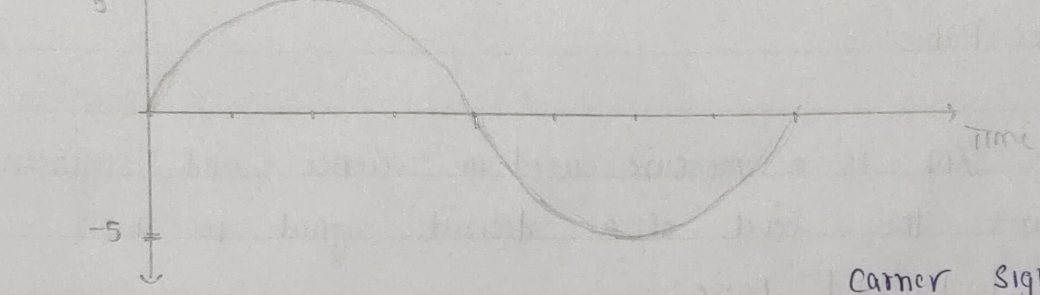
```
subplot(4,1,3);
plot(t,y);
xlabel('time');
ylabel('Amplitude');
title('Frequency modulated signal')
```

(4)

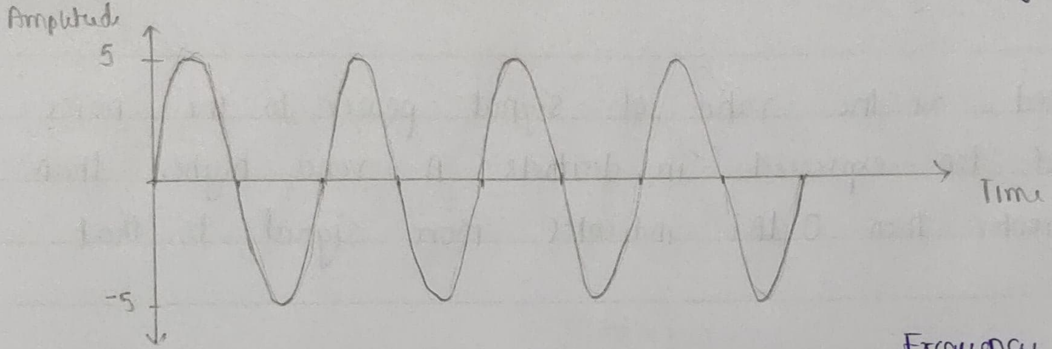
```
subplot(4,1,4);
plot(t,z);
xlabel('time');
ylabel('amplitude');
title('Frequency modulated signal with AWGN');
```

FM Signal under AWGN

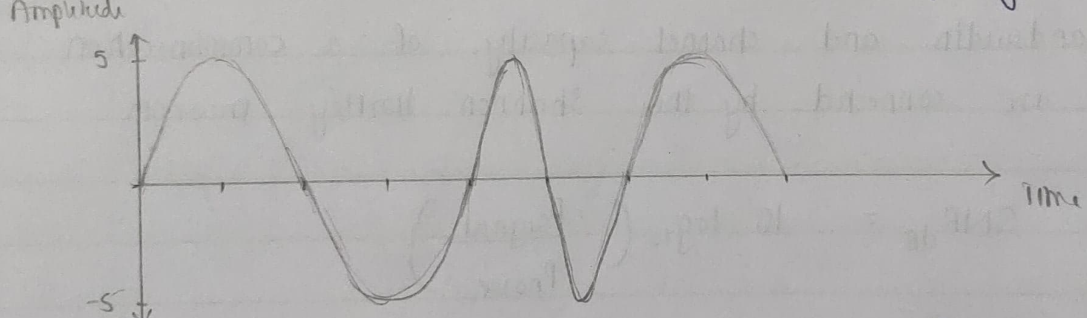
Message Signal



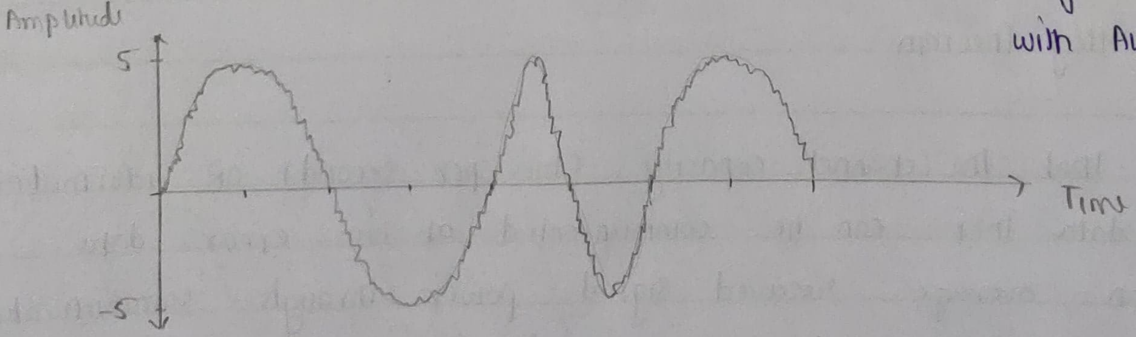
Carrier Signal



Frequency Modulated Signal



Frequency Modulated signal with AWGN



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FM in Different SNR

```
clc; clear all;
t= 0: 0.001 : 1;
vm = 10;        fm = 2;
vc = 5;         fc = 25;
fd = 10
m = vm * sin( 2*pi*fm * t);
c= vc* sin( 2*pi* fc*t);
amp = vc + vm*sin( 2* pi*fm*t);
y = vc*sin ( 2*pi*fc*t + fd.*cos(2*pi* fm*t) );
y1 =awgn ( y, 1, 'measured' );
y2 = awgn ( y, 10, 'measured');
y3 = awgn ( y, 100, 'measured' );
```

①
```
subplot (4,1,1);
plot ( t, y);
xlabel ('time')
ylabel('amplitude')
title ('Frequency modulated signal')
```

②
```
subplot (4,1,2);
plot( t, y1);
xlabel ('time');
ylabel ('amplitude');
title ('snr 10');
```

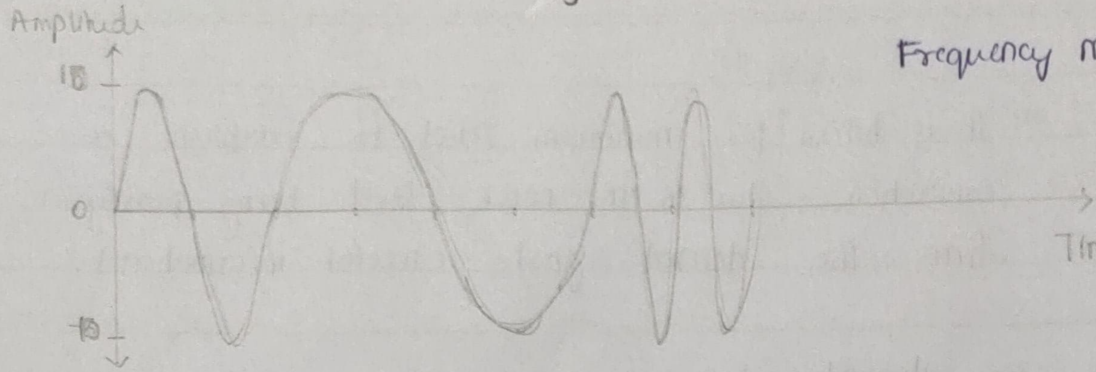
③
```
subplot (4,1,3);
plot ( t, y2);
xlabel ('time');
ylabel ('amplitude');
title (' AWGN [snr 100] FM');
```

④
```
subplot (4,1,4);
plot ( t, y3);
xlabel ('time');
ylabel ('amplitude');
title ('AWGN [snr 1000] FM');
```

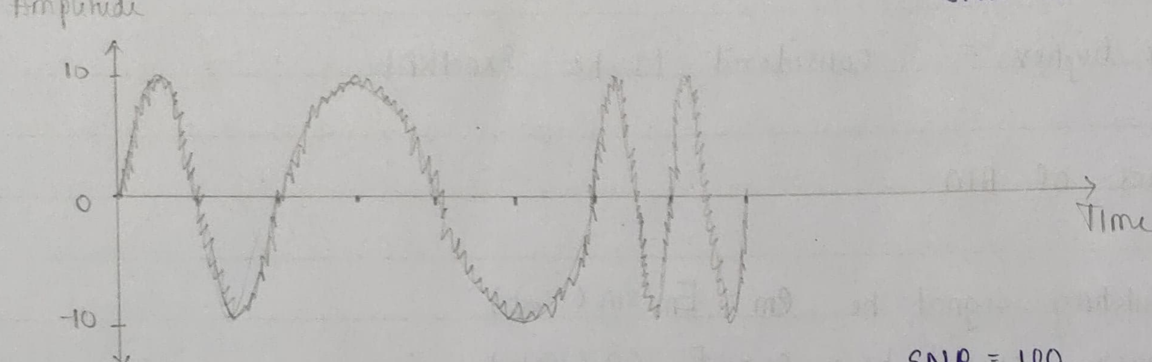
Teacher's Signature:

FM signal with different SNR

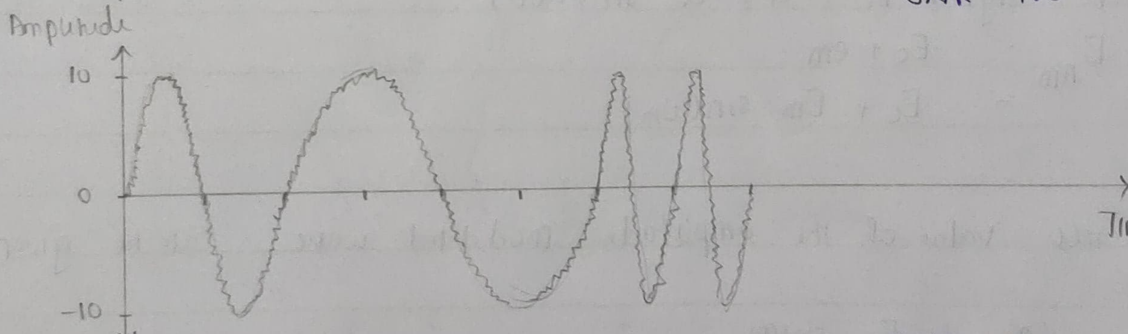
Frequency modulated Signal



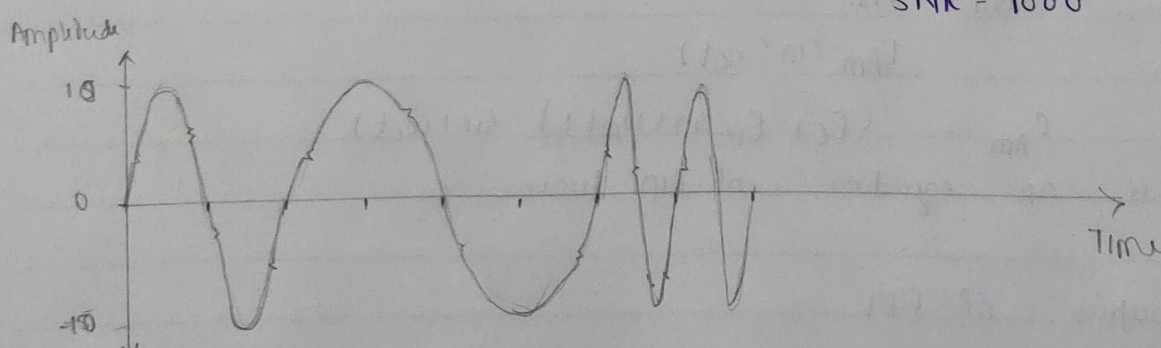
SNR = 10



SNR = 100



SNR = 1000



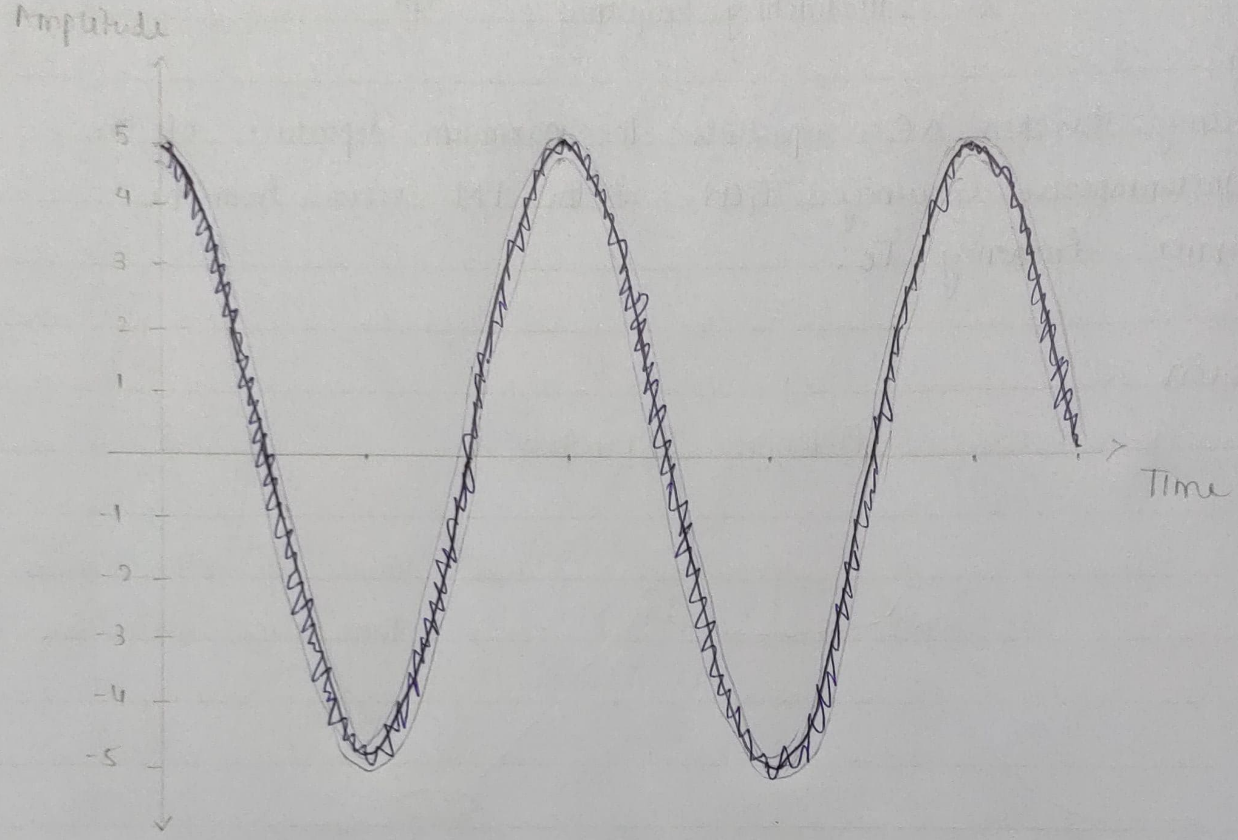
[U19CS012]

Moving Average Filter

```
clear all;    clc;
close all;
fs = 500000;
fm = 10000;
t = 1:200;
x = 5* cos( 2*pi* (fm/fs)*t );
z = awgn(x,5);
plot (x, 'g', 'linewidth', 1.5);
hold on;
plot (z);  hold on;
for i: 1:194;
    y(i) = ( z(i) + z(i+1) + z(i+2) + z(i+3) + z(i+4) + z(i+6))/6;
end
plot (y, 'r', 'linewidth', 1.5);
legend( 'Actual', 'Noisy', 'Filtered' );
title ('moving Average Filter', 'fontsize', 12);
xlabel ('--> time in 2us');
ylabel (' --> volts');
```

> CONCLUSION: We have successfully studied the effect of AWGN on transmission of Amplitude modulation (AM) and Frequency Modulation (FM).

Use of moving average filter to retrieve the signal by averaging the noise fluctuation



Moving Average Filter

— = Actual — = Noisy - Filtered

EXPERIMENT - 8 [U19CS012]

SINGLE SIDE BAND (SSB-SC) MODULATION SCHEME

AIM: Write and simulate a program for single side-band (SSB) modulation scheme. Draw the message / carrier waveforms and resultant modulated signal in time domain and frequency domain. Show input / output using matlab code in virtual mode.

APPARTUS: MATLAB

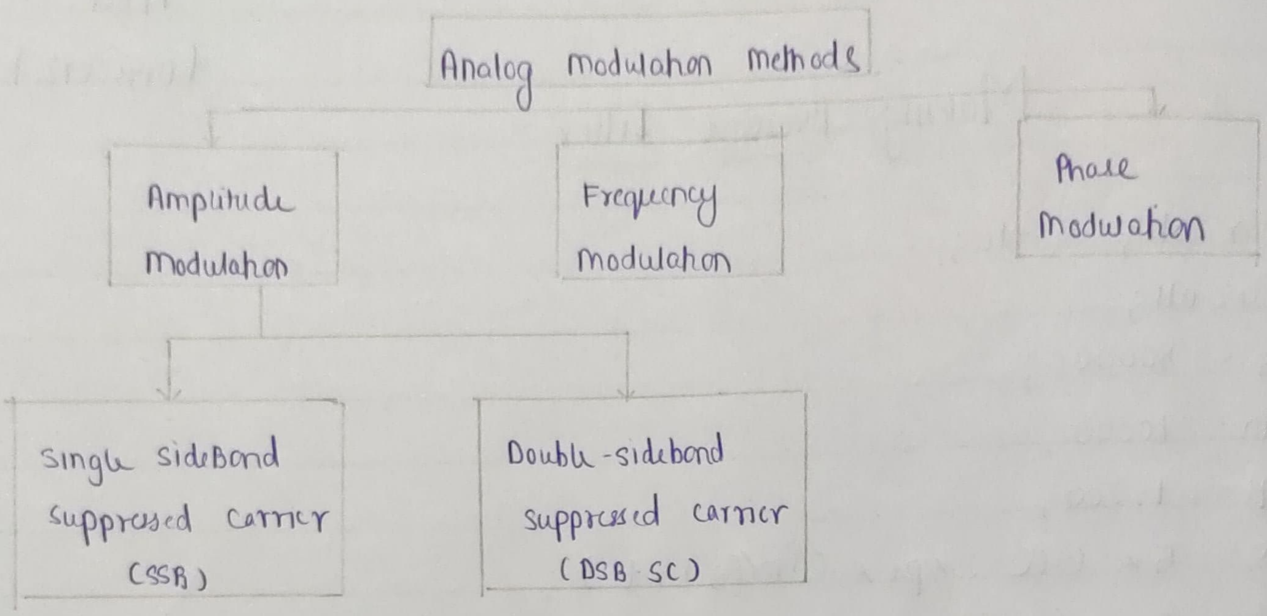
THEORY:

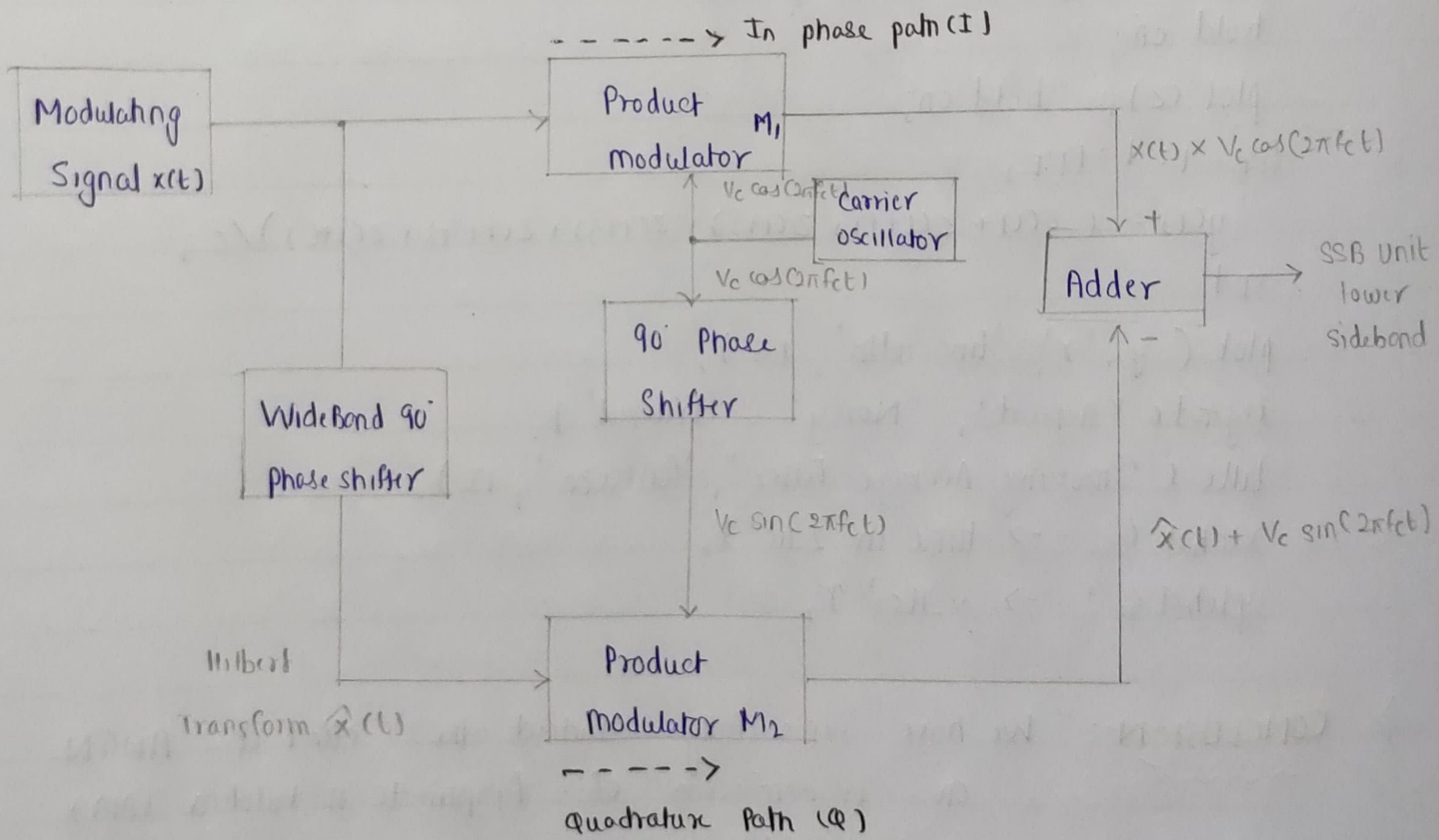
1.) Modulation: Modulation is a process by which some characteristics of a carrier wave is varied in accordance with a modulating (message) signal.

2) Analog modulation: It is a kind of modulation, where the message signal and the carrier wave both are analog in Nature.

3.) Single Side Band (SSB-SC) modulation:

- SSB-SC is a type of Amplitude Modulation
- In conventional A.M, we have two side band and the one carrier wave (no information is contained by the carrier)
- In SSB-SC modulation, only one side band is transmitted because





Generation of SSB-SC : By Hilbert Transform method both USB (upper side band) and LSB (lower side band) having the same information.

- Therefore, the transmission bandwidth is reduced to half and also required less power compare to other method of A.M.

4.) There are two methods of generation of SSB-SC :

1. Frequency discrimination method
2. Hilbert transform method or Phase discrimination method

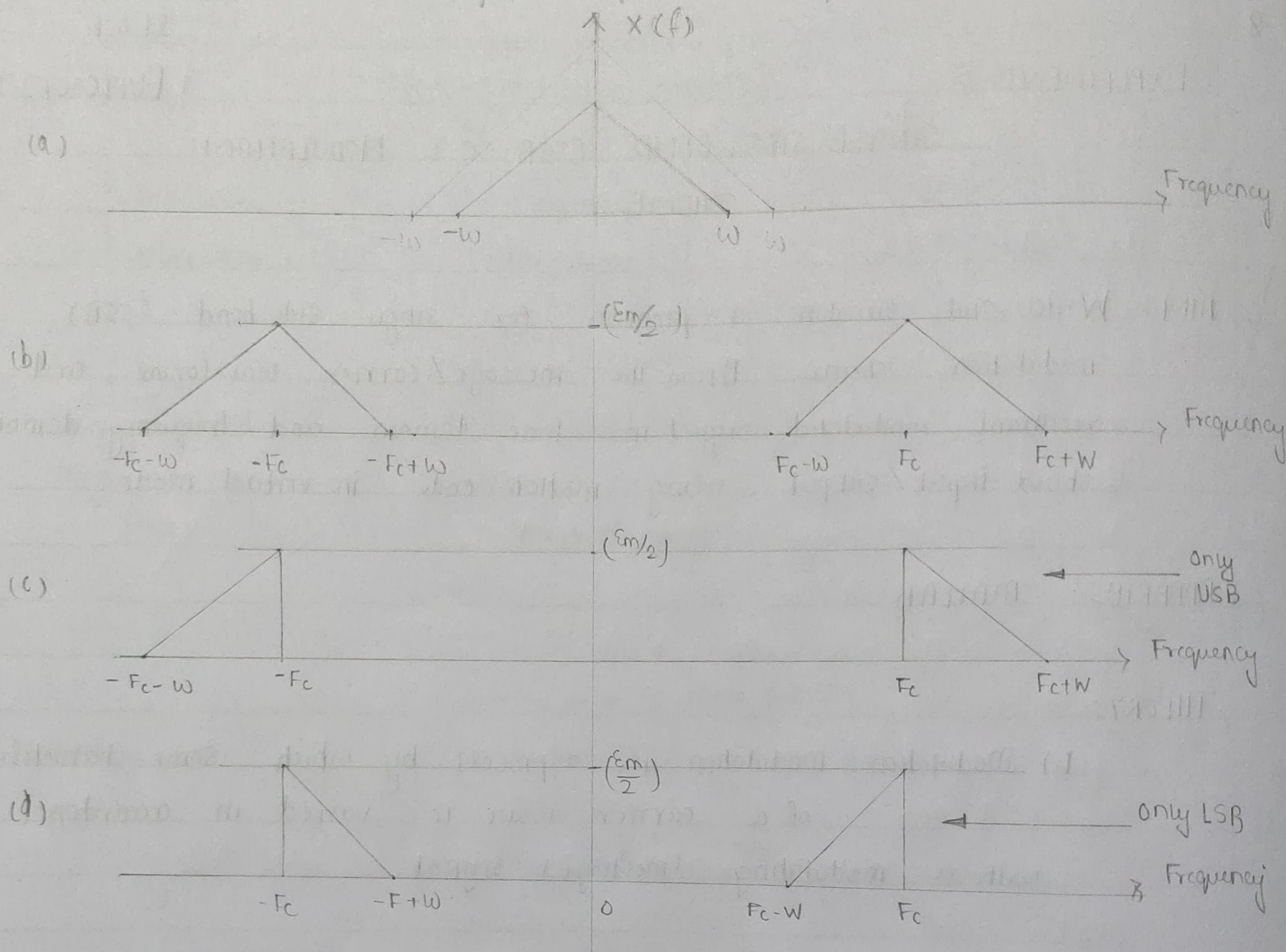
5.) Applications :

→ In point-to-point communications

→ Radar Communication

→ Where the power saving and low bandwidth requirements are important

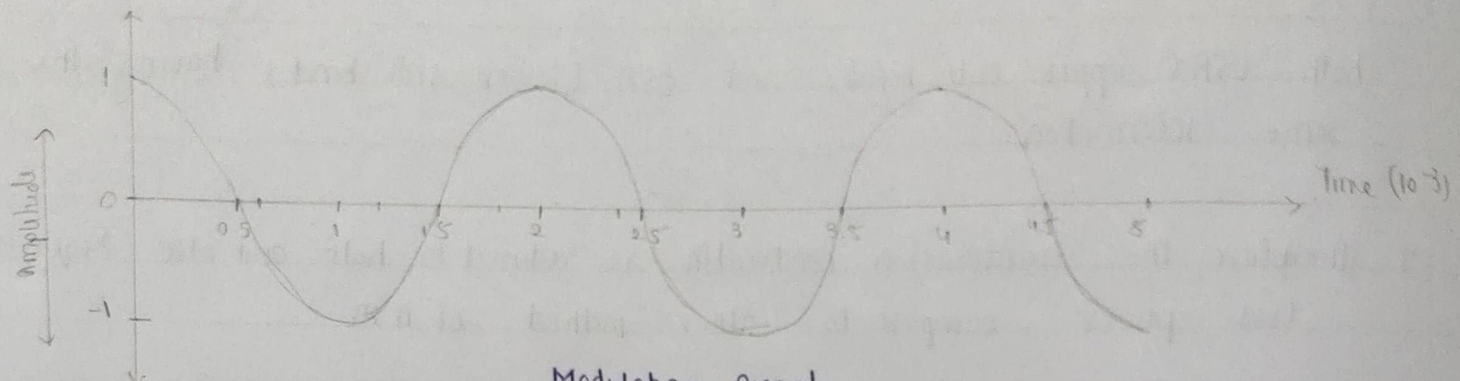
→ In many voice applications.



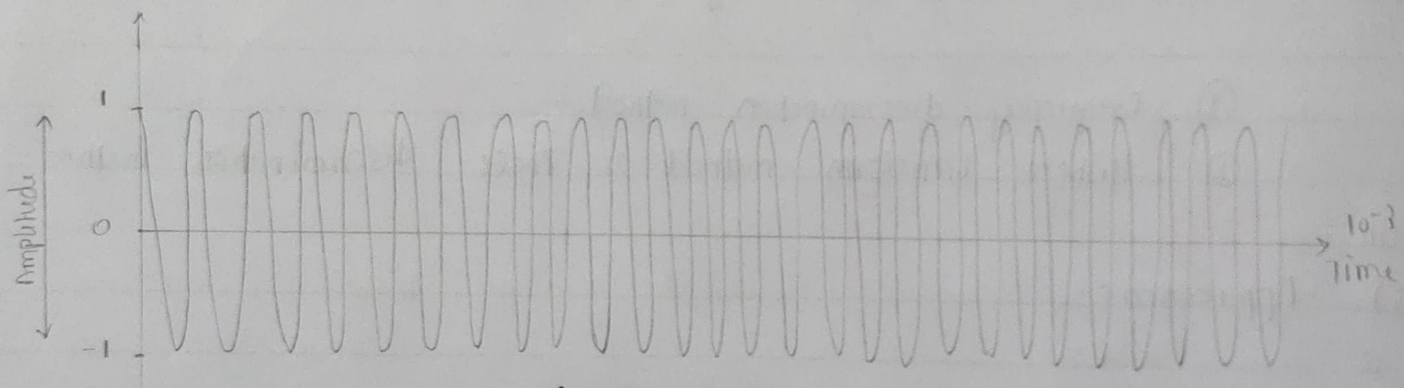
(a) Spectrum of message Signal
(b) Spectrum of DSB- SC wave
(c) spectrum of DSB- SC with only USB Transmission
(d) Spectrum of DSB-SC with only LSB Transmission

MATLAB Code :

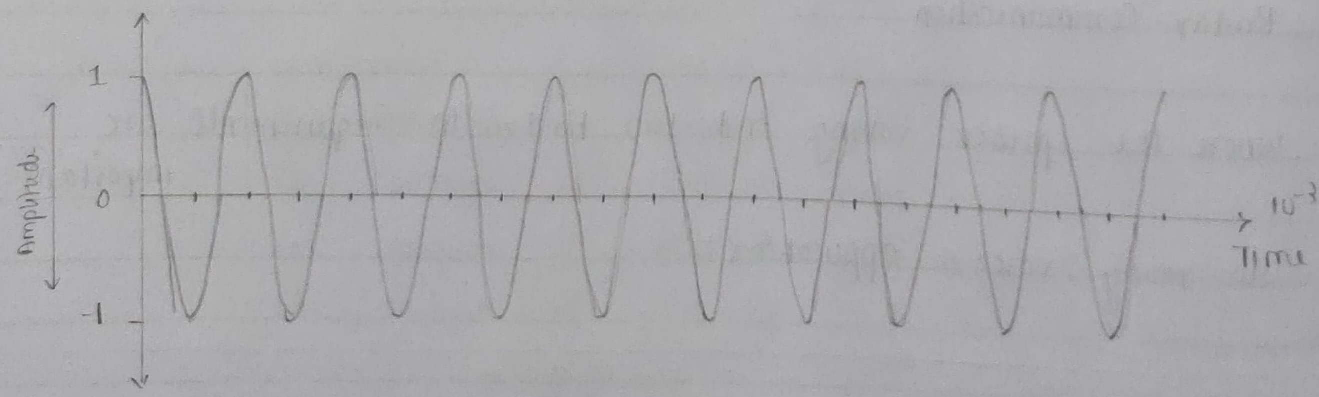
```
clc;
clear all;
close all;
am = 1 ;                          % amplitude of modulating signal
ac = 1 ;                          % amplitude of carrier signal
fm = 500;                         % modulating signal frequency
fc = 5000;                        % carrier frequency
fs = 100000;                      % sampling frequency
ts = 1/fs;                        % sampling interval
N = 10000;                        % no. of samples
t = (-N/2 : 1: (N/2-1))*ts;       % time interval
m = am * cos(2*pi* fm*t);         % modulating signal
mh = am * sin(2*pi* fm*t);        % hilbert transformation of message signal
c = ac * cos(2*pi* fc*t);         % carrier signal
ch = ac * sin(2*pi* fc*t);        % hilbert transform of carrier signal
st = m.*c - mh.*ch;               % SSB-SC signal

% time domain of all signals
subplot (3,2,1)
plot( t, m, 'red', 'linewidth', 1.5 );
axis ( [0 0.005 -2.5 2.5] );
xlabel ('time');
ylabel ('amplitude');
title ('modulating signal');
grid on;
```



Modulating signal



Carrier signal



Modulated Signal

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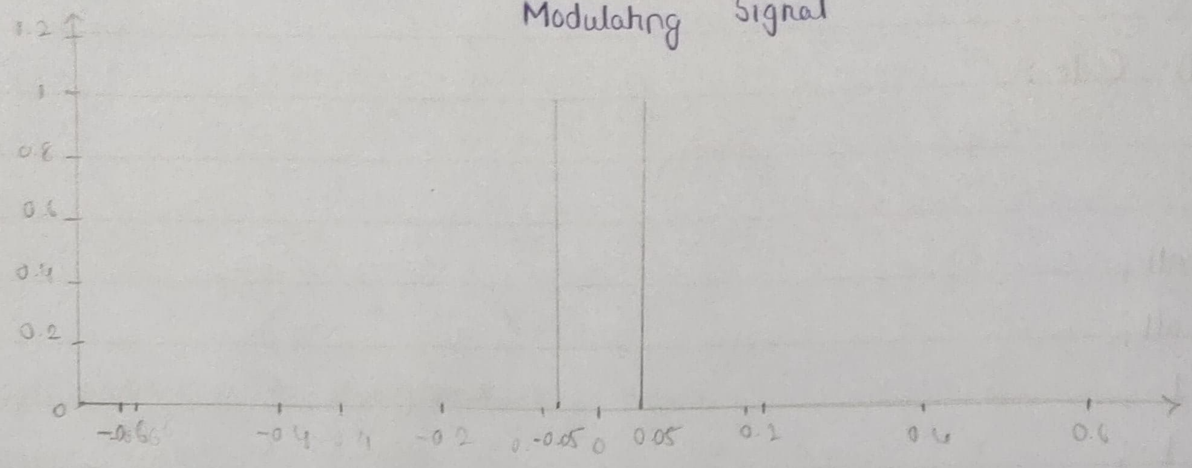
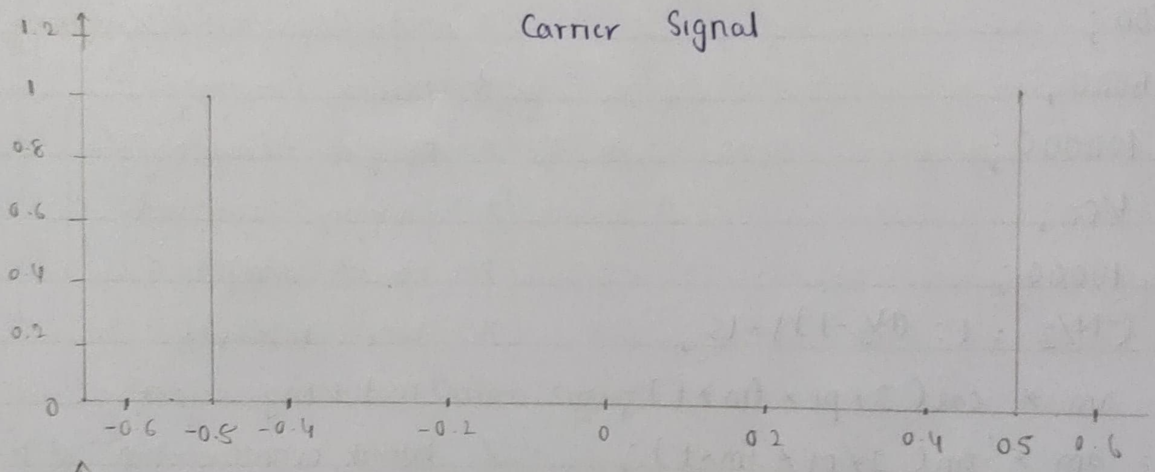
```
subplot ( 3, 2, 3)
plot ( t, c, 'black', 'line width', 1.5 );
axis ( [ 0  0.005   -2.5   2.5] ) ;
xlabel ('time') ;
ylabel ('amplitude' ) ;
title ( 'carrier signal' );
grid on ;

subplot (3, 2, 5)
plot ( t, st, 'blue', 'Linewidth', 1.5 );
axis ( [ 0 0.005 -2.5 2.5] )
xlabel ('time') ;
ylabel ('amplitude') ;
title ( 'modulated signal' ) ;
grid on ;

% spectrum of all signals
f = ( - N/2 ; 1: (N/2 - 1)) * fs / N;
M = abs ( (2/N) * fftshift ( fft(m)) ) ;
C = abs ( (2/N) * fftshift ( fft (c)) ) ;
SF = abs ( (2/N) * fftshift ( fft (st)) ) ;
subplot (3, 2, 2) ;
plot (f, M/max(M), 'red', 'Linewidth', 1.5 );
axis ( [ -2*fc   2*fc   -0.1   1.1] ) ;
xlabel ('Frequency') ;
ylabel ('amplitude') ;
title ('modulating signal') ;
grid on;

subplot ( 3, 2, 4) ;
plot( f, C/max(C), 'black', 'linewidth', 1.5) ;
axis ( [  -2*fc   2*fc   -0.1   1.1] ;
xlabel ( 'frequency') ;
ylabel ( ~~'carrier~~ 'amplitude' ) ;
title ( 'carrier signal' ) ;     grid on ;
```

Teacher's Signature:

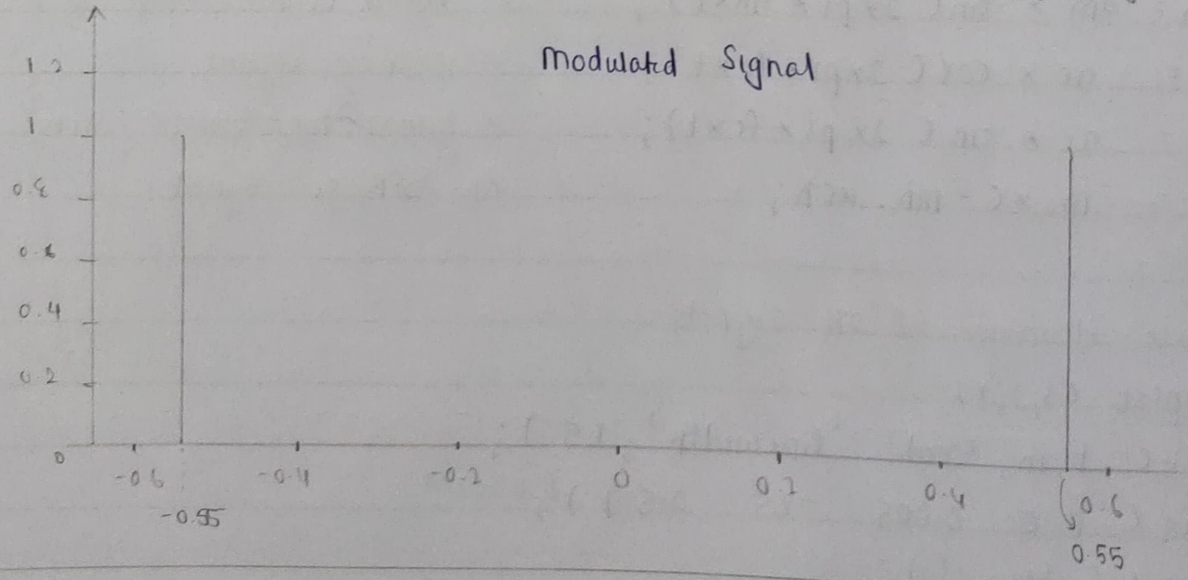
Modulating Signal



Carrier Signal



Modulated Signal



[U19CS012]

```
subplot (3,2,6);
plot ( f, SF/max (SF) , 'blue', 'linewidth' , 1.5);
axis ( [ -2*fc   2*fc   -0.1  1.1] );
xlabel ( 'frequency' );
ylabel ( 'amplitude' );
title ( 'modulating signal' );
grid on;
```

CONCLUSION : We successfully observed single side Band (SSB-SC) modulation scheme and as we come to know that here low Bandwidth is required for transmission. Hence, we also save power.

—×—

EXPERIMENT 9 [U19CS012]

QAM MODULATION AND DE MODULATION

> AIM: Study of 16 QAM Modulation and Demodulation technique with consetellation ~~dig~~ diagram and waveforms.

> APPARATUS: MATLAB

> THEORY: ① QAM: Quadrature Amplitude Modulation or QAM is a form of modulation which is widely used for modulating data signals onto a carrier used for radio communications.

- QAM is a signal in which two carriers shifted in phase by 90° are modulated & the resultant output consists of both Amplitude and phase variations.

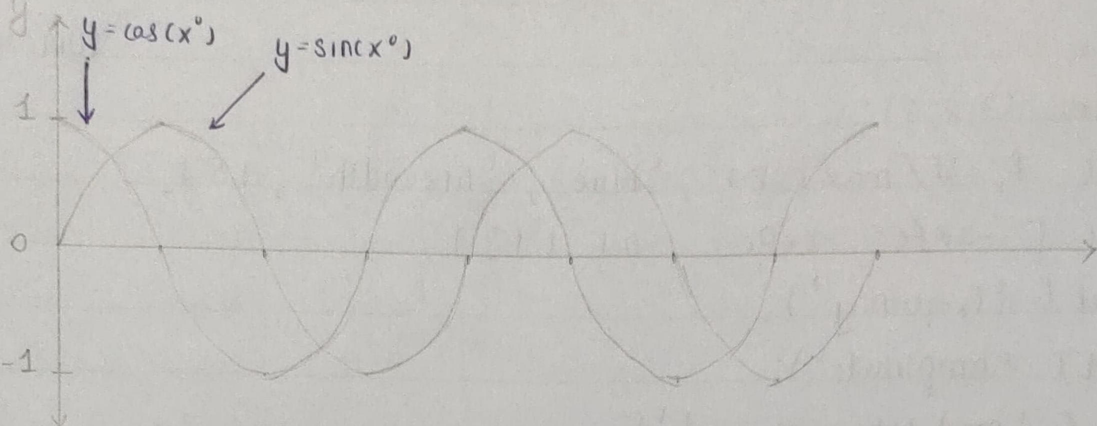
- Hence it may also be considered as a mixture of amplitude and phase modulation. QAM is both an analog and digital modulation technique.

② Main parameters to be considered while designing any communication system
- Transmission Power
- Transmission Bandwidth

- Although the SSB-SC systems are most power and bandwidth efficient but still their performance lags in the noisy environment.

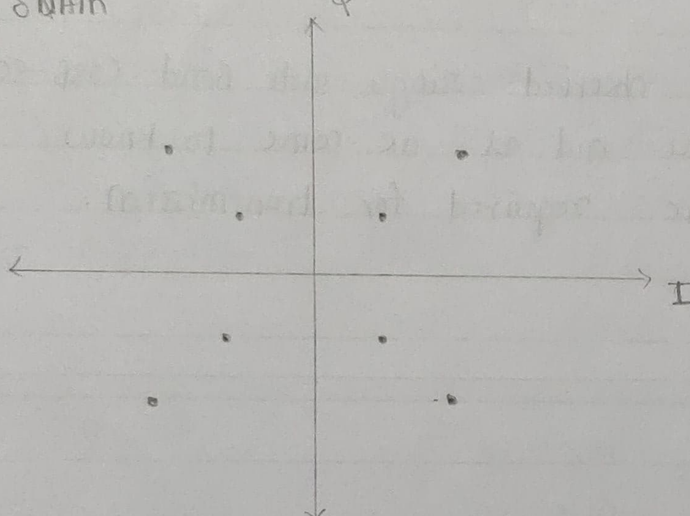
Teacher's Signature:

Quadrature = Sine wave + Cosine wave

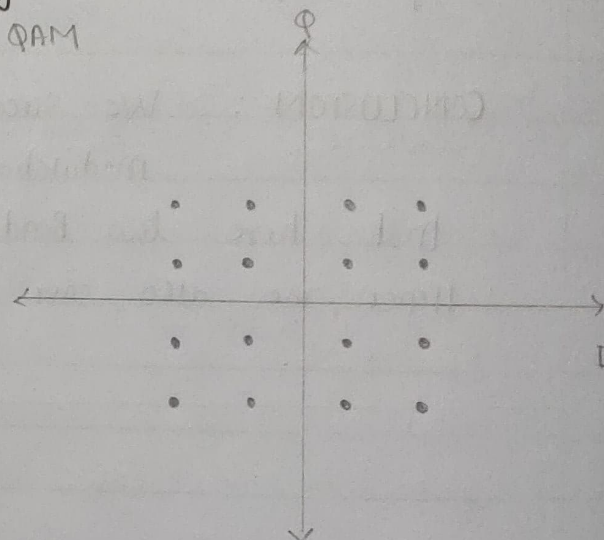


Types of QAM- constellation Diagram

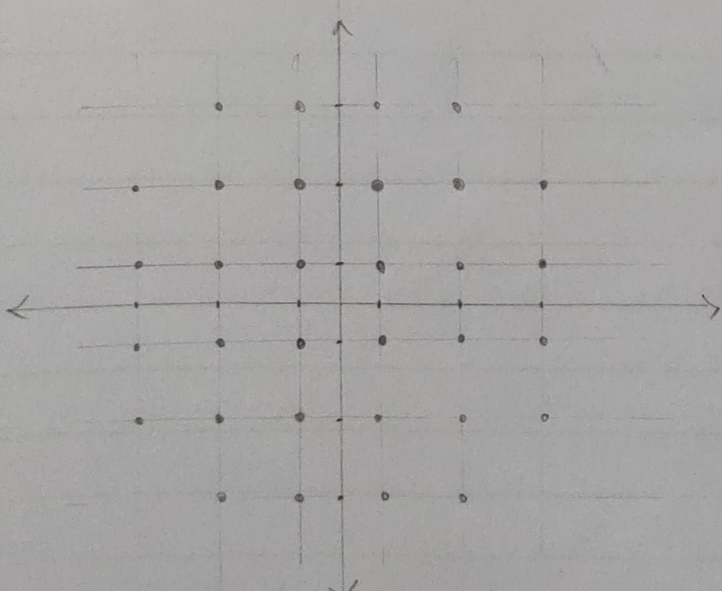
8 QAM



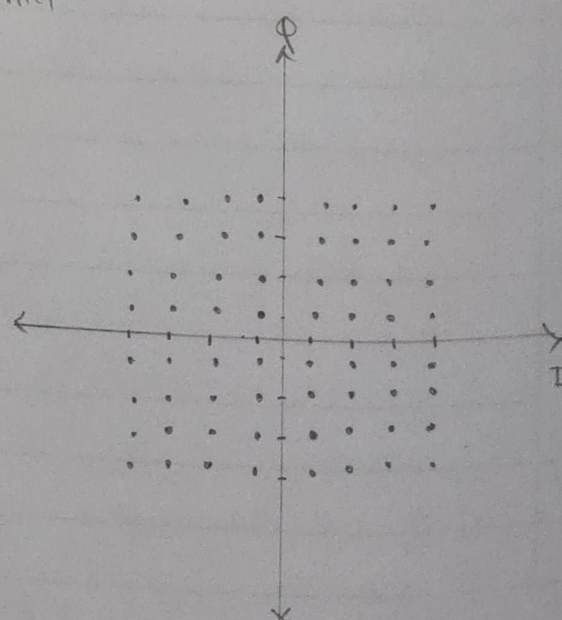
16 QAM



32 QAM



64 QAM



3.) Why QAM?

- The main aim is to save bandwidth. Two modulated signal occupies the same transmission channel.
- A motivation for the use of QAM comes from the fact that a straight amplitude modulated signal occupies twice the bandwidth of the modulating signal.

- This is very wasteful of the available frequency spectrum

- QAM places two independent double sideband suppressed carrier signals in the same spectrum.

4.) Type of QAM

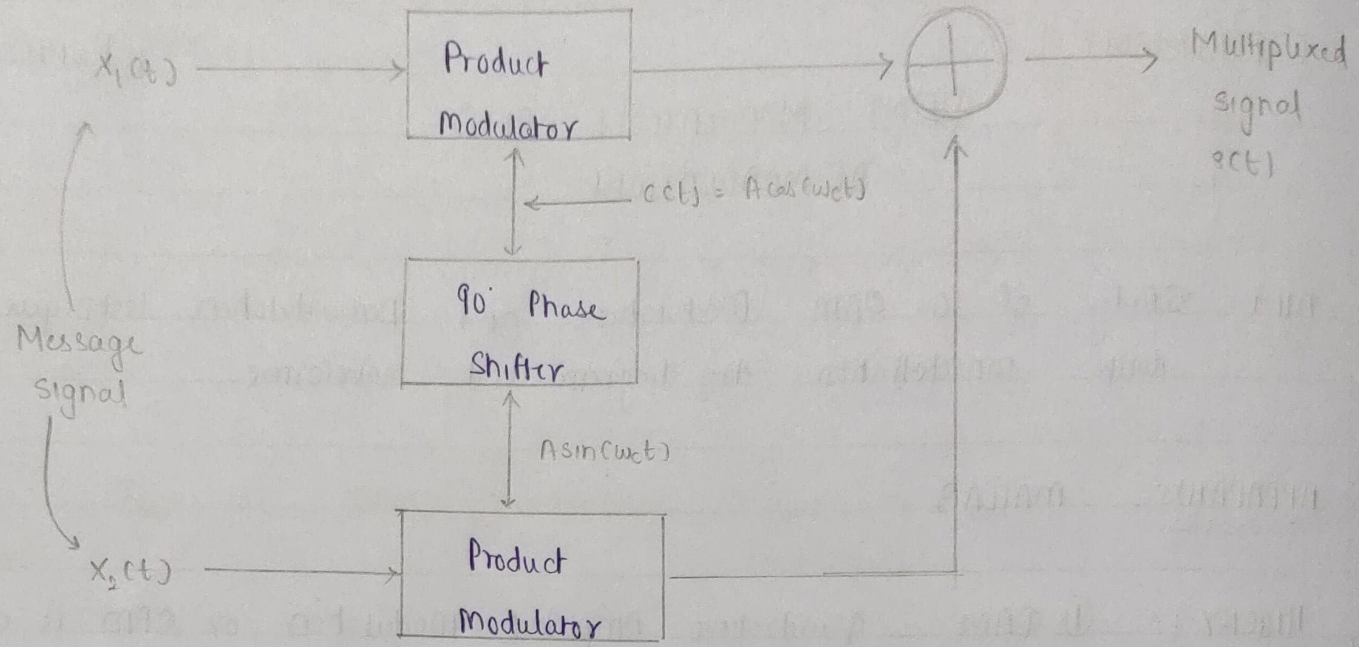
- A variety of forms of QAM are available which include:

- 16 QAM
- 32 QAM
- 64 QAM
- 128 QAM
- 256 QAM

5.) QAM Modulation

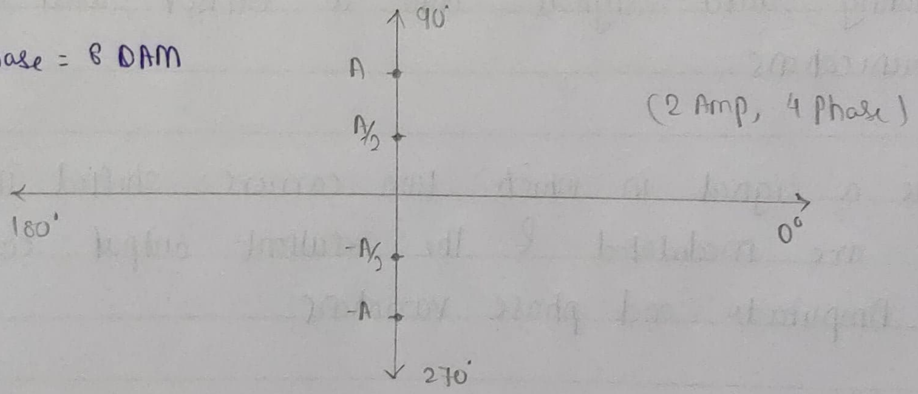
- QAM theory states that both Amplitude and phase changes within a QAM signal.
- The basic way in which a QAM signal can be generated is to generate two signals that are 90° out of phase with each other and them sum them.

QAM Modulation

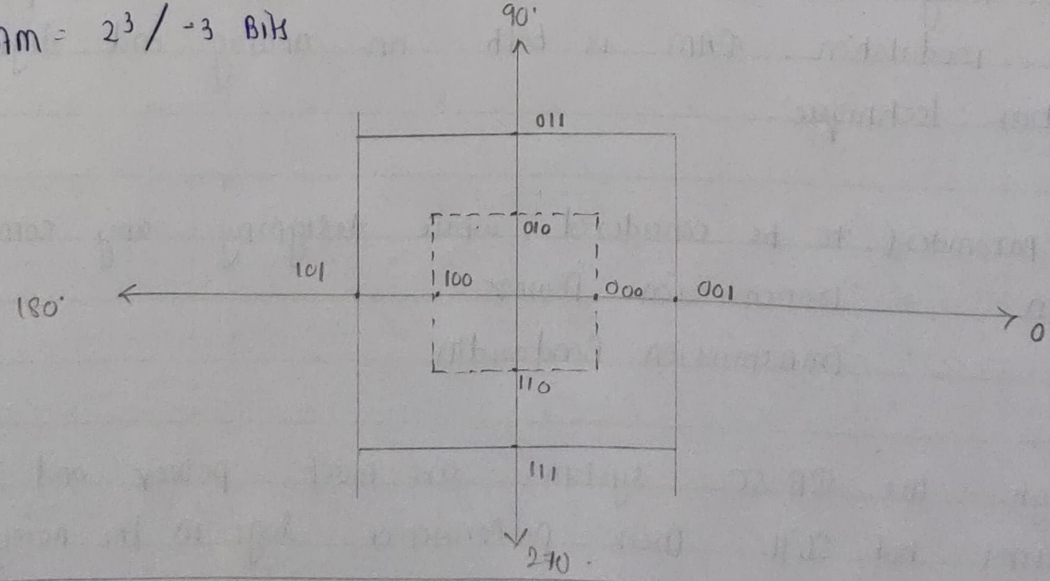


$$s(t) = x_1(t)\, A\cos(w_c t) + x_2(t)\, A\sin(w_c t)$$

2 Amp × 4 phase = 8 QAM



8 QAM = 2^3 / -3 Bits



- The I and Q signals can be represented by the equation below:

$$I = A\cos(\phi)$$

$$Q = A\sin(\phi)$$

- This signals will not overlap with each other because they are orthogonal.

- It is possible to transmit two DSB-SC signal with in bandwidth of "2fm"

- It provides bandwidth effeciency.

- Gives better performance than SSB and also improves data rate.

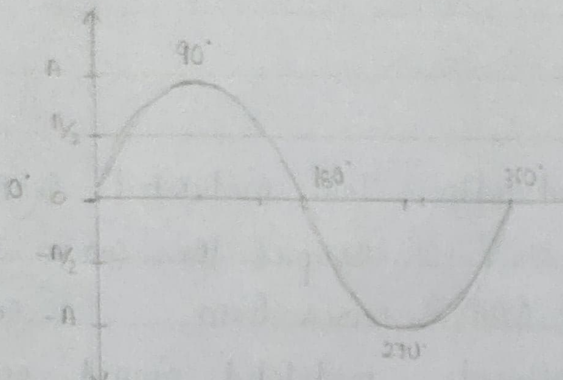
6.) QAM Demodulation

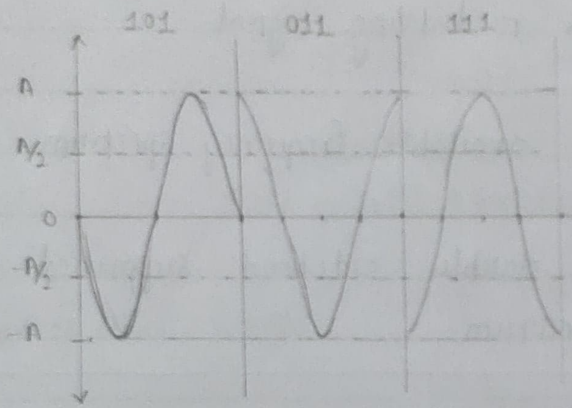
- The QAM demodulator is very much the reverse of the QAM modulator
- The signals enter the system, they are split and each side is applied to a mixer.

7.) Bit error Rate (Received Bits)

- While higher order modulation rates are able to offer much faster data rates and higher levels of spectral effeciency for the radio communication system, this comes at a price.
- The higher order modulation schemes are considerable less relient to noise and interference.

Phasor Diagram

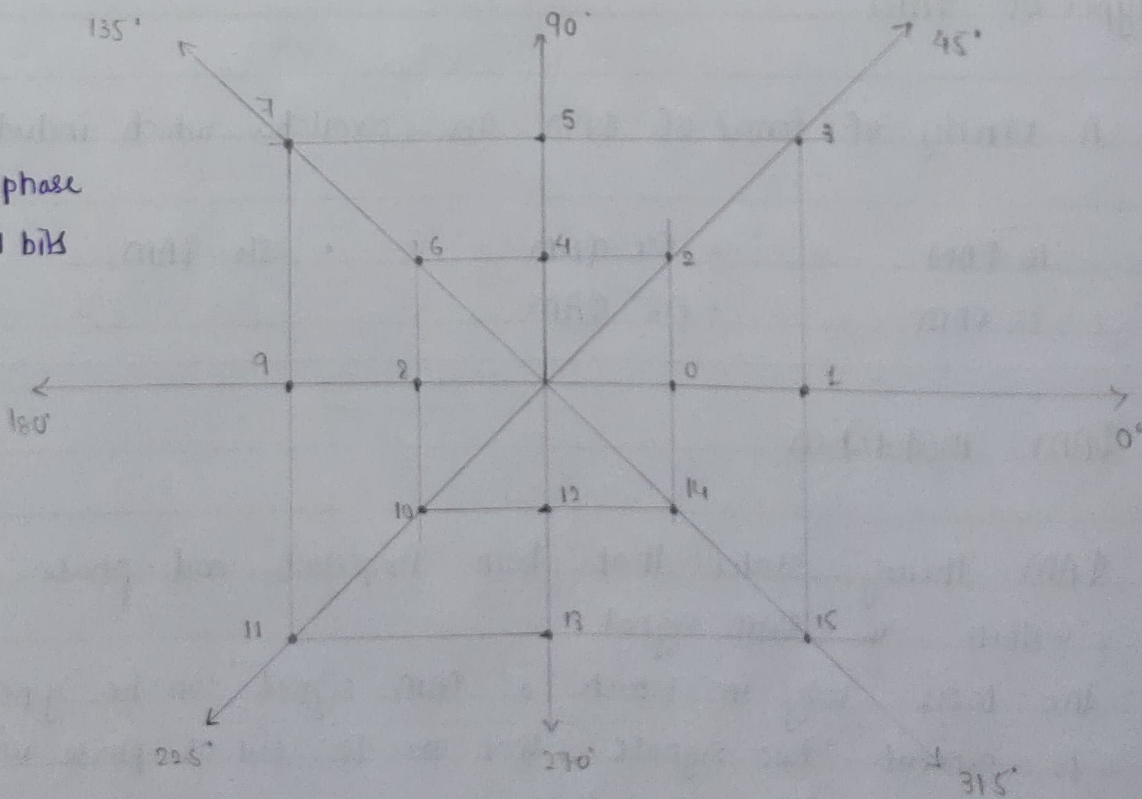




16 QAM

2 Amp * 8 phase

$= 2^4$ / 4 bits



- Many radio communications systems now use dynamic adaptive modulation techniques. They sense the channel conditions and adapt the modulation scheme to obtain the highest data rate for the given conditions.

- M-QAM technique provides better bit error rate performance than M-PSK modulation techniques.

8.) Advantages

- The advantage of using QAM is that it is a higher order form of modulation ; As a result, it is able to carry more bits of information per symbol

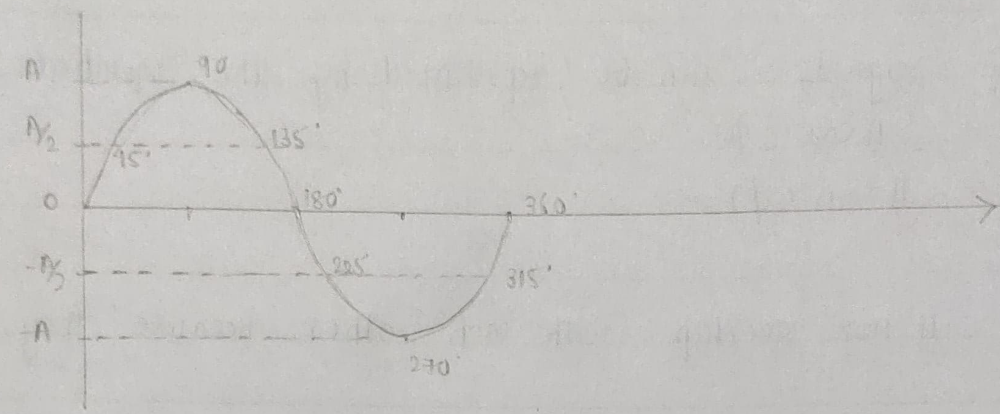
- By selecting a higher order format of QAM, the data rate of a link can be increased.

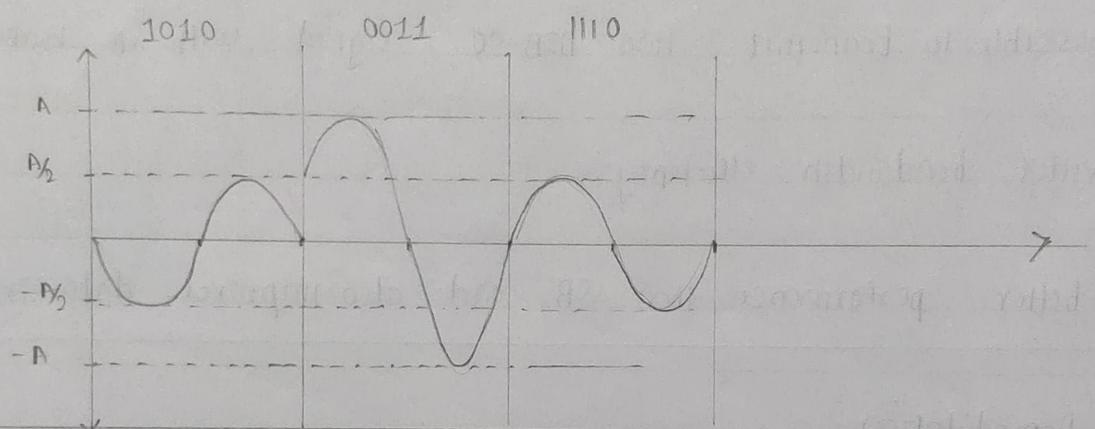
- Bit rate ~~of~~ is increased without increasing the Bandwidth.

9.) Applications

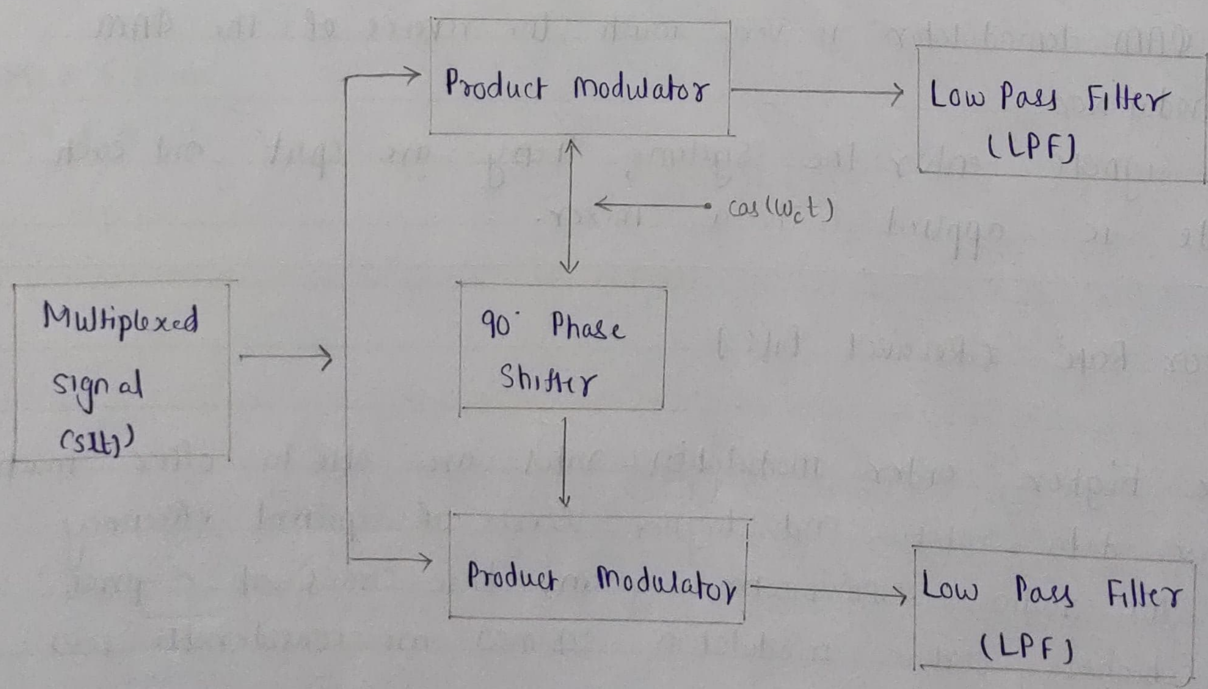
- Quadrature multiplexing is used in color television to multiplex the so-called chrominance signals which carry the information about colors.
- QAM scheme is used on telephone lines for data transmission
- Ultra high capacity Microwave Backhaul systems also use 1024-QAM.

Phasor Diagram





QAM Demodulation



EXPT. NO. NAME: Page No.: 9 Date:

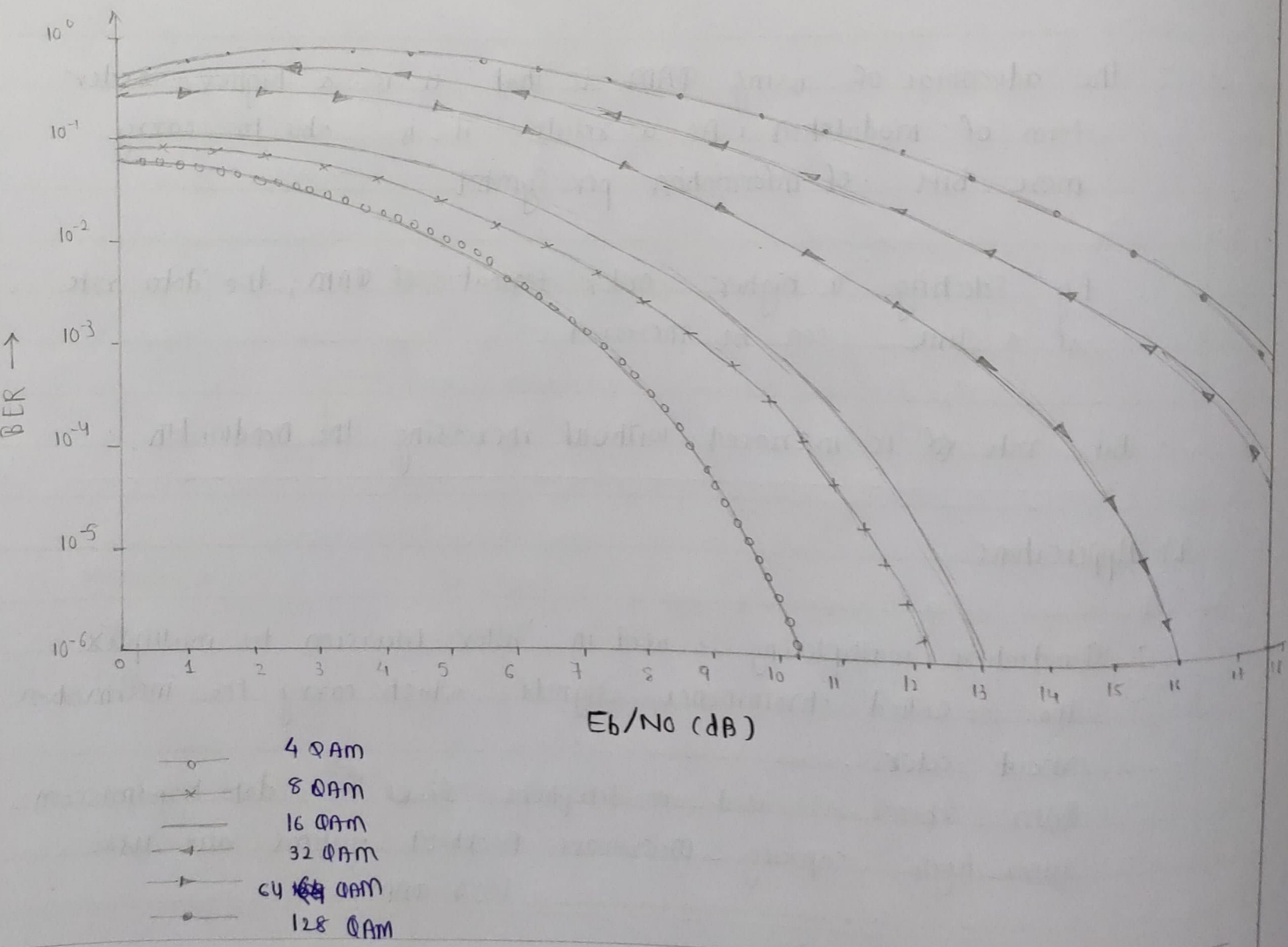
> MATLAB Code:

```
clc;
clear all;
close all;
M = 16 ;
x =  (0 : M-1);
y= qammod (x, M);
scatterplot (y);
% Z = qamdemod (y, M, pi/4);
%  scatterplot (z);
ber_1 = [] ;                            ((3 * log2(M) * EbNo ) / 2*(M-1))
for  EbNOdB  =   0 : 20;
        EbNO =   10 ^ ( EbNOdB/ 10);
        ber  =  (1/ log2 (M)) * (2*(1- sqrt (1/m)) * erfc ( sqrt (^) ) )
        ber_1 = [ ber_1  ber];
end
EbNO dB =  0 : 20;
figure
semilogy  (EbNodB, ber_1 (1,:) , 'ro-');
xlabel ( 'Eb/ N0 (dB)');
ylabel ( 'BER' );
title ( 'BER  of 16-QAM' );
axis ( [0  16  10^-6  10^0] );
grid on;
```

Teacher's Signature:

Modulation	Bit Per Symbol	Symbol Rate
BPSK	1	1 Bit Rate
QPSK	2	1/2 Bit Rate
8PSK	3	1/3 Bit Rate
16 PSK	4	1/4 bit Rate
32 PSK	5	1/5 bit Rate
64 PSK	6	1/6 bit Rate

"BER Values" using MATLAB



[U19CS012]

```
% BER comparision of various M-QAM
clc; clear; close all;
M = [ 4 8 16 64 128 256 ]
for i=1: length (M)
        ber_th1 = [];
for EbNodB = 0:20
        EbNO = 10.^ ( EbNodB / 10);
        ber = ( 1/ log2 (M(i))) * (2* (1-sqrt (1/m(i)))) *
                erfc (sqrt ( (3* log2 (M(i)) * EbNO) / 2*( m(i) -1))) )
        ber_thi = [ ber_th1 ber];
end
    berth = [ ber_th; ber_thi];
end
EbNodB = 0:20;
semilogy ( EbNodB, ber_th (1, :), 'ro-'); hold on
semilogy ( EbNodB, ber_th (2, :), 'g+-' ); hold on
semilogy ( EbNodB, ber_th (3, :), 'y.-' ); hold on
semilogy ( EbNodB, ber_th (4, :), 'k>-'); hold on
semilogy ( EbNodB, ber_th (5, :), 'c<-'); hold on
semilogy ( EbNodB, ber_th (6, :), 'mx-'); hold on

xlabel ('Eb/NO(dB)'); ylabel ('BER'); axis ([0 20 10^-6 10^0 ]);
```

>> CONCLUSION: We successfully examined the 16- Quadrature Amplitude Modulation (16-QAM) and Demodulation scheme.

We also evaluated 16 BER values for different QAM using MATLAB.

-x-

Teacher's Signature:

EXPERIMENT 10

[U19CS012]

PULSE CODE MODULATION AND DEMODULATION

AIM: To demonstrate the pulse code modulation (PCM) and demodulation technique. To show the sampled, quantized / encoded and decoded time domain signal for different bit-codes.

SOFTWARE: MATLAB

THEORY: 1.> Pulse Code Modulation (PCM)

- PCM is a technique, which is used to convert an analog signal into digital signal.

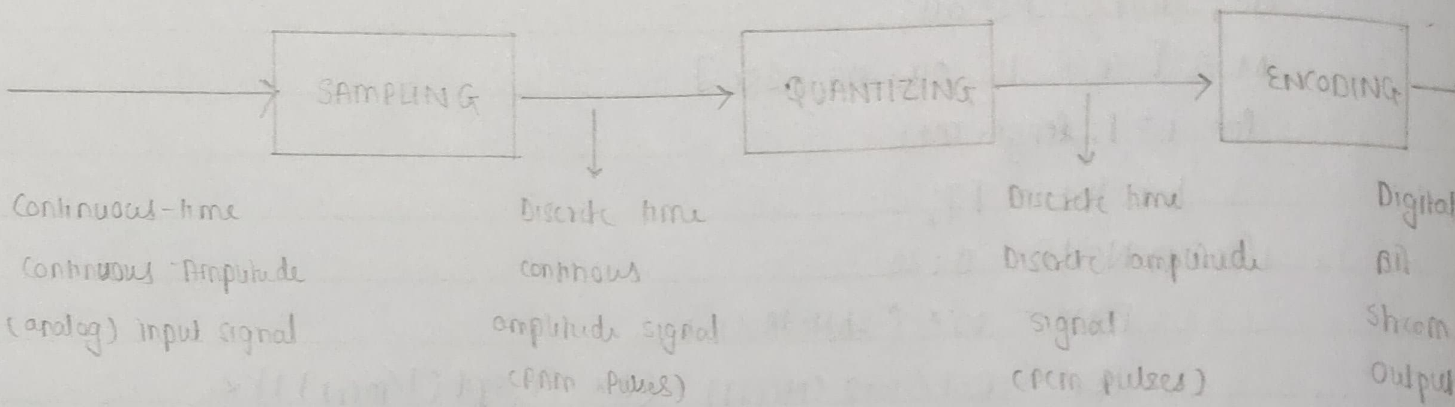
- PCM is a preferred method of communication within public switched telephone network (PSTN)

- A PCM stream is determined by two following steps:

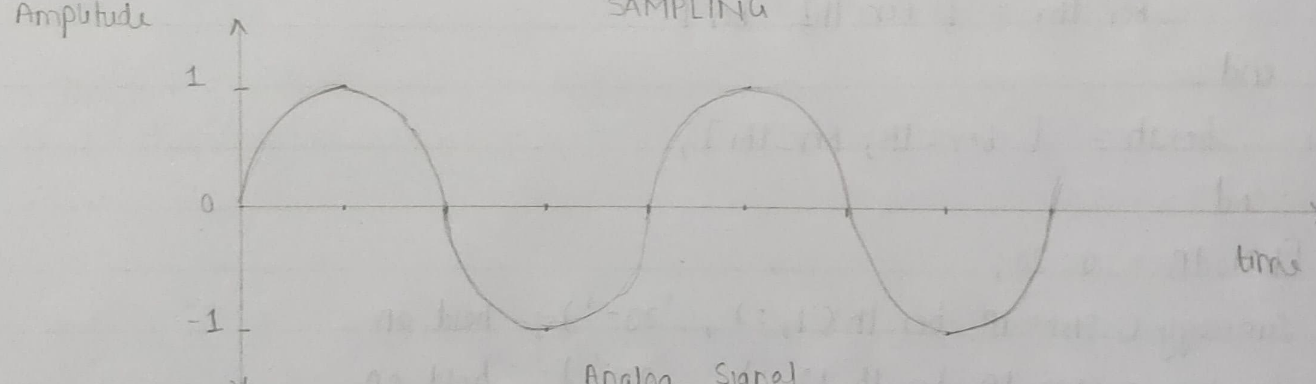
  a) Sampling Rate: which the number of times per second that samples are taken.

  b) Bit depth: which determines the number of possible digital values that can be used to represent each sample.

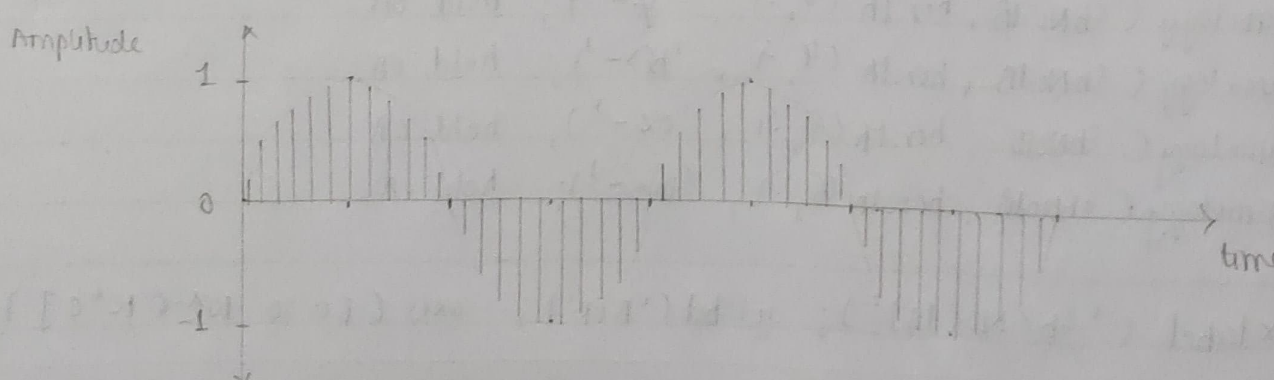
- Hence, the output of PCM resembles a binary sequence.



SAMPLING



Analog Signal



Discrete Time signal

2.) Reasons for Digital Transmission

- Less susceptible to interference cause by noise due to discrete level.

- Easy to detect errors due to discrete levels

- Easy to Encrypt (Higher Security)

- Simpler to store digital data

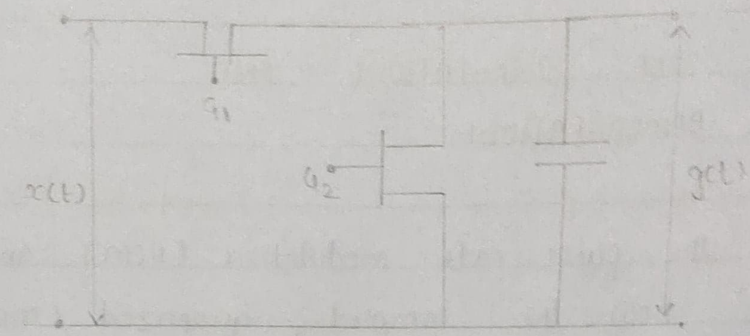
3.) Sampling

- Sampler extract samples of a continuous signal.
- Sampler produces samples that are equivalent to the instantaneous value of the continuous signal at the specified various points.
- The sampling process generates Flat-top Pulse Amplitude Modulated (PAM) signal.

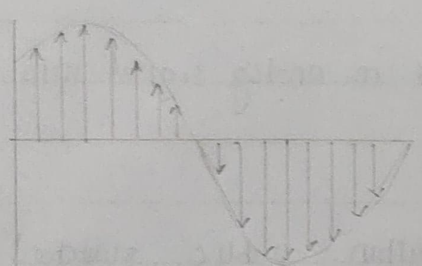
4.) Quantization

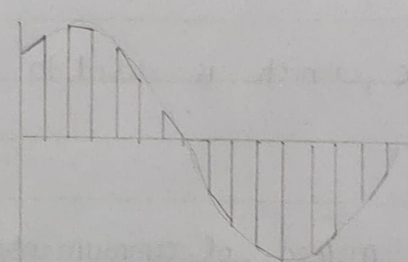
- Quantization is done by dividing the range of possible values of the analog samples into some different levels and assigning the center value of each level to any sample in the quantization interval.
- Quantization approximates the analog signal values with the nearest quantization values.

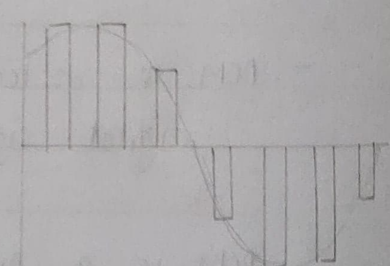
Flat-top PAM



Instantaneous Sampling	Natural Sampling	Flat-top Sampling
It is not practical method	This method is used practically	This is also used practically.
Sample rate = infinity	Sample Rate satisfied Nyquist	Sample rate satisfied Nyquist

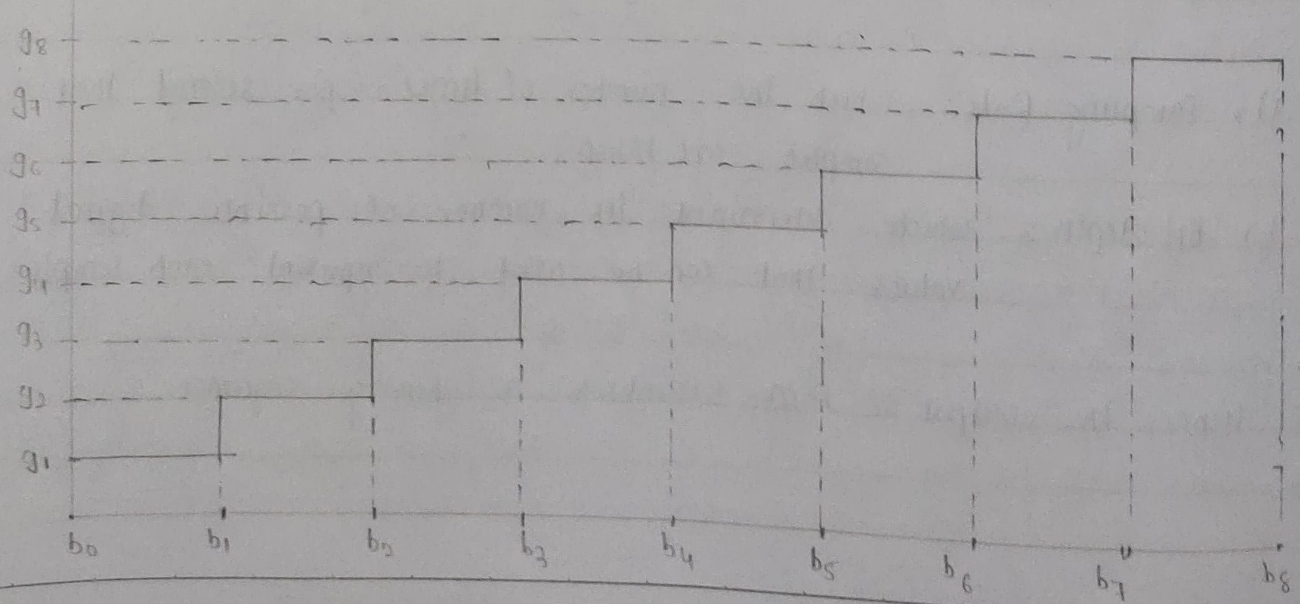






Uniformly Quantized Signal

A/D output = n bits per sample (quantization level $M = 2^m$)



EXPT. NO. | NAME: | Page No.: 5 | Date:

5.) Pulse Code Modulation (PCM) is a method of converting an Analog signal into a digital signal. (A/D conversion)

- PCM produces a series of number or digits instead of pulse train.
- Each one of these digits, in binary code, represents the approximate amplitude of the signal sample at that instant.

6.) Concluding Remark for PCM

- In PCM Transmitter, the signal x(t) is first passed through the low pass filter of cut-off frequency Fm Hz.

- This low pass filter blocks all the Frequency components above Fm Hz. This means that now the signal x(t) is band-limited to Fm Hz.

- The Sample and Hold circuit then samples this signal at the rate of Fs.

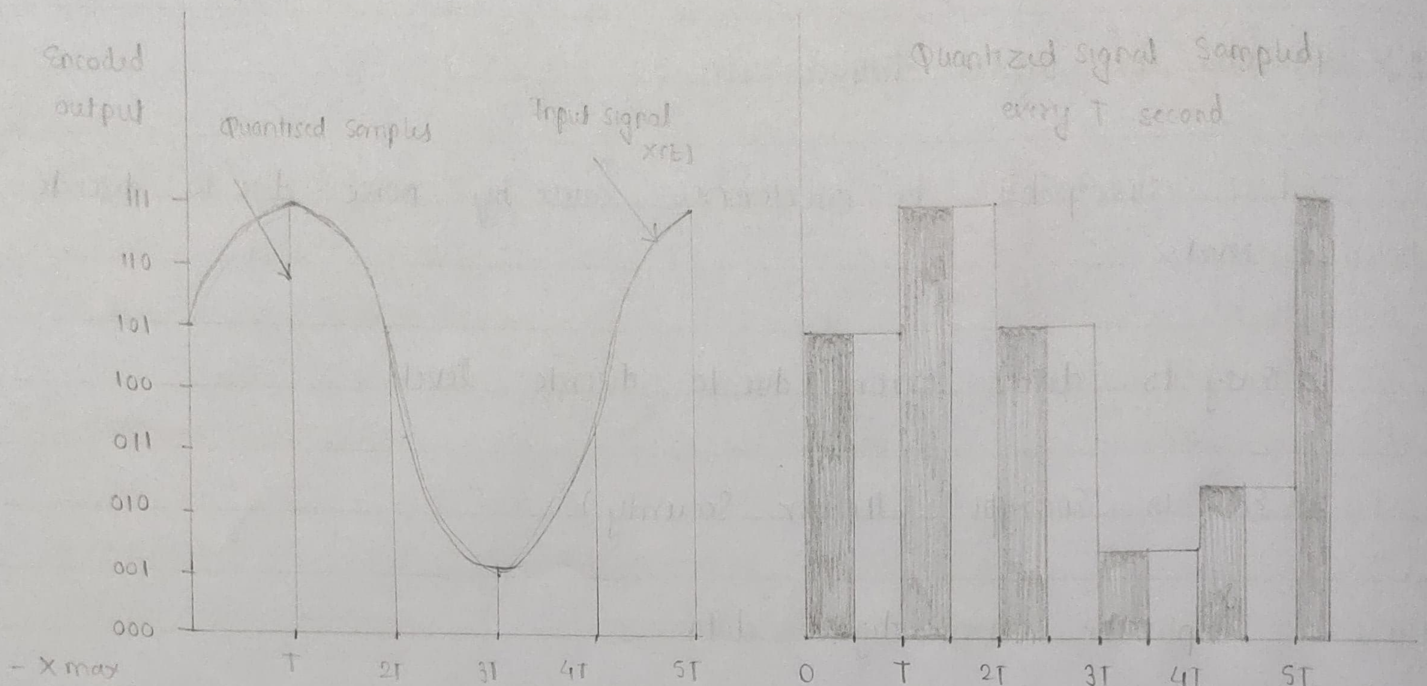
- Sampling Frequency Fs is selected sufficiently above Nyquist rate to avoid Aliasing.

- The Output from sample and hold circuit is denoted by x(nTs)

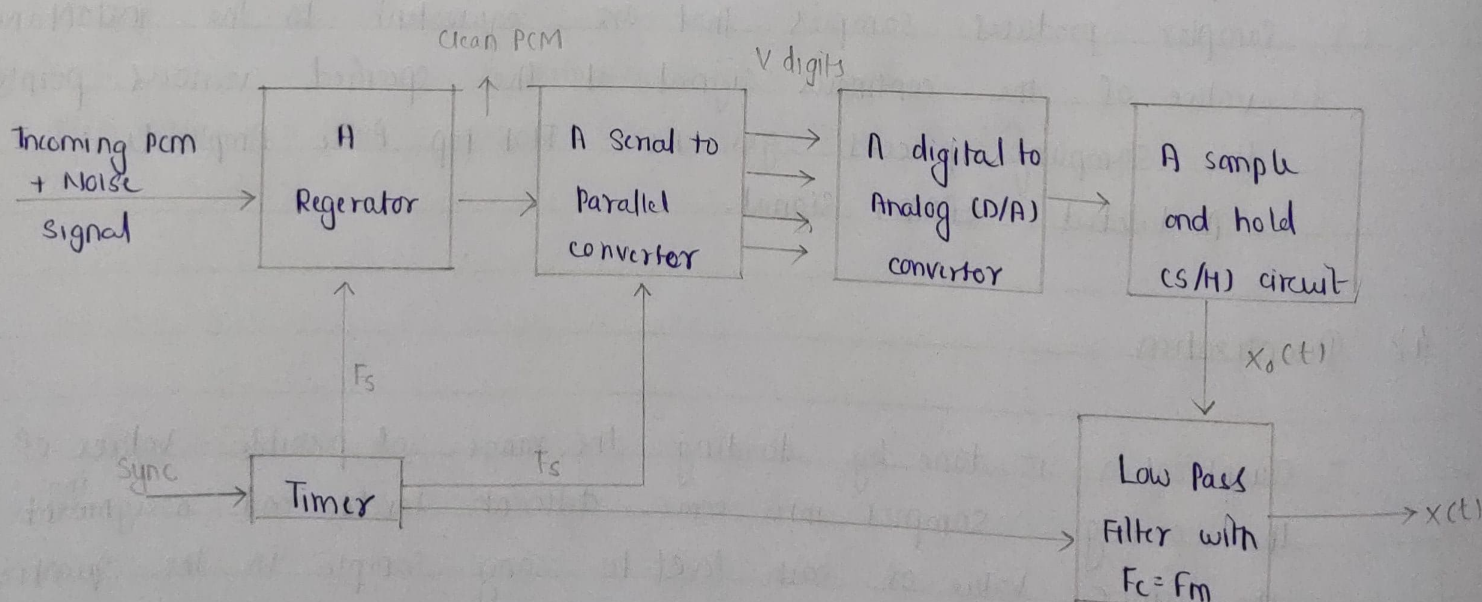
Teacher's Signature:

Transmitter

Figure : 1 Quantization of a sampled Analog signal



PCM Receiver



- This signal $x(nTs)$ is discrete in time and continuous in Amplitude
- A q-level quantizer compares input $x(nTs)$ with its fixed digital levels.
- Quantized signal is then encoded in PCM output using encoder.

7.) PCM Standards

- There are two standards of PCM ① The European standard
  ② The American Standard
- They differ slightly in the detail of their working but the principles are the same.

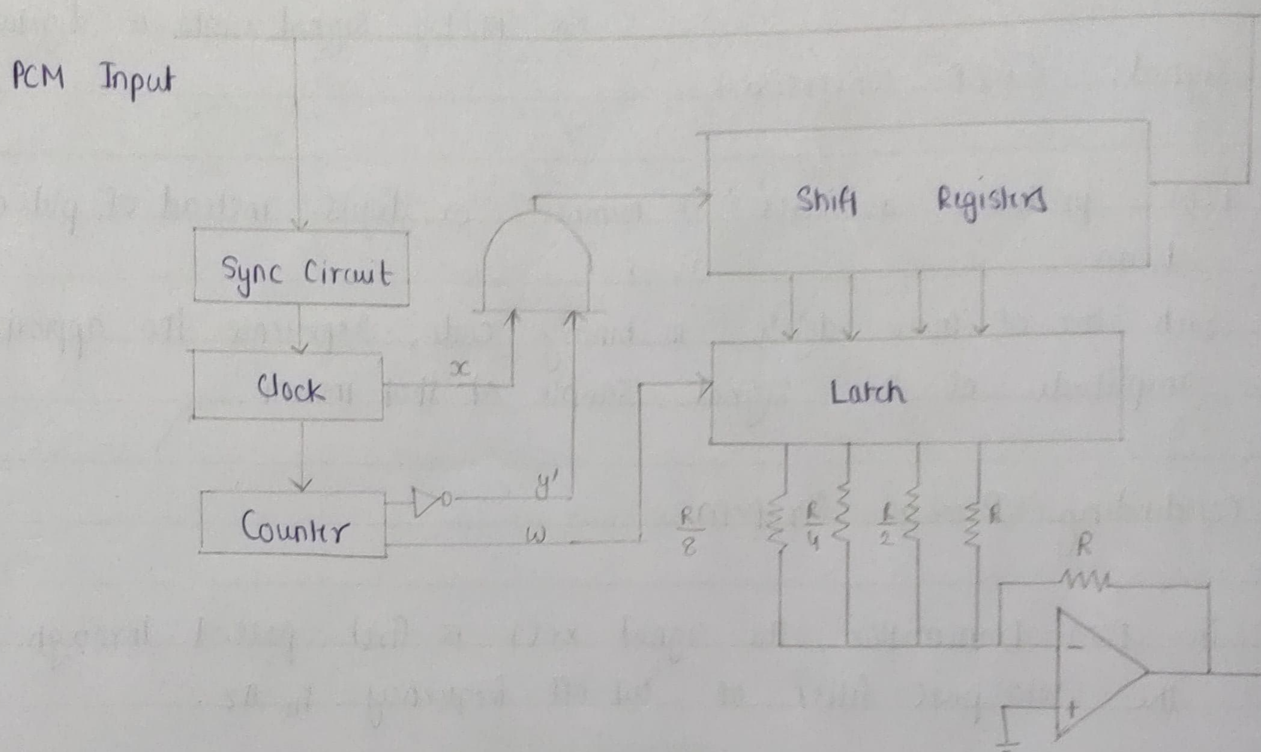
- European PCM = 30 channels
- North American PCM = 24 channels
- Japanese PCM = 24 channels

In India, we follow the European PCM of 30 channels system working.

8.) Applications

- In Compact Disks
- Digital Telephony
- Digital Audio Applications

PCM Demodulator



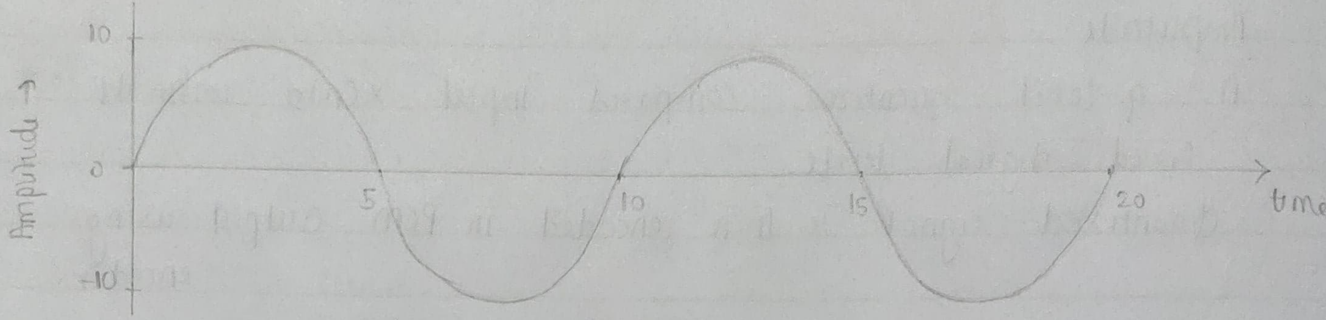
EXPT. NO. NAME: Page No.: 1 Date:

MATLAB CODE: ① Sampling

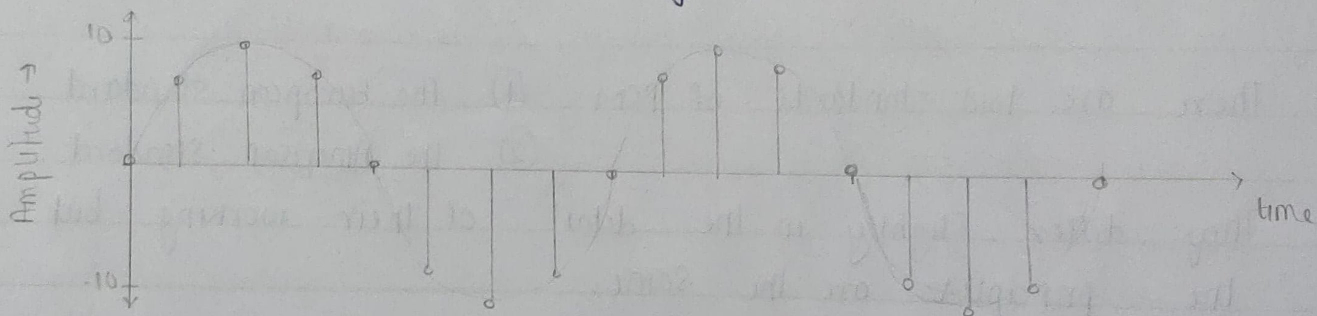
```
n = input ('Enter n value for n-bit pcm system:');
n1 = input('Enter number of samples in a period :');
 L = 2^n;
%% Signal generation
% x = 0:1/100 : 4*pi;
% y = 8 * sin(x)
% subplot (2,2,1);
% plot (x,y); grid on;
 % SAMPLING OPERATION
  x = 0: 2*pi/n1 : 4*pi;
  s = 8 * sin(x) ;
  subplot( 3,1,1);
  plot (s);
  title('Analog Signal');
  ylabel(' Amplitude-->');
  xlabel('Time -->');
  subplot (3,1,2);
  stem(s);
  grid on;
  title ('Sampled Signal');
  ylabel ('Amplitude -->');
  xlabel ('Time -->');
```

Teacher's Signature:

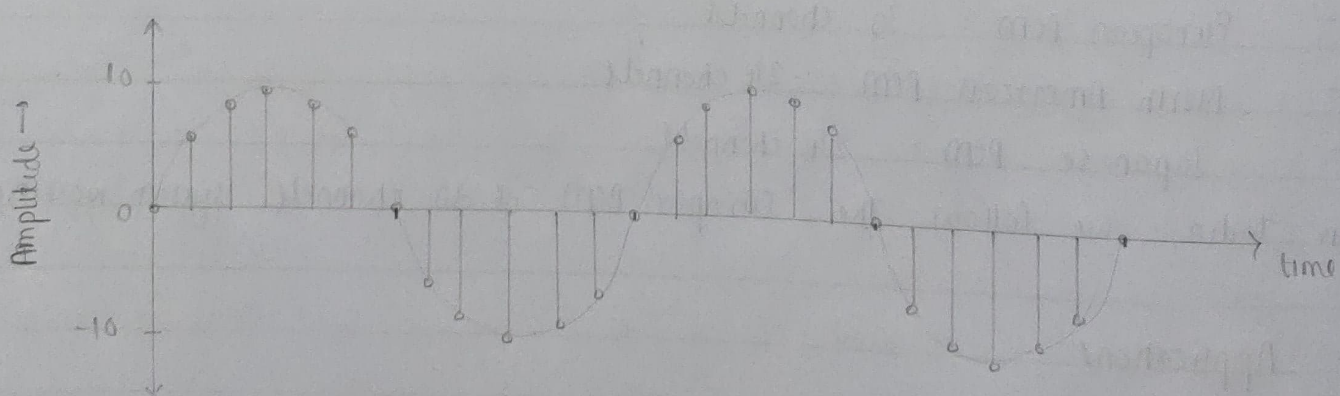
Outputs from MATLAB



Analog signal ↑



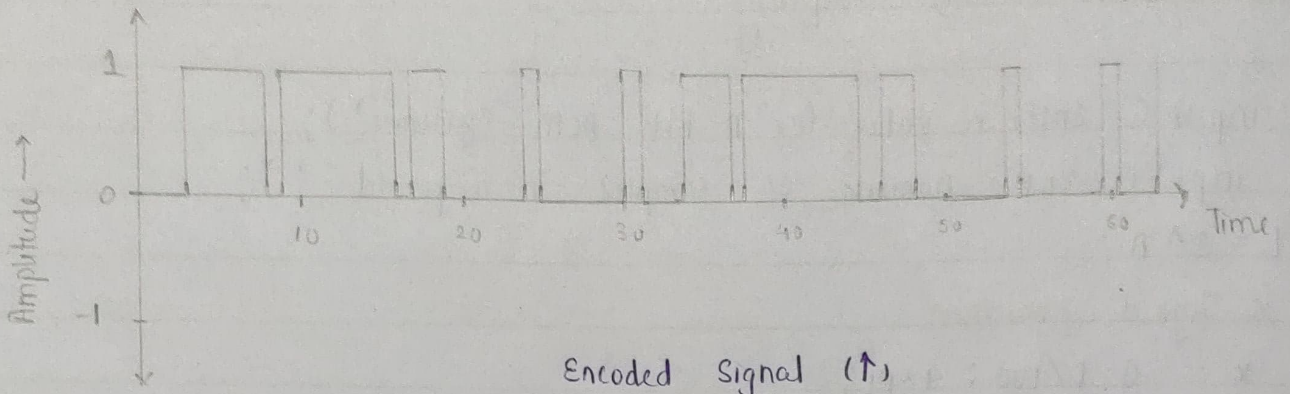
Sampled signal (↑)



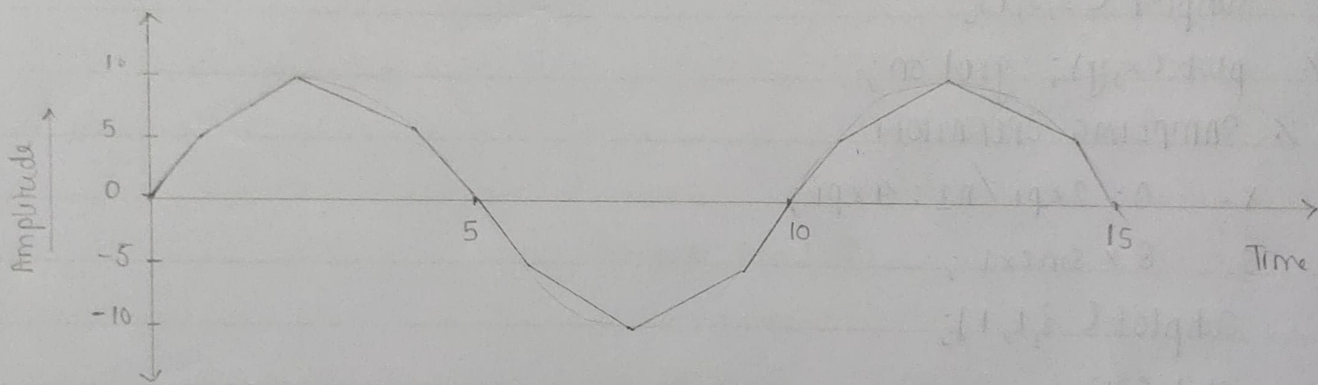
Quantized Signal

② Quantization Process

```
vmax = 8
vmin = - vmax;                          % level are between vmin & vmax with diff del
del = (vmax - vmin)/L;
part = vmin: del: vmax;
code = vmin - (del/2) : del : vmax + (del/2);
[ind,q] = quantiz( s, part, code);      % quantization Process

l1 = length(ind);
l2 = length(q);

for i=1: l1                             % to make index as binary decimal so started from 0 to N
    if (ind(i)~=0)
        ind(i) = ind(i)-1;
    end
    i=i+1;
end

for i=1: l2                             % to make quantize value in between lvl
    if ( q(i) == vmin - (del/2) )
        q(i) = vmin + (del/2);
    end
subplot(3,1,3);                         % Display Quantize Values
stem(q); grid on;
title('Quantized Signal');
ylabel('Amplitude -->');
xlabel('Time -->');
```



Encoded Signal (↑)



Demodulated Signal

EXPT. NO. NAME: Page No.: 13 Date:

③ Encoding

```
% Encoding Process
figure
code = de2bi( ind, 'left-msb');          % convert decimal to binary
k=1;
for i=1: l1
    for j=1:n
        coded(k) = code(i,j);            % convert code matrix to coded row
        j=j+1;
        k=k+1;
    end
    i=i+1;
end
subplot(2,1,1); grid on;
stairs( coded);
axis ([0 100 -2 3]);                     % Display the encoded signal
title('Encoded Signal');
ylabel('Amplitude -->');
xlabel('Time -->');
```

Teacher's Signature:

[U19CS012]

④ Demodulation of PCM signal

```
quant = reshape ( coded, n, length(coded)/n );
index = bi2de ( quant', 'left-msb' );      % getback index in decimal
q= del * index + Vmin + (del/2) ;          % getback quantized values
subplot ( 2,1,2 );   grid on;
plot(q);
title ('Demodulated Signal');              % Plot Demodulated signal
ylabel ('Amplitude -->');
xlabel ('Time -->');
```

> CONCLUSION: (1) We successfully demonstrated the Pulse Code Modulation (PCM) and demodulation technique.

(2) We observed block diagrams for receiver and transmitter of PCM signals. In later stage we also observed demodulation circuit which consist of Shift Registers, Latch and opamp.

(3) In the last phase, we executed MATLAB code and observed sampling, Quantization, Encoding and Demodulation waves and drawn them.

—×—

Teacher's Signature:

EXPERIMENT - 11 [U19CS012]

DELTA MODULATION

AIM: To demonstrate the delta modulation (DM) and demodulation Technique.

SOFTWARE: MATLAB

THEORY: ① Delta Modulation

- It is a technique used to convert analog-to-digital and digital-to-analog signal.

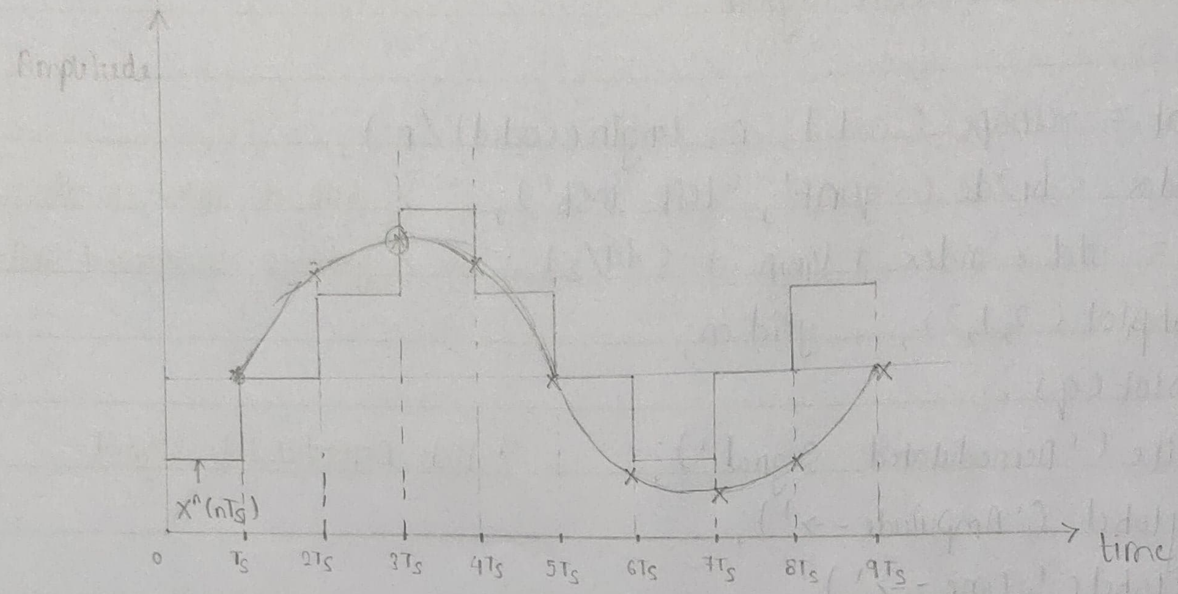
- In this modulation signal is sent in differential form, the data is encrypted/ transmitted in 1 bit.

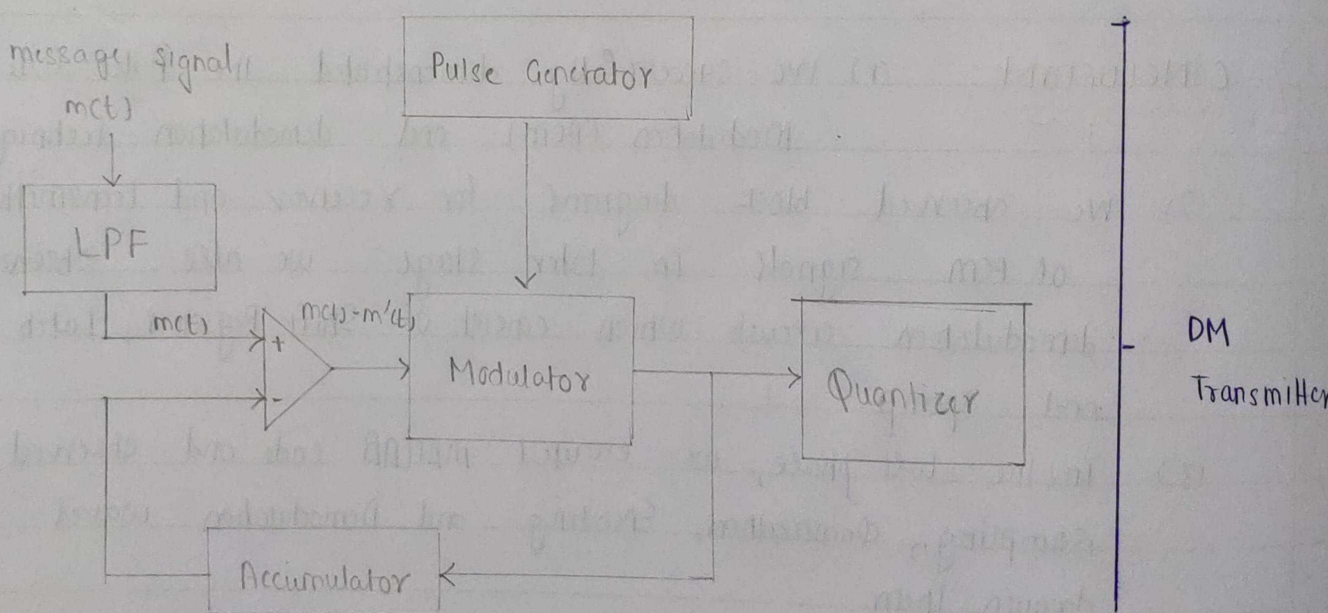
- The analog signal is approximated with series of segments and each segment is compared to original analog to determine the change in relative Amplitude.

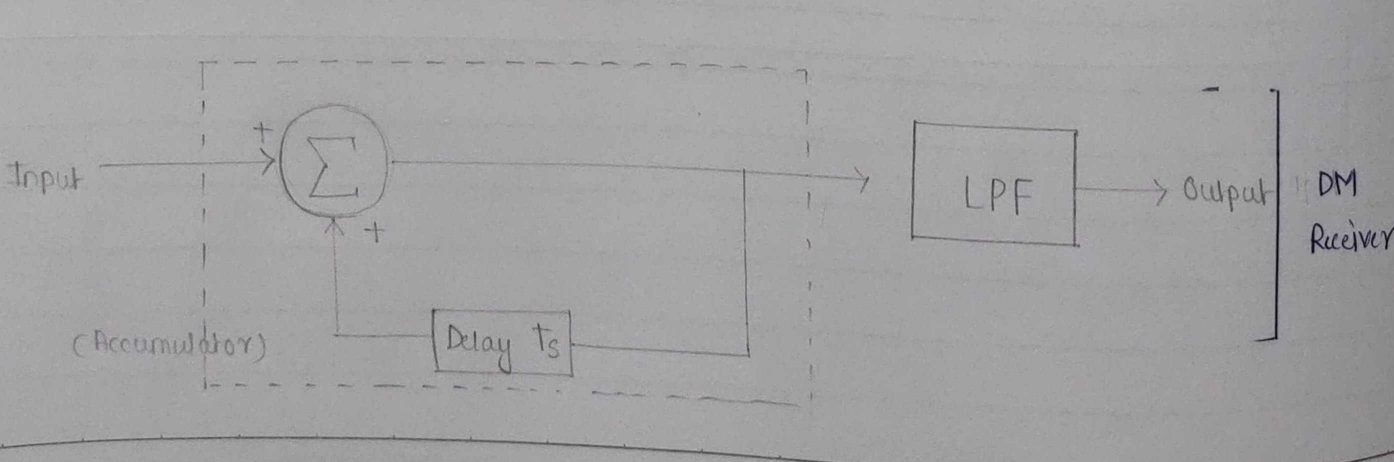
- Hence, only changes in information is sent and if no change occurs it remains on the same state.

- This is the simplified form of Differential Pulse Code Modulation and also called as 1 bit (2 level) version of DPCM.

- It provides a staircase approximation of over-sampled base-band signal. Here, the difference between the present sample and


Amplitude
x^(nTs)
0
Ts
2Ts
3Ts
4Ts
5Ts
6Ts
7Ts
8Ts
9Ts
time


message signal
m(t)
LPF
Pulse Generator
m(t)
+
-
m(t)-m'(t)
Modulator
Quantizer
Accumulator
DM
Transmitter


Input
+
+
Σ
Delay Ts
(Accumulator)
LPF
Output
DM
Receiver

previous approximate sample is quantized into two levels ie Δ (delta).

- This is used for voice transmission.

2.) Operating Principle

- The operating principle of DM is such that, a comparision between present and previously sampled value is performed, the difference of which decides the increment or decrement in the transmitted values.

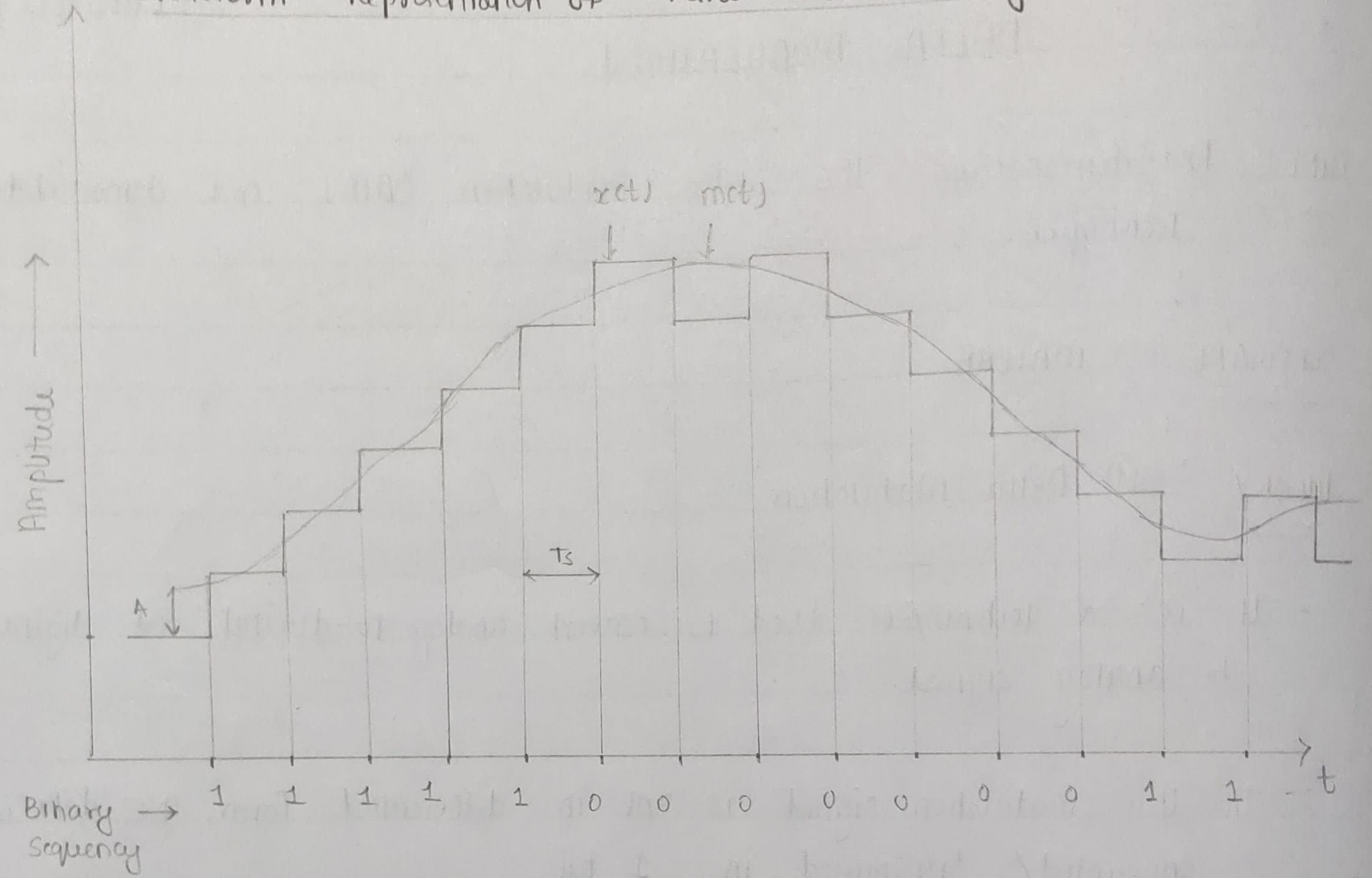
- When the two sample values are compared, either we get difference having a positive polarity or negative polarity.

- If the difference polarity is positive, then the step of the signal denoted by Δ is increased by 1. As against in case when difference polarity is negative then step of the signal is decreased ie. reduction in Δ.

- When $+\Delta$ is noticed i.e. increases the step size, then 1 is transmitted, However, in case of $-\Delta$ ie decrease in step size, 0 is transmitted.

Teacher's Signature:

Waveform Representation of Delta Modulated Signal



3.> Advantages of Delta Modulation

- Due to transmission of 1 bit per sample, it permits low channel bandwidth as well as signaling rate.

- ADC is not required. Thus permits easy generation and detection.

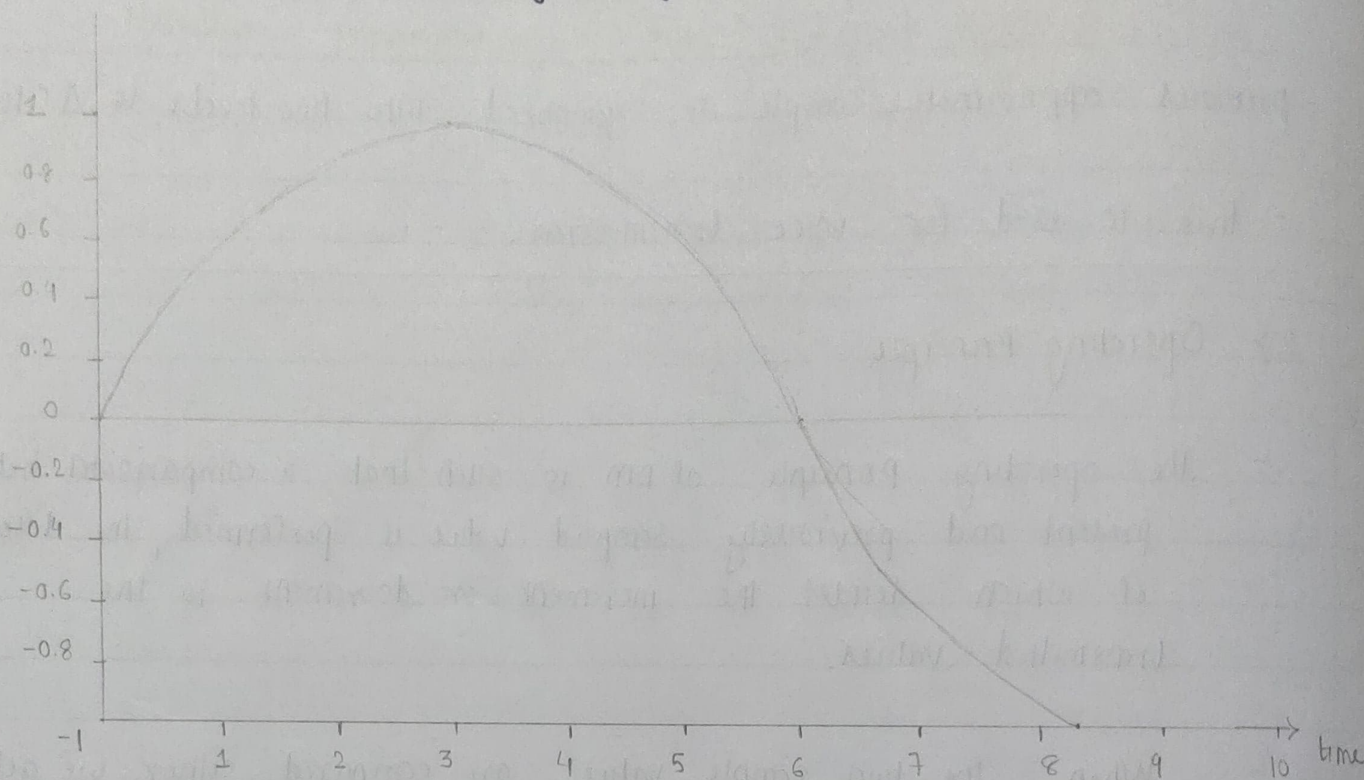
4.> Disadvantages of Delta Modulation

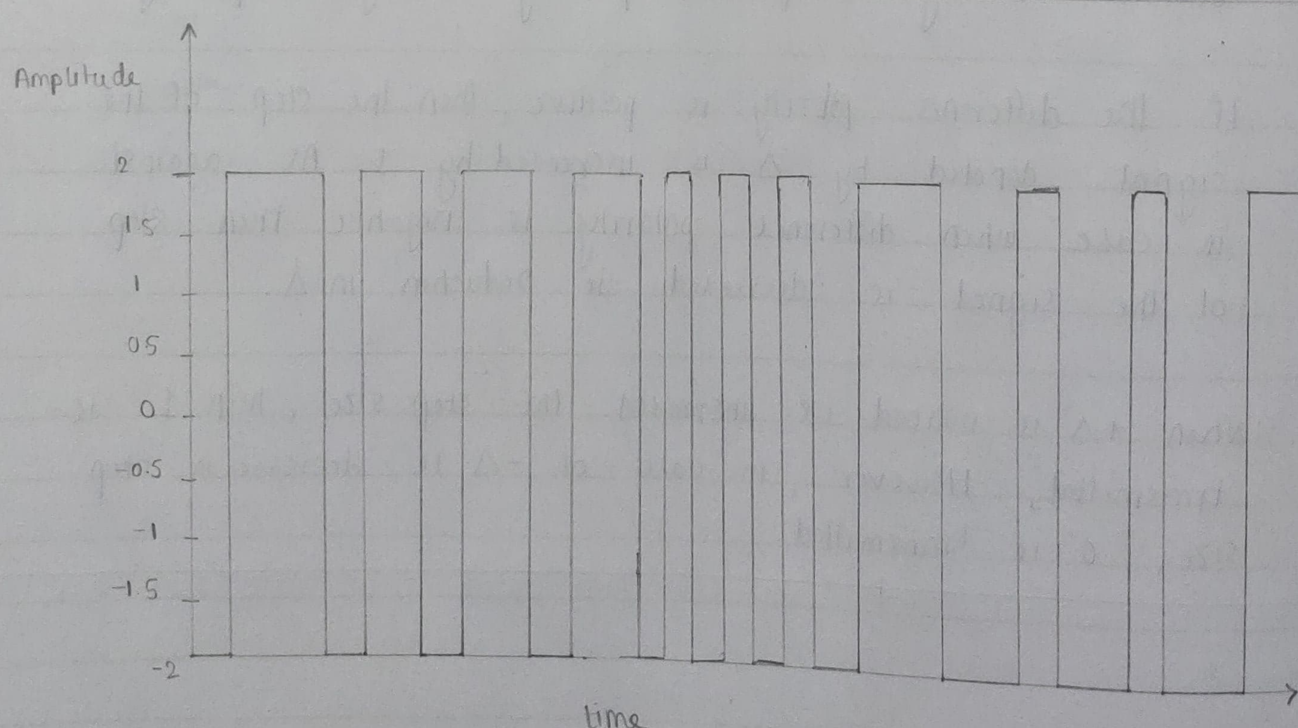
- Delta modulation leads to drawbacks such as slope overload distortion (when Δ is small) and granular noise (when Δ is large.

5.> Application of Delta Modulation

- It is widely used in radio communication devices and digital voice storage and voice transmission.

original signal





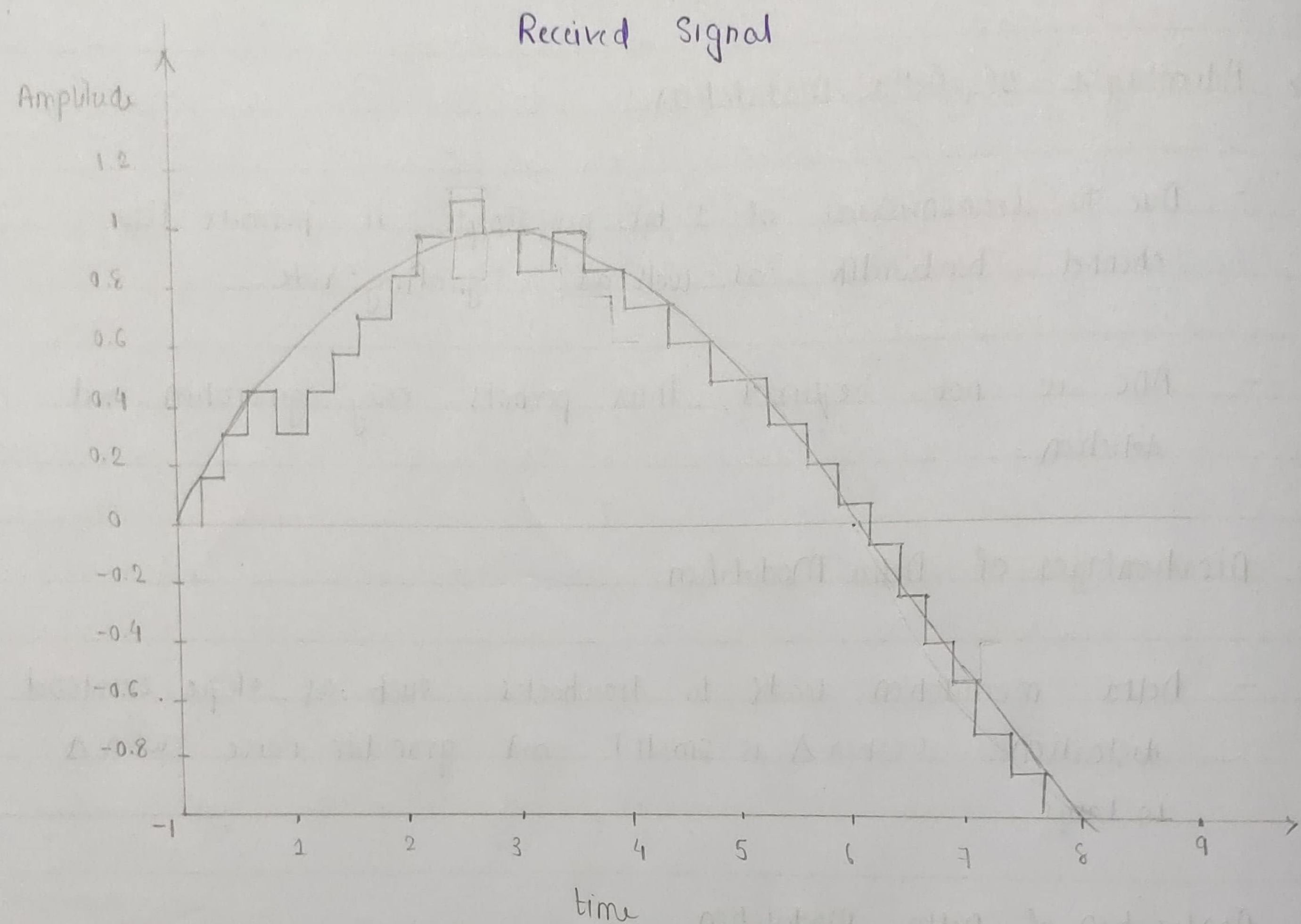
DM output

EXPT. NO. | NAME: | Page No.: | Date:

6) MATLAB CODE :

```
%% Delta Modulation (DM)
    predictor = [0 1];
    partition = [-1:.1:.9];
    step = 0.2
    partition = [0]
    codebook = [-1* step step];
% DM Quantizer
   t = [0: pi/20 : 2*pi];
   x = 1.1 * sin( 2*pi*0.1*t); % Analog Signal
% Quantize x(t) using DPCM
   encoded_x = dpcmenco (x, codebook, partition, predictor);
% Try to recover x from modulated signal
   decoded_x = dpcmdeco ( encoded_x, codebook, predictor);
% Plots
   figure
   plot(t,x);
   xlabel('time');
   title('original signal');
    figure
    stairs (t, 10,* codebook (encodedx +1) .'g');
    xlabel ('time');
    title ('DM output');
    figure
     plot (t,x)
    hold;
```

Teacher's Signature:


Received Signal
Amplitude
1.2
1
0.8
0.6
0.4
0.2
0
-0.2
-0.4
-0.6
-0.8
-1
1
2
3
4
5
6
7
8
9
time

[U19CS012]

```
stairs( t, decoded x );
grid;
xlabel('time');
title('received signal');
```

> CONCLUSION: We have successfully understood and demonstrated the delta modulation (DM) and demodulation technique. and also verified it with sampled, quantized /encoded and decoded time domain signal using MATLAB in virtual LAB mode.

X

END OF DCOM JOURNAL

SUBMITTED BY:

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