LABORATORY JOURNAL

Submitted in partial fulfillment of the requirement
For the Subject

"DIGITAL COMMUNICATION"

(EC 209)

: Prepared & Submitted By :

Ms. Madhumitha Ichapuram
(Admission No. U19CS081)

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

: Laboratory Teachers :

Prof. N. B. Kanirkar
Prof. M. C. Patel
Mr. Abhishek Tripathi

(Aug to Dec - 2020)

ELECTRONICS ENGINEERING DEPARTMENT
Sardar Vallabhbhai National Institute of Technology
Surat-395 007, Gujarat, INDIA.

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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 **Ms. Madhumitha Ichapuram** bearing **Admission No.U19CS081** 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

MADHUMITHA ICHAPURAM

Admission Number: U19CS081

B.Tech. II (CSE) 3rd Semester

EXPERIMENT-1

SPECTRUM ANALYSER AND OBSERVE SPECTRUM

AIM-

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

APPARATUS-

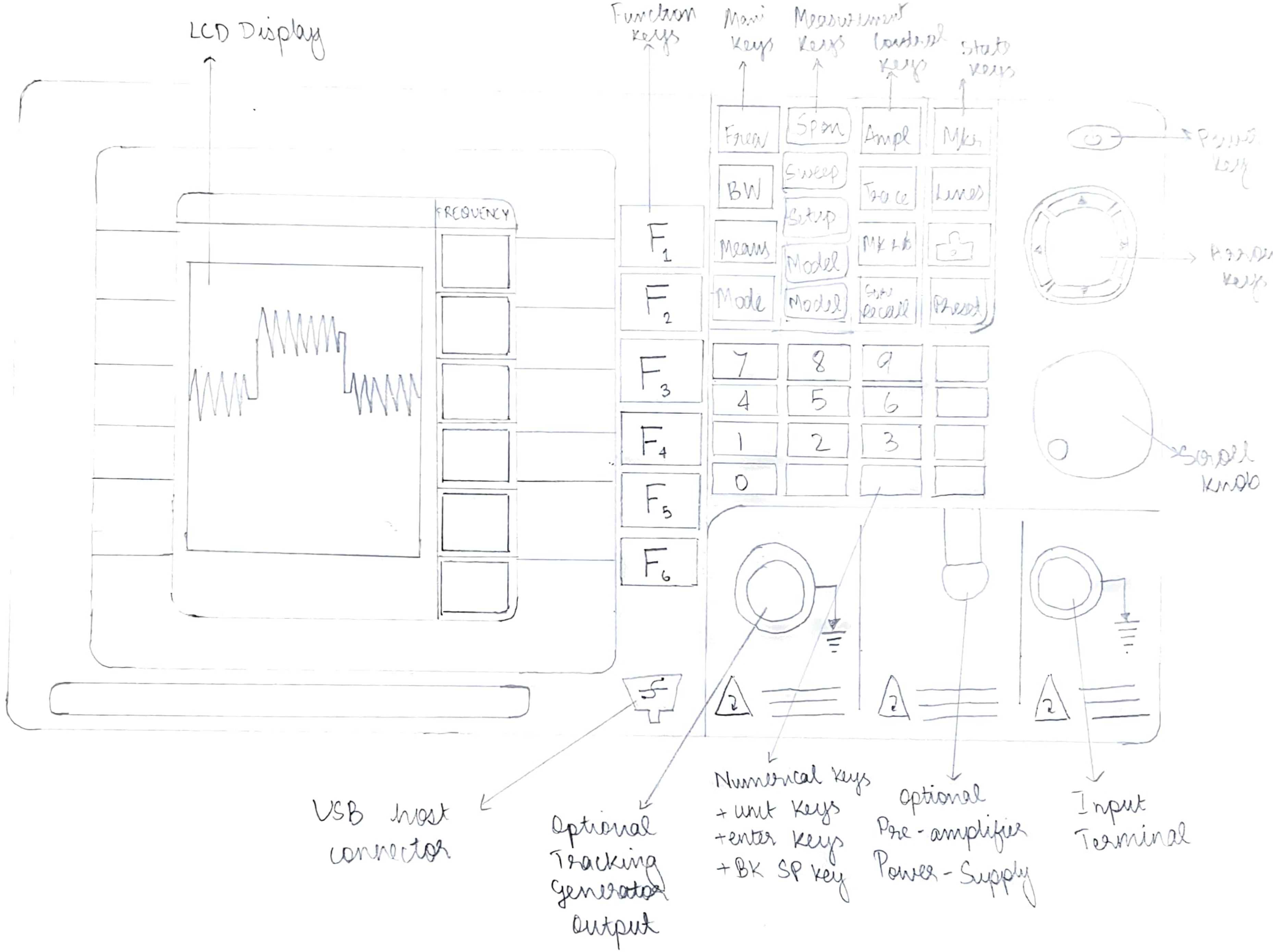
Spectrum Analyzer (9 KHz - 3GHz) Function generator

THEORY-

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. To the casual observer, a spectrum analyzer looks like an oscilloscope and, in fact, some lab instruments can functions either as oscilloscopes or spectrum analyzers.

A spectrum analyzer can be used to determine whether or not a wireless transmitter is working according to federally defined standards for purity of emissions. Output signals at frequencies other than the intended communications frequency appear as vertical lines (pips) on the display. A spectrum analyzer can also be used to determine, by direct observation, the bandwidth of a digital or analog signal.

A spectrum analyzer interface is a device that can be connected to a wireless receiver or a personal computer to allow visual detection and analysis of electromagnetic signals over a defined band of frequencies


LCD Display
Function Keys
Main Keys
Measurement Keys
Control Keys
State Keys
Power Key
Arrow Keys
Scroll Knob
FREQUENCY
Freq
BW
Meas
Mode
Span
Sweep
Setup
Model
Model
Ampl
Trace
Mk Lk
Sav Recall
Mkr
Lines
Preset
7
8
9
4
5
6
1
2
3
0
F1
F2
F3
F4
F5
F6
USB host connector
Optional Tracking Generator Output
Numerical keys + unit keys + enter keys + BK SP key
Optional Pre-amplifier Power-Supply
Input Terminal

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 operation
- Autoset
- 9 kHz - 3 GHz frequency range

FREQUENCY SELECTION AND THEIR SELECTION METHODS-

(1) Frequency -
- Frequency/span: The frequency key, together with span key sets the frequency scale
- View Signal (Center & Span): Center-and-span method defines the center frequency & the left/right bandwidth (span) to locate the signal
- Setting frequency adjustment step: Frequency adjustment step defines the arrow keys resolution for center, start and stop frequency.

Panel Operation:
- Press frequency key
- press F4 (step)
- Enter the value using numerical and unit keys, arrow keys and scroll nope.

(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 nope.

(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 the 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 (Key)
- Enter the value using numerical and unit keys, arrow keys and scroll nope.

(7) Set stop frequency -
Panel Operation:
- Press frequency key
- press F 3 (stop)
- Enter the value using numerical and unit keys, arrow keys and scroll nope.

(8) Full or zero span –

- Full or zero span setting sets the span to extreme values: 3GHz (full) or 0 KHz (zero) they provide faster ways to ~~provide~~ view signals in certain situations such as in time domain (0 span) for viewing modulation or in full span for viewing signals with unknown frequencies

(9) Display full frequency span –

Panel Operation:

- Press the span key
- Press F2 (full span)
- Range: 3GHz (fixed)
- Full span also sets these parameters to fixed values
- Center frequency: 1.5 GHz
- Start frequency: 0 KHz
- Stop frequency: 3GHz

(10) Zero span display –

- Zero span display can be obtained by pressing F3 key
- Start frequency and stop frequency remains same as that of center frequency
- Note: last span setting can be recalled by F4 Key

AMPLITUDE SELECTION AND SETTINGS METHODS –

(1) Amplitude –

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

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

(3) Set reference amplitude -
- The reference level defines the amplitude at the top of the displayed range

Panel Operation:
- Press amplitude key
- Press F1 (Reference Level)
- Enter the value using numerical and unit keys, arrow keys and scroll knob. Arrow keys and scroll knob, scroll knob resolution: vertical scale

Range:
- dBm -110 to +20dBm, 0.1dB resolution
- dBmV -63.1 to 66.99 dBmV, 0.01dB resolution
- dBμV -3.01 to 126.99 dBμV, 0.01dB 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)
- ~~Sel~~ Select and press the unit from F1 (dBm), F2 (dBmV) and F3 (dBμV)
- Press F6 (return) to go back to previous menu
- dBmV −110 to +20 dBm, 0.1 dB resolution
- dBμV −3.01 to 126.99 dBμV, 0.01 dB resolution
- Set external offset level

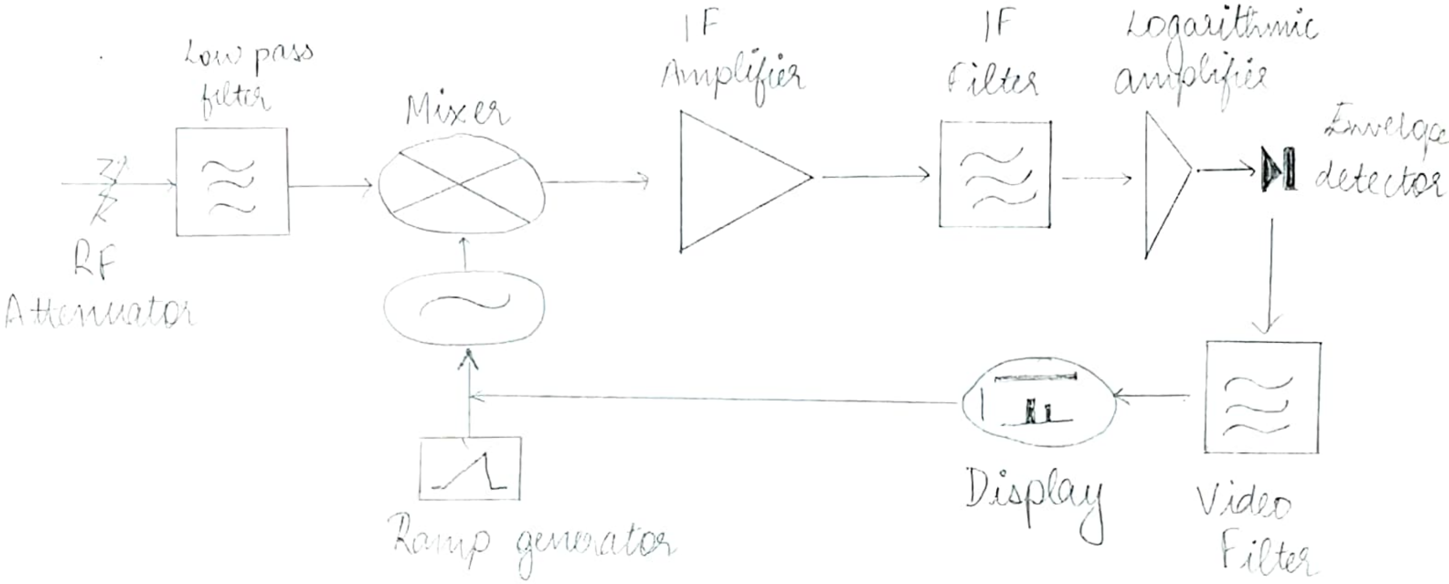
(5) Background –
- External offset compensates the amplitude gain or loss caused by an external network or device.

Panel Operation:
- Press amplitude key
- Press F4 (external gain)
- Enter the value using numerical and unit keys, arrow keys and scroll knob.

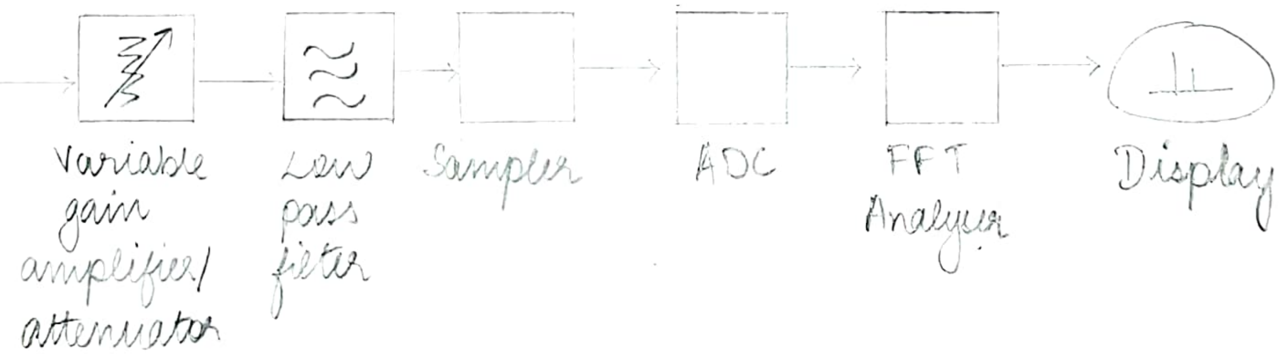
Range:
- −20 dB to +20 dB, 0.1 dB resolution

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

BLOCK DIAGRAM:



(Superheterodyne or swept frequency spectrum analyzer block diagram)



OBSERVATIONS -

WAVEFORM: SINE

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

WAVEFORM: SQUARE

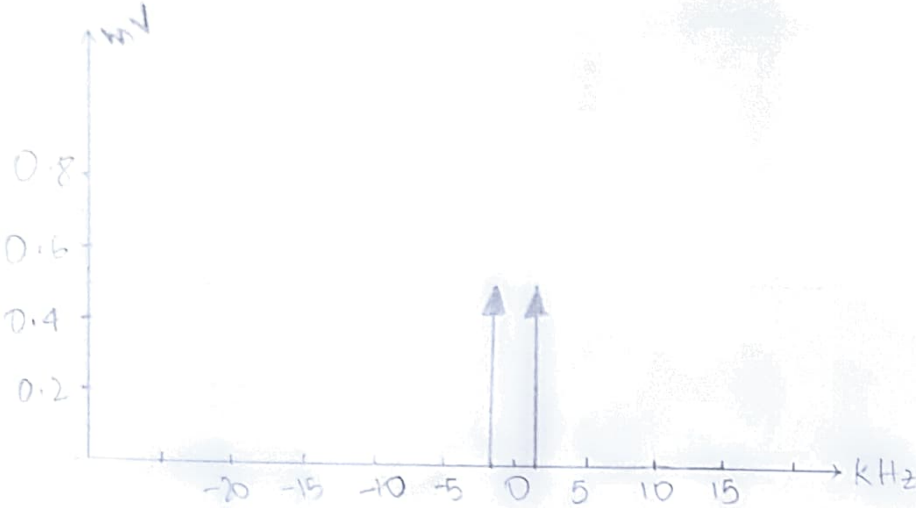
S. No	Frequency (kHz)	Amplitude (mV)
1	2	1
2	2.5	1.2
3	3	1.5
4	4	1.6
5	5	2

CONCLUSION -

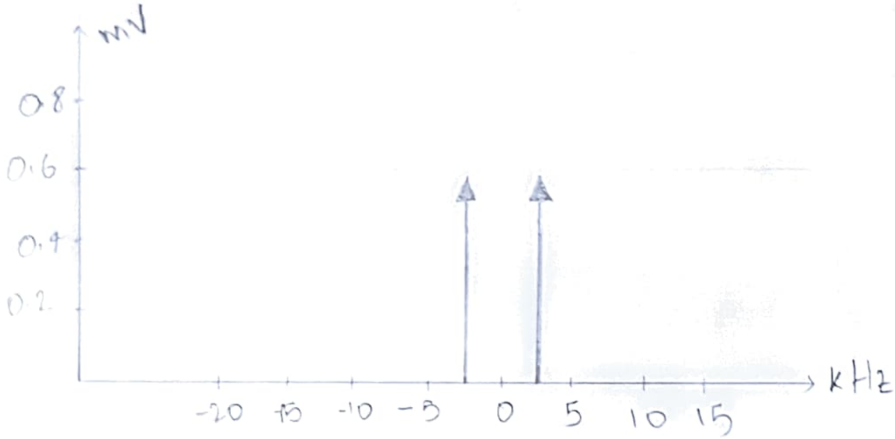
Hence, successfully verified and analyzed the spectrum of sine and square waveform for different frequencies and amplitude

WAVEFORM : SINE

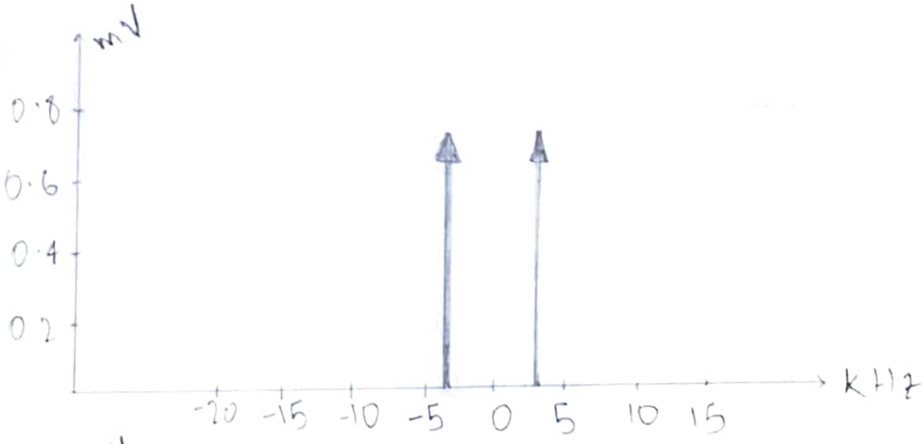
1)



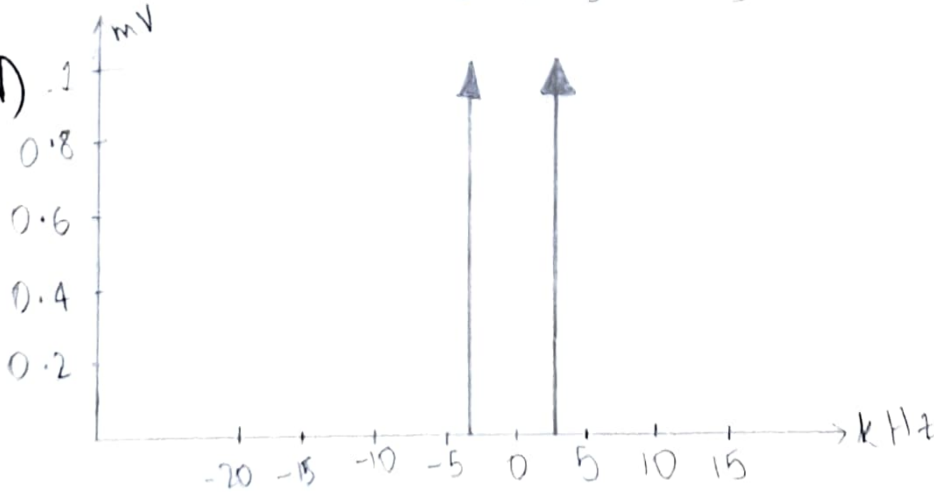
2)



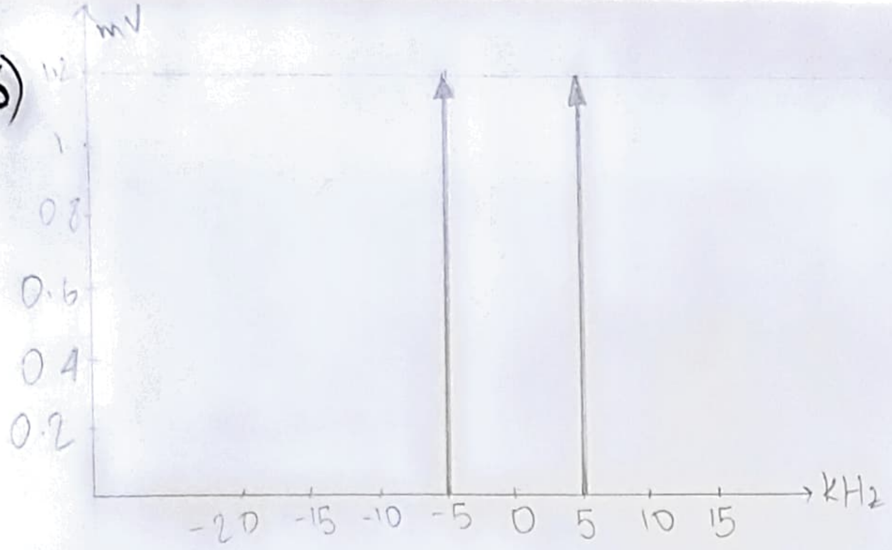
3)



4)

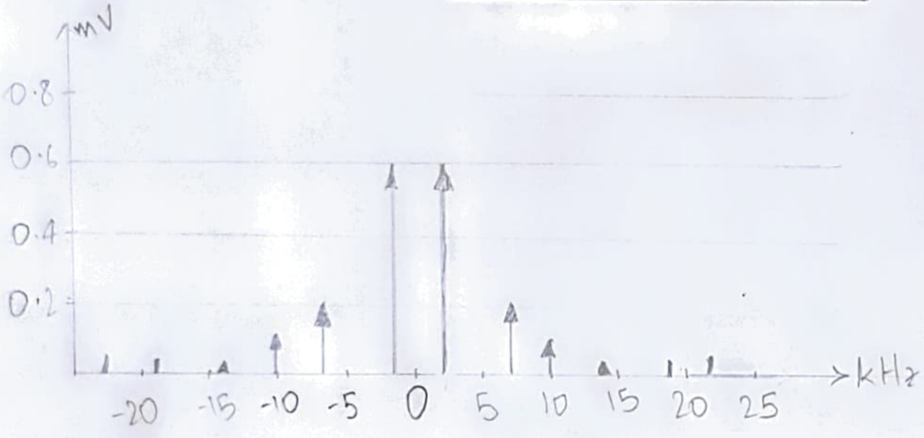


5)

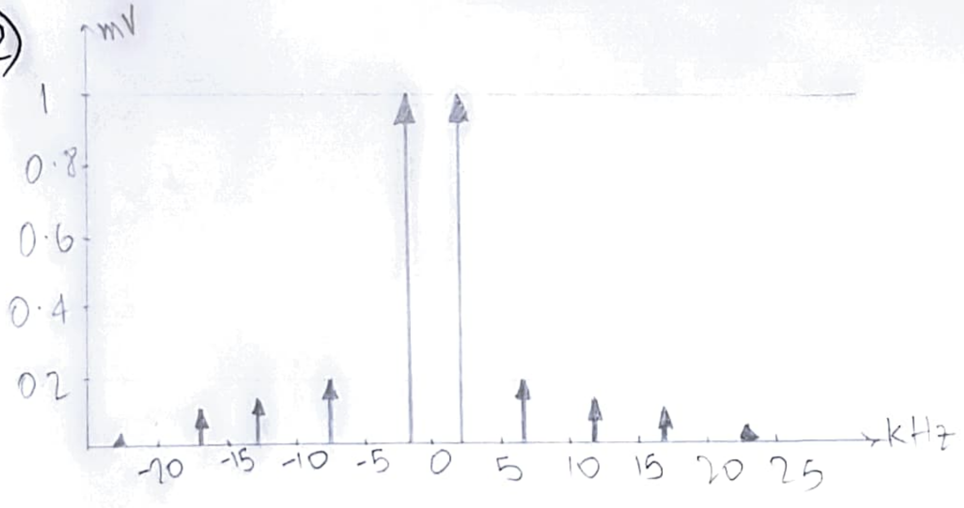


WAVEFORM : SQUARE

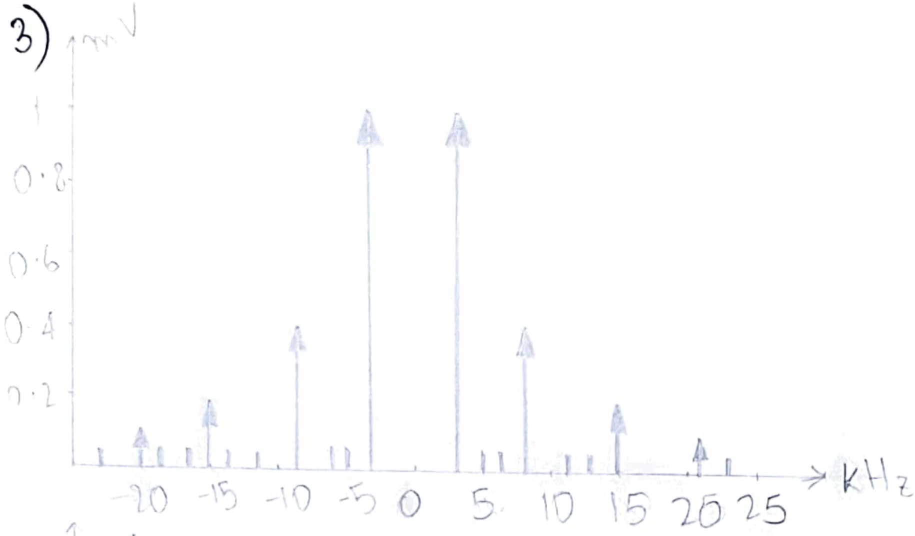
1)



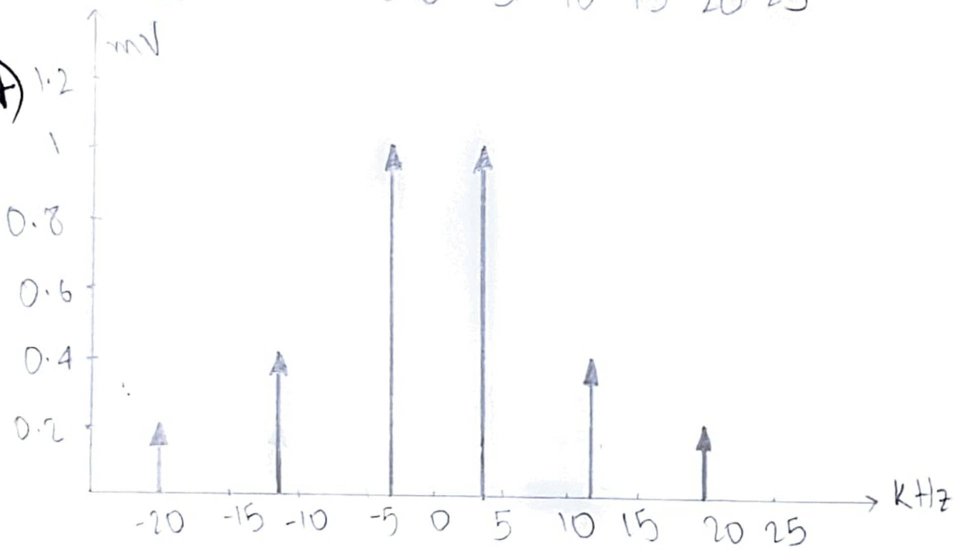
2)



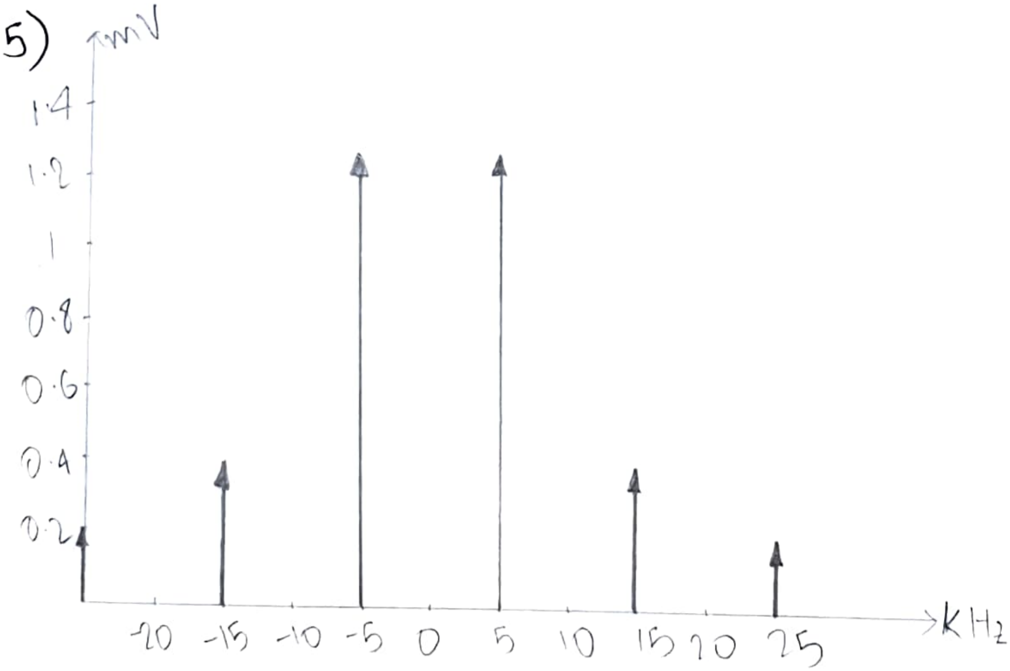
3)


mV
1
0.8
0.6
0.4
0.2
-20
-15
-10
-5
0
5
10
15
20
25
KHz

4)


mV
1.2
1
0.8
0.6
0.4
0.2
-20
-15
-10
-5
0
5
10
15
20
25
KHz

5)


mV
1.4
1.2
1
0.8
0.6
0.4
0.2
-20
-15
-10
-5
0
5
10
15
20
25
KHz

EXPERIMENT-2

AIM-
To perform sampling and reconstruction of signal and obtain its waveforms. Also verify the Nyquist criteria by varying sampling frequency.

APPARATUS-
Nyquist Applet

THEORY-
A continous time (or analog) signal can be stored in a digital computer, in the form of equidistant discrete points or samples. The higher the sampling rate (or sampling frequency, f_s), 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 has continous time varying feature of the signal.

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

Technique that can be used for Analog to Digital Conversion:
Pulse Code Modulation (PCM)
It has following three processes
- Sampling
- Quantization
- Encoding

SAMPLING -

- It is the process of measuring the instantaneous values of continous time signal in discrete form
- Sample is a piece of data taken from the whole data which is continous in the ~~form~~ time domain.
- When a source generates an analog signal and if that has to be digitized, having 1's and 0's, i.e, high or low, the signal has to be discretized in time
- This discretization of analog signal is called sampling

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

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

Nyquist Rate -

- Suppose that a ~~data~~ 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

$$f_s = 2w$$

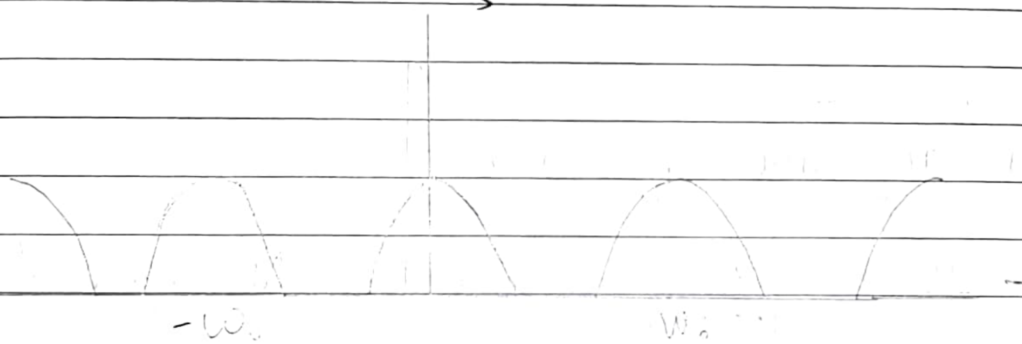
where f_s is the sampling rate and w is the highest frequency

- This rate of sampling is called Nyquist rate
- A theorem called sampling theorem.

Condition 1 —

- Oversampling → If a sampled at higher rate than 2ω in the frequency domain $(f_s > \omega)$

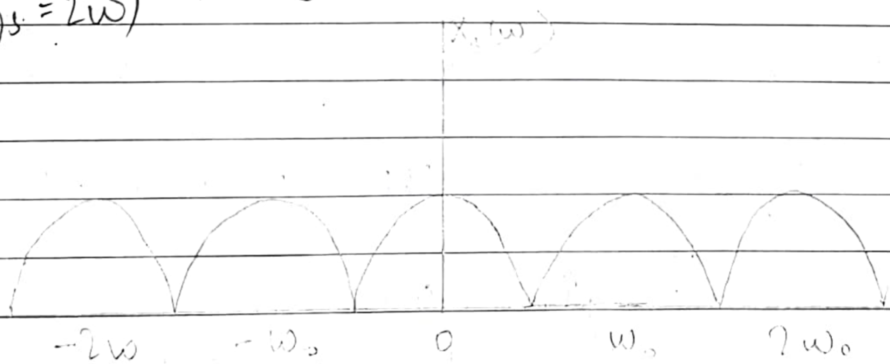
$$X_s(\omega) = \frac{1}{T_s} \sum_{n=-\infty} X(\omega - n\omega_0)$$



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

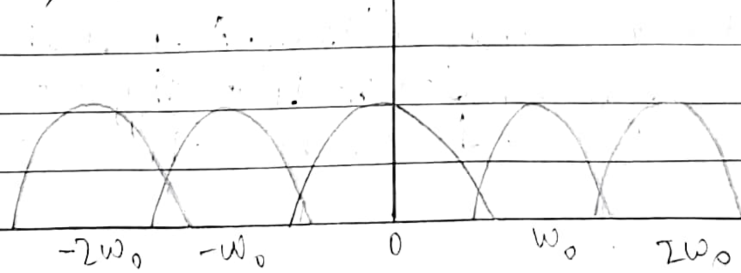
Condition 2 –

- If the sampling rate is equal to twice the highest frequency $(f_s = 2\omega)$



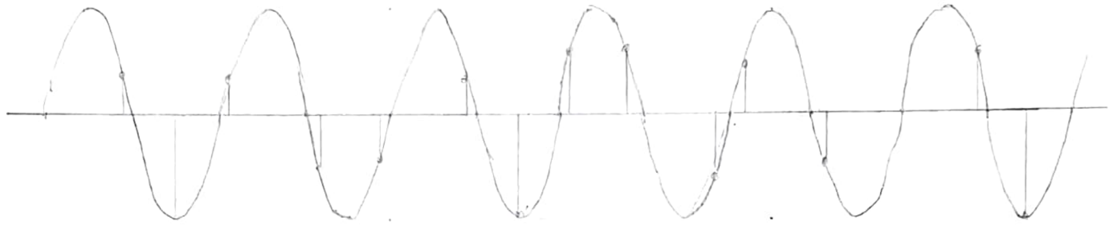
Condition 3 –

- If the sampling rate is less than twice the highest frequency $(f_s < 2\omega)$

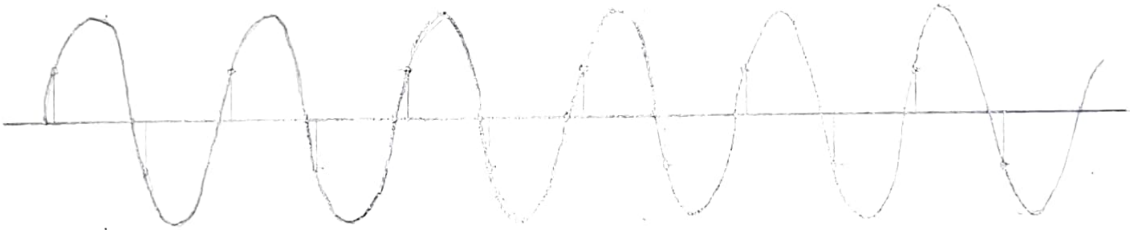


Sampling of a sinusoidal signal of frequency f_s at different sampling rates f_s. With dashed lines are shown the alias frequencies occuring when $f_s/f < 2$

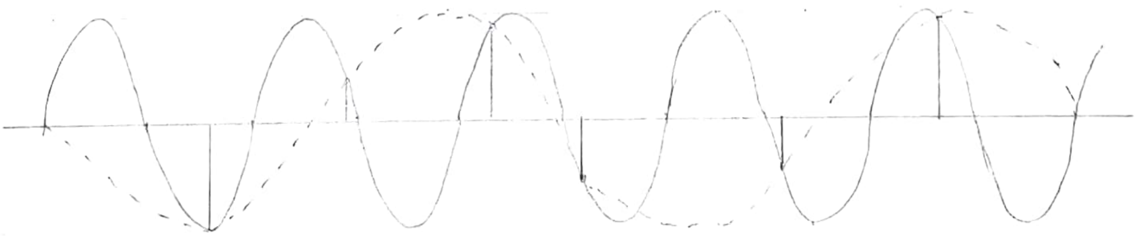
$\frac{f_s}{f} = 2.6$



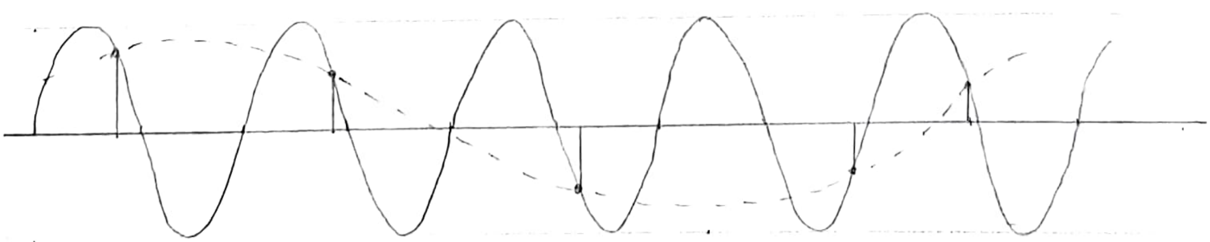
$\frac{f_s}{f} = 2.0$



$\frac{f_s}{f} = 1.4$



$\frac{f_s}{f} = 0.8$



QUANTIZATION -

- The method of sampling chooses a few points on the analog signal and then these points are joined to round off the value to a near stabilized value. Such a process is called quantization
- Quantization of a signal produces the closest representable value.

ENCODING -

The digitization of the analog signal is done by the 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

→ Designing this filter is to determine the bandwidth required in the acquisition system

→ The maximum frequency of the input signal should be less than or equal to half of the sampling rate

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

If f_s is the sampling frequency, then the critical frequency (or Nyquist limit) f_N is defined as equal to $f_s/2$.
Any sinusoidal component of the signal of frequency f' higher than f_N (eg $f' = f_N + \Delta f$) is not only lost, but is reintroduced in the sampled signal by folding at frequency f_N an alias sinusoidal component of frequency $f' = f_N - \Delta f$

This ~~fq~~ effect is known as aliasing. A sinusoidal signal (blue) of frequency f is sampled at four different sampling frequencies $f_s = 2.6F, 2.0F, 1.4F$ & $0.8F$

In two first cases the sampling rates are adequate enough for the accurate reconstruction of the original sinusoidal signal whereas in the last two, subsampling occurs, and the collected points ~~ø~~ may be considered as belonging to signals of lower frequencies. The alias frequencies due to subsampling can be calculated ~~by~~ by the following equation

Alias frequency :- $f' = |f - k f_s|$
where $k = 1, 2$

Hence when $f_s/f = 1.4$, the alias frequency is F'
$= |F - 1 \times 1.4F| = 0.4f$
whereas when $f_s/f = 0.8$, the alias frequency is F'
$= |F - 1 \times 0.8F| = 0.2F$

Problems arising due to aliasing :- The effect of the sampling frequency on the spectrum of a signal consisting of infinite number of sinusoidal components.

In Fig, the Nyquist frequency is sufficiently higher from the maximum frequency (f_{max}) component of the signal and the stored signal is not distorted.
The opposite occurs in Fig where $f_{max} > f_N$ and all frequency components higher than f_N are not only lost, but they are also folded at f_N and they are added to the other sinusoidal components, corrupting thus the stored signal due to aliasing.

A typical application:- For the digitilization of sound, a sampling rate of about 6 kHz is sufficient for telephony, since normal human voice does not contain an appreciable amount of frequency components higher than 2.5–3 kHz. However, a sampling rate of about 40 kHz is needed for the digitalization of music, since frequency components of about 15-20 kHz are common and needed for achieving fidelity of sound reconstruction.

Notes -

(1) In practice the sampling rate f_s is commonly selected in the range $2.5 \times f_{max} - 3 \times f_{max}$. For digital recording of music in CD, a sampling rate of 44.1 kHz is commonly used.

(2) Prior to sampling, the signal must pass through a low-pass filter which will remove all unnecessary components (e.g. noise) higher than f_{max}, preventing thus the "contamination" of the stored signal by their aliased frequencies.

1) SIGNAL FREQUENCY = 10 Hz

SAMPLING FREQ (Hz)	ALIAS FREQ (Hz)
7	3
10	0
15	5
20	–
22	–

2) SIGNAL FREQUENCY = 20 Hz

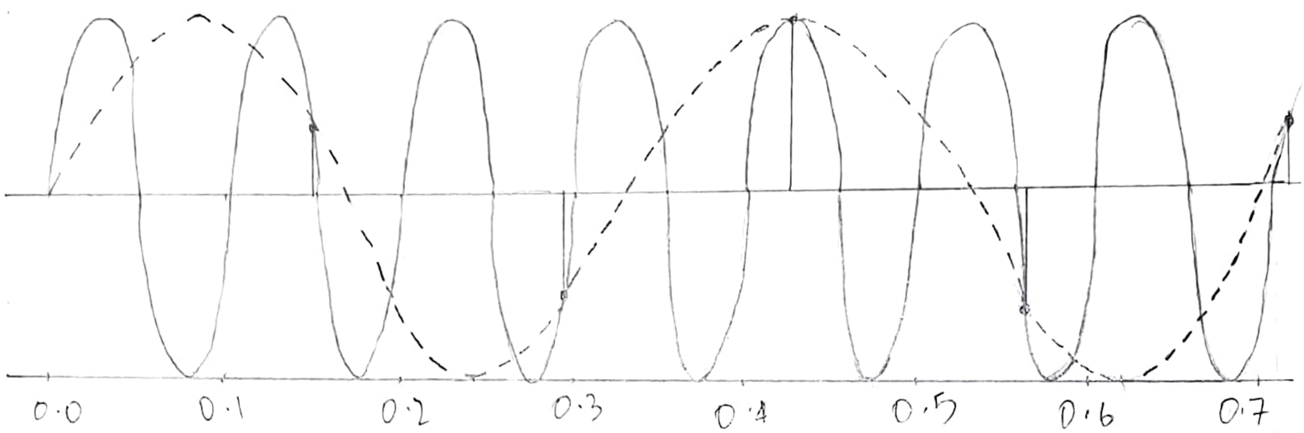
SAMPLING FREQ (Hz)	ALIAS FREQ (Hz)
19	0
20.1	0.1
30	10
40	–
42	–

CONCLUSION -

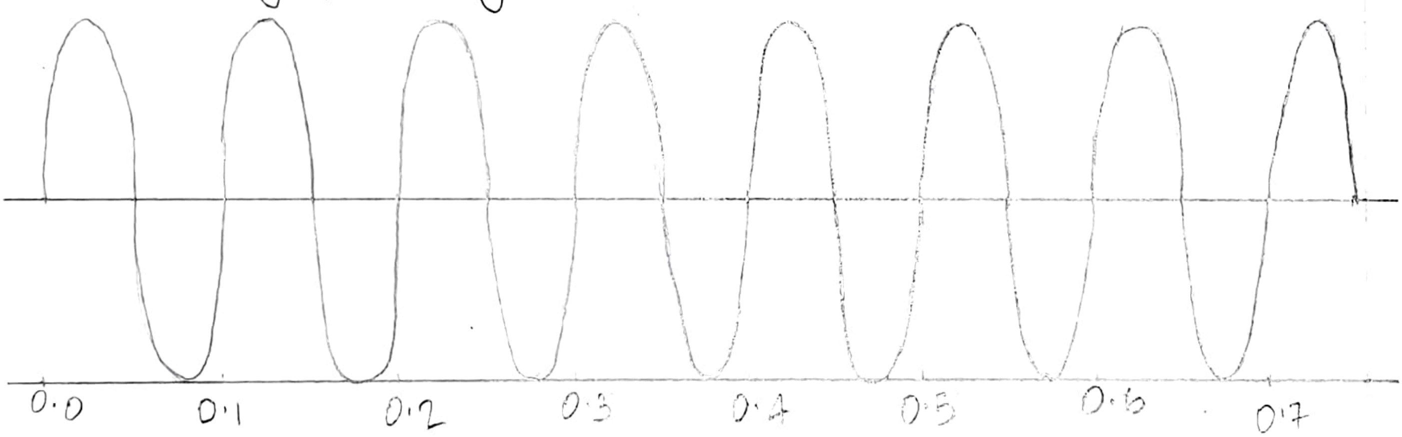
Successfully examined the sampling and reconstruction of a signal and verified the Nyquist criteria by varying the sampling frequency.

SIGNAL FREQUENCY: 10Hz

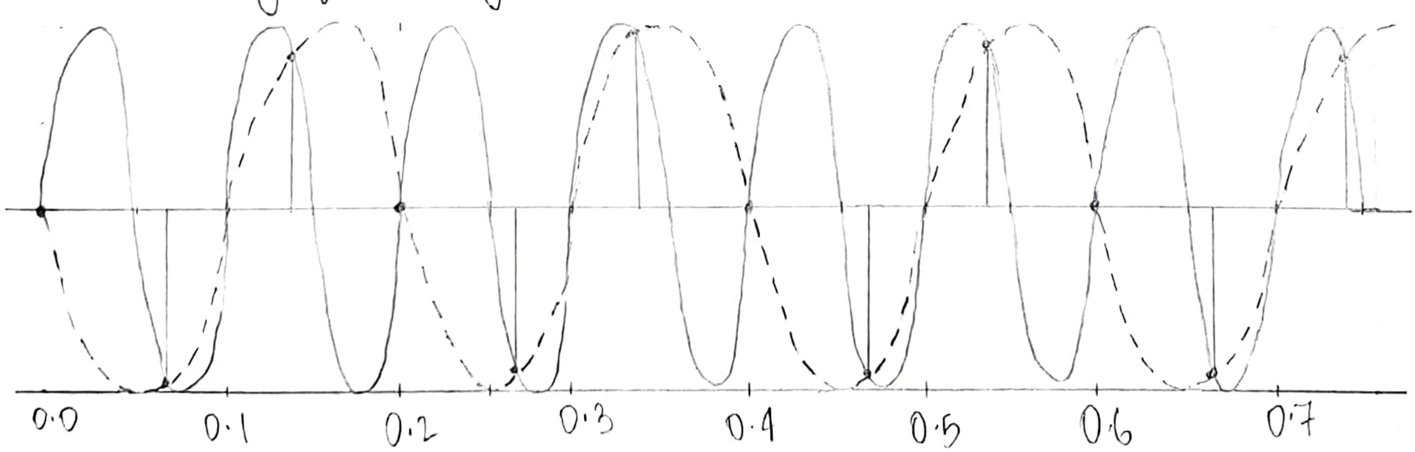
1) Sampling frequency = 7Hz



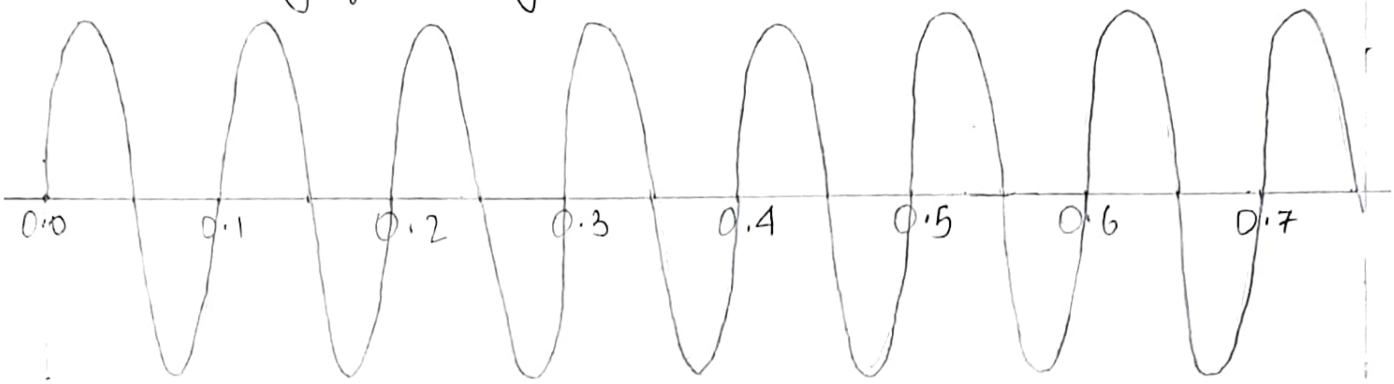
2) Sampling frequency = 10Hz



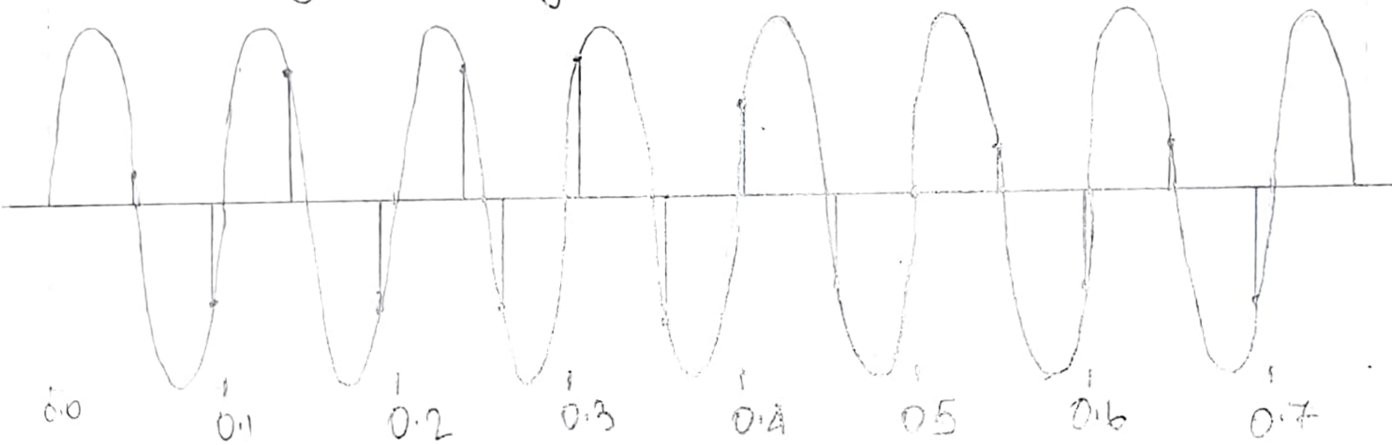
3) Sampling frequency = 15Hz



4) Sampling frequency = 20Hz

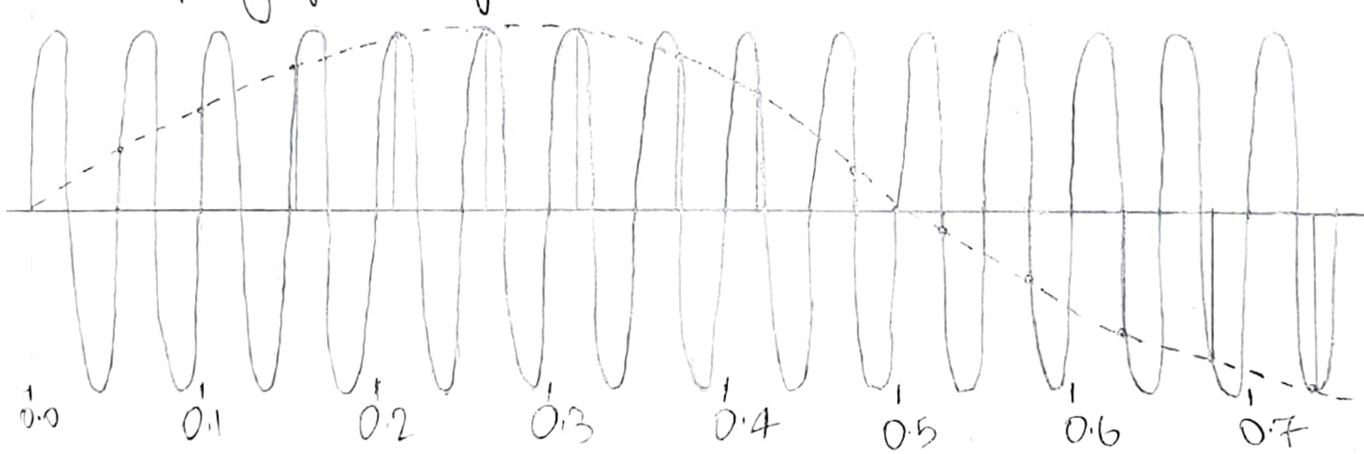


5) Sampling frequency = 22Hz

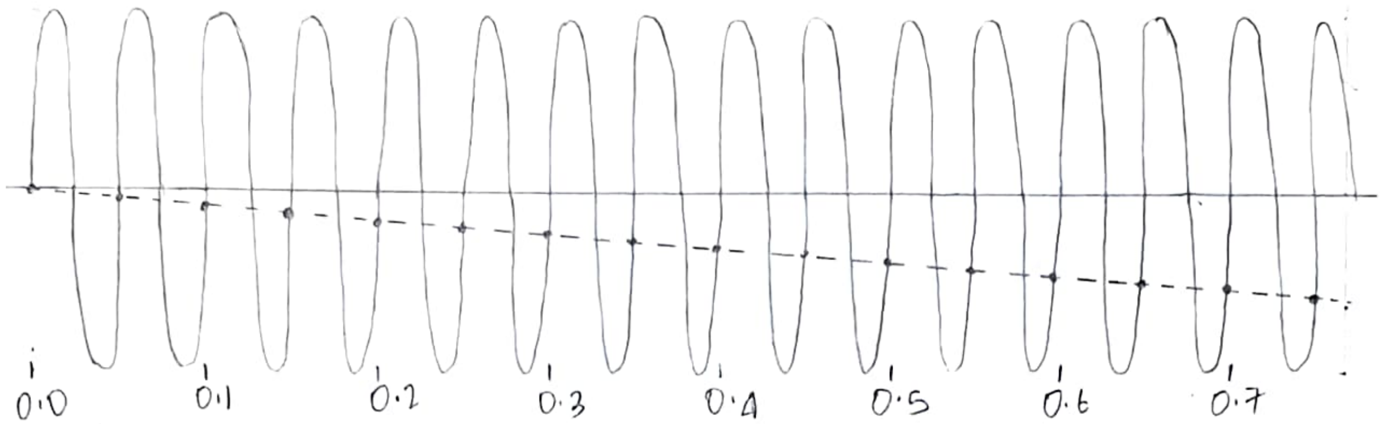


SIGNAL FREQUENCY = 20Hz

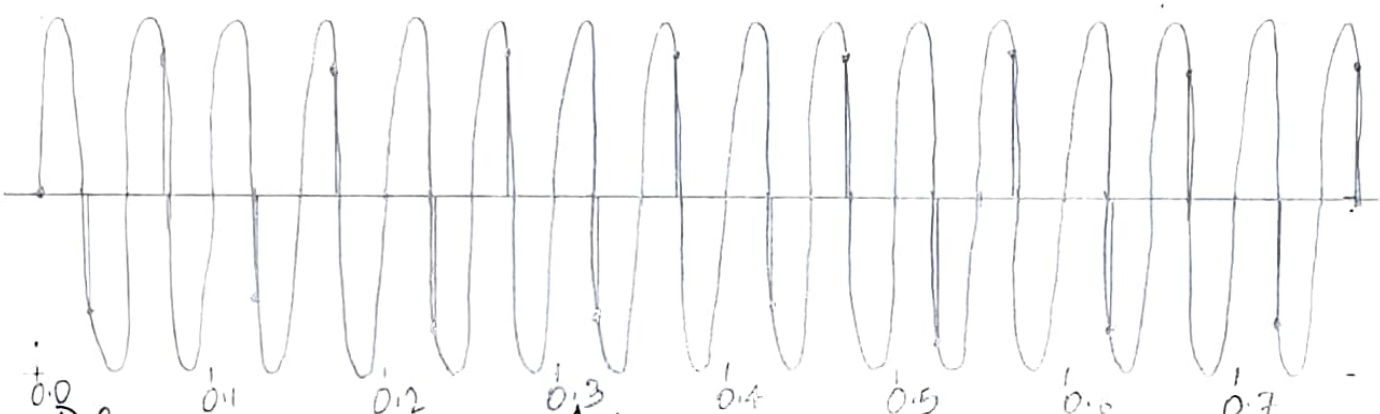
1) Sampling frequency = 19Hz



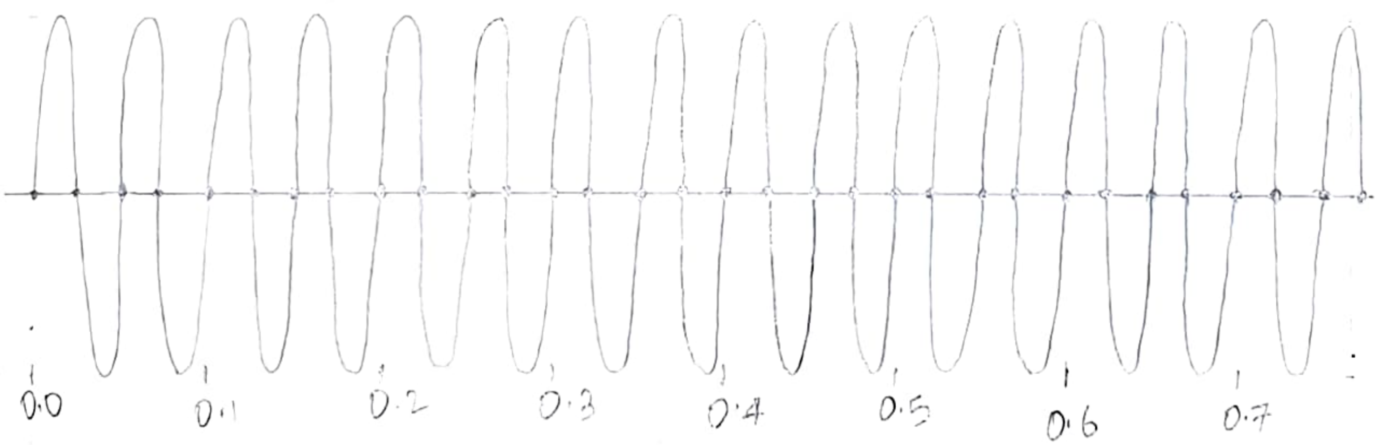
2) Sampling frequency = 20.1 Hz



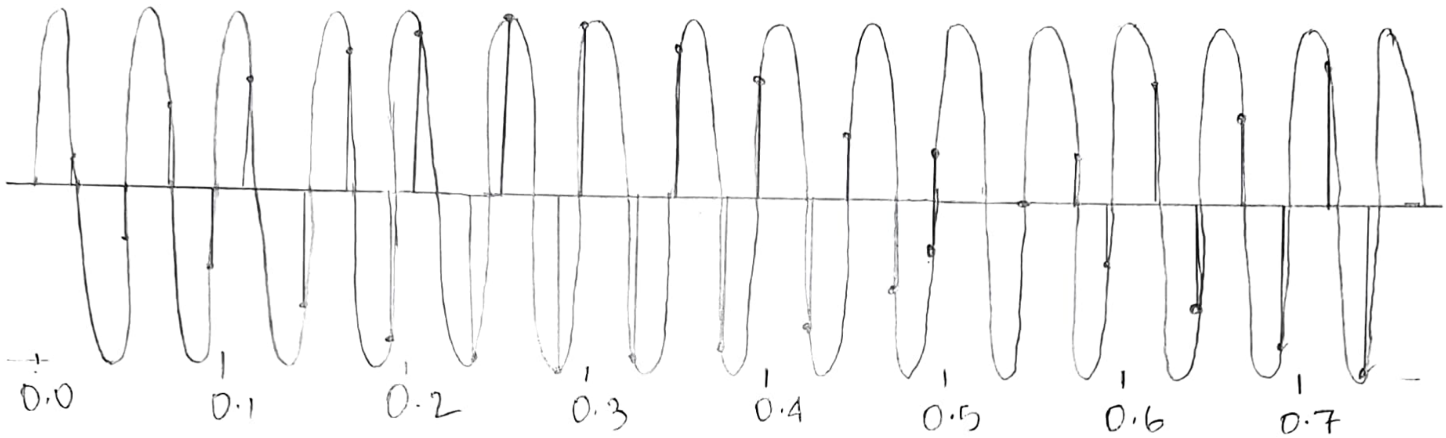
3) Sampling frequency = 30Hz



4) Sampling frequency = 40Hz



5) Sampling frequency = 42 Hz



EXPERIMENT-3

AIM-
Study of an amplitude modulated (AM) scheme, depth of modulation, waveforms, spectra and trapezoidal display

APPARATUS-
Online simulation tools
1. labAlive AM analyzer
2. Envelope detector
3. synchronous detector

THEORY-
To modulate a signal means to regulate or adjust some parameters of a high frequency carrier wave with a low frequency message signal.

When that parameter is amplitude, it is called amplitude modulation and the signal we get is known as amplitude modulated signal.
Modulated carrier wave is given by.

$$e(t) = \underbrace{[E_{c\,max} + e_m(t)]}_{\text{Envelop of modulated wave}} \cdot \cos(2\pi f_c t + \phi_c)$$

where $E_{c\,max}$ = max amplitude of carrier wave
$e_m(t)$ = message signal
f_c = carrier frequency

Classification of AM -
1. Double side band supressed carrier (DSB-SC)
2. Double side band with carrier (AM)
3. Single side band (SSB)
4. Vestigial side band (VSB)

- Depth of modulation (m) -

Ratio of max. amplitude of message signal and carrier signal is known as depth of modulation (m).
The magnitude of 'm' can be measured directly from AM display itself

$$m = \frac{E_m}{E_c} \rightarrow (1)$$

Max. amplitude of modulated wave, $a = E_c + E_m \rightarrow (2)$
Min amplitude of modulated wave, $b = E_c - E_m \rightarrow (3)$

From eq (2) & (3),

$$E_c = \frac{a+b}{2}, \quad E_m = \frac{a-b}{2}$$

so,

$$\boxed{m = \frac{a-b}{a+b}}$$

- Significance of 'm'

$m < 1 \rightarrow$ under modulated wave (no zero crossing)
$m > 1 \rightarrow$ over modulated wave (not accepted)
$m = 1 \rightarrow$ perfect modulation

- Envelope detection -

At the receiver section, envelope detection, process is used for detecting the modulating signal.

This process is a perfect process for the double side band with carrier signal.

~~This process is a~~

In this process, we use rectifier and low pass filter. Rectifier allows the positive pattern and blocks the negative pattern. After rectifier, we remove the higher frequency component using low pass filter.

- **Synchronous detector** –
If there is double side band supressed carrier, then we use synchronous detection method at receiver end.

In this process, we use a local carrier which has same frequency of the original carrier frequency (f_c). Local carrier multiplies with signal and then we pass this signal from low pass filter.

The main disadvantage of this process is to match the f_c and the local carrier frequency.

- **Trapezoid method** –
It is used to calculate 'm' in the time domain. The scope is placed in XY mode –
X – modulating signal
Y – modulated signal

Modulation index is then calculated from the vertical edge length using

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

As modulation index (m) increases, the ratio between the vertical trapezoid edge increases.

PROCEDURE-

In this Online mode of practical, we perform these processes-

(1) After executing the AM analyzer simulation, click on the $\hat{S}$ in the AM modulator window

(2) For DSB with carrier, on the DC and for DSB with suppressed carrier off the DC

(3) For the different value of m observe the transmitted signal (oscilloscope and spectrum analyzer)

OBSERVATION-

1) Double sideband with carrier (DC offset = ON)

	m (modulation index)
m<1	0.5
	0.8
m=1	1
m>1	1.2
	1.5

2) Double side band suppressed carrier (DC offset = OFF).

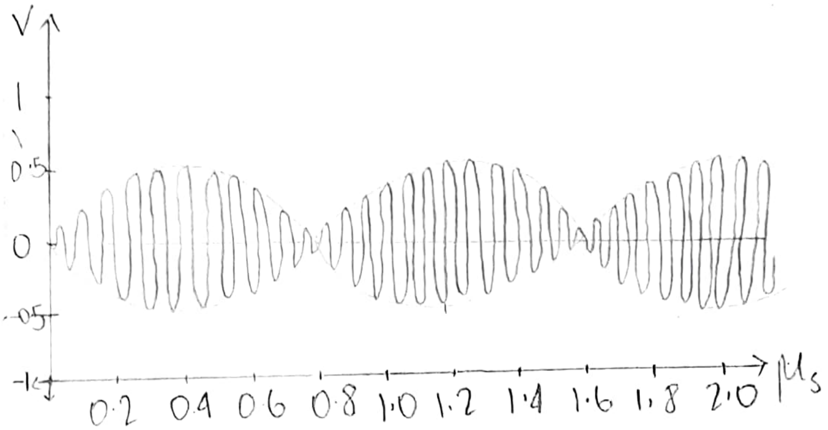
	m (modulation index)
m<1	0.5
	0.8
m=1	1
m>1	1.2
	1.5

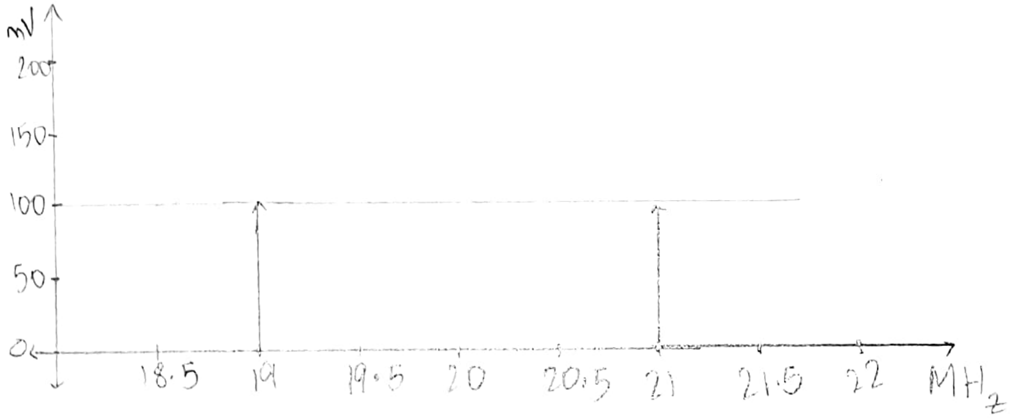
CONCLUSION-

Successfully observed and verified AM signals for double sideband with and without carrier by changing m as m>1, m=1, m<1 using LabAcive software

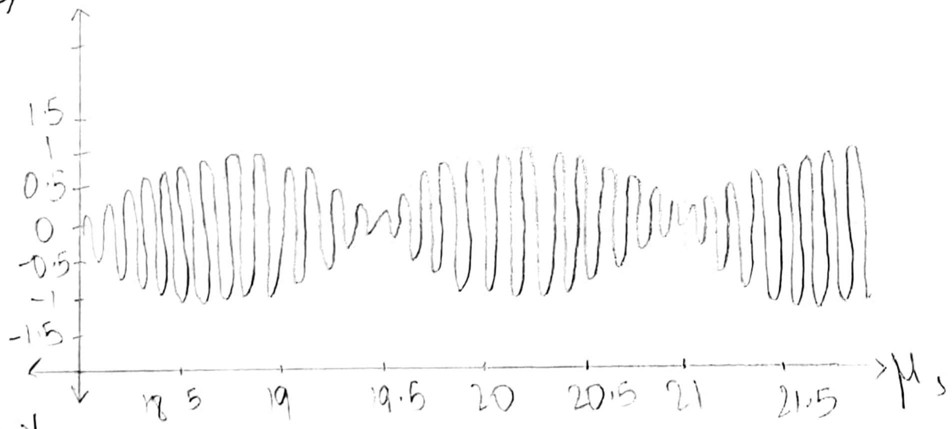
DOUBLE SIDEBAND SUPPRESSED CARRIER

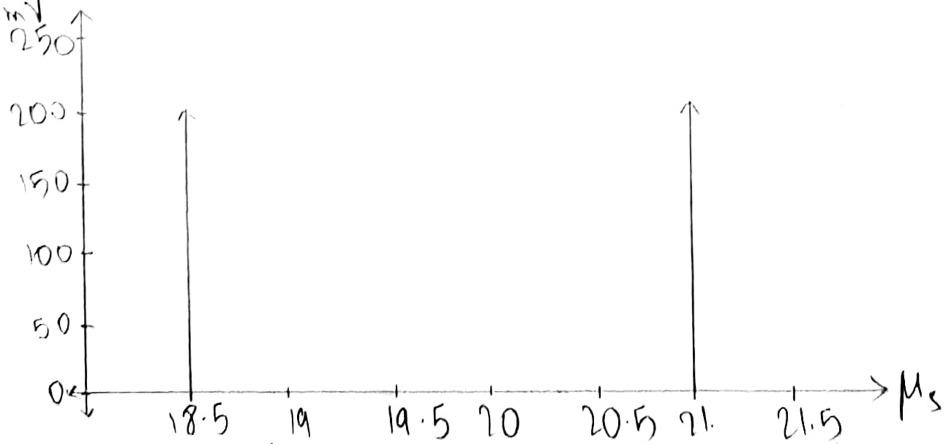
1) m=0.5



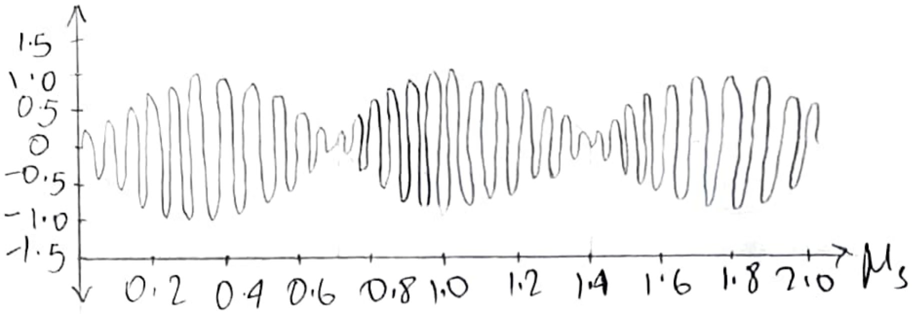


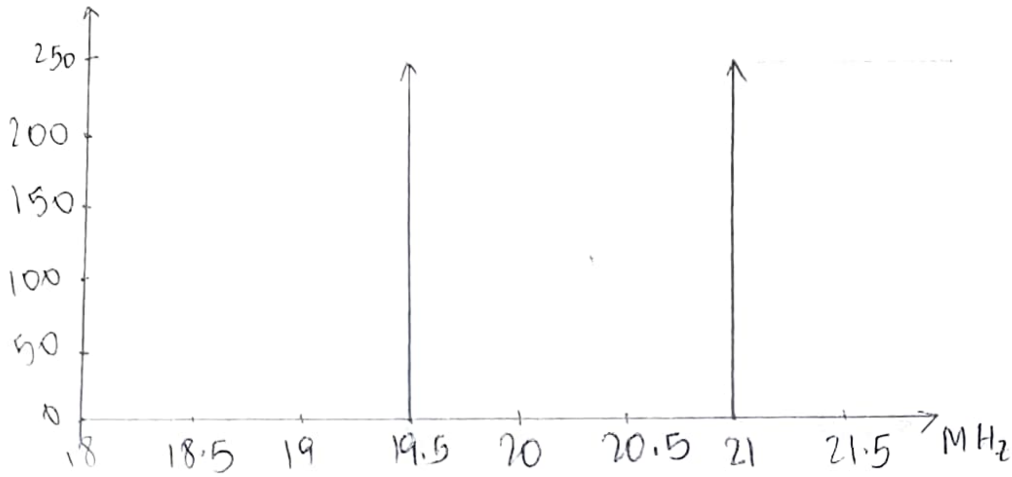
2) m=0.8



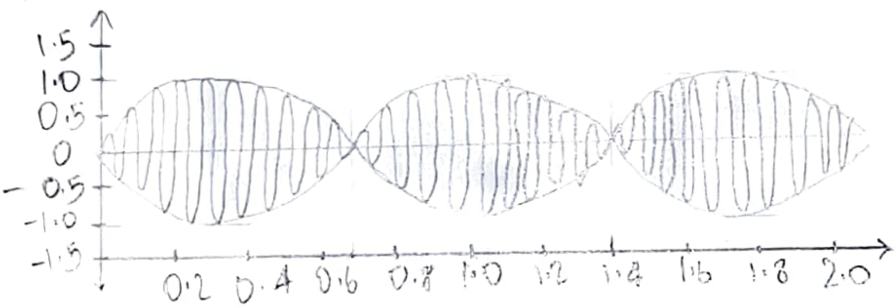


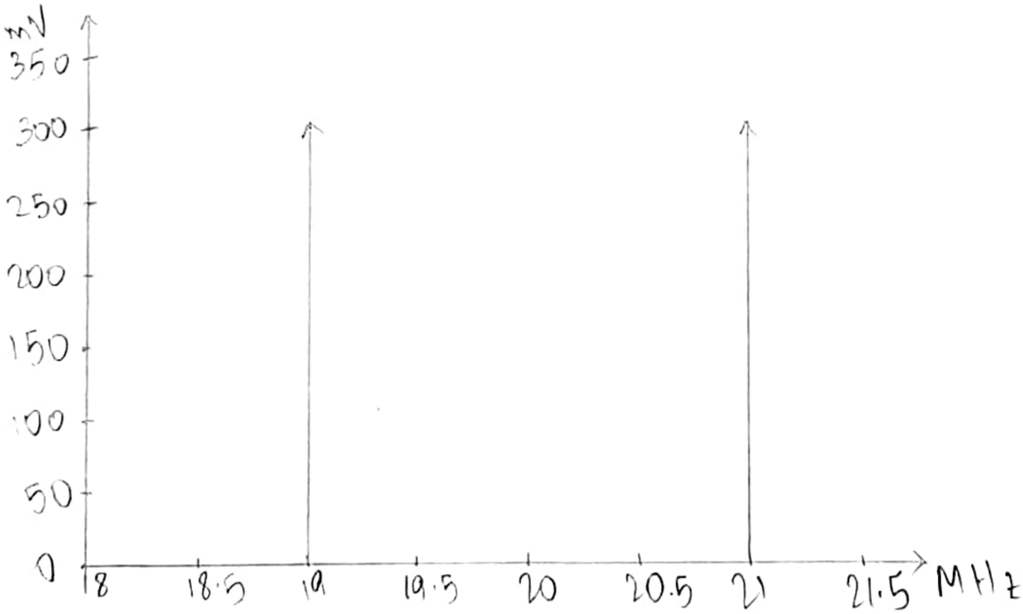
3) $m = 1$


1.5
1.0
0.5
0
-0.5
-1.0
-1.5
0.2 0.4 0.6 0.8 1.0 1.2 1.4 1.6 1.8 2.0
μs

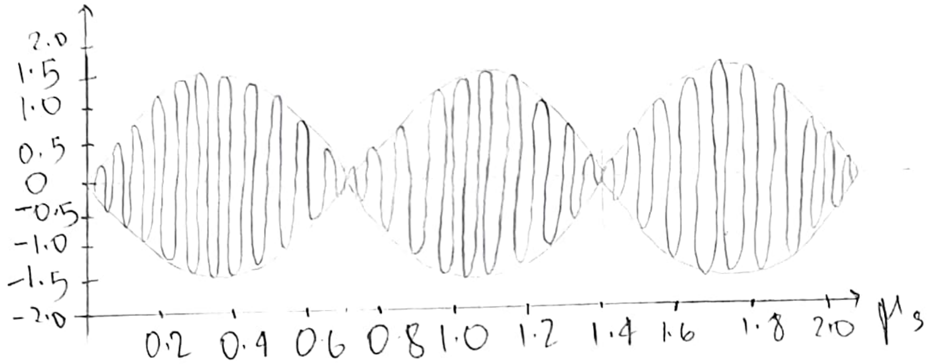

250
200
150
100
50
0
18 18.5 19 19.5 20 20.5 21 21.5
MHz

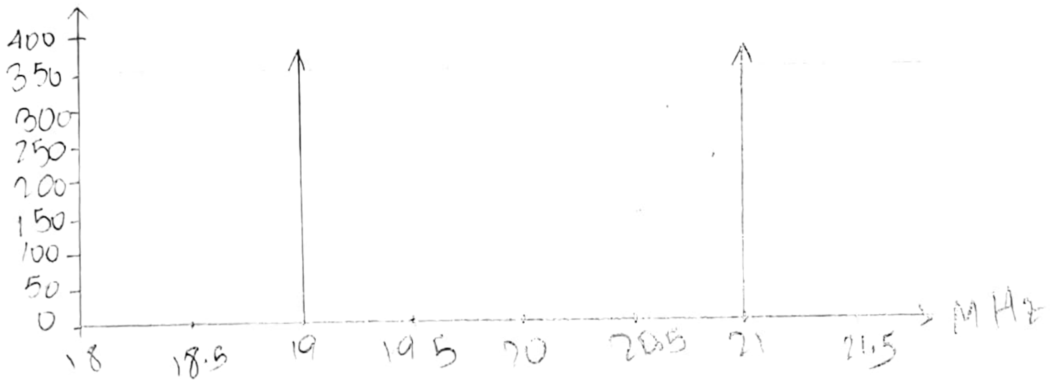
4) $m = 1.2$


1.5
1.0
0.5
0
-0.5
-1.0
-1.5
0.2 0.4 0.6 0.8 1.0 1.2 1.4 1.6 1.8 2.0


mV
350
300
250
200
150
100
50
0
18 18.5 19 19.5 20 20.5 21 21.5
MHz

5) $m = 1.5$


2.0
1.5
1.0
0.5
0
-0.5
-1.0
-1.5
-2.0
0.2
0.4
0.6
0.8
1.0
1.2
1.4
1.6
1.8
2.0
μs


400
350
300
250
200
150
100
50
0
18
18.5
19
19.5
20
20.5
21
21.5
MHz

EXPERIMENT-4

AIM-

To study frequency modulation (FM) and frequency demodulation with its applications

APPARATUS -

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

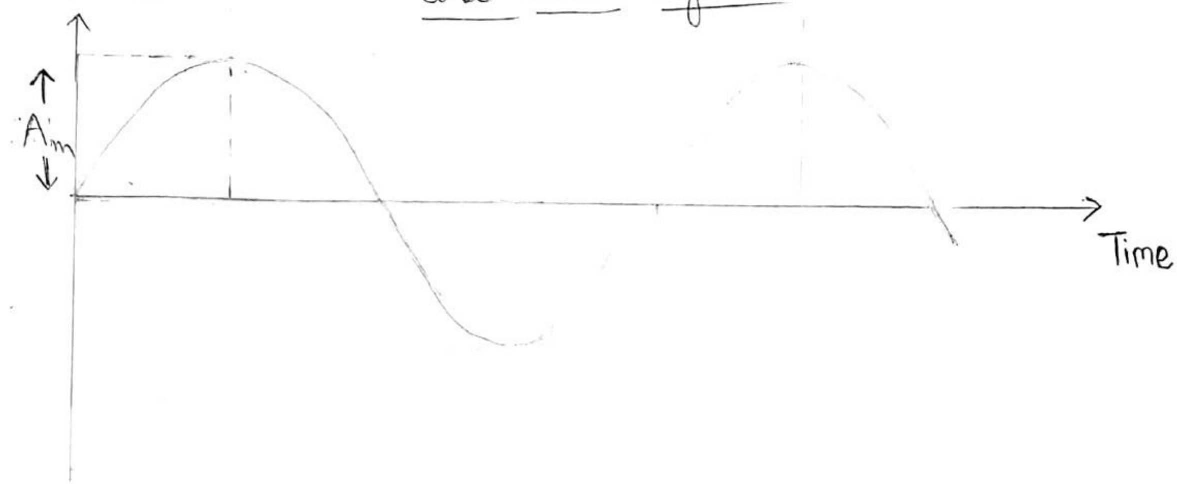
(3) Angle modulation is further divided into frequency and phase modulation

- Frequency modulation is the process of varying the frequency of the carrier signal linearly with message signal.
- Phase modulation is the process of varying the phase of carrier signal linearly with signal

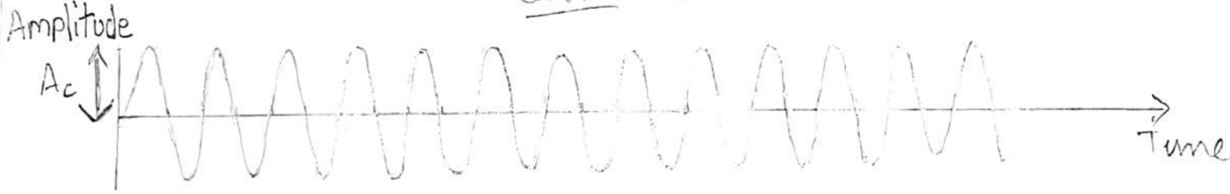
(4) The frequency of modulated wave increases, when the amplitude of the modulating or message signal increases. Similarly, the frequency of modulated wave decreases, when the amplitude of the modulating signal decreases.

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

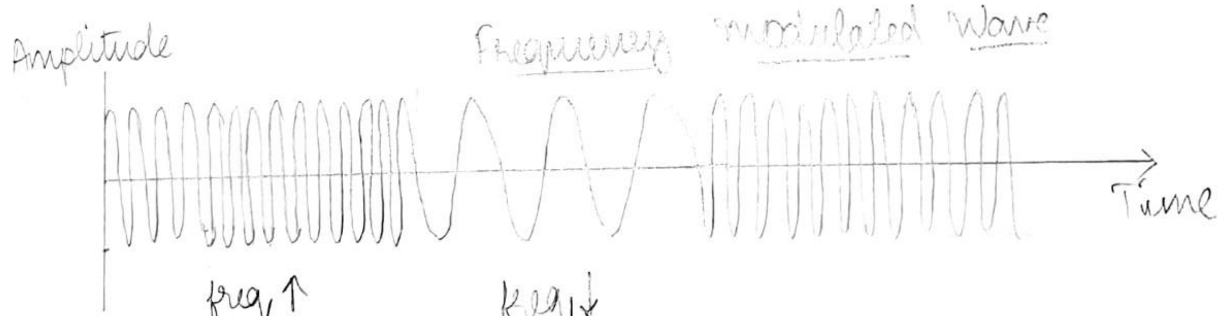
Base band Signal



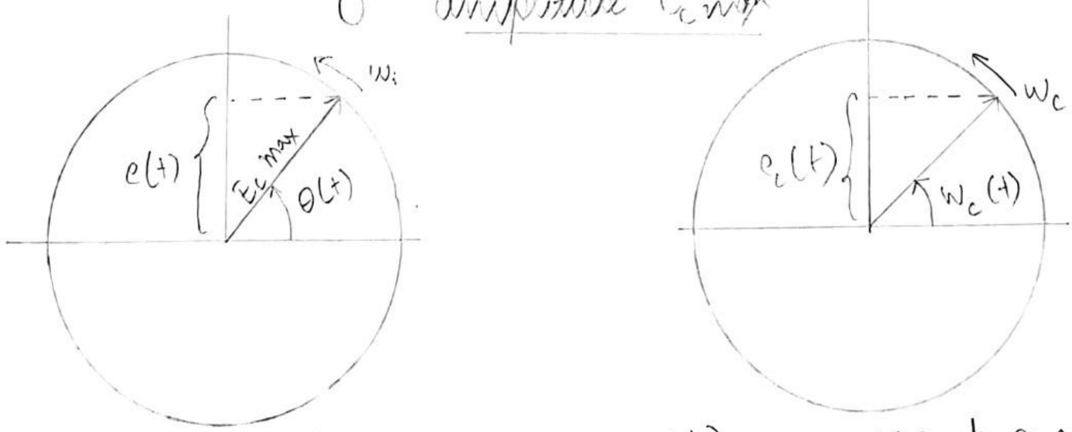
Carrier Wave



Frequency modulated Wave



Rotating phasor representation of carrier of amplitude $E_{c\,max}$



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

(b) at constant angular velocity (ω_c)

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

$$\boxed{f_i = f_c + (k_f)\, m(t)} \rightarrow (1)$$

carrier frequency — frequency sensitivity — message signal

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

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

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

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

Substitute f_i from eqn (1),

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

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

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

$$\boxed{s(t) = A_c \cos\left(2\pi f_c t + 2\pi k_f \int m(t) dt\right)} \rightarrow (4)$$

(Equation of FM wave)

(7) Finally, equation of FM wave

$$s(t) = A_c \cos\left(2\pi f_c t + 2\pi k_f \int m(t)\, dt\right) \rightarrow (4)$$

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

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

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

(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

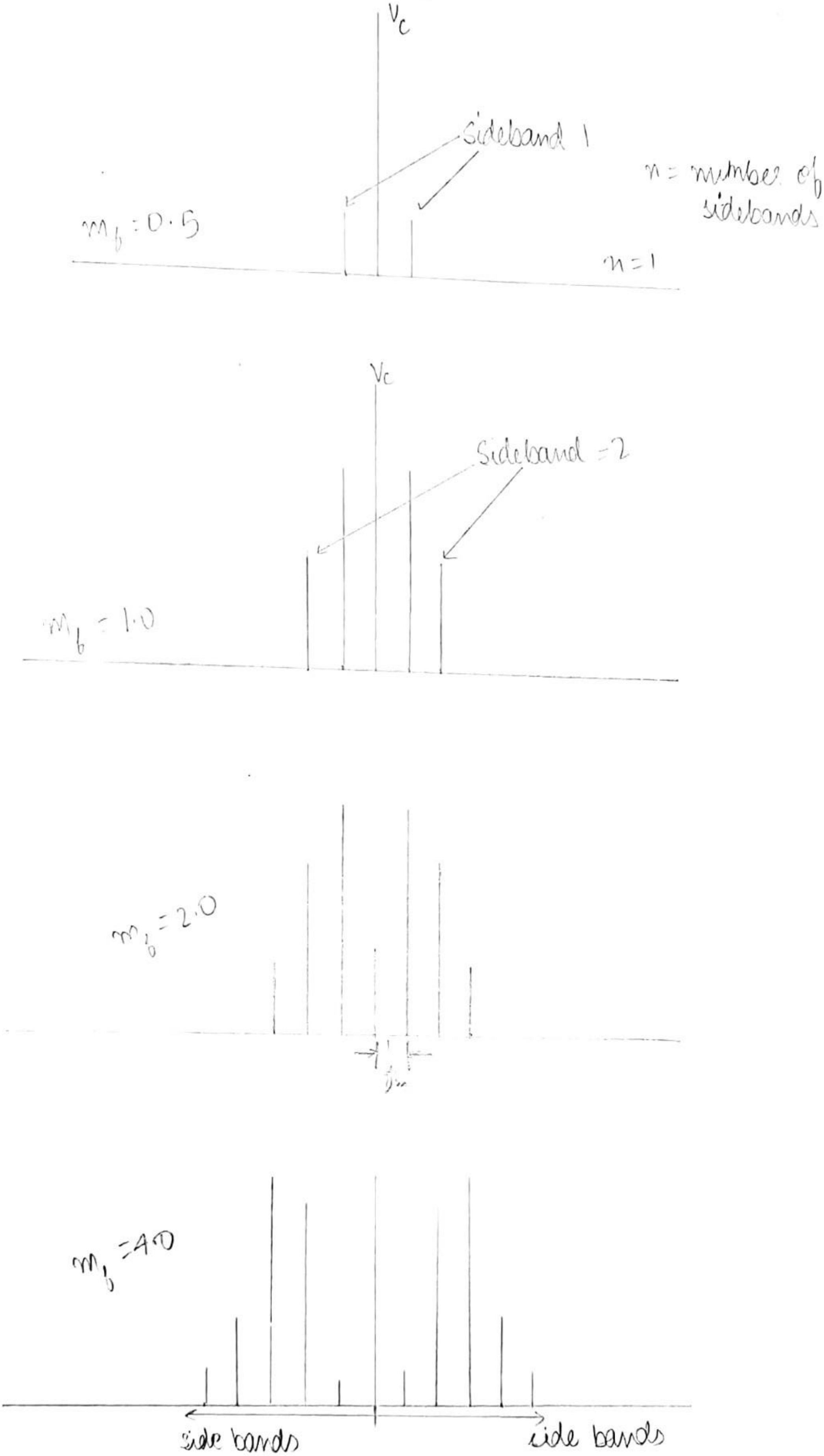
(9) FM can be divided into narrowband FM and wideband FM based on values of modulating index (β)

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

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

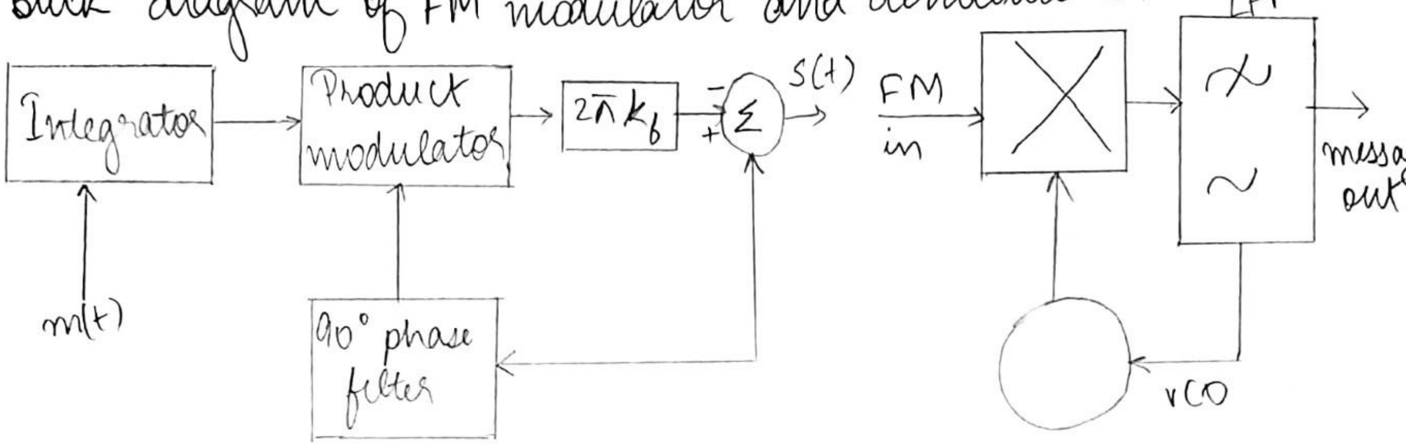
	MHz	f_d from carrier
f_c	100	0
f_{max}	105	+5 MHz
f_{min}	95	-5 MHz

(11) FM signal spectrum is quite complex and will have infinite number of sideband as shown in figure. This figure gives an idea, how the spectrum expands as the modulation index decreases.



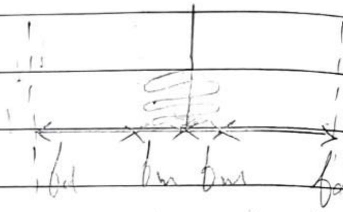
How spectrum FM varies with m_f

Block diagram of FM modulator and demodulator



Sidebands are separated from carrier by $f_c \pm f_m$, $f_c \pm 2f_m$, $f_c \pm 3f_m$ and so on

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



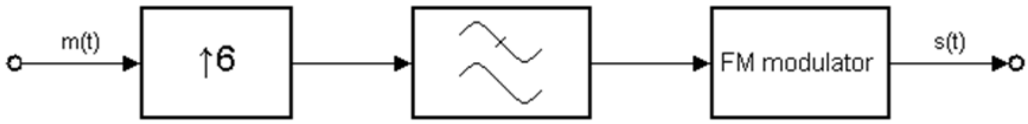
(12) In FM, carrier amplitude is constant,
∴ Transmitted power is constant & transmitted power does not depend on modulation index

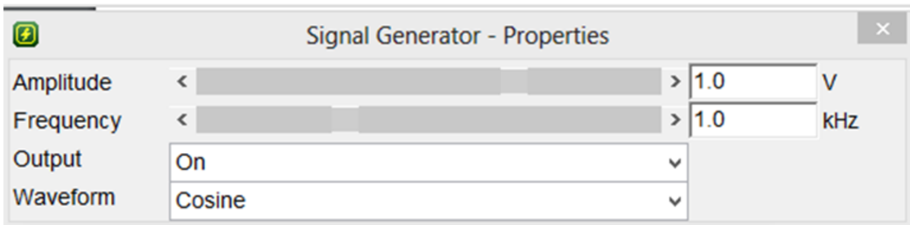
(13) FM has better noise immunity. FM is rugged/robust against noise. Therefore, the quality of FM will be good even in presence of noise.

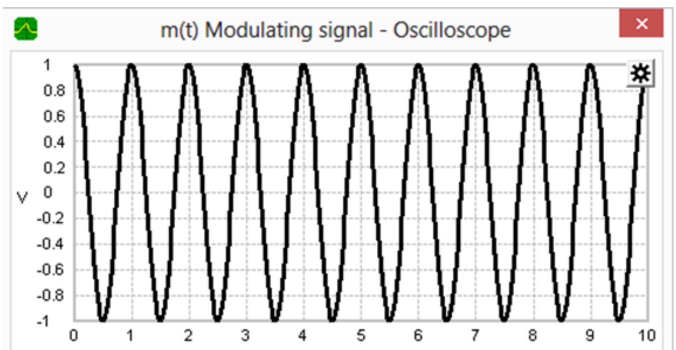
(14) Applications and advantages of FM –
(a) FM is resilient to noise and interference. Therefore, it is used for high quality broadcast transmission

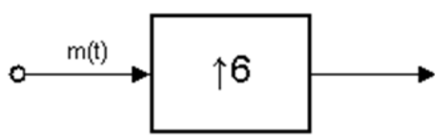
(b) FM is ideal for mobile radio transmission application including more general two way radio communication or portable applications where signal levels are likely to vary considerably

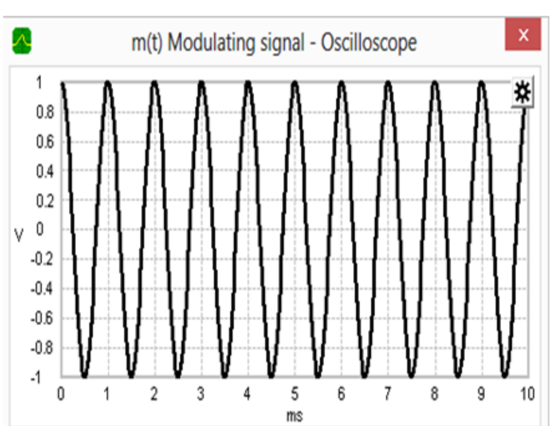
(c) Radar, telemetry, observing infants for seizure through EEG, music (magnetic tape record system) synthesis

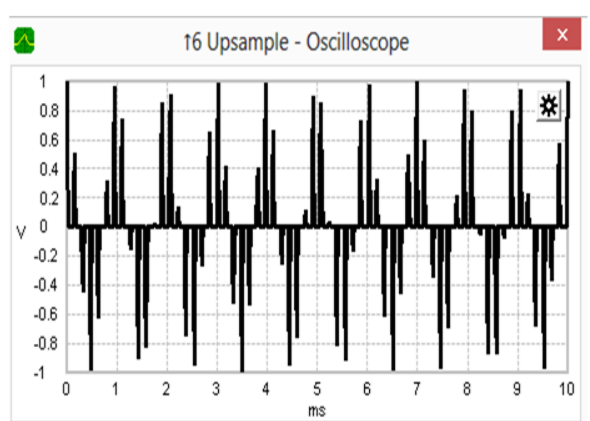

m(t)
↑6
FM modulator
s(t)

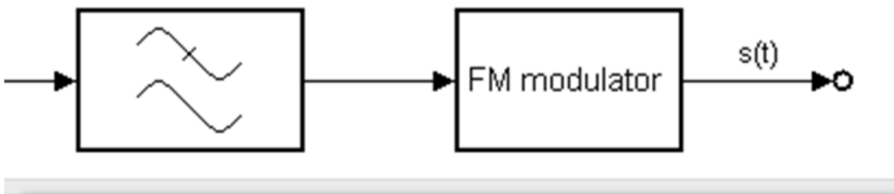

Signal Generator - Properties
Amplitude
1.0
V
Frequency
1.0
kHz
Output
On
Waveform
Cosine


m(t) Modulating signal - Oscilloscope
V
ms

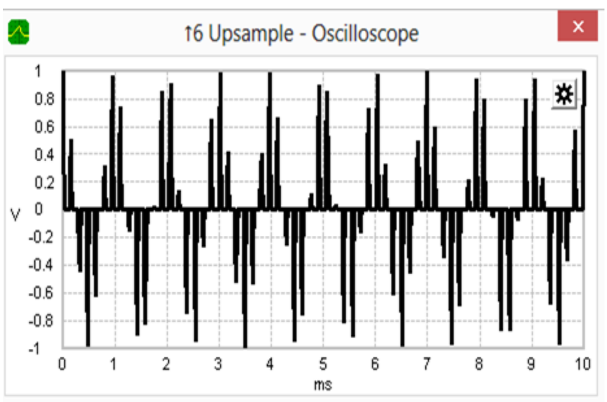

m(t)
↑6


m(t) Modulating signal - Oscilloscope
V
ms

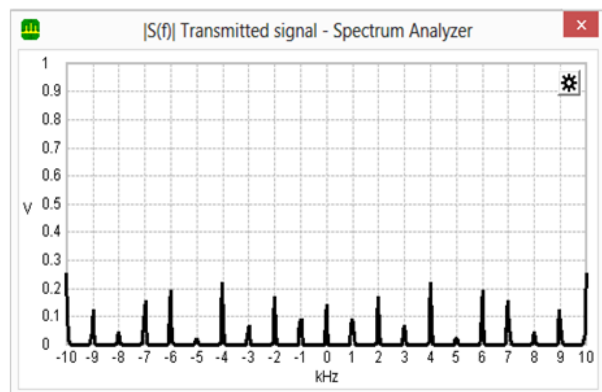

↑6 Upsample - Oscilloscope
V
ms



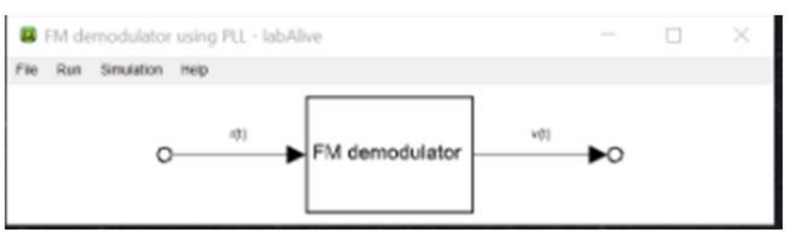
FM modulator
s(t)



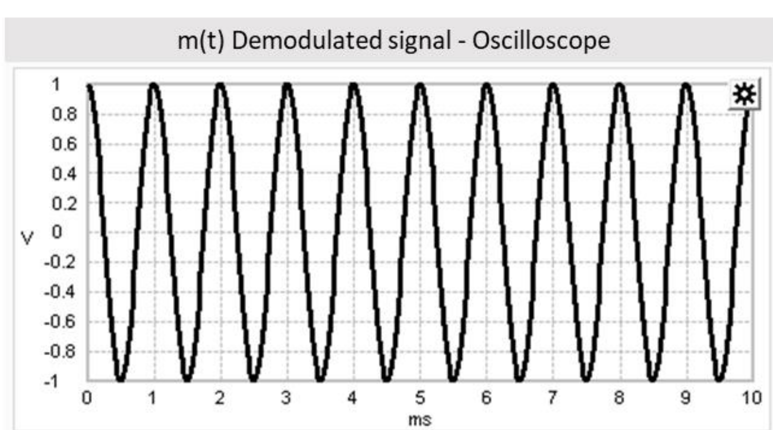
↑6 Upsample - Oscilloscope
V
ms



|S(f)| Transmitted signal - Spectrum Analyzer
V
kHz



FM demodulator using PLL - labAlive
File Run Simulation Help
FM demodulator



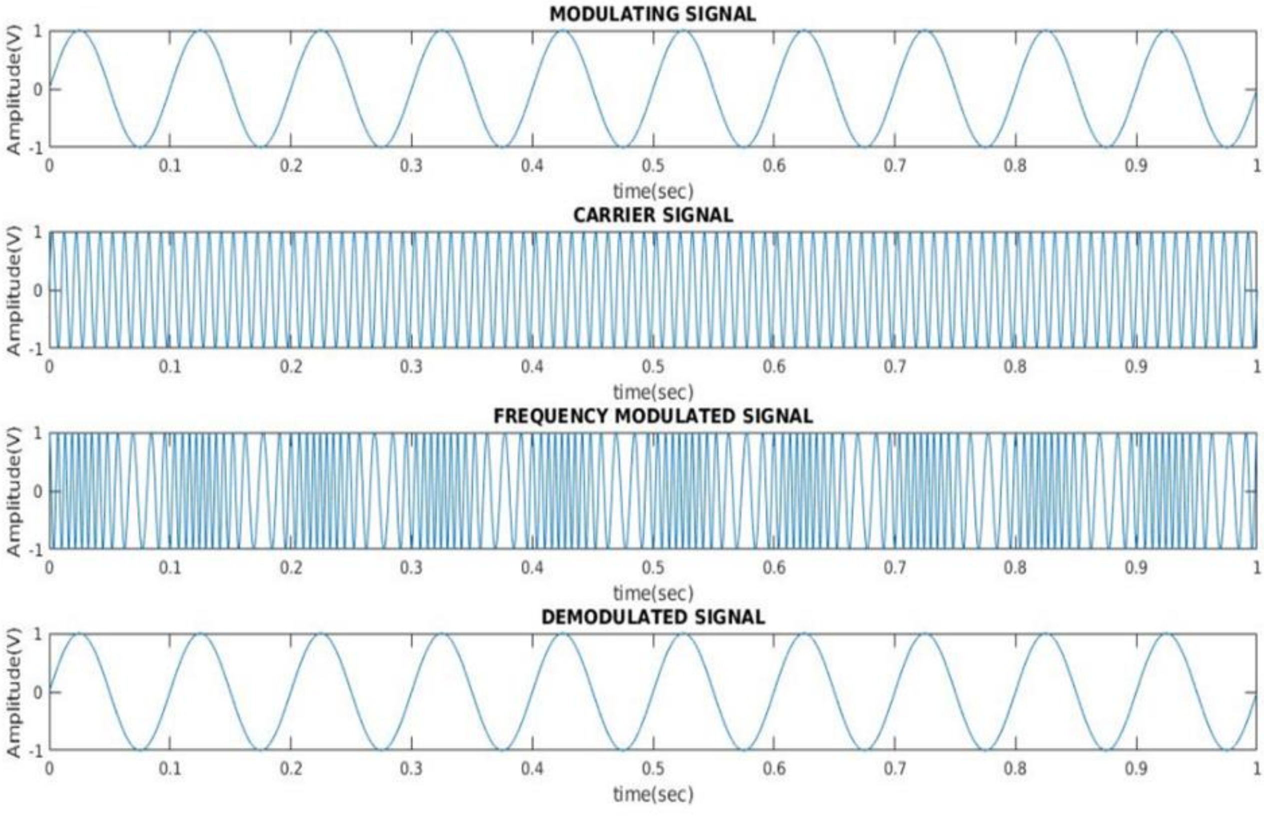
m(t) Demodulated signal - Oscilloscope
V
ms

MatLab Code

```
% Plot the frequency modulated signal
fc=100;
fm=5;
ts=1/(10*fc);
fs=(1/ts);
kf=2; % Frequency deviation sensitivity
wc=2*pi*fc;
t=0:ts:2;
m=sin(2*pi*fm*t);
y=cos(wc*t+(kf*2*pi*cumsum(m)).*ts);
figure(1)
subplot(211)
plot(t,m)
title('Input signal')
subplot(212)
plot(t,y)
title('FM modulation of input signal')
```

FM Modulation

```
% Plot the frequency response
mf=fftshift(fft(m))*ts;
delta=fs/length(mf);
f=-fs/2:delta:fs/2-delta;
figure(2)
subplot(211)
plot(f,abs(mf))
title('Magnitude spectrum of input signal')
a=fftshift(fft(y))*ts;
delta=fs/length(a);
f=-fs/2:delta:fs/2-delta;
subplot(212)
plot(f,abs(a))
title('Magnitude spectrum of the fm')
```


MODULATING SIGNAL
CARRIER SIGNAL
FREQUENCY MODULATED SIGNAL
DEMODULATED SIGNAL
Amplitude(V)
time(sec)
1
0
-1
0
0.1
0.2
0.3
0.4
0.5
0.6
0.7
0.8
0.9
1

EXPERIMENT-5
PULSE MODULATION

AIM-
To examine pulse amplitude modulation (PAM), pulse position modulation (PPM) and pulse width modulation (PWM) and verify and draw the resultant waveforms. Illustrate the circuit diagram for PAM and PWM. Show and draw output waveform using MATLAB code using virtual mode.

SOFTWARE- MATLAB

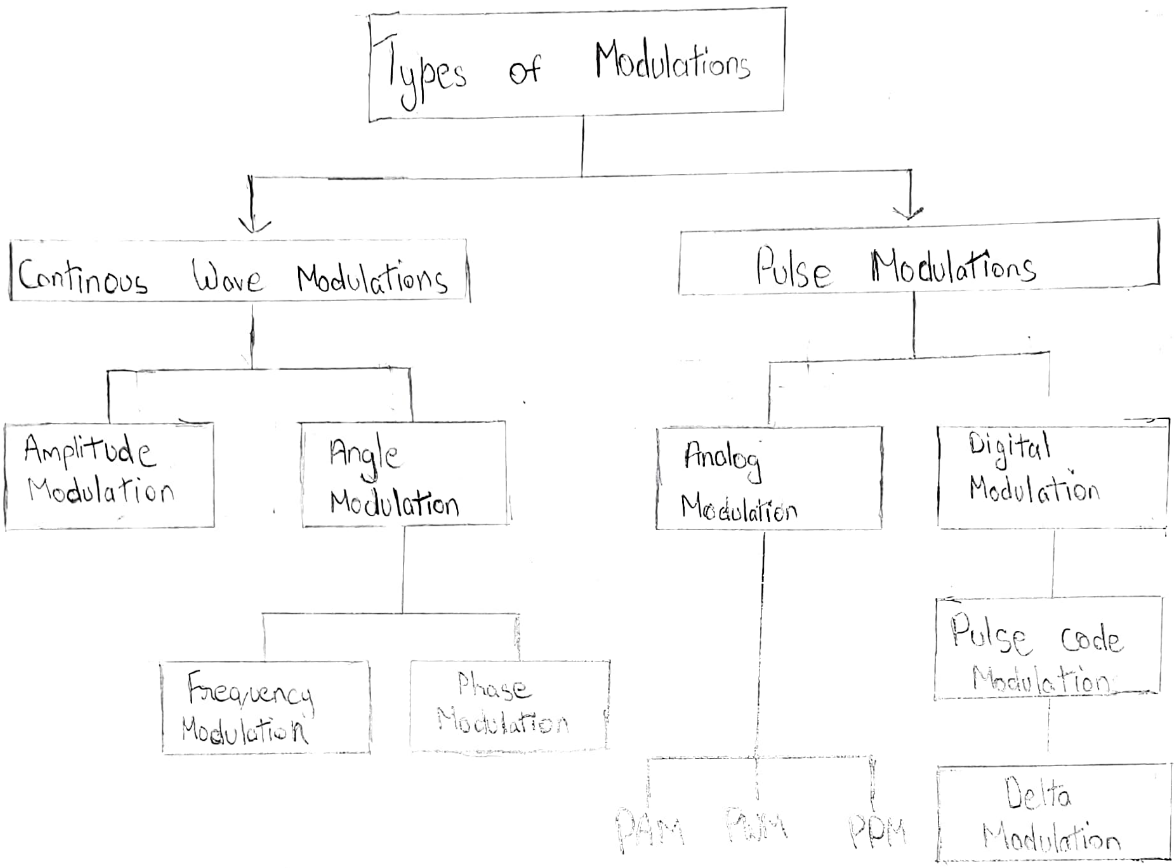
THEORY-
Pulse modulation is a type of modulation in which the signal is transmitted in the form of pulses. In pulse modulation, continous signals are sampled at regular intervals

Pulse Modulation is further divided into analog and digital modulation and further analog and digital modulation is subdivided in PAM, PWM, PPM (analog) and PCM, DM (digital)

Analog part consists:
- Pulse amplitude modulation (PAM)
- Pulse width modulation (PWM)
- Pulse position modulation (PPM)

Pulse Amplitude Modulation (PAM)-
In pulse amplitude modulation (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 out the path of the whole wave

Block Diagram Showing Basic Classification



PAM Signal Generation-

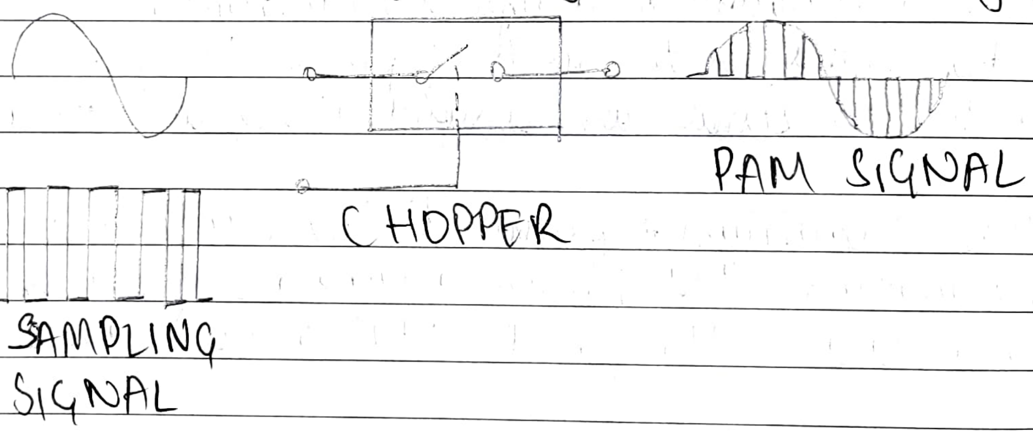
We can generate PAM signal by two types of sampling possible

(a) Natural sampling (b) Flat-top samping

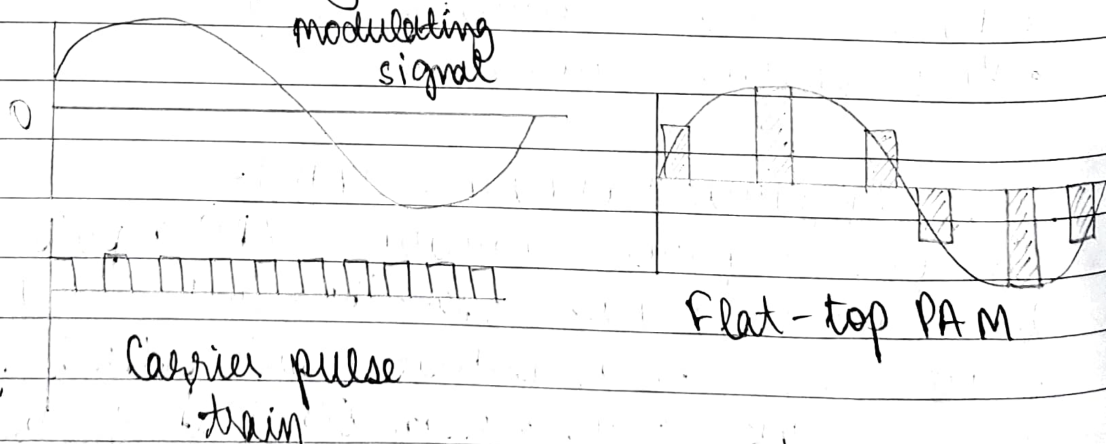
(a) Natural Sampling-

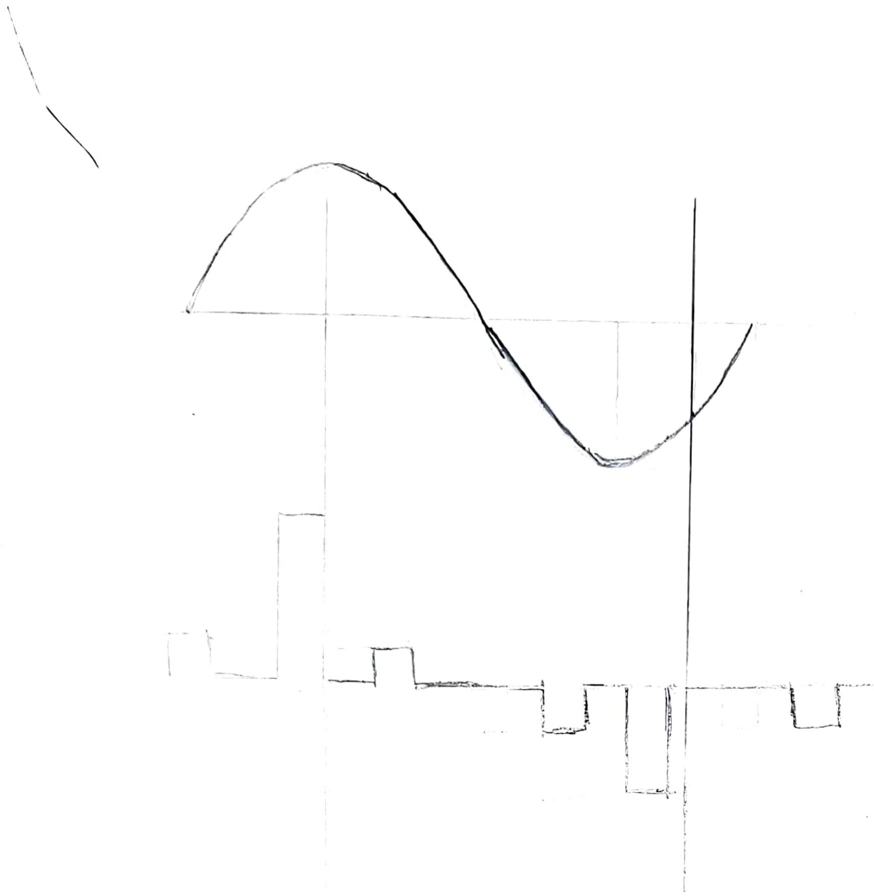
For a PAM signal produced with natural sampling, the sampled signal follows the waveform of the input signal during the time that each sample is taken

Generation of PAM signal by natural sampling

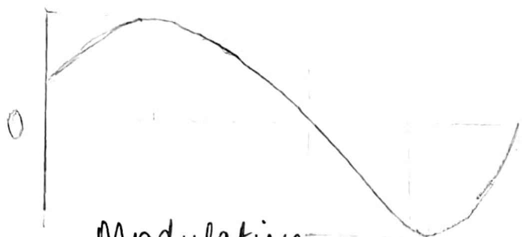


Flat-top sampling

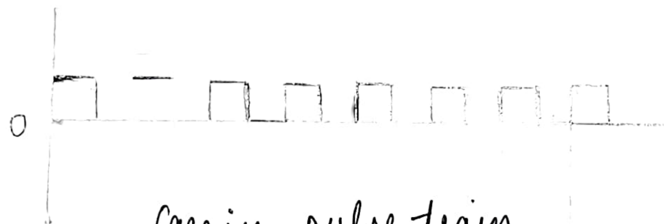




Natural Sampling:



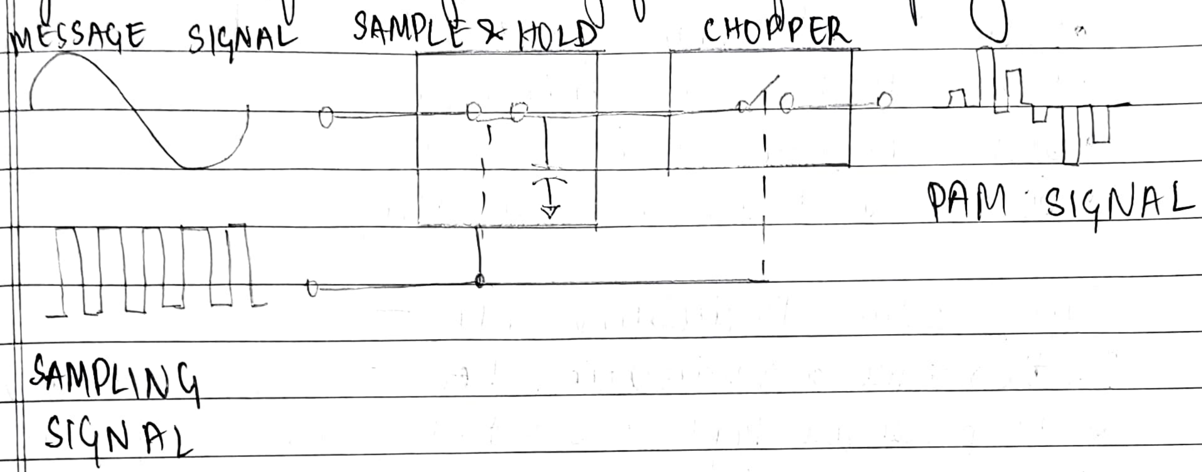
Modulating signal



Carrier pulse train

Natural PAM

Generation of PAM signal by flat-top sampling



A sample-and-hold circuit is used to hold the amplitude of each pulse at a constant level

In PAM -

- Amplitude of the pulse proportional to amplitude of modulating signal
- Bandwidth of the transmission channel depends on the pulse width
- Instantaneous power of the transmitter varies
- Noise interference is high
- System is complex to implement
- Similar to amplitude modulation

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 the analog signal.

Properties of PDM/PWM -

- Width of the pulse is proportional to amplitude of modulating signal.
- Bandwidth of the transmission channel depends on the rise

time of the pulse
- Instantaneous power of the transmitter varies
- Noise interference is minimum
- System is simple to implement
- Similar to frequency modulation

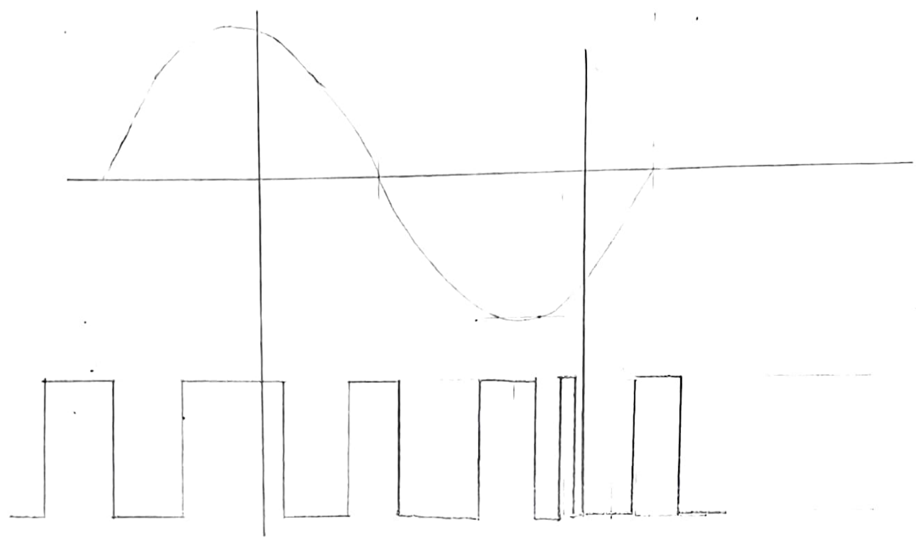
Pulse Postion Modulation (PPM) —

In this type 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

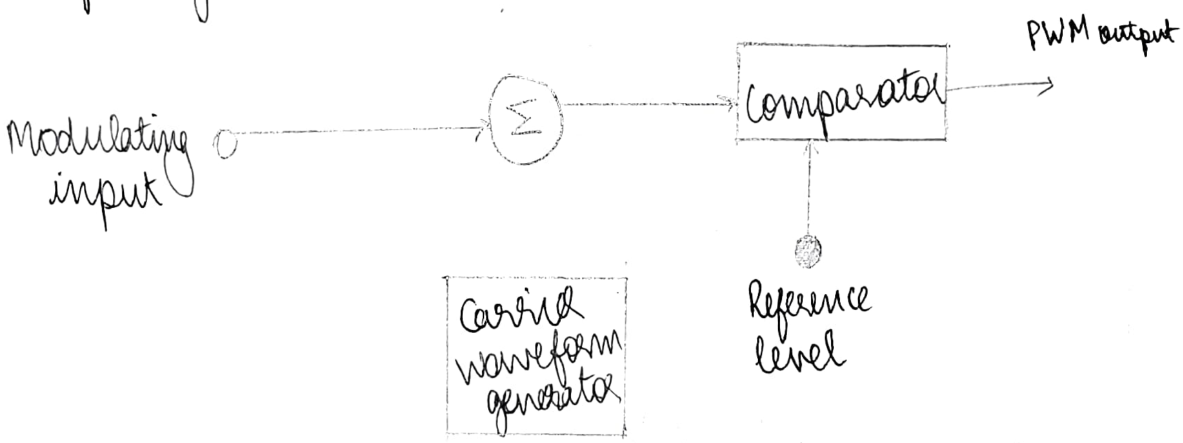
In PPM (Properties) —
- The relative position of the pulse is proportional to amplitude of modulating signal
- Bandwidth of the transmission channel depends on the rising time of the pulse.
- Instantaneous power of the transmitter remains constant
- Noise interference is minimum
- System is simple to implement
- Similar to phase modulation

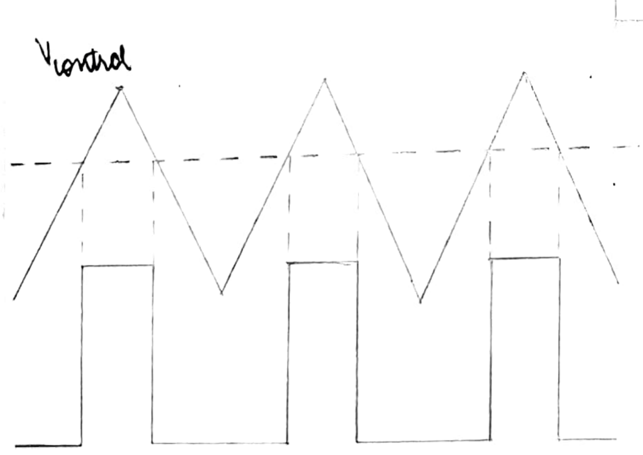
% PAM using Ideal sampling

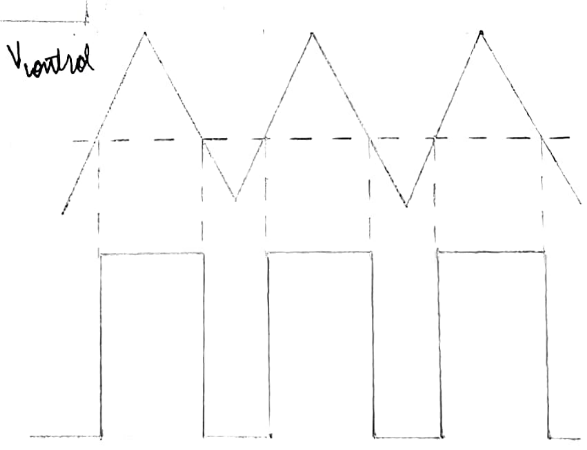
```
clc;
close all;
clear all.
a = input('Enter the amplitude = ');
f = input('Enter the frequency = ');
t = 0:0.02:2;
```



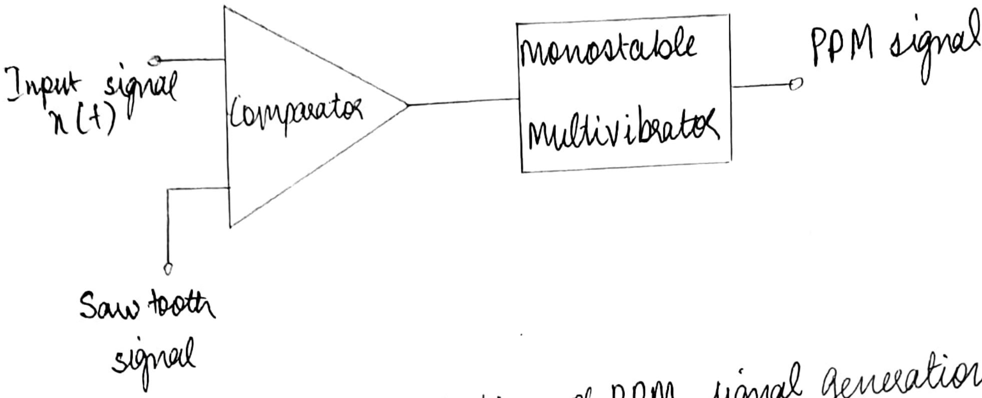
PWM signal generation



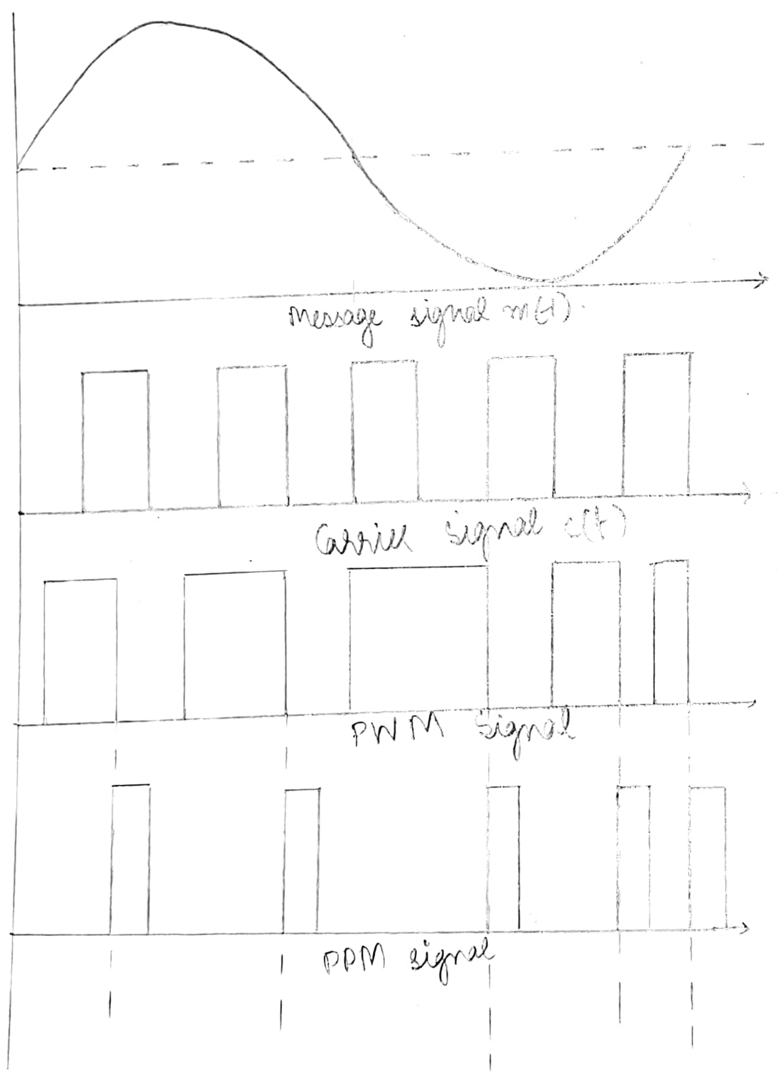




Generation of PPM signal



Waveform representation of PPM signal generation



```
x1 = 1 % generation of an impulse signal
x2 = a* sin (2*pi * f * t); % generation of sinewave
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 wave plot
stem (t,y);
title ('PAM Wave');
xlabel ('Time');
ylabel ('amplitude');
```

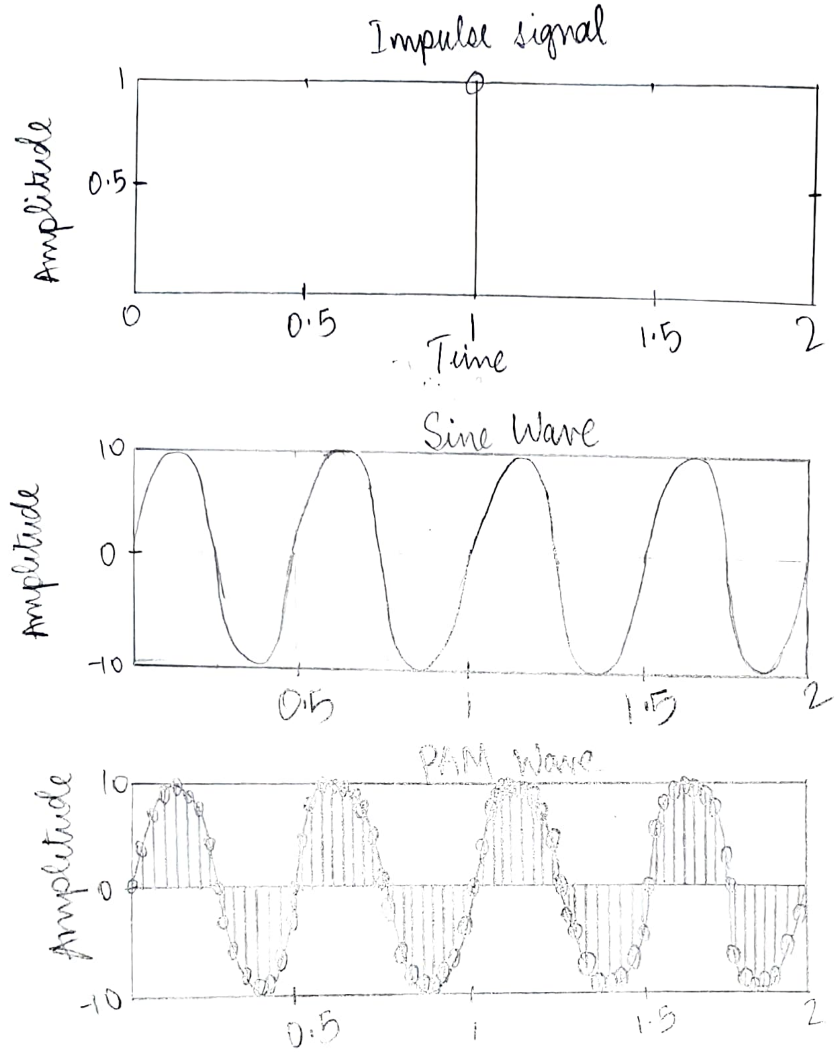
% PAM using natural sampling

```
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;
```

Amplitude = 10

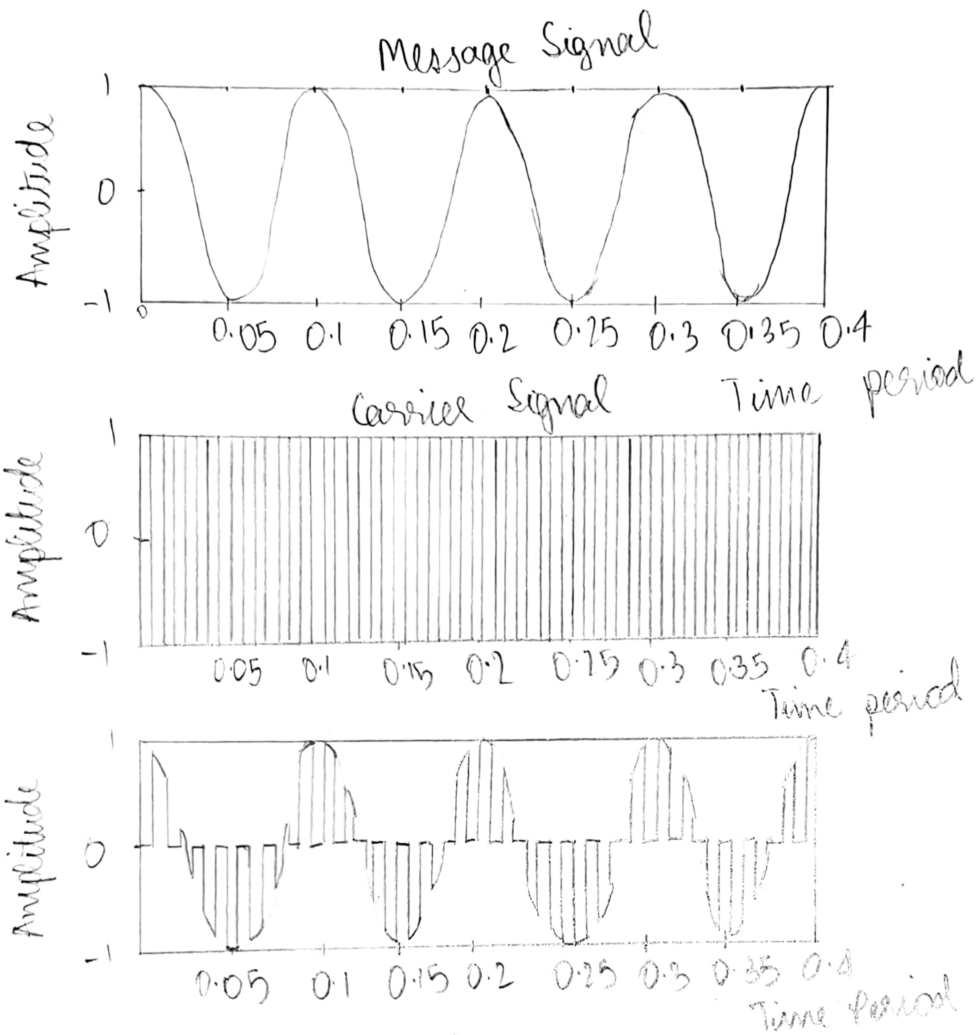
Frequency = 2

% PAM Ideal Sampling



```
tt = [];
for i=1: length
    if Mod_sgl (i) == 0;
        tt= [tt, Mod_sgl (i)];
    else
        tt= [tt, Mod_sgl(i)+2];
    end
end
figure (1)
subplot (4,1,1)
plot (t, Msg_sgl);
title ('Message signal');
xlabel ('Time period');
ylabel ('Amplitude');
subplot (4,1,2);
plot (t, carr_sgl);
title ('carrier signal');
xlabel ('Time period');
ylabel ('Amplitude');

subplot (4,1,3);
plot (t, Mod_sgl);
title ('PAM modulated signal')
xlabel ('Time period');
ylabel ('Amplitude');
% subplot (4,1,4);
% plot (t, tt);
```

% PAM Natural Sampling



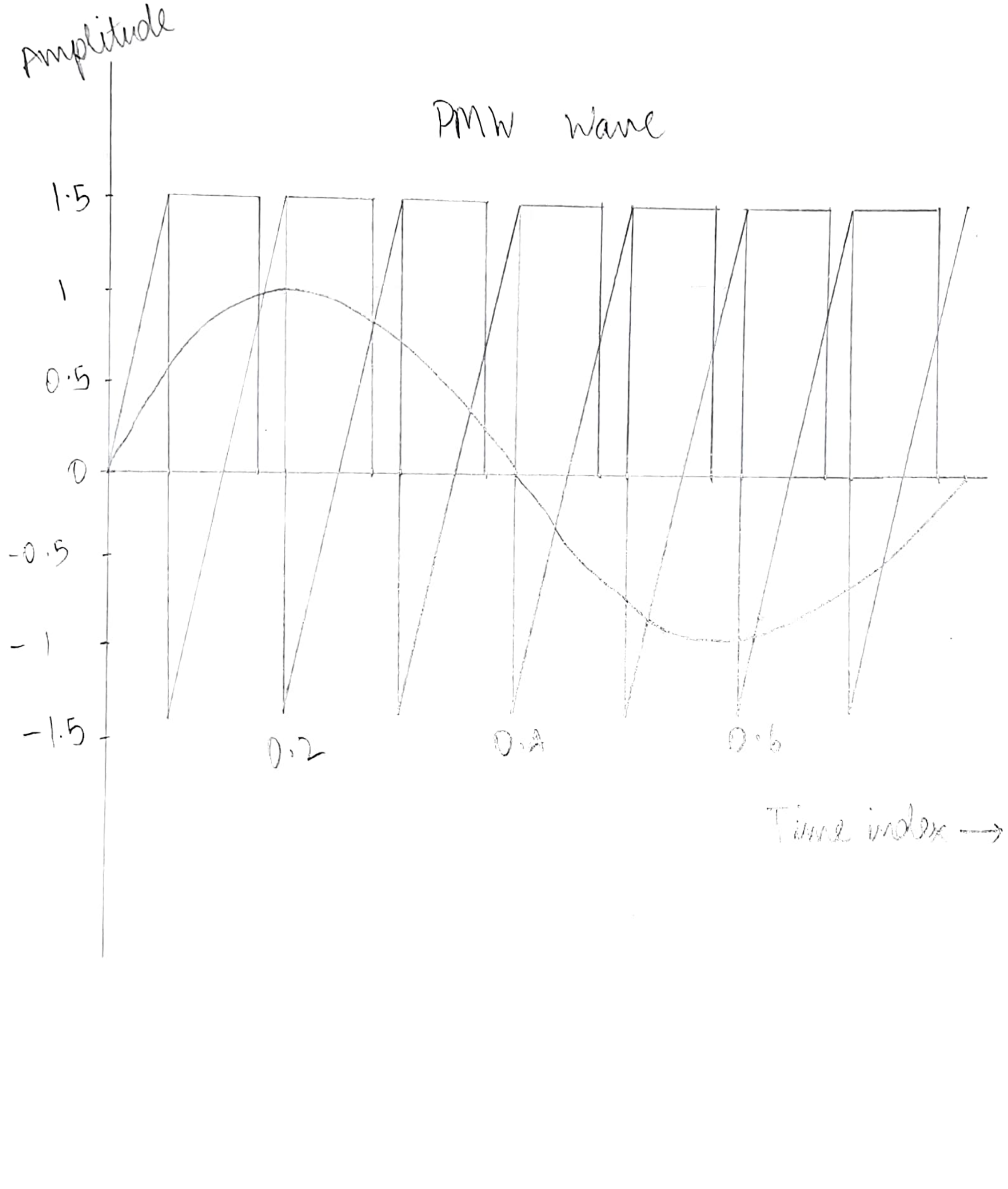
PAM Modulated Signal

%PWM signal

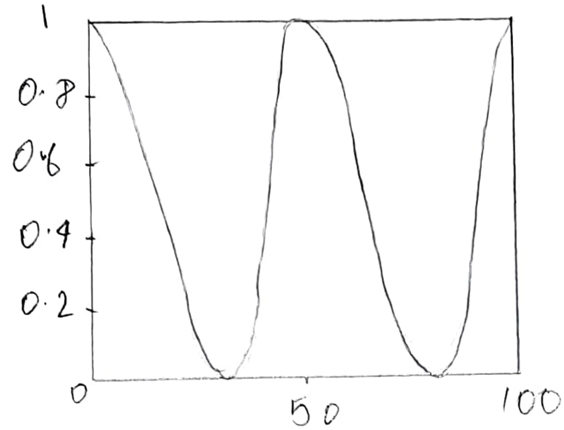
```
clc;
close all;
clear all;
t=0:0.0001:1;
s=sawtooth(2*pi*10*t+pi);
m=0.75*sin(2*pi*1*t);
n=length(s);
for i=1:n
    if (m(i)>=s(i))
        pwm(i)=1;
    else if (m(i)<=s(i))
        pwm(i)=0;
        end
end
plot(t,pwm,'g',t,m,'r',t,s,'b');
ylabel('Amplitude');
axis([0,1-1.5 1.5]);
xlabel('Time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

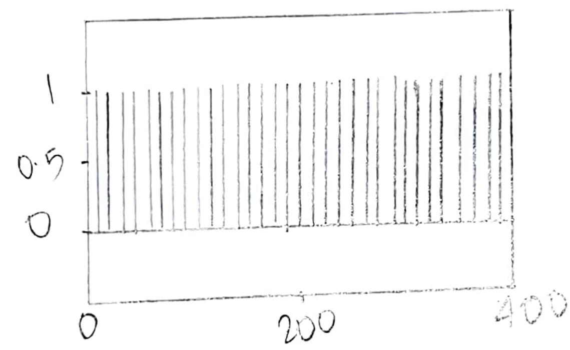
% PPM signal

```
clc;
close all;
clear all;
fc=1000;
fs=10000;
fm=200;
t=0:1/fs:((2/fm)-(1/fs));
```


Amplitude
PMW wave
1.5
1
0.5
0
-0.5
-1
-1.5
0.2
0.4
0.6
Time index →

PPM signal





```
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 ([0 500- 0.2 1.2]);
title ('PPM');
```

Advantages of Pulse Modulation –

(a) The pulses are quite short as compared to the time in between so a pulse modulated wave remains off most of the time

(b) The time interval between pulses may be filled with sample values from other messages, so we can send many messages at a time on a pulse communication system

(c) One of the chief advantages of PM is that if we combine pulse modulation with continuous modulation (AM, FM, PM) we can obtain "multichannel" communication system, a desirable feature feature for data-transmission

CONCLUSION –

Successfully performed PAM PWM, PPM and analysed and verified its input and output waveforms using MATLAB software.

EXPERIMENT-6

STUDY ASK, FSK AND PSK MODULATION

AIM-
To study the amplitude shift keying (ASK), frequency shift keying (FSK) and phase shift keying (PSK) modulation technique and verify waveform

SOFTWARE - MATLAB/Simulink

THEORY-
In case of amplitude shift keying the amplitude of the resultant output (modulated) depend upon the input data. This is also a type of amplitude modulation which represent the binary data in form of the variation in amplitude of signal.

ASK is a digital modulation technique defined as the process of shifting the amplitude of the carrier signal between two levels, depending on whether 1 or 0 is to be transmitted. Let the message signal be binary sequence of 1's and 0's. It can be represented as function of time as follows

$$V_m = \begin{cases} V_m & \text{(when symbol is 1)} \\ 0 & \text{(when symbol is 0)} \end{cases}$$

Let the carrier be defined as

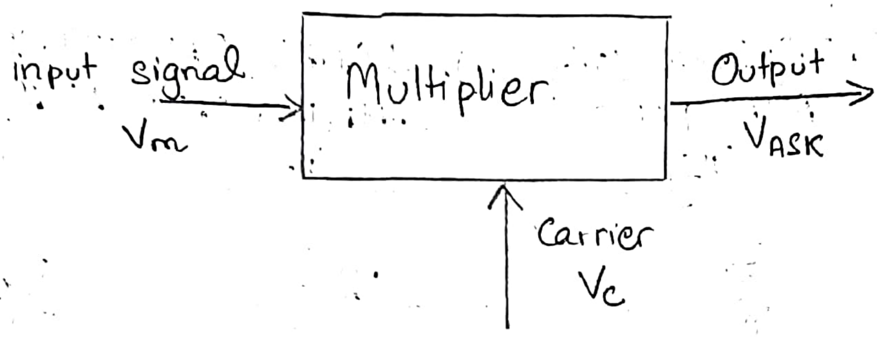
$$V_c = V_c \cos w_c t$$

then corresponding ASK signal is given by product of V_m & V_c as

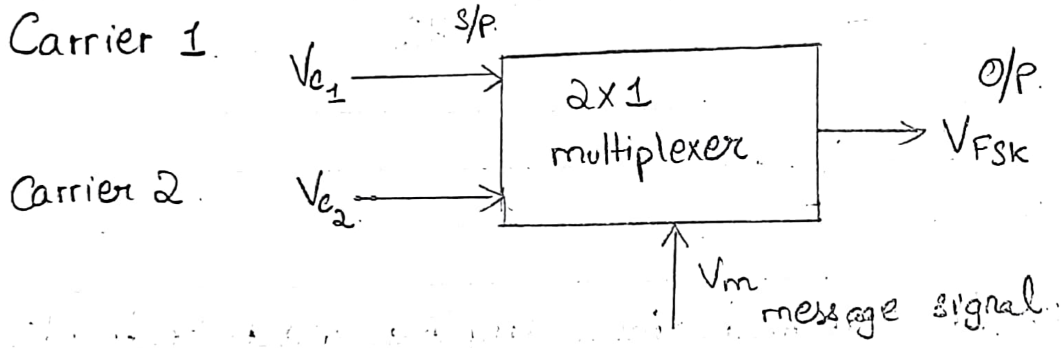
$$V_{ASK} = V_m V_c \cos w_c t \quad \text{when symbol is 1}$$
$$= 0 \quad \text{when symbol is 0}$$

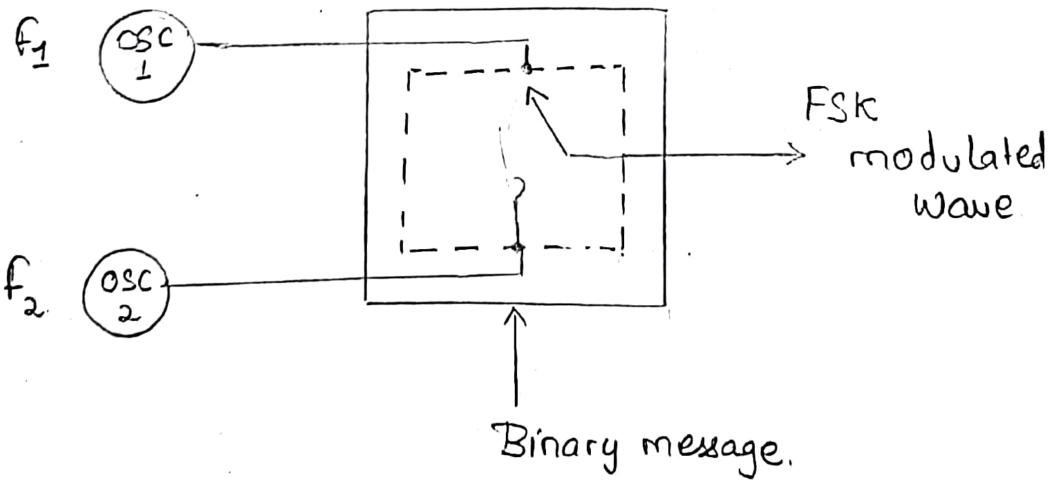
BLOCK DIAGRAM

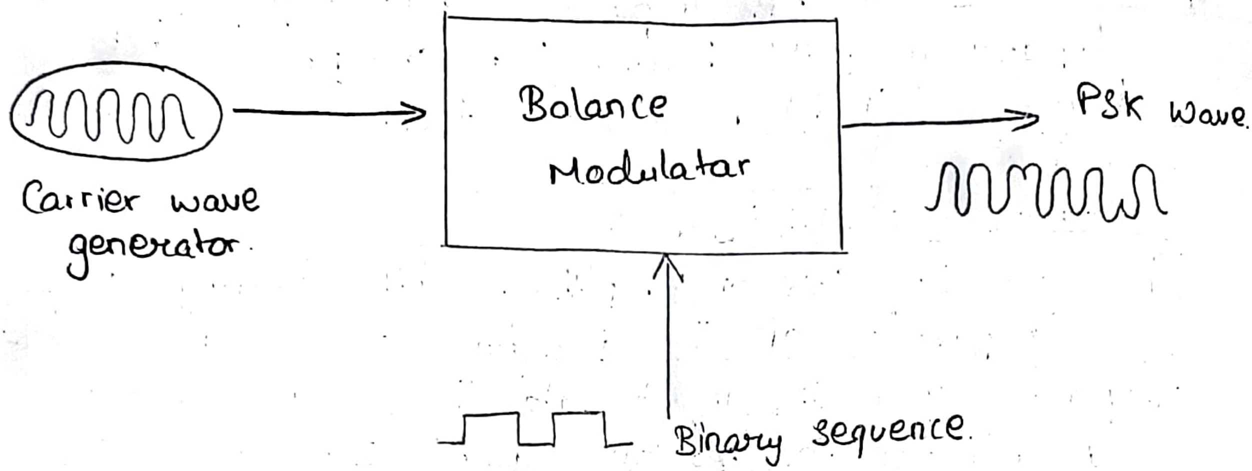
1) Block diagram of generation of ASK signal.



2) Block diagram of FSK Generator.







Block Diagram showing generation of PSK.

FSK

(ii) In case of frequency shift keying output signal will be either high or low, depending upon the input data applied. FSK is the digital modulation technique in which the frequency of the carrier signal varies according to discrete digital changes. FSK is a scheme of frequency modulation

Let V_m be the message signal

$$V_m(t) = V_m$$

Let the two carriers be defined as

$$V_{c_1} = V_c \cos \omega_{c_1} t$$

$$V_{c_2} = V_c \cos \omega_{c_2} t$$

then corresponding FSK signal is defined as

$$V_{FSK} = V_m V_c \cos \omega_{c_1} t \quad \text{when symbol is 1}$$

$$= V_m V_c \cos \omega_{c_2} t \quad \text{when symbol is 0}$$

(iii) Phase shift key (PSK) is the digital modulation technique in which the phase of carrier signal is changed by varying the sine and cosine input at a time. The phase of output signal get shifted depending upon the input

MATLAB code for ASK

```
clc;
clear all;
close all;
fc = input('Enter the value of carrier frequency');
fp = input('Enter the value of frequency for binary message sig
```

```
amp = input('Enter the amplitude for Am & Ac');
t = 0 : 0.001 : 1
amp = amp/2
m = amp * square (2 * pi * fp * t) + amp;
c = amp * sin (2 * pi * fc * t);
ask = c * m;
subplot (3, 1, 1)
plot (t, m);
xlabel ('time');
ylabel ('Amplitude');
subtitle('message signal');
subplot (3, 1, 2);
plot (t,c);
xlabel ('time');
ylabel('Amplitude');
title ('carrier');
subplot (3,1,3);
plot (t, ask);
xlabel (Time);
ylabel ('amplitude');
title ('ASK modulated signal');
```

MATLAB code for FSK

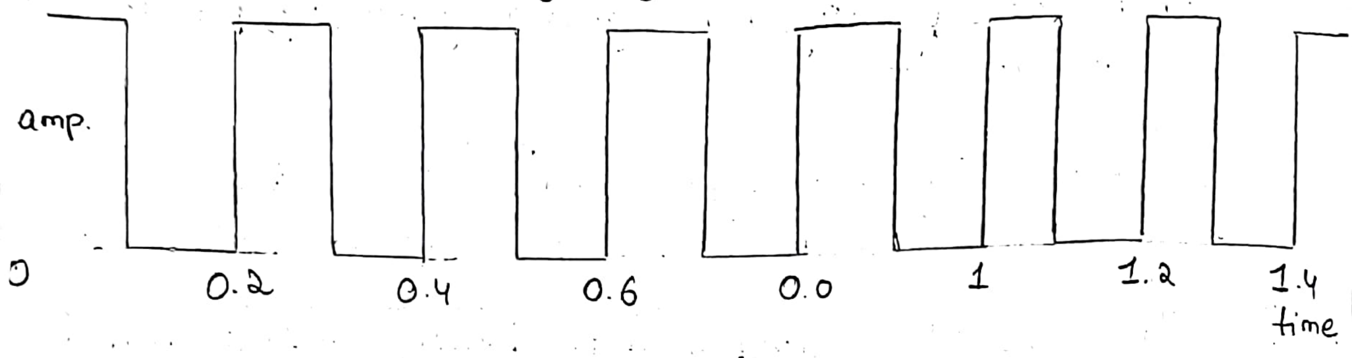
```
clc;
clear all;
close all;
fc1 = input ('Enter the frequency for 1st carrier signal');
fc2 = input ('Enter the frequency for 2nd carrier signal');
fp = input ('Enter the frequency for binary message signal');
amp = input ('Enter the amplitude for message & carrier');
```

1)

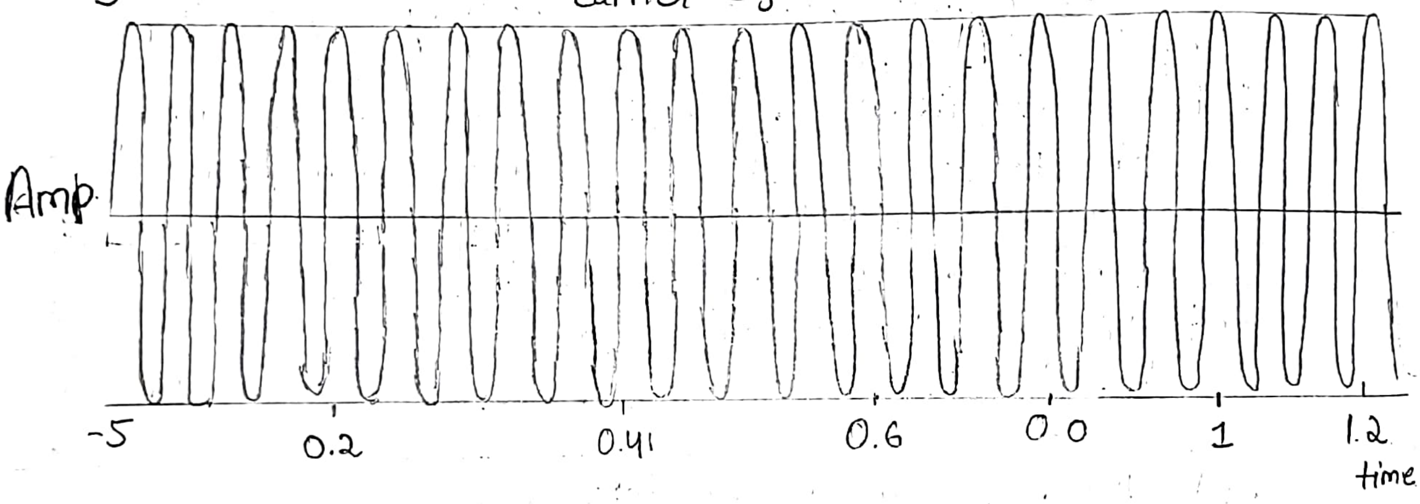
ASK

$f_c = 20\,Hz$
$f_p = 5\,Hz$
amp = 5.

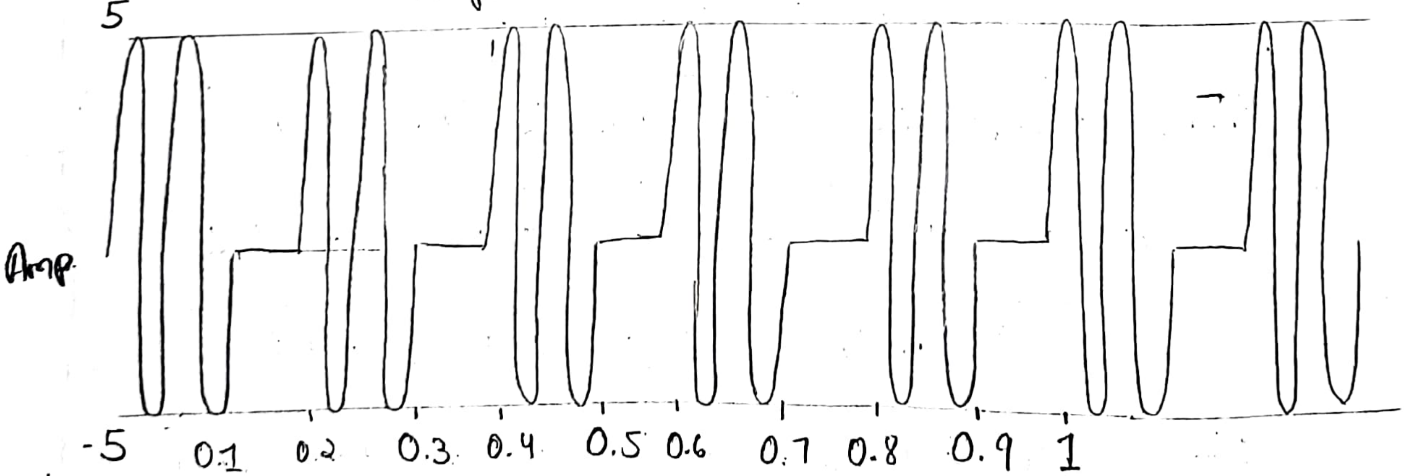
message Signal



Carrier Signal



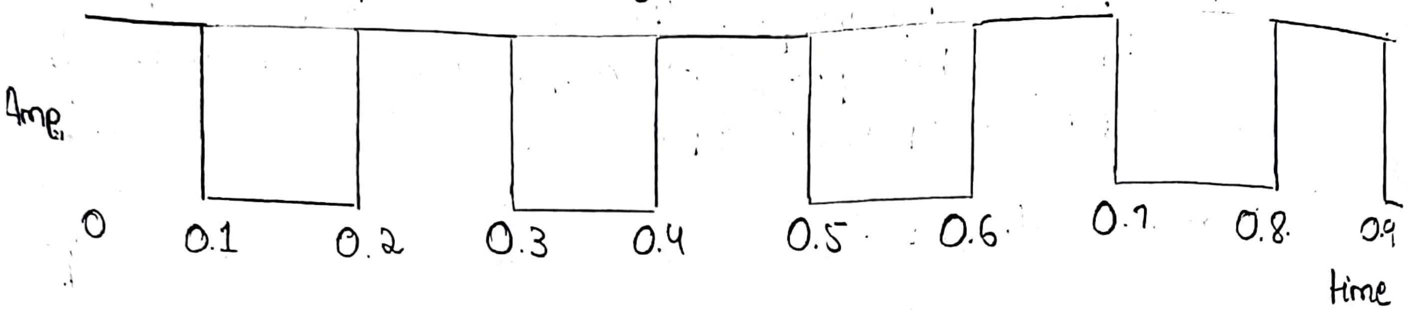
Amplitude shift keying Signal



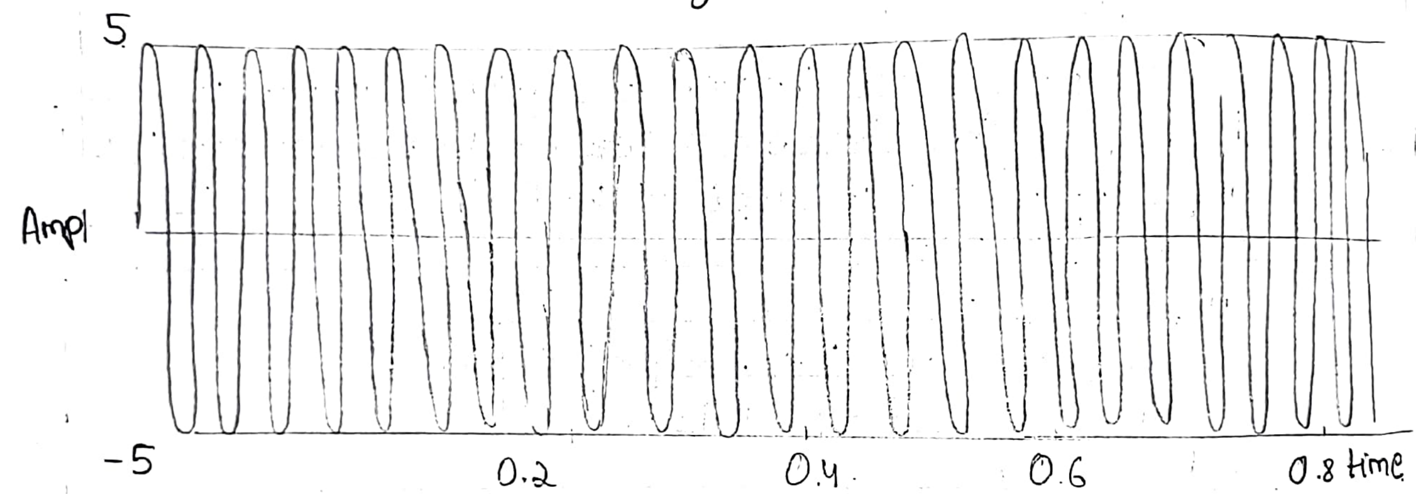
2) ASK.

$f_c = 45 Hz$.
$f_p = 5 Hz$
amp = 5

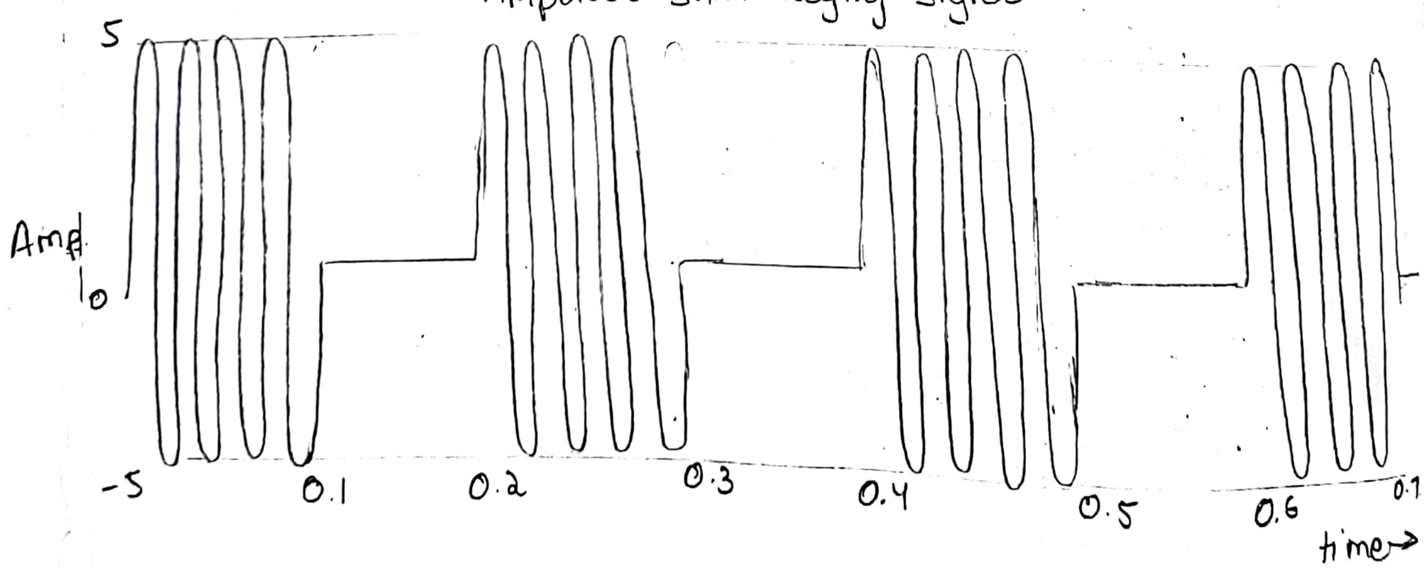
Message Signal



Carrier Signal



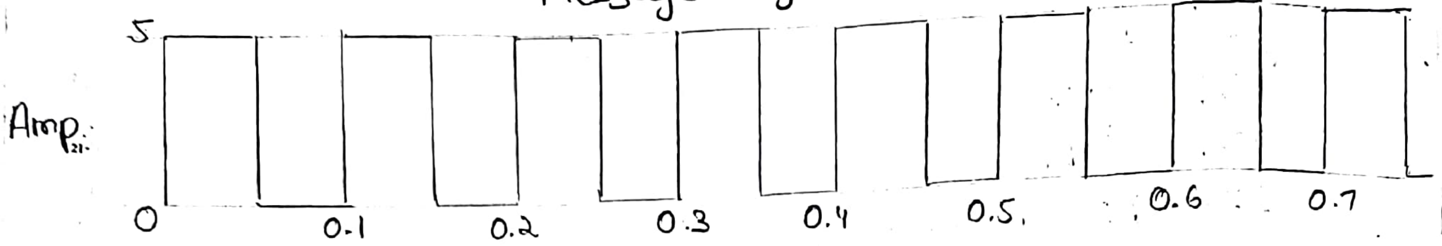
Amplitude shift keying signal



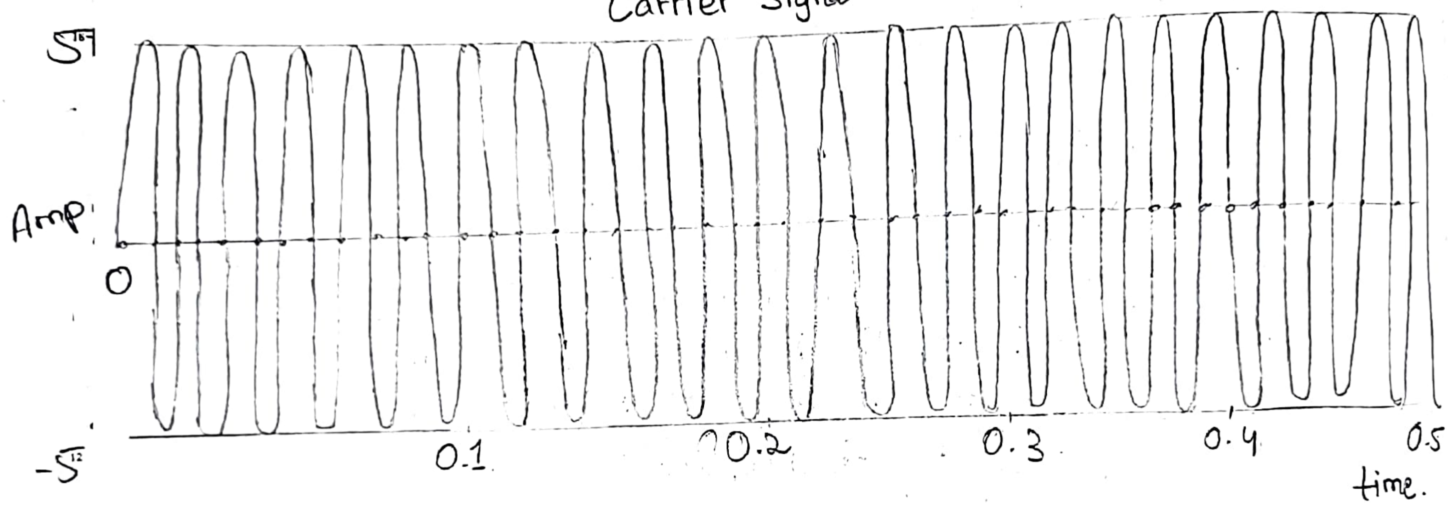
3) ASK

$f_c = 50Hz$
$f_p = 10Hz$
amp = 5.

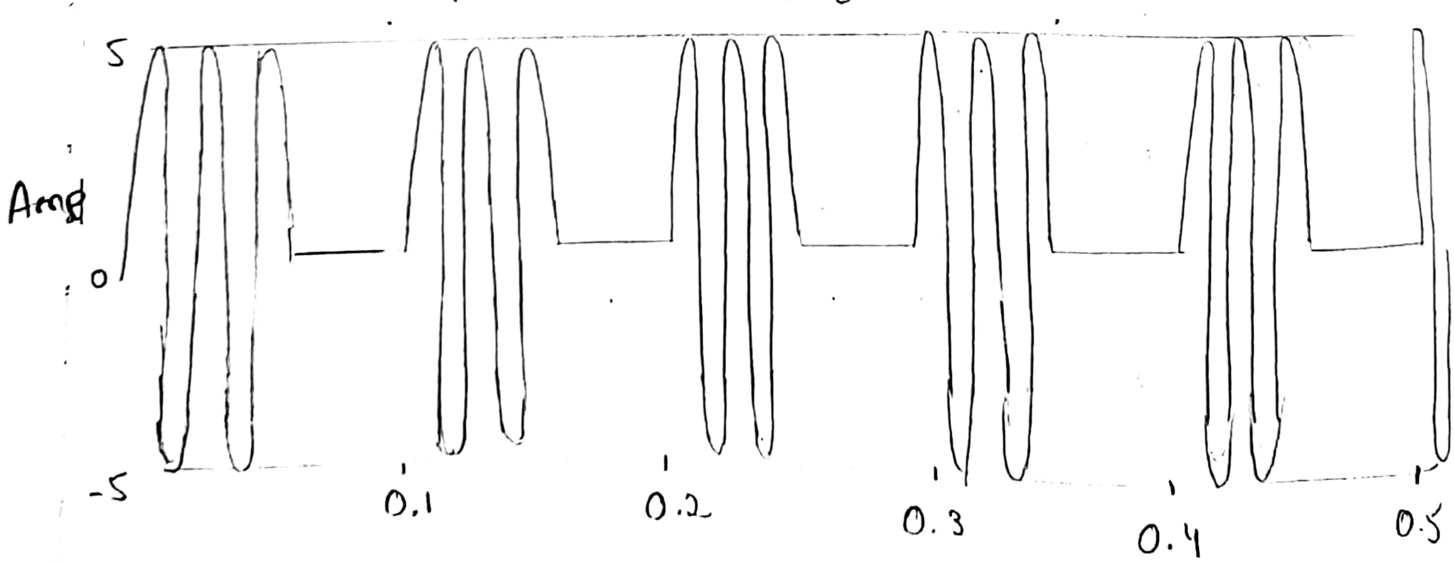
Message Signal



Carrier Signal



Amplitude Shift keying Signal



```
c1 = amp * sin(2* pi * fc1 * t);
c2 = amp * sin(2* pi * fc2 * t);
m = (amp/2) * square(2*pi*fp*t) + (amp/2)
for s = 0:1000
    if m(s+t) == 0
        mm(s+1) = c2(s+1);
    else
        mm(s+1) = c1(s+1);
    end
end
subplot(4,1,1)
plot(t, m)
xlabel('time')
ylabel('amplitude')
title('message signal')
subplot(4,1,2)
plot(t, c1)
xlabel('time')
ylabel('amplitude')
title('1st carrier signal')
subplot(4,1,3)
plot(t, c2)
xlabel('time')
y label('Amplitude')
title('FSK modulated signal')
```

MATLAB code for PSK

```
clc;
close all;
clear all;
```

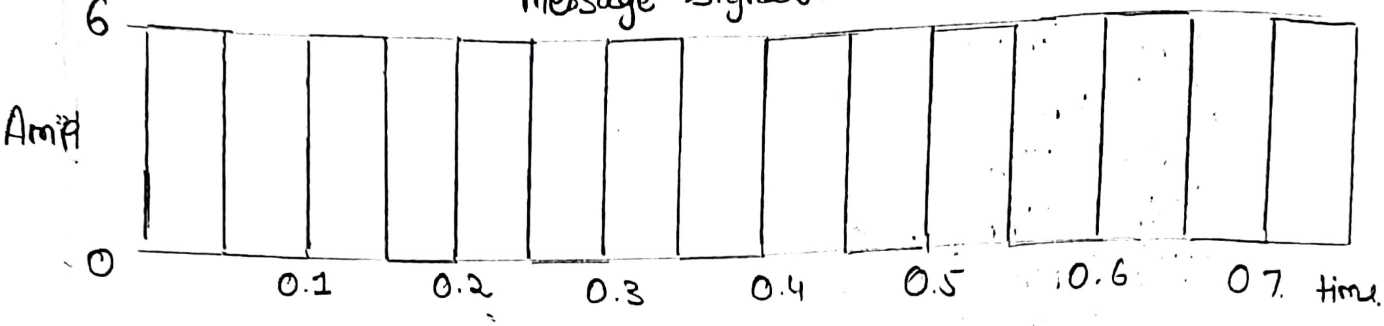
```
fc1 =
fp =
amp=
amp = amp
      2
t = 0:0.001:1;
g = amp * sin(2* ps * fc1 * t);
subplot(3,1,1);
plot(t, (i));
xlabel('time');
ylabel('amplitude);
grid on;
m = square(2* ps * fp * t);
subplot(3,1,2);
plot(t, m);
xlabel('time');
ylabel('amplitude');
title('Binary message signal');
grid on;
 for i=0:1000
     if (m(s+1) ==1)
         s(s+1) = c1(s+1)
     else
         s(s+1) = -c1(s+1)
     end
 end
subplot(3,1,3);
plot(t, s)
xlabel('Time');
ylabel('amplitude');
```

1) FSK

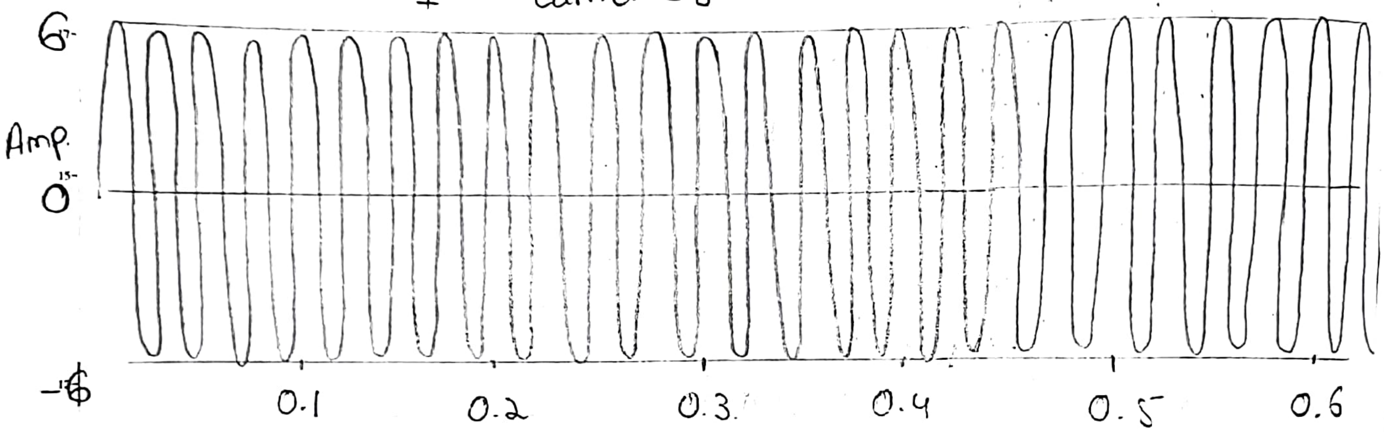
$f_{c_1} = 50Hz$ $f_{c_2} = 20Hz$

$f_p = 10Hz$ amp=6

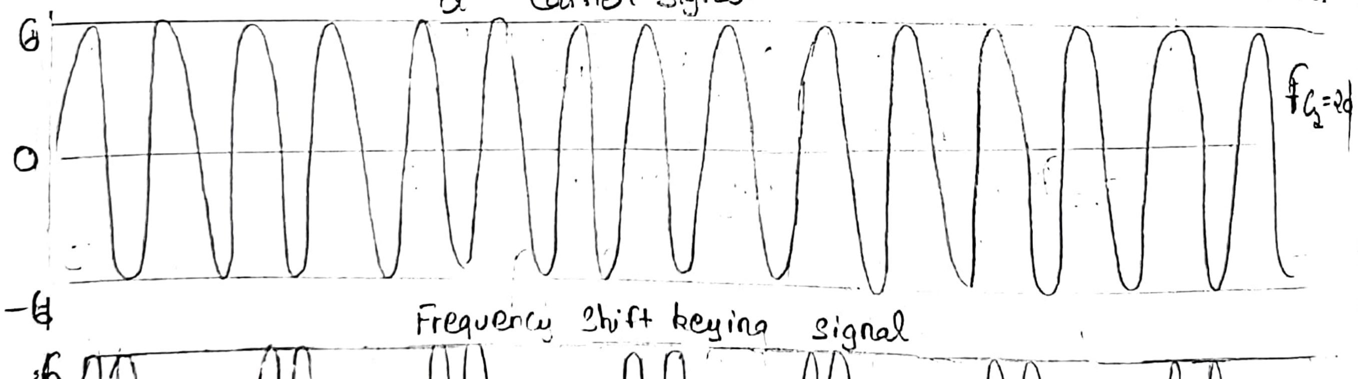
message Signal



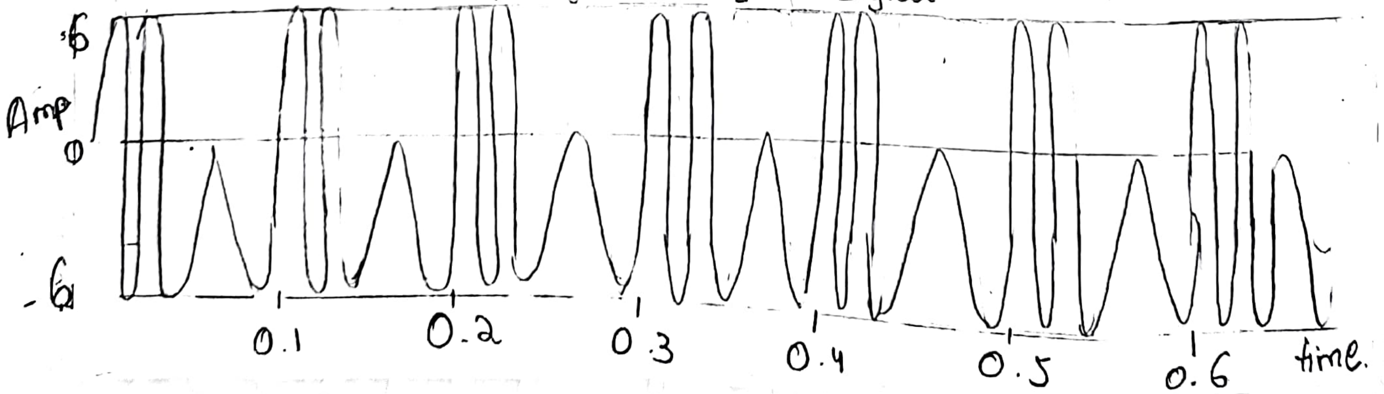
1st Carrier Signal



2nd Carrier signal



Frequency shift keying signal

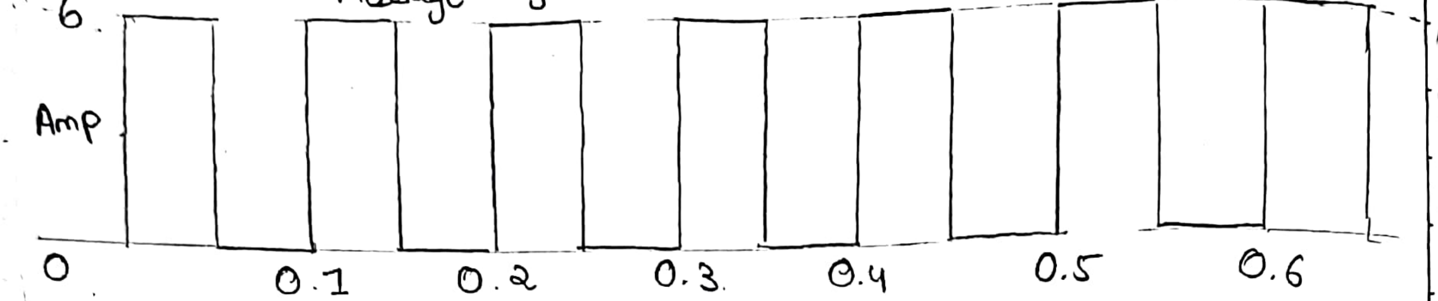


2) FSK

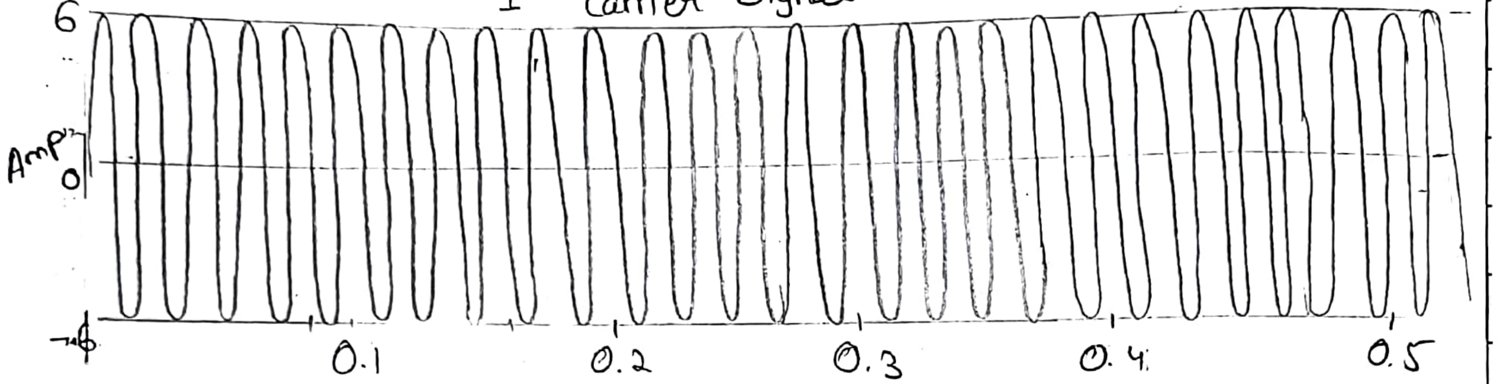
$f_{c_1} = 50\,Hz$ $f_{c_2} = 30\,Hz$

$f_p = 10\,Hz$ amp $= 6$

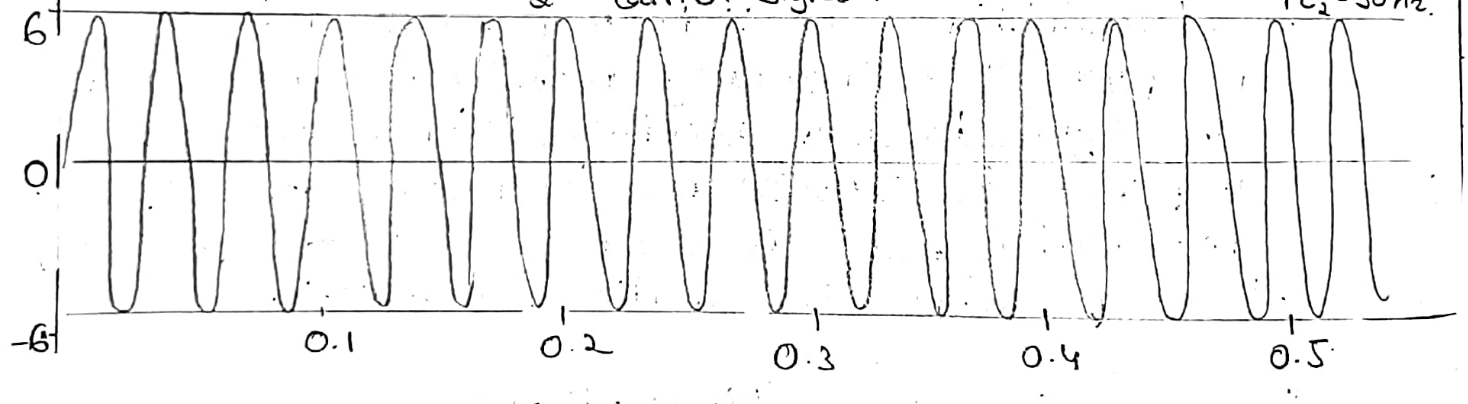
Message Signal



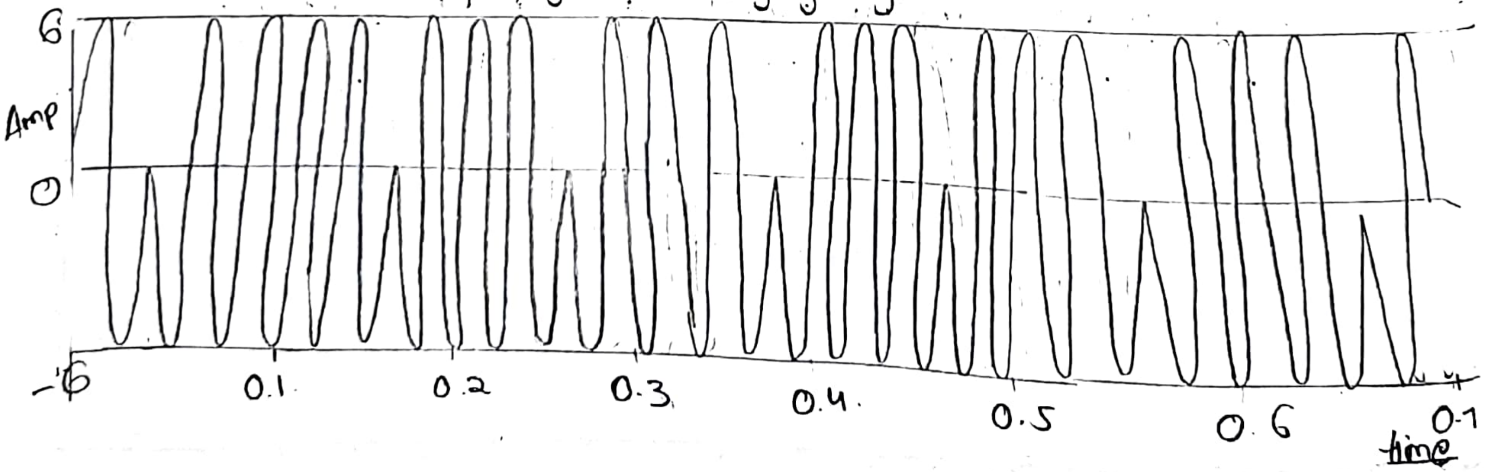
1st Carrier Signal. $f_{c_1} = 50\,Hz$



2nd Carrier Signal. $f_{c_2} = 30\,Hz$.



Frequency Shift keying Signal.

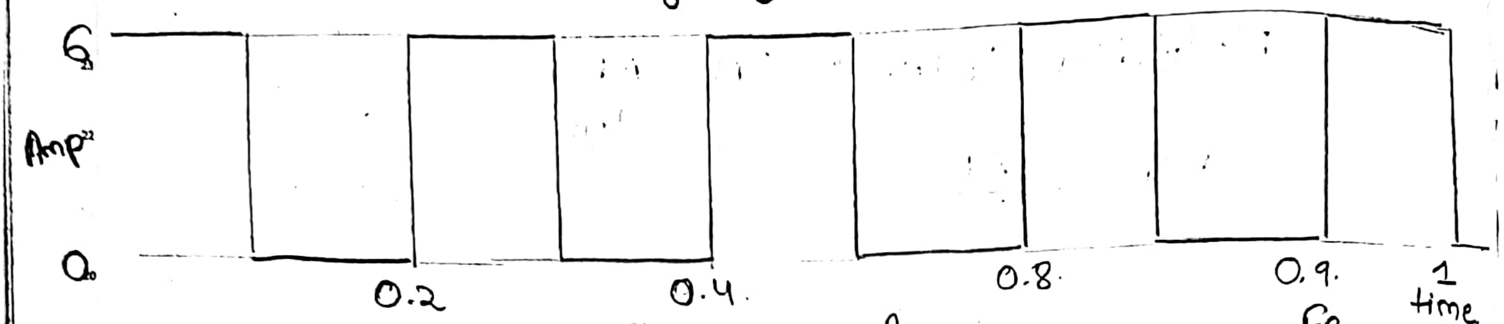


3) FSK.

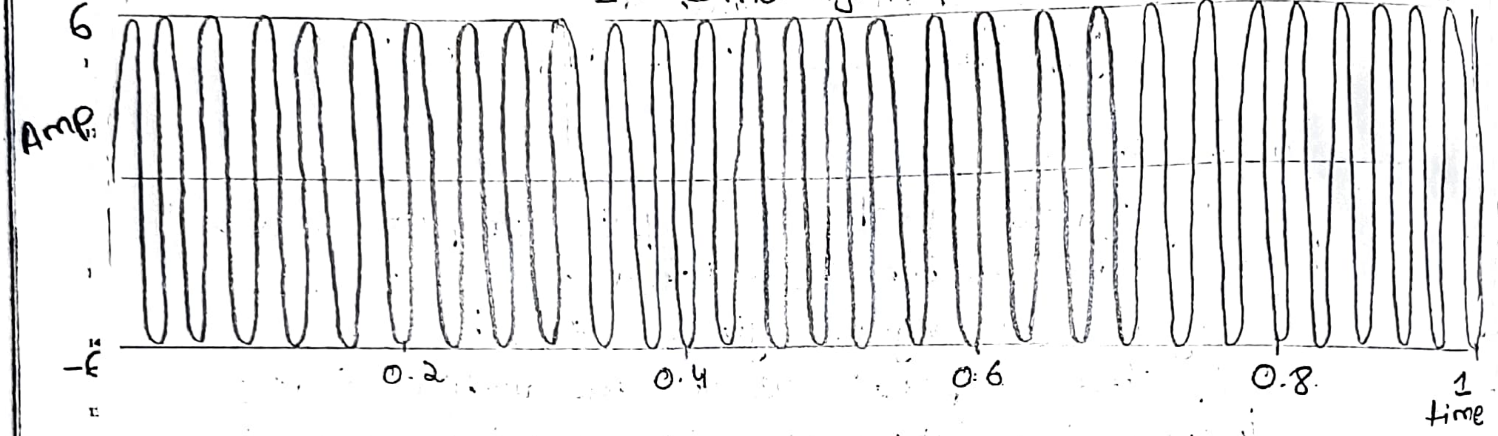
$f_{c_1} = 30Hz$ $\quad$ $f_{c_2} = 10Hz$

$f_p = 5Hz$ $\quad$ amp = 6

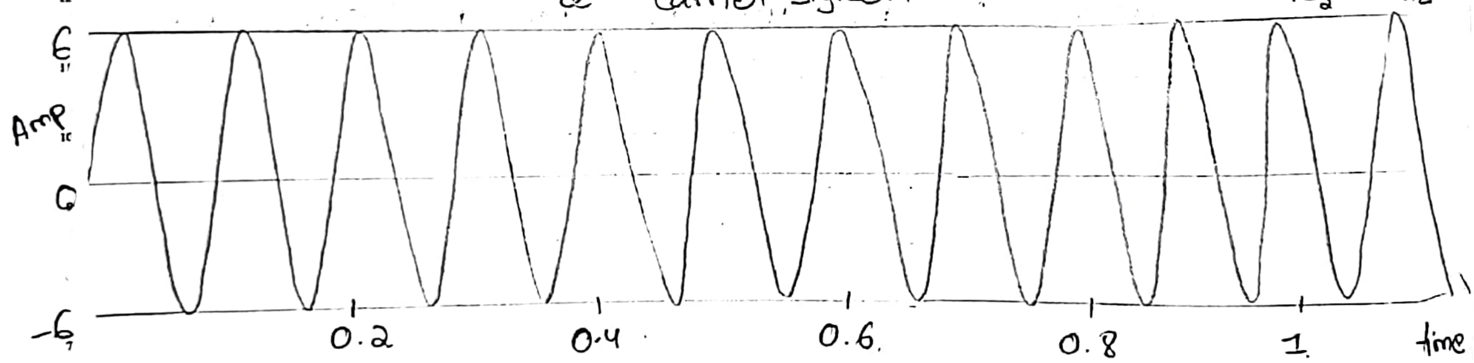
Message Signal.



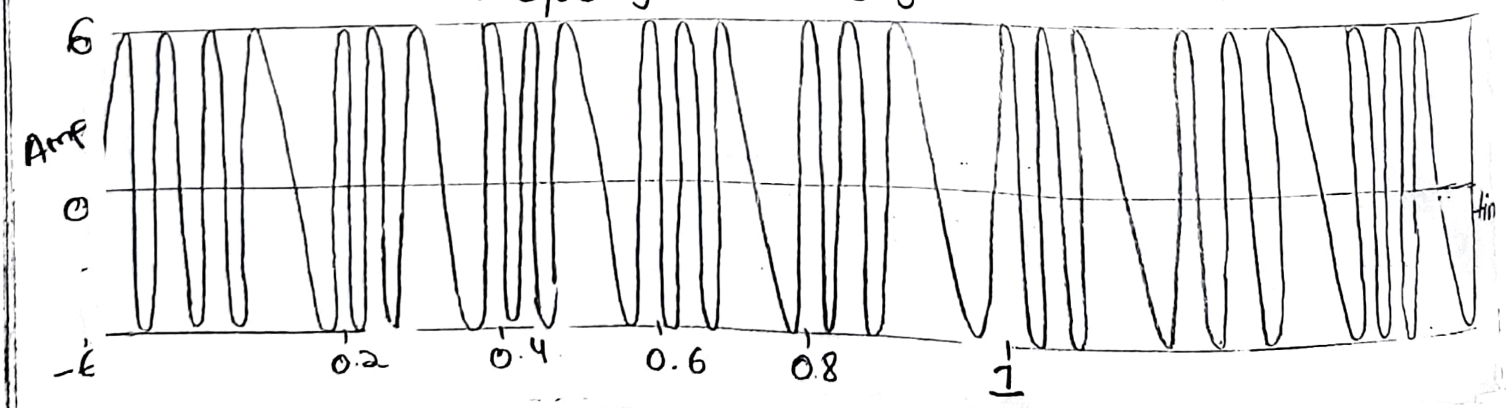
1st Carrier Signal. $\quad$ $fc_1 = 30Hz$



2nd Carrier Signal. $\quad$ $fc_2 = 10Hz$



Frequency Shift keying Signal



title ('Modulated wave')
grid on',

ADVANTAGES-

ASK:

- It offers high bandwidth efficiency
- Has simple receiver design
- ASK modulation & demodulation is comparitively inexpensive
- Can be used to transmit digital data over optical fiber

FSK:

- Easy to implement, high data rate, provides high SNR
- used for long distance communication
- power requirement is constant and FSK has good sensitivity
- It has better immunity than ASK method, so the probability of error-free reception of data is high

PSK:

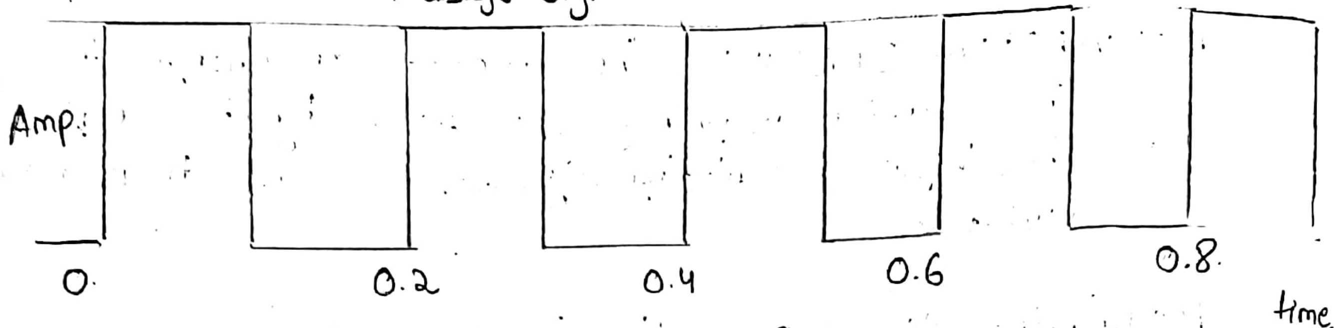
- It carries data over RF signal more efficient compared to the other modulation types. Hence it is more power efficient modulation compared to ASK and FSK
- It is less susceptible to errors compared to ASK and occupies same bandwidth as ASK.
- It allows information to be transmitted in the radio communication in a way more efficiently as compared to that of FSK

D) **PSK.**

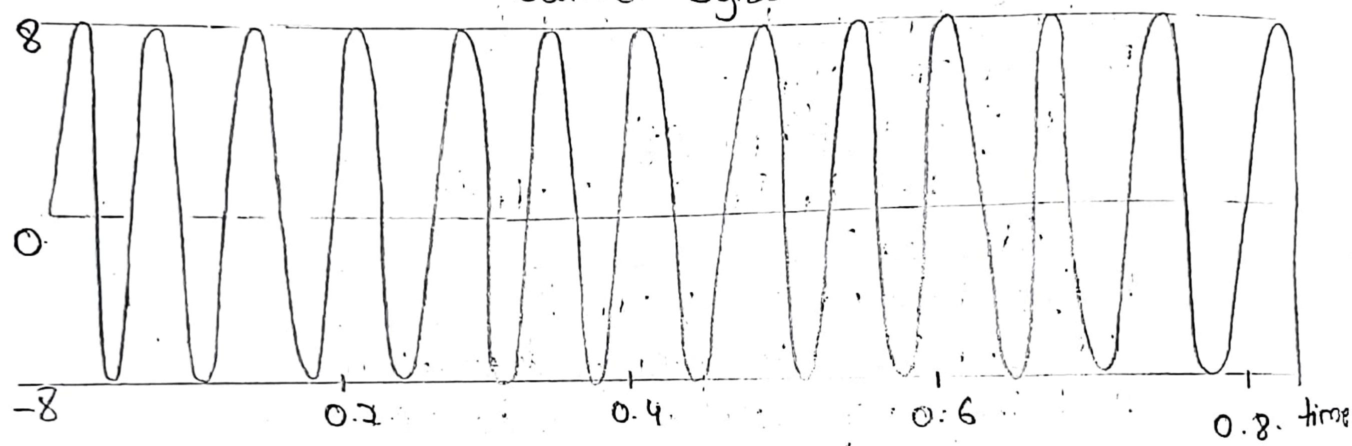
$f_c = 15Hz$ $f_p = 5Hz$

amp = 8

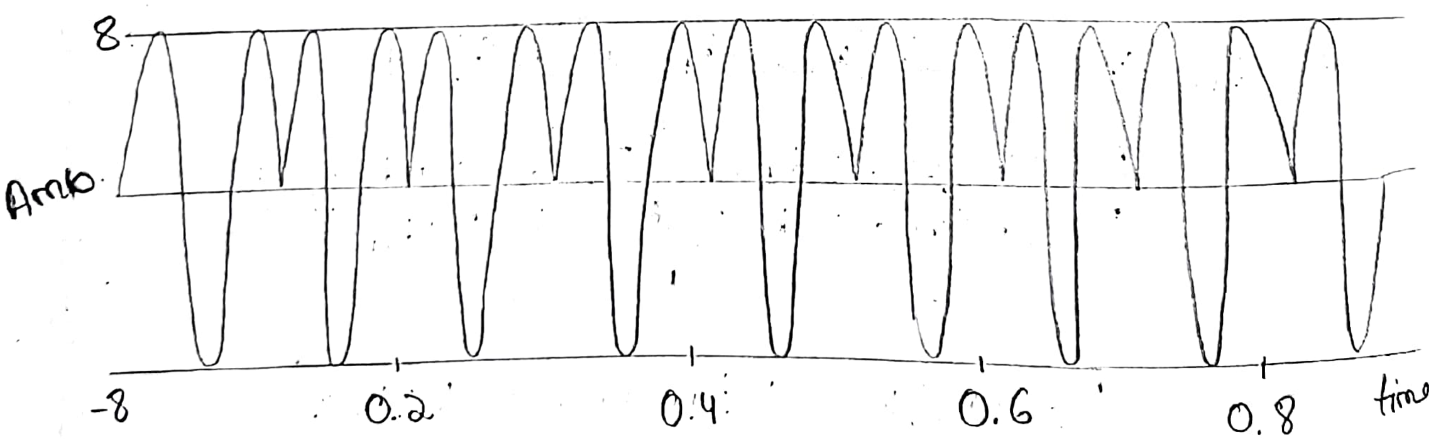
Message Signal



Carrier signal



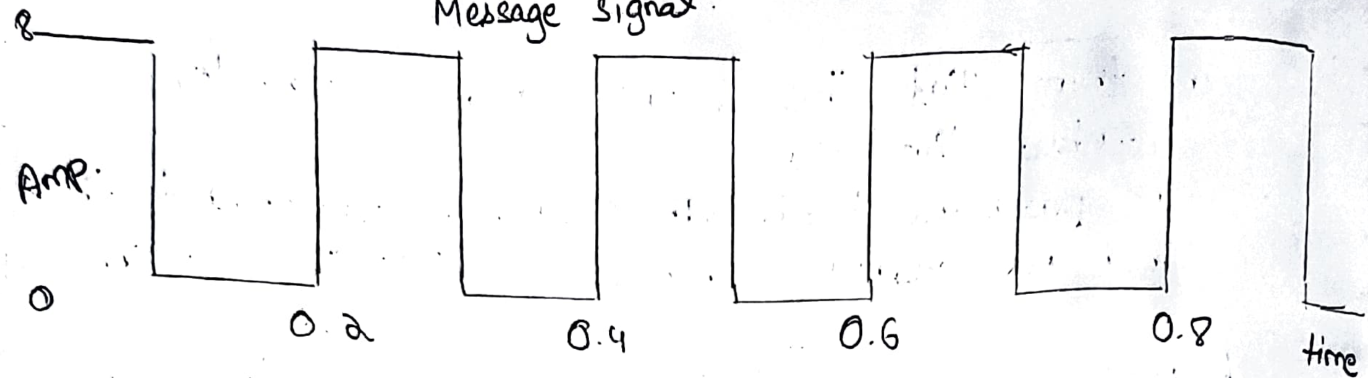
Phase shift keying Signal



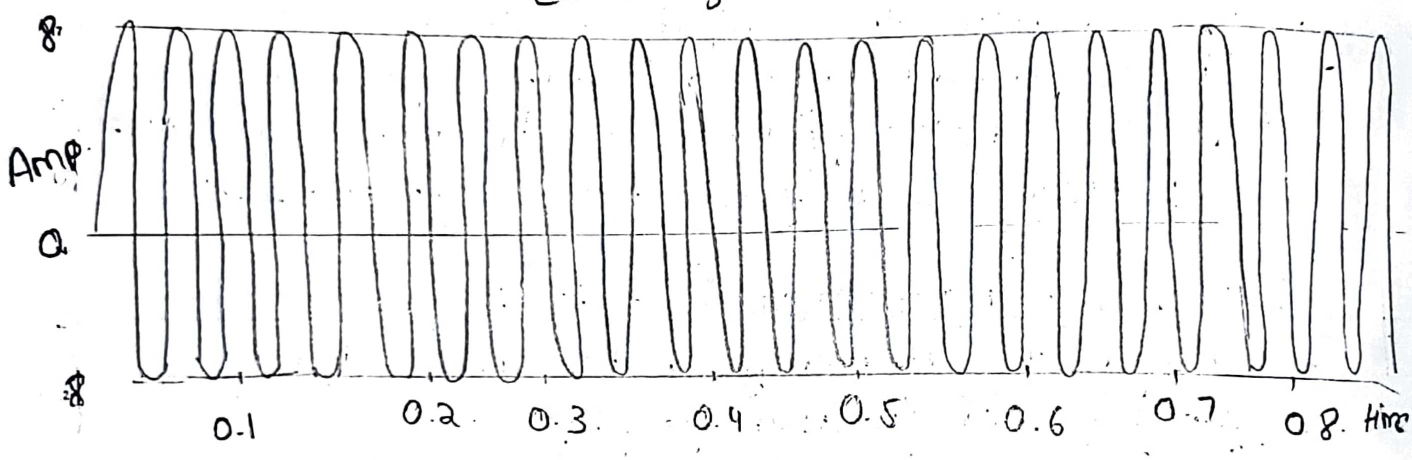
2) **PSK** $f_c = 2.5\,Hz$. $f_p = 5\,Hz$.

$amp = 8$.

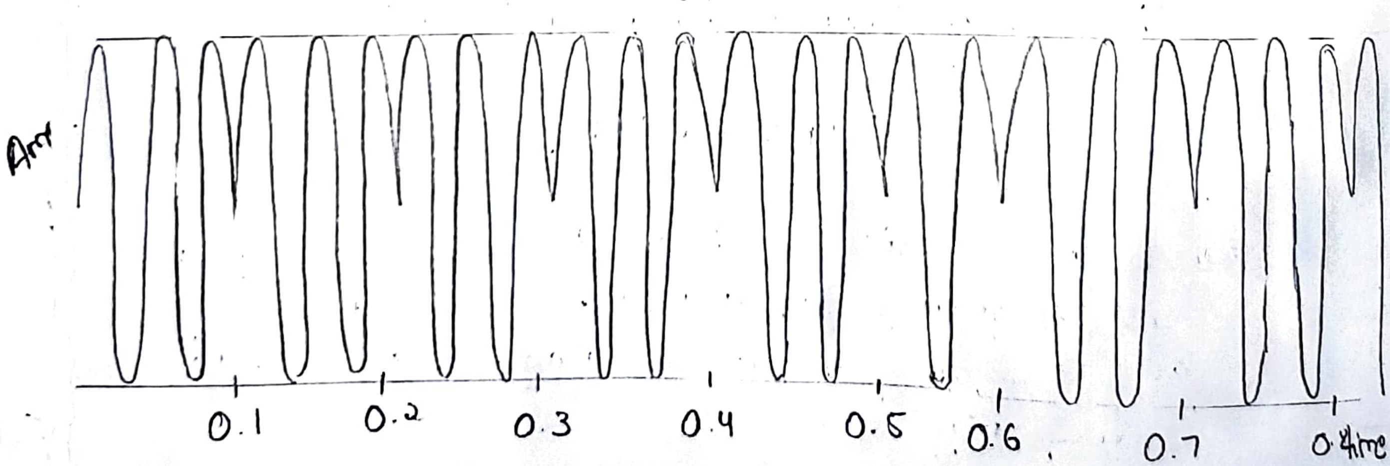
Message signal.



Carrier Signal.



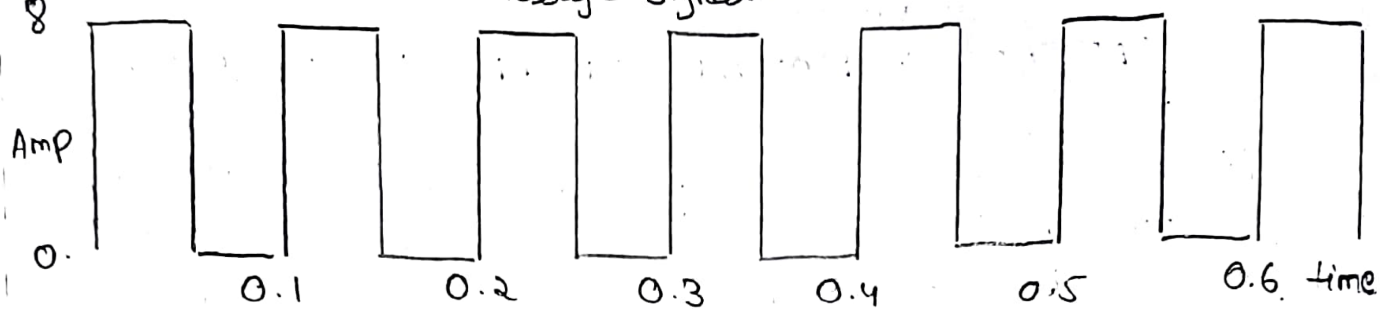
Phase shift Keying Signal



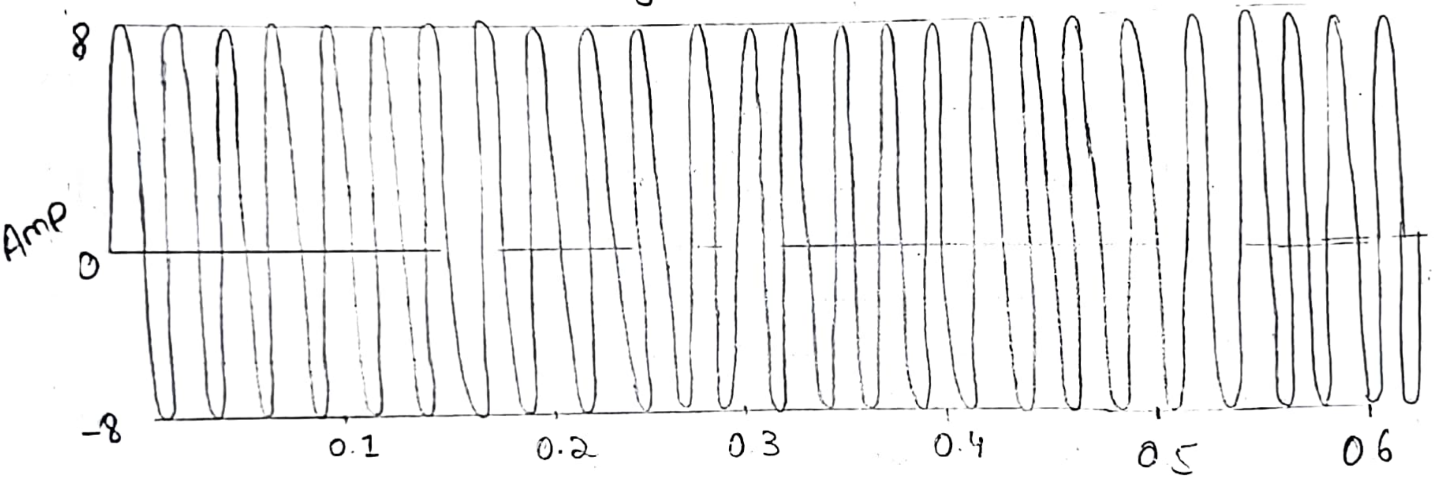
3) PSK $f_c = 40Hz$ $f_p = 10Hz$

amp = 8

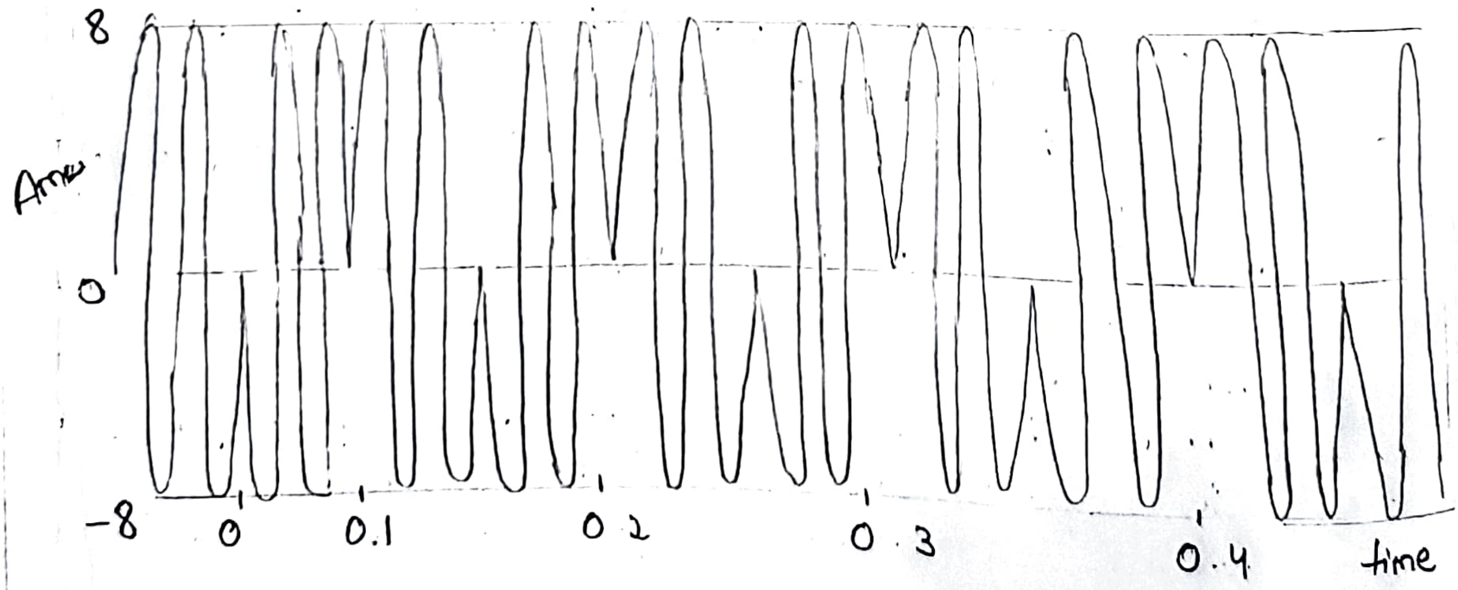
Message Signal.



Carrier Signal.



Phase shift keying.



CONCLUSION -
successfully performed ASK, FSK, PSK modulation technique and verified their waveforms using the MATLAB software

EXPERIMENT-7

EFFECT OF AWGN ON AM AND FM

AIM-
Examining how addition of noise in modulated signal (AM, FM) distort the original message signal

OBJECTIVE-
To study the transmission amplitude modulated (AM) and frequency modulated (FM) signal under the Additive Gaussian noise channel (AWGN) on AM and FM signal using the MATLAB / Simulink and draw the distorted waveform for different signal to noise ratio (SNR) values

SOFTWARE - MATLAB/Simulink

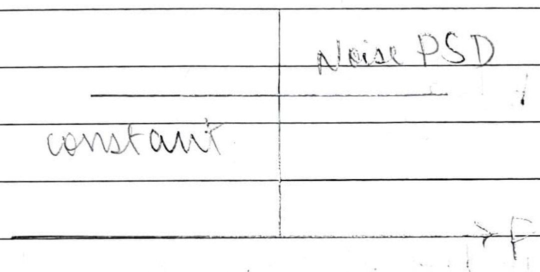
THEORY-
AWGN is a basic noise model used to mimic the effect of many random processes that occur in nature channel produces Additive White Gaussian Noise

Additive - The received signal equeals the transmit signal plus some noise where the noise is statiscally independent of signal

$$r(t) = s(t) + w(t) \rightarrow \text{noise}$$

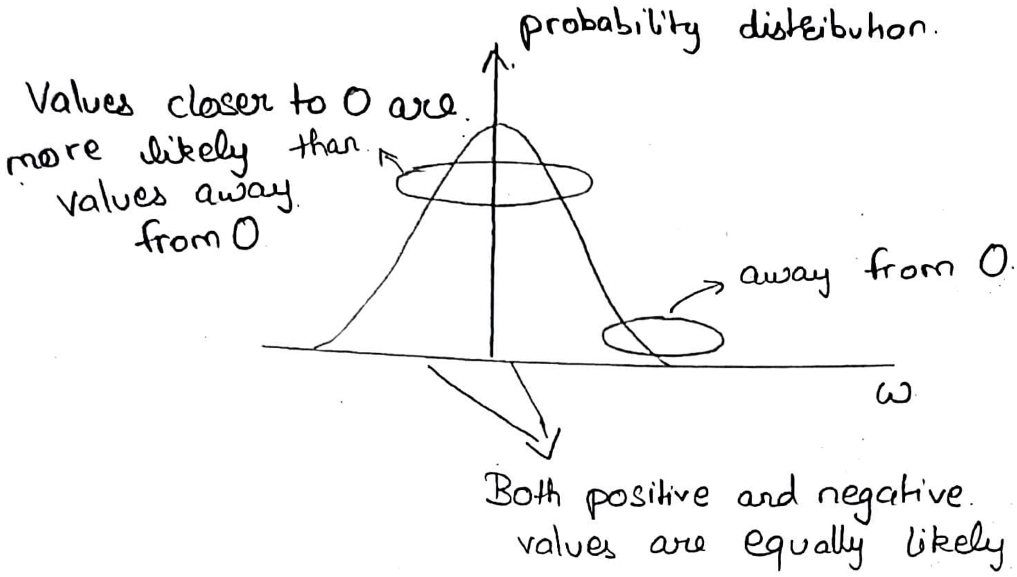
$s(t) \downarrow$ message signal

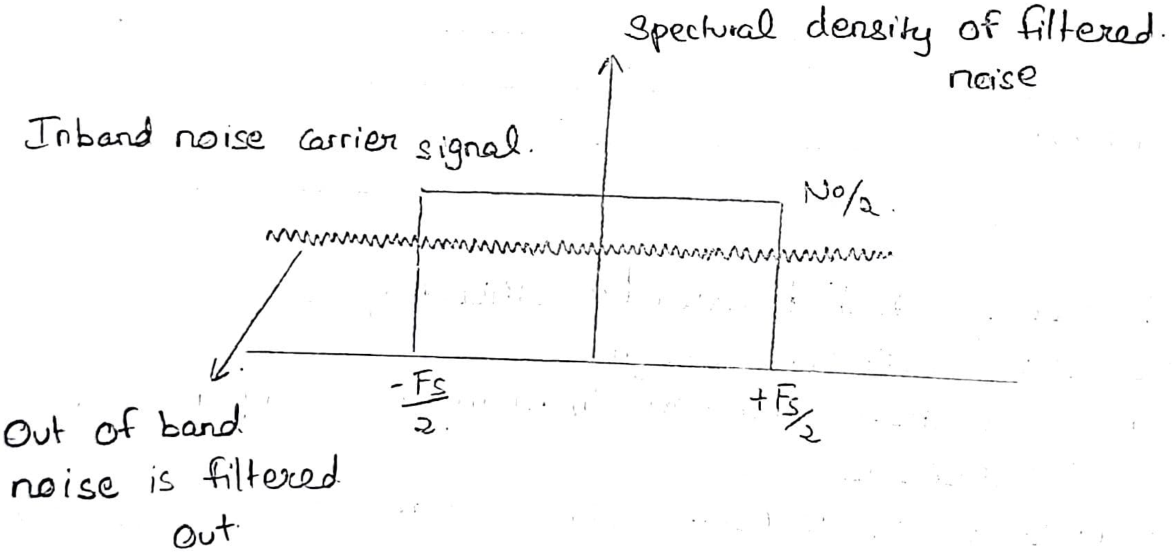
White - It refers that the noise has some power distribut at every frequency or it has uniform power across the frequency band for the information system. It is an analogy to the color white ~~bas~~ which has uniform emission at all frequency in the visible spectrum if we focus a beam of light for each colour on the visible spectrum onto a single spot that combination would result in a beam of white light, as consequence, power spectral density (PSD) of white noise is constant for all frequency ranging from $-\infty$ to $+\infty$, as shown in figure



Gaussian – Gaussian distribution, or a normal distribution, has an average of zero in time domain, and is represented as a bell-shaped curve. The probability distribution of the noise sample is Gaussian with a zero mean. The value close to zero has a higher chance of occurence 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 frequency of interest in wireless communication (a few KHz to hundreds of GHz) but not for higher frequencies

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 power to noise power. Often expressed in decibel. A ratio higher than 1:1 (greater than 0dB) indicate more signal than noise. SNR, bandwidth and channel capacity of communication channel are connected by Shannon-Hartley theorem


probability distribution.
Values closer to 0 are more likely than values away from 0
away from 0.
ω
Both positive and negative values are equally likely


Spectral density of filtered noise
Inband noise carrier signal.
No/2.
Out of band noise is filtered out
- Fs/2
+ Fs/2

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

Shannon-Hartley theorem - It indicates that channel capacity (bits per second) or information rate of data that can be communicated at low error rate using an average received signal power through communication channel subject to AWGN of power

$$C = B\log_2\left(1+\frac{S}{N}\right)$$

where B is bandwidth of channel in Hertz. We can see that it is related to SNR. Different case for SNR Values.

5dB - 10dB → is minimum level to establish a connection due to noise level being nearly indistinguishable from desired signal.

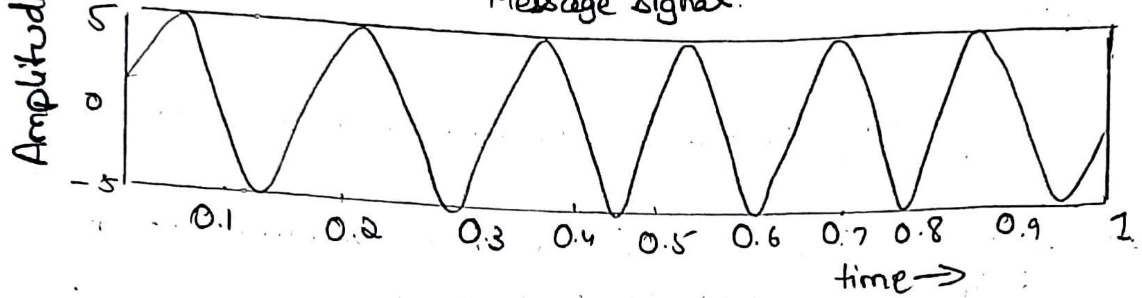
25dB - 40dB → deemed to be good
41dB a higher → deemed excellent

AWGN over AM (amplitude modulation)

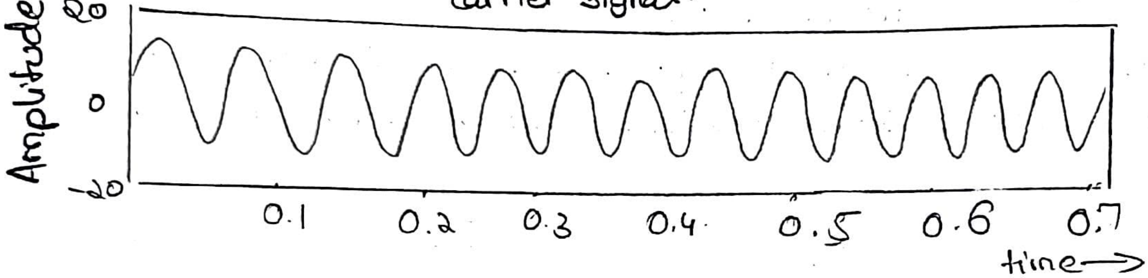
Let $e_m(t) = e_m \sin \omega_m t$ be message signal and
$e_c(t) = e_c \sin \omega_c t$ be carrier signal
then $e_m(t) = (e_c + e_m(t)) \sin \omega_c t$
$e_m(t) = (e_c + e_m \sin \omega_m t) \sin \omega_c t$

AWGN effect over AM

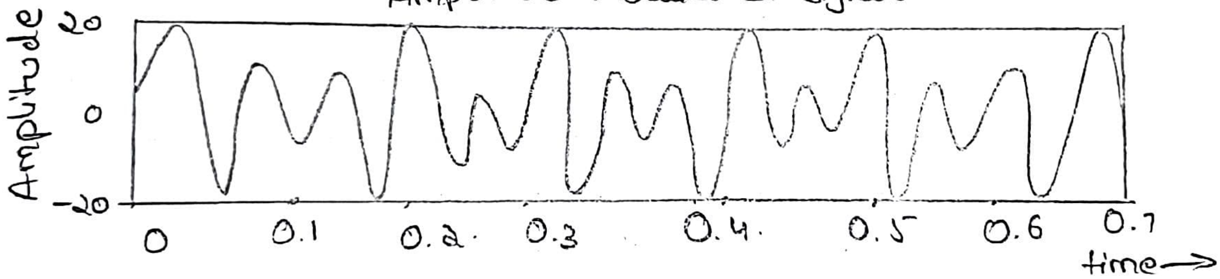
Message signal.



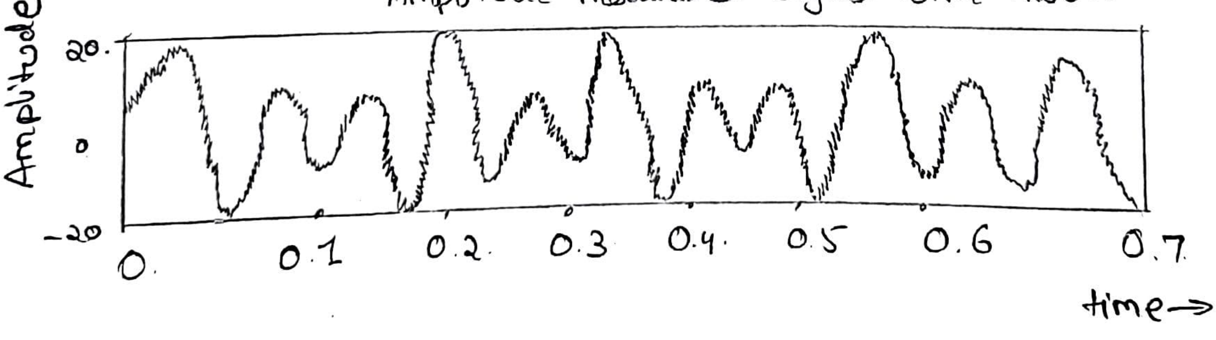
Carrier signal.



Amplitude Modulated Signal



Amplitude modulated signal with AWGN.



Matlab code for performing the AWGN effect over AM

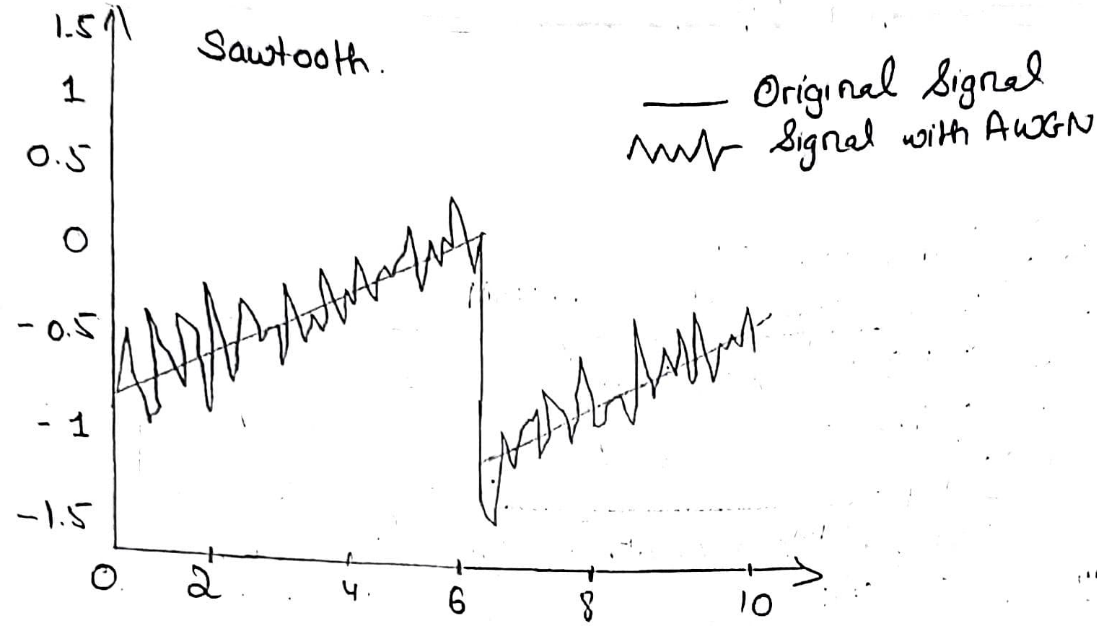
1) clc;
clear all;
t=0:0.001:1;
V_m=6;
V_c=12;
f_m=5;
f_c=30;
m = V_m* sin (2* pi * f_m*t);
c = V_c * sin (2* pi* f_c*t);
amp= V_c + V_m * sin (2* pi * f_c*t);
y = awgn(am, 15, 'measured');
subplot (4,1,1);
plot (t, m);
xlabel ('time');
ylabel('amplitude');
title (' carrier signal ');
subplot (4, 1, 3); plot (t, am);
xlabel ('time');
ylabel ('amplitude');
title ('amplitude modulated signal');
subplot (4,1,4);
plot (t, y);
xlabel ('time');
ylabel ('amplitude');
title ('amplititude modulated signal with AWGN');

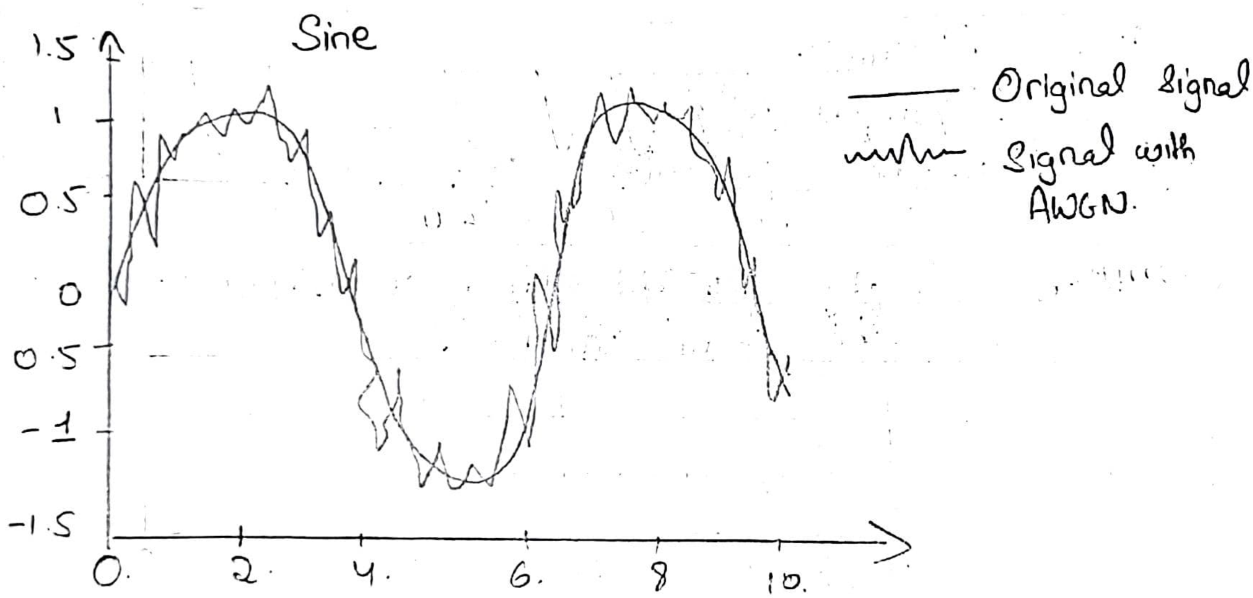
Matlab code for AWGN effect on different functions i.e. sine, cosine, sawtooth

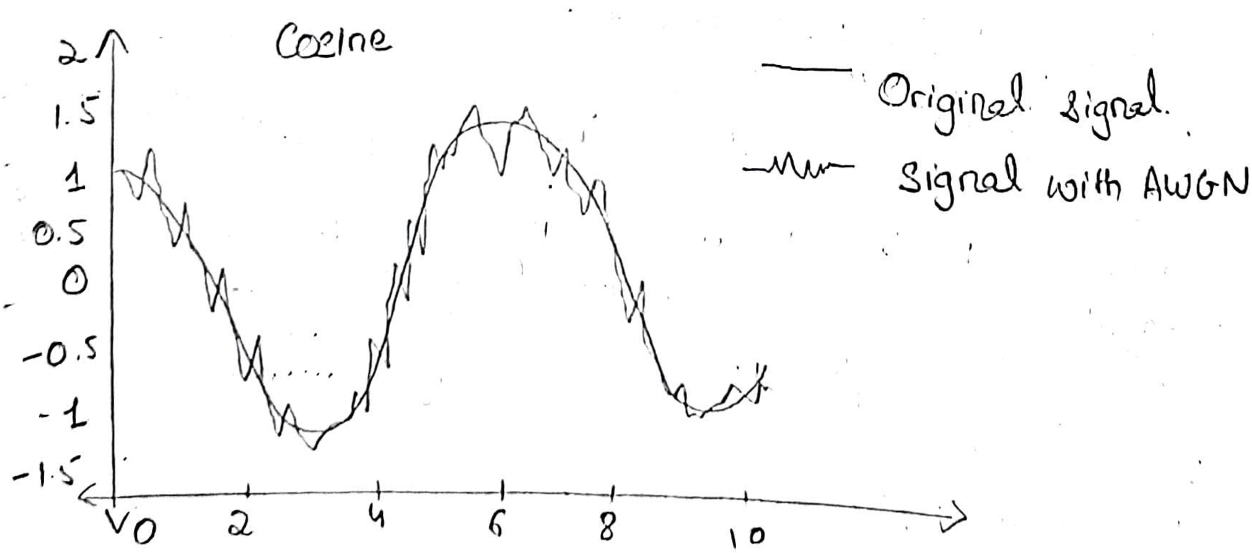
```
clc;
clear all;
t =(0:0.1:10)
x = sawtooth(t);
y =awgn(x,10,'measured');
plot (t,[x y])
legend ('original signal', 'signal with AWGN')

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')
```

Matlab code for AM signal with different SNR values

```
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* f * t);
amp= amp* sin (2* pi *fc* t);
```


Sawtooth.
Original Signal
Signal with AWGN
1.5
1
0.5
0
-0.5
-1
-1.5
0
2
4
6
8
10


Sine
Original Signal
Signal with AWGN.
1.5
1
0.5
0
0.5
-1
-1.5
0.
2.
4.
6.
8
10.

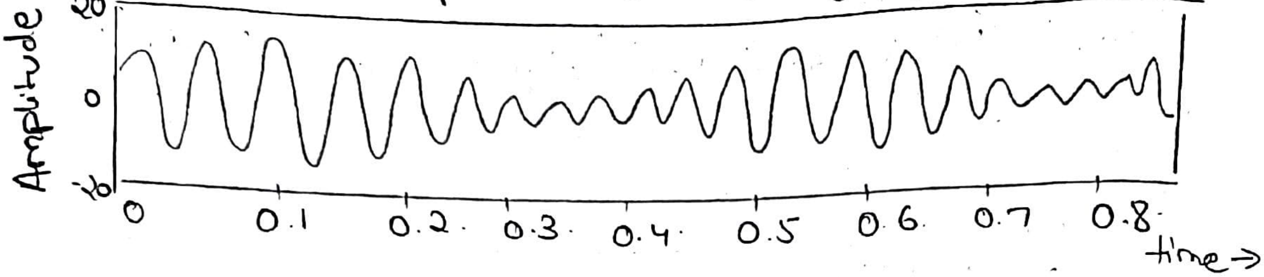

Cosine
Original signal.
Signal with AWGN
2
1.5
1
0.5
0
-0.5
-1
-1.5
0
2
4
6
8
10

```
y1 = awgn (am, 10, 'measured');
y2 = awgn (am, 100, 'measured');
y3 = awgn (am, 1000, 'measured');
subplot (4,1,1);
plot (t, am);
xlabel ('time');
ylabel ('amplitude'); title ('amplitude modulated signal');
subplot (4,1,2);
plot (t, y1);
xlabel ('time');
y label ('amplitude');
title ('amplitude modulated signal with AWGN [snr 10]');
subplot (4,1,3);
plot (t, y2);
xlabel ('time');
ylabel ('amplitude');
title ('amplitude modulated signal with AWGN [snr 100]');
subplot (4,1,4);
plot (t, y3);
xlabel ('time');
ylabel ('amplitude');
title ('amplitude modulated signal with AWGN [snr 1000]');
```

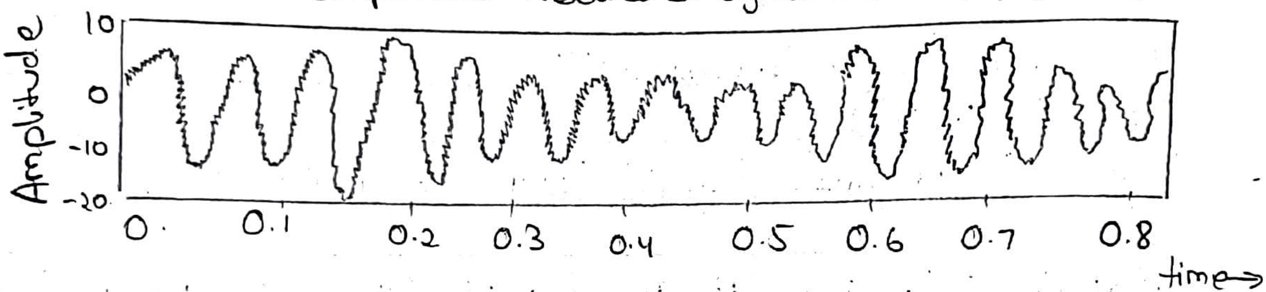
Matlab code for FM signal under AWGN

```
clc;
clear all;
t = 0:0.001:1;
Vm = 5;
Vc = 5;
fm = 2;
fc = 25;
```

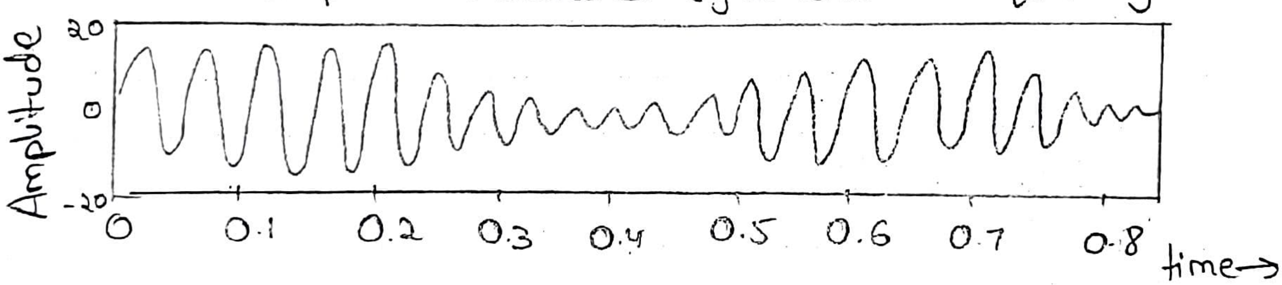
amplitude modulated signal



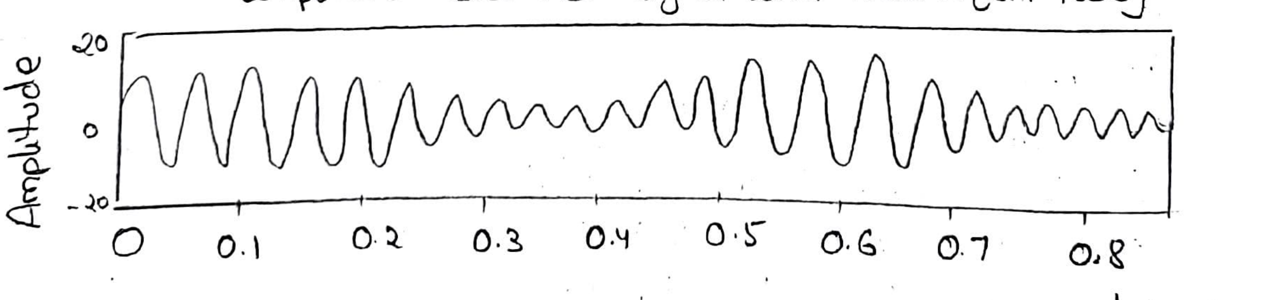
amplitude modulated signal with AWGN [snr 10]



amplitude modulated signal with AWGN [snr 100]



amplitude modulated signal with AWGN [snr 1000]



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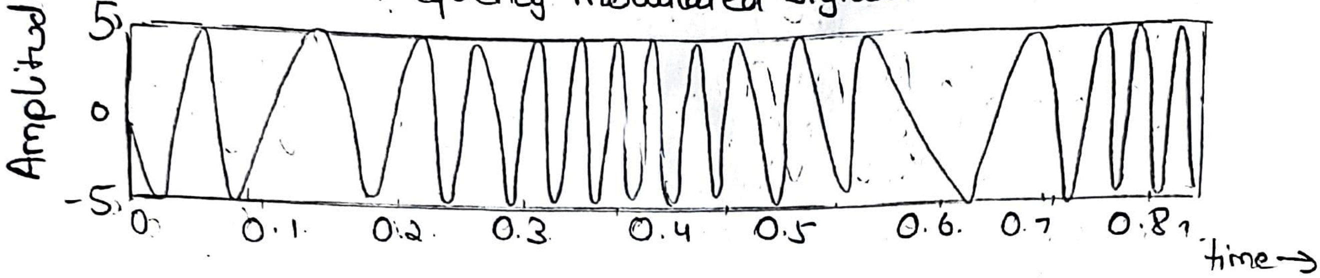
```
fd = 5;
msg = Vm* sin(2* pi * fm * t);
C = Vc* sin(2* pi* fc* t);
y = Vc* sin(2* pi * fc* t + fd*cos(2* pi* fm*t));
z = awgn(y, 5 'measured');
subplot (4, 1, 1);
plot (4, 1, 1);
plot (t, msg);
x label ('time');
y label ('amplitude');
title ('carrier signal');
subplot (4, 1, 3); plot (t, y);
x label ('time');
y label ('amplitude');
title ('Frequency modulated signal');
subplot (4, 1, 4);
plot (t, z);
x label ('time');
y label ('amplitude');
title ('Frequency modulated signal with AWGN');
```

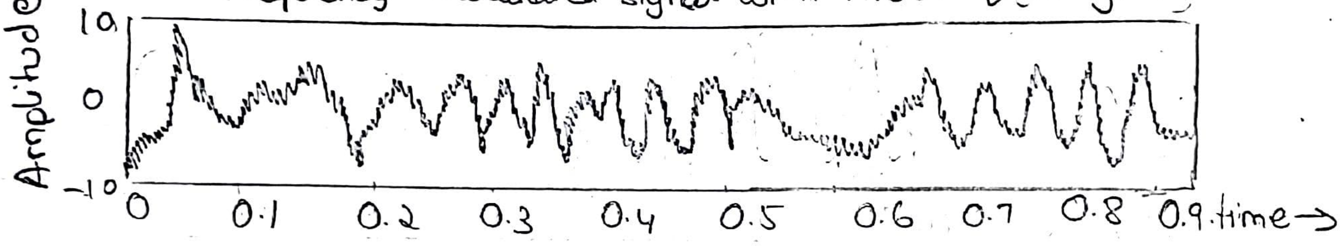
<u>Matlab code for different SNR in FM</u>

```
clc;
clear all;
t = 0:0.001:1;
Vm = 10;
Vc = 5;
fm = 2;
fc = 25;
fd = 10;
m = Vm * sin(2* pi * fm * t);
```

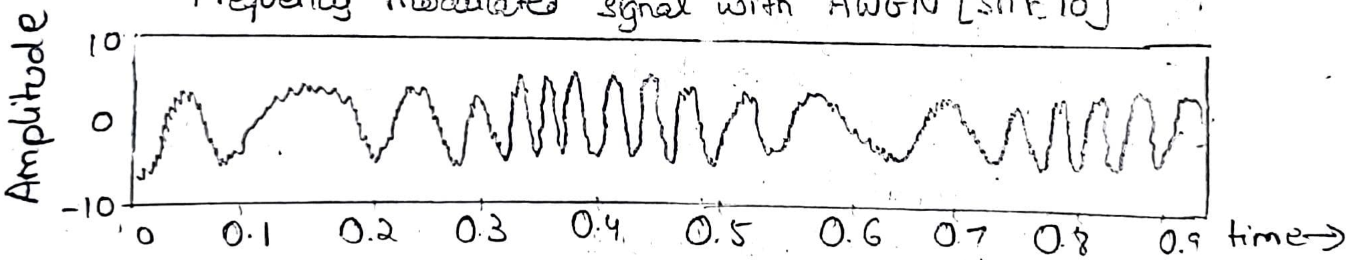
Frequency modulated signal



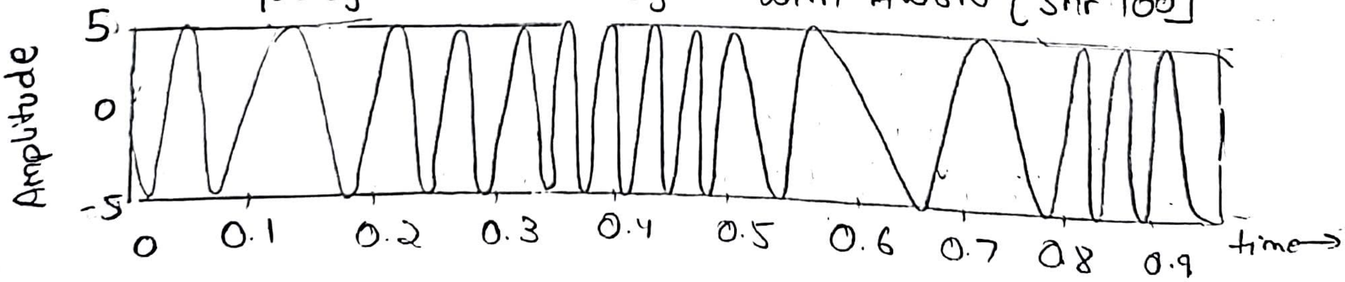
Frequency modulated signal with AWGN [snr 1]



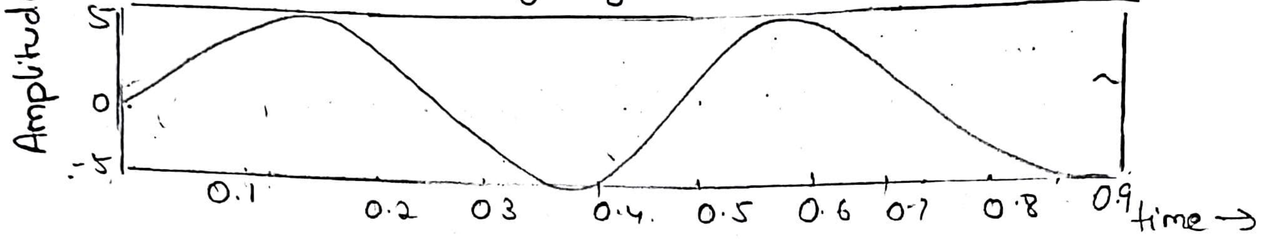
Frequency modulated signal with AWGN [snr 10]



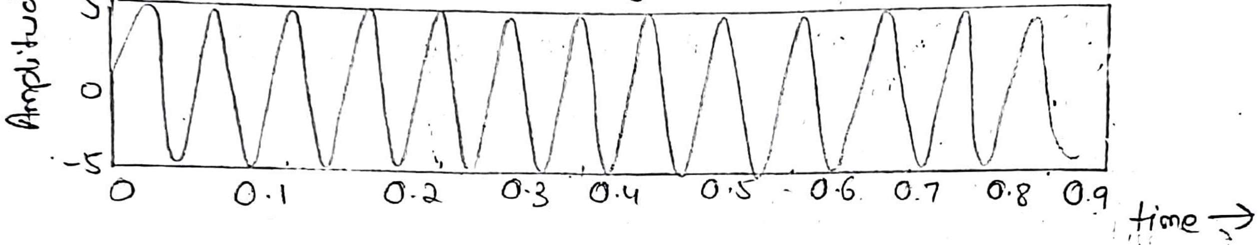
Frequency modulated signal with AWGN [snr 100]



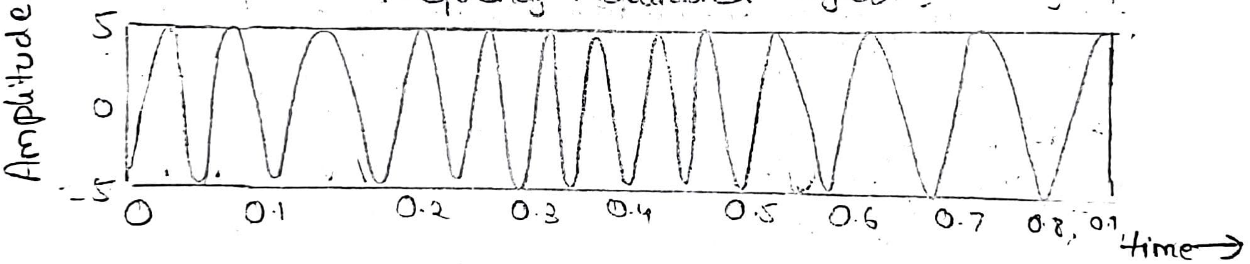
Message Signal



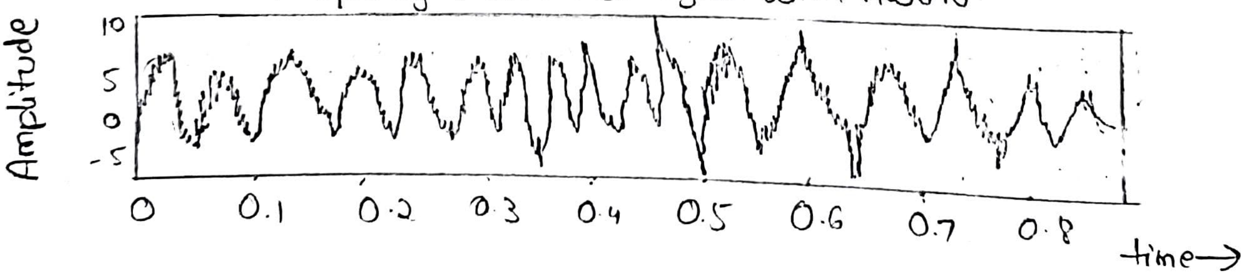
Carrier Signal



Frequency Modulated Signal



Frequency Modulated Signal with AWGN

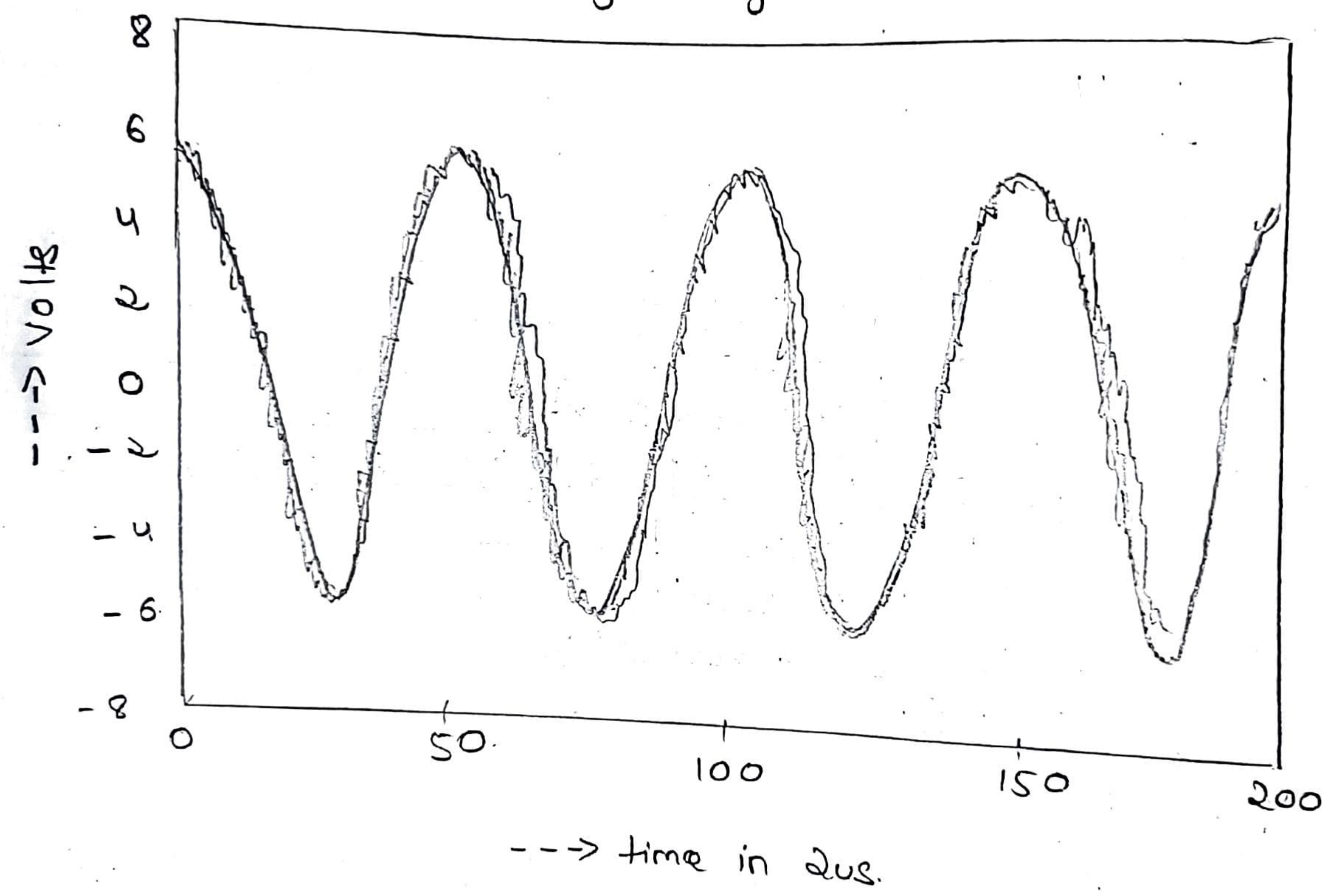


```
c = Vc*sin(2*pi*fc*t);
amp=Vc+Vm*sin(2*pi*fm*t);
y = Vc* sin(2*pi*fc*t+fd*cos(2*pi*fm*t));
y1=awgn(y,1,'measured');
y2=awgn(y,10,'measured');
y3=awgn(y,100,'measured');
subplot(4,1,1);
plot(t,y);
xlabel('time');
ylabel('amplitude'); title('Frequency modulated signal');
subplot(4,1,2);
plot(t,y1);
xlabel('time');
ylabel('amplitude');
title('Frequency modulated signal with AWGN [snr 1]');
subplot(4,1,3);
plot(t,y2);
xlabel('time');
ylabel('amplitude');
title('Frequency modulated signal with AWGN [snr 10]');
subplot(4,1,4)
plot(t,y3);
xlabel('time');
ylabel('amplitude');
title('Frequency modulated signal with AWGN [snr 100]');
```

Matlab code for moving average filter to retrieve the signal by averaging the noise fluctuation

```
% moving average filter
clear all;
close all;
clc;
fs = 500000;
fm = 10000;
t = 1:200;
x = 5* cos(2* pi*(fm/fs)*t);
z = awgn(x,5);
% average white gaussian noise to the input with S/N =5
plot(x,'g','linewidth',1.5);
hold on;
plot(z);
hold on;
for i =1:194;
```

Moving Average Filter.



Noisy.
Actual
Filtered

EXPERIMENT-8

SINGLE SIDE BAND MODULATION SCHEME

AIM-

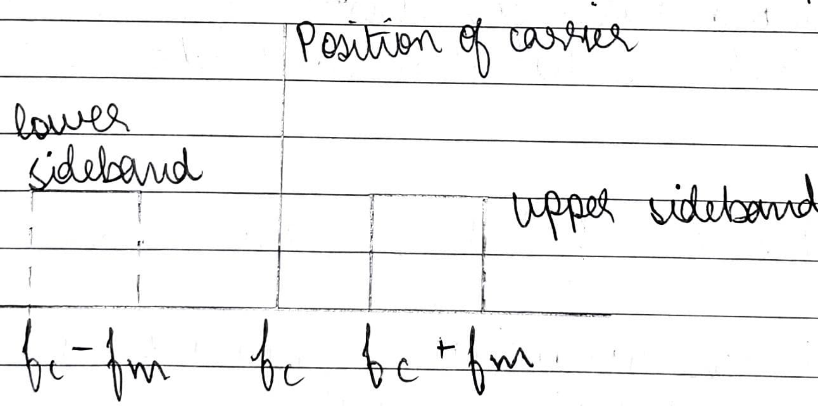
To write and stimulate a program for single side-band (SSB) modulation scheme. Draw the message/carrier waveforms and resultant modulated signal in time domain and frequency domain. Show the input/output waveforms using Matlab code/Simulink in virtual mode

APPARATUS-

MATLAB software

THEORY-

The process of suppressing one of the sidebands along with the carrier and transmitting a single sideband is called single sideband suppressed carrier system or simply SSBC.



Here, the carrier and the lower sideband are suppressed is used for transmitting the lower sideband. This single sideband suppressed carrier or SSBSC system, which transmits a single sideband has high power, as the power allotted for both the carrier and other sideband is utilized in transmitting this single sideband. In SSBSC, only one sideband is transmitted because both USB and LSB contain same information.

Mathematical Expression -

Let Modulating signal $\Rightarrow m(t) = A_m \cos 2\pi f_m t$
Let carrier signal $\Rightarrow c(t) = A_c \cos 2\pi f_c t$

SSBSC wave, $s(t) = m(t)c(t)$

$s(t) = \frac{A_m A_c}{2} \cos[2\pi(f_m + f_c)t]$ for upper sideband

$s(t) = \frac{A_m A_c}{2} \cos[2\pi(f_c - f_m)t]$ for lower sideband

Bandwidth of SSBSC wave
In DSBSC modulated wave, the wave contains two sidebands and its bandwidth is $2f_m$. Since SSBSC modulated wave contains only one sideband, its bandwidth is half of the bandwidth of DSBSC modulated wave

Bandwidth of SSBSC wave $= f_m$

Therefore the bandwidth required is same as that required for the modulating signal

$\Rightarrow$ Power Calculator
As SSBSC wave eqn $s(t) = \frac{A_m A_c}{2} \cos[2\pi(f_c + f_m)t]$ for USB

$s(t) = \frac{A_m A_c}{2} \cos[2\pi(f_c - f_m)t]$ for LSB

Power of SSBSC is equal to the power of any one sideband frequency components

$$P = P_{USB} = P_{LSB}$$

As $P = \frac{(V_{rms})^2}{R} = \frac{(V_m/\sqrt{2})^2}{R}$

$$P_{USB} = \frac{(A_m A_c / 2\sqrt{2})^2}{R} = \frac{A_m^2 A_c^2}{8R}$$

$$P_{LSB} = \frac{A_m^2 A_c^2}{8B}$$

$$P_{SSBSC} = \frac{A_m^2 A_c^2}{8R}$$

Therefore, the power required is less than that required for DSBSC wave

Method of generation of SSBSC -

1) Frequency discrimination Method -

In this method, first we will generate DSBSC wave with the help of the product modulator. Then, apply this DSBSC wave as an input of band pass filter. The band pass filter produces an output which is SSBSC wave. Select the frequency range of band pass' filter as the spectrum can be turned to either USB or LSB frequencies to get respective SSBSC wave having USB or LSB.

2) Phase discrimination method or Hilbert transform Method -

The block diagram consists of two product modulators, two -90° phase shifters. The modulating signal $A_m \cos(2\pi f_m t)$ are applied to product modulator

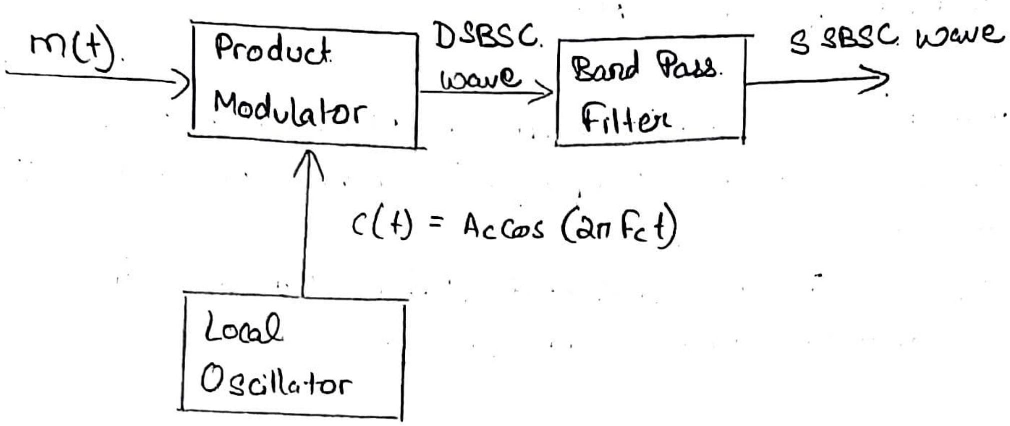


Fig :- Frequency Discrimination Method

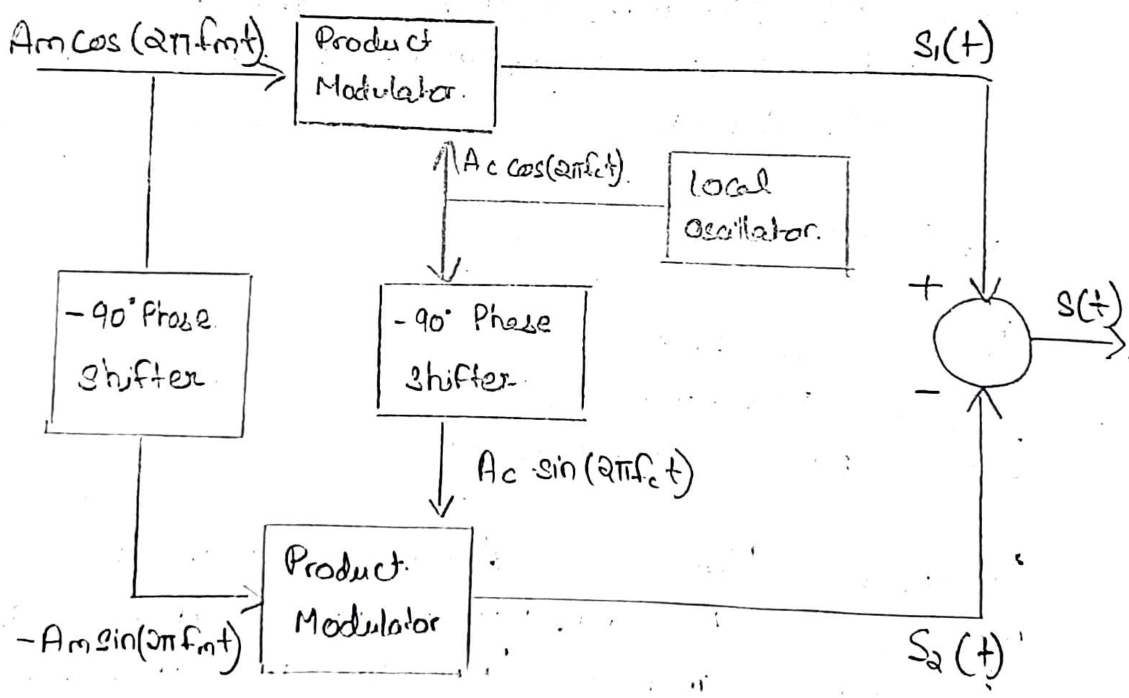


Fig :- Phase Discrimination Method

The output: ~~of modulating and carrier signals passed thro~~

$S_1(f) = A_m A_c \cos(2\pi f_m t) \cos(2\pi f_c t)$

$s_1(f) = \frac{A_m A_c}{2} \{ \cos[2\pi(f_c + f_m)t] + \cos[2\pi(f_c - f_m)t] \}$

The output of modulating and carrier signals passed through -90° phase shifter and then product modulator:

$S_2(t) = A_m A_c \cos(2\pi f_m t - 90) \cos(2\pi f_c - 90°)$

$s_2(t) = \frac{A_m A_c}{2} \{ \cos[2\pi(f_c - f_m)t] - \cos[2\pi(f_c + f_m)t] \}$

Adding $S_1(t)$ and $s_2(t)$

$S(t) = A_m A_c \cos[2\pi(f_c - f_m)t] \Rightarrow$ lower sideband

subtract $S_2(t)$ from $S_1(t)$

$S(t) = A_m A_c \cos[2\pi(f_c + f_m)t] \rightarrow$ upper sideband

Therefore, choosing correct polarities of input at summer block will get SSBSC having upper or a lower sideband

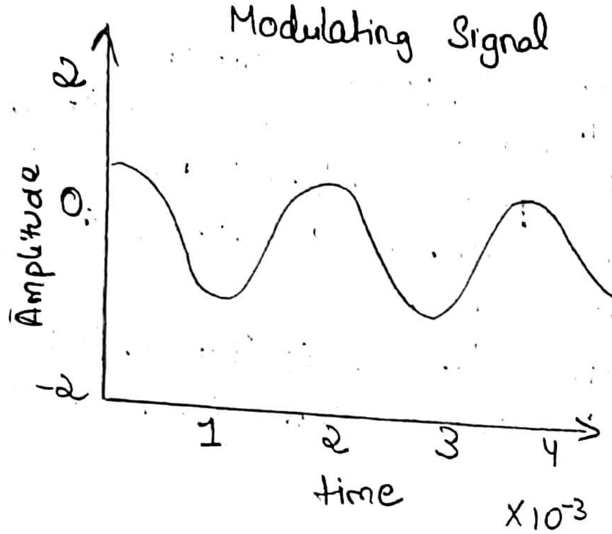
Matlab code-

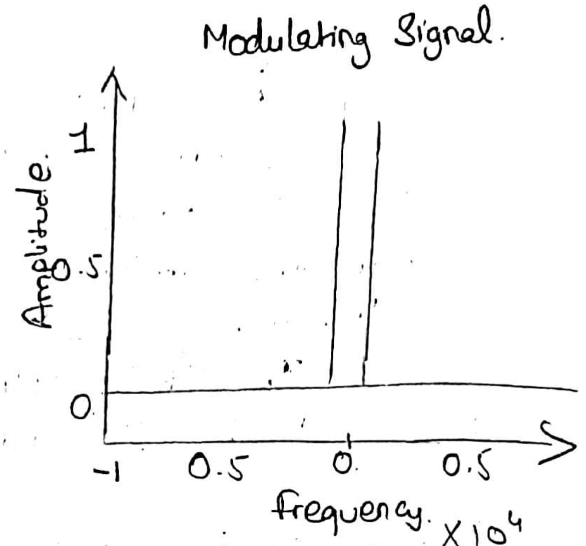
```
clc;
clear;
close all;
am = 1;       % amplitude of modulating signal
ac = 1;       % amplitude of carrier signal
fm = 500;     % modulating signal frequency
fc = 5000;    % carrier frequency
```

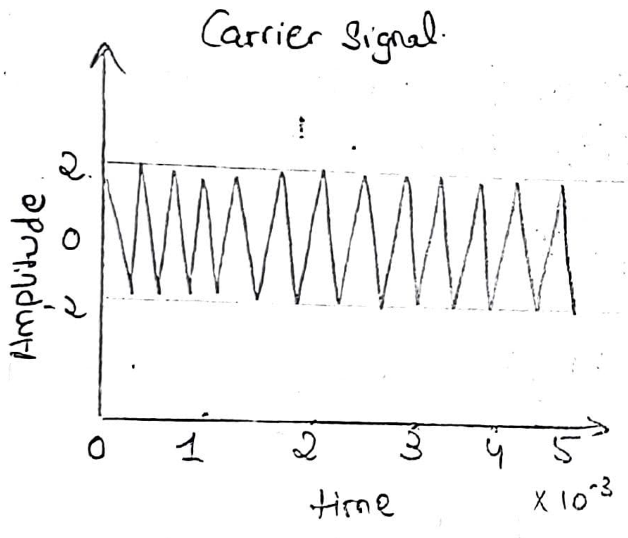
```
fs =100000;          % sampling frequency
ts = 1/fs;           % sampling interval

N = 10000;           % number 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
c =ac*cos(2*pi*fc*t);        % carrier signal
ch =ac*sin(2*pi*fc*t);       % hilbert transform for carrier
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;

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

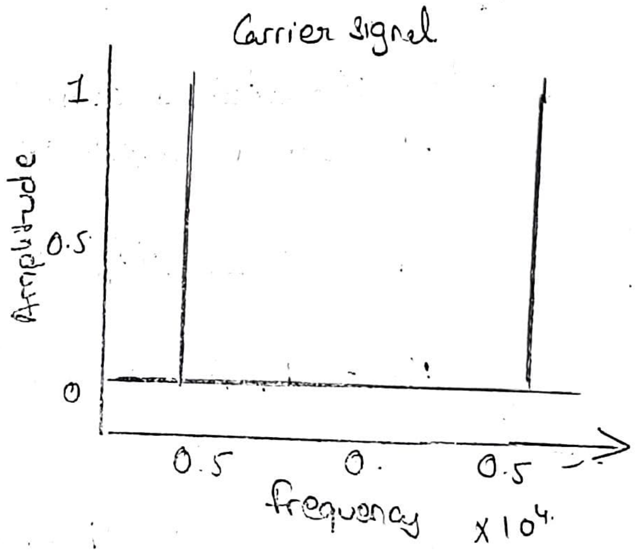
```
subplot (3,2,5);
plot (t, st, 'blue', 'linewidth', 1.5);
axis ([0 0.0005 -2.5 2.5]);
xlabel ('time');
ylabel ('amplitude');
title ('modulated signal');
grid on;

% spectrums 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 ('amplitude');
title ('carrier signal');
grid on;
```

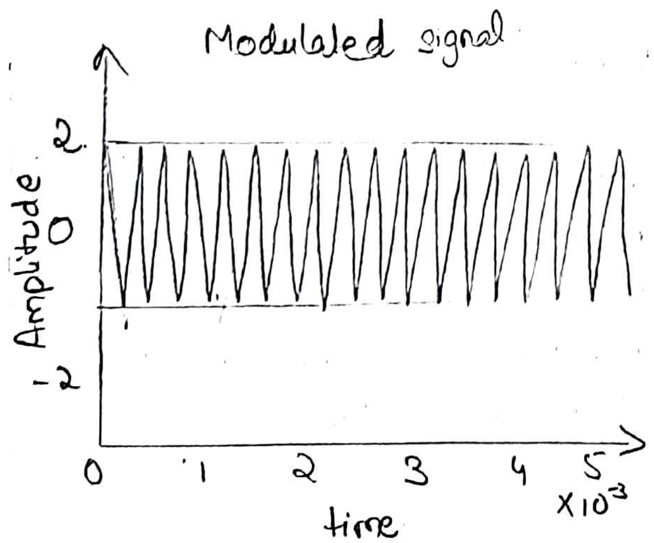
1) $f_m = 500$, $f_c = 5000$

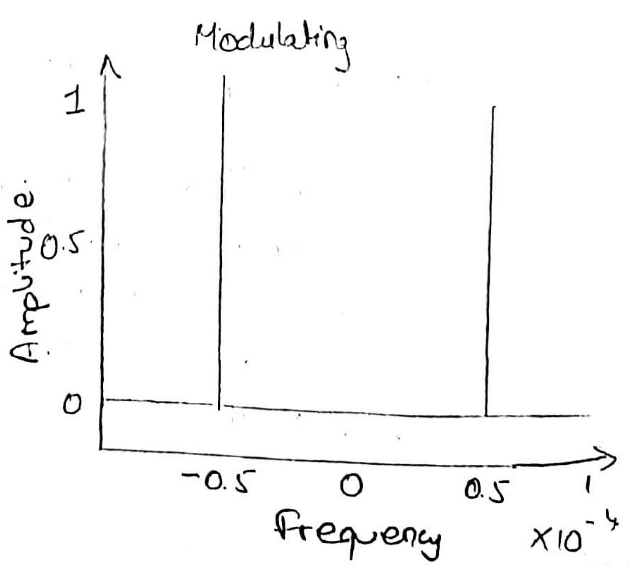




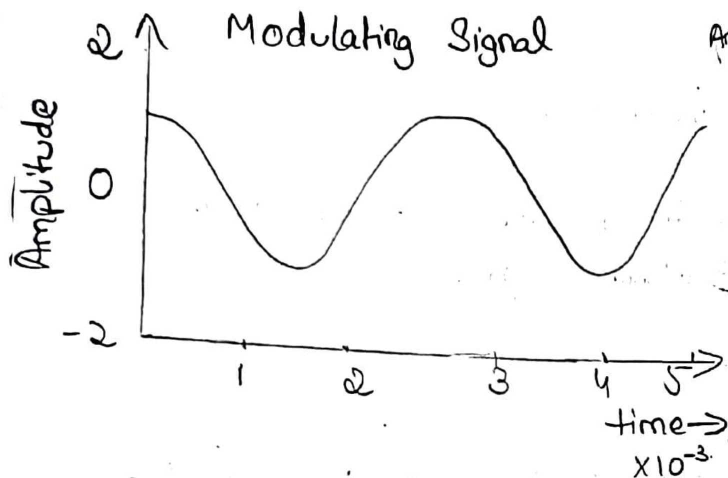


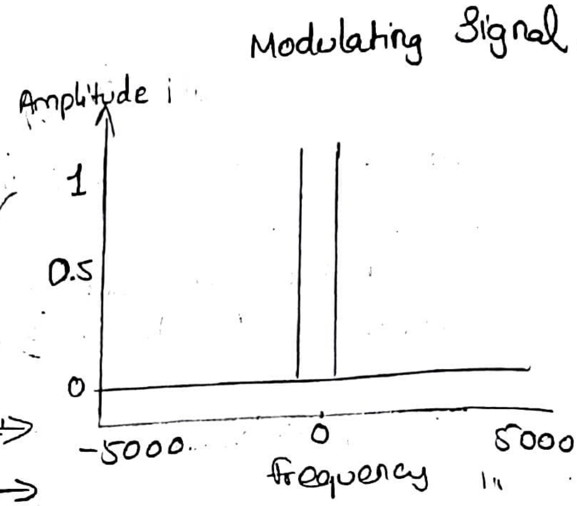


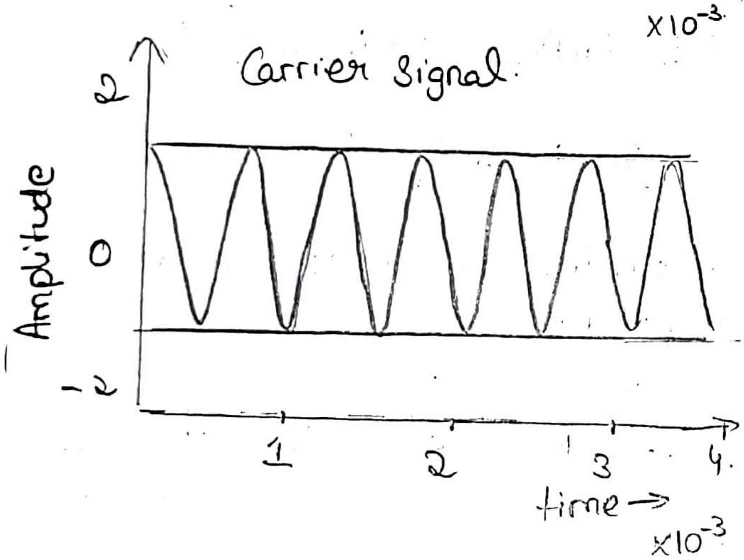


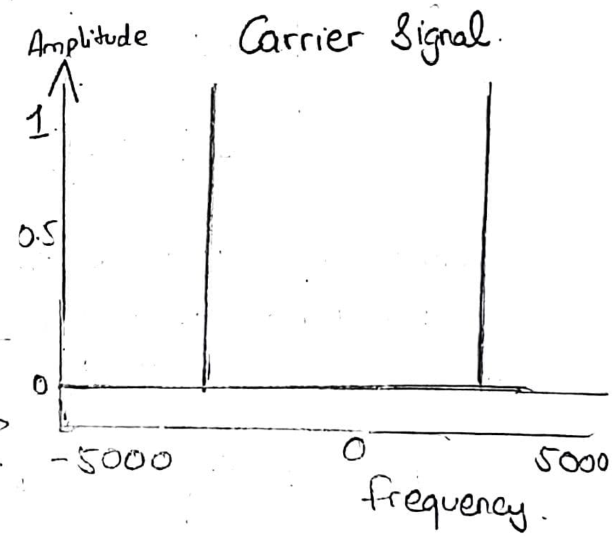


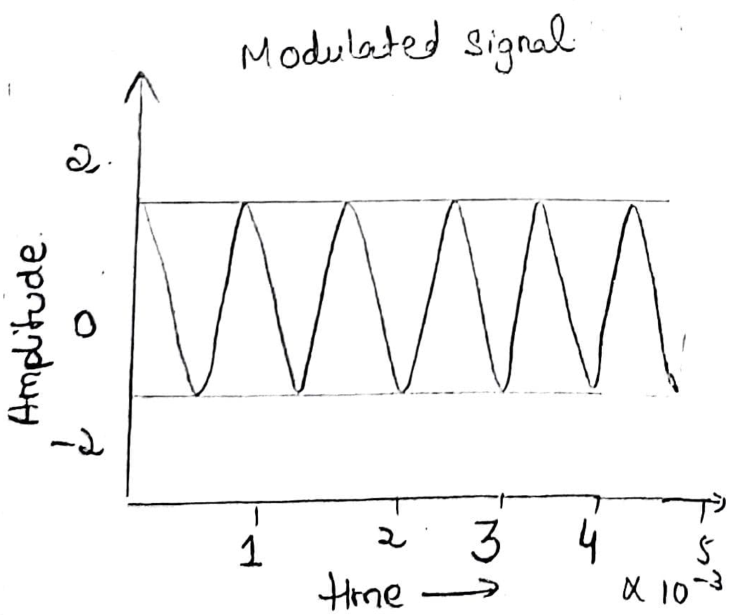
2) $f_m = 400$, $f_c = 3000$.

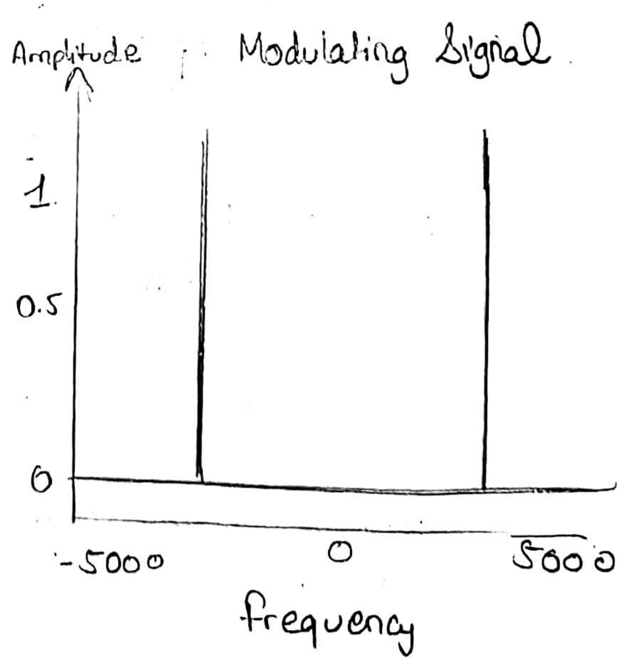






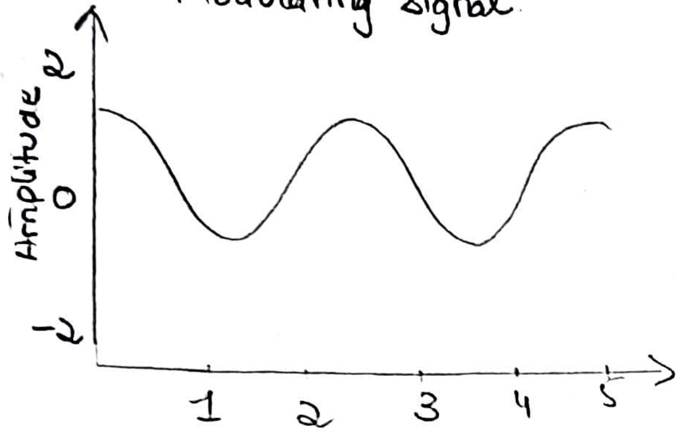




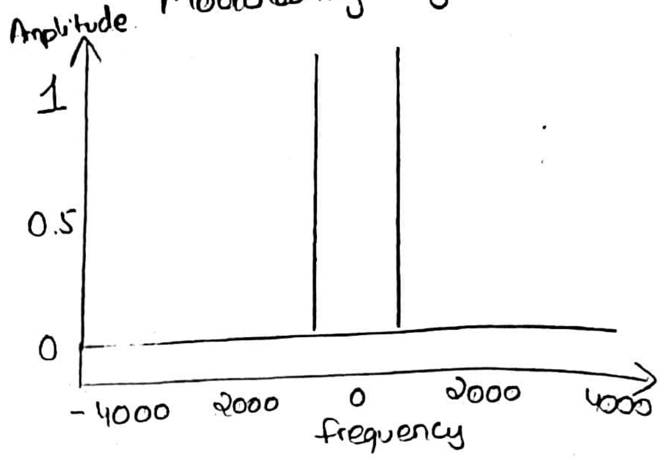


3) $f_m = 400$ $f_c = 2000$

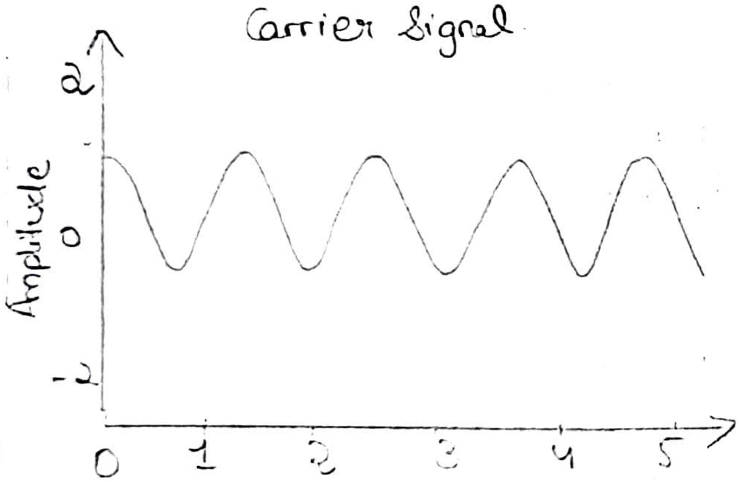
Modulating Signal



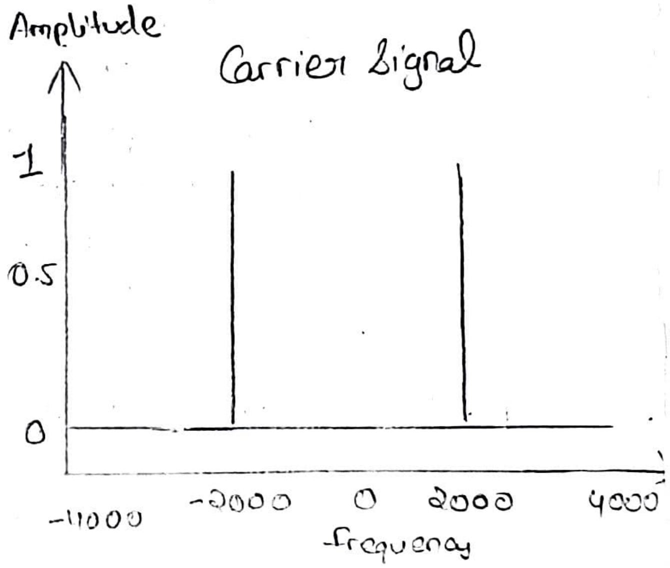
Modulating Signal



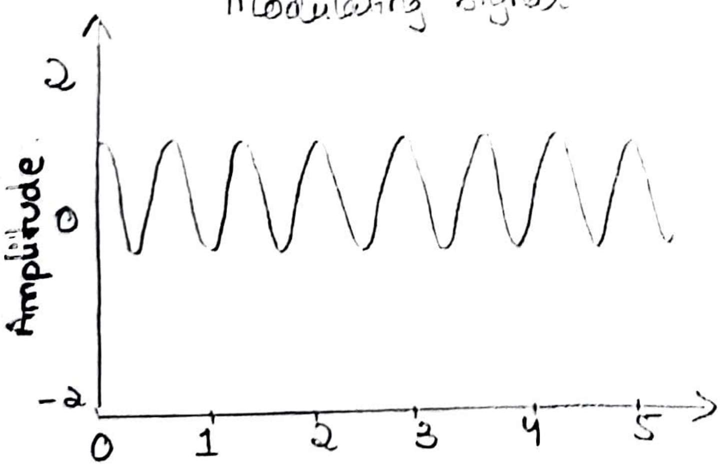
Carrier Signal



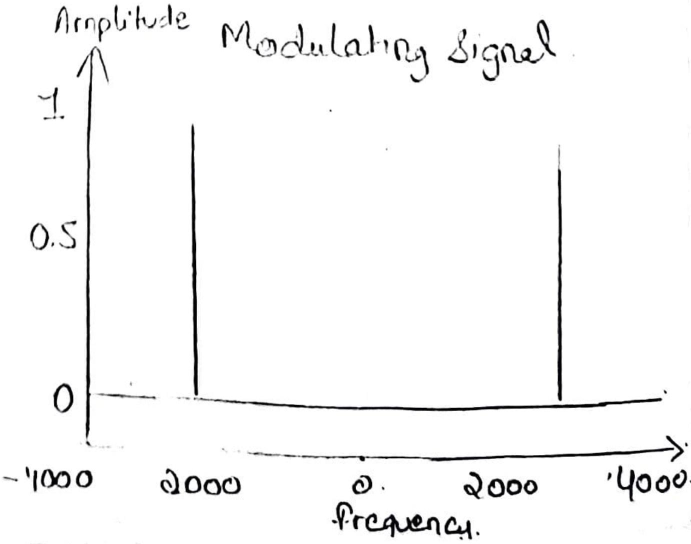
Carrier Signal



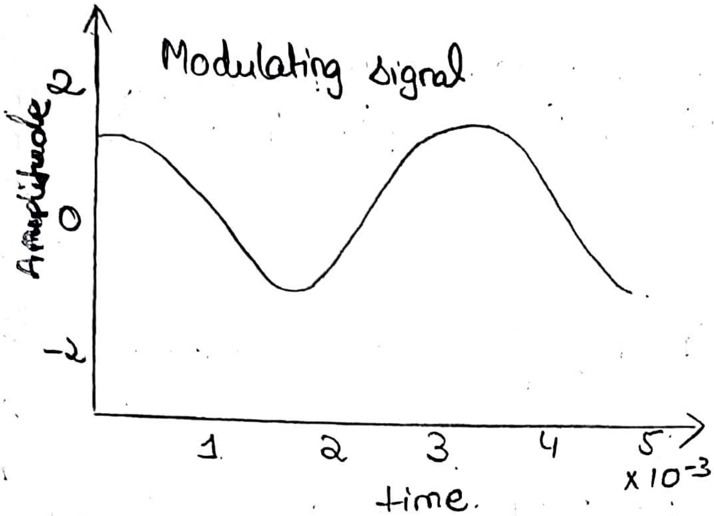
Modulating Signal

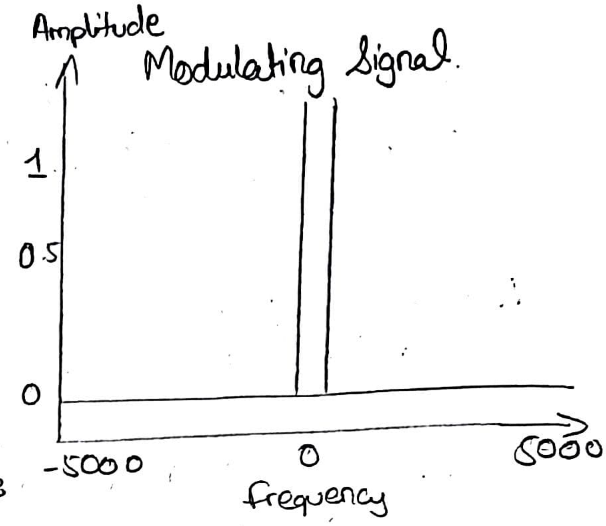


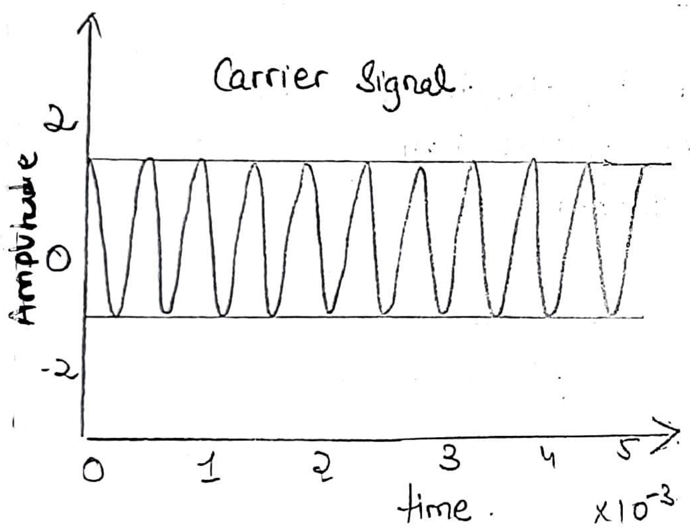
Modulating Signal

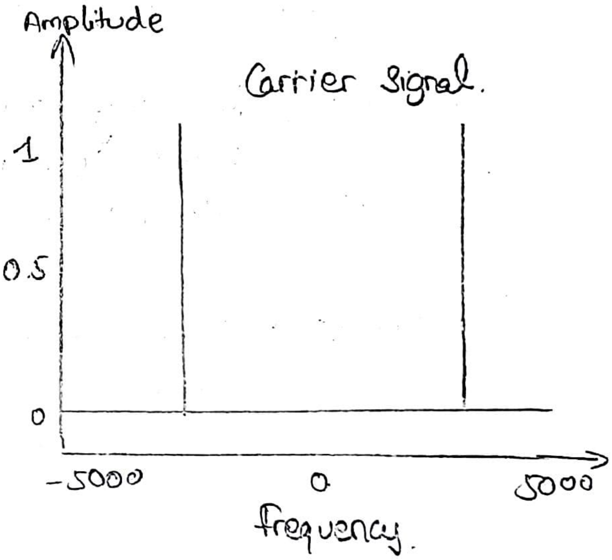


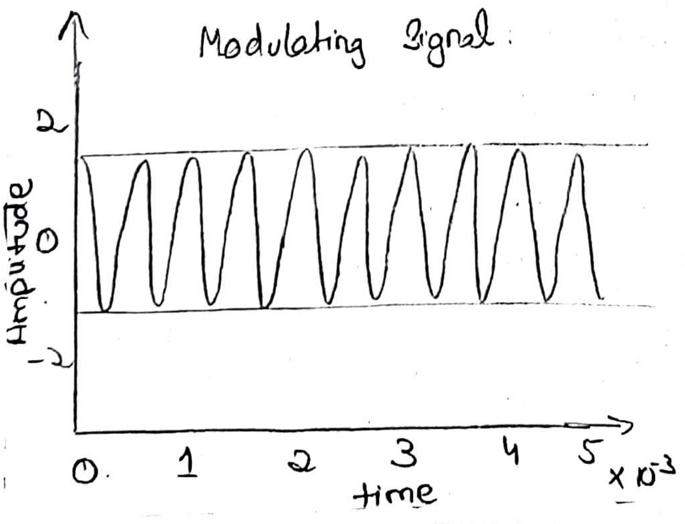
4) $f_m = 300$ $f_c = 3000$

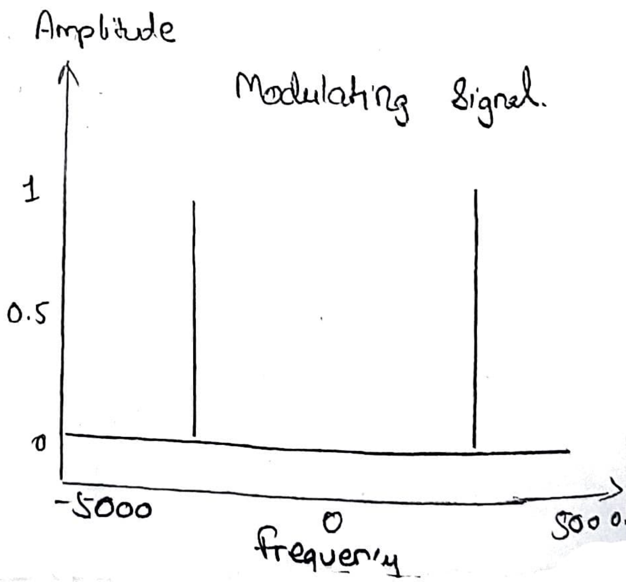






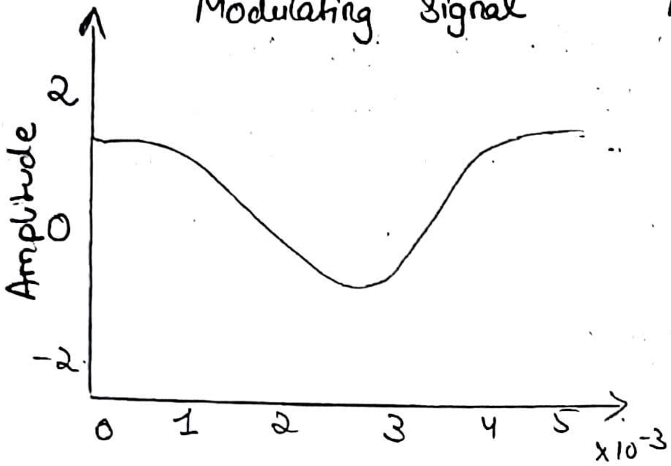




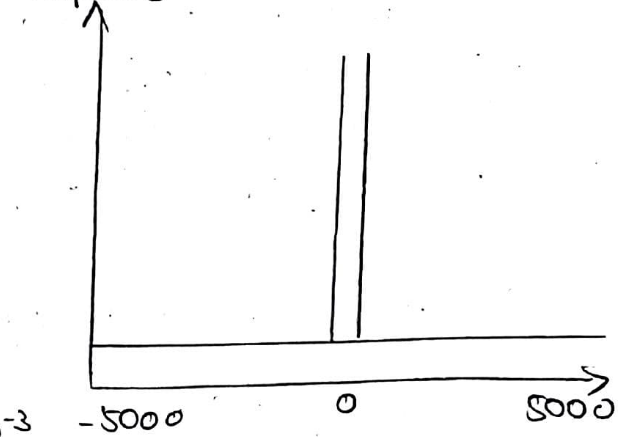


5) $f_m = 200$ $f_c = 3000$

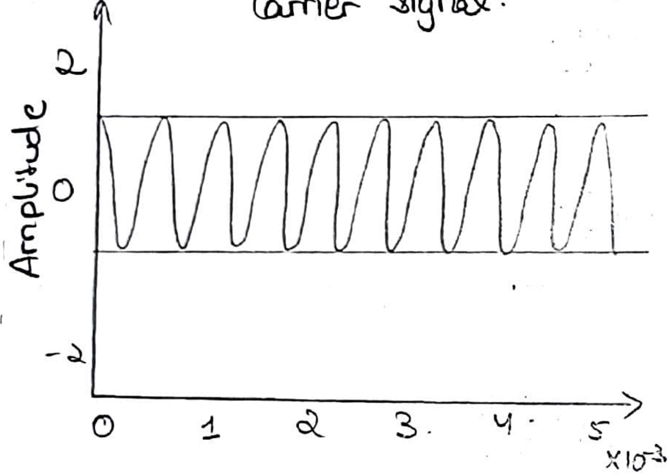
Modulating Signal



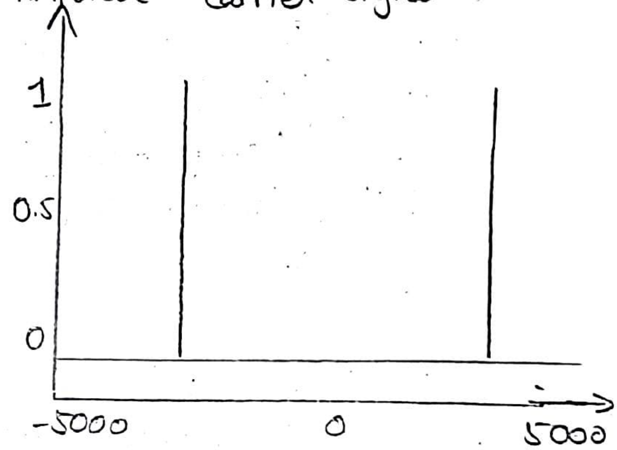
Modulating Signal



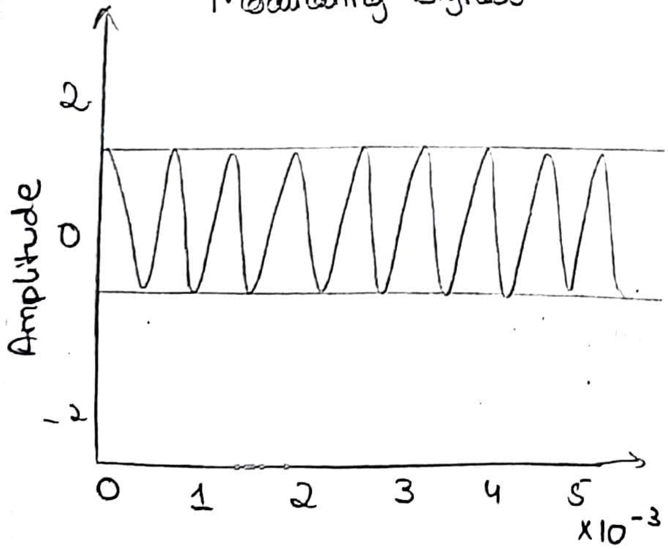
Carrier Signal.



Carrier Signal.



Modulating Signal



Modulating Signal

```
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-

Thus we stimulated program for SSB modulation & drew the message/carrier waveforms and resultant modulated signal in time domain and frequency domain

ADVANTAGES-

(a) Bandwidth or spectrum space occupied is lesser than AM and DSB signal
(b) Transmission of more signal is allowed
(c) Power is conserved
(d) High power signal can be transmitted
(e) Less amount of noise is present
(f) Signal fading is less likely to occur

DISADVANTAGES-

(a) The generation and detection of SSB signal is a complete process
(b) Quality of the signal gets affected unless the SSB transmitter and receiver have an excellent frequency stability

APPLICATIONS-

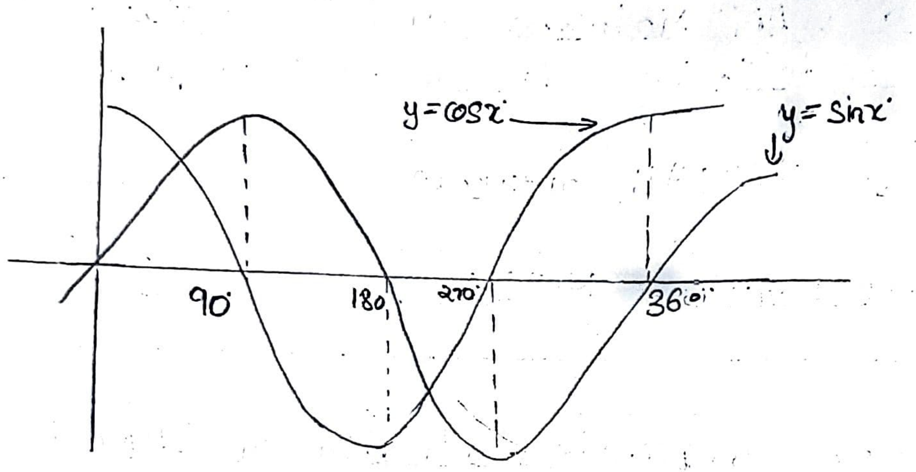
(a) For power saving requirements and low bandwidth requirements
(b) In land, air and mainline mobile communication

EXPERIMENT-9

AIM-
16 QAM Modulation and demodulation technique with constellation diagram and waveforms

APPARATUS-
MATLAB software

THEORY-
- 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 and the resultant output consist 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.
- The main aim of QAM 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



Quadrature = Sine wave + Cosine wave

signals in the same spectrum

Types of QAM

- A variety of forms of QAM are available which include
16 QAM
32 QAM
64 QAM
128 QAM
256 QAM

- Quadrature amplitude theory states that both amplitude and phase change 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 then sum them

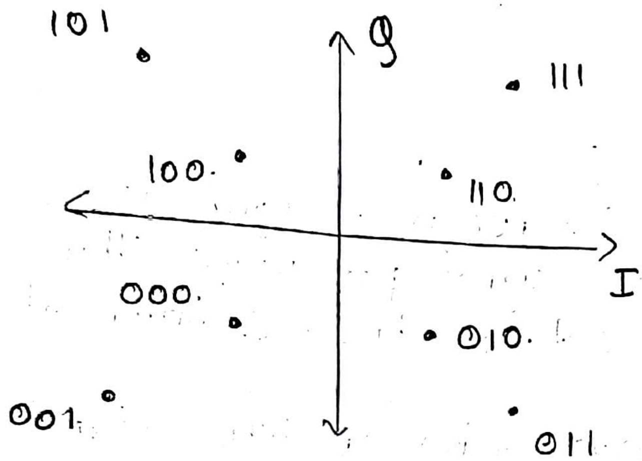
- The I and Q, signals can be represented by the equations below

$$I = A\cos(\psi)$$
$$Q = A\sin(\psi)$$

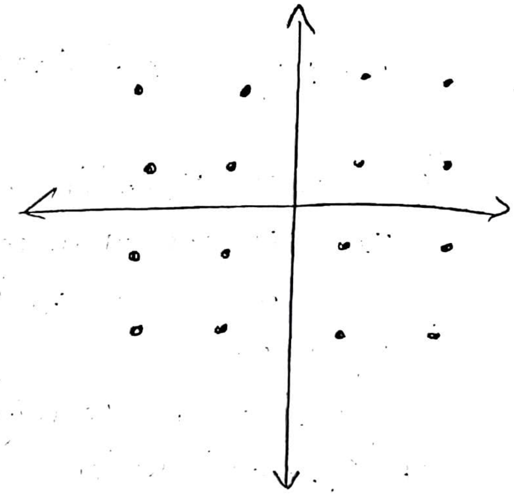
- These signal will not overlap with each other because they are orthogonal

- It is possible to transmit two DSB-SC signals with in a bandwidth of $2f_m$.
- Improves data rate
- Provide bandwidth efficiency
- The QAM demodulator is very much the reverse of the QAM modulator
- Gives better performance than SSB

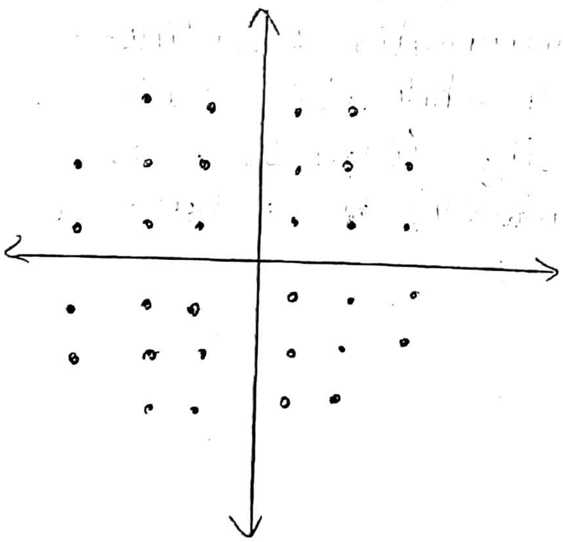
Types of QAM - CONSTELLATION DIAGRAMS



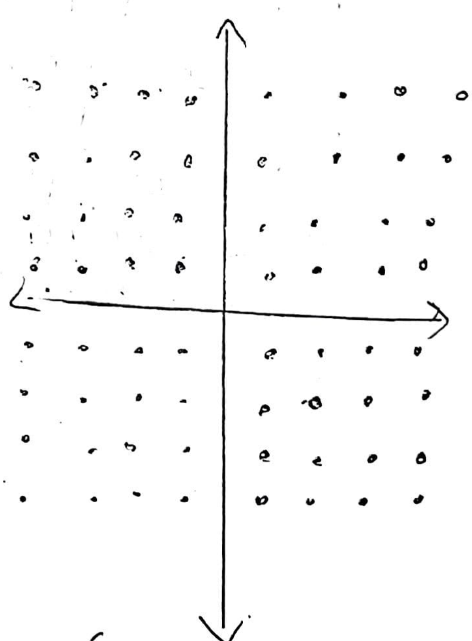
(8 QAM)



(16 QAM



(32 QAM)



(64 QAM)

- The signals enter the system, they are split and each side is applied to a mixer.

Bit Error Rate

- While higher order modulation rates are able to offer much faster data rates and higher levels of spectral efficiency for the radio communications system, this comes at a price.

- The higher order modulation schemes are considerably less resilient to noise and interference

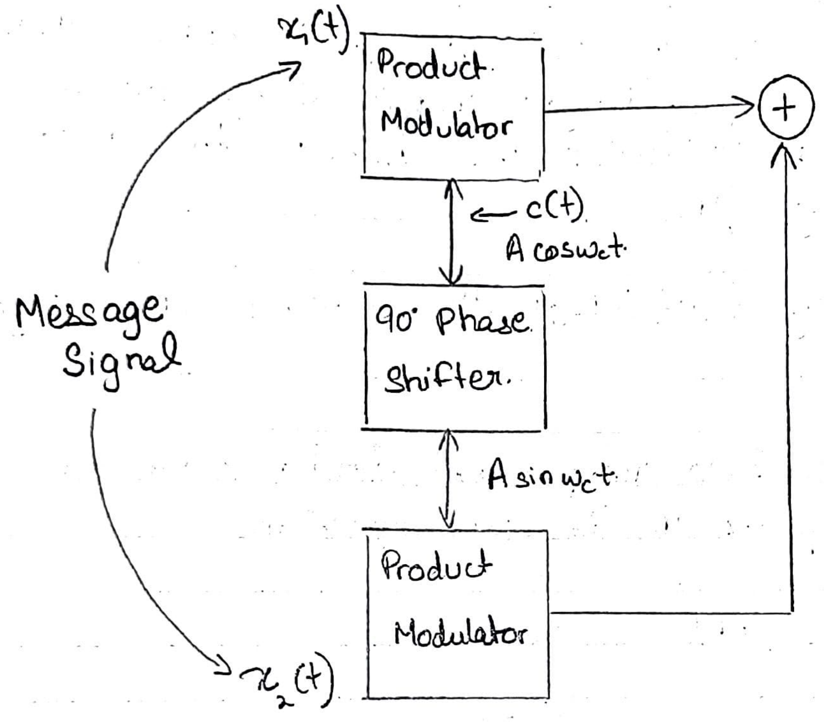
- Many radio communications systems now use dynamic adaptive modulation techniques. They sense the channel conditions and adapt the modulation scheme to obtain the highest data for the given conditions.

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

MATLAB CODE-

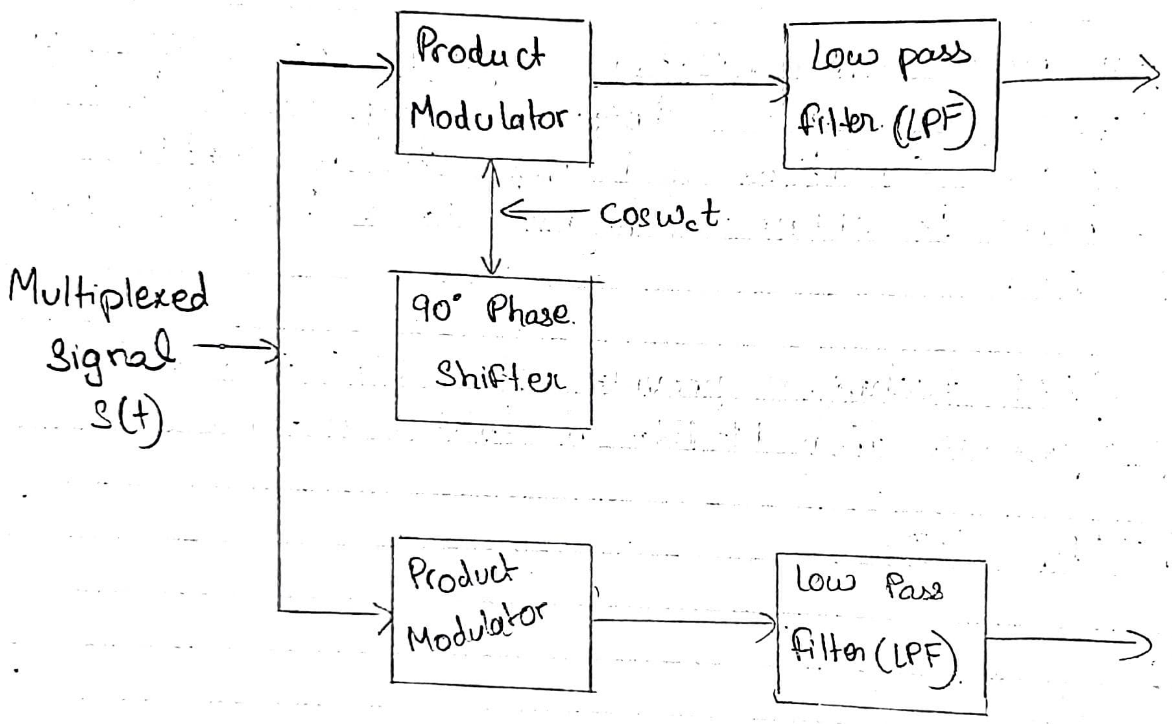
```
clc;
clear all;
close all;
M=16;
x=(0:M-1);
y=qammod(x,M);
Scatterplot(y);
Z=qamdemod(y,M,pi/4);
Scatterplot(z);
ber_1=[];
```

1) QAM Modulation.



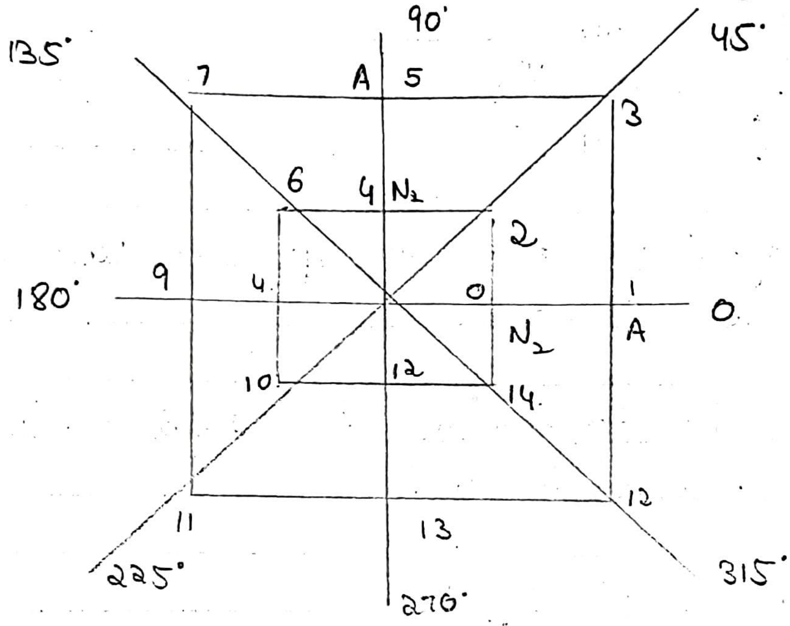
$$S(t) = x_1(t)\, A\cos w_c t + x_2(t)\, A\sin w_c t$$

QAM Demodulation:

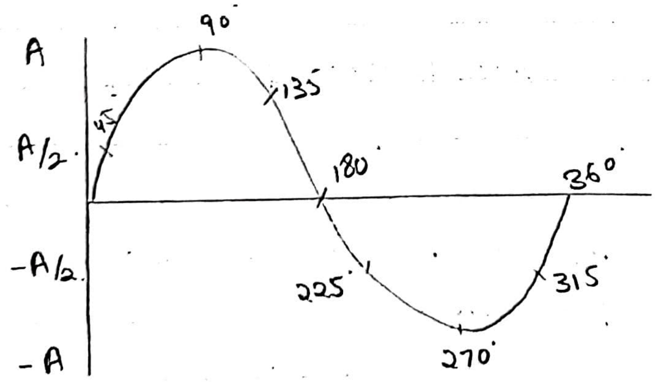


16 QAM Waveforms.

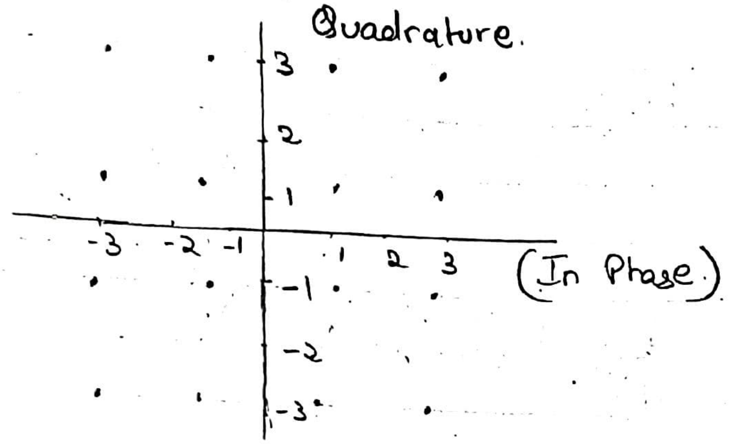
2 Amplitudes x 8 phase $(16 = 2^4 \Rightarrow 4 \text{ bits})$



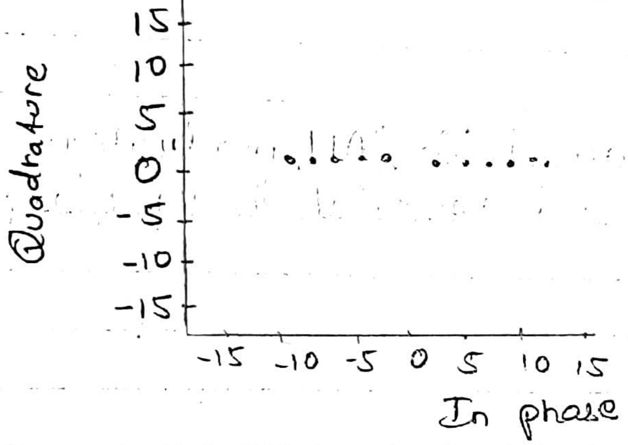
Phasor Diagram.



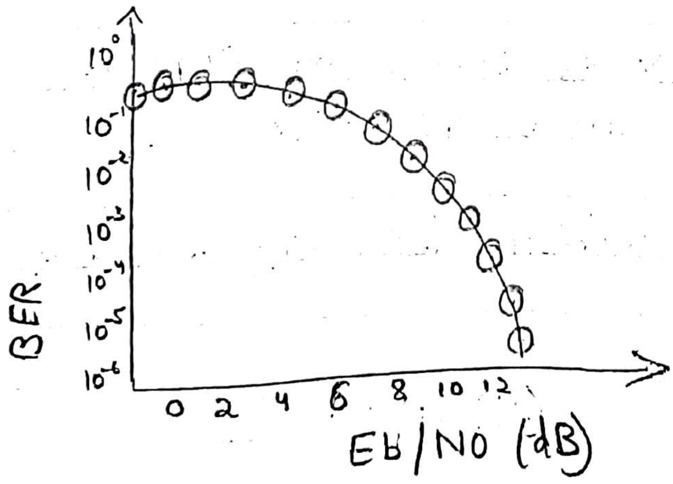
Scatter plot of 16-QAM Modulation:-



Scatter plot of 16-QAM Demodulation:-



BER of 16-QAM:-



```
for EbNOdB=0:20
    EbNO=10^(EbNOdB/10);
    ber(1/log2(M))*(2*(1-sqrt(1/M))*erfc(sqrt(3*log2(M)
            *EbNo)/(2*(M-1))));
    ber_1=[ber_1 ber];
end
EbNOdB=0:20;
figure
semilogy(EbNOdB, ber_1(1.), 'ro-');
xlabel('Eb/NO (dB)');
Ylabel('BER');
title('BER of 16-QAM');
axis([0 16 10^ 10^6]);
grid on;
```

CONCLUSION-

Successfully examined 16-QAM modulation and demodulation scheme and illustrated the input/output waveforms using MATLAB

ADVANTAGES-

- 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 is increased without increasing the band width

DISADVANTAGES -

- As states are more closer as shown in the figure, QAM modulation is more susceptible to the noise. Due to this QAM receivers of other modulation types

- As QAM uses amplitude component of signal to represent binary data linearly need to be maintained and hence linear amplifier is needed which consumes more power

APPLICATIONS -

- Quadratic 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

- Ultra-high capacity Microwave Backhaul systems also use 1024-QAM

EXPERIMENT-10

PULSE CODE MODULATION AND DEMODULATION

AIM-

To demonstrate the pulse code modulation (PCM) and demodulation technique show the sampled quantized/encoded and decoded time-domain signal for different bit codes. Show the input/output waveforms using MATLAB code/ Simulink in virtual mode

SOFTWARE - Matlab

THEORY-

PCM (Pulse code modulation) is a techinque that is used to convert an analog signal to digital signal. In this the amplitude of an analog signal is converted to a binary value represented as a series of pulses. PCM is a preferred method of communication within the public switched telephone network (PSTN)

PCM is determined by two following steps -

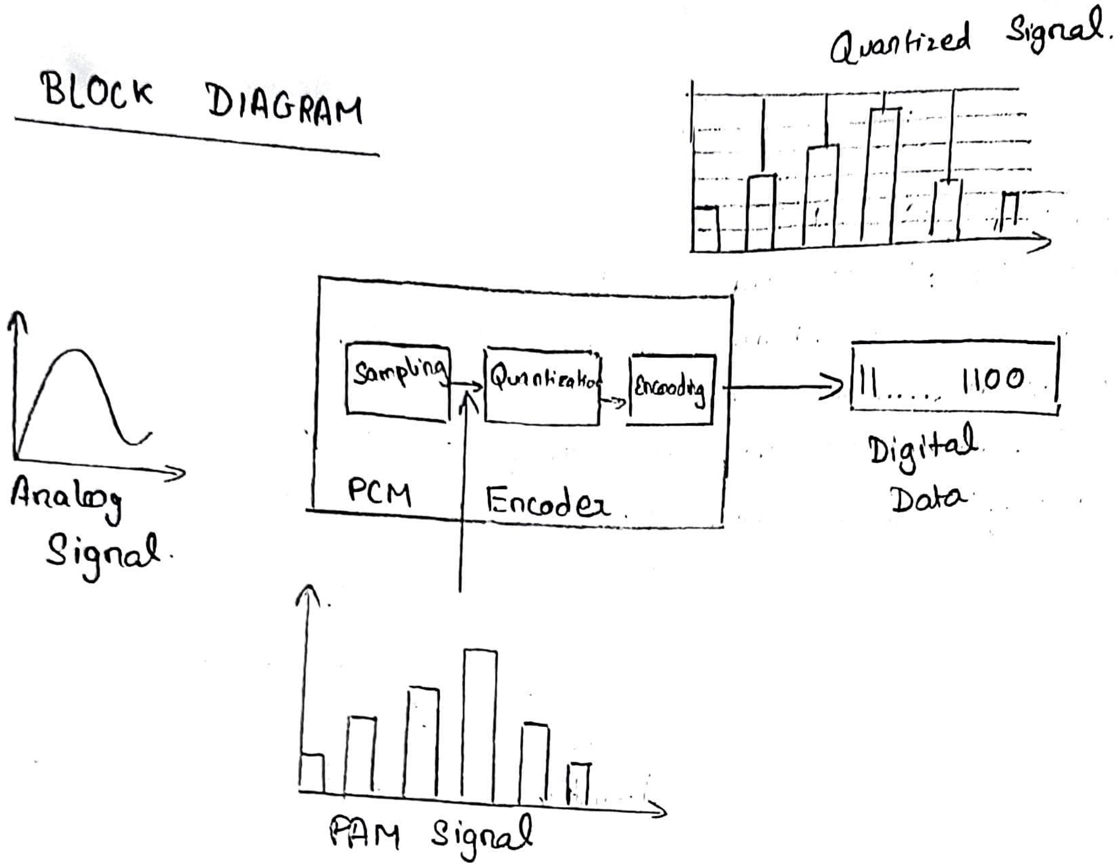
Sampling rate - which is the number of times per second that samples are taken

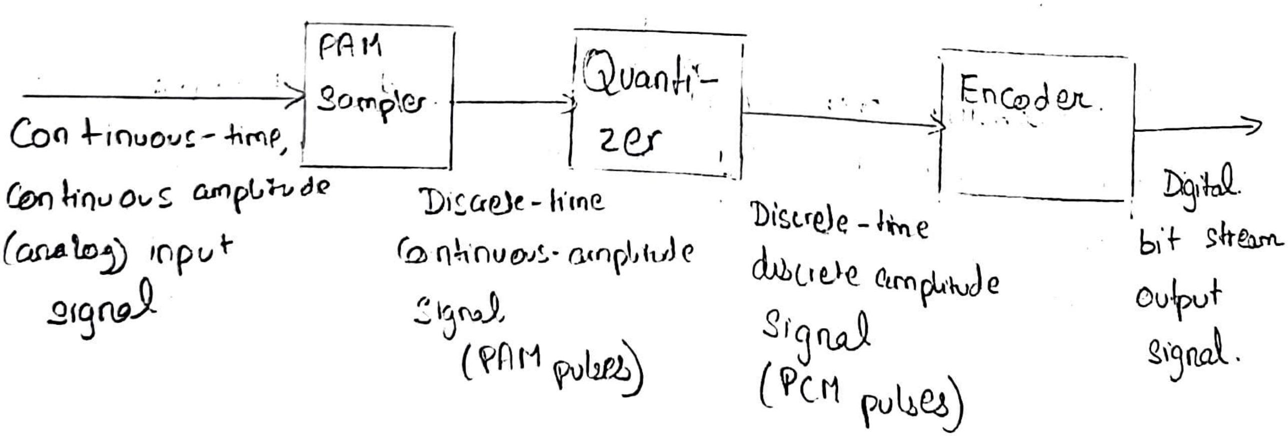
Bit depth - Which determines the number of possible digital values that can be used to represent each sample.

Hence PCM resembles output of binary sequence

The transmitter section of a pulse code modulator circuit consists of sampling, quantizing and encoding.

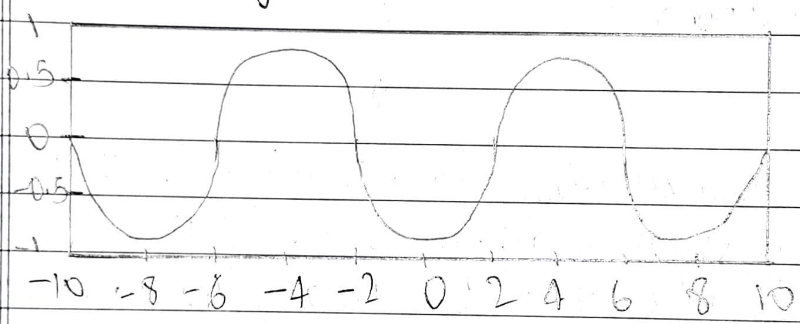
BLOCK DIAGRAM



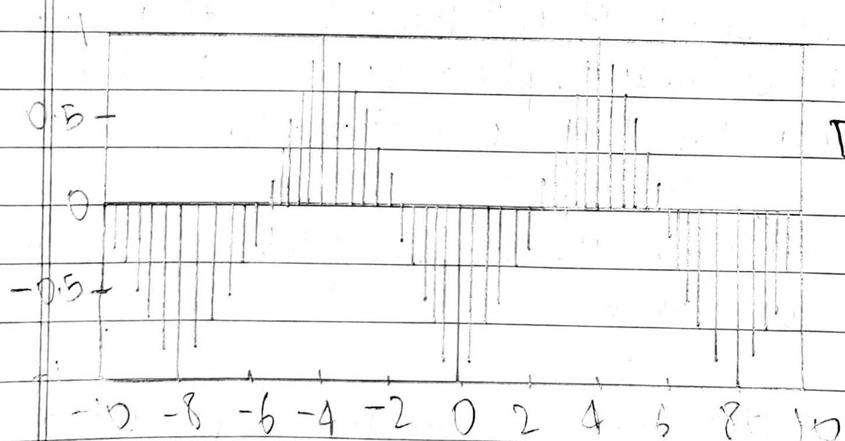


Sampling –

- The samples extracts of a continuous signal
- Sampler produces samples that are equivalent to the instantaneous value of the continous signal at the specified various points
- The sampling generates flat-top pulse amplitude modulated (PAM) signal



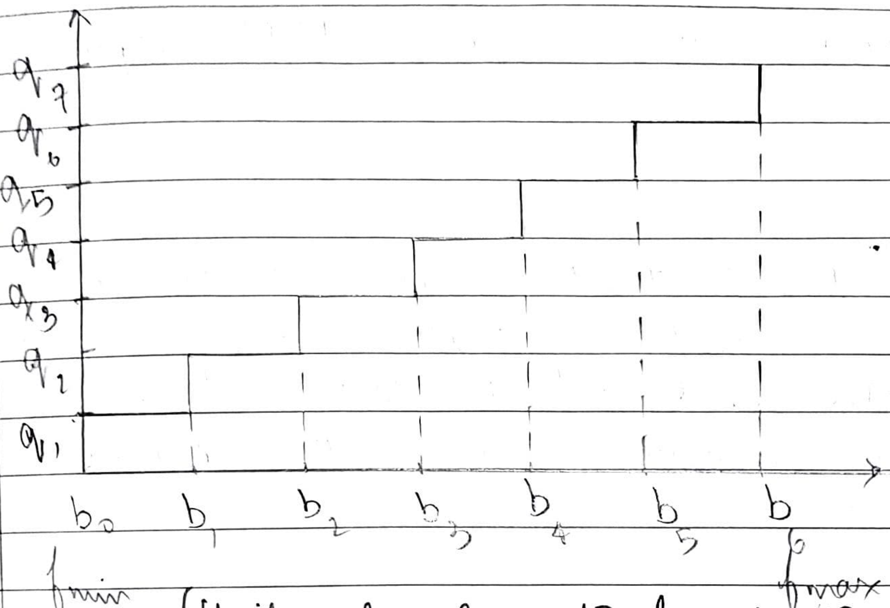
Analog signal



Discrete time signal

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 sample values with the nearest quantization values



(Uniformly quantized signal)

Encoder -

An encoder performs the conversion of the quantized signal into binary codes. This unit generates a digitally encoded signal which is a sequence of binary pulses that acts as the modulated output.

Process in PCM transmitter -

- In PCM transmitter, the signal $x(t)$ is first passed through the low pass filter of cut off frequency f_m Hz. This low pass filter blocks all frequency components above f_m Hz.

- The sample and hold circuit then samples this signal at the rate of f_s sampling frequency f_s is selected sufficiently above Nyquist rate to avoid aliasing. The output from the sample and hold circuit is denoted by $x(nT_s)$.

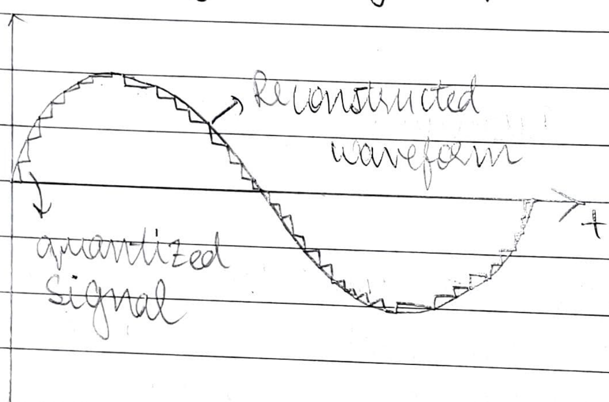
- The signal $x(nT_s)$ is discrete in time and continous in amplitude

- A q-level quantizer compares input $x(nT_s)$ with its fixed digital levels

- Quantized signal is then encoded in PCM output using encoder

PCM receiver-

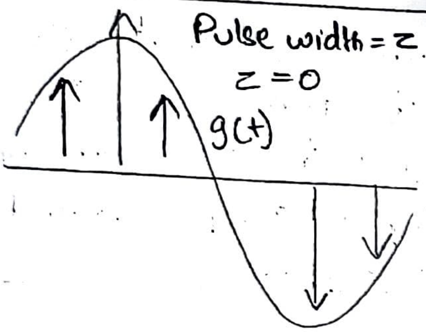
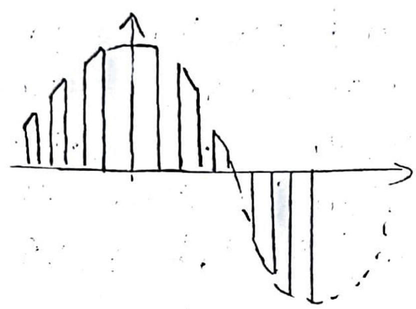
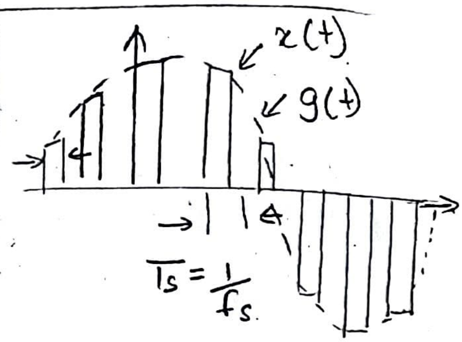
The process done at the transmitter is somewhat reversed at the receiver in order to generate the original analog message. The figure below shows the reconstruction of the analog message signal at the receiver



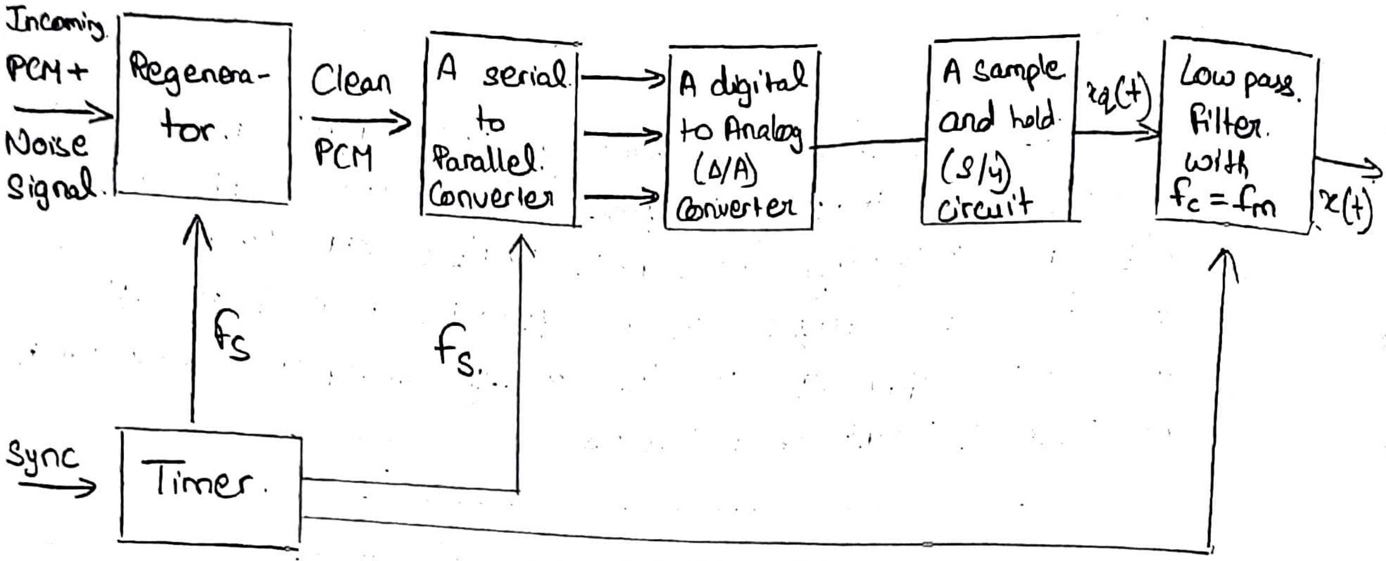
Reconstruction of analog signal at the receiver

Matlab code-

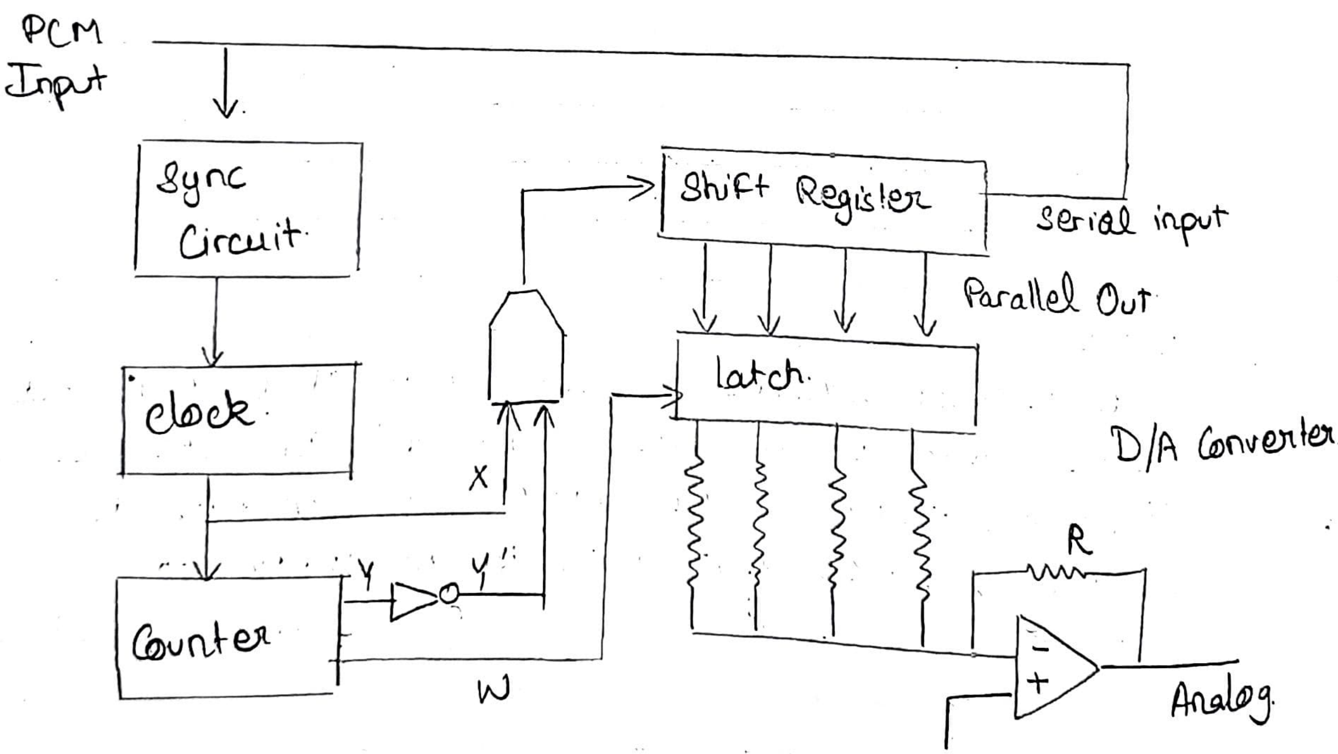
```
% sampling
clc;
clear all;
close all;
n= input('Enter n value for n-bit PCM system: ');
n1= input('Enter number of samples in a period: ');
l=2^n;
%% signal generation
% y = 8* sin (x); % Amplitude of signal is 8V
% x = 0:1/100:4* pi;
% subplot (2,2,1);
```

Instantaneous Sampling.	Natural Sampling.	Flat - Top Sampling.
		
• It is not a practical Method.	• This method is used practically.	• This method is also used practically
• Sample rate is infinity.	• Sample rate satisfies. Nyquist criteria	• Sample rate satisfies Nyquist criteria.

PCM Receiver Block Diagram.



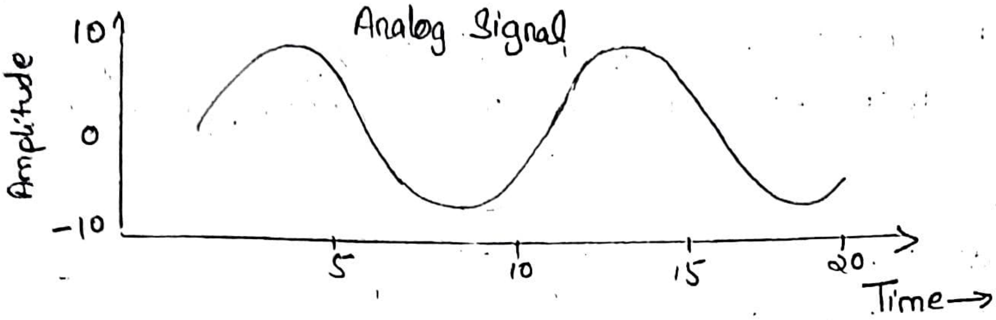
PCM Demodulator.

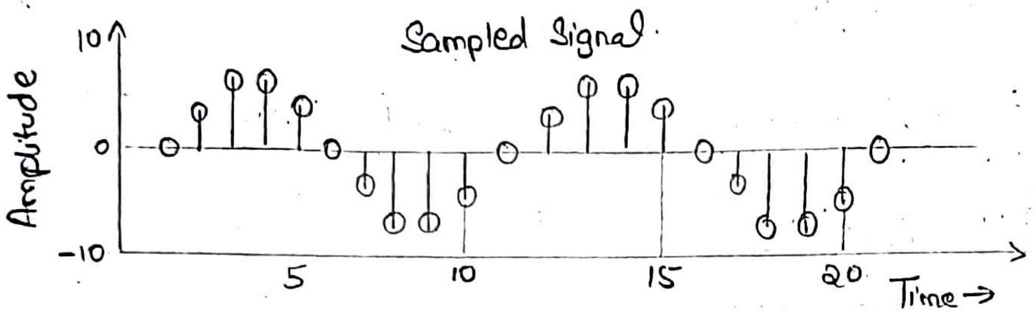


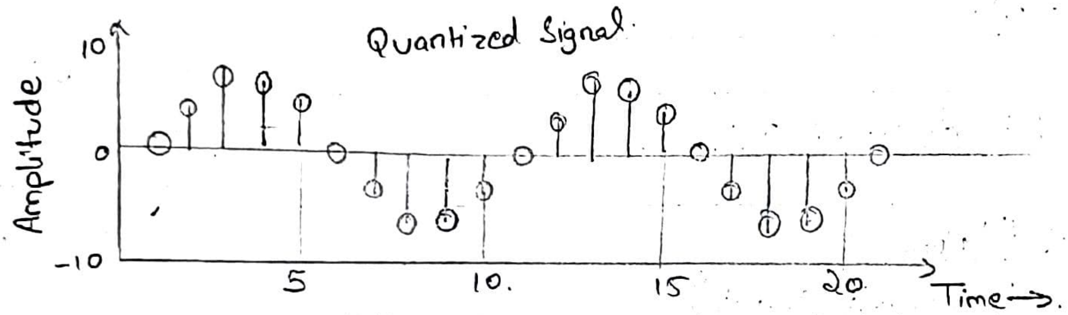
```
%plot(x,y); grid on;
% sampling operation
x=0:2*pi/n1=4*pi;
s = 8*sin(x);
subplot(3,1,1);
plot(s);
title('analog signal');
ylabel('amplitude -->');
xlabel('Time --->');
subplot(3,1,2);
stem(s); grid on; title('sampled signal');
ylabel('amplitude -->'); xlabel('Time -->');
% Quantization process
Vmax = 8;
Vmin = Vmax
del = (Vmax - Vmin)/L   % level between Vmin and Vmax
                          with % difference of del
part = Vmin:del:Vmin;
code = Vmin-(del/2):del:Vmax+(del/2);
                    % contain quantized values
[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
```

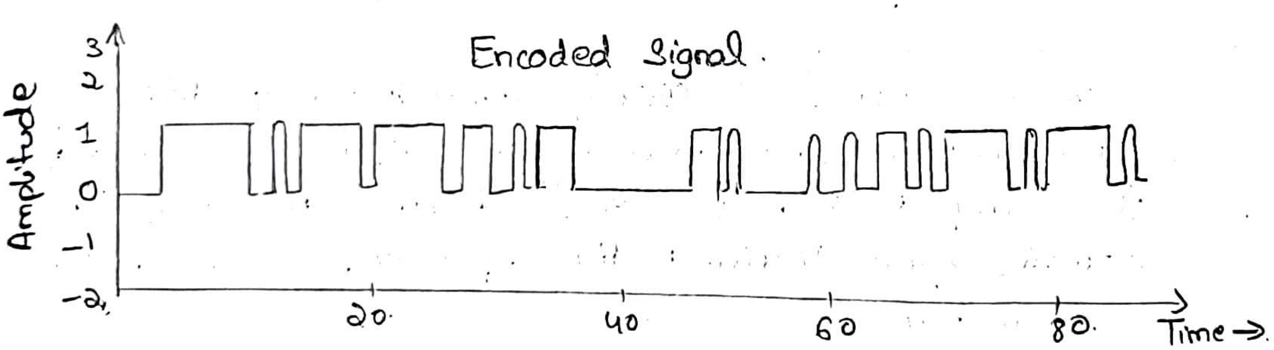
```
subplot (3,1,3); %Display the quantize values
stem (q); grid on;
title ('Quantized signal');
ylabel ('amplitude -->');
xlabel ('Time -->');

% 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 a coded row vector
        j = j+1;
        k = k+1;
    end
    i = i+1;
end
subplot (2,1,1); grid on
stairs (coded); % Display the encoded signal
axis ([0 100 -2 3]); title ('Encoded signal');
ylabel ('Amplitude -->');
xlabel ('Time -->');

% Demodulation of PCM signal
qint = reshape (coded, n, length (coded)/n);
index = bi2de (qint, 'left-msb');
% Getback the index in decimal form
q = del * index + Vmin + (del/2);
% Getback quantized values
```

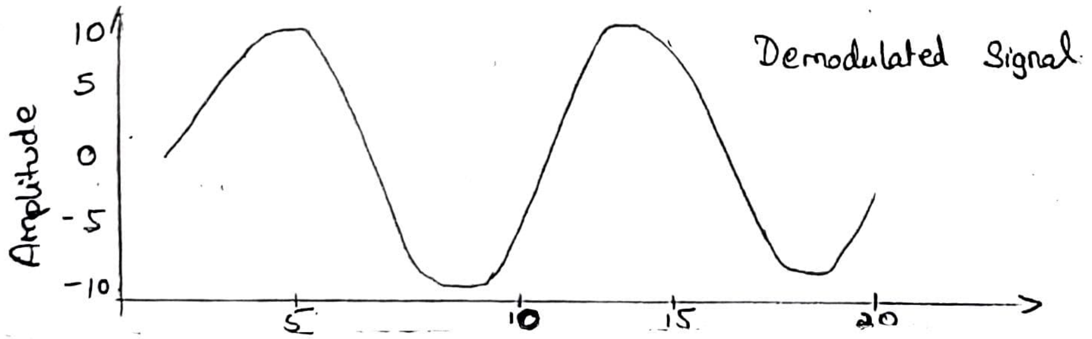
(1) $n = 6$.
no. of samples = 10



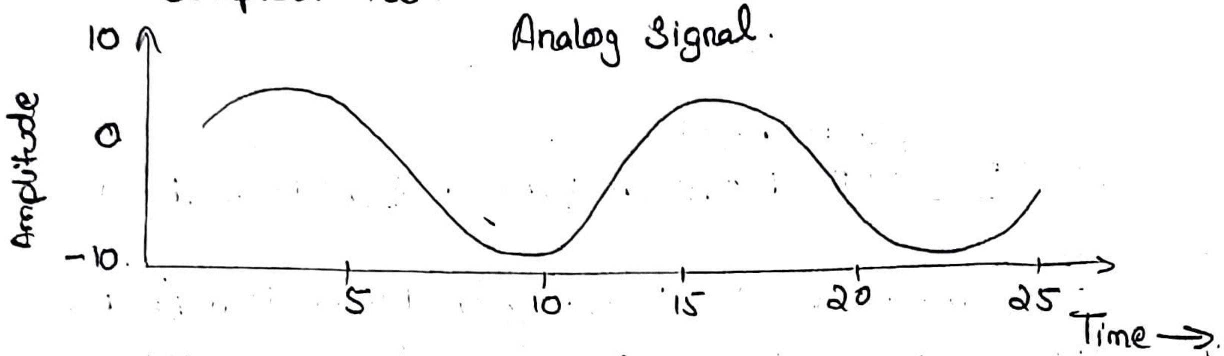


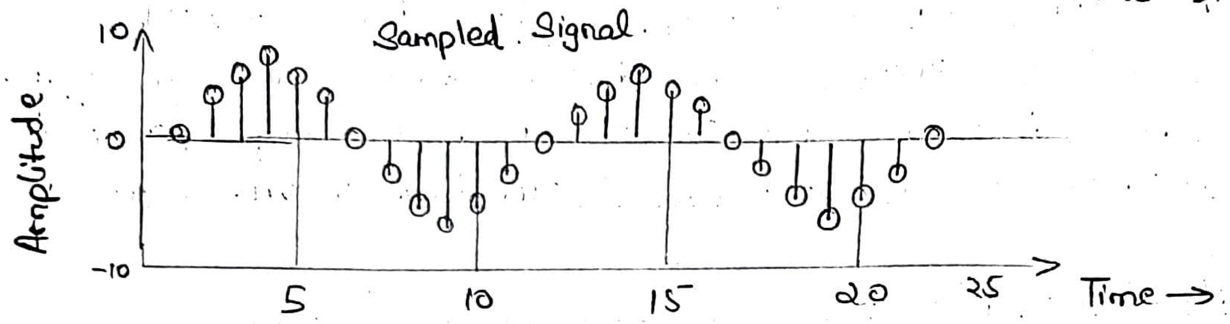


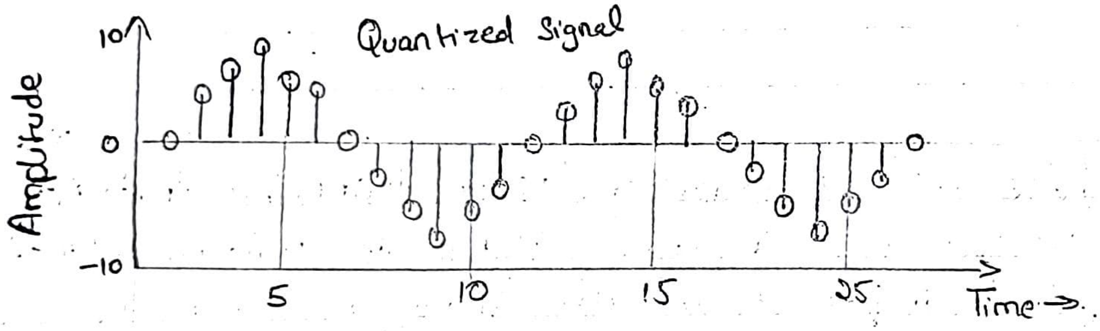


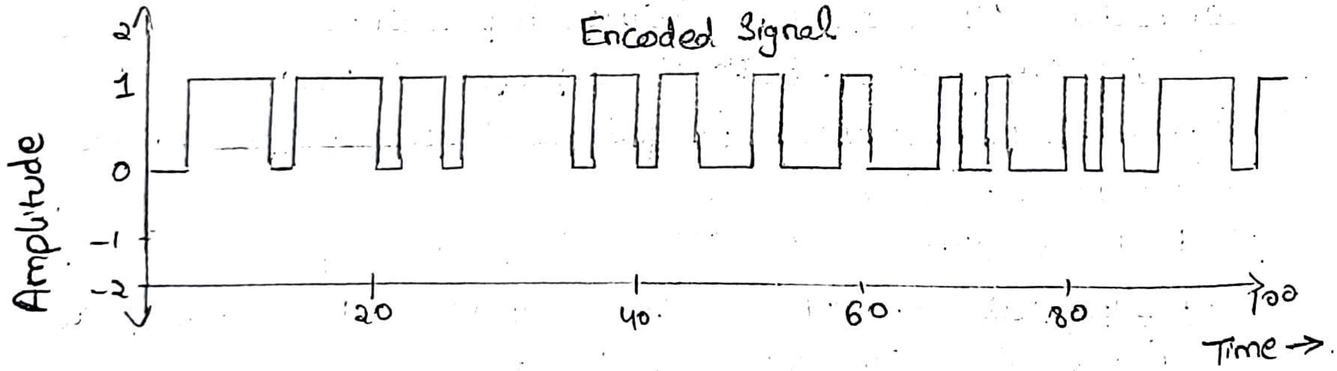


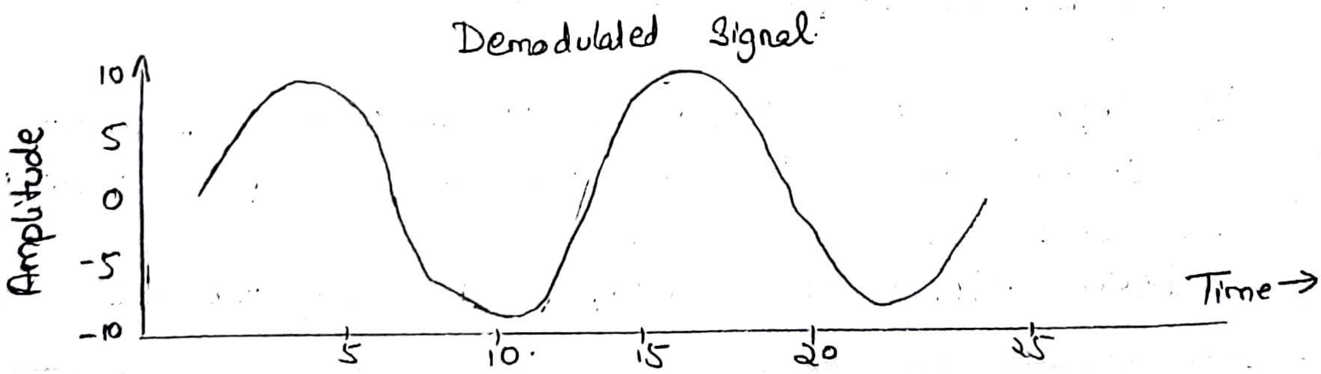
2) n=8.
no. of samples:- 12.







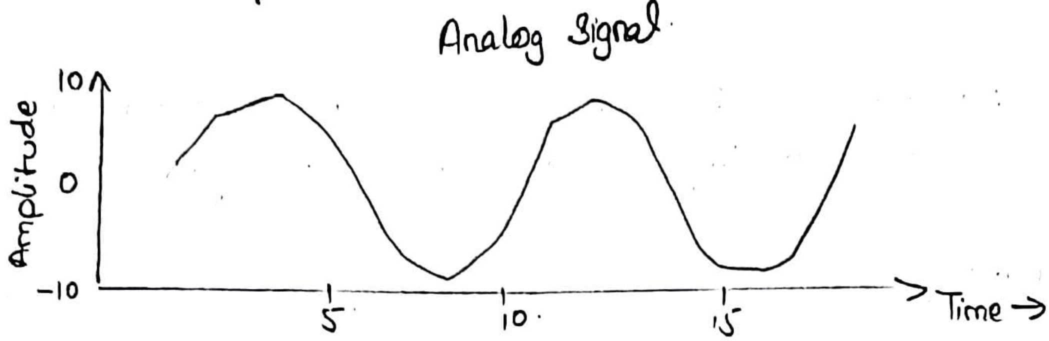


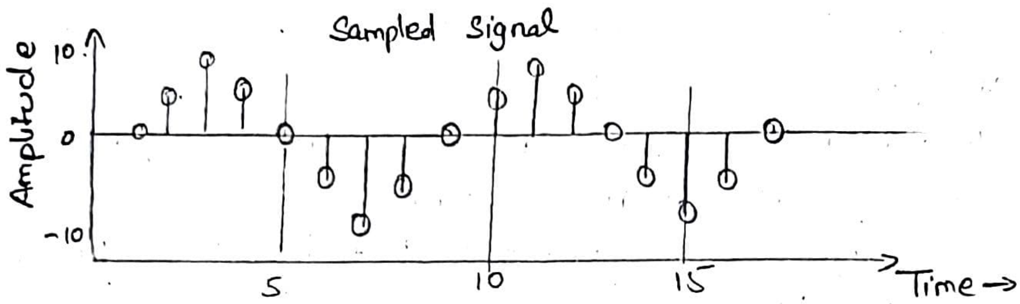


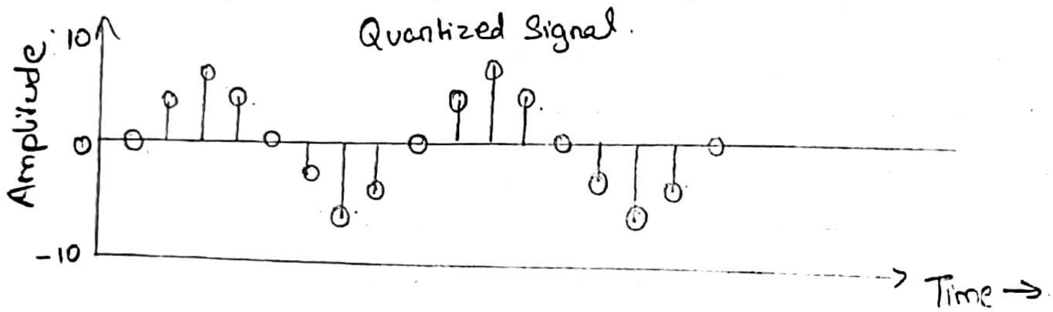
3) n=4

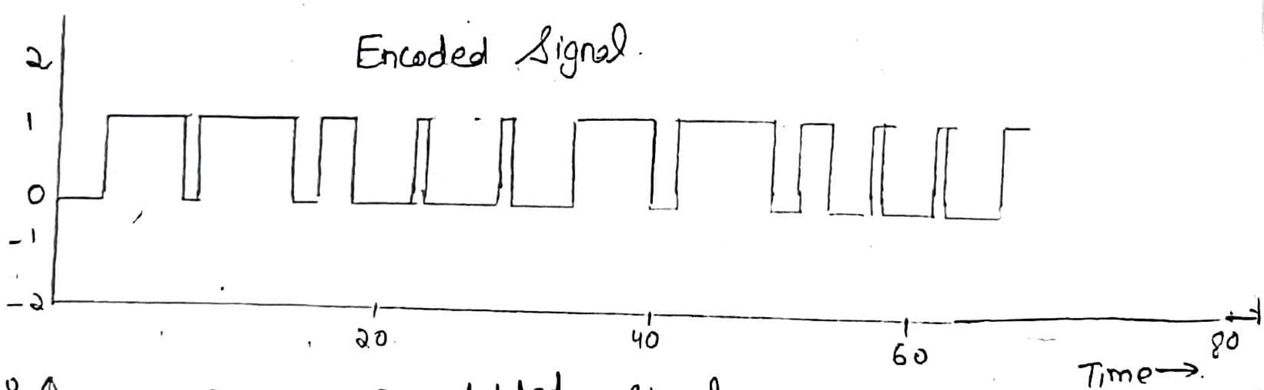
no. of samples = 8.

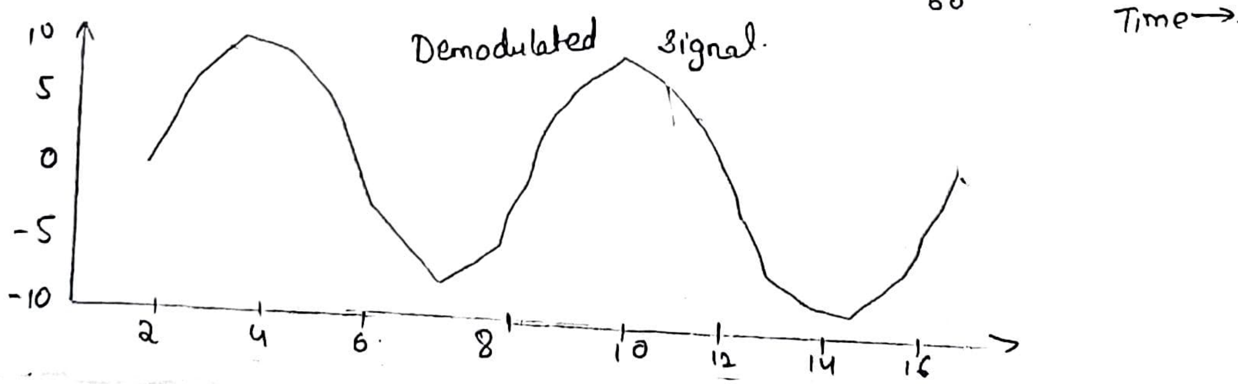
127











```
subplot (2, 1, 2); grid on;
plot (q); % plot demodulated signal
ylabel ('Amplitude -->');
xlabel ('Time -->');
```

ADVANTAGES OF PCM-

1. Immune to channel induced noise and distortion
2. Repeaters can be employed along the transmitting channel
3. Encoders allow secured data transmission
4. It ensures uniform transmission quality.

DISADVANTAGES OF PCM-

1. Pulse code Modulation increases the transmission bandwidth
2. A PCM system is somewhat more complex than another system

APPLICATIONS-

1. In compact disk
2. Digital telephony
3. Digital radio applications

CONCLUSION-

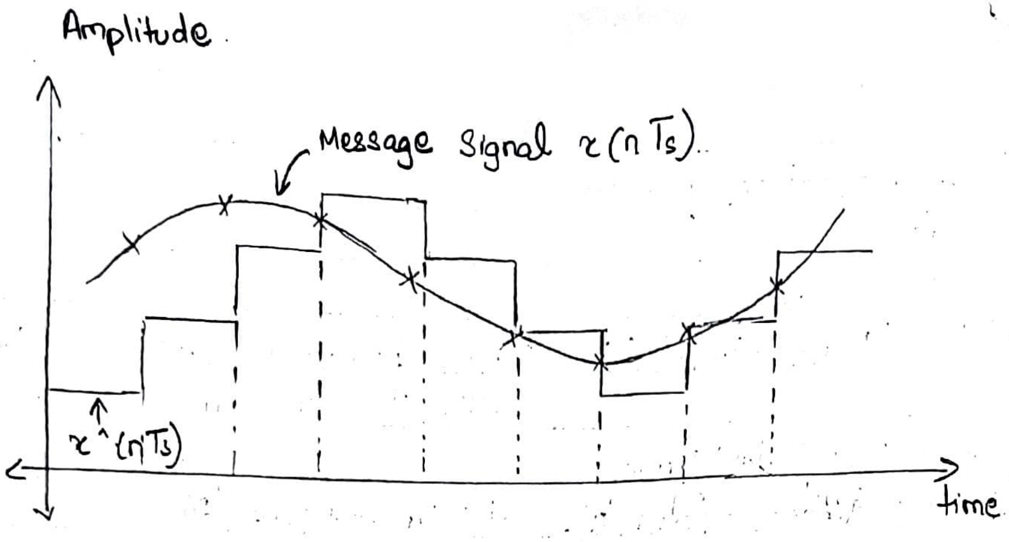
Hence, we demonstrated Pulse Code Modulation (PCM) and demodulation technique and showed the sample quantized/encoded and decoded time-domain signal for different bit codes

EXPERIMENT-11
DELTA MODULATION

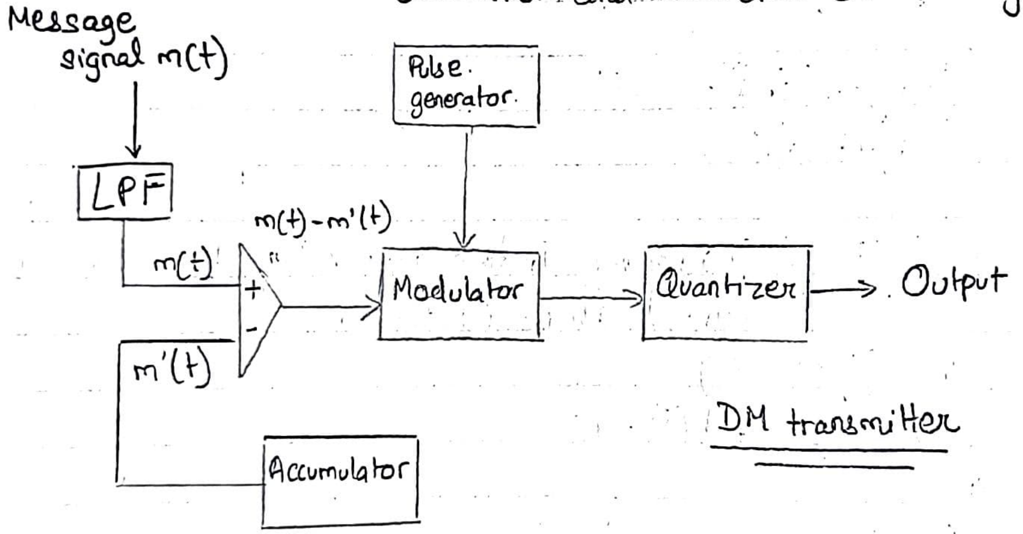
AIM-
To demonstrate the delta modulation (DM) and demodulation technique. Show the sampled quantized/ encoded and decoded time-domain signal. Show the input / output waveforms using Matlab code/ Simulink in virtual mode.

APPARATUS-
MATLAB software

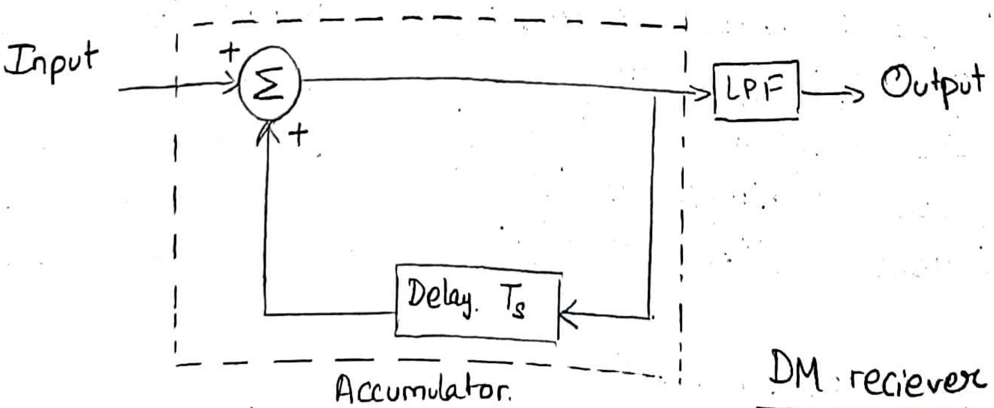
THEORY-
Delta modulation is a technique used to convert analog to digital and digital to analog signal. In this modulation signal is sent in differential form, the data is encrypted / transmitted in 1 bit. The analog signal is approximated with series of segments and each segment is compared to original analog to determine the change in relative amplitude. Hence only change 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 previous approximated sample is quantized into two levels i.e. $\pm\Delta$ (delta). This is used for voice transmission.



Generation and detection of DM signal:-

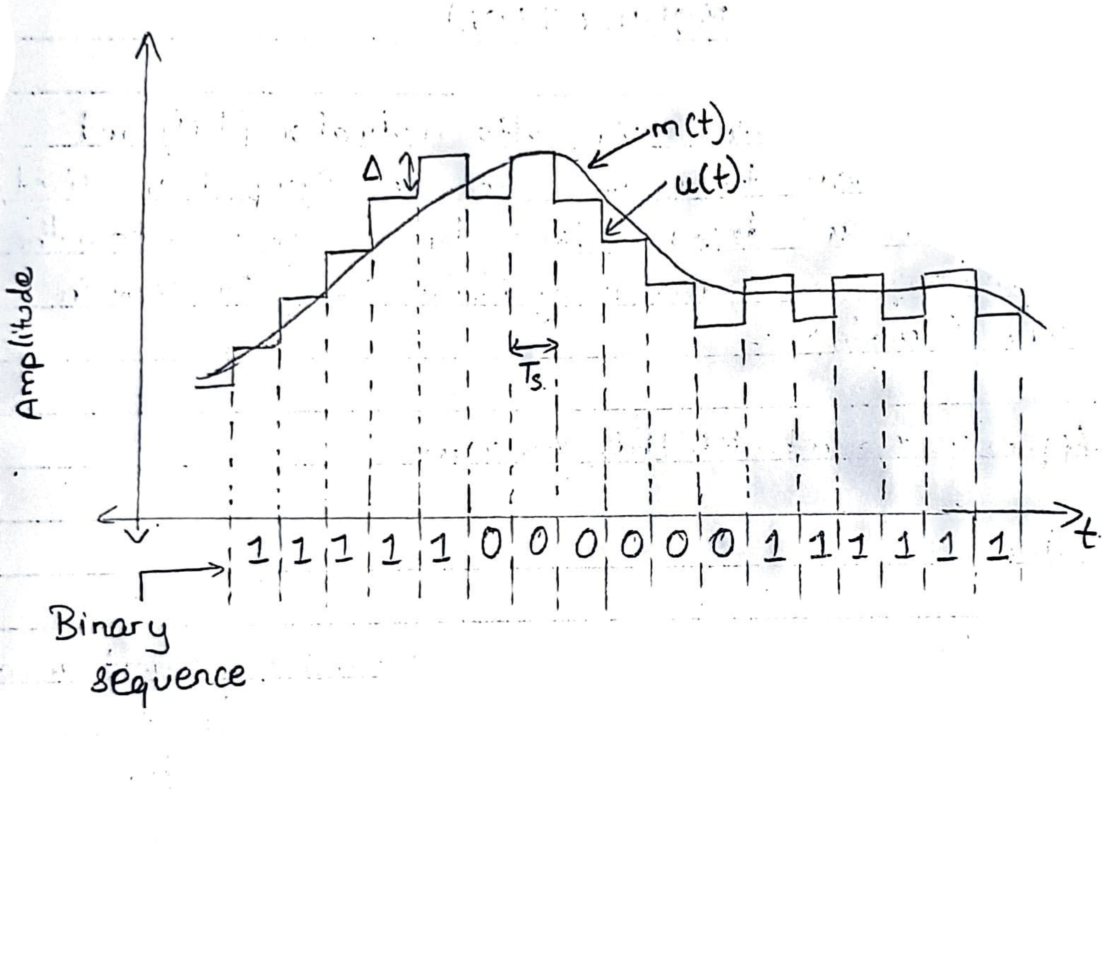


DM transmitter



DM reciever

Waveform representation of Delta Modulated Signal:-



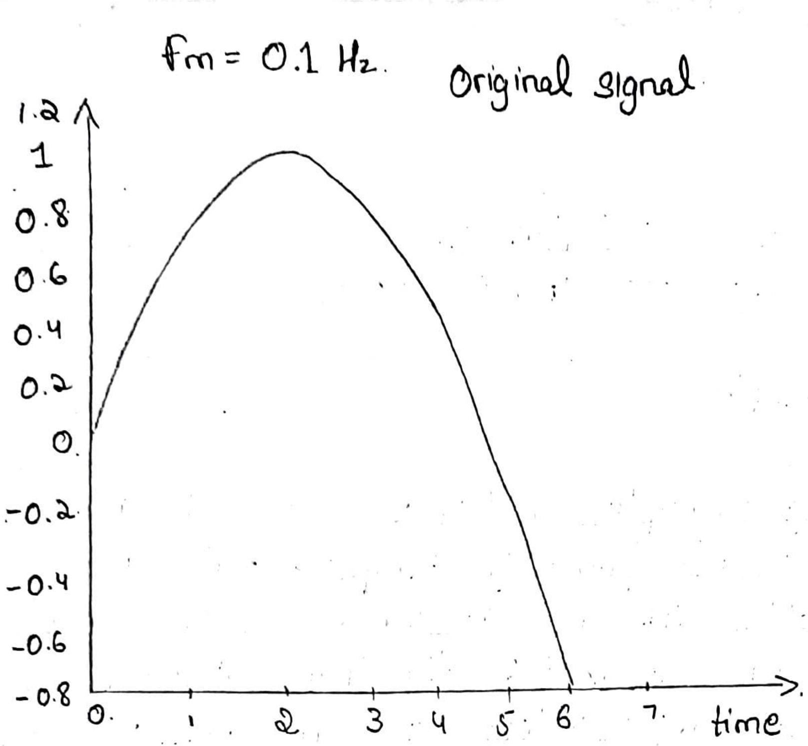
Operating Principle -

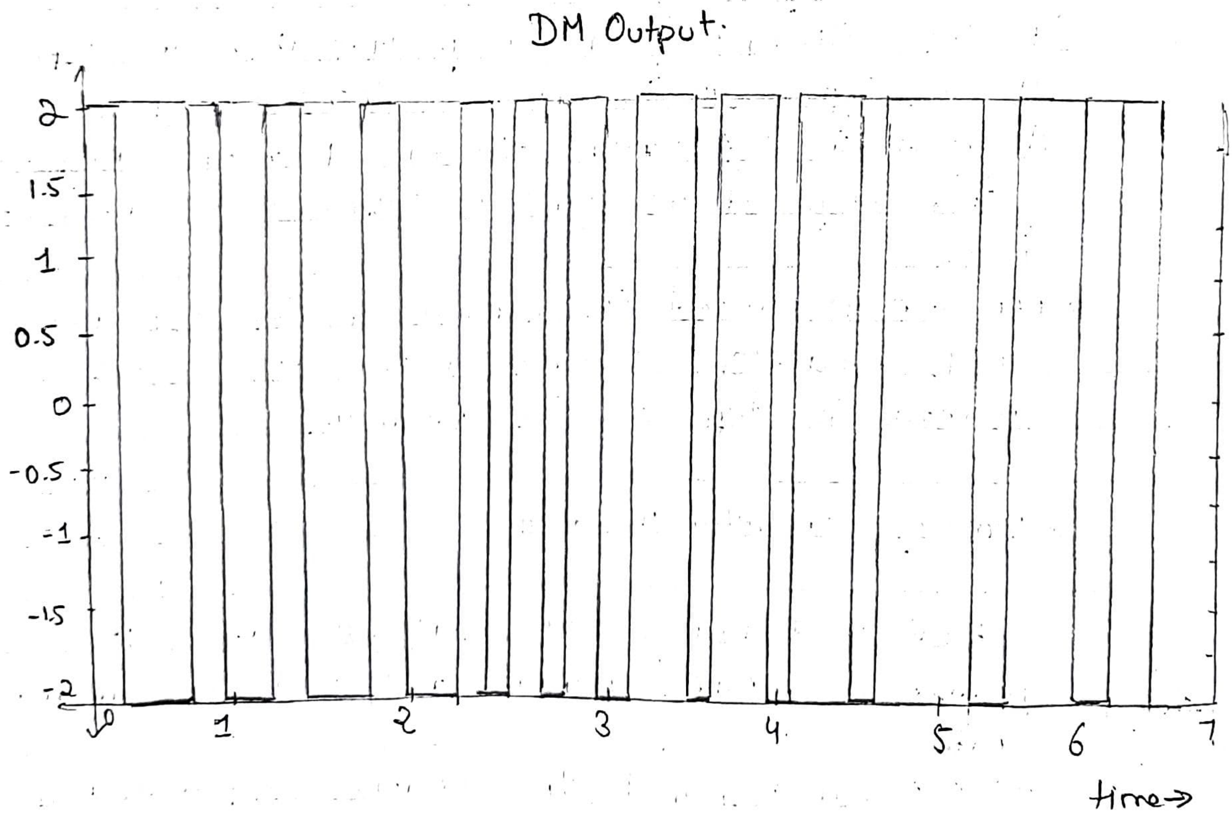
The operating principle of DM is such that, a comparison between present and previously sampled value is performed, the difference of which decides the increment or decrement in the transmitted value. When the two sample values are compared, either we get difference having a positive or negative polarity. 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 i.e. reduction in Δ. When +Δ is noticed i.e. increase in step size, then 1 is transmitted. However, in the case of −Δ i.e, decrease in step size, 0 is transmitted.

Matlab code -

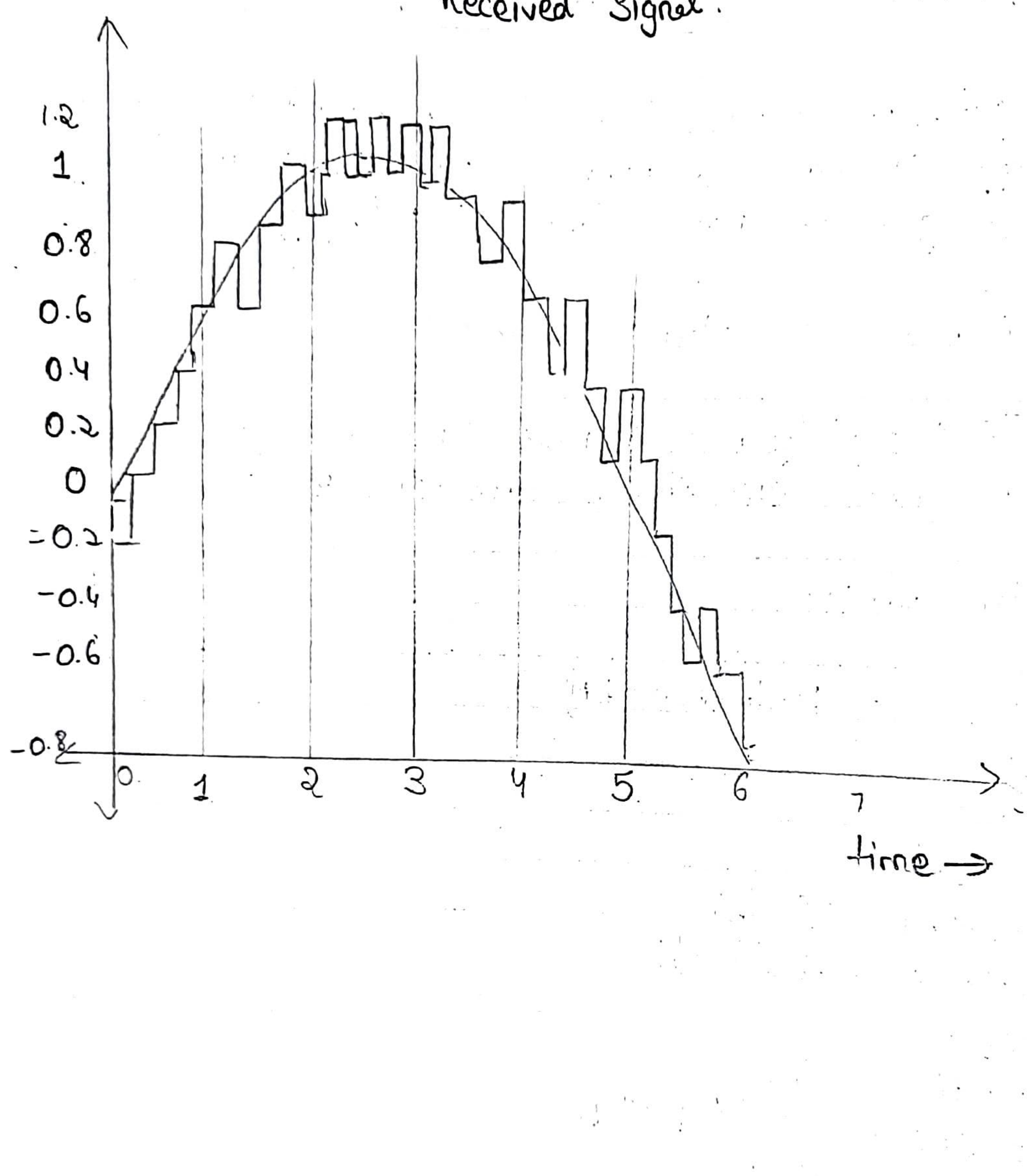
```
%% Delta modulation (DM)
clc;
clear;
close all;
predictor = [0 1];
partition = [-1:.1:.9];
step = 0.2;
partition = [0];
codebook = [-1* step step];
% DM quantizer
t = [0 : pi/20:2* pi];
x = 1.1* sin(2* pi* 0.1* t);
% Quantize x(t) using DPCM
encoded_x = dpcmenco(x, codebook, partition, predictor)
% Try to recover x from the modulated signal
decoded_x = dpcmdeco(encoded_x, codebook, predictor);
```

$f_m = 0.1$ Hz.


Original signal
1.2
1
0.8
0.6
0.4
0.2
0
-0.2
-0.4
-0.6
-0.8
0
1
2
3
4
5
6
7
time

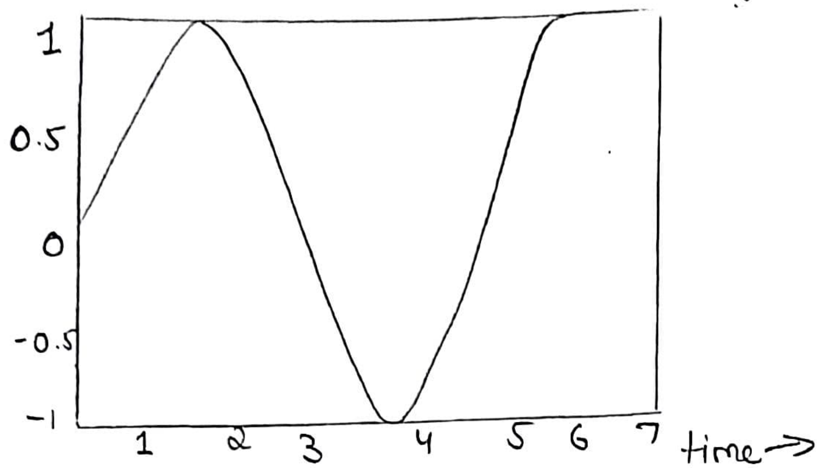

DM Output
2
1.5
1
0.5
0
-0.5
-1
-1.5
-2
0
1
2
3
4
5
6
7
time

Received Signal.

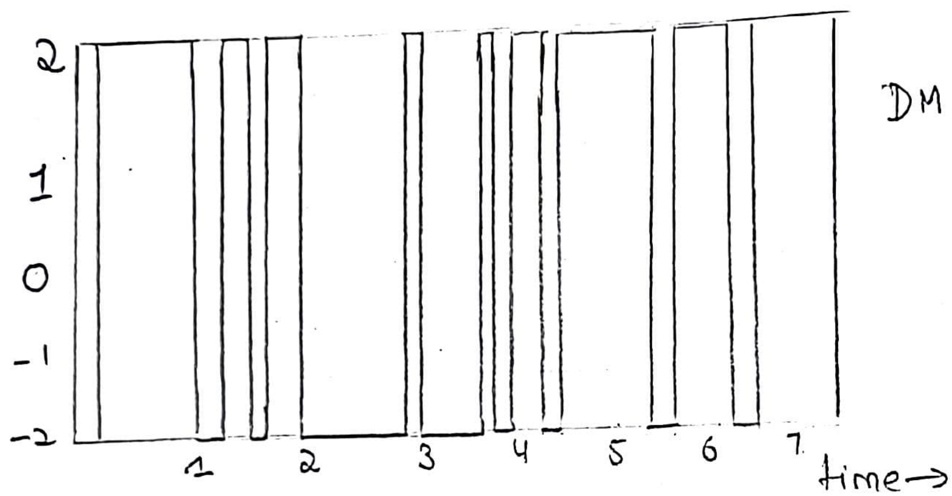


$f_m = 0.2.$

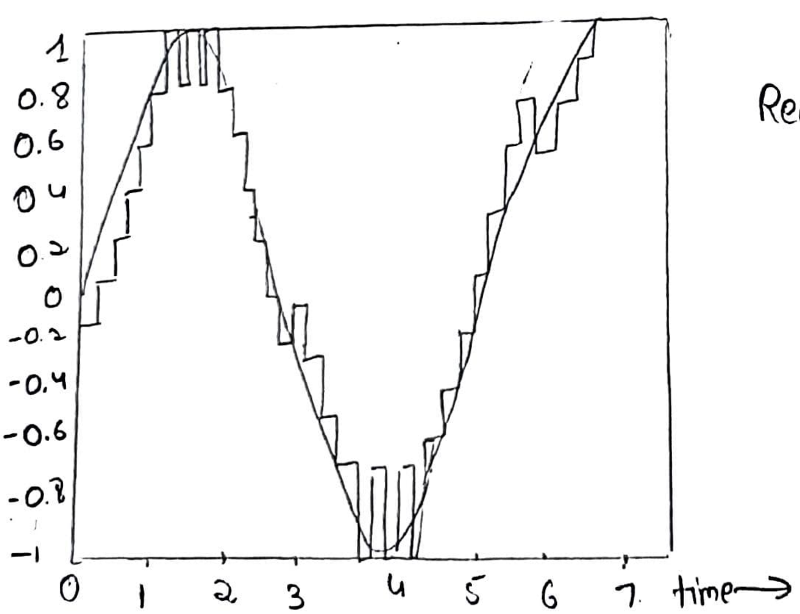
Orginal Signal



DM Signal



Received Signal.



```
figure
plot (t, x);
xlabel ('time');
title ('original signal');
figure
stairs (t, 10*codebook (encoded_x+1), 'y');
xlabel ('time');
title ('DM output');
figure
plot (t, x);
hold;
stairs (t, decoded_x);
grid;
xlabel (' time');
title ('received signal');
```

ADVANTAGES-

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 & detection

DISADVANTAGES-

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

CONCLUSION-

Hence, successfully demonstrated the delta modulation and demodulation techniques, verified the sampled quantized/encoded and decoded time domain signal. Verified the input/output waveform using matlab.