EXPERIMENT - 1

SPECTRUM ANALYSER AND OBSERVE SPECTRUM

Aim :- To study spectrum analyser and observe the spectrum of sinusoidal signal and square wave.

APPARATUS :- Spectrum Analyzer (9KHz - 3GHz) Function generator

Theory :- (1) A spectrum analyzer is a laboratory instrument that displays signal amplitude (strength) as it varies by signal frequency. The frequency appears on the horizontal axis, and the amplitude is displayed on the vertical axis.

(2) To the casual observer, a spectrum analyzer looks like an oscilloscope and in fact, some lab instruments ~~are~~ can function either oscilloscope or spectrum analyzer. A spectrum analyzer can be used to determine whether or not a wireless transmitter is working according to federally defined standards for purity of emissions.

(3) Output signals at frequencies other than the intended communications frequency appears as vertical lines (pips) on the display. A spectrum analyzer can also be used to determine, by direct observation, the bandwidth of a digital or analog signal.

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

Features OF LAB INSTRUMENTS GSP-830 (GWINSTEK)
→ 5 markers with delta marker and peak functions
→ 3 traces
→ Split windows with seperate settings.
→ 6.4" TFT colour LED, 640×480 resolution.
→ AC / DC / battery - multi-made power operator
→ Autoset
→ 3 KHz - 9 ~~GHz~~ frequency range

FREQUENCY SELECTION AND THEIR SELECTION METHODS
(1) FREQUENCY :-
→ Frequency / Span : The frequency key, together with span key sets the span scale.
→ View signal (Center and span) : Center and span methods defines the center frequency and 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 mobe.

(2) Range: 9 KHz to 3 GHz

(3) Set center frequency:
→ Panel operation: → Panel frequency key.
→ Press F1 (center)
→ Enter the value using numerical and unit keys, arrow keys and scroll nobe.

(4) Set frequency span:
→ Panel operation: → Press span key.
→ Press F1 (span)
→ Enter the value using numerical and unit keys, arrow keys and scroll nobe

(5) View signal (start and stop)
→ Start and stop method defines the beginning and end of the frequency range.
→ Arrow keys and scroll knobe resolution: 1/10 of span

(6) Set start frequency.
→ Panel operation: → Press frequency key.
→ Press F2 (Start)
→ Enter the value using numerical and unit keys, arrow keys and scroll nobe.

→

(7) Set stop frequency.
→ Panel operation: → Press frequency key.
→ Press F3 (stop)
→ Enter the value using numerical and unit keys, arrow keys and scroll mode.

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

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

(10) Zero span display.
→ It can be obtained by pressing F3 key.
→ Start and stop frequency remains same as center freq.
→ Note: Last span setting can be recalled by F4 key.

AMPLITUDE SELECTION AND SETTING METHODS

(1) Amplitude :-

Amplitude key sets vertical attribute of the display, including the upper limit (extreme value), vertical range (amplitude scale) vertical unit and compensation for external gain or less (extreme offset).

(2) Set vertical scale.

'Vertical display scale is defined by reference amplitude, amplitude range, measurmment unit and extreme gain / loss.

(3) Set reference amplitude

The reference level defines the amplitude at the top of the display range.

→ Panel operations :-

→ Press amplitude key.

→ Press F1 (reference level)

→ Enter the value using numerical and unit, arrow keys and scroll knob.

Arrow keys, scroll knob, scroll ~~note~~ knob resolution: Vertical scale

→ Range :-

dBm : -110 to +20 dBm, 0.1 dBm resolution.

dBmV : -63.01 to 66.99 dBmV, 0.01 dB resolution.

dBμV : -3.01 to 126.99 dBμV, 0.01 dB resolution

(4) Select amplitude scale.
→ Panel operation :
→ Press amplitude key.
→ Press F_2 (scale dB/div)
→ Repeatedly to select the scale.

→ Range : 10, 5, 2, 1 dB/div.

→ Panel operation.
→ Press amplitude key.
→ Press F_3 (units)
→ Select and press units from F1 (dBm), F2 (dBm) & F_3 (dBuV)
→ Press F6 (return) to go back to previous menu.
→ dBm = -110 to +20 dBm, 0.1 dBm resolution.
→ dBmV = -63.1 to 66.99 dBmV, 0.01 dB resolution
→ dBuV = -3.01 to 126.99 dBuV, 0.01 dB resolution
Set external offset level.

(5) Background
External offset compensates the amplitude gain or loss caused by an external network or devices.
→ Panel operations :
1) Press amplitude key.
2) Press F4 (external gain)
3) Enter the value using numerical and unit keys, arrow keys and scroll knob.

→ Range :- -20dB 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 will generate 10MHz signal with 10dB amplitude.

→ CONCLUSION :-

Hence, we have successfully verified and analysed the spectrum of sinusodial signal and square wave to different frequency and amplitude

EXPERIMENT - 2
SAMPLING AND RECONSTRUCTION : NYQUIST CRITERIA

Aim: To perform sampling and reconstruction of signal and obtain its waveforms. Also verify nyquist criteria.

Apparatus : Nyquist Applet (software)

THEORY :

(1) The continue - time signal can be stored in a digital computer, in the form of discrete (equidistant) points or samples.

The higher the sampling rate (sampling frequency), the more accurate would be the stored information and the signal reconstruction from its samples.

However, high sampling rate produces a large volume of data to be stored and makes necessary the use of very fast analog to digital converter.

→ Analog signal :- It is continous time varying feature of the signal.

→ Digital signal :- It represent data as sequence of discrete values of any time, It can only take any one of the finite number of values.

→ The technique that can be used for analog to digital conversion is Pulse code modulation.

→ It has 3 Processes

- Sampling
- Quantization
- Encoding

(2) Sampling.

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

→ When a source generates an analog signal and if that has to be digitalized, having 1's and 0's, the signal has to be discritized in time. The discritization 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 a 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.

(3) Nyquist Rate

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

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

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

F_S = Sampling rate W = Highest frequency.

This is nyquist rate and theorem is called sampling theorem.

(A) Condition 1 : Over sampling $(F_S > 2W)$

If sampled at highes rate the 2W is the frequency domain.

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

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

(B) Condition 2 :

→ If the sampling rate is equal to twice the frequency $F_S = 2W$.

→ The information is retrived without any loss. Hence, this is also a good sampling rate.

→

(c) Condition 3 : Under sampling

$F_s < 2W$.

→ The below pattern shows overlapping of information which leads to mixing up and loss of information. This unwanted phenomena of over-lapping is called Aliasing.

→ Aliasing :- A high frequency component is taking on the identity of a low frequency component in the spectrum of sampled version.

The effect of aliasing is reduced by.

1) The signal needs to be sampled at a rate slightly higher than the nyquist rate.
2) In the transmitter section of PCM, a low pass anti-aliasing filter is employed to eliminate the unwanted high frequency components.

(4) Quantization :- The method of sampling chooses few points on the analog signal and then these points are joined by round off the value of a near stablized value is called quantization.

(5) Encoding :

→ The ~~digitization~~ digitization of analog signal is done by encoding.

→ After each sampleng is quantized, the number of bits per sample is decided.
→ Each sample is changed to an n bit code.
→ Encoding is also used to minimize the bandwidth.

(6) Anti-Aliasing filter.
→ Designing this filter is to determine the bandwidth required in the ~~acquisim~~ aquisitian system. The max. frequency of the input signal should be less than ~~of~~ or equal to half of sampling rate.
→ This sets the cuttoff frequency of the low-pass filter.
→ The order of a filter affects the stepness of the transition region roll-off and hence the width of the transition region.

→ A first order filter has a roll-off of 20dB/decade, which means any signal having frequency above cutt-off frequency will be attneuated at this rate.

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

CONCLUSION
Therefore, sampling and reconstruction of the signal has been ~~por~~ performed successfully on Nyquist Applet and Nyquist Criteria has been Verified.

EXPERIMEN - 3
AMPLITUDE MODULATION

AIM :- Study of an amplitude modulated (A.M) scheme, depth of modulation, waveforms, spectra and trapezodial display.

APPARATUS : Lab live software.

THEORY : 1) Classification of A.M modulation.

- Double side bend supressed carrier (DSB-SC)
- Double side Band with carrier (AM)
- Single side Band (SSB)
- Vestigal side band (VSB)

2) AM

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

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

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

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

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

3) Measurement of 'm'

- The magnitude of 'm' can be measured directly from the AM signal itself.
- Maximum and minimum amplitudes of the transmission signals envelope, determine the modulation depth :

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

max Amplitude of modulated wave ; $a = A_m + A_c$
min Amplitude of modulated wave , $b = A_c - A_m$

$$\therefore A_c = \frac{a+b}{2} \quad , A_m = \frac{a-b}{2}$$

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

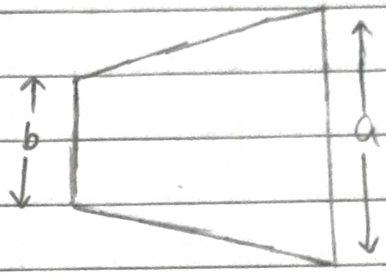
4) Envelope detector :- The non-coherent detection doesn't require a carrier recovery circuit. In its simplified form, it consist of rectifier diode and a low pass filter.

5) Synchronous detector :- AM without a carrier. Envelope detection can't be deployed because the transmitted signal's envelope changes sign Transmit spectrum of DSB-SC.

6> Trapezoid Method

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

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



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

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

PROCEDURE

In Lab-Alive Software.

1) We will first execute the AM analyser simulator.
2) Then click on $\hat{s}$ in the AM modulation window.
3) For D.S.B with carrier click on D.C and for D.S.B with suppressed carrier off the DC output.
4) For the different value of m observe the transmitted signal. using oscilloscope and spectrum analyser.

$\longrightarrow$

Teacher's Signature :

OBSERVATION

Double side band with carrier.

Sr No	AM(V) Message	FM (MHz) message	AC (V) carrier	Fc (MHz) carrier	A(V) $\hat{s}$	F(MHz) $\hat{s}$	μ
1							
2							
3.							

Double sideband with suppressed carrier.

Sr No	Message		Carrier		Modulation.
	AM(V)	Fm(MHz)	A_c(V)	F(MHz)	Index (μ)

CONCLUSION

We observe that using envelope detector we can detect double side band with carrier but synchronous detector is needed for double sideband suppressed carrier. We also observe that information lies in sidebands and in carrier. Using DBS, we can minimize power usage.

EXPERIMENT-4

FREQUENCY MODULATION AND DEMODULATION.

Aim: To study frequency modulation (FM) and frequency demodulation with its application.

Apparatus Required: Labaline software, Matlab software.

THEORY:

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

2) The standard equation of the angle modulated wave is

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

where A_c = Amplitude of the modulated wave / carrier signal.

$\theta_i(t)$ = Angle of modulated wave.

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

1) Frequency modulation → is the process of varying the frequency of the carrier signal linearly with message signal.

2) Phase modulation → is the process of varying the phase of carrier signal linearly with message signal.

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

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

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

$$\boxed{f_i = \underbrace{f_c}_{\text{Carrier frequency}} + \underbrace{(K_F)}_{\text{Frequency sensitivity}}\, \underbrace{m(t)}_{\text{Message signal}}} \quad \text{—— (1)}$$

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

$$\left[\omega_i = \frac{d(\theta_i)}{dt}\right]$$

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

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

Substitute f_i from equation (1)

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

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

Substitute $\theta_i(t)$ value in standard equation of angle modulated wave

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

7) Finally, equation of F.M. wave.

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

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

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

$$\boxed{\beta = \frac{K_f A_m}{F_m} = \frac{\Delta F}{F_m} = \text{modulating index.}}$$

8) The difference between FM modulated frequency (instantaneous) and normal frequency is termed as Frequency Deviation.
It is denoted by $[\Delta F = f_i - f_c = K_f A_m]$

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

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

10) Bandwidth $= 2 * (f_m + \Delta f)$

11) In FM, carrier amplitude is constant.
∴ Transmitted power is constant.
and transmitted power does not depend on modulation index.

12) Applications and Advantages of FM
→
→ FM is resilient to noise and interface. ∴ It is used for high quality broadcast transmission.
→ FM is ideal for mobile radio communication application ~~using~~ including more general two way radio communication or portable applications where signal levels are likely to very considerably.
→ Rador, Telemetry, Observing infants for through EEG.

MAT-LAB CODE

```
clc; clear all; close all;
fs = 5000;
fc = 200;
t = (0:1/fs:0.4);
m = sin(2*pi*10*t);
 % + sin(2*pi*30*t);
fDev = 100; % Frequency deviation value.
%% FM Modulation
y = fmmod(m, fc, fs, fDev);
%% Plotting the Base band signal
```

→

```
subplot(311);
Plot(t,m);
title('Modulation (Baseband signal');
xlabel('Time-->');
ylabel('Amplitude-->');
%% Plotting the FM signal.
subplot(312);
Plot(t,y);
title('FM Modulated signal');
xlabel('Time-->');
ylabel('Amplitude-->');

%% Frequency demodulation
z=fmdemod(y,fc,fs,fDev);
subplot(313);
Plot(t,z,'r');
title('Recieved signal (originally transmitted via FM)');
xlabel('Time-->');
ylabel('Amplitude-->');
```

CONCLUSION.

We have successfully verified and understand the concept of frequency modulation and demodulation using MATLAB and also learnt applications of FM.

EXPERIMENT-5

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

AIM: To examine PAM, PPM and PWM and verify and draw the resultant waveform.

APPARATUS: MATLAB Software (online)

THEORY:

1) Pulse modulation is a type of modulation in which the signal is transmitted in the form of pulses. Pulse modulation is further divided into analog and Digital communication and further analog and digital is subdivided in PAM, PWM, PPM, PCM, DM.

2) Pulse amplitude modulation (PAM)
In PAM, a pulse signal is used to sample an analog signal. The result is the train of constant ~~wid~~ width pulses. The amplitude of each pulse is propotional to the amplitude of the message signal at the time of sampling.

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

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

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

3) Pulse width Modulation (PWM)

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

4) Pulse Position Modulation (PPM)

→ In this type of modulation, both the amplitude and the width of the pulse are kept constant.

→ PPM is further modification of PWM

5) Comparision of PAM, PWM and PPM.

Sr No.	PAM	PWM	PPM
1)	Amplitude of the Pulse is propotional to amplitude of modulating signal.	Width of the pulse is propotional to amplitude of modulating signal.	Relative position of Pulse is propotional to amplitude of modulating signal

2) Band width of the transmission channel depends on the pulse width.	Here, it depends on the rise of time of the pulse.	Depends on rising time of the pulse.
3) Instantaneous Power of transmitter varies.	Varies.	Constant.
4) Noise Interference is high	Minimum	Minimum.
5) System is complex to implement	Simple to implement	Simple to implement.
6) Similar to amplitude modulation	Frequency Modulation.	Phase Modulation.

CONCLUSION

We successfully examined PAM, PPM, PWM and also verified their waveforms. we also illustrated circuits for PAM & PWM. We performed our experiment successfully using MATLAB.

MAT-LAB CODE

→

```
% PAM Signal.
clc; clear all; close all;
fc=100
fm= fc/10.
fs = 100*fc
t=0:1/fs:4/fm;
Msg_sgl = cos(2*pi*fm*t);
Carr - sgl = 0.5* square (2*pi*fc*t) +0.5
Mod-sgl = Msg-sgl * Carr_sgl;
tt [];
for i=1: length (mod_sgl);
        if Mod-sgl(i)==0;
                ~~tt[tt~~ tt = [tt, Mod-sgl(i)];
        else
                tt =[tt ;Mod-sgl(i)+2];
        end.
end.
figure (1)
subplot (4,1,1)
Plot (t, Msg-sgl ., 'm');
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(r)) ;
title ('PAM Modulated signal') ;
xlabel ('Time period') ;
ylabel ('Amplitude') ;

% PPM signal
clc; clearall; close all;
fc=1000;
fs = 10000;
fm=200;
t=0:1/fs:((2/fm)-(1/fs));
x=0.5*cos(2*pi*fm*t)+0.5;
y=modulate(x,fc,fs,'PPM');
subplot(2,1,1);    Plot(x);
title('Msg signal');
subplot(2,1,2);  Plot(y);
axis([0 500 -0.2 1.2]);
title('PPM');

% PWM-I signal (Natural sampling)
clc;  clear all;  close 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
```

```
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,'o'; t,m,'r'; t,s,'b');
ylabel('Amplitude');
axis([0 1 -1.5 1.5]);
xlabel('Time p index');
title('PWM wave');
grid on;

% PWM-2 signal {
clc; clear all; close all;
t = 0 : 0.0001 : 1;
s = Sawtooth(2 x Pi x 10 x t + Pi);
m = 0.75 x sin(2 x Pi x 1 x 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
```

```
subplot (3,1,1)
Plot (t, m, 'm');
y label ('Amplitude');
axis ([0 1 -1.5 1.5]);
x label ('Time index');
grid on;

Subplot (3,1,2)
Plot (t, s, 'b');
y lablel ('Amplitude');
axis ([0 1 -1.5 1.5]);
x label ('Time index');
title ('PWM wave');
grid on;

Subplot (3,1,3)
Plot (t, Pwm, 'r');
ylabel ('Amplitude');
axis ([0 1 -1.5 1.5]);
x label ('Time index');
grid on;
```

EXPERIMENT - 6.

AMPLITUDE SHIFT KEYING (ASK)
FREQUENCY SHIFT KEYING (FSK)
PHASE SHIFT KEYING (PSK).

Aim: To study ASK, FSK and PSK modulation technique and verify waveforms.

APPARATUS: MATLAB.

THEORY:

1) MODulation:- It is a process by which same charactastics of a carrier wave is varied in accordance with a modulationg (message) signal.

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

The ASK, FSK and PSK are analogus to AM, FM and PM. The difference is that it is digital and that is analog in nature.

2) ASK.

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

Application of ASK : 1) Wireless Base station.
2) Low frequency RF application.
3) Industrial Network Devices.

3) FSK.
If the frequency of sinusodial carrier wave is varied depending on the input signal, then it is known as FSK.
Applications of FSK :- 1) High frequency radio transmission

4) PSK.
In PSK, phase of the carrier wave (analog in nature) is switched as per the input digital signal.
Applications of PSK : 1) It is widely use for wireless LANs, RFID and bluetooth communication.

MATLAB CODE

```
% ASK signal
clc; clear all; close all;
fc=input ('Enter the frequency of sine wave carrier:');
fp=input ('Enter the frequency of periodic binary Pulse:');
amp=input ('Enter the amplitude (for carrier and binary):');
t=0:0.001:1;
c=amp.*sin(2xpixfcxt);
Subplot (3,1,1).
Plot (t,c);
xlabel ('Time');
ylabel ('Amplitude');
title ('Carrier wave');
```

Teacher's Signature :

```
m = amp/2 x square (2xPixfpxt) +(amp/2);
subplot (3,1,2)
Plot (t,m);
xlabel ('Time');
Ylabel ('Amplitude');
title ('Binary message pulse');
w = c x m;
subplot (3,1,3)
Plot (t,w).
xlabel (' Time');
Ylabel ('Amplitude');
title ('Amplitude shift keyed signal');

% FSK signal.
clc ;, clearall ; close all;
fc1 = input ('Enter the freq. of 1st sine wave carrier');
fc2 = input ('Enter the freq of 2nd sine wave carrier');
fp = input ('Enter the freq of periodic binary pulse');
amp = input ('Enter the amplitude (For both Carrier and
                        binary pulse message):');
amp = amp/2;
t = 0:0.001:1;
c1 = amp x sin (2xPixfc1xt);
c2 = amp xsin (2x Pi x fc2 xt);
subplot (4,1,1);
plot (t,c1);
Xlabel ('Time');   ylabel ('Amplitude');
title ('Carrier Wave 1');
```

```
Subplot (4,1,2) ;    Plot (t,c2) ;
xlabel ('Time') ;    ylabel ('Amplitude');
title ('Carrier 2 wave');
m = amp x square (2xPi x fp x t) + amp;
subplot (4,1,3)
Plot (t,m)
xlabel ('Time') ;
ylabel ('Amplitude') ;
title ('Binary message pulse') ;
for i=0:1000
    if m(i+1)==0
        mm(i+1) = c2(i+1) ;
    else
        mm(i+1) = c1(i+1) ;
    end
end.
subplot (4,1,4)
Plot (t,mm)
xlabel ('Time') ;
ylabel ('Amplitude') ;
Title ('Modulated wave') ;

% PSK signal.
clc ; clear all ; close all ;
fc = input ('Enter the freq of sine wave carrier') ;
fp = input ('Enter the freq. of periodic binary pulse:') ;
amp = input ('Enter the amplitude:') ;
```

```
t=0:0.001:1;
c= amp x sin (2xPi xfc xt);
subplot (3,1,1); Plot (t,c);
Xlabel ('Time'); Ylabel ('Amplitude');
Title ('carrier wave');
grid on;
m = square (2x Pi x fp x t);
subplot (3,1,2); Plot (t,m);
Xlabel ('Time'); Ylabell ('Amplitude');
title ('Binary message pulse');
w=cxm;
subplot (3,1,3)
Plot (t,w);
x label ('Time); Ylabel ('Amplitude');
grid on;
```

CONCLUSION

We have successfully studied ASK, PSK and FSK. modulation technique and verified their waveforms using MATLAB. We also observe ~~that~~ the schematic diagrams for ASK, FSK and PSK.