HAND WRITTEN LABORATORY JOURNAL

Submitted in partial fulfillment of the requirement
For the Subject

"DIGITAL COMMUNICATION"

(EC 209)

: Prepared & Submitted By :

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

(July to Dec - 2021)

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DEPARTMENT OF ELECTRONICS ENGINEERING

Academic Year : 2021-22

SUB : DIGITAL COMMUNICATION (EC209)

CERTIFICATE

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

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1.	
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3.	

July -Dec. 2021.

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

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ONLINE MODE

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- Submitted By

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Admission Number : U20CS100
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Date: 22/10/21
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EXPERIMENT-7

EFFECT OF AWGN ON AM AND FM

AIM: To study the transmission of Amplitude modulated (AM) and frequency modulated (FM) Signal under the Additive Gaussian noise channel (AWGN). Examine the effects of the AWGN on AM and/or FM signal using the matlab and draw the distorted waveforms for different signal to noise ratio (SNR) value. Show the input/output waveforms using matlab Code in virtual mode.

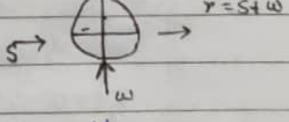
APPARATUS: MATLAB (software)

THEORY: 1> Additive white Gaussian Noise (AWGN)

A Basic Noise model used to mimic the effect of many random process that occur in nature. channel produces Additive white Gaussian noise.

ⓐ Additive: The received signal Equals the transmit Signal plus some noise, where the noise is statistically independent of the signal.

$$r(t) = s(t) + \omega(t)$$



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

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

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

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

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

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

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

2) Signal to Noise Ratio.

→ The SNR or S/N is a measure used in science and Engineering that compares the level of a desired Signal to the level of background noise.

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

→ SNR, bandwidth and channel Capacities of a Communication channel are connected by the shannon - Hartley theorem.

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

Shannon - Hartley Theorem.

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

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

→ It is related to signal to noise (SNR) or carrier to noise (CNR) [linear power Ratio].

→ 5 dB - 10 dB = It is below the minimum level to establish a Connection due to the noise level being nearly indistguistable from the desired signal (useful Information)

→ 25 dB - 40 dB = deemed to be good

→ 41 dB or higher = Considered to be excellent.

3) Mathematics of AM.

→ let modulating signal to be $e_m = E_m \sin(\omega_m t)$
Carrier signal be $e_c = E_c \sin(\omega_c t)$

$\therefore E_{AM} = E_c + e_m$

$= E_c + E_m \sin(\omega_m t)$

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

$e_{AM} = E_{AM} \sin(\theta)$

$= E_{AM} \sin(\omega_c t)$

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

This is an equation of AM wave.

4) Mathematics of FM.

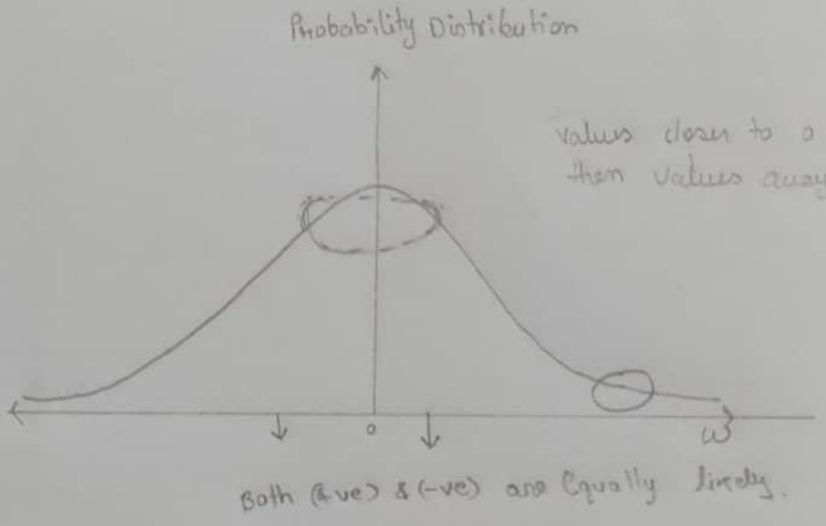
$S(t) = E_s \sin(\omega_c t + m_s \sin(\omega_m t))$

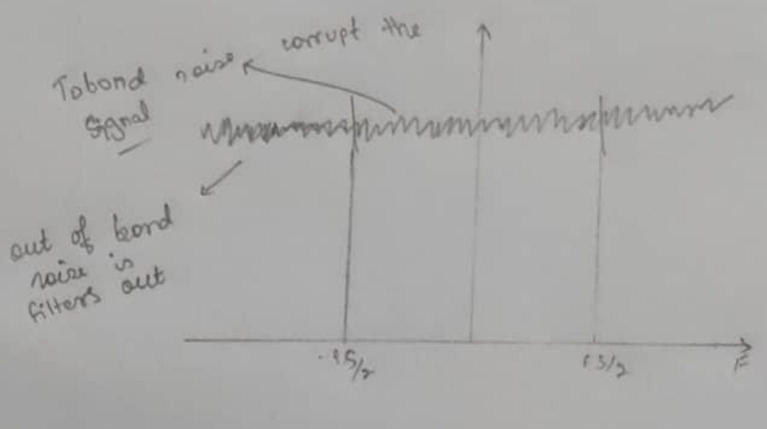
This is the expression for FM wave, m_f = modulation index.

$$m_F = \frac{\text{frequency deviation}}{\text{modulating frequency}} = \frac{\Delta f}{f_m}$$

→ frequency deviation Δf = represents the maximum departure of the instantaneous frequency $f_i(t)$ of the FM wave from the Carrier frequency F_C.

⇒ Gassion





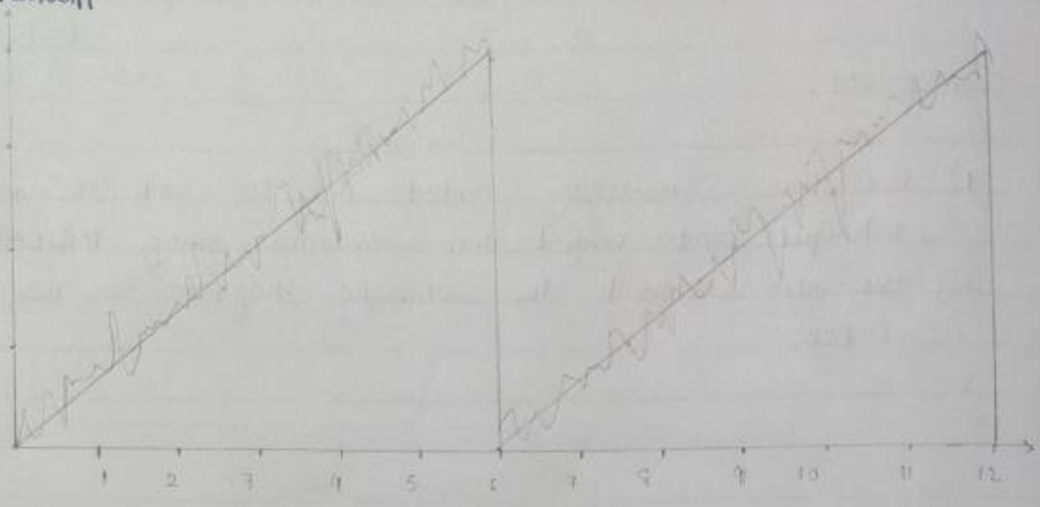
MATLAB CODE :

AWGN in different function.

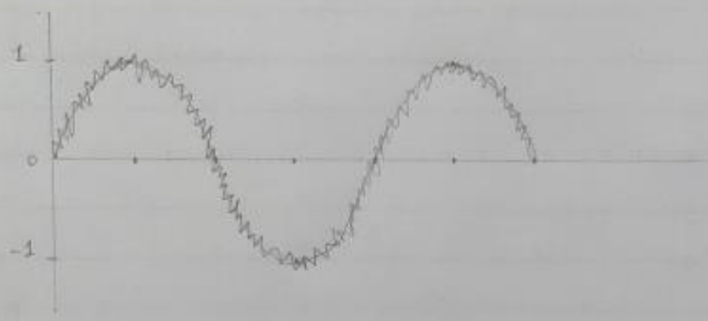
```
clc ;
clear all;
t = 0 : 0.1 :10) ;
x = sawtooth (t);
y = awgn (x, 10, 'measured ') ;
Plot (t, [x, y]);
legend ('original signal', 'signal with AWGN');

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

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

AWGN Effect on Different functions

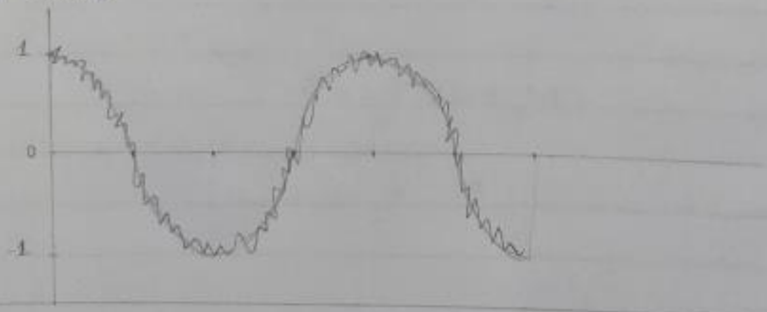
1) Sawtooth



2) Sine



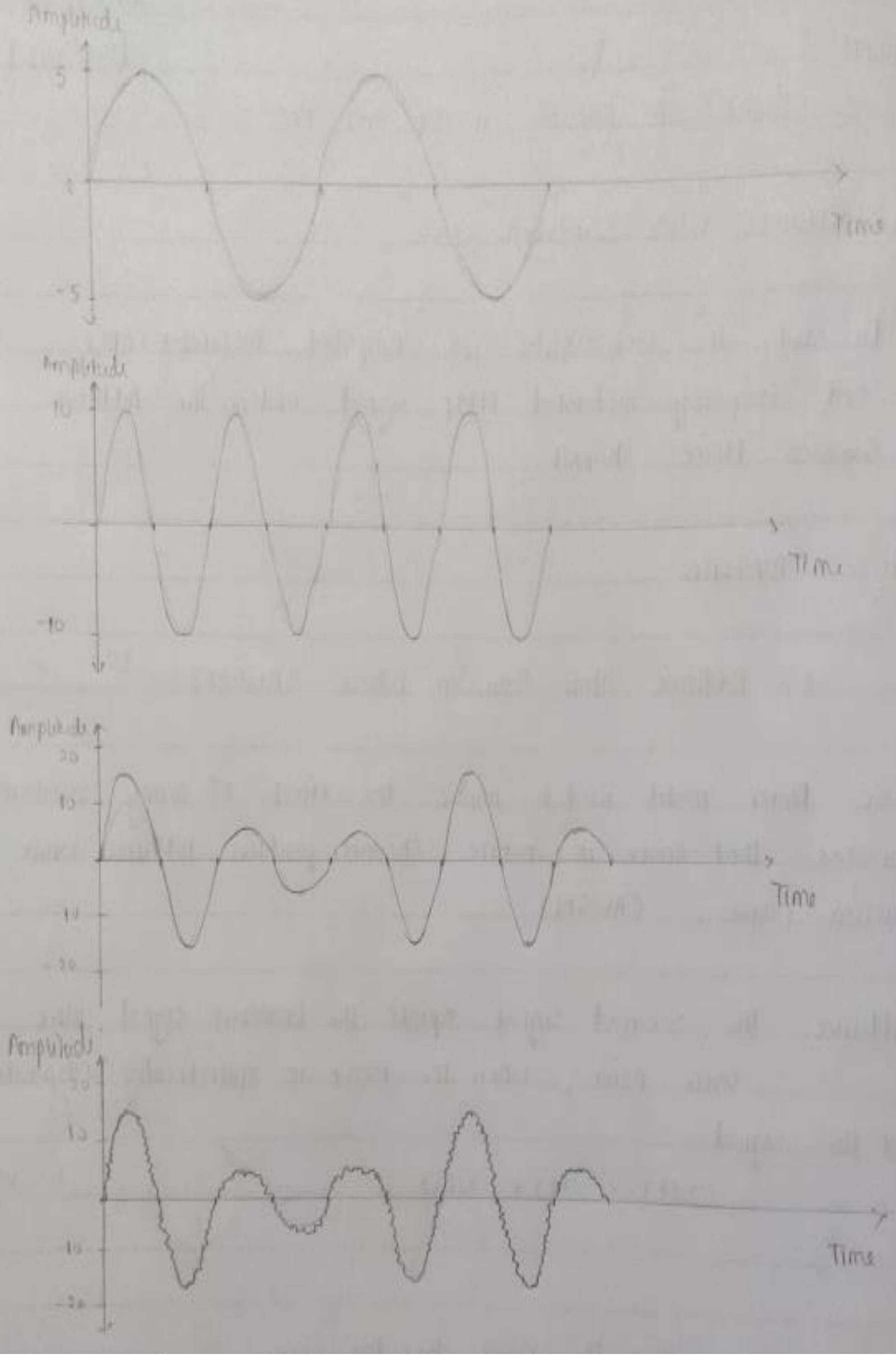
3) Cosine



AWGN in AM

```
clc;
clear all;
t=0:0:001:1;
Vm=5;
Vc=10;
fm=2;
fc=25;

m= Vm * sin (2* pi + fm *t);
c = Vc * sin (2* pi * fc *t );
amp = Vc + Vm * sin(2*pi*fm*t);
am = amp * sin (2* pi* fc*t );
Y = awgn (am , 10, 'measured');

(I) subplot (4,1,1);
Plot (t,m);
X label ('time')
Xlabel ('time')
Y label ('Amplitude')

(II) subplot (4,1,2);
Plot (t,c);
X label ('time');
Y label ('amplitude');

(III) Subplot (4,1,3);
Plot (t, am);
X label ('time');
Y label ('amplitud)

(IV) Subplot (4,4,4);
Plot (t,Y);
Xlabel ('time')
Ylabel ('amplitude).

title ('amplitude modulated signal with AWGN');
```

AM signal under AWGN



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PAGE NO.: 8
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AWGN in different SNR

```
clc; clear all;
t=0 : 0.001 : 1;
Vm=5; Fm=2;
Vc=10; Fc=25;
m = Vm * sin(2*pi*(fm*t));
c = Vc * sin(2*pi*fc*t);
amp = Vc + Vm * sin(2*pi*fm*t);
am = amp * sin(2*pi*fc*t);
Y1 = awgn(am, 10, 'measured');
Y2 = awgn (am, 10, 'measured');
Y2 = awgn (am, 100, 'measured');
Y3 = awgn (am, 1000, 'measured');
```

(i)
```
Subplot (4,1,1);
Plot (t, am);
Xlabel ('time')
Y label ('amplitude');
title ('amplitude modulated signal');
```

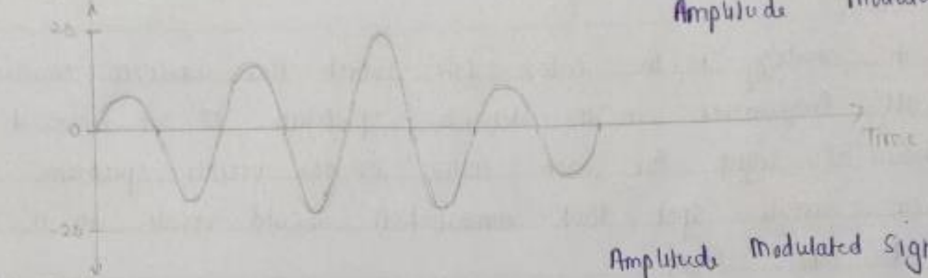
(ii)
```
Subplot (4,1,2);
Plot (t, Y1);
Xlabel ('amplitude');
Y label ('amplitude');
title ('Am signal with
AWGN [snr 10]');
```

(iii)
```
Subplot (4,1,3);
Plot (t,Y2);
xlabel ('amplitude');
title ('Am signal with
AWGN [SNR 100]');
```

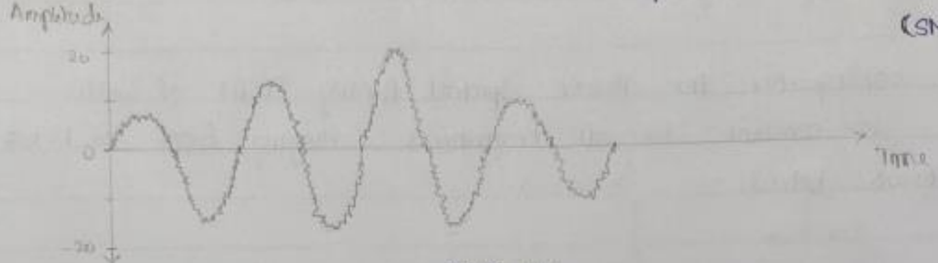
(iv)
```
subplot (4,1,4);
plot (t, Y3);
Xlabel ('time')
Y label ('amplitude')
title ('Am signal
with AWGN
[SNR 1000]);
```

AM Signal with different SNR values

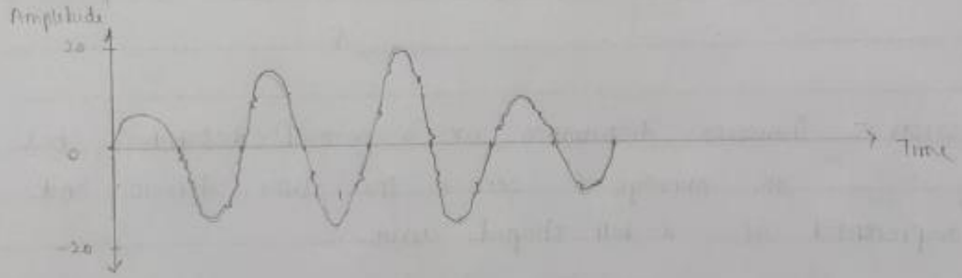
Amplitude Modulated signal



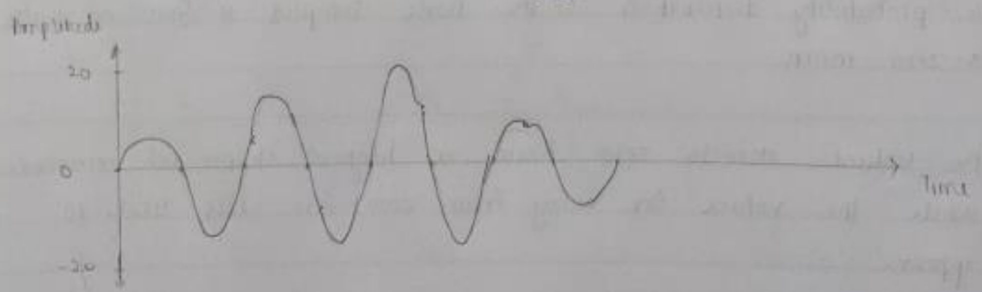
Amplitude Modulated Signal with (AWGN)
(SNR = 10)



SNR = 100



SNR = 1000



AWGN in FM

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

```
(i) Subplot (4,1,1)
Plot (t, msg)
xlabel ('time');
ylabel ('amplitude');
title ('message signal');
```

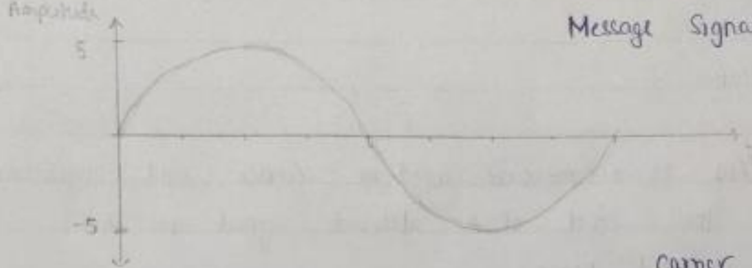
```
(ii) subplot (4,1,2);
Plot (t,c);
ylabel ('amplitude');
xlabel ('time');
title ('carrier signal');
```

```
(iii) Subplot (4,1,3);
Plot (t,Y);
Xlabel ('time');
Ylabel ('Amplitude');
title (' frequency modulated signal').
```

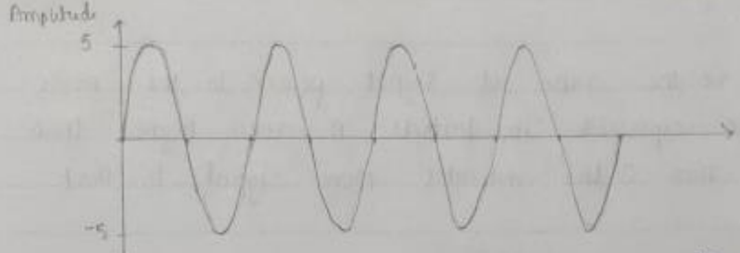
```
(iv) Subplot (4,1,4);
Plot (t, z);
Xlabel ('time')
Y label ('Amplitude')
title ('frequency modulated signal with AWGN');
```

FM Signal under AWGN

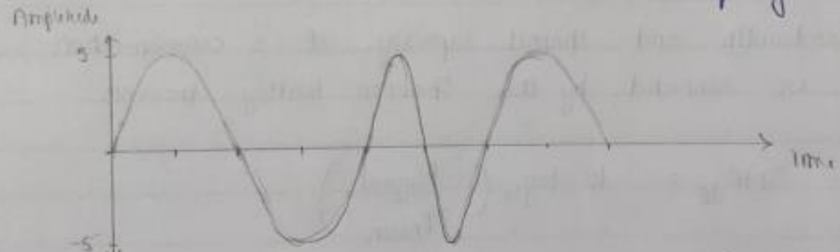
Message Signal



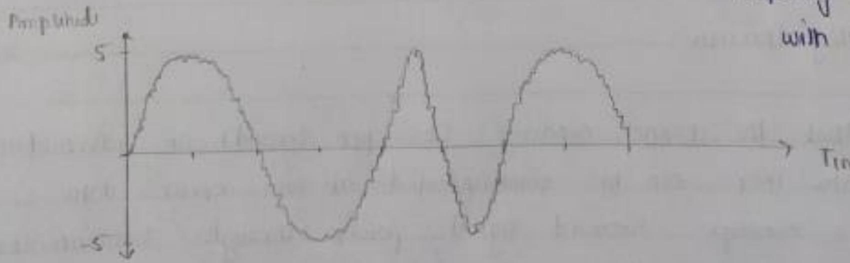
Carrier signal



Frequency modulated Signal



Frequency Modulated signal with AWGN



FM in different SNR

```
clc; clear all;
t = 0:0.001:1;
Vm = 10;        fm = 2;
Vc = 5;         fc = 25;
fd = 10
m = Vm * sin (2*pi * fm *t);
C = Vc * sin (2* pi * fc *t);
amp = vc + Vm* sin (2* pi * fm *t);
Y = Vc *sin (2*pi*fc*t + fd* cos (2*pi* fm *t));
Y1 = awgn (Y, 1, 'measured');
Y2 = awgn (Y, 10, 'measured');
Y3 = awgn (Y, 100, 'measured');
```

(i)
```
Subplot (4,1,1);
Plot (t,Y);
xlabel ('time')
Ylabel ('amplitude')
title ('frequency modulated
        signal')
```

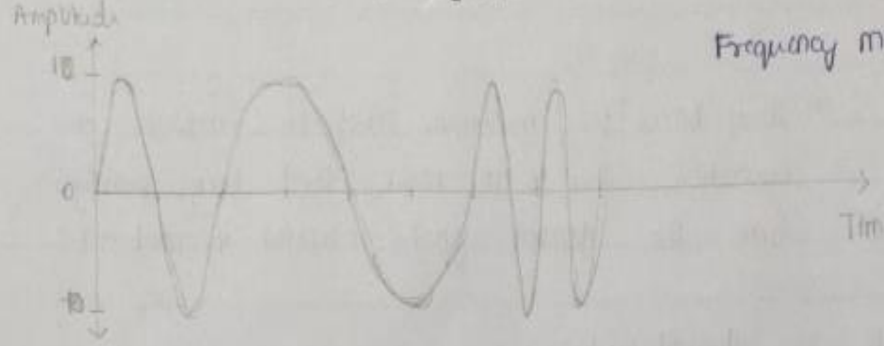
(ii)
```
SubPlot (4,1,2);
plot (t,Y1);
X label ('time');
Y label ('amplitude');
title ('snr 10');
```

(iii)
```
Subplot (4,1,3);
Plot (t,Y2);
Xlabel ('time');
Ylabel ('Amplitude');
title ('AWGN [SNR 100] FM');
```

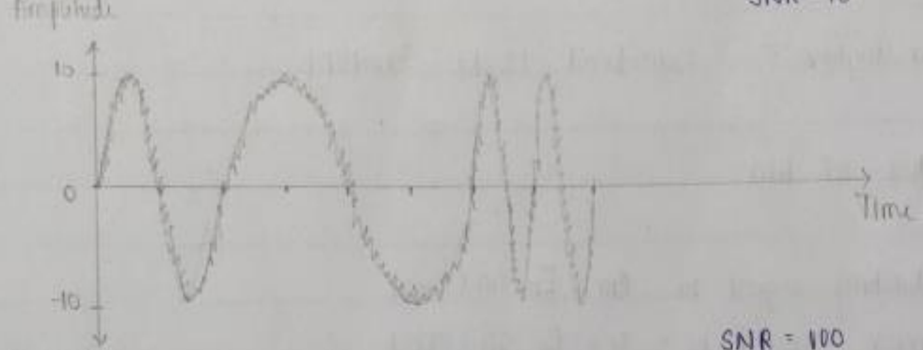
(iv)
```
Subplot (4,1,4);
Plot (t, Y3);
X label ('time');
Y label ('Amplitude')
title ('AWGN [SNR 1000] FM');
```

FM signal with different SNR

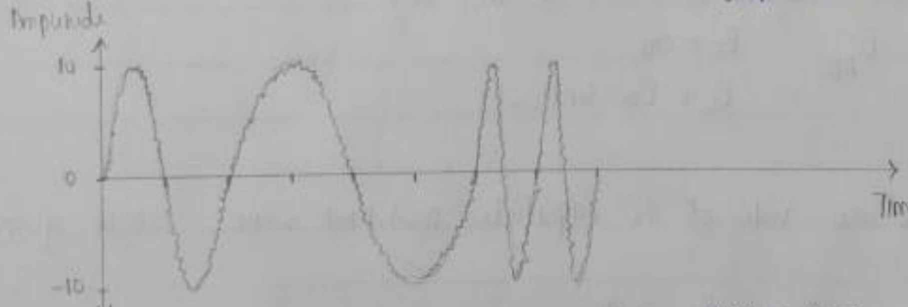
Frequency modulated Signal



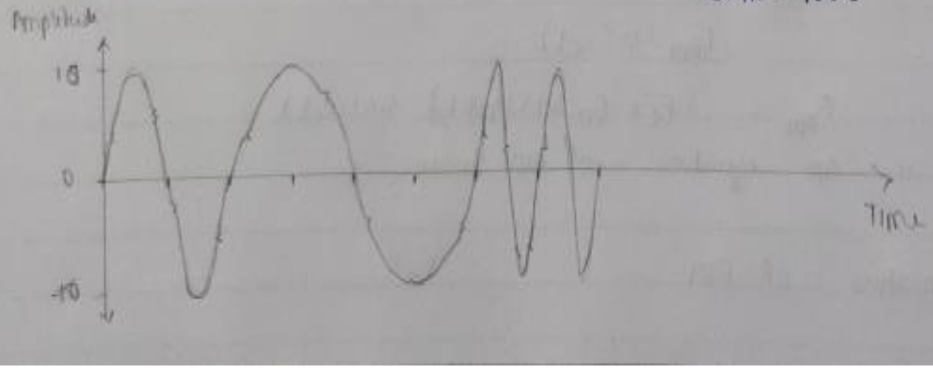
SNR = 10



SNR = 100



SNR = 1000



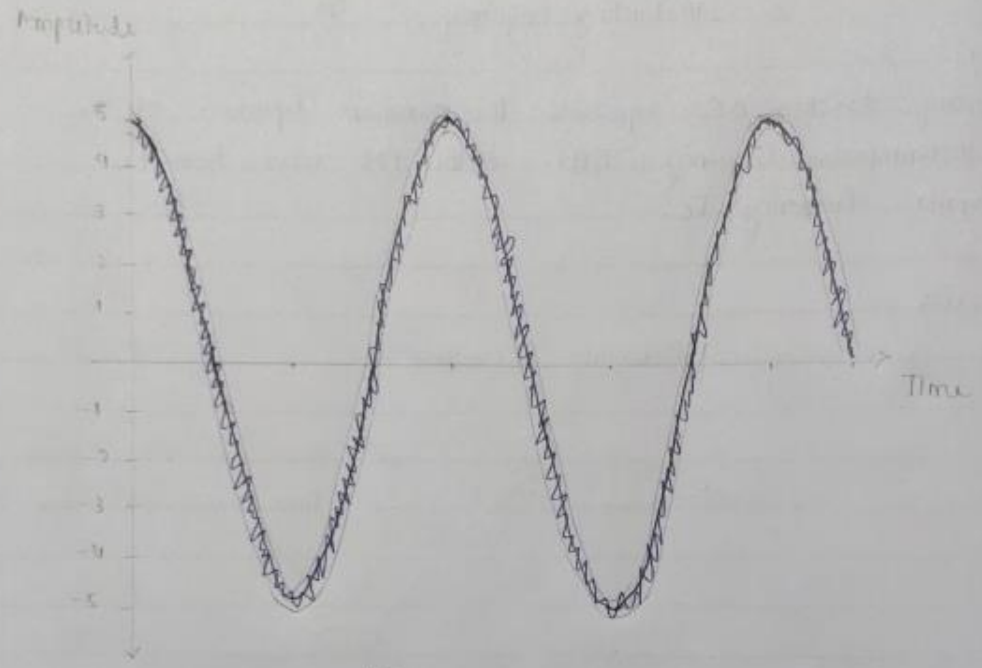
Moving Average Filter

```
clear all ;    clc;
close all ;
fs = 500000 ;
fm = 10000 ;
t = 1: 200;
x = 5 * cos ( 2* pi * (fm/fs) *t) ;
z = awgn (x ,5) ;
Plot (x, 'g', 'linewidth ', 1.5 );
hold on ;
Plot (z) ;  hold on;
for 1:1 :194 ;
y (i) = (z(i) +z(i+1)+ z (i+2) + z (i+3)+ z (i+4)+
  z(i+6) ) /6 ;
  end

Plot (y,'r', 'linewidth', 1.5);
legend ('Actual', 'noisy','filtered');
title ('moving Average filter ;'frontsize ', 12);
x label ('--> time in 2 us');
y label ('---> Volts ');
```

CONCLUSION : we have successfully studied the effect of AWGN on transmission of Amplitude modulation (AM) and frequency modulation (FM).

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



Moving Average Filter

— = Actual — = Noisy - Filtered

EXPERIMENT - 8

SINGLE SIDE BAND (SSB - SC) MODULATION SCHEME

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

APPARATUS: MATLAB

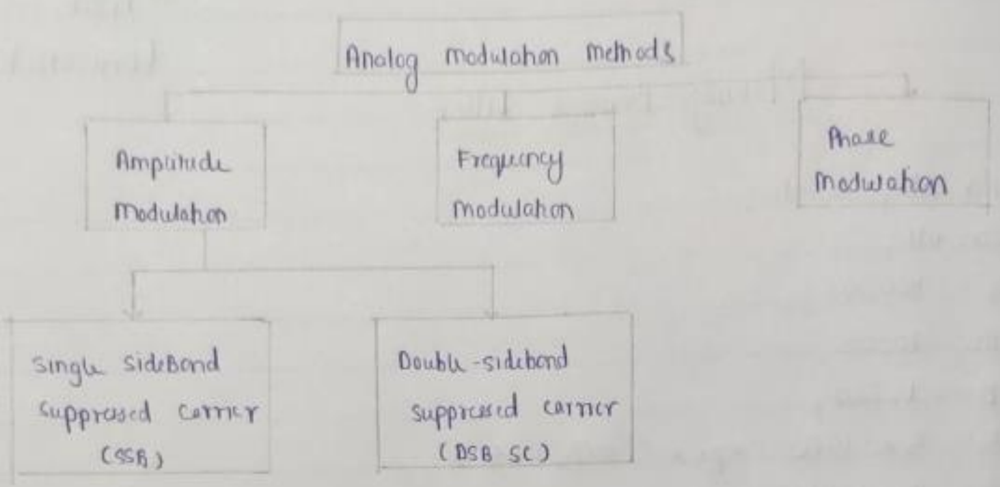
THEORY: 1> modulation: modulation is a process by which some characteristics of a carrier wave is varied in accordance with a modulating (message) signal.

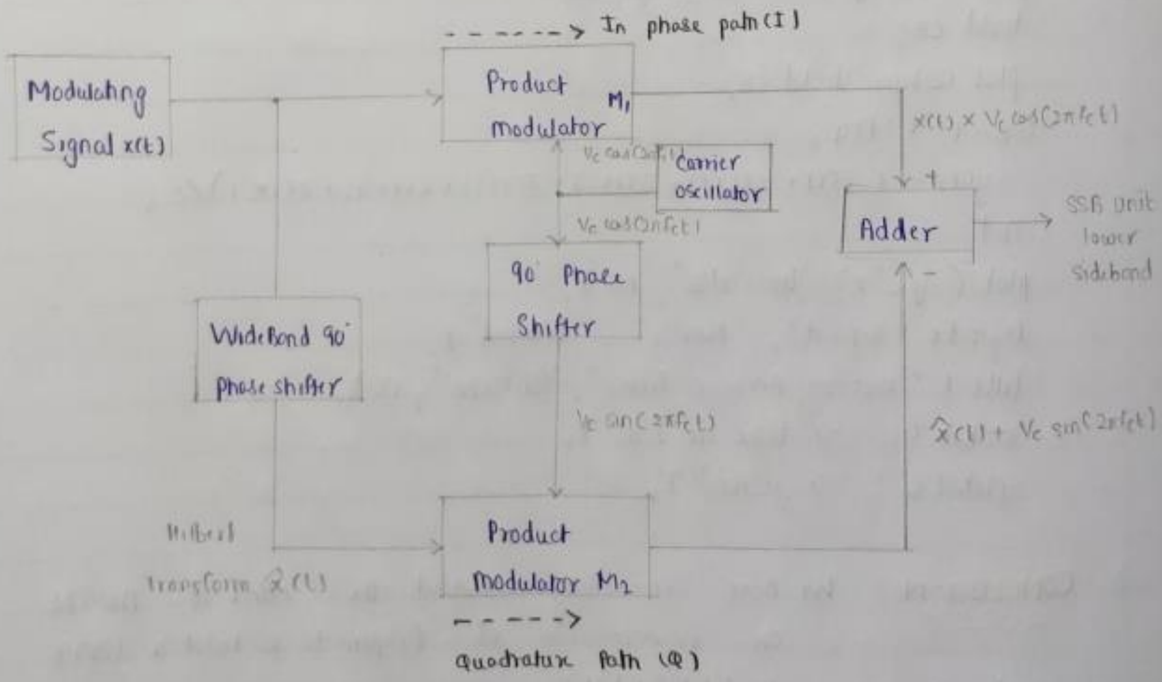
2> Analog modulation: It is a kind of modulation, where the message signal and the Carrier wave both are analog in nature.

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

→ SSB - SC is a type of Amplitude modulation.

→ In Converted AM, we have two side Band and the One carrier wave (no information is contained by the Carrier)





Generation of SSB-SC : By Hibert Transform method

→ In SSB-SC modulation, only one side band is transmitted because both USB (upper side Band) and LSB (lower Side Band) having the same information.

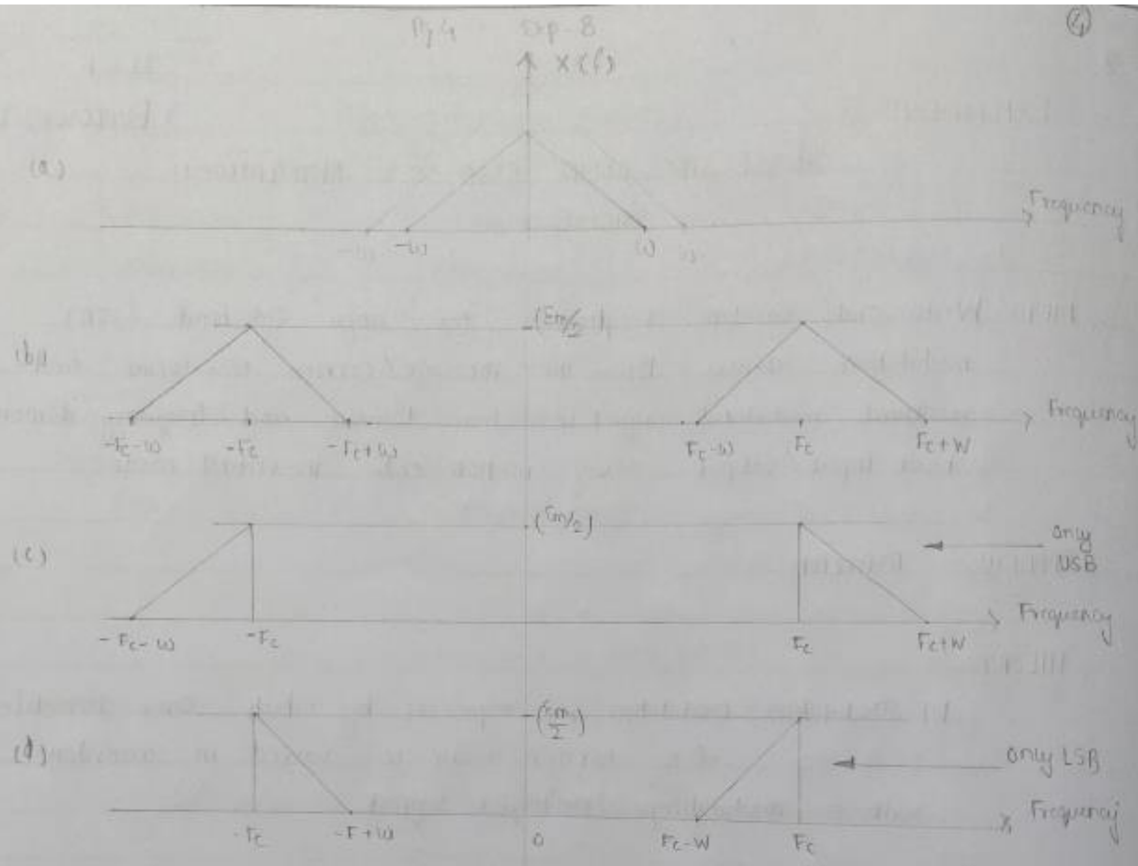
→ Therefore, the transmission bandwidth is reduced to half and also required less power compare to other method of A.M.

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

(i) frequency discrimination method
(ii) Hilbert transform method or phase discrim. method.

5) Applications:

→ In point-to-point Communications
→ Radar communication
→ Where the power saving and low bandWidth requirements are important
→ In many voice applications.

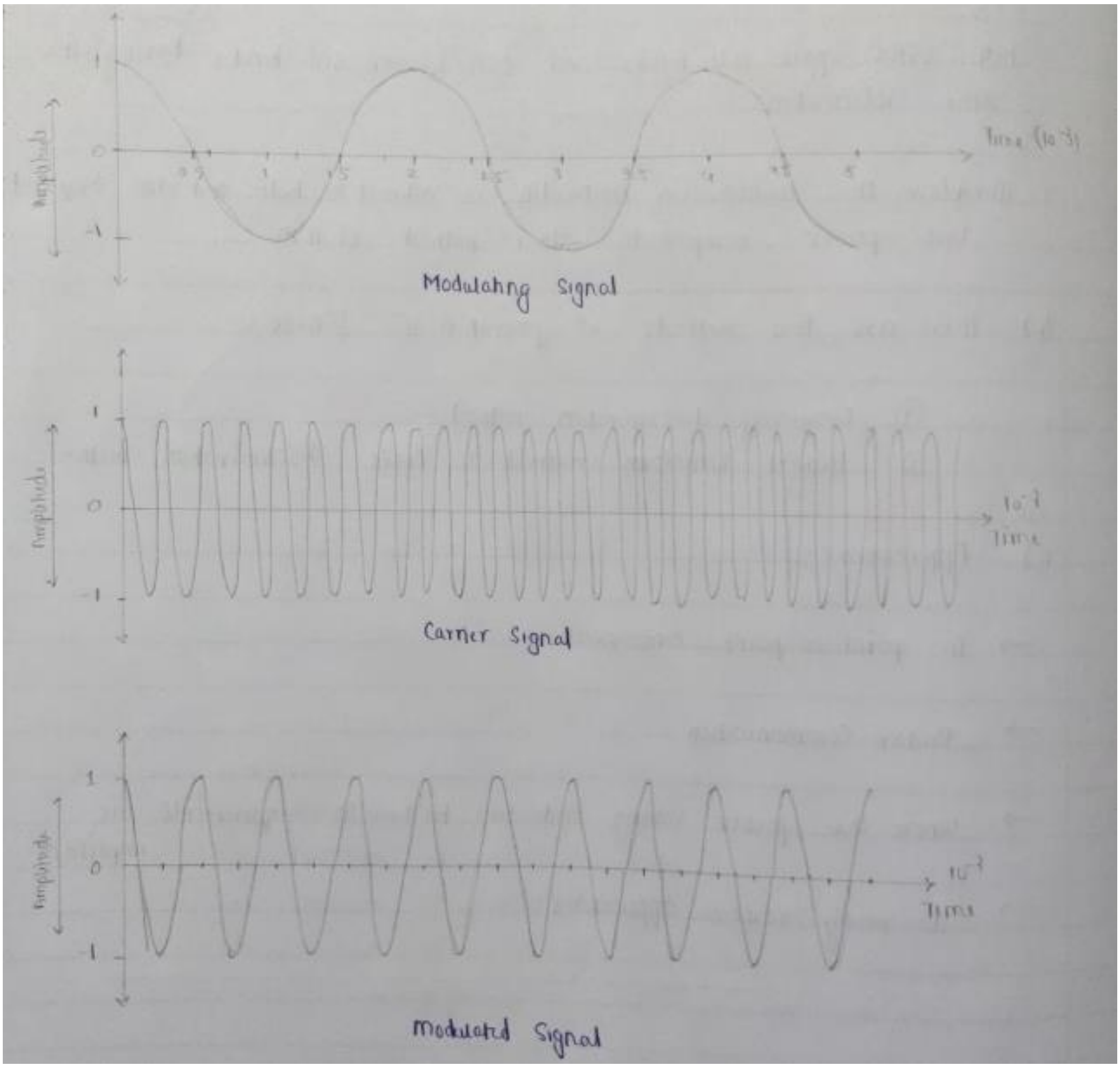


(a) Spectrum of message Signal
(b) Spectrum of DSB-SC wave
(c) spectrum of DSB-SC with only USB Transmission
(d) Spectrum of DSB-SC with only LSB Transmission

MATLAB CODE

```
clc;
clear all;
close all;
am =1;
ac =1;
fm =500;
fc = 5000;
fs = 100000;
ts = 1/fs;
N = 10000;
t= (-N/2 :1 : (N/2-1)) *ts;
m= am * cos (2* pi *fm *t);
mh= am * sin (2*pi * fm*t);
c = ac * cos (2* pi * fc *t );
ch = ac* sin (2* pi * fc *t);
st = m*c - mh.* ch;

% 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;
sub
```



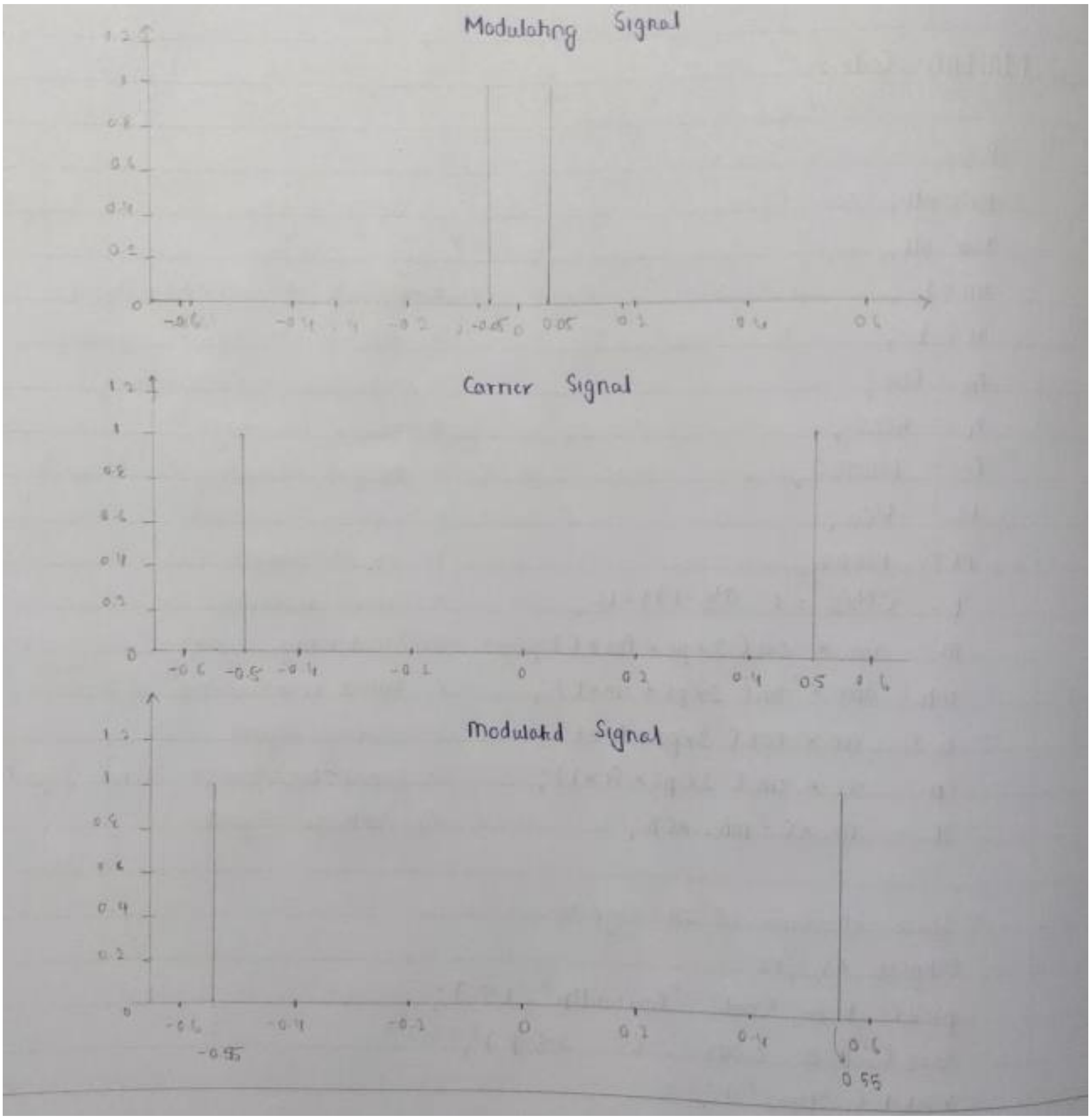
Amplitude
Time (10^-3)
Modulating Signal
Amplitude
Time
Carrier Signal
Amplitude
Time
Modulated Signal

```
Subplot (3,2,3)
Plot (t,c,'black','line width',1.5);
axis ([0 0.005 -2.5 2.5]);
X label ('time');
Y label ('amplitude');
title ('carrier signal');
grid on;

subplot (3,2,5)
Plot (t,st,'blue','line width',1.5);
axis ([0 0.005 -2.5 2.5]);
X label ('time');
Y label ('amplitude');
title ('modulated signal');
grid on;

% Spectrum of all signals

f = (-N/2 : 1: (N/2-1)) * fs/N;
M = abs ((2/N) * fft shift (fft(m)));
C = abs ((2/N) * fft shift (fft(c)));
Sf = abs ((2/N) * fft shift (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');
Y label ('Amplitude');
title ('modulating signal');
grid on;
```


Modulating Signal
Carrier Signal
Modulated Signal
-0.55
0.55

```
Subplot (3,2,4);
 Plot (f, C/max(c), 'black', 'linewidth', 1.5);
 axis ([-2*fc 2*fc -0.1 1.1 ];
  x label ('freq');
  y label ('amplitude');
  title ('carrier Signal');
  grid on;

Subplot (3,2,6);
Plot (f, SF/max(sf), 'blue', 'lineWidth', 1.5);
axis ([-2*FC 2*FC -0.1 1.1]);
xlabel ('frequency');
y label ('amplitude');
title ('modulating signal');
grid on;
```

CONCLUSION :

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

Experiment - 9

PULSE 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 in virtual mode.

APPARATUS : MATLAB (software)

THEORY :

1) Pulse code modulation (PCM)

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

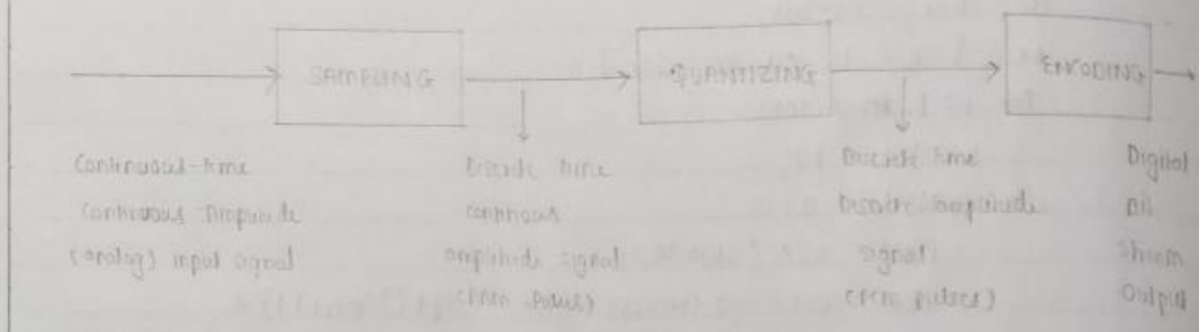
→ PCM is a preferred method of communication within public switched telephone network (PSTN)

→ A PCM stream is determined by two following steps :

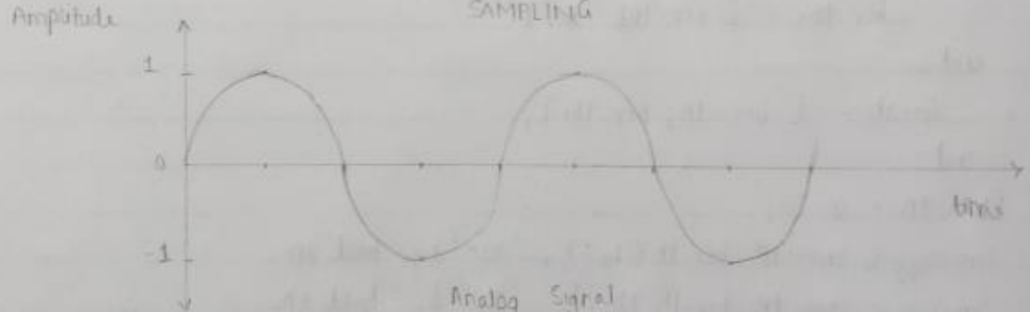
a) Sampling Rate : Which the number of times per second that samples are taken.

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

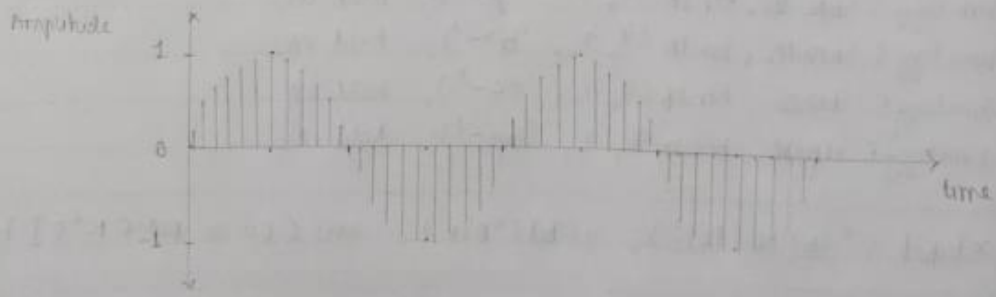
- Hence, the output of PCM resembles a binary sequence.



SAMPLING



Analog Signal



Discrete Time Signal

2) Reasons for Digital Transmission

→ less susceptible to interference cause by noise due to discreate level.

→ Easy to detect errors due to discreate levels.
→ Easy to Encrypt (Higher Security)
→ Simpler to store digital data

3) Sampling

→ Sampler extract samples of a Continuous signal.
→ Sampler produces samples that are equivalent to the instantaneous value of the Continuous signal at the specified various points.
→ The sampling process generates flat-top pulse amplitude modulated (PAM) signal.

4) Quantization

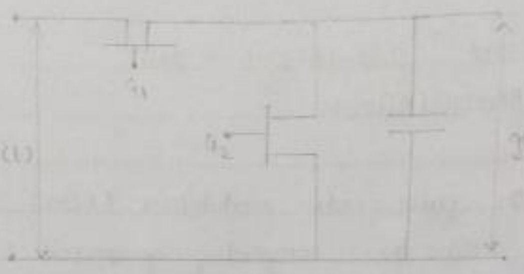
→ Quantization is done by dividing the range of possible values the analog samples into some different levels and assigning the center values of each level to any sample in the quantization interval.

→ Quantization approximates the analog signal values with the nearest Quantization values.

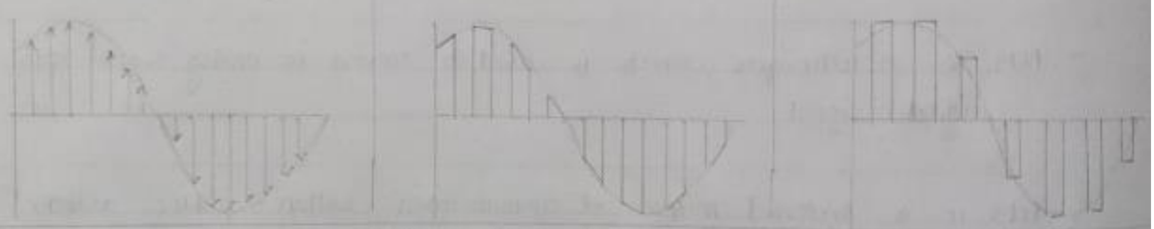
5) Pulse code modulation (PCM)

→ It is a method of converting an analog signal into a d.B.

Flat-top PAM

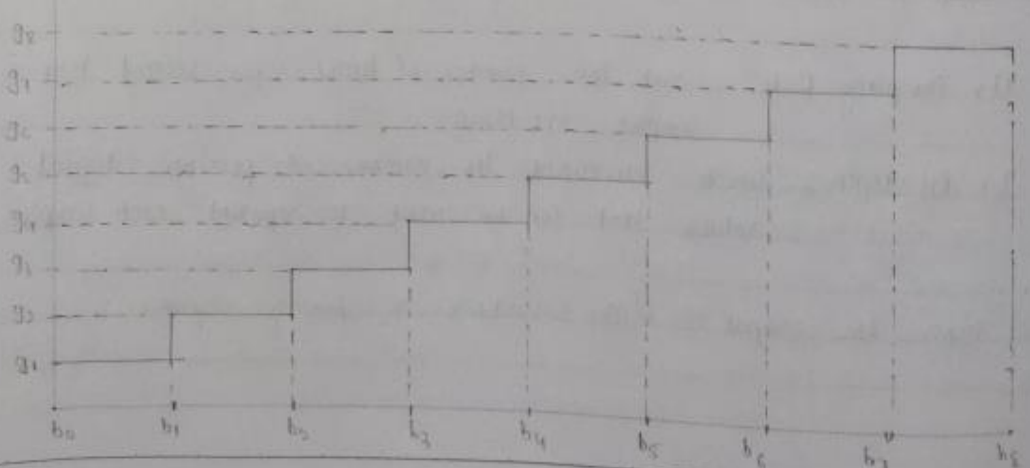


Instantaneous Sampling	Natural Sampling	Flat-top Sampling
It is not practical method	This method is used practically	This is also used practically
Sample rate = infinity	Sample rate satisfied Nyquist	Sample rate satisfied Nyquist



Uniformly Quantized Signal

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



→ PCM Produces a series of number or digits instead of pulse train.

→ Each one of these digits, in binary code, represents the approximate amplitude of the signal sample at the instant.

6) Concluding Remark for PCM.

→ In PCM Transmitter, the signal $x(t)$ is first Passed through the low pass filter of cut-off frequency f_m Hz.

→ This low pass filter blocks all the frequency Components above f_m Hz. This means That now that signal $x(t)$ is band-limited to 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.

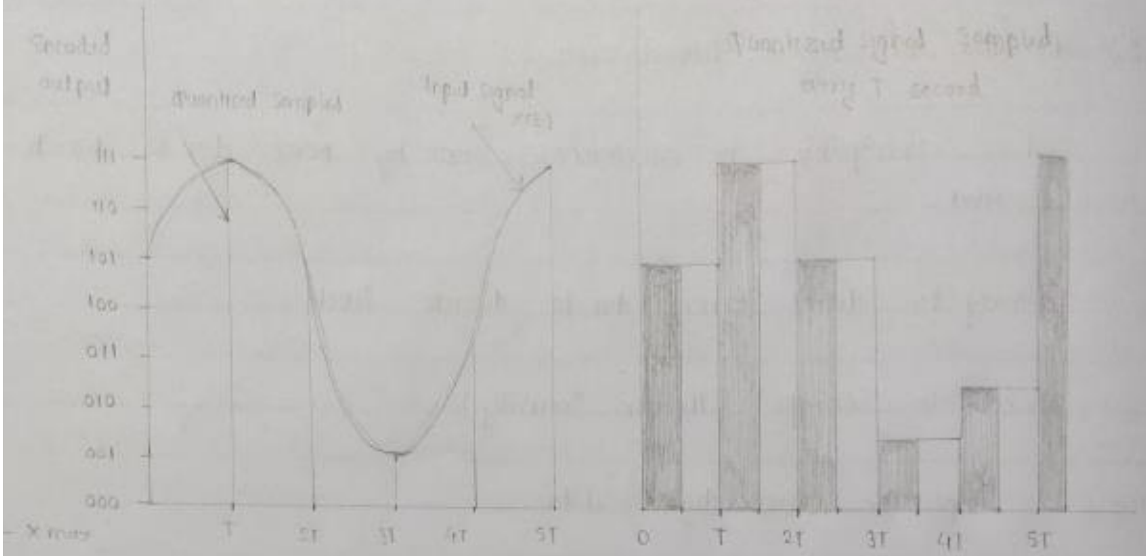
→ The output from sample & hold circuit is denoted by $x(nT_s)$

→ This signal $x(nT_s)$ is discrete in time & continuous in Amplitude.

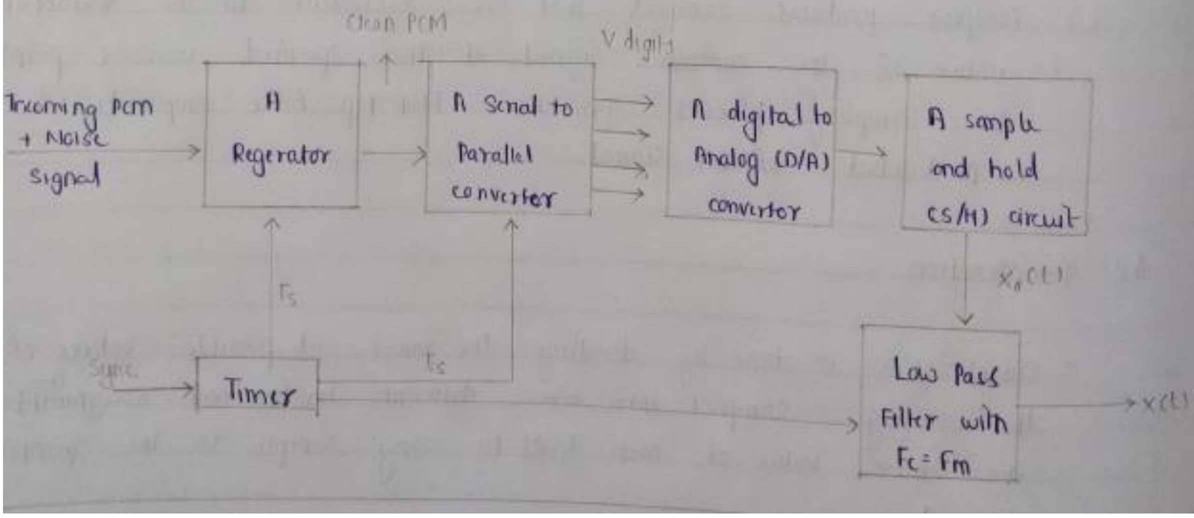
→ A q-level quantizer compares input $x(nT_s)$ with its Fixed digital level.

Transmitter

Figure : 1 Quantization of a sampled Analog signal



PCM Receiver



7) Pcm standard

→ There are two standards of PCM :- (i) The european standard
(ii) The American standard.

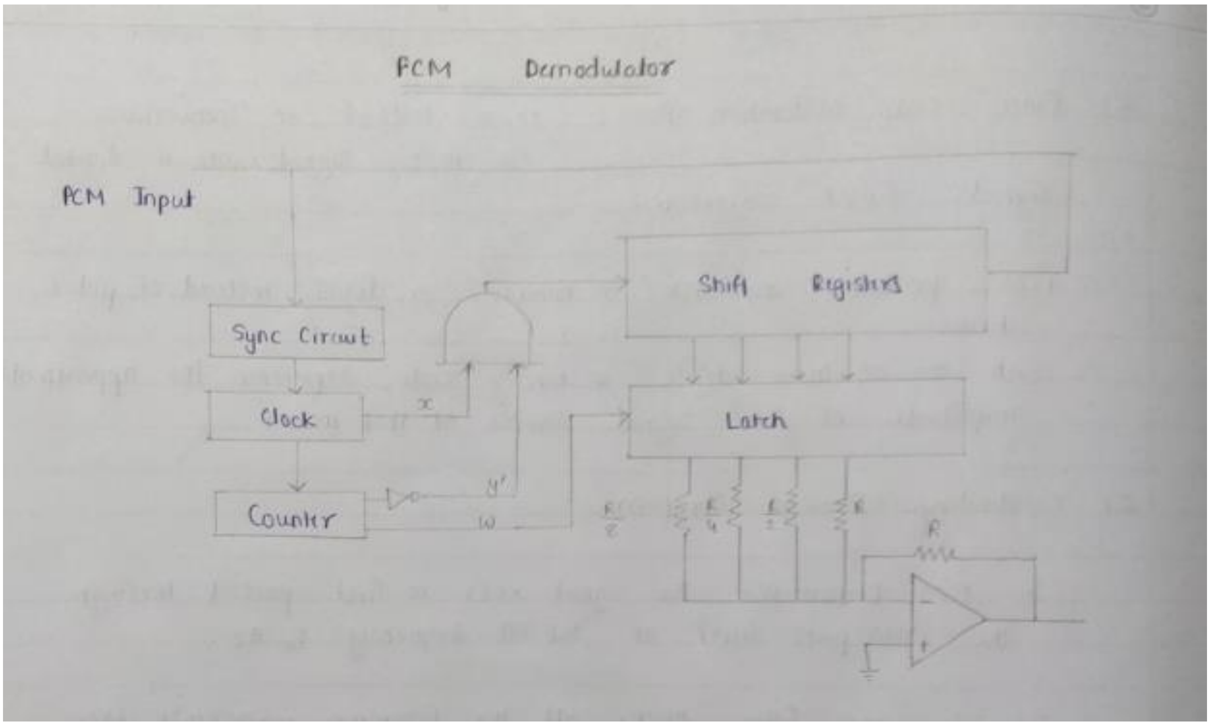
→ The differ slightly in the detail of their working but the principles are the same.

→ European pcm = 30 channels
→ North American pcm = 24 channels.
→ Japanese pcm = 24 channels.

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

8) Applications.

→ In compact Disks.
→ Digital Telephony
→ Digital Audio Applications.


PCM Demodulator
PCM Input
Sync Circuit
Clock
Counter
Shift Registers
Latch
x
y'
w
R/8
R/4
R/2
R

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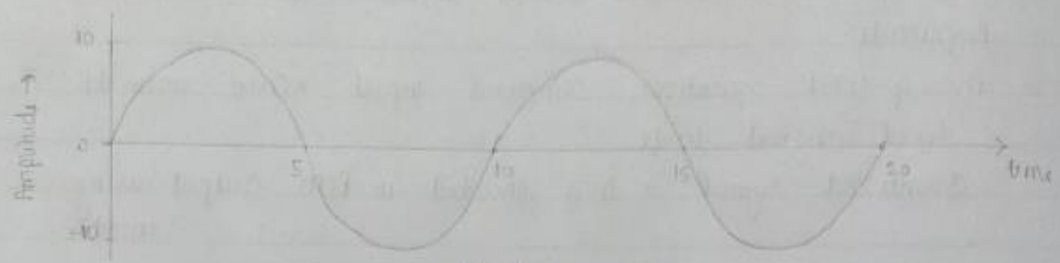
MATLAB CODE :

① sampling.

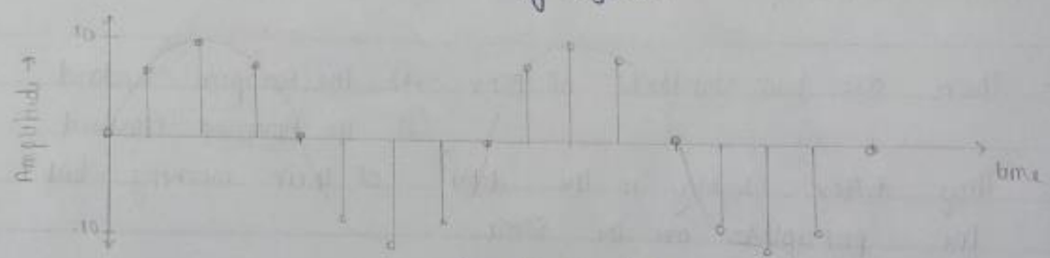
```
n = input ('Enter n value for n-bit Pcm system:');
n1 = input ('Enter number of samples in a period:');
L = 2^n ;

% % signal generation
% x = 0 : 1/100 : 4*pi ;
% y = 8*sin (x)
%subplot (2,2,1) ;
%plot (x,y) ;
% grid on ;
% SAMPLING OPERATION.
    x = 0 : 2*pi / n1 : 4*pi ;
    s = 8* sin (x) ;
   subplot (3,1,1) ;
    Plot (s) ;
   title ('Analog signal') ;
   ylabel (' Amplitude --> ') ;
   xlabel ('Time --> ') ;
  subplot
    stem (s) ;
   grid on ;
   title ('sampled signal') ;
   ylabel ('Amplitude --> ') ;
   xlabel ('Time --> ') ;
```

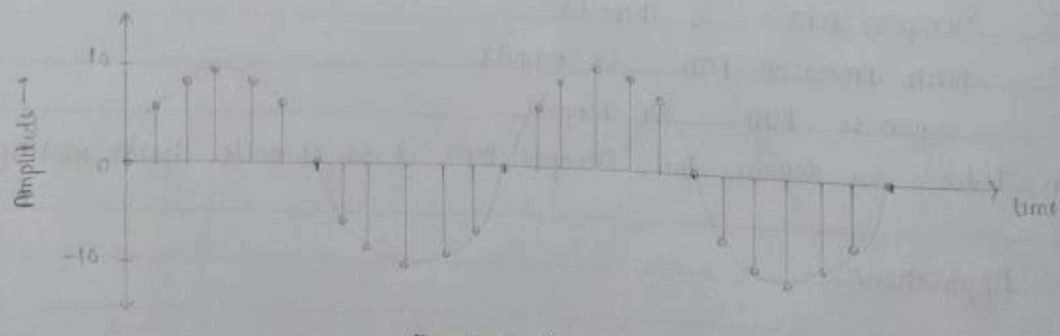
Outputs from MATLAB



Analog signal ↑



Sampled signal (↑)



Quantized Signal

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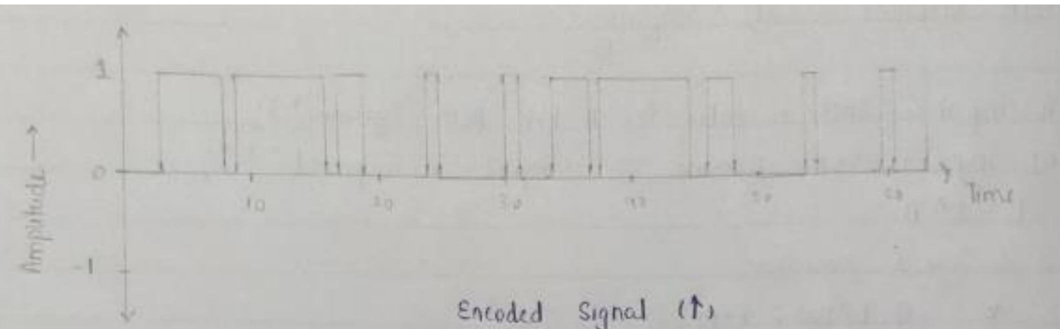
2) Quantization Process .

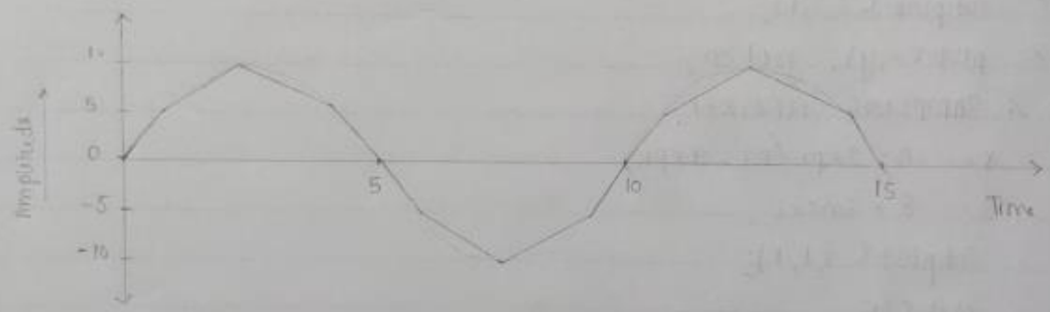
```
Vmax = 8
Vmin = - Vmax;          % level are b/w Vmin & Vmax with diff del.
del = (Vmax-Vmin) /L;
Part = Vmin : del : Vmax ;
Code = Vmin - (del/2) : del : Vmax + (del/2);
[ind, q] = quantiz (s, Part, code);  % Quantization process.

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

for i = 1 : l1          % to make quantize value in b/w lvl.
  if (ind (i) ~ = 0)
     ind(i) = ind (i) -1 ;
  end
end
Subplot (3,1,3);        % Display Quantize Values.
stem (q) ; grid on;
title ('Quantized signal');
Ylabel ('Amplitude -->');
x label ('Time ->');
```



Encoded Signal (↑)



Demodulated Signal

3) Encoding

```
% Encoding Process
figure.
Code = de2bi(ind,'left-msb');  % convert decimal to binary
K=1;
for i=1:l1
  for j=1:n
   Coded (K) = code(i,j);   % convert coded matrix to coded row
   j= j+1;
   K= K+1;
  end
  i= 1+1;
 end
Subplot (2,1,1); grid on;
Stairs (coded)
axis ([0 100 -2 3]);         % Display the encoded signal
title ('Encoded signal');
Ylabel ('Amplitude ->');
Xlabel ('Time -->');
```

4) Demodulation of pcm signal.

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

CONCLUSION :

(i) We successfully demonstrated the pulse code modulation (PCM) & demodulation technique.

(ii) We observed block diagrams for receiver and transmitter of PCM signals. In later stage we also observed demodulation circuit which consist of shift Registers, latch & opamp.

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

EXPERIMENT-10

Date: 25/11/21

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DELTA MODULATION

AIM : To demonstrate the delta modulation (DM) and demodulation technique. show the sampled, quantized / encoded and decoded time domain signal

APPARATUS : MATLAB (software)

Theory : ① Delta modulation.

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

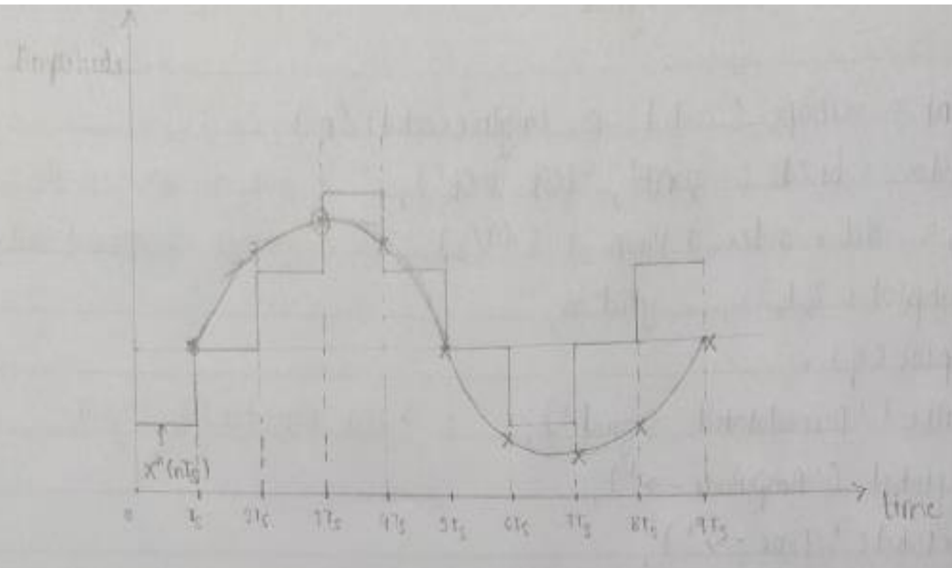
→ In this modulation signal is sent in different form, the data is encrypted / transmitted in 1 bit.

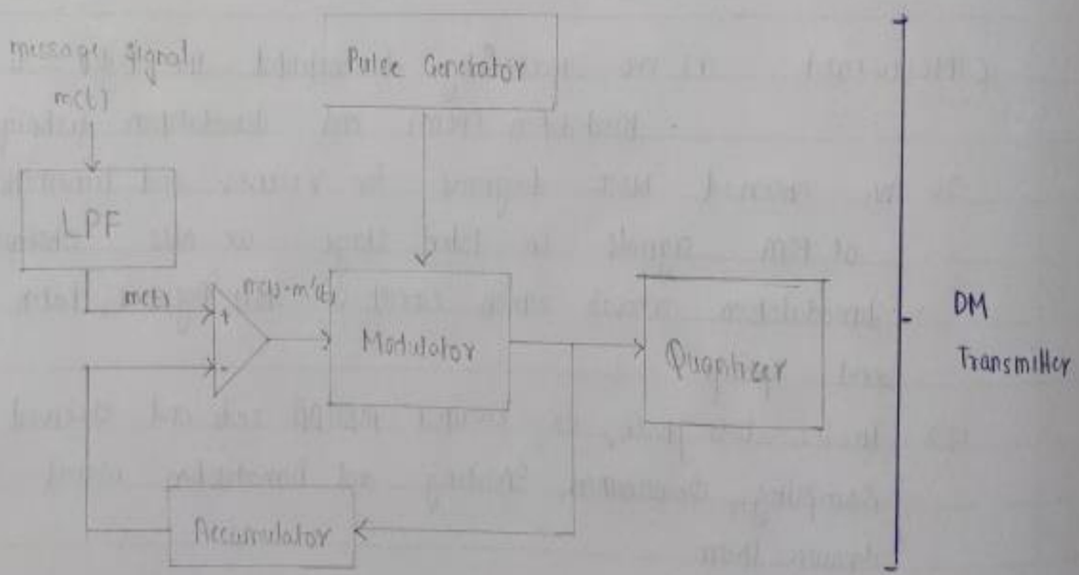
→ The analog signal is approximated with series of segments and each segment is compared to original analog to determine the change in relative Amplitude.

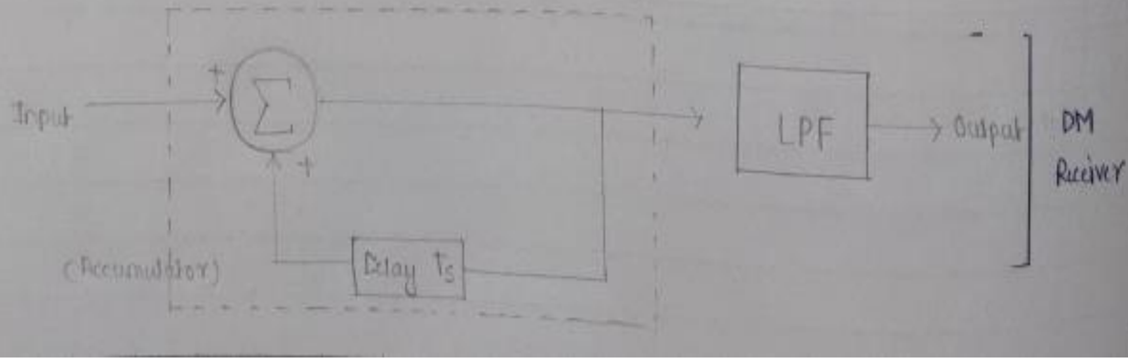
→ Hence, only changes in information is sent & if no change occurs it remains on the same state.

→ 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


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


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


Input
Σ
Delay Ts
(Accumulator)
LPF
Output
DM
Receiver

base-band signal. Here the difference b/w the present sample & previous approximate is quantized into two levels $\pm\Delta$ (delta).

- This is used for Voice transmission.

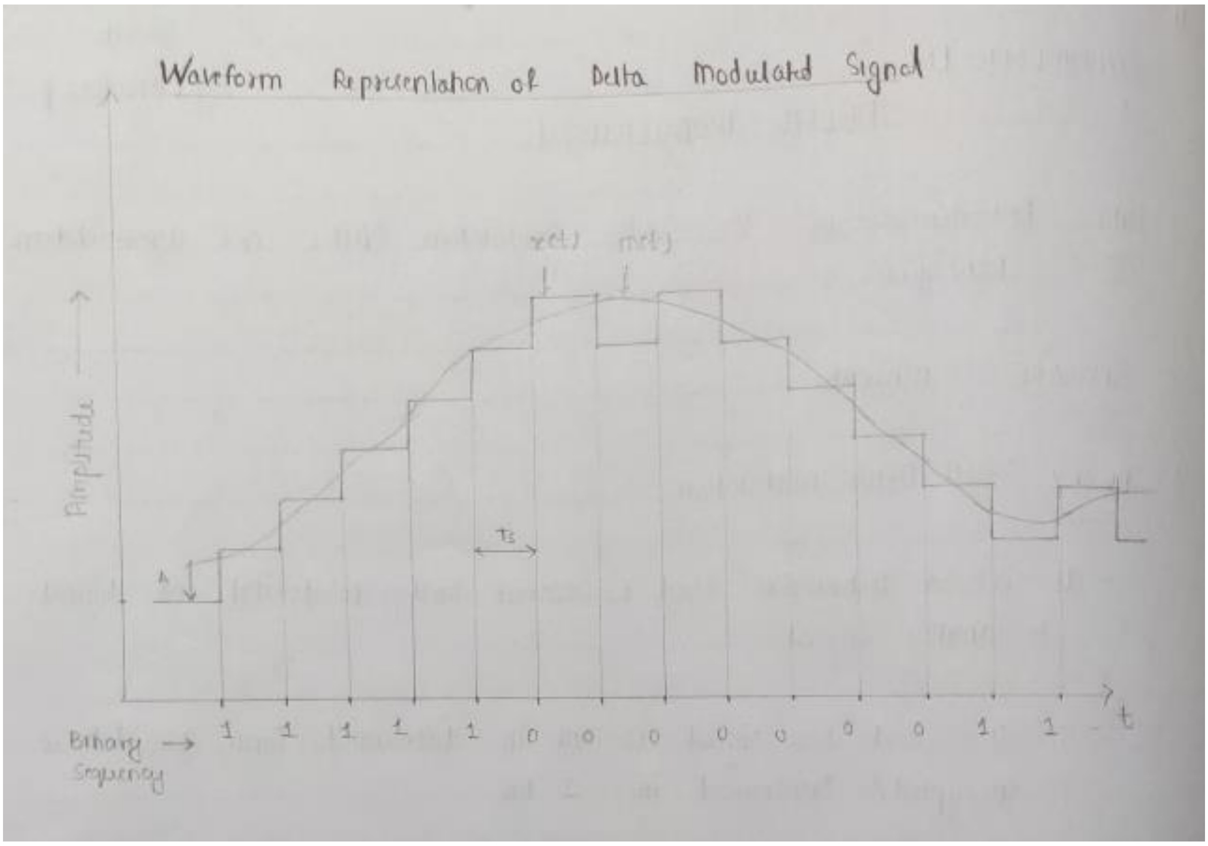
2) Operating principle

→ The operating principle of DM is such that, a Comparison b/w present and previously sampled Value is performed, the difference of which decides the increment or decrement in the transmitted values.

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

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

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



Waveform Representation of Delta Modulated Signal
Amplitude →
x(t)
m(t)
Ts
Δ
t
Binary → Sequency
1 1 1 1 1 0 0 0 0 0 0 0 1 1

3) Advantages of delta modulation.

→ Due to transmission of 1 bit per ~~second~~ sample, it permits low channel bandwidth as well as Signaling rate.

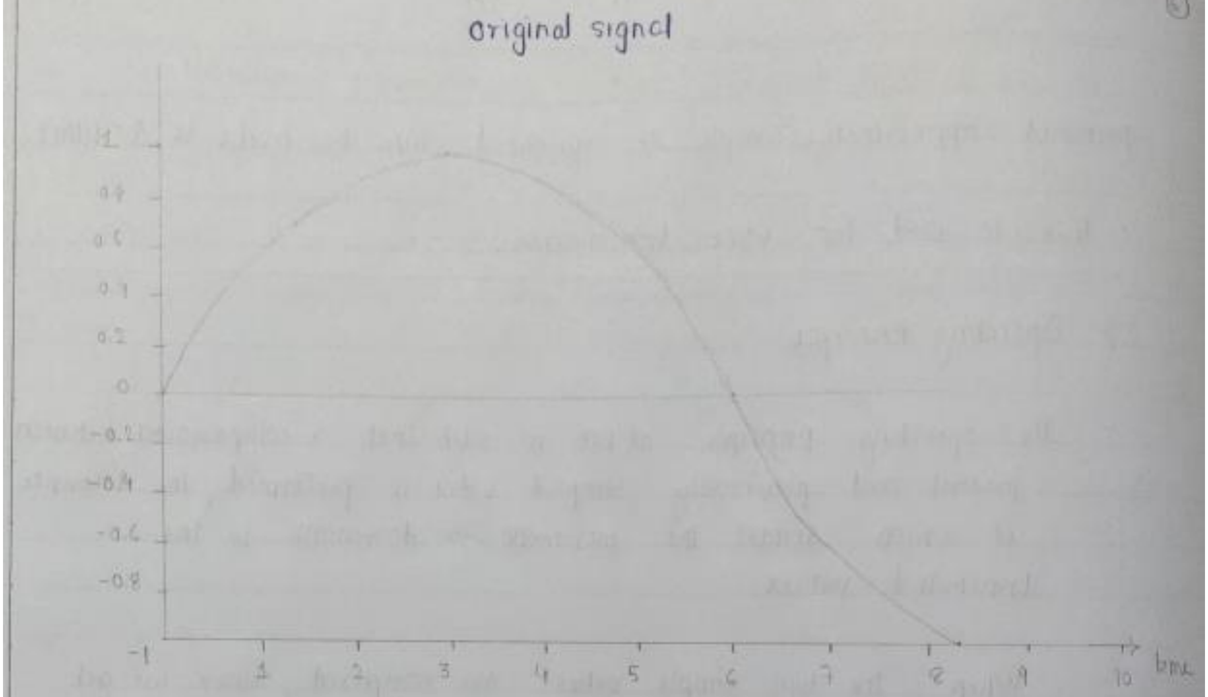
→ ADC is not required thus permits easy generation and detection.

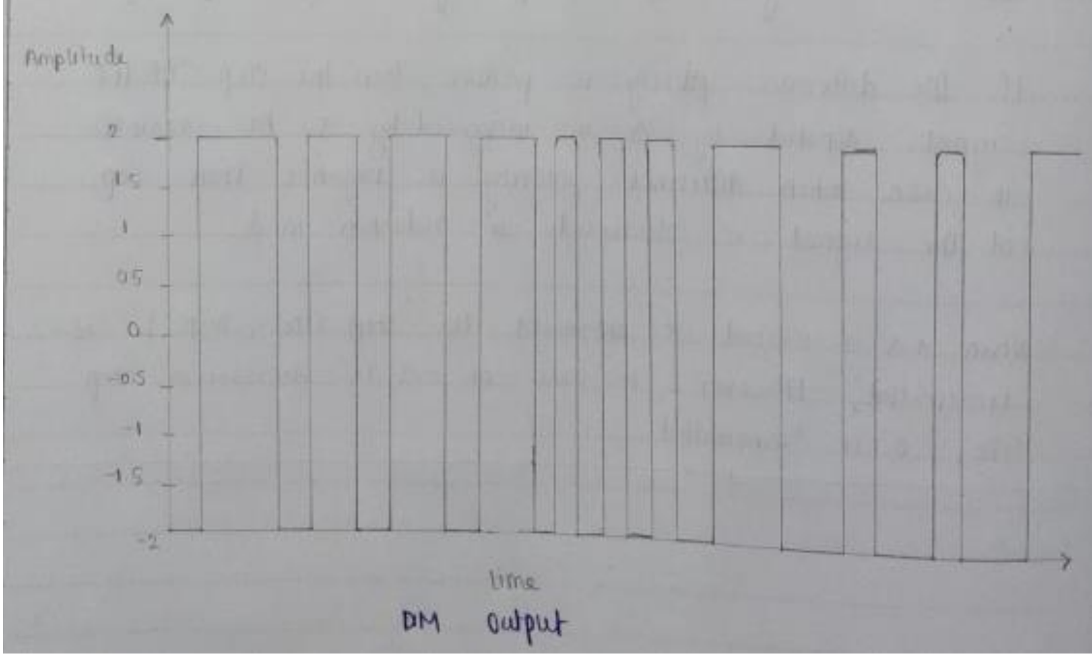
4) Disadvantages of Delta modulation

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

5) Application of Delta modulation.

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


original signal
1
0.8
0.6
0.4
0.2
0
-0.2
-0.4
-0.6
-0.8
-1
1
2
3
4
5
6
7
8
9
10
time


Amplitude
2
1.5
1
0.5
0
-0.5
-1
-1.5
-2
time
DM output

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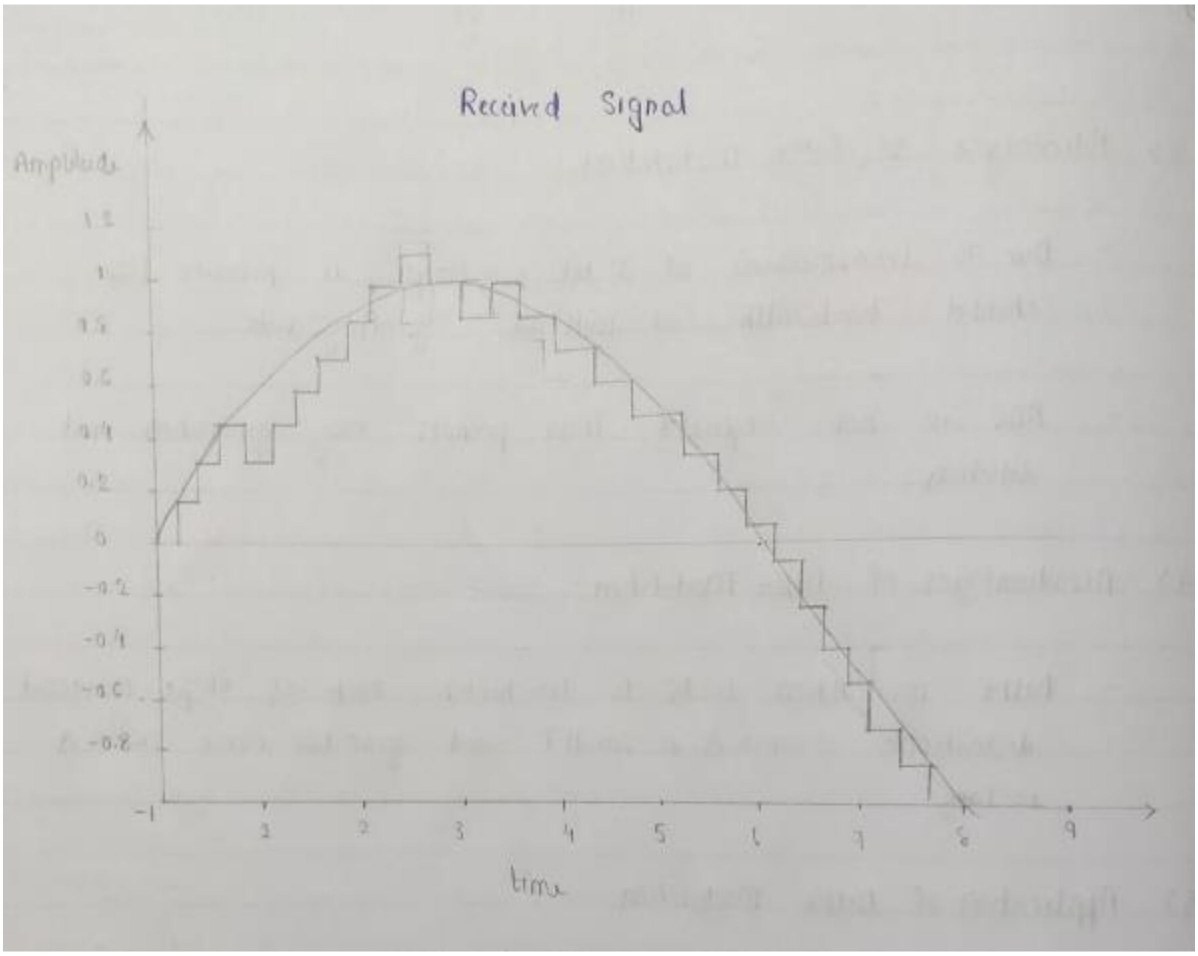
MATLAB CODE

```
%.%. Delta Modulation (DM)

Predictor = [01]
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 sin (2*Pi* 0.1 *t); %. Analog signal

%. Quantize x(t) using DPCM
encoded_x = dpcmenco (x, codebook, Partition, Predictor);
%. Try to recover x from modulated signal.
decoded_x = dpcmdeco (encoded_x, codebook, predictor);
%. plots
figure
Plot (t,x);
xlabel ('time');
title ('original signal');
figure
stairs (t, 10* codebook(encodex +1) . 'g');
x label ('time');
title ('DM output');
figure.
Plot (t,x)
hold ;
```


Received Signal
Amplitude
time

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```
stairs (t, decoded_x);
  grid on;
  xlabel ('time');
  title ('received signal');
```

CONCLUSION :

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

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EXPERIMENT - 11

QAM MODULATION AND DEMODULATION

AIM: To examine the 16-Quadrature Amplitude Modulation (16-QAM) and demodulation scheme. Draw the 16-QAM m-ary mapped signal & modulated waveform.

APPARATUS: MATLAB (software)

THEORY:

① QAM - Quadrature Amplitude modulation or QAM is a form of modulation which is widely used for modulating data signals onto a carrier used for radio communication.

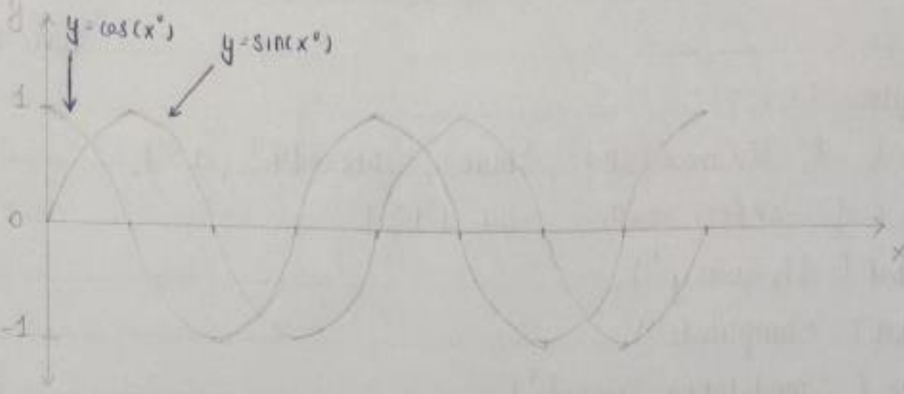
- QAM is a signal in which two carrier shifted in phase by 90° are modulated & the resultant output consists of both Amplitude and phase variations.

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

② Main parameters to be considered while designing any communication system:

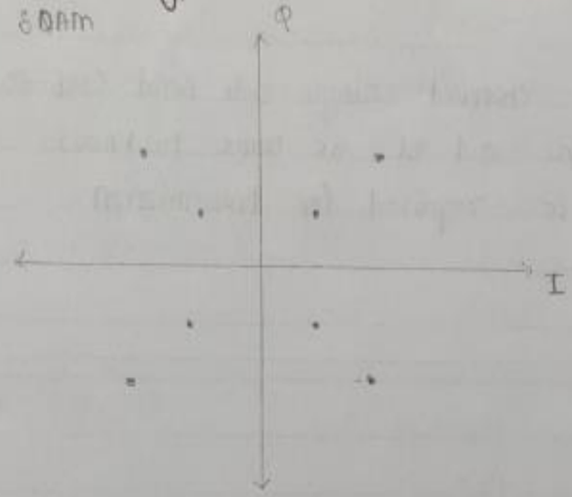
- Transmission Power
- Transmission Bandwidth

Quadrature = Sine wave + Cosine wave

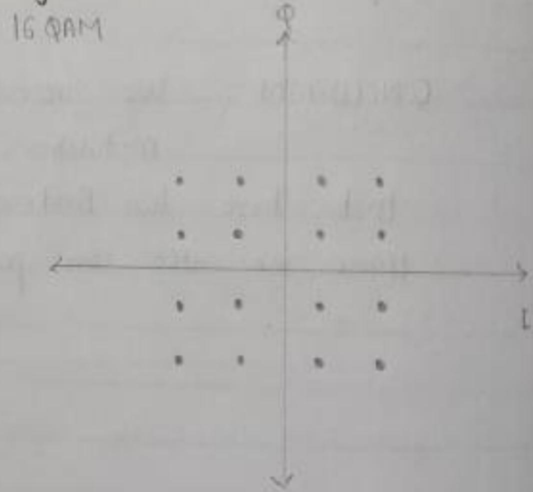


Types of QAM - constellation Diagram ↴

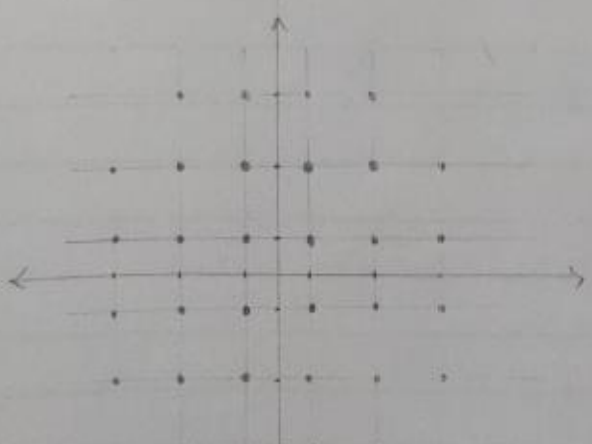
8 QAM



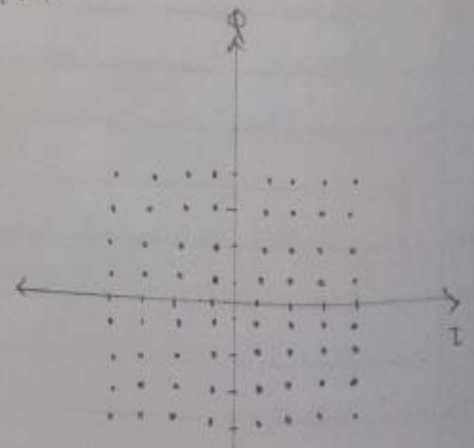
16 QAM



32 QAM



64 QAM



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

3) Why QAM ?

- The main aim is to save bandwidth : Two modulated signal occupies the same transmission channel.

- A motivation for the use of QAM comes from the fact that a straight amplitude modulated signal occupies twice the bandwidth of the modulating signal.

- This is very wasteful of the available frequency spectrum.

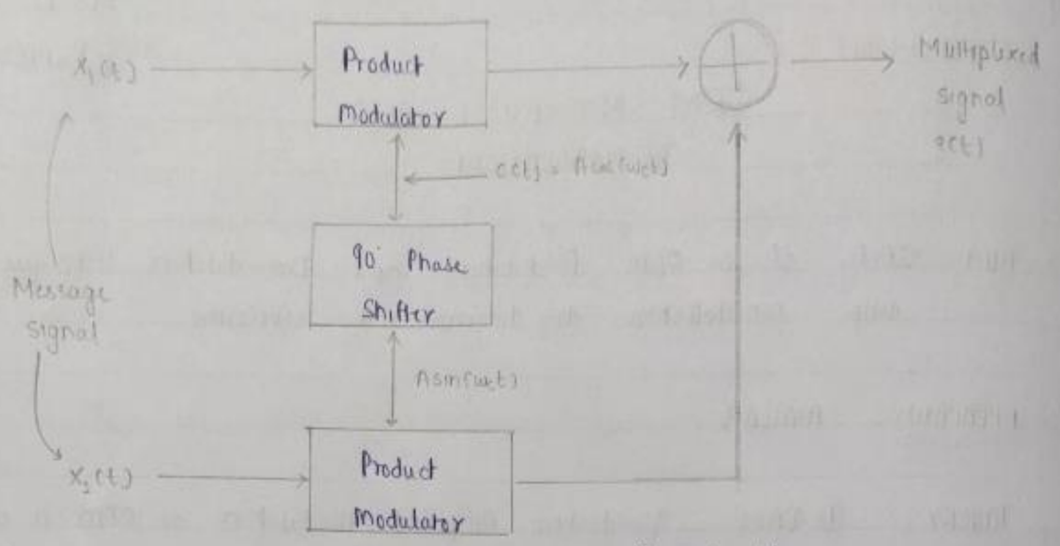
- QAM places two independent double sideband suppressed carrier signals in the same spectrum

4) Types of QAM

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

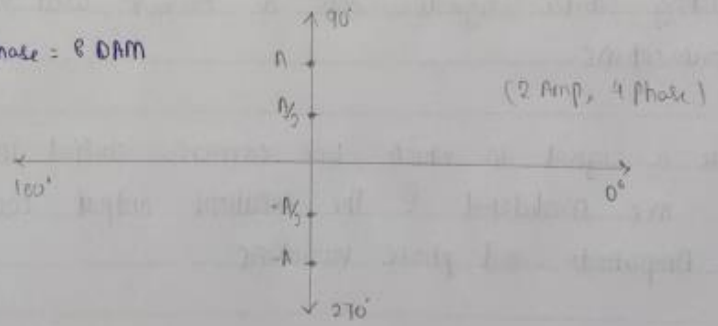
  - 16 QAM
  - 32 QAM
  - 64 QAM
  - 128 QAM
  - 256 QAM

QAM Modulation

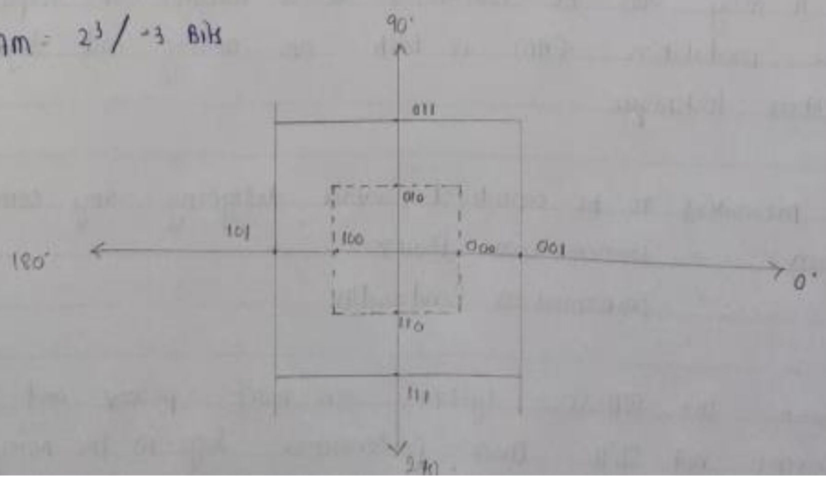


$$s(t) = x_1(t)\, A\cos(w_c t) + x_2(t)\, A\sin(w_c t)$$

2 Amp × 4 phase = 8 QAM



8 QAM = 2^3 / →3 Bits



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5) QAM modulation.

- QAM theory states that both Amplitude and phase changes within a QAM signal.

- The basic way in which a QAM signal can be generated is to generated two signals that are 90° out of Phase with each other & then sum them.

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

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

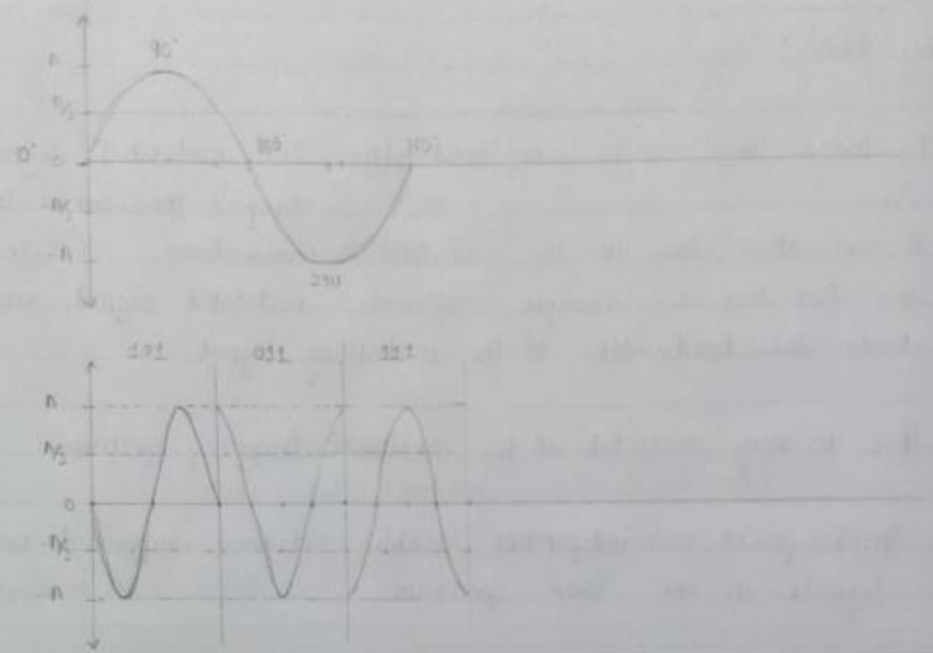
- This signals will not overlap with each other because they are orthogonal

- It is Possible to transmit two DSB-SC signal with in bandwidth of "$2f_m$".

- It provides bandwidth efficiency.

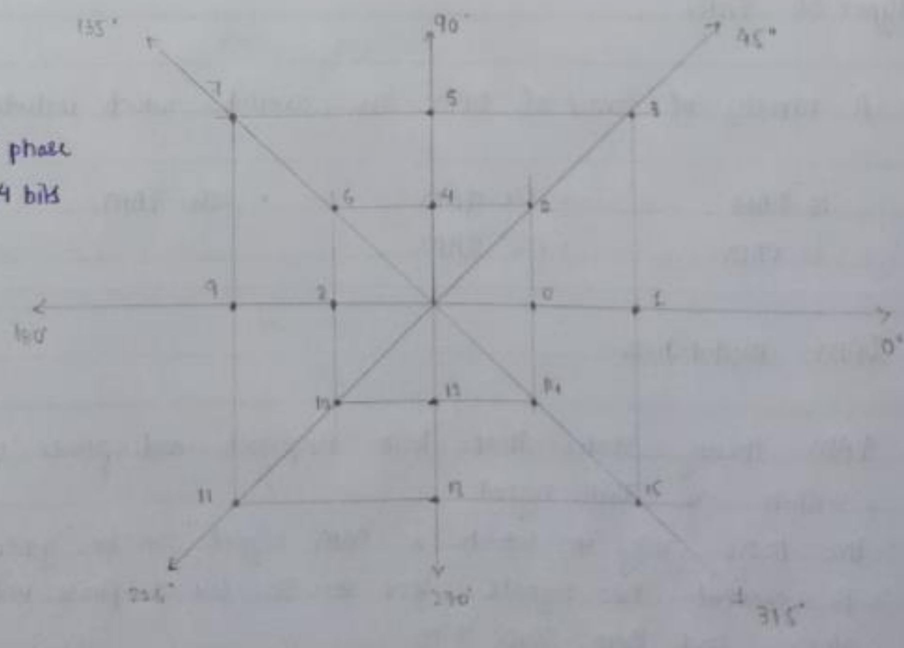
- Gives better performance than SSB & also improves data rate.

Phasor Diagram



16 QAM

2 Amp * 8 phase
= 2^4 / 4 bits



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6) QAM Demodulation

- The QAM demodulator is very much the reverse of the QAM modulator.

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

7) Bit error Rate (Received Bits)

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

- The higher order modulation schemes are considered less relent to noise and interference.

- Many radio communications systems now use dynamic adoptive modulation techniques. They sense the Channel Conditions and adopt the modulation scheme to obtain the heighest data rate for the given conditions.

- M-QAM technique provides better bit error rate performance than m-psk modulation technique.

8) Advantages.

- The advantages of using QAM is that it is a higher order form of modulation. As a result,

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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 Bandwidth.

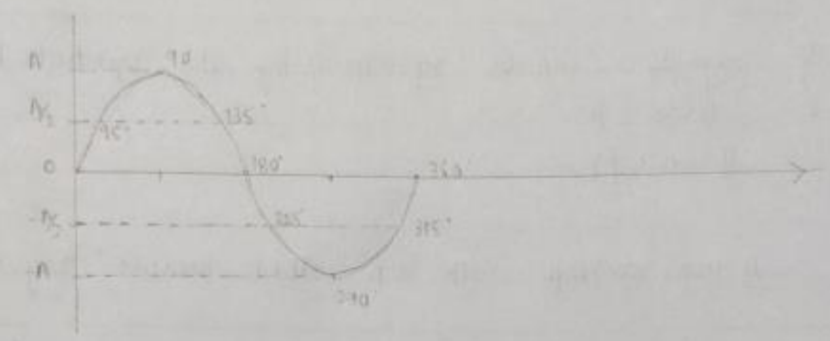
9) Applications

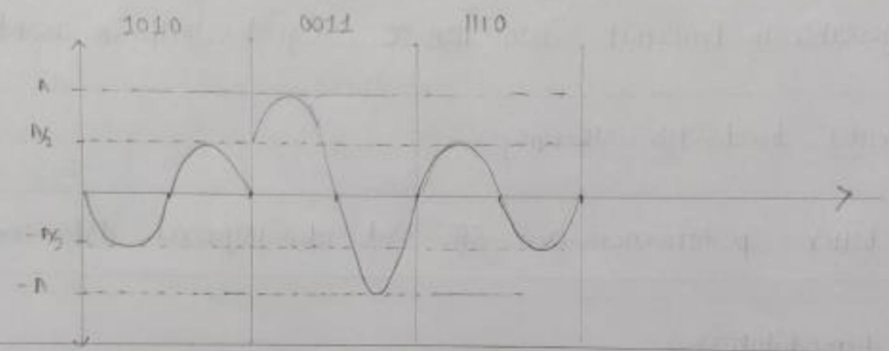
- Quadrature multiplexing is used in color television to multiplex, the so-called chrominance signals which carry the information about colors.

- QAM schme is used on telephone lines for data transmission.

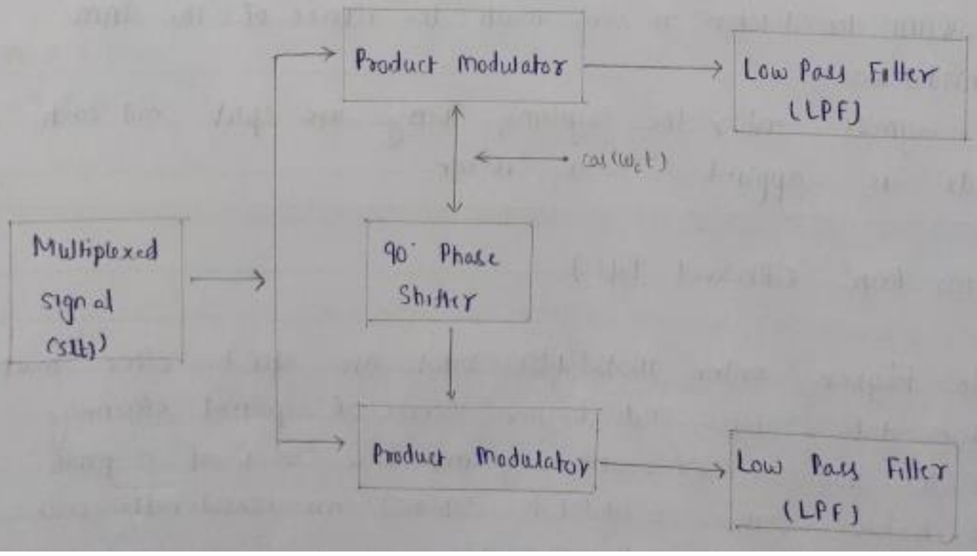
- ultra high Capacity microwave Backhall Systems all use 1024-QAM.

Phasor Diagram





QAM Demodulation

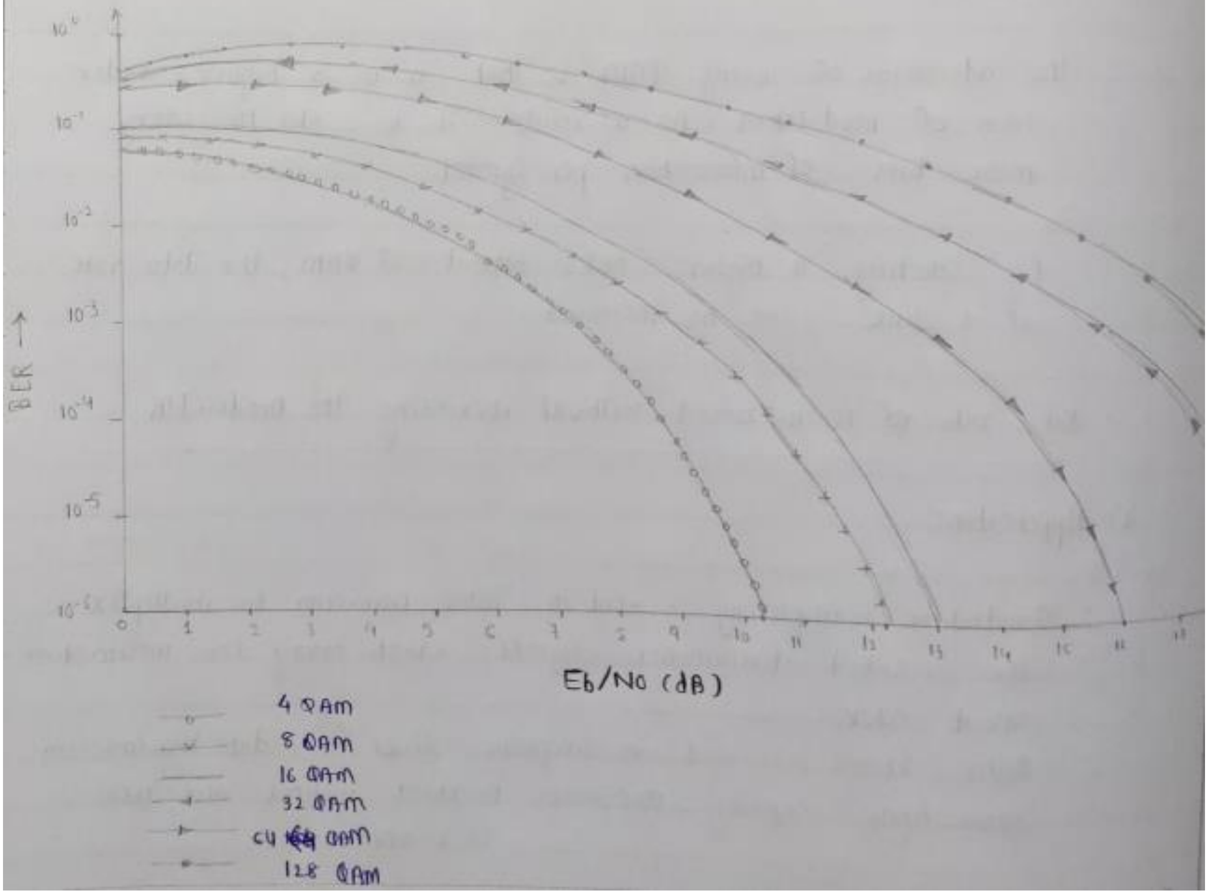


MATLAB CODE

```
ck;
clear all;
close all;
M=16;
x = (0 : m-1);
Y = qommod (x,m);
 scatter Plot (Y);
% z= qam demod (Y, M, Pi/4);
% scatter plot (z);
     ber-1 = [] ;
     ber for  EbNodB =  0:20;
              EbNO = 10^ (EbNOdB/10).;
              ber =(1/log2(m)) * (2*(1-sqrt(1/m)) * Cerg (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'),
Y label ('BER');
title ('BER of 16-QAM');
axis ([0 16 10^-6 10^0]);
grid on;
```

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

"BER Values" using MATLAB



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% BER comparison of various M-QAM

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

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

Conclusion: We successfully examined the 16-Quadratic amplitude modulation (16-QAM) & Demodulation scheme.