EXPERIMENT : No. 7 Effect of AWGN on AM and FM. | Page No. | Date

* 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 Gussian noise channel (AWGN) on AM and FM. signal using the MATLAB /Simulink and draw the distortied 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.

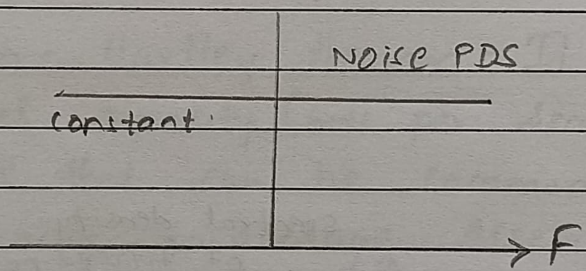
- Additive - The received signal equals the transmit. signal plus some noise where the noise is statiscally independent of signal.

$r(t) = s(t) + w(t)$ → noise.
↓
message signal.

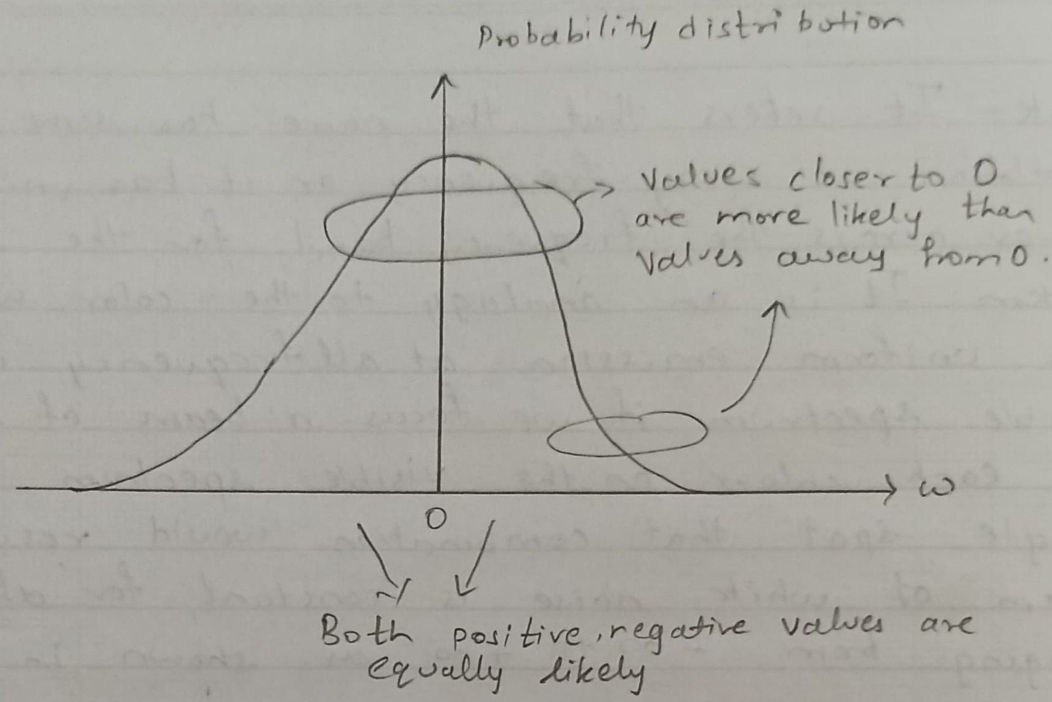
Sundaram

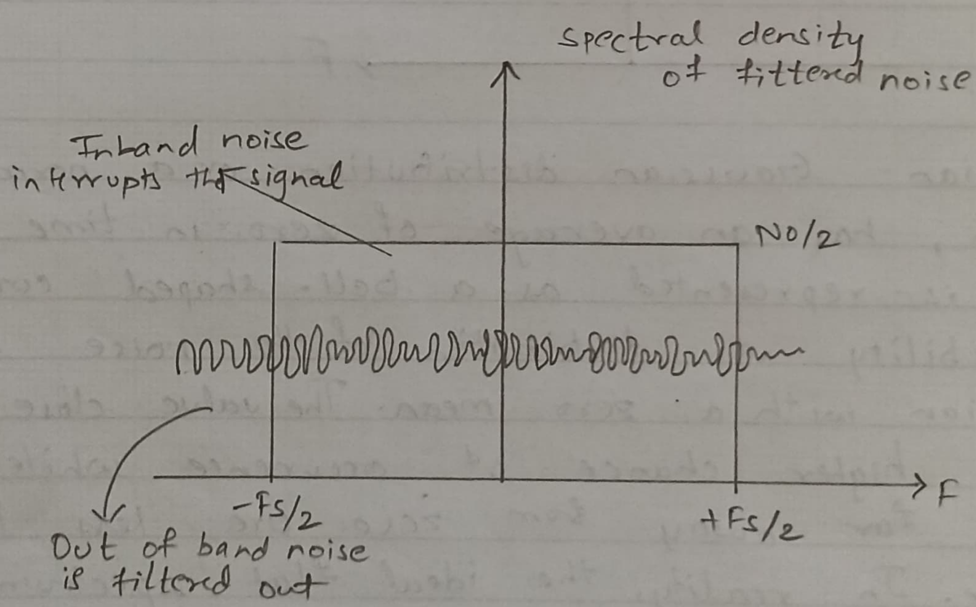
Teacher's Sign. : ____________

- White - It refers that the noise has some 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 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 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 of 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 hundereds of GHz) but not for higher frequencies.


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


spectral density of filtered noise
Inband noise interrupts the signal
N0/2
-Fs/2
+Fs/2
F
Out of band noise is filtered out

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- Signal to Noise Ratio - The SNR 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 0 dB) indicates more signal than noise. SNR, bandwidth and channel capacity of communication channel are connected by shannon - Hartley theorem.

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

- Shannon - Hartley theorem - It indicates that channel capacity (bits per second) or information rate of date 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.

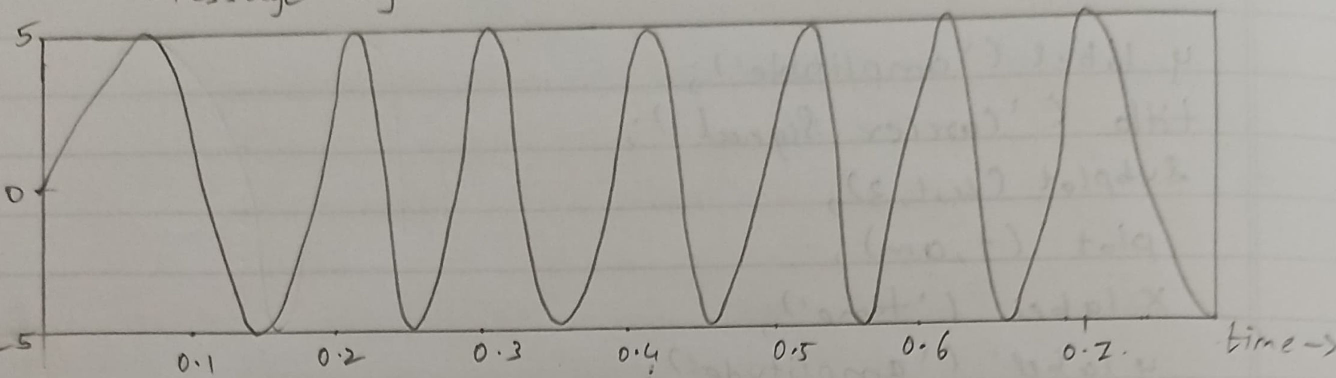
5dB - 10dB → is minimum level to establish a connection due to noise level being nearly indistinguishable from desired signal.

25 dB - 40dB → deemed to be good.

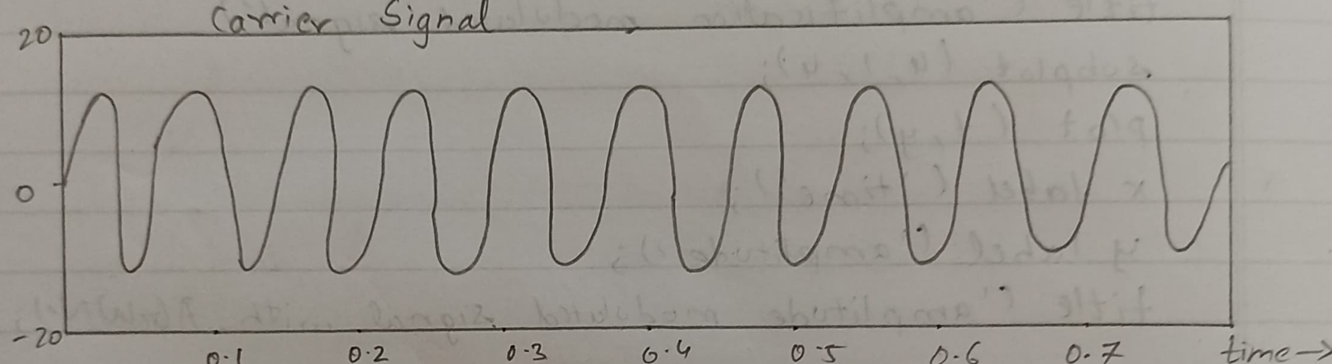
Teacher's Sign. :

AWGN Effect over AM

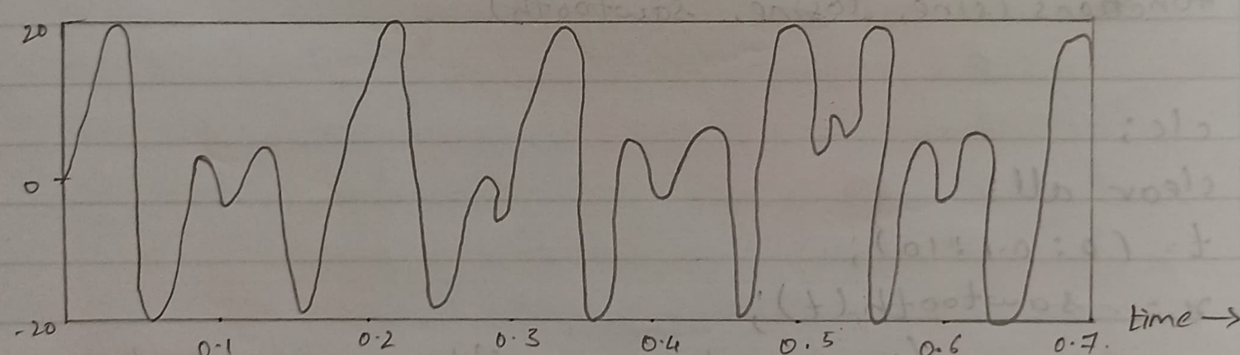
Message Signal



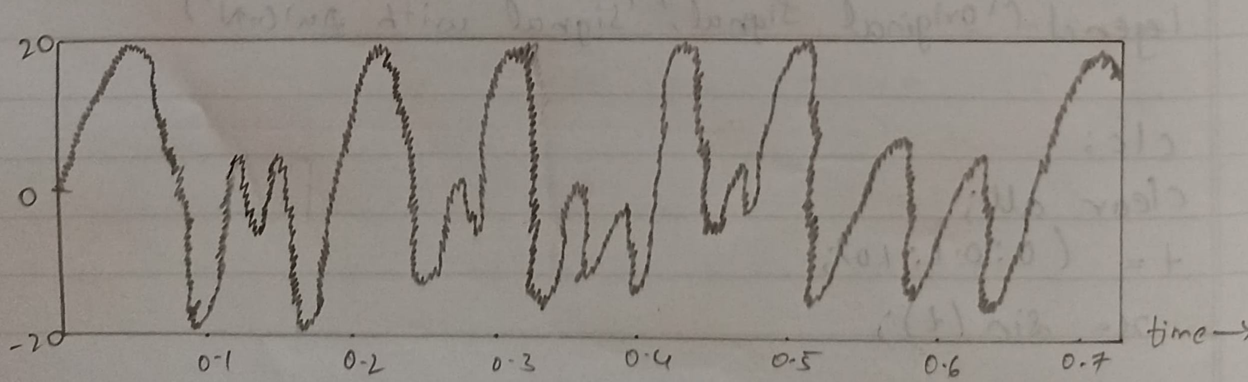
Carrier Signal



Amplitude Modulated Signal



Amplitude Modulated Signal Using AWGN.



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41 dB or higher → deemed excellent.

- AWGN over AM (amplitude modulation)

Let $E_m(t) = E_m \sin \omega_m t$ be message signal &

$E_c(t) = E_c \sin \omega_c t$ be carrier signal.

then $e_m(t) = (E_c + E_m(t)) \sin \omega_c t$

$e_m(t) = (E_c + E_m \sin \omega_m t) \sin \omega_c t$.

* MATLAB code for performing the AWGN effect over 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,     , 'measured');
 subplot (4,1,1);
 plot (t,m);
 xlabel ('time');
```

Teacher's Sign. :

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```
y label ('amplitude');
title ('Carrier Signal');
subplot (4,1,3);
plot (t,am);
x label ('time');
y label ('amplitude');
title ('amplification modulated signal')
subplot (4,1,4);
plot (t,y);
x label ('time');
y label ('amplitude');
title ('amplitude modulated signal with AGWN');
```

* MATLAB Code for AWGN effect on different functions. (sine, cosine, sawtooth)

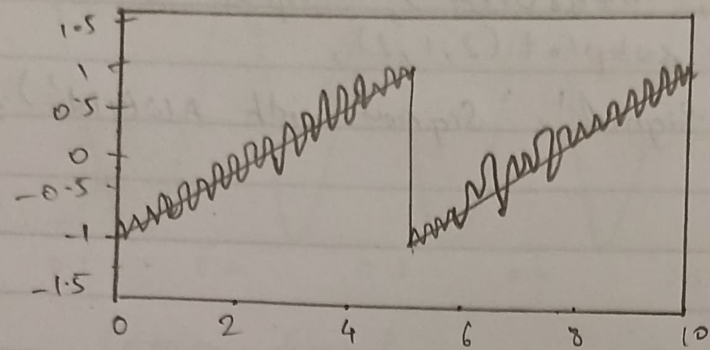
```
clc;
clear all;
t= (0:0.1:10);
x= sawtooth(t);
y= awgn (x,10,'measured');
plot(t,[x,y])
legend ('original Signal,' 'Signal with AWGN')

clc;
clear all;
t= (0:0.1:10);
x= sin(t);
```

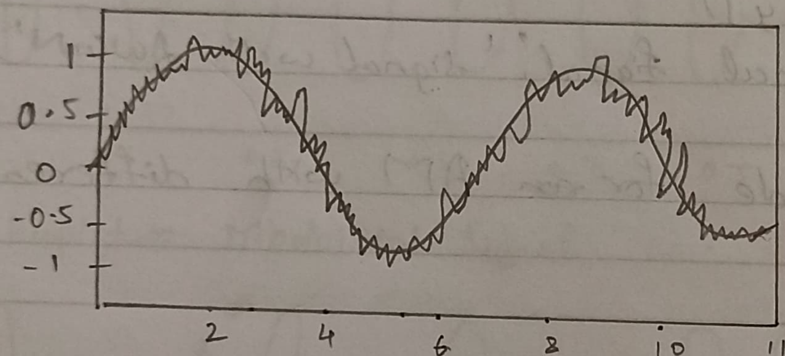
Teacher's Sign. :

AWGN Effect On different functions

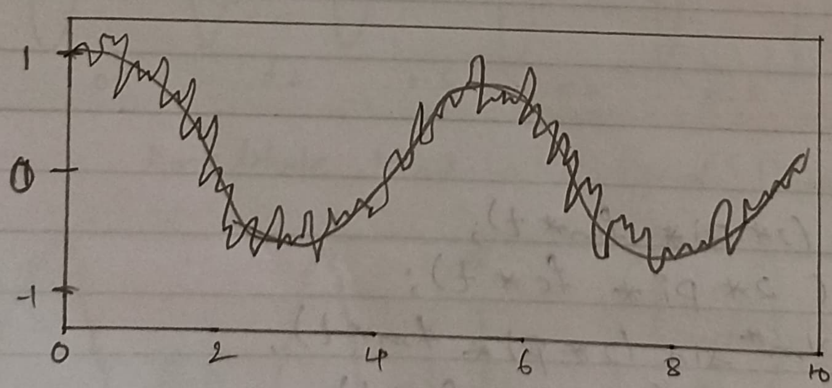
- Saw tooth:



- Sine -



- Cosine -



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```
y = awgn (x, 10, 'measured'); subplot (2,1,1)
plot( t,  x  ]) subplot (2,1,2);
plot (t,y)
```

```
clc;
clear all;
t = (0:0.1:10);
x = cos(t);
y = awgn (x, 10, 'measured'); subplot (2,1,1)
plot (t,  x  ])
legend ('Original Signal', 'signal with AWGN')
plot (t,y)
```

* MATLAB code for AM 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* fc * t);
amp = Vc + Vm* sin (2* pi* fm* t);
am = amp.* sin (2* pi* fc* t);
y1 = awgn( am, 5, 'measured');
y2 = awgn( am, 10, 'measured');
```

Teacher's Sign. :

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```
y3 = awgn (am, 50, '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');
ylabel ('amplitude');
title ('Amplitude modulated signal with AWGN [snr 5]');
subplot (4,1,4);
plot( t, y3);
xlabel ('time');
ylabel ('amplitude');
title ('amplitude modulated signal with AWGN [snr 10]');
```

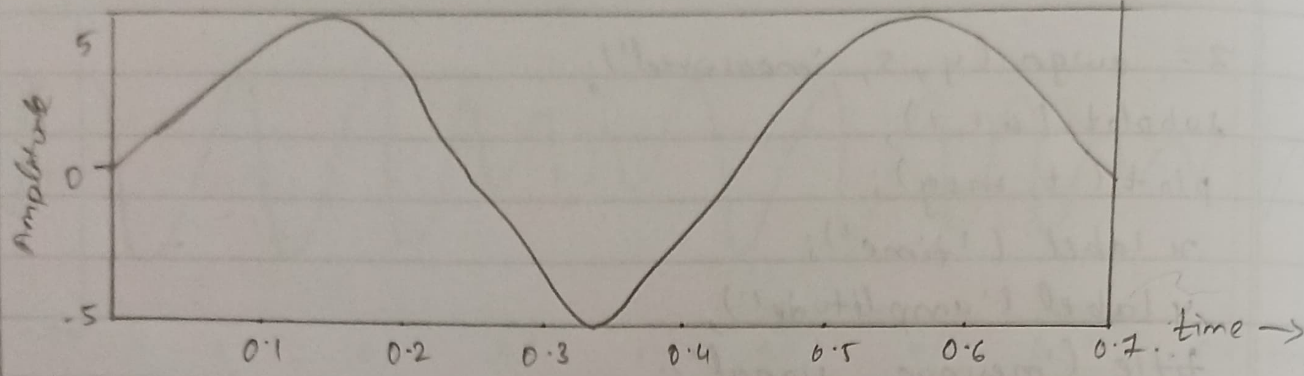
* MATLAB Code for FM signal under AWGN:

```
clc;
clear all;
t = 0:0.001:1;
vm = 5;
vc = 5;
fm = 2;
fc = 25
fd = 5;
msg = vm * sin (2 * pi * fm * t);
c = vc * sin (2 * pi * fc * t);
y = vc * sin (2 * pi * fc * t, fd), cos(2 * pi * fm * t));
```

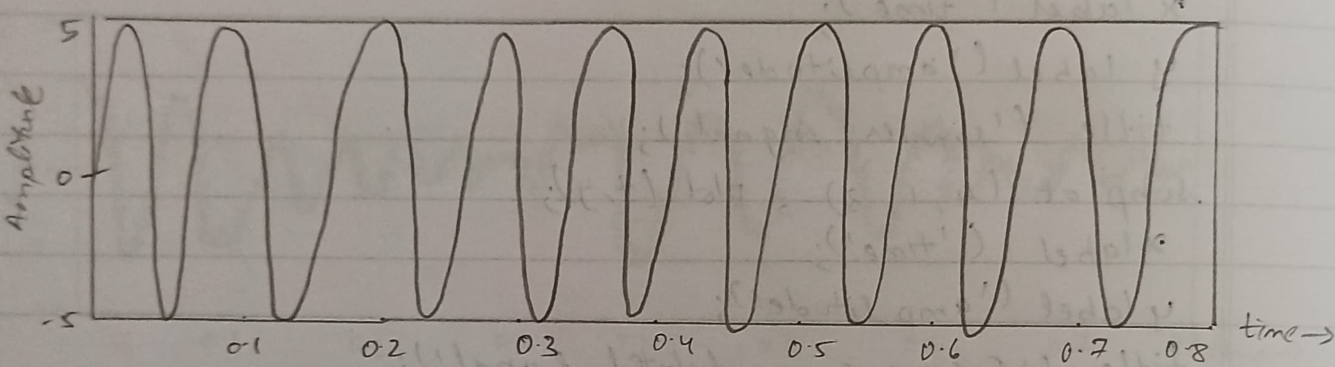
Teacher's Sign. :

FM under AWGN

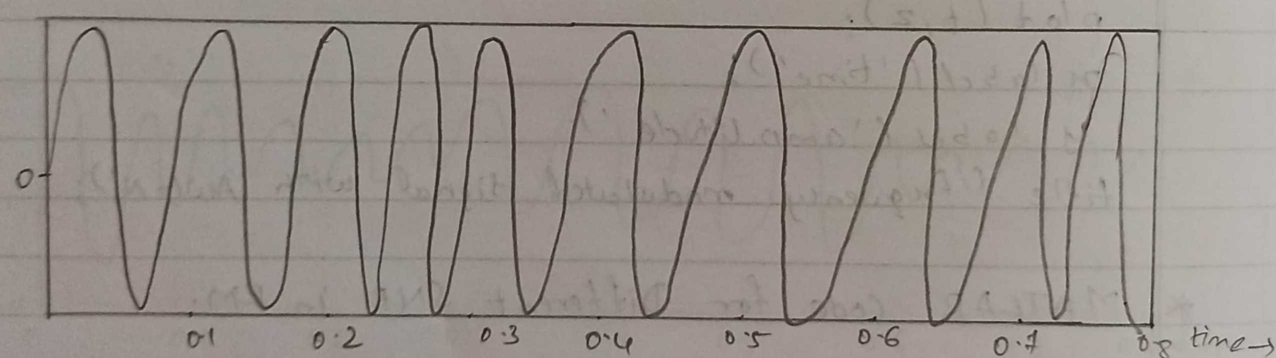
Message signal



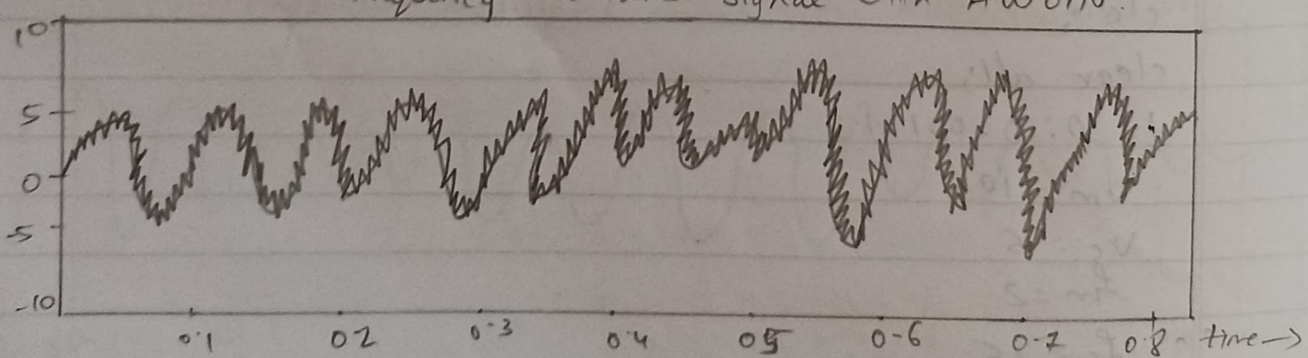
Carrier Signal



Frequency Modulated Signal.



Frequency Modulated Signal with AWGN.



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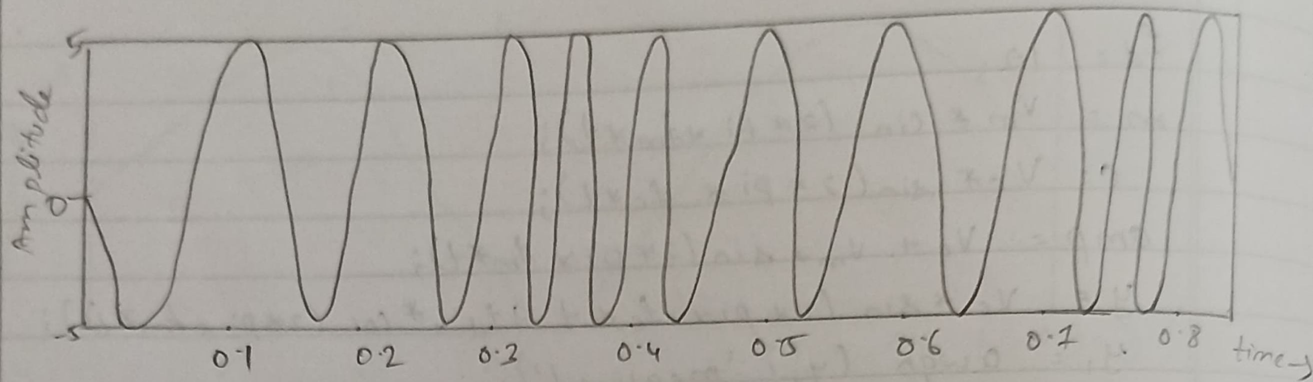
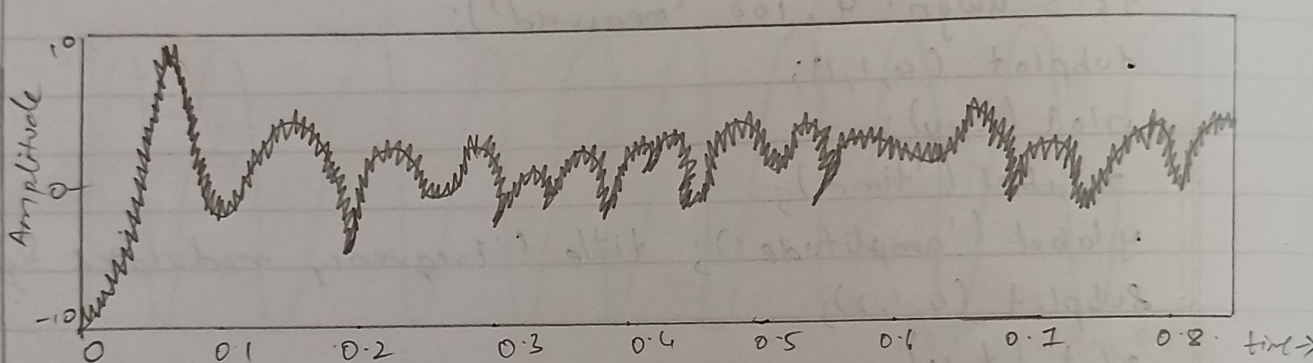
```
fd = 10;
m = vm * sin (2* pi *fm*t);
c = vc* sin(2 * pi * fc*t);
amp = vc + vm * sin (2*pi* fm*t);
y = vc * sin (2 * pi* fc * t + fd * cos (2*pi* fm*t));
y1 = awgn (y, 1, 'measured');
y2 = awgn( y, 10, 'measured');
y3 = awgn( y, 100, 'measured');
subplot (4,1,1);
plot (t, y);
xlabel ('time');
ylabel ('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]');
```

AWGN for different SNR in FM

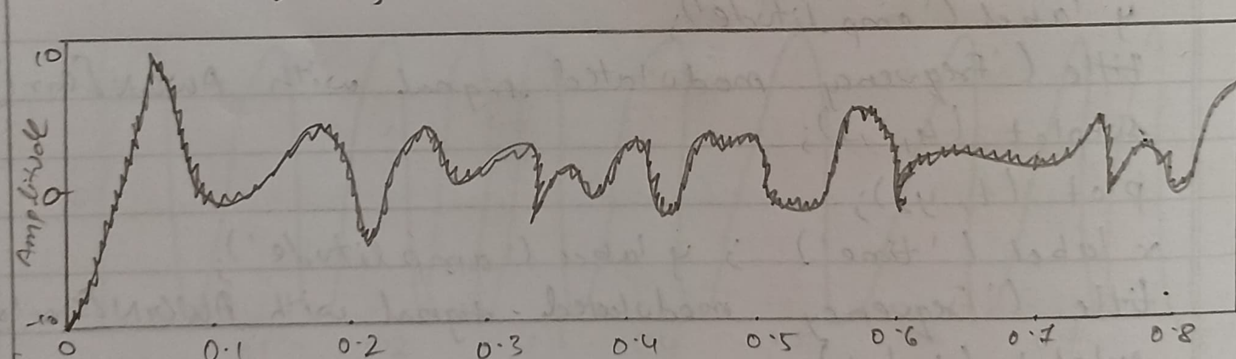
Frequency Modulated signal



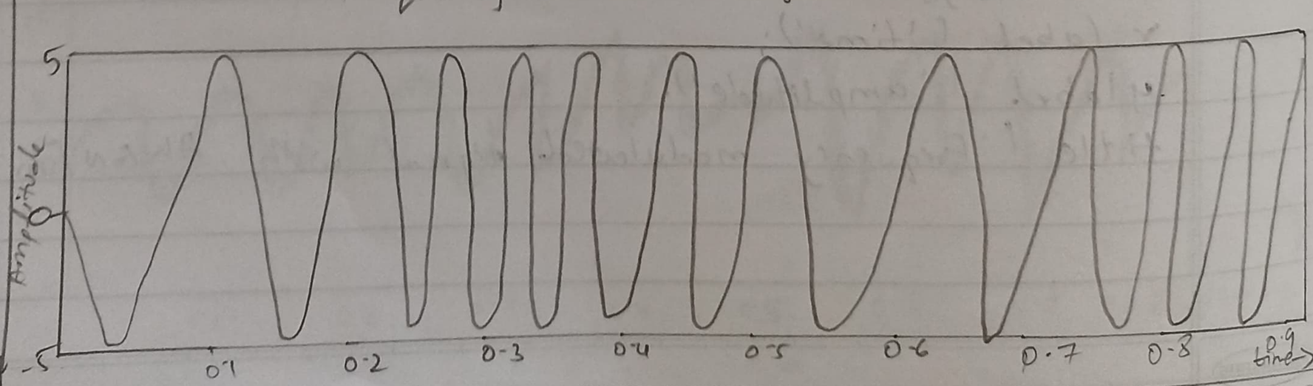
Frequency modulated signal with AWGN [Snr 1]



Frequency modulated signal with AWGN [Snr 10]



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 while Gaussian noise to
                    the input with S/N = 5
plot (x, 'g', 'line width', 1.5);
hold on;
plot(z)
hold on;
for i=1: 194;
y(i) = (z(i) + z(i+1) + z(i+2) + z(i+3) + z(i+4)+ z(i+6))
                                                        /6
end
plot (y, 'r', 'linewidth', 1.5);
legend ('Actual', 'Noisy', 'filtered');
title ('moving Average filter', ' font size', 12);
x label ('--->time in 2us');
y label ('--> Volts');
```

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* Advantages of AWGN :

- The noise is additive i.e the received signal is equal to the transmitted signal plus noise. This gives the most widely used equality in communication systems.
- The result of summing a large number of different & independent factors which allows us to apply an important result from probability and statistics, called the central limit theorem.

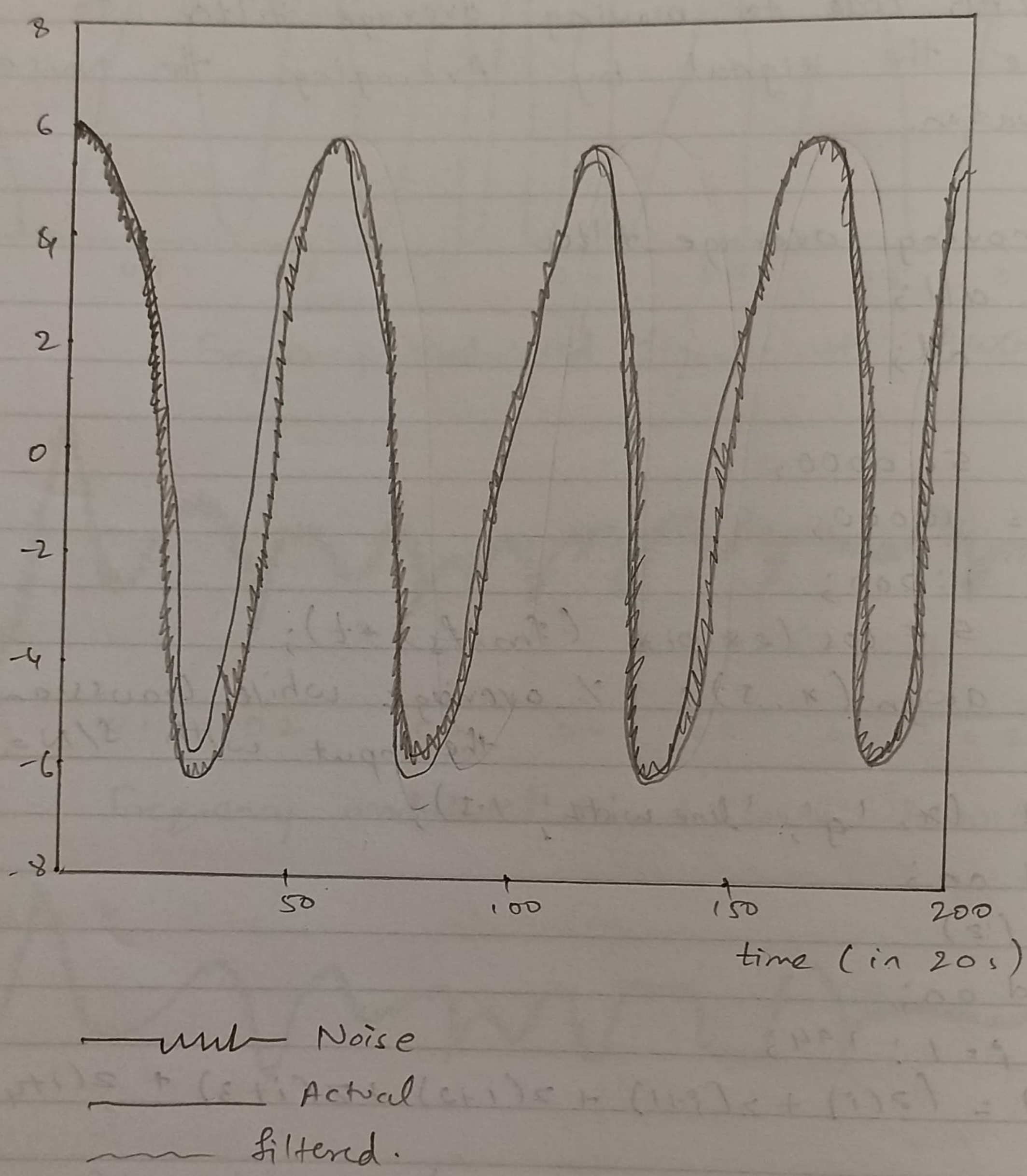
* Applications -

It is often used as a channel model in which the only implement to communication is a linear addition of wideband or white noise with a constant spectral density (expressed as watts per hertz of bandwidth) and a gaussian distribution of amplitude.

* Conclussion -

Successfully observed the AM and FM under AWGN using MATLAB software and drew the waveforms for different signal to noise values.

Moving Average Filter.



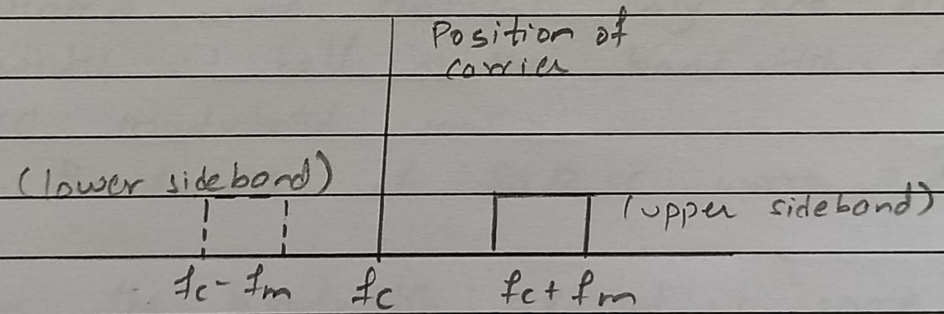
EXPERIMENT : No. 8 Single Side Band (SSB-SC) Modulating Scheme.

* Objective:- Write and simulate a program for single side-band (SSB) modulating 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.

* Software :- MATLAB

* Theory -

The process of suppressing one of the sidebands along with the carrier and transmitting a single side band is called as single side band suppressed carrier system or simple SSBCC



Here, the carrier and the lower sideband are suppressed. Hence, the upper sideband is used for transmission. Similarly, we can suppress the carrier and the upper sideband with the transmission of the lower sideband. This is because in SSBSC both upper side band and lower side band have the same information.

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* Mathematical Expression -

Let Modulating signal → $m(t) = A_m \cos 2\pi f_m t$.
Let Carrier signal signal → $c(t) = A_c \cos 2\pi f_c t$.

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 side band.

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

* Bandwidth of SSBSC :-

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

∴ Bandwidth of SSBSC wave $= f_m$.

Therefore, the bandwidth required is same as the required for the modulating signal.

* Power Calculations -

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

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

Teacher's Sign. :

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Power of SSBSC is equal to the power of any one side-band frequency components:

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

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

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

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

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

Therefore, the power required is these than required for DSBSC wave.

- Generation of SSBSC

There are two methods for the generation of SSBSC:
1) Frequency discrimation method.
2) Hilbert transform method or phase discrimation method.

1) Frequency Discrimation 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.

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The band pass filter produces output, which is SSBSC wave.

Select the frequency range of band pass ~~filt~~ filter as the spectrum of desired SSBSC wave. This means the band pass filter can be tuned 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 product modulators, two 90° phase shifters.
The modulating signal $A_m \cos(2\pi f_m t)$ and carrier signal $A_c \cos(2\pi f_c t)$ are applied to product modulator. The output -

$$s_1(t) = A_m A_c \cos(2\pi f_m t) \cos(2\pi f_c t)$$

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

Adding $s_1(t)$ and $s_2(t)$,

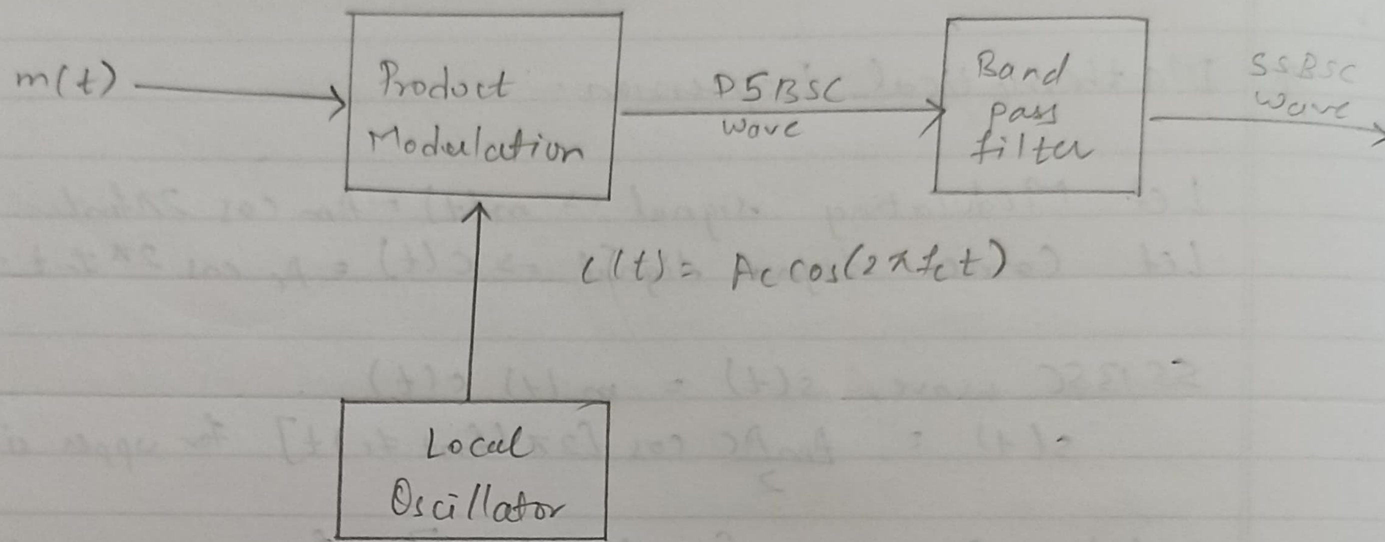
$$s(t) = A_m A_c \cos[2\pi(f_c - f_m)t] \Rightarrow \text{Lower side band}$$

Subtract $s_2(t)$ from $s_1(t)$,

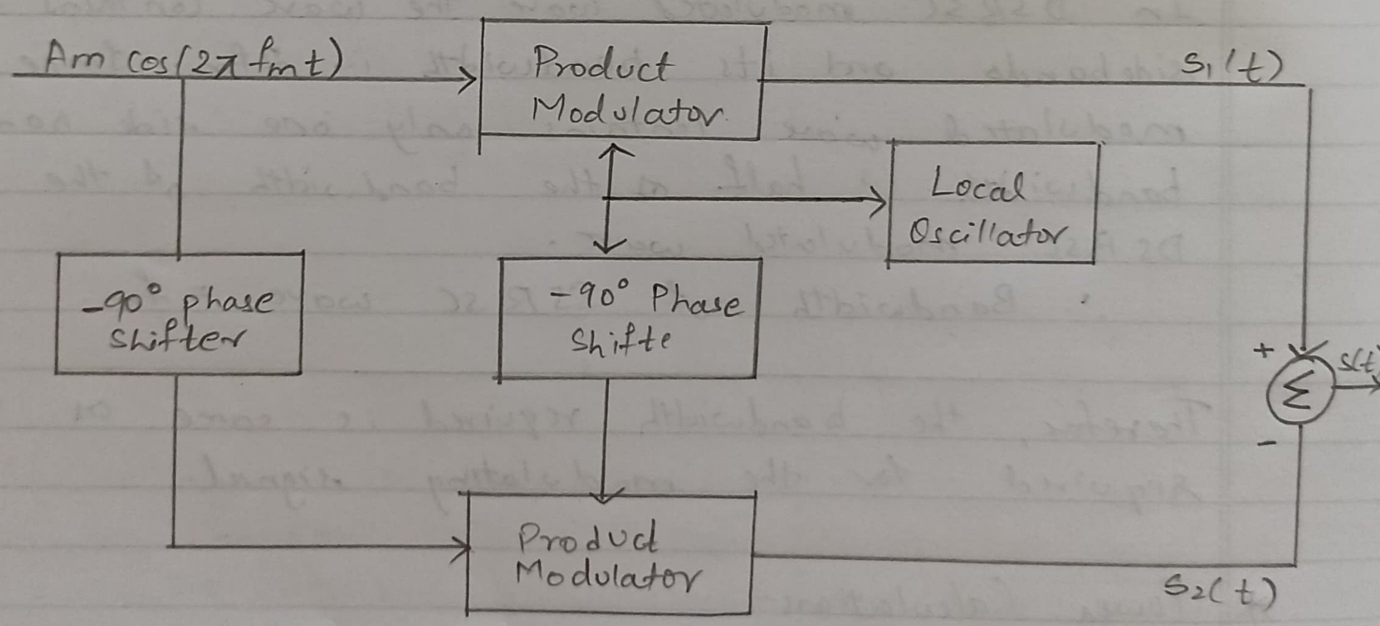
$$s(t) = A_m A_c \cos[2\pi(f_c + f_m)t] \rightarrow \text{Upper side band.}$$

Therefore, choosing correct polarities of input of summer block will get SSBSC having upper side band or lower sideband.

Frequency Determination Method :-



Phase Discrimation method :-



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* MATLAB code-

```
clc;
clear all;
close all;
am = 1;          % amplitude of modulating signal
ac = 1;          % amplitude of carrier signal
fm = 500;        % modulating signal frequency
fc = 5000;       % carrier frequency
fs = 100000;     % sampling frequency
ts = 1/fs;       % sampling Interval
N = 1000;        % No of samples.
t = (-N/2:1:(N/2 -1)) * ts;      % time interval
m = am * cos (2*pi* fm*t);   % modulating signal
mh = am* sin (2*pi* fm*t);   % hilbert transformation message signal
c = ac* cos (2*pi* fc*t);    % carrier signal
ch = ac * sin (2*pi* fc *t);   % hilbert tranformation carrier signal
st = m * c mh.*ch;   % SSBSC 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;
```

Sundaram Teacher's Sign. :

```
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 0.005  -2.5  2.5]);
xlabel ('time');
ylabel ('amplitude');
title ('modulated signal');
grid on;

% spectrum of all signals
f = (-N/2 : 1 : (N/2 - 1)) * fs/N;
M = abs ((2/N)* fftshift (fft(m)));
C = abs ((2/N) * fftshift(fft(st)));
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;
```

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```
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;

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;
```

* Observations:

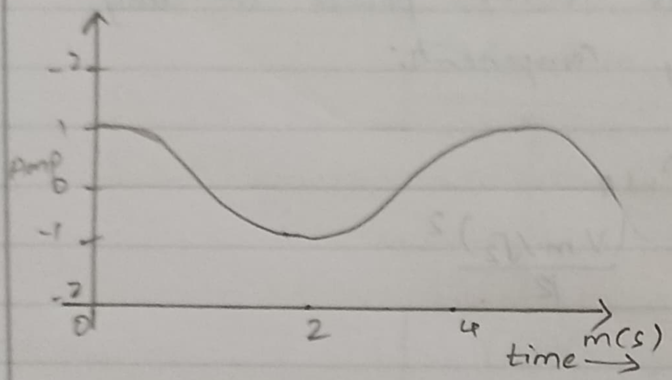
Sampling freq = 100kHz   No. of samples = 10000
Amplitude of carrier = 1   Amplitude of modulating signal = 1

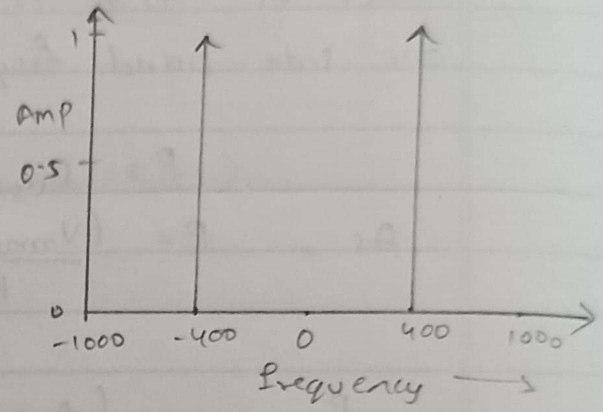
Modulating freq (f_m)	Carrier freq (f_c)	SSBSC freq
400	2000	2400
200	3000	3200
300	4000	4300
500	2000	2500
500	5000	5500

Teacher's Sign. :

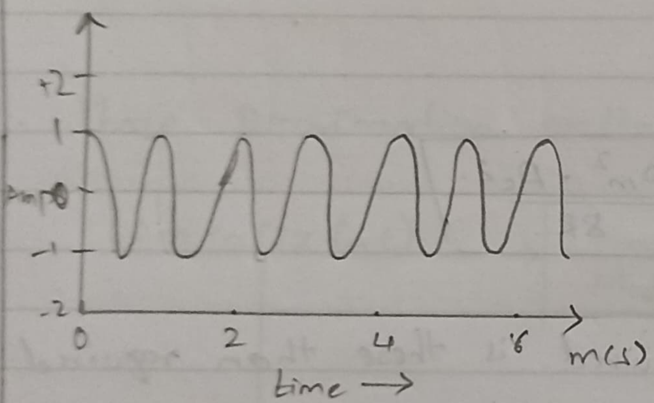
1) $f_m = 400$, $f_c = 2000$

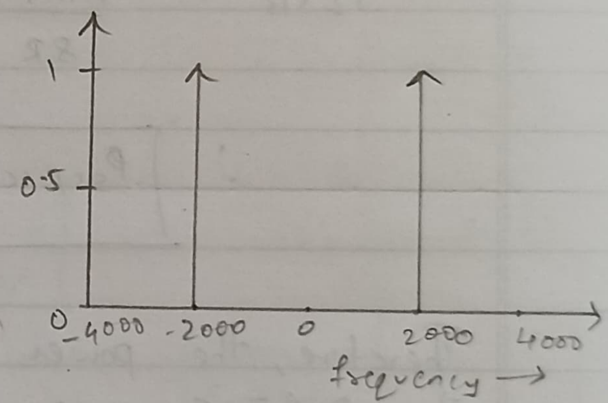
Modulating Signal



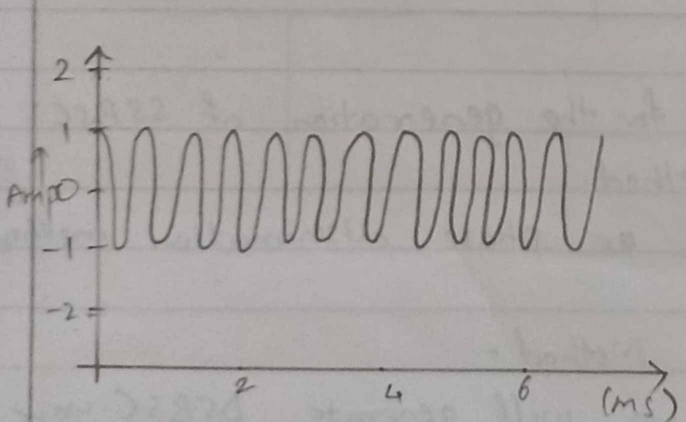


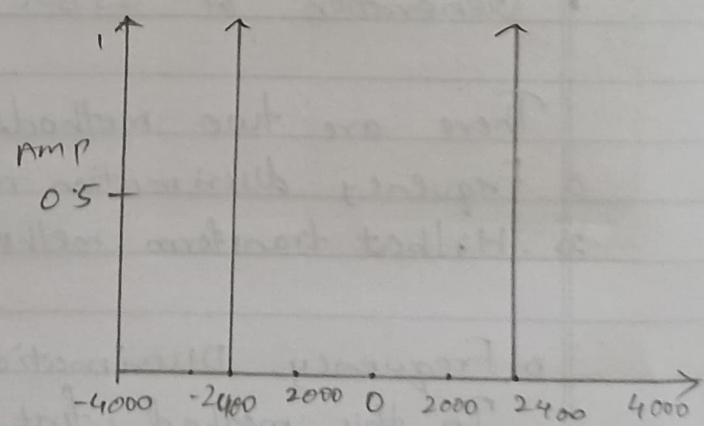
Carrier Signal





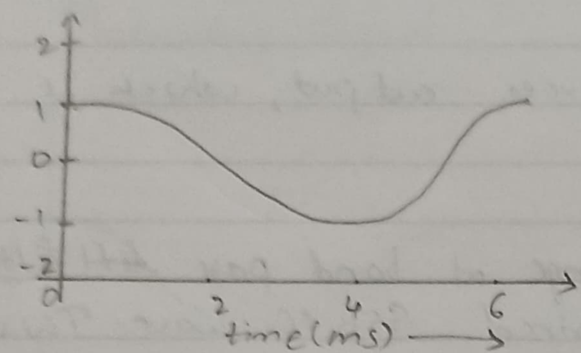
Modulated Signal

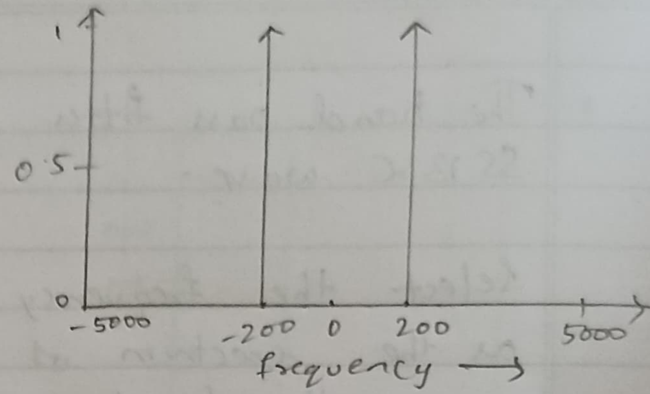




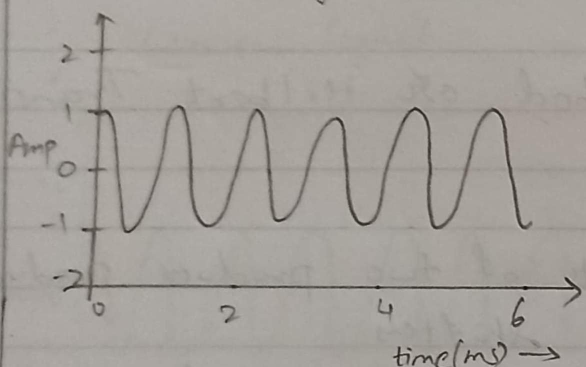
2) $f_m = 200, \; f_c = 3000$

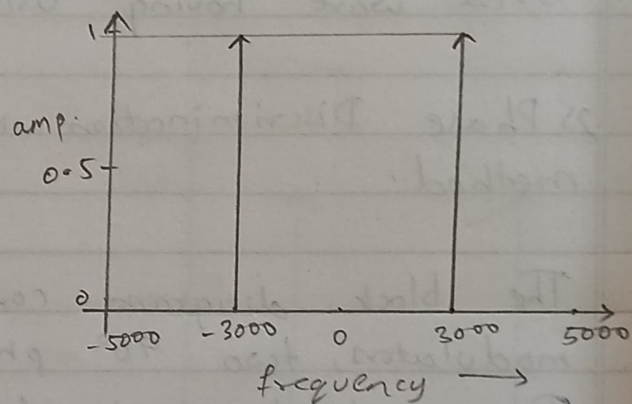
Modulating signal



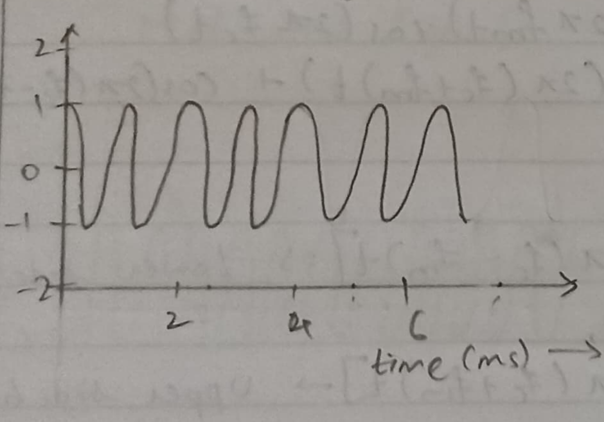


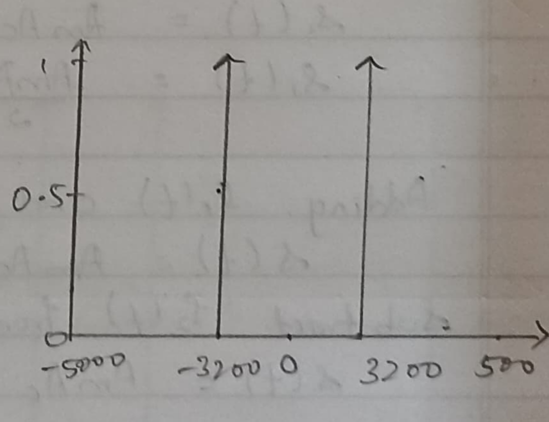
Carrier signal





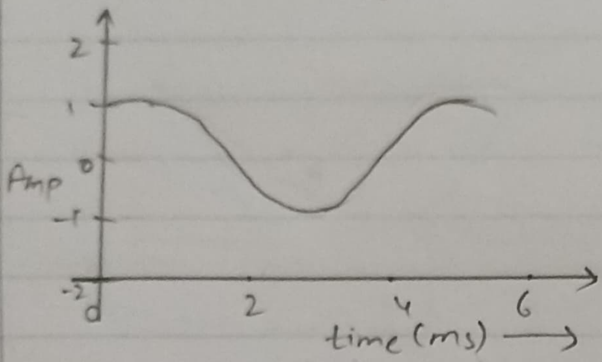
Modulated signal

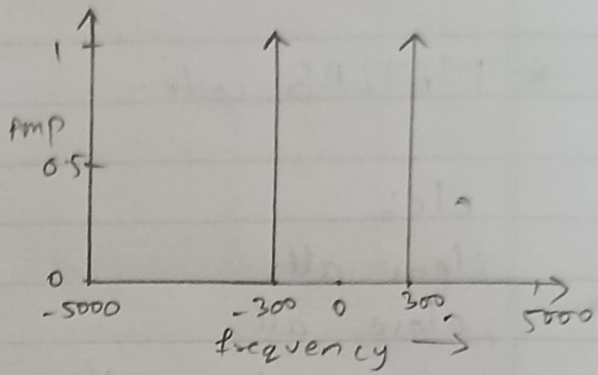




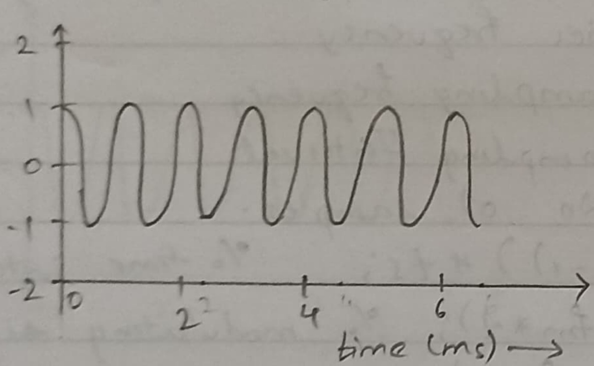
3) $f_m = 300$ $f_c = 4000$

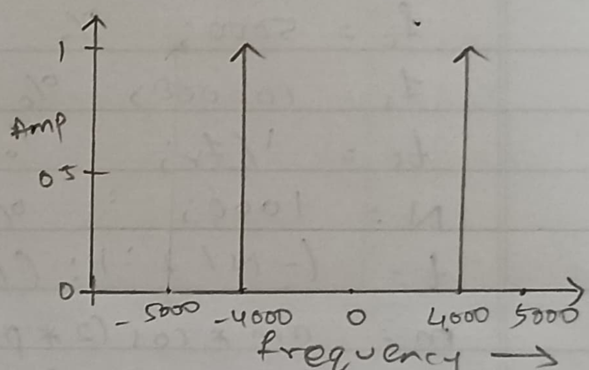
Modulating Signal



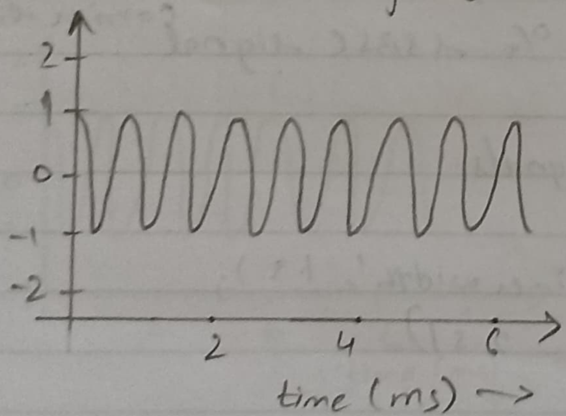


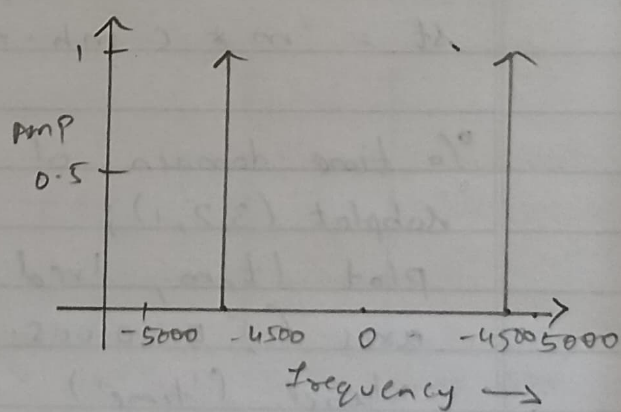
Carria signal.





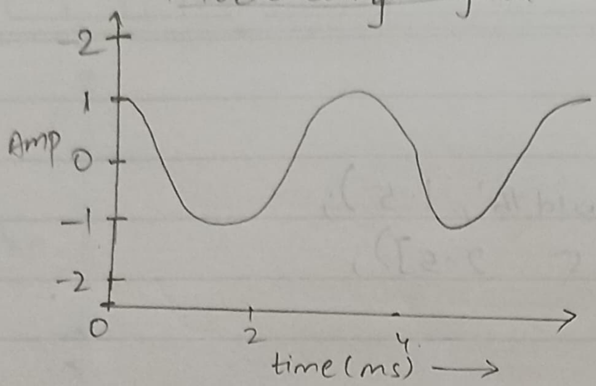
Modulated Signal

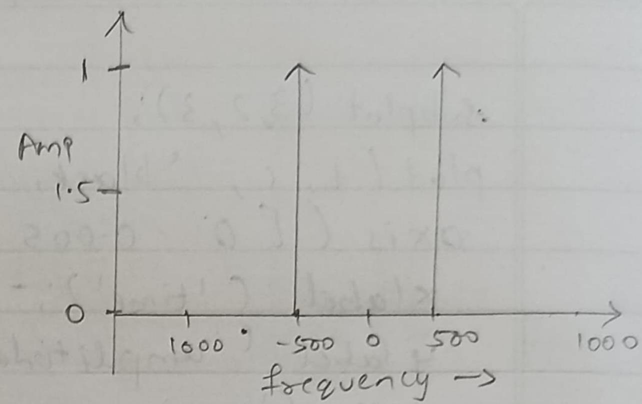




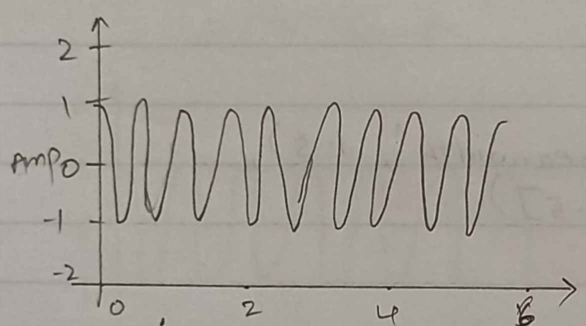
4) $f_m = 500$, $f_c = 2000$

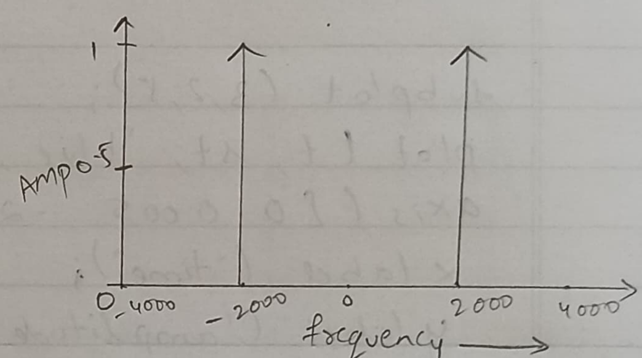
Modulating Signal



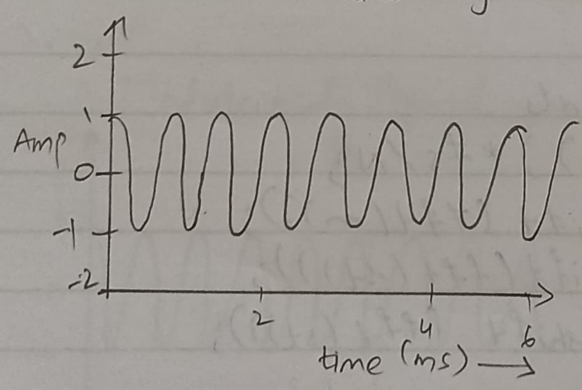


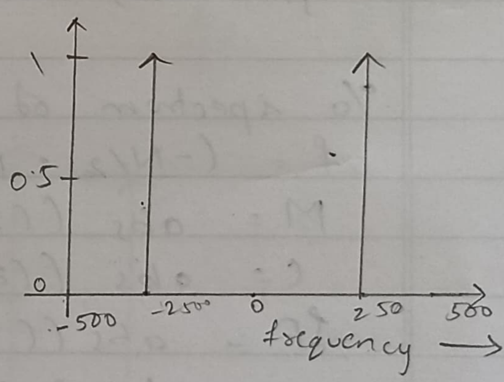
Carrier Signal





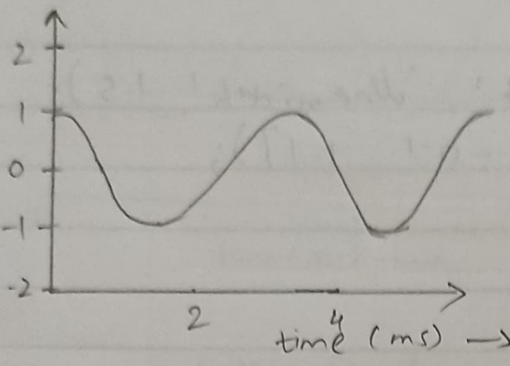
Modulated Signal

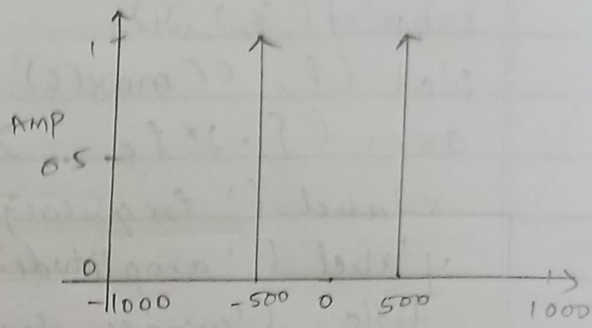




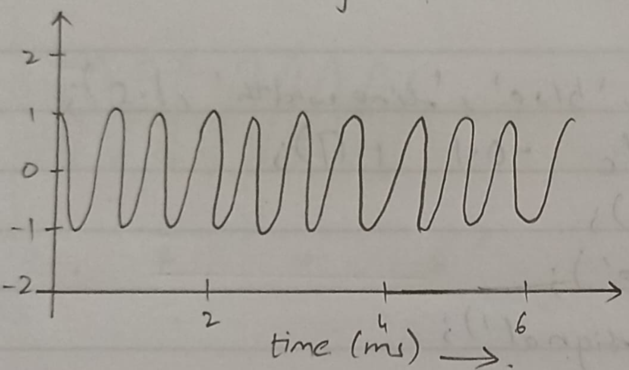
5) $f_m = 500$, $f_c = 5000$

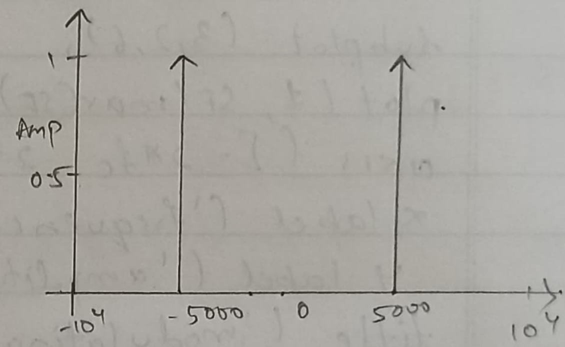
Modulating Signal



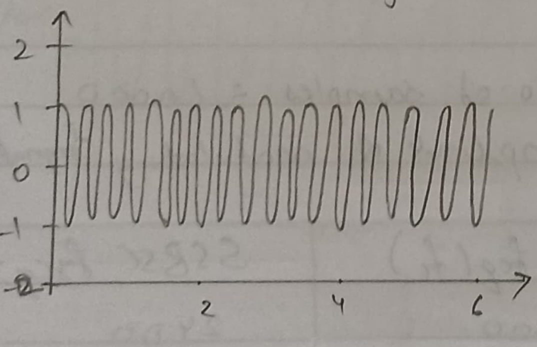


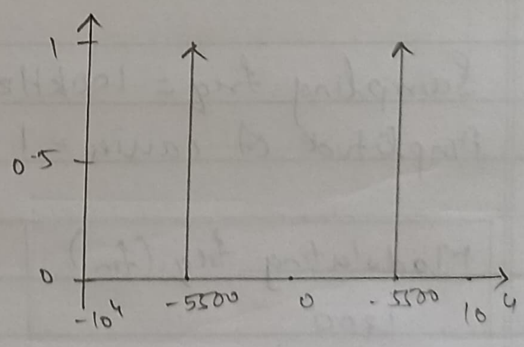
Carrier Signal





Modulated Signal





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* Advantages

1) Bandwidth or spectrum space occupied is less than AM & DSB signal.
2) Power is conserved and high power signal can be transmitted
3) Less amount of noise is present.
4) Signal fading is less likely to occur.

* Disadvantages

1) The generation & detection of SSB is a complex process
2) Quality of the signal gets affected unless the SSB transmitter & receiver have an excellent frequency stability

* Applications:-

1) For power saving requirements & low bandwidth requirements
2) In land, air and maritime mobile communication.
3) In point-to-point communications.
4) In radio communications.
5) In television, telemetry and radar communication.
6) In military communications, such a amateur radio.

* Conclusion-

Successfully observed and simulated SSB scheme and implemented the waveforms of message / carrier and resultant modulated signal in the time and frequency domain using MATLAB Software.

EXPERIMENT : No. 9 Pulse Code Modulation and Demodulation.

Page No. Date

* Objective - 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.

* Software :- MATLAB

* Theory :-

PCM (Pulse Code modulation) is a technique that is used to convert all analog signal to digital signal. In this the amplitude of an analogue signal is converted to a binary value represented as a series of pulses. PCM is a preferred method of communication within the public switch telephon network (PSTN)

PCM is determined by two following steps -

- Sampling rate: Which is the number of times per second that samples are taken.

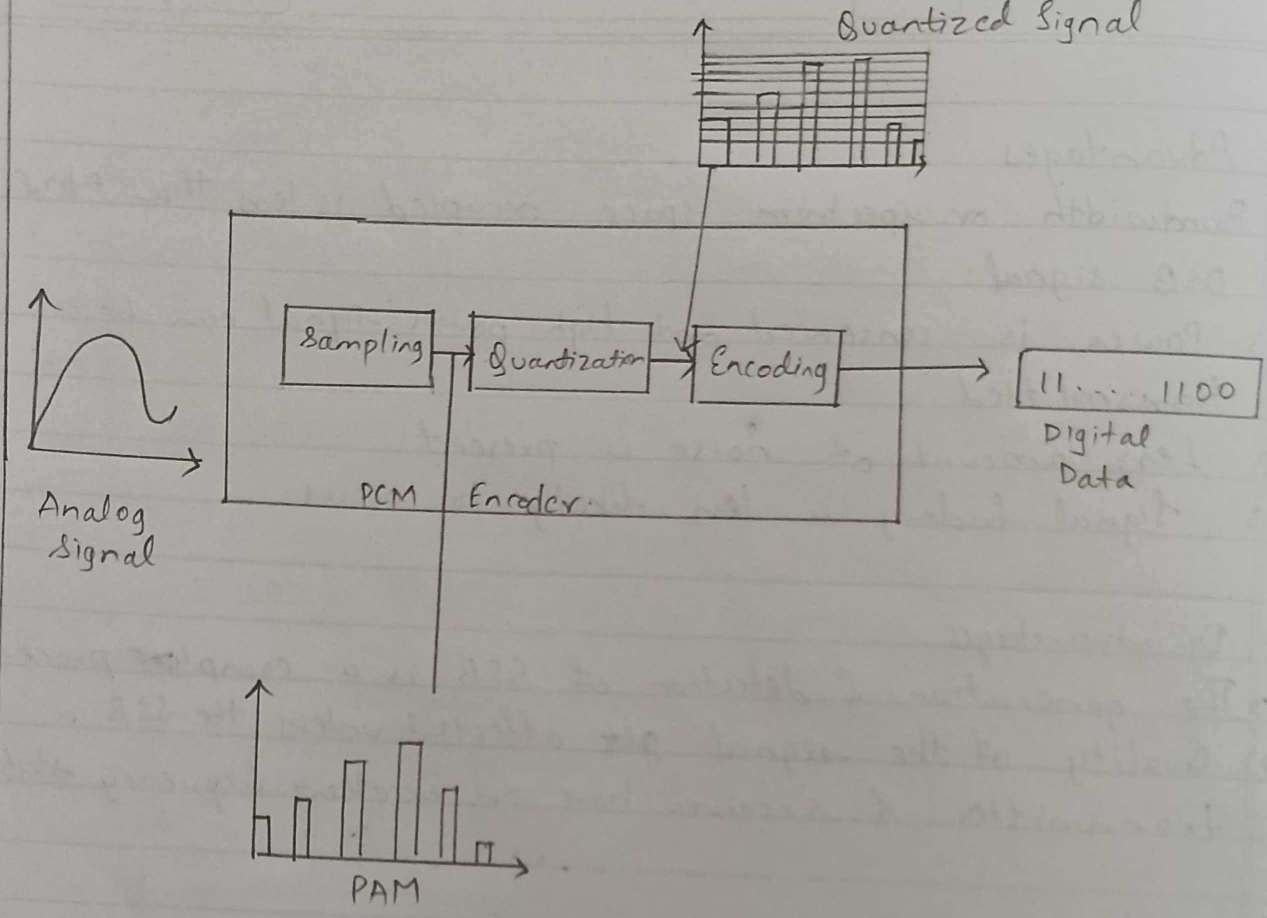
- Bit Depth: Which determines the number of possible digital values that can be used to represent each sample.

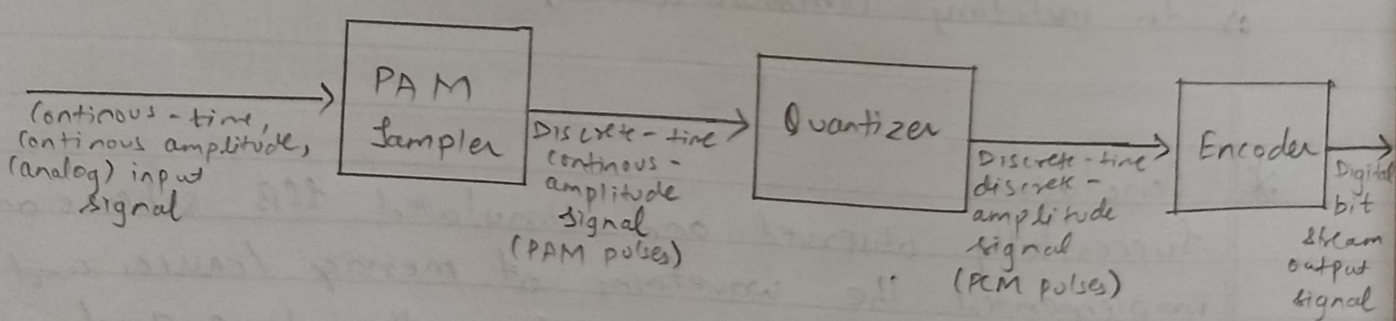
Hence PCM resumbles output of binary sequence.

Sundaram

Teacher's Sign. :

BLOCK DIAGRAM

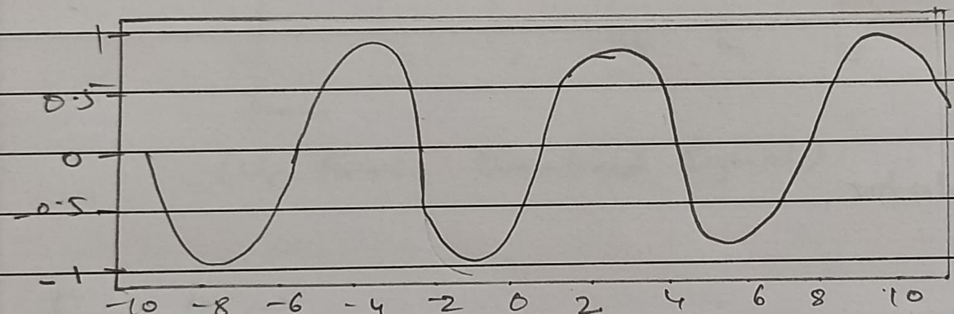




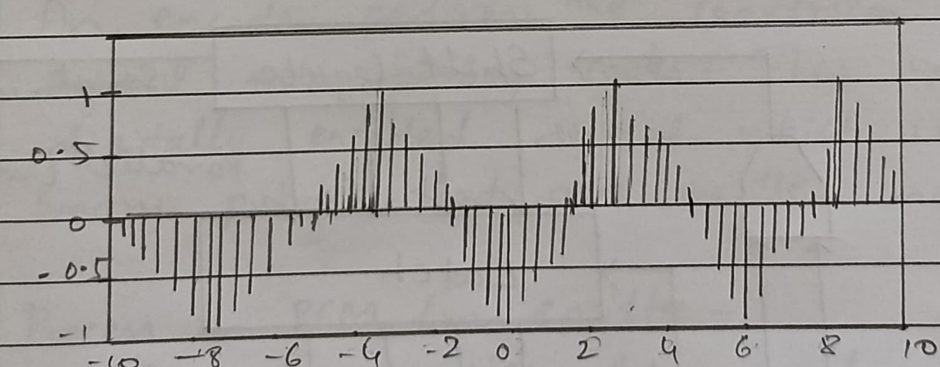
The transmitter section of a pulse code Modulator circuit consists of sampling, Quantizing and encoding

Sampling -

- The sampler extracts samples of a continous signal.
- Sampler produces samples that are equivalent to the instantaneous value of the continous signal at the specified various points.
- The sampling process generates flat-top pulse Amplitude Modulated (PAM) signal.



Analog signal

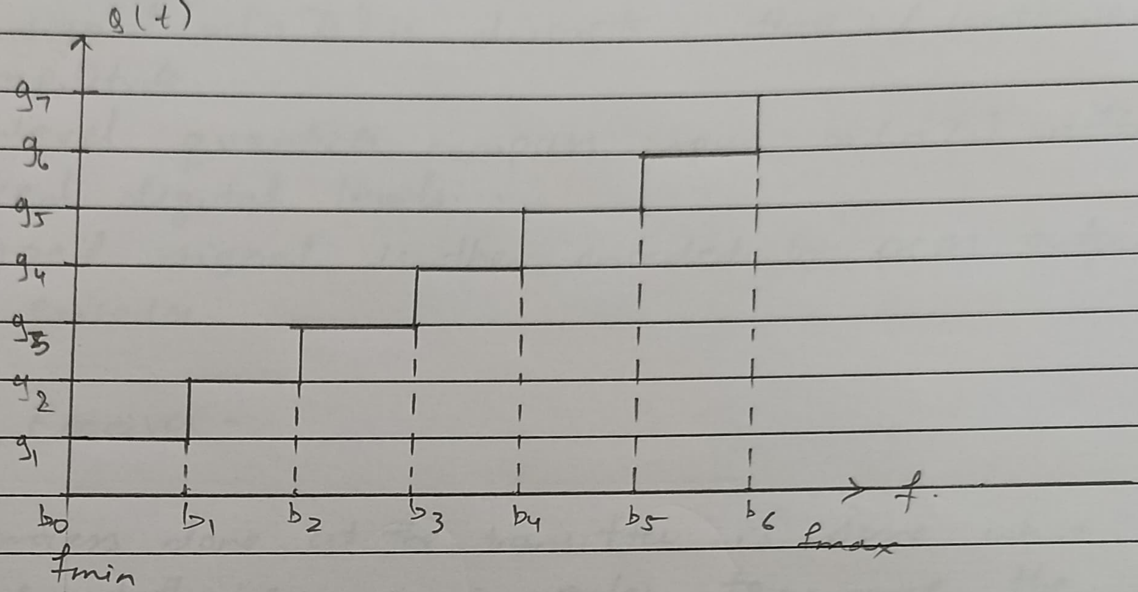


Discrete time signal

* 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 generats a digitally encoded signal which is a sequence of binary pulses that acts as the modulated output.

Process in PCM transmittee -

-> In PCM transmittee, 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 than samples this signal at the rate of f_s. sampling freq f_s is selected sufficiently above Nyquist rate to avoid aliasing.

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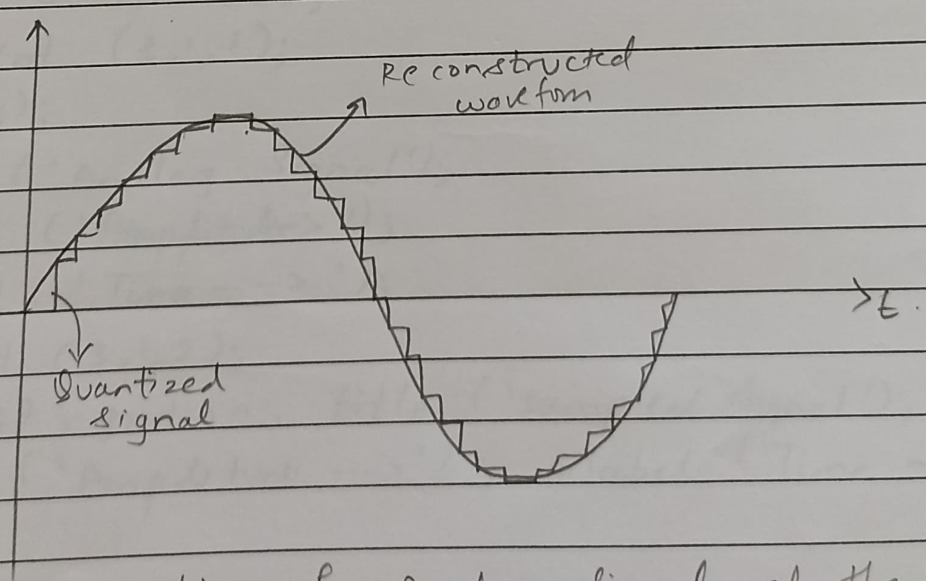
The output from the sample and hold circuit is denoted by $x(nT_s)$.

→ 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 levels.
- Quantized signal is then encoded in PCM output using encoder.

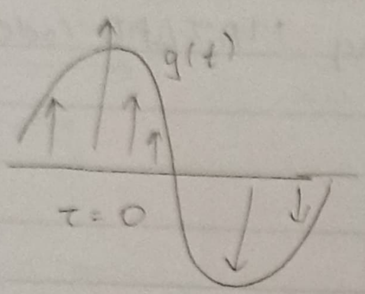
* PCM Receiver -

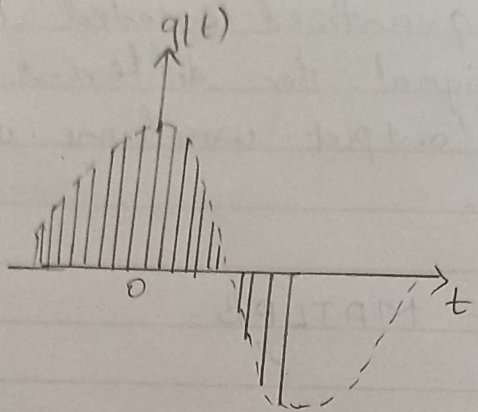
The process done at the transmitter is some what reversed at the receiver in order to generate the original analogue message. The figure below shows the reconstruction of the analog message signal at the reciever.

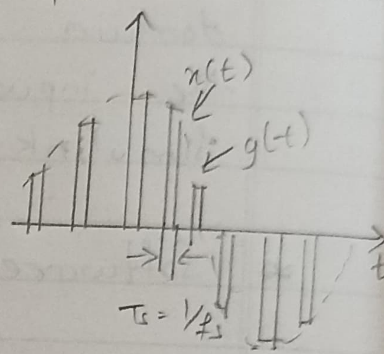


(Reconstruction of Analog signal at the Receiver)

Instantaneous Sampling	Natural Sampling	Flat - Top Sampling
Pulse width = τ $\tau = 0$		Pulse width = τ

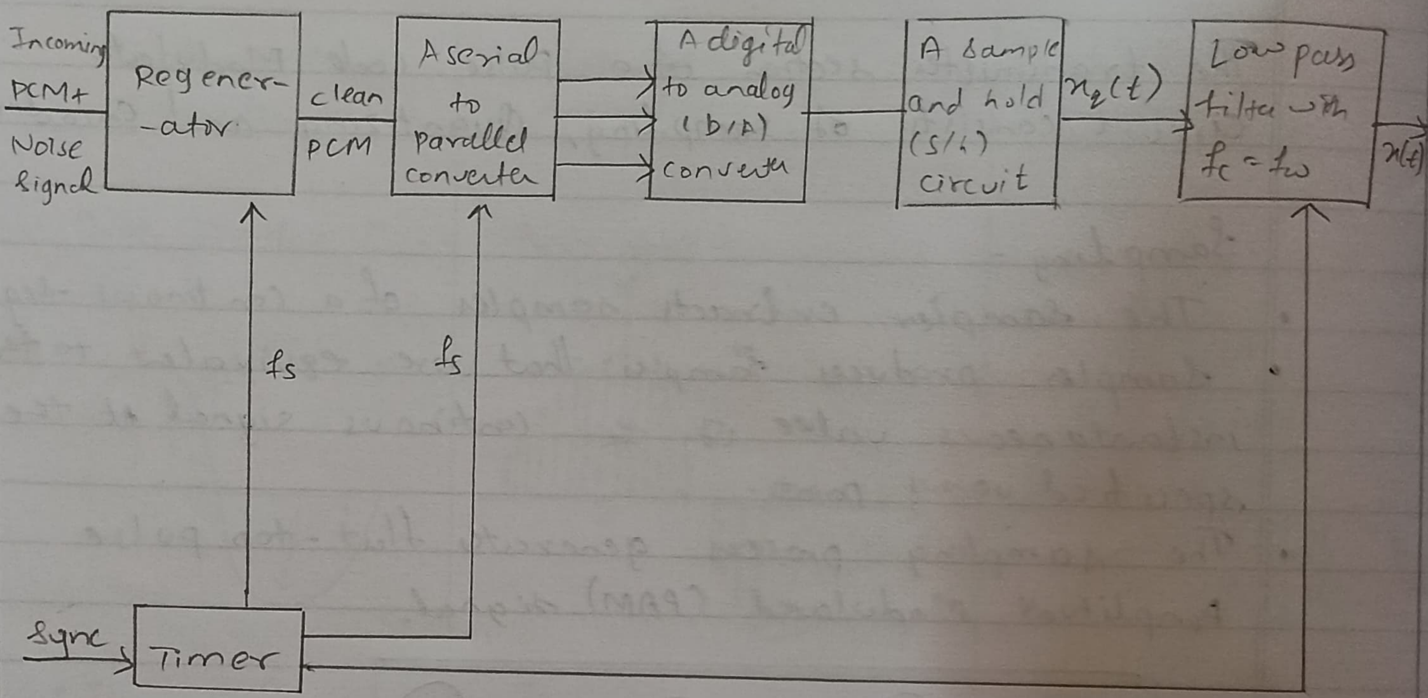




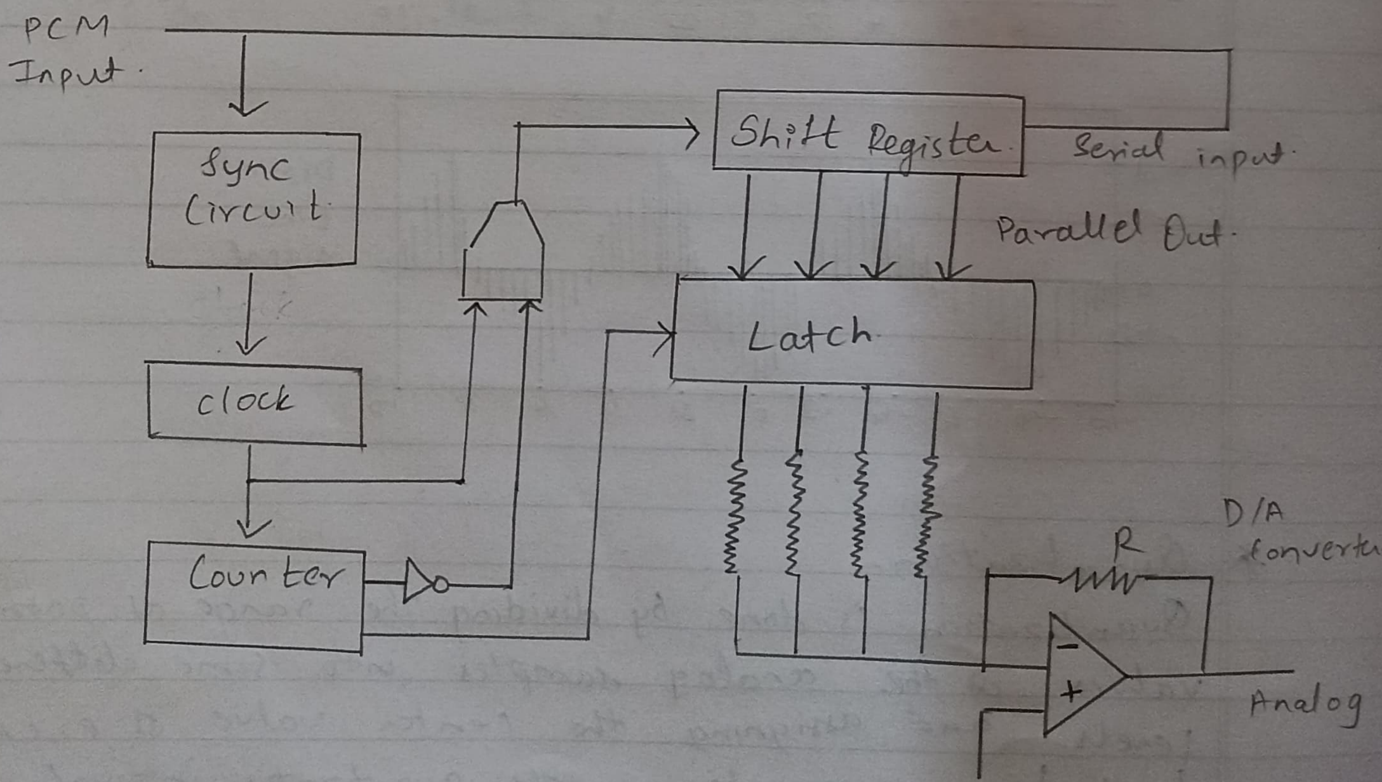


Instantaneous Sampling	Natural Sampling	Flat - Top Sampling
• It is not a practical method	• This method is used practically	• This method is also used practically
• Sample rate is infinity	• Sample rate satisfies Nyquit criteria.	• Sample rate satisfies Nyquist criteria.

PCM Receiver Block Diagram.



PCM Demodulator



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* 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);
% 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;
```

Teacher's Sign. :

EXPERIMENT : No. Page No. Date

```
Vmin = -Vmax
del = (Vmax - Vmin)/L; % Level all between Vmin & Vmax with
                       % difference of del
part = Vmin : del : Vmax;
code = Vmin - (del/2) : del : Vmax + (del/2); % Contain quantized value
[ind,q] = quantiz (s, part, code); % Quantization process

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

for  i = 1:11   % To make index as binary decimal so
                  started from 0 to N
    if (ind(i) ~= 0)
            ind (i) = ind (i)-1;
    end
    i = i+1;
end
    for  i = 1:12
        if (q(i) == Vmin -(del/2))
                q(i) = Vmin + (del/2));
        end
    end
subplot (3,1,3); % Display the Quantize values.
stem (q); grid on;
title ('Quantized Signal');
ylabel ('Amplitude -->');
Xlabel ('Time -->');

% Encoding process
figure
```

Teacher's Sign. :

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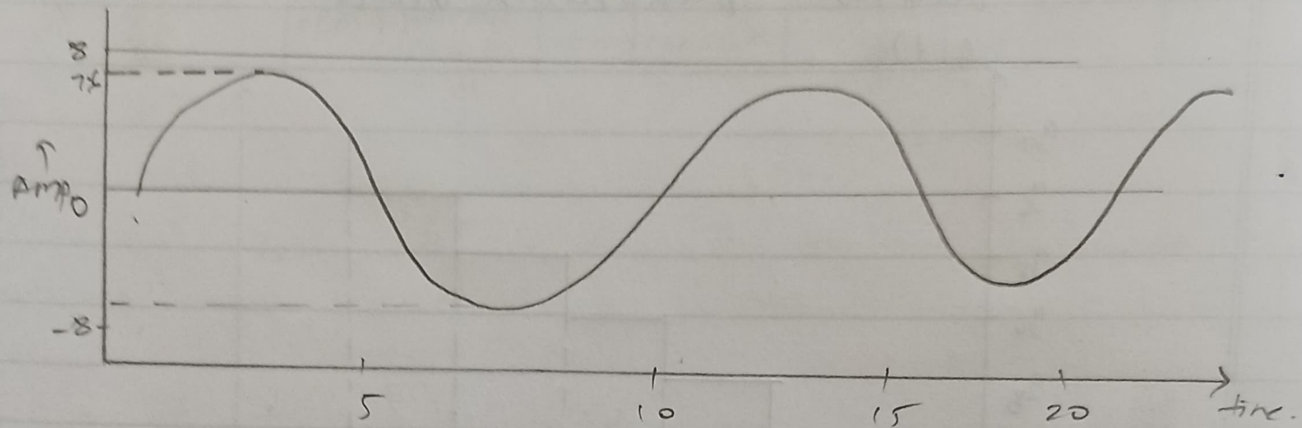
```
code= dezbi (ind, 'left-msb'); % Convert decimal to binary
k=1;
for i= 1:11
  for j= 1:n
    coded (k)= code(i,j) % convert code matrix to a
                           coded row vector.
    j=j+1;
    k= k+1;
  end
  i= i+1;
end
subplot (2,1,1); grid on;
stairs (coded);     % Display the encoded signal
axis ([0 100 -2 3]); title ('Encoded signal');
ylabel ('Amplitude -->');
xlabel ('Time -->');

% Demodulation of PCM Signal.
qunt = reshape (coded, n, length (coded)/n);
index = bi2de (qunt', 'left-msb'); % getback the
                                     index in decimal form
q= del*index + vmin+ (del/2); % Getback quantized value
subplot (2,1,2); grid on
plot(q);              % plot demodulated signal
ylabel ('Amplitude -->');
xlabel ('Time -->');
```

Teacher's Sign. :

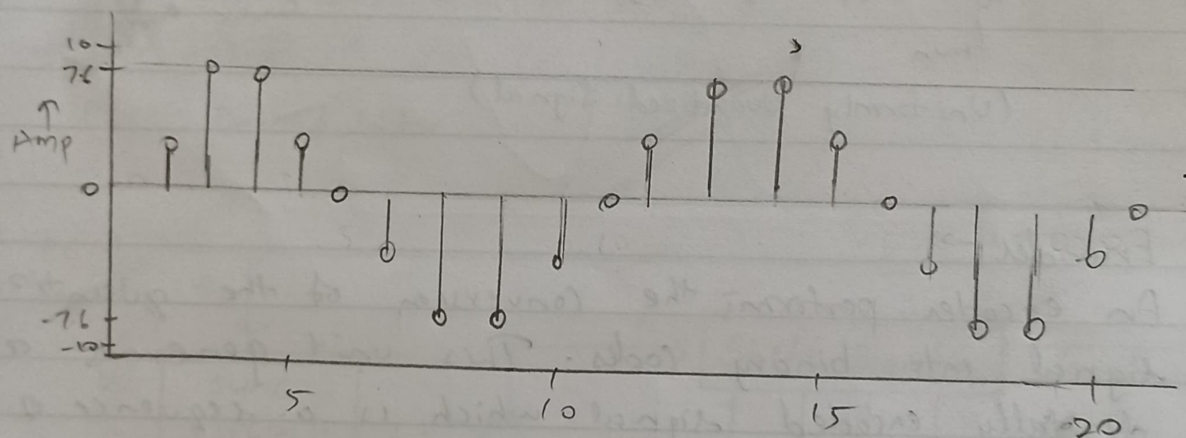
1) 5-bit system

No of samples in a period = 10

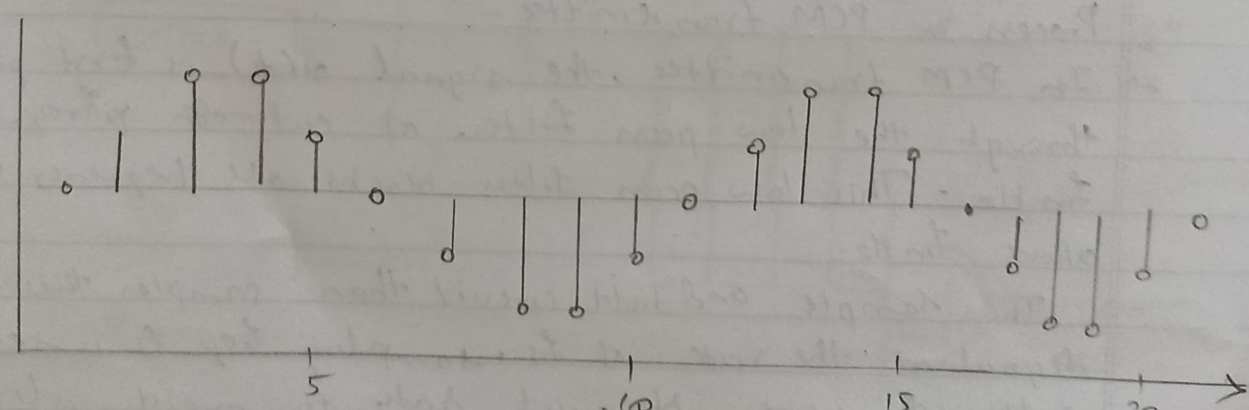
Analog signal.



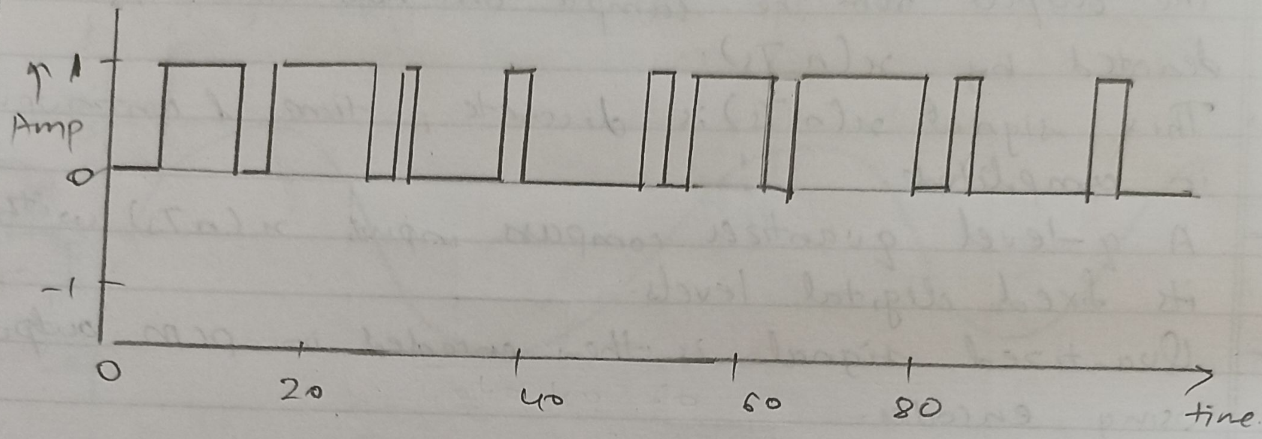
Sampled Signal



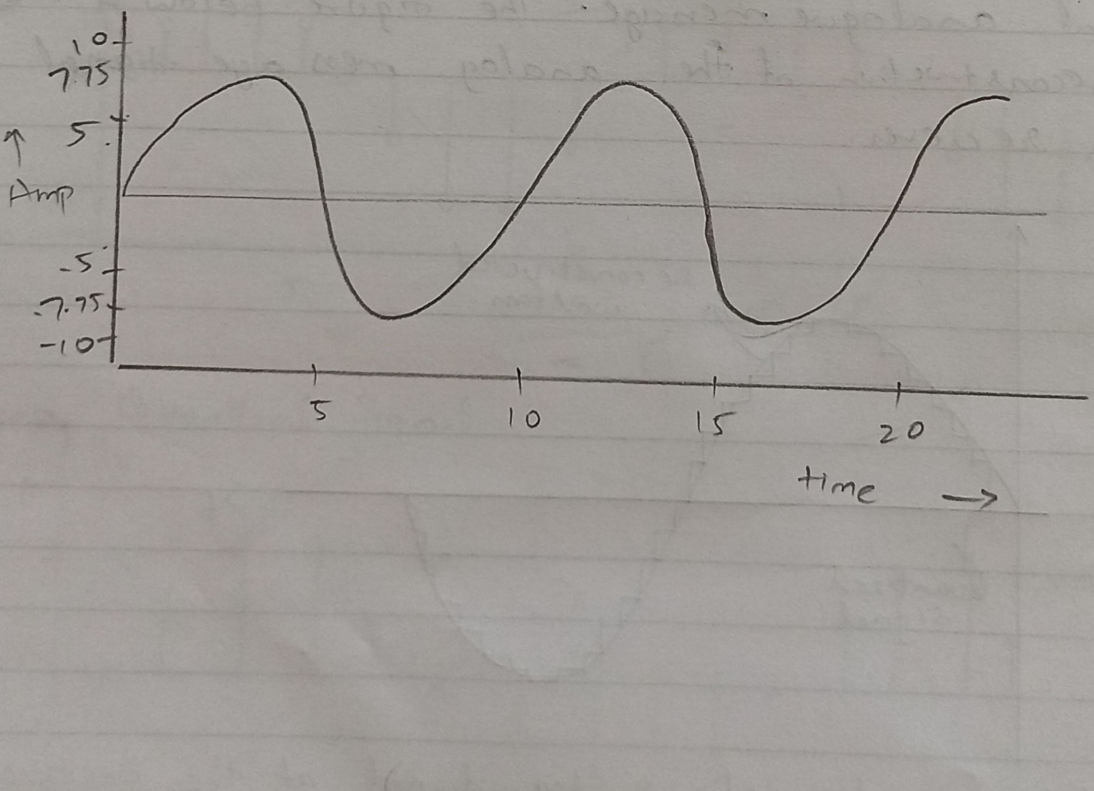
Quantized Signal.



Encoded signal



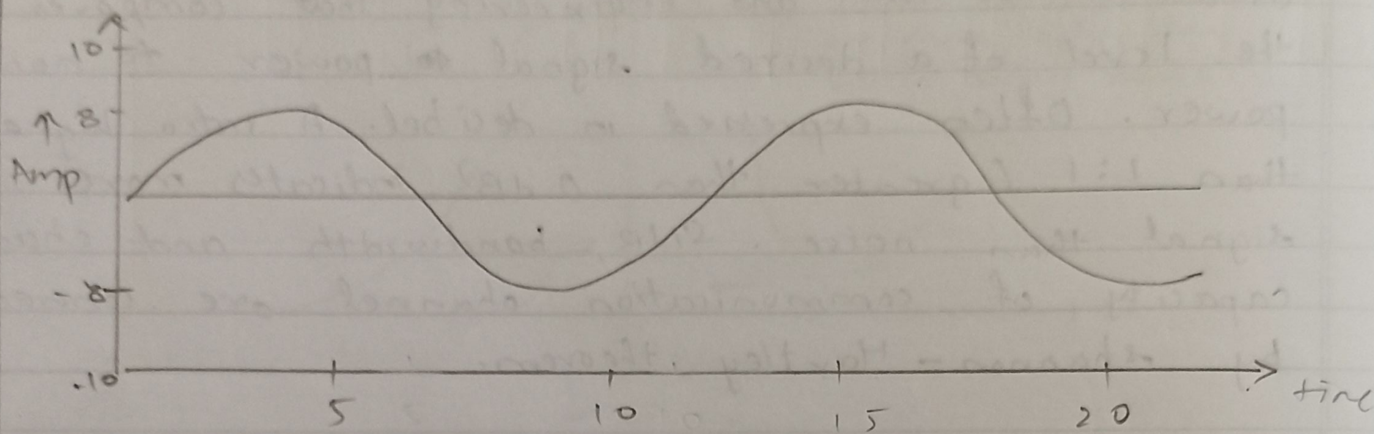
Demodulated Signal



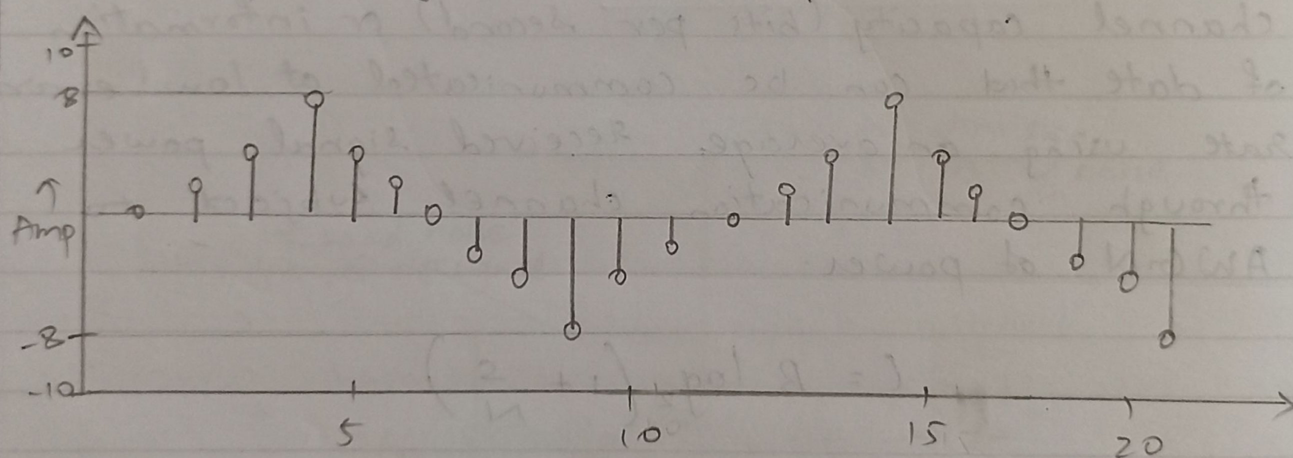
2) 6 bit system

Number of samples in a period = 12.

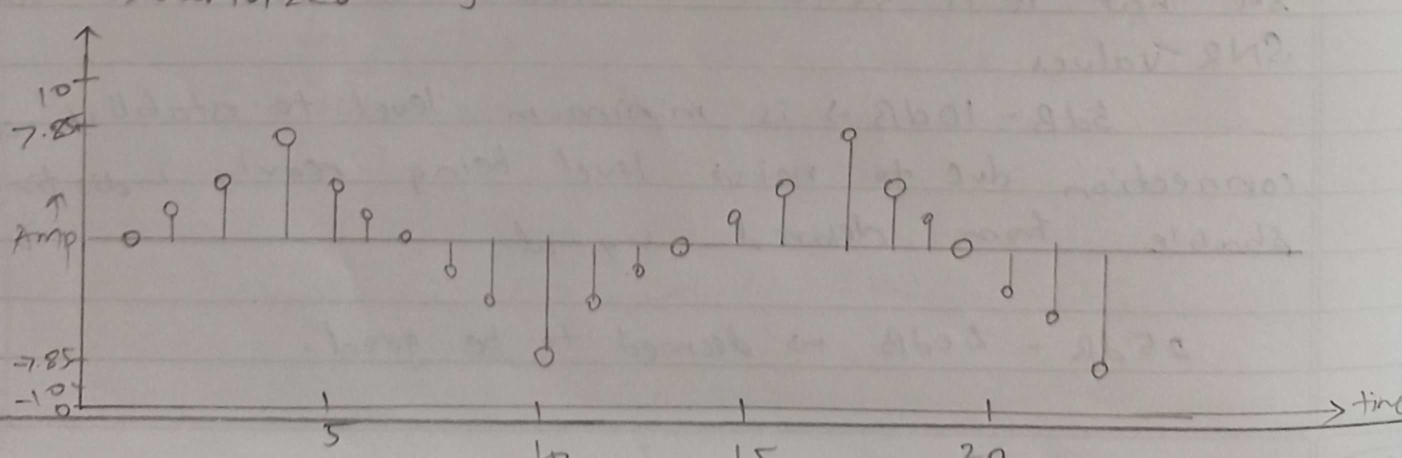
Analog System



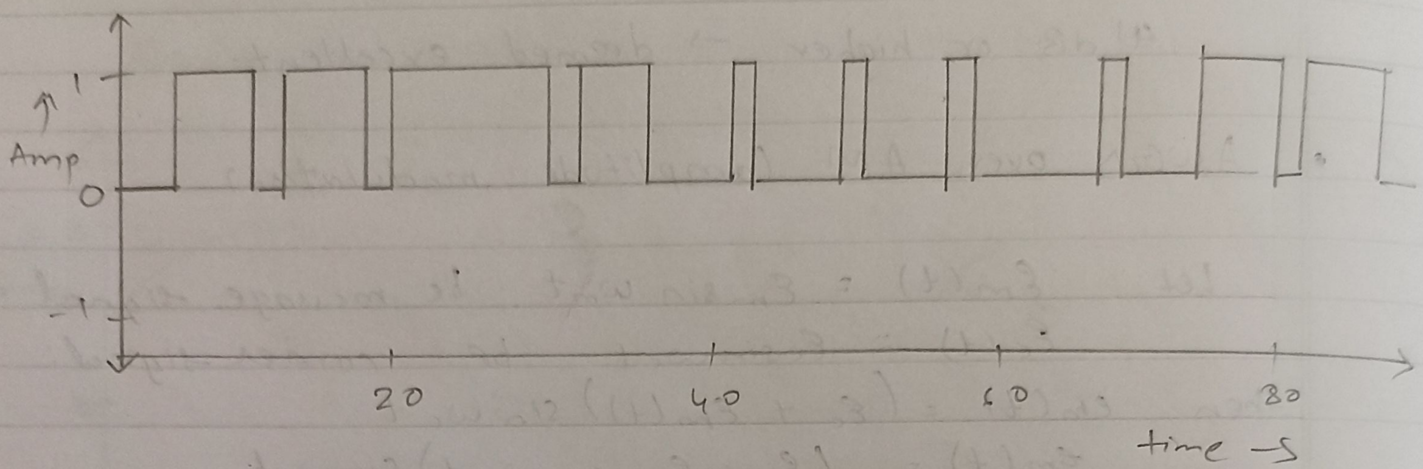
Sampled Signal



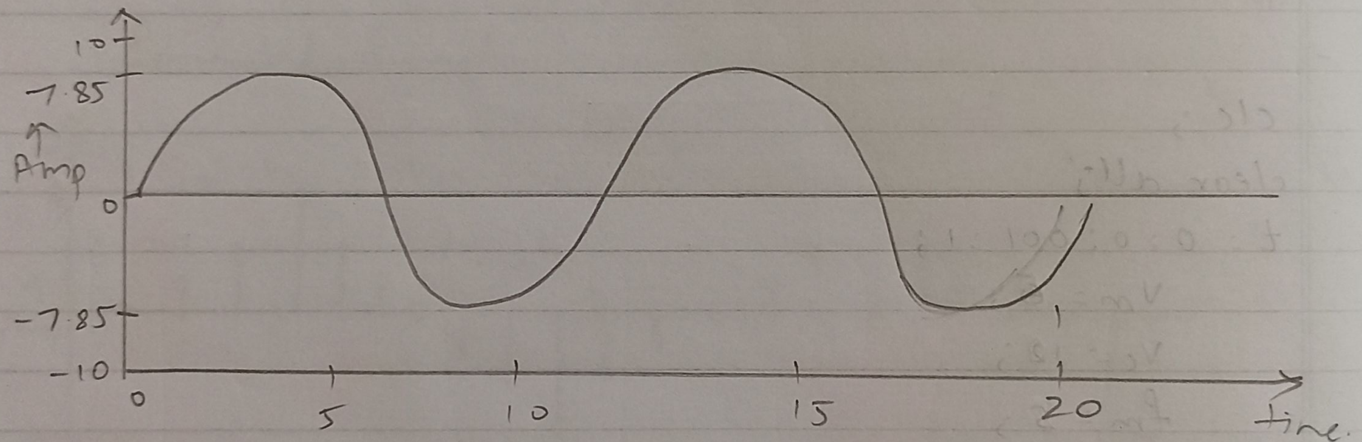
Quantized Signal.



Encoded Signal



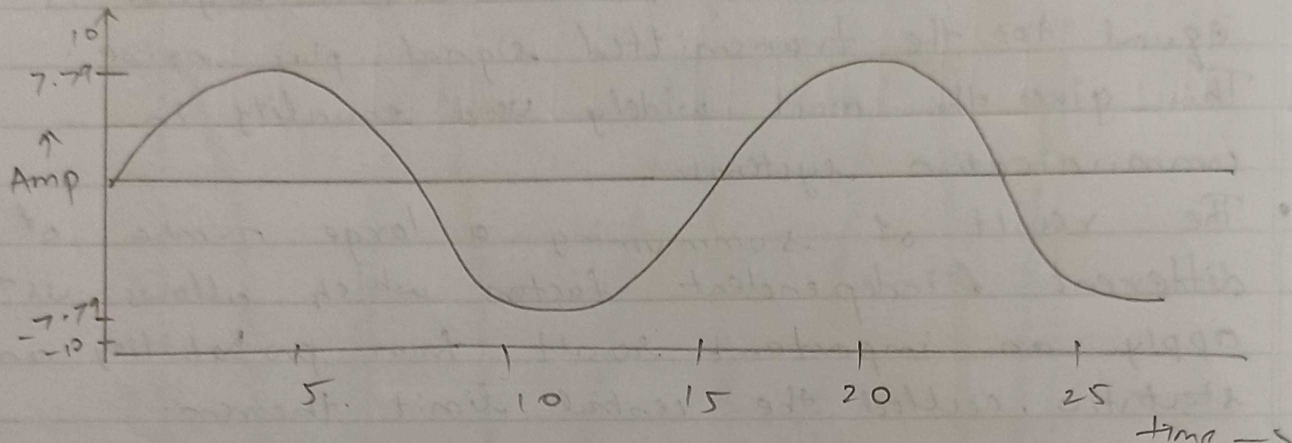
Demodulated Signal



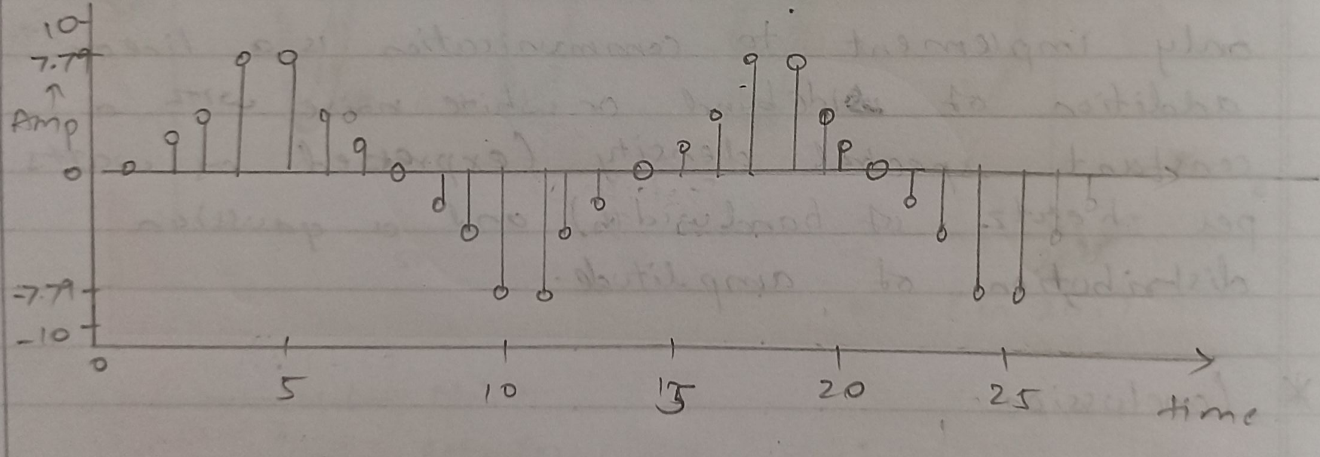
3) 8-bit system

Number of samples in a period = 14

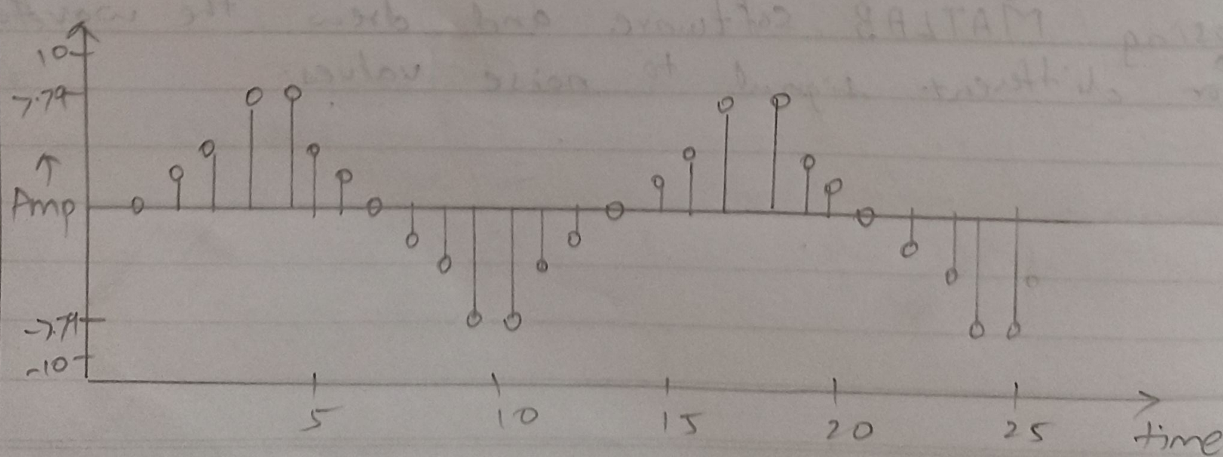
Analog Signal



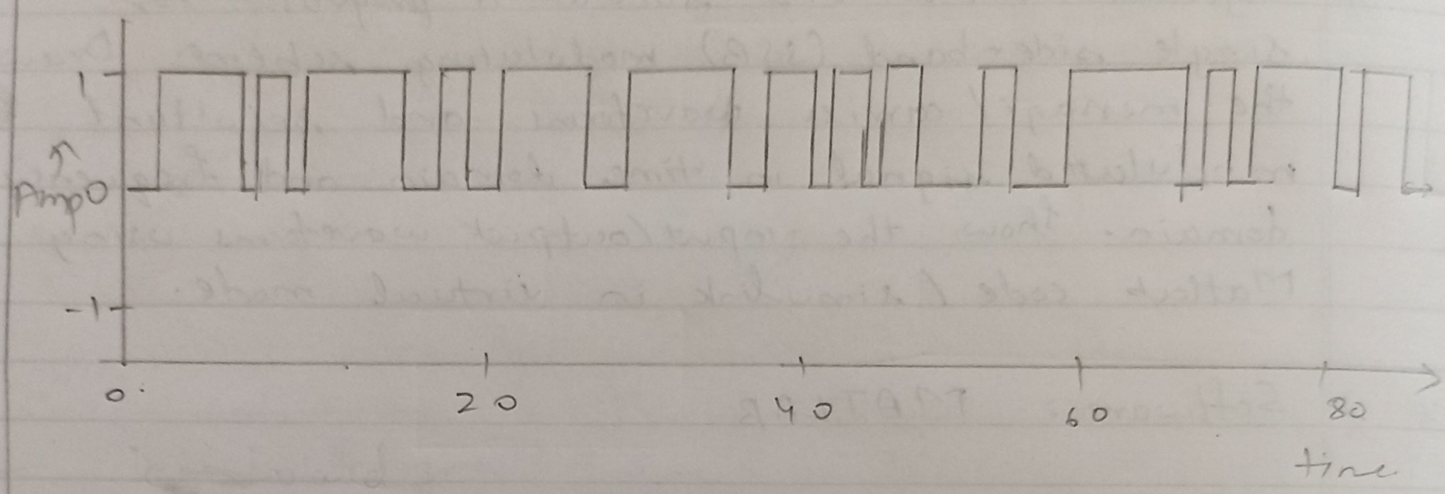
Sampled Signal



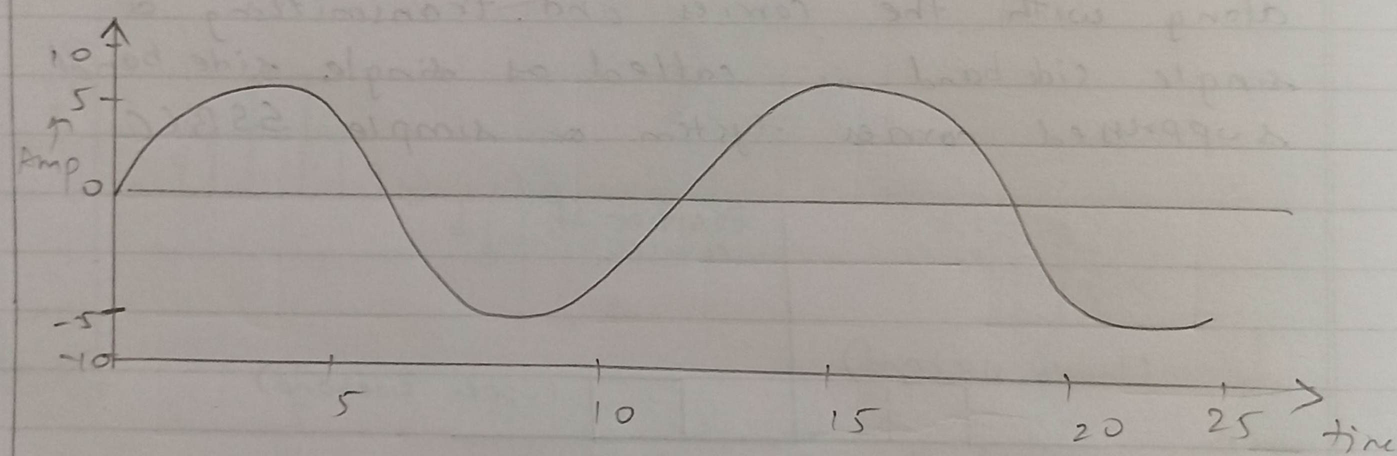
Quantized Signal



Encoded Signal



Demodulated Signal



EXPERIMENT : No. Page No. Date

* Advances of PCM:

1) Immune to channel induced noise and distortion.
2) Repeates 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 increses the transmission bandwidth
2) A PCM system is somewhat more complex that another system.

* Application-

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

* Conclusion:
Successfully demonstrated the Pulse Code Modulation technique and observed sampled, quantized, encoded and decoded time domain signal for different bit codes using matlab software.

EXPERIMENT : No. 10 DELTA Modulation.

Page No. | Date

* 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 work.

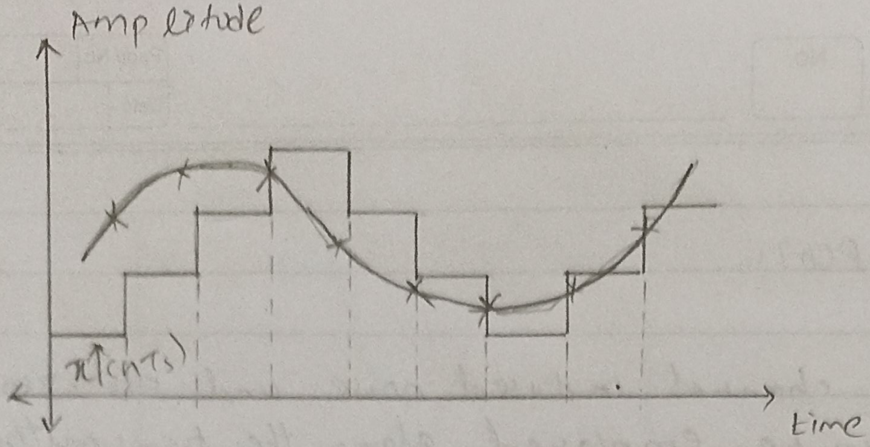
* Apparatus- MATLAB software.

* Theory

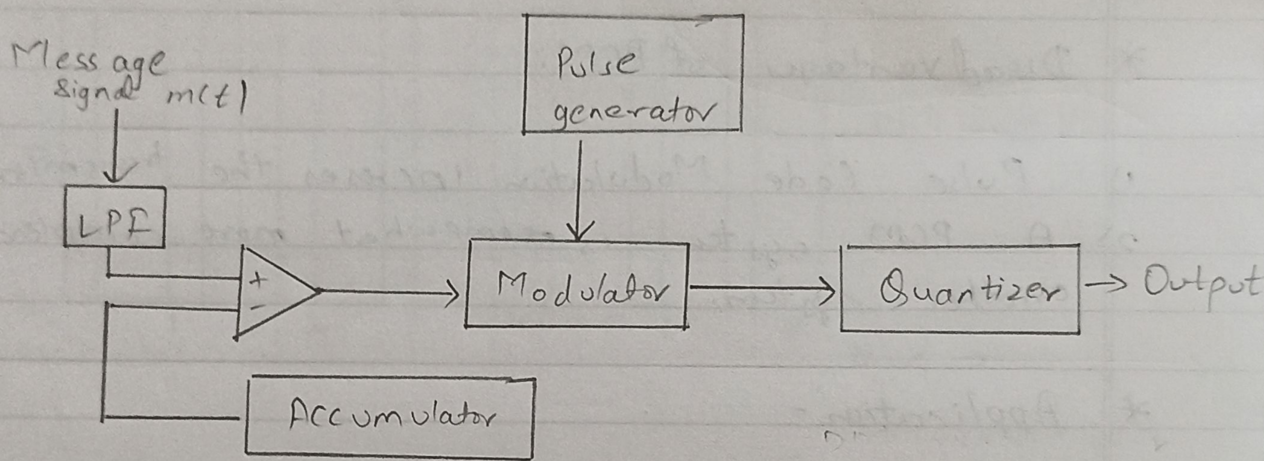
Delta modulation is a technique used to convert analog to digital and digital to analog signal. In this modulation signal is sent in differential form the data is encrypted / transmitted in 1 bit. The analog signal is approximated with series of segments and each signal is approximated with series of 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 one 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 the levels i.e $\pm\Delta$ (delta). This is used for voice transmission.

Sundaram

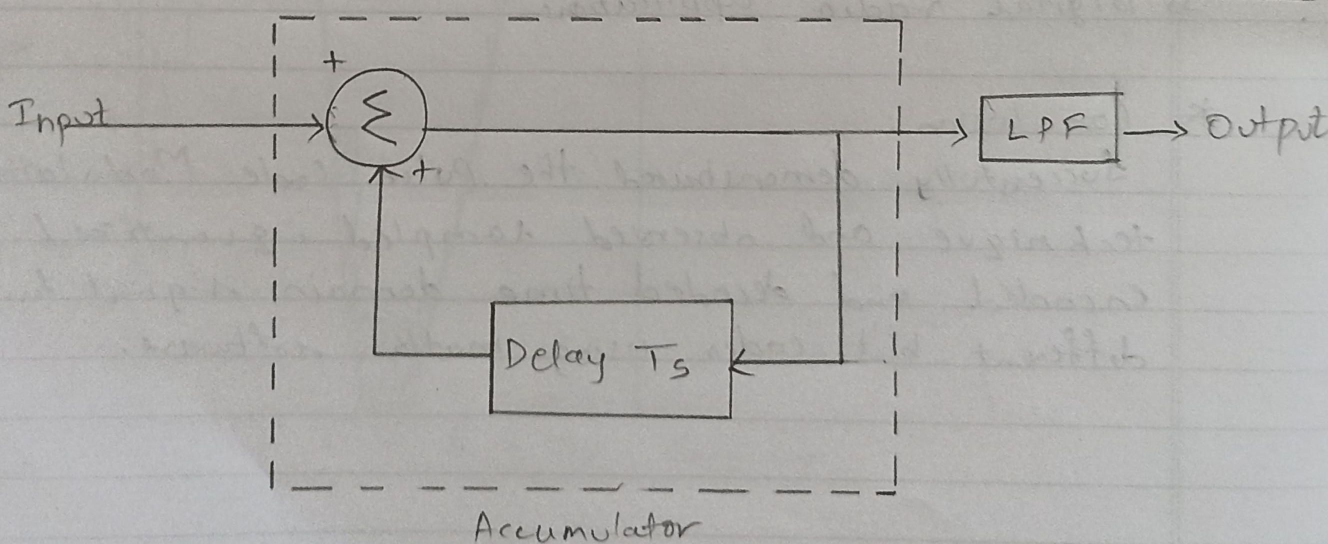
Teacher's Sign. :



Generation &
detection of DM signal

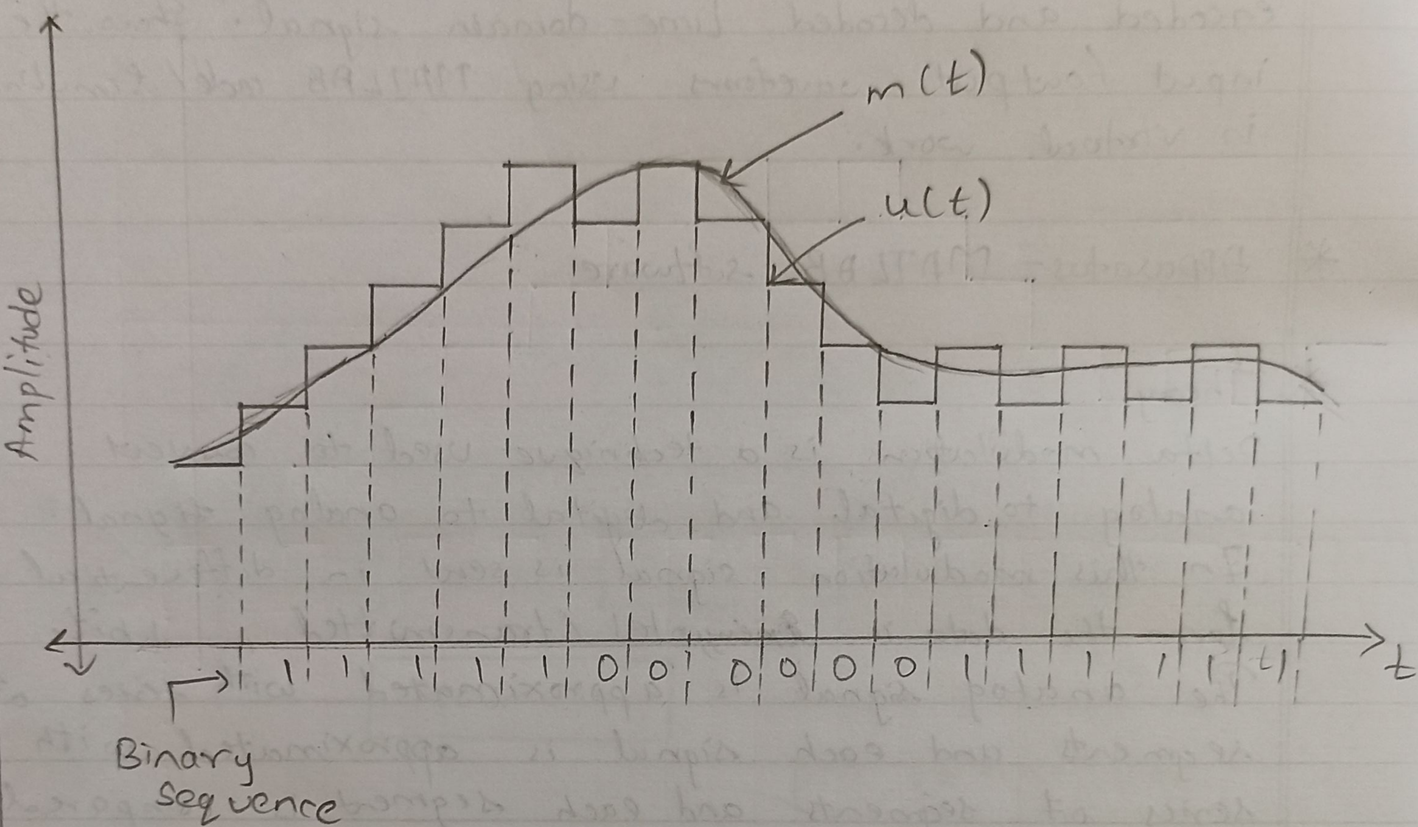


DM transmitter.



DM reciever

Waveform representation of Delta Modulated Signal



* Operating Principle -

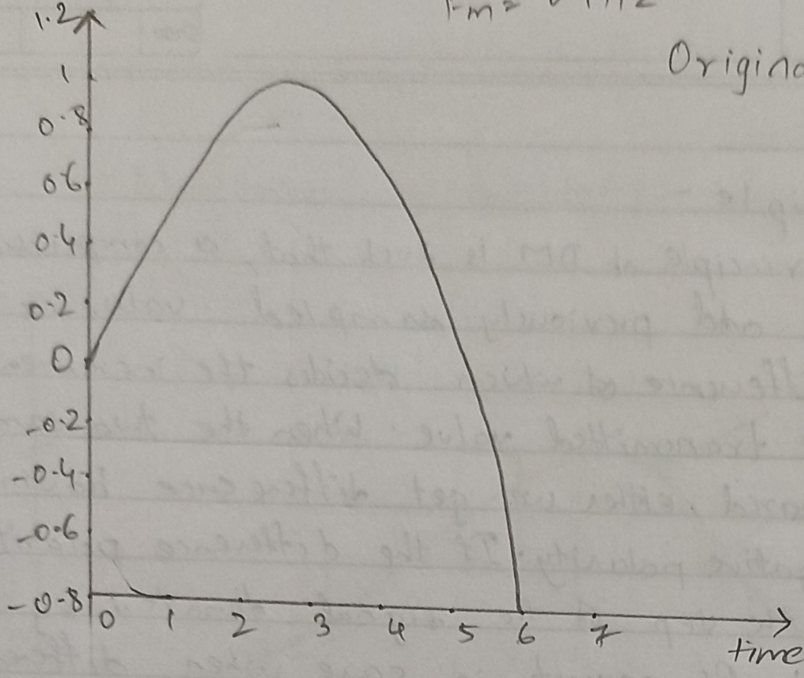
The operating principle of DM is such that, a comparision between present and previously sampled value is performed, the diffeuence of which decides the increment or decrement in the transmitted value. When the two sample values are compared, either we get difference having a positive or negative polarity. 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 them 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.

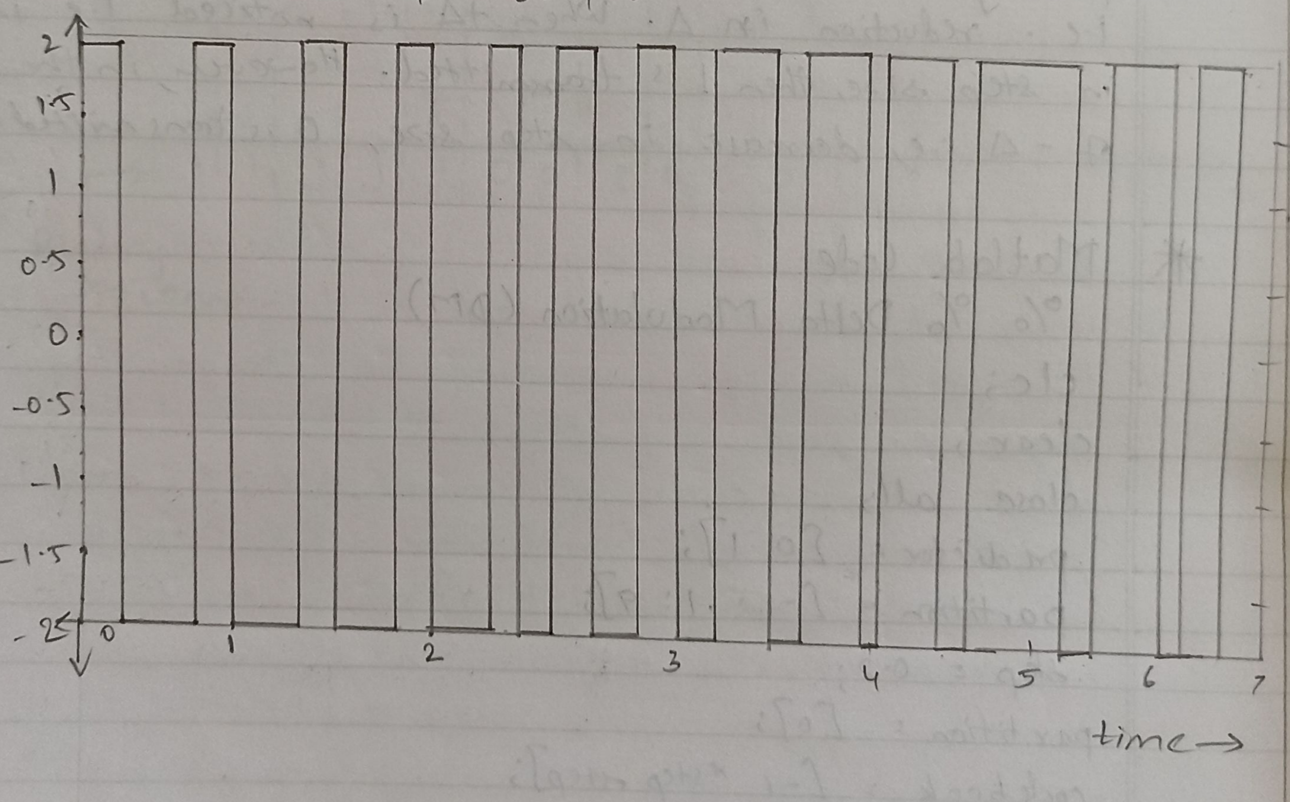
```
% % Delta Modulation (DM)
clc;
clear;
close all;
predictor = [0 1];
partition = [-1 : .1 : 9];
step = 0.2;
partition = [0];
codebook = [-1 *step step];
% DM quantizer
t = [0 : pi /20:2*pi];
x = 1.1*sin(2*pi * 0.1*t);
% Quantize x(t) using DPCM.
encodedx = dpcmenco (x, codebook, partition, predicor)
```

Fm = 0.1Hz

Original Signal

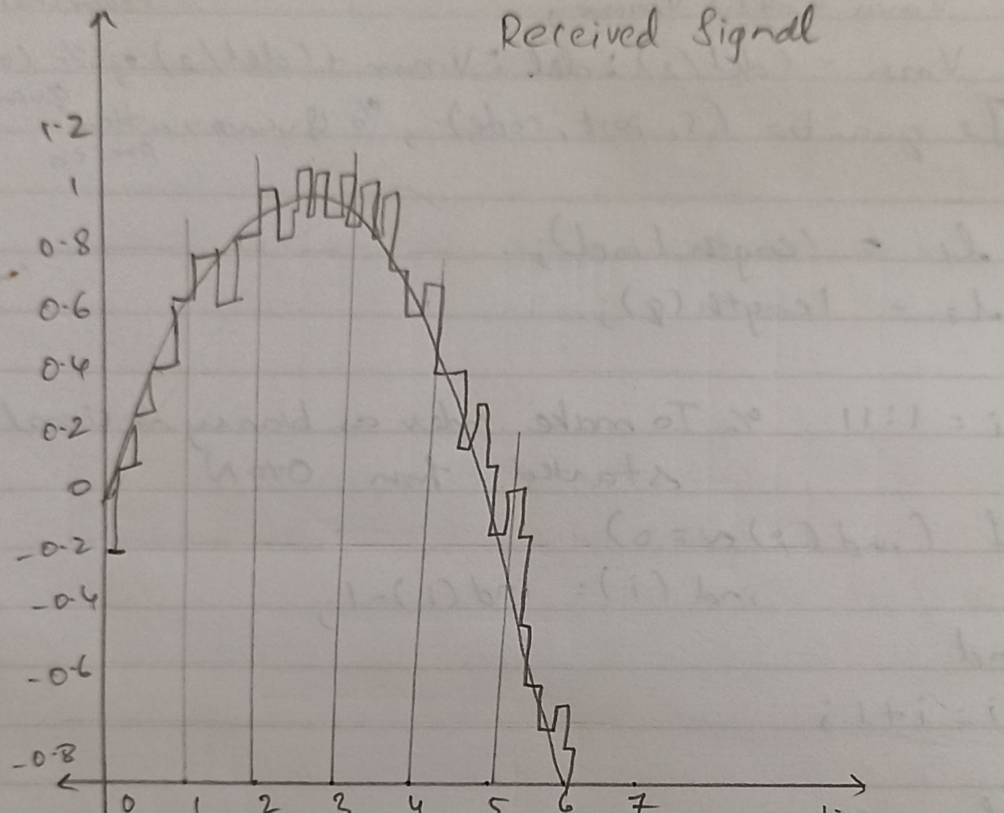


DM Output



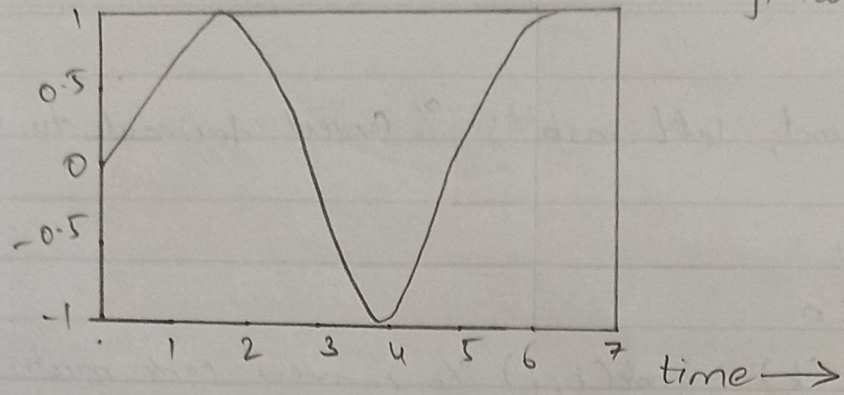
X X X

Received Signal

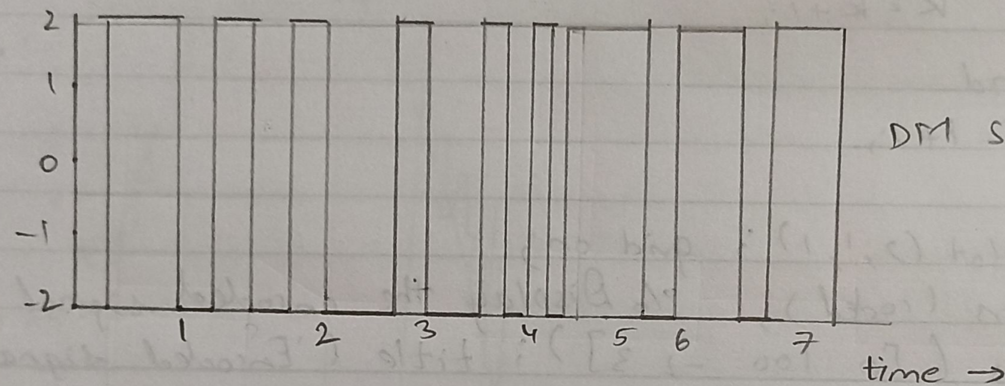


Fm = 0.2

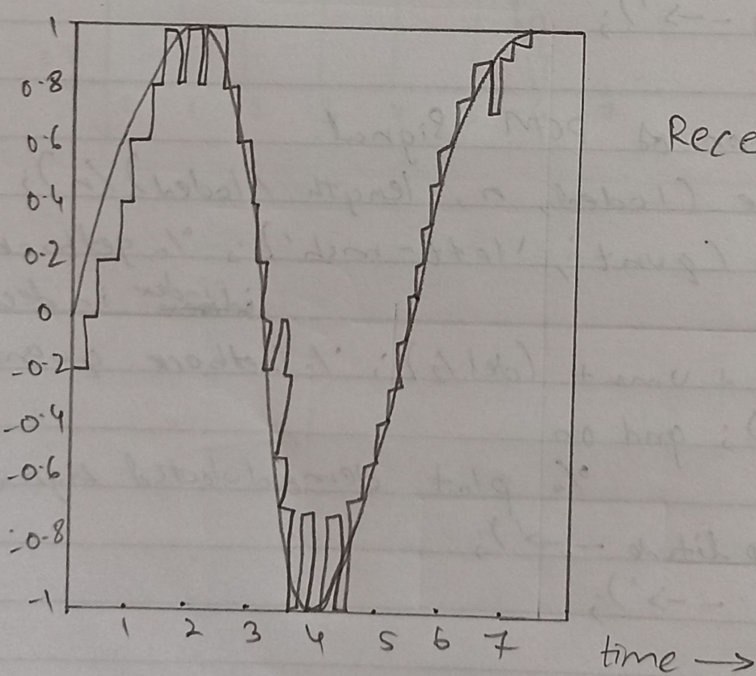
Original Signal



DM Signal



Received Signal



EXPERIMENT : No. Page No. Date

```
% Try to recover x from the modulated signal
decoded x = dpcmdeco (encoded_x, codebook, prediction);
figure
plot(t,x);
xlabel('time'); title ('original signal');
figure
stairs (t, 10*codeblook (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 channel bandwidth as well as sampling 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) & granular (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.

Teacher's Sign. :

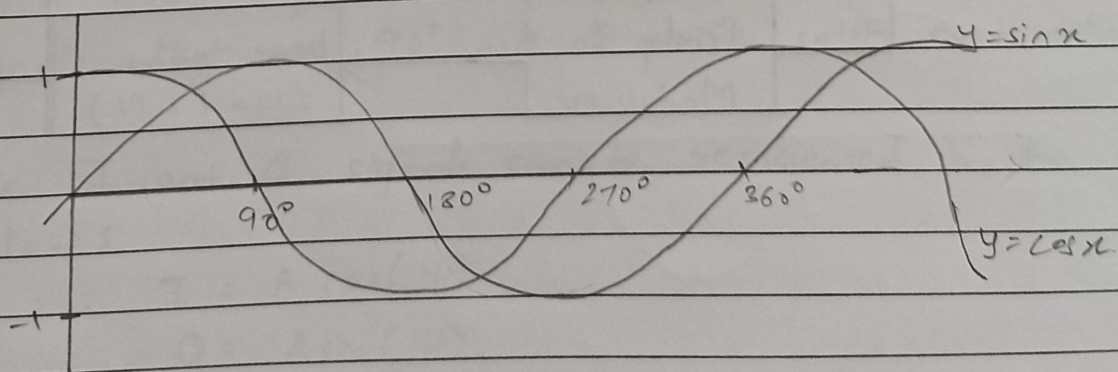
EXPERIMENT : No. 11 16-Quadrature Amplitude Modulation & Demodulation.

Page No. | Date

* Objective:- To examine the 16-Quadrature Amplitude Modulation and Demodulation scheme. Draw the 16-QAM m-ary mapped signal and modulated warfoms. Evaluate the BER values for it. Show the input/output waveforms using MATLAB code/simulink in virtual mode.

* Software- MATLAB

* 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 carrier shifted in phase by 90° are modulated and the resultant output consists of both amplitude and phase variations. Hence, it may be considered as a mixture of amplitude and phase modulation. QAM is both an analog and digital modulation technique.



Quadrature = sine wave + cosine wave.

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The main aim of QAM is to save bandwidth. Two modulated signals 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 too independent double sideband and suppressed carrier signals in the same spectrum.

- Types of QAM:

A variety of forms of QAM are available which include

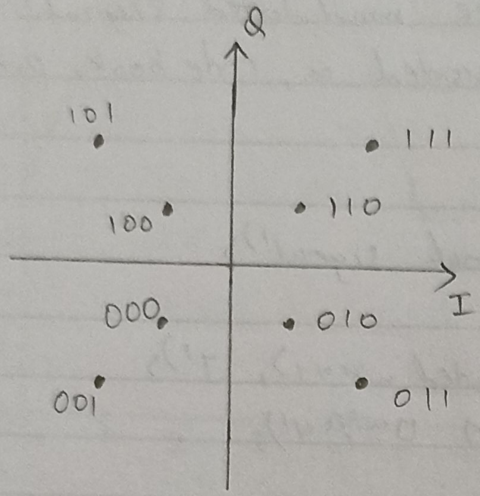
→ 16 QAM
→ 32 QAM
→ 64 QAM
→ 128 QAM
→ 256 QAM

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

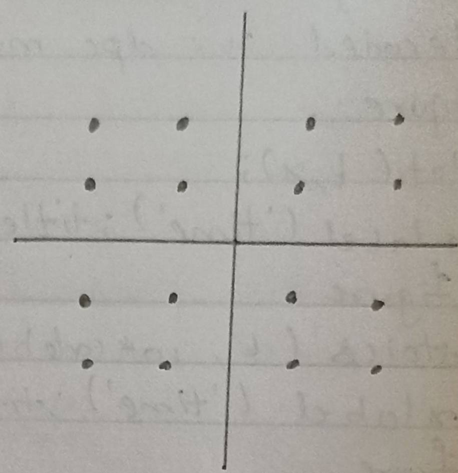
The I and Q signals can be represented by the equations below:

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

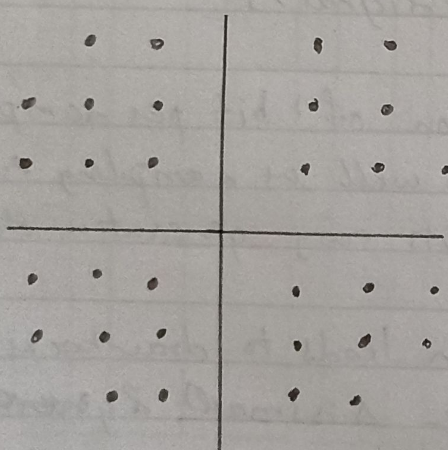
Types of QAM - Constellation Diagrams



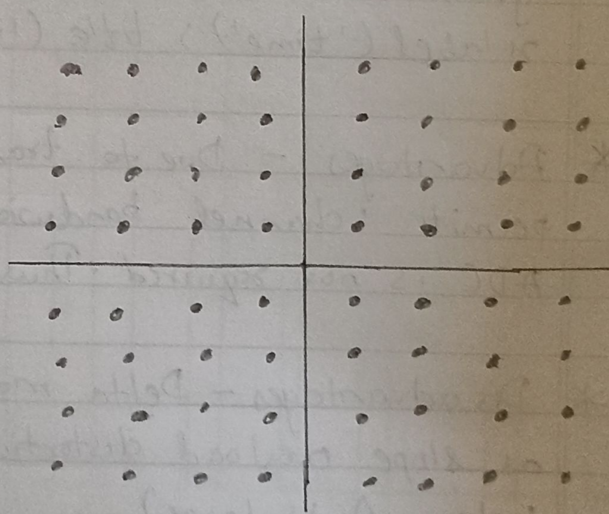
(8 QAM)



(16 QAM)



(32 QAM)



(64 QAM)

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These signal will not overlap each because they are orthogonal. It is possible to transmit two DSB-SC signals with a bandwidth of 2fm.

→ Provide bandwidth efficiency
→ Gives better performance
→ Improves data rate.

- QAM Demodulation:

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

- Bit Error Rate (Received Bits)

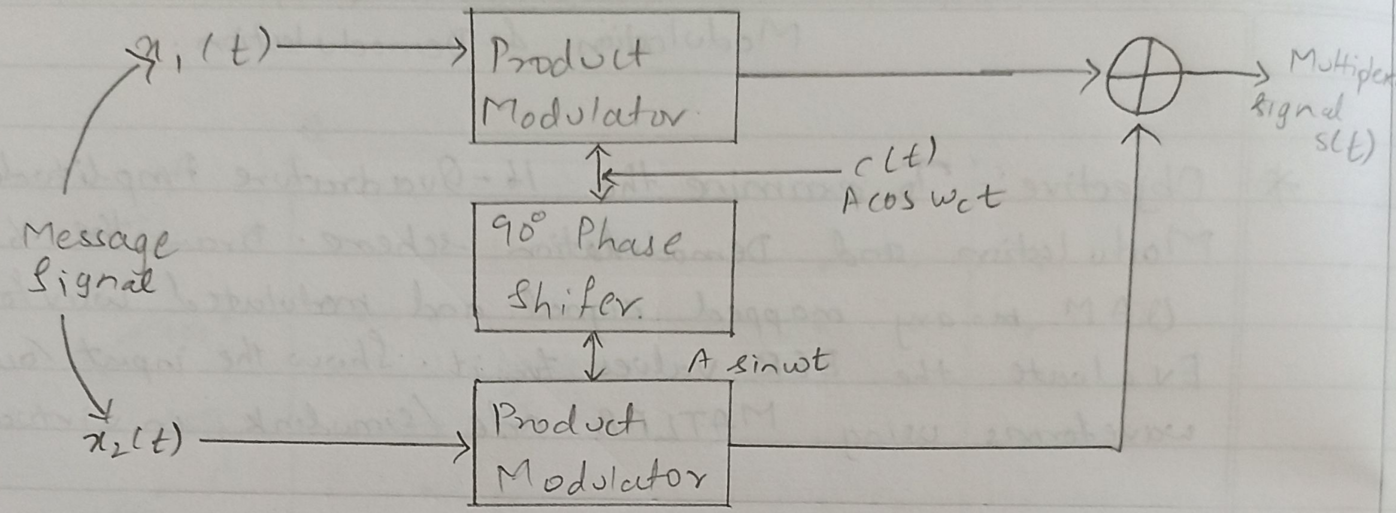
→ While higher order modulation rates ~~are~~ able to offer much faster data rates and higher levels of spectral efficiency for the radio communication system, this comes at a price.

- The higher order modulation schemes ~~are~~ considerably less ~~resistient~~ to noise and interferences.
- Many radio communications systems now use dynamic adaptive modulation techinques. They sense the channel conditions and adapt the modulation scheme to obtain the higher data rate for given conditions.
- M-QAM technique provide better bit error rate performance than M-PSK modulation technique.

Sundaram Teacher's Sign. :

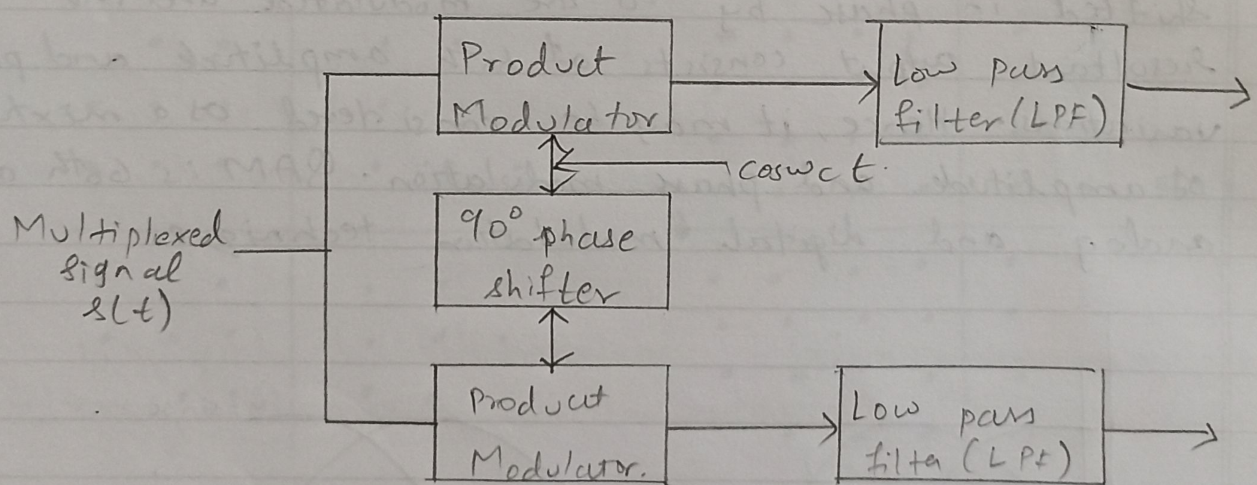
BLOCK DIAGRAMS

1) QAM Modulations



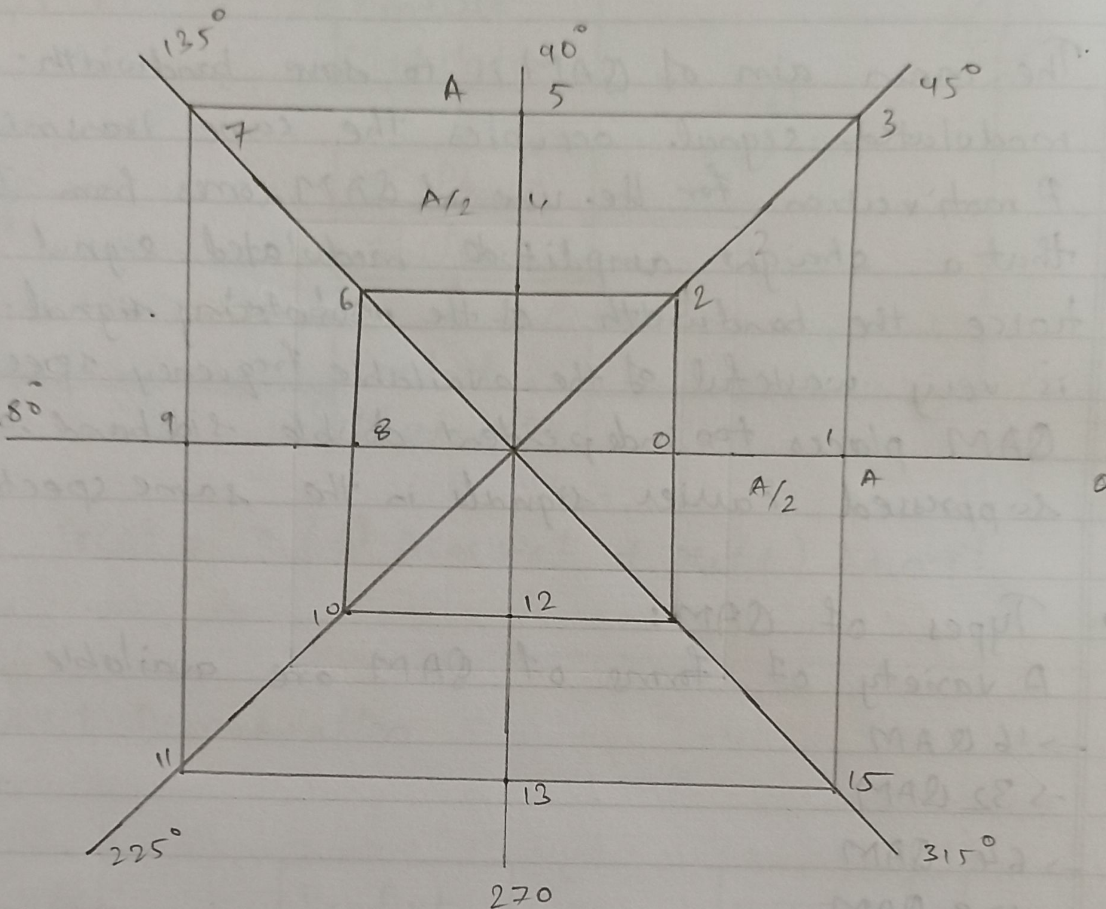
$$s(t) = x_1(t)\, A\cos w_c t + x_2(t)\, A\sin w_c t$$

2) QAM Demodulation

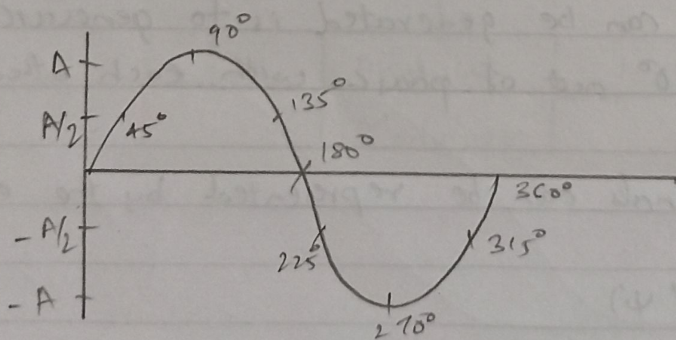


16 QAM Waveforms

2 Amplitude × 8 phase $(16 = 2^4 = 4 \text{ bits})$



Phasor Diagram:



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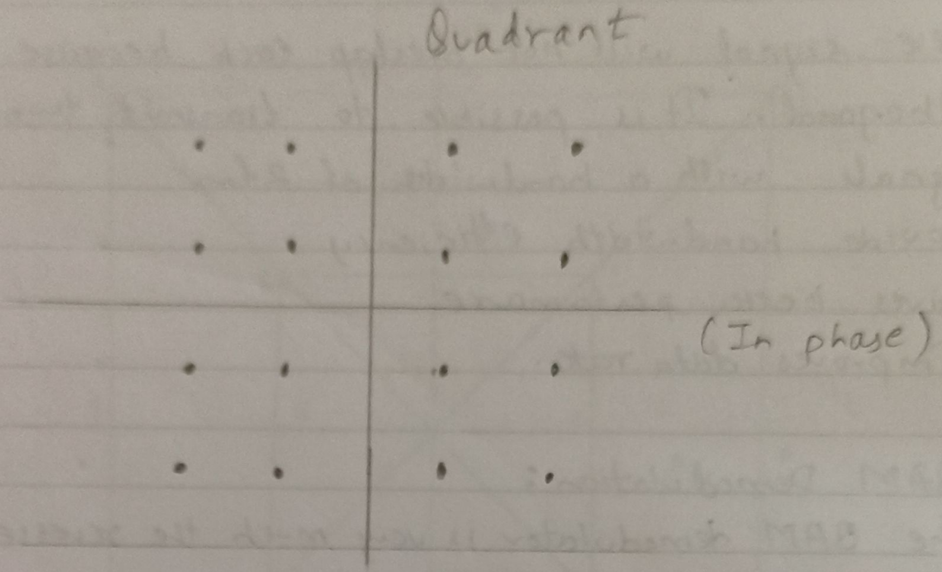
Date

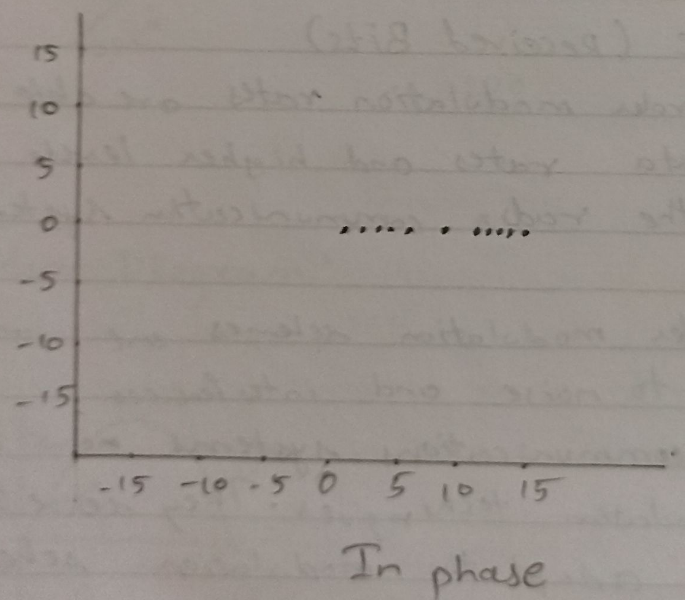
* MATLAB Code -

```
clc;
clear all;
close all;
M = 16;
 x = (0:M-1);
y = scatterplot (y);
% z = qamdemod (y, M, pi/y)
% scatterplot (z)
ber 1 = [ ]
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)))
    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^-6 10^0]);
grid on;
```

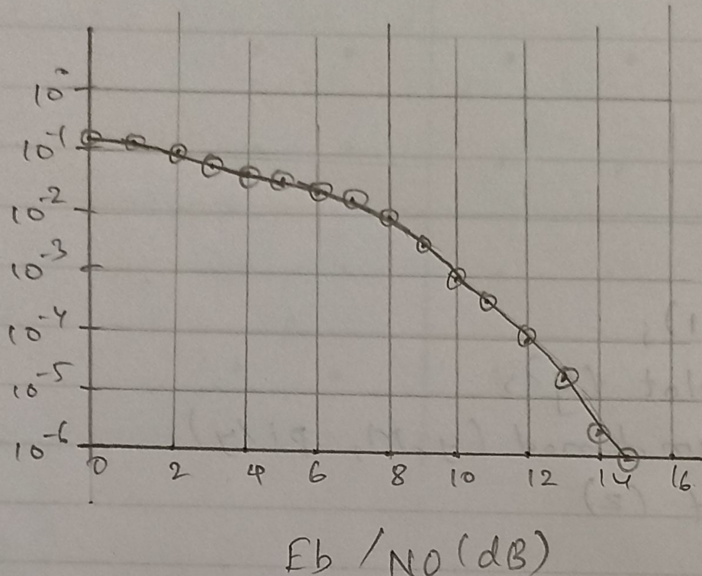
Scatter plot of 16-QAM Modulation:



Scatter plot for 16- QAM Demodulating



BER of 16-QAM:



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* Advantages:

- The advantage of using QAM is that it is higher order form of modulation. As a result it is able to carry more bits of unifomation persymbol.
- 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.

* Applications

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

Modulation	Bits per symbol	System Rate.
BPSK	1	1 bitrate
QPSK	2	1/2 bitrate
8PSK	3	1/3 bitrate
16QAM	4	1/4 bit rate
32QAM	5	1/5 bitrate
64QAM	6	1/6 bitrate.

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