BPIT

Communication System

Practical file

(IV SEMESTER)

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CSE-A

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INDEX

Experiment-1

Aim:

To study the function of Amplitude Modulation & Demodulation (under modulation, perfect modulation & over modulation) and also to calculate the modulation index.

Theory:

Amplitude modulation (AM) is defined as a process in which the amplitude of the carrier wave c(t) is varied about a mean value, linearly with the base band signal m(t). An AM wave may thus be described, in its most general form, as a function of time as follows.

$$S(t)=A_C\,[1+K_a m(t)]\,\mathrm{Cos}\,(2\,\pi f_C t)$$

Where K_a	———	Amplitude Sensitivity of the modulator
S(t)	———	Modulated signal
^AC	———	Carrier Amplitude
m(t)	———	Message Signal

The amplitude of $K_a m(t)$ is always less than unity, that is

$$|K_a m(t)| < 1 \text{ for all } t$$

It ensures that the function 1+Kam(t) is always positive. When the amplitude sensitivity Ka of the modulator is large enough to make |Kam(t)| >1 for any t, the carrier wave becomes over modulated, resulting in carrier phase reversals. Whenever the factor 1+Kam(t) crosses zero. The modulated wave then exhibits envelope distortion as shown in fig. below.

The absolute maximum value of Kam(t) multiplied by 100 is referred to as the percentage modulation.

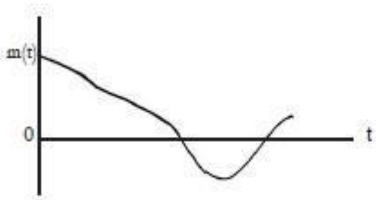
$$\text{or percentage modulation} = \frac{V_{max} - V_{min}}{V_{max} + V_{min}} \times 100$$

The carrier frequency fc is much greater than the highest frequency component w of the message signal m(t), that is fC >>W Where W is the message bandwidth.

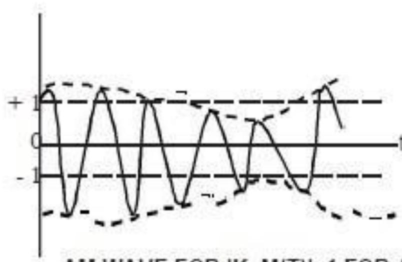
If this condition is not satisfied, envelope cannot be visualized (and therefore detected) satisfactorily.

PHYSITECH'S modulation and demodulation trainer has a carrier generator, which generates carrier wave of 100 KHz when the trainer is switched on.

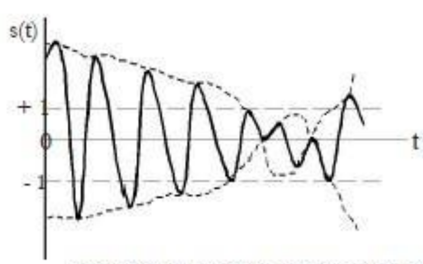
The blocks, carrier generator, modulator and demodulator are provided with built in supplies, no supply connections are to be given externally.



BASE BAND SIGNAL M(T)



AM WAVE FOR $|K_A M(T)|<1$ FOR ALLT

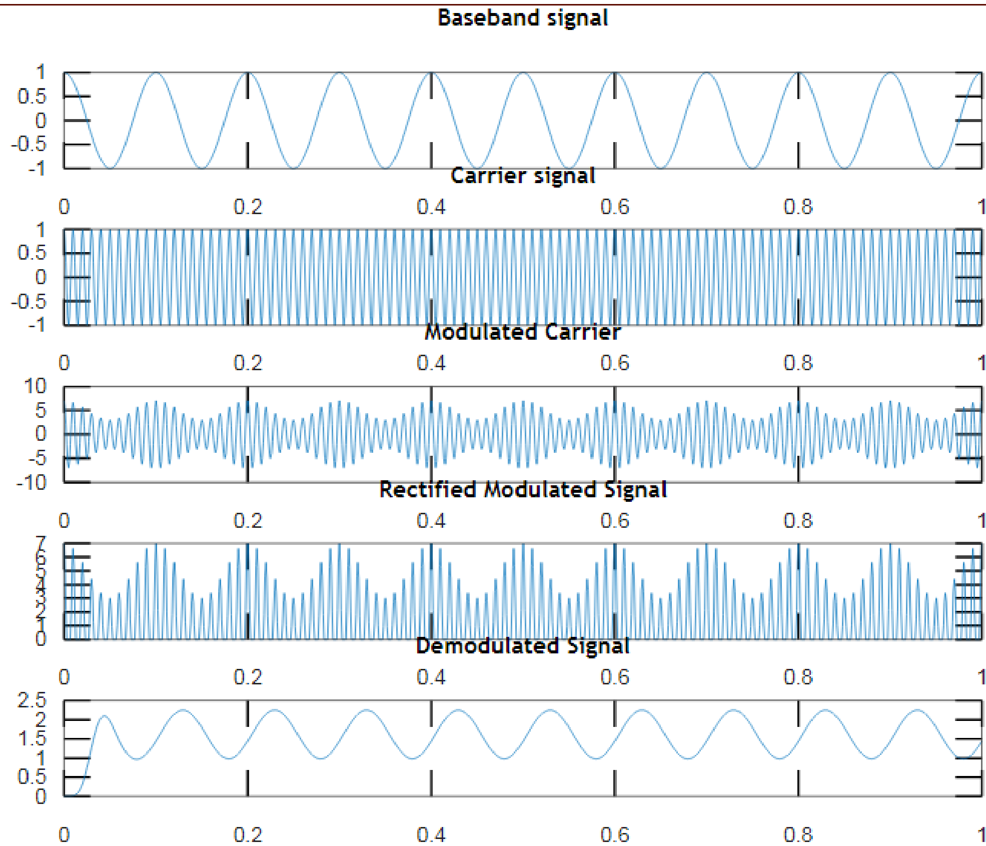


AM WAVE FOR $|K_A M(T)|>1$ FOR SOME

Code :

```
A=input('enter the carrier signal peak')  B=input('enter the
baseband signal peak')  f1=input('enter the baseband signal
frequency')  f2=input('enter the carrier signal frequency')
fs=input('enter the sampling frequency')

t=0:0.001:1;
M=cos(2*pi*f1*t);
N=cos(2*pi*f2*t);
m=B/A;
O=A*(1+m*M).*N;
O1=O;  for
i=1:length(t)  if
O1(i)<=0  O1(i)=0; end
end

[den, num]=butter(2,2*pi*f1/fs);

M1=filter(den,num,O1);

M11=filter(den,num,M1);

M12=filter(den,num,M11);
M13=filter(den,num,M12);
subplot(5,1,1)  plot(t,M)
title('Baseband signal')
subplot(5,1,2)  plot(t,N)
title('Carrier signal')
subplot(5,1,3)  plot(t,O);
title('Modulated Carrier')
subplot(5,1,4)

plot(t,O1)

title('Rectified Modulated Signal')
subplot(5,1,5)  plot(t,M13)

title('Demodulated Signal')
```

Output :


Baseband signal
Carrier signal
Modulated Carrier
Rectified Modulated Signal
Demodulated Signal
0
0.2
0.4
0.6
0.8
1

Experiment-2

Aim:

To study Balanced modulator for DSB-SC modulation, demodulation.

Theory:

In Balanced modulator, two non-linear devices are connected in the balanced mode, so as to suppress the carrier wave.

The Balanced Modulator consists of summing devices (operational amplifiers) and two matched nonlinear elements. If x(t) is band limited to fx and if fc>2fx, then the band pass filter output will be the desired product signal.

Figure shows IC that has been specifically designed for use as balanced modulators. Figure is the 1496 balanced modulator which is manufactured by Motorola, National, and Signetics. This device uses a differential amplifier configuration. Its carrier suppression is rated at a minimum of -50dB with a typical value -65dB at 500 KHz.

PHYSITECH'S trainer contains a balanced modulator using a 1496 integrated circuit. You will verify that it does suppress the carrier and also adjust it for optimum carrier suppression.

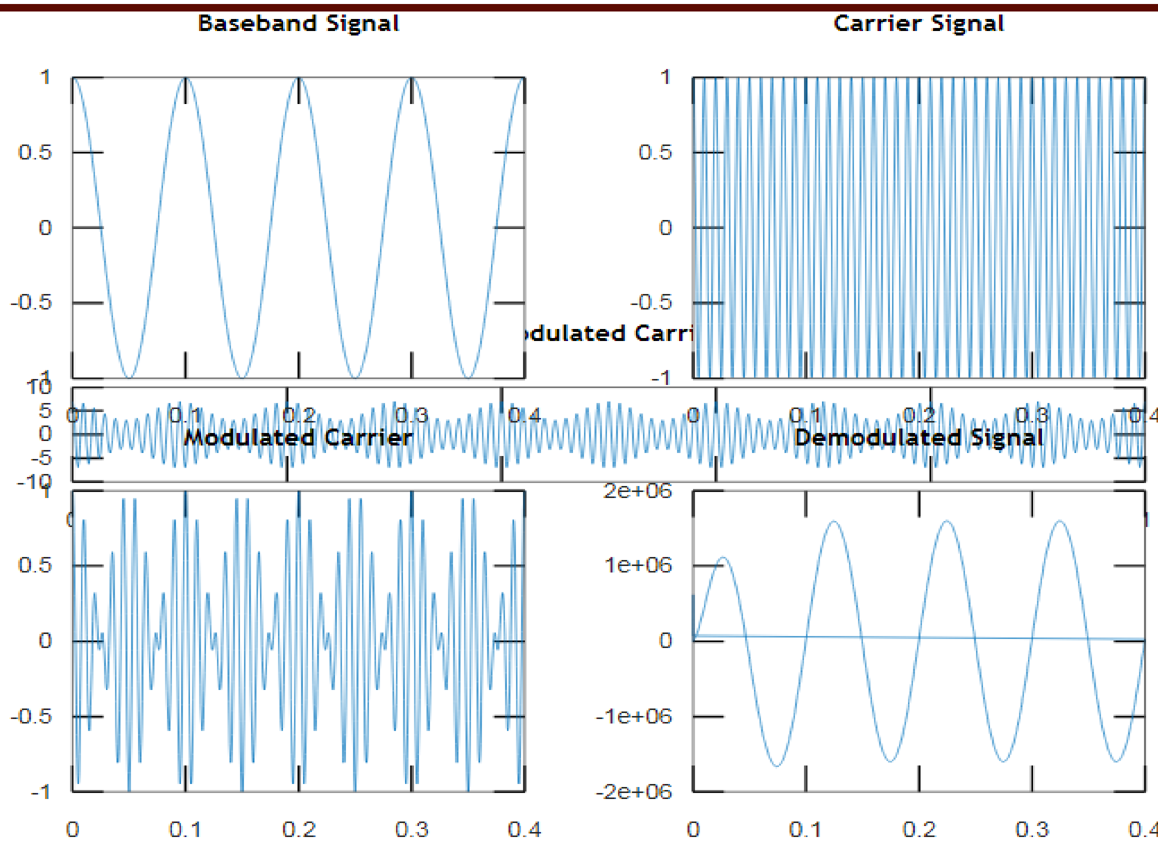
Code:

```
f1=input('enter the baseband signal frequency')

f2=input('enter the carrier signal frequency')

T=input('enter the duration over which the signal is to be plotted')

fs=input('enter the sampling frequency')

t=0:T/fs:T;

M=cos(2*pi*f1*t);

N=cos(2*pi*f2*t);

O=M.*N;

P=O.*N;

C=input('enter the value of the capacitor of the filter')
```

```
R=1/(2*pi*f1*C); H=(1/(R*C))*exp(-t/(R*C));

h=conv(H,conv(P,H));

t1=t;

for i=length(t)+1:length(h)

   t1(i)=0;

end

subplot(2,2,1)

plot(t,M)

title('Baseband Signal')

subplot(2,2,2)

plot(t,N)

title('Carrier Signal')

subplot(2,2,3)

plot(t,O)

title('Modulated Carrier')

subplot(2,2,4)

plot(t1,h)

title('Demodulated Signal')
```

Output:

```
enter the baseband signal frequency> 10
f1 = 10
enter the carrier signal frequency> 100
f2 = 100
enter the duration over which the signal is to be
plotted> 0.4000
T = 0.4000
enter the sampling frequency> 1000
fs = 1000
enter the value of the capacitor of the filter>
1.3330e-08
C = 1.3330e-08
```


Baseband Signal
Carrier Signal
Modulated Carrier
Demodulated Signal

Experiment-3

Aim:

To study the functioning of frequency modulation & demodulation and to calculate the modulation index.

Theory:

FM is a system in which the amplitude of the modulated carrier is kept constant, while its frequency and rate of change are varied by the modulating signal.

By the definition of FM, the amount by which the carrier frequency is varied from its unmodulated value, called the deviation, is made proportional to the instantaneous amplitude of the modulating voltage. The rate at which this frequency variation changes or takes place is equal to the modulating frequency.

FM is that form of angle modulation in which the instantaneous frequency $f_i(t)$ is varied linearly with the message signal m(t), as

$$f_i(t) = f_C + k_f\, m(t)$$

The term f_c represents the frequency of the unmodulated carrier, and the constant K_f represents the frequency sensitivity of the modulator expressed in Hertz per volt.

Unlike AM, the spectrum of an FM signal is not related in a simple manner to that of modulating signal, rather its analysis is much more difficult than that of an AM signal.

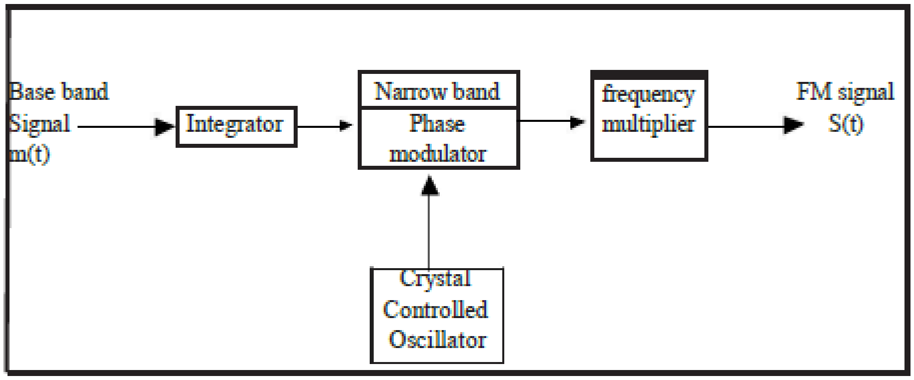
<u>Generation of FM signals:</u>

There are essentially two basic methods of generating frequency modulated signals, namely, direct FM and indirect FM. In the direct method, the carrier frequency is directly varied in accordance with the input base band signal, which is readily accomplished using a voltage-controlled oscillator. In the indirect method, the modulating signal is first used to produce a narrow band FM signal, and frequency multiplication is next used to increase the frequency deviation to the desired level.

The indirect method is the preferred choice for FM when the stability of carrier frequency is of major concern as in commercial radio broad casting.

Indirect FM :

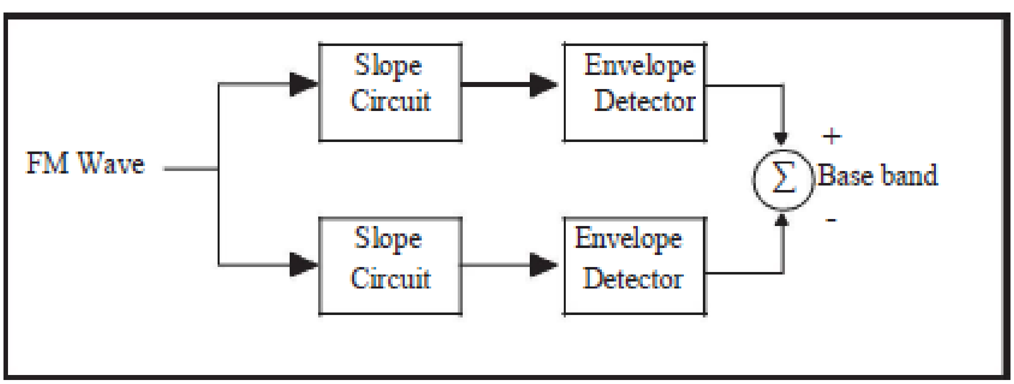
A simplified block diagram of an indirect FM system is shown in fig below.



Demodulation of fm signals:

Frequency demodulation is the process that enables us to recover the original modulating signal from a frequency – modulated signal. Here we describe a direct method of frequency demodulation involving the use of popular device known as a frequency discriminator, whose instantaneous output amplitude is directly proportional to the instantaneous frequency of the input FM signal.

Basically, the frequency discriminator consists of a slope circuit followed by an envelope detector.



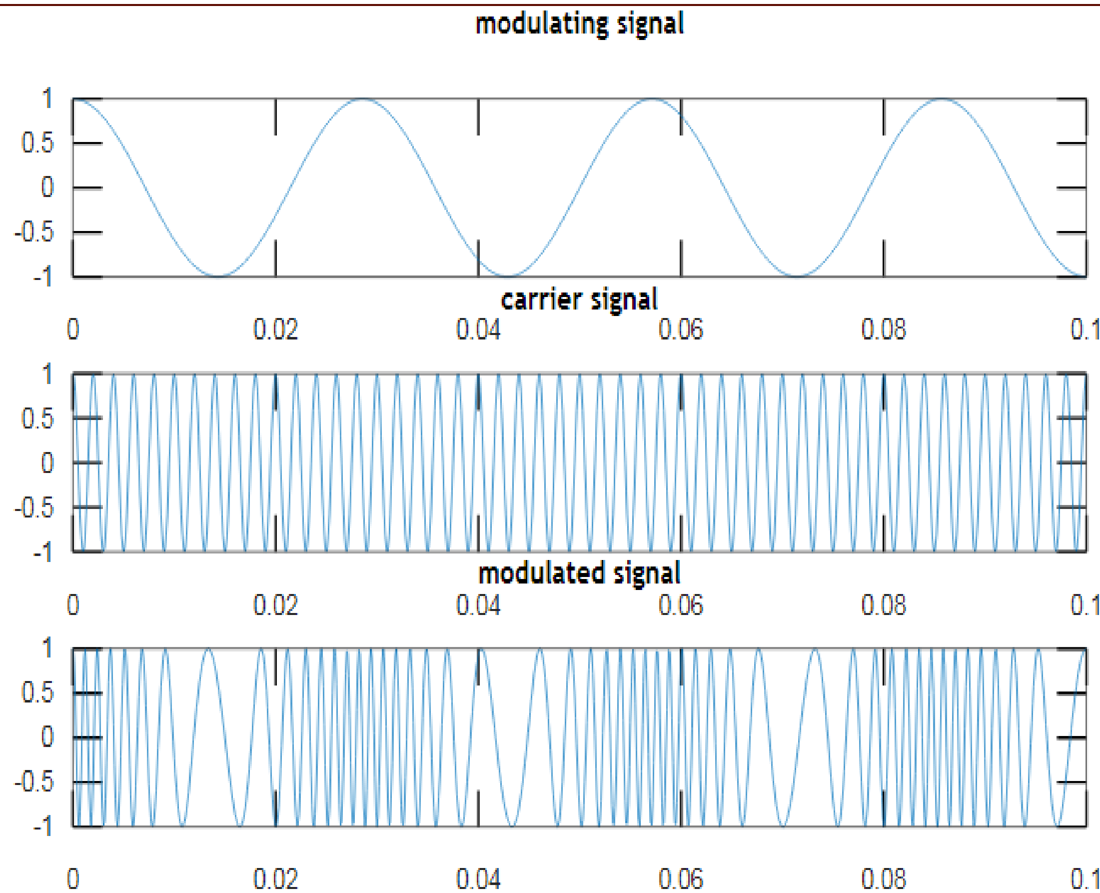
FREQUENCY DISCRIMINATOR

Code:

```
clc;
clear all;
close all;
fs=10000;
ac=1;
am=1;
fm=35;
fc=500;
b=10;
t=(0:0.1*fs)/fs;
wc=2*pi*fc;
wm=2*pi*fm;
mt=am*cos(wm*t);
subplot(4,1,1);
plot(t,mt);
title('modulating signal');
ct=ac*cos(wc*t);
subplot(4,1,2);
plot(t,ct);
title('carrier signal');
st=ac*cos((wc*t)+b*sin(wm*t));
subplot(4,1,3);
plot(t,st);
title('modulated signal');
```

```
d=demod(st,fc,fs,'fm');

subplot(4,1,4);

plot(t,d);

title('demodulated signal');
```

Output:



Experiment-4

Aim:

-To generate SSB using phase method and demodulation of SSB signal using Synchronous detector.

Theory:

The phase shift method makes use of two balanced modulators and two phase shift networks as shown in fig. One of the modulators receives the carrier signal shifted by 900 and the modulating signal with 00 (sine)phase shift, whereas the other receives modulating signal shifted by 900 (co-sine) and the carrier (RF) signal with 00 phase shift voltage.

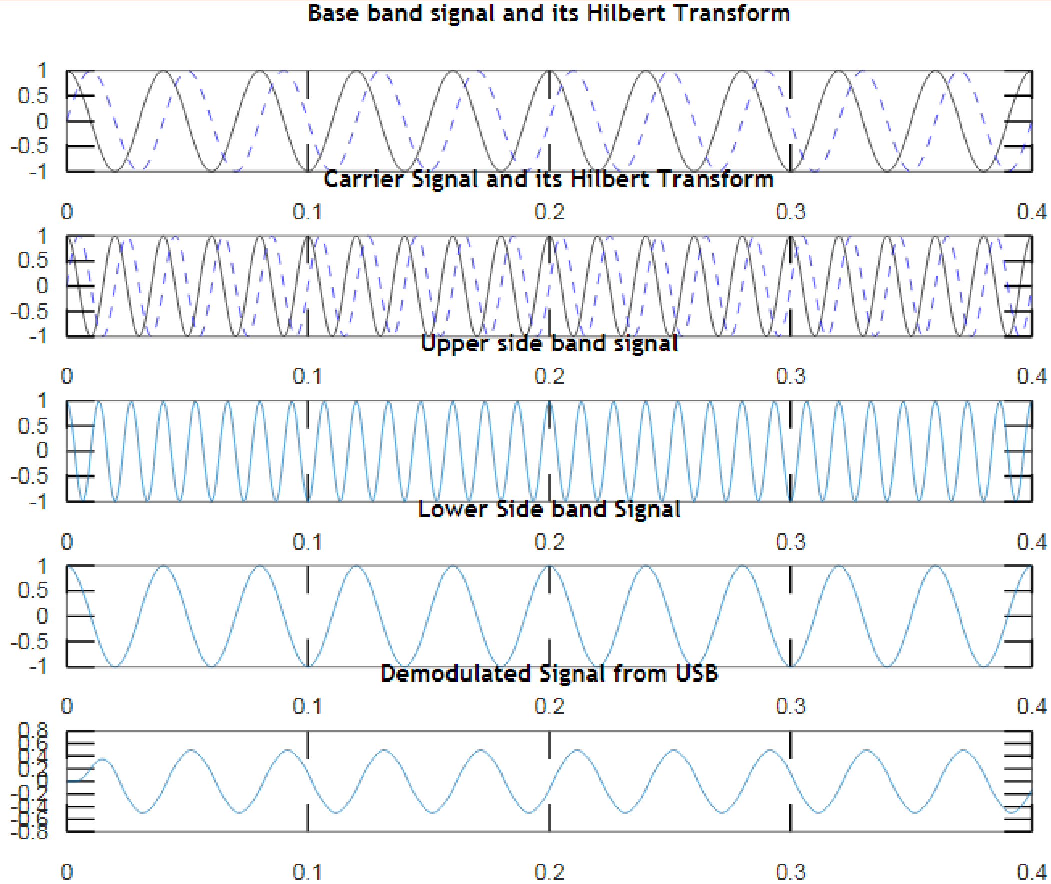
Both modulators produce an output consisting only of sidebands. It will be shown that both upper sidebands lead the input carrier voltage by 900. One of the lower sidebands leads the reference voltage by 900, and the other lags it by 900. The two lower sidebands are thus out of phase, and when combined in the adder, they cancel each other. The upper sidebands are in phase at the adder and therefore they add together and gives SSB upper side band signal. When they combined in the subtractor, the upper side bands are cancelled because in phase and lower side bands add together and gives SSB lower side band signal.

Code:

```
f1=input('enter the base band signal frequency')  f2=input('enter
the carrier signal frequency')  t=0:0.001:0.4;

fs=input('enter sampling frequency')

M=cos(2*pi*f1*t);

N=cos(2*pi*f2*t);

DSB1=M.*N;

M1=cos(2*pi*f1*t-(pi/2));
```

```
N1=cos(2*pi*f2*t-(pi/2));

DSB2=M1.*N1;

USB=DSB1-DSB2;

LSB=DSB1+DSB2;  subplot(5,1,1)

plot(t,M,'k',t,M1,'--b')

title('Base band signal and its Hilbert Transform')

subplot(5,1,2)  plot(t,N,'k',t,N1,'--b')

title('Carrier Signal and its Hilbert Transform')

subplot(5,1,3)  plot(t,USB)  title('Upper side band

signal')  subplot(5,1,4)  plot(t,LSB)

title('Lower Side band Signal')

USBMULT=USB.*N;

[den num]= butter(2,(2*pi*f1)/fs);  Filter1=filter(den,num,USBMULT);

Filter2=filter(den,num,Filter1);

Filter3=filter(den,num,Filter2);

Filter4=filter(den,num,Filter3);

subplot(5,1,5)  plot(t,Filter4)

title('Demodulated Signal from USB')
```

Output:


Base band signal and its Hilbert Transform
Carrier Signal and its Hilbert Transform
Upper side band signal
Lower Side band Signal
Demodulated Signal from USB
1
0.5
0
-0.5
-1
0
0.1
0.2
0.3
0.4

Experiment-5

Aim:

To study the Pulse Amplitude modulation and de-modulation and their waveforms.

Theory:

Pulse Amplitude Modulation (PAM) is the simplest and most basic form of analog pulse modulation, In PAM, the amplitudes of regularly spaced pulses are varied in proportional to the corresponding sample values of a continuous message signal; the pulses can be of a rectangular form or some other appropriate shape.

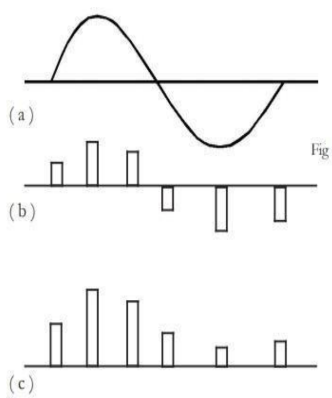


Fig : **Pulse amplitude modulation**

a. Signal

b. Double-Polarity PAM

c. Single-Polarity PAM

PAM is a pulse modulation system in which the signal is sampled at regular intervals, and each sample is made proportional to the amplitude of the signal at the instant of sampling.

The pulses are then sent by either wire or cable, or else are used to modulate a carrier.

The two

types of PAM are shown in fig. above. The two types are Double-polarity PAM, and single-

polarity PAM. The largest pulse represents the greatest positive signal amplitude sampled, while

the smallest pulse represents the largest negative sample. The time duration of each pulse may be quite short, and the time interval between pulses may be relatively long. In single-polarity PAM,

in which a fixed dc level is added to the signal, to ensure that the pulses are always positive. The ability to use constant-amplitude pulses is a major advantage of pulse modulation and since PAM does not utilize constant amplitude pulses, it is infrequently used. When it is used, the pulses frequency modulate the carrier. If a radio frequency is pulse-amplitude modulated instead of simply being amplitude modulated, much less power is required for the transmission of information because the transmitter is actually switched off between pulses. This is one advantage of pulse modulation.

It is very easy to generate and demodulate PAM. In a generator the signal to be converted to PAM s fed to one input of an AND gate. Pulses at the sampling frequency are applied to the other input of the AND gate to open it during the wanted time intervals. The output of the gate then consists of pulses at the sampling rate, equal in amplitude to the signal voltage at each instant. The pulses are then passed through a pulse-shaping network, which gives them flat tops.

Frequency modulation is then employed, so that the system becomes PAM-FM.

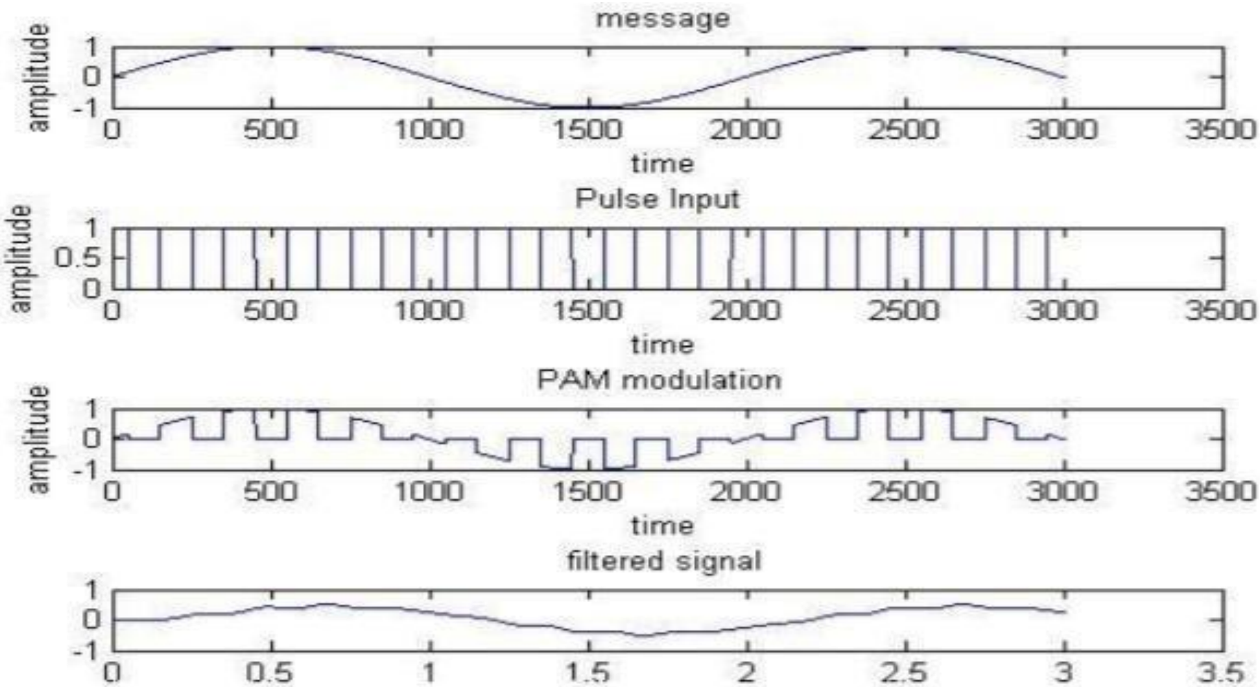
In the receiver, the pulses are first recovered with a standard FM de-modulator. They are then fed to an ordinary diode detector, which is followed by a low-pass filter. If the cutoff frequency of this filter is high enough to pass the highest signal frequency, but low enough to remove the Sampling frequency ripple, an undistorted replica of the original signal is reproduced.

Code:

```
% pulse amplitude
modulation close all

clear
all clc

t = 0 : 1/1e3 : 3; % 1 kHz sample freq for 1
sec d = 0 : 1/5 : 3;
```

```
x = sin(2*pi/4*2*t);

 %message signal figure;

plot(x);
title('message');
xlabel('time');
ylabel('amplitude')
;

y = pulstran(t,d,'rectpuls',0.1); %generation of pulse
input subplot(4,1,2)

plot(y);

title('Pulse Input ');

xlabel('time');
ylabel('amplitude');
z=x.*y; % PAM
output
subplot(4,1,3)

plot(z);

title('PAM modulation
'); xlabel('time');
ylabel('amplitude');

[den,
num]=butter(1,2*pi*0.5/1000);
s11=filter(den,num,z);
s12=filter(den,num,s11);
subplot(4,1,4)

plot(t,s12)
axis([0 3.5 -1
1]);

title('filtered signal')
```

OUTPUT:



message
amplitude
1
0
-1
0
500
1000
1500
2000
2500
3000
3500
time
Pulse Input
amplitude
1
0.5
0
0
500
1000
1500
2000
2500
3000
3500
time
PAM modulation
amplitude
1
0
-1
0
500
1000
1500
2000
2500
3000
3500
time
filtered signal
1
0
-1
0
0.5
1
1.5
2
2.5
3
3.5

Experiment-6

Aim:

To study the Pulse Width Modulation (PWM) and Demodulation process and record the corresponding waveforms.

Theory:

The Pulse-width modulation of PTM is also called as Pulse-duration modulation (PDM), or pulse length modulation (PLM). In this modulation, the pulses

have a constant amplitude and a variable time duration. The time duration (or width) of each pulse is proportional to the instantaneous amplitude of the modulating signal.

In this system, as shown in fig. below, we have a fixed amplitude and starting time of watch pulse, but the width of each pulse is made proportional to the amplitude of the signal at that instant.

In this case, the narrowest pulse represents the most negative sample of the original signal and the widest pulse represents the most positive sample.

When PDM is applied to radio transmission, the carrier frequency has constant amplitude, and the transmitter on time is carefully controlled in some

circumstances, PDM can be more accurate than PAM. One example of this is in magnetic tape recording, where pulse widths can be recorded and reproduced with less error than pulse amplitudes.

PWM or PPM are not used in telephony. To use PWM or PPM in such an application, we have to ensure that full scale modulation will not cause a pulse from one message signal to enter a time slot belonging to another message signal.

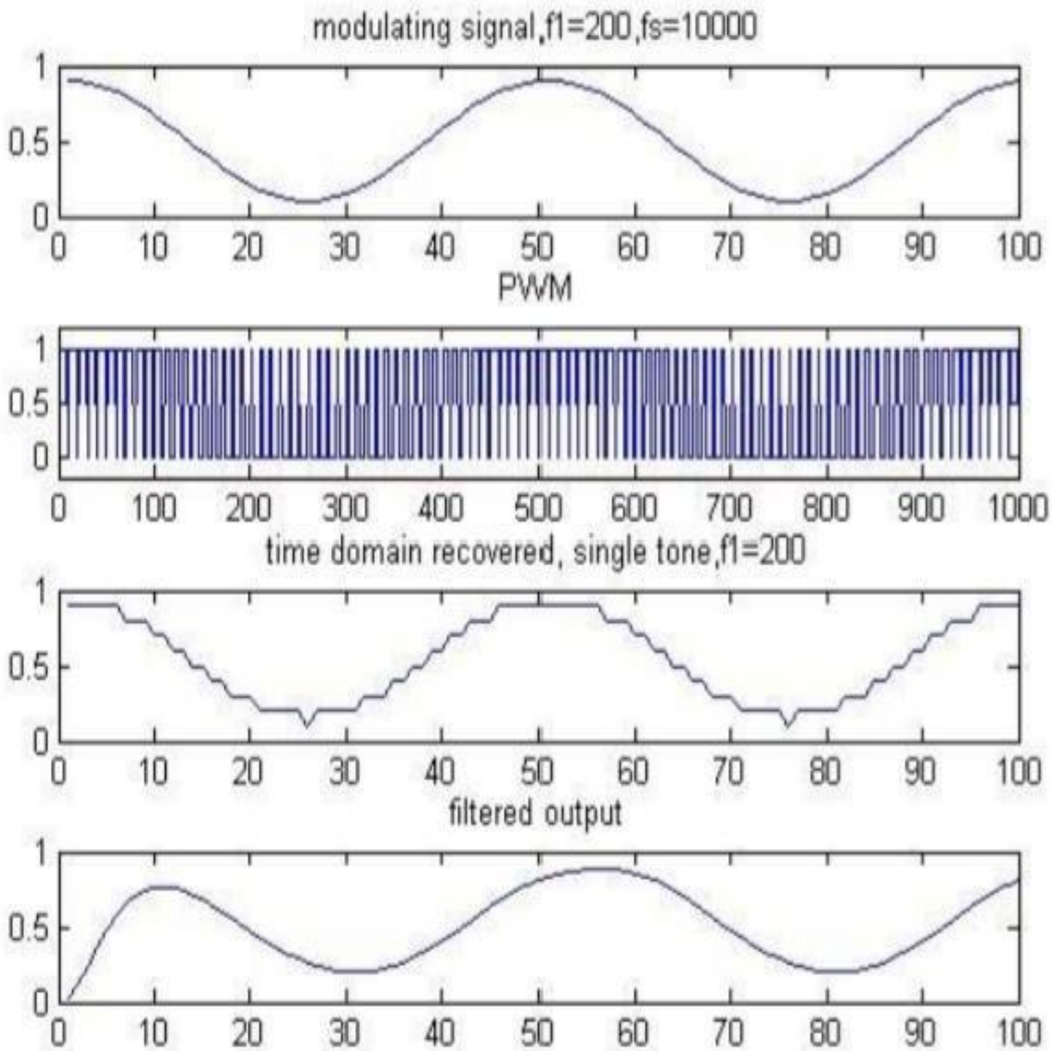
This restriction results in a wasteful use of time space in telephone systems that are characterized by high peak factors.

Code:

```
% pulse width modulation & demodulation
close all

clear all

clc fc=1000; fs=10000; f1=200;

t=0:1/fs:((2/f1)-(1/fs));
x1=0.4*cos(2*pi*f1*t)+0.5;

%modulation
y1=modulate(x1,fc,fs,'pwm');
subplot(411);

plot(x1);

axis([0 100 0 1]);

title('modulating
signal,f1=200,fs=10000') subplot(412);

plot(y1);

axis([0 1000 -0.2 1.2]);

title('PWM')

%demodulation
x1_recov=demod(y1,fc,fs,'pwm');
[den, num]=butter(1,2*pi*f1/fs);
s11=filter(den,num,x1_recov);
s12=filter(den,num,s11);
subplot(413);

plot(x1_recov);

title('time domain recovered, single tone,f1=200')
axis([0 100 0 1]);

subplot(414);
plot(s12);
```

```
title('filtered
output') axis([0 100
0 1]);
```

Output:



Experiment-7

Aim:

To study the functioning of Pre-Emphasis and De-Emphasis circuits.

Theory:

Frequency modulation is much more immune to noise than amplitude modulation and is significantly more immune than phase modulation. The threshold effect is more serious in FM as compared to AM, because in FM, the signal to noise ratio at the input of a detector, at which threshold effect starts, is higher. Lower the threshold level, better is the system because threshold can be avoided at a comparatively lower ratio, and a small signal is needed to avoid threshold for an equivalent noise power. Hence, it is desirable to lower the threshold level in the FM receivers. The process of lowering the threshold level is known as threshold improvement, or threshold reduction. Two methods are used for the improvement of the threshold.

Pre-Emphasis and De-Emphasis circuits.

FMFB (Frequency Modulation with Feed Back.)

PRE-EMPHASIS AND DE-EMPHASIS:

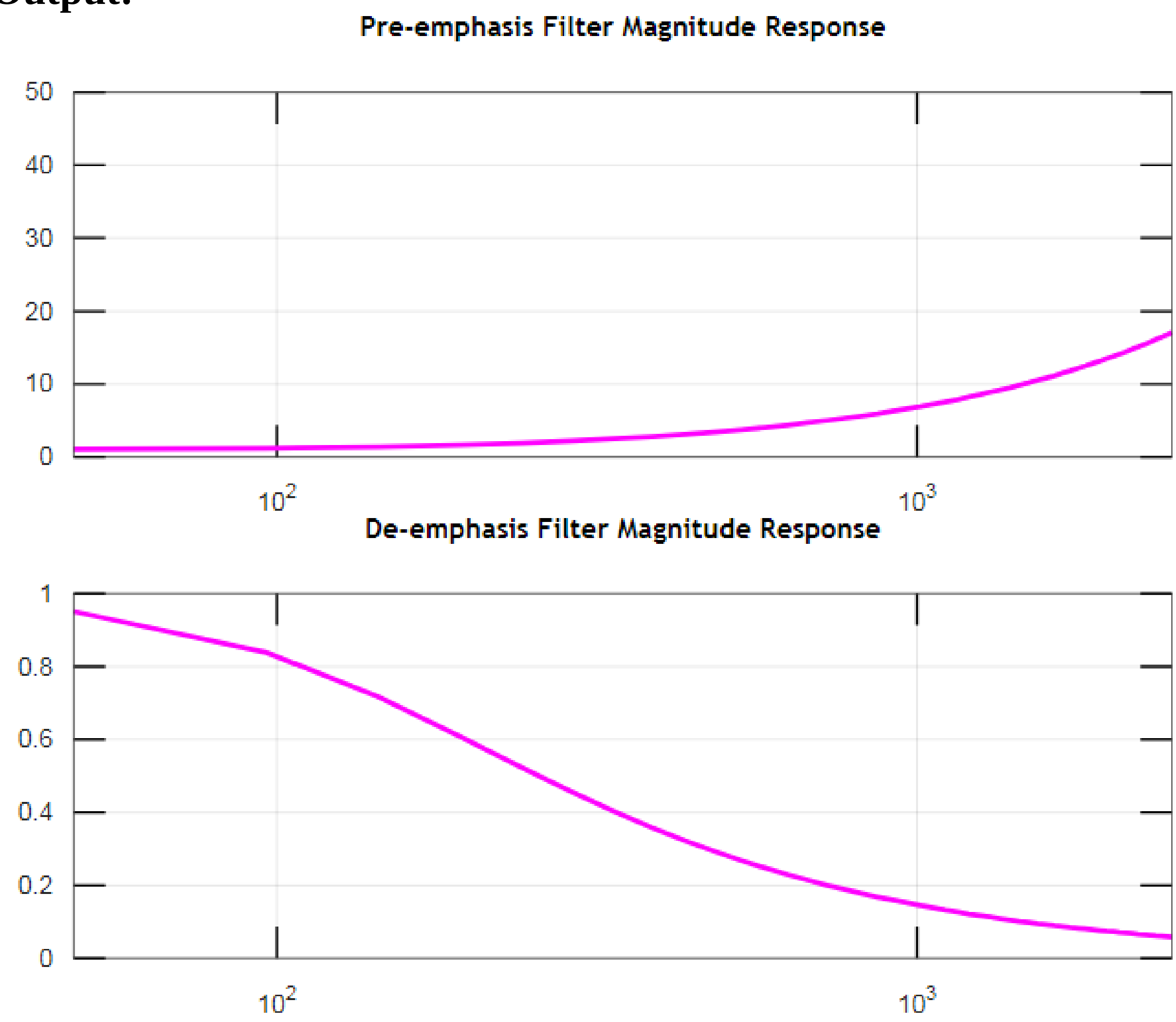
The noise triangle shows, noise has a greater effect on the higher modulating frequencies than on the lower ones. Thus, if the higher frequencies were artificially boosted at the transmitter and correspondingly cut at the receiver, an improvement in noise immunity could be expected, thereby increasing the signal-to-noise ratio. This boosting of the higher modulating frequencies, in accordance with a prearranged curve, is termed pre-emphasis, and the compensation at the receiver is called de-emphasis.

Code:

```
clc; close
all; clear
all;

num_samples = 2^13;
fs=5000;

Ts=1/fs;
fm1=20;
fm2=30;
fc=200;

t=(0:num_samples-1)*Ts;

f=(-num_samples/2:num_samples/2-1)*fs/num_samples; mt=sin(2*pi*fm1*t);

Mf=fftshift(abs(fft(mt)));
f_cutoff_pe=15;
Wn_pe=f_cutoff_pe/(fs/2);
[b_pe,a_pe]=butter(1,Wn_pe);
[H_pe,W]=freqz(a_pe,b_pe);
a_de=b_pe;

b_de=a_pe;
[H_de,W]=freqz(a_de,b_de);
mt_pe=filter(a_pe,b_pe,mt);
Mf_pe=fftshift(abs(fft(mt_pe)));
figure(1) subplot(211);plot(t,mt)

axis([0 .6 min(mt)-1 max(mt)+1])

grid on; title('Modulating Signal (Time Domain)')

subplot(212);plot(f,Mf)

grid on;axis([-50 50 0 max(Mf)+100])
title('Modulating Signal (Frequency Domain)')
figure(2)
```

```
subplot(211)
semilogx(W*pi*(fs/2),abs(H_pe),'m','linewidth',2) axis([0
fs/2 0 50])

grid on;title('Pre-emphasis Filter Magnitude Response')
subplot(212) semilogx(W*pi*(fs/2),abs(H_de),'m','linewidth',2)
axis([0 fs/2 0 1])

grid on;title('De-emphasis Filter Magnitude Response')
```

Output:



Experiment-8

Aim:

To study the Pulse Position Modulation (PPM) and demodulation process and record corresponding waveforms.

Theory:

Pulse position modulation (PPM) is more efficient than PAM or PDM for radio transmission. In PPM all pulses have the same constant amplitude and narrow pulse width. The position in time of the pulses is made to vary in proportion to the amplitude of the modulating signal

The simplest modulation process for pulse position modulation is a PDM system with the addition of a monostable multivibrator. The monostable is arranged so that it is triggered by the trailing edges of the PDM pulses. Thus, the monostable output is a series of constant-width, constant amplitude pulses which vary in position according to the original signal amplitude.

Code:

```
% pulse position modulation
close all

clear all

clc

fc=100;

fs=1000;

f1=80;

t=0:1/fs:((2/f1)-(1/fs));

x1=0.4*cos(2*pi*f1*t)+0.5;
```

```
%modulation
y1=modulate(x1,fc,fs,'ppm');
subplot(311);
plot(x1);
axis([0 15 0 1]);
title('modulating
signal,f1=80,fs=1000')
subplot(312);
plot(y1);
axis([0 250 -0.2 1.2]);
title('PPM')
%demodulation
x1_recov=demod(y1,fc,fs,'ppm');
subplot(313);
plot(x1_recov);
title('time domain recovered, single tone,f1=80')
axis([0 15 0 1]);
```

Output: