Communication Systems

Practical File

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EXP-1

Aim : Generation of DSB-SC AM Signal using Balance Modulator.

Theory :

Amplitude modulation: Modulation is a process of translating information signal from low band frequency to high band frequency that is suits the transmission medium. Information signal is usually of low frequency, so it cannot travel far. It needs a carrier signal of higher frequency for long distance destination. The inputs are carrier and information (modulating) signals while the output is called the modulated signal. Amplitude Modulation (AM) refers to the modulation technique where the carrier's amplitude is varied in accordance to the instantaneous value of the modulating or baseband signal's amplitude

Balanced modulator consists of two identical AM modulators. These two modulators are arranged in a balanced configuration in order to suppress the carrier signal. Hence, it is called as Balanced modulator.

The same carrier signal $c(t)=A_c\cos(2\pi f_c t)c(t)=A_c\cos(2\pi f_c t)$ is applied as one of the inputs to these two AM modulators. The modulating signal $m(t)m(t)$ is applied as another input to the upper AM modulator. Whereas, the modulating signal $m(t)m(t)$ with opposite polarity, i.e., $-m(t)-m(t)$ is applied as another input to the lower AM modulator.

Output of the upper AM modulator is

$$s_1(t)=A_c[1+k_a m(t)]\cos(2\pi f_c t)s_1(t)=A_c[1+k_a m(t)]\cos(2\pi f_c t)$$

Output of the lower AM modulator is

$$s_2(t)=A_c[1-k_a m(t)]\cos(2\pi f_c t)s_2(t)=A_c[1-k_a m(t)]\cos(2\pi f_c t)$$

We get the DSBSC wave $s(t)s(t)$ by subtracting $s_2(t)s_2(t)$ from $s_1(t)s_1(t)$. The summer block is used to perform this operation. $s_1(t)s_1(t)$ with positive sign and $s_2(t)s_2(t)$ with negative sign are applied as inputs to summer block. Thus, the summer block produces an output $s(t)s(t)$ which is the difference of $s_1(t)s_1(t)$ and $s_2(t)s_2(t)$.

$$\Rightarrow s(t)=A_c[1+k_a m(t)]\cos(2\pi f_c t)-A_c[1-k_a m(t)]\cos(2\pi f_c t)\Rightarrow s(t)=A_c[1+k_a m(t)]\cos(2\pi f_c t)-A_c[1-k_a m(t)]\cos(2\pi f_c t)$$

$$\Rightarrow s(t)=A_c\cos(2\pi f_c t)+A_c k_a m(t)\cos(2\pi f_c t)-A_c\cos(2\pi f_c t)+\Rightarrow s(t)=A_c\cos(2\pi f_c t)+A_c k_a m(t)\cos(2\pi f_c t)-A_c\cos(2\pi f_c t)+$$

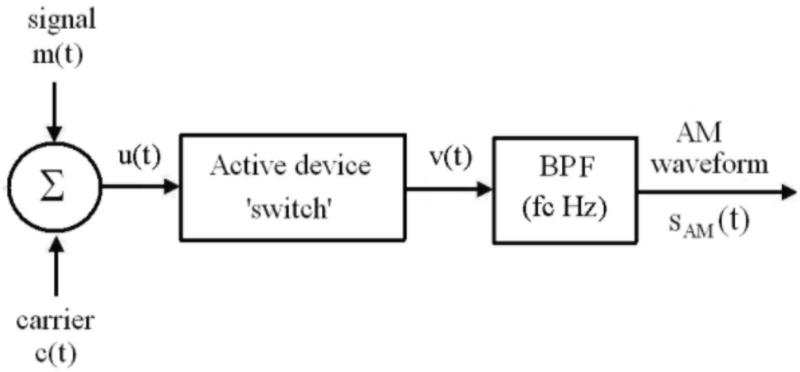
$A_c k_a m(t)\cos(2\pi f_c t)A_c k_a m(t)\cos(2\pi f_c t)$
$\Rightarrow s(t)=2A_c k_a m(t)\cos(2\pi f_c t)\Rightarrow s(t)=2A_c k_a m(t)\cos(2\pi f_c t)$

We know the standard equation of DSBSC wave is

$$s(t)=A_c m(t)\cos(2\pi f_c t)s(t)=A_c m(t)\cos(2\pi f_c t)$$

By comparing the output of summer block with the standard equation of DSBSC wave, we will get the scaling factor as $2k_a$

Diagram :



Source Code :

```
close all
clear all
clc
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 :

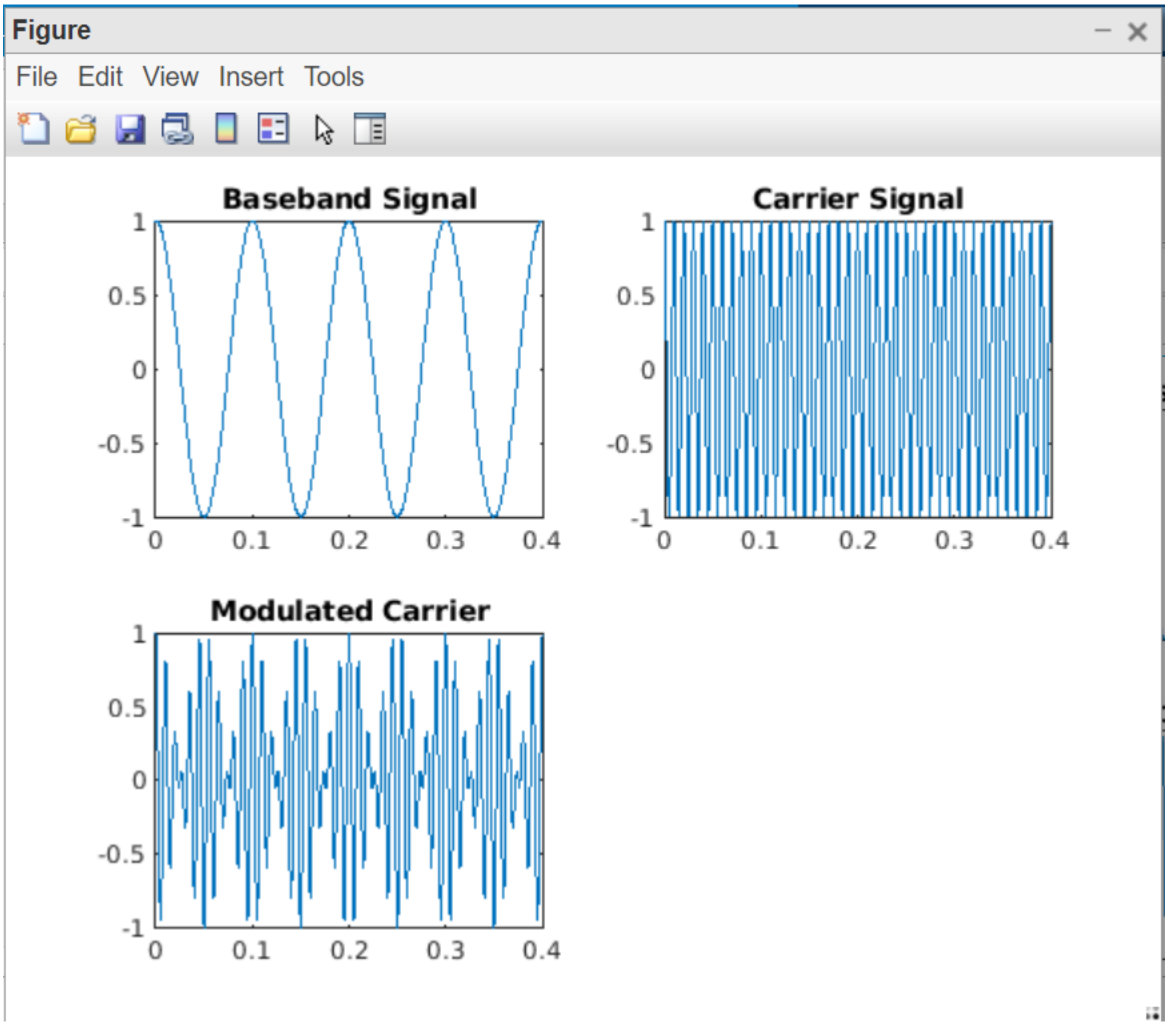


Figure
File Edit View Insert Tools
Baseband Signal
1
0.5
0
-0.5
-1
0 0.1 0.2 0.3 0.4
Carrier Signal
1
0.5
0
-0.5
-1
0 0.1 0.2 0.3 0.4
Modulated Carrier
1
0.5
0
-0.5
-1
0 0.1 0.2 0.3 0.4

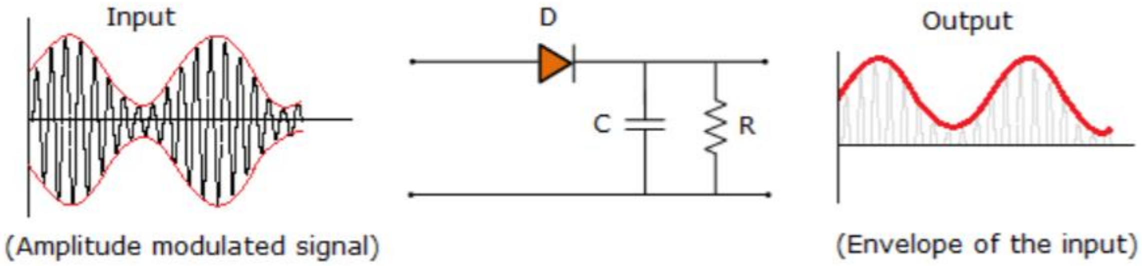
EXP-2

Aim: To Study Amplitude Demodulation by linear diode detector.

Theory :

Diode Detector: The function of the diode detector is to extract the audio signal from the signal at the output of the IF amplifiers. It is similar to a half wave rectifier i.e. converting an input to a DC output. Diode conducts every time the input signal applied to its anode is more positive than the voltage on the top of the capacitor. When the voltage falls below the capacitor voltage, the diode ceases to conduct and voltage across the capacitor leaks away until next time the input signal is able to switch it again. Operating principle: Let us assume that the operation is absent in the circuit, the detector output wave would be half rectified modulated signal. Now let us consider that the capacitor is introduced in the circuit. 1. For +ve half cycle: The diode conducts and capacitor is charged to peak value of carrier voltage. 2. For –ve half cycle: The diode doesn't conducts, that means i/p carrier voltage is disconnected from the R-C circuit therefore capacitor starts discharging through the resistor R with the time constant t =RC. If T is suitably chosen, the voltage across capacitor C will not fall during the small period of –ve half cycle and by that time the next +ve half cycle appears. This cycle again charges the capacitor voltage. Hence the output voltage across the capacitor C is a spiky modulating as baseband signal. This means that the voltage across the C is same as envelope of modulated carrier signal. These spikes are introduced because of charging and discharging of capacitor. Time constant RC should be large so that C discharges negligibly small.

Diagram :

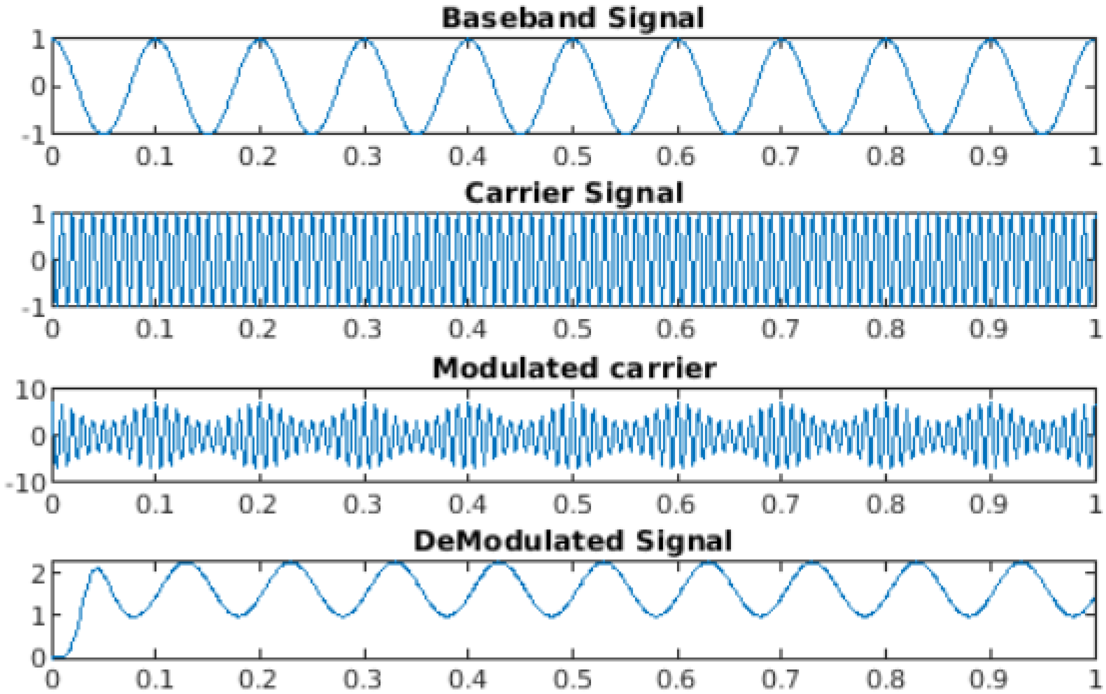


Source Code:

```
A=input('enter the carrier signal peek');
B=input('enter the baseband signal peek');
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,M13)
title('DeModulated Signal')
```

Output:



Baseband Signal
1
0
-1
0
0.1
0.2
0.3
0.4
0.5
0.6
0.7
0.8
0.9
1
Carrier Signal
1
0
-1
0
0.1
0.2
0.3
0.4
0.5
0.6
0.7
0.8
0.9
1
Modulated carrier
10
0
-10
0
0.1
0.2
0.3
0.4
0.5
0.6
0.7
0.8
0.9
1
DeModulated Signal
2
1
0
0
0.1
0.2
0.3
0.4
0.5
0.6
0.7
0.8
0.9
1

EXP-3

Aim: To Study the generation of SSB-AM signal.

Theory :

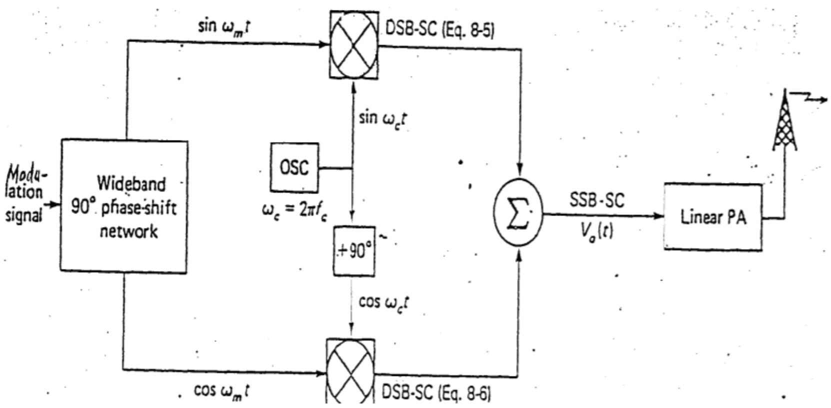
An SSB signal is produced by passing the DSB signal through a highly selective band pass filter. This filter selects either the upper or the lower sideband. Hence transmission bandwidth can be cut by half if one sideband is entirely suppressed. This leads to single-sideband modulation (SSB). In SSB modulation bandwidth saving is accompanied by a considerable increase in equipment complexity.

Single sideband, SSB modulation is basically a derivative of amplitude modulation, AM. By removing some of the components of the ordinary AM signal it is possible to significantly improve its efficiency.

It is possible to see how an AM signal can be improved by looking at the spectrum of the signal. When a steady state carrier is modulated with an audio signal, for example a tone of 1 kHz, then two smaller signals are seen at frequencies 1 kHz above and below the main carrier.

If the steady state tones are replaced with audio like that encountered with speech of music, these comprise many different frequencies and an audio spectrum with frequencies over a band of frequencies is seen.

Diagram :

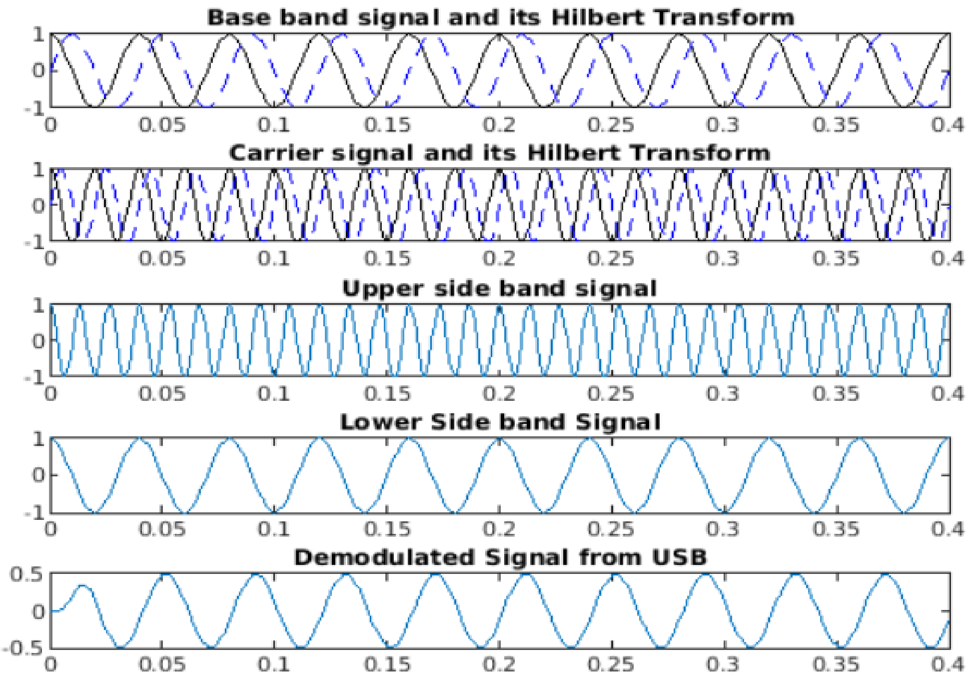


Source Code:

```
f1=input('enter the baseband 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:



EXP-4

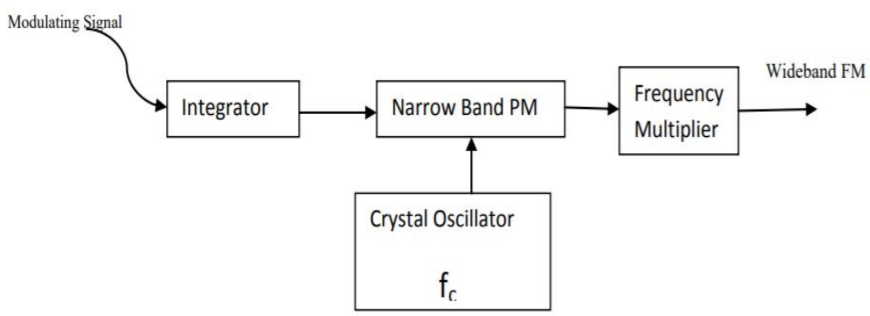
Aim: To Study the functioning of frequency modulation and demodulation and to calculate the modulating index.

Theory :

In frequency modulation instantaneous frequency ωi is varied linearly with a message or base band signal x(t) about an unmodulated carrier frequency ωc. That means the instantaneous value of the angular frequency ωi will be equal to the carrier frequency ωc plus a time varying component proportional to the baseband signal x(t). Instantaneous frequency is given by: ωi = ωc + kf .x(t) kf = frequency sensitivity General expression for FM wave: S(t) = A cos[ωct + kf ∫x(t) dt] 11

(a). Varactor Modulator : The variations in capacitance form part of the tuned circuit that is used to generate the FM signal to be transmitted. The tuned circuit sets the operating frequency of the oscillator and the varactor which is effectively in parallel with the tuned circuit. C1 is a DC blocking capacitor to provide DC isolation between the oscillator and the collector of the transmitter. L1 is an RF choke which allows the information signal to pass through the varactor but blocks the RF signal. (b). Reactance Modulator: As in the circuit diagram , the left hand side is the varactor modulator which generates the un-modulated carrier. The capacitor C and the resistor R are two components used for the phase shifting and together with the transistor for the voltage controlled capacitor. This voltage- controlled capacitor is actually in parallel with the tuned circuit

Diagram :

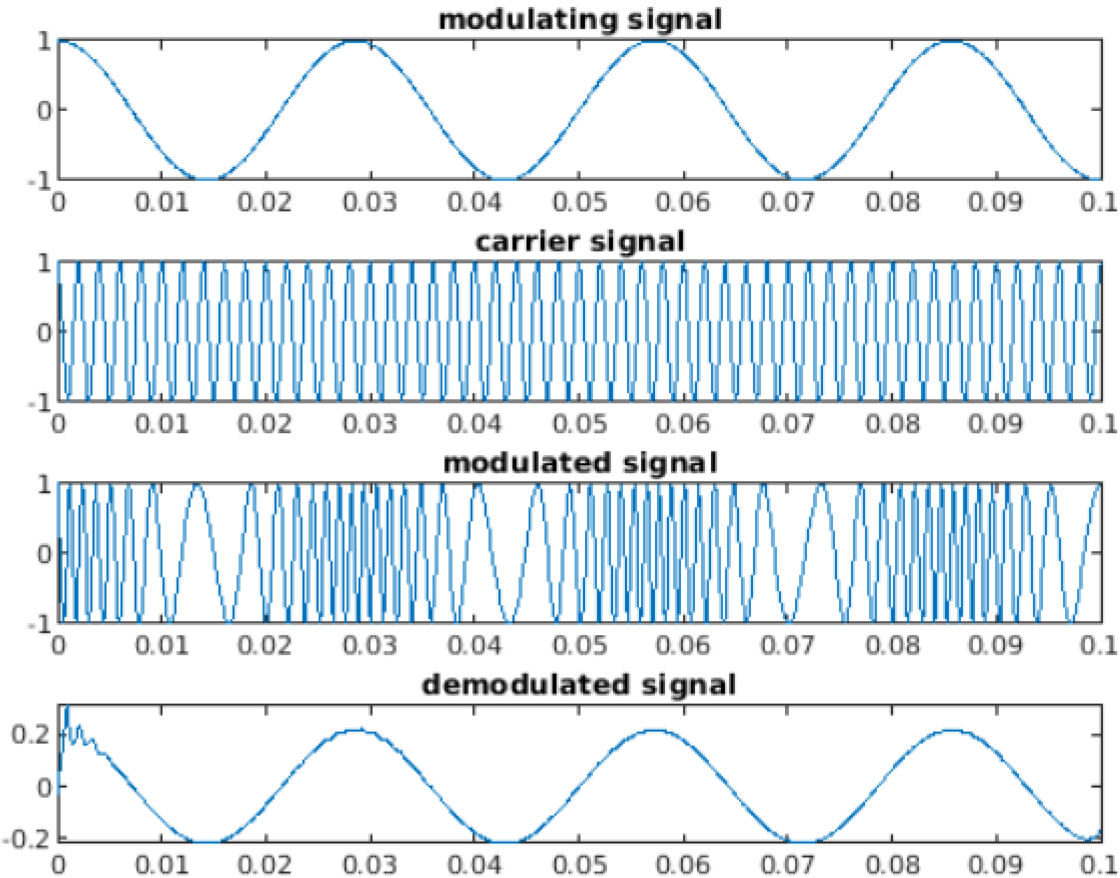


Source Code:

```
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:



EXP-5

Aim: To Study the functioning of frequency 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 comparitively lower ratio, and a small signal is needed to avoid threshold for an equivalent noise power. Hence, it is desirable to lower the thresholdlevel 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.

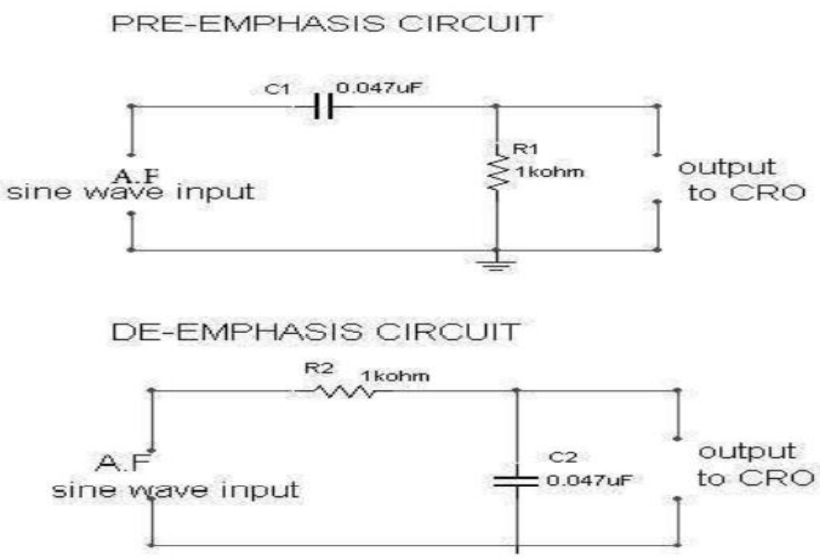
Pre-Emphasis and De- Emphasis circuits

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 is 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.

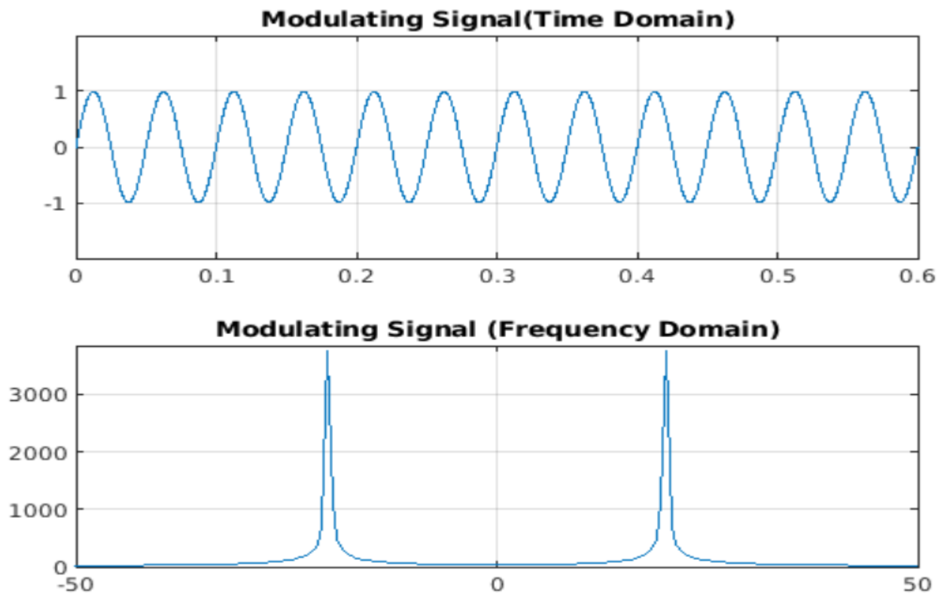
Diagram :

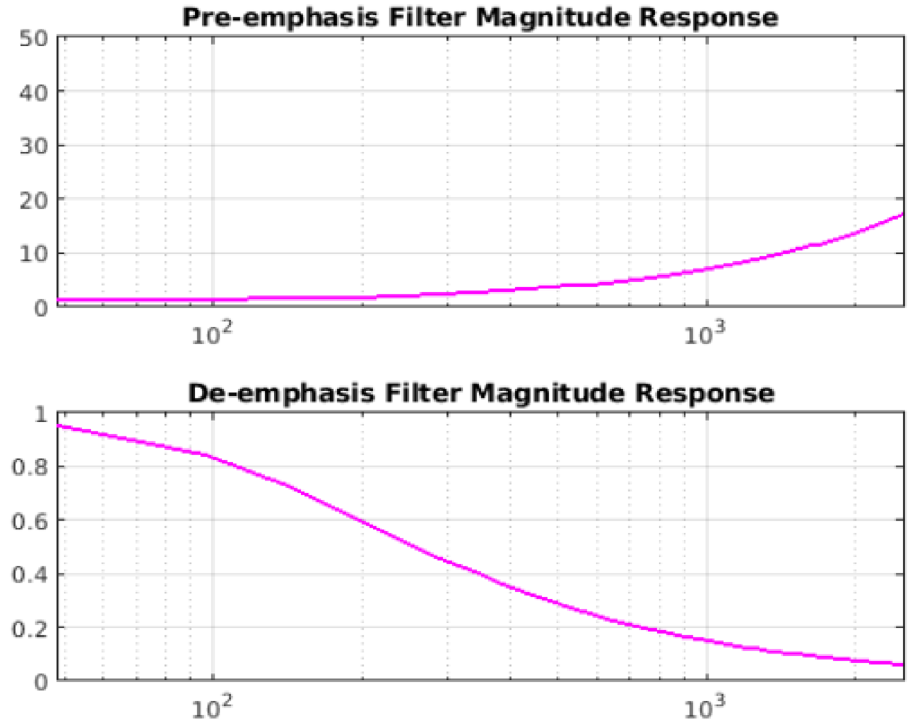


Source Code:

```
close all
clear all
clc
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:


Modulating Signal(Time Domain)
1
0
-1
0
0.1
0.2
0.3
0.4
0.5
0.6
Modulating Signal (Frequency Domain)
3000
2000
1000
0
-50
0
50


Pre-emphasis Filter Magnitude Response
50
40
30
20
10
0
10^2
10^3
De-emphasis Filter Magnitude Response
1
0.8
0.6
0.4
0.2
0
10^2
10^3

EXP-6

Aim: To Study the Pulse Amplitude Modulation And De-modulation and their waveforms.

Theory :

In PAM, amplitude of pulses of carrier pulse train is varied in accordance with the modulating signal. Fig. explains the principle of PAM. A signal i.e. baseband is shown in fig. and carrier pulse train f(t) is also shown. The frequency of carrier train is decided by sampling theorem. A pulse amplitude modulated signal fc(t) is shown. It can be seen that the amplitude of pulse depends upon the value of f(t) during the time of pulse.

Diagram :

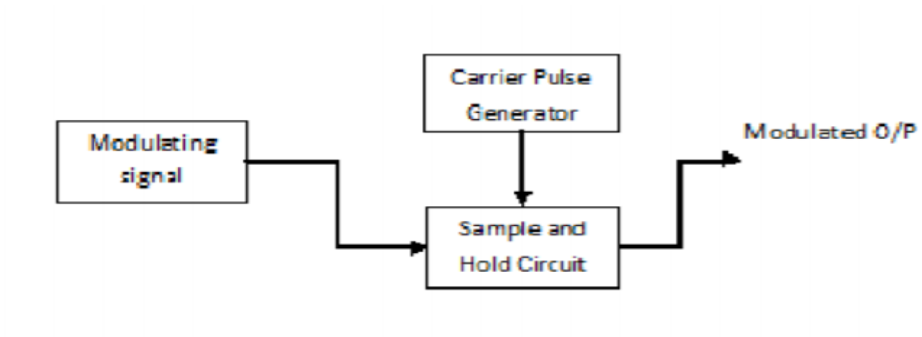


Fig.1: PAM Modulator

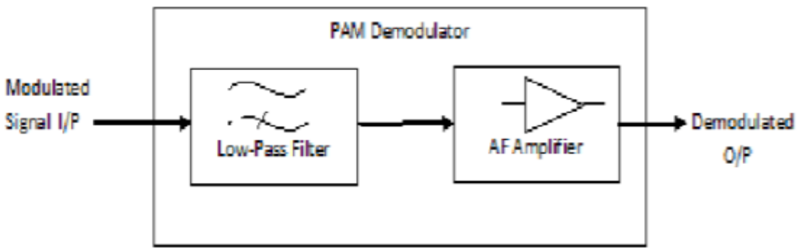
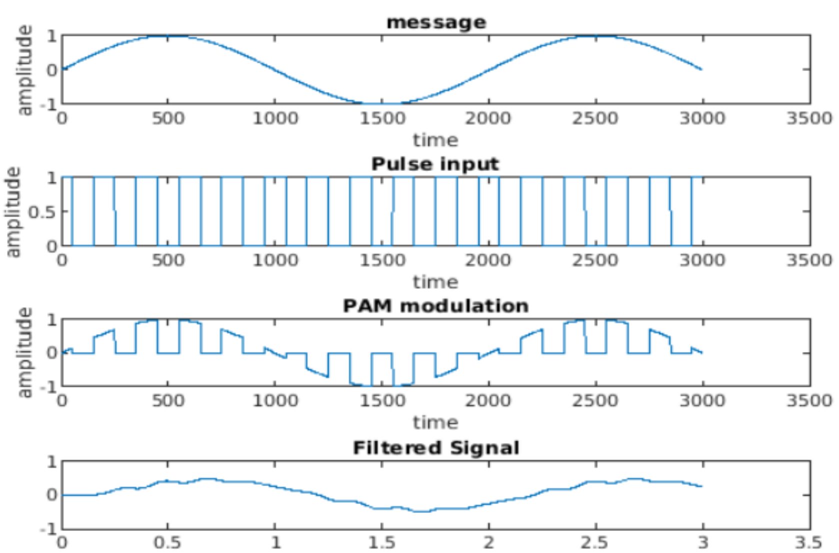


Fig.2: PAM De-Modulator

Source Code :

```
close all
clear all
clc
t = 0 : 1/1e3 : 3;
d = 0 : 1/5 : 3;
x= sin(2*pi/4*2*t);
figure;
subplot(4,1,1)
plot(x);
title('message');
xlabel('time');
ylabel('amplitude');
y = pulstran(t,d,'rectpuls',0.1);
subplot(4,1,2)
plot(y);
title('Pulse input');
xlabel('time');
ylabel('amplitude');
z=x.*y;
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 :



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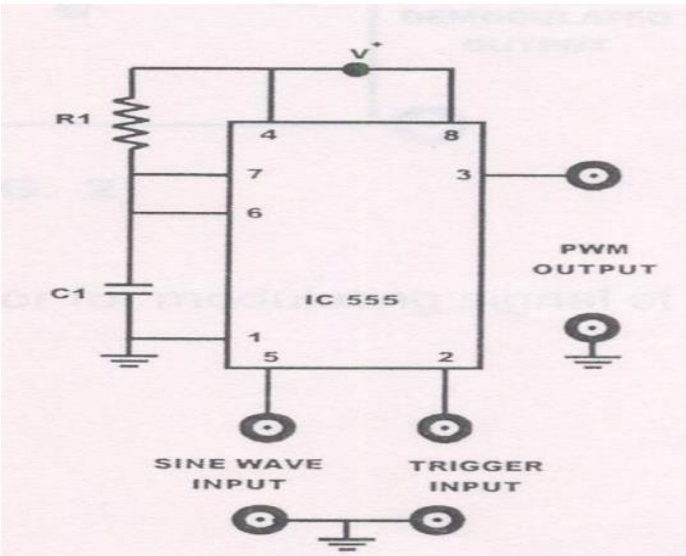
EXP-7

Aim: To Study the Pulse Width Amplitude Modulation And De-modulation and record their corresponding waveforms.

Theory :

The PWM is also known as pulse duration modulation. It modulates the time parameter of the pulses. The width of PWM pulses varies. The amplitude is constant; width of the pulse is proportional to the amplitude of the modulating signal. Bandwidth on transmission channel depends on rise time of the pulse. The demodulation circuit used is a simple filter circuit that demodulator the PWM signal and gives the original message input.

Diagram :

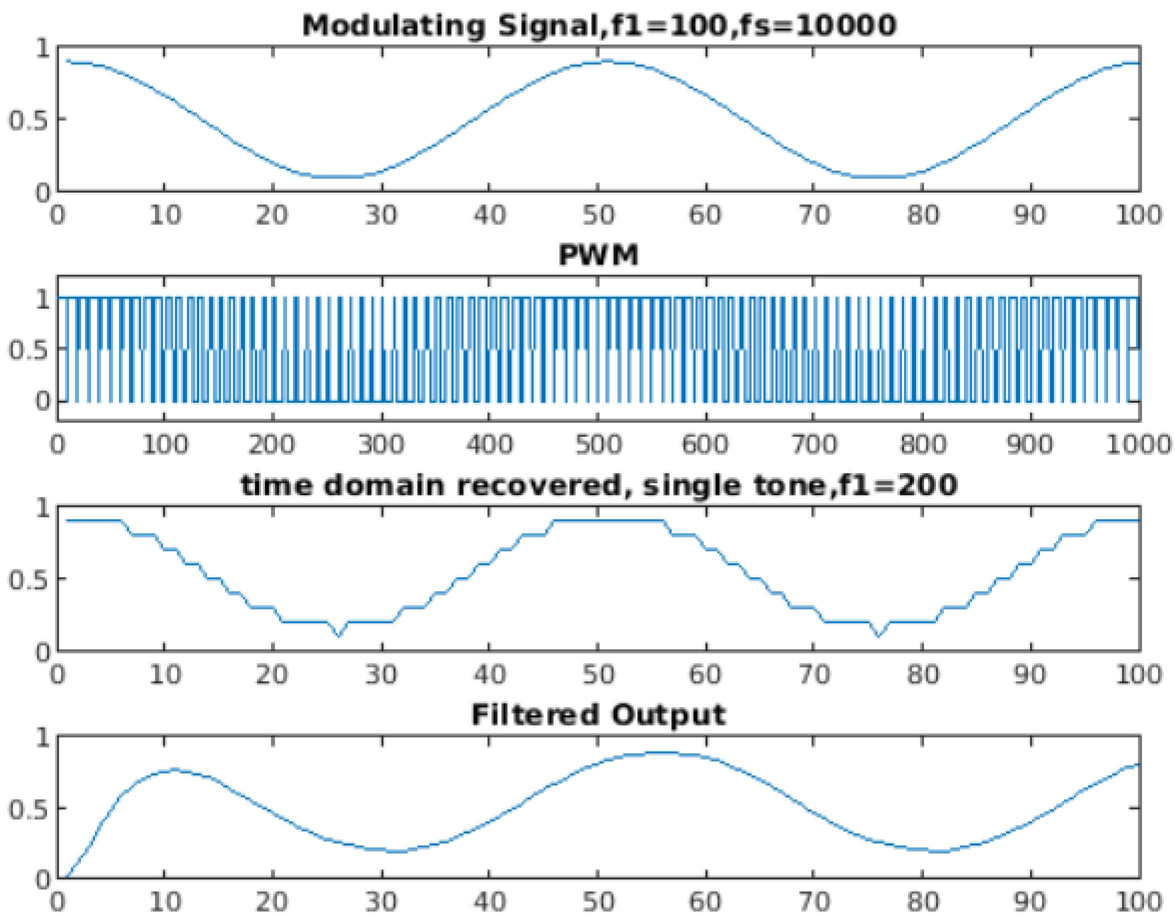


Source Code :

```
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;
y1=modulate(x1,fc,fs,'pwm');
subplot(4,1,1)
plot(x1)
axis([0 100 0 1]);
title('Modulating Signal,f1=100,fs=10000')
```

```
subplot(4,1,2)
plot(y1)
axis([0 1000 -0.2 1.2]);
title('PWM')
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(4,1,3)
plot(x1_recov)
title('time domain recovered, single tone,f1=200')
axis([0 100 0 1]);
subplot(4,1,4)
plot(s12)
title('Filtered Output')
axis([0 100 0 1]);
```

Output :



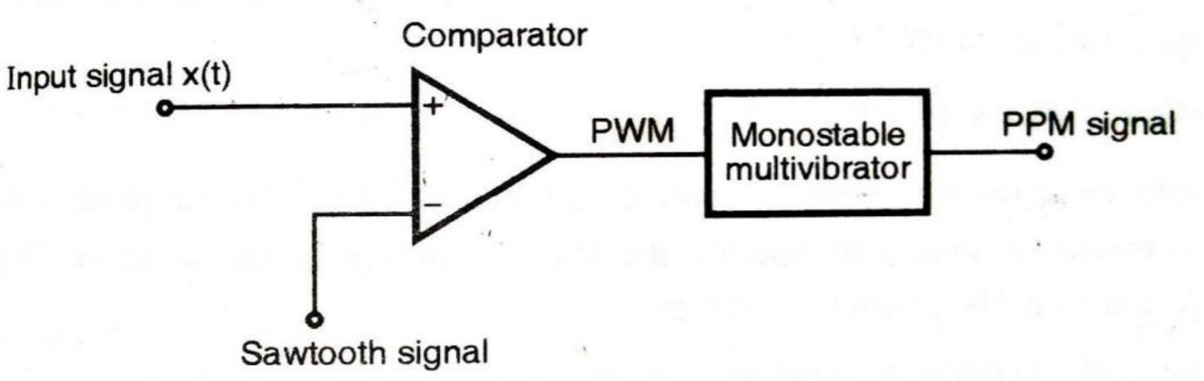
EXP-8

Aim: To Study the Pulse Position Modulation And Demodulation and record their corresponding waveforms.

Theory :

In Pulse Position Modulation, both the pulse amplitude and pulse duration are held constant but the position of the pulse is varied in proportional to the sampled values of the message signal. Pulse time modulation is a class of signaling techniques that encodes the sample values of an analog signal on to the time axis of a digital signal and it is analogous to angle modulation techniques. The two main types of PTM are PWM and PPM. In PPM the analog sample value determines the position of a narrow pulse relative to the clocking time. In PPM rise time of pulse decides the channel bandwidth. It has low noise interference.

Diagram :



Source Code :

```
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 :