ANALOG COMMUNICATION LAB

LABORATORY MANUAL

DEPARTMENT OF ELECTRONICS AND COMMUNICATIONS ENGG

MALLA REDDY COLLEGE OF ENGINEERING AND TECHNOLOGY
(Sponsored by CMR Educational Society)
(Affiliated to JNTU, Hyderabad)
Secunderabad-14.

CONTENTS

S.NO	NAME OF THE EXPERIMENT	PAGE NO
1.	Amplitude modulation and demodulation	1
2.	DSB-SC Modulator & Detector	9
3.	SSB-Sc Modulator & Detector (Phase Shift Method)	13
4.	Frequency modulation and demodulation.	19
5.	Spectrum analyzer and analysis of AM and FM signals	23
6.	Pre-emphasis & De-emphasis.	26
7.	Frequency Division Multiplexing & De multiplexing	32
8.	Verification of Sampling Theorem	36
9.	Pulse Amplitude Modulation & Demodulation	44
10.	Pulse Width Modulation & Demodulation	51
11.	Pulse Position Modulation & Demodulation	57
12.	Frequency Synthesizer	63
13.	AGC Characteristics	67
14.	PLL as FM Demodulator	72

EXPERIMENT NO-1 DATE:

AMPLITUDE MODULATION &DEMODULATION

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

APPARATUS :

1. Amplitude Modulation & De modulation trainer kit.
2. C.R.O (20MHz)
3. Function generator (1MHz).
4. Connecting cords & probes.
5. PC with windows(95/98/XP/NT/2000)
6. MATLAB Software with communication toolbox

THEORY:

Modulation is defined as the process of changing the characteristics (Amplitude, Frequency or Phase) of the carrier signal (high frequency signal) in accordance with the intensity of the message signal (modulating signal).

Amplitude modulation is defined as a system of modulation in which the amplitude of the carrier is varied in accordance with amplitude of the message signal (modulating signal).

The message signal is given by the expression.

Em(t) =Em cosWmt

Where Wm is -----> Angular frequency

Em --------→ Amplitude

Carrier voltage Ec(t)= Ec cosWct

E(t)=Ec + KaEm cosWmt

KaEm cosWmt -----→ change in carrier amplitude

Ka-----→ constant

The amplitude modulated voltage is given by

E=E(t) cosWct

From above two equations

E= (Ec+KaEm cosWmt) cosWct.

E= (1+KaEm/Ec cosWmt) Ec cosWct

E= Ec(1+Ma cosWmt)cosWct

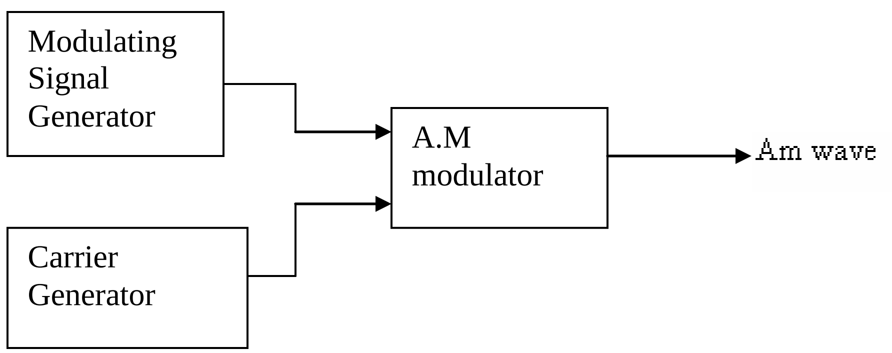
Where Ma-----→ depth of modulation/ modulation index/modulation factor

Ma=KaEm/Ec

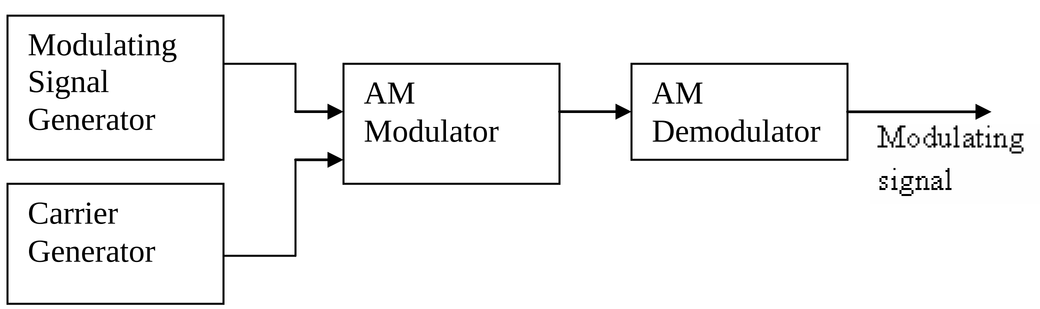
100* Ma gives the percentage of modulation.

BLOCK DIAGRAM:

Modulation



Demodulation



PROGRAM:

% program for AM modulation and demodulation

close all

clear all

```
clc
fs=8000;
fm=20;
fc=500;
Am=1;
Ac=1;
t=[0:0.1*fs]/fs;
m=Am*cos(2*pi*fm*t);
c=Ac*cos(2*pi*fc*t);
ka=0.5;
u=ka*Am;
s1=Ac*(1+u*cos(2*pi*fm*t)).*cos(2*pi*fc*t);
subplot(4,3,1:3);
plot(t,m);
title('Modulating or Message signal(fm=20Hz)');
subplot(4,3,4:6);
plot(t,c);
title('Carrier signal(fc=500Hz)');
subplot(4,3,7);
plot(t,s1);
title('Under Modulated signal(ka.Am=0.5)');
Am=2;
ka=0.5;
u=ka*Am;
s2=Ac*(1+u*cos(2*pi*fm*t)).*cos(2*pi*fc*t);
subplot(4,3,8);
plot(t,s2);
title('Exact Modulated signal(ka.Am=1)');
Am=5;
ka=0.5;
u=ka*Am;
```

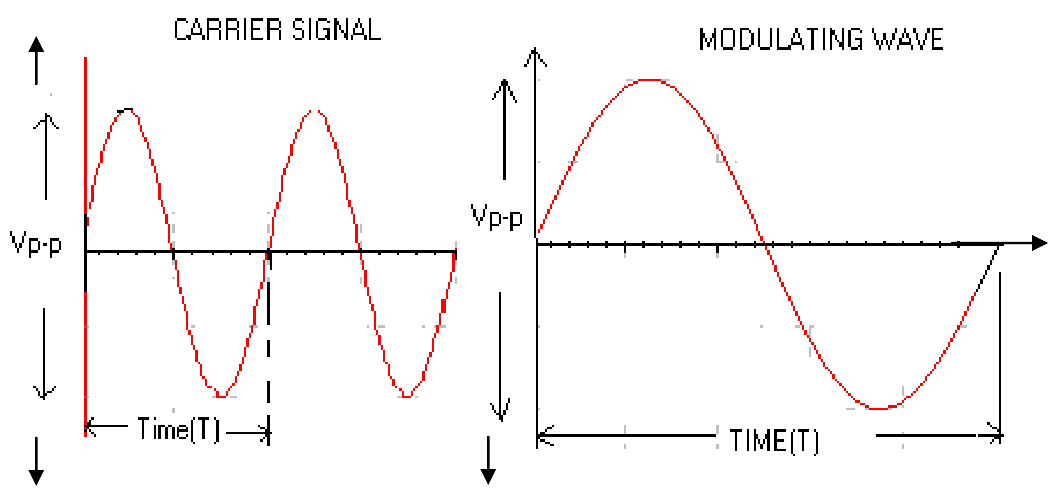
```
s3=Ac*(1+u*cos(2*pi*fm*t)).*cos(2*pi*fc*t);
subplot(4,3,9);
plot(t,s3);
title('Over Modulated signal(ka.Am=2.5)');
r1= s1.*c;
[b a] = butter(1,0.01);
mr1= filter(b,a,r1);
subplot(4,3,10);
plot(t,mr1);
title(' deModulated signal for(ka.Am=0.5)');
r2= s2.*c;
[b a] = butter(1,0.01);
mr2= filter(b,a,r2);
subplot(4,3,11);
plot(t,mr2);
title(' deModulated signal for(ka.Am=1)');
r3= s3.*c;
[b a] = butter(1,0.01);
mr3= filter(b,a,r3);
subplot(4,3,12);
plot(t,mr3);
title(' deModulated signal for(ka.Am=2.5)');
```

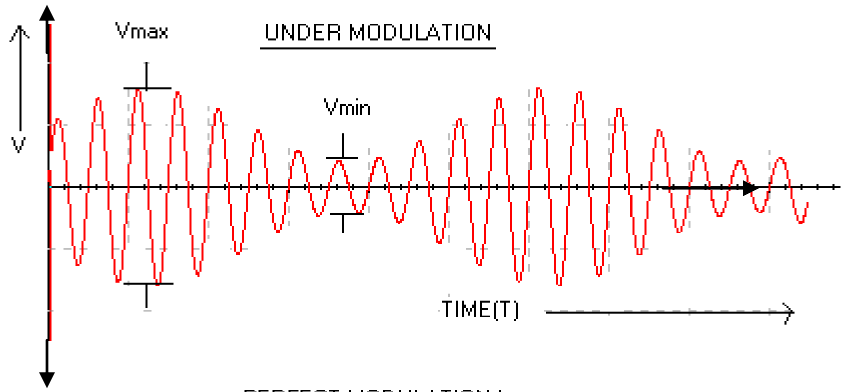
PROCEDURE:-

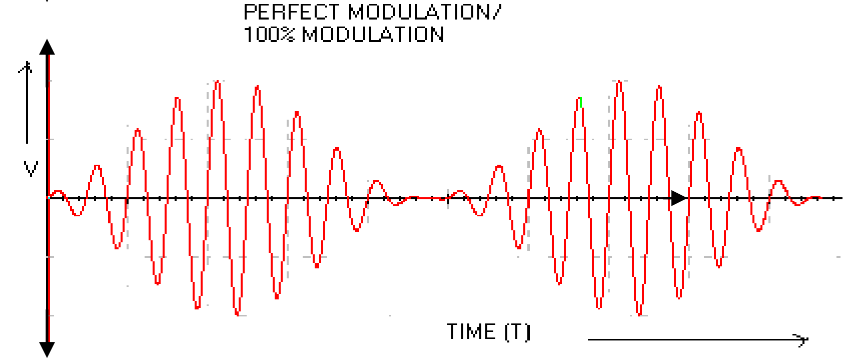
1. Connect the AC Adapter to the mains and the other side to the Experimental Trainer. Switch ‘ON’ the power.
2. Observe the carrier and modulating waveforms and note their frequencies. (Carrier frequency is around 100 KHz and amplitude is variable from 0 -8Vp-p, modulating signal is 1KHz).
3. Connect the carrier and modulating signals to the modulator circuit.
4. Observe the amplitude modulated wave.

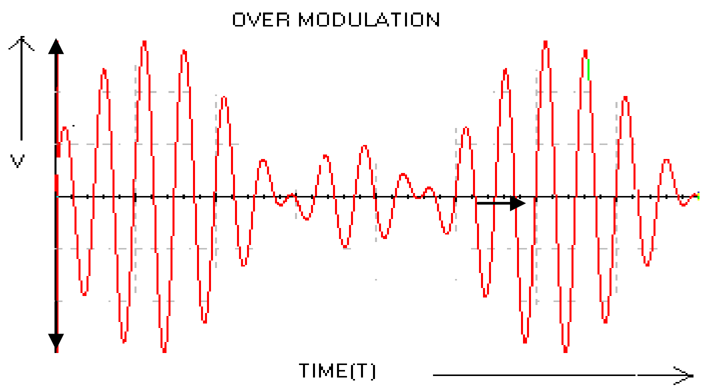
5. Connect Carrier I/P to ground and apply a 2V peak to peak AF Signal input to (modulating I/P) and adjust P1 in anti-clock wise position to get minimum A.C output.
6. Connect modulating I/P to ground and apply a 3V peak to peak carrier signal to carrier I/P and adjust P2 in clock wise direction to get minimum A.C ouyput..
7. Connect modulating input &carrier input to ground and adjust P3 for zero D.C output.
8. Make modulating i/p 2 Vpp and carrier i/p 3 Vpp peak to peak and adjust potentiometer P4 for maximum output.
9. Calculate maximum and minimum points on the modulated envelope on a CRO and calculate the depth of modulation.
10. Observe that by varying the modulating voltage, the depth of modulation varies.
11. During demodulation connect this AM output to the input of the demodulator.
12. By adjusting the RC time constant (i.e., cut off frequency) of the filter circuit we get minimum distorted output.
13. Observe that this demodulated output is amplified has some phase delay because of RC components.
14. Also observe the effects by changing the carrier amplitudes.
15. In all cases, calculate the modulation index.

<u>EXPECTED WAVEFORMS</u>:-

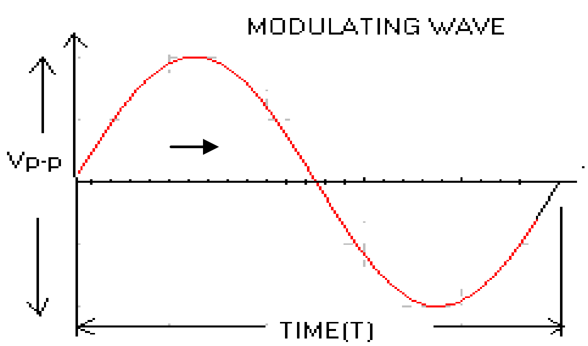








Demodulated signal



OBSERVATIONS:

Modulation

	Vc (V)	Vm (V)	Vmax (V)	Vmin (V)	m=(Vmax-Vmin)/ (Vmax+Vmin)	m= Vm/Vc
Under modulation						
Perfect modulation						
Over modulation						

Demodulation

Modulating signal Frequency	Demodulated output signal frequency

RESULT:

QUESTIONS

1. Define AM and draw its spectrum?
2. Draw the phase's representation of an amplitude modulated wave?
3. Give the significance of modulation index?
4. What are the different degrees of modulation?
5. What are the limitations of square law modulator?
6. Compare linear and nonlinear modulators?
7. Compare base modulation and emitter modulation?
8. Explain how AM wave is detected?
9. Define detection process?
10. What are the different types of distortions that occur in an envelop detector? How can they be eliminated?
11. What is the condition of for over modulation?

12. Define modulation & demodulation?
13. What are the different types of linear modulation techniques?
14. Explain the working of carrier wave generator.
15. Explain the working of modulator circuit.

EXPERIMENT NO-2 DATE:

DSB-SC MODULATOR & DETECTOR

AIM: To study the working of the Balanced Modulator and demodulator.

APPARATUS:

1. Balanced modulator trainer kit
2. C.R.O (20MHz)
3. Connecting cords and probes
4. Function generator (1MHz)
5. PC with windows (95/98/XP/NT/2000)
6. MATLAB Software with communication toolbox

THEORY:

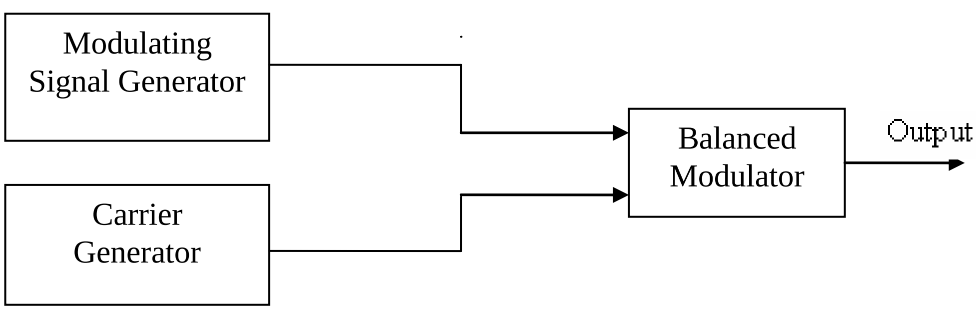
Balanced modulator circuit is used to generate only the two side bands DSB-SC. The balanced modulation system is a system is a system of adding message to carrier wave frequency there by only the side bands are produced. It consists of two AM modulators arranged in a balanced configuration. The AM modulator is assumed to be identical. The carrier input to the two modulators is same.

If we eliminate or suppress the carrier then the system becomes suppressed carrier DSB-SC. In this we need reinsert the carrier is complicated and costly. Hence the suppressed carrier DSB system may be used in point to point communication system.

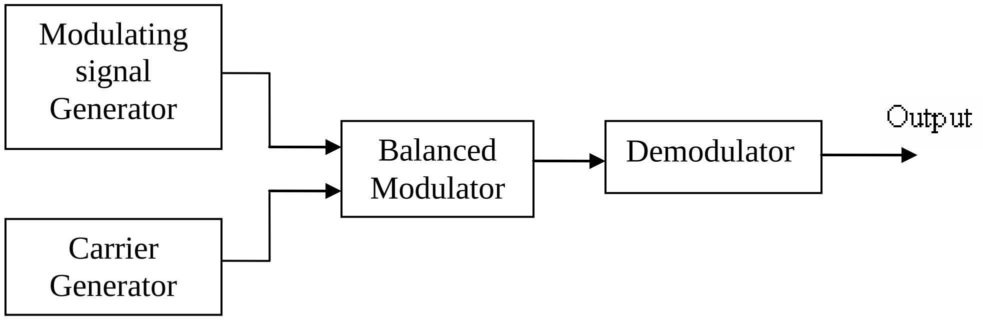
Generation of suppressed carrier amplitude modulated volt balanced modulator may be of the following types.

1. Using transistors or FET.
2. Using Diodes

BLOCK DIAGRAM: Modulation



Demodulation



PROGRAM:

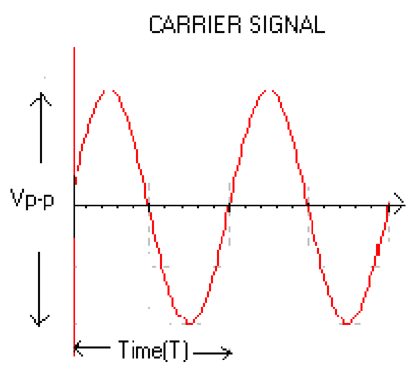
```
% program for dsbsc modulation and demodulation
close all
clear all
clc
t =0:0.000001:.001;
Vm= 1;
Vc= 1;
fm = 2000;
fc= 50000;
m_t = Vm*sin(2*pi*fm*t);
subplot(4,1,1);
plot(t,m_t);
c_t = Vc*sin(2*pi*fc*t);
subplot(4,1,2);
plot(t,c_t);
subplot(4,1,3);
s_t = m_t.*c_t;
hold on;
plot(t,s_t);
plot(t,m_t,'r:');
plot(t,-m_t,'r:');
hold off;
```

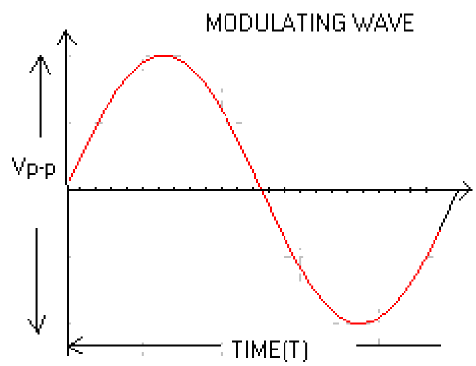
```
r = s_t.*c_t;
[b a] = butter(1,0.01);
mr= filter(b,a,r);
subplot(4,1,4);
plot(t,mr);
```

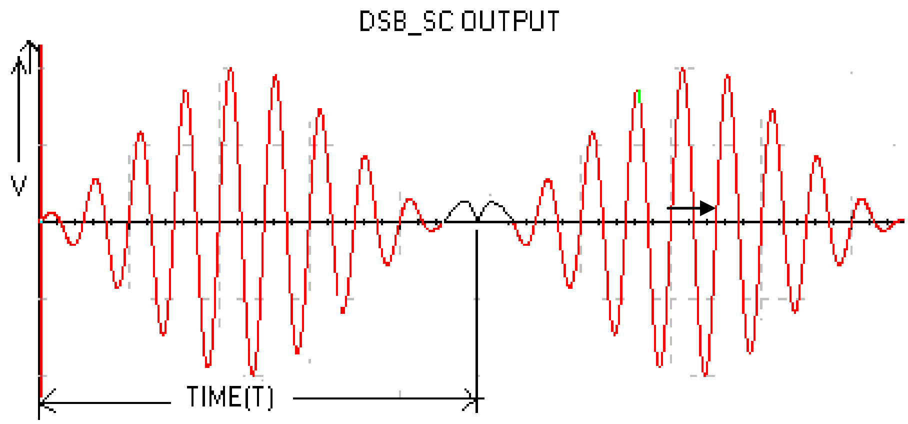
PROCEDURE:-

1. Connect the circuit as per the given circuit diagram.
2. Switch on the power to the trainer kit.
3. Apply a 100KHz, 0.1 peak sinusoidal to the carrier input and a 5KHz, 0.1 peak sinusoidal to the modulation input.
4. Measure the output signal frequency and amplitude by connecting the output to CRO.
5. And note down the output signals.

EXPECTED WAVEFORMS:-







OBSERVATIONS:

Carrier Signal		Message signal		Modulated signal output		Demodulated Signal output	
Fc(Hz)	Vc(volts)	Fm(Hz)	Vm(v)	Fo(Hz)	Vo(v)	F(Hz)	V(v)

RESULT:

QUESTIONS

1. What are the two ways of generating DSB_SC?
2. What are the applications of balanced modulator?
3. What are the advantages of suppressing the carrier?
4. What are the advantages of balanced modulator?
5. What are the advantages of Ring modulator?
6. Write the expression for the output voltage of a balanced modulator?
7. Explain the working of balanced modulator and Ring Modulator using diodes.

EXPERIMENT.NO-3 **DATE:**

SSB-SC MODULATOR & DETECTOR

(PHASE SHIFT METHOD)

AIM:- To generate SSB using phase method and detection of SSB signal using Synchronous detector.

APPARATUS:-

1. SSB trainer kit
2. C.R.O (20MHz)
3. Patch cards
4. CRO probes

THEORY:

AM and DSBSC modulation are wasteful of band width because they both require a transmission bandwidth which is equal to twice the message bandwidth In SSB only one side band and the carrier is used. The other side band is suppressed at the transmitter, but no information is lost. Thus the communication channel needs to provide the same band width, when only one side band is transmitted. So the modulation system is referred to as SSB system.

The base band signal may not be recovered from a SSB signal by the Use of a diode modulator. The bae band signal can be recovered if the spectral component of the output i.e either the LSB or USB is multiplied by the carrier signal.

Consider the modulating signal

$$M(t)=A_m \cos W_{mt}$$

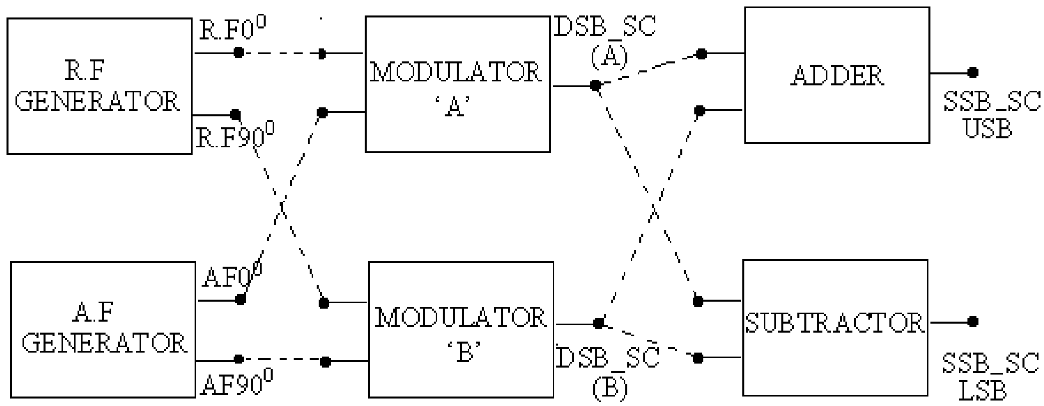
$$C(t)=A_c \cos W_{ct}$$

$$M(t)c(t)= A_c A_m \cos W_{mt} \cos W_{ct}$$

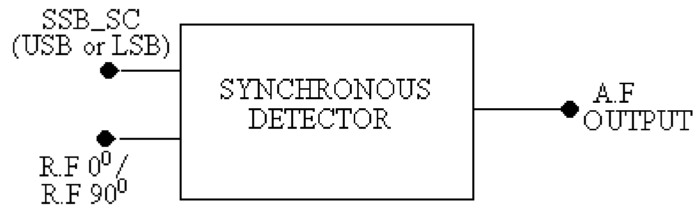
The above signal when passed through a filter, only one of the above component is obtained which lays the SSB signal.

BLOCK DIAGRAM: -

SSB MODULATION



SSB DEMODULATION/SYNCHRONOUS DETECTOR



PROGRAM:-

```
% program for ssb modulation and demodulation
close all
clear all
clc
fs=8000;
fm=20;
fc=50;
Am=1;
Ac=1;
t=[0:0.1*fs]/fs;
subplot(5,1,1);
m1=Am*cos(2*pi*fm*t);
plot(t,m1);
title('Message Signal');
```

```
m2=Am*sin(2*pi*fm*t);
subplot(5,1,2)
c1=Ac*cos(2*pi*fc*t);
plot(t,c1)
title('Carrier Signal');
c2=Ac*sin(2*pi*fc*t);
subplot(5,1,3)
% Susb=0.5* Am*cos(2*pi*fm*t).* Ac*cos(2*pi*fc*t) -- 0.5* Am*sin(2*pi*fm*t).* Ac*sin(2*pi*fc*t);
Susb=0.5*m1.*c1-0.5*m2.*c2;
plot(t,Susb);
title('SSB-SC Signal with USB');
subplot(5,1,4);
Slsb=0.5*m1.*c1+0.5*m2.*c2;
plot(t,Slsb);
title('SSB-SC Signal with LSB');
r = Susb.*c1;
subplot(5,1,5);
[b a] = butter(1,0.0001);
mr= filter(b,a,r);
plot(t,mr);
title('demodulated output');
```

PROCEDURE:-

SSB MODULATION

1. Connect the Adaptor to the mains and the other side to the Experimental Trainer Switch 'ON' the power.

2. (a) Connect carrier f_c 90^0 to A_{in} of Balanced Modulator –A and adjust its amplitude to 0.1Vpp.

(b). Connect modulating signal f_m 0^0 5Vpp to B_{in} of the Balanced Modulator-A.

3. Observe the DSB-A output on CRO.

4. Connect fc 0^0 at 0.1 Vpp at C_{in} of Balanced Modulator B. Connect fm 90^0 at 5 Vpp at D_{in} of Balanced Modulator B.

5. Connect the DSB-A output and DSB-B output to the summing amplifier. Observe the output (SSB output) on the spectrum analyzer. This gives single side band (upper) only while the lower side band is cancelled in the summing Amplifier.

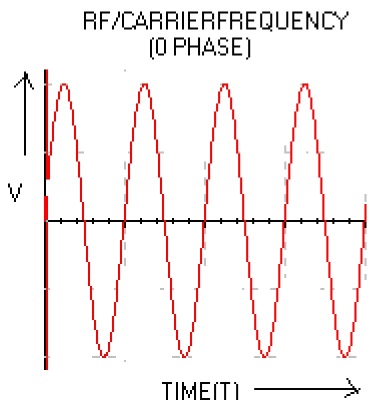
SSB DEMODULATION

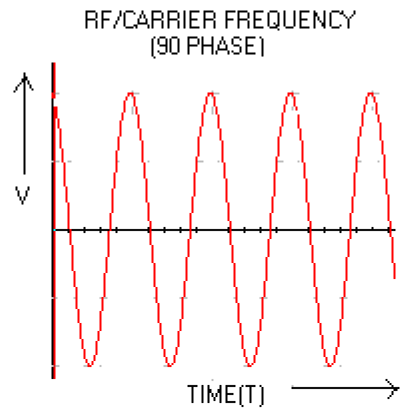
1. Connect the carrier fc 0^0 and SSB output to the synchronous detector.
2. Connect the demodulator output on the oscilloscope which is the recovered modulating signal.

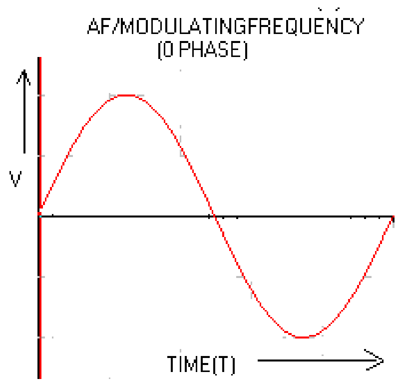
OBSERVATIONS:

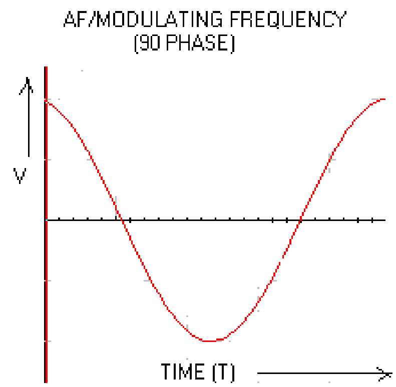
Carrier signal		Modulating signal		Balanced modulator-A		Balanced modulator-B		Adder/ Subtractor Output		Synchronous detector	
Fc	**Vc**	**Fm**	**Vm**	**Vmax**	**Vmin**	**Vmax**	**Vmin**	**Vmax**	**Vmin**	**Fd**	**Vd**

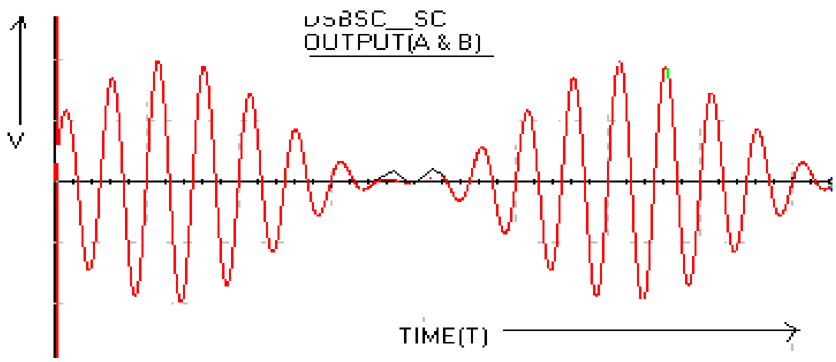
EXPECTED WAVE FORMS: -

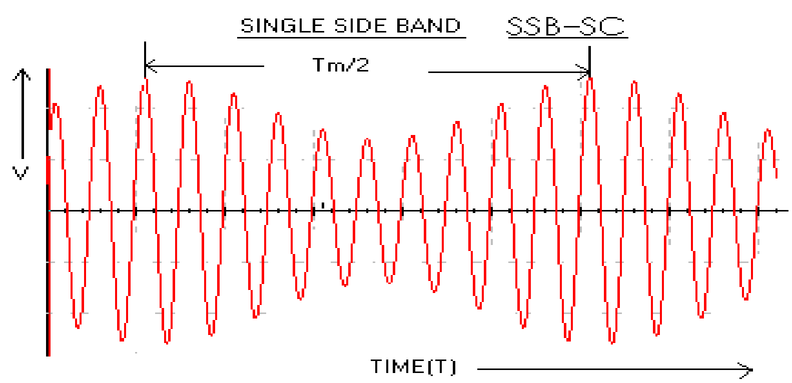




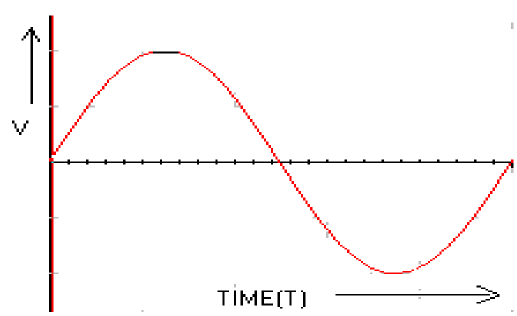








SSB DEMODULATED OUTPUT



RESULT:

QUESTIONS

1. What are the different methods to generate SSB-SC signal?
2. What is the advantage of SSB-SC over DSB-SC?
3. Explain Phase Shift method for SSB generation.
4. Why SSB is not used for broadcasting?

SSB DETECTION

5. Give the circuit for synchronous detector?
6. What are the uses of synchronous or coherent detector?
7. Give the block diagram of synchronous detector?
8. Why the name synchronous detector?

EXPERMENT NO-4 DATE:

FREQUENCY MODULATION AND DEMODULATION

AIM: To study the process of frequency modulation and demodulation and calculate the depth of modulation by varying the modulating voltage.

APPARATUS :

1. FM modulation and demodulation kit
2. Dual trace CRO.
3. CRO probes
4. Patch cards.
5. PC with windows(95/98/XP/NT/2000)
6. MATLAB Software with communication toolbox

THEORY:

The modulation system in which the modulator output is of constant amplitude, in which the signal information is super imposed on the carrier through variations of the carrier frequency.

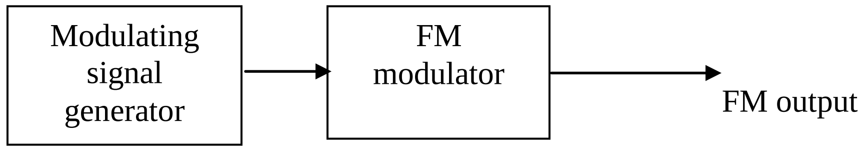
The frequency modulation is a non-linear modulation process. Each spectral component of the base band signal gives rise to one or two spectral components in the modulated signal. These components are separated from the carrier by a frequency difference equal to the frequency of base band component. Most importantly the nature of the modulators is such that the spectral components which produce decently on the carrier frquency and the base band frequencies.The spetral components in the modulated wave form depend on the amplitude.

The modulation index for FM is defined as

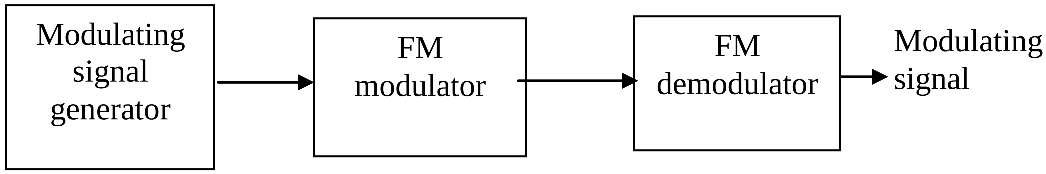
Mf= max frequency deviation/ modulating frequency.

BLOCK DIAGRAM:

Modulation



Demodulation



PROGRAM:-

```
% program for fm modulation and demodulation
close all
clear all
clc
%fm=35HZ,fc=500HZ,Am=1V,Ac=1V,B=10
fs=10000;
Ac=1;
Am=1;
fm=35;
fc=500;
B=10;
t=(0:.1*fs)/fs;
wc=2*pi*fc;
wm=2*pi*fm;
m_t=Am*cos(wm*t);
subplot(4,1,1);
plot(t,m_t);
title('Modulating or Message signal(fm=35Hz)');
c_t=Ac*cos(wc*t);
subplot(4,1,2);
plot(t,c_t);
title('Carrier signal(fm=500Hz)');
s_t=Ac*cos((wc*t)+B*sin(wm*t));
subplot(4,1,3);
```

```
plot(t,s_t);
title('Modulated signal');
d=demod(s_t,fc,fs,'fm');
subplot(4,1,4);
plot(t,d);
title('demodulated signal');
```

PROCEDURE:

1. Switch on the experimental board.
2. Observe the FM modulator output without any modulator input which is the carrier signal and note down its frequency and amplitude.
3. Connect modulating signal to FM modulator input and observe modulating signal and FM output on two channels of the CRO simultaneously.
4. Adjust the amplitude of the modulating signal until we get less distorted FM output.
5. Apply the FM output to FM demodulator and adjust the potentiometer in demodulation until we get demodulated output.

OBSERVATIONS:

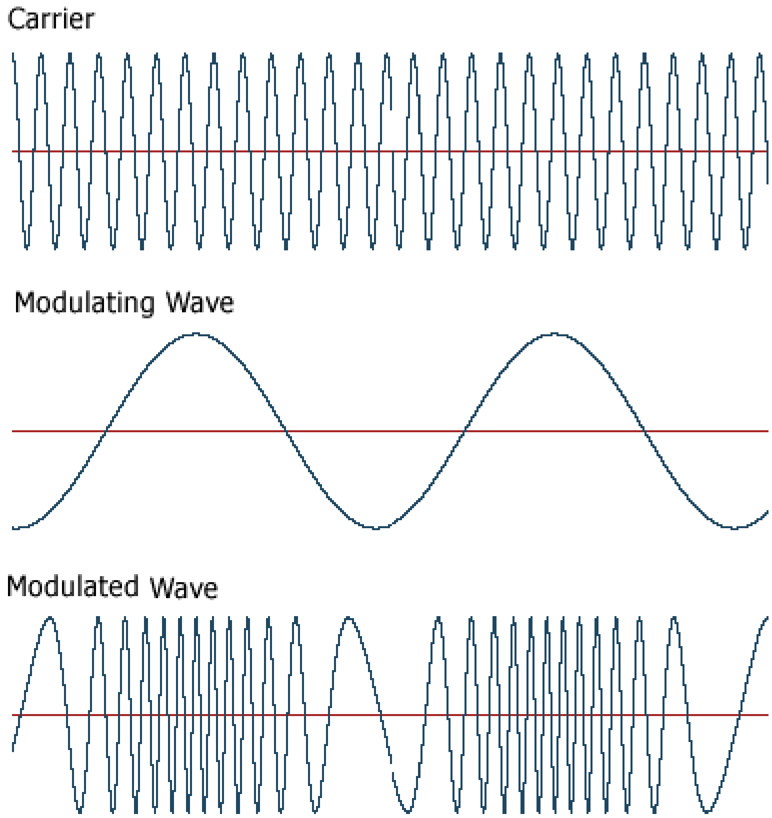
Modulation

Vm	F1	F2	Frequency deviation Fd (f1-f2)	Modulating index (f1-f2)/Fm	Band width= 2(Fd+Fm)

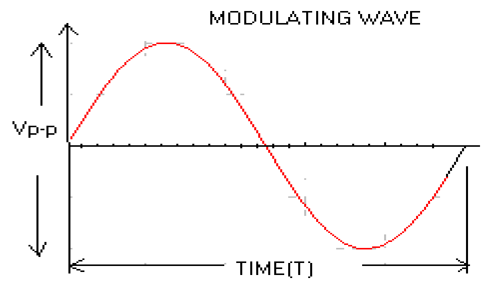
Demodulation

Modulating signal frequency	Demodulating signal frequency

EXPECTED WAVEFORMS:-



Demodulated signal



RESULT:

QUESTIONS

1. Define FM & PM.
2. What are the advantages of Angle modulation over amplitude modulation?
3. What is the relationship between PM and FM?
4. With a neat block diagram explain how PM is generated using FM.

EXPERIMENT NO-5 DATE:

STUDY OF SPECTRUM ANALYZER AND ANALYSIS OF AM AND FM SIGNALS

AIM: To verify the spectrum of AM and FM signals using spectrum analyzer.

APPARATUS / SOFTWARE REQUIRED:

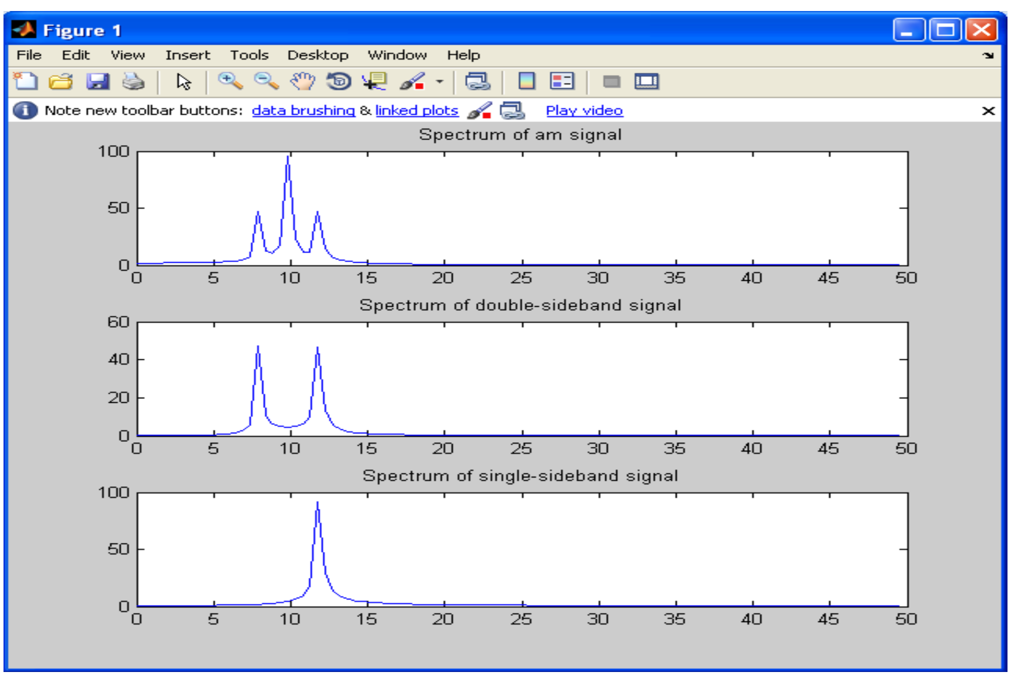
1. PC with windows(95/98/XP/NT/2000)
2. MATLAB Software with communication toolbox

PROGRAM:

```
%program of spectrum analyzer and analysis of am and fm signals
close all
clear all
clc
Fs = 100;          %sampling frq
t = [0:2*Fs+1]'/Fs;
Fc = 10;           % Carrier frequency
x = sin(2*pi*2*t);     % message signal
Ac=1;
% compute spectra of am
xam=ammod(x,Fc,Fs,0,Ac);
zam = fft(xam);
zam = abs(zam(1:length(zam)/2+1));
frqam = [0:length(zam)-1]*Fs/length(zam)/2;
% compute spectra of dsbsc
ydouble = ammod(x,Fc,Fs, 3.14,0);
zdouble = fft(ydouble);
zdouble = abs(zdouble(1:length(zdouble)/2+1));
frqdouble = [0:length(zdouble)-1]*Fs/length(zdouble)/2;
% compute spectra of ssb
ysingle = ssbmod(x,Fc,Fs,0,'upper');
zsingle = fft(ysingle);
```

```
zsingle = abs(zsingle(1:length(zsingle)/2+1));
frqsingle = [0:length(zsingle)-1]*Fs/length(zsingle)/2;
% Plot spectrums of am dsbsc and ssb
figure;
subplot(3,1,1); plot(frqam,zam);
title('Spectrum of am signal');
subplot(3,1,2); plot(frqdouble,zdouble);
title('Spectrum of double-sideband signal');
subplot(3,1,3); plot(frqsingle,zsingle);
title('Spectrum of single-sideband signal');
% spectrum of fm
xfm=fmmod(x,Fc,Fs,10);
zfm = fft(xfm);
zfm = abs(zfm(1:length(zfm)/2+1));
frqfm = [0:length(zfm)-1]*Fs/length(zfm)/2;
figure;
plot(frqfm,zfm);
title('Spectrum of fm signal');
```

EXPECTED WAVEFORMS:



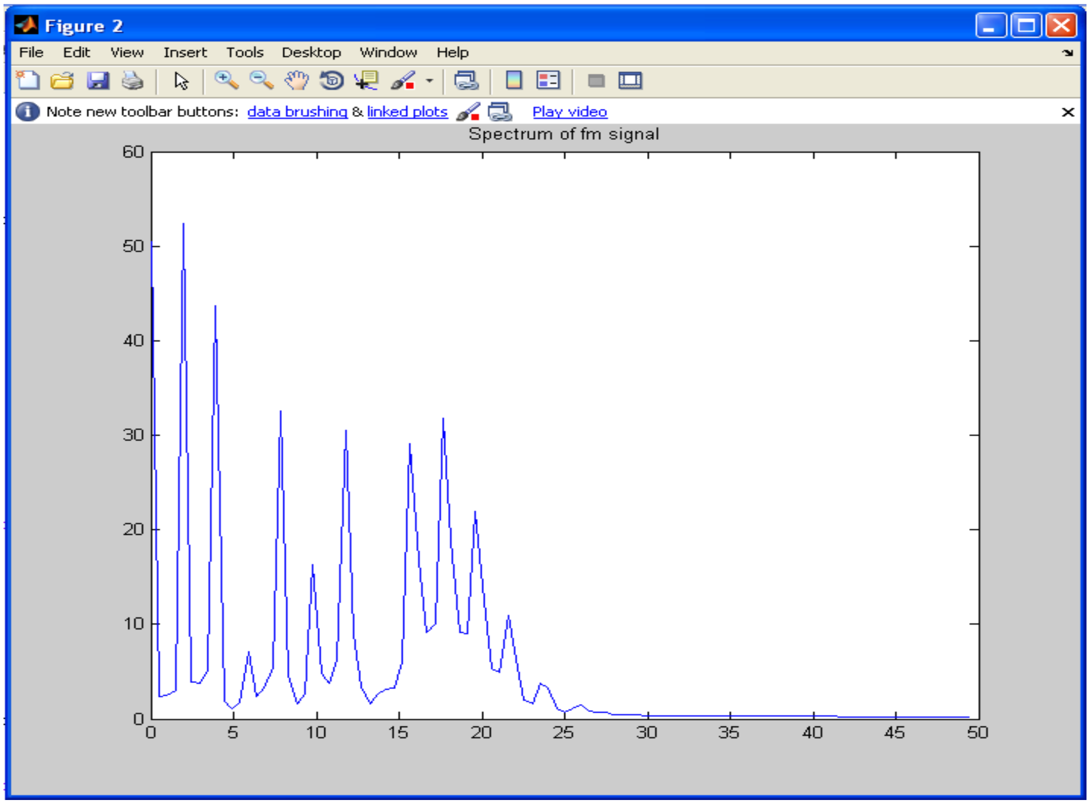


Figure 2
File Edit View Insert Tools Desktop Window Help
Note new toolbar buttons: data brushing & linked plots Play video
Spectrum of fm signal
0 10 20 30 40 50 60
0 5 10 15 20 25 30 35 40 45 50

RESULT:

EXPERIMENT.NO-6 DATE:

PRE-EMPHASIS & DE-EMPHASIS

AIM: To study the frequency response of Pre-Emphasis and De-Emphasis circuits.

APPARATUS:

1. Pre-emphasis & De-emphasis trainer kits.
2. C.R.O (20 MHz)
3. Function generator (1MHz).
4. Patch chords and Probes.
5. PC with windows (95/98/XP/NT/2000)
6. MATLAB Software with communication toolbox

THEORY:

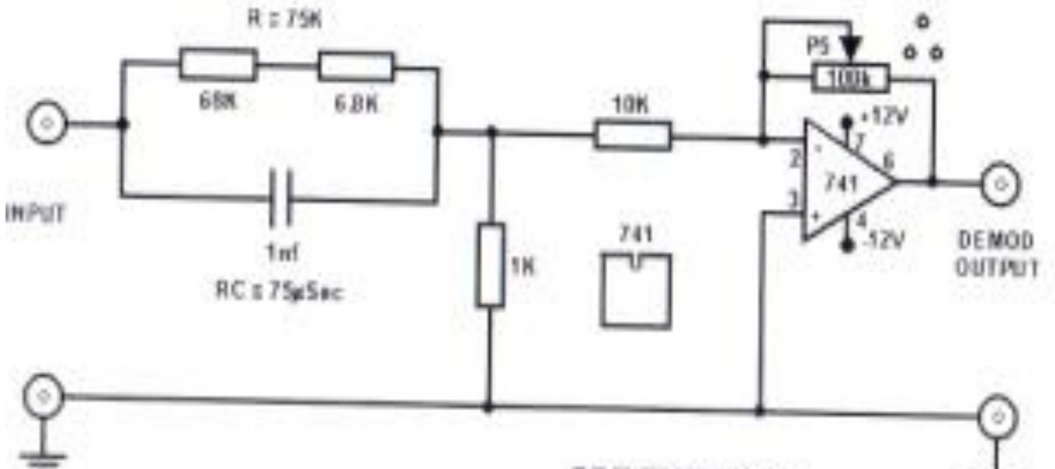
Frequency modulation is much immune to noise than amplitude modulation and significantly more immune than phase modulation. A single noise frequency will affect the output of the receiver only if it falls with in its pass band.

The noise has a greater effect on the higher modulating frequencies than on lower ones. Thus, if the higher frequencies were artificially boosted at the transmitter and correspondingly cut at the receiver, improvement in noise immunity could be expected. This booting of the higher frequencies, in accordance with a pre-arranged curve, is termed pre-emphasis, and the compensation at the receiver is called de-emphasis.

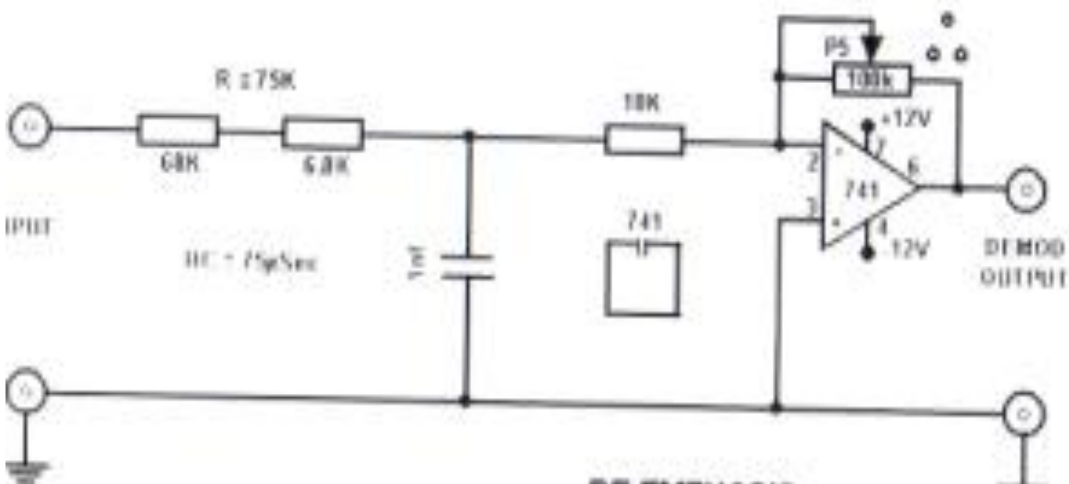
If the two modulating signals have the same initial amplitude, and one of them is pre-emphasized to (say) twice this amplitude, whereas the other is unaffected (being at a much lower frequency) then the receiver will naturally have to de-emphasize the first signal by a factor of 2, to ensure that both signals have the same amplitude in the output of the receiver. Before demodulation, I.e. while susceptible to noise interference the emphasized signal had twice the deviation it would have had without pre-emphasis, and was thus more immune to noise. Alternatively, it is seen that when this signal is de-emphasized any noise sideband voltages are de-emphasized with it, and therefore have a correspondingly lower amplitude than they would have had without emphasis again their effect on the output is reduced.

Apart from that, it would be difficult to introduce pre-emphasis and de-emphasis in existing AM services since extensive modifications would be needed, particularly in view of the huge numbers is receivers in use.

CIRCUIT DIAGRAM:



PRE EMPHASIS



DE EMPHASIS

PROGRAM:-

% program for Pre-Emphasis and De-Emphasis

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')
figure(3)
subplot(211)
plot(t,mt_pe);
axis([0 .6 min(mt_pe)-1 max(mt_pe)+1]);
title('preemphasised signal time domain')
subplot(212);
plot(f,Mf_pe);
title('pre-emphasised signal frequency domain');
grid on;axis([-50 50 0 max(Mf_pe)+100])
```

PROCEDURE:

I-PRE-EMPHASIS

1. Connect the circuit as per the circuit diagram
2. Apply a sine wave to the input terminals of 2 $V_{P-P\ (Vi)}$

3. By varying the input frequency with fixed amplitude, note down the output amplitude (V_o) with respect to the input frequency.

4. Calculate the gain using the formula

   Gain = 20 log (V_O/ V_I) db

   Where V_O = output voltage in volts.

   V_I = Input voltage in volts.

   And plot the frequency response.

II-DE-EMPHASIS

1. Connect the circuit as per circuit diagram.
2. Repeat steps 2, 3 & 4 of Pre-Emphasis to de-emphasis also.

EXPECTED WAVEFORMS

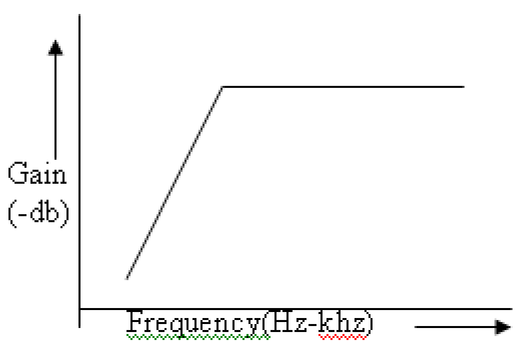


Fig: Pre-emphasis

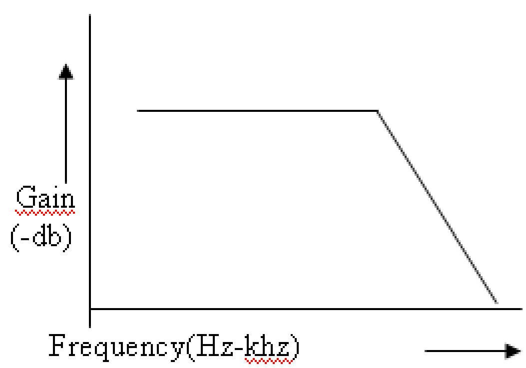


Fig: De-emphasis

TABLE:-

Pre-emphasis

Frequency(f)	Vin	Vo	Vo/Vin	Gain in db (20logVo/Vin)

De-emphasis

Frequency(f)	Vin	Vo	Vo/Vin	Gain in db (20logVo/Vin)

RESULT :

QUESTIONS

1. What is the need for pre-emphasis?
2. Explain the operation of pre-emphasis circuit?
3. Pre emphasis operation is similar to high pass filter explain how?
4. De emphasis operation is similar to low pass filter justify?
5. What is de-emphasis?
6. Draw the frequency response of a pre-emphasis circuit?
7. Draw the frequency response of a de-emphasis circuit?
8. Give the formula for the cutoff frequency of the pre-emphasis circuit?
9. What is the significance of the 3db down frequency?

EXPERIMENT NO-7 DATE:

FREQUENCY DIVISION MULTIPLEXING

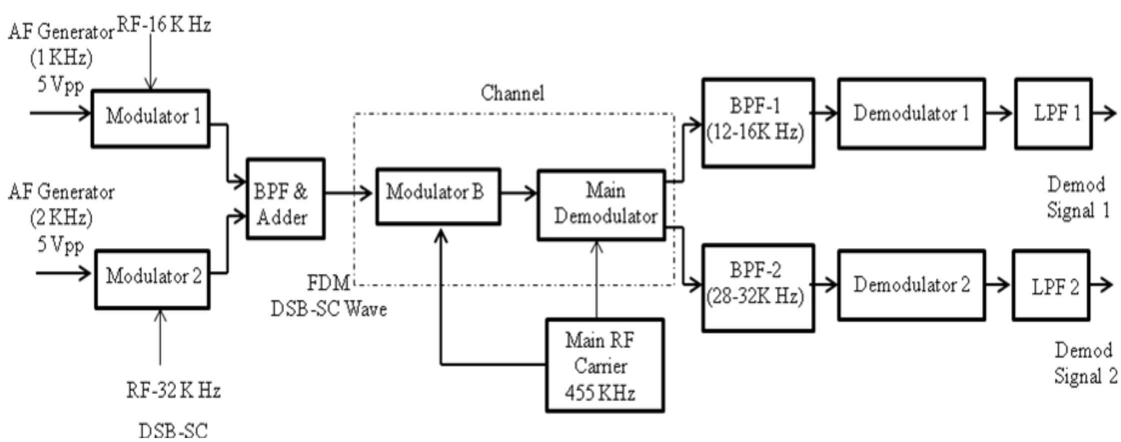
& DE MULTIPLEXING

AIM: To study the frequency division multiplexing and De multiplexing Techniques.

APPARATUS/SOFTWARE REQUIRED:

1. FREQUENCY DIVISION MULTIPLEXING & DEMULTIPLEXING Trainer Kit.
2. C.R.O (30 MHz)
3. Patch chords.
4. PC with windows (95/98/XP/NT/2000)
5. MATLAB Software

BLOCK DIAGRAM:



PROGRAM:

```
%program for frequency division multiplexing and demultiplexing
close all
clear all
clc
Fs = 100;         % sampling freq
t = [0:2*Fs+1]'/Fs;
x1 = sin(2*pi*2*t); % signal 1 signal
z1 = fft(x1);
z1=abs(z1);
```

```
x2 = sin(2*pi*10*t); % signal 2 signal
z2 = fft(x2);
z2=abs(z2);
figure;
subplot(4,1,1); plot(x1);
title('signal 1');xlabel('time');ylabel('amplitude');
subplot(4,1,2); plot(x2);
title('signal 2');xlabel('time');ylabel('amplitude');
subplot(4,1,3); plot(z1);
title('Spectrum of signal 1');xlabel('freqency');ylabel('magnitude');
subplot(4,1,4); plot(z2);
title('Spectrum of signal 2');xlabel('freqency');ylabel('magnitude');
% freqency multiplexing
z=z1+z2;
figure;
plot(z);
title('frequency multiplexed signals');
figure;
% freqency demultiplexing
f1=[ones(10,1); zeros(182,1);ones(10,1)];%applying filter for signal 1
dz1=z.*f1;
d1 = ifft(dz1);
subplot(2,1,1)
plot(t*100,d1);
f2=[zeros(10,1); ones(182,1);zeros(10,1)];% applying filter for signal 2
dz2=z.*f2;
d2 = ifft(dz2);
title('recovered signal 1');xlabel('time');ylabel('amplitude');
subplot(2,1,2)
plot(t*100,d2);
title('recovered signal 2');xlabel('time');ylabel('amplitude');
```

PROCEDURE:

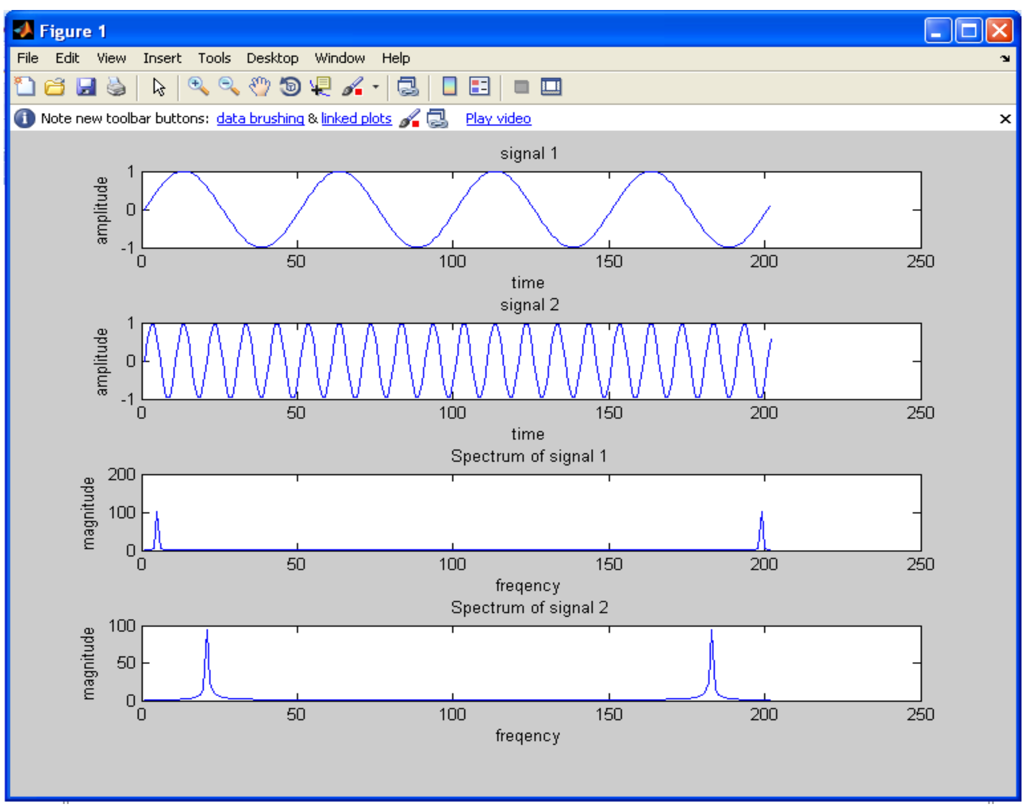
FDM Multiplexing:

1. Connect the circuit as shown in the figure.
2. Switch ON the power supply.
3. Set the amplitude of each modulating signal as 5Vp-p and frequency of each AF signal to 1kHz and 2kHz respectively.
4. Monitor the outputs at Tp1(signal-1), Tp2(signal-2),Tp10(RF-16kHz),Tp12(RF-32KHz),Tpq(modulation-1),Tp11(Modulator-2),Tp17(BPF & adder)
5. Set output frequency of RF oscillator to 455 kHz and amplitude to10Vp-p.
6. Monitor the output at Tp18 the FDM DSB-SC wave will be observed.

FDM DeMultiplexing & LPF:

1. Connect the Tp18 to Tp22 and observe the output of main demodulator at Tp23.
2. Connect the main demodulator output to the BPF1 (28-32 kHz) and BPF1 (12-16 kHz).
3. Connect the output of BPF,s to the respective demodulator and then to LPF,s.
4. Monitor the demodulated signal1 and at TP32 and demodulated signal2 atTP39.

EXPECTED WAVEFORMS:



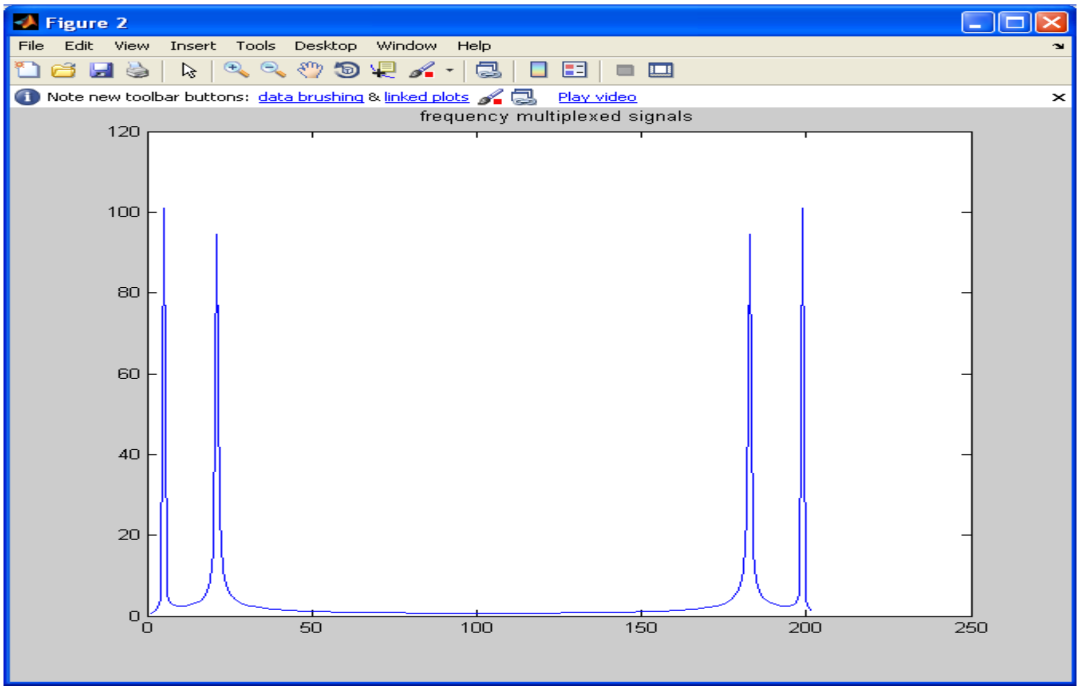


Figure 2
File Edit View Insert Tools Desktop Window Help
Note new toolbar buttons: data brushing & linked plots Play video
frequency multiplexed signals
120
100
80
60
40
20
0
0 50 100 150 200 250

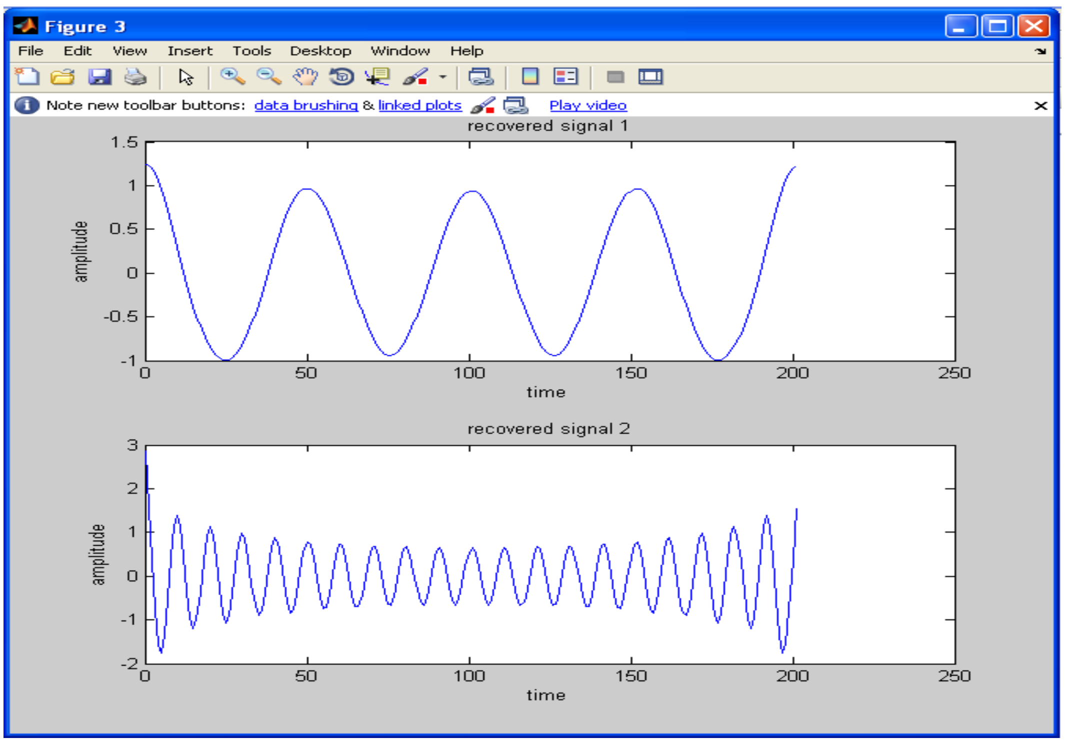


Figure 3
File Edit View Insert Tools Desktop Window Help
Note new toolbar buttons: data brushing & linked plots Play video
recovered signal 1
amplitude
time
recovered signal 2
amplitude
time

RESULT:

EXPERIMENT NO-8 DATE:

VERIFICATION OF SAMPLING THEOREM

AIM:

1. To study the sampling theorem and its reconstruction.

2. To study the effect of amplitude and frequency variation of modulating signal on the output.

3. To study the effect of variation of sampling frequency on the demodulated output.

APPARATUS:

1. Sampling and reconstruction Trainer Kit.
2. C.R.O(30Mhz)
3. Patch cords.
4. PC with windows(95/98/XP/NT/2000)
5. MATLAB Software with communication toolbox

THEORY:

Pulse Modulation is used to transmit analog information. In this system continuous wave forms are sampled at regular intervals. Information regarding the signal is transmitted only at the sampling times together with synchronizing signals.

At the receiving end, the original waveforms may be reconstituted from the information regarding the samples.

Sampling Theorem Statement:

A band limited signal of finite energy which has no frequency components higher than f_m Hz, is completely described by specifying the values of the signal at instants of time separated by ½ f_m seconds.

The sampling theorem states that, if the sampling rate in any pulse modulation system exceeds twice the maximum signal frequency, the original signal can be reconstructed in the receiver with minimum distortion.

$F_s > 2f_m$ is called Nyquist rate.

Where f_s – sampling frequency

F_m – Modulation signal frequency.

If we reduce the sampling frequency f_s less than f_m, the side bands and the information signal will overlap and we cannot recover the information signal simply by low pass filter. This phenomenon is called fold over distortion or aliasing.There are two methods of sampling. (1) Natural sampling (2) Flat top sampling.

Sample & Hold circuit holds the sample value until the next sample is taken. Sample & Hold technique is used to maintain reasonable pulse energy.The duty cycle of a signal is defined as the ratio of Pulse duration to the Pulse repetition period. The duty cycle of 50% is desirous taking the efficiency into account.

Circuit Description:-

Pulse and Modulating Signal Generator:-

A 4.096 MHz clock is used to derive the modulating signal, which is generated by an oscillator circuit comprising a 4.096MHz crystal and three 74HC04(U9) inverter gates. This 4.096MHz clock is then divided down in frequency by a factor of 4096, by binary counter 74HC4040(U10), to produce 50% duty cycle, 1KHz square wave on pin no.1 of U10, and 2KHz square wave on pin no.15. the frequency is selectable by means of SW1. this input of fourth order low pass filter U11(TL072) is used to produce sine wave from the square wave. The amplitude of this sine wave can be varied.

The square wave which is generated by the oscillator is buffered by inverter 74HC04(U9), to produce 32KHz square wave at pin no. 4 of the 74HC4040. This pulse is given to the monostable multi(U4) to obtain the 16KHz and 32KHz square wave at the output which are selected by the frequency pot.

Sampling Circuit:-

The IC DG211(U3) is used as analog switch which is used in pulse amplitude modulation in this circuit. The modulation signal & pulse signal are given as the input to TL074(U2), 7400(U1) IC’s respectively. These IC output are fed to the inputs of the DG211.

The sampled output is available at the pin no.2 of DG211 and it is buffered by using TL074(U2) and then output is available at TP5.

Similarly the sample & hold output and the flat top output are available at pin no15 & 10 of DG211 respectively. These are buffered by TL074(U2) and then output is available at TP6 & TP7 respectively.

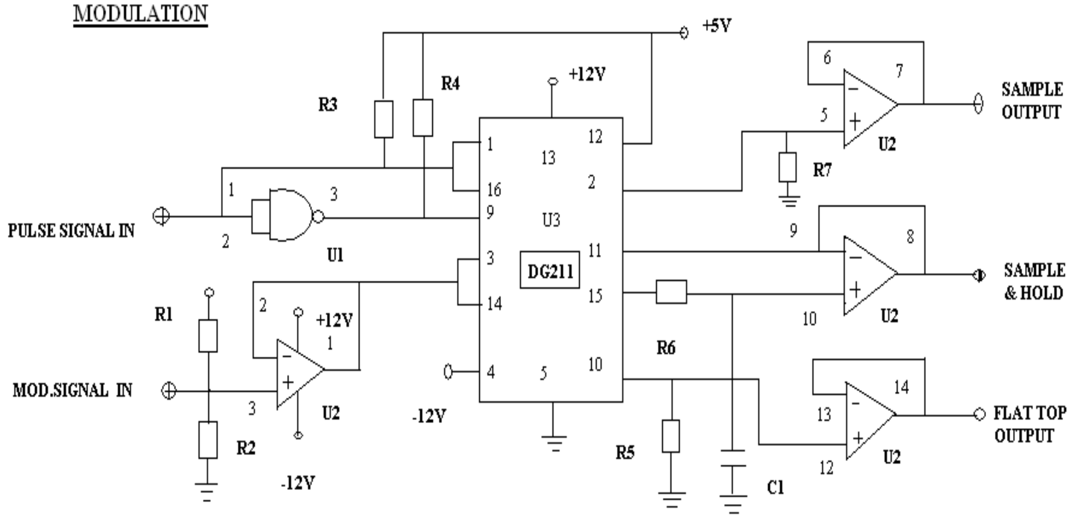
Reconstruction Circuit:-

The demodulation section comprises of a fourth order low pass filter and an AC amplifier. The TL074 (U5) is used as a low pass filter and AC amplifier. The output of the modulator is given as the input to the low pass filter.

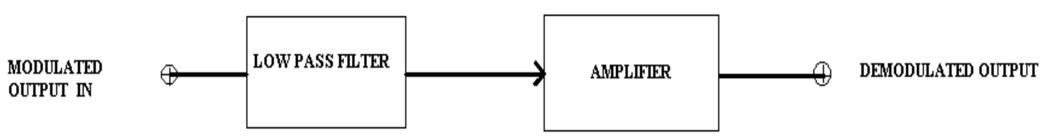
The low pass filter output is obviously less and it is fed to the AC amplifier which comprises of a single op amp and whose output is amplified.

CIRCUIT DIAGRAM:

MODULATION



DEMODULATION



PROGRAM:-

```
%program for verification of sampling theorem
close all;
clear all
clc
```

```
t=-10:.01:10;
 T=4;
 fm=1/T;
 x=cos(2*pi*fm*t);      % input signal
 subplot(2,2,1);
 plot(t,x);
 xlabel('time');ylabel('x(t)');title('continous time signal');
 grid;
 n1=-4:1:4;
fs1=1.6*fm;
fs2=2*fm;
fs3=8*fm;
%discrete time signal with fs<2fm
x1=cos(2*pi*fm/fs1*n1);
subplot(2,2,2);
stem(n1,x1);
xlabel('time');ylabel('x(n)');
title('discrete time signal with fs<2fm');
hold on
subplot(2,2,2);
 plot(n1,x1)
  grid;
  %discrete time signal with fs=2fm
  n2=-5:1:5;
  x2=cos(2*pi*fm/fs2*n2);
    subplot(2,2,3);
 stem(n2,x2);
 xlabel('time');ylabel('x(n)');
title('discrete time signal with fs=2fm');
 hold on
subplot(2,2,3);
```

```
plot(n2,x2)
%discrete time signal with fs>2fm
 grid;
 n3=-20:1:20;
x3=cos(2*pi*fm/fs3*n3);
subplot(2,2,4);
stem(n3,x3);
xlabel('time');ylabel('x(n)');
title('discrete time signal with fs>2fm');
hold on
subplot(2,2,4);
plot(n3,x3)
 grid;
```

PROCEDURE:

Sampling:-

1. Connect the circuit as shown in diagram 1 .

   a. The output of the modulating signal generator TP1 is connected to modulating signal input TP4 of the sampling circuit keeping the frequency switch in 1KHz position, and amplitude knob to max position.

   b. The output of pulse generator TP2 is connected to sampling pulse input TP3 of the sampling circuit keeping the frequency switch in 16KHz position.(Adjust the duty cycle pot to mid position i.e.50%).

2. Switch ON the power supply.
3. Observe the outputs of sampling, sampling and hold, flat top output at TP7, TP8 and TP9 respectively. By varying the amplitude pot also observe the effect on outputs.
4. By varying Duty cycle pot observe the effect on sampling outputs (Duty cycle is varying from 10-15%).

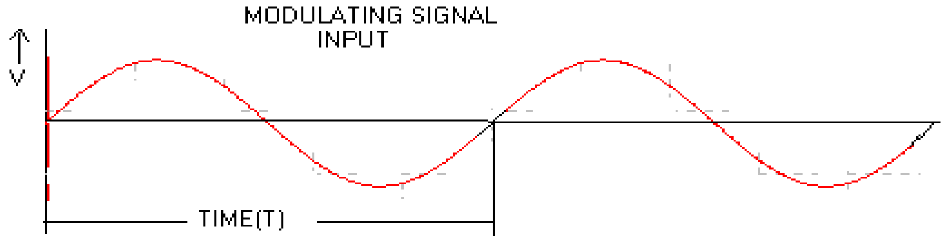
5. Vary the switch position in the pulse generator circuit to 32 KHz and now observe the outputs at TP7, TP8 and TP9.By varying the amplitude pot also observe the effect on outputs.
6. Now, vary the switch position in modulating signal generator to 2 KHz and repeat all the above steps 3&4.
7. Switch OFF the power supply.

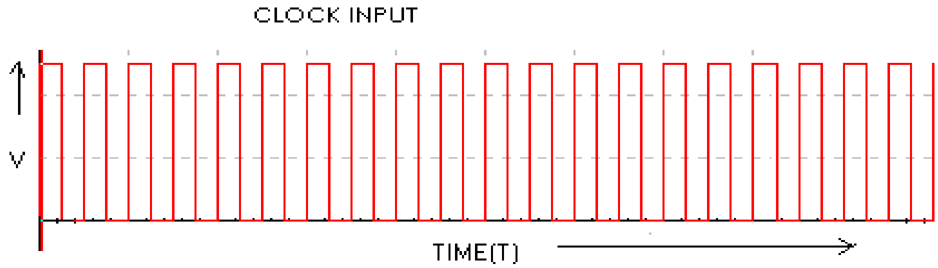
Reconstruction:-

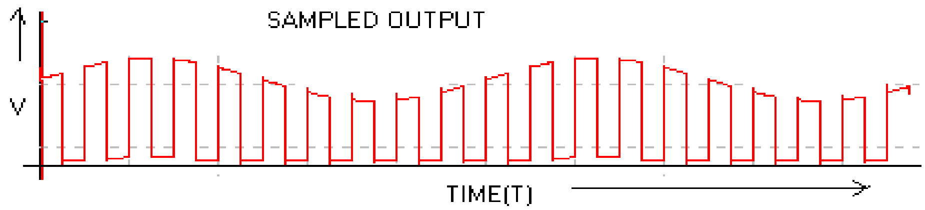
1. Connect the circuit as shown in diagram 2.
    a. . The output of the modulating signal generator TP1 is connected to modulating signal input TP4 of the sampling circuit keeping the frequency switch in 1 KHz position, and amplitude knob to max position.
    b. The output of pulse generator TP2 is connected to sampling pulse input TP3 of the sampling circuit keeping the frequency switch in 16KHz position.(Adjust the duty cycle pot to mid position i.e.50%).
    c. Connect the sample output from TP7 to the input of low pass filter TP10.
    d. Output of low pass filter from TP11 to input of AC amplifier TP12, keep the gain pot in AC amplifier to max position.
2. Switch ON the power supply.
3. Observe the output of AC amplifier at TP13. The output will be the replica of the input. By varying the gain pot observe the demodulating signal amplification.
4. Similarly connect the sample and hold output and flat top output to TP10 and observe reconstructed the signal.
5. Vary the switch position in the sampling frequency circuit to 32KHz and now repeat the steps 3&4.
6. Vary the switch position in the modulating signal generator to 2KHz and repeat all the above steps 3 to 5.
7. Switch OFF the power supply.

EXPECTED WAVEFORMS:

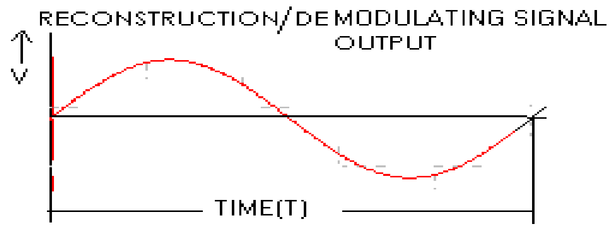
Below Wavforms for **fs > 2fm**

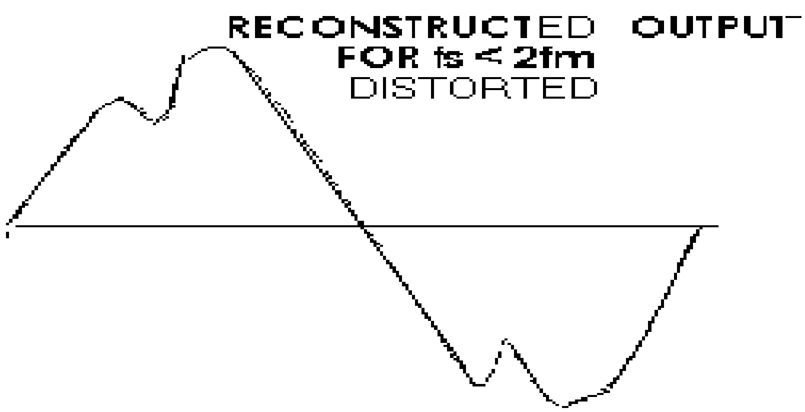






Demodulated Output





RESULT:

QUESTIONS

1. What are the types of sampling?
2. State sampling theorem?
3. What happens when fs < 2 fm?
4. How will be the reconstructed signal when fs >= 2fm?
5. Explain the operation of sampling circuit?
6. Explain the operation of re-construction circuit?
7. Who formalized the sampling theorem?
8. What are the applications of the above theorem?
9. Is the sampling theorem basis for the modern digital communications?
10. Is the voice signal sampling of 8000 Hz, follows sampling theorem in Land line Telephone Exchange.

EXPERIMENT NO-9 DATE:

PULSE AMPLITUDE MODULATION

AIM:- 1.To study the Pulse amplitude modulation & demodulation Techniques.

2. To study the effect of amplitude and frequency variation of modulating signal on the output.

APPARATUS:-

1. Pulse amplitude modulation & demodulation Trainer Kit.
2. Dual trace CRO.
3. Patch chords.
4. PC with windows(95/98/XP/NT/2000)
5. MATLAB Software with communication toolbox

THEORY:-

Pulse modulation is used to transmit analog information. In this system continuous wave forms are sampled at regular intervals. Information regarding the signal is transmitted only at the sampling times together with syncing signals.

At the receiving end, the original waveforms may be reconstituted from the information regarding the samples.

The pulse amplitude modulation is the simplest form of the pulse modulation. PAM is a pulse modulation system is 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 cables are used to modulated carrier.

The two types of PAM are i) Double polarity PAM, and ii) the single polarity PAM, in which a fixed dc level is added to the signal to ensure that the pulses are always positive. Instantaneous PAM sampling occurs if the pulses used in the modulator are infinitely short.

Natural PAM sampling occurs when finite-width pulses are used in the modulator, but the tops of the pulses are forced to follow the modulating waveform.

Flat-topped sampling is a system quite often used because of the ease of generating the modulated wave.

PAM signals are very rarely used for transmission purposes directly. The reason for this lies in the fact that the modulating information is contained in the amplitude factor of the pulses, which can be easily distorted during transmission by noise, crosstalk, other forms of distortion. They are used frequently as an intermediate step in other pulse-modulating methods, especially where time-division multiplexing is used.

Circuit description:-

Pulse and Modulation Signal Generator:-

A 4.096 MHz clock is used to derive the modulating signal, which is generated by an oscillator circuit comprising a 4.096MHz crystal and three 74HC04(U9) inverter gates. This 4.096MHz clock is then divided down in frequency by a factor of 4096, by binary counter 74HC4040(U10), to produce 50% duty cycle, 1 KHz square wave on pin no.1 of U10, and 2KHz square wave on pin no.15. the frequency is selectable by means of SW1. this goes to input of fourth order low pass filter U11(TL072) is used to produce sine wave from the square wave. The amplitude of this sine wave can be varied.

The square wave which is generated by the oscillator is buffered by inverter 74HC04(U9), to produce 32KHz square wave at pin no.4 of the 74HC4040(U10). This pulse is given to the monostable multi to obtain the 16 KHz and 32 KHz square wave at the output which are selected by the frequency pot.

Modulation:-

The ICDG211 (U3) is used as a pulse amplitude modulation in this circuit. The modulation signal & pulse signals are given to TL074 (U2) & 7400(U1) IC's respectively. These outputs are fed to the inputs the D4211 (U3).

The sampled output is available at the pin no 2 of DG211 and it is buffered by using TL074 (U2) and then output is available at TP5.

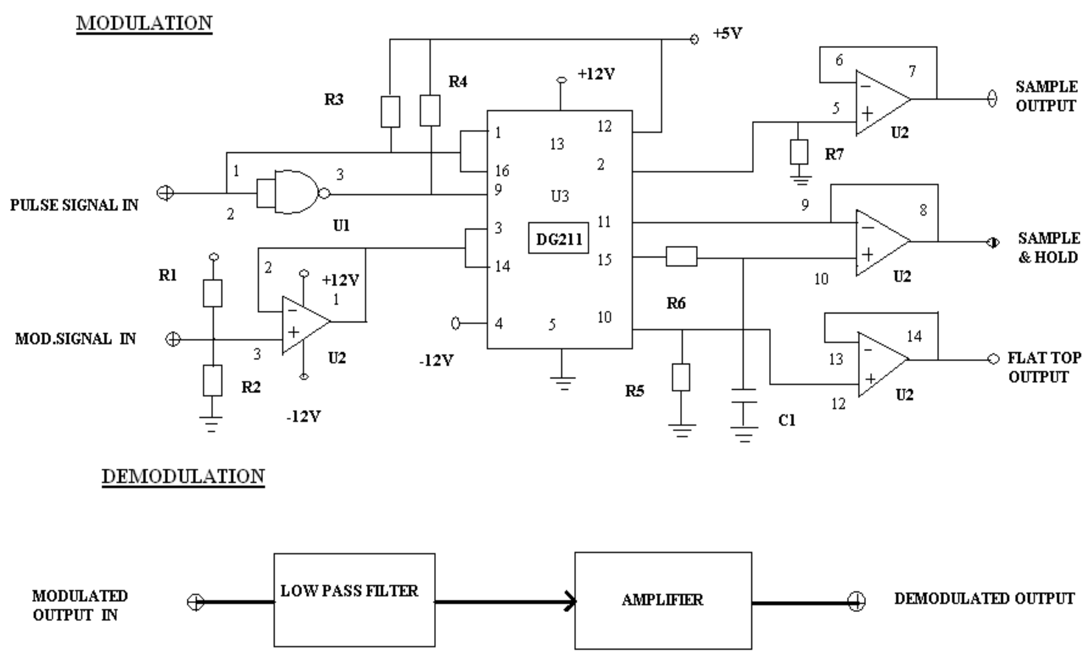
Similarly the sample & hold output and the flat top output are available at pin no.15 &10 of DG211 respectively. These are buffered by TL074 (U2) and then output is available at TP6&TP7 respectively.

Demodulation:-

The demodulation section comprises of fourth order low pass filter and an AC amplifier. The TL074 (U5) is used as a low pass filter and AC amplifier. The output of the modulator is given as the input to the low pass filter.

The low pass filter output is obviously less and it is fed to the AC amplifier which comprises of a single op amp and whose output is amplified.

CIRCUIT DIAGRAM:



PROGRAM:-

```
% pulse amplitude modulation
close all
clear all
clc
t = 0 : 1/1e3 : 10;        % 1 kHz sample freq for 1 sec
d = 0 : 1/5 : 10;
x = 5+sin(2*pi/4*2*t);     %message signal
figure;
subplot(3,1,1)
plot(x);
```

```
title('message');
xlabel('time');ylabel('amplitude');
y = pulstran(t,d,'rectpuls',0.1); %generation of pulse input
subplot(3,1,2)
plot(y);
title('Pulse Input ');
xlabel('time');ylabel('amplitude');
z=x.*y;                 % PAM output
subplot(3,1,3)
plot(z);
title('PAM modulation ');
xlabel('time');ylabel('amplitude');
```

PROCEDURE:

Double Polarity:-

Modulation:-

1. Connect the circuit as shown in diagram 1.
    a. The output of the modulating signal generator is connected to the modulating signal input TP2 keeping the frequency switch in 1KHz position, and amplitude knob to max position
    b. 16KHz pulse output to pulse input TP1.(Keep the frequency in minimum position in pulse generator block).
2. Switch ON the power supply.
3. Monitor the outputs at TP5, TP6& TP7. And observe the outputs also by varying amplitude pot (Which is in modulation signal generator block).
4. Now vary the frequency selection which position in modulating signal generator block to 2 KHz, amplitude pot to max position.
5. Observe the output at TP5, TP6& TP7 and observe the outputs also by varying amplitude pot (Which is in modulation signal generator block).
6. Repeat all the above steps for the pulse frequency 32KHz (By varying the frequency pot in the pulse generator block).
7. Switch OFF the power supply.

Single Polarity PAM:-

8. Connect the circuit as shown in diagram 2.

a. The output of the modulating signal generator is connected to the modulating signal input TP2 keeping the frequency switch in 1KHz position, and amplitude knob to max position

b. 16KHz pulse output to pulse input TP1 .

9. Switch ON the power supply.

10. Repeat above step 3 to 6 and observe the outputs.

11. Vary DC output pot until you get single polarity PAM at TP5, TP6, TP7.

12. Switch OFF the power supply.

Demodulation:-

1. Connect the circuit as shown in diagram 3.

a. The output of the modulating signal generator is connected to the modulating signal input TP2 keeping the frequency switch in 1KHz position, and amplitude knob to max position

b. 16KHz pulse output to pulse input TP1.

c. Sample output, sample and hold output and flat top outputs Respectively to the input of low pass filter(TP9) and LPF output (TP10) to AC amplifier input(TP11).

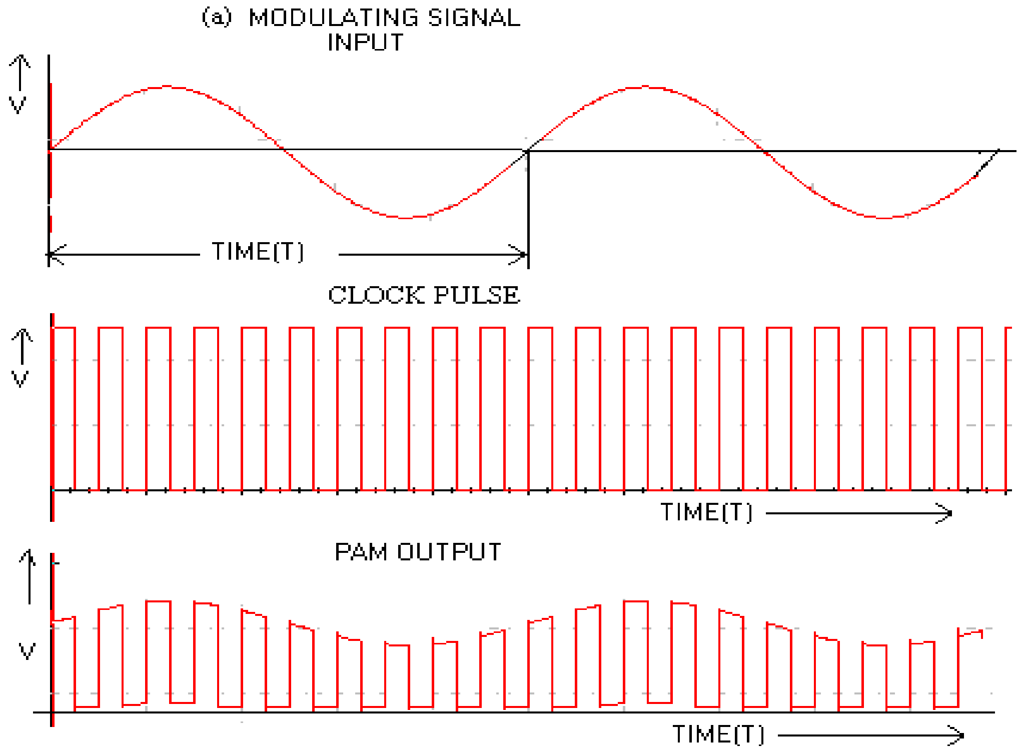
2. Observe the output of LPF and AC amplifier at TP10,TP12 respectively, corresponding to inputs from TP5,TP6 &TP7. The outputs will be the true replica of the input.

3. Now, set the switch position in modulating signal generator to 2KHz and observe the outputs at TP10&TP12 respectively, corresponding to inputs from TP5,TP6& TP7.

4. Vary the frequency of pulse to 32KHz (By varying the frequency pot(Put in max position) in pulse generator block) and repeat the above steps 2&3.

5. Switch OFF the power supply.

EXPECTED WAVEFORMS



RESULT:

QUESTIONS

1. TDM is possible for sampled signals. What kind of multiplexing can be used in continuous modulation systems?

2. What is the minimum rate at which a speech signal can be sampled for the purpose of PAM?

3. What is cross talk in the context of time division multiplexing?

4. Which is better, natural sampling or flat topped sampling and why?

5. Why a dc offset has been added to the modulating signal in this board? Was it essential for the working of the modulator? Explain?

6. If the emitter follower in the modulator section saturates for some level of input signal, then what effect it will have on the output?

7. Derive the mathematical expression for frequency spectrum of PAM signal.

8. Explain the modulation circuit operation?

9. Explain the demodulation circuit operation?

10. Is PAM & Demodulation is sensitive to Noise?

EXPERIMENT NO-10 DATE:

PULSE WIDTH MODULATION & DEMODULATION

AIM:

1. To study the Pulse Width Modulation (PWM) and Demodulation Techniques.

2. To study the effect of Amplitude and Frequency of Modulating Signal on PWM output.

APPARATUS:

1. PWM trainer kit
2. C.R.O(30MHz)
3. Patch Chords.
4. PC with windows(95/98/XP/NT/2000)
5. MATLAB Software with communication toolbox

THEORY:-

Pulse modulation is used to transmit analog information. In this system continuous wave forms are sampled at regular intervals. Information regarding the signal is transmitted only at the sampling times together with synchronizing signals.

At the receiving end, the original waveforms may be reconstituted from the information regarding the samples.

The pulse Width Modulation of the PTM is also called as the Pulse Duration Modulation (PDM) & less often Pulse length Modulation (PLM).

In pulse Width Modulation method, we have fixed and starting time of each pulse, but the width of each pulse is made proportional to the amplitude of the signal at that instant.

This method converts amplitude varying message signal into a square wave with constant amplitude and frequency, but which changes duty cycle to correspond to the strength of the message signal.

Pulse-Width modulation has the disadvantage, that its pulses are of varying width and therefore of varying power content. This means that the transmitter must be powerful enough to handle the maximum-width pulses. But PWM still works if synchronization between transmitter and receiver fails, whereas pulse-position modulation does not.

Pulse-Width modulation may be generated by applying trigger pulses to control the starting time of pulses from a mono stable multivibrator, and feeding in the signal to be sampled to control the duration of these pulses.

When the PWM signals arrive at its destination, the recovery circuit used to decode the original signal is a sample integrator (LPF).

CIRCUIT DESCRIPTION:-

Pulse & Modulating Signal Generator:-

A 4.096MHz clock is used to derive the modulating signal, which is generated by an oscillator circuit comprising a 4.096MHz crystal and three 74HC04(U9) inverter gates. This 4.096MHz clock is then divided down in frequency by a factor of 4096, by binary counter 74HC4040(U2), to produce 50% duty cycle, 1KHz square wave on pin no.1 of U4, and 2KHz square wave on pin no.15. the frequency is selectable by means of SW1. This goes to input of fourth order low pass filter U3 is used to produce sine wave from the square wave. The amplitude of this sine wave can be varied.

The square wave which is generated by the oscillator is buffered by inverter 74HC04, to produce 32KHz square wave at pin no.4 of the 74HC4040(U2). This pulse is given to the monostable multi to obtain the 16KHz and 32KHz square wave at the output which are selected by the frequency pot.

Modulation:-

The PWM circuit uses the 555 IC (U1) in monostable mode. The Modulating signal input is applied to pin no.5 of 555IC, and there Pulse input is applied to pin no.2.

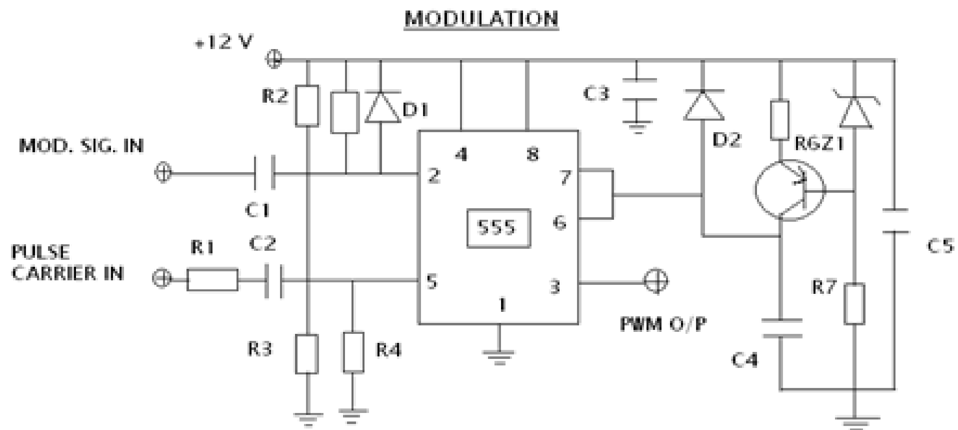
The output of PWM is taken at the pin no.3 of 555IC i.e., TP3.

Demodulation:-

The demodulation section comprises of a fourth order low pass filter and an AC amplifier. The TL074(U5) is used as a low pass filter and an AC amplifier. The output of the modulator is given as the input to the low pass filter.

The low pass filter output is obviously less and it is feed to the AC amplifier which comprises of a single op amp and whose output is amplified.

CIRCUIT DIAGRAM:



PROGRAM:-

```
% 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(311);
plot(x1);
axis([0 50 0 1]);
title('original signal taken mesage,f1=500,fs=10000')
subplot(312);
plot(y1);
axis([0 500 -0.2 1.2]);
```

```
title('PWM')
 %demodulation
x1_recov=demod(y1,fc,fs,'pwm');
subplot(313);
plot(x1_recov);
title('time domain recovered, single tone,f1=200')
axis([0 50 0 1]);
```

PROCEDURE:

Modulation:-

1. Connect the circuit as shown in the diagram 1.
    a. The output of the modulating signal generator is connected to the modulating signal input TP2 keeping the frequency switch in 1KHz position, and amplitude knob to max position
    b. 16KHz pulse output (by varying the frequency pot (put it min position) in pulse generator block) from pulse generator to pulse input(TP1).
2. Switch ON the power supply.
3. Observe the output of pulse width modulation block at TP3.(By varying the amplitude pot).
4. Vary the modulating signal generator frequency by switching the frequency selector switch to 2 KHz.
5. Now, again observe the PWM output at TP3.(By varying the amplitude pot).
6. Repeat the above steps (3 to 5) for the pulse frequency of 32KHz(by varying the frequency pot(put it in max position) in pulse generator block).
7. Switch OFF the power supply.

Demodulation:-

8. Connect the circuit as shown in diagram 2.

a. The output of the modulating signal generator is connected to the modulating signal input TP2 keeping the frequency switch in 1KHz position, and amplitude knob to max position.

b. 16KHz pulse output (put frequency pot minimum) from pulse generator block to pulse input TP1.

c. PWM output to LPF input.

d. LPF output to AC amplifier input.

9. Switch ON the power supply.
10. Observe the output of low pass filter and AC amplifier respectively at TP6 & TP8. The output will be the true replica of the input.
11. Now vary the position of the switch in modulating signal generator to 2 KHz and observe the outputs at TP6 & TP8.
12. Repeat the steps 10& 11 for pulse frequency 32 KHz (By varying the frequency pot (put in max). in pulse generator block). Observe the output waveforms.
13. Switch OFF the power supply.

EXPECTED WAVEFORMS

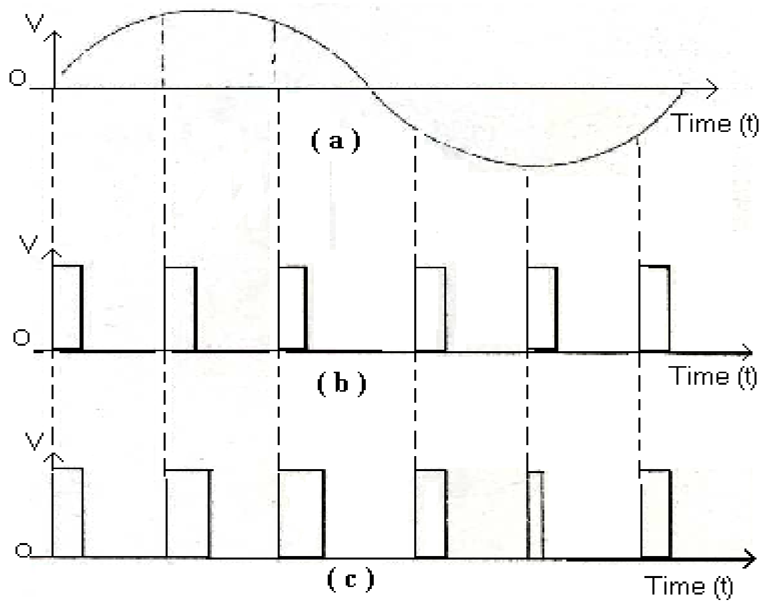


Fig (2) PULSE WIDTH MODULATION
(a) Signal
(b) Unmodulated pulses
(c) PWM

RESULT:

QUESTIONS

1. An audio signal consists of frequencies in the range of 100Hz to 5.5KHz.What is the minimum frequency at which it should be sampled in order to transmit it through pulse modulation?

2. Draw a TDM signal which is handling three different signals using PWM?

3. What do you infer from the frequency spectrum of a PWM signal?

4. Clock frequency in a PWM system is 2.5 kHz and modulating signal frequency is 500Hzhowmany pulses per cycle of signal occur in PWM output? Draw the PWM signal?

5. Why should the curve for pulse width Vs modulating voltage be linear?

6. What is the other name for PWM?

7. What is the disadvantage of PWM?

8. Will PWM work if the synchronization between Tx and Rx fails?

9. Why integrator is required in demodulation of PWM?

10. What kind of conversion is done in PWM generation?

EXPERIMENT NO-11 DATE:

PULSE POSITION MODULATION AND DEMODULATION

AIM:

1. To study the generation Pulse Position Modulation (PPM) and Demodulation.
2. To study the effect of Amplitude and the frequency of modulating signal on its output and observe the wave forms.

APPARATUS:

1. Pulse Position Modulation (PPM) and demodulation Trainer Kit.
2. C.R.O(30MHz)
3. Patch chords.
4. PC with windows(95/98/XP/NT/2000)
5. MATLAB Software with communication toolbox

THEORY:-

Pulse Modulation is used to transmit analog information in this system continuous wave forms are sampled at regular intervals. Information regarding the signal is transmitted only at the sampling times together with synchronizing signals.

At the receiving end, the original waveforms may be reconstituted from the information regarding the samples. Pulse modulation may be subdivided in to two types analog and digital. In analog the indication of sample amplitude is the nearest variable. In digital the information is a code.

The pulse position modulation is one of the methods of the pulse time modulation.PPM is generated by changing the position of a fixed time slot.

The amplitude& width of the pulses is kept constant, while the position of each pulse, in relation to the position of the recurrent reference pulse is valid by each instances sampled value of the modulating wave. Pulse position modulation into the category of analog communication. Pulse-Position modulation has the advantage of requiring constant transmitter power output, but the disadvantage of depending on transmitter receiver synchronization.

Pulse-position modulation may be obtained very simply from PWM. However, in

PWM the locations of the leading edges are fixed, whereas those of the trailing edges are not. Their position depends on pulse width, which is determined by the signal amplitude at that instant. Thus, it may be said that the trailing edges of PWM pulses are, in fact, position-modulated. This has positive-going narrow pulses corresponding to leading edges and negative-going pulses corresponding to trailing edges. If the position corresponding to the trailing edge of an un modulated pulse is counted as zero displacement, then the other trailing edges will arrive earlier or later. They will therefore have a time displacement other than zero; this time displacement is proportional to the instantaneous value of the signal voltage. The differentiated pulses corresponding to the leading edges are removed with a diode clipper or rectifier, and the remaining pulses, is position-modulated.

Circuit Description:-

Modulating Signal Generator:-

A 4.096 MHz clock is used to derive the modulating signal, which is generated by an oscillator circuit comparing a 4.096MHz crystal and three 74HC04(U9) inverter gates. This 4.096 MHz clock is then divided down in frequency by a factor of 4096, by binary counter 74HC4040(U4), to produce 50% duty cycle, 1 KHz square wave on pin no.1 of U4, and 2 KHz square wave on pin no.15. The frequency is selectable by means of SW1. This goes to input of fourth order low pass filter U3 (TL072) is used to produce sine wave from the square wave. The amplitude of this sine wave can be varied.

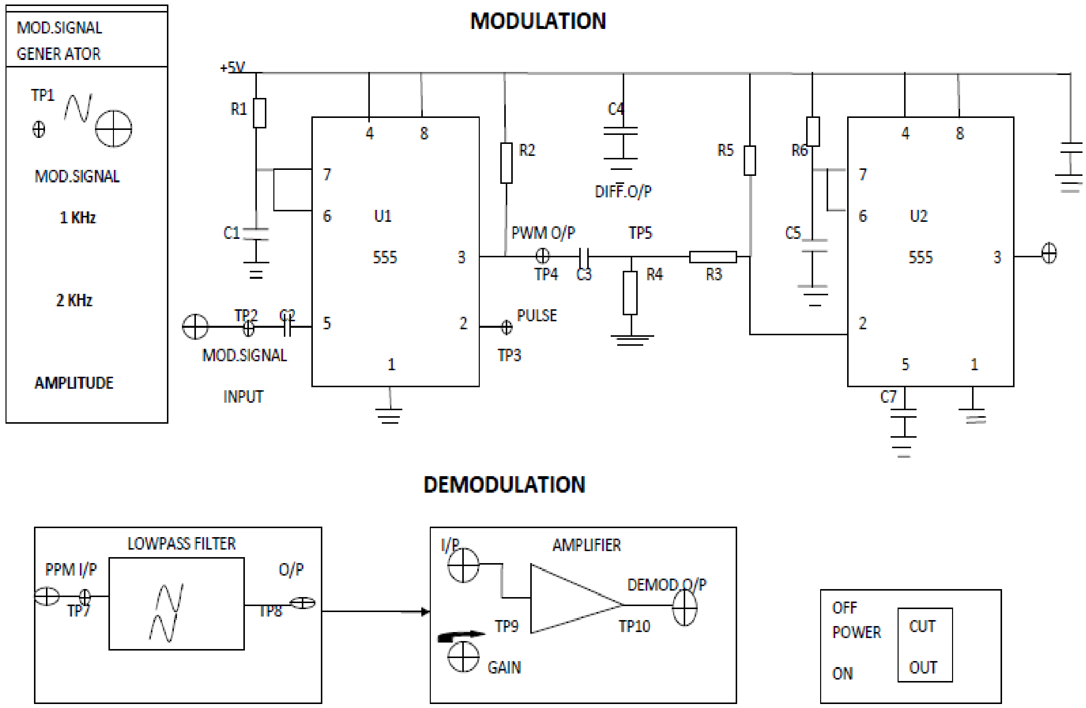
Modulation:-

The circuit uses the IC 555(U1) a Mono stable Multivibrator to perform the pulse position Modulation action.

The Modulating signal is given to Pin No. 5 at Pin No.2 the pulse is 32 KHz which is connected internally.

The PWM is available at TP2; this PWM output is differentiated by using differentiated circuit. This differentiated output is available at TP8. This differentiated output is fed to the 555 IC (U2) (Mono stable Mode) Pin No.2. The PPM output is available at TP3.

CIRCUIT DIAGRAM:



PROGRAM:-

```
% 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('original signal taken mesage,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]);
```

PROCEDURE:

Modulation:

1. Connect the circuit as shown in diagram 1.
   a. Connect the modulating signal generator output to modulating signal input (TP1) in PPM block.
   b. Keep the switch in 1 KHz position and amplitude pot in max position.
2. Switch ON the power supply
3. Observe the PWM output at TP2, and the differentiated output signal at TP8.
4. Now, monitor the PPM output at TP3.
5. Try varying the amplitude and frequency of sine wave by varying amplitude pot.
6. Repeat Step 5 for frequency of 2 KHz and observe the PPM output.
7. Switch OFF the power supply.

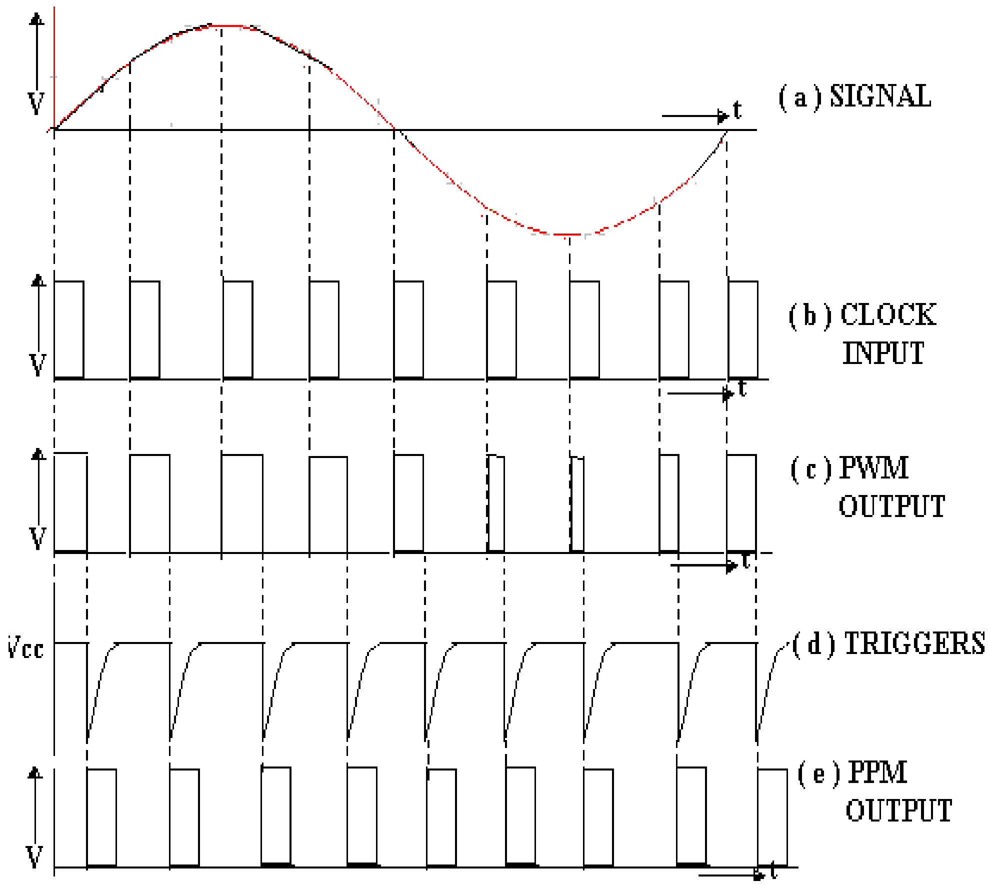
Demodulation:-

8. Connect the circuit as shown in diagram2.
   a. Connect the modulating signal generator output to modulating signal input (TP1) in PPM block.
   b. Keep the switch in 1 KHz position and amplitude pot in max position.
   c. Connect the PPM output (TP3) to input of LPF(TP4).
9. Switch ON the power supply
10. Observe the demodulated signal at the output of LPF at TP5.

11. Thus the recovered signal is true replica of the input signal
12. a. As the output of LPF has less amplitude, connect the output of LPF to the input of an AC amplifier (TP5 to TP6).

    b. Observe the demodulated out put on the oscilloscope at TP7 and also observe the amplitude of demodulated signal by varying gain pot. This is amplitude demodulated output.
13. Repeat the steps (7 to 9) for the modulating signal for frequency 2 KHz.
14. Switch OFF the power supply.

EXPECTED WAVEFORMS:



RESULT:

QUESTIONS:

1. What is the advantage of PPM over PWM?
2. Is the synchronization is must between Tx and Rx
3. Shift in the position of each pulse of PPM depends on what?

4. Can we generate PWM from PPM?
5. Why do we need 555 timers?
6. Does PPM contain derivative of modulating signal compared to PWM?
7. For above scheme, do we have to use LPF and integrator in that order?
8. If we convert PPM to PWM & then detect the message signal, will the o/p has less distortion?
9. Is synchronization critical in PPM?
10. How robust is the PPM to noise?

EXPERIMENT.NO-12 DATE:

FREQUENCY SYNTHESIZER

AIM: To study the operation of frequency synthesizer using PLL

APPARATUS :

1. Frequency synthesizer trainer Kit.
2. Dual trace C.R.O (20 MHZ)
3. Digital frequency counter or multimeter
4. Patch chords
5. PC with windows(95/98/XP/NT/2000)
6. MATLAB Software with communication toolbox

THEORY:

PLL stands for 'Phase locked loop' and it is basically a closed loop frequency control system, whose functioning is based on phase sensitive detection of phase difference between the input and output signals of controller oscillator.

Before the input is applied the PLL is in free running state. Once the input frequency is applied the VCO frequency starts change and phase locked loop is said to be in captured mode. The VCO frequency continues to change until it equals the input frequency and PLL is then in the phase locked state. When phase locked the loop tracks any change in the input frequency through its repetitive action.

Frequency Synthesizer:

The frequency divider is inserted between the VCO and the phase comparator. Since the output of the divider is locked to the input frequency f_{in}, VCO is running at multiple of the input frequency. The desired amount of multiplication can be obtained by selecting a proper divide by N network. Where N is an integer. For example $f_{out} = 5\ f_{in}$ a divide by N=10, 2 network is needed as shown in block diagram. This function performed by a 4 bit binary counter 7490 configured as a divide by 10, 2 circuit. In this circuit transistor Q1 used as a driver stage to increase the driving capacity of LM565 as shown in fig.b.

To verify the operation of the circuit, we must determine the input frequency range and then adjust the free running frequency F_{out} of VCO by means of R_1 (between

10^{th} and 8^{th} pin) and CI (9^{th} pin), so that the output frequency of the 7490 driver is midway within the predetermined input frequency range. The output of the VCO now should $5F_{in}$.

Free running frequency(f_0):

Where there is no input signal applied, it is in free running mode.

$F_0 = 0.3 / (R_tC_t)$ where R_t is the timing resistor

C_t is the timing capacitor.

Lock range of PLL(f_L)

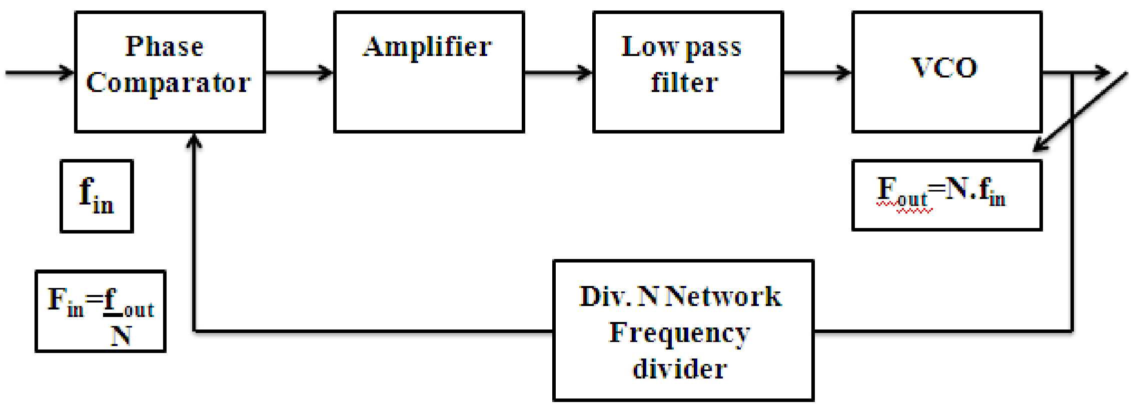
$F_L = \pm 8f_0/V_{cc}$ where f_0 is the free running frequency

$=2V_{CC}$

Capture range (f_C)

$$f_c = \frac{1}{2\pi}\sqrt{\frac{2\pi f_L}{3.6X10^3XC_c}}$$

CIRCUIT DIAGRAM:



PROGRAM:-

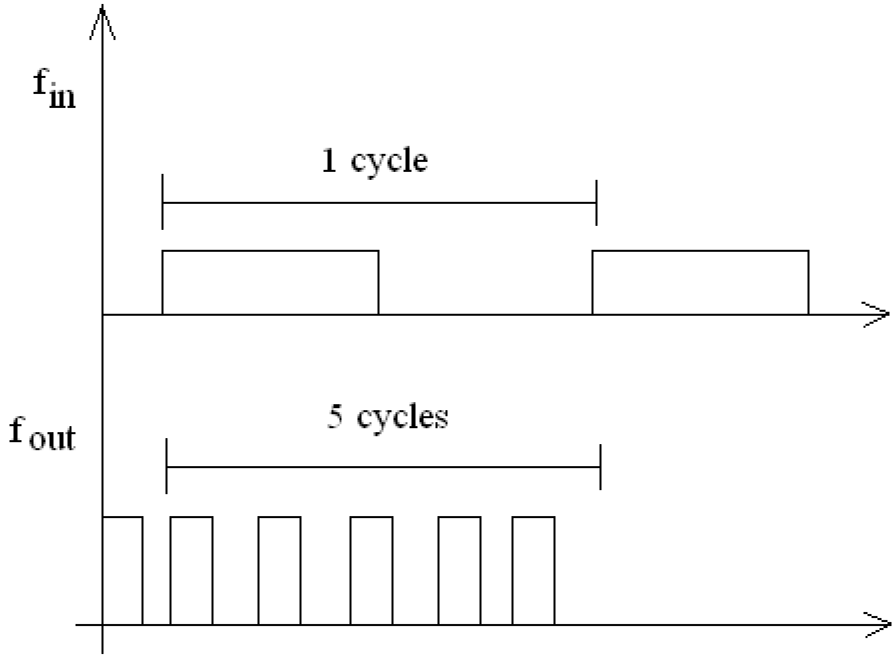
```
% program for frequency synthesizer
close all;
clear all;
clc
fs = 10000;
t = 0:1/fs:1.5;
f=50;
```

```
x1 = square(2*pi*f*t);
subplot(3,1,1)
plot(t,x1); axis([0 0.2 -1.2 1.2])
xlabel('Time (sec)');ylabel('Amplitude');
title('Square wave input with freq=50HZ');
t = 0:1/fs:1.5;
x2 = square(2*pi*2*f*t);
subplot(3,1,2)
plot(t,x2); axis([0 0.2 -1.2 1.2])
xlabel('Time (sec)');ylabel('Amplitude');
title('frequency multiplication by a factor of 2');
x3 = square(2*pi*f/2*t);
subplot(3,1,3)
plot(t,x3); axis([0 0.2 -1.2 1.2])
xlabel('Time (sec)');ylabel('Amplitude');
title('frequency division by a factor of 2');
```

PROCEDURE:

1. Switch on the trainer and verify the output of the regulated power supply i.e. $\pm$ 5V. These supplies are internally connected to the circuit so no extra connections are required.
2. Observe output of the square wave generator using oscilloscope and measure the range with the help of frequency counter, frequency range should be around 1 KHz to 10 KHz.
3. Calculate the free running frequency range of the circuit (VCO output between 4th pin and ground). For different values of timing resistor R1 (to measure R_t switch off the trainer and measure R_t value using digital multimeter between given test points) . and record the frequency values in tabular 1. $F_{out} = 0.3 /(R_tC_t)$ where R_t is the timing resistor and C_t is the timing capacitor =0.01 µf.
4. Connect 4th pin of LM 565 (F_{out}) to the driver stage and 5th pin (Phase comparator) connected to 11th pin of 7490. Output can be taken at the 11th pin of the 7490. It should be divided by the 10, 2 times of the f_{out}.

EXPECTED WAVEFORMS:



F_{in} KHz	F_{out} =N f_{in} KHz	Divided by 10,2

RESULT:

QUESTIONS:

1. What are the applications of PLL?
2. What is PLL?
3. Define Lock range of a PLL?
4. What is a VCO?
5. What are the applications of frequency synthesizer?
6. What is meant by the free running frequency of PLL?
7. What is the operation of a frequency synthesizer?
8. Which block is mainly used in frequency synthesizer?

EXPERIMENT NO-13 DATE:

AGC CHARADTERISTICS

AIM: To study the operation of AGC in communication system.

APPARATUS:

1. Trainer Kit
2. Dual trace oscilloscope
3. Digital multi meter.
4. PC with windows(95/98/XP/NT/2000)
5. MATLAB Software with communication toolbox

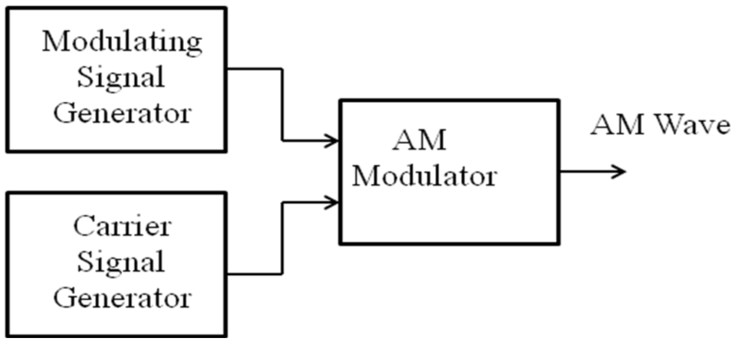
THEORY:

A Simple AGC is a system by means of which the overall gain of a radio receiver is varied automatically with the changing strength of the received signal, to keep the output substantially constant. A dc bias voltage, derived from the detector. The devices used in those stages are ones who’s trans-conductance and hence gain depends on the applied bias voltage or current. It may be noted in passing that, for correct AGC operation, this relationship between applied bias and trans-conductance need not to be strictly linear, as long as trans-conductance drops significantly with increased bias.

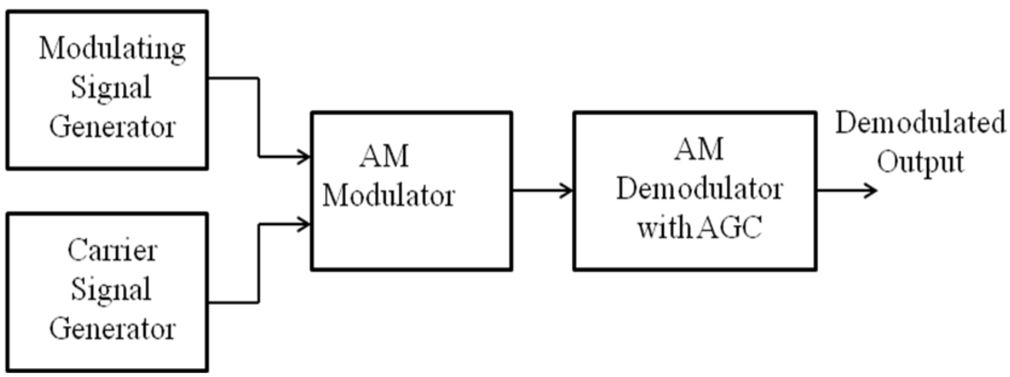
All modern receivers are furnished with AGC, which enables tuning to stations of varying signal strengths without appreciable change in the size of the output signal thus AGC “irons out” input signal amplitude variations, and the gain control dose not have to be re adjusted every time the receiver is tuned from one station to another, except when the change in signal strengths is enormous. In addition, AGC helps to smooth out the rapid fading which may occur with long-distance short-wave reception and prevents the overloading of last IF amplifier which might otherwise have occurred.

BLOCK DIAGRAM:

AM Modulator:



Demodulator:



PROGRAM:

```
% program for AGC
close all
clear all
clc
Fs = 100e3;     %sampling freq
t = 0:1/Fs:.1-1/Fs;             % time variable
Am=2;
fm = 100;  %fm 100 Hz
m = cos(2*pi*fm*t);     %message signal
Fc = 0.5e3;
% am modulation
 Ac = 8;
c=Ac.*cos(2*pi*Fc*t);   %carrier signal
```

```
figure;
% ploting message and carrier signals
subplot(2,1,1);
plot(c);
title('carrier');xlabel('time');ylabel('amplitude');
subplot(2,1,2);
plot(m);
title('message');xlabel('time');ylabel('amplitude');
figure;
% ploting AM modulated output
s = ammod(m,Fc,Fs,0,Ac);
subplot(2,1,1);
plot(s);
title('am modulation ');xlabel('time');ylabel('amplitude');
z = amdemod(s,Fc,Fs,0,Ac);
subplot(2,1,2);
plot(z);
title('am demodulation ');xlabel('time');ylabel('amplitude');
```

PROCEDURE:

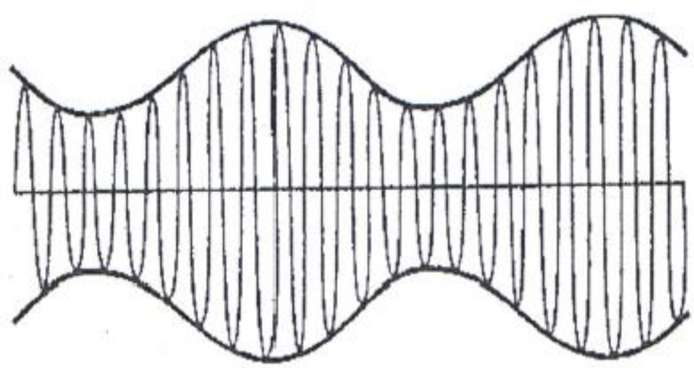
1. As the circuit is already wired you just have to trace the circuit according to the Circuit diagram given above Fig1.1.
2. Connect the trainer to the mains and switch on the power supply.
3. Measures the output voltages of the regulated power supply circuit i.e. +12v and -12v,+6@150ma.
4. Observe outputs of RF and AF signal generator using CRO, note that RF voltage is approximately 50mVpp of 455KHz frequency and AF voltage is 5Vpp of 1KHZ frequency.
5. Now vary the amplitude of AF signal and observe the AM wave at output, note the Percentage of modulation for different value of AF signal.

$$\%\text{Modulation} = (B-A)/(B+A) \times 100$$

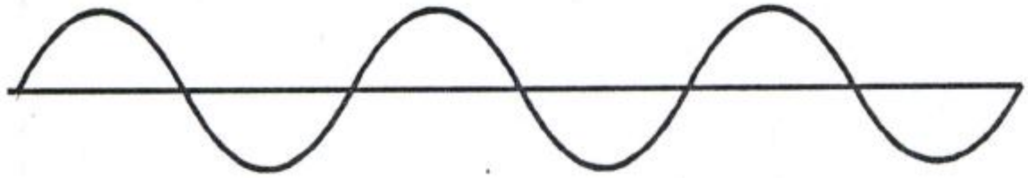
6. Now adjust the modulation index to 30% by varying the amplitudes of RF & AF Signals simultaneously.
7. Connect AM output to the input of AGC and also to the CRO channel-1.
8. Connect AGC link to the feedback network through OA79 diode
9. Now connect CRO channel-2 at output. The detected audio signal of1 KHz will be observed.
10. Calculate the voltage gain by measuring the amplitude of output signal (Vo) waveform, using formula A = Vo/Vi.
11. Now vary input level of 455 KHz IF signal and observe detected 1 KHz audio Signal with and without AGC link. The output will be distorted when AGC link removed i.e. there is no AGC action.
12. This explains AGC effect in Radio circuit.

EXPECTED WAVEFORMS:

.: AF Modulated RF Input.



: Detected Output With AGC:



RESULT:

QUESTIONS

1. What is the need for AGC in communications receivers?

2. Mention different types of AGC and suggest the best one ?

EXPERIMENT.NO-14 DATE:

PLL AS FM DEMODULATOR

AIM: To study the characteristics of PLL and calculate its capture range, lock range and free running VCO frequency.

APPARATUS:

1. PLL Trainer Kit
2. C R O (20MHz)
3. Digital Multimeter
4. PC with windows(95/98/XP/NT/2000)
5. MATLAB Software with communication toolbox

THEORY:

Phase Locked Loop is a versatile electronic servo system that compares the phase and frequency of a given signal with an internally generated reference signal. It is used in various applications like frequency multiplication, FM detector, AM modulator & De modulator and FSK etc.,

Free running frequency (f_0):

When there is no input signal applied to pin no:2 of PLL,it is in free running mode and the free running frequency is determined by the circuit elements R_t and C_t and is given by

$F_0 = 0.3/(R_tC_t)$ where R_t is the timing resistor

C_t is the timing capacitor

Lock range of PLL (f_L):

Lock range of PLL is in the range of frequencies in which PLL will remain lock, and this is given by

$f_L = \pm 8f_0 /V_{CC}$ Where f_0 is the free running frequency

$V_{CC} = V_{CC} -(- V_{CC})$

$= 2 V_{CC}$

Capture range(f_C):

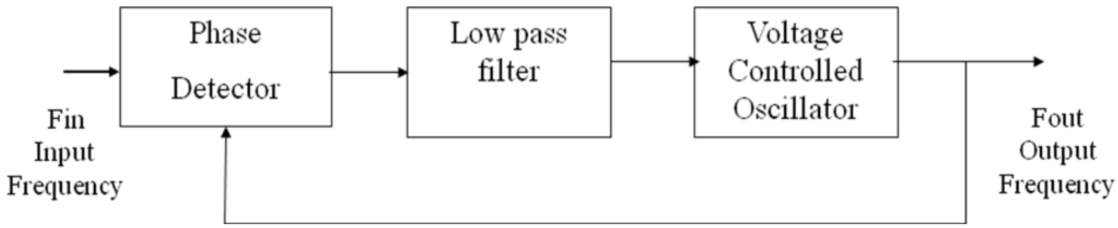
The capture range of PLL is the range of frequencies over which PLL acquires the lock. This is given by

$$f_C = \frac{1}{2\Pi}\sqrt{\frac{2\Pi f_L}{3.6x10^3 xC_C}}$$ Where f_L is the lock range and

C_C is filter capacitor

$R = 3.6 \text{ X } 10^3$

PLL BLOCK DIAGRAM:



PROGRAM:

```
close all;
clear all;
reg1 =0;
reg2 =0;
reg3 = 0;
eta =sqrt(2)/2;
theta =2*pi*1/100;
Kp = [(4*eta*theta)/(1+2*eta*theta+theta^2)];
Ki = [(4*theta^2)/(1+2*eta*theta+theta^2)];
d_phi_1 = 1/20;
n_data = 100;
for nn =1:n_data
phi1= reg1 +d_phi_1;
phi1_reg(nn) = phi1;
s1 =exp(j*2*pi*reg1);
s2 =exp(j*2*pi*reg2);
s1_reg(nn) =s1;
s2_reg(nn) =s2;
t =s1*conj(s2);
```

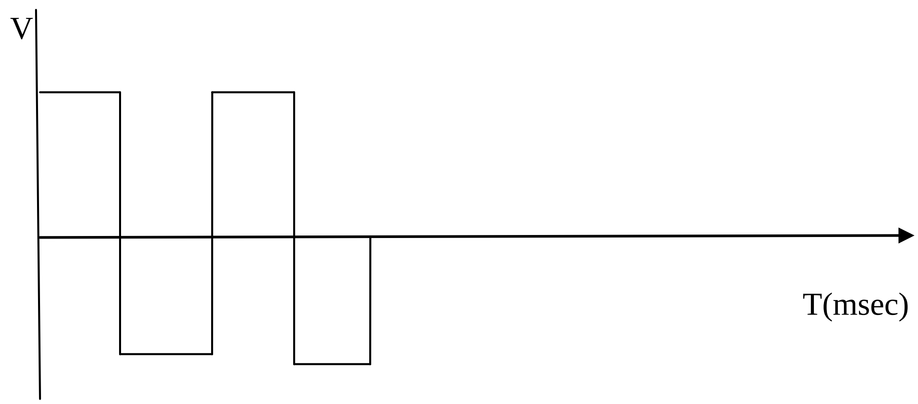
```
phi_error =atan(imag(t)/real(t))/(2*pi);
phi_error_reg(nn) = phi_error;
sum1 =Kp*phi_error + phi_error*Ki+reg3;
reg1_reg(nn) =reg1;
reg2_reg(nn) = reg2;
reg1 =phi1;
reg2=reg2+sum1;
reg3 =reg3+phi_error*Ki;
phi2_reg(nn) =reg2;
end
figure(1)
plot(phi1_reg);
hold on
plot(phi2_reg,'r');
hold off;
grid on;
title('phase plot');
xlabel('Samples');
ylabel('Phase');
figure(2)
plot(phi_error_reg);
title('phase Error of phase detector');
grid on;
xlabel('samples(n)');
ylabel('Phase error(degrees)');
figure(3)
plot(real(s1_reg));
hold on;
plot(real(s2_reg),'r');
hold off;
grid on;
```

```
title('Input signal & Output signal of VCO');
xlabel('Samples');
ylabel('Amplitude');
axis([0 n_data -1.1 1.1]);
```

OBSERVATIONS:

Theoritical values	Practical values
1.free running frequency	
Fo=1.2/4R1C1	
2.Lock in Frequency range	

EXPECTED WAVEFORM:



PROCEDURE:

1. Switch ON the experimental board by connecting the power card to the AC mains.
2. Then check the VCO output at pin 4.
3. This is a square waveform. The frequency of the wave from depends on C_T (0.01 μf) and R_5 (variable 10KΩ potentiometer).
4. Next Short pin 4 and pin 5. and give any signal of variable frequency and observe VCO output.
5. Change the input frequency and observe the VCO output on the CRO.

6. Between some frequencies, the VCO output is locked to the input signal frequency. This can be observed by increasing of decreasing the frequency of the VCO output by changing input frequency.
7. Before or after that frequency range, VCO output is not locked.
8. By changing the potentiometer provided on the board, locking frequency range can be changed.

RESULT:

QUESTIONS

1.What are the applications of PLL?

2.What is a PLL?

3.What is a VCO?

4.Define the lock range of a PLL?

5.Define the capture range of PLL?

6.Give the expression for free running frequency f_0 of a PLL?

7.What is meant by the free running frequency of a PLL?

8. Give the formulae for the lock range and capture range of the PLL?