PVP Siddhartha Institute of Technology

Kanuru, Vijayawada

Department of ECE

Communication Theory Lab Manual

(20EC3452)

Department of Electronics & Communication engineering
PVP Siddhartha Institute of Technology
Affiliated to JNTU Kakinada,
Approved by AICTE, New Delhi Accredited By NBA, NAAC A^+
ISO9001:2015 Certified Institute

List of Experments

Experiment No 1
Amplitude Modulation and Demodulation

Aim:

1. To generate amplitude modulated wave and determine the percentage modulation.
2. To demodulate the modulated wave using envelope detector.

Experimental Requirements:

SNo	Experimental Requirements	Range	Quantity
1.	Amplitude modulation & Demodulation kit	------	1
2.	Function Generator	(0-1)MHz	1
3.	CRO & Probes	(0-20)MHz	1
4.	Connecting Wires	-----------	7

Theory:

Amplitude Modulation is defined as a process in which the amplitude of the carrier wave c(t) is varied linearly with the instantaneous amplitude of the message signal m(t).The standard form of an amplitude modulated (AM) wave is defined by

$$S(t)=A_c(1+K_a\, m(t))\cos(2\pi f_c t)$$

where *Ka* is a constant called the amplitude sensitivity of the modulator. The demodulation circuit is used to recover the message signal from the incoming AM wave at the receiver. An envelope detector is a simple and yet highly effective device that is well suited for the demodulation of AM wave, for which the percentage modulation is less than 100%.Ideally, an envelop detector produces an output signal that follows theenvelop of the input signal wave form exactly; hence, the name. Some version of this circuit is used in almost all commercial AM radio receivers. The Modulation Index is defined as,

$$m = \frac{(E_{max} - E_{min})}{(E_{max} + E_{min})}$$

where Emax and Emin are the maximum and minimum amplitudes of the modulated wave.

Procedure:

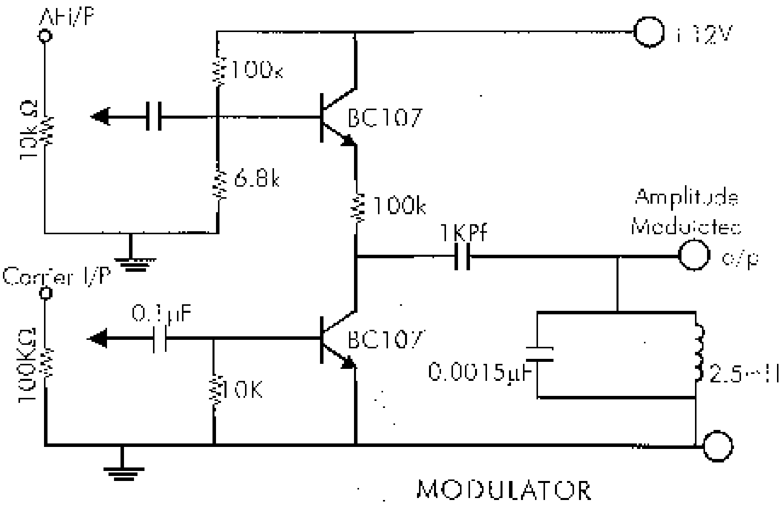
1. Switch on the trainer kit and check the o/p of carrier generator on oscilloscope.
2. Apply the 1KHz (2vp-p) A.F modulating signal to the AM modulation at AF i/p terminal
3. Connect the carrier signal (RF) at the carrier i/p of the modulator.
4. Connect the modulating (AF) signal to CH 1 and modulated signal (i.e, o/p of AM modulator) to CH 2 of a dual trace oscilloscope. Observe the o/p.
5. Calculate the maxima and minima points of modulated wave (o/p) on the CRO and thecalculate the depth of modulation using the formula.
6. Vary the modulating frequency and amplitude and observe the effects of the o/p modulated waveform.

$$\text{Modulation index}(\mu) = \frac{V_{max} - V_{min}}{V_{max} + V_{min}}$$

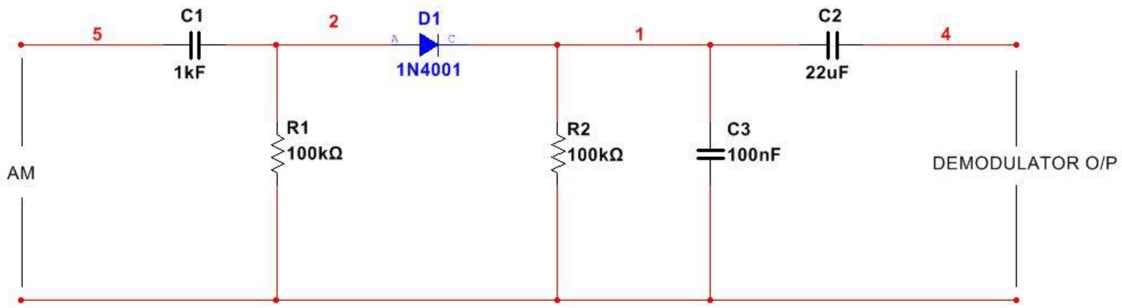
$$\%\ \text{Modulation} = \frac{V_{max} - V_{min}}{V_{max} + V_{min}} \times 100$$

7. The depth of modulation can be varied by varying the potentiometer provided at AFinput.
8. Repeat step 5 for 100% modulation, under modulation & over modulation.
9. Connect the o/p of the modulation circuit to the i/p of demodulator circuit and observe theo/p.
10. Connect the modulated signal (i/p demodulator) to CH 1 and (o/p of demodulator) to CH2
11. Observe the WAVEFORMS.

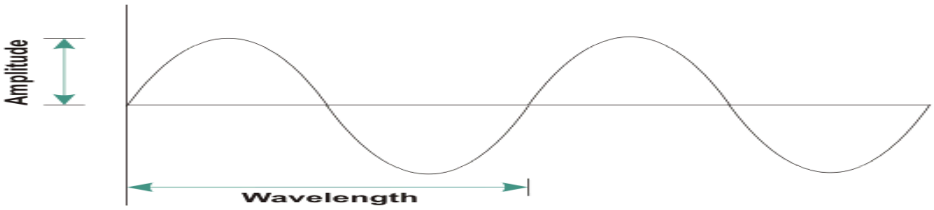
Circuit Diagram: **Modulator**



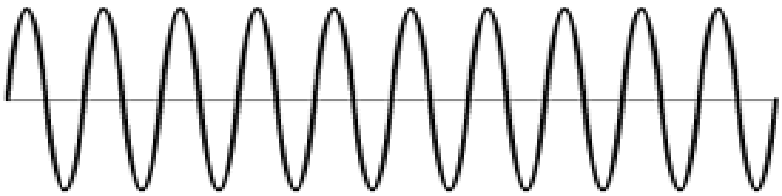
Demodulator



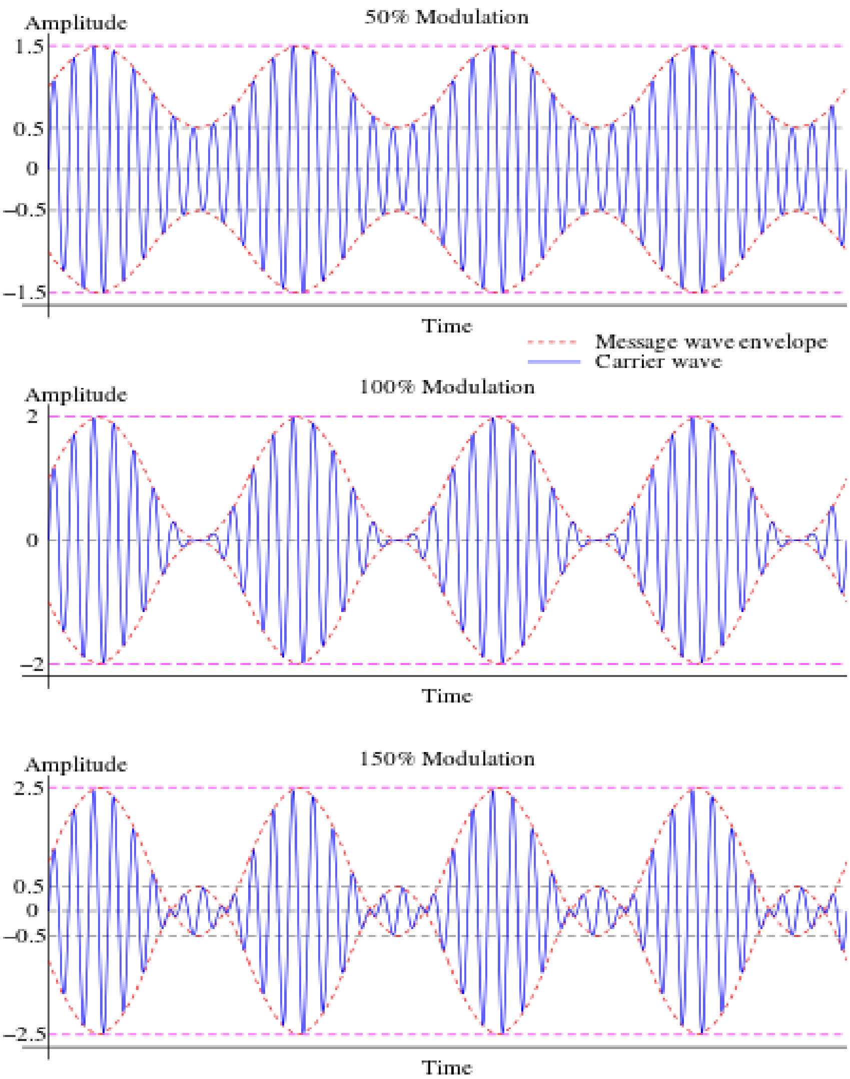
Expected Waveforms:



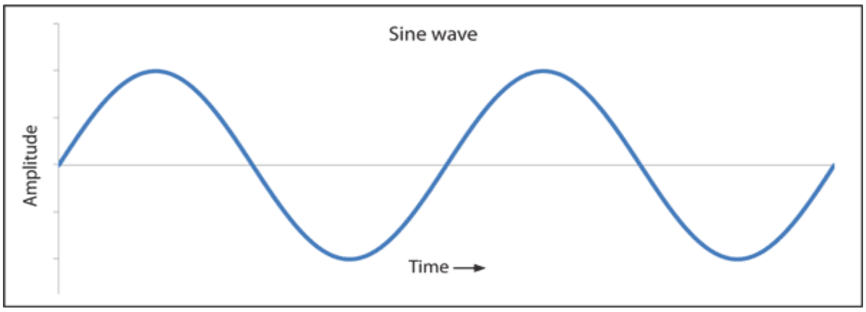
Message signal



Carrier signal



Types of Amplitude modulated waveforms



Demodulated signal

Tabular Column:

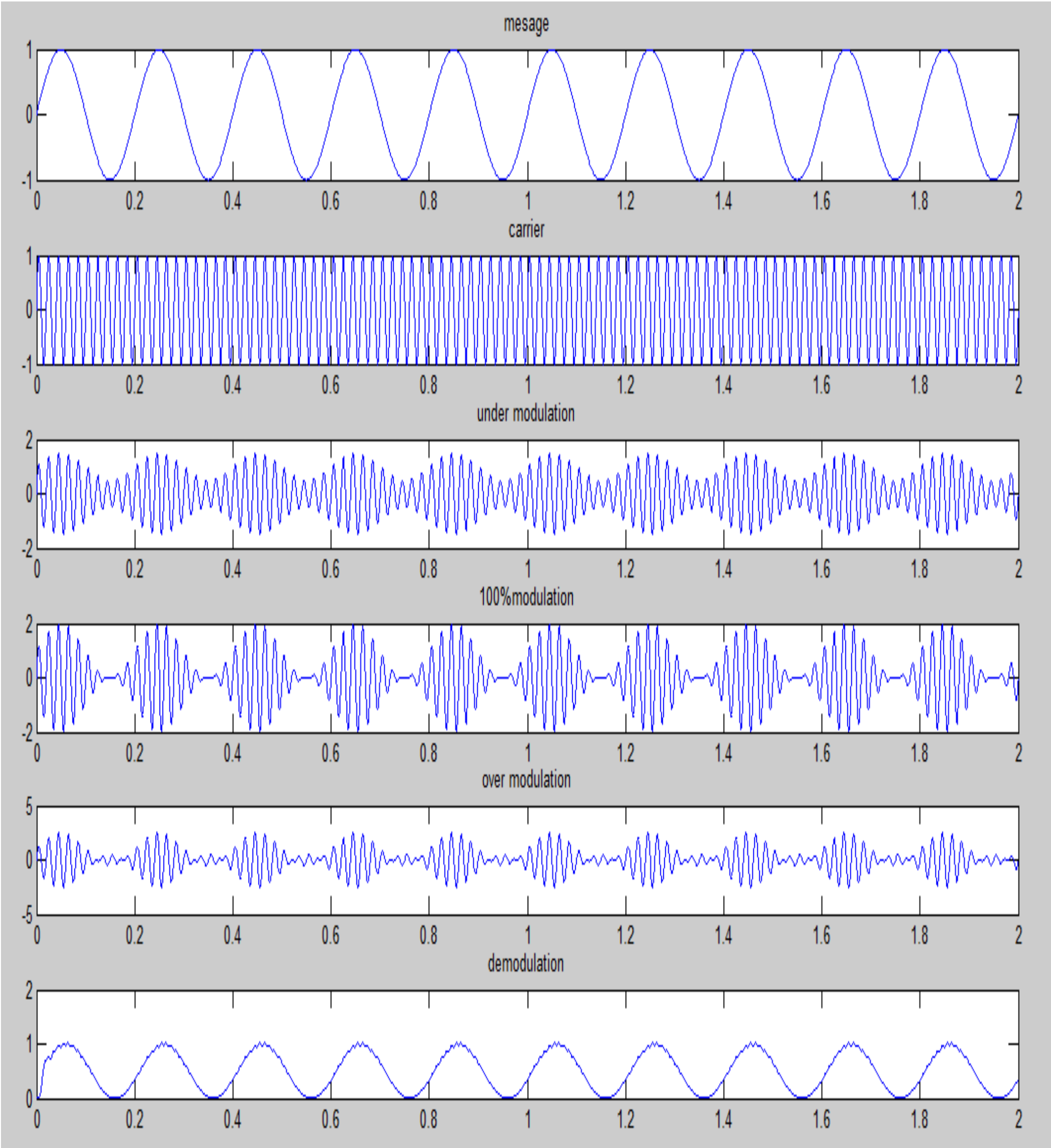
S. No	Vmax (Volts)	Vmin (Volts)	Theoritical μ= Vm/Vc	μ = $\frac{Vmax-Vmin}{Vmax+Vmin}$

Matlab Program:

```
clc;
clear all;
close all;
t=[0:0.001:2];
f1=5;
m=sin(2*pi*f1*t);
subplot(6,2,[1,2]);
plot(t,m);
title('mesage');
f2=50;
c=sin(2*pi*f2*t);
subplot(6,2,[3,4]);
plot(t,c);
title('carrier');
```

```
m1=0.5;

s1=(1+(m1*m)).*c;

subplot(6,2,[5,6]);

plot(t,s1);

title('under modulation');

m2=1;

s2=(1+(m2*m)).*c;

subplot(6,2,[7,8]);

plot(t,s2);

title('100%modulation');

m3=1.5;

s3=(1+(m3*m)).*c;

subplot(6,2,[9,10]);

plot(t,s3);

title('over modulation');

s5=s2.*c; [b,a]=butter(5,0.1);

s4=filter(b,a,s5);

subplot(6,2,[11,12]);

plot(t,s4);

title('demodulation' );
```

Expected Waveforms:



Precautions:

1. Check the connections before giving the power supply
2. Observations should be done carefully.

Pre Lab Questions:

1. Why modulation is an essential process of communication system?
2. Explain Block diagram of Communication system?
3. Explain need for modulation?
4. Define Amplitude modulation?

5. How carrier is differing from message?
6. What should be the modulation index of AM?

Post Lab Questions:

1. What are the distortions that are likely to be present in the demodulated outputwhen diode detector is used?
2. Explain how negative peak clipping occurs in the demodulated signal when diodedetector is used?
3. Explain under modulation, 100% modulation, over modulation?
4. Explain High level modulation?
5. Write the formulae to calculate practical modulation index?

Result:

Thus the depth of modulation is calculated using hardware kits and Matlab simulation program.

Experiment No 2
DSB-SC Modulation and Demodulation

Aim:

To generate AM-Double Side Band Suppressed Carrier (DSB-SC) signal using Balanced Modulator.

Experimental Requirements:

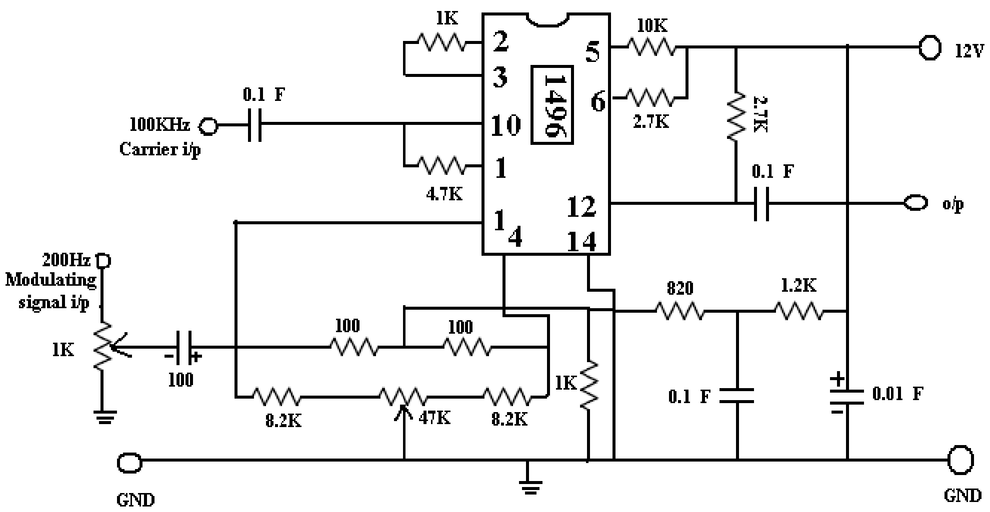
S.No	Experimental Requirements	Range	Quantity
1	Balanced modulator Trainer kit	--------	1
2	Function Generator	(0-1) MHz	1
3	C.R.O.	(0-20) MHz	1
4	Connecting wires.	-------	Required

Theory:

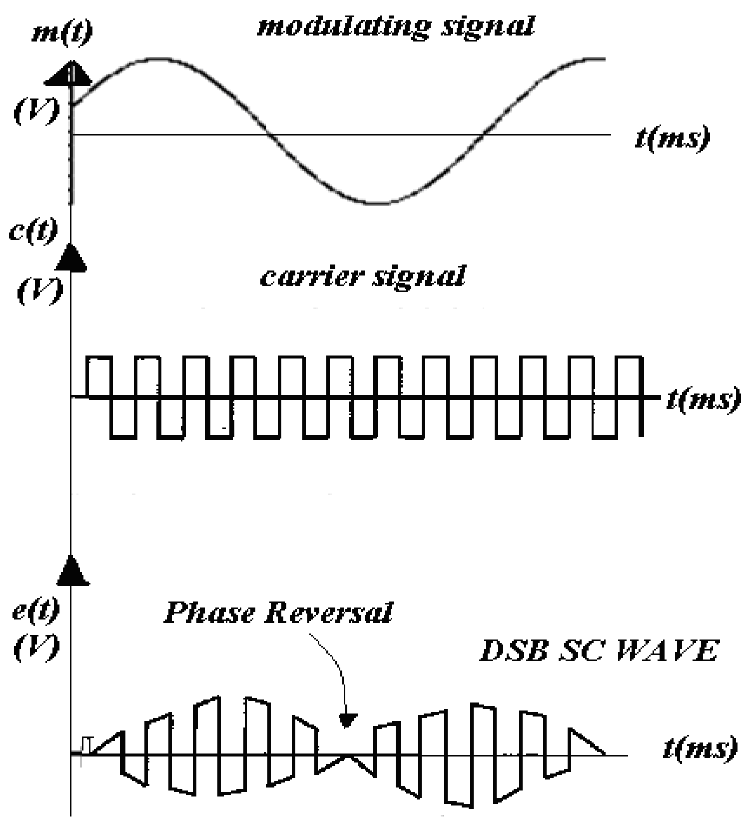
Balanced modulator is used for generating DSB-SC signal. A balanced modulator consists of two standard amplitude modulators arranged in a balanced configuration so as to suppress the carrier wave. The two modulators are identical except the reversal of sign of the modulating signal applied to them.

Circuit Diagram:

Balanced Modulator:



Expected Waveforms:



Procedure:

1. Switch on the balanced modulator trainer kit
2. Connect 200 Hz sine wave, and 100 KHz square wave from the function generators.Adjust R1 (1K linear pot). Connect oscilloscope to the output.
3. Vary R1 (1K) both clockwise and counter clockwise .Observe the output.
4. Disconnect the sine input to R1(1K) . The output should now be close to zero.
5. Increase the oscilloscope‟s vertical input sensitivity to measure the output voltage, E out carrier only.
6. Set the vertical input control to 1V /cm .Connect the sine input to R1 (1K) andadjust R1 for maximum output without producing clipping. Measure the peak side band output voltage Epk side bands = ------------
7. Calculate the carrier suppression in db.
   Suppression (db) = -20 log (Epk sideband/Eout carrier only)

Precautions:

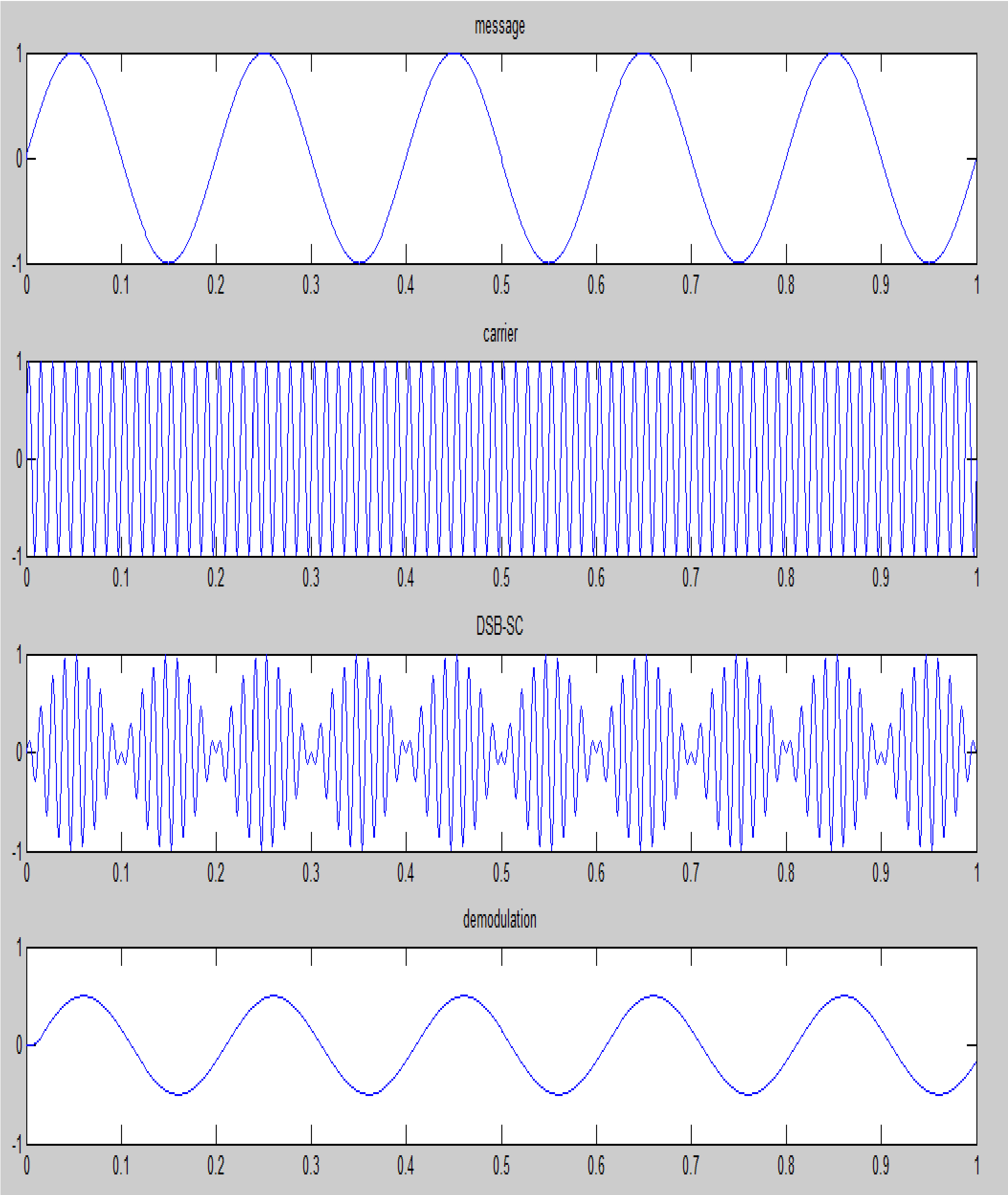
1. Check the connections before giving the power supply
2. Observations should be done carefully.

Matlab Program:

```
clc;
clear all;
close all;
t=[0:0.001:1];
f1=5;
m=sin(2*pi*f1*t);
subplot(4,2,[1,2]);
plot(t,m);
title('message');
f2=80;
c=sin(2*pi*f2*t);
subplot(4,2,[3,4]);
plot(t,c);
title('carrier');
s=m.*c;
subplot(4,2,[5,6]);
plot(t,s);
title('DSB-SC');
s1=s.*c;
[b,a]=butter(5,0.1);
s2=filter(b,a,s1);
subplot(4,2,[7,8]);
plot(t,s2);
```

```
title('demodulation');
```

Expected Waveforms:



Result:

The DSB-SC modulator is demonstrated and carrier suppression is calculated.

Experiment No 3

Frequency Modulation and Demodulation

Aim:

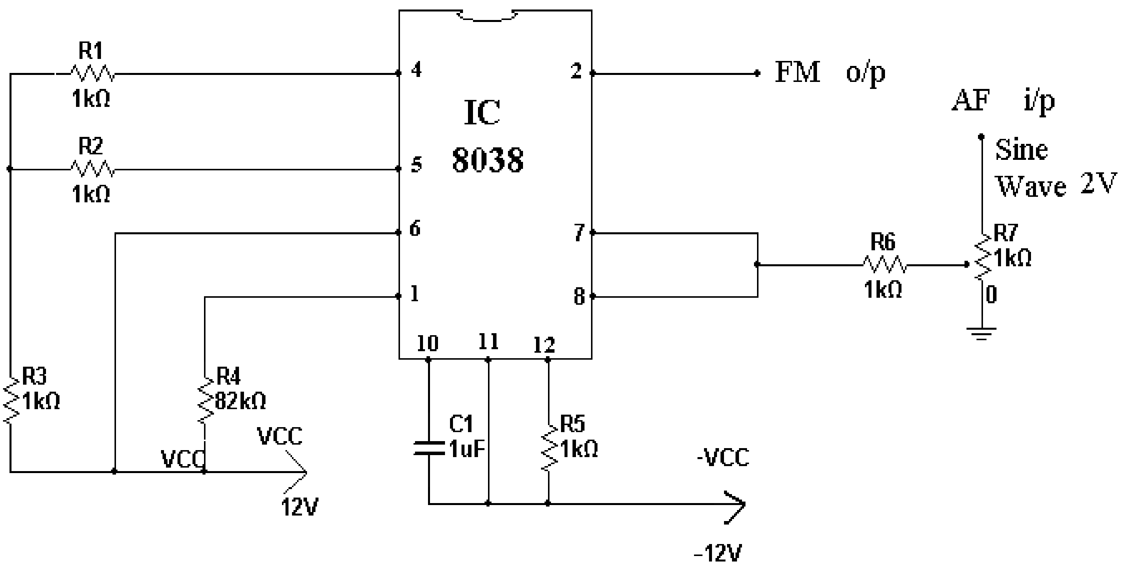
1. To generate frequency modulated signal and determine the modulation index andbandwidth for various values of amplitude and frequency of modulating signal.
2. To demodulate a Frequency Modulated signal using FM detector.

Experimental Requirements:

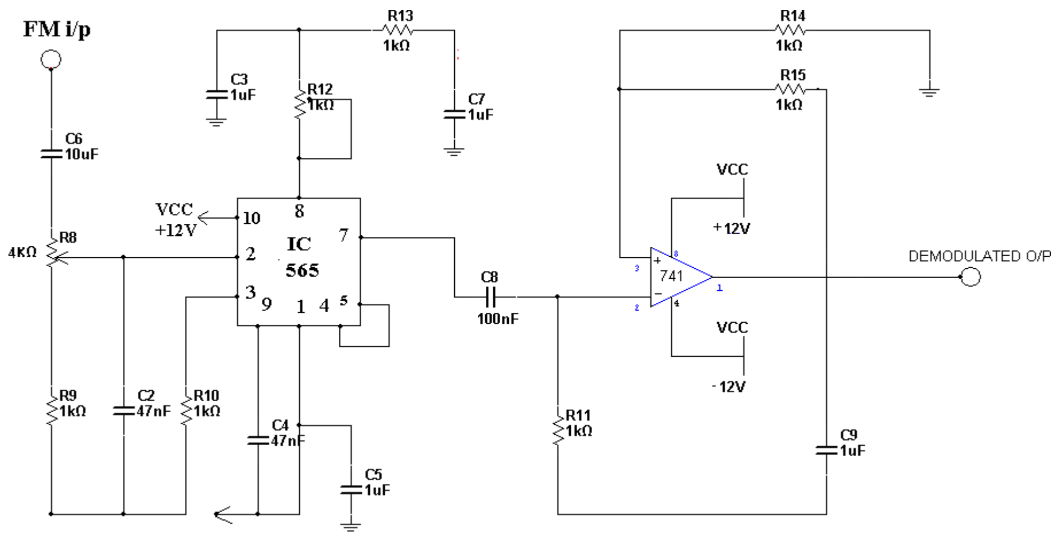
S. No	Experimental Requirements	Range	Quantity
1	Frequency modulation and Demodulation Trainer kit	--------	1
2	Function Generator	(0-1) MHz	1
3	C.R.O.	(0-20) MHz	1
4	Connecting wires.	-------	Required

Circuit Diagram:

FM Modulator



FM Demodulator



Theory:

Frequency modulation is a system in which the frequency of the carrier is varied in accordance with the signal amplitude. Let's assume for the moment that the carrier of the transmitter is at its resting frequency (no modulation) of 100MHz and we apply a modulating signal. The amplitude of the modulating signal will cause the carrier to deviate from this resting frequency by a certain amount. If we increase the amplitude of this signal, we will increase the deviation to a maximum of 75 kHz as specified by the FCC. If we remove the modulating voltage, the carrier shifts back to resting frequency (100MHz). From this we can say that the deviation of the carrier is proportional to the amplitude of the modulating voltage. The shift in the carrier frequency from its resting point compared to the amplitude of the modulating voltage is called the deviation ratio (a deviation ratio of 5 is the maximum) allowed in commercially broadcast FM) The rate at which the carrier shifts from its resting point to a no resting point is determined by the frequency of the modulating signal. The interaction between the amplitude and frequency of the modulating signal on the carrier is complex and requires the use of Bessel's function to analyze the results). If the modulating signal is 15kHz at a certain amplitude and the carrier shift is 75 kHz, the transmitter will produce eight significant sidebands. This is known as the maximum deviation ratio. If the frequency deviation of the carrier is known and the frequency of the modulating signal is known then

Modulation index = freq dev / freq AF

Procedure:

Modulation:

1. Switch on the frequency modulation trainer kit.
2. Connect oscilloscope to the FM o/p & observe the carrier frequency without any AFinput.
3. Now observe the frequency-modulated o/p on the CRO and adjust the amplitude of theAF signal to get clear frequency modulated waveform.
4. Apply a 1 KHz (2Vp-p) sine wave (AF) to the i/p of frequency modulator at AF input.
5. Vary the modulating signal frequency f_m and amplitude & observe the effects on themodulated WAVEFORMS.

Demodulation:

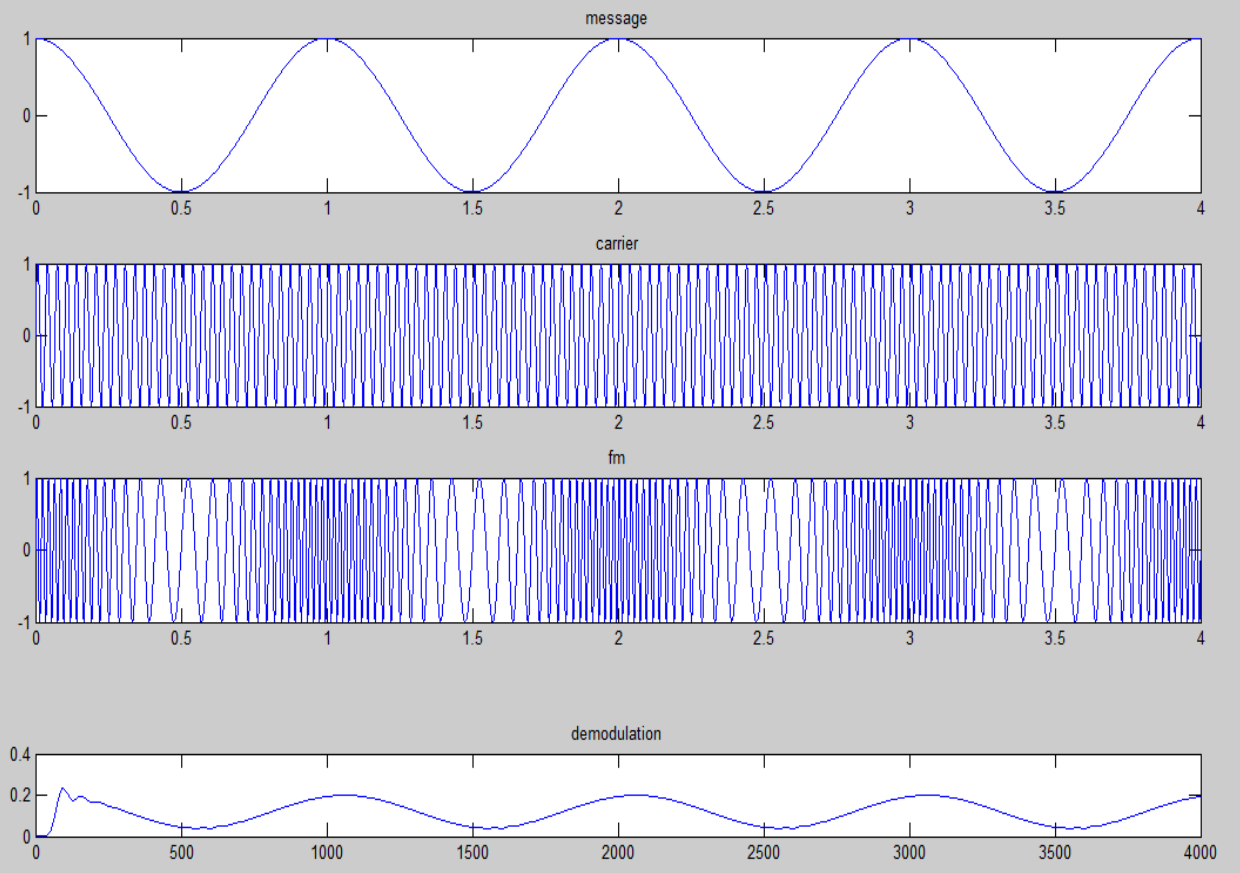
1. Connect the FM output to the input of the FM Demodulator. Observe the output of the demodulator on the CRO (Vary the potentiometer provide in the demodulator section).

Matlab Program:

```
clc;

clear all;

close all;

t=[0:0.001:4];

f1=1;

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

subplot(4,2,[1,2]);

plot(t,m);

title('message');

f2=30;
```

```
c=sin(2*pi*f2*t);
subplot(4,2,[3,4]);
plot(t,c);
title('carrier');
mf=20;
s=sin((2*pi*f2*t)+(mf*sin(2*pi*f1*t)));
subplot(4,2,[5,6]);
plot(t,s);
title('fm');
syms t1;
 x=diff(s);
y=abs(x);
[b,a]=butter(10,0.033);
s1=filter(b,a,y);
subplot(6,2,[11,12]);
plot(s1);
title('demodulation');
```

Expected Waveforms:



FM Modulation:

S.No.	Modulating Signal Voltage (V)	Carrier Freq (KHz)	Change In Freq(KHz)	Freq Dev (KHz)	Mf = Freq dev/fm

FM Demodulation:

S.No.	Mod-Voltage (E_m) in mv	Modulationg Frequency in (KHz)	Demodulated Signal Voltage	Demodulated Signal Frequency

Pre Lab Questions:

1. Why modulation is an essential process of communication system?
2. Explain Block diagram of Communication system?
3. Define Frequency modulation?
4. What are the advantages of FM over AM
5. What are applications of FM?

Post Lab questions:

1. Define Modulation Index?
2. Define Frequency Deviation?
3. When the amplitude of modulating signal increases then the effect on freq deviation?
4. Compare AM & FM
5. Compare NBFM & WBFM.

Lab Assignment:

1. Generate PM output using Frequency modulation?
2. Observe the spectrum and calculate BW?

Result:

The process of frequency modulation and demodulation is demonstrated and thefrequency deviation and modulation index is calculated

Experiment No 4
Pre-Emphasis and De-Emphasis

Aim:

a) To observe the effects of pre-emphasis on given input signal.

b) To observe the effects of De-emphasis on given input signal.

Experimental Requirements:

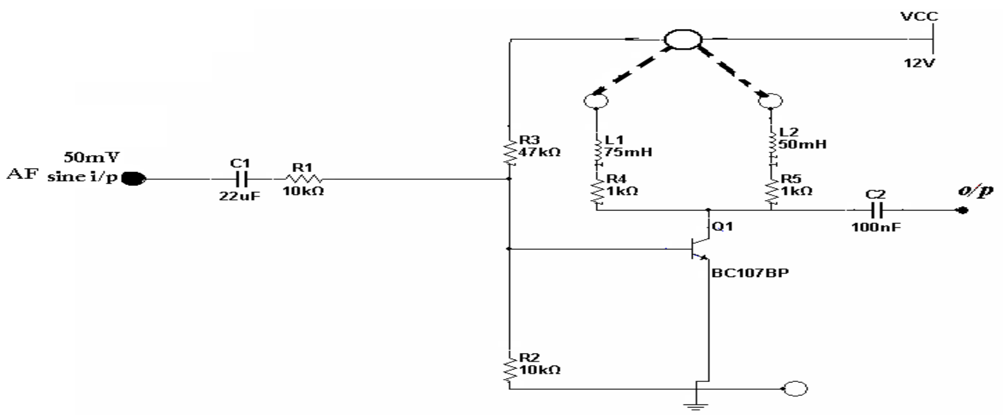
S. No.	Experimental Requirements	Range	Quantity
1	Pre - Emphasis and De - Emphasis Trainer kit	--------	1
2	Function Generator	(0-1) MHz	1
3	C.R.O.	(0-20) MHz	1
4	Connecting wires.	-------	Required

Theory:

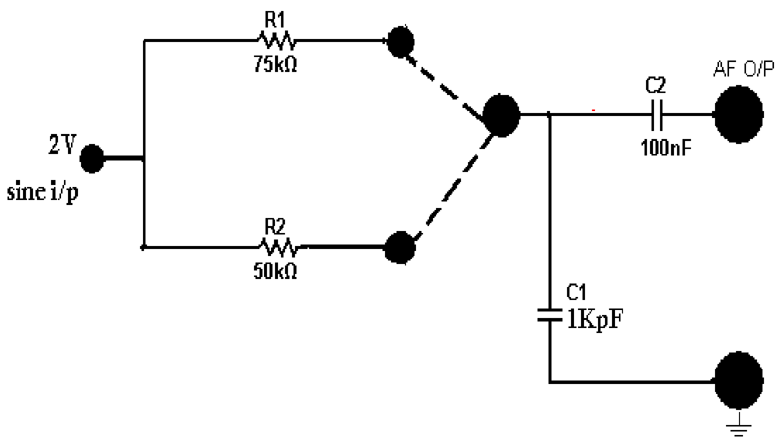
The noise triangle shows that noise has greater effect on the higher frequencies than on lower ones. Thus if higher frequencies were artificially boosted at the transmitter and correspondingly cut at the receiver, an improvement in noise immunity could be expected. This boosting of the higher modulating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

Circuit Diagram:

Pre – Emphasis:



De – Emphasis:



Procedure:

Pre-Emphasis:

1. Switch on pre-emphasis and De- emphasis trainer
2. Connect AF signal to the input of the pre-emphasis circuit (say 75 μ sec)
3. Connect CH I input of CRO to the input of the pre-emphasis network .
4. Adjust the AF signal to the required amplitude level (say 4mv,6m --------)
5. Observe the output waveform on CRO CH I by connecting either75 or 50mH.
6. By varying the AF signal frequency (keeping amplitude constant) in

steps note down the corresponding i/p and o/p voltage in tabulated form as shown below.

7. Plot the graph between frequency (X-axis) and o/p voltage (Y-axis).
8. From the graph note the frequency at which o/p voltage is 70.7 %of i/p voltageand compare it with theoretical frequency. if (theoretical) = $1/2\pi RC$ (or) $R/2\pi L$
9. Repeat the above steps for 50 μ sec ,for pre-emphasis network.Where RC (or) L/R is time constant .

   L = 75mH; R=1kΩ for 75 μ sec, L=50mH; R=kΩ for 50 μ sec

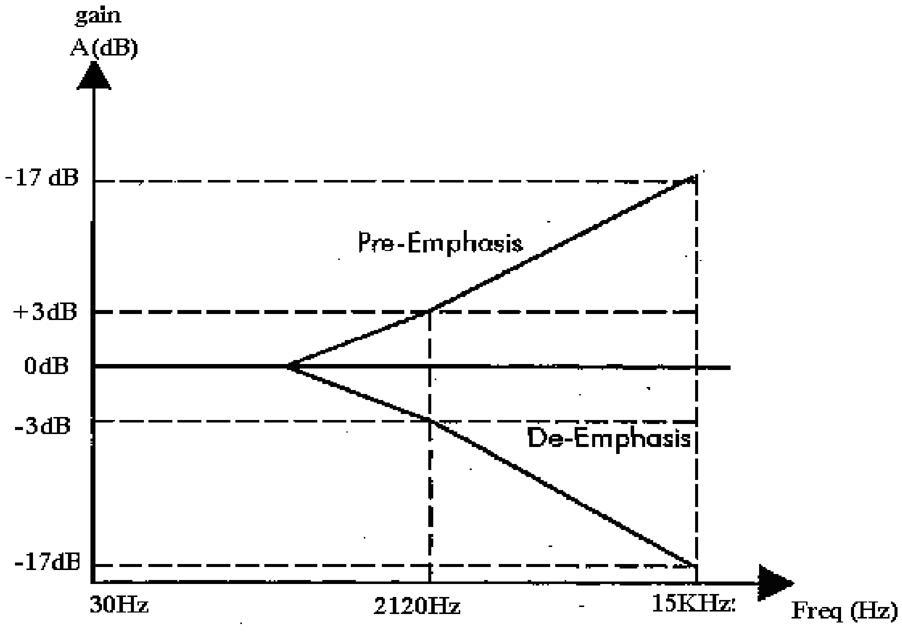
De-Emphasis:

1. Connect the o/p of pre-emphasis to the i/p of the De-emphasis circuit.
2. Connect CH I i/p of CRO to the i/p of De-emphasis network & i/p to the o/p of De- emphasis network.
3. By varying the AF signal frequency (keeping amplitude constant) in steps note down thecorresponding i/p & o/p voltages in tabulated form as shown below.
4. Plot the graph between log frequency on X – axis and attenuation on Y-axis to show theemphasis curve.
5. From the graph note the frequency at which the o/p voltage is 70.7 % of i/p voltage andcompare it with the theoretical frequency.
6. Repeat above steps for 50 μ sec.
7. The theoretical frequency (f) = $1/2\pi RC$

   R = 75 kΩ ; C = 1nf; Time constant = 75 μ sec

   R = 50kΩ ; C = 1nf; Time constant = 50 μ sec

Expected Graph:



75-μs Emphasis Curves

Tabular Column:

PRE EMPHASIS				DE EMPHASIS			
Vi=50mV				Vi=2V			
Frequency (Hz)	I/p voltage (Vpp)	O/p voltage (Vpp)	Attenuation (db)=20log (Vp/Vi)	Frequency (Hz)	I/p voltage (Vpp)	O/p voltage Vpp)	Attenuation (db)=20log (Vp/Vi)
100HZ				100HZ			
200				200			
400				400			
600				600			
800				800			
1k				1k			
2k				2k			
3k				3k			
4k				4k			
5k				5k			
6k				6k			
7k				7k			
8k				8k			

9k				9k			
10k				10k			
12k				12k			
14k				14k			
16k				16k			
18k				18k			
20k				20k			

Precautions:

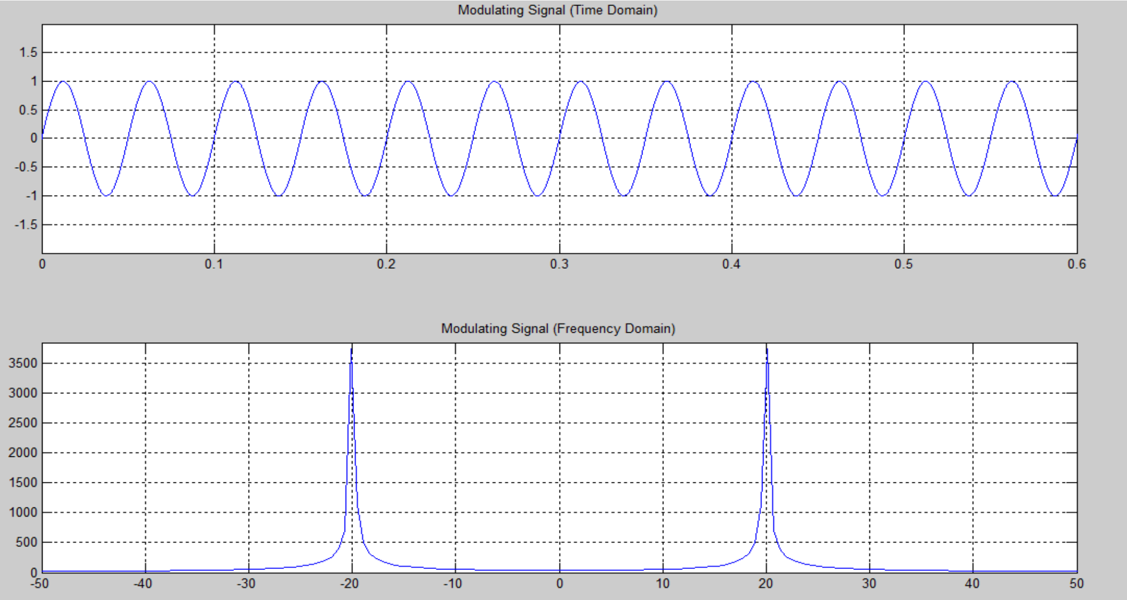
1. Check the connections before giving the power supply
2. Observation should be done carefully

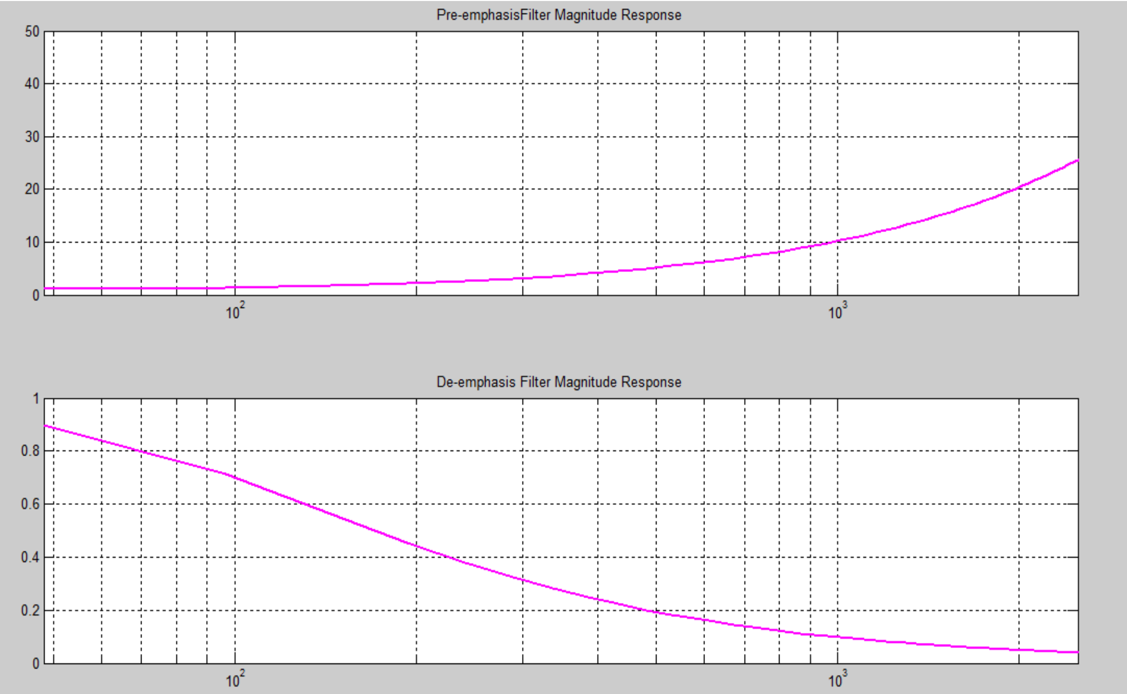
Matlab Program:

```
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=10;

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-emphasisFilter 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')
```

Expected Waveforms:





Pre Lab Questions:

1. Define Frequency Deviation?
2. Define Modulation Index?
3. Define Frequency modulation?
4. What are the advantages of FM over AM
5. Explain high pass & low pass filters

Post Lab Questions:

1. Explain Pre-emphasis
2. Explain De-emphasis
3. Define noise triangle?
4. Define 3-dB frequency
5. Why FM having greater noise immunity?

Lab Assignment:

1. Observe pre-emphasis output for L=100mH
2. Observe de-emphasis output for c=100µF

Result:

The frequency response curve of pre-emphasis and de-emphasis is demonstrated

Experiment No 5
Study of Spectrum Analyser

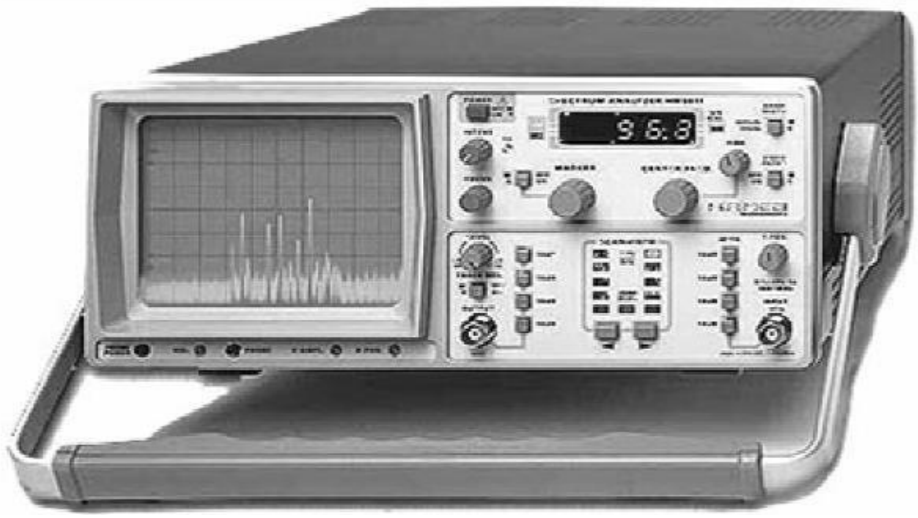
Aim: To study the operation of spectrum analyzer.

Experimental Requirements:

S.No	Experimental Requirements	Range	Quantity
1	Spectrum Analyzer	------	1
2	Any modulation system kit		1
3	Connecting wires.	-------	Required

Introduction:

To analyze the AM and FM waveform using spectrum analyzer. The oscilloscope is the most common device used to display the signals, with time as x-axis. The signals which require time as x-axis, to display them are time domain signals. The signals which require frequency as x-axis, to display them are called frequency domainsignals. Frequency domain display of signal consists of information of energy distributed of the signal. The analysis of such a frequency domain display of the signal is called spectral analysis of the signal. Thus the study of the energy distribution across the frequency spectrum if a given signal is defined as the spectral analysis. The instrument which graphically provides the energy distribution of a signal as a function of frequency on its CRT is called spectrum analyzer.



HAMEG 5010 SPECTRUM ANALYZER

It provides a calibrated graphical display with the frequency on horizontal axis and the signal component on the vertical axis, the sinusoidal components of which, the signal ismade up of, are displayed as the vertical lines against frequency coordinates. The frequency of each vertical line gives the absolute amplitude if the component while the horizontal location gives that particular frequency.

The analysis of electrical signals is a fundamental problem for many engineers and scientists. Even if the immediate problem is not electrical, the basic parameters of interest are often changed into electrical signals by means of transducers. The rewardsfor transforming physical parameters to electrical signals are great, as many instruments are available for the analysis of electrical signals in the time and frequency domain. The traditional way of observing electrical signals is to view them in time domain using oscilloscope. The time domain is used to recover relative timing and phase information which is needed to characterize electrical circuit behavior. How ever, not all circuit can be uniquely characterized from just time domain information. Circuit elements such as amplifiers, oscillators, mixers, modulators, detectors and filters are best characterized by their frequency response information. This frequency information is best obtained by viewing electrical signals in frequencydomain. To display the signal in the frequency domain requires a device that can discriminate between frequency domains is the spectrum analyzer. It graphicallydisplays the voltage or power as a function of frequency on a CRT. In the time domain, all frequency components of a signal are seen summed together. In the frequency domain, complex signals are separated in to their frequency components, and power level at each frequency is displayed. The frequency domain is a graphical representation of signal amplitudes as a function of frequency. The frequency domain contains information not found in time domain.

Types of Spectrum Analysers:

There are two basic types of spectrum analyzers, swept-tuned and real time analyzers. The swept-tuned analyzers are tuned by electrically sweeping them over their frequency range. Therefore the frequency components of a spectrum are sampled sequentially in time. This enables periodic and random signals to be displayed, but makes it impossible to display transient response. Real time analyzers, on the other hand, simultaneously display the amplitude of all signals in the frequency range of theanalyzer: hence the name real-time. This preserves the time dependency between signals which permits information to be displayed.

Real time analyzers are capable of displaying transient response as well as periodic and random signals. The swept tuned analyzers are usually of the TRF (tuned radio frequency) or super heterodyne type. A TRF analyzer consists of a frequency range, a detector to produce vertical deflection on a CRT, and a horizontal scan generator used to synchronize the tuned frequency to the CRT horizontal deflection. It is a simple, inexpensive analyzer with wide frequency coverage, but lacks resolution and sensitivity. Because TRF analyzers have swept filter they are limited in sweep width.

Applications of Spectrum Analyser:

1. **Modulation measurements**:
When the frequency scan of spectrum analyzer is set to zero and x-axis is representingtime instead of frequency, it operates as a fixed tuned receiver to measure amplitude against time. This is called its synchroscope mode. When analyzer is tuned to carrier frequency with bandwidth at least twice that of modulation frequency and with a linear display, the envelop of an AM signal is observed. Measuring the peak VP and through VT, modulation index can be determined. When operated in normal mode, two sidebands separated from the carrier by modulation frequency fm are observed. The modulation index can be calculated from the sidebands and carrier amplitude. Similarly it can be used to calculate the distortion occurring in modulation process. The sideband configuration in frequency modulation enables observer to calculate the frequency modulation index.
2. **Continuous wave signal frequency stability**
The frequency drift of a signal can be measured by observing the excursions of the signal across the display. Over period of minutes, it gives long term stability while over period of seconds it gives short term stability.
3. **Harmonic distortion measurement**
The distortion affects the frequency components of a signal to be transmitted. The harmonics appear as the additional signals in the spectrum analyzer at multiples of the carrier frequency. To keep it low, its measurement plays an important role. Thespectrum analyzer can be used to make such distortion measurements.
4. **Noise measurement**
The noise can be measured with very straightforward method using the spectrum analyzer. Similarly the measurement of impulse noise also can be measured using spectrum analyzer. The examples of impulse noise in the generation of voltage spikes due to engine

ignition and electric motor commutation.

5. Examining Pulse Modulation

This is the first application of spectrum analyzer. The spectrum analyzer can be used to Measure or evaluate the quality of the pulse modulation. The difficult task ofmeasuring pulse Modulation of radar transmitters is possible due to spectrumanalyzer. Apart from these common applications it is used in the following applications as well.

1. In the fields of biomedical electronics, geological surveying, oceanography. It is used to analyze the water and air pollution.
2. It is used to measure the antenna pattern.
3. It is used to tune the parametric amplifier
4. It is used to examine the vibration signals from the automobiles, airplanes, space vehicles bridges and other mechanical systems. It provides useful information about mechanical integrity, unbalance and bearing, gear wear.
5. It finds number of applications in the field of electronic testing related to trouble shooting and quality control.

Experiment No 6

Single Side Band Modulation and Demodulation

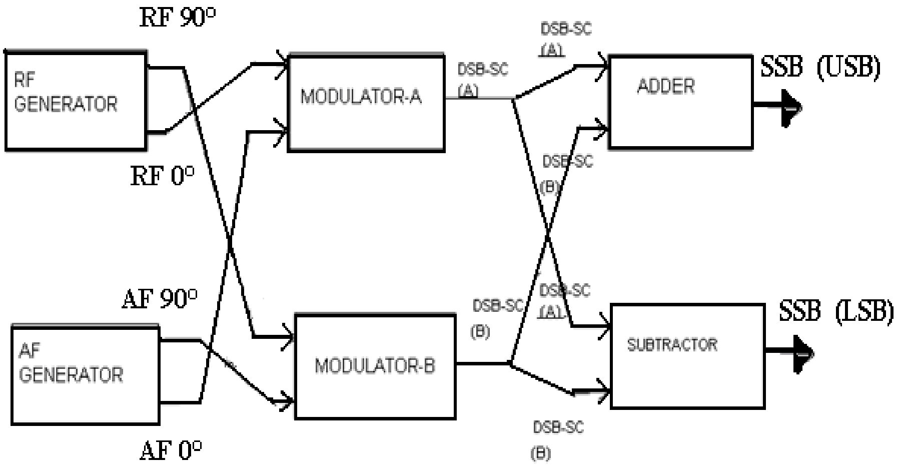
Aim:

To generate the SSB modulated wave using Phase shift method and demodulate the SSB modulated wave.

Experimental Requirements:

S. No	Experimental Requirements	Range	Quantity
1	Single Side Band trainer kit	--------	1
2	C.R.O.	(0-20) MHz	1
3	Connecting wires.		10

Circuit Diagram:



Procedure:

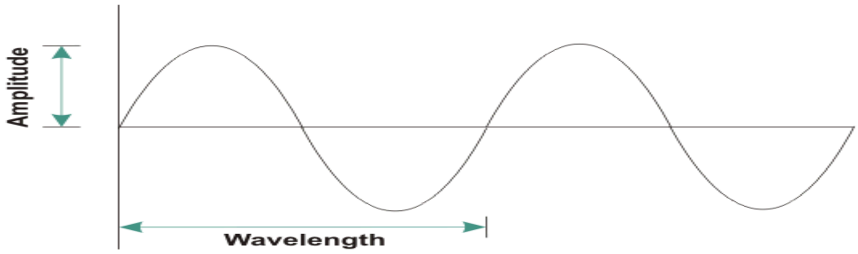
1. Switch on the trainer and measure the output of the regulated power supply i.e., ±12V and -8V.
2. Observe the output of the RF generator using CRO. There are 2 outputs from the RF generator, one is direct output and another is 90o out of phase with the direct output. The output frequency is 100 KHz and the amplitude is ≥ 0.2VPP. (Potentiometers are provided to vary the output amplitude).
3. Observe the output of the AF generator, using CRO. There are 2 outputs from theAF generator, one is direct output and another is 90o out of phase with the direct output.

Aswitch is provided to select the required frequency (2 KHz, 4KHz or 6 KHz). AGC potentiometer is provided to adjust the gain of the oscillator (or to set the output to good shape). The oscillator output has amplitude 10VPP. This amplitude can be varied using the potentiometers provided.

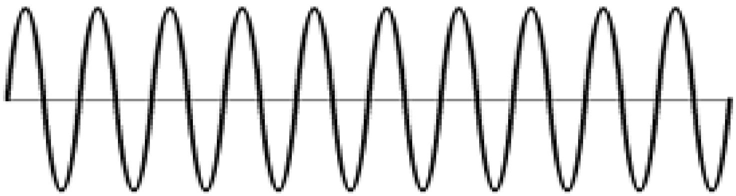
4. Measure and record the RF signal frequency using frequency counter. (or CRO).
5. Set the amplitudes of the RF signals to 0.1 Vp-p and connect direct signal to one balancedmodulator and 90o phase shift signal to another balanced modulator.
6. Select the required frequency (2KHz, 4KHz or 6KHz) of the AF generator with the helpof switch and adjust the AGC potentiometer until the output amplitude is 10 VPP (when amplitude controls are in maximum condition).
7. Measure and record the AF signal frequency using frequency counter (or CRO).
8. Set the AF signal amplitudes to 8 Vp-p using amplitude control and connect to thebalanced modulators.
9. Observe the outputs of both the balanced modulators simultaneously using Dual trace oscilloscope and adjust the balance control until desired output wave forms (DSB-SC).
10. To get SSB lower side band signal, connect balanced modulator output (DSB_SC) signalsto subtractor.
11. Measure and record the SSB signal frequency.
12. Calculate theoretical frequency of SSB (LSB) and compare it with the practical value LSB frequency = RF frequency – AF frequency
13. To get SSB upper side band signal, connect the output of the balanced modulator to thesummer circuit.
14. Measure and record the SSB upper side band signal frequency.
15. Calculate theoretical value of the SSB(USB) frequency and compare it with practicalvalue.

USB frequency = RF frequency + AF frequency generator, one is direct output andanother is 90° out of phase with the direct output. A switch is provided to select the required frequency (2 KHz, 4KHz or 6 KHz). AGC potentiometer is provided to adjust the gain of the oscillator (or to set the output to good shape). The oscillator output has amplitude 10VPP. The amplitude can be varied using the potentiometers provided. USB frequency = RF frequency + AF frequency

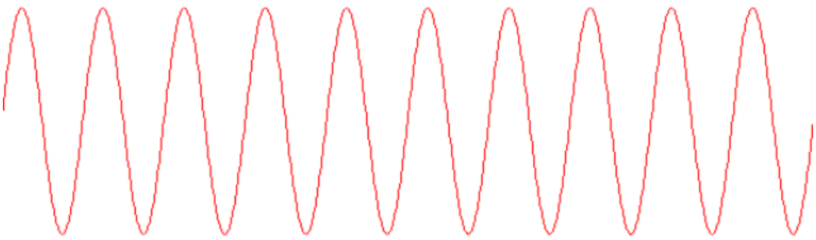
Expected Waveforms:



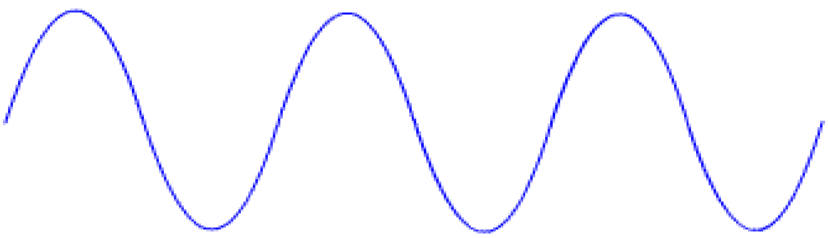
Message signal



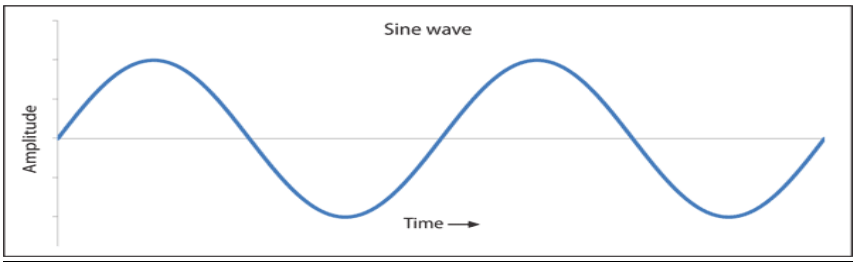
Carrier Signal



SSB USB Signal



SSB LSB Signal



Demodulated signal

Tabular Column:

Signal	Amplitude (Volts)	Frequency (KHz)
Message Signal		
Carrier Signal		
SSB (LSB)		
SSB (USB)		

Matlab Program:

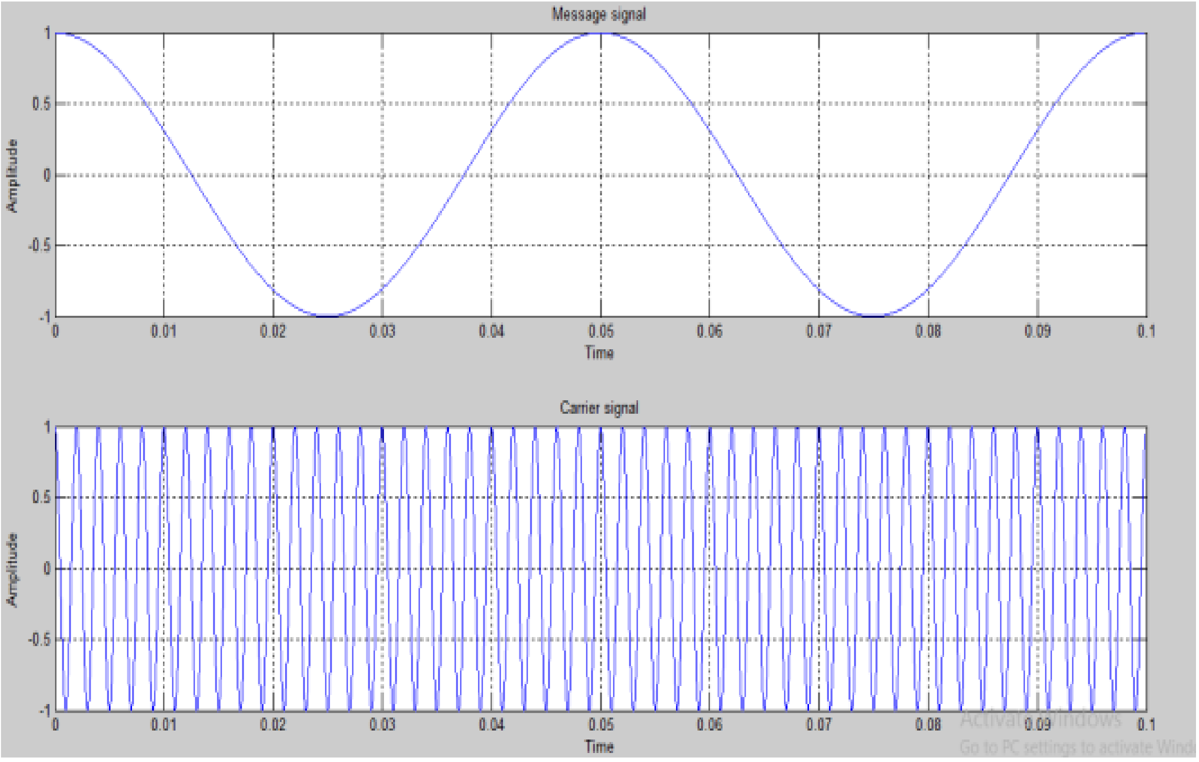
```
fm=20;
fc=500;
vm=1;
vc=1;
mu=0.1;
t=0:0.00001:0.0999;
f=0:1:9999;
m=vm*cos(2*pi*fm*t); %% message
mp=vm*sin(2*pi*fm*t);
c=vc*cos(2*pi*fc*t); %% carrier
cp=vc*sin(2*pi*fc*t);
ss1=m.*c;
ss2=mp.*cp;
```

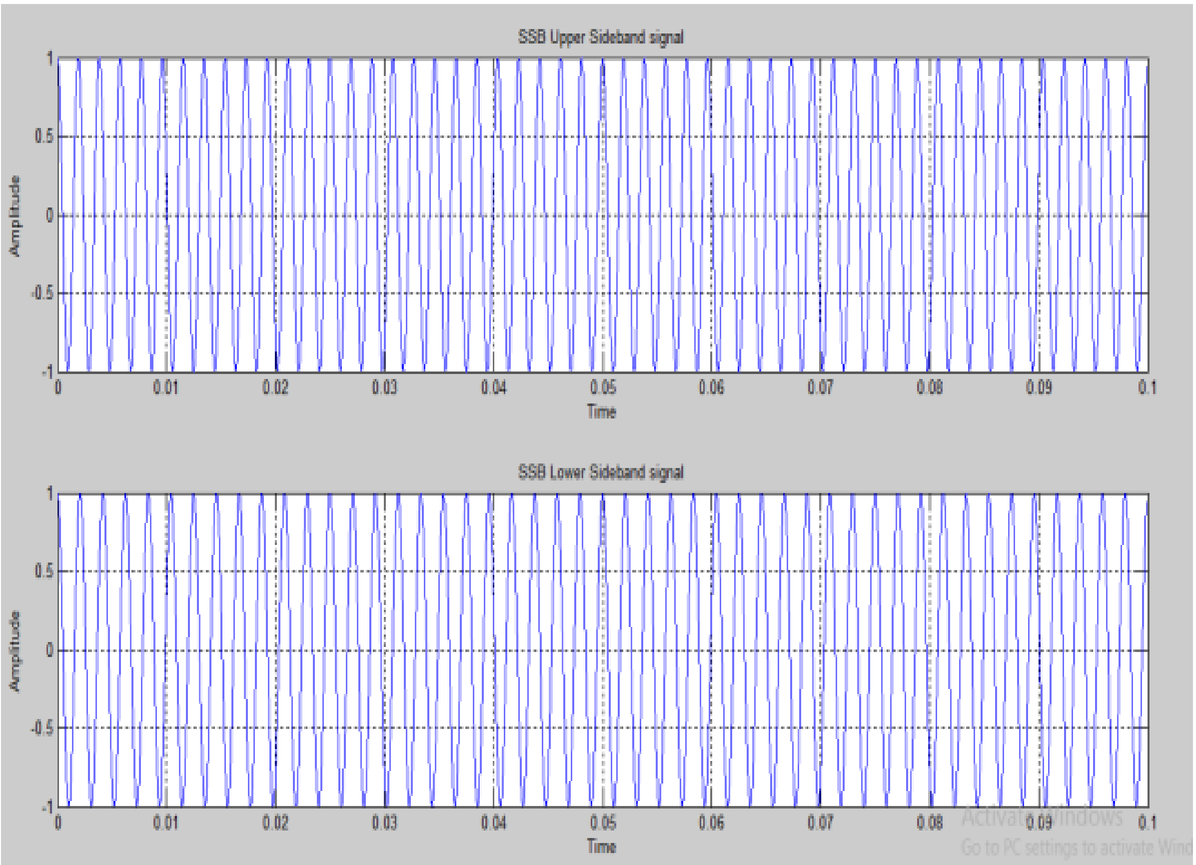
```
upper=ss1-ss2; %% upper sideband signal

lower=ss1+ss2; %% lower sideband signal

Vfupper=abs(fft(upper,10000))/10000; %% upper frequency spectrum

Vflower=abs(fft(lower,10000))/10000; %% lower frequency spectrum

%%% demodulator using Synchronous detector %%%%

%% upper demodulated

vudemod=c.*upper;

[b a] = butter(2,0.002);

upperdemod= filter(b,a,vudemod);

%% lower demodulated

vldemod=c.*lower;

[b a]=butter(2,0.002);

lowerdemod=filter(b,a,vldemod);

figure(1)

subplot(211);

plot(t,m)

xlabel('Time') ;

ylabel('Amplitude');

title('Message signal');

grid;

subplot(212);

plot(t,c)

xlabel('Time');

ylabel('Amplitude');

title('Carrier signal');

grid;

figure(2)
```

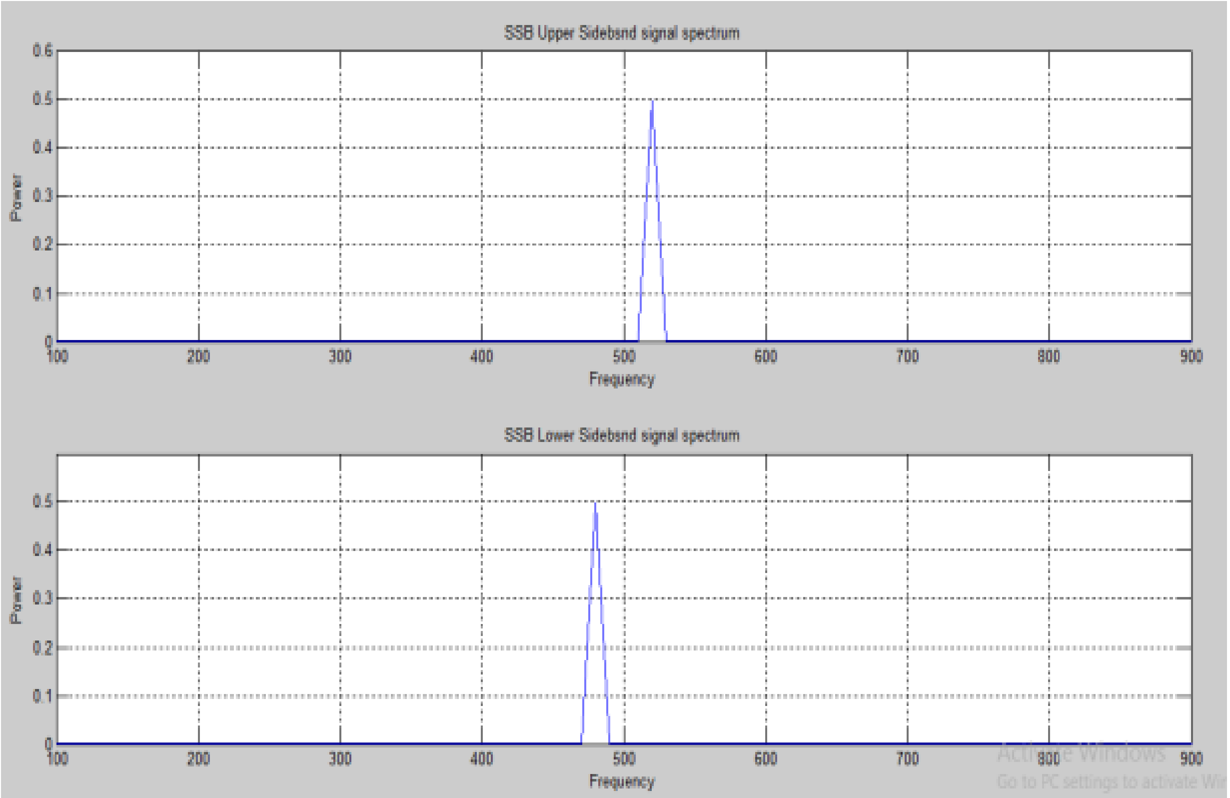
```
subplot(211);
plot(t,upper)
xlabel('Time');
ylabel('Amplitude');
title('SSB Upper Sideband signal');
grid
subplot(212);
plot(t,lower)
xlabel('Time');
ylabel('Amplitude');
title('SSB Lower Sideband signal');
grid
figure(3)
subplot(211);
plot(f*10,Vfupper)
axis([(fc-20*fm) (fc+20*fm) 0 0.6]);
xlabel('Frequency');
ylabel('Power');
title('SSB Upper Sidebsnd signal spectrum');
grid
subplot(212);
plot(f*10,Vflower)
axis([(fc-20*fm) (fc+20*fm) 0 0.6]);
xlabel('Frequency');
ylabel('Power');
title('SSB Lower Sidebsnd signal spectrum');
grid
```

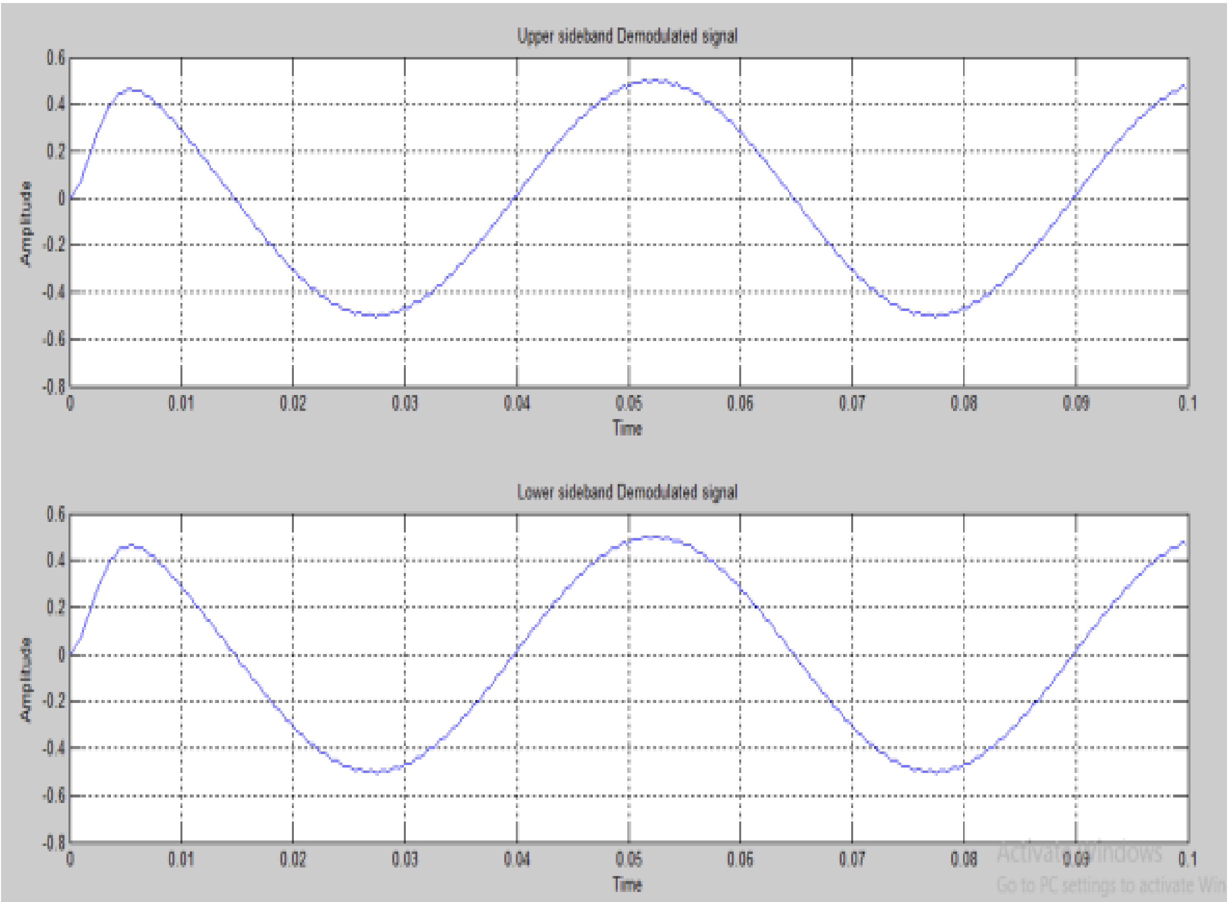
```
figure(4)
subplot(211);
plot(t,upperdemod)
xlabel('Time');
ylabel('Amplitude');
title('Upper sideband Demodulated signal');
grid;
subplot(212);
plot(t,lowerdemod)
xlabel('Time');
ylabel('Amplitude');
title('Lower sideband Demodulated signal');
grid;
```

Expected Waveform:






SSB Upper Sidebsnd signal spectrum
Power
Frequency
SSB Lower Sidebsnd signal spectrum
Power
Frequency


Upper sideband Demodulated signal
Amplitude
Time
Lower sideband Demodulated signal
Amplitude
Time

Precautions:

1. Check the connections before giving the power supply
2. Observation should be done carefully.

Pre Lab Questions:

1. Why modulation is an essential process of communication system?
2. Explain Block diagram of Communication system?
3. Explain need for modulation?
4. Define Amplitude modulation?

Post Lab Questions:

1. Explain Phase shift method for generation of SSBSC
2. Write Power equation of SSBSC
3. Mention applications of SSBSC
4. Explain advantages of phase shift method
5. Explain COSTAS loop

Lab Assignment:

1. Generate SSBSC using filter method?
2. Observe the spectrum and calculate BW?

Result:

The SSB modulated wave using Phase shift method is generated and SSB Modulated wave is demodulated.

Experiment No 7

Time Division Multiplexing

Aim:

To study the operation of Time-Division multiplexing.

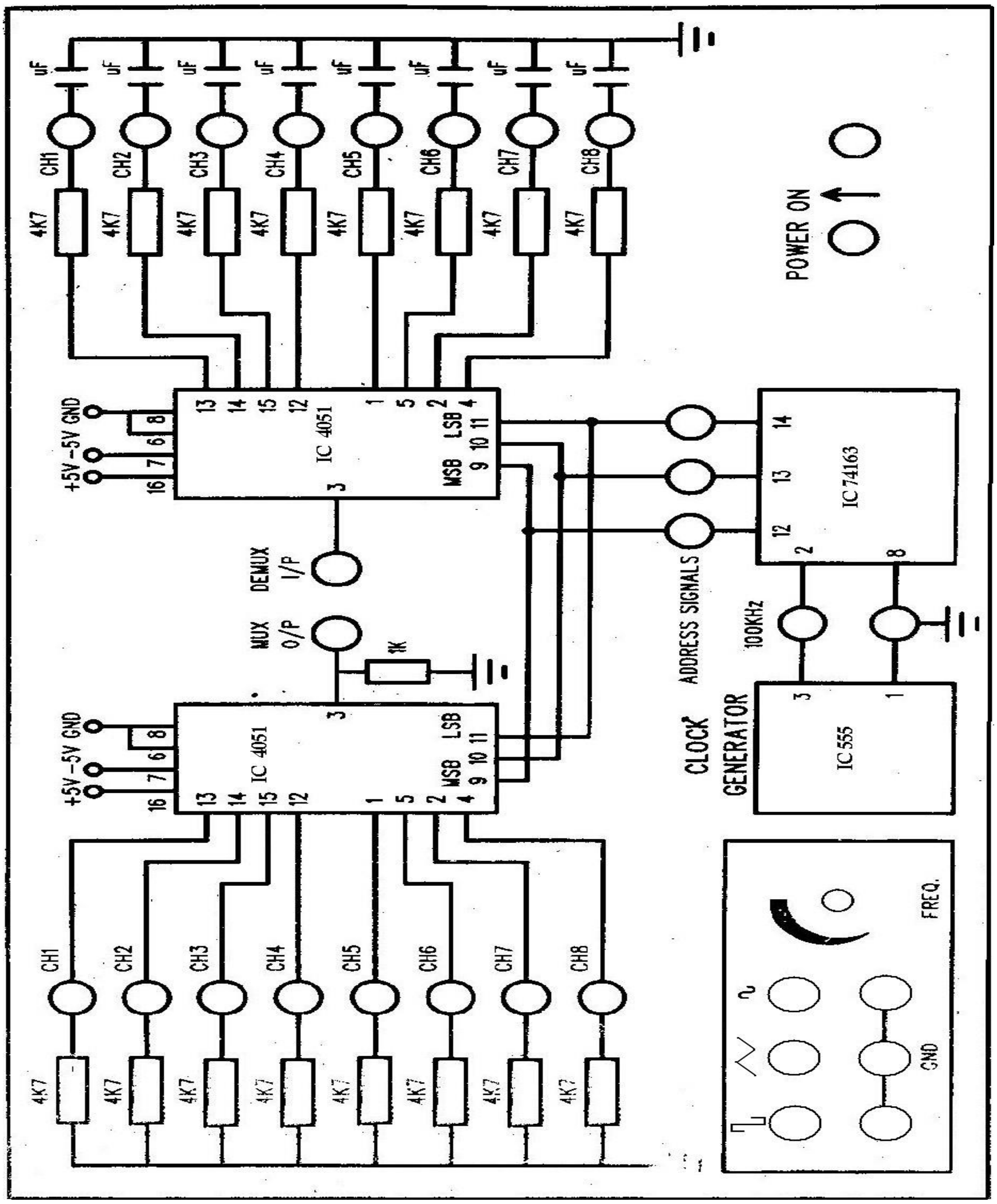
Experimental Requirements:

S. No	Experimental Requirements	Range	Quantity
1	Time-Division Multiplexing and Demultiplexing trainer Kit.	--------	1
2	C.R.O.	(0-20) MHz	1
3	Connecting wires.	-------	07

Theory:

The TDM system is highly sensitive to dispersion in the common channel, that is, to variations of amplitude with frequency or lack of proportionality of phase with frequency. Accordingly, accurate equalization of both magnitude and phase response of the channel is necessary to ensure a satisfactory operation of the system. The primary advantage of TDM is that several channels of information can be transmitted simultaneously over a single cable. In the CIRCUIT DIAGRAM the 555 timer is used as a clock generator. This timer is a highly stable device for generating accurate time delays. In this circuit this timer generates clock signal, which is of 100 KHz frequency(approximately). This clock signal is connected to the 74163 IC. 74163 IC is a synchronous preset-able binary counter. It divides the clock signal frequency into three parts and those are used as selection lines for multiplexer and Demultiplexer. In built signal generator is provided with sine, square and triangle outputs with variable frequency. These three signals can be used as inputs to the multiplexer. IC 4051 is a 8 to 1 analog multiplexer. It selects one-of eight signal sources as a RESULTof aunique three-bit binary code at the select inputs. Again IC 4051 is wired as 1 to 8 Demultiplexer. Demux input receives the data source and transmits the data signals ondifferent channels.

Circuit Diagram:



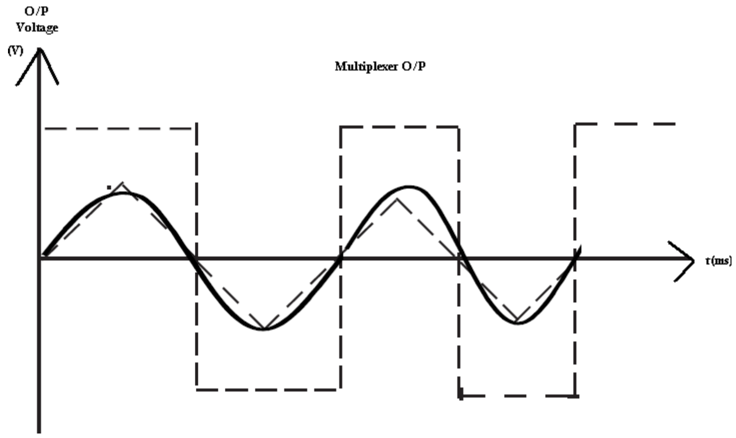
Procedure:

1. Switch on Time division multiplexing and demultiplexing trainer.
2. Connect the sine wave to ch1 , square wave to ch2 and Triangle wave form toCh3Terminals of 8 to 1 multiplexer.
3. Observe the Multiplexer output on channel 1 of a CRO.
4. Connect Mux output to demux input.

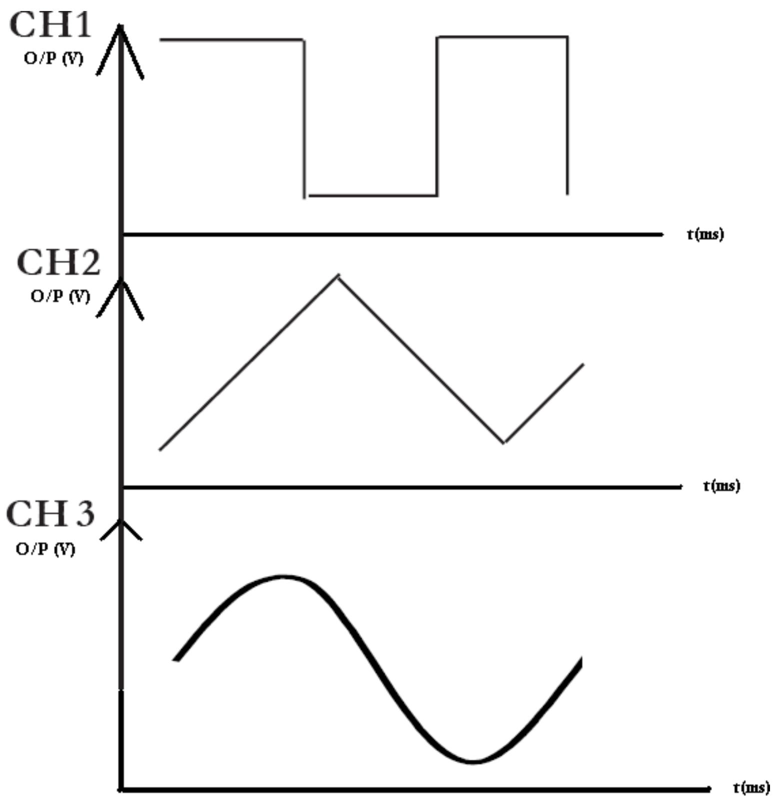
5. Observe corresponding signal

<u>Expected Waveforms:</u>

<u>Multiplexer O/P:</u>



<u>De-Multiplexer O/P:</u>



Precautions:

1. Check the connections before giving the power supply
2. Observation should be done carefully
3. Connect the circuit properly.
4. Apply the voltages wherever required.
5. Do not apply stress on the components.

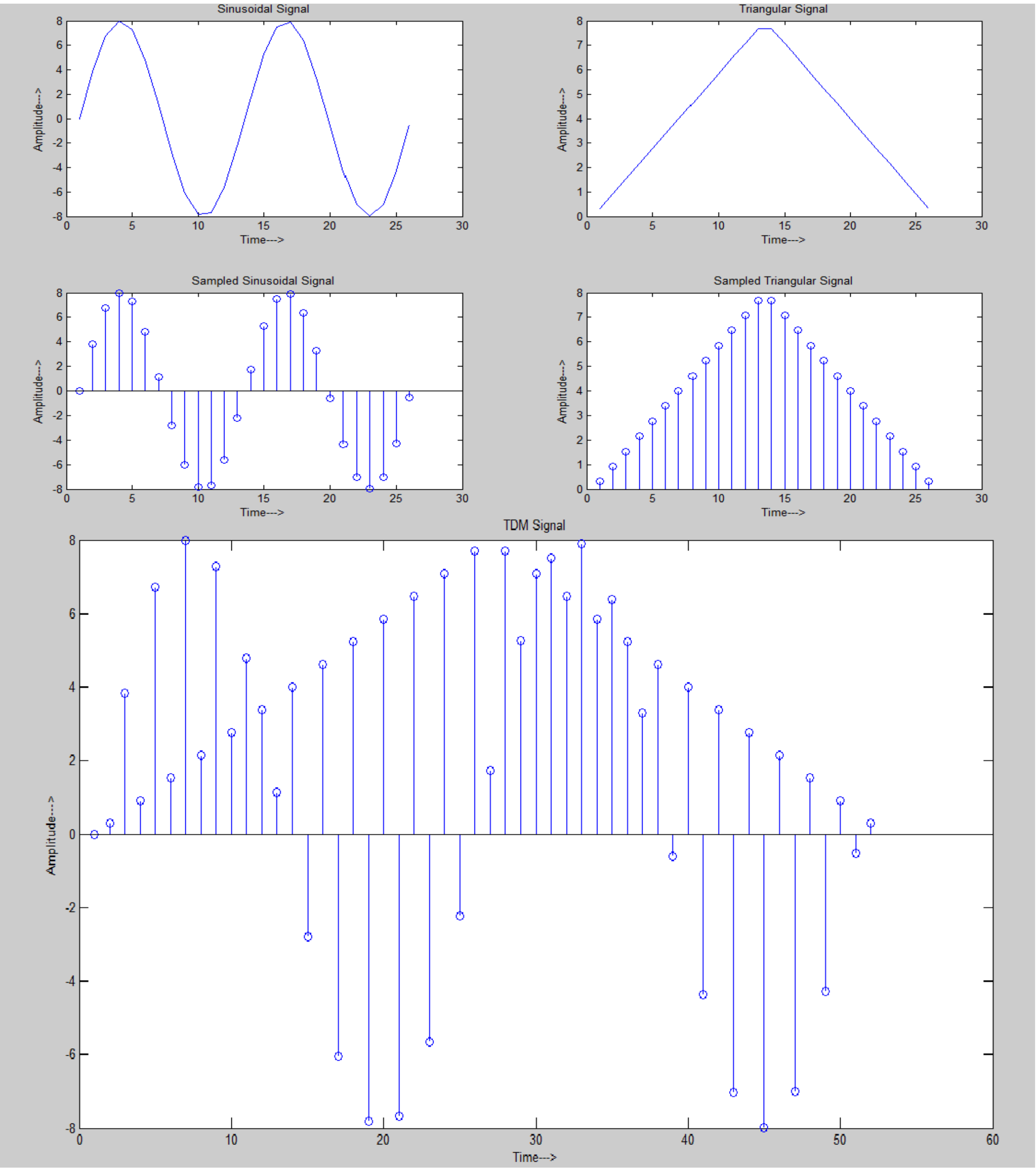
Matlab Program:

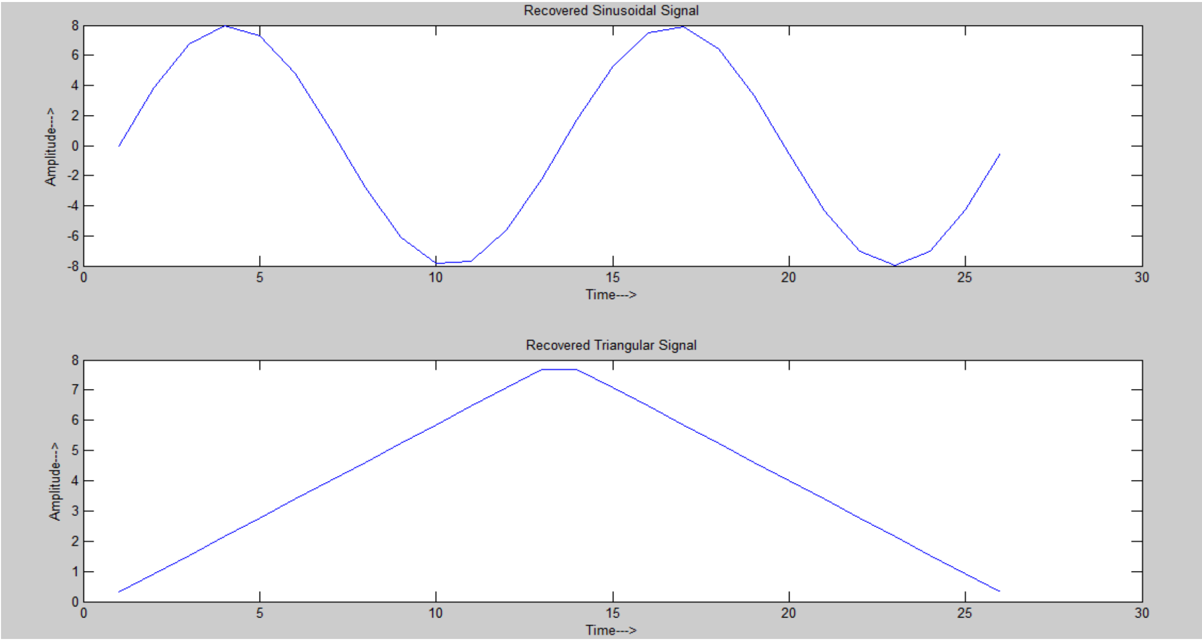
```
clc;
close all;
clear all;
% Signal generation
x=0:.5:4*pi;
% siganal taken upto 4pi sig1=8*sin(x);
% generate 1st sinusoidal signal l=length(sig1);
sig2=8*triang(l);
% Generate 2nd traingular Sigal
% Display of Both Signal subplot(2,2,1);
plot(sig1);
title('Sinusoidal Signal');
ylabel('Amplitude--->');
xlabel('Time--->');
subplot(2,2,2); plot(sig2);
title('Triangular Signal');
ylabel('Amplitude--->');
xlabel('Time--->');
% Display of Both Sampled Signal subplot(2,2,3);
stem(sig1);
```

```
title('Sampled Sinusoidal Signal');
ylabel('Amplitude--->');
xlabel('Time--->');
subplot(2,2,4);
stem(sig2);
title('Sampled Triangular Signal');
ylabel('Amplitude--->');
xlabel('Time--->');
l1=length(sig1);
l2=length(sig2);
for i=1:l1
sig(1,i)=sig1(i);
% Making Both row vector to a matrix sig(2,i)=sig2(i);
end
% TDM of both quantize signal tdmsig=reshape(sig,1,2*l1);
% Display of TDM Signal figure
stem(tdmsig);
title('TDM Signal');
ylabel('Amplitude--->');
xlabel('Time--->');
% Demultiplexing of TDM Signal demux=reshape(tdmsig,2,l1);
for i=1:l1
sig3(i)=demux(1,i);
% Converting The matrix into row vectors sig4(i)=demux(2,i);
end
% display of demultiplexed signal figure
subplot(2,1,1) plot(sig3);
```

```
title('Recovered Sinusoidal Signal');

ylabel('Amplitude--->');

xlabel('Time--->');

subplot(2,1,2) plot(sig4);

title('Recovered Triangular Signal');

ylabel('Amplitude--->');

xlabel('Time--->');
```

Expected Waveform:





Pre Lab Questions:

1. Explain multiplexing?
2. Explain different types of multiplexing?
3. What are the advantages of multiplexing?

Post Lab Questions:

1. Explain Time-division multiplexing
2. Differentiate FDM & TDM
3. What is the BW of TDM
4. Explain TDM Generation

Lab Assignment:

1. Observe TDM output at different channels?
2. Observe TDM output for 3 inputs using mat lab code

Result:

The operation of time division multiplexing is studied.

Experiment No 8
Pulse Amplitude Modulation and Demodulation

Aim:

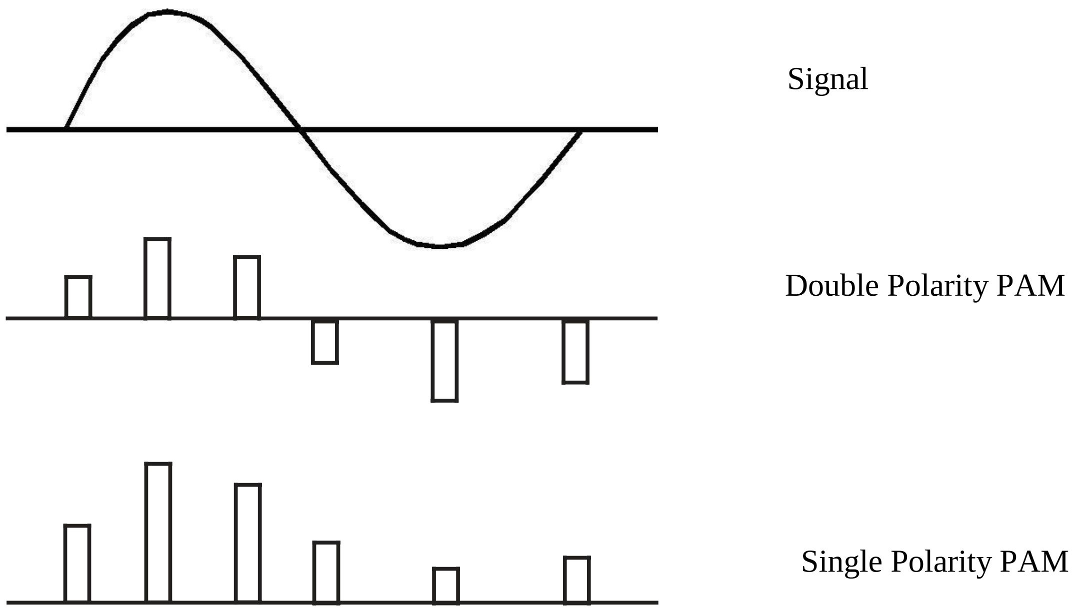
To study the Pulse Amplitude modulation and de-modulation and their waveforms.

Experimental Requirements:

PC Loaded with MatLab

Theory:

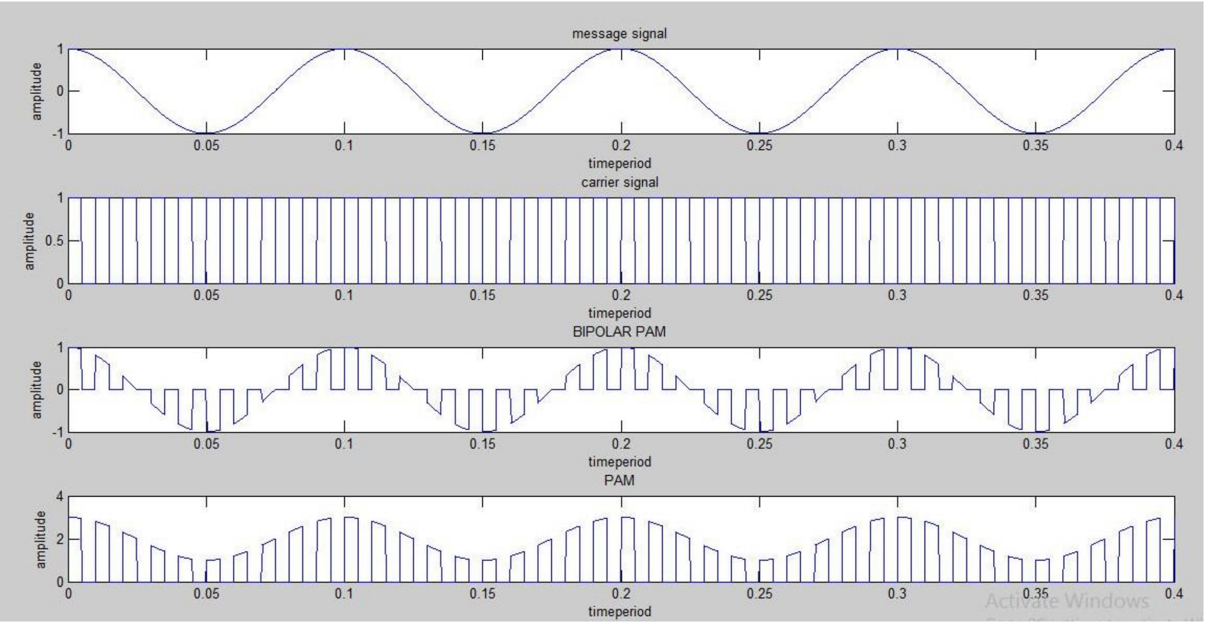
Pulse Amplitude Modulation (PAM) is the simplest and most basic form of analog pulse modulation, In PAM, the amplitudes of regularly spaced pulses are varied in proportional to the corresponding sample values of a continuous message signal; the pulses can be of a rectangular form or some other appropriate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

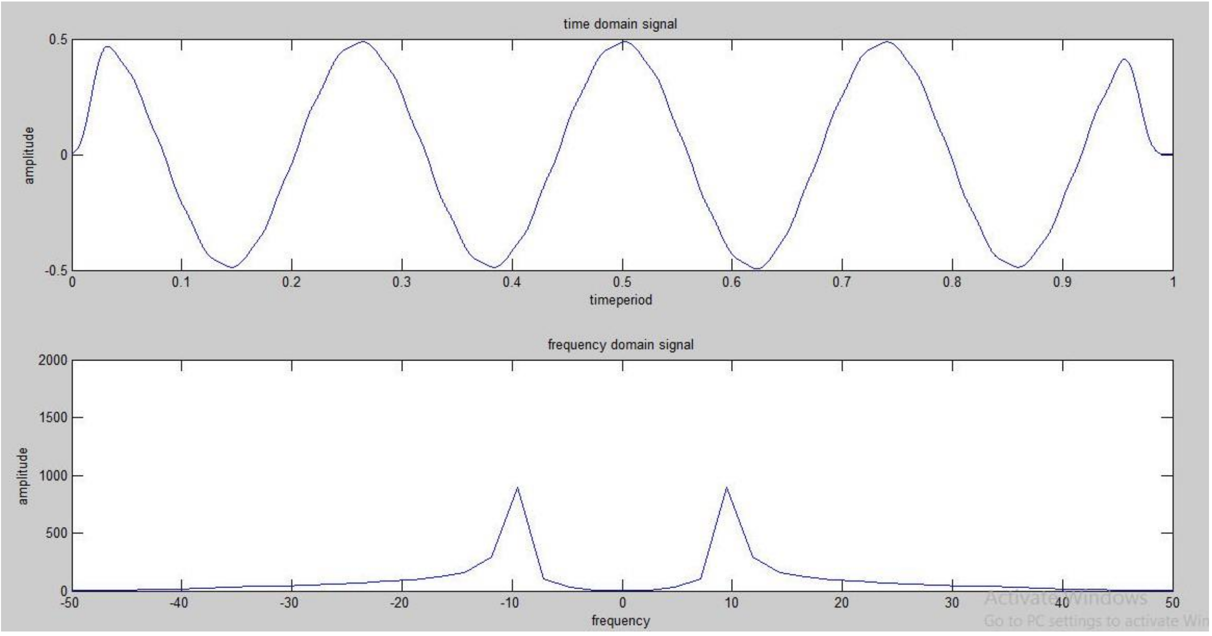


PULSE AMPLITUDE MODULATION

PAM is a pulse modulation system in which the signal is sampled at regular intervals, and each sample is made proportional to the amplitude of the signal at the instant of sampling. The pulses are then sent by either wire or cable, or else are used to modulate a carrier. The two types of PAM are shown in fig. above. The two types are Double-polarity PAM, and single- polarity PAM. The largest pulse represents the greatest positive signal amplitude sampled, while the smallest pulse represents the largest negative sample. The time duration of each pulse may be quite short, and the time interval between pulses may be relatively long. In single-polarity PAM, in which a fixed dc level is added to the signal, to ensure that the pulses are always positive. The ability to use constant-amplitude pulses is a major advantage of pulse modulation and since PAM does not utilize constant amplitude pulses, it is infrequently used. When it is used, the pulses frequency modulate the carrier.

If a radio frequency is pulse-amplitude modulated instead of simply being amplitude modulated, much less power is required for the transmission of information because the transmitter is actually switched off between pulses. This is one advantage of pulse modulation. It is very easy to generate and demodulate PAM. In a generator the signal to be converted to PAM s fed to one input of an AND gate. Pulses at the sampling frequency are applied to other input of the AND gate to open it during the wanted time intervals. The output of the gate then consists of pulses at the sampling rate, equal in amplitude to the signal voltage at each instant. The pulses are then passed through a pulse-shaping network, which gives them flat tops. Frequency modulation is then employed, so that the system becomes PAM-FM. In the receiver, the pulses are first recovered with a standard FM de-modulator. They are then fed to an ordinary diode detector, which is followed by a low-pass filter. If the cutoff frequency of this filter is high enough to pass the highest signal frequency, but low enough to remove the Sampling frequency ripple, an undistorted replica of the original signal is reproduced.





Matlab Program:

```
clc;

clear all;

close all;

fc=100;

fm=fc/10;

fs=100*fc; t=0:1/fs:4/fm;

mt=cos(2*pi*fm*t);

ct=0.5*square(2*pi*fc*t)+0.5;

st=mt.*ct;

tt=[ ];


%single sided PAM for i=1:length(st);

if st(i)==0;

tt=[tt,st(i)];
```

```
else tt=[tt,st(i)+2];
end
end
figure(1)
subplot(4,1,1);
plot(t,mt);
title('message signal');
xlabel('timeperiod');
ylabel('amplitude');
subplot(4,1,2);
plot(t,ct);
title('carrier signal');
xlabel('timeperiod');
ylabel('amplitude');
subplot(4,1,3);
plot(t,st);
title('BIPOLAR PAM');
xlabel('timeperiod');
ylabel('amplitude');
subplot(4,1,4);
plot(t,tt);
title('PAM');
xlabel('timeperiod');
ylabel('amplitude');

%demodulation dt=st.*ct;
dt_frequency=fftshift(abs(fft(dt)));
```

```
filter=fir1(200,fm/fs,'low');

original_t_signal=conv(filter,dt);

original_f_signal=fftshift(abs(fft(original_t_signal)));

t1=0:1/(length(original_t_signal)-1):1;

f=-fs/2:fs/(length(original_f_signal)-1):fs/2; figure(2)

subplot(2,1,1);

plot(t1,original_t_signal);

title('time domain signal');

xlabel('timeperiod');

ylabel('amplitude');

subplot(2,1,2);

plot(f,original_f_signal);

title('frequency domain signal');

xlabel('frequency');

ylabel('amplitude');

axis([-50 50 0 2000]);
```

Experiment No 9
Pulse Position Modulation and Demodulation

Aim:

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

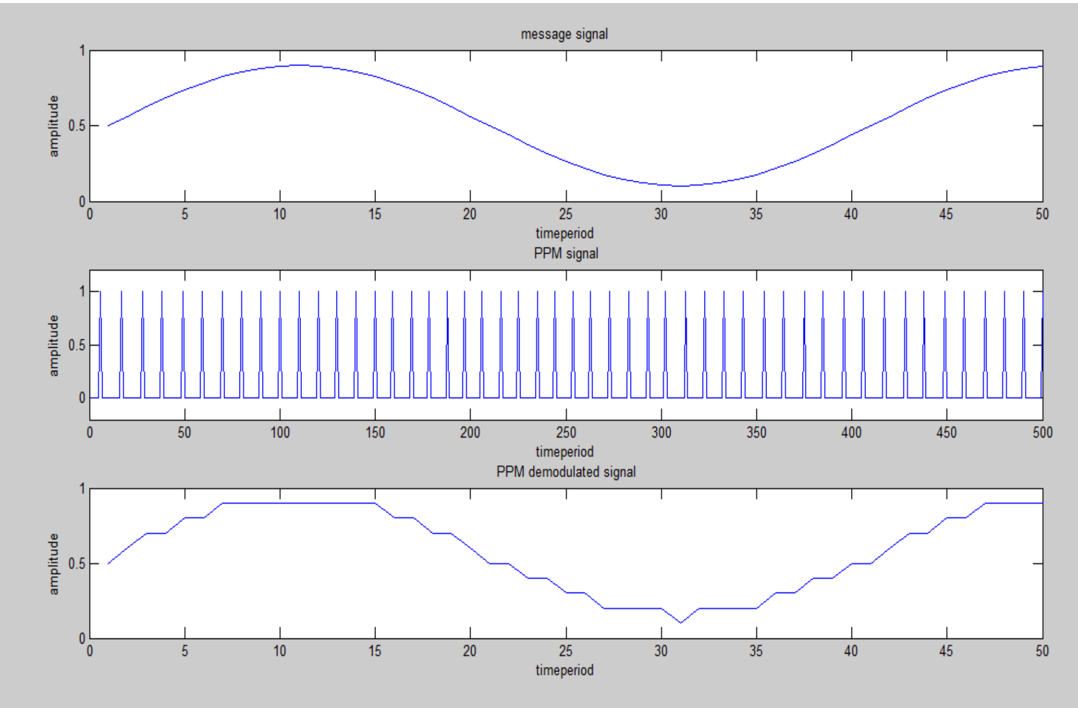
Experimental Requirements:

PC loaded with MatLab

Theory:

Pulse position modulation (PPM) is more efficient than PAM or PDM for radio transmission. In PPM all pulses have the same constant amplitude and narrow pulse width. The position in time of the pulses is made to vary in proportion to the amplitude of the modulating signal. The simplest modulation process for pulse position modulation is a PDM system with the addition of a monostable multivibrator. The monostable is arranged so that it is triggered by the trailing edges of the PDM pulses. Thus, the monostable output is a series of constant-width, constant amplitude pulses which vary in position according to the original signal amplitude .

Expected Wave Forms:



Matlab Code:

```
clc;
clear all;
close all;
fc=4000;
fs=40000;
fm=1000;
t=0:1/fs:(2/fm-1/fs);
mt=0.4*sin(2*pi*fm*t)+0.5;
st=modulate(mt,fc,fs,'PPM');
dt=demod(st,fc,fs,'PPM');
figure
subplot(3,1,1);
plot(mt);
title('message signal');
xlabel('timeperiod');
ylabel('amplitude');
axis([0 50 0 1])
subplot(3,1,2);
plot(st); title(' PPM signal');
xlabel('timeperiod');
ylabel('amplitude');
axis([0 500 -0.2 1.2])
subplot(3,1,3);
plot(dt);
title(' PPM demodulated signal');
xlabel('timeperiod');
ylabel('amplitude');
axis([0 50 0 1])
```

Experiment No 10
Agc Characteristics using Matlab

Aim:

To study the AGC Characteristics.

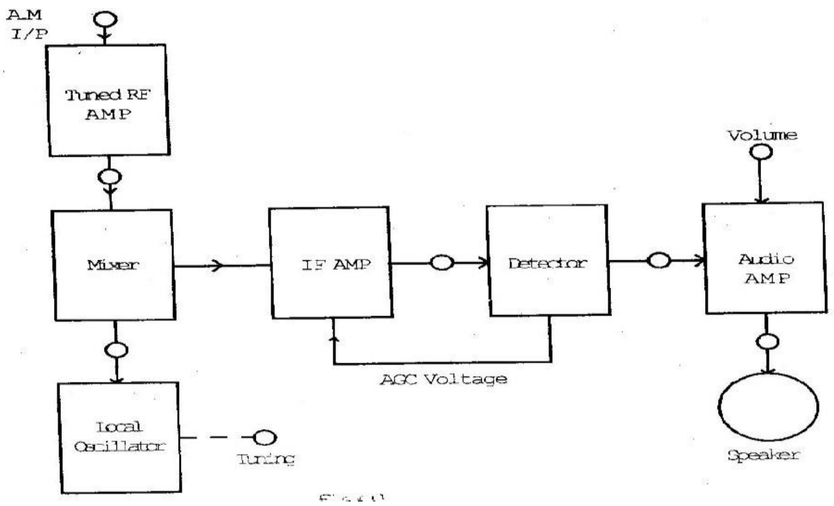
Experimental Requirements:

PC loaded with MatLab

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. The devices used in those stages are ones whose transconductance and hence gain depends on the applied bias voltage or current. It may be noted that, for correct AGC operation, this relationship between applied bias and transconductance need not to be strictly linear, as long as transconductance 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 does not have to be re adjusted every time the receiver is tuned from one station to another, except when the change in signal strength is enormous.

Block Diagram:



Expected Waveforms:

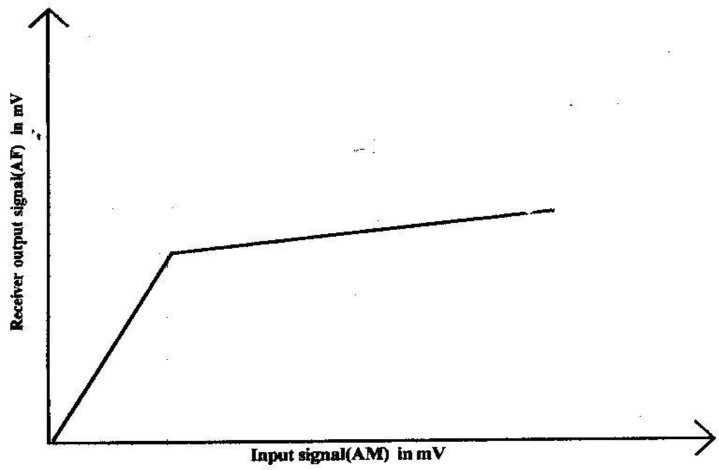


Fig.2. AGC characteristics curve

Procedure:

1. As the circuit is already wired you just have to trace the circuit according to theCIRCUIT DIAGRAM given above
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 isapproximately 50mVpp of 455 KHz frequency and AF voltage is 5Vpp of1 KHz frequency.
5. Now vary the amplitude of AF signal and observe the AM wave at output, note thepercentage of modulation for different values of AF signal.% Modulation= (Emax -Emin)/(Emax+Emin) × 100
6. Now adjust the modulation index to 30% by varying the amplitudes of RF & AFsignals 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 of 1 KHz will be observed.
10. Calculate the voltage gain by measuring the amplitude of output signal (Vo) waveform, using Formula A =Vo/V i

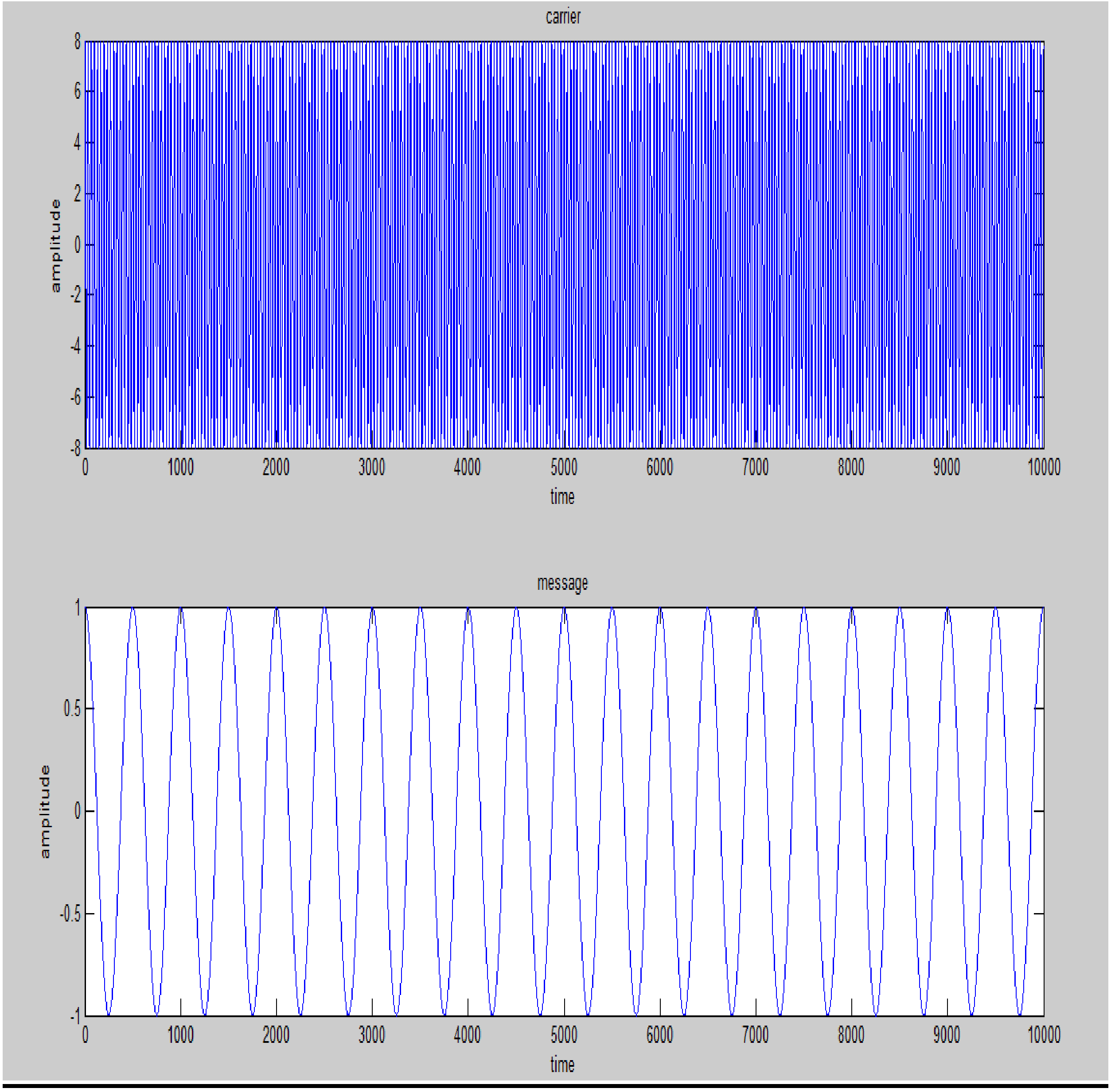
11. Now vary input level of 455 KHz IF signal and observe detected 1 KHz audio signal with and Without AGC link. The output will be distorted when AGC link removed i.e. thereis no AGC action.12. This explains AGC effect in Radio circuit

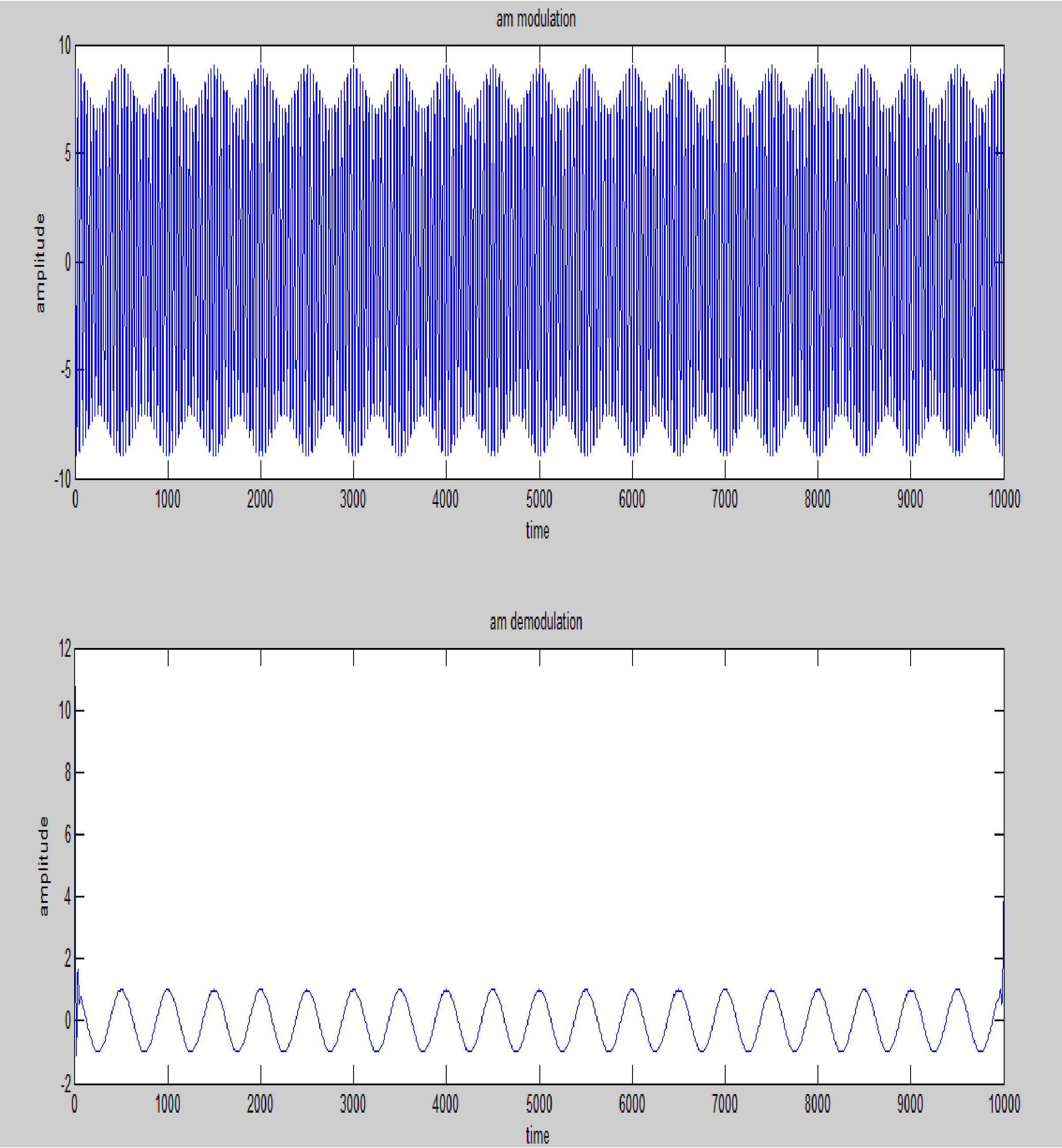
Matlab Program:

```
close all

clear all

clc

Fs = 100e3;

%sampling freq

t = 0:1/Fs:.1-1/Fs;

% time variable Am=2;

fm = 200;

%fm 200 Hz

m = cos(2*pi*fm*t);

%message signal Fc = 3e3;

% 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');
```

Expected Waveforms:





Tabular Column:

Signal Type	Frequency	Amplitude
Modulating Signal		
Carrier Signal		
Modulated Signal		
De modulated Signal(without AGC)		
De modulated Signal(with AGC)		

Pre Lab Questions:

1. Classify receivers
2. Explain Super heterodyne working principle.
3. List out the advantages and disadvantages of TRF receiver

Post Lab Questions:

1. Define Sensitivity and Selectivity.
2. Define Image frequency rejection ratio.
3. Define image frequency.
4. Define Image frequency rejection ratio.

Lab Assignment:

1. Observe TRF Receiver characteristics?

Result:

Thus AGC characteristics was studied and wave forms was observed.

Experiment No 11
Phase Locked Loop

Aim:

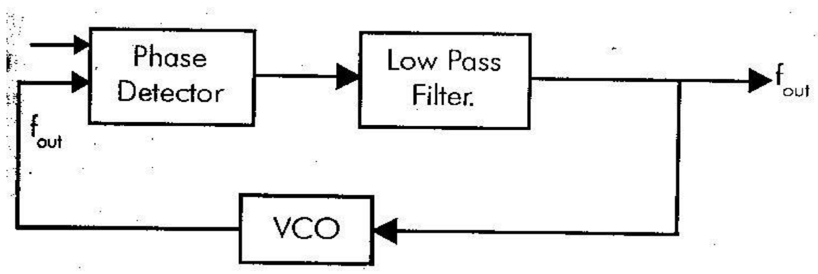
To compare the theoretical and practical values of capture range and lock range of phase locked loop.

Experimental Requirements:

Sl.No	Experimental Requirements	Range	Quantity
1	PLL Trainer Kit.	--------	1
2	C.R.O.	(0-20) MHz	1
3	Function Generator (0-1MHz)		
3	Connecting wires.	-------	07

Theory:

A phase locked loop is basically a closed loop system designed to lock the output frequency and phase to the frequency and phase of an input signal. It is commonly abbreviated as PLL. PLL"s are used in applications such as frequency synthesis, frequency modulation/demodulation, AM detection, tracking filters, FSK demodulator, tone detector etc. The block diagram of PLL is as shown below



PLL consists of

1. Phase detector
2. Low pass filter
3. Voltage controlled oscillator (VCO)

Procedure:

1. Connect the circuit as per the CIRCUIT DIAGRAM on the breadboard.
2. Without giving input signal, find out the output signal frequency, which is called free running frequency, F0

3. Now apply 1V, 1 KHz sinusoidal signal as input and slowly increase the input frequency andnote down the corresponding output frequency
4. When input and output frequencies are equal, then note down it as F1
   Now increase the input frequency slowly and the output frequency will also follow the input frequency. This follow up will continue until a certain frequency point F2 Note down the value of F2.Continue to increase the input frequency and then the output frequency will be back to F0.
5. Now decrease the input frequency slowly and at one point input and output frequencies willbe equal. Note down this point as *F3*.
6. Continue to decrease the input frequency. The output frequency will also follow once again, this follow up continues up to F4. Note down this frequency value and decrease the input frequency further. Then the output frequency will once again back to only.
7. Calculate the theoretical and practical values of free-running frequency lock range and capture range and compare them.

   Free running frequency f0 = 1.2/4R1C1

   Lock range fL = ±8 f0/V

   Where V = +V-(-V) = 12-(-12) = 24

   Theoretical lock rangef = f0± fL

   Capture RangefC = ± (fL/2π R1C1)1/2

Circuit Diagram :

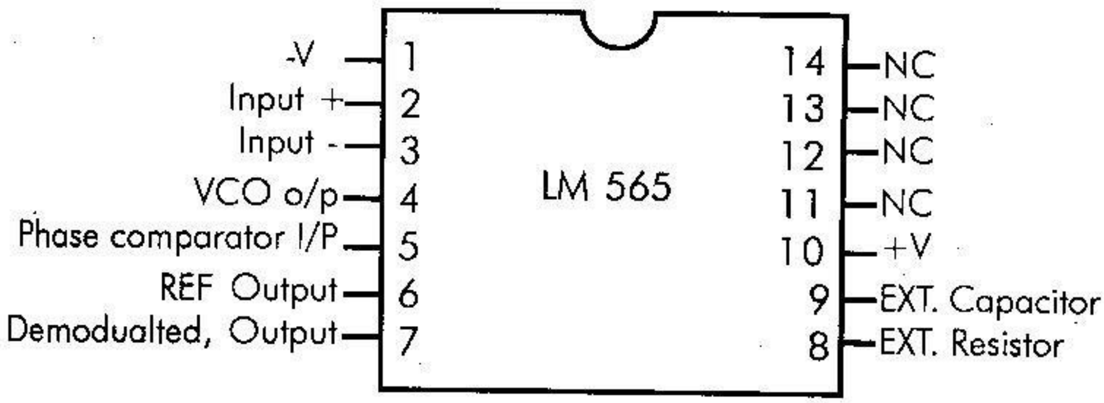


Fig.1 Pin Diagram of LM 565

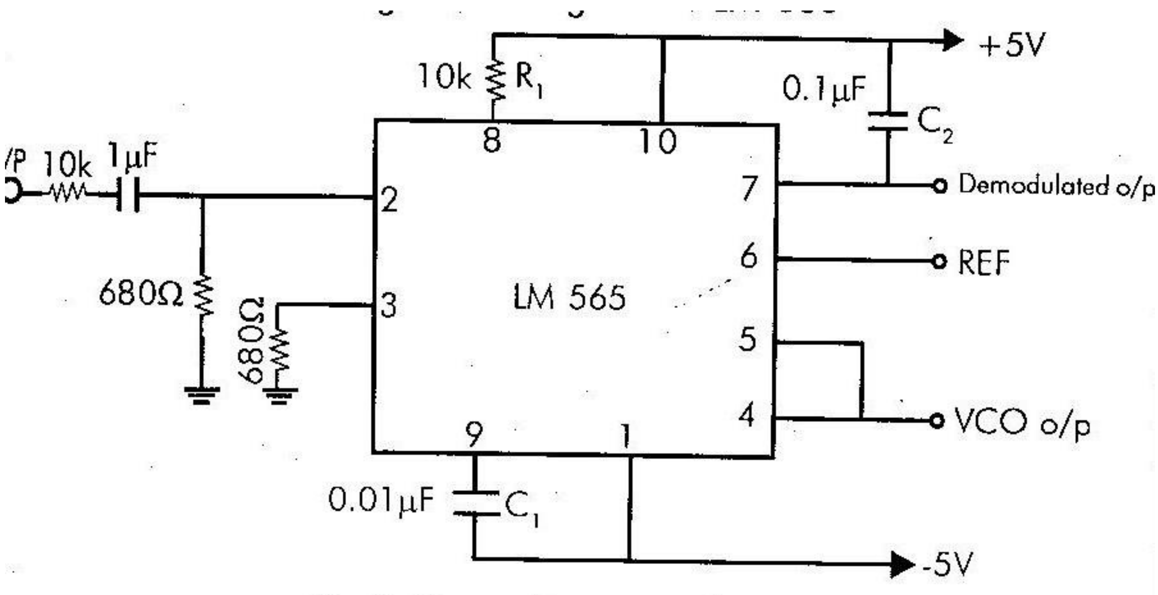


Fig.2 Circuit Diagram of LM 565

Matlab Program:

clear all;

close all;

f=1000;

%Carrier frequency fs=100000;

%Sample frequency N=5000;

%Number of samples Ts=1/fs;

t=(0:Ts:(N*Ts)- Ts);

%Create the message signal f1=100;

%Modulating frequency msg=sin(2*pi*f1*t);

kf=.0628;

%Modulation index

%Create the real and imaginary parts of a CW modulated carrier to be tracked.

Signal=exp(j*(2*pi*f*t+2*pi*kf*cumsum(msg)));

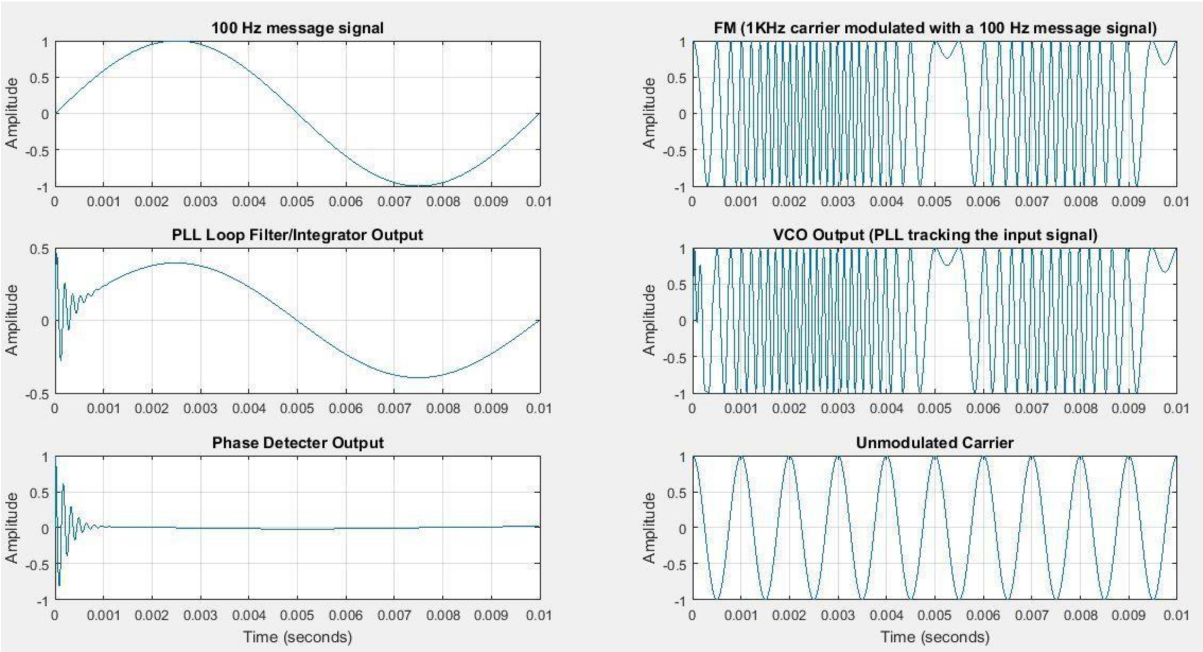
%Modulated carrier Signal1=exp(j*(2*pi*f*t));

%Unmodulated carrier

```
%Initilize PLL Loop phi_hat(1)=30;

e(1)=0;

phd_output(1)=0;

 vco(1)=0;

%Define Loop Filter parameters(Sets damping) kp=0.15;

%Proportional constant

ki=0.1;

%Integrator constant

%PLL implementation for n=2:length(Signal)

vco(n)=conj(exp(j*(2*pi*n*f/fs+phi_hat(n-1))));

%Compute VCO phd_output(n)=imag(Signal(n)*vco(n));

%Complex multiply VCO x Signal input e(n)=e(n-1)+(kp+ki)*phd_output(n)-
ki*phd_output(n-1);

%Filter integrator phi_hat(n)=phi_hat(n-1)+e(n);

%Update VCO

end;

%Plot waveforms startplot = 1;

endplot = 1000;

figure(1);

subplot(3,2,1);

plot(t(startplot:endplot), msg(startplot:endplot));

title('100 Hz message signal');

%xlabel('Time (seconds)');

ylabel('Amplitude'); grid;

figure(1); subplot(3,2,2);

plot(t(startplot:endplot), real(Signal(startplot:endplot)));
```

```
title('FM (1KHz carrier modulated with a 100 Hz message signal)');
%xlabel('Time (seconds)');
ylabel('Amplitude'); grid;
figure(1) subplot(3,2,3);
plot(t(startplot:endplot), e(startplot:endplot));
title('PLL Loop Filter/Integrator Output');
%xlabel('Time (seconds)');
 ylabel('Amplitude'); grid;
subplot(3,2,4);
plot(t(startplot:endplot), real(vco(startplot:endplot)));
title('VCO Output (PLL tracking the input signal)');
%xlabel('Time (seconds)');
ylabel('Amplitude');
grid;
subplot(3,2,5);
plot(t(startplot:endplot), phd_output(startplot:endplot));
title('Phase Detecter Output');
xlabel('Time (seconds)');
ylabel('Amplitude');
grid;
subplot(3,2,6);
plot(t(startplot:endplot), real(Signal1(startplot:endplot)));
title('Unmodulated Carrier');
xlabel('Time (seconds)');
ylabel('Amplitude');
grid;
```

Waveforms:



Result:

Thus the theoretical and practical values of lock range and capture range for PLL are calculated andcompared.

Experiment No 12

Verification of Sampling Theorem

Aim:

To verify sampling theorem

Experimental Requirements:

PC loaded with MatLab

Theory:

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.

Let m(t) be a signal whose highest frequency component is fm. Let the value of m(t) be obtained at regular intervals separated by time T far less than (1/2 fm) The sampling is thus periodically done at each TS seconds. Now the samples m (nTS) where n is an integer which determines the signals uniquely. The signal can be reconstructed from these samples without distortion.

Time Ts is called the SAMPLING TIME.

The minimum sampling rate is called NYQUIST RATE.

The validity of sampling theorem requires rapid sampling rate such that at least two samples are obtained during the course of the interval corresponding to the highest frequency of the signal under analysis.

Let us consider an example of a pulse modulated signal, containing speech information, as is used in telephony. Over standard telephone channels the frequency range of A.F. is from 300 Hz to 3400Hz. For this application the sampling rate taken is 8000 samples per second. This is an Inter-national standard. We can observe that the pulse rate is more than twice the highest audio frequency used in this system. Hence the sampling theorem is satisfied and the resulting signal is free from sampling error.

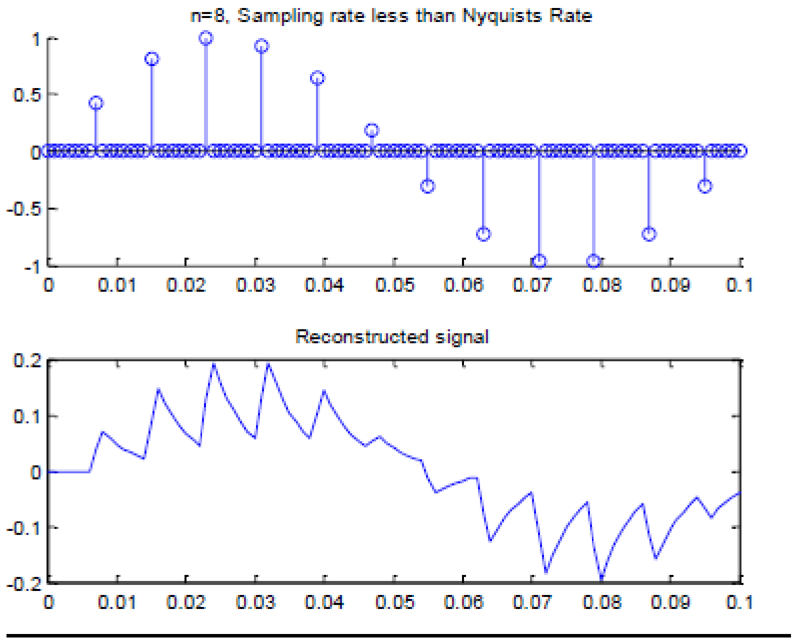
Matlab Program:

```
t=0:0.001:0.1;

t1=zeros(1,length(t));

f=input('enter the baseband signal frequency') x=sin(2*pi*f*t);

n=input('enter the integer which decides the sampling frequency') for i=1:length(t)
```

```
if n*i<=length(t) t1(n*i)=1;
end end
s1=x.*t1;
[den,num]=butter(1,2*pi*f/1000);
s11=filter(den,num,s1);
subplot(2,1,1)
stem(t,s1);
title('n=8, Sampling rate less than Nyquists Rate') subplot(2,1,2)
plot(t,s11) title('Reconstructed signal')
```

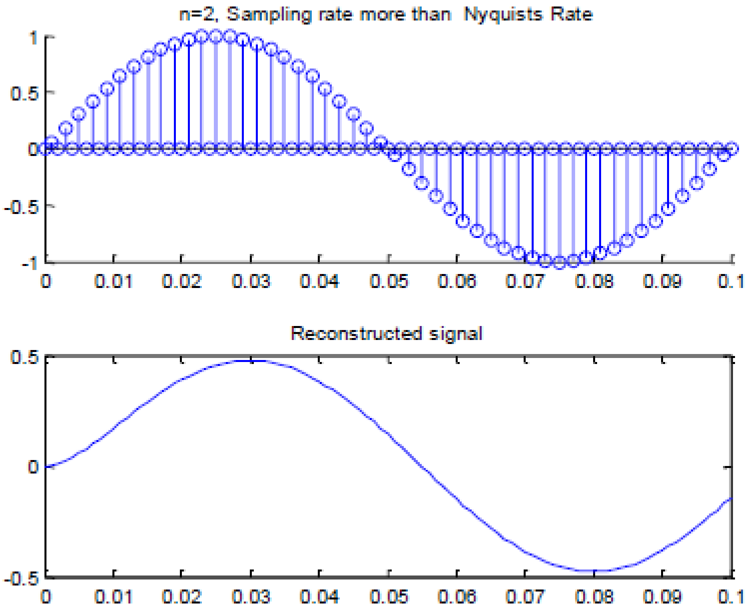
f =10

Enter the integer which decides the sampling frequencyn =8



Enter the baseband signal frequency f=10

Enter the integer which decides the sampling frequency n= 2



Result:

Enter the baseband signal frequency