PART-A

Wiring Experiments

1. Design of active second order Butterworth low pass and high pass filters.
2. Amplitude Modulation and Demodulation of
   (a) Standard AM and (b) DSBSC (LM741 and LF398 ICs can be used)
3. Frequency modulation and demodulation
4. Design and test Time Division Multiplexing and Demultiplexing of two band limited signals.
5. Design and test i) Pulse sampling, flat top sampling and reconstruction.
   ii)Pulse amplitude modulation and demodulation.
6. Design and test BJT/FET Mixer
7. Pulse Code Modulation and demodulation
8. Phase locked loop Synthesis

EXPERIMENT: 01:

Design active second order Butterworth low pass and high pass filters

AIM: *Design a second order active Butterworth low pass filter having upper cut off frequency 1 KHz, also determine its frequency response using IC 741.*

APPARATUS REQUIRED:

1. OP-AMP IC741
2. Resistor 10KΩ, 1.6KΩ (2nos), 5.6 KΩ
3. Capacitor 0.1 µF (2nos)
4. CRO, RPS DUAL (0-30) V, bread board , connecting wires,...

THEORY: A filter is a frequency selective circuit that allows only a certain band of frequency component of an input signal to pass through and blocks other frequency components. An active filter network is obtained by interconnecting passive elements and active element. Op-amps are used in active filters to provide amplification and gain control. A low pass filter allows only low frequency signals and suppresses high frequency signals. The range of frequency varies from dc to cut off frequency f_L. The frequency range below cut off frequency is called pass band and frequency range beyond f_L is called stop band. A high pass filter allows only high frequency signals and suppresses low frequency signals. The range of frequency beyond cut off frequency f_H is called pass band and range of frequency from dc to f_H is called stop band.

Butterworth filter is the best compromise between attenuation and phase response. It has no ripple in the pass band or the stop band, and because of this is sometimes called a maximally flat filter. The Butterworth filter achieves its flatness at the expense of a relatively wide transition region from pass band to stop band, with average transient characteristics.

The Butterworth filter is normalized for a –3 dB response at ωo = 1. The values of the elements of the Butterworth filter are more practical and less critical than many other filter types.

An improved filter response can be obtained by using a second order active filter. A second order filter consists of two RC pairs and has a roll-off rate of -40 dB/decade. A general second order filter (Sallen Kay filter) is used to analyze different LP, HP, BP and BS filters.

DESIGN:

Second order active Low Pass filter (see fig.1)

1) Choose high cut-off frequency f_H, say 1KHz
2) The design can be simplified by selecting $R_2 = R_3 = R$ and $C_2 = C_3 = C$ and choose a value of **C = 0.1µF**.
3) Calculate the value of R from the equation, $F_H = 1/(2\pi RC)$

   R = 1.6 KΩ
4) To guarantee Butterworth response gain must be equal to 1.586.

   For n = 2, α (damping factor) = 1.414,

   Passband gain = $A_F = 3 - \alpha = 3 - 1.414 = 1.586$.

$$A_F = 1 + \frac{R_F}{R_i} \qquad 1.586 = 1 + \frac{R_F}{R_i} \qquad 0.586 = \frac{R_F}{R_i}$$

$$0.586\,R_i = R_F$$

5) Let R_i = **10K Ω**, then R_F = **5.8KΩ**

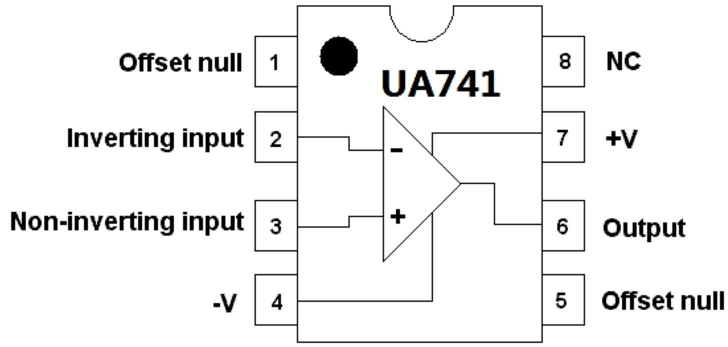
Second order active High Pass filter: (see fig.2)

Similarly for HPF select lower cut-off frequency, **f_L=1KHz** and design using the same values of R and C.

[NOTE: i) Decade is a tenfold increase (multiply by 10) or tenfold decrease (divide by 10).
For example, 2 to 20Hz represents one decade.

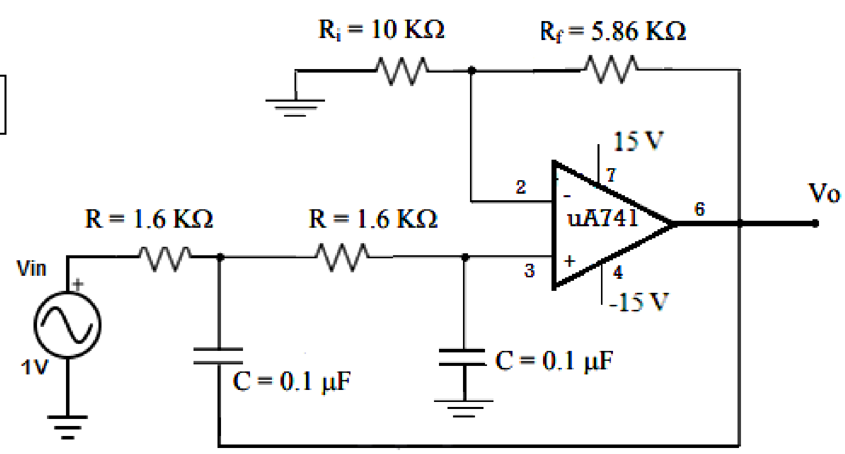
ii) Octave is a doubling (multiply by 2) or halving (divide by 2) of the frequency scale.
For example, 10 to 20Hz represents one octave, while 2 to 16Hz is three octaves]

PIN DIGRAM of OPAMP:



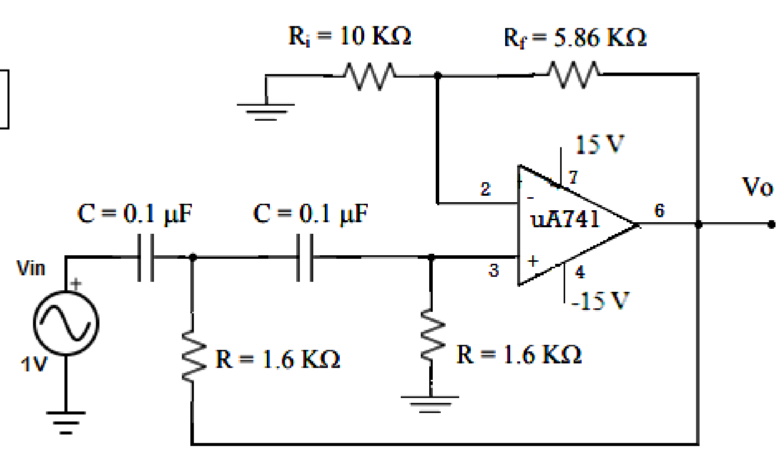
CIRCUIT DIAGRAM: Second order LPF

Fig.1



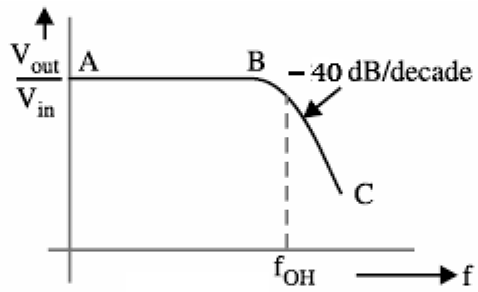
CIRCUIT DIAGRAM: Second order HPF

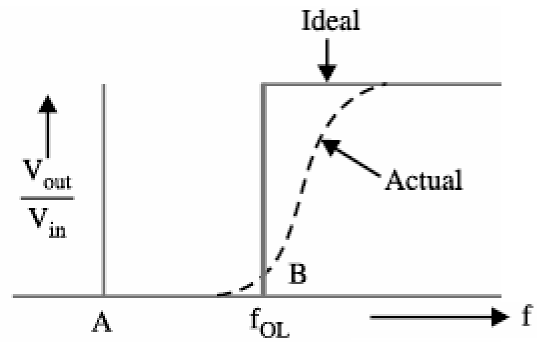
Fig.2



EXPECTED WAVEFORM

LOW PASS FILTER **HIGH PASS FILTER**





Frequency response of 2nd order LPF and HPF

PROCEDURE:

1. *The connections are made as shown in the circuit diagram.*
2. *The input signal 1 V peak-peak (sine wave) is applied to the second order RC filter circuit with the non-inverting terminal.*
3. *The supply voltage to OPAMP is switched ON and the o/p voltages are recorded through CRO by varying different frequencies from 10 Hz to 100 KHz and tabulate the readings for the input/output amplitudes corresponding to different frequencies.*
4. *Calculating Gain using the formula and plotting the frequency response characteristics using Semi-log graph sheet with gain in dB on y-axis and frequency in Hz on x-axis and find out 3dB line for fc.*

TABULATION: V_{in} = 1V

LPF				HPF			
Frequency Hz	Vo(p-p)	A_V = Vo(p-p) / Vin(p-p)	A_V (dB) = 20*log A_V	Frequency Hz	Vo(p-p)	A_V = Vo(p-p) / Vin(p-p)	A_V (dB) = 20*log A_V
100				800			
200				900			
500				1K			
700				3K			
900				5K			
1K				7K			
2K				10K			
3K				20K			

Result: Thus the second order Active Low Pass filter is designed and its frequency response characteristic curves are drawn.

EXPERIMENT: 02A:

Amplitude Modulation and Demodulation of Standard AM

AIM: *1. To generate amplitude modulated wave and determine the percentage modulation.*
2. To Demodulate the modulated wave using envelope detector.

Apparatus Required:

Transistor (BC 107)
Diode (0A79)
Resistors
Capacitor
IF Transformer
CRO 20MHz
Function Generator 1MHz
Regulated Power Supply 0-30V, 1A

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 K_a is 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 the envelop of the input signal wave form exactly; hence, the name. Some version of this circuit is used in almost all commercial AM radio receivers.

$$\text{The Modulation Index is defined as, } m = \frac{(E_{max} - E_{min})}{(E_{max} + E_{min})}$$

MODULATION

Design:

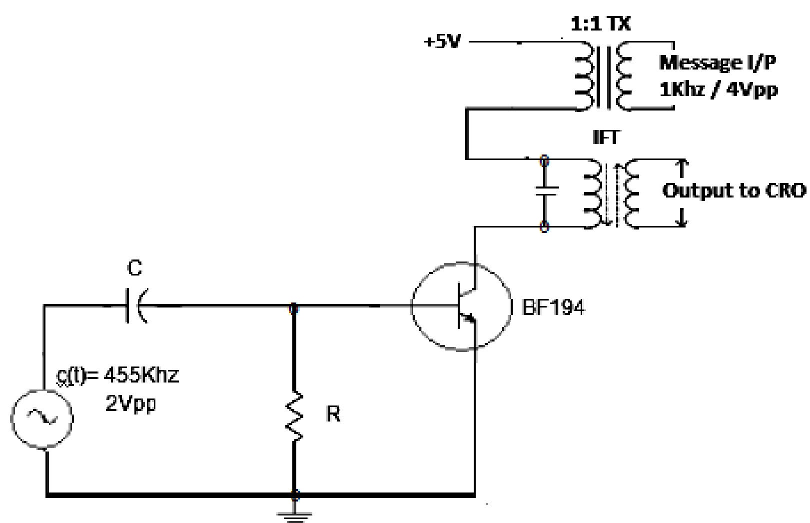
Given: F_{IFT} = 455 KHz, T = 2.19µs

Assume, RC>>T, RC = 100T

Choose **C=0.1µF**

R=2.19KΩ, choose **R= 2.2 KΩ**

CIRCUIT DIAGRAM:
Modulation:



DEMODULATION: Design: Given: Fc = 455 KHz, Fm =1 KHz

1/Fm>RC>1/Fc

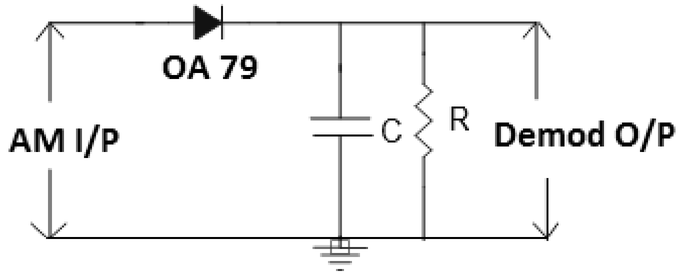
1ms> RC > 2.2 μs

Let RC =100/Fc

Choose **C= 0.1μF**

R =2.19KΩ. choose R= **2.2 KΩ**

Circuit diagram:



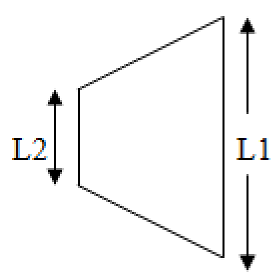
PROCEDURE:

1. *Rig up the circuit as per the circuit diagram.*
2. *Adjust the carrier frequency to about fc=455Khz & fine tune the signal to get maximum output.*
3. *Keeping the carrier amplitude constant vary the modulating signal voltage in appropriate steps & measure the modulation index using the formula m= Emax - Emin / Emax +Emin*
4. *Obtain the trapezoidal pattern & calculate the modulation index using formula m=L1-L2/L1+L2*
5. *Tabulate the results & draw the graph of modulation index Vs modulating voltage amplitude.*
6. *Rig up the demodulation circuit & observe the demodulated O/P.*

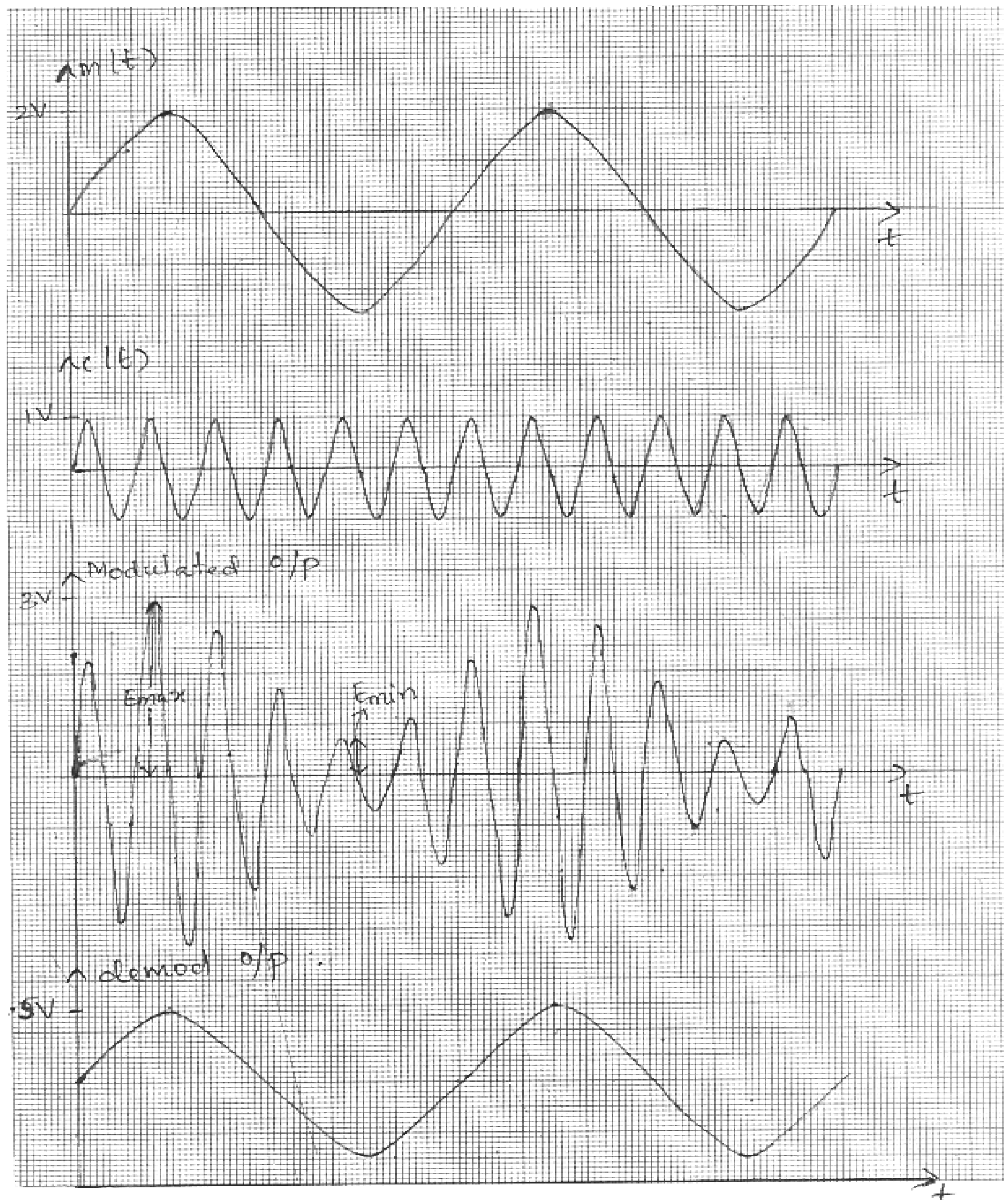
Tabulations:

Message Amp (V)	E_{max} (V)	E_{min} (V)	m= E_{max}-E_{min}/ E_{max}+E_{min}	L1 (cm)	L2 (cm)	m=L1-L2/L1+L2

Trapezoidal pattern to measure μ



Waveforms



RESULT: Modulation index is calculated using two different methods.

EXPERIMENT: 2B:

Amplitude Modulation and Demodulation of DSBSC (LM741 and LF398 ICs can be used)

AIM: *To generate AM-Double Side Band Suppressed Carrier (DSB-SC) signal.*

Apparatus Required:

IC 1496 Wide frequency response up to 100 MHz, Internal power dissipation – 500mW (max)

Resistors 6.8KΩ, 10 KΩ, 3.9 KΩ, 1KΩ ,51 KΩ

Capacitors 0.1 μ F

Variable Resistor (Linear Pot) 0-50KΩ

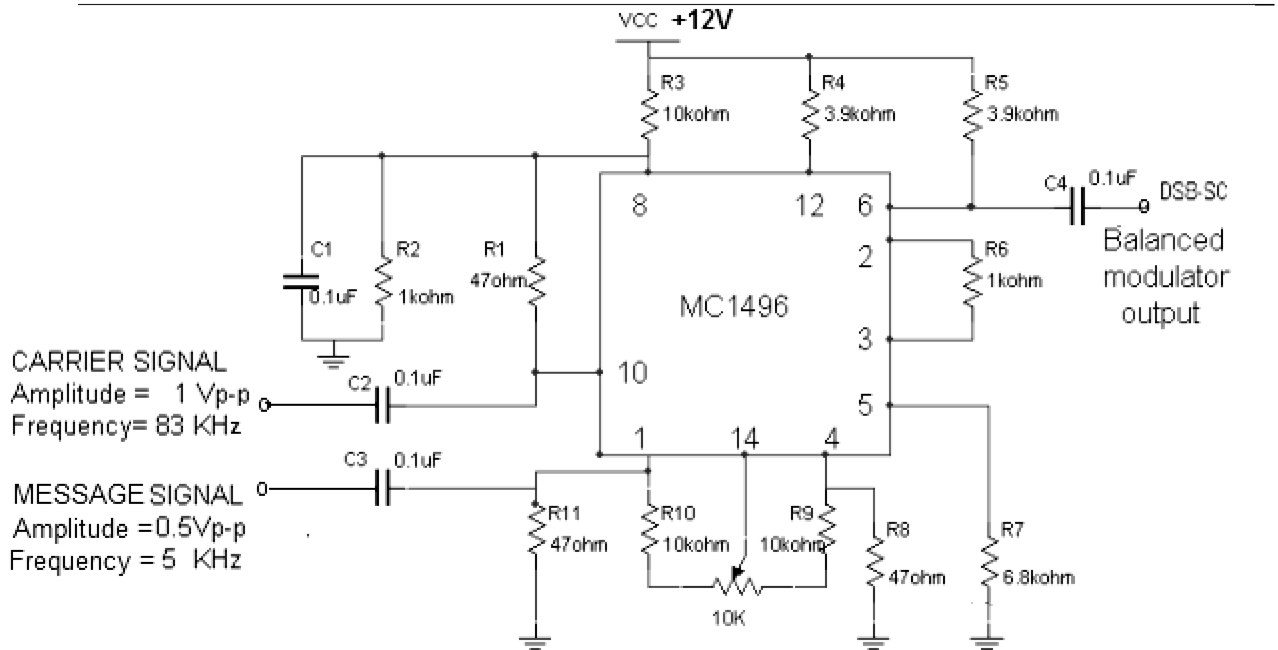
CRO 100MHz

Function Generator 1MHz	2
Regulated Power Supply 0-30 v, 1A	1

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. The IC MC 1496 is used as Modulator in this experiment. MC 1496 is a monolithic integrated circuit balanced modulator/Demodulator, is versatile and can be used up to 200 MHz.

CIRCUIT DIAGRAM



Procedure: Generation of DSB-SC waveform

1. Connect the circuit diagram as shown in Fig.1.

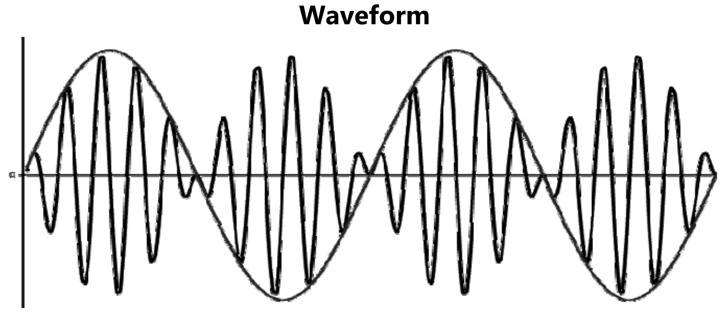
2. For the above circuit apply the modulating signal (AF) frequency in between 1Khz to 5Khz having 0.4 $V_{P\text{-}P}$ as message signal to pin no.1 and a carrier signal (RF) of 100KHz having a 0.1 $V_{P\text{-}P}$ as carrier to pin no.10.

3. Adjust the RF carrier null potentiometer to observe a DSB-SC waveform at the output terminal on CRO and plot the same.

4. Repeat the above process by varying the amplitude and frequency of AF. But RF is maintained constant.

5. Observe the DSB-SC waveform at pin no.12.

Waveform



RESULT: Amplitude Modulation and Demodulation of DSBSC is verified

EXPERIMENT: 03:

FREQUENCY MODULATION & DEMODULATION

AIM: 1. *To design & conduct an experiment to generate FM wave using IC 8038 & to find the parameters the modulation index β , the bandwidth of operation B_T & maximum frequency deviation δ .* 2. *To demodulate a Frequency Modulated signal using FM detector.*

Apparatus required:

IC 8038 Power dissipation – 750mW, Supply voltage - ±18V or 36V total
IC 565 Power dissipation -1400mw, Supply voltage - ±12V
Resistors, Capacitors
CRO 100MHz, Function Generator 1MHz
Regulated Power Supply 0-30 v, 1A

Theory:

The process, in which the frequency of the carrier is varied in accordance with the instantaneous amplitude of the modulating signal, is called "Frequency Modulation". The FM signal is expressed as

$s(t) = A_c \cos(2\pi f_c + \beta \sin(2\pi f_m t))$ Where A_C is amplitude of the carrier signal,

f_C is the carrier frequency β is the modulation index of the FM wave

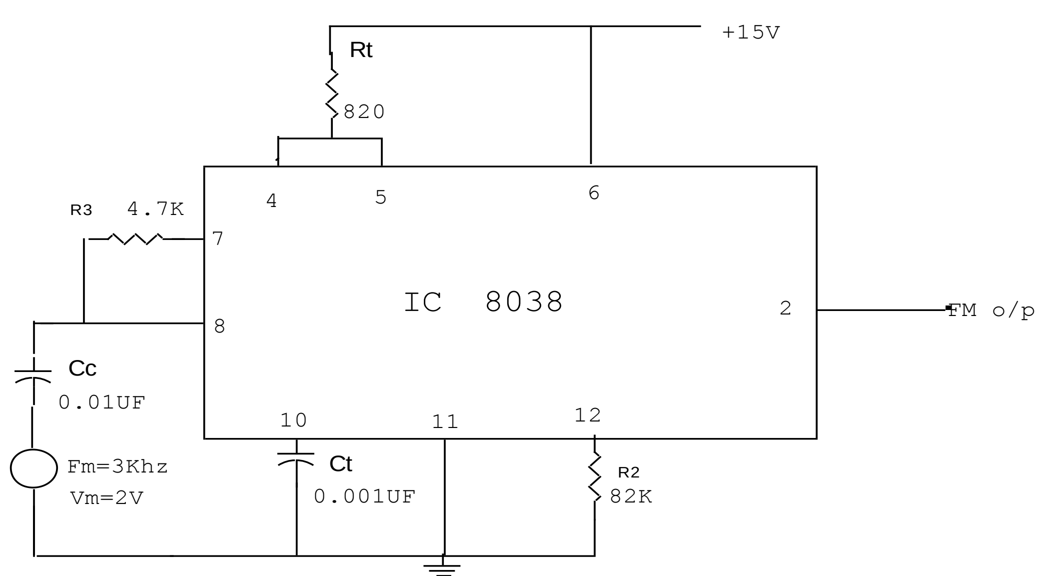
There are essentially two basic methods of generating frequency-modulated signal, namely direct FM and indirect FM. In direct FM the carrier signal is varied directly in accordance with the input base band signal, which is readily accomplished using a voltage-controlled oscillator. In the indirect method the modulating signal is first use to produce a narrow band FM signal, and frequency multiplication is next used to increase the frequency deviation to desired level. The indirect method is preferred choice for FM when the stability of the carrier is of major concern as in commercial radio broadcasting.

Modulation index = frequency deviation / modulating signal frequency

DESIGN:

Given: Fc = 170 KHz We have Fc = 0.159/Rt*Ct
Choose Ct = 0.001µF
Therefore, Rt = 0.159/Fc*Ct
Rt = 935Ω, Choose Rt = 820Ω

CIRCUIT DIAGRAM:



TABULAR COLUMN

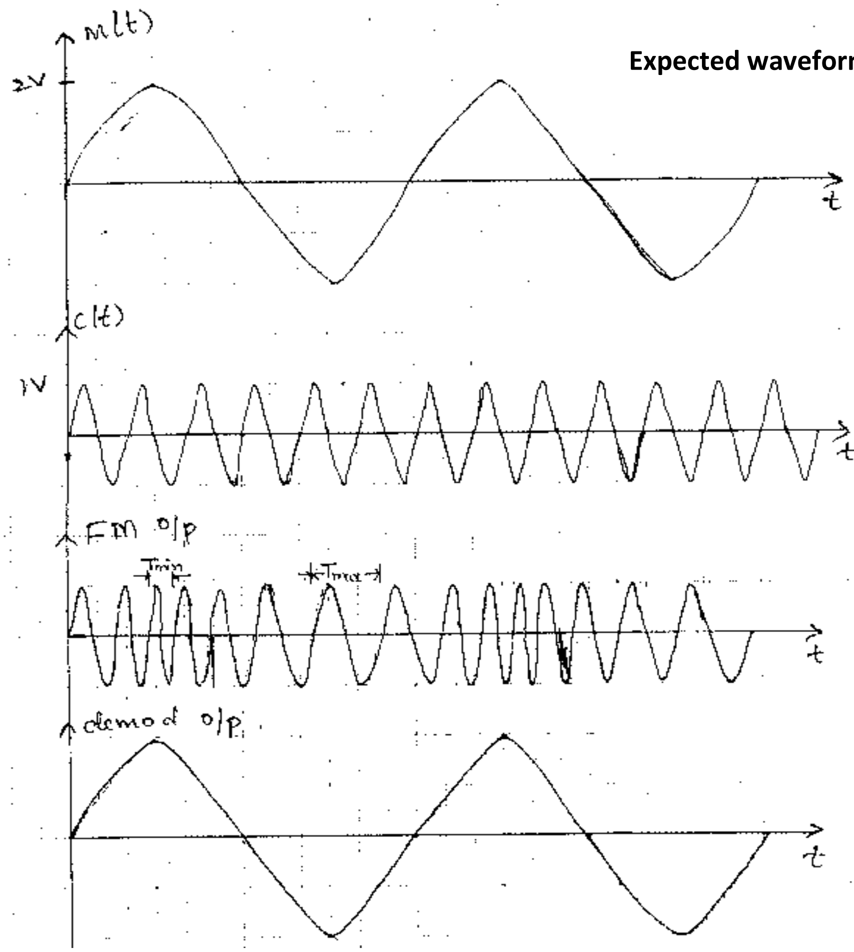
Fc(hz)	Fm(hz)	Vm(V)	Fcmax (hz)	Fcmin(hz)	δ1(hz)	δ2(hz)	β= δ/Fm	BT=2(δ+Fm)(hz)

$\delta 1 = F_{max} - Fc$, $\delta 2 = Fc - F_{min}$ δ = max of $\delta 1$ or $\delta 2$

PROCEDURE:

1. *Rig up the circuit as per the circuit diagram.*
2. *Switch OFF the message signal & note down the carrier frequency Fc.*
3. *Then Switch ON the message signal & note down the message frequency fm.*
4. *Adjust the amplitude of the message signal & observe the FM waveform.*
5. *Find Fmax & Fmin from the FM waveform.*
6. *Calculate the maximum frequency deviation δ , modulation index β & bandwidth of operation B_T.*

Expected waveforms



RESULT:

Fc (prac) = Fc(theo) = 170Khz. Fmax = β =

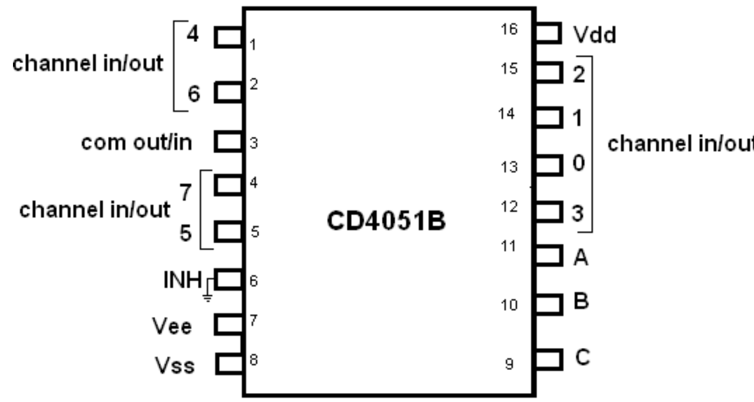
Fmin = δ = B_T =

EXPERIMENT: 4

Design and test Time Division Multiplexing and Demultiplexing of two band limited signals

AIM: *To conduct an experiment to study TDM for two band limited signals for two different signal frequencies.*

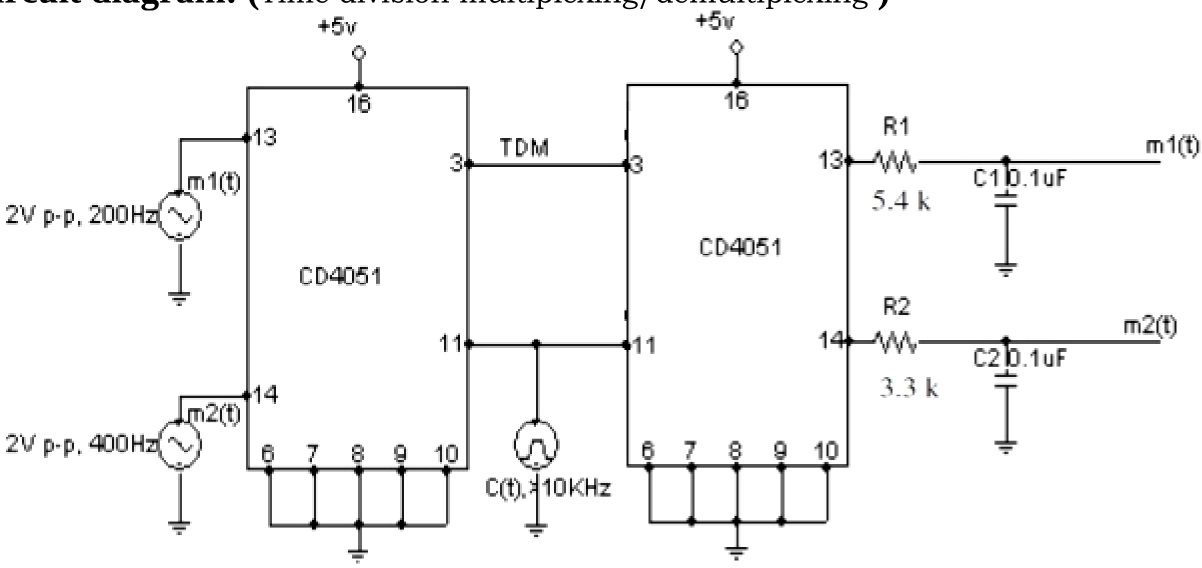
Pin configuration of CD4051B



Theory:

Time-division multiplexing (TDM) is a method of transmitting and receiving independent signals over a common signal path by means of synchronized switches at each end of the transmission line so that each signal appears on the line only a fraction of time in an alternating pattern. It is used when the bit rate of the transmission medium exceeds that of the signal to be transmitted. Two message signals are sine or triangular wave generated from frequency generator and they are time division multiplexed when square wave has ON and OFF Cycles. The multiplexed output is viewed on the CRO.

Circuit diagram: (Time division multiplexing/demultiplexing)



Procedure:

1. *Apply m1(t) and m2(t) whose frequencies are f1 (500 Hz sine wave, with DC offset) and f2 (1000 Hz triangular wave, with DC offset) to the pin13 and pin14 respectively.*
2. *Connect square wave signal of 10khz at pin11 using signal generator.*
3. *Observe TDM output signal at pin3 of multiplexing circuit.*
4. *Connect TDM output signal to the TDM input of demultiplexing ciruit at pin number 3 of 2nd CD4051*
5. *Observe the demultiplexed signals.*

OBSERVATION TABLE:

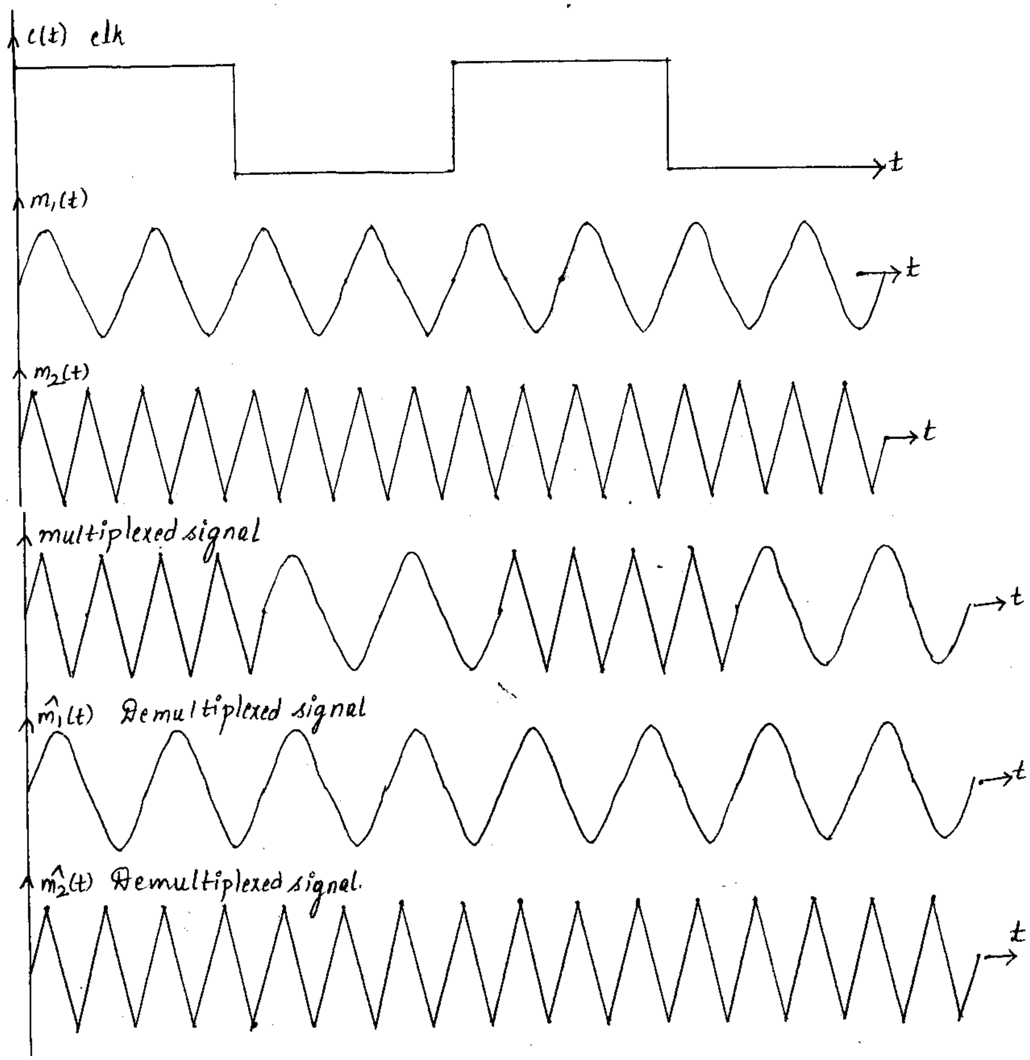
Multiplexing:

Signal	Amplitude	Time Period	Frequency
AF Signal 1			
AF Signal 2			
Clk			
TDM o/p AF_1			
TDM o/p AF_2			

De- multiplexing:

Signal	Amplitude	Time Period	Frequency
AF Signal 1			
AF Signal 2			

TD-Mux and TD-demux wave forms of two bandlimited signals



EXPERIMENT – 5 A
Design and test i) Pulse sampling, flat top sampling and reconstruction.

AIM: *To Design and test Pulse sampling, flat top sampling and reconstruction*

Apparatus required:

Op-amps: IC µA741
Transistor SL100
Resistors
Capacitor
Signal generators
Dual Power supplies
CRO
Connecting wires/probes

Theory:

The analog signal can be converted to a discrete time signal by a process called sampling. The sampling theorem for a band limited signal of finite energy can be stated as, ‘’a band limited signal of finite energy, which has no frequency component higher than W Hz is completely described by specifying the values of the signal at instants of time separated by 1/2W seconds.’’
It can be recovered from knowledge of samples taken at the rate of 2W per second.

DESIGN: Flat top sampling

RC<< $T_{M,}$ where T_M = 3.3ms, by assuming f_m = 300Hz

Let RC = 1ms,

Choose **C= 0.1µF** , R = 10KΩ

Sampling Circuit using transistor

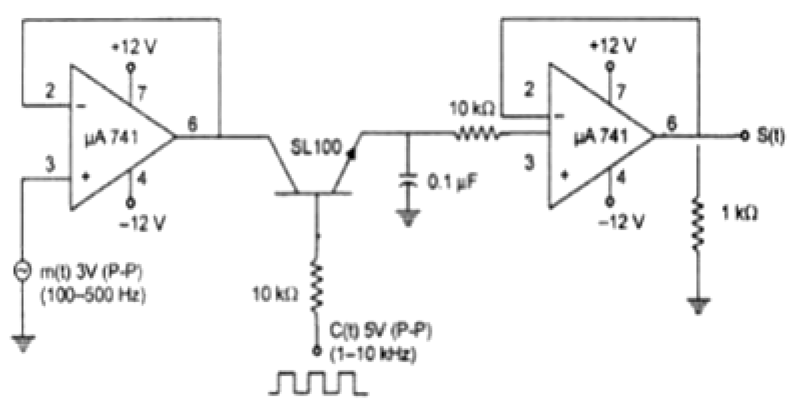


Fig.1

Design of Demodulation circuit:

$f = 1/(2\pi RC)$ =500Hz
Choose **C= 0.1µF**, R = 3.1KΩ ~ **3.3KΩ**

CIRCUIT DIAGRAM

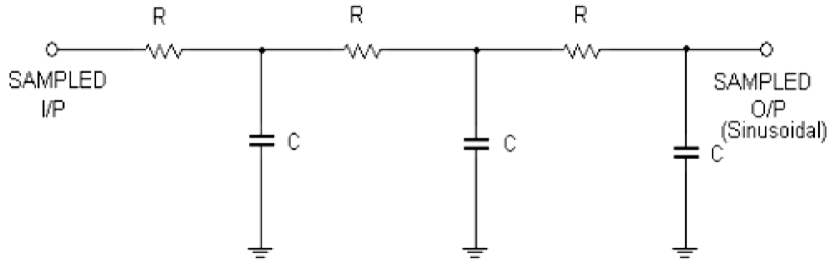


Fig.2

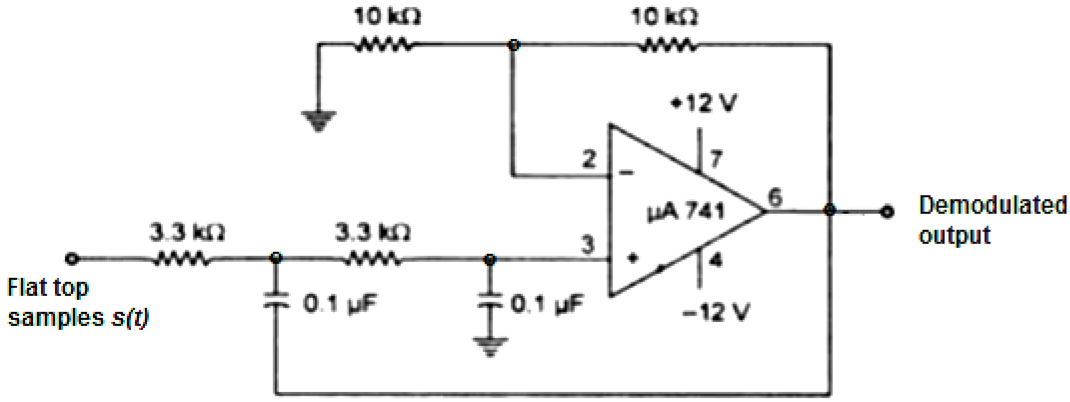
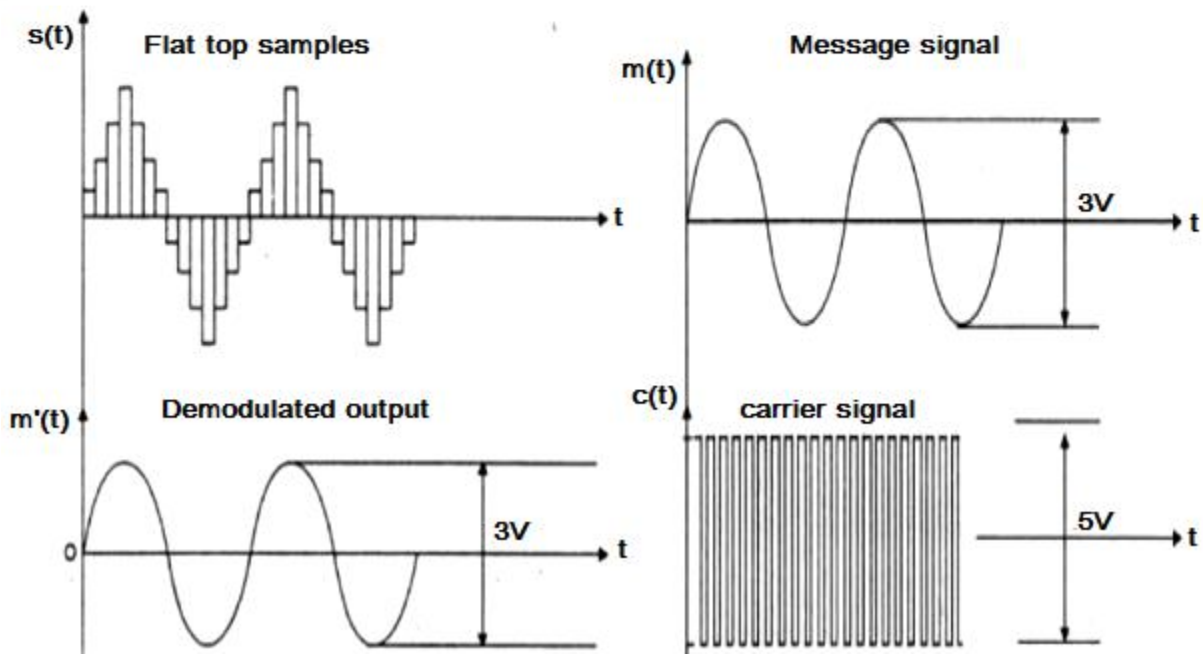


Fig.3

Expected Waveforms:



Procedure:

1. The circuit is connected as per the circuit diagram shown in the fig .1

2. Connect the power supply by setting +12V and -12V.

3. Apply the sinusoidal signal of approximately 3V (p-p) at 100-500 Hz frequency and pulse signal of 5V (p-p) with frequency between 100Hz and 10 KHz.

4. Connect the sampling circuit output and AF signal to the two inputs of oscilloscope

5. Initially set the sampling frequency to 200Hz and observe the output on the CRO. Now vary the amplitude of modulating signal and observe the output of sampling circuit. Note that the *amplitude of the sampling pulses will be varying in accordance with the amplitude of the modulating signal.*

6. *Design the reconstructing circuit. Depending on sampling frequency, R & C values are calculated using the relations Fs = 1/Ts, Ts = RC. Choosing an appropriate value for C, R can be found using the relation R=Ts/C*
7. *Connect the sampling circuit output to the reconstructing circuit shown in Fig.2 or Fig.3*
8. *Observe the output of the reconstructing circuit (AF signal) for different sampling frequencies. The original AF signal would appear only when the sampling frequency is 200Hz to 500Kz.*

RESULT: The sampling theorem is verified & various waveforms plotted.

EXPERIMENT – 5 B
Pulse amplitude modulation and demodulation

AIM: *To design & conduct an experiment to generate PAM signal to verify the sampling theorem & also to demodulate the PAM signal & also to plot the relevant waveforms.*

Apparatus required:

Transistor SL100
Resistors
Capacitor
Signal generators
Power supplies
CRO
Connecting wires/probes

Theory:

PAM is the simplest form of data modulation .The amplitude of uniformly spaced pulses is varied in proportion to the corresponding sample values of a continuous message m (t). A PAM waveform consists of a sequence of flat-topped pulses. The amplitude of each pulse corresponds to the value of the message signal x (t) at the leading edge of the pulse. The pulse amplitude modulation is the process in which the amplitudes of regularity spaced rectangular pulses vary with the instantaneous sample values of a continuous message signal in a one-one fashion

DESIGN: Modulation:

Given: Ic = 1mA, hfe = 100, Vce (sat) = 0.3V, Vbe (sat) = 0.7V, Fm = 100 Hz

Vm (t) = IcRc + Vce(sat)

Let Vm (t) = 2.5V+3V dc shift = 5.5V, Then Rc = 5.2KΩ

Vc (t) =IbRb + Vbe (sat)

Vc(t)=2Vp-p, Let Ib = Ic/hfe = 10µA, then Rb = 30KΩ

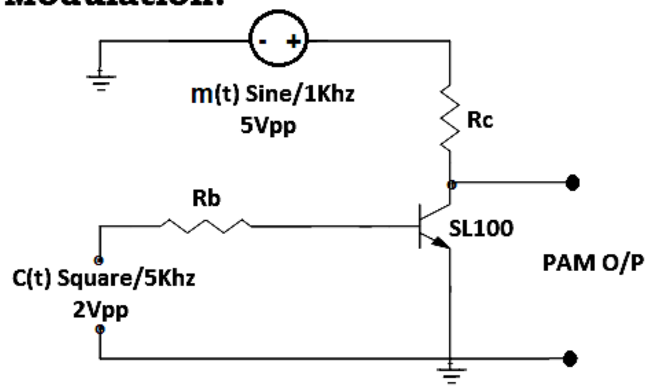
Demodulation: Filter:

Cut off frequency Fo =500hz

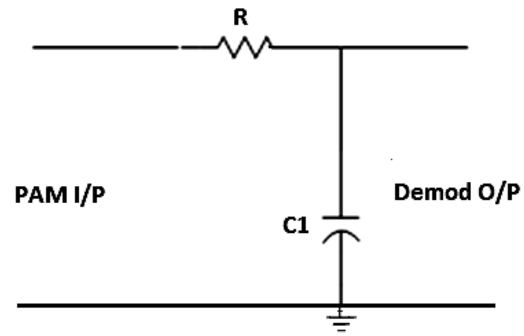
Fo = 1/2πRC, Let **C = 0.1µF,** then R = 3.3KΩ

CIRCUIT DIAGRAM:

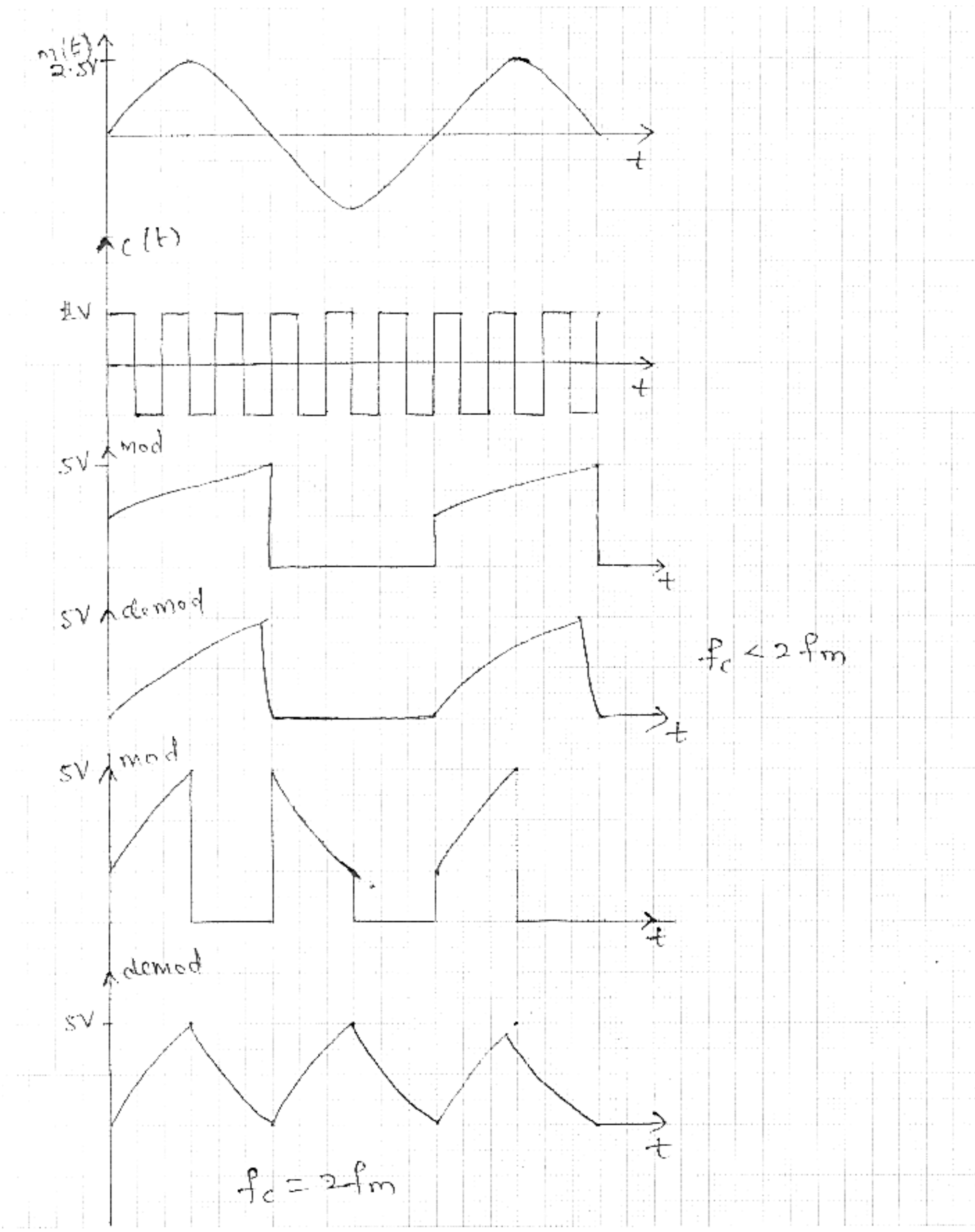
Modulation:



Demodulation:



PAM Modulation & Demodulation Waveforms:



PROCEDURE:

1. *Rig up the circuit as per the circuit diagram.*
2. *Initially Apply square wave carrier of 2Vp-p with frequency fc=5 KHz.*
3. *Apply sine wave modulating signal of 5Vp-p amplitude & 3V dc shift with frequency Fm=100 Hz.*
4. *Observe the PAM output.*
5. *Observe the demodulated signal at the output of low pass filter.*
6. *Plot the various waveforms.*
7. *Repeat the steps from 2 to 6 for fc<2fm, fc=2fm & fc>2fm.*

RESULT: PAM is verified & various waveforms plotted.

EXPERIMENT – 6

Design and test BJT/FET Mixer

Aim: *To design and obtain the characteristics of a mixer circuit.*

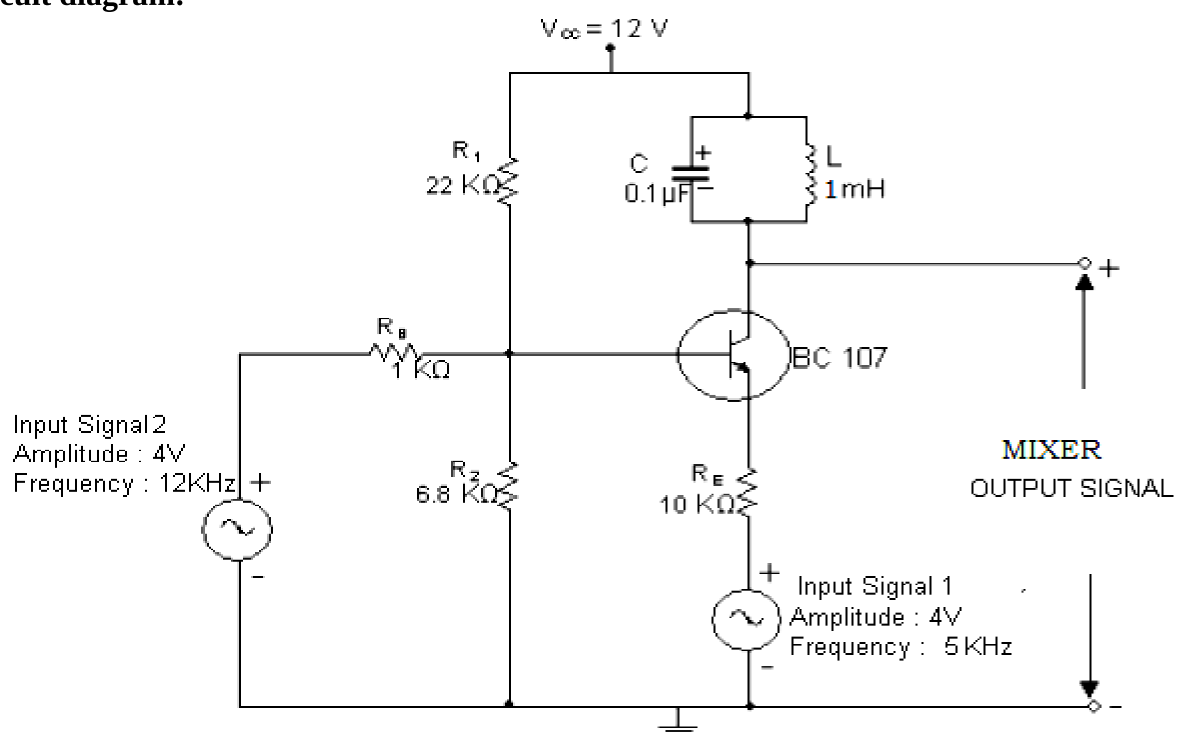
Apparatus Required:

Transistors (BC 107)
Resistors 1 KΩ, 6.8 KΩ, 10KΩ 1 each
Capacitor 0.1 μ F
Inductor 1mH
CRO 20MHZ
Function Generator 1MHz
Regulated Power Supply 0-30v, 1A

Theory:

The mixer is a nonlinear device having two sets of input terminals and one set of output terminals. Mixer will have several frequencies present in its output, including the difference between the two input frequencies and other harmonic components.

Circuit diagram:



Procedure:

1. Connect the circuit as shown in Fig. Assume C = 0.1µF and calculate value of L1 using

$$f = \frac{1}{2\pi\sqrt{L_1 C_1}} \quad \text{where } f = 7KHz$$

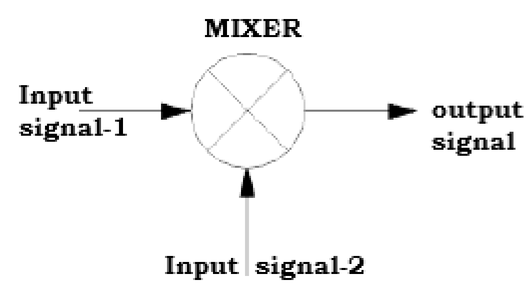
2. Apply the input signals at the appropriate terminals in the circuit.

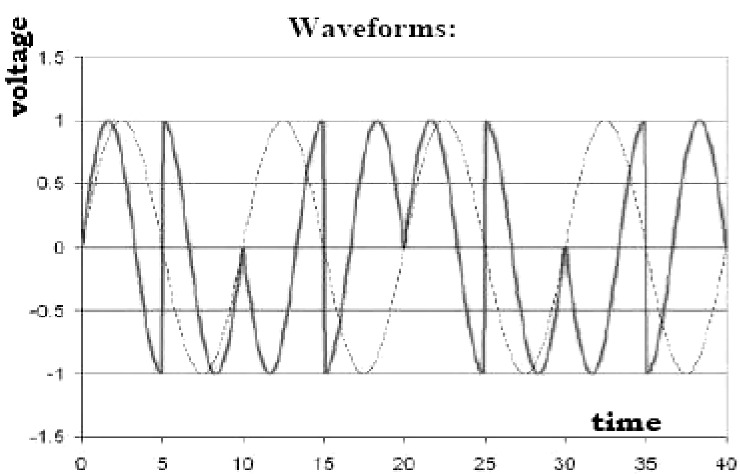
3. Note down the frequency of the output signal, which is same as difference frequency of given signals.

Sample readings:

Signal	Amplitude (Volts)	Frequency(KHz)
Input signal1	4	5
Input signal 2	4	12
Output signal	9	7

Block Diagram





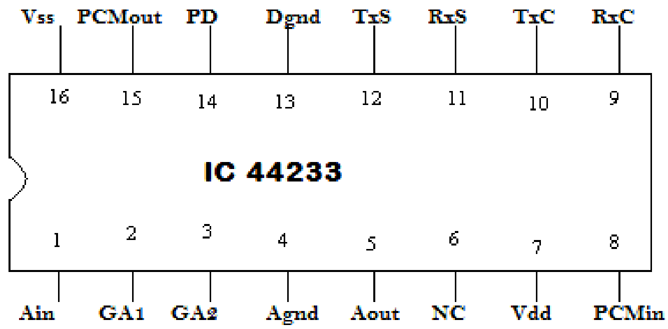
RESULT: Mixer is verified & waveforms plotted.

EXPERIMENT –7

Pulse Code Modulation and demodulation

AIM: *To study the performance of the given CODEC chip in implementing generation and detection of PCM wave.*

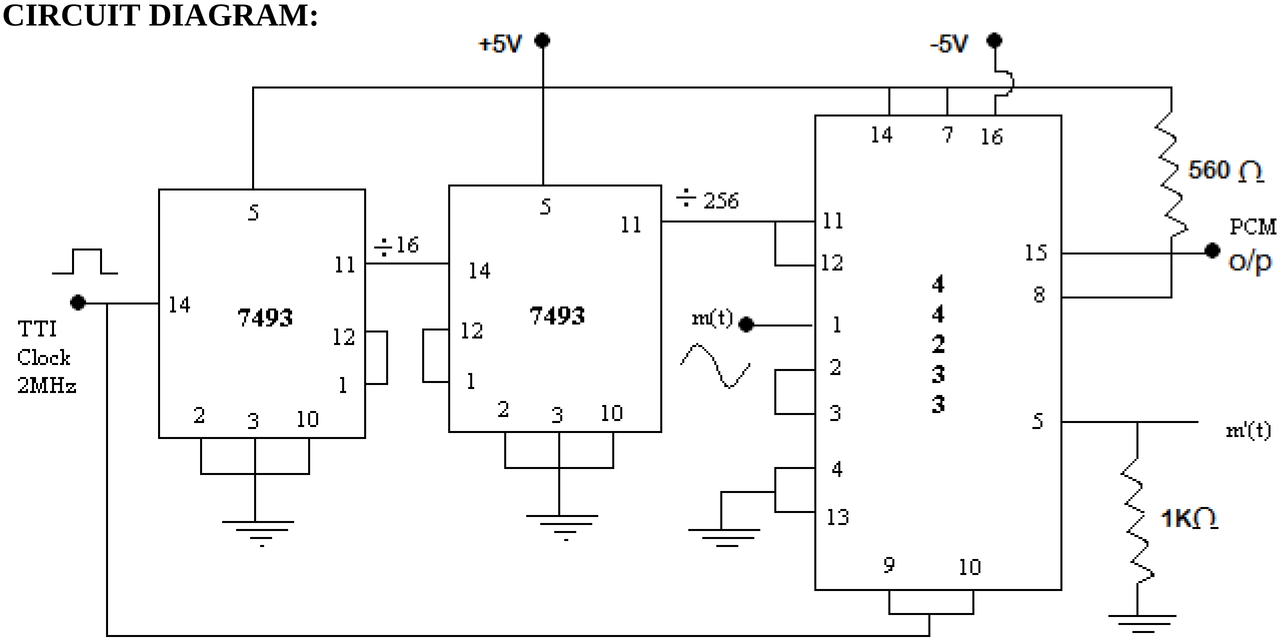
Pin Configuration of IC 44233



Theory:

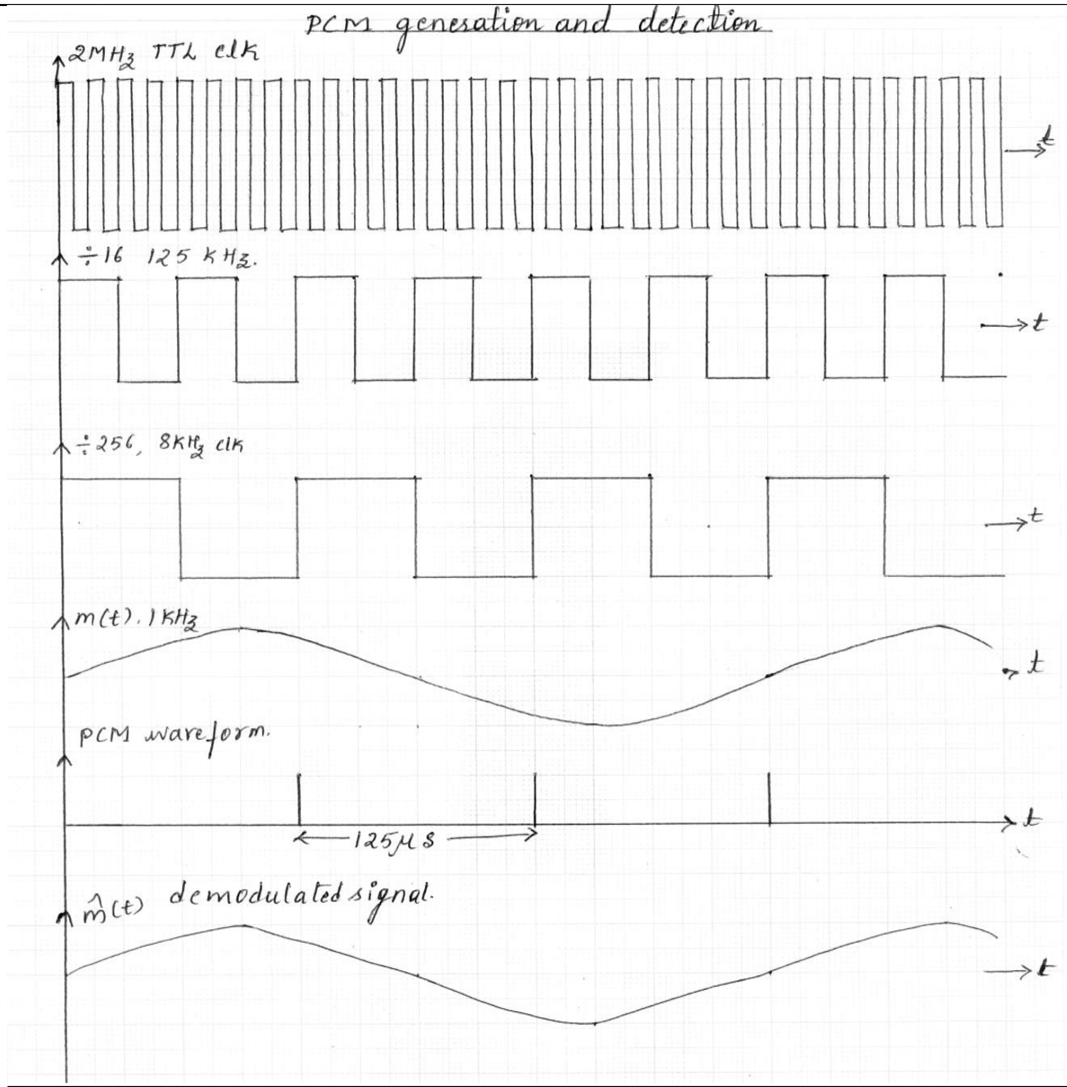
In Pulse code modulation (PCM) only certain discrete values are allowed for the modulating signals. The modulating signal is sampled, as in other forms of pulse modulation. But any sample falling within a specified range of values is assigned a discrete value. Each value is assigned a pattern of pulses and the signal transmitted by means of this code. The electronic circuit that produces the coded pulse train from the modulating waveform is termed a coder or encoder. A suitable decoder must be used

CIRCUIT DIAGRAM:



Procedure for PCM generation and detection

1. *Connections are made as per circuit diagram given in the figure.*
2. *Power supplies are switched on and applied the specific voltages.*
3. *TTL clocks of 2 MHz is applied to the counter IC 7493 at pin number 14 and observe the output using a CRO at pin number 11 that should be 125 kHz (divided by 16 of 2MHz).*
4. *Check the output at pin number 11 of the 2nd IC 7493, which will be approximately 8 kHz (divided by 256 of 2 MHz).*
5. *Apply a sinusoidal message frequency of 1 kHz, 1V at pin No.1 of IC 44233.*
6. *Observe the PCM output at pin No.8 of IC 44233. (Change the time range of CRO to convenient range to observe the frame time (50 μs range) and the 8-bit word length (0.5 μs range)).*
7. *Observe the demodulated output at pin number 5 of IC 44233 and compare it with original analog message. Observe the changes at the PCM output and demodulated output by changing the frequency and amplitude of the message signal.*



OBSERVATIONS: **PCM Modulation / Demodulation**

	Amplitude	Time period
AC input		
Sample and hold circuit		
Clock signal(4KHz)		
Clock signal(64KHz)		
PCM Output		
Demodulated output		

EXPERIMENT – 8

Phase Locked Loop (PLL)

Aim: *To study phase lock loop and its capture range, lock range and free running VCO*

Theory:

PLL has emerged as one of the fundamental building block in electronic technology. It is used for the frequency multiplication, FM stereo detector , FM demodulator , frequency shift keying decoders, local oscillator in TV and FM tuner. It consists of a phase detector, a LPF and a voltage controlled oscillator(VCO) connected together in the form of a feedback system. The VCO is a sinusoidal generator whose frequency is determined by a voltage applied to it from an external source. In effect, any frequency modulator may serve as a VCO. The output frequency of the VCO is directly proportional to the input DC level. The VCO frequency is compared with the input frequencies and adjusted until it is equal to the input frequency. In short, PLL keeps its output frequency constant at the input frequency.

Thus, the PLL goes through 3 states: 1. Free running state, 2. Capture range / mode & 3. Phase lock state.

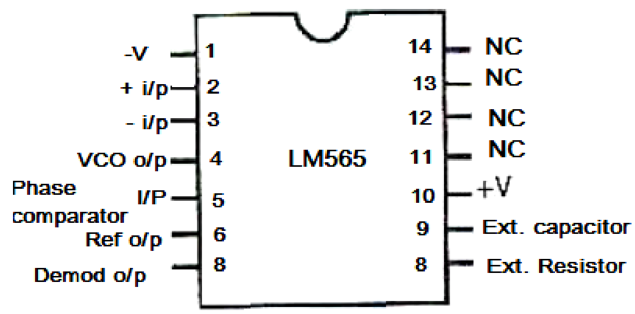
Before input is applied, the PLL is in the free running state. Once the input frequency is applied, the VCO frequency starts to change and the PLL is said to be the capture range/mode. The VCO frequency continues to change (output frequency) until it equals the input frequency and the PLL is then in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

Lock Range or Tracking Range:

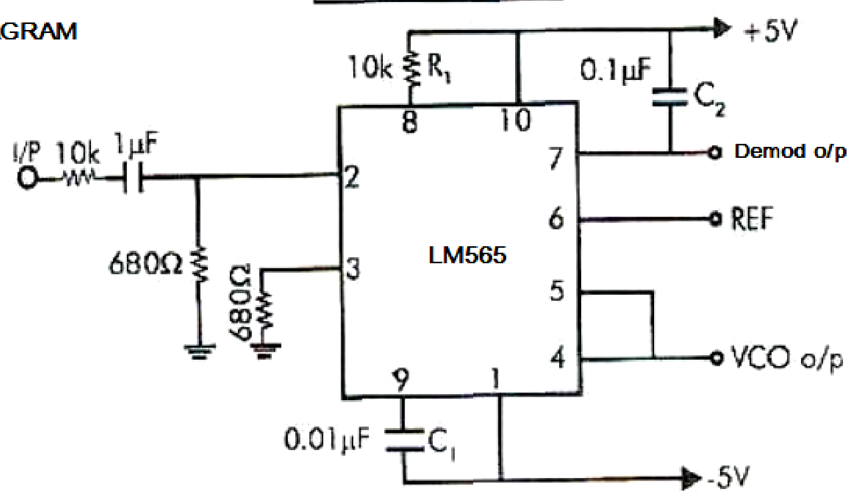
It is the range of frequencies in the vicinity of ‘f O’ over which the VCO, once locked to the input signal, will remain locked.

Capture Range: (f $_C$) : Is the range of frequencies in the vicinity of ‘f O’ over which the loop will acquire lock with an input signal initially starting out of lock.

LM565 IC PIN diagram



CIRCUIT DIAGRAM



Procedure:

1. *Connect + 5V to pin 10 and connect -5V to pin 1of LM 565.*
2. *Remaining connections are as shown in the circuit diagram.*
3. *Without giving input signal, measure (f_O) free running frequency.*
4. *Connect pin 2 to oscillator or function generator through a 1 μ f capacitor, adjust the amplitude around 2Vpp. Connect output to the second channel of the CRO.*
5. *By varying the frequency in different steps observe that of one frequency the wave form will be phase locked. Change R-C components to shift VCO center frequency and see how lock range of the input varies.*

PART-B

Simulation using MATLAB

1. Illustration of (a) AM modulation and demodulation and display the signal and its spectrum. (b) DSB-SC modulation and demodulation and display the signal and its spectrum.
2. Illustration of FM modulation and demodulation and display the signal and its spectrum.
3. Illustrate the process of sampling and reconstruction of low pass signals. Display the signals and its spectrums of both analog and sampled signals.
4. Illustration of Delta Modulation and the effects of step size selection in the design of DM encoder.

General Instructions for all the MATLAB programs

1. Click on the MATLAB icon on the desktop.
2. MATLAB window open.
3. Click on the 'FILE' Menu on the menu bar.
4. Click on NEW M-File from the File Menu.
5. An editor window opens, start typing commands.
6. Now SAVE the file in a directory in the format USN
7. Then Click on Run

EXPERIMENT – 1 A

Illustration of AM modulation and demodulation and display the signal and its spectrum.

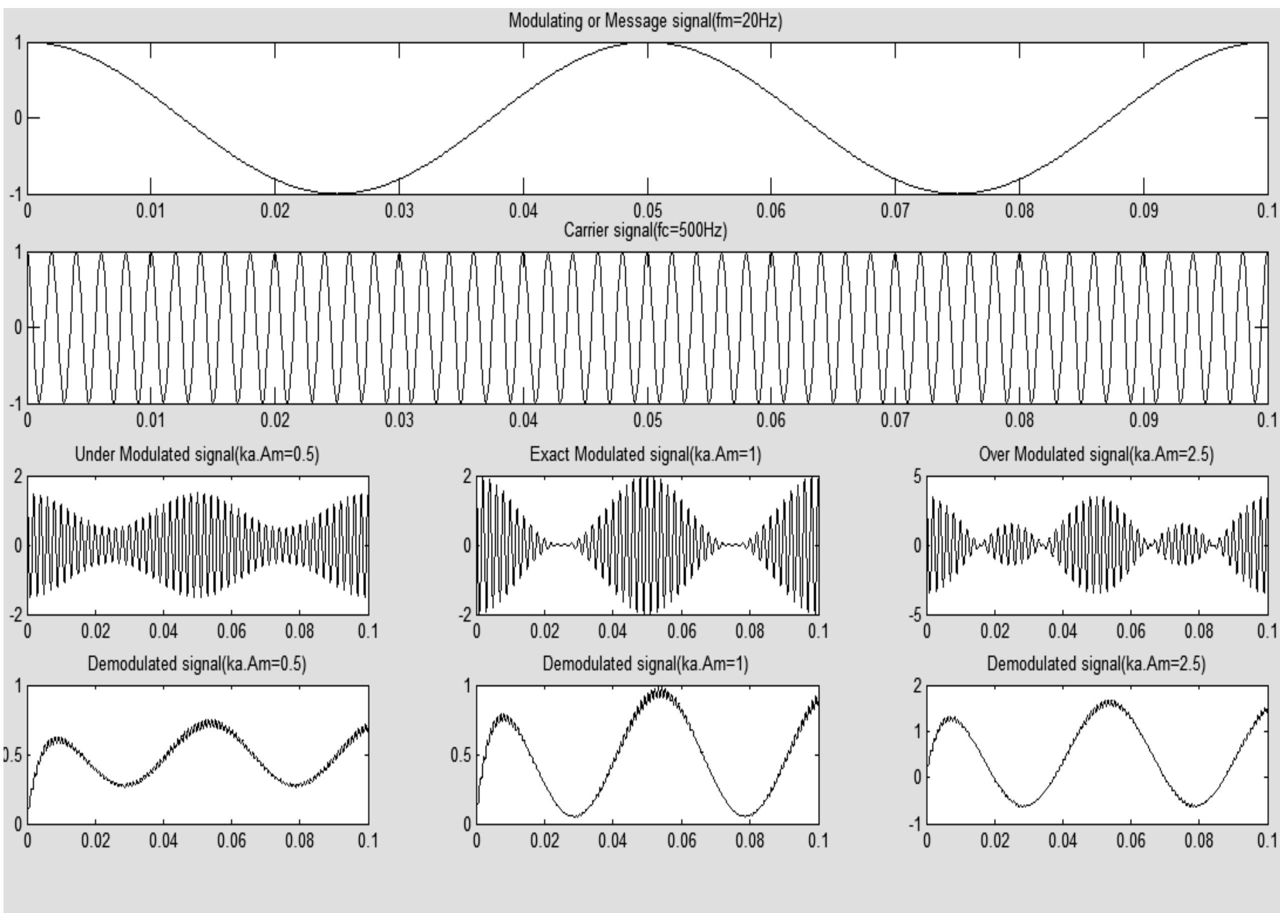
Aim: 1. To generate AM wave and to demodulate the modulated wave using MATLAB program
2. To display time domain signals and spectrum using MATLAB program.

MATLAB Program to generate AM wave and to demodulate the modulated wave

```
clc;
clear all;
close all;
fs=8000; fm=20; fc=500; Am=1; Ac=1;
t=[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;
%Demodulated signal
[b a] = butter(1,0.01);
mr1= filter(b,a,r1);
```

```
subplot(4,3,10);
plot(t,mr1);
title('Demodulated signal(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(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(ka.Am=2.5)');
```

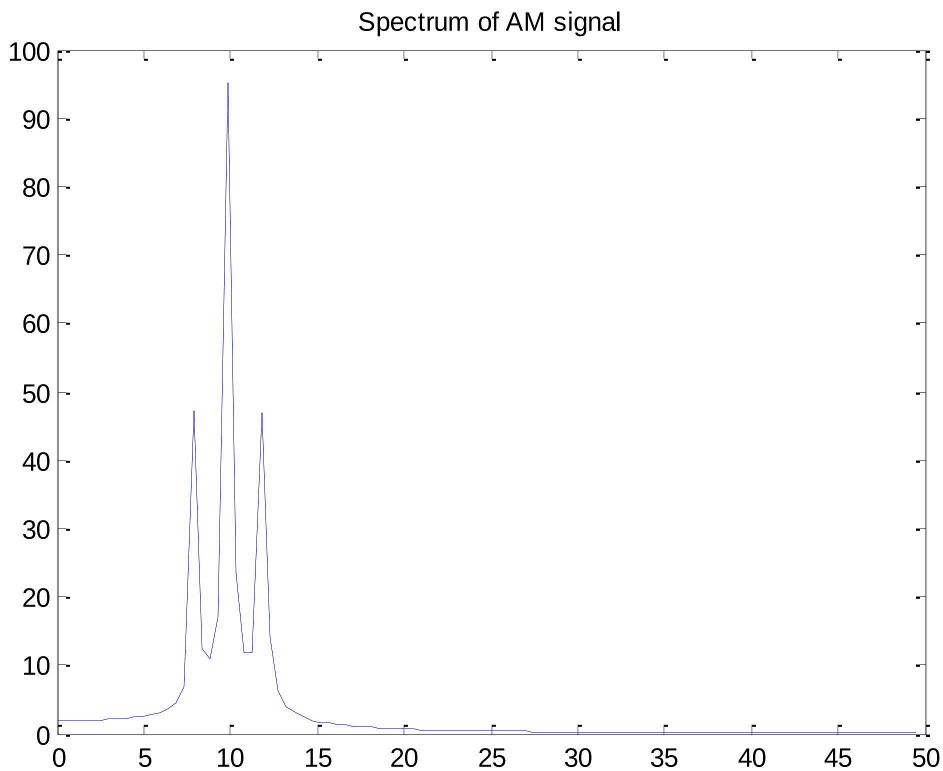
EXPECTED WAVEFORM



SPECTRUM OF AM

```
%program of spectrum analyzer and analysis of AM
close all
close all
clear all
clc
```

```
Fs = 100; %sampling frequency
t = [0:2*Fs+1]'/Fs;
Fc = 10; % Carrier frequency
x = sin(2*pi*2*t); % message signal
Ac=1; % Carrier amplitude

% 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;

% Plot spectrum of AM
figure;
plot(frqam,zam);
title('Spectrum of AM signal');
```



Result: MATLAB program is executed and AM modulation, demodulation and spectrum is plotted & verified.

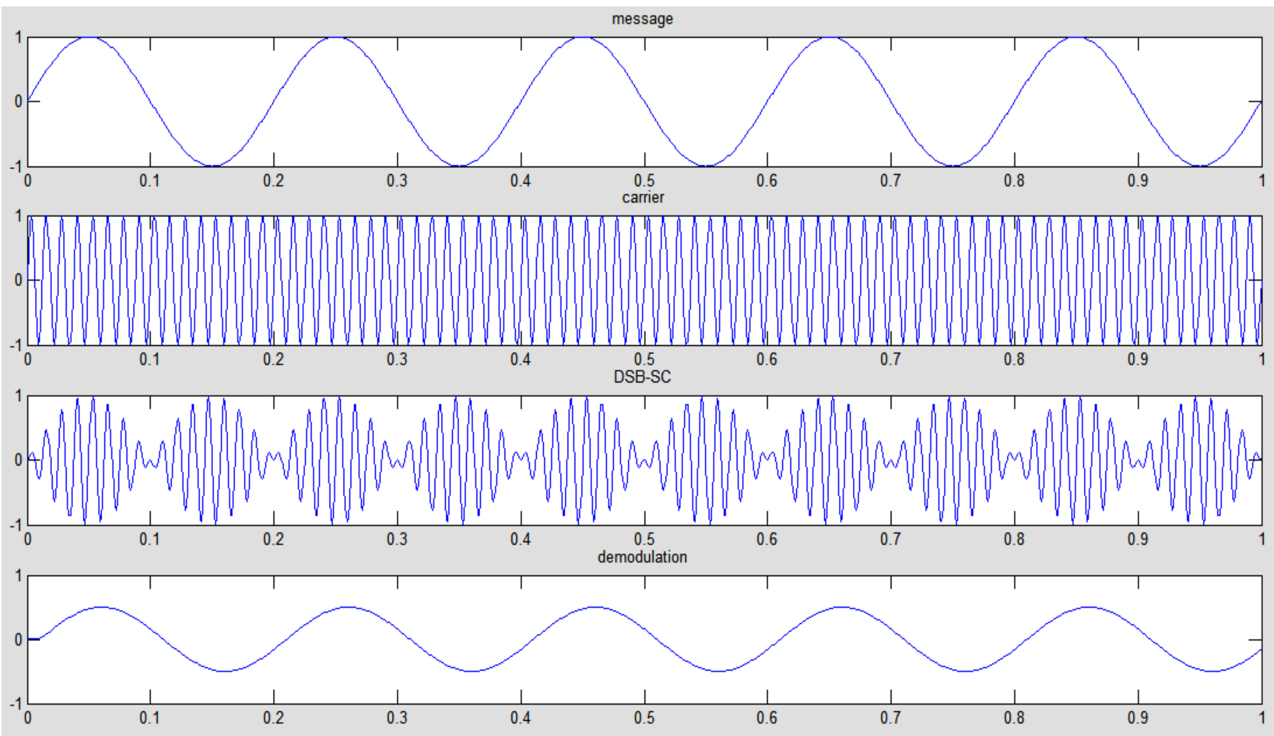
EXPERIMENT – 1 B

Illustration of DSB-SC modulation and demodulation and display the signal and its spectrum.

AIM: To generate **and demodulate DSB-SC wave and display the signal**

```
clc;
clear all;
close all;
t=[0:0.001:1];
f1=5;
m=sin(2*pi*f1*t);
```

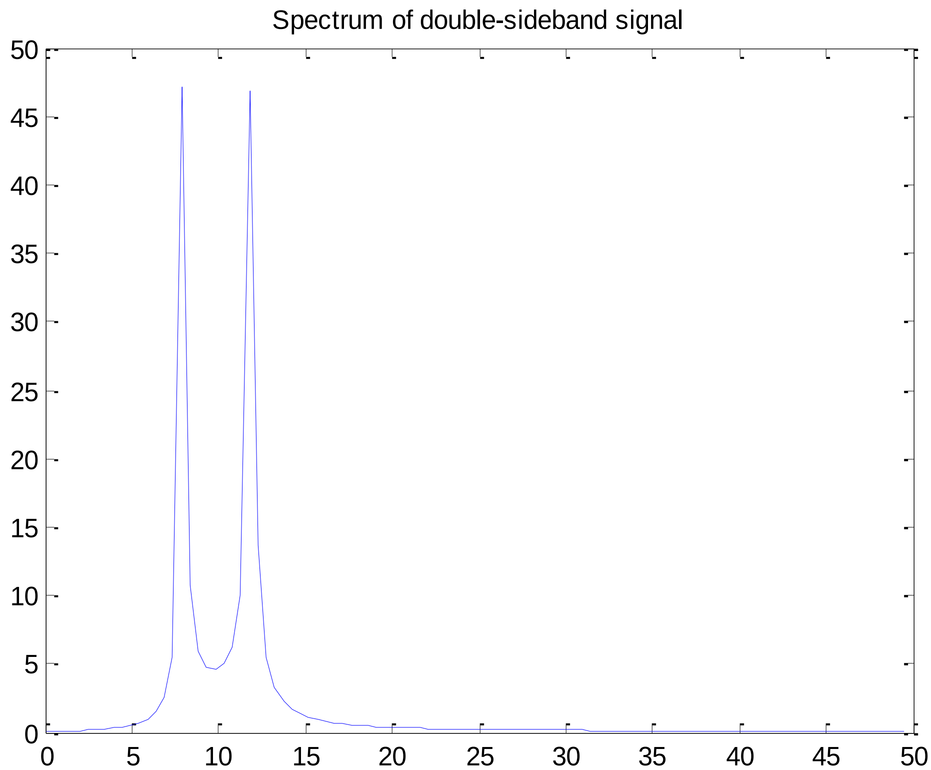
```
subplot(4,2,[1,2]);
plot(t,m);
title('message');
f2=80;
c=sin(2*pi*f2*t);
subplot(4,2,[3,4]);
plot(t,c);
title('carrier');
s=m.*c;
subplot(4,2,[5,6]);
plot(t,s);
title('DSB-SC');
s1=s.*c;
[b,a]=butter(5,0.1);
s2=filter(b,a,s1);
subplot(4,2,[7,8]);
plot(t,s2);
title('demodulation');
```



SPECTRUM OF DSB-SC wave

```
%program of spectrum analyzer and analysis of DSB-SC
close all
clear all
clc
Fs = 100; %sampling frequency
t = [0:2*Fs+1]'/Fs;
Fc = 10; % Carrier frequency
x = sin(2*pi*2*t); % message signal
Ac=1; % Carrier amplitude

% compute spectra of DSB-SC wave
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;

% Plot spectrums of am dsbsc
figure;
plot(frqdouble,zdouble);
title('Spectrum of double-sideband signal');
```



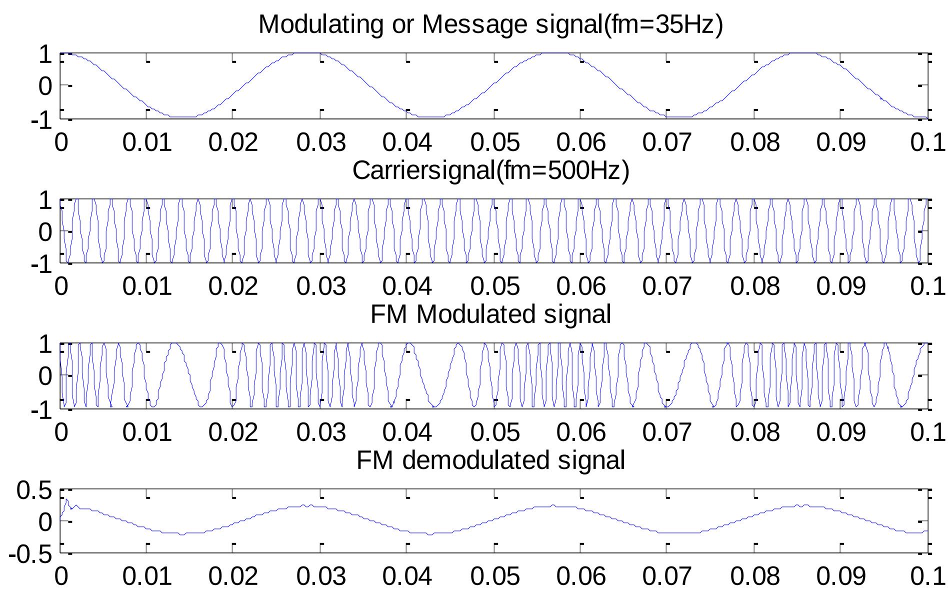
EXPERIMENT – 2

Illustration of FM modulation and demodulation and display the signal and its spectrum.

Aim: To Illustrate FM modulation and demodulation and display the signal and its spectrum using MATLAB

```
%The frequency modulation (FM)waveform in time and frequency domain.
%fm=35HZ,fc=500HZ,Am=1V,Ac=1V,B=10
fs=10000;
Ac=1;
Am=1;
fm=35;
fc=500;
B=10;
t=(0:0.1*fs)/fs;
wc=2*pi*fc;
wm=2*pi*fm;

m_t=Am*cos(wm*t);% Message signal
subplot(5,1,1);
plot(t,m_t);
title('Modulating or Message signal(fm=35Hz)');
```

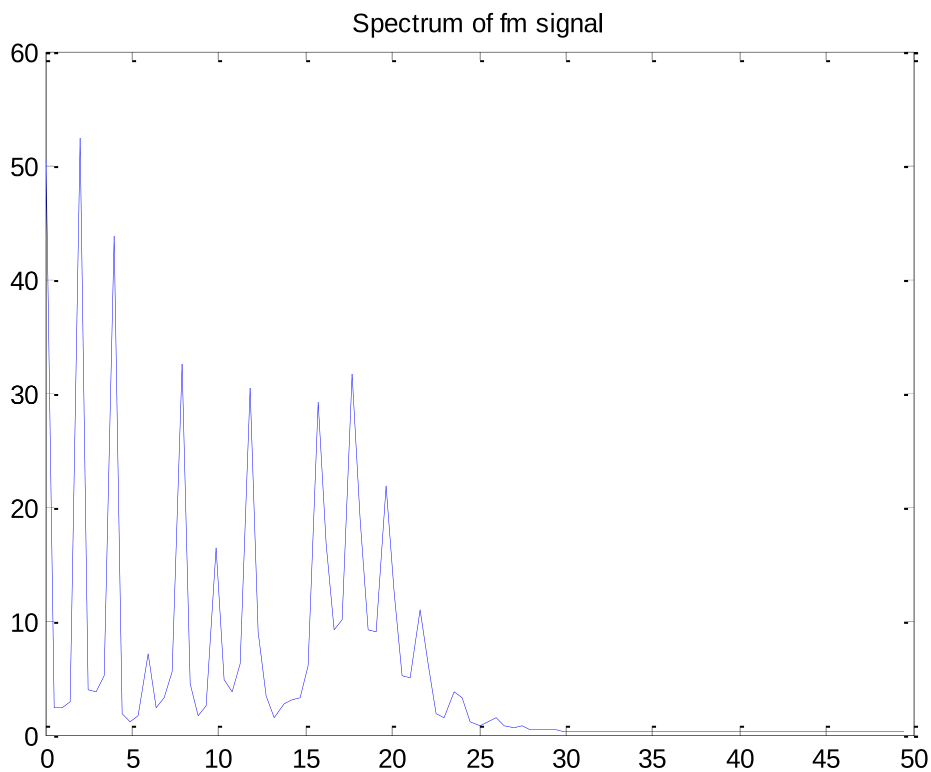
```
c_t=Ac*cos(wc*t);% Carrier signal
subplot(5,1,2);
plot(t,c_t);
title('Carrier signal(fm=500Hz)');

s_t=Ac*cos((wc*t)+B*sin(wm*t));% FM Modulated signal
subplot(5,1,3);
plot(t,s_t);
title('FM Modulated signal');

%DEMODULATION
d=demod(s_t,fc,fs,'fm');
subplot(5,1,4);
plot(t,d);
title('FM demodulated signal');
```



SPECTUM OF FM WAVE:

```
%program of spectrum analyzer and analysis of Fm
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;
% 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');
```



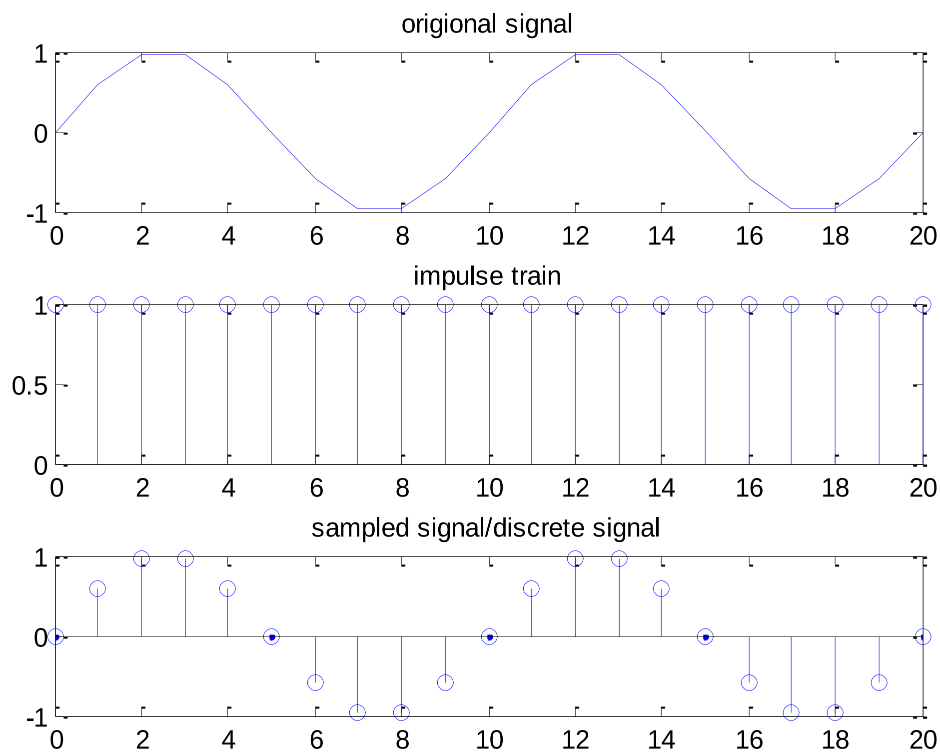
EXPERIMENT – 3

Illustrate the process of sampling and reconstruction of low pass signals. Display the signals and its spectrums of both analog and sampled signals.

To illustrate the process of sampling and reconstruction of low pass signals

```
clc
clear all
close all
m=1;
N=20;
n=0:m:N;
d=(n==0:m:N);
f=100;
fs=1000;
b=sin(2*pi*(f/fs)*n);
y=d.*b;
subplot(3,1,1)
plot(n,b);
title('origional signal');
subplot(3,1,2)
stem(n,d);
```

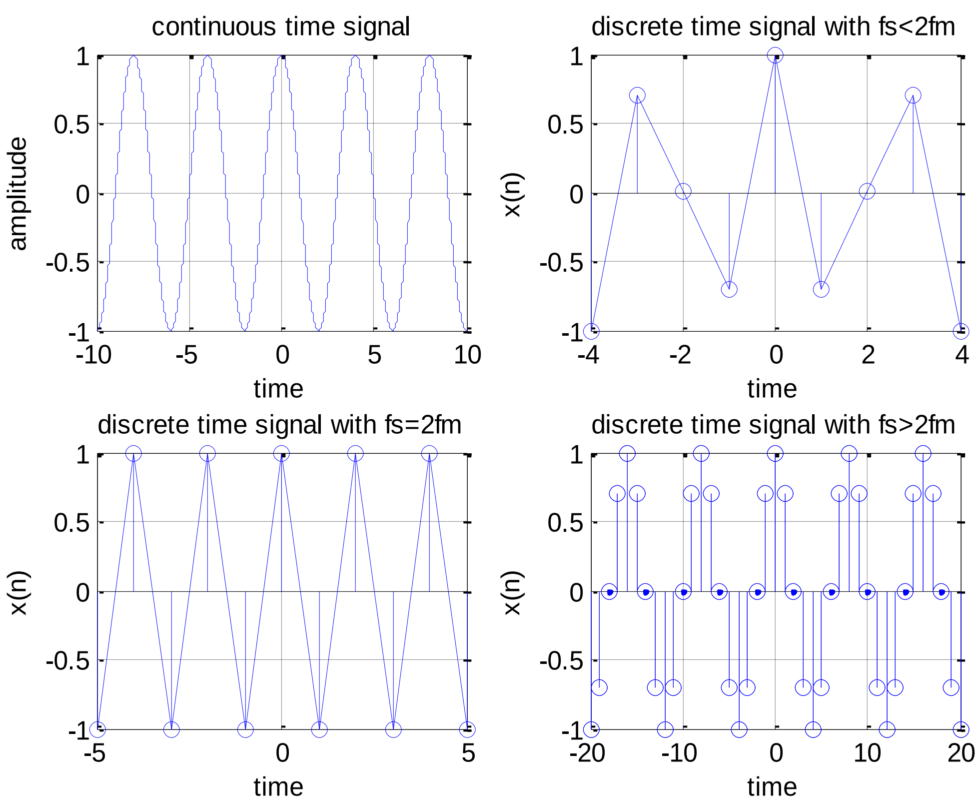
```
title('impulse train');
subplot(3,1,3)
stem(n,y, 'r');
title('sampled signal/discrete signal');
```



Effect of Nyquist rate

```
close all;
clear all;
clc;
t=-10:.01:10;
T=4;
fm=1/T;
x=cos(2*pi*fm*t);
subplot(2,2,1);
plot(t,x);
grid
title('continuous time signal');
xlabel('time');
ylabel('amplitude');
n1=-4:1:4;
fs1=1.6*fm;
fs2=2*fm;
fs3=8*fm;
x1=cos(2*pi*fm/fs1*n1);
subplot(2,2,2);
stem(n1,x1);
title('discrete time signal with fs<2fm');
xlabel('time');
ylabel('x(n)');
hold on
```

```
subplot(2,2,2);
plot(n1,x1);
grid
n2=-5:1:5;
x2=cos(2*pi*fm/fs2*n2);
subplot(2,2,3);
stem(n2,x2);
title('discrete time signal with fs=2fm');
xlabel('time');
ylabel('x(n)');
hold on
subplot(2,2,3);
plot(n2,x2);
grid
n3=-20:1:20;
x3=cos(2*pi*fm/fs3*n3);
subplot(2,2,4);
stem(n3,x3)
title('discrete time signal with fs>2fm');
xlabel('time');
ylabel('x(n)');
hold on
subplot(2,2,4);
plot(n3,x3);
grid
```



EXPERIMENT – 4

Illustration of Delta Modulation and the effects of step size selection in the design of DM encoder

```
%delta modulation = 1-bit differential pulse code modulation (DPCM)
predictor = [0 1]; % y(k)=x(k-1)
%partition = [-1:.1:.9];codebook = [-1:.1:1];
step=0.2; %SFs>=2pifA
partition = [0];codebook = [-1*step step]; %DM quantizer
 t = [0:pi/20:2*pi];
x = 1.1*sin(2*pi*0.1*t); % Original signal, a sine wave

% Quantize x(t) using DPCM.
encodedx = dpcmenco(x,codebook,partition,predictor);
 % Try to recover x from the modulated signal.
decodedx = dpcmdeco(encodedx,codebook,predictor);
distor = sum((x-decodedx).^2)/length(x) % Mean square error
 % plots
figure,
subplot(2,2,1);plot(t,x);
xlabel('time');title('original signal');
subplot(2,2,2);
stairs(t,10*codebook(encodedx+1),'--');
xlabel('time');title('DM output');
subplot(2,2,3);plot(t,x);
hold;
stairs(t,decodedx);
grid;xlabel('time');title('received signal');
```