ADVANCED ELECTRONIC COMMUNICATION SYSTEM LABORATORY MANUAL

IV SEMESTER B.Sc PAPER IV

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DEPARTMENT OF ELECTRONICS

LAB RULES

- **Students are not allowed to carry bags inside the lab.**
- **Loitering and throwing litter in the lab is strictly prohibited.**
- **Usage of mobile phones within the lab is restricted.**
- **The required lab equipment and components are to be used only with lab assistant supervision.**
- **The students should maintain the lab manuals and up-to-date instructions given by the lab in-charge.**
- **Use the lab equipment only after instructions given by the lab in-charge.**
- **While leaving the lab, return the electronic components used in the experiments and arrange the chairs.**
- **A student must take care of lab equipment and is expected to protect it from getting damaged/vandalized.**
- **Do not change the settings of equipment without lab in-charge permission.**
- **Students should maintain lab observation during the lab hours.**
- **Fans and lights should be switched off after use.**

Sl.no	Date	Experiment Name	Marks	Remarks	Signature of faculty
1.					
2.					
3.					
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12.					
13.					

INDEX

Experiment No: 1

AMPLITUDE MODULATOR AND DEMODULATOR

Aim: To construct an amplitude modulator and de-modulator circuit and to display the output wave form and also determine the modulation index.

Components: Transistor, CRO, multimeter, signal generators, diode(OA79), Resistors Capacitors, inducatance box, power supply, connecting wires.

Theory: In amplitude modulation, amplitude of the high frequency carrier wave is varied in accordance with the instantaneous voltage of the modulating signal. In this circuit the amplitude modulation takes place in emitter section. The net emitter potential varies in accordance with the instantaneous voltage of the modulating signal given by $V_E=V_E+V_m\sin\omega t$. The emitter bias current and hence the gain of the amplifier varies in accordance with the modulating voltage.

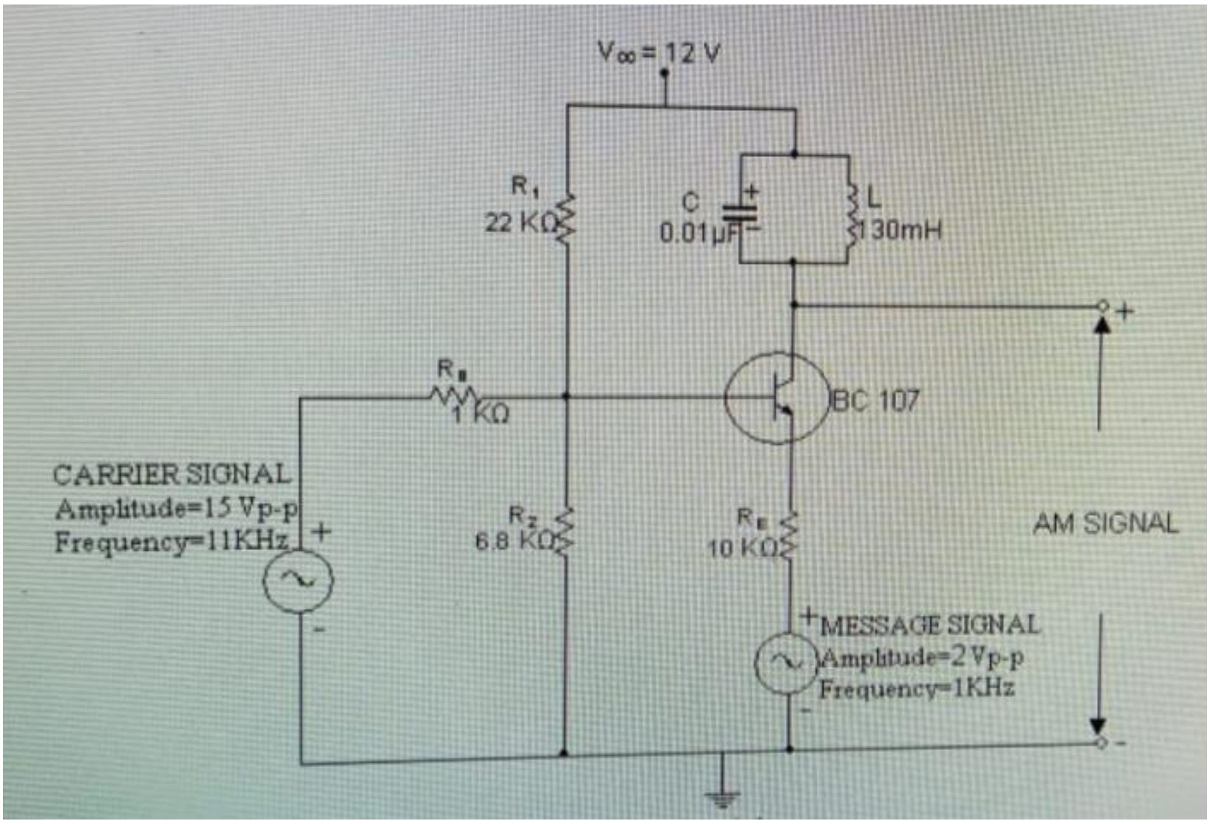
Procedure:

- Rig up the circuit as shown in fig i.
- Amplitude and the frequency of carrier wave input ia adjusted to 10V, 455kHz.
- The frequency of modulating signal is set to 1KHz and the V_m is set to low value.
- The AM output is observed in CRO & the modulation index I is calculated using the formula $m_a = E_{max} – E_{min} / E_{max} + E_{min}$.
- Rig up the circuit as shown in fig ii.
- Amplitude modulated wave with a certain m_a less than 100% is applied at the input of the second circuit and the demodulated output is observed in CRO.

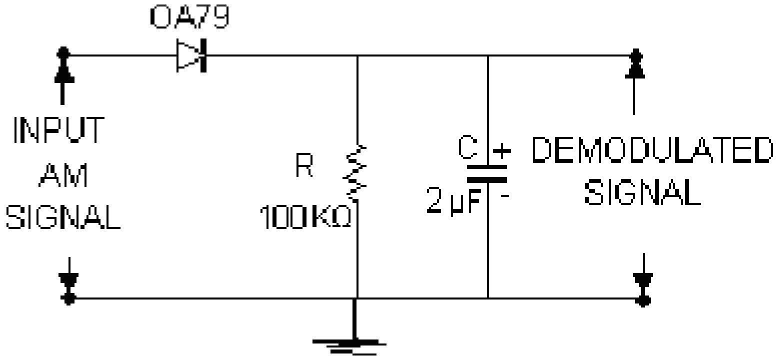
Result: Amplitude modulator and de modulator circuits are constructed and the output waveforms are observed.

CIRCUIT DIAGRAM:

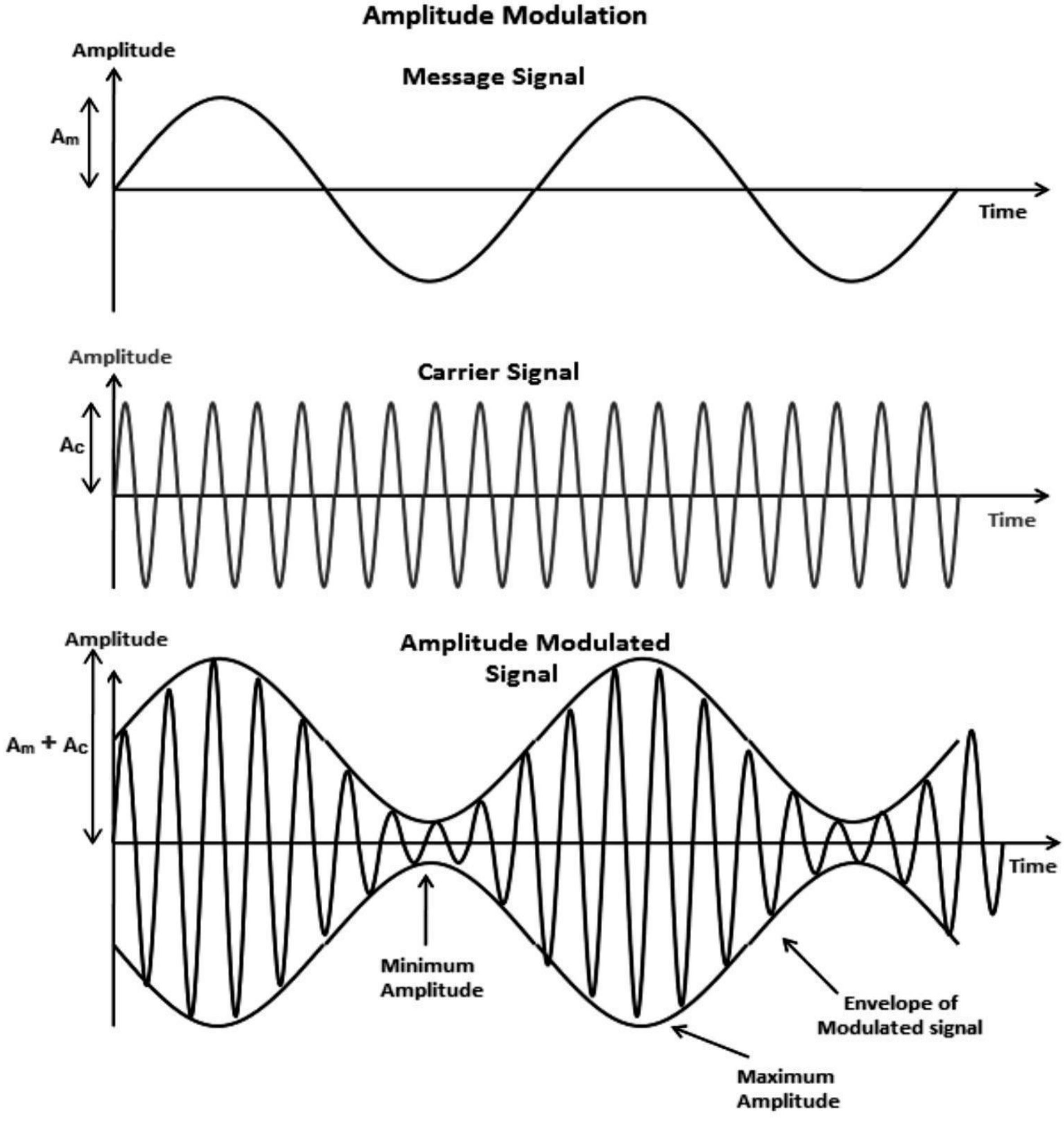
Modulator circuit



<u>De-modulator circuit</u>



Waveforms:



Tabular Column:

Carrier amplitude,Ec(V)= ;Carrier Frequency ,fc= ;

Modulating signal frequency, fm=

E_m(V)	E_{max}(V)	E_{min}(V)	$m_{a(theoretical)}$	$m_{a(practical)}$

Experiment No:2

AUDIO CROSS OVER NETWORK

Aim: To construct and study the frequency response of three way audio cross over network using resistive loads.

Components:

CRO, signal generator, decade inductance box, decade capacitance box and decade resistance box.

Principle: To realize a realistic audio output of high fidelity stereo systems cross over networks are used. These networks provide the desired audio effects.

Three way audio cross over network generally connected at the output of audio power amplifiers separates audio frequencies (20Hz to 20KHz) into three groups called Bass frequencies (low frequencies of audio range) Mid range frequencies and Treble frequencies (high frequencies of audio range). In this network, the first branch provides low frequency which is called woofer frequency (300Hz to 1200Hz), second branch provides mid frequency (1.2KHz to 3.2KHz) and the last branch provides high frequency called tweeter frequency (above 3.2KHz). The circuit is actually a combination of low pass filter, Band pass filter and High pass filter.

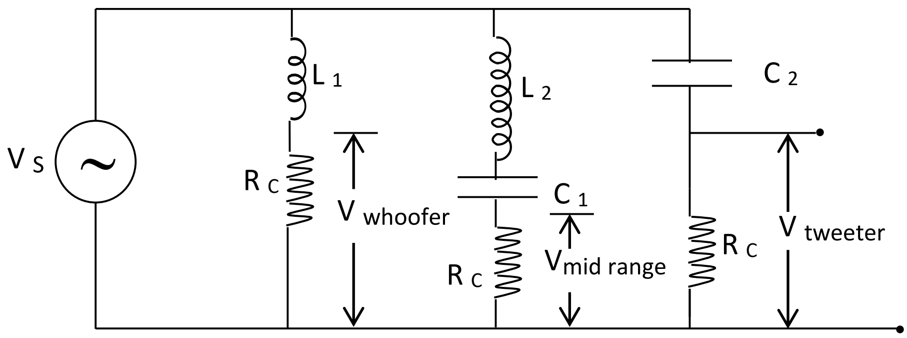
Procedure:

- Circuit connections are made stage by stage as shown in the diagram.
- Input voltage is set at 4V or 5V.
- First section (ie) the woofer section is constructed. By varying the frequency from 100Hz to 10KHz the output voltage is measured across the resistor.
- Second section (ie) the mid frequency section is constructed and repeat step 3
- Third section (ie) the tweeter section is constructed and repeat step 3
- A graph of output voltage versus frequency is plotted for each section.

Result:

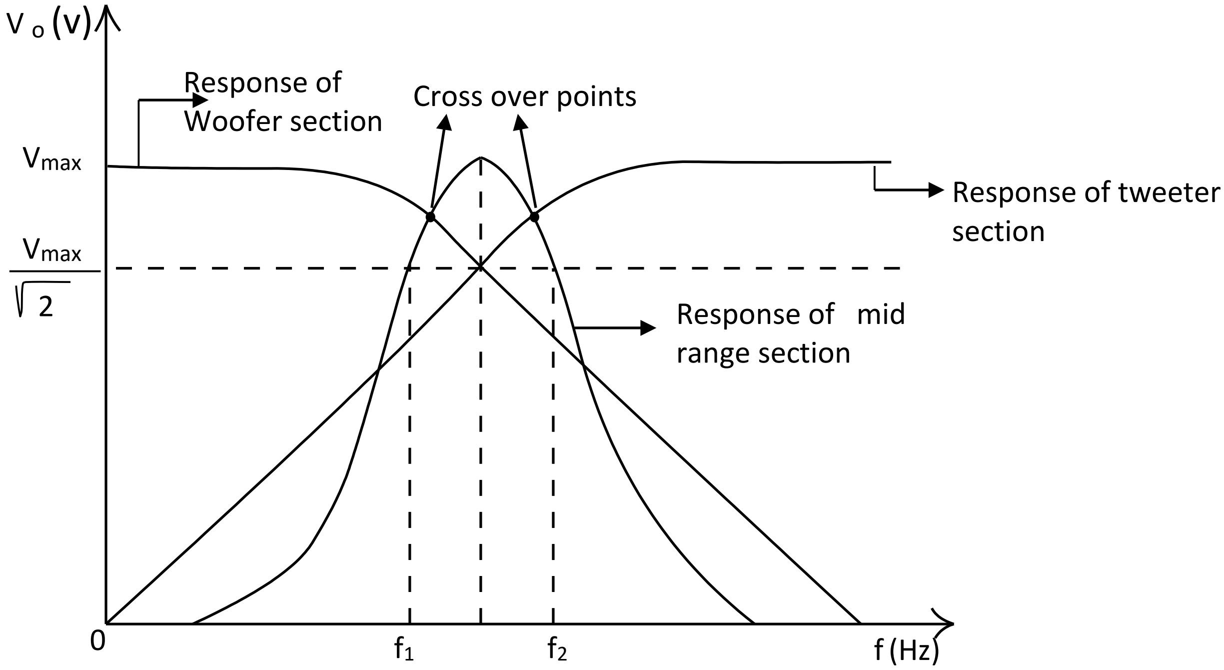
Audio cross over network is constructed and its operation is studied. The frequency response curve of woofer, midrange and tweeter networks is plotted.

CIRCUIT DIAGRAM



$R_c = 500\Omega$
$C_1 = 0.132\ \mu F$
$C_2 = 0.099\ \mu F$
$L_1 = 66.35mH$
$L_2 = 39.81mH$

FREQUENCY RESPONSE OF THREE WAY CROSSOVER NETWORK



DESIGN

Let $R_c = 500\Omega$;

Assume the tweeter section cut off frequency, $f_2 = 3.2KHz$ and woofer section cut off frequency as

$$f_1 = 1.2KHz$$

$$f_1 = \frac{R_c}{2\pi L_1}$$

$$L_1 = \frac{500}{2 \times 3.14 \times 1.2 \times 1000} = 66.35mH$$

Assume the mid frequency of midrange section as $f_m = 2.2KHz$;Bandwidth,BW=2 KHz

Quality factor of the midrange section, $Q = \frac{f_m}{BW} = 1.1$

$$Q = \frac{1}{2\pi f_m C_1 R_c} = \frac{2\pi f_m L_2}{R_c} \text{ -------- (1)}$$

$$C_1 = \frac{1}{2\pi\, f_m QR} = \frac{1}{2 \times 3.14 \times 2.2 \times 1000 \times 1.1 \times 500} = 0.132\mu F$$

$$f_2 = \frac{1}{2\pi R_c C_2} = 3.2KHz$$

$$C_2 = \frac{1}{2\pi R_c f_2} = \frac{1}{2 \times 3.14 \times 500 \times 3.2 \times 1000} = 0.099\mu F$$

From equation (1),

$$L_2 = \frac{Q\ R_c}{2\pi\ f_m} = \frac{1.1 \times 500}{2 \times 3.14 \times 2.2 \times 1000} = 39.81mH$$

OBSERVATIONS

Vin = 4 V

Frequency (Hz)	Woofer O/P(V)	Mid range O/P (V)	Tweeter O/P (V)
100			
200			
300			
400			
500			
600			
700			
800			
1k			
2k			
3k			
4k			
5k			
6k			
7k			
8k			
9k			
10k			
15k			
20k			
25k			
30k			
35 k			
40 k			
45 k			
50 k			

Experiment No:3

PULSE AMPLITUDE MODULATOR AND DEMODULATOR

AIM:

a) To construct a Pulse Amplitude Modulator and to sketch the output waveforms.

b) To construct a PAM demodulator and to study its performance.

COMPONENTS AND EQUIPMENT REQUIRED:

Function Generator (2), CRO, Transistor - BC107, Resistor - 10KΩ, 1KΩ, 680Ω, Capacitor – 0.01µF.

PRINCIPLE:

Pulse amplitude modulation is a kind of analog modulation where the carrier pulse is amplitude modulated at discrete time intervals corresponding to a continuous modulating signal i.e., the amplitude of the carrier pulse is varied with respect to the modulating signal at the instant of sampling.

The circuit is a simple emitter follower. In the absence of carrier signal, output follows the input. Base of the transistor is coupled with the modulating signal and carrier at the collector end. Amplitude of the carrier is kept high to bias the transistor into the cut off region. When the pulse signal is high, the transistor circuit behaves as an emitter follower and the output follows the input-modulating signal. When the clock signal is low, the transistor is in cut off and the output is zero.

A PAM de-modulator is a RC circuit which filters the high frequency carrier wave and at the output the signal is recovered back.

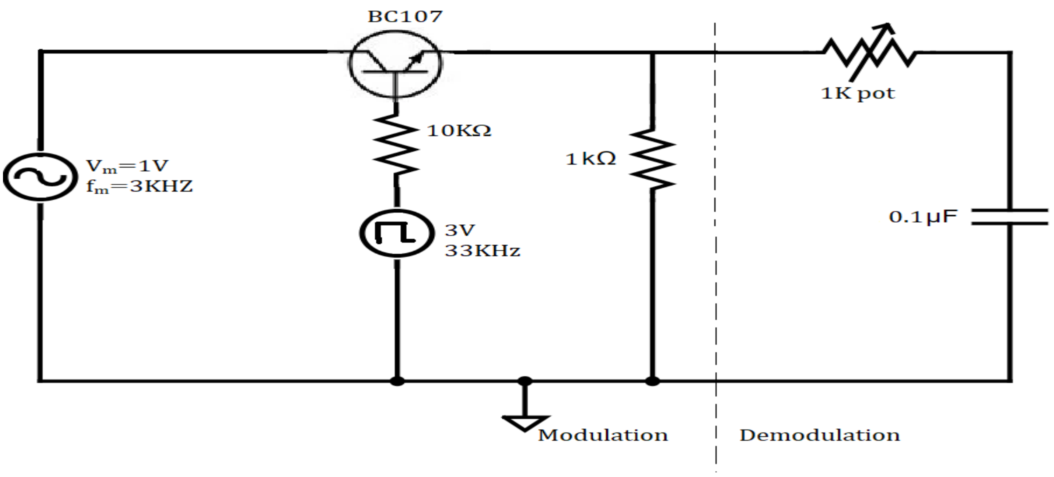
PROCEDURE:

Circuit connections are made as shown. The modulating voltage is adjusted to 1V, 3 KHz and the carrier voltage (square wave) is adjusted 3 V, 33 KHz frequency and individual waveforms are observed. The output waveforms are observed at the emitter. A demodulator circuit is connected at the output of the modulator. The demodulated waveform is observed. The input output waveforms are traced on a graph sheet.

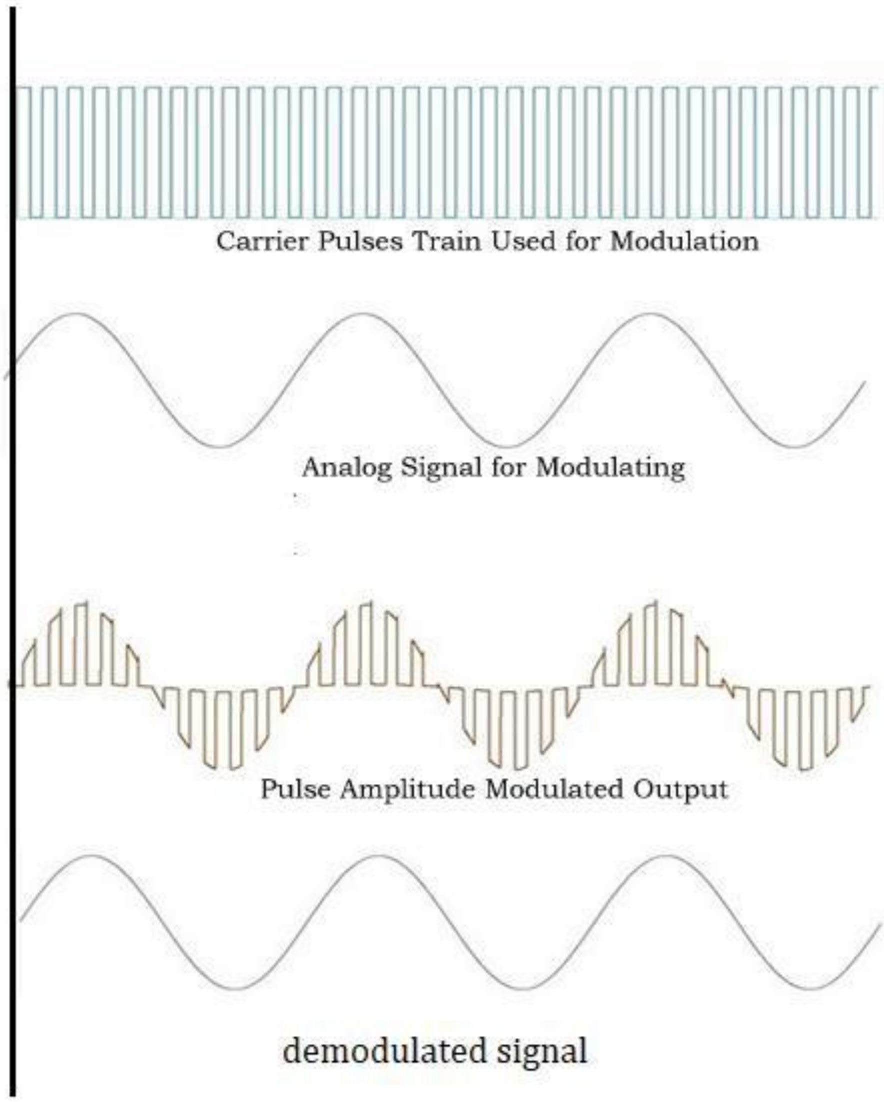
RESULT:

A Pulse Amplitude Modulator has been constructed using transistor and output waveforms are observed. A de-modulator circuit is connected at the output of the modulator and waveforms are studied

Circuit diagram:



Output waveform:



Experiment No:4

SIMULATION OF AMPLITUDE MODULATION AND DEMODULATION USING MATLAB

Aim:

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

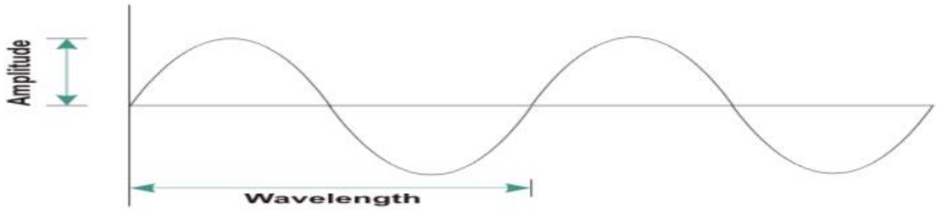
Theory:

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 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 envelope 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,

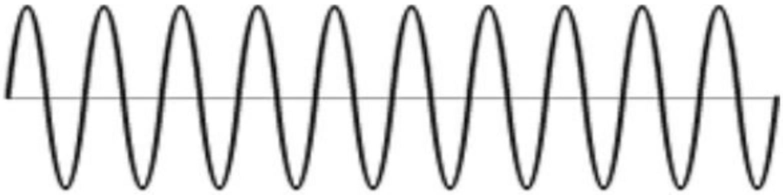
$$m = \frac{(E_{max} - E_{min})}{(E_{max} + E_{min})}$$

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

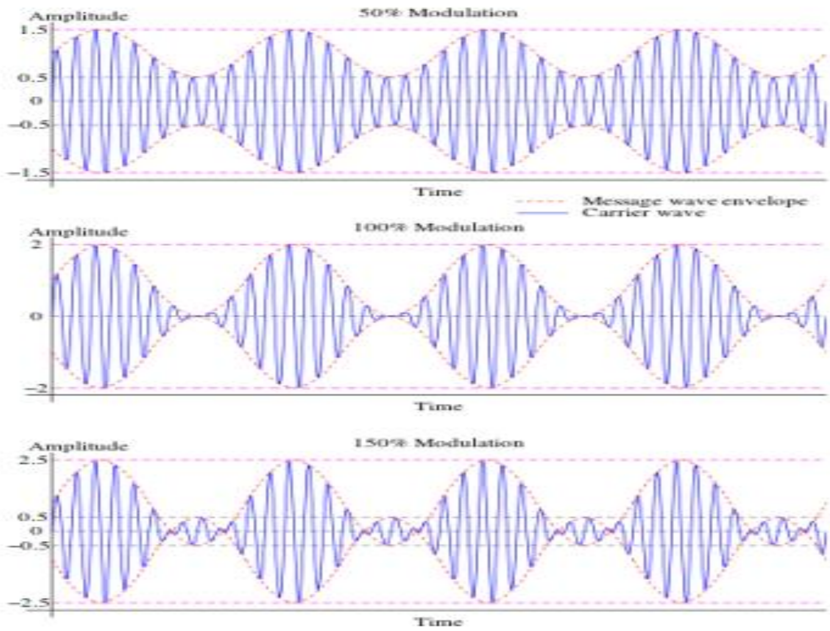
Expected Wave Forms:



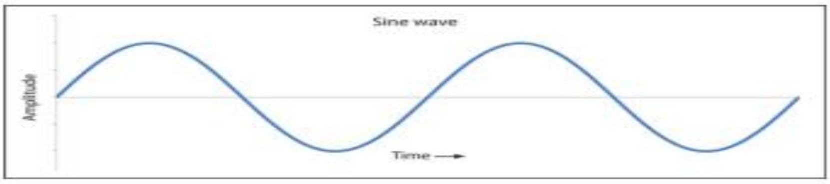
Message signal



Carrier signal



Types of Amplitude modulated waveforms



Demodulated signal

Tabular Column:

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

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' )
```

Result: Thus the depth of modulation is calculated using matlab simulation program

Experiment No:5

SIMULATION OF FREQUENCY MODULATION AND DEMODULATION USINGMATLAB

AIM:

1. To generate frequency modulated signal and determine the modulation index and bandwidth for various values of amplitude and frequency of modulating signal.

2. To demodulate a Frequency Modulated signal using FM detector.

Theory -:

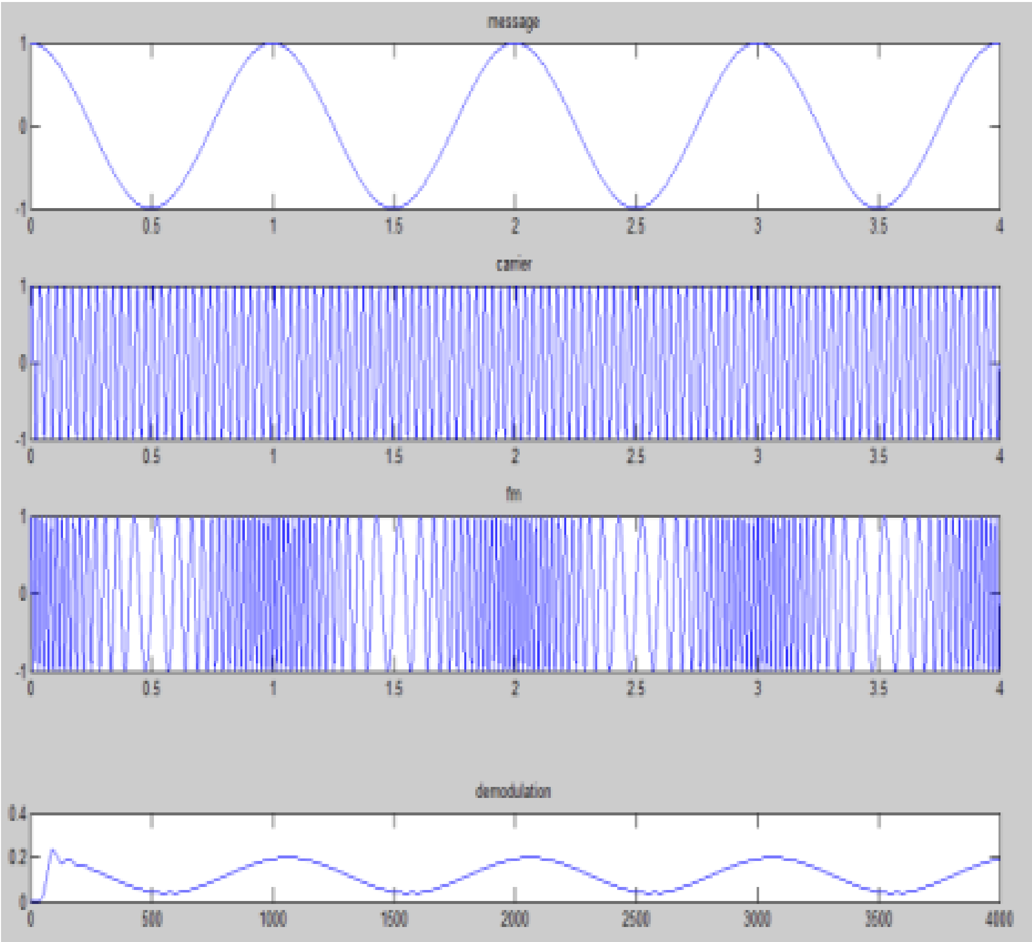
Frequency modulation is a system in which the frequency of the carrier is varied in accordance with the signal amplitude. Lets assume for the moment that the carrier of the transmitter is at its resting frequency (no modulation) of 100MHz and we apply a modulating signal. The amplitude of the modulating 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

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');
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	Modulating Frequency in (kHz)	Demodulated Signal voltage	Demodulated Signal frequency

Result-:

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

Experiment No:6

ASK GENERATION AND DETECTION

Aim: To construct and study the ASK Modulator and Demodulator circuits using a Bipolar junction transistor and IC741 respectively and to draw the waveforms.

Components: Power supply (0-15) V (2), AFO (2), CRO, IC741, Transistor SL100, Diode 0A79, Resistors220KΩ, 33KΩ, 3.3KΩ, 10KΩ, Capacitors-0.1µF

Principle:

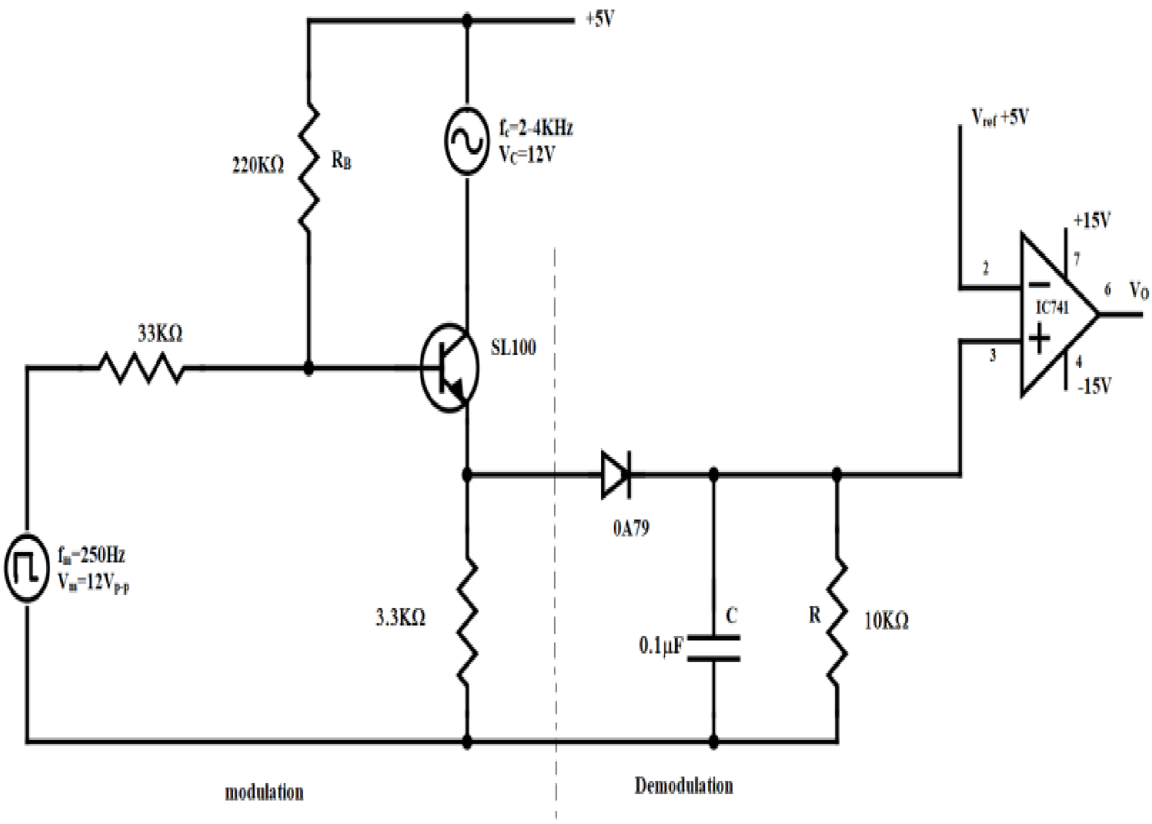
If the information signal is digital and if the amplitude of carrier is varied in proportional to the information signal, a digitally modulated signal called amplitude shift keying is produced ASK is also referred to as ON-OFF keying, because the carrier is either ON or OFF.
In the ASK modulator circuit, binary input is the modulating signal. The sine wave at the collector is the carrier signal. When the binary input is HIGH, the transistor is switched ON and the collector potential varies in accordance with the instantaneous voltage of carrier. As a result, the emitter current varies accordingly. The output waveform across the emitter hence contains a few cycles of carrier wave. When the binary input is LOW the transistor is driven into OFF state and the emitter current is zero. Hence the output is zero (ie) the carrier wave is switched OFF.

In the ASK demodulator circuit, the diode detector followed by bypass capacitor produces slow varying signal following the discharge path of capacitor with small ripples. The op-amp is used as non inverting comparator (ie) the reference voltage is applied at the inverting input. Whenever the rectified output voltage at the non inverting input crosses the reference voltage, the output of op amp toggles. As a result the output of demodulator exactly follows the binary input of modulator.

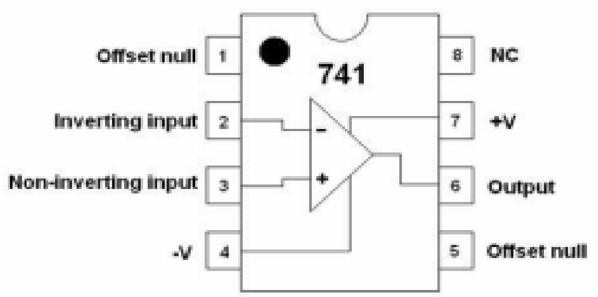
Procedure:

- Rig up the circuit as shown in fig.1
- A square wave of frequency 250Hz and amplitude around 12V(p-p) is applied at the input
- The sinusoidal carrier wave is set to around 12V(p-p) at 2.4KHz frequency.
- By adjusting the controls of CRO, binary input and ASK output waveforms are simultaneously displayed and drawn on graph sheet.
- ASK demodulator circuit is constructed as shown in fig 2.
- The ASK modulated output is applied at the input of demodulator circuit.
- The final demodulated output waveform is displayed on the CRO and is drawn on a graph sheet.

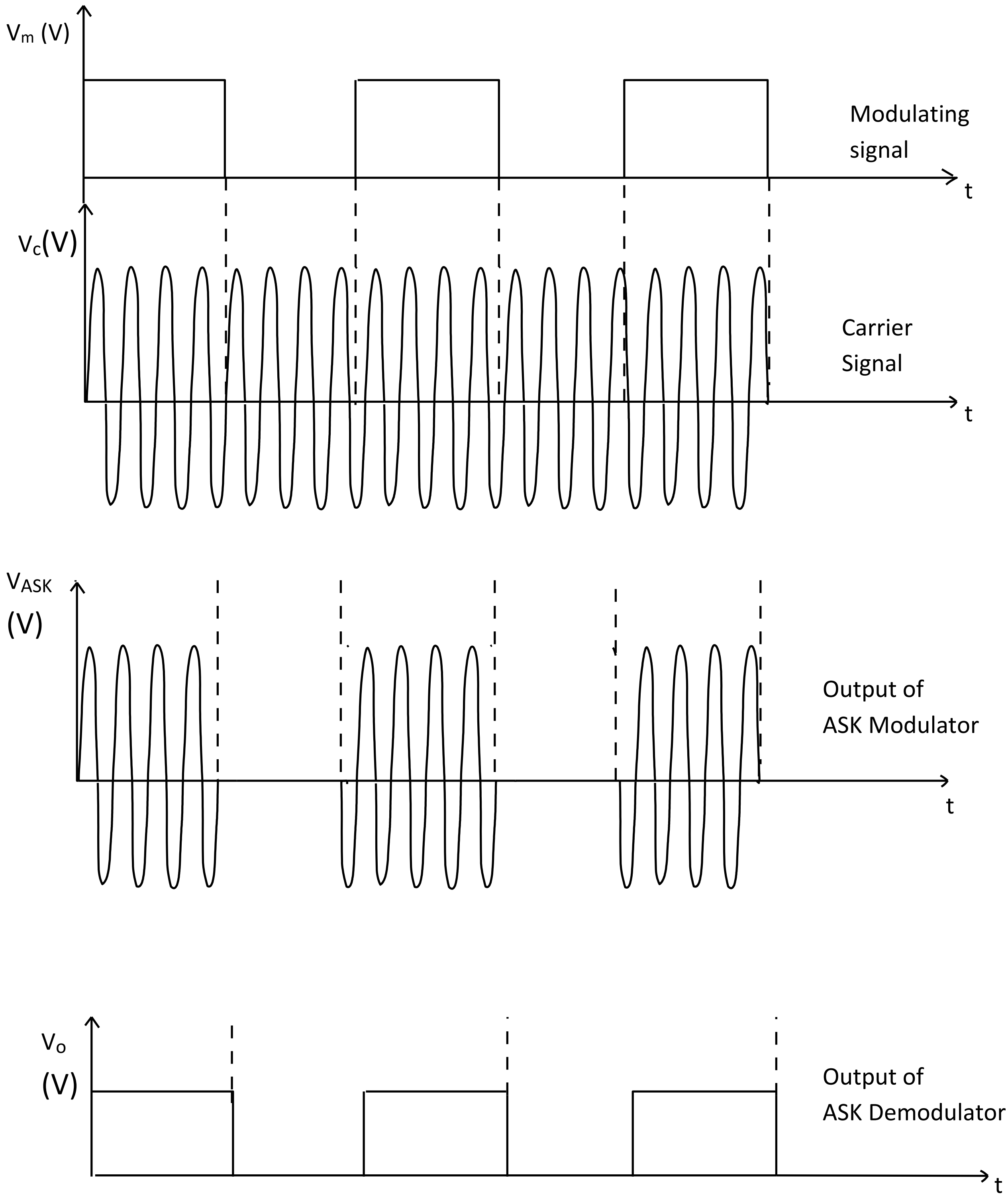
CIRCUIT DIAGRAM



PIN DIAGRAM OF IC 741



WAVE FORMS OF ASK MODULATOR AND DEMODULATOR



Result:

The ASK modulator and demodulator circuits are constructed and the waveforms are drawn on a graph sheet. It is found that the demodulated output is exactly similar to the binary input.

Experiment No – 7

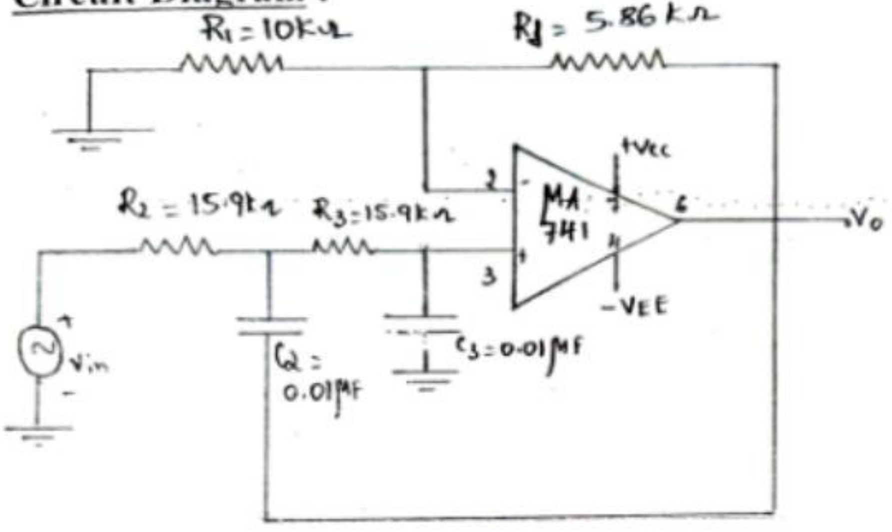
SECOND ORDER ACTIVE FILTER

A) SECOND ORDER LOW PASS FILTER

AIM: To design an active low pass Butterworth second order filter and to draw its frequency response.

Apparatus Required: CRO, Signal generator, Resistances, capacitors, IC μA741, probes and connecting wires.

Circuit Diagram :



Design:

Let f_H=1 KHz.

Pass Band gain A_f=$(1+R_f/R_1)$=1.56

Choose R_F=0.586 R_1

Let R_1=10 KHz.

R_F = 5.86KΩ.

Let R= R_1= R_2 = R_3

R=1/2π f_HC Where C= C_2 = C_3

and C=0.01μF

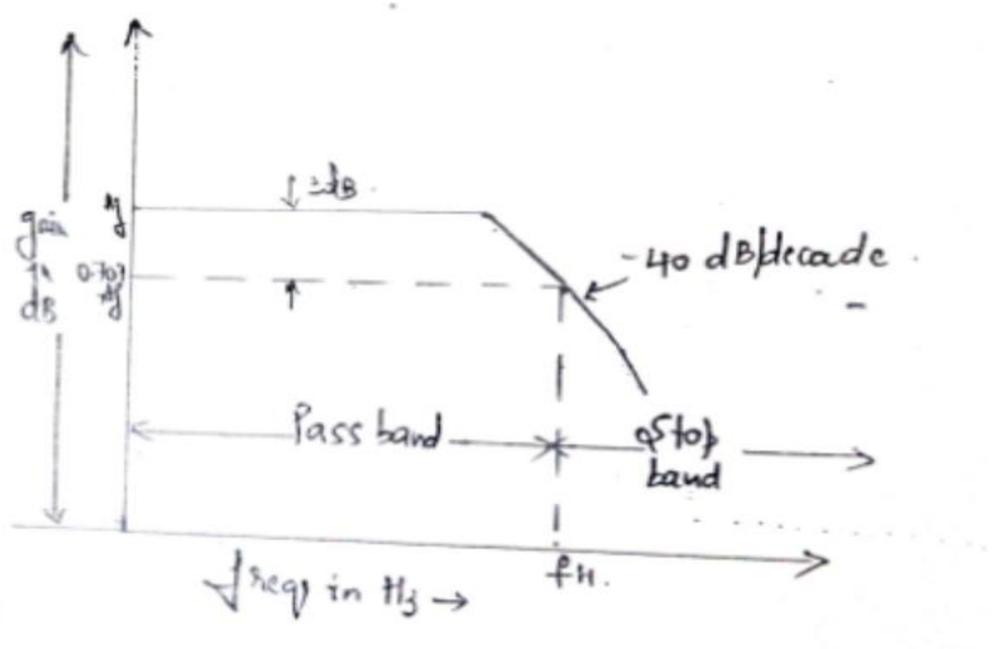
R= R_2 = R_3=15.986KΩ.

Input Voltage, $V_{in} = 1\ V$

Tabular Column for Low Pass Filter

Frequency (Hz)	Output Voltage(V_o)	Gain Magnitude$\frac{V_o}{V_{in}}$	Gain in db
100			
200			
300			
400			
500			
600			
700			
800			
1k			
2k			
3k			
4k			
5k			
6k			
7k			
8k			
9k			
10k			

Model Graph:

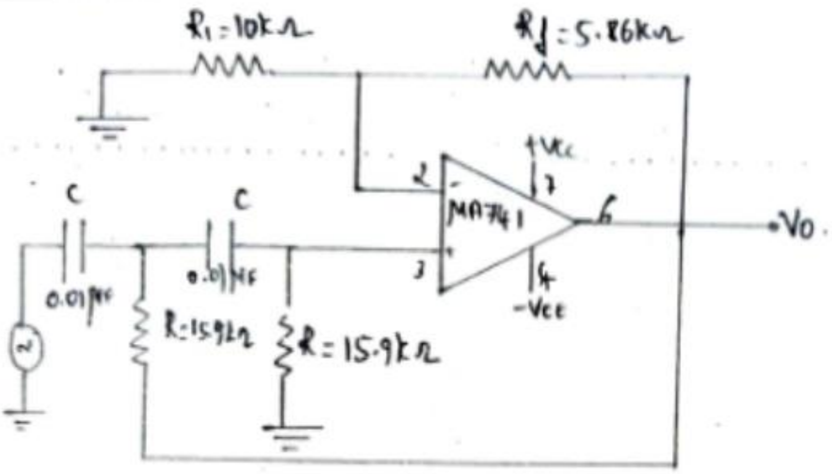


B) SECOND ORDER HIGH PASS FILTER

AIM: To design an active high pass Butterworth second order filter and to draw its frequency response.

Apparatus Required: CRO, Signal generator, Resistances, capacitors, IC µA741, probes and connecting wires.

Circuit Diagram:



Design:

Let f_L=1KHz.

Pass Band gain A_f=$(1+R_f/R_1)$=1.56

Choose R_f=0.586 R_1

Let R_1=10KHz.

R_f = 5.86KΩ.

Let R= R_1= R_2 = R_3

R=1/2π f_LC Where C= C_2 = C_3

and C=0.01µF

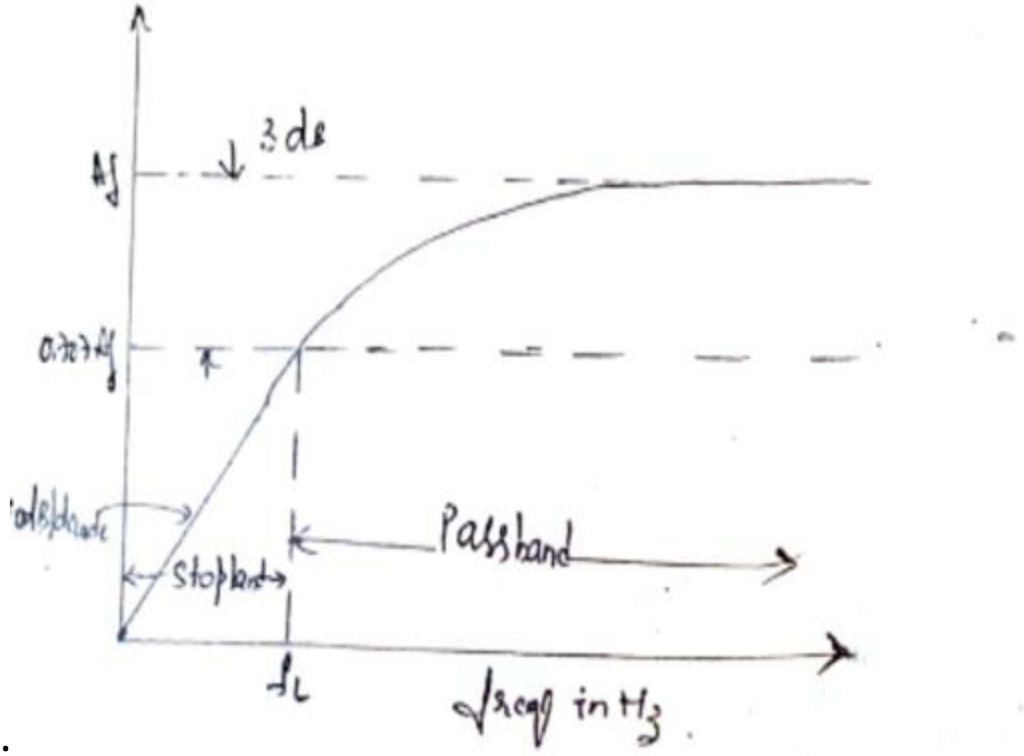
R= R_2 = R_3=15.986KΩ.

Tabular Column for High Pass Filter

Input Voltage, $V_{in} = 1\ V$

Frequency (Hz)	Output Voltage(V_o)	Gain Magnitude$\frac{V_o}{V_{in}}$	Gain in db
100			
200			
300			
400			
500			
600			
700			
800			
1k			
2k			
3k			
4k			
5k			
6k			
7k			
8k			
9k			
10k			

Model Graph:



Theory-:

The practical response of Second Order Low Pass Butterworth Filter must be very close to an ideal one. In case of low pass filter, it is always desirable that the gain rolls off very fast after the cut off frequency, in the stop band. In case of first order filter, it rolls off at a rate of 20 dB/decade. In case of second order filter, the gain rolls off at a rate of 40 dB/decade. Thus, the slope of the frequency response after $f = f_H$ is — 40 dB/decade, for a second order low pass filter.

The second order high pass Butterworth filters produces a gain roll off at the rate of + 40 dB/decade in the stop band. This filter also can be realized by interchanging the positions of resistors and capacitors in a second order low pass Butterworth filters.

Results-:

Designed and obtained the frequency response of second order Low pass and high pass filter.

High pass filter-:

Cutoff frequency =

Bandwidth=

Low pass filter-:

Cutoff frequency =

Bandwidth

Experiment No – 8

BAND ELIMINATION FILTER

Aim: To design an active band reject filter circuit and observe its frequency response.

Components Required:
OP –AMP 741, Resistors,Capacitors,DC power supply (for biasing),Signal Generator,Trainer Board,Oscilloscope,DMM,Wires

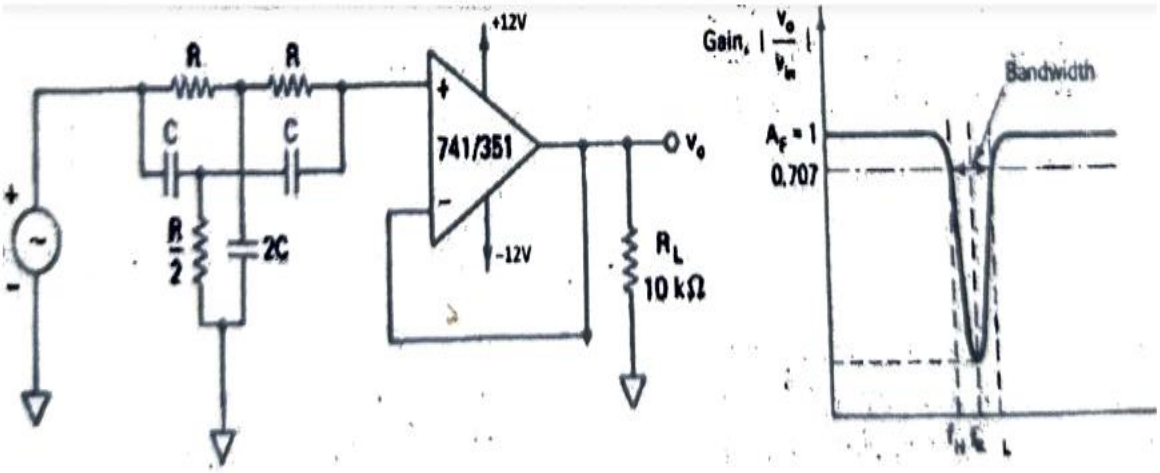
Principle:

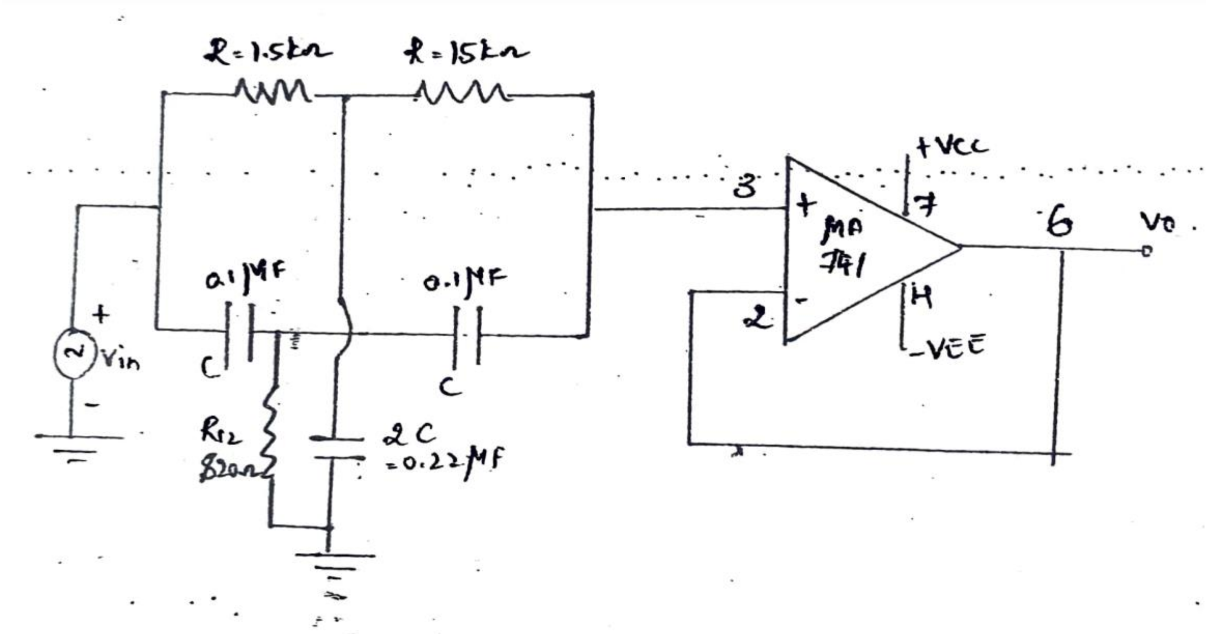
This kind of filter passes all frequencies above and below a particular range set by the component values. Stopband filters can be constructed using a low-pass and a high pass filter. However, rather than the cascaded configuration used for the pass-band filter, a parallel arrangement is required. A low-frequencyf1 can pass through the low-pass filter, and a higher-frequency f2 can use the parallel path. However, frequency such as fo in the reject-band is higher than the lowpass critical frequency and lower than the high-pass critical frequency and is therefore prevented from contributing to the levels of Vo above0.707Vmax.

Procedure -:

1. Rig up the circuit as shown in the figure.
2. Give a 1 V sinusoidal signal as the input from the signal generator
3. Vary the frequency from 100 Hz to 2 KHz
4. Observe the change in amplitude of output sine wave
5. Calculate the gain in dB

CIRCUIT DIAGRAM





Design-:

$$f_r = \frac{1}{2\pi RC} = 1 \text{ KHz};$$

$$R = 1.59\text{K}\Omega(\text{use } 1.5\text{K}\Omega \text{ std})$$

$$\frac{R}{2} = 795\Omega(\text{use } 820\Omega \text{ std})$$

$$\text{C} = 0.1\mu\text{F}; 2\text{C} = .2\mu\text{F}(\text{use } 0.22\mu\text{F std})$$

Tabular Column-:

Input Voltage,$V_i = 1\ V$

Frequency,Hz	Output Voltage,V_o	Gain=$\frac{V_o}{V_i}$	Gain in dB

Result-:

Designed and constructed band elimination filter and observed the frequency response.The following results were obtained-:

1. Rejecting frequency,f_r=--------------------
2. Lower cut off frequency,f_l=---------------
3. Higher cut off frequency,f_h=--------------
4. Bandwidth(in Hz)=--------------

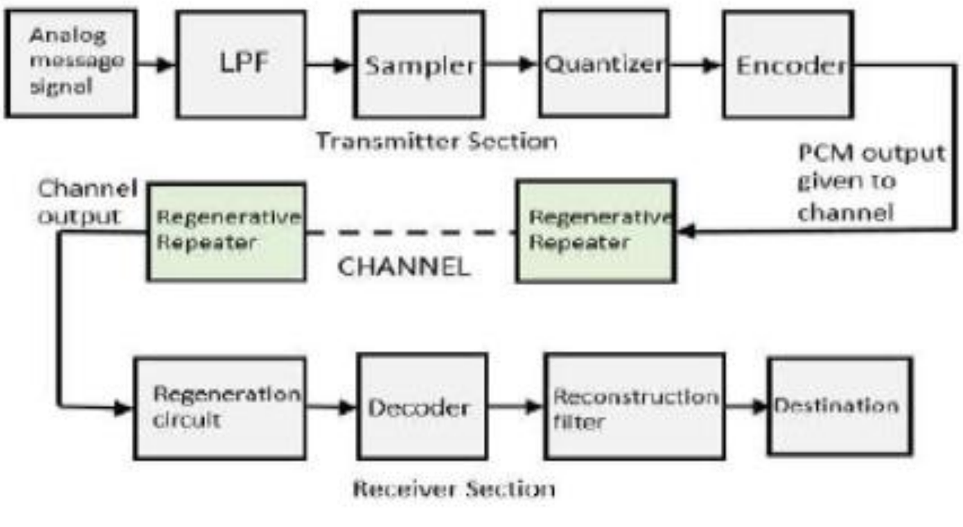
Experiment No: 9

SIMULATION OF PULSE CODE MODULATION AND DEMODULATION SYSTEM USING MATLAB

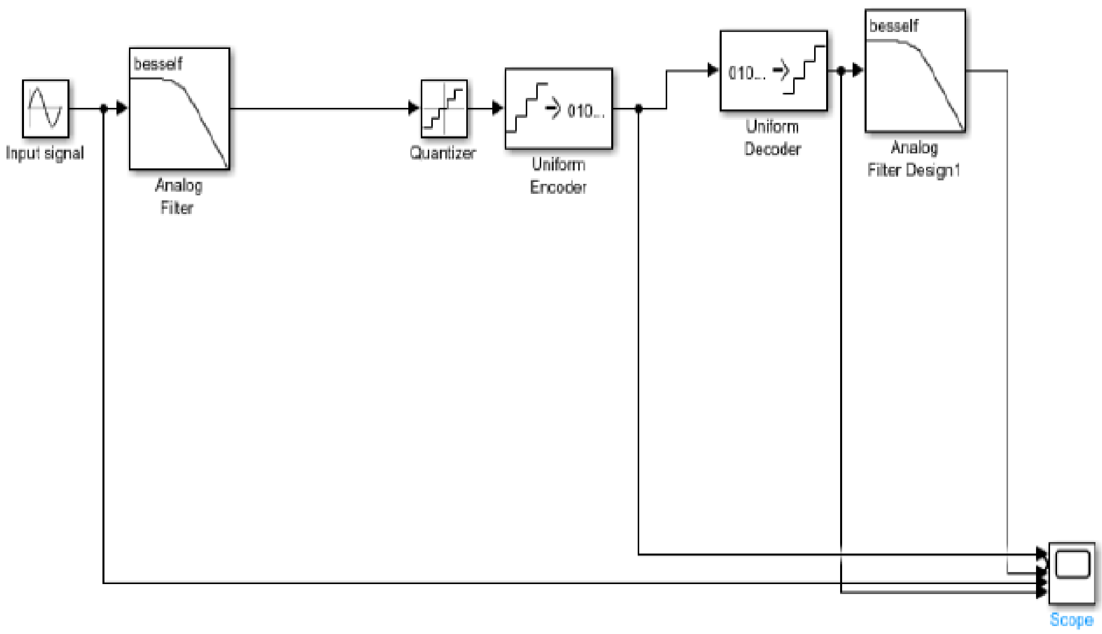
Aim: Simulate the Pulse code modulation and demodulation system and display the waveforms.

Theory: Pulse-code modulation (PCM) is a method used to digitally represent sampled analog signals. It is the standard form of digital audio in computers, compact discs, digital telephony and other digital audio applications. In a PCM stream, the amplitude of the analog signal is sampled regularly at uniform intervals, and each sample is quantized to the nearest value within a range of digital steps.

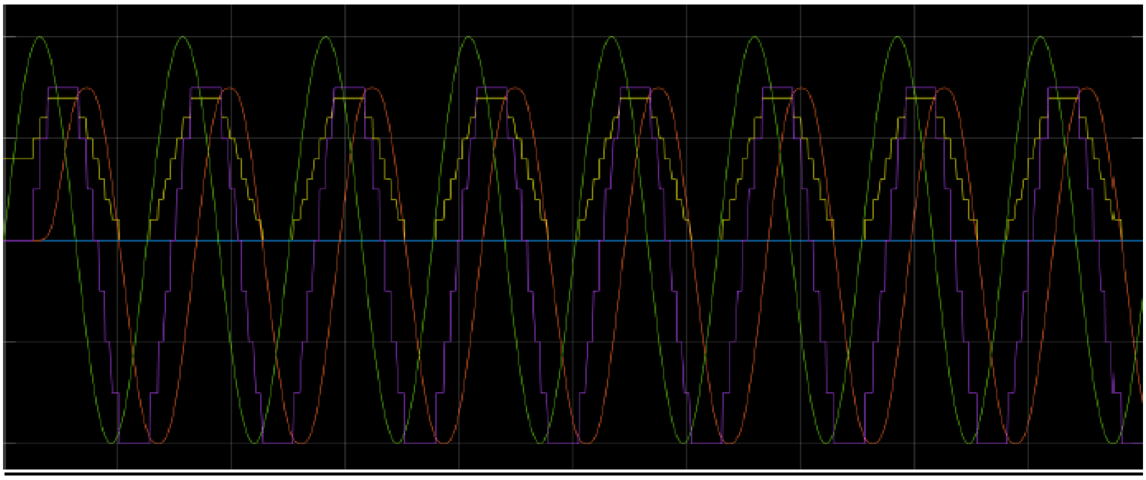
To recover the original signal from the sampled data, a "demodulator" can apply the procedure of modulation in reverse. After each sampling period, the demodulator reads the next value and shifts the output signal to the new value. As a result of these transitions, the signal has a significant amount of high-frequency energy caused by aliasing. To remove these undesirable frequencies and leave the original signal, the demodulator passes the signal through analog filters that suppress energy outside the expected frequency range.



Simulink Diagram



Waveforms-:



Result-:PCM modulation and demodulation simulated using MATLAB/Simulink

Experiment No:10

DPSK MODULATION AND DEMODULATION

AIM:

To perform the simulation of DPSK modulation and demodulation using MATLAB.

EQUIPMENTS REQUIRED:

1. Operating system:Windows XP
2. Software:MATLAB

THEORY:

Differential phase shift keying (DPSK), a common form of phase modulation conveys data by changing the phase of carrier wave. In DPSK, there is no absolute carrier phase reference, instead transmitted signal itself used as phase reference. For example, in differentially encoded BPSK a binary '1' may be transmitted by adding 180° to the current phase and a binary '0' by adding 0° to the current phase. For a signal that has been differentially encoded, there is an obvious alternative method of demodulation. Instead of demodulating as usual and ignoring carrier-phase ambiguity, the phase between two successive received symbols is compared and used to determine what the data must have been. When differential encoding is used in this manner, the scheme is known as differential phase-shift keying (DPSK).

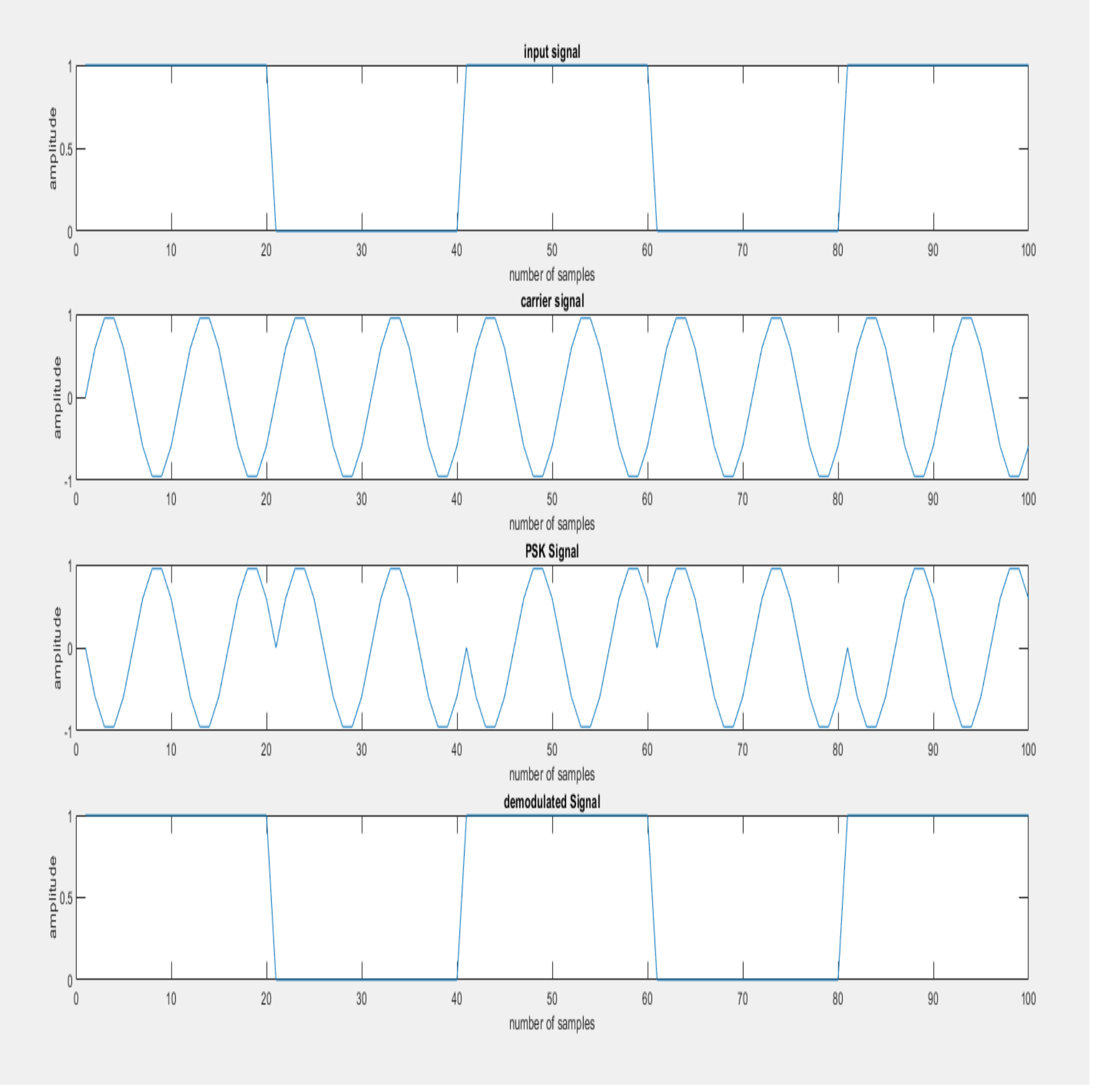
ALGORITHM:

1. Define the number of input samples.
2. Generate the binary input sequence.
3. Generate the carrier signal with frequency f1
4. Generate DPSK signal.
5. Reconstruct the original message signal
6. Plot the message, carrier, modulated and demodulated signal.

MATLAB CODE-:

```
clc;
clear all;
close all;
n=100;
x=[ones(1,20) zeros(1,20) ones(1,20) zeros(1,20) ones(1,20)];
subplot(4,1,1);
plot(x);
title('input signal');
xlabel('number of samples');
ylabel('amplitude');
f=1*10^6;
fs=10*10^6;
for i=0:n-1
    d(i+1)=sin(2*pi*(f/fs)*i);
end
subplot(4,1,2);
plot(d);
title('carrier signal');
xlabel('number of samples');
ylabel('amplitude');
for i=0:n-1
if(x(i+1)==0)
x(i+1)=sin(2*pi*(f/fs)*i);
else
x(i+1)=sin(2*pi*(f/fs)*i+pi);
end
end
subplot(4,1,3);
plot(x);
title('PSK Signal');
xlabel('number of samples');
ylabel('amplitude');
for i=0:n-1
if(x(i+1)==sin(2*pi*(f/fs)*i))
x(i+1)=0;
else
x(i+1)=1;
end
end
subplot(4,1,4);
plot(x);
title('demodulated Signal');
xlabel('number of samples');
ylabel('amplitude');
```

Waveform-:



Result-:

DPSK transmitter and receiver waveforms are simulated in MATLAB