MALLA REDDY ENGINEERING COLLEGE

(AUTONOMOUS)

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DEPARTMENT OF ECTRONICS AND COMMUNICATION ENGINEERING

LAB MANUAL

ANALOG COMMUNICATIONS LAB

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Principal

ANALOG COMMUNICATIONS LAB

LIST OF EXPERIMENTS:

1. Amplitude modulation and demodulation.
2. DSB-SC Modulator & Detector
3. SSB-SC Modulator & Detector (Phase Shift Method)
4. Study of spectrum analyser and analysis of AM and FM Signals
5. Pre-emphasis & de-emphasis.
6. Time Division Multiplexing & De multiplexing
7. Verification of Sampling Theorem
8. Pulse Amplitude Modulation and Demodulation
9. Pulse Width Modulation & Demodulation
10. Pulse Position Modulation & Demodulation
11. AGC Characteristics
12. Radio Receiver

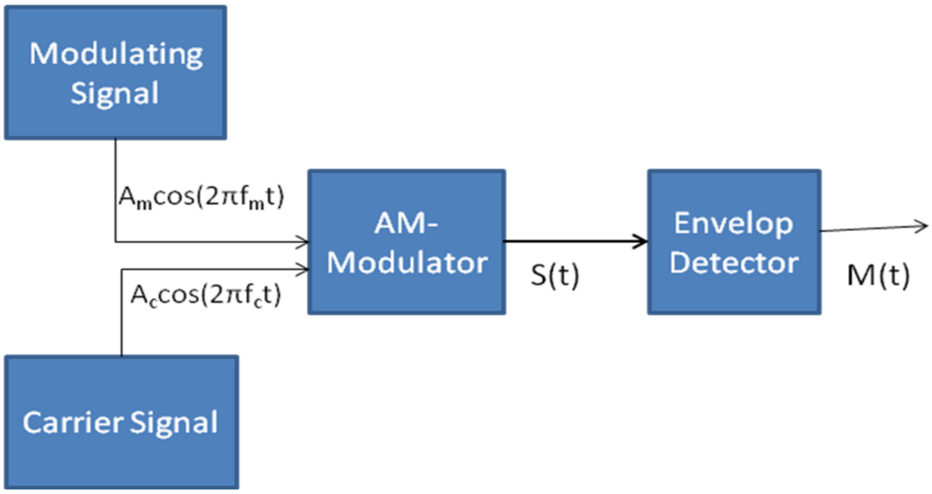
1. AMPLITUDE MODULATION & DEMODULATION

AIM: To calculate the depth of modulated wave by changing the amplitude of the modulating signal and demodulated the same and simulate using MATLAB.

APPARATUS:

1 AM Trainer Kit
2. CRO
3. Connecting wires
4. CRO Probes

Block Diagram:



THEORY:

Amplitude modulation is defined as a process in which the amplitude of the carrier wave is varied about a mean value linearly with the base band signal

It ensure that the function 1+ka m(t) is always positive .when the amplitude sensitivity Ka of the modulator is large enough to make Ka m(t)>1 for any t, the carrier wave becomes over modulated, resulting in carrier phase reversals whenever the factor 1+Kam(t) crosses zero the modulated wave then exhibits envelope distortion.
The absolute maximum value of K am (t) multiplied by 100 is referred to as the percentage modulation.

% modulation = (Vmax- Vmin/ Vmax+Vmin) X100.

PROCEDURE:

1. Connect the circuit diagram as shown in the circuit diagram
2. Apply the 28 KHz carrier signal and amplitude of 1.5V p-p to the input AM modulation to 10k pot.
3. Apply the 800Hz of modulating signal to the AM modulation at 100k pot
4. Apply power supply of 12V as shown in circuit diagram
5. Observe the amplitude modulation wav synchronization with the modulating signal on a dual trace CRO
6. Adjust the 10K linear pot for carrier suppression and 100k pot for proper modulation
7. Now by varying the amplitude of modulating signal the depth of the modulation varies
8. Calculate the maximum and minimum points of modulated wave under CRO and calculate the depth of modulation

   $m = V_{max} - V_{min} / V_{max} + V_{min}$
9. During the modulation give this AM o/p to the point of demodulation circuit.

Observations

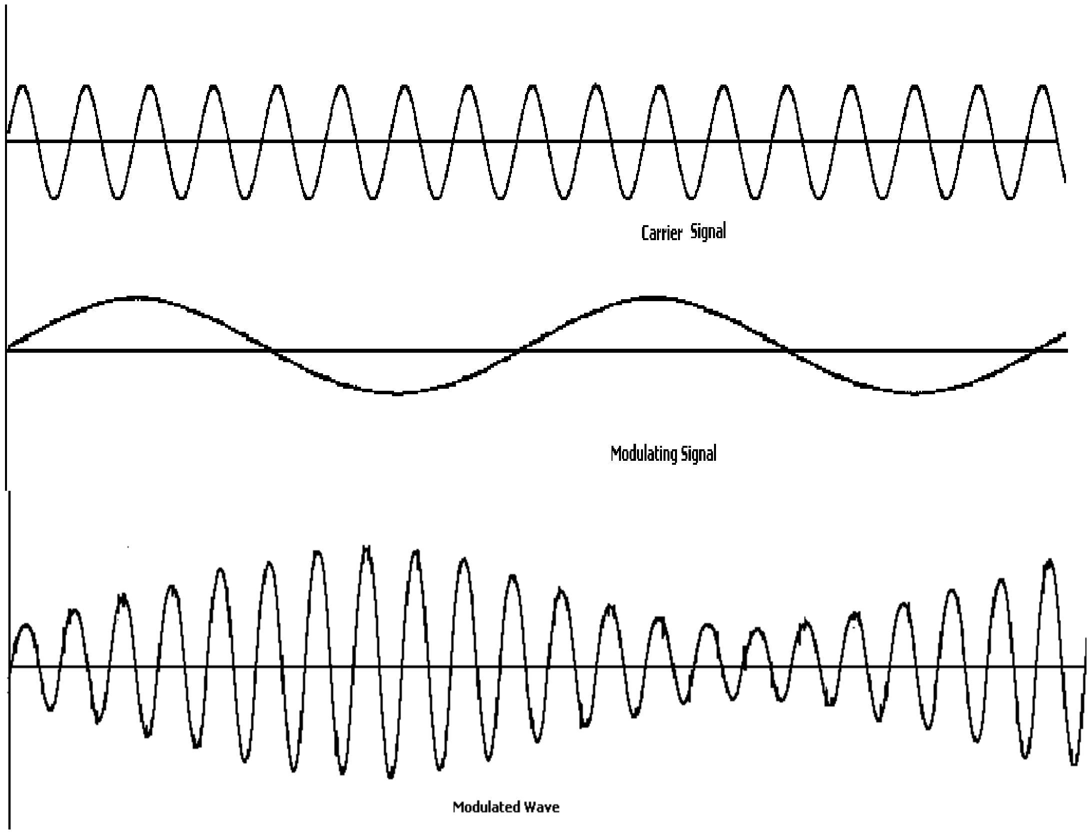
Table 1: **f_m= 1KHz, f_c=11KHz, A_c=15 V p-p.**

S.No.	V_m(Volts)	E_{max}(volts)	E_{min} (Volts)	m	%m (m x100)

Table 2: **A_m= 4 Vp-p f_c =11KHz, A_c=15 V p-p.**

S.No.	f_m(KHz)	E_{max}(volts)	E_{min}(Volts)	m	%m (m x100)

EXPECTED WAVE FORMS:

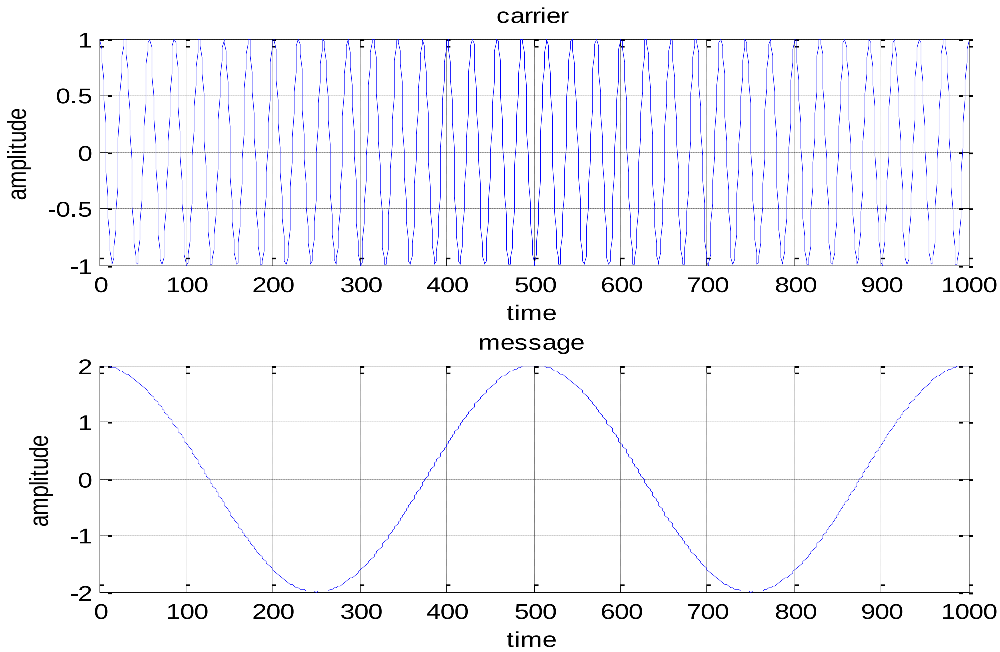


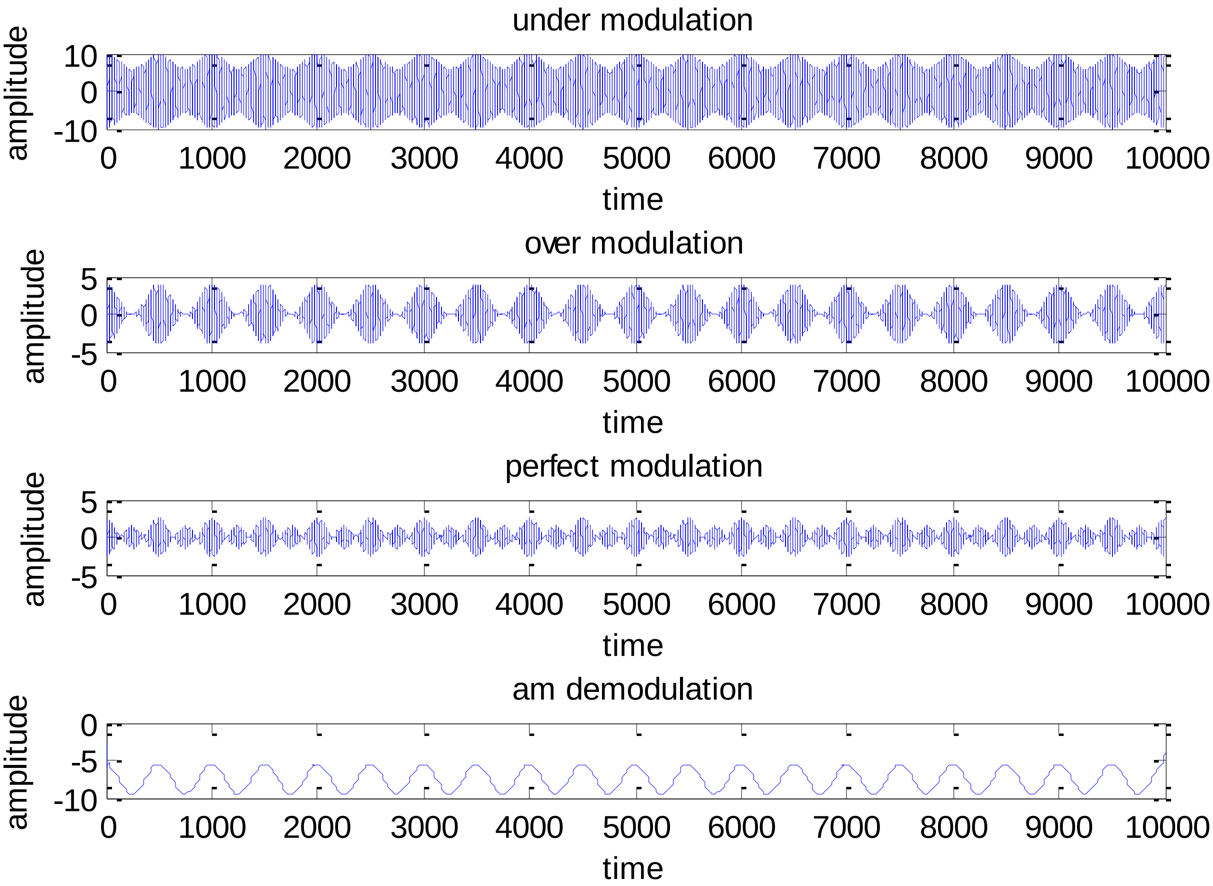
MATLAB CODE:

```
close all;
clear all;
clc;
fs=100e3
t=0:1/fs:.1-1/fs;
am=2;
fm=200;
m=am.*cos(2*pi*fm*t);
fc=3.5e3
ac=8;
c=ac.*cos(2*pi*fc*t);
figure;
subplot(2,1,1);
plot(c);
grid
title('carrier');
xlabel('time');
ylabel('amplitude');
subplot(2,1,2);
plot(m);
title('message');
xlabel('time');
ylabel('amplitude');
figure;
s=ammod(m,fc,fs,0,ac);
```

```
subplot(4,1,1);
plot(s);
title('under modulation');
xlabel('time');
ylabel('amplitude');
ac=2
s=ammod(m,fc,fs,0,ac);
subplot(4,1,2);
plot(s);
title('over modulation');
xlabel('time');
ylabel('amplitude');
ac=0.5
s=ammod(m,fc,fs,0,ac);
subplot(4,1,3);
plot(s);
title('perfect modulation');
xlabel('time');
ylabel('amplitude');
ac=8
z=amdemod(s,fc,fs,0,ac)
subplot(4,1,4);
plot(z);
title('am demodulation');
xlabel('time');
ylabel('amplitude');
```

WAVEFORMS:





RESULT:

AM signal is generated and original signal is demodulated from AM signal. Depth of Modulation is calculated for various amplitude levels of modulating signal and simulated using MATLAB.

VIVA VOICE:

1. Define AM and draw its spectrum?

2. Draw the phase’s representation of an amplitude modulated wave?

3. Give the significance of modulation index?

4. What is the different degree of modulation?

5. What are the limitations of square law modulator?

6. Compare linear and nonlinear modulators?

7. Compare base modulation and emitter modulation?

8 .Explain how AM wave is detected?

10. Define detection process?

11. What are the different types of distortions that occur in an envelope detector?

2. DSB-SC MODULATOR & DETECTOR

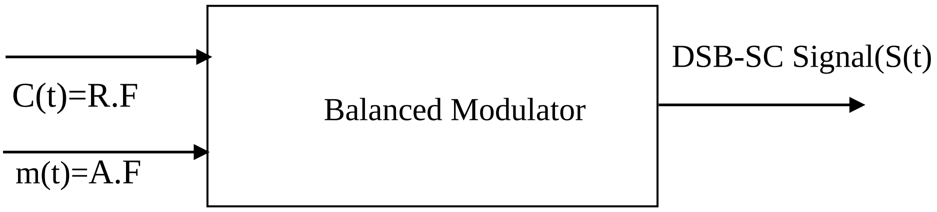
AIM: 1.To verify the frequency doubling using balance modulator

2. To perform DSB-SC modulation using balance modulator
3. Simulate and study using MATLAB.

APPARATUS:

1. AM Trainer Kit
2.CRO2
3.Connecting wires

CIRCUIT:



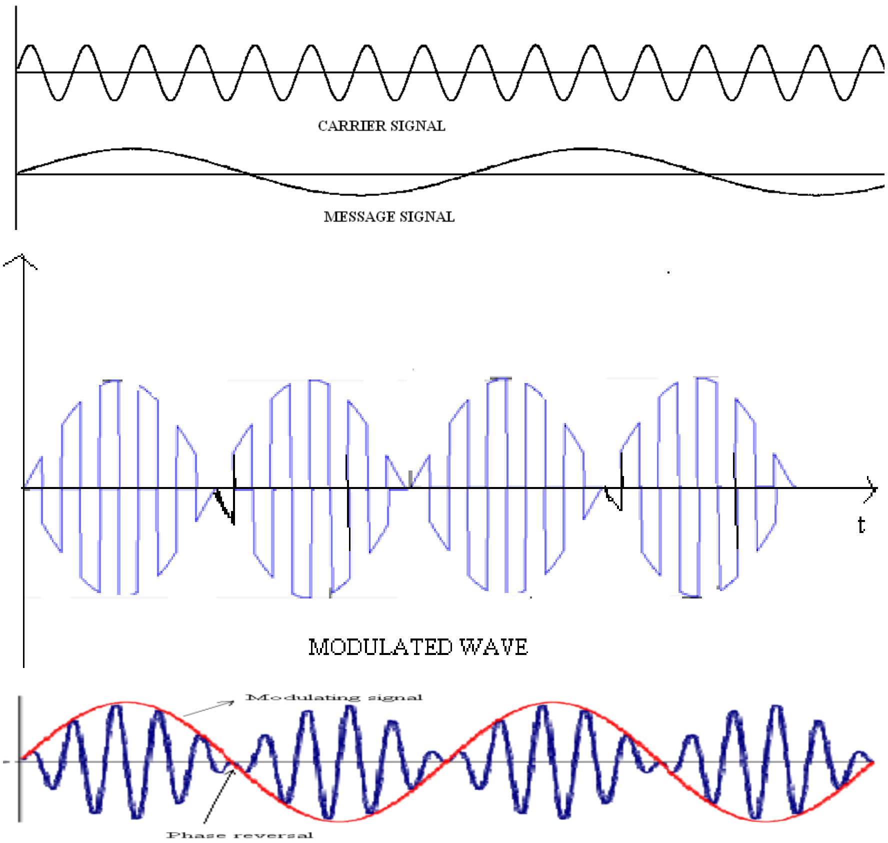
THEORY:

In Balanced modulator, two non-linear devices are connectd in the balanced modulator, so as to suppress the carrier. The Balanced modulator consists of summing device and two matched non-linear elements. If x(t) is band limited to fx and if fc >2fx,then the band pass filter output will be the desired product signal. In simple non-linear circuits the undesired non-linear terms are eliminated by a band pass filter. But in balanced modulator the undesired non-linear terms are automatically balanced out and at the output we get only the desired terms.

PROCEDURE:

1. Switch on the trainer kit and measure the output voltage of power supply
2. Apply the 4 KHz carrier signal and 100Hz of modulating signal observe the o/p of CRO.
3. Adjust the carrier i/p frequency more than that of modulating signal frequency
4. Observe the DSB – SC modulation and phase reversion on CRO
5. Draw the wave form of DSB-SC signal with phase reversal

EXPECTED WAVEFORMS:

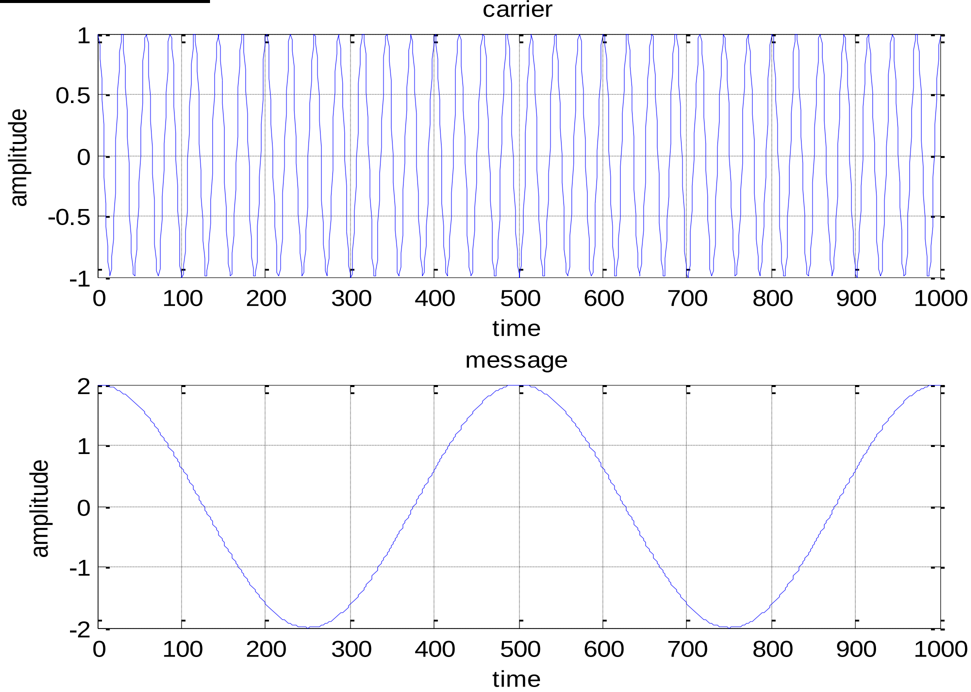


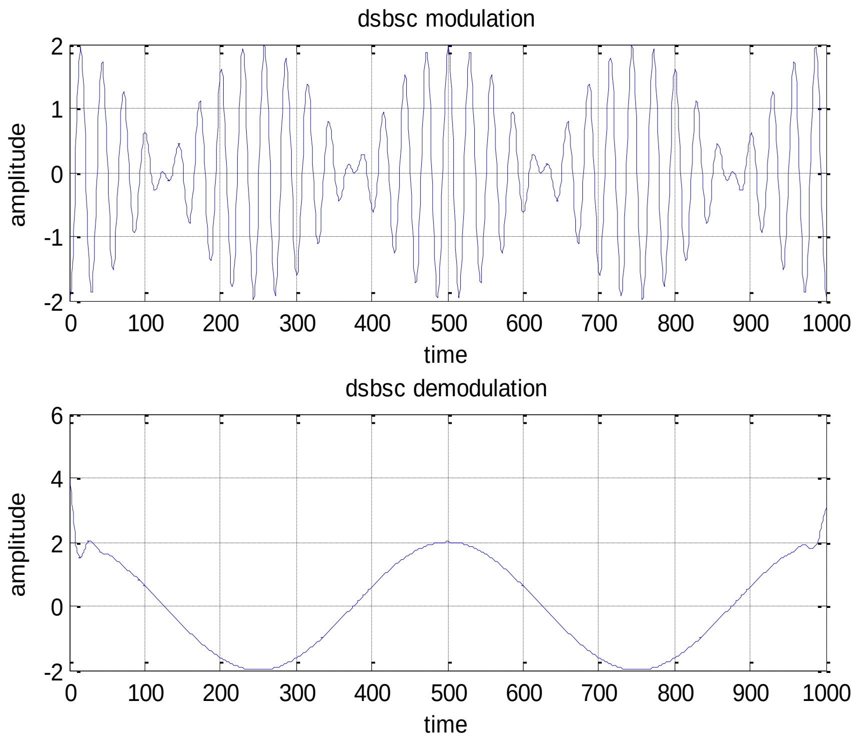
MATLAB CODE:

```
clc
close all
clc
fs=100e3
t=0:1/fs:.01-1/fs
am=2
fm=200
m=am.*cos(2*pi*fm*t)
fc=3.5e3
ac=1
c=ac.*cos(2*pi*fc*t)
figure;
```

```
subplot(2,1,1)
plot(c)
grid
title('carrier');
xlabel('time');
ylabel('amplitude');
subplot(2,1,2)
plot(m)
grid
title('message');
xlabel('time');
ylabel('amplitude');
s=ammod(m,fc,fs,3.14,0)
figure;
subplot(2,1,1)
plot(s)
grid
title('dsbsc modulation');
xlabel('time');
ylabel('amplitude');
z=amdemod(s,fc,fs,3.14,0)
subplot(2,1,2)
plot(z)
grid
title('dsbsc demodulation');
xlabel('time');
ylabel('amplitude');
```

WAVEFORMS:





RESULT:

The frequency doubling and DSB-SC modulation are observed and verified and simulated using MATLAB.

VIVA VOICE:

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

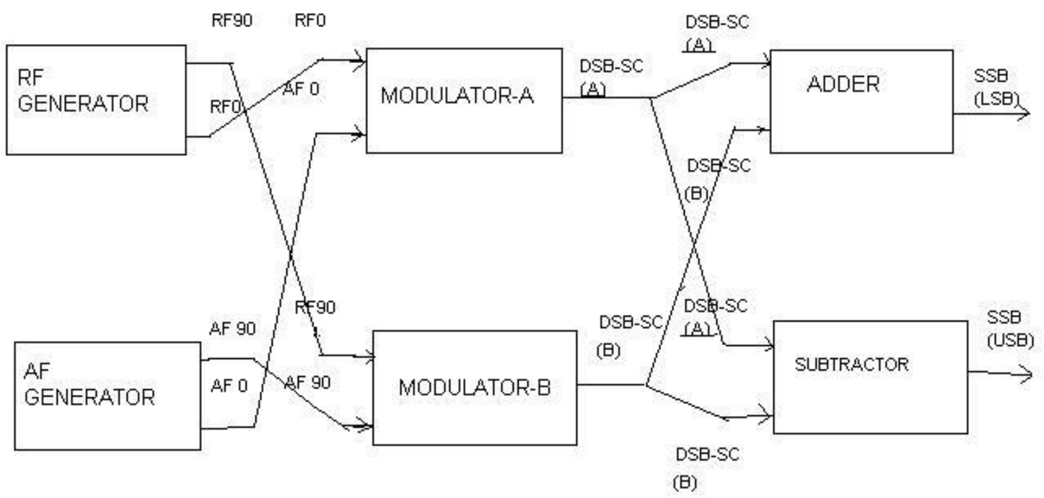
3. SSB-SC MODULATOR & DETECTOR

AIM: To generate SSB – SC signal and detect the original signal and simulate with matlab.

APPARATUS:

1. Trainer Kit
2. CRO

CIRCUIT:



SSB-SC DETECTION

THEORY:

The shift method makes use of two balanced modulators and two phase shift networks. One of the modulators receives the carrier signal shifted by 90 degrees and the modulating signal with 0 degree phase shift where as receives modulating signal shifted by 90 degrees and the carrier signal with 0 degrees phase shift.

Both modulators produce an o/p consisting only of side bands. Both USB's leads the i/p carrier voltage by 90 degrees. One of the lower side band leads the reference voltage by 90 degrees and the other lags it by 90 degrees. The two lower side bands are thus out of phase and when combined in the adder they cancel each other .The USB are in phase at the adder and therefore they add together and gives USB. When they combined in the subtract or the USB cancel because in phase and LSB are add together and gives LSB.

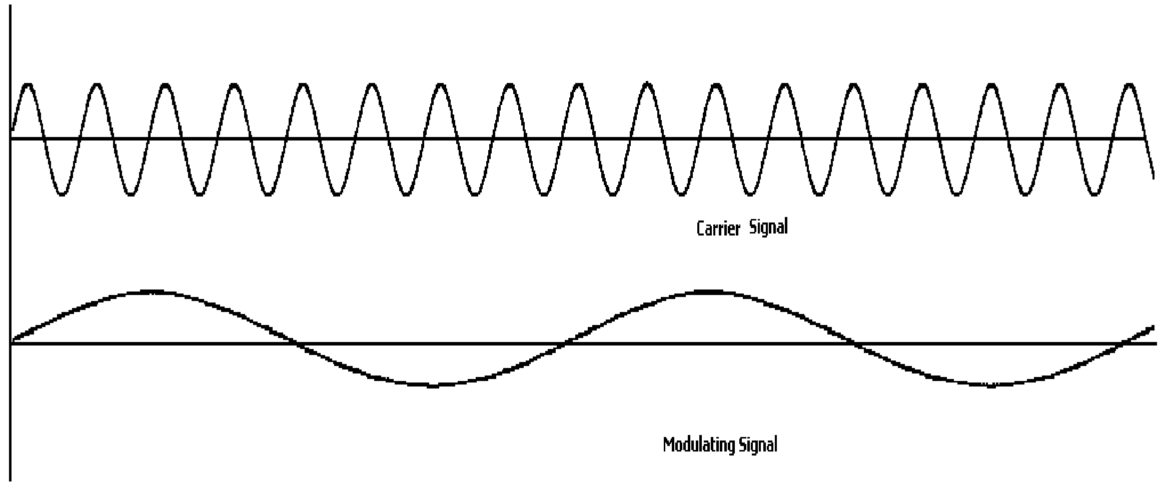
The base band signal m(t) can be uniquely recovered from a DSB-SC signal s(t) by first multiplying s(t) with a locally generated sine wave carrier and then low pass filtering the product. It is assumed that the local oscillator signal is exactly coherent or synchronous, in both frequency and phase with the carrier wave c(t) used in the balanced modulator to generate s(t). This method of demodulation is known as coherent detection or synchronous detection.

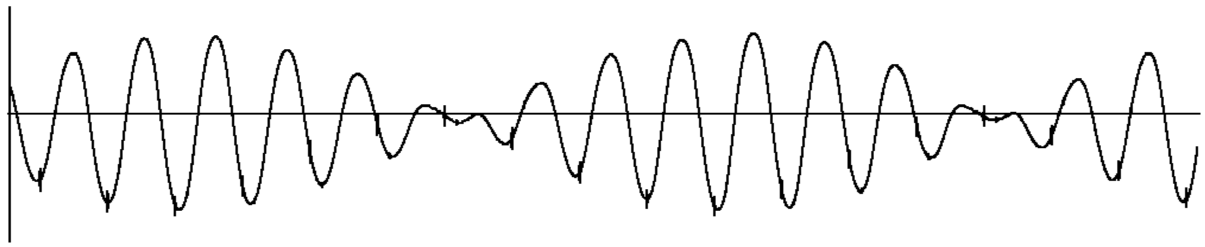
In this unit IC MC 1496 is used as synchronous demodulator. The MC 1496 is a monolithic balanced modulator/ balanced demodulator, is versatile and can be used up to 200MHz. On board generated carrier is used as synchronous signal.

PROCEDURE:

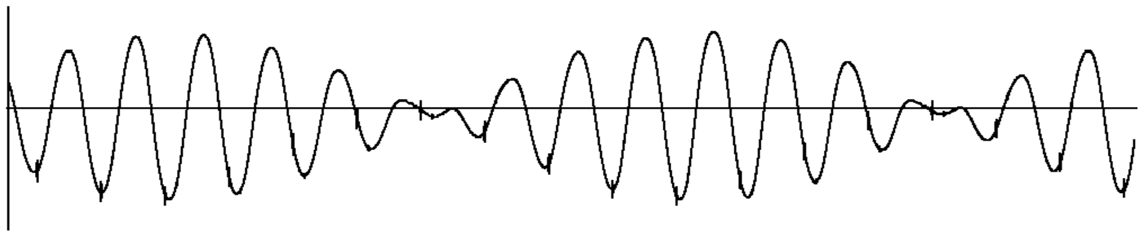
1. Connect the i/p carrier and modulating signals to the signal to the balance modulator A&B as indicated on the kit
2. Switch on the trainer and observe the o/p of the balance modulator A&B individually, it is a DSB – SC wave
3. Connect the o/p of the balance modulator A to on i/p of the summer and the o/p of balance modulator B to the other i/p of the summer
4. Now observe the o/p, it is SSB – SC wave
5. Adjust the amplitude and frequency of the carrier and modulating wave to get a clear wave form.

WAVE FORMS:

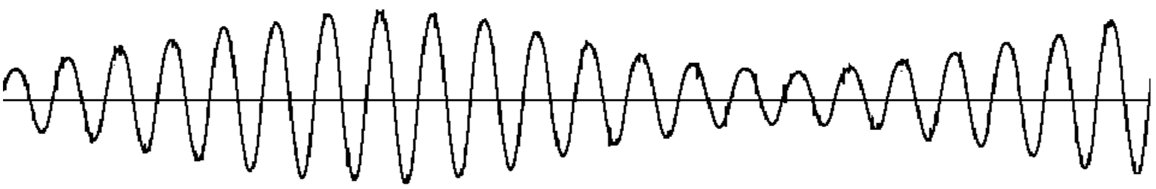




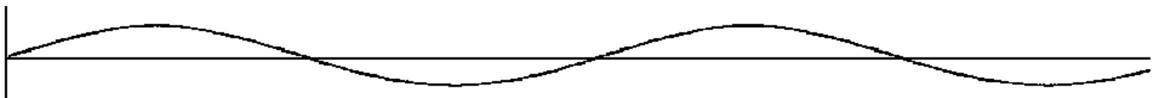
DSB-SC (A)



DSB-SC (B)



SSB SIGNAL



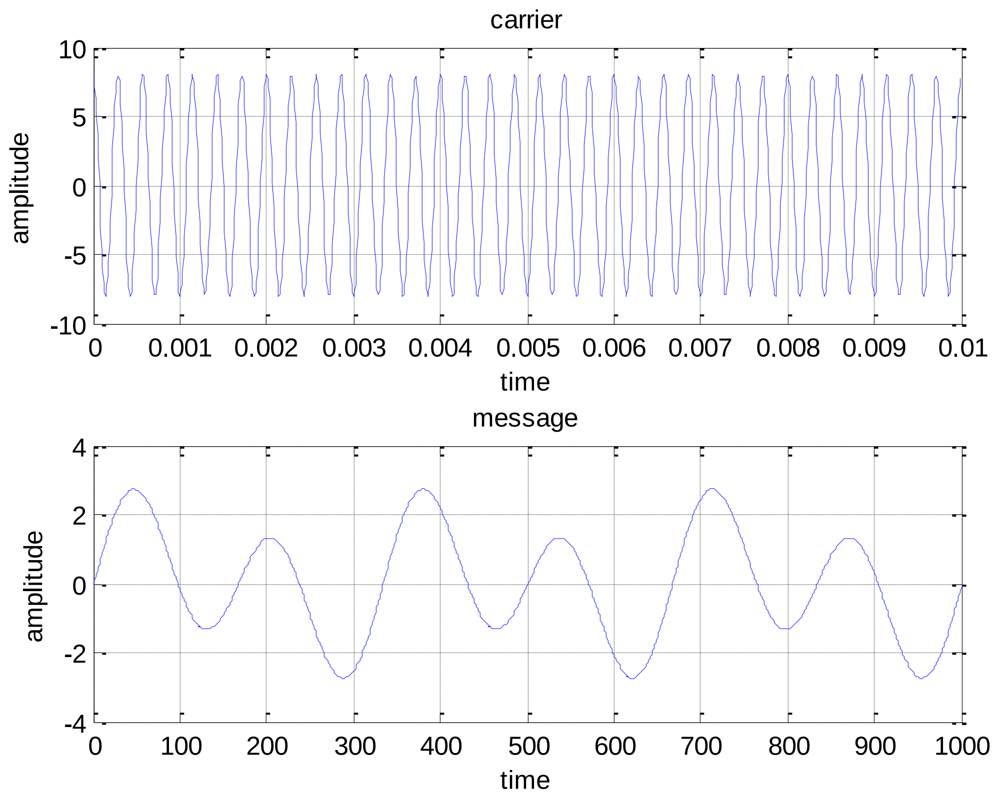
DEMODULATED SIGNAL

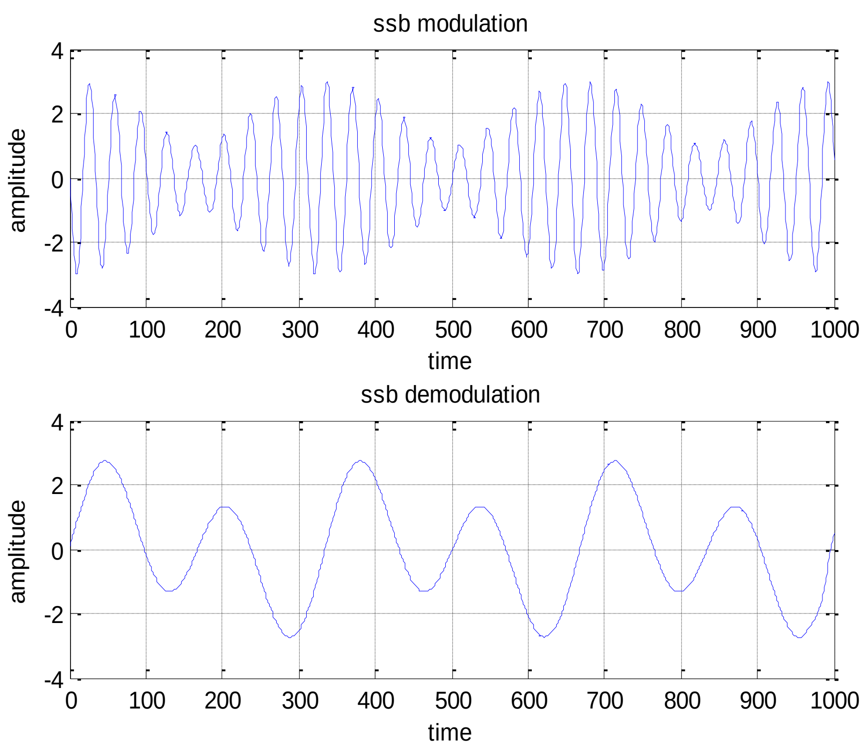
MATLAB CODE:

```
close all
clear all
clc
fs=100e3
t=0:1/fs:.01-1/fs
fc=3.5e3
am=2
fm=200
m=sin(2*pi*300*t)+2*sin(2*pi*600*t)
ac=8
c=ac.*cos(2*pi*fc*t)
figure
subplot(2,1,1)
plot(t,c)
grid
title('carrier');
xlabel('time');
ylabel('amplitude');
```

```
 subplot(2,1,2)
 plot(m)
 grid
 title('message');
 xlabel('time');
 ylabel('amplitude');
 figure
s=ssbmod(m,fc,fs,0)
subplot(2,1,1)
plot(s)
grid
title('ssb modulation');
xlabel('time');
ylabel('amplitude');
z=ssbdemod(s,fc,fs,0)
subplot(2,1,2)
plot(z)
grid
title('ssb demodulation');
xlabel('time');
ylabel('amplitude');
```

WAVEFORMS:





RESULT:

SSB- SC wave is generated by using two balanced modulators and a summer circuit and detected and simulated with MATLAB.

VIVA VOICE:

1. What are the two ways of generation of SSB wave?

\2. What are the features of filter method generation of SSB?

3. What are the advantages of phase shift method of SSB generation?

4. What are the disadvantages of phase shift method of SSB generation?

5. What are the advantages of SSB-SC AM?

6. What are the disadvantages of SSB-SC AM?

7. What are the applications of SSB-SC AM

4. STUDY OF SPECTRUM ANALYZER

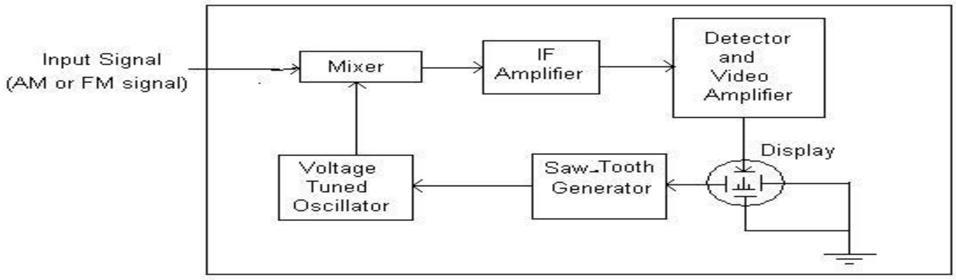
AIM:

To observe the spectrum of AM and FM signals and obtain the power levels in dBm of fundamental frequency components by using spectrum Analyzer.

APPARATUS REQUIRED:

Name of the Component/Equipment	Specifications	Quantity
Spectrum analyzer	LPT-2250 Spectrum analyzer	1
AM/FM generator	0.1MHz-110MHz	1
CRO	30MHz	1

Block diagram:



Theory:

A spectrum analyzer provides a calibrated graphical display on its CRT with frequency on the horizontal axis and amplitude on the vertical axis. Displayed as vertical lines against these coordinates are sinusoidal components of which the input signal in composed. The height represents the absolute magnitude, and horizontal location represents the frequency. This instrument provide a display of the frequency spectrum over a given frequency band.

Procedure:

1. AM signal is given to the spectrum analyzer.
2. Adjust the zero marker to carrier frequency and measure spectrum of AM.
3. For different values of f_c and f_m, observe the spectrum of AM.
4. Now remove AM signal and give FM signal to the spectrum analyzer.
5. Adjust the zero marker to carrier frequency and observe spectrum of FM.
6. Plot the spectrums of FM and AM.

Observation Table:

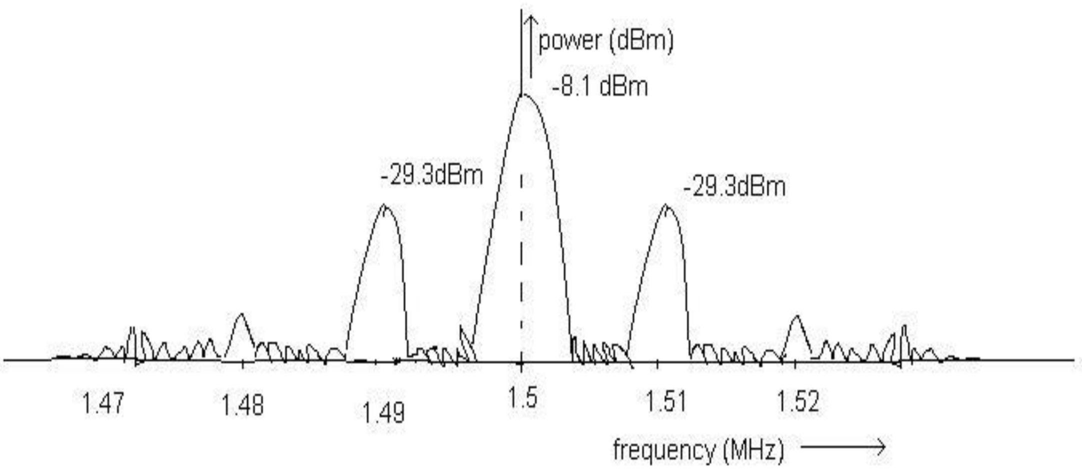
Table1: Readings for AM signal

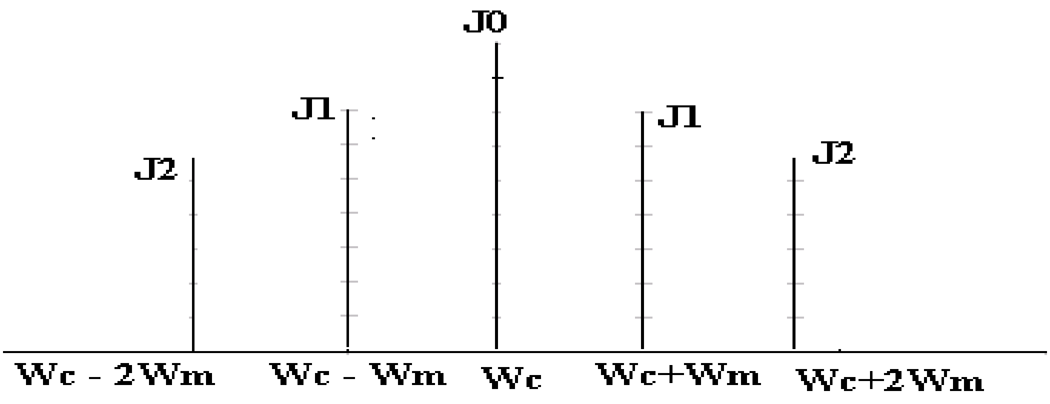
S.No.	f_c (MHz)	f_m (KHz)	(f_m+ f_c) (MHz)	(f_c - f_m) (MHz)
1				

Table2: Readings for FM signal

S.No.	f_c (MHz)	f_m (KHz)	(f_m+ f_c) (MHz)	(f_c - f_m) (MHz)
1				

EXPECTED WAVE FORMS:

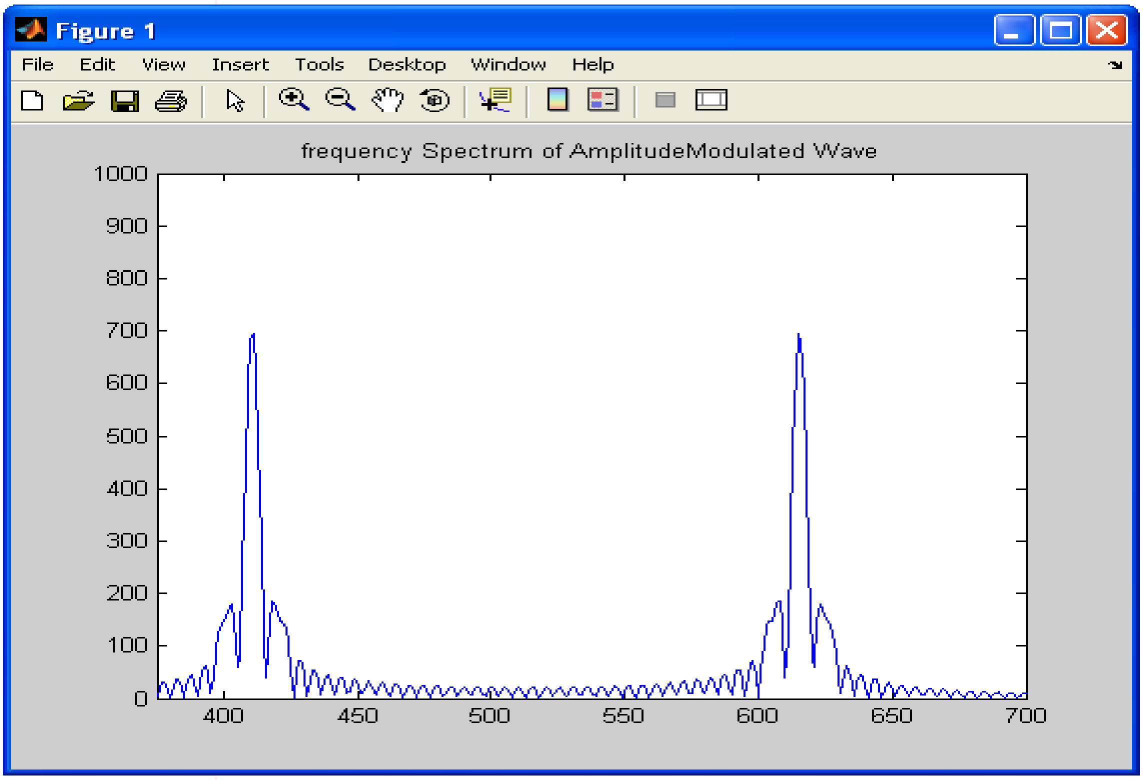




Frequency spectrum of FM wave

MATLAB CODE:

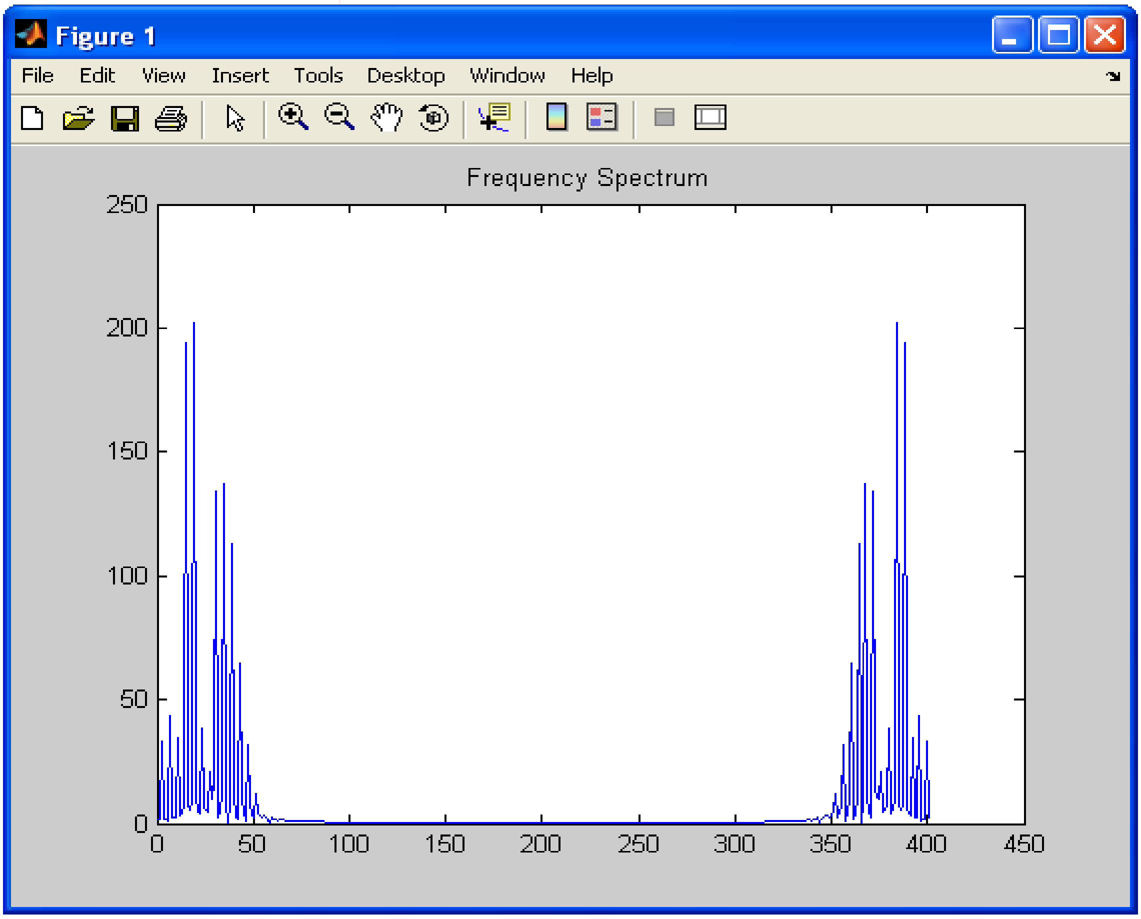
```
t=0:0.0001:0.02
fc=1000
Ec=7
Carr=Ec*sin(2*pi*fc*t)
fm=100
Em=3
Mod=Em*sin(2*pi*fm*t)
Am=(Ec+Mod).*(sin(2*pi*fc*t))
FA=fft(Am,1024)
subplot(1,1,1)
plot(fftshift(abs(FA)))
axis([375 700 0 1000])
title('frequency Spectrum of AmplitudeModulated Wave')
```



FREQUENCY SPECTRUM OF FREQUENCY MODULATED WAVE:

```
Am=1
Ac=2
fc=500
fm=200
fs=400
kf=30
dt=1/fs
T=20e-3
```

```
t=0:T/fs:T
mod=Am*cos(2*pi*fm*t)
FM=Ac*cos(2*pi*fc*t+(2*pi*kf*(cumsum(mod)*dt)))
FFM=fft(FM)
plot(abs(FFM))
title('Frequency Spectrum')
```



RESULT: AM – FM spectrum analyzer is studied and simulated using matlab.

5. PRE-EMPHASIS AND DE-EMPHASIS

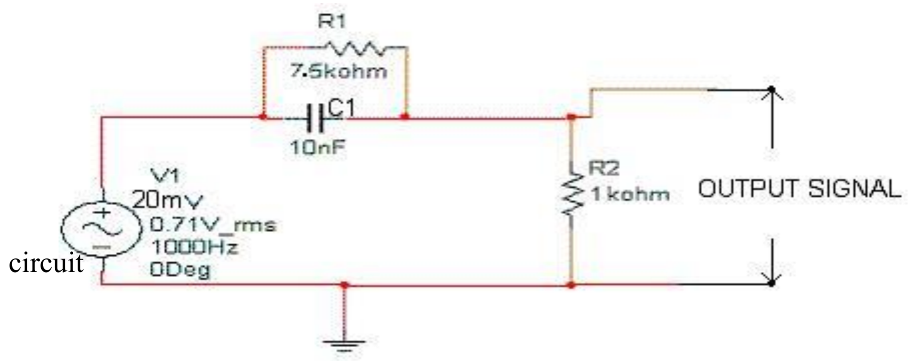
AIM:

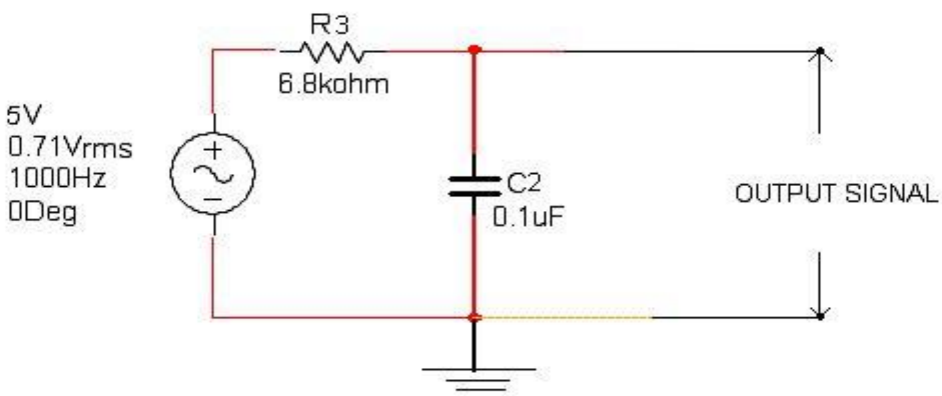
To obtain the characteristics of pre-emphasis and de-emphasis and simulate with MATLAB.

APPARATUS:

1. Pre-emphasis and de-emphasis trainer kit
2. CRO
3. Signal Generator

CIRCUIT:





THEORY:

The noise triangle shoes noise has a greater effect on the higher modulating frequencies than on the lower ones thus if the higher frequencies were artificially boosted at the transmitter and correspondingly cut at the receiver, an improvement of noise immunity could be expected, there by increasing signal to noise ratio.

This boosting of higher modulating frequencies in accordance with a prearranged curve

is termed as pre-emphasis and the compensation at the receiver is called de-emphasis.
If emphasis were applied to amplitude modulation some improvement would also result, but it is not as great as in FM because the highest modulating frequencies in AM are no more affected by noise than any others. Apart from that it would be difficult to introduce pre emphasis and de emphasis in existing AM services since extensive modifications would be needed particularly in view of the huge numbers of receivers in use.

PROCEDURE:

Pre-emphasis:

1. Apply input signal, through the signal generator and set the amplitude of the input signal at 100mv (p-p) with the help op the CRO.
2. Take DIB and set the value of 750mH and connect the circuit and switch on the trainer
3. Connect the i/p signal to the CH- I and o/p to CH- II of CRO
4. Now vary the frequency of the i/p signal from 100Hz to 10 KHz according to table given and note down the o/p voltage
5. Draw the graph frequency V_s gain in db on a semi log sheet.

$$L/R = 750mH / 3.3K$$

De-emphasis:

1. Adopt the same procedure 1 to 5 as above

TABULAR FORM:

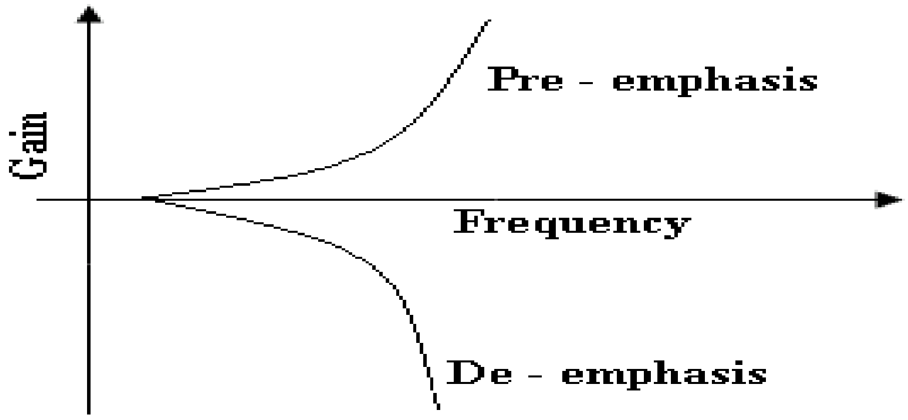
Pre-emphasis: (i/p signal voltage 100mV)

i/p signal frequency (Hz)	o/p voltage	Gain in db 20 log V_0/V_i
100		
200		
300		
400		
900		
1K		
2K		
3K		
4K		
5K		

De-emphasis:

i/p signal frequency (Hz)	o/p voltage	Gain in db 20 log V_0/V_i
100		
200		
300		
400		
500		
600		
700		
800		
900		
1K		
2K		
3K		
4K		
5K		

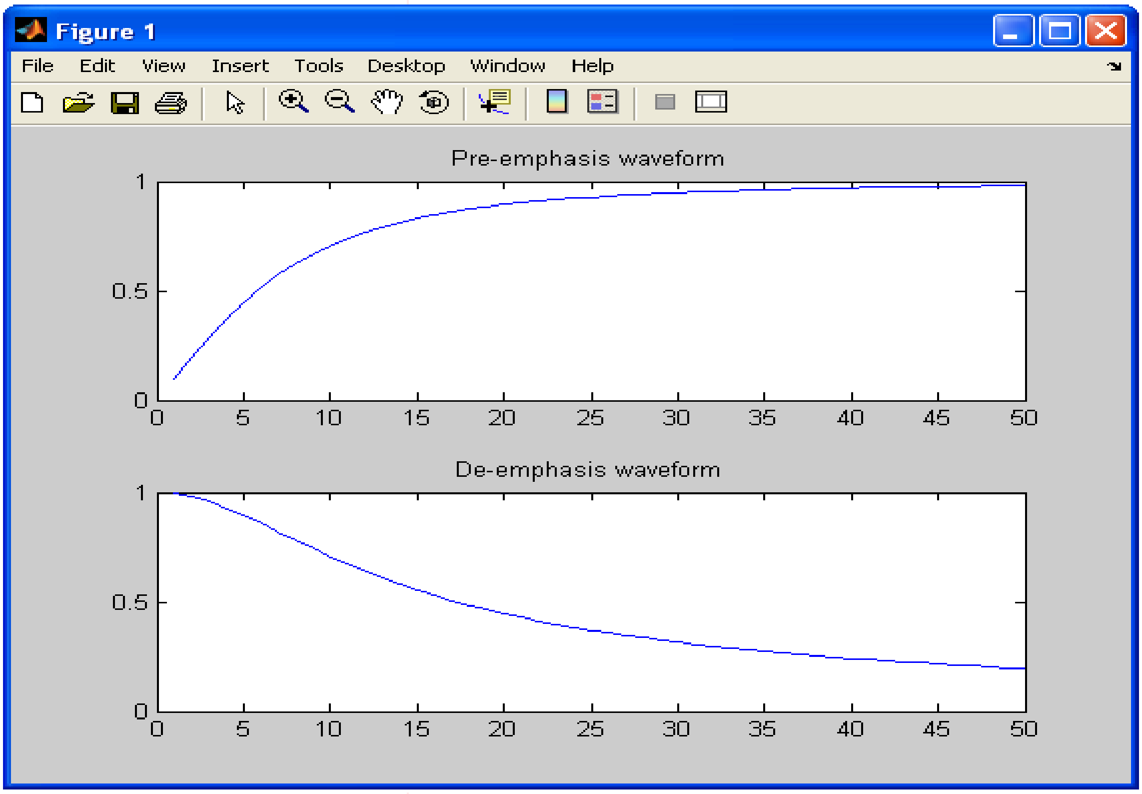
EXPECTED WAVEFORMS:



MATLAB CODE:

```
f1=10;
for f=1:50
x(f)=(1/sqrt(1+(f1/f)^2));
f2(f)=f;
end
subplot(2,1,1);
plot(f2,x);
title('Pre-emphasis waveform')
for f=1:50
y(f)=(1/sqrt(1+(f/f1)^2));
f3(f)=f;
end
subplot(2,1,2);
plot(f3,y);
```

```
title('De-emphasis waveform')
```



RESULT:

The characteristics of Pre-emphasis and De-emphasis networks are obtained and simulated using matlab.

VIVA VOICE:

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

6. TIME DIVISION MULTIPLEXING & DEMULTIPLEXING

AIM:

To study the Time Division Multiplexing and De-Multiplexing using Pulse Amplitude Modulation and Demodulation and to reconstruct the signals at the Receiver using Filters and simulate.

APPARATUS:

1. Time Division Multiplexer Trainer kit
2. Dual trace Oscilloscope
3. Patch chords

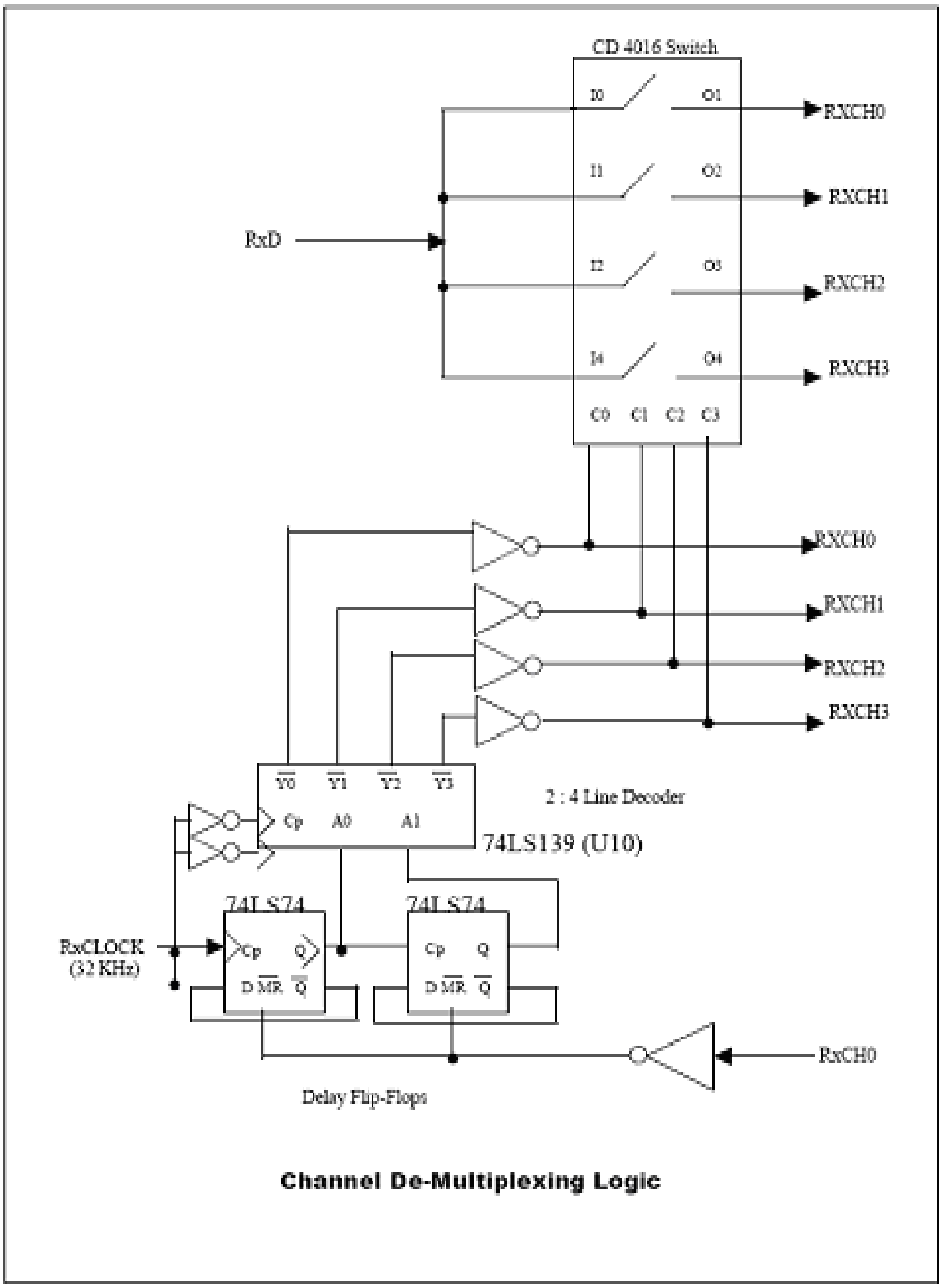
THEORY:

The Time Division Multiplexed PAM signals are conveyed over a single line. At the Receiver, the multiplexed signals are to be demultiplexed to yield the respective individual signals. A Demultiplexer at the receiver will perform the above task to yield in signals. Successive Low Pass filters filter out the high frequency components to recover the original signals. The figure below shows the demultiplexing logic implemented. The Demultiplexer at the receiver again employs the CD4016 switch for demultiplexing the multiplexed signal RxD. The switch extracts the individual signals depending upon the control signals, which are again generated by the 2:4 line decoders.

For achieving the synchronization b/w the transmitter and the receiver, the clocks for all the devices have start at the same time. Hence, the TXCLOCK is sent along with the data on another line as depicted in the fig. For frame synchronization purposes, the channel identification information in the form of one of the channels RXCHO is sent on another line, which marks the starting of the frame and starts the flip-flops at the beginning of the frame. This method calls for an additional two lines, which is very impractical and uneconomical for longer distance communications.

Hence other methods of deriving the clock and the identification information from the data itself are realized, which are discussed in depth in the coming exercises.

Circuit Diagram

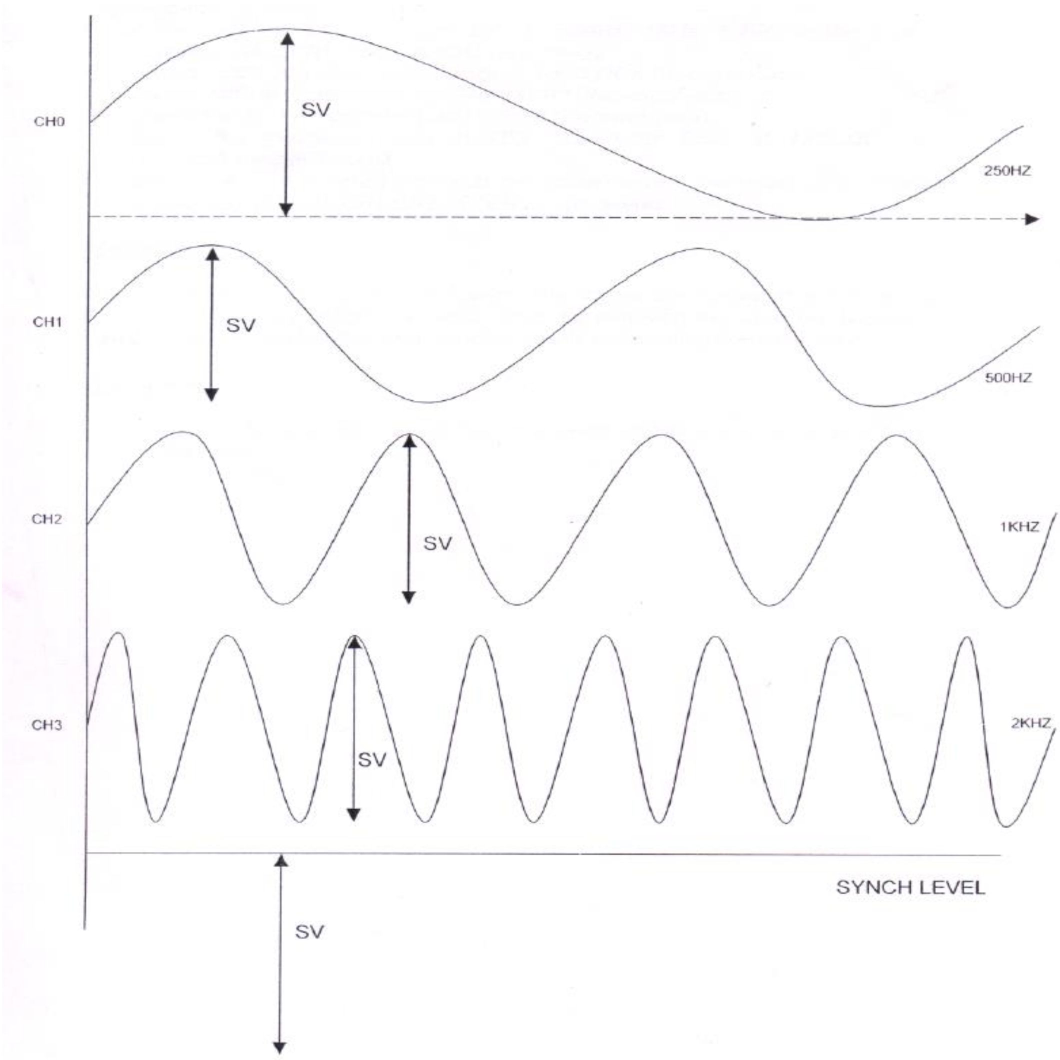


EXPERIMENTAL PROCEDURE:

- Connect the four channel inputs **250 Hz, 500Hz, 1KHz, 2 KHz** to the input of transmitter **CHO, CH1, CH2** and **CH3** respectively.
  Connect **TXCLOCK** (Transmitter Clock) to **RXCLOCK** (Receiver Clock).
  Connect **TXCH0** (Transmitter Sync) to **RXCH0** (Receiver Sync).
  Connect the **TXD** (Transmitter Data) to **RXD** (Receiver Data).

- Observe the multiplexed data at **TDX**, Transmitter Clock at **TXCLOCK** and Transmitter Syncat **TXCHO**.
  Observe the Demultiplexed signals at the receiver across the output of fourth order low pass filter at **CHO, CH1,CH2** and **CH3** respectively.

EXPECTED WAVEFORMS



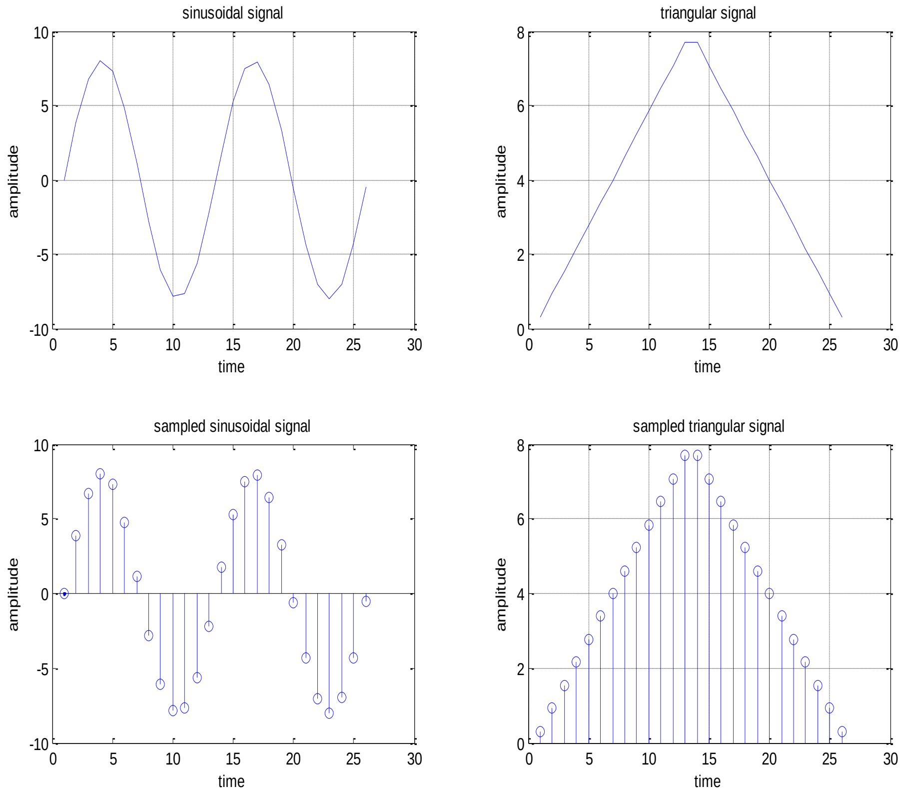
MATLAB CODE:

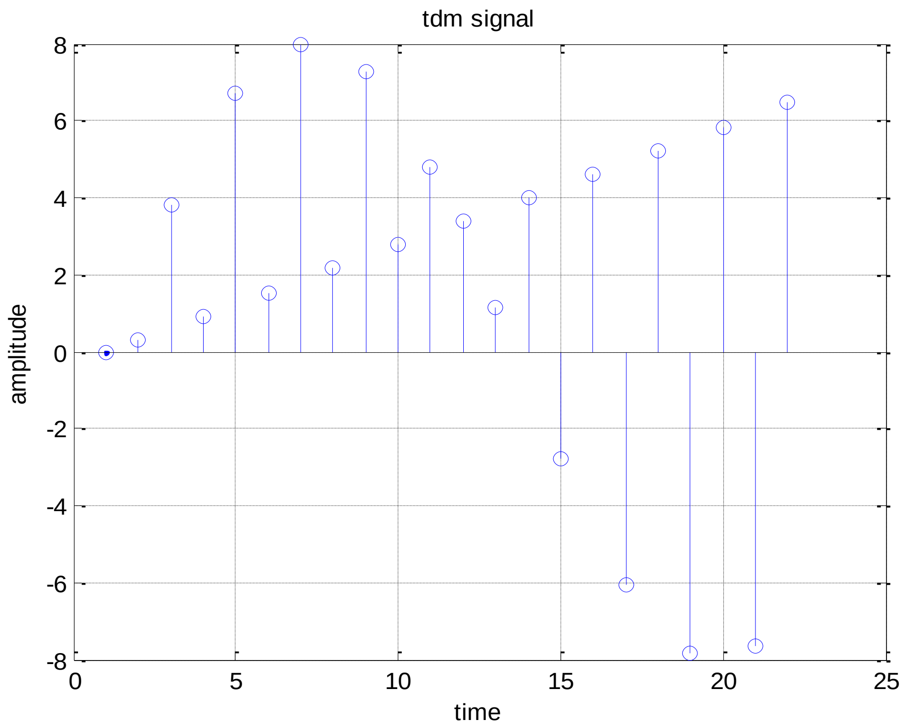
```
close all
clear all
clc
x=0:.5:4*pi
sig1=8*sin(x)
l=length(sig1)
sig2=8*triang(l)
subplot(2,2,1)
plot(sig1)
grid
```

```
title('sinusoidal signal');
xlabel('time');
ylabel('amplitude');
subplot(2,2,2)
plot(sig2)
grid
title('triangular signal');
xlabel('time');
ylabel('amplitude');
subplot(2,2,3)
stem(sig1)
title('sampled sinusoidal signal');
xlabel('time');
ylabel('amplitude');
subplot(2,2,4)
stem(sig2)
grid
title('sampled triangular signal');
xlabel('time');
ylabel('amplitude');
l1=length(sig1)
l2=length(sig2)
for i=1:11
    sig(1,i)=sig1(i)
    sig(2,i)=sig2(i)
end
tdmsig=reshape(sig,1,2*11)
figure
subplot(1,1,1)
stem(tdmsig)
grid
title('tdm signal');
xlabel('time');
ylabel('amplitude');
demux=reshape(tdmsig,2,11)
for i=1:11
    sig3(i)=demux(1,i)
    sig4(i)=demux(2,i)
end
figure
subplot(2,1,1)
plot(sig3)
grid
title('recovered sinusoidsl signal');
xlabel('time');
ylabel('amplitude');
subplot(2,1,2)
plot(sig4)
grid
title('recovered triangular signal');
xlabel('time');
```

```
ylabel('amplitude');
```

WAVEFORMS:





OBSERVATIONS:

From the above set up, we can observe that the signals are recovered at the receiver faithfully and are very distinct from each other. By removing the other two lines apart from the TXD, we find that the reconstructed signals suffer from severe distortion.

RESULT:

From the above observations, we conclude that synchronization is a very critical aspect of any TDM system and simulated with MATLAB.

<u>VIVA VOICE:</u>

1. Draw the TDM signal with 2 signals being multiplexed over the channel?
2. Define guard time & frame time?
3. Explain block schematic of TDM?
4. How TDM differ from FDM?
5. What type of filter is used at receiver end in TDM system?
6. What are the applications of TDM?
7. If two signal band limited to 3 kHz, 5KHz & are to be time division multiplexed. What is the maximum permissible interval between two successive samples.
8. Is the bandwidth requirement for TDM & EDM will be same?

7. VERIFICATION OF SAMPLING THEOREM

AIM:

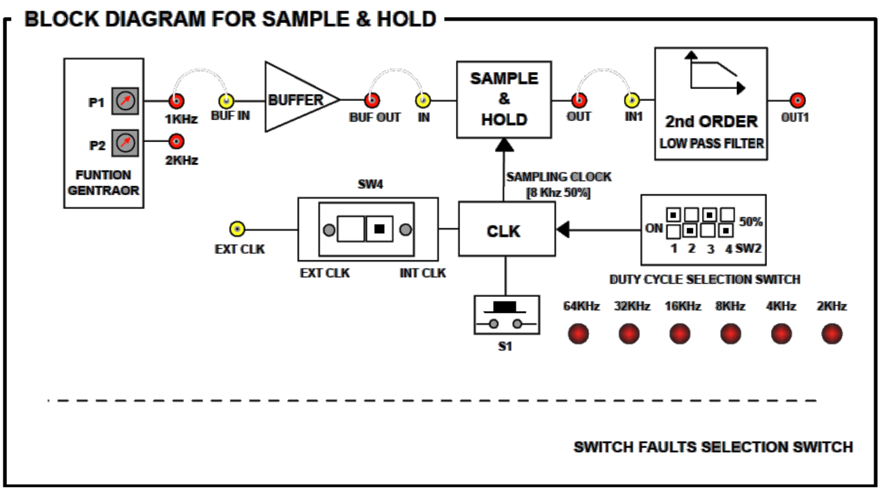
1.To study the signal sampling and its re-construction techniques.
2.To observe the difference between "Sampling Amplifier" and "Sample Hold Amplifier" outputs and its effects on reconstruction.
3.To simulate with MATLAB.

APPARATUS:

1. Analog signal sampling and reconstruction unit kit
2.Dual trace Oscilloscope
3.Patch chords

THEORY:

As Sampling theorem states that as long as the sampling rate is at least equal to twice the signal frequency, the signal can be faithfully reconstructed from the samples. This can be verified here by selecting the different sampling signals.The sampling is achieved by an electronic switch that turns ON and OFF at the different sampling rates and the samples of the analog signal is being transmitted during the sampling times. This is best illustrated with the help of the diagram given below



The analog switch CD 4016 switches at the rate of the sampling rate selected. These sampling clock acts as the control input and whenever the switch is "ON", i.e, during the TON of the sampling frequency, the base band signal from the unity gain is latched to the output .The switch changes its state at the rate of the sampling frequency selected. At the reconstruction side, a low pass filter gathers all the samples to produce the original source signal.

EXPERIMENTAL PROCEDURE:

- Connect the **2KHz** 5V p-p signal generated onboard to the **ANALOG INPUT**, by means of the patch-cords provided.
- Connect the sampling frequency signal in the **INTERNAL** mode, by means of the shorting pin provided.

- Observe the output of sampling amplifier at **SAMPLE OUTPUT**.
- Connect **SAMPLE OUTPUT** to the **INPUT** of the second order and fourth order low
- pass filter.

OBSERVATIONS:

We observe that, by sampling a 2KHz signal at 4KHz the signal is re-constructed; Even increasing the sampling frequency it can be observed that the re-construction will be similar to 4KHz

II)SAMPLE HOLD AMPLIFIER

THEORY:

In practice, the sampling is accomplished by means of high speed switching transistor circuits. Accordingly, we find that the resulting sampled waveform deviates from the ideal form of instantaneous sampling because the operation of a physical switching circuit, however fast, still requires a nonzero interval of time.The sampling process takes the form of natural sampling or flat-op sampling. In both cases, the spectrum of the sampled signal is scaled by the ratio of T/Ts, where T is the sampling-pulse duration and Ts is the sampling period. Typically, this ratio is quite small, with the result that the signal power at the output of the low-pass filter in the receiver is correspondingly small.We may obviously remedy this situation by the use of amplification, which may be quite large. A more attractive approach, however, is to use a simple ***sample-and-hold circuit.*** The circuit, shown below (Fig. a) consists of an amplifier of unity gain and low output impedance, a switch, and a capacitor. Refer to fig.b for the sample & hold waveform.

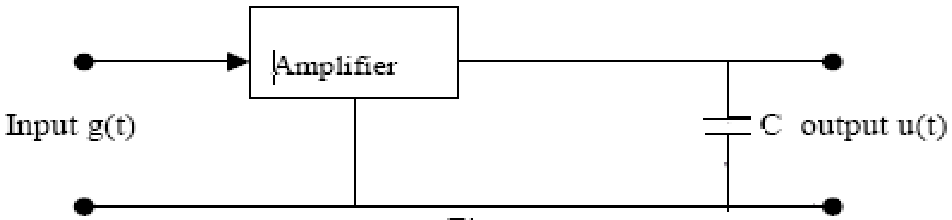


Fig. a

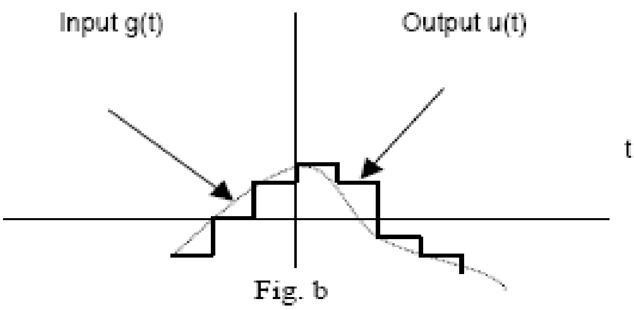


Fig. b

The switch is timed to close only for the small duration T of each sampling pulse, during which time the capacitor rapidly charges up to a voltage level equal to that of the input sample. When the switch is open, the capacitor retains its voltage level until the next closure of the switch.Thus the ***sample-and –hold circuit***, in its ideal form, produces an output waveform that represents a ***staircase*** interpolation of the original signal

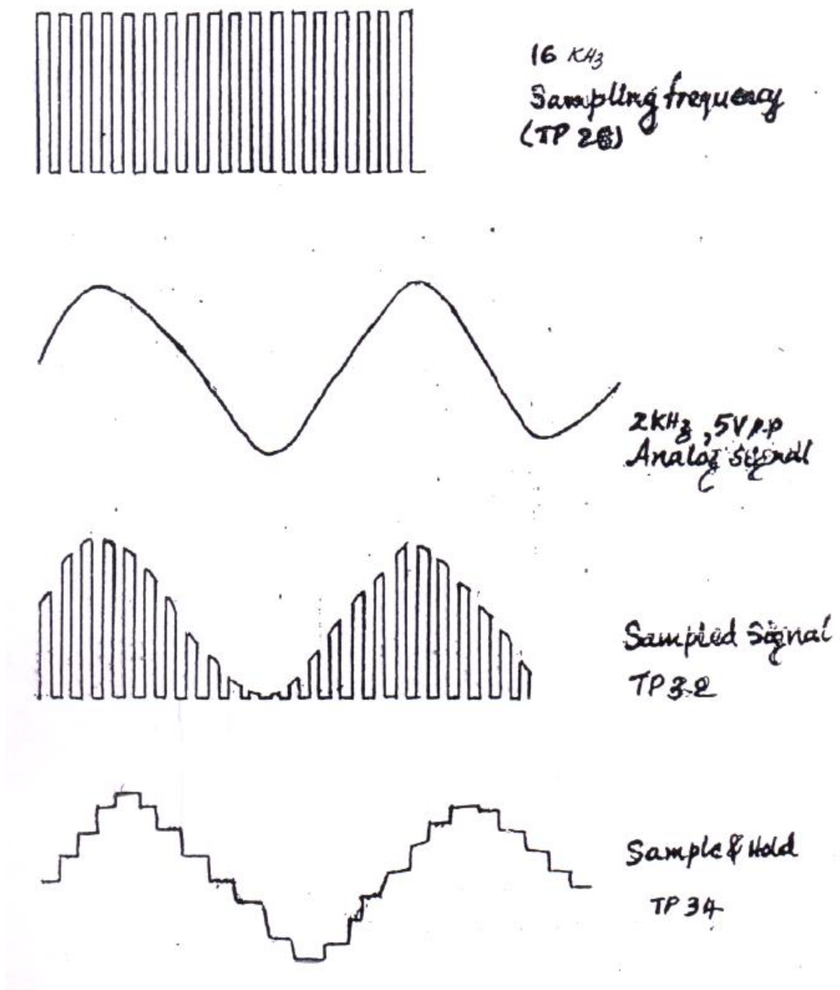
EXPERIMENTAL PROCEDURE:

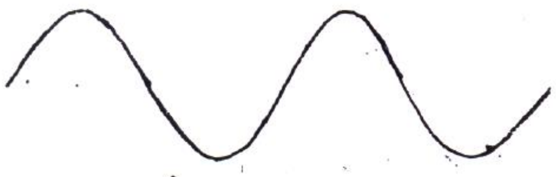
- Connect the **2KHz** 5V p-p signal generated onboard to the **ANALOG INPUT**, by means of the patch-cords provided.
- Observe the output of sampling Amplifier waveform at **SAMPLE OUTPUT**.
- Connect this **SAMPLED OUTPUT** to the input of the **FOURTH ORDER** Low pass filter.
- Observe the reconstructed signal at the **OUTPUT** of the fourth order low pass filter.
- Observe the output of sample hold amplifier at **SAMPLE HOLD OUTPUT**.
- Observe the reconstructed signal at the **OUTPUT** of the fourth order low pass filter.

OBSERVATIONS:

From the observations, we find the signal is re-constructed more faithfully in the case of sample hold amplifier rather in case of sampling amplifier, where the re-constructed signal suffers an amplitude distortion.

EXPECTED WAVEFORMS:





(a) 2KHz. Input analog Signal

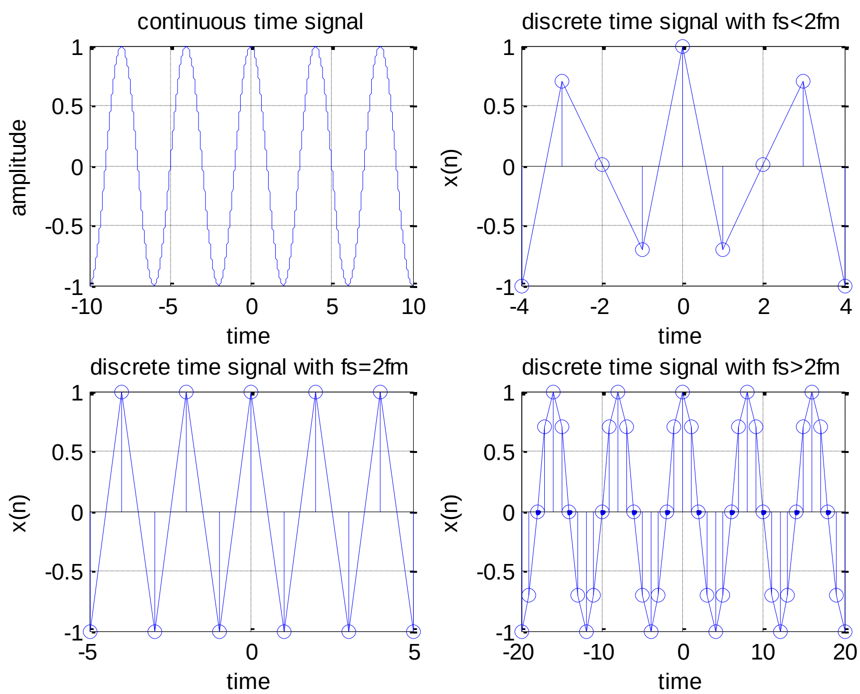
(b) Reconstructed o/p at TP38

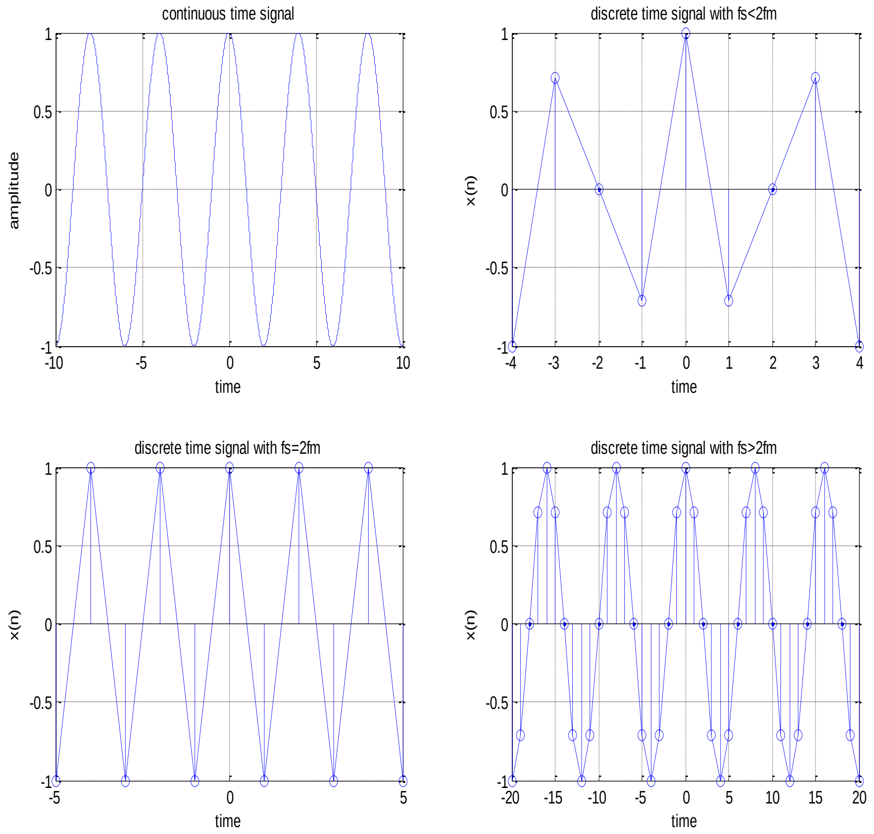
MATLAB CODE:

```
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
```

WAVEFORMS:





RESULT:

i)From the above, we infer that for signal reconstruction with no distortion, Nyquist Criterion has to be satisfied and hence we prove the sampling theorem. If Nyquist Criteria is not satisfied, or if the signal is not band-limited, then the spectral overlap called “aliasing” occurs, causing higher frequency signal to show up at lower frequencies in the recovered signal

ii)Hence, we infer that the use of sample hold logic compensates for the poor amplitude at the input of the low pass filter resulting in faithful reconstruction.

iii) simulation results has been verified.

VIVA VOICE:

1.What are the types of sampling?

2.State sampling theorem?

3.What happens when fs < 2 fm ?

4.How will be the reconstructed signal when fs >= 2fm?

5.Explain the operation of sampling circuit?

6. Explain the operation of re-construction circuit

8. PULSE AMPLITUDE MODULATION & DEMODULATION

AIM:

a)To study the techniques and generate the waveforms of Pulse Amplitude Modulation with

i)Natural Sampling.

ii)Flat top sampling

b)To study the demodulation techniques and generate the waveforms for PAM signals

c) To simulate the generation detection using MATLAB.

APPARATUS:

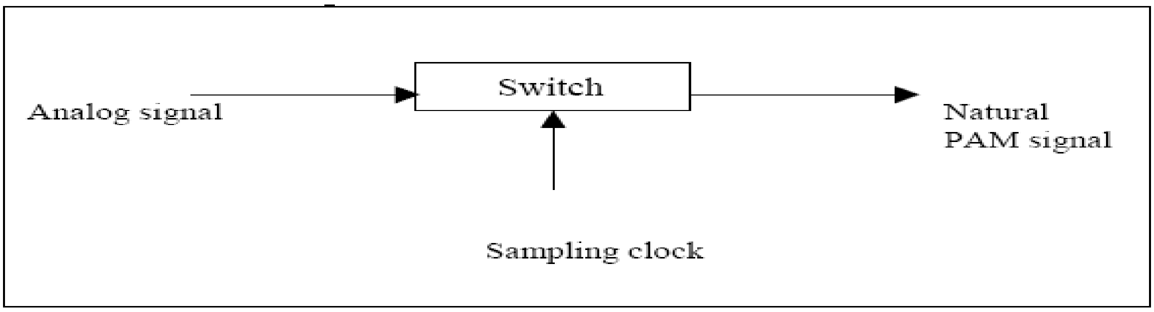
1. PAM trainer kit
2. Dual trace Oscilloscope
3. Patch chords

I) NATURAL SAMPLING.

THEORY:

It forms an excellent introduction to pulse modulation in general 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 to the instant of sampling. The pulses are then sent by either wire or cable, or else are used to modulate a carrier.As shown in the figure, the two types are double polarity PAM, which is self-explanatory and single polarity PAM, in which a fixed dc level is added to the signal to ensure that the pulses are always positive. As will be seen, 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.It is very easy to generate a PAM signal. The figure below shows the logic used.The sampling clock can be any of the four different clocks of 32 KHz,16KHz,8 KHz and 4 KHz that are generated on the board and can be selected by the help of a switch provided.

BLOCK DIAGRAM OF PAM GENERATOR:



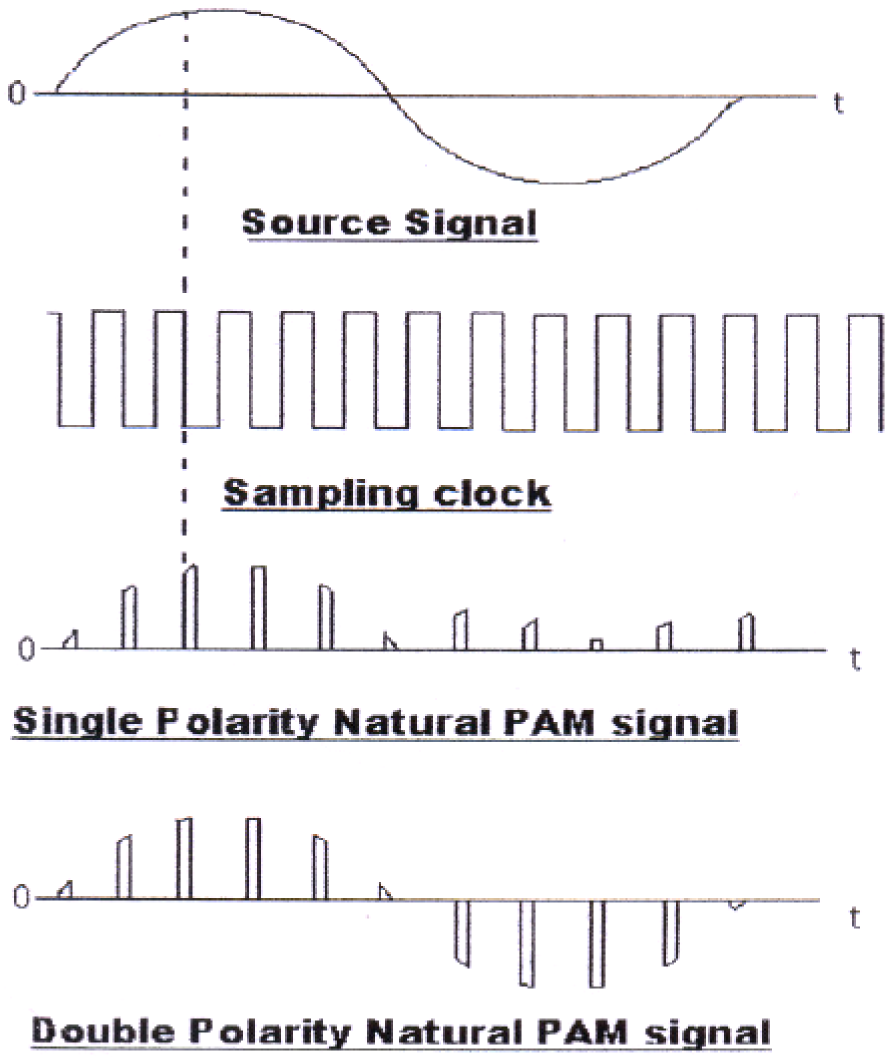
In a generator, the signal to be converted to PAM is fed to one I/P of an AND gate. Pulses at the sampling frequency are supplied to the other I/P of AND gate to open it during the watched time intervals. The output of the gate then consists of pulses at the sampling rate, equal in amplitude to the signal voltage at each instant.

Experimental Procedure:

- Connect signal source 1 KHz (A1) to (A) as shown in the interconnection diagram with the help of the patch cords given.
- Select sampling frequency to 8 KHz
- Select natural sampling by pushing the switch to the extreme left as shown in the figure.

- Adjust pulse width potentiometer to extreme anti clockwise.
- Connect the oscilloscope with the input analog signal A1 and with the PAM modulator output E1.

EXPECTED WAVEFORMS



Natural Sampling PAM Modulator

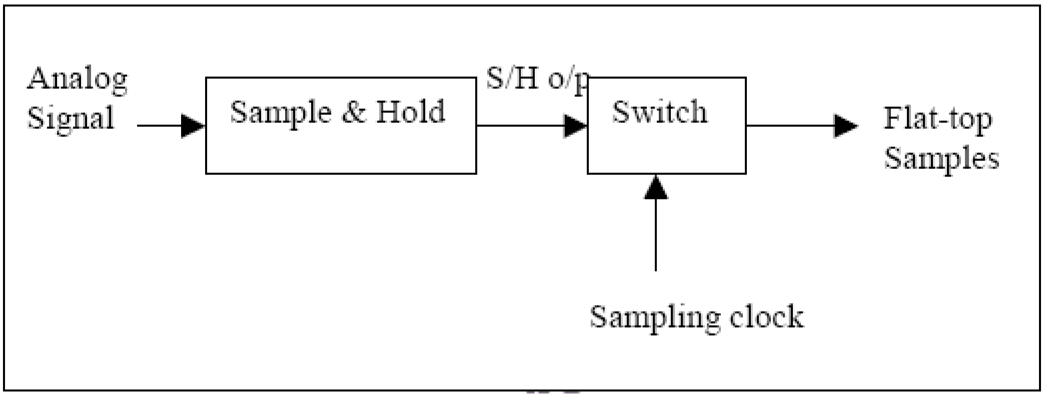
Observations:

- The PAM modulator output will be a product of input analog signal and regularly spaced pulse train.
- Note the pulse width of PAM and by varying the pulse width potentiometer, record the pulse width of PAM.
- Determine the minimum and maximum duty cycle of pulse
- Determine the minimum and maximum duty cycle of pulse at 4 KHz, 16 KHz and 32KHz.
- Repeat the above experiment by connecting 2 KHz (A2) to (A)

II) FLAT TOP SAMPLING.

THEORY:

In Flat top sampling, the amplitude of the samples remain constant at an instant of time unlike in natural sampling where the amplitude of the samples vary in accordance with the amplitude of the input signal that is to be sampled. The generation of flat top samples involves two stages as shown in the figure given below.

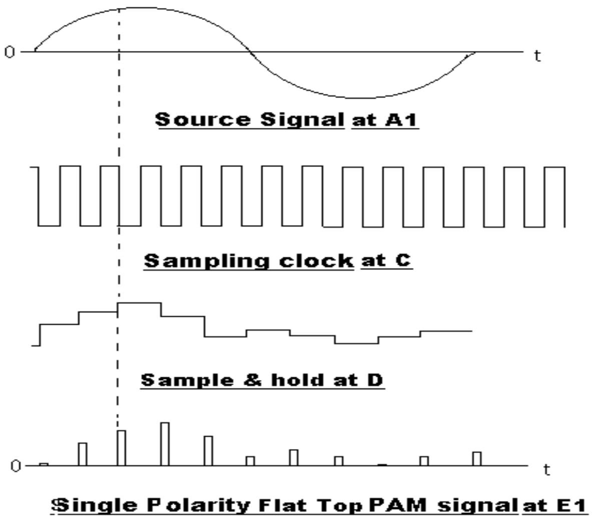


For generating the Flat top samples, the signal to be sampled is first fed to a Sample and hold amplifier which generates the staircase waveform as represented in the above graph at the point D. The sampling clock selected determines the hold period of the S/H waveform. The resulting waveform is then passed on to a electronic switch (sampler), which latches the samples of the S/H waveform for the period determined by the duty cycle of the input sampling clock. The resulting samples are flat-topped corresponding to the flat portions of the input S/H wave.

EXPERIMENTAL PROCEDURE:

- Connect signal Source 1KHz(A1) to (A)
- Select sampling frequency to 8 KHz
- Select flat sampling by moving switch to the extreme right as shown in the figure.
- Adjust the pulse width potentiometer to extreme anti clockwise
- Connect the oscilloscope with the input analog signal A1 and with the PAM modulator output E1.

The figure below shows the different waveforms for the generation of Flat-top samples.



OBSERVATIONS:

- Observe the S/H waveform at the point D and the sampling clock at the point C
- Observe the differences between the Flat topped samples and the natural samples.

b) DEMODULATION OF PAM SIGNALS

THEORY:

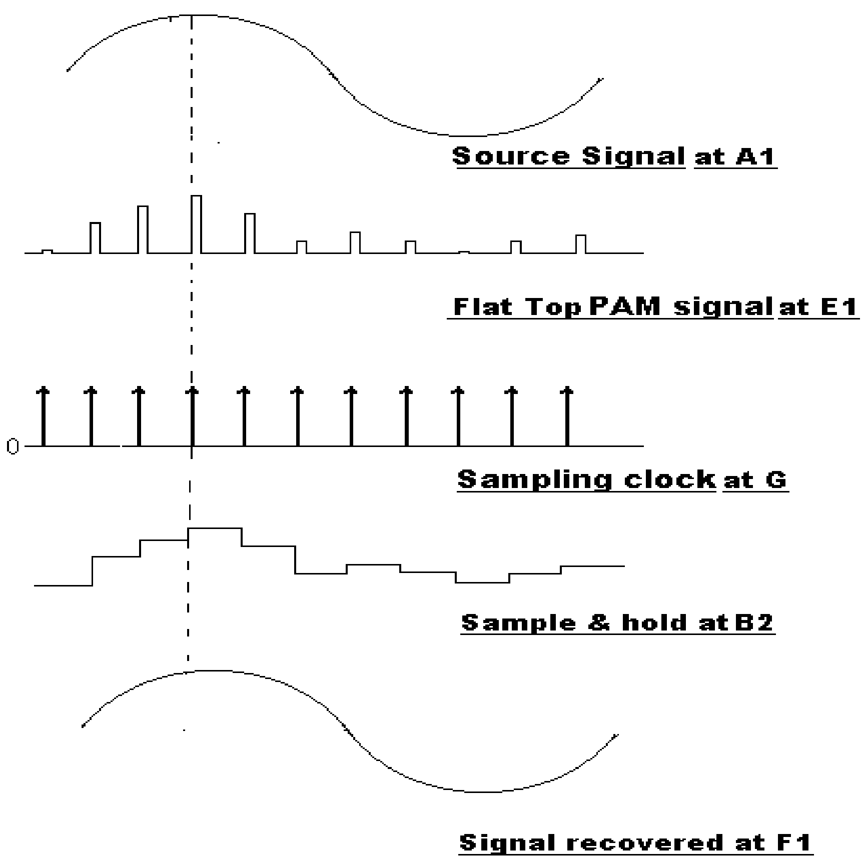
To demodulate the PAM signals, a low pass filter is enough. This does not result in reconstruction of analog signals with sufficient amplitude.

The PAM pulses coming from the transmitter are sampled by a sampling signal, which is regenerated at the receiver itself. The sampler output is kept at a steady level until the following sample arrives, thereby generating a step signal that approximates the starting signal. The reconstruction of analog signal, by passing step signal through LPF will have wider amplitude than the signal reconstructed directly from the PAM pulses.

EXPERIMENTAL PROCEDURE:

- Connect the signal source 1 KHz (A1) to (A), PAM output (E1) to receiver input (E) and PAM demodulator output (B2) to LPF input (B).
- Select sampling frequency to 8 KHz.
- Select flat sampling.
- Connect the oscilloscope with the signal B1 and sampling pulse regenerator output G.

WAVEFORMS:



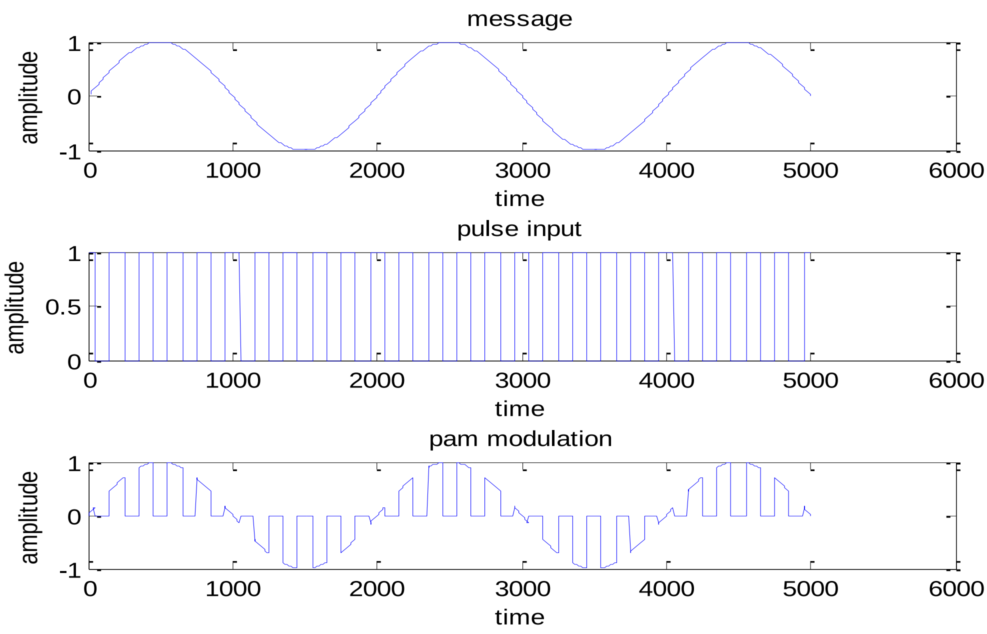
OBSERVATIONS:

- Observe the relative position of PAM pulses and sampling pulses.
- Record the PAM demodulator output at B2.
- Vary the phase adjust potentiometer gradually till the sampling pulses are in middle of PAM pulses.
- Record the PAM demodulator output at B2, which will be a step signal.
- Observe the reconstructed output at F1.
- Vary the pulse width of PAM pulses and observe the reconstructed output at F1.
- Repeat the above procedure at other sampling frequencies.
- Explain the distortion if the sampling frequency is 4 KHz.

MATLAB CODE:

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

WAVEFORMS:



RESULT:

The techniques of Pulse Amplitude Modulation with natural sampling ,flat top sampling and the demodulation techniques for PAM signals have been observed.

VIVA VOICE:

1. TDM is possible for sampled signals. What kind of multiplexing can be used in continuous modulation systems?
2. What is the minimum rate at which a speech signal can be sampled for the purpose of PAM
3. What is cross talk in the context of time division multiplexing?
4. Which is better, natural sampling or flat topped sampling and why?
5. Why a dc offset has been added to the modulating signal in this board? Was it essential? for the working of the modulator? Explain.
6. If the emitter follower in the modulator section saturates for some level of input signal, then what effect it will have on the output
7. Study about the frequency spectrum of PAM signal and derive mathematical expression for it?
8. Explain the modulation circuit operation?

9. PULSE WIDTH MODULATION& DEMODULATION

AIM:

To study and generate the waveforms for PWM modulation and demodulation and simulated using matlab.

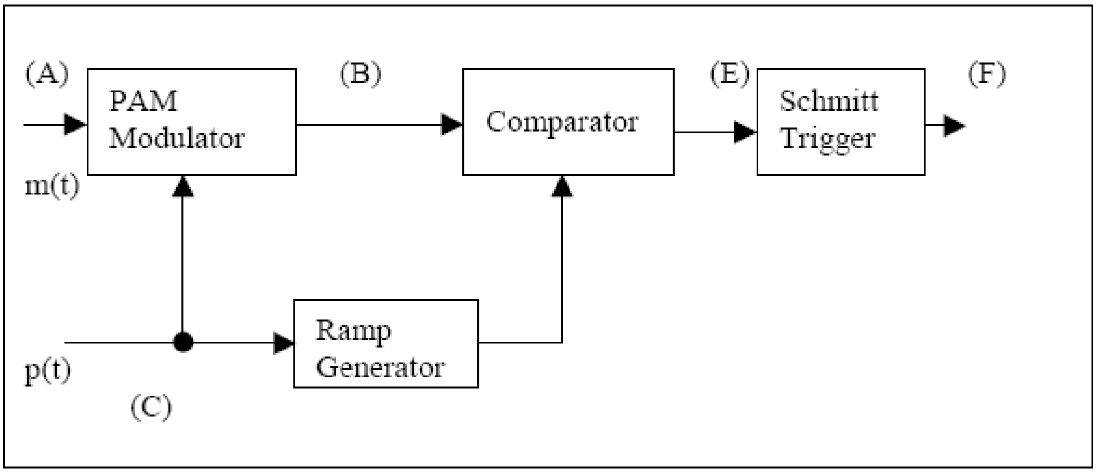
APPARATUS:

1. PWM trainer kit
2. Dual trace Oscilloscope
3. Patch chords

I) GENERATION OF PWM SIGNALS

THEORY:

Pulse Width Modulation is a type of pulse modulation where the duration of the pulse varies in proportion with the sample values of the analog signal. The generation of PWM signal involves of first sampling the analog signal to its PAM form. The figure below shows the block diagram for generating the PWM signal.

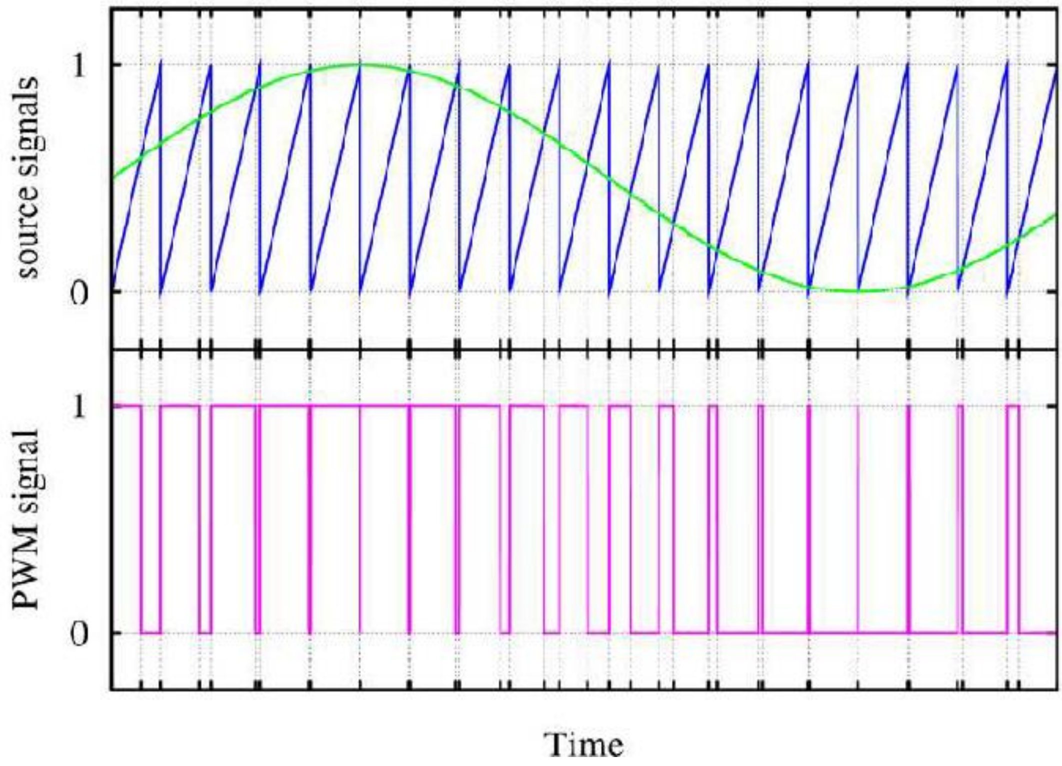


The modulating signal m (t) is applied to the input of a PAM modulating circuit, to generate the PAM signal (B). The same pulse train (C) which supplies the PAM modulator is used to gate on a ramp generator, to generate a train of ramp pulses (D) which all will have equal slopes, amplitudes and durations. These ramp pulses are fed to a comparator where they are compared with the PAM pulses. Depending upon the comparison, the comparator generates the varying –width pulses. These pulses gate a Schmitt trigger circuit to generate the varying-width rectangular pulses of the PWM wave (F). The waveforms below illustrate the different signals at the monitoring points on the board.

EXPERIMENTAL PROCEDURE:

- Connect signal source 1 KHz (A1) to (A).
- Select sampling frequency to 8 KHz.
- Select flat sampling.
- Connect the oscilloscope with the input analog signal A1 and with the RAMP generator J.
- Adjust the RAMP generator potentiometer, such that the peak-to-peak of RAMP is uniformly higher than the input analog signal on both positive and negative cycle.

WAVEFORMS:



OBSERVATIONS:

- Observe the PWM Modulator output E2 along with input analog signal A1.
- Fine tune RAMP generator potentiometer, hence the negative peak of analog signal results in minimum pulse width of PWM.
- Hence in PWM, pulse-width-modulation, the duration of regularly spaced rectangular
- pulses vary in direct proportion to the sample values of analog signal.
- Repeat the above procedure at 16 and 32 KHz.
- Study the PWM pattern by varying the amplitude of analog signal.

ii)DEMODULATION OF PWM SIGNALS.

THEORY:

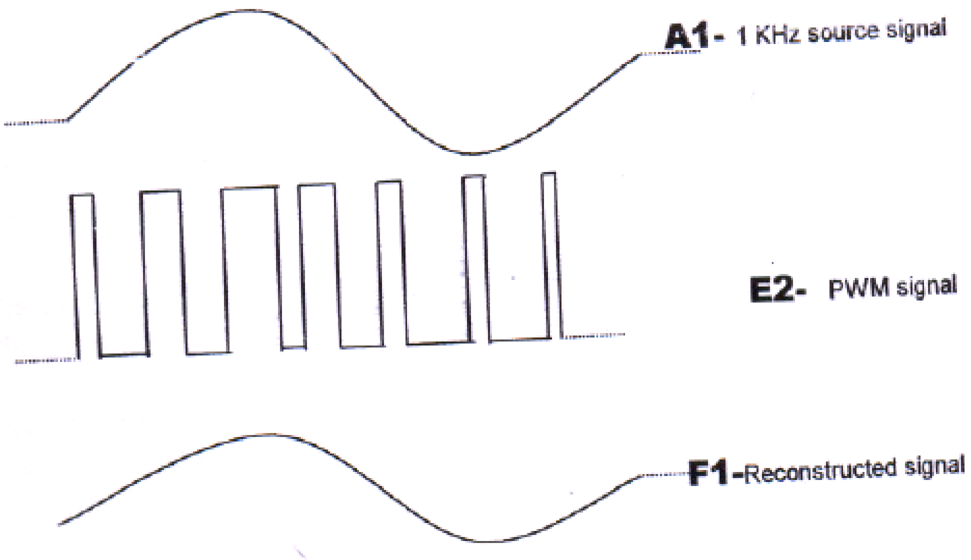
Since in Pulse Width Modulation, the pulse widths vary in direct proportion to the amplitude of the signal to be modulated, a low pass filter is enough for recovering the original signal from the pulses.

The PWM signals are first amplified in an amplifier and then passed to a low pass filter. The Low pass filter gathers all the samples together and regenerates the original signal. The figure below shows the interconnection diagram as well as the block diagram for the demodulation of the PWM signals.

EXPERIMENTAL PROCEDURE:

- Connect signal source 1 KHz (A1) to (A), PWM output (E2) to receiver input (E) and PWM demodulator output (B1) to LPF input (B).
- Select sampling frequency to 8 KHz.
- Select flat sampling.
- Connect the oscilloscope with LPF output F1

EXPECTED WAVEFORMS



PWM Demodulator

OBSERVATIONS:

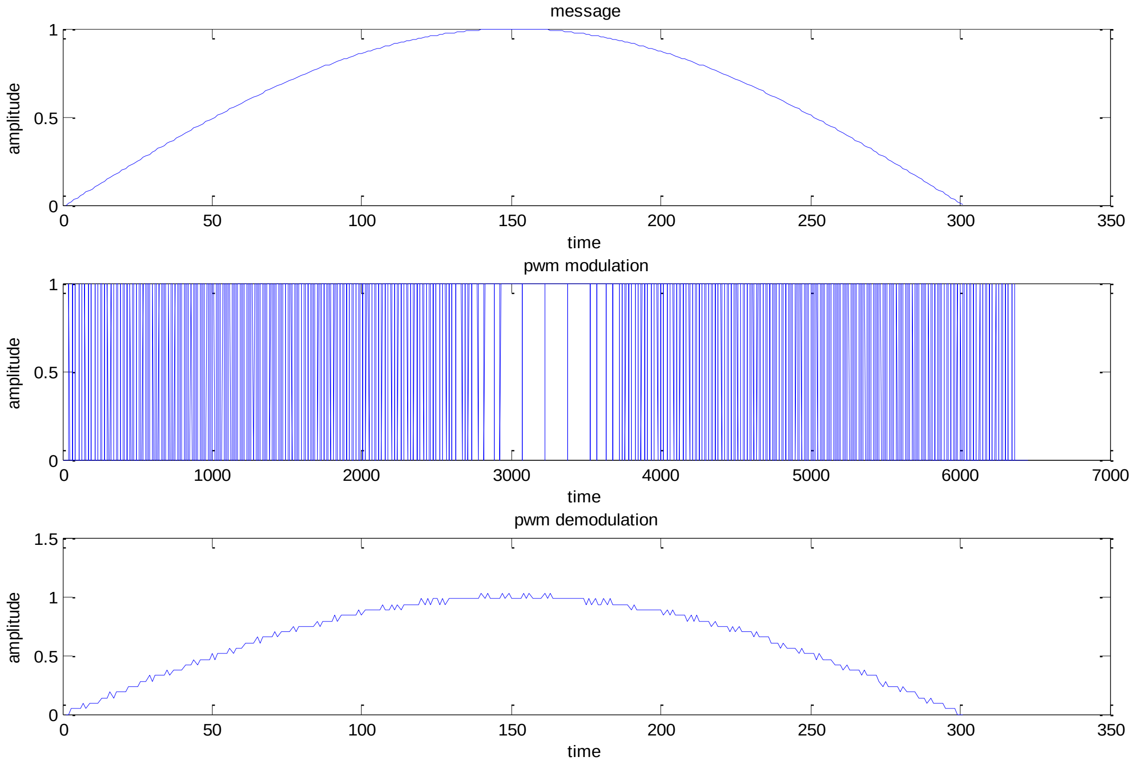
- Observe the reconstructed analog signal at F1, by gradually varying the RAMP generator potentiometer.
- Observe RAMP generator and input analog signal at the condition when the signal F1 is proper sinewave.
- Establish the condition between RAMP generator output and input analog signal for the reconstruction of analog signal.
- With the established condition, repeat the above procedure at 16, and 32 KHz
- sampling frequency and at 2 KHz and (0.5 + 1 + 2) KHz input analog signal.
- Explain the distortion of the reconstructed analog signal at 4 KHz sampling frequency.

MATLAB CODE:

```
close all
clear all
clc
fs=150
t=[0:2*fs]/fs
fc=7
x=sin(pi/4*2*t)
s=modulate(x,fc,fs,'pwm','centered')
figure
```

```
subplot(3,1,1)
plot(x)
title('message');xlabel('time');ylabel('amplitude');
subplot(3,1,2)
plot(s)
title('pwm modulation');xlabel('time');ylabel('amplitude');
z=demod(s,fc,fs,'pwm','centered')
subplot(3,1,3)
plot(z)
title('pwm demodulation');xlabel('time');ylabel('amplitude');
```

WAVEFORMS:



RESULT:

Generation and demodulation of PWM signals have been observed and simulation results verified using MATLAB..

VIVA VOICE:

1. An audio signal consists of frequencies in the range of 100Hz to 5.5KHz.What is the minimum frequency at which it should be sampled in order to transmit it through pulse modulation?
2. Draw a TDM signal which is handling three different signals using PWM
3. What do you infer from the frequency spectrum of a PWM signal?

4. Clock frequency in a PWM system is 2.5 kHz and modulating signal frequency is 500Hzhowmany pulses per cycle of signal occur in PWM output? Draw the PWM signal
5. Why should the curve for pulse width Vs modulating voltage be linear
6. What is the other name for PWM?
7. What is the disadvantage of PWM?
8. Will PWM work if the synchronization between Tx and Rx fails

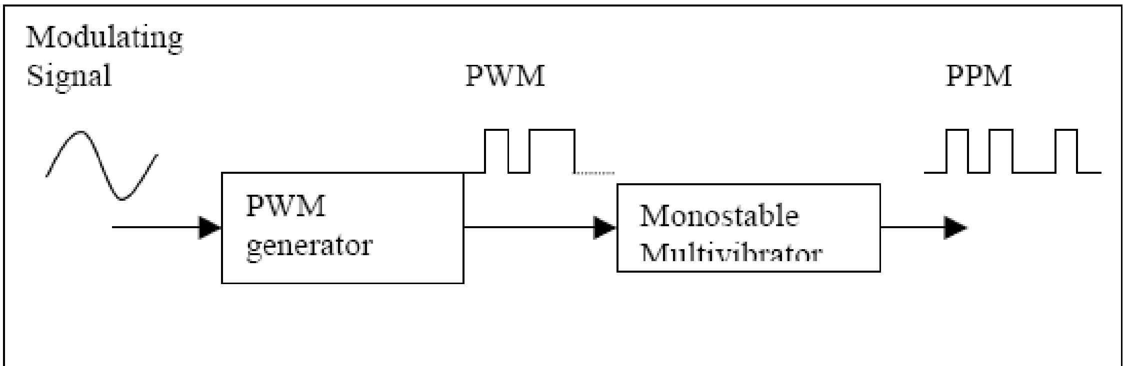
10. PULSE POSITION MODULATION & DEMODULATION

AIM:
To study and generate the waveforms of PPM Modulation and demodulation and simulated using matlab..

APPARATUS:
1. PPM trainer kit
2.Dual trace Oscilloscope
3.Patch chords

I)GENERATION OF PULSE POSITION MODULATION

THEORY:
Pulse position Modulation is another type of modulation, where the position of the pulses varies in accordance of the amplitude of the modulating signal. Pulse Position Modulation (PPM) signals can be readily generated from the PWM signals by using the modulated edge of the PWM pulse to trigger a mono-stable multi-vibrator circuit, which generates fixed width, fixed amplitude pulses when triggered. The figure below shows the block diagram of a PPM generator.



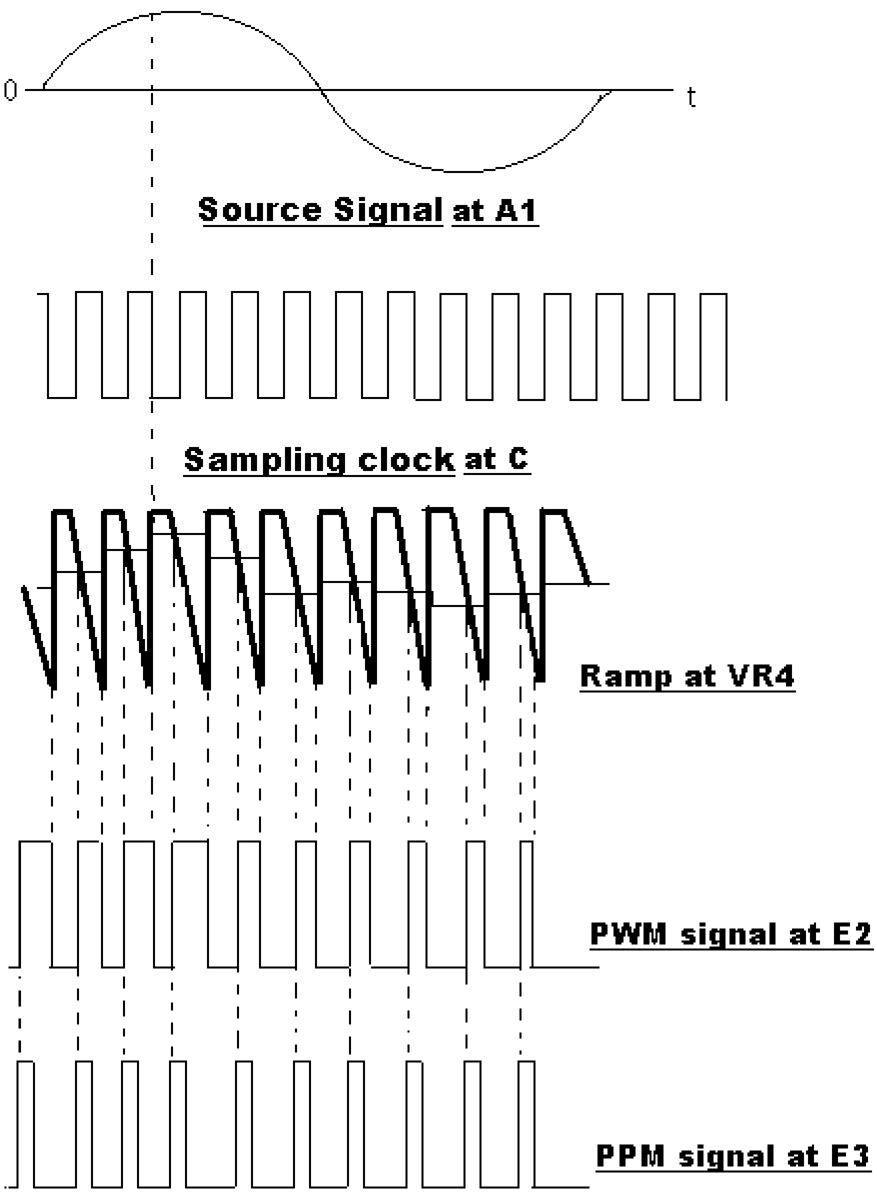
PPM Generator

The PWM signals generated are applied as trigger pulses to a Mono-stable Multi-vibrator.The Mono-stable Multi-vibrator triggers each time at the positive going edge of the PWM signals thereby generating the required PPM signals of constant width and amplitude.

EXPERIMENTAL PROCEDURE:
- Connect signal source 1 KHz (A1) to (A).
- Select sampling frequency to 8 KHz.
- Select flat sampling.
- Repeat the procedure of PWM Modulation to derive proper PWM.

EXPECTED WAVEFORMS

The waveform above illustrates the generation of PPM signals.



OBSERVATIONS:

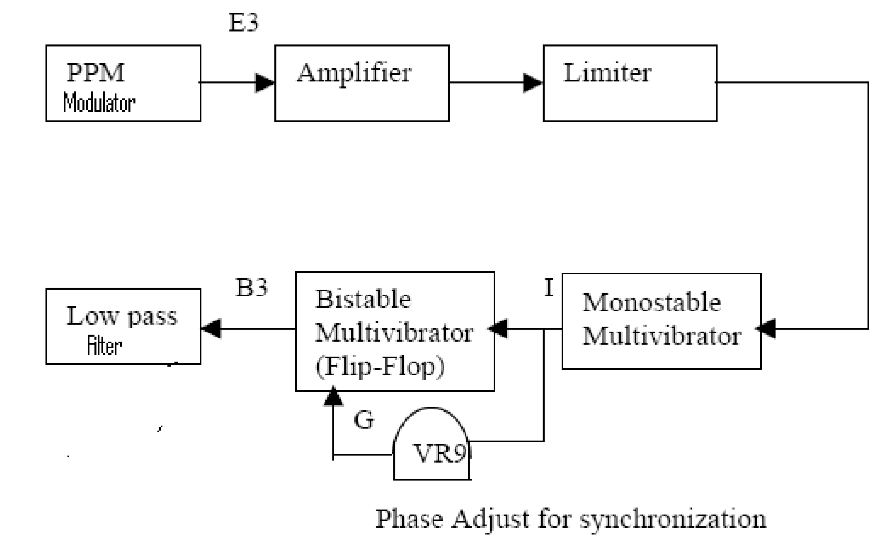
- Observe the waveform of input analog signal source A1 along with PPM modulator output E3.
- In PPM, pulse position Modulation, the position of pulse is proportional to the modulating analog signal amplitude.
- Observe the waveform of PWM modulator output E2 along with PPM modulator output E3.
- It can be compared that, in PWM, long pulses expend considerable power during the pulse while bearing no additional information. If this unused power is subtracted from PWM, so that only time transitions are preserved, we obtain a more efficient pulse position modulation (PPM).
- Select other sampling frequency, repeat procedure 1 to 3
- Study the PPM pattern by varying the amplitude of analog signal. Explain the PPM pattern of minimum analog signal.

- Repeat the above procedure at 2 KHz and (0.5 + 1 + 2) KHz input analog signal.

II)DEMODULATION TECHNIQUES OF PPM SIGNALS

THEORY:

For demodulation of PPM, the direct demodulation method as PWM can be used. But, the demodulated analog signal shows a very low amplitude, for PPM pulses are very narrow and much spaced out. Converting the PPM signal into a PWM, with subsequent filtering through a low pass filter performs a more effective PPM demodulation. The figure below shows the block diagram of a PPM Modulator.



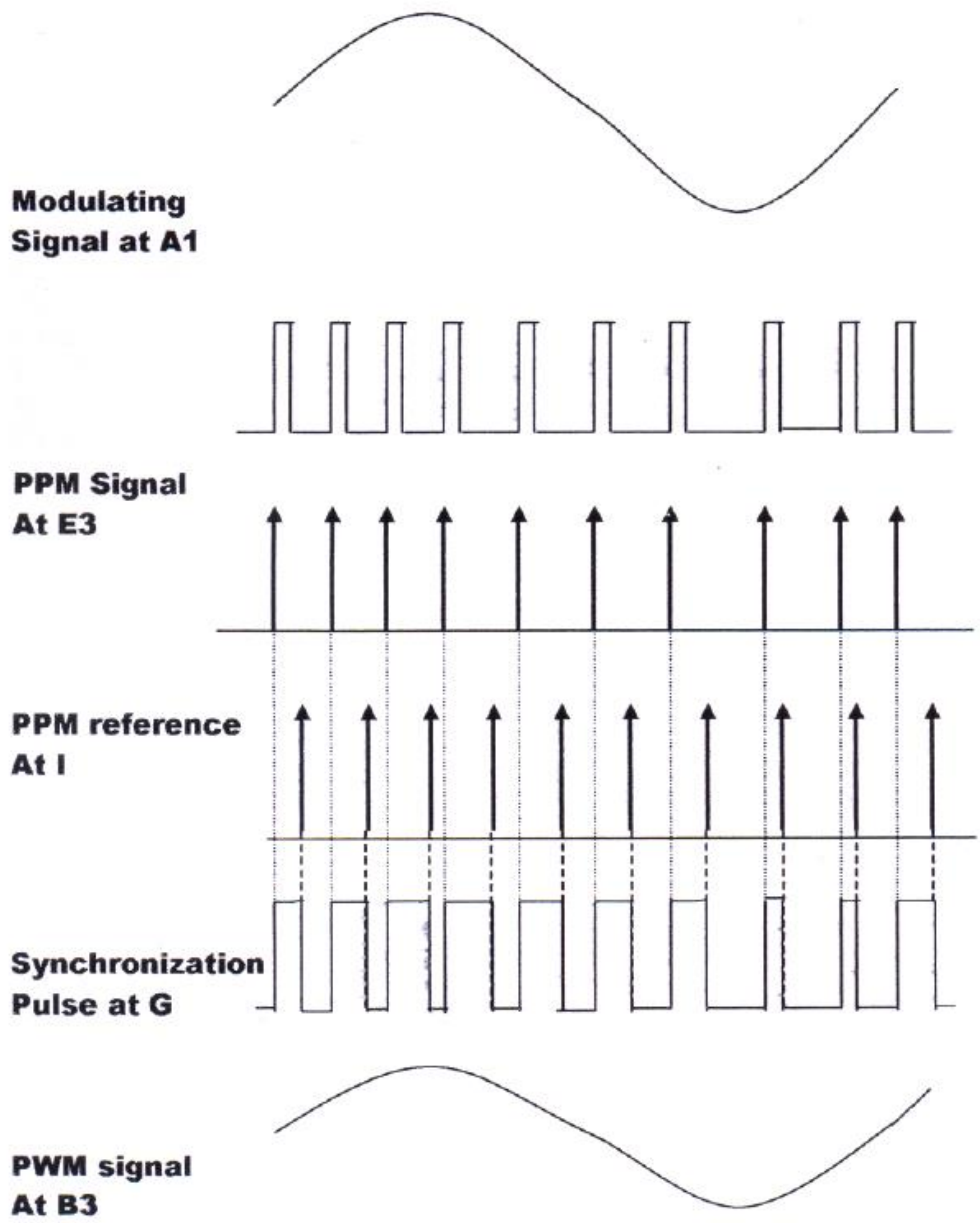
PPM Modulator

Since the PPM pulses are very narrow, they are first amplified and then fed to a limiter.From the limiter PPM output, a Mono stable Multi vibrator on the leading edges of the PPM signals generates a fixed-period reference pulse (I).

EXPERIMENTAL PROCEDURE:

- Connect signal source 1 KHz (A1) to (A), PPM output (E3) to receiver input (E) and PPM demodulator output (B3) to LPF input (B)
- Select sampling frequency to 8 KHz
- Select flat sampling.

EXPECTED WAVEFORMS



OBSERVATIONS:

- Repeat the procedure of PPM Modulation, to derive proper PPM.
- Observe the waveform of PPM demodulator output B3 along with LPF output F1.
- Gradually vary the phase adjust potentiometer to the proper reconstruction of analog signal. Fine tune RAMP generator, if required for finer reconstructed signal.
- Establish the necessary condition for reconstruction.
- In a PPM demodulator output the PWM, the adjacent pulse widths on either side of
- any reference pulse will have gradual width variation from width of reference pulse in

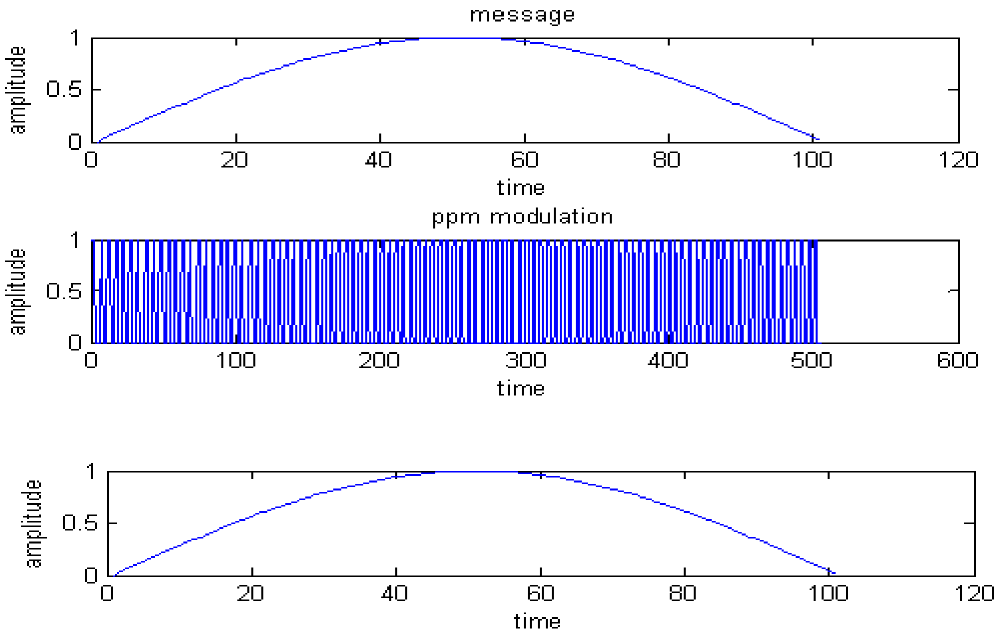
the same order of a variation of input sine wave.

- There have to be definite width during –ve peak of input analog sine wave.
- Observe the reconstruction at F1 by varying the amplitude.
- Repeat the above procedure at 8 KHz and by feeding 2 KHz and (0.5 + 1 + 2) KHz analog signal.

MATLAB CODE:

```
close all;
clear all;
clc;
fs=50;
t=[0:2*fs]'/fs
fc=10;
x=sin(pi/4*2*t);
s=modulate(x,fc,fs,'ppm',0.5);
figure
subplot(3,1,1);
plot(x);
title('message');xlabel('time');ylabel('amplitude');
subplot(3,1,2);
plot(s);
title('ppm modulation');xlabel('time');ylabel('amplitude');
z=demod(s,fc,fs,'ppm',0.5)'
subplot(3,1,3);
plot(z);
title('ppm demodulation');xlabel('time');ylabel('amplitude');
```

WAVEFORM:



PPM demodulation

RESULT:

Generation and demodulation of PPM signals have been observed and studied using matlab..

VIVA VOICE:

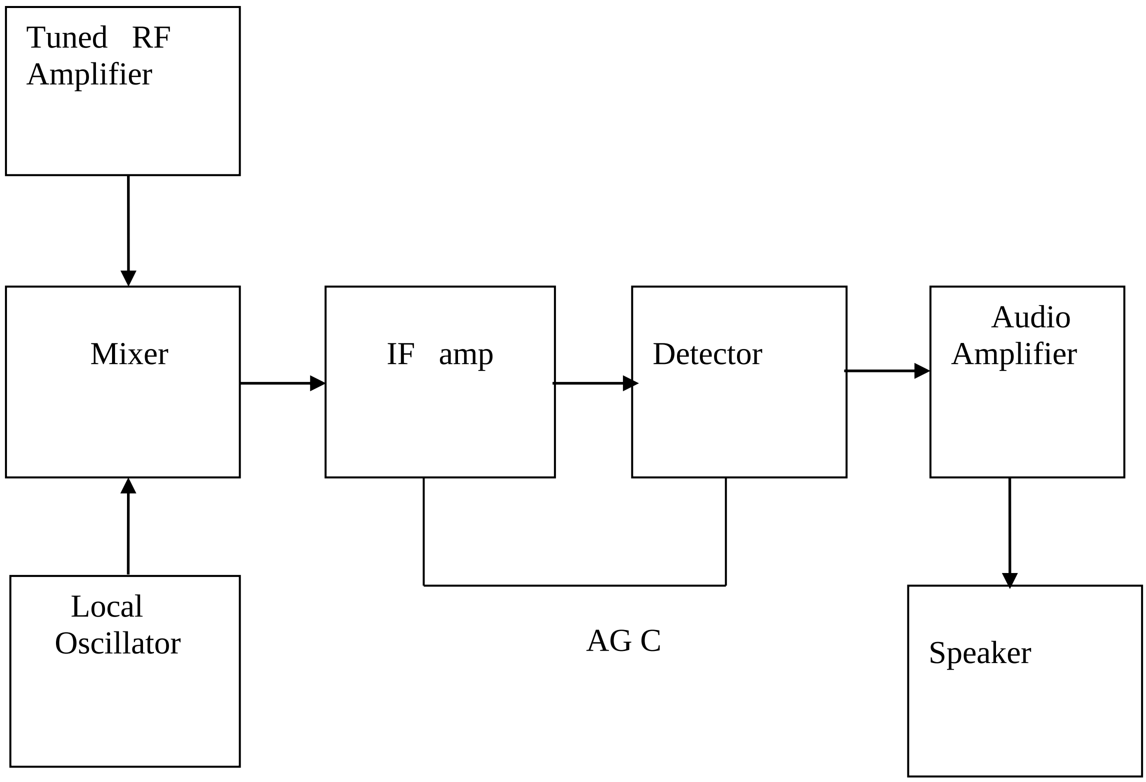
1. What is the advantage of PPM over PWM?
2. Is the synchronization is must between Tx and Rx
3. Shift in the position of each pulse of PPM depends on what?
4. Can we generate PWM from PPM?
5. Why do we need 555 timer?
6. Does PPM contain derivative of modulating signal compared to PWM?

11. AGC CHARACTERISTICS

AIM:
To obtain the AGC characteristics of a radio receiver

APPARATUS:
1. Radio receiver trainer kit
2. Dual trace CRO
3. Connecting patch cords

CIRCUIT:



THEORY:

The AGC signal is used as a bias signal to reduce the gain of the RF and the IF amplifiers to prevent detector overload a strong signals. AGC is a system beans of which the overall gain of a radio receiver is a varied automatically with the changing strength of a received signal to keep the output substantially constant.

PROCEDURE:

1. Select a carrier frequency of 1000 kHz. AF frequency 1 kHz and apply AM signal to the receiver.
2. connect CRO at the o/p of the audio amplifier.
3. Tune the mixer local oscillator for maximum AF signal o/p at detector o/p and measure the audio signal.
4. Increase the RF level in appropriate steps and note down corresponding o/p AF signal amplitude.
5. plot the AF o/p vs RF i/p on graph.

TABULAR FORM:

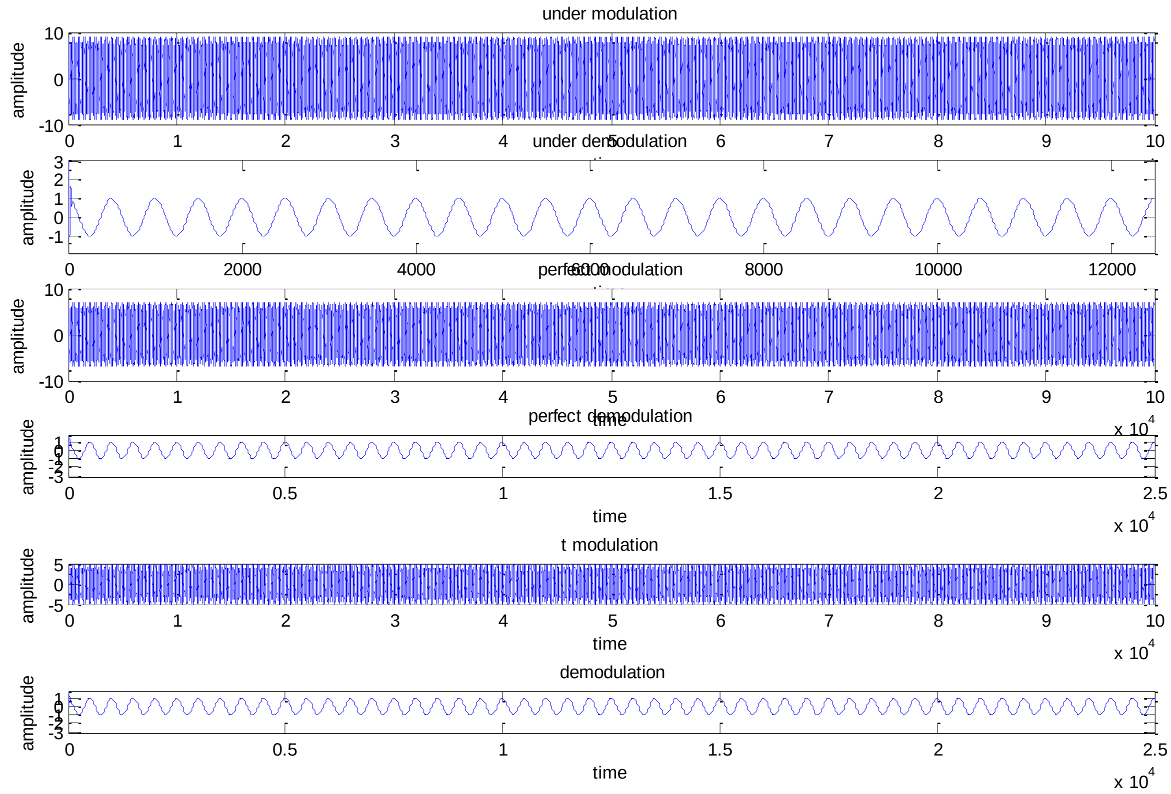
S.No	Voltage of Input Signal	Voltage of Input Signal

EXPECTED WAVEFORM:

MATLAB CODE:

```
close all;
clear all;
clc;
fs=100e3;
t=0:1/fs:1-1/fs;
```

```
am=2;
fm=200;
m=cos(2*pi*fm*t);
fc=3.5e3;
ac=8;
c=ac.*cos(2*pi*fc*t);
figure;
subplot(2,1,1);
plot(c);
title('carrier');
xlabel('time');ylabel('amplitude');
subplot(2,1,2);
plot(m);
title('message');
xlabel('time');ylabel('amplitude');
s=ammod(m,fc,fs,0,ac);
subplot(6,1,1);
plot(s);
title('under modulation');
xlabel('time');ylabel('amplitude');
z=amdemod(s,fc,fs,0,ac);
 subplot(6,1,2);
plot(z);
title('under demodulation');
xlabel('time');ylabel('amplitude');
ac=6;
s=ammod(m,fc,fs,0,ac);
subplot(6,1,3);
plot(s);
title('perfect modulation');
xlabel('time');ylabel('amplitude');
z=amdemod(s,fc,fs,0,ac);
 subplot(6,1,4);
plot(z);
title('perfect demodulation');
xlabel('time');ylabel('amplitude');
ac=4;
s=ammod(m,fc,fs,0,ac);
subplot(6,1,5);
plot(s);
title('t modulation');
xlabel('time');ylabel('amplitude');
z=amdemod(s,fc,fs,0,ac);
subplot(6,1,6);
plot(z);
title(' demodulation');
xlabel('time');ylabel('amplitude');
```



RESULT:

The AGC characteristics are obtained.

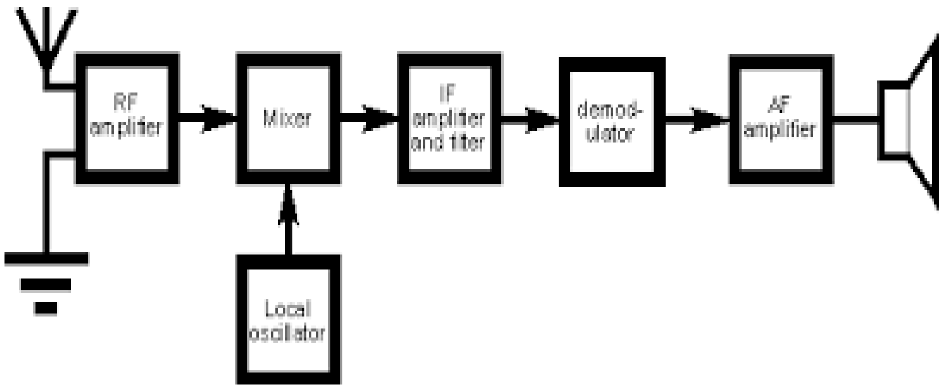
12. RADIO RECEIVER

AIM: To examine the principle diagram of the operation of AM Radio Receiver

APPARATUS:

- Dual-trace Oscilloscope
- Function generator
- Frequency meter.
- Power supply mod.PSI-PSU

CIRCUIT:



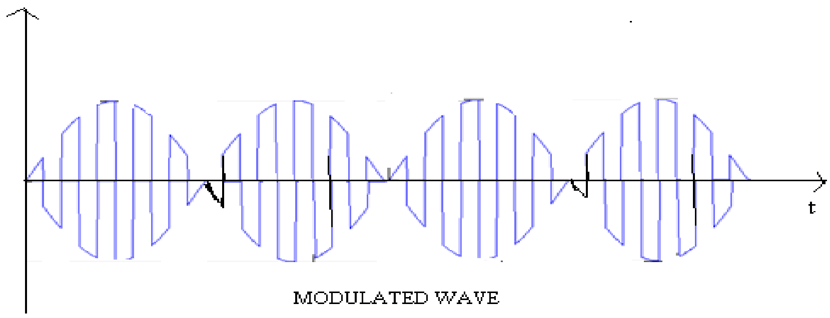
THEORY:

The purpose of receiver is to convert the modulated radio frequency signal is picked up by the space in which it traveled electromagnetic wave and sent through a transmit ion line to the electronic circuit of the receiver in order to be de-modulated . Operation of block of receiver: The filter and RF amplifier remove the channel we do, not want to receive from the useful signal and increases its amplitude level as the RF signal can be different , the input filter must change its characteristics . Typically this occurs automatically without the user intervention by means of D.C control circuits. The frequency converter translates the frequency from RF channel frequency that is to be received to IF . it employs a frequency stabilized oscillator with a PLL circuit The filter and IF amplifier cleans the useful signal from any inter-modulation products or noise and increases its amplitude level. As the IF is always the same the filter does not need regulation calibration and can be a commercial component optimized for this purpose. The demodulator must receive or extract the information contained into the IF signal. The frequency spectrum of the IF signal depends on the kind of used modulation and on the same information the IF frequency is always equal and dose not depend on the frequency of the RF channel that is to be used. This optimizes the modulation and filtering circuits.

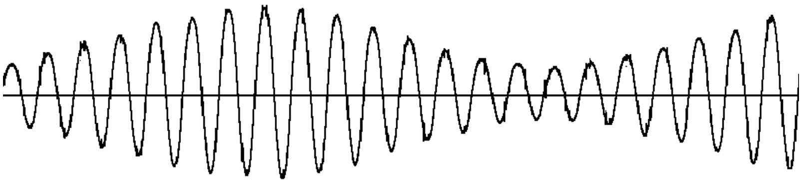
PROCEDURE:

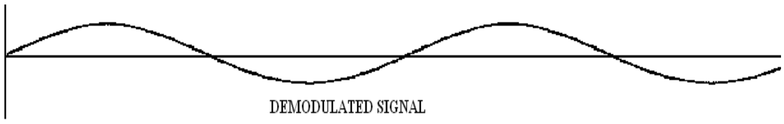
- Set the switch SW1 of the antenna / Cable section to cable.
- Set switch SW2 of the RF filter section to by pass in the AGC / Level meter section .
- In the AGC/LEVEL METER section :Turn switch SW6 off. Turn the D.C source trimmer completely clockwise to obtain the maximum gain of the voltage controlled amplifier.
- In the LOCAL Oscillator ½ section (Local Oscillator 1)
- Set switch SW8 to FM
- Set switch SW10 to PLL to obtain the automatic control of the frequency of the local oscillator (in these condition the frequency is fixed)
- Set switch SW4 of the 10.7MHz if filter section of CERAMIC .
- Set switch SW7 of the IF amplifier / FM demodulator section to FM/SSB(to enable the operation of the local oscillator 1)
- Insert a sine signal to the input CABLE in with amplitude of 1 Vpp and frequency of 1 MHz using an external generator.
- Connect the Oscilloscope and check the presence and amplitude of the signal in the following test point.

EXPECTED WAVE FORMS:



MODULATED WAVE





DEMODULATED SIGNAL

RESULT: The Radio receiver characteristics are obtained.