Madhav Institute of Technology and Science, Gwalior

(A Govt. Aided UGC Autonomous & NAAC Accredited Institute Affiliated to R.G.P.V., Bhopal, M.P.)

Department Of Electronics Engineering

Lab Manual

Analog Communication

(140413/200413)

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1	To generate amplitude modulated wave and determine thepercentage modulation
2	To generate amplitude demodulated wave and determine thepercentage modulation.
3	To generate AM-Double Side Band Suppressed Carrier (DSB-SC) signal
4	To generate SSB-SC -Double Side Band Suppressed Carrier(DSB-SC) signal
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6	Verify the generation and detection of DSB-SC Signal usingMATLAB
7	Verify the generation and detection SSB-SC signal usingMATLAB
8	Verify the generation and detection of FM Signal usingMATLAB.

Analog Communication
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Course Outcomes

After completing the lab, students will be able to -

CO1	**Differentiate** modulation and demodulation techniques.
CO2	**Calculate** the modulation index for a given modulated wave.
CO3	**Generate** AM, DSB, SSB and FM signals.

MADHAV INSTITUTE OF TECHNOLOGY AND SCIENCE, GWALIOR

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Department Of Electronics Engineering

Analog Communication Lab

Experiment No...................

Date....................................

Co-workers:- 1..

2...

3..

4...

Title:-

....

EXPERIMENT No.1
AMPLITUDE MODULATION

AIM:- To generate Amplitude Modulated (AM) wave and calculate the modulation index.

APPARATUS REQUIRED:- (i) C.R.O. (ii) CRO Probe (ii) DSB/SSB Transmitter (ST 2201) Trainer kit (iv) Connecting leads.

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

$$s(t) = A_c\left[1 + K_a m(t)\cos(2\pi f_c t)\right] \quad \ldots\ldots\ldots(1)$$

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 the envelope 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,

$$m = \frac{(E_{max} - E_{min})}{(E_{max} + E_{min})} \quad \ldots\ldots\ldots..(2)$$

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

PROCEDURE:-

1. Switch on the trainer kit and ensure that audio input select switch should be in INT position and mode switch in DSB position.

2. Turn the balance pot, in the balanced modulator & band pass filter circuit 1 block, to its fully clockwise position. It is this block that we will use to perform *amplitude modulation.*

3. Now, monitor the two inputs to the balanced modulator & band pass filter circuits 1 block, at TP1 and TP9. Signal at TP1 is audio frequency sine wave which is modulating input and TP9 carries a sine wave of 1MHz and 120mVpp approx. This is the carrier input.

4. Next, examine the output of balanced modulator & band pass filter circuit 1 block at TP3, along with modulating signal at TP1. The percentage modulation can be calculated using the formula.

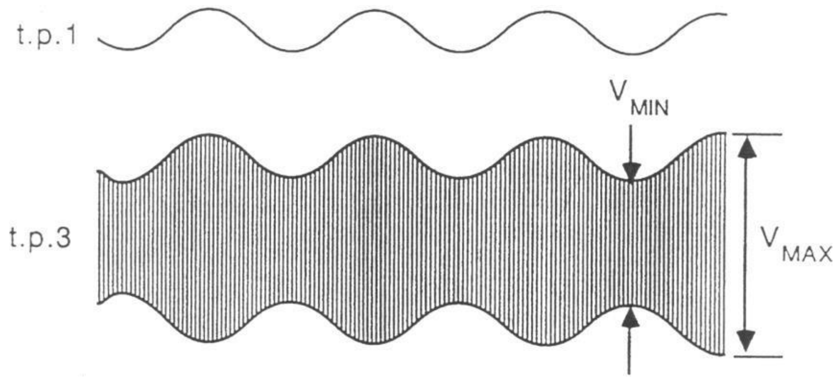


Figure 1 Modulating and AM wave

The output from the balanced modulator & band pass filter circuit 1 block (at TP3) is AM waveform, which has been formed by amplitude-modulating the 1MHz carrier sinewave with the audio-frequency sinewave from the audio oscillator.

5. To determine the depth of modulation, measure the maximum amplitude (Vmax) and the minimum amplitude (Vmin) of the AM waveform at TP3, and use the following formula:

$$\mu = \frac{Vmax - Vmin}{Vmax + Vmin}$$

Where Vmax and Vmin are the maximum and minimum amplitudes of AM wave.

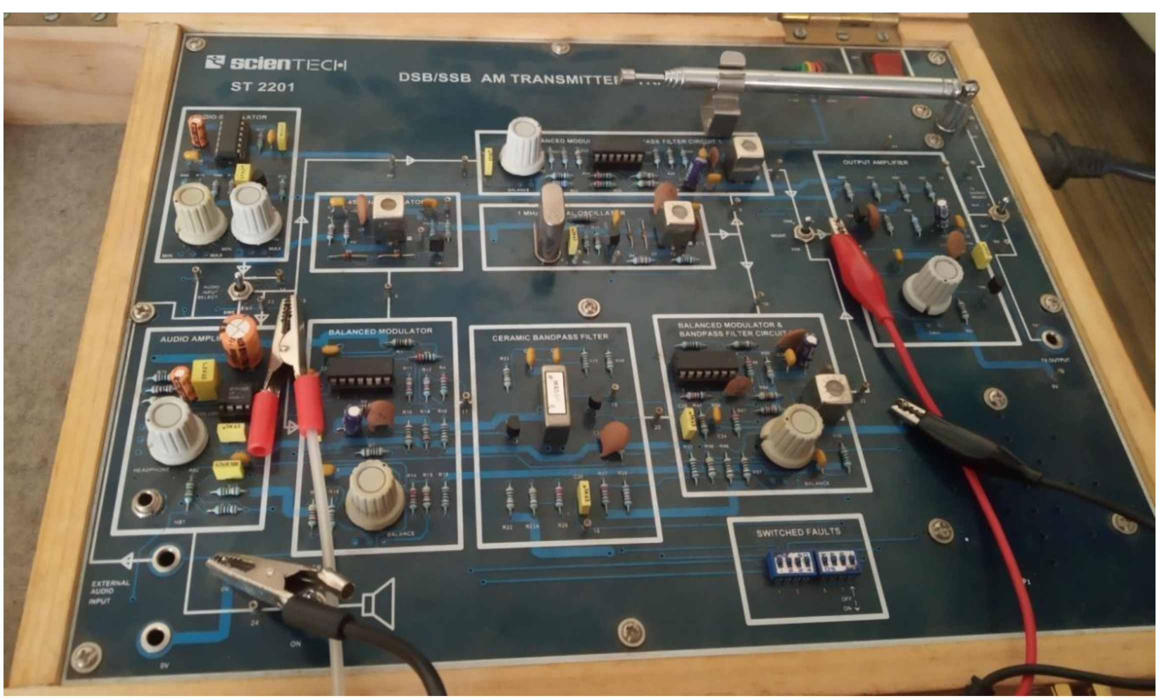


Figure 2 : Trainer kit connections

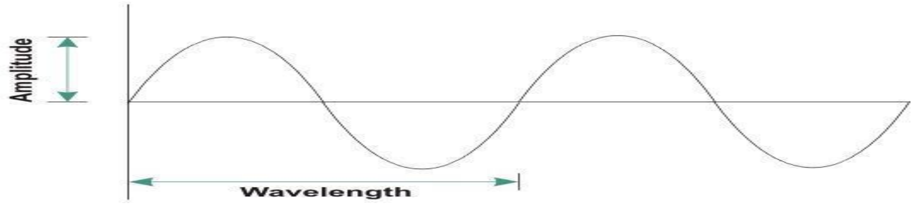


Figure 3: Message signal

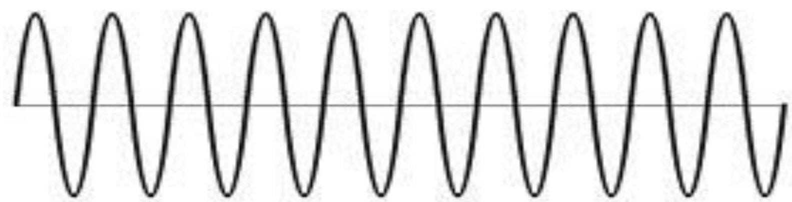


Figure 4: Carrier signal

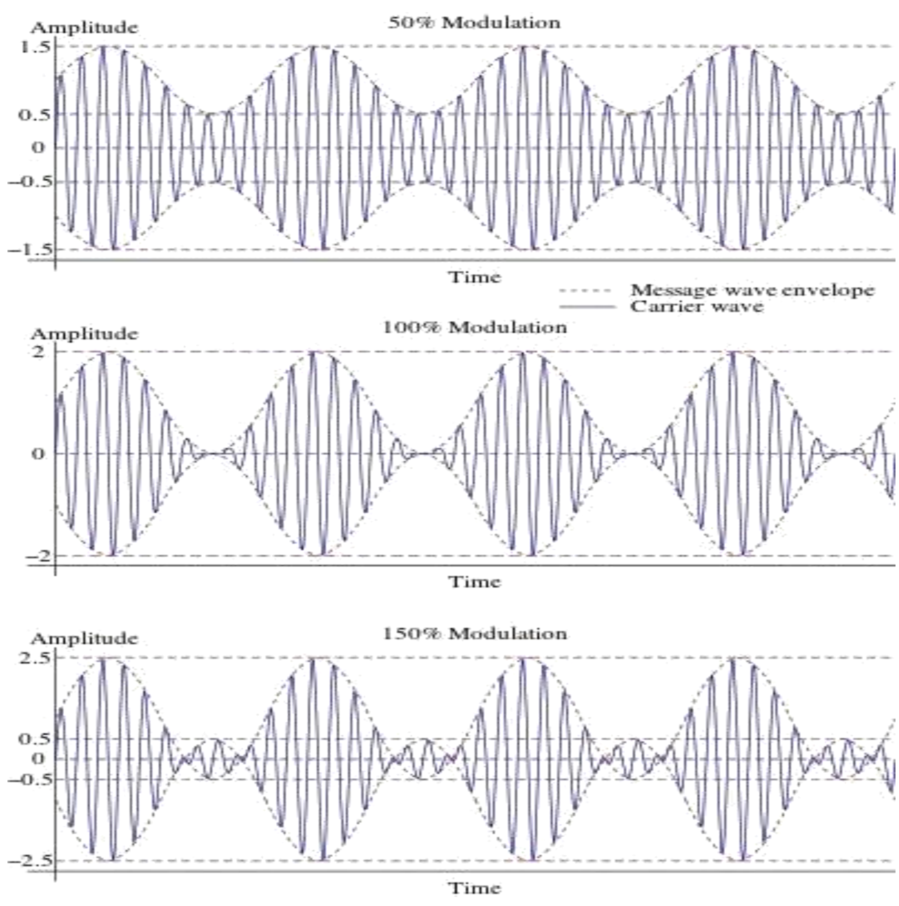


Figure 5: Types of Amplitude modulated waveforms

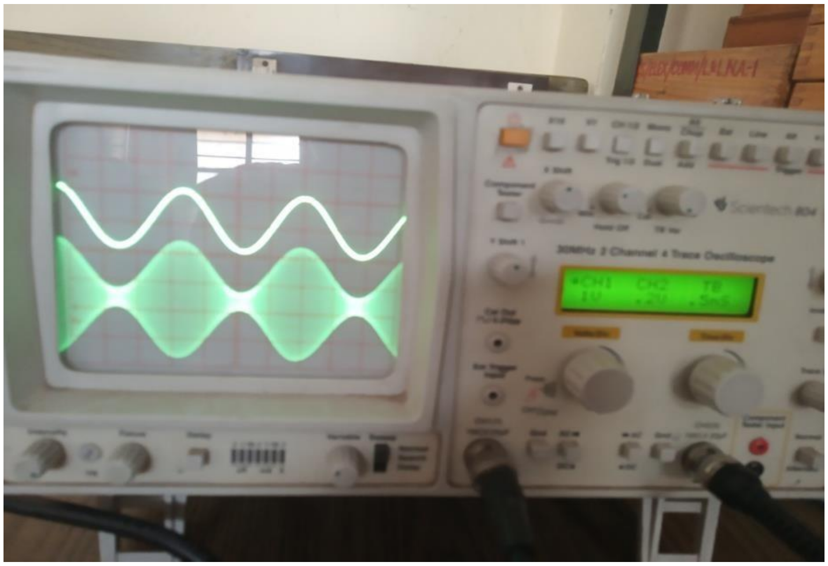


Figure 6 : Observed waveform

Observations:

Amplitude of modulating signal____

Frequency of modulating signal____

Amplitude of carrier signal____

Frequency of carrier signal____

Amplitude of demodulating signal____

Frequency of demodulating signal____

Calculation:

S. No	Vmax (Volts)	Vmin (Volts)	Theoritical μ= Vm/Vc	μ = Vmax-Vmin / Vmax+ Vmin

RESULT:- The amplitude modulated wave has been generated and the modulation index is calculated as……………

PRECAUTIONS:-

1. Do not use open ended wires for connecting to 230 V power supply.
2. Before connecting the power supply plug into socket, ensure power supply should be switched off
3. Ensure all connections should be tight before switching on the power supply.
4. Take the reading carefully.
5. Power supply should be switched off after completion of experiment.

LAB QUESTIONS:-

Where the modulation index lies?
What is the range of audio frequencies?
Why modulation is an essential process of communication system?
Define Amplitude modulation draw its spectrum.
Explain need for modulation.
Q.6 How carrier is differing from message?
Q.7. Explain how negative peak clipping occurs in the demodulated signal when diode detector is used?
Q.8.Explain under modulation, 100% modulation, over modulation
Give the significance of modulation index.

EXPERIMENT No.2
AMPLITUDE DEMODULATION

AIM:- To demodulate the AM signal and observe peak diagonal clipping effect.

APPARATUS REQUIRED:- (i) C.R.O. (ii) CRO Probe (ii) DSB/SSB Transmitter (ST 2201) and Receiver Trainer (ST 2202) (iv) Connecting leads

THEORY:-

The AM Transmitter:
The transmitter circuits produce the amplitude modulated signals which are used to carry information over the transmission to the receiver. The main parts of the transmitter are shown in Fig.1. In Fig.2, we can see that the peak-to-peak voltage in the AM waveform increase and decrease in sympathy with the audio signal.

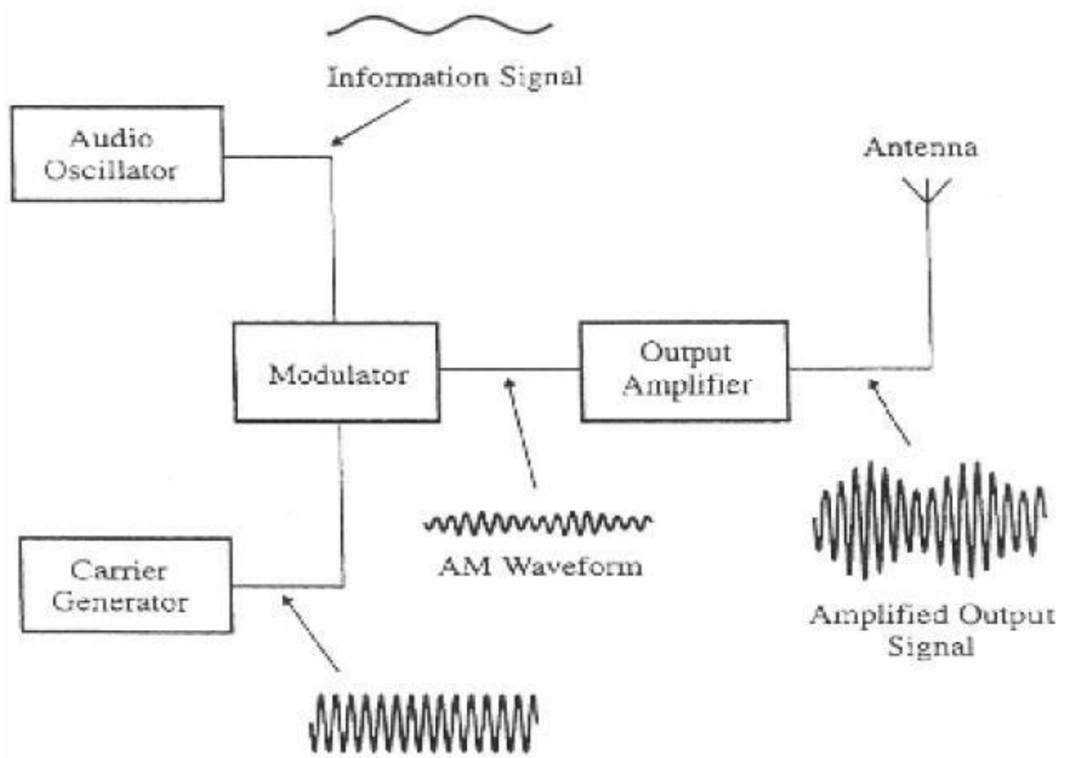


Fig. 1: AM Transmitter System

To emphasize the connection between the information and the final waveform, a line is sometimes drawn to follow the peaks of the carrier wave as shown in Fig.2. This shape, enclosed by a dashed line in out diagram, is referred to as an 'envelope', or a
'modulation envelope'. It is important to appreciate that it is only a guide to emphasize of the AM waveform.

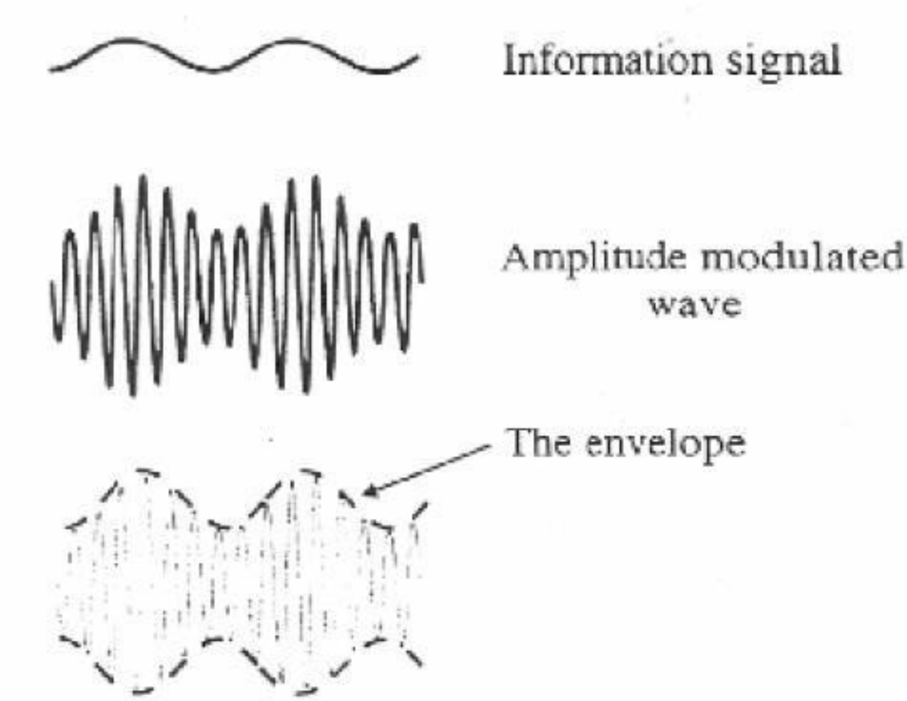
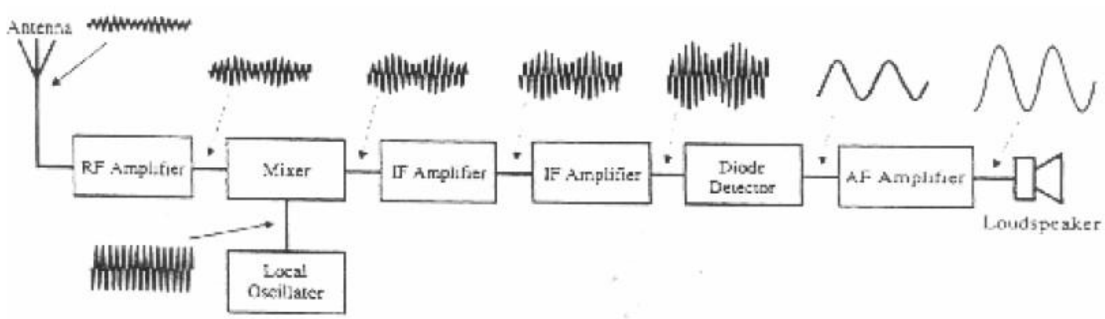


Figure 2: Waveforms in AM transmitter

AM Reception: The 'em' wave from the transmitting antenna will travel to the receiving antenna carrying the information with it. The stages of AM reception are shown in Fig. 3. :

Figure 3: AM Reception



Envelope Detector:

The simplest form of envelope detector is diode detector. The function of the diode detector is to extract the audio signal from the signal at the output of the IF amplifiers. It performs this task in a very similar way to a half wave rectifier converting an AC input to a DC output. Fig.4 shows a simple circuit diagram of the diode detector.

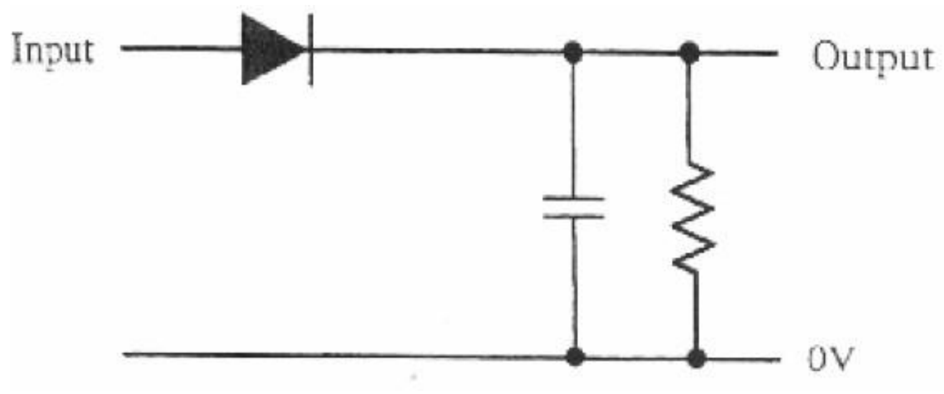


Figure 4: Diode Detector

In Fig.4, the diode conducts every time the input signal applied to its anode is more positive than the voltage on the top plate of the capacitor.

When the voltage falls below the capacitor voltage, the diode ceases to conduct and the voltage across the capacitor leaks away until the next time the input signal is able to switch it on again. See fig. 5

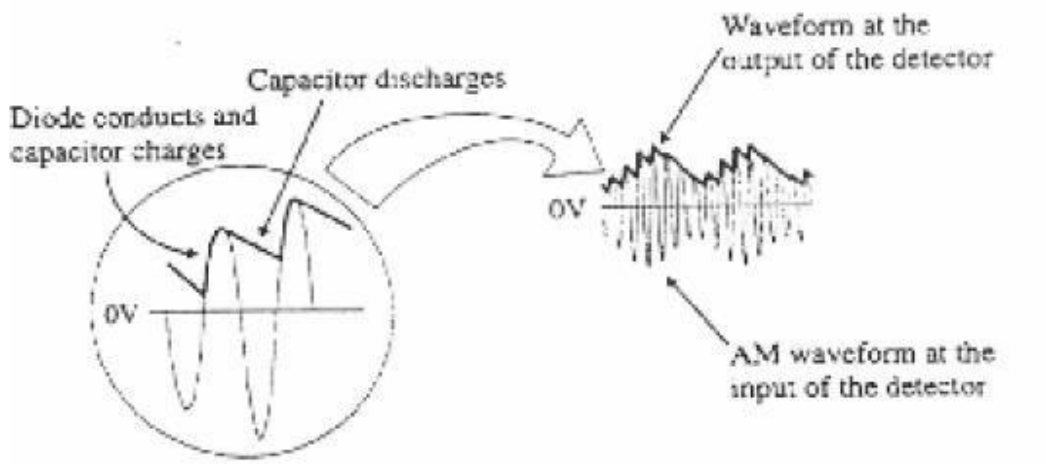


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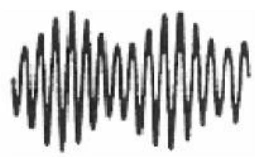
Fig. 5 Clipping in Diode Detector

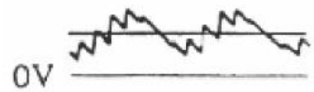
The result is an output which contains three components :

1. The wanted audio information signal.
2. Some ripple at the IF frequency.
3. A positive DC voltage level.

At the input to the audio amplifier, a low pass filter is used to remove the IF ripple and a capacitor blocks the DC voltage level. Fig.6 shows the result of the information signal passing through the diode detector and audio amplifier. The remaining audio signals are then amplified to provide the final output to the loudspeaker.



The input to the diode detector from the last IF amplifier



Output of diode detector includes : a DC level, the audio signal, ripple at IF frequency

Output after filtering

Figure 6: Output of Diode Detector and output Filter

<u>PROCEDURE:</u>-

1. Position the **ST2201** & **ST2202** modules, with the **ST2201** board on the left, and a gap of about three inches between them.
2. Ensure that the following initial conditions exist on the **ST2201** board.
   a. Audio oscillator's amplitude pot in fully clockwise position.
   b. Audio input select switch in INT position.
   c. Balance pot in balanced modulator & band pass filter circuit 1 block, in full clockwise position;
   d. Mode switch in DSB position.
   e. Output amplifier's gain pot in full counter-clockwise position.
   f. TX output select switch in ANT position:
   g. Audio amplifier's volume pot in fully counter-clockwise position.
   h. Speaker switch in ON position.
   i. On-board antenna in vertical position, and fully extended.
3. Ensure that the following initial conditions exist on the **ST2102** board:
   a. RX input select switch in ANT position.
   b. R.F. amplifier's tuned circuit select switch in INT position.
   c. R.E amplifier's gain pot in fully clock-wise position;
   d. AGC switch in INT position.
   e. Detector switch in diode position.
   f. Audio amplifier's volume pot in fully counter-clockwise position.
   g. Speaker switch in ON position.
   h. Beat frequency oscillator switch in OFF position.
   i. On-board antenna in vertical position, and fully extended.
4. Turn on power to the modules.
5. On the **ST2202** module, slowly turn the audio amplifier's volume pot clockwise, until sounds can be heard from the on-board loudspeaker. Next, turn the vernier tuning dial until a broad cast station can be heard clearly, and adjust the volume control to a comfortable level.

6. The first stage or 'front end' of the **ST2202** AM receiver is the R.F amplifier stage. This is a wide - bandwidth tuned amplifier stage, which is tuned into the wanted station by means of the tuning dial. Examine the envelope of the signal at the R.F. amplifier's output (at t.p. 12), with an a.c. - coupled oscilloscope channel. Note that:

   a. The amplifier's output signal is very small in amplitude (a few tens of millivolts at the most). This is because one stage of amplification is not sufficient to bring the signal's amplitude up to a reasonable level.

   b. Only a very small amount of amplitude modulation can be detected, if any.

7. The next stage of the receiver is the mixer stage, which mixes the R.F. amplifier's output with the output of a local oscillator. The Frequency of the local oscillator is also tuned by means of the tuning dial, and is arranged so that its frequency is always 455 KHz above the signal frequency that the R.F. amplifier is tuned to. This fixed frequency difference is always present, irrespective of the position of the tuning dial, and is arranged so that its frequency is always 455 KHz above the signal frequency that the
R.F. amplifier is tuned to. Examine the output of the local oscillator block, and check that its frequency varies as the tuning dial is turned.

8. The operation of the mixer stage is basically to shift the wanted signal down to the IF frequency, irrespective of the position of the tuning dial. The end result of this process is that the carrier frequency of the selected AM station is shifted down to 455 KHz (the IF Frequency), and the sidebands of the AM signal are now either side of 455 KHz.

9. Examine the output of the mixer block (t.p.20) with an a.c. coupled oscilloscope channel, and note that main frequency component present changes as the tuning dial is turned.

10. These IF amplifiers are basically tuned amplifiers which have been pre- tuned to the IF frequency-they have a bandwidth just wide enough to amplify the 455 KHz carrier and the AM sidebands either side of it. Any frequencies outside this narrow frequency band will not be amplified. Examine the output of IF amplifier 1 (at. t.p. 24) with an a.c.-coupled oscilloscope channel, and note that:

   a. The overall amplitude of the signal is much larger than the signal amplitude at the mixer's output, indicating that voltage amplification has occurred.

   b. The dominant component of the signal is now at 455 KHz, irrespective of any particular station you have tuned into. This implies that the wanted signal, at the IF frequency, has been amplified to a level where it dominates over the unwanted components.

   c. The envelope of the signal is modulated in amplitude, according to the sound information being transmitted by the station you have tuned into.

11. Examine the output of IF amplifier 2 (t.p.28) with an a.c.-coupled oscilloscope channel, noting that the amplitude of the signal has been further amplified by this second IF amplitude of the signal has been further amplified by this second IF amplifier stage. IF amplifier 2 has once again preferentially amplified signals around the IF frequency (455 KHz), so that:

   a. The unwanted local oscillator and sum components from the mixer are now so small in comparison, that they can be ignored totally,

   b. Frequencies close to the I F frequency, which are due to stations close to the wanted station, are also strongly attenuated.

The resulting signal at the output of IF amplifier 2 (t.p.28) is therefore composed almost entirely of a 455 KHz carrier, and the A.M. sidebands either side of it carrying the wanted audio information.

12. The next step is extract this audio information from the amplitude variations of the signal at the output of IF amplifier 2. This operation is performed by the diode detector block, whose output follows the changes in the amplitude of the signal at its input. To see how this works, examine the output of the diode detector block (t.p.31), together with the output from. IF amplifier 2 (at t.p.28). Note that the signal at the diode detector's output:

· Follows the amplitude variations of the incoming signal as required:

· Contains some ripple at the IF frequency of 455 KHz, and

· The signal has a positive DC offset, equal to half the average peak to peak amplitude of the incoming signal.

13. The final stage of the receiver is the audio amplifier block contains a simple low- pass filter which passes only audio frequencies, and removes the high frequency ripple from the diode detector's output signal. This filtered audio signal is applied to the input of an audio power amplifier, which drives on board loudspeaker (and the headphones, if these are used). The final result is the sound you are listening to the audio signal which drives the loudspeaker can be monitored at t.p. 39 (providing that the audio amplifier block's volume pot is not in its minimum volume position). Compare this signal with that at the diode detector's output (t.p. 31), and note how the audio amplifier block's low pass filter has 'cleaned up' the audio signal.

14. Presently, the gain of **ST2201'**s output amplifier block is zero, so that there is no output from the Transmitter. Now turn the gain pot in **ST2201**'s output amplifier block to its fully clockwise (maximum gain) position, so that the transmitter generates an AM signal. On the **ST2201** module, examine the transmitter's output signal (t.p.13), together with the audio modulating signal (t.p.1), triggering the 'scope with the signal'. Since **ST2201** TX output select switch is in the ANT position, the AM signal at t.p.13 is fed to the transmitter's antenna. Prove this by touching **ST2201**'s antenna, and nothing that the loading caused by your hand reduces the amplitude of the AM waveform. at t.p.13. The antenna will propagate this AM signal over a maximum distance of about 1.4 feet. We will now attempt to receive the propagated AM waveform with the **ST2201/ ST2202** board, by using the receiver's on board antenna.

15. On the **ST2201** module, turn the volume pot (in the audio amplifier block) clockwise, until you can hear the tone of the audio oscillator’s output signal, from the loudspeaker on the board.

16. On the **ST2201/ST2202** receiver, adjust the volume pot so that the receiver's output can be clearly heard. Then adjust the receiver's tuning dial until the tone generated at the transmitter is also clearly audible at the receiver.

17. In the output of diode detector peak diagonal clipping can be observed at low values of time constant of tuning circuitry.

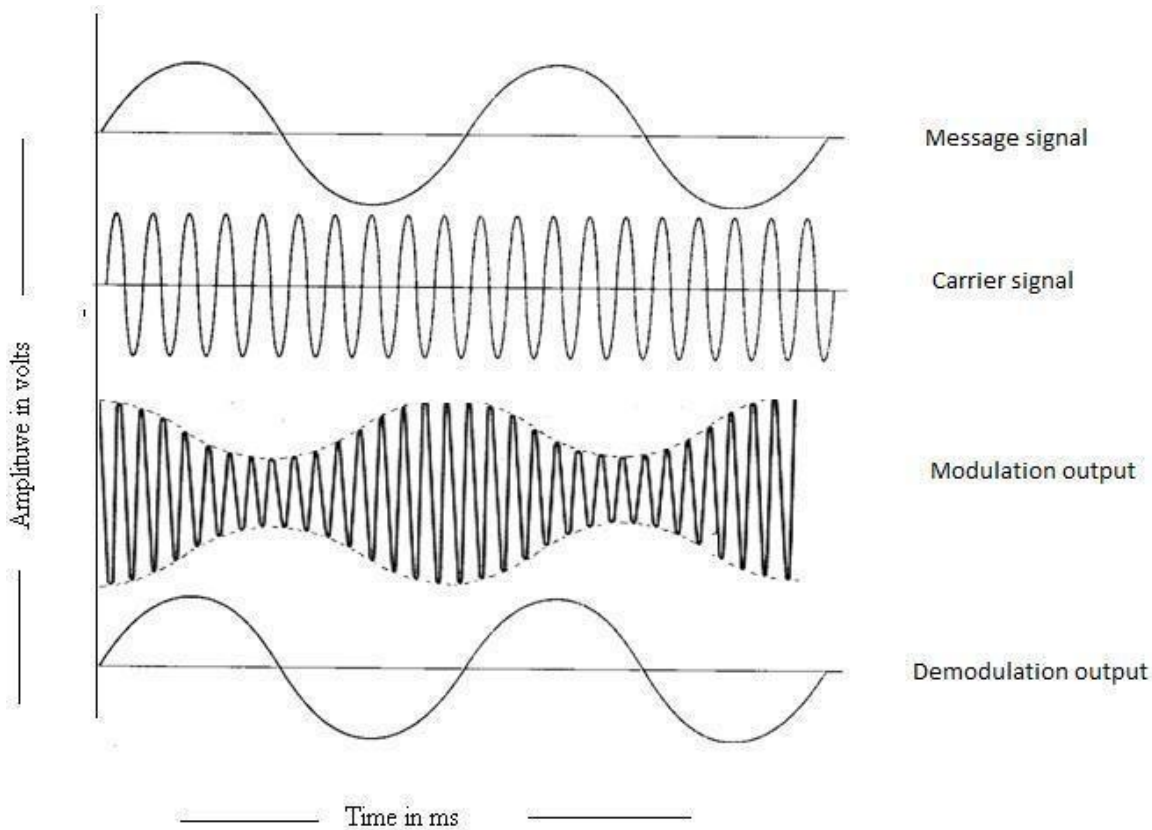


Figure 7: Expected waveforms

RESULT:-

AM signal has been demodulated using envelope detector and peak diagonal clipping effect has been observed.

PRECAUTIONS:-

1. Do not use open ended wires for connecting 230 V power supply.
2. Before connecting the power supply plug into socket, ensure power supply should be switched off.
3. Ensure all connections should be tight before switching on the power supply.
4. Take the reading carefully.
5. Power supply should be switched off after completion of experiment.

LAB QUESTIONS:-

Q. 1. What is amplitude modulation?
Q. 2. Which are the three discrete frequencies in AM?
Q. 3 How many sidebands in AM?
Q. 4. Which circuit is used as LPF?
Q. 5. Which are the two methods of AM generation?
What is diagonal clipping?
What is the unit of modulation index in AM?
Q. 8. Where the modulation index lies?
Q. 9. What happens in case of over modulation?
Q. 10. How DSBSC can be converted into conventional AM?

EXPERIMENT No.3
DSB-SC GENERATION

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

APPARATUS REQUIRED:- (i) C.R.O. (ii) CRO Probe (ii) DSB/SSB Transmitter (ST 2201) Trainer kit (iv) Connecting leads.

THEORY:-

A double sideband suppressed carrier signal, or DSBSC, is defined as the modulating signal and the carrier wave.

$$DSBSC = E.\cos\mu t \,.\, \cos\omega t \qquad (1)$$

Generally, and in the context of this experiment, it is understood that: $\omega >> \mu$ (2) Equation (3) can be expanded to give:

$$\cos\mu t \,.\, \cos\omega t = (E/2)\cos(\omega - \mu)t + (E/2)\cos(\omega + \mu)t \qquad (3)$$

Equation (3) shows that the product is represented by two new signals, one on the sum frequency ($\omega + \mu$), and one on the difference frequency ($\omega - \mu$) – see Figure 1.

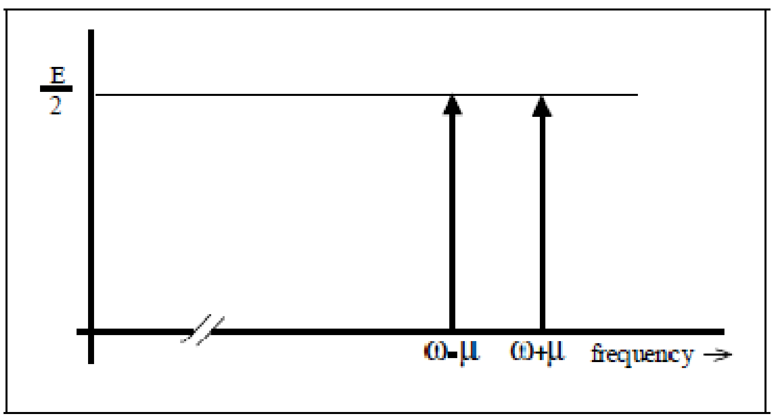


Figure 1: Spectral components

Remembering the inequality of eqn. (2) the two new components are located close to the frequency ω rad/s, one just below, and the other just above it. These are referred to as the lower and upper sidebands respectively.

These two components were derived from a ‘carrier’ term on ω rad/s, and a message on μ rad/s. Because there is no term at carrier frequency in the product signal it is described is a double sideband *suppressed* carrier (DSBSC) signal.

The term ‘carrier’ comes from the context of ‘double sideband amplitude modulation’ (commonly abbreviated to just AM).

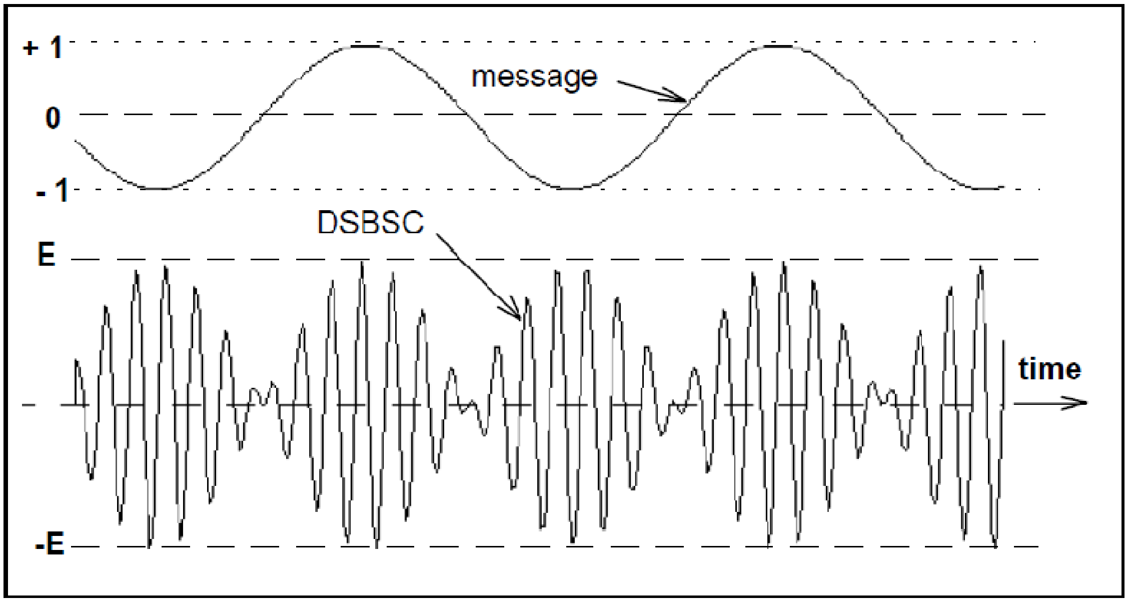


Figure 2: DSBSC – seen in the time domain

Notice the waveform of the DSBSC in Figure 2, especially near the times when the message amplitude is zero. The fine detail differs from period to period of the message. This is because the ratio of the two frequencies μ and ω has been made non-integral. Although the message and the carrier are periodic waveforms (sinusoids), the DSBSC itself need not necessarily be periodic.

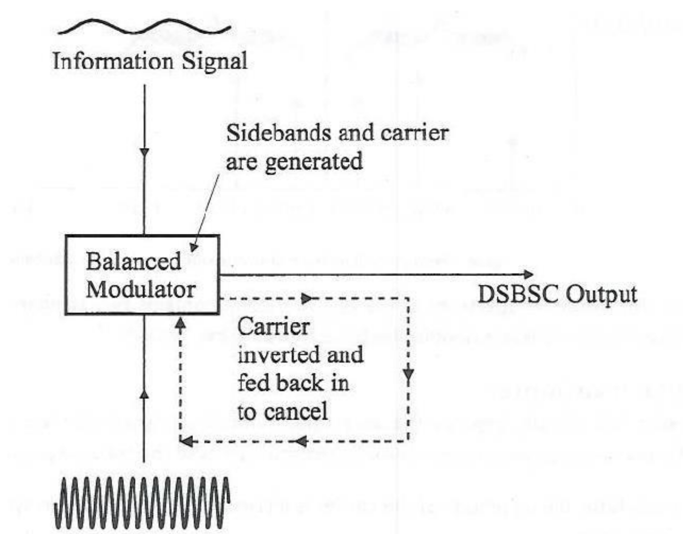


Figure 3: DSBSC Generation using balanced modulator

By removing the carrier from an AM waveforms, the generation of double sideband suppressed carrier (DSBSC) AM is generated.

Properties of DSB-SC Modulation:

(a) There is a 180 phase reversal at the point where m(t) goes negative. This is typical of DSB-SC modulation.

(b) The bandwidth of the DSB-SC signal is double that of the message signal, that is, *BW =2B* (Hz)

(c) The modulated signal is centered at the carrier frequency ω with two identical sidebands (double-sideband) – the lower sideband (LSB) and the upper sideband (USB). Being identical, they both convey the same message component.

(d) The spectrum contains no isolated carrier. Thus the name suppressed carrier.

(e) The 180 phase reversal causes the positive (or negative) side of the envelope to have a shape different from that of the message signal, see Figure 2.

A balanced modulator has two inputs: a single-frequency carrier and the modulating signal. For the modulator to operate properly, the amplitude of the carrier must be sufficiently greater than the amplitude of the modulating signal (approximately six to seven times greater.

PROCEDURE:-

1. Switch on the trainer kit and ensure that audio input select switch should be in INT position and mode switch in DSB position.
2. Now, monitor the two inputs to the balanced modulator & band pass filter circuits 1 block, at TP1 and TP9. Signal at TP1 is audio frequency sine wave which is modulating input and TP9 carries a sine wave of 1MHz and 120mVpp approx. This is the carrier input.
3. Next, examine the output of balanced modulator & band pass filter circuit 1 block at TP3, along with modulating signal at TP1.

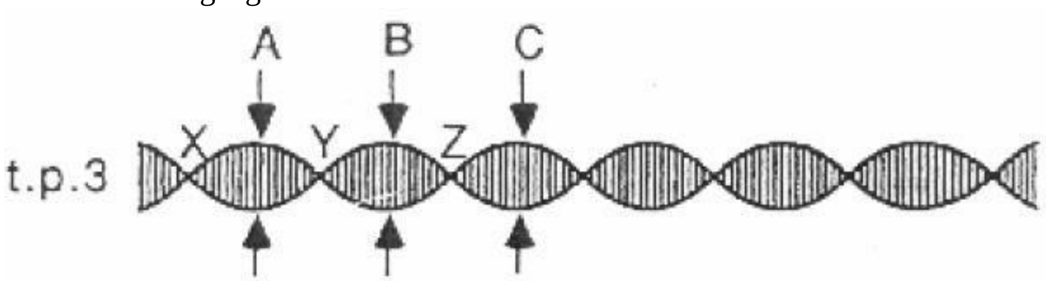


Figure **4**: Output of BPF

4. Now turn the balance pot in the balanced modulator & band pass filter circuit 1 block, until the signal at TP3 is as shown in figure 4. The balance pot varies the amount of the 1 MHz carrier component, which is passed from the modulator's output. By adjusting the pot until the peaks of the waveform (A, B, C and so on) have the same amplitude, we are removing the carrier component altogether. We say that the carrier has been 'balanced out' (or 'suppressed') to leave only the two sidebands. The waveform at TP3 is known as a double-side suppressed carrier (DSBSC) waveform.
5. Now, examine the output from the output amplifier block (TP13), together with the audio modulating signal (at TP1).

Observations:

Amplitude of modulating signal -------------------------------

Frequency of modulating signal--------------------------------

Amplitude of carrier signal ________________________

Frequency of carrier signal ________________________

Frequency of Balanced detector output signal---------------

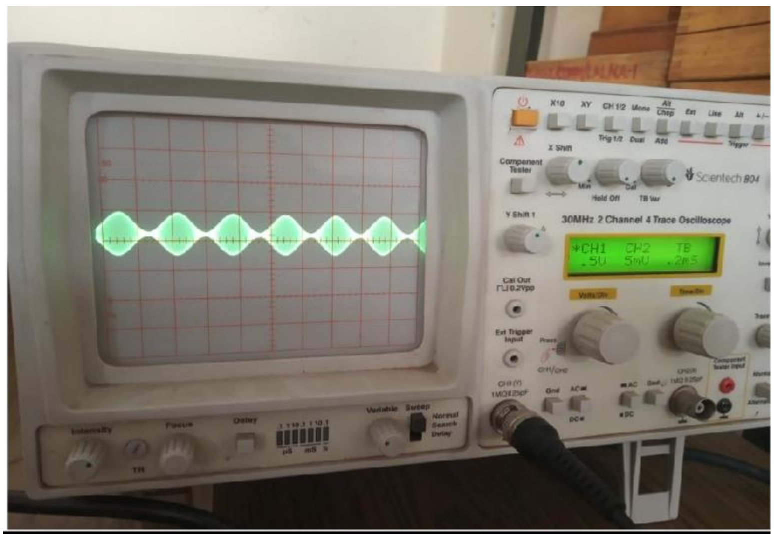


Figure 5. Observed DSBSC waveform

RESULT:-

The DSBSC signal has been generated.

WAVE FORMS OBSERVED:-

Draw wave forms as observed on CRO and label the different waveforms appropriately.

PRECAUTIONS:-

1. Do not use open ended wires for connecting to 230 V power supply.
2. Before connecting the power supply plug into socket, ensure power supply should be switched off
3. Ensure all connections should be tight before switching on the power supply.
4. Take the reading carefully.
5. Power supply should be switched off after completion of experiment.

LAB QUESTIONS:-

Q. 1. What is DSBSC?
Q. 2. Which are the discrete frequencies in DSBSC?
Q. 3 In DSBSC, how many sidebands are there?
Q. 4. Mention advantages of DSBSC over DSBFC.
Q. 5. Which type of carrier is used in Ring modulator?
Q. 6. Write the methods of DSBSC generation.
Q. 7. What is the BW of DSBSC for a single tone modulating signal with frequency *w*?
Q. 8. Where the modulation index lies?
What is the range of audio frequencies?

EXPERIMENT No.4
SSB-SC GENERATION

AIM:- To generate Single Sideband Suppressed Carrier (SSB-SC) signal.

APPARATUS REQUIRED:- (i) C.R.O. (ii) CRO Probe (ii) DSB/SSB Transmitter (ST 2201) and Receiver Trainer (ST 2202) (iv) Connecting leads

THEORY:-

Single Sideband Suppressed Carrier (SSB-SC) modulation was the basis for all long distance telephone communications up until the last decade. It was called "L carrier." It consisted of groups of telephone conversations modulated on upper and/or lower sidebands of contiguous suppressed carriers. The groupings and sideband orientations (USB, LSB) supported hundreds and thousands of individual telephone conversations.

SSB Transmitter:

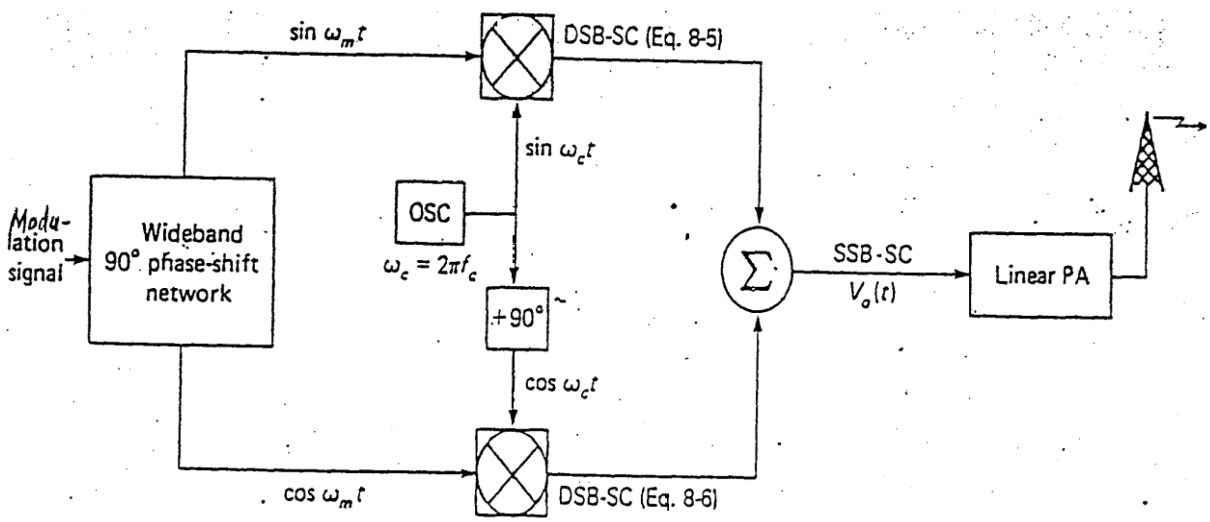


Figure 1: SSB Transmitter

A double side band transmission was the first method of modulation developed and for broadcast stations, is still the most popular. Indeed , for medium and long range broad cast stations is still the most popular .In medium and long range broad cast stations is still the most popular .The reason for such wide spread use is that the receiver design can be simple and reliable. Radio is also used for communications in which the signal is addressed to a receiving station or a group of station .For this type of communication other system are used, one of which is investigated.

PROCEDURE:-

1. Turn on power to the **ST2201** board and ensure that the audio input select switch in INT position and mode switch in SSB position.
2. Turn the audio oscillator block's amplitude pot to its fully clockwise (MAX) position, and examine the block's output (t.p.14) on an oscilloscope. This is the audio frequency sine wave which will be used as out modulating signal and adjust the frequency pot for an audio frequency of 2 KHz, approx. (mid-way).
3. To achieve signal- sideband amplitude modulation, we will utilize the following three blocks on the **ST2201** module.
    a. Balanced modulator.
    b. Ceramic band pass filter
    c. Balanced modulator & band pass filter circuit 2.
4. Monitor the two inputs to the balanced modulator block, at t.p.15 and t.p.6 noting that:
    a. The signal t.p. 15 is the audio frequency sine wave from the audio oscillator block. This is the modulating input to the balanced modulator block.
    b. The signal at t.p. 6 is a sinewave whose frequency is slightly less than 455 KHz. It is generated by the 455 KHz oscillator block, and is the carrier input to the balanced modulator block.
5. Next, examine the output of the balanced modulator block (at t.p.17), together with the modulating signal at t.p.15 trigger the oscilloscope on the modulating signal. Check that the waveforms are as shown Fig. 2.

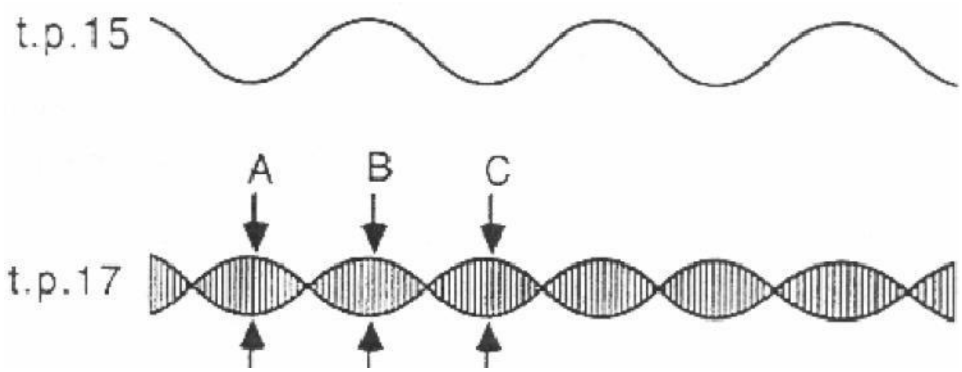


Figure 2: Modulating and Modulated Signal waveforms

Note that it may be necessary to adjust the balanced modulator block's balance pot, in order to ensure that the peaks of t.p.17's waveform envelope (labeled A, B, C etc. in the above diagram) all have equal amplitude.

6. The DSBSC output from the balanced modulator block is next passed on to the ceramic filter block, whose purpose is to pass the upper sideband, but block the lower sideband.
7. Monitor the output of the ceramic band pass filter block (at t.p. 20) together with the audio modulating signal (at t.p.15). Note that the envelope of the signal at t.p. 20 now has fairly constant amplitude, as shown in Fig.3.

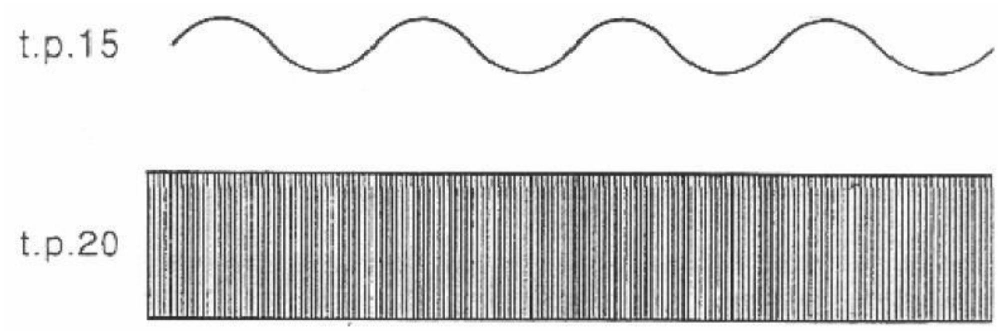


Figure 3: Input Audio Signal and SSB output Signal

If the amplitude of the signal at t.p. 20 is not reasonably constant, adjust the balance pot in the balance modulator block to minimize variations in the signal's amplitude. If the constant-amplitude waveform still cannot be obtained, then the frequency of the 455 KHz oscillator needs to be trimmed.

8. Now examine the output of the balanced modulator & band pass filter circuit 2 blocks (t.p.22), and check that the waveform is a good sine wave of frequency approximately 1.45MHz. This indicates that only the upper sideband is being passed by the block. Monitor the 1.455 MHz SSB signal (at t.p. 22) together with the audio modulating signal (t.p. 15).

9. Examine the final SSB output (at t.p. 22) together with the output from the output amplifier block (t.p. 13). Note that the final SSB waveform appears, amplified slightly, at t.p. 13.

Observations:

Amplitude of AF Gr 0^0 phase signal =... V.
Frequency of AF Gr 0^0 phase signal =... HZ.
Amplitude of AF Gr90^0 phase signal =... V.
Frequency of AF Gr90^0 phase signal =... HZ.
Amplitude of RF Gr 0^0 phase signal =... V.
Frequency of RF Gr 0^0 phase signal =... HZ.
Amplitude of RF Gr90^0 phase signal =... V.
Frequency of RF Gr90^0 phase signal =... HZ.
Amplitude of SSB (USB) signal =... v.
Frequency of SSB (USB) signal =... HZ.
Amplitude of SSB (LSB) signal =... v.
Frequency of SSB (LSB) signal =... HZ.

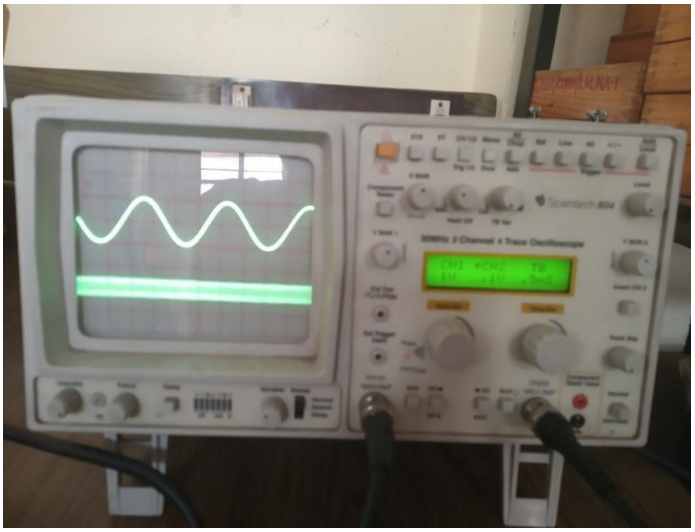


Figure 4: Observed SSBSC waveform

RESULT:-

The SSB signal has been generated using balanced modulator.

PRECAUTIONS:-

1. Do not use open ended wires for connecting to 230 V power supply.
2. Before connecting the power supply plug into socket, ensure power supply should be switched off
3. Ensure all connections should be tight before switching on the power supply.
4. Take the reading carefully.
5. Power supply should be switched off after completion of experiment

LAB QUESTIONS:-

Q.1.What is the most commonly used demodulator?
Q.2.What is AGC?
What is the use of AGC?
What is the required oscillator frequency in AM receiver?
Q.5.What is the use of pilot carrier in SSB?
Q.6.What are the methods of SSB generation?
Q.7.What are the advantages of SSB over DSB?
Q.8.Which type of modulation is used in India for video transmission?
Q.9.Which filter is used in SSB generation?
Q.10.How AM signals with large carrier are detected?

EXPERIMENT No.5
FREQUENCY MODULATION

AIM:- To generate FM signal using Varactor & reactance modulation.

APPARATUS REQUIRED☹i)C.R.O. (ii) CRO Probe (ii) FM Modulation and Demodulation Trainer (ST 2203) (iv) Connecting leads

THEORY:

FM Using Varactor Modulator:
The variations in capacitance form part of the tuned circuit that is used to generate the FM signal to be transmitted. Varactor modulator is shown in fig 1.

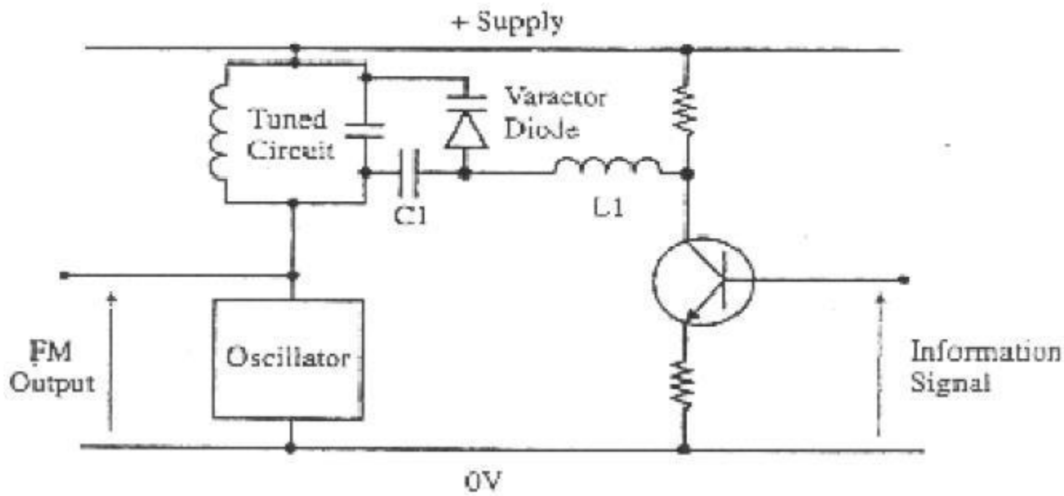


Figure 1: FM generation using Varactor Modulator

We can see the tuned circuit which sets the operating frequency of the oscillator and the varactor which is effectively in parallel with the tuned circuit. Two other components which may not be immediately obvious are C1 and L1. C1 is a DC blocking capacitor to provide DC isolation between the oscillator and the collector of the transmitter. L1 is an RF choke which allows the information signal through to the varactor but blocks the RF signals.

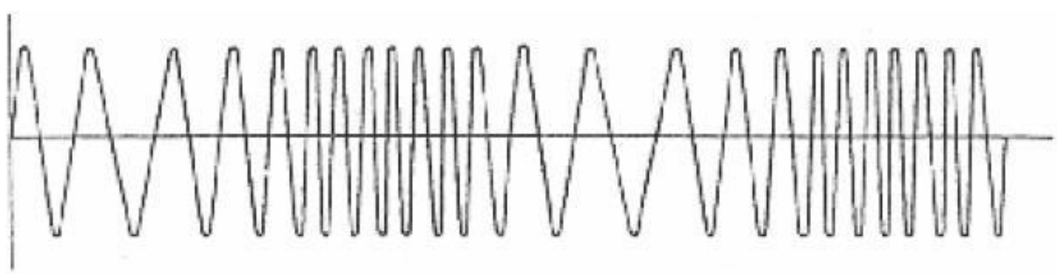


Figure 2: FM modulated wave

<u>FM Using Reactance Modulator:</u> In fig. 3, the left hand half is the previous varactor modulator simply an oscillator and a tuned circuit, which generates the un-modulated carrier. The capacitor C and the resistor R are the two components used for the phase shifting, and together with the transistor, form the voltage controlled capacitor. This voltage-controlled capacitor is actually in parallel with the tuned circuit.

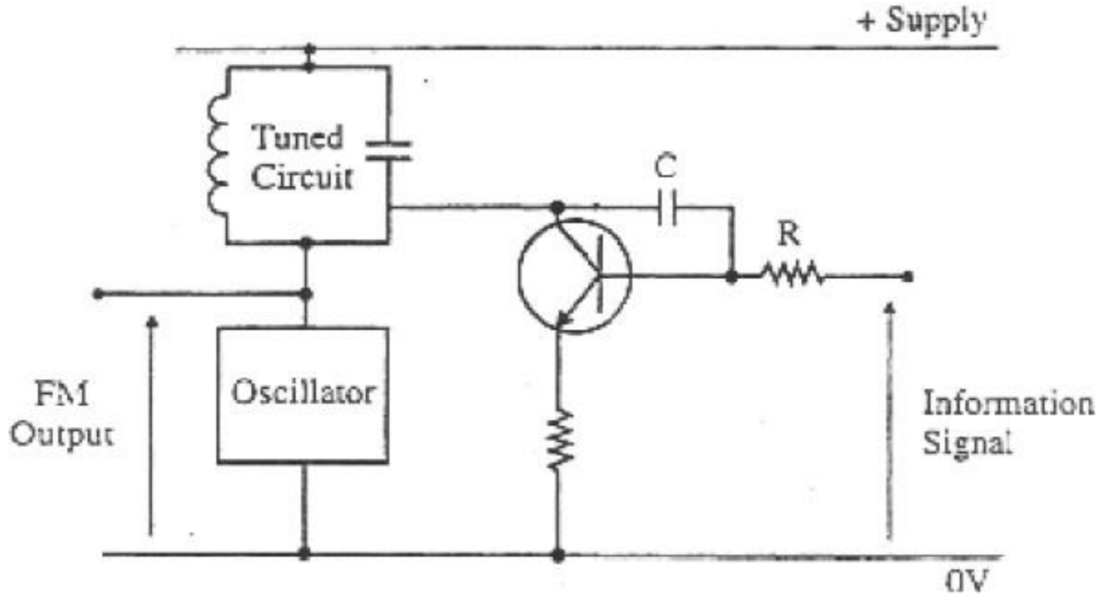


Figure 3: FM using Reactance Modulation.

In the next part, the two supply lines are connected together. We can justify this by saying that the output of the DC power supply always includes a large smoothing capacitor to keep the DC voltages at a steady value. This large capacitor will have a very low reactance at the frequencies being used in the circuit less than a milliohm. We can safely ignore this and so the two supply lines can be assumed to be joined together. Remember that this does not affect the DC potentials, which remain at the normal supply voltages.

BLOCK DIAGRAM:-

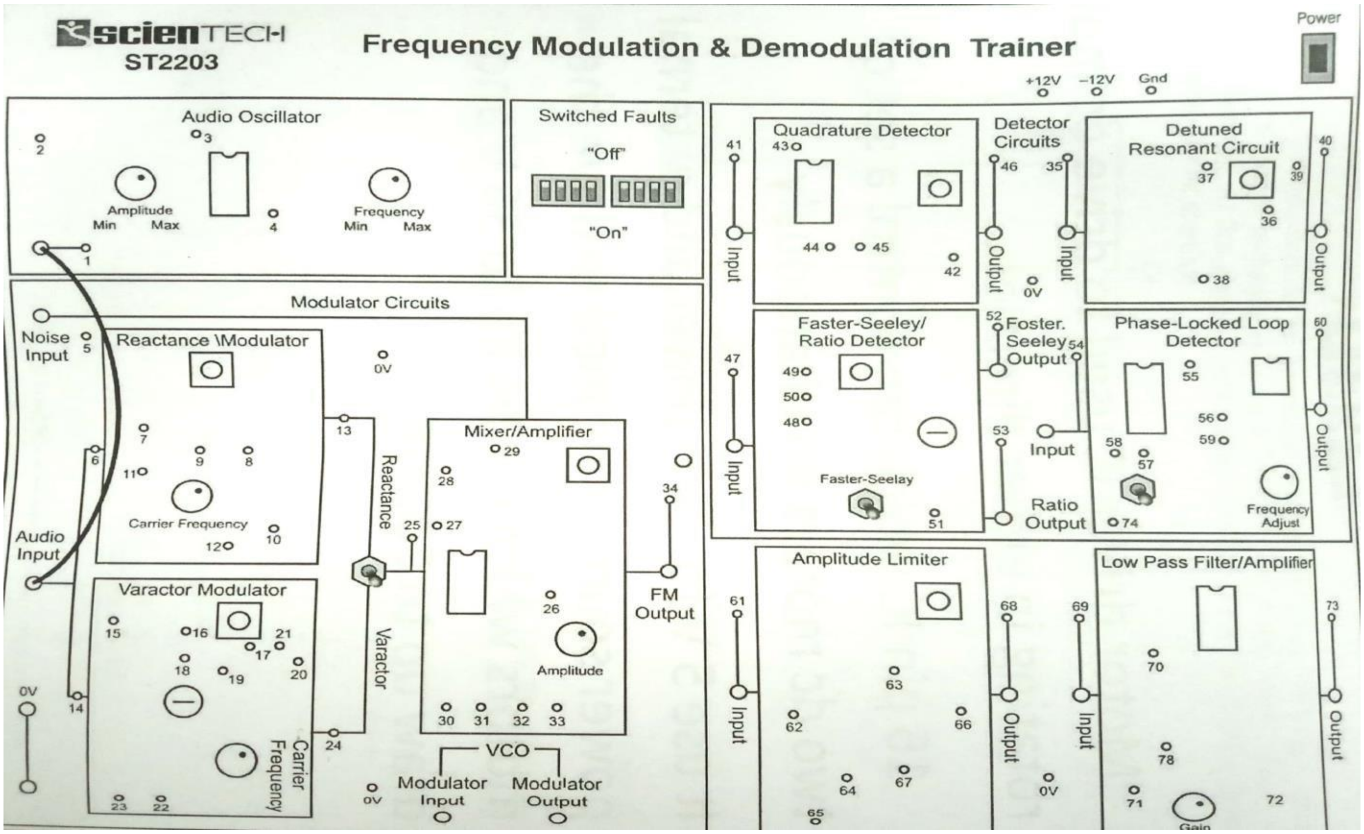


Figure 4: Block Diagram of FM Trainer kit

PROCEDURE:-

1. Ensure that the following initial conditions exist on the **ST2202** board.
    a. All Switched Faults in ‘Off’ condition.
    b. Amplitude potentiometer (in mixer amplifier block) in fully clockwise position.
    c. VCO switch (in phase locked loop detector block) in ‘Off’ position.
2. Make the connections as shown in fig 3.
3. Switch On the power and turn the audio oscillator block’s amplitude potentiometer to its fully clockwise position, and examine the block’s output TP1 on an Oscilloscope. This is the audio frequency sine wave.
4. Connect the output socket of the audio oscillator block to the audio input socket of the modulator circuit’s block.

For FM Varactor Modulator

5. Set the reactance / varactor switch to the varactor position. This switch selects the varactor modulator and also disables the reactance modulator to prevent any interference between the two circuits.
6. The output signal from the varactor modulator block appears at t.p. 24 before being buffered and amplified by the mixer / amplifier block, any capacitive loading (e.g. due to oscilloscope probe) may slightly affect the modulators output frequency. In order to avoid this problem we monitor the buffered FM output signal the mixer / amplifier block at t.p.34.
7. Put the varactor modulator’s carrier frequency pot in its midway position, and then examine t.p. 34. Note that it is a sine wave of approximately 1.2 Vp-p, centered on 0V. This is our FM carrier, and it is un-modulated since the varactor modulators audio input signal has zero amplitude.

8. The amplitude of the FM carrier (at t.p.34) is adjustable by means of the mixer/amplifier block's amplitude pot, from zero to its pot level. Try turning this pot slowly anticlockwise, and note that the amplitude of the FM signal can be reduced to zero. Return the amplitude pot to its fully clockwise position.
9. Try varying the carrier frequency pot and observe the effects.
10. Also, see the effects of varying the amplitude and frequency pots in the audio oscillator block.
11. Turn the carrier frequency pot in the varactor modulator block slowly clockwise and note that in addition to the carrier frequency increasing there is a decrease in the amount of frequency deviation that is present.
12. Return the carrier frequency pot to its midway position, and monitor the audio input (at t.p.6) and the FM output (at p.34) triggering the oscilloscope on the audio input signal. Turn the audio oscillator's amplitude pot throughout its range of adjustment, and note that the amplitude of the FM output signal does not change. This is because the audio information is contained entirely in the signals frequency and not in its amplitude.

For FM Reactance Modulator:

5. Put the reactance /varactor switch in the reactance position. This switches the output of the reactance modulator through to the input of the mixer/amplifier block ~ and also switches off the varactor modulator block to avoid interference between the two modulators.
6. The output signal from the reactance modulator block appears at tp.13, before being buffered and amplified by the mixer/amplifier block. Although the output from the reactance modulator block can be monitored directly at tp.13, any capacitive loading affect this point (e.g. due to an oscilloscope probe) may slightly affect the modulator's output frequency. In order to avoid this problem we will monitor the buffered FM output signal from the mixer/amplifier block at t.p. 34.
7. Put the reactance modulator's pot in its midway position (arrow pointing towards top of PCB) then examine t.p. 34. **Note** that the monitored signal is a sine wave of approximately 1.2V peak/peak centered on 0 volts D.C. This is our FM carrier, and it is presently un-modulated since the reactance modulator's audio input signal has, zero amplitude.
8. The amplitude of the FM carrier (at t.p.34) is adjustable by means of the mixer/amplifier block's amplitude pot, from zero to its present level. Try turning this pot slowly anticlockwise, and note that the amplitude of the FM signal can be reduced to zero. Return the amplitude pot to its fully clockwise position.
9. The frequency of the FM carrier signal (at t.p.34) should be approximately 455Khz at the moment This carrier frequency can be varied from 453Khz to 460Khz (approx.) by adjusting the carrier frequency pot in the reactance modulator block. Turn this pot over its range of adjustment and note that the frequency of the monitored signal can be seen to vary slightly. Note also that the carrier frequency is maximum when the pot is in fully clockwise position.
10. Try varying the amplitude & frequency pot in audio oscillators block, and also sees the effect of varying the carrier frequency pot in the mixer/amplifiers block.
11. Monitor the audio input (at t.p.6) and the FM output (at t.p.34) triggering the oscilloscope on the audio input signal. Turn the audio oscillator's amplitude pot throughout its range of adjustment and note that the amplitude of the FM output signal does not change. This is because the audio information is contained entirely in the signal's frequency, and not in its amplitude.

OBSERVATIONS:

Amplitude of modulating signal________________________
Frequency of modulating signal__________________________
Amplitude of carrier signal____________________________
Frequency of carrier signal_____________________________
Frequency deviation ____________________________

RESULT:-

Frequency modulated signal is generated by using varactor and reactance modulator.

PRECAUTIONS:-

1. Do not use open ended wires for connecting to 230 V power supply.
2. Before connecting the power supply plug into socket, ensure power supply should be switched off
3. Ensure all connections should be tight before switching on the power supply.
4. Take the reading carefully.
5. Power supply should be switched off after completion of experiment.

LAB QUESTIONS:-

How many types of FM are there? Write their names.
What frequency deviation in FM?
Which is the useful parameter for determination of bandwidth?
How many sidebands are there in FM?
Q.1. Which sidebands are ignored in FM?
Which are significant sidebands?
What is CCIR?
What is the indirect method of FM generation?
Classify FM on the basis of bandwidth.
Which one is better in terms of noise immunity AM or FM?

EXPERIMENT No.6
FREQUENCY DEMODULATION

AIM:- To Detect FM Signal using Foster-Seeley method.

APPARATUS REQUIRED:- (i)C.R.O. (ii) CRO Probe (ii) FM Modulation and Demodulation Trainer (ST 2203) (iv) Connecting leads

THEORY:-

A FM receiver is very similar to an AM receiver. The most significant change is that the demodulator must now extract the information signal from a frequency rather than amplitude modulated wave.

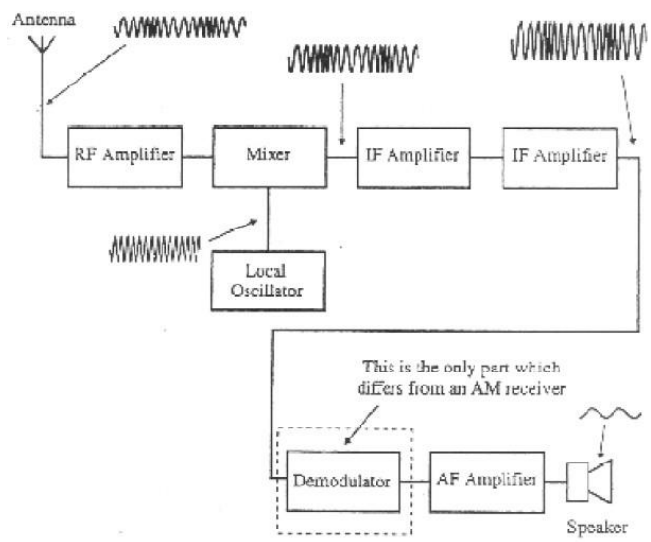


Figure 1: FM Receiver

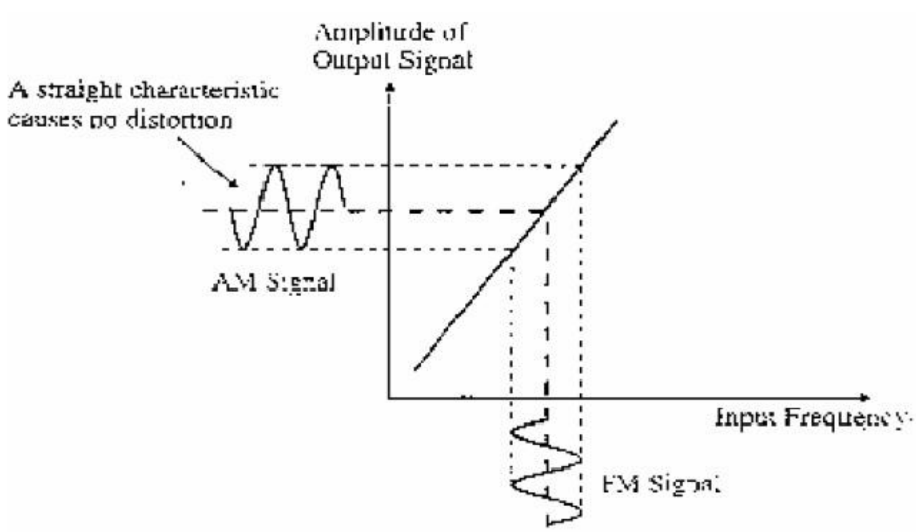
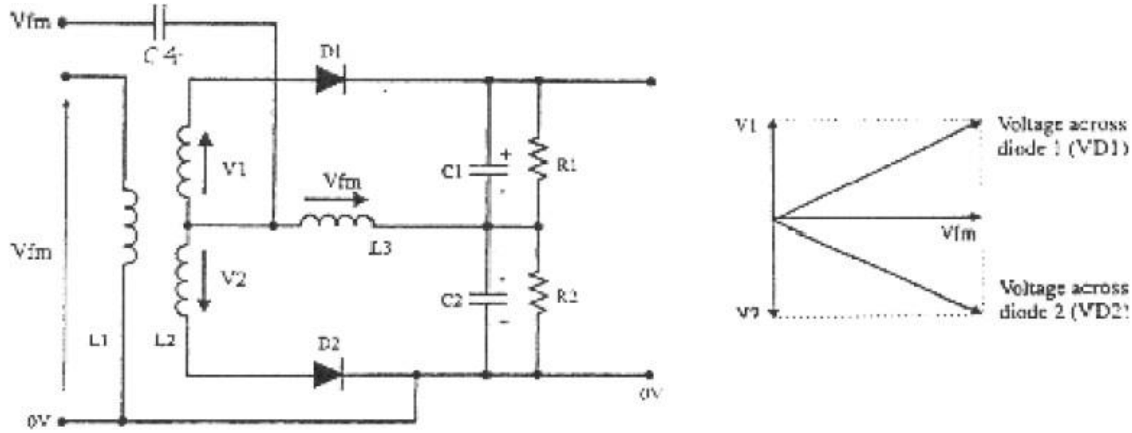


Figure 2: Voltage/Frequency Characteristics

FOSTER SEELEY DETECTOR

The foster Seeley circuit is shown in fig. 3. At first glance, it looks rather complicated but it becomes simpler if we consider it a bit at a time.



Circuit diagram **Phasor diagram**

Figure 3: Foster –Seeley Detector

When the input signal is un-modulated: We will start by building up the circuit a little at a time. To do this, we can ignore many of the companies we may recognize immediately that it consist of two envelope detectors like half wave rectifiers are fed from the center-tapped coil L2. With reference to the center-tap, the two voltages V1 and V2 are in anti-phase as shown by the arrows. The output voltage would be zero volts since the capacitor voltages are in anti-phase and are equal in magnitude. After adding two capacitors: The next step is to add two capacitors and see their effect on the phase of the signals. L1 and L2 are magnetically tightly coupled and by adding C3 across the centre-tapped coil, they will form a parallel tuned circuit with a resonance frequency equal to the un-modulated carrier frequency. Capacitor C5 will shift the phase of the input signal by 90° with reference to the voltage across L1 and L2. The voltages are shown as Va and Vb in the phasor diagram given in figure 39. Using the input signal Vfm as the reference, the phasor diagrams now look the way shown in figure 4. C4 is not important. It is only a DC blocking capacitor and has negligible impedance at the frequencies being used. But what it does do is to supply a copy of the incoming signal across L3. The entire incoming signal is dropped across L3 because C1 and C2 also have negligible impedance. If we return to the envelope detector section, we now have two voltages being applied to each diode. One is V1 or V2 and the other is the new voltage across L3, which is equal to Vfm. When the input Frequency changes: If the input frequency increased above its un-modulated value, the phasor of Va would fall below 90° due to the parallel tuned circuit becoming increasingly capacitive. This would result in a larger total voltage being applied across D1 and a reduced voltage across D2. Since the capacitor C1 would now charge to a higher voltage, the final output from the circuit would be a positive voltage. Conversely, if the frequency of the FM input signal decreased below the unmodulated value, the phase shift due to capacitor C5 increases above 90° as the parallel tuned circuit becomes slightly inductive. This causes the voltage across diode D2 to increase and the final output from the demodulator becomes negative. The effect of noise is to change the amplitude of the incoming FM signal resulting in a proportional increase and decrease in the amplitude of diode voltages VD1 and VD2 and the difference in voltage is the demodulated output, the circuit is susceptible to noise interference and should be preceded by a noise limiter circuit.

BLOCK DIAGRAM:-

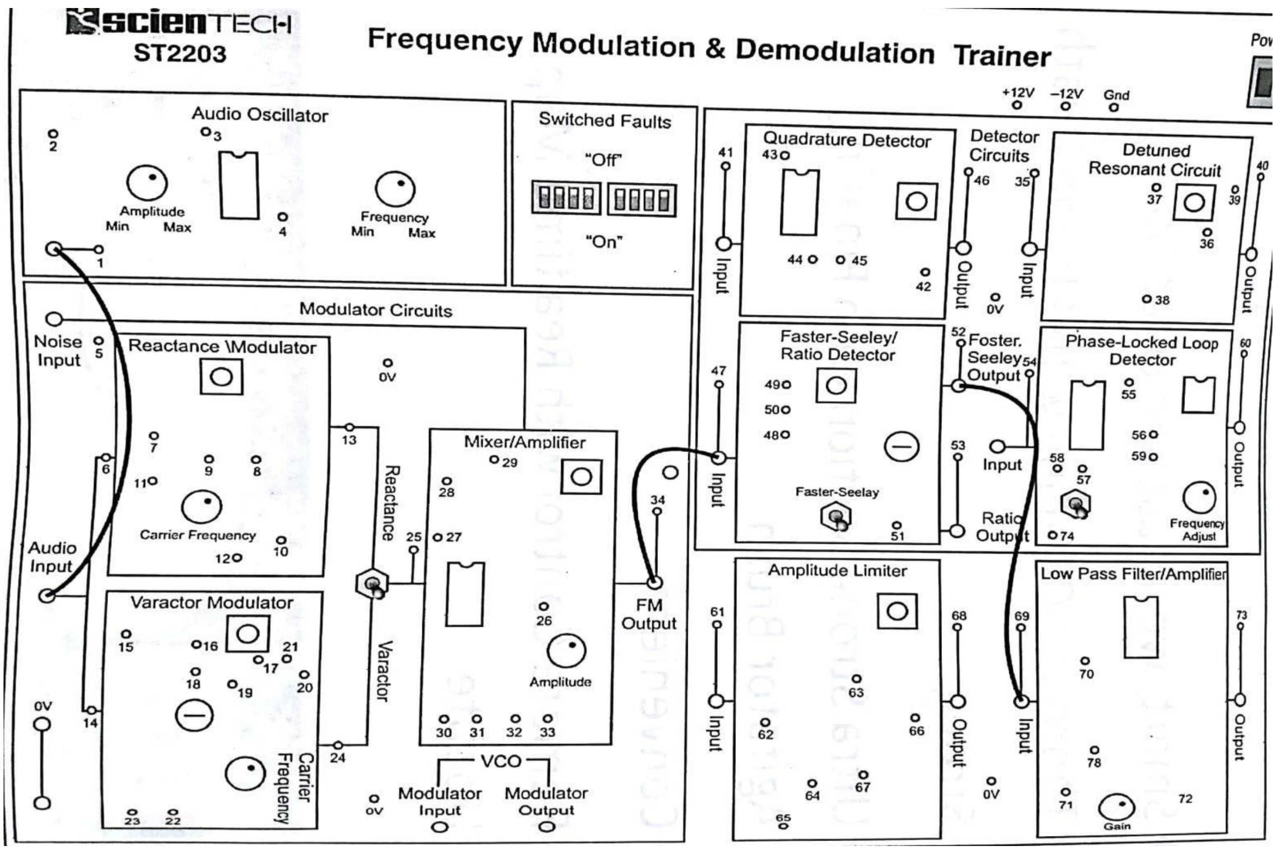


Figure 4: Connections for FM Demodulation using Foster-Seelay Detector

PROCEDURE:-

1. Ensure that the following initial conditions exist on the **ST2203** module:

- **a.** All switched faults OFF;
- **b.** Audio amplifier block's amplitude pot in fully clockwise (MAX) position.
- **c.** Audio amplifier block's frequency pot in fully counter-clockwise (MIN) position.
- **d.** Amplitude pot (in the mixer/amplifier block) in fully clockwise position.
- **e.** VCO switch (in phase-locked loop detector block) in OFF position.

2. Make connection as shown in figure 4.

3. Turn on power to the **ST2203** module.

4. We will now investigate the operation of the foster-Seeley detector on the **ST2203** module. In the Foster-Seeley / ratio detector block, select the Foster-Seeley detector by putting the switch in the Foster-Seeley position.

5. Initially, we will use the varactor modulator to generate our FM signal, since this is the more linear of the two modulators, as fast as its frequency/voltage characteristic is concerned. To select the varactor modulator, put the reactance/ varactor switch in the varactor position.

Ensure that the varactor modulator's carrier frequency pot is in the midway position.

6. The audio oscillator's output signal (which appears at t.p.1) is now being used by the varactor modulator, to frequency-modulate a 455Khz carrier sine wave. As we saw earlier, this FM waveform appears at the FM output socket from the mixer/amplifier block. You will probably need to have an X-expansion control on your oscilloscope.

7. Now monitor the audio input signal to the varactor modulator block (at t.p. 14) together with the foster-seeley output from the foster-seeley/ratio detector block (at t.p. 52), triggering the oscilloscope on t.p. 14. The signal at t.p. 52 should contain two components:

- A sine wave at the same frequency as the audio signal at t.p. 14.
- A High frequency ripple component of small amplitude.

8. The low-pass filter/amplifier strongly attenuates this high-frequency ripple component, and blocks any small D.C. offset voltage that might exist at the detector's output. Consequently, the signal at the output of the low-pass filter/ amplifier block (a t.p. 73) should very closely resemble the original audio modulating signal.

9. Monitor the audio input to the varactor modulator (at t.p. 14) and the output of the low pass filter / amplifier block (at t.p. 73) and adjust the gain pot (in the low pass filter/ amplifier block) until the amplitudes of the monitored audio waveforms are the same.

10. Adjust the audio oscillator block's amplitude and frequency pots, and compare the original audio signal with the final demodulated signal.

Observations:

Amplitude of modulating signal_________________________
Frequency of modulating signal___________________________
Amplitude of carrier signal______________________________
Frequency of carrier signal______________________________
Amplitude of demodulating signal_______________________
Frequency of demodulating signal________________________

RESULT:-

FM signal is being demodulated by Foster-Seeley Method.

PRECAUTIONS:-

1. Do not use open ended wires for connecting to 230 V power supply.
2. Before connecting the power supply plug into socket, ensure power supply should be switched off
3. Ensure all connections should be tight before switching on the power supply.
4. Take the reading carefully.
5. Power supply should be switched off after completion of experiment.

LAB QUESTIONS:-

Q.1.How many types of FM are there? Write their names.
Q.2.What frequency deviation in FM?
Which is the useful parameter for determination of bandwidth?
What are different methods of FM detection?
Q.5.Which sidebands are ignored in FM?
Q.6.Which are significant sidebands?
Q.7.What is basic principle of FM detection?
Q.8.What is the indirect method of FM generation?
Q.9.What is the direct method of FM generation?
Q.10.What is the function of amplitude limiter?

EXPERIMENT 7

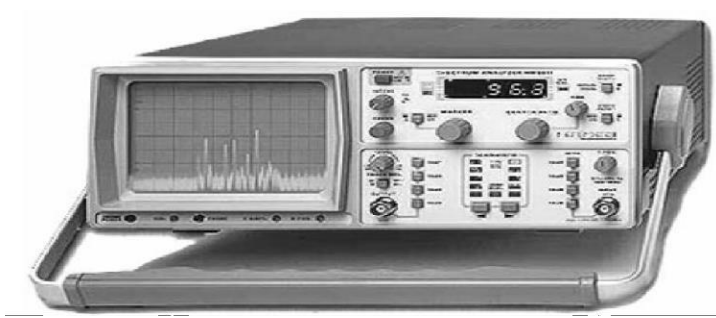
Spectrum of AM Signal using Spectrum analyzer

AIM:
To study the spectrum of AM signals using spectrum analyzer.

EQUIPMENT / COMPONENTS REQUIRED: Spectrum Analyzer, Any modulation system kit, Connecting wires.

THEORY:

To analyze the AM and FM waveform using spectrum analyzer. The oscilloscope is the most common device used to display the signals, with time as x-axis. The signals which require time as x-axis, to display them are time domain signals. The signals which require frequency as x-axis, to display them are called frequency domain signals. Frequency domain display of signal consists of information of energy distributed of the signal. The analysis of such a frequency domain display of the signal is called spectral analysis of the signal. Thus the study of the energy distribution across the frequency spectrum if a given signal is defined as the spectral analysis. The instrument which graphically provides the energy distribution of a signal as a function of frequency on its CRT is called spectrum analyzer. It provides a calibrated graphical display with the frequency on horizontal axis and the signal component on the vertical axis, the sinusoidal components of which, the signal is made up of, are displayed as the vertical lines against frequency coordinates. The frequency of each vertical line gives the absolute amplitude if the component while the horizontal location gives that particular frequency.



HAMEG 5010 SPECTRUM ANALYZER

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

Types Of Spectrum Analyzers

There are two basic types of spectrum analyzers, swept-tuned and real time analyzers. The swept-tuned analyzers are tuned by electrically sweeping them over their frequency range. Therefore the frequency components of a spectrum are sampled sequentially in time. This enables periodic and random signals to be displayed, but makes it impossible to display transient response. Real time analyzers, on the other hand, simultaneously display the amplitude of all signals in the frequency range of the analyzer: hence the name real-time. This preserves the time dependency between signals which permits information to be displayed. Real time analyzers are capable of displaying transient response as well as periodic and random signals. The swept tuned analyzers are usually of the TRF (tuned radio frequency) or super heterodyne type. A TRF analyzer consists of a frequency range, a detector to produce vertical deflection on a CRT, and a horizontal scan generator used to synchronize the tuned frequency to the CRT horizontal deflection. It is a simple, inexpensive analyzer with wide frequency coverage, but lacks resolution and sensitivity. Because TRF analyzers have swept filter they are limited in sweep width.

Applications Of Spectrum Analyzers

1. **Modulation measurements**:
When the frequency scan of spectrum analyzer is set to zero and x-axis is representing time instead of frequency, it operates as a fixed tuned receiver to measure amplitude against time. This is called its synchroscope mode. When analyzer is tuned to carrier frequency with bandwidth at least twice that of modulation frequency and with a linear display, the

envelop of an AM signal is observed. Measuring the peak VP and through VT, modulation index can be determined. When operated in normal mode, two sidebands separated from the carrier by modulation frequency fm are observed. The modulation index can be calculated from the sidebands and carrier amplitude. Similarly it can be used to calculate the distortion occurring in modulation process. The sideband configuration in frequency modulation enables observer to calculate the frequency modulation index.

2. **Continuous wave signal frequency stability**
   The frequency drift of a signal can be measured by observing the excursions of the signal across the display. Over period of minutes, it gives long term stability while over period of seconds it gives short term stability.

3. **Harmonic distortion measurement**
   The distortion affects the frequency components of a signal to be transmitted. The harmonics appear as the additional signals in the spectrum analyzer at multiples of the carrier frequency. To keep it low, its measurement plays an important role. The spectrum analyzer can be used to make such distortion measurements.

4. **Noise measurement**
   The noise can be measured with very straightforward method using the spectrum analyzer. Similarly the measurement of impulse noise also can be measured using spectrum analyzer. The examples of impulse noise in the generation of voltage spikes due to engine ignition and electric motor commutation.

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

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

MODEL WAVEFORM:

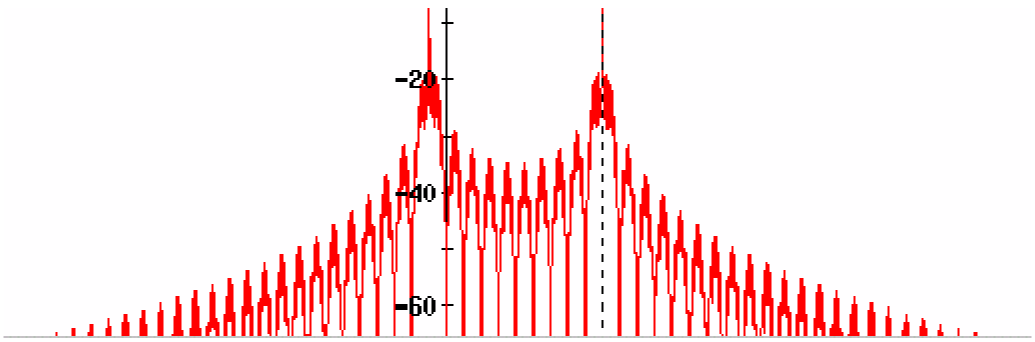


Figure1: AM Spectral analysis signal

PRECAUTIONS:

1. Check for loose contacts of wires and components.
2. Keep all the control knobs in the minimum position.
3. Before switch on the power supply get the circuit connections verified by the teacher.
4. Adjust the control knobs smoothly.
5. After taking the readings bring back all the control knobs to minimum position.
6. Switch off the power supply before leaving the experimental table.

CONCLUSION:

The spectrum of AM Signals are observed using spectrum analyzer and plotted.

LAB QUESTIONS:

Define a spectrum.
Is it possible to visualize the time domain signals using spectrum analyzer?
How can we select the central frequency?
How many side bands appear for a conventional AM signal?
What are the major components required to apply the AM signals to the spectrum analyzer?
Define a marker?
Is it possible to visualize the time domain signals using spectrum analyzer?
Draw the Spectrum of AM wave?
Draw the Spectrum of DSB and SSB wave?
What are frequencies components present in AM wave?
What are the major components required to apply the AM signals to the spectrum analyzer?

Experiments using MATLAB

EXPERIMENT 8
Generation and Detection of AM Signals

Aim:-

To generate Amplitude Modulation and Demodulation using MATLAB Software for different modulation indices.

Apparatus Required

a) Hardware Tools: Computer system

b) Software Tool: MATLAB 7.0 and above version

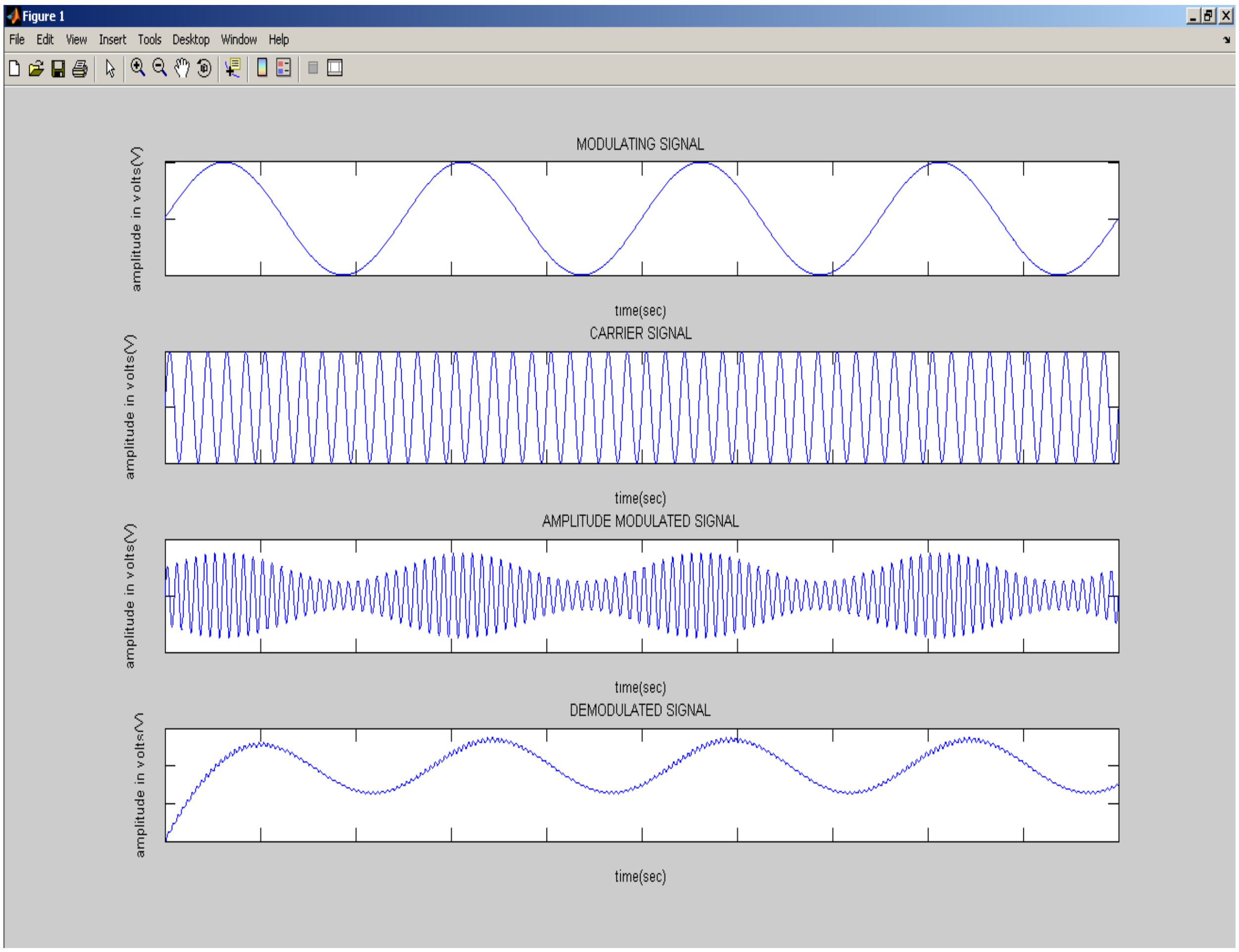
MATLAB PROGRAM:

AM without functions:

```
Clc;
clear all;
close all;
t=linspace(0,0.02,10000);%defining time range for the signal fc=5000;%frequency of carrier signal
fm=200;%frequency of message signals
fs=40000;%samplingfrequency--------- fs>=2(fc+BW)
Am=5;%amplitude of the message signal Ac=10;%amplitude of the carrier signal
m=Am/Ac%modulation index for the AM wave wc=2*pi*fc*t;%carrier frequency in radians wm=2*pi*fm*t;%message frequency in radians ec=Ac*sin(wc);%carrier signal
em=Am*sin(wm);%messagesignal
y=Ac*(1+m*sin(wm)).*sin(wc);%amplitude modulated signal
 z=y.*ec; %in synchronous detection the AM signal is multiplied with carrier signal and passed throughLPF
 z1=conv(z,exp(-t/0.000795));% the LPF filter response in time domain is given by exp(-t/RC), the cut off frequency for filter should befm=200
%F=1/(2*pi*R*C), replacing F=200, and
%assuming R=1k ohm then C=0.795MICROFARAD
%so RC=0.000795
%we will get the demodulated signal by convolving the AM signal with LPF response
l=10000;
subplot(4,1,1),plot(t(1:l),em(1:l))
xlabel('time(sec)'); ylabel('amplitude in volts(V)'); title('MODULATING SIGNAL');
subplot(4,1,2),plot(t(1:l/2),ec(1:l/2)) xlabel('time(sec)'); ylabel('amplitude in volts(V)');
title('CARRIER SIGNAL');
subplot(4,1,3),plot(t(1:l),y(1:l)) xlabel('time(sec)'); ylabel('amplitude in volts(V)');
title('AMPLITUDE MODULATED SIGNAL');
```

```
subplot(4,1,4),plot(t(1:l),z1(1:l)) xlabel('time(sec)'); ylabel('amplitude in volts(V)');
title('DEMODULATED SIGNAL');
```

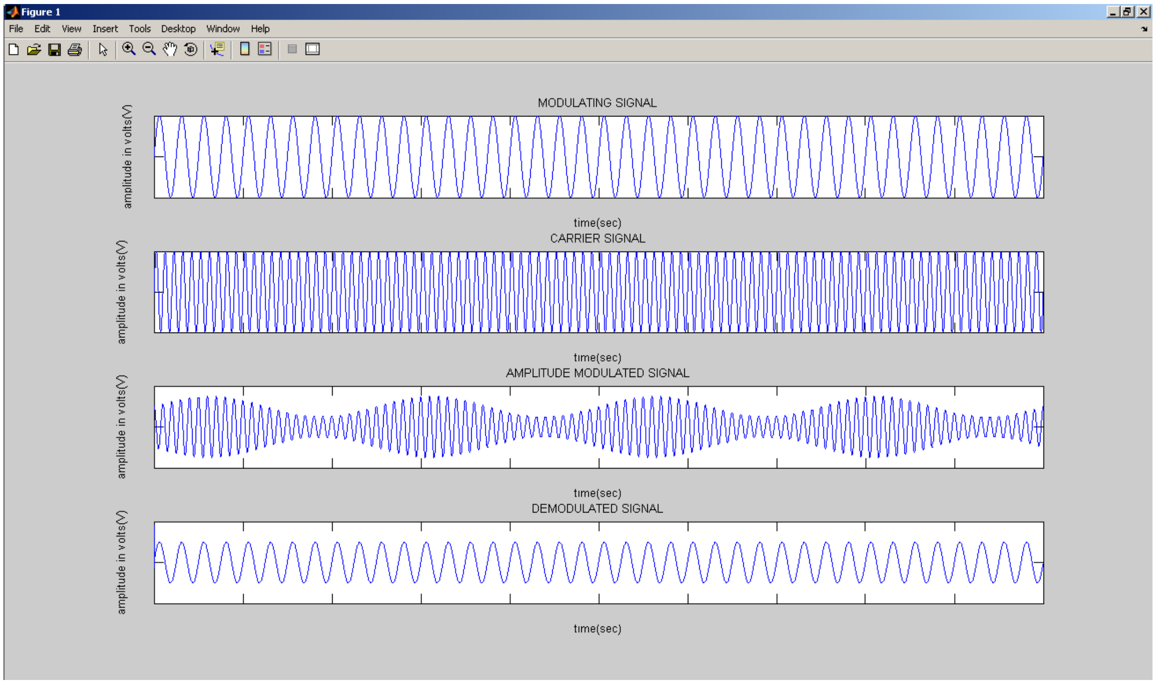
<u>OUTPUT WAVEFORMS:</u>



AM with functions:

```
Clc; Clear all; closeall ;
t=linspace(0,0.2,100000);%defining time range for the signal fc=1000;%frequency of carrier signal
fm=200;%frequency of message signal fs=100000;%samplingfrequency
 ______________________________fs>=2(fc+BW)
Am=5;%amplitude of the message signal Ac=10;%amplitude of the carrier signal
m=Am/Ac%modulation index for the AM wave wc=2*pi*fc*t;%carrier frequency in radians wm=2*pi*fm*t;%message frequency in radians ec=Ac*sin(wc);%carrier signal
em=Am*sin(wm);%messagesignal
y=ammod(em,fc,fs,0,Ac);%amplitude modulated signal
z=amdemod(y,fc,fs,0,Ac);%demodulated AM signal l=100000;
subplot(4,1,1),plot(t(1:l),em(1:l)) xlabel('time(sec)'); ylabel('amplitude in volts(V)');
title('MODULATING SIGNAL');
subplot(4,1,2),plot(t(1:l/2),ec(1:l/2)) xlabel('time(sec)'); ylabel('amplitude in volts(V)');
title('CARRIER SIGNAL');
subplot(4,1,3),plot(t(1:l),y(1:l))
axis([0 0.02 -20 20])%setting axis dimensions xlabel('time(sec)');
ylabel('amplitude in volts(V)');
title('AMPLITUDE MODULATED SIGNAL');
subplot(4,1,4),plot(t(1:l),z(1:l)) xlabel('time(sec)'); ylabel('amplitude in volts(V)');
title('DEMODULATED SIGNAL');
```

OUTPUT WAVEFORMS:



RESULT:

Amplitude modulated wave is observed for different modulation indices.

EXPERIMENT 9
Generation and Detection of DSB-SC Signals

AIM

To perform the AM DSB-SC signal Generation and Detection using Matlab.

APPARATUS REQUIRED

a) Hardware Tools: Computer system

b) Software Tool: MATLAB 15.0 or Upgraded version

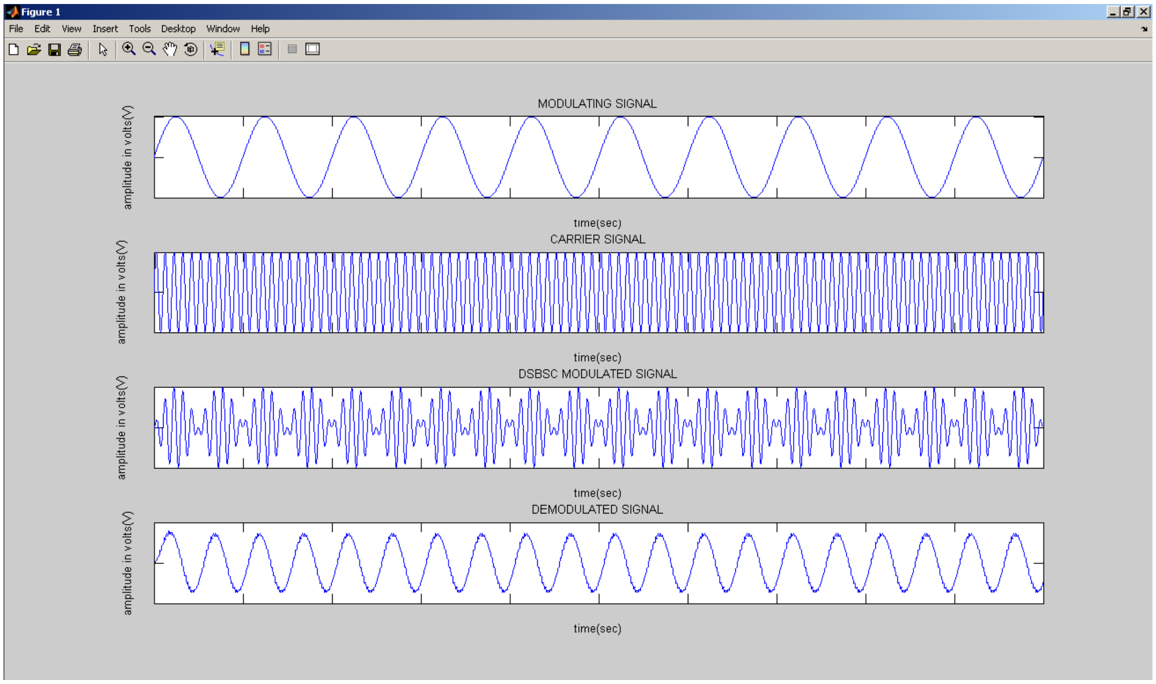
PROGRAM:

AM-DSBSC without functions:

```
Clc; clear all; close all;
t=linspace(0,0.02,100000);%defining time range for the signal fc=10000;%frequency of carrier signal
fm=1000;%frequency of message signaql fs=40000;%samplingfrequency fs>=2(fc+BW)
Am=5;%amplitude of the message signal Ac=10;%amplitude of the carrier signal
m=Am/Ac;%modulation index for the AM wave wc=2*pi*fc*t;%carrier frequency in radians
wm=2*pi*fm*t;%message frequency in radians ec=Ac*sin(wc);%carrier signal
em=Am*sin(wm);%messagesignal
y=em.*ec;
z=y.*ec; %in synchronous detection the AM signal is multiplied with carrier signal and passed through LPF
z1=conv(z,exp(-t/0.000159));% the LPF filter response in time domain is given by exp(-t/RC), the cut off frequency for filter should be fm=200
%F=1/(2*pi*R*C), replacing F=200, and
%assuming R=1k ohm then C=0.159MICROFARAD
%so RC=0.000159
%we will get the demodulated signal by
%convolving the AM signal with LPF response l=100000;
subplot(4,1,1),plot(t(1:l/2),em(1:l/2)) xlabel('time(sec)');
ylabel('amplitude in volts(V)'); title('MODULATING SIGNAL');
subplot(4,1,2),plot(t(1:l/2),ec(1:l/2)) xlabel('time(sec)'); ylabel('amplitude in volts(V)');
title('CARRIERSIGNAL');
subplot(4,1,3),plot(t(1:l/2),y(1:l/2)) xlabel('time(sec)'); ylabel('amplitude in volts(V)');
title('DSBSC MODULATED SIGNAL');
```

```
subplot(4,1,4),plot(t(1:l),z1(1:l)) xlabel('time(sec)'); ylabel('amplitude in volts(V)');
title('DEMODULATED SIGNAL');
```

OUTPUT WAVEFORMS:

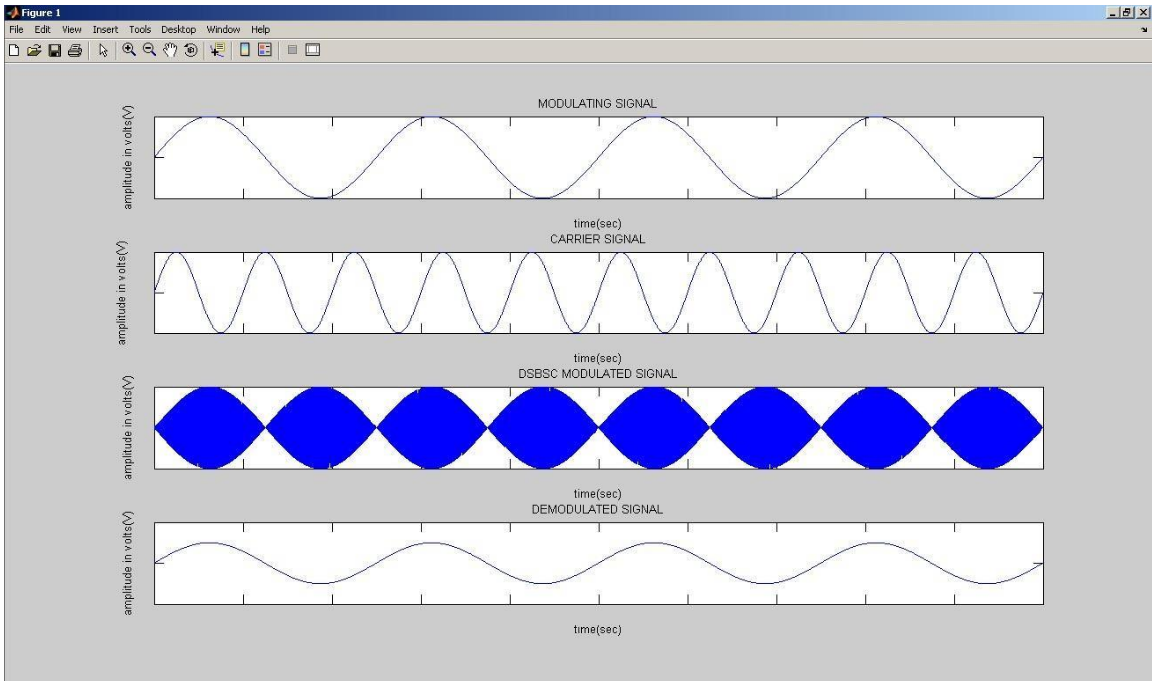


AM-DSBSC with functions:

```
clcclearallcloseall
t=linspace(0,0.02,10000);%defining time range for the signal fc=1000;%frequency of
carrier signal
fm=200;%frequency of message signal fs=10000;%samplingfrequency      fs>=2(fc+BW)
Am=5;%amplitude of the message signal
Ac=10;%amplitude of the carrier signal m=Am/Ac%modulation index for the AM wave
wc=2*pi*fc*t;%carrier frequency in radians wm=2*pi*fm*t;%message frequency in radians
ec=Ac*sin(wc);%carrier signal em=Am*sin(wm);%messagesignal
y=modulate(em,fc,fs,'amdsb-sc');%amplitude modulated signal z=demod(y,fc,fs,'amdsb-
sc');%demodulated AM signal l=10000;
subplot(4,1,1),plot(t(1:l),em(1:l)) xlabel('time(sec)'); ylabel('amplitude in volts(V)');
title('MODULATING SIGNAL');
subplot(4,1,2),plot(t(1:l/2),ec(1:l/2)) xlabel('time(sec)'); ylabel('amplitude in volts(V)');
```

```
title('CARRIER SIGNAL');
subplot(4,1,3),plot(t(1:l),y(1:l))
axis([0 0.02 -5 5])%setting axis dimensions xlabel('time(sec)');
ylabel('amplitude in volts(V)'); title('DSBSC MODULATED SIGNAL');
subplot(4,1,4),plot(t(1:l),z(1:l)) xlabel('time(sec)'); ylabel('amplitude in volts(V)');
title('DEMODULATED SIGNAL');
```

OUTPUT WAVEFORMS:



RESULT:

DSB-SC modulated wave is observed using MATLAB software.

EXPERIMENT 10
Generation of SSB-SC Signals

AIM

To perform the AM SSB-SC signal Generation using Matlab Simulink.

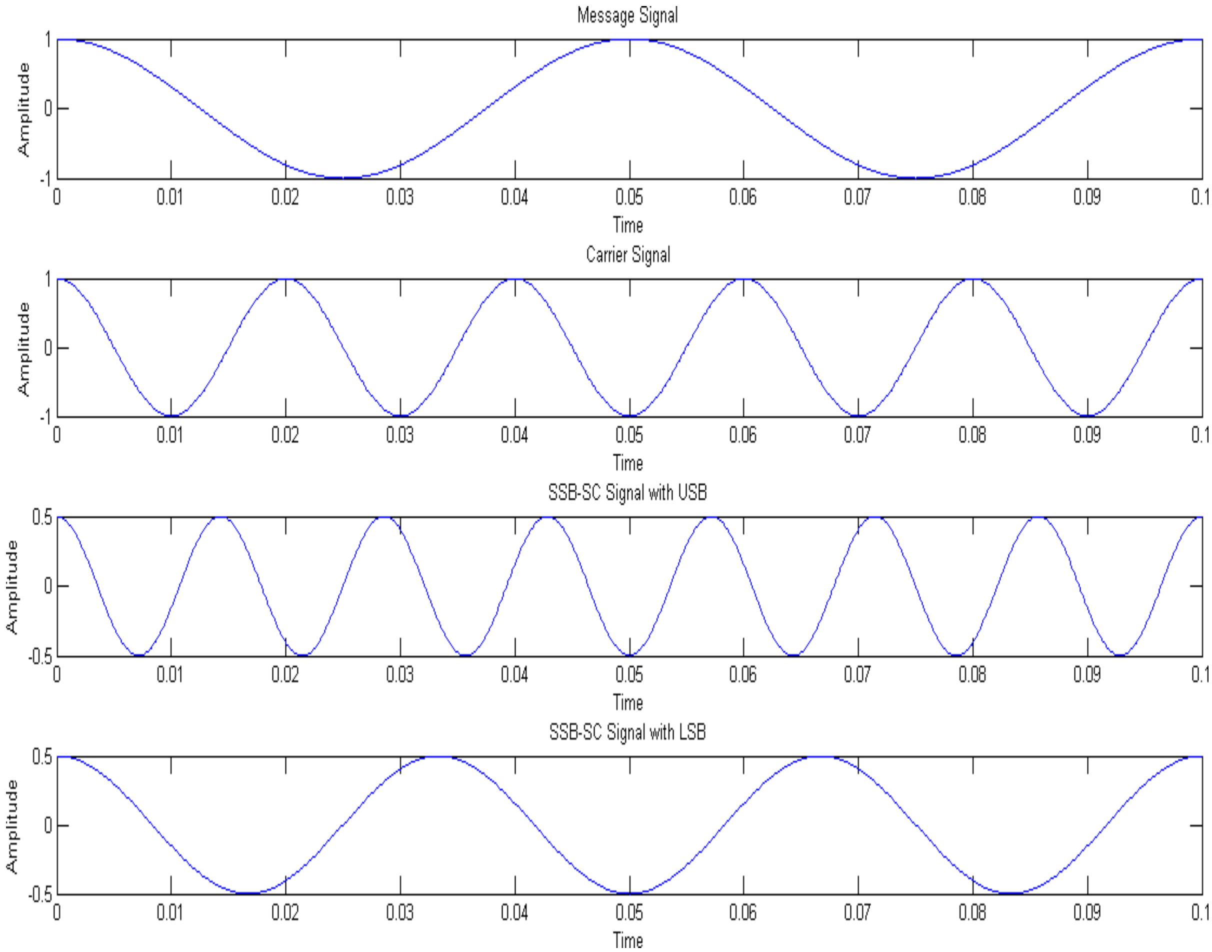
APPARATUS REQUIRED

a) Hardware Tools: Computer system
b) Software Tool: MATLAB 15.0 and above version

MATLAB PROGRAM

```
s=8000;
fm=20; fc=50; Am=1; Ac=1;
t=[0:.1*fs]/fs;
subplot(4,2,1);
m1=Am*cos(2*pi*fm*t);
plot(t,m1);
title('Message Signal');
m2=Am*sin(2*pi*fm*t);
subplot(4,2,2);
c1=Ac*cos(2*pi*fc*t);
plot(t,c1);
title('Carrier Signal');
c2=Ac*sin(2*pi*fc*t);
subplot(4,2,3);
Susb=0.5*m1.*c1-0.5*m2.*c2;
plot(t,Susb);
title('SSB-SC Signal with USB');
subplot(4,2,4);
Slsb=0.5*m1.*c1+0.5*m2.*c2;
plot(t,Slsb);
title('SSB-SC Signal with LSB');
 r = Susb.*c1;
[b a] = butter(1,0.0001);
mr= filter(b,a,r);
subplot(4,2,5);
plot(t,mr);
```

OUTPUT WAVE FORMS



RESULT:

The SSBSC wave has been generated by using a MATLAB Software.

EXPERIMENT 15
Time Division Multiplexing & DeMultiplexing

AIM

To perform the Time Division Multiplexing using Matlab.

Apparatus Required

a) Hardware Tool**s**: Computer system

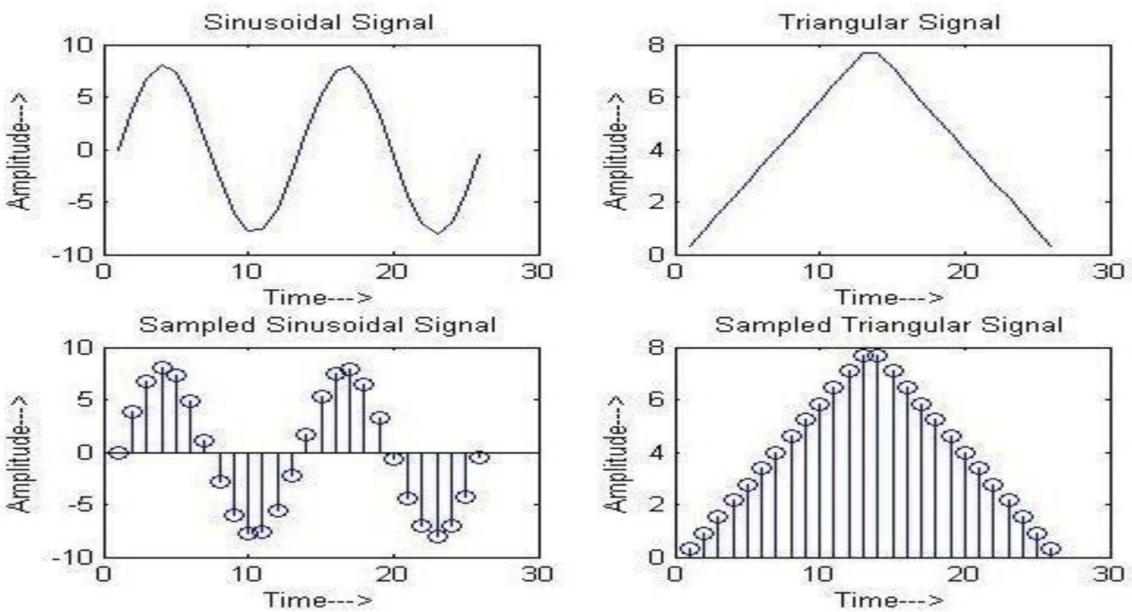
b) Software Tool: MATLAB 15.0 or Upgraded Version

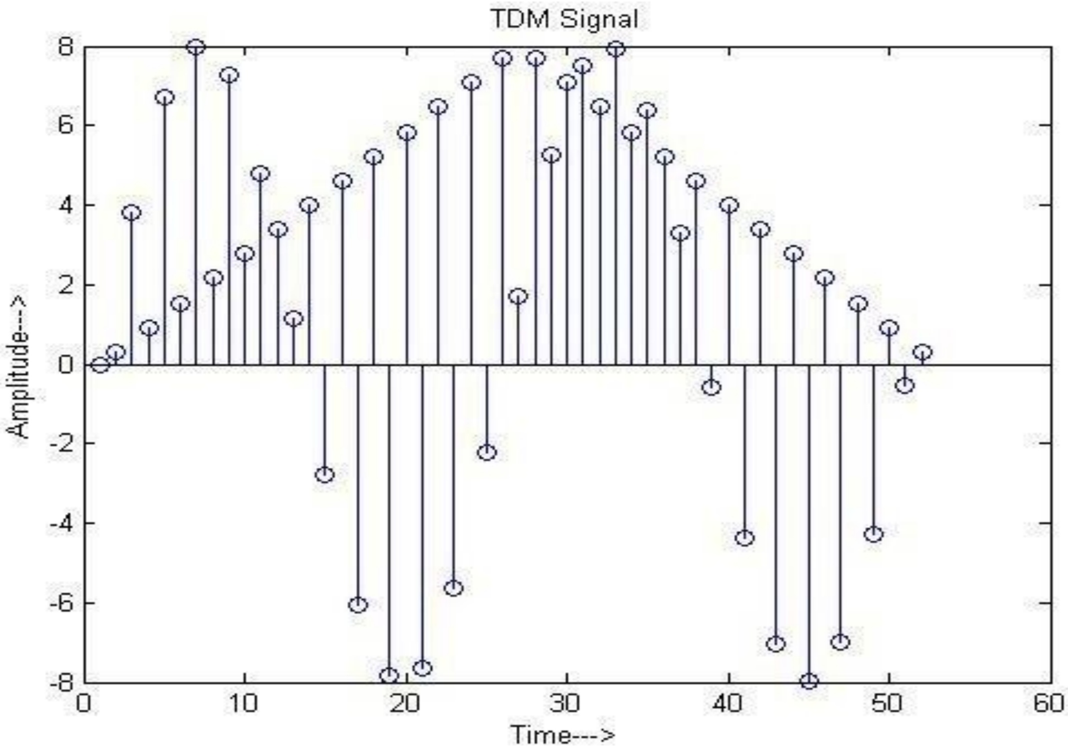
PROGRAM:

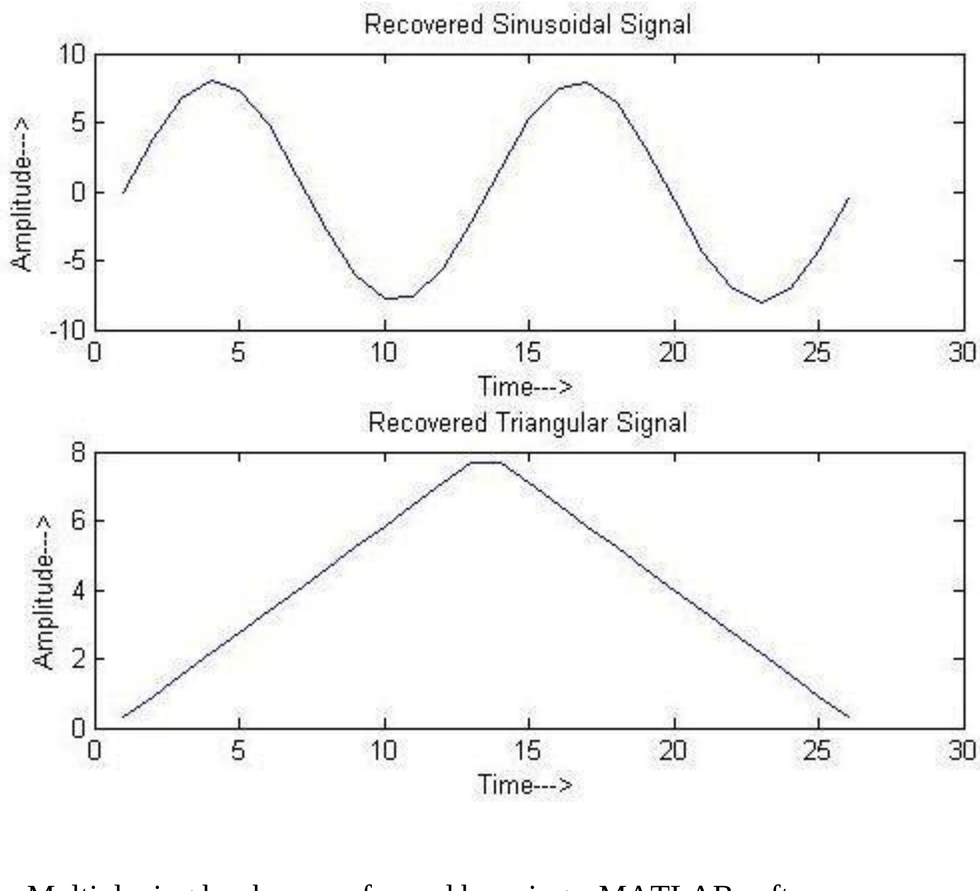
```
clc; close all; clear all;
% Signal generation
x=0:.5:4*pi;    % signal taken up to 4pi
sig1=8*sin(x); % generate 1st sinusoidal signal
l=length(sig1);
sig2=8*triang(l);         % Generate 2nd triangular Signal
% Display of Both Signal
subplot(2,2,1); plot(sig1); title('Sinusoidal Signal');
ylabel('Amplitude--->');
xlabel('Time--->');
subplot(2,2,2); plot(sig2); title('Triangular Signal');
ylabel('Amplitude--->');
xlabel('Time--->');
% Display of Both Sampled Signal
 subplot(2,2,3);
stem(sig1);
title('Sampled Sinusoidal Signal'); ylabel('Amplitude--->');
xlabel('Time--->'); subplot(2,2,4); stem(sig2);
title('Sampled Triangular Signal'); ylabel('Amplitude--->');
xlabel('Time--->'); l1=length(sig1); l2=length(sig2);
for i=1:l1
sig(1,i)=sig1(i);           % Making Both row vector to a matrix sig(2,i)=sig2(i);
end
% TDM of both quantize signal
 tdmsig=reshape(sig,1,2*l1);
% Display of TDM Signal figure
stem(tdmsig); title('TDM Signal'); ylabel('Amplitude--->');
xlabel('Time--->');
% Demultiplexing of TDM Signal
demux=reshape(tdmsig,2,l1);
```

```
for i=1:l1
sig3(i)=demux(1,i);     % Converting The matrix into row vectors sig4(i)=demux(2,i);
end
% display of demultiplexed signal figure
subplot(2,1,1);
 plot(sig3);
title('Recovered Sinusoidal Signal'); ylabel('Amplitude--->');
xlabel('Time--->');
subplot(2,1,2);
 plot(sig4);
title('Recovered Triangular Signal');
ylabel('Amplitude--->');
xlabel('Time--->');
```

OUTPUT WAVE FORMS







RESULT:

The time division Multiplexing has been performed by using a MATLAB software.

LAB QUESTIONS

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 2 signal band limited to 3 kHz, 5 KHz & are to be time division multiplexed. What is the maximum permissible interval between 2 successive samples.?

8. Is the bandwidth requirement for TDM & FDM will be same?

9. Is TDM system is relatively immune to interference with in channels (inter channel cross talk) as compared to FDM?

10. Is the FDM susceptible to harmonic distortion compared to TDM?

11. In what aspects, TDM is superior to FDM?

EXPERIMENT 12
Frequency Division Multiplexing & DeMultiplexing

AIM

To perform the Frequency Division Multiplexing with AM DSB-SC signals using Matlab.

APPARATUS REQUIRED

a) Hardware Tools: Computer system

b) Software Tool: MATLAB 15.0 and above version

THEORY

When several communications channels are between the two same point's significant economics may be realized by sending all the messages on one transmission facility a process called multiplexing. Applications of multiplexing range from the vital, if prosaic, telephone networks to the glamour of FM stereo and space probe telemetry system. There are two basic multiplexing techniques

1. Frequency Division Multiplexing (FDM)
2. Time Division Multiplexing (TDM)

The principle of the frequency division multiplexing is that several input messages individually modulate the subcarriers fc1, fc2 etc. after passing through LPFs to limit the message bandwidth. We show the subcarrier modulation as SSB, and it often is but any of the CW modulation techniques could be employed or a Mixture of them. The modulated signals are then summed to produce the baseband signal with the spectrum, the designation "baseband" is used here to indicate that the final carrier modulation has not yet taken place.

The major practical problem of FDM is cross talks, the unwanted coupling of one message into another. Intelligible cross talk arises primarily because of non linearity's in the system, which cause 1 message signal to appear as modulation on subcarrier. Consequently, standard practice calls for negative feedback to minimize amplifier non linearity in FDM systems

PROGRAM:

```
%program for frequency division multiplexing and demultiplexing
 close all; clear all; clc;
Fs = 100; % sampling freq t = [0:2*Fs+1]'/Fs;
x1 = sin(2*pi*2*t); % signal 1 signal z1 = fft(x1);
z1=abs(z1);
x2 = sin(2*pi*10*t); % signal 2 signal z2 = fft(x2);
z2=abs(z2); figure;
subplot(4,1,1); plot(x1);
title('signal 1');xlabel('time');ylabel('amplitude'); subplot(4,1,2); plot(x2);
title('signal 2');xlabel('time');ylabel('amplitude'); subplot(4,1,3); plot(z1);
title('Spectrum of signal 1');xlabel('freqency');ylabel('magnitude'); subplot(4,1,4); plot(z2);
title('Spectrum of signal 2');xlabel('freqency');ylabel('magnitude');
% freqency multiplexing
z=z1+z2; figure; plot(z);
title('frequency multiplexed signals'); figure;
% freqency demultiplexing
f1=[ones(10,1); zeros(182,1);ones(10,1)];%applying filter for signal 1 dz1=z.*f1;
d1 = ifft(dz1); subplot(2,1,1) plot(t*100,d1);
f2=[zeros(10,1); ones(182,1);zeros(10,1)];% applying filter for signal 2 dz2=z.*f2;
d2 = ifft(dz2);
title('recovered signal 1');xlabel('time');ylabel('amplitude'); subplot(2,1,2)
plot(t*100,d2);
title('recovered signal 2');xlabel('time');ylabel('amplitude');
```

OUTPUT WAVEFORMS:

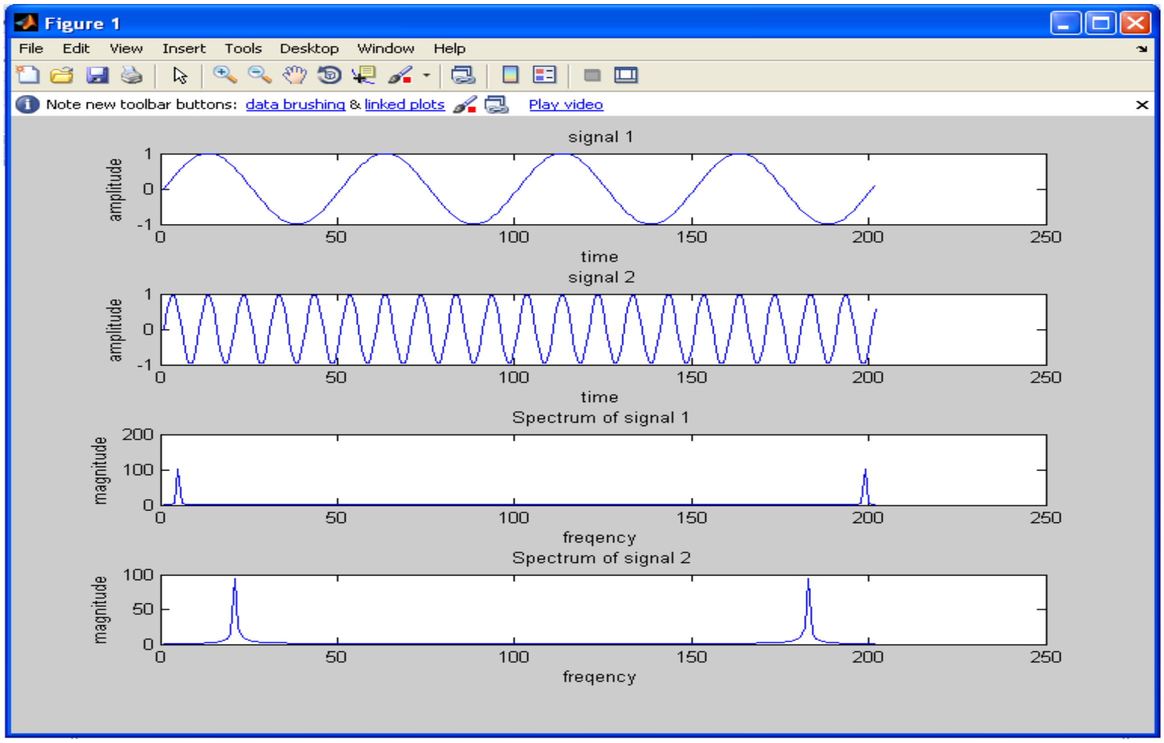


Figure 1: Signal 1 and 2 with their spectrums

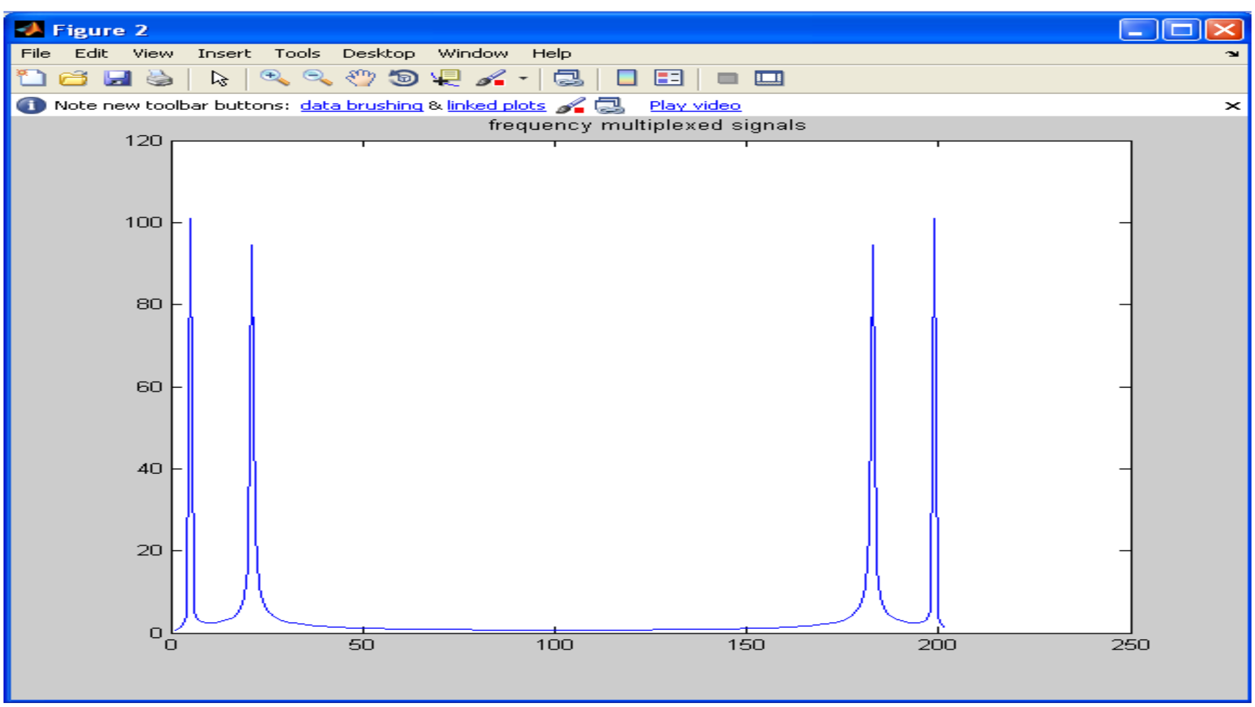


Figure 2: Frequency multiplexed signals

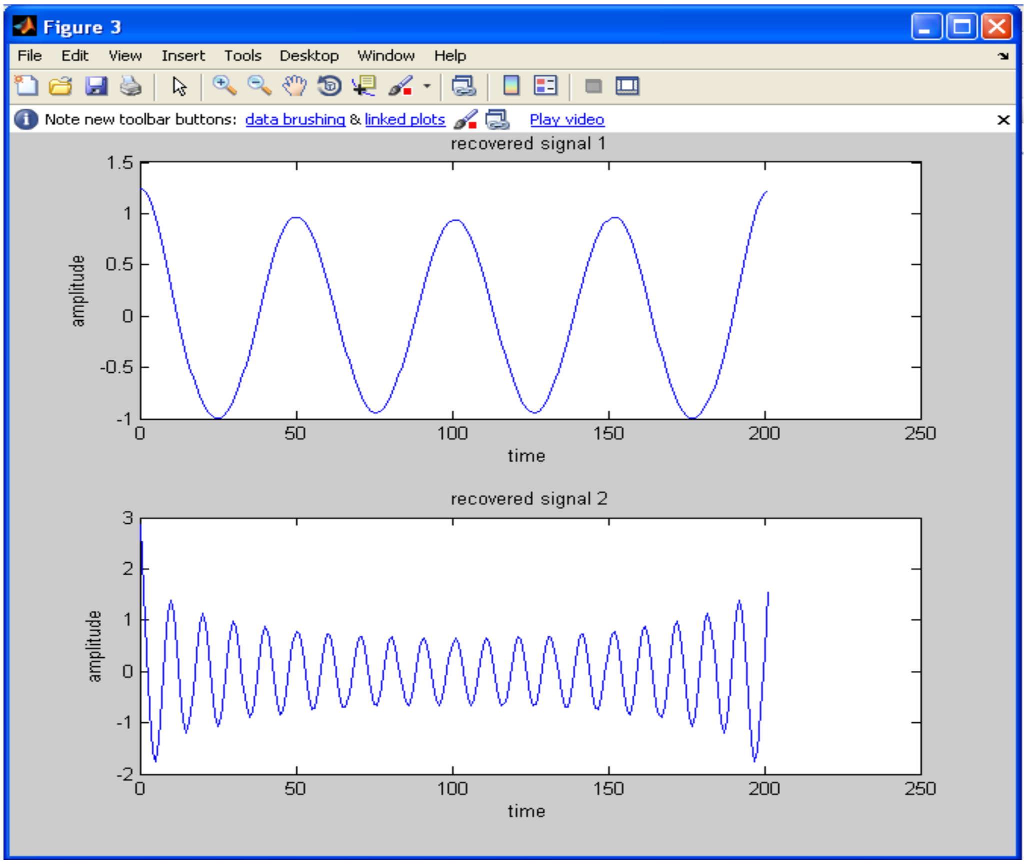


Figure 3: Recovered signal 1 and 2

RESULT:

The frequency division Multiplexing has been performed by using a MATLAB software.

LAB QUESTIONS

1. Draw the FDM signal with 2 signals being multiplexed over the channel?
2. Explain block schematic of FDM?
3. How TDM differ from FDM?
4. What are the applications of FDM?
5. Is the bandwidth requirement for TDM & FDM will be same?
6. Is TDM system is relatively immune to interference with in channels (inter channel cross talk) as compared to FDM?
7. Is the FDM susceptible to harmonic distortion compared to TDM?
8. In what aspects, TDM is superior to FDM?

EXPERIMENT 13
Generation and Detection of FM Signals

AIM

To perform the Frequency Modul**a**tion **s**ignal Gene**r**ation and Detection u**s**ing Matlab.

APPARATUS REQUIRED

a) Hardware Tools: Computer system

b) Software Tool: MATLAB 15.0 and above version.

FM with functions:

```
Clc;
clear all;
close all;
Fs = 8000; % Sampling rate of signal
 Fc = 100; % Carrier frequency
t = linspace(0,1,10000); % Sampling times
x = sin(2*pi*10*t) % Channel 1
dev = 50; % Frequency deviation in modulated signal
 y = fmmod(x,Fc,Fs,dev); % Modulate both channels.
z = fmdemod(y,Fc,Fs,dev); % Demodulate both channels.
subplot(411);
plot(t,x);
xlabel('time(sec)');
 ylabel('amplitude in volts(V)');
 title('MODULATING SIGNAL');
subplot(412);
plot(t,sin(2*pi*Fc*t));
xlabel('time(sec)');
ylabel('amplitude in volts(V)');
title('CARRIER SIGNAL');
subplot(413);
plot(t,y);
xlabel('time(sec)');
 ylabel('amplitude in volts(V)');
title('FREQUENCY MODULATED SIGNAL');
subplot(414);
plot(t,z);
xlabel('time(sec)');
ylabel('amplitude in volts(V)');
```

```
title('DEMODULATED SIGNAL');
```

OUTPUT WAVEFORMS:

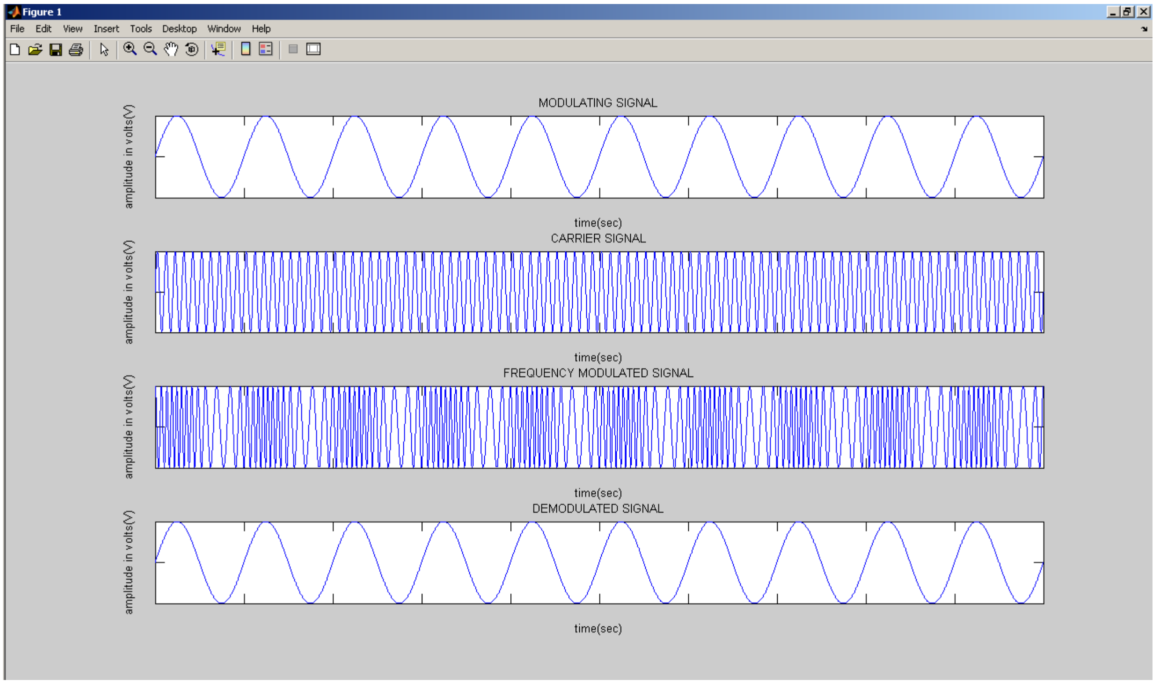


Figure 1: Frequency modulated and demodulated wave

RESULT:

The FM wave has been generated by using a MATLAB Software.

EXPERIMENT 14
Spectral Characteristics of AM & FM

AIM

To verify the Spectral Components of AM and FM using Matlab Simulink.

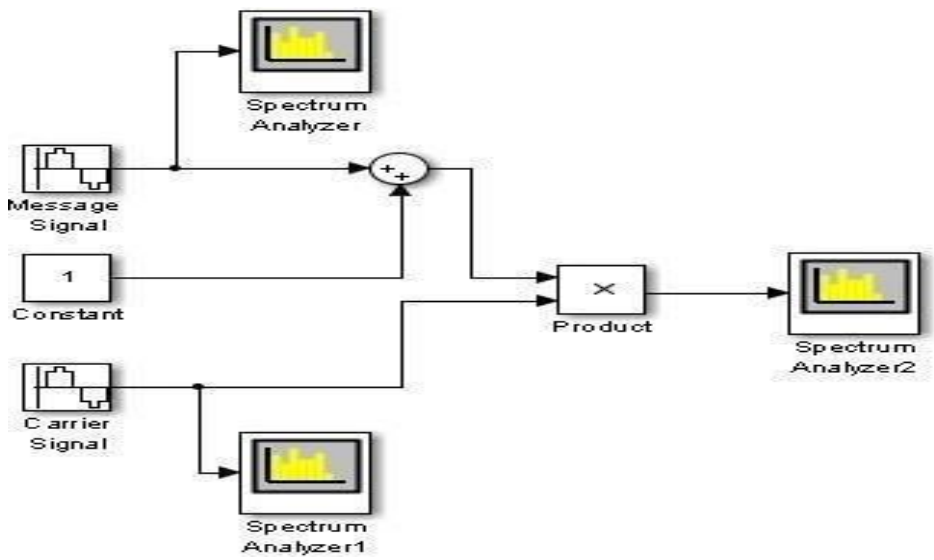
APPARATUS REQUIRED

a) Hardware Tools: Computer system

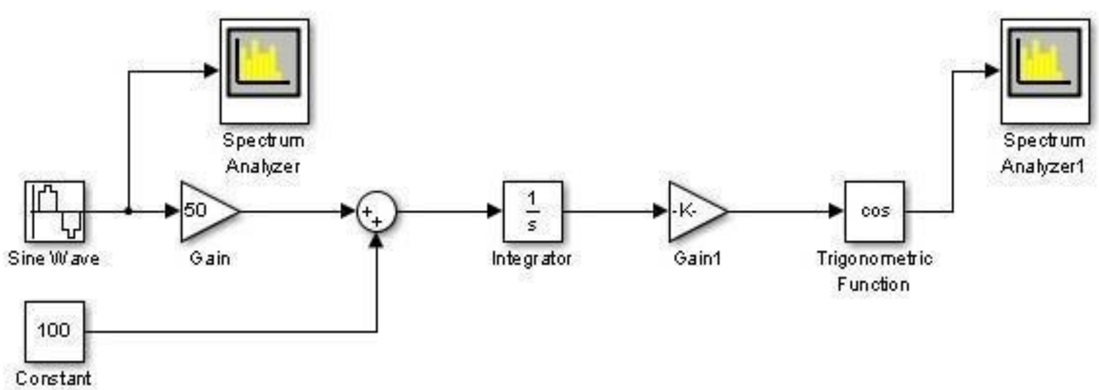
b) Software Tool: MATLAB 15.0 or Upgraded version.

SIMULINK MODEL

Amplitude Modulation Setup



Frequency Modulation Setup



PROCEDURE

1. Switch on the computer and click on the MATLAB icon.
2. Go to start at the bottom of the command window, then select “Simulink” then go to library browser and drag it into creating file. (or) Once you open the Matlab then click on the Simulink

icon . Go to file and select new and then select model. You will get a new window.

3. Arrange the functional blocks as shown in Simulink model.
4. Assign required parameters to each functional block.
5. Observe the outputs on scope.

PARAMETERS

For Amplitude Modulation:

Set the Message signal amplitude = 1V and frequency = 50 Hz

Set the Carrier signal amplitude = 1V and frequency = 300 Hz

For Frequency Modulation:

As per the setting of frequency modulation setup of experiment.

OUTPUT

For AM

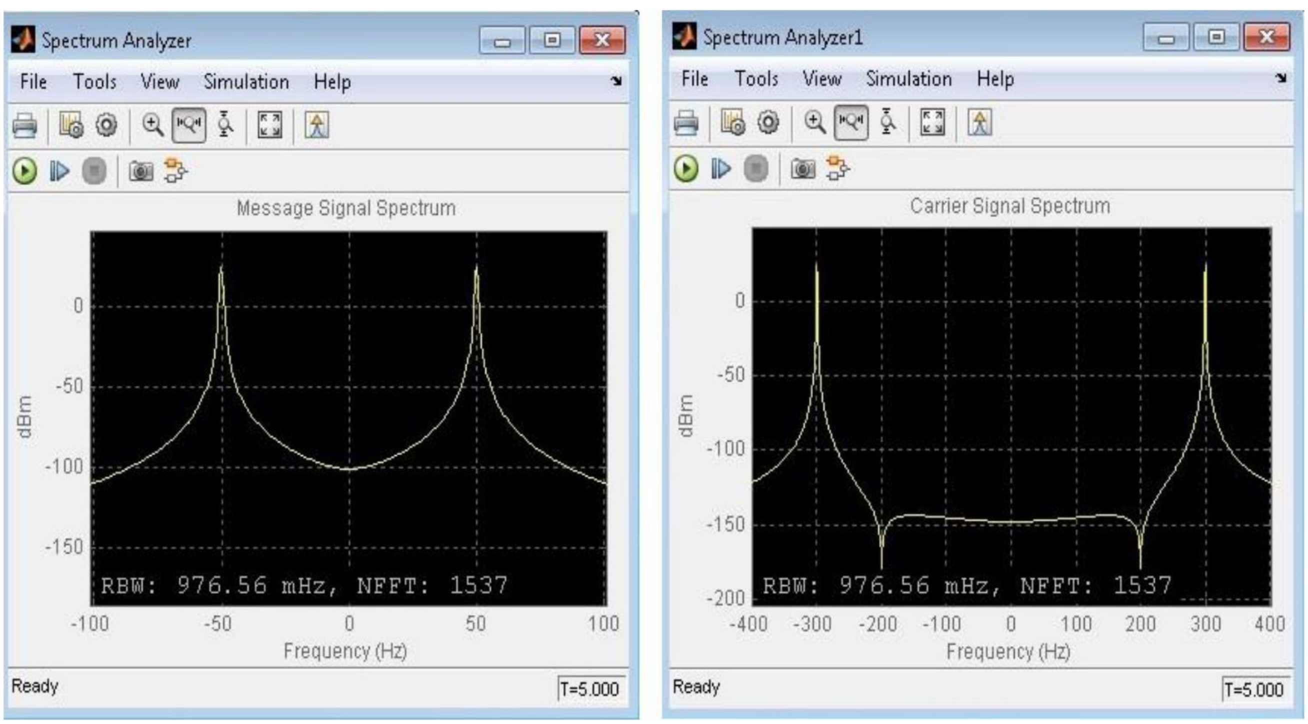


Figure 1: Message and carrier signal spectrum

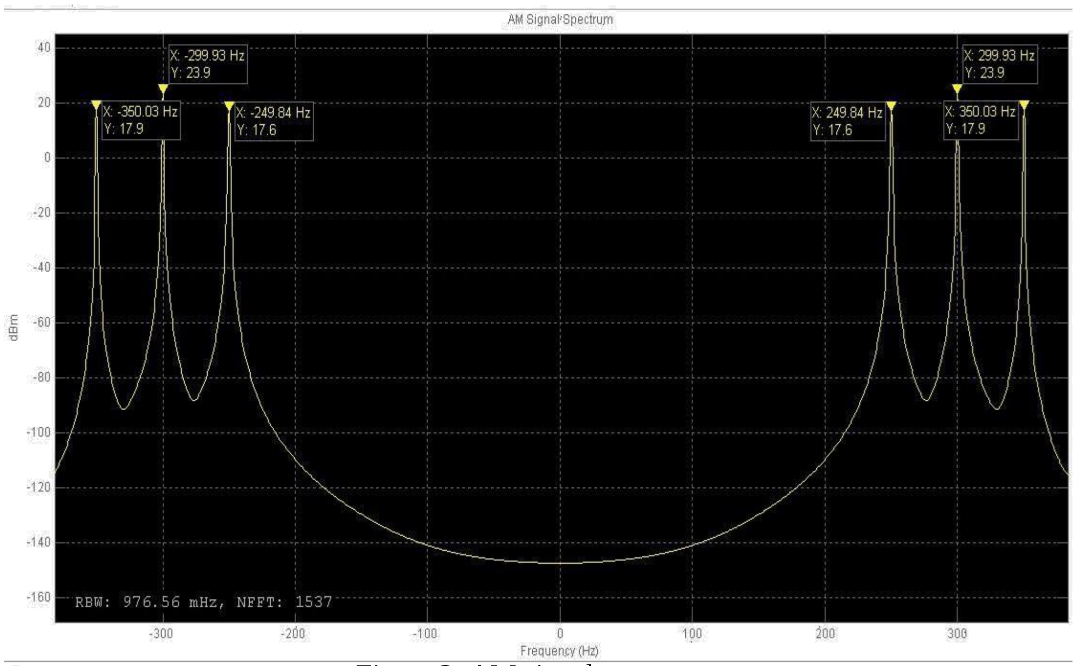


Figure 2: AM signal spectrum

For FM

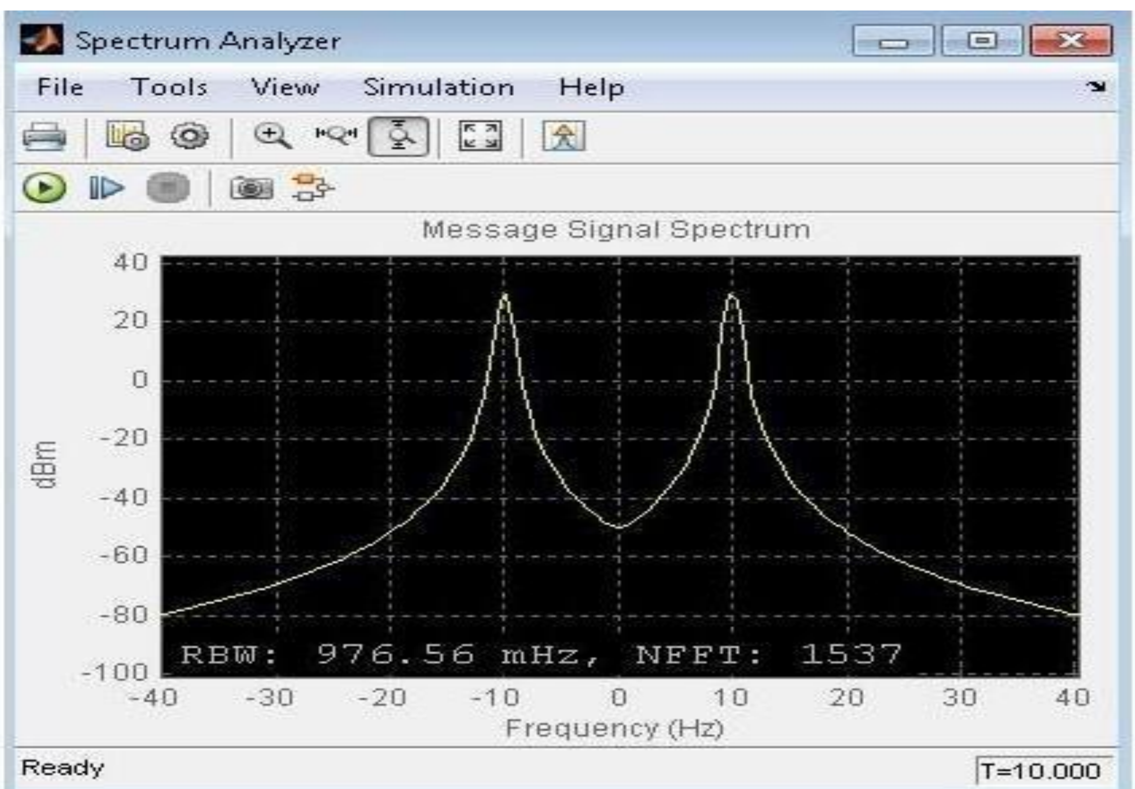


Figure 3: Message signal spectrum

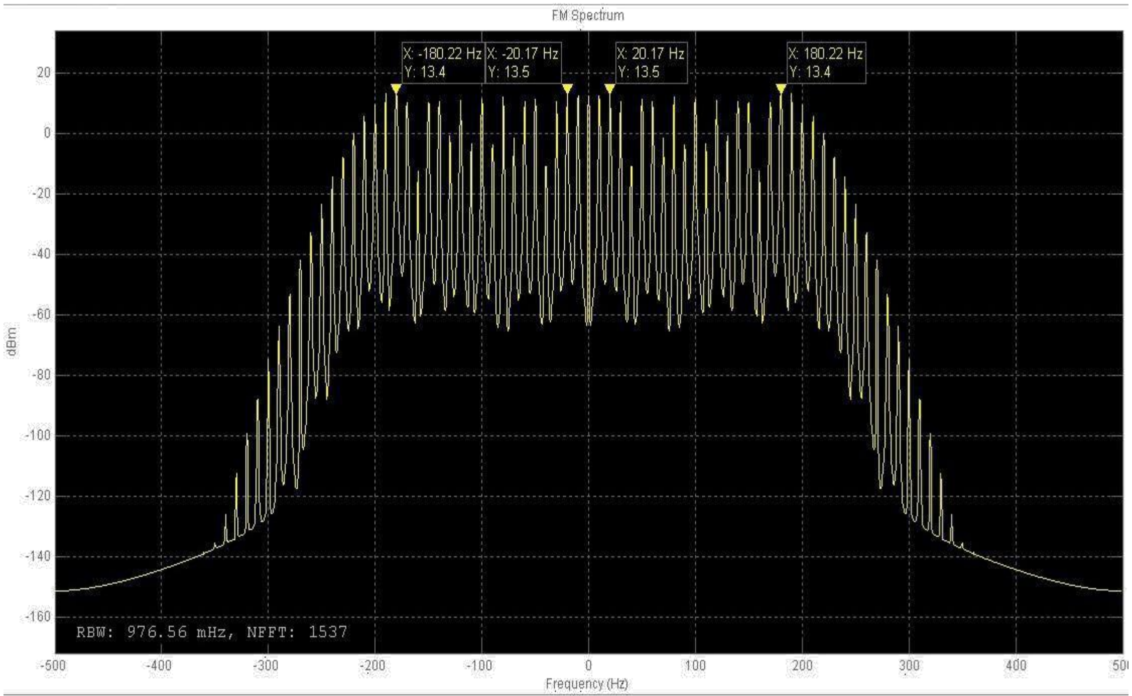


Figure 4: FM Spectrum

RESULT:

Spectrum of AM and FM gas been observed using MATLAB.