DRONACHARYA College of Engineering

LAB MANUAL

ANALOG COMMUNICATION SYSTEM (EC-417-F)

IV SEM (ECS)

DEPARTMENT OF ELECTRONICS & COMPUTER ENGG
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CONTENTS

SUBECT: ANALOG COMMUNICATION LAB (EC – 417-F)

EXPERIMENT 1

AIM: To study Amplitude Modulation (AM) using a transistor and determine Modulation Index or Depth of Modulation'm_a'.

APPARATUS REQUIRED:
(i) C.R.O.(ii) Transistor (BC -107) (iii)Diode (0A79) (iv) Resistor (v) Capacitor (vi) inductor (vii) Audio frequency Generator (viii) Connecting leads.

THEORY:
A message signal (audio or video signal) cannot be transmitted over long distances directly due to **attenuation, distortion, interference and noise** in any medium either wired or wireless. A suitable high frequency signal called carrier signal needs to be **modulated** by the low frequency message signal so that the message can be transmitted efficiently over long distances.

MODULATION:
Modulation is a scheme which alters some characteristics (amplitude, frequency or phase) of the high frequency signal, called the **carrier signal** in accordance with the instantaneous value of the low frequency message signal called the **modulating signal.** The resultant signal is called **modulated signal**. The carrier signal is periodic and continuous wave, it is termed as **Analog Modulation** and when the carrier signal is in the form of pulse, it is termed as **Digital Modulation.** When the amplitude of carrier signal is varied in accordance with the instantaneous value of amplitude of the message signal, the resultant modulation is termed as **Amplitude Modulation.**

AMPLITUDE MODULATION (AM):
It is a process in which the **maximum amplitude** of the **carrier wave** is **varied linearlyin accordance with instantaneous amplitude of modulating signal** or base band signal. The waves can be voltage or current signals. The waveforms of the carrier wave, modulating wave and the resultant modulated wave are shown below. If the base band signal consists of single frequency it is called as **single tone modulation.** If the base band signal consists of more than one frequency it is called as **multi tone modulation**.

EXPRESSION FOR AM WAVE AND FREQUENCY COMPONENTS :

Single tone modulating signal $X(t) = V_m \cos 2\pi f_m t = V_m \cos w_m t$…(1)

Carrier signal $C(t) = V_c \cos 2\pi f_c t = V_c \cos w_c t$ …..........(2)

Then the **modulated signal** is given by

$$S(t) = (V_c + V_m \cos w_m t)\cos w_c t \quad \text{……...............(3)}$$

On simplification we get

$$S(t) = V_c \{ 1+ (V_m / V_c)\cos w_m t \}\cos w_c t$$
$$= V_c \{ 1+ m_a \cos w_m t \}\cos w_c t$$
$$= V_c \cos w_c t + V_c m_a \cos w_m t \cos w_c t$$

$= V_c \cos w_c t + V_{c/2} m_a \cos (w_m + w_c)t + V_c/2 m_a \cos (w_m - w_c)t$

$= V_c \cos 2\pi f_c t + V_{c/2} m_a (\cos 2\pi (f_c + f_m)t + \cos 2\pi (f_c - f_m)t)$

……………………………………………………………………(4)

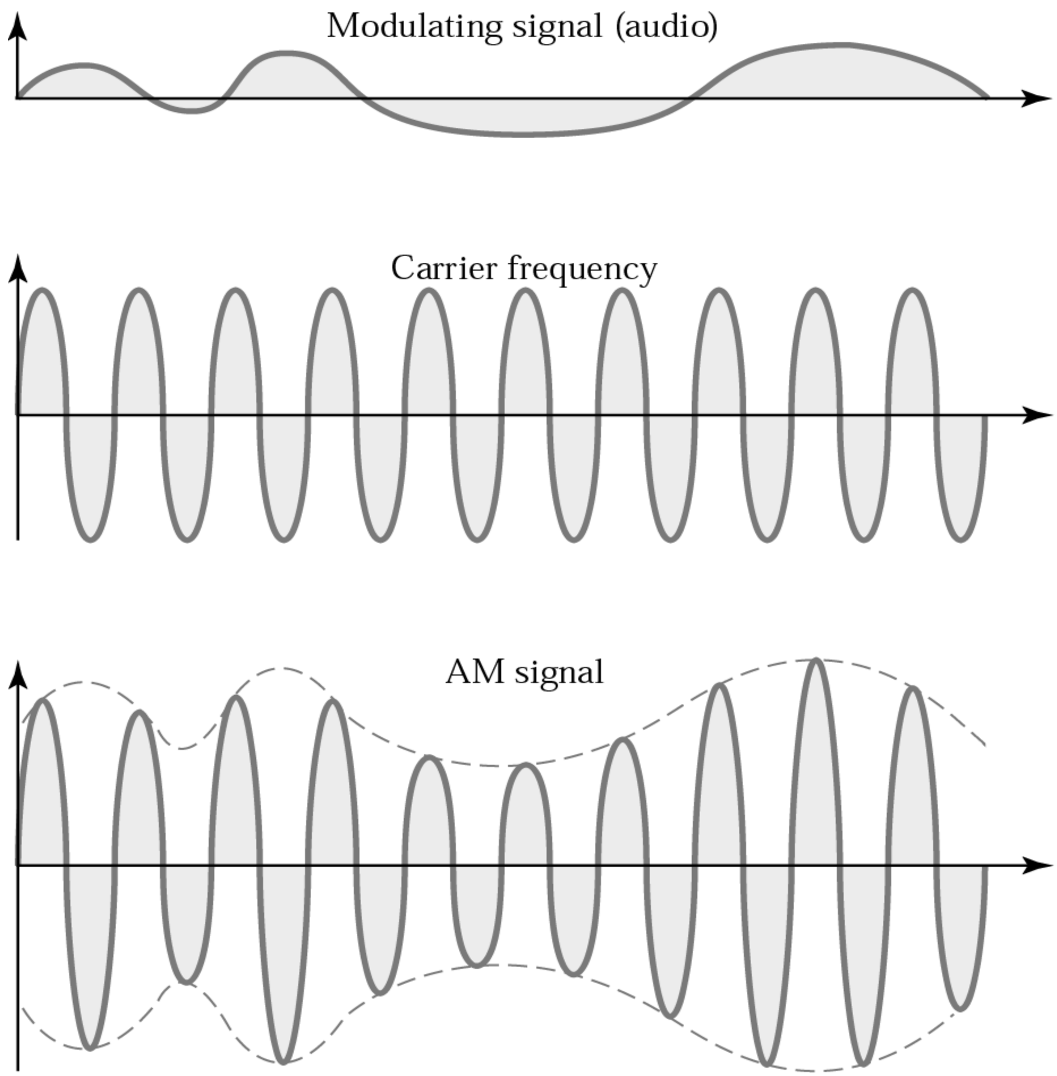


Figure 1: Waveforms of Modulating Signal, Carrier Signal and AM Signal

FREQUENCY SPECTRUM:

It is evident from the equation **(4)** that the frequency spectrum of the resultant modulated signal in case of **single tone modulation** consists of the original carrier frequency f_c as well as additional frequencies $(f_c + f_m)$ called as **upper side frequency** and $(f_c - f_m)$ called as **lower side frequency**. In case of **multi tone modulation** the base band contains a range of frequencies from ($f_m + \Delta f_m$) . In this case the frequency spectrum of the resultant modulated signal consists of the original carrier frequency f_c, an **upper side band** of frequencies $(f_c + f_m)$ to $(f_c + f_m + \Delta f_m)$ and a **lower side band** of frequencies $(f_c - f_m)$ to $(f_c - f_m - \Delta f_m)$.

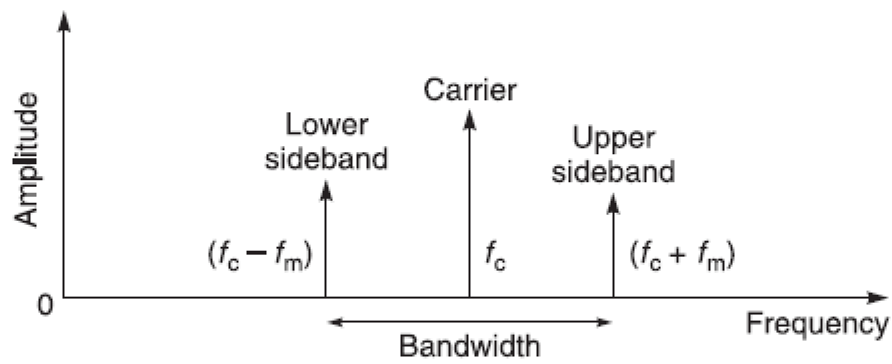


Figure 2: Frequency Spectrum (Carrier with two sidebands)

MODULATION INDEX:

The ratio of maximum amplitude of the modulating signal to the maximum amplitude of carrier wave is defined as the **amplitude modulation index** and denoted by 'm_a'. The modulation index is also known as **depth of modulation** or **degree of modulation** or **modulation factor.** Normally the value of 'm_a' lies between **0** and **1.** The modulation index is given by expression

$m_a = V_m / V_c$

where V_m = Maximum amplitude of the modulating signal

and V_c = Maximum amplitude of the carrier wave

It can be seen from the resultant of AM(single tone modulation) waveform

$V_m = (V_{max} - V_{min})/2$……………………………(5)

$V_c = (V_{max} + V_{min})/ 2$……………………………..(6)

From (5) & (6)

$m_a = (V_{max} - V_{min})/ (V_{max} + V_{min})$ ………………….(7)

OVERMODULATION:

If the value of 'm_a' exceeds **1,** then the percentage modulation is greater than **100** and the base band signal is not preserved in the envelope. In this case the base band signal recovered from the envelope by the demodulator of a receiver will be distorted. This type of distortion is called envelope distortion and AM signal is called **overmodulated**

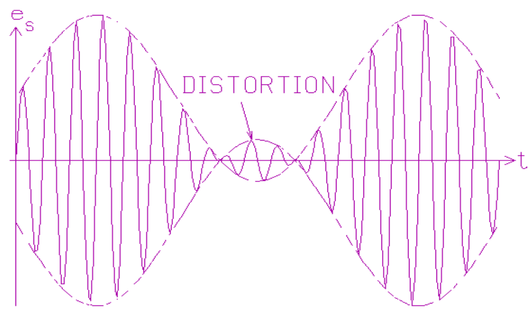


Figure 3: OverModulation (m_a> 1)

UNDERMODULATION:
If the value of 'm_a' is below**1,** then the percentage modulation is less than **100**. This type of AM signal is called **undermodulated**.

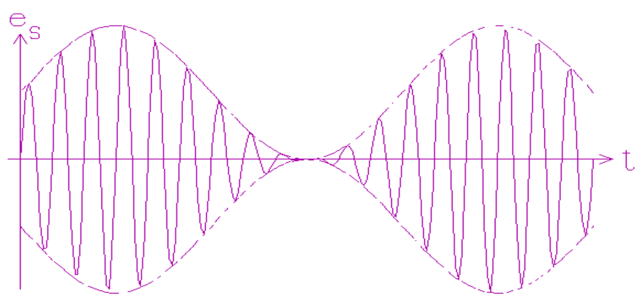


Figure 4: Unity Modulation (m_a = 1)

METHODS OF MODULATION & DEMODULATION:
Amplitude modulation is carried out by a circuit utilizing the nonlinear characteristic of solid state device like a **diode** called **Square law modulator** or collector circuit of a transistor called as **collector modulator.** It can be seen from the waveforms of the carrier wave, modulating wave and the resultant modulated wave that the envelope of the resultant waveform is identical to the modulating wave and thus utilized by the AM receiver for recovery of original message i.e. the modulating signal.

BENEFITS OF MODULATION:

(i) Efficient radiation
(ii) Reduced antenna height
(iii) Transmission from multiple sources using Multipliexing
(iv) Less interference and noise
(v) Strong signals

BLOCK DIAGRAM:

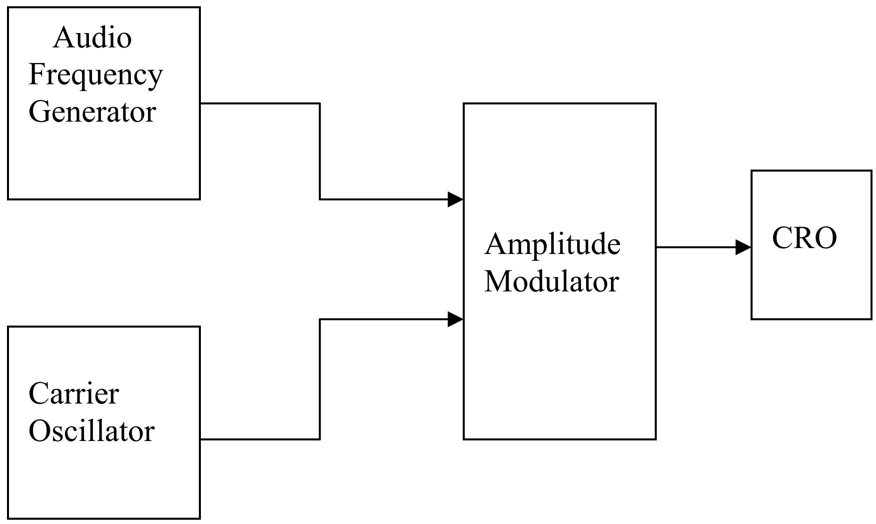


Figure 5: Block Diagram of Amplitude Modulation

PROCEDURE:

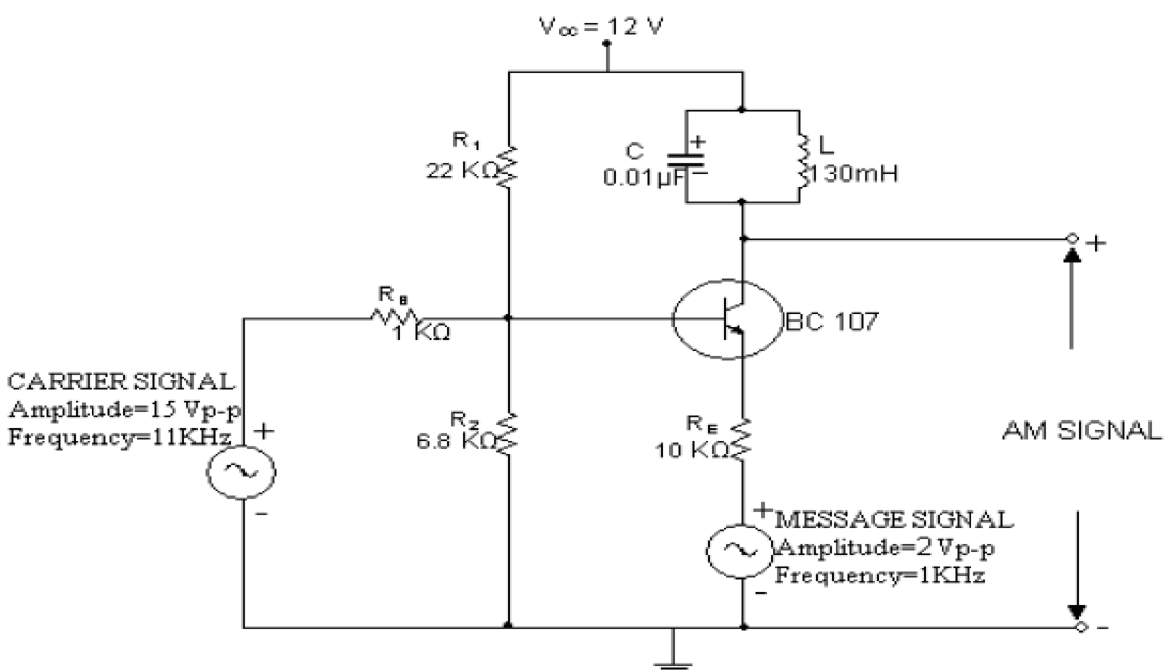


Figure 6: CircuitDiagram of Amplitude Modulation

1. Make the connection according to the circuit diagram shown in Figure 6.

2. Switch on the power supply.Connect **1 KHz, 2VAF Generator** to CRO and note the frequency and amplitude of AF output. .
3. Apply sinusoidal signal of 1 KHz frequency and amplitude 2 Vp-p as modulating signal and carriersignal of frequency 11 KHz and amplitude 15 Vp-p.
4. Observe the carrier signal, modulating signal and AM signal in CRO.
5. Now slowly increase the amplitude of the modulating signal up to 7V in steps of 1V and note down values of Vmaxand Vmin.
6. Calculate **amplitude modulation index**m_a=.$(V_{max} - V_{min}) / (V_{max} + V_{min})$
7. Change the amplitude **of RF carrier** to different values keeping the **the amplitude of AF Generator** constant and note down the corresponding values of V_{max} and V_{min}
8. Repeat step No.7 and tabulate the results.

OBSERVATION TABLE:

S.NO.	AFSignal		RF Signal.		V_{max}.	V_{min}.	Modulation index
	Frequency	Amplitude	Frequency	Amplitude			
1 2 3 4 5							

SAMPLE CALCULATION:

m_a = (Vmax. – Vmin) / (Vmax. + Vmin)

m_a = (10 -6) / (10 +6)

= .25

RESULT:

Generation of AM Wave with different amplitude and frequency of modulating signal and carrier signal using a transistor is observed and modulation index is calculated.

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.

QUIZ / ANSWERS:

Q. 1. What is modulation?
Ans. Modulation is a process in which the amplitude of the carrier is made proportional to the instantaneous amplitude of the modulating signal.

Q. 2. Which are the three discrete frequencies in AM?
Ans. (1) Carrier frequency (2) lower sideband frequency (3) upper sideband frequency

Q. 3 How many sidebands in AM?
Ans. There are two sidebands in AM i.e. LSB and USB.

Q. 4.Explain why modulation is necessary or desirable.
Ans.Practicability of antenna Size, Narrowbanding, Multiplexing, Less Interference, Strong Signals.

Q. 5. Name the circuit that causes one signal to modulate another,and give the names of the two signals applied tothis circuit.
Ans. The circuit is called modulator and the signal applied are modulating signal and carrier signal.

Q. 6. In AM, how does the carrier vary in accordance withthe information signal?
Ans. The amplitude of carrier signal varies according to instantaneous amplitude of modulating signal.

Q. 7. True or false? The carrier frequency is usually lowerthan the modulating frequency.
Ans. False, Carrier frequency is much higher than modulating signal.

Q. 8. What is the outline of the peaks of the modulated signalcalled, and what shape does it have?
Ans. The outline of the modulated signal is called Envelop and its shape is same as modulating signal.

Q. 9. True or false? The carrier frequency remains constantduring AM.
Ans. True, frequency of modulated carrier signal remains constant in original carrier signal.

Q. 10. What mathematical operation does an amplitude modulatorperform?
Ans. Multiplication and addition.

EXPERIMENT 2

AIM: To study envelope detector for 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 Amplitude Modulation Transmitter:
The transmitter circuits produce the amplitude modulated signals which are used tocarry information over the transmission to the receiver. The main parts of thetransmitter are shown in Fig.11.In Fig.11 & 12, we can see that the peak-to-peak voltage in the AM waveformincrease and decrease in sympathy with the audio signal.

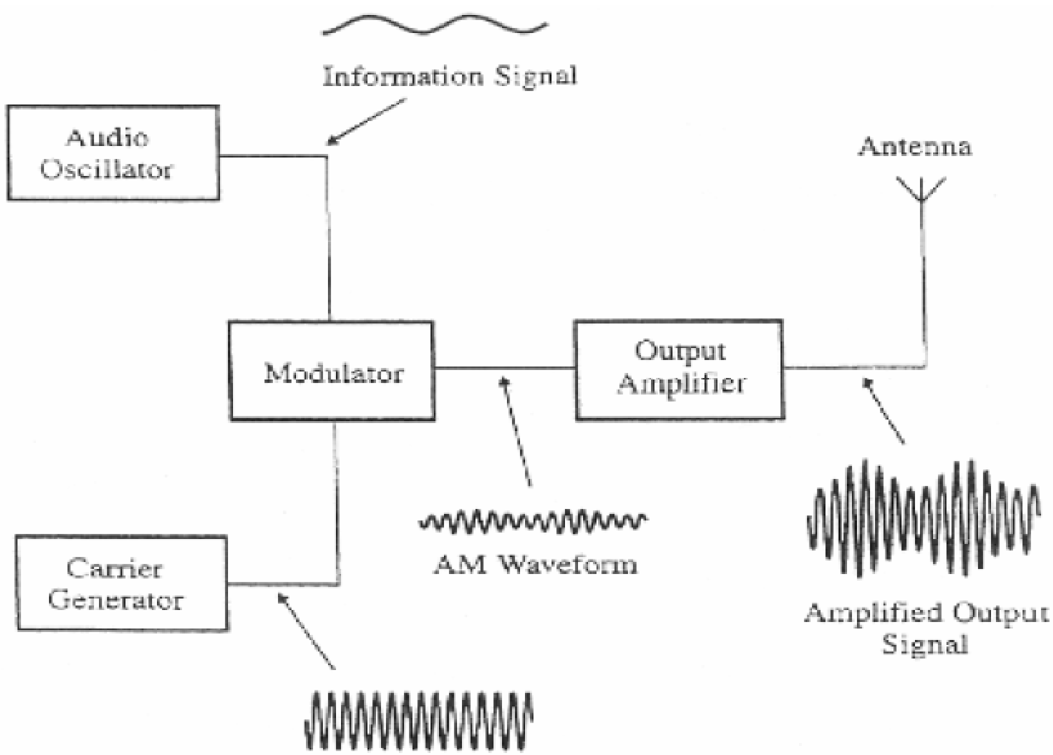


Fig. 1: AM Transmitter System

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.12. 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.

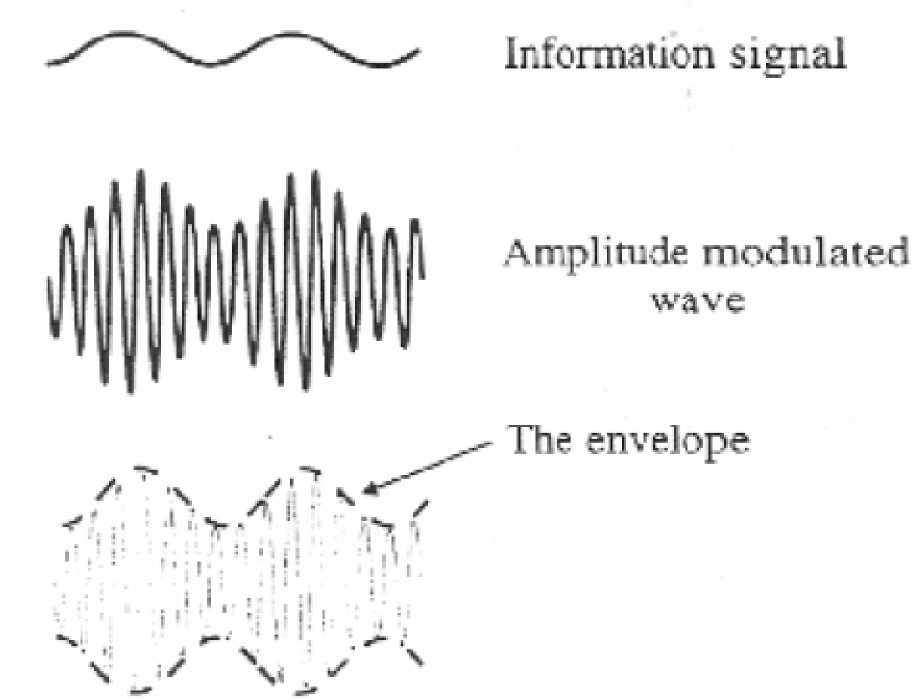


Figure 2: Waveforms in AM transmitter

AM Reception:
The AM 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. :

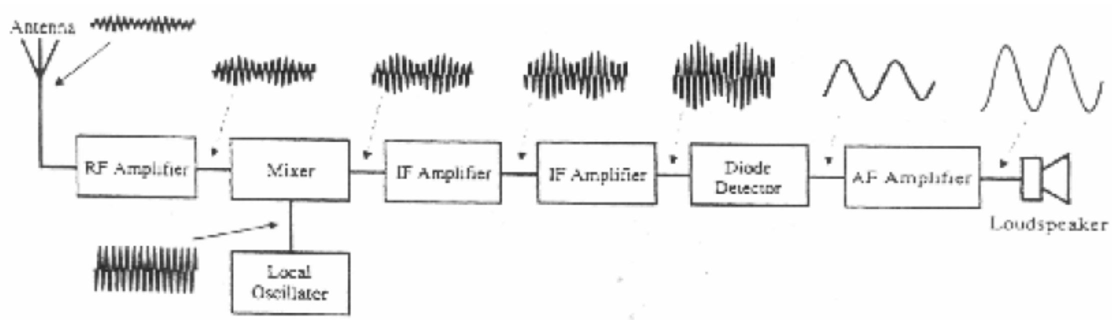


Figure 3: AM Reception

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 theoutput of the IF amplifiers. It performs this task in a very similar way to a half waverectifier converting an AC input to a DC output. Fig.4 shows a simple circuitdiagram of the diode detector.

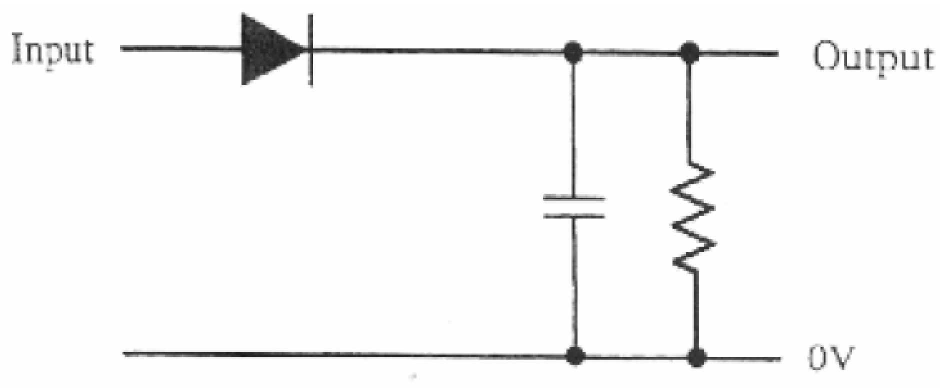


Figure 4: Diode Detector

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

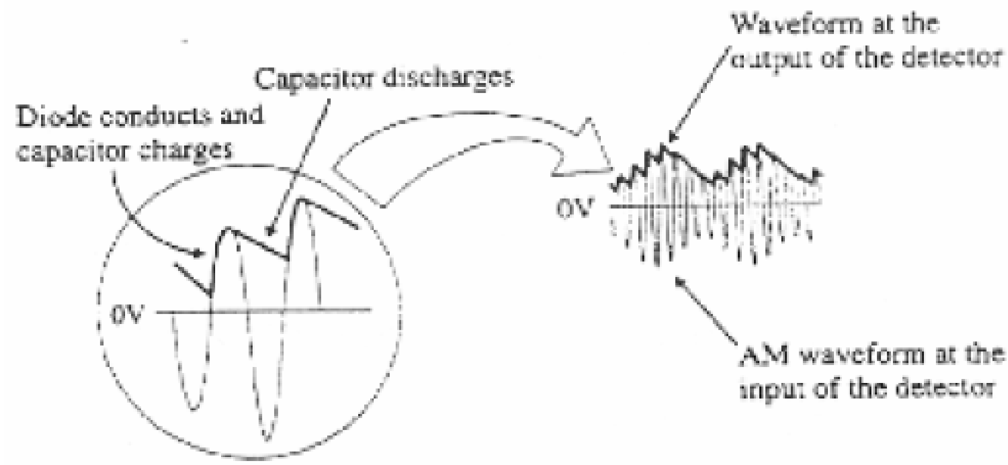
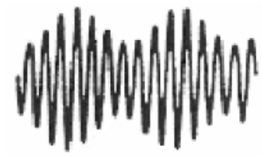


Fig. 5 Clipping in Diode Detector

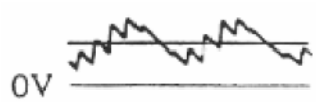
The result is an output which contains three components :

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 rippleand a capacitor blocks the DC voltage level. Fig.6shows the result of theinformation signal passing through the diode detector and audio amplifier. The remaining audio signals are then amplified to provide the final output to theloudspeaker.



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

PROCEDURE:

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, infull 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.

Note: *If desired, headphones (supplied with the module) may be used instead ofthe on-board loudspeaker. To use the headphones, simply plug the headphonejack into the audio amplifier block's headphones socket, and adjust controlledblock's volume pot.*

6. The first stage or 'front end' of the **ST2202** AM receiver is the R.F amplifierstage. This is a wide - bandwidth tuned amplifier stage, which is tuned into thewanted station by means of the tuning dial.Once it has been tuned into the wanted station, the R.F. amplifier, having littleselectivity, will not only amplify, but also those frequencies that are close to thewanted frequency. As we will see later, these nearby frequencies will beremoved by subsequent stages of the receiver, to leave only the wanted signal.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 ofmillivolts at the most). This is because one stage of amplification is notsufficient 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. This is because there are many unwanted frequencies getting throughto the amplifier output, which tend to 'drown out' the wanted AM Signal.You may notice that the waveform itself drifts up and down on the scopedisplay, indicating that the waveform's average level is changing. This is due tothe operation of the AGC circuit, which will be explained later.

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 thelocal oscillator is also tuned by means of the tuning dial, and is arranged so thatits 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 itsfrequency is always 455 KHz above the signal frequency that the R.F. amplifieris tuned to. This fixed frequency difference is always present, irrespective of theposition of the tuning dial, and is known as the intermediate frequency (IF forshort). This frequency relationship is shown below, for some arbitrary positionof the tuning dial.

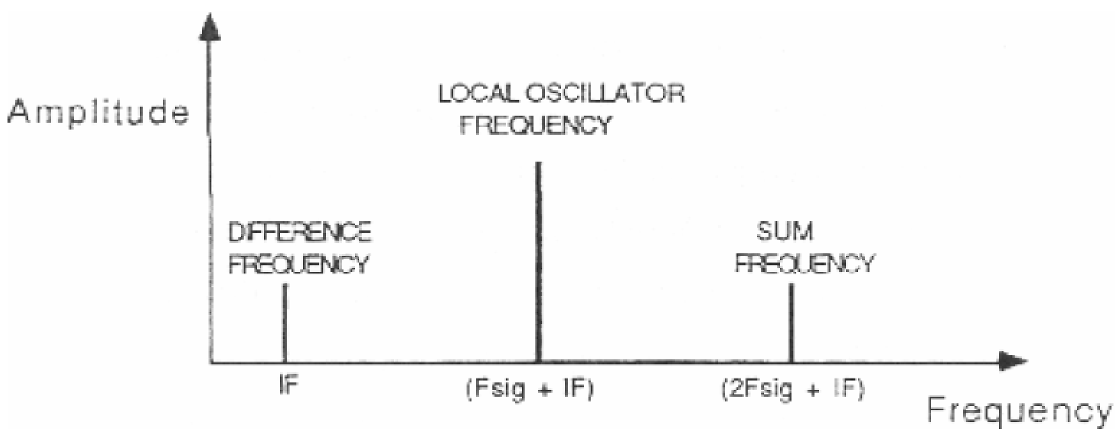


Figure 7: Frequency Contents in DSB AM

Examine the output of the local oscillator block, and check that its frequencyvaries as the tuning dial is turned. Re-time the receiver to a radio station.

8. The operation of the mixer stage is basically to shift the wanted signal down tothe IF frequency, irrespective of the position of the tuning dial. This is achievedin two stages.

a. By mixing the local oscillator's output sinewave with the output from theR.F. amplifier block. This produces three frequency components:

The local oscillator frequency = (f sig + IF)

The sum of the original two frequencies, f sum = (2 f sig + IF)

The difference between the original two frequencies,

b. By strongly attenuating all components. Except the difference frequency,IF this is done by putting a narrow-bandwidth band pass filter on themixer's output.

The end result of this process is that the carrier frequency of the selected AMstation is shifted down to 455 KHz (the IF Frequency), and the sidebands of theAM signal are now either side of 455 KHz.

9. Note that, since the mixer's band pass filter is not highly selective, it will notcompletely remove the local oscillators and sum frequency components fromthe mixer's output. this is the case particularly with the local oscillatorcomponent, which is much larger in amplitude than the sum and differencecomponents.Examine the output of the mixer block (t.p. 20) with an a.c. coupledoscilloscope channel, and note that the main frequency component presentchanges as the tuning dial is turned. This is the local oscillator component,which still dominates the mixer's output, in spite of being attenuated by themixer's band pass filter.

10. Tune in to a strong broadcast station again and note that the monitored signalshows little, if any, sign of modulation. This is because the wanted component,which is now at the IF frequency of 455 KHz, is still very small in component,which is now at the IF frequency of 455 KHz, is still very small in comparisonto the local oscillator component.What we need to do now is to preferentially amplify frequencies around 455KHz, without amplifying the higher-frequency local oscillator and SUMcomponents.This selective amplification is achieved by using two IF amplifier stages, IFamplifier 1 and IF amplifier 2, which are designed to amplify strongly a narrowband of frequencies around 455 KHz, without amplifying frequencies on eitherside of this narrow band.These IF amplifiers are basically tuned amplifiers which have been pre-tuned tothe 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 thisnarrow frequency band will not be amplified.Examine the output of IF amplifier 1 (at. t.p. 24) with an a.c.-coupledoscilloscope channel, and note that:

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

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

c. The envelope of the signal is modulated in amplitude, according to thesound 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 oscilloscopechannel, noting that the amplitude of the signal has been further amplified bythis second IF amplitude of the signal has been further amplified by this secondIF amplifier stage.IF amplifier 2 has once again preferentially amplified signals around the IFfrequency (455 KHz), so that:

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

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

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

12. The next step is extract this audio information from the amplitude variations ofthe signal at the output of IF amplifier 2. This operation is performed by thediode detector block, whose output follows

the changes in the amplitude of thesignal 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 atthe 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 peakamplitude of the incoming signal. We will see how we make use of thisoffset later on, when we look at automatic gain control (AGC).
13. The final stage of the receiver is the audio amplifier block contains a simplelow-pass filter which passes only audio frequencies, and removes the highfrequencyripple from the diode detector's output signal. This filtered audiosignal is applied to the input of an audio power amplifier, which drives on boardloudspeaker (and the headphones, if these are used). The final result is thesound 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 minimumvolume 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 'cleanedup' the audio signal.You may notice that the output from the audio amplifier block (t.p. 39) isinverted with respect to the signal at the output of the diode detector (t.p. 31)this inversion is performed by the audio power amplifier IC, and in no wayaffects the sound produced by the receiver.
14. Now that we have examined the basic principles of operation of the **ST2202**receiver for the reception and demodulation of AM broadcast signals, we willtry receiving the AM signal from the **ST2201** transmitter.Presently, the gain of **ST2201'**s output amplifier block is zero, so that there isno output from the Transmitter. Now turn the gain pot in **ST2201**'s outputamplifier block to its fully clockwise (maximum gain) position, so that thetransmitter 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 thesignal'.Since **ST2201** TX output select switch is in the ANT position, the AM signal att.p.13 is fed to the transmitter's antenna. Prove this by touching **ST2201**'santenna, and nothing that the loading caused by your hand reduces theamplitude of the AM waveform. at t.p.13.The antenna will propagate this AM signal over a maximum distance of about1.4 feet. We will now attempt to receive the propagated AM waveform with the**ST2201/ ST2202** board, by using the receiver's on board antenna.
Note: *If more than one* **ST2201** *transmitter/receiver system is in use at onetime, it is possible that there may be interference between nearby transmitters ifantenna propagation is used. To eliminate this problem, use a cable betweeneach transmitter/receiver pair, connecting it between* **ST2201**'*s TX outputsocket and* **ST2201/ST2202**'*s RX input socket. If you do this, make sure that thetransmitter's TX output select switch, and the receiver's RX input select switch,are both in the SKT position, then follow the steps below as though antennapropagation were being used.*
15. On the **ST2201** module, turn the volume pot (in the audio amplifier block) clock-wise, until you can hear the tone of the audio oscillator’s output signal,from the loudspeaker on the board.
Note: If desired, headphones may be used instead of the loudspeaker on theboard. To use the headphones, simply plug the headphone jack into the audioamplifier block's headphones socket, and put the speaker switch in the OFFposition. The volume from the headphones is still controlled by the block'svolume pot. Turn the volume pot to the full counter-clockwise (minimumvolume) position.
16. On the **ST2201/ST2202** receiver, adjust the volume pot so that the receiver'soutput can be clearly heard. Then adjust the receiver's tuning dial until the tonegenerated at the transmitter is also clearly audible at the receiver (this should bewhen the tuning dial is set to about 55-65 and adjust the receiver's

volume potuntil the tone is at a comfortable level.Check that you are tuned into the transmitter's output signal, by varying**ST2201**'s frequency pot in the audio oscillator block, and nothing that the tonegenerated by the receiver changes.
The **ST2201/2202** receiver is now tuned into AM signal generated by the**ST2201** transmitter. Briefly check that the waveforms, at the outputs of thefollowing receiver blocks, are as expected:
R. F. Amplifier (t.p.12)
Mixer (t.p.20)
I.F. Amplifier 1 (t.p.24)
I.F. Amplifier 2 (t.p.28)
Diode Detector (t.p.31)
Audio Amplifier (t.p.39)
17. By using the microphone, the human voice can be used as transmitter's audiomodulating signal, instead of using **ST2201'**s audio oscillator block. Use DSBand not DSBSC.Connect the microphone's output to the external audio input on the **ST2201**board, and put the audio input select switch in the EXT position.
18. In the output of diode detector peak diagonal clipping can be observed at low values of time constant of tuning circuitry.

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.

QUIZ / ANSWERS:

Q. 1. What are the methods for the generation of AM Wave?
Ans. The methods of AM generations are classified as:
a) Low Level AM modulation
b) High Level AM modulation

Q. 2. Which do you mean by low level AM generation?
Ans. A very small power is associated with the carrier signal and modulating signal and hence output power of modulation is low.

Q. 3. Which do you mean by high level AM generation?
Ans. Modulation is done at high power level, therefore the carrier signal and modulating signal must be at high power levels, and hence output power of modulation is high.

Q. 4.Why do we need wide-band and narrow-band power amplifiers in high level AM modulation?
Ans. Wide-band power amplifier is required to preserve all the frequency components of the modulating signal and narrow-band power amplifier is required for a fixed –frequency carrier signal.

Q. 5. What is square law diode modulation?
Ans. Modulation is done at the transmitter. The square law modulation circuit makes use of non-linar current voltage characteristics of diode.

Q. 6. What is square law demodulation?
Ans. DeModulation is done at the receiver end where the original signal is recovered from the received signal. The square law detector circuit makes use of non-linar current voltage characteristics of diode and for detecting of modulated signal of small magnitude (below 1 volt).

Q. 7. What is the transmission efficiency of AM wave?
Ans. Transmission efficiency is amount of useful power in the modulated signal.
Transmission efficiency $\acute{\eta} = P_s/P_t * 100$ where P_t is total transmitted power and P_s in power in sidebands.

EXPERIMENT 3

AIM: To generate DSB-SC AM signal using balanced modulator.

APPARATUS REQUIRED:
(i) C.R.O. (ii) CRO Probe (ii)DSB/SSB Transmitter (ST 2201) and Receiver (ST2202) Trainer (iv) Connecting leads.

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

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

Generally, and in the context of this experiment, it is understood that: $\omega >> \mu$ (2)

Equation (3) can be expanded to give:

$\cos\mu t \cdot \cos\omega t = (E/2)\cos(\omega - \mu)t + (E/2)\cos(\omega + \mu)t$ (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.

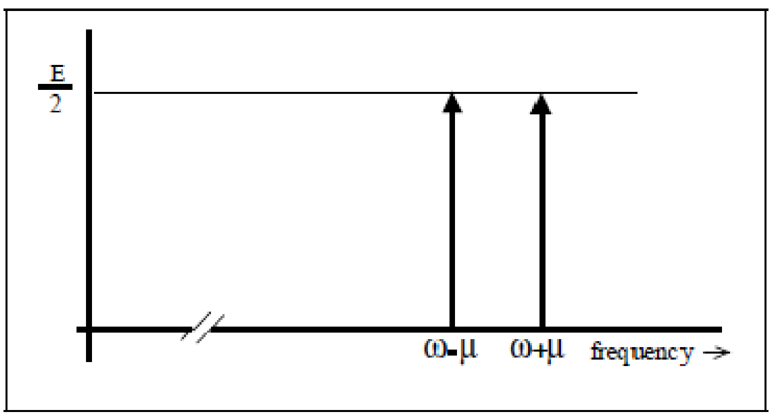


Figure 1: Spectral components

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 referredas 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).

The time domain appearance of a DSBSC (eqn. 1) in a text book is generally as shown in Figure 2.

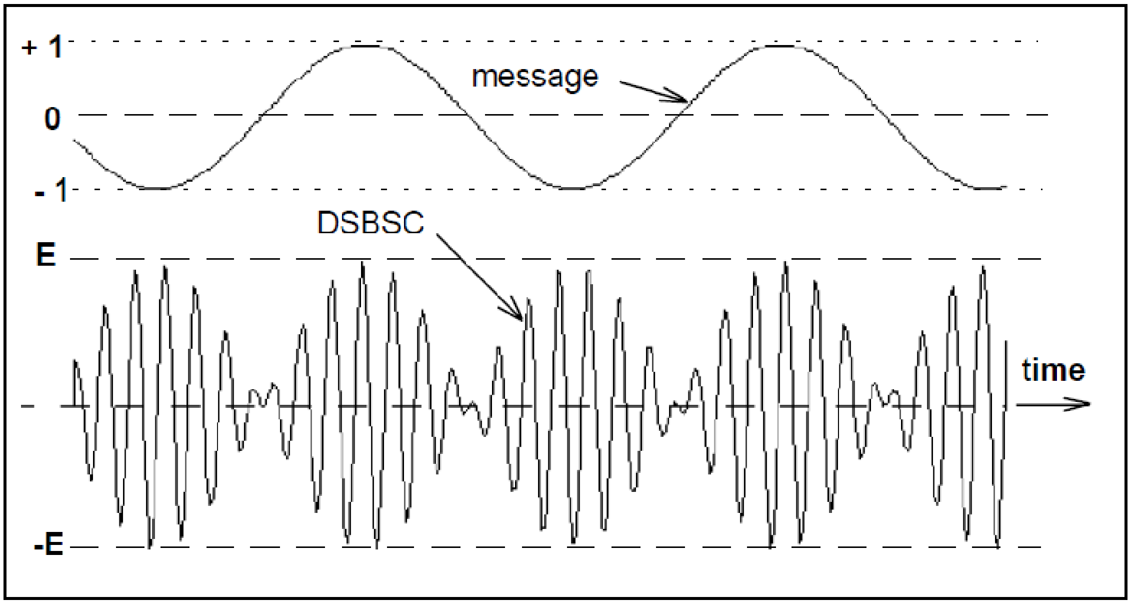


Figure 2: DSBSC - seen in the time domain

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.

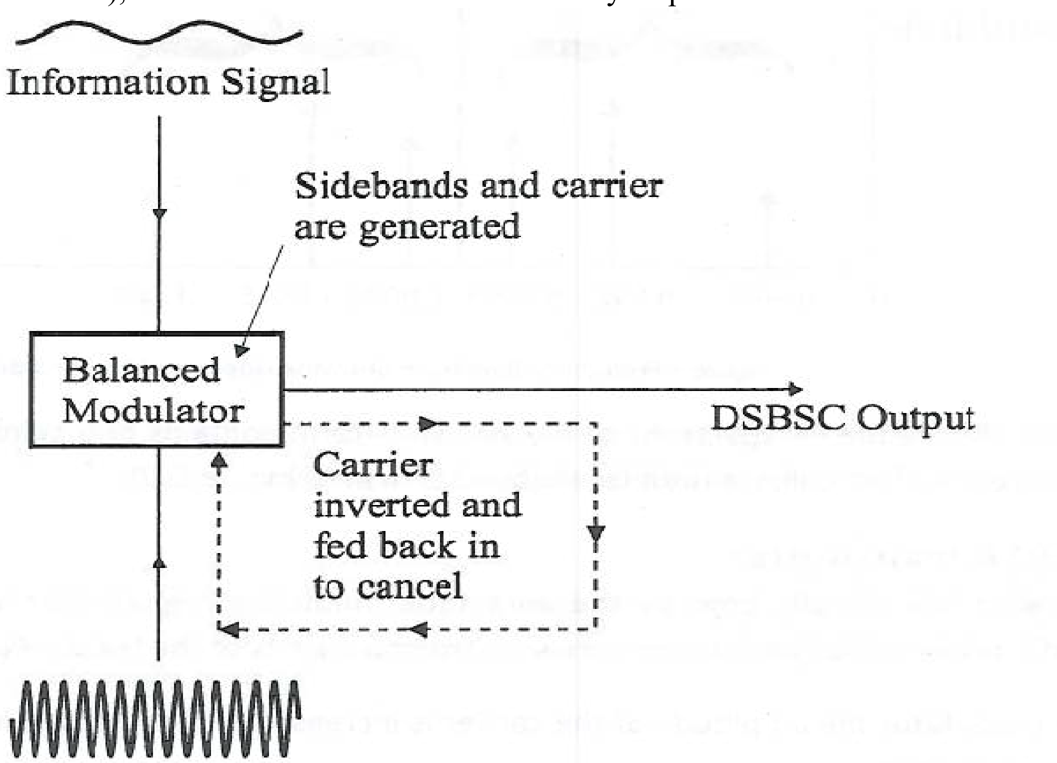


Figure 3: DSBSC Generation using balanced modulator

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, $BW_{DSB\text{-}SC} = 2B$ (Hz).

(c) The modulated signal is centered at the carrier frequency ω_c 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. Ensure that the following initial conditions exist on the board.

a. Audio input select switch is in INT position.
b. Mode switch is in DSB position.
c. Output amplifier's gain pot is in full clockwise position.
d. Speaker's switch is in OFF position.

2. Turn on power to the ST2201 board.

3. Turn the audio oscillator block's amplitude pot to its full 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 our modulating signal. Note that the sine wave's frequency can be adjusted from about 300Hz to approximately 3.4KHz, by adjusting the audio oscillator's frequency potmeter. Also note that the amplitude of this audio modulating signal can be reduced to zero by turning the Audio oscillator's amplitude potmeter to its fully counterclockwise (MIN) position. Return the amplitude present to its max position.

4. Turn the balance potmeter in the balanced modulator & band pass filter circuit-1block to its fully clockwise position. It is this block that we will use to perform*double-side band amplitude modulation.*

5. Monitorthe two inputs to the balanced modulator & band pass filtercircuits block, at t.p.1 and t.p.9. Note that:

a. The signal at t.p.1 is the audio-frequency sinewave from the audiooscillator block. This is the modulating input to our double-sidebandmodulator.

b. Test point 9 carries a sine wave of 1MHz frequency and amplitude120mVpp approx. This is the carrier input to our double-sidebandmodulator.

6. Next, examine the output of the balanced modulator & band pass filter circuit-1block (at t.p.3), together with the modulating signal at t.p.1 Trigger theoscilloscope on the t.p. 1 signal.The output from the balanced modulator & band pass filter circuit 1 block (at

t.p. 3) is a DSBFC AM waveform, which has been formed byamplitude-modulating the 1MHz carrier sinewave with the audio-frequencysinewave from the audio oscillator.

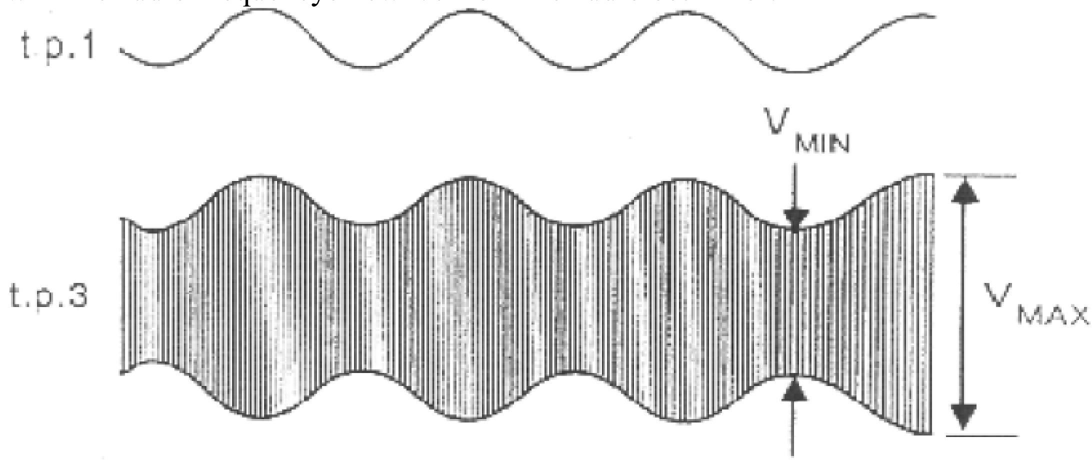


Figure 4: DSB FC (AM) waveforms

7. Now vary the amplitude and frequency of the audio-frequency sinewave, byadjusting the amplitude and frequency present in the audio oscillator block.Note the effect of varying amplitude and frequency of the modulating audio signal on the amplitude modulated waveform.The amplitude and frequency of the two sidebands can be reduced tozero by reducing the amplitude of the modulating audio signal to zero. Do thisby turning the amplitude pot to its MIN position, and note that the signal at t.p3 becomes an un-modulated sine wave of frequency 1 MHz, indicating that onlythe carrier component now remains. Return the amplitude pot to its maximumposition.
Now turn the balance pot in the balanced modulator & band pass filter circuit 1block, until the signal at t.p. **3**looks as shown in Fig.**5**

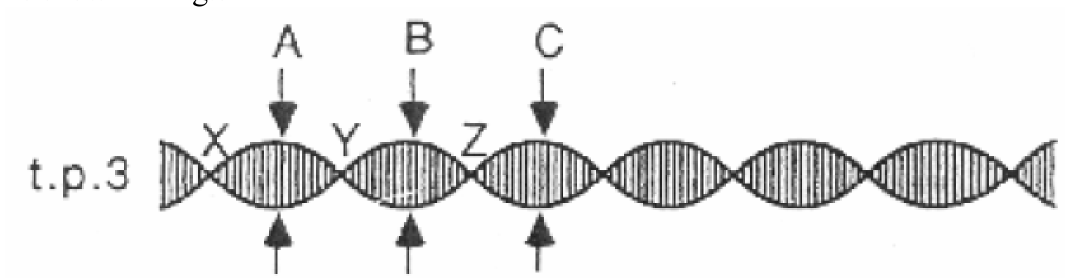


Figure **5**: Output of BPF

The balance pot varies the amount of the 1 MHz carrier component, which ispassed from the modulator's output.By adjusting the pot until the peaks of the waveform (A, B, C and so on) havethe same amplitude, we are removing the carrier component altogether.We say that the carrier signal has been 'balanced out' (or 'suppressed') to leave only thetwo sidebands.
Note that once the carrier signal has been balanced out or suppressed, the amplitude of waveform at t.p.3 could be zero at points X, Y; Z etc. If this is not the case,it is because one of the sidebands is being amplified more than the other. Toremove this problem, the band pass filter in the balanced modulator & band passfilter circuit 1 block must be adjusted so that it passes both sidebands equally.This is achieved by carefully trimming transformer T1, until the waveform'samplitude is as close to zero as possible at the minimum points.The waveform at t.p.3 is known as a double-side suppressed carrier

(DSBSC)waveform, and its frequency spectrum is as shown in Figure.1.Note that only the two sidebands remain there, the carrier component has beenremoved.
8. Change the amplitude and frequency of the modulating audio signal (byadjusting the audio oscillator block's amplitude and frequency pots), and notethe effect that these changes on the DSBSC waveform. The amplitudes of thetwo sidebands can be reduced to zero by reducing the amplitude of themodulating audio signal to zero. Do these by turning the amplitude present to itsMIN position, and note that the monitored signal becomes a D C level indicating that there are now no frequency components present. Return theamplitude pot to its MAX position.
9. Examine the output from the output amplifier block (t.p.13), together with theaudio modulating signal (at t.p.1), triggering the scope with the audiomodulating signal. Note that the DSBSC waveform appears, amplified slightlyat t.p.13, as we will see later, it is the output amplifier's output signal which willbe transmitted to the receiver.
10. By using the microphone, the human voice can be used as the modulatingsignal, instead of using ST2201's audio oscillator block.Connect the module's output to the external audio input on the ST2201 board,and put the audio input select switch in the ext position.The input signal to the audio input module may be taken from an externalmicrophone or from a cassette recorder, by choosing the appropriate switchsetting on the module.

RESULT:-
TheDSBSC signal has been generated using balanced modulator.

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 readings carefully.
5. Power supply should be switched off after completion of experiment.

QUIZ / ANSWERS:

Q. 1. What is DSB-SC?
Ans. DSB-SC means the transmission of both the sidebands (Lower Side-band and Upper Side-band) and the carrier is totally removed from the modulated signal.

Q. 2. What are the frequencies of both side-bands in AM?
Ans. (1) lower sideband frequency $(\omega_c - \omega_m)$ (2) upper sideband frequency$(\omega_c + \omega_m)$.

Q. 3. Which is the difference between DSBFC, DSBSC and DSBRC?
Ans. In DSBFC, carrier signal along with both sidebands are transmitted.
In DSBSC, both sidebands are transmitted and carrier signal is removed.
In DSBRC, a little information about carrier signal is transmitted with both sidebands.

Q. 4. Write the methods of DSBSC generation.
Ans. (1) Balanced Modulator (2) Ring Modulator

Q. 5. Write the methods of DSBSC detection.
Ans. (1) Synchronous detection (2) Envelop Detection after inserting Carrier.

Q. 6. What is the draw-back in synchronous detection of DSB-SC?
Ans. The phase and frequency of locally generated carrier signal in synchronous detector is very critical. We get an attenuated and distorted output signal at the receiver end.

Q. 7. What is the pilot carrier in DSB-SC AM?
Ans. A small amount of carrier signal called pilot carrier is transmitted along with modulated signal from the transmitter.

Q. 8. What is the use of pilot carrier?
Ans. Pilot carrier at the receiver end provides phase locking and thus provides synchronization.

Q. 11. Where is the use of DSB-SC?
Ans. Television

EXPERIMENT 4

AIM: To generate SSB-SC AM 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.

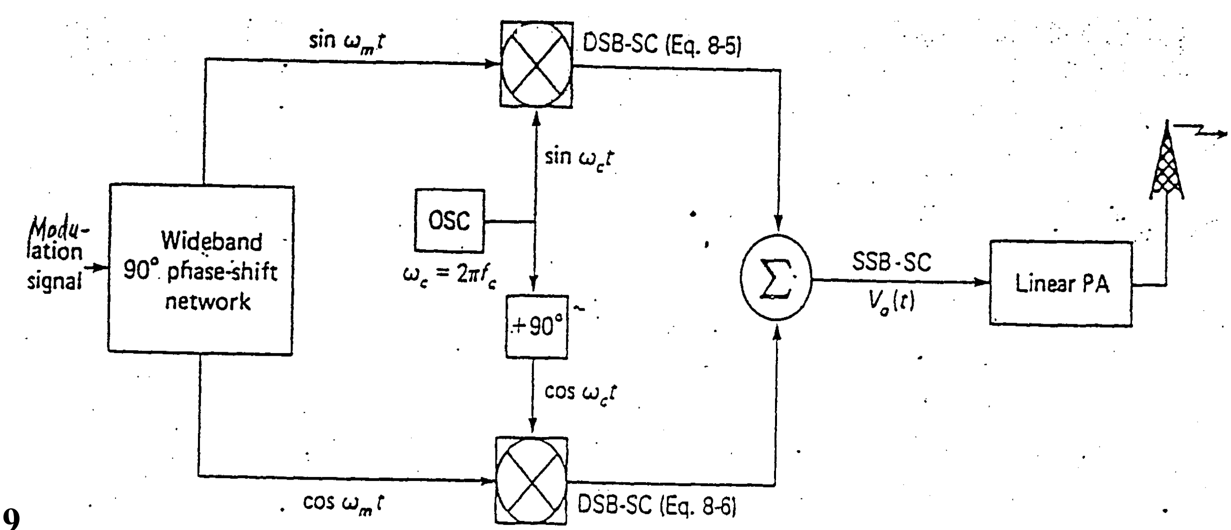


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Figure 1: SSB Transmitter

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. Ensure that the following initial conditions exist on the board:

a) Audio input select switch is in INT position.

b) Mode switch is in SSB position.

c) Output amplifier's gain pot is in fully clockwise position.

d) Speaker switch is in OFF position.

2. Turn on power to the **ST2201** board.

3. 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. Note that the sine wave's frequency can be adjusted from about 300Hz to approximately 3.4 KHz, by adjusting the audio oscillator's frequency pot.

Note:*That the amplitude of this audio modulating signal can be reduced to zero, by turning the audio oscillator's pot to its fully counter-clockwise (MIN) position. Leave the amplitude pot on its full clockwise position, and adjust the frequency pot for an audio frequency of 2 KHz, approx. (mid-way).*

4. 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.

We will now examine the operation of each of these blocks in detail.

5. 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 455KHz. It is generated by the 455 KHz oscillator block, and is the carrierinput to the balanced modulator block.

6. Next, examine the output of the balanced modulator block (at t.p.17), togetherwith the modulating signal at t.p.15 trigger the oscilloscope on the modulatingsignal. Check that the waveforms are as shown Fig. 2.

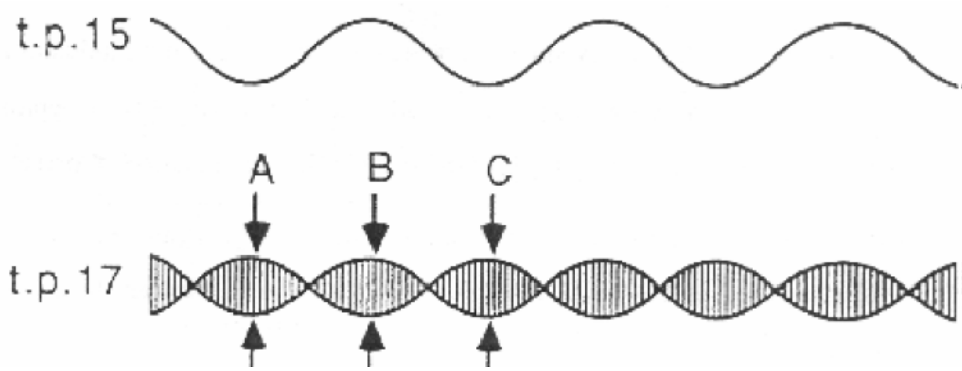


Figure 2: Modulating and Modulated Signal waveforms

Note that it may be necessary to adjust the balanced modulator block's balancepot, 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.You will recall that the waveform at t.p.17 was encountered in the previousexperiment this is a double-sideband suppressed carrier (DSBSC) AMwaveform, and it has been obtained by amplitude-modulating the carriersinewave at t.p. 6 of frequency fc with the audio-frequency modulating signal att.p. 15 of frequency fm, and then removing the carrier component from theresulting AM signal, by adjusting the balance pot. The frequency spectrum ofthis DSBSC waveform is shown in Fig.3.

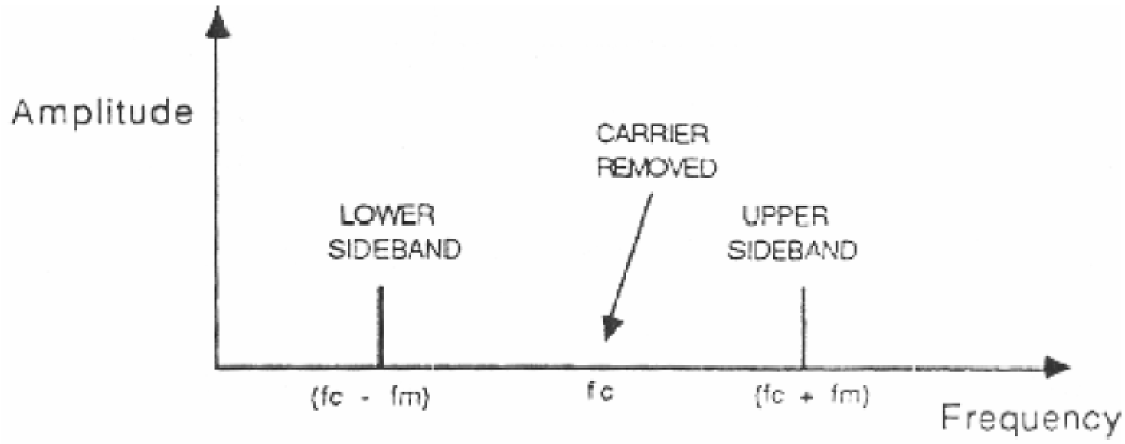


Figure 3:DSBSC Sidebands

7. The DSBSC output from the balanced modulator block is next passed on to theceramic filter block, whose purpose is to pass the upper sideband, but block thelower sideband. We will now investigate how this is achieved.First note that the ceramic band pass filter has a narrow pass band centeredaround 455 KHz.It was mentioned earlier that the frequency of the carrier input to the balancedmodulator block has been arranged to be slightly less than 455 KHz. In fact, thecarrier chosen so that, whatever the modulating frequency fm, the uppersideband (at fc+fm) will fall inside the filter's pass band, while the lowersideband (at fc-fm) always falls outside.Consequently, the upper sideband will suffer little attenuation, while the lowersideband will be heavily attenuated to such an extent that it can be ignored. This is shown in the frequency spectrum in fig 4.

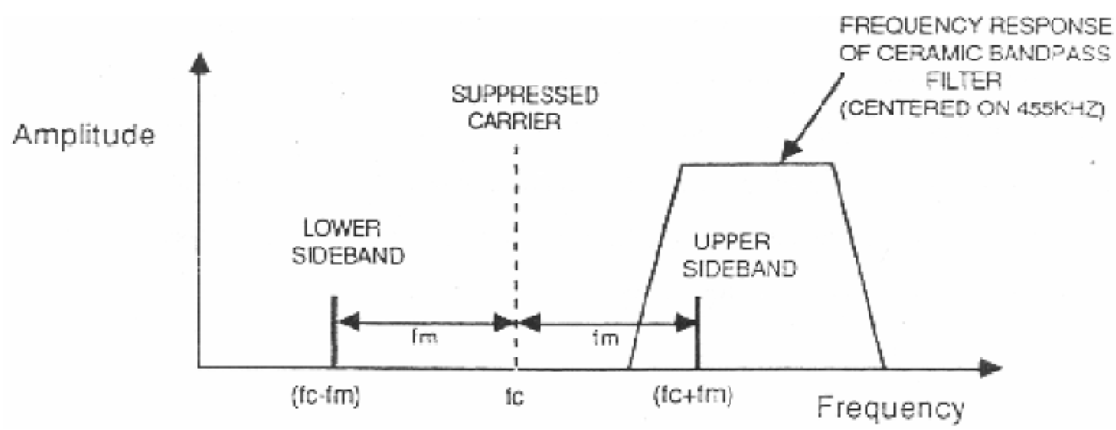


Figure 4: Frequency Response of Ceramic BPF

8. Monitor the output of the ceramic band pass filter block (at t.p. 20) togetherwith the audio modulating signal (at t.p.15) using the later signal to trigger theoscilloscope. Note that the envelope of the signal at t.p. 20 now has fairlyconstant amplitude, as shown in Fig.5.

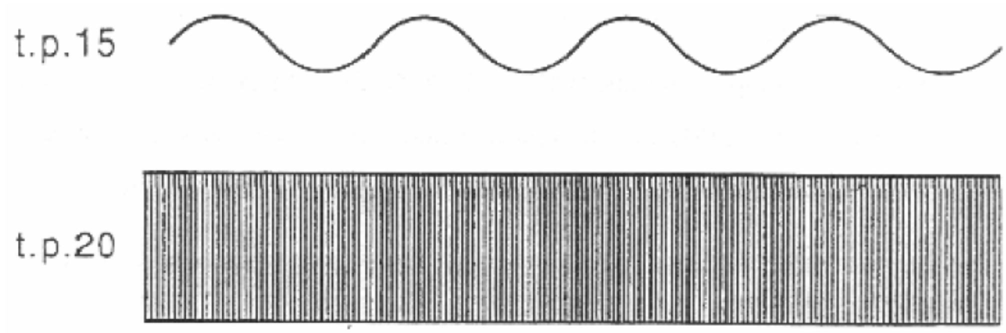


Figure 5: Input Audio Signal and SSB output Signal

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

9. Now, trigger the oscilloscope with the ceramic band pass filter's output signal (t.p.20) and note that the signal is a good, clean sinewave, indicating that thefilter has passed the upper sideband only.Next, turn the audio oscillator block's frequency pot throughout its range. Notethat for most audio frequencies, the waveform is a good, clean sinewave,indicating that the lower sideband has been totally rejected by the filter. For lowaudio frequencies, you may notice that the monitored signal is not such a puresinusoid. This is because the upper and lower sidebands are now very close toeach other, and the filter can no longer completely remove the lower sideband.

Nevertheless, the lower sideband's amplitude is sufficiently small comparedwith the upper sideband, that its presence can be ignored. Since the uppersideband dominates for all audio modulating frequencies, we say that singlesideband (SSB) amplitude modulation has taken place.

***Note:** If the monitored waveform is not a good sinewave at higher modulatingfrequencies (i.e. when the frequency pot is near the MAX position), then it islikely that the frequency of the 455 KHz oscillator needs to be trimmed*

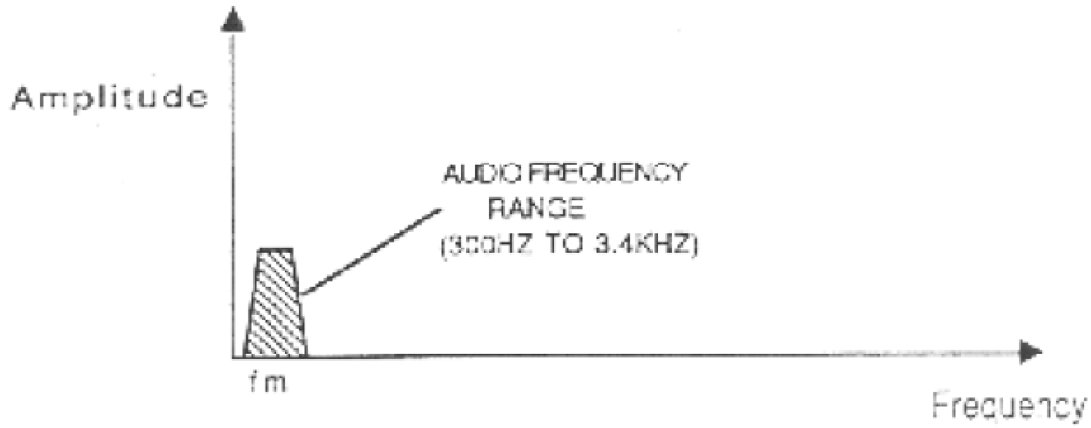
10. Note that there is some variation in the amplitude of the signal at the filter'soutput (t.p. 20) as the modulating frequency changes. This variation is due tothe frequency response of the ceramic band pass filter, and is best explained byconsidering the spectrum of the filter's input signal at the MIN and MAXpositions of the frequency pot, as shown in Fig. 4.

a. Modulating frequency fm = 300Hz (pot in MIN position)

b. Modulating frequency fm = 3.4 KHz (pot in MAX position)

Notice that, since the upper sideband cuts rising edge of the filter's frequencyresponse when fm = 300 Hz, there will be a certain amount of signal attenuation then the frequency pot is in its 'MIN' position.

11. Note that, by passing only the upper side band of frequency (fc+fm), all wehave actually done is to shift out audio modulating signal of frequency fm up infrequency by an amount equal to the carrier frequency fc. This is shown in Fig.7.



(a). Range of frequencies available from audio oscillator

(b). Corresponding range of output frequencies from ceramic band passfilter block

Figure 7: Range of frequency output from audio oscillator and ceramic BPF

12. With the audio oscillator block's frequency pot roughly in its midway position(arrowhead pointing towards the top), turn the block's amplitude pot to its MINposition, and note that the amplitude of the signal at the ceramic band passfilter's output (t.p. 20) drops to zero.This highlights one on the main advantages of SSB amplitude modulation ifthere is no modulating signal, then the amplitude of the SSB waveform drops tozero, so that no power is wasted.Return the amplitude pot to its MAX position.

13. The particular filter we are using has a pass bandcentered on 455 KHz, and this is why we have arranged for the wanted uppersideband to also be at about 455 KHz.However, there is a disadvantage of this type of filter is the range of frequencies that the filter will pass is fixed during the filter's manufacture, and cannot subsequently be altered.Note that since there is a large gap between the upper and lower sidebands (agap of about 910 KHz), a band pass filter with a very sharp response is notneeded to reject the lower sideband, a simple tuned circuit band pass filter isquite sufficient.

14. Now examine the output of the balanced modulator & band pass filter circuit 2block (t.p.22), and check that the waveform is a good sinewave of frequencyapproximately 1.45MHz.This indicates that only the upper sideband is being passed by the block. Checkthat the waveform is reasonably good sinusoid for all audio modulatingfrequencies (i.e. all positions of the audio oscillator’s frequency pot). If this isnot the case, it may be that the balance pot (in the balanced modulator & bandpass filter circuit 2 blocks) needs adjusting, to remove any residual carriercomponent at 1 MHz. If a reasonably clean sinewave still cannot be obtained forall audio frequencies, then the response of the tuned circuit band pass filterneeds adjusting. This is achieved by adjusting transformer T4 in the balancedmodulator & band-pass filter circuit 2 blockWhen the modulating audio signal is swept over its entire range (arange of 3.4 KHz – 300 Hz = 3.1 KHz), the SSB waveform at t.p. 22 sweepsover the same frequency range. So single-sideband modulation has simplyserved to shift our range of audio frequencies up so they are centered on1.455 MHz.

15. Monitor the 1.455 MHz SSB signal (at t.p. 22) together with the audiomodulating signal (t.p. 15), triggering the scope with the later. Reduce theamplitude of the audio modulating signal to zero (by means of the audiooscillator block's amplitude pot), and note that the amplitude of the SSB signalalso drops to zero, as expected. Return the amplitude pot to its MAX position.

16. Examine the final SSB output (at t.p. 22) together with the output from theoutput amplifier block (t.p. 13). Note that the final SSB waveform appears,amplified slightly, at t.p. 13. As we still see later, it is the output signal whichwill be transmitted to the receiver.

17. By using the microphone the human voice can be used as the audio modulatingsignal, instead of using **ST2201**'s audio oscillator block. Connect themicrophone to the external audio input on the **ST2201** board, and put the audioinput select switch in the EXT position.

The input signal to the audio input select may be taken from an externalmicrophone (supplied with the module) of from a cassette recorder, by choosingthe appropriate switch setting on the module.

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

QUIZ / ANSWERS:

Q.1. What is the most commonly used demodulator?

Ans. Diode detector.

Q.2. What is AGC?

Ans. AGC stands for automatic gain control.

Q.3. What is the use of AGC?

Ans. AGC circuit is used to prevent overloading receiver and also reduce the effect of fluctuations in the received signal strength.

Q.4. What is the required oscillator frequency in AM receiver?

Ans. The required oscillator frequency in AM receiver is always higher than the signal frequency.

Q.5. What is the use of pilot carrier in SSB?
Ans. For frequency stabilization.

Q.6. Which circuit is not be used to demodulate SSB?
Ans. Phase discriminator.

Q.7. What is the advantage of SSB over DSB?
Ans. Transmitter circuit is more stable, giving batter reception.

Q.8.Which type of modulation is used in India for video transmission?
Ans. Amplitude Modulation.

Q.9. Which filter is used in SSB generation?
Ans. Mechanical filters.

Q.10. How AM signals with large carrier are detected?
Ans. By using envelope detector

EXPERIMENT 5

AIM: To generate Frequency modulated signal using Voltage Control Oscillator.

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

THEORY:
Frequency modulation is a form of angle modulation in which the amplitude of the modulated carrier is kept constant while its frequency and its rate of change are varied by themodulating signal . In FM the instantaneous angular frequencyW_i is varied linearly in accordance with the instantaneous magnitude of base band signal X(t) , about an un-modulated carrier frequency (also called as resting frequency) Wc and the rate at which the carrier shifts from its resting point to its non resting point isdetermined by the frequency of modulating signal while keeping the amplitude of the carrier wave constant.

Carrier signal $C(t) = A\text{Sin}(W_c t + \theta_0) = A\text{Sin}\,\Phi$...........(1)

where W_c is the frequency of Carrier wave in radians/secondand
Φ in radians = Total phase angle of the unmodulated carrier = $(W_c t + \theta_0)$….(2)
In FM whilethe amplitude A remains constant, instantaneous value of Φ changes. If $W_i(t)$ = Instantaneous value of angular velocity , and
and Φ_i =Instantaneous phase angle of FM wave,

then $W_i(t) = d\Phi_i / dt$, ..…………….….…(3)

and $\Phi_i = \int W_i(t)\,dt$ ..………………….(4)

Therefore FM wave can be represented as $S(t) = A\text{Sin}\,\Phi_i$…….……….(5)

Modulating voltage Signal = X(t) volts ……….….………….(6)

Then instantaneous angular frequency of an FM signal is given by

$d\Phi_i / dt = W_i(t) = W_c + K_f X(t)$….…..………….(7)

where K_f = Constant of proportionality = frequency sensitivity of the modulator in Hertz per volt

Therefore Frequency variation = $|K_f X(t)|$ ………………….………….(8)

Since the value of $W_c$is assumed to be fixed,

$\Phi_i = \int W_i(t)\,dt = \int [W_c + K_f X(t)]\,dt = W_c t + K_f \int X(t)\,dt$ ……………(9)

Frequency Deviation:
It is the amount by which carrier frequency is varied from its unmodulated value and it is same as frequency variation.
Max Frequency deviation $\Delta W = |K_f X(t)|_{max}$……………………(10)
Very often we write $\Delta W = \delta$;
Maxium allowed deviation =75 khz

Frequency Modulation Indexm$_f$:
It is the ratio of frequency deviation ΔW in rad/sec to the angular frequency of modulating signal W_m orfrequency deviation in Hertz/secto to the modulating frequency in Hertz/sec.
Thus $m_f = \Delta W / W_m = \delta / W_m$ if δ is given in rad /Sec ……………..(11)
If δ is givenin Hertz/Sec then $m_f = \delta / f_m$ …………………..……(12)

Mathematicalexpression for FM wave
$S(t) = A\text{Sin}\ \Phi_i = A\text{Sin}\ [W_c t + K_f \int X(t)\,dt]$ …………………….(13)

For Single tone FM

$$X(t)=V_m \text{Cos}\ W_c t \ldots\ldots(14)$$

Thus $\Phi_i = W_c t + K_f \int V_m \text{Cos}\ W_m t\,dt = W_c t + \frac{K_f V_m}{W_m} \text{Sin}\ W_m t$

$$W_m = W_c t + \underline{\Delta W}\ \text{Sin}\ W_m t = W_c t + m_f\, \text{Sin}\ W_m t$$

Thus $S(t) = A\ \text{Sin}\ [W_c t + K_f \int X(t)\,dt]$

$$= A\ \text{Sin}\ [W_c t + m_f\, \text{Sin}\ W_m t] = \ldots\ldots(15)$$

Deviation Ratio:
It is the ratio of deviation in carrier frequency to the maximum modulating frequency. In single tone FM, modulation index and the deviation ratio will be the same. If the modulating signal(AF) is 15 kHZ at a certain amplitude and the carrier shift is 75kHZ, the transmitter will produce eight(8) significant sidebands as shown in the table above. The corresponding deviation ratio / modulation index is known as Maximum Deviation Ratio.However in multi tone FM, the amplitude of highest frequency component may not necessarily be maximum. Modulation index will be different for each signal frequency component. The deviation ratio in this case will not be equal to any particular modulation index.

Frequency Spectrum:
Analysis of equation (15) which is a sine function of another sine function shows:

$$S(t) = A\{ J_0.(m_f) \sin W_c t + J_1(m_f)\{\sin(W_c t + W_m t) + \sin(W_c t - W_m t)\}$$
$$+J_2(m_f)\{\sin(W_c t + 2W_m t) + \sin(W_c t - 2W_m t)\}$$
$$+J_3(m_f)\{\sin(W_c t + 3W_m t) + \sin(W_c t - 3W_m t)\}$$
$$+J4(m_f)\{\sin(W_c t + 4W_m t) + \sin(W_c t - 4W_m t)]+\ldots\ldots]$$

The output consists of a carrier and an apparently infinite number of pairs of side bands having an amplitude coefficient J_n (m_f), which is a Bessel function of m_f and of the order n denoted by the subscript. Values of these coefficients are available readily in table form as well as in graphic form as shown below.
Analysis of FM waveforms Wave forms of carrier, modulating signal, modulated signal as well as graphical form of plot of J_n (m_f) versus values of m_f are shown below. It can be seen that:
1. Unlike AM, FM output contains carrier component of frequency f_c as well as infinite number of side bands separated from the carrier frequency by f_m, $2f_c$, $3f_c$,……and thus have a recurrence frequency of f_m.

2. The values of each J_n coefficient, which represent the amplitude of a pair of side bands, fluctuates on either side of zero, gradually diminishes with increasing value of m_f like damped oscillations. The values of J_n coefficients also eventually decrease, but only past increased value of n. As the value of mf increases, the valueof J_0decreasesfrom 1 and the values of J_1 to J_n increases from0 andfluctuate around mean value of 0.
3. The modulation index determines how many side band components have significant values.
4. Unlike AM, in FM, while the amplitude of modulated signal remains constant, the value of the carrier component decreases with increase in m_f like a damped oscillation. It means that while the total transmitted power remains constant in FM, the number side bands of significant amplitude (and therefore the effective band width) increase with increase inm_f .This increases the immunity to noise in FM unlike AM.
5. As the value of m_f increases, The carrier component becomes zero for certain values of modulation index, called eigen values which are approximately 2.4,5.5.8.6.11.8 and so on. These disappearances of carrier for specific values of $m_{f,}$ form a handy basis for measuring deviation.

BLOCK DIAGRAM:

The audio oscillator supplies the information signal and could, if we wish, be replacedby a microphone and AF amplifier to provide speech and music instead of thesinewave signals that we are using with **ST2203**.The FM modulator is used to combine the carrier wave and the information signal inmuch the same way as in the AM transmitter. The only difference in this case is thatthe generation of the carrier wave and the modulation process is carried out in thesame block. It is not necessary to have the two processes in same block, but in ourcase, it is. The output amplifier increases the power in the signal before it is applied tothe antenna for transmission just as it did in the corresponding block in the FMtransmitter.

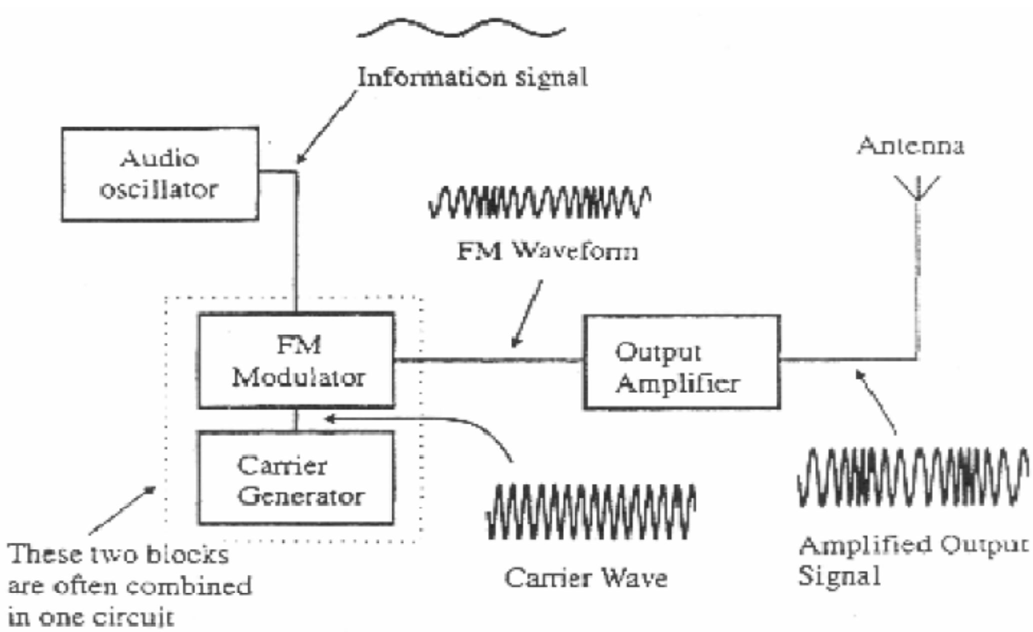


Figure 1: FM Transmitter

Controlling the VCO:
To see how the VCO is actually controlled, let us assume that it is running at the samefrequency as an un-modulated input signal. The input signal is converted into a square wave and, together with the VCO output,forms the two inputs to an Exclusive - OR gate.Remember that the Exclusive - OR gate provides an output whenever the two inputsare different in value and zero output whenever they are the same. The provided an output from the Exclusive -OR gate with an on-off ratio of unity and anaverage voltage at the output of half of the peak value. Now let us assume that the FM signal at the input decreases in frequency (see fig. 34).The period of the 'squared up' FM signal increases and the mean voltage level fromthe Exclusive -OR gate decreases. The mean voltage level is both the demodulatedoutput and the control voltage for the VCO. The VCO frequency will decrease untilits frequency matches the incoming FM signal.

PROCEDURE:
1. Ensure that the following initial conditions exist on the **ST2202** board.
    - **a.** All switches are turned off.
    - **b.** Amplitude pot (in mixer amplifier block) is in fully clockwise position.
    - **c.** VCO switch isin 'ON' position.
2. Turn the audio oscillator block's amplitude pot to its fully clockwise position,and examine the block's output t.p.1 on an oscilloscope. This is the audiofrequency sinewave, which will be used as our modulating signal. Note that thesinewave's frequency can be adjusted from about 300Hz to approximately3.4KHz, by adjusting the audio oscillator's frequency pot.
3. Connect the output of audio oscillator to VCO section’s MOD In socket.
4. Turn ON the power supply.
5. Observe the modulating signal and modulated output at the VCO’s MOD OUT socket by using CRO.
4. Calculate $m_f = \delta / f_m$.
5. Vary the modulating frequency keeping carrier freq constant and repeat steps 3 & 4.
6. Vary the carrier frequency keeping modulator freq constant and repeat steps 3 & 4.
7. Tabulate the results.

OBSERVATION TABLE:

1 **2** **3** **4** **5**				

SAMPLE CALCULATION:-

$$m_f = \delta / f_m$$
$$= 2 \times 8.3 \times 10^3 / 1000$$
$$= 16.6$$

Waveforms:

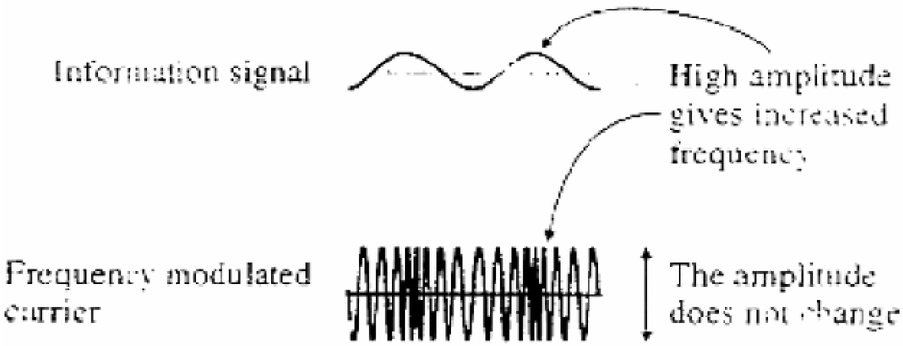


Figure 2: Modulating and FM Modulated signal

RESULT:
Frequency modulated wave using VCO is observed on CRO and m_f is calculated.

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.

QUIZ / ANSWERS:
Q.1. How many types of FM are there? Write their names.
Ans. There two types of FM i.e. narrow band FM and wideband FM.

Q.2. What is frequency deviation in FM?
Ans. The maximum change in instantaneous frequency from the average is known as frequency deviation.

Q.3. Which is the useful parameter for determination of bandwidth?
Ans. Frequency deviation is the useful parameter for determination of bandwidth.

Q.4. How many sidebands are there in FM?
Ans. Theoretically, there is infinite number sidebands in FM.

Q.5. Which sidebands are ignored in FM?
Ans. The sidebands with small amplitude are ignored in FM.

Q.6. Which are the significant sidebands?
Ans. The sidebands having considerable amplitudes i.e. more than or equal to 1% of the carrier amplitude are known as significant sidebands.

Q.7. What is CCIR?
Ans. CCIR stands for Consultative Committee for International Radio.

Q.8. What is the indirect method of FM generation?
Ans. Armstrong method.

Q.9.What is the direct method of FM generation?
Ans. The parameter variation method.

Q.10. What is VCO?
Ans. VCO stands for voltage controlled oscillator whose frequency is controlled by modulating voltage.

EXPERIMENT 6

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.

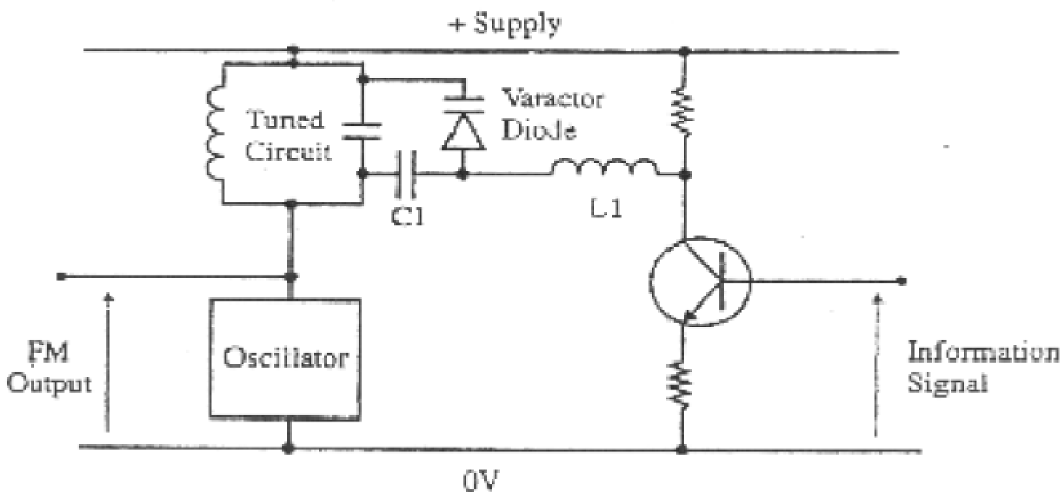


Figure 1: FM generation using Varactor Modulator

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

<u>The operation of the varactor modulator:</u>

1. The information signal is applied to the base of the input transistor and appearsamplified and inverted at the collector.

2. This low frequency signal passes through the RF choke and is applied across thevaractor diode.

3. The varactor diode changes its capacitance in sympathy with the informationsignal and therefore changes the total value of the capacitance in the tunedcircuit.

4. The changing value of capacitance causes the oscillator frequency to increaseand decrease under the control of the information signal. The output is thereforea FM signal.Before we start the study of varactor/ reactance modulation techniques we shall studya simple VCO circuit.

Simply connect the audio output to the socket labeled VCO modulation in andobserve the FM modulated waveform on the oscilloscope at the VCO modulation outterminal. Keep the amplitude of audio output to approx 4 V p-p and frequency 2 kHzapprox. Observe a stable FM modulated waveform on CRONow turn the timebase speed of CRO little higher and you will observe the samewaveforms as under (like Bessel function).

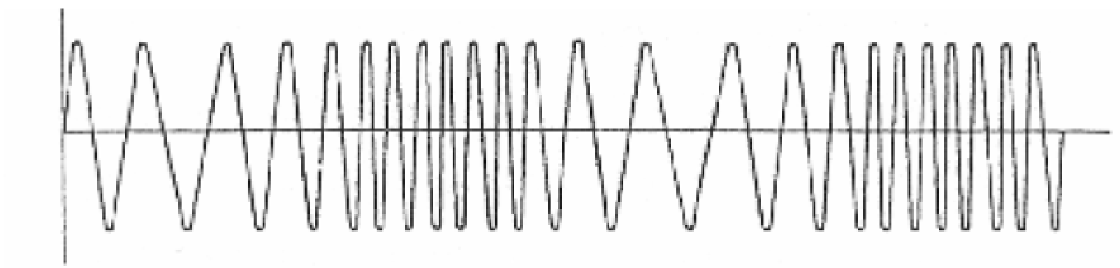


Figure 2:FM modulated wave

Now disconnect the audio amplifier's output from modulation IN and connect it toaudio IN, keep the reactance/varactor switch in varactor position. Observe the outputof mixer - amplifier circuit. Keep the oscilloscope in X10 position now observe thefull waveform by shifting the X position. It is as shown in fig. Mark the resemblancebetween the output of VCO and the Varactor modulator. They are same.The freq. modulation in VCO was more because the Frequency difference betweenthe carrier and the modulating signal was very less.

<u>FM Using Reactance Modulator:</u>In fig. 3, the left hand half is the previous varactor modulator simply an oscillatorand a tuned circuit, which generates the un-modulated carrier. The capacitor C and theresistor R are the two components used for the phase shifting, and together with thetransistor, form the voltage controlled capacitor. This voltage-controlled capacitor isactually in parallel with the tuned circuit. This is not easy to see but figure 18 may behelpful.In the first part of the figure, the capacitor and associated components have beenreplaced by the variable capacitor, shown dotted.

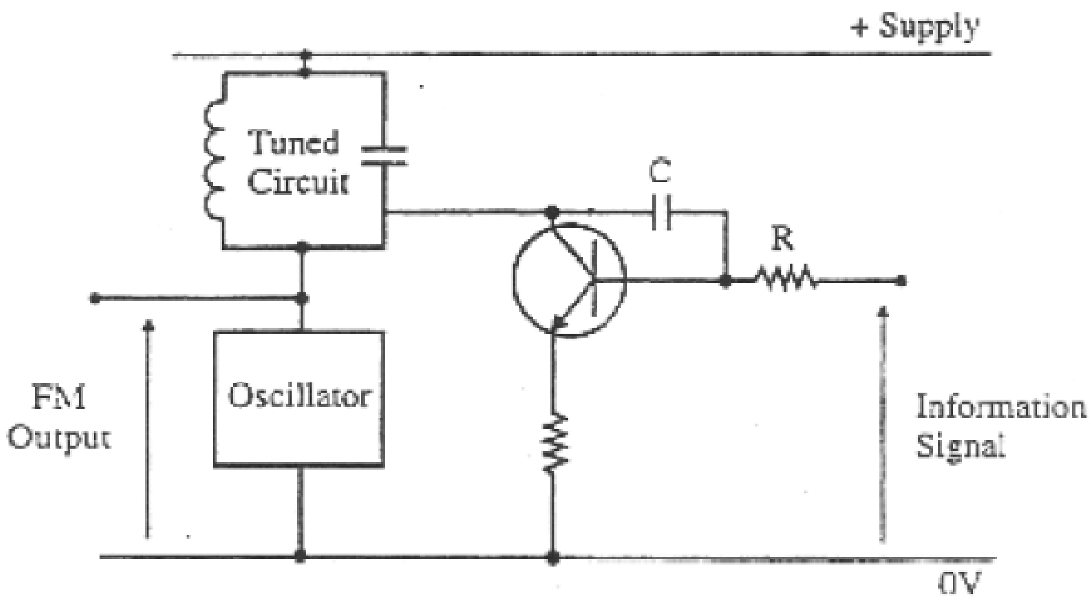


Figure 3: FM using Reactance Modulation.

In the next part, the two supply lines are connected together. We can justify this bysaying that the output of the DC power supply always includes a large smoothingcapacitor to keep the DC voltages at a steady value.This large capacitor will have a very low reactance at the frequencies being used inthe circuit less than a milliohm. We can safely ignore this and so the two supply linescan be assumed to be joined together. Remember that this does not affect the DCpotentials, which remain at the normal supply voltages.If the two supply voltages are at the same AC potential, the actual points ofconnection do not matter and so we can redraw the circuit as shown in the third part.

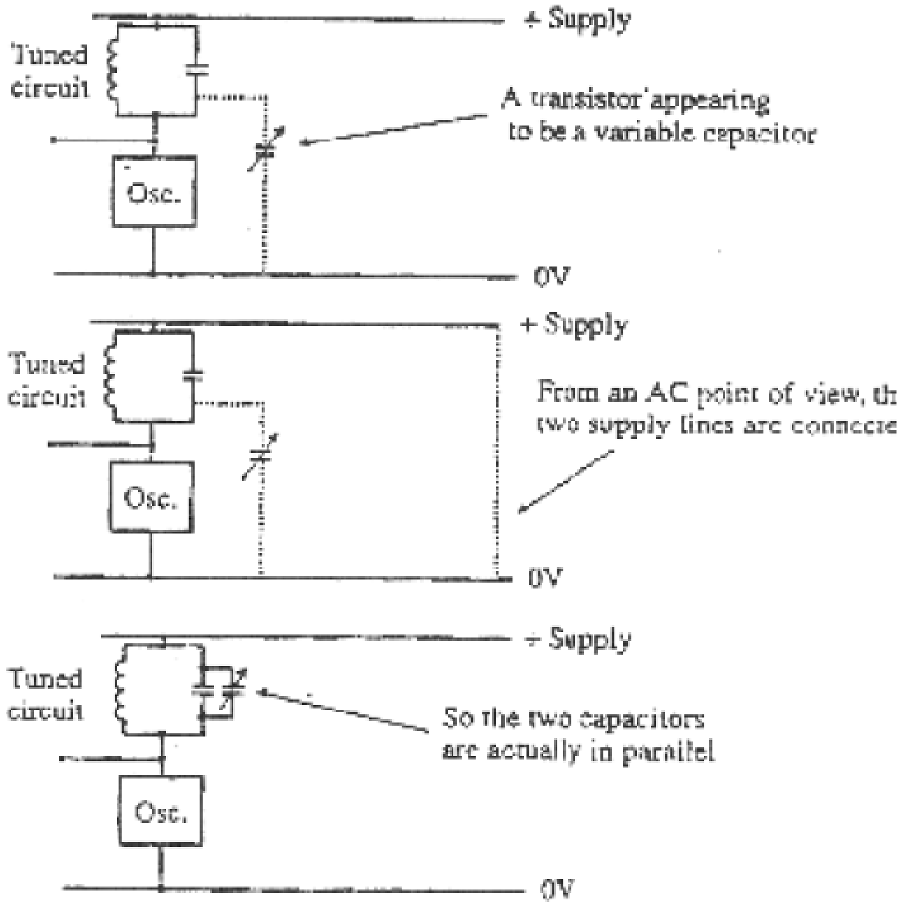


Figure 4: VCO using capacitor

<u>Operation of the Reactance Modulator:</u>

1. The oscillator and tuned circuit provide the un-modulated carrier frequency andthis frequency is present on the collector of the transistor.

2. The capacitor and the resistor provide the 90° phase shift between the collectorvoltage and current. This makes the circuit appear as a capacitor.

3. The changing information signal being applied to the base has the same effectas changing the bias voltage applied to the transistor and, this would have theeffect of increasing and decreasing the value of this capacitance.

4. As the capacitance is effectively in parallel with the tuned circuit the variationsin value will cause the frequency of resonance to change and hence the carrierfrequency will be varied in sympathy with the information signal input.

Block Diagram:

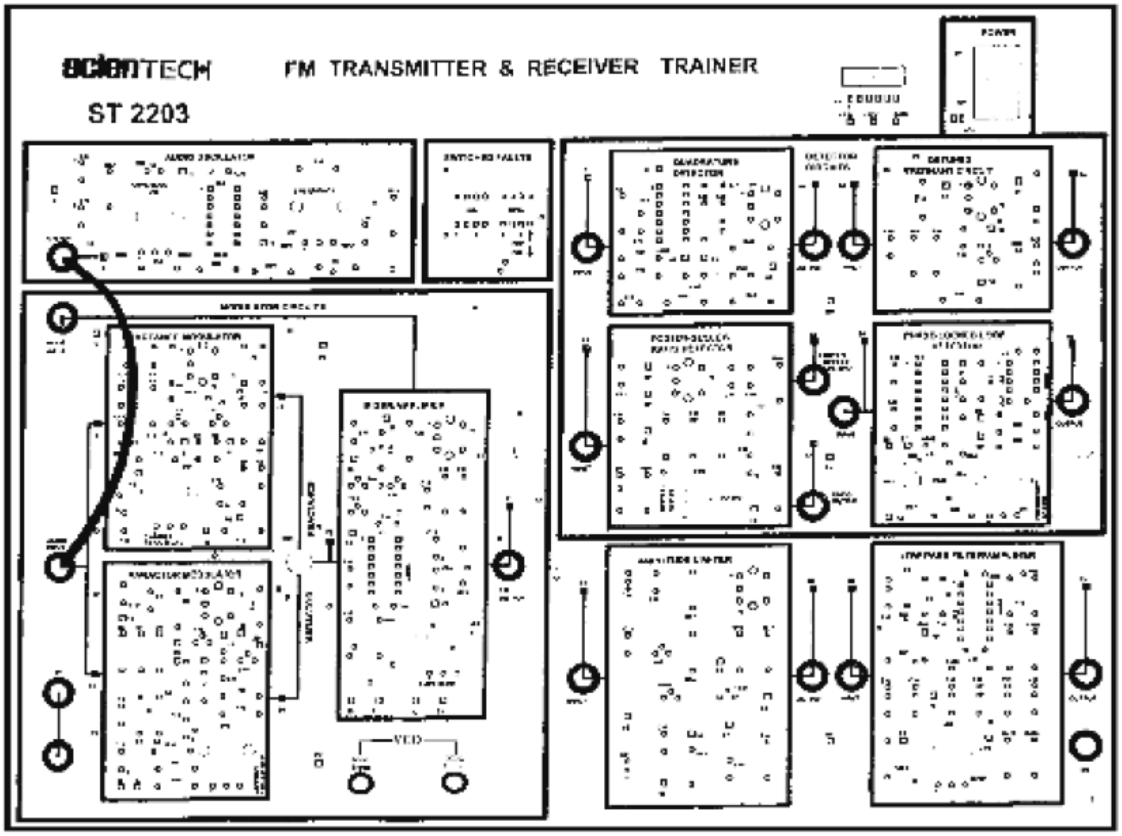


Figure 3: Block Diagram of FM Trainer kit

PROCEDURE:

1. Ensure that the following initial conditions exist on the **ST2202** board.

- **a.** All switched faults off.
- **b.** Amplitude pot (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.

4. Turn the audio oscillator block's amplitude pot to its fully clockwise position,and examine the block's output t.p.1 on an oscilloscope. This is the audiofrequency sinewave, which will be used as our modulating signal. Note that thesinewave's frequency can be adjusted from about 300Hz to approximately3.4KHz, by adjusting the audio oscillator's frequency pot.Note also that the amplitude of this modulating signal is adjusted by audiooscillator amplitude pot. Leave the amplitude pot in min. position.

5. Connect the output socket of the audio oscillator block to the audio input socketof the modulator circuit's block.

For FM Varactor Modulator

6. Set the reactance / varactor switch to the varactor position. This switch selectsthe varactor modulator and also disables the reactance modulator to prevent anyinterference between the two circuits.
7. The output signal from the varactor modulator block appears at t.p. 24 beforebeing buffered and amplified by the mixer / amplifier block, any capacitiveloading (e.g. due to oscilloscope probe) may slightly affect the modulatorsoutput frequency. In order to avoid this problem we monitor the buffered FMoutput signal the mixer / amplifier block at t.p.34.
8. Put the varactor modulator's carrier frequency pot in its midway position, andthen examine t.p. 34. Note that it is a sinewave of approximately 1.2 Vp-p,centered on 0V. This is our FM carrier, and it is un-modulated since the varactormodulators audio input signal has zero amplitude.
9. The amplitude of the FM carrier (at t.p.34) is adjustable by means of themixer/amplifier block's amplitude pot, from zero to its pot level. Try turningthis pot slowly anticlockwise, and note that the amplitude of the FM signal canbe reduced to zero. Return the amplitude pot to its fully clockwise position.
10. Try varying the carrier frequency pot and observe the effects.
11. Also, see the effects of varying the amplitude and frequency pots in the audiooscillator block.
12. Turn the carrier frequency pot in the varactor modulator block slowly clockwiseand note that in addition to the carrier frequency increasing there is a decreasein the amount of frequency deviation that is present.
13. Return the carrier frequency pot to its midway position, and monitor the audioinput (at t.p.6) and the FM output (at p.34) triggering the oscilloscope on theaudio input signal. Turn the audio oscillator's amplitude pot throughout its rangeof adjustment, and note that the amplitude of the FM output signal does notchange. This is because the audio information is contained entirely in thesignals frequency and not in its amplitude.
14. By using the optional audio input module **ST2108** the human voice can be usedas the audio modulating signal, instead of using **ST2203**'s audio oscillatorblock. If you have an audio input module, connect the module's output to theaudio input socket in the modulator circuit’s block. The input signal to the audioinput module may be taken from an external microphone be (supplied with themodule) or from a cassette recorder, by choosing the appropriate switch settingon the module.

For FM Reactance Modulator:

6. Put the reactance /varactor switch in the reactance position. This switches theoutput of the reactance modulator through to the input of the mixer/amplifierblock~ and also switches off the varactor modulator block to avoid interferencebetween the two modulators.
7. The output signal from the reactance modulator block appears at tp.13, beforebeing buffered and amplified by the mixer/amplifier block. Although the outputfrom the reactance modulator block can be monitored directly at tp.13, anycapacitive loading affect this point (e.g. due to an oscilloscope probe) mayslightly affect the modulator's output frequency.In order to avoid this problem we will monitor the buffered FM output signalfrom the mixer/amplifier block at t.p. 34.
8. Put the reactance modulator's pot in its midway position (arrow pointingtowards top of PCB) then examine t.p. 34.**Note** that the monitored signal is a sinewave of approximately 1.2V peak/peakcentered

on 0 volts D.C. This is our FM carrier, and it is presently un-modulatedsince the reactance modulator's audio input signal has, zero amplitude.

9. The amplitude of the FM carrier (at t.p.34) is adjustable by means of themixer/amplifier block's amplitude pot, from zero to its present level. Try turningthis pot slowly anticlockwise, and note that the amplitude of the FM signal canbe reduced to zero.Return the amplitude pot to its fully clockwise position.

10. The frequency of the FM carrier signal (at t.p.34) should be approximately455Khz at the moment This carrier frequency can be varied from 453Khz to460Khz (approx.) by adjusting the carrier frequency pot in the reactancemodulator block.Turn this pot over its range of adjustment and note that the frequency of themonitored signal can be seen to vary slightly. Note also that the carrierfrequency is maximum when the pot is in fully clockwise position.

11. Try varying the amplitude & frequency pot in audio oscillators block, and alsosees the effect of varying the carrier frequency pot in the mixer/amplifiersblock.

12. Monitor the audio input (at t.p.6) and the FM output (at t.p.34) triggering theoscilloscope on the audio input signal. Turn the audio oscillator's amplitude potthroughout its range of adjustment and note that the amplitude of the FM outputsignal does not change. This is because the audio information is containedentirely in the signal's frequency, and not in its amplitude.

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.

QUIZ / ANSWERS:

Q.1. How are the types of FM? Write their names.

Ans. There two types of FM i.e. narrow band FM and wideband FM.

Q.2. What frequency deviation in FM?

Ans. The maximum change in instantaneous frequency from the average is known as frequency deviation.

Q.3. Which is the useful parameter for determination of bandwidth?

Ans. Frequency deviation is the useful parameter for determination of bandwidth.

Q.4. How many sidebands are there in FM?

Ans. Theoretically is infinite number sidebands in FM.

Q.5. Which sidebands are ignored in FM?

Ans. The sidebands with small amplitude are ignored in FM.

Q.6. Which are significant sidebands?
Ans. The sidebands having considerable amplitudes i.e. more than or equal to 1% of the carrier amplitude are known as significant sidebands.

Q.7. What is CCIR?
Ans. CCIR stands for Consultative Committee for International Radio.

Q.8. What is the indirect method of FM generation?
Ans. Armstrong method.

EXPERIMENT 7

AIM: To Detect or demodulate FM Signal using PLL & Foster-Seelay 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 thatthe demodulator must now extract the information signal from a frequency rather thanamplitude modulated wave.

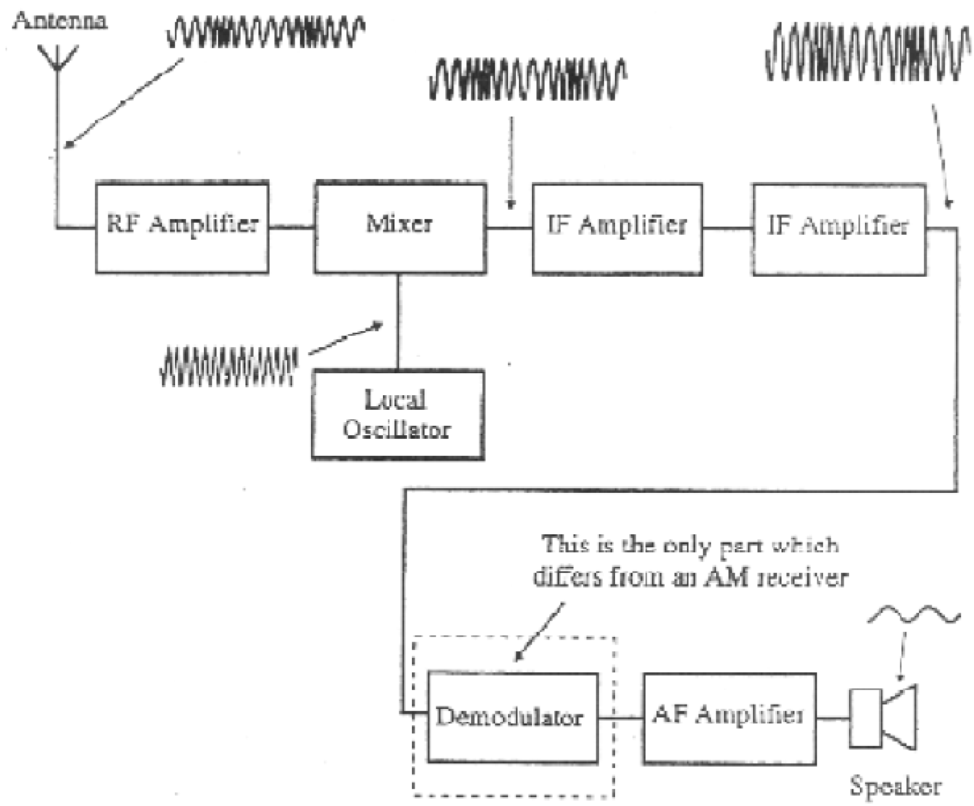


Figure 1: FM Receiver

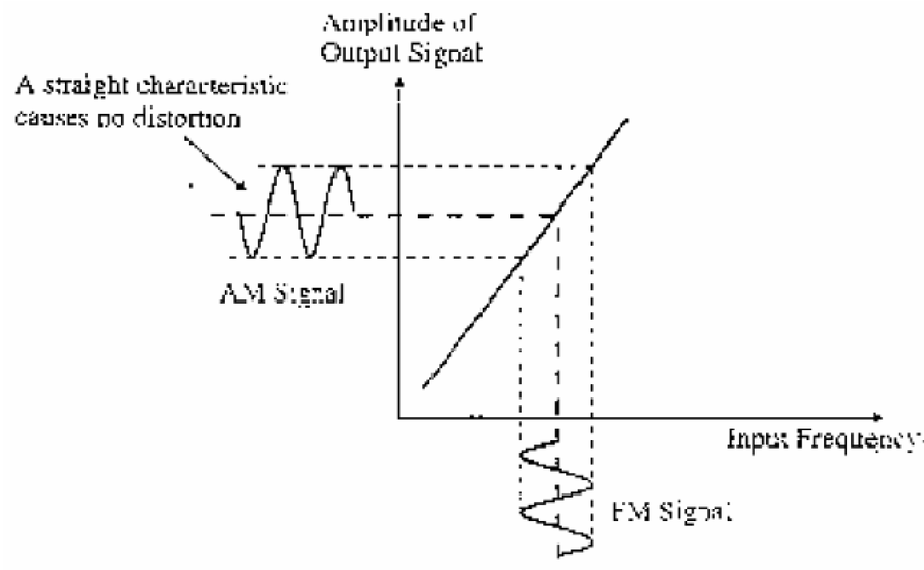


Figure 2: Voltage/Frequency Characteristics.

The basic requirement of any FM demodulator is therefore to convert frequencychanges into changes in voltage, with the minimum amount of distortion. To achievethis, it should ideally have a linear voltage/frequency characteristic, similar to that shown in figure 2. A 'demodulator' can also be called a 'discriminator' or a 'detector'.

PHASE LOCK LOOP DETECTOR

This is a demodulator that employs a phase comparator circuit. It is a very gooddemodulator and has the advantage that it is available, as a self-contained integratedcircuit so there is no setting up required. You plug it in and it works. For thesereasons, it is often used in commercial broadcast receivers. It has very low levels ofdistortion and is almost immune from external noise signals and provides very lowlevels of distortion. Altogether it is a very nice circuit.

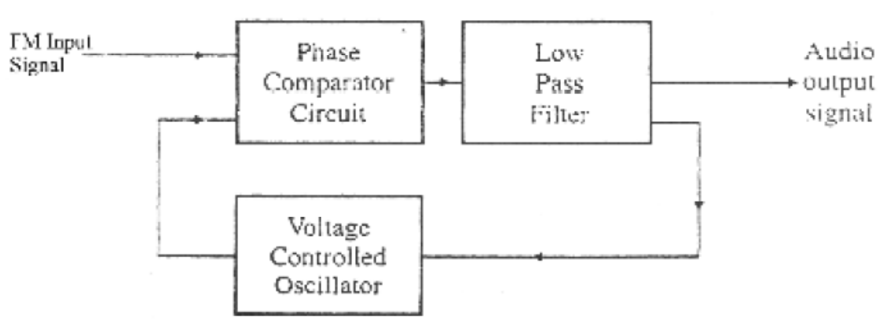
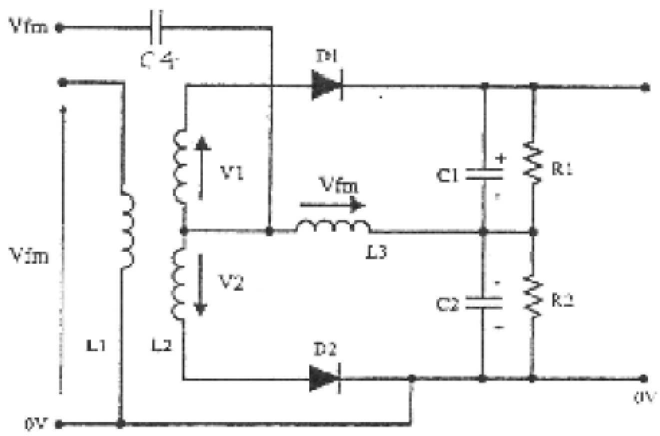


Fig. 3 Phase Lock Loop Detector

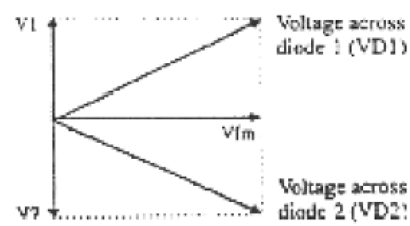
The overall action of the circuit may, at first, seem rather pointless. As we can see infig 3, there is a voltage-controlled oscillator (VCO). The DC output voltage from theoutput of the low pass filters controls the frequency of this oscillator. Now this DCvoltage keeps the oscillator running at the same frequency as the original input signaland 90° out of phase. And if we did, then why not just add a phase shifting circuit atthe input to give the 90° phase shift? The answer can be seen by imagining whathappens when the input frequency changes - as it would with a FM signal. If the input frequency increases and decreases, the VCO frequency is made to follow it. To dothis, the input control voltage must increase and decrease. These change of DCvoltage level that forms the demodulated signal. The AM signal then passes through asignal buffer to prevent any loading effects from disturbing the VCO and then throughan audio amplifier if necessary. The frequency response is highly linear as shown in figure 2.

FOSTER SEELEY DETECTOR

The foster Seeley circuit is shown in fig. 4. At first glance, it looks rather complicated but it becomes simpler ifwe consider it a bit at a time.



Circuit diagram



Phasor diagram

Figure 4: Foster –Seelay 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 ignoremany of the companieswe may recognize immediately that it consist of two envelope detectors like halfwave 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 outputvoltage would be zero volts since the capacitor voltages are in anti-phase and areequal 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-tappedcoil, they will form a parallel tuned circuit with a resonance frequency equal to theun-modulated carrier frequency.Capacitor C5 will shift the phase of the input signal by 90° with reference to thevoltage across L1 and L2. The voltages are shown as Va and Vb in the phasordiagram given in figure 39. Using the input signal Vfm as the reference, the phasordiagrams now look the way shown in figure 4. C4 is not important. It is only a DC blocking capacitor and hasnegligible impedance at the frequencies being used. But what it does do is to supply acopy of the incoming signal across L3. The entire incoming signal is dropped acrossL3 because C1 and C2 also have negligible impedance.If we return to the envelope detector section, we now have two voltages being appliedto each diode. One is V1 or V2 and the other is the new voltage across L3, which isequal to Vfm. When the input Frequency changes:If the input frequency increased above its un-modulated value, the phasor of Vawould fall below 90° due to the parallel tuned circuit becoming increasinglycapacitive.This would result in a larger total voltage being applied across D1 and areduced voltage across D2. Since the capacitor C1 would now charge to a highervoltage, the final output from the circuit would be a positive voltage.Conversely, if the frequency of the FM input signal decreased below the unmodulatedvalue, the phase shift due to capacitor C5 increases above 90° as theparallel tuned circuit becomes slightly inductive. This causes the voltage across diodeD2 to increase and the final output from the demodulator becomes negative. Theeffect of noise is to change the amplitude of the incoming FM signal resulting in aproportional increase and decrease in the amplitude of diode voltages VD1 and VD2and the difference in voltage is the demodulated output, the circuit is susceptible tonoise interference and should be preceded by a noise limiter circuit.

BLOCK DIAGRAM:

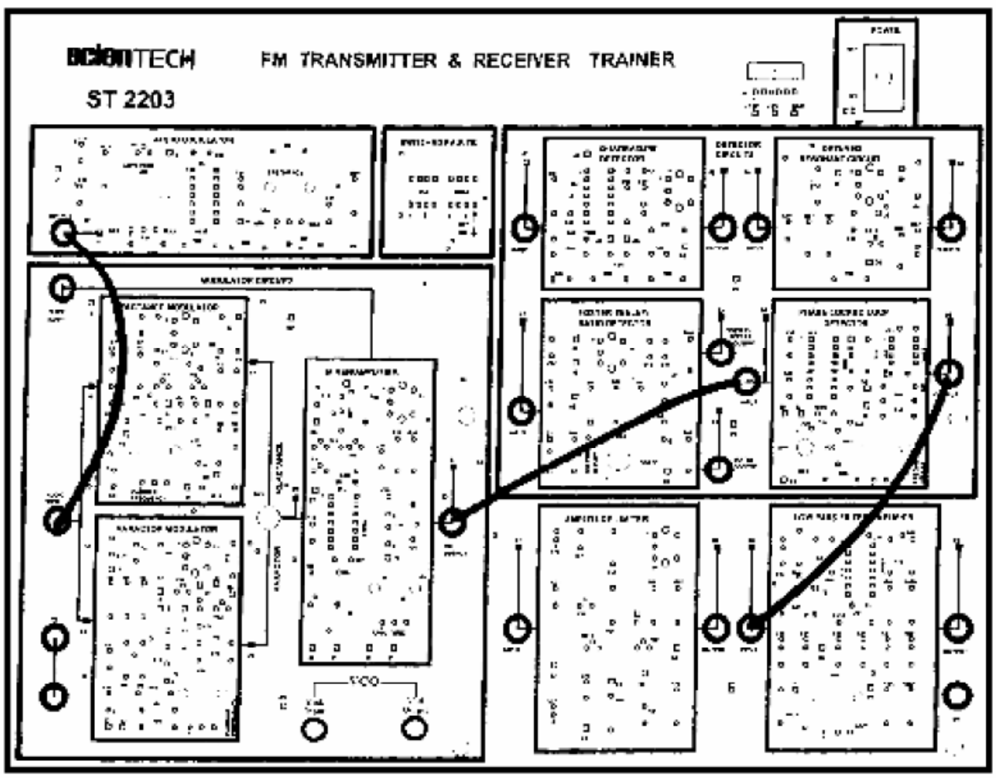


Figure 5: Connections for FM Demodulation using PLL

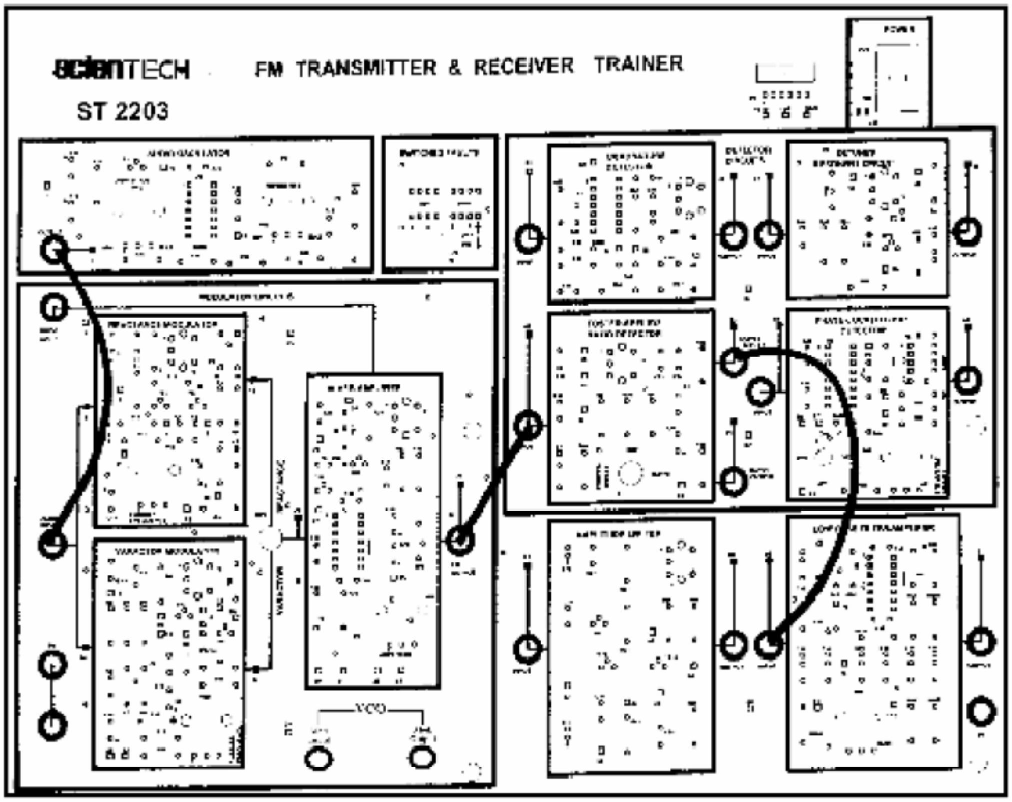


Figure 6: Connections for FM Demodulation using Foster-Seelay Detector

PROCEDURE:

FM Detection using PLL:

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. Ensurethat the following initial conditions exist on the **ST2203** 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 ON position.

2. Make the connections shown in figure 5.

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

4. Now monitor the audio input signal to the varactor modulator block (at t.p.14)together with the output from the phase-locked loop detector block (at t.p.60),triggering the oscilloscope in t.p.14. The signal at t.p.68 should contain threecomponents:

- A positive D.C. offset voltage.
- A sinewave at the same frequency as the audio signal at t.p.14.
- A high - frequency ripple component.

5. The low pass filter/amplifier block strongly attenuates the high-frequency ripplecomponent at the detector's output and also blocks the D.C. offset voltage.Consequently the signal at the output of the low- pass filter/amplifier block (att.p.73) should be very closely resemble the original audio making signal, if notthen slowly adjust the freq. adjust pot of PLL block.

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

FM Detection using Foster-Seelay Detector:

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 42

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, sincethis is the more linear of the two modulators, as fast as its frequency/voltagecharacteristic 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 midwayposition.

6. The audio oscillator's output signal (which appears at t.p.1) is now being usedby the varactor modulator, to frequency-modulate a 455Khz carrier sinewave.As we saw earlier, this FM waveform appears at the FM output socket from themixer/amplifier block. You will probably need to have an X-expansion controlon 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)A sinewave at the same frequency as the audio signal at t.p. 14.

b) A High frequency ripple component of small amplitude.

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

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

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

RESULT:

FM signal is being demodulated by using PLL and Foster-Seelay 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.

QUIZ / ANSWERS:

Q.1. How many types of FM are there? Write their names.
Ans. There two types of FM i.e. narrow band FM and wideband FM.

Q.2. What frequency deviation in FM?
Ans. The maximum change in instantaneous frequency from the average is known as frequency deviation.

Q.3. Which is the useful parameter for determination of bandwidth?
Ans. Frequency deviation is the useful parameter for determination of bandwidth.

Q.4. How many sidebands are there in FM?
Ans. Theoretically is infinite number sidebands in FM.

Q.5. Which sidebands are ignored in FM?
Ans. The sidebands with small amplitude are ignored in FM.

Q.6. Which are significant sidebands?
Ans. The sidebands having considerable amplitudes i.e. more than or equal to 1% of the carrier amplitude are known as significant sidebands.

Q.7. What is CCIR?
Ans. CCIR stands for Consultative Committee for International Radio.

Q.8. What is the indirect method of FM generation?
Ans. Armstrong method.

Q.9. What is the direct method of FM generation?
Ans. The parameter variation method.

EXPERIMENT 8

AIM: To observe the effects of pre-emphasis and De-emphasis on given input 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:

The noise has a effect on the higher modulating frequencies than on the lower ones. Thus, if thehigher frequencies were artificially boosted at the transmitter and correspondingly cut at the receiver, animprovement in noise immunity could be expected, thereby increasing the SNR ratio. This boosting ofthe higher modulating frequencies at the transmitter is known as **pre-emphasis** and the compensation atthe receiver is called **de-emphasis**.

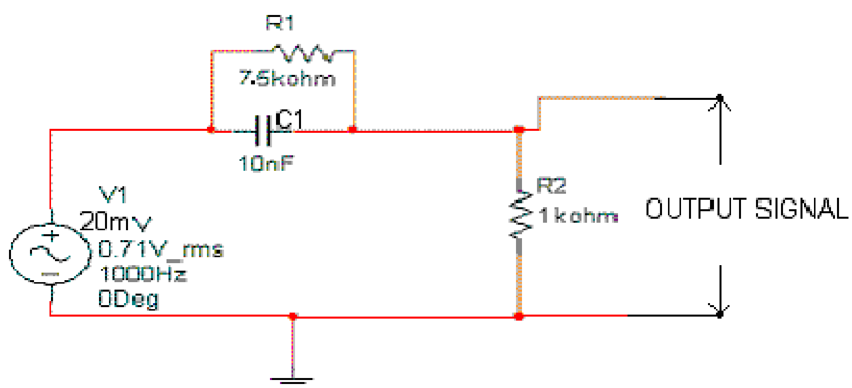


Figure 1: Pre-empasis Circuit

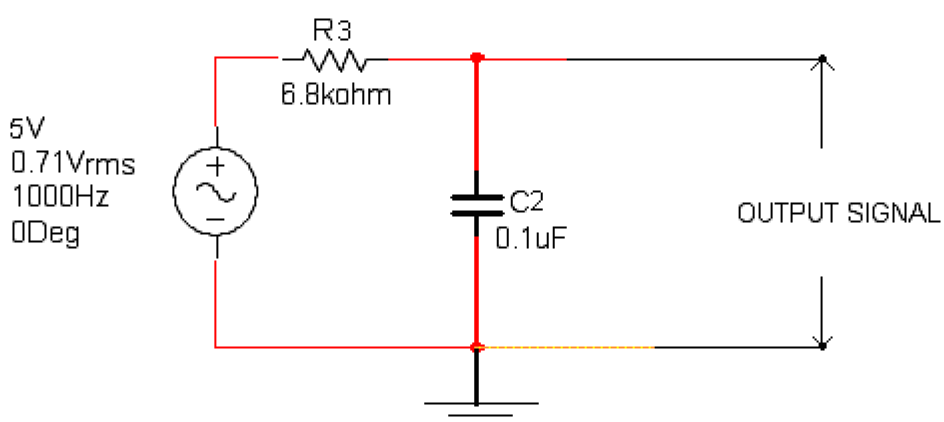


Figure 2: De-emphasis Circuit

PROCEDURE:

1. Connect the circuit as per circuit diagram as shown in Fig.1.
2. Apply the sinusoidal signal of amplitude 20mV as input signal to pre emphasis circuit.
3. Then by increasing the input signal frequency from 500Hz to 20KHz, observe the output voltage(vo) and calculate gain (20 log (vo/vi).
4. Plot the graph between gain Vs frequency.
5. Repeat above steps 2 to 4 for de-emphasis circuit (shown in Fig.2). by applying the sinusoidalsignal of 5V as input signal.

OBSERVATION TABLE:

RESULT:

Effect of pre-emphasis and de-emphasis on input signal is studied.

Table1: Pre-emphasis $V_i = 20mV$

Frequency(KHz)	Vo(mV)	Gain in dB(20 log V_o/V_i)

Table2: De-emphasis $V_i = 5v$

Frequency(KHz)	Vo(Volts)	Gain in dB(20 log V_o/V_i)

PRECAUTIONS:

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

QUIZ / ANSWERS:

Q.1. What do you mean by pre-emphasis?

Ans. In an FM system the higher frequencies contribute more tothe noise than the lower frequencies. Because of this all FMsystems adopt a system of preemphasis where the higherfrequencies are increased in amplitude before being used tomodulate the carrier. The transfer function sketched aboveis used for a preemphasis circuit for FM signals in the FMband. The Time T = 75µs. For FM systems in the FM band m ~ 5
resulting in a S/N improvement of 19dB. With preemphasisthis can be increased by 4dB for a total of 23dB.

Q.2. What do you mean by de-emphasis?

Ans. At the receiver the higher frequencies must be deemphasisedin order to get back the origunal baseband signal. Thetransfer function of the deemphasis circuit is shown above.

EXPERIMENT 9

AIM: To Study Super heterodyne AM receiver and measurement of receiver parameters viz.sensitivity, selectivity & fidelity.

APPARATUS REQUIRED:

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

BLOCK DIAGRAM:

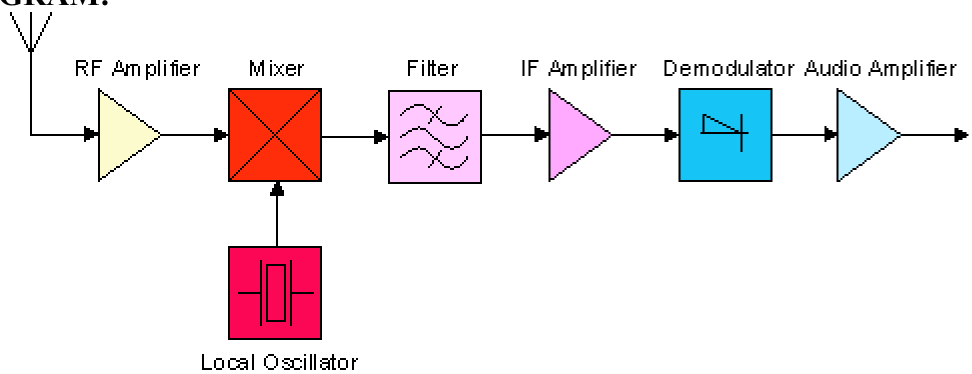


Figure 1: Superhetrodyne Receiver

THEORY:

The principle of operation of the superheterodyne receiver depends on the use of heterodyning or frequency mixing. The signal from the antenna is filtered sufficiently at least to reject the *image frequency* (see below) and possibly amplified. A local oscillator in the receiver produces a sine wave which mixes with that signal, shifting it to a specific intermediate frequency (IF), usually a lower frequency. The IF signal is itself filtered and amplified and possibly processed in additional ways. The demodulator uses the IF signal rather than the original radio frequency to recreate a copy of the original modulation (such as audio).

Figure 1. shows the minimum requirements for a single-conversion superheterodyne receiver design. The following essential elements are common to all superhet circuits: a receiving antenna, a tuned stage which may optionally contain amplification (RF amplifier), a variable frequency local oscillator, a frequency mixer, a band pass filter and intermediate frequency (IF) amplifier, and a demodulator plus additional circuitry to amplify or process the original audio signal (or other transmitted information).To receive a radio signal, a suitable antenna is required. This is often built into a receiver; especially in the case of AM broadcast band radios. The output of the antenna may be very small, often only a few microvolts. The signal from the antenna is tuned and may be amplified in a so-called radio frequency (RF) amplifier, although this stage is often omitted. One or more tuned circuits at this stage block frequencies which are far removed from the intended reception frequency. In order to tune the receiver to a particular station, the frequency of the local oscillator is controlled by the tuning knob (for instance). Tuning of the local oscillator and the RF stage may use a variable capacitor, or varicap diode. The tuning of one (or more) tuned circuits in the RF stage must track the tuning of the local oscillator.

Mixer stage

The signal is then fed into a circuit where it is mixed with a sine wave from a variable frequency oscillator known as the local oscillator (LO). The mixer uses a non-linear component to produce both sum and difference beat frequencies signals, each one containing the modulation contained in the desired signal. The output of the mixer may include the original RF signal at f_d, the local oscillator signal at f_{LO}, and the two new frequencies f_d+f_{LO} and f_d-f_{LO}. The mixer may inadvertently produce additional frequencies such as 3rd- and higher-order intermodulation products. The undesired signals are removed by the IF bandpass filter, leaving only the desired offset IF signal at f_{IF} which contains the original modulation (transmitted information) as the received radio signal had at f_d.

Intermediate frequency stage

The stages of an intermediate frequency amplifier are tuned to a particular frequency not dependent on the receiving frequency; this greatly simplifies optimization of the circuit.[6] The IF amplifier (or *IF strip*) can be made highly selective around its center frequency f_{IF}, whereas achieving such a selectivity at a much higher RF frequency would be much more difficult. By tuning the frequency of the local oscillator f_{LO}, the resulting difference frequency f_{LO} - f_d (or f_d-f_{LO} when using so-called low-side injection) will be matched to the IF amplifier's frequency f_{IF} for the desired reception frequency f_d. One section of the tuning capacitor will thus adjust the local oscillator's frequency f_{LO} to $f_d + f_{IF}$ (or. less often, to $f_d - f_{IF}$) while the RF stage is tuned to f_d. Engineering the multi-section tuning capacitor (or varactors) and coils to fulfill this condition across the tuning range is known as *tracking*.Other signals produced by the mixer (such as due to stations at nearby frequencies) can be very well filtered out in the IF stage, giving the superheterodyne receiver its superior performance. However, if f_{LO} is set to $f_d + f_{IF}$, then an incoming radio signal at $f_{LO} + f_{IF}$ will *also* produce a heterodyne at f_{IF}; this is called the *image frequency* and must be rejected by the tuned circuits in the RF stage. The image frequency is $2f_{IF}$ higher (or lower) than f_d, so employing a higher IF frequency f_{IF} increases the receiver's *image rejection* without requiring additional selectivity in the RF stage.Usually the intermediate frequency is lower than the reception frequency f_d, but in some modern receivers (e.g. scanners and spectrum analyzers) it is more convenient to first convert an entire band to a much higher intermediate frequency; this eliminates the problem of *image rejection*. Then a tunable local oscillator and mixer convert that signal to a second much lower intermediate frequency where the selectivity of the receiver is accomplished. In order to avoid interference to receivers, licensing authorities will avoid assigning common IF frequencies to transmitting stations. Standard intermediate frequencies used are 455 kHz for medium-wave AM radio,10.7 MHz for broadcast FM receivers, 38.9 MHz (Europe) or 45 MHz (US) for television, and 70 MHz for satellite and terrestrial microwave equipment.

Bandpass filter

The IF stage includes a filter and/or multiple tuned circuits in order to achieve the desired **selectivity**. This filtering must therefore have a band pass equal to or less than the frequency spacing between adjacent broadcast channels. Ideally a filter would have a high attenuation to adjacent channels, but maintain a flat response across the desired signal spectrum in order to retain the quality of the received signal. This may be obtained using one or more dual tuned IF transformers or a multipole ceramic crystal filter.

Demodulation:

The received signal is now processed by the **demodulator** stage where the audio signal (or other **baseband** signal) is recovered and then further amplified. AM demodulation requires the simple **rectification** of the RF signal (so-called **envelope detection**), and a simple RC low pass filter to remove remnants of the intermediate frequency. FM signals may be detected using a discriminator, **ratio detector**, or **phase-locked loop**. **Continuous wave** (**morse code**) and **single sideband** signals require a **product detector** using a so-called **beat frequency oscillator**, and there are other techniques used for different types of **modulation**. The resulting audio signal (for instance) is then amplified and drives a loudspeaker. When so-called high-side injection has been used, where the local oscillator is at a *higher* frequency than the received signal (as is common), then the frequency spectrum of the original signal will be reversed. This must be taken into account by the demodulator (and in the IF filtering) in the case of certain types of modulation such as **single sideband**.

RECEIVER CHARACTERISTICS:

The important characteristics of receivers are sensitivity, selectivity, & fidelity described as follows:

Sensitivity

The sensitivity of radio receiver is that characteristic which determines the minimumstrength of signal input capable of causing a desired value of signal output. Therefore,expressing in terms of voltage or power, sensitivity can be defined as the minimumvoltage or power at the receiver input for causing a standard output.In case of amplitude-modulation broadcast receivers, the definition of sensitivity hasbeen standardized as "amplitude of carrier voltage modulated 30% at 400 cycles,which when applied to the receiver input terminals through a standard dummyantenna will develop an output of 0.5 watt in a resistance load of appropriate valuesubstituted for the loud speaker".

Selectivity

The selectivity of a radio receiver is that characteristic which determines the extent towhich it is capable of differentiating between the desired signal and signal of otherfrequencies.

Fidelity

This is defined as the degree with which a system accurately reproduces at its output the essential characteristics of signals which is impressed upon its input.

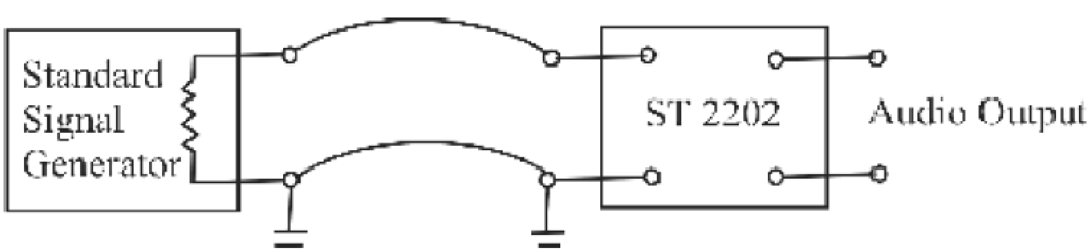


Figure 2: Setup for Determining Reciever Characteristics

Determination of receiver characteristics
A laboratory method for the measurement of receiver characteristics is shown in Fig. 2. We use here an artificial signal to represent the voltage that is induced in thereceiving antenna. This artificial signal is applied through 'dummy' antenna, which inassociation antenna with which the receiver is to be used. Substituting the resistanceload of proper value for the loudspeaker and measuring the audio frequency powerdetermine the receiver output.

Sensitivity
Sensitivity is a determined by impressing different RF voltages in series with astandard dummy antenna and adjusting the intensity of input voltage until standardoutputs obtained at resonance for various carrier frequencies. Sensitivity is expressedin microvolt.

Selectivity
Selectivity is expressed in the form of a curve that give the carrier signal strength withstandard modulation that is required to produce the standard test output plotted as afunction off resonance of the test signal.The receiver is tuned to the desired frequency and manual volume control is set formaximum value. At standard modulation, the signal generator is set at the resonantfrequency of the receiver. The carrier output of the signal generator is varied until thestandard test output is obtained. At the same tuning of receiver, the frequency ofsignal generator is varied above and below the frequency to which the receiver istuned. For every frequency, the signal generator voltage, applied to the receiver input, is adjusted to give the standard test output from the receiver

Fidelity
Fidelity is the term expressing the behavior of receiver output with modulationfrequency of input voltage. To obtain a fidelity curve, the carrier frequency of thesignal generator adjusted to resonance with the receiver, standard 400 cyclesmodulation is applied, the signal generator carrier level is set at a convenient arbitrarylevel and the manual volume control of the receiver is adjusted to give the standardtest output. The modulation frequency is then varied over the audio range, keepingdegree of modulation constant.
PROCEDURE:
(a) To Plot Selectivity of Receiver
1. Setting on ST2202
   a. Set the detector in diode mode.
   b. AGC on.
   c. Set the volume control full clockwise.
2. Apply AM signal with 400 Hz modulating frequency and 30% modulation taken from AM generator into Rx input socket.
3. Set the input carrier frequency to suitable value that lies within the AM band (525 KHz - 1600 KHz). Also set signal level to 100mV.
4. Tune the Receiver using tuning control. Also adjust gain potentiometer provided in R.F. amplifier section of ST2202 so as to get unclipped demodulated signal at detector's output (output of audio amplifier).
5. Note the voltage level at receiver's final output stage i.e. audio amplifier's output on CRO (voltage at resonance (Vr)).

6. Now gradually offset the carrier frequency in suitable steps of 5 KHz or 10 KHz below and above the frequency adjusted in step 2 without changing the tuning of receiver while maintaining the input signal level.
7. Now record the signal level at output of audio amplifier for different input carrier frequency, on CRO (i.e. voltage off resonance (Vi)).
8. Tabulate the readings as under:

Carrier Frequency	Output Voltage	Ratio = 20 log (Vi / Vr) dB

9. Plot the curve between ratio and carrier frequency.

(b) To Plot Sensitivity of Receiver

1. Setting on ST2202 :
   a. Set the detector in diode mode.
   b. AGC on.
   c. Set the volume control fully clockwise.
2. Apply AM signal, with 400 Hz modulating signal and 30% modulation, taken from AM generator into Rx input socket.
3. Set the input carrier frequency so as to lie within the AM Band (525 KHz – 1600 KHz). Also tune the detector to that carrier frequency using tuning control.(You will hear atone)
4. Set the input AM level to 100 mV. Also adjust the gain potentiometer provided in R.F. amplifier section of ST2202 so as to get unclipped demodulated signal at detectors output.
5. Record input carrier frequency & signal level at the final output stage i.e. output of audio amplifier (observed on CRO).
6. Change the input carrier frequency & also tune the receiver to that frequency & repeat step 4.
7. Tabulate the collected readings as under:

Carrier frequency	Output (pp)

8. Plot the graph between carrier frequency & output level.

(c) To Plot Fidelity of Receiver

1. Setting on ST220:
   a. Set the detector in diode mode.
   b. AGC on.
   c. Set the volume control fully clockwise.
2. Apply AM signal of 100mV with 400Hz modulating signal and 30% modulation, into Rx input.
3. Select a suitable carrier frequency that lies within AM Band (525 KHz - 1600 KHz). Tune the ST2202 receiver to that frequency using tuning control. Also adjust gain potentiometer provided in R.F. amplifier section so as to get unclipped demodulated signal at detector's output.

4. Note the demodulated signal level (Vr) at the final output stage i.e. output of audio amplifier (on CRO) for the applied AM signal with 400Hz modulating signal.
5. Now vary the modulating signal frequency over audio range (300 Hz-3 KHz) in suitable steps say 100Hz. Note the corresponding output level (Vi) at the output of audio amplifier (on CRO).
6. Tabulate readings as under:

Carrier frequency	Modulating frequency	Output Voltage

Relative response = 20 log (Vi / Vr) dB
7. Plot the graph between modulating frequency and relative response.

RESULT:
Superhetrodyne receiver has been studied and plot for receiver parameters viz. sensitivity, selectivity and fidelity has been drawn.

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.

EXPERIMENT 10

AIM:-To study Pulse Amplitude Modulation.

APPARATUS REQUIRED:
CRO, experimental kit, power supply, connecting leads.

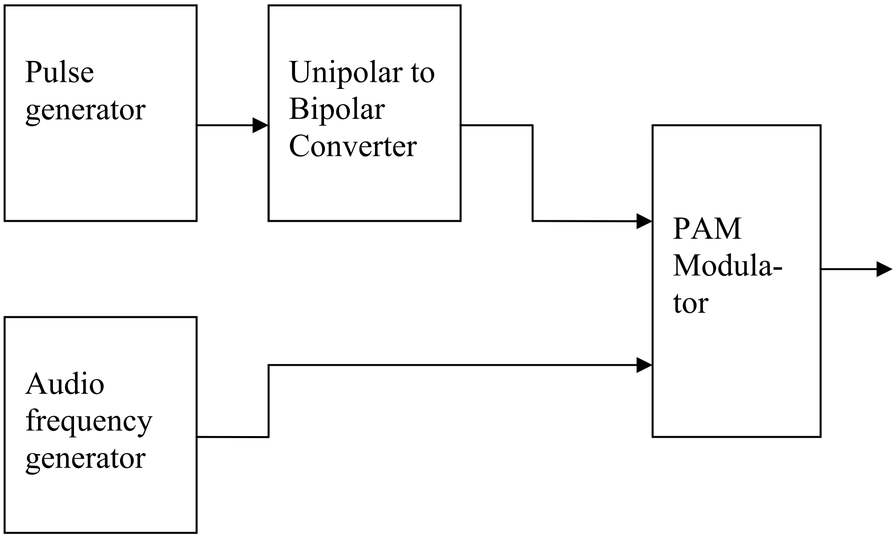
BRIEF THEORY:

Pulse Modulation

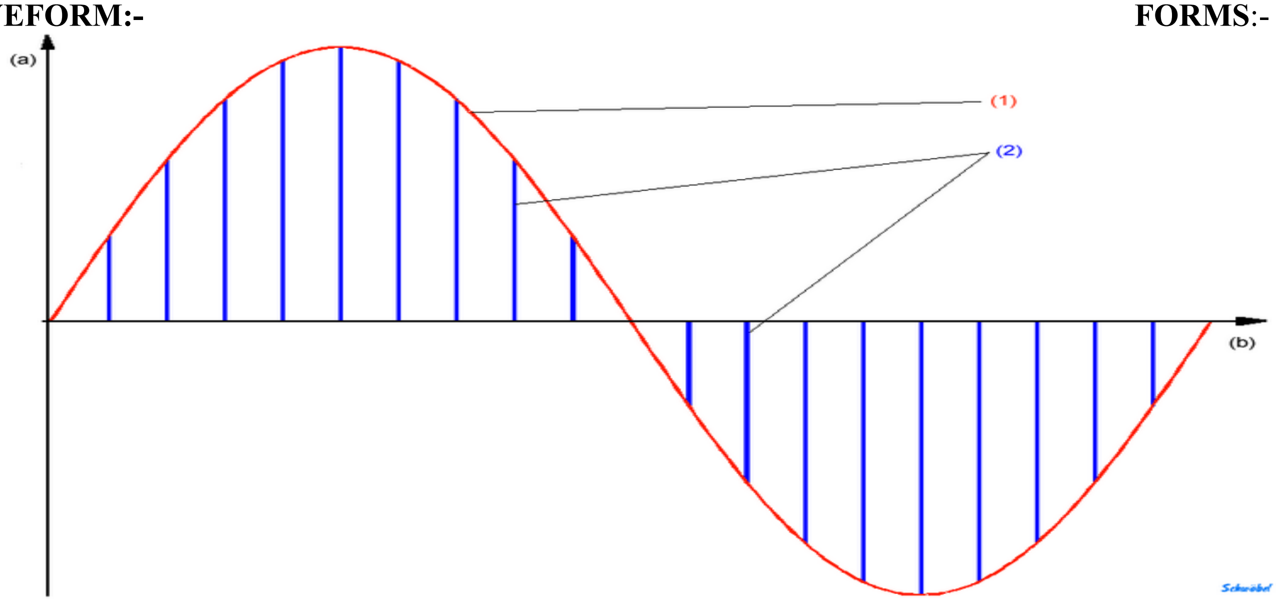
We know that in Analog modulation systems, some parameter of a sinusoidal carrier (continuous in time domain) is varied according to the instantaneous value of the modulating signal. But in pulse modulation methods, the carrier is no longer a continuous signal but consists of a train of uniform pulses having a defined PRF (Pulse Repetition Frequency). The continuous modulating message signal waveforms are sampled at regular intervals. Information regarding the signal is transmitted only at the sampling times, together with any synchronizing pulses that may be required. At the receiving end, the original wave form may be reconstituted with negligible distortion from the information regarding the samples, if these samples are taken with minimum sufficient frequency.

In Pulse Modulation some parameter of the pulsed carrier is varied according to the instantaneous value of the modulating signal. Pulse modulation may be broadly subdivided into two categories: Analog&Digital. In the former, the indication of sample amplitude may be infinitely variable, while in the latter a code which indicates the sample amplitude to the nearest pre-determined level is sent. Pulse-Amplitude&Pulse-Time Modulation are both analog while the Pulse-code and Delta modulation are both digital. All the modulation systems have sampling in common, but they differ from each other in the manner of indicating the sample amplitude.In PAM the signal is sampled at a regular intervals and each sample is made proportional to the instant of sampling.In single polarity PAM is fixed, AC level is acted to ensure that all the pulse are +Ve going. The frequency spectrum is decaying but with decaying amplitude. The rate of decay depends upon the width of the pulses. As the pulses are made wider, the spectrum decays faster.

BLOCK DIAGRAM



WAVEFORM:- **FORMS:-**



Principle of PAM; (1) original Signal, (2) PAM-Signal, (a) Amplitude of Signal, (b) Time

PROCEDURE:

1. Make the connection according to the block diagram.
2. Connect pulse generator to the unipolar to bipolar converter
3. Connect the audio frequency of 2 KHz, 2V to modulator.
4. Connect the modulator output to CRO.
5. Observe output on CRO.

RESULT:
Pulse modulated waveform is obtained on CRO.

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.

QUIZ / ANSWERS:-

Q.1. What is pulse amplitude modulation?
Ans. Amplitude of the sampled pulse train (Carrier Signal) is varied according to the instantaneous amplitude of modulating signal.

Q.2. How many types of pulse modulation?
Ans. There are two types of PM i.e. Pulse Amplitude Modulation (PAM) and
Pulse Time Modulation (PTM).

Q.3. How many types of pulse time modulation?
Ans. There are two types of PTM i.e. Pulse Width Modulation (PWM) and
Pulse Position Modulation (PPM).

Q.4. What is base band signal?
Ans. When the basic message signal is transmitted without a frequency translation means modulation is known as base band signal.

Q.5. How many types of pulse amplitude modulation?
Ans. There are two types of PAM i.e. Single Polarity and Double Polarity.

Q.6. Write the demodulation method of Plato sample signal.
Ans. (1) using an equalizer (2) using holding circuit.

Q.7. Which filter is used in PAM demodulator circuit?
Ans. Second order low pass filter.

Q.8. Write the two method of multiplexing
Ans. (1) FDM (2) TDM

Q.10. What is the merit of flat top sampling?
Ans. The tops of pulses are flat thus the pulses have constant amplitude with in the pulse Interval.

EXPERIMENT 11

AIM:- To study the pulse width modulation.

APPARATUS REQUIRED:
CRO, experimental kit, power supply, connecting leads

BRIEF THEORY:
Pulse width modulation is a part of PTM modulation. The PWM is also called PDM (pulse duration modulation) and sometimes it is also called PLM (pulse length modulation).

In PWM width of each pulse depends on the instantaneous value of the base band signal at the sampling instant. In pulse width modulation continuous waveform is sampled at regular intervals and the width of each pulse is kept proportional to the magnitude of signal at that instant in PWM. In pulse width modulation pulse is varied accordance with the modulating signal but the amplitude and starting time of each pulse is fixed .In PWM, the information about the base band signal lies in the trailing edge of the pulse

PWM has the disadvantage, when compared with PPM that its pulses are of varying width and therefore of varying power content .This means that transmitter must be powerful enough to handle the maximum- width pulses, although the average power transmitted is perhaps only half of the peak power. PWM still works if synchronization between transmitter and receiver fails.

Generation and Demodulation of PWM
PWM may be generated by applying trigger pulses to control the duration of these pulses.The emitter coupled mono-stable multi-vibrator is used as voltage to time converter, since its gate width is dependent on the voltage to which the capacitor C is charged .If this voltage is varied in accordance with a signal voltage, a series of rectangular pulses will be obtained, with widths varying as required. The demodulation of pulse width modulation is a simple process. PWM is fed to an integrating circuit from which a signal emerges whose amplitude at any time is proportional to the pulse width at that time.

BLOCK DIAGRAM:

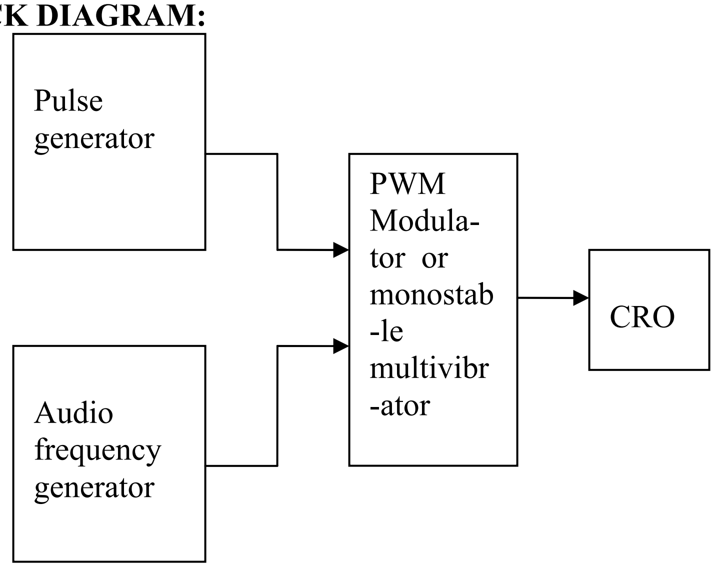


Figure 1: Block diagram of pulse width modulation

PROCEDURE:

1. Make the connection according to the block diagram.
2. Connect the audio frequency of 2 KHz, 2V to modulator.
3. Connect the modulator output to CRO.
4. Switch ON the power supply.
5. Observe output on CRO.

OUTPUT WAVE FORMS:-

```
           _      _      _      _      _      _      _       _
          | |    | |    | |    | |    | |    | |    | |     | |
Clock     | |    | |    | |    | |    | |    | |    | |     | |
        __| |____| |____| |____| |____| |____| |____| |_____| |____



                 _      __     ____          ____  _
Data            | |    |  |   |    |        |    | | |
                | |    |  |   |    |        |    | | |
        ________| |____|  |___|    |________|    |_| |___________



Data       0      1      2      4      0      4      1      0
```

RESULT:

Pulses width modulated wave is obtained on CRO.

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.

QUIZ:

Q.1 Which modulation is PWM?

Ans. Analog Modulation

Q.2 What are two categories of Pulse Modulation.?

Ans.! Analog Modulation 2. Digital Modulation

Q.3 What is the unit of signaling speed?

Ans. Baud

Q.4 What is the disadvantage of PWM?

Ans. Due to varying of pulses width power contents of PWM also varying

Q.5 Which multivibrator is used for PWM?
Ans. Monostable Multivibrator

Q.6 Which circuit is used for PWM demodulator?
Ans. Integrating circuit.

Q.7 What is difference between PAM and PWM ?
Ans. In PAM, amplitude of pulse is varied according to modulating signal and in PWM,width is varied of pulses.

Q.8 How PWM may be generated?
Ans. PWM may be generated applying trigger pulses to contol the starting time of pulses from a monostable multivibrator.

Q.9 What is the use of sampling theorem?
Ans. Sampling Theorem is used to determine minimum sampling speed.

Q.10 What is the world wide standard sampling rate?
Ans. Eight thousand samples per second.

EXPERIMENT 12

AIM:-To study the pulse position modulation.

APPARATUS REQUIRED:
CRO, experimental kit, power supply, connecting leads.

THEORY:
In pulse position modulating the amplitude of pulse is kept constant and position of the pulse in relation to the position of the reference pulse or synchronize pulse is varied by each sample value of modulating signal.
PPM may be obtained from PWM. In PWM each pulse has a leading edge and a trailing edge but the location of leading edge are fixed where trailing edge are depends on the pulse width. Thus PPM may be obtained from PWM by simply getting side of the leading edger and slots tops of PWM pulses.In pulse position modulating the amplitude of pulse is kept constant and position of the pulse in relation to the position of the reference pulse or synchronize pulse is varied by each sample value of modulating signal.In PWM each pulse has a leading edge and a trailing edge but the location of leading edge are fixed where trailing edge are depends on the pulse width. The trailing edges of PWM pulses are in fact position modulated. Thus PPM may be obtained from PWM by simply getting rid of the leading edge and slots tops of PWM pulses.In comparison with PWM, PPM has the advantage of requiring constant transmitter power output, but the disadvantage of depending on transmitter –receiver synchronization.

Generation and demodulation of PPM:
PPM may be generated from PWM easily. First of all, PWM pulses are generated and then they are differentiated. The result is another pulse train which has positive going narrow pulses corresponding to leading edges and negative going narrow pulses corresponding to trailing edges. If the position corresponding to the trailing edges of an un-modulated PWM pulse is counted as zero displacement, then the trailing edges of a modulated pulse will arrive earlier or later. An unmodulated PWM pulse is one that is obtained when the instantaneous signal value is zero. The differentiated pulses corresponding to the leading edges are removed with a diode clipper and the remaining pulses are nothing but position modulated output.When the PPM is demodulated in the receiver, it is again first converted into PWM by using flip-flop or bistable multivibrator. One input of the multivibrator receives trigger pulses from a local generator which is synchronized by trigger pulses received from the transmitter, and these triggers are used to switch off one of the stages of the flip-flop. The PPM pulses are fed to the other base of the flip-flop and switch that stage ON. The period of time during which this particular stage is OFF, depends on the time difference between the two triggers, so that the resulting pulse has a width that depends on the time displacement of each individual PPM pulse. The PWM pulse train thus obtained is a demodulated output.

BLOCK DIAGRAM:

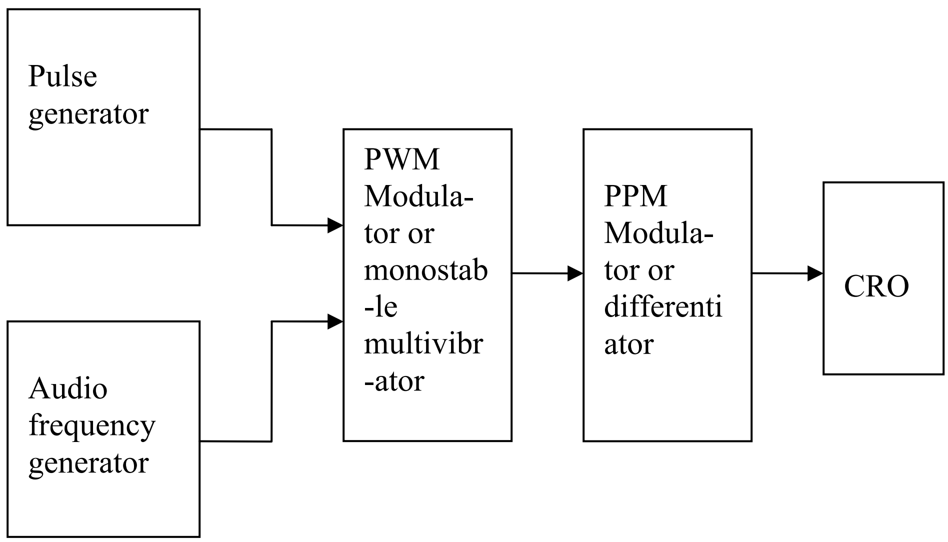
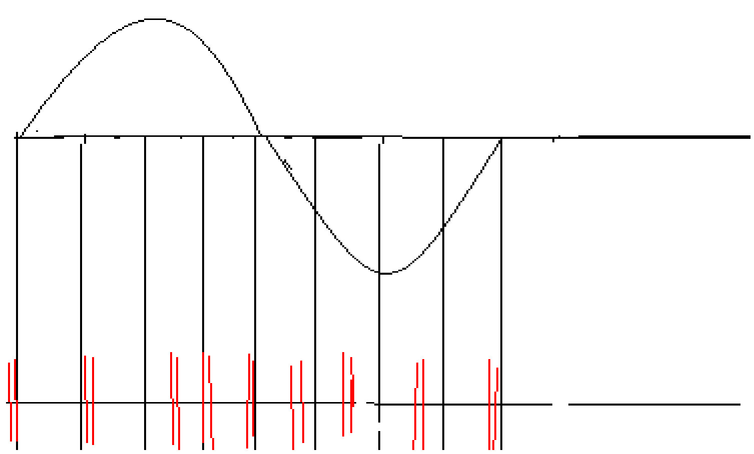


Figure 1: Block diagram of Pulse Position Modulation

PROCEDURE:
1. Make the connection according to the block diagram.
2. Connect the audio frequency of 2 KHz, 2V to modulator.
4. Connect the PWM output to the PPM modulator.
4. Connect the PPM modulator output to CRO.
5. Switch ON the power supply.
6. Observe output on CRO

OUTPUT WAVEFORMS:



RESULT:
The Pulse Position Modulated wave is obtained on CRO.

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

QUIZ:

Q.1 What is advantage of PPM?
Ans. It has no varying width of pulse so power content are not varying.

Q.2 What is PPM?
Ans. In PPM the position of pulses is varied and width and amplitude are constant.

Q.3 Which Multivibrator is used for PPM De-modulator?
Ans. Bi-stable Multivibrator.

Q.4 What is the difference between PPM & PWM?
Ans. In PWM, the width is varied and in PPM, the position is varied according to modulating signal.

Q5. Which filter is used in PPM demodulator?
Ans. Second order low pass filter.

Q6. In which category of PM is PPM?
Ans. Analog Modulation.

Q7. Which modulation is similar to PDM?
Ans. Phase modulation.

Q8. At which factor the band-width of PPM depends?
Ans. Bandwidth of transmission channel depends on rising time of the pulse.

Wires.

EXPERIMENT 13

AIM: -Study of Amplitude Shift Keying .

APPARATUS REQUIRED: -ASK modulation kit, CRO and connecting leads.

THEORY

The binary ASK System was one of the earliest forms of digital modulation used in wireless telegraphy. This simplest form of digital modulation is no longer used widely in digital communication .Nevertheless it serves as a useful model which helps in understanding certain concepts. In an ASK system, binary symbol 1 is represented by transmitting a sinusoidal carrier wave of fixed amplitude Ac and fixed frequency f_c for the bit duration T_b seconds whereas binary symbol 0 is represented by switching off the carrier for T_b seconds. This signal can be generated by switching off the carrier of a sinusoidal oscillator on and off for the prescribed periods indicated by the modulating pulse train. For this reason the scheme is also known as on-off keying (OOK).

Let the sinusoidal carrier be represented by

$e_c(t)=A_c \cos(2\pi f_c t)$

Then, the binary ASK signal can be represented by a wave s(t) given by

$S(t) = A_c \cos(2\pi f_c t)$ symbol 1

$=0,$ symbol 0

A typical ASK waveform is illustrated in figure for a binary data represented by {10110101}

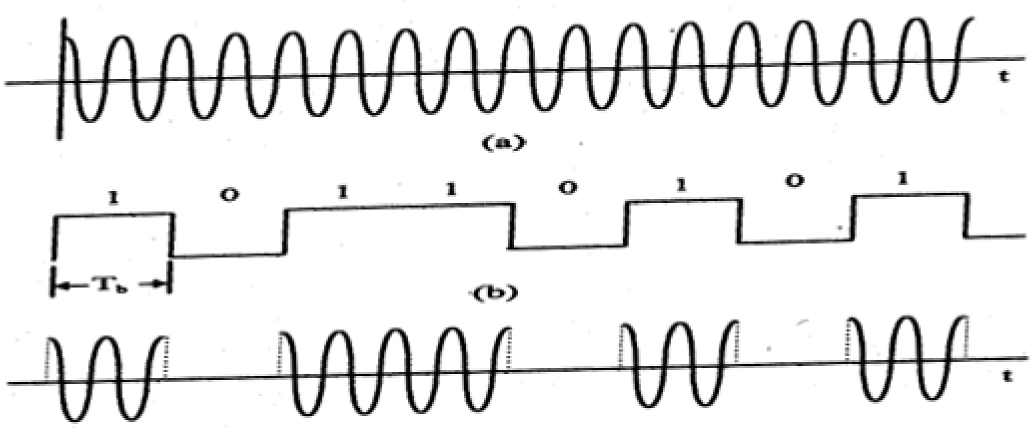


Figure1: ASK wave forms: (a) Unmodulated carrier (b) Unipolar bit sequence (c) ASK wave.

Generation Of ASK Signal

ASK signal can be generated by applying the incoming binary data (represented in unipolar form) and the sinusoidal carrier to the two inputs of a product modulator (balanced modulator) The resulting output is the ASK wave. This is illustrated in figure modulation causes a shift of the baseband signal spectrum.

The ASK signal which is basically the product of the binary sequence and the carrier signal .

Block Diagram:-

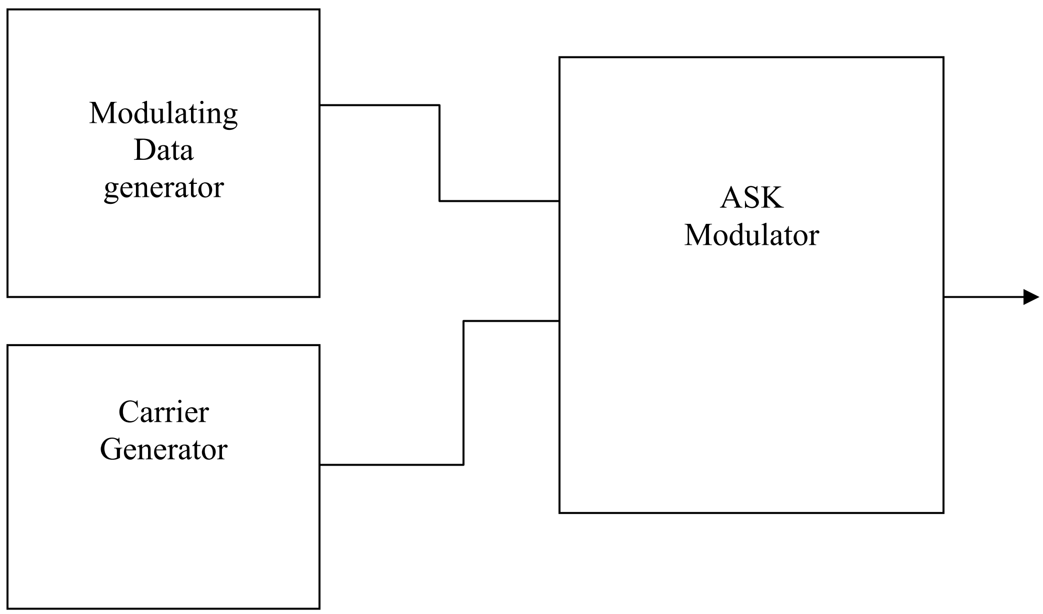


Figure 2: Block diagram for ASK Generation

PROCEDURE:

1. Make the connection according to the circuit diagram.
2. Connect the modulator output to CRO.
3. Observe output on CRO.

RESULTS:- ASK output is obtained on CRO.

PRECAUTIONS:-

I. 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.

EXPERIMENT 14

AIM:-To study the Phase Shift Keying.

APPARATUS REQUIRED:
CRO, experimental kit, power supply, connecting leads.

BRIEF THEORY:

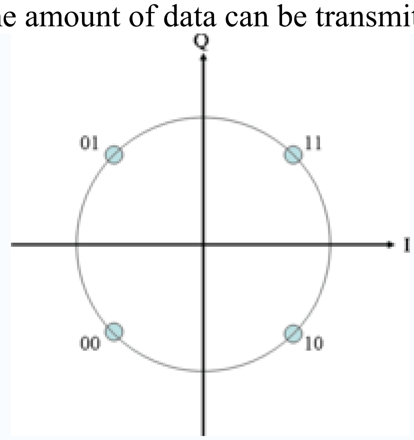
Phase-shift KeyingPSK
PSK involves the phase change at the carrier sine wave between 0 to 180 in accordance with the data stream to be transmitted .

PSK modulator is similar to ASK modulator both used balanced modulator to multiply the carrier with balanced modulator signal. The digital signal with applied to modulation input for PSK generation is bipolar i.e. equal positive and negative voltage level.

When the modulating input is positive the out put at modulator is a line wave in phase with the carrier input whereas for positive voltage level, the output of modulator is a sine wave which is switched out of phase by 180 from the carrier input.

Quadrature Phase-shift Keying (QPSK)

QPSK:- in QPSK each pair at consecutive data bit is treated as a two bit code which is switch the phase of the carrier sine wave between one at four phase 90º apart. The four possible combinations at bib it code are 0º, 01, 10, and 11 each code represents either a phase of 45º, 185º, 225º, and 315º lagging, relative to the phase at the original un modulated carrier QPSK offers an advantage over PSK is a no carrier that how each phase represents a two bit code rather than a single bit. This means that either we can charge phase per sec. or the same amount of data can be transmitted with.



Constellation diagram for QPSK with Gray coding. Each adjacent symbol only differs by one bit.

Sometimes known as quaternary or quadriphase PSK or 4-PSK, QPSK uses four points on the constellation diagram, equispaced around a circle. With four phases, QPSK can encode two bits per symbol, shown in the diagram with Gray coding to minimize the BER — twice the rate of BPSK. Analysis shows that this may be used either to double the data rate compared to a BPSK system while maintaining the bandwidth of the signal or to maintain the data-rate of BPSK but halve the bandwidth needed.

Although QPSK can be viewed as a quaternary modulation, it is easier to see it as two independently modulated quadrature carriers. With this interpretation, the even (or odd) bits are used to modulate the in-phase component of the carrier, while the odd (or even) bits are used to modulate the quadrature-phase component of the carrier. BPSK is used on both carriers and they can be independently demodulated

BLOCK DIAGRAM:-

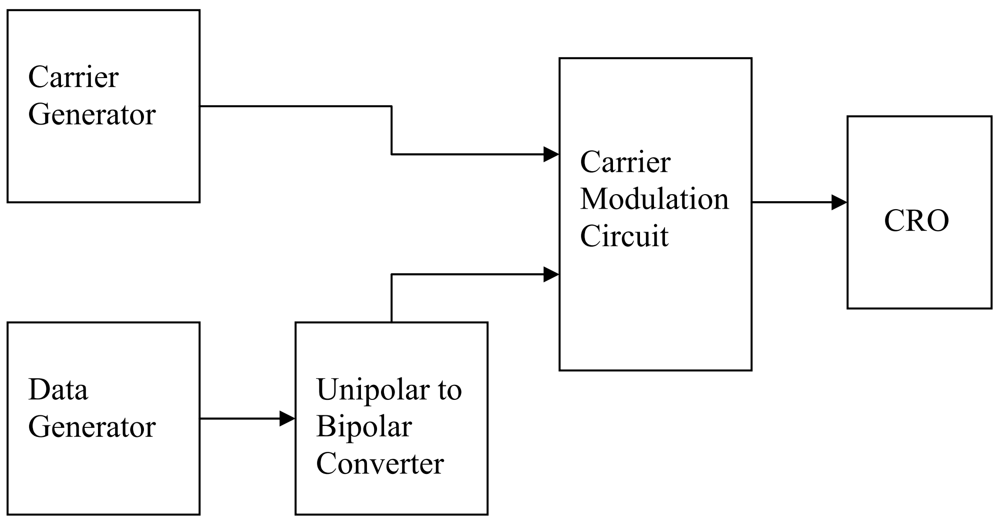
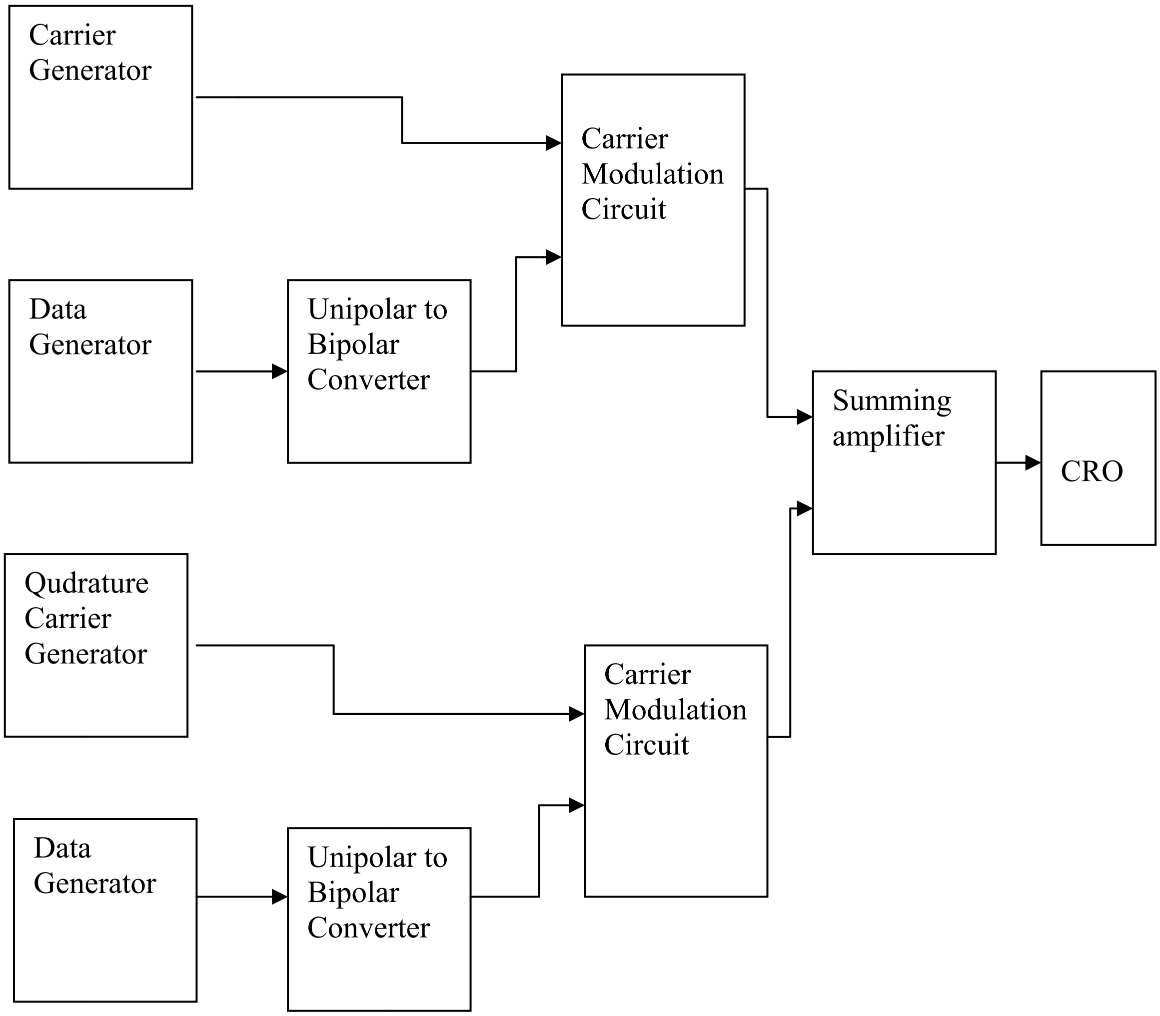
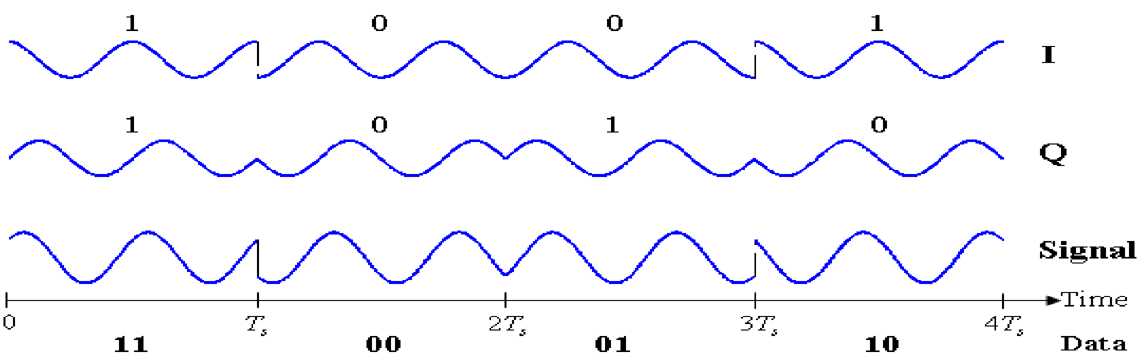


Figure 1: Block diagram of PSK



PROCEDURE:
1. Make the connection according to the block diagram.
2. Connect the modulator output to CRO.
3. Observe output on CRO.

WAVE FORM:



RESULT:
PSK and QPSK output is obtained on CRO.

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

QUIZ:

Q.1 What is PSK?
Ans. It is one of the basic digital modulation technique. Here the phase of the carrier is switched depending upon the input digital signal. This is similar to the Phase Modulation and has constant Amplitude envelope.

Q.2 What is the disadvantage of PSK?
Ans. It needs a complicated synchronizing circuit of the receiver.

Q.3 What is BPSK?
Ans. BPSK is Binary Phase Shift Keying. Here binary symbol 1& 0 modulate the phase of the carrier. The phase of carrier change by 180°

Q.4 How BPSK is generated?
Ans. It can be generated by applying carrier signal and base-band signal as modulating signal to a balanced modulator.

Q.5 What is the advantage of PSK?
Ans. Error rate is less than DPSK.

Q.6 What is the difference between QPSK and BPSK?
Ans. In BPSK phase shift is 180° where as in PSK the phase shift is 45°.

Q.7 What is QPSK?
Ans. In QPSK two successive bits are combined. This combination of two bits forms four distinguishing symbols. When the symbol is changed to next symbol the phase of carrier is changed by 45°.

Q.8 How QPSK is generated?
Ans. The input binary sequence is first converted to a bipolar NRZ type of signal called b(t) than it is divided by demultipluxer and added together after insertion of carrier. The generates QPSK signal.
Q.9 How QPSK is detected?

Ans. Basically QPSK receiver uses a synchronous reception. A coherent carrier applied to the two synchronous demodulator, each consists of a multiplier and an integrator. The output is detected original signal.

Q.10 What is DPSK?
Ans. DPSK is differential phase shift keying and is a non-coherent versus of PSK.DPSK does not need a coherent carrier at the demodulator. The input sequence of binary bits is modified such that the next bit depends upon the previous bit.