CODE OF CONDUCT FOR THE LABORATORIES

➢ All students must observe the Dress Code while in the laboratory.

➢ Sandals or open-toed shoes are NOT allowed.

➢ Foods, drinks and smoking are NOT allowed.

➢ All bags must be left at the indicated place.

➢ The lab timetable must be strictly followed.

➢ Be PUNCTUAL for your laboratory session.

➢ Program must be executed within the given time.

➢ Noise must be kept to a minimum.

➢ Workspace must be kept clean and tidy at all time.

➢ Handle the systems and interfacing kits with care.

➢ All students are liable for any damage to the accessories due to their own negligence.

➢ All interfacing kits connecting cables must be RETURNED if you taken from the lab supervisor.

➢ Students are strictly PROHIBITED from taking out any items from the laboratory.

➢ Students are NOT allowed to work alone in the laboratory without the Lab Supervisor

➢ USB Ports have been disabled if you want to use USB drive consult lab supervisor.

➢ Report immediately to the Lab Supervisor if any malfunction of the accessories, is there.

Before leaving the lab

➢ Place the chairs properly.

➢ Turn off the system properly

➢ Turn off the monitor.

➢ Please check the laboratory notice board regularly for updates.

GENERAL LABORATORY INSTRUCTIONS

You should be punctual for your laboratory session and should not leave the lab without the permission of the teacher.

Each student is expected to have his/her own lab book where they will take notes on the experiments as they are completed. The lab books will be checked at the end of each lab session. Lab notes are a primary source from which you will write your lab reports.

Organization of the Laboratory

It is important that the programs are done according to the timetable and completed within the scheduled time.

You should complete the prelab work in advance and utilize the laboratory time for verification only.

The aim of these exercises is to develop your ability to understand, analyze and test them in the laboratory.

A member of staff and a Lab assistant will be available during scheduled laboratory sessions to provide assistance.

Always attempt program first without seeking help.
When you get into difficulty; ask for assistance.

Assessment

The laboratory work of a student will be evaluated continuously during the semester for 25 marks. Of the 25 marks, 15 marks will be awarded for day-to-day work. For each program marks are awarded under three heads:

- ➢ Prelab preparation – 5 marks
- ➢ Practical work – 5marks, and
- ➢ Record of the Experiment – 5marks

Internal lab test(s) conducted during the semester carries 10 marks.

End semester lab examination, conducted as per the JNTU regulations, carries 50 marks.

At the end of each laboratory session you must obtain the signature of the teacher along with the marks for the session out of 10 on the lab notebook.

Lab Reports
Note that, although students are encouraged to collaborate during lab, each must individually prepare a report and submit.

They must be organized, neat and legible.

Your report should be complete, thorough, understandable and literate.

You should include a well-drawn and labeled engineering schematic for each circuit investigated

Your reports should follow the prescribed format, to give your report structure and to make sure that you address all of the important points.

Graphics requiring- drawn straight lines should be done with a straight edge. Well drawn free-hand sketches are permissible for schematics.

Space must be provided in the flow of your discussion for any tables or figures. Do not collect figures and drawings in a single appendix at the end of the report.

Reports should be submitted within one week after completing a scheduled lab session.

Presentation

Experimental facts should always be given in the past tense.

Discussions or remarks about the presentation of data should mainly be in the present tense.

Discussion of results can be in both the present and past tenses, shifting back and forth from experimental facts to the presentation.

Any specific conclusions or deductions should be expressed in the past tense.

Report Format:

Lab write ups should consist of the following sections:

Aim:

Equipments:

Circuit Diagram:

Theory:

Procedure:

Expected Waveform:

Observation and Calculations:

Results:

Conclusions:

.

Note: Diagrams if any must be drawn neatly on left hand side.

LIST OF EXPERIMENTS

1. Amplitude Modulation and Demodulation

2. DSB – SC Modulator & Detector

3. SSB-SC Modulator & Detector (Phase Shift Method)

4. Frequency Modulation and Demodulation

5. Study of Spectrum analyzer and Analysis of AM and FM Signals

6. Pre-emphasis & De-emphasis

7. Time Division Multiplexing & De multiplexing

8. Frequency Division Multiplexing & De multiplexing

9. Verification of Sampling Theorem

10. Pulse Amplitude Modulation & Demodulation

11. Pulse Width Modulation & Demodulation

12. Pulse Position Modulation & Demodulation

13. Frequency Synthesizer

14. AGC Characteristics

15. PLL as FM Demodulator

Annexure I : Additional Experiments

1. Characteristics of Mixer

2. Digital Phase Detector

3. Coastas Receiver

EXPERIMENT NO : 1

AMPLITUDE MODULATION & DEMODULATION

OBJECTIVE:

To study

1. Amplitude modulation process
2. Measurement of Modulation Index
3. Effect of 100% Modulation, over modulation, under modulation
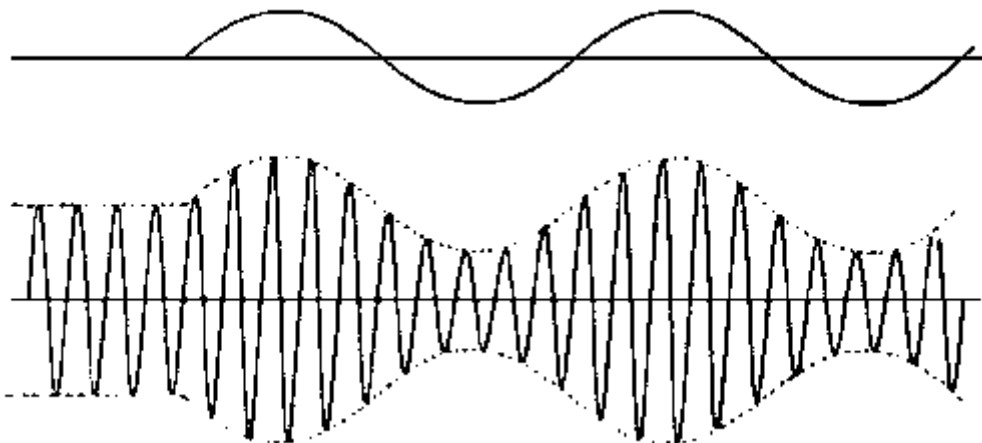4. Demodulation Process

PRE-LAB:

1. Study the theory of Amplitude Modulation and Demodulation
2. Draw the circuit diagram for Modulation and Demodulation
3. Draw the expected wave forms for different values of modulation index

COMPONENTS:

1. Amplitude Modulation and Demodulation Trainer Kit   1 No.
2. CRO Dual Trace   1 No.
3. Connecting Probe and Cords

BRIEF THEORY:

If you connect a long wire to the output terminals of your Hi-Fi amplifier and another long wire to the input of another amplifier, you can transmit music over a short distance. DON'T try this. You could blow up your amplifier.


A radio wave can be transmitted long distances. To get our audio signal to travel long distances we piggyback it onto a radio wave. This process is called MODULATION.

The radio wave is called the CARRIER. The audio signal is called the MODULATION. At the receiving end the audio is recovered by a process called DEMODULATION. From the diagram below, it can be seen that when the carrier is modulated, its amplitude goes above and below its unmodulated amplitude. It is about 50% modulated in the

diagram.
The maximum percentage modulation possible is 100%. Going above this causes distortion.
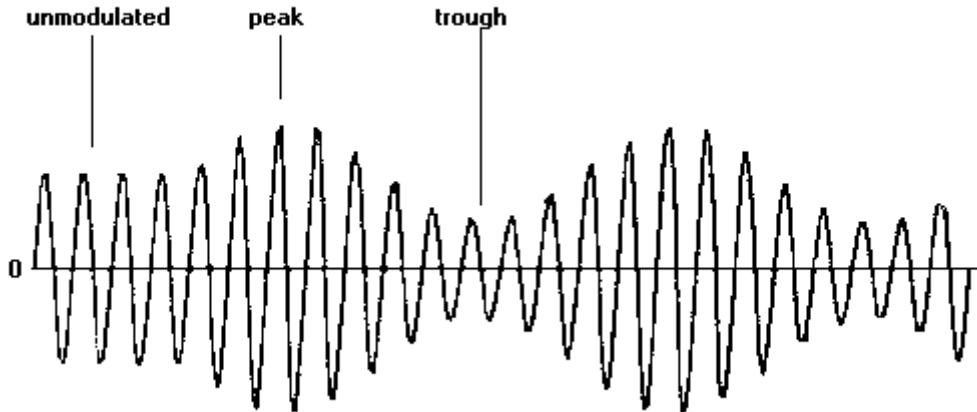
Most broadcasters limit modulation to 80%.Modulating the carrier frequency with an audio frequency produces two new frequencies. At this point it would be a good idea to read the page on MIXERS. These new frequencies are called the upper and lower SIDEBANDS. The upper sideband is the carrier frequency plus the audio frequency. The lower side band is the carrier frequency minus the audio frequency.Since the audio signal is not a single frequency but a range of signals (usually 20 Hz to 20 KHz) the sidebands are each 20Hz to 20 KHz wide.
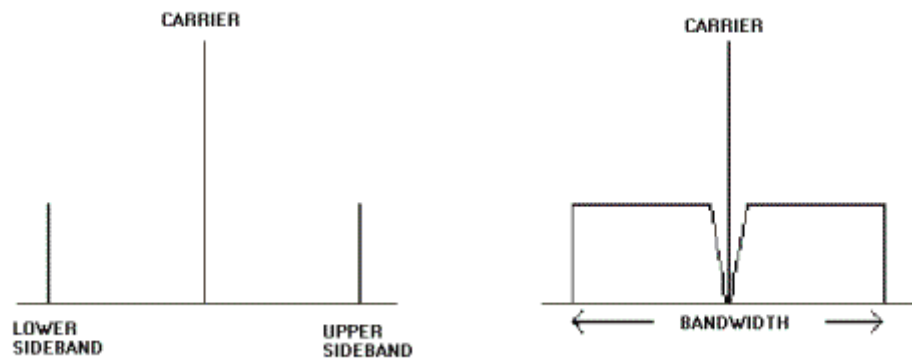

If you tune across a station in the Medium Wave Band you will find that it takes up space in the band. This is called the signal BANDWIDTH. This is the space taken by the upper and lower sidebands. In the the example given above it would be 40 KHz. Since the Medium Wave is only 500 KHZ wide there would only be space for about 12 stations. Therefore the bandwidth of stations is limited to 9 KHz, which limits the audio quality. If there are two stations too close together, their sidebands mix and produce HETERODYNE whistles. Since both sidebands carry
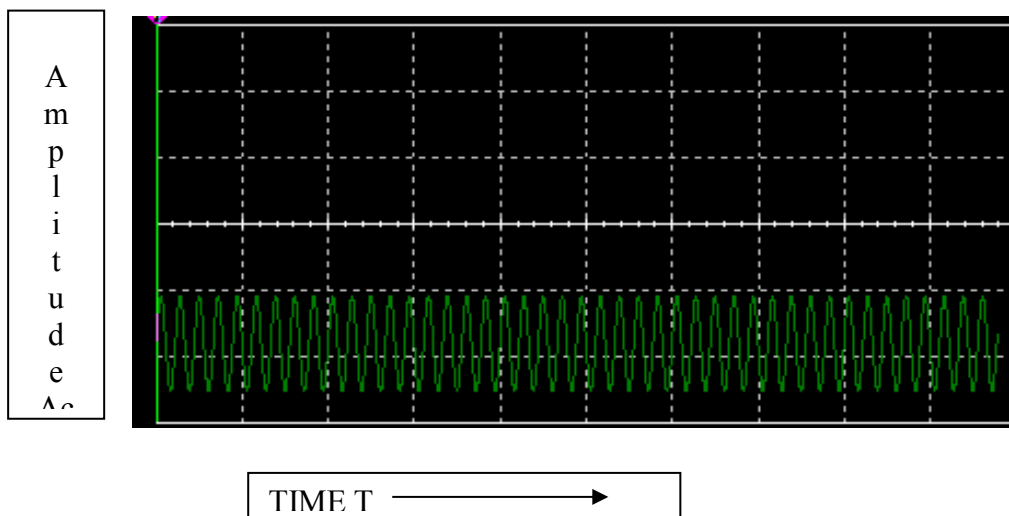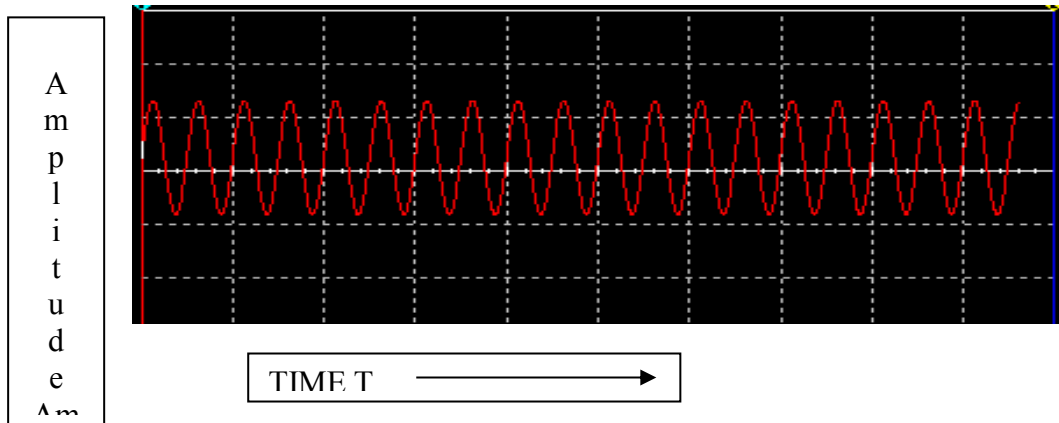
CIRCUIT DIAGRAMS :
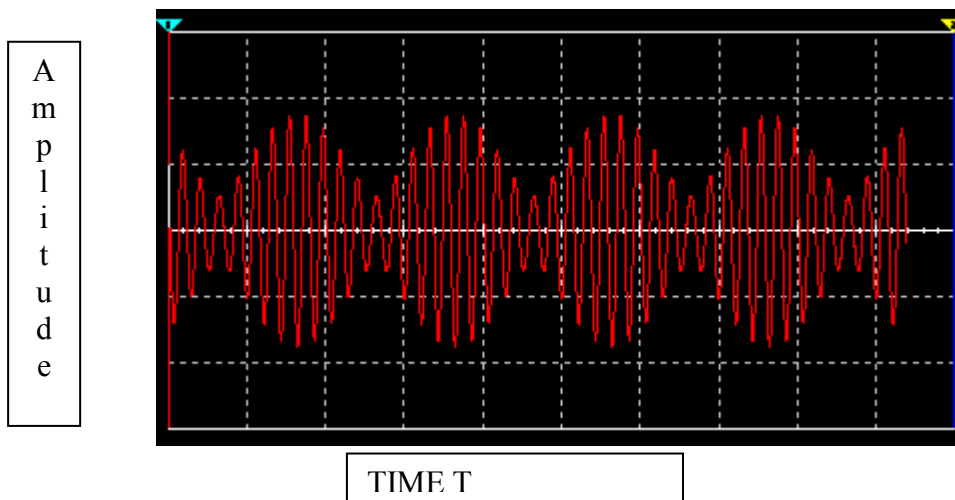
AM MODULATION
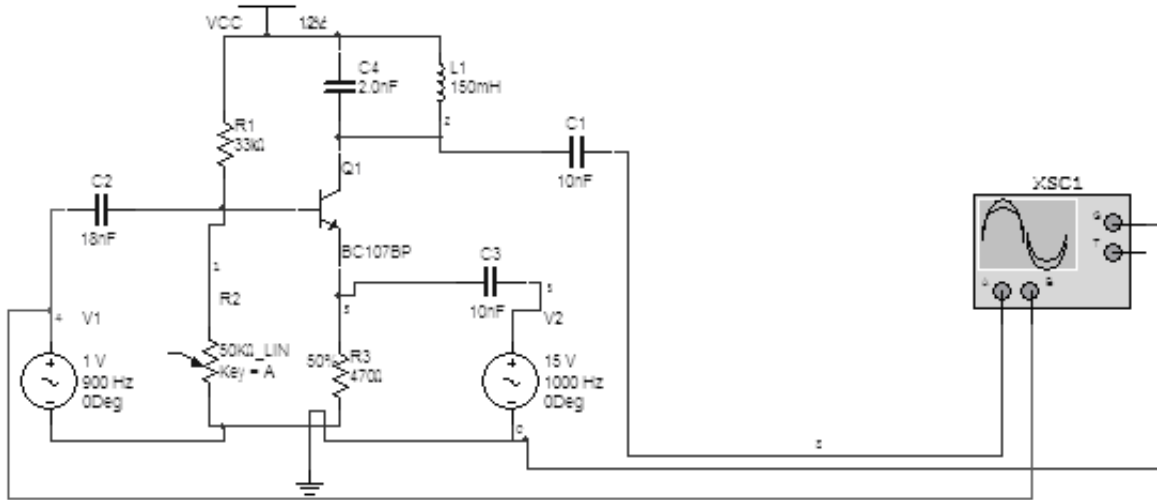
AM DEMODULATION

CARRIER SIGNAL

TIME T ⟶
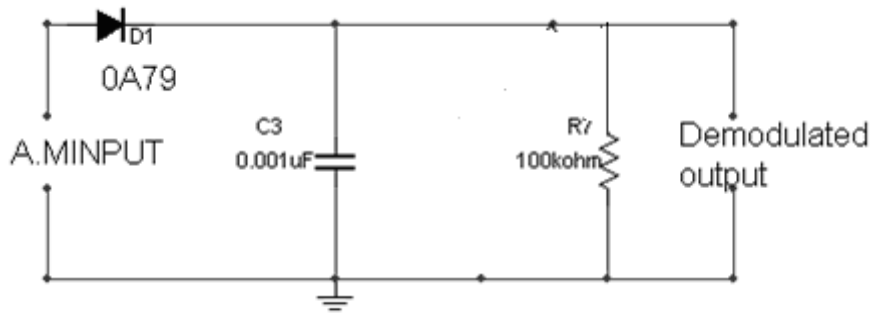
MODULATING SIGNAL

AM MODULATED OUTPUT

AM DEMODULATION OUTPUT

PROCEDURE

A.   MODULATION:

    i.   Switch 'ON' the Trainer Kit.

    ii.   Generate the carrier and modulating signal such that carrier frequency must be greater than modulating frequency.

    iii.   Apply the carrier and modulating signal into AM modulator input,.

    vi.   Trace the envelope of the modulated waveform.

    v.   Repeat step(vi) duly adjusting the modulating signal for under modulation and over modulation

B.   DEMODULATION

    i.   Apply the modulated AM output into Demodulation input.
    ii.   Trace the demodulated output waveform.

    POST- LAB

i.   Observe the frequency and amplitude of the carrier and modulating signal.

ii.   Observe the frequency and maximum peak-peak amplitude(A_{max}) and minimum peak-peak amplitude (A_{min}) of the AM signal in Table:I

iii.   Calculate the modulation index Ч=A_{max}-A_{min}/A_{max}+$_A$min

iv   Plot the out put waveforms Time T V_s Voltage V_o

v.   Write result and conclusion

CONCLUSION: Hence the AM modulation and demodulation is verified.

QUESTIONS: **1.** what is meant by modulation?
                  2. What is meant by modulation index?
                  3. What is meant by under modulation and over modulation?

EXPERIMENT NO: 2

DSB – SC Modulator & Demodulator

OBJECTIVE:

To study the balanced modulator using IC 1496

PRE-LAB:

1. Study the theory of balanced modulator

2. Draw the circuit diagram

3. Draw the expected waveforms

4. Study the datasheet of IC 1496

EQUIPMENT:

1. Function generator

2. Dual trace CRO
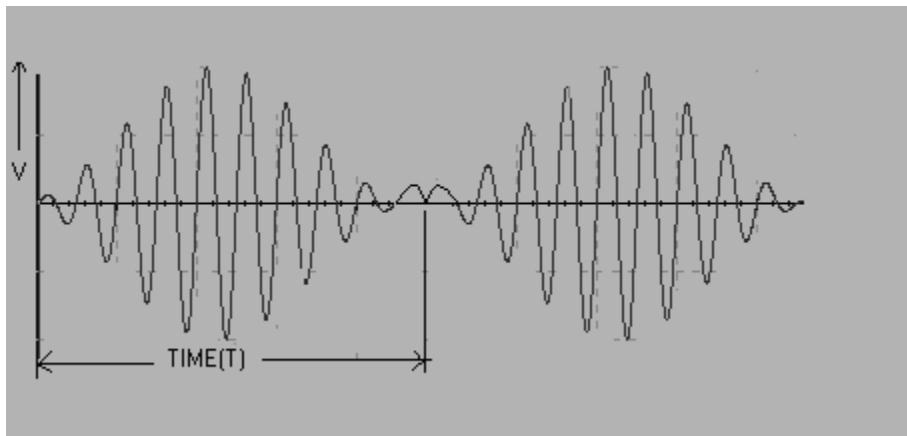
3. Balanced modulator trainer kit

THEORY:

In a balanced modulator, a signal is modulated using two carriers that are 180 degrees out of phase. The resulting signals are then combined in such a way that the carrier components cancel, leaving a DSB-SC (double sideband, suppressed carrier) signal.
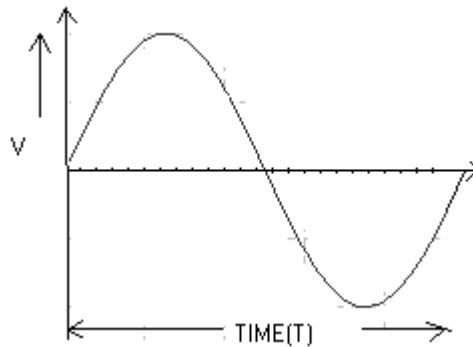
CIRCUIT DIAGRAM:

CARRIER SIGNAL

MODULATING WAVE
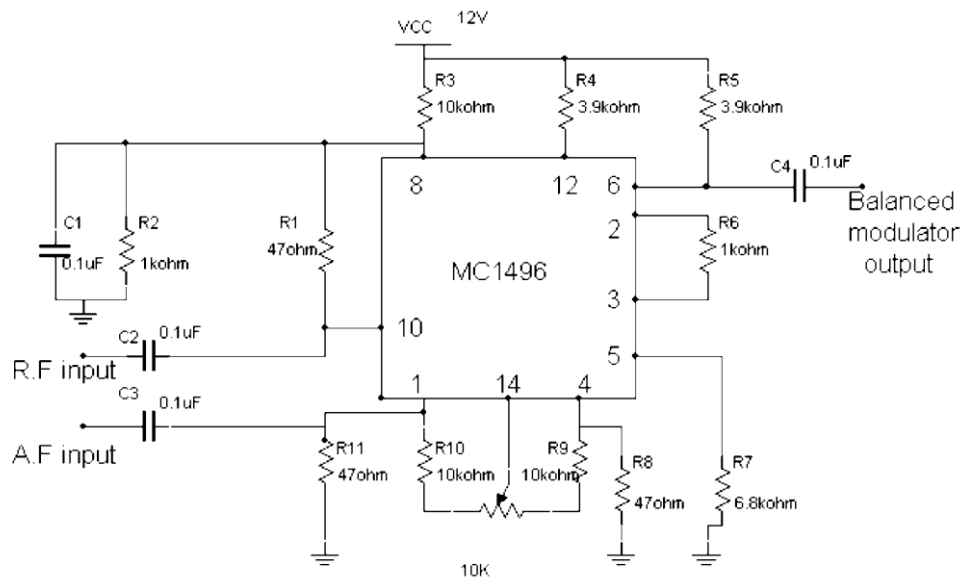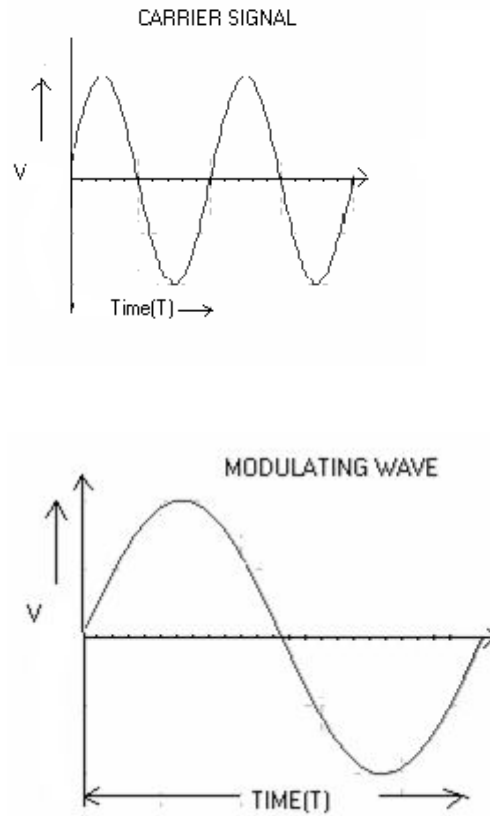
BALANCED MODULATOR OUTPUT

TIME(T)

BALANCED DEMODULATOR OUTPUT

PROCEDURE:

1. Switch ON the trainer kit

2. Generate the carrier and modulating signal

3. Apply the modulating signal and the carrier signal to the balanced modulator   input

4. Observe the balanced modulator output

POST LAB

1 .Observe the frequency and amplitude of the carrier and modulating signal

2. Observe the frequency and amplitude of the balanced modulator output

3. Observe the quadrature null effect

4. Plot the output waveforms Time T V_S Voltage V

5. Write result and conclusions

CONCLUSIONS: The balanced modulator output is verified

QUESTIONS: 1. what is the efficiency for DSB SC signal?
2. What are the other types of balanced modulators?
3. What is meant by pilot carrier?

EXPERIMENT NO: 3

SSB – SC MODULATOR & DEMODULATOR

OBJECTIVE:

       To study the Single side band modulation by using Phase shift method and demodulation using synchronous detector method.

PRE-LAB:

1. Study the theory of SSB
2. Draw the circuit diagram and expected waveforms

EQUIPMENTS:

1. Function generator 2
2. Dual trace CRO 1
3. SSB system modulation and demodulation trainer kit
4. Patch Cards/Connecting wires
5. Frequency counter/Multi meter

THEORY:

    Single-sideband modulation (**SSB**) is a refinement of amplitude modulation that more efficiently uses electrical power and bandwidth. It is closely related to vestigial sideband modulation (VSB).Amplitude modulation produces a modulated output signal that has twice the bandwidth of the original baseband signal. Single-sideband modulation avoids this bandwidth doubling, and the power wasted on a carrier, at the cost of somewhat increased device complexity. SSB was also used over long distance telephone lines, as part of a technique known as frequency-division multiplexing (FDM). FDM was pioneered by telephone companies in the 1930s. This enabled many voice channels to be sent down a single physical circuit, for example in L-carrier. SSB allowed channels to be spaced (usually) just 4,000 Hz apart, while offering a speech bandwidth of nominally 300–3,400 Hz. Amateur began serious experimentation with SSB after World War II. The Strategic Air Command established SSB as the radio standard for its bombers in 1957.[4]It has become a de facto standard for long-distance voice radio transmissions since then
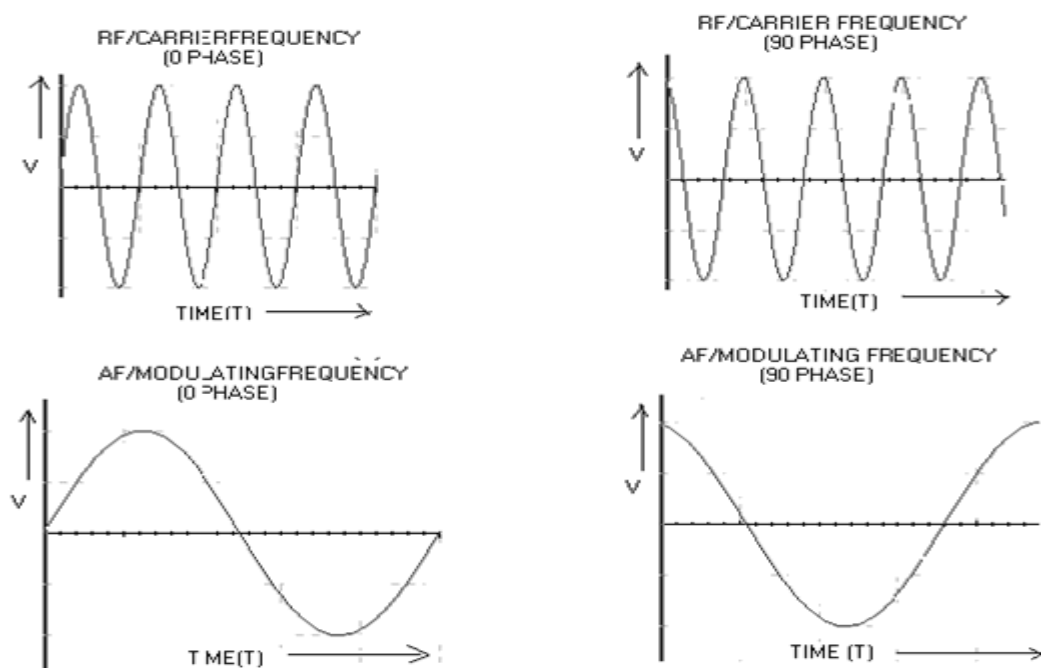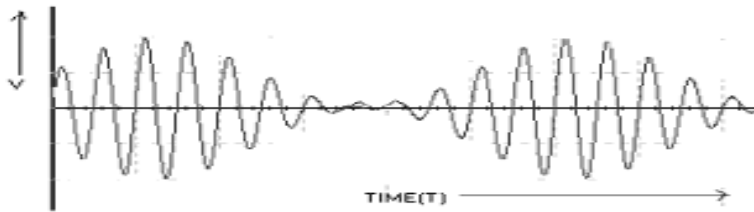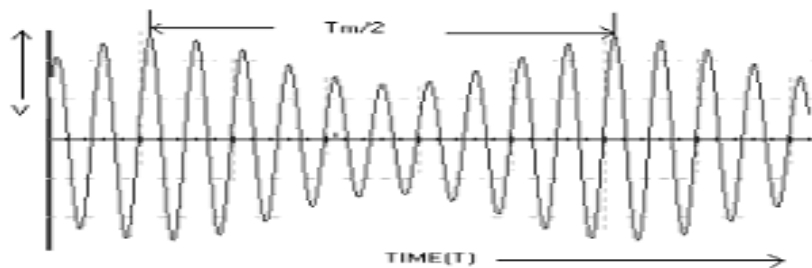
BLOCK DIAGRAM OF SSB MODULATOR:
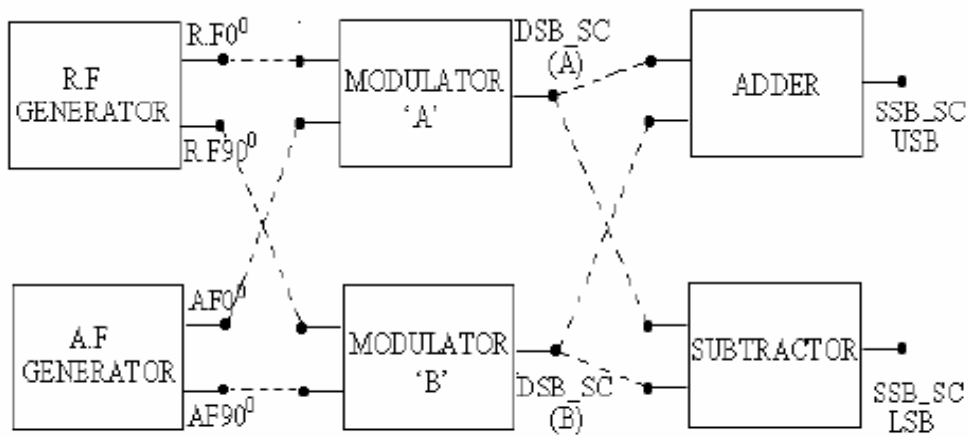
SSB-DEMODULATOR SYSTEM
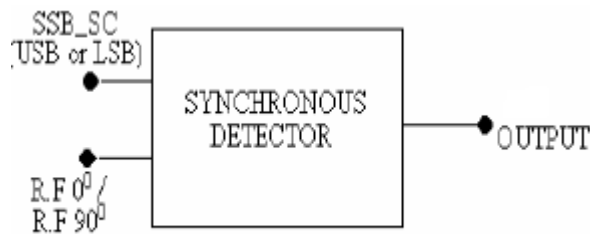
EXPECTED WAVEFORMS

DSB-SC 0UTPUT

SSB-SC OUTPUT

PROCEDURE:

    a.   **Modulation**

1. Switch ON the trainer kit

2. Observe the output of RF generator using CRO.There are two outputs from the RF generator one is direct output and another is 90 deg phase shift with the direct output

3. Observe the output of AF generator using CRO.There are two outputs from the AF generator one is direct output and another is 90 deg phase shift with the direct output

4. Set the amplitude of the RF signal to 0.1vpp and connect the 0 deg phase shift signal to one BM and 90 deg angle phase shift signal to the second BM

5. Set the AF Signal amplitude to 8vpp and connect to the BM

6. Observe the outputs of both balanced modulators simultaneously and adjust the balance control until get the output waveforms (DSB SC).

7. To get LSB of SSB connect both balanced modulators outputs to SUBTRACTED circuit

8. Measure the frequencies of LSB using multimeter.

9. Calculate the theoretical frequency of the SSB (LSB) and compare it with the practical value.

            LSB=RE Frequency –AF Frequency

10. To get USB of SSB signal, connect both balanced modulators outputs to ADDER circuit

11. Measure the frequencies of USB

12. Calculate the theoretical value of the SSB Upper Side Band (SSB-USB) frequency and compare it with practical value

            USB=RF Frequency+AF Frequency

b.   Demodulation (Synchronous detector)

1. Connect the SSB signal from the summer or sub tractor to the SSB signal input of the synchronous detector and RF signal (0º) to the RF input of the synchronous detector.

2. Observe the Detector output using CRO and compare it with the AF signal (Modulating Signal).

3. Observe the SSB signal for the different frequencies of the AF signal (modulating signal)

TABULATION: RF Frequency (Fc) =--------------------KHz.

S.No	AF.sig Freq(Fm)	USB		LSB	
		Practical Freq	Theoretical Freq=Fc+Fm	Practical Freq	Theoretical Freq=Fc-Fm

POST LAB

1. Observe the frequency and amplitude of the RF carrier both (90^0 & 0^0) phase , AF modulating signal (90^0 & 0^0) phase, DSB-SC, SSB-SC, LSB and USB of the SSB signal

2. Observe the frequency components and bandwidth of SSB

3. Plot the output waveforms above mentioned.

4. Write result and conclusion

CONCLUSIONS: Hence various outputs of SSB modulator are obtained.

QUESTIONS:
   1. What are the advantages of SSB over AM?
   2. What is the main drawback of SSB?
   3. What is meant by VSB?

EXPERIMENT NO: 4

FREQUENCY MODULATION AND DEMODULATION

OBJECTIVE:
To study
1. Frequency modulation process
2. Measurement of Modulation Index
3. Measurement of Frequency deviation
4. Demodulation Process

PRE-LAB:
1. Study the data sheet of IC LM 2206, LM 565
2. Study the theory of FM modulation and demodulation
3. Draw the circuit diagram for modulation and demodulation
4. Draw the expected wave forms

EQUIPMENT:
1. Frequency modulation and demodulation trainer kit   1 No.
2. Dual trace CRO   1 No.
3. Connecting probe and cords

THEORY:

    In telecommunications, **frequency modulation** (**FM**) conveys information over a carrier wave by varying its frequency (contrast this with amplitude modulation, in which the amplitude of the carrier is varied while its frequency remains constant). In analog applications, the instantaneous frequency of the carrier is directly proportional to the instantaneous value of the input signal. Digital data can be sent by shifting the carrier's frequency among a set of discrete values, a technique known as keying. The instantaneous frequency of the oscillator and is the frequency deviation, which represents the maximum shift away from f_c in one direction, assuming $x_m(t)$ is limited to the range ±1. Although it may seem that this limits the frequencies in use to $f_c \pm f_\Delta$, this neglects the distinction between instantaneous frequency and spectral frequency. The frequency spectrum of an actual FM signal has components extending out to infinite frequency, although they become negligibly small beyond a point. Carson's rule of thumb, Carson's rule states that nearly all (~98%) of the power of a frequency-modulated signal lies within a bandwidth
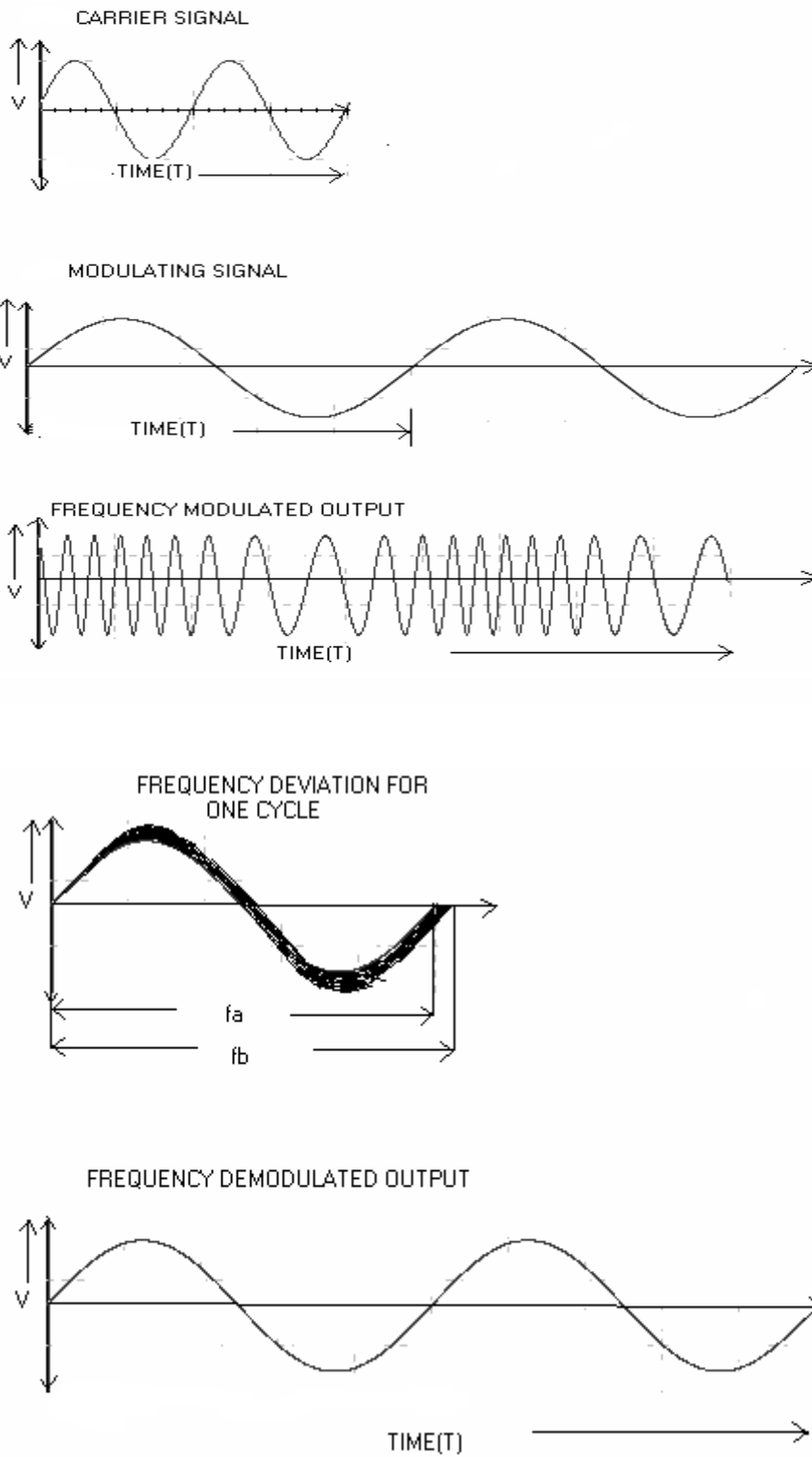
CIRCUIT DIAGRAM :

MODULATION
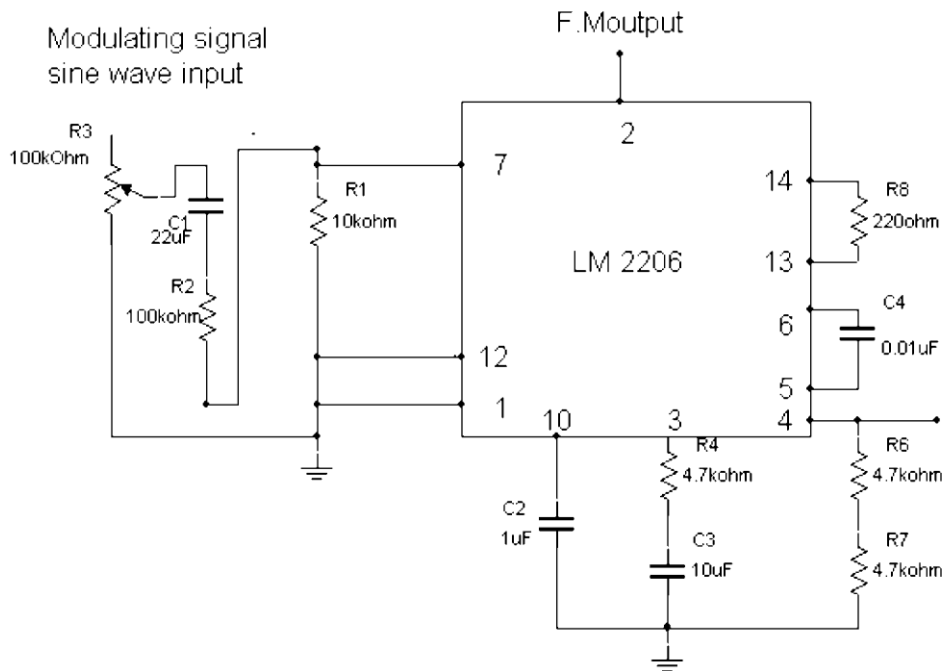
DEMODULATION
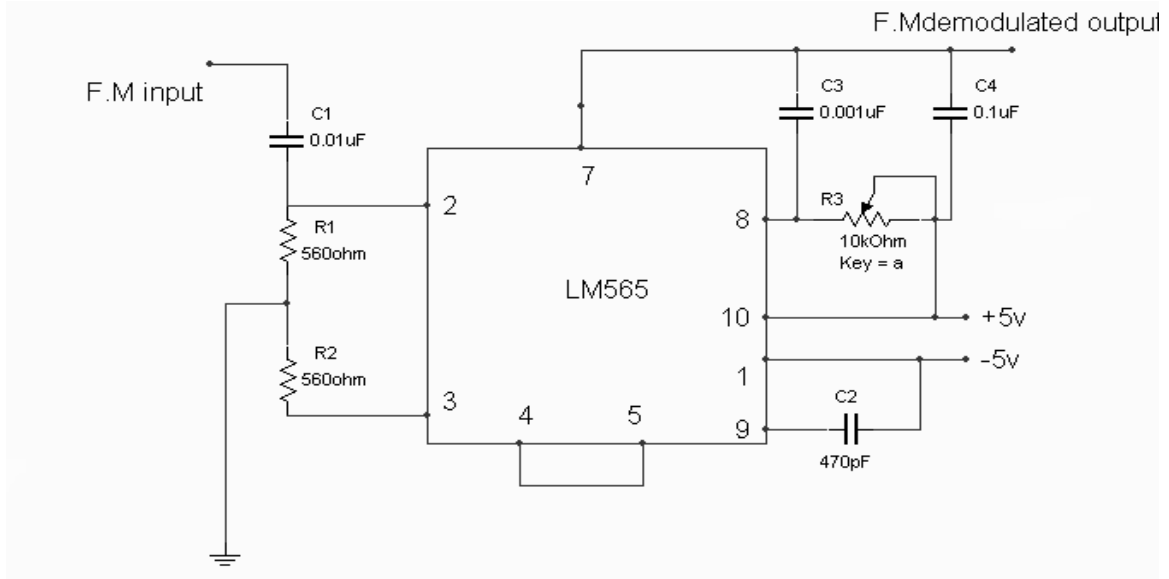
OUT PUT WAVEFORMS

CARRIER SIGNAL

MODULATING SIGNAL

FREQUENCY MODULATED OUTPUT

FREQUENCY DEVIATION FOR
ONE CYCLE

fa

fb

FREQUENCY DEMODULATED OUTPUT

TIME(T)

PROCEDURE:

    a. **Modulation:**

      i.   Switch ON the trainer kit

      ii. Generate the carrier and modulating signal

      iii. apply the modulating signal to the frequency modulator

      iv. Trace the FM output

    b. **Demodulation:**

      i.   apply the modulated FM output into Demodulation input

   ii. Trace the demodulated output waveform

TABULATION

S.No	Amplitude of Modulating Signal(Am)	F_{min}	Frequency Deviation $\Delta f = f_c - f_{min}$ or $f_c - f_{max}$	Modulation Index $\beta = \Delta f / f_m$

POST LAB

1. Observe the frequency and amplitude of the carrier and modulating signal
2. Calculate the frequency deviation by observing F_{max} and F_{min}

    $\delta = F_c - F_{max}$ or $\delta = F_c - F_{min}$

3. Calculate modulation index $\beta = \delta / Fm$
4. Change the amplitude of the modulating signal and repeat steps 1 to 3 for different values of A_m and note down the readings in table-1
5. Plot the output waveforms Time T V_S Voltage V_o
6. Write result and conclusion

CONCLUSIONS: Hence the frequency modulated output is obtained

QUESTIONS: 1. what is meant by angle modulation?
                 2. How FM is different from AM
                 3. What is meant by Carson Rule?
                 4. What is the maximum modulation index?

EXPERIMENT NO : 5

STUDY OF SPECTRUM ANALIZER AND ANALYSIS OF AM & FM SIGNALS

INTRODUCTION
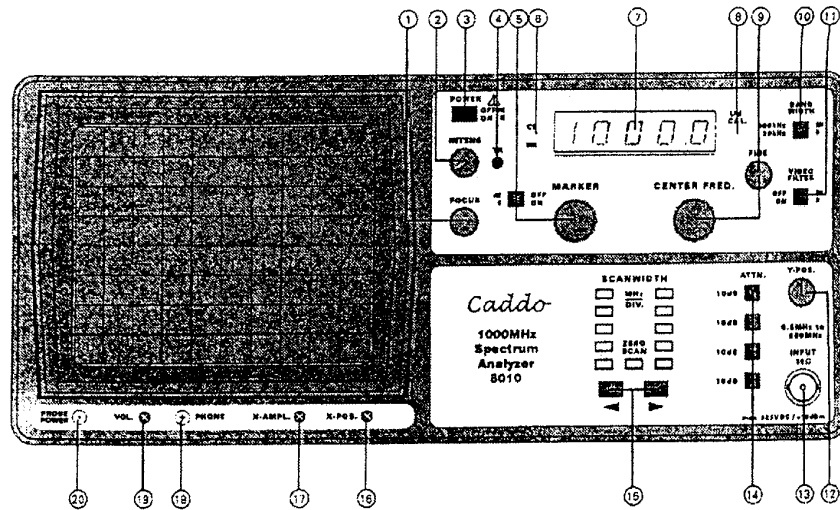
- Frequency Range 0.15MHz - 1050MHz.

- 4½ Digit Display   (Center & Marker Frequency, 0.1MHz resolution)

- -100 to +13dBm Amplitude Range, 20kHz, 400kHz and Video-Filter

- Tracking-Generator (8010TG only):

- Frequency range: 0.15MHz - 1050MHz.

- Output Voltage: + 1dBm to -50dBm (50Ω)

The spectrum analyzer permits the detection of spectrum components of electrical signals in the frequency range of 0.15 to 1050MHz. The detected signal and its content have to be repetitive. In contrast to an oscilloscope operated in Yt mode, where the amplitude is displayed on the time domain, the spectrum analyzer displays amplitude on the frequency domain (Yf). The individual spectrum components of "a signal" become visible on a spectrum analyzer. The oscilloscope would display the same signal *as* one resulting waveform.

The spectrum analyzer works according to the triple superhet receiver principle. The signal to be measured (fin = 0.15MHz to 1050MHz) is applied to the 1st mixer where it is mixed with the signal of a variable voltage controlled oscillator

(f_{LO} 1350MHz - 2350MHz). This oscillator is called the 1st LO (local oscillator). The difference between the oscillator and the input frequency (f_{LO} -f_{in}=1st IF) is the first intermediate frequency, which passes through a waveband filter tuned to a center frequency of 1350MHz. It then enters an amplifier, and this is followed by two additional mixing stages, oscillators and amplifiers. The second IF is 29.875MHz and the third is 2.75MHz. In the third IF stage, the signal can be selectively transferred through a filter with 400kHz or 20kHz bandwidth before arriving at an AM demodulator. The logarithmic output (video signal) is transferred directly, or via a low pass filter to another amplifier. This amplifier output is connected to the Y deflection plates of the CRT.

The X deflection is performed with a ramp generator voltage. This voltage can also be superimposed on a dc voltage which allows for the control of 1 st LO. The spectrum analyzer scans a frequency range depending on the ramp height. This span is determined by the scanwidth setting. In ZERO SCAN mode only the direct voltage controls the 1 st LO.

The 8010TG also includes a tracking generator. This generator provides sine wave voltages within the frequency range of 0.15 to 1050MHz. The tracking generator frequency is determined by the first oscillator (1st LO) of the spectrum analyzer section. Spectrum analyzer and tracking generator are frequency synchronized.

Evolution of the 8005/8005TG has led to the development of 8010/8010TG Spectrum Analyzer/Tracking Generator which now extends operation over 1 GHz (frequency range 0.15 to 1050 MHz). Both fine and coarse center frequency controls, combined with a scanwidth selector provide simple frequency domain measurements from 100 kHz/div. to 100 MHz/Div. Both models include a 4½ digit numeric LED readout that can selectively display either the centre or marker frequency. The 8010TG includes a tracking generator.

These instruments are suitable for pre-compliance testing during development prior to third party testing. A near-field sniffer probe set, can be used to locate cable and PC board emission "hot spots" and evaluate EMC problems at the breadboard and prototype level. The Spectrum Analyzer is an excellent solution for RF leakage/radiation detection, CATV/ MATV system troubleshooting, cellular telephone/ pocket pager test, and EMC diagnostics.

Caddo

1GHz SPECTRUM ANALYSER
Model 8010 / 8010TG

Technical Specifications:

FREQUENCY:

Frequency Range:	150KHz to 1050MHz (-3dB).
Frequency:	
Resolution Displayed:	Resolution 100KHz, (4 ½ digit)
Center Frequency	
Display Accuracy:	±100KHz.
Marker Accuracy:	0.1% Span +100KHz.
Frequency Stability (Drift):	<150KHz / hr.
Frequency Span:	Zero span 0Hz/div, and 100KHz/div to 100MHz/div in 1-2-5 sequence.
Accuracy:	±10%.
RBW (-3dB):	Resolution 20KHz & 250KHz.
Video Bandwidth, VBW:	400KHz.
Sweep Time (Fixed):	23ms.

AMPLITUDE:

Amplitude Range:	-100dBm to +13dBm.
Screen Display Range:	80dB (10dB/div)., 8 div
Reference Level:	-27dBm to +13dBm (in 10dB steps). -27,-17,-7,+3 and +13dBm
Accuracy(Ref.Level):	±2dB.
Average Noise Level:	-103dBm (250KHz RBW).
Frequency Response	
(Relative to 250MHz,	
ATTN 10dB):	±2dB.
Intermodulation	
(3rd order)	
(Spurious Response):	<-75dBc (2 signals, -27dBm each Freq. Distance >3MHz).
Harmonic Distortion	
(2nd, 3rd):	<-75dBc.
Absolute Amplitude Accuracy:±2.5dB.	

INPUT:

Input Impedance:	50 ohms.
Input Connector:	BNC.
Input Attenuator:	0 to 40 dB (4 x 10 dB steps).
Input Attenuator Accuracy:	±2dB / 10dB step.
Max. Input Level:	+10dBm, (0dB attenuation) DC : ±25V

TRACKING GENERATOR:
(Only for 8010TG)

Output Frequency Range:	150KHz to 1050MHz.
Output Power Level:	-50dBm to +1dBm (in 10dB steps & Variable).
Output Attenuator:	0 to 40 dB (4 x 10 dB steps).
Output Attenuator	
Accuracy:	±1dB.
Output Flatness	
(150KHz to 1050MHz):	±1.5dB.
Spurious Outputs	
Harmonic Spurs & Non	
harmonic Spurs(RFI):	<20dBc.
Output Impedance:	50 ohms (BNC female).

MISCELLANEOUS:

AM/Demodulator:	Earphones
Load Impedance:	> 8W.

GENERAL:

Display:	CRT 6 inch, 8 x 10 div internal graticule.
Trace Rotation:	Adjustable on front panel.
Line Voltage:	220V ±10%, 50-60 Hz.
Power Consumption:	Approx. 25W.
Operating Temp.:	+10ºC to +40ºC.
Protective System:	Safety class 1 (IEC 1010-1).
Cabinet:	285 (W) x 125 (H) x 380 (D) mm.
Weight:	Approx. 7 Kg.
ACCESSORIES SUPPLIED:	Instruction Manual, Power Cord.
OPTIONAL ACCESSORIES :	Sniffer Probe: (1. High Impedance Probe 2. Magnetic Field Probe 3. Electric Field Probe)

Features

- Frequency Range : 150KHz to 1050MHz
- 4½ Digit Display : Centre & Marker Frequency, 0.1MHz Resolution
- Amplitude Range : -100dBm to +13dBm
- Filters : 20KHz, 250KHz & Video Filter.
- Built-in Tracking Generator : 0.15MHz to 1050MHz (Only for Model 8010TG)
- Output Power : +1dBm to -50dBm (50 ohms)

(Subject to Change)

ScienTECH®
Technologies Pvt. Ltd.
94-101, Electronic Complex, Pardesipura, Indore- 452 010 India.
Tel.: 91-731-2570301/02, 4211100, Fax : 91-731-2555643
Website : www.caddo.bz, www.scientech.bz,
Email: info@scientech.bz

EXPERIMENT NO: 6
PRE EMPHASIS AND DE EMPHASIS

OBJECTIVE:

To study the operation of Pre-emphasis and De-emphasis circuits

PRE-LAB:

1.   Study the theory of Pre-emphasis and De-emphasis circuits

2.   Draw the circuit diagram

3.   Draw the expected waveforms

COMPONENTS:

1.   Resistors 100kΩ, 100Ω

2.   Capacitors 1μf, 100μf

EQUIPMENT:

1.   Function generator 1
2.   Dual trace CRO 1

THEORY:

Pre-emphasis

Improving the signal to noise ratio by increasing the magnitude of higher frequency signals with respect to lower frequency signals

De-emphasis

" Improving the signal to noise ratio by decreasing the magnitude of higher frequency signals with respect to lower frequency signals"

Transmitters that employ a true FM modulator require a pre emphasis circuit before the modulator fore the true FM modulator doesn't automatically pre emphasize the audio like a transmitter that uses a phase modulator. A separate circuit is not necessary for pre emphasis in a transmitter that has a phase modulator because the phase modulator applies pre emphasis to the transmitted audio as a function of the phase modulator. The receivers De emphasis circuitry takes the unnatural sounding pre emphasized audio and turns it back into its original response. Pre emphasized (discriminator) audio is however available directly from the audio demodulation (discriminator) circuitry. In linking systems, many choose to eliminate the emphasis circuitry to allow better representation of retransmitted

signals. Since the signal has already been pre emphasized (by the user that is transmitting,) and since the receiver you are listening to takes care of the de emphasis.

CIRCUIT DIAGRAMS

PRE-EMPHASIS
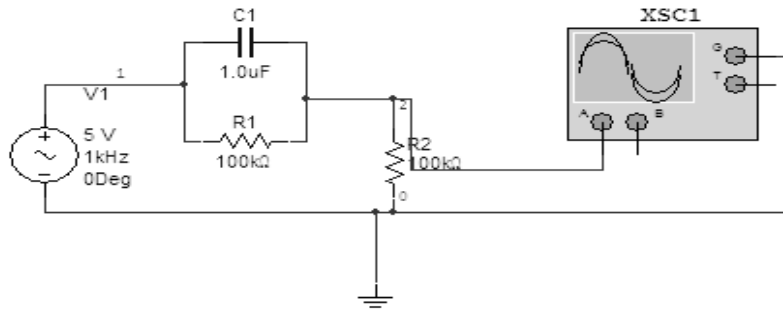
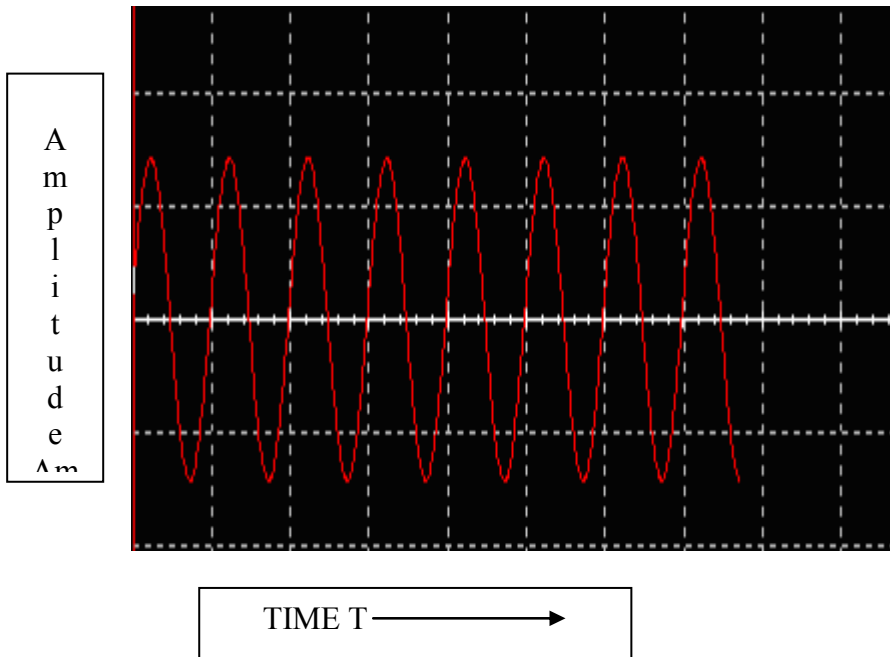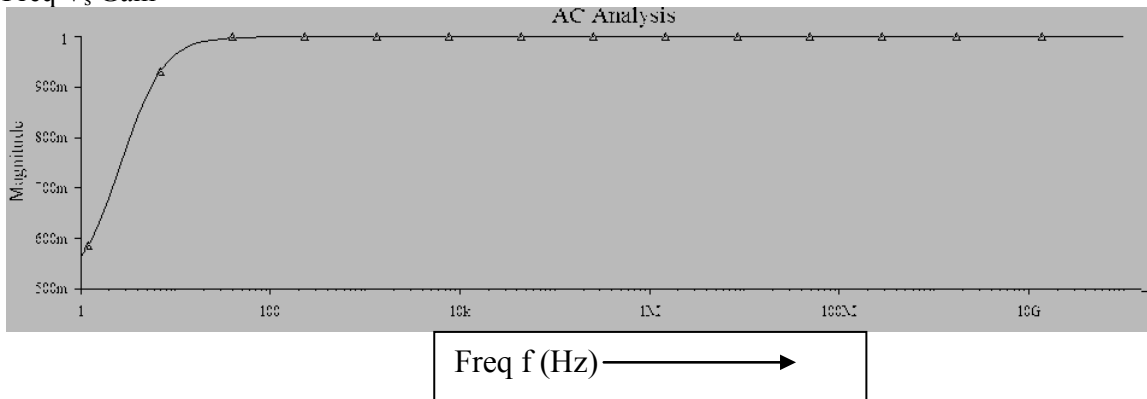**DE-EMPHASIS**
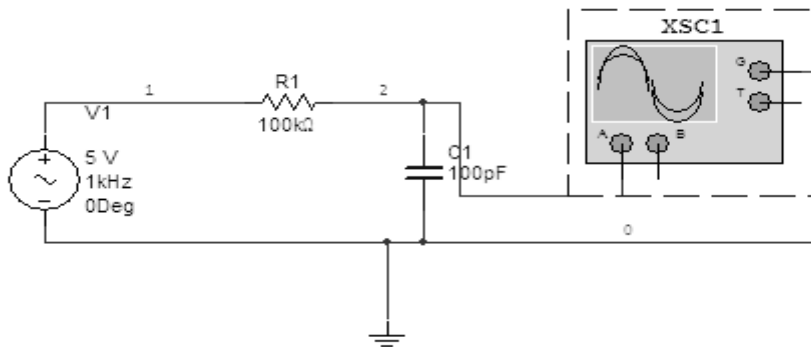
EXPECTED WAVEFORMS:
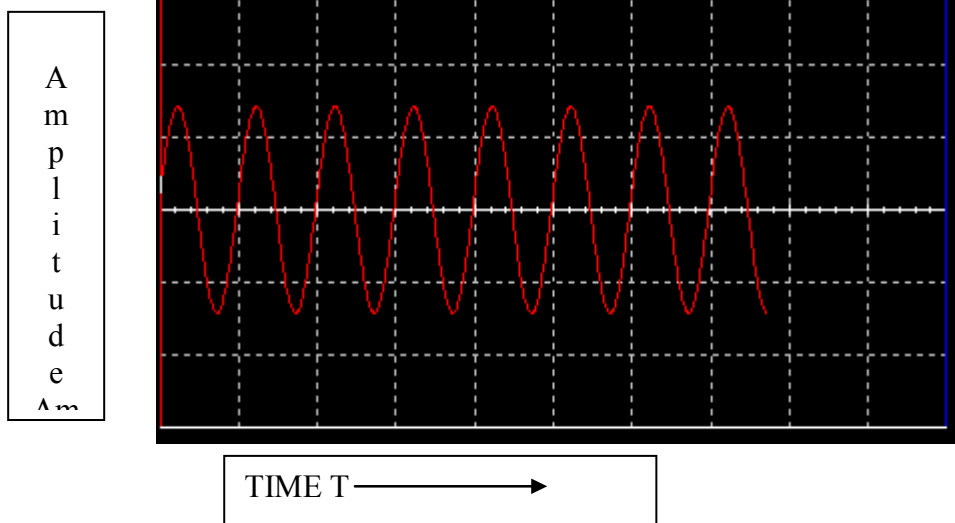
INPUT

OUTPUT

TIME T ⟶

PRE-EMPHASIS

Freq V$_s$ Gain

Freq f (Hz) ⟶

DE-EMPHASIS

Freq V$_s$ Gain
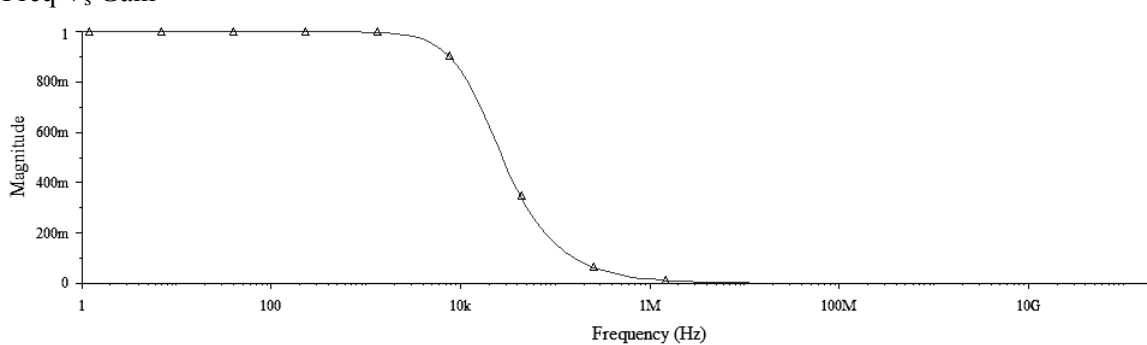
TABLE

PRE EMPHASIS

S.No	Input Freq	Input Voltage V_i (Volts)	Output Voltage V_o (Volts)	Gain=V_o/V_i	Gain in dB 20log(V_o/V_i)

DE-EMPHASIS

S.No.	Input Freq	Input Voltage V_i (Volts)	Output Voltage V_o (Volts)	Gain=V_o/V_i	Gain in dB 20log(V_o/V_i)

PROCEDURE:

    a.   **Pre-emphasis**

    1.   Connect the circuit as per the circuit diagram

    2.   Apply a constant sine wave input across the input terminal of fixed amplitude(V_i)

    3.   By varying the input frequency ,note down the output amplitude (Vo) with respect to input frequency in Table

    4.   Calculate the gain using the formula

    5.   Gain =20log (V_o)/ (V_i) db

    b.   **De-Emphasis**

    1.   Connect the circuit as per the circuit diagram

    2.   Repeat the steps 2, 3 &4

POST LAB

    1.   Observe the output amplitude for both the pre-emphasis and De-emphasis circuits

    2.   Calculate the gain in db

    3.   Plot the frequency versus gain curves on logarithmic graphs for both the circuits

    4.   Plot the input and output waveforms Time T V_S Voltage V_o

    5.   Write the result and conclusion

CONCLUSIONS: Obtained the pre emphasis and de-emphasis outputs

QUESTIONS:
    1. Observe the out put signal is disturbed by noise what action has to be taken and measure the output voltage.
    2. Where we use the pre emphasis circuit.
    3. Where we use the de emphasis circuit.
    4.   What are the advantages of pre emphasis and de emphasis circuits?

EXPERIMENT NO : 7

TIME DIVISION MULTIPLEXING AND DEMULTIPLEXING

AIM: To study Time Division Multiplexing and its waveforms.

PRE LAB WORK :

- ➢ Study the operation of Time division multiplexer and demultiplexer
- ➢ Draw the block diagram of TDM
- ➢ Draw the expected graphs of all necessary waveforms (CH1,CH2,TDM OP and Demultiplexed OP)

EQUIPMENT:

1)   Experimental board on TDM
2)   Dual Trace C.R.O.
3)   Probes

THEORY:

Time Division Multiplexing (TDM) is a technique for transmitting serial messages on a signal transmission channel by dividing the time division into slots. One slot for each message. The concept of TDM comes from sampling theorem which enables use to transmit the information contained in the band limited Signal using sampling of the signal taken uniformly or a rate slightly higher then the Nyquist rate TDM enables the joint utilization at a common transmission channel by a number of independent message sources without mutual interference.

The different I/p message signals, all band limited in WHz by the I/P low pass filters, and sequentially sampled at the transmitters by a rotary switch or a comparator. The switch makes are complete revolution in Ts ≤ ½ W, extracting are sample from each I/P. The commutator O/P is a PAM waveform containing individual message sample.

It there are N number of I/P the pulse to Pulse spacing will be $\underline{Ts} = \underline{1,}$ while the spacing

N   Nfs

b/w successive samples from each I/P is called a frame.

At the receiver a similar rotatory switch, the decommutator separates the samples and distributes them to a bank of low-pass filters which in turn is usually electronic and synchronization signals are provided to keep the distribute in step with the commutant.

EXPECTED GRAPHS:

PROCUDURE:

1. Connect the O/P of the experimental board to one of the channels on C.R.O.
2. Switch ON the experimental board.
3. Observe the O/P waveforms.
4. Vary the values of the resistors R1, R2 and R3 alternative and observe the O/P on C.R.O.

RESULT:

The process of the Time Division Multiplexing and the waveforms has been studied.
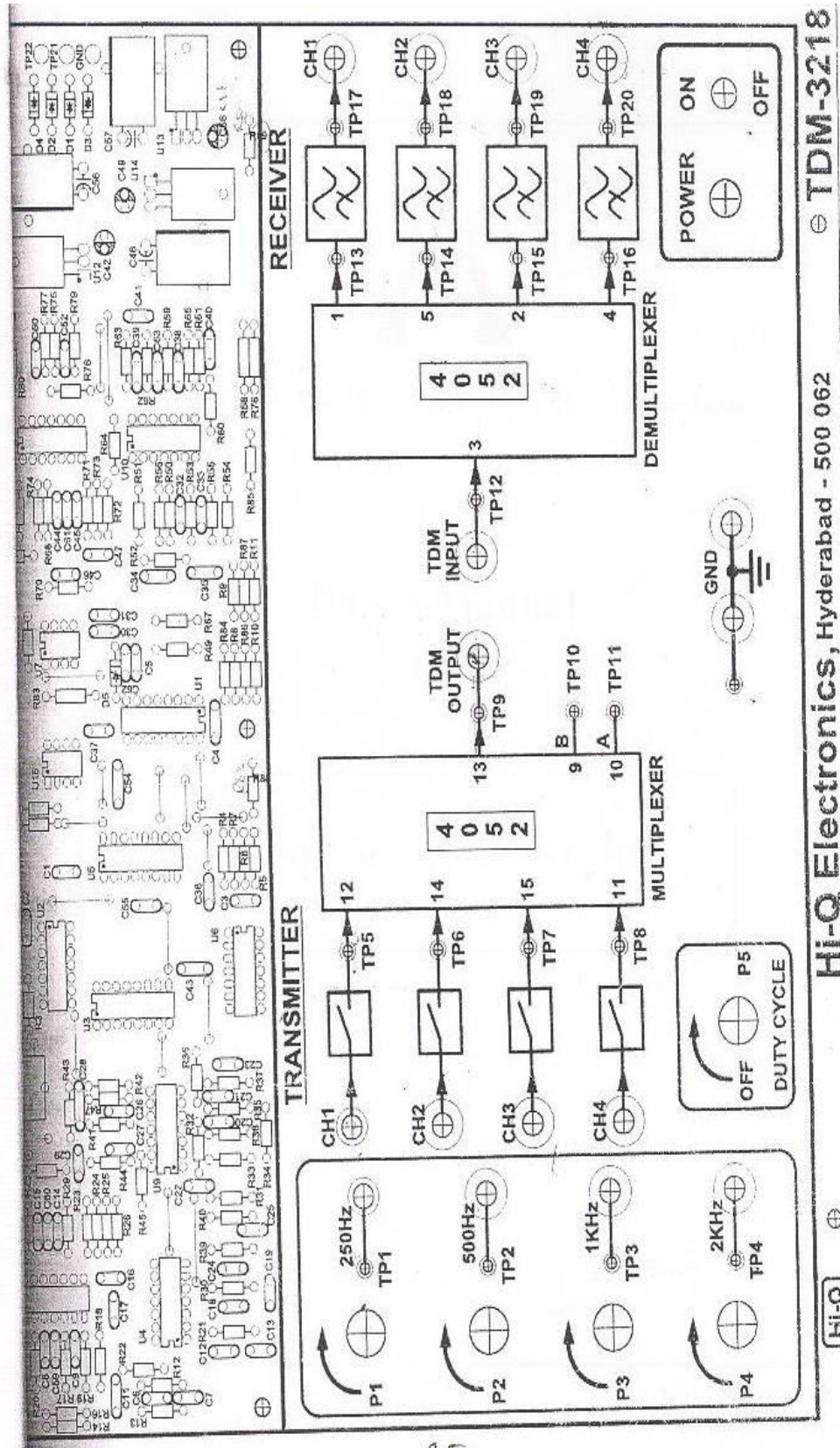
CONCLUSION

Figure 1 Time Division Multiplexing & De multiplexing

EXPERIMENT NO: 9
VERIFICATION OF SAMPLING THEOREM

AIM: To study the effect of sampling on the transmission of information through PWM.

PRE LAB WORK:

- ➢ Study the sampling theorem
- ➢ Draw the expected graphs of all necessary waveforms (message signal, carrier wave, PWM o/p, demodulated o/p etc

EQUIPMENT:

1) Experimental Board on study of sampling theorem.
2) Dual trace C.R.O. (0 – 20 MHz)
3) Function Generator
4) Connecting wires.

THEORY:

The principle of sampling can be explained using the switching sampler.   The switch periodically shifts below b/w two constants at the rate of $fs = 1 / Ts$ Hz staying on the I/P constant for each sampling period.   The o/p $Xs(t)$ of the samples consists of segments of $x(t)$ and $Xc(t)$ can be represented as

$$Xs(t) = X(t)\ S(t)$$

Where $S(t)$ is sampling or switching function.   There are a number of differences b/w the ideal sampling and reconstruction techniques described in the proceeding sections of the actual signal.

PROCEDURE:

1) Observe the internal clock and measure its frequency and amplitude.
2) Give the clock to the clock I/P terminal.
3) Connect the modulating signal to the I/P.
4) Check the condition $fs > 2fm$ and observe the de-modulated wave.
5) Check the condition $fs < 2fm$ and $fs = 2fm$ and draw the O/P waveforms.

RESULT:

CONCLUSION:
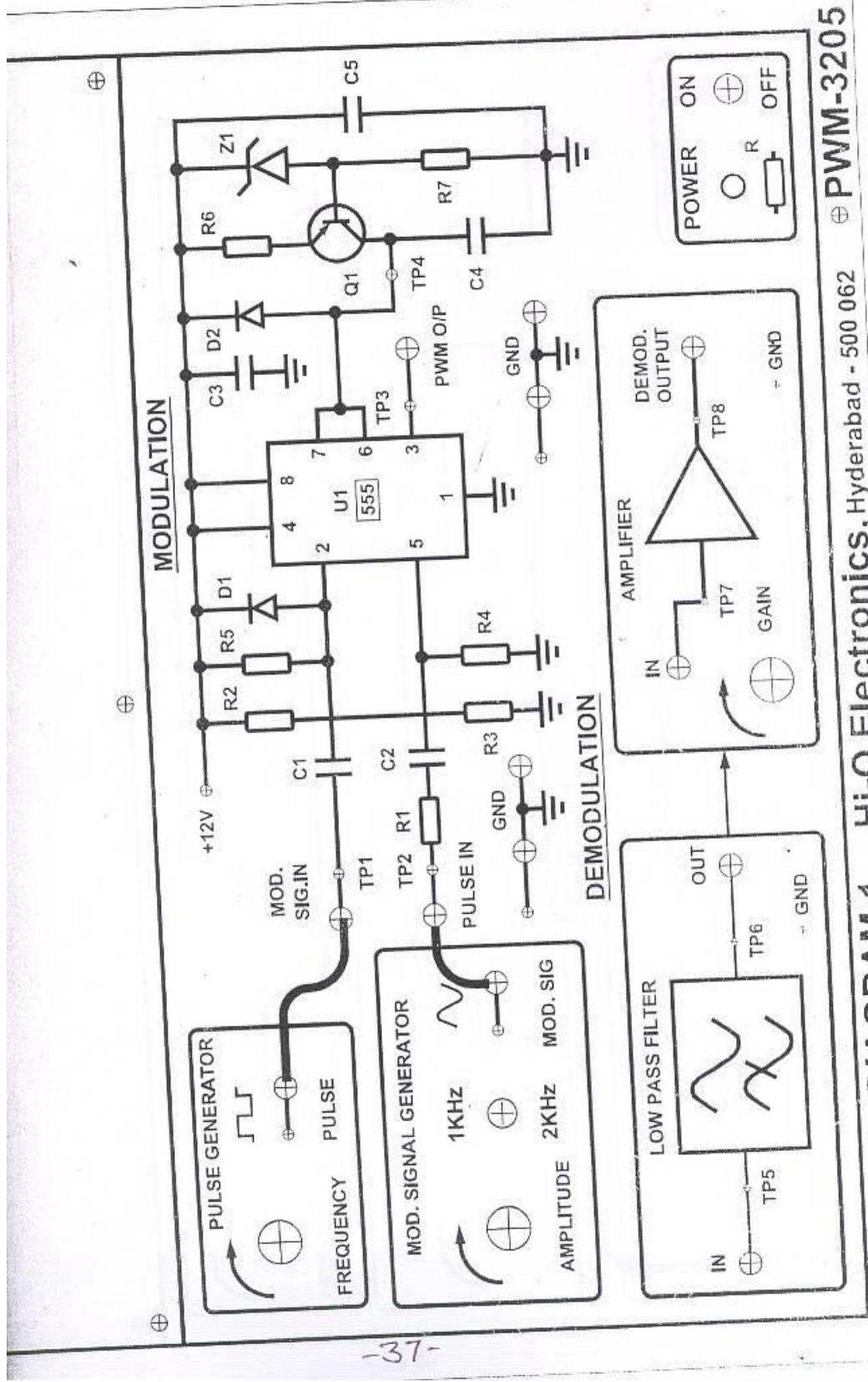
Figure 2 Sampling Theorem

EXPERIMENT NO: 10

PULSE AMPLITUDE MODULATION AND DEMODULATION

AIM:

1) To study the PAM modulation and its waveforms
2) To study the de-modular of PAM
3) To study the effect of sampling frequency on de-modulation O/P.

PRE LAB WORK :

➢ Study the operation Pulse Amplitude Modulator and Demodulator.
➢ Draw the block diagram of PAM modulator and demodulator,
➢ Draw the expected graphs of all necessary waveforms (message signal, carrier wave, PAM o/p, demodulated o/p etc.)

EQUIPMENT:

1) Exp. Board on study of PAM.
2) C.R.O.
3) Functional Generator
4) Connecting wires and chords

THEORY:

In PAM, the amplitude of the carrier wave of higher frequency follow the amp of the message signal. Within this (Modulation Sampling) we have two more types of sampling those are

       (1)   Natural Sampling

       (2)   Flat Top Sampling

Basically, in Natural Sampling, the carrier waves are train of pulses of a certain frequency.   After sampling, the top of the Pulse follow the message signal.   Due to this following of top of Pulses with the message signals cause a distortion in De-modulation.

In Flat top sampling also we use train of pulses for sampling, but only the starting point of each pulse follow the message signal, remaining width of the pulse will (follow) be flat.   But here we go for Natural sampling.

EXPECTED WAVEFORMS:

PROCEDURE: (MODULATION):

1.   Observe the clock on C.R.O. and measure its frequency and amp. (freq $\cong$ 5KHz)
2.   Connect the clock to clock I/P terminals in modulation.
3.   Put control switch in   a.c. position and set amplitude = OV. This provides DC at Modulating signal terminal.
4.   Connect the modulating signal or I/P to the modulator.
5.   Connect one channel of C.R.O. at 'Ti'   and Gnd. Observe that the pulses are of same height. Height is adjustable by varying offset.   This is PAM signal with DC I/P. Measure amplitude and frequency of Pulses.
6.   Increase the amplitude of modulation signal.   Observe the amplitude and frequency of this sinusoidal modulating signal.
7.   Observe PAM signal.

     a)   How many Pulses are there in one modulating cycle?
     b)   It is natural or Flat Top PAM.
     c)   It is Single or dual polarity PAM?

8.   Observe the effect of D.C. offset on PAM waveform.

DE-MODULATION:

         1)   Generate undistorted PAM waveform.
         2)   Connect the modulating signal to de-modulation I/P.
         3)   Observe the detected O/P and measure its frequency which must be same as modulating signal.

EFFECT OF SAMPLING FREQUENCY:

1)   Connect internal sinusoidal signal to modulating signal I/P of the modulator.
2)   Connect an external functional Generator at the circuit terminals of the modulator.
3)   Get the PAM waveform using the above circuit.
4)   Adjust the modulating amplitude offset and frequency to get an undistorted frequency.
5)   Connect the PAM to de-modulator I/P.
6)   Observe the demodulated signal for various frequencies of clock
     (fs > 2fm,   < 2fm, = 2 fm)

RESULT:

CONCLUSION:
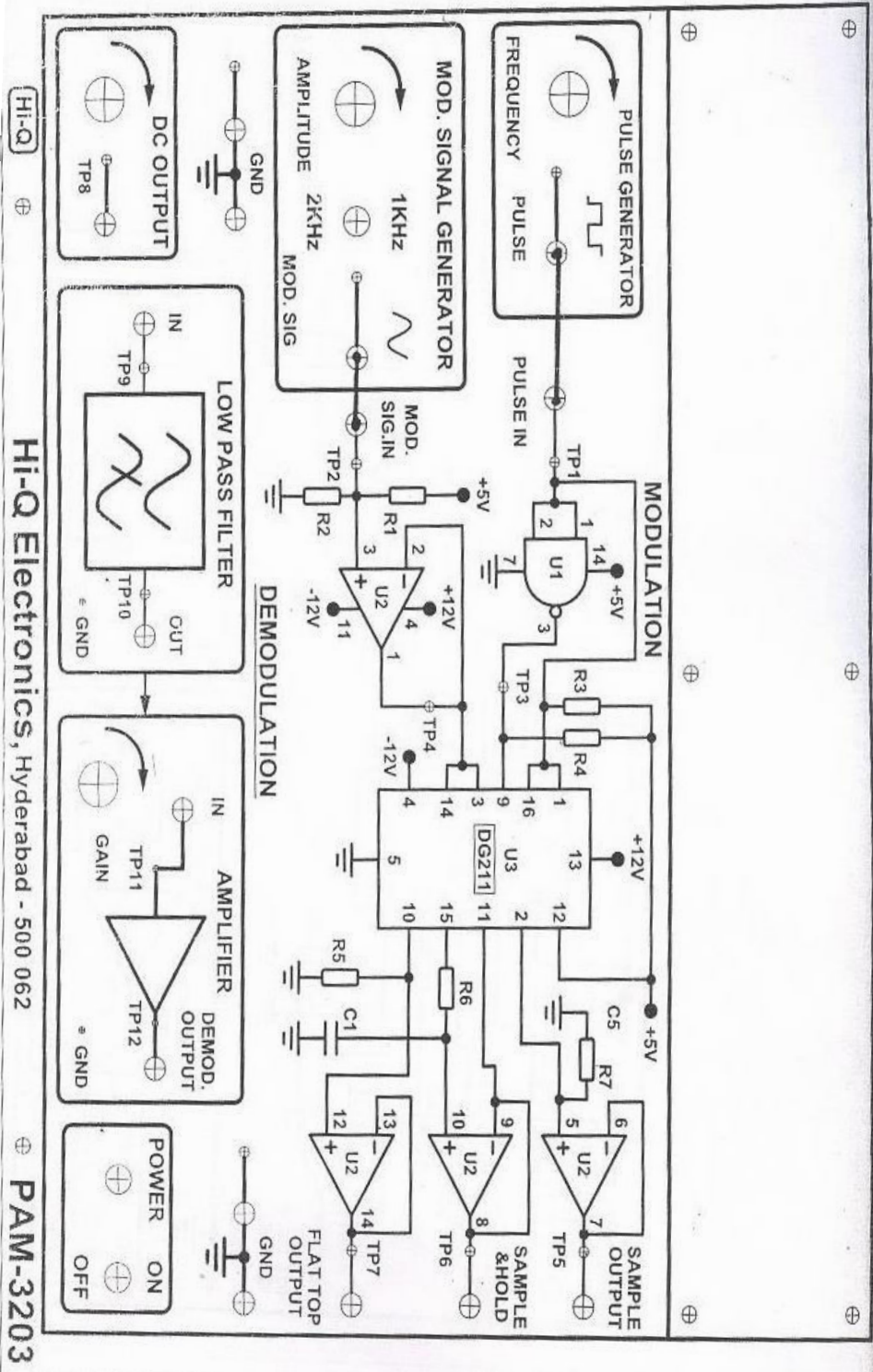
Figure 3 Pulse Amplitude Modulation

EXPERIMENT NO : 11
PULSE WIDTH MODULATION AND DEMODULATION

AIM:

1) To study the PWM process and the corresponding waveforms.
2) To study the effect of sampling frequency on the transmission of information through PWM.

PRE LAB WORK :

➤ Study the operation Pulse Width Modulator and Demodulator.
➤ Draw the block diagram of PWM modulator and demodulator,
➤ Draw the expected graphs of all necessary waveforms (message signal, carrier wave, PWM o/p, demodulated o/p etc

EQUIPMENT:

1) Experimental Board
2) Dual trace C.R.O.
3) AF generator
4) Multimeter

THEORY:

In PWM, the samples of the message signal are used to vary the duration of the individual pulses. The pulse width may be varied by varying the time of occurrence of the leading edge, trailing edge of both edges of the pulse in accordance with the sampled value of the modulating wave PWM can also be generated by using emitter follower monostable multivibrator is an excellent voltage to time converter, since its gate width is dependent on the voltage to which the capacitor is changed. If this voltage can be varied in accordance with a signal voltage, a series of rectangular pulses can be obtained with width varying as required.

The demodulated of PWM is quite simple PWM is passed through a low pass filter. The reconstruction is associated with a certain amount of Distortion caused by the cross modulator products that fall in the signal band.

EXPECTED WAVEFORM:

PROCEDURE:

STUDY OF MODULATION:

1) Observe the clock on C.R.O. and measure its frequency (fs).
2) Observe the sine wave modulating signal on C.R.O. and observe the d.c. shift produced in this by varying the offset nob.   Measure the frequency Variation limits of this signal.
3) Apply clock at the circuit input terminal.
4) Apply modulating signal by keeping the control switch in d.c. position.
5) Observe the modulated wave.
6) Plot the graph b/w pulse width and the applied d.c. voltage.
7) Put the control switch in sine wave position and adjust the modulating signal amplitude, frequency and d.c. offset to get the stationary waveform of PWM.
8) Note the number of pulses in one circuit of modulating signal.

STUDY OF DEMODULATION:

1) Generate the stationary PWM waveform using sin wave modulating signal.
2) Connect the modulator O/P to PWM demodulator.
3) Connect Channel (1) to Modulating Signal and channel (2) to demodulated wave.

EFFECT OF SAMPLING FREQUENCY:

1) Use an external square wave and connect it to the clock input.
2) Set the frequency of square wave at frequency 2fm.
3) Observe the demodulated wave
4) Change the frequency of square wave to fs < 2fm, fs > 2fm and observe the demodulated wave.

RESULT:

CONCLUSION:
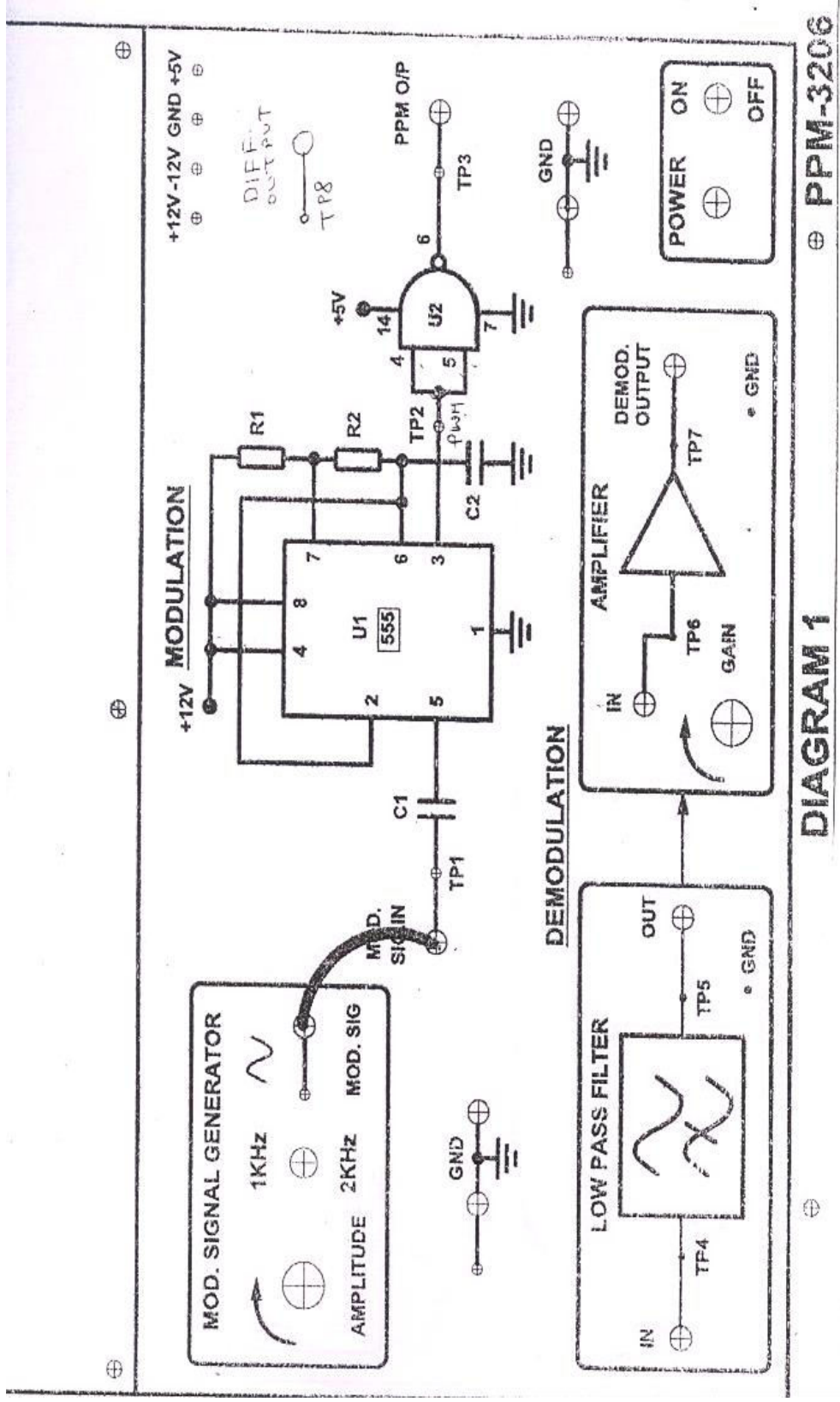
Figure 4 Pulse Width Modulation

EXPERIMENT NO : 12

PULSE POSITION MODULATION AND DEMODULATION

AIM:

1) To study the PPM Modulation and the corresponding waveforms.
2) To study the de-modulation of PPM.
3) To see the effect of sampling frequency on the transmission of information through PPM.

PRE LAB WORK :

➢ Study the operation Pulse Position Modulator and Demodulator.
➢ Draw the block diagram of PPM modulator and demodulator,
➢ Draw the expected graphs of all necessary waveforms (message signal, carrier wave, PPM o/p, demodulated o/p etc

EQUIPMENT:

1) Experimental Board for PPM
2) Dual Phase Oscilloscope
3) AF Generator

THEORY:

In PDM, long pulses expand considerable power duration.   If the pulse which bearing no additional information.   If this un used power is subtracted from PDM, so that only time transitions are preserved. We observe a more efficient type of pulse modulation known as pulse position modulation.   Here in the Pulse Position Modulated wave the position of the Pulse relative to its unmodulated time of occurrence is varied in accordance with the message signal.

We generate a PPM wave from PWM wave by using a Monostable Multivibrator.   The device has one absolutely stable state and one quest – stable state into which it is triggered by an externally applied Pulse. The monostable multivibrator is designed to trigger on the trailing edges of the duration – modulated pulse is varied in accordance with the message signal.

The fixed pulse duration of the PPM wave at the monostable multivibrator O/P can be set by appropriately choosing the resistance – capacitance combination in the timing circuit of the device.

EXPECTED WAVEFORM:

PROCEDURE:

STUDY OF MODULATION:

- Observe the clock and measure its frequency (fs).
- Apply the clock at the clock I/P terminal in the modulator.
- Apply the modulating signal keeping control switch in DC position.
- Observe the PPM O/P.
- Connect the channel – (1) on C.R.O. to clock and channel (2) to PPM wave O/P.
- Compare the position of two waveforms.
- Vary the DC offset and observe the relative shift between the clock and PPM O/P.
- Measure the time shift between clock and PPM wave for various settings of offset.
- Draw the graph between time shift and DC voltage.
- Put the control switch in AC position adjusts the amplitude and frequency and offset of sinusoidal modulating signal to get the stationary PPM wave.
- Note the number of pulses in one cycle of the modulated wave

STUDY OF DEMODULATION:

- Generate a stationery PPM wave for a sinusoidal modulating signal.
- Apply the PPM wave to the input of de-modulator
- Connect the Channel – (1) of C.R.O. to the modulating signal and channel – 2 to the de-modulator output.
- Compare the two waveforms
- Note the frequency of demodulated wave.
- It should be same as the modulating signal.
- Note down the waveform.

RESULT:

CONCLUSION:

MODULATION
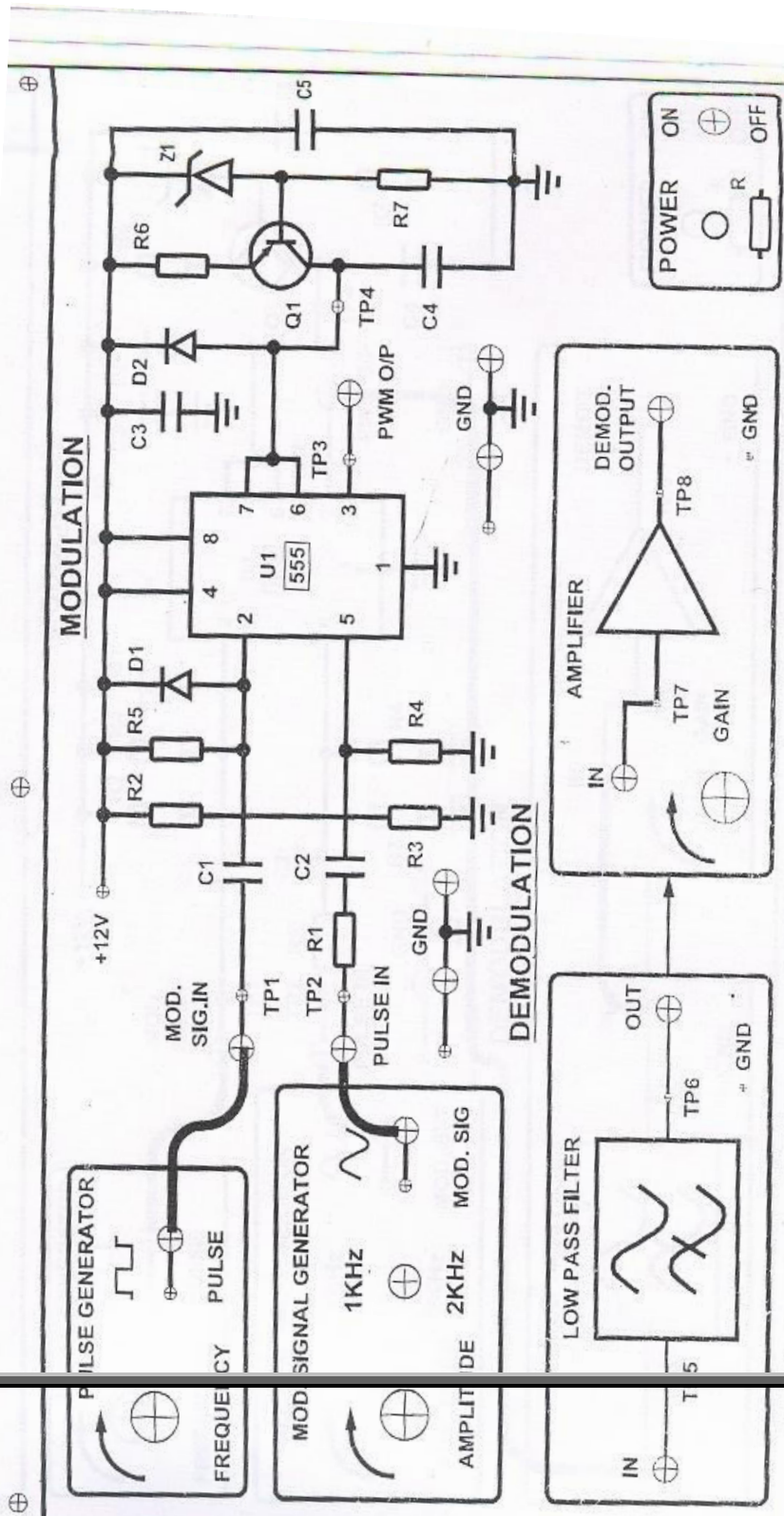
DEMODULATION

DIAGRAM 1 Hi-Q Electronics, Hyderabad - 500 062 ⊕ PWM-3205

Figure 5 Pulse Position Modulation

EXPERIMENT NO: 13

FREQUENCY SYNTHESIZER

AIM

To study the operation of frequency synthesizer using PLL

PRE-LAB:

1. Study the data sheet of IC 7404,IC 4017,IC 565 and IC 4046
2. Study the theory of frequency synthesizer
3. Draw the circuit diagram and expected waveforms

EQUIPMENTS

1. Frequency synthesizer trainer   AET -26A
2. Dual trace C.R.O (20Mhz)
3. Digital frequency counter or multimeter
4. Patch chords

THEORY:

A **frequency synthesizer** is an electronic system for generating any of a range of frequencies from a single fixed time base or oscillator. They are found in many modern devices, including radio receivers, mobile telephones, radiotelephones, walkie-talkies, CB radios, satellite receivers, GPS systems, etc. A frequency synthesizer can combine frequency multiplication, frequency division, and frequency mixing (the frequency mixing process generates sum and difference frequencies) operations to produce the desired output signal.

BLOCK DIAGRAM:
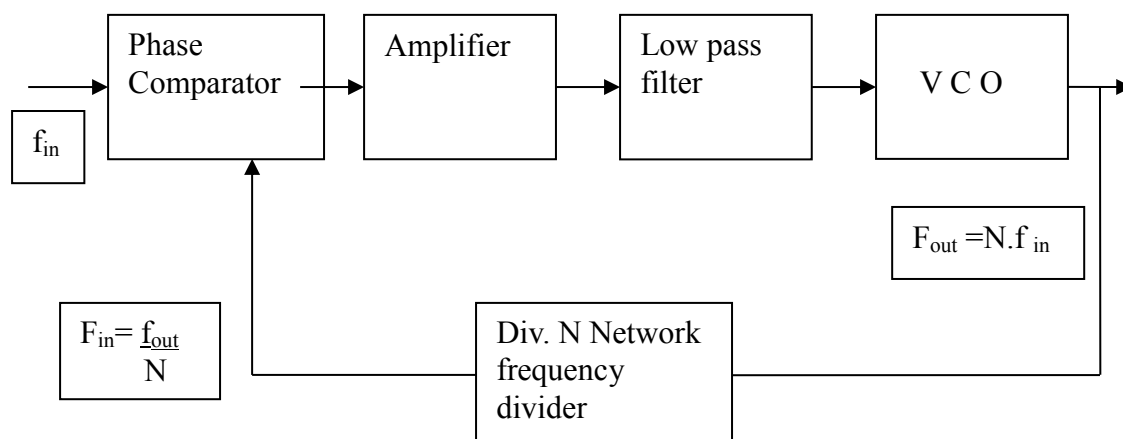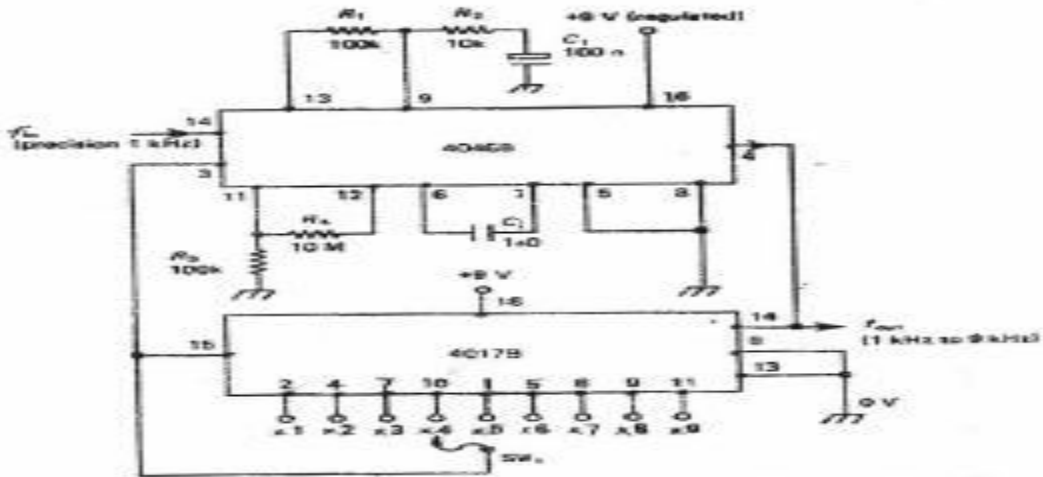
SIMPLE 1KHZ-9KHZ FREQUENCY SYNTHESIZER

CIRCUIT DAIGRAM
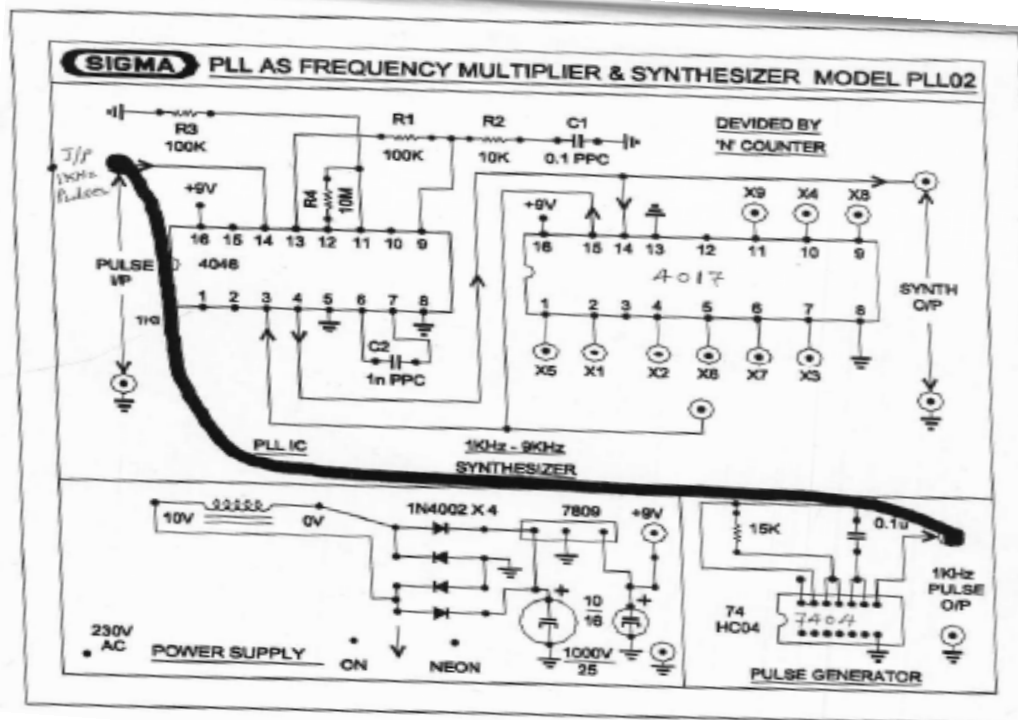
PROCEDURE:

1. Switch on the trainer ad verify the output of the regulated power supply i.e. $\pm$ 5V. These supplied are internally connected to the circuit so no extra connections are required.

2. Observe output of the square wave generator using oscilloscope and measure the range with the help of frequency counter, frequency range should be around 1KHz to 10KHz.

3. Calculate the free running frequency range of the circuit (VCO output between 4th pin and ground 1. For different values of timing resistor R_1 (to measure R_t value using digital multimeter between given test points 1. And record the frequency values in table 1

    F_{out} = 0.3 / ($R_t C_t$) where R_t is the timing resistor and Ct is the timing capacitor = 0.01 μ f.

4.   Connect 4th pin of LM 565 (F_{out}) to the driver stage and 5th pin (Phase comparator) connected to 11th pin of 7404. Output can be taken at the 11th pin of the 7490. It should be divided by the 2 & 3 times of the f_{out}

TABULATION:

Fin KHz	Fout = N x fin KHz	Divided by 3, 2

EXPECTED WAVEFORMS:

Γ01.: Input Frequency (1KHz)

A
m
p
l
i
T
u
d
e
Ac

Γ02.: Output Frequency X 2 = (2KHz)

Γ03.: Output Frequency X 3 = (3KHz)

TIME T
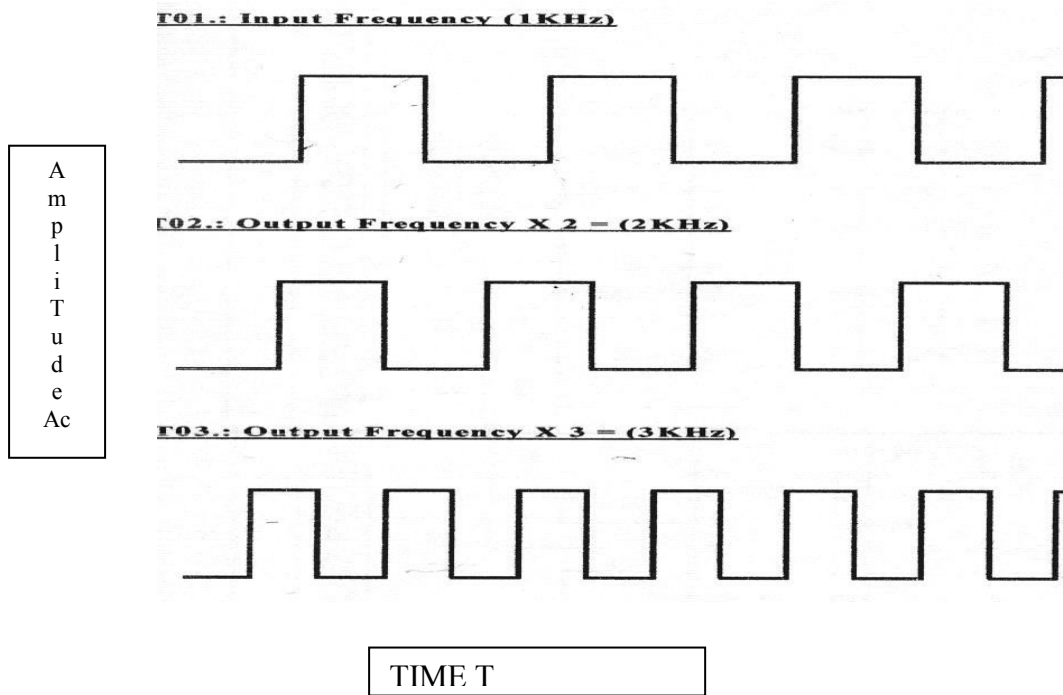
POST LAB:

1.   Observe the synthesized output in table
2.   Calculate the output frequency
3.   Plot the output waveforms
4.   Write result and conclusions

CONCLUSIONS:

The various outputs of frequency synthesizer are verified.

QUESTIONS:

1.   What are the applications of frequency synthesizer?

2. What is the difference between heterodyning and synthesizer?

EXPERIMENT NO: 14

AGC CHARATERISTICS

AIM

To observe the effect of automatic gain control(AGC)

PRE LAB:

1. To study the theory of AGC.
2. Draw the Block Diagram, circuits diagram and expected waveforms

EQUIPMENTS:

1. AGC Trainer Kit
2. CRO
3. Patch Chords
4. Digital Multimeter

THEORY:

Automatic gain control (AGC) is an adaptive system found in many electronic devices. The average output signal level is fed back to adjust the gain to an appropriate level for a range of input signal levels. For example, without AGC the sound emitted from an AM radio receiver would vary to an extreme extent from a weak to a strong signal; the AGC effectively reduces the volume if the signal is strong and raises it when it is weaker. AGC algorithms often use a PID controller where the P term is driven by the error between expected and actual output amplitude.

Block Diagram :
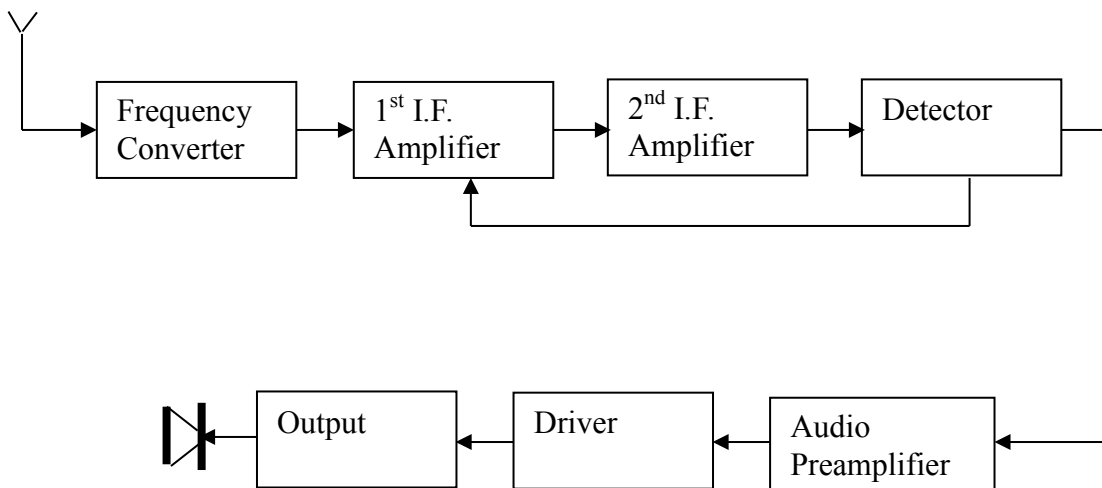
CIRCUIT DIAGRAM
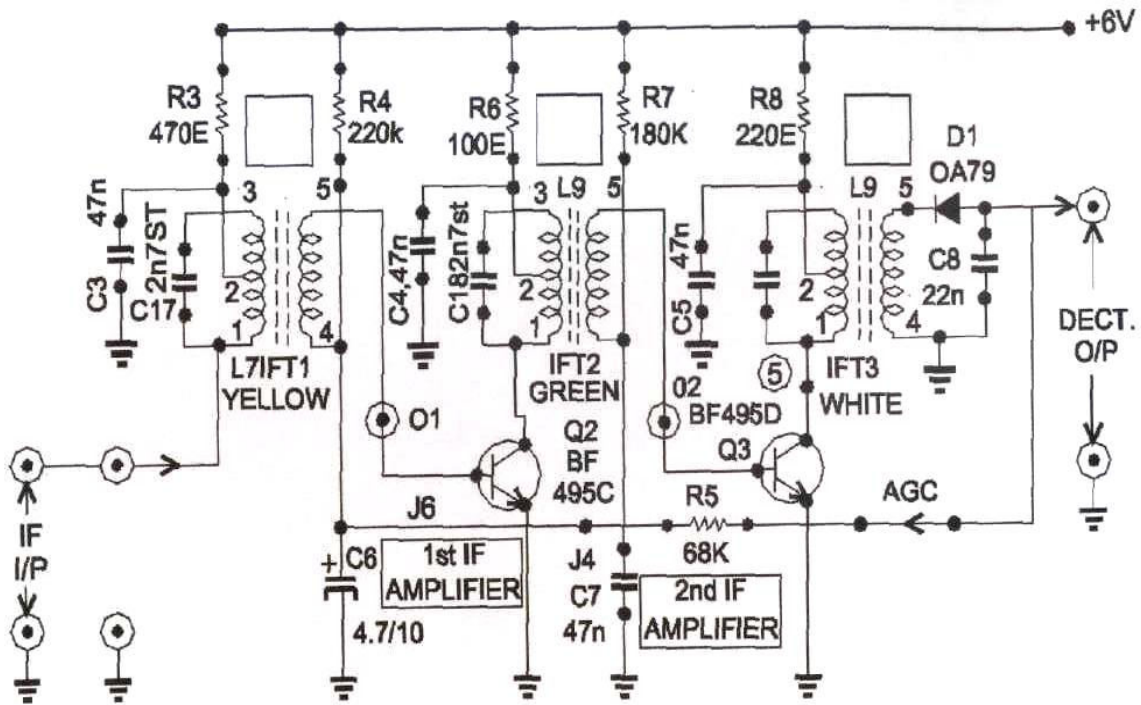
PROCEDURE:

1. Connect 455 KHz IF Signal Function Generator connect AGC link. Connect CRO channel-1 at input terminals.

2. Connect CRO channel 2 at collector of Q3. The amplified signal will be observed.

3. Calculate the voltage gain by measuring the amplitude of output signal (V_o) waveform, using formula $A=V_o/V_i$.

4. Now connect CRO-2 channel at output. The detected audio signal of 455 Khz will be observed.

5. Now vary input level of 455 KHz IF signal and observe detected 455 KHz audio signal with and without AGC link. The output will be distorted when AGC link is removed i.e., there is no AGC action.

6. This explains AGC effect in Radio circuit.

TABULATION:

S.No.	Input Voltage V_i (P-P)	With out AGC		With AGC	
		Output Voltage V_o (P-P)	Gain $A= V_o/V_i$	Output Voltage V'_o (P-P)	Gain $A'= V'_o/V_i$

EXPECTED WAVEFORMS:

AM Modulated RF Input

Amplitude Ac

+0.2V

+0.2V
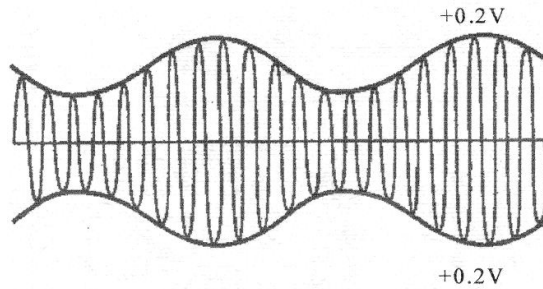
TIME T

Amplitude Ac

Detected Output with AGC
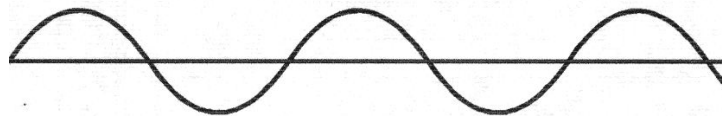
TIME T

Amplitude Ac

Detected Output without AGC
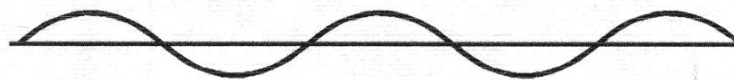
TIME T

POST-LAB:

1. Observe the input voltage and output voltage for with AGC and Without AGC in Table 1
2. Calculate Gain $a = v_o/v_i$ for both cases.
3. Plot the output waveforms Time T Vs Voltage V_o and input voltage V_i Vs Gain A.
4. Write Result and Conclusions

CONCLUSIONS: The AGC characteristics are verified.

QUESTIONS:
1. what is meant by AGC?
2. What is the difference between simple AGC and delayed AGC?
3. From which part of receiver AGC is obtained?

EXPERIMENT NO: 15

PHASE LOCKED LOOP

OBJECTIVE:

To study the Phase locked loop using 565 IC and to take input frequency given by an external signal source

PRE-LAB:

To Study

1. The data sheet of IC 565
2. The theory of Phase locked loop
3. About free running frequency locking range and capture range
4. Draw the circuit diagram and expected waveforms

EQUIPMENT:

1. Function generator 2
2. Dual trace CRO 1
3. Phase locked loop trainer kit

THEORY:

A **phase-locked loop** or **phase lock loop** (PLL) is a control system that generates a signal that has a fixed relation to the phase of a "reference" signal. A phase-locked loop circuit responds to both the frequency and the phase of the input signals, automatically raising or lowering the frequency of a controlled oscillator until it is matched to the reference in both frequency and phase. A phase-locked loop is an example of a control system using negative feedback Phase-locked loops are widely used in radio, telecommunications, computers and other electronic applications. They may generate stable frequencies, recover a signal from a noisy communication channel, or distribute clock timing pulses in digital logic designs such as microprocessors. Since a single integrated circuit can provide a complete phase-locked-loop building block, the technique is widely used in modern electronic devices, with output frequencies from a fraction of a cycle per second up to many gigahertz.

BLOCK DIAGRAM
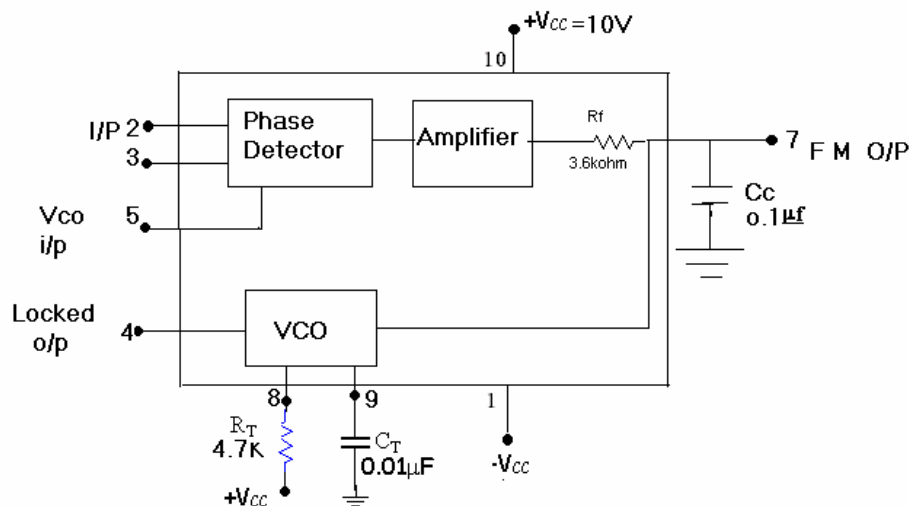
CIRCUIT DIAGRAM:
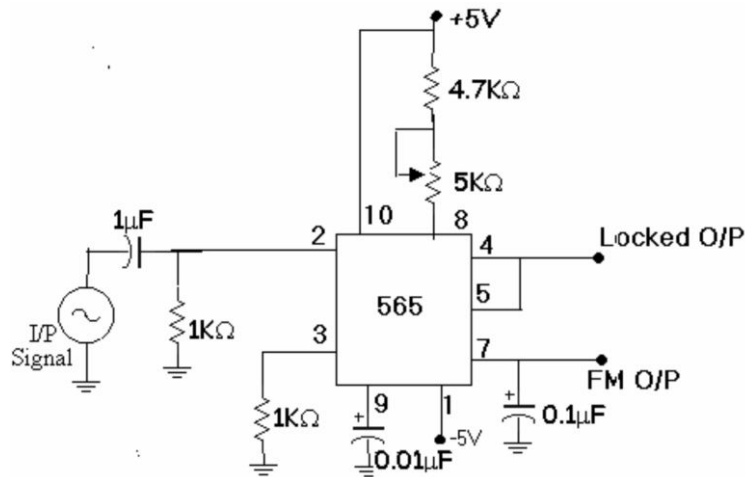
PROCEDURE:

1. Switch ON the trainer kit
2. Check the VCO output at pin 4 of IC 565 that is a square wave form. The frequency of the VCO Output depends on C_T (0.01µf) and R_T (4.7kΩ).
3. Next short the pins 4&5 and give signal of variable frequency and observe VCO output
4. Change the input frequency and observe the VCO output on the CRO.
5. Between some frequencies the VCO output is locked to the input signal frequency This can be observed by increasing or decreasing the frequency of the VCO output by changing input frequency
6. Before or after that frequency range the VCO output is not locked
7. Calculate Lock range (F_L) and capture range (F_C) by using the following formulas
   $F_L = 8fo/VCO$, $f_o = 0.3/R_{tCT}$
   $F_C = 1/2\pi \sqrt{2\pi f2/3.6 * 10^3 C_2}$

Table1:

S.No	Locking Frequency		Frequency Range
	From	To	

CHARTERISTICES
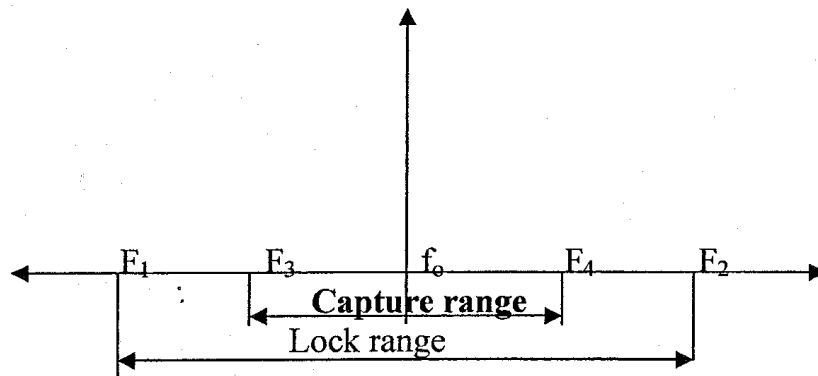
CONCLUSIONS: Obtained the lock range of PLL

QUESTIONS:

      1. Observe the VCO output when pins 4&5 short and give square wave as signal
      2. Define lock range and capture range
      3. What are the basic applications of PLL?

ANNEXURE-I

EXPERIMENT NO: 1

CHARACTERISTICS OF MIXER

AIM

To study the functioning of a frequency mixer.

PRE LAB:
1.   To study the theory of Mixer
2.   Draw the circuit diagram and expected waveforms

EQUIPMENTS:
1.   Frequency mixer trainer kit.
2.   C.R.O (20MHz)
3.   Connecting cords and probes.
4.   Function generator (1MHz).

THEORY:

In receivers using the super heterodyne principle, a signal at variable frequency is converted to a fixed lower frequency, IF, before detection. IF is called the intermediate frequency In typical AM (Amplitude Modulation, e.g. as used on medium wave) home receivers, that frequency is usually 455 kHz; for FM VHF receivers, it is usually 10.7 MHz. An intermediate frequency (IF) is a frequency to which a carrier frequency is shifted as an intermediate step in transmission or reception. Frequency modulation (FM) is a form of modulation that represents information as variations in the instantaneous frequency of a carrier wave. Very high frequency (VHF) is the radio frequency range from 30 MHz (wavelength 10 m) to 300 MHz (wavelength 1 heterodyne receivers "mix" all of the incoming signals with an internally generated waveform called the local oscillator. The user tunes the radio by adjusting the set's oscillator frequency, in the mixer stage of a receiver, the local oscillator signal multiplies with the incoming signals, which shifts them all down in frequency. The one that shifts is passed on by tuned circuits, amplified, and then demodulated to recover the original audio signal. The oscillator also shifts a "copy" of each incoming signal up in frequency by amount Those very high frequency "images" are all rejected by the tuned circuits in the IF stage. The Super heterodyne receiver (or to give it its full name, The Supersonic Heterodyne Receiver usually these days shortened to superhet) was invented by Edwin Armstrong in 1918.

BLOCK DIAGRAM

```
┌──────────────┐
│ Carrier      │
│ Freq(Fx)     │
└──────────────┘
                        ┌────────────────────┐          ┌────────────┐      ┌───────┐
                   ───→ │ Non linear circuit │   ───→   │ Filter     │      │ Fx-Fy │
                        │ (Amp BC-547/107)   │          │            │      └───────┘
┌──────────────┐   ───→ │                    │          │            │ ────→
│ Modulating   │        └────────────────────┘          └────────────┘
│ Freq (Fy)    │                                                            ┌───────┐
└──────────────┘                                                            │ Out put│
                                                                            └───────┘
```
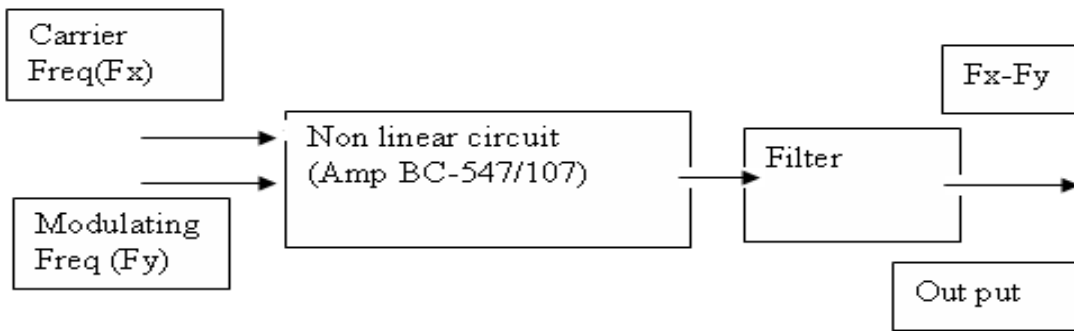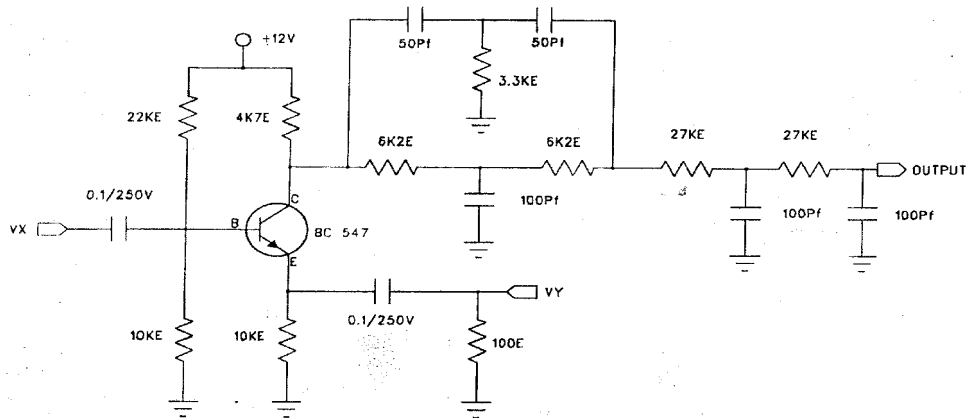
CIRCUIT DIAGRAM

EXPECTED WAVEFORMS

INPUT SIGNAL Fx

A m p l i T u d e Ac

TIME T ──────────→

INPUT SIGNAL Fy

A
m
p
l
i
T
u
d
e

TIME T ——————→

MIXER OUTPUT(Fy-Fx)

A
M
P
L
I
T
U
D
E

TIME T ——————→
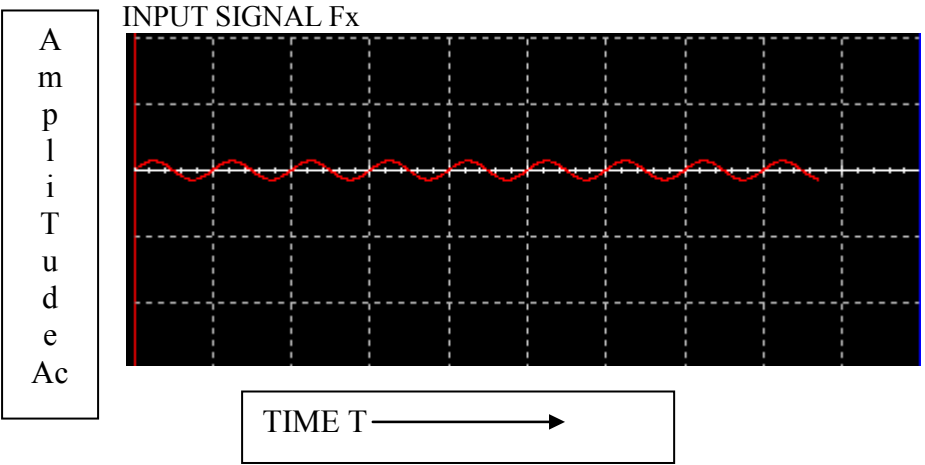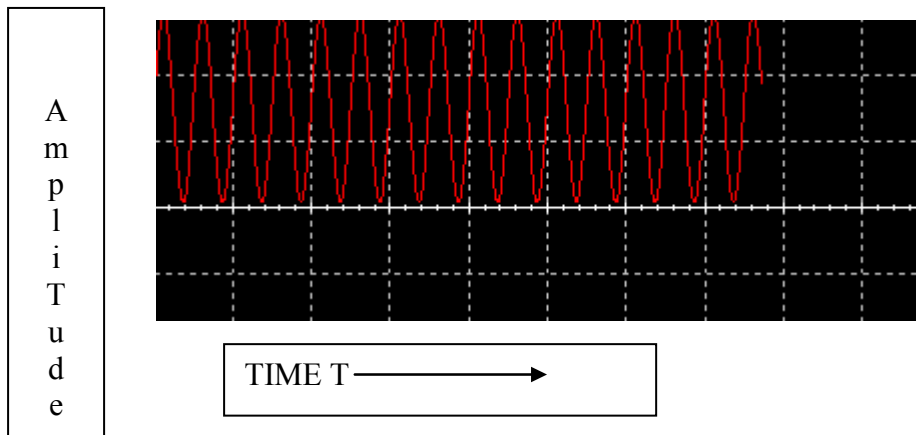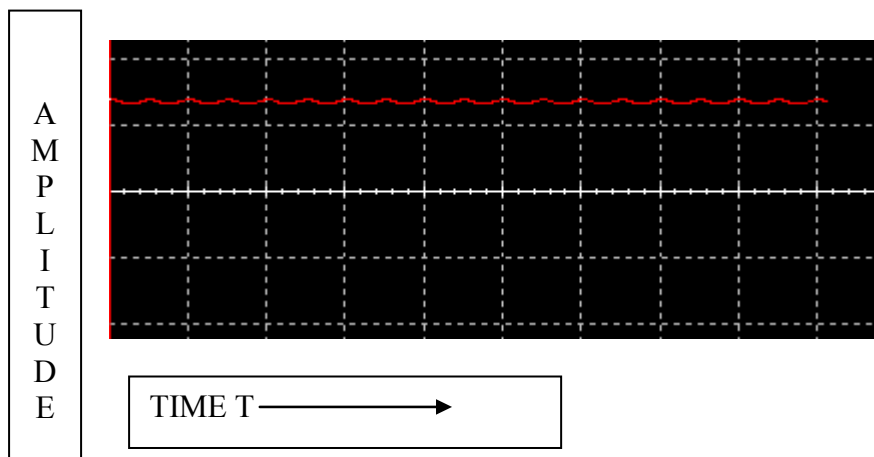
PROCEDURE:

1.   Connect the circuit as per the given circuit diagram.

2.   Switch on the power supply of trainer kit.

3.   Apply a sine wave at input Fx of 2 VP-P amplitude and 100 KHz frequency.

4.   Apply a sine wave at input FY of 2 VP-P amplitude and 100 KHz frequency.

5.   Observe the output waveform on the CRO.

6.   Repeat the steps 3,4 and 5 by changing the values of Fx once greater than and less than FY in a steps of 5Khz (in the range 80KHz to 120KHz)

TABULATION

S.No	Freq (Fy) (kHz)	Amp(V)	Freq(Fx) (kHz)	Amp(V)	Freq(Fy-Fx) (kHz)	Amp(V)

POST LAB:

1. Observe the amplitude and time. Of inputs and output in table
2. Draw the expected waveforms
3. Verify the output signal obtained with the theoretical value.
4. Plot the graphs for Time T V_S input signal (F_X) Voltage V ,input signal (F_Y) Voltage V and output Signal F(x-y) voltage V.
5. Write result and conclusions

CONCLUSIONS:

Obtained the output of the mixer for various inputs.

QUESTIONS:

1. what are the different types of mixtures available?

2. What is meant by heterodyning?

3. What is meant by IF?

EXPERIMENT NO: 2

DIGITAL PHASE DETECTOR

OBJECTIVE:

To detect the phase difference between two square wave signals using digital phase detector.

PRE LAB:

1. To study the theory of digital phase detector and EX-OR gate operation and truth table.

2. Draw the circuit diagram and expected output and input waveforms

EQUIPMENT:

1. Digital phase detector trainer kit

2. Regulated power supplies

3. C.R.O (20MHz)

4. Connecting cords & probes

5. Digital frequency counter or multimeter

THEORY:

Phase detector is a frequency mixer or analog multiplier circuit that generates a voltage signal which represents the difference in phase between two signal inputs. It is an essential element of the phase-locked loop (PLL).Detecting phase differences is very important in many applications, such as motor control, radar and telecommunication systems, servo mechanisms, and demodulators.

Electronic phase detector

Some signal processing techniques such as those used in radar may require both the amplitude and the phase of a signal, to recover all the information encoded in that signal. One technique is to feed an amplitude-limited signal into one port of a product detector and a reference signal into the other port; the output of the detector will represent the phase difference between the signals. If the signal is different in frequency from the reference, the detector output will be periodic at the difference frequency. Phase detectors for phase-locked loop circuits may be classified in two types. A Type I detector is designed to be driven by analog signals or square-wave digital signals and produces an output pulse at the difference frequency. The Type I detector always produces an output waveform, which must be filtered to control the phase-locked loop variable frequency oscillator (VCO). A type II detector is sensitive only to the relative timing of the edges of the input and reference pulses, and produces a constant output proportional to phase difference when both signals are at the same frequency. This output will tend not to produce ripple in the control voltage of the VCO

BLOCK DIAGRAM OF PLL
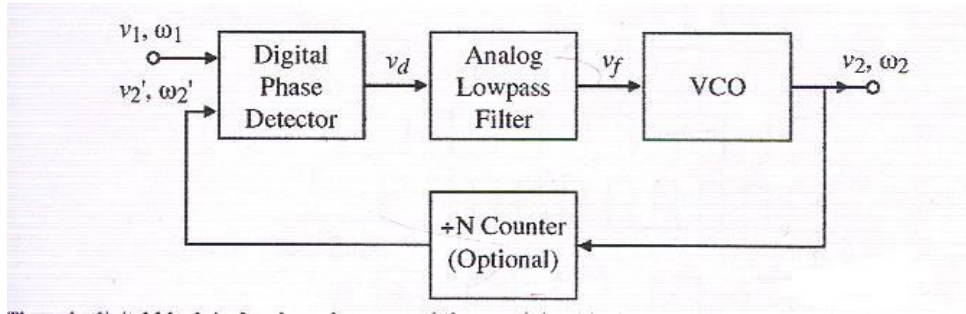
CIRCUIT DIAGRAM
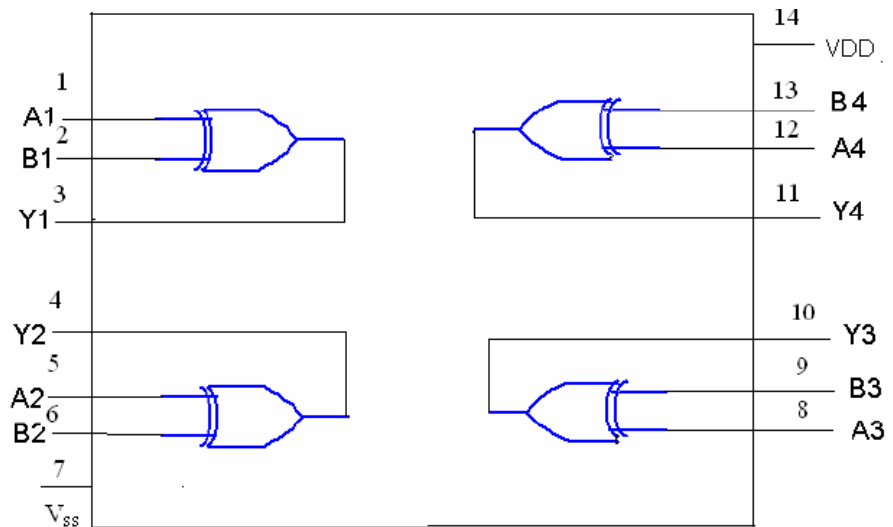
CD4070 quad 2-input exclusive OR gate   VDD: 3 to 15V
VSS: Ground

EX-OR phase detector logic diagram
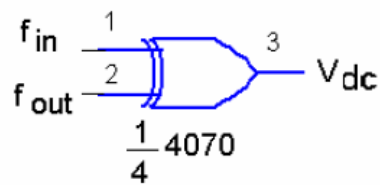
EXPECTED WAVEFORMS

EX-OR PHASE DETECTOR
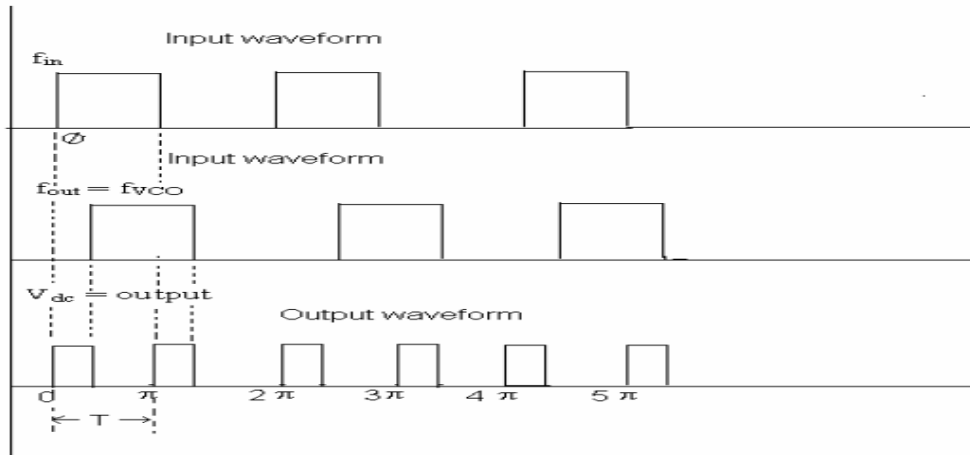
EDGE TRIGGERED OUTPUT
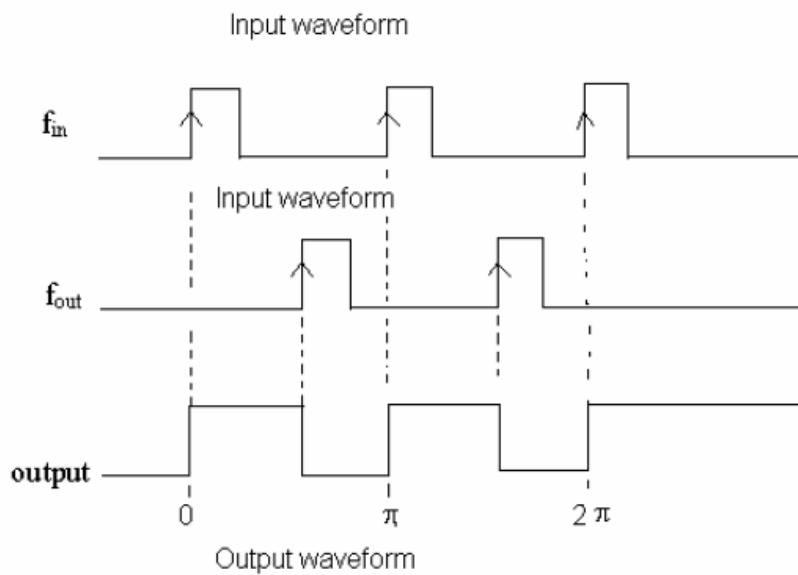
PROCEDURE

1. Switch on the trainer kit

2. Observe the output of the square wave generator available on the trainer kit using CRO and measure the range with the help of frequency counter, the frequency range should be around 2 KHz to 13 KHz.

3. Calculate the free running range of the VCO output i.e. between 4th pin of IC PLL 565 and ground. For different values of timing resistor Rt, fout is given by Fout =0.3/ Rt Ct where Ct: timing capacitor = 0.01 μF, Rt : timing resistor

4. Connect the square wave to the input of IC PLL 565 and short 4th and 5th pin of PLL. Vary the input frequency of the square wave, when the PLL is locked that is connected to

one input fout EX-OR phase detector. The other input fin of EX-OR phase detector is the coming from inbuilt of square wave generator.

5. Connect the pulse generator output to the input of IC 565 PLL and short 4th & 5th pin of PLL. Vary the input frequency of the square wave when the PLL is locked that is connected to one input of Edge triggered phase detector input i.e. fout. The other input fin of edge triggered phase detector is the pulse input coming from the inbuilt pulse generator.

6. The dc output voltage of the exclusive-OR phase detector is a function of the phase difference between its two inputs fin and fout.

POST LAB:

1. Observe the Voltage and Time of the two input square waves
2. Observe difference of two inputs as square wave output.
3. Plot the inputs and output waveforms Time T V_S Voltage V

CONCLUSIONS:

The phase detector outputs are verified

QUESTIONS:

1. What is the function of phase detector?

2. What is the condition for getting maximum output?

3. Advantages of digital phase detector over analog phase detector

EXPERIMENT NO: 3

COASTAS RECEIVER

AIM

To demodulate a DSB-SC signal using coastas receiver.

PRE LAB:

1.   To study the operation of coastas receiver.
2.   Draw the circuit diagram and expected waveforms

EQUIPMENTS:

1.   Communications trainer kit.
2.   DSB-SC modulator trainer kit.
3.   C.R.O (20MHz)
4.   Connecting cords and probes.
5.   Function generator (1MHz).

THEORY:

This loop, and its variations, is much-used as a method of carrier acquisition (and simultaneous message demodulation) in communication systems, both analog and digital.
It has the property of being able to derive a carrier from the received signal, even when there is no component at carrier frequency present in that signal (eg, DSBSC). The requirement is that the amplitude spectrum of the received signal be symmetrical about this frequency.

The Costas loop is based on a pair of quadrature modulators - two multipliers fed with carriers in phase-quadrature. These multipliers are in the in-phase (I) and quadrature phase (Q) arms of the arrangement. Each of these multipliers is part of separate synchronous demodulators. The outputs of the modulators, after filtering, are multiplied together in a third multiplier, and the lowpass components in this product are used to adjust the phase of the local carrier source - a VCO - with respect to the received signal. The operation is such as to maximise the output of the I arm, and minimize that from the Q arm. The output of the I arm happens to be the message, and so the Costas loop not only acquires the carrier, but is a (synchronous) demodulator as well. A complete analysis of this loop is non-trivial. It would include the determination of conditions for stability, and parameters such as lock range, capture range, and so on.

BLOCK DIAGRAM
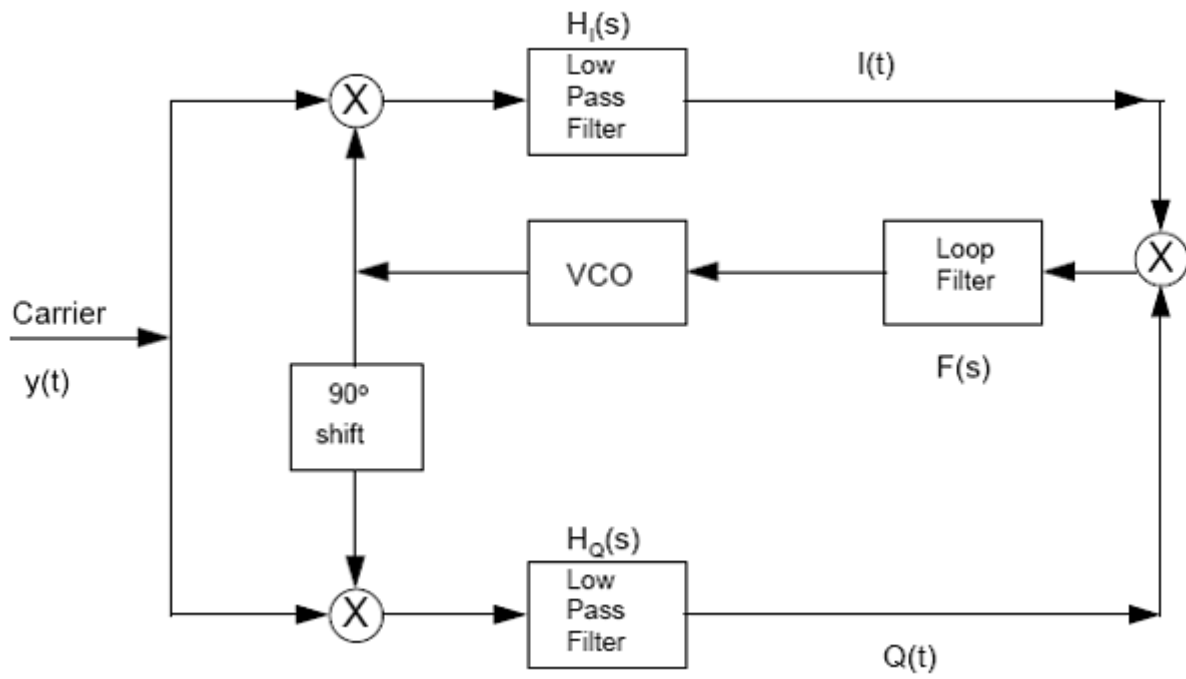
EXPECTED WAVEFORMS
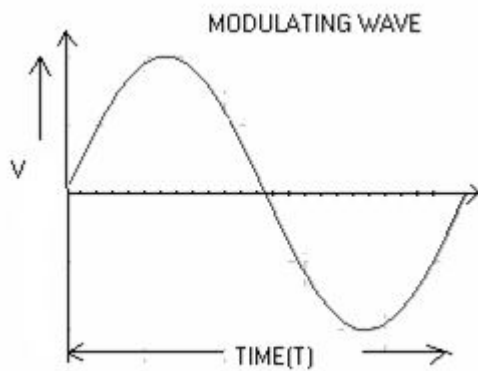
DSB-SC modulated output
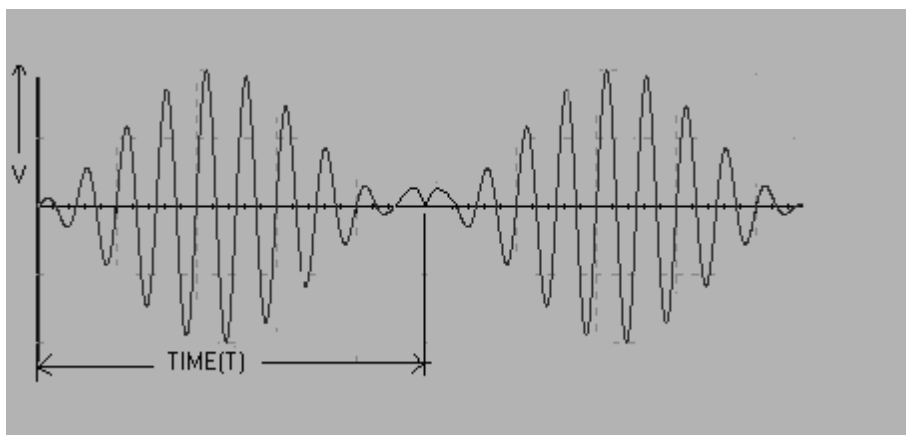
Demodulated output of the coastas loop :
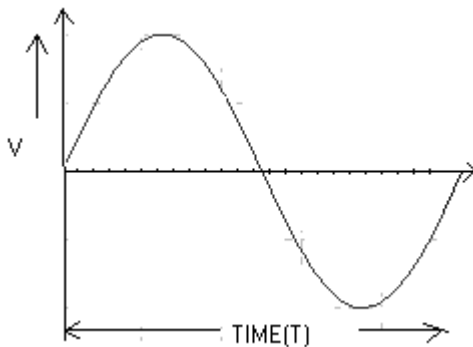
PROCEDURE:

1. Switch ON the DSB-SC trainer kit

2. Generate the carrier and modulating signal.

3. Generate the DSB-SC modulated signal.

4. Connect the required blocks of the coastas receiver in the communications kit.

5. Apply the generated DSB-SC signal to the above connected coastas receiver.

6. Observe that the output of the coastas receiver is the demodulated wave of DSB-SC signal.

POST LAB

1 .Observe the frequency and amplitude of the carrier and modulating signal

2. Observe the frequency and amplitude of the balanced modulator output

3. Observe the lock condition of the coastas loop.

4.   observe the output of the coastas demodulator.

5. Write result and conclusions

CONCLUSIONS: The balanced modulator output and the operation of coastas receiver is verified

QUESTIONS: 1. what is the operation of coastas loop?

2. What happens when the locking condition of the coastas loop occurs?

3. Coastas loop can be used for demodulation of other modulation techniques also. What are they?

MATLAB PROGRAMMES

INDEX

1 INTRODUCTION

What Is Simulink?

Simulink is a software package for modeling, simulating, and analyzing dynamic systems. It supports linear and nonlinear systems, modeled in continuous time, sampled time, or a hybrid of the two. Systems can also be multirate, i.e., have different parts that are sampled or updated at different rates.

Tool for Simulation

Simulink encourages you to try things out. You can easily build models from scratch, or take an existing model and add to it. You have instant access to all the analysis tools in MATLAB®, so you can take the results and analyze and visualize them. A goal of Simulink is to give you a sense of the fun of modeling and simulation, through an environment that encourages you to pose a question, model it, and see what happens.

Simulink is also practical. With thousands of engineers around the world using it to model and solve real problems, knowledge of this tool will serve you well throughout your professional career.

Tool for Model-Based Design

With Simulink, you can move beyond idealized linear models to explore more realistic nonlinear models, factoring in friction, air resistance, gear slippage, hard stops, and the other things that describe real-world phenomena. Simulink turns your computer into a lab for modeling and analyzing systems that simply wouldn't be possible or practical otherwise, whether the behavior of an automotive clutch system, the flutter of an airplane wing, the dynamics of a predator-prey model, or the effect of the monetary supply on the economy.

For modeling, Simulink provides a graphical user interface (GUI) for building models as block diagrams, using click-and-drag mouse operations. With this interface, you can draw the models just as you would with pencil and paper (or as most textbooks depict them). This is a far cry from previous simulation packages that require you to formulate differential equations and difference equations in a language or program. Simulink includes a comprehensive block library of sinks, sources, linear and nonlinear components, and connectors. You can also customize and create your own blocks. For information on creating your own blocks, see the separate Writing S-Functions guide.

Models are hierarchical, so you can build models using both top-down and bottom-up approaches. You can view the system at a high level, then double-click blocks to go down through the levels to see increasing levels of model detail. This approach provides insight into how a model is organized and how its parts interact.
After you define a model, you can simulate it, using a choice of integration methods, either from the Simulink menus or by entering commands in the MATLAB Command Window. The menus are particularly convenient for interactive work, while the command-line approach is very useful for running a batch of simulations (for example, if you are doing Monte Carlo simulations or want to sweep a parameter across a range of values). Using scopes and other display blocks, you can see the simulation results while the simulation is running. In addition,

you can change many parameters and see what happens for "what if" exploration. The simulation results can be put in the MATLAB workspace for post processing and visualization.

Model analysis tools include linearization and trimming tools, which can be accessed from the MATLAB command line, plus the many tools in MATLAB and its application toolboxes. And because MATLAB and Simulink are integrated, you can simulate, analyze, and revise your models in either environment at any point

1. Generation of cosine wave in time domain and frequency domain

```
Clc; clear all; close all;
x = -5:0.001:5;
t = 0:1/4000:1;
time = cos(2*3.14*1000*t);
y1 = cos(2*3.14*1000*x);
subplot(2,1,1);
plot(x,y1)
axis([-5 5 -3 3]);
grid on
title('Time domain');
xlabel('Time ');
ylabel('Amplitude');

% now create a frequency vector for the x-axis and plot the magnitude and
phase

subplot(2,1,2);
fre = abs(fft(time));
f = (0:length(fre) - 1)'*4000/length(fre);
plot(f,fre);
grid on
title('Frequency domain (Spectrum)');
xlabel('Freq ');
ylabel('Amplitude');
```

WAVEFORMS:
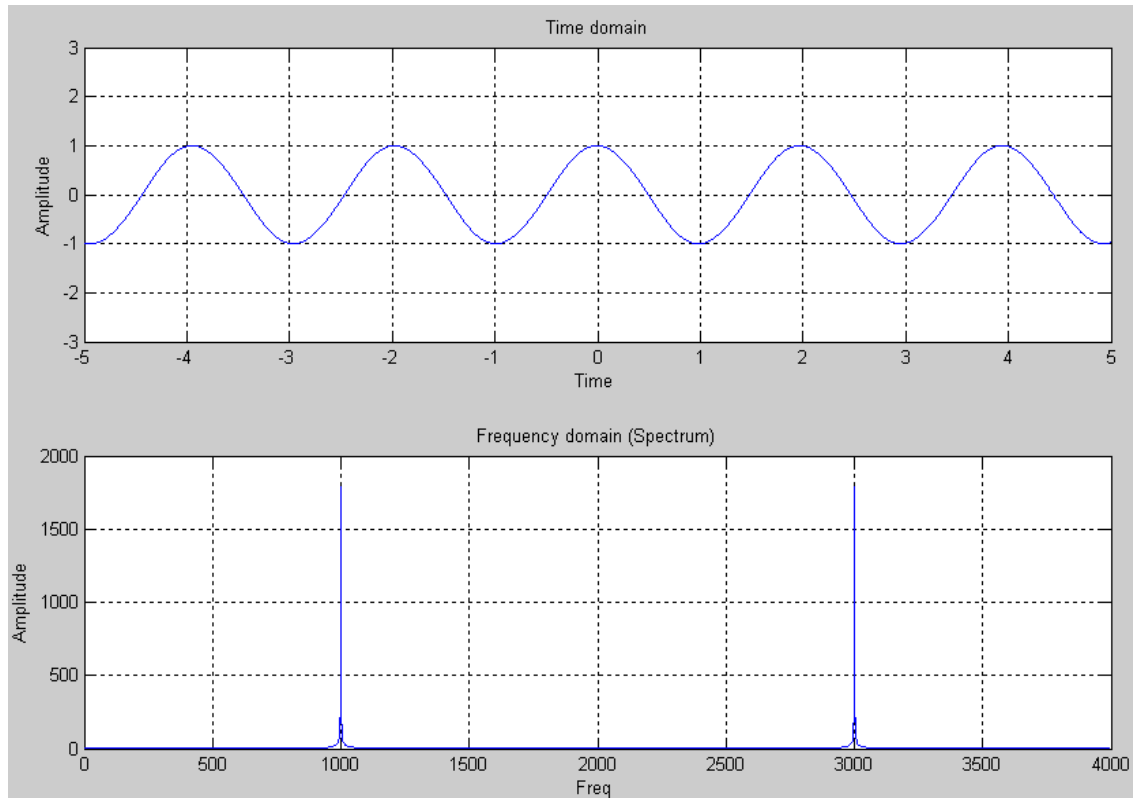
2. Generation of square wave in time domain and frequency domain

```
clc;
clear all;
close all;
x = -5:0.001:5;
Fs = 399;
t = 0:1/Fs:1;
time = SQUARE(2*pi*1000*t);
y1 = SQUARE(2*3.14*1000*x);
subplot(2,1,1);
plot(x,y1)
axis([-5 5 -3 3]);
grid on
xlabel('Time domain');
ylabel('Amplitude');

% now create a frequency vector for the x-axis and plot the magnitude and
phase

subplot(2,1,2);
fre = abs(fft(time));
f = (0:length(fre) - 1)'*Fs/length(fre);
plot(f,fre);
grid on
xlabel('Freq domain');
ylabel('Amplitude');
```

WAVEFORMS:
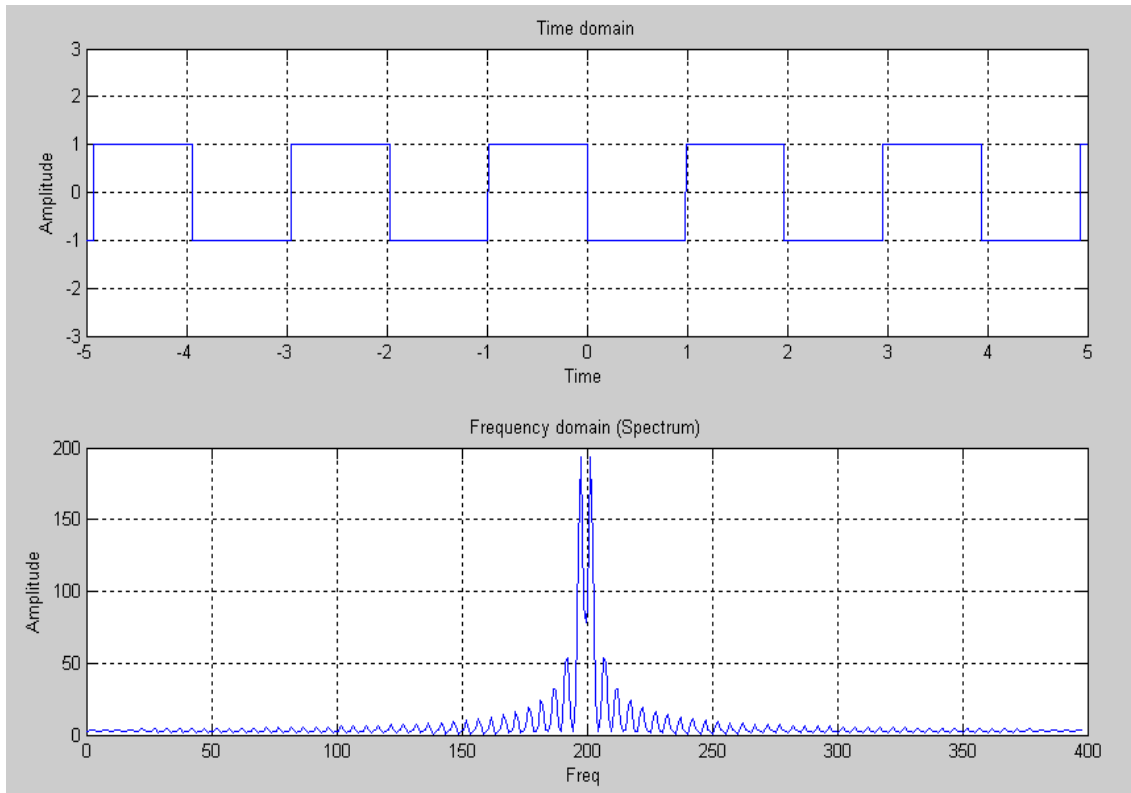
2. AMPLITUDE MODULATION AND DEMODULATION

PROGRAM 1:

```
clc;
clear all;
close all;
Ac=1;                  %Carrier Amplitude
Fc=0.4;                %Carrier frequency
Fm=0.05;               %baseband frequency
Fs=10;                 %sampling
t=0:0.1:200;
mt=cos(2*pi*Fm*t);

%%%%%%%%%%%%%%%%%%%%% under modulation %%%%%%%%%%%%%%%%%%%%%%%%%%%

mu=0.5;
st=Ac*(1+mu*mt).*cos(2*pi*Fc*t);
subplot(2,1,1);
plot(t,st);
hold on;
plot(t,Ac*(mu*mt+ones(1,length(mt))),'r');
plot(t,-Ac*(mu*mt+ones(1,length(mt))),'r');
hold off;
title('Modulation Index = 0.5 under modulation
Ac(1+0.5*cos(2*pi*0.05*t))cos(2*pi*0.4*t)');
xlabel('time (s)');ylabel('amplitude');
st_fft=fft(st);
st_fft=fftshift(st_fft);
st_fft_fre=5*linspace(-1,1,length(st_fft));
subplot(2,1,2);
plot(st_fft_fre,abs(st_fft));
title('spectrum with Modulation Index = 0.5');
xlabel('Frequency (Hz)');axis([-1 1 0 1000*Ac+100]);

%%%%%%%%%%%%%%%%%%%%% total modulation %%%%%%%%%%%%%%%%%%%%%%%%%%%

figure;
mu=1;
st=Ac*(1+mu*mt).*cos(2*pi*Fc*t);
subplot(2,1,1);
plot(t,st);
hold on;
plot(t,Ac*(mu*mt+ones(1,length(mt))),'r');
plot(t,-Ac*(mu*mt+ones(1,length(mt))),'r');
hold off;
title('Modulation Index = 1 modulation
Ac(1+1*cos(2*pi*0.05*t))cos(2*pi*0.4*t)');
xlabel('time (s)');ylabel('amplitude');
st_fft=fft(st);
st_fft=fftshift(st_fft);
st_fft_fre=Fs/2*linspace(-1,1,length(st_fft));
subplot(2,1,2);
plot(st_fft_fre,abs(st_fft));
title('spectrum with Modulation Index = 1');
xlabel('Frequency (Hz)');axis([-1 1 0 1000*Ac+100]);
```

```
%%%%%%%%%%%%%%%%%%%% over Modulation %%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%

figure;
mu=2;
st=Ac*(1+mu*mt).*cos(2*pi*Fc*t);
subplot(2,1,1);
plot(t,st);
hold on;
plot(t,Ac*(mu*mt+ones(1,length(mt))),'r');
plot(t,-Ac*(mu*mt+ones(1,length(mt))),'r');
hold off;title('Modulation Index = 2 over modulation
Ac(1+2*cos(2*pi*0.05*t))cos(2*pi*0.4*t)');
xlabel('time (s)');ylabel('amplitude');
st_fft=fft(st);
st_fft=fftshift(st_fft);
st_fft_fre=Fs/2*linspace(-1,1,length(st_fft));
subplot(2,1,2);
plot(st_fft_fre,abs(st_fft));
title('spectrum with Modulation Index = 2');
xlabel('Frequency (Hz)');axis([-1 1 0 1000*Ac+100]);
```

WAVEFORMS :
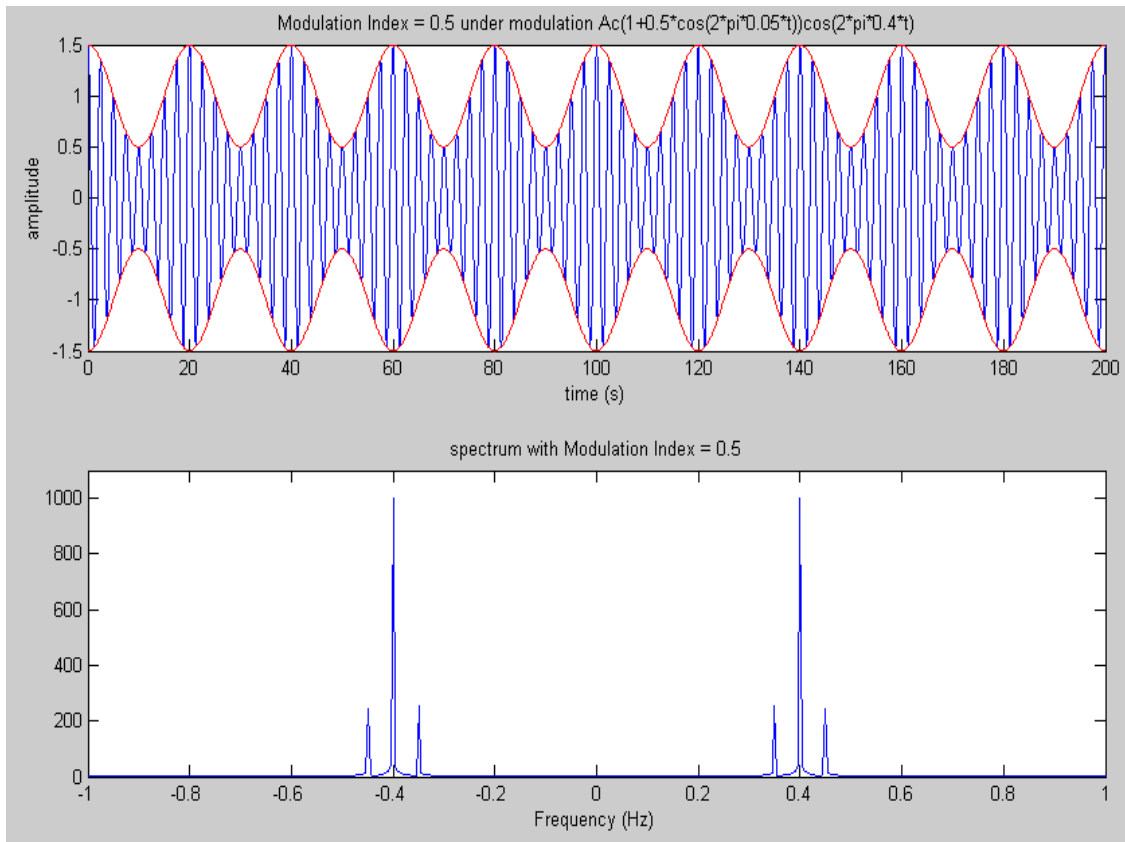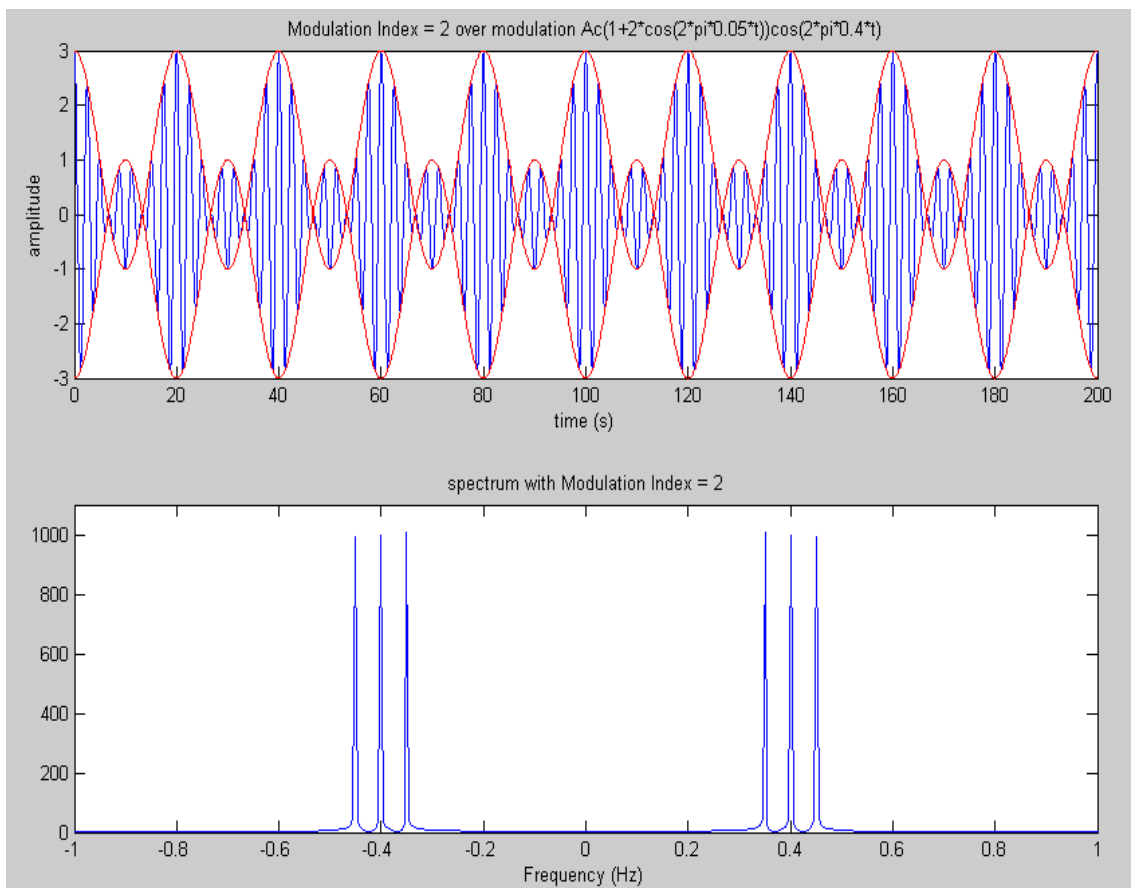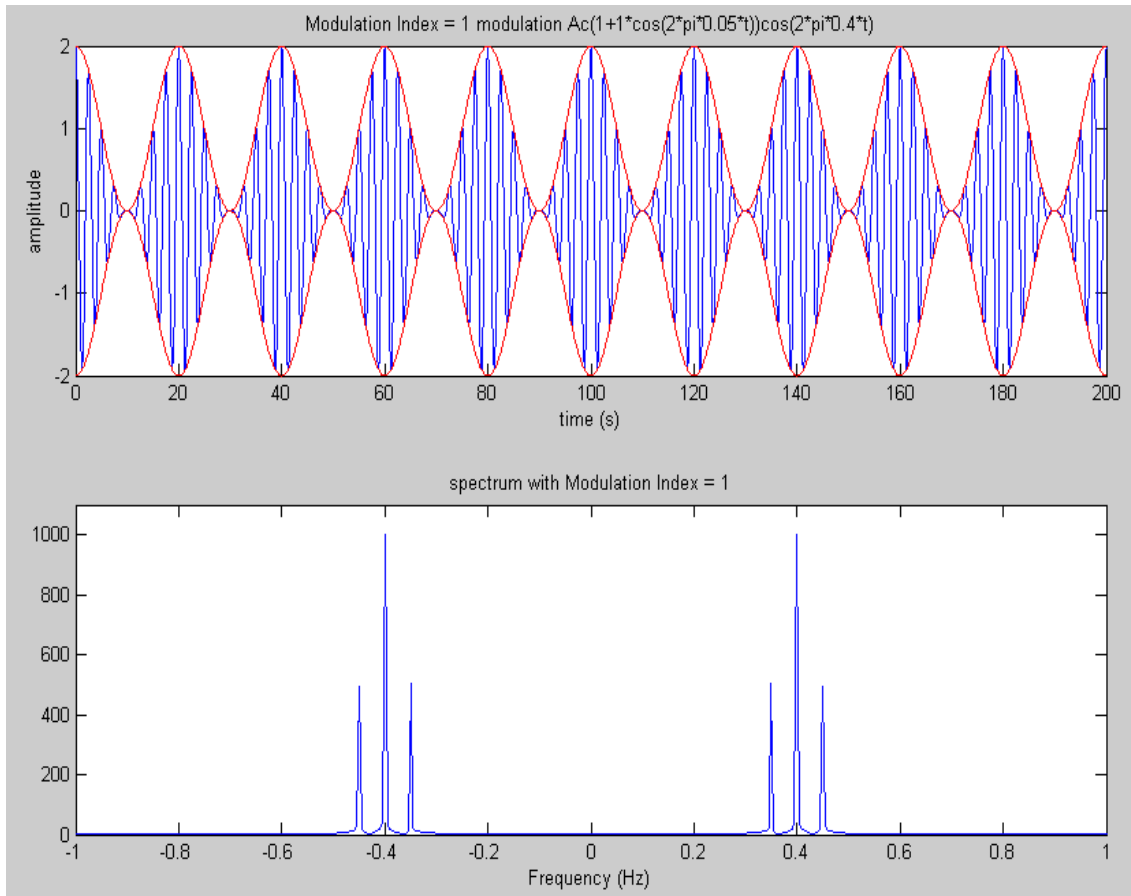
Modulation Index = 1 modulation Ac(1+1*cos(2*pi*0.05*t))cos(2*pi*0.4*t)

spectrum with Modulation Index = 1

Modulation Index = 2 over modulation Ac(1+2*cos(2*pi*0.05*t))cos(2*pi*0.4*t)

spectrum with Modulation Index = 2

PROGRAM 2:

```
% task 1
fc=154000;

% task 2
fm=fc/10;
fs=100*fc;
t=0:1/fs:4/fm;
xc=cos(2*pi*fc*t);
xm=cos(2*pi*fm*t);
figure(1)
subplot(2,1,1),plot(t,xc);
title('carrier signal of 154 khz');
xlabel('time (sec)');
ylabel('amplitude');
subplot(2,1,2),plot(t,xm);
title('message signal of 15.4 khz');
xlabel('time (sec)');
ylabel('amplitude');
% DSB-SC MODULATION
mu = input(' enter the Modulation Index value :: ');
z1=(1+mu*xm).*xc;
figure(2)

% task 3.1
subplot(2,1,1),plot(t,z1);
title('AMPLITUDE MODULATION IN TIME DAOMAIN');
xlabel('time (sec)');
ylabel('amplitude');

% task 3.2
l1=length(z1);
f=linspace(-fs/2,fs/2,l1);
Z1=fftshift(fft(z1,l1)/l1);
subplot(2,1,2),plot(f,abs(Z1));
title('AMPLITUDE  MODULATION IN FREQUENCY DOMAIN');
xlabel('frequency(hz)');
ylabel('amplitude');
axis([-200000 200000 0 0.3]);

% task 3.3 demodulation
s1=z1.*xc;
S1=fftshift(fft(s1,length(s1))/length(s1));
figure(3)
plot(f,abs(S1));
title(' demodulated signal IN FREQUENCY DOMAIN before filtring');
xlabel('frequency(hz)');
ylabel('amplitude');
axis([-200000 200000 0 0.3]);
hold on
Hlp=1./sqrt(1+(f./fc).^(2*100));
plot(f,Hlp,'r');
title(' frequency response of low pass filter');
xlabel('frequency(hz)');
ylabel('amplitude');
axis([-200000 200000 0 2]);

% task 3.4
E1=Hlp.*S1;
figure(4)
subplot(2,1,1),plot(f,E1);
```
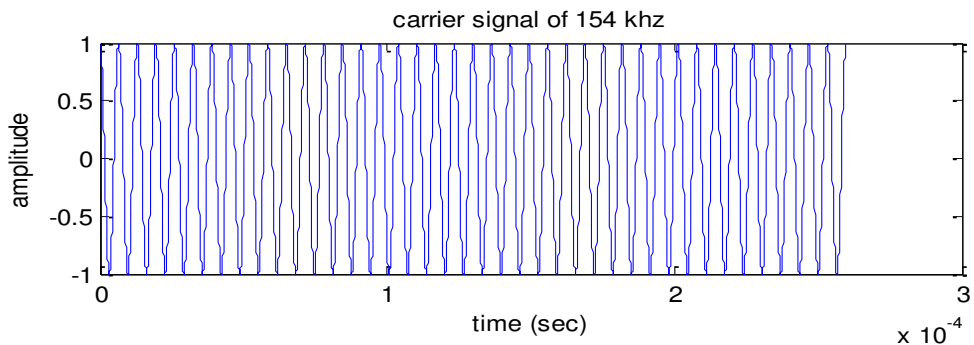
```
title(' Recover signal IN FREQUENCY DOMAIN after filtring');
xlabel('frequency(hz)');
ylabel('amplitude');
axis([-200000 200000 0 0.3]);
e1=ifft(ifftshift(E1))*length(E1);
subplot(2,1,2),plot(t,(1/0.5)*e1);
title(' Recover signal IN Time DOMAIN after filtring');
xlabel('time(sec)');
ylabel('amplitude');
```

WAVEFORMS:

carrier signal of 154 khz

message signal of 15.4 khz

enter the Modulation Index value :: 0.5

AMPLITUDE MODULATION IN TIME DAOMAIN

AMPLITUDE MODULATION IN FREQUENCY DOMAIN

frequency response of low pass filter

Recover signal IN FREQUENCY DOMAIN after filtring

Recover signal IN Time DOMAIN after filtring

enter the Modulation Index value :: 1

AMPLITUDE MODULATION IN TIME DAOMAIN

AMPLITUDE MODULATION IN FREQUENCY DOMAIN

frequency response of low pass filter

Recover signal IN FREQUENCY DOMAIN after filtring

Recover signal IN Time DOMAIN after filtring

enter the Modulation Index value :: 2

AMPLITUDE MODULATION IN TIME DAOMAIN

AMPLITUDE MODULATION IN FREQUENCY DOMAIN

frequency response of low pass filter

Recover signal IN FREQUENCY DOMAIN after filtring

Recover signal IN Time DOMAIN after filtring
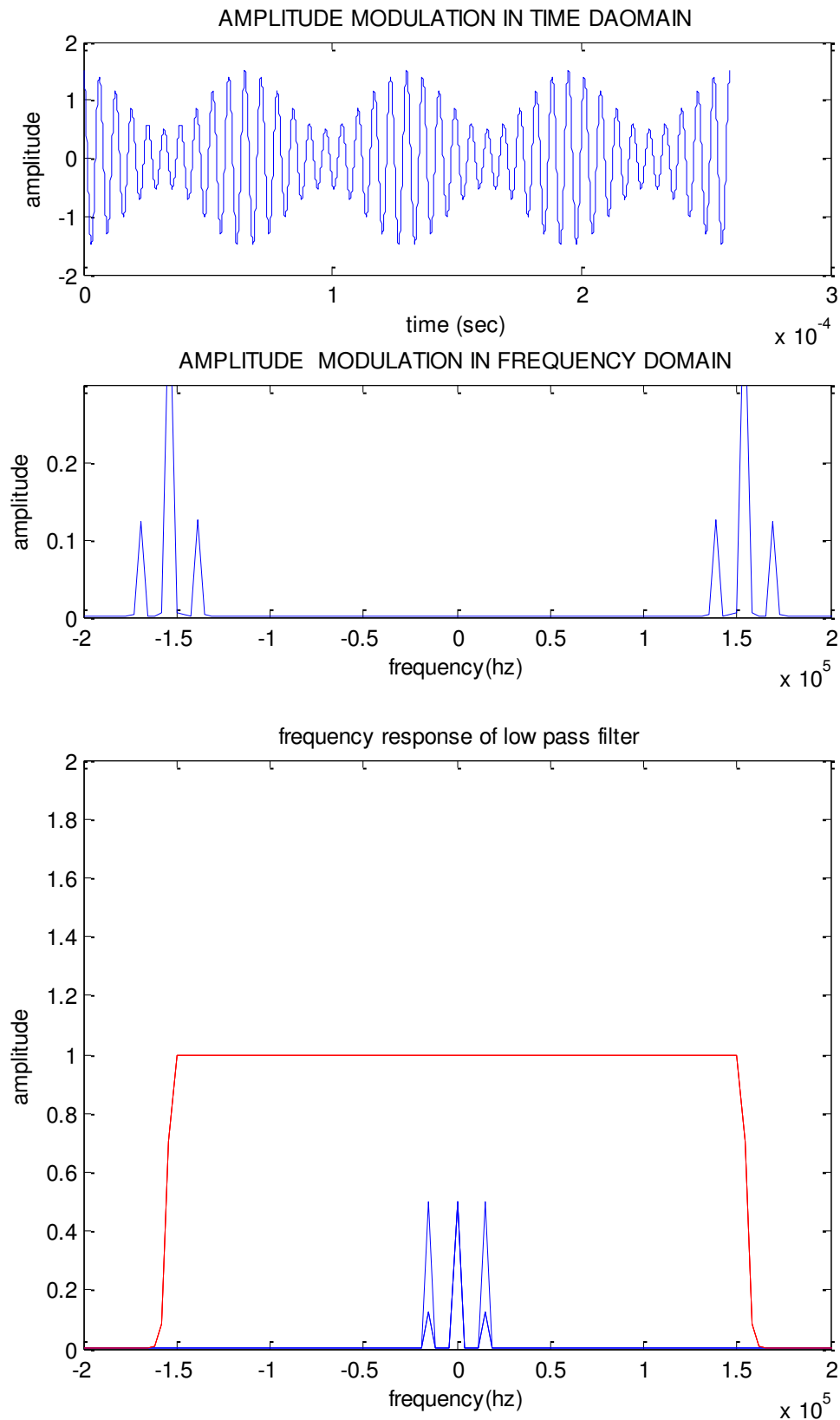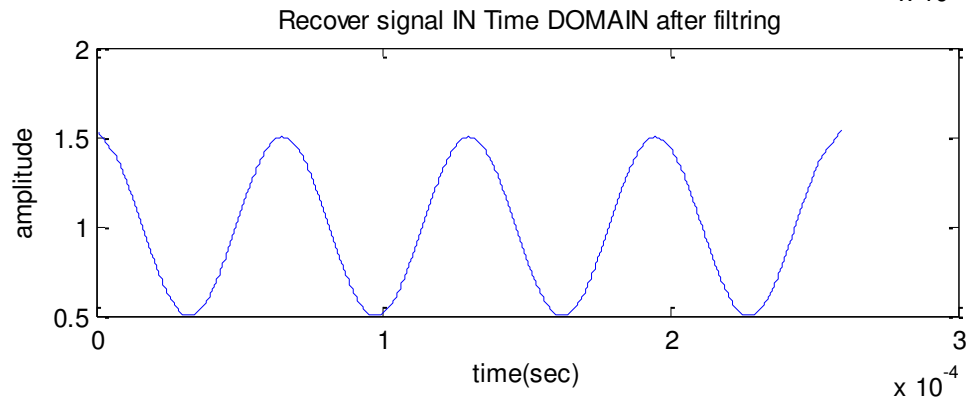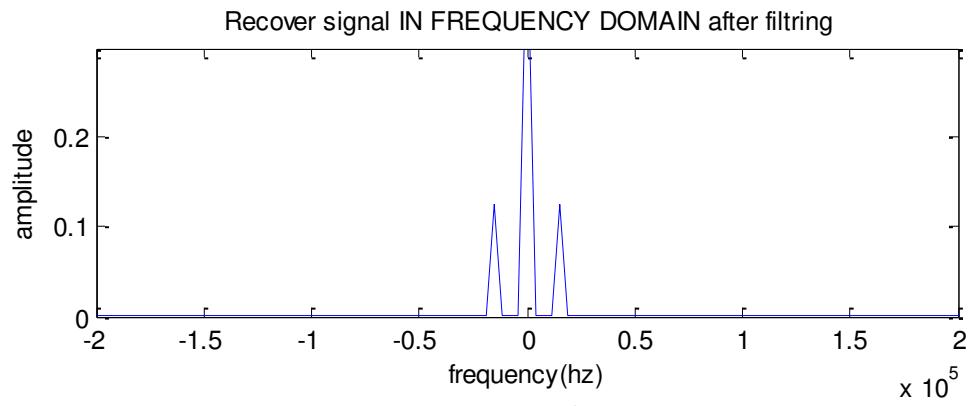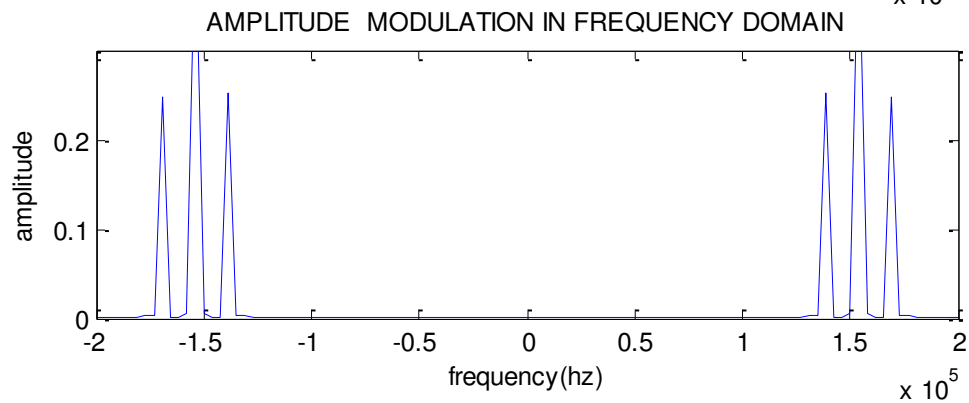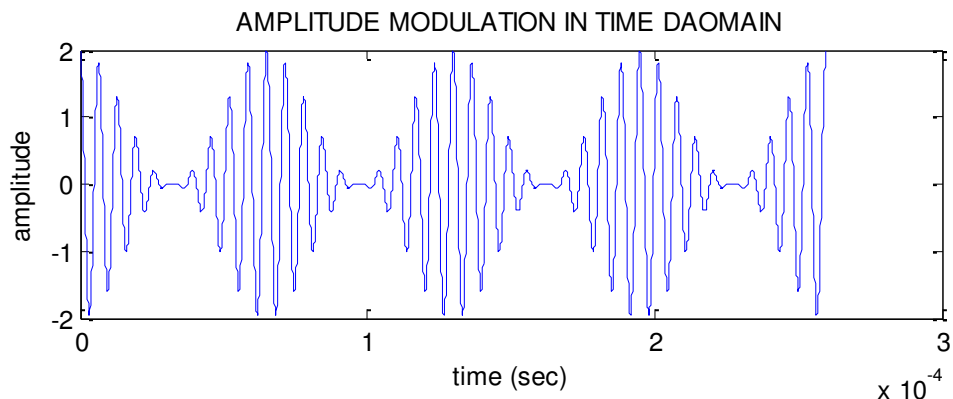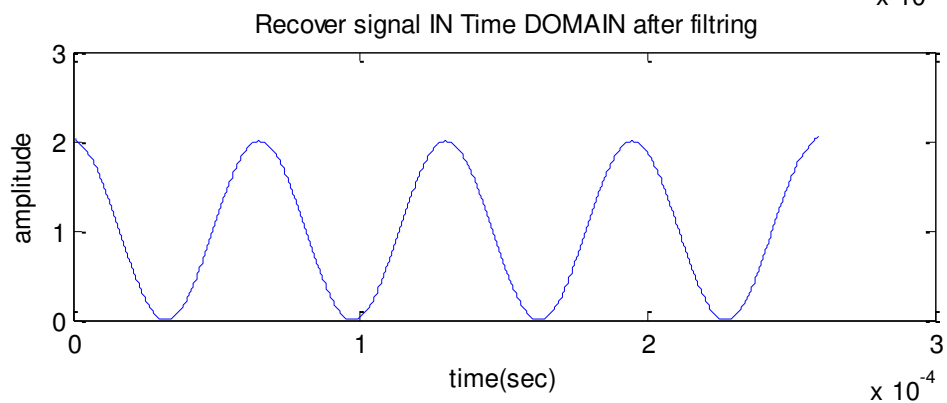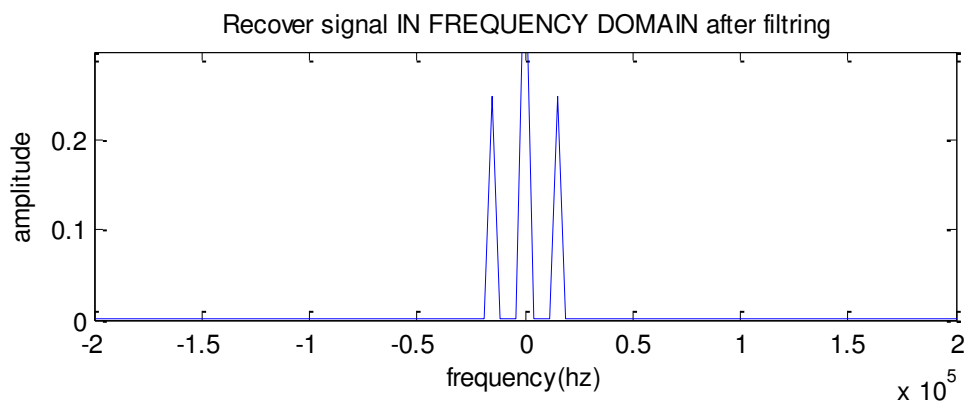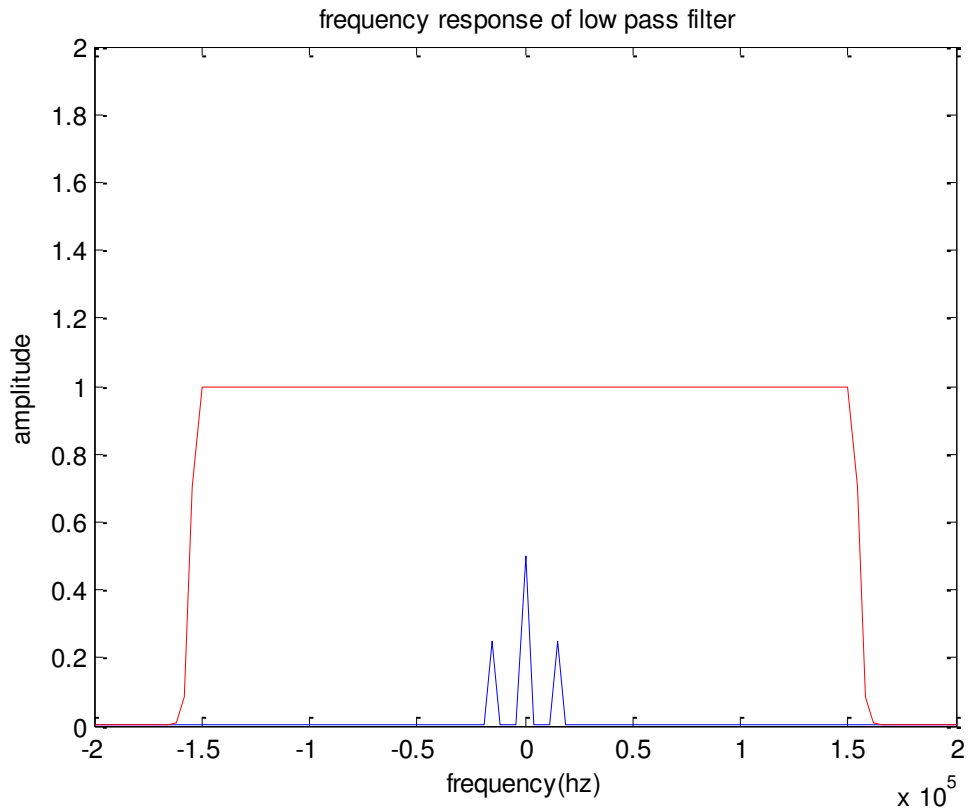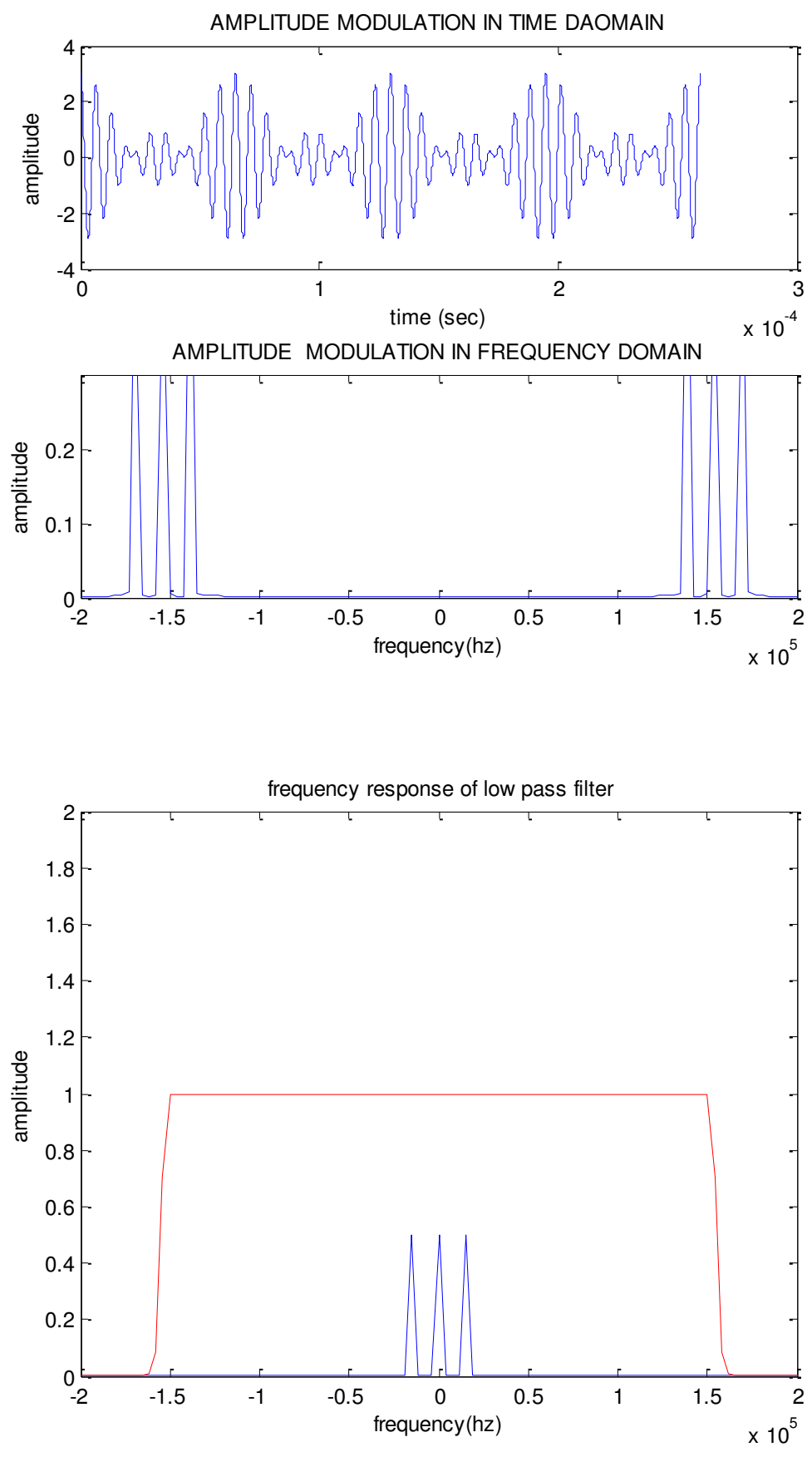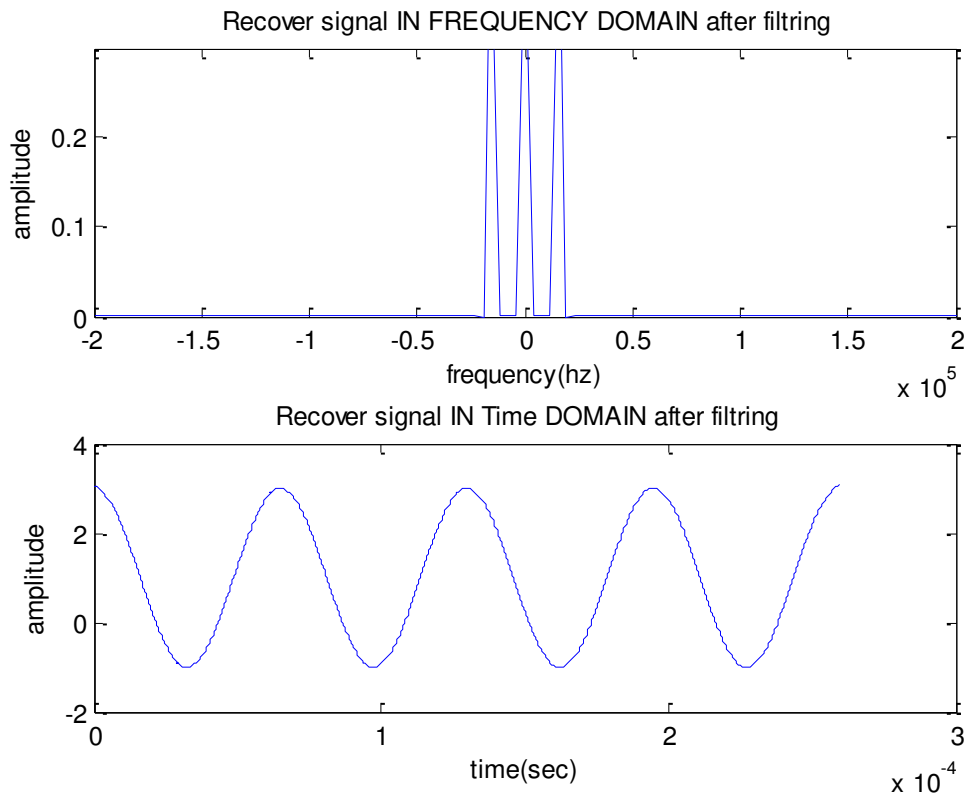
3. DOUBLE SIDEBAND – SUPRESSED CARRIER MODULATION AND DEMODULATION

PROGRAM 1:

```
clc;
clear all;
close all;
Ts = 199;
subplot(4,1,1);
t = 0:1/Ts:1;
m = cos(2*pi*1000*t);
plot(t,m);
title('Modulating Signal ');
xlabel('Time');
ylabel('Amplitude');
grid on

% plot of the carrier signal

subplot(4,1,2);
c = cos(2*pi*5000*t);
plot(t,c);
title('Carrier Signal ');
xlabel('Time');
ylabel('Amplitude');
grid on

% plot of the DSB signal with Suppresed carrier intime domain

subplot(4,1,3);
d = m.*c;
plot(t,d);
title('DOUBLE SIDEBAND – SUPRESSED CARRIER  SIGNAL ');
xlabel('Time');
ylabel('Amplitude');
grid on

% freq. domain of the DSB signal.

subplot(4,1,4);
fre = abs(fft(d));
f = (0:length(fre) - 1)'*Ts/length(fre);
plot(f,fre);
axis([0 100 0 50]);
grid on
title('DOUBLE SIDEBAND – SUPRESSED CARRIER  SPECTRUM ');
xlabel('Freq domain');
ylabel('Amplitude');
```

WAVEFORMS:

Modulating Signal

Carrier Signal

DOUBLE SIDEBAND – SUPRESSED CARRIER SIGNAL

DOUBLE SIDEBAND – SUPRESSED CARRIER SPECTRUM
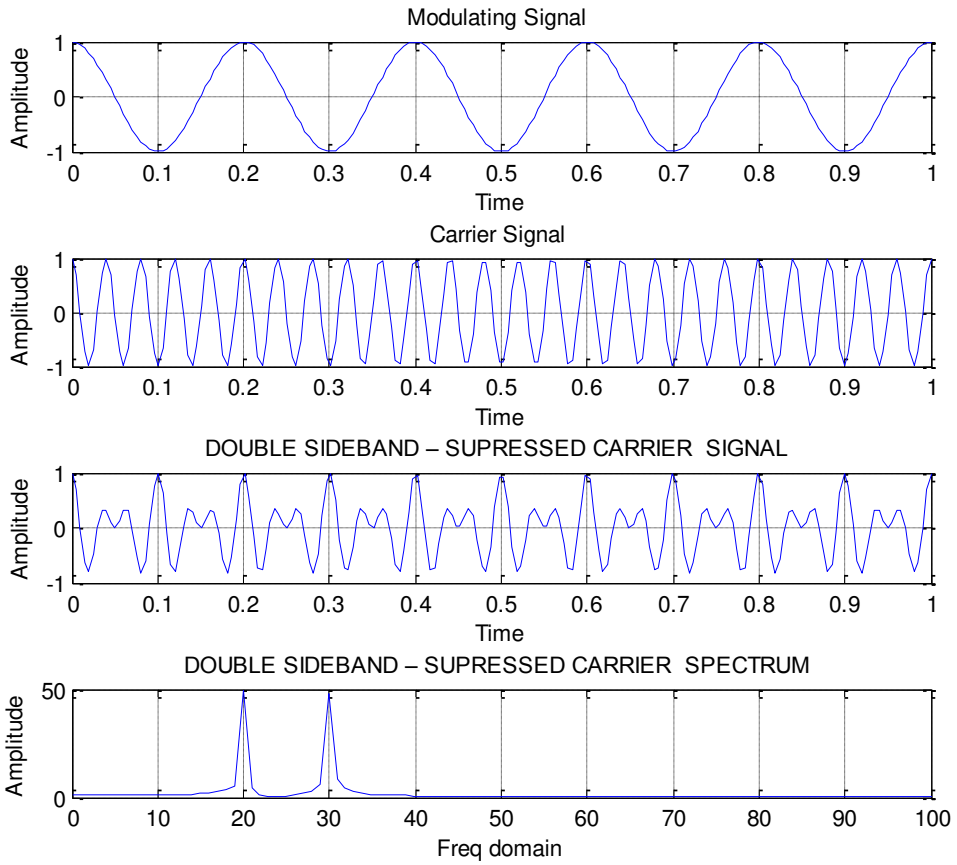
PROGRAM 2:

```
% task 1
fc=154000;
% task 2
fm=fc/10;
fs=100*fc;
t=0:1/fs:4/fm;
xc=cos(2*pi*fc*t);
xm=cos(2*pi*fm*t);
figure(1)
subplot(2,1,1),plot(t,xc);
title('carrier signal of 154 khz');
xlabel('time (sec)');
ylabel('amplitude');
subplot(2,1,2),plot(t,xm);
title('message signal of 15.4 khz');
xlabel('time (sec)');
ylabel('amplitude');
% DSB-SC MODULATION
z1= xm.*xc;
figure(2)
% task 3.1
subplot(2,1,1),plot(t,z1);
title('DSB-SC MODULATION IN TIME DAOMAIN');
xlabel('time (sec)');
ylabel('amplitude');
% task 3.2
l1=length(z1);
f=linspace(-fs/2,fs/2,l1);
Z1=fftshift(fft(z1,l1)/l1);
subplot(2,1,2),plot(f,abs(Z1));
title('DSB SC MODULATION IN FREQUENCY DOMAIN');
xlabel('frequency(hz)');
ylabel('amplitude');
axis([-200000 200000 0 0.3]);
% task 3.3 demodulation
s1=z1.*xc;
S1=fftshift(fft(s1,length(s1))/length(s1));
figure(3)
plot(f,abs(S1));
title(' demodulated signal IN FREQUENCY DOMAIN before filtring');
xlabel('frequency(hz)');
ylabel('amplitude');
axis([-200000 200000 0 0.3]);
hold on
Hlp=1./sqrt(1+(f./fc).^(2*100));
plot(f,Hlp,'g');
title(' frequency response of low pass filter');
xlabel('frequency(hz)');
ylabel('amplitude');
axis([-200000 200000 0 2]);
% task 3.4
E1=Hlp.*S1;
figure(4)
subplot(2,1,1),plot(f,E1);
title(' Recover signal IN FREQUENCY DOMAIN after filtring');
xlabel('frequency(hz)');
ylabel('amplitude');
axis([-200000 200000 0 0.3]);
e1=ifft(ifftshift(E1))*length(E1);
subplot(2,1,2),plot(t,(1/0.5)*e1);
```

```
title(' Recover signal IN Time DOMAIN after filtring');
xlabel('time(sec)');
ylabel('amplitude');
```

WAVEFORMS:

carrier signal of 154 khz

message signal of 15.4 khz

DSB-SC MODULATION IN TIME DAOMAIN

DSB SC MODULATION IN FREQUENCY DOMAIN

frequency response of low pass filter

Recover signal IN FREQUENCY DOMAIN after filtring

Recover signal IN Time DOMAIN after filtring

4. SINGLE SIDE BAND - SUPRESSED CARRIER MODULATION AND DEMODULATION

PROGRAM 1:

```
clc;
clear all;
close all;
Ts = 199;
subplot(4,1,1);
t = 0:1/Ts:1;
xm = cos(2*pi*1000*t);
plot(t,xm);
title('Modulating Signal ');
xlabel('Time');
ylabel('Amplitude');
grid on

% plot of the carrier signal

subplot(4,1,2);
c = cos(2*pi*5000*t);
plot(t,c);
title('Carrier Signal ');
xlabel('Time');
ylabel('Amplitude');
grid on

subplot(4,1,3);
% helbert transform of messeage signal
xh=cos((2*pi*1000*t)-pi/2);
% SSB UPPER SIDE BAND MODULATION
d= (xm.*cos(2*pi*5000*t))-(xh.*sin(2*pi*5000*t));
plot(t,d);
title('SINGLE SIDEBAND – SUPRESSED CARRIER  SIGNAL ');
xlabel('Time');
ylabel('Amplitude');
grid on

% freq. domain of the DSB signal.

subplot(4,1,4);
fre = abs(fft(d));
f = (0:length(fre) - 1)'*Ts/length(fre);
plot(f,fre);
axis([0 100 0 50]);
grid on
title('SINGLE SIDEBAND – SUPRESSED CARRIER  SPECTRUM ');
xlabel('Freq domain');
ylabel('Amplitude');
```

WAVEFORMS:
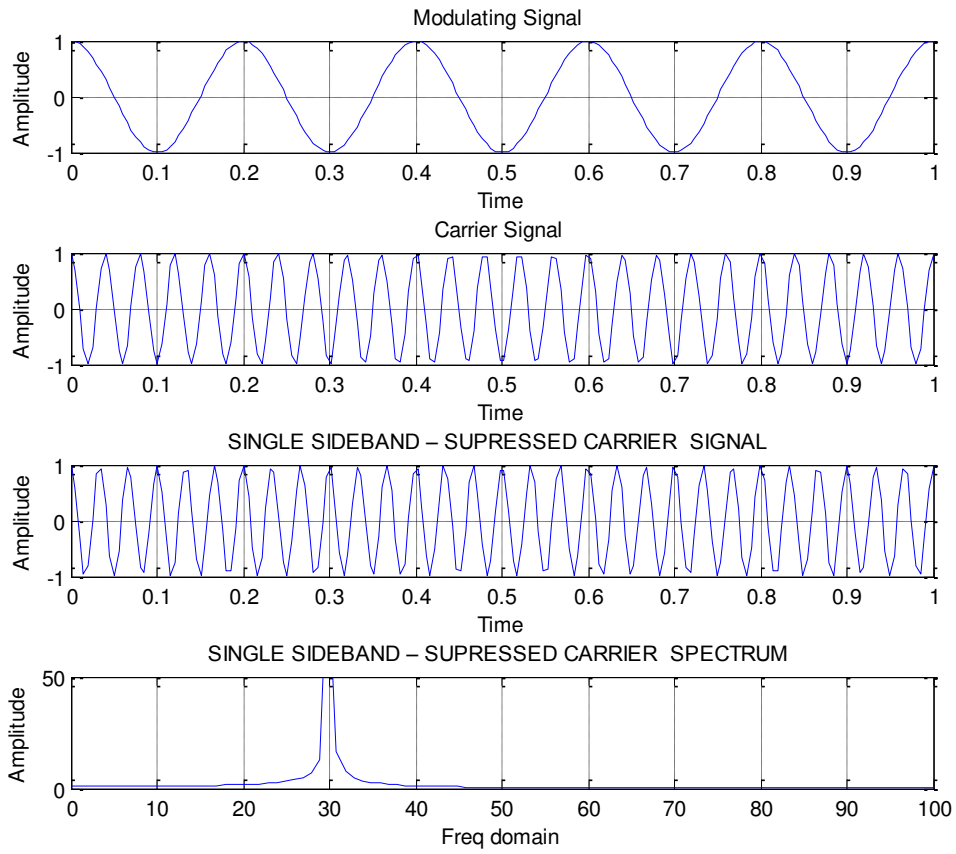
PROGRAM 2:

```
% task 1
fc=154000;

% task 2
fm=fc/10;
fs=100*fc;
t=0:1/fs:4/fm;
xc=cos(2*pi*fc*t);
xm=cos(2*pi*fm*t);

%task 4.1
xh=cos((2*pi*fm*t)-pi/2);% helbert transform which is the phase shift of
pi/2 in messeage signal
% SSB UPPER SIDE BAND MODULATION
z2= (xm.*cos(2*pi*fc*t))-(xh.*sin(2*pi*fc*t));
figure(5)
subplot(2,1,1),plot(t,z2);
title('SSB USB MODULATION IN TIME DAOMAIN');
xlabel('time (sec)');
ylabel('amplitude');
l2=length(z2);
f=linspace(-fs/2,fs/2,l2);
Z1=fftshift(fft(z2,l2)/l2);
subplot(2,1,2),plot(f,abs(Z1));
title('SSB USB MODULATION IN FREQUENCY DOMAIN');
xlabel('frequency(hz)');
ylabel('amplitude');
axis([-400000 400000 0 1]);
% apply local carrier for demodulation
z1= z2.*xc;
figure(6)

% task 4.2
subplot(2,1,1),plot(t,z1);
title('DEMODULATED SSB in time domain');
xlabel('time (sec)');
ylabel('amplitude');
l1=length(z1);
f=linspace(-fs/2,fs/2,l1);
Z1=fftshift(fft(z1,l1)/l1);
subplot(2,1,2),plot(f,abs(Z1));
title('DEMODULATED SSB IN FREQUENCY DOMIAN');
xlabel('frequency(hz)');
ylabel('amplitude');
axis([-400000 400000 0 0.3]);

% task 4.3
figure(7)
plot(f,abs(Z1));
title('FILTERING SIGNAL');
xlabel('frequency(hz)');
ylabel('amplitude');
hold on
Hlp=1./sqrt(1+(f./fc).^(2*100));
plot(f,Hlp,'g');
title(' frequency response of low pass filter');
xlabel('frequency(hz)');
ylabel('amplitude');
```

```
axis([-400000 400000 0 2]);

%task 4.4
E1=Hlp.*Z1;
figure(8)
subplot(2,1,1),plot(f,E1);
title(' Recover signal IN FREQUENCY DOMAIN after filtring');
xlabel('frequency(hz)');
ylabel('amplitude');
axis([-400000 400000 0 0.3]);
e1=ifft(ifftshift(E1))*length(E1);
subplot(2,1,2),plot(t,(1/0.5)*e1);
title(' Recover signal IN Time DOMAIN after filtring');
xlabel('time(sec)');
ylabel('amplitude');
```

WAVEFORMS :

DEMODULATED SSB in time domain

DEMODULATED SSB IN FREQUENCY DOMIAN

frequency response of low pass filter

Recover signal IN FREQUENCY DOMAIN after filtring

Recover signal IN Time DOMAIN after filtring
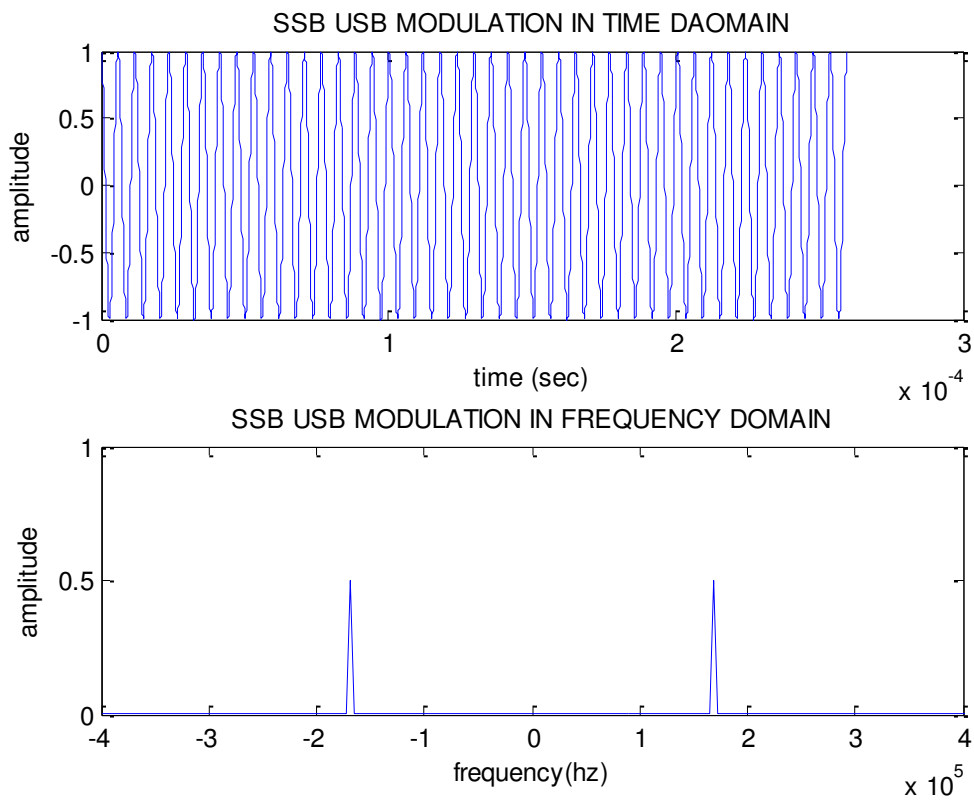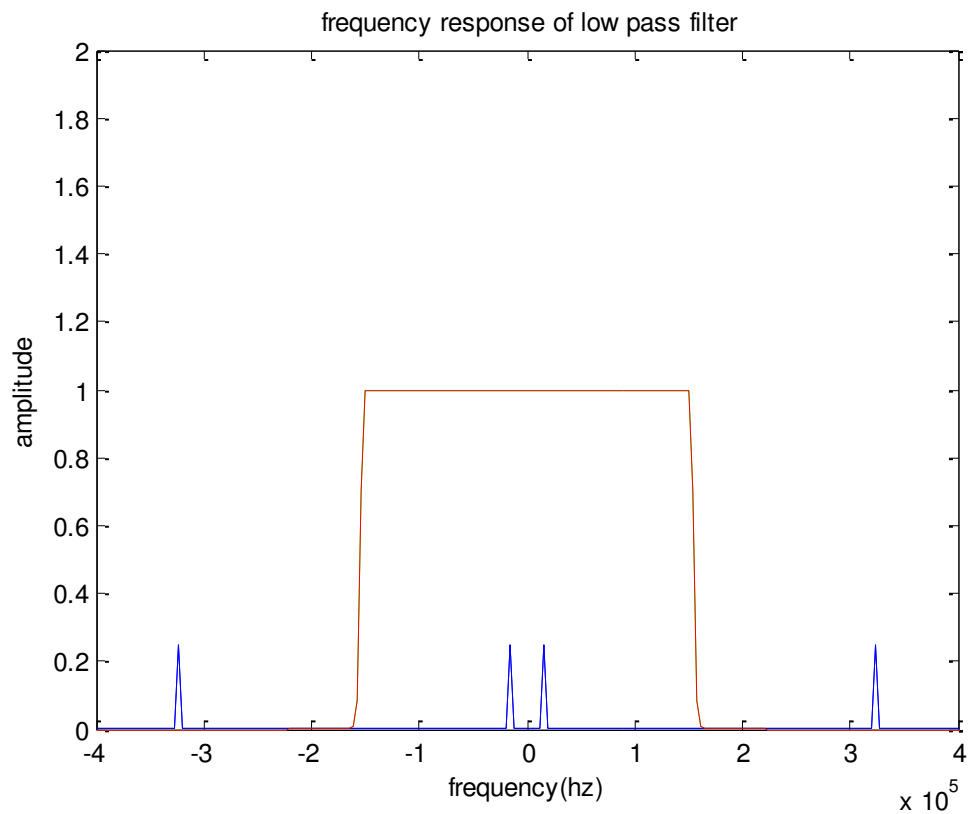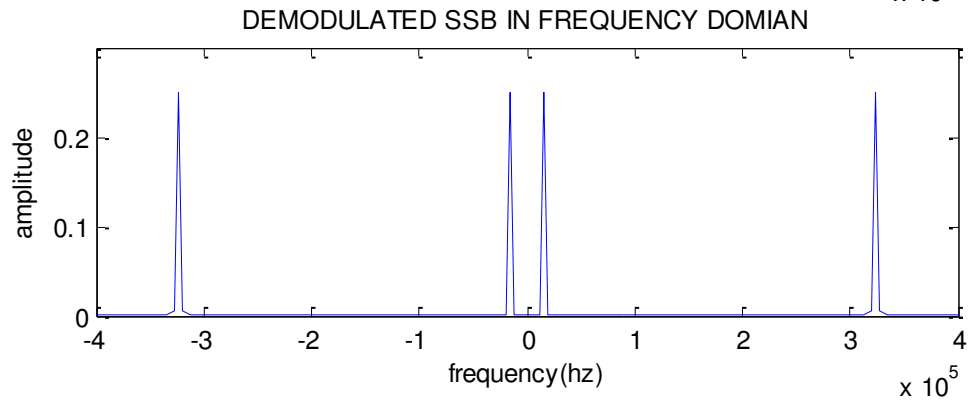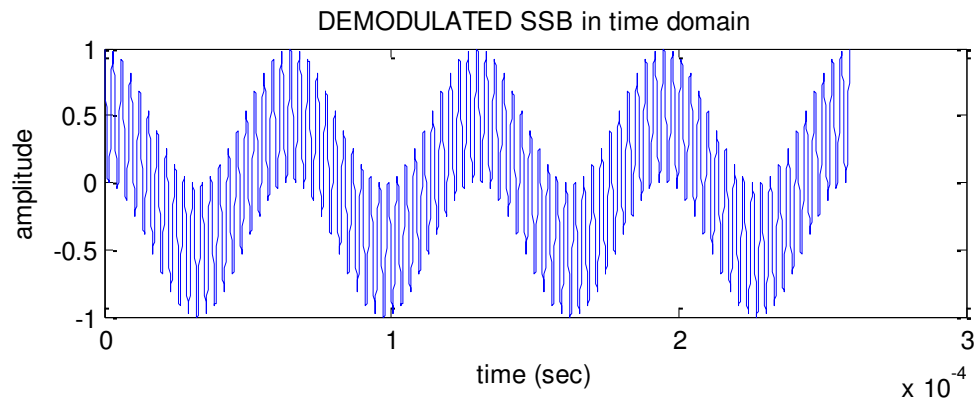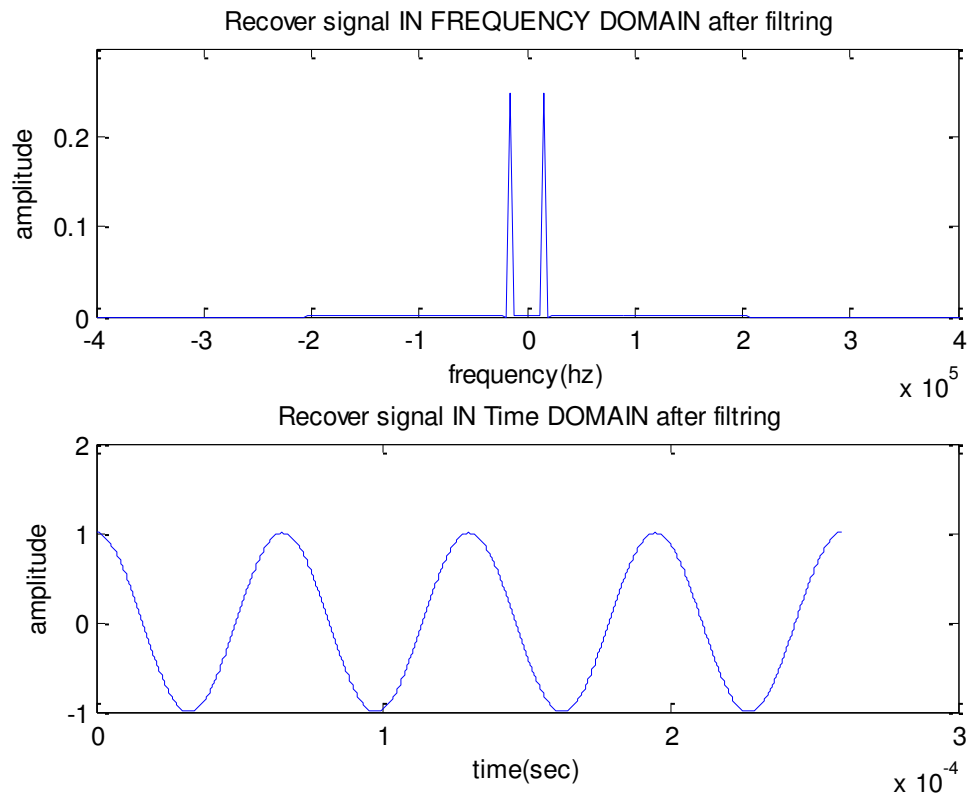
5. FREQUENCY MODULATION AND DEMODULATION

```matlab
clc;
clear all;
close all;

%% Frequency MODULATION of Triangular Signal
t = -0.04:1.e-4:0.04;
Ta = 0.01;
fc = 200;                               % Frequency of carrier
msg = msg1(t,Ta);        % Calling Function to Generate Msg Signal

%% MODULATION
kf = 160*pi;
m_int = kf*1.e-4*cumsum(msg);       % Integrating Msg
fm = cos(2*fc*pi*t + m_int);      % fm = cos(2*pi*fc*t + integral(msg))

%% DEMODULATION (Using Differentiator)
dem = diff(fm);
dem = [0,dem];
rect_dem = abs(dem);

%% Filtering out High Frequencies
N = 80;                                 % Order of Filter
Wn = 1.e-2;                             % Pass Band Edge Frequency.
a = fir1(N,Wn);          % Return Numerator of Low Pass FIR filter
b = 1;                       % Denominator of Low Pass FIR Filter
rec = filter(a,b,rect_dem);

%% Finding frequency Response of Signals

fl = length(t);
fl = 2^ceil(log2(fl));
f = (-fl/2:fl/2-1)/(fl*1.e-4);

mF = fftshift(fft(msg,fl));  % Frequency Response of Message Signal

fmF = fftshift(fft(fm,fl));  % Frequency Response of FM Signal

% Frequency Response of Rectified FM Signal
rect_demF = fftshift(fft(rect_dem,fl));

% Frequency Response of Recovered Message Signal
recF = fftshift(fft(rec,fl))

%% Plotting signal in time domain
figure;
subplot(2,1,1);
plot(t,msg);
title('Message Signal');
xlabel('{\it t} (sec)');
ylabel('Amplitude)');
grid;

subplot(2,1,2);
plot(t,fm);
```

```
title('FM');
xlabel('{\it t} (sec)');
ylabel('Amplitude');
grid;


figure;
subplot(2,1,1);
plot(t,rect_dem);
title('Rectified FM');
xlabel('{\it t} (sec)');
ylabel('Amplitude')
grid;


subplot(2,1,2);
plot(t,rec);
title('Recovered Signal');
xlabel('{\it t} (sec)');
ylabel('Amplitude');
grid;


%% Plotting Freq Response of Signals
figure;
subplot(2,2,1);
plot(f,abs(mF));
title('Freq Response of MSG');
xlabel('f(Hz)');
ylabel('Amplitude');
grid;
axis([-600 600 0 200]);


subplot(2,2,2);
plot(f,abs(fmF));
title('Freq Response of FM');
grid;
xlabel('f(Hz)');
ylabel('Amplitude');
axis([-600 600 0 300]);


subplot(2,2,3);
plot(f,abs(rect_demF));
title('Freq Response of Rectified FM');
xlabel('f(Hz)');
ylabel('Amplitude');
grid;
axis([-600 600 0 100]);


subplot(2,2,4);
plot(f,abs(recF));
title('Freq Response of Recovered Signal');
xlabel('f(Hz)');
ylabel('Amplitude');
grid;
axis([-600 600 0 100]);
```

Function to give message signal

```
% Function of Msg(tri)
function a = msg1(t,Ta)


t1 = -0.02:1.e-4:0;
```

97

```
t2 = 0:1.e-4:0.02;

m1 = 1 - abs((t1+Ta)/Ta);
m1 = [zeros([1 200]),m1,zeros([1 400])];
m2 = 1 - abs((t2-Ta)/Ta);
m2 = [zeros([1 400]),m2,zeros([1 200])];
a = m1 - m2;

end
```

WAVEFORMS :

Message Signal

FM

```
m1 = 1 - abs((t1+Ta)/Ta);
m2 = 1 - abs((t2-Ta)/Ta);
```
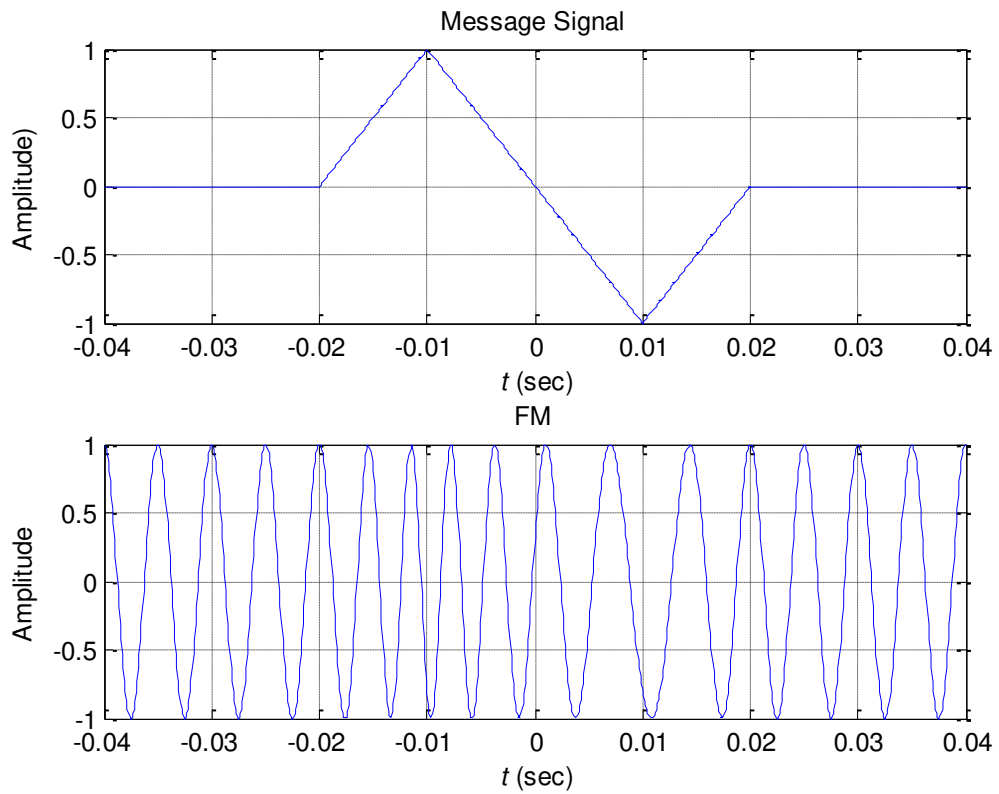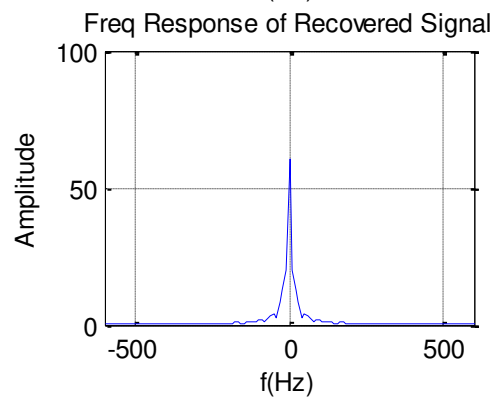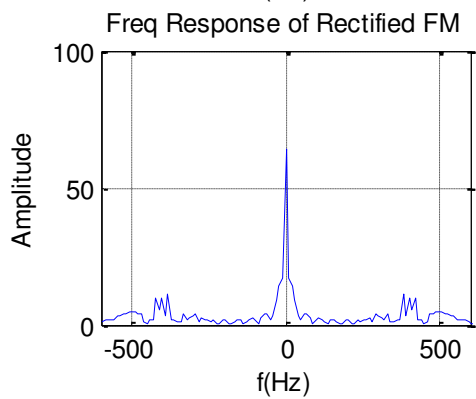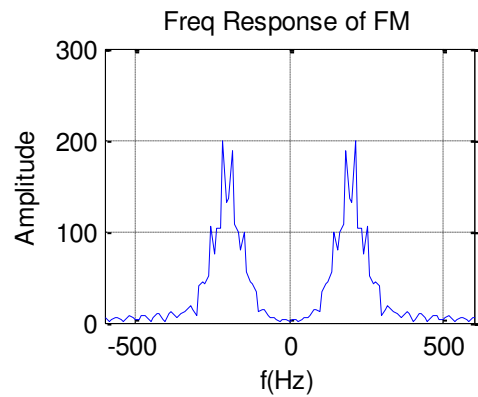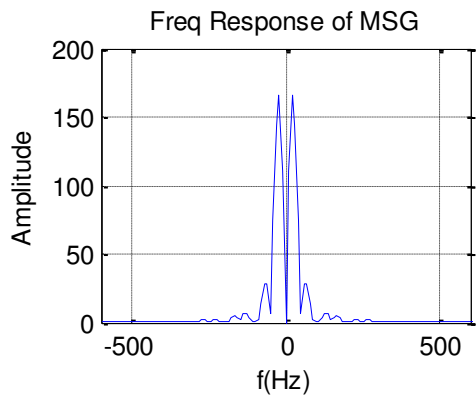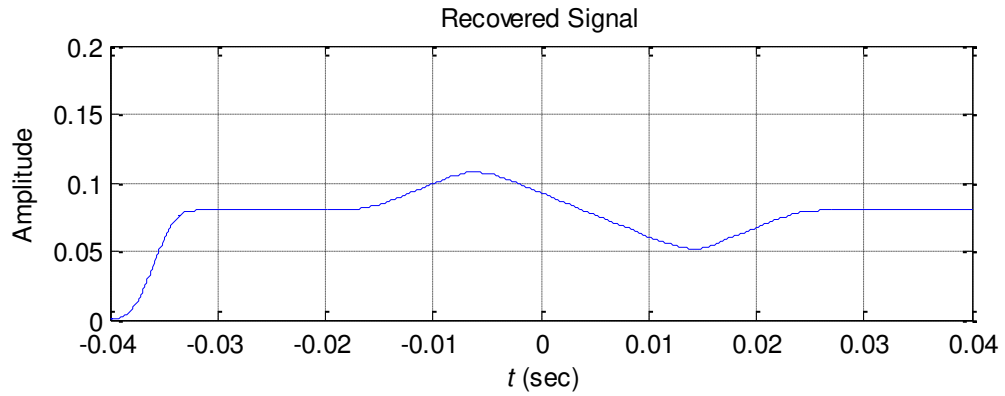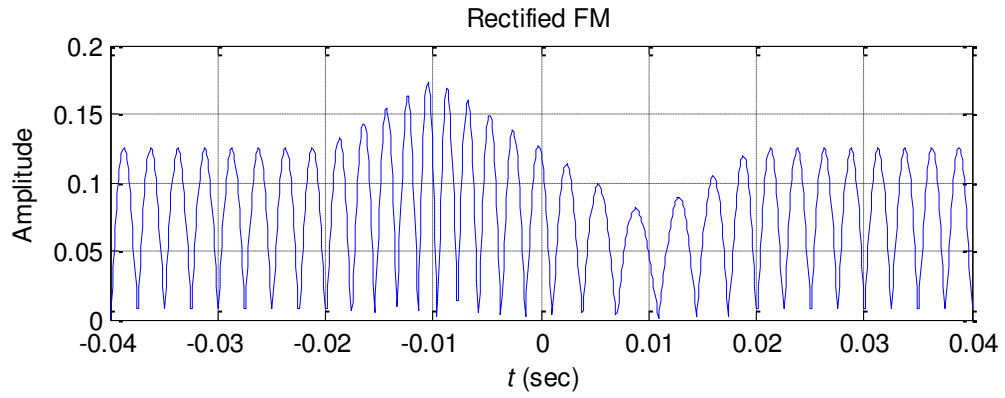
6. PHASE MODULATION AND DEMODULATION

```
clear all
close all
t = 0:0.01:1;                    % time variable
fc = 5;                          % carrier frequency

%------------------------------------------
% create message signal m(t)
m = sin(2*pi*t);
%------------------------------------------

kp  = pi/2;                      % phase deviation constant

%-------------------------------------------------
% modulating the carrier with the message signal
carrier = cos(2*pi*fc*t);
modulated = cos(2*pi*fc*t + kp*m);
%-------------------------------------------------

%-------------------------------------------------
% Plotting the signals
plot(t,m,'b',t,carrier,'r',t,modulated,'k--')
axis([0 1 -1.5 1.5]);
xlabel('Time(seconds)');
ylabel('Amplitude(volt)');
title('Phase modulation');
legend('Message','Carrier','Modulated');
```
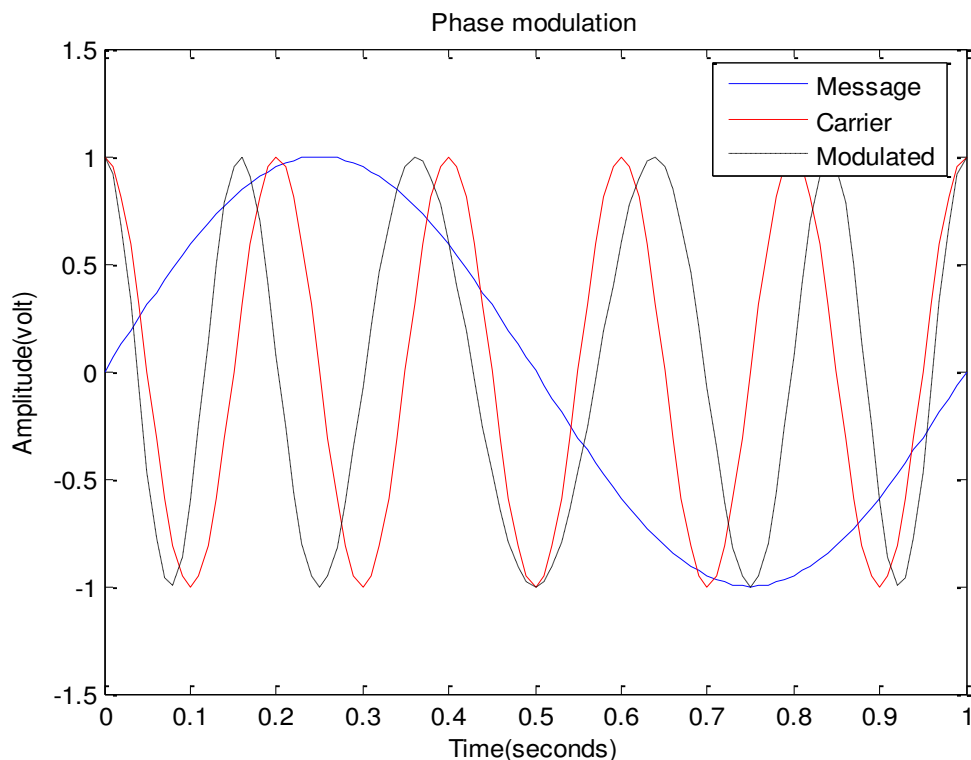
WAVEFORMS :

7. TIME DIVISION MULTIPLEXING AND DEMULTIPLEXING

```
clc;
close all;
clear all;
% Signal generation
x=0:.5:8*pi;
sig1=8*sin(x-pi);              % generate 1st sinusoidal signal
l=length(sig1);
sig2=8*square(x);              % Generate 2nd traingular Sigal

% Display of Both Signal
subplot(2,2,1);
plot(sig1);
title('Sinusoidal Signal');
ylabel('Amplitude--->');
xlabel('Time--->');
subplot(2,2,2);
plot(sig2);
title('Triangular Signal');
ylabel('Amplitude--->');
xlabel('Time--->');

% Display of Both Sampled Signal
subplot(2,2,3);
stem(sig1);
title('Sampled Sinusoidal Signal');
ylabel('Amplitude--->');
xlabel('Time--->');
subplot(2,2,4);
stem(sig2);
title('Sampled Triangular Signal');
ylabel('Amplitude--->');
xlabel('Time--->');
l1=length(sig1);
l2=length(sig2);
 for i=1:l1
  sig(1,i)=sig1(i);
  sig(2,i)=sig2(i);
 end

% TDM of both quantize signal
tdmsig=reshape(sig,1,2*l1);
% Display of TDM Signal
figure
stem(tdmsig);
title('TDM Signal');
ylabel('Amplitude--->');
xlabel('Time--->');

% Demultiplexing of TDM Signal
 demux=reshape(tdmsig,2,l1);
 for i=1:l1
  sig3(i)=demux(1,i);
  sig4(i)=demux(2,i);
 end


 % display of demultiplexed signal
 figure
 subplot(2,1,1)
```
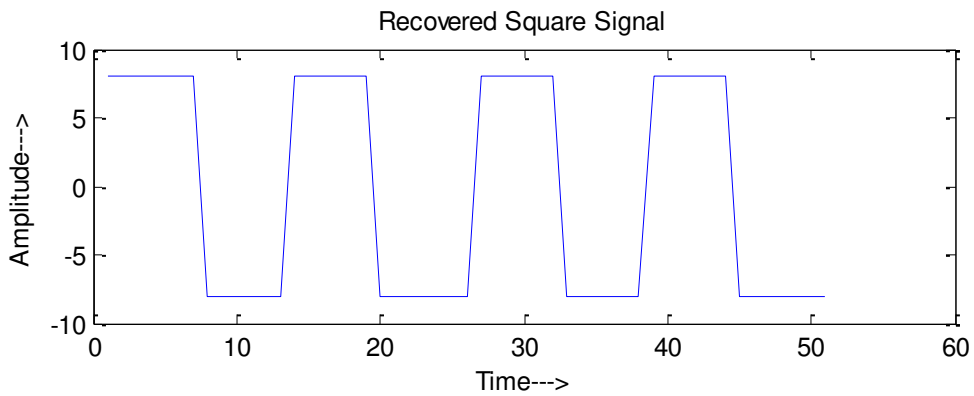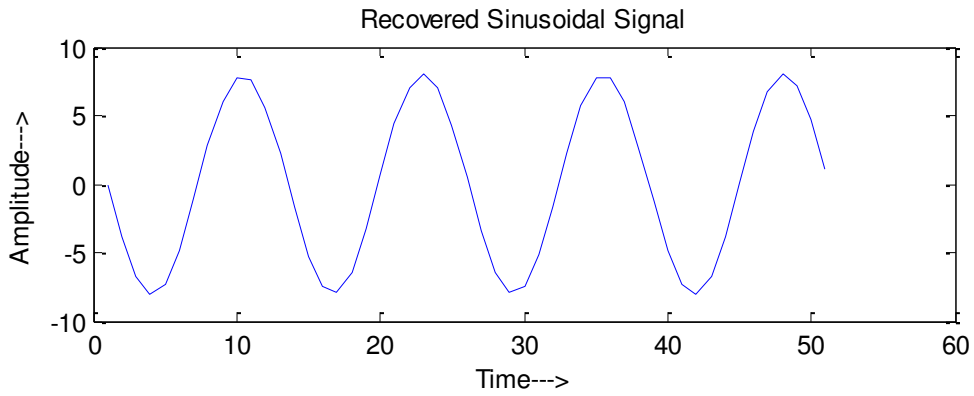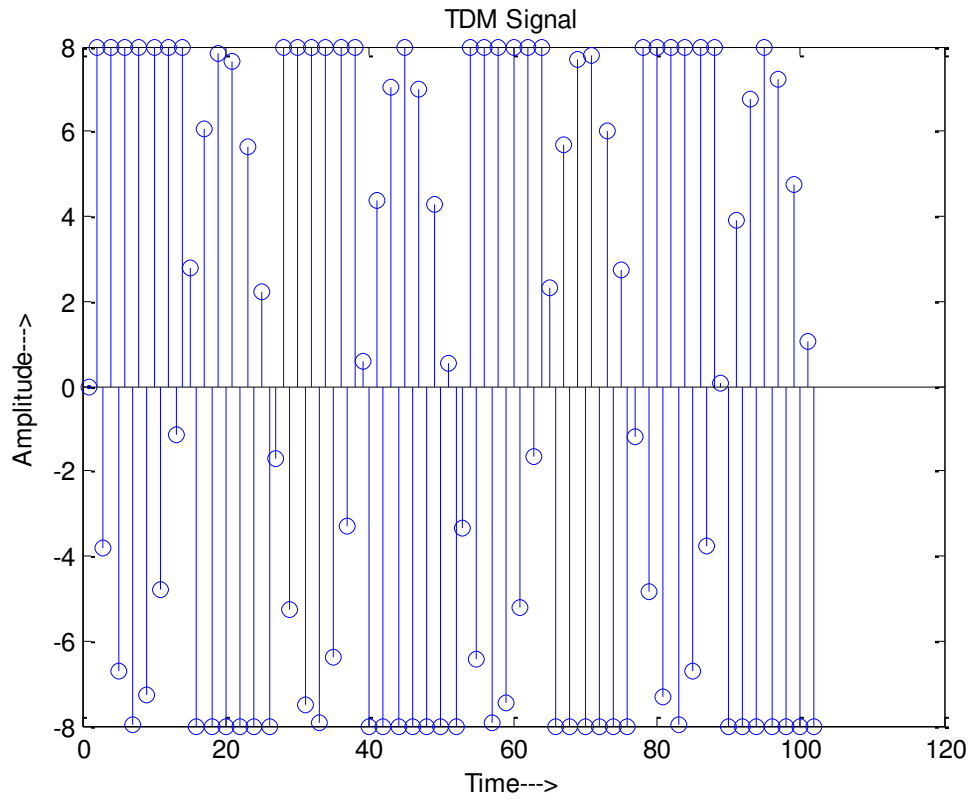
```
plot(sig3);
title('Recovered Sinusoidal Signal');
ylabel('Amplitude--->');
xlabel('Time--->');
subplot(2,1,2)
plot(sig4);
title('Recovered Triangular Signal');
ylabel('Amplitude--->');
xlabel('Time--->');
```

WAVEFORMS :

TDM Signal

Recovered Sinusoidal Signal

Recovered Square Signal
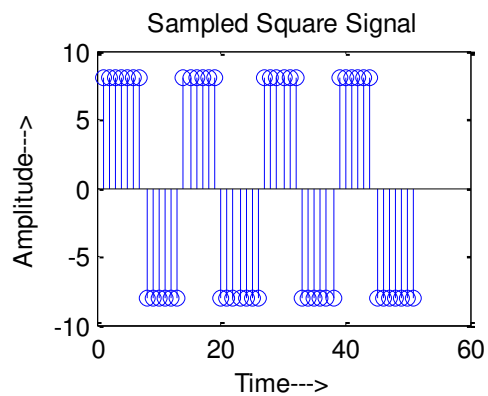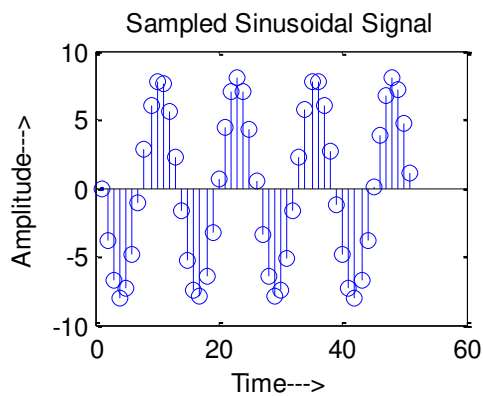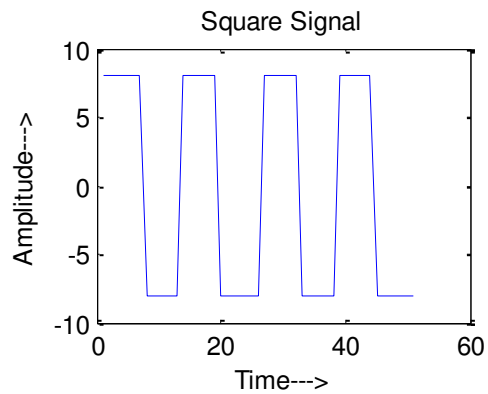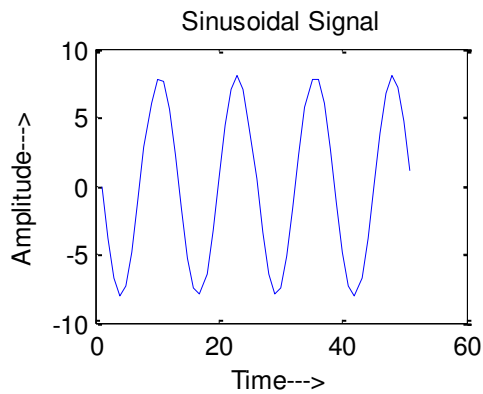
8. FREQUENCY DIVISION MULTIPLEXING AND DEMULTIPLEXING

9. VERIFICATION OF SAMPLING THEOREM

```
clear all;
close all;
clc;

Fs = input('Enter Sampling Frequency in Hz ....:: ');
Fm = input('Enter Message Frequency in Hz .....:: ');

t = 0:0.01:2;
msg = sin(2*pi*t);

k = 2*Fs/Fm;
step = ceil(201/k);
samp_msg = [];
for i = 1:201,
    if (mod(i,step) == 0)
        samp_msg = [samp_msg,msg(i)];
    else
        samp_msg = [samp_msg,zeros(1)];
    end;
end;

subplot(2,1,1);
plot(msg);
xlabel('---> Time');
ylabel('---> Amplitude');
title('Msg signal');

subplot(2,1,2);
stem(samp_msg);
xlabel('---> Time');
ylabel('---> Amplitude');
if Fs < 2*Fm
    title('Under Sampling');
else if Fs == 2*Fm
        title('Critical Sampling');
    else
        title('Over Sampling');
    end;
end;
```
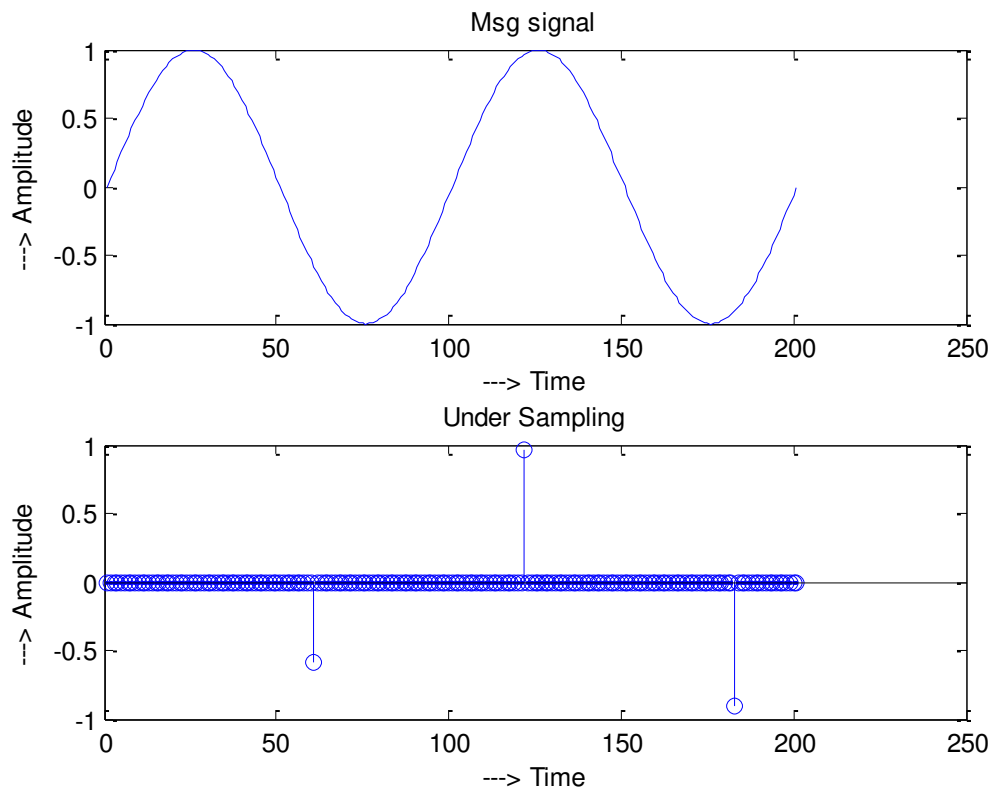
WAVEFORMS :

Enter Sampling Frequency in Hz ....:: 1200
Enter Message Frequency in Hz .....:: 60

Msg signal
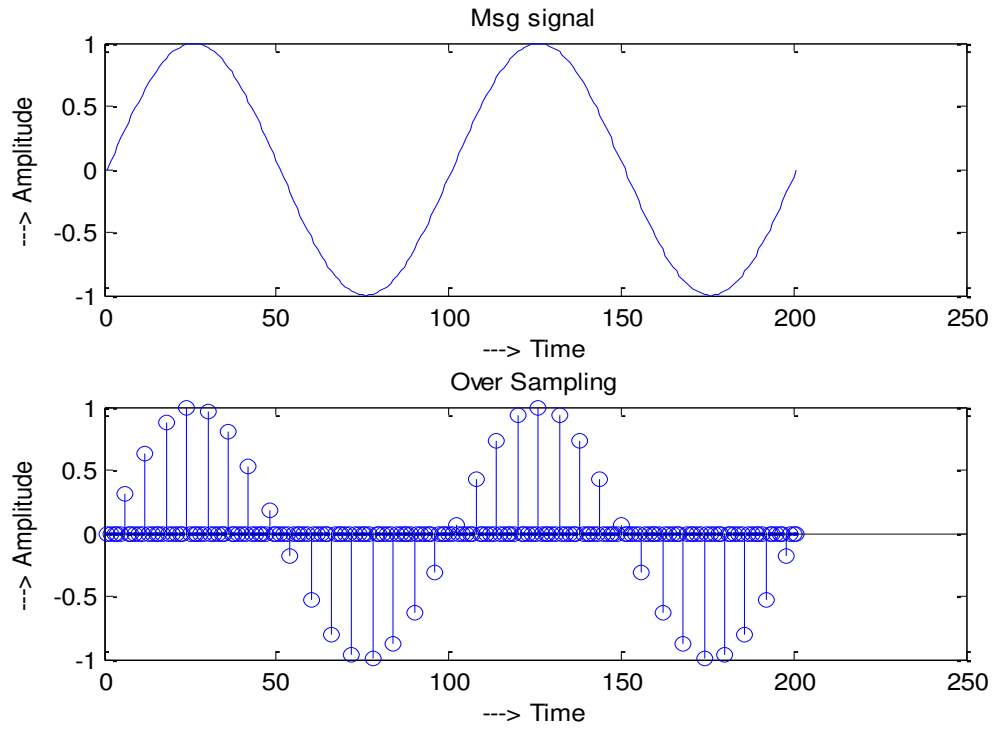
Over Sampling

Enter Sampling Frequency in Hz ....:: 100
Enter Message Frequency in Hz .....:: 60

Msg signal

Under Sampling

10. PULSE AMPLITUDE MODULATION AND DEMODULATION

```matlab
clear all;
close all;
clc;

t = 0.001:0.001:3;
msg = sin(2*pi*t);
subplot(3,1,1)
plot(msg);
title('Message Signal ');
xlabel('Time');
ylabel('Amplitude');
x = [ones(1,50),zeros(1,100)]
car  = [];
for i = 1:20,
    car = [car,x];
end
subplot(3,1,2);
plot(car);
title('Carrier Signal ');
xlabel('Time');
ylabel('Amplitude');
mm = msg.*car;
tt = [];
for i =1:3000
    if mm(i) == 0;
        tt = [tt,mm(i)];
    else
        tt = [tt,mm(i)+2];
    end
end
subplot(3,1,3);
stem(tt,'.r');
title('PAM signal ');
xlabel('Time');
ylabel('Amplitude');
```
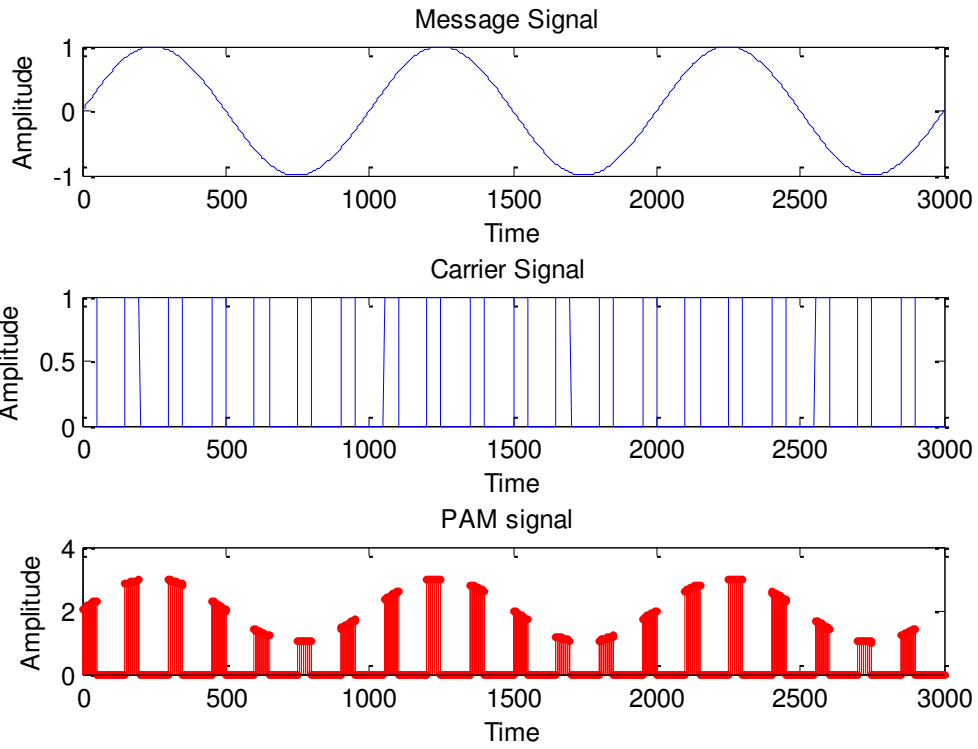
WAVEFORMS :

Message Signal

Carrier Signal

PAM signal

11. PULSE WIDTH MODULATION AND DEMODULATION

```
fc = 1000;
fs = 10000;
f1 = 200;
f2 = 300;
t = 0:1/fs:((2/f1)-(1/fs));
x1 = 0.4*cos(2*pi*f1*t)+0.5;
%x2 = 0.2*(cos(2*pi*f1*t)+cos(2*pi*f2*t))+0.5;


%%%%%%%%%%%%%%%%%%%MODULATION%%%%%%%%%%%%%%%%%%%%%%%%%%%%%


y1 = modulate(x1,fc,fs,'pwm');
subplot(2,1,1); plot(x1);
title('Original single tone message f1 = 200 , fs = 10000');
subplot(2,1,2); plot(y1);
title('PWM on cycle of f1 = 200 , fc = 1000');
axis([0,500,-0.2,1.2]);
figure
fx1 = abs(fft(y1,1024));
fx1 = [fx1(514:1024) fx1(1:513)];
f = (-511*fs/1024):(fs/1024):(512*fs/1024);
subplot(2,1,1); plot(f,fx1);
title('Frequency description single tone f1 = 200 , fc = 1000');


%%%%%%%%%%%%% DEMODULATION%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%


x1_recov = demod(y1,fc,fs,'pwm');
subplot(2,1,2); plot(x1_recov );
title('Time domain recovered single tone f1 = 200 ');
```
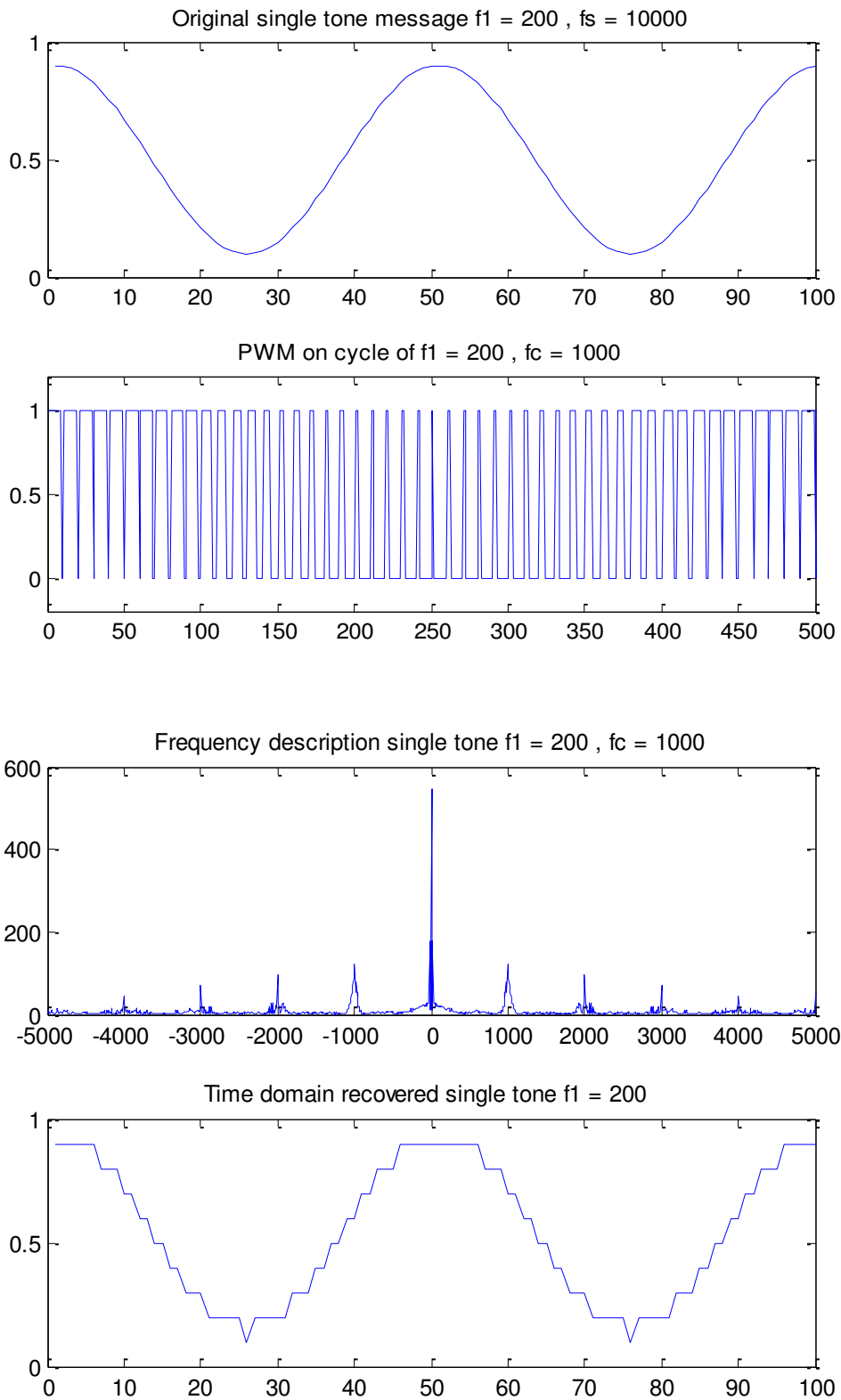
WAVEFORMS :

Original single tone message f1 = 200 , fs = 10000

PWM on cycle of f1 = 200 , fc = 1000

Frequency description single tone f1 = 200 , fc = 1000

Time domain recovered single tone f1 = 200

12. PULSE POSITION MODULATION AND DEMODULATION

```
fc = 1000;
fs = 10000;
f1 = 200;
f2 = 300;
t = 0:1/fs:((2/f1)-(1/fs));
x1 = 0.4*cos(2*pi*f1*t)+0.5;
%x2 = 0.2*(cos(2*pi*f1*t)+cos(2*pi*f2*t))+0.5;


%%%%%%%%%%%%%%%%%%MODULATION%%%%%%%%%%%%%%%%%%%%%%%%%%%%%


y1 = modulate(x1,fc,fs,'ppm');
subplot(2,1,1); plot(x1);
title('Original single tone message f1 = 200 , fs = 10000');
subplot(2,1,2); plot(y1);
title('PPM on cycle of f1 = 200 , fc = 1000');
axis([0,500,-0.2,1.2]);
figure
fx1 = abs(fft(y1,1024));
fx1 = [fx1(514:1024) fx1(1:513)];
f = (-511*fs/1024):(fs/1024):(512*fs/1024);
subplot(2,1,1); plot(f,fx1);
title('Frequency description single tone f1 = 200 , fc = 1000');


%%%%%%%%%%%%% DEMODULATION%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%


x1_recov = demod(y1,fc,fs,'ppm');
subplot(2,1,2); plot(x1_recov );
title('Time domain recovered single tone f1 = 200 ');
```
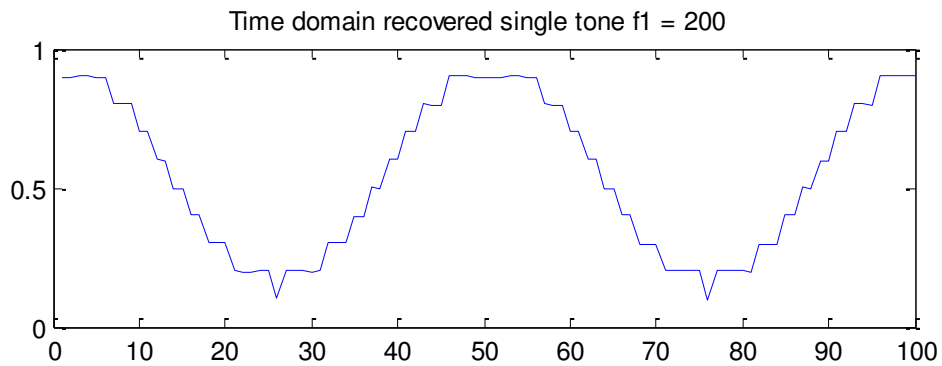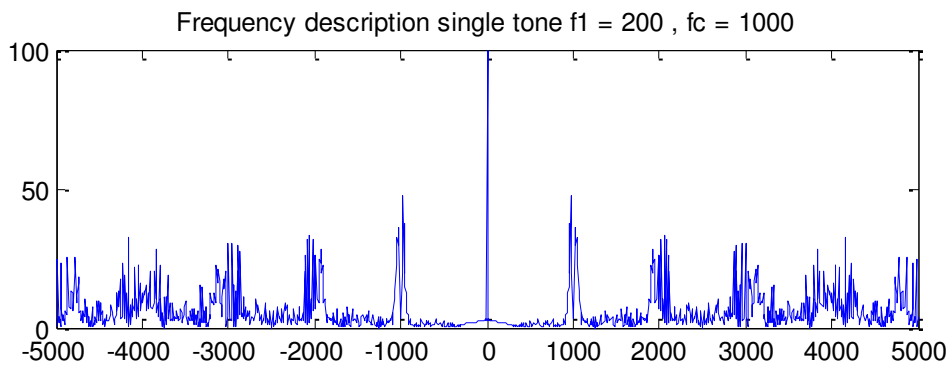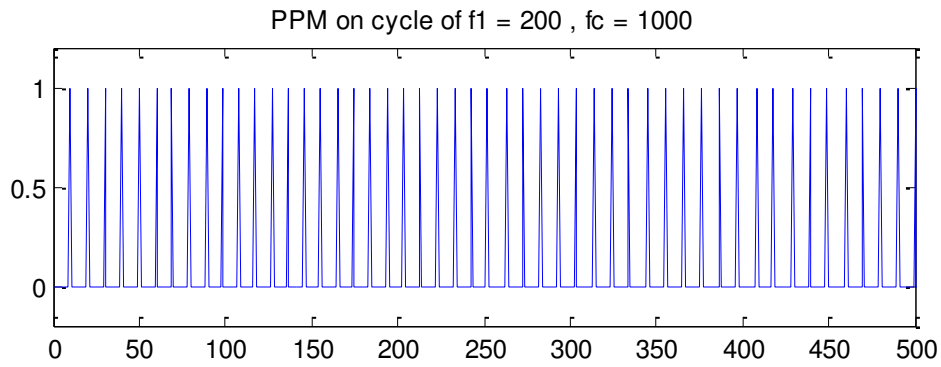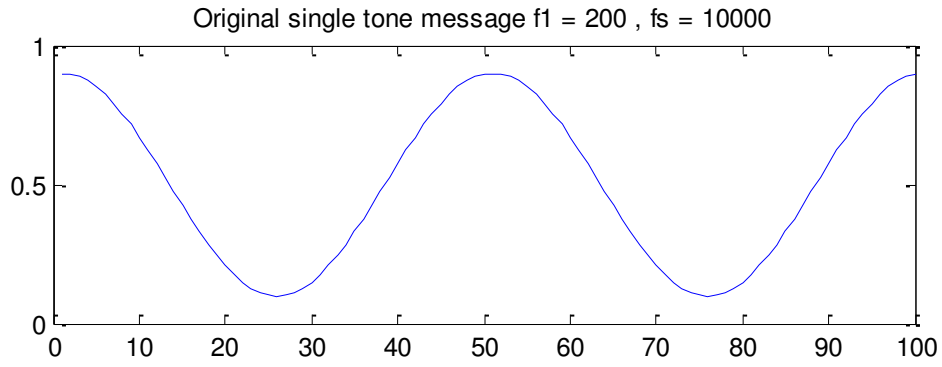
 WAVEFORMS :

Original single tone message f1 = 200 , fs = 10000

PPM on cycle of f1 = 200 , fc = 1000

Frequency description single tone f1 = 200 , fc = 1000

Time domain recovered single tone f1 = 200

13. PHASE LOCKED LOOP AS FM DEMODULATOR

```
%This m-file demonstrates a PLL which tracks and demodulates an FM carrier.

clear all;
close all;
f=1000;          %Carrier frequency
fs=100000;       %Sample frequency
N=5000;          %Number of samples
Ts=1/fs;
t=(0:Ts:(N*Ts)- Ts);
     %Create the message signal
f1=100;          %Modulating frequency
msg=sin(2*pi*f1*t);
kf=.0628;        %Modulation index
                         %Create the real and imaginary parts of a CW %
                         modulated carrier to be tracked.
Signal=exp(j*(2*pi*f*t+2*pi*kf*cumsum(msg)));   %Modulated carrier
Signal1=exp(j*(2*pi*f*t));               %Unmodulated carrier
     %Initilize PLL Loop
phi_hat(1)=30;
e(1)=0;
phd_output(1)=0;
vco(1)=0;
     %Define Loop Filter parameters(Sets damping)
kp=0.15;                  %Proportional constant
ki=0.1;                   %Integrator constant
     %PLL implementation
for n=2:length(Signal)
vco(n)=conj(exp(j*(2*pi*n*f/fs+phi_hat(n-1)))); %Compute VCO
phd_output(n)=imag(Signal(n)*vco(n));      %Complex multiply VCO x %Signal
                                           input
                                 %Filter integrator
e(n)=e(n-1)+(kp+ki)*phd_output(n)-ki*phd_output(n-1); phi_hat(n)=phi_hat(n-
1)+e(n);%Update VCO
end;
          %Plot waveforms
startplot = 1;
endplot = 1000;

figure(1);
subplot(3,2,1);
plot(t(startplot:endplot), msg(startplot:endplot));
title('100 Hz message signal');
ylabel('Amplitude');
grid;

figure(1);
subplot(3,2,2);
plot(t(startplot:endplot), real(Signal(startplot:endplot)));
title('FM (1KHz carrier modulated with a 100 Hz message signal)');
ylabel('Amplitude');
grid;

figure(1)
subplot(3,2,3);
plot(t(startplot:endplot), e(startplot:endplot));
title('PLL Loop Filter/Integrator Output');
```

```
ylabel('Amplitude');
grid;

subplot(3,2,4);
plot(t(startplot:endplot), real(vco(startplot:endplot)));
title('VCO Output (PLL tracking the input signal)');
ylabel('Amplitude');
grid;

subplot(3,2,5);
plot(t(startplot:endplot), phd_output(startplot:endplot));
title('Phase Detector Output');
xlabel('Time (seconds)');
ylabel('Amplitude');
grid;

subplot(3,2,6);
plot(t(startplot:endplot), real(Signal1(startplot:endplot)));
title('Unmodulated Carrier');
xlabel('Time (seconds)');
ylabel('Amplitude');
grid;
```

WAVEFORMS :