SRM VALLIAMMAI ENGINEERING COLLEGE

(An Autonomous Institution)

S.R.M Nagar, Kattankulathur – 603 203

Department of Electronics and Communication Engineering

Laboratory Manual

EC3567 – ANALOG AND DIGITAL COMMUNIATION LABORATORY

Regulation - 2023

Semester/Branch	:	**V semester ECE**
Academic Year	:	**2025 - 2026 (ODD SEMESTER)**
Prepared By	:	**Dr. R. Dhananjeyan,** *Assistant Professor* / ECE
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		Dr. K. Lekha, *Assistant Professor* / ECE

VISION OF THE INSTITUTE

Educate to excel in social transformation

MISSION OF THE INSTITUTE

- To contribute to the development of human resources in the form of professional engineers and managers of international excellence and competence with high motivation and dynamism, who besides serving as ideal citizen of our country will contribute substantially to the economic development and advancement in their chosen areas of specialization.
- To build the institution with international repute in education in several areas at several levels with specific emphasis to promote higher education and research through strong institute-industry interaction and consultancy.

VISION OF THE DEPARTMENT

To excel in the field of electronics and communication engineering and to develop highly competent technocrats with global intellectual qualities.

MISSION OF THE DEPARTMENT

M1: To train the students with the state-of-art technologies and to develop innovative solutions to cater to the societal needs.

M2: To orient the students towards global career with universal moral values and professional ethics.

PROGRAM OUTCOMES

1. Engineering knowledge: Apply the knowledge of mathematics, science, engineering fundamentals, and an engineering specialization to the solution of complex engineering problems.

2. Problem analysis: Identify, formulate, review research literature, and analyze complex engineering problems reaching substantiated conclusions using first principles of mathematics, natural sciences, and engineering sciences.

3. Design/development of solutions: Design solutions for complex engineering problems and design system components or processes that meet the specified needs with appropriate consideration for the public health and safety, and the cultural, societal, and environmental considerations.

4. Conduct investigations of complex problems: Use research-based knowledge and research methods including design of experiments, analysis and interpretation of data, and synthesis of the information to provide valid conclusions.

5. Modern tool usage: Create, select, and apply appropriate techniques, resources, and modern engineering and IT tools including prediction and modelling to complex engineering activities with an understanding of the limitations.

6. The engineer and society: Apply reasoning informed by the contextual knowledge to assess societal, health, safety, legal and cultural issues and the consequent responsibilities relevant to the professional engineering practice.

7. Environment and sustainability: Understand the impact of the professional engineering solutions in societal and environmental contexts, and demonstrate the knowledge of, and need for sustainable development.

8. Ethics: Apply ethical principles and commit to professional ethics and responsibilities and norms of the engineering practice.

9. Individual and team work: Function effectively as an individual, and as a member or leader in diverse teams, and in multidisciplinary settings.

10. Communication: Communicate effectively on complex engineering activities with the engineering community and with society at large, such as, being able to comprehend and write effective reports and design documentation, make effective presentations, and give and receive clear instructions.

11. Project management and finance: Demonstrate knowledge and understanding of the engineering and management principles and apply these to one's own work, as a member and leader in a team, to manage projects and in multidisciplinary environments.

12. Life-long learning: Recognize the need for, and have the preparation and ability to engage in independent and life-long learning in the broadest context of technological change.

PROGRAM SPECIFIC OUTCOME (PSOs)

1. Ability to solve societal problems by applying knowledge acquired in the core domain.
2. Ability to apply technologies like VLSI, Embedded and Wireless Communications to meet the industrial requirements.

Syllabus

EC3567 ANALOG AND DIGITAL COMMUNICATION LABORATORY L T P C

0 0 3 1.5

OBJECTIVES:

- To study the AM & FM Modulation and Demodulation.
- To learn and realize the effects of sampling and TDM.
- To understand the pulse modulation techniques.
- To analyze Digital Modulation Schemes.
- To explore the detection techniques in digital communication.
- To implement channel coding and error control coding Schemes in the communication system.

LIST OF EXPERIMENTS

1. AM- Modulator and Demodulator
2. FM - Modulator and Demodulator
3. Signal sampling and TDM.
4. Pulse Code Modulation and Demodulation.
5. Pulse Modulation and Demodulation- PAM, PPM, PWM
6. Digital Modulation – ASK, PSK, FSK.
7. Delta Modulation and Demodulation.
8. Simulation of ASK, FSK, and BPSK Generation and Detection Schemes.
9. Simulation of QPSK and QAM Generation and Detection Schemes.
10. Simulation of Linear Block and Cyclic Error Control coding Schemes.

TOPIC BEYOND THE SYLLABUS EXPERIMENT

1. Simulation of Differential Phase Shift Keying.

Total: 45 Periods

OUTCOMES:

On completion of this lab course, the student would be able to

- Design AM, FM & Digital Modulators for specific applications.
- Acquire knowledge in sampling, time division multiplexing and pulse modulation techniques.
- Simulate & analyze the various functional modules of Communication system.
- Demonstrate their knowledge in base band signaling schemes through implementation of digital modulation schemes.
- Understand the significance of demodulation techniques in various digital communication schemes.
- Apply various channel coding & error control coding schemes to demonstrate their capabilities towards the improvement of the noise performance of Communication system.

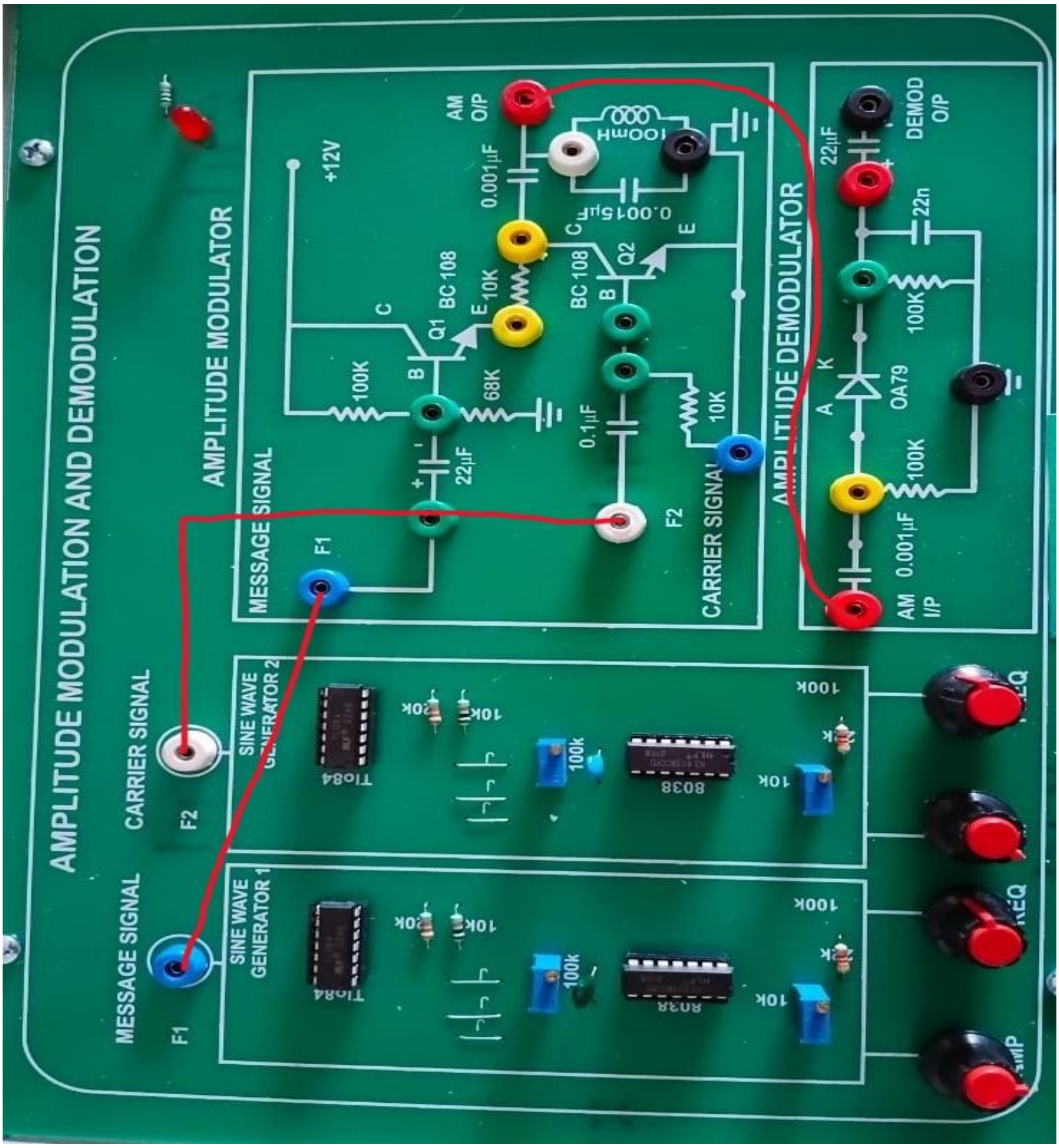
COURSE OUTCOMES - PROGRAM OUTCOMES MATRIX

CO	PO												PSO	
	1	2	3	4	5	6	7	8	9	10	11	12	1	2
EC3567.1	3	3	3	3	2	1	2	-	3	2	1	1	2	-
EC3567.2	3	3	3	3	2	1	2	-	3	2	1	1	3	-
EC3567.3	3	3	3	3	3	1	2	-	3	2	1	1	2	2
EC3567.4	3	3	3	3	3	1	2	-	3	2	1	1	3	3
EC3567.5	3	3	3	3	2	1	2	-	3	2	1	1	2	2
EC3567.6	3	3	3	3	2	1	2	-	3	2	1	1	3	2
AVG	3	3	3	3	2	1	2	-	3	2	1	1	3	2

LAB Requirements for a Batch of 30 students (3 students per experiment):

1. Kits for Signal Sampling & TDM, AM, FM, PCM and PM.
2. CROs/DSOs – 15 Nos, Function Generators – 15 Nos.
3. MATLAB or equivalent software package for simulation experiments
4. PCs - 15 Nos

List of Experiments



OBSERVATION:

Modulation	Message Signal	Carrier Signal	Vmax (Volts)	Vmin (Volts)	Modulation Index $\frac{V_{max} - V_{min}}{V_{max} + V_{min}}$
Under Modulation	Amplitude : Timeperiod :	Amplitude : Timeperiod :			
Critical Modulation	Amplitude : Timeperiod :	Amplitude : Timeperiod :			
Over Modulation	Amplitude : Timeperiod :	Amplitude : Timeperiod :			

Ex. No. 1

AMPLITUDE MODULATOR AND DEMODULATOR

AIM :

To study and understand amplitude modulation & demodulation circuit and to calculate modulation index for various modulating voltages.

APPARATUS REQUIRED:

1. AM trainer kit
2. CRO
3. Patch chords

PRE LAB QUESTIONS:

1. What is meant by modulation?
2. Summarize Amplitude modulation and Demodulation.
3. Define modulation index.
4. What is the difference between DSB, SSB, and conventional AM?
5. What are the components of an AM signal?

THEORY:

Amplitude Modulation is a process by which amplitude of the carrier signal is varied in accordance with the instantaneous value of the modulating signal, but frequency and phase of carrier wave remains constant.

The modulating and carrier signal are given by

$$V_m(t) = V_m \sin \omega_m t$$

$$V_C(t) = V_C \sin \omega_C t$$

The modulation index is given by, $m_a = V_m / V_C$.

$$V_m = V_{max} - V_{min} \text{ and } V_C = V_{max} + V_{min}$$

The amplitude of the modulated signal is given by,

$$V_{AM}(t) = V_C (1+m_a \sin \omega_m t) \sin \omega_C t$$

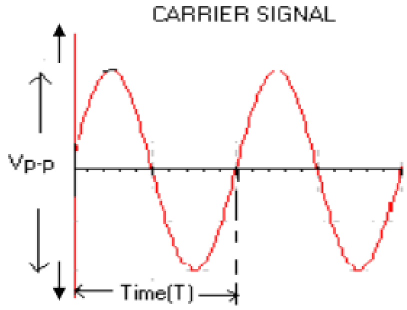
Where

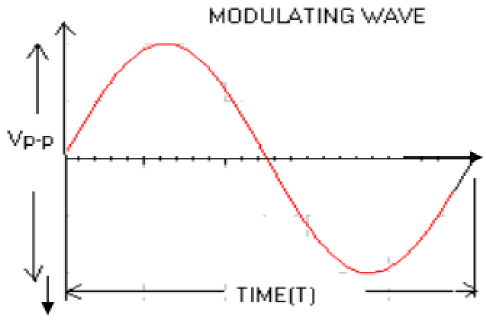
V_m = maximum amplitude of modulating signal
V_C = maximum amplitude of carrier signal

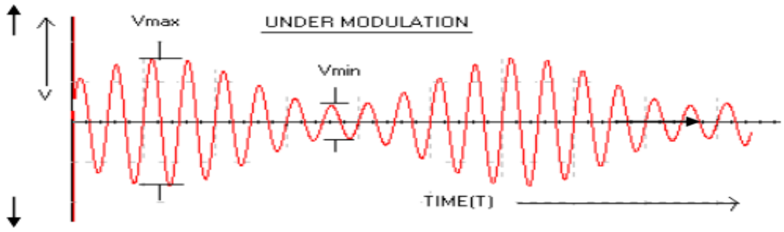
V_{max} = maximum variation of AM signal

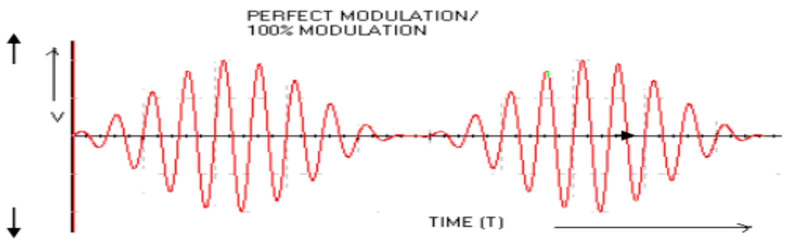
V_{min} = minimum variation of AM signal

MODEL GRAPH:

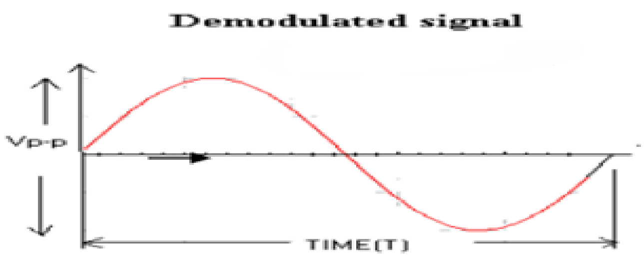

CARRIER SIGNAL
Vp-p
Time(T)


MODULATING WAVE
Vp-p
TIME(T)


UNDER MODULATION
Vmax
Vmin
TIME(T)


PERFECT MODULATION/
100% MODULATION
TIME (T)


OVER MODULATION
TIME(T)


Demodulated signal
Vp-p
TIME(T)

PROCEDURE:

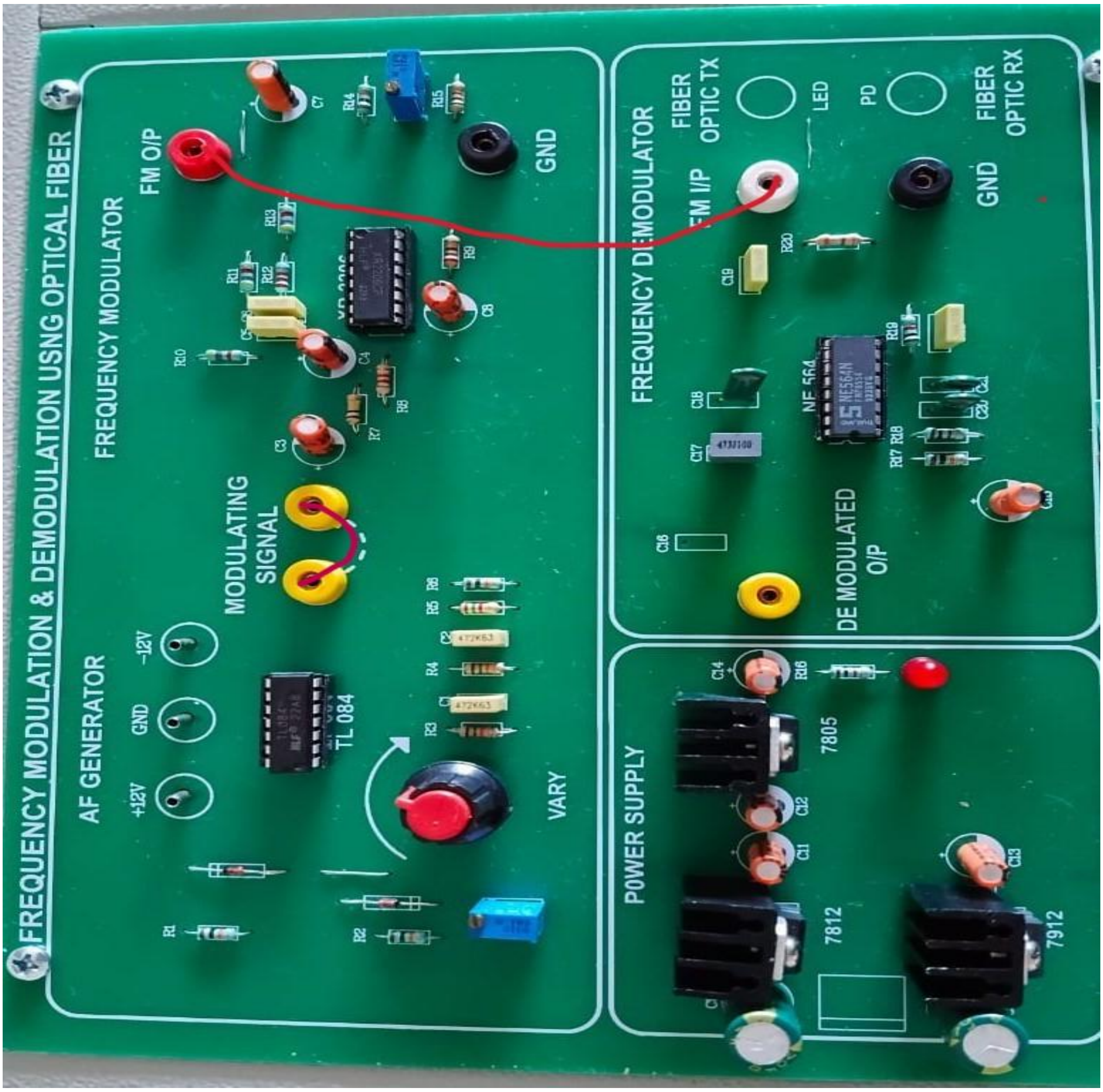
1. The circuit wiring is done as shown in diagram
2. Measure the time period and the amplitude of the carrier signal at the point F2 by adjusting the AMP and FREQ knob in that block.
3. Set the frequency of the message signal around 1kHz using the FREQ knob below the F1 block
4. To measure the under modulation , set the amplitude of the message signal less than the amplitude of the carrier signal using the AMP knob under the F1 block.
5. To measure the critical modulation , set the amplitude of the message signal equal to the amplitude of the carrier signal using the AMP knob under the F1 block.
6. To measure the under modulation , set the amplitude of the message signal greater than the amplitude of the carrier signal using the AMP knob under the F1 block.
7. To observe the amplitude modulated signal connect F1 to F1 in amplitude modulator block and F2 to F2 in amplitude modulator block and AM signal at AM O/P.
8. amplitude modulator stage and the amplitude and time duration of the AM wave are noted down.
9. Calculate the modulation index by using the formula and tabulate the observed values.
10. To perform demodulation connect AM O/P in the amplitude modulator block to the AM I/P in the amplitude demodulator block, note the amplitude and time period of the demodulated signal.

POST LAB QUESTIONS:

1. How the modulation index was observed?
2. As related to AM, what is over modulation, under modulation and 100%modulation?
3. What type of AM signals was generated in the experiment?
4. What distortions were noticed in the demodulated AM signal?
5. How did carrier frequency affect the modulated signal?

RESULT:

Thus the amplitude modulation and demodulation was observed and the modulation index was calculated for various modulating voltages.



OBSERVATION:

SIGNAL	AMPLITUDE	TIME PERIOD
Input Signal		
Carrier Signal		
Fm Signal		
Demodulated Signal		

Ex. No. 2

FREQUENCY MODULATOR AND DEMODULATOR

AIM

To study and understand frequency modulation & demodulation characteristics and to calculate modulation index for the frequency modulated signal.

APPARATUS REQUIRED:

1. FM trainer kit
2. CRO
3. Patch chords

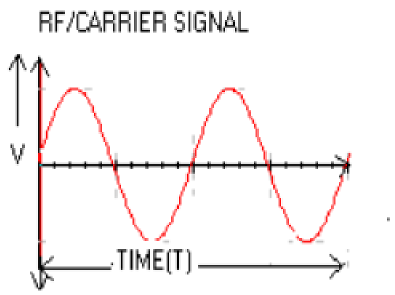
PRE LAB QUESTIONS:

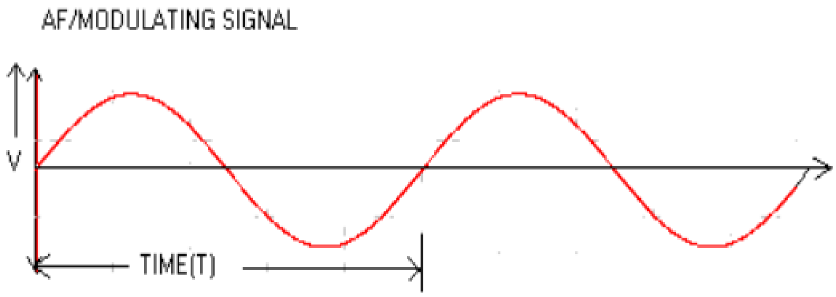
1. What is FM?
2. Define frequency deviation in FM.
3. What are the advantages of FM?
4. Define modulation index for FM.
5. Why is FM considered more noise-immune than AM?

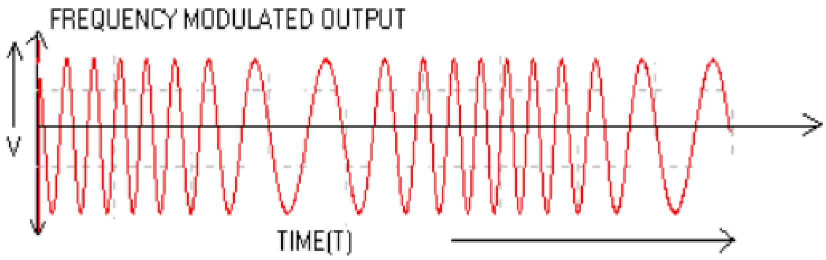
THEORY:

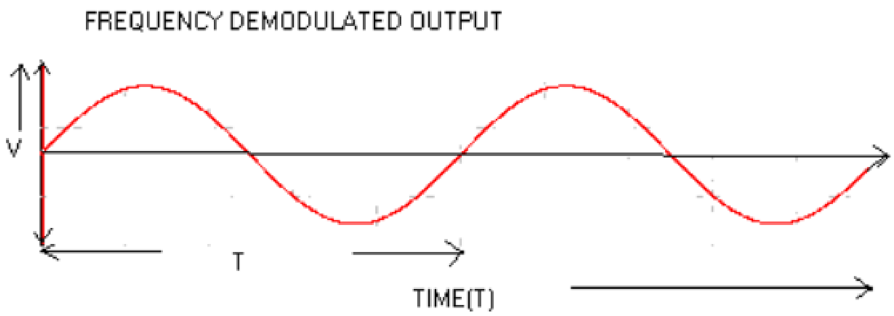
Frequency modulation (FM) is a form of modulation that represents information as variations in the instantaneous frequency of a carrier wave. (Contrast this with amplitude modulation, in which the amplitude of the carrier is varied while its frequency remains constant.) In analog applications, the carrier frequency is varied in direct proportion to changes in the amplitude of an input signal. Shifting the carrier frequency among a set of discrete values can represent digital data, a technique known as frequency-shift keying. FM is commonly used at VHF radio frequencies for high-fidelity broadcasts of music and speech (see FM broadcasting). Normal (analog) TV sound is also broadcast using FM. A narrowband form is used for voice communications in commercial and amateur radio settings. The type of FM used in broadcast is generally called WideBandFM (WBFM). In two- way radio, narrowband (NBFM) is used to conserve bandwidth. In addition, it is used to send signals into space.

MODEL GRAPH:









FORMULA:

Modulation Index

$$m_f = \frac{\Delta f}{f_m}$$

Δf is frequency deviation and fm is modulating frequency

PROCEDURE:

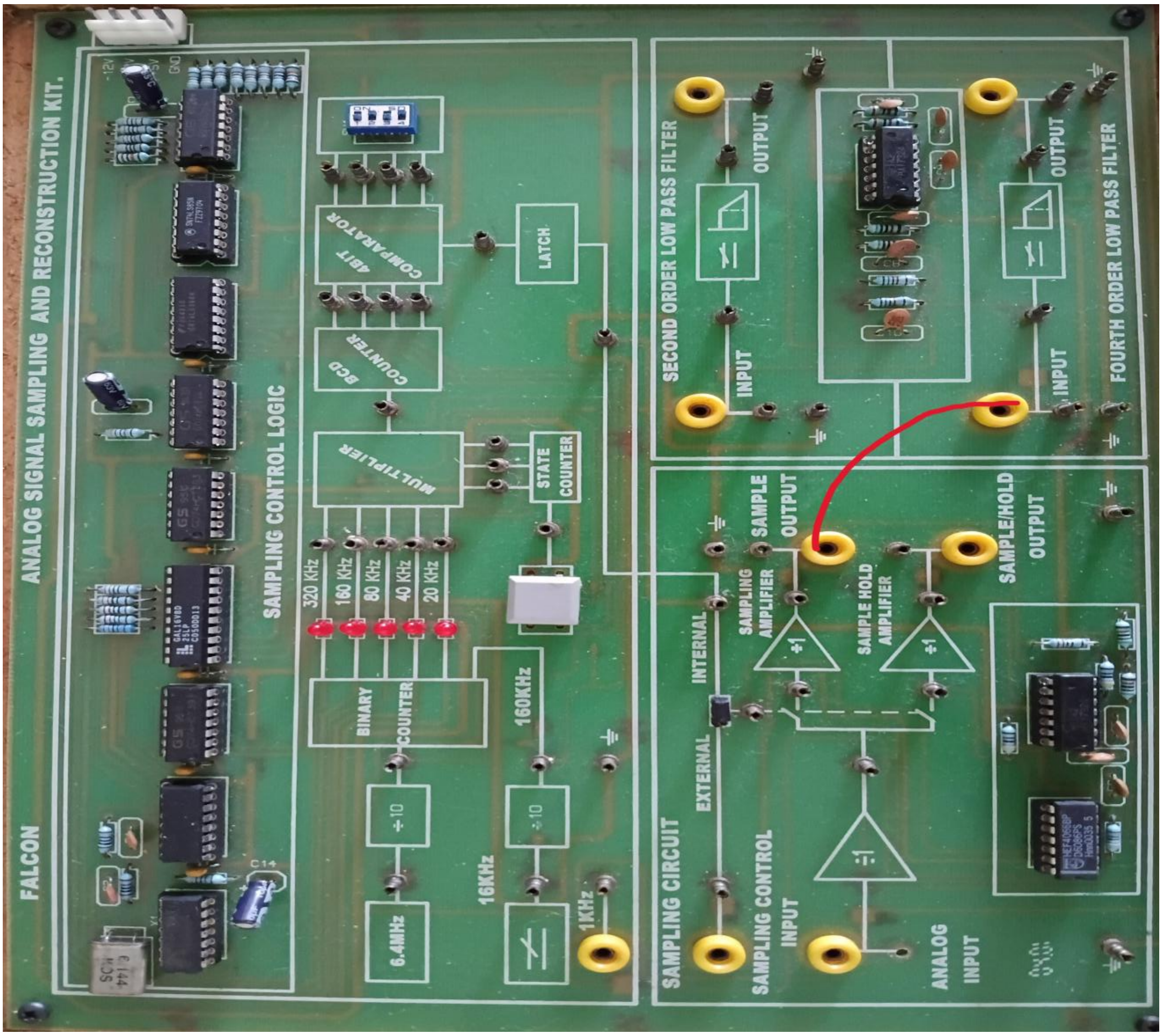
1. The circuit wiring is done as shown in diagram
2. Measure the carrier signal amplitude and the time period at FM O/P point before connecting the modulating signal input.
3. Connect the modulating input as per the connection in the diagram and vary the amplitude of the modulating signal until the frequency modulated signal is observed at FM O/P. Then measure the modulating signal amplitude and time period.
4. Observe the frequency demodulated amplitude and time period at the point Demodulated output by connecting the FM O/P point to the FM I/P.

POST LAB QUESTIONS:

1. What type of FM generation was done in the kit?
2. What do you mean by narrow band and wideband FM?
3. Was the demodulated output same as the message signal?
4. How did varying message signal amplitude affect frequency deviation?
5. Mention the Carson's rule for FM bandwidth estimation

RESULT:

Thus the Frequency modulation and demodulation was observed and the frequency deviation was observed in the modulated signal.



OBSERVATION:

SIGNAL	AMPLITUDE (Volts)	TIME PERIOD (sec)
Message Signal		
Sampling pulse		
Sampled Signal		
Reconstructed Signal		

Ex. No: 3(a)

SIGNAL SAMPLING

AIM:

To study and experimentally verify the sampling and reconstruction concept using Sampling Trainer Kit model.

APPARATUS AND COMPONENTS REQUIRED:

1. Sampling trainer kit, Patch chords (Or)
2. Transistor BC107 -1, Resistors – 10kΩ -2, 1.5 kΩ – 2, Capacitor – 1µF -2, Function generator- 2
3. CRO & probes

PRE LAB QUESTIONS:

1. What is Nyquist rate?
2. Explain sampling theorem?
3. What is sample and hold circuit?
4. Define aliasing in sampling.
5. What is the significance of the sampling frequency?

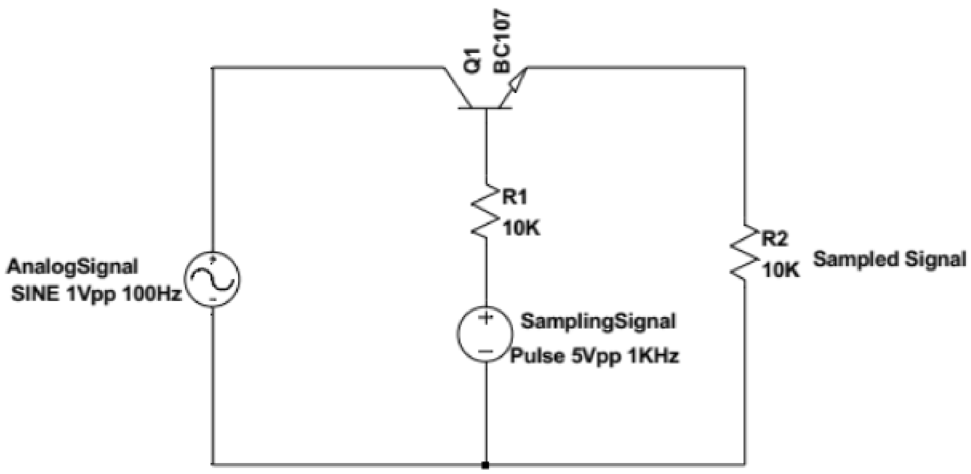
THEORY

The Nyquist criterion states that a continuous signal band limited to fm Hz, can be completely represented by and reconstructed from, the samples taken at a rate greater than or equal to 2 fm samples per second. This minimum frequency is called as "Nyquist Rate". Thus, for the faithful reconstruction of the information signal from its samples, it is necessary that the sampling rate, fs must be greater than 2fm.

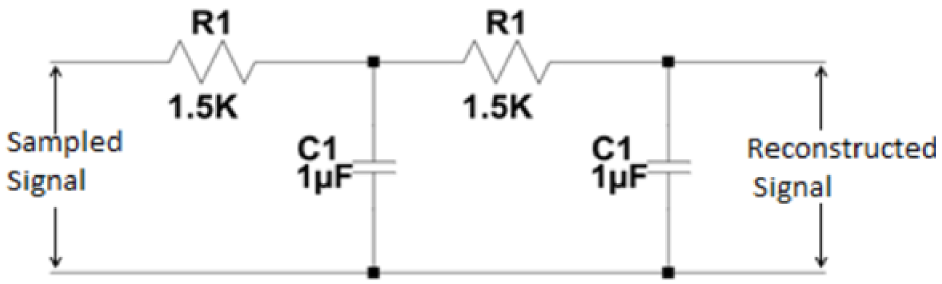
Aliasing: If the information signal is sampled at a rate, lower than that stated by Nyquist criterion, than there is an overlap between the information signal and the side bands of the harmonics. Thus the lower and the higher frequency components get mixed and cause unwanted signals to appear at the demodulator output. This phenomenon is termed as Aliasing or Fold-over Distortion.

CIRCUIT DIAGRAM:

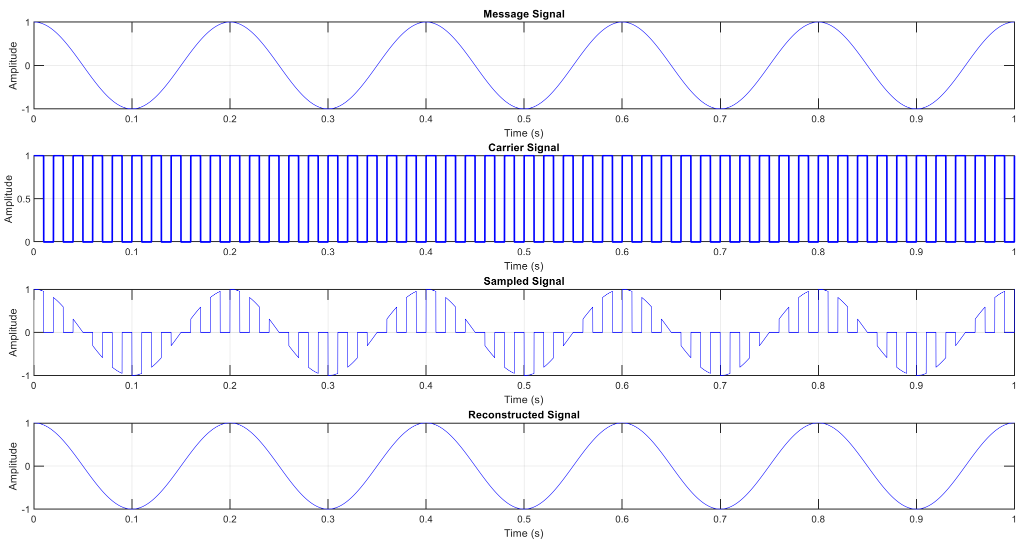
SAMPLING CIRCUIT



RECONSTRUCTION CIRCUIT



MODEL GRAPH:



PROCEDURE:

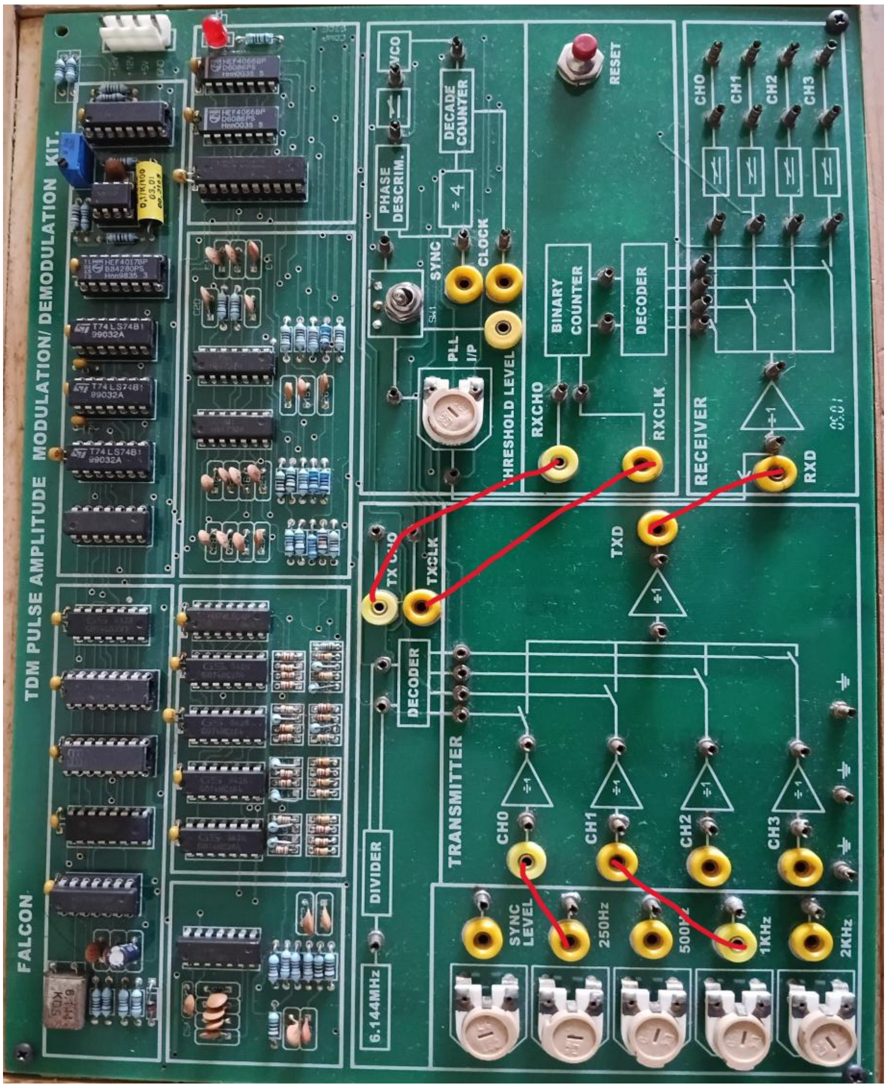
1. Connect the components as per the circuit diagram and observe the sampled signal and reconstructed signal.
2. Record the amplitude and time period of the analog signal, sampling signal, sampled signal and reconstructed signal.
3. Connect the square signal of amplitude 5 V, 1kHz frequency to the sampling control point and sinusoidal signal of amplitude 1V, 500Hz to the analog input, observe the sampled signal at the sample output point.
4. Connect the sample output as input to the input of fourth order low pass filter and observe th reconstructed signal.
5. Measure all the signal amplitude and time period and plot the same.

POST LAB QUESTIONS:

1. Is the output natural or flat top sampled?
2. For 5KHz signal what would be the minimum sampling rate?
3. How was the original analog signal reconstructed from samples?
4. What was the observed effect of quantization error?
5. How did sample-and-hold circuit behavior influence the output?

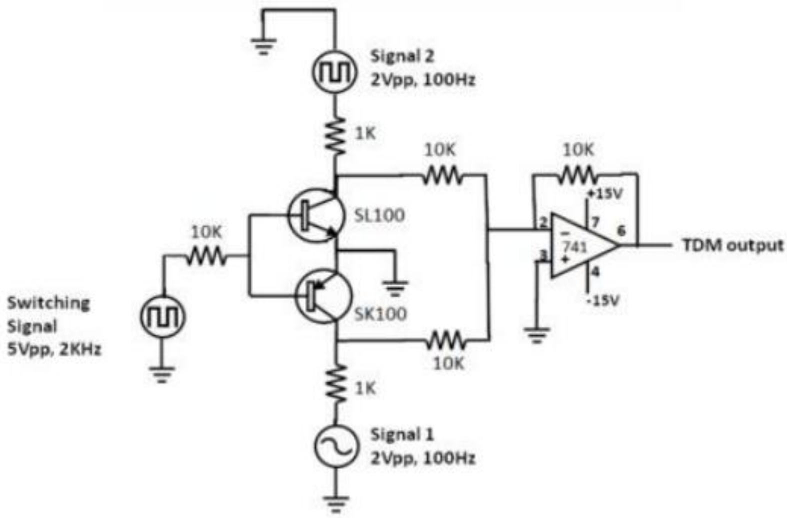
RESULT:

Thus the Sampling and reconstruction were performed and graphs were plotted.



CIRCUIT DIAGRAM :

TIME DIVISION MULTIPLEXER



Ex. Np. 3(b)

TIME DIVISION MULTIPLEXING

AIM:

To study the performance of Time division multiplexing and de-multiplexing using the trainer Kit.

APPARATUS REQUIRED:

1. TDM Trainer Kit & Patch Chords (or)
2. Transistor : SL100-1, Sk100-1, Resistors: 10 kΩ -4, 1 kΩ 2, IC741-1, power supply -1, function generator -3
3. CRO & Probes

PRE LAB QUESTIONS:

1. What is time division multiplexing?
2. What is meant by multiplexing?
3. List the advantages of multiplexing.
4. What are the requirements for synchronization in TDM?
5. What type of signals are best suited for TDM?

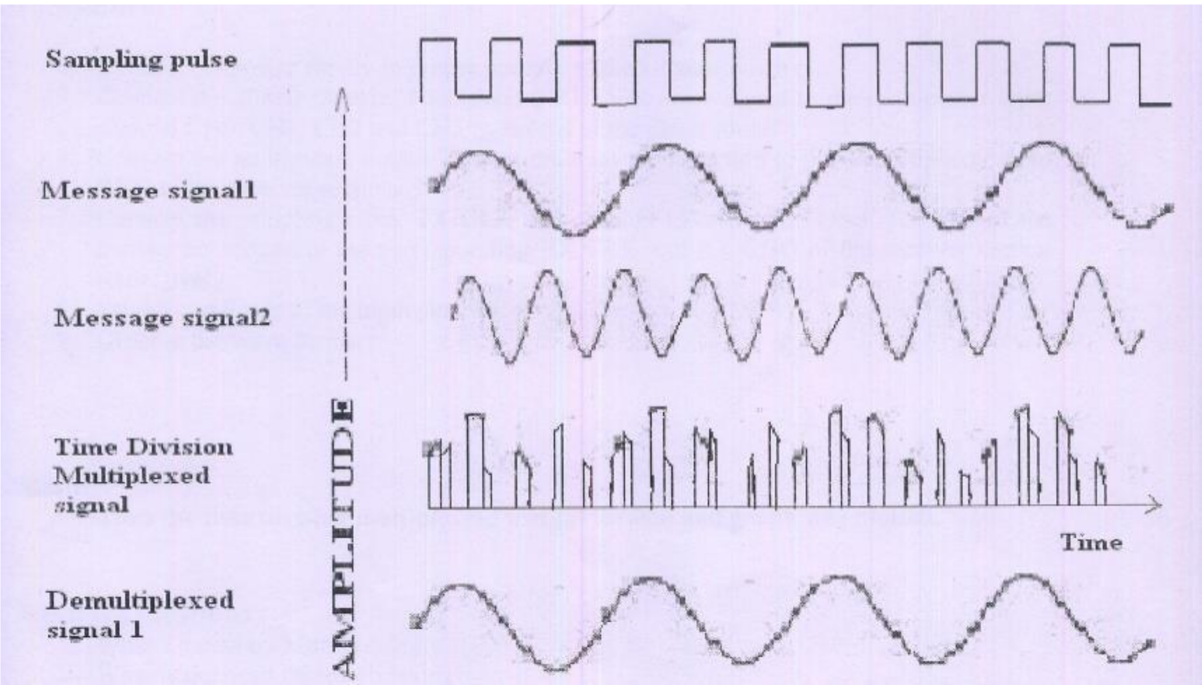
THEORY:

Time-Division Multiplexing (TDM) is a technique used to transmit multiple independent message signals over a single communication channel by dividing time into distinct slots. Each signal is assigned a specific time slot during which it can transmit its data. The message signals are first filtered using low-pass pre-alias filters to limit their bandwidth, ensuring efficient sampling. A commutator (electronic switch) samples each filtered signal at a rate slightly above twice its highest frequency (Nyquist rate) and interleaves these samples sequentially within a fixed time frame. This interleaved sequence forms the TDM signal, which is then modulated for transmission. At the receiver, a demodulator extracts the original pulses, and a synchronized de-commutator separates and routes them to the corresponding low-pass reconstruction filters for signal recovery. Synchronization between the transmitter and receiver is crucial for accurate de-multiplexing. TDM effectively allows joint channel utilization, increasing communication efficiency without mutual interference among signals,

OBSERVATION:

SIGNAL	AMPLITUDE (Volts)	TIME PERIOD (sec)
Message Signal		
Channel 1 signal		
Channel 2 signal		
Channel 3 signal		
Channel 4 signal		
TDM signal		
Demultiplexed		

MODEL GRAPH:



PROCEDURE:

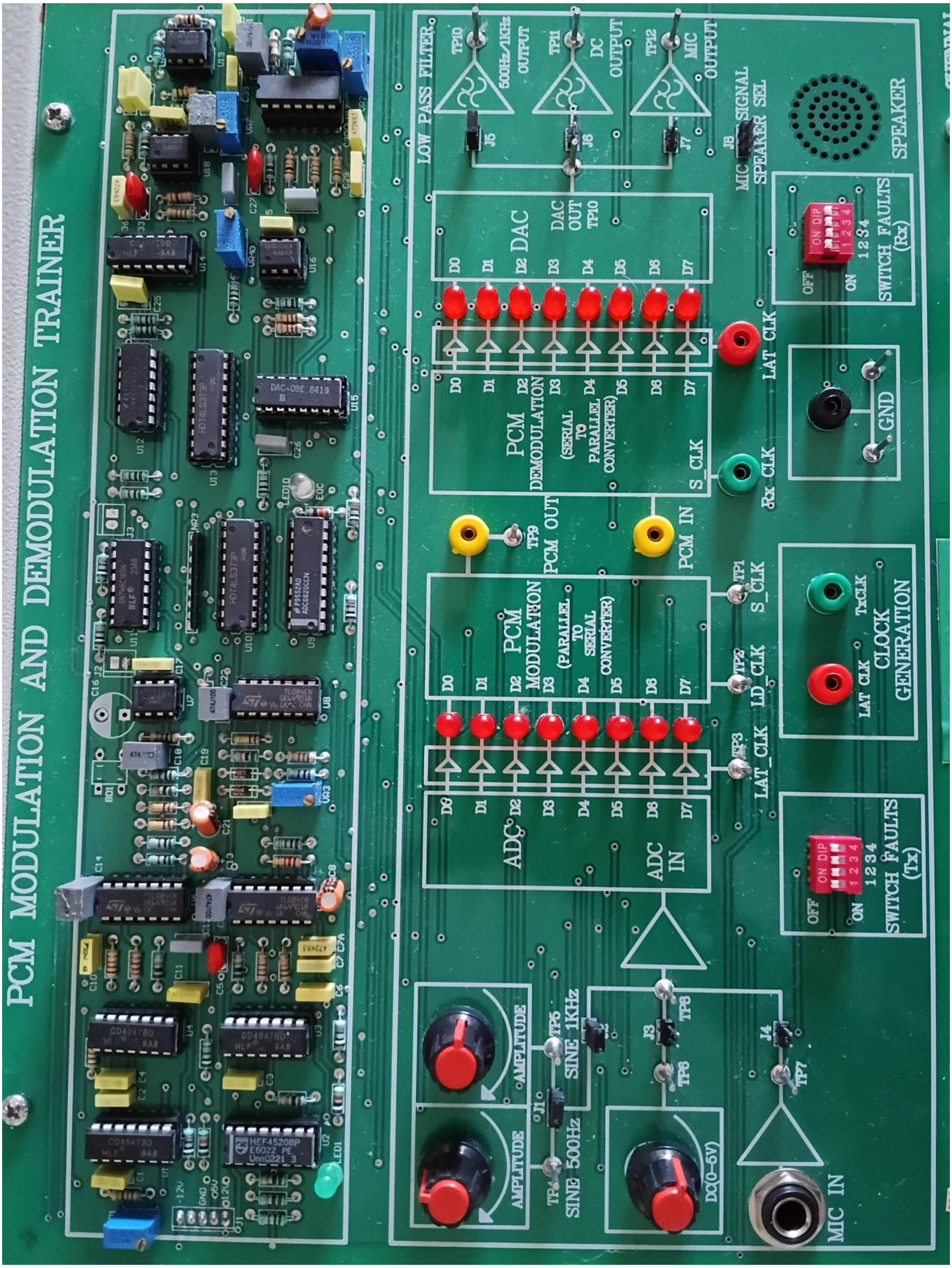
1. Connect the points as per the diagram of kit.
2. Out of the four channels to visualize the output clearly choose the Ch1 and Ch3 input 250Hz and 1kHz respectively.
3. Observe the interleave pulses measure the amplitude and time period at TxD point.
4. Connect Tx CH0 to Rx CH0 , Tx Clk to Rx Clk and TxD to RxD.
5. Observe the demultiplexed signal at Ch1 and Ch3 at the receiver end.
6. The procedure can be repeated by providing all the channel inputs.
7. Alternatively connect the components as per the circuit diagram and note the sampling pulse and message signal 1 &2 and observe the time division multiplexed output.

POST LAB QUESTIONS:

1. What is the advantage of TDM?
2. Compare FDM and TDM.
3. Mention how many message signals were multiplexed.
4. Summarize how synchronization was maintained between sender and receiver.
5. How the time slots assigned to different channels?

RESULT:

Thus, Time division multiplexing and de-multiplexing was performed and the respective waveforms were plotted


PCM MODULATION AND DEMODULATION TRAINER
LOW PASS FILTER
TP10
500Hz/1KHz OUTPUT
TP11
DC OUTPUT
TP12
MIC OUTPUT
SIGNAL SEL
MIC
SPEAKER
DAC
DAC OUT TP10
SWITCH FAULTS (Rx)
PCM DEMODULATION (SERIAL TO PARALLEL CONVERTER)
LAT CLK
S_CLK
Rx CLK
GND
TP9
PCM OUT
PCM IN
PCM MODULATION (PARALLEL TO SERIAL CONVERTER)
S_CLK
LD_CLK
LAT_CLK
TxCLK
CLOCK GENERATION
ADC
ADC IN
SWITCH FAULTS (Tx)
AMPLITUDE
SINE 1KHz
SINE 500Hz
DC(0-5V)
MIC IN

Ex. No. 4

PULSE CODE MODULATION AND DEMODULATION

AIM:

To perform pulse code modulation and demodulation and to observe and plot the corresponding waveforms

APPARATUS REQUIRED:

1. PCM trainer kit
2. CRO
3. Patch chords
4. Probes

PRE LAB QUESTIONS:

1. What is PCM?
2. List the advantages of PCM.
3. Summarize the application of PCM.
4. What are the steps involved in generating the PCM signal?
5. How does PCM differ from delta modulation?

THEORY:

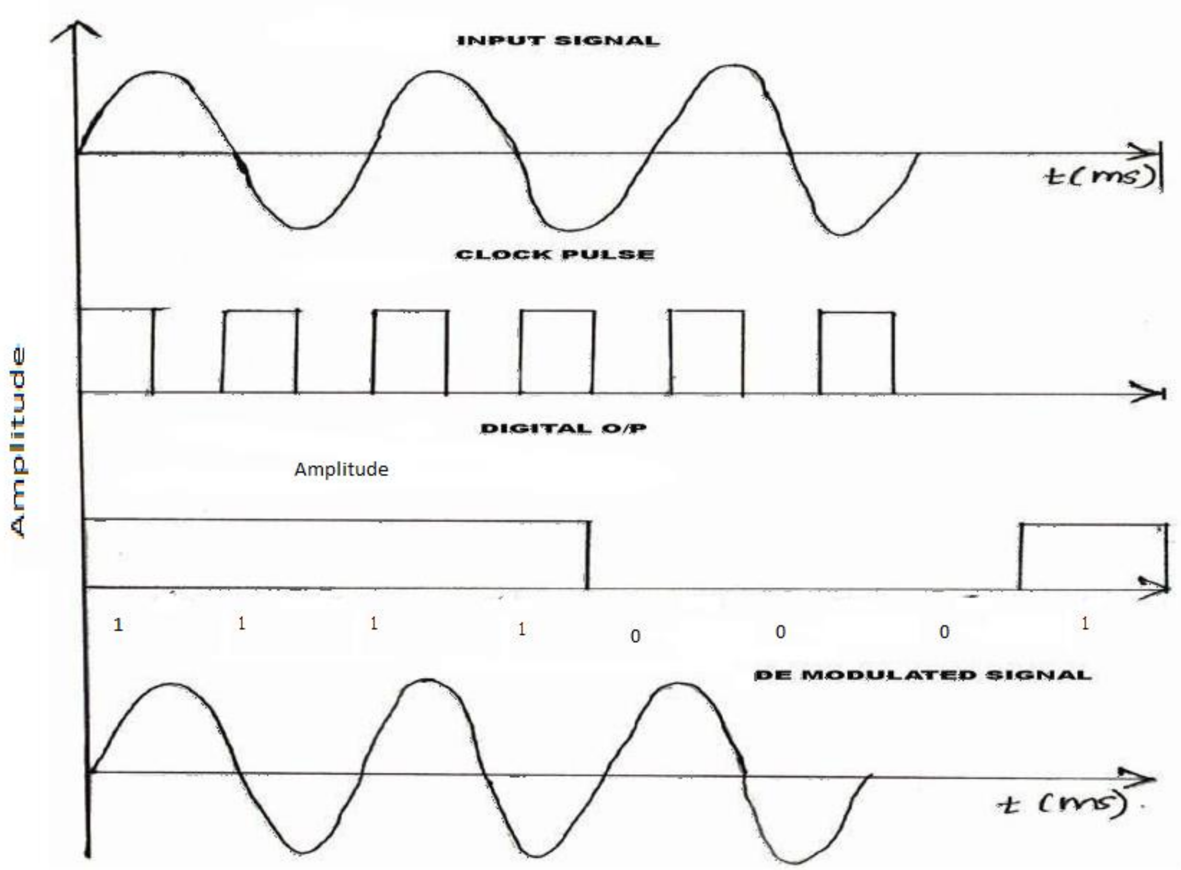
Pulse code modulation is a process of converting an analog signal into digital. The voice or any data input is first sampled using a sampler (which is a simple switch) and then quantized. Quantization is the process of converting a given signal amplitude to an equivalent binary number with fixed number of bits. This quantization can be either mid-tread or mid-raise and it can be uniform or non- uniform based on the requirements. For example in speech signals, the higher amplitudes will be less frequent than the low amplitudes.After quantization the signal is digital and the bits are passed through a parallel to serial converter and then launched into the channel serially.

At the demodulator the received bits are first converted into parallel frames and each frame is de-quantized to an equivalent analog value. This analog value is thus equivalent to a sampler output. This is the demodulated signal.

OBSERVATION :

S.No	Name of the signal	Amplitude in V	Time period (sec)
1	Message Signal		
2	Carrier Signal		
3	Modulated Signal		
4	Demodulated Signal		

MODEL GRAPH:



PROCEDURE:

1. Connect the jumper J1 towards TP5 and give sine signal with 1KHz as input to the PCM process.
2. Connect the LAT CLK and Tx CLK in the transmitter side to the LAT CLK and Rx CLK in the receiver side for synchronization.
3. By varying the amplitude of the message signal reflected in the glowing LEDs , the same process can be repeated by replacing sine wave with DC signal and by connecting the jumper J3.
4. As synchronization is established, the receiver LEDs. will vary same as the transmitter LEDs.
5. Connect the PCM OUT to PCM IN for demodulation and connect the jumper J5 to observe the amplitude and time period of the PCM demodulated signal at the output of low pass filter.

POST LAB QUESTIONS:

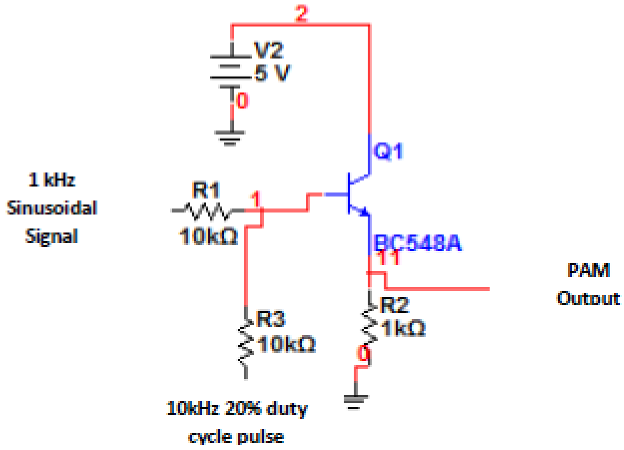
1. Mention the expression for transmission bandwidth in a PCM system?
2. How did the quantization levels affect signal quality?
3. What is the expression for quantization noise/error in PCM system?
4. How was the original analog signal reconstructed?
5. What are the disadvantages of PCM?

RESULT:

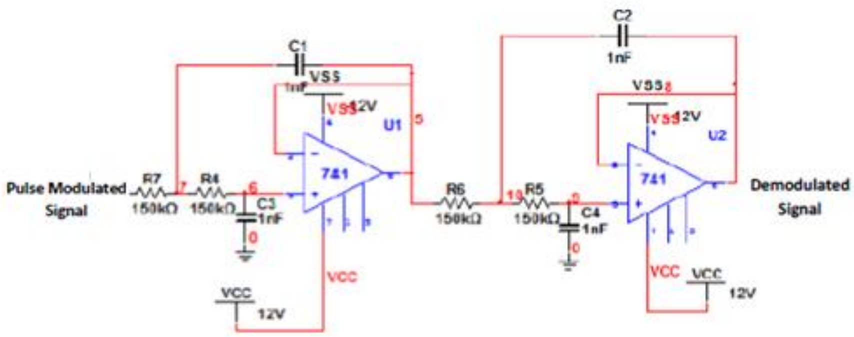
Thus the pulse code modulation and demodulation was performed using the trainer kit and graphs were plotted.

CIRCUIT DIAGRAM :

Pulse Amplitude Modulation



Pulse Amplitude Demodulation



OBSERVATION:

Signal	Amplitude (V)	Time period (s)
Message Signal		
Carrier Signal		
Pulse amplitude modulated signal		
Pulse amplitude demodulated signal		

Ex. No. 5(a)

PULSE AMPLITUDE MODULATION AND DEMODULATION

AIM: To perform pulse amplitude modulation and demodulation and observe the signals.

APPARATUS & COMPONENTS REQUIRED :

Function generator, Connecting probes, Connecting wires, Power supply, CRO,Transistor (BC 548), AFO, IC 741, Resistors, Capacitors.

PRE LAB QUESTIONS:

1. What is Pulse Amplitude Modulation (PAM)?
2. What are the types of PAM?
3. Explain the difference between natural sampling and flat-top sampling.
4. Why is a carrier signal used in modulation?
5. What is the role of a sample-and-hold circuit in PAM?

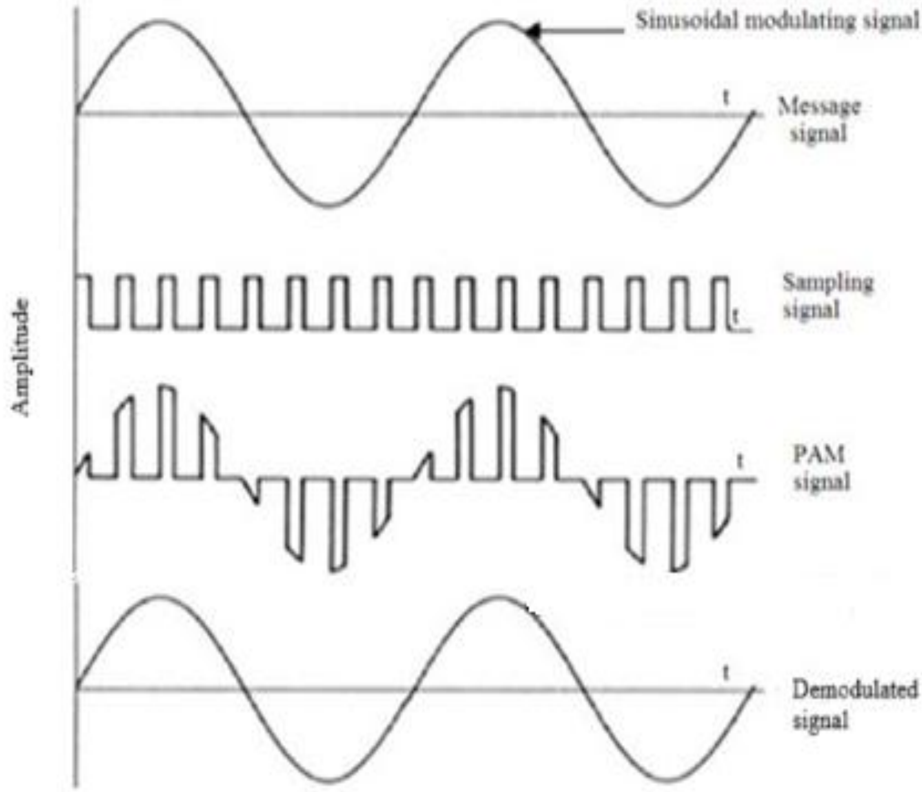
THEORY:

Pulse amplitude modulation is a scheme, which alters the amplitude of regularly spaced rectangular pulses in accordance with the instantaneous values of a continuous message signal. Then amplitude of the modulated pulses represents the amplitude of the intelligence. A train of very short pulses of constant amplitude and fast repetition rate is chosen, the amplitude of these pulse is made to vary in accordance with that of a slower modulating signal the result is that of multiplying the train by the modulating signal the envelope of the pulse height corresponds to the modulating wave. The PAM wave contains upper and lower side band frequencies besides the modulating and pulse signals. The demodulated PAM waves, the signal is passed through a low pass filter having a cut off frequencies equal to the highest frequency in the modulating signal.

PROCEDURE:

1. Connect the components as per the circuit diagram, set the amplitude and frequency of the message signal and carrier signal as specified.
2. Observe the amplitude and time period of the pulse amplitude modulated signal and note the same.
3. The PAM signal is given as an input to the demodulator circuit and the demodulation is performed to retrieve the message signal.
4. Observe the amplitude and time period of the demodulated signal.

MODEL GRAPH:



Sinusoidal modulating signal
t
Message
signal
Sampling
signal
t
Amplitude
t
PAM
signal
t
Demodulated
signal

POST LAB QUESTIONS:

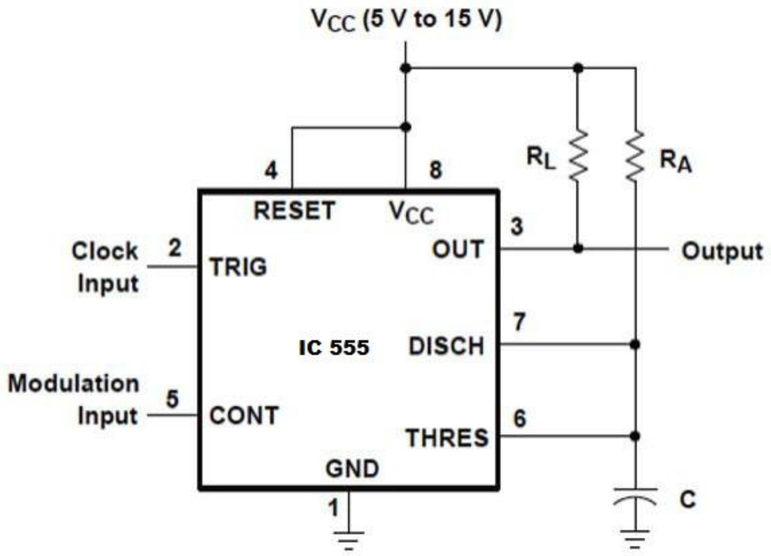
1. How does changing the sampling frequency affect the PAM signal?
2. What is the role of a low-pass filter in PAM demodulation?
3. Compare PAM modulation and demodulation techniques.
4. What are the practical applications of PAM?
5. What is the difference between single polarity and double polarity PAM?

RESULT:

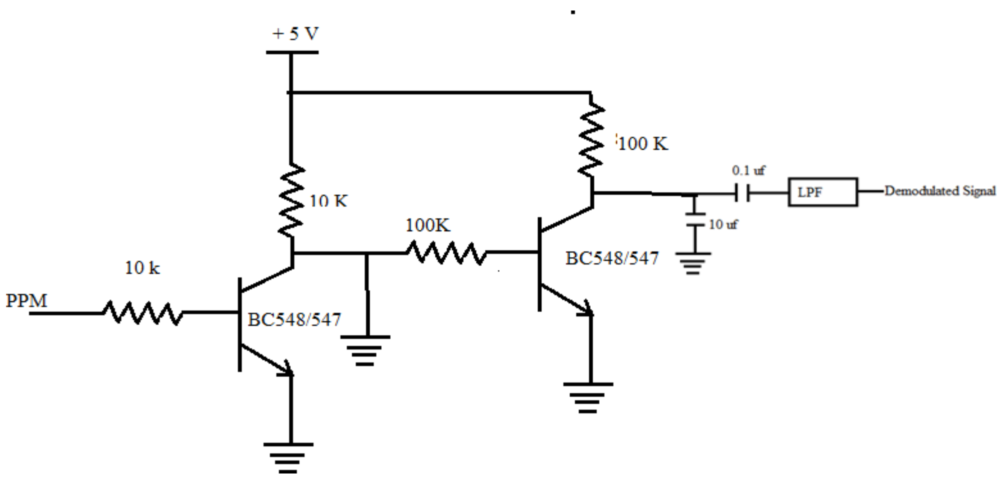
Thus, the circuit was implemented for pulse amplitude modulation and demodulation and the output was observed.

CIRCUIT DIAGRAM :

Pulse Width Modulation



Pulse Width Demodulation



Ex. No. 5(b)

PULSE WIDTH MODULATION AND DEMODULATION

AIM:

To perform pulse width modulation and demodulation techniques and observe the output waveforms.

APPARATUS AND COMPONENTS REQUIRED:

Function generator, Breadboard, multi meter, DC power Supply, Resistor (5.6 kΩ, 6.8 kΩ, 10 kΩ , 100 kΩ), Capacitors (10nf, 0.1µf), Transistor (BC547/548), IC 555.

PRE LAB QUESTIONS:

1. What is Pulse Width Modulation (PWM)?
2. How does PWM differ from PAM and PPM?
3. What is the advantage of PWM over other pulse modulation techniques?
4. Summarize the role of duty cycle in PWM.
5. Why is PWM used in motor control and power electronics?

THEORY:

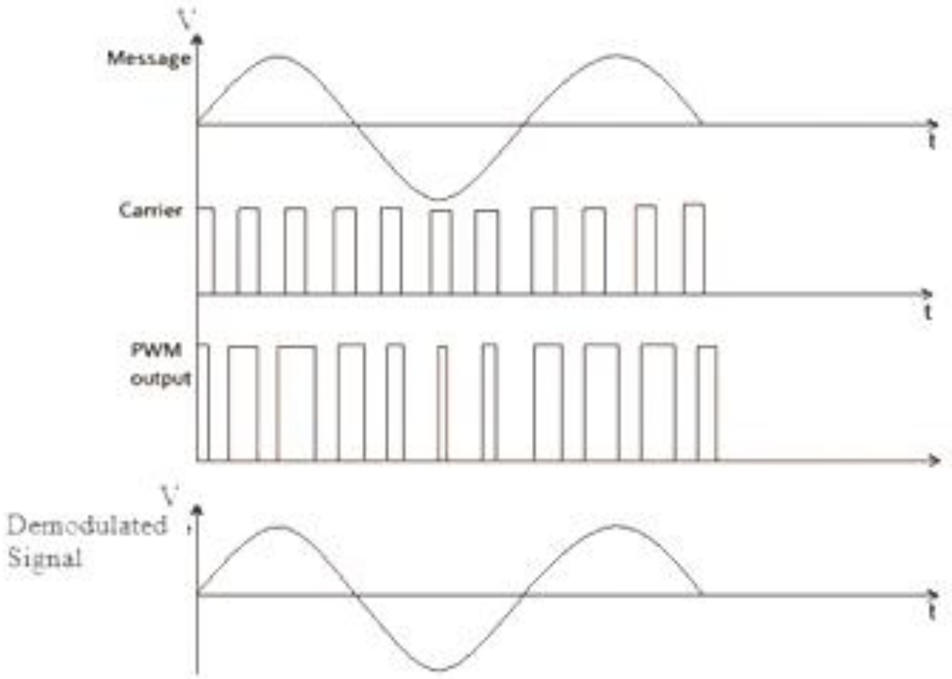
Pulse width Modulation (PWM) is a modulation techniques that control the width of the pulse, formally the Pulse Modulation based on modulator signal information. Its main use to allow the control the power supplied to electric device especially to inertial load.The main advantage of PWM is that power loss in the switching device is very low. When a switch is off there is practically no current and when it is on and power is transferred to the load, there is a voltage of voltage and current is thus in both case close to zero. PWM also work with digital control, which become of their on/off nature can easily set the duty the duty cycle.

The 555 timer IC is an integrated circuit (chip) used in a variety of timer, pulse generation, and oscillator applications. The 555 can be used to provide time delays, as an oscillator, and as a flip-flop element. Derivatives provide up to four timing circuits in one package. By applying a voltage to the control voltage input one can alter the timing characteristics of the device. In most applications, the control voltage input is not used. It is usual to connect a 10 nF capacitor between pin 5 and 0 V to prevent interference. The control voltage input can be used to build an astable multivibrator with a frequency-modulated output.

OBSERVATION:

Signal	Amplitude (V)	Time period (s)
Message Signal		
Carrier Signal		
Pulse width modulated signal		
Pulse width demodulated signal		

MODEL GRAPH



PROCEDURE:

1. Connect the components as per the circuit diagram, set the amplitude and frequency of the message signal and carrier signal as specified.
2. Observe the amplitude and time period of the pulse width modulated signal and note the same.
3. The PWM signal is given as an input to the demodulator circuit and the demodulation is performed to retrieve the message signal.
4. Observe the amplitude and time period of the demodulated signal.

POST LAB QUESTIONS:

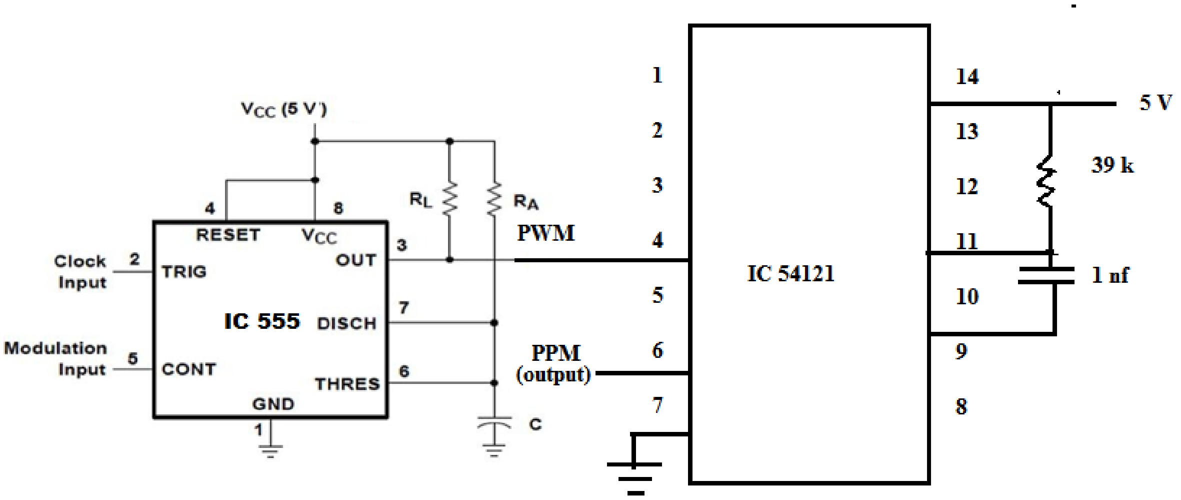
1. How does changing the duty cycle affect the PWM signal?
2. What is the role of a low-pass filter in PWM demodulation?
3. Compare PWM modulation and demodulation techniques.
4. What are the practical applications of PWM?
5. How does PWM contribute to digital communication?

RESULT:

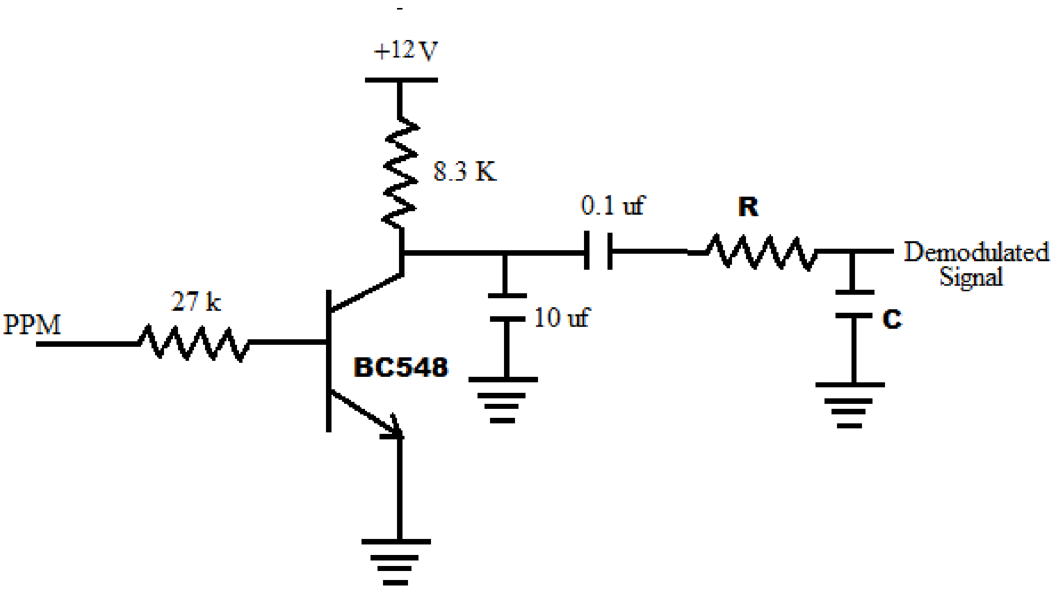
Thus the circuit was implemented for pulse width modulation and demodulation and the output was observed.

CIRCUIT DIAGRAM:

PULSE POSITION MODULATION



PULSE POSITION DEMODULATION



Ex. No. 5(c)

PULSE POSITION MODULATION AND DEMODULATION

Aim:

To perform Pulse Position Modulation and Demodulation and to observe the output waveforms.

Apparatus and Components Required:

DC Power Supply, CRO, Function generator, IC 555, IC 54121, Transistor (BC 548), IC 741, Resistors, Capacitors.

PRE LAB QUESTIONS:

1. What is Pulse Position Modulation (PPM)?
2. How does PPM differ from PAM and PWM?
3. What is the advantage of PPM over other pulse modulation techniques?
4. What is the sampling theorem, and how does it apply to PPM?
5. What is the Nyquist rate, and why is it important in PPM?

THEORY:

In pulse position modulation, the amplitude and width of the pulse are kept constant, while the position of each pulses are kept constant, while the position of a reference pulse is changed according to the instantaneous sampled value of the modulating circuit. Thus, the transmitter and receiver is in synchronization. As the amplitude and width of the pulse are constant, the transmitter handles constant power output which is a definite advantage over PWM.

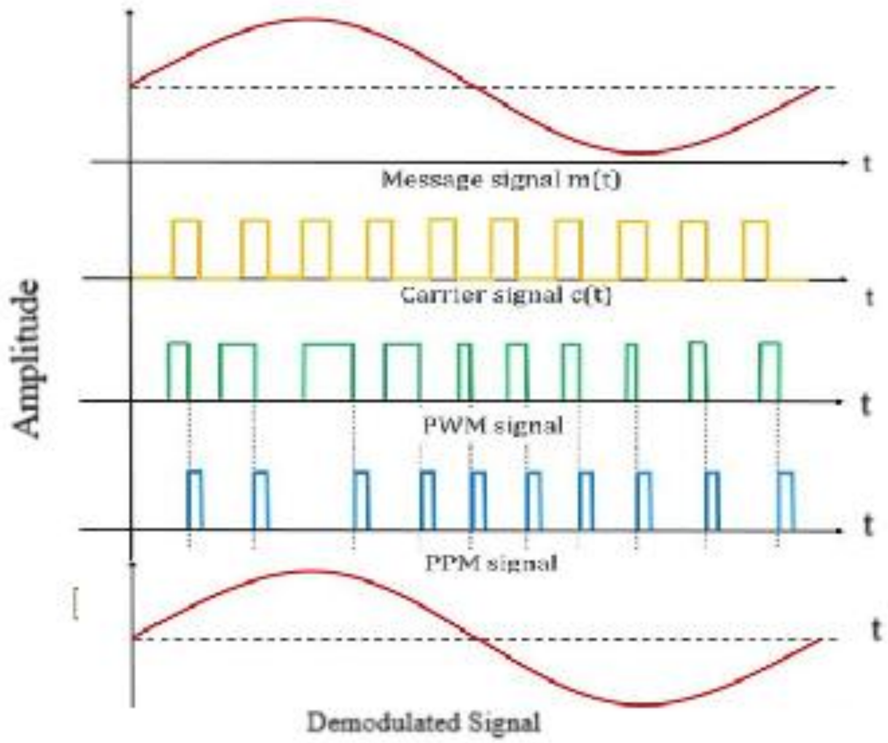
Pulse position modulator: It consist of a PWM generator followed by the monostable multivibrator. Since in PPM , output remain high for fixed duration from trailing edges of the PWM signal, the trailing edge of the PWM signal is used as a trigger input for the monostable multivibrator.

DM54/54121: It is a monostable multivibrator having both positive and negative edge triggering with complementary outputs. An internal 2kΩ resistor is provided for design convenience. A single external capacitor is used. Input (A) is active low trigger transition input and input (B) is an active high Schmitt trigger input that allows jitter free triggering input.

OBSERVATION:

Signal	Amplitude (V)	Time period (s)
Message Signal		
Carrier Signal		
Pulse position modulated signal		
Pulse position demodulated signal		

MODEL GRAPH:



PROCEDURE:

1. Connect the components as per the circuit diagram, set the amplitude and frequency of the message signal and carrier signal as specified.
2. Observe the amplitude and time period of the pulse position modulated signal and note the same.
3. The PPM signal is given as an input to the demodulator circuit and the demodulation is performed to retrieve the message signal.
4. Observe the amplitude and time period of the demodulated signal.

POST LAB QUESTIONS:

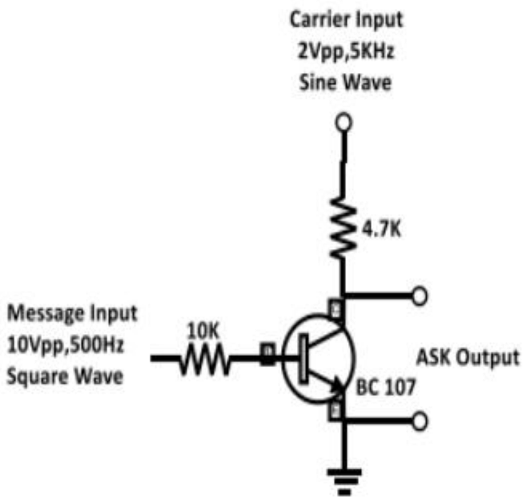
1. What observations did you make about the PPM waveform?
2. How does changing the pulse width affect the PPM signal?
3. What is the role of a low-pass filter in PPM demodulation?
4. What are the practical applications of PPM?
5. What challenges did you face during the PPM experiment?

RESULT:

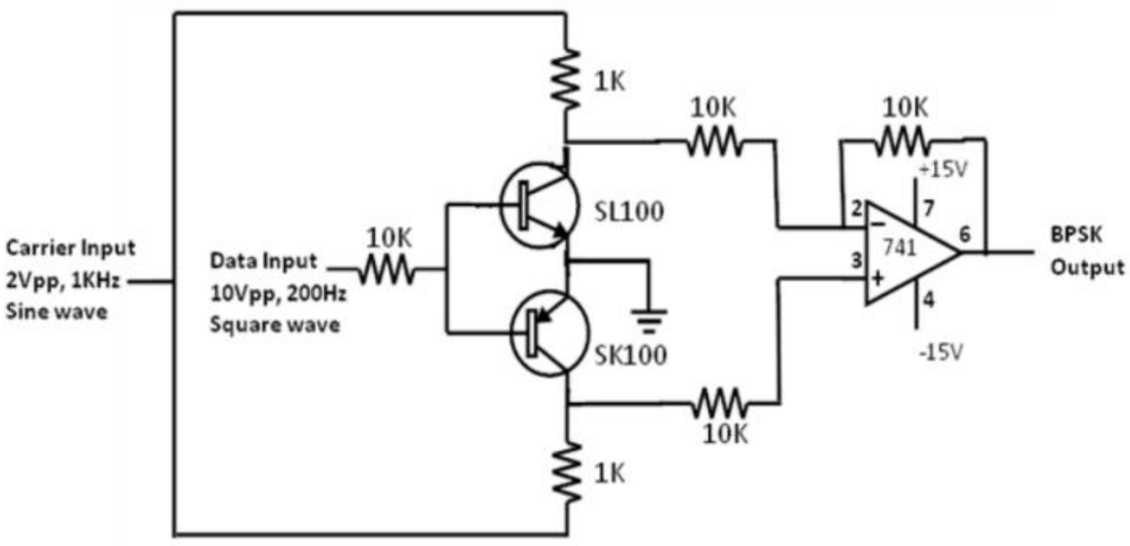
Thus the circuit was implemented for pulse position modulation and demodulation and the output was observed.

CIRCUIT DIAGRAM:

AMPLITUDE SHIFT KEYING MODULATOR



PHASE SHIFT KEYING MODULATOR



Ex. No:6

DIGITAL MODULATION - ASK, PSK, FSK

AIM:

To perform digital modulation techniques such as Amplitude Shift Keying (ASK), Phase Shift Keying (PSK) and Frequency Shift Keying (FSK) and to observe the modulated waveforms.

APPARATUS AND COMPONENTS REQUIRED:

FOR ASK - Transistor: BC107- 1, Resistor : 4.7 k, 10 k , function generation -2, CRO, power supply-1, bread board, wires, probes

FOR PSK – Transistor : Sl100, SK100, Resistor: 10kΩ -4, 1kΩ -2, function generator -2, CRO, probes, bread board, wires

FOR FSK – Resistor – 470Ω,10kΩ -2, 4.7kΩ, Transistor – 2N2222A, Capacitor – 0.01µF, Function generator, CRO, bread board, wires, probes

PRE-LAB QUESTIONS :

1. What is digital modulation?
2. Why is PSK more robust than ASK in terms of noise immunity?
3. How is binary data represented in ASK?
4. How does PSK differ from ASK and FSK?
5. What is meant by coherent detection in PSK?

THEORY:

Any digital modulation scheme uses a finite number of distinct signals to represent digital data.

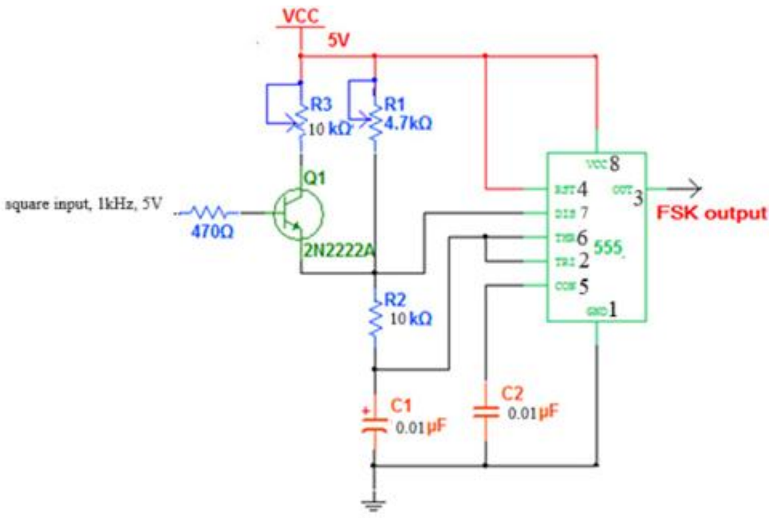
ASK (Amplitude Shift Keying) modulates digital data by varying the amplitude of a carrier wave while keeping frequency and phase constant. Binary ASK uses two amplitude levels - high for '1' and low (often zero) for '0'. It's simple to implement but highly susceptible to noise and amplitude variations in the channel.

PSK (Phase Shift Keying) encodes data by shifting the phase of the carrier signal. Binary PSK (BPSK) uses two phase states 180° apart, while QPSK uses four phases to transmit two bits per symbol. PSK offers better noise immunity than ASK since phase information is more robust against amplitude distortions.

FSK (Frequency Shift Keying) represents digital bits using different carrier frequencies. Binary FSK assigns one frequency to '0' and another to '1'. It provides good noise resistance and works well in noisy environments, making it popular for radio communications and modems.

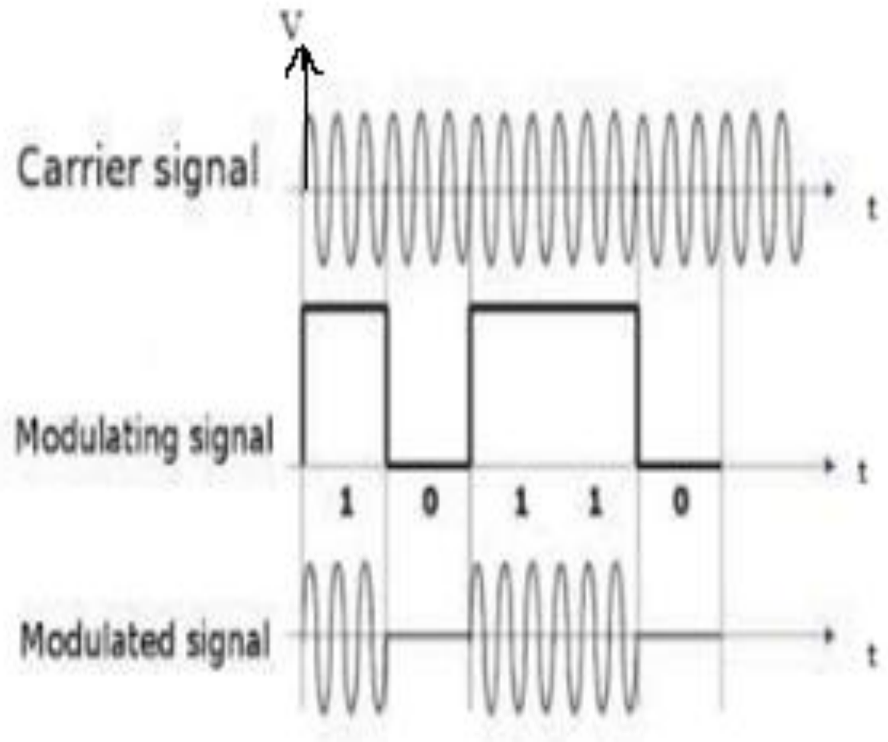
as data input.

FREQUENCY SHIFT KEYING MODULATOR

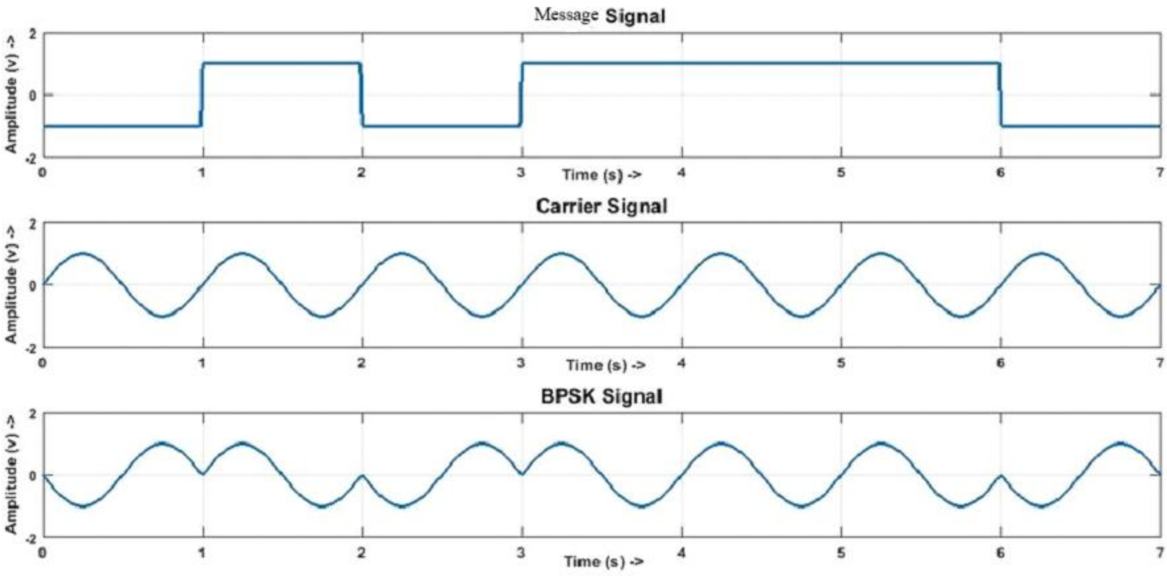


MODEL GRAPH:

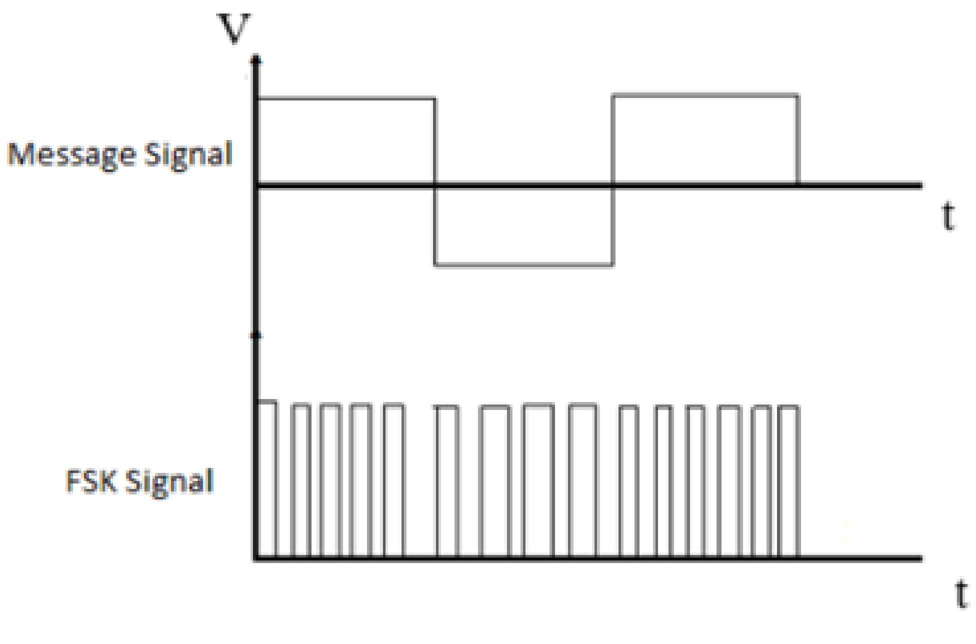
AMPLITUDE SHIFT KEYING MODULATION



PHASE SHIFT KEYING MODULATION



FREQUENCY SHIFT KEYING MODULATION



OBSERVATION:

Modulation	Signal	Amplitude (V)	Time period (s)
ASK	Message Signal		
	Carrier Signal		
	Modulated Signal		
PSK	Message Signal		
	Carrier Signal		
	Modulated Signal		
FSK	Message Signal		
	Carrier Signal		
	Modulated Signal		

PROCEDURE:

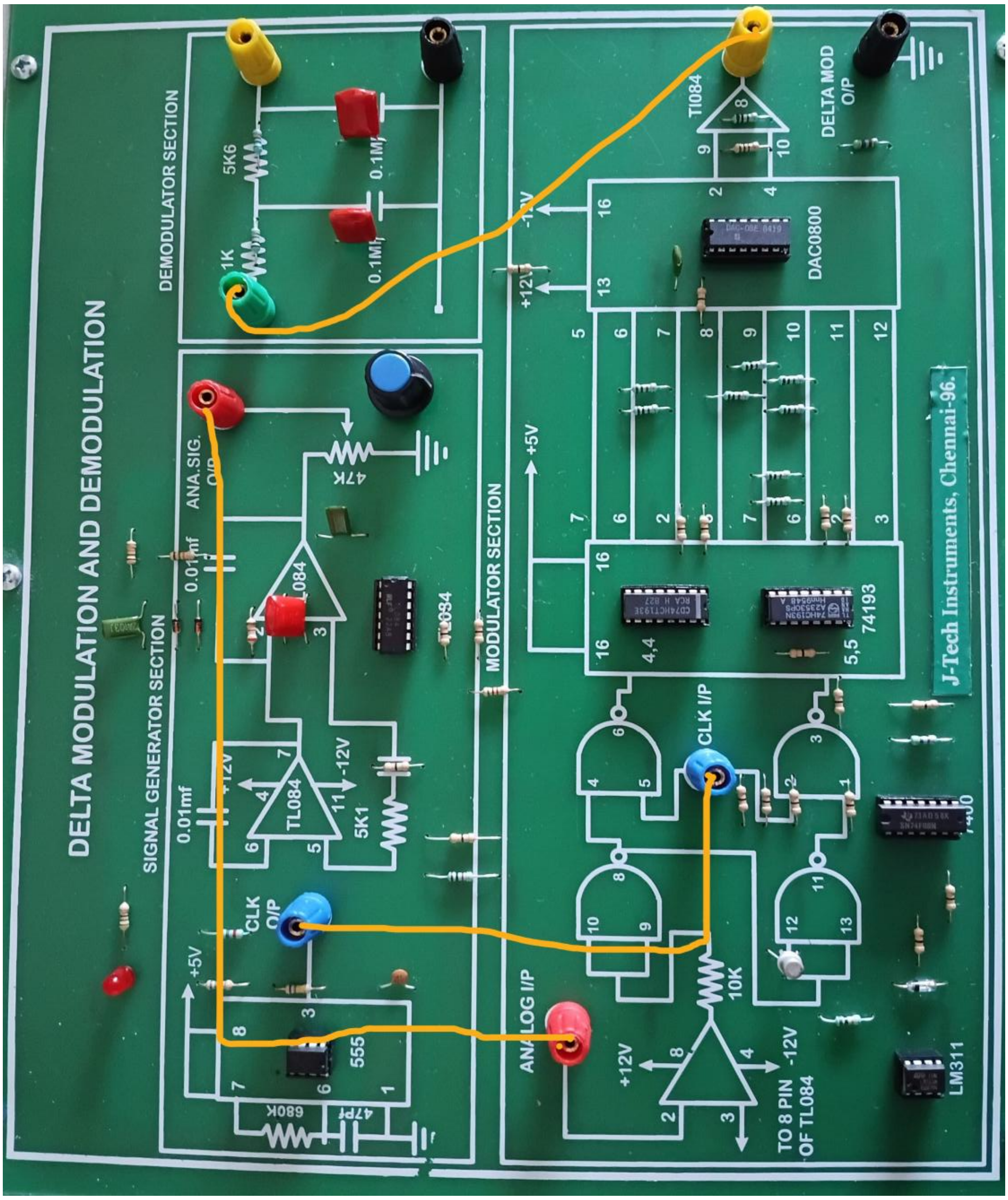
1. For ASK set 2Vpp, 5 KHz sinusoidal signal as carrier and the 10Vpp, 500Hz square wave as carrier signal .
2. Similarly connect the components as per the circuit diagram for PSK and FSK and observe the carrier wave, message signal and the modulated waveform.
3. For the FSK , to measure the carrier signal, turn off the message signal(square signal) and observe the signal in pin 3 of IC 555.
4. Observe carrier signal, message signal and modulated signal for all the modulations from the point as mentioned in the circuit diagram.

POST LAB QUESTIONS:

1. How did the frequency separation affect signal clarity?
2. What challenges were faced in achieving phase synchronization?
3. What is On-Off Keying (OOK)?
4. Mention the difference between coherent and non-coherent FSK detection.
5. Write the bandwidth required for ASK transmission.

RESULT:

Thus the ASK PSK and FSK modulation was performed and the output waveform was observed.



OBSERVATION:

SIGNAL	AMPLITUDE (Volts)	TIME PERIOD (sec)
Input		
Integrated output		
Delta Modulated Wave		
Delta Demodulated Wave		

Ex. No:7

DELTA MODULATION AND DEMODULATION

AIM:

To perform delta modulation and demodulation and to observe the corresponding waveforms.

APPARATUS REQUIRED

1. Delta Modulator kit
2. CRO and Probes
3. Patch chords

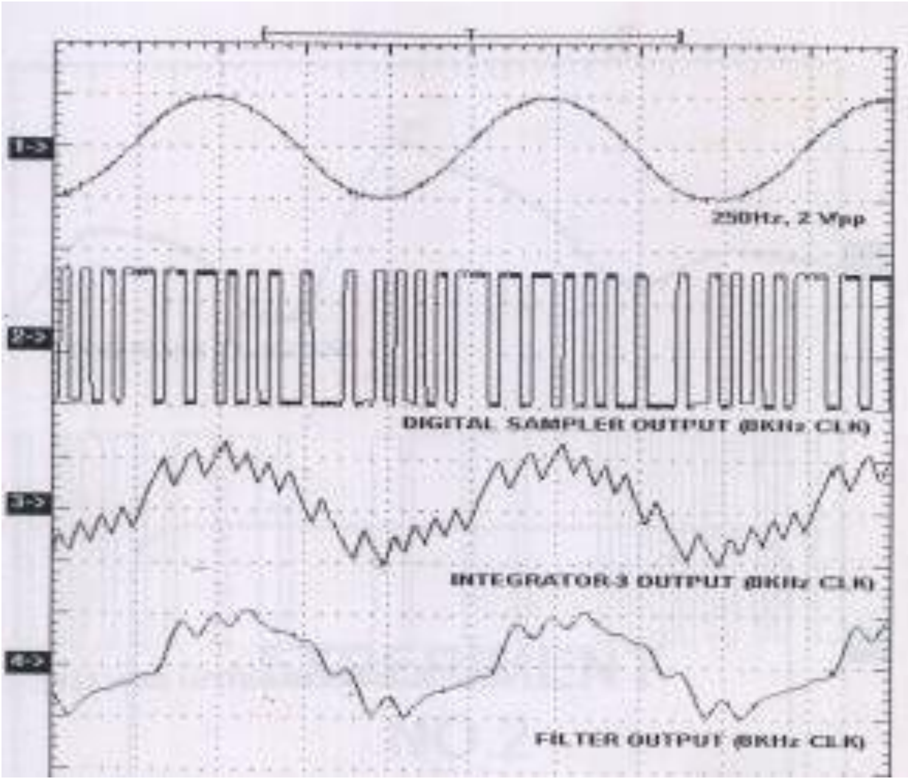
PRE LAB QUESTIONS:

1. What is delta modulation and how does it differ from conventional PCM?
2. What is the purpose of the integrator circuit in delta modulation?
3. What causes slope overload distortion in delta modulation?
4. What is granular noise in delta modulation?
5. How does the sampling frequency affect delta modulation performance?

THEORY:

Delta Modulation (DM) is a simplified form of Pulse Code Modulation (PCM) where each sample of the message signal is encoded into just one bit, significantly reducing the bit rate. It operates on the principle of encoding the difference between the current and previous samples, rather than the absolute value. If the current sample is larger than the previous one, a '1' is transmitted; if smaller, a '0' is transmitted. At the transmitter, this comparison is achieved using a comparator and a feedback loop involving a Digital-to-Analog Converter (DAC) and an up/down counter, which track and generate a reference signal. The encoded single-bit stream is transmitted to the receiver, where an up-counter and DAC reconstruct the staircase waveform. This is then passed through an operational amplifier (Op-Amp) and a Low Pass Filter (LPF) to smooth the signal and retrieve the original message. Delta Modulation offers efficient transmission but may suffer from slope overload or granular noise.

MODEL GRAPH:


1->
250Hz, 2 Vpp
2->
DIGITAL SAMPLER OUTPUT (8KHz CLK)
3->
INTEGRATOR-3 OUTPUT (8KHz CLK)
4->
FILTER OUTPUT (8KHz CLK)

PROCEDURE:

1. As per the diagram, connect the clock signal and analog input in the delta modulation section.
2. Note the time period and amplitude of the clock signal and analog input.
3. Measure the delta modulated signal output at Delta MOD O/P point .
4. Connect the Delta MOD O/P to the input of the demodulator section to retrieve the message signal.
5. Modulation and demodulation process can be analyzed by varying analog message signal input.

POST LAB QUESTIONS:

1. How to over come slope overload distortion?
2. What is the other name of Granular noise?
3. What is meant by staircase approximation?
4. What happened to the output signal quality when you increased the input signal amplitude beyond a certain threshold?
5. How did varying the step size affect both slope overload and granular noise in your experiment?

RESULT: Thus the analog message signal in its digital form was transmitted using Delta modulator and the original analog message signal was reconstructed at the receiver by the Delta demodulator.

Ex. No:8

SIMULATION OF ASK, FSK, AND BPSK GENERATION AND DETECTION SCHEMES

AIM:

To perform generation and detection of ASK, FSK, BPSK schemes using MATLAB.

SOFTWARE REQUIRED:

MATLAB software

PRE-LAB QUESTIONS :

1. What is ASK modulation?
2. Compare the performance of ASK, FSK and BPSK.
3. What is Constellation in the digital modulation techniques?
4. How does PSK differ from ASK and FSK?
5. How binary data is represented in FSK ?

THEORY:

Any digital modulation scheme uses a finite number of distinct signals to represent digital data.

ASK (Amplitude Shift Keying) modulates digital data by varying the amplitude of a carrier wave while keeping frequency and phase constant. Binary ASK uses two amplitude levels - high for '1' and low (often zero) for '0'. It's simple to implement but highly susceptible to noise and amplitude variations in the channel.

PSK (Phase Shift Keying) encodes data by shifting the phase of the carrier signal. Binary PSK (BPSK) uses two phase states 180° apart, while QPSK uses four phases to transmit two bits per symbol. PSK offers better noise immunity than ASK since phase information is more robust against amplitude distortions.

FSK (Frequency Shift Keying) represents digital bits using different carrier frequencies. Binary FSK assigns one frequency to '0' and another to '1'. It provides good noise resistance and works well in noisy environments, making it popular for radio communications and modems.

ALGORITHM:

1. Start.
2. Get the data bits and compute its length.
3. Generate carrier signal , compute ASK modulated signal by multiplying the carrier signal with the input bits and coherent demodulation is performed to retrieve the message bits.
4. For FSK modulation, generate two carrier signals and multiply with input bits bit 1 with Vc1 and bit 0 with Vc2 to get the modulated signal.
5. Coherent demodulation is performed to retrieve the message bits from the FSK modulated signal.
6. In PSK modulation, Generate the BPSK waveforms with their corresponding carrier frequency.
7. Plot the output waveform.
8. Stop

PROGRAM FOR ASK

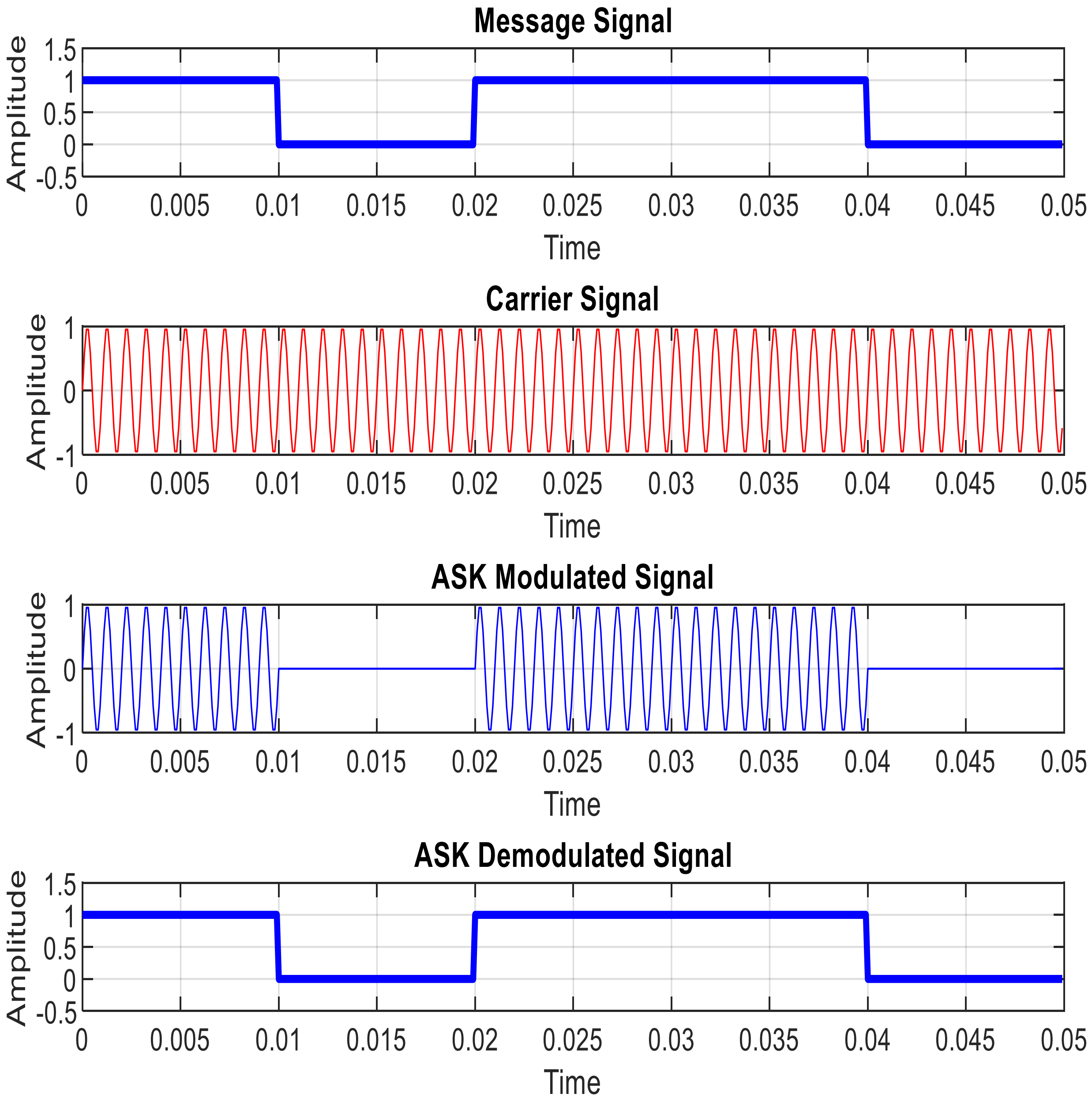
```
clc; clear; close all;
% Parameters
data = [1 0 1 1 0];      % Binary message
Fs = 10000;              % Sampling frequency
Fc = 1000;               % Carrier frequency
Tb = 0.01;               % Bit duration
t_bit = 0:1/Fs:Tb-1/Fs;  % Time for one bit
% Initialize
msg_signal = [];
modulated = [];
carrier = [];
t = [];
% Modulation
for bit = data
    msg = bit * ones(1, length(t_bit));        % Message segment
    car = sin(2*pi*Fc*t_bit);                  % Carrier
    mod = msg .* car;                          % ASK modulation
    % Accumulate
    msg_signal = [msg_signal msg];
```

```
 carrier = [carrier car];
    modulated = [modulated mod];
    t = [t t_bit + length(t)/Fs];              % Time vector
end
% Coherent Demodulation
received = modulated .* carrier;
% Low-pass filter (moving average)
N = round(Fs*Tb/10);  % smoothing window
demod_filtered = filter(ones(1,N)/N, 1, received);
% Bit decisions
samples_per_bit = length(t_bit);
received_bits = [];
for i = 1:length(data)
    segment = demod_filtered((i-1)*samples_per_bit + 1 : i*samples_per_bit);
    avg_val = mean(segment);
    if avg_val > 0.3
        received_bits = [received_bits 1];
    else
        received_bits = [received_bits 0];
    end
end
% Reconstruct demodulated signal for plotting
demod_signal = [];
for bit = received_bits
    demod_signal = [demod_signal bit * ones(1, samples_per_bit)];
end
% Plotting
figure;
subplot(4,1,1);
plot(t, msg_signal,'b', 'LineWidth', 2);
title('Message Signal');
xlabel('Time'); ylabel('Amplitude'); ylim([-0.5 1.5]); grid on;
subplot(4,1,2);
```

ASK OUTPUT:

Original bits: 1 0 1 1 0

Demodulated bits: 1 0 1 1 0



```
  plot(t, carrier, 'r');
  title('Carrier Signal');
  xlabel('Time'); ylabel('Amplitude'); grid on;
  subplot(4,1,3);
  plot(t, modulated, 'b');
  title('ASK Modulated Signal');
  xlabel('Time'); ylabel('Amplitude'); grid on;
  subplot(4,1,4);
  plot(t, demod_signal, 'b','LineWidth',2);
  title('ASK Demodulated Signal');
  xlabel('Time'); ylabel('Amplitude'); ylim([-0.5 1.5]); grid on;
  % Display result
  disp(['Original bits:     ' num2str(data)]);
  disp(['Demodulated bits:  ' num2str(received_bits)]);
```

PROGRAM FOR FSK:

```
clc; clear; close all;
% Parameters
data = [1 0 1 1 0];
Fs = 10000;           % Sampling frequency
Tb = 0.01;            % Bit duration
t_bit = 0:1/Fs:Tb-1/Fs;
F0 = 500;             % Frequency for bit 0
F1 = 1000;            % Frequency for bit 1
% Initialize
msg_signal = [];
modulated = [];
t = [];
car1 = cos(2*pi*F1*t_bit);
car2 = cos(2*pi*F0*t_bit);
% Modulation
for bit = data
    msg = bit * ones(1, length(t_bit));
    if bit == 1
```

```
        car = sin(2*pi*F1*t_bit);
    else
        car = sin(2*pi*F0*t_bit);
    end
    mod = car;
        msg_signal = [msg_signal msg];
    modulated = [modulated mod];
    t = [t t_bit + length(t)/Fs];
end
car1 = sin(2*pi*F1*t);
car2 = sin(2*pi*F0*t);
% Demodulation
demodulated_bits = [];
demod_signal = [];
for i = 1:length(data)
    segment = modulated((i-1)*length(t_bit) + 1 : i*length(t_bit));
        % Correlate with both frequencies
    corr0 = sum(segment .* sin(2*pi*F0*t_bit));
    corr1 = sum(segment .* sin(2*pi*F1*t_bit));

    if corr1 > corr0
        bit = 1;
    else
        bit = 0;
    end
        demodulated_bits = [demodulated_bits bit];
    demod_signal = [demod_signal bit * ones(1, length(t_bit))];
end
% Plotting
figure;
subplot(5,1,1);
plot(t, msg_signal, 'LineWidth', 1.5);
title('Message Signal'); ylim([-0.5 1.5]); grid on;
```

```
subplot(5,1,2);
plot(t, car1, 'LineWidth', 1.5);
title('Carrier 1 Signal'); grid on;

subplot(5,1,3);
plot(t, car2, 'LineWidth', 1.5);
title('Carrier 2 Signal');  grid on;

subplot(5,1,4);
plot(t, modulated);
title('FSK Modulated Signal','b', 'LineWidth',1.5); grid on;

subplot(5,1,5);
plot(t, demod_signal, 'b', LineWidth=1.5);
title('FSK Demodulated Signal'); ylim([-0.5 1.5]); grid on;
disp(['Original bits:     ' num2str(data)]);
disp(['Demodulated bits:  ' num2str(demodulated_bits)]);
```

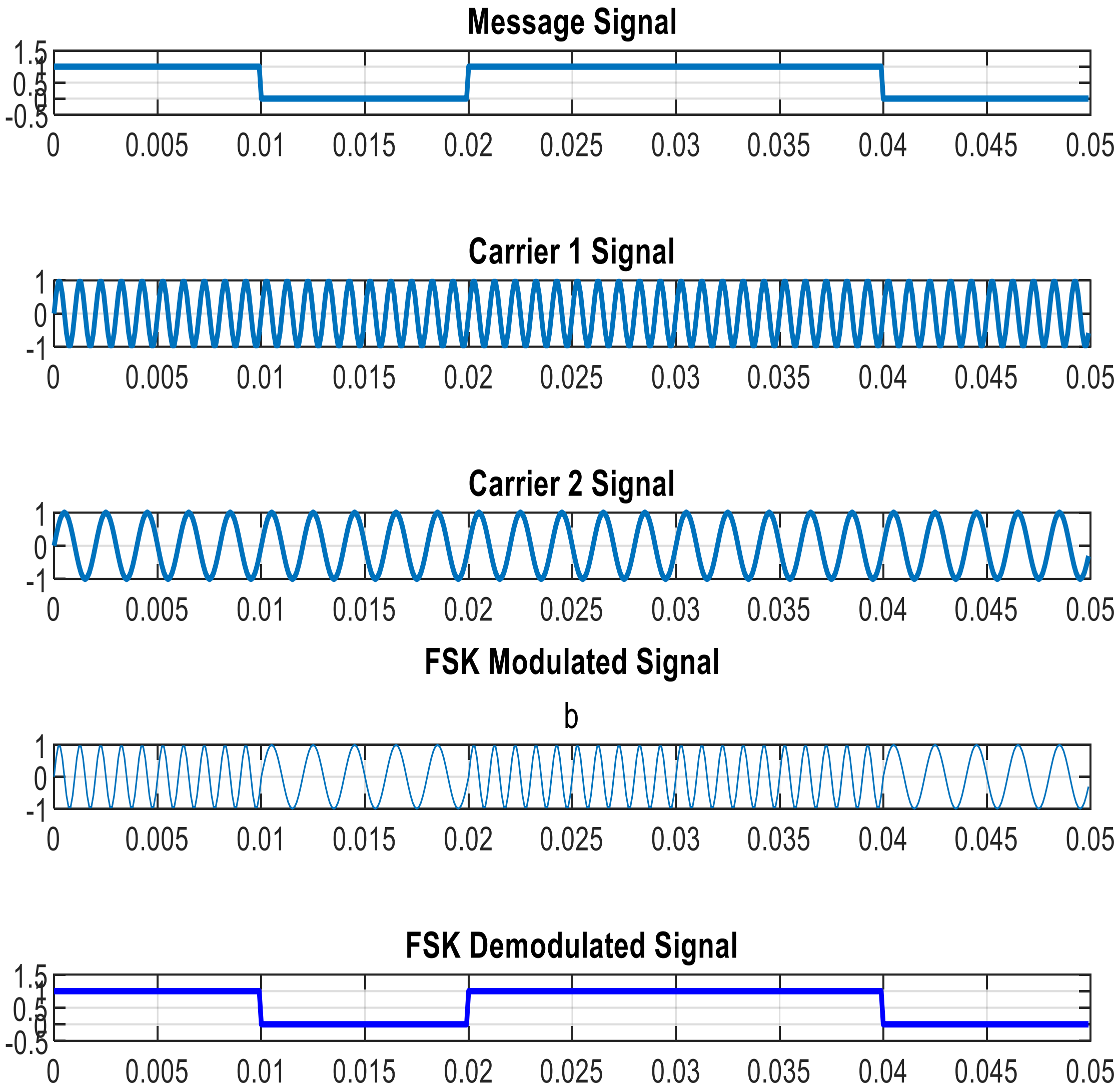
PROGRAM FOR PSK

```
clc; clear; close all;
% Parameters
data = [1 0 1 1 0];
Fs = 10000;              % Sampling frequency
Tb = 0.005;
t_bit = 0:1/Fs:Tb-1/Fs;
Fc = 1000;               % Carrier frequency
% Initialize
msg_signal = [];
modulated = [];
t = [];
```

FSK OUTPUT:

Original bits: 1 0 1 1 0

Demodulated bits: 1 0 1 1 0



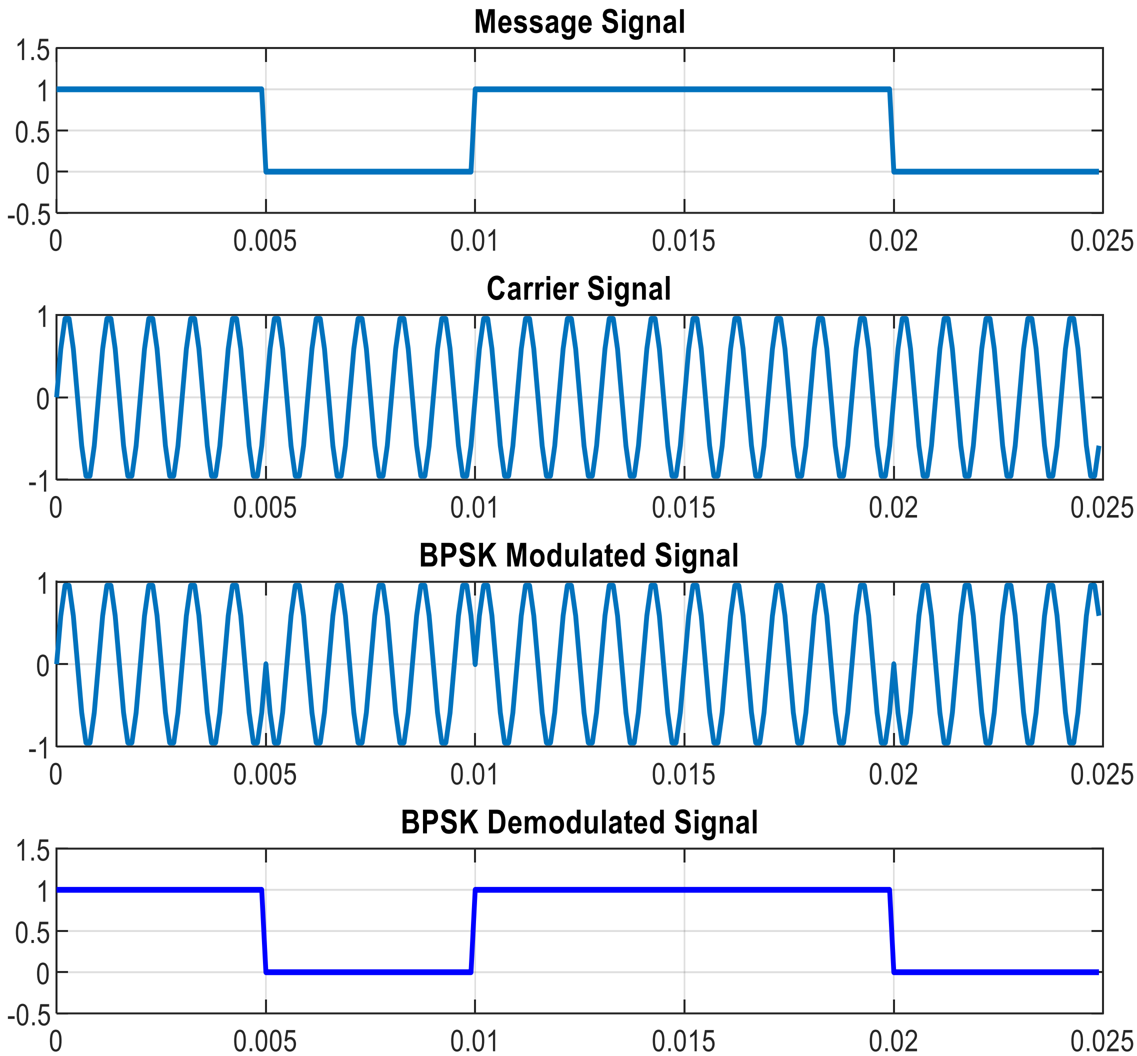
```
% Modulation
for bit = data
    msg = bit * ones(1, length(t_bit));
    if bit == 1
        car = sin(2*pi*Fc*t_bit);      % 0 phase
    else
        car = -sin(2*pi*Fc*t_bit);     % 180° phase shift
    end
    mod = car;
        msg_signal = [msg_signal msg];
    modulated = [modulated mod];
    t = [t t_bit + length(t)/Fs];
end
 car1 = sin(2*pi*Fc*t);
% Coherent Demodulation
demod_raw = modulated .* cos(2*pi*Fc*t);
% Low-pass filter
N = round(Fs*Tb/10);
demod_filtered = filter(ones(1,N)/N, 1, demod_raw);
% Threshold detection
demodulated_bits = [];
demod_signal = [];
samples_per_bit = length(t_bit);
for i = 1:length(data)
    segment = demod_filtered((i-1)*samples_per_bit + 1 : i*samples_per_bit);
    avg = mean(segment);
    bit = avg > 0;
    demodulated_bits = [demodulated_bits bit];
    demod_signal = [demod_signal bit * ones(1, samples_per_bit)];
end
% Plotting
figure;
subplot(4,1,1);
plot(t, msg_signal, 'LineWidth', 1.5);
```

```
title('Message Signal'); ylim([-0.5 1.5]); grid on;
subplot(4,1,2);
plot(t, car1, 'LineWidth', 1.5);
title('Carrier Signal'); grid on;

subplot(4,1,3);
plot(t, modulated,'LineWidth', 1.5);
title('BPSK Modulated Signal'); grid on;
subplot(4,1,4);
plot(t, demod_signal, 'b','LineWidth', 1.5);
title('BPSK Demodulated Signal'); ylim([-0.5 1.5]); grid on;
disp(['Original bits:     ' num2str(data)]);
disp(['Demodulated bits:  ' num2str(demodulated_bits)]);
```

PSK OUTPUT:

Original bits: 1 0 1 1 0

Demodulated bits: 1 0 1 1 0



POST-LAB QUESTIONS:

1. Plot Constellation diagram for ASK signal.
2. Obtain FSK Constellation with binary mapping.
3. Generate the BPSK signal for the data 11101110.
4. How did the frequency separation affect signal clarity?
5. What was the observed difference between ASK and AM?

RESULT:

Thus the ASK PSK and FSK modulation and demodulation are performed using MATLAB.

Ex. No:9

SIMULATION OF QPSK AND QAM GENERATION AND DETECTION SCHEMES

AIM:

To perform generation and detection of QPSK and QAM schemes using MATLAB.

SOFTWARE REQUIRED:

MATLAB

PRE LAB QUESTIONS:

1. Define QPSK.
2. How many bits are represented by each QPSK symbol?
3. What is the bandwidth requirement of QPSK compared to BPSK?
4. Summarize the principle of QAM.
5. What is the significance of Gray coding in QAM?

THEORY :

Quadrature Phase Shift Keying (QPSK) is a digital modulation technique where data is transmitted by varying the phase of the carrier signal. QPSK uses four distinct phase shifts (e.g., 45°, 135°, 225°, and 315°), allowing it to encode two bits per symbol. This improves bandwidth efficiency compared to Binary PSK (BPSK), while maintaining a similar noise performance. QPSK is widely used in wireless systems, satellite communications, and 4G/5G technologies.

Quadrature Amplitude Modulation (QAM) combines both amplitude and phase modulation to transmit data. Unlike QPSK, which varies only the phase, QAM varies both the amplitude and the phase of the carrier. Common QAM schemes like 16-QAM, 64-QAM, or 256-QAM can carry 4, 6, or 8 bits per symbol respectively. QAM offers high spectral efficiency, making it ideal for high-speed data transmission in broadband systems. However, it is more sensitive to noise and requires better signal quality than QPSK.

ALGORITHM:

QPSK:

1. Input Bitstream Generation:Generate a random binary sequence and map 0 → -1 to prepare for BPSK-style modulation.
2. Bitstream Separation and Upsampling: Separate the bitstream into odd (I-channel) and even (Q-channel) bits, then upsample both to match the sinusoidal carrier length.

3. QPSK Modulation: Multiply I and Q streams with cosine and sine carriers respectively, and add them to form the QPSK modulated signal.
4. Demodulation via Correlation: Multiply the received QPSK signal with the same carriers, integrate (sum) over symbol duration, and make bit decisions based on sign.
5. Output and Visualization: Reconstruct the demodulated bitstream, compare it with the original bits, plot all signals, and display the constellation diagram.

QAM:

1. Generate random bits and group them into symbols based on $\log_2(M)$.
2. Map symbols to QAM constellation points using predefined I/Q values.
3. Transmit symbols through an AWGN channel at a specified SNR.
4. Detect received symbols using thresholding based on I/Q signs.
5. Plot constellations to visualize transmission and reception.

PROGRAM FOR QPSK;

```
clc;
clear all;
close all;
 ---- USER INPUT ----
cvx = input('Enter Length of Random Bit Sequence (Even Number): ');
if mod(cvx,2) ~= 0
   error('Bit length must be even for QPSK.');
end
% ---- MESSAGE GENERATION ----
d = round(rand(1,cvx));  % Binary sequence
disp('Input Bit Sequence:');
disp(d);
l = cvx;
x = 0:0.01:l*2*pi;       % Time vector
cc = cos(x);             % Cosine carrier
cs = cos(x + pi/2);      % Sine carrier (90 deg)
k = length(cc);
```

```
k1 = floor(k / l);       % Samples per bit
% ---- BPSK Mapping ----
d(d==0) = -1;
% ---- SEPARATE I & Q STREAMS ----
dd1 = zeros(1,l); dd2 = zeros(1,l);
i = 1; j = 1;
while (i < l) && (j < l)
   dd1(j:j+1) = d(i);       % Odd bits
   dd2(j:j+1) = d(i+1);     % Even bits
   i = i + 2;
   j = j + 2;
end
% ---- UPSAMPLING TO MATCH SINUSOIDAL PERIOD ----
t = 1;
for i = 1:l
   for j = 1:k1
      dd(t) = d(i);
      d1(t) = dd1(i);
      d2(t) = dd2(i);
      t = t + 1;
   end
end
% ---- MODULATION ----
len = length(d1);
cc = cc(1:len);
cs = cs(1:len);
qcc = cc .* d1;     % I-component
qcs = cs .* d2;     % Q-component
qp = qcc + qcs;     % QPSK modulated signal
% ---- DEMODULATION ----
```

```
numSymbols = l / 2;
demodBits = zeros(1, l);
for sym = 1:numSymbols
    idx_start = (sym-1)*2*k1 + 1;
    idx_end = idx_start + 2*k1 - 1;

    symbol = qp(idx_start:idx_end);
    I = symbol .* cc(idx_start:idx_end);
    Q = symbol .* cs(idx_start:idx_end);
        I_sum = sum(I);
    Q_sum = sum(Q);
        demodBits(2*sym - 1) = I_sum > 0;
    demodBits(2*sym)     = Q_sum > 0;
end
% ---- PLOTTING ----
figure;
subplot(7,1,1);
stairs(dd,'LineWidth',1.5);
xlim([0 5000]);
ylim([-2 2]); grid on;
title('Input Bit Stream');
subplot(7,1,2);
stairs(d1,'LineWidth',1.5);
xlim([0 5000]);
ylim([-2 2]); grid on;
title('Odd Bit Stream (I)');
subplot(7,1,3);
stairs(d2,'LineWidth',1.5);
xlim([0 5000]);
ylim([-2 2]); grid on;
```

QPSK OUTPUT:

Enter Length of Random Bit Sequence (Even Number): 8

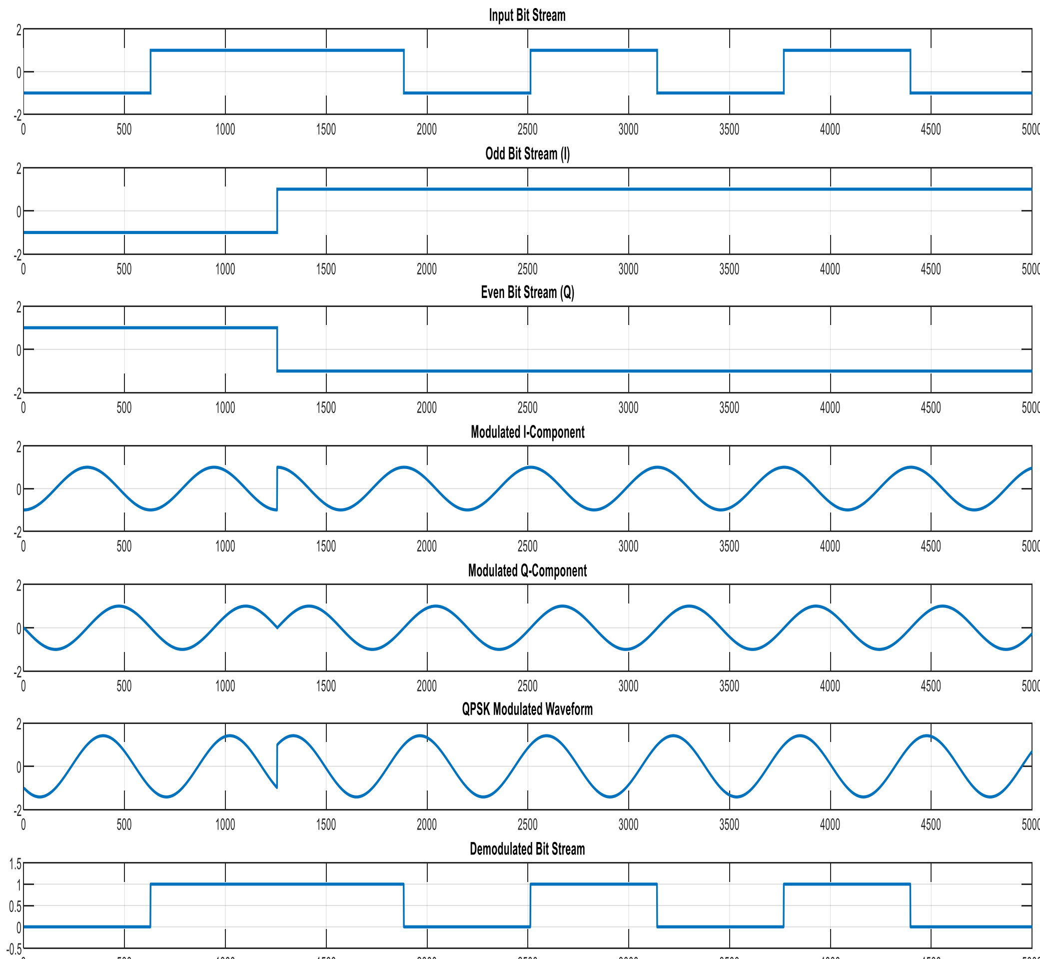
Input Bit Sequence:

0 1 1 0 1 0 1 0

Demodulated Bits:

0 1 1 0 1 0 1 0

Number of Bit Errors: 0

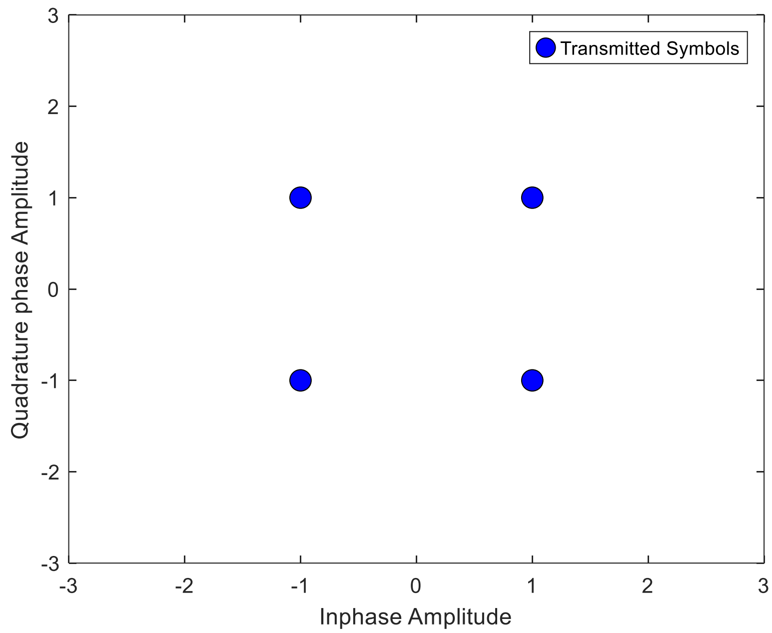


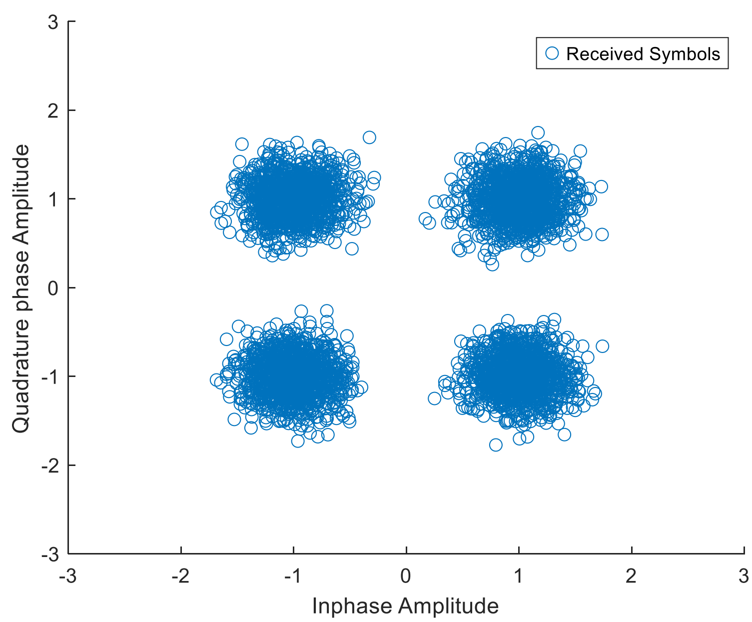
```
title('Even Bit Stream (Q)');
subplot(7,1,4);
plot(qcc,'LineWidth',1.5);
xlim([0 5000]);
ylim([-2 2]); grid on;
title('Modulated I-Component');

subplot(7,1,5);
plot(qcs,'LineWidth',1.5);
xlim([0 5000]);
ylim([-2 2]); grid on;
title('Modulated Q-Component');
subplot(7,1,6);
plot(qp,'LineWidth',1.5);
xlim([0 5000]);
ylim([-2 2]); grid on;
title('QPSK Modulated Waveform');
subplot(7,1,7);
% Stretch demodBits to same time axis as 'dd' for fair comparison
demodPlot = repelem(demodBits, k1);
plot(demodPlot(1:length(dd)), 'LineWidth', 1.5);
xlim([0 5000]);
ylim([-0.5 1.5]); grid on;
title('Demodulated Bit Stream');
disp(['Demodulated Bits:']);
disp(demodBits);
% ---- BIT ERROR CHECK ----
originalBits = (d == 1);  % Convert -1 to 0
num_errors = sum(originalBits ~= demodBits);
disp(['Number of Bit Errors: ', num2str(num_errors)]);
```

PROGRAM FOR QAM:

```
clc; clear all; close all;
M = 4;
k = log2(M);  % Number of bits per symbol
rng(10)
N =10000;
InputBits = randi([0 1],1,N);
InputSymbol_matrix = reshape(InputBits,length(InputBits)/k,k);
InputSymbols_decimal = bi2de(InputSymbol_matrix);
for n= 1:N/k
if InputSymbols_decimal(n)==0
QAM(n)= complex(1,1);
elseif InputSymbols_decimal(n)==1
QAM(n)= complex(-1,1);
elseif InputSymbols_decimal(n)==2
QAM(n)= complex(1,-1);
else
QAM(n)= complex(-1,-1);
end
end
%Transmission of 4QAM data over AWGN channel
snrdB = 10;
Y=awgn(QAM,snrdB); %received signal
%Threshold Detection
for n= 1:N/k
if (real(Y(n))>0 && imag(Y(n))>0)
Z(n)=complex(1,1);
elseif (real(Y(n))>0 && imag(Y(n))<0)
Z(n)=complex(1,-1);
elseif (real(Y(n))<0 && imag(Y(n))>0)
Z(n)=complex(-1,1);
```

QAM OUTPUT:


Transmitted Symbols
Quadrature phase Amplitude
Inphase Amplitude
3
2
1
0
-1
-2
-3


Received Symbols
Quadrature phase Amplitude
Inphase Amplitude
3
2
1
0
-1
-2
-3

```
else
Z(n)=complex(-1,-1);
end
end
figure(1)
plot(QAM,'ok','MarkerSize',10,'MarkerFaceColor', 'b')
xlim([-3, 3]);
ylim([-3, 3]);
xlabel('Inphase Amplitude')
ylabel ('Quadrature phase Amplitude')
legend('Transmitted Symbols')
figure(2)
scatter(real(Y), imag(Y))
xlim([-3, 3]);
ylim([-3, 3]);
xlabel('Inphase Amplitude')
ylabel ('Quadrature phase Amplitude')
legend('Received Symbols')
```

OST LAB QUESTIONS:

1. How many distinct phases are used in QPSK? List their values in degrees?
2. How did QPSK's performance compare to other modulation schemes?
3. What is the phase separation between adjacent QPSK symbols?
4. How was the QAM constellation affected by noise?
5. What distortion was observed in higher-order QAM?

RESULT:

The quadrature phase shift keying and quadrature amplitude modulation and demodulation was performed and the corresponding plots are obtained.

Ex. No:10 (a)

SIMULATION OF LINEAR BLOCK ERROR CONTROL CODING SCHEME

AIM:

To write a program in MATLAB for Linear Block coding technique.

SOFTWARE REQUIRED:

MATLAB

PRE-LAB QUESTIONS:

1. What is linear block coding ?
2. Define error controlling technique.
3. How to compute syndrome table?
4. What is a Hamming code?
5. What are (n, k) block codes?

THEORY:

In coding theory, a linear code is an error-correcting code for which any linear combination of code words is also a codeword. Linear codes are traditionally partitioned into block codes and convolutional codes, although turbo codes can be seen as a hybrid of these two types. Linear codes allow for more efficient encoding and decoding algorithms than other codes. Linear codes are used in forward error correction and are applied in methods for transmitting symbols (e.g., bits) on a communications channel so that, if errors occur in the communication, some errors can be corrected or detected by the recipient of a message block. The codewords in a linear block code are blocks of symbols which are encoded using more symbols than the original value to be sent. A linear code of length *n* transmits blocks containing *n* symbols.

ALGORITHM:

Encoder:

1. Get the Generator Matrix of Linear Block codes with n=7 and k=4
2. Obtain the data which is going to be encoded .

OUTPUT:

enter the generator matrix[1 0 0 0 1 0 1;0 1 0 0 1 1 1;0 0 1 0 1 1 0; 0 0 0 1 0 1 1]

```
g =

     1     0     0     0     1     0     1
     0     1     0     0     1     1     1
     0     0     1     0     1     1     0
     0     0     0     1     0     1     1

G=
the order of the linear block for the given generator matrix:

n =

     7

k =

     4

the possible codewords are:

c =
     0     0     0     0     0     0     0
     0     0     0     1     0     1     1
     0     0     1     0     1     1     0
     0     0     1     1     1     0     1
     0     1     0     0     1     1     1
     0     1     0     1     1     0     0
     0     1     1     0     0     0     1
     0     1     1     1     0     1     0
     1     0     0     0     1     0     1
     1     0     0     1     1     1     0
     1     0     1     0     0     1     1
     1     0     1     1     0     0     0
     1     1     0     0     0     1     0
     1     1     0     1     0     0     1
     1     1     1     0     1     0     0
     1     1     1     1     1     1     1

the minimum hamming distance dmin for given block code:

d_min =

     3
```

3. Multiply the data with Generator Matrix to obtain the encoded sequence.
4. Calculate the minimum distance between the adjacent codes.

Decoder:

1. Get the received code word.
2. Generate ‘h’ matrix from Generator matrix.
3. Multiply the received codeword with the transpose of ‘h’ matrix to obtain the syndrome.
4. If syndrome =0 then display the correct code word else correct the error and display the code word.

PROGRAM:

```
clc;
clear all;
%Input generator matrix
g=input('enter the generator matrix')
disp('G=')
disp('the order of the linear block for the given generator matrix:')
[n,k]=size(transpose(g))
for i=1:2^k
   for j=k:-1:1
    if rem(i-1,2^(-j+k+1))>=2^(-j+k)
        u(i,j)=1;
    else u(i,j)=0;
    end
   end
end
u;
disp('the possible codewords are:');
c=rem(u*g,2)
disp('the minimum hamming distance dmin for given block code:')
d_min=min(sum((c(2:2^k,:))'))
```

```
enter the received codeword:[1 0 0 0 1 0 0]

r =

     1     0     0     0     1     0     0

Hamming code

ht =

     1     0     1
     1     1     1
     1     1     0
     0     1     1
     1     0     0
     0     1     0
     0     0     1

syndrome of given codeword is:

s =

     0     0     1

Warning: Colon operands must be real scalars. This warning will become an error in a future release.
> In blockcode (line 28)

the error is in bit:

i =

     7

the corrected codeword is:

r =

     1     0     0     0     1     0     1
```

```
%code word
r=input('enter the received codeword:')
p=[g(:,n-k+2:n)];
h=[transpose(p),eye(n-k)];
disp('Hamming code')
ht=transpose(h)
disp('syndrome of given codeword is:')
s=rem(r*ht,2)
for i=1:1:size(ht)
   if(ht(i,1:3)==s)
      r(i)=1-r(i);
      break;
   end
end
disp('the error is in bit:');
i
disp('the corrected codeword is:')
r
```

POST LAB QUESTIONS :

1. Summarize how block coding help with error detection and correction?
2. How was syndrome decoding performed?
3. Mention the observed trade-off between redundancy and efficiency?
4. How single error correction is achieved using Hamming codes?
5. What was the observed trade-off between redundancy and efficiency?

RESULT

Thus the simulation of linear block code (7, 4) was done using MATLAB

Ex.No:10(b) SIMULATION OF CYCLIC ERROR CONTROL CODING SCHEME

AIM:

To simulate an error control coding based on cyclic codes using MATLAB.

SOFTWARE USED:

MATLAB

PRE-LAB QUESTIONS:

1. What is error control coding ?
2. Mention the significance of the cyclic code
3. Compare linear block code with cyclic code.
4. What is a generator polynomial?
5. List the steps in encoding and decoding cyclic codes.

ALGORITHM:

1. Generate the possible polynomial for cyclic codes.
2. Choose any one of the polynomial to obtain the Generator matrix.
3. Get the input which is going to be encoded.
4. Multiply the data with the Generator Matrix to obtain the encoded sequence.
5. Decode the received sequence using the Generator Matrix again.

PROGRAM:

```
close all; clear all; clc;
%  Generator Matrix
p=cyclpoly(7,4,'all')
x=input('Choose the polynomial');
v=p(x,:);
[pm,gm]=cyclgen(7,v,'sys'); %Generate non systematic codes
input=0:1:15;
u=dec2bin(input);
```

OUTPUT:

```
p =

     1     0     1     1
     1     1     0     1

Choose the polynomial1
The Possible Codewords are :

c =

     0     0     0     0     0     0     0
     0     1     1     0     0     0     1
     1     1     0     0     0     1     0
     1     0     1     0     0     1     1
     1     1     1     0     1     0     0
     1     0     0     0     1     0     1
     0     0     1     0     1     1     0
     0     1     0     0     1     1     1
     1     0     1     1     0     0     0
     1     1     0     1     0     0     1
     0     1     1     1     0     1     0
     0     0     0     1     0     1     1
     0     1     0     1     1     0     0
     0     0     1     1     1     0     1
     1     0     0     1     1     1     0
     1     1     1     1     1     1     1

The Minimum Distance between the code words is=

l1 =

     3

message to be encoded

m =

     1     0     1     1

the decoded sequence is

msg =

     1     0     1     1
```

```
disp('The Possible Codewords are :')
c = rem(u*gm,2)
for i=1:15
l(i)=length(find(c(i,:)~=c(i+1,:)));
end
disp('The Minimum Distance between the code words is= ')
l1=min(l)
disp('message to be encoded')
m=[1 0 1 1]
r= encode(m,7,4);
% Decoding using Toolbox
 rs=[1 1 1 0 1 1 1];
disp('the decoded sequence is');
msg=decode(r,7,4)
```

POST LAB QUESTIONS:

1. What makes cyclic codes a subset of linear block codes?
2. How was the CRC implemented?
3. How did you verify the correctness of decoded data?
4. What generator polynomial was used?
5. Where are cyclic codes commonly used?

RESULT :

Thus the cyclic code is used to implement the error detection and perform error correction in the receiver on decoding the received message.

Ex. No:11

SIMULATION OF DIFFERENTIAL PHASE SHIFT KEYING GENERATION AND DETECTION

AIM:

To perform the differential phase shift keying (dpsk) generation and detection using MATLAB.

SOFTWARE REQUIRED:

MATLAB

PRE-LAB QUESTIONS :

1. What is Differential PSK and how does it differ from standard PSK?
2. Why is DPSK preferred in systems without carrier synchronization?
3. How does DPSK encode binary information?
4. What is the advantage of using differential encoding?
5. What are the limitations of DPSK?

THEORY:

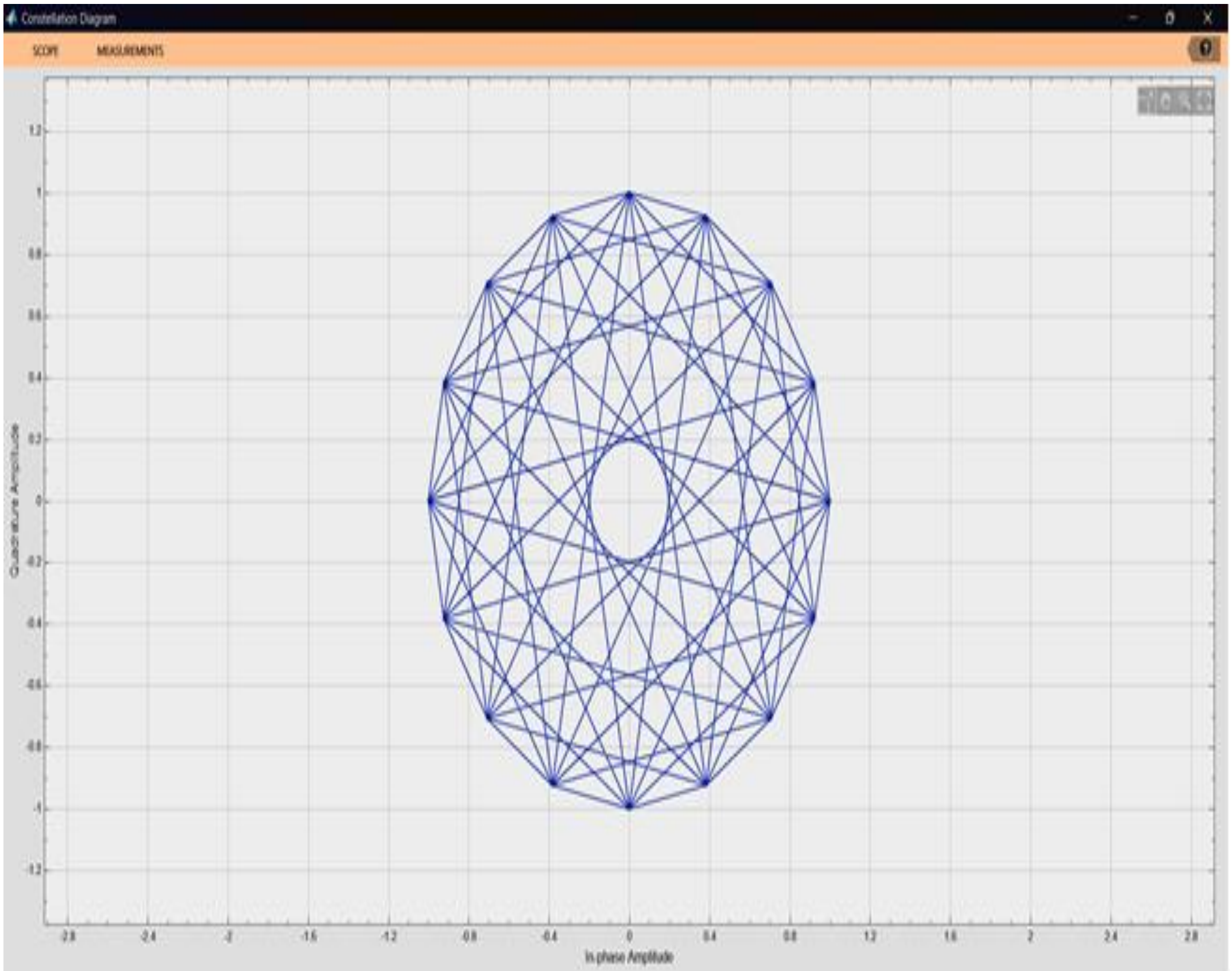
Differential Phase Shift Keying (DPSK) is a digital modulation technique where data is encoded in the phase difference between successive signal elements, rather than the absolute phase. Unlike conventional Phase Shift Keying (PSK), DPSK does not require a reference oscillator at the receiver, making it more robust to phase ambiguities and simplifying receiver design. In DPSK, a binary '1' might represent a phase shift (e.g., 180°), while a '0' indicates no phase change. This relative encoding reduces synchronization complexity. DPSK is widely used in optical and wireless communications due to its efficiency and resilience to noise. However, its performance can be slightly inferior to coherent detection schemes in terms of bit error rate (BER) under certain conditions.

ALGORITHM:

1. For Initialization set modulation order M = 4 (i.e., 4-DPSK).
2. Generate a random message vector x of 1000 symbols, each in the range [0, M−1].
3. Use dpskmod to apply DPSK modulation on the message x with a phase rotation of π/8.
4. Create a constellation diagram object and display the DPSK signal y on the constellation diagram.

OUTPUT :

No. of Symbol error = 0



5. Assign y to txSig as the transmitted signal.Simulate a random phase rotation on the received signal rxSig using a random phase offset.
6. Use dpskdemod to demodulate the received signal rxSig using DPSK.
7. Compute symbol errors between original and demodulated data using symerr.
8. Display the number of symbol errors.

%PROGRAM FOR DPSK:

```
clc;
clear all; close all;
M = 4; % Alphabet size
x = randi([0 M-1],1000,1); % Random message
y = dpskmod(x,M,pi/8);
cd = comm.ConstellationDiagram( ...
   ShowTrajectory=true, ...
   ShowReferenceConstellation=false);
cd(y)
txSig = y;      % Modulate
rxSig = txSig*exp(2i*pi*rand());
dataOut = dpskdemod(rxSig,M);
errs = symerr(x,dataOut)
errs = symerr(x(2:end),x(2:end))
disp(['No. of Symbol error =', num2str(errs)])
```

POST LAB QUESTIONS:

1. What is non coherent detection?
2. Illustrate the constellation diagram of DPSK.
3. Was the demodulated signal more or less reliable than BPSK?
4. How does DPSK modulation differ from conventional PSK modulation ?
5. What is the primary component used in a DPSK modulator ?

RESULT :

The differential phase shift keying modulation and demodulation was performed and plotted the constellation.