ST.ANNE'S

COLLEGE OF ENGINEERING AND TECHNOLOGY

(Approved by AICTE, New Delhi. Affiliated to Anna University, Chennai)

(An ISO 9001 **:**2015 Certified Institution)

ANGUCHETTYPLAYAM, PANRUTI – 607 106.

DEPARTMENT OF

ELECTRONICS AND COMMUNICATION ENGINEERING

LAB MANUAL

ACADEMIC YEAR: AUG 2022 – NOV 2022

SUBJECT CODE/NAME:

EC8561 COMMUNICATION SYSTEMS LABORATORY

YEAR/SEM: III/V   `   **BATCH**: 2019 - 2023

AS PER ANNA UNIVERSITY, CHENNAI REGULATION 2017

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EC8561   COMMUNICATIONSYSTEMSLABORATORY   L T P C 0 0 4

LIST OFEXPERIMENTS:

1. Signal Sampling and reconstruction
2. Time Division Multiplexing
3. AM Modulator and Demodulator
4. FM Modulator and Demodulator
5. Pulse Code Modulation and Demodulation
6. Delta Modulation and Demodulation
7. Line coding schemes
8. Simulation of ASK, FSK, and BPSK generation schemes
9. Simulation of DPSK, QPSK and QAM generation schemes
10. Simulation of signal constellations of BPSK, QPSK and QAM
11. Simulation of ASK, FSK and BPSK detection schemes
12. Simulation of Linear Block and Cyclic error control coding schemes
13. Simulation of Convolutional coding scheme
14. Communication link simulation   **TOTAL: 60 PERIODS**

BLOCK DIAGRAM OF SAMPLING PROCESS

Basic Sampling Process
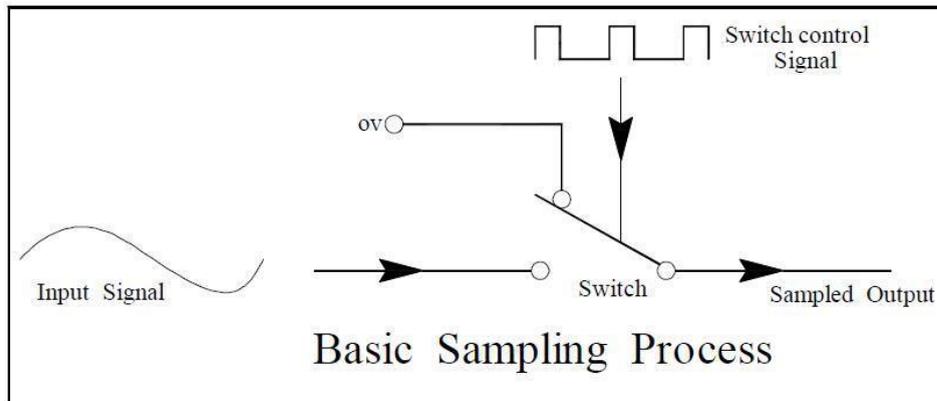
BLOCK DIAGRAM OF RECONSTRUCTION PROCESS
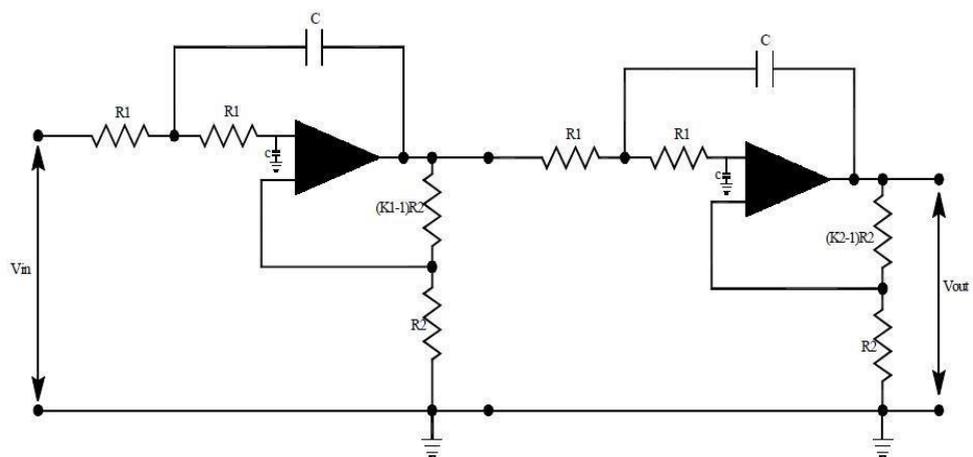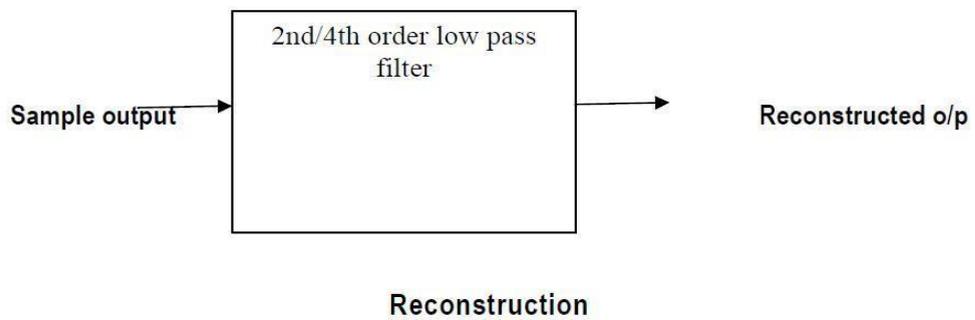
Reconstruction

FOURTH ORDER BUTTERWORTH LOW PASS FILTER

1. ANALOG SAMPLING ANDRECONSTRUCTION

AIM:

To study the process of sampling and to reconstruct the signals at the receiver using filters.

APPARATUS REQUIRED:

1. Communication trainerkit:
2. PowerCable
3. Patchcords.
4. CRO(60MHz)

THEORY OF SAMPLING

In analog communication systems like AM, FM, the instantaneous value of the information signal is used to change certain parameter of the carrier signal.Pulse modulation systems differ from these systems in a way that transmit a limited no.of discrete states of a signal at predetermined time sampling can be defined as measuring the value of an information signal at predetermined time intervals. The rate at which the signal is sampled is known as the sampling rate or sampling frequency. It is the major parameter which decides the quality of the reproduced signal. If the signal is sampled quite frequently (whose limit is specified by Nyquist Criterion), then it can be reproduced exactly at the receiver with no distortion.

Needs of Sampling Process

It however the message signal happens to be converted into digital form before it can be transmitted by digital means. The sampling process is the first process performed in analog to digital conversion. Two other process, quantizing and encoding are also involved in this conversion.

NYQUIST CRITERION (SAMPLING THEOREM)

The Nyquist Criterion states that a continuous signal band limited to Fm Hz can be completely represented by and reconstructed from the sample taken at a rate greater than or equal to 2Fm samples/second. This minimum sampling frequency is called as NYQUIST RATE i.e. for faithful reproduction of information signal $Fs \geq 2Fm$.

ALIASING

If the signal is sampled at a rate lower than stated by Nyquist criterion, then there is an overlap between the information signal and the sidebands of the harmonics. Thus the higher and the lower frequency components get mixed and causes unwanted signals to appear at the demodulator output. This phenomenon is turned as Aliasing or Fold over Distortion.To avoid aliasing using anti aliasing filter or the signal must satisfied the nyquist criterion ($Fs \geq 2$ Fm)

LOW PASS FILTERS

The PAM system the message is recovered by a low pass filter. The type of filter used is very important, as the signal above the cut-off frequency would affect the recovered signal if they are not attenuated sufficiently.

MODEL GRAPH
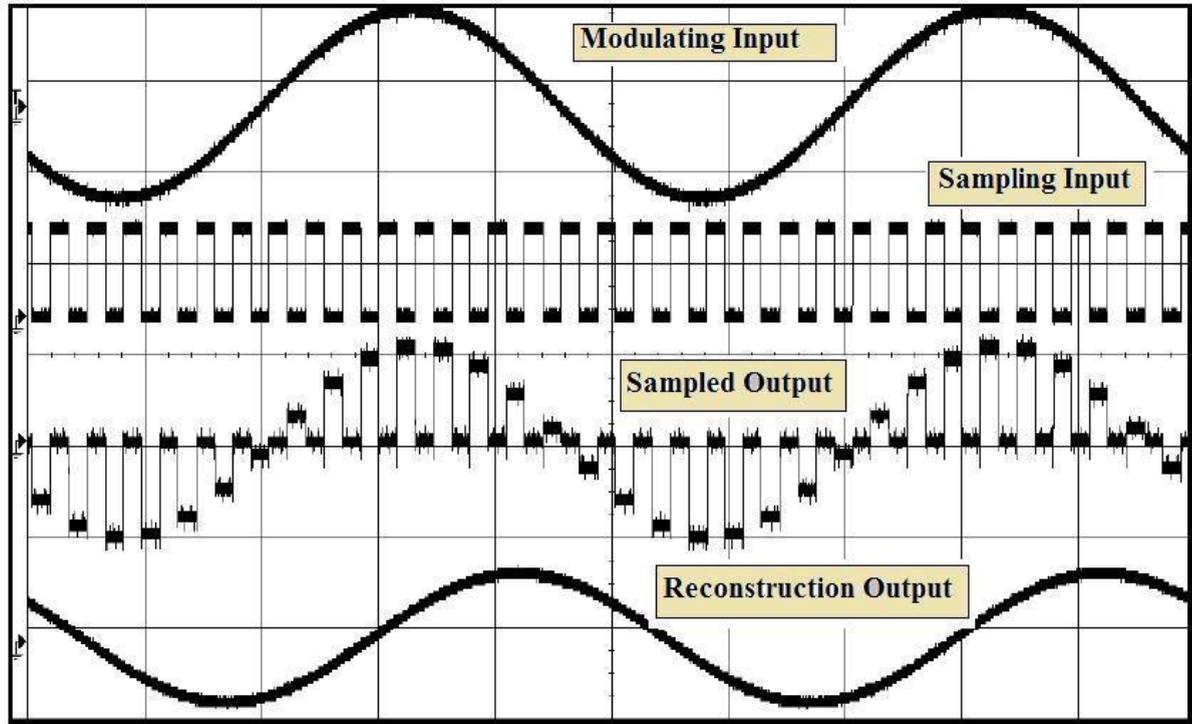
TABULATION

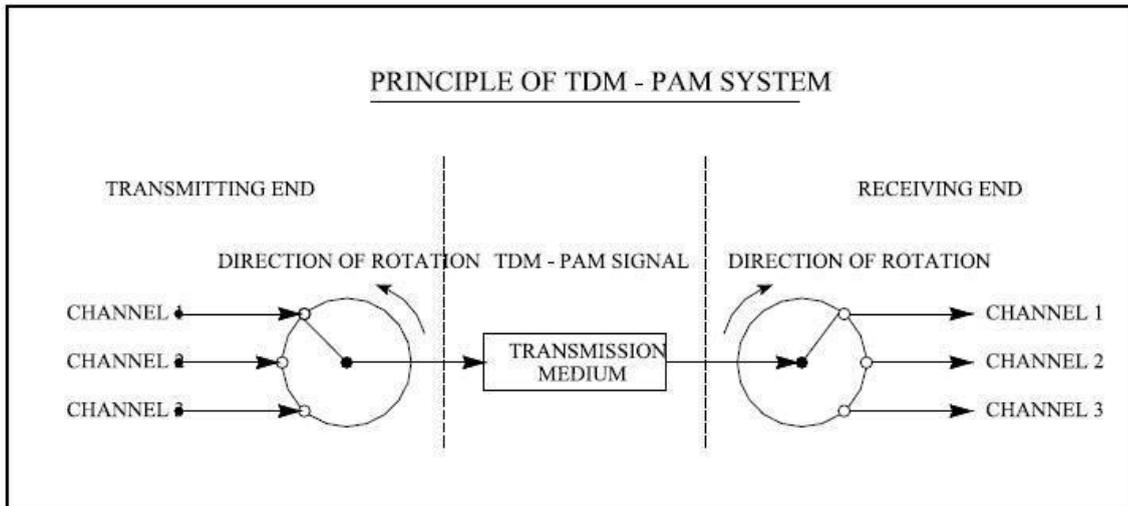| S.NO	SIGNAL	AMPLITUDE(V)	TIME(ms)	FREQUENCY(HZ)
1	Modulating Input			
2	Sampling Input			
3	Sampled Output			
4	Flat top output			
5	Sample & Hold output			
6	Reconstruction Output			

1. Connection are given as per the given blockdiagram.
2. To give an modulating input and sampled input (square wave form) to the inputblock.
3. To verify the output usingCRO.
4. The output as given to the input of de modulated block and taken the outputreading
5. Plot thegraph

RESULT:

   Thus the continuous-time signals are sampled and reconstructed from the samples at the receiver by original signal.
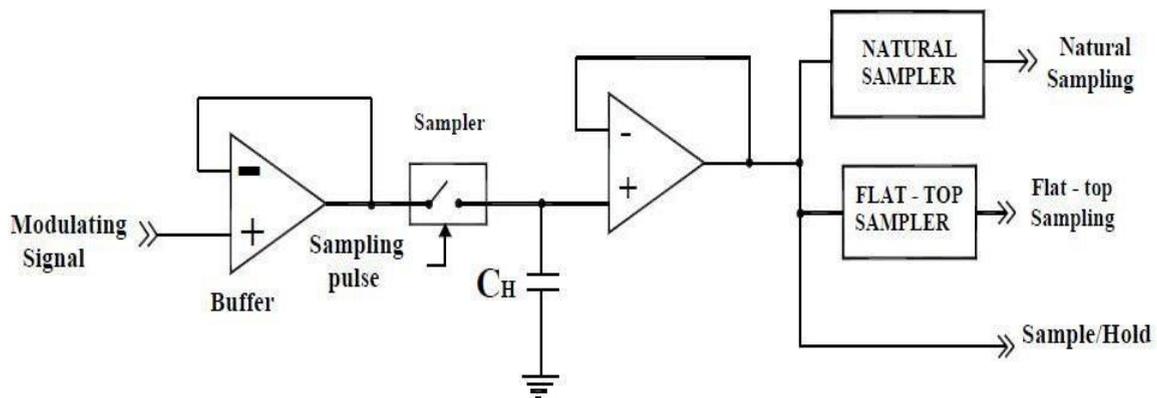
TDM USING PAM, MODULATION & DEMODULATION

PRINCIPLE OF TDM - PAM SYSTEM

TABULATION

S.NO	SIGNAL	AMPLITUDE(V)	TIME(ms)	FREQUENCY(HZ)
1	Modulating Input channel-1			
2	Modulating Input channel-2			
3	Modulating Input channel-3			
4	Mux output			
5	De Modulating Input channel-1			
6	De Modulating Input channel-2			
7	De Modulating Input channel-3			
8	Sync pluse			

BLOCK DIAGRAM FOR PAM MODULATION

2.TDM USING PAM, MODULATION & DEMODULATION

AIM

To Perform the time division multiplexing using PAM Modulation and Demodulation using the trainer kit.

APPARATUS REQUIRED
  i. TDMTrainer.
  ii. CRO.
  iii. Patch Chords.
  iv. CROprobe

THEORY

MULTIPLEXER

Multiplexing is the process of combining signals from different information sources so that they can be transmitted to a common channel. This is under taken by a multiplexer. A digital multiplexer is a combinational circuit that selects data from 2_n input lines (or) group of lines and transmit them through a single output line (or group of lines). Multiplexing is advantageous in cases where it is impracticable and uneconomical to provide separate links for the different information sources. The two most commonly used methods of multiplexing are,
  i. Frequency DivisionMultiplexing
  ii. Time DivisionMultiplexing.

TIME DIVISION MULTIPLEXING

It is the process of taking the samples from different information signals, in time domain so that they can be transmitted over the same channel. The main fact in the TDM technique is that there are large intervals between the message samples. The samples from the other sources are placed with in these time intervals. Thus every sample is separated from other in time domain.

Here, each signal is sampled over one sampling interval and transmitted one after the other along a common channel. But the receiving end has to follow some constraints. i. It must receive and show the signal as the transmitted.
ii. It must start at the same time as the transmitting end and establish electrical contact with the same channel of the inputchannel.

When the two conditions are met then the receiver end is said to be in synchronization with the transmitter end. If the 1st condition is not met then the samples different sources would get mixed out the receiver end and if the 2nd condition is not met then the information from source '1' will be received by same other channel which is not intending to accept the information from that particular channel.

PULSE AMPLITUDE MODULATION

In pulse amplitude modulation, the amplitude of the high frequency pulse is changed in accordance with the amplitude of the message signal. It is very easy to generate and recover pulse amplitude modulated signal. In pulse amplitude modulation, the pulse amplitude is made proportional to the modulating signals amplitude. This is the simplest pulse modulation to create in that a simple sampling of the modulating signal at a periodic rate can be used to generate the pulses, which are subsequently used to modulate a high frequency carrier.
There are three basic sampling techniques used to create a PAM signal.
  1. Naturalsampling
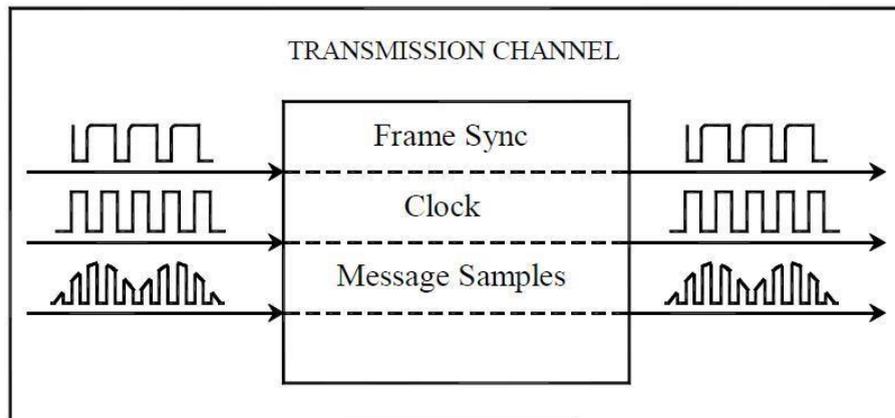 2. Flat-topsampling
  3. Sample andHold
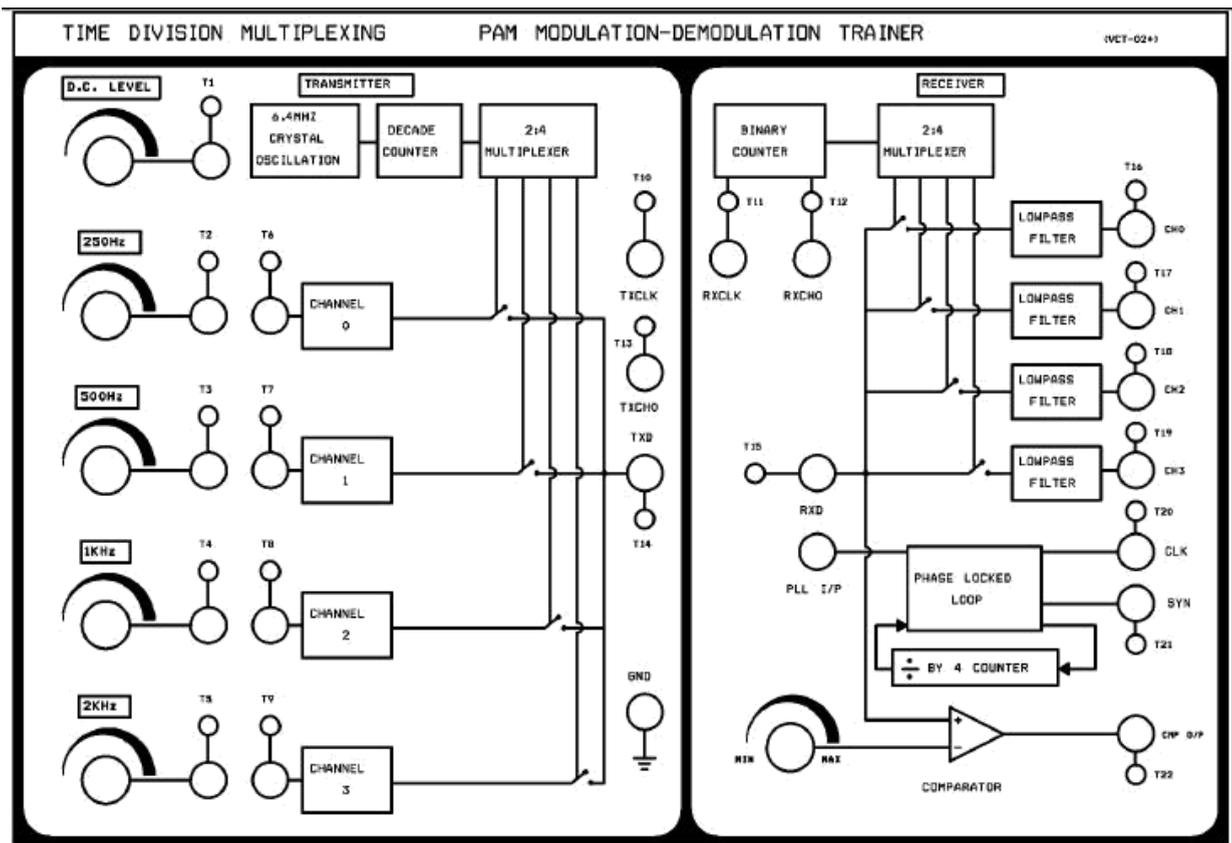
MODEL GRAPH
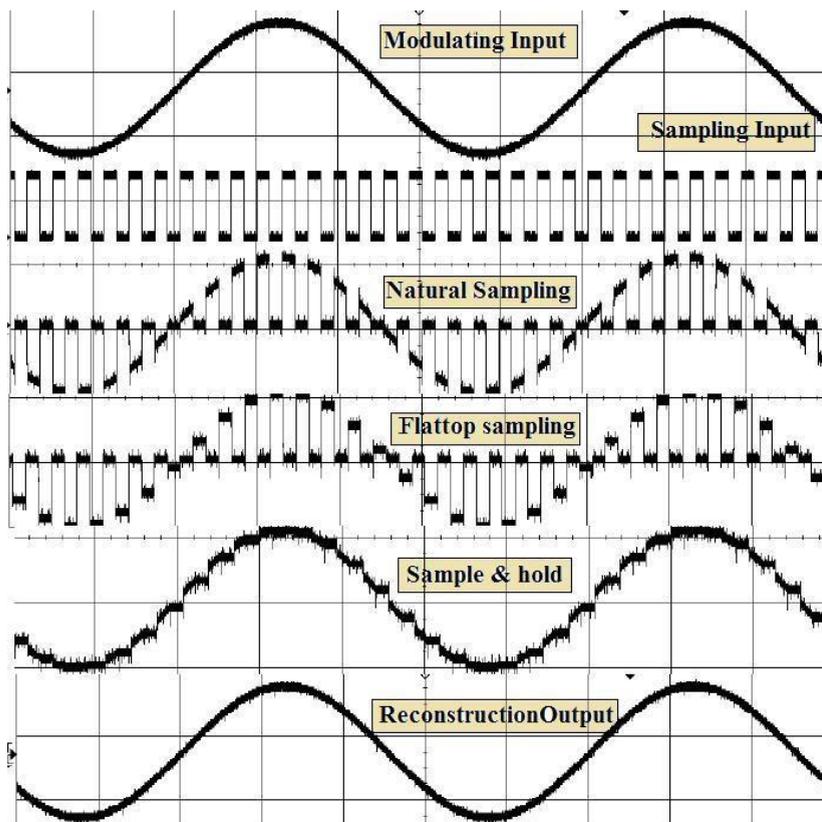
THREE WIRE SYSTEM

This mode of operation provides three links to be given directly from the transmitter end to the receiver end for the transmission of signals. This is illustrated below.

In this mode of operation, the signals from all the channels are taken to the multiplexer and are combined with the carrier signal produced by oscillator and the counter and then it is finally multiplexed and sent through the transmission data signal (TXD) to the receiving data(RXD). The clock signal is sent through CLK channel. The signals at the transmitter side are received at the receiving side at respective points and the signals are demultiplexed and sent to respective channels and the output is viewed out there.
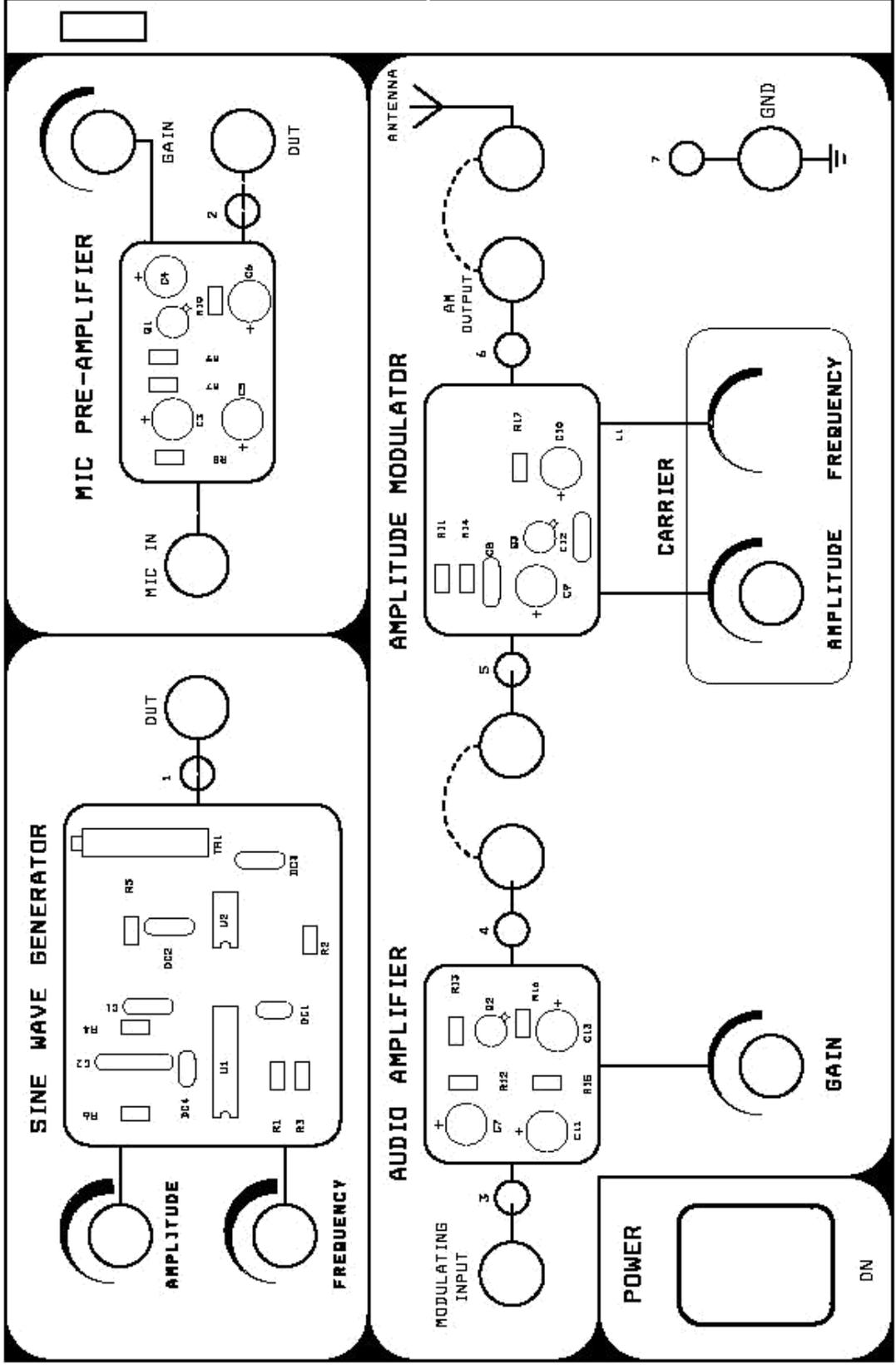
PROCEDURE

1. Switch ON the power supply to theboard.
2. Make initial settings on VCT- 02 asfollows.
    a) Set all sine wave voltages to2V,
    b) Make the wiring connections as in wiring diagram which is provided at the
      end of thisexperiment.
3. Display the multiplexed signal at test point T14 on channel 1 and 250Hz sinewave at test point T2 on channel 2 of oscilloscope, note downwaveforms.
4. Display the 500Hz sinewave at test point T3 on channel 2 in place of 250Hz, identify sampled version of this sinewave in TDM signal and notedown.
5. Similarly observe 1KHz and 2KHz waveforms at test point T4 and T5 respectively on oscilloscope and notedown.
6. Display the TDM waveform (test point T14) on channel 1 and channel synchronization signal (test point T13) on channel 2 of oscilloscope and note downwaveforms.
7. Display 250Hz sinewave at test point T2 on channel 1 and output sinewave at test point T16 on channel 2 of oscilloscope and note downwaveforms.
8. Similarly, observe input and output 500Hz, 1KHz and 2KHz sine waves on oscilloscope and notedown.

RESULT

Thus the Perform time division multiplexing using PAM Modulation and Demodulation using the trainer kit and understand the concept using graph.

AM TRANSMITTER TRAINER (VCT-08)

MIC PRE-AMPLIFIER

GAIN
OUT
MIC IN

SINE WAVE GENERATOR

OUT
AMPLITUDE
FREQUENCY

AMPLITUDE MODULATOR

ANTENNA
AM OUTPUT
GND

CARRIER
AMPLITUDE   FREQUENCY

AUDIO AMPLIFIER

MODULATING INPUT
GAIN

POWER
ON

Vi Microsystems Pvt. Ltd., Chennai-96.

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3. AMPLITUDE MODULATION AND DEMODULATION

AIM

To study the AM Transmission and Reception using AM techniques (sine wave).

APPARATUS REQUIRED
1. AM trainerKit
2. CRO.
3. Patchchords.
4. BNC to P8003cables.

THEORY

In radio transmission, it is necessary to send audio signal (eg. music, speech etc) from a broad casting station over great distances to a receiver. The audio signal cannot be sent directly over the air for appreciable distance. Even if audio signal is converted into electrical signal, the latter cannot be sent very far without employing large amount of power. The energy of a wave is directly proportional to its frequency. At audio frequencies, (20Hz to 20KHz), the signal power is quite small and radiation is not practicable. For it to be practicable, the frequency should be above 20KHz. If audio signal is to be transmitted properly, some means must be devised which will permit transmission to occur at high frequencies while it simultaneously allows the carrying of audio signal. This is achieved by super imposing electrical audio signal on high frequency carrier. The resultant waves are known as modulated waves or radio waves and the process is called "Modulation". At the radio receiver, the audio signal is extracted from the modulated wave by the process called"Demodulation".
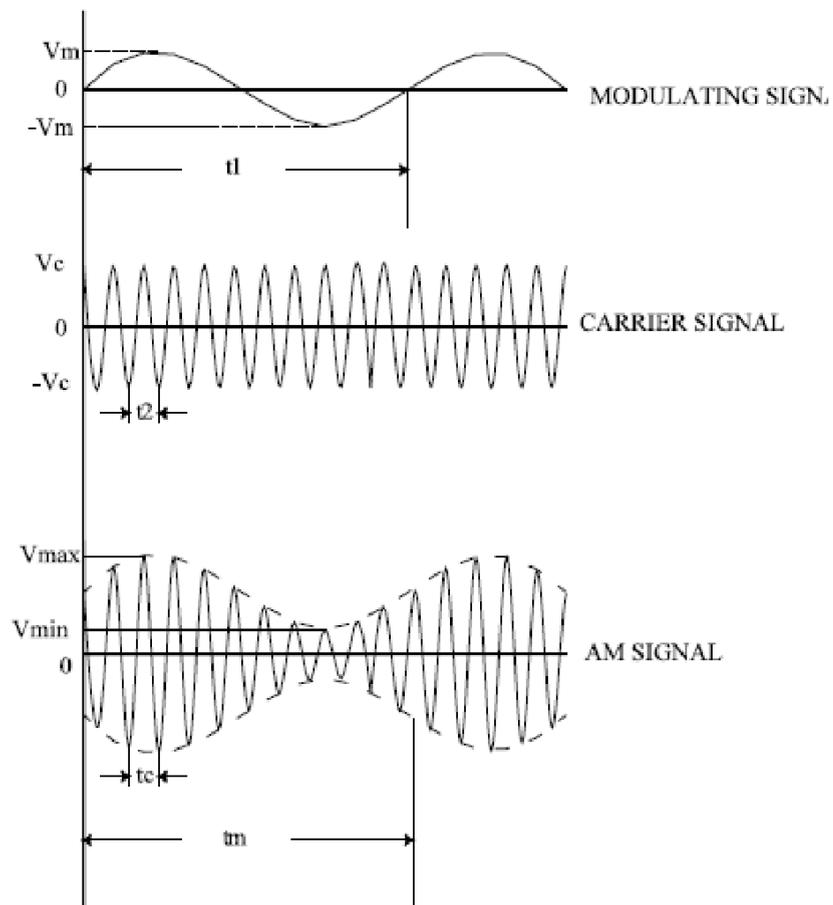
MODULATION

The main problem is that a high frequency carrier wave is used to carry the audio signal and that we need to know how the audio signal should be "added" to the carrier wave. The solution lies in changing some characteristic of carrier wave in accordance with the signal. Under such conditions, the audio signal will be contained in the resultant wave. This process is called modulation. This modulation is of three types namely amplitude modulation, frequency modulation, phase modulation. Here, we shall be discussing about amplitude modulation only.

DEMODULATION

In demodulation process, the modulated signal is to be passed through andemodulater to get the original audio signal. The demodulator used may be an envelope detector. The envelope detector will demodulate the modulator signal and therefore reproduce the original message signal.
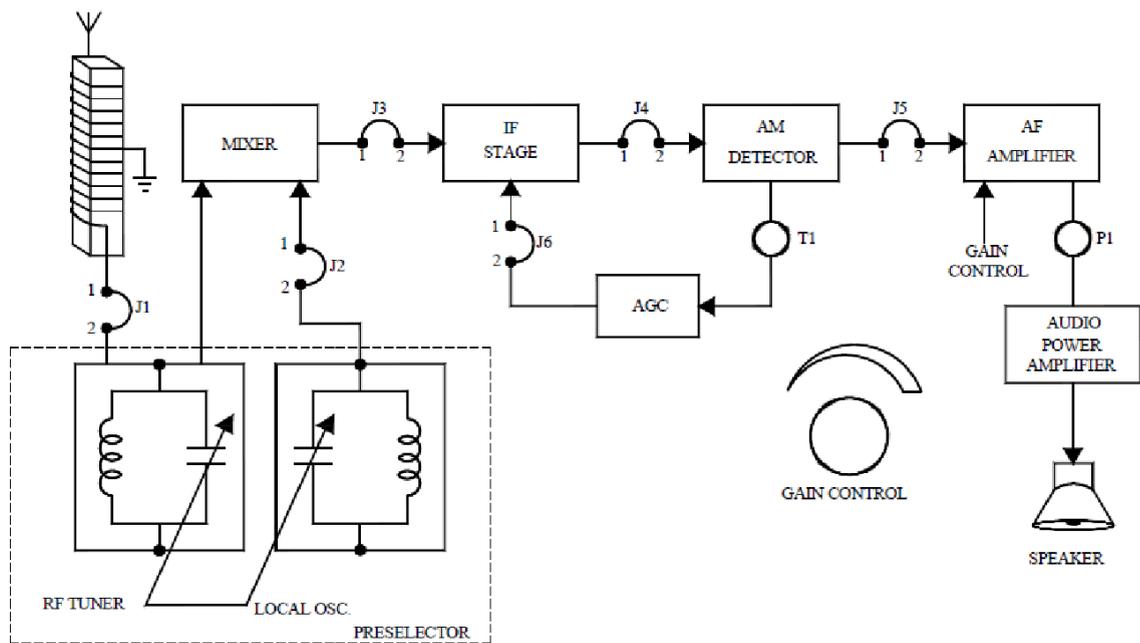
TRANSMITTER

MODULATION FACTOR

        An important consideration in amplitude modulation is to describe the depth of modulation ie, the extent to which the amplitude of carrier wave is changed by the signal. This is described by a factor called modulation factor which may be defined as follows.
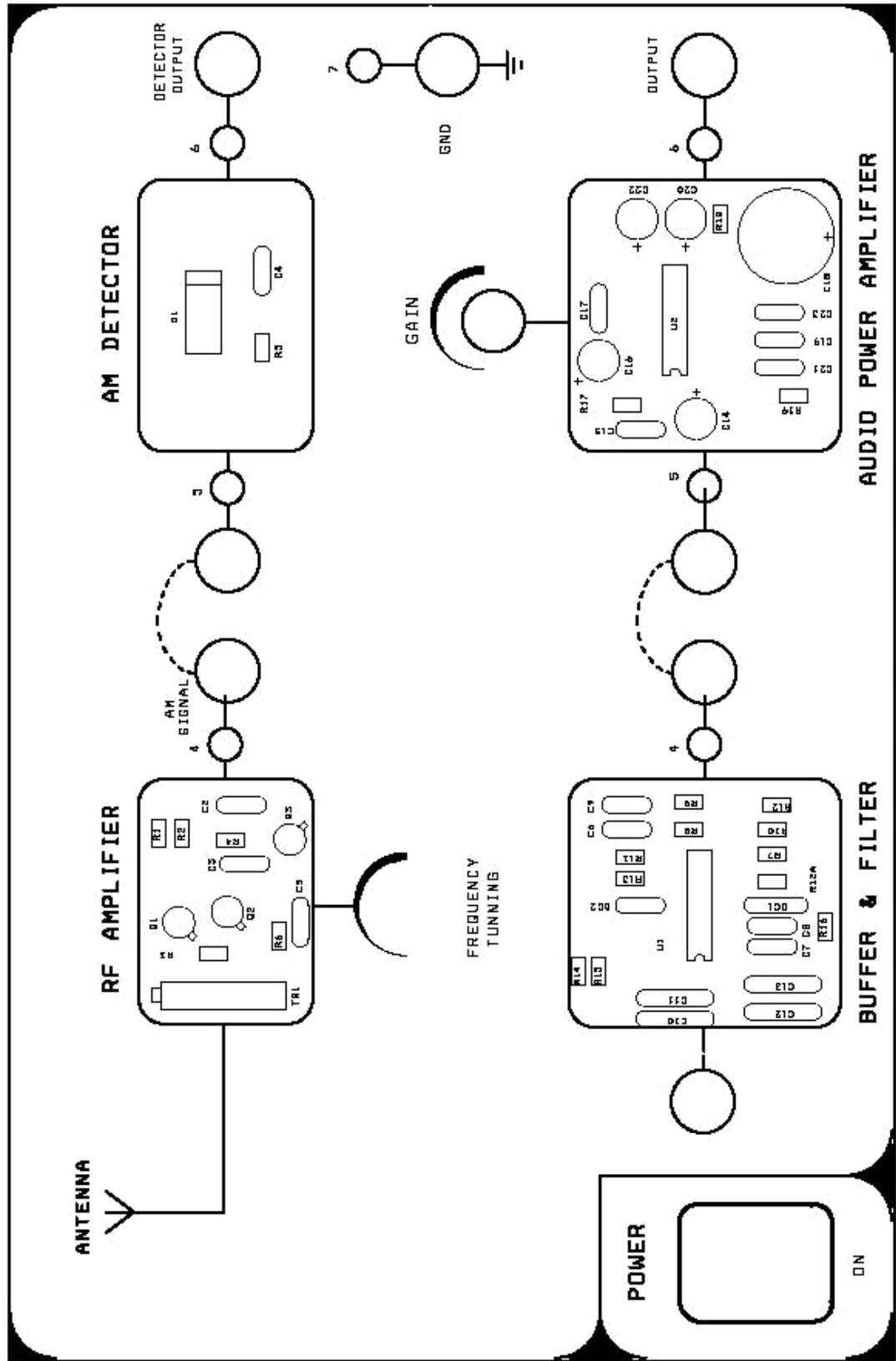
The ratio of change of amplitude of carrier wave to the amplitude of normal carrier wave is called the modulation factor (m) ie,

$$\text{Modulation factor (m)} = \frac{\text{Amplitude change of carrier wave}}{\text{Normal carrier wave (unmodulated)}}$$

$$= \frac{V_{max} - V_{min}}{V_{max} + V_{min}} \times 100$$

PROCEDURE

1. The circuit wiring is done as shown in the wiringdiagram.
2. A Modulating signal input is given to the amplitude modulator from the on-board sine wave generator.
3. Modulating signal input to the amplitude modulator can also be given from an external function generator or an audio frequencyoscillator.
4. If an external signal source with every low voltage level(below 100 mV) is used then this signal can be amplified using the audio amplifier before connecting to the input of the amplitudemodulator.
5. The amplitude and the time duration of the carrier signal are observed and noted down from the output of the amplitude modulator by keeping the amplitude knob of the sinewave generator at zero position.
6. Now increase the amplitude of the modulating signal to the requiredlevel.
7. The amplitude and time duration of the modulating signal are observed using a CRO andtabulated.
8. Finally the amplitude modulated output is observed from the output of the amplitude modulator stage and the amplitude and time duration of the AM wave are noteddown.
9. Patch the modulated signal to the telescopic whip antenna inreceiver.
10. The receiver circuit wiring is also done as shown in the wiringdiagram.
11. The carrier frequency knob in the transmitter side is kept at a middleposition.
12. The frequency tuning knob in the receiver is tuned slowly from one end to the other till the point where the demodulated signal is obtained with minimum distortion &noise.
13. Now the amplitude and time duration of the received signal are noteddown.
14. From the tabulated values the modulating signal, carrier signal, AM signal, & demodulated message signal are plottedneatly.
15. The depth of modulation is alsocalculated.
16. The same experiment can be repeated for various values of carrierfrequency

AM RECEIVER TRAINER (VCT-09)

AM DETECTOR

DETECTOR OUTPUT

GND

RF AMPLIFIER

AM SIGNAL

ANTENNA

FREQUENCY TUNNING

GAIN

OUTPUT

AUDIO POWER AMPLIFIER

BUFFER & FILTER

Vi Microsystems Pvt. Ltd., Chennai-96.

REV:0

POWER

ON

TABULATION

Transmitter

Signal	Amplitude	Time	Frequency
Modulating Signal	$V_m =$	$t_1 =$	$f_1 =$
Carrier Signal	$V_c =$	$t_2 =$	$f_2 =$
Amplitude Modulated wave	$V_{max} =$ V_{min}	$t_m =$ $t_c =$	$f_m =$ $f_c =$

Receiver

Signal	Amplitude	Time	Frequency
Amplitude Modulated Signal	$V_{max} =$ V_{min}	$t_m =$ $t_c =$	$f_m =$ f_c
Demodulated Signal	$V =$	$t =$	$f =$

RESULT

Thus the study of AM Transmission and Reception using AM techniques (sine wave) verified successfully and also calculate the % of modulation index.

POWER ON

FM TRANSMITTER TRAINER   VCT-12

VI Microsystems Pvt. Ltd.,
Chennai-96, INDIA.

ANT

AUDIO AMPLIFIER

O/P T3

AUDIO I/P T2

GAIN

FREQUENCY MODULATOR

ANT I/P T9
RF O/P T8

AMPLITUDE P2

VCO

MOD. I/P T7

FREQUENCY P1

AUDIO OSCILLATOR

T1

FREQUENCY

AMPLITUDE

MIC PRE-AMPLIFIER

O/P T5

GND T6

MIC IN T4
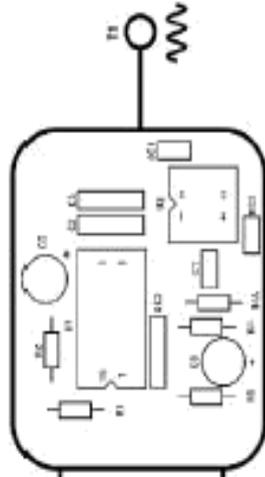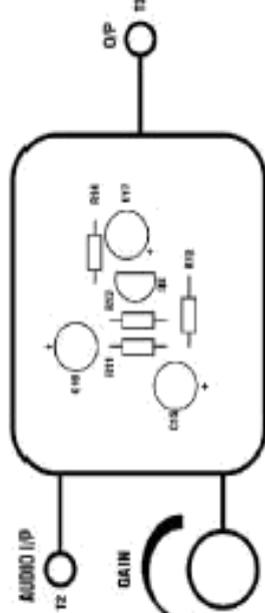
GAIN
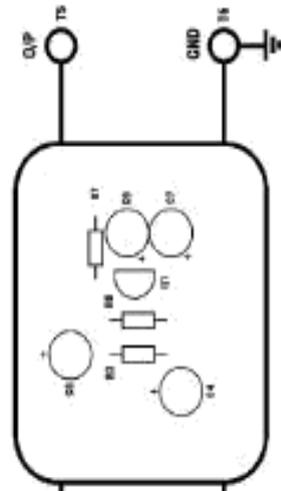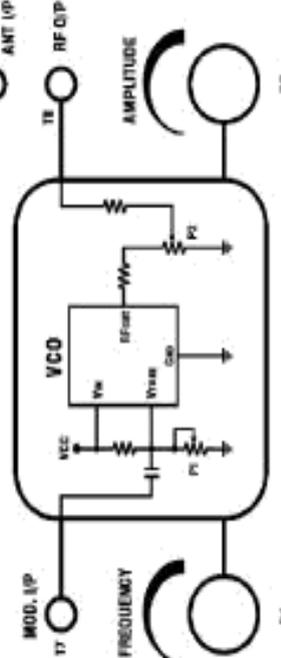
4.FREQUENCY MODULATION AND DEMODULATION

AIM

To transmit a tone (sinusoidal signal) using trainerkit and receive the signal back after demodulator using trainerkit

APPARATUSREQUIRED

1. FMTransmitter
2. FMReceiver
3. 1000 MHz Spectrum analyser and CRO
4. Mic&Speaker
5. Patchchords

PROCEDURE

1. Connect the test point T1 and T7 of VCT - 12 using 2mm patchchord.
2. Switch ON thetrainer.
3. Tune the amplitude control POT of *Audio oscillator* section to fully anticlockwise direction (zero amplitude), now test point T7 is virtually connected toground.
4. Connect the test point T8 to oscilloscope or spectrum analyser, set frequency control POT at *Frequency modulator* section to minimum position (fully anticlockwisedirection).
5. Turn Amplitude control POT P2 to fully clockwise Direction (Maximum amplitude). Note down carrier frequency of VCO which is observed using oscilloscope (or) spectrumanalyser.
6. Slowly turn frequency control POT P1 towards clockwise direction and observe the VCO‟s frequency which is increasing on spectrum analyser, set POT P1 to maximum position note down carrier frequency of VCO which is observed inspectrum.
7. Set back carrier frequency to 100 MHz using P1 potentiometer. Display the test point T1 on oscilloscope, increase amplitude control POT in Audio oscillator section and set sinusoidal amplitude to 1Vpp, turn frequency control POT in *Audio oscillator* section and set frequency of sinusoidal signal to1KHz.
8. Display the frequency modulated waveform at test point T8 on spectrum analyser. Reduce the spectrum analyser band and observe the spectrum of FMsignal.
9. Connect the test point T8 and T9 using 2 mm patch chord to transmit FM signal through whip antenna. 10. Make the following settings and connections on (Refer connection diagram provided at the end of this experiment (Fig) Set RF Tunner to minimum position (fully anticlockwisedirection)

VI Microsystems Pvt. Ltd.,
Chennai-96, INDIA.

FM RECEIVER TRAINER   VCT-13

POWER   ON

AUDIO POWER AMPLIFIER

FM RECEIVER

RF TUNER

SPEAKER
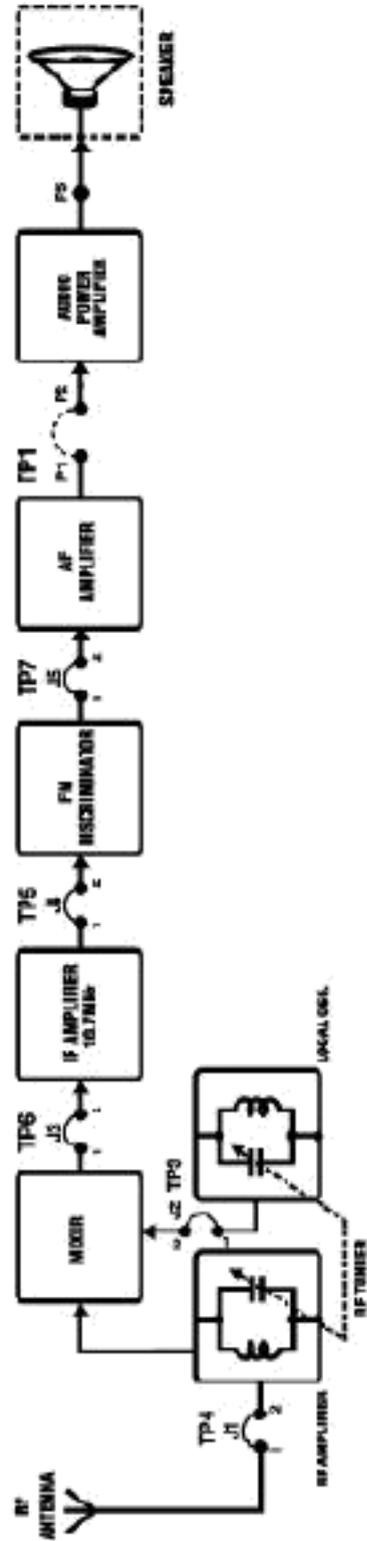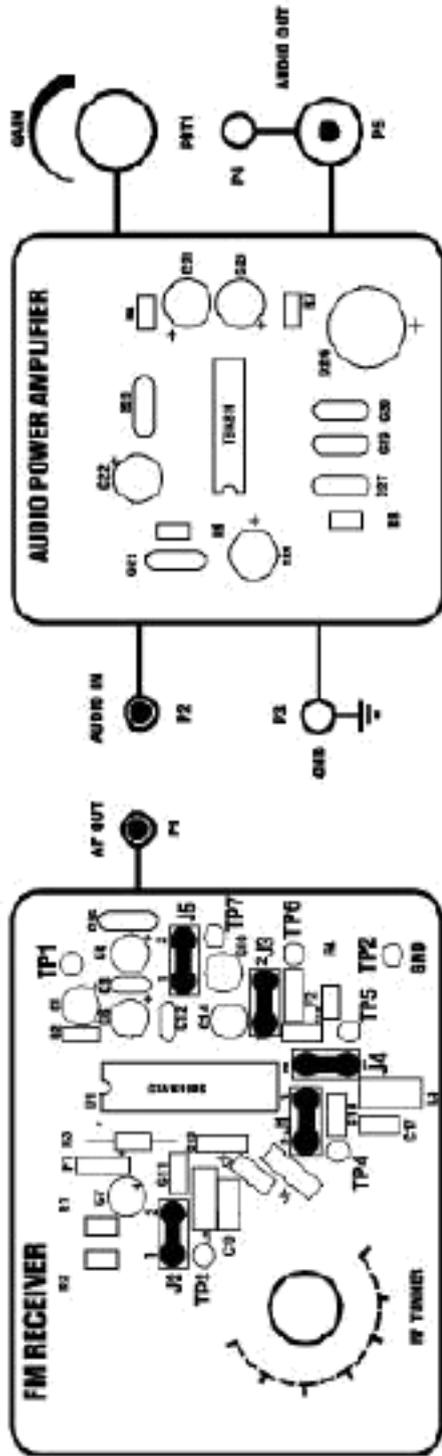
AUDIO POWER AMPLIFIER

AF AMPLIFIER

FM DISCRIMINATOR

IF AMPLIFIER 10.7MHz

MIXER

LOCAL OSC.

RF TUNER

RF AMPLIFIER

RF ANTENNA

b. Set Gain control POT1 to minimum position (fully anticlockwisedirection)

c. Insert Jumper J1, J2, J3, J4, and J5 at respectiveplace.

11. Connect the test point P1 to channel 1 of oscilloscope and switch ON the trainer.

12. Turn RF tunner to clock wise direction slowly, at one point oscilloscope displays 1KHz of tone. Measure the local oscillator frequency of input post J2 jumper by using BNC to P8003 cable. Calculate IF Frequencyby,

IF Frequency = Receiver Local Oscillator Frequency - Transmitter Carrier Frequency

13. Connect the test points P1 and P2 using 2mm patch chord, connect the given speaker at EP Socket P5. Turn Gain control POT1 to clockwise direction, you will hear 1KHz of tone.

14. Vary the frequency control POT of *Audio Oscillator* section at VCT-12 and observe the tone variation on speaker which is connected atVCT-13.

FREQUENCY MODULATION

There are two basic methods for generating frequency modulated signal

namely, i. Direct FM

ii. Indirect FM.

In Direct method the carrier frequency is directly varied in accordance with input baseband signal which is readily accomplished using a voltage control oscillator (VCO).
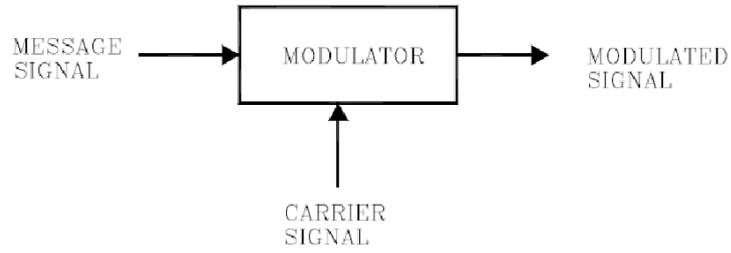
In Indirect method the modulating signal is first used to produce a narrowband FM signal and frequency multiplication is next used to increase the frequency deviation to the desired level. The indirect method is the preferred choice for frequency modulation when the stability of carrier frequency is of major concern as in commercial radio broadcasting.

Frequency modulation is a process in which the instantaneous frequency of the sinusoidal signal is varied in accordance with the incoming message signal. FM signal is a non linear function of modulating signal therefore simply it makes the frequency modulation a non-linear process. Consequently unlike amplitude modulation the spectrum of an FM signal is not related in a simple manner to that of modulating signal rather its analysis is much more difficult than that of AMsignal.

DEMODULATION

Demodulation is a reverse process in which the receiver recreates the original message signal from a degraded version of transmitted signal after the propagation through the channel

TRANSMITTER

MESSAGE SIGNAL → [MODULATOR] → MODULATED SIGNAL

↑ CARRIER SIGNAL

RECEIVER

MODULATED SIGNAL → [DEMODULATOR] → ESTIMATE OF MESSAGE SIGNAL

AMPLITUDE

CARRIER SIGNAL

MODULATING SIGNAL
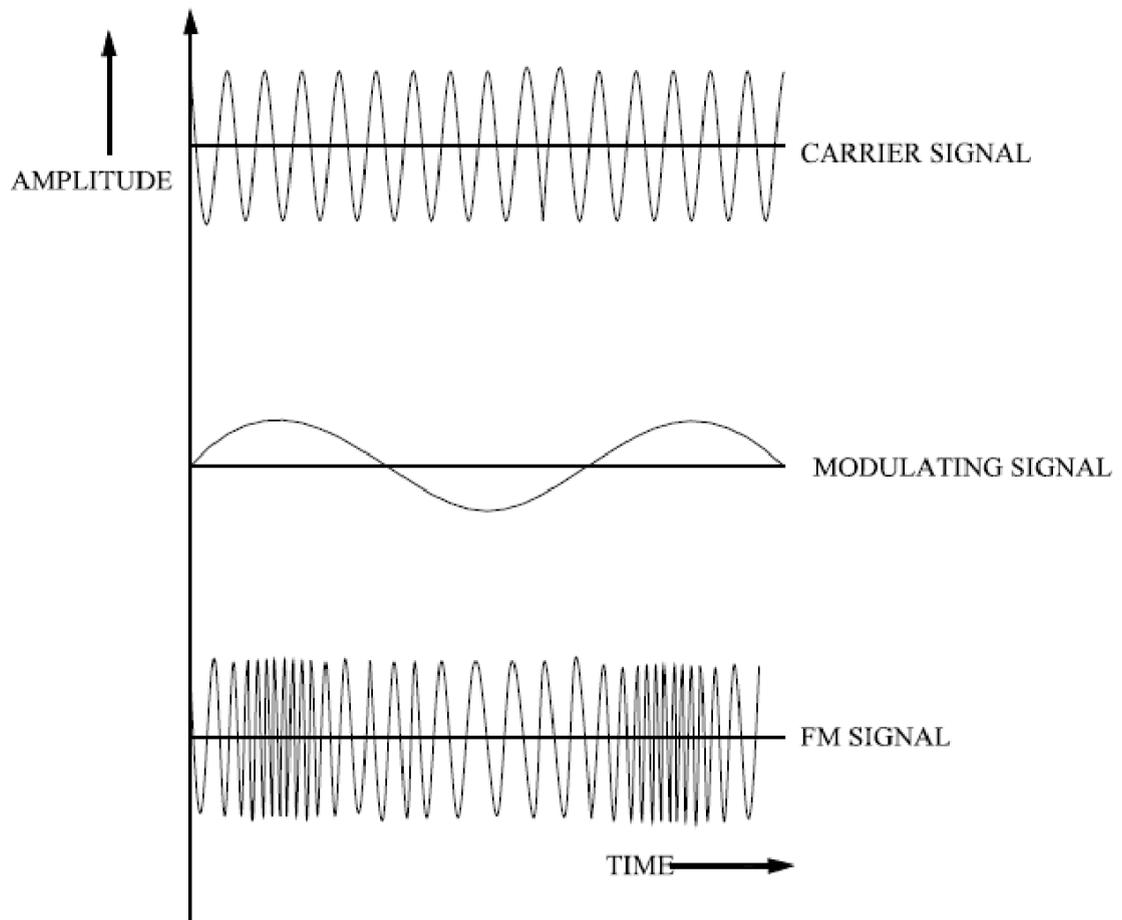
FM SIGNAL

TIME

TABULATION:

S.NO	SIGNAL	AMPLITUDE(V)	TIME(ms)	FREQUENCY(HZ)
1	Modulating Input			
2	carrier Input			
3	FM Output			
4	De mod output			

RESULT

Thus the transmit a tone (sinusoidal signal) using trainer kit and receive the signal back after demodulator using trainer kitsuccessfully.

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BLOCK DIAGRAM OF PULSE CODE MODULATION
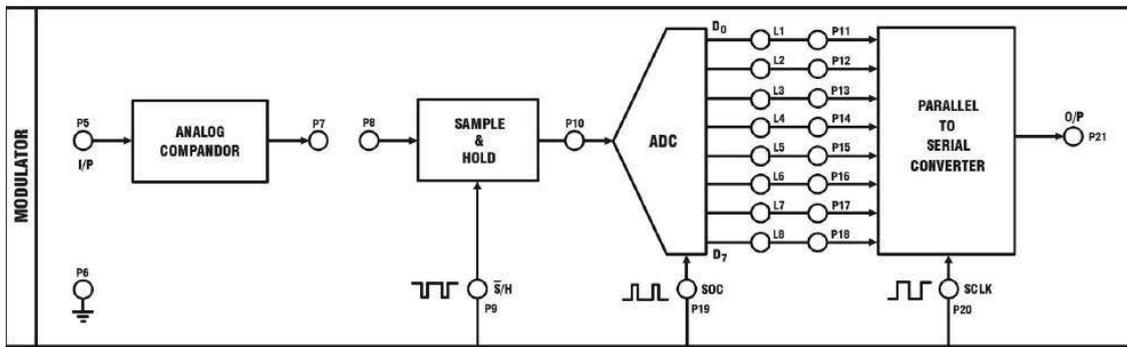
BLOCK DIAGRAM OF PULSE CODE DEMODULATION
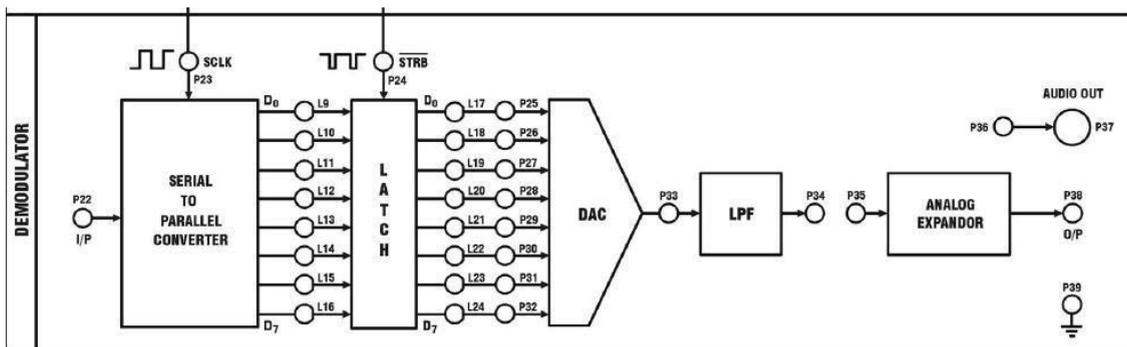
TABULATION

S.NO	SIGNAL	AMPLITUDE(V)	TIME(ms)	FREQUENCY(HZ)
1	Modulating Input			
2	Sampled input			
3	PCM Output			
4	Demodulated output			

5.PULSE CODE MODULATION AND DEMODULATION

AIM

To Study the conceptof pulse code modulation and demodulation..

EQUIPMENTS REQUIRED

PCM trainer Kit
Two channel 20Mhz oscilloscope.
Patch chords and oscilloscope probe.

THEORY

In pulse code modulation, each analog sample converted into eight bit code and they are transmitted in serial form. The PCM system consists of a sample/hold circuit, analog to digital converter and parallel to serial converter. The 1KHz on board sinewave signal can be used for studying modulation and demodulation purpose. External sinewave can also be feed to the modulator section from an external function generator which will be useful for studying frequency response of the system.

PROCEDURE

a) Study of pulse code modulation and demodulation

1. Make wiring connection on VCT - 07 as shown in figure 3.2 (or) simplyconnect the test points P1 to P8 and P21 to P22 using patch chords provided with this training kit.
2. Ensure that all switches in switched faults block in OFF position and all potentiometers POT1 and POT2 in minimumposition.
3. Keep 8KHz of samplingrate.
4. Display the modulating signal at test point P1 using a probe on channel1 of oscilloscope. Increase sinewave amplitude by rotating POT1 in clockwise $_{pp}$direction and set sinewave amplitude to 3V and notedown.
5. Displays the sample / Hold output waveform on channel 2 of oscilloscope and note down the waveform, amplitude level of thesignal.
6. Replace channel 1 waveform by modulator output serial data (test point P21 and compare it with the sample signal on channel 2, every sample has been transmitted with corresponding 8-bits of data. Note down the modulator outputwaveform.
7. Plot all the noted waveforms such as modulating signal, S/H output and modulator output on a linear graphsheet.
8. Replace the channel 2 waveform by digital to analog converter (test point P33) waveform which is the recovered sampled analog signal, note down thewaveform.
9. Observe the recovered sinewave at test point P34, note down waveforms. Plot all the noted waveforms such as DAC signal, and recovered sinewave on a linear graphsheet.

RESULT

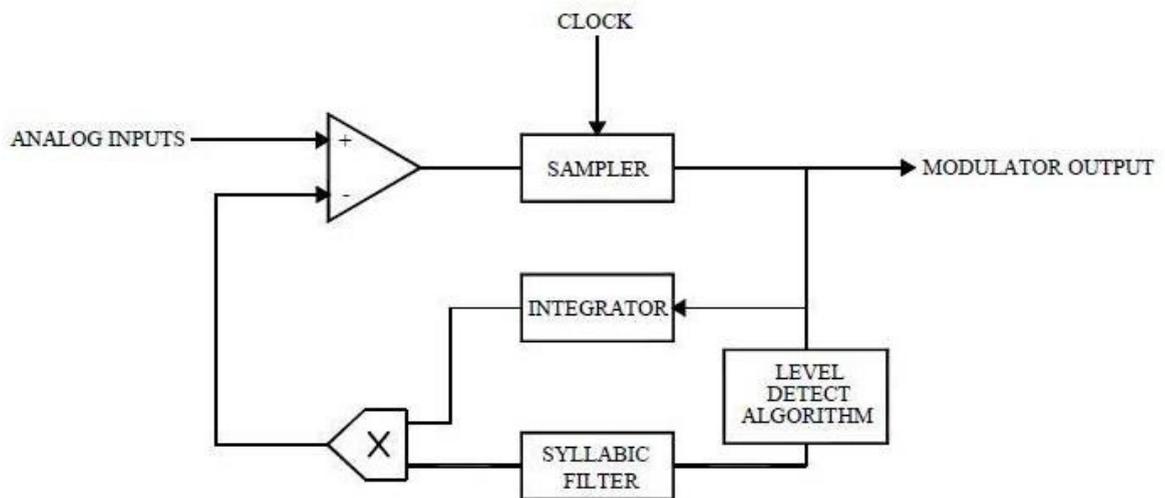Thus the Study of pulse code modulation and demodulation verified successfully.

DELTA MODULATOR
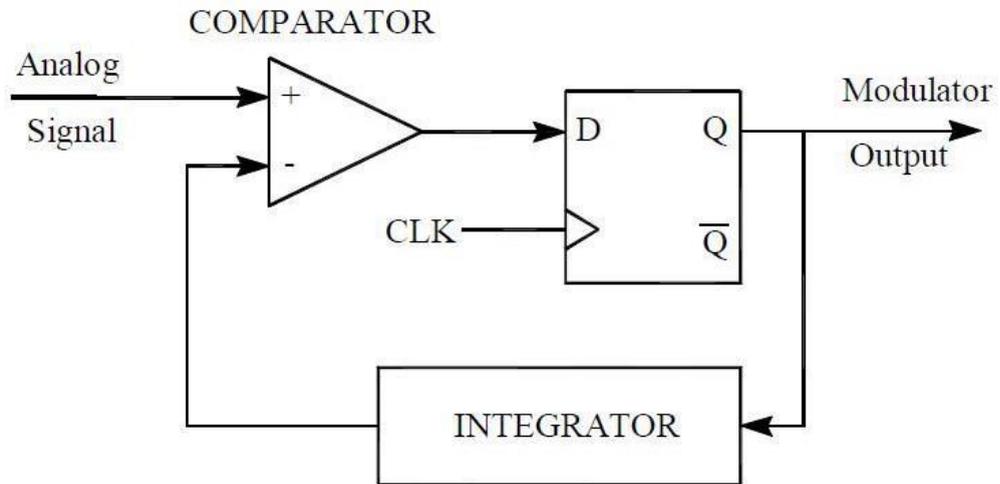
DELTA DEMODULATOR

ADAPTIVE DELTA MODULATOR

6.DELTA / ADAPTIVE DELTA MODULATION AND DEMODULATION

AIM

To study the delta / Adaptive delta modulation and demodulation

EQUIPMENTS REQUIRED

a. DM / ADMkit

b. Two Channel 20MHz Oscilloscope

c. Patch Chords, Oscilloscopeprobe

THEORY

DELTA MODULATOR

The modulator comprises of comparator, quantizer and Integrator. The input base band sinusoidal signal and its quantized approximated signals (feedback signal from integrator) are applied to comparator. A comparator as its name suggests simply makes a comparison between inputs. The comparator gives a TTL signal is then latched into a D-flipflop which is clocked by selected clock rates. The binary data stream from the flip flop is transmitted to receiver and is fed to the integrator. The integrator output is then connected to the negative terminal of the voltage comparator.

DELTA DEMODULATOR

The demodulator comprises of simple, integrator and low pass filter. The receive delta modulator signal is applied to integrator, its output tries to follow the analog signal. The integrator output contains sharp edges which is smoothened out by the 4_{th} order low pass filter.

DISTORTION

The distortion in delta modulation can be broken into two distinct areas, quantization noise and idle channel noise (during zero input signal). The two major parameters s which affect the distortion in delta modulation are the sampling rate f and step size $s_{,,})$". Ideally the step size)should be small as possible and sampling rate f as large as possible. Practical and economic considerations limits the minimum step size of $s_{,,})$" and the maximum sampling rate of f. These limitations give rise to two types of distortions, slope overload noise and granular noise.

Idle channel noise

Usually delta modulation in an idle state (there is no or zero input signal), generates series of one"s and zero"s consecutively. This generates a square waveform at the output of accumulator (or) feedback section called *step size*. The frequency of the step size is equal to that of sampling clock of the DM system. This unnecessary square waveform doesn"t affect the decoder because of frequency of the step size (which will be attenuated due to low passfilter).

Slope overload and Granular noise

In normal delta modulator operation, the encoder is able to track the input within an error not more than). The system is said to be slope overload if the error exceeds), where two or more steps are required to achieve the input level. The slope overload can be reduced by increasing step size)of the system. The system exhibits granular noise if the error falls by), where two or more steps are required to achieve the input level. The granular noise can be reduced by decreasing step size)of the system
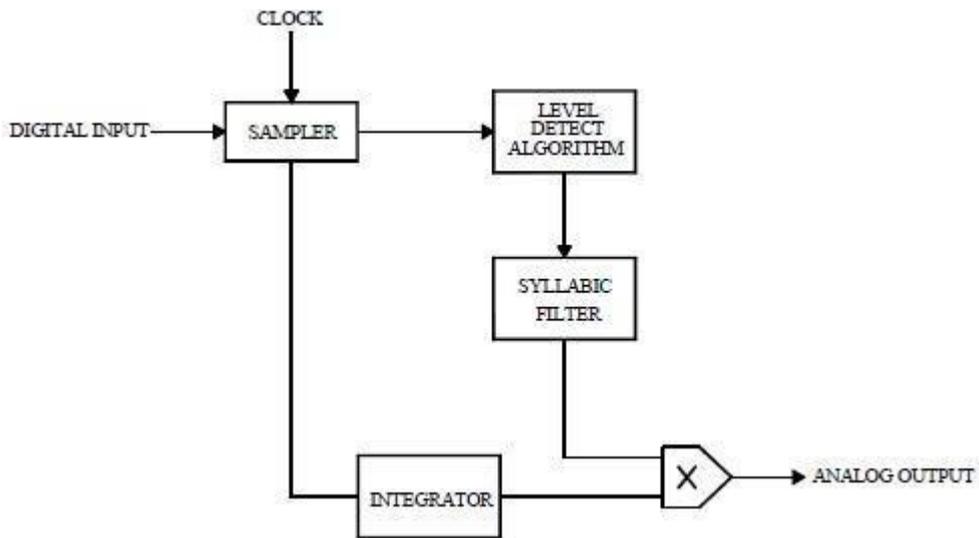
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ADAPTIVE DELTA DEMODULATOR

TABULATION

S.NO	SIGNAL	AMPLITUDE(V)	TIME(ms)	FREQUENCY(HZ)
1	Modulating Input			
2	Sampled input			
3	Sampled output			
4	DM output			
5.	Demodulated output			

MODEL GRAPH

Granular Noise

Slope overload
Noise
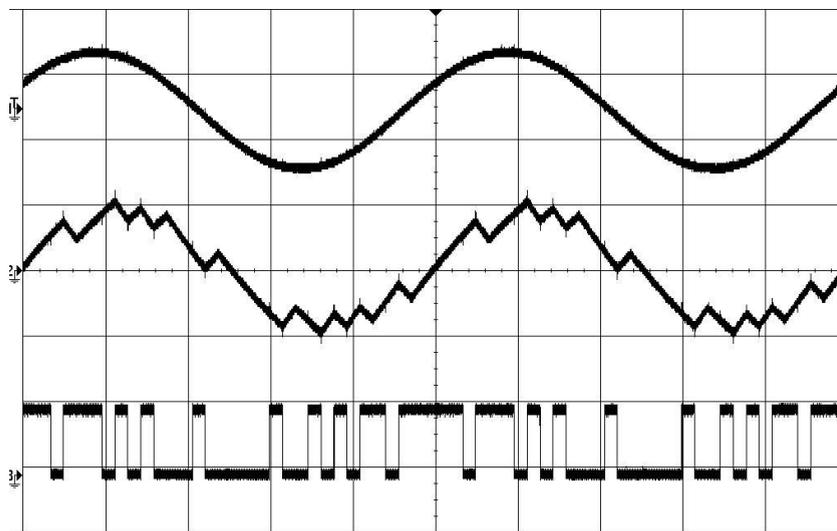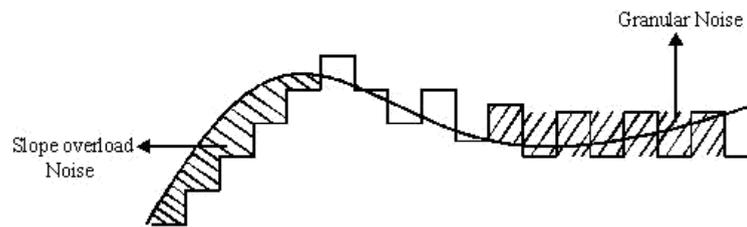
The step size)should be selected to bring a trade - off between slope overload and granular noise. Anyhow the error between the input and the recovered signals can be appreciably reduced by using low pass filter.

PROCEDURE

1. Plug-in AC power cord into 230V, 5A Mains powersupply.
2. Ensure that the following initial conditions exist onVCT-50:
      a. Keep all switches in OFFposition.
      b. Keep all potentiometer controls in minimumposition.
3. Wiring connections do as follows inVCT-50:
      a. Connect point P1 (1KHz sine wave) to input of DM section(P9).
      b. Connect point P2 (Clock signal) 32KHz to clock input of DMsection(P14).
      c. Connect point P12 (Integrator output to the comparator input(P10)).
      d. Connect output (P15) of delta modulator to the input of delta demodulator
        section(P19).
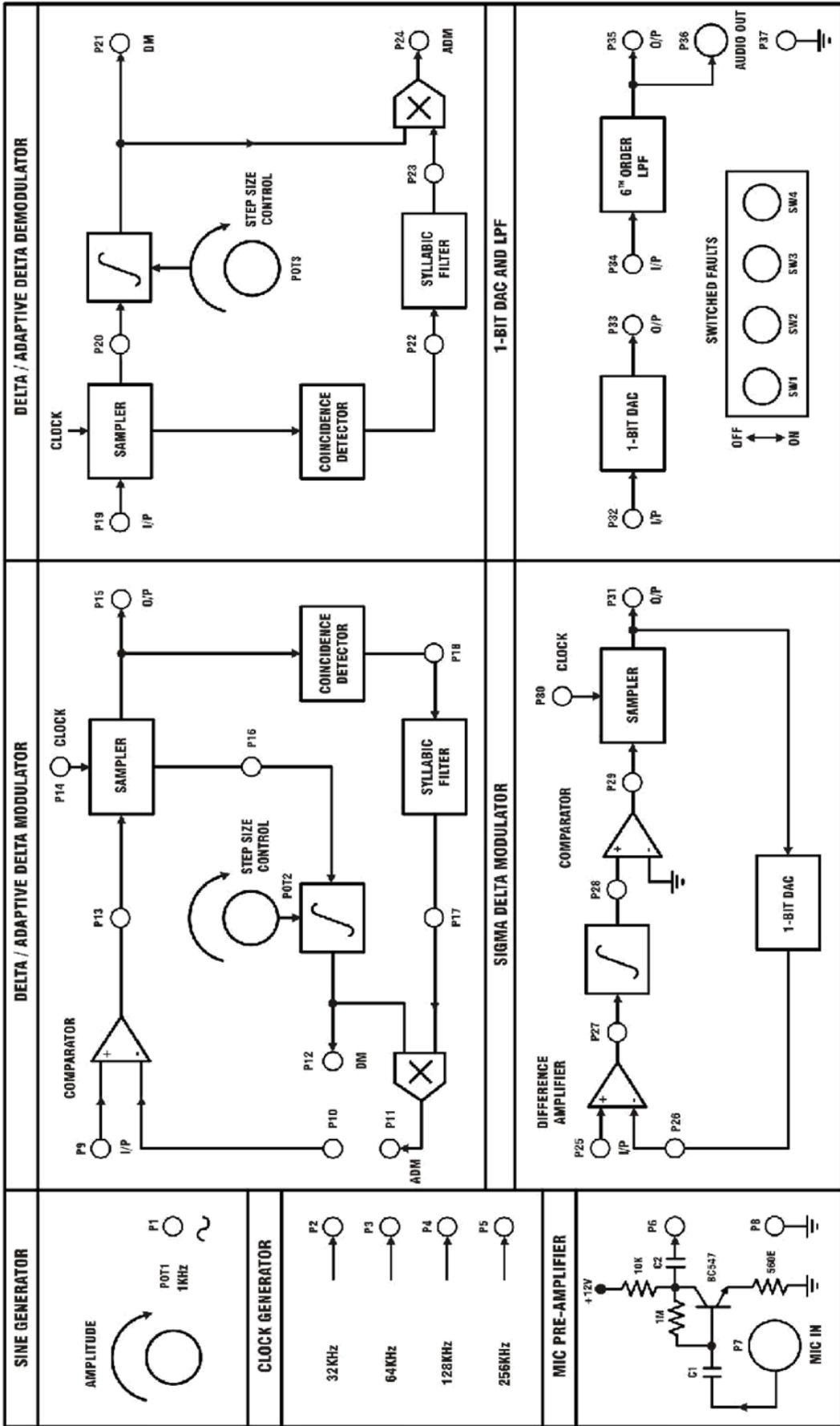   e. Connect point P21 to input of low pass filter(P34). (Refer the wiring diagram
        included in the end of thisexperiment)
4. Switch ON the power supply to theboard
5. Connect the test point P1 with oscilloscope, turn POT1 in clockwise direction and set the amplitude of sine wave to2V$_{pp}$.
6. Observe the integrator output waveform (P12) on channel 2 of oscilloscope, turn step size control potentiometer (POT2) in clockwise direction so as get the waveform as shown in output waveform figure. Note down the integrator outputwaveform.
7. Display the data modulator output waveform (P15) on channel 1 of oscilloscope in place of sinewave. Note down modulator waveform with respect to the integratorwaveform.
8. Display the received digital signal in test point P19 on channel 1 of oscilloscope and its integrated output waveform (P21) on channel 2 of oscilloscope. Turn POT 3 in clockwise direction and set waveform amplitude to 2V$_{pp}$. Note down both signals plot the all above waveform on a linear graphsheet.
9. Observe the final demodulated waveform (P35) on oscilloscope and plot in graphsheet.
10. Do the above procedure for other clock rates of 64KHz and 128KHz, explore the changes in demodulated signal.

RESULT
  Thus the study of delta / Adaptive delta modulation and demodulation verified sucessfully

DELTA / ADAPTIVE DELTA / SIGMA DELTA
MODULATION & DEMODULATION

VCT-50

DELTA / ADAPTIVE DELTA DEMODULATOR

DELTA / ADAPTIVE DELTA MODULATOR

SIGMA DELTA MODULATOR

1-BIT DAC AND LPF

SINE GENERATOR

CLOCK GENERATOR

MIC PRE-AMPLIFIER

Vi Microsystems Pvt. Ltd., Chennai-96.

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

Data

Unipolar NRZ

Unipolar RZ

Polar NRZ

Polar RZ

Bipolar NRZ

Bipolar RZ or AMI

Manchester
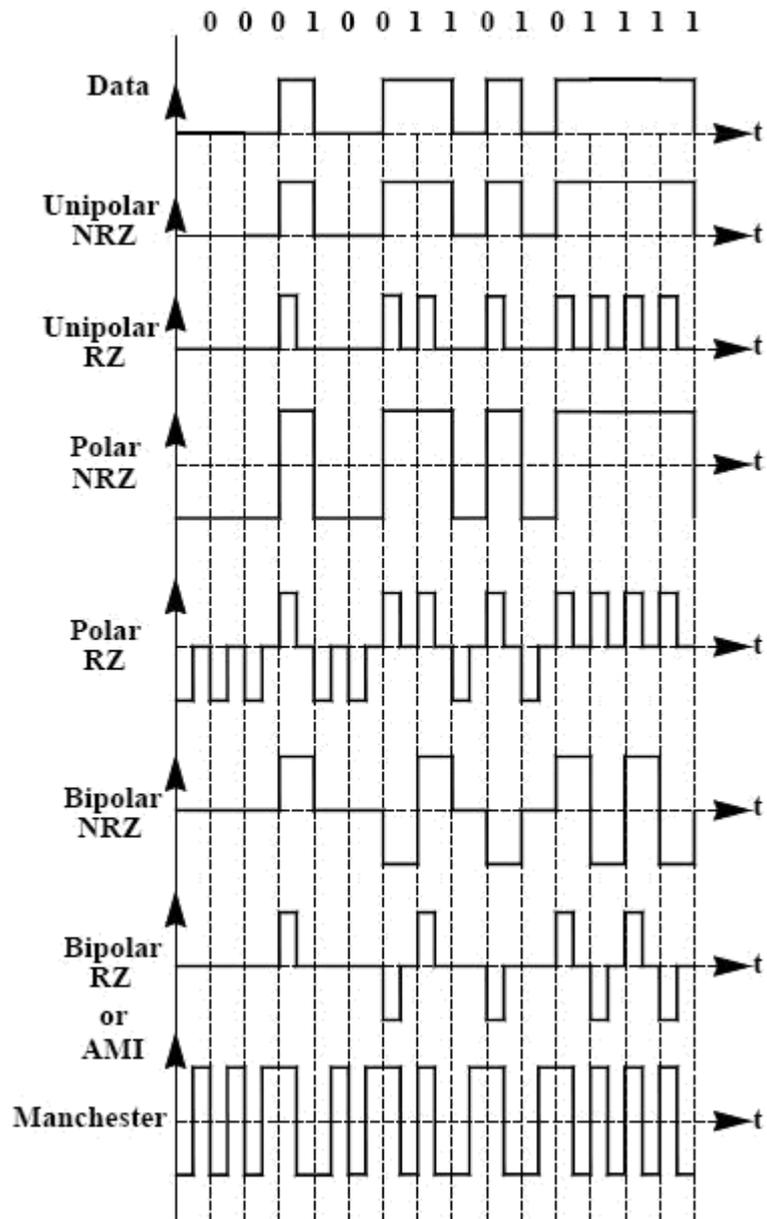
7.LINE CODING AND DECODING TECHNIQUE

AIM

To perform various type of line coding &Decoding technique

CHARACTERISTICS OF LINECODES

1. TransmissionBandwidth
2. Timing
3. DC Content
4. PowerSpectrum
5. Power Efficiency
6. Probability oferror
7. Transparency

Unipolar RZ

In this line code, a binary „ 1' is represented by a non-zero voltage level during a portion of the bit duration, usually for half of the bit period, and a zero voltage level for rest of the bit duration. A binary „0' is represented by a zero voltage level during the entire bit duration.

The main advantage of unipolar RZ are case of generation requires single power supply and which allows simple timing recovery. A number of disadvantages exists for this line code. It has a non-zero DC component and non-zero DC content, which can load to DC wander. A long string of „0's will back pulse transition and could load to loss of synchronization. There is no error detection capability. The bandwidth requirement is also higher than non-return to zero signal.

Polar RZ

In this scheme, a binary „1' is represented by alternating positive voltage levels, which return to zero for a portion of the bit duration, generally half the bit period. A binary „0's is represented by a negative voltage levels and return to zero for half bit duration.

This code has no DC component and zero DC content, completely avoiding +ve DC wander problem. Timing recovery is rather easy by squaring, or full-wave rectifying. It requires low bandwidth. The obvious disadvantage is that the error rate performance is worst. A long string of 0's or 1's could not appear and so improves in synchronization, and two power supplies are required for this code.

Polar NRZ

In this line code, a binary 1 is represented by a positive voltage +v and a binary 0 is represented by a negative voltage -v over the full bit period. This code is also referred to as NRZ(L), since a bit is represented by maintaining a level during its entire period. This code can also be represented by assigning negative voltage for logic 1 and positive voltage for logic 0.

The advantage of polar NRZ includes a low-bandwidth requirements, very good error probability, and great reduced DC because the waveform has a zero DC component. A major disadvantage of this code that there is no error detection capability and that a long string of 1's or 0's could result in loss of synchronization and power supplies are required to generate this code.

Bipolar NRZ:

In this scheme, a binary „1' is represented by positive and negative voltage levels in alternating mark level in full bit period. A binary „0' is represented by a zero voltage levels during entire bit duration. This code also called as alternate mark inversion (AMI) since 1's are represented by alternating positive and negative pulses.

This code has no DC component and zero DC content, completely avoiding the DC wander problem. Because of the alternative polarity pulses for binary 1's, this code has error detection and hence correction also possible. A long string of 0's could result in loss of synchronization, and two power supplies are required for this code.

Bipolar RZ

In this scheme, a binary „1' is represented by alternating positive and negative voltage a levels for a half bit period duration and maintaining zero for other of period. A binary „0' is represented by a zero voltage levels during entire bit duration. This code also called as AMI. This code has no DC component and zero DC Conant, completely avoiding the DC wander problem. Because of alternative polarity pulses for binary 1's, this code has error detection and hence correction also possible. A long string of 0's could result in loss of synchronization, and two process supplier and required for this code.

Manchester Coding

In this scheme, a binary 1 is represented by a pulse that has positive voltage during the first-half of the bit duration and negative voltage during second-half of the bit duration. A binary „0' is represented by a pulse that has negative voltage during first-half of the bit duration and positive voltage during second-half of the bit duration.

The advantage of this code includes a zero DC content and so avoiding DC-wandering problems. The code having alternation positive and negative pulses and so timing recovery is simple and it has good error rate performance. The main disadvantage of this scheme is larger bandwidth. It has no error detection possibility.

RESULT:

Thus the study Line coding schemes verified sucessfully

FSK MODULATION AND DEMODULATION
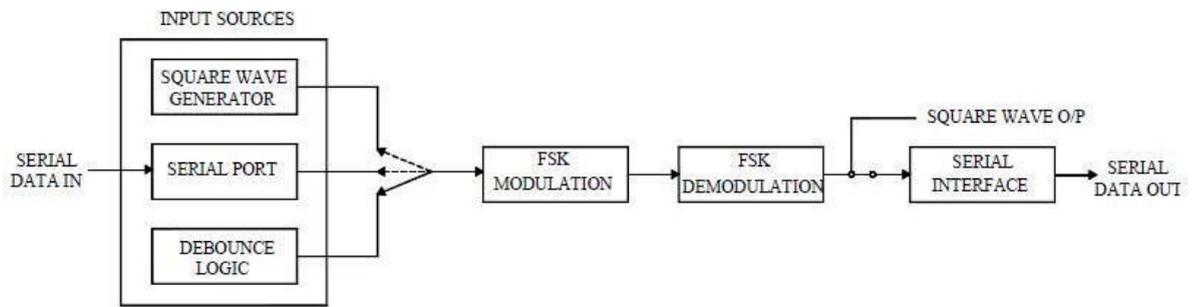
Block Diagram Of FSK Modulation And Demodulation

8. ASK, FSK, BPSK GENERATION & DETECTION SCHEMESIMULATION

AIM

To study the ASK,FSK,BPSK Modulation and demodulation using SCILAB code & observe the output waveform.

THEORY

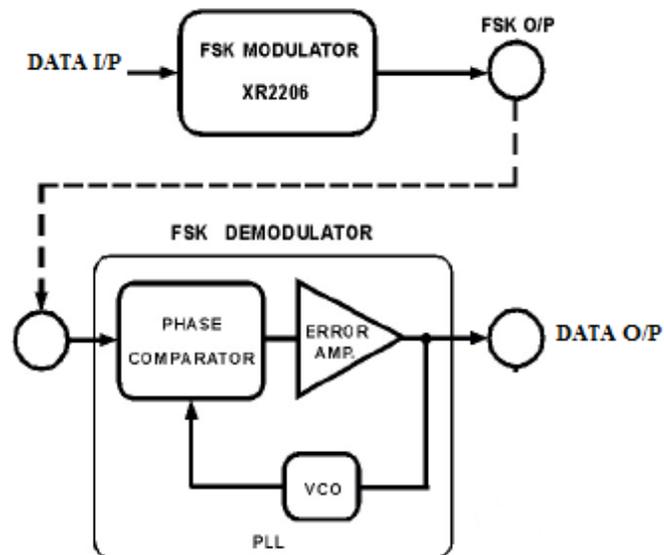ASK signal has a well defined envelope. Thus it is amenable to demodulation by an envelope detector. Some sort of decision-making circuitry is necessary for detecting the message. The signal is recovered by using a correlator and decision making circuitry is used to recover the binary sequence.

FSK is one method used to overcome the bandwidth limitation of the telephone system so that digital data can be sent over the phone lines. The basic idea of FSK is to represent 1s and 0s by two different frequencies within the telephone bandwidth. The standard frequencies for a full duplex 300 baud FSK Modulator & Demodulator in the originate modes are 1070 Hz for a 0 (called a space) and 1270 Hz for a 1 (called a mark). In the answer mode, 2025 Hz is a 0 and 2225 Hz is a 1. The relationship of these FSK frequencies and the telephone bandwidth is illustrated in figure 1. Signals in both the originate and answer bands can exist at the same time on the phone line and they do not interfere with each other because of the frequencyseparation.

PSK is a digital modulation scheme, which is analog to phase modulation. In binary phase shift keying two output phases are possible for a single carrier frequency ("binary" meaning"2"), one output phase represents a logic 1 and the other a logic 0. As the input digital binary signal changes its state, the phase of output carrier shifts between two angles that are 180° out of phase. Other names for PSK are phase reversal keying (PRK) and biphase modulation.

MODEL GRAPH:

ASK

FSK
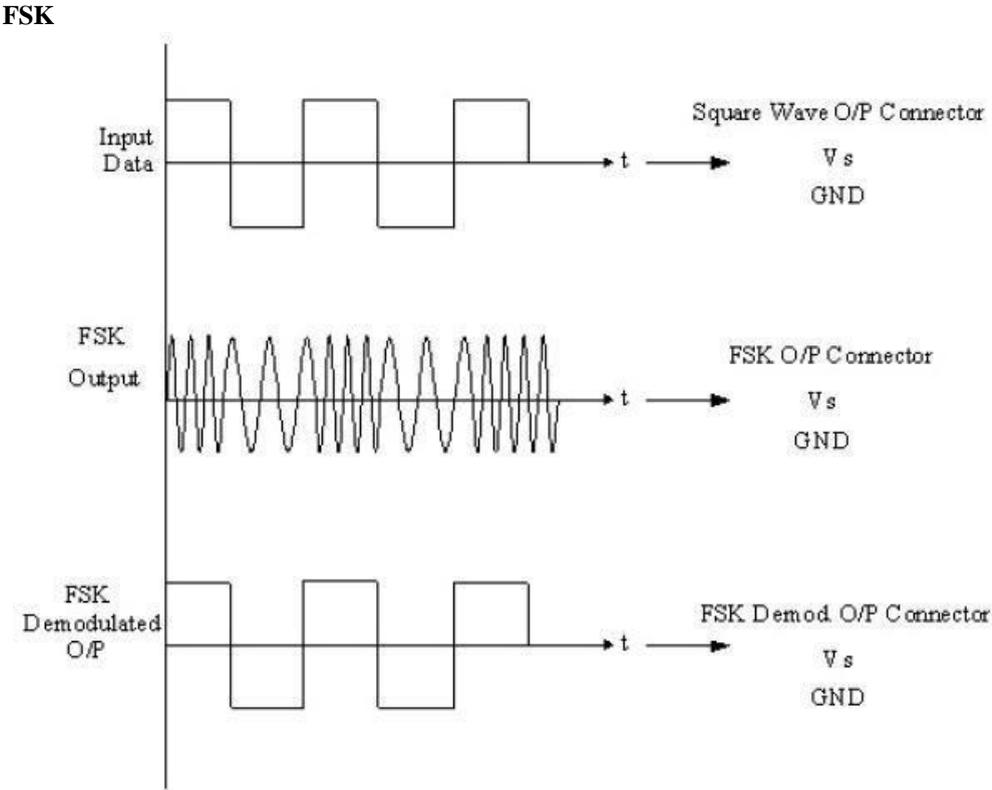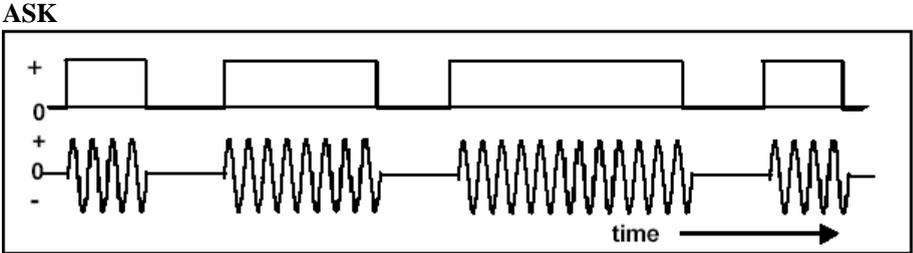
PSK
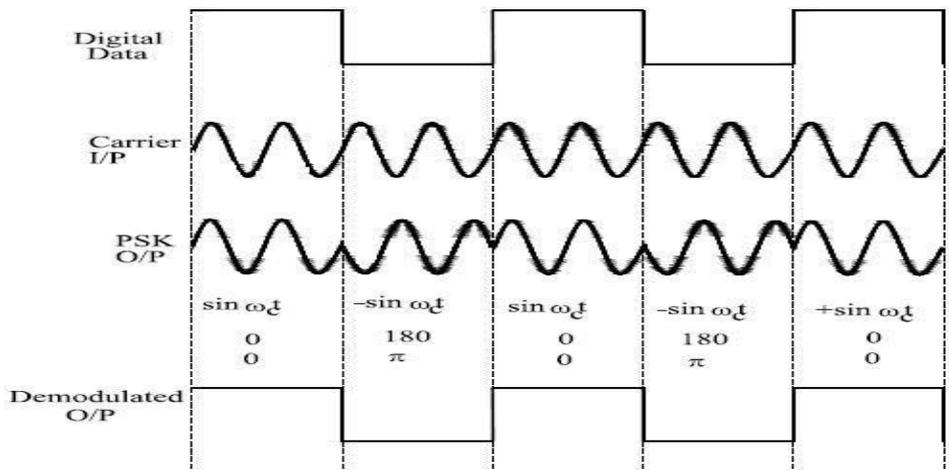
SCILAB CODE FOR ASK:

```
clearall;
clc;
f=input('enter the analog carrier frequency in Hz:');
t=0:0.00001:0.1;
x=cos(2*%pi*f*t);
message=[];
carrier=[];
I=input('enter the digital binary data:');
fori=1:length(I)
t=[0:.00001:0.1]
ifI(i)>0.5
m(i)=1;
m_s=ones(1,length(t));
carrier=[carrier,x]
else
m(i)=0;
m_s=zeros(1,length(t));
carrier=[carrier,x]
end
message=[message,m_s];
end
//generation of ask
Xask=[];
forn=1:length(I)
if((I(n)==1)&(n==1))
Xask=[x,Xask];
elseif((I(n)==0)&(n==1))
Xask=[zeros(1,length(x)),Xask];
elseif((I(n)==1)&(n~=1))
Xask=[Xask,x];
elseif((I(n)==0)&(n~=1))
Xask=[Xask,zeros(1,length(x))];
end
end
subplot(4,1,1)
plot(message)
```

```
xtitle('Y15EC805   (Binary Message Signal)')
xlabel('Time--->')
ylabel('Amplitude--->')
subplot(4,1,2)
plot(carrier);
xtitle('Carrier signal')
xlabel('Time--->')
ylabel('Amplitude--->')
subplot(4,1,3)
plot(Xask)
xtitle('Amplitude Shift Keying')
xlabel('Time--->')
ylabel('Amplitude--->')
subplot(4,1,4);
plot(message)
xtitle('Demodulated signal')
xlabel('Time--->')
ylabel('Amplitude--->')*/
//demodulation
demod=[]
xdm=[]
fori=1:length(I)
ifi>=2
xdemod=sum(Xask(1:(i*length(t))))-sum(Xask(1:((i-1)*length(t))));
else
xdemod=sum(Xask(1:(i*length(t))));
end
ifxdemod>0
demod(:,i)=1
xdm=[xdm,ones(1,length(t))]
else
demod(:,i)=0
xdm=[xdm,zeros(1,length(t))]
end
end
disp('demodulated data:',demod);
```

```
if(demod==I)
subplot(4,1,4);
plot(xdm)
xtitle('Demodulated Signal')
xlabel('Time--->')
ylabel('Amplitude--->')
else
disp('invalid');
end
```

OUTPUT

enter the analog carrier frequency in Hz:50

enter the digital binary data:[1 0 1 0 1]

  "demodulated data:"

  1. 0. 1. 0. 1.

WAVEFORM
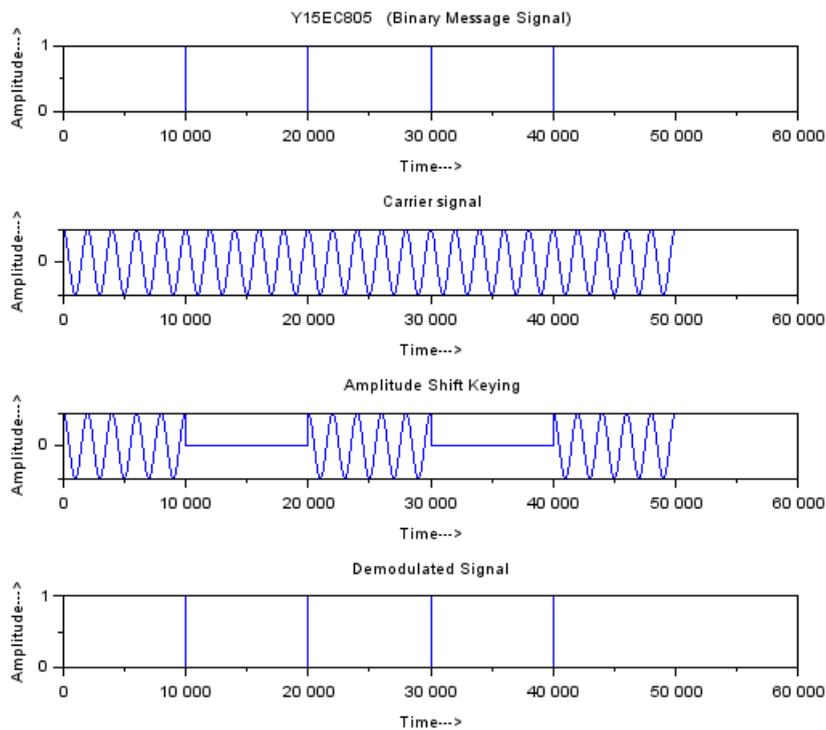
SCILAB CODE FOR FSK

```
clear all
clc
f=input('Enter frequency in Hz:')
t=0:0.0001:1
x1=cos(2*%pi*f*t);
x2=cos(2*%pi*(2*f)*t);
message=[];
carrier1=[];
```

```
carrier2=[];
I=input('Enter binary data:')
fori=1:length(I)
if(I(i))==1
m_s=ones(1,length(t));
else
m_s=zeros(1,length(t));
end
message=[message,m_s];
   carrier1=[carrier1,x1]
   carrier2=[carrier2,x2]
end
fsk=[];
for n=1:length(I)
if(I(n)==1)
fsk=[fsk,x2];
else
fsk=[fsk,x1]
end
end
subplot(4,1,1);
plot(message);
xtitle('Message Signal');
xlabel('Time---->');
ylabel('Amplitude--->');
subplot(4,1,2);
plot(carrier1);
xtitle('Carrier Signal');
xlabel('Time---->');
ylabel('Amplitude--->');
subplot(4,1,3);
plot(fsk);
xtitle('Modulated Signal');
xlabel('Time---->');
ylabel('Amplitude--->');
//demodulation
xdemod1=[];
xdemod2=[];
demod=[];
xdemod1=fsk.*carrier1;
xdemod2=fsk.*carrier2;
xdemod=xdemod2-xdemod1;
fori=1:length(I)
ifi==1
    s=sum(xdemod(1:length(t)))
else
    s=sum(xdemod((i-1)*length(t):i*length(t)))
end
if(s>0)
demod=[demod,zeros(1,length(t))]
else
demod=[demod,ones(1,length(t))]
end
```

end
subplot(4,1,4);
plot(demod);
xtitle('DeModulated Signal');
xlabel('Time---->');
ylabel('Amplitude--->');

OUTPUT
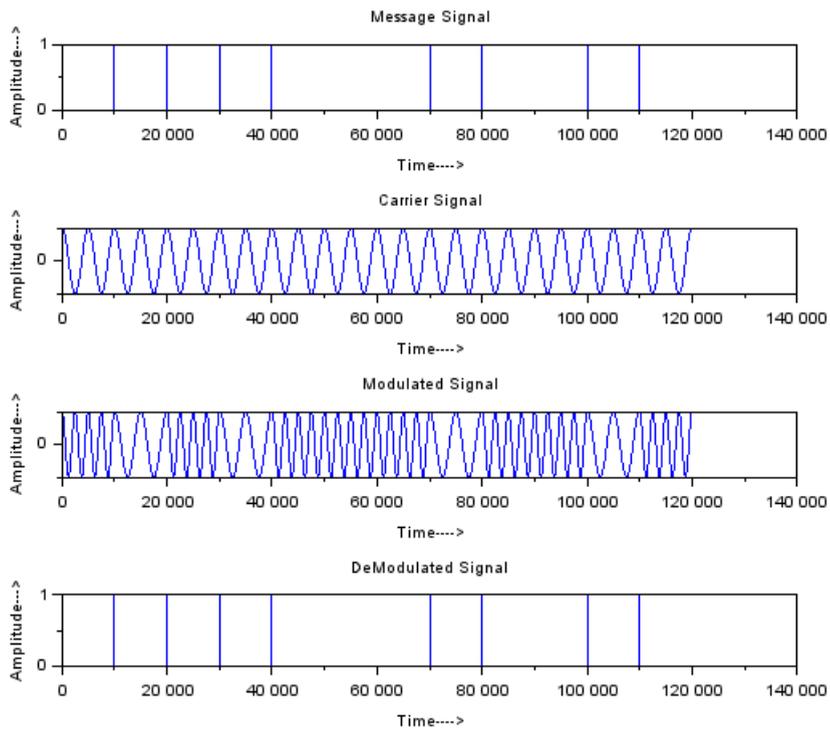
BLOCK DIAGRAM OF PSK MODULATOR
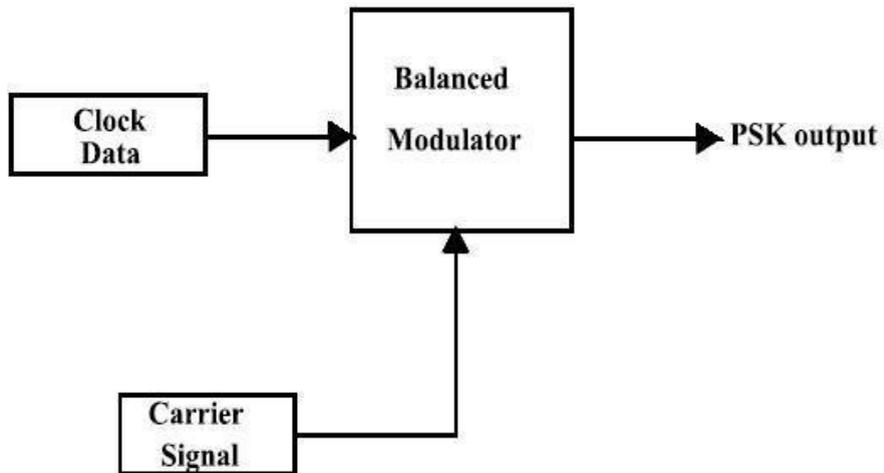
BLOCK DIAGRAM OF PSK DEMODULATOR
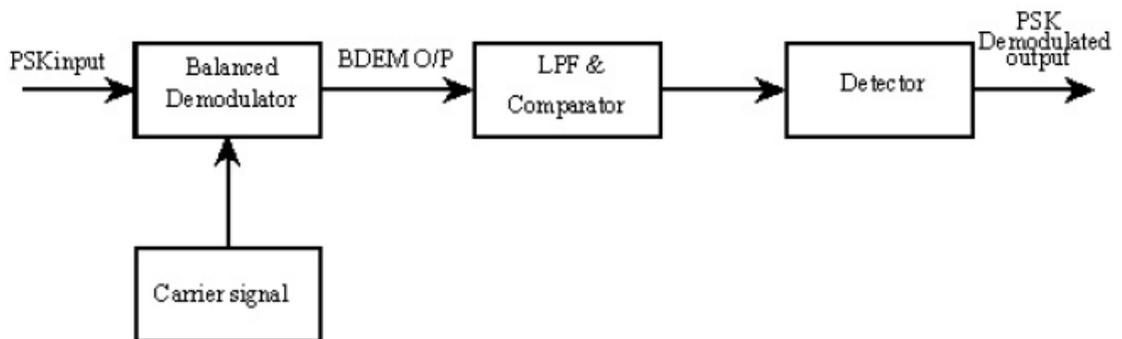
```
clear all
clc
f=input('enter frequency in Hz:')
t=0:0.0001:1
x1=sin(2*%pi*f*t);
x2=sin((2*%pi*f*t)+%pi)
message=[];
carrier1=[];
carrier2=[];
I=input('Enter binary data:')
fori=1:length(I)
if(I(i))==1
m_s=ones(1,length(t));
else
m_s=zeros(1,length(t));
end
message=[message,m_s];
    carrier1=[carrier1,x1]
    carrier2=[carrier2,x2]
end
psk=[];
for n=1:length(I)
if(I(n)==1)
psk=[psk,x2];
else
psk=[psk,x1]
end
end
subplot(4,1,1);
plot(message,"r");
xtitle('Y15EC847   Message Signal');
xlabel('Time---->');
ylabel('Amplitude--->');
subplot(4,1,2);
plot(carrier1,"g");
xtitle('Carrier Signal');
xlabel('Time---->');
ylabel('Amplitude--->');
subplot(4,1,3);
plot(psk);
xtitle('Modulated Signal');
xlabel('Time---->');
ylabel('Amplitude--->');
//demodulation
xdemod1=[];
xdemod2=[];
demod=[];
xdemod1=psk.*carrier1;
xdemod2=psk.*carrier2;
xdemod=xdemod2-xdemod1;
fori=1:length(I)
ifi==1
    s=sum(xdemod(1:length(t)))
else
```

```
        s=sum(xdemod((i-1)*length(t):i*length(t)))
end
if(s>0)
demod=[demod,zeros(1,length(t))]
else
demod=[demod,ones(1,length(t))]
end
end
subplot(4,1,4);
plot(demod);
xtitle('DeModulated Signal');
xlabel('Time---->');
ylabel('Amplitude--->');
```

OUTPUT
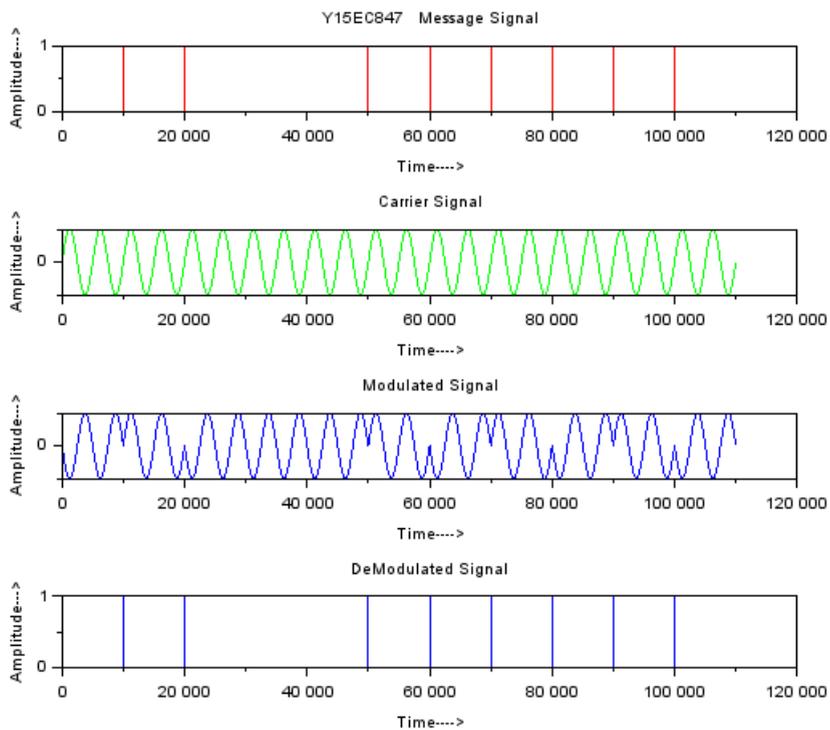
RESULT

ASK,FSK,BPSK modulation and Demodulation using scilab code successfully simulated and waveforms are generated.

BLOCK DIAGRAM OF QPSK MODULATION

BINARY I/P - 1   I - CHANNEL   BALANCED MODULATOR

$\sin\omega_c t$

CARRIER OSCILLATOR

SUMMER → QPSK O/P

90° PHASE SHIFT

$\cos\omega_c t$

BINARY I/P - 2   Q - CHANNEL   BALANCED MODULATOR
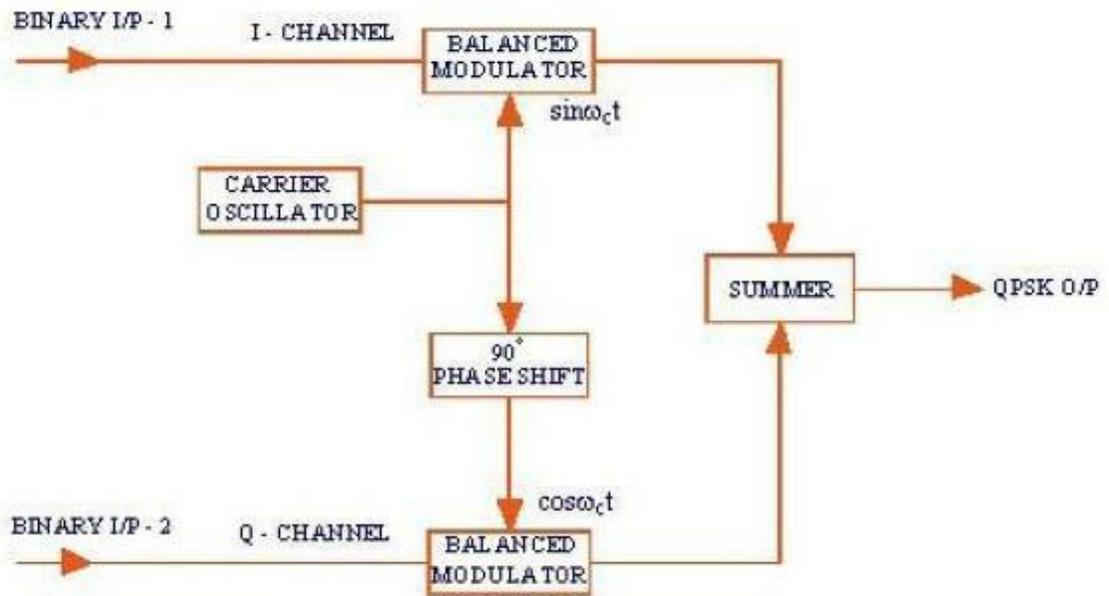
BLOCK DIAGRAM OF QPSK DEMODULATION

I-CHANNEL   BALANCED MODULATOR → LPF → COMPARATOR → DETECTOR → BINARY DATA-1

$\sin\omega_c t$

CARRIER RECOVERY

QPSK I/P SIGNAL

90° PHASE SHIFT

$\cos\omega_c t$

Q-CHANNEL   BALANCED MODULATOR → LPF → COMPARATOR → DETECTOR → BINARY DATA-2
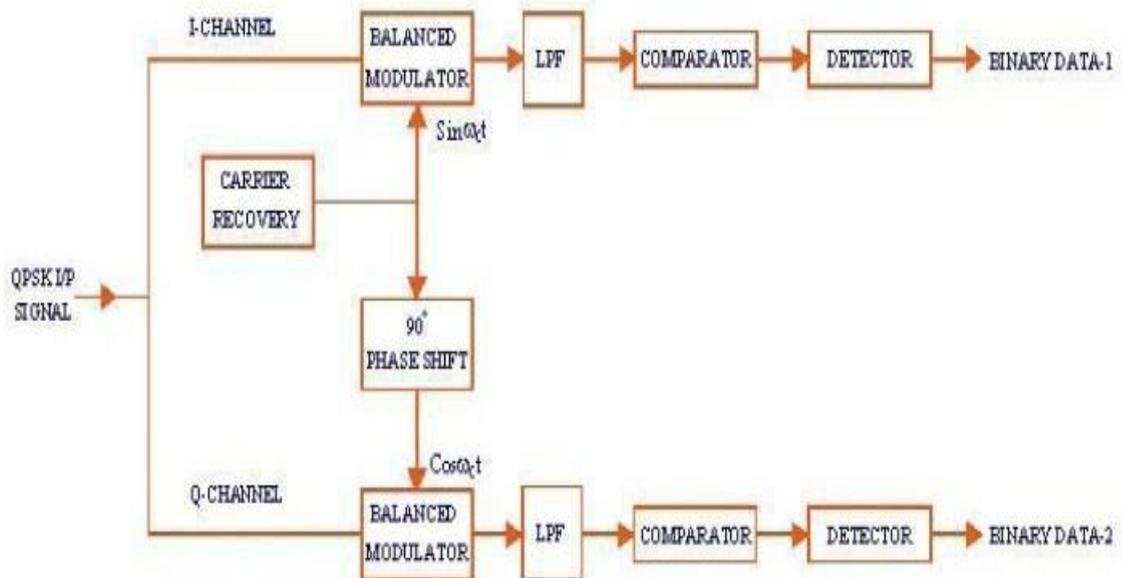
8. SIMULATION OF DPSK, QPSK, QAM GENERATIONSCHEMES

AIM:

To generate a DPSK,QPSK,QAM modulated waveform and to simulate using SCILAB

THEORY

QPSK MODULATOR

Quaternary phase shift keying (QPSK), or quadrature PSK as it is sometimes called, is another form of angle-modulated, constant-amplitude digital modulation. QPSK is an M-ary encoding technique where M = 4 (hence, the name "quaternary," meaning "4"). With QPSK four output phases are possible for a single carrier frequency. Because there are four different output phases, there must be four different input conditions. Because the digital input to a QPSK modulator is a binary (base 2) signal, to produce four different input conditions, it takes more than a single input bit. With two bits, there are four possible conditions: 00, 01, 10 and 11. Therefore, with QPSK, the binary input data are combined into groups of two bits called dibits. Each dibit code generates one of the four possible output phases. Therefore, for each two - bit dibit clocked into the modulator, a single output change occurs. Therefore, the rate of change at the output (baud rate) is onehalf of the input bitrate.

A block diagram of QPSK modulator is shown in above Figure. Two bits (a dibit) are clocked into the bit splitter. After both bits have been serially inputted, they are simultaneously parallel outputted. One bit is directed to the I channel and the other to the Q channel. The 1- bit modulates a carrier that is in phase with the reference oscillator (hence, the name "I" for "in phase" channel), and the Q bit modulates a carrier that is 90° out of phase or in quadrature with the reference carrier (hence, the name "Q" for "quadrature" channel).

DPSK SCILAB CODE

```
clc ;
bk = [1 ,0 ,0 ,1 ,0 ,0 ,1 ,1]; // input digital sequence
fori = 1: length (bk)
if(bk(i) ==1)
bk_not (i) =~1;
else
bk_not (i)= 1;
end
end
dk_1 (1) = bool2s ( 1&bk (1) ); // initial value of differential encoded sequence
dk_1_not (1) = bool2s (0&bk_not(1));
dk (1) = bitxor ( dk_1 (1) ,dk_1_not (1)) // f i r s t b i t o f dpsk encoder
fori =2: length (bk)
dk_1 (i) = dk(i -1);
dk_1_not (i) = ~dk(i -1);
dk(i) = bitxor ( bool2s ( dk_1 (i)&bk(i)),bool2s (dk_1_not (i)&bk_not (i)));
end
fori =1: length (dk)
if(dk(i) ==1)
dk_radians (i)=0;
elseif (dk(i) ==0)
dk_radians (i)=%pi;
end
end
disp (' Table 7.3 Illustrating the Generation of DPSK Signal')
disp ( '' )
disp (bk , '( bk )')
bk_not = bk_not ';
disp (bk_not , ' (bk not)')
dk = dk ';
disp (dk , 'Differentially encoded sequence (dk)')
dk_radians = dk_radians ';
disp (dk_radians, 'Transmitted phase in radians')
```

OUTPUT

" Table 7.3 Illustrating the Generation of DPSK Signal"

  ""

  1. 0. 0. 1. 0. 0. 1. 1.

 "(bk)"

  0. 1. 1. 0. 1. 1. 0. 0.

" (bk not)"

  1. 0. 1. 1. 0. 1. 1. 1.

 "Differentially encoded sequence (dk)"

  0. 3.1415927 0. 0. 3.1415927 0. 0. 0.

<u>**SCILAB CODE FORQPSK**</u>

SCILAB CODE

```
clear ;
clc ;
close ;
M =4;
i = 1:M;
t = 0:0.001:1;
fori = 1:M
s1(i ,:) = cos (2* %pi *2* t)*cos ((2*i -1) *%pi /4) ;
s2(i ,:) = -sin (2* %pi *2* t)*sin ((2*i -1) *%pi /4) ;
end
S1 =[];
S2 = [];
S = [];
Input_Sequence=[0 ,1 ,1 ,0 ,1 ,0 ,0 ,0];
m = [3 ,1 ,1 ,2];
fori =1: length (m)
S1 = [S1 s1(m(i) ,:) ];
S2 = [S2 s2(m(i) ,:) ];
end
S = S1+S2;
```

figure

subplot (3 ,1 ,1)

a = gca ();

a.x_location = "origin";

plot (S1)

title ('Binary PSK wave o f Odd-numbered bits of input sequence')

subplot (3 ,1 ,2)

a = gca ();

a.x_location = "origin";

plot (S2)

title ('Binary PSK wave o f Even-numbered bits of input sequence')

subplot (3 ,1 ,3)

a = gca ();

a.x_location = "origin";

plot (S)

title ('QPSK waveform ')

OUTPUT
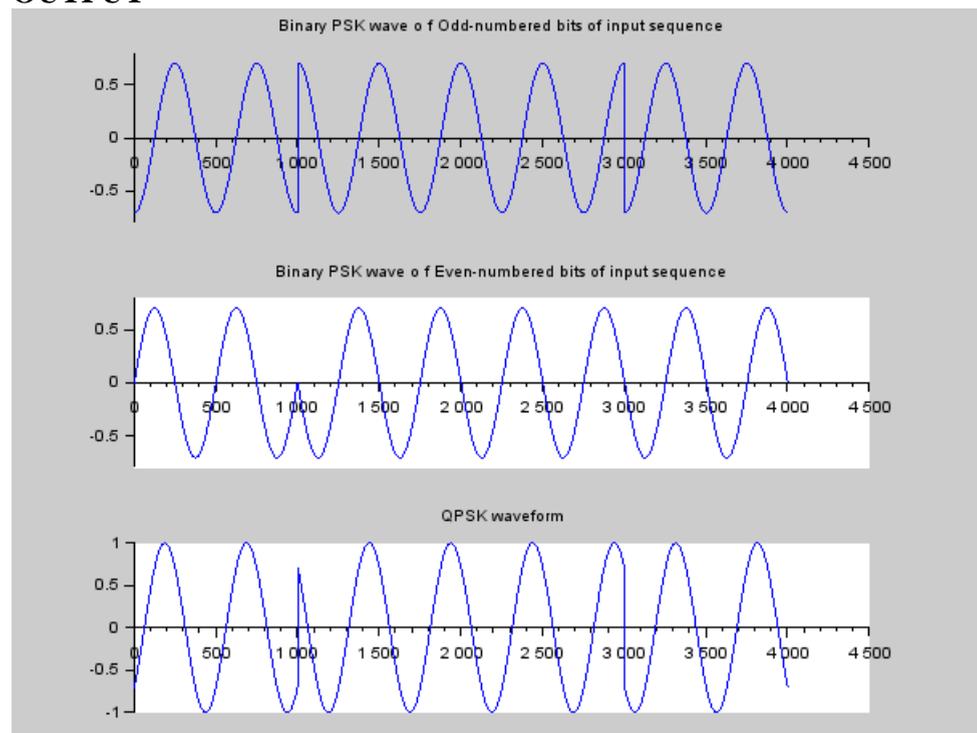
QAM GENERATION

SCILAB CODE

```
//Program Title: Study and generate 8-QAM waveforms
clear;
clc;
close;
T=3;//One Symbol period
t=0:0.01:T;// Sampling Matrix for one symbol period
f=1/T;// Carrier frequency (cycles per bit period)
I=[000001010011100101110111];//data stream giving tribits equivalent to 0,1,2,3,4,5,6,7
//Polar NRZ Converter
I_PNRZ=[]//empty matrix for Polar NRZ data
forn=1:length(I)
ifI(n)==0then
I_PNRZ=[I_PNRZ,-1]
else
I_PNRZ=[I_PNRZ,1]
end
end
I_Carrier=sqrt(2/T)*cos(2*%pi*f*t);// In phase carrier
Q_Carrier=sqrt(2/T)*sin(2*%pi*f*t);// Quadrature phase carrier
//Generation of 8-QAM Waveform
z=0;//Starting point of plot on x-axis
forn=1:3:length(I_PNRZ)
Q_Bit=I_PNRZ(n)//Set Q Bit Value
I_Bit=I_PNRZ(n+1)//Set I Bit Value
C_Bit=I_PNRZ(n+2)//Set C Bit Value
ifC_Bit==-1then//Set PAM, Product of C with I or Q
QC=0.5*Q_Bit//Set half amplitude
IC=0.5*I_Bit//Set half amplitude
else
QC=Q_Bit//Set full amplitude
IC=I_Bit//Set full amplitude
end
subplot(3,1,1)//QC Plot
```

```scilab
a=gca();
a.data_bounds=[0,-1.5;length(I_PNRZ),1.5];
a.x_location="origin";
a.grid=[1,1];
title('Q-PAM')
plot((t+z),Q_Carrier*QC);//Q_Carrier * Q-PAM (Q Balance Modulator)
plot((t+z),QC,'r');//Q-PAM Output


subplot(3,1,2)//IC Plot
a=gca();
a.data_bounds=[0,-1.5;length(I_PNRZ),1.5];
a.x_location="origin";
a.grid=[1,1];
title('I-PAM')
plot((t+z),I_Carrier*IC);//I_Carrier * I-PAM (I Balance Modulator)
plot((t+z),IC,'r');//I-PAM Output


subplot(3,1,3)//8-QAM Plot
a=gca();
a.data_bounds=[0,-1.5;length(I_PNRZ),1.5];
a.x_location="origin";
a.grid=[1,1];
title('8-QAM')
plot((t+z),(I_Carrier*IC)+(Q_Carrier*QC));//I-PAM + Q-PAM (Adder)
plot((t+z),I_Carrier,'r');//I Carrier for reference
plot(((t/3)+z),Q_Bit,'c');//Q Bit for reference
plot(((t/3)+1+z),I_Bit,'b');//I Bit for reference
plot(((t/3)+2+z),C_Bit,'m');//C Bit for reference
z=z+3;//Move starting point of plot on x-axis by 3 bits (1 symbol) period
end


OUTPUT
```

Q-PAM

I-PAM

8-QAM
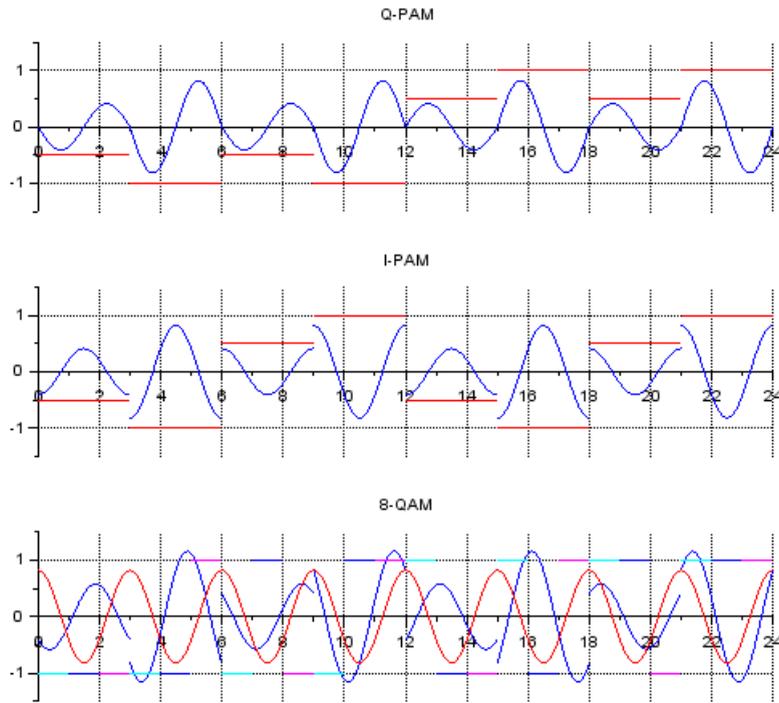
RESULT:

DPSK,QPSK,QAM simulated using SCILAB and generated relevant waveforms.

EXPERIMENTNO10   SIGNAL CONTELLATION FOR BPSK, QPSK &QAM

AIM:

To plot the constellation diagram of digital modulation system BPSK, QPSK & QAM using

SCILAB

SOFTWARE USED:

SCILAB

THEORY:

A constellation diagram is a representation of a signal modulated by an arbitrary digital modulation scheme. It displays the signal as a two dimensional scatter diagram in the complex plane at symbol sampling instants. It can also be viewed as the possible symbols that may be selected by a given modulation scheme as points in the complex plane.

SCILAB CODE: BPSK
BPSK

```
clear

clc ;

close ;

M =2;

i = 1:M;

y = cos (2* %pi +(i -1)* %pi);

annot = dec2bin ([length(y)-1: -1:0] ,log2(M));

disp (y, ' coordinates of message points ' )

disp (annot , 'Message points ' )

figure ;

a = gca ();

a.data_bounds = [ -2 , -2;2 ,2];

a.x_location = "origin";

a.y_location = "origin";

plot2d (real (y (1) ),imag (y (1) ) ,-9)

plot2d (real (y (2) ),imag (y (2) ) ,-5)

xlabel ('In-Phase');

ylabel ('Quadrature');

title (' Constellation for BPSK')

legend ([ ' message point 1 ( binary 1) ' ; ' message point 2 ( binary 0) '] ,5)
```

OUTPUT

Constellation for BPSK
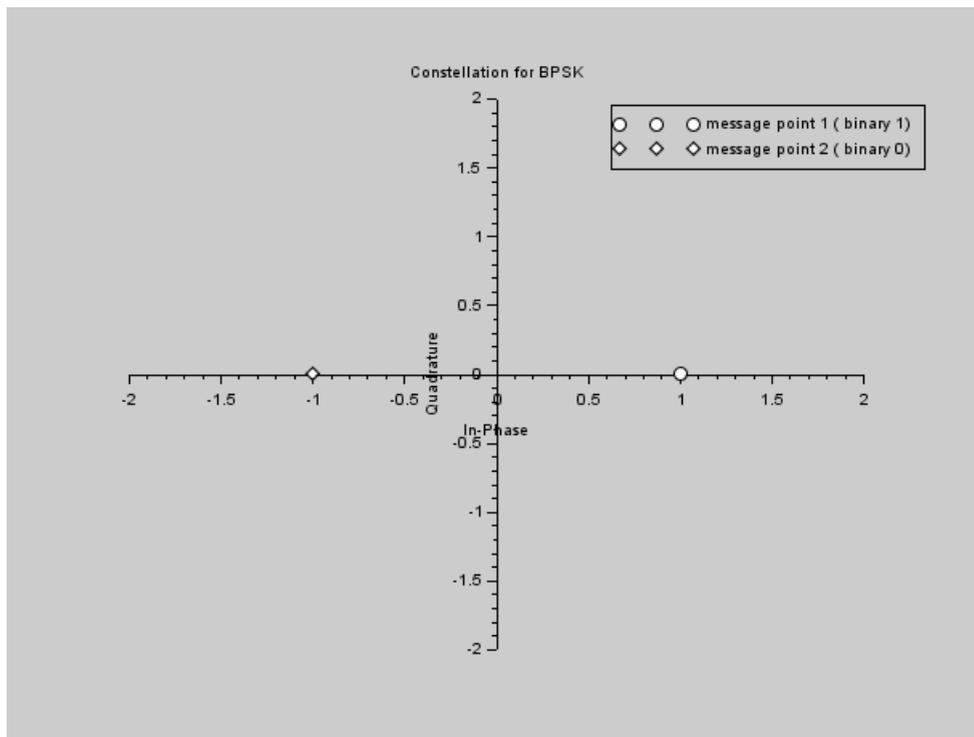
 SCILAB CODE FOR QPSK

```
clear
clc;
close;
M =4;
i = 1:M;
y =cos((2*i -1) *%pi/4)-sin((2*i -1)*%pi/4)*%i;
annot = dec2bin([0:M-1],log2(M));
disp (y, 'coordinates of message points')
disp (annot,'dibits value')
figure;
a = gca();
a.data_bounds = [ -1 , -1;1 ,1];
a.x_location = "origin";
a.y_location = "origin";
plot2d (real (y (1) ),imag (y (1) ) ,-2)
plot2d (real (y (2) ),imag (y (2) ) ,-4)
plot2d (real (y (3) ),imag (y (3) ) ,-5)
plot2d (real (y (4) ),imag (y (4) ) ,-9)
xlabel ('In-Phase');
ylabel ('Quadrature');
```

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title ('Constellation for QPSK')

legend (['message point 1 (dibit 1 0) ' ; ' message point 2 (dibit 0 0) ' ; ' message point 3 (dibit 0 1) ' ;
'message point 4 (dibit 1 1) '],5)

OUTPUT
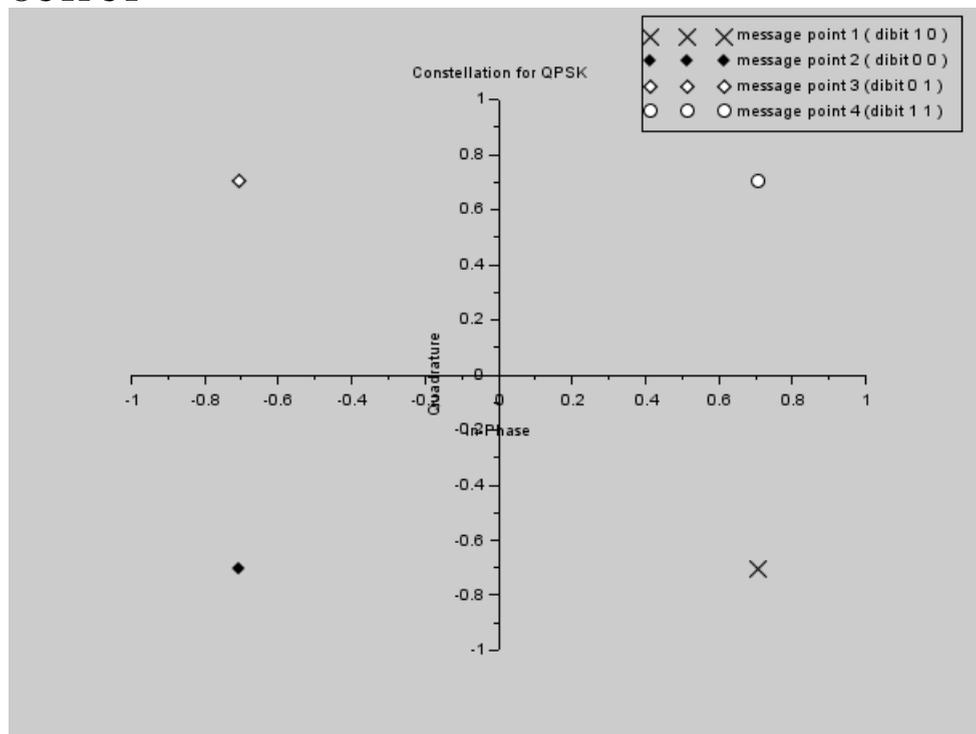
RESULT:
Thus the constellation diagrams of digital modulation system BPSK, QPSK & QAM are
simulated & plotted in SCILAB

EXPERIMENTNO11(A).   IMPLEMENTATION OF LINEAR BLOCKCODES

AIM:

Construct a (7, 4) linear block code whose generator matrix is given by,

$$G = \begin{bmatrix} 1000111 \\ 0100110 \end{bmatrix}$$

Determine all code words and the minimum weight of the code.

SOFTWARE REQUIRED:

SCILAB 6.1.0

THEORY:

Linear block codes:

Its one of the error control coding. Linear codes means that sum of any two code vector gives another code vector. Also it is a systematic code. Block codes in which the message bits are transmitted in unaltered form are called systematic code.

Consider an (n, k) linear block code in which „k" is a message bit, „n" is block length and b=n-k is a parity check bit.

bo,b1,b2…............................ bn-k-1	mo,m1,m2.m3…................................ mk-1

Structure of code word

Message Vector m= [mo,m1,m2...................mk-1]

Parity check vector b= [b0,b1, b2,… .... bn-k-1]

Code vector X= [Xo,X1,X2…...............Xn-1]

b=m x P

Where,

$$P = \begin{pmatrix} P_{00} \ P_{10} \ldots \ldots \ldots \ldots \ldots P_{n-k-1,0} \\ P_{01} . \\ \quad . \quad . \quad \quad \quad \quad . \\ \quad . \quad . \quad \quad \quad \quad . \end{pmatrix}$$

Define the k by n generator matrix G= {P: Ik}

Define the (n-k) by k sub matrix H= [Ik : P^T]

Parity check vector b= m x P

Code vectorX=

Message vector	check vector

ALGORITHM:

• From the given (n, k) block code assign the values of „k" nothing

   but Number of messagebit.

• Assign the given generatormatrix.

• Compute the check vector, then arrange code vector by

   combining Message and checkvector.

• Find the weight of the code that is by finding minimum hamming

   weight Of the Code which is nothing but number of non zero bits in a

   codeVector.

• Find the minimum weight of the code, from that we can understandthat

the given block code can able to detect and correct how many bits.

• Display the all possible code vector and weight of thecode.

SCILAB CODE FOR LINEAR BLOCK CODES:

```
//Note: Details of scilab software version and OS version used:
//Tested on OS: Windows 7 SP1, 64 bit and Windows XP SP3, 32 bit
//Scilab version: 5.4.1 (Tested on both 32 bit and 64 bit versions)
//Program Title: Linear Block Codes (7,4)
clc;
clear;
k=4;//Information message length
n=7;//Coded word length
P=[110;011;111;101]//Parity Matrix
disp(P,'Parity Matrix P')
G=[Peye(k,k)]//Generator Matrix to create code word in P1P2P3D1D2D3D4 format
disp(G,'Generator Matrix G')
H=[eye(n-k,n-k);P]'//Parity Check Matrix
disp(H,'Parity Check Matrix H')
//All_M = All 16 posibilities for Information Message Matrix
All_M=[0000;0001;0010;0011;
0100;0101;0110;0111;
1000;1001;1010;1011;
1100;1101;1110;1111]
//Calculate all 16 posibilecodewords
CodedMat=All_M*G;
CodedMat=modulo(CodedMat,2);//Convert generated code into binary
disp(CodedMat,'Codewords Matrix')
//Calculate Hamming Distances
HamDist=sum(CodedMat,'c')//Sum over the rows of CodedMat(column of values)
disp(HamDist,'Hamming Distances');
//Find Minimum non-zero Hamming Distance
[row,col]=find(HamDist==0);//find elements that are zero
HamDist(row,:)=[];//Remove all rows that are zero (replace by null)
MinHamDist=min(HamDist)//Find Minimum non-zero Hamming Distance
disp(MinHamDist,'Minimum Non-Zero Hamming Distance')
```

```
//Calculate Error Detection Capability
ErrDetCap=MinHamDist-1;
disp(ErrDetCap,'Error Detection Capability');
//Calculate Error Correction Capability
ErrCorCap=(MinHamDist-1)/2;
disp(ErrCorCap,'Error Correction Capability');
//Generate random message
RandMessage=modulo(round(16*rand()),16)+1//Get random number between 1 to 16
M=All_M(RandMessage,:)//Select a random row from Message Matrix All_M as Information Message
disp(M,'Information Message M')
C=CodedMat(RandMessage,:)//Select a random row from Coded Matrix CodedMat as Coded Message
disp(C,'Coded Message C')
//Transmit random message
R=C//Create recieved code word
//Generate error at random bit position
ErrPos=modulo(round(8*rand()),8)//Get random number between 0 to 7
ifErrPos==0then
//Do nothing, as '0' means no error
else
ifR(ErrPos)==0then
R(ErrPos)=1//Invert bit at Erroneous Bit Position
else
R(ErrPos)=0//Invert bit at Erroneous Bit Position
end
end

disp(R,'Recieved Code word R')


//Error Correction


S=R*H'//Find Syndrome Matrix
S=modulo(S,2);//Convert Syndrome Matrix into binary
disp(S,'Syndrome Matrix R*H(transpose)')


ifS==[000]then//[0 0 0] indicates no error
disp(R,'Recieved Code without error')
```

```
disp(R(4:7),'Recieved Information Message')//Extract and display  Message from code word
else
//Find erroneous bit position
//Here we find colomn within H matrix with pattern simmilar to Syndrome Matrix
//The position number of that colomn is equivalent to erroneous bit position

ErrPos=1//Initiallize erroneous bit position
d=[H(:,ErrPos)]'//Transpose of first coloumn of H matrix
//(Transpose is used to convert colomn to row as syndrome is in row format)

while((d(1)<>S(1))|(d(2)<>S(2))|(d(3)<>S(3)))do//Check element wise inequallity for any element (OR condition)
ErrPos=ErrPos+1//Increament erroneous bit position (Point to next colomn)
d=[H(:,ErrPos)]'//Transpose of next coloumn of H matrix
end

disp(ErrPos,'Erroneous Bit Position')

//Error correction
ifR(ErrPos)==0then
R(ErrPos)=1//Invert bit at Erroneous Bit Position
disp(R,'Recieved Code with error corrected')
disp(R(4:7),'Recieved Information Message')//Extract and display  Message from code word
else
R(ErrPos)=0//Invert bit at Erroneous Bit Position
disp(R,'Recieved Code with error corrected')
disp(R(4:7),'Recieved Information Message')//Extract and display Message from code word
end
end
```

OUTPUT

> 1. 1. 0.
>
> 0. 1. 1.
>
> 1. 1. 1.
>
> 1. 0. 1.

"Parity Matrix P"

1. 1. 0. 1. 0. 0. 0.

0. 1. 1. 0. 1. 0. 0.

1. 1. 1. 0. 0. 1. 0.

1. 0. 1. 0. 0. 0. 1.

"Generator Matrix G"

1. 0. 0. 1. 0. 1. 1.

0. 1. 0. 1. 1. 1. 0.

0. 0. 1. 0. 1. 1. 1.

"Parity Check Matrix H"

0. 0. 0. 0. 0. 0. 0.

1. 0. 1. 0. 0. 0. 1.

1. 1. 1. 0. 0. 1. 0.

0. 1. 0. 0. 0. 1. 1.

0. 1. 1. 0. 1. 0. 0.

1. 1. 0. 0. 1. 0. 1.

1. 0. 0. 0. 1. 1. 0.

0. 0. 1. 0. 1. 1. 1.

1. 1. 0. 1. 0. 0. 0.

0. 1. 1. 1. 0. 0. 1.

0. 0. 1. 1. 0. 1. 0.

1. 0. 0. 1. 0. 1. 1.

1. 0. 1. 1. 1. 0. 0.

0. 0. 0. 1. 1. 0. 1.

0. 1. 0. 1. 1. 1. 0.

1. 1. 1. 1. 1. 1. 1.

"Codewords Matrix"

0.

3.

4.

3.

3.

4.

3.

4.

3.

4.

3.

4.

4.

3.

4.

7.

"Hamming Distances"

3.

"Minimum Non-Zero Hamming Distance"

2.

"Error Detection Capability"

1.

"Error Correction Capability"

0. 0. 1. 1.

"Information Message M"

0. 1. 0. 0. 0. 1. 1.

"Coded Message C"

0. 1. 0. 0. 0. 0. 1.

"Recieved Code word R"

1. 1. 1.

"Syndrome Matrix R*H(transpose)"

6.

"Erroneous Bit Position"

0. 1. 0. 0. 0. 1. 1.

"Recieved Code with error corrected"

    0.   0.   1.   1.

 "Recieved Information Message"

RESULT:

All possible code vector and weight of the given linear block code is found. From the values of dmin=3 the given linear block code is found that is hamming code.

EXPERIMENTNO12.(B)   IMPLEMENTATION OF CYCLIC CODE GENERATION

AIM:

To simulate the generates Matrix, Code word, Parity check Matrix and error syndrome for a (7, 4) cyclic code using SCILAB

APPARATUS REQUIRED:

1. Personalcomputer.
2. SCILAB 6.1.0.

THEORY:

Error control coding is the processor of adding redundant list to the information bits, So on to simulate two level objectives at the receiver. Error detection and correction. A block code is linear if any linear combination of its code words a code is cyclic, if any cyclic shift of a code and is also a code word. They are usually denoted by (n, k) in which the first position of k bits is always identical to the message sequence to the transmitted. The block length is denoted by n.

ALGORITHM:

CYCLIC CODES

Initialize the message bits (k) and block length (m)

Select the message bits

Generate the polynomial

Encode the message bits

Introduce and in the encoded messagebits

Decode the original message from the RX message

Display the Encoded & Decodedmessage

SPECIFICATIONS FOR THE (7, 4) CYCLIC CODES

PARAMETERS	SPECIFICATIONS	DIMENSIONS
Message bits, "M"	The message bits	4 bits
cyclic	Used at the TX for Encoding generation at the TX	4 bits
Code word „X"	The cyclic code	7 key matrix

CYCLIC CODE GENERATION USING MAT LAB CODINGS:

Cyclic Coding using Scilab code

1. Encoder for the (7,4) Cyclic Hamming Code

```
clc;
close;
clear;
D = poly (0, 'D' );
g = 1+D+0+D ^3; // generator polynomial
m = (D^3) *(1+0+0+ D^3); //message sequence
[r,q] = pdiv (m,g);
p = coeff (r);
disp (r, ' remainder in polynomial form ' )
disp (p, ' Parity bits are : ' )
disp ( ' Table 8.3 Contents of the Shift Register in the Encoder of fig 8.7 for Message Sequence (1 0 0 1 ) ')
disp ( '-----------------------------------------------------------------------')
disp ( ' Shift Input Register Contents ' )
disp ( '----------------------------------------------' )
disp ( ' 1 1 1 1 0 ' )
disp ( ' 2 0 0 1 1 ' )
disp ( ' 3 0 1 1 1 ' )
disp ( ' 4 1 0 1 1 ' )
disp ( '-------------------------' )
```

OUTPUT
D +D²
" remainder in polynomial form "

  0. 1. 1.

" Parity bits are : "

" Table 8.3 Contents of the Shift Register in the Encoder of fig 8.7 for Message Sequence (1 0 0 1) "

  "--"

" Shift Input Register Contents "

  "---"

" 1 1 1 1 0 "

" 2 0 0 1 1 "

" 3 0 1 1 1 "

" 4 1 0 1 1 "

2. syndrome calculator for the(7,4) Cyclic Hamming Code

```
clc;
D=poly(0,'D');
g=1+D+0+D^3;// g e n e r a t o r polynomial
C1=0+D+D^2+D^3+0+0+D^6;// e r r o r f r e ecodeword
C2=0+D+D^2+0+0+0+D^6;//middl e b i t i s e r r o r
[r1,q1]=pdiv(C1,g);
S1=coeff(r1);
S1=modulo(S1,2);
disp(r1,' remainder in polynomial form ')
disp(S1,' Syndrome bits for error free codeword are : ')
[r2,q2]=pdiv(C2,g);
S2=coeff(r2);
S2=modulo(S2,2);
disp(r2,' remainder in polynomial form for erroredcodeword ')
disp(S2,' Syndrome bits for erroredcodeword are : ')
```

 OUTPUT

  $2D +2D^2$

" remainder in polynomial form "

  0. 0. 0.

" Syndrome bits for error free codeword are : "

$1 + 3D + 2D^2$

" remainder in polynomial form for erroredcodeword "

   1.   1.   0.

" Syndrome bits for erroredcodeword are : "

RESULT:

Thus the simulation for cyclic code is done using SCILAB

EXP NO: 12SIMULATION FOR CONVOLUTIONAL CODING SCHEMES

AIM:

To simulate the generates Matrix, Code word, Parity check Matrix and error syndrome for a (7, 4) cyclic code using SCILAB.

APPARATUS REQUIRED:

1. Personalcomputer.

2. SCILAB 6.1.0software.

Scilab code Solution for Convolutional Coding Technique

```
clear ;
close ;
clc ;
 g1 = input ('Enter the input Top Adder Sequence:= ' )
 g2 = input ('Enter the input Bottom Adder Sequence:= ')
 m = input ('Enter the message sequence:= ')
x1 = round ( convol (g1 ,m));
x2 = round ( convol (g2 ,m));
x1 = modulo (x1 ,2);
x2 = modulo (x2 ,2);
 N = length (x1);
fori =1: length (x1)
x(i ,:) =[x1(N-i +1) ,x2(N-i +1)];
end
 x = string (x)
disp (x, ' x=')
```

OUTPUT

Enter the input Top Adder Sequence:= [1,1,1]

Enter the input Bottom Adder Sequence:= [1,0,1]

Enter the message sequence:= [1,1,0,0,1]

```
" x="
       "1"  "1"
       "1"  "0"
       "1"  "1"
       "1"  "1"
       "0"  "1"
```

RESULT:

Thus the simulation for convolutional code is done using SCILAB

Digital Communication block diagram
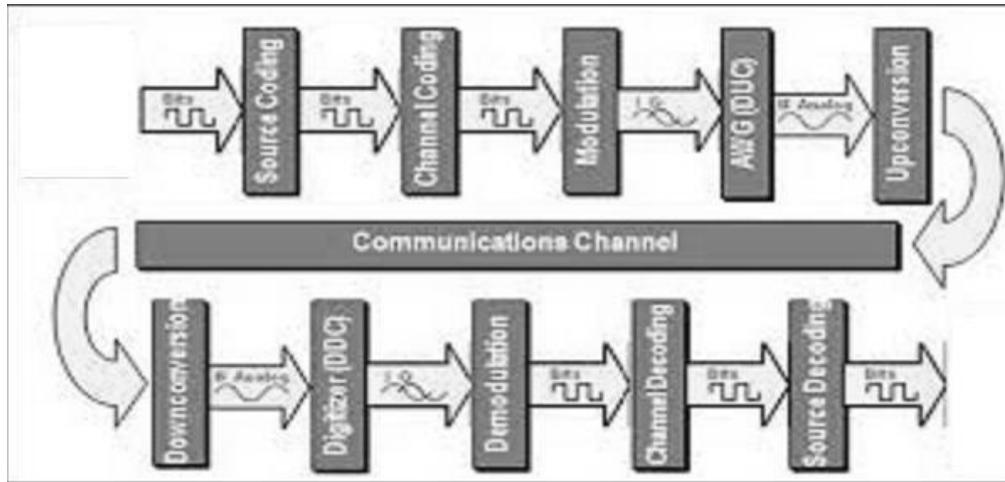
EXPT.NO 13	**Digital communication link simulation using Simulink**	DATE

AIM:
 To study the performance of digital communication link using SCILAB.

SOFTWARE REQUIREMENT:
 PC with SCILAB 6.1.0 software

RESULT:
             Thus the simulation for communication simulink is done using SCILAB.

EC8561 COMMUNICATION SYSTEM LAB

VIVA QUESTIONS & ANSWERS

Experiment no-1   SAMPLING ANDRECONSTRUCTION

1. State samplingtheory.

A continuous time signal can be completely represented in its samples and recovered back if the sampling frequency is twice of the highest frequency content of the signal.

$$f_s \geq 2f_m$$

2. What is meant by Nyquist rate and Nyquistinterval?

$f_s = 2f_m$ is called the Nyquist rate. $T_s = {}^1/_{2f_m}$ is called Nyquist interval.

3. Define the samplingprocess.

Sampling is the process of converting the continuous time continuous amplitude signal into discrete time continuous amplitude signal.

4. What is meant byquantisation?

Quantisation is the process of converting the discrete time continuous amplitude signal into discrete time discrete amplitudesignal.

5. What is meant by undersampling?

If the sampling rate is less than the nyquist rate, then that sampling is called under sampling.

6. What is meant by oversampling?

If the sampling rate is greater than nyquist rate, then that sampling is called over sampling.

7. What is meant by nyquistsampling?

If the sampling rate is exactly equal to the nyquist rate, then that sampling is called nyquist sampling.

8. What are the types ofsampling?

a.Naturalsampling   b. Flattopsampling   c.Idealsampling   d. Sample andhold

9. What is meant byreconstruction?

The process of reconstructing the continuous time signal back from discrete time signal is called reconstruction.

10. What is meant by aliasing? When does itoccur?

Aliasing is the overlapping of adjacent spectrums is known as aliasing. Aliasing results, if the sampling theorem is not satisfied.

11. Give an example for sample and hold circuit.

A switch followed by a grounded capacitor can be used as a sample and hold circuit. The switch samples the signal and the capacitor holds that value.

12. How to implement the sampler?

Sampler can be implemented by using a switch.

Experiment no-2   TIME DIVISION MULTIPLEXING

1. What is meant bymultiplexing?

Multiplexing is the process of simultaneously transmitting two or more individual signals over a single communication channel.

2. Explain the operation ofTDM.

In TDM, each signal is transmitted only for a short time period. The time is divided into time slots and each time slot is mapped to one signal.

3. What are the applications ofTDM?

Time division multiplexing is used in network data communication as well as in GSM mobile communications.

4. Differentiate TDM andFDM.

In TDM, the time is divided into time slots and each time slot is assigned to a signal. In FDM, the frequency band is divided into sub-bands and each frequency sub band is assigned to a signal.
TDM is based on time sharing. FDM is based on frequency sharing.

5. What is meant byCDMA?

In CDMA, many transmitters are allowed to transmit in a single channel without any interference. Each transmitter is assigned with a unique code and spread spectrum modulation techniques are used. CDMA stands for Code Division Multiple Access.

6. What is meant bySDMA?

SDMA stands for Space Division Multiple Access.

7. Define synchronousmultiplexer.

In synchronous digital multiplexers, a single clock drives all the sources, so all the sources will operate at the same bit rate.

Experiment no-3   AMPLITUDE MODULATION

1. Define amplitudemodulation.
In amplitude modulation, the amplitude of the carrier signal is varied according to the modulating signal.
2. What is the range of carrier used inAM?
500 Hz to 1500 KHz.

3. Which modulation is used for videosignals?

Amplitude modulation is used for video signals.

4. What are the types of amplitudemodulation?

   a. double sideband modulation(DSB-AM)

b. double sideband suppressed carrier modulation (DSBSC-AM)

c. single sideband carrier modulation(SSBAM)

d. single sideband suppressed carrier modulation (SSBSC-AM)

e. vestigial sideband modulation (VSBAM)

5. Define modulation index ofAM.

Modulation index indicates the amount of amplitude variation around the un-modulated carrier.

Modulation index, $h = \dfrac{M}{A}$

M is the peak amplitude value of modulating signal.

A is the peak amplitude value of carrier signal.

6. What are the advantages ofAM?

The AM transmitter and receiver are simple.

7. What are the disadvantages ofAM?

AM is more susceptible to noise.

8. What are the applications ofAM?

AM radio broadcasting., AM video transmission.

9. What are the types of continuous wave analogmodulation?

Amplitude modulation

Angle modulation

10. What isVCO?

In Voltage Controlled Oscillator, the frequency of the oscillator is varied according to the applied voltage.

11. What is the role of mixer in AMdemodulator?

Mixer is used to convert the radio frequency (RF) signal into intermediate frequency (IF) signal.

12. What is the use of two stageamplifier?

The two stage amplifier is used for improving the gain.

Experiment no-4   FREQUENCYMODULATION

1. Define frequencymodulation.

The frequency of the carrier signal is varied according to the amplitude of the modulating signal.

2. What is the range of carrier used inFM?

88 MHz to 108 MHz.

3. What are the types of frequencymodulation?

- Narrow band FM(h<1)
- Wide band FM(h≥1)

4. Define modulation index ofFM.

Modulation index of FM indicates the amount of frequency variation around the un-modulated carrier frequency.

$h = \dfrac{\Delta f}{f_m}$

where Δf is the peak frequency deviation and fm denotes the maximum frequency component present in the modulating signal.

5. What are the advantages ofFM?

Immunity to noise

6. What are the disadvantages of FM?

Bandwidth requirement is more in FM.

Transmitter and receiver are verycomplex.

7. What are the applications ofFM?
- For recording luminance component of thevideo.
- Audio signalbroadcasting.

8. What are the types of anglemodulation?

Frequency modulation and phase modulation.

9. What is the role of a limiter in FMsystem?

To limit the amplitude variations in a FM signal.

Experiment No–5   PULSE CODE MODULATION

1. Define Pulse codemodulation?

Each and every quantized samples will be encoded with sequence of zeros and ones with 'n' bits within sampling interval (Ts), So the bit duration will be Ts/n. as no. of bits (n) increases error decreases but bandwidth increases

2. How bits are needed to encode N differentlevels?

$Log_2 N$

3. Define stepsize?

It is the difference between two successive no.s (levels), i.e. voltage difference between one digital level and the next digital level. (Ex: 000 & 001, 1011 & 1100)

How to calculate Step size in PCM?

step size$(\Delta) = (V_{FS}-V_{min})/L$

where $L = 2^n$, n = no. of bits, VFS = full scale voltage

4. Define Quantizationerror?

It is the difference between sampled signal and Quantized signal

5. Define Differential pulse codemodulation?

In DPCM, The difference between the consequent samples will be encoded with sequence of zeros and ones

Experiment no-6   DELTA AND ADAPTIVE DELTAMODULATION

1. What is the reason to use deltamodulation?
Data rate and bandwidth required are more in PCM. So delta modulation is used.
2. How many bits are transmitted per sample in DM?

Only one bit per sample is transmitted in DM.
3. What are the two types of errors present inDM?

Slope over load distortion and granular noise.
4. Define slope overload distortion with an illustrativewaveform.

When the slope of the input signal is larger than the slope of staircase waveform, the staircase waveform can't follow the input signal variations. This error is known as slope overload distortion.
5. Define granular noise with an illustrativewaveform.

When the slope of the input signal is smaller than the slope of staircase waveform the granular noise occurs.

6. How to reduce the slope overload distortion?

The slope overload distortion can be minimized by increasing the step size of the quantizer.

7. How to reduce the granular noise?

This noise can be controlled by decreasing the step size of the quantizer

8. Explain the operation ofDM.

In delta modulation present sample value is compared with the previous sample value. If present sample value is greater than previous sample value, then '1' is transmitted. If present sample value is less than previous sample value, then '0' is transmitted.

9. What is the reason to go forADM?

To control the slope overload distortion and granular noise ADM is used.

10. Explain the operation ofADM.

The step size of the quantizer is made adaptive to the variations of input signal. If the input signal varies slowly, step size is decreased. If the input signal varies sharply, step size is increased.

11. What does the term adaptive mean inADM?

The step size of the quantizer is made adaptive to the variations of input signal.

12. What is the condition to avoid slope overload distortion inDM?

The slope overload distortion will not occur if $A > {}^{\Delta}\!/_{(2\pi fT)}$

Where Δ is the step size, f is the frequency of the modulating signal, T is the sampling period, A is the amplitude of the modulating signal.

13. Differentiate DM andADM.

In DM, the step size of the quantizer is fixed, but in ADM the step size is adaptive. The SNR of ADM is better than that of DM.

14. What is the use of higher order filters at the receiverside?

For making the frequency response of practical filter closer to ideal response.

Experimentno-7   **LINE CODINGTECHNIQUES**

1. Define linecoding.

Line coding is the waveform representation of digital data.

2. Differentiate uni-polar, bi-polar and polarsignalling.

- In uni-polar signalling, symbol '0' is represented by 0 volts. Symbol '1' is represented by a positivepulse.

- In polar signalling, symbol '0' is represented by negative pulse. Symbol '1' is represented by a positivepulse.

- In bi-polar signalling, symbol '0' is represented by 0 Volts. Symbol '1' is represented by alternative positive and negativepulse.

3. Differentiate NRZ and RZsignalling.

In Non return to zero (NRZ) signalling, the pulses are transmitted for the entire bit duration but in Return to zero (RZ) signalling, the pulses are transmitted only for half bitduration.

4. What is meant by NRZ-L, NRZ-M andNRZ-S?

The NRZ-Level is just like the uni-polar NRZ. In NRZ-M, the change in level occurs for symbol'1' and no change for symbol '0'. In NRZ-S, the change in level occurs for symbol'0' and no change for symbol'1'.

5. What are the types of bi-phase codingtechniques?

Bi-Φ-level, bi-Φ-mark and bi-Φ-space

6. What is meant by Bi-Φ-level, bi-Φ-mark and bi-Φ-spacesignalling?

- In Bi-Φ-level signalling, symbol '1' is represented by positive to negative transition at half bit interval and symbol '0' is represented by negative to positive transition at half bit interval.

- In bi-Φ-mark signalling, symbol '1' is represented by both starting transition and half bit interval transition and symbol '0' is represented by only startingtransition.

- In bi-Φ-space signalling, symbol '0' is represented by both starting transition and half bit interval transition and symbol '1' is represented by only startingtransition.

7. What is Manchester code? Draw the Manchester format for the data stream10110?

In Manchester code each bit of data is signified by at least one transition. Manchester encoding is therefore considered to be self-clocking, which means that accurate clock recovery from a data stream is possible. In addition, the DC component of the encoded signal is zero. Although transitions allow the signal to be self-clocking, it carries significant overhead as there is a need for essentially twice the bandwidth of a simple NRZ or NRZI encoding.

8. State any four desirable properties of linecode
- The PAM signal should have adequate timing content,
- The PAM signal should immune to channel noise andinterference
- The PAM signal should allow error detection and errorcorrection
- The PAM signal should be transparent to digital data beingtransmitted

Experiment no - 8 DIGITAL MODULATION TECHNIQUES (ASK, FSK, BPSK)

1. Give examples for digital modulationtechniques.

Amplitude shift keying (ASK), Frequency shift keying (FSK), Phase shift keying (PSK), Quadrature phase shift keying (QPSK), Quadrature amplitude modulation (QAM).

2. What is meant by pass-bandtransmission?

The transmission of the modulated signal is called pass band transmission because of its spectrum located at high frequency band. This spectrum is similar to the response of the band-pass filter.

3. What is meant by base-bandtransmission?

The transmission of the message signal without any modulation is called as baseband transmission because its spectrum is located nearby zero frequency. This spectrum is similar to the frequency response of low pass filter.

4. What is the need formodulation?

- For placing the signal at low noisefrequency.

- For reducing the antennasize.

5. Define ASK, FSK, PSK.

- ASK: In ASK, the digital data modulates the amplitude of thecarrier.

- FSK: In FSK, the digital data modulates the frequency of thecarrier.

- PSK: In PSK, the digital data modulates the phase of thecarrier.

6. What isQPSK?

In QPSK, the binary data sequence is converted to an odd bit sequence and an even bit sequence. Four different phase delays are assigned to the four combinations of two bits.

For 10, the phase delay is 45®. For 00, the phase delay is 135®. For 01, the phase delay is 225®. For 11, the phase delay is 315®.

7. What are the types ofQPSK?

Non offset QPSK, Offset QPSK (OQPSK), (π/4) QPSK.

8. What is meant by coherentdetection?

In coherent detection, a local carrier is generated at the receiver with the same frequency and phase as the transmitter clock.

9. What is meant by non-coherentdetection?

In non-coherent detection, there is no need of generating local carrier at the receiver.

Experiment no-9   DIGITAL MODULATION TECHNIQUES (DPSK, QPSK,QAM)

1.   What is meant byDPSK?

Differential Phase Shift Keying (DPSK) is an alternative form of digital modulation where the binary input information is contained in the difference between two successive signaling elements rather than the absolute phase.

2.   What is meant by Probability of error & Bit ErrorRate?

Probability of error P (e) & Bit Error Rate (BER) are often used interchangeably, although in practice they do have slightly different meanings. P (e) is a theoretical expectation of the bit error rate for a given system. BER is an empirical record of a systems actual bit error performance.

3. What is meant by antipodalsignaling?

The phase relationship between signaling elements for BPSK (i.e., 180 degree out of phase) is the optimum-signaling format, referred to as antipodal signaling, and occurs only when two binary signal levels are allowed and when one signal is the exact negative of the other.

4. Give the formula for the error distance of thePSK.

The error distance of the PSK is given by, d = (2 sin 180/M)*D Where, d – error distance, M – number of phases D – peak signal amplitude

5. What does 8-QAM & 16-QAM means? Eight QAM is an M-ary encoding technique

where M=8. The output signal from an 8-QAM is not constant amplitude. Sixteen QAM is also an M-ary system whereM=16.

6. What are all the types of FSK systems & explainthem?

There are two types of FSK system. -Non coherent FSK -Coherent FSK -With non-coherent FSK, the transmitter and receiver are not frequency or phase synchronized. -With coherent

FSK, local receiver reference signals are in frequency and phase lock with the transmitted signals.

7. Compare the bandwidth efficiency of BPSK and QPSK modulatedsignals.

The bandwidth efficiency of BPSK is 1 bit per cycle, where as that of QPSK is 2 bits per cycle. The bandwidth efficiency of QPSK is more because it encodes the signal with 4 different phase shifts. Therefore it combines two successive bits.

8. What is meant by peak frequency deviation?

Peak frequency deviation is the product of the binary input voltage and the deviation sensitivity of the VCO. ñG = vm (t) * k1 Where, ñG= Peak frequency deviation (Hz) vm(t)= peak binary modulating signal voltage (volts) k1 = deviation sensitivity (Hz per volt)

9. DefineQPSK.

QPSK is Quadrature phase –shift keying. In QPSK the phase of the carrier takes on one of the four equally spaced values Such as ë/4, 3ë/4, 5ë/4 and 7ë/4.

10. DefineDibit.

A unique pair of bits is called a dibit. Gray encoded set of dibits 10, 00, 01 & 11

11. Give the two basic operation of DPSKtransmitter.

1. Differential encoding of the input binary wave 2. Phase –shift keying hence, the name differential phase shiftkeying

12. What are the three broad types ofsynchronization?

1. Carrier synchronization 2. Symbol & Bit synchronization 3. Framesynchronization.

13. What is carriersynchronization?

The carrier synchronization is required in coherent detection methods to generate a coherent reference at the receiver. In this method the data bearing signal is modulated on the carrier in such a way that the power spectrum of the modulated carrier signal contains a discrete component at the carrier frequency.

14. What are the two methods for carriersynchronization?

1. Carrier synchronization using Mth Power loop 2. Costas loop for carriersynchronization

15. What is called symbol or bitsynchronization?

In a matched filter or correlation receiver, the incoming signal is sampled at the end of one bit or symbol duration. Therefore the receiver has to know the instants of time at which a symbol or bit is transmitted. That is the instants at which a particular bit or symbol status and when it is ended. The estimation of these times of bit or symbol is called symbol or bit synchronization.

16. What are the two methods of bit and symbolsynchronization?

1) Closed loop bit synchronization 2) Early late gate synchronizer

Experimentno-12   ERROR CONTROL CODING (Linear code and Cycliccode)

1. What is the need for error controlcoding?

For error detection and error correction of the code word.

2. What is meant by linearcode?

If sum of two code vector results in another valid code vector, then the code is said to be linear code.

3. What is meant by cycliccode?

If cyclic shift in a code word results in another valid code vector, then the code is said to be cyclic code.

4. Name some error control codingtechniques.

Linear block code, binary cyclic code, convolutional code, CRC codes, RS codes, BCH codes.

5. Name the code which is used only for errordetection.

CRC code can perform error detection but error correction is not possible.

6. In the linear block code program, what is the use of function'eye'?

'eye' is a library function used for initialising an identity matrix.

7. What is the meaning for errorpattern?

Error pattern which is also known as error vector is the vector containing zeros and ones as elements. Zero represents 'no error' on that position and one represents 'error' on that position. The size of the error pattern is same as that of the codevector.

8. Differentiate systematic and non-systematiccoding?

In a systematic code word, it is possible to distinguish the message bits and check bits in the code word.

In a systematic code word, it is not possible to distinguish the message bits and check bits in the code word.

Experimentno-13   CONVOLUTONAL CODINGSCHEMES

1. Mention is the properties of cycliccodes

Linearity property The sum of any two code word is also a valid code word Cyclic property Every cyclic shift of a valid code vector produces another valid code vector

2. Define hammingdistance.

The hamming distance between two code vectors is equal to the number of elements in which they differ. For example, let the two code words be, X = (101) and Y= (110) These two code words differ in second and third bits. Therefore the hamming distance between X and Y is two.

3. What is meant by transparency with respect to linecodes

The line code is said to be transparent if the synchronization between the transmitter and receiver is maintained for any type of input data sequence.

4. **Define hamming distance and calculate its value for two code words 11100 and 11011**

The hamming distance between two code vectors is equal to the number of elements in which they differ. For example, let the two code words be, X = (11100) and Y= (11011) D= 2 These two code words differ in second and third bits. Therefore the hamming distance between X and Y istwo.

5. **What is Golaycodes?**

Golay code is the (23,12) cyclic code whose generating polynomial is, G(p) = p11+p9+p7+p6+p5+p+1 This code has minimum distance of dmin = 7. This code can correct upto 3 errors. But Golay code cannot be generalized to other combinations of n and k.

6. **What is meant by systematic and non-systematiccodes?**

In a Systematic block code, message bits appear first and then check bits. In the nonsystematic code, message and check bits cannot be identified in the code vector.

7. **What is meant by linearcode?**

A code is linear if modulo-2 sum of any two code vectors produces another code vector. This means any code vector can be expressed as linear combination of other code vectors.

8. **What are the error detection and correction capabilities of hamming codes?**

The minimum distance (dmin) of hamming codes is „3". Hence it can be used to detect double errors or correct single errors. Hamming codes are basically linear block codes with dmin =3.

9. **What is meant by cyclic codes?**

Cyclic codes are the subclasses of linear block codes. They have the property that a cyclic shift of one codeword produces another code word.

10. **What are the advantages and disadvantages of convolutionalcodes?**

Advantages: 1. The decoding delay is small in convolutional codes since they operate o smaller blocks of data.

2. The storage hardware required by convolutional decoder is less since the block sizes are smaller.

Disadvantages: 1. Convolutional codes are difficult to analyze since their analysis is complex.

2. Convolutional codes are not developed much as compared to block codes.