

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**

REGULATION 2017

ACADEMIC YEAR: 2019 - 2020

**EC8561 COMMUNICATION SYSTEMS
LABORATORY**

STUDENT MANUAL

STUDENT NAME :

REGISTER NO :

SEMESTER :

YEAR :

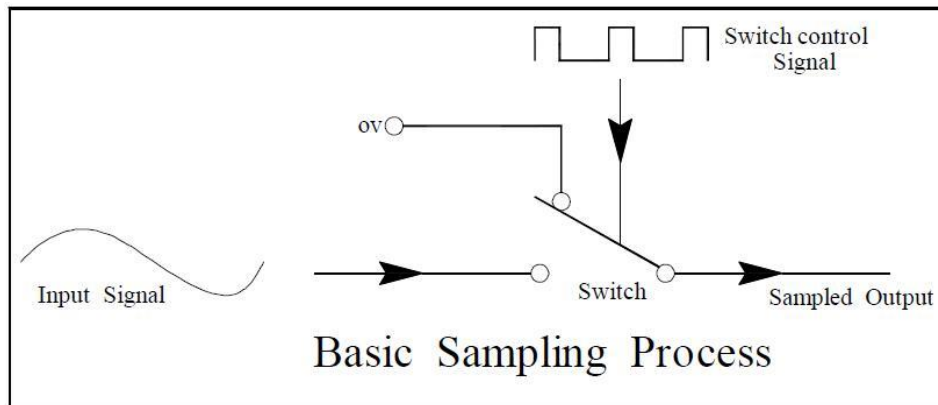
Faculty In-charge

Mr. S. DURAI RAJ AP/ECE

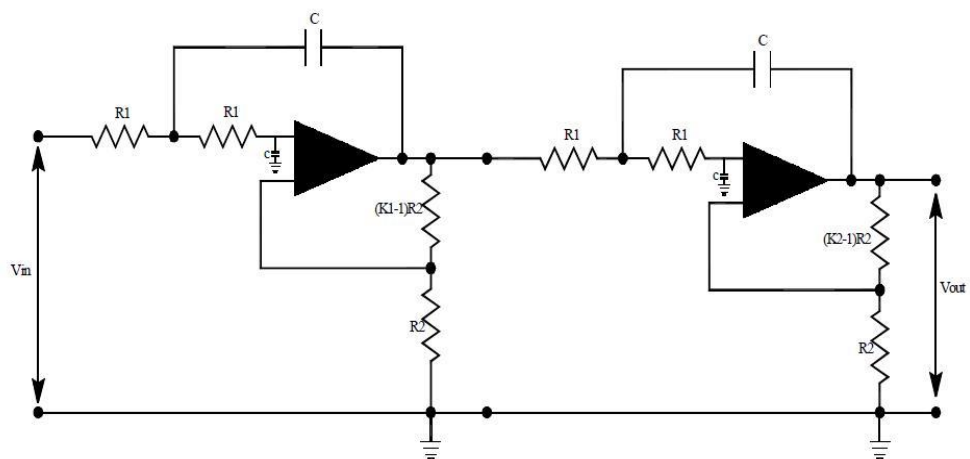
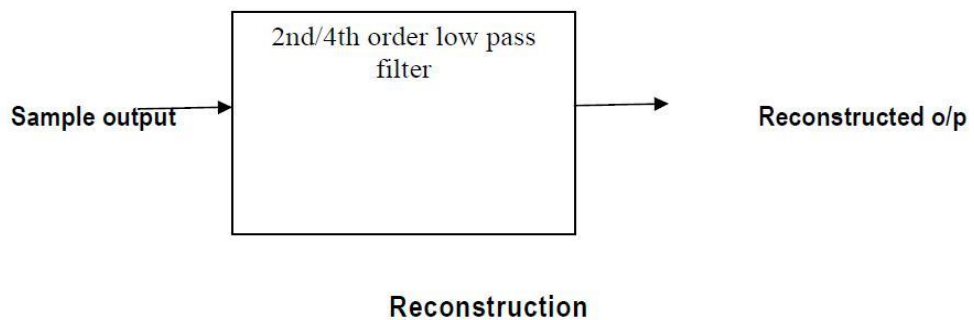
LIST OF EXPERIMENTS:

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



BLOCK DIAGRAM OF RECONSTRUCTION PROCESS



FOURTH ORDER BUTTERWORTH LOW PASS FILTER

1. ANALOG SAMPLING AND RECONSTRUCTION

AIM:

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

APPARATUS REQUIRED:

1. Communication trainer kit:
2. Power Cable
3. Patch cords.
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 F_m Hz can be completely represented by and reconstructed from the sample taken at a rate greater than or equal to $2F_m$ samples/second. This minimum sampling frequency is called as NYQUIST RATE i.e. for faithful reproduction of information signal $F_s \geq 2 F_m$.

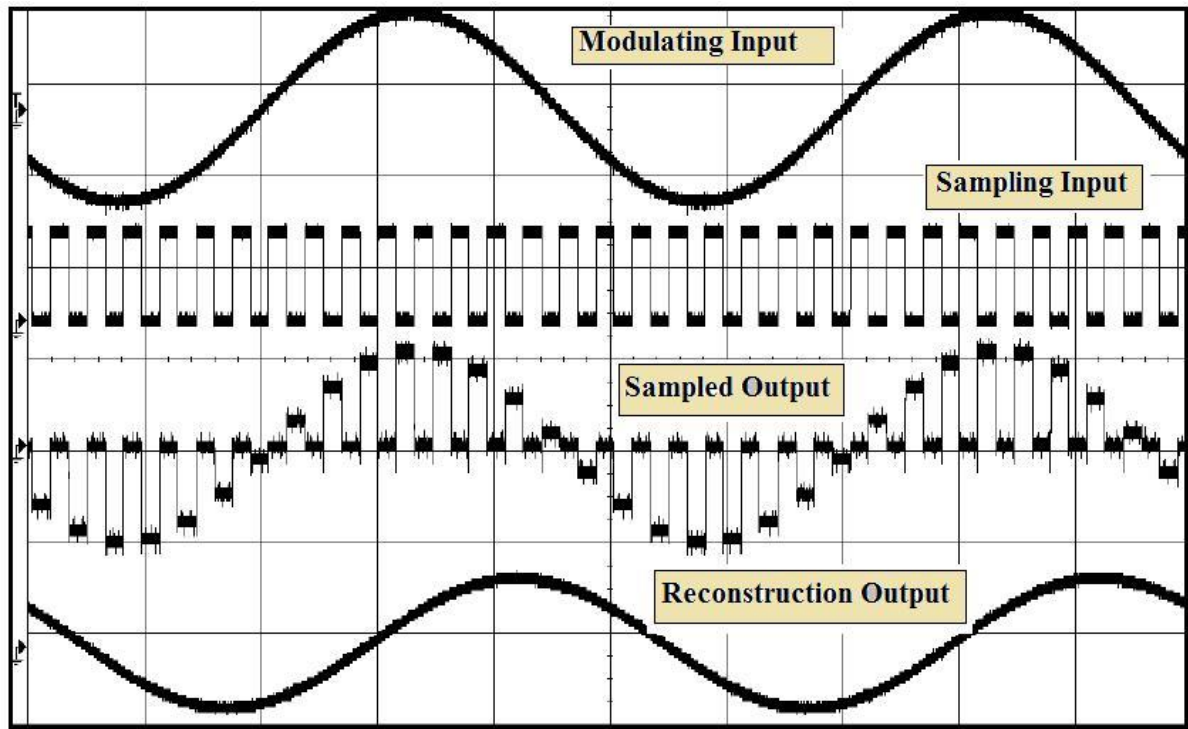
ALIASING

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 ($F_s \geq 2 F_m$)

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			

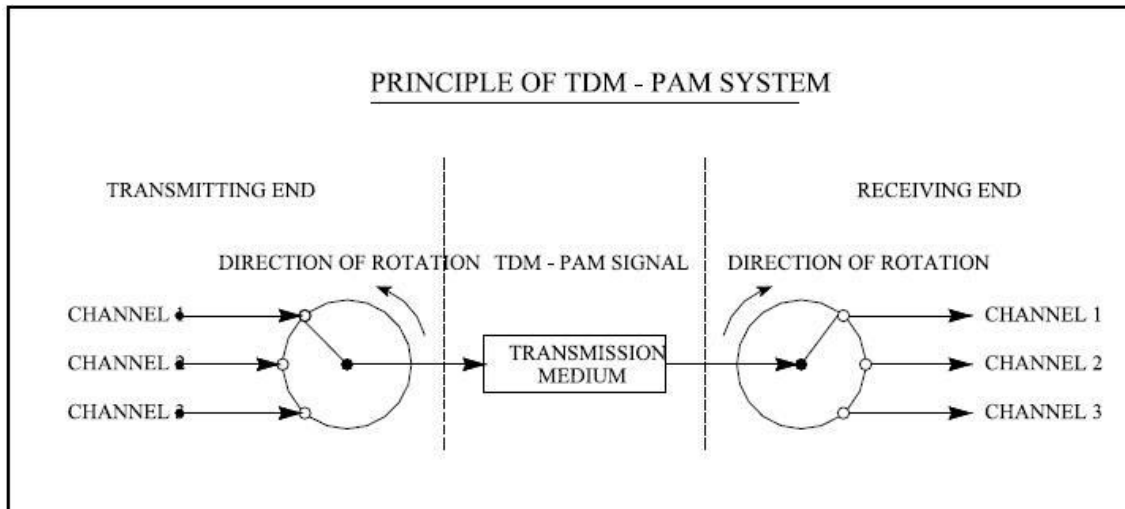
PROCEDURE

1. Connection are given as per the given block diagram.
2. To give an modulating input and sampled input (square wave form) to the input block.
3. To verify the output using CRO.
4. The output as given to the input of de modulated block and taken the output reading
5. Plot the graph

RESULT:

Thus the continuous-time signals are sampled and reconstructed from the samples at the receiver by original signal.

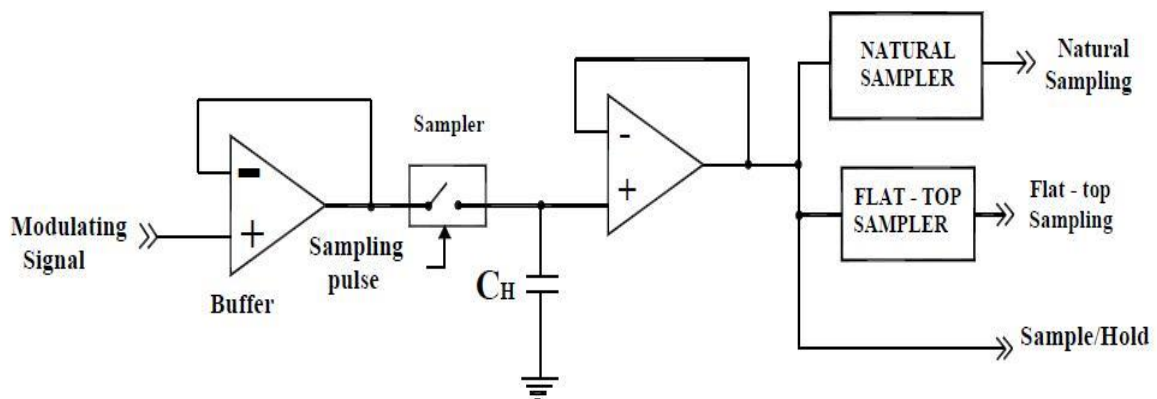
TDM USING PAM, MODULATION & DEMODULATION



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. TDM Trainer.
- ii. CRO.
- iii. Patch Chords.
- iv. CRO probe

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 Division Multiplexing
- ii. Time Division Multiplexing.

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 input channel.

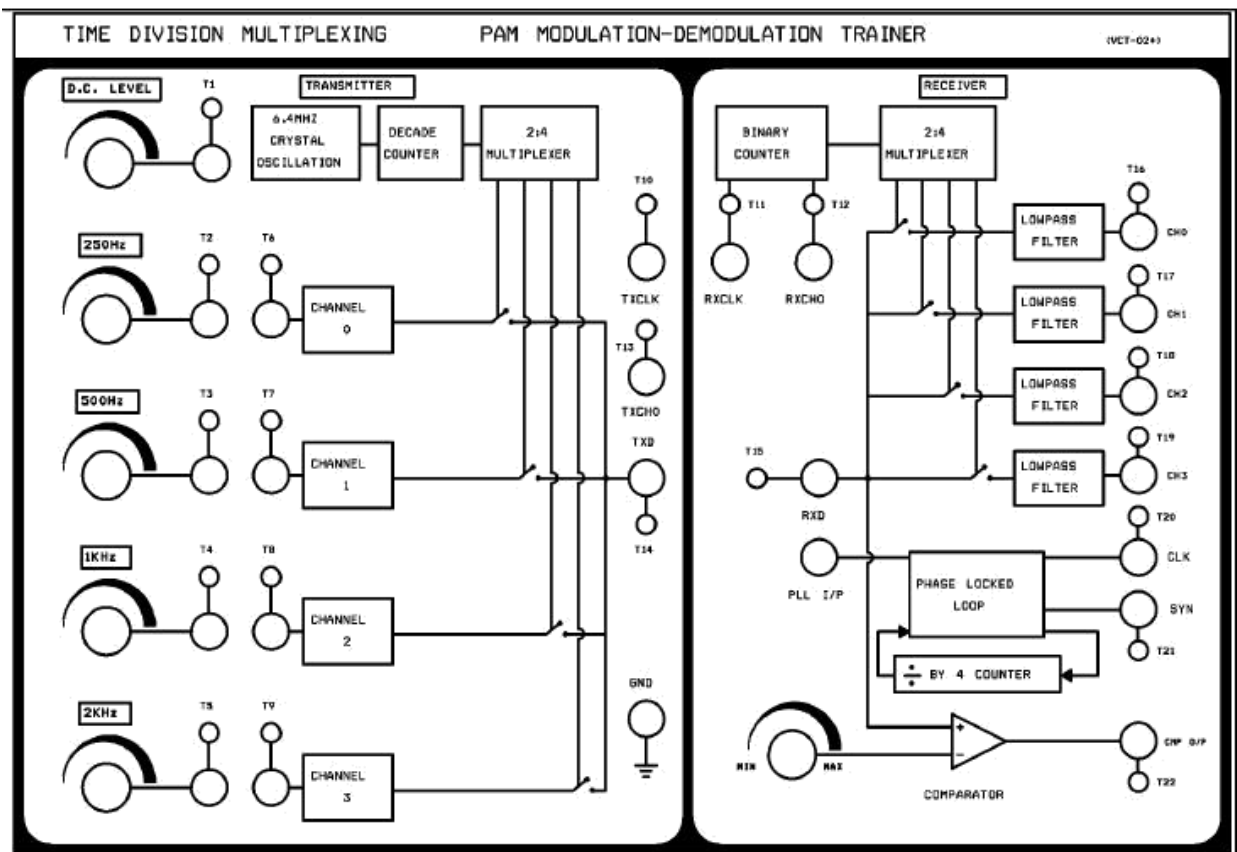
When the two conditions are met then the receiver end is said to be in synchronization with the transmitter end. If the 1st condition is not met then the samples different sources would get mixed out the receiver end and if the 2nd condition is not met then the information from source '1' will be received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

PULSE AMPLITUDE MODULATION

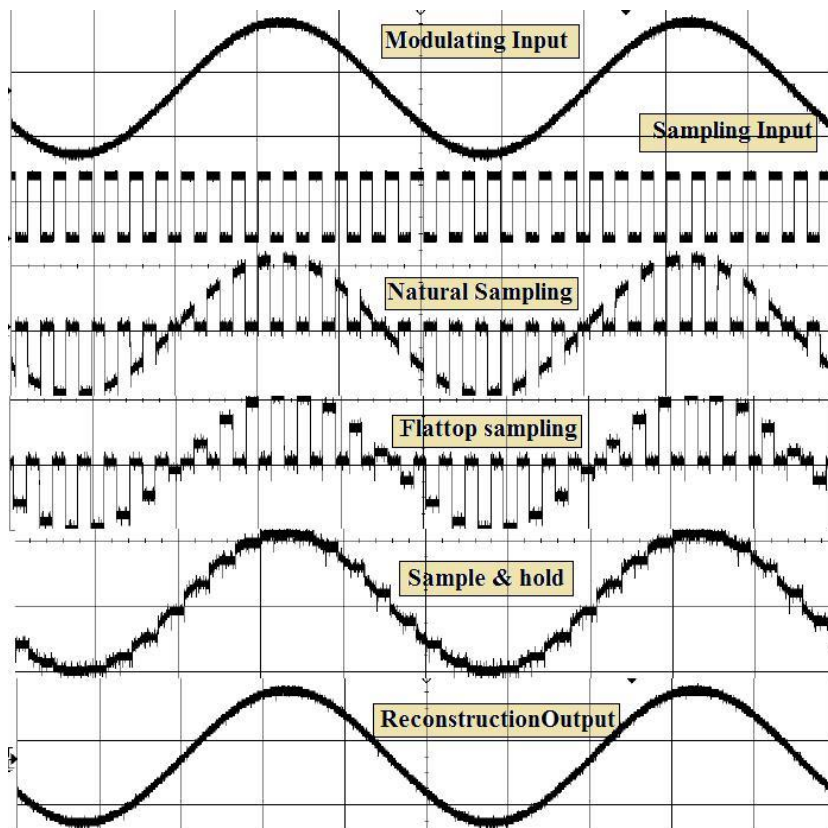
In pulse amplitude modulation, the amplitude of the high frequency pulse is changed in accordance with the amplitude of the message signal. It is very easy to generate and recover pulse amplitude modulated signal. In pulse amplitude modulation, the pulse amplitude is made proportional to the modulating 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. Natural sampling
2. Flat-top sampling
3. Sample and Hold

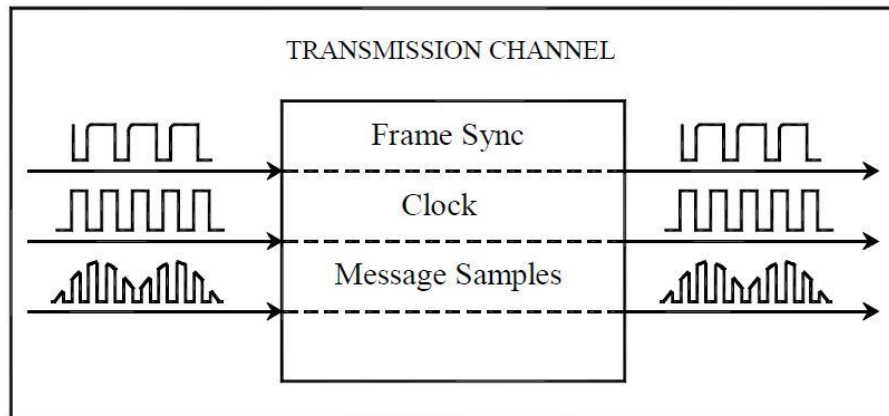


MODEL GRAPH



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.

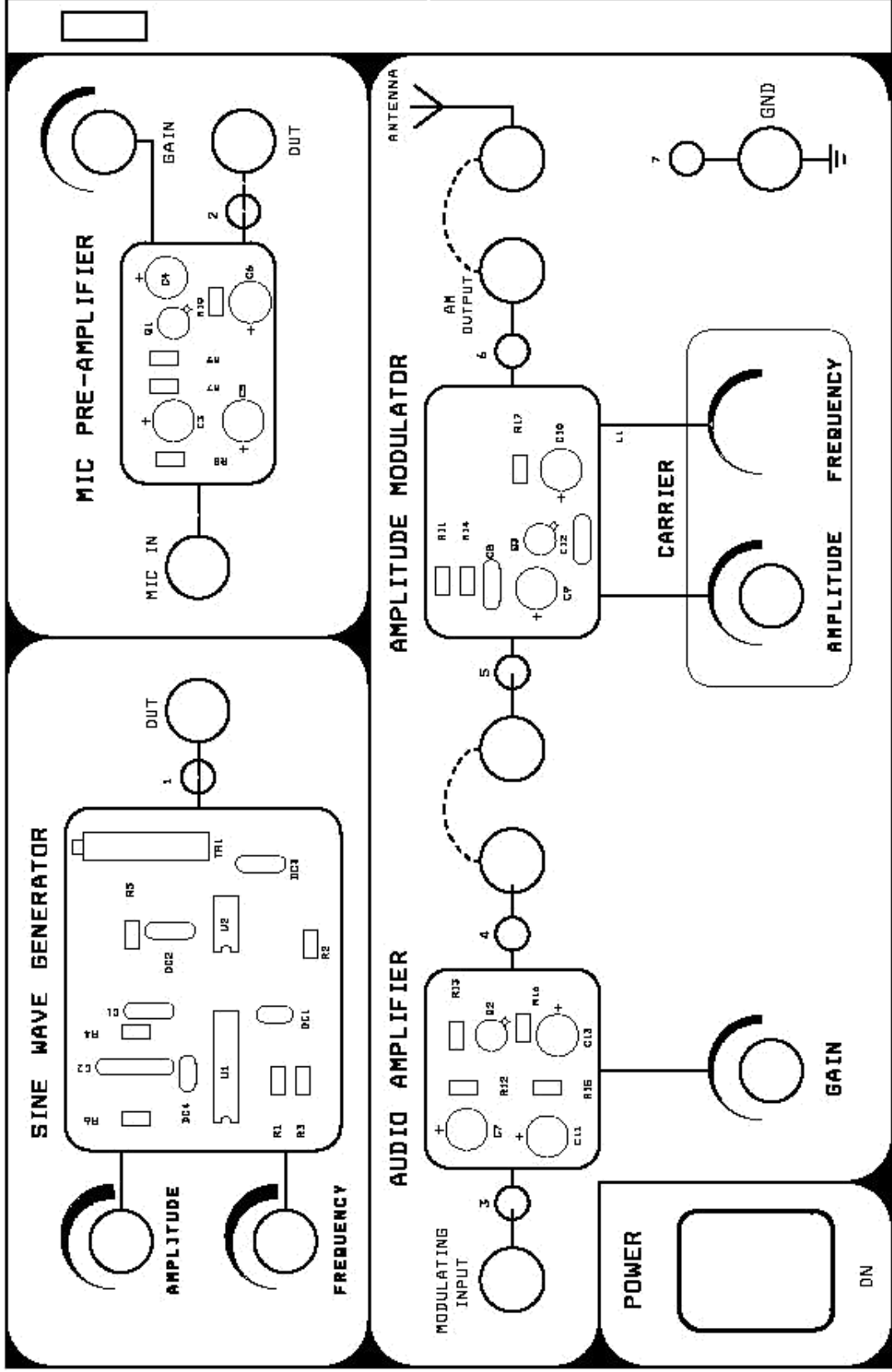
PROCEDURE

1. Switch ON the power supply to the board.
2. Make initial settings on VCT- 02 as follows.
 - a) Set all sine wave voltages to 2V,
 - b) Make the wiring connections as in wiring diagram which is provided at the end of this experiment.
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 down waveforms.
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 note down.
5. Similarly observe 1KHz and 2KHz waveforms at test point T4 and T5 respectively on oscilloscope and note down.
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 down waveforms.
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 down waveforms.
8. Similarly, observe input and output 500Hz, 1KHz and 2KHz sine waves on oscilloscope and note down.

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)



Vi Microsystems Pvt. Ltd., Chennai-96.

3. AMPLITUDE MODULATION AND DEMODULATION

AIM

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

APPARATUS REQUIRED

1. AM trainer Kit
2. CRO.
3. Patch chords.
4. BNC to P8003 cables.

THEORY

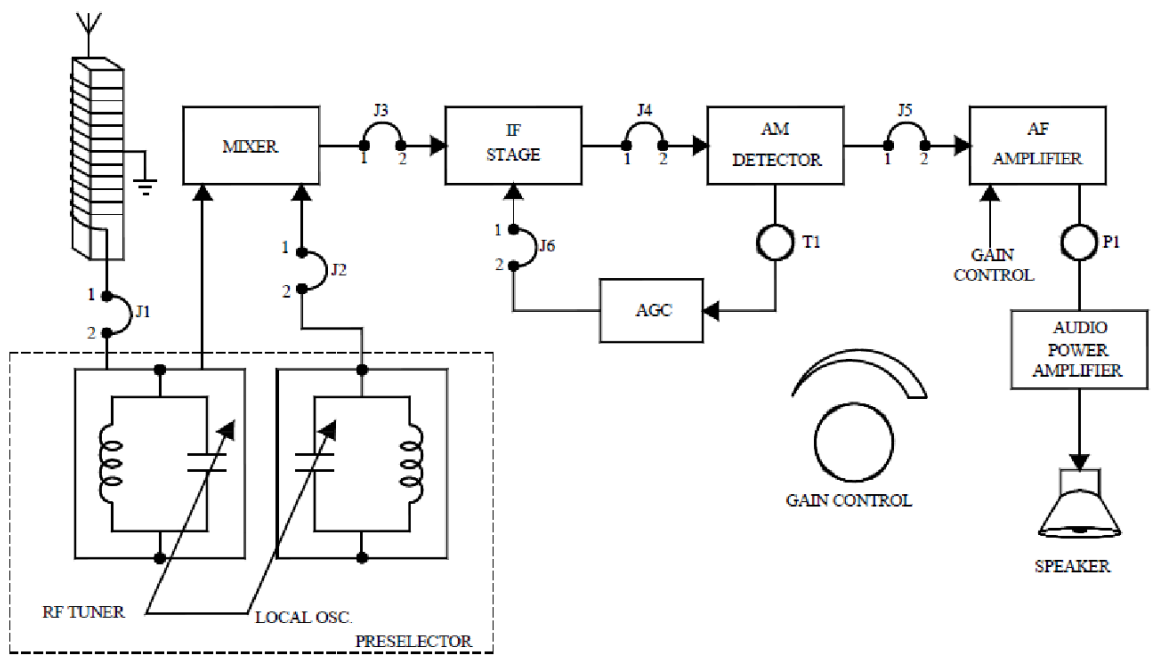
In radio transmission, it is necessary to send audio signal (eg. music, speech etc) from a broad casting station over great distances to a receiver. The audio signal cannot be sent directly over the air for appreciable distance. Even if audio signal is converted into electrical signal, the latter cannot 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

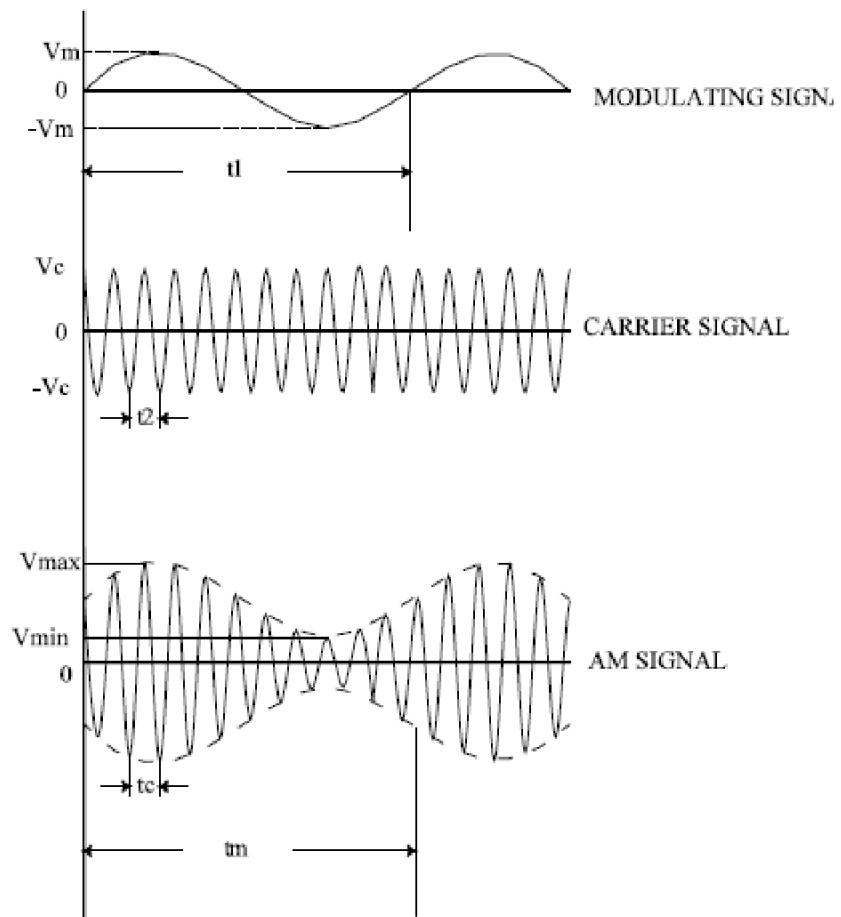
The main problem is that a high frequency carrier wave is used to carry the audio signal and that we need to know how the audio signal should be "added" to the carrier wave. The solution lies in changing some characteristic of carrier wave in accordance with the signal. Under such conditions, the audio 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 an demodulator 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.

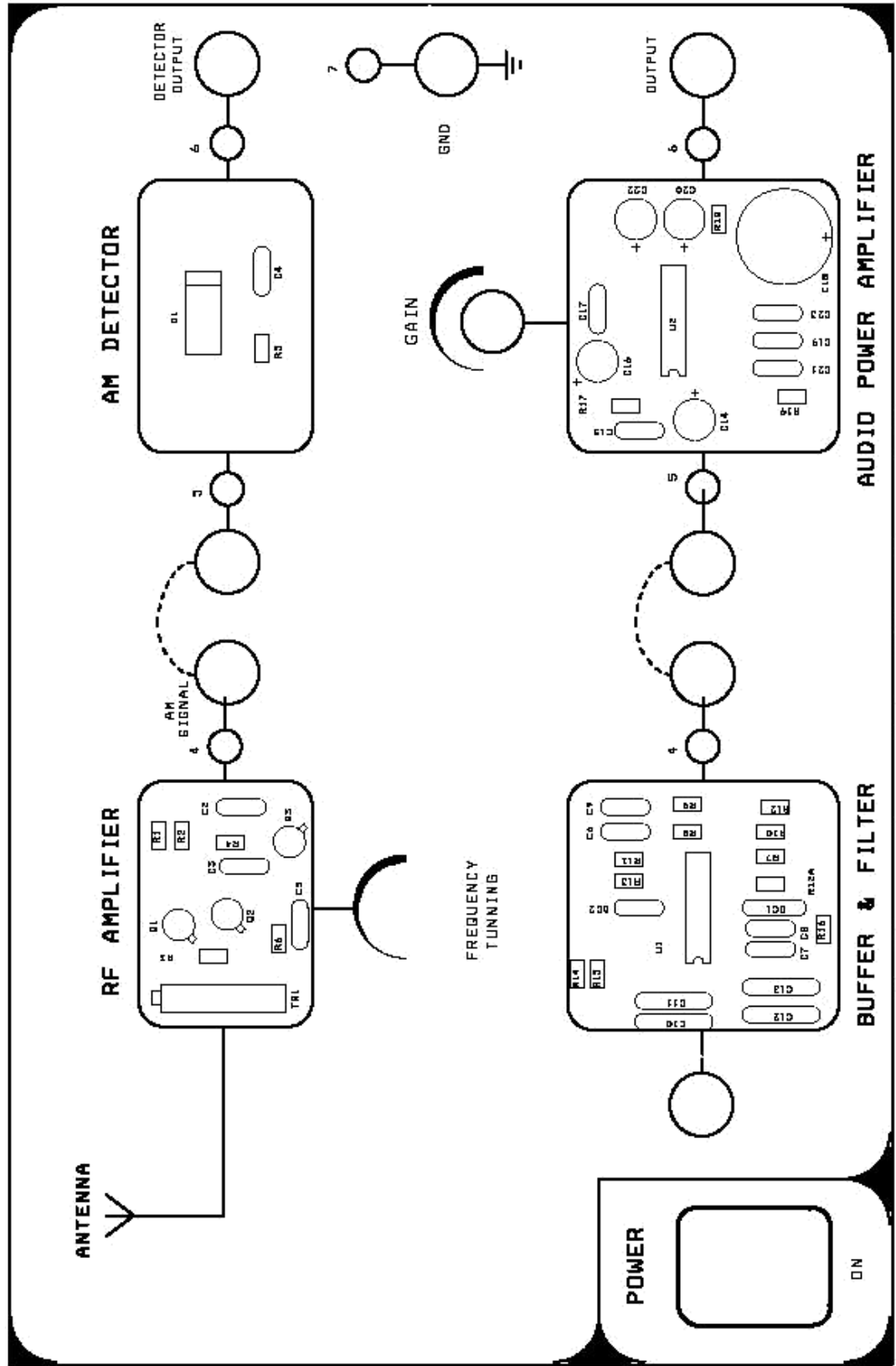
The ratio of change of amplitude of carrier wave to the amplitude of normal carrier wave is called the modulation factor (m) ie,

$$\begin{aligned}\text{Modulation factor (m)} &= \frac{\text{Amplitude change of carrier wave}}{\text{Normal carrier wave (unmodulated)}} \\ &= \frac{V_{\max} - V_{\min}}{V_{\max} + V_{\min}} \times 100\end{aligned}$$

PROCEDURE

1. The circuit wiring is done as shown in the wiring diagram.
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 frequency oscillator.
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 amplitude modulator.
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 required level.
7. The amplitude and time duration of the modulating signal are observed using a CRO and tabulated.
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 noted down.
9. Patch the modulated signal to the telescopic whip antenna in receiver.
10. The receiver circuit wiring is also done as shown in the wiring diagram.
11. The carrier frequency knob in the transmitter side is kept at a middle position.
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 noted down.
14. From the tabulated values the modulating signal, carrier signal, AM signal, & demodulated message signal are plotted neatly.
15. The depth of modulation is also calculated.
16. The same experiment can be repeated for various values of carrier frequency

AM RECEIVER TRAINER (VCT-09)



Vi Microsystems Pvt. Ltd., Chennai-96.

REV:0

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.



V.I. Microsystems Pvt. Ltd.,
Chennai-60, INDIA.

FM TRANSMITTER TRAINER

VCT-12

POWER

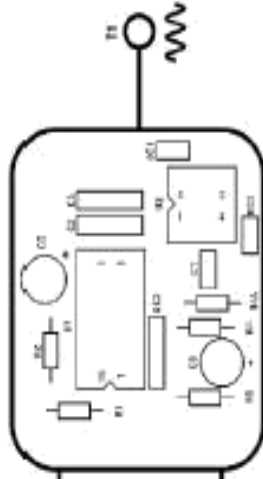
ON

AUDIO OSCILLATOR

FREQUENCY



AMPLITUDE

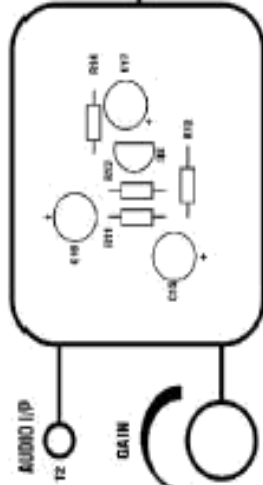


AUDIO AMPLIFIER

AUDIO I/P



GAIN



O/P

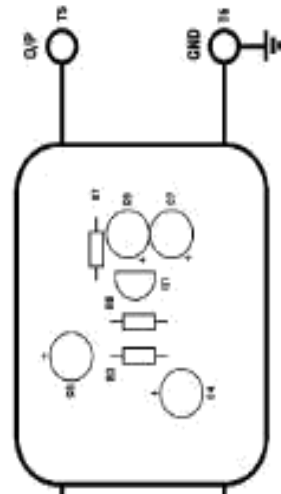


MIC PRE-AMPLIFIER

MIC IN



GAIN

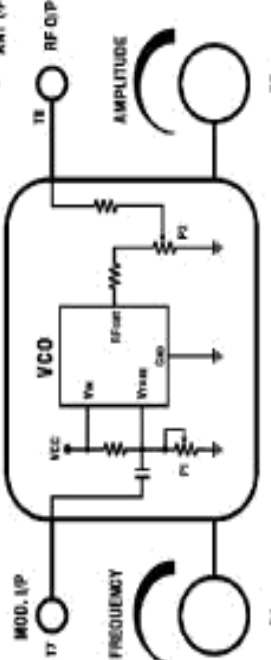


FREQUENCY MODULATOR

MOD. I/P



FREQUENCY



ANT I/P



RF O/P



AMPLITUDE



P2

P1

4.FREQUENCY MODULATION AND DEMODULATION

AIM

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

APPARATUS REQUIRED

1. FM Transmitter
2. FM Receiver
3. 1000 MHz Spectrum analyser and CRO
4. Mic & Speaker
5. Patch chords

PROCEDURE

1. Connect the test point T1 and T7 of VCT - 12 using 2mm patch chord.
2. Switch ON the trainer.
3. Tune the amplitude control POT of *Audio oscillator* section to fully anticlockwise direction (zero amplitude), now test point T7 is virtually connected to ground.
4. Connect the test point T8 to oscilloscope or spectrum analyser, set frequency control POT at *Frequency modulator* section to minimum position (fully anticlockwise direction).
5. Turn Amplitude control POT P2 to fully clockwise Direction (Maximum amplitude). Note down carrier frequency of VCO which is observed using oscilloscope (or) spectrum analyser.
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 in spectrum.
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 to 1KHz.
8. Display the frequency modulated waveform at test point T8 on spectrum analyser. Reduce the spectrum analyser band and observe the spectrum of FM signal.
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 anticlockwise direction)



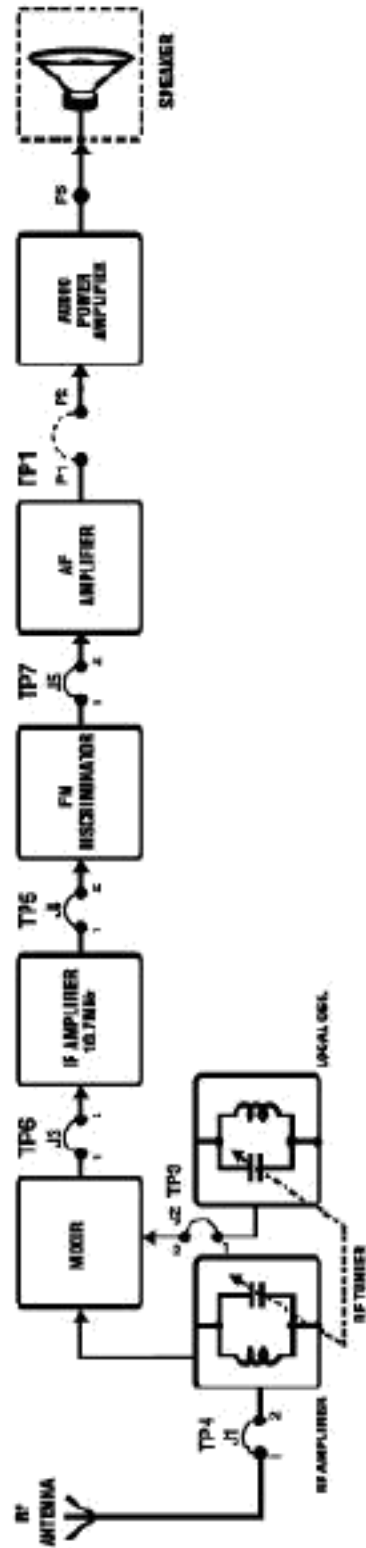
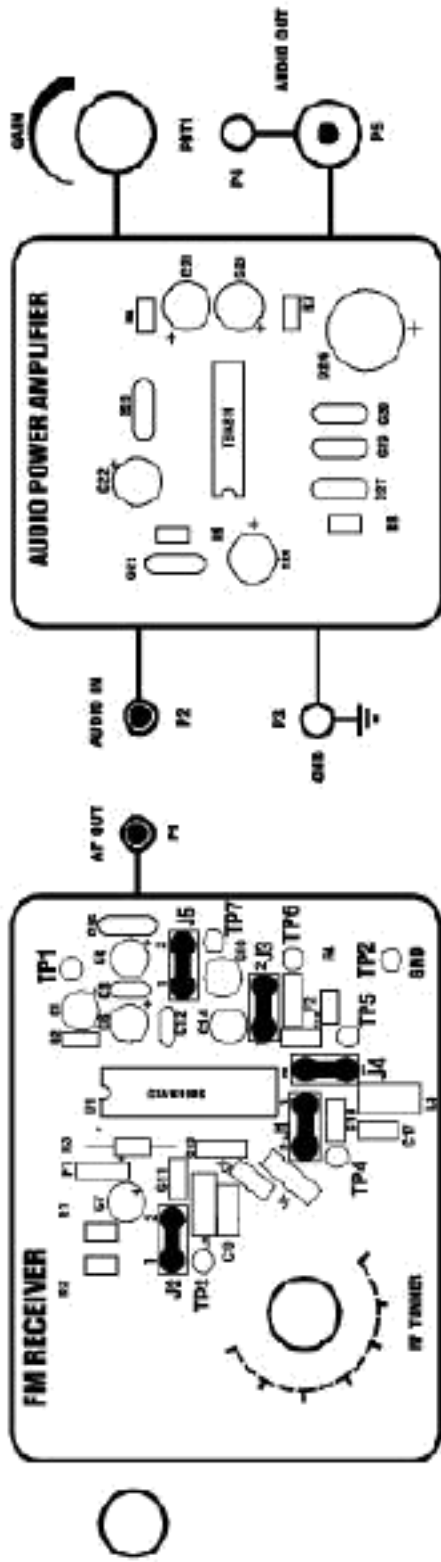
V.I. Microsystems Pvt. Ltd.
Chennai-66, INDIA.

FM RECEIVER TRAINER

VCT-13

POWER

OM



- b. Set Gain control POT1 to minimum position (fully anticlockwise direction)
 - c. Insert Jumper J1, J2, J3, J4, and J5 at respective place.
 - 11. Connect the test point P1 to channel 1 of oscilloscope and switch ON the trainer .
 - 12. Turn RF tuner 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 Frequency by,
- 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 at VCT-13.

FREQUENCY MODULATION

There are two basic methods for generating frequency modulated signal namely, i. Direct FM
ii. Indirect FM.

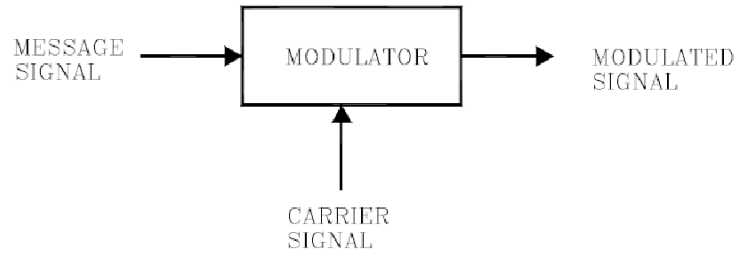
In Direct method the carrier frequency is directly varied in accordance with input baseband signal which is readily accomplished using a voltage control oscillator (VCO). In Indirect method the modulating signal is first used to produce a narrowband FM signal and frequency multiplication is next used 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 AM signal.

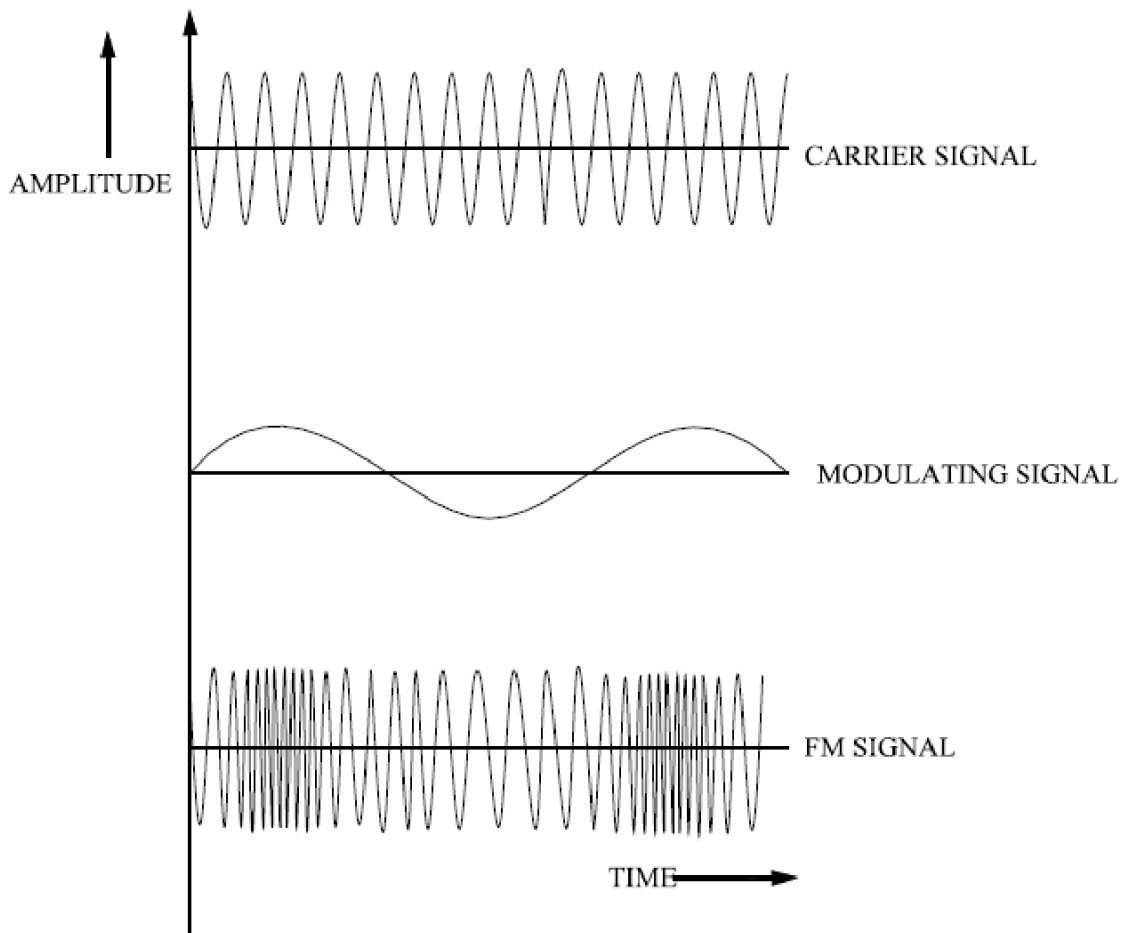
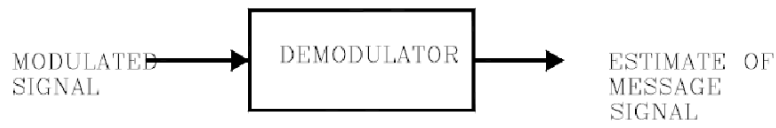
DEMODULATION

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



RECEIVER



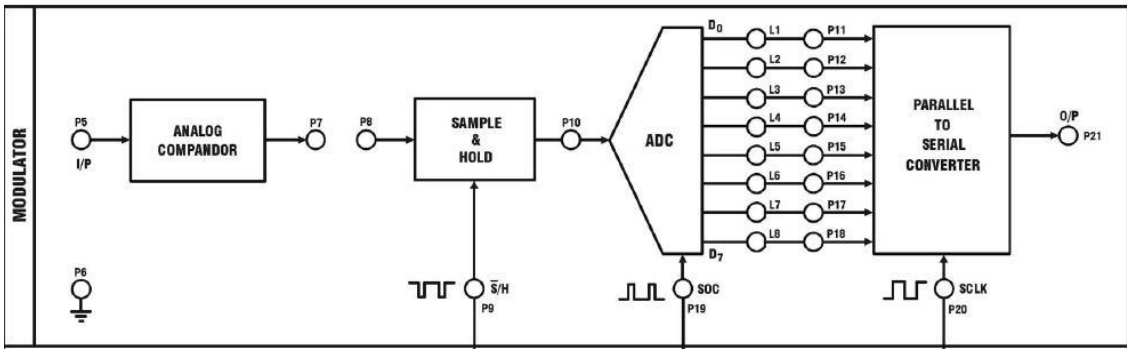
TABULATION:

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

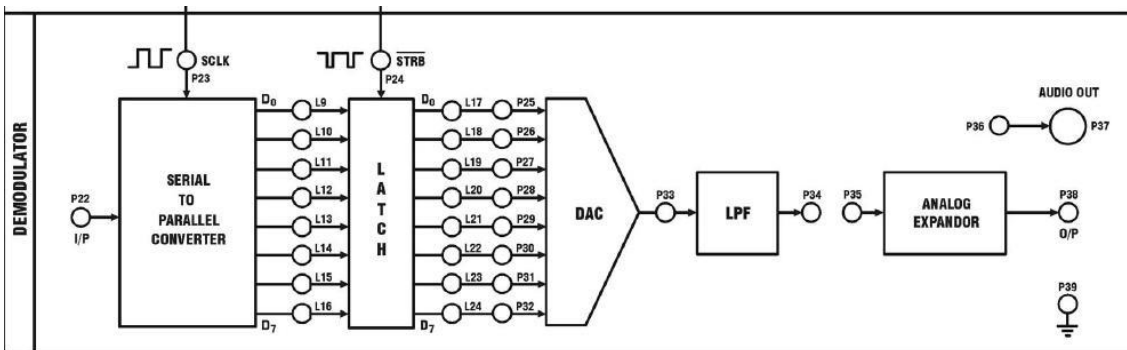
RESULT

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

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 concept of 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 is 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 fed to the modulator section from an external function generator which will be useful for studying frequency response of the system.

PROCEDURE

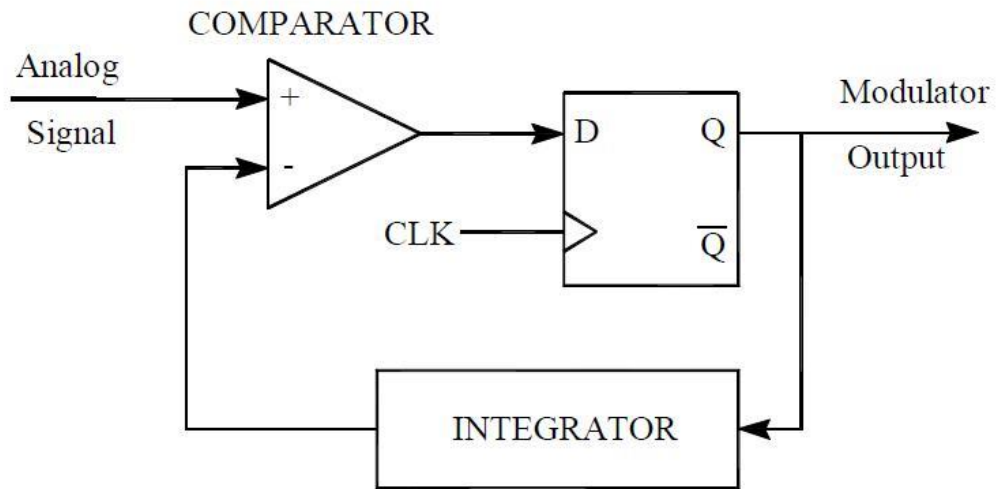
a) Study of pulse code modulation and demodulation

1. Make wiring connection on VCT - 07 as shown in figure 3.2 (or) simply connect 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 minimum position.
3. Keep 8KHz of sampling rate.
4. Display the modulating signal at test point P1 using a probe on channel 1 of oscilloscope. Increase sinewave amplitude by rotating POT1 in clockwise direction and set sinewave amplitude to 3V and note down.
5. Displays the sample / Hold output waveform on channel 2 of oscilloscope and note down the waveform, amplitude level of the signal.
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 output waveform.
7. Plot all the noted waveforms such as modulating signal, S/H output and modulator output on a linear graph sheet.
8. Replace the channel 2 waveform by digital to analog converter (test point P33) waveform which is the recovered sampled analog signal, note down the waveform.
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 graph sheet.

RESULT

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

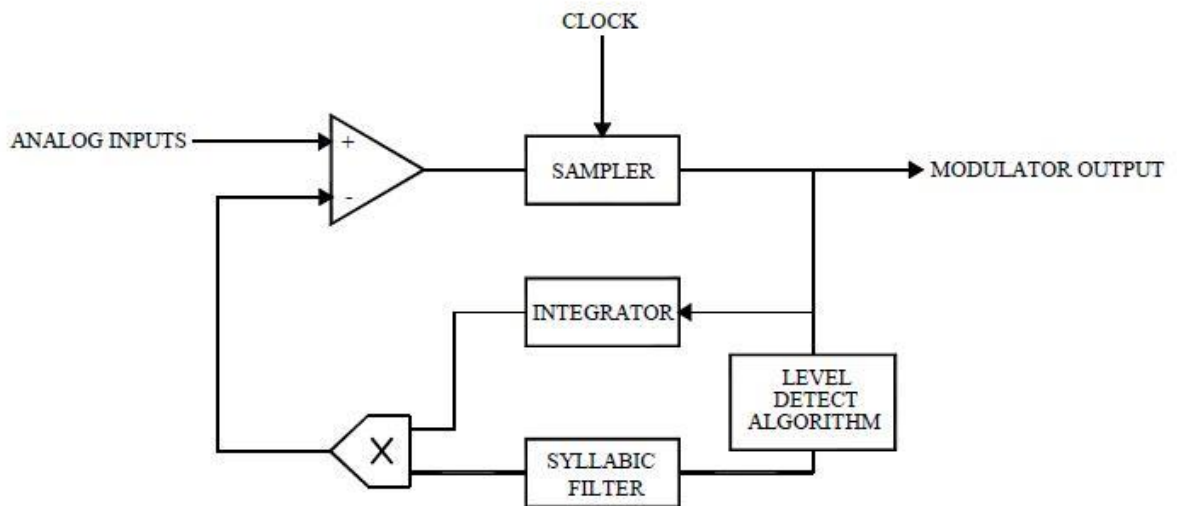
DELTA MODULATOR



DELTA DEMODULATOR



ADAPTIVE DELTA MODULATOR



6.DELTA / ADAPTIVE DELTA MODULATION AND DEMODULATION

AIM

To study the delta / Adaptive delta modulation and demodulation

EQUIPMENTS REQUIRED

- DM / ADM kit
- Two Channel 20MHz Oscilloscope
- Patch Chords, Oscilloscope probe

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 smoothed out by the 4th 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 which affect the distortion in delta modulation are the sampling rate f and step size Δ . Ideally the step size Δ should be small as possible and sampling rate f as large as possible. Practical and economic considerations limits the minimum step size of Δ and the maximum sampling rate of f . These limitations give rise to two types of distortions, slope overload noise and granular noise.

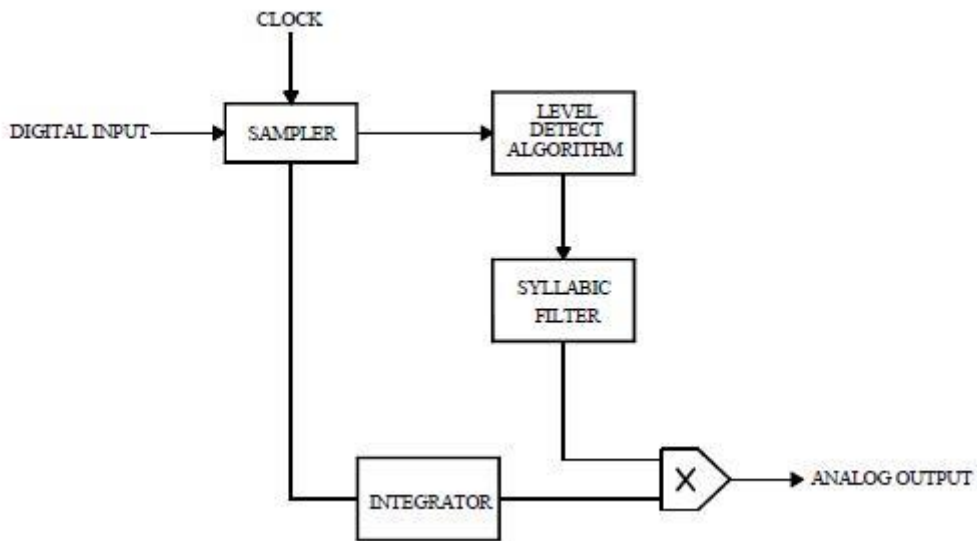
Idle channel noise

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 pass filter).

Slope overload and Granular noise

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

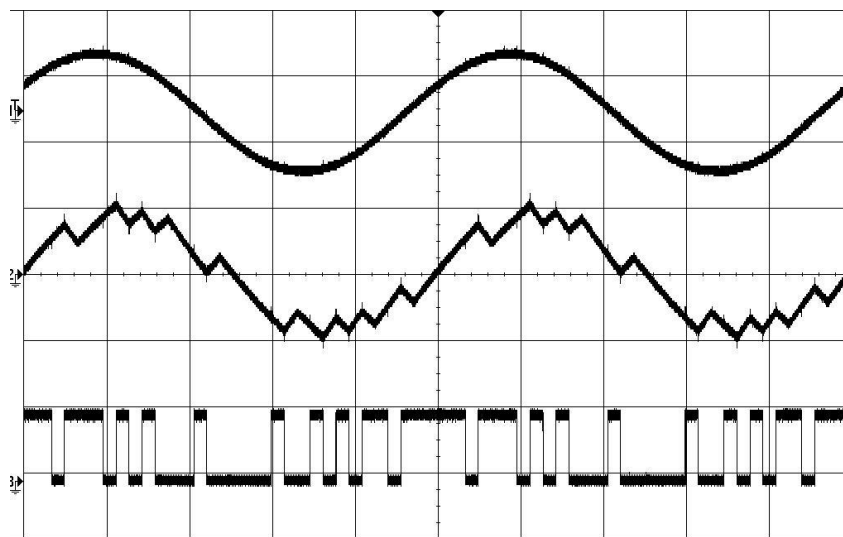
ADAPTIVE DELTA DEMODULATOR

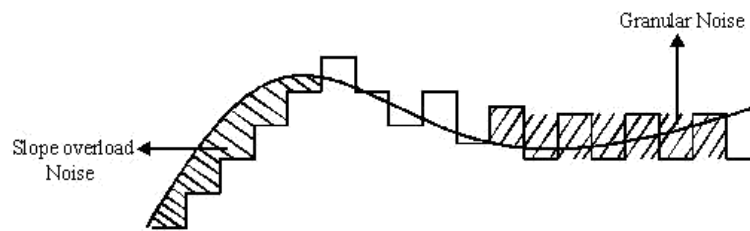


TABULATION

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





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

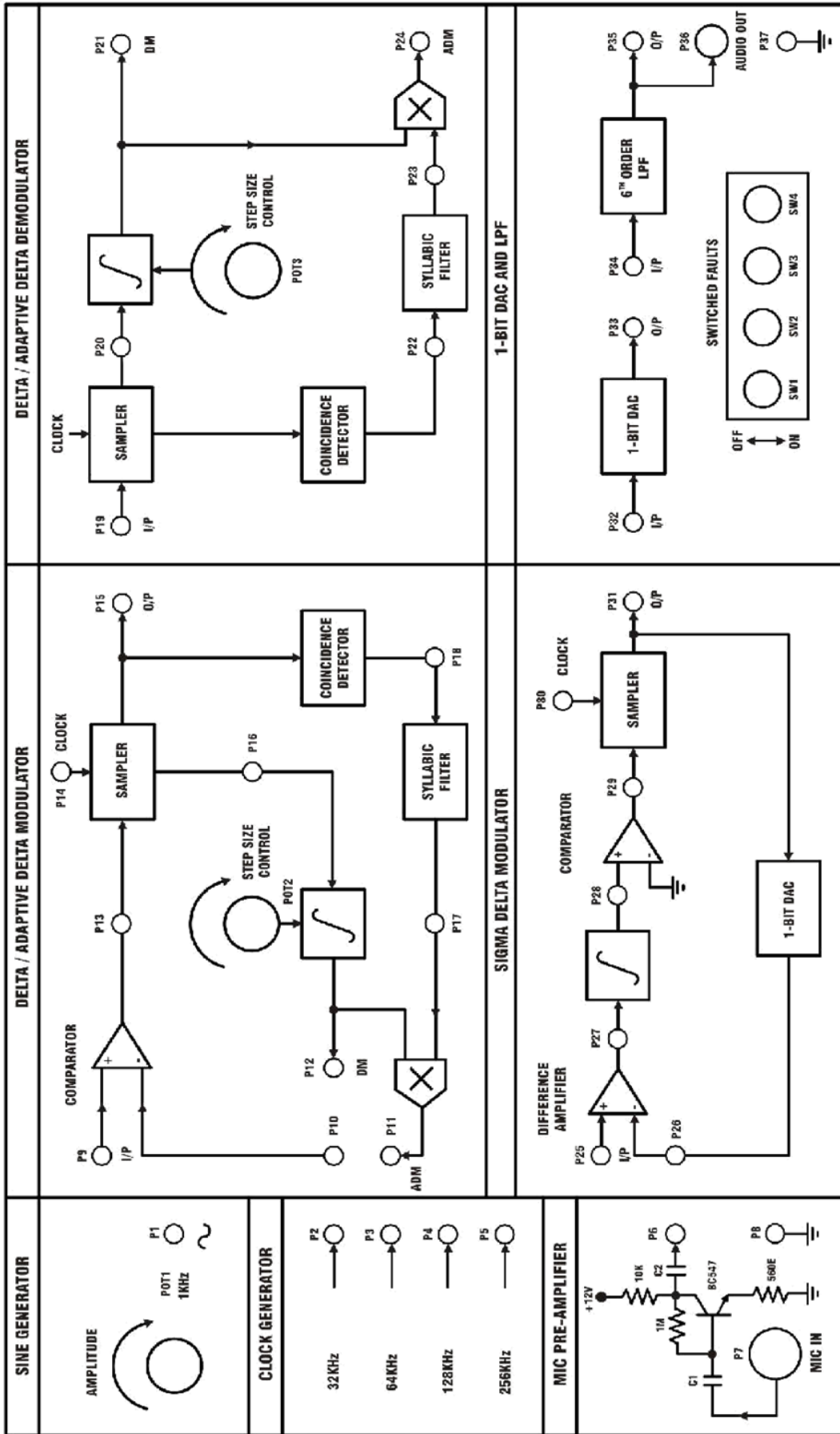
1. Plug-in AC power cord into 230V, 5A Mains power supply.
2. Ensure that the following initial conditions exist on VCT-50:
 - a. Keep all switches in OFF position.
 - b. Keep all potentiometer controls in minimum position.
3. Wiring connections do as follows in VCT-50:
 - a. Connect point P1 (1KHz sine wave) to input of DM section(P9).
 - b. Connect point P2 (Clock signal) 32KHz to clock input of DM section(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 this experiment)
4. Switch ON the power supply to the board
5. Connect the test point P1 with oscilloscope, turn POT1 in clockwise direction and set the amplitude of sine wave to $2V_{pp}$.
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 output waveform.
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 integrator waveform.
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 graph sheet.
9. Observe the final demodulated waveform (P35) on oscilloscope and plot in graph sheet.
10. Do the above procedure for other clock rates of 64KHz and 128KHz, explore the changes in demodulated signal.

RESULT

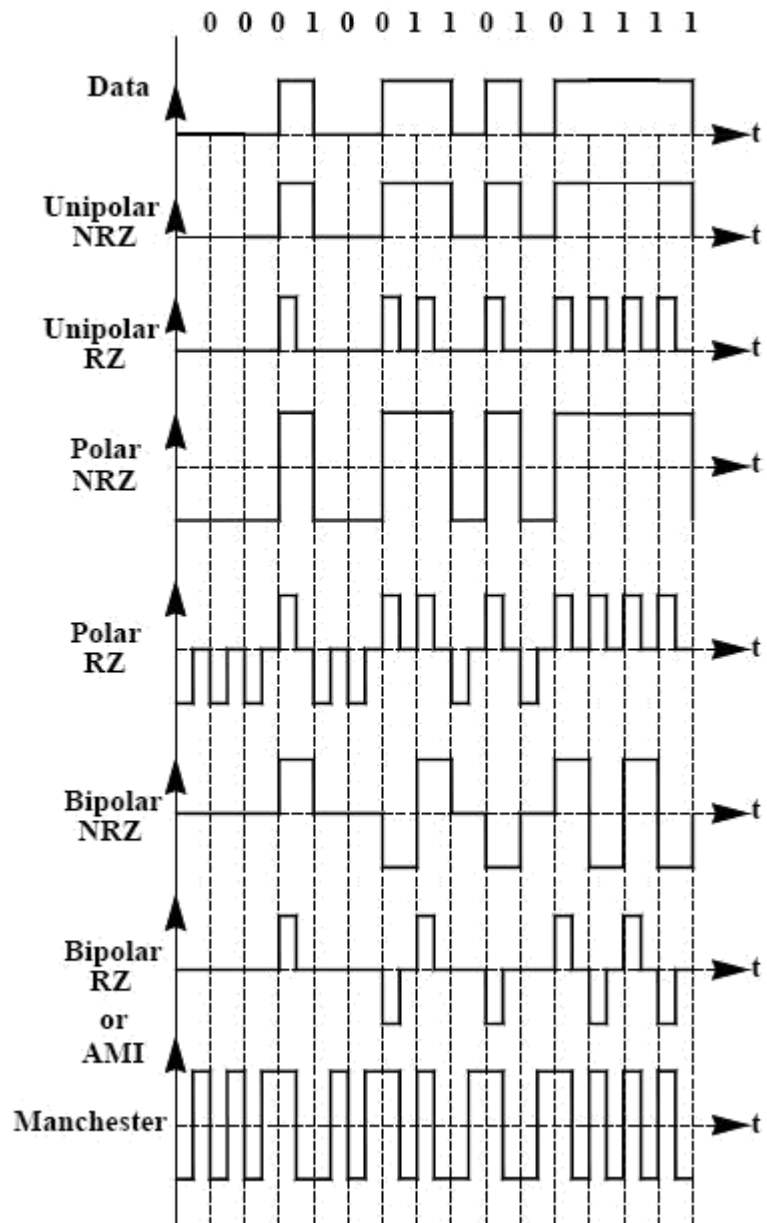
Thus the study of delta / Adaptive delta modulation and demodulation verified sucessfully

DELTA / ADAPTIVE DELTA / SIGMA DELTA MODULATION & DEMODULATION

VCT-50



Vi Microsystems Pvt. Ltd., Chennai-96.



7.LINE CODING AND DECODING TECHNIQUE

AIM

To perform various type of line coding & Decoding technique

CHARACTERISTICS OF LINE CODES

1. Transmission Bandwidth
2. Timing
3. DC Content
4. Power Spectrum
5. Power Efficiency
6. Probability of error
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 lead to DC wander. A long string of „0's will block pulse transition and could lead 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

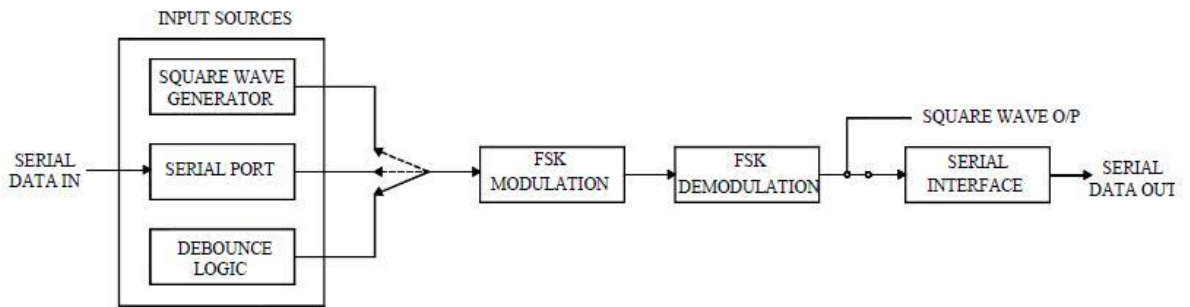
In this scheme, a binary 1 is represented by a pulse that has positive voltage during the first-half of the bit duration and negative voltage during second-half of the bit duration. A binary „0' is represented by a pulse that has negative voltage during first-half of the bit duration and positive voltage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

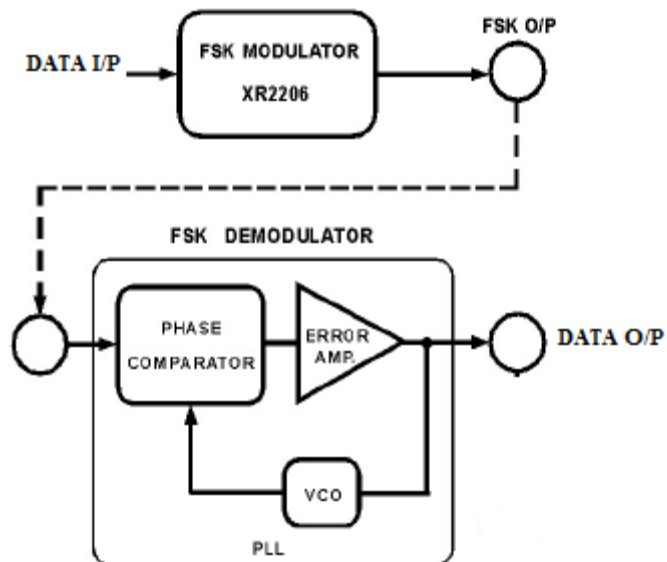
RESULT:

Thus the study Line coding schemes verified sucessfully

FSK MODULATION AND DEMODULATION



Block Diagram Of FSK Modulation And Demodulation



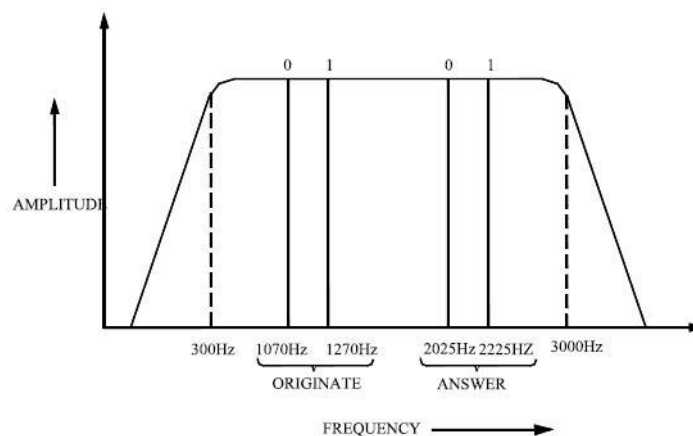
8. ASK, FSK, BPSK GENERATION SCHEME SIMULATION

AIM

To study the FSK,PSK ,DPSK Modulation using MATLAB code & observe the output waveform.

THEORY

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 frequency separation.



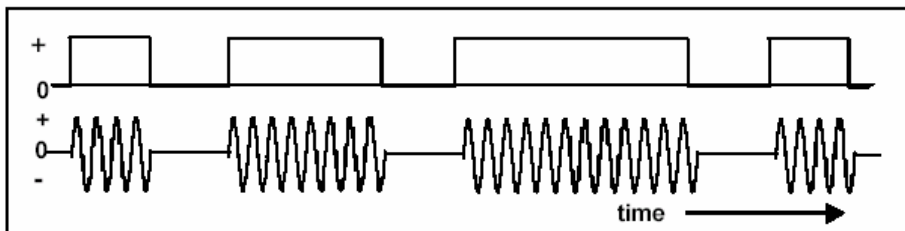
WORKING OF FSK

In FSK, the carrier frequency is shifted in steps or levels corresponding to the levels of the digital modulating signal. In the case of a binary signal, two carrier frequencies are used, one corresponding to binary '0' (i.e space) and the other to a binary 1 (i.e mark). An example of a digital data stream converted to FSK by modulation & demodulation is shown in figure 2.

PROGRAM FOR ASK :

```
%Modulation ASK
g = [1 0 1 0 1];
f = 2;
t=0:2*pi/99:2*pi;
cp=[];sp=[];
mod=[];mod1=[];bit=[];
for n=1:length(g);
    if g(n)==0;
        die=zeros(1,100);
        se=zeros(1,100);
    else g(n)==1;
        die=2*ones(1,100);
        se=ones(1,100);
    end
    c=sin(f*t);
    cp=[cp die];
    mod=[mod c];
    bit=[bit se];
end
ask=cp.*mod;
subplot(2,1,1);plot(bit,'LineWidth',1.5);grid on;
title('Binary Signal');
axis([0 100*length(g) -2.5 2.5]);
subplot(2,1,2);plot(ask,'LineWidth',1.5);grid on;
title('ASK modulation');
axis([0 100*length(g) -2.5 2.5]);
```

MODEL GRAPH:



TABULATION

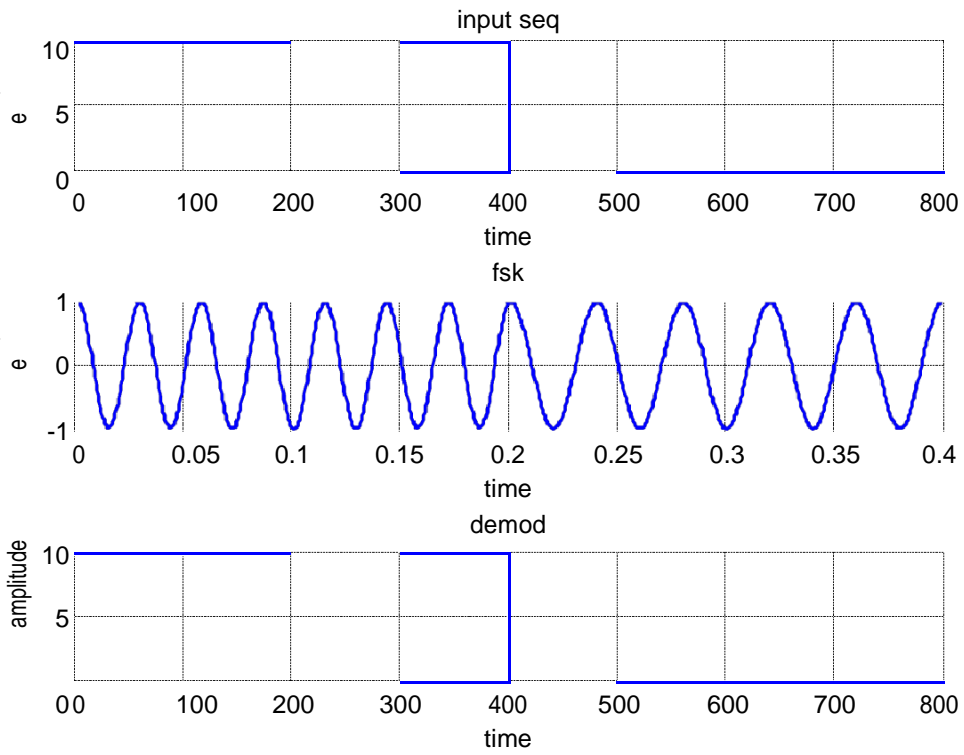
S.NO	SIGNAL	AMPLITUDE(V)	TIME(ms)	FREQUENCY(HZ)
1	Modulating Input			
2	Carrier signal			
3	FSK Output			
4	Demodulated output			

The baud rate is the number of changes of the transmitted data. This can be determined by taking the reciprocal of the time of the shortest pulse transmitted. A FSK Modulator & Demodulator sends and receives serial data at a rate of 300 bits or 300 baud. For a 300 baud data stream, maximum frequency occurs when the data stream has 0's and 1's alternatively and the frequency of this will be 150 Hz. As mentioned earlier the telephone network has a bandwidth between 300Hz & 3000Hz. So the maximum frequency of 300 baud data stream falls out of the bandwidth range of the telephone lines. This prevents sending digital data in its pure form over the phone lines. FSK Modulator & Demodulator is one method used to overcome the bandwidth limitation of the telephone network for digital data transmission.

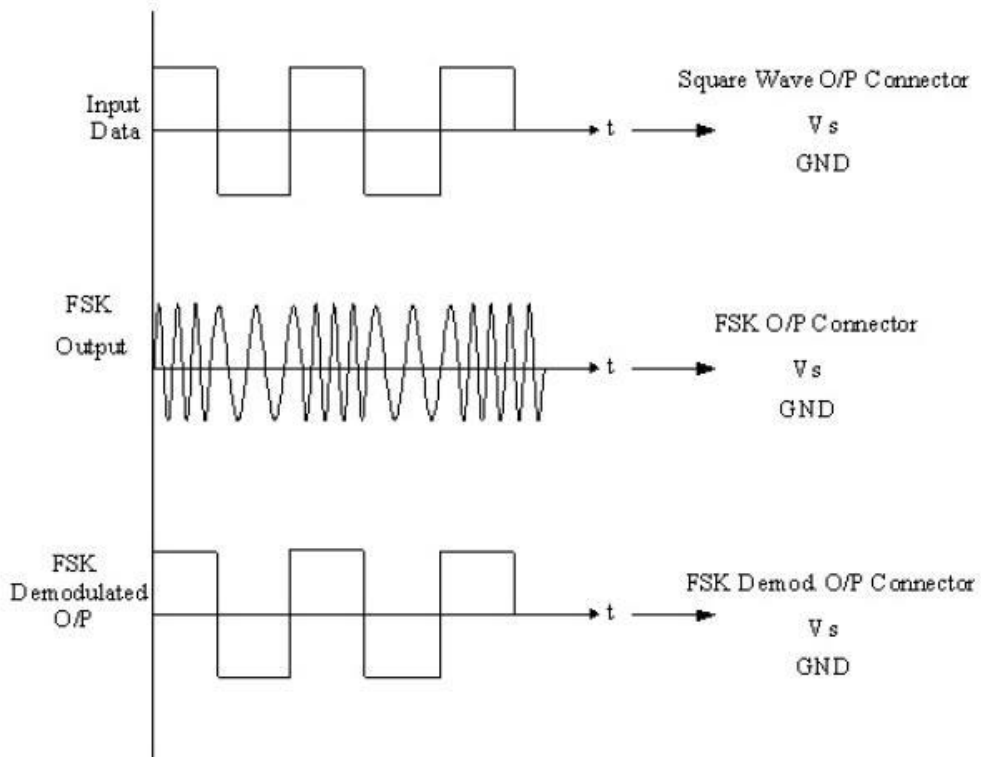
As mentioned earlier in FSK, the standard frequency for a space is either 1070Hz or 2025 Hz depending on the FSK Modulator & Demodulator mode and that of a mark is either 1270 Hz or 2225 Hz. All these frequencies come under the permissible frequency range of the telephone lines. Thus the bandwidth limitation of the telephone line is overcome by the use of FSK.

MATLAB PROGRAM FSK

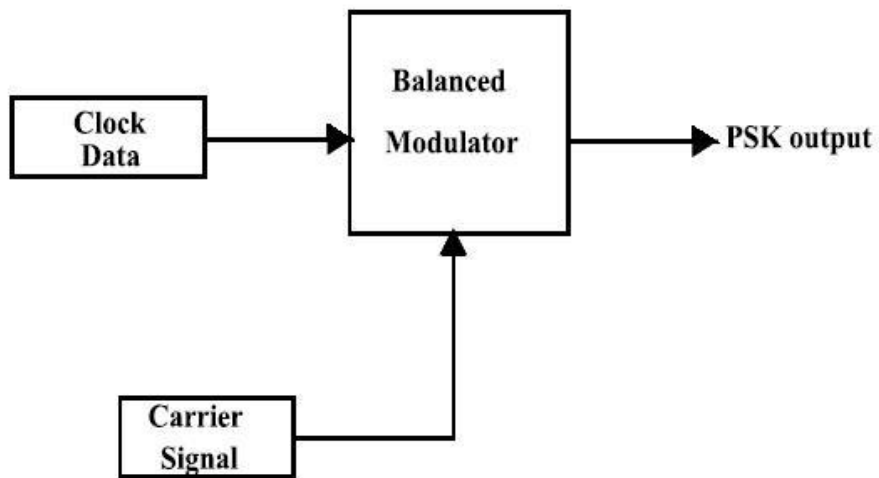
```
clear all;
close all;
X=input('enter the seq');
fs=input('enter the sampling freq');
fc=input('enter the carrier freq');
fd=input('enter the fd');
m=2;
y=MODMAP(X,fd,fs,'fsk',m);
subplot(3,1,1);
plot(y,'linewidth',1.5);
grid on;
xlabel('time');
ylabel('amplitude');
title('input seq');
[y,t]=dmod(X,fc,fd,fs,'fsk',m);
subplot(3,1,2);
plot(t,y,'linewidth',1.5);
grid on; xlabel('time');
ylabel('amplitude');
title('fsk');
z=ddemod(y,fc,fd,fs,'fsk',m);
z1=MODMAP(z,fd,fs,'fsk',m);
subplot(3,1,3);
plot(z1,'linewidth',1.5);
grid on;
xlabel('time');
ylabel('amplitude');
title('demod');
```



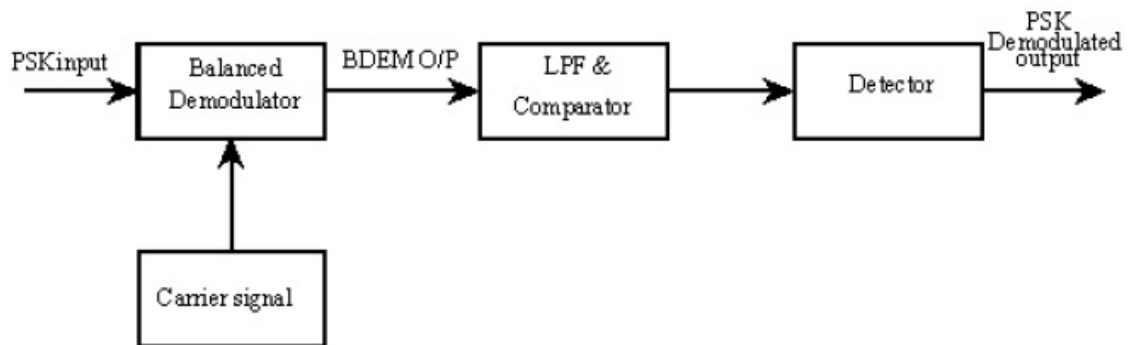
MODEL GRAPH:



BLOCK DIAGRAM OF PSK MODULATOR



BLOCK DIAGRAM OF PSK DEMODULATOR



PSK MODULATOR AND DEMODULATOR

THEORY

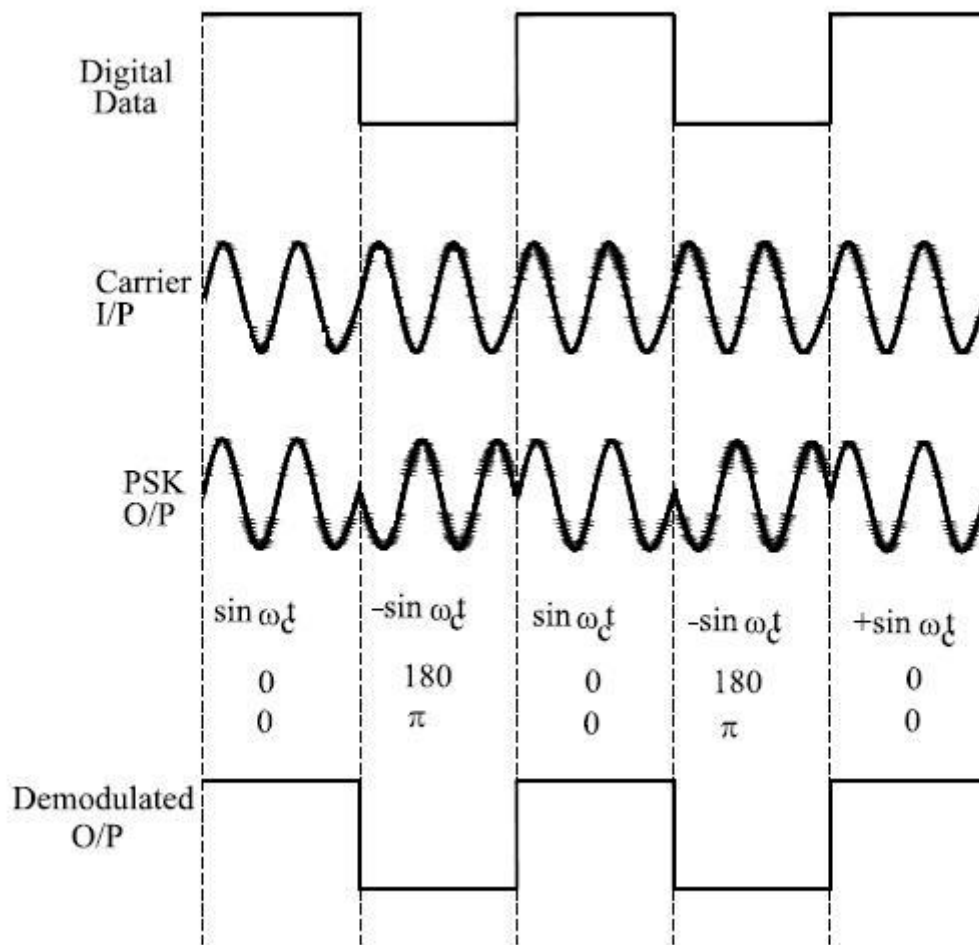
Communication is a process of conveying information from one place to other. Some of the examples for communication systems are face to face in meeting (or) conferences, often requiring travel, are increasing using „teleconferring“. Similarly, teleshopping and telebanking will provide services by electronic communications and newspapers may be replaced by electronic news services. The source originates a message such as human voice, a television picture, a teletype message or data. If the data is non electrical (human voice, teletype message, television picture) it must be converted by an input transducer into an electric waveform referred to as the baseband signal (or) message signal.

Modulation may be defined as a process by which any characteristics of a wave is varied as a function of the instantaneous value of another wave. In essence then the transmission takes place at high frequency (the carrier) which has been modified to “carry” the lower frequency information. The low-frequency information is often called “intelligence signal” (or) message signal. It follows that once this information is received the intelligence must be removed from the high frequency carrier a process known as demodulation (i.e) the process of removing intelligence signal from the high frequency carrier is called demodulation.

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.

A clock source (i.e.) Data Generator, Generate clock signals' (i.e.) 0 's and 1's and 8 bit data generator produce 8 bit data signals. i.e the 0's and 1's. In PSK modulation scheme, the input digital signal is used to switch the carrier phase - different phase depending on the source symbol. A balanced modulator is a product modulator, the output signal is the product of the two input signals. In binary communication the carrier phase is switched between two levels 0 and 1. The phase of the carrier signal is shifted with respect to modulating input is called PSK. Generally mixing of modulating input and carrier produces modulated output. We make use of MC 1496 modulator IC for PSK modulator. Offset adjustment is provided externally to produce correct PSK waveform.

MODEL GRAPH

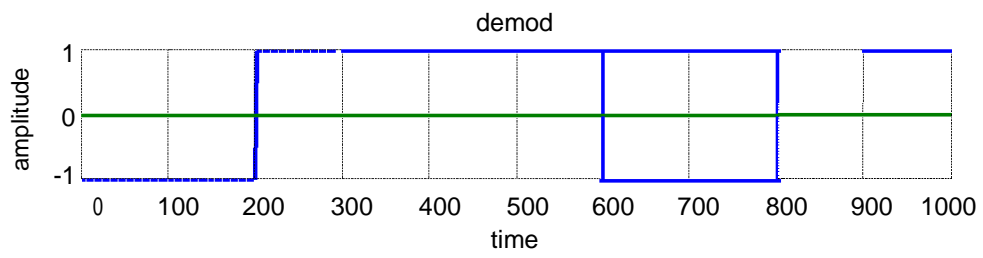
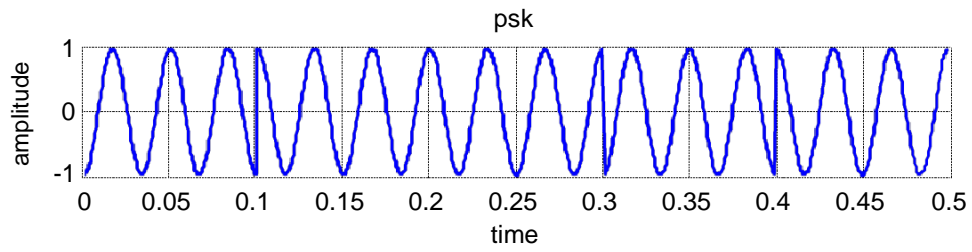
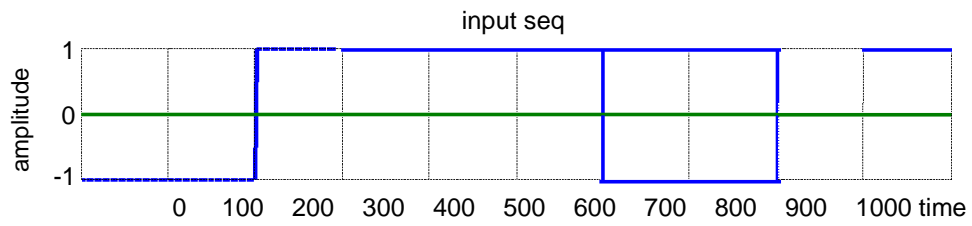


TABULATION

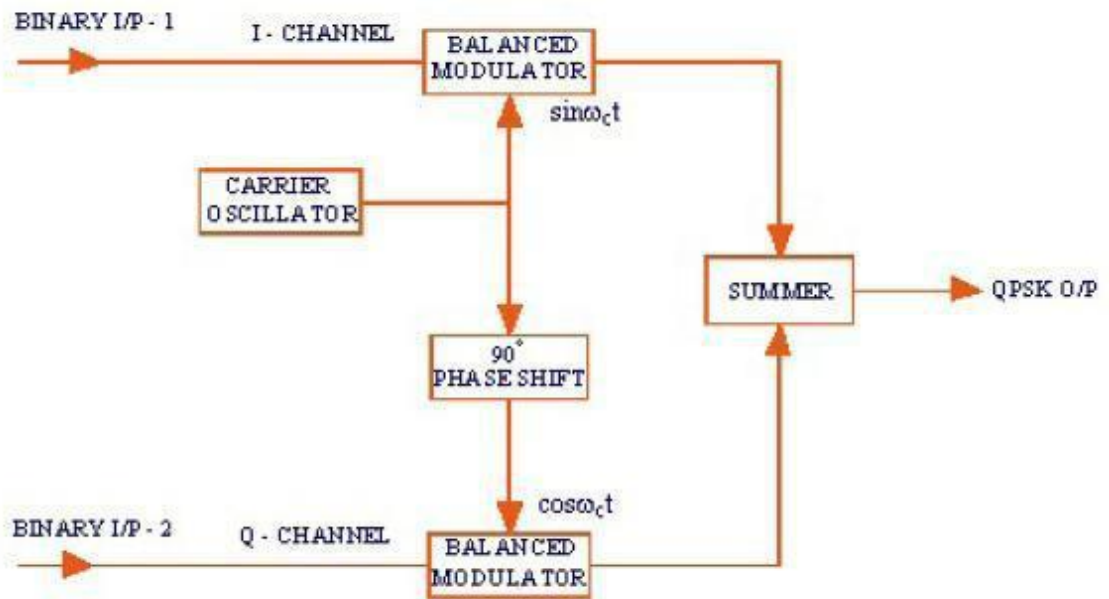
S.NO	SIGNAL	AMPLITUDE(V)	TIME(ms)	FREQUENCY(HZ)
1	Modulating Input			
2	Carrier signal			
3	PSK Output			
4	Demodulated output			

MATLAB PROGRAM PSK

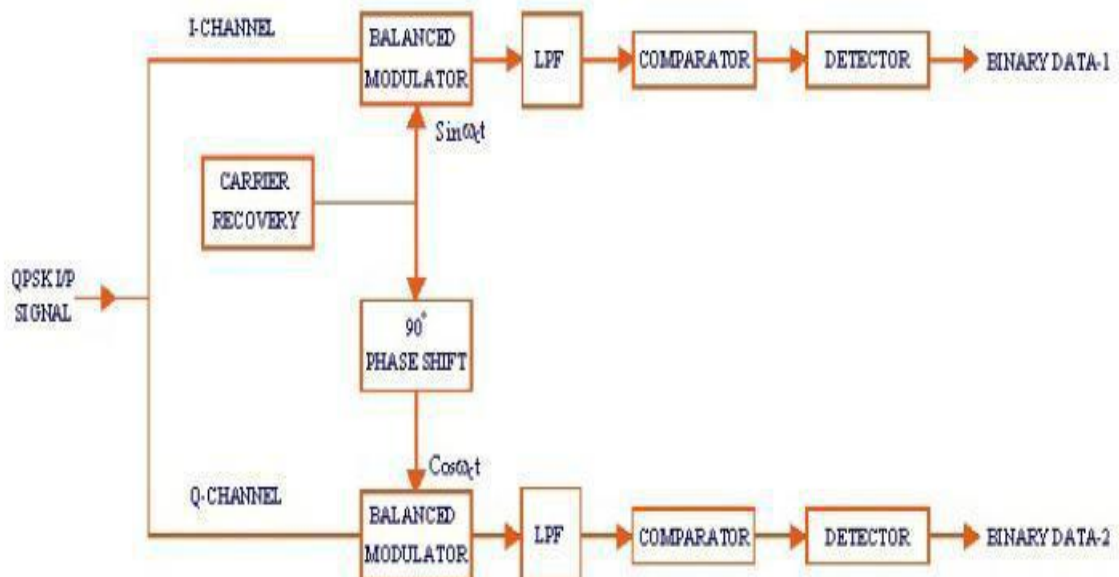
```
clear all;
close all;
X=input('enter the seq');
fs=input('enter the sampling freq');
fc=input('enter the carrier freq');
fd=input('enter the fd');
m=2;
y=MODMAP(X,fd,fs,'psk',m);
subplot(3,1,1);
plot(y,'linewidth',1.5);
grid on;
xlabel('time');
ylabel('amplitude');
title('input seq');
[y,t]=dmod(X,fc,fd,fs,'psk',m);
subplot(3,1,2);
plot(t,y,'linewidth',1.5);
grid on; xlabel('time');
ylabel('amplitude');
title('psk');
z=ddemod(y,fc,fd,fs,'psk',m);
z1=MODMAP(z,fd,fs,'psk',m);
subplot(3,1,3);
plot(z1,'linewidth',1.5);
grid on;
xlabel('time');
ylabel('amplitude');
title('demod');
```



BLOCK DIAGRAM OF QPSK MODULATION



BLOCK DIAGRAM OF QPSK DEMODULATION



9.SIMULATION OF DPSK, QPSK, QAM GENERATION SCHEMES

AIM:

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

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 bit rate.

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 I - 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).

TABULATION

S.NO	SIGNAL	AMPLITUDE(V)	TIME(ms)	FREQUENCY(HZ)
1	Modulating Input1			
2	Modulating Input2			
3	Carrier signal1			
4	Carrier signal2			
5	QPSK Output			
6	Demodulated output1			
7	Demodulated output2			

OUTPUT

MATLAB PROGRAM qpsk

```
%MATLAB Script for a Binary PSK with two Phases
% Clear all variables and close
all figures clear all;
close all;
% The number of bits to send - Frame Length
N=input('enter the number of bits to be modulated : N = ');
% Generate a random
bit stream bit_stream
= round(rand(1,N));
% 4 PHASE SHIFTS

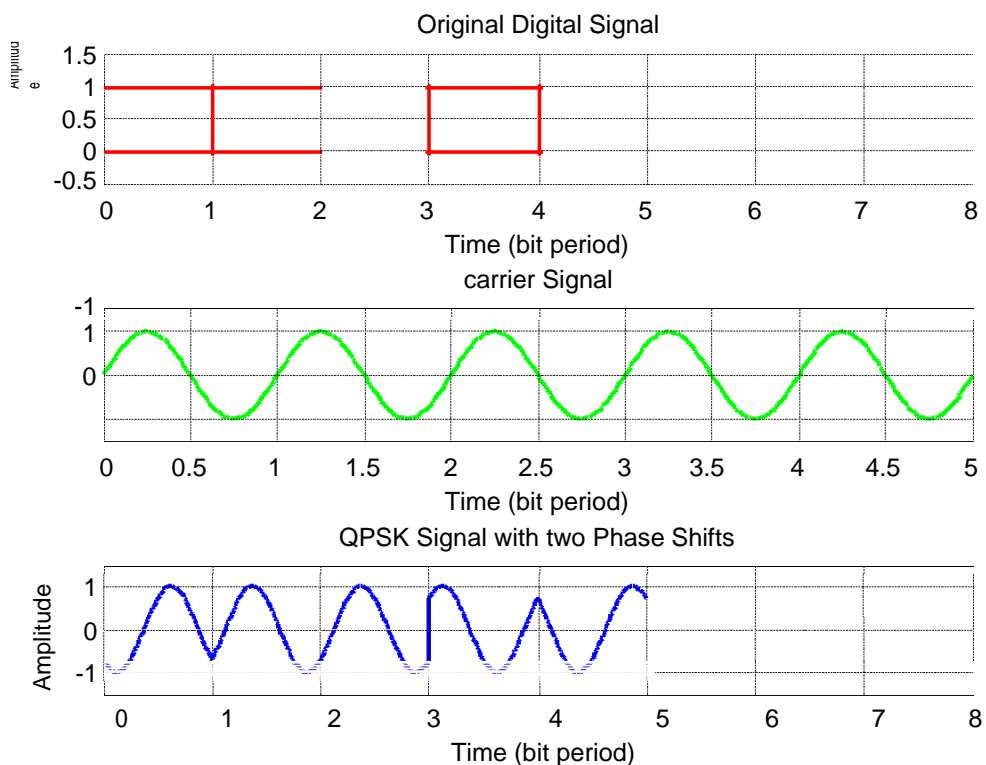
% Frequency          of
Modulating Signal f = 1;
%f --> time period
% Sampling rate of sine wave - This will define the
resolution fs = 100;
% Time for one bit
t = 0: 1/fs : 1;
% This time variable is just for
plot time = [];
QPSK_signal = [];
Digital_signal =
[];
carrier_signal=[];
for ii = 1: 2: length(bit_stream)
jj = ii + 1;
%Code for generation of Original Digital Signal Digital_signal =
[Digital_signal
(bit_stream(ii)==0)*zeros(1,length(t))+(bit_stream(jj)==1)*ones(1,length(t)
)
)];

%Code for generation of carrier signal
carrier_signal=[carrier_signal
(sin(2*pi*f*t))];
%Code for generating QPSK signal
modulated signal if bit_stream(ii)==0
if bit_stream(jj)==0
bit0 = (bit_stream(ii)==0)*sin(2*pi*f*t +
P1); QPSK_signal = [QPSK_signal (bit0)];
else
bit0 = (bit_stream(ii)==0)*sin(2*pi*f*t +
P2); bit1 = (bit_stream(jj)==0)*sin(2*pi*f*t
+ P2); QPSK_signal = [QPSK_signal (bit0+bit1)
];
end
end
if bit_stream(ii)==1
if bit_stream(jj)==0
bit1 = (bit_stream(ii)==0)*sin(2*pi*f*t +
P3); bit0 = (bit_stream(jj)==0)*sin(2*pi*f*t
+ P3); QPSK_signal = [QPSK_signal (bit1+bit0)
]; else
bit11 = (bit_stream(jj)==1)*sin(2*pi*f*t +
P4); QPSK_signal = [QPSK_signal (bit11) ];
end
```

```

end
time = [time t];
t = t + 1;
end
% Plot the Original Digital Signal
subplot(3,1,1);
plot(time,Digital_signal,'r','LineWidth',2) ;
xlabel('Time (bit period)');
ylabel('Amplitude');
title('Original Digital
Signal'); axis([0 8 -0.5 1.5]);
grid on;
% Plot the carrier Signal subplot(3,1,2);
plot(time,carrier_signal,'g','LineWidth',2) ;
xlabel('Time (bit period)');
ylabel('Amplitude');
title('carrier Signal');
axis([0 time(end) -1.5
1.5]); grid on;
% Plot the QPSK
Signal
subplot(3,1,3);
plot(time,
QPSK_signal,'LineWidth',2);
xlabel('Time (bit period)');
ylabel('Amplitude');
title('QPSK Signal with two Phase
Shifts'); axis([0 8 -1.5 1.5]);
grid on;

```

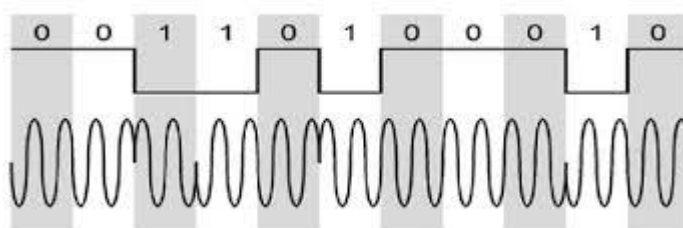


MATLAB PROGRAM QPSK:

Program

```
N = 10^4 % number of bits or symbols
rand('state',100); % initializing the rand() function
randn('state',200); % initializing the randn() function
ip = rand(1,N)>0.5; % generating 0,1 with equal probability
ipD = mod(filter(1,[1 -1],ip),2); % differential encoding y[n]=y[n-1]+x[n]
s = 2*ipD-1; % BPSK modulation 0 -> -1; 1 -> 0
n = 1/sqrt(2)*[randn(1,N) + j*randn(1,N)]; % white gaussian noise, 0dB variance
Eb_N0_dB = [-3:10]; % multiple Eb/N0 values
for ii = 1:length(Eb_N0_dB)
y = s + 10^(-Eb_N0_dB(ii)/20)*n; % additive white gaussian noise
ipDHat_coh = real(y) > 0; % coherent demodulation
ipHat_coh = mod(filter([1 -1],1,ipDHat_coh),2); % differential decoding
nErr_dbpsk_coh(ii) = size(find([ip - ipHat_coh]),2); % counting the number of errors 10
end
simBer_dbpsk_coh = nErr_dbpsk_coh/N;
theoryBer_dbpsk_coh = erfc(sqrt(10.^(Eb_N0_dB/10))).*(1 - .5*erfc(sqrt(10.^(Eb_N0_dB/10))));
close all
figure
semilogy(Eb_N0_dB,theoryBer_dbpsk_coh,'b.-');
hold on
semilogy(Eb_N0_dB,simBer_dbpsk_coh,'mx-');
axis([-2 10 10^-6 0.5])
grid on
legend('theory', 'simulation');
xlabel('Eb/No, dB')
ylabel('Bit Error Rate')
title('Bit error probability curve for coherent demodulation of DBPSK')
```

Modal Graphs



MATLAB PROGRAM FOR QAM

```
M = 16; % M determine the size of signal constellation
k = log2(M); % k determine the number of bits per symbol
n = 3e4; % n determine the number of bits to process
nSyms = n/k; % nSyms represent the number of symbols
hMod = modem.qammod(M); % We will create a 16-QAM modulator
hMod.InputType = 'Bit'; % Then accept bits as inputs
hMod.SymbolOrder = 'Gray'; % accept bits as inputs
hDemod = modem.qamdemod(hMod); %
x = randi([0 1],n,1); % Random binary data stream
tx = modulate(hMod,x);
EbNo = 0:10; % In dB
SNR = EbNo + 10*log10(k);
rx = zeros(nSyms,length(SNR));
bit_error_rate = zeros(length(SNR),1);
for i=1:length(SNR)
    rx(:,i) = awgn(tx,SNR(i),'measured');
end
rx_demod = demodulate(hDemod,rx);
for i=1:length(SNR)
    [~,bit_error_rate(i)] = biterr(x,rx_demod(:,i));
end
theoryBer = 3/(2*k)*erfc(sqrt(0.1*k*(10.^(EbNo/10))));
figure;
semilogy(EbNo,theoryBer,'-',EbNo, bit_error_rate, '^');
grid on;
legend('Theory BER', 'simulation');
xlabel('Eb/No, dB');
ylabel('Bit Error Rate');
title('Bit error probability curve for 16-QAM modulation');
```

RESULT:

EXPERIMENT NO 10 SIGNAL CONTELLATION FOR BPSK, QPSK & QAM

AIM:

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

SOFTWARE USED:

MATLAB

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.

PROGRAM: BPSK

```
M=2;
k=log2(M);
n=3*1e5;
nsamp=8;
X=randint(n,1);
xsym = bi2de(reshape(X,k,length(X)/k).','left-msb');
Y_psk= modulate(modem.pskmod(M),xsym);
psk = Y_psk;
EbNo=30;
SNR=EbNo+10*log10(k)-10*log10(nsamp);
Ynoisy_psk = awgn(Ytx_psk,SNR,'measured');
Yrx_psk = Ynoisy_psk;
h1=scatterplot(Yrx_psk(1:nsamp*5e3),nsamp,0,'r'); hold on;
scatterplot(Yrx_psk(1:5e3),1,0,'k',h1);
title('constellation diagram BPSK');
legend('Received signal', 'signal constellation');
axis([-5 5 -5 5]);
hold off;
```

Program for QPSK & QAM:

```
clc;
clear all;
close all;
M=16;
k=log2(M);
n=3*1e5;
nsamp=8;
X=randint(n,1);
```

```

xsym = bi2de(reshape(X,k,length(X)/k).','left-msb');
Y_qam= modulate(modem.qammod(M),xsym);
Y_qpsk= modulate(modem.pskmod(M),xsym);
Ytx_qam = Y_qam;
Ytx_qpsk = Y_qpsk;
EbNo=30;
SNR=EbNo+10*log10(k)-10*log10(nsamp);
Ynoisy_qam = awgn(Ytx_qam,SNR,'measured');
Ynoisy_qpsk = awgn(Ytx_qpsk,SNR,'measured');
Yrx_qam = Ynoisy_qam;
Yrx_qpsk = Ynoisy_qpsk;
h1=scatterplot(Yrx_qam(1:nsamp*5e3),nsamp,0,'r. ');
hold on;
scatterplot(Yrx_qam(1:5e3),1,0, 'k*',h1);
title('constellation diagram 16 QAM');
legend('Received signal' , 'signal constellation');
axis([-5 5 -5 5]);
hold off;
h2=scatterplot(Yrx
qpsk(1:nsamp*5e3),nsamp,0,'r. '); hold on;
scatterplot(Yrx_qpsk(1:5e3),1,0, 'k*',h2);
title('constellation diagram 16 PSK');
legend('Received signal' , 'signal constellation');
axis([-5 5 -5 5]);
hold off;

```

RESULT:

Thus the constellation diagrams of digital modulation system BPSK, QPSK & QAM are simulated & plotted in MATLAB

EXPERIMENT NO 11 SIMULATION FOR ASK,FSK,BPSK DETECTION

AIM:

To simulation of ASK, FSK, BPSK detection using MATLAB

SOFTWARE USED:

MATLAB

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.

PROGRAM: ASK

```
t1=0;
t2=Tb
for i=1:N
t=[t1:Tb/100:t2]
%correlator
x=sum(c.*ask_sig(i,:));
%decision device
if x>0 demod(i)=1;
else demod(i)=0;
end t1=t1+(Tb+.01);
t2=t2+(Tb+.01);
end
%plot demodulated binary data bits subplot(5,1,5);
stem(demod); title('ASK demodulated signal');
xlabel('n-->');
ylabel('b(n)');
grid on
```

Program for PSK:

```
%PSK Demodulation
t1=0;t2=Tb
for i=1:N t=[t1:.01:t2]
%correlator
x=sum(c.*bpsk_sig(i,:));
%decision device
if x>0 demod(i)=1;
else demod(i)=0;
t1=t1+1.01;
t2=t2+1.01;
```

```

end
%plot the demodulated data bits
subplot(5,1,5);
stem(demod);
title('demodulated data');
xlabel('n-->');
ylabel('b(n)');
grid on

```

Program for FSK DETECTION:

```

t1=0;
t2=Tb
for i=1:N
t=[t1:(Tb/100):t2]
%correlator
x1=sum(c1.*fsk_sig1(i,:));
x2=sum(c2.*fsk_sig2(i,:));
x=x1-x2; %decision device if x>0
demod(i)=1;
else
demod(i)=0;
end
t1=t1+(Tb+.01);
t2=t2+(Tb+.01);
end
%Plotting the demodulated data bits
subplot(3,2,6);
stem(demod);
title(' demodulated data');
xlabel('n---->');
ylabel('b(n)'); grid on;

```

RESULT:

Thus the Detection of ASK, FSK, PSK are simulated & plotted in MATLAB

EXPERIMENT NO 12(B). IMPLEMENTATION OF LINEAR BLOCK CODES

AIM:

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

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

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

SOFTWARE REQUIRED:

MATLAB 7.0 software

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.

$b_0, b_1, b_2, \dots, b_{n-k-1}$	$m_0, m_1, m_2, m_3, \dots, m_{k-1}$
	1

Structure of code word

Message Vector $m = [m_0, m_1, m_2, \dots, m_{k-1}]$

Parity check vector $b = [b_0, b_1, b_2, \dots, b_{n-k-1}]$

Code vector $X = [X_0, X_1, X_2, \dots, X_{n-1}]$

$b = m \times P$

Where,

$$P = \begin{pmatrix} P_{00} & P_{10} & \dots & P_{n-k-1,0} \\ P_{01} & \cdot & & \cdot \\ \cdot & \cdot & & \cdot \\ \cdot & \cdot & & \cdot \end{pmatrix}$$

Define the k by n generator matrix $G = \{P: I_k\}$

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

Parity check vector $b = m \times P$

Code vector $X =$

Message vector	check vector
----------------	--------------

ALGORITHM:

- From the given (n, k) block code assign the values of „ k “ nothing but Number of message bit.
- Assign the given generator matrix.
- Compute the check vector, then arrange code vector by combining Message and check vector.
- 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 code Vector.
- Find the minimum weight of the code, from that we can understand that

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

- Display the all possible code vector and weight of the code.

LINEAR BLOCK CODES:

CODING

```
clc;
```

```
clear;
```

```
k=4;
```

```
for i=1:2^k
```

```
for j=k:-1:1
```

```
    if rem((i-1),2^(-j+k+1))>=2^(-j+k)
```

```
        u(i,j)=1;
```

```
    else
```

```
        u(i,j)=0
```

```
    end
```

```
    echo off;
```

```
end
```

```
end
```

```
echo on;
```

```
G=[1 0 0 0 1 1 1;
```

```
    0 1 0 0 1 1 0;
```

```
    0 0 1 0 1 0 1;
```

```
    0 0 0 1 0 1 1]
```

```
c=rem(u*G,2);
```

```
disp(c);
```

```
w_min=min(sum((c(2:2^k,:))));
```

```
disp(w_min);
```

LINEAR BLOCK CODES:

OUTPUT:

```
G=[1 0 0 0 1 1 1;
```

```
0 1 0 0 1 1 0;
```

```
0 0 1 0 1 0 1;
```

```
0 0 0 1 0 1 1];
```

```
C=rem(U*G,2);
```

```
disp(C);
```

```
0 0 0 0 0 0 0
```

```
0 0 0 1 0 1 1
```

```
0 0 1 0 1 0 1
```

```
0 0 1 1 1 1 0
```

```
0 1 0 0 1 1 0
```

```
0 1 0 1 1 0 1
```

```
0 1 1 0 0 1 1
```

```
0 1 1 1 0 0 0
```

```
1 0 0 0 1 1 1
```

```
1 0 0 1 1 0 0
```

```
1 0 1 0 0 1 0
```

```
1 0 1 1 0 0 1
```

```
1 1 0 0 0 0 1
```

```
1 1 0 1 0 1 0
```

```
1 1 1 0 1 0 0
```

```
1 1 1 1 1 1 1
```

RESULT:

All possible code vector and weight of the given linear block code is found. From the values of $d_{min}=3$ the given linear block code is found that is hamming code.

EXPERIMENT NO 12.(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 MATLAB.

APPARATUS REQUIRED:

1. Personal computer.
2. MATLAB software.

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 message bits

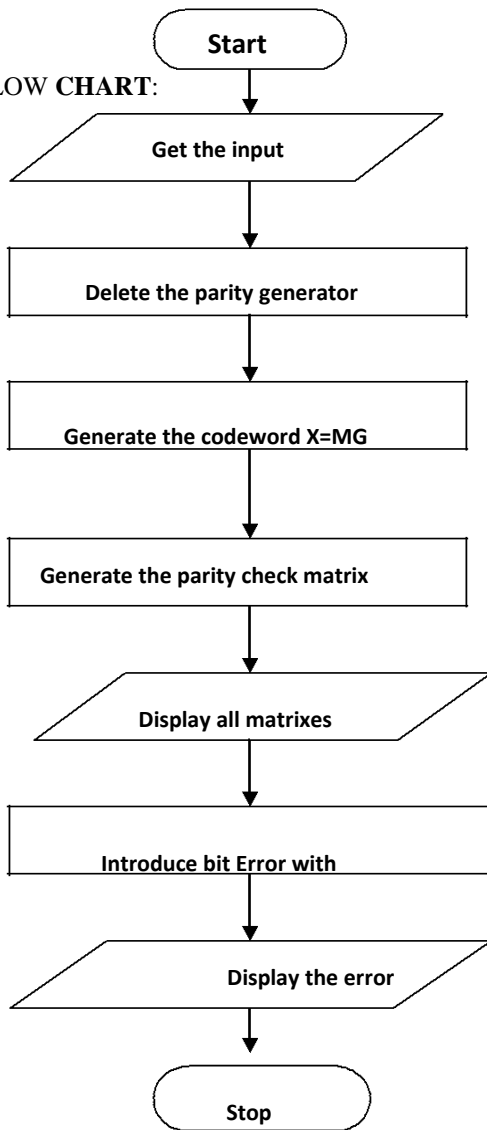
Decode the original message from the RX message

Display the Encoded & Decoded message

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

FLOW CHART:



CYCLIC CODE GENERATION USING MAT LAB

CODINGS:

%ENCODING

clc;

n=7; % CODE LENGTH

k=4; % NUMBER OF MESSAGE BITS

disp('MESSAGE'); % RANDOM MESSAGE GENERATION

m=randint(2,k,[0,1]);

disp(m);

disp('POLYNOMIAL'); % GENERATOR POLYNOMIAL

pol=cyclpoly(n,k,'min');

disp(pol);

disp('CODE VECTOR'); % CODE VECTOR GENERATION

code=encode(m,n,k,'CYCLIC/FMT',pol);

disp(code);

disp('ERROR'); % RANDOM ERROR GENERATION

e=randerr(2,n,[1 0;0.8 0.2]);

disp(e);

disp('RECEIVED MATRIX'); % RECEIVED MATRIX

r=rem(plus(code,e),2);

disp(r);

[newmsg err cc]=decode(r,n,k,'CYCLIC'); % DECODING OF RECEIVED MESSAGE

disp('DECODED RECEIVED VECTOR');

disp(cc);

disp('DECODED MESSAGE');

disp(newmsg);

CYCLIC CODE GENERATION USING MAT LAB

OUTPUT:

MESSAGE

0 1 1 0
0 0 0 1

POLYNOMIAL

1 0 1 1

CODE VECTOR

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

ERROR

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

RECEIVED MATRIX

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

DECODED RECEIVED VECTOR

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

DECODED MESSAGE

0 1 1 0
0 0 0 1

MATLAB PROGRAM CYCLIC CODES

```
clc;
clear all;
close all;
n=7;
k=4;
g=[1 0 1 1];
d= input ('enter the data seq');
c=encode(d,n,k,'cyclic',g);
c1(1)=c(4);c1(2)=c(5);c1(3)=c(6);c1(4)=c(7);c1(5)=c(1);
c1(6)=c(2);c1(7)=c(3);
disp('code word');
disp(c);
pol=cyclpoly(7,4);
[parmat,genmat,k]=cyclgen(7,pol);
trt=syndtable(parmat);
recd=input('enter the received codeword');
syndrome=rem(recd * parmat',2);
syndromel=bi2de(syndrome,'left-msb');
errorvect=trt(1+syndromel,:);
disp('errorvect');
disp(errorvect);
correctedcode=rem(errorvect+recd,2);
disp('correctedcode');
disp(correctedcode);
r=correctedcode;
m1(1)=r(5);m1(2)=r(6);m1(3)=r(7);m1(4)=r(1);m1(5)=r(2);
m1(6)=r(3);m1(7)=r(4);
m=decode(m1,n,k,'cyclic',g);
disp('messageword');
disp(m);

enter the data seq[1 0 0 1]
      code word1   1   0   1   0   0 1

enter the received codeword[1 1 1 1 1 1 1]
errorvect
      0   0   0   0   0   0   0

correctedcode
      1   1   1   1   1   1   1

messageword
      1   1   1   1
```

RESULT:

Thus the simulation for cyclic code is done using MATLAB

EXP NO: 13 SIMULATION 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 MATLAB.

APPARATUS REQUIRED:

1. Personal computer.
2. MATLAB software.

PROGRAM FOR CONVOLUTIONAL CODE:

```
for n = 1:length(EbNoVec)
    % Convert Eb/No to SNR
    snrdB = EbNoVec(n) + 10*log10(k*rate);
    % Noise variance calculation for unity average signal power.
    noiseVar = 10.^(-snrdB/10);
    % Reset the error and bit counters
    [numErrsSoft,numErrsHard,numBits] = deal(0);

    while numErrsSoft < 100 && numBits < 1e7
        % Generate binary data and convert to symbols
        dataIn = randi([0 1],numSymPerFrame*k,1);

        % Convolutionally encode the data
        dataEnc = convenc(dataIn,trellis);

        % QAM modulate
        txSig = qammod(dataEnc,M,'InputType','bit','UnitAveragePower',true);

        % Pass through AWGN channel
        rxSig = awgn(txSig,snrdB,'measured');

        % Demodulate the noisy signal using hard decision (bit) and
        % soft decision (approximate LLR) approaches.
        rxDataHard = qamdemod(rxSig,M,'OutputType','bit','UnitAveragePower',true);
        rxDataSoft = qamdemod(rxSig,M,'OutputType','approxllr',...
            'UnitAveragePower',true,'NoiseVariance',noiseVar);

        % Viterbi decode the demodulated data
        dataHard = vitdec(rxDataHard,trellis,tbl,'cont','hard');
        dataSoft = vitdec(rxDataSoft,trellis,tbl,'cont','unquant');

        % Calculate the number of bit errors in the frame. Adjust for the
        % decoding delay, which is equal to the traceback depth.
        numErrsInFrameHard = biterr(dataIn(1:end-tbl),dataHard(tbl+1:end));
        numErrsInFrameSoft = biterr(dataIn(1:end-tbl),dataSoft(tbl+1:end));

        % Increment the error and bit counters
        numErrsHard = numErrsHard + numErrsInFrameHard;
        numErrsSoft = numErrsSoft + numErrsInFrameSoft;
        numBits = numBits + numSymPerFrame*k;

    end

    % Estimate the BER for both methods
    berEstSoft(n) = numErrsSoft/numBits;
    berEstHard(n) = numErrsHard/numBits;
end
```

Plot the estimated hard and soft BER data. Plot the theoretical performance for an uncoded 64-QAM channel.

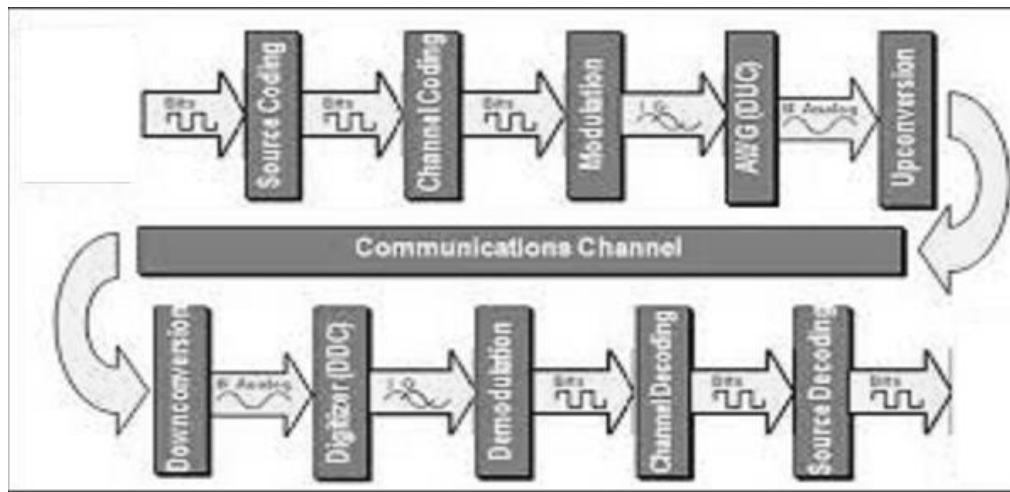
```
semilogy(EbNoVec,[berEstSoft berEstHard],'-*')
hold on
semilogy(EbNoVec,berawgn(EbNoVec,'qam',M))
legend('Soft','Hard','Uncoded','location','best')
grid
xlabel('Eb/No (dB)')
ylabel('Bit Error Rate')
```

RESULT:

Thus the simulation for convolutional code is done using MATLAB

\

Digital Communication block diagram



EXPT.NO 14	Digital communication link simulation using Simulink	DATE

AIM:

To study the performance of digital communication link using Matlab.

SOFTWARE REQUIREMENT:

PC with MATLAB software

THEORY:

Simulink is a block diagram environment for multidomain simulation and Model-Based Design. It supports system-level design, simulation, automatic code generation, and continuous test and verification of embedded systems. Simulink provides a graphical editor, customizable block libraries, and solvers for modeling and simulating dynamic systems. It is integrated with MATLAB, enabling you to incorporate MATLAB algorithms into models and export simulation results to MATLAB for further analysis.

KEY FEATURES:

1. Graphical editor for building and managing hierarchical block diagrams.
2. Libraries of predefined blocks for modeling continuous-time and discrete-time systems.
3. Simulation engine with fixed-step and variable-step ODE solvers.
4. Scopes and data displays for viewing simulation results.
5. Project and data management tools for managing model files and data.
6. Model analysis tools for refining model architecture and increasing simulation speed.
7. MATLAB Function block for importing MATLAB algorithms into models.
8. Legacy Code Tool for importing C and C++ code into models.

OUTPUT:

RESULT:

Thus the simulation for communication simulink is done using MATLAB

EC8561 COMMUNICATION SYSTEM LAB

VIVA QUESTIONS & ANSWERS

Experiment no - 1 SAMPLING AND RECONSTRUCTION

1. State sampling theory.

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 Nyquist interval?

$f_s = 2f_m$ is called the Nyquist rate. $T_s = 1/2f_m$ is called Nyquist interval.

3. Define the sampling process.

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

4. What is meant by quantisation?

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

5. What is meant by under sampling?

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

6. What is meant by over sampling?

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

7. What is meant by nyquist sampling?

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

8. What are the types of sampling?

a. Natural sampling b. Flat top sampling c. Ideal sampling d. Sample and hold

9. What is meant by reconstruction?

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

10. What is meant by aliasing? When does it occur?

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 by multiplexing?

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

2. Explain the operation of TDM.

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 of TDM?

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

4. Differentiate TDM and FDM.

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 by CDMA?

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 by SDMA?

SDMA stands for Space Division Multiple Access.

7. Define synchronous multiplexer.

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 amplitude modulation.

In amplitude modulation, the amplitude of the carrier signal is varied according to the modulating signal.

2. What is the range of carrier used in AM?

500 Hz to 1500 KHz.

3. Which modulation is used for video signals?

Amplitude modulation is used for video signals.

4. What are the types of amplitude modulation?

- a. double sideband modulation (DSB-AM)
- b. double sideband suppressed carrier modulation (DSB SC-AM)
- c. single sideband carrier modulation(SSB AM)
- d. single sideband suppressed carrier modulation (SSB SC-AM)
- e. vestigial sideband modulation (VSB AM)

5. Define modulation index of AM.

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

$$\text{Modulation index, } h = \frac{M}{A}$$

M is the peak amplitude value of modulating signal.

A is the peak amplitude value of carrier signal.

6. What are the advantages of AM?

The AM transmitter and receiver are simple.

7. What are the disadvantages of AM?

AM is more susceptible to noise.

8. What are the applications of AM?

AM radio broadcasting., AM video transmission.

9. What are the types of continuous wave analog modulation?

Amplitude modulation

Angle modulation

10. What is VCO?

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

11. What is the role of mixer in AM demodulator?

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

12. What is the use of two stage amplifier?

The two stage amplifier is used for improving the gain.

Experiment no - 4 FREQUENCY MODULATION

1. Define frequency modulation.

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

2. What is the range of carrier used in FM?

88 MHz to 108 MHz.

3. What are the types of frequency modulation?

- Narrow band FM ($h < 1$)
- Wide band FM ($h \geq 1$)

4. Define modulation index of FM.

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

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

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

5. What are the advantages of FM?

Immunity to noise

6. What are the disadvantages of FM?

Bandwidth requirement is more in FM.

Transmitter and receiver are very complex.

7. What are the applications of FM?

- For recording luminance component of the video.
- Audio signal broadcasting.

8. What are the types of angle modulation?

Frequency modulation and phase modulation.

9. What is the role of a limiter in FM system?

To limit the amplitude variations in a FM signal.

Experiment No – 5

PULSE CODE MODULATION

1. Define Pulse code modulation?

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

2. How bits are needed to encode N different levels?

$$\log_2 N$$

3. Define step size?

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?

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

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

4. Define Quantization error?

It is the difference between sampled signal and Quantized signal

5. Define Differential pulse code modulation?

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

Experiment no - 6 DELTA AND ADAPTIVE DELTA MODULATION

1. What is the reason to use delta modulation?

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. Draw the DM waveform for an analog input signal.

4. What are the two types of errors present in DM?

Slope over load distortion and granular noise.

5. Define slope overload distortion with an illustrative waveform.

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.

6. Define granular noise with an illustrative waveform.

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

7. How to reduce the slope overload distortion?

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

8. How to reduce the granular noise?

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

9. Explain the operation of DM.

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.

10. What is the reason to go for ADM?

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

11. Explain the operation of ADM.

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.

12. What does the term adaptive mean in ADM?

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

13. What is the condition to avoid slope overload distortion in DM?

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

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.

14. Differentiate DM and ADM.

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.

15. What is the use of higher order filters at the receiver side?

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

Experiment no -7

LINE CODING TECHNIQUES

1. Define line coding.

Line coding is the waveform representation of digital data.

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

- In uni-polar signalling, symbol '0' is represented by 0 volts. Symbol '1' is represented by a positive pulse.
- In polar signalling, symbol '0' is represented by negative pulse. Symbol '1' is represented by a positive pulse.
- In bi-polar signalling, symbol '0' is represented by 0 Volts. Symbol '1' is represented by alternative positive and negative pulse.

3. Differentiate NRZ and RZ signalling.

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 bit duration.

4. What is meant by NRZ-L, NRZ-M and NRZ-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 coding techniques?

Bi- Φ -level, bi- Φ -mark and bi- Φ -space

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

- 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 starting transition.
- 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 starting transition.

7. What is Manchester code? Draw the Manchester format for the data stream 10110?

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 line code

- The PAM signal should have adequate timing content,
- The PAM signal should be immune to channel noise and interference
- The PAM signal should allow error detection and error correction
- The PAM signal should be transparent to digital data being transmitted

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

1. Give examples for digital modulation techniques.

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-band transmission?

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-band transmission?

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 for modulation?

- For placing the signal at low noise frequency.
- For reducing the antenna size.

5. Define ASK, FSK, PSK.

- ASK: In ASK, the digital data modulates the amplitude of the carrier.
- FSK: In FSK, the digital data modulates the frequency of the carrier.

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

6. What is QPSK?

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 of QPSK?

Non offset QPSK, Offset QPSK (OQPSK), $(\pi/4)$ QPSK.

8. What is meant by coherent detection?

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-coherent detection?

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 by DPSK?

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 Error Rate?

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 antipodal signaling?

The phase relationship between signaling elements for BPSK (i.e., 180° 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 the PSK.

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 where $M=16$.

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

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 modulated signals.

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. $\Delta f = v_m(t) * k_1$ Where, Δf = Peak frequency deviation (Hz) $v_m(t)$ = peak binary modulating signal voltage (volts) k_1 = deviation sensitivity (Hz per volt)

9. Define QPSK.

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

10. Define Dibit.

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 DPSK transmitter.

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

12. What are the three broad types of synchronization?

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

13. What is carrier synchronization?

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 carrier synchronization?

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

15. What is called symbol or bit synchronization?

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 symbol synchronization?

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

Experiment no -12 ERROR CONTROL CODING (Linear code and Cyclic code)

1. What is the need for error control coding?

For error detection and error correction of the code word.

2. What is meant by linear code?

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 cyclic code?

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 coding techniques.

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

5. Name the code which is used only for error detection.

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 error pattern?

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 code vector.

8. Differentiate systematic and non-systematic coding?

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

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

Experiment no -13 CONVOLUTIONAL CODING SCHEMES

1. Mention the properties of cyclic codes

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 hamming distance.

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 line codes

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 is two.

5. What is Golay codes?

Golay code is the (23,12) cyclic code whose generating polynomial is, $G(p) = p^{11} + p^9 + p^7 + p^6 + p^5 + p + 1$ This code has minimum distance of $d_{min} = 7$. 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-systematic codes?

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 linear code?

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 (d_{min}) 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 $d_{min} = 3$.

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 convolutional codes?

Advantages: 1. The decoding delay is small in convolutional codes since they operate on 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.