ST.ANNE'S

COLLEGE OF ENGINEERING & TECHNOLOGY

ANGUCHETTYPALAYAM, PANRUTI – 607 110

EC6512 COMMUNICATION SYSTEMS LABORATORY

FOR B.E ELECTRONICS AND COMMUNICATION ENGINEERING STUDENTS

AS PER ANNA UNIVERSITY CHENNAI SYLLABUS **2013 REGULATION**

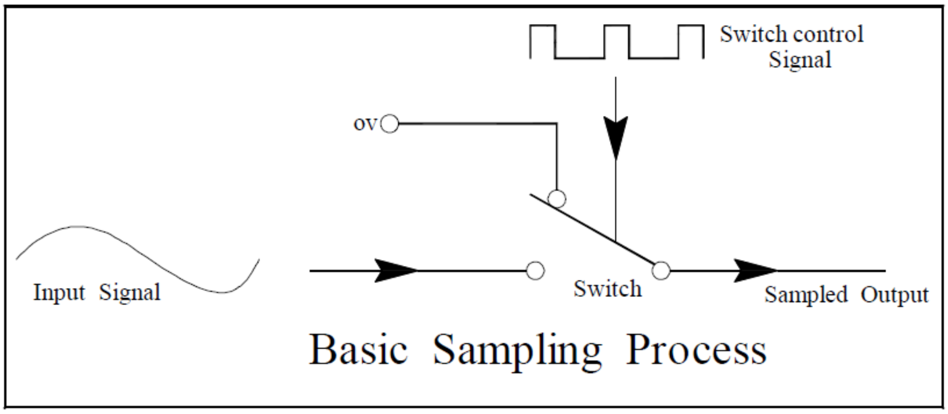
DEPARTMENT OF ELECTRONICS AND COMMUNICATION ENGINEERING

EC6512 COMMUNICATION SYSTEMS LABORATORY

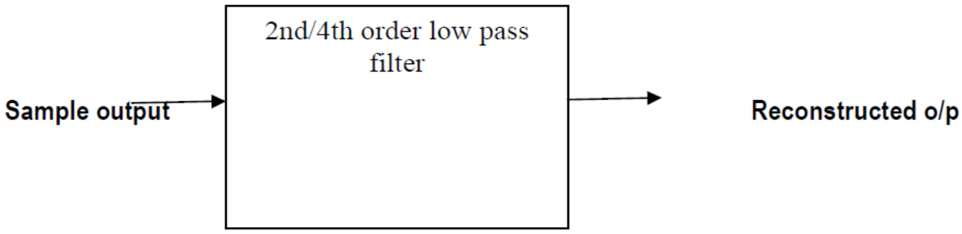
LIST OF EXPERIMENTS

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. Observation (simulation) of signal constellations of BPSK, QPSK and QAM
8. Line coding schemes
9. FSK, PSK and DPSK schemes (Simulation)
10. Error control coding schemes - Linear Block Codes (Simulation)
11. Communication link simulation
12. Equalization – Zero Forcing & LMS algorithms(simulation)

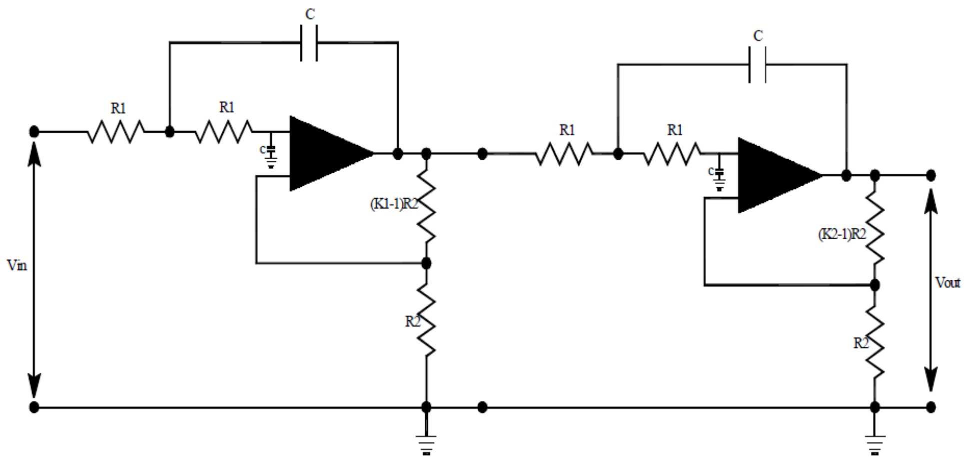
BLOCK DIAGRAM OF SAMPLING PROCESS



BLOCK DIAGRAM OF RECONSTRUCTION PROCESS



Reconstruction



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 Criterian), 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 $F_s \geq 2 F_m$.

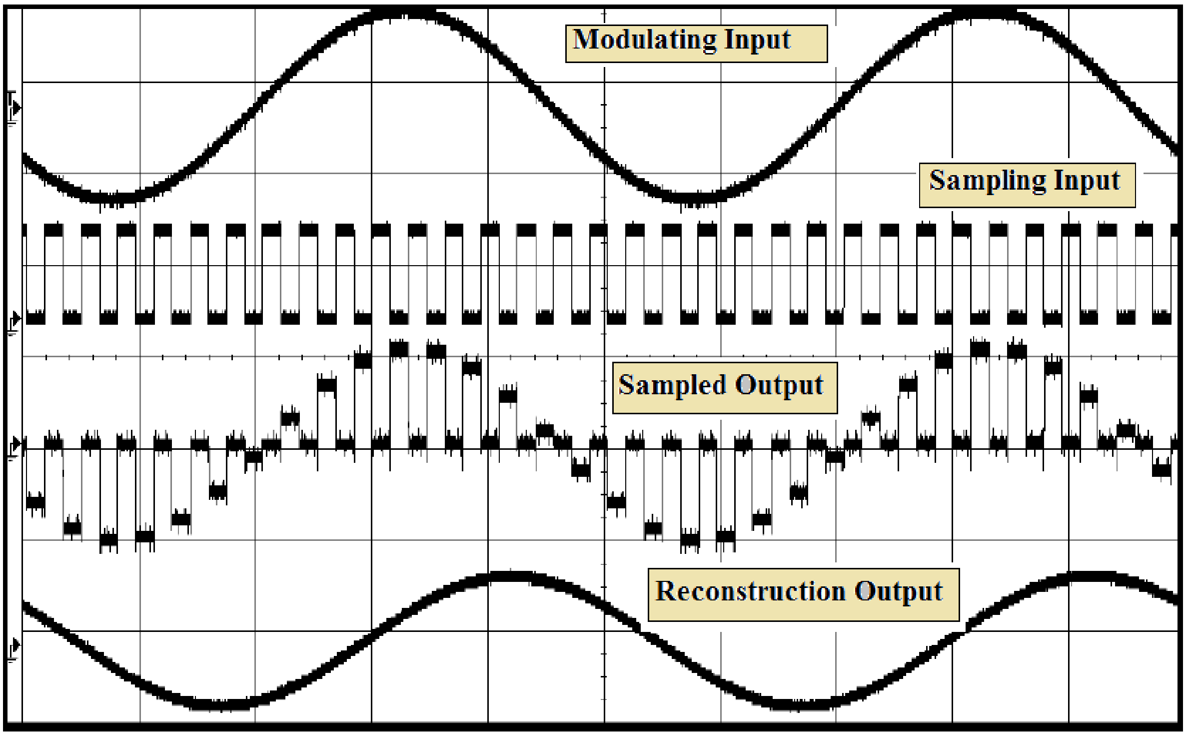
ALIASING

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			

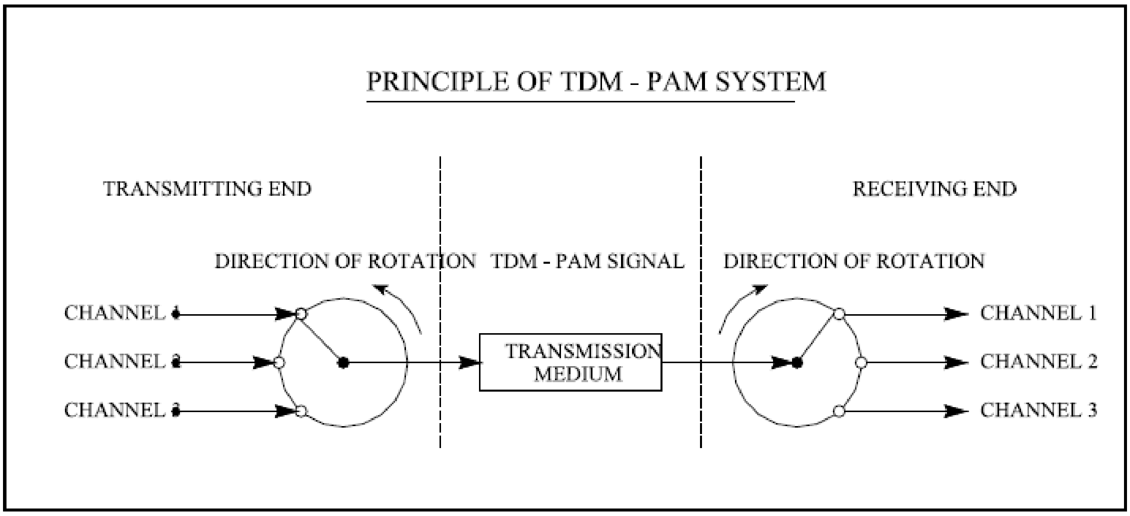
PROCEDURE

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.

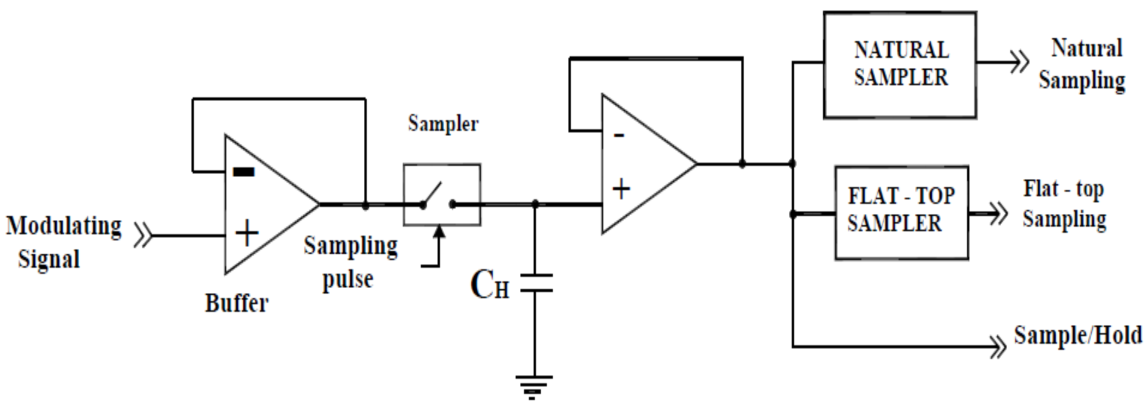
TDM USING PAM, MODULATION & DEMODULATION



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.

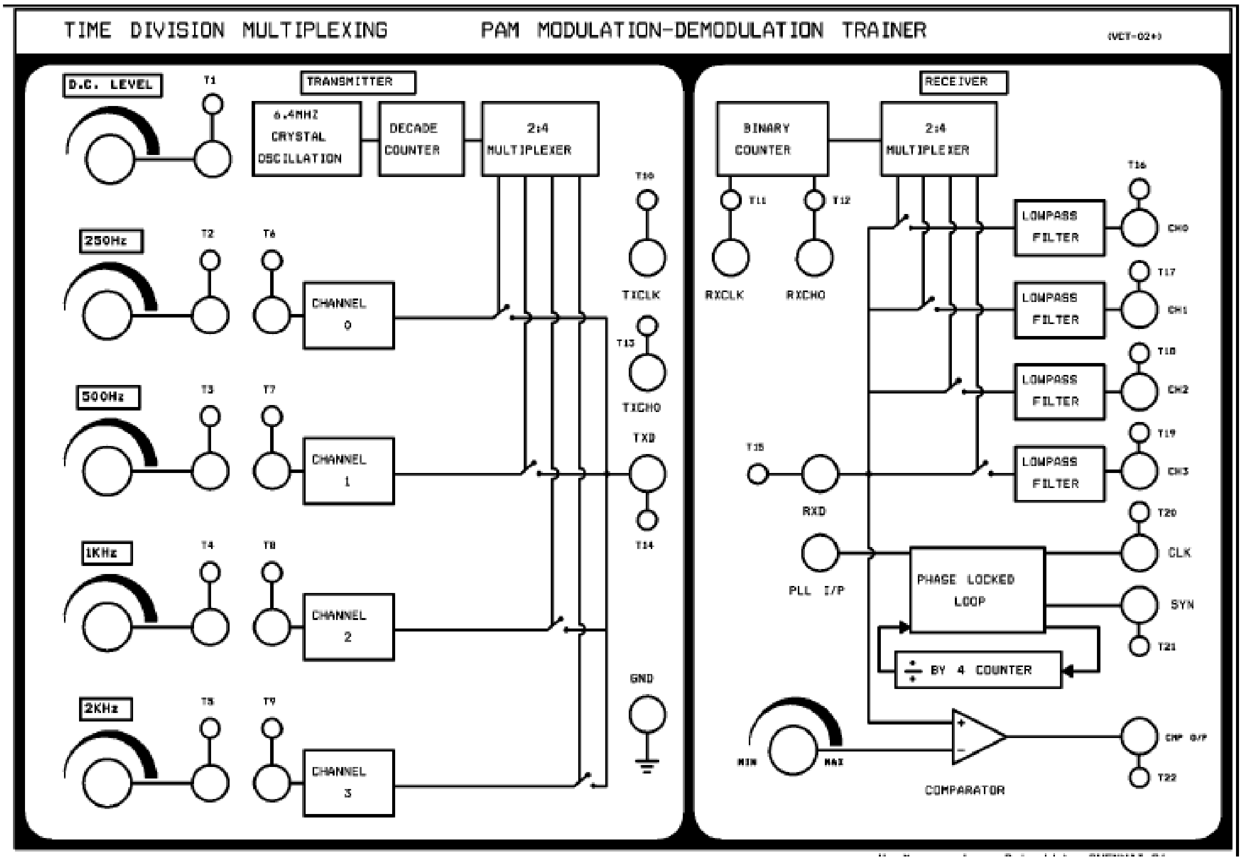
Here, each signal is sampled over one sampling interval and transmitted one after the other along a common channel. But the receiving end has to follow some constraints.
i. It must receive and show the signal as the transmitted.
ii. It must start at the same time as the transmitting end and establish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

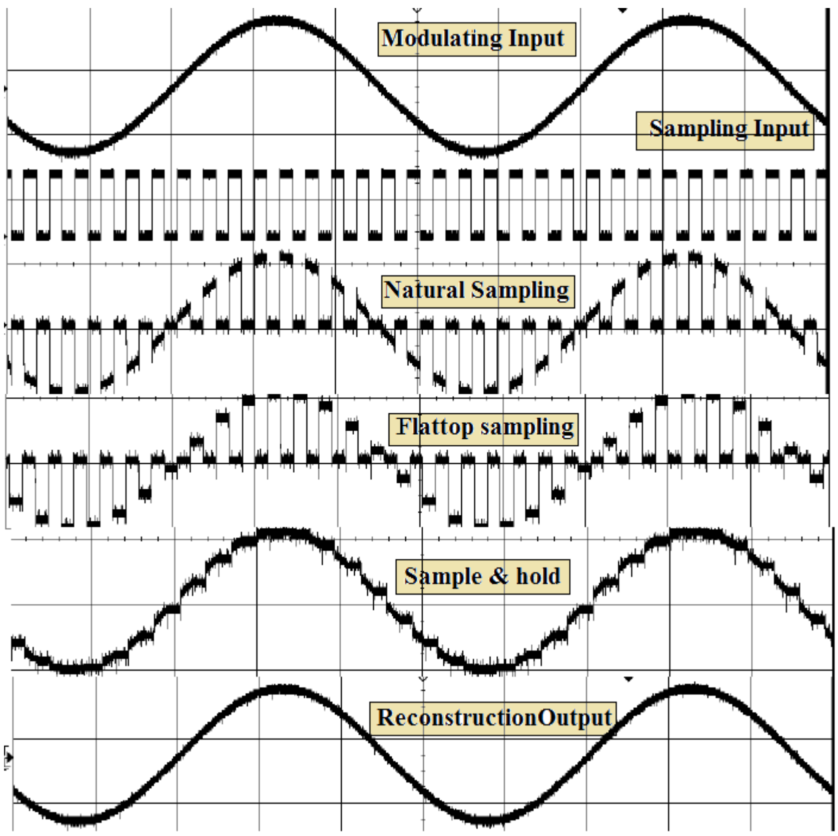
PULSE AMPLITUDE MODULATION

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

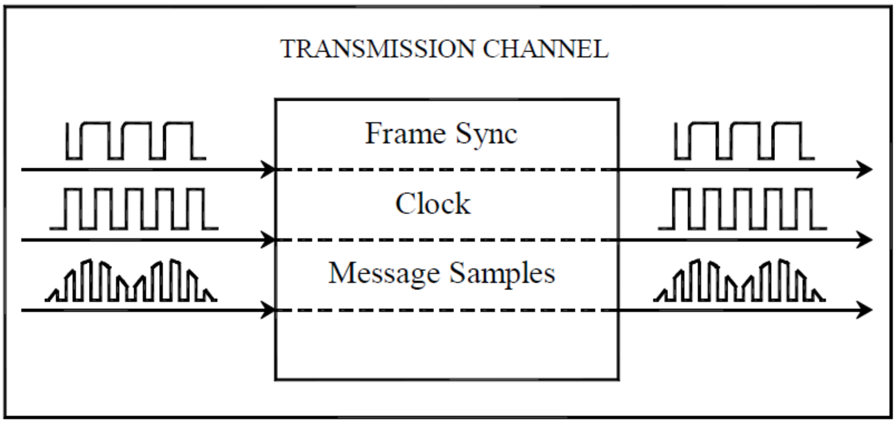

TIME DIVISION MULTIPLEXING PAM MODULATION-DEMODULATION TRAINER
TRANSMITTER
D.C. LEVEL
6.4MHZ CRYSTAL OSCILLATION
DECADE COUNTER
2:4 MULTIPLEXER
250Hz
500Hz
1KHz
2KHz
CHANNEL 0
CHANNEL 1
CHANNEL 2
CHANNEL 3
TXCLK
TXCHO
TXD
GND
RECEIVER
BINARY COUNTER
2:4 MULTIPLEXER
RXCLK
RXCHO
LOWPASS FILTER
CH0
CH1
CH2
CH3
RXD
PLL I/P
PHASE LOCKED LOOP
CLK
SYN
BY 4 COUNTER
COMPARATOR
MIN
MAX

MODEL GRAPH


Modulating Input
Sampling Input
Natural Sampling
Flattop sampling
Sample & hold
ReconstructionOutput

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.



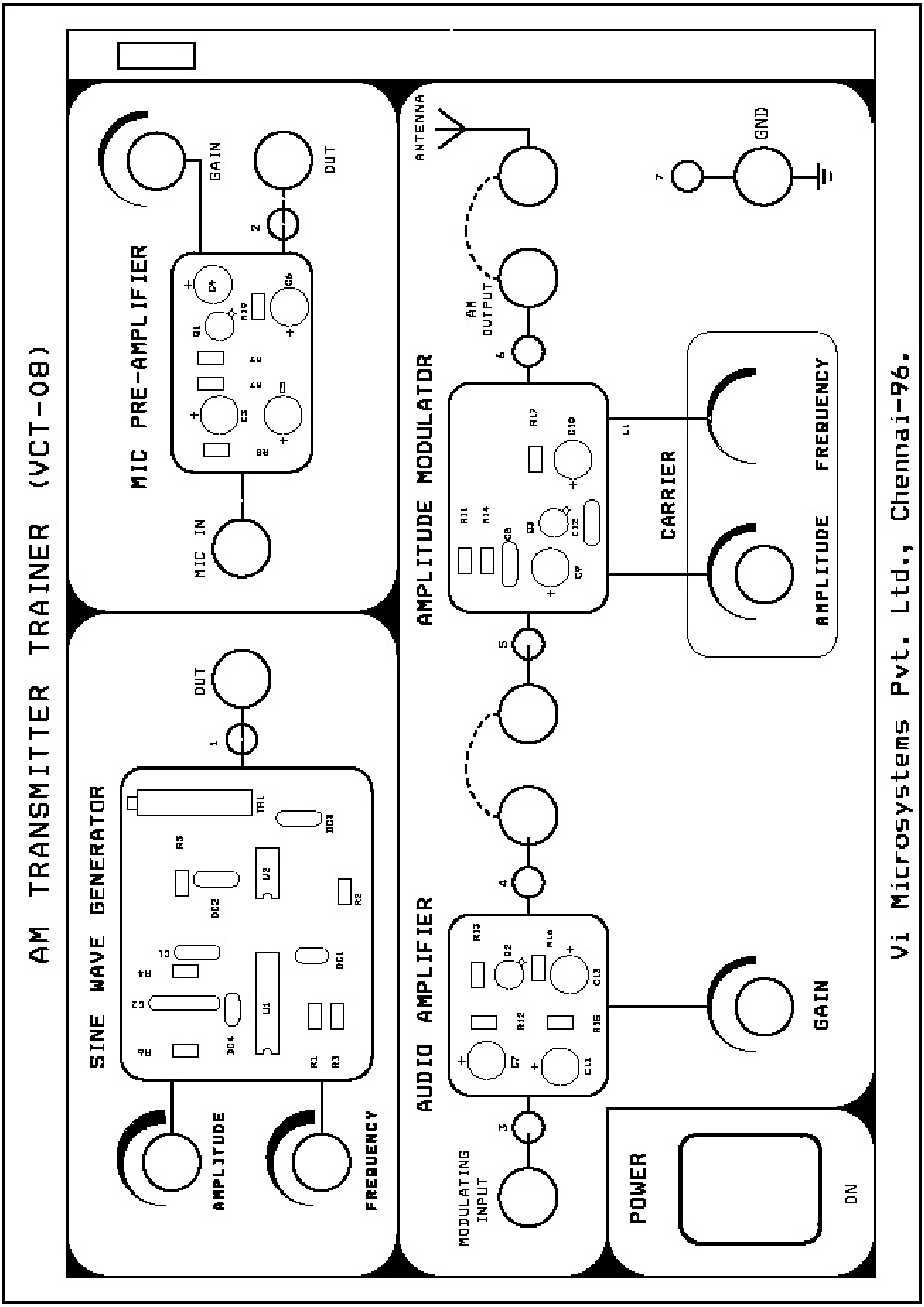
In this mode of operation, the signals from all the channels are taken to the multiplexer and are combined with the carrier signal produced by oscillator and the counter and then it is finally multiplexed and sent through the transmission data signal (TXD) to the receiving data(RXD). The clock signal 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)
SINE WAVE GENERATOR
AMPLITUDE
FREQUENCY
OUT
1
MIC PRE-AMPLIFIER
MIC IN
GAIN
OUT
2
AUDIO AMPLIFIER
MODULATING INPUT
3
4
GAIN
AMPLITUDE MODULATOR
5
6
AM OUTPUT
ANTENNA
7
GND
CARRIER
AMPLITUDE
FREQUENCY
POWER
ON
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

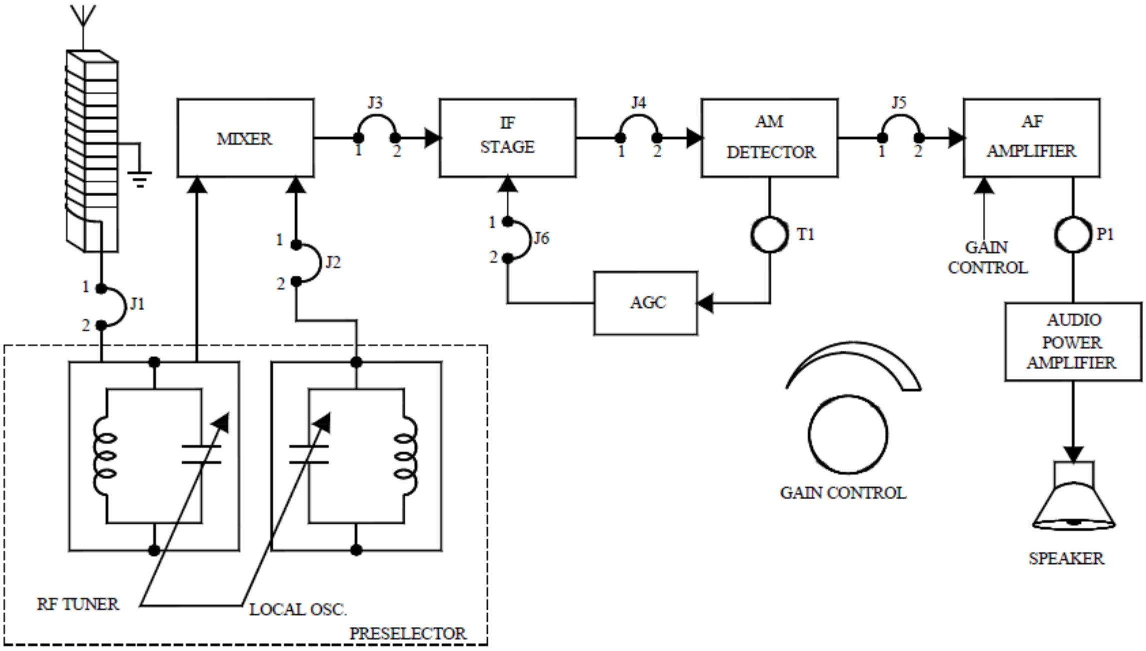
In radio transmission, it is necessary to send audio signal (eg. music, speech etc) from a broad casting station over great distances to a receiver. The audio signal cannot be sent directly over the air for appreciable distance. Even if audio signal is converted into electrical signal, the latter cannot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

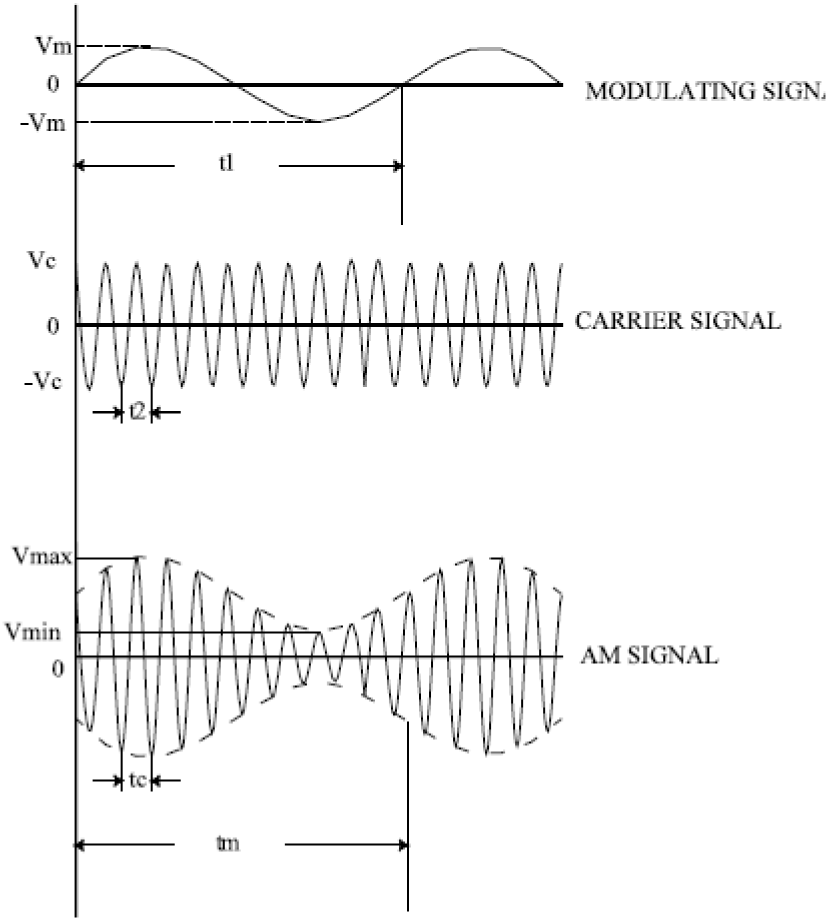
The main problem is that a high frequency carrier wave is used to carry the audio signal and that we need to know how the audio signal should be "added" to the carrier wave. The solution lies in changing some characteristic of carrier wave in accordance with the signal. Under such conditions, the audio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

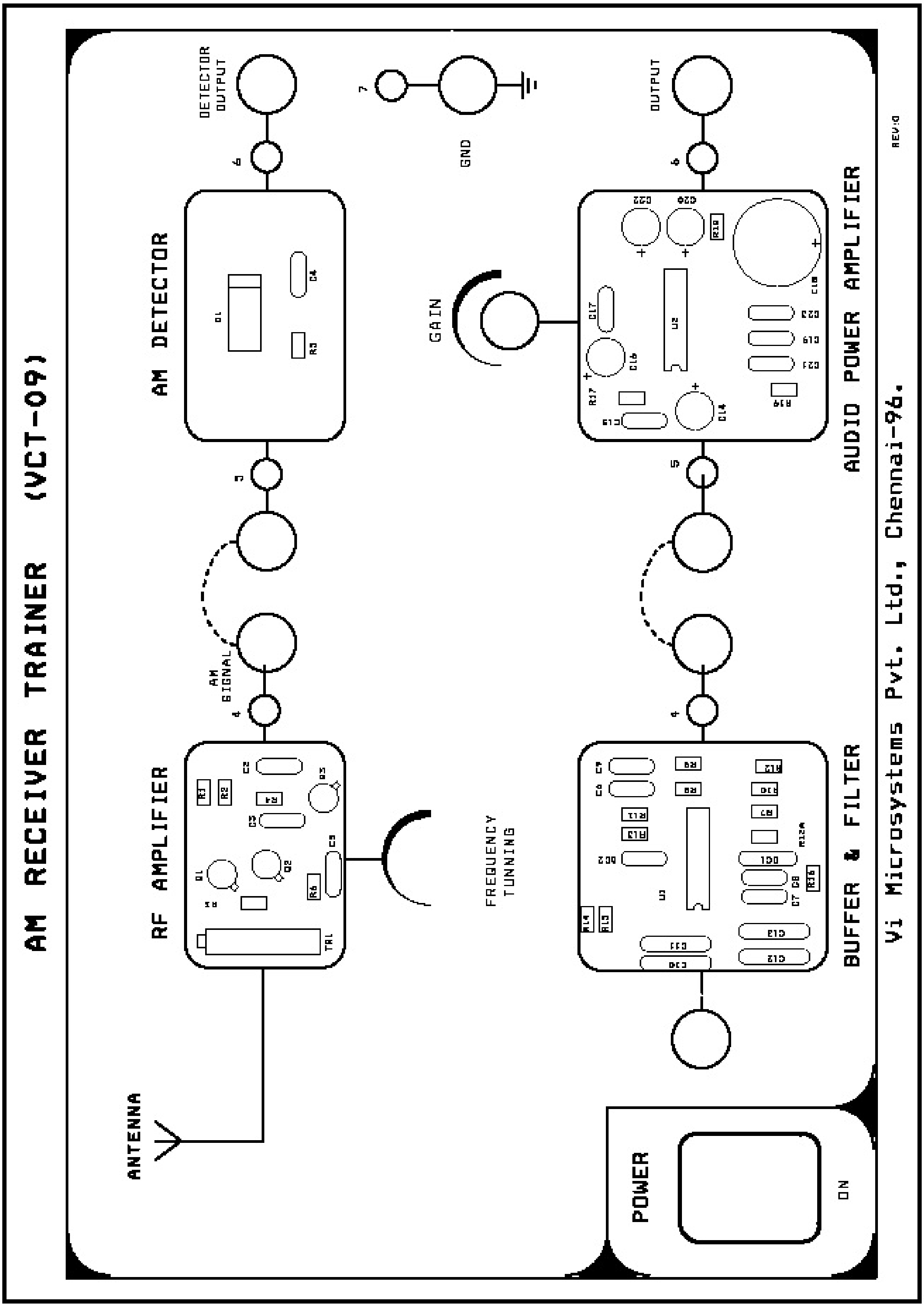
The ratio of change of amplitude of carrier wave to the amplitude of normal carrier wave is called the modulation factor (m) ie,

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

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

PROCEDURE

1. The circuit wiring is done as shown in the 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)
RF AMPLIFIER
AM DETECTOR
BUFFER & FILTER
AUDIO POWER AMPLIFIER
Vi Microsystems Pvt. Ltd., Chennai-96.
ANTENNA
AM SIGNAL
DETECTOR OUTPUT
GND
OUTPUT
GAIN
FREQUENCY TUNNING
POWER
ON

TABULATION

Transmitter

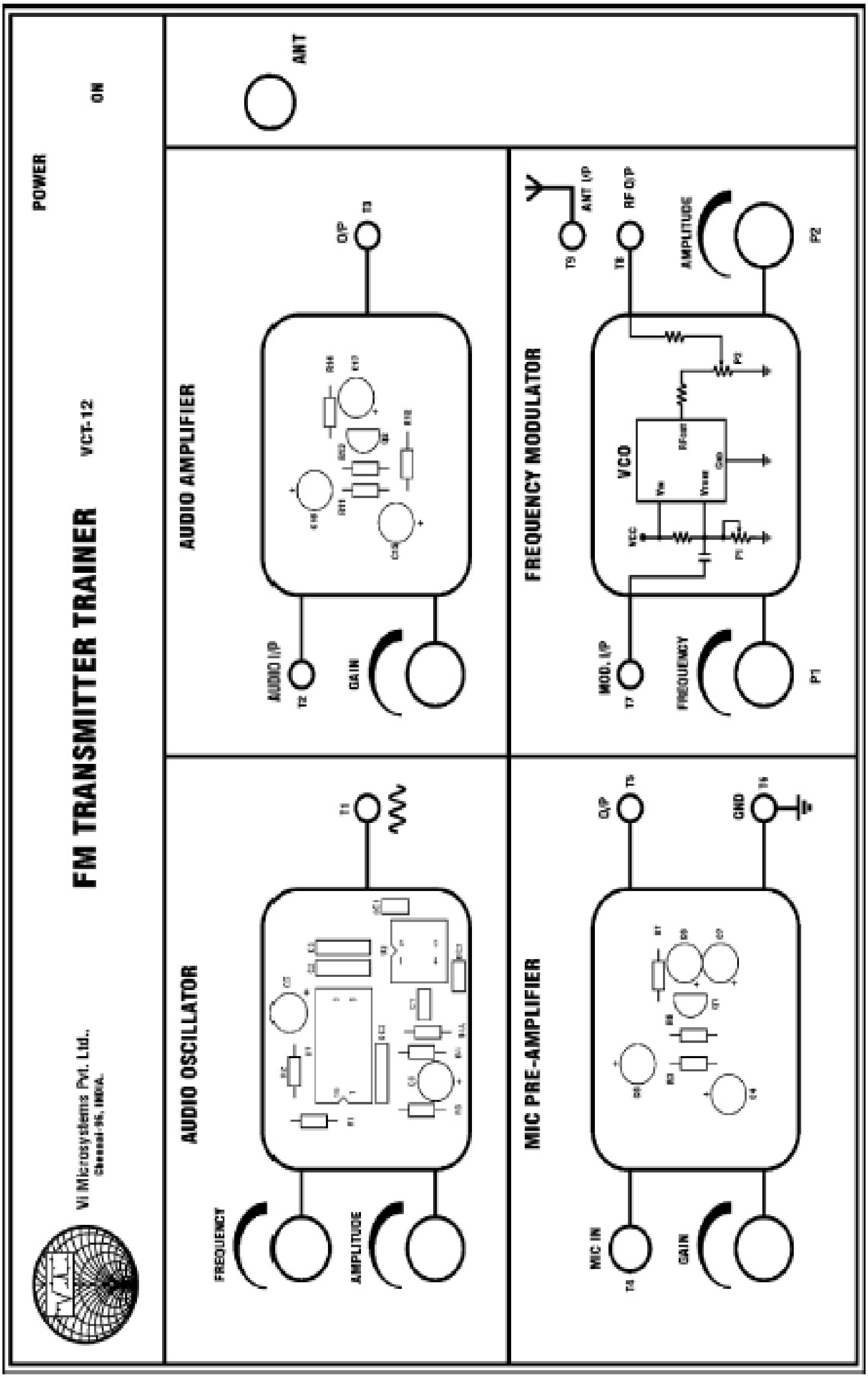
Signal	Amplitude	Time	Frequency
Modulating Signal	Vm =	t_1 =	f_1 =
Carrier Signal	Vc =	t_2 =	f_2 =
Amplitude Modulated wave	Vmax = Vmin	t_m = t_c =	f_m = f_c =

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.


Vi Microsystems Pvt. Ltd.,
Chennai-96, INDIA.
FM TRANSMITTER TRAINER
VCT-12
POWER
ON
ANT
AUDIO OSCILLATOR
FREQUENCY
AMPLITUDE
T1
AUDIO AMPLIFIER
AUDIO I/P
T2
GAIN
O/P
T3
MIC PRE-AMPLIFIER
MIC IN
T4
GAIN
O/P
T5
GND
T6
FREQUENCY MODULATOR
MOD. I/P
T7
FREQUENCY
P1
VCO
VCC
AMPLITUDE
P2
RF O/P
T8
ANT I/P
T9

4.FREQUENCY MODULATION AND DEMODULATION

AIM

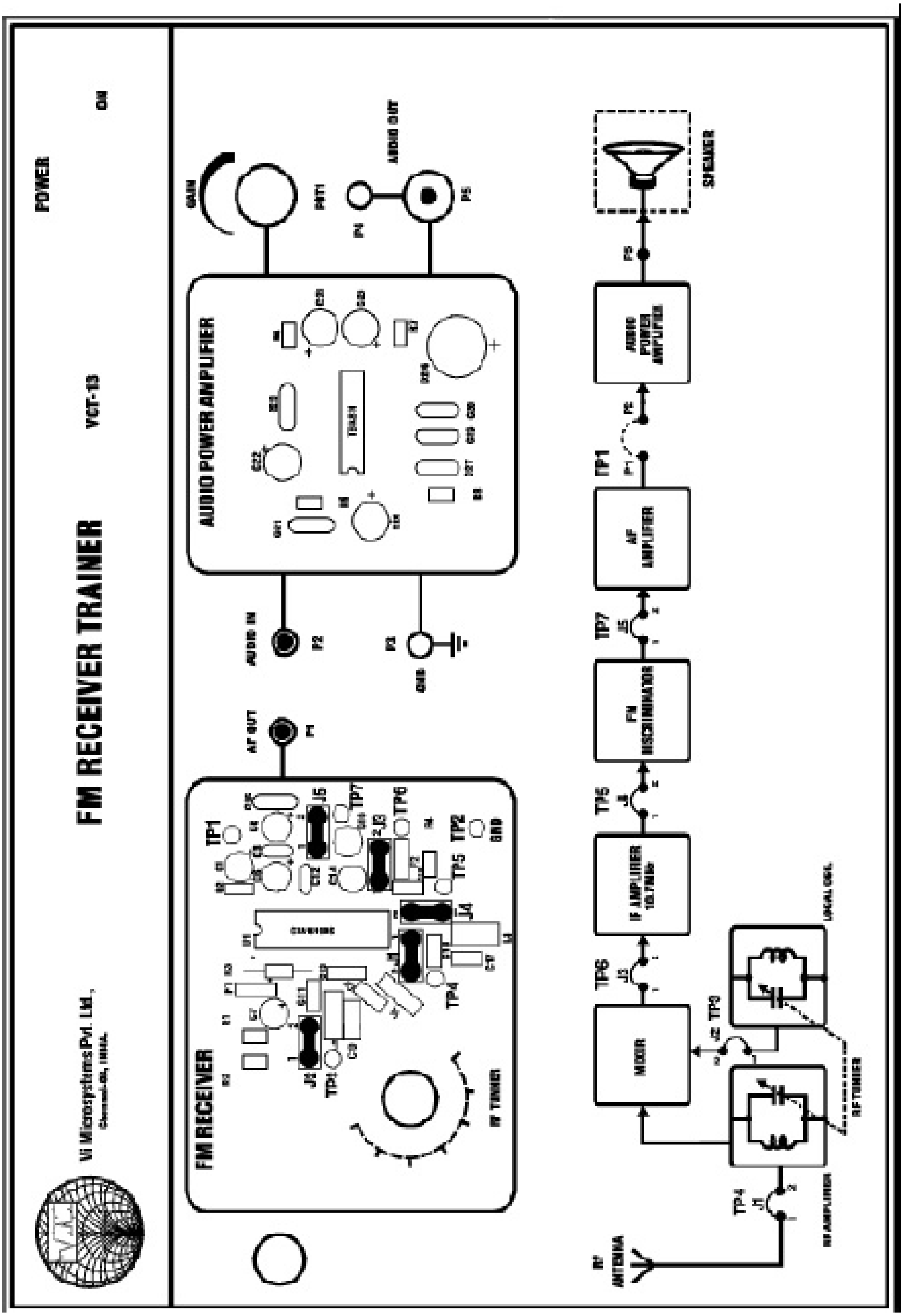
To transmit a tone (sinusoidal signal) using trainerkit and receive the signal back after demodulator using trainer kit

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)



FM RECEIVER TRAINER
VCT-13
POWER
ON
Vi Microsystems Pvt. Ltd.,
FM RECEIVER
AUDIO POWER AMPLIFIER
GAIN
AUDIO OUT
AF OUT
AUDIO IN
GND
TP1
TP2
TP3
TP4
TP5
TP6
TP7
RF ANTENNA
RF AMPLIFIER
RF TUNER
LOCAL OSC.
MIXER
IF AMPLIFIER 10.7 MHz
FM DISCRIMINATOR
AF AMPLIFIER
AUDIO POWER AMPLIFIER
SPEAKER

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 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 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.

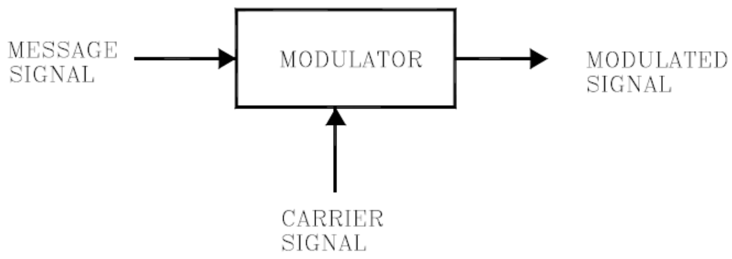
In Direct method the carrier frequency is directly varied in accordance with input baseband signal which is readily accomplished using a voltage control oscillator (VCO).
In Indirect method the modulating signal is first used to produce a narrowband FM signal and frequency multiplication is next used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

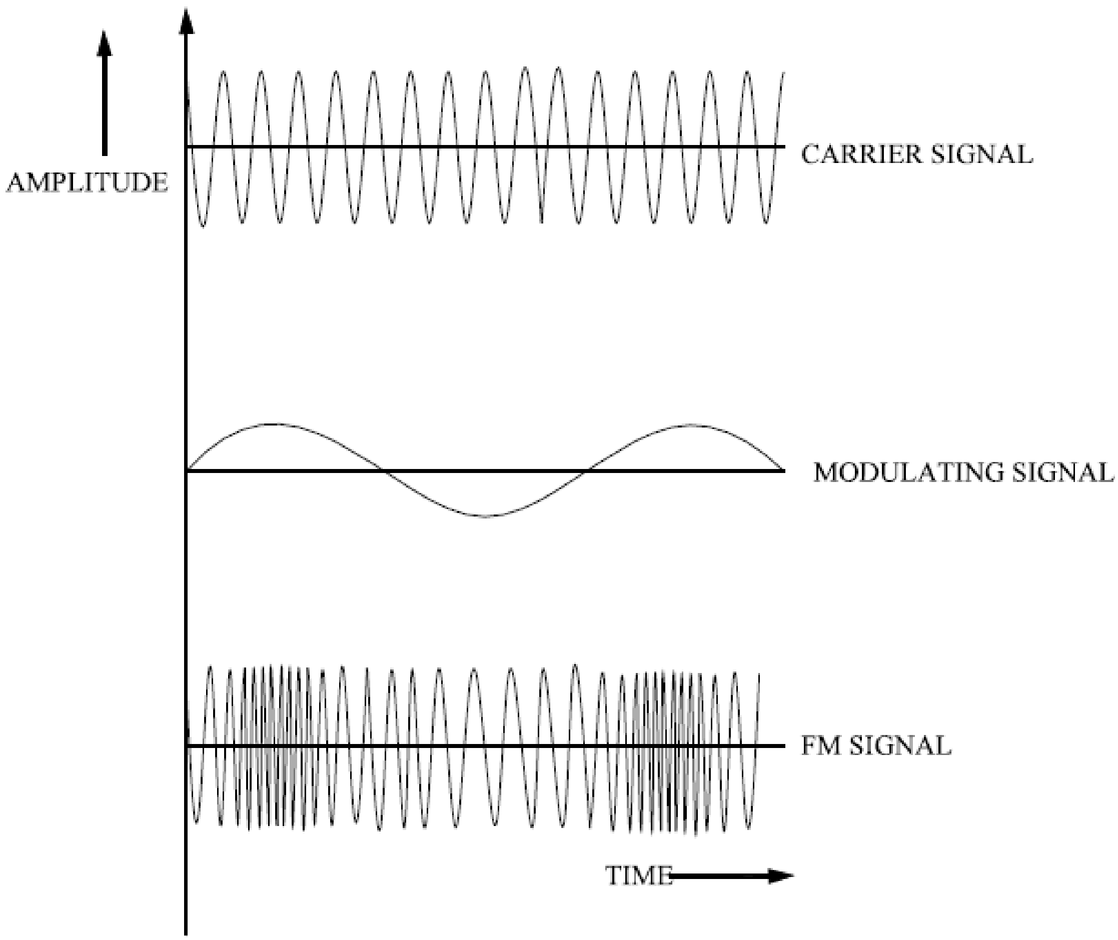
DEMODULATION

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



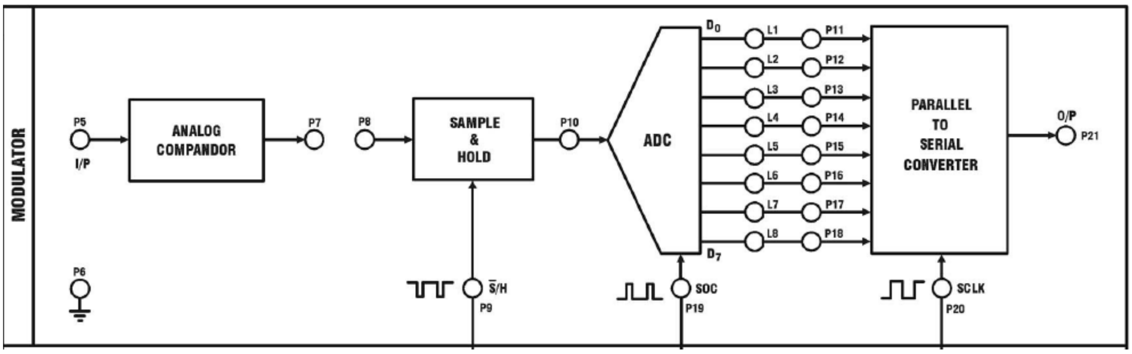
TABULATIO
N

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

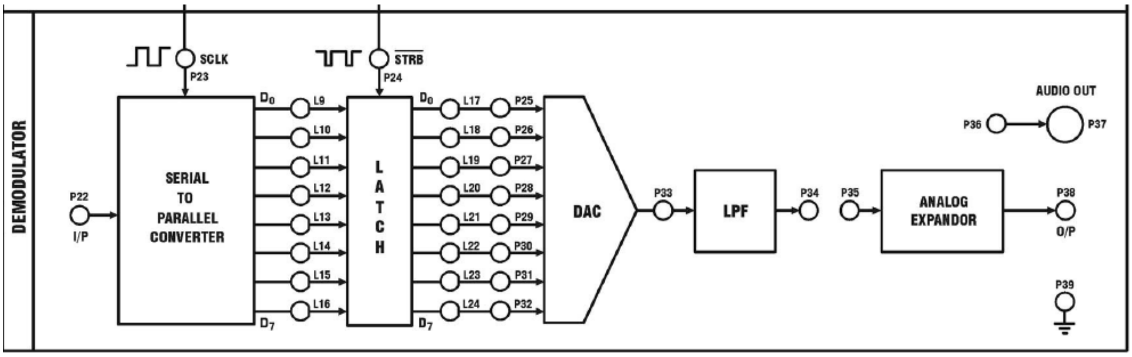
RESULT

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 conceptof pulse code modulation and demodulation..

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

THEORY

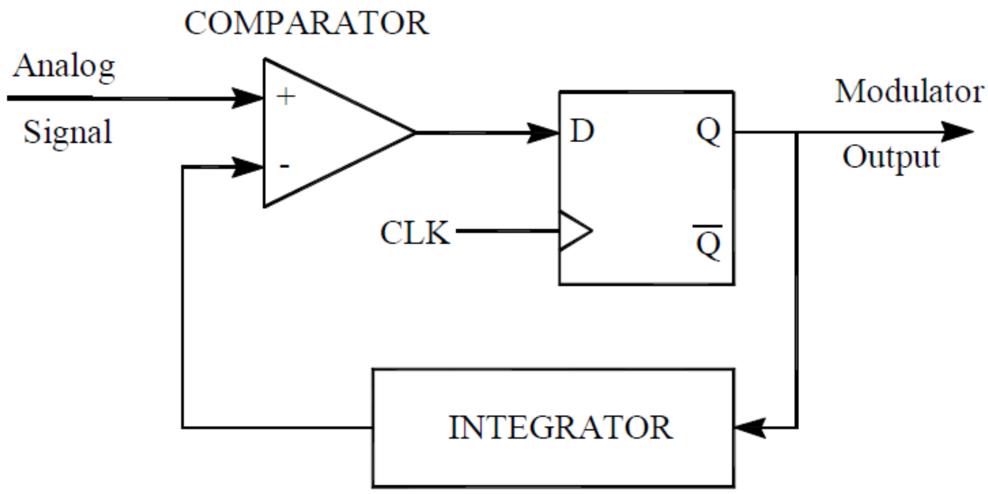
In pulse code modulation, each analog sample converted into eight bit code and they are transmitted in serial form. The PCM system consists of a sample/hold circuit, analog to digital converter and parallel to serial converter. The 1KHz on board sinewave signal can be used for studying modulation and 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) 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 channel1 of oscilloscope. Increase sinewave amplitude by rotating POT1 in clockwise $_{pp}$ 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.

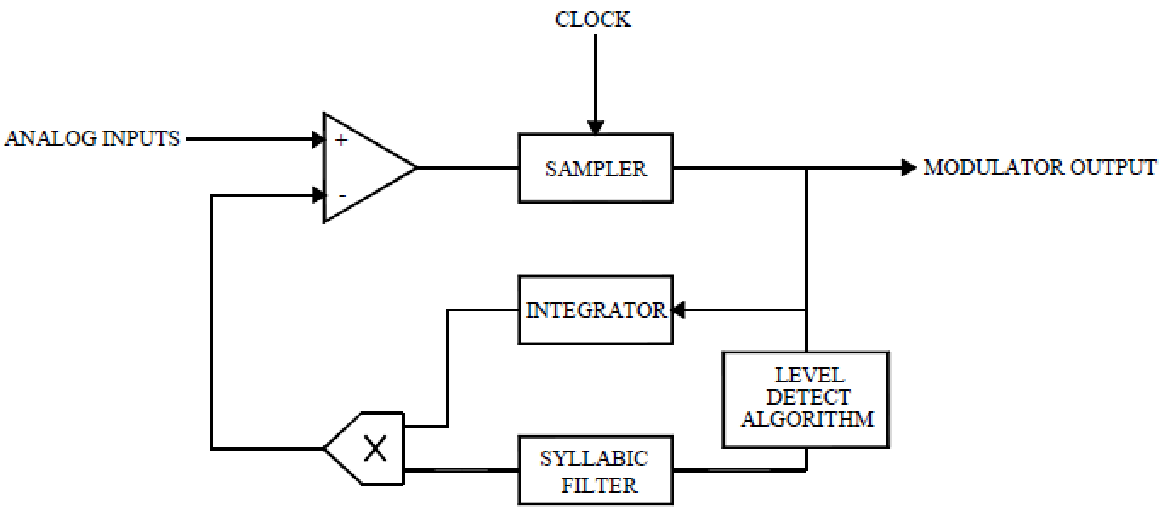
RESULT
Thus the Study of pulse code modulation and demodulation verified successfully.

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

a. DM / ADM kit
b. Two Channel 20MHz Oscilloscope
c. 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 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.

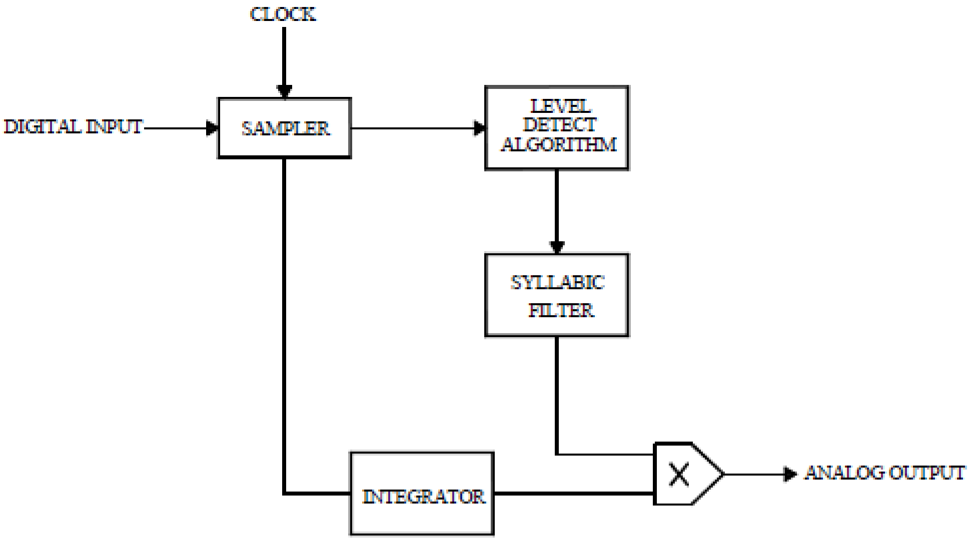
Idle channel noise

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 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). 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

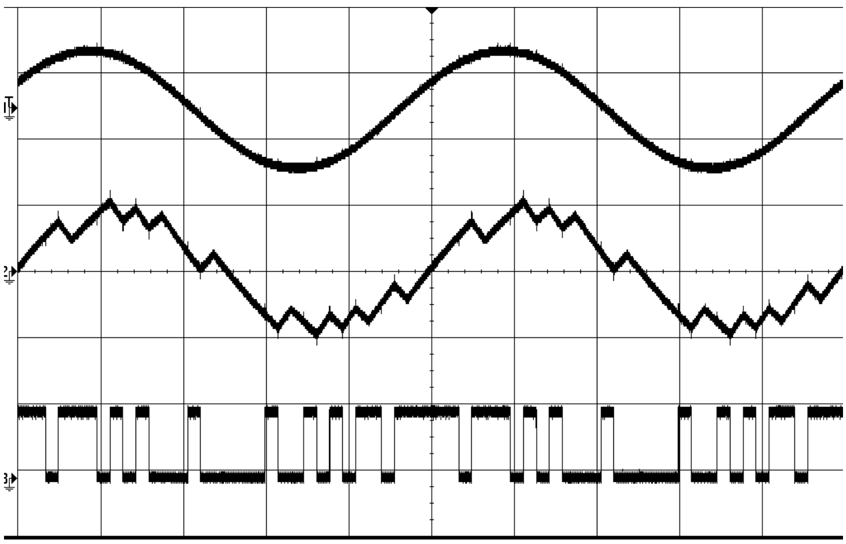
ADAPTIVE DELTA DEMODULATOR

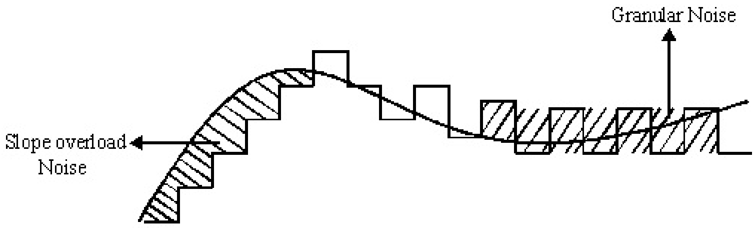


TABULATION

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





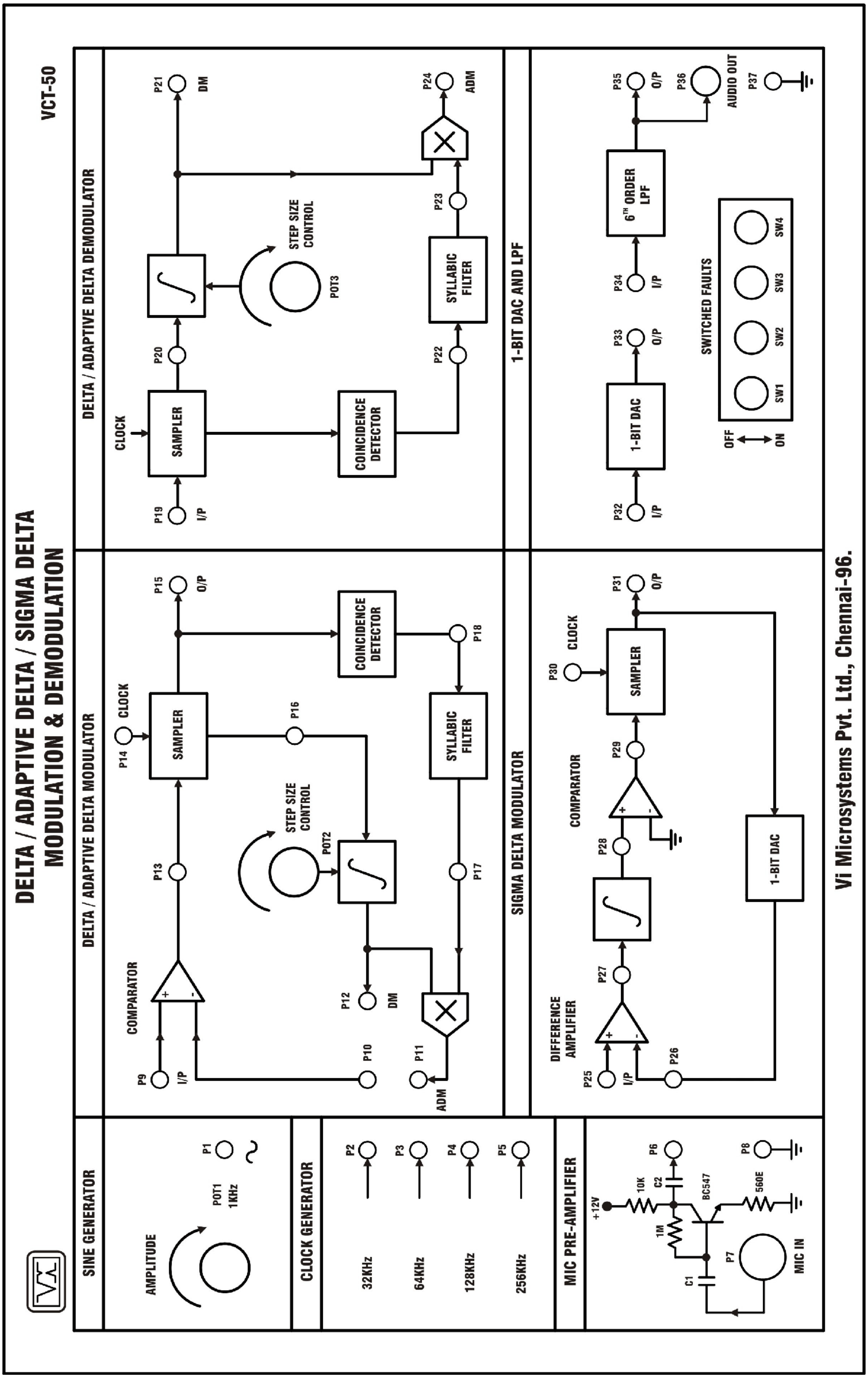
The step size) should be selected to bring a trade - off between slope overload and granular noise. Anyhow the error between the input and the recovered signals can be appreciably reduced by using low pass filter.

PROCEDURE

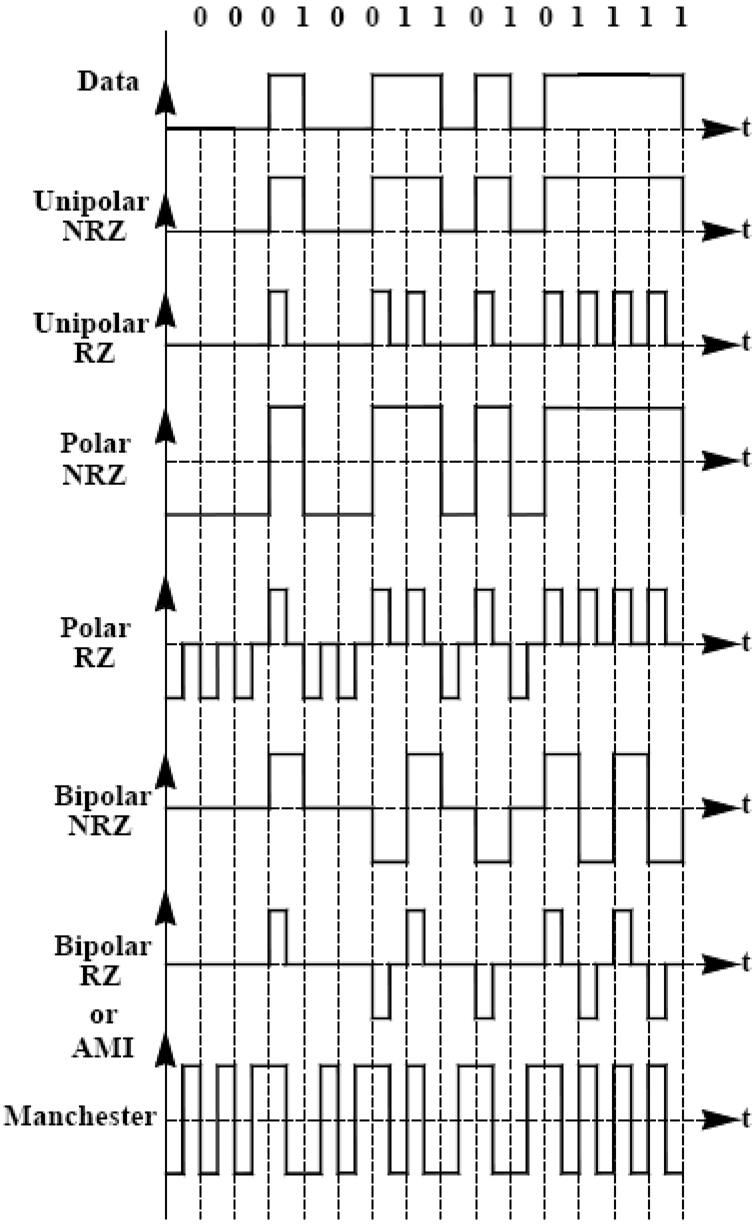
1. Plug-in AC power cord into 230V, 5A Mains power supply.
2. Ensure that the following initial conditions exist on VCT-50:
    a. Keep all switches in OFF position.
    b. Keep all potentiometer controls in minimum position.
3. Wiring connections do as follows in VCT-50:
    a. Connect point P1 (1KHz 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
SINE GENERATOR
AMPLITUDE
POT1
1KHz
P1
CLOCK GENERATOR
32KHz
64KHz
128KHz
256KHz
P2
P3
P4
P5
MIC PRE-AMPLIFIER
+12V
10K
C2
P6
1M
BC547
560E
P8
C1
P7
MIC IN
DELTA / ADAPTIVE DELTA MODULATOR
COMPARATOR
P9
I/P
P10
P11
ADM
P12
DM
P13
STEP SIZE
CONTROL
POT2
P14
CLOCK
SAMPLER
P16
P15
O/P
COINCIDENCE
DETECTOR
P18
SYLLABIC
FILTER
P17
SIGMA DELTA MODULATOR
DIFFERENCE
AMPLIFIER
P25
I/P
P26
P27
P28
COMPARATOR
P29
P30
CLOCK
SAMPLER
P31
O/P
1-BIT DAC
DELTA / ADAPTIVE DELTA DEMODULATOR
P19
I/P
CLOCK
SAMPLER
P20
STEP SIZE
CONTROL
POT3
P21
DM
COINCIDENCE
DETECTOR
P22
SYLLABIC
FILTER
P23
P24
ADM
1-BIT DAC AND LPF
P32
I/P
1-BIT DAC
P33
O/P
P34
I/P
6TH ORDER
LPF
P35
O/P
P36
AUDIO OUT
P37
SWITCHED FAULTS
OFF
ON
SW1
SW2
SW3
SW4
Vi Microsystems Pvt. Ltd., Chennai-96.


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
t

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 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

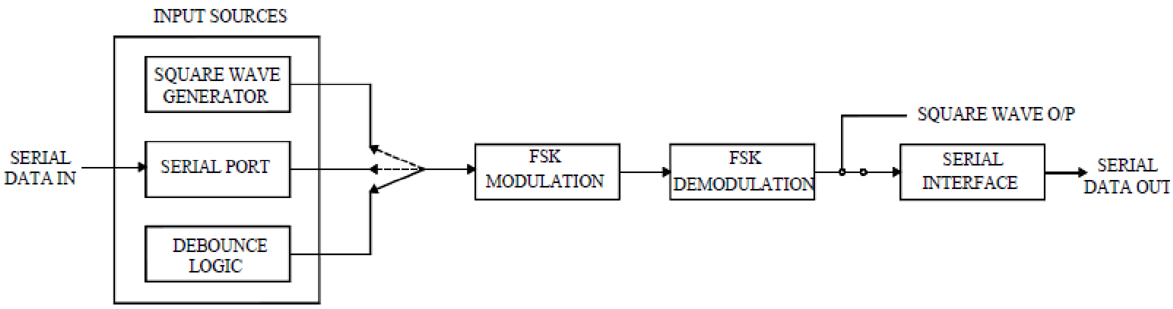
In this scheme, a binary „1' is represented by alternating positive and negative voltage a levels for a half bit period duration and maintaining zero for other of period. A binary „0' is represented by a zero voltage levels during entire bit duration. This code also called as AMI. This code has no DC 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

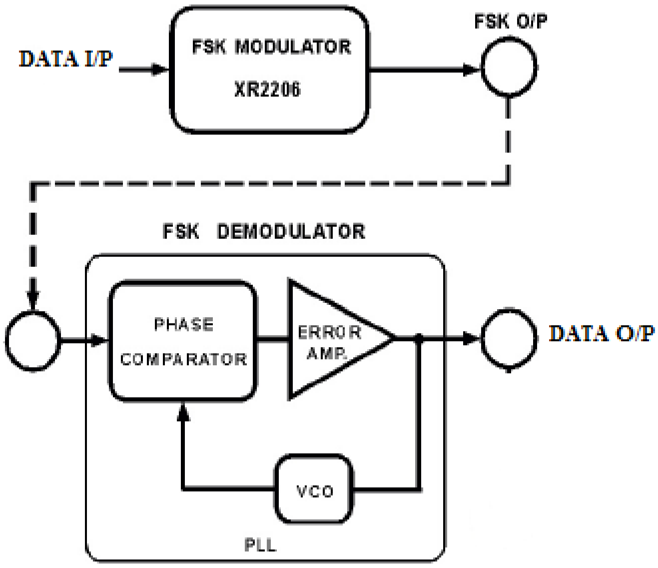
In this scheme, a binary 1 is represented by a pulse that has positive voltage during the first-half of the bit duration and negative voltage during second-half of the bit duration. A binary „0' is represented by a pulse that has negative voltage during first-half of the bit duration and positive voltage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

FSK MODULATION AND DEMODULATION



<u>Block Diagram Of FSK Modulation And Demodulation</u>



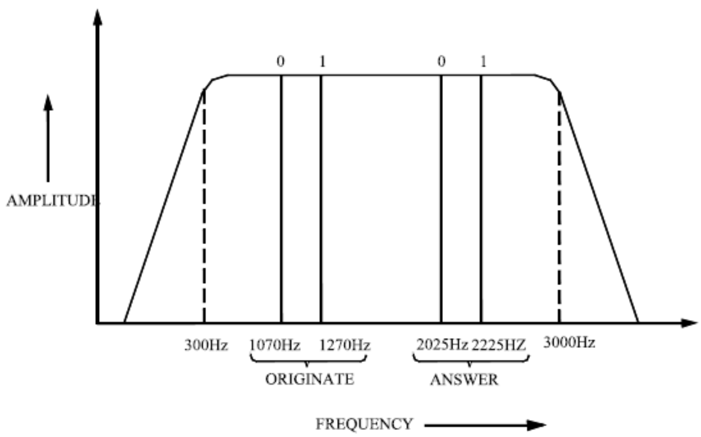
8.FSK ,PSK,DPSK SCHEME SIMULATION

AIM

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

THEORY

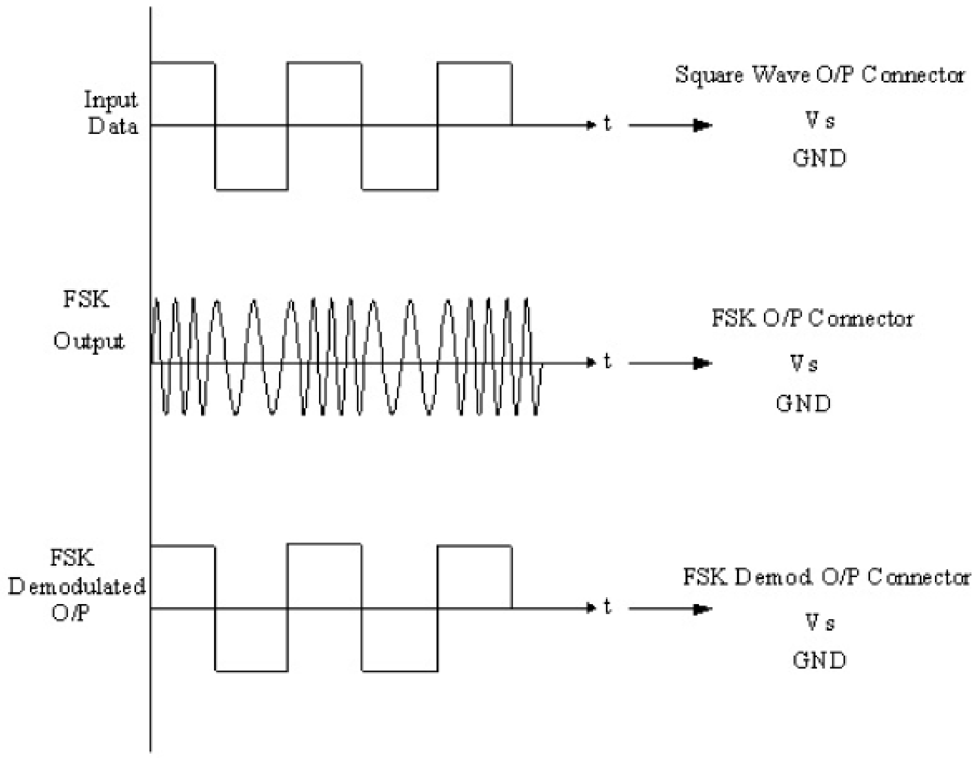
FSK is one method used to overcome the bandwidth limitation of the telephone system so that digital data can be sent over the phone lines. The basic idea of FSK is to represent 1s and 0s by two different frequencies within the telephone bandwidth. The standard frequencies for a full duplex 300 baud FSK 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

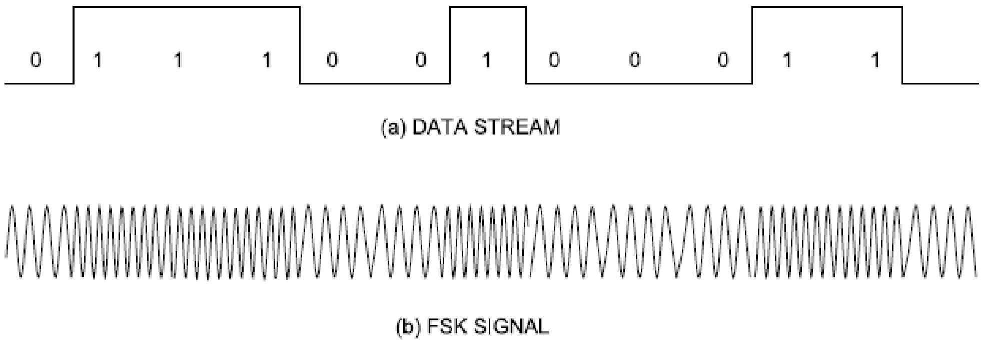
In FSK, the carrier frequency is shifted in steps or levels corresponding to the levels of the digital modulating signal. In the case of a binary signal, two carrier frequencies are used, one corresponding to binary '0' (i.e space) and the other to a binary 1 (i.e mark). An example of a digital data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

MODEL GRAPH



TABULATION

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



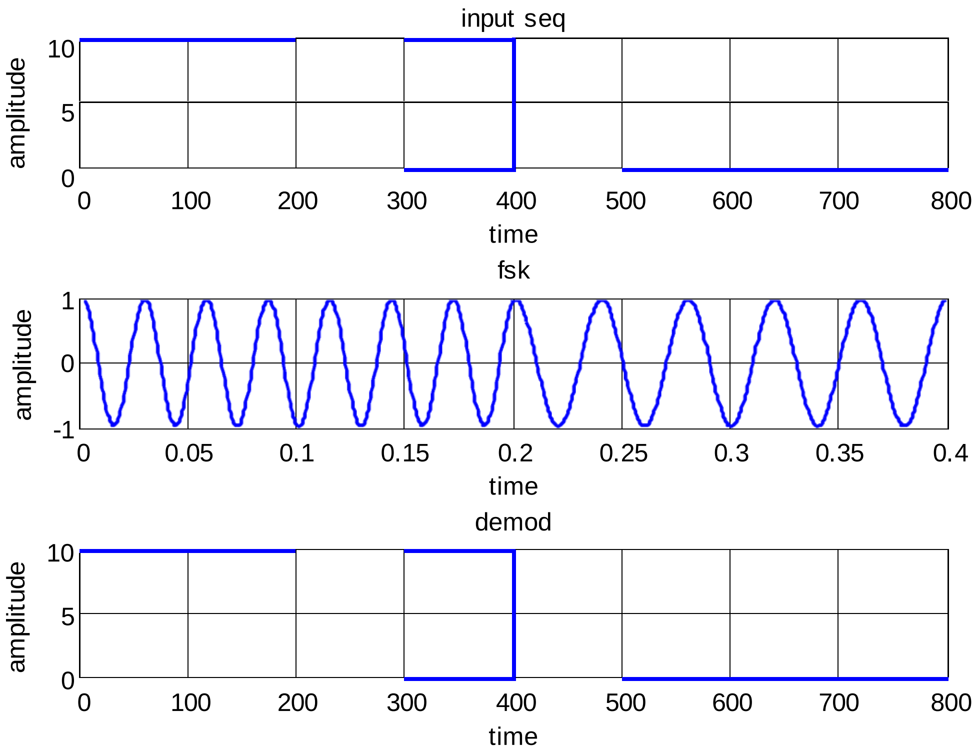
The baud rate is the number of changes of the transmitted data. This can be determined by taking the reciprocal of the time of the shortest pulse transmitted. A FSK Modulator & Demodulator sends and receives serial data at a rate of 300 bits or 300 baud. For a 300 baud data stream, maximum frequency 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.

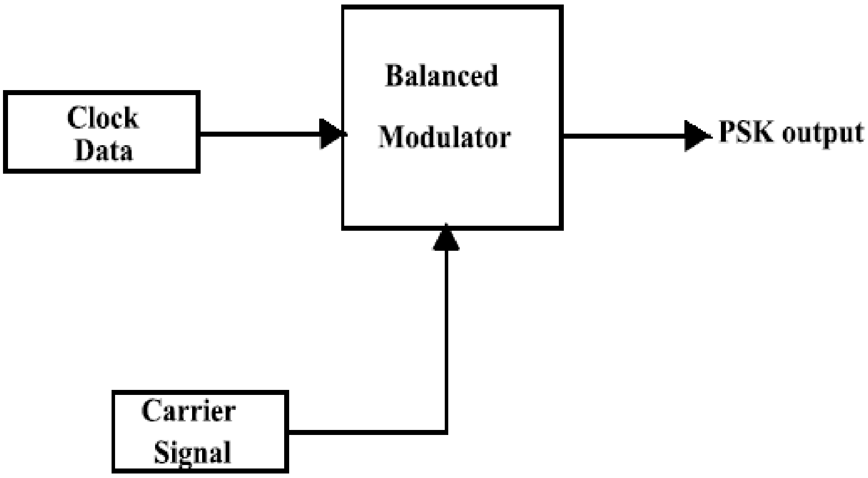
.

MATLAB PROGRAM FSK

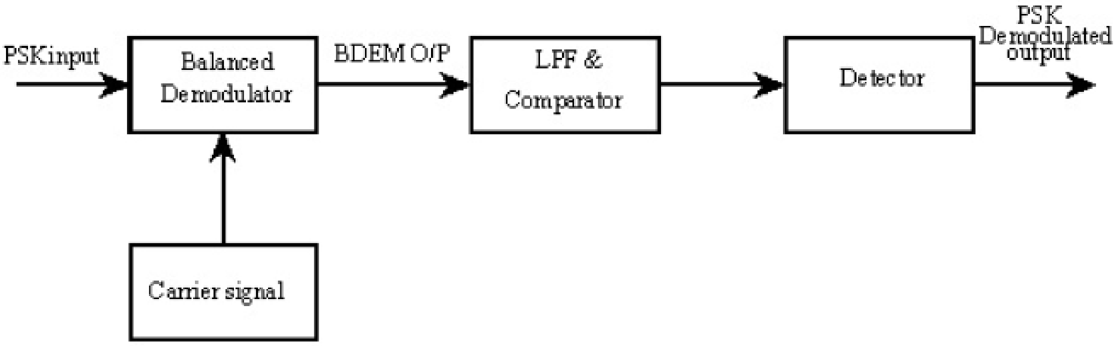
```
clear all;
close all;
X=input('enter the seq');
fs=input('enter the sampling freq');
fc=input('enter the carrier freq');
fd=input('enter the fd');
m=2;
y=MODMAP(X,fd,fs,'fsk',m);
subplot(3,1,1);
plot(y,'linewidth',1.5);
grid on;
xlabel('time');
ylabel('amplitude');
title('input seq');
[y,t]=dmod(X,fc,fd,fs,'fsk',m);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



BLOCK DIAGRAM OF PSK MODULATOR



BLOCK DIAGRAM OF PSK DEMODULATOR



PSK MODULATOR AND DEMODULATOR

THEORY

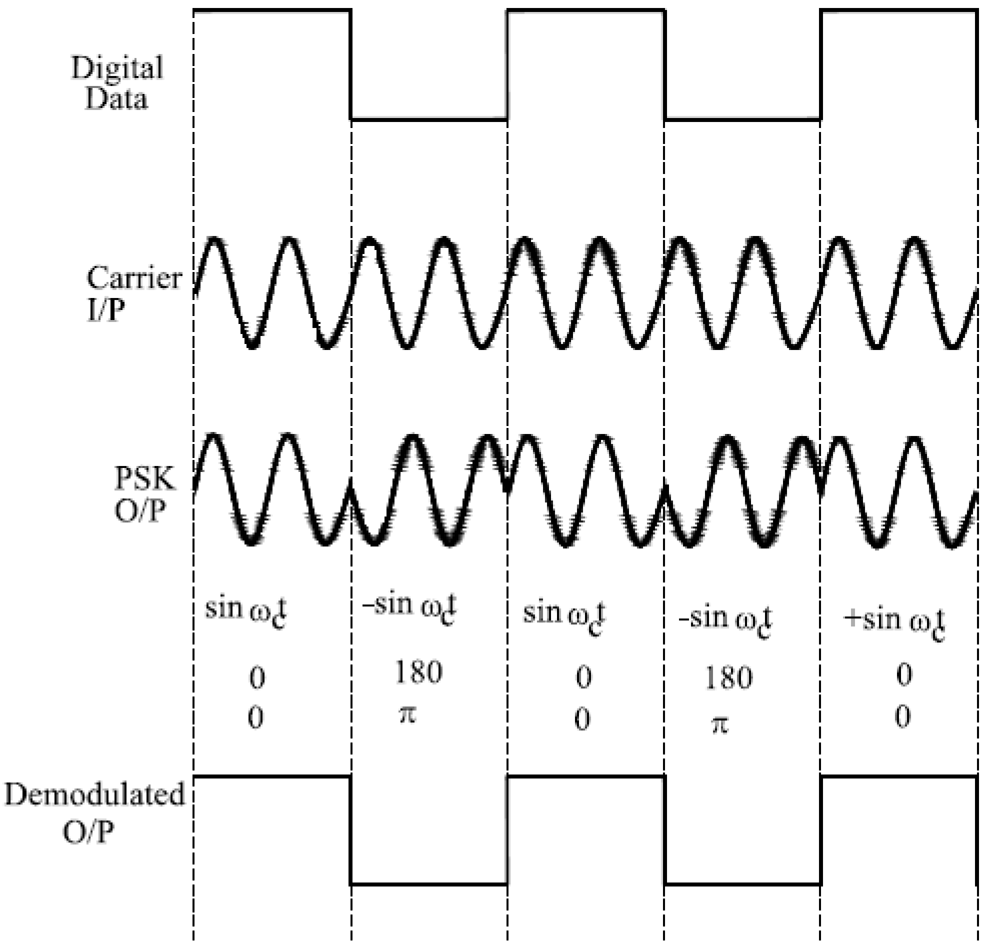
Communication is a process of conveying information from one place to other. Some of the examples for communication systems are face to face in meeting (or) conferences, often requiring travel, are increasing using „teleconferring". Similarly, teleshopping and telebanking will provide services by electronic 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 and1. 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

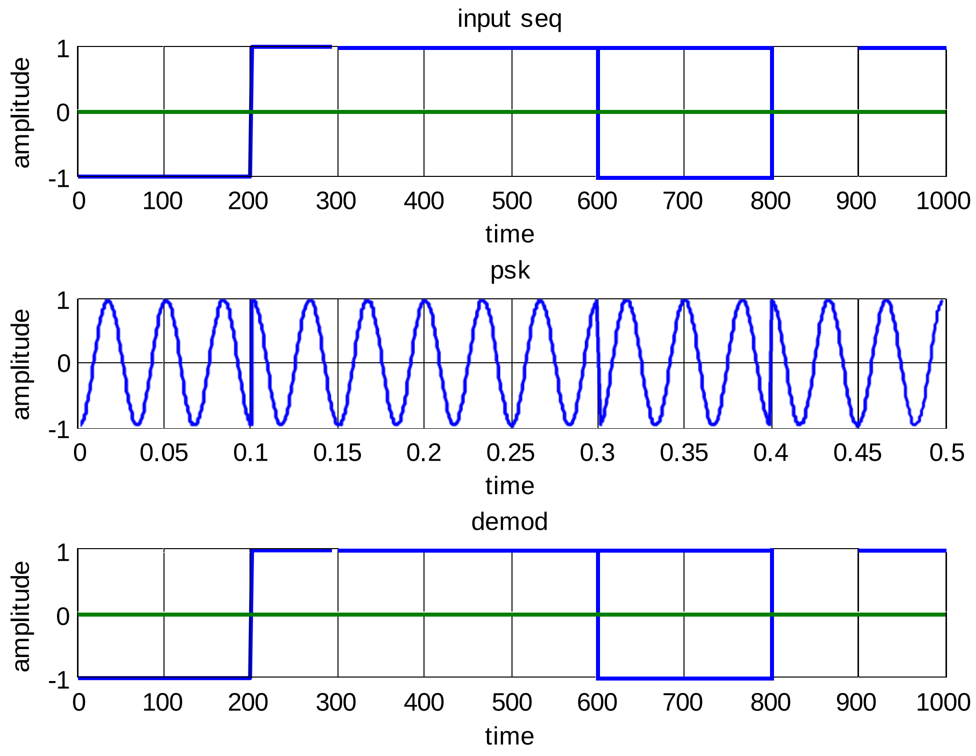


TABULATION

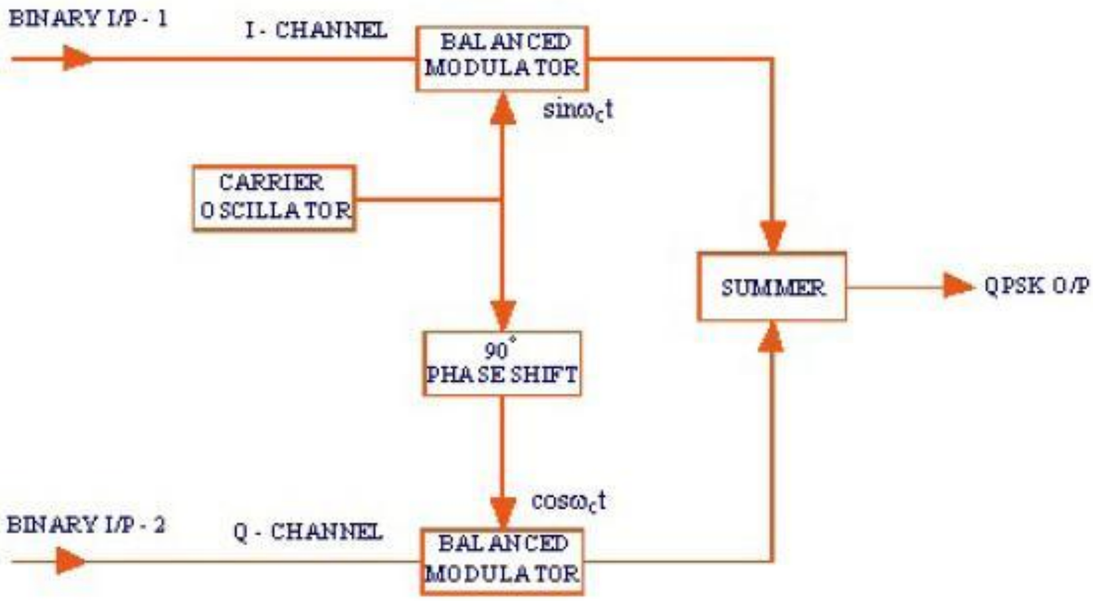
S.NO	SIGNAL	AMPLITUDE(V)	TIME(ms)	FREQUENCY(HZ)
1	Modulating Input			
2	Carrier signal			
3	PSK Output			
4	Demodulated output			

MATLAB PROGRAM PSK

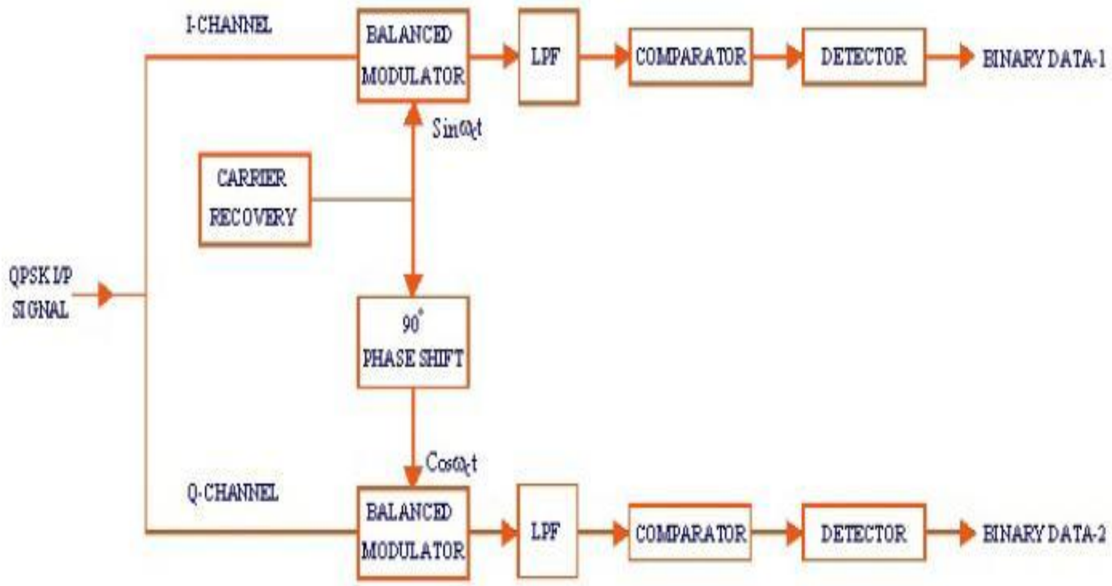
```
clear all;
close all;
X=input('enter the seq');
fs=input('enter the sampling freq');
fc=input('enter the carrier freq');
fd=input('enter the fd');
m=2;
y=MODMAP(X,fd,fs,'psk',m);
subplot(3,1,1);
plot(y,'linewidth',1.5);
grid on;
xlabel('time');
ylabel('amplitude');
title('input seq');
[y,t]=dmod(X,fc,fd,fs,'psk',m);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


input seq
amplitude
1
0
-1
0
100
200
300
400
500
600
700
800
900
1000
time
psk
amplitude
1
0
-1
0
0.05
0.1
0.15
0.2
0.25
0.3
0.35
0.4
0.45
0.5
time
demod
amplitude
1
0
-1
0
100
200
300
400
500
600
700
800
900
1000
time

BLOCK DIAGRAM OF QPSK MODULATION



BLOCK DIAGRAM OF QPSK DEMODULATION



9.QPSK MODULATOR & DEMODULATOR

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 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).

QPSK DEMODULATOR

The block diagram of a QPSK receiver is shown in Figure. The input QPSK signal given to the I and Q product detectors and the carrier recovery circuit. The carrier recovery circuit reproduces the original transmit carrier oscillator signal. The recovered carrier must be frequency and phase coherent with the transmit reference carrier. The QPSK signal is demodulated in the I and Q product detectors, which generate the original I and Q data bits. The output of the product detectors are fed to the bit combining circuit, where they are converted from parallel I and Q data channels to a single binary output data stream.

TABULATION

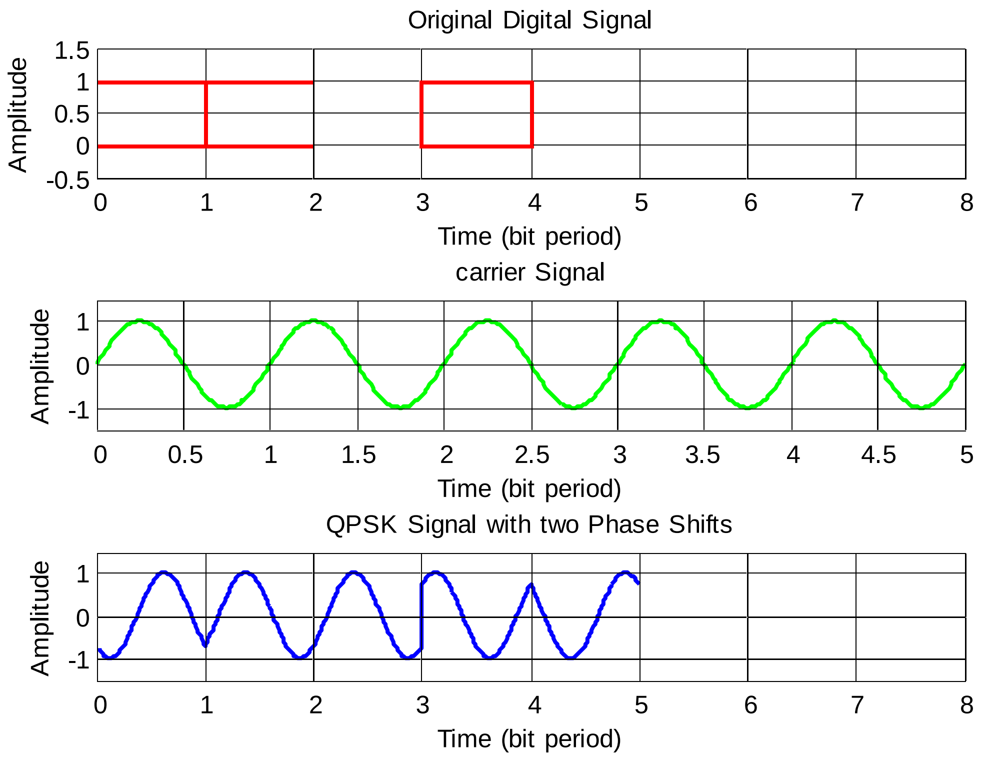
S.NO	SIGNAL	AMPLITUDE(V)	TIME(ms)	FREQUENCY(HZ)
1	Modulating Input1			
2	Modulating Input2			
3	Carrier signal1			
4	Carrier signal2			
5	QPSK Output			
6	Demodulated output1			
7	Demodulated output2			

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
P1 = pi/4; %45degrees phase shift
P2 = 3/4*pi; %135 degrees phase shift
P3 = 5/4*pi; %225 degree phase shift
P4 = 7/4*pi; %315 degree phase shift
% Frequency of Modulating
Signal f = 1; %f --> time
period
% Sampling rate of sine wave - This will define the resoultion
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 genearting QPSK signal modulated
signal if bit_stream(ii)==0
if bit_stream(jj)==0
bit00 = (bit_stream(ii)==0)*sin(2*pi*f*t + P1);
QPSK_signal = [QPSK_signal (bit00)];
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;
```



10 .MATLAB CODING FOR DIGITAL COMMUNICATION

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.

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

Structure of code word

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

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

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

$b = m \times P$

Where,

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

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 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
```

```
W_min=min(sum((C(2:2^k,:))'));

disp(W_min);
```

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.

11.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

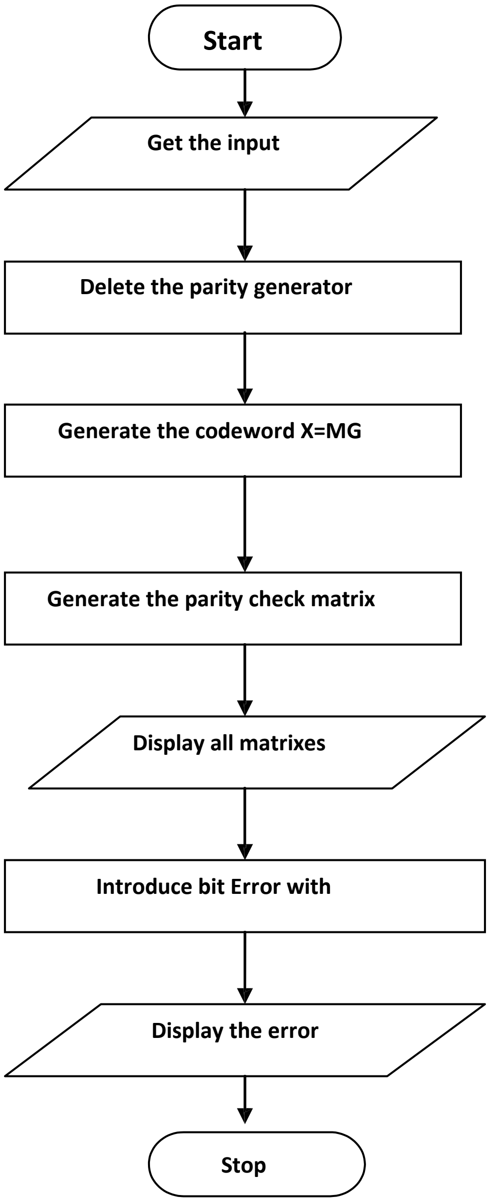
Decode the original message from the RX message

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:


Start
Get the input
Delete the parity generator
Generate the codeword X=MG
Generate the parity check matrix
Display all matrixes
Introduce bit Error with
Display the error
Stop

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);
```

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);
syndrome1=bi2de(syndrome,'left-msb');
errorvect=trt(1+syndrome1,:);
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
```

INFERENCE:

Cyclic codes posses a well defined mathematically structure, Which to the development of very efficient decoding scheme for them, Linear codes has capacity of the correcting and detecting end bit

RESULT:

Thus the simulation for cyclic code is done using MATLAB

12.SIMULATION OF OFDM TRANSCEIVER USING MATLAB

AIM

To simulate OFTM transceiver using MATLAB

APPARATUS REQUIRED

1. Personal computer
2. MATLAB software

THEORY

The transmitter first converts the input data from a serial stream to parallel sets. Each set of data contains one symbol, Si, for each subcarrier. Before performing the Inverse Fast Fourier Transform (IFFT), this example data set is arranged on the horizontal axis in the frequency domain as shown in Figure 2. This symmetrical arrangement about the vertical axis is necessary for using the IFFT to manipulate this data.

An inverse Fourier transform converts the frequency domain data set into samples of the corresponding time domain representation of this data. Specifically, the IFFT is useful for OFDM because it generates samples of a waveform with orthogonal frequency components.

Then, the parallel to serial block creates the OFDM signal by sequentially outputting the time domain samples. The channel simulation will allow examination of the effects of noise, multipath, and clipping. By adding random data to the transmitted signal, simple noise can be simulated. Multipath simulation involves adding attenuated and delayed copies of the transmitted signal to the original. This simulates the problem in wireless communication when the signal propagates on many paths. For example, a receiver may see a signal via a direct path as well as a path that bounces off a building. Finally, clipping simulates the problem of amplifier saturation. This addresses a practical implementation problem in OFDM where the peak to average power ratio is high.

The receiver performs the inverse of the transmitter. First, the OFDM data are split from a serial stream into parallel sets. The Fast Fourier Transform (FFT) converts the time domain samples back into a frequency domain representation. The magnitudes of the frequency components correspond to the original data. Finally, the parallel to serial block converts this parallel data into a serial stream to recover the original input data.

ALGORITHM:

TRANSMITTER

1. Initialize the size of the matrix
2. Convolute the data with encoding.
3. Interleave the encoded data
4. Convert binary to decimal
5. Modulate the data using QAM.
6. Pilot insertion using IFFT.
7. Add cyclic extension.
8. Calculate the value of SNR.

RECEIVER

1. Remove the cyclic extension
2. Take FFT
3. Pilot synchronization
4. Convert decimal to binary.
5. Do the interleave process.
6. Decode the data
7. Calculate the value of BER

Plot the values for BER vs SNR .

SIMULATION OF OFDM TRANSCEIVER USING MATLAB

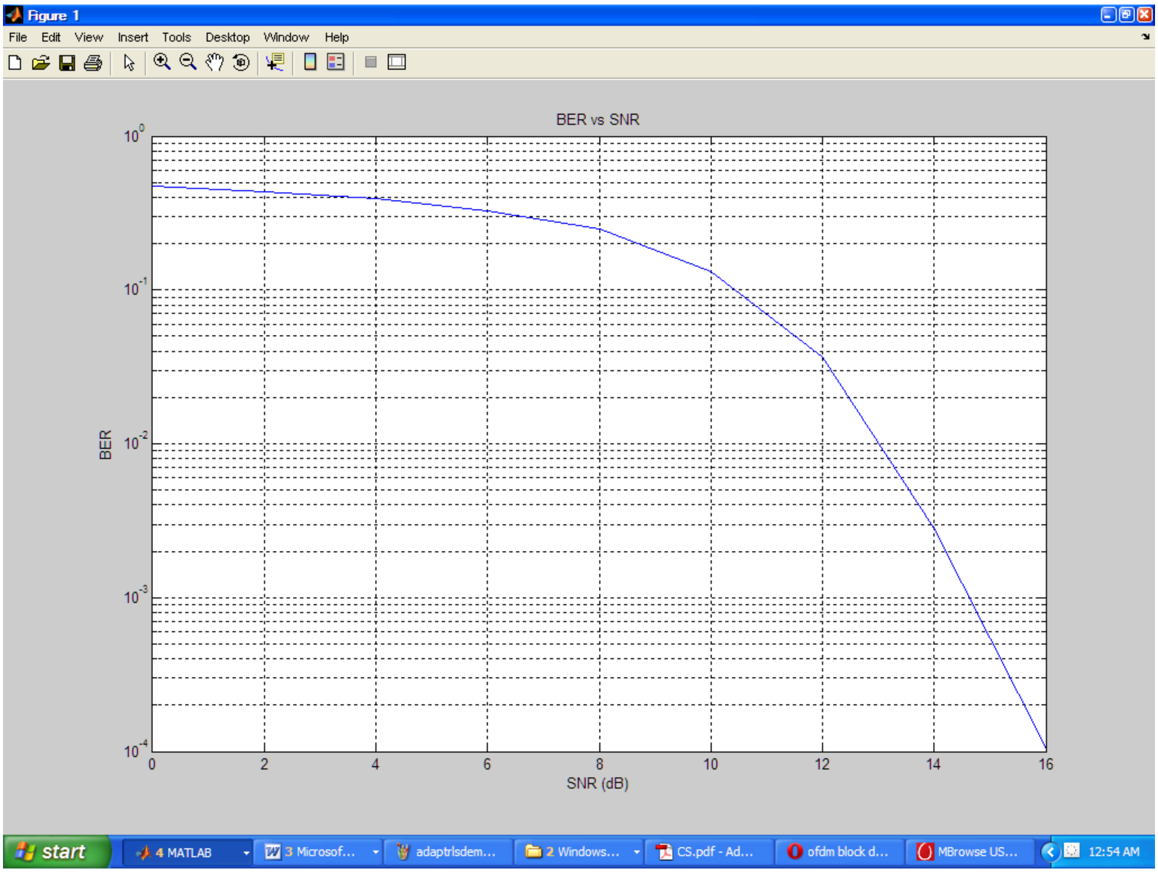
CODING

```
close all
clear all
clc
t_data=randint(9600,1)';
x=1;
si=1; %for BER rows
for d=1:100;
data=t_data(x:x+95);
x=x+96;
k=3;
n=6;
s1=size(data,2);  % Size of input matrix
j=s1/k;

% Convolutionally encoding data
constlen=7;
codegen = [171 133];    % Polynomial
trellis = poly2trellis(constlen, codegen);
codedata = convenc(data, trellis);


%Interleaving coded data

s2=size(codedata,2);
j=s2/4;
matrix=reshape(codedata,j,4);

intlvddata = matintrlv(matrix',2,2)'; % Interleave.
intlvddata=intlvddata';


% Binary to decimal conversion

dec=bi2de(intlvddata','left-msb');


%16-QAM Modulation
```

```
M=16;
y = qammod(dec,M);
% scatterplot(y);


% Pilot insertion

lendata=length(y);
pilt=3+3j;
nofpits=4;

k=1;

for i=(1:13:52)

    pilt_data1(i)=pilt;

    for j=(i+1:i+12);
        pilt_data1(j)=y(k);
        k=k+1;
    end
end

pilt_data1=pilt_data1';    % size of pilt_data =52
pilt_data(1:52)=pilt_data1(1:52);     % upsizing to 64
pilt_data(13:64)=pilt_data1(1:52);    % upsizing to 64

for i=1:52

    pilt_data(i+6)=pilt_data1(i);

end

% IFFT

ifft_sig=ifft(pilt_data',64);


% Adding Cyclic Extension

cext_data=zeros(80,1);
cext_data(1:16)=ifft_sig(49:64);
for i=1:64

    cext_data(i+16)=ifft_sig(i);

end


% Channel

 % SNR

 o=1;
for snr=0:2:50

ofdm_sig=awgn(cext_data,snr,'measured'); %AWGN
% figure;
% index=1:80;
% plot(index,cext_data,'b',index,ofdm_sig,'r'); %plot both
signals
```

```
% legend('Original Signal to be Transmitted','Signal with AWGN');


%                    RECEIVER
%Removing Cyclic Extension

for i=1:64

    rxed_sig(i)=ofdm_sig(i+16);

end


% FFT

ff_sig=fft(rxed_sig,64);

%%
% Pilot Synch%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%%


for i=1:52

    synched_sig1(i)=ff_sig(i+6);

end

k=1;

for i=(1:13:52)

    for j=(i+1:i+12);
        synched_sig(k)=synched_sig1(j);
        k=k+1;
    end
end

% scatterplot(synched_sig)


% Demodulation
dem_data= qamdemod(synched_sig,16);


% Decimal to binary conversion

bin=de2bi(dem_data','left-msb');
bin=bin';


% De-Interleaving


deintlvddata = matdeintrlv(bin,2,2); % De-Interleave
deintlvddata=deintlvddata';
deintlvddata=deintlvddata(:)';



%Decoding data
```

```
n=6;
k=3;
decodedata =vitdec(deintlvddata,trellis,5,'trunc','hard');  %
decoding datausing veterbi decoder
rxed_data=decodedata;

% Calculating BER
rxed_data=rxed_data(:)';
errors=0;


c=xor(data,rxed_data);
errors=nnz(c);



BER(si,o)=errors/length(data);
o=o+1;

 end % SNR loop ends here
 si=si+1;
end % main data loop

% Time averaging for optimum results

for col=1:25;        %%%change if SNR loop Changed
    ber(1,col)=0;
for row=1:100;
        ber(1,col)=ber(1,col)+BER(row,col);
    end
end
ber=ber./100;

%%
figure
i=0:2:48;
semilogy(i,ber);
title('BER vs SNR');
ylabel('BER');
xlabel('SNR (dB)');
grid on
```



RESULT

Thus the SIMULATION OF OFDM TRANSCEIVER was done using MATLAB Software.

13.DIGITAL DATA TRANSMISSION THROUGH FIBER OPTIC LINK

AIM:

To study fiber optic digital link at 850 nm and the relationship between the input signal and received signal.

EQUIPMENT REQUIRED:

1. DL-03 transmitter & DL-03 receiver.
2. Power supply
3. 20MHz dual channel oscilloscope
4. 10 MHz function generator
5. 1 meter glass fiber cable

THEORY:

Fiber optic links can be used for transmission of digital as well as analog signals. Basically a fiber optic link contains three main elements, A transmitter, an optical fiber & a receiver. The transmitter module takes the input signal in electrical form & then transforms it into optical (light) energy containing the same information. The optical fiber is the medium, which carries this energy to the receiver. At the receiver, light is converted back into electrical form with the same pattern a originally fed to the transmitter.

TRANSMITTER:

Fiber optic transmitters are typically composed of a buffer, drive & optical source. The buffer electronics provides both an electrical connection & isolation between the transmitter & the electrical system supplying the data. The driver electronics provides electrical power to the optical source in a fashion that duplicates the pattern of data being fed to the transmitter. Finally the optical source (LED) converts the electrical current to light energy with the same pattern. The LED supplied with this link operates outside the visible light spectrum.

Its optical output is centered at near infrared spectrum. Its optical output is centered at near infrared wavelength of 850nm. The emission spectrum is broad, so a faint red glow can be usually being seen when the LED is switched on in a dark room.

The LED 0PF322A used in the link is coupled to the transistor driver in common emitter mode. The driver is preceded by the digital buffer circuit. A TTL compatible digital signal can be applied to this buffer. Buffer circuit is nothing but NAND gate; transistor driver is switched between saturation and cutoff states on load line by this signal hence modulating current flowing through LED. This turns on LED when transistor is in cutoff state optical signal is then carried over by the optical fiber.

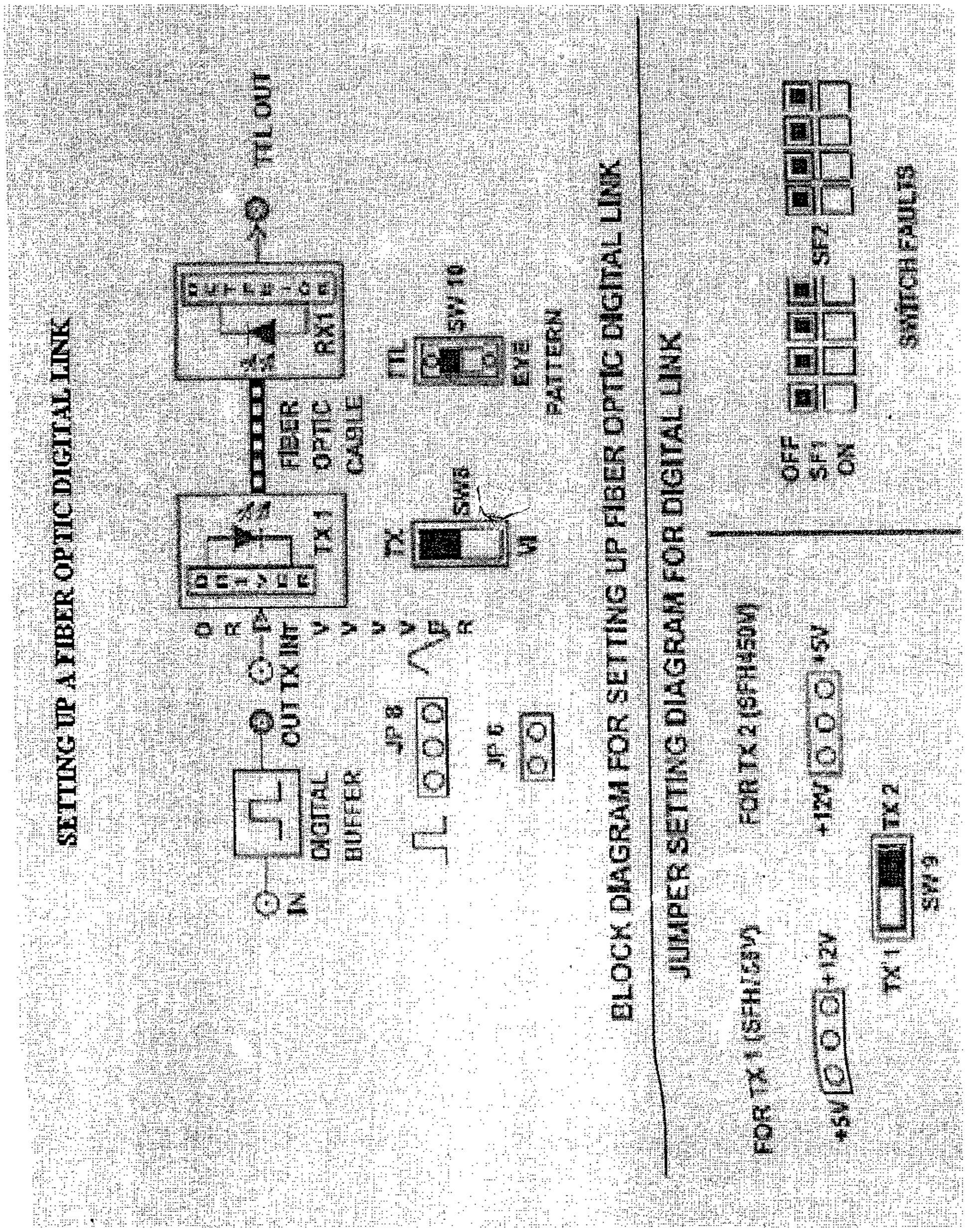
RECEIVER:

The function of the receiver is to convert the Optical energy into electrical form, which is then conditioned to reproduce the transmitted electrical signal in this original form. The detectors usually come in various types and one has to select proper detector depending on the nature of the application. The parameters usually considered in this case of detector it's responsively peak wavelength &

Response time. The detector used in this link has TTL type of output. This means that only two intensity levels of light are detected, presence of light or its absence.

PROCEDURE:

1. Insert the BT connector at one end of the fiber into the receptacle in which the LED is housed. Similarly, fit the ST connector with which the other end of the fiber has been terminated into the receptacle in which the PIN photodiode in housed.
2. Connect the power supply cables with proper polarity to kit. While connecting this, Ensure that the power supply is OFF. Now switch on the power supply.
3. Feed the TTL signal of about 1KHz square wave, to IN post of buffer selection.
4. Observe the received signal on CR0 at output post.
5. To measure the digital bandwidth of the link, vary the frequency of the input Signal from 100Hz onwards and observe the effect on receiver signal



Transmitter side:

Amplitude in (V)	Frequency in (KHz)	Time period (ms)

Receiver side:

Amplitude in (V)	Frequency in (KHz)	Time period (ms)

RESULT:

Thus the digital data transmission using optical fiber was performed.