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DEPARTMENT OF ELECTRONICS AND COMMUNICATION ENGINEERING

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EC3461- COMMUNICATION SYSTEMS LABORATORY MANUAL

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EC3461- COMMUNICATION SYSTEMS LABORATORY SYLLABUS

COURSE OBJECTIVES : **L T P C**
0 0 3 1.5

- To study the AM & FM Modulation and Demodulation.
- To learn and realize the effects of sampling and TDM.
- To understand the PCM & Digital Modulation.
- To Simulate Digital Modulation Schemes.
- To Implement Equalization Algorithms and Error Control Coding Schemes.

LIST OF EXPERIMENTS

1. AM- Modulator and Demodulator
2. FM - Modulator and Demodulator
3. Pre-Emphasis and De-Emphasis.
4. Signal sampling and TDM.
5. Pulse Code Modulation and Demodulation.
6. Pulse Amplitude Modulation and Demodulation.
7. Pulse Position Modulation and Demodulation and Pulse Width Modulation and Demodulation.
8. Digital Modulation – ASK, PSK, FSK.
9. Delta Modulation and Demodulation.
10. Simulation of ASK, FSK, and BPSK Generation and Detection Schemes.
11. Simulation of DPSK, QPSK and QAM Generation and Detection Schemes.
12. Simulation of Linear Block and Cyclic Error Control coding Schemes.

TOTAL: 45 PERIODS

COURSE OUTCOMES:

At the end of the laboratory course, the student will be able to understand the:
CO1:Design AM, FM & Digital Modulators for specific applications.
CO2:Compute the sampling frequency for digital modulation**.**
CO3:Simulate & validate the various functional modules of Communication system.
CO4:Demonstrate their knowledge in base band signaling schemes through implementation of digital modulation schemes.
CO5:Apply various channel coding schemes & demonstrate their capabilities towards the improvement of the noise performance of Communication system.

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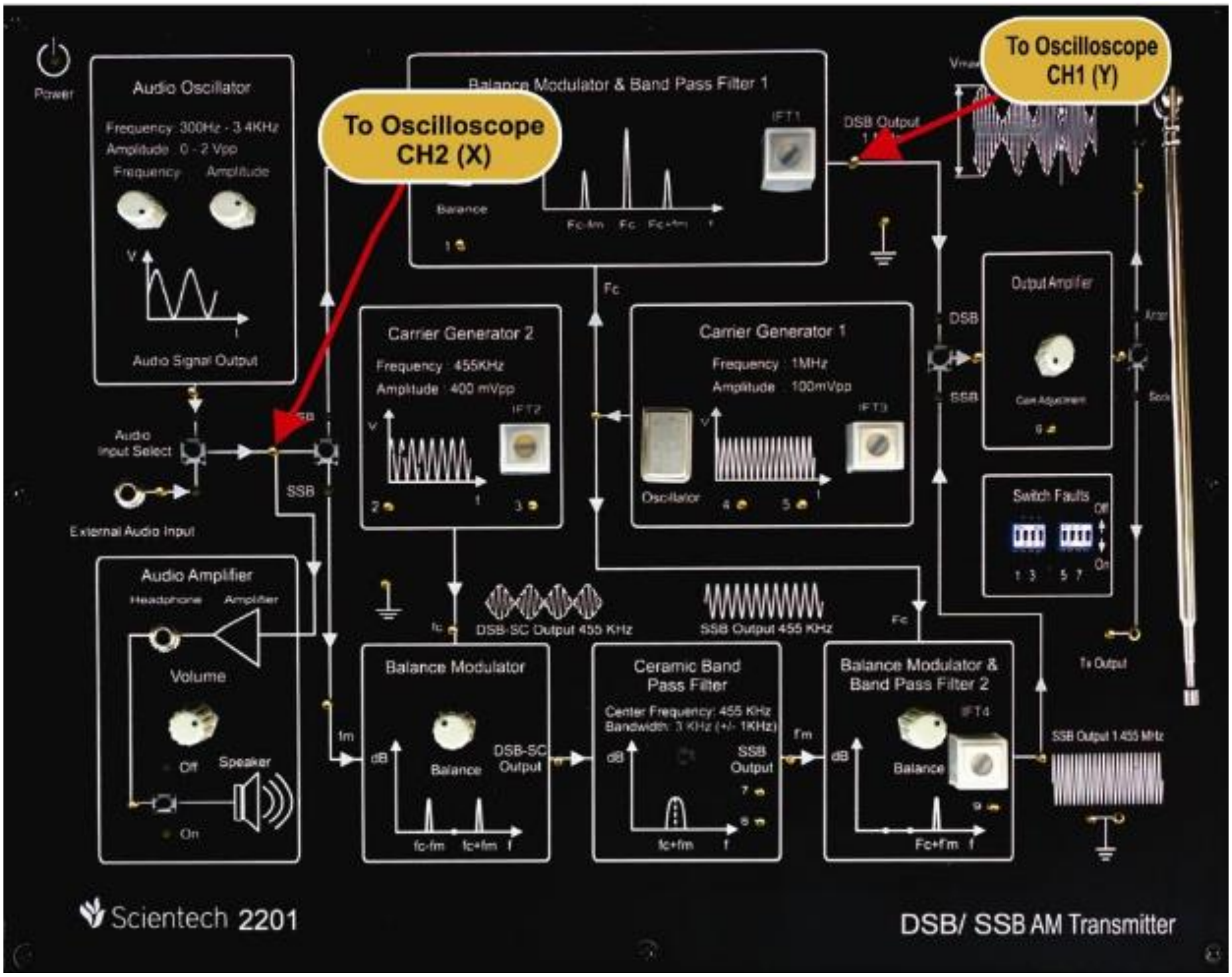
CIRCUIT DIAGRAM- Modulation



EX .NO: 1
DATE:

AM- Modulator and Demodulator

AIM:

To design and verify the characteristics of a amplitude modulator and a corresponding demodulator circuit.

EQUIPMENTS REQUIRED:

Equipments	Range	Quality
AM transmitter andReceiver Kit	-	1
CRO	-	1
Batch Cards	-	As Required
Probes		

THEORY:

Modulation is the process of changing the carrier signal amplitude or frequency or phase value in accordance with the message signal to be transmitted. Various modulation methods have been developed for transmission of signals as effectively as possible, with minimum possible distortion. The comparison of the effectiveness of these modulation methods may be based upon the signal power to noise power measured at the output of the receiver.

In AM transmission, the carrier signal is modulator, so that its amplitude various with the changing amplitude of the modulating signal. Frequency and phase of the carrier signal remains the same, only the amplitude changes to follow variation in the information. The BW of AM signal is equal to twice the BW of the modulating signal and cause a range between central values. The carrier frequency, the BW of an audio signal is usually 5 KHz; therefore on AM station needs a minimum BW of 10 KHz. The demodulator signal is taken across the capacitor.

PROCEDURE-MODULATION :

Step 1.Ensure that the following initial conditions exist on the board.

a. Audio input select switch should be in INT position to select onboard generated audio signal as a modulating signal.

b.Mode switch in DSB position to connect the DSB signal to Output Amplifier section.

c. Output amplifier's gain potentiometer in full clockwise position for maximum amplification.

d. Speakers switch in OFF position.

Step 2. Turn on power to the Scientech 2201 board.

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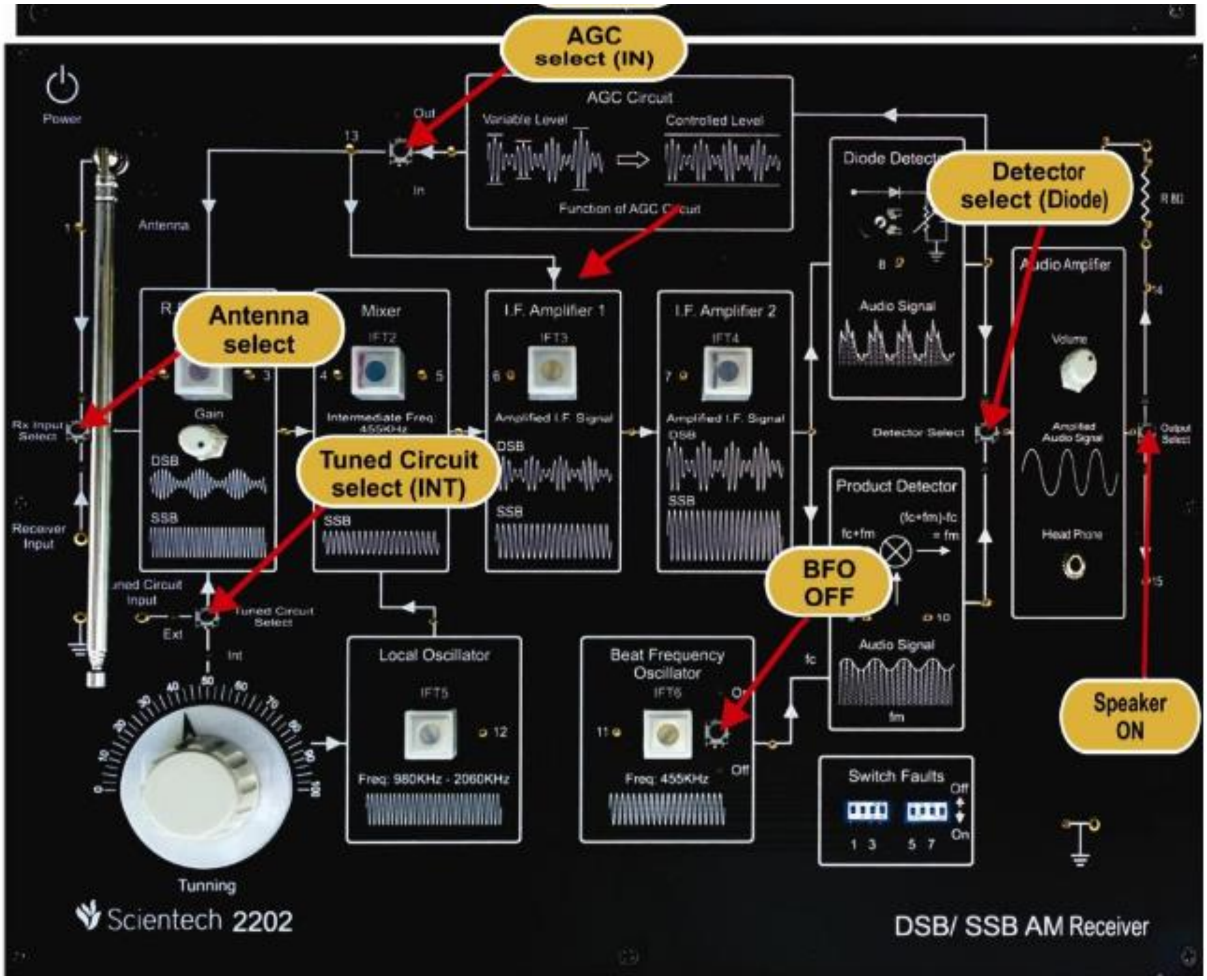
CIRCUIT DIAGRAM- DeModulation



Step 3. Set the amplitude and frequency of the audio signal to its maximum position using the respective Amplitude and Frequency control pots. This is the audio frequency sine wave is the modulating signal input to Balanced Modulator and Band Pass Circuit 1 with 1 MHz carrier input from 1 MHz Crystal oscillator block.

Step 4. Balanced Modulator and Band Pass Filter Circuit 1 generate 'Double Side Band Amplitude Modulation'. Balance pot is used to vary the depth of modulation AM waveform. Initially turn the pot to its maximumposition. The output from the balanced modulator & band pass filter circuit 1 block is a Double Sideband. AM waveform, which has been formed by amplitude - modulating the 1MHz carrier sine wave with the audio -frequency sine wave from the audio oscillator.

Step 5. To determine the depth of modulation, measure the maximum amplitude (Vmax) and the minimum amplitude (V min) of the AM waveform, and use the followingformula:

Percentage Modulation = (Vmax–Vmin / Vmax +Vmin)X 100%

Step 6.Now connect modulated waveform to the CH1(Y) input of the Oscilloscope and the modulating audio signal to the CH2(X) input of the Oscilloscope.

Step 7 Now vary the frequency of the modulating audio signal by varying the frequency pot of audio oscillator block and observe the effect on AM waveform. The frequency of envelop also varies with respect to the modulating audio signal frequency.

Step 8

Now vary the amplitude of the modulating audio signal by varying the amplitude pot in the audio oscillator block and observe the effect on AM waveform. The amplitude of two sidebands can be reduced to zero by reducing the amplitude of the modulating audio signal to zero.

PROCEDURE-DEMODULATION

This experiment investigates the reception and demodulation of AM waveforms by the Scientech 2201/ Scientech 2202 module. Both AM broadcast signals, and AM transmissions from Scientech 2201, will be examined, and the operation of automatic gain control at the receiver will be investigated.

Step 9.Position the Scientech 2201 & Scientech 2202 modules, with the Scientech 2201 board on the left, and a gap of about three inches between them.

Step 10.Ensure that the following initial conditions exist on the Scientech 2201 board.

a.Audio oscillator's amplitude pot in fully clockwise position.

b.Audio input select switch in INT position.

c. Balance pot in balanced modulator & band pass filter circuit 1 block, in full clockwise position;

d. Mode switch in DSB position.

e.Output amplifier's gain pot in full clockwise position.

f.TX output select switch in ANT position:

g.Audio amplifier's volume pot in fully counter-clockwise position.

h. 'Speaker' switch in ON position.

i. On -board antenna in vertical position, and fully extended.

Step 11. Ensure that the following initial conditions exist on the Scientech 2202 board:

a.RX input select switch in ANT position.

b.R.F. amplifier's tuned circuit select switch in 'INT' position.

c.R.F. amplifier's gain pot in fully clock-wise position;

d. AGC switch in 'IN' position.

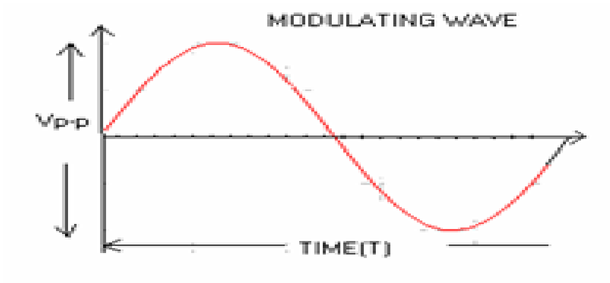
e.Detector switch in 'Diode'position.

f. Audio amplifier's volume pot in fully counter-clockwise position.

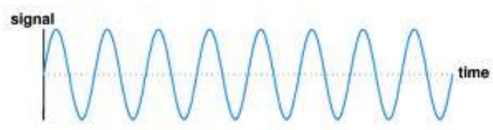
g.'Speaker' switch in ON position.

Model Graph:

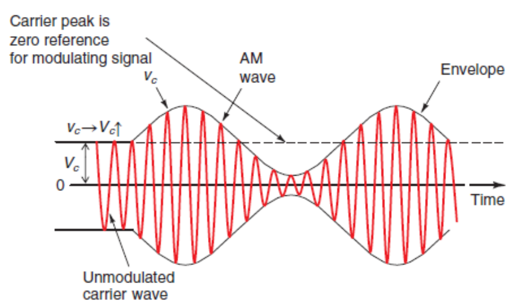
Input Signal



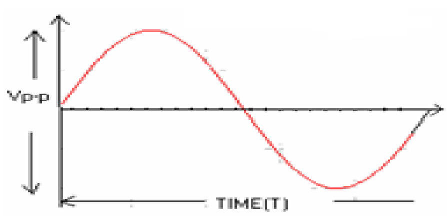
Carrier Signal:



Amplitude Modulated Signal:



Demodulated Signal:



TABULATION:

Signal	Amplitude(volts)	Time(ms)
Carrier		
Message		
Modulated O/P	Emax= Emin=	
Demodulated O/P		

h.Beat frequency oscillator switch in OFF position.

i. On - board antenna in vertical position, and fully extended.

Step 12 Turn on power to the modules.

Step 13 On the Scientech 2201 module, examine the transmitter's output signal, together with the audio modulating Audio signal. [CH1(Y) – 1V; CH2(X) – 0.2V Time base – 0.1 mS]

Since Scientech 2201 TX output select switch is in the ANT position, the AM signal at the output is fed to the transmitter's antenna. Prove this by touching Scientech 2201's antenna, and nothing that the loading caused by your hand reduces the amplitude of the AM waveform at the output.

Step 14 On the Scientech 2201 module, turn the volume pot (in the audio amplifier block) clockwise, until you can hear the tone of the audio oscillator's output signal, from the loudspeaker on the board.

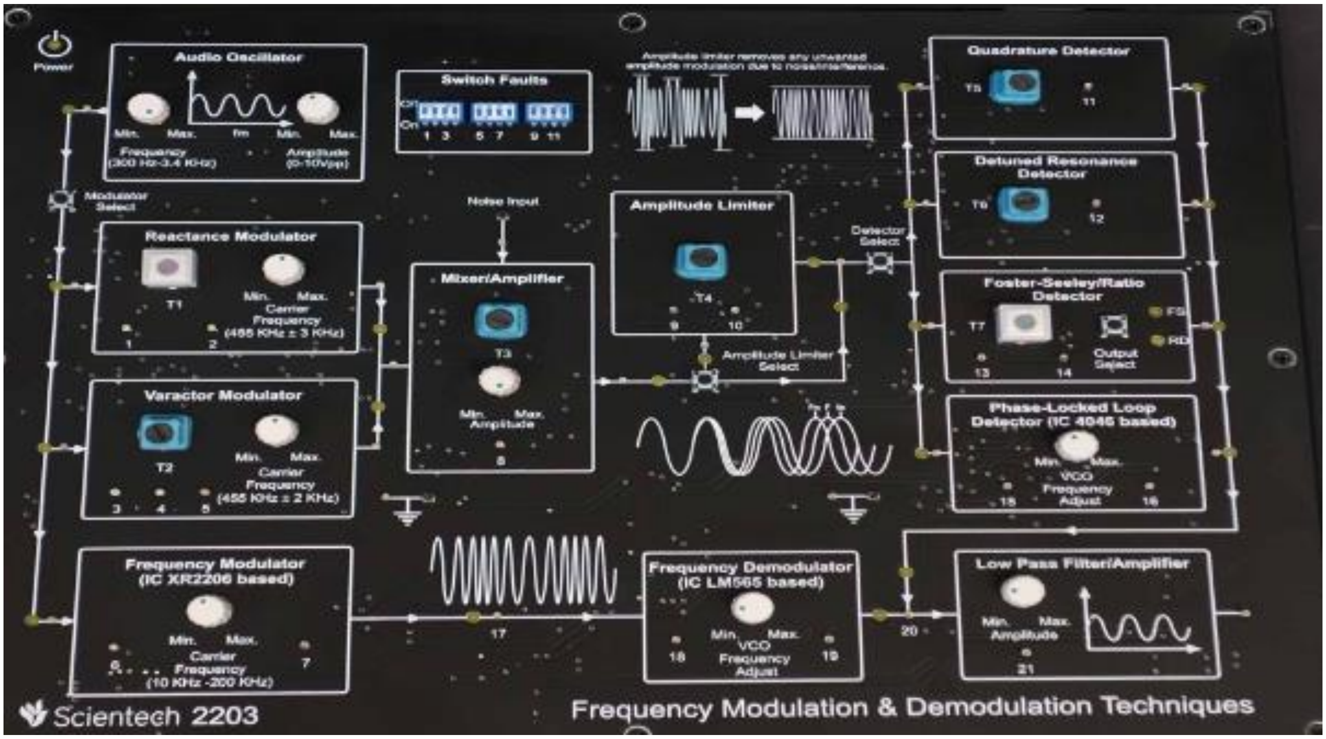
Step 15 On the Scientech 2202 receiver, adjust the volume pot so that the receiver's output can be clearly heard. Then adjust the receiver's tuning dial until the tone generated at the transmitter is also clearly audible at the receiver (this should be when the tuning dial is set to about 55-65 and adjust the receiver's volume pot until the tone is at a comfortable level.

Step 16 Check that you are tuned into the transmitter's output signal, by varying Scientech 2201's frequency pot in the audio oscillator block, and nothing that the tone generated by the receiver changes. [CH1(Y) – 1V; CH2(X) – 2V Time base – 0.1 mS]

RESULT:

Thus the characteristics of amplitude modulation and demodulation circuits are studied and the waveforms are observed and plotted.

CIRCUIT DIAGRAM



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EX: NO: 2
DATE:

FM - Modulator and Demodulator

AIM:

To design and verify the characteristics of a frequency modulator and the corresponding demodulator circuit.

EQUIPMENTS REQUIRED:

Equipments	Range	Quality
FM transmitter andReceiver Kit	-	1
CRO	-	1
Batch Cards	-	As Required
Probes		

THEORY:

Frequency modulation is the phenomenon in which amplitude is constant but the frequency will be design changed or varied accordance with the incoming message signal.FM is the process that enables us to recovers the original signal from a frequency modulated signal.

The objective is to produce a transfer characteristics that is the inverse of frequency modulator, which can be realized by a direct method of frequency demodulation involves the use of a popular device known as frequency discriminator whose instantaneous frequency of input FM signal.

Frequency modulation is the process of varying the frequency of a carrier wave in proportion to the instantaneous amplitude of the modulating signal without any variation in the amplitude of the carrier wave. Because the amplitude of the wave remains unchanged, the power associated with an FM wave is constant.

The FM wave as a matter of fact, contains an infinte number of side bands, each side band is separated from the next by fm . However , out of these , there are only a few side bands which carry

PROCEDURE

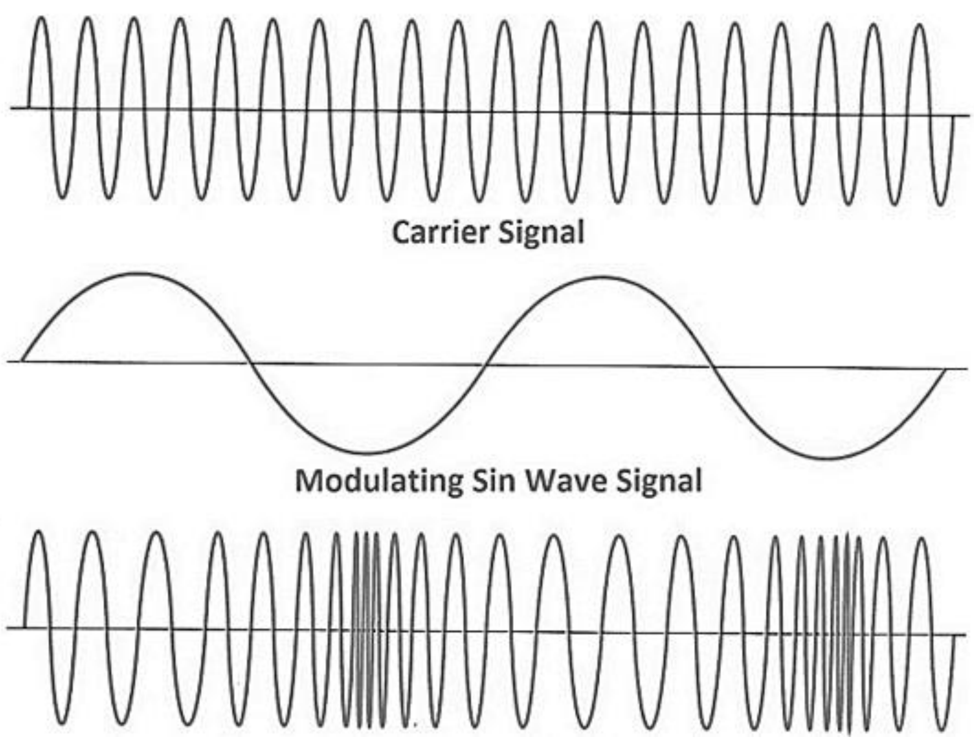
Step 1 Ensure that the following initial conditions exist on the ST2203 board.
(a)All switched faults off.
(b)Amplitude pot (in mixer amplifier block) in fully clockwise position.
(c)VCO switch in 'ON' position.
Step 2 Turn the audio oscillator block's amplitude pot to its fully clockwise position, and examine the block's output t.p.1 on an oscilloscope. This is the audio frequency sine wave, which will be used as our modulating signal. Note that the sine wave's frequency can be adjusted from about 300Hz to approximately 3.4 KHz, by adjusting the audio oscillator's frequency pot.
Step 3 Connect the output of audio oscillator to VCO section’s MOD In socket.
Step 4 Turn ON the power supply.
Step 5 Observe the modulating signal and modulated output at the VCO’s MOD OUT socket by using CRO.

TABULATION:

Signal	Amplitude (v)	Time period (ms)
Carrier signal		
Message signal		
Modulated signal		
Demodulated signal		

MODEL GRAPH:

WAVE FORMS OBSERVED:-



Step 6 Calculate mf = δ / fm.
Step 7 Vary the modulating frequency keeping carrier frequency constant and repeat steps 3 & 4.
Step 8 Vary the carrier frequency keeping modulator frequency constant and repeat steps 3 & 4.
Step 9 Tabulate the results.

PROCEDURE :- FM Detection using PLL:
Step 1 Ensure that the following initial conditions exist on the ST2203 module:
(a)All switched faults off.
(b)Audio amplifier block's amplitude pot in fully clockwise (MAX) position.
(c) Audio amplifier block's frequency pot in fully counter clockwise. Ensure that the following initial conditions exist on the ST2203 clockwise (MIN) position.
(d) Amplitude pot (in the mixer/amplifier block) in fully clockwise position.
(e) VCO switch (in phase-locked loop detector block) in ON position.
Step 2 Make the connections shown in figure 4.
Step 3 Turn on power to the ST2203 module.
Step 4 Now monitor the audio input signal to the varactor modulator block (at t.p.14) together with the output from the phase-locked loop detector block (at t.p.60), triggering the oscilloscope in t.p.14. The signal at t.p.68 should contain three components:
- A positive D.C. offset voltage.
- A sine wave at the same frequency as the audio signal at t.p.14.
- A high - frequency ripple component.

Step 5 The low pass filter/amplifier block strongly attenuates the high-frequency ripple component at the detector's output and also blocks the D.C. offset voltage. Consequently the signal at the output of the low-pass filter/amplifier block (at t.p.73) should be very closely resemble the original audio making signal, if not then slowly adjust the freq. adjust pot of PLL block.
Step 6 Adjust the audio oscillator block's amplitude and frequency pots, and compare the original audio signal with the final demodulated signal.
FM Detection using Foster-Seelay Detector:
Step 1 Ensure that the following initial conditions exist on the ST2203 module:
(a)All switched faults OFF;
(b)Audio amplifier block's amplitude pot in fully clockwise (MAX) position.
(c)Audio amplifier block's frequency pot in fully counter-clockwise (MIN) position.
(d)Amplitude pot (in the mixer/amplifier block) in fully clockwise position.
(e)VCO switch (in phase-locked loop detector block) in OFF position.
Step 2 Make connection as shown in figure 5.
Step 3 Turn on power to the ST2203 module.
Step 4 We will now investigate the operation of the foster-Seeley detector on the ST2203 module. In the Foster-Seeley / ratio detector block, select the Foster-Seeley detector by putting the switch in the Foster-Seeley position.
Step 5 Initially, we will use the varactor modulator to generate our FM signal, since this is the more linear of the two modulators, as fast as its frequency/voltage characteristic is concerned. To select the varactor modulator, put the reactance/ varactor switch in the varactor position. Ensure that the varactor modulator's carrier frequency pot is in the midway position.
Step 6 The audio oscillator's output signal (which appears at t.p.1) is now being used by the varactor modulator, to frequency-modulate a 455Khz carrier sine wave. As we saw earlier, this FM waveform appears at the FM output socket from the mixer/amplifier block. You will probably need to have an X-

expansion control on your oscilloscope.

Step 7 Now monitor the audio input signal to the varactor modulator block (at t.p. 14) together with the foster-seeley output from the foster-seeley/ratio detector block (at t.p. 52), triggering the oscilloscope on t.p. 14. The signal at t.p. 52 should contain two components:

- A sine wave at the same frequency as the audio signal at t.p. 14.
- A High frequency ripple component of small amplitude.

Step 8 The low-pass filter/amplifier strongly attenuates this high-frequency ripple component, and blocks any small D.C. offset voltage that might exist at the detector's output. Consequently, the signal at the output of the low-pass filter/ amplifier block (at t.p. 73) should very closely resemble the original audio modulating signal.

Step 9 Monitor the audio input to the varactor modulator (at t.p. 14) and the output of the low pass filter / amplifier block (at t.p. 73) and adjust the gain pot (in the low pass filter/ amplifier block) until the amplitudes of the monitored audio waveforms are the same.

Step 10 Adjust the audio oscillator block's amplitude and frequency pots, and compare the original audio signal with the final demodulated signal.

RESULT:

Thus the frequency modulation and demodulation characteristics was designed and verified successfully.

EX: NO: 3
DATE:

PRE EMPHASIS &DE-EMPHASIS

AIM:

To construct and verify pre-emphasis and de-emphasis network and plot the waveform.

APPARATUS REQUIRED:

Equipments	Range	Quality
Pre Emphasis &De-Emphasis Kit	-	1
CRO	-	1
Batch Cards	-	As Required
Probes		

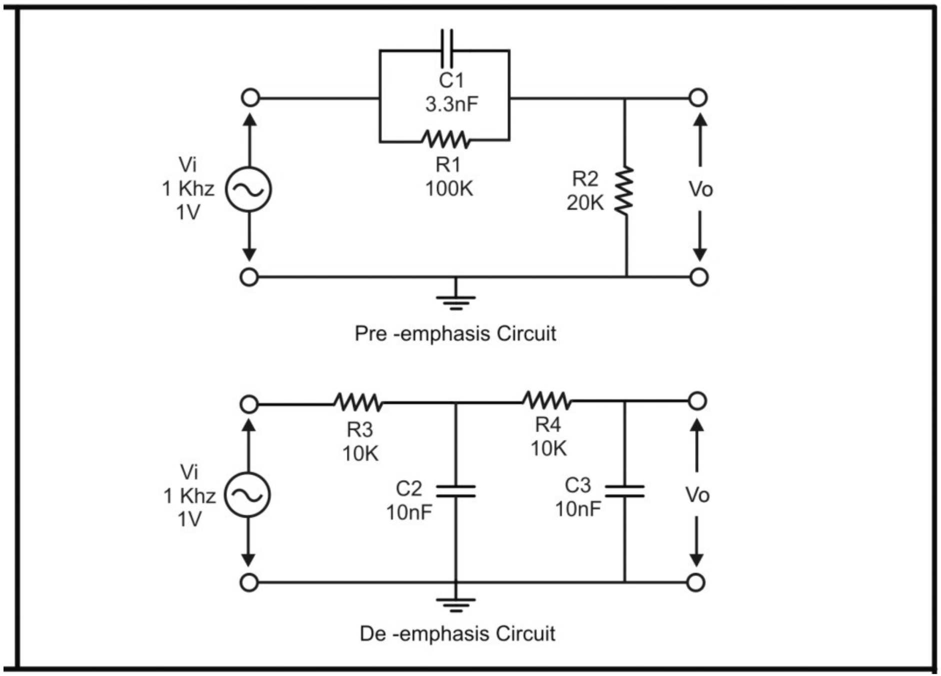
THEORY:

In processing electronic audio signals, pre-emphasis refers to a system process designed to increase, within a band of frequencies, the magnitude of some (usually higher) frequencies with respect to the magnitude of other (usually lower) frequencies in order to improve the overall signal-to-noise ratio by minimizing the adverse effects of such phenomena as attenuation distortion or saturation of recording media in subsequent parts of the system.

Pre-emphasis is commonly used in telecommunications, digital audio recording, record cutting, in FM broadcasting transmissions, and in displaying the spectrograms of speech signals.

In telecommunication, de-emphasis is a system process designed to decrease, within a band of frequencies, the magnitude of some (usually higher) frequencies with respect to the magnitude of other (usually lower) frequencies in order to improve the overall signal-to-noise ratio by minimizing the adverse effects of such phenomena as attenuation differences or saturation of recording media in subsequent parts of the system.

Pre Emphasis &De-Emphasis Circuit Diagram

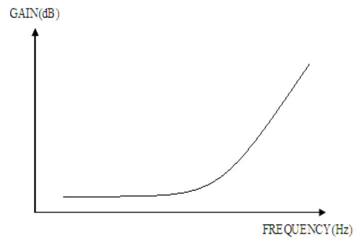


TABULATION for Pre Emphasis **:**

V_i =

Frequency(Hz)	Output voltage(V_0 volts)	Gain (dB)=20 log (V_0/V_i)

MODEL GRAPH:

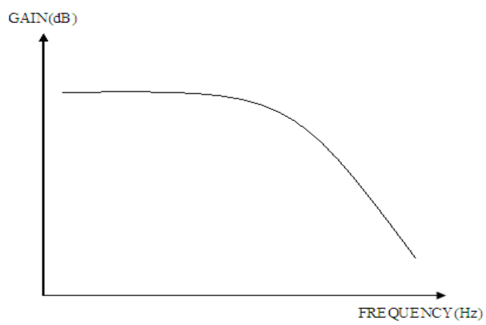


TABULATION for De-Emphasis:

Vi=

Frequency(Hz)	Output voltage(V_0 volts)	Gain (dB)=20 log (V_0/V_i)

MODEL GRAPH:



In serial data transmission, de-emphasis has a different meaning, which is to reduce the level of all bits except the first one after a transition. That causes the high frequency content due to the transition to be emphasized compared to the low frequency content which is de-emphasized. This is a form of transmitter equalization; it compensates for losses over the channel which is larger at higher frequencies.

PROCEDURE:

1. Connections are made as per the circuit diagram.
2. Set input signal amplitude using function generator.
3. Vary the input signal frequency from 0Hz to 100kHz in regular steps.
4. Note down the corresponding output voltage.
5. Plot the graph: Gain(dB) vs Frequency(Hz).

RESULT:

Thus Pre- emphasis and de-emphasis circuit is constructed and verified.

EX .NO: 4a
DATE:

SAMPLING AND HOLDING

AIM:

To study and verify the signal sampling and reconstruction unit.

APPARATUS REQUIRED:

Equipments	Range	Quality
Sampling Kit	-	1
CRO	-	1
Batch Cards	-	As Required
Probes		

THEORY:

Sampling is the process within converts the analog signal into digital signal. This process may involve three immediate steps known as sampling, Quantizing and decoding. Sampling is the process of converting continuous time signal into discrete time signal. It is very convenient to represent the sampling operation rate and of at the different sampling rate and the samples of the analog signal are being transmitted during the sampling period. The switch classes for a very short internal between the successive samples is‘t’ seconds and the sampling frequency is given by,

$$F = 1/T \text{ Hz}$$

The discrete time signal obtained by sampling is processed and then converted into continuous time signal. According to sampling theorem, the sampling frequency must be at least twice the highest frequency present in the input signal. I.e. f= 2fm. To avoid aliasing, very high sampling frequency is often choosed, but sampling at very high frequency introduces numerical error.

The output of antialiasing filter is fed to sample and hold circuit. If samples the analog input signal at uniform intervals and hold circuit allows ADC to operate slowing at step by step.

Procedure:

A. Setup for sampling and Reconstruction signal

Initial set up of trainer:

1. Duty cycle selector switch position: Position 5
2. Sampling selector switch: Internal position
3. Connect the power cord to the trainer. Keep the power switch in ‘Off’ position.
4. Connect 1 KHz Sine wave to signal Input as shown in figure 1.1.
5. Switch‘On’the trainer's power supply & Oscilloscope.
6. Connect BNC connector to the CRO and to the trainer’s output port.
7. You can observe the process of step-by-step generating sine waveConnect the sample and hold output to square wave of 1 KHz at TP3, TP4, TP5 and TP6

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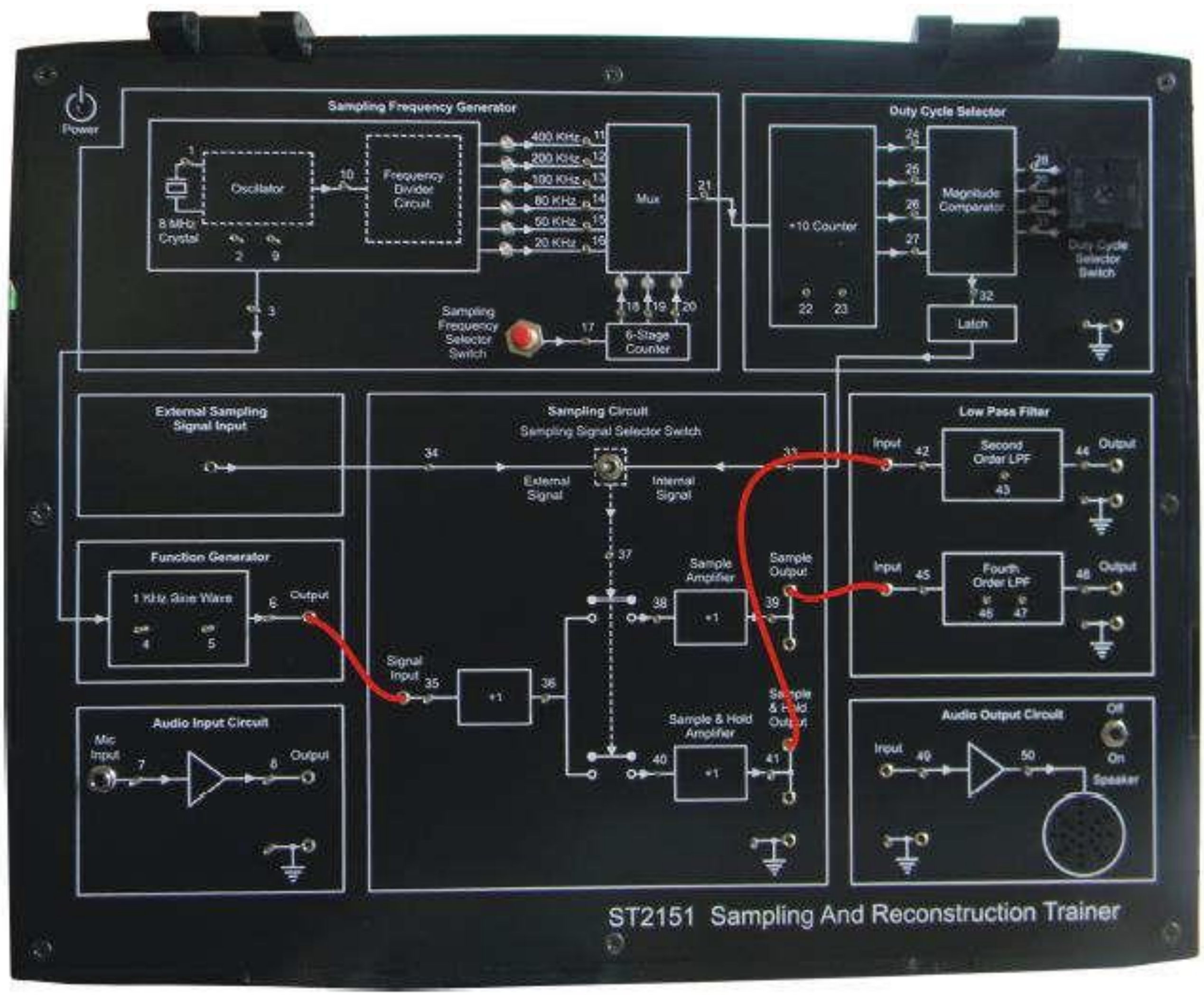


B. Setup for effect sample amplifier and sample and hold Amplifier on reconstruction filter

Initial set up of trainer:

1. Duty cycle selector switch position: Position 5
2. Sampling selector switch: Internal position
3. Connect the power cord to the trainer. Keep the power switch in 'Off'position.
4. Connect 1 KHz Sine wave to signal Input.
5. Switch 'On'the trainer's power supply & Oscilloscope.
6. Connect BNC connector to the CRO and to the trainer's output port.
7. Select sampling frequency of 8 KHz by Sampling Frequency Selector Switch pressed till 80 KHz signal LED glows.
8. Observe 1 KHz sine wave and Sample Output (TP39) on oscilloscope. The display shows 1 KHz sine wave being sampled at 8 KHz, so there are 8 samples for every cycle of the sine wave.
9. Connect Sample Output to Fourth Order low pass filter Input as shown in figure 1.2 Observe the filtered output (TP48) on the oscilloscope. The display shows the reconstructed 1 KHz sine wave
10. Similarly observe the sampled 1 KHz sine wave at and Sample and Hold Output (TP41) on oscilloscope. The display shows 1 KHz sine wave being sampled and hold signal at 8 KHz.
11. Connect Sample and Hold Output to Second Order low pass filter Input and observe the filtered output (TP44) on oscilloscope. The display shows the reconstructed

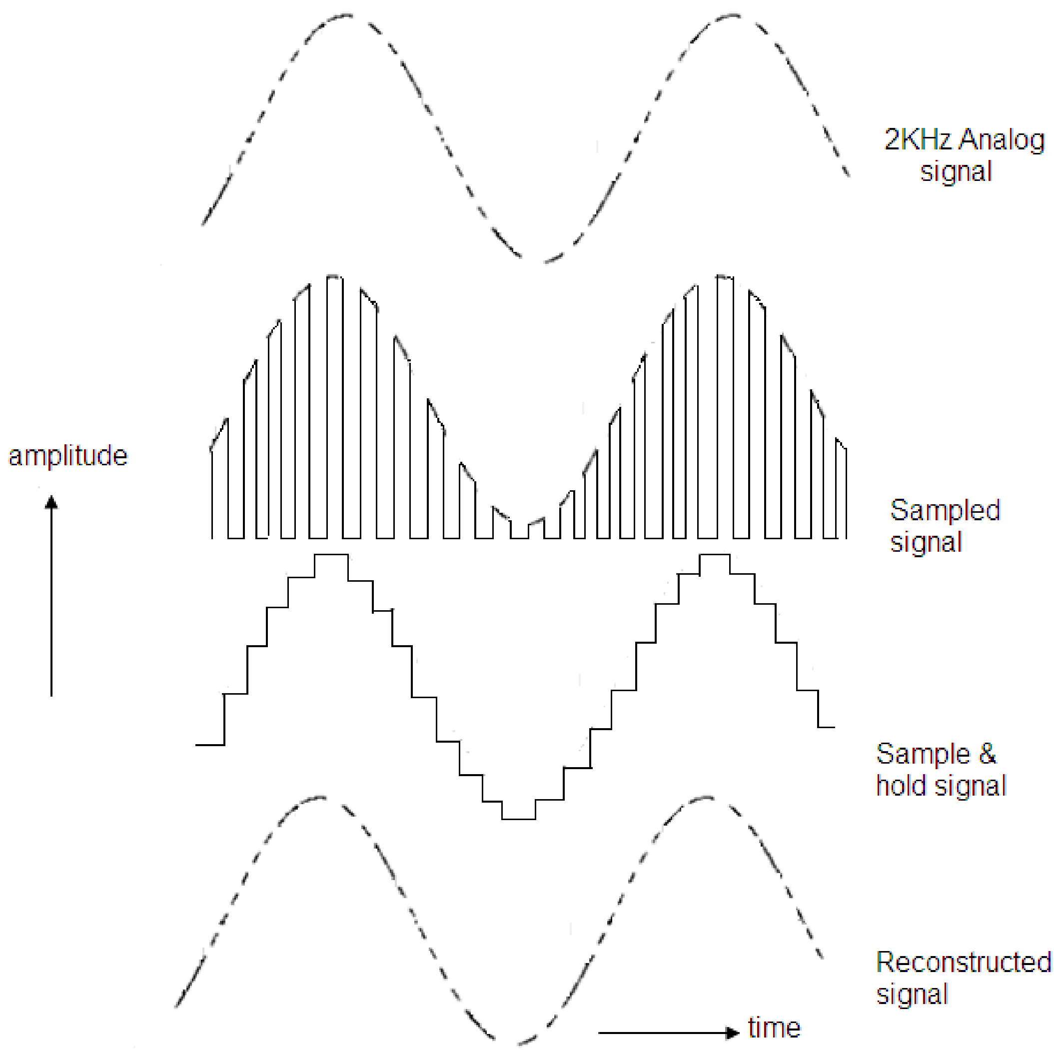
C. Set up for sample/hold circuit on reconstructed signal.

Initial set up of trainer:

Duty cycle selector switch position: Position 5 Sampling selector switch: Internal

1. Connect the power cord to the trainer. Keep the power switch in 'Off' position.
2. Connect 1 KHz Sine wave to input signal.Connect BNC connector to the CRO and to the trainer's output port.
3. Connect Sample Output to fourth Order low pass filter Input and Sample and Hold
4. Output to second Order low pass filter Input
5. Switch 'On'the trainer's power supply & Oscilloscope.(Turning 'On'the supply will randomly select the sampling frequency).
6. Select sampling frequency of 8 KHz by Sampling Frequency Selector Switch pressed till 80 KHz signal LED glows.
7. Vary the position of Duty Cycle Selector Switch from 0% to 90% (position 0 to 9) and observe the Sample Output (TP39) and Sample and hold Output (TP41).
8. Also observe variation of output signal with the change in duty cycle at low pass filter outputs (TP44 and TP48). Compare the output of the two low pass filters

MODEL GRAPH:



TABULATION:

Signal	**Amplitude(V)**	**Time period (ms)**
Analog input Signal		
Sampling Signal		
Sampled Output		
Sample and Hold Output		
Reconstructed Signal		

RESULT:

Thus the Sampling and Reconstructed Unit was studied and verified.

EX: NO: 4b
DATE:

TIME DIVISION MULTIPLEXING AND DEMULTIPLEXING

Aim:

To study the Time Division Multiplexing (TDM) and draw its waveforms.

Apparatus Required:

1. DCL 02 TDM kit
2. Digital Storage Oscilloscope (DSO)
3. Power supply
4. Patch cords

Procedure:

1. The connections are given as per the block diagram.
2. Connect the power supply in proper polarity to the kit and & switch it on.
3. Set the amplitude of the sine wave as desired.
4. Observe the following waveforms at the

a. Input Channel

b. Multiplexer Output (TXD)

c. Reconstructed Signal (OUT0, OUT1,OUT2, OUT3) and plot it on graph paper

Block diagram

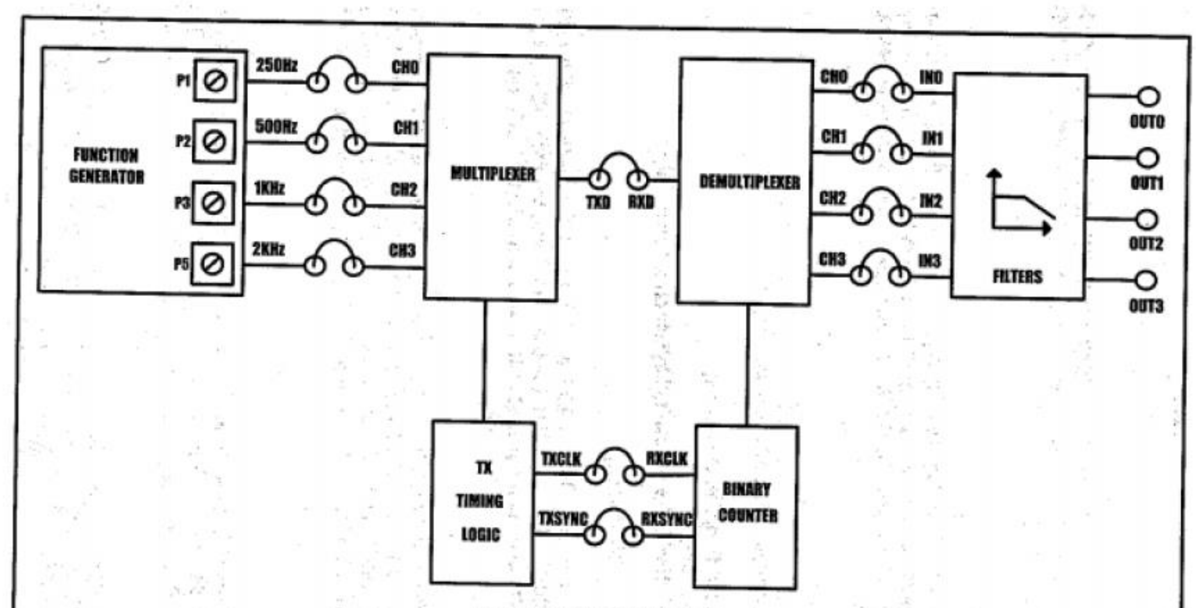
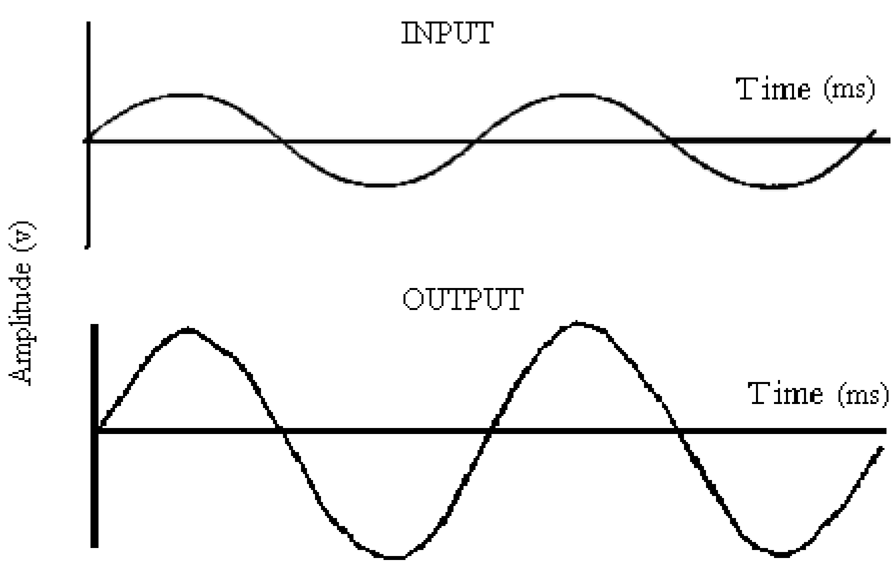


FIG.1 BLOCK DIAGRAM FOR EXP.1

Model graph:



Tabulation:

Signal	Amplitude(volts)	Time(ms)
Carrier Signal		
Message Signal 1		
Message Signal 2		
Modulated o/p	Emax= Emin=	
Demodulated Signal 1		
Demodulated Signal 2		

RESULT

Thus the TDM Modulation and Demodulation are verified in the hardware kit and its waveforms are studied.

EX: NO: 5
DATE:

PULSE CODE MODULATION AND DEMODULATION.

AIM:

To study and verify the pulse code modulation technique.

EQUIPMENTS REQUIRED:

Equipments	Range	Quality
PCM Kit	-	1
CRO	-	1
Multimeter		
patch Cards	-	As Required
Probes		

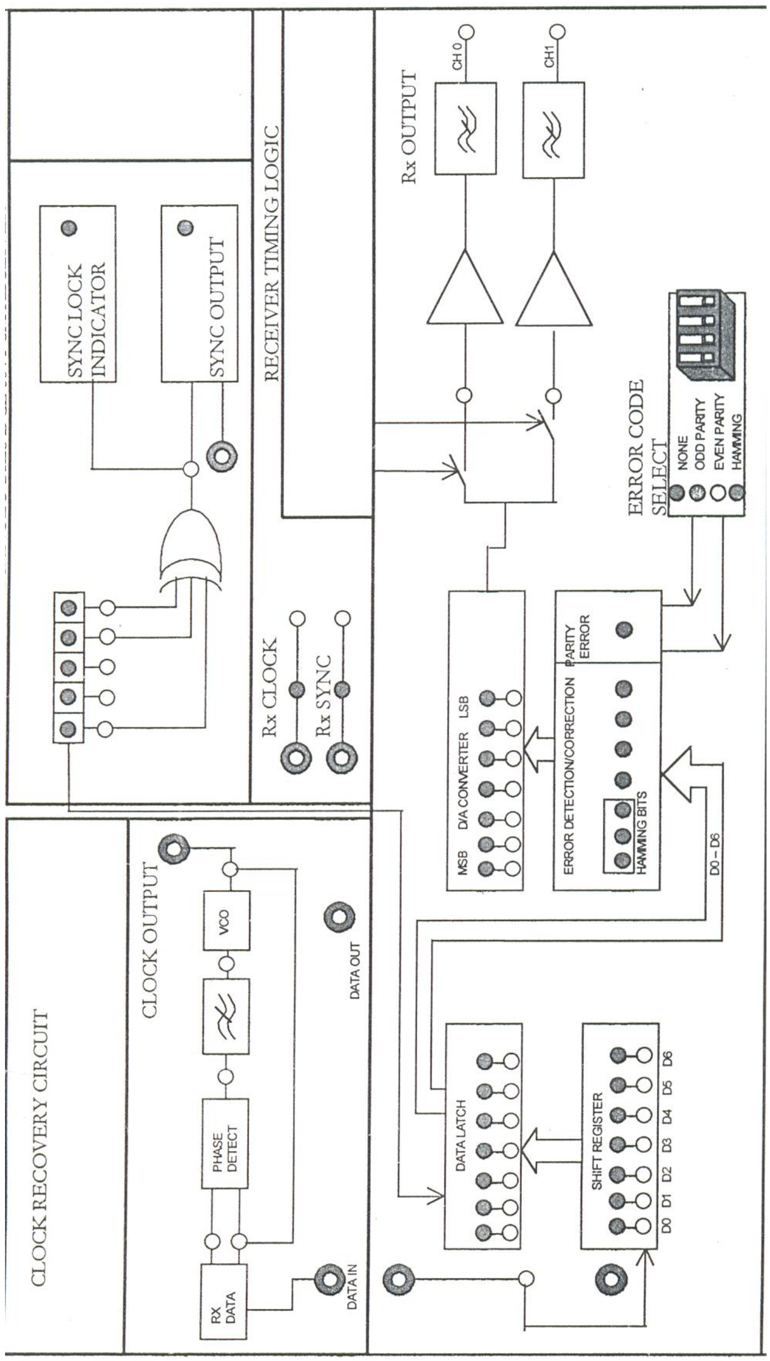
THEORY:

PCM system are complex in that message signal is subjected to large number of operations. The essential operation in the transmitter of a PCM system are sampling, Quantizing and encoding. The sampling, Quantizing and encoding operations are usually performed in the same circuit which is called analog to digital computer regeneration of impaired signal occurs at intermediate points along the transmission path as indicated middle point at this receiver the essential operation consists of one last stage of regeneration followed by decoding, then the demodulation of the train of quantized samples as in the bottom part of decoding and reconstructions are usually performed in the same circuit, called a digital to analog converter when TDM is used it becomes necessary to synchronize the receives to the overall system to operate, satisfactory. It is noteworthy that PCM is not modulation in the conventional sense. The term modulation usually refers to the variation of some characteristics of a carrier wave in accordance with an information bearing signal. The only part of PCM that conforms to that definition is sampling which is basic to PCM, introduces a signal distortion that has to counter part in conventional modulation.

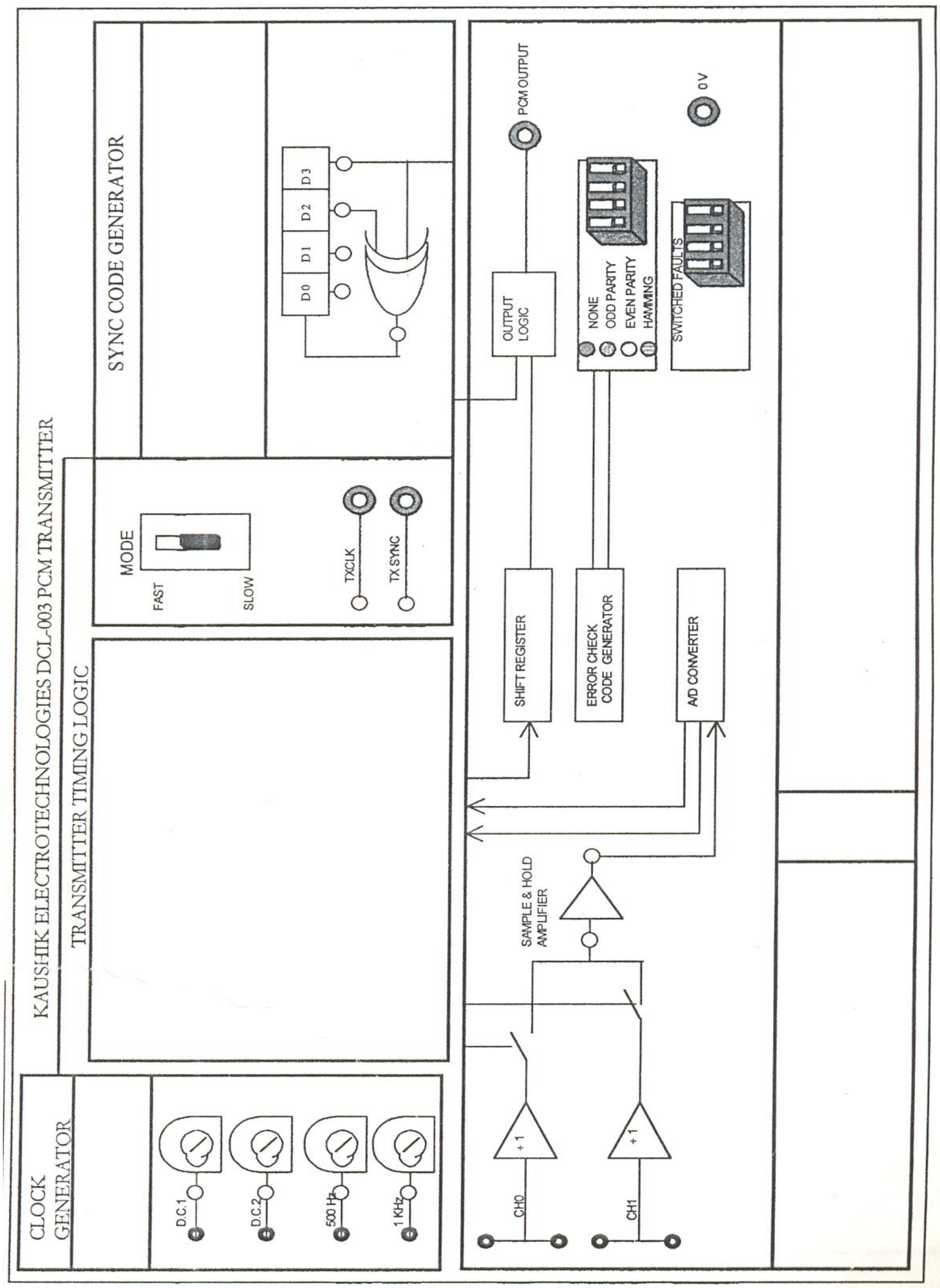
PROCEDURE:

1. Connections are given as per the circuit diagram.
2. Set the speed selection into low mode.
3. Vary the analog input signal from minimum to maximum range and note down the voltage value from Multimeter and Binary code from A/D converter.
4. Plot the graph.

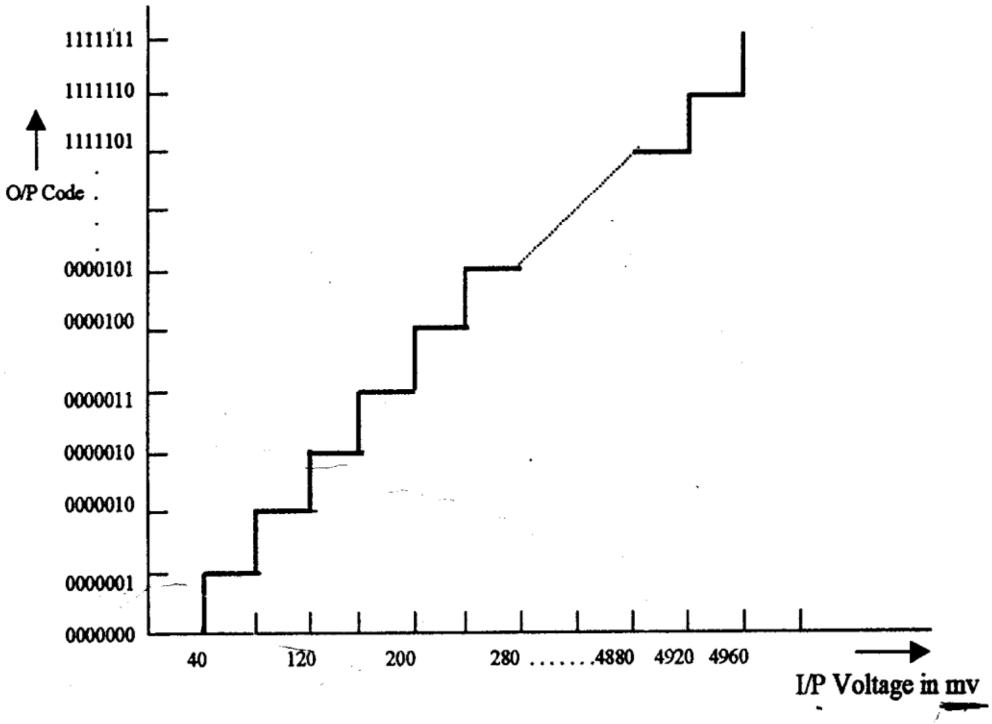
PCM Transmitter



PCM Receiver:



MODEL GRAPH:



A/D Conversion O/P code Vs Input analog voltage

TABULATION:

Voltage (V)	Binary code (bits)

RESULT:

Thus the pulse code modulation technique was studied and verified.

EX. NO:6
DATE:

PULSE AMPLITUDE MODULATION AND DEMODULATION.

AIM:

To study and verify the pulse Amplitude modulation and Demodulation techniques.

EQUIPMENTS REQUIRED:

Equipments	Range	Quality
Pulse modulation kit Kit	-	1
CRO	-	1
Patch Cards	-	As Required
Probes		

THEORY:

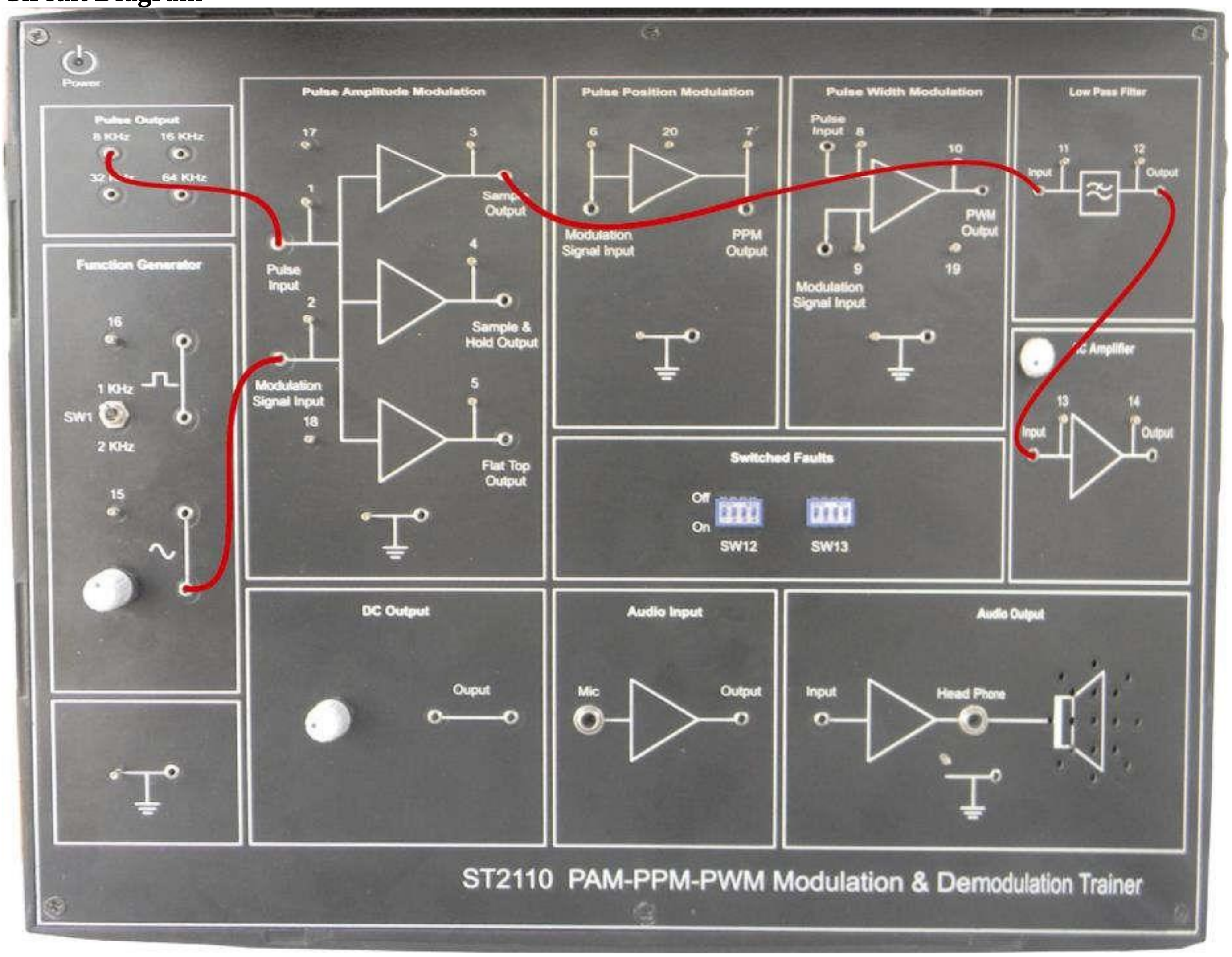
(1). Pulse amplitude modulation:

In flat top sampling, the amplitude of the signals remains at an instant of time unlike in natural sampling when the amplitude of the sample vary in accordance with the sample amplitude. For generating the flat top sampling the signal to be sampled is first fed into a sample and hold amplifiers circuit, which generates the staircase waveform as represents in the graph at point. The sampling clock selected determine the hold period of the sample and hold. The resulting waveform is then passed into a electronic switch latches the samples of sample / hold waveforms for the period determined by the duty cycle of the input sampling circuit.

Procedure:

1. Connect the circuit as shown in Figure 2.1.
    a. Output of sine wave to modulation signal IN in PAM block keeping the switch in 1 KHz position.
    b. 8 KHz pulse output to pulse input.
    c. Output of low pass filter to input of AC amplifier.
    d. Keep the gain pot in Connect the sample output to low pass filter input.
    e. AC amplifier block in anticlockwise position.
2. Switch‘On’th powersupply& oscilloscope.
3. Observe the outputs at TP (3 & 5) these are natural & flat top outputs respectively.
4. Observe the difference between the two outputs.
5. Vary the amplitude potentiometer and frequency change over switch & observe the effect on the two outputs.
6. Vary the frequency of pulse, by connecting the pulse input to the 4 frequencies available i.e. 8, 16, 32, 64 kHz in Pulse output block.
7. Switch‘On’faultNo.1,2,3,4one by one & observe their effect on Pulse Amplitude Modulation output and try to locate them.
8. Monitor the output of AC amplifier. It should be a pure sine wave similar to input.
9. Vary the amplitude of input, the amplitude of output will vary.
10. Similarly connect the sample & hold & flat top outputs to low pass filter and see the demodulated waveform at the output of AC amplifier.
11. Switch‘On’theswitched faults No.1,2,3,4,5&8 one by one and see their effects on output.

Circuit Diagram



TABULATION:

Signal	Amplitude(volts)	Time period (ms)
Input signal		
Train pulse or Carrier signal		
PAM signal		
Output signal		

Demodulation

Step 11 Connect the Sample output to the input of low pass filter. Observe the output of the Low pass filter TP (12) together with Modulation signal input TP (2).

Step 12 Observe the output of the AC Amplifier TP (14) together with Modulation signal input TP (2). Vary the Gain of AC Amplifier to get the unclipped output. Vary the amplitude of input; the amplitude of output will vary.

Step 13 Connect the Flat top output to the input of low pass filter Observe the output of the Low pass filter TP (12) together with Modulation signal input TP (2).

Step 14 Observe the output of the AC Amplifier TP (14) together with Modulation signal input TP (2). Vary the Gain of AC Amplifier to get the unclipped output. Vary the amplitude of input; the amplitude of output will vary.

Step 15 Connect the Sample & Hold outputto the input of low pass filter Observe the output of the Low pass filter (12) together with Modulation signal input TP (2).

Step 16 Observe the output of the AC Amplifier TP (14) together with Modulation signal input TP (2). Vary the Gain of AC Amplifier to get the unclipped output. Vary the amplitude of input; the amplitude of output will vary.

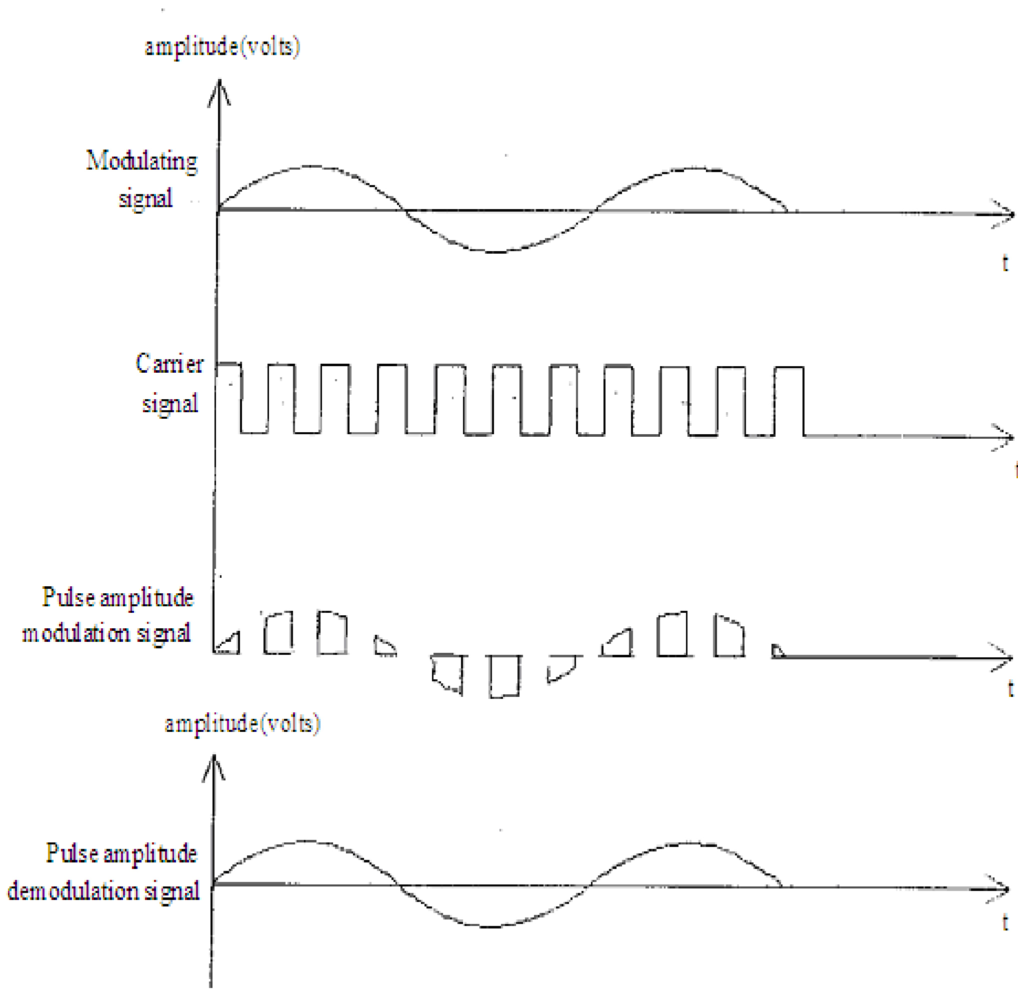
Step 17 Vary the Amplitude Potentiometer and frequency change over switch & observe the effect on the output.

Step 18 Vary the frequency of pulse, by connecting the pulse input to the 4 frequencies available i.e. 8, 16, 32, 64 kHz in Pulse output block.

Step 19 Switch ‘On’ fault No. 1, 2, 3, 4 one by one & observe their effect on Pulse Amplitude Modulation output and try to locate them.

Step 20 Switch ‘Off’ the Power Supply.

MODEL GRAPH:


amplitude(volts)
Modulating signal
t
Carrier signal
t
Pulse amplitude modulation signal
t
amplitude(volts)
Pulse amplitude demodulation signal
t

RESULT:

Thus the pulse amplitude modulation and demodulation techniques was verified and output was plotted successfully.

EX. NO: 7
DATE:

PULSE POSITION MODULATION AND DEMODULATION AND PULSE WIDTH MODULATION AND DEMODULATION.

AIM:

To study and verify the different types of pulse modulation techniques.

EQUIPMENTS REQUIRED:

Equipments	Range	Quality
Pulse modulation kit	-	1
CRO	-	1
Multimeter		
Patch Cards	-	As Required
Probes		

THEORY:

(1). Pulse width modulation:

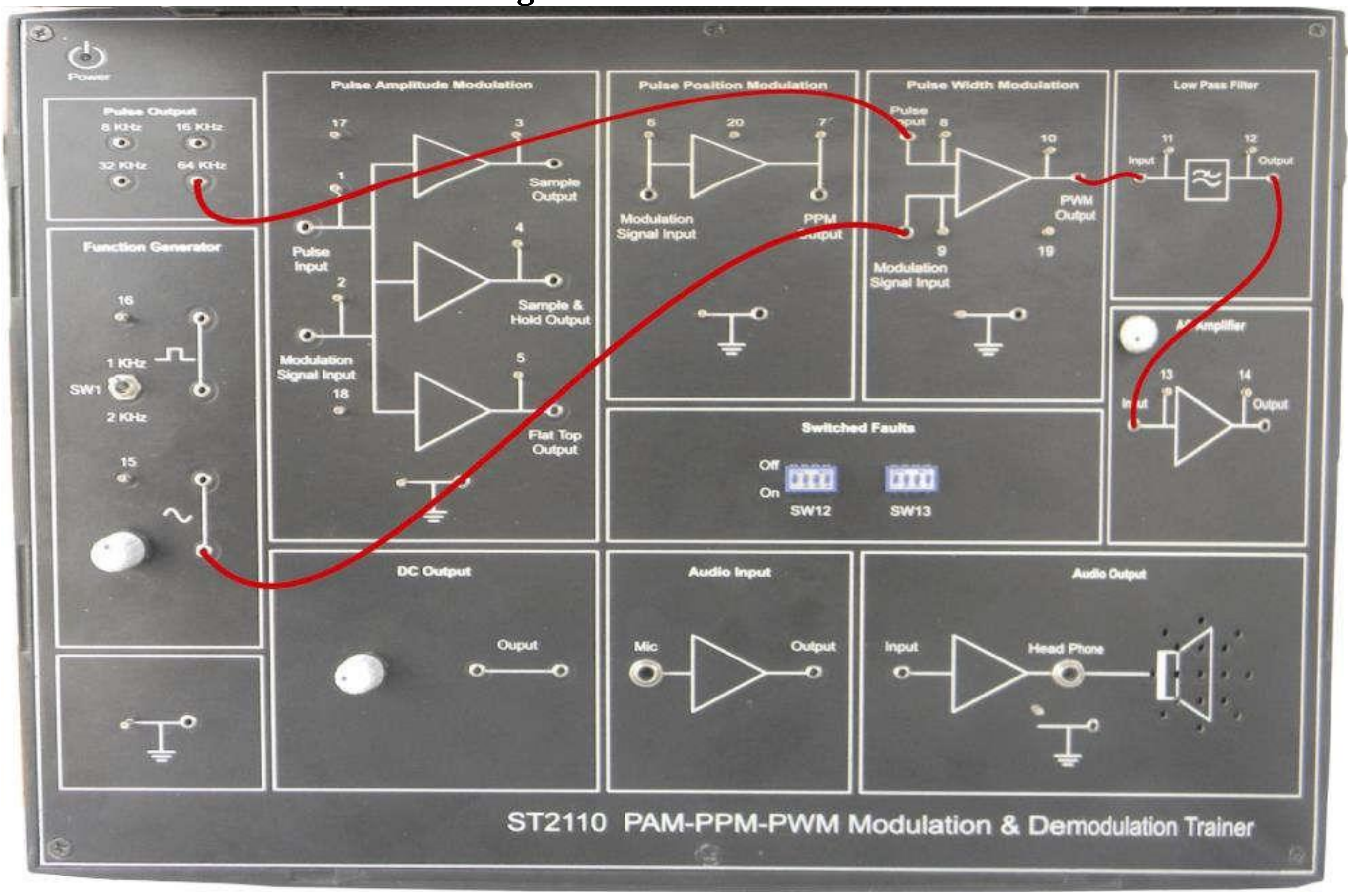
PWM is a type of pulse modulation where the duration of the pulse varies in proportion with the sample values of the analog signals. The generation of PWM signals involves of first sampling the analog signal into PAM form. The modulating signal m(t) is applied to the input of a PAM modulating circuit to generate the PAM signal. The same pulse gain which supplies the PAM modulator is used to gate ON a ramp generator to generate a train of slopes and amplitudes. The pulse duration only varied when we change the input message signal.

(2). Pulse position modulation:

PPM is another type of modulation. The pulse positions various in accordance with the amplitude of the modulating signal. PPM signals can be readily generated acts as applied trigger pulse to a monostable multivibrator. The monostable trigger generates are applied as trigger pulse to monostable multivibrator.

The Monostable trigger each time at the positive going edge of the PWM signals by generating the PPM signals constant amplitude width for demodulation of PPM. The direct demodulation of PPM, PWM can be used. But the demodulated analog signal shows a very low amplitude for PPM pulses are very narrow and much spaced out converting the PPM signals into PWM, with subsequent filtering through a LPF perform a maximum effective PPM signals.

Circuit Diagram-Pulse Width Modulation



(1) Pulse width modulation:

Signal	Amplitude(volts)	Time period (ms)
Input signal		
Train pulse or carrier signal		
PWM signal		
Output signal		

Procedure:

1. Connect the circuit as shown in Figure
   a. 1 KHz sine wave output of function generator to modulation input of PWM
b. 64 KHz square wave output to pulse input. c. Output of PWM to input of low pass filter.
d. Output of low pass filter to input of AC Amplifier.

2. Switch 'On' the power supply & oscilloscope.

3. Observe the output of low pass filter and AC amplifier respectively to demodulation of pulse width demodulation waveform in detail. understand the

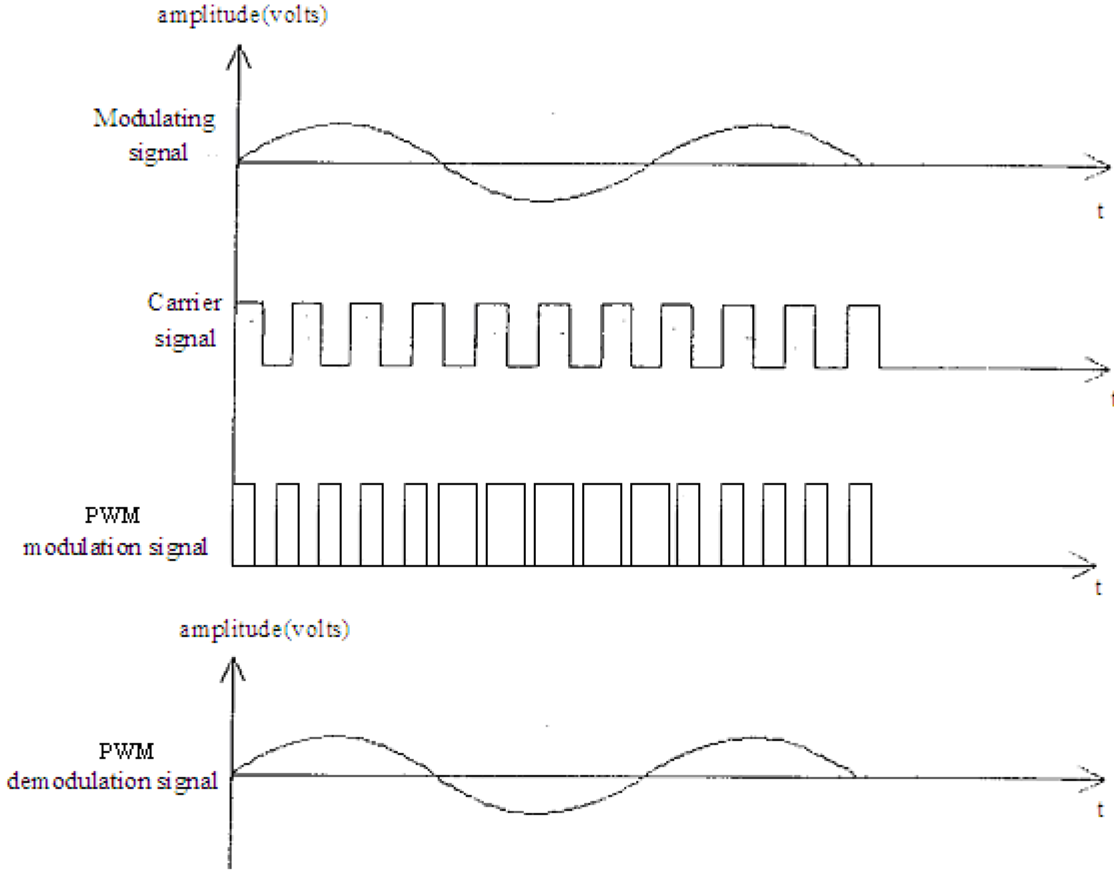
4. Vary the amplitude and frequency of sine wave and observe its effect on the demodulated waveform.

5. Now, connect the pulse input in the pulse width modulation block to the different frequencies available on board viz. 8, 16, 32 KHz and observe their demodulated waveforms.

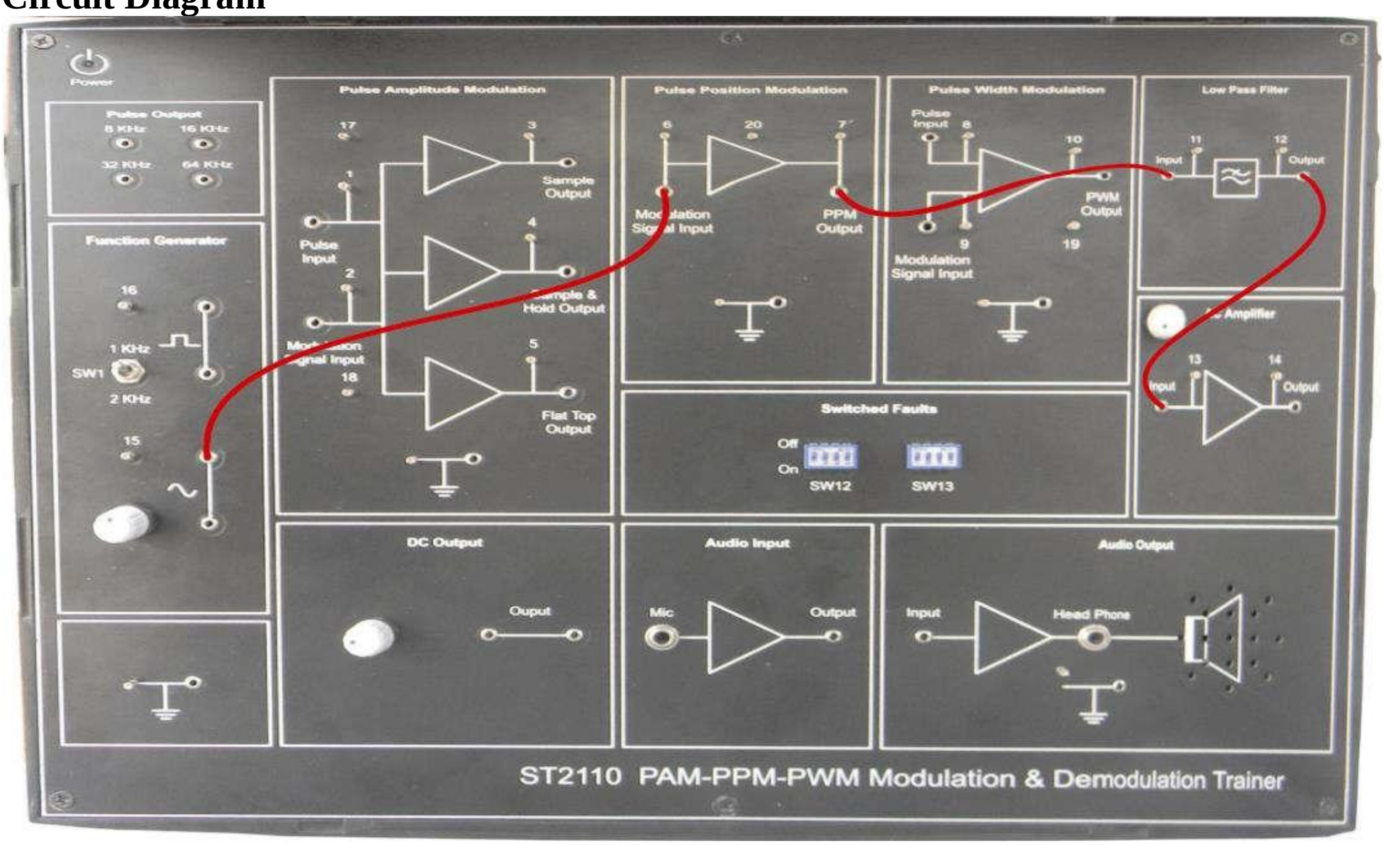
6. Try varying the amplitude of sine wave signal; you will observe that the output signal varies similarly.

7. Switch 'On' fault no, 1, 2, 5 & 8 one by one at a time. Observe their effects on final output and try to locate them.

MODEL GRAPH:



Pulse Position Modulation
Circuit Diagram



Procedure:

1. Connect the circuit as shown in Figure 2.3 and also described below for clarity.

a. Sine wave of 1 KHz to input of PPM block.

b. Output PPM block to input of low pass filter.
c. Output of low pass filter to input of AC amplifier.

d. Keep the gain potentiometer in amplifier block at maximum position.

2. Switch 'On' the power supply & oscilloscope.

3. Observe the waveform at the TP12 output of low pass filter block.

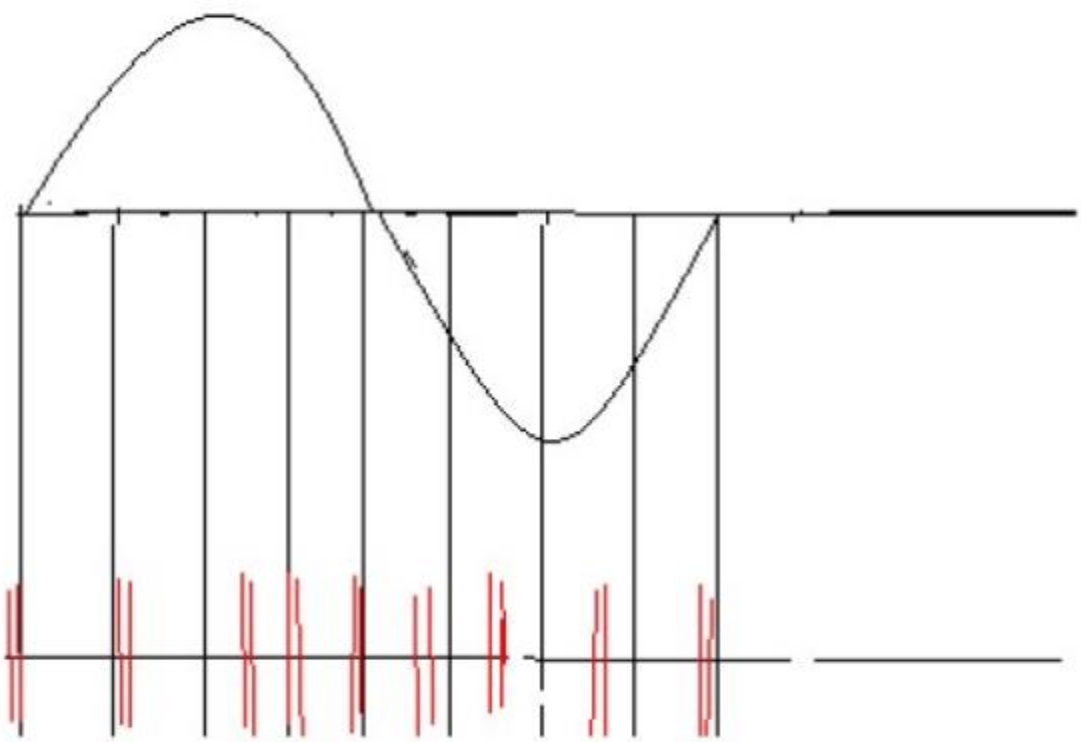
4. Then observe the delnodulated output at TP14 output of AC amplifier.

5. Switch 'On' fault No. 1, 2, 6 & 8 one by one & observes their effect on demodulated waveform & tries to locate them.

(2) Pulse position modulation:

Signal	Amplitude(volts)	Time period (ms)
Input signal		
Train pulse or Carrier signal		
PPM signal		
Output signal		

MODEL GRAPH: PWM



RESULT:

Thus the different types of pulse modulation techniques was verified and output was plotted Successfully.

Ex.No : 8

Date :

DIGITAL MODULATION – ASK, PSK, FSK

AIM:

To study and verify the Digital Modulation & Demodulation techniques.

APPARATUS REQUIRED:

Equipments	Range	Quality
Digital modulation technique trainer kit	-	1
CRO	-	1
Multimeter		
Patch Cards	-	As Required
Probes		

THEORY:

(1). Amplitude Shift Keying:

Amplitude Shift Keying or ON – OFF keying is the simplest digital modulation technique.

In this method, there is only one unit energy carrier input binary Sequence, the ASK waveform can be represented as

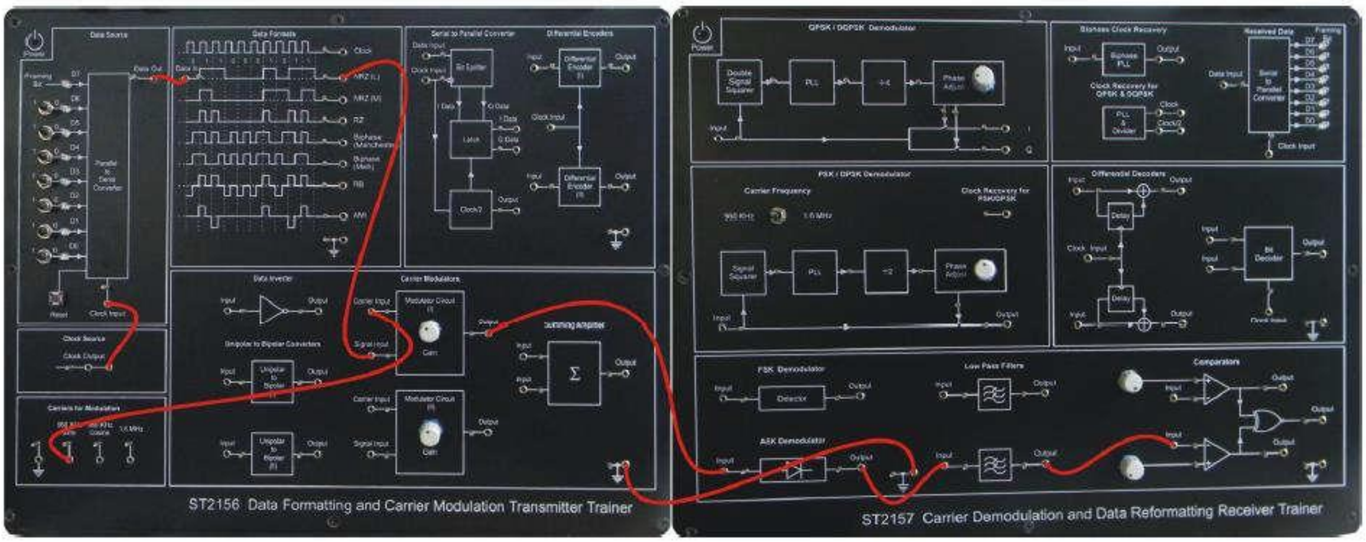
$$S(t) = \sqrt{2}ps \cos(2\pi f_c t)$$

To transmit symbol '0' the signal S (t) =0 that is no signal is transmitted. S (t) contains some complete cycle of carrier frequency 'f'. Thus symbol '1' is pulse transmitter and '0' is no pulse transmitted. Thus ASK looks like an ON-OFF of the signal. Hence it is also known as ON-OFF keying. The input binary sequence is applied to the product modulation. It passes the carrier when input is '1'.It blocks the carrier when input is '0'. The ASK signal is applied to the correlator consisting of multiplier and integrator. The locally generated coherent carrier is applied to the multiplier.

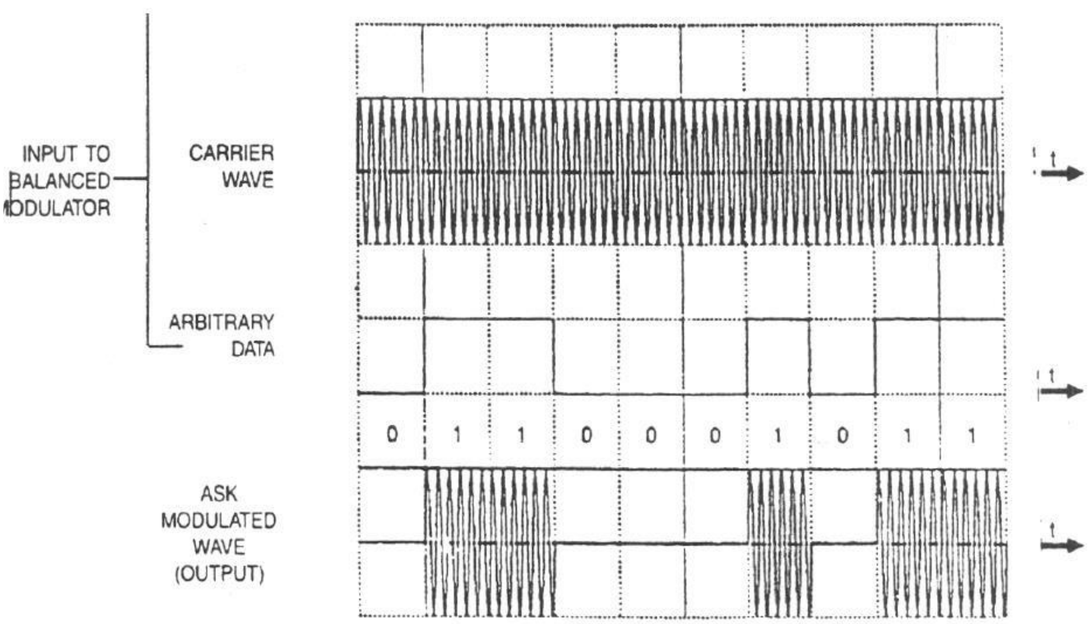
Procedure:

1. Connect the power supplies of ST2156 and ST2157 but do not turn on the power supplies until connections are made for this experiment.
2. Make the connections
3. Switch 'ON' the power.
4. On ST2156,connect oscilloscope CH1 to 'ClockI n'and CH2 to 'Data In' and observe the waveforms.
5. On ST2156, connect oscilloscope CH1 to 'NRZ (L)' and CH2 to 'Output' of modulator Circuit (l) on ST2156 and observe the waveforms.
6. Vary the gain potentiometer of modulator circuit (l) on ST2156 to adjust the amplitude of ASK Waveform.
7. On ST2156, connect oscilloscope CH1 to 'NRZ (L)' and CH2 to 'Output' of comparator on ST2157 and observe the waveforms.

Circuit Diagram



Model Graph:



(2) Frequency Shift Keying:

In FSK the frequency of the carrier is shifted to the binary symbol. The phase of the carrier is unaffected. We have two different frequency signals according to binary symbols.

If b (t) = ‘1’: S_H (t) = √2ps cos ($2\pi f_c t$)

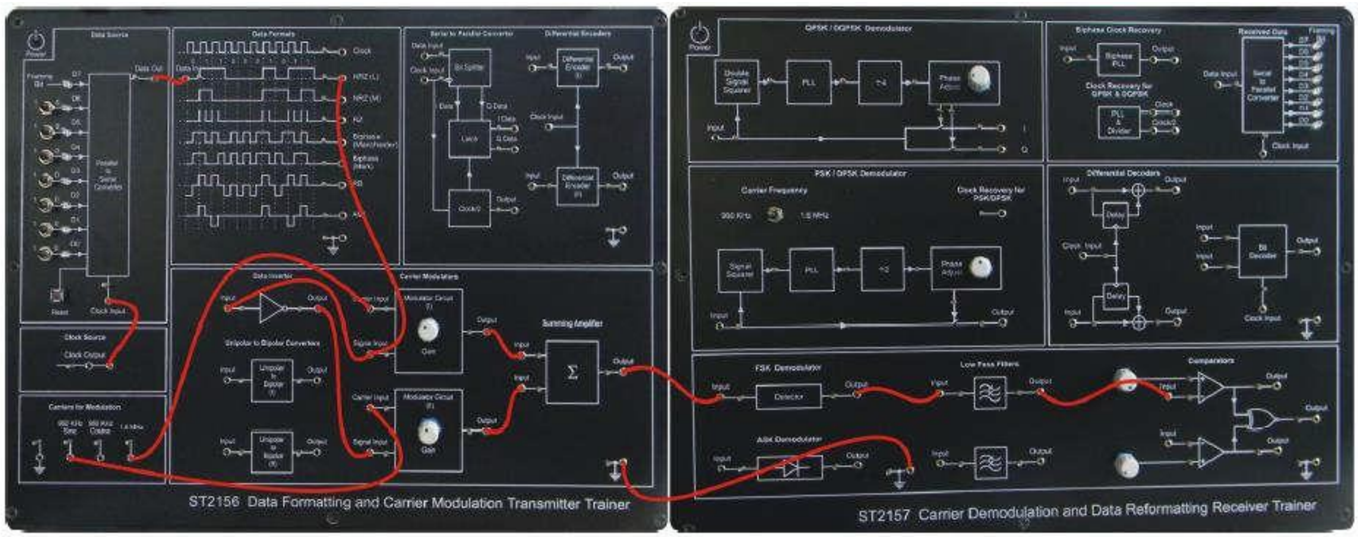
If b (t) = ‘0’: S_L (t) = √2ps cos ($2\pi\ f_c t$)

Thus there is increase or decrease in frequency by Ω. In this FSK, there are transmitter and receiver. In transmitter we know that input sequence b (t) is same as P_H (t). An inverter is added after b (t) to get P_L (t). P_H (t) and P_L (t) are unipolar signals. The level shifter converts the +1 level to √psTb zero level is unaffected.

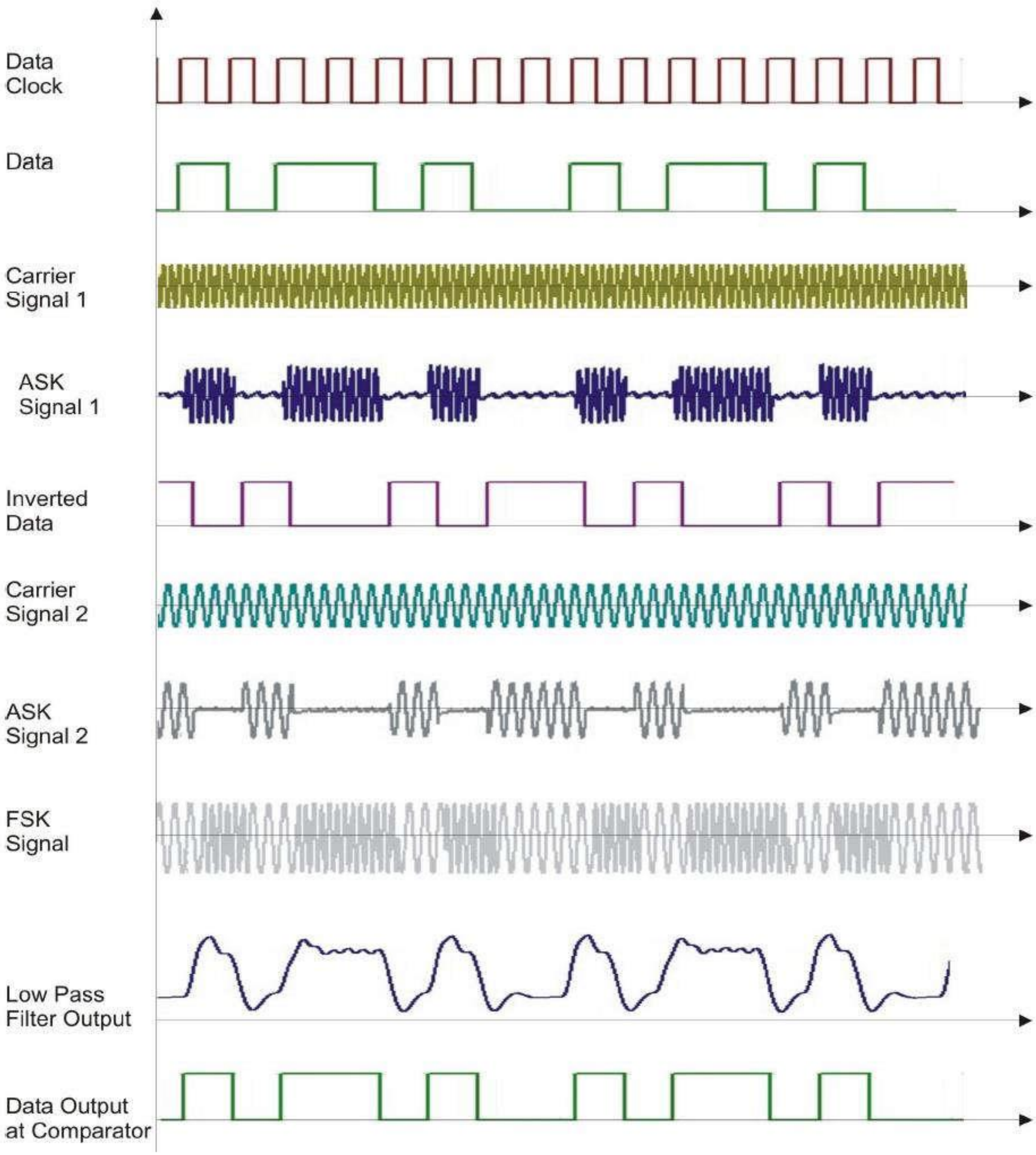
Procedure:

1. Connect the power supplies of ST2156 and ST2157 but do not turn on the power supplies until connections are made for this experiment.
2. Make the connections as shown in the figure
3. Switch 'ON' the power.
4. On ST2156,connect oscilloscope CH1 to ‘Clock In’ and CH2 to ‘Data In’ and observe the waveforms.
5. On ST2156, connect oscilloscope CH1 to‘NRZ (L)’ and CH2 to‘Output’ of Summing Amplifier on ST2156 and observe the waveforms.
6. Adjust the potentiometers of both the Modulator Circuit (l) & (ll) onST2156 to adjust The amplitude of FSK waveform at Summing Amplifier’s output on ST2156.
7. On ST2156, connect oscilloscope CH1 to ‘NRZ (L)’ and CH2 to ‘Output’ of comparator on ST2157 and observe the waveforms.

Circuit diagram:



Model Graph:


Data Clock
Data
Carrier Signal 1
ASK Signal 1
Inverted Data
Carrier Signal 2
ASK Signal 2
FSK Signal
Low Pass Filter Output
Data Output at Comparator

(3). Phase Shift Keying:

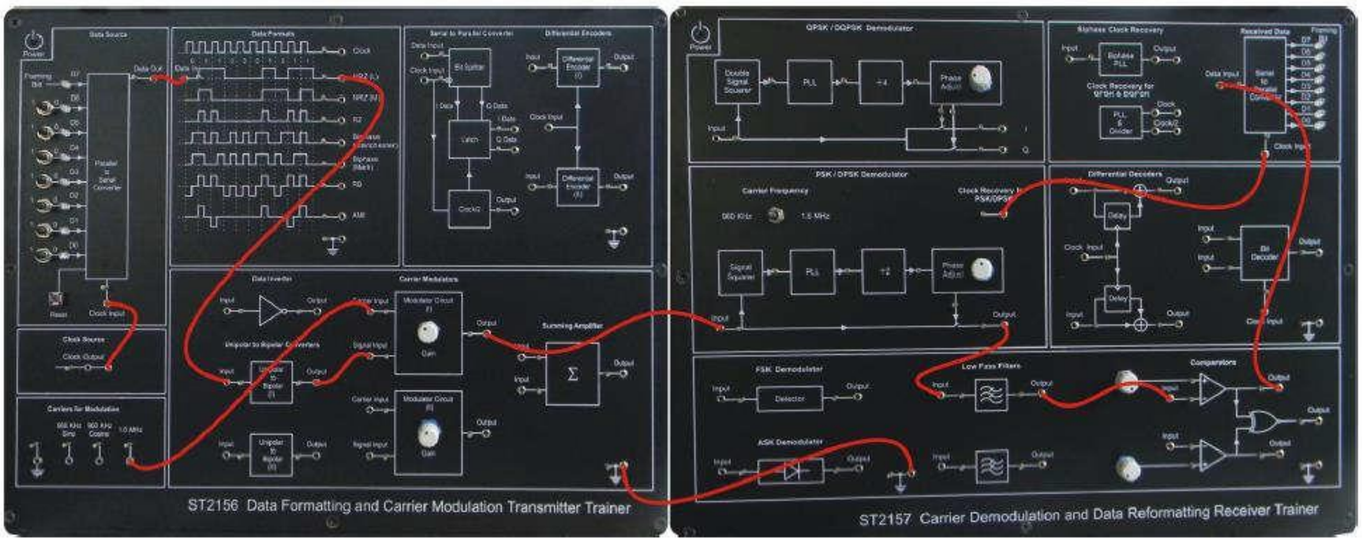
In communication systems, we know that there are two main resources i.e., transmission power and channel bandwidth depends upon the bit rate or signaling rate f_b. In digital band pass transmission. This carrier is transmitted over a channel. If two or more bits are combined in same symbols then the signaling rate is reduced. Therefore the frequency of the carrier required is also reduced. This reduces the channel band width.

In PSK two successive bits in the data sequence are grouped together. Thus reduce the bit rate of signaling rate and hence reduces the band width of the channel. In binary PSK we know that when signal changes the level. The phase of the carrier is changed by 180 degree. Since there were only two symbols in BPSK, the phase shift occurs in two levels only in PSK, two successive bits are

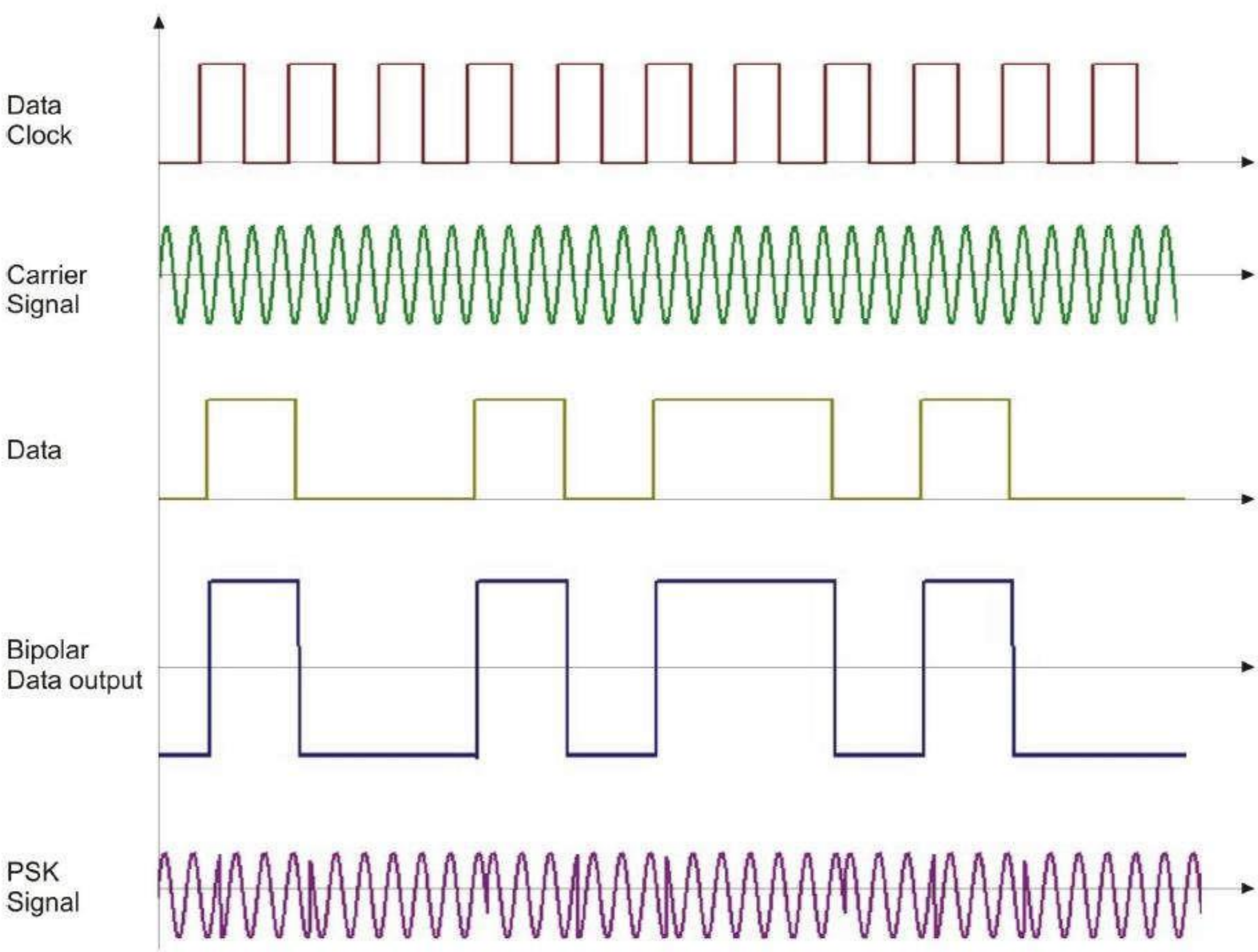
Procedure:

1. Connect the power supplies of ST2156 and ST2157 but do not turn on the power supplies until connections are made for this experiment.
2. Make the connections as shown in the figure 7.1.
3. Switch 'ON' the power.
4. On ST2156, connect oscilloscope CH1 to 'Clock In' and CH2 to 'Data In' and observe the waveforms.
5. On ST2156, connect oscilloscope CH1 to 'NRZ (L)' and CH2 to 'Output' of Modulator Circuit (1) on ST2156 and observe the waveforms.
6. Adjust the 'Gain' potentiometer of the Modulator Circuit (1) on ST2156 to adjust the amplitude of PSK waveform at output of Modulator Circuit (1) on ST2156.
7. Now on ST2157 connect oscilloscope CH1 to 'Input' of PSK demodulator and connect CH2 one by one to output of double squaring circuit, output of PLL, output of Divide by four (2) observe the wave forms.
8. On ST2157 connect oscilloscope CH1 to output of Phase adjust and CH2 to 'output' of PSK demodulator and observe the waveforms. Set all toggle switch to 0 and compare the waveform now vary the phase adjust potentiometer and observe its effects on the demodulated signal waveform. (Note: If there is problem in setting the waveform with potentiometer then toggle the switch given in PSK demodulator block two to three times to get the required waveform).

Circuit diagram:



MODEL GRAPH:



TABULATION:

(1) Amplitude Shift Keying:

S.NO	SIGNAL	AMPLITUDE(V)	TIME PERIOD(ms)
1.	Message Signal		
2.	Carrier Signal		
3.	ASK Modulated Signal		
4.	ASK Demodulated Signal		

(2) Frequency Shift Keying:

S.NO	SIGNAL	AMPLITUDE(V)	TIME PERIOD(ms)
1.	Message Signal		
2.	Carrier Signal		
3.	FSK Modulated Signal		
4.	FSK Demodulated Signal		

(3) Phase Shift Keying:

S.NO	SIGNAL	AMPLITUDE(V)	TIME PERIOD(ms)
1.	Message Signal		
2.	Carrier Signal		
3.	PSK Modulated Signal		
4.	PSK Demodulated Signal		

RESULT:

Thus the Digital Modulation and Demodulation techniques was studied and verified.

EX. NO: 9
DATE:

DELTA MODULATION AND DEMODULATION

AIM:

To study and verify the characteristics of the Delta modulation and demodulation systems.

EQUIPMENTS REQUIRED:

Equipments	Range	Quality
DM Kit	-	1
CRO	-	1
Batch Cards	-	As Required
Probes		

THEORY:

Delta modulation is an encoding process where the logic levels of the transmitted pulses indicate whether the decoded output should rise or fail at each pulse.The delta encoding process samples, quantizes and encodes the intelligence signal into a digital signal. The instantaneous voltage of an signal is compared to the feedback signal. The result of the comparision is quantized &encoded & appears as a logic 1 or logic 0 depending on which sample voltage is greater. The encoded logic levels make up the digital signal. Delta modulation require simple hardware for encoding an intelligence Signal.

The encoding process consists of a digital sampler and an integrator as shown in the figure below: the digital sampler consists of a comparator and D-type flip flop. The intelligence signal drives the non-inverting input of the comparator. The feedback signal from the integrator drives the inverting input of the comparator. During each clock signal the comparator compares the present sample voltage of the intelligence signal with the feedback signal. The feedback signal is an approximate voltage of the previous intelligence signal sample. If the intelligence signal is greater than the feedback signal.The comparator outputs a logic1 to the input of the D-type FF.

The D-type FF will set the Q output to +5V on the leading edge of the next clock pulse. If the intelligence signal is less than the feedback signal the comparator outputs an (-Ve) signal to the D-FF. The output of the D-FF is 0V on the leading edge of the next clock pulse. The output of the D-FF is the digital signal.

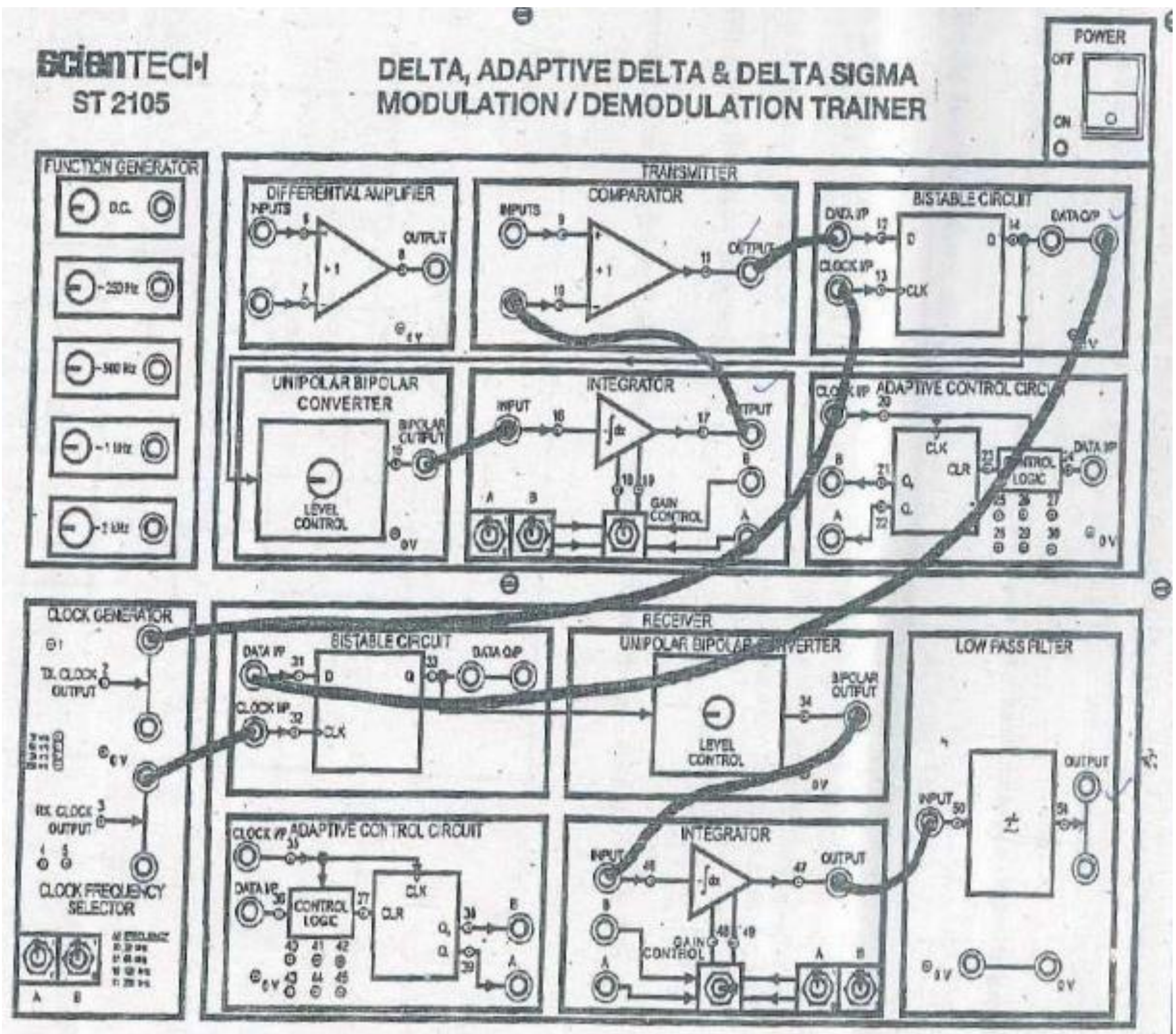
The digital signal contains the information needed by an integrator to generate the approximate intelligence signal (feedback signal). The integrator outputs an upward – sloping ramp as the feedback signal when the digital signal is ay logic 1.When the digital signal is at logic 0, the integrator outputs a download – sloping ramp as the feedback signal. The digital signal is the difference between the intelligence and feedback signals.

PROCEDURE :

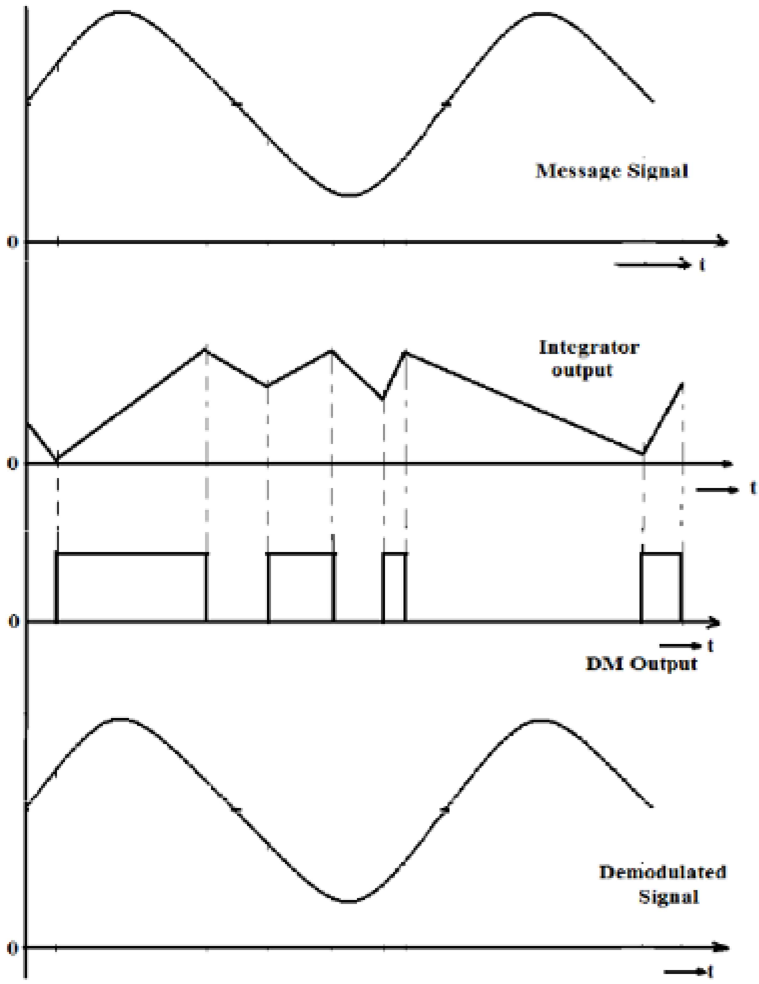
1.Connect the comparator O/P node 11 of transmitter to the I/P node 12 of bistable circuit
2. Connect the comparator I/P node 10 to O/P node 17 of integrator
3. Connect the O/ P node 15 of bipolar convertor to I/P node 16 of integrator.
4. Connect the TX clock O/P node 2 of clock generator to clock I/P node13 of bi-stable circuit
5.Connect I/P node 31 of bi- stable circuit of receiver to O/P node 14 of bi-stable circuit of transmitter.

6.Connect the receiver clock O/P node 3 to input node 32 of bi-stable circuit of receiver.
7. Connect the I/P node 46 of integrator of receiver to bipolar O/P node 34 of bipolar converter of receiver.
8. Connec t the O/P node 4v 7 of integrator of receiver to I/P node 50 of low pass filter
9. Connect the CRO to the I/P node 10 of comparator of transmitter and trace the waveform
10. Connect the CRO at O/P node of integrator & trace the waveform.
11. Connect th e CRO at O/P node of bi-stable circuit and trace the waveform
12. Connect the CRO at O/P node of LPF and trace the waveform.

CIRCUIT DIAGRAM



MODEL GRAPH:



Waveforms of Delta Modulation & Demodulation

TABULATION:

S.NO	TYPE OF SIGNAL	AMPLITUDE (V)	TIME PERIOD (ms)
1.			
2.			
3.			
4.			
5.			
6.			
7.			
8.			

RESULT:

Thus the Delta modulation and demodulation characteristics was studied and verified.

EX. NO: 10
DATE:

SIMULATION OF ASK, FSK AND BPSK

AIM

To Simulate the ASK, FSK and BPSK using MATLAB 7.8 Software.

SOFTWARE / EQUIPMENTS REQUIRED

- PC having MATLAB SOFTWARE

PROCEDURE

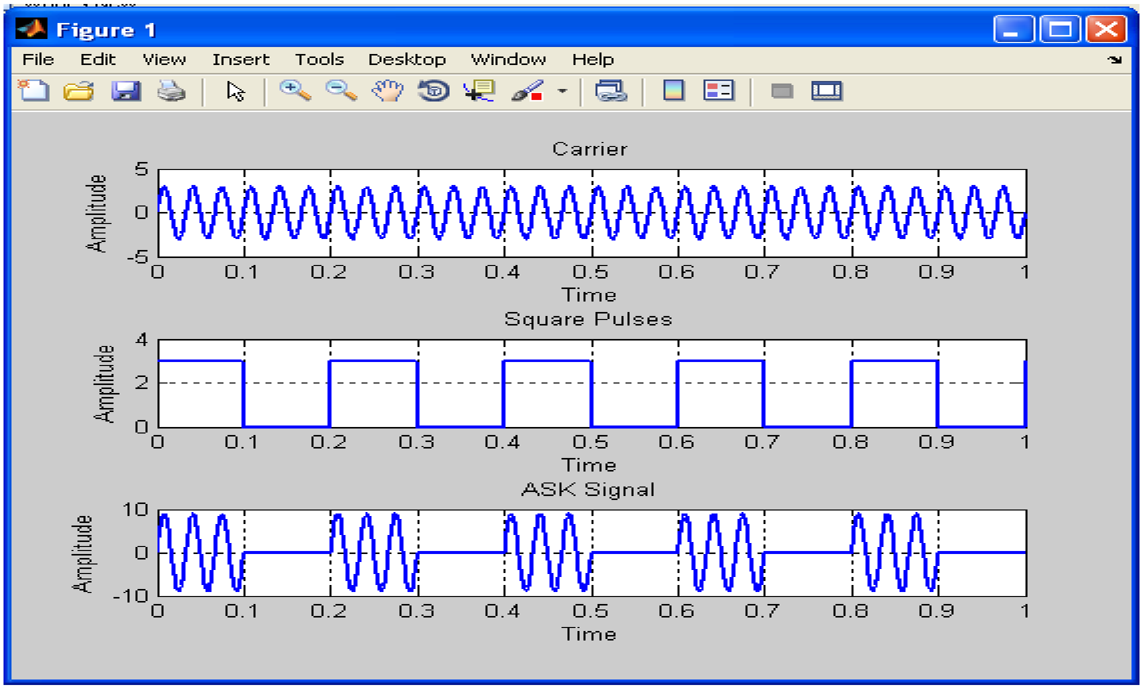
1. Start the program
2. Enter the MATLAB 7.8 software
3. Create a new file and type a program in edit file and save the program
4. Execute the program and plot the waveform
5. Stop the program

MATLAB PROGRAM FOR ASK

```
F1=input('Enter the frequency of carrier=');
F2=input('Enter the frequency of pulse=');
A=3; %Amplitude
t=0:0.001:1;
x=A.*sin(2*pi*F1*t); %Carrier Sine wave
u=A/2.*square(2*pi*F2*t)+(A/2); %Square wave message
v=x.*u;
subplot(3,1,1);
plot(t,x);
xlabel('Time');
ylabel('Amplitude');
title('Carrier');
grid on;
subplot(3,1,2);
plot(t,u);
xlabel('Time');
ylabel('Amplitude');
title('Square Pulses');
grid on; subplot(3,1,3);
plot(t,v);
xlabel('Time');
ylabel('Amplitude');
title('ASK Signal');
grid on;
```

OUTPUT WAEFORM:

Enter the frequency of carrier=30
Enter the frequency of pulse=5



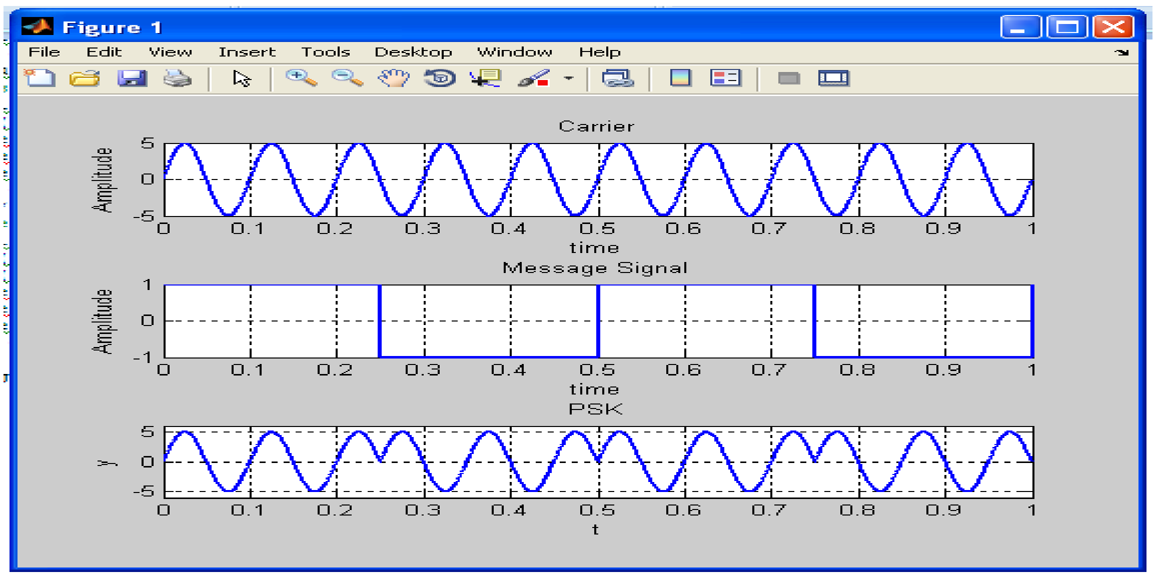
MATLAB PROGRAM FOR PSK

```
clear all;
clc;
close all;
set(0,'defaultlinelinewidth',2);
A=5;
t=0:.001:1;
f1=input('Carrier Sine wave frequency =');
f2=input('Message frequency =');
x=A.*sin(2*pi*f1*t);
%Carrier Sine
subplot(3,1,1);
plot(t,x);
xlabel('time');
ylabel('Amplitude');
title('Carrier');
grid on;
u=square(2*pi*f2*t);
%Message signal
subplot(3,1,2);
plot(t,u);
xlabel('time');
```

```
ylabel('Amplitude');
title('Message Signal');
grid on;
v=x.*u;
%Sine wave multiplied with square wave
subplot(3,1,3);
plot(t,v);
axis([0 1 -6 6]);
xlabel('t');
ylabel('y');
title('PSK');
grid on;
```

OUTPUT WAEFORM:

Carrier Sine wave frequency =10
Message frequency =2



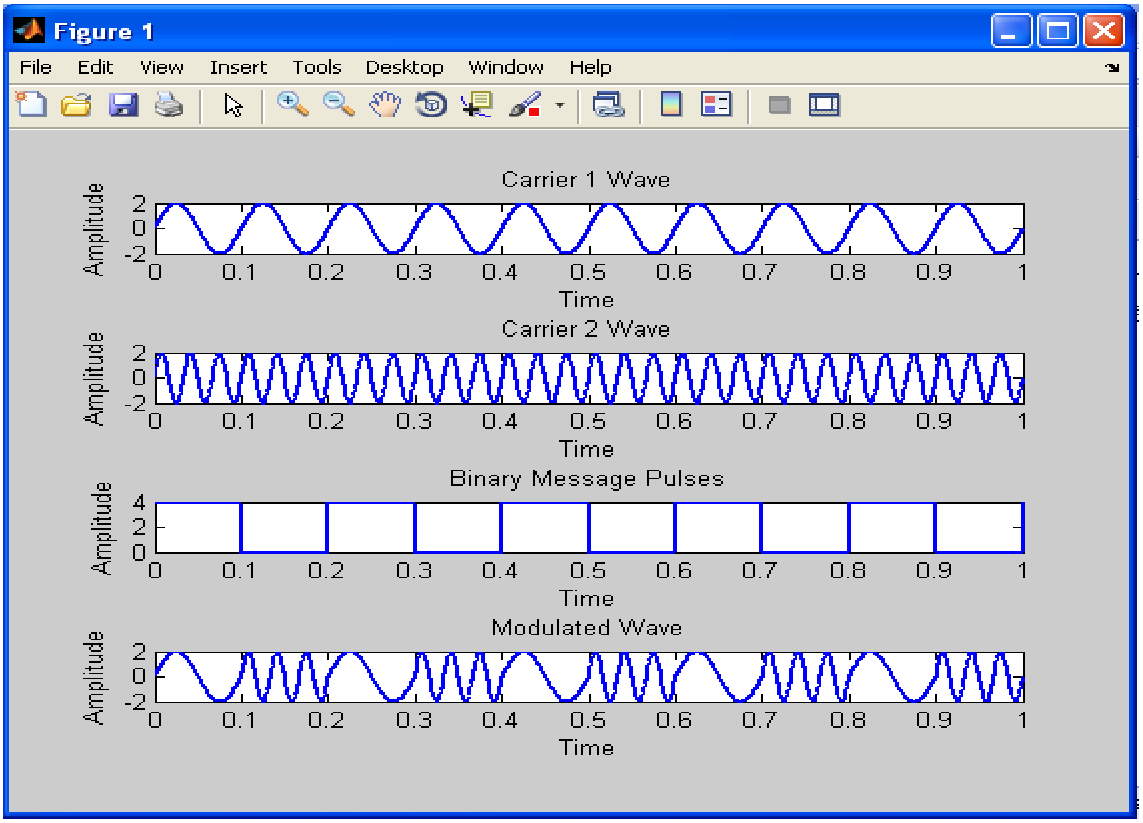
MATLAB PROGRAM FOR FSK

```
clc %for clearing the command window
close all %for closing all the window except command window
clear all %for deleting all the variables from the memory
fc1=input('Enter the freq of 1st Sine Wave carrier:');
fc2=input('Enter the freq of 2nd Sine Wave carrier:');
fp=input('Enter the freq of Periodic Binary pulse (Message):');
amp=input('Enter the amplitude (For Both Carrier & Binary Pulse Message):');
amp=amp/2;
t=0:0.001:1; % For setting the sampling interval
c1=amp.*sin(2*pi*fc1*t);% For Generating 1st Carrier Sine wave
c2=amp.*sin(2*pi*fc2*t);% For Generating 2nd Carrier Sine wave
subplot(4,1,1); %For Plotting The Carrier wave
plot(t,c1)
xlabel('Time')
```

```
ylabel('Amplitude')
title('Carrier 1 Wave')
subplot(4,1,2) %For Plotting The Carrier wave
plot(t,c2)
xlabel('Time')
ylabel('Amplitude')
title('Carrier 2 Wave')
m=amp.*square(2*pi*fp*t)+amp;%For Generating Square wave message
subplot(4,1,3) %For Plotting The Square Binary Pulse (Message)
plot(t,m)
xlabel('Time')
ylabel('Amplitude')
title('Binary Message Pulses')
for i=0:1000 %here we are generating the modulated wave
   if m(i+1)==0
      mm(i+1)=c2(i+1);
      else
      mm(i+1)=c1(i+1);
   end
end
subplot(4,1,4) %For Plotting The Modulated wave
plot(t,mm)
xlabel('Time')
ylabel('Amplitude')
title('Modulated Wave')
```

OUTPUT WAEFORM:

Enter the freq of 1st Sine Wave carrier:10
Enter the freq of 2nd Sine Wave carrier:30
Enter the freq of Periodic Binary pulse (Message):5
Enter the amplitude (For Both Carrier & Binary Pulse Message):4



RESULT:

Thus the simulation of ASK, FSK and BPSK has been executed and verified using MATLAB 7.8 Software.

EX. NO: 11
DATE:

SIMULATION OF DPSK, QPSK AND QAM

AIM

To Simulate the DPSK, QPSK and QAM using MATLAB 7.8 Software.

SOFTWARE / EQUIPMENTS REQUIRED

- PC having MATLAB SOFTWARE

PROCEDURE

1. Start the program
2. Enter the MATLAB 7.8 Software
3. Create a new file and type a program in edit file and save the program
4. Execute the program and plot the waveform
5. Stop the program

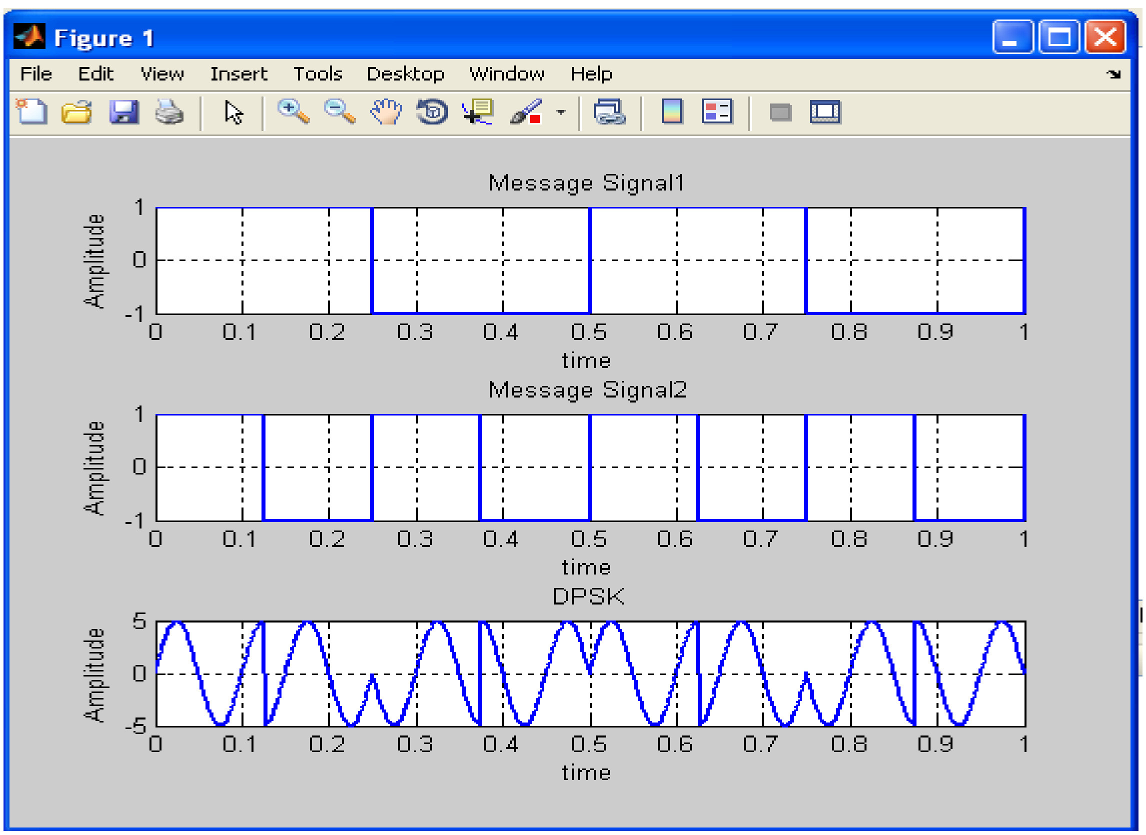
MATLAB PROGRAM FOR DPSK

```
clear all;
clc;
close all;
set(0,'defaultlinelinewidth',2);
A=5;
t=0:.001:1;
f1=input('Message frequency1 =');
f2=input('Message frequency2 =');
f3=input('Carrier sine wave frequency =');
v=square(2*pi*f1*t);
%Message signal1
subplot(3,1,1);
plot(t,v);
xlabel('time');
ylabel('Amplitude');
title('Message Signal1');
grid on;
 u=square(2*pi*f2*t);
%Message signal2
subplot(3,1,2);
plot(t,u);
xlabel('time');
ylabel('Amplitude');
title('Message Signal2');
grid on;
```

```
x=A.*sin(2*pi*f3*t); %Carrier Sine
subplot(3,1,3);
plot(t,x);
xlabel('time');
ylabel('Amplitude');
title('Carrier Signal');
grid on;
 z=x.*u; %Sine wave multiplied with square wave
subplot(3,1,3);
plot(t,z);
xlabel('time');
ylabel('Amplitude');
title('DPSK');
grid on;
```

OUTPUT WAVEFORM:

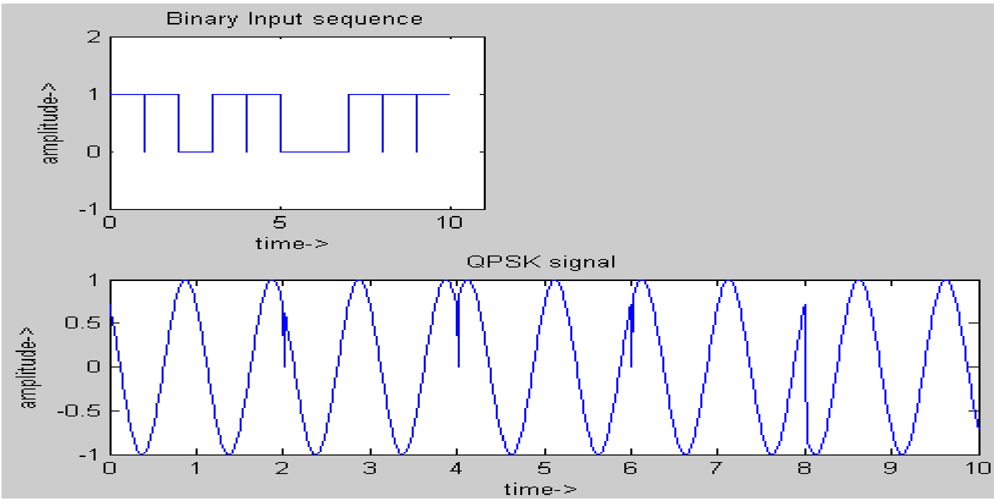
Message frequency1 =2
Message frequency2 =4
Carrier sine wave frequency =10



MATLAB PROGRAM FOR QPSK

```
x=randint(1,10);%binary input generation
%NRZ pulse shaping
i=1;
t=0:.01:length(x);
for k=1:length(t)
if t(k)<=i
y(k)=x(i);
else
i=i+1;
end
end
subplot(221);plot(t,y);axis([0 length(x)+1 -1 2]);title('Binary Input sequence');
xlabel('time->');ylabel('amplitude->');
%Mapping
yy=[];
for i=1:2:length(x)
if x(i)==i&&x(i+1)==1
    y=(1/sqrt(2))+j*(1/sqrt(2));
elseif x(i)==1 && x(i+1)==1
    y=-(1/sqrt(2))+j*(1/sqrt(2));
elseif x(i)==1 && x(i+1)==0
    y=(1/sqrt(2))-j*(1/sqrt(2));
end
yy=[yy y];
end
%inphase and quadrature carrier generation
T=2;
i=1;
t=0:.01:length(x);%time duration of the signal is bit duration
c1=sqrt(2/T)*cos(2*pi*1*t);
c2=sqrt(2/T)*sin(2*pi*1*t);
m=2:2:length(x);
for j=1:length(t);
    if t(j)<=m(i)
        op(j)=real(yy(i))*c1(j)-imag(yy(i))*c2(j);
    else
        i=i+1;
    end
end
subplot(212);plot(t,op);xlabel('time->');ylabel('amplitude->');title('QPSK signal');
```

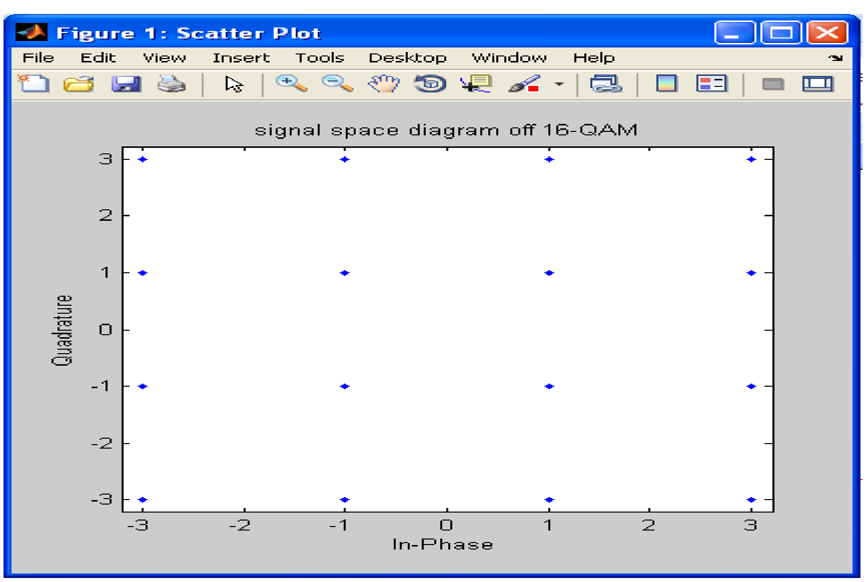
OUTPUT WAVEFORM:

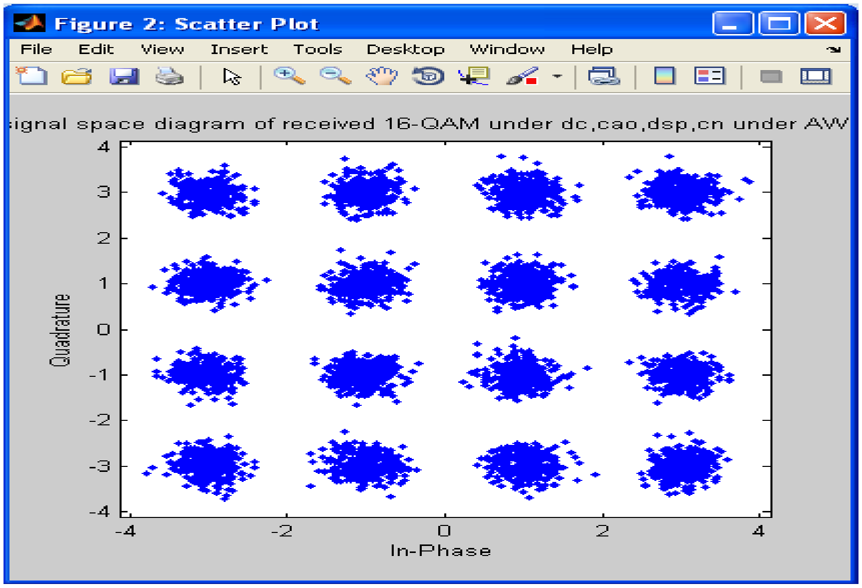


MATLAB PROGRAM FOR QAM

```
M=16;
x=randint (5000, log2(M));
x1=bi2de (x,'left-msb');
xmod=qammod (x1,M);
scatterplot (xmod);
title('signal space diagram off 16-QAM');
snr=10;
rx=awgn(xmod,snr);
scatterplot(rx);
title('signal space diagram of received 16-QAM under dc,cao,dsp,cn under AWGN');
```

OUTPUT WAVE FORM





RESULT:

Thus the simulation of DPSK, QPSK and QAM has been executed and verified using MATLAB 7.8 Software.

SSM COLLEGE OF ENGINEERING,KOMARAPALAYAM-638 183
DEPARTMENT OF ELECTRONICS AND COMMUNICATION ENGINEERING

EX. NO: 12
DATE:

SIMULATION OF ERROR CODING SCHEMES (LINEAR BLOCK CODES)

AIM

To simulate error control coding scheme linear block codes using MATLAB 7.8 Software.

SOFTWARE / EQUIPMENTS REQUIRED

- PC having MATLAB SOFTWARE

PROCEDURE

1. Start the program
2. Enter the MATLAB 7.8 Software
3. Create a new file and type a program in edit file and save the program
4. Execute the program and plot the waveform
5. Stop the program

MATLAB PROGRAM FOR LINEAR BLOCK CODE

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

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

Output:

```
enter the generator matrix:[1 0 0 0 1 0 1; 0 1 0 0 1 1 1; 0 0 1 0 1 1 0; 0 0 0 1 0 1 1]
G=the order of the linear block code for given generator matrix is: n = 7
 k = 4
syndrome of a given codeword is:
 s = 0 0 1
the error is in bit: i = 7
 the corrected codeword is: r = 1 0 0 0 1 0 1
the possible code words are:
 c = 0 0 0 0 0 0 0 0 0 0 1 0 1 1 0 0 1 0 1 1 0 0 0 1 1 1 0 1 0 1 0 0 1 1 1 0 1 0 1 1 0 0 0 1 1 0 0 0 1 0 1 1 1 0 1 0 1 0 0 0 1 0 1
1 0 0 1 1 1 0 1 0 1 0 0 1 1 1 0 1 1 0 0 0 1 1 0 0 0 1 0 1 1 0 1 0 0 1 1 1 1 0 1 0 0 1 1 1 1 1 1 1

the minimum hamming distance dmin for given block code d_min = 3
enter the received code word:[1 0 0 0 1 0 0]
r =1 0 0 0 1 0 0
hamming code
ht = 1 0 1 1 1 1 1 1 0 0 1 1 1 0 0 0 1 0 0 0 1
```

CYCLIC CODE

```
%Generation of parity check matrix and generator matrix for a (7, 4) Hamming code. [h,g,n,k] =
hammgen(3);
%Generation of parity check matrix for the generator polynomial
g(x) = 1+x+x3
h1 = hammgen(3,[1 0 1 1])
 %Computation of code vectors for a cyclic code clc; close all;
n=7; k=4;
msg=[1 0 0 1; 1 0 1 0; 1 0 1 1];
 code = encode(msg,n,k,'cyclic');
msg code %Syndrome decoding clc;
close all;
q=3;
n=2^q-1;
k=n-q;
 parmat = hammgen(q); % produce parity-check matrix
trt = syndtable(parmat)% produce decoding table
recd = [1 0 1 1 1 1 0 ] %received vector
syndrome = rem(recd * parmat',2);
 syndrome_de = bi2de(syndrome, 'left-msb'); %convert to decimal
disp(['Syndrome = ',num2str(syndrome_de),..... ' (decimal), ',num2str(syndrome),' (binary) ']);
corrvect = trt(1+syndrome_de, :);%correction vector correctedcode= rem(corrvect+recd,2);
parmat
corrvect
correctedcode
```

```
recd =

     1     0     1     1     1     1     0

Syndrome = 7 (decimal), 1  1  1 (binary)

parmat =

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


corrvect =

     0     0     0     0     0     1     0


correctedcode =

     1     0     1     1     1     0     0
```

RESULT:

Thus the error control coding scheme –linear block code and cyclic error control code has been executed and verified using MATLAB 7.8.