SRM Institute of Science and Technology Faculty of Engineering and Technology Department of Electronics and Communication Engineering
18ECC205J ANALOG AND DIGITAL COMMUNICATION LAB **Fifth Semester, 2021-22 (odd semester)**

Name :Pushpal Das

Register No. :RA1911004010565.

Day / Session : 2/3&4

Venue :Online (G-meet)

Title of Experiment :1.AMPLITUDE MODULATION AND DEMODULATION. Date of Conduction :14th July , 2021

Date of Submission :27th July, 2021

Particulars Max. Marks	**Marks Obtained**
Pre-lab questions 05	
In-lab experiment 10	
Post-lab questions 15	
Total 30	

REPORT VERIFICATION

Date :

Staff Name : Dr. P. Malarvezhi / Mrs. Harisudha

Signature :

1. AMPLITUDE MODULATION AND DEMODULATION

1.1 Objective

To construct an amplitude modulator circuit using transistor with V_c=50mv, V_m=8v to satisfy under modulation condition and generate amplitude modulated signal. Calculate the modulation index and also demodulate using envelope detector and reconstruct the modulating signal. Simulate Amplitude Modulation (AM) wave in time domain using SCILAB.

1.2 Hardware Required

S.No	Equipment/Component name Specifications/Value	Quantity
1	Cathode Ray Oscilloscope (0 – 20MHz)	1
2	Audio Frequency Oscillator (0-2) MHz	2
3	Regulated power supply (0 -30V), 1A	1
4	Resistors 1.5K Ω 10 K Ω 20 K Ω 100 K Ω	2 3 1 2

5 Capacitors 0.1 µf

1

0.01 µf

1

0.001 µf

3

22 µf

1

6 Semiconductor Device(Transistor) BC108 1 7 Semiconductor Device(Diode) OA79 1

1.3 Theory

Modulation is defined as the process by which some characteristics of a carrier signal is varied in accordance with a modulating signal. The base band signal is referred to as the modulating signal and the output of the modulation process is called as the modulation signal.

AM Modulation Circuit Diagram

Specifications

R1 = R2 = R5 = 10KΩ; R3 = 1.5KΩ; R4 = 20KΩ; C1 = 0.01μF; C2 = 0.001μF; C3 = 0.1 μf; Vc = 50mV; fc = 500KHZ; Vm = 8V; fm = 1KHZ; V_{CC} = 30V

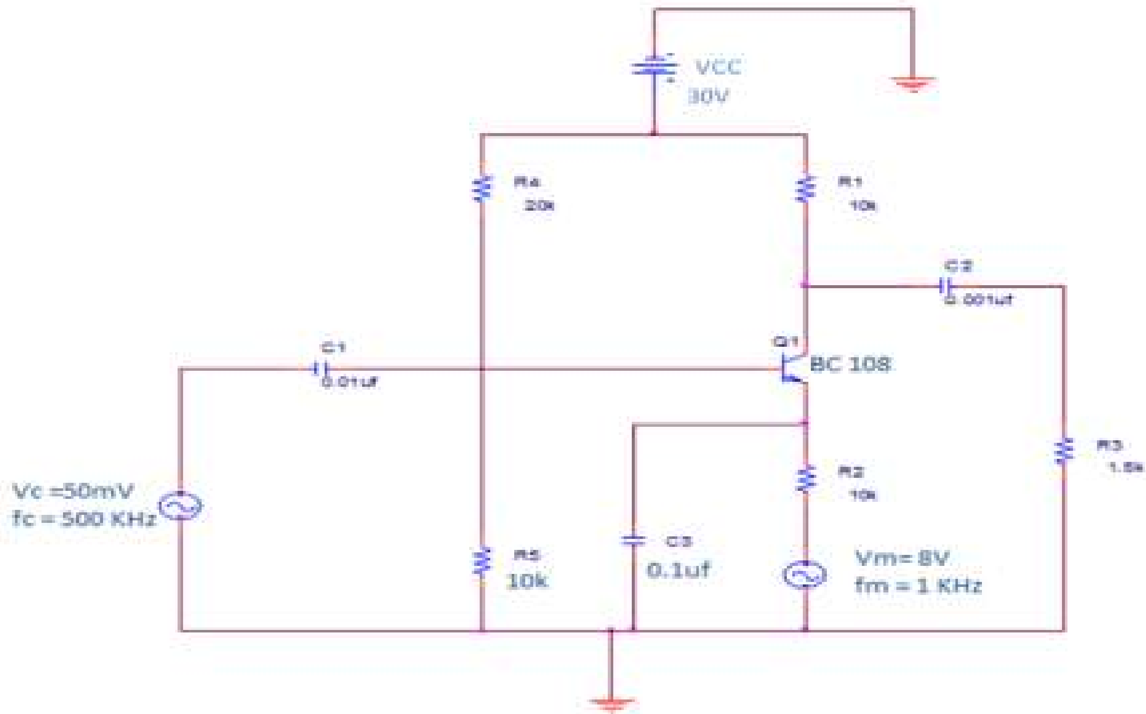


Fig. 1.1 AM Modulator Circuit

1.3.1 Amplitude Modulation

Amplitude modulation is defined as the process in which amplitude of the carrier wave is varied in accordance with the instantaneous values of the modulating signal. The envelope of the modulating wave has the same shape as the base band signal provided the following two requirements are satisfied

1. The carrier frequency f_c must be much greater then the highest frequency components f_m of the message signal m (t)

i.e. $f_c >> f_m$

2. The modulation index must be less than unity. If the modulation index is greater than unity, the carrier wave becomes over modulated.

1.3.2 Amplitude Demodulation

The process of detection provides a means of recovering the modulating Signal from modulating signal. Demodulation is the reverse process of modulation. The envelope detector circuit is employed to separate the carrier wave and eliminate the side bands. Since the envelope of an AM wave has the same shape as the message, independent of the carrier frequency and phase, demodulation can be accomplished by extracting envelope.

An increased time constant RC results in a marginal output follows the modulation envelope. A further increase in time constant the discharge curve become horizontal if the rate of modulation envelope during negative half cycle of the modulation voltage is faster than the rate of voltage RC combination ,the output fails to follow the modulation resulting distorted output is called as diagonal clipping : this will occur even high modulation index.

The depth of modulation at the detector output greater than unity and circuit impedance is less than circuit load (R_l>Z_m) results in clipping of negative peaks of modulating signal. It is called “negative clipping “

AM Demodulation Circuit Diagram

Specifications

C1=0.001µf, C2=22 µf, C3=0.001µf, R1=100KΩ and R2=100KΩ.

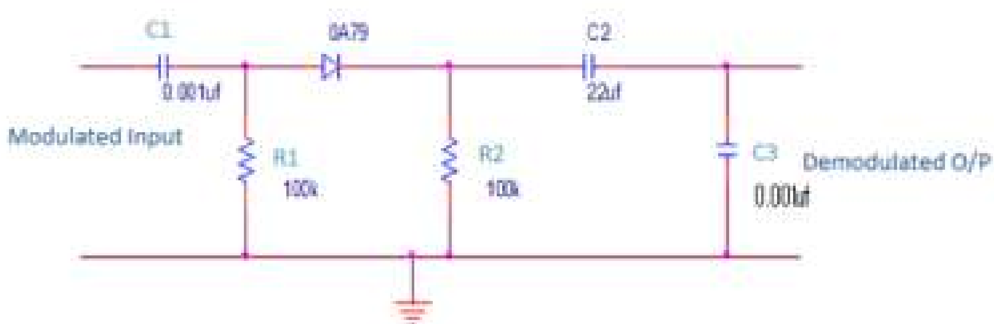


Fig. 1.2 AM Demodulator Circuit

1.4 Pre Lab Questions

1. Define Modulation.
2. Why Modulation is necessary for communication system.
3. What is Baseband signal?
4. Differentiate analog and Digital Modulation.
5. Define Amplitude Modulation and Demodulation?

1.5 Lab Procedure

1.5.1 Amplitude Modulation

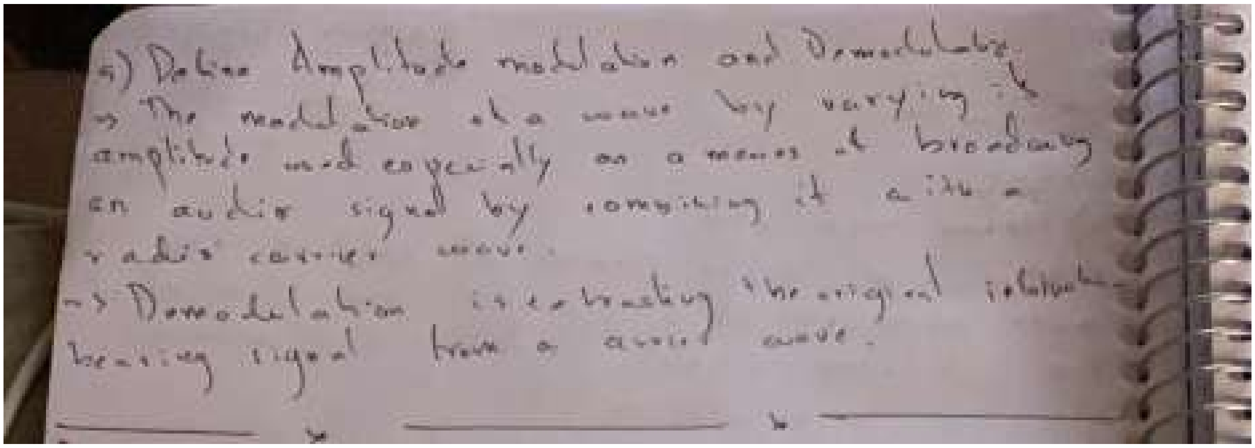
1. The circuit connection is made as shown in the circuit.
2. The power supply is connected to the collector of the transistor. 3. Set the input signal f_m as 1 KHz and 8 volt sinusoidal signal in AFO 4. Set the carrier signal f_c as 500 KHz and 50 millivolt sinusoidal signal in AFO 5. The Amplitude Modulated Output is taken from the collector of the Transistor. 6. Note down E_{max} and E_{min} from the Output waveform.
7. Calculate modulation index using the formula.

Modulation index m = $\frac{\text{�������� − ��������}}{\text{�������� + ��������}}$

8. Plot the input signals and obtained AM output waveforms in the graph sheet

1.5.2 Amplitude Demodulation

1. The circuit connections are made as shown in the circuit diagram.
2. The amplitude modulated signal from AM generator is given as input to the demodulator circuit.
3. The demodulated output is observed on the CRO
4. Plot the obtained AM demodulated output waveforms in the graph sheet


5) Define Amplitude modulation and Demodulation
→ The modulation of a wave by varying its amplitude used especially as a means of broadcasting an audio signal by combining it with a radio carrier wave.
→ Demodulation is extracting the original information-bearing signal from a carrier wave.

1.6 Observation - Hardware

Signal name	Amplitude Frequency Time period
Modulating signal	
Carrier signal	
Modulated signal	
Demodulated signal	

1.6.1 Model graph

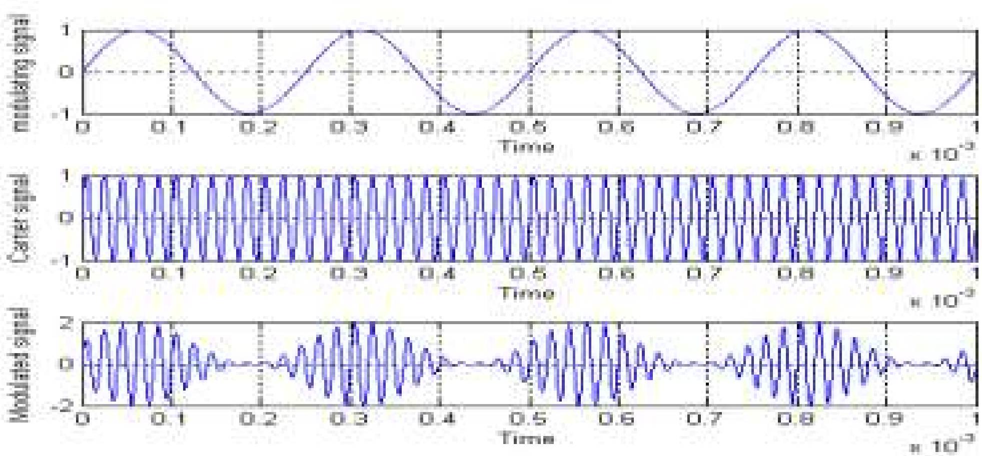


Fig. 1.3 Amplitude Modulation waveforms

Fig. 1.4 AM Demodulated Waveform

1.7. VIRTUAL LAB

1.7.1 Equipment / Apparatus

SCILAB Software 6.0.2

1.7.2 Exercise

With the following data, use SCILAB to generate and display an Amplitude Modulation signal.

Carrier frequency fc = 5 kHz
Amplitude Carrier frequency = Ac = 9 V
Sampling time = 100 ms
Modulating frequency =500 Hz
Amplitude Modulating Signal Vm = 4.5 V

Solution

```
// generate carrier signal
fc = 5000;
Ac = 9;
t = linspace(0,10*(10^(-3)),500);
Vc = Ac*sin(((2*%pi)*fc)*t);
subplot(411)
plot(t,Vc)

// generate modulating signal
fm = 500;
Am = 4.5;
Vm = Am*sin(((2*%pi)*fm)*t);
subplot(412)
plot(t,Vm)

// generate modulation signal with index modulation m = 0.5
```

```
m = Am/Ac;
Vt = (Ac*(1+m*sin(((2*%pi)*fm)*t))) .*sin(((2*%pi)*fc)*t);
subplot(413)
plot(t,Vt)
```

1.7.3 Observation – Software

Signal name	Amplitude Frequency Time period
Modulating signal	
Carrier signal	
Modulated signal	
Demodulated signal	

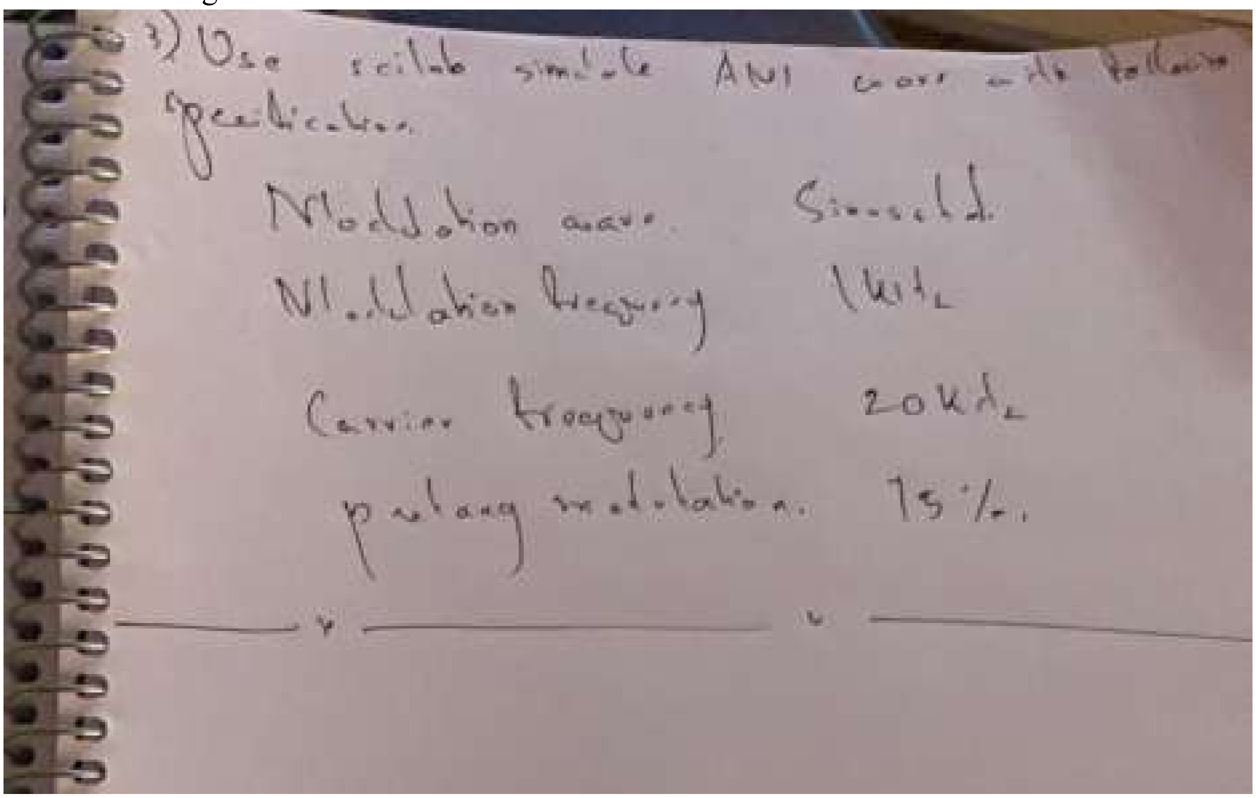
1.7.4 Model graph

Fig. 1.3 Simulated Amplitude Modulation waveforms

1.8 Post Lab Questions

1. What will happen, if modulation index is greater than 100%?
2. What happens to AM signal if $m_a<1$ & $m_a=1$?
3. Use SCILAB to produce AM wave with the following specification

   Modulating Wave Sinusodal
   Modulation Frequency 1kHz
   Carrier frequency 20kHz
   Percentage Modulation 75%



SCILAB OUTPUT :

1.9 Lab Result

Thus the amplitude modulation and demodulation were performed and the modulation index for various modulating voltage were calculated and simulated using Scilab.

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Name : Pushpal Das

Register No. : RA1911004010565

Day / Session : 2/3&4

Venue : Online(G-Meet)

Title of Experiment : 2. DSB-SC MODULATION AND DEMODULATION

Date of Conduction : 22 JULY 2021

Date of Submission : 04 AUG 2021

Particulars	**Max. Marks**	**Marks Obtained**
Pre-lab questions	05	
In-lab experiment	10	
Post-lab questions	15	
Total	**30**	

REPORT VERIFICATION

Date : 02/08/2021

Staff Name : Dr. P. Malarvizhi / Mrs. Harisudha

Signature :

2. DSB-SC MODULATION AND DEMODULATION

2.1 OBJECTIVE

To generate DSB-SC AM signal using balanced modulator. Calculate the modulation index and reconstruct the modulating signal using synchronous detector. Simulate DSBSC modulation in time domain using SCILAB.

2.2 HARDWARE REQUIRED:

S.No	Equipment/Component name	Specifications/Value	Quantity
1	Cathode Ray Oscilloscope	(0 – 20MHz)	1
2	Audio Frequency Oscillator	(0-2) MHz	2
3	Scientech Trainer kit	Scientech 2209	1

2.3 THEORY

Double Sideband-Suppressed carrier modulation, in which the transmitted power consists of only the upper and lower sidebands. It is basically an amplitude modulation wave without the carrier, therefore reducing power waste, giving it 50% efficiency. This is an increase compared to normal AM transmission, (DSB) has a maximum efficiency of 33.333%, since 2/3 of the power is in the carrier which carries no intelligence, and each sideband carries the same information. Single Side Band (SSB) Suppressed Carrier is 100% efficient.

2.3.1 BALANCED MODULATOR

The principle of operation of a balanced modulator states that if two signals at different frequencies are passed through a non-linear resistance then at the output, we get an AM signal with suppressed carrier. The device having a non-linear resistance can be a diode or a JFET or even a bipolar transistor.

BLOCK DIAGRAM

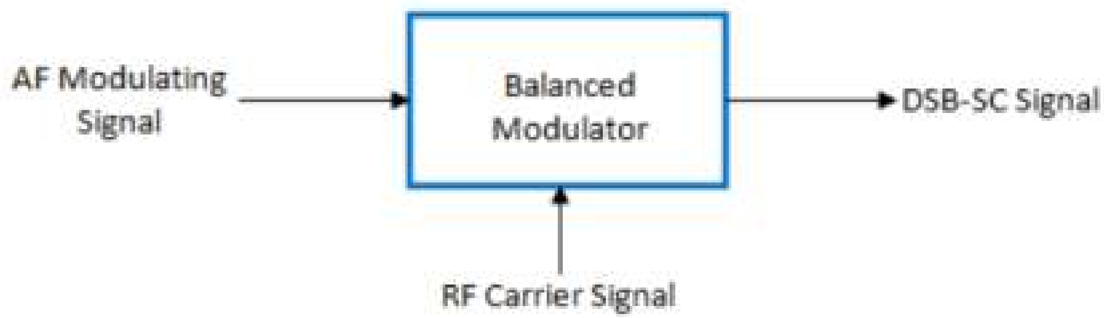


Fig 2.1 Balanced modulator

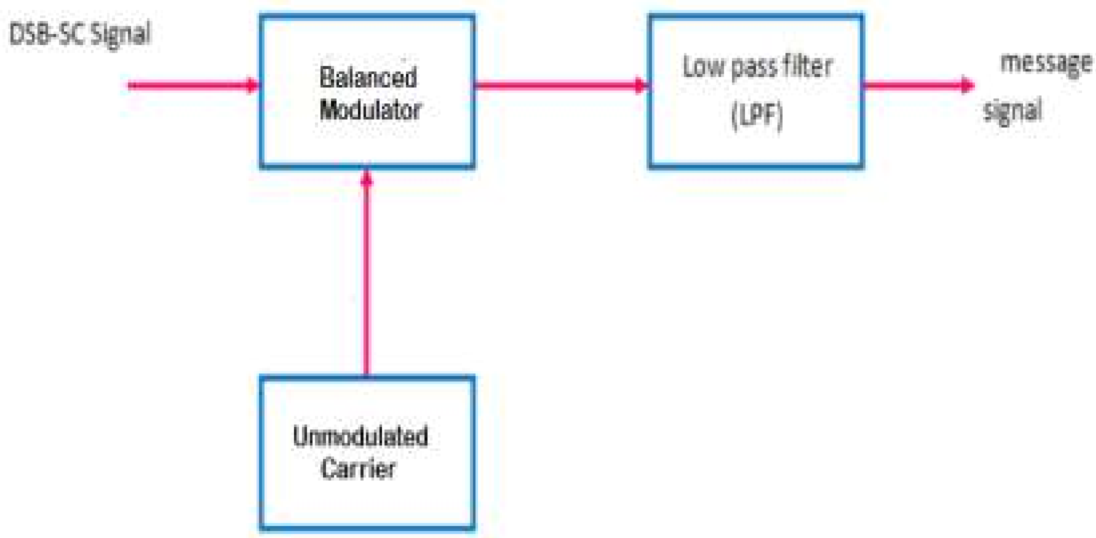


Fig 2.2 Synchronous detector

2.3.2 SYNCHRONOUS DETECTOR

Synchronous detection is used for the detection or demodulation of amplitude modulation (AM). Demodulation is performed by multiplying the modulated carrier by a sine wave that is phase locked to the incoming carrier. Another characteristic of synchronous detectors is that they are phase sensitive. The amplitude of the demodulated signal is a function of the relative phases of the incoming carrier and the carrier generated inside the receiver. In any receiver a key element is the detector. Its purpose is to remove the modulation from the carrier to give the audio frequency representation of the signal. This can be amplified by the audio amplifier ready to be converted into audible sound by headphones or a loudspeaker.

When an AM signal or already modulated signal is applied to linear input of another balance modulator (AM generator), and an unmodulated Carrier is applied to the switching input, neglecting the higher order harmonics of the switching waveform, We obtain the output that is has one side modulation only. And when we pass it to the Low pass Filter, the low pass Filter will only pass the lower frequency and block all other higher frequency. Hence over message signal which is of lower frequency then the carrier signal will pass from Low pass filter and the carrier frequency will be blocked.

2.4 PRE LAB QUESTIONS

1. What is the difference between DSB-SC and SSB-SC?
2. What are the applications of DSBSC?
3. Write the generation methods of DSB-SC AM.
4. Draw the spectrum of DSB-SC AM signal in which the modulating signal $E_m \cos\omega_m t$ modulates a carrier signal $E_c \cos\omega_c t$.
5. What is the percentage of power saving for DSBSC when compared with AM having 100% depth of modulation?

Pre Lab:-

Experiment 2

RA1911004010555
Pushpal Das

Pre-lab

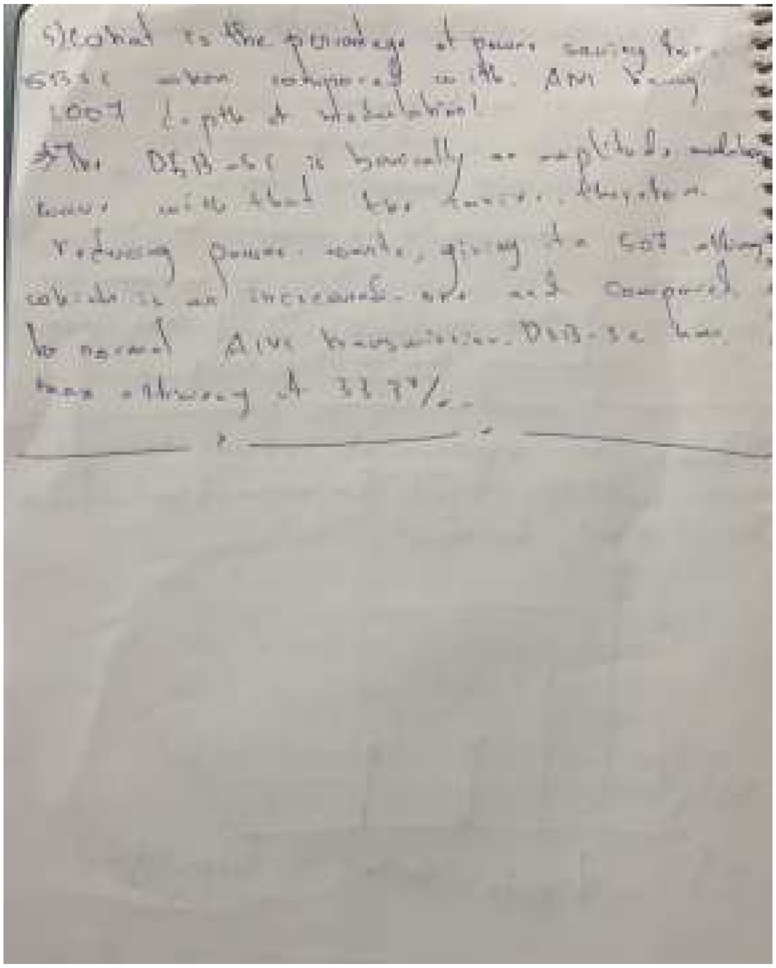
1) What is the difference b/w DSB SC and SSB-SC?

→ DSB-SC is double side band suppressed carrier modulation and SSB-SC is single band suppressed carrier modulation. When amplitude modulation is carried out, the modulating signal spectrum is shifted to the carrier frequency band and SSB-SC is DSB-SC ~~sent~~ with the only side band.

2) What are the applications of DSBSC?

→ During the transmission of binary data DSB-SC system is used in phase shift keying method.

→ In order to transmit 2 channel stereo signals, DSB signals are used in Television and FM broadcasting.


[illegible] to the percentage of power saving for DSB-SC when compared with AM having 100% depth of modulation.
The DSB-SC is basically an amplitude modulation wave with [illegible] the carrier [illegible] reducing power waste, giving it a 50% efficiency which is an increase over [illegible] compared to normal AM transmitter. DSB-SC has max efficiency of 33.3%.

MODEL GRAPH

DSB-SC MODULATION

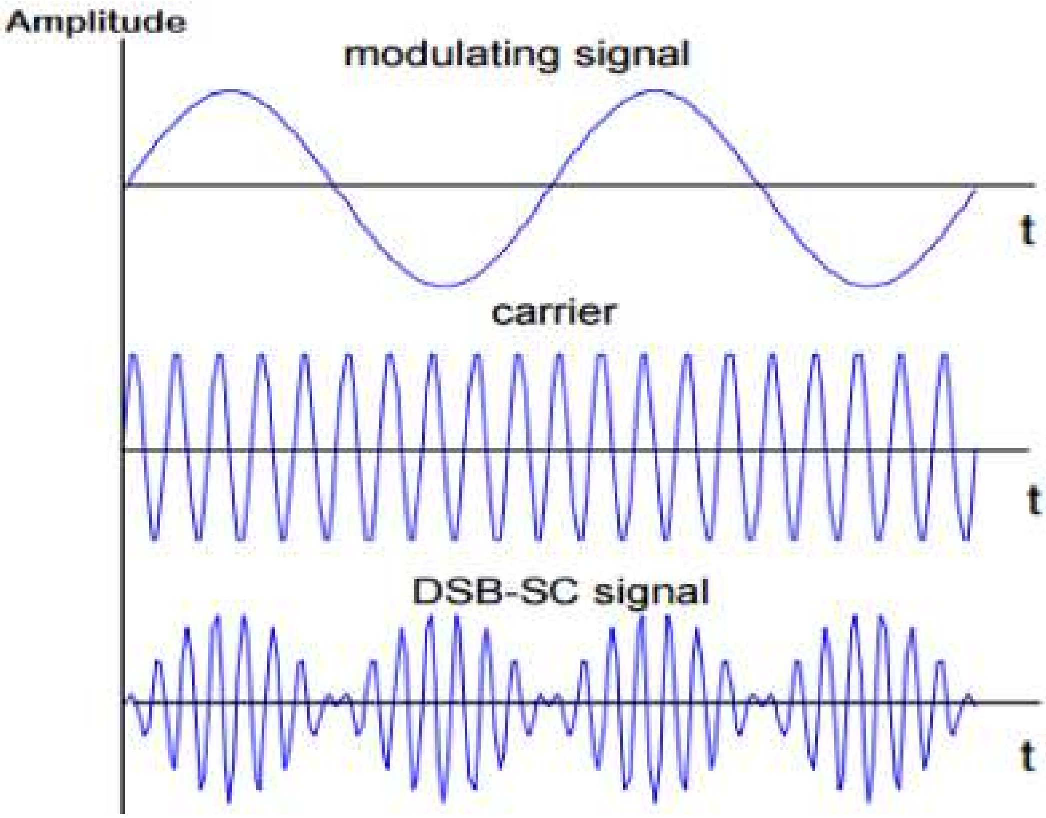


Fig 2.3 DSB-SC modulation waveforms

DSB-SC DEMODULATION

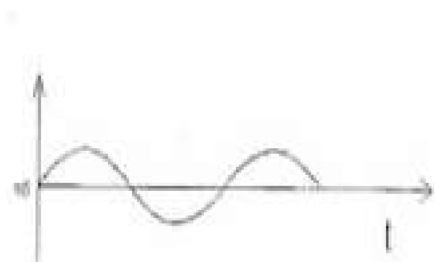


Fig 2.4 DSB-SC demodulation waveform

2.5 LAB PROCEDURE

2.5.1 DSB-SC MODULATION

1. Set the input signal f_m as 2 KHz and 400mV sinusoidal signal in AFO
2. Set the carrier signal f_c as 100 KHz and 400mV sinusoidal signal in AFO
3. The Double sideband suppressed carrier Amplitude Modulated wave is taken from the output of the balanced modulator.

4. Note down E_{max} and E_{min} from the Output waveform.

5. Calculate modulation index using the formula.

Modulation index $m = \frac{Emax - Emin}{Emax + Emin}$

6. Vary the Carrier Adjust pot of first balance modulator to see the 100% modulation. 7. Plot the input signals and obtained DSB-SC AM signal in graph sheet.

2.5.2 DSB-SC DEMODULATION

1. Give the output of First Balance modulator (DSBSC-AM) into message input of second Balanced Modulator.
2. Feed the same carrier signal in Second Balance Modulator also.
3. Observe the Double Modulated output from balanced modulator 2 and Vary the 'Carrier Adjust' Pot 'P2' for perfect one-side modulated wave.
4. Now feed this Double Modulated signal at Low Pass Filter for perfect message signal.
5. Plot the demodulated signal in graph sheet.

2.6 OBSERVATION

DSB-SC MODULATION

Modulating signal				**Carrier signal**			
Signal Type	**Amplitude**	**Time Period**	**Frequency**	**Signal Type**	**Amplitude**	**Time Period**	**Frequency**
Sine wave	2V	5000Hz	0.0002s	Sine wave	10V	5000Hz	0.0002s

Modulated Output			
Signal Type	**E_{max}**	**E_{min}**	**Modulation index**
AM	15	5	0.5

2.7. VIRTUAL LAB

2.7.1 Equipment / Apparatus

SCILAB Software 6.0.2

2.7.2 Exercise

With the following data, use SCILAB to generate and display a DSBSC signal.

Carrier frequency fc = 5 kHz
Amplitude Carrier frequency = Ac = 10 V
Sampling time = 10 ms
Modulating frequency =500 Hz
Amplitude Modulating Signal Vm = 5 V

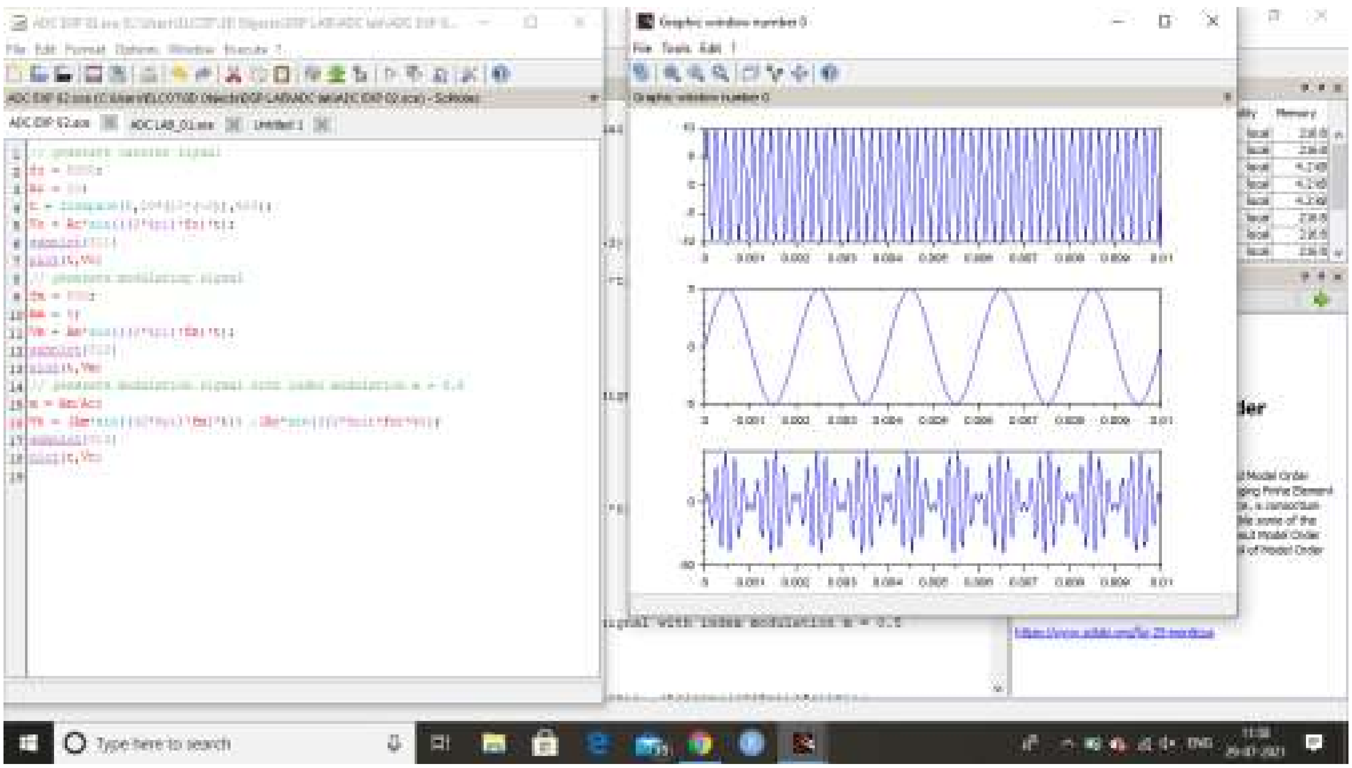
Solution

```
// generate carrier signal
fc = 5000;
Ac = 10;
t = linspace(0,10*(10^(-3)),500);
Vc = Ac*sin(((2*%pi)*fc)*t);
subplot(311)
plot(t,Vc)

// generate modulating signal
fm = 500;
Am = 5;
Vm = Am*sin(((2*%pi)*fm)*t);
subplot(312)
plot(t,Vm)

// generate modulation signal with index modulation m = 0.5
m = Am/Ac;
Vt = (Am*sin(((2*%pi)*fm)*t)) .*(Ac*sin(((2*%pi)*fc)*t));
subplot(313)
plot(t,Vt)
```

Simulation Code And Output:-



2.7.3 Observation – Software

Signal name	Amplitude	Frequency	Time period
Modulating signal	5V	500HZ	0.0002S
Carrier signal	10V	5000HZ	0.0002S
Modulated signal	02V	5000HZ	0.0002S

2.7.4 Model graph

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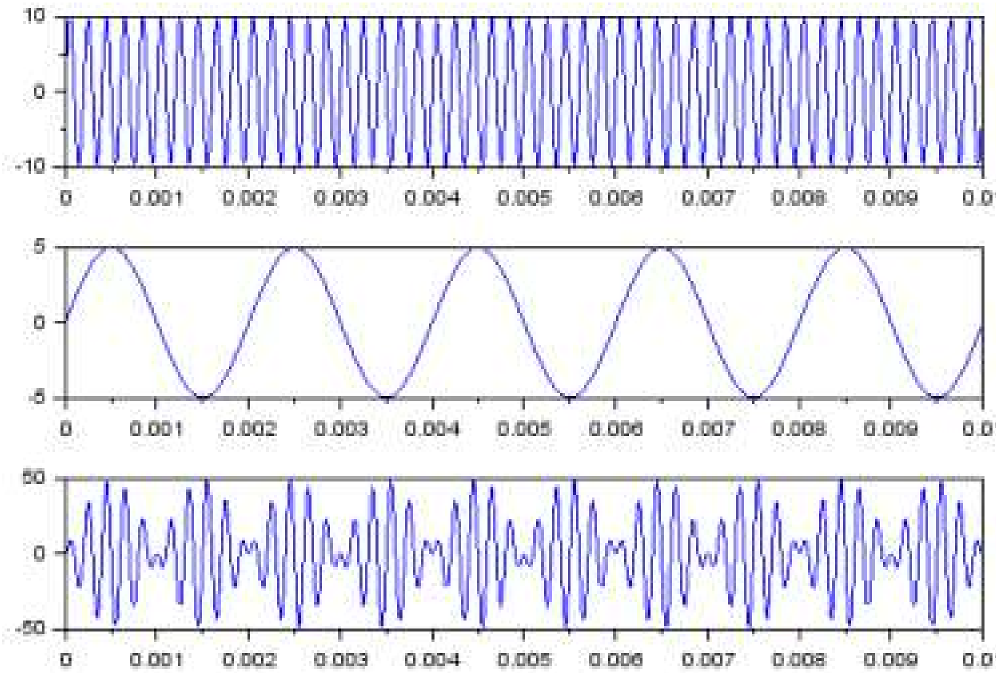


Fig. 1.3 Simulated DSBSC modulated waveforms

2.8 POST LAB QUESTIONS

1. Write the general equation of DSBSC signal.
2. What is quadrature null effect?
3. Use SCILAB to produce AM wave with the following specification

Modulating Wave Sinusodal
Modulation Frequency 1kHz
Carrier frequency 10kHz

Post Lab: -

Post Lab

1) Write the general equation of DSBSC signal

→ $m(t) = A_m \cos(2\pi f_m t)$

$c(t) = A_c \cos(2\pi f_c t)$

Then, the equation of DSB SC signal

$s(t) = m(t) \times c(t)$

$S(t) = A_m A_c \cos(2\pi f_m t) \cos(2\pi f_c t)$

Bandwidth $= (f_c + f_m) - (f_c - f_m) = 2 f_m$

Power saving $= \frac{D.P.}{T} = 66.67\%$

Efficiency = 100%

2) What is quadrature null effect?

→ Quadrature null effect is a phenomenon in amplitude modulation, demodulation circuit occurs if carrier signal generated from local oscillator is not synchronised in phase with i/p signal. The demodulated signal amplitude will be zero, when $\phi = \pm 90$. This is called as quadrature null effect.

3) Use scilab to produce AM wave with the following specifications.

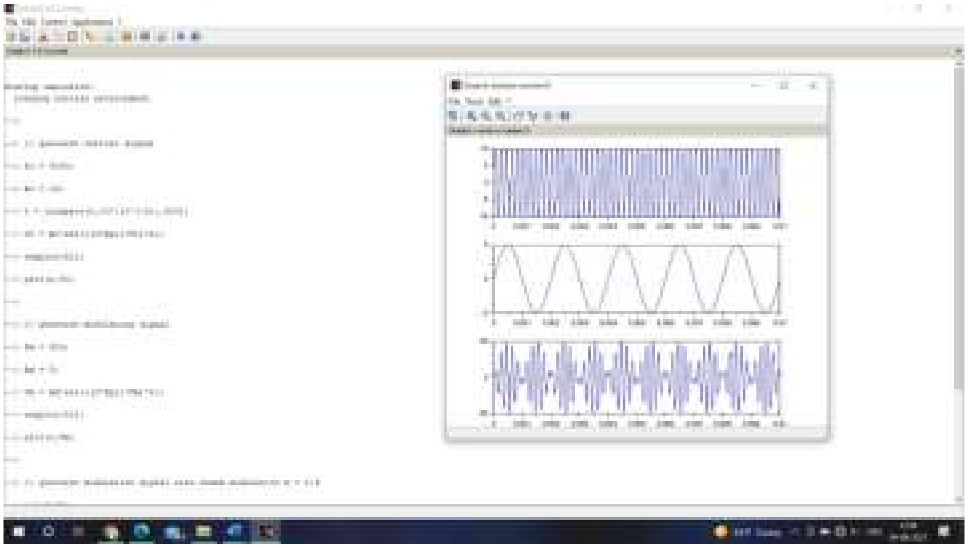
Modulating wave: Sinusoidal

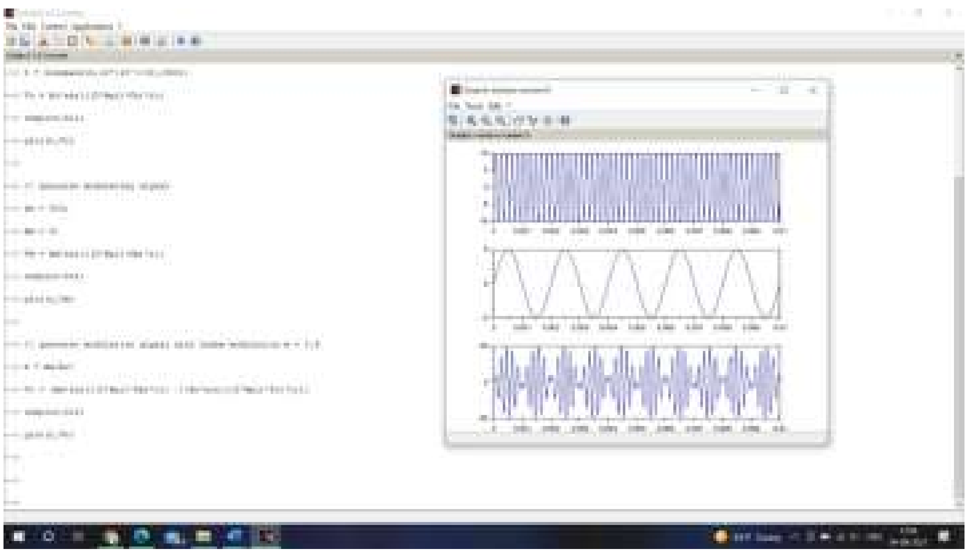
Modulating frequency 1kHz

Carrier frequency 10kHz

3.

SCILAB OUTPUT





2.9 LAB RESULT

Thus the DSB-SC modulation and demodulation was performed and the modulation of DSBSC signal was simulated using Scilab for the given carrier and modulating signal frequency.

SRM Institute of Science and Technology

Faculty of Engineering and Technology

Department of Electronics and Communication Engineering

18ECC205J ANALOG AND DIGITAL COMMUNICATION LAB

Fifth Semester, 2021-22 (odd semester)

Name : **Pushpal Das**

Register No. : **RA1911004010565**

Day / Session : **2/3&4**

Venue : **Online(G-Meet)**

Title of Experiment : **3. FREQUENCY MODULATION & DEMODULATION**

Date of Conduction : **05 AUG 2021**

Date of Submission : **12 AUG 2021**

Particulars	Max. Marks	Marks Obtained
Pre-lab questions	05	
In-lab experiment	10	
Post-lab questions	15	
Total	**30**	

REPORT VERIFICATION

Date : **11/08/2021**

Staff Name : **Dr. P. Malarvizhi / Mrs. Harisudha**

Signature :

3. FREQUENCY MODULATION & DEMODULATION

3.1 Objective

To perform frequency modulation and demodulation process using audio frequency signal with adjustable amplitude & frequency (300Hz-3.4 KHz) with the help of Trainer kit. Simulate the Frequency Modulation (FM) wave in time domain using SCILAB.

3.2 Hardware Required

S.No	Equipment/Component name	Specifications/Value	Quantity
1	Cathode Ray Oscilloscope	(0 – 20MHz)	1
2	Audio Frequency Oscillator	(0-2) MHz	2
3	FM Trainer kit	ST8203	1

3.3 Theory

Frequency modulation is a process of changing the frequency of a carrier wave in accordance with the slowly varying base band signal. The main advantage of this modulation is that it can provide better discrimination against noise. The frequency of the carrier is made to increase as the voltage in the information signal increases and to decreases in frequency as it reduces. Higher the amplitude gives increased frequency.

3.3.1 Frequency Modulator

The FM modulator is used to combine the carrier wave and the information signal. The varactor and reactance modulator are two different methods to perform frequency modulation process. In varactor modulator when the information signal applied to the varactor diode, the capacitance will be increased and decreased in sympathy with the incoming signal. The changing value of capacitance causes the oscillator frequency to increase and decrease under the control of the information signal. The output is therefore a FM signal. In reactance modulator, the changing information signal being applied to the base has the same effect as changing the bias voltage applied to the transistor and this would have the effect of increasing and decreasing the value of this capacitance. As the capacitance is parallel with the tuned circuit the variations in value will cause the frequency of resonance to change and hence the carrier frequency will be varied in sympathy with the information signal input.

Block Diagram

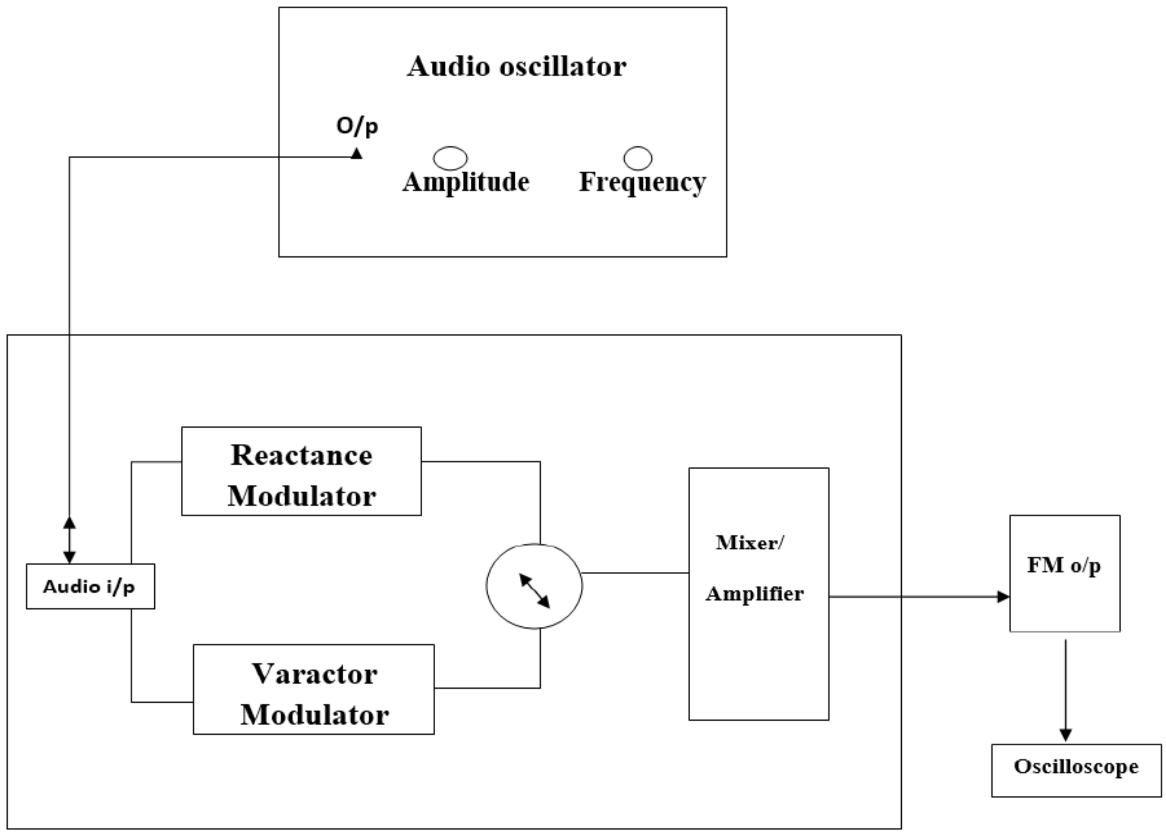


Fig 3.1 Frequency Modulation

3.3.2 FM Demodulator

The extraction of information signal from the modulated signal is done by demodulators. The basic requirement of FM demodulator is therefore to convert frequency changes into changes in voltage, with the minimum amount of distortion. The demodulator converts FM into AM and then to the base band signal (information/audio signal). Foster Seeley uses double tuned RF transformer to convert frequency variations in the received FM signal to amplitude variations. These amplitude variations are then rectified and filtered to provide a dc output voltage. This voltage varies in both the amplitude and polarity as the input signal varies in frequency.

Block Diagram

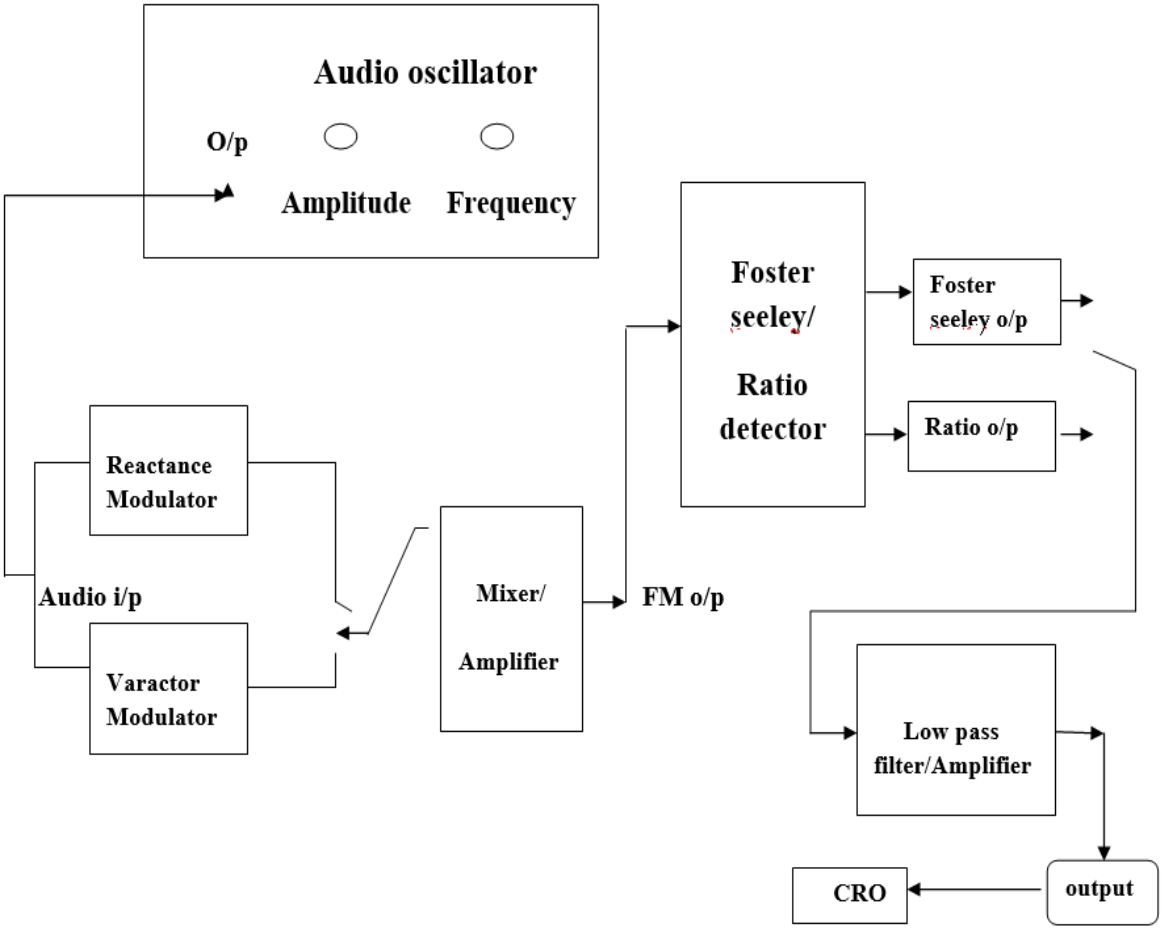


Fig 3.2 Frequency demodulation

3.4 Pre-Lab Questions

1. What are the different methods of generating FM signal?
2. A frequency modulated wave is given as $s(t)=10\sin(2\pi x10^{8}t+5\sin400\pi t)$, identify carrier frequency and maximum frequency deviation of the signal.
3. Give the Carson's rule to calculate bandwidth of the system
4. What is the commercial FM frequency range?
5. When a single tone modulating signal $\cos(15\pi10^{3}t)$ frequency modulates a carrier of 10MHz and produces a frequency deviation of 75KHz calculate the modulation index of FM?

<u>PRE LAB:-</u>

Experiment-03

RA1911004010565
Pushpal Das

Prelab

1) What are the different modes of generating FM signal?

⇒ FM signals can be generated by using direct frequency modulation which is achieved by inputting modulation which is achieved by inputting a message directly into a voltage controlled oscillator or by using indirect frequency modulation, which is achieved by integrating a message signal oscillator, the result of which is transmitted through a frequency multiplier by to produce FM signal.

2) A frequency modulated wave is given as $s(t) = 10 \sin(2\pi \times 10^8 t + 5 \sin 400\pi t)$, identify carrier frequency and maximum frequency deviation of the signal.

Soln

$$\phi_m(t) = 10 \sin\left[(2\pi \times 10^8 t) + (5 \sin 400\pi t)\right]$$

$$= A_c \sin\left[2\pi f_c t + \beta \sin 2\pi f_m t\right]$$

$A_c = 10$

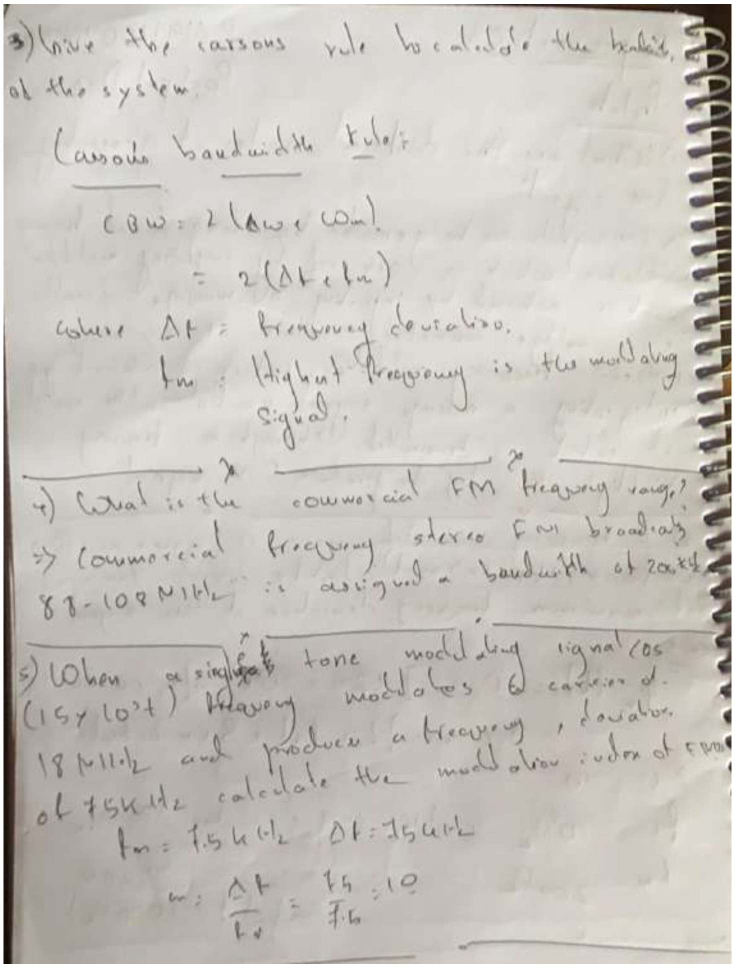
$f_c = 10^8 Hz$

$f_m = 200 Hz$

Frequency deviation $\Delta f = \beta f_m$

$= 5 \times 200$

$\boxed{\Delta f = 1000 Hz}$



3) Give the carsons rule to calculate the bandwidth of the system.

Carson's bandwidth rule:

$CBW = 2(\Delta\omega + \omega_m)$

$= 2(\Delta f + f_m)$

where Δf = frequency deviation.

f_m = Highest frequency in the modulating signal.

4) What is the commercial FM frequency range?

=> Commercial frequency stereo FM broadcasting 88-108 MHz is assigned a bandwidth of 200kHz

5) When a single tone modulating signal $\cos(15\times10^3 t)$ frequency modulates a carrier of 18 MHz and produces a frequency deviation of 75kHz calculate the modulation index of FM

$f_m = 7.5\text{ kHz}$ $\Delta f = 75\text{kHz}$

$m = \frac{\Delta f}{f_m} = \frac{75}{7.5} = 10$

3.5 Lab Procedure

3.5.1 Frequency Modulator

1. Turn on power to the trainer kit and set the 5 volts amplitude and frequency of 1KHz in the audio oscillator (i.e modulating signal) to the trainer kit-ST8203 module.
2. Connect the output socket of the audio oscillator block to the audio input socket of the modulator circuit's block.
3. Select reactance/varactor modulator switch in position. These switches the output of the modulator through to the input of the mixer/amplifier block and the FM output appears once it is connect to the display (Digital/Analog oscilloscope).
4. Adjust the audio oscillators amplitude pot throughout its range note time period of modulated signal for one cycle as T1 and T2 using oscilloscope.
5. Measure the frequency of the FM output signal with F1 and F2 (using F=1/T).This is because the audio information is contained entirely in the signal's frequency and not in its amplitude.

3.5.2 Frequency Demodulator

1. Make the connection as per the block diagram.
2. Turn on power to the ST8203 module.
3. FM output from mixer/amplifier block is connected to the input of Foster Seeley detector or ratio detector block.
4. Select the Foster Seeley or ratio detector switch in position.
5. Connect the input of Low pass filter /amplifier block with the output from the Foster Seeley/ratio detector block.
6. Measure the amplitude and frequency of audio signal (demodulated signal).
7. Adjust the audio oscillator block’s amplitude and frequency pots and compare the original audio signal with the final demodulated signal.

3.6.1 Model Graph

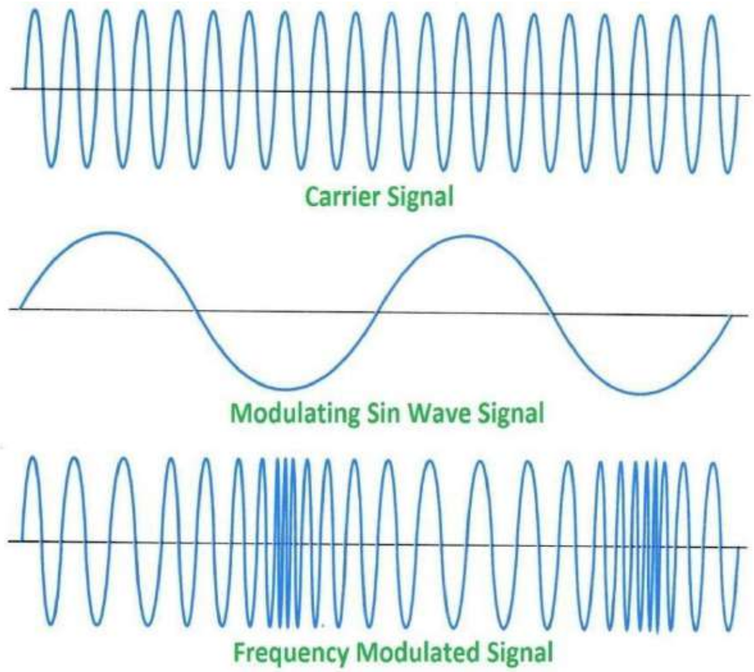


Fig 3.3 Frequency modulated waveforms

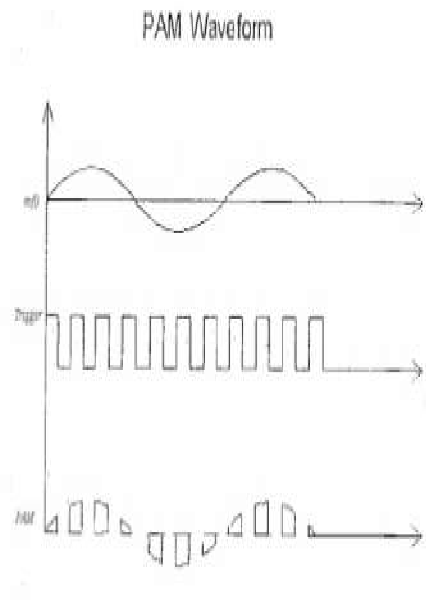


Fig. PAM Model Graph

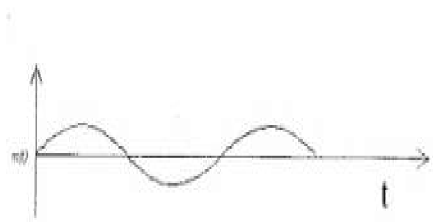


Fig. PAM Demodulation Model Graph

Fig 3.4 Frequency demodulated waveforms

3.7. VIRTUAL LAB

3.7.1 Equipment / Apparatus

SCILAB Software 6.0.2

3.7.2 Exercise

With the following data, use SCILAB to generate and plot time domain representation of Frequency Modulation signal

Carrier frequency fc = 1 kHz

Amplitude Carrier frequency = Ac = 9 V
Modulating frequency =300 Hz
Amplitude Modulating Signal Vm = 4.5 V

Solution

```
t= 0:1/10000:0.02; // declare time interval
Ec = 9; // amplitude of carrier signal
Em = 4.5; // amplitude of modulating signal
fc = 1000; // carrier frequency
fm = 300; // modulating frequency

//Carrier signal
Vc = Ec *cos (((2*%pi)*fc)*t);

//Modulating signal
Vm = Em * cos (((2*%pi)*fm)*t);

m1 = 1; // modulation index

Vfm = Ec*cos(((( 2*%pi)*fc)*t)+m1*sin(((2*%pi)*fm)*t));  //Frequency modulation signal

// plot signal
subplot (311);
plot (t, Vm);
title("Modulating signal");
xlabel('Time - s');
ylabel('Amplitude');
subplot (312);
plot (t,Vc);
title("Carrier signal");
xlabel('Time - s');
ylabel('Amplitude');
subplot (313);
plot (t,Vfm);
title("Modulated-wave");
xlabel('Time - s');
ylabel('Amplitude');
```

CODE:-

ADC EXP03.sce (C:\Users\ELCOT\3D Objects\DSP LAB\ADC lab\ADC EXP03.sce) - SciNotes

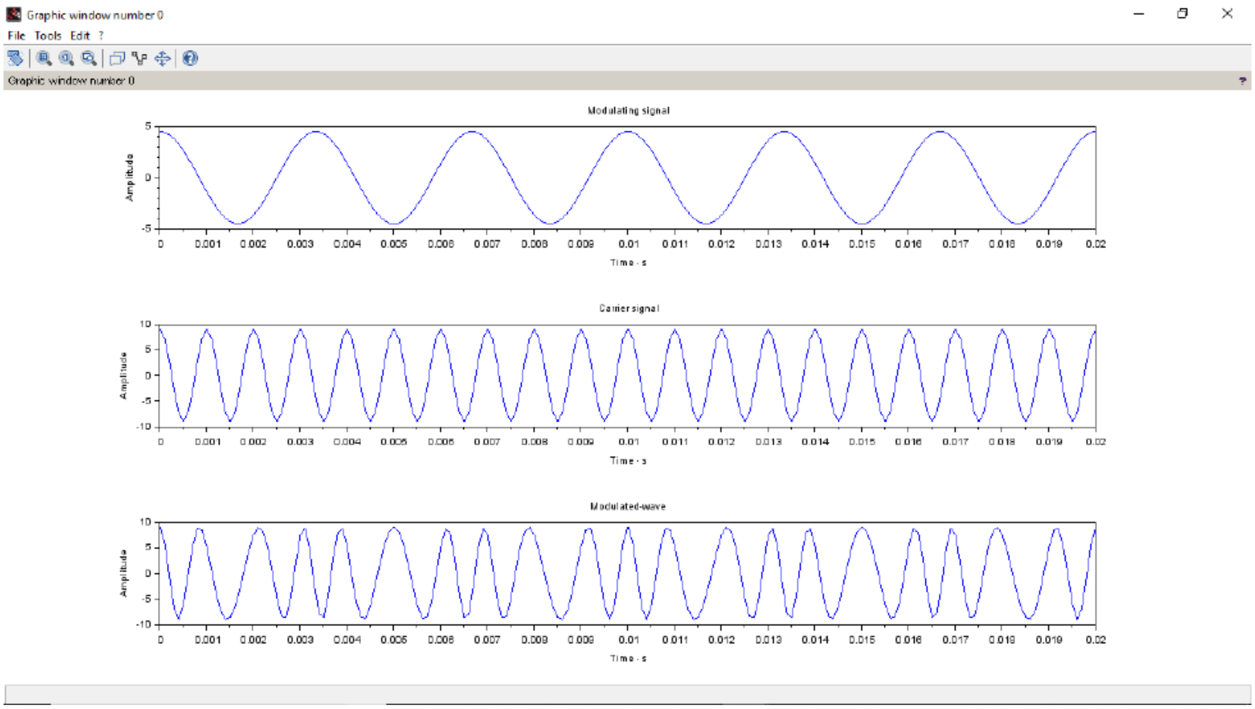
File Edit Format Options Window Execute ?

ADC EXP03.sce (C:\Users\ELCOT\3D Objects\DSP LAB\ADC lab\ADC EXP03.sce) - SciNotes

ADC EXP03.sce

```
t= 0:1/10000:0.02; // declare time interval
Ec = 9; // amplitude of carrier signal
Em = 4.5; // amplitude of modulating signal
fc = 1000; // carrier frequency
fm = 300; // modulating frequency

//Carrier signal
Vc = Ec *cos (((2*%pi)*fc)*t);

//Modulating signal
Vm = Em * cos (((2*%pi)*fm)*t);

ml = 1; // modulation index

Vfm = Ec*cos(((( 2*%pi)*fc)*t)+ml*sin(((2*%pi)*fm)*t)); //Frequency modulation signal

// plot signal
subplot (311);
plot (t, Vm);
title("Modulating signal");
xlabel('Time - s');
ylabel('Amplitude');
subplot (312);
plot (t,Vc);
title("Carrier signal");
xlabel('Time - s');
ylabel('Amplitude');
subplot (313);
plot (t,Vfm);
title("Modulated-wave");
xlabel('Time - s');
ylabel('Amplitude');
```

CODE SIMULATION OUTPUT:-



3.7.3 Observation – Software:-

<table>
<tr><th>Signal Name</th><th>Amplitude</th><th colspan="2">Frequency</th><th colspan="2">Time period</th></tr>
<tr><td>Modulating signal (input signal)</td><td>4.5V</td><td colspan="2">300Hz</td><td colspan="2">3.33ms</td></tr>
<tr><td>Carrier signal</td><td>9.16V</td><td colspan="2">1KHz</td><td colspan="2">1ms</td></tr>
<tr><td rowspan="2">Modulated signal (output signal)</td><td rowspan="2">9.0V</td><td>F1</td><td>F2</td><td>T1</td><td>T2</td></tr>
<tr><td>300Hz</td><td>1KHz</td><td>3.3ms</td><td>1ms</td></tr>
</table>

3.7.4 Model Graph

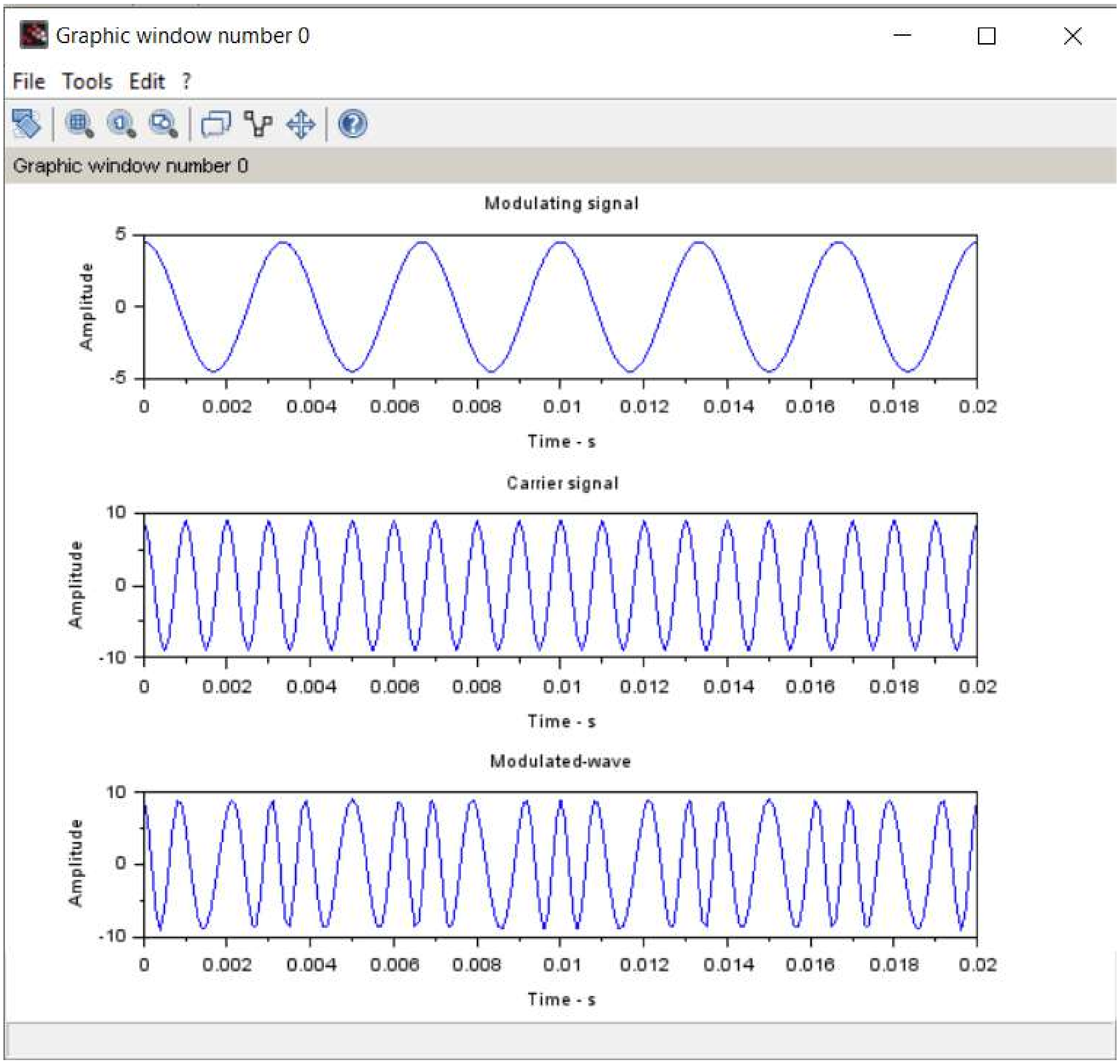


Fig. 3.5 Simulated Amplitude Modulation waveforms

3.8 Post Lab Questions

1. Consider when two modulating signal produces a modulation index of 100 with same frequency deviation of 75KHz. Find its frequency and amplitude of modulating signal assume K_f=15KHz per volt.

2. List the applications of FM technique.

3. A carrier signal, vct=10 cos[8π* 103]t is frequency modulated by a modulating signal, vmt=5 cos30π*101t. Given deviation sensitivity is 15 Hz /volt. Plot FM signal in time domain using SCILAB.

Post-lab.

1) Consider when two modulating ~~frequency~~ signals produce a modulation index of 100 with same frequency deviation of 75 kHz. Find its frequency and amplitude of modulating signal assum $k_f = 15\text{kHz}$.

Soln

$k_f = 15 \text{ kHz/volt}$

$\Delta f = k_f A_m$

$A_m = \frac{\Delta f}{k_f} = \frac{75}{15} = 5V$

$m = \frac{\Delta f}{f_m} \Rightarrow f_m \Rightarrow \frac{\Delta f}{m}$

$f_m = \frac{75}{100} = 750 \text{ Hz}$

2) List of applications of FM technique.

=> FM can be used for the broadcasting of F.M radio

=> This helps in larger signal to noise ratio.

=> Used in telemetry, radar etc

3) A carrier signal, is frequency modulated by a modulating signal. Give deviation sensitivity is 15 kHz/volt. Plot F.M signal in time domain using suitabl

POST LAB 03:-

CODE:-

EXP 03 POST LAB.sce (C:\Users\ELCOT\3D Objects\DSP LAB\ADC lab\EXP 03 POST LAB.sce) - SciNotes

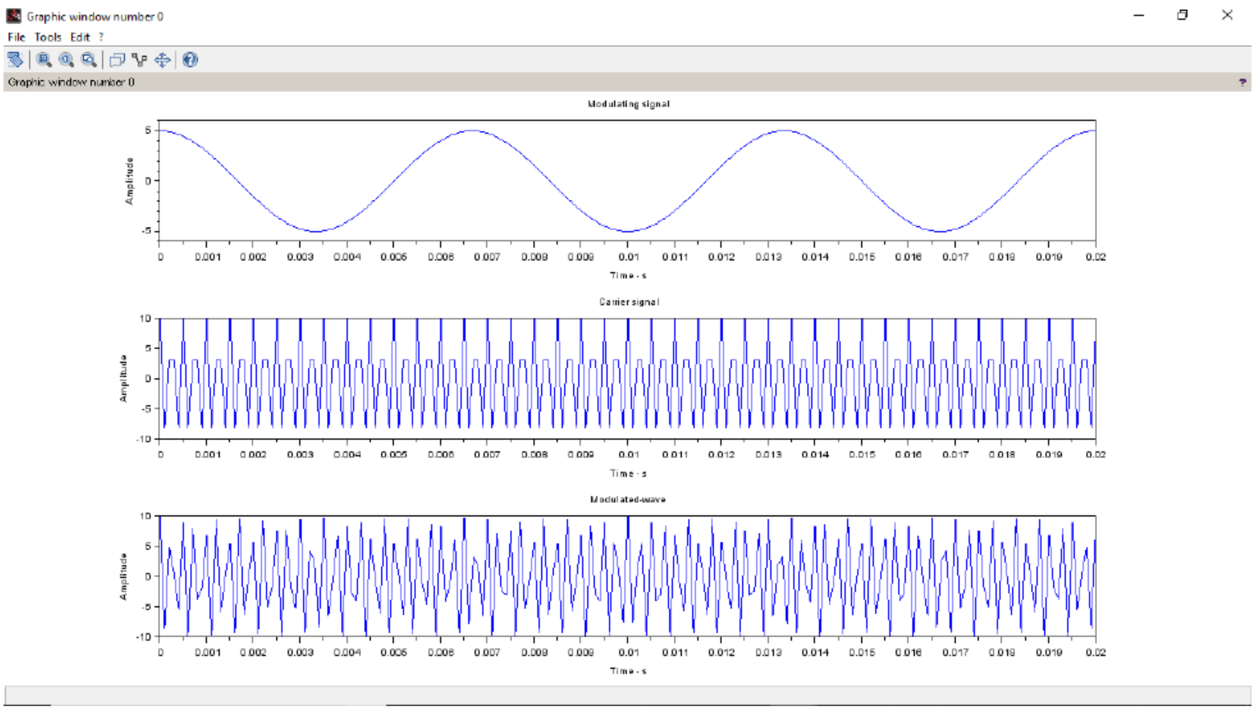
File Edit Format Options Window Execute ?

EXP 03 POST LAB.sce (C:\Users\ELCOT\3D Objects\DSP LAB\ADC lab\EXP 03 POST LAB.sce) - SciNotes

ADC EXP03.sce | EXP 03 POST LAB.sce

```
t= 0:1/10000:0.02; // declare time interval
Ec = 10; // amplitude of carrier signal
Em = 5; // amplitude of modulating signal
fc = 4000; // carrier frequency
fm = 150; // modulating frequency

//Carrier signal
Vc = Ec *cos (((2*%pi)*fc)*t);

//Modulating signal
Vm = Em * cos (((2*%pi)*fm)*t);

ml = 0.2; // modulation index

Vfm = Ec*cos(((( 2*%pi)*fc)*t)+ml*sin(((2*%pi)*fm)*t)); //Frequency modulation signal

// plot signal
subplot (311);
plot (t, Vm);
title("Modulating signal");
xlabel('Time --s');
ylabel('Amplitude');
subplot (312);
plot (t,Vc);
title("Carrier signal");
xlabel('Time --s');
ylabel('Amplitude');
subplot (313);
plot (t,Vfm);
title("Modulated-wave");
xlabel('Time --s');
ylabel('Amplitude');
```

SIMULATION OUTPUT:-



3.9 LAB RESULT

Thus the Frequency modulation and demodulation process in a trainer kit was performed and the modulation of FM signal was simulated using Scilab for the given carrier and modulating signal.

SRM Institute of Science and Technology Faculty of Engineering and Technology Department of Electronics and Communication Engineering
18ECC205J ANALOG AND DIGITAL COMMUNICATION LAB **Fifth Semester, 2021-22 (odd semester)**

Name : Pushpal Das

Register No. : RA1911004010565

Day / Session : 2/3&4

Venue : Online(G-Meet)

Title of Experiment : 4. PRE –EMPHASIS AND DE-EMPHASIS CIRCUITS USING FM Date of

Conduction : 08 AUG 2021

Date of Submission : 23 AUG 2021

Particulars	Max. Marks	Marks Obtained
Pre-lab questions	05	
In-lab experiment	10	
Post-lab questions	15	
Total	**30**	

REPORT VERIFICATION

Date : 23/08/2021

Staff Name : Dr. P. Malarvizhi / Mrs. Harisudha

Signature :

4. PRE –EMPHASIS AND DE-EMPHASIS CIRCUITS USING

FM 4.1 OBJECTIVE

To design a pre emphasis circuit to boost the input signal level for a FM transmitter for a cut off frequency of 1KHz. Attenuate the boosted high frequency signals at the receiver side using a deemphasis circuit with a cutoff frequency of 1.6KHz. Analyze the frequency response characteristics of pre emphasis and de emphasis circuits.

4.2 HARDWARE REQUIRED

S.No	Equipment/Component name	Specifications/Value	Quantity
1	Cathode Ray Oscilloscope	(0 – 20MHz)	1
2	Audio Frequency Oscillator	(0-2) MHz	1
3	Regulated power supply	(0 -30V), 1A	1
4	Resistors	1K Ω 2K Ω 10K Ω 68K Ω 100K Ω	1 1 2 1 1
5	capacitors	0.1 µf 0.001 µf	2 3
6	Semiconductor Device(Transistor)	Q2N2222	1
7	Decade Inductance Box	0.3H	1

PRE – EMPHASIS CIRCUIT DIAGRAM

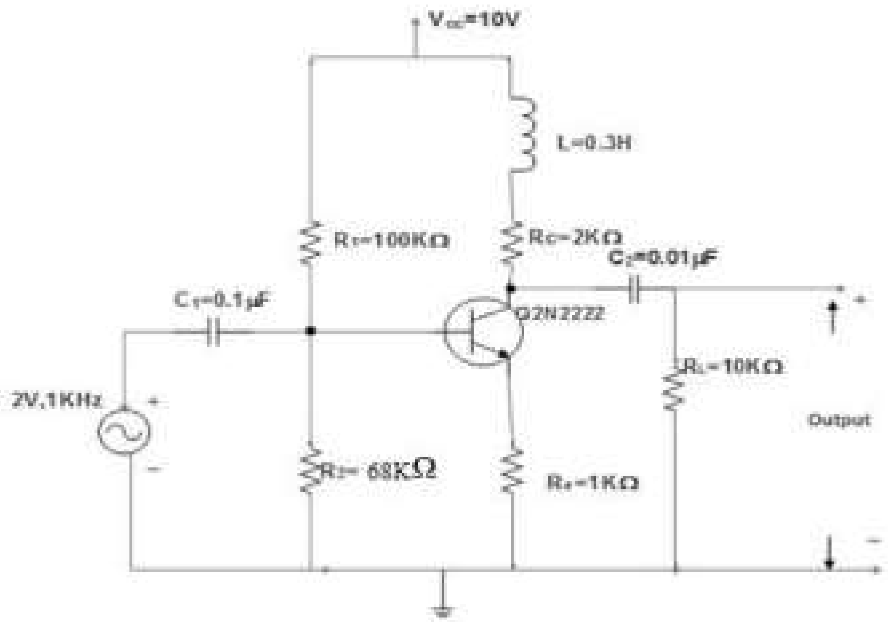


Fig. 4.1 Pre Emphasis Circuit

DE – EMPHASIS CIRCUIT DIAGRAM

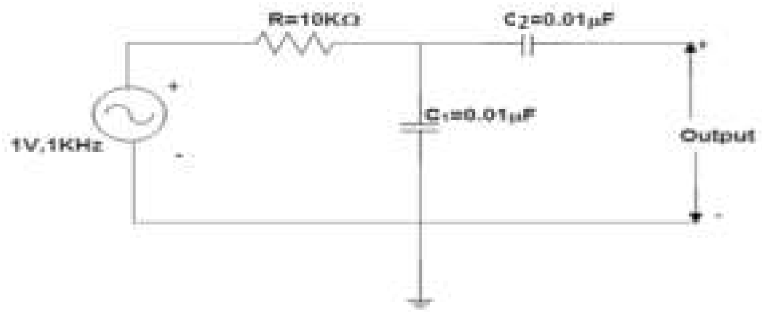


Fig. 4.2 De Emphasis Circuit

4.3 THEORY

During the transmission over a channel, the received signal contains interference (high frequency noise). For demodulated FM signals, the interference power increases as the frequency goes up. Thus,

de-emphasis is applied to the demodulated signal to decrease the power of the interference in high frequency. However, in order to keep the high frequency component of the demodulated message, pre-emphasis must be applied to the message before going through the FM modulator. **4.3.1 PRE-EMPHASIS**

Pre-emphasis refers to boosting the relative amplitudes of the modulating voltage for higher audio frequencies. Pre-emphasis is done at the transmitting side of the frequency modulator. Signals with higher modulation frequencies have lower SNR. In order to compensate this, the high frequency signals are emphasised or boosted in amplitude at the transmitter section of a communication system prior to the modulation process. That is, the pre- emphasis network allows the high frequency modulating signal to modulate the carrier at higher level, this causes more frequency deviation.

The circuit consist of a transistor, resistor and an inductor. It is basically a high pass filter or Differentiator. A pre-emphasis circuit produces a constant increase in the amplitude of the modulating signal with an increase in frequency.

The cut off frequency is determined by the RC or L/R time constant of the network. Normally, the cut off frequency occurs at the frequency where capacitive reactance or inductive reactance equals R.

The cut off frequency is given by the formula

$$f_c = \frac{R}{2\pi L}$$

By the use of an active pre-emphasis network we can reduce the signal loss and distortion with the increase of SNR. Also the output amplitude of the network increases with frequencies above cut off frequency.

4.3.2 DE-EMPHASIS

De-emphasis is the complement of pre-emphasis, in the anti-noise system called emphasis. This circuit is used to attenuate the high frequency signal that is boosted at the transmitter section. The circuit is placed at the receiving side. It acts as a low pass filter. The cut off frequency is given by the formula

$$f_c = \frac{1}{2\pi RC}$$

The circuit consists of a passive network consisting of a resistor and a capacitor. It is basically a low pass filter or integrator. The pre- emphasis network in front of the FM modulator and a de-emphasis network at the output of the FM demodulator improves the Signal to Noise Ratio for higher modulating signal frequencies, thus producing a more uniform SNR at the output of demodulator.

4.4 DESIGN

4.4.1 PRE- EMPHASIS

The cut off frequency is given by the formula

$f_c = R/(2\pi L)$

Let f_c =1KHz

Assume R=2KΩ

Therefore $L = R/(2\pi f_c)$

$L = 2000/(2*3.14*1000)$

$L = 0.3H$

4.4.2 DE- EMPHASIS

The cut off frequency is given by the formula

$f_c = 1/(2\pi RC)$

Let f_c =1.6KHz

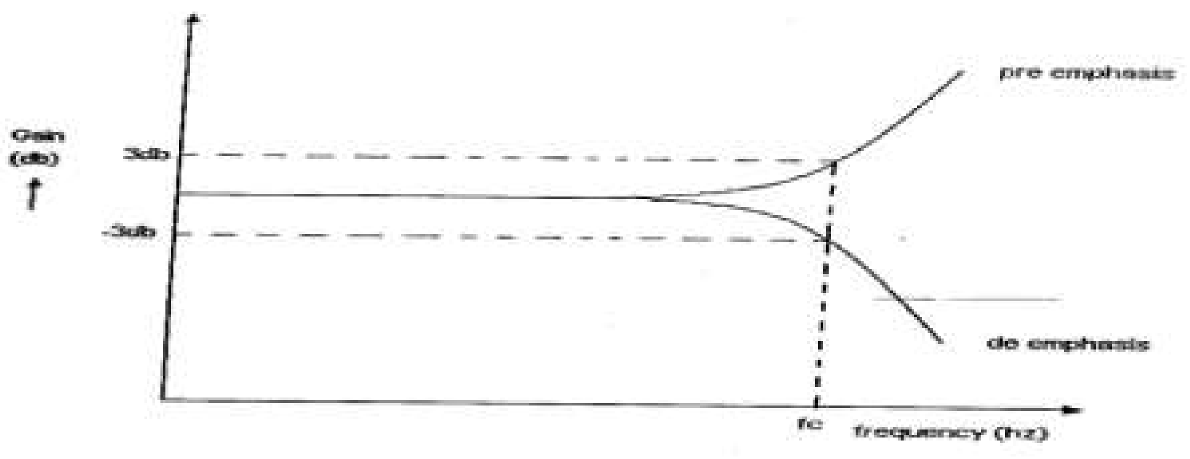
Assume R=10KΩ

Therefore $C_1 = 1/(2\pi f_c R)$

$C_1 =1/(2*3.14*1600*10000)$

$C_1 =0.01\mu F$

MODEL GRAPH



OBSERVATION FOR PRE-EMPHASIS:

Vi=

Frequency(Hz)	Vo	Gain= Vo/Vi	Gain in dB=20log(Vo/Vi)

OBSERVATION FOR DE-EMPHASIS

Vi=

Frequency(Hz)	Vo	Gain= Vo/ Vi	Gain in dB =20log(Vo/Vi)

4.5 PRE LAB QUESTIONS

▪ What is meant by threshold effect?

1. What is pre-emphasis?

2. How the threshold effect can be avoided?

3. What is fidelity?

4. What is sensitivity and selectivity?

Prelab Exp-04

RA1911004010665

Pushpa-[illegible]

1) Threshold effect is an effect in a dependent variable that doesn't occur until a certain level or threshold is reached in an independent variable.

2) A process of increasing the amplitude at certain frequencies relative to others in a signal in order to help them overcome noise.

3) Threshold reduction can be realized by reduction in FM receivers may be achieved by using an FM demodulator with negative feedback (commonly referred to as an FMFB demodulator) or by using a phase-locked loop demodulator.

4) Fidelity of a receiver is its ability to reproduce the exact replica of the transmitted signals at that receiver output.

5) Sensitivity

The ability of a receiver to detect the weakest possible signal is known as sensitivity.

Selectivity:

=> Selectivity refers to the detector's ability to filter out signals that are not the signals.

4.6 LAB PROCEDURE

1. The circuit connections are made as shown in the circuit diagram for the pre-emphasis and de-emphasis circuits.

1. A power supply of 10V is given to the pre-emphasis circuit.

2. Set the input voltage at 2V, 1 KHz for pre-emphasis and 1V, 1 KHz for de-emphasis using AFO.

3. For this constant value of input voltage the values of the frequency is varied and the output voltage is noted on the CRO.

4. A graph is plotted between gain and frequency in a semilog graph sheet for both pre emphasis and de-emphasis outputs.

4.7 VIRTUAL LAB

4.7.1 Equipment / Apparatus

SCILAB Software 6.0.2

4.7.2 Exercise

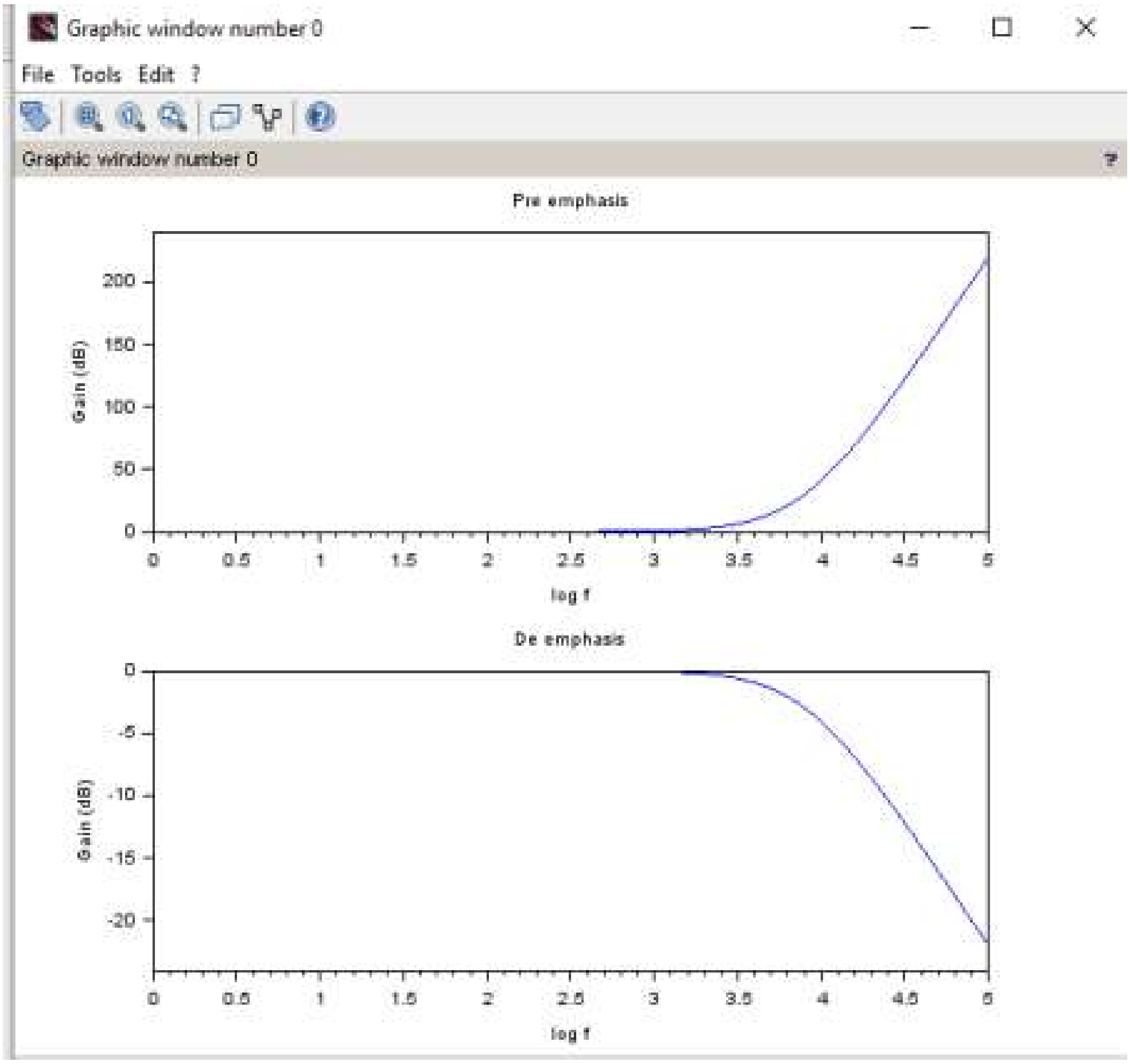
Use Scilab software to plot the Pre emphasis and Deemphasis curve with frequency Versus Gain .Cut off frequency =1 KHz.

4.7.3 Solution:

```
clc ;
clear ;
xdel ( winsid () ) ;//xdel(0)//xdel()
r=200;
R=2000;
j=sqrt(-1);
C=0.1*10^-6;

for i=1:100000
 k(i)=log10(i)
 f=i;
 w=2*3.1414*f;
 w1=1/(r*C);
 pre_emp(i)=(R/r)*20*log10(sqrt(1+(w/w1)^2))
 de_emp(i)=-20*log10(sqrt(1+(w/w1)^2))
end
```

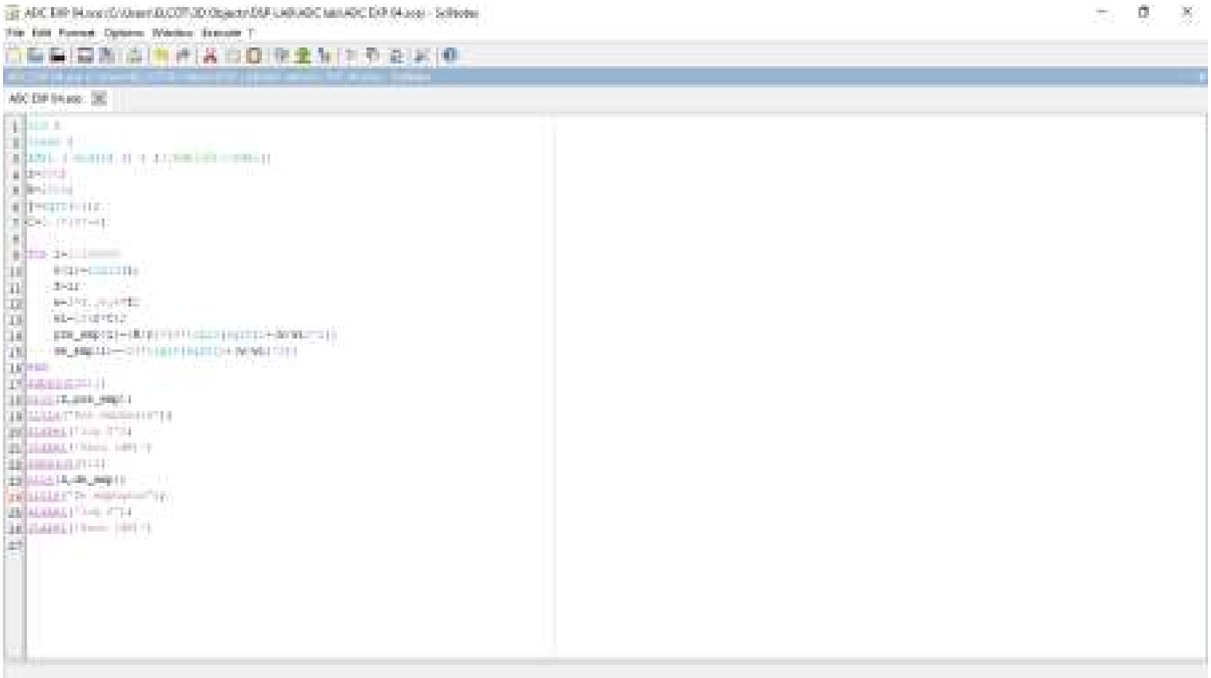
```
subplot(211)
plot(k,pre_emp);
title("Pre emphasis");
xlabel('log f');
ylabel('Gain (dB)')
subplot(212)
plot(k,de_emp);
title("De emphasis");
xlabel('log f');
ylabel('Gain (dB)')
```

Model Graph :

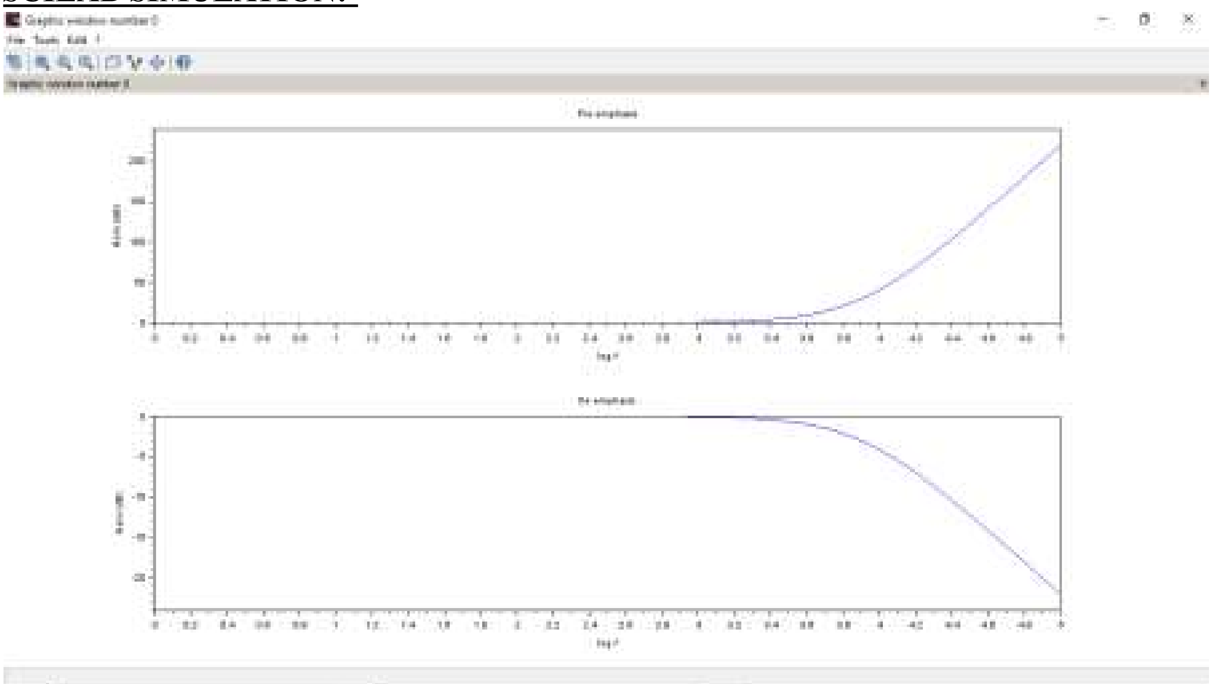


Simulated Pre emphasis and De emphasis waveforms

SCILAB CODE :-



SCILAB SIMULATION:-



4.8 POST LAB QUESTIONS

1. What is de-emphasis?
2. How to reduce the noise during transmission in FM ?

3. What should be the time constant for de-emphasis circuit?

4. Why pre-emphasis is done after modulation?

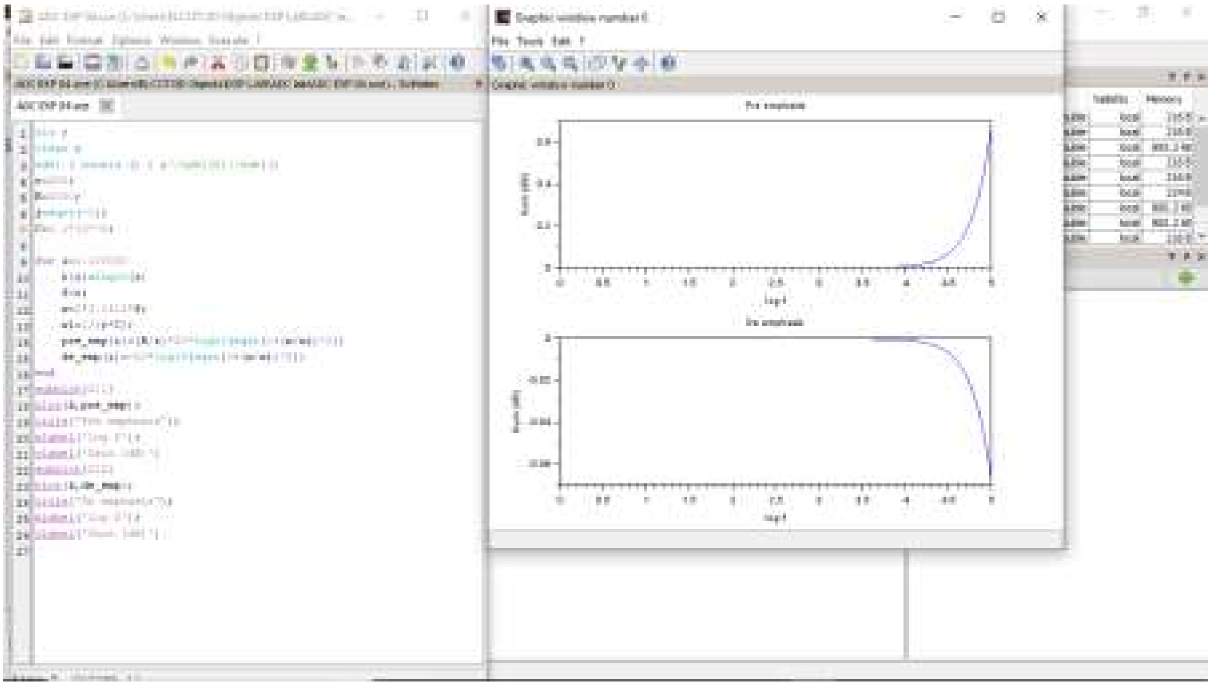
5. List some applications of pre-emphasis circuit.

6. Write a program in scilab to design a circuit to boost the baseband signal amplitude in the FM transmitter for the cut off frequency f_c =10 KHz

4) Pre emphasis in FM is done before modulation. The high frequency energy in the audio is boosted to compensate for the high frequency noise in FM and then after modulation in the receiver, the high frequency energy is reduced back to normal. This process improves the signal to noise ratio in FM signal.

5) Pre emphasis is commonly used in telecommunication, digital audio recording, record cutting, in FM broadcasting transmission and in displaying the spectrograms of speech signals.

6.SCI LAB O/P:-



4.9 LAB RESULT

The Pre-emphasis and De-emphasis graphs were plotted using Scilab.

Laboratory Report Cover Sheet

SRM Institute of Science and Technology Faculty of Engineering and Technology Department of Electronics and Communication Engineering
18ECC205J ANALOG AND DIGITAL COMMUNICATION **Fifth Semester, 2021-22 (odd semester)**

Name : Pushpal Das

Register No. : RA1911004010565

Day / Session : 2 nd/ FN

Venue : ONLINE(G meet)

Title of Experiment : (a) Pulse Amplitude Modulation (PAM) & Demodulation (B) PULSE POSITION MODULATION AND DEMODULATION

(C) PULSE WIDTH MODULATIONS AND DEMODULATION

Date of Conduction : 20/08/2021

Date of Submission : 20/09/2021

Particulars	**Max. Marks**	**Marks Obtained**
Pre-lab questions	05	
In-lab experiment	10	
Post-lab questions	10	
Record Submission	05	
Total	**30**	

REPORT VERIFICATION

Date :26 AUG 2021

Staff Name : **Dr. P. Malarvizhi / Mrs. Harisudha**

Signature :

5 (a) Pulse Amplitude Modulation (PAM) & Demodulation 5a.1

Objective

Study of Pulse Amplitude Modulation & Demodulation with Sample, Sample & Hold & Flat Top

5a.2 Equipment required

Scientech 2110 with Power Supply cord
Scientech Oscilloscope with connecting probe
Connecting cords

5a.3 Wiring Diagram

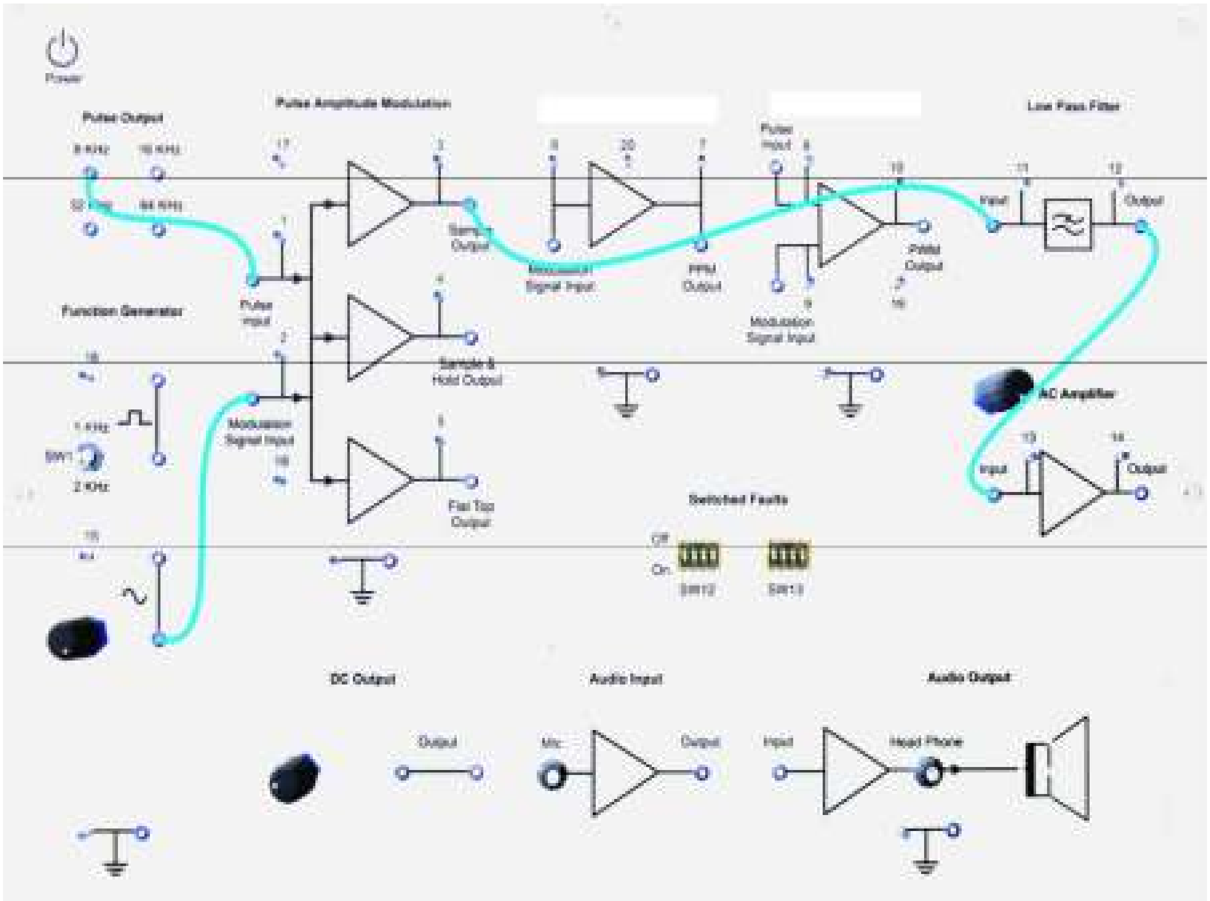


Figure 5a.1 Pulse Amplitude Modulation and Demodulation

5a.4 Procedure

1. Connect the circuit as shown in Figure 1.1
2. Output of sine wave to modulation signal input in PAM block keeping the switch in 1 KHz position.
3. 8 KHz pulse output to pulse input

4. Output of low pass filter to input of AC Amplifier. Keep the gain pot in AC Amplifier block in anti clock wise position.
5. Switch ‘On’ the Power Supply & Oscilloscope.
6. Observe the outputs at TP (3) together with Modulation signal input TP (3) and Pulse input TP (1). This is a Natural sampling output.
7. 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).
8. 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.
9. Observe the Flat Top output at TP (5), together with Modulation signal input TP (2) and Pulse input TP (1). This is Flat Top Sampling output.
10. 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).
11. 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.
12. Observe the output of Sample & Hold circuit at TP4, together with Modulation signal input TP (2) and Pulse input TP (1). This is Sample & Hold output.
13. Connect the Sample & Hold 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). 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.
15. Vary the amplitude potentiometer and frequency change over switch & observe the effect on these three outputs.
16. 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.

5a.5 Model Graph

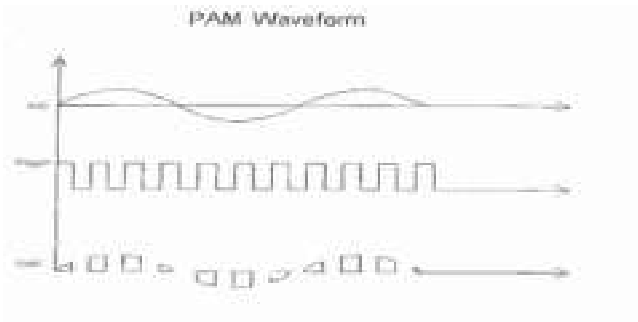
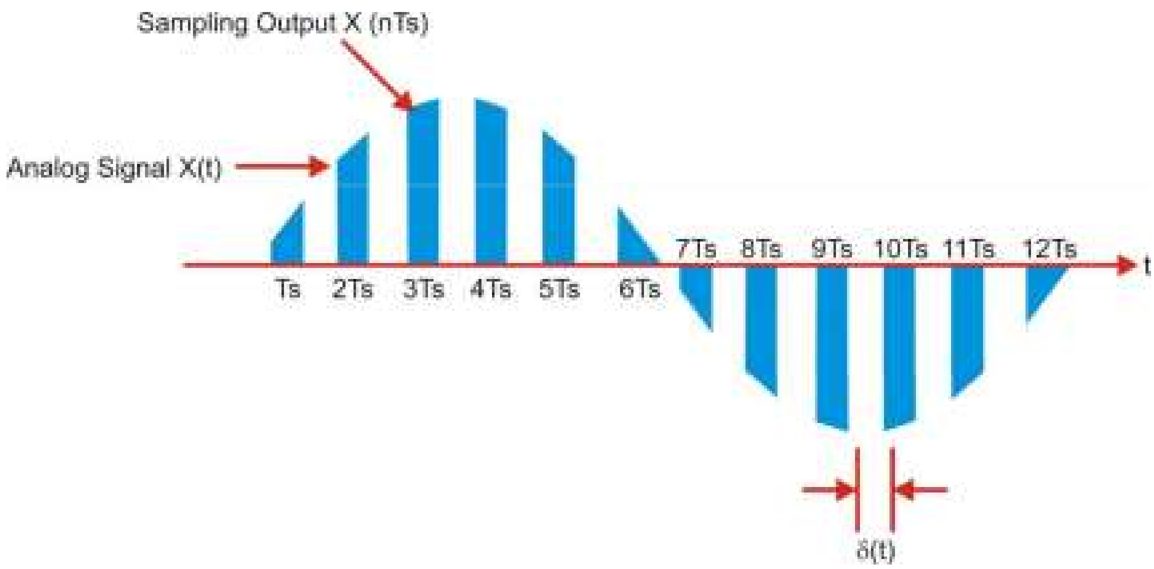
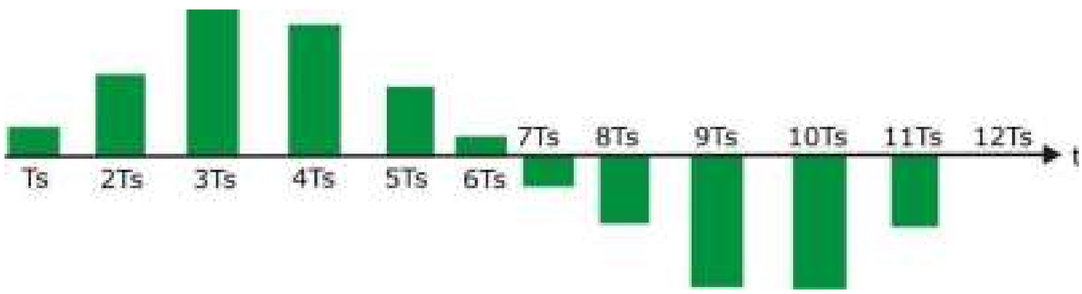


Fig. 5a.2 PAM Modulated Waveform

Natural Sampling Output



<u>Flat top sampling Output</u>



5a.6 Tabulation

Signal	Amplitude (V)	Time Period (ms)
Sinusoidal waveform		
Pulse Carrier waveform		
Sample Output		
Sample and Hold Output		
Flat Top Output		
Demodulated waveform		

5a.7 Lab Result

Thus the PAM and its demodulation were performed and graphs were plotted.

(B) PULSE POSITION MODULATION AND DEMODULATION

5b.1 Objective

To analyze a PPM system and interpret the modulated and demodulated waveforms

5b.2 Hardware Required

PPM Modulator and Demodulator Trainer Kit
Power supply Cord
Digital Oscilloscope

5b.3 Introduction

In Pulse Position Modulation both the pulse amplitude and pulse duration are held constant but the position of the pulse is varied in proportional to the sample values of the message signal .Pulse time modulation is a class of signaling techniques that encodes the sample values of the analog signal on to the time axis of a digital signal and it is analogous to angle modulation technique. The two types of PTM are PWM and PPM. In PPM the analog sample value determines the position of a narrow pulse relative to the clocking time. In PPM rise time of pulse decides the channel bandwidth it has low noise interference.

5b.4 Procedure

1. Connect the circuit as shown in Figure 2.1
2. A modulating signal is given to the PPM modulator
3. The amplitude and the time duration of the modulating signal are observed using CRO. 4. PPM output is observed from the output of PPM modulator stage and the amplitude and time duration of the PPM wave are noted down.
5. For Demodulation process, PPM signal is applied to the filter circuit and then to amplifier as shown in fig 2.3
6. After demodulation the original signal is recovered.

5b.5 Wiring Diagram

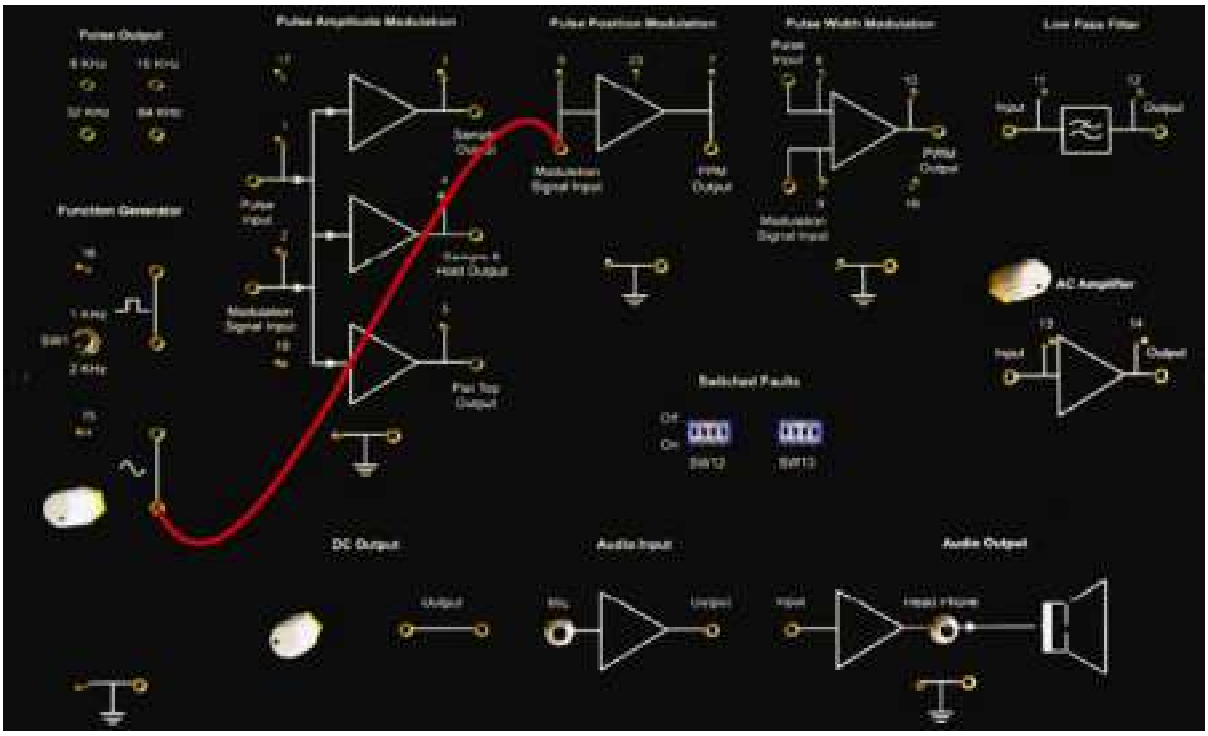


Figure 5b.2.1 Wiring Diagram for Modulation

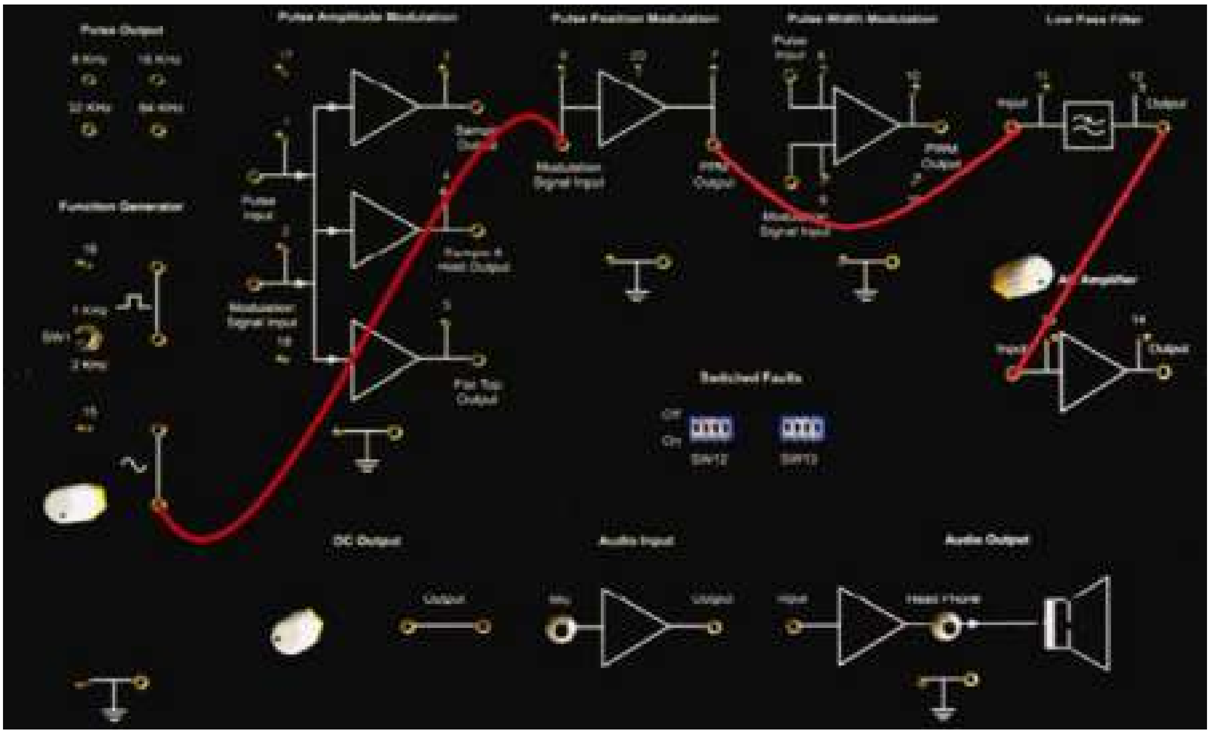
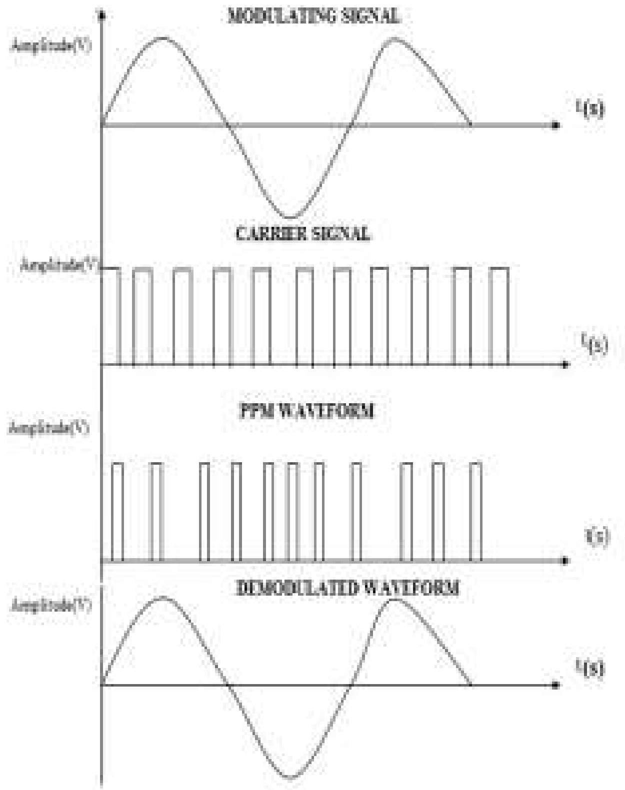


Figure 5b.2.2 Wiring Diagram for Demodulation

5b.6 Model Graph:



5b.7 PPM Modulation

Name of the Signal	Amplitude(V)	Time Period(ms)	Frequency(Hz)
Message signal			
Carrier Signal			
Modulated Signal			

PPM Demodulation

Name of the Signal	Amplitude(V)	Time Period(ms)	Frequency(Hz)
LPF Signal			
Demodulated signal			

5b.8 Lab Result

Thus the Pulse Position modulation and demodulation were performed and graphs were plotted.

5 (C) PULSE WIDTH MODULATIONS AND DEMODULATION

5c.1 Objective

Study of PWM using different Sampling Frequency and its demodulation

5c.2 Equipment Required

Scientech 2110 with Power Supply cord
Scientech Oscilloscope with connecting probe
Connecting cords

5c.3 Wiring Diagram

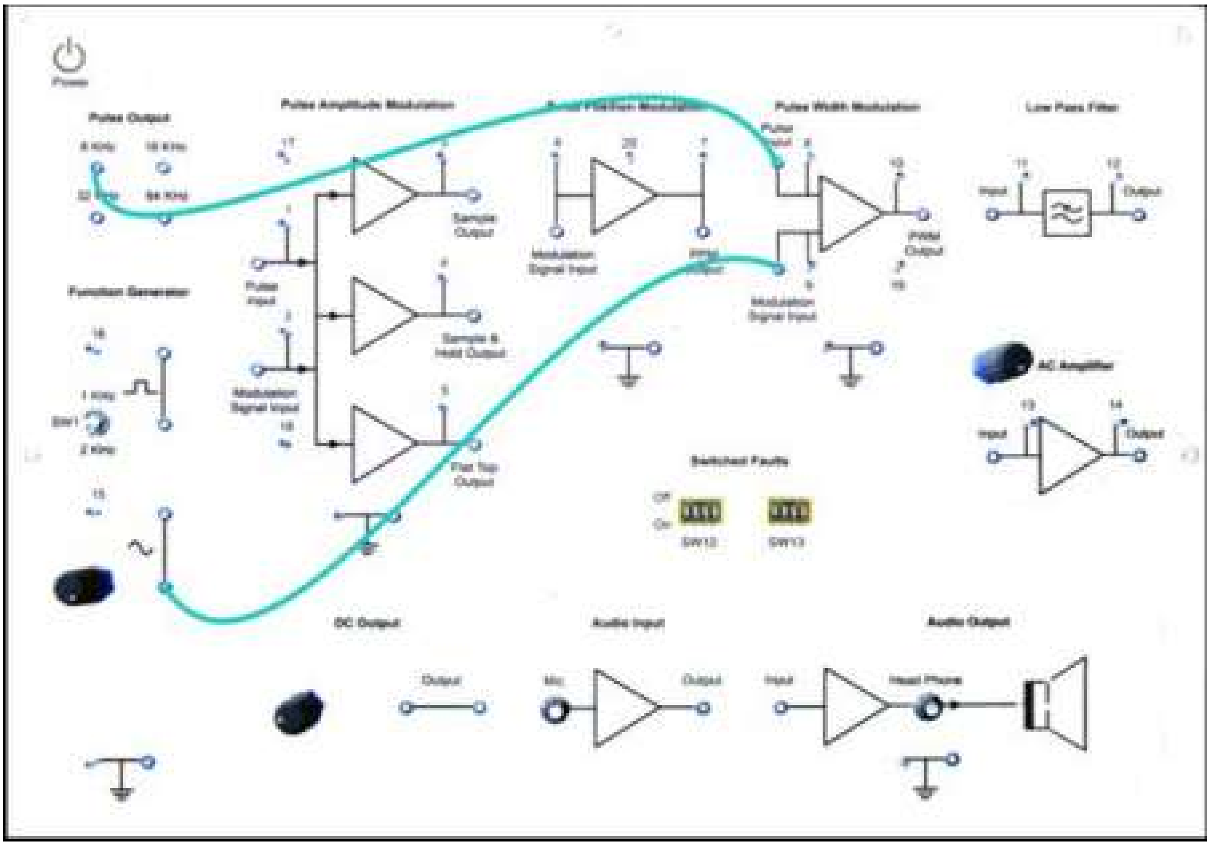


Figure 5c.3.1 Wiring Diagram for PWM Modulation

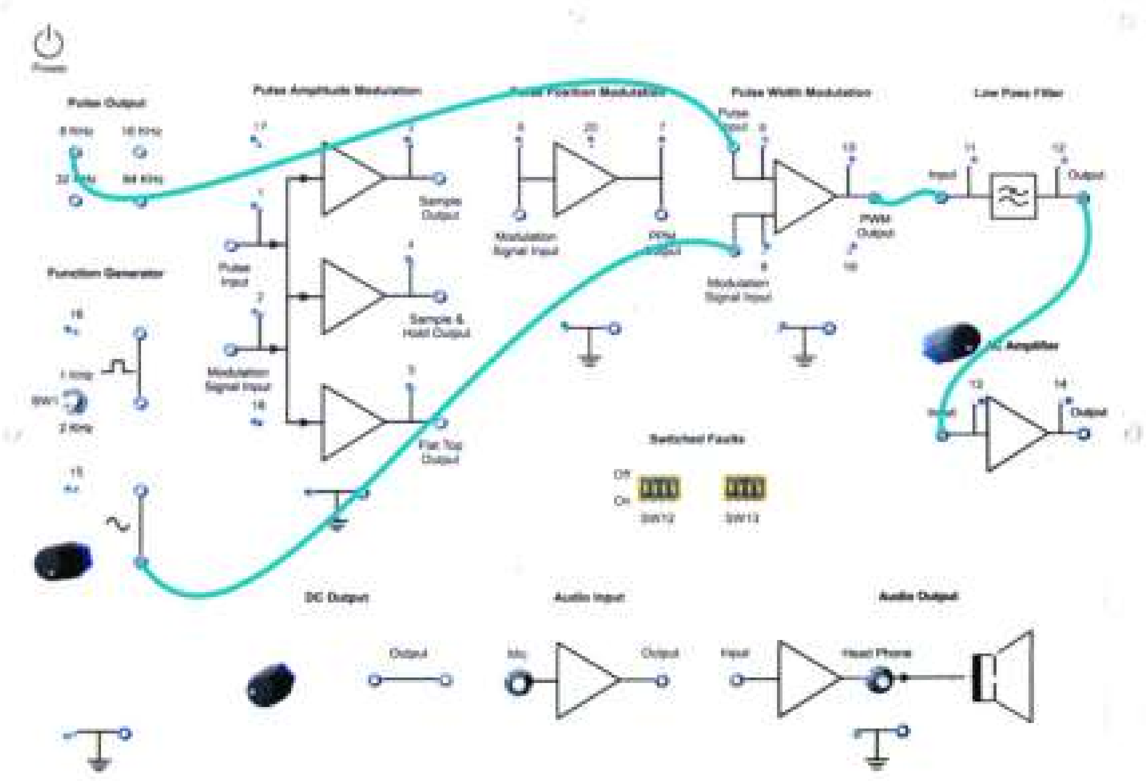


Figure 5c.3.2 Wiring Diagram for PWM Demodulation

5c.4 Procedure

1. Connect the circuit as shown in the Figure 1.3.1
2. 1 KHz sine wave output of Function Generator block to modulation input of PWM block.
3. 8 KHz square wave output to pulse input of PWM block.
4. Switch ‘On’ the Power Supply & Oscilloscope.
5. Observe the PWM outputs at TP (10) together with Modulation signal input TP (9). 6. Vary the amplitude of sine wave and see its effect on width of pulse output. 7. Also, change the frequency of the pulse by connecting the pulse input to different pulse frequencies viz. 8 KHz, 16 KHz, 32 KHz and see the variations in the PWM output. 8. Switch ‘On’ fault No. 1, 2, & 5 one by one & observes their effect on PWM Output and tries to locate them.
9. Output of low pass filter to input of AC Amplifier.
10. Switch ‘On’ the Power Supply & Oscilloscope.
11. Observe the waveform at the output of Low pass filter TP12 together with modulation signal input (9).
12. Then observe the demodulated output at the output of AC Amplifier TP14 together with modulation signal input TP (9). Vary the Gain of AC Amplifier to get the unclipped output. Vary the amplitude of input; the amplitude of output will vary.
13. Try varying the amplitude of sine wave signal; you will observe that the output signal varies similarly.
14. 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.
15. Switch ‘Off’ the Power Supply

5c.5 Model Graph

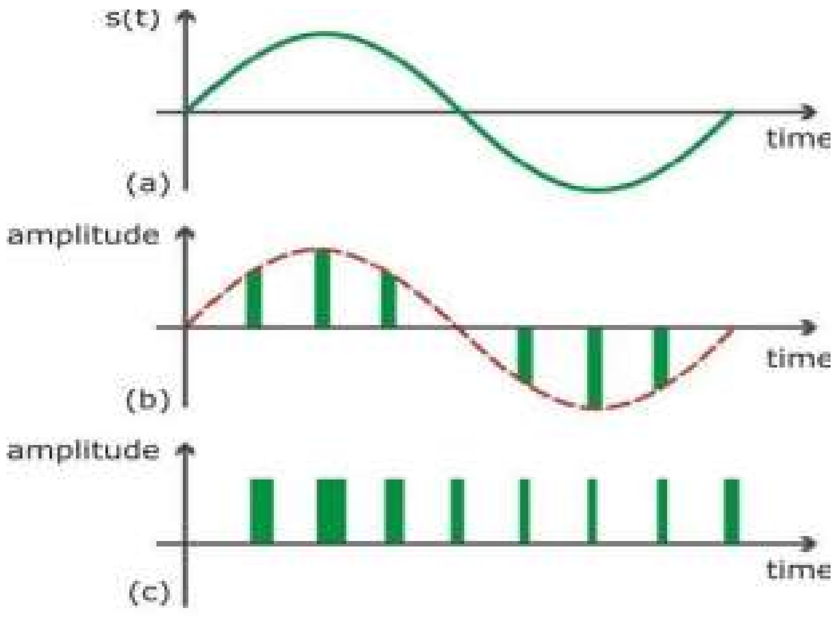


Figure 5c.3 (a) Input Waveform (b) PAM Waveform (c) PWM Waveform

5c.6 Tabulation

Signal	Frequency	Amplitude
Sinusoidal waveform		
Pulse waveform		
Demodulated waveform		

5c.7 Pre Lab Questions

1. What is sampling?
2. What is sample and hold circuits?
3. Why Is Flat Top sampling more preferred than natural sampling?
4. What is Pulse Amplitude Modulation (PAM)?
5. What do you mean by Pulse-Time Modulation?

PRE LAB :-

4) What is pulse modulation (PAM)?

=> Pulse amplitude modulation is an analog modulation scheme in which the amplitude of the carrier pulse varies proportional to the instantaneous amplitude of the message signal.

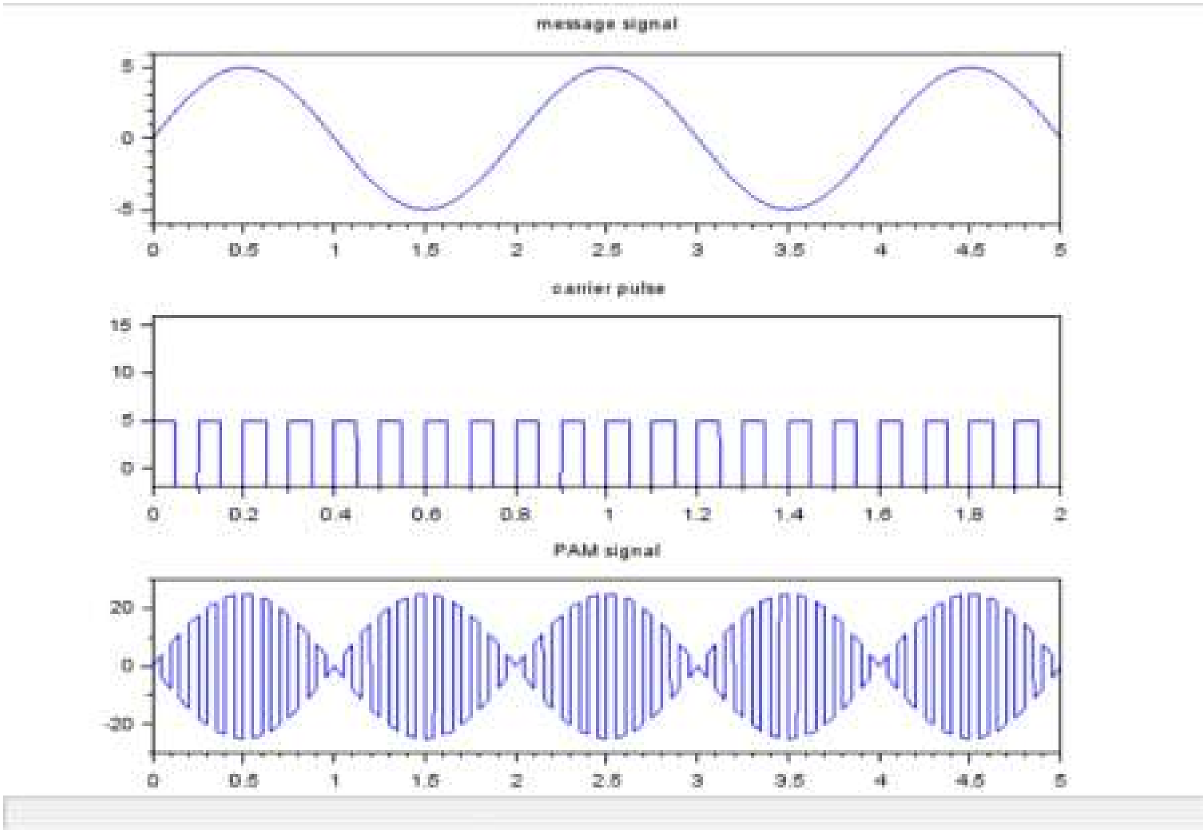
5) What do you mean by pulse time modulation.

=> In pulse time modulation, the pulse will be the same amplitude. However one of the amplitude of the sampled signal. The modulation of the time intervals b/w successive pulses of constant duration and amplitude in accordance with a signal.

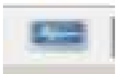
5c.8 SCILAB PROGRAM:

```
clc;
clear;
xdel ( winsid ());
// generate modulating signal
fm = 2;
Am = 5;
t = 0:0.001:5;
Vm = Am*sin(((2*%pi)*fm)*t/4);
subplot(311)
plot(t,Vm)
title('message signal');
// generate carrier signal
fc = 10;
Ac = 5;
Vc = Ac*squarewave (((2*%pi)*fc)*t);
subplot(312)
plot(t,Vc)
title('carrier pulse');
h=gca();
h.data_bounds=[0,-1;2,3*Ac]
// generate pam
pam_mod=Vm.*Vc
subplot(313)
plot(t,pam_mod);
title('PAM signal');
```

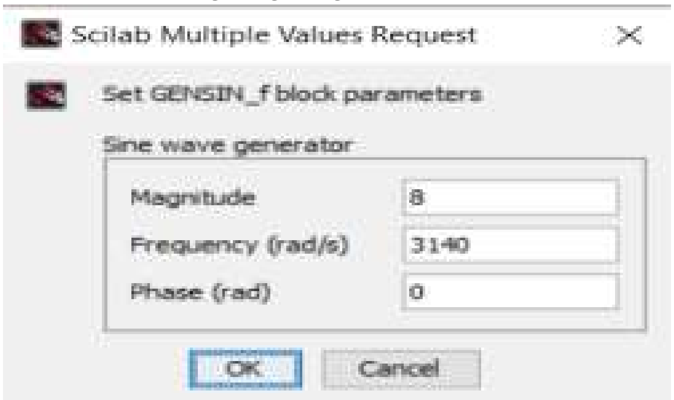
OUTPUT:



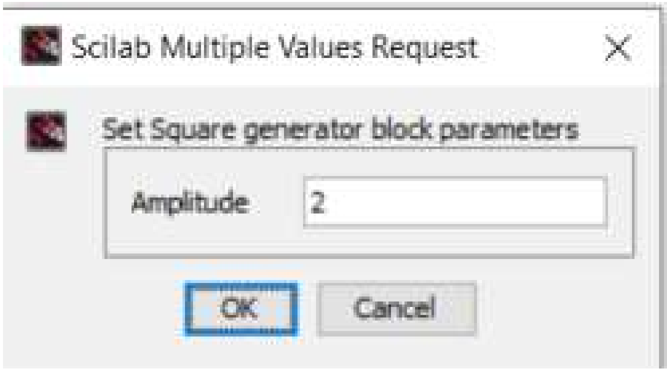
Generation of PAM in Scilab by using XCOS

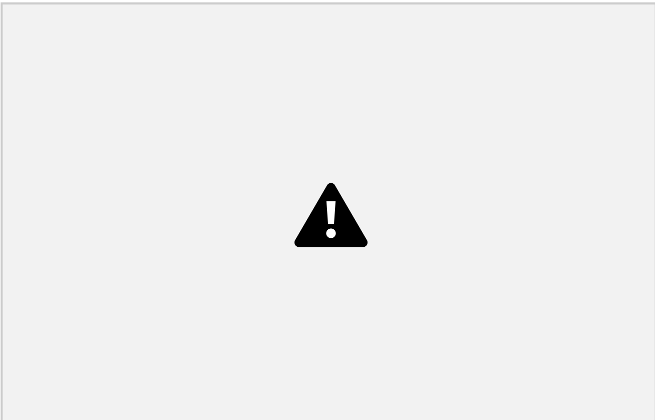
1.Click the XCOS icon 
2.In the palette browser, click source, choose the sine and square wave generator and clock
3.In the sink choose the cscope & In mathematical operation select PROD-f (ie multiplier).Then connect all the blocks as shown below.

4.click on the message signal generator, set the values of message signal as

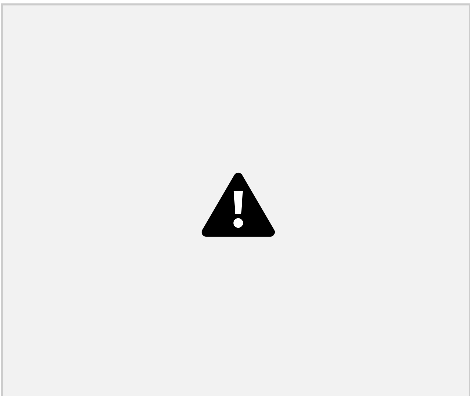


5. click on the carrier signal generator, set the values of carrier signal as

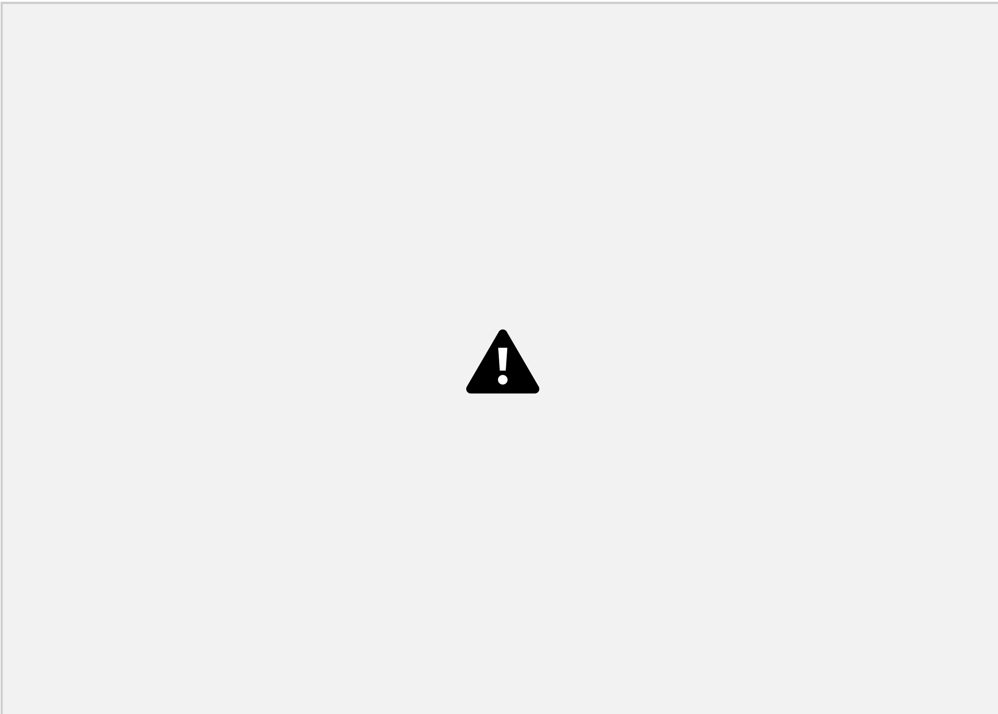




6.Set the cscope clock value as



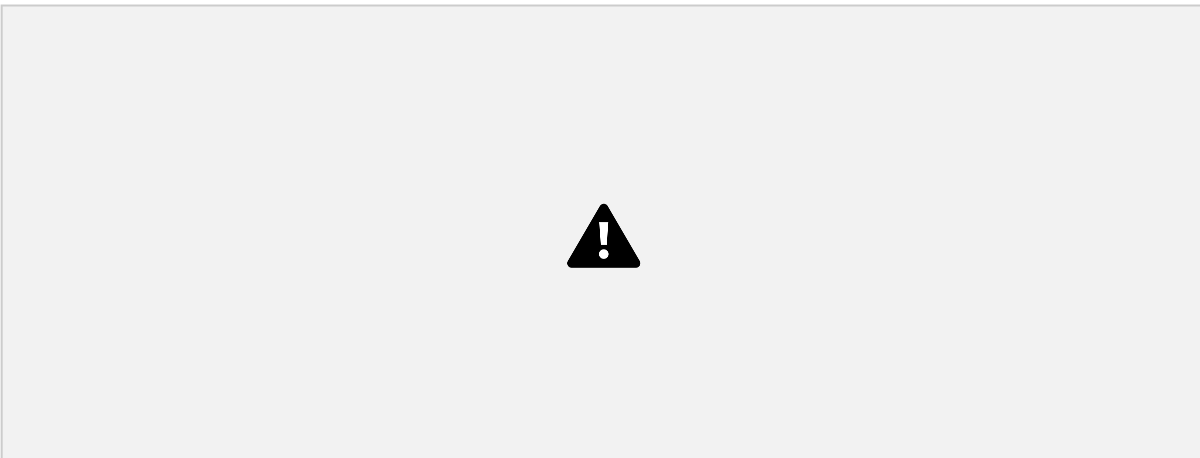
7. Save and execute the PAM generator



8.The generated pam signal will be displayed in the graphical

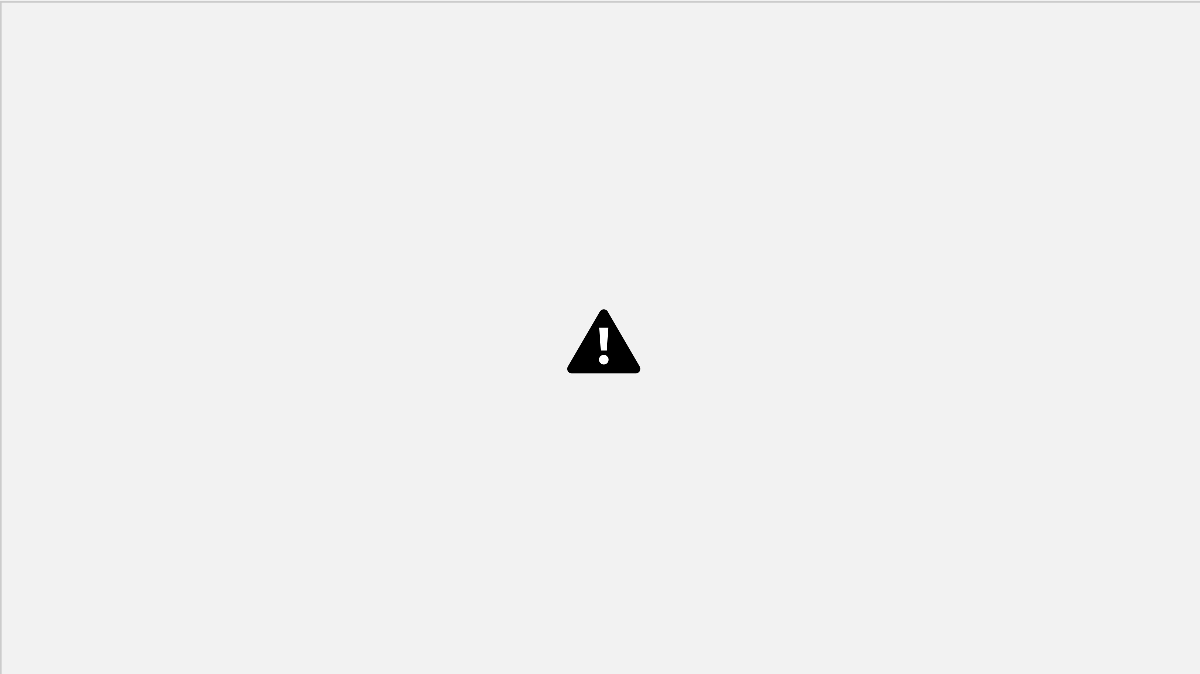
window 9.Vary the parameter of message and carrier ,check your

plot.

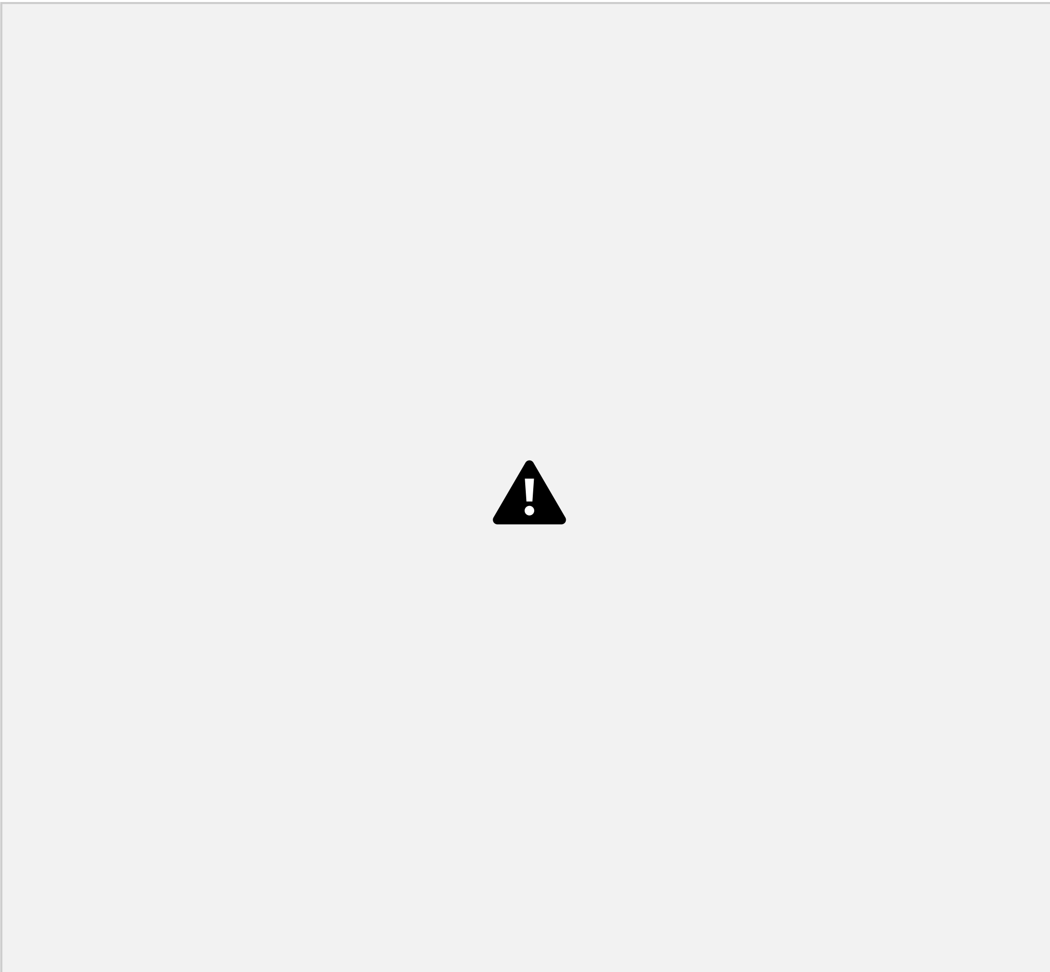


<u>SCI LAB CODE:-</u>

SCILAB SIMULATION O/P:-



SCILAB CODE(XCOS):-

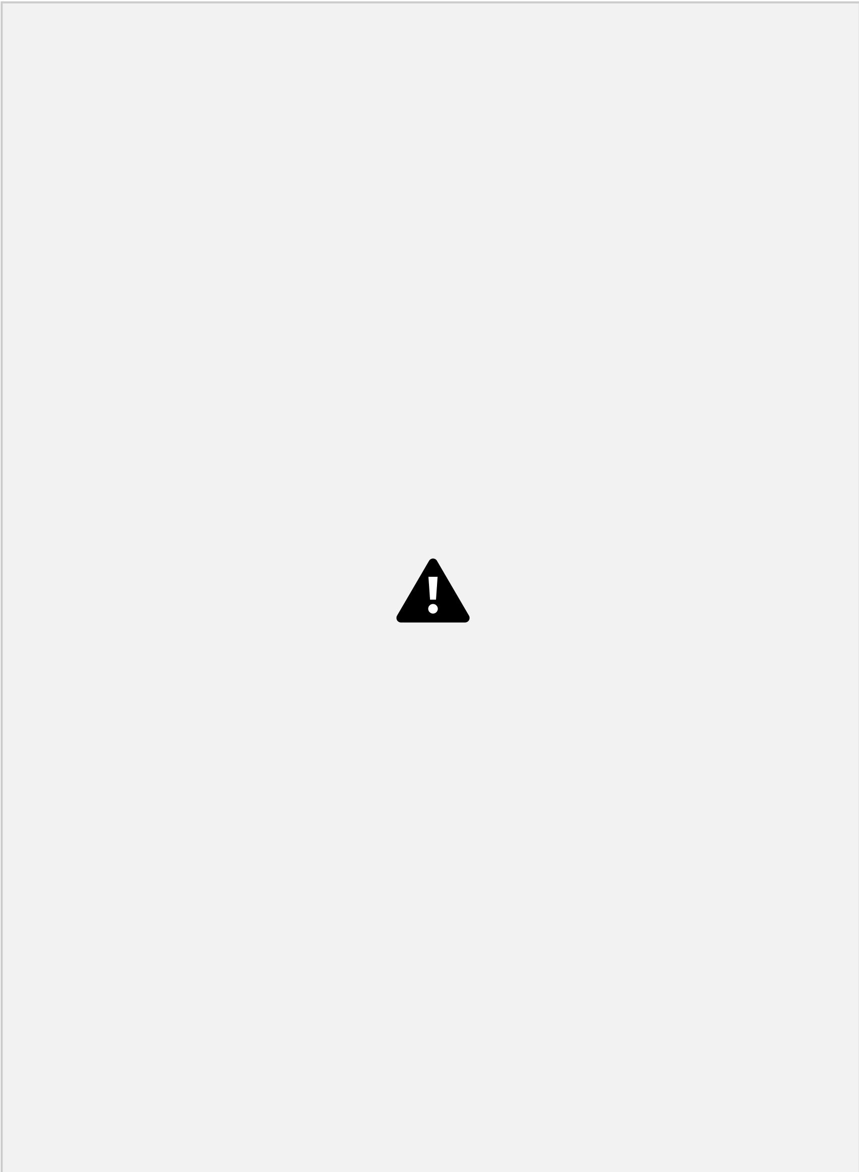


SCILAB SIMULATION O/P:-

5c.9 Postlab Questions

1.What are the different types of PTM systems?
2.What is Pulse duration modulation (PDM)?
3.Explain the principle of PPM?
4.What are the Drawbacks of PAM.
5.Compare different types of Pulse Modulation

POST LAB:-



5c.10 Lab Result

Thus the PWM and its demodulation were performed and the graphs were plotted.

Laboratory Report Cover Sheet

SRM Institute of Science and Technology Faculty of Engineering and Technology Department of Electronics and Communication Engineering
18ECC205J ANALOG AND DIGITAL COMMUNICATION **Fifth Semester, 2021-22 (odd semester)**

Name : Pushpal Das

Register No. : RA1911004010565

Day / Session : 2nd / FN

Venue : ONLINE(G meet)

Title of Experiment : PULSE CODE MODULATION AND DEMODULATION Date of

Conduction : 26/08/2021

Date of Submission : 20/09/2021

Particulars	**Max. Marks**	**Marks Obtained**
Pre-lab questions	05	
In-lab experiment	10	
Post-lab questions	10	
Record Submission	05	
Total	**30**	

REPORT VERIFICATION

Date :01 SEP2021

Staff Name : Dr. P. Malarvizhi / Mrs. Harisudha

Signature :

6 PULSE CODE MODULATION AND DEMODULATION

6.1 Objective

To analyze PCM system and interpret the modulated and demodulated waveforms for various sampling frequency and to find the Signal to Quantization Noise Ratio of PCM system.

6.2 Hardware Required

- Pulse code modulator & transmitter (ST2153), TDM Pulse code
- Demodulator & receiver (ST2154),
- DSO,
- Testing probes and patch chords/connecting wire

6.3 Introduction

Pulse-code modulation (PCM) is a method used to digitally represent sampled analog signals. Analog voice data must be translated into a series of binary digits before they can be transmitted. With PCM, the amplitude of the wave to be transmitted is sampled at regular intervals and translated into a binary number.

In Pulse code modulation (PCM) only certain discrete values are allowed for the modulating signals. The modulating signal is sampled, as in other forms of pulse modulation. But any sample falling within a specified range of values is assigned a discrete value. Each value is assigned a pattern of pulses and the signal is transmitted by means of this code. The electronic circuit that produces the coded pulse train from the modulating waveform is termed a coder or encoder. A suitable decoder must be used at the receiver in order to extract the original information from the transmitted pulse train.

Time Division Multiplexing (TDM) is a type of digital (or rarely analog) multiplexing in which switching takes place between two or more signals (mostly PCM signals), serially in time. In this the time domain is divided into several recurrent time slots of fixed length, one for each sub-channel.

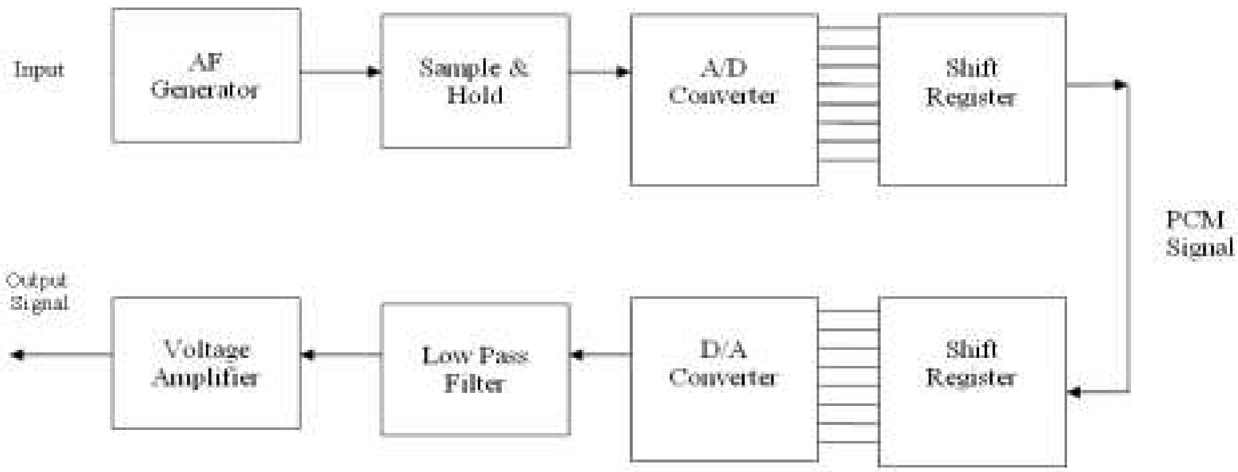
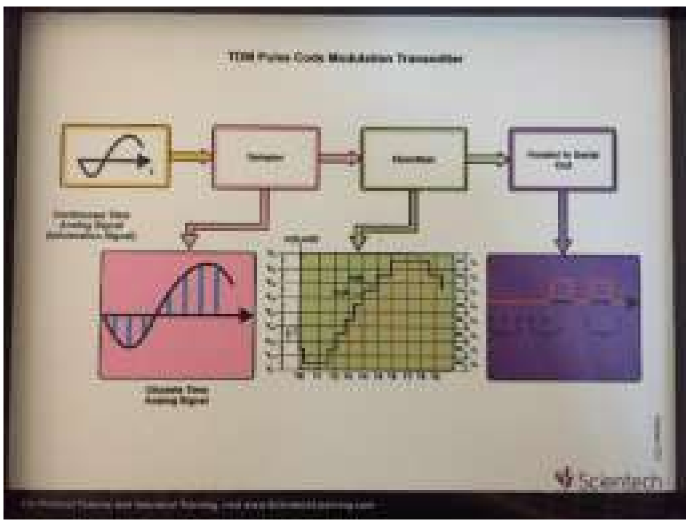
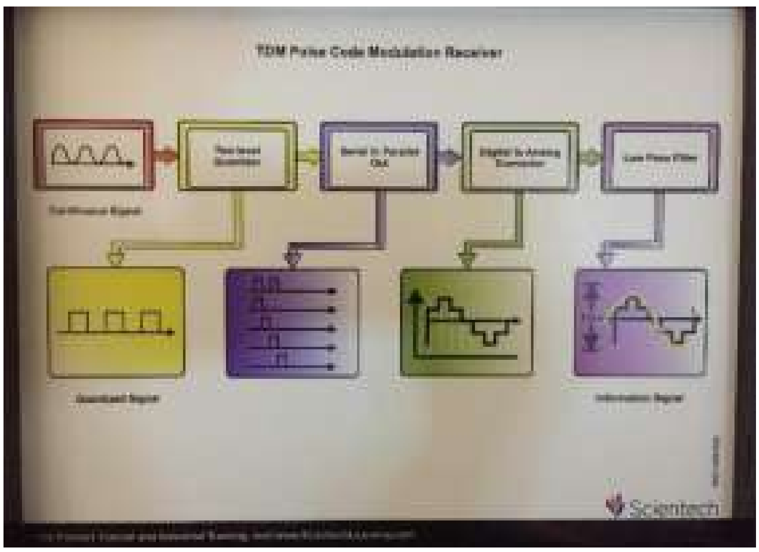


Figure 6..1 Block Diagram of PCM Modulator and Demodulator





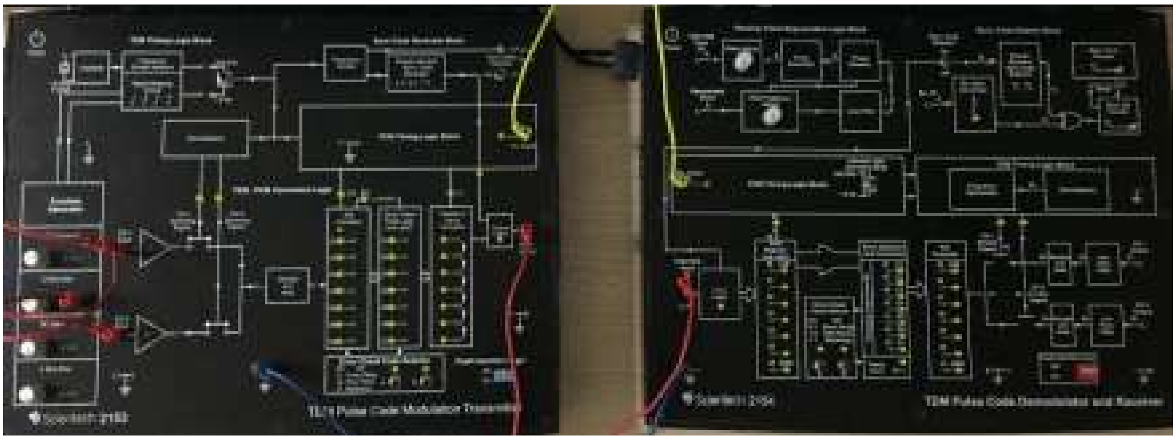


Figure 6.2 Connections between different modules of TDM PCM Transmitter and Receiver (Scientech 2153 & 2154)

6.4 Pre-Lab Questions

1. State sampling theorem.
2. What are the various steps involved in A/D conversion?
3. What is the significant function of Sample and Hold Circuit.

6.5 Procedure

6.5.1 PCM Operation (with AC input)

1. Connect the modulator trainer to the mains and switch on the power supply. 2. Observe the output of the Sine generator using CRO, it should be a sine wave of 2KHz frequency with tuned amplitude.

3. Observe the Sample and Hold output using CRO.

4. Observe the output of the TDM PCM output using CRO.

6.5.2 PCM Operation (with DC input)

Modulation

5. Set DC source to some value say 4.4V with the help of the Knob and connect it to the A/D converter input and observe the output LED's

6. Note down the digital code i.e. output of the A/D converter and compare with the theoretical value.

Theoretical value can be obtained by:

$$A\ D\ Input\ voltage = = \ {}^{/}X\ Y$$

where 1 *LSB Value* (10) (2)

1 LSB Value = $V_{ref}/2^n$

Since V_{ref} = 5 V and n=8

1 LSB Value = 0.01953

For example:

A/D Input voltage = 4.4 V

= $225.28_{(10)}$1

= $1110\ 0001_{(2)}$

So digital output is 1110 0001

7. Observe the Sample and Hold output using CRO.
8. Observe the output of the TDM PCM output using CRO.

Repeat the above steps for other Sampling frequencies.

Note: From this waveform you can observe the LSB bit enters the output first.

6.5.3 Demodulation

9. Connect TDM PCM signal of the modulator to the demodulators with the help of patch chord.
10. Connect transmitter clock to the receiver clock.
11. Observe the Sample and Hold output using CRO.
12. Observe the output of the LPF output using CRO.

Sample work sheet

1. Modulating signal : 4.4 V
2. A/D Output (theoretical) : $1110\ 0001_{(2)}$
3. A/D Output (practical) : $1110\ 0001_{(2)}$
4. S-P Output : $1110\ 0001_{(2)}$
5. D/A Converter output : 4.4 V

6.6 Model Graph

i) With AC Input

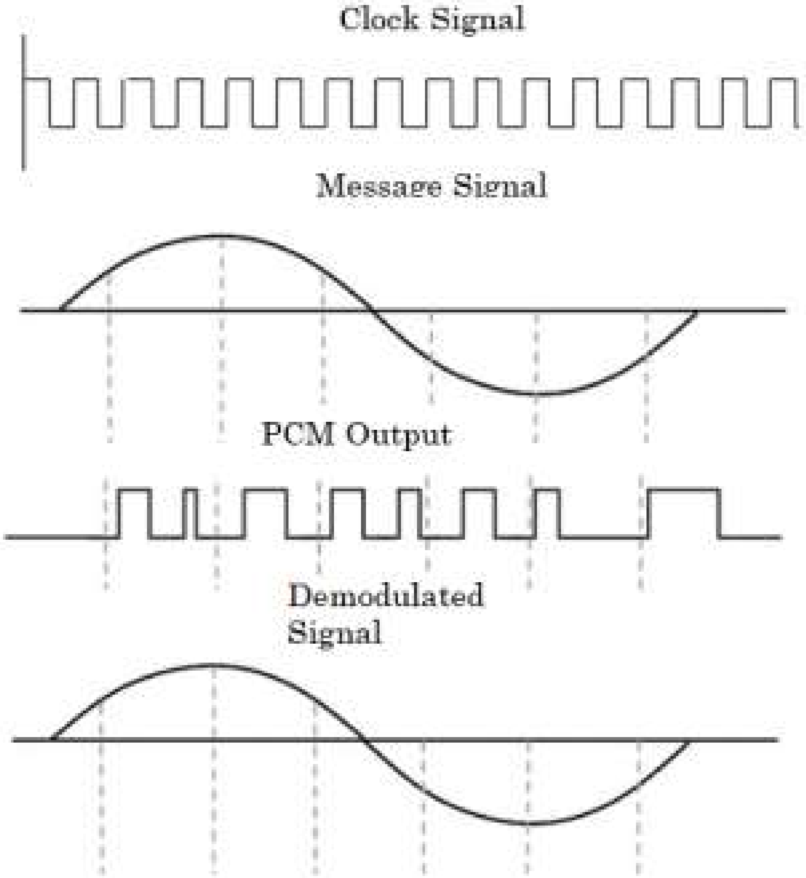


Figure 6.3 Model Graph of PCM Waveform with AC input

ii)With DC Input

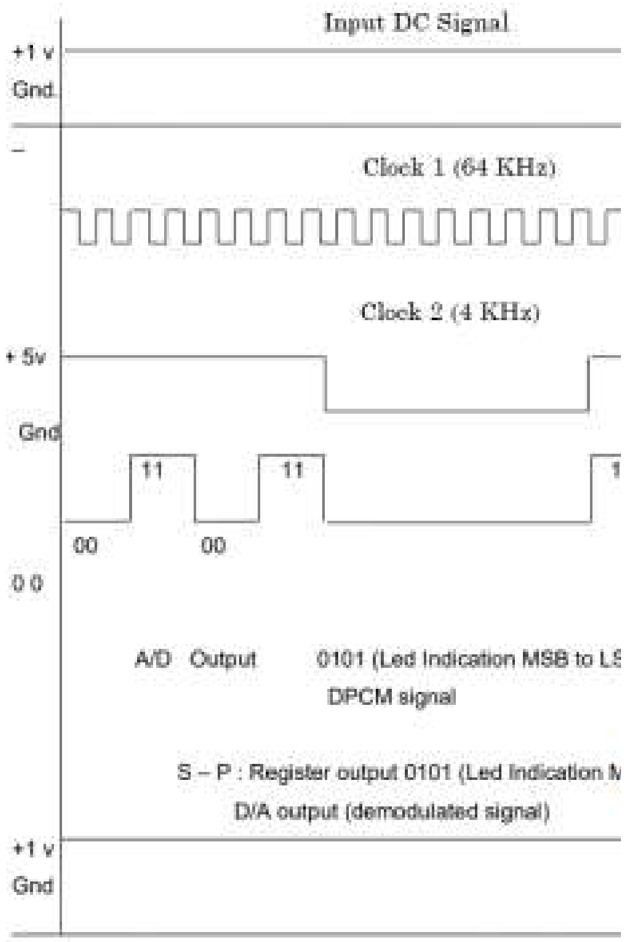


Figure 6.4 PCM Waveform with DC input

6.7 Observation

Signal Amplitude Time

e

Time Period

Period Signal Amplitud

PCM Modulation (With AC input) PCM Modulation (With DC input)

AC input

Clock Signal

Sample and hold signal

PCM Output

	Clock Signal
	PCM Outpu

PCM Demodulation (with AC input) PCM Demodulation (with DC input)

D/A Converter output Signal LPF output signal Demodulated output

	LPF output sig

Demodulated

Output

6.8 SIMULATION LAB

6.8.1 Software

Open Source SCILAB Software 6.0.2

6.8.2 Problem Statement

Consider a signal x(t) = sin 2π5t is to be quantized into 8 levels. The number of samples is 1000. Write a command using Scilab to plot the following signal: (i) Original signal

(ii) Quantized signal

(iii)Determine the encoding bit for each Sample.

Program

```
// Experiment 6 : // Pulse Code Modulation and Demodulation

// Generate PCM Signal with quantization levels of 8 and 1000 samples for the signal x(t)

clc;
close;
clear;

//Generate Message Signal for the specified frequency and amplitude
t = 0:0.001:1;
f=input('Enter the frequency of message signal');
x = sin(2*%pi*f*t);
//Plotting of Signals
//Message Signal
figure(1)
subplot(4,1,1);
plot(t*2*%pi,x);
title('Messgae Signal');
xlabel('time in seconds');
ylabel('Amplitude');

//Quantization

L = input('Enter the Quantization Levels');
xmax = max(abs(x));
xq = x/xmax;
en_code = xq;
d = 2/L;
q = d*[0:L-1];
q = q-((L-1)/2)*d;
for i = 1:L
```

```
    xq(find(((q(i)-d/2)<=xq)&(xq<=(q(i)+d/2))))=q(i).*ones(1,length(find(((q(i)-
d/2)<=xq)&(xq<=(q(i)+d/2)))));
    en_code(find(xq == q(i)))= (i-1).*ones(1,length(find(xq== q(i))));
end
xq = xq*xmax;
disp(xq);
// Plotting of Sampled version of the message signal and quantization steps
subplot(4,1,2);
plot2d3(t*2*%pi,xq,5);
title('Sample Signal');
xlabel('time in seconds');
ylabel('Amplitude');
subplot(4,1,3);
plot2d2(t*2*%pi,x);
title('Quantized Signal');
xlabel('time');
ylabel('Steps');
xq=xq*xmax;
//Visualization of Message and Quantized Signal
figure(2)
plot(t*2*%pi,x);
plot2d2(t*2*%pi,xq,5);
title('Message and Quantized Signal');
xlabel('time in seconds');
ylabel('Amplitude');

//Encoding Process
n = log2(L);
c = zeros(length(x),n);
for i = 1:length(x)
            for j = n:-1:0
    if(fix(en_code(i)/(2^j)) = =1)
    c(i,(n-j)) =1;
    en_code(i) = en_code(i)-2^j;
    end
    end
end
disp('Encoded Bits');
disp(c)
```

Note:

Students should use comment statements to include the Experiment title, Course Code, Student name and register number, Date of performing the experiment, Scilab version 6.0.2
(for example: // Experiment 5 – Pulse Code Modulation and Demodulation)

Appropriate Comment statement should be written wherever necessary. (eg: //Generation of message signal)

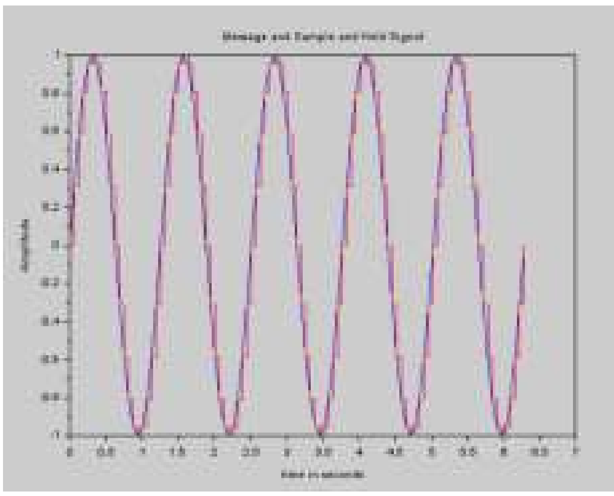
Observation – Simulation

PCM Modulation (With AC input)

Signal	**Amplitude**	**Time Period**	**Frequency**
AC input	1v	1.25s	0.8hz
Sample and Hold signal	0.87v	1.25s	0.8hz

6.8.3 Model graph

Figure (1) Figure(2)



Samples of Encoded Bits:

```
Encoded Bits

   0.   0.   0.
   0.   0.   0.
   0.   0.   0.
   0.   0.   0.
   0.   0.   0.
   0.   0.   1.
   0.   0.   0.
   0.   0.   0.
   0.   0.   0.
   0.   0.   0.
   0.   0.   0.
   0.   0.   0.
   0.   0.   0.
   0.   0.   0.
   0.   0.   0.
   0.   0.   0.
   0.   0.   0.
   0.   0.   0.
   0.   0.   0.
   0.   0.   0.
   0.   0.   0.
   0.   0.   0.
   0.   0.   0.
   0.   0.   0.
   0.   0.   0.
```

6.9 Determination of Signal to Quantization Noise ratio of PCM signal *Write a Scilab code to determine the Signal to quantization noise ratio for a TV signal operating at 4.2 MHz is transmitted using binary PCM system using 512 quantization levels. Also, Determine*

(a) Code word-length and (b) The PCM bandwidth

Program

```
//PCM - Signal to Quantization Noise Ratio
clear;
clc;
// Frequency of operation
fm=input('Enter the frequency in MHz :'); // /Enter the frequency as per the
specification
f_m = fm*10^6; //television signal of 4.2 MHz to Hz
// Enter the levels as per the specification
q=input('Enter the Quantization Levels : ');
//Calculations
//Number of bits and quantization levels are related in binary PCM as q = 2^v
//where v is code word length
```

```
v = log2(q); //code word length
BW = 2*v*f_m; //transmission channel bandwidth
//output signal to noise ratio which is less than or equal to obtained value
SQNR_dB = 1.8 + 6*v;
//results
printf("\n\n Code word length = %.2d bits",v);
printf("\n\n Transmission bandwidth = %.2f Hz",BW);
printf("\n\n.Output signal to quantization noise ratio = %.2f dB",SQNR_dB);
```

Output

```
Code word length = 09 bits
Transmission bandwidth = 75600000.00 Hz
Output signal to quantization noise ratio = 55.80 dB
```

6.10 Post Lab Questions

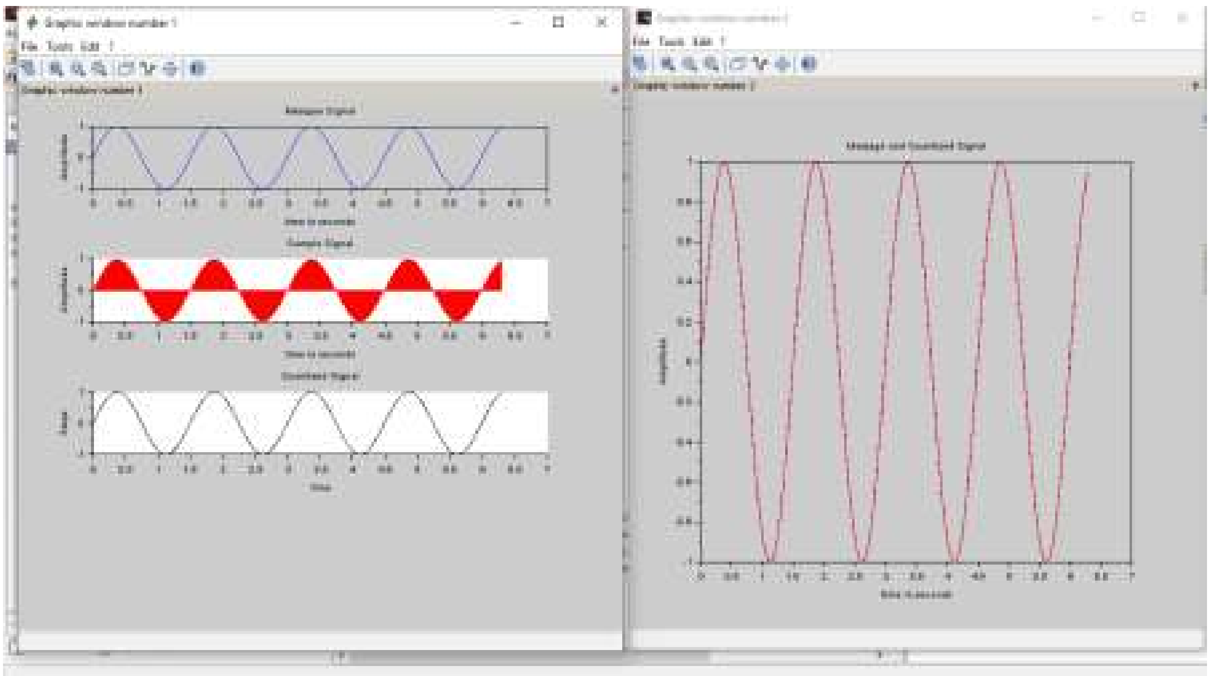
1. A signal is sampled 8 kHz and it is quantized using 8 – bit uniform quantizer assuming SNRq for the sinusoidal signal. Find the bit rate (R_b) and SNR of PCM. Use Scilab code to determine the parameters.
2. Find the A/D Converter output for input DC voltage of 3.6V.

6.11Simulation

1.PCM(Sci Lab):-

```
// Generate PCM Signal with quantisation levels of 8 and 1000 samples for the signal x(t)
clc;
close;
clear;
//Generate Message Signal for the specified frequency and amplitude
t = 0:0.001:1;
f=input('Enter the frequency of message signal');
x = sin(2*%pi*f*t);
//Plotting of Signals
//Message Signal
figure(1)
subplot(4,1,1);
plot(t*2*%pi,x);
title('Message Signal');
//Quantization
L = input('Enter the Quantization Levels');
xmax = max(abs(x));
xq = x/xmax;
en_code = xq;
d = 2/L;
xlabel('time in seconds');
ylabel('Amplitude');
q = d*[0:L-1];
q = q-((L-1)/2)*d;
for i = 1:L
    xq(find(((q(i)-d/2)<=xq)&(xq<=(q(i)+d/2))))=q(i).*ones(1,length(find(((q(i)-d/2)<=xq)&(xq<=(q(i)+d/2)))));
    en_code(find(xq == q(i)))= (i-1).*ones(1,length(find(xq== q(i))));
end
xq = xq*xmax;
disp(xq);
// Plotting of Sampled version of the message signal and quantisation steps
subplot(4,1,2);
plot2d3(t*2*%pi,xq,5);
title('Sample Signal');
```

SIMULATION O/P:-



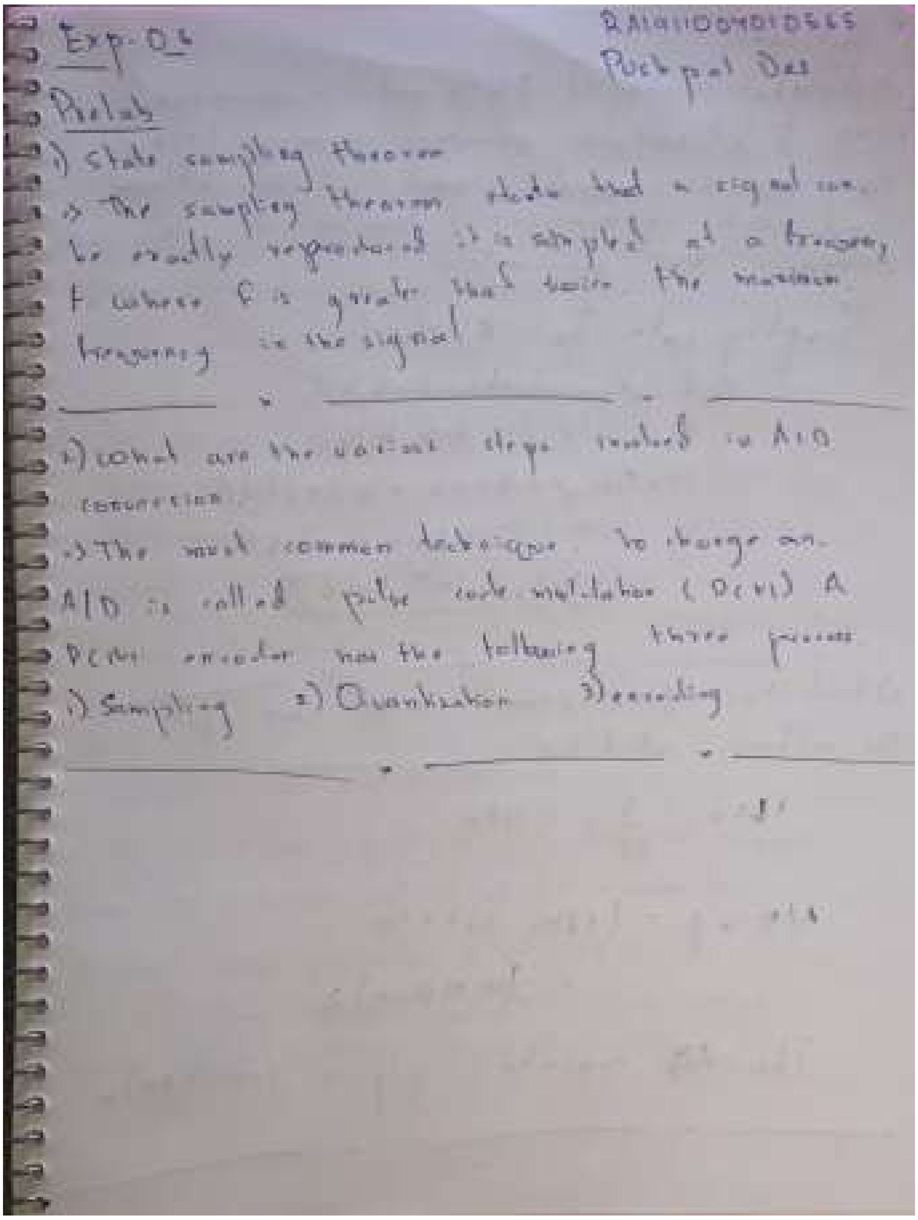
```
"Encoded Bits"

  0.   1.   1.
  1.   0.   0.
  1.   0.   0.
  1.   0.   0.
  1.   0.   0.
  1.   0.   0.
  1.   0.   0.
  1.   0.   0.
  1.   0.   0.
  1.   0.   1.
  1.   0.   1.
  1.   0.   1.
  1.   0.   1.
  1.   0.   1.
  1.   0.   1.
  1.   0.   1.
  1.   0.   1.
  1.   1.   0.
  1.   1.   0.
  1.   1.   0.
  1.   1.   0.
  1.   1.   0.
  1.   1.   0.
  1.   1.   0.
  1.   1.   0.
  1.   1.   0.
  1.   1.   0.
  1.   1.   1.
  1.   1.   1.
  1.   1.   1.
  1.   1.   1.
```

2.DETERMINATRION OF SIGNAL TO QUANTIZATION NOISE RATIO OF PCM SIGNAL

```
//EXP 6-Pulse code Modulation and Demodulation

//PCM - Signal to Quantisation Noise Ratio
clear;
clc;
// Frequency of operation
fm=input('Enter the frequency in MHz:');  // /Enter the frequency as per the specification
f_m = fm*10^6;   //Television signal of 4.2 MHz  to Hz
// Enter the levels as per the specification
q=input('Enter the Quantisation Levels:');
//Calculations
//Number of bits and quantisation levels are related in binary PCM as q = 2^v
//where v is code word length
v = log2(q);    //code word length
BW = 2*v*f_m;  //transmission channel bandwidth
//output signal to noise ratio which is less than or equal to obtained value
SQNR_dB = 1.8 + 6*v;
//results
printf("\n\n  Code word length  = %.2d bits",v);
printf("\n\n  Transmission bandwidth = %.2f Hz",BW);
printf("\n\n Output signal to quantization noise ratio = %.2f dB",SQNR_dB);
```

```
Enter the frequency in MHz: 4.2

Enter the Quantization Levels: 512



  Code word length  = 09 bits

  Transmission bandwidth = 75600000.00 Hz

.Output signal to quantization noise ratio = 55.80 dB
-->
```

6.12 Pre lab Answers:



Exp-06

RA1011004010565
Pushpal Das

Prelab

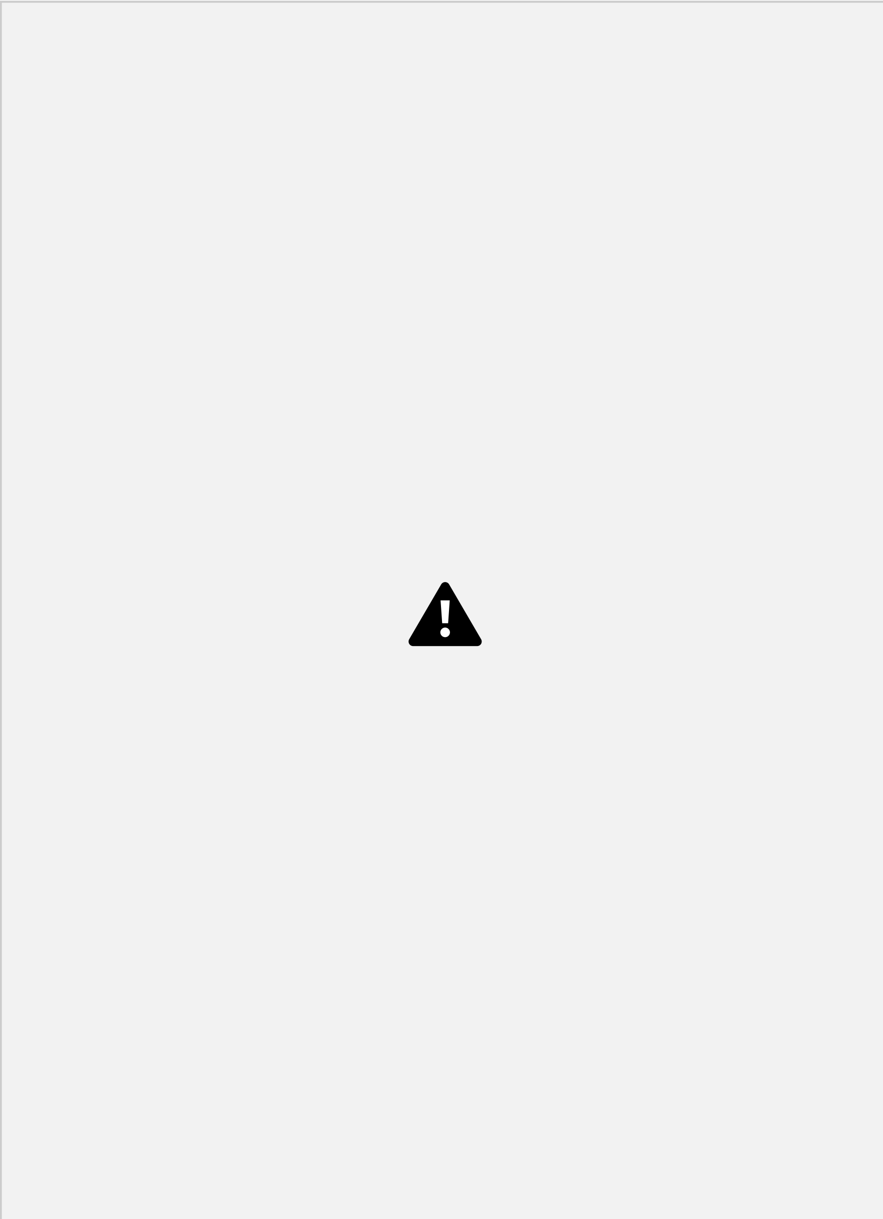
1) State sampling theorem

=> The sampling theorem states that a signal can be exactly reproduced if it is sampled at a frequency F where F is greater than twice the maximum frequency in the signal.

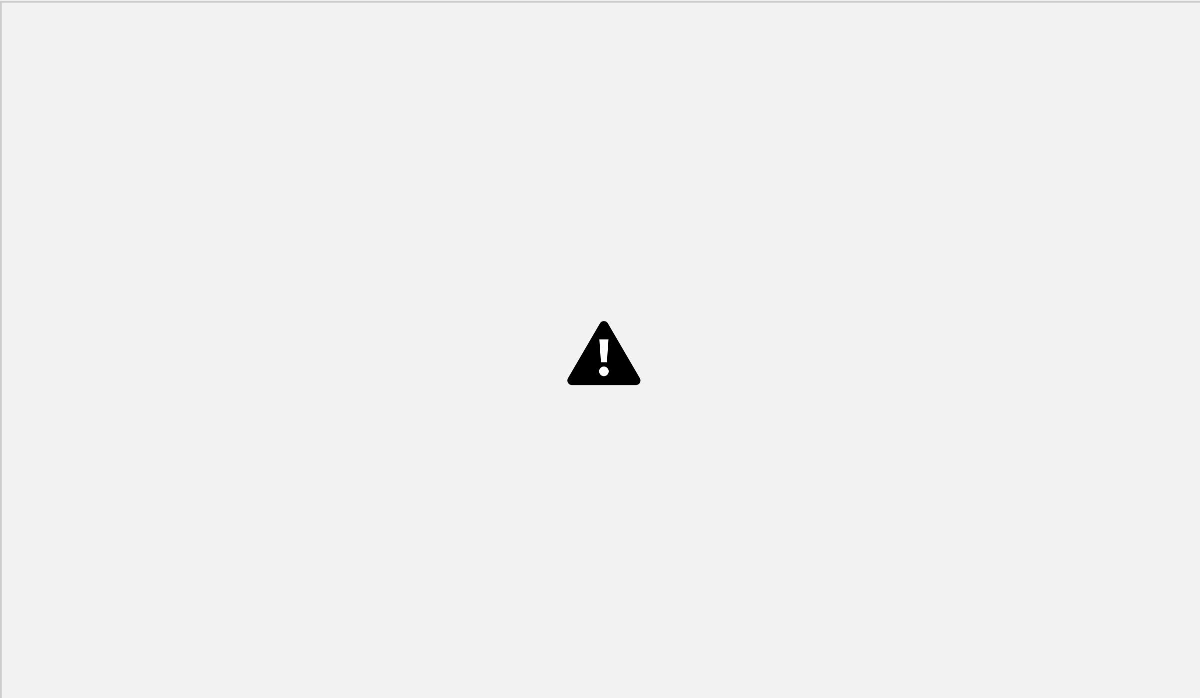
2) What are the various steps involved in A/D conversion

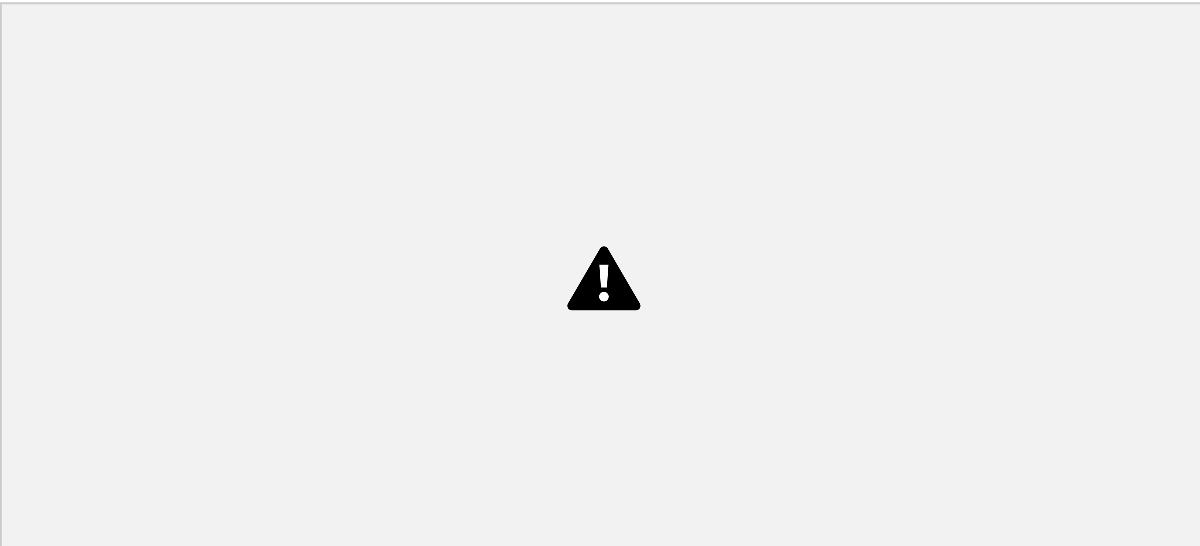
=> The most common technique to change an A/D is called pulse code modulation (PCM). A PCM encoder has the following three process
1) Sampling 2) Quantization 3) Encoding

6.13 Post lab Answers:



POST LAB(Q.01):-





6.14 Result

Thus, the PCM modulation is observed for specified quantization levels and Signal to Quantization noise ratio is determined through Simulation.

SRM Institute of Science and Technology Faculty of Engineering and Technology Department of Electronics and Communication Engineering
18ECC205J ANALOG AND DIGITAL COMMUNICATION Fifth Semester, 2021-22 (odd semester)

Name : Pushpal Das

Register No. : RA1911004010565

Day / Session : 2ND / FN

Venue : Online (G Meet)

Title of Experiment : 7.DIFFERENTIAL PULSE CODE MODULATION AND DEMODULATION

Date of Conduction : 03/09/2021

Date of Submission : 20/09/2021

Particulars	**Max. Marks**	**Marks Obtained**
Pre-lab questions	05	
In-lab experiment	10	
Post-lab questions	10	
Record Submission	05	
Total	**30**	

REPORT VERIFICATION

Date : 13 SEP 2021

Staff Name : Dr. P. Malarvizhi / Mrs. Harisudha

Signature :

7 DIFFERENTIAL PULSE CODE MODULATION AND DEMODULATION

7.1 Objective

To analyze DPCM system and to interpret the modulated and demodulated waveforms for various sampling frequency and simulate DPCM wave in Scilab.

7.2 Hardware Required

Adcl-07 Kit

20 MHz Dual Trace Oscilloscope

Connecting Chords

Power Supply

Note: Keep the Switch faults in Off Position

7.3 Introduction

Differential PCM is quite similar to ordinary PCM. Each word in this system indicates the difference in amplitude, positive or negative, between this sample and the previous sample. Thus the relative value of each sample is indicated rather than, the absolute value in normal PCM.

Block Diagram

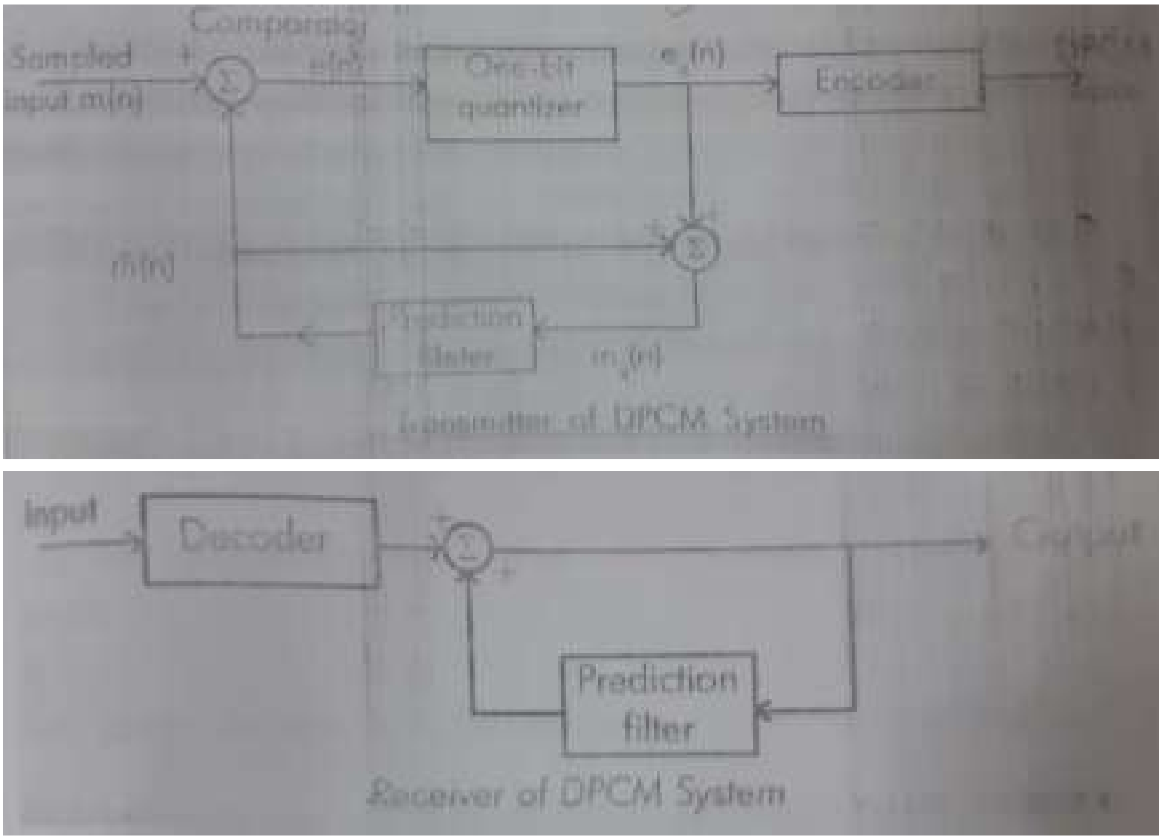


Figure 7.1 Block diagram of DPCM System

Figure 7.2 DPCM trainer kit

7.4 Prelab Questions

1. Compare between DPCM and PCM.
2. State the significance of predictor in DPCM?
3. State the significance of accumulator in DPCM?
4. Justify how DPCM is used for speech compression?

<u>PRE LAB :-</u>

Exp-01

RA1411004010565
Pushpa 1 Das

1) Compare DPCM and PCM

DPCM	PCM
• Stands for Differential pulse code modulation	• Stands for pulse code modulation
• DPCM feedback is provided	• In PCM, feedback is not provided
• More efficient than PCM	• It is less efficient than DPCM
• DPCM is simple in terms of complexity	• PCM is complex than DPCM in terms of complexity

---

2) What is the significance of predictor in DPCM?

=> The predictor produces the assumed samples from the previous output of the transmitter circuit

=> The same predictor circuit is used in the decoder to reconstruct the original input

---

3) What is the significance of accumulator in DPCM?

=> The significance of accumulator in DPCM, a quantizer precede the differencing of adjacent quantized samples, the decoder's an accumulator, which is correctly initialized exactly restore the quantized signals.

4) Why DPCM is used for speech compression?

=> The DPCM technique mainly used speech, img, & audio signal compression. DPCM conducted on signals with the correlation b/w successive samples leads to good compression ratios.

=> Therefore, it can be utilized for lossless compression.

7.5 Procedure

7.5.1 DPCM modulation

1. Refer to the block diagram and carry out the following Connections and switch settings.

2. Connect power supply in proper polarity to the kit ADCL-07 and switch it ON. 3. Keep the clock frequency at 512 KHz by changing the jumper position of JP1 in the clock generator section

4. Keep the amplitude of the onboard sine wave, of frequency 500Hz to 1Vpp DPCM modulation.

5. Connect the 500Hz sine wave to the lN post of Analog Buffer.

6. Connect OUT post of Analog Buffer to lN post of DPCM modulator section. 7. Observe

the sample output at the given test point the input signal is sampled at the clock frequency of 16 KHz.

8. Observe the linear predictor output at the PREDICTED OUT post of the Linear predictor in the DPCM modulator section.
9. Observe the differential pulse code modulate data (DPCM) at the DPCM OUT post of the DPCM modulator section.
10. Observe the DPCM data at DPCM OUT post by varying input signal from to 2V.

7.5.2 DPCM demodulation

1. Connect the DPCM modulated data from the DPCM OUT post of the DPCM Modulator to the lN post of the DPCM demodulator.
2. Observe the demodulated data at the output of summation block.
3. Observe the integrated demodulated data at the DEMOD OUT post of the DPCM demodulator.
4. Connect the demodulated data from the DEMOD OUT post of the DPCM demodulator to the lN post of the low-pass filter.
5. Observe the reconstructed signal at the OUT post of the filter. Use RST switch for clear observation of output.
6. Now, simultaneously reduce the clock frequencies from 512 KHz to 256 KHz,128 KHz and 64 KHz by changing the jumper position of JP1 and observe the difference in the DPCM modulated and demodulated data. As the frequency of clock decreases DPCM Demodulated data at DEMOD OUT becomes distorted.
7. Observe various waveform as mentioned below

7.6 Model Graph

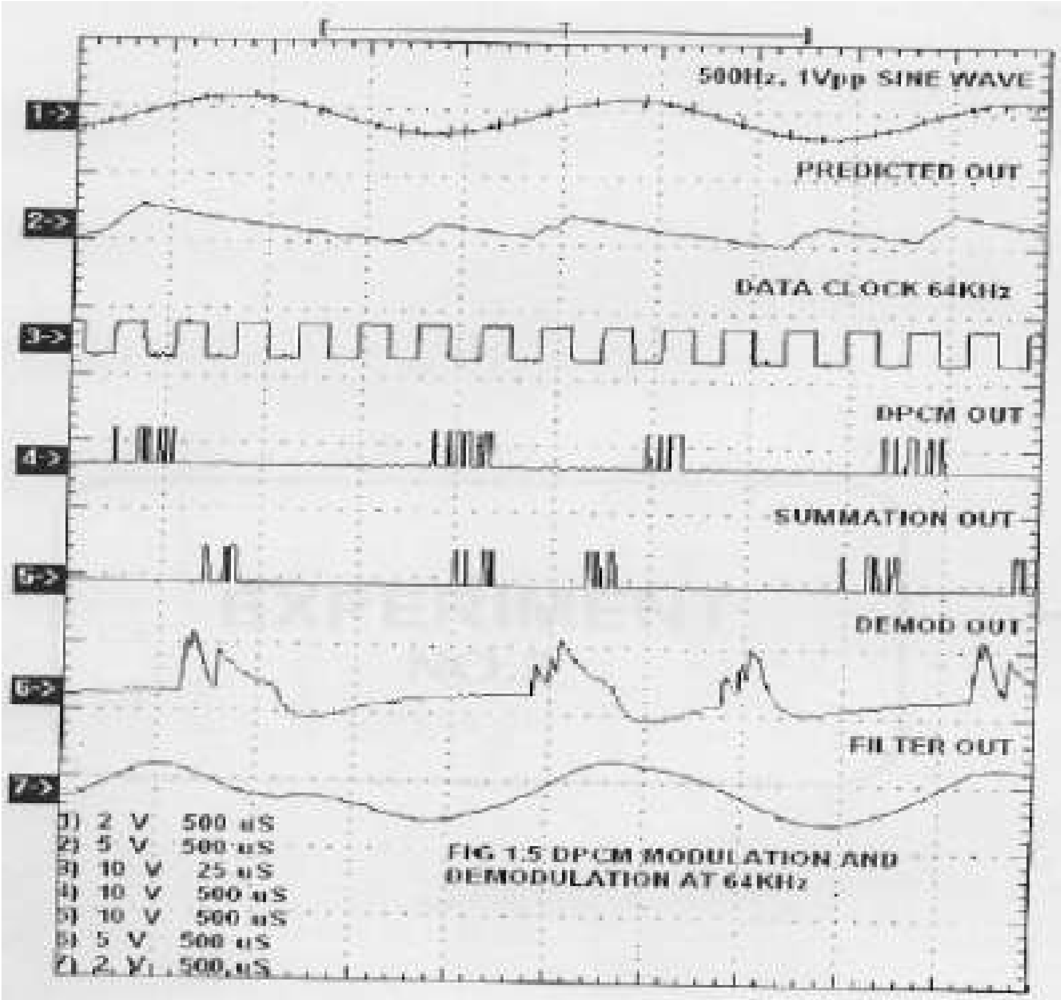


Figure 7.2 DPCM Operation (With Ac Input) Modulation & Demodulation

Observation

ON KIT ADCL-07

Observe the following waveforms on the oscilloscope and plot on the paper.

1. 500 Hz , 1 V pp input sine wave.
2. Sampled out at the provided test point SAMPLER OUT
3. Linear predictor out at PREDICTED OUT post.
4. DPCM data at DPCM OUT post
5. Line interface out at the given output test point of line interface block in DPCM Demodulator
6. Demodulated DPCM data at the output test point of summation block in DPCM demodulator.

7. Integrated demodulated data at the DEMOD OUT post of the DPCM demodulator
8. Reconstructed sine wave at the OUT post of the filter
9. Observe the data at different clock rates.

DPCM Operation - with AC input

Modulation		
	Amplitude	**Time Period**
AC Input		
Clock – 1 Output		
Sample and Hold Output		
DPCM Output		

Demodulation		
	Amplitude	**Time Period**
DPCM Input		
D/A Converter Output		
LPF Output		
Demodulated output Prediction Filter Output		

7.7 VIRTUAL LAB

Software : SCILAB 6.0.2

Exercise

Perform differential pulse code modulation using SCILab for the following specifications.

Amplitude of the signal : 2 V, message frequency is 4Hz, sampling frequency is 80Hz

7.7.1. Scilab Code

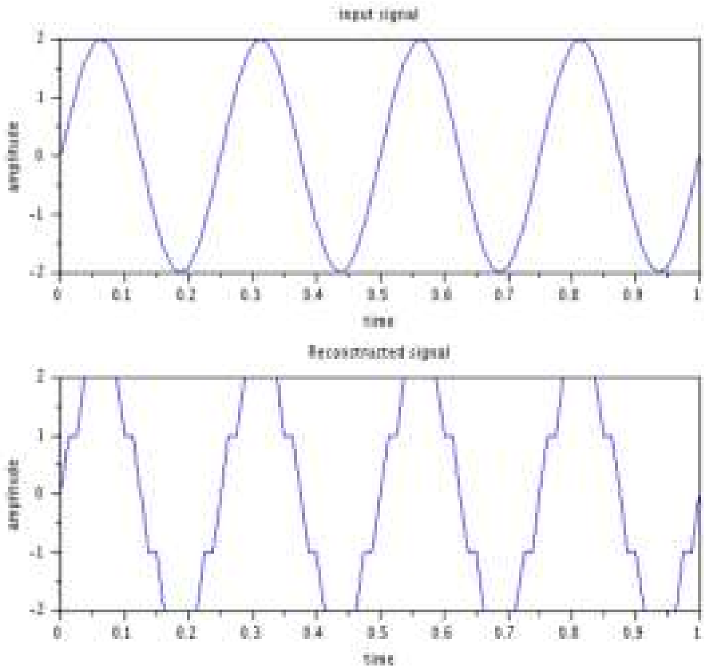
```
clc;
clear;
fm=4;
```

```
fs=20*fm;
am=2;
t=0:1/fs:1;
x=am*sin(2*%pi*fm*t);
subplot(2,1,1);
plot(t,x);
xlabel('time');
ylabel('amplitude');
title('input signal');
for n=1:length(x)
 if n==1
 e(n)=x(n);
 eq(n)=round(e(n))
 xq(n)=eq(n)
 else
 e(n)=x(n)-xq(n-1)
 eq(n)=round(e(n))
 xq(n)=eq(n)+xq(n-1)
 end
end
subplot(2,1,2);
plot(t,xq);
xlabel('time');
ylabel('amplitude');
title('quantized signal');
```

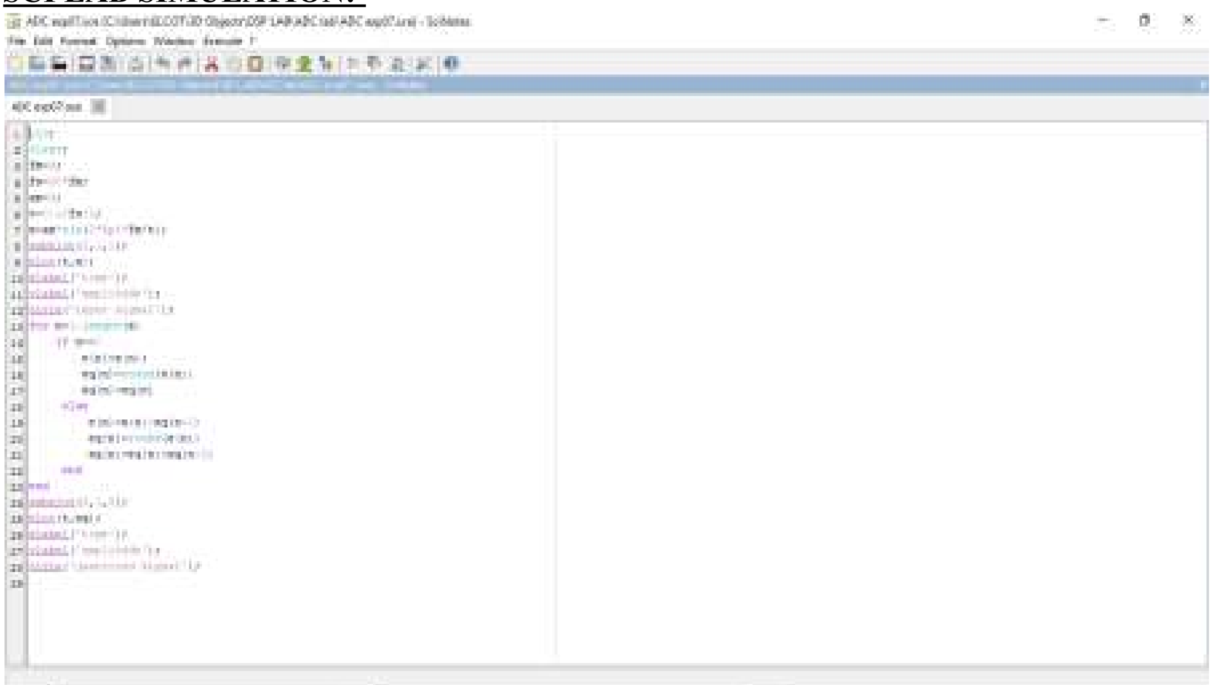
Observations

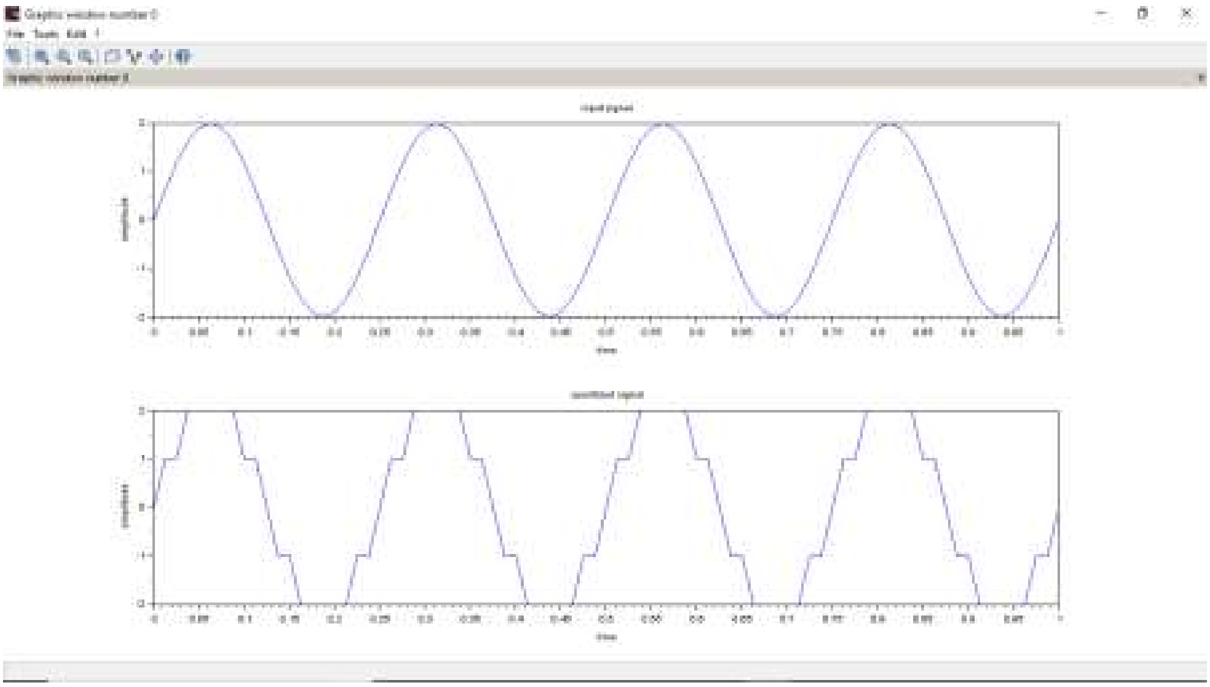
Signal	**Amplitude**	**Time period**
Input Signal	**2V**	**0.25ms**
Reconstructed Signal	**2V**	**0.25ms**

7.7.2 Model graph



SCI LAB SIMULATION:-





7.8 Post Lab Questions

1. What is the need for compression? Mention the types of compression.
2. List the communication standards which use DPCM.
3. Name the circuit used to achieve synchronization between transmitter and receiver. 4. Use Scilab to perform differential pulse code modulation for the input signal of amplitude 10V and the frequency of 5K.

<u>**POST LAB:-**</u>

Post-lab

1) What is the need for compression? Mention the types of compression.

=> Compressed files require significantly less storage capacity than uncompressed files.

=> Compressed files also required less time for transfer while consuming less network bandwidth.

=> This can be also help with costs and also increase productivity.

---

2) List the communication standards which are open.

=> Microphone.

=> Audio compression

=> Lines in circuit.

=> Compressed audio format

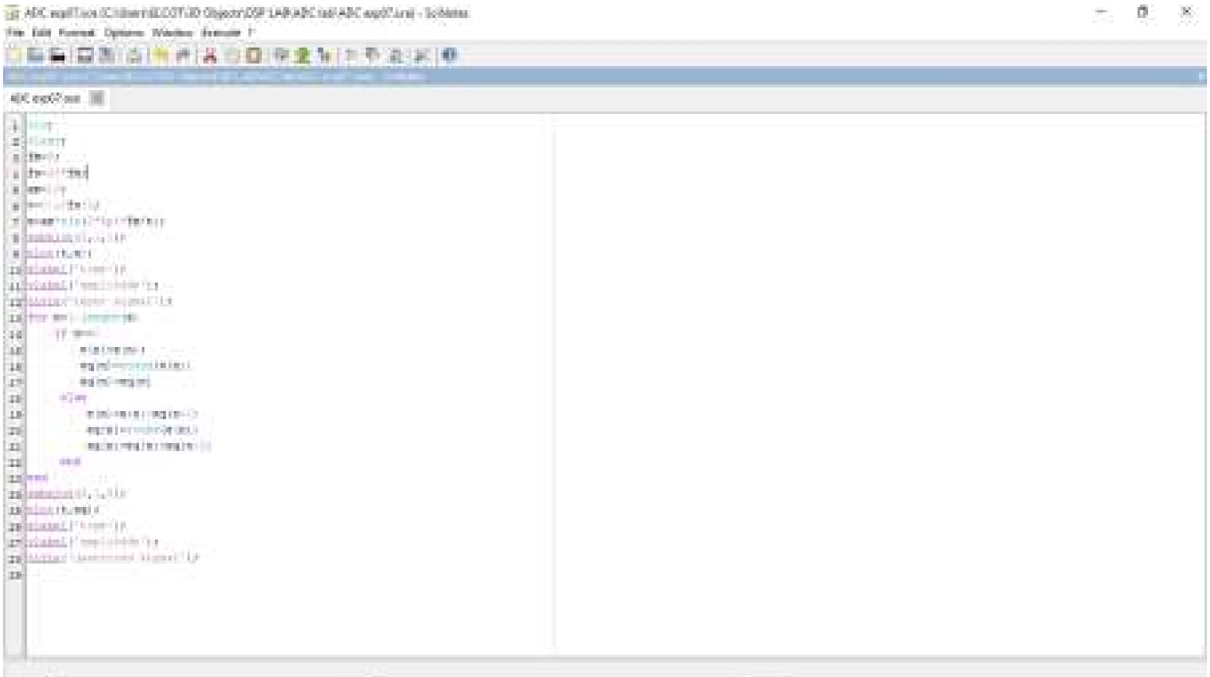
=> Learn quantization.

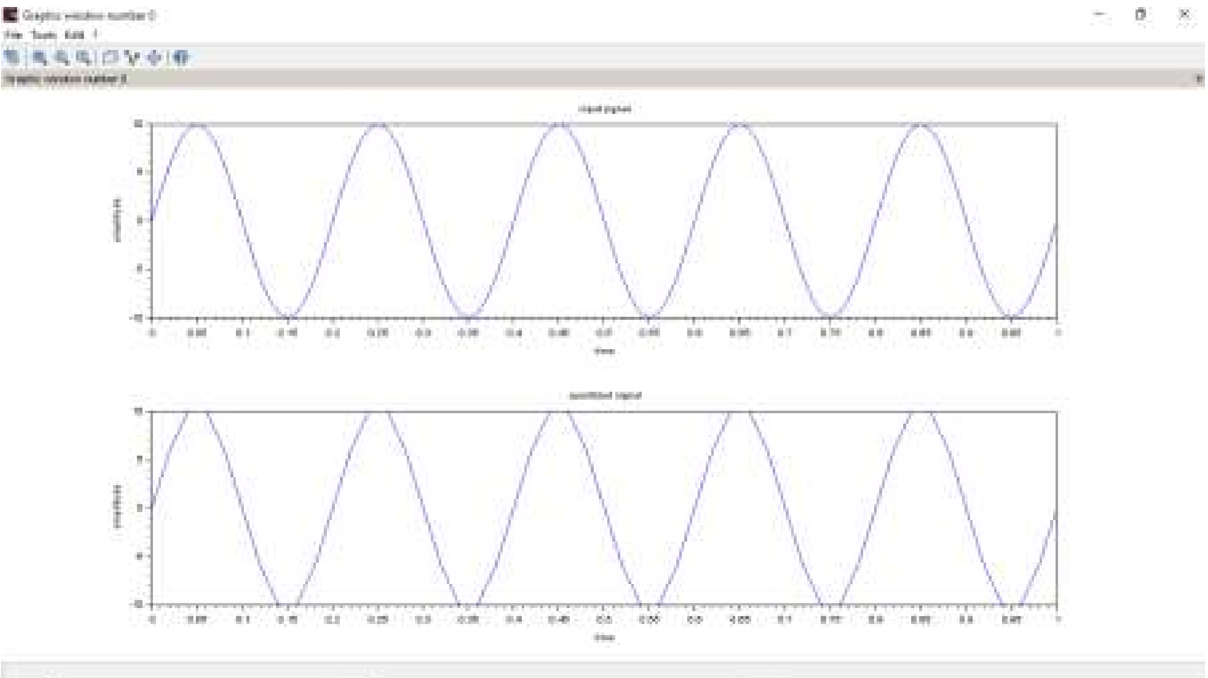
---

3) Name the circuit used to achieve synchronization b/w transmitter and receiver.

→ The key i/p device is used to send a request to the transmitter through control circuit viz. The transmitter then sends a signal, to be received by receiver 100 for synchronization.

POST LAB-04 (SCI LAB SIMULATION):-





7.9 Lab Result

Thus the Differential Pulse code modulation and demodulation were performed using the trainer kit and Scilab.

Laboratory Report Cover Sheet

SRM Institute of Science and Technology
Faculty of Engineering and Technology
Department of Electronics and Communication Engineering

18ECC205J ANALOG AND DIGITAL COMMUNICATION

Fifth Semester, 2020-21 (odd semester)

Name : **Pushpal Das**

Register No. : **RA1911004010565**

Day/ Session : **2ND / FN**

Venue : **Online (G Meet)**

Title of Experiment : **Delta Modulation and Demodulation**

Date of Conduction : **10 SEP 2021**

Date of Submission : **29 SEP 2021**

Particulars	Max. Marks	Marks Obtained
Pre-lab questions	05	
In-lab experiment	10	
Post-lab questions	15	
Total	**30**	

REPORT VERIFICATION

Date : **29 SEP 2021**

Staff Name : **Dr. P. Malarvezhi / Mrs.Harisudha**

Signature :

8 DELTA MODULATION AND DEMODULATION

8.1 Objective

To analyze a Delta modulation system and interpret the modulated and demodulated waveforms. Simulate delta modulation (DM) wave in SCI lab.

8.2 Hardware Required

PCM Modulator trainer- AET-73M
PCM Demodulator trainer-AET-73D
Storage Oscilloscope
Digital Multimeter
Co-axial cables (standard accessories with AET-73 trainer)

8.3 Introduction

Delta Modulation is a form of pulse modulation where a sample value is represented as a single bit. This is almost similar to differential PCM, as the transmitted bit is only one per sample just to indicate whether the present sample is larger or smaller than the previous one. The encoding, decoding and quantizing process become extremely simple but this system cannot handle rapidly varying samples. This increases the quantizing noise.

Block Diagram

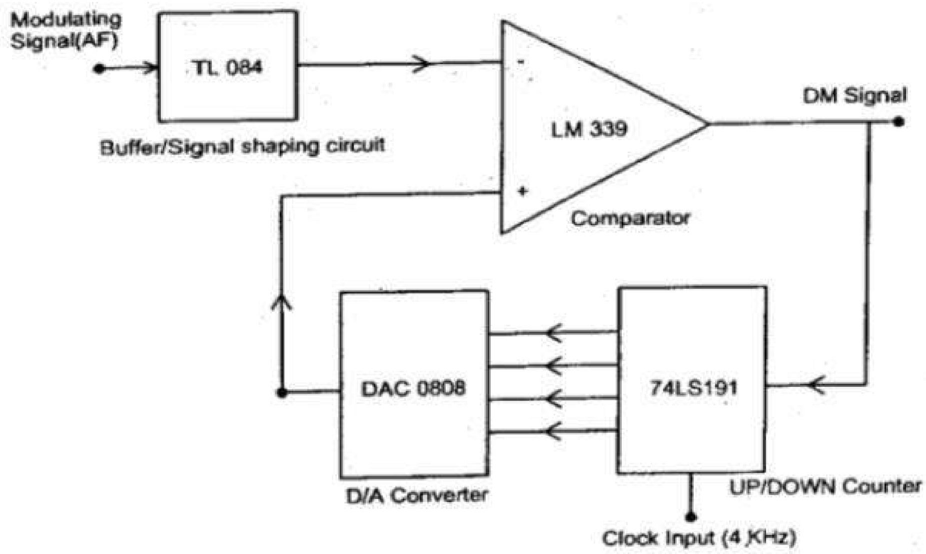


Figure 8.1 DM Modulator

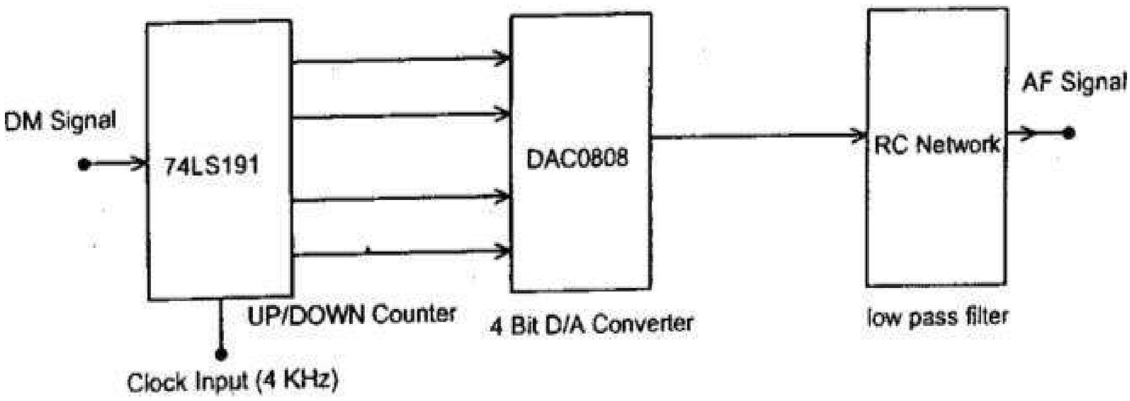


Figure 8.2. DM Demodulator

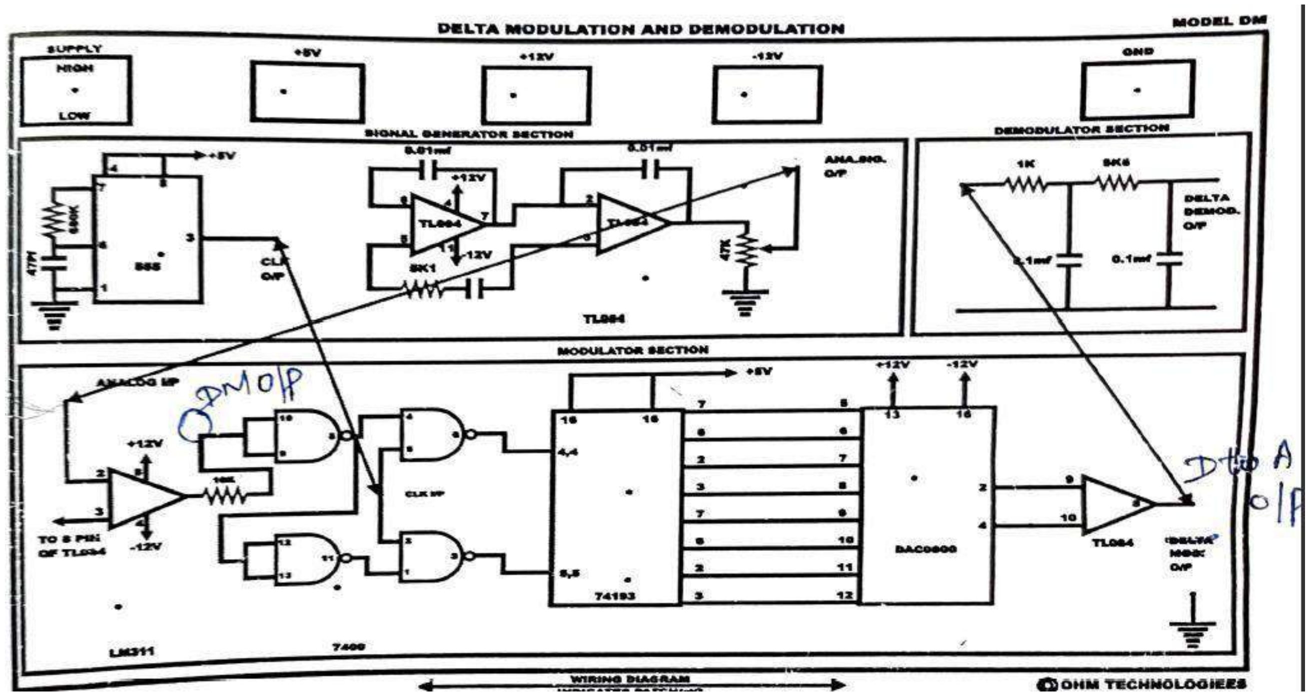


Figure 8.3. DM Wiring diagram

8.4 Procedure

8.5.1 DM Modulator

1. Study the theory of operation
2. Connect the trainer (AET-73M)–
3. ObservetheoutputofAFgeneratorusingCRO;itshouldbeaSinewaveof100Hz frequency with 3Vppamplitude.
4. Verify the output of the DC source with multimeter/scope; output should vary 0to +4V
5. Observe the output of the clock generator using Crotchety should be 4 KHz frequency of square wave with 5 Up amplitude.

Note: This clock signal is internally connected to the up/down counter so no external connection is required.

8.5.2 DM With DC Voltage as modulating signal:

1. ConnectDCsignalfromtheDCsourcetotheinvertinginputofthecomparatorandset some voltage says3V.
2. Observe and plot the signals at D/A converter output (i.e. non-inverting input of the comparator), DM signal using CRO and compare them with the waveforms given in figure.
3. Connect DM signal (from 73M) to the DM input of the demodulator.
4. Connect clock (4KHz) from modulator (73M) to the clock input of the demodulator (73D). Connect clock input of UP/DOWN counter (in 73D) to the clock from transmitter with the help of springs provided.
5. Observe digital output (LED indication) of the UP/DOWN counter (in 73 D) and compare it with the output of the UP/DOWN (in 73M) .By this you can notice that the both the outputs are same.

6. Observe and plot the output of the D/A converter and compare it with the waveforms given in figure.
7. Measure the demodulated signal (i.e. output of the D/A converter 73D with the help of multimeter and compare it with the original signal 73 M. From the above observation you can notice that both the voltages are equal and there is no loss in process of modulation, transmission and demodulation.
8. Similarly you can verify the DM operation for different values of modulating signal.

8.5.3 DM With AF Voltage as modulating signal:

1. ConnectAFsignalfromtheAFsourcetotheinvertinginputofthecomparatorandset some voltage says3V.
2. Observe and plot the signals at D/A converter output (i.e. non-inverting input of the comparator), DM signal using CRO and compare them with the waveforms given in figure.
3. Connect DM signal (from 73M) to the DM input of the demodulator.
4. Connect clock (4 KHz) from modulator (73M) to the clock input of the demodulator (73D).
5. ConnectclockinputofUP/DOWNcounter(in73D)totheclockfromtransmitterwith the help of springs provided.
6. Observe and plot the output of the D/A converter and compare it with the waveforms given in figure.
7. Observe and sketch the D/A output.
8. Connect D/A output to the LPF input.
9. Observe the output of the LPF/Amplifier and compare it with the original modulating signal(AET-73M).
10. From the above observation you can verify that there is no loss in information in conversion and transmission process.
11. Disconnect clock from transmitter (AET-73M) and connect to local oscillator (i.e. clock generator output from AET-73D) with remaining setup as it is. Observe demodulated signal output and compare it with the previous result. This signal islittle bit distorted in shape. This is because lack of synchronization between clock at transmitter and clock at receiver.

8.5 ModelGraph

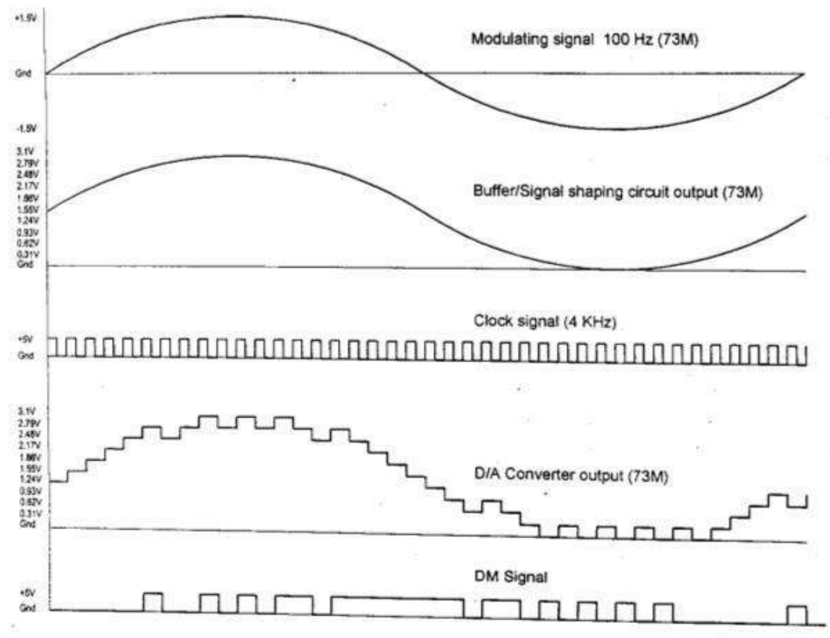


Fig 5.3 DM Waveform / Timing Diagram (Modulation) AF Input

Figure 8.4 DM Waveforms for AC input signal

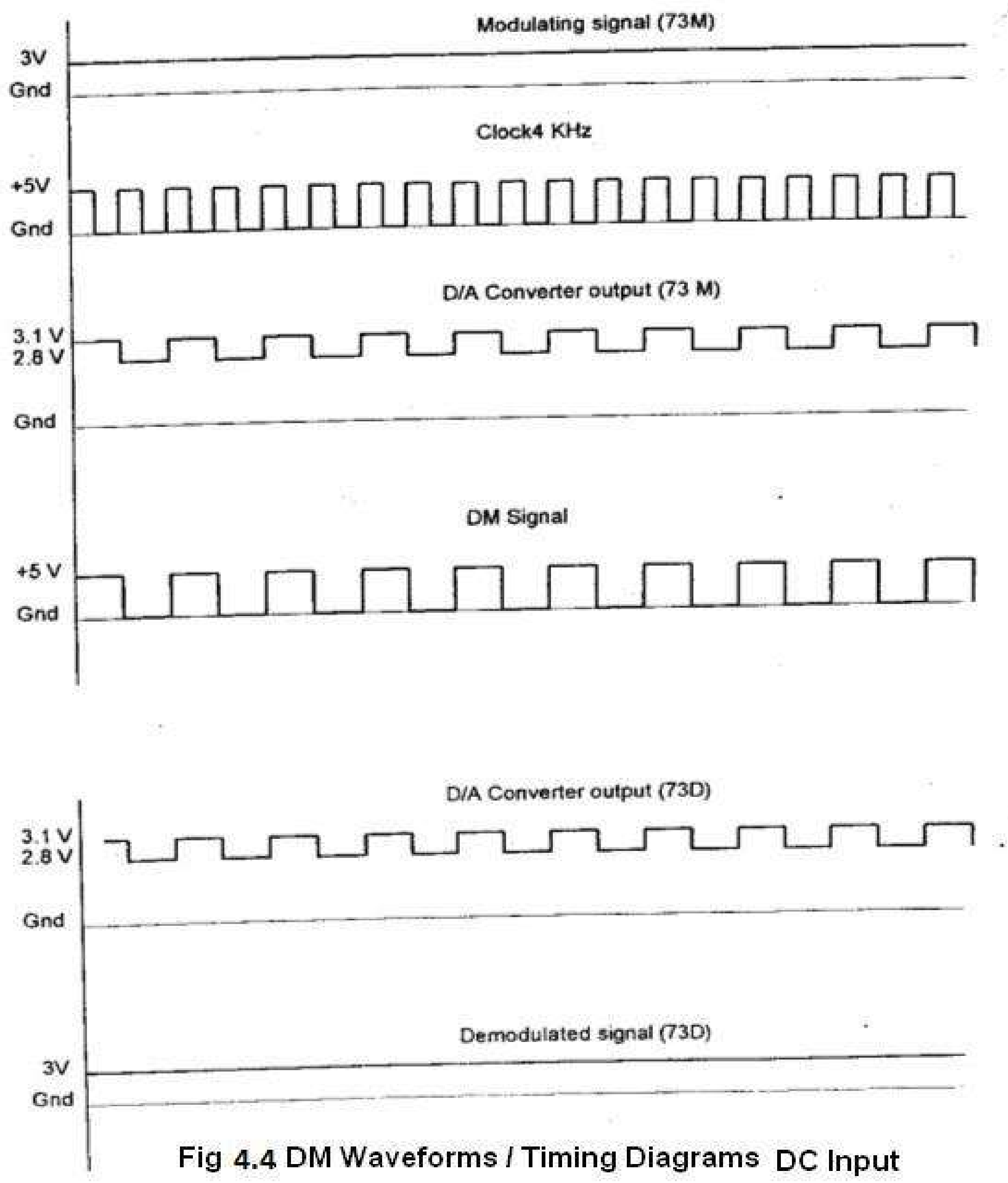


Figure 8.5 DM Waveforms/Timing Diagrams with DC input

Observation

DM Modulation (With AC input)

	Amplitude	Time Period
AC input		
D/A Converter Output		
Clock signal		
DM Output		

DM Demodulation (with AC input)

	Amplitude	Time Period
DM input		
D/A Converter output Signal		

Demodulated Output		
Clock signal		

DM Modulation (With DC input)

	Amplitude	Time Period
DC input		
D/A Converter Output		
Clock signa		
DMOutput		

DM Demodulation (With DC input)

	Amplitude	Time Period
DM input		
D/A Converter output Signal		
Demodulated Output		
Clock signal		

8.6 Virtual lab

Software: SCI lab6.0.2

Problem statement

Perform delta modulation using SCI lab for the following specifications
Amplitude of the signal: 12V
Step size=0.5 V

8.7.1 SCILAB code

```
A=12;
del=1;
pi=3.14;
t=0:2*pi/100:2*pi;
x=A*sin(t);
plot(x)
y=[0];
xr=0;
for i=1:length(x)-1
   if xr(i)<=x(i)
      d=1;
      xr(i+1)=xr(i)+del;
   else
      d=0;
      xr(i+1)=xr(i)-del;
   end
   y=[y d];

end
plot2d2(xr);
xlabel('Time');
ylabel('Amplitude');
title('DM modulation');
```

Observations

Signal	Amplitude	Time period
Input signal		
Modulated signal		

Model graph

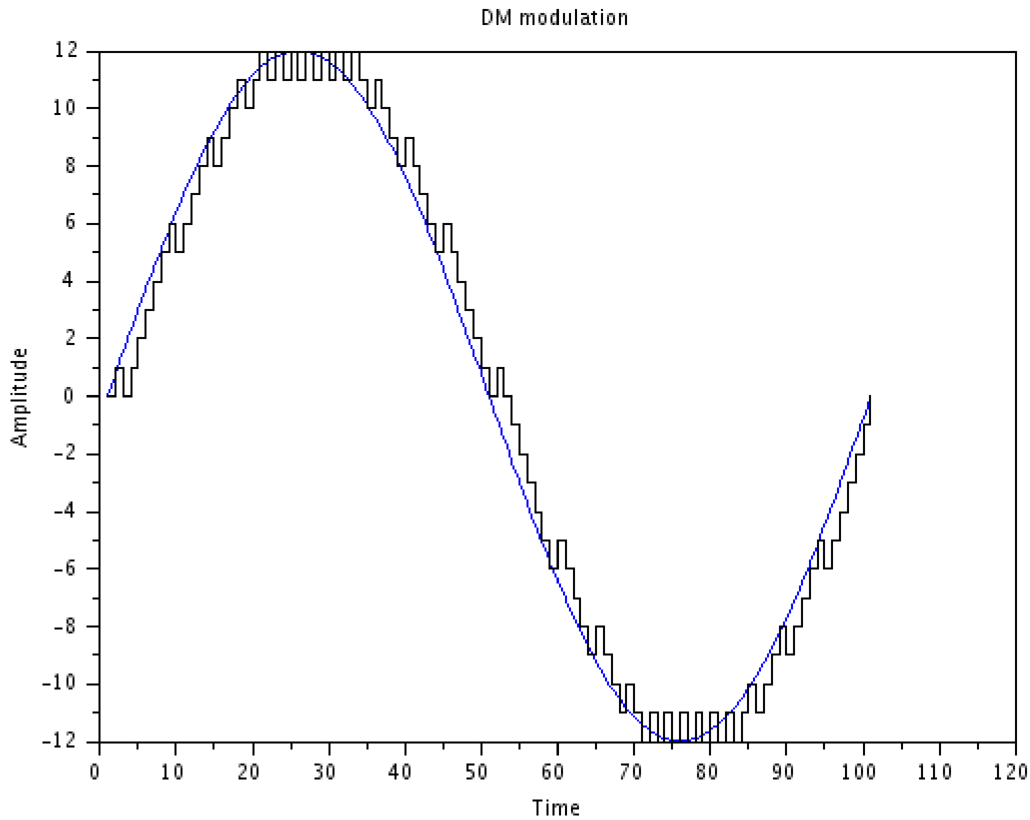
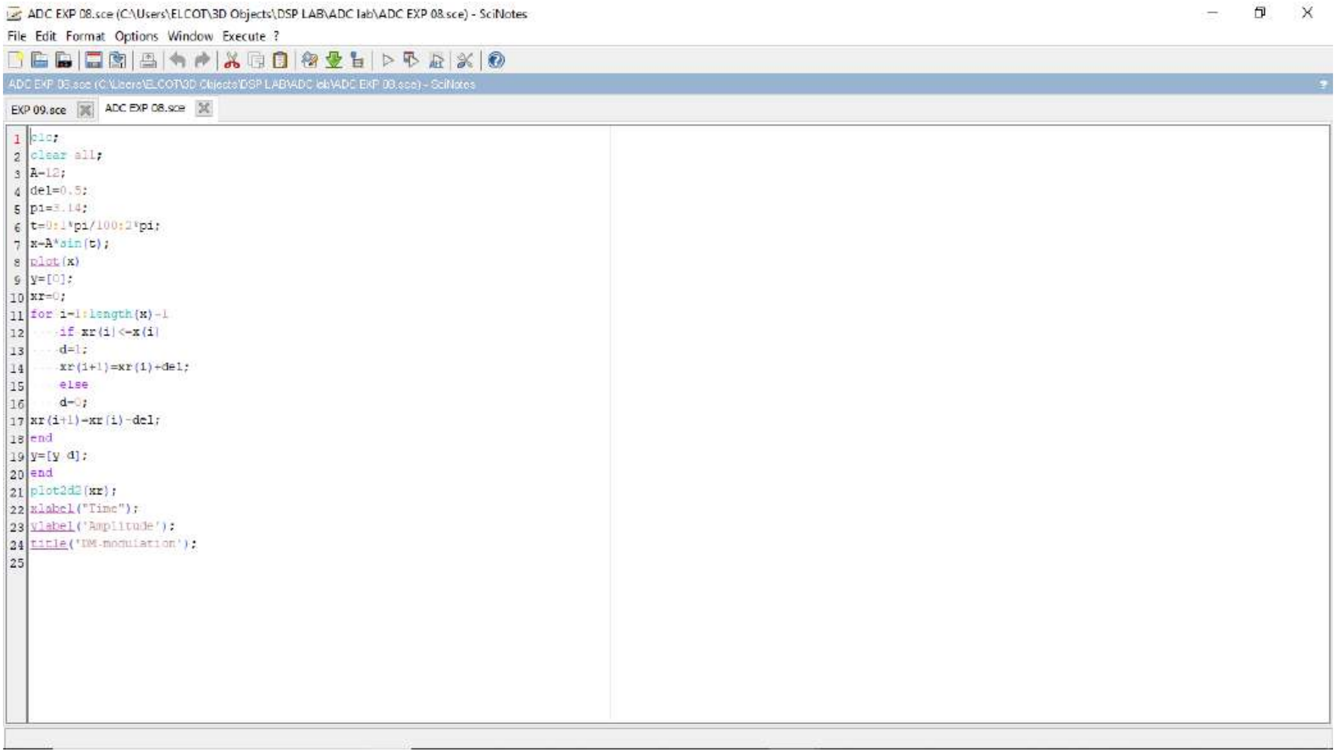
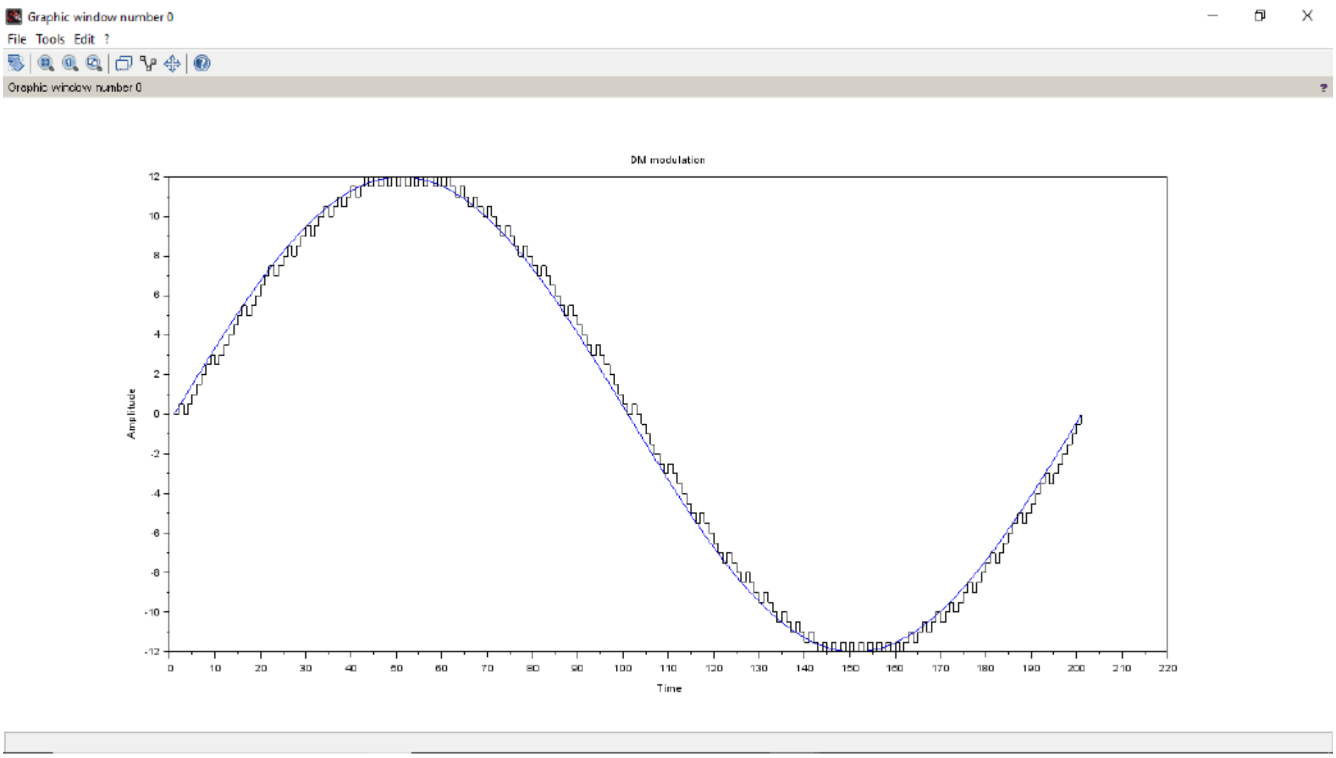


Figure 8.6 Input and DM modulated signal

SCI LAB SIMULATION:-





Observations

Signal	Amplitude	Time period
Input signal	10	55ms
Modulated signal	11	55ms

Prelab Questions

1. What is granular noise?
2. What is slope over distortion?
3. What happens to the output signal if the variation of the message signals
   (i) Greater than the step size (ii) less than the step size

Exp. 08

RA1911004010565

Pushpal Das,

Pre-lab

1) What is granular noise?

=> Granular noise is the manifestation of random signals when the variation of the input signal is smaller than the step size.

2) What is slope over distortion?

=> The distortion that arises because of larger dynamic range of the input signal is known as slope overload distortion.

3) What happens to the output signal if the variation of the message signal is

(i) Greater than the step size.

=> If message signal is greater than step signal size then the granular noise occurs

(ii) less than the step size.

=> If message signal is less than the step size than slope overload distortion occurs

Post Lab Questions:-

1. Compare DPCM & Delta modulation.
2. How to reduce the quantization noise that occurs in DM?
3. A band pass signal has a spectral range that extends from 20 to 82 KHz. Find the acceptable sampling frequency.

Post-lab

1) Compare DPCM & Delta modulation.

Delta modulation	DPCM
Only one bit is used per sample.	More than one bit are used.
It has poor SNR	It has fair SNR.
The step size is fixed	The number of levels are fixed.
It is less efficient	It is more efficient

2) How to reduce the quantization noise that occurs in DM?

⇒ The process of over sampling to reduce the ADC quantization noise is straight forward. An analog signal is digitised at an sample rate. That is higher than the minimum work rate needed to satisfy the nyquist and then the low pass filtered.

3) A bandpass signal has a special angle range that extends from 20 to 82 KHz. find the acceptable sampling frequency

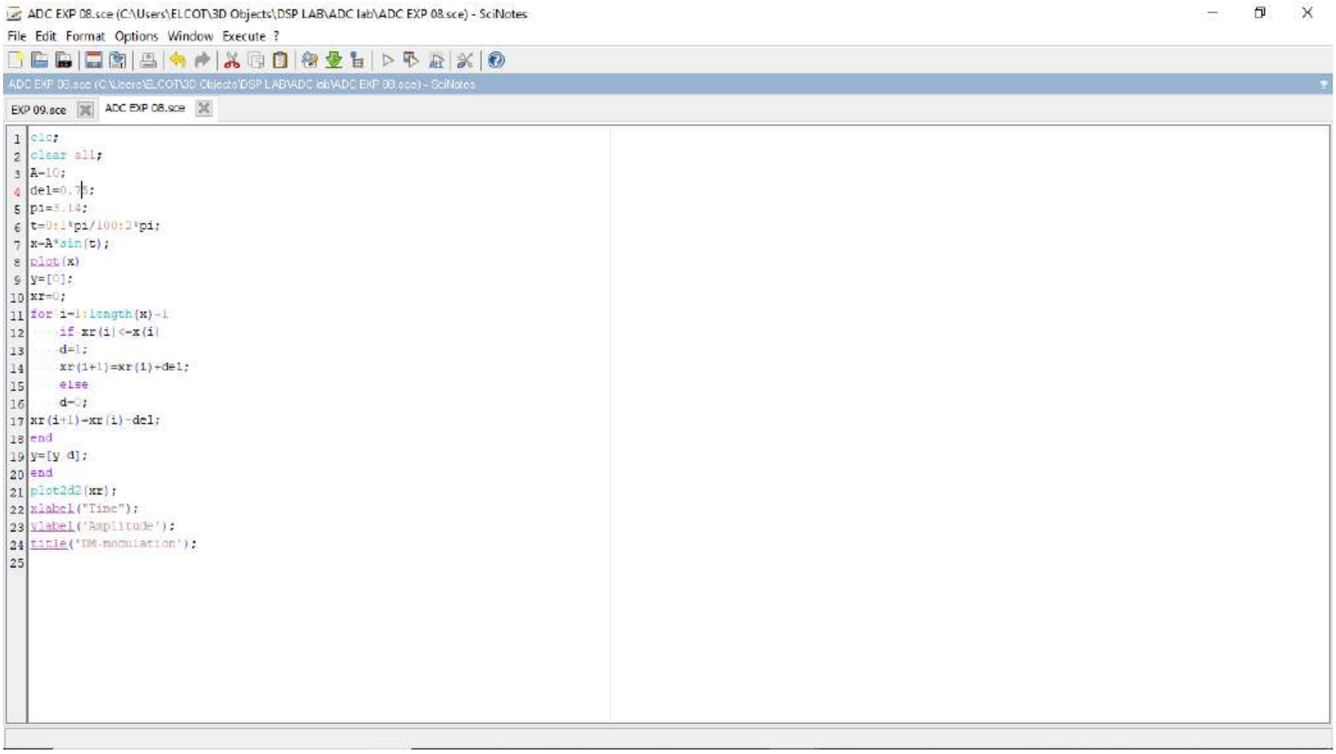
Soln// Bandwidth = 2fm

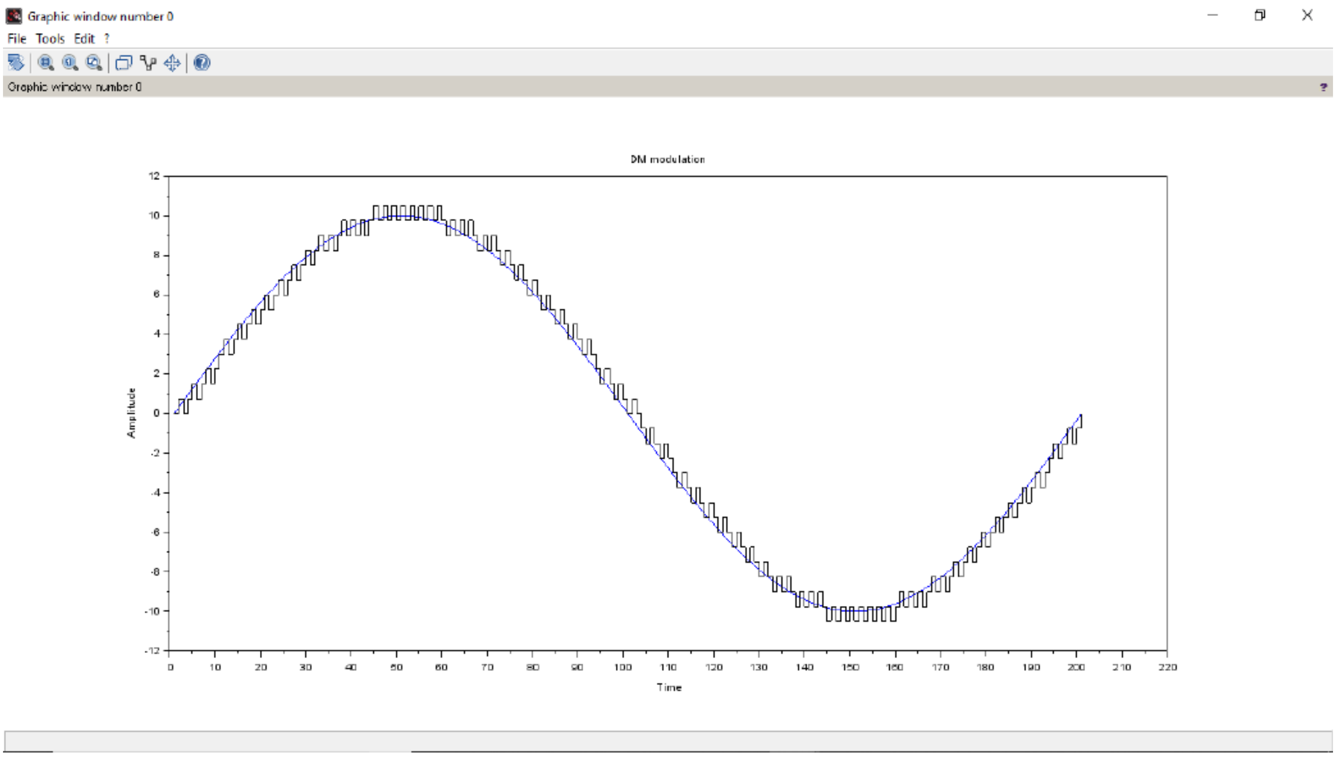
= 82 KHz - 20 KHz.

= 62 KHz.

Minimum sampling rate = 2 × BW = 2 × 62 = 124 KHz

4. Use SCILAB to perform delta modulation for
(i) Input signal of amplitude10V
(ii) Step size of0.75V





Result

Thus, the Delta modulation and demodulation were performed using the trainer kit and DM modulation is performed using SCILAB.

Laboratory Report Cover Sheet

SRM Institute of Science and Technology
Faculty of Engineering and Technology
Department of Electronics and Communication Engineering

18ECC205J ANALOG AND DIGITAL COMMUNICATION

Fifth Semester, 2020-21 (odd semester)

Name : **Pushpal Das**

Register No. : **RA1911004010565**

Day/ Session : **2ND / FN**

Venue : **Online (G Meet)**

Title of Experiment : **9.PSK MODULATION AND DEMODULATION**

Date of Conduction : **18 SEP 2021**

Date of Submission : **29 SEP 2021**

Particulars	**Max. Marks**	**Marks Obtained**
Pre-lab questions	05	
In-lab experiment	10	
Post-lab questions	15	
Total	**30**	

REPORT VERIFICATION

Date : **29 SEP 2021**

Staff Name : **Dr. P. Malarvezhi / Mrs.Harisudha**

Signature :

9. PSK MODULATION AND DEMODULATION

9.1 Objective

To analyze a PSK modulation system and interpret the modulated and demodulated waveforms. Simulate PSK modulated waveform and also find the Probability of Error using SCILAB.

9.2 Hardware Required

PSK Trainer Kit

Dual Trace oscilloscope-POS-2020

Software Required

Scilab 6.1.0

9.3 Introduction

Phase shift keying is a modulation/data transmitting technique in which phase of the carrier signal is shifted between two distinct levels. In a simple PSK(ie., Binary PSK) un-shifted carrier Acosωt is transmitted to indicate a 1 condition, and the carrier shifted by 180^0ie., – Acosωt is transmitted to indicate as 0 condition.

S(t)= Acos 2πfct for Binary 1

Acos (2πfct +π) for Binary 0

WIRING /TRAINER KITDIAGRAM

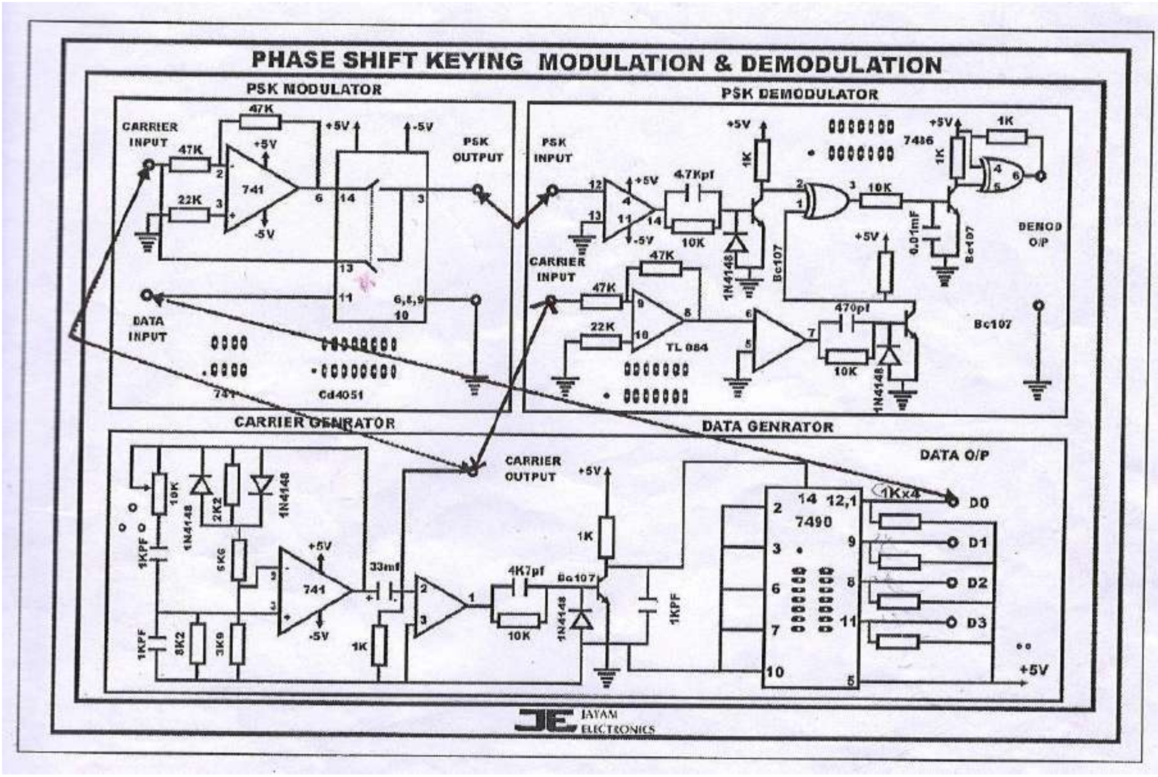


Fig 9.1 Wiring Diagram for PSK Modulation and Demodulation

9.4 Pre Lab Questions

1. What is BPSK?
2. Draw the truth table,Phasor diagram,Constellation diagram of BPSK .
3. What are Antipodal signals?
4. Write the advantages of BPSK.
5. Compare binary PSK with QPSK.

6. 9.5 Lab Procedure

1. Connect the trainer to mains and switch on the power supply.
2. Observe the output of the carrier generator using CRO,it should be an 17KHZ sine with 5Vpp amplitude.
3. Observe the various data signals(4KHZ and 8KHZ) using CRO

Modulation

4. Connect carrier signal to carrier input of the PSKmodulator.
5. Connect data signal say 4KHZ from data source to data input of the modulator.

6. Keep CROin dual mode and connect CH1 input of the CRO to data signal and CH2 to the output of the PSK modulator.
7. Observe the PSKoutput signal with respect to data signal and plot the waveforms.

Demodulation

8. Connect the PSK output to the PSK input of the demodulator.
9. Connect carrier to the carrier input of the PSKdemodulator.
10. Keep CROin dual mode and connect CH1 to data signal (at modulator) and CH2 to the output of the demodulator.
11. Compare the demodulated signal with the original signal. By this we can notice that there is no loss in modulation and demodulation process
12. Repeat the steps 5 to 11 with different data signal ie., 8KHZ

9.6 Model Graph

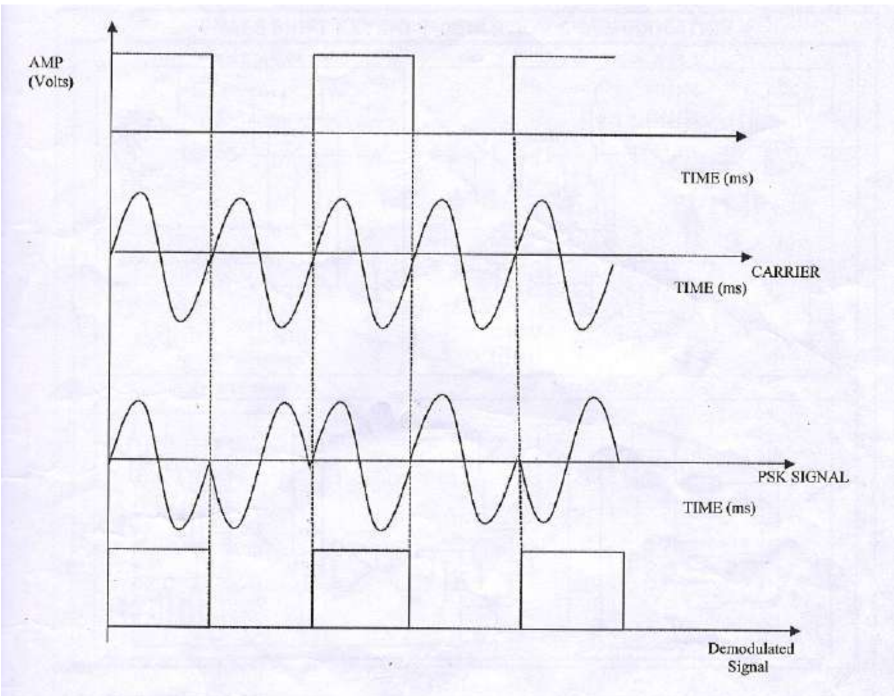


Figure 9.2 PSK Waveforms for different data input signals

9.7 Observation-Hardware

PSK Modulation

Signal Name	Amplitude	Time Period
Carrier signal		
Data source For 4KHz For 8KHz		
Modulated output For 4KHz For 8KHz		

PSK Demodulation

Signal Name	Amplitude	Time Period
Demodulated output For 4KHz For 8KHz		

9.8 VIRTUAL LAB

9.8.1 Software Required

SCILAB software 6.1.0

9.8.2Exercise

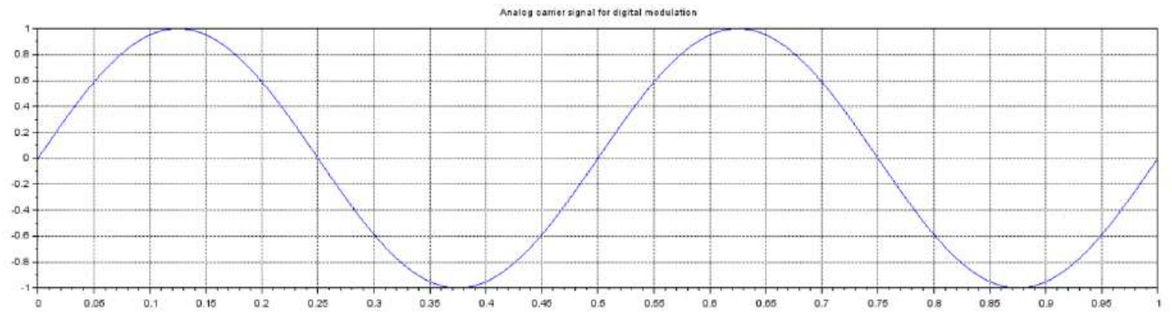
Use SCILAB to produce PSK waveform with the following specifications.
Analog carrier frequency = 2KHz, Input Binary data= [1, 0,1,0]

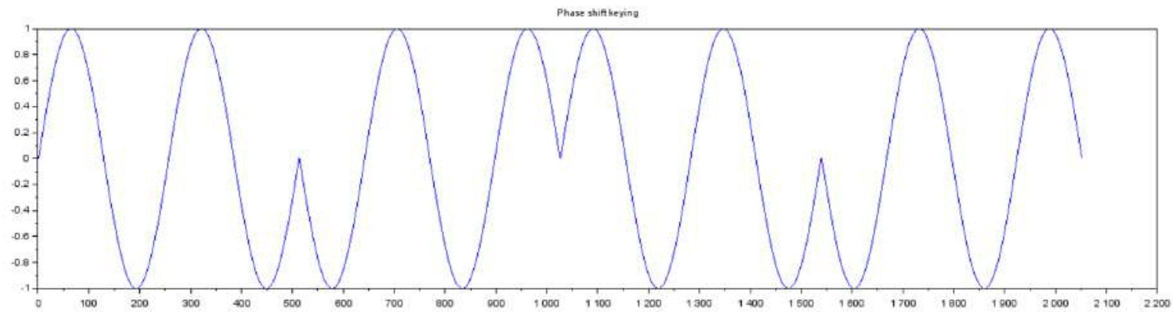
Solution

```
clear;
clc;
close;
f=input('Enter the analog carrier frequency in KHz' );
t=0:1/512:1;
x=sin(2*%pi*f*t);
i=input('Enter the input binary data');
disp(' Binary information at Transmitter :');
disp(i);
xpsk=[];
```

```
x1=sin(2*%pi*f*t);
x2=-sin(2*%pi*f*t);
for n=1:length(i)
   if(i(n)==1)
   xpsk=[xpsk,x1];
   elseif(i(n)==0)
   xpsk=[xpsk,x2];
end
end
subplot(2,1,1)
plot(t,x)
xtitle('Analog carrier signal for digital modulation');
   xgrid
   subplot(2,1,2);
   plot(xpsk)
   xtitle('Phase shift keying')
```

Model Graph





In console window enter the analog carrier frequency and input binary data .

```
Enter the analog carrier frequency in KHz 2

Enter the input binary data [1,0,1,0]


 " Binary information at Trans mitter :"

  1.   0.   1.   0.
```

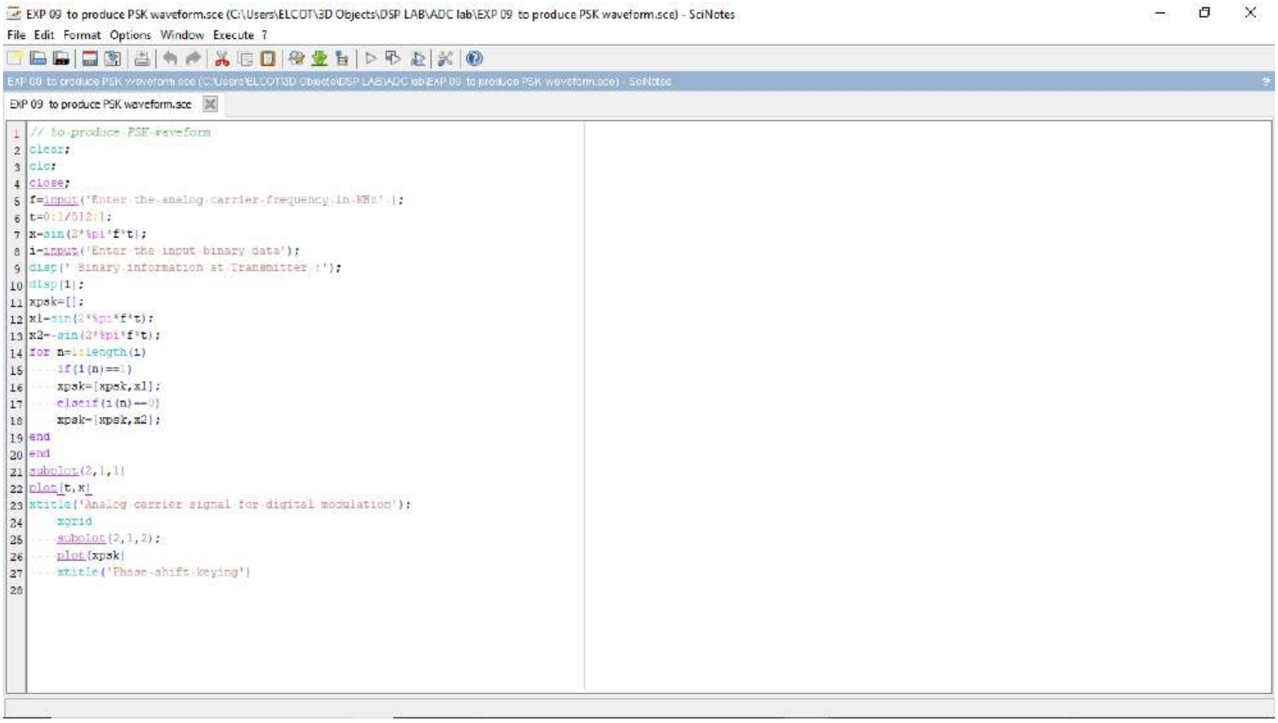
9.8.3 Exercise: The binary receiver system receives a bit rate of 1Mbps.The waveform amplitude is 5mV and the noise power spectral density (N0/2) is $0.5x10^{-11}$ W/Hz. By using Scilab to find bit error Probability for PSK.

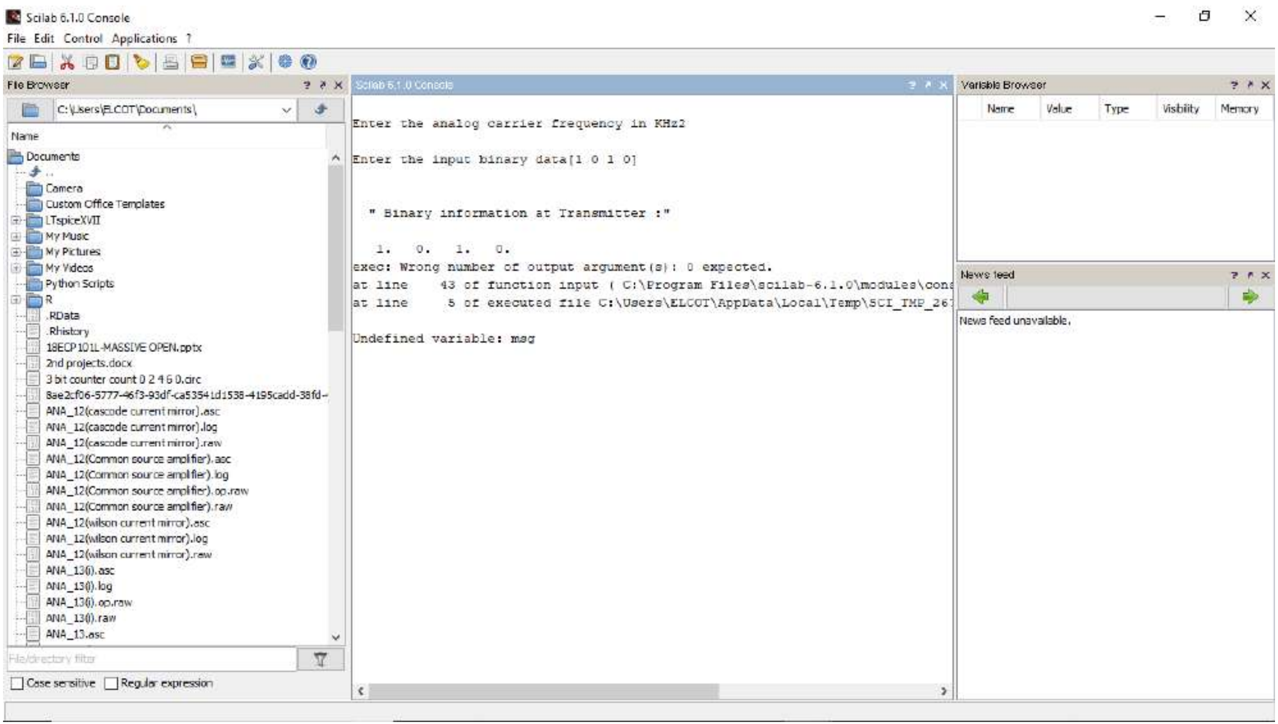
```
clc;
clear;
A=5*10^-3;
Tb=1*10^-6;
fb=1/Tb;
N0=1*10^-11; // Noise power spectral density
Eb=(A^2*Tb)/2;//Eb=bit energy
z=sqrt(Eb/N0);
Pe=1/2*erfc(z) //bit error probability
disp("Bit Error probability of PSK is  P(e)")
disp(Pe);
```

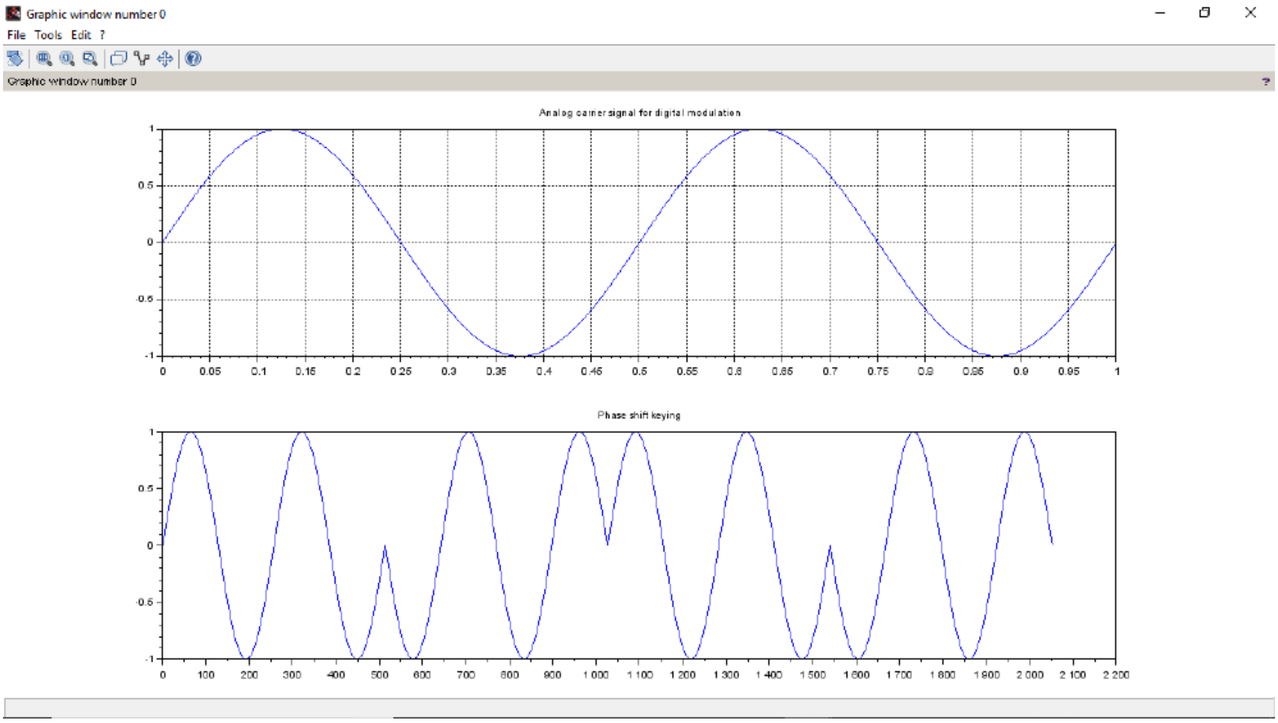
Output:

```
"Bit Error probability of PSK is P(e)"
 0.0569231
```

SCI LAB CODE & SIMULATION:-


EXP 09 to produce PSK waveform.sce (C:\Users\ELCOT\3D Objects\DSP LAB\ADC lab\EXP 09 to produce PSK waveform.sce) - SciNotes
File Edit Format Options Window Execute ?
EXP 09 to produce PSK waveform.sce
// to produce PSK waveform
clear;
clc;
close;
f=input('Enter the analog carrier frequency in KHz');
t=0:1/512:1;
x=sin(2*%pi*f*t);
i=input('Enter the input binary data');
disp(' Binary information at Transmitter :');
disp(i);
xpsk=[];
x1=sin(2*%pi*f*t);
x2=-sin(2*%pi*f*t);
for n=1:length(i)
if(i(n)==1)
xpsk=[xpsk,x1];
elseif(i(n)==0)
xpsk=[xpsk,x2];
end
end
subplot(2,1,1)
plot(t,x)
xtitle('Analog carrier signal for digital modulation');
xgrid
subplot(2,1,2);
plot(xpsk)
xtitle('Phase shift keying')


Scilab 6.1.0 Console
File Edit Control Applications ?
File Browser
C:\Users\ELCOT\Documents\
Scilab 6.1.0 Console
Enter the analog carrier frequency in KHz2
Enter the input binary data[1 0 1 0]
" Binary information at Transmitter :"
1. 0. 1. 0.
exec: Wrong number of output argument(s): 0 expected.
at line 43 of function input (C:\Program Files\scilab-6.1.0\modules\cons
at line 5 of executed file C:\Users\ELCOT\AppData\Local\Temp\SCI_TMP_26
Undefined variable: msg
Variable Browser
Name Value Type Visibility Memory
News feed
News feed unavailable.
Case sensitive Regular expression



9.9 Post Lab Questions

1. What is bit error rate and give the expression for bit error Probability of BPSK.

2. Show the PSK modulated waveform for the message 10010110.
3. Use SCILAB to produce PSK waveform with the following specifications.

   Analog carrier frequency = 4 KHz, Input Binary data = [1, 1, 1, 0, 1]

Pre Lab Ans:-

RA1911004010565

Pushpal Das

~~ABE~~

Exp - 09.

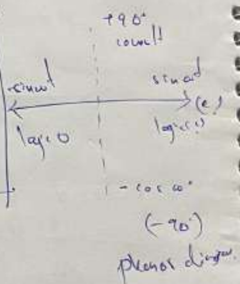
1) What is BPSK?

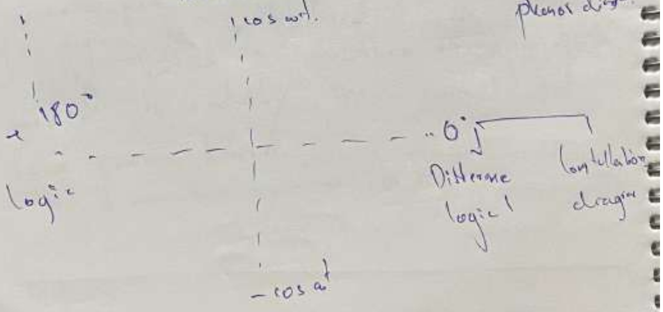
=> Binary phase shift keying is a two phase modulation scheme, where the 0's and 1's in a binary msg are represented by two different states in carrier signal.

2) Draw the truth table, phasor diagram, constillation diagram of BPSK

Binary I/P	O/P phase.
logic 0	180°
logic 1	0°

Truth table.





3) What are antipodal signal?

→ Antipodal signal is total signal 180° opposite to each other. One signal have value on -1 and other on 1 (mathematical form)↓.

$-T_c \quad \sqrt{T_c} \quad \sqrt{T_c} \quad +T_c$

$\longleftarrow\!\!-\!\!-\!\!-\!\!-\!\!-\!\!-\!\!-\!\!\longrightarrow$

logic 0 Binary logic 1

Antipodal signal

---

4) Write the advantages of BPSK.

- It is most robust modulation technique due to the fact that binary 180° are seperated by 180 degree phase shift of the carrier.
- Due to this property, BPSK modulated data can travel longer distance when transmitted from base station.

---

5) Compare binary PSK with QPSK.

→ QPSK has advantages of having double data rate compare to BPSK. This is due to support of two bits per carrier in QPSK compare to one bit per carrier in the case of BPSK.

Post lab Ans:-

Post lab

1) What is bit error rate. and give the expression for bit error probability of BPSK.

=> The ratio of number of error bits and totals number of bits transmitted during a specific period.

probability density function = $p(z)$

$$\Rightarrow \frac{z}{\sigma^2} e^{\frac{z^2}{2e^2\sigma^2}}, \quad z \geq 0$$

$$\sigma^2 = 6(z^2) = P_n = \frac{1}{\sqrt{2\sigma^2}} e^{\frac{(n-x)^2}{2\sigma^2}}$$

=> bit rate error probability for given value. efficient bit energy to noise ratio.

$$= \frac{|z|^2 E_b}{N_a}$$

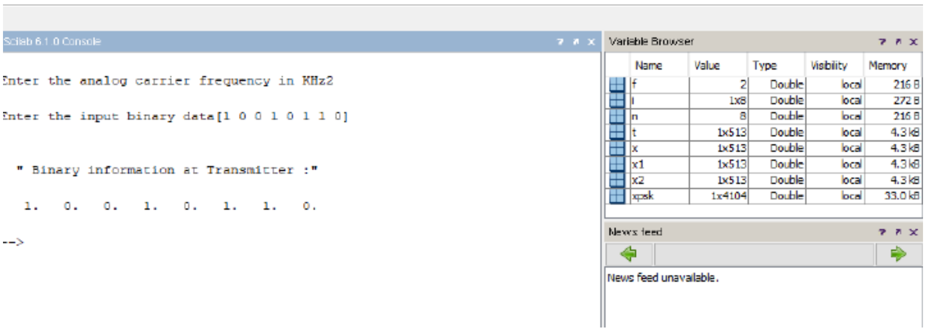
$$P_{b12} = \frac{1}{2} erfc\left(\sqrt{\frac{|z|^2 E_b}{N_a}}\right)$$

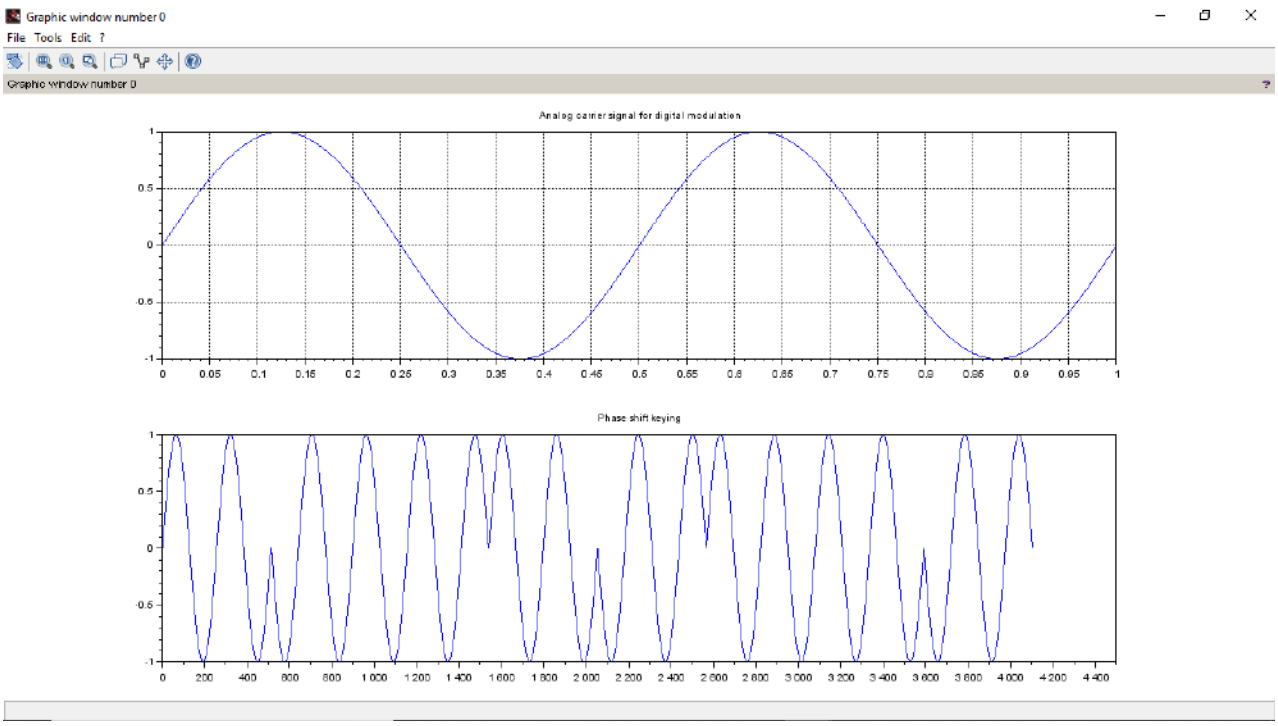
Hence probability error is

$$P_s = \frac{1}{2}\left(1 - \frac{l_b(N_a)}{E_0 |N_a|^4}\right)$$

Post Lab 02:-

2.Show the PSK modulated waveform for the message 10010110.

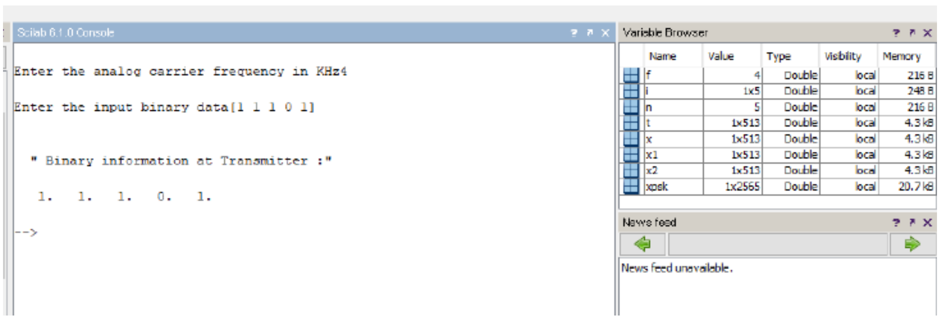


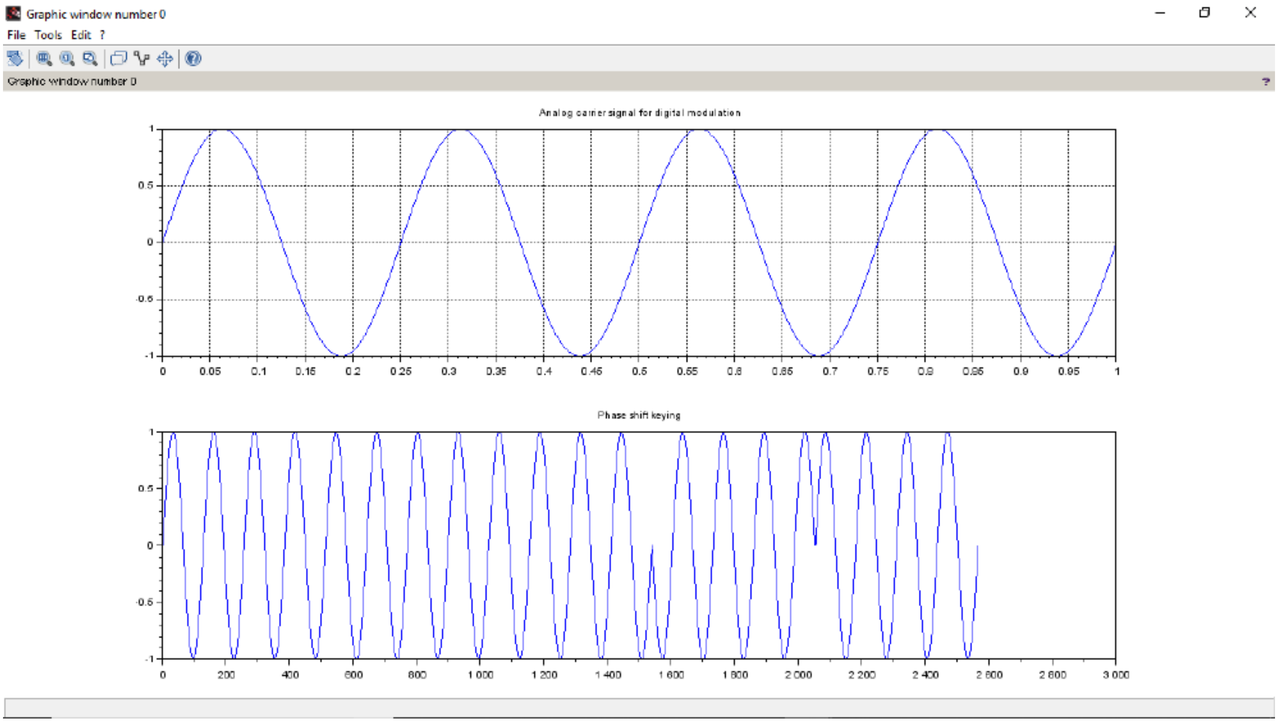


Post Que.03:-

Use SCILAB to produce PSK waveform with the following specifications.

Analog carrier frequency = 4 KHz, Input Binary data = [1, 1, 1, 0, 1]





Lab Result

Thus the PSK modulation and demodulation were performed and simulated to find Probability of error by using SCILAB.

Laboratory Report Cover Sheet

SRM Institute of Science and Technology
College of Engineering and Technology
Department of Electronics and Communication Engineering

18ECC205J ANALOG AND DIGITAL COMMUNICATION
Fifth Semester, 2021-22 (ODD semester)

Name : **Pushpal Das**

Register No. : **RA1911004010565**

Day / Session : **2^{ND} / FN**

Venue : **Online (G meet)**

Title of Experiment : **Simulation of Quadrature Phase Shift Keying (QPSK) . Modulation and Demodulation**

Date of Conduction : **23 SEP 2021**

Date of Submission : **29 SEP 2021**

Particulars	Max. Marks	Marks Obtained
Pre-lab questions	05	
In-lab experiment	10	
Post-lab questions	10	
Record Submission	05	
Total	**30**	

REPORT VERIFICATION

Date : **29 SEP 2021**

Staff Name : **Dr. P. Malarvezhi / Mrs.Harisudha**

Signature :

Experiment No.10 Simulation of Quadrature Phase Shift Keying (QPSK) Modulation and Demodulation

Objective: To generate and demodulate quadrature phase shifted (QPSK) signal using SCI lab.

Algorithm:

QPSK modulation

1. Generate quadrature carriers.
2. Start FOR loop
3. Generate binary data, message signal (bipolar form)
4. Multiply carrier 1 with odd bits of message signal and carrier 2 with even bits of message signal
5. Perform addition of odd and even modulated signals to get the QPSK modulated signal
6. Plot QPSK modulated signal.
7. End FOR loop.
8. Plot the binary data and carriers.

QPSK demodulation

1. Start FOR loop
2. Perform correlation of QPSK modulated signal with quadrature carriers to get two decision variables x1 and x2.
3. Make decisions on x1 and x2 and multiplex to get demodulated binary data.

   If x1>0and x2>0, choose ‘11’. If x1>0and x2<0, choose ‘10’. If x1<0and x2>0, choose ‘01. If x1<0and x2<0, choose ‘00’.
4. End FOR loop
5. Plot demodulated data

Program

```
clc ;
clear ;
xdel ( winsid () )
```

```
Tb=1;
t=0:(Tb/100):Tb;
 fc=1;
 c1=sqrt(2/Tb)*cos(2*%pi*fc*t);
 c2=sqrt(2/Tb)*sin(2*%pi*fc*t);
 N=16;
 m=rand(1,N);
 t1=0;t2=Tb;
 for
 i=1:2:(N-1
 )
    t=[t1:(Tb/100):t2]

 if
    m(i)>0.
    5
    m(i)=1;
 m_s=ones(1,length(t));
 else
 m(i)=0;

 m_s=-1*ones(1,length(t))
 ; end
 odd_sig(i,:)=c1.*m_s;
 if m(i+1)>0.5
 m(i+1)=1;
 m_s=ones(1,length(t));
 else
 m(i+1)=0;

 m_s=-1*ones(1,length(t))
 ; end
 even_sig(i,:)=c2.*m_s;
 qpsk =
 odd_sig+even_sig;
```

```
subplot(3,2,4);
plot(t,qpsk(i,:));
title('QPSK signal');
xlabel('t     >');
```

```
ylabel('s(t)');


t1=t1+(Tb+.01)
;
t2=t2+(Tb+.01)
;
```

```
end
subplot(3,2,1)
; for N=1:16;
   plot2d3(N,m(N),style=2)
end
title('binary
data bits');
xlabel('n
>');

ylabel('b(n)');

subplot(3,2,2);
plot(t,c1);
title('carrier signal-1');
xlabel('Time (sec)');
ylabel('Amplitude
(volts)'); subplot(3,2,3);
plot(t,c2);

title('carrier signal-2');
xlabel('Time (sec)');
ylabel('Amplitude
(volts)'); t1=0;
t2=Tb;

 for i=1:N-1
   t=[t1:(Tb/100):t2]
   x1=sum(c1.*qpsk(i,:));


   x2=sum(c2.*qpsk(i,:))
   ; if (x1>0&&x2>0)
      demod(i)=1;
      demod(i+1)=1
      ;

elseif
   (x1>0&&x2<0)
```

```
    demod(i)=1;
    demod(i+1)=0;
elseif
```

```
    (x1<0&&x2<0)
    demod(i)=0;
    demod(i+1)=0;
elseif

    (x1<0&&x2>0
    ) demod(i)=0;
```

```
    demod(i+1)=1;
    end
t1=t1+(Tb+.01)

;
t2=t2+(Tb+.01)
;

end
subplot(3,2,5)
; for N=1:16;
    plot2d3(N,demod(N),style=2)
end
title('qpsk
demodulated bits');
xlabel('n                    >');
ylabel('b(n)');
```

Simulated Results:

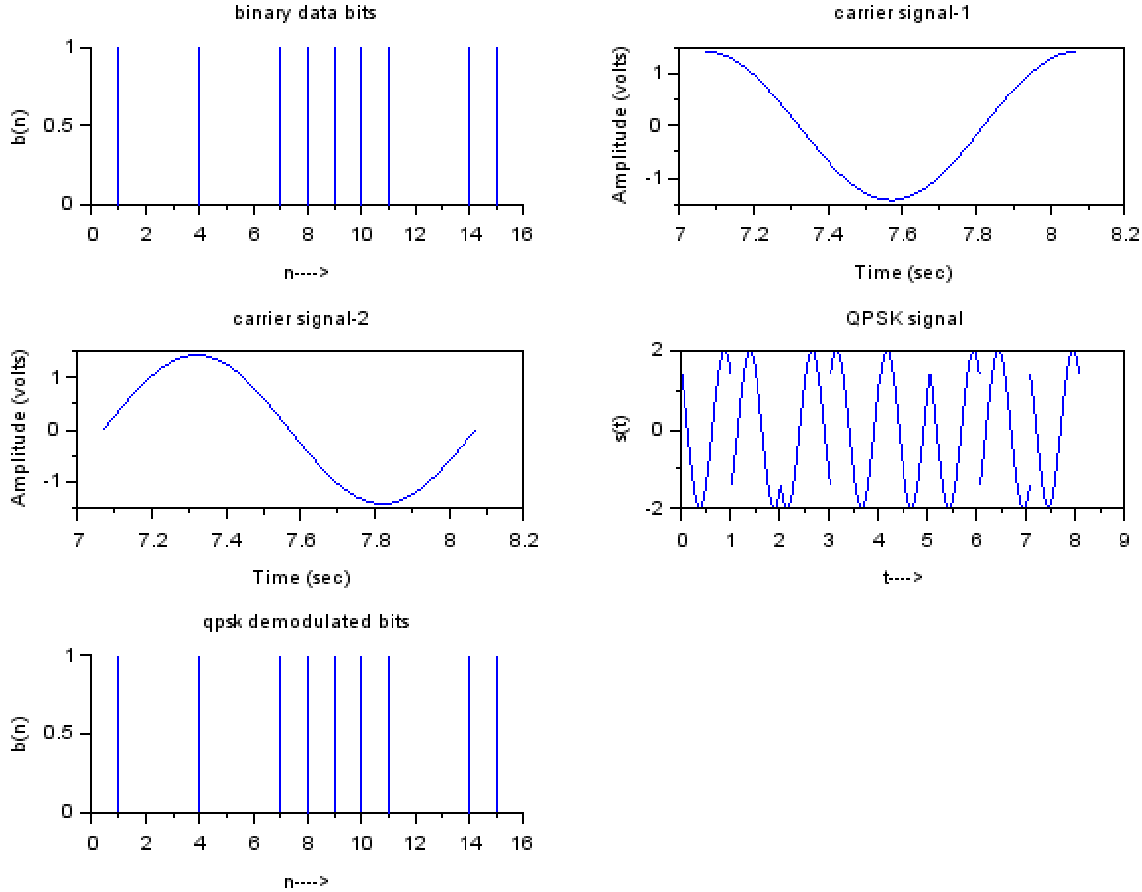
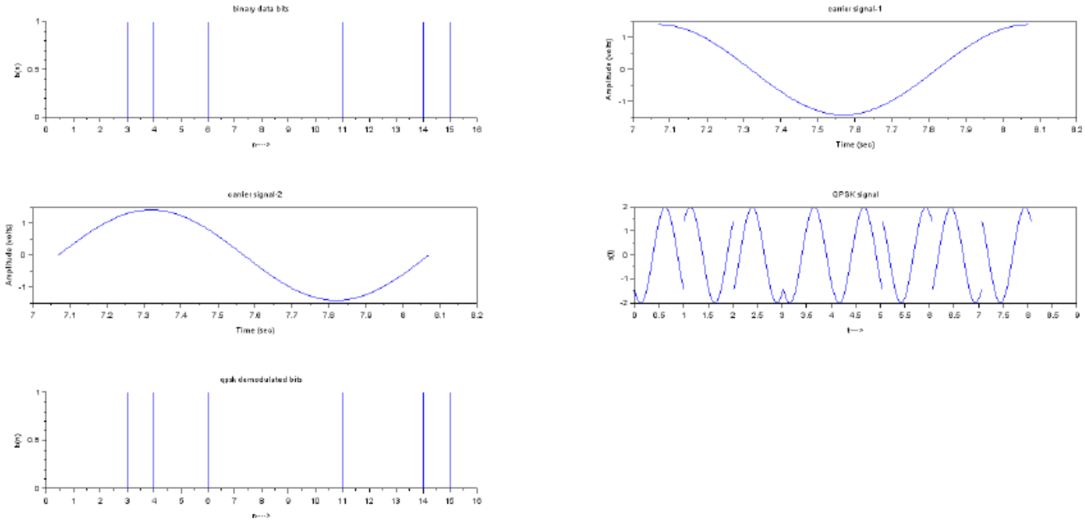


Fig. 7.2 Waveform of QPSK Modulation and Demodulation in SCI lab

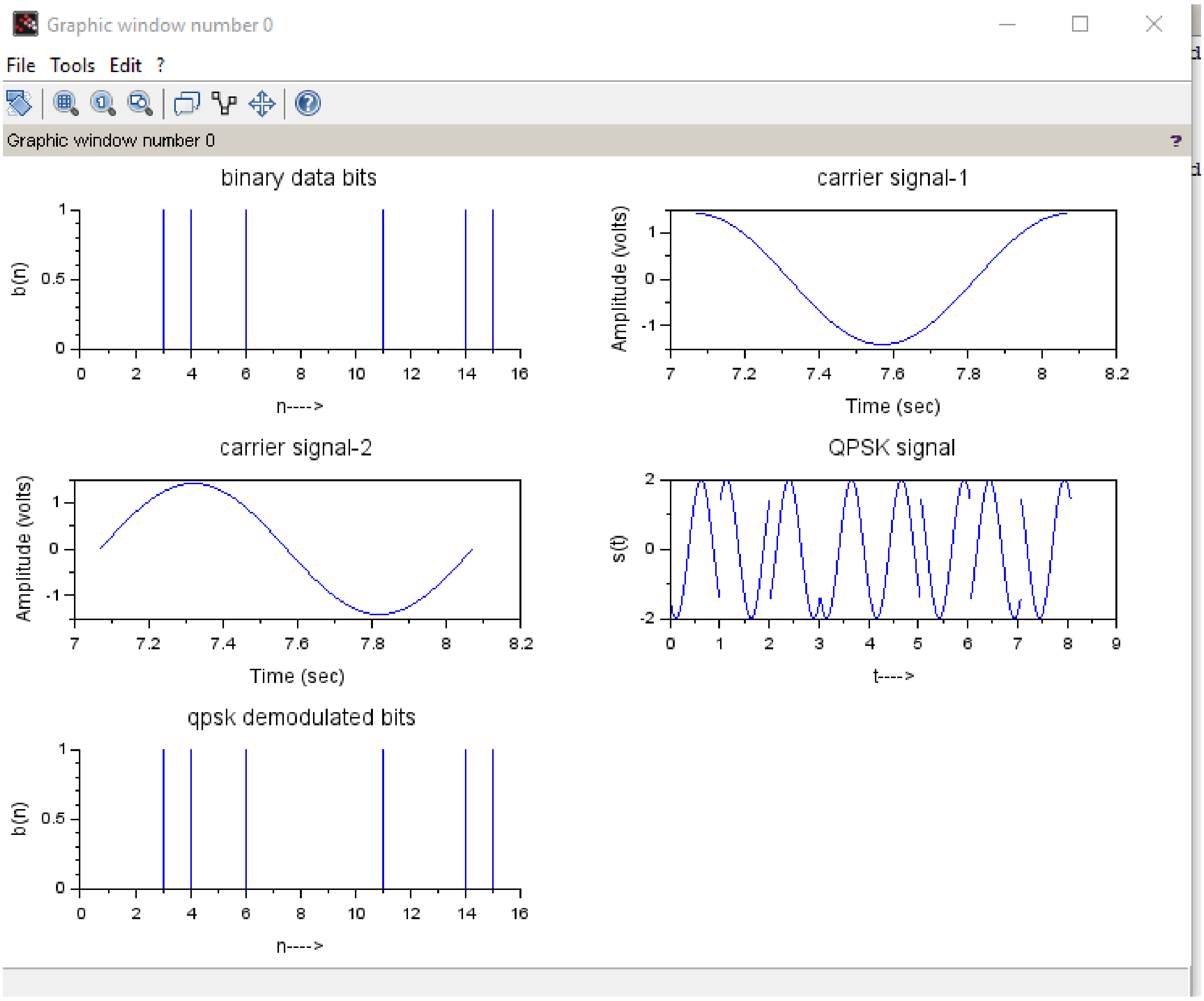


SCILAB CODE:-

```
clc ;
clear ;
xdel ( winsid () ) ;
Tb=1;
t=0:(Tb/100):Tb;
fc=1;
c1=sqrt(2/Tb)*cos(2*%pi*fc*t);
c2=sqrt(2/Tb)*sin(2*%pi*fc*t);
N=16;
m=rand(1,N);
t1=0;t2=Tb;
for i=1:2:(N-1)
    t=[t1:(Tb/100):t2]
if m(i)>0.5
    m(i)=1;
m_s=ones(1,length(t));
else
m(i)=0;
m_s=-1*ones(1,length(t));
end
odd_sig(i,:)=c1.*m_s;
if m(i+1)>0.5
m(i+1)=1;
m_s=ones(1,length(t));
else
m(i+1)=0;
m_s=-1*ones(1,length(t));
end
even_sig(i,:)=c2.*m_s;
qpsk = odd_sig+even_sig;
subplot(3,2,4);
plot(t,qpsk(i,:));
title('QPSK signal');
xlabel('t---->');
```

```
labels
t1=t1+ Tb+
t2=t2+ Tb+
end
subplct.: ,  );
N=
N,m
N',style=
title
xlabel¿
labels
subplct.: , ,
plct t,cl
q title
xlabel(
labels
#ubplCt': , ,
plCt (t,c2 :
title
xlabel
labels
t1= t= t1           t2
t2=t0;  x1=.  \c1 .  o fi\
^  ‹c2 .'ps t,‹ 1,
2= iS .:x1:›
x2:›
demc  i'=
:-1s:-i demcl i+
**x2 : ›
3emc3(i =
3emc3 i+

:-1s:-iS .:xl.:
six2< jj     demcd
i =
i     demcd(i+
:-1s:-iS  xl :
-x2:. ›
3emc3(i
= 3emc3
t2=t2+(Tb+
end :-a:l
subplot(2,2, i
t1=t1+ Tb+
N=
(N,demod(N),style=
<- n:l
tltle
xlabel¿
ylabel
```

OUTPUT:-

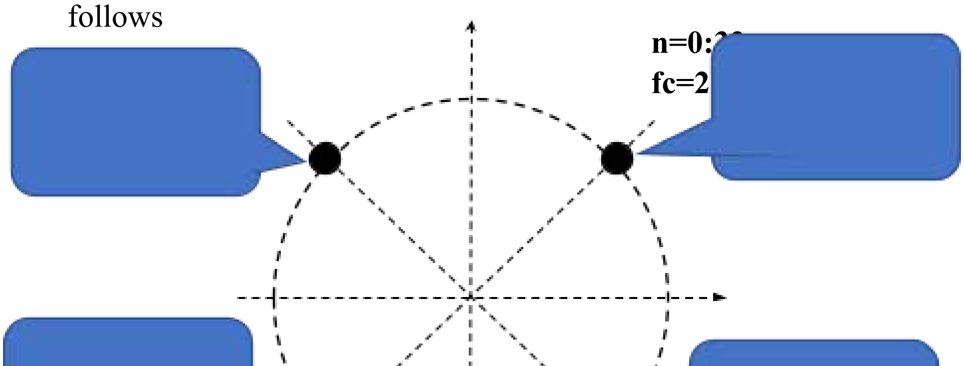


Pre-lab Questions:

1. Draw the signal space diagram of coherent QPSK system
2. By using the BER equation, justify how the QPSK system supersedes the BPSK system?

Post-lab Questions:

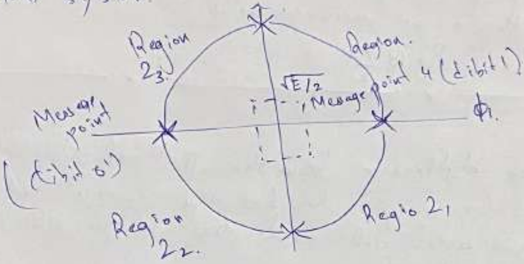
1. State the difference between the plot and subplot?
2. Compare QPSK and OQPSK.
3. Make QPSK waveform using SCIlab as follows



EXP-10

Prelab

1) Draw the signal space diagram of coherent QPSK system.



2) By using BER equation justify how the QPSK system supercedes the BPSK system.

* QPSK modulation consists of the two BPSK modulation of in-phase and quadrature components of the signal.

$$P_b = Q\sqrt{\frac{2E_b}{N_0}}$$

The symbol probability error (SER) is the probability of either branch has a bit error

$$P_s = 1 - [1 - Q(M_s)]^2, \text{ where } M_s = 2E_b$$

We can use the union bound to give an upper bound on SER of QPSK, condition that the symbol zero is sent the probability of error is bounded by the error of probabilities of $0 \to 1$, $0 \to 2$, & $0 \to 3$

$$P_s \le Q(d_{01}/\sqrt{2N_0}) + Q(d_{02}/\sqrt{2N_0}) + Q(d_{03}/\sqrt{2N_0})$$

$$\le 2Q(A/\sqrt{N_0}) + Q(\sqrt{2}A/\sqrt{2N_0})$$

since $M_s = 2M_b = \frac{A^2}{N_0}$ for $z > 0$, $Q(z) = \frac{1}{2\sqrt{2\pi}} e^{-z^2/2}$

$$P_s \le \frac{3}{\sqrt{2\pi M_s}} e^{-0.5 M_s}$$

Using gray coding and assuming that for high SNR, the error occurs only for the neighbours

$$P_b \approx \frac{P_s}{2}$$

Post-lab

1) State the difference b/w the plot and subplot?

→ Subplot (m, n, p) divides the current figure into m×n grid and creates axes in the position specified by p.

→ Plot (x, y) creates a 2-D line plot of the data in an x-y axis

2) Compare QPSK & OQPSK.

QPSK → Phase shift of ±90° & ±180°

→ Amplitude variation of the order of 30 dB

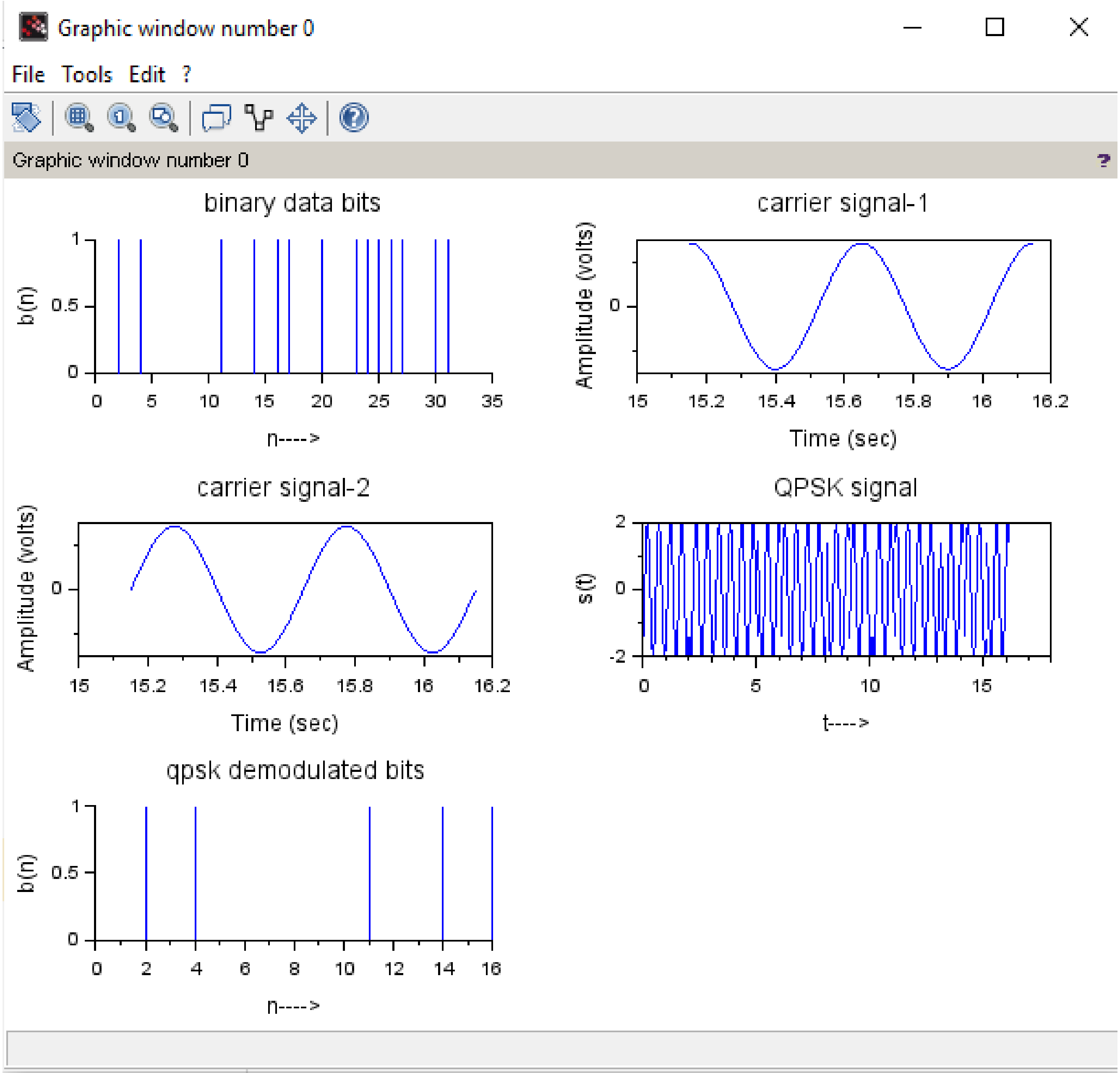
OQPSK → phase change of ±90°

→ Amplitude variation are of the order 3 dB

POST LAB ANS 03:-

```
clc ;
clear ;
xdel ( winsid () ) ;
Tb=1;
t=0:(Tb/100):Tb;
fc=2;
c1=sqrt(2/Tb)*cos(2*%pi*fc*t);
c2=sqrt(2/Tb)*sin(2*%pi*fc*t);
N=32;
m=rand(1,N);
t1=0;t2=Tb;
for i=1:2:(N-1)
    t=[t1:(Tb/100):t2]
if m(i)>0.5
    m(i)=1;
m_s=ones(1,length(t));
else
m(i)=0;
m_s=-1*ones(1,length(t));
end
odd_sig(i,:)=c1.*m_s;
if m(i+1)>0.5
m(i+1)=1;
m_s=ones(1,length(t));
else
m(i+1)=0;
m_s=-1*ones(1,length(t));
end
even_sig(i,:)=c2.*m_s;
qpsk = odd_sig+even_sig;
subplot(3,2,4);
plot(t,qpsk(i,:));
title('QPSK signal');
xlabel('t---->');
```

```
ylabel.:

#*'tl+.:
Tb+
t2=t2+.:T   ,i);
b+    N=
p c— n:l         ¿N,m¿N
subplct¿    ,style=
title
¿ xlabel
•:
q¿          ,2);
ylabel•:
g¿ subplct•;
, g° plct •;
t, cl.';
subplot(3,2, );
plot(t,c2);
xlabel•:
title ylabel
xlabel•:
ylabel
•:
, i=  N-
t2=Tb:   Tb       t2
l      •;cl. 'qpsk
x1=    i;
x2=    :c2. 'jpsk
i, it {x1:• ssx2:•
demcd•:i;=
:-is:-if demcd•:i+
x2 :
3emc3•:i'=
3emc3•:i+

¿g :-1s:-iS .:x1.:
six2<
demcd i =
demcd(i+
:-1s:-iS x1 :
-x2:. >
3emc3{i
= 3emc3
t2=t2+(Tb+
end :-a:l
subplot(Tb+ ,  i
N=
(N,demod(N),style=
<- n:l
tltle
g
xlabel¿
ylabel
```



Result:-The program for QPSK modulation and demodulation has been simulated in the SCI lab and necessary graphs are plotted.

Laboratory Report Cover Sheet

SRM Institute of
Science and
Technology
College of Engineering
and Technology

Department of Electronics and Communication Engineering

18ECC205J ANALOG AND DIGITALCOMMUNICATION **Fifth Semester, 2021-22 (ODD semester)**

Name : Pushpal Das

Register No. : RA1911004010565

Day / Session : 2^{ND} / FN

Venue : Online (G meet)

Title of Experiment : DPSK Modulation and Demodulation Date of Conduction : 01 OCT 2021

Date of Submission : 18 OCT 2021

Particulars	**Max. Marks**	**Marks Obtained**
Pre-lab questions	05	
In-lab experiment	10	
Post-lab questions	10	
Record Submission	05	

Total	**30**	

REPORT VERIFICATION

Date : 11 OCT 2021

Staff Name : Dr. P. Malarvezhi / Mrs.Harisudha

Signature :

11. Differential Phase Shift Keying (DPSK) Modulation and Demodulation

11.1 Objective

To construct a DPSK modulation and demodulation system and interpret the modulated and demodulated waveforms.
Simulate the DPSK modulation and demodulation using SCILAB.

11.2 Hardware Required

1. DPSK Trainer Kit – ST8112
2. Dual Trace oscilloscope-POS-2020
3. Digital Multimeter

11.3 Theory

DPSK system may be viewed as the non coherent version of the PSK. It eliminates the need for coherent reference signal at the receiver by combining two basic operations at the transmitter (i) Differential encoding of the input binary wave and (ii) Phase shift keying Hence the name differential phase shift keying [DPSK]. To send symbol 0 we phase advance the current signal waveform by 180^0 and to send symbol 1 we leave the phase of the current signal unchanged.

In the differential encoding at the transmitter input starts with an arbitrary first bit serving as reference and thereafter the sequence is generated using

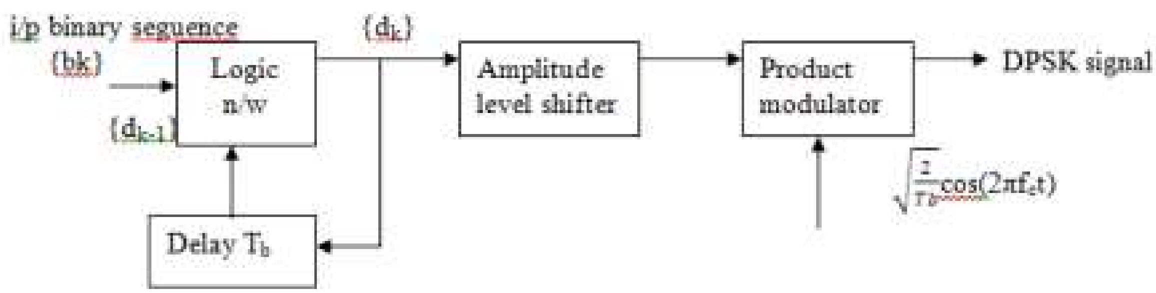
$$d_k = d_{k-1}b_k + \bar{d}_{k-1}\bar{b}_k \qquad \text{modulo-2}$$

d_{k-1} �� previous value of differentially encoded digit.
b_k �� i/p binary digit at time kT_b.
d_{k-1}, b_k �� logical inversion.

Assuming reference bit added to {dk} is a'1'. {dk} is thus generated and used to phase shift key a carrier with phase angles 0 and π.

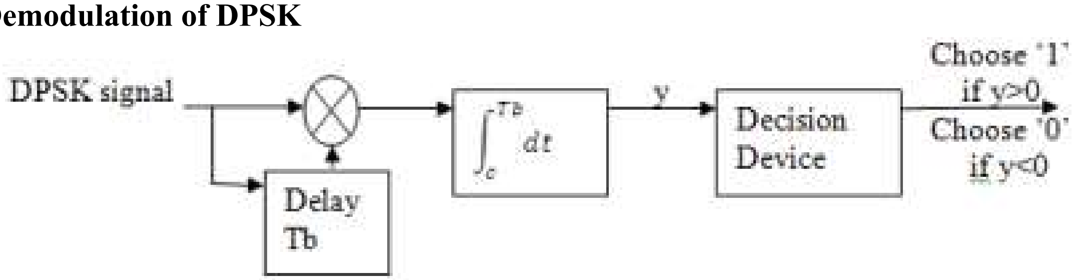
$\{b_k\}$		1	0	0	1	0	0	1	1
$\{\bar{b}_k\}$		0	1	1	0	1	1	0	0
$\{d_{k-1}\}$		1	1	0	1	1	0	1	1
$\{\bar{d}_{k-1}\}$		0	0	1	0	0	1	0	0
$\{b_k d_{k-1}\}$		1	0	0	1	0	0	1	1
$\{\bar{b}_k \bar{d}_{k-1}\}$		0	0	1	0	0	1	0	0
Differentially encoded sequence $\{d_k\}$	1	1	0	1	1	0	1	1	1
Transmitted phase (*radians*)	0	0	π	0	0	π	0	0	0

Modulation of DPSK



The transmitter consists of a logic network and a one-bit delay element interconnected so as to convert a input binary sequence $\{b_k\}$ into a differentially encode sequence $\{d_k\}$. This sequence is amplitude shifted and then used to modulate a carrier wave of frequency f_c, to produce the desired DPSK wave.

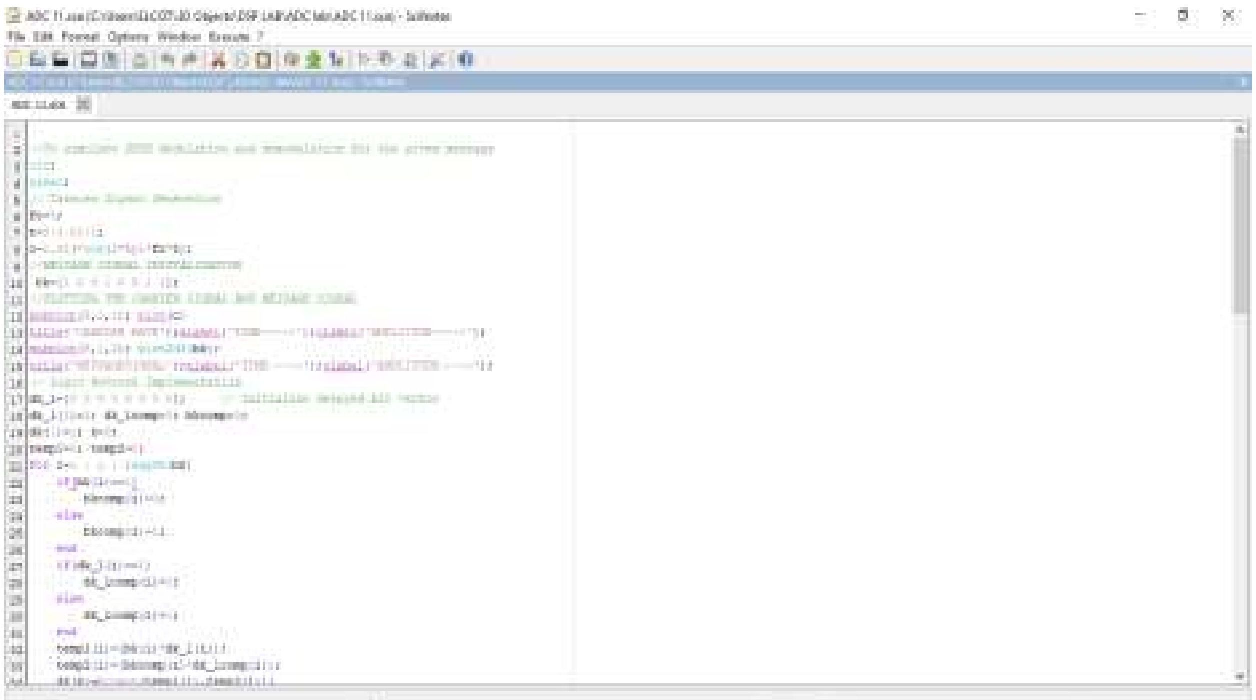
Demodulation of DPSK

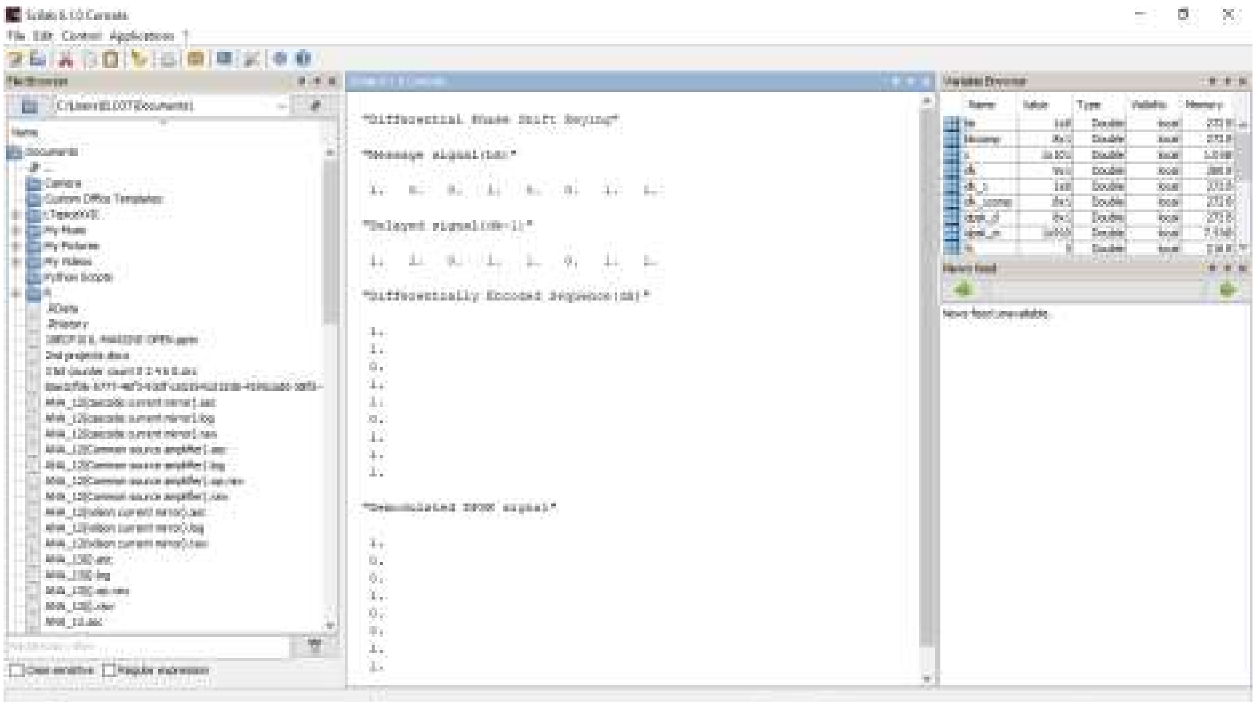


At the receiver input, the received DPSK signal plus noise is passed through a band pass filter centered at freq f_c. The filter input & a delayed version of it (delay T_b) are applied to the correlator.

The correlator output is compared with a threshold of generated voltage. If the correlator output is positive the phase difference between waveform received during the pertinent pair of bit intervals lies inside the range $-\pi/2$ to $\pi/2$ & receiver decides in favor of symbol 1. If correlator o/p is negative, the phase diffference lies outside the range $-\pi/2$ to $\pi/2$ and receiver decides in favor of symbol 0.

SCI LAB CODE & SIMULATION:-





Observation- Software:-

	D0	D1	D2	D3	D4	D5	D6	D7
Input Data Bit	**1**	**0**	**0**	**1**	**0**	**0**	**1**	**1**
Differentially encoded bits	**1**	**1**	**0**	**1**	**1**	**0**	**1**	**1**
DPSK Demodulated Bit	**1**	**0**	**0**	**1**	**0**	**0**	**1**	**1**

11.4 Pre-Lab Questions

1. Why is DPSK preferred over PSK
2. For the message signal 11001110, write the differential encoded signal.
3. DPSK uses coherent form of PSK.
   a) True
   b) False

PRE LAB:-

Exp - 11

RA1911004010565
Pushpal Das

Postlab

1) Why is DPSK preferred over PSK?

=> DPSK is preferred over PSK because a coherent carrier is not required to generate to the receiver. Hence no circuit for generation of carrier is avoided. & bandwidth for DSP is less compared to PSK.

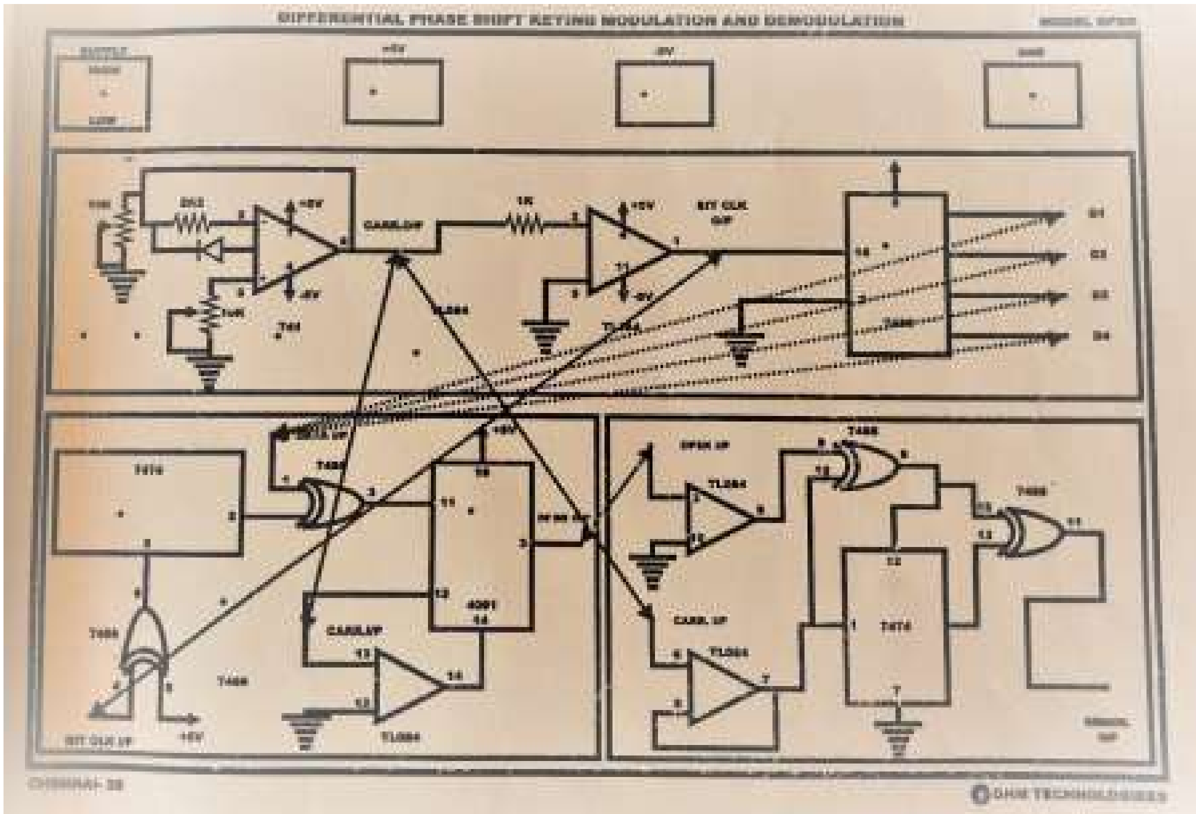
2) For the message signal 11001110, write the differential encoded signal.

=> Message signal
11001110.

3) DPSK uses coherent for it PSK.

=> False

11.5 Wiring Diagram



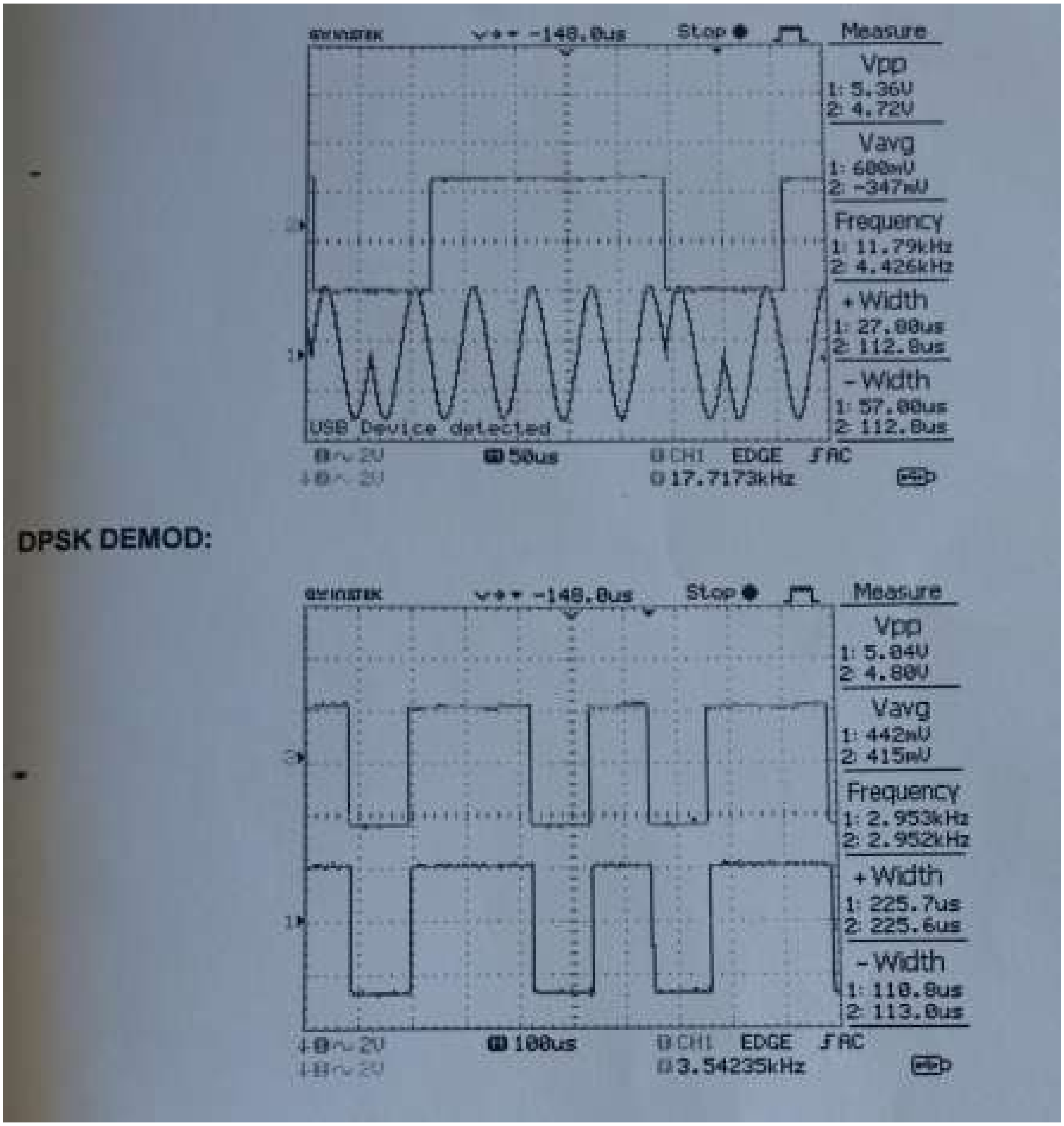
11.6 Procedure

1. Connect the AC Supply to the Kit.
2. Make connections and settings as shown in Block Diagram.
3. Connect the carrier signal output from carrier block to the 'Carrier Input' post of PSK Modulator block. Connect the 'D_0' from data block to the data input post of DPSK Modulator block.
4. Switch ON the power.
5. Connect the DPSK MOD O/P to the DPSK MOD I/P of DPSK demodulator section. 2.

Connect the carrier signal output from carrier block to the 'Carrier Input' post of DPSK demodulator section.

3. Change the data I/P to D_1, D_2, D_3 and observe the DPSK
4. O/P changes accordingly.
5. Observe the following waveforms on oscilloscope.
    a. Carrier signal O/P.
    b. Data O/P D_0, D_1, D_2, D_3.
    c. DPSK MOD O/P at DPSK Modulator.
    d. DPSK Demodulated signal at DPSK demodulator BLOCK.

11.7 Model Graph



11.2 Model Graph

11. 8 Observation

DPSK (Modulation)

Data source	Amplitude (V)	Time Period (ms)
D_0		

D_1		
D_2		

D_3		
Carrier signal		
Modulated output		
D_0		
D_1		
D_2		
D_3		

Demodulation

Demodulated output	**Amplitude (V)**	**Time Period (ms)**
D_0		
D_1		
D_2		
D_3		

11.9 VIRTUAL LAB
11.9.1 Equipment / Apparatus

SCILAB Software 6.0.2

11.9.2 Exercise

For the given 8-bit data [10010011] and carrier signal of Cos(2*π*5*t) generate the DPSK Modulated , Demodulated Signal and obtain the corresponding output Bit stream.

11.9.3 Solution

Algorithm

Initialization commands

DPSK modulation

1. Generate carrier signal.
2. Initialize binary bits, bk
3. Plot carrier and message signal
4. Start FOR loop
5. Take complement of bk and delayed bit dk
6. Obtain the differentially encoded sequence by using the logical xor operation
7. End FOR loop
8. Plot the differentially encoded sequence
9. Modulate the differentially encoded sequence by assigning carrier without phase shift for ‘1’ and carrier with 180 degree phase shift for’0’. 10. Plot the DPSK signal.

DPSK demodulation

1. Start FOR loop
2. Perform correlation of DPSK modulated signal with delayed version of itself to get decision variable y
3. Make decision on y. If y>0, choose ‘1’ else choose ‘0’
4. End FOR loop
5. Plot demodulated data

Program

```
// Experiment Number :
//Write a program to perform DPSK Modulation and Demodulation
// 18ECC205J: Analog and Digital Communication Laboratory //
Scilab 6.0.2
// B. Tech ECE : III Year V Sem
// Student Name :
// Register Number :

//To simulate DPSK Modulation and demodulation for the given message
clc;
clear;
// Carrier Signal Generation
fc=5;
t=0:0.01:1;
c=1.414*cos(2*%pi*fc*t);
//MESSAGE SIGNAL INITIALIZATION
```

```
bk=[1 0 0 1 0 0 1 1];
//PLOTTING THE CARRIER SIGNAL AND MESSAGE SIGNAL
subplot(5,1,1); plot(c)
title('CARRIER WAVE');xlabel('TIME---->');ylabel('AMPLITUDE---->');
subplot(5,1,2); plot2d3(bk);
title('MESSAGESIGNAL');xlabel('TIME---->');ylabel('AMPLITUDE---->')
; // Logic Network Implementation
dk_1=[0 0 0 0 0 0 0 0]; // Initialise delayed bit vector
dk_1(1)=1; dk_1comp=0; bkcomp=0;
dk(1)=1; k=2;
temp1=0; temp2=0;
for i=1 : 1 : length(bk)
 if(bk(i)==1)
 bkcomp(i)=0;
 else
 bkcomp(i)=1;
 end
 if(dk_1(i)==1)
 dk_1comp(i)=0;
 else
 dk_1comp(i)=1;
 end
 temp1(i)=(bk(i)*dk_1(i));
 temp2(i)=(bkcomp(i)*dk_1comp(i));
 dk(k)=bitxor(temp1(i),temp2(i));
 if(i<8)
 dk_1(i+1)=dk(k);
 k=k+1;
 end
end
 //plotting the differentially encoded signal
 subplot(5,1,3);
plot2d3(dk);
title('DIFFERENTIALLY ENCODED SIGNAL');
xlabel('TIME---->');
ylabel('AMPLITUDE---->');

// Generate DPSK modulated signal from the differentially encoded signal
```

```
dpsk_m=0;
for k=1:1:9
 if(dk(k)==1)
 dpsk_m=[dpsk_m c];
 else
 dpsk_m=[dpsk_m -c];
 end
end
 //plotting the DPSK modulated signal
 subplot(5,1,4);
plot(dpsk_m);
title('DPSK MODULATED SIGNAL');
xlabel('TIME---->');
ylabel('AMPLITUDE---->');
// DPSK DEMODULATION
// Implementing the auto correlator
dpsk_d=0; x=1; z=length(t);
for j=1:1:8
 y=0;
for l=x:1:(z-1)
 y=y+(dpsk_m(l)*dpsk_m(l+length(t)));
end
//Implementing the decision device
if(y<0)
 dpsk_d(j)=0;
else
 dpsk_d(j)=1;
end
x=x+length(t);
z=z+length(t);
end

//plotting the demodulated signal

subplot(5,1,5);
plot2d3(dpsk_d);
title('RECEIVED SIGNAL AFTER DPSK DEMODULATION');
xlabel('TIME---->');
```

```
ylabel('AMPLITUDE---->');

// Display all the signals in the SCILAB console
disp("Differential Phase Shift Keying")
disp("Message signal(bk)")
disp(bk)
disp("Delayed signal(dk-1)")
disp(dk_1)
disp("Differentially Encoded Sequence(dk)")
disp(dk)
disp("Demodulated DPSK signal")
disp(dpsk_d)
```

11.10 Observation- Software

	D0	D1	D2	D3	D4	D5	D6	D7
Input Data Bit								
Differentially encoded bits								
DPSK Demodulated Bit								

11.11 Model Graph

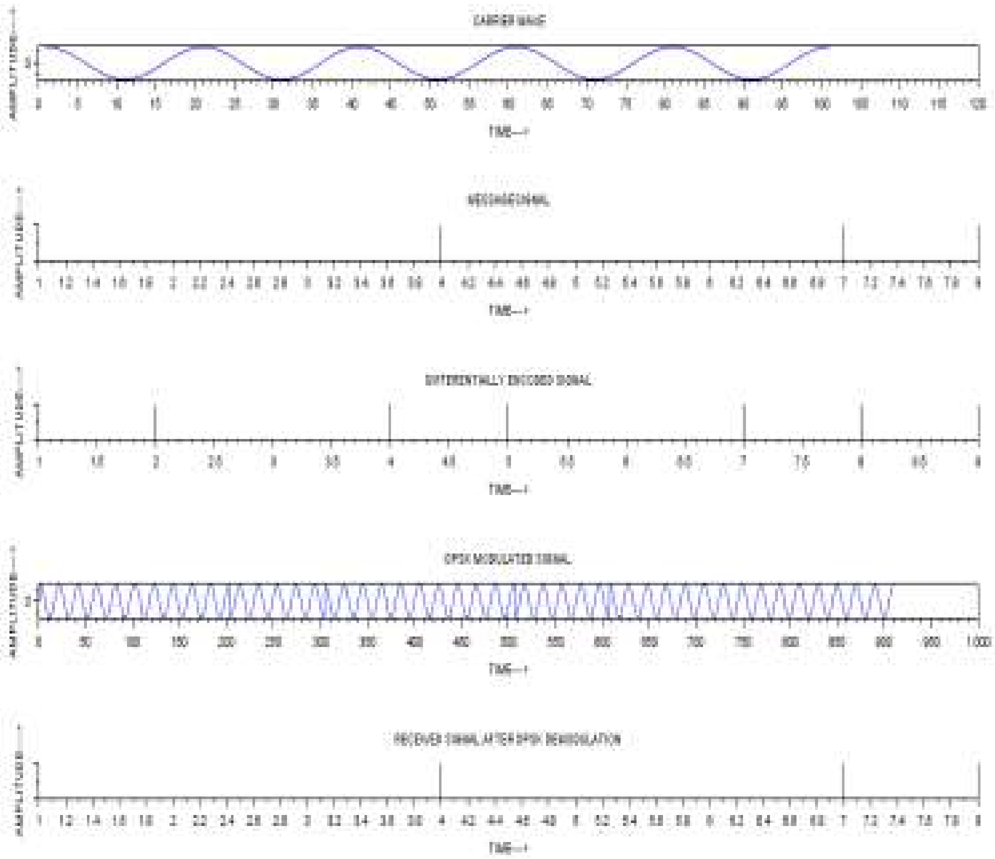


Fig 11.6 Simulated DPSK Modulated and Demodulated waveforms

11.12 Output Obtained

```
"Differential Phase Shift Keying"

"Message signal(bk)"

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

"Delayed signal(dk-1)"

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

"Differentially Encoded Sequence(dk)"

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

"Demodulated DPSK signal"

 1.
 0.
 0.
 1.
 0.
 0.
 1.
 1.
```

11. 12 Post Lab Questions

1. Compare DPSK and QPSK modulation scheme.
2. Generate a random stream of data of length N = 15 (use rand syntax) and obtain the DPSK Modulated, Demodulated Signal and corresponding output Bit stream using SCILAB. 3. Differentiate between Coherent and Non-coherent modulation
4. The energy efficiency of DPSK is _________ to coherent PSK.
    a) Superior b) Same c) Zero d) Inferior
5. How many phases does DPSK transmit?

POST LAB:-

11.11 Lab Result

Thus, the DPSK modulation and demodulation has been simulated in SCILAB and necessary graphs were plotted.

Laboratory Report Cover Sheet

SRM Institute of Science and Technology

Faculty of Engineering and Technology

Department of Electronics and Communication Engineering **18ECC205J**

ANALOG AND DIGITAL COMMUNICATION **Fifth Semester, 2020-21 (odd semester)**

Name : Pushpal Das

Register No. : RA1911004010565

Day/ Session :2ND / FN

Venue : Online (G Meet)

Title of Experiment : BER Performance Analysis of various modulation

Date of Conduction : 08 OCT 2021

Date of Submission : 18 OCT 2021

Particulars	**Max. Marks**	**Marks Obtained**
Pre-lab questions	05	
In-lab experiment	10	
Post-lab questions	15	
Total	**30**	

REPORT VERIFICATION

Date : 13 OCT 2021

Staff Name : Dr. P. Malarvezhi / Mrs.Harisudha

Signature :

12 BER Performance Analysis of various modulation

12.1 Objective:

To analyze the BER performance of a Communication System using BASK and QPSK scheme.

12.2 BER Performance analysis of BASK Scheme

```
clc;
    clear;
    A=1*10^-3;
    Tb=0.2*10^-3;
    fb=1/Tb;
    fc=5*fb;
    N0=2*10^-11;// power sepctral density
    Eb=(A^2*Tb)/2;//Eb=bit energy
    x=sqrt(Eb/2*N0);
    Pa=erfc(x)/2'//bit error probability
    disp("Error probability of BASK is P(a)")
   disp(Pa);
```

output:

"Error probability of BASK is P(a)"

0.5

12.3 BER Performance comparision of BPSK and QPSK Scheme

```
clc;
clear;
xdel(winsid());
sym=10000;//No .of symbols
M=4;
qpsk_mod=[];i_phase=[];
data1=grand(1,sym,"uin",0,1);//Random Symbol generation from 0 to 1 with
uniform //distribution
for j=1:2:length(data1)// Seperation of I & Q component
i_phase=2*data1(j)-1;// BPSK modulation of I phase component
q_phase=2*data1(j+1)-1;//BPSK modulation of Q phase component
```

```
temp=i_phase+%i*q_phase;//combining of I phase and Q phase component for QPSK
```



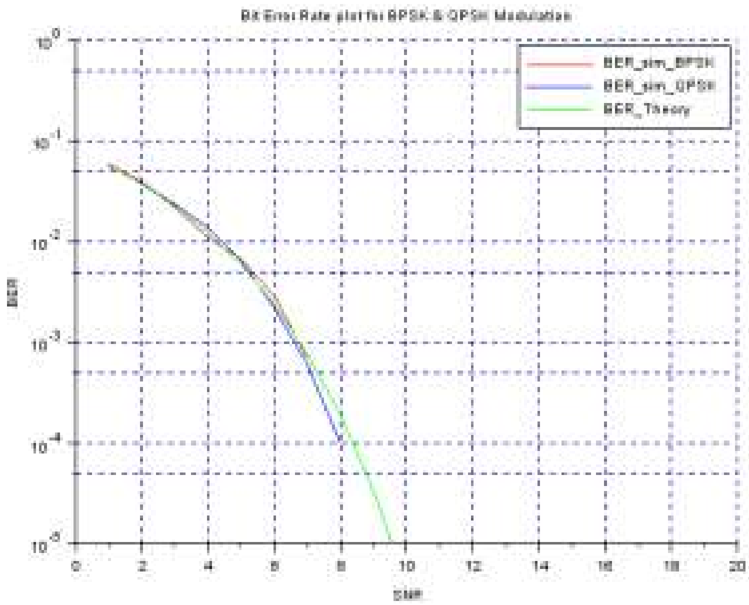
```
//modulation
qpsk_mod=[qpsk_mod temp];//QPSK modulated signal
end
bpsk_mod=2*data1-1;//BPSK Modulated signal
snr=1:10;//Signal to Noise Ratio
for k=1:1:length(snr)
H=1/sqrt(2)*(rand(1,length(qpsk_mod),'normal')+%i*(rand(1,length(qpsk_mod),'normal')))
; noise1=1/sqrt(2)*(10^(-
(k/20)))*(rand(1,length(qpsk_mod),'normal')+%i*(rand(1,length(qpsk_mod),'normal')));//White
Gaussian Noise generation for QPSK
noise=1/sqrt(2)*(10^(-
(k/20)))*(rand(1,length(bpsk_mod),'normal')+%i*(rand(1,length(bpsk_mod),'normal')));//White
Gaussian Noise generation for QPSK
rec1_qpsk=qpsk_mod+noise1;//QPSK modulated signal over AWGN channel
rec1_bpsk= bpsk_mod+noise;//BPSK modulated signal over AWGN channel
rec_data_qpsk=[];rec_data_bpsk=[];
rec1_i=real(rec1_qpsk);//Seperation of I phase and Q phase comopnent of
//received QPSK modulated signal
rec1_q=imag(rec1_qpsk);
//
for i=1:length(rec1_i)//QPSK Demodulation: BPSK demodulation of I phase and
Q //phase components
if rec1_i(i)>=0
demod_out_i=1;
else rec1_i(i)<0
demod_out_i=0;
end
if rec1_q(i)>=0
demod_out_q=1;
else rec1_q(i)<0
demod_out_q=0;
end
rec_data_qpsk=[rec_data_qpsk demod_out_i demod_out_q];//QPSK
Demodulated //signal
end

for i=1:1:length(data1)//BPSK Demodulation
if real(rec1_bpsk(i))>=0
demod_out_bpsk=1;
else real(rec1_bpsk(i))<0
demod_out_bpsk=0;
end
rec_data_bpsk=[rec_data_bpsk demod_out_bpsk];//BPSK Demodulated
signal end

errA=0;errB=0;
for i=1:sym
```

```
if rec_data_qpsk(i)==data1(i)
errA=errA;
else
```

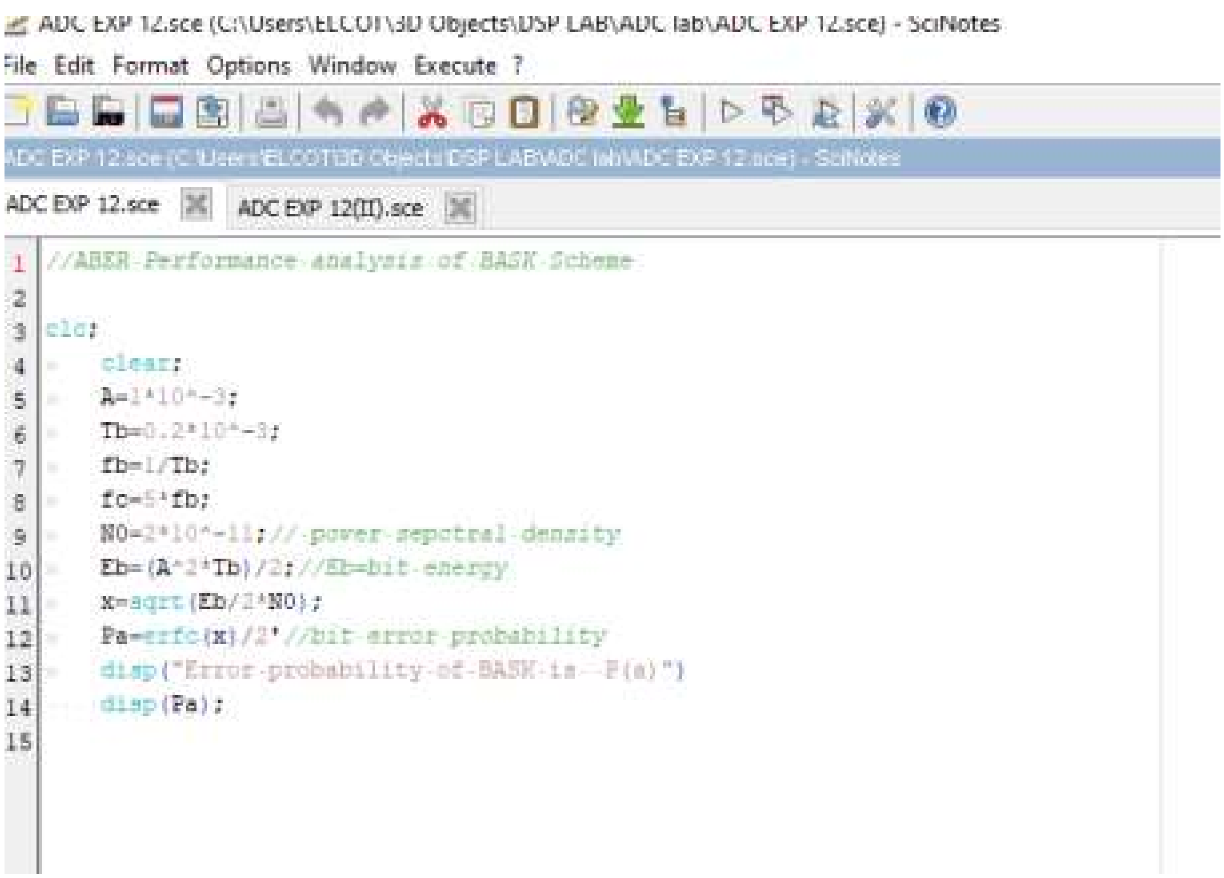


```
errA=errA+1;
end
end
BER_qpsk(k)=errA/sym;// BER of QPSK
for i=1:sym
if rec_data_bpsk(i)==data1(i)
errB=errB;
else
errB=errB+1;
end
BER_bpsk(k)=errB/sym;//BER of BPSK
end
theoryBer = 0.5*erfc(sqrt(10.^(snr/10))); //Theoritical BER of BPSK &
QPSK end
// end
snr=1:1:10;
plot2d(snr,BER_bpsk,5,logflag="nl");//plot simulated BER of BPSK over AWGN channel
plot2d(snr,BER_qpsk,2,logflag="nl");//plot simulated BER of QPSK over AWGN channel
plot2d(snr,theoryBer,3,logflag="nl");//Plot theoritical BER of QPSK and BPSK over AWGN
//channel
mtlb_axis([0 20 10^-5 0.5]);//axis
xgrid(10);
xtitle( 'Bit Error Rate plot for BPSK & QPSK Modulation', 'SNR', 'BER') ;//title of
//plot
legend(['BER_sim_BPSK';'BER_sim_QPSK';'BER_Theory']);//legend
//This experiments results plot of bit error rate(BER) comparison of simulated BPSK //over AWGN
channel,simulated QPSK over AWGN channel and theoritical BER of BPSK //and QPSK // It will take
few minutes to get plots as 100000 bits are applied as an input to get //better plots
```

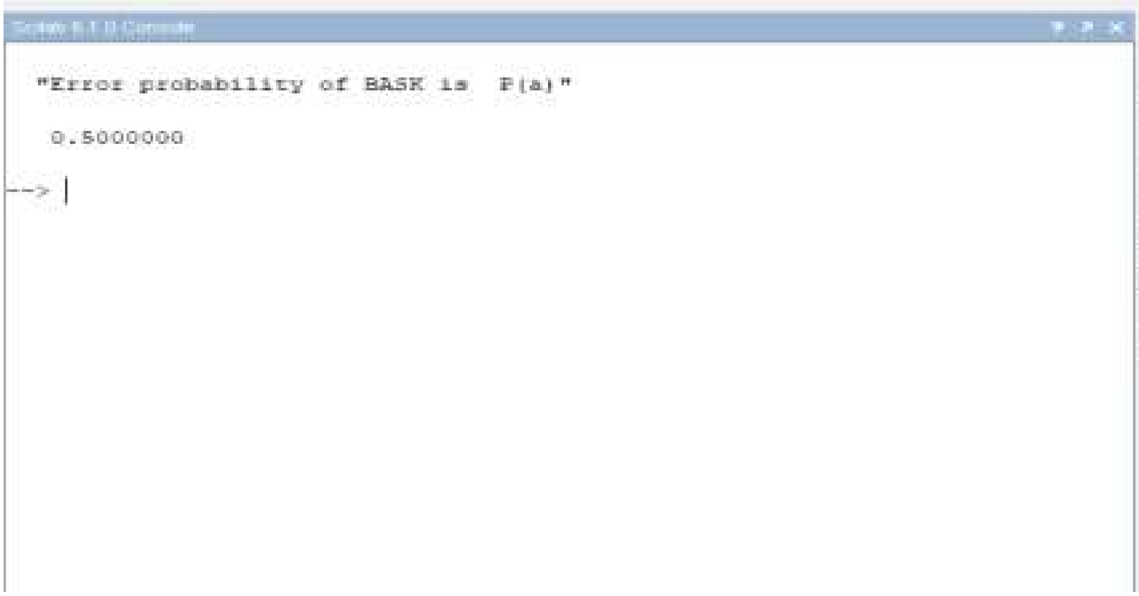
Output



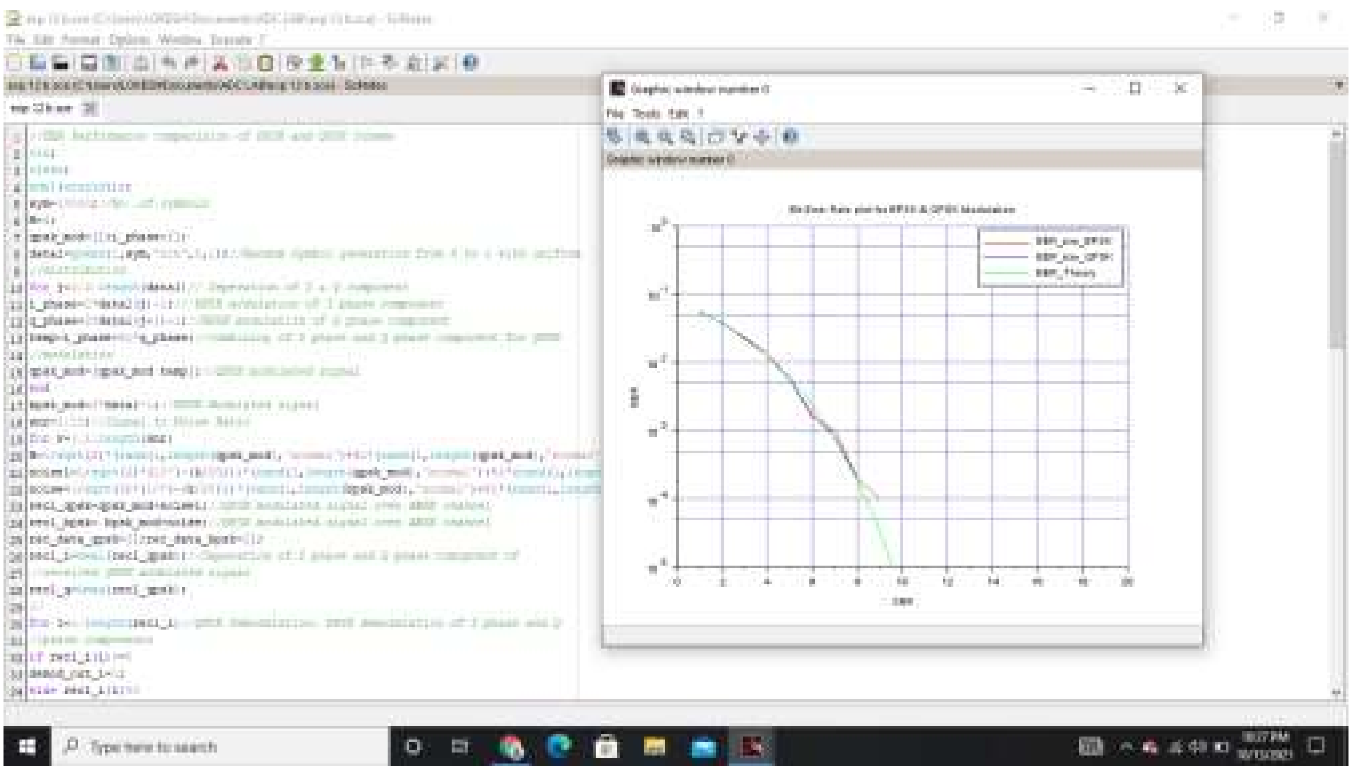
SCI LAB CODE &SIMULATION:-




```
//ABER Performance analysis of BASK Scheme

clc;
    clear;
    A=1*10^-3;
    Tb=0.2*10^-3;
    fb=1/Tb;
    fc=5*fb;
    N0=2*10^-11;// power sepctral density
    Eb=(A^2*Tb)/2;//Eb=bit energy
    x=sqrt(Eb/2*N0);
    Pa=erfc(x)/2'//bit error probability
    disp("Error probability of BASK is - P(a)")
    disp(Pa);
```



5



12.5 PRELAB:

1. What are quad bits?
2. Compute the bit rate for a 1000 baud16-QAM signal
3. In QAM, which parameters of the carrier signal are varied?

<u>PRE LAB ANS:-</u>

Exp. 12

RA1911004010565
Pushpal Das

Prelab

1) What are quad bits?

A group of four bits used in QAM modulation.
A set of four bits.

2) Compute the bit rate for 1000 baud 16-QAM signal.

A 16-QAM signal has 4 bits per signal unit since $\log_2 16 = 4$. Thus $(1000 \times 4) = 4000$ bps

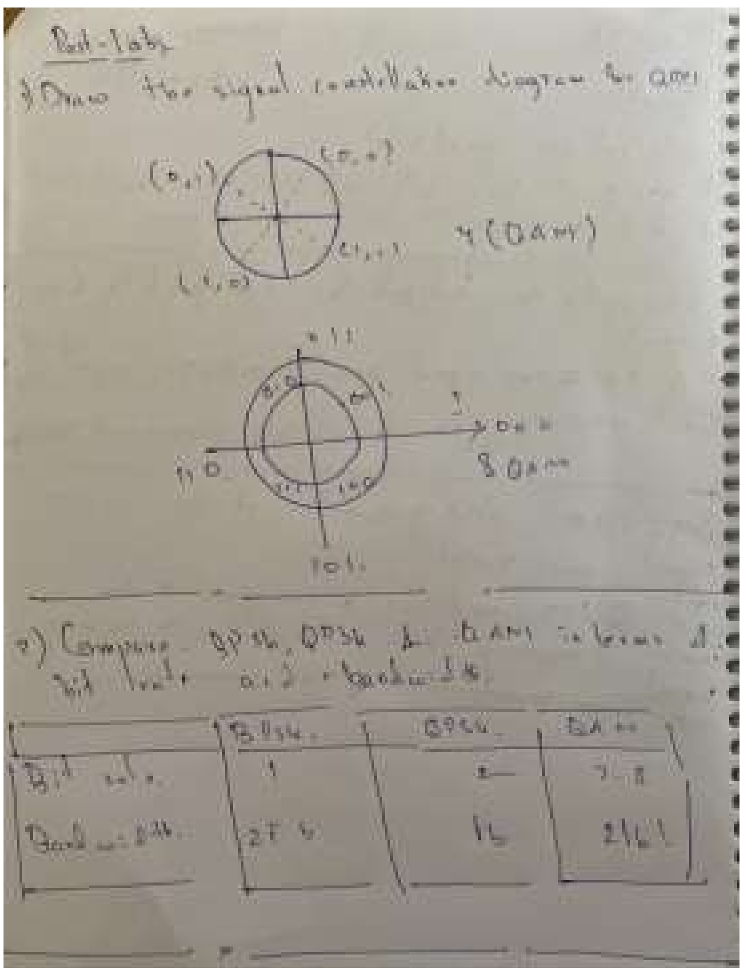
3) In QAM, which parameters of the carrier signal are varied?

In QAM, amplitude & the phase of the carrier frequency both varied with the msg signal.

12.6 POSTLAB:

1. Draw the signal constellation diagram for QAM
2. Compare BPSK, QPSK and QAM in terms of bit rate and bandwidth **<u>POST LAB ANS :-</u>**

7



12.7 Result

Thus the BER performance analysis of communication system using BPSK, BASK and QPSK was performed in SCILAB.