COURSE PLAN

A. COURSE OVERVIEW

Degree:	BE	Programme:	ECE
Academic Year:	2024-25	Semester:	IV
Course Title:	Principles Of Communication Systems	Course Code:	BEC402
L-T-P-S:	3-0-2-0	Duration of SEE	3 Hrs
Total Contact Hours:	40 Hrs+10Hrs	SEE Marks:	50*
CIE Marks:	50	IA Test	30
Credits:	4	Components	20
Lesson Plan Author:	Mrs. Nisha G R	Nisha	Date 25/01/2025
Checked By:	Mr. Nirupama K	निरुपमा	Date 25/01/2025

*The SEE will be conducted for 100 marks and proportionally reduced to 50 marks.

B. PREREQUISITES

- Applied Physics (BPHYE102)
- Basic Electronics(BBEE103)
- Introduction to Electronics Communication(BESCK104C)
- AV Mathematics-III for EC Engineering(BMATEC301)
- Electronic Principles and Circuits (BEC303)

C. COURSE DESCRIPTION

i) Course Outcomes

At the end of the course, the student will be able to;

1. Identify and associate the random variables and random process in Communication system design and noise modeling.
2. Design simple systems for generating and demodulating AM and SSB signals.
3. Understand the concepts of Angle modulation and Design simple systems for generating and demodulating frequency modulated signals.
4. Analyze pulse modulation and sampling techniques.
5. Describe the ideal condition, practical considerations of the signal representation for baseband transmission of digital signals.

ii) Relevance of the Course

- Communication Laboratory (BECL404)
- Digital Communication
- Microwave Theory and Antennas
- Communication Engineering
- Optical and wireless Communication

iii) Applications areas

- Satellite communication
- Telecommunications
- 3G communication
- Hard-Line Technology
- Portable Radios, Mobile Radios and in repeaters

Nisha | निरुपमा | [signature]

Prepared by: Nisha G R | Checked by:Nirupama K | HOD

COURSE PLAN

D1. ARTICULATION MATRIX, CO v/s PO

Mapping of CO to PO												
	POs											
COs	1	2	3	4	5	6	7	8	9	10	11	12
1 Identify and associate the random variables and random process in Communication system design and noise modeling.	2	2	3	1	1	1	-	-	-	-	1	1
2 Design simple systems for generating and demodulating AM and SSB signals.	2	3	3	1	2	1	-	-	-	-	1	1
3 Understand the concepts of Angle modulation and Design simple systems for generating and demodulating frequency modulated signals.	2	3	2	1	1	1	-	-	-	-	1	1
4 Analyze pulse modulation and sampling techniques.	2	2	2	1	2	1	-	-	-	-	1	1
5 Describe the ideal condition, practical considerations of the signal representation for baseband transmission of digital signals.	2	2	2	1	2	1	-	-	-	-	1	1

Note: Mappings in the Tables D1 (above) and D2 (below) are done by entering in the corresponding cell the Correllation Levels in terms of numbers. For Slight (Low): 1, Moderate (Medium): 2, Substantial (High): 3 and for no correllation: " - ".

D2. ARTICULATION MATRIX, CO v/s PSO

Mapping of CO to PSO		
	PSOs	
COs	1	2
1 Identify and associate the random variables and random process in Communication system design and noise modeling.	-	3
2 Design simple systems for generating and demodulating AM and SSB signals.	-	3
3 Understand the concepts of Angle modulation and Design simple systems for generating and demodulating frequency modulated signals.	-	3
4 Analyze pulse modulation and sampling techniques.	-	3
5 Describe the ideal condition, practical considerations of the signal representation for baseband transmission of digital signals.	-	3

E. MODULE PLANS

MODULE – I

Title:	Random Variables and Processes	Appr. Time:	8 Hrs
MO:			RBT

COURSE PLAN

At the end of the Module, the student will be able to:	
1. Learn Probability and Random Variables	L1
2. Understand Statistical Averages and Random Processes	L2
3. Find Mean, Correlation and Covariance Functions	L2
4. understand random process	L2
5. Understand Gaussian process	L2

Lesson Schedule:

Lecture No.	Portion to be covered	CO
1	Introduction, Probability	CO1
2	Conditional Probability	CO1
3	Random variables	CO1
4	Statistical Averages: Function of a random variable, Moments	CO1
5	Random Processes, Mean,	CO1
6	Correlation & Covariance function: Properties of autocorrelation function,	CO1
7	Cross–correlation functions,	CO1
8	Gaussian Process: Gaussian Distribution Function.	CO1

Application Areas:

- Speech and audio communication

Review Questions / Questions Appeared in the Previous Years (CO):

1	What is conditional probability? Prove that P(B)= P(A/B).P(B) /P(A)(CO1)
2	With an example, explain what is meant by statistical averages (CO1)
3	What do you mean by probability density function? Prove that the total volume under the surface of a probability density function (pdf) is always 1. (CO1)
4	Define mean, autocorrelation and auto-covariance function (CO1)
5	Define a random variable. Illustrate the relationship between sample space, random variable and probability.(CO1)
6	Define the autocorrelation and crosscorrelation functions. State the properties of auto correlation function (CO1)
7	What is binary symmetric channel? Obtain a posteriori probabilities for the binary symmetric channel using transition probability diagram (CO1)
8	Explain the following terms : i) Moments ii) central moment iii) mean iv) covarience (CO1)
9	Explain the properties of joint distribution function (CO1)
10	The random variable 'Y' is a function of another random variable 'X' such that Y=cos(X) and 'X' is uniformly distributed in the interval (-Π, Π) i,e $f_x(x) = \frac{1}{2\pi} \quad -\pi < x < \pi$; $= 0$ otherwise. Find out the mean value of 'y'. (CO1)
11	Consider the random variable X defined by probability density function $f_x(x) = \begin{cases} k & \text{a constant for } 2 \le x \le 4 \\ 0 & \text{elsewhere} \end{cases}$ Determine : i) The constant K ii) $F_X(x)$. (CO1)

12	A random variable has probability function : (CO1) $$f(x)=\begin{cases}\frac{5(1-x^4)}{4} & ;\quad 0\le x\le 1\\ 0 & ;\quad \text{elsewhere}\end{cases}$$ Find : i)$E(x)$ ii) $E(4x+2)$ iii) $E(x^2)$.

MODULE – II

Title:	Amplitude Modulation Fundamentals, AM Circuits & Frequency Division Multiplexing	Appr. Time:	8 Hrs

MO:	RBT
At the end of the Module, the student will be able to:	
1 Define Modulation, overmodulation and explain how to alleviate its effects	L1
2 Calculate the modulation index and percentage of modulation of an AM signal, given the amplitudes of the carrier and modulating signals.	L2
3 Explain how the power in an AM signal is distributed between the carrier and the sideband, and then compute the carrier and sideband powers, given the percentage of modulation.	L2
4 Compute sideband frequencies, given carrier and modulating signal frequencies.	L2
5 Compare time-domain, frequency-domain, and phasor representations of an AM signal.	L2
6 Explain what is meant by the terms DSB and SSB and state the main advantages of an SSB signal over a conventional AM signal.	L2

Lesson Schedule:

Lecture No.	Portion to be covered	CO
1	AM Concepts, Modulation index and Percentage of Modulation	CO2
2	Sidebands and the frequency domain, AM Power	CO2
3	Single Sideband Modulation	CO2
4	Amplitude Modulators: Diode Modulator, Transistor Modulator, collector Modulator	CO2
5	Amplitude Demodulators: Diode Detector	CO2
6	Balanced Modulators (Lattice type)	CO2
7	Transmitter-Multiplexer, Receiver-Demultiplexer.	CO2
8	Superheterodyne receiver, Frequency Conversion: Mixing Principles, JFET Mixer.	CO2

Application Areas:

- Radio broadcasting
- Analog Telephony
- TV broadcast

Review Questions / Questions Appeared in the Previous Years (CO):

1	Write an AM wave expression in time and in frequency domain. Also, draw AM waveform. (CO2)
2	An AM signal with a carrier of 1kW has 200W in each side band. What is the percentage of modulation? (CO2)
2	The message signal $m(t) = 1/(1+t^2)$. Determine and sketch the modulated wave for amplitude modulation with the following values : i) u= 50% , ii)u= 100% (CO2)

4	Derive an equation for SSB modulated wave for which upper side-band is retained. (CO2)
5	State the four main benefits of SSB over conventional AM.(CO2)
6	An AM wave displayed on an oscilloscope has values of Vmax = 4.8 and Vmin = 2.5 as read from the grati-cule. What is the percentage of modulation?(CO2)
7	What is the bandwidth of an AM signal whose carrier is 2.1 MHz modulated by a 1.5-kHz square wave with significant harmonics up to the fifth? Calculate all the upper and lower sidebands produced.(CO2)
8	An AM transmitter puts a carrier of 6 A into an antenna whose resistance is 52 V. The transmitter is modulated by 60 percent. What is the total output power?(CO2)
9	Describe the two basic ways in which amplitude modulator circuits generate AM(CO2)
10	How is the output power of an SSB transmitter expressed?(CO2)

MODULE – III

Title:	Fundamentals of Frequency Modulation, FM Circuits & Communication Receiver	Appr. Time:	8 Hrs
MO:			RBT
At the end of the Module, the student will be able to:			
1 Compare and contrast frequency modulation and phase modulation.			L2
2 Calculate the modulation index			L2
3 Understand Angle modulation concept			L2
4. Ability to generate FM signal using different modulation techniques			L2
5. Understand the difference between the linear and nonlinear modulation			L2
6. Analyze FM demodulation circuit operation for balanced frequency discriminator			
Lesson Schedule:			

Lecture No.	Portion to be covered	CO
1	Basic Principles of Frequency Modulation,	CO3
2	Principles of Phase Modulation	CO3
3	Modulation index and sidebands	CO3
4	Noise Suppression effects of FM	CO3
5	Frequency Modulation versus Amplitude Modulation	CO3
6	Frequency Modulators: VCO	CO3
7	Frequency Demodulators: Slope Detectors, Phase Locked Loops.	CO3
8	Transmitter Fundamentals: Transmitter Configurations, .	CO3

Application Areas:

- Commercial radio broadcasting
- Cellular radio broadcasting
- Microwave and satellite communication systems
- Broadcasting music and speech
- Magnetic tape-recording systems

Review Questions / Questions Appeared in the Previous Years(CO):

1	Explain a typical FM transmitter using indirect FM with a phase modulator.(CO3)
2	What are the maximum and minimum frequencies of a 16-MHz crystal with a stability of 200 ppm?(CO3)
3	Explain the basic principles of FM and PM(CO3)
4	What is the deviation ratio of TV sound if the maximum deviation is 25 kHz and the maximum modulating frequency is 15 kHz?(CO3)

5	Explain FM signal spectra. (a) Modulation index of 0 (no modulation or sidebands). (b) Modulation index of 1. (c) Modulation index of 2. (d) Modulation index of 0.25 (NBFM) (CO3)
6	What is the maximum bandwidth of an FM signal with a deviation of 30 kHz and a maximum modulating signal of 5 kHz as determined by (a) Carrier and sideband amplitudes for different modulation indexes of FM signals based on the Bessel functions. and (b) Carson's rule?(CO3)
7	Explain noise suppression effects of FM?(CO3)
8	Explain the advantages and disadvantages of FM?(CO3)
9	Explain the operation of VCO?(CO3)
10	Explain the operation of slope detector circuit?(CO3)
11	Explain the operation of quadrature detector(CO3)
12	Explain the operation of Phase locked loop.(CO3)

MODULE – IV

Title:	Digital communication Techniques	Appr. Time:	8 Hrs
MO:			RBT
At the end of the Module, the student will be able to:			
1 Explain how quantizing error occurs, describe the techniques used to minimize it,			L2
2 List the advantages and disadvantages of the three most common types of analog-to-digital converters.			L1
3 Explain oversampling and undersampling and state their advantages and disadvantages			L2
4 Explain why pulse-code modulation has superseded pulse-amplitude modulation (PAM)			L2
5 Compare various pulse modulation methods			L2
Lesson Schedule:			
Lecture No.	Portion to be covered		CO
1	Digital Representation of Analog Signals: Introduction, Why Digitize Analog Sources?		CO4
2	The Sampling process,		CO4
3	Pulse Amplitude Modulation, Time-Division Multiplexing,		CO4
4	Pulse Position Modulation: Generation and Detection of PPM wave.		CO4
5	The Quantization Process.		CO4
6	Pulse Code Modulation: Sampling, Quantization, Encoding, line Codes,		CO4
7	Differential encoding, Regeneration,		CO4
8	Decoding, filtering, multiplexing.		CO4
Application Areas:			
• Image processing • Digital signal processing			
Review Questions / Questions Appeared in the Previous Years (CO):			
1	State sampling theorem. Write the mathematical form of sampled signal and explain the steps to reconstruct the signal g(t) from the sequence of sample value. (CO4)		
2	What is aperture effect? Briefly explain how to overcome this effect. (CO4)		
3	Briefly explain the following pulse modulation with waveform: i) PAM ii) PWM iii) PPM.(CO4)		
4	With neat block diagram, explain the generation of PAM wave. (CO4)		
5	Explain the following terms:		

	i) Under sampling ii) Over sampling iii) Nyquist rate. (CO4)
6	Discuss bandwidth-noise trade-off. (CO4)
7	What are the advantages of digital signal over analog signal. (CO4)
8	Explain the generation and detection of PPM waves. (CO4)
9	Describe the effect of noise in pulse position modulation. (CO4)
10	Explain the concept of TDM system with block diagram. (CO4)

MODULE – V

Title:	Noise & Baseband Transmission of Digital signals	Appr. Time:	8 Hrs
MO:			RBT
At the end of the Module, the student will be able to:			
1Explain how noise interferes with signals both before and after they reach the receiver.			L2
2 Describe different types of noise			L2
3 Explain why multiplexing techniques are necessary in telemetry, telephone systems, radio and TV broadcasting, and Internet access			L2
4 Compare frequency division multiplexing with time-division multiplexing			L2
5 Trace the steps in the transmission and reception of multiplexed signals.			L2

Lesson Schedule:

Lecture No.	Portion to be covered	CO
1	Introduction	CO5
2	Intersymbol Interference, Eye Pattern,	CO5
3	Nyquist criterion for distortionless Transmission,	CO5
4	Baseband M-ary PAM Transmission.	CO5
5	Signal to Noise Ratio	CO1
6	External Noise,	CO1
7	Internal Noise,	CO1
8	Expressing Noise Levels, Noise in Cascade Stages.	CO1

Application Areas:

- Telephone Network

Review Questions / Questions Appeared in the Previous Years (CO):

1	Explain signal to noise ratio?(CO1)
2	Explain different types of noise.(CO1)
3	An RF amplifier has an S/N ratio of 8 at the input and an S/N ratio of 6 at the output. What are the noise factor and noise figure?(CO1)
4	Explain Intersymbol interference with block diagram(CO5)
5	With a neat diagram explain eye diagram.(CO5)
6	State and explain Nyquist criterion for distortionless Transmission(CO5)
7	Explain how to overcome the practical difficulties encountered with ideal Nyquist channel.(CO5)
8	Explain Baseband M-ary PAM Transmission.(CO5)
9	State Intersymbol interference (CO5)
10	Explain ideal Nyquist channel(CO5)

F. LABORATORY CONTENTS:

Expt No.	Title of the Experiments	RBT	CO
1	Basic Signals and Signal Graphing: a) unit Step, b) Rectangular, c) standard triangle d) sinusoidal and e) Exponential signal.	L3	CO2,3
2	Illustration of signal representation in time and frequency domains for a rectangular pulse.	L3	CO2,3
3	Amplitude Modulation and demodulation: Generation and display the relevant signals and its spectrums.	L3	CO2
4	Frequency Modulation and demodulation: Generation and display the relevant signals and its spectrums.	L3	CO3
5	Sampling and reconstruction of low pass signals. Display the signals and its spectrum.	L3	CO4
6	Time Division Multiplexing and demultiplexing	L3	CO3
7	PCM Illustration: Sampling, Quantization and Encoding	L3	CO4
8	Generate a)NRZ, RZ and Raised cosine pulse, b) Generate and plot eye diagram	L3	CO4
9	Generate the Probability density function of Gaussian distribution function.	L3	CO1
10	Display the signal and its spectrum of an audio signal.	L3	CO2,3
11	Open ended experiment – 1	L3	
12	Open ended experiment – 2	L3	

EXPERIMENTS

1. EXPERIMENT NO:1

2. TITLE: BASIC SIGNALS AND SIGNAL GRAPHING: A) UNIT STEP, B) RECTANGULAR, C) STANDARD TRIANGLE D) SINUSOIDAL AND E) EXPONENTIAL SIGNAL.

3. LEARNING OBJECTIVES:
- To write and simulate a MATLAB program for various basic signal and display the signal.

4. AIM:
- To simulate program for various basic signal
- To display the signal.

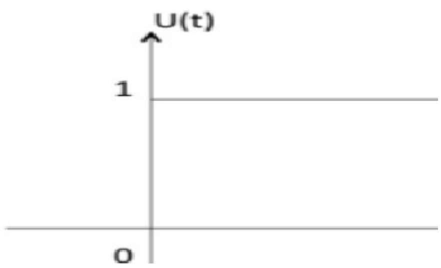
5. MATERIAL / EQUIPMENT REQUIRED:
- MATLAB software

6. THEORY :

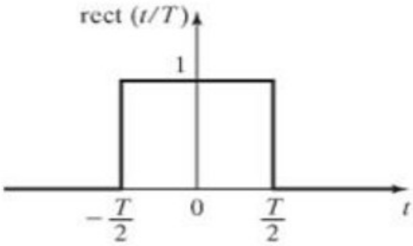
1) Unit Step Function

Unit step function is denoted by u(t). It is defined as

$$u(t) = \begin{cases} 1 & t \geqslant 0 \\ 0 & t < 0 \end{cases}$$

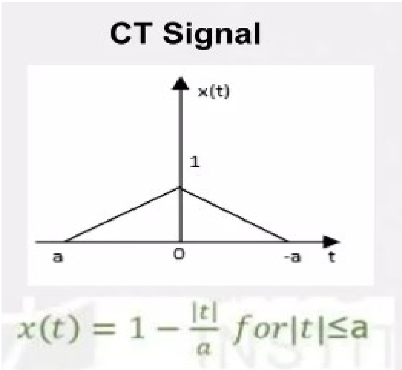


2) Rectangular Function



$$\mathrm{rect}(t/T) = \begin{cases} 1, & -T/2 < t < T/2 \\ 0, & \text{otherwise} \end{cases}$$

3) Triangular Function



4) Sinusoidal Function

Sinusoidal are the most smooth signals with no abrupt variation in their amplitude, the amplitude witnesses gradual change with time. Sinusoidal signals can be defined as a periodic signal with waveform as that of a sine wave. The amplitude of sine wave increase from a value of 0 at 0° angle to a maximum value of 1 at 90° , it further reaches its minimum value of -1 at 270° and then return to 0 at 360° . After any angle greater than 360° , the sinusoidal signal repeats the values so we can say that time period of sinusoidal signal is 2π i.e. 360°.

If we consider a sinusoidal signal x(t) having an amplitude A, frequency f, and phase of quantity then we can represent the signal as

$$x(t)=A \sin(2\pi ft+ \theta)$$

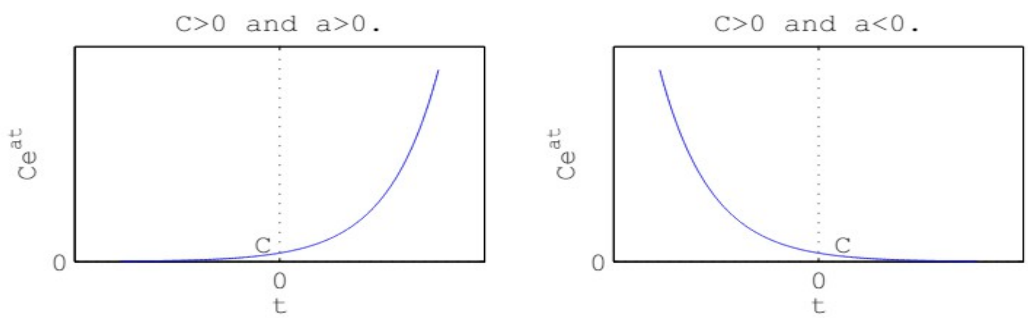
5) Exponential Function

Continuous-time complex exponential and sinusoidal signals:

$$x(t) = Ce^{at}$$

where C and a are in general complex numbers.

Real exponential signals: C and a are reals.



7. FORMULA / CALCULATIONS:

8. PROCEDURE / PROGRAMME / ACTIVITY:

```
%Generating a unit step sequence
clc;clear all;close all;
t1=-10:0.01:0;
t2=0:0.01:20;
t=[t1 t2];
x1=zeros(size(t1));
x2=ones(size(t2));
x=[x1 x2];
subplot(5,1,1);
plot(t,x);
axis([-10 20 0 2]);
xlabel('n'),ylabel('x(n)');
title('Unit Step Sequence')
% generate rectangular pulse
t3=0:0.01:10;
t4=10:0.01:20;
rec=[t1 t3 t4];
x1=zeros(size(t1));
x3=ones(size(t3));
x4=zeros(size(t4));
x5 = [x1 x3 x4];
subplot(5,1,2)
plot(rec,x5);
axis([-10 20 0 2]);
xlabel('Time');ylabel('Amplitude');
title('rectangular Sequence');
% generate triangular signal
T = 10*(1/50);
fss = 1000;
tt = 0:1/fss:T-1/fss;
xxx = sawtooth(2*pi*50*tt,1/2);
subplot(5,1,3);
plot(tt,xxx);
xlabel('Time');ylabel('Amplitude');
title('Triangular');
grid on
% Generation of a sinusoidal sequence
Fs=1000;
f=50;
tn=[0:100];
xx=sin(2*pi*f*tn/Fs);
subplot(5,1,4)
plot(tn,xx);
title('Sinusoidal Sequence');
xlabel('samples');
```

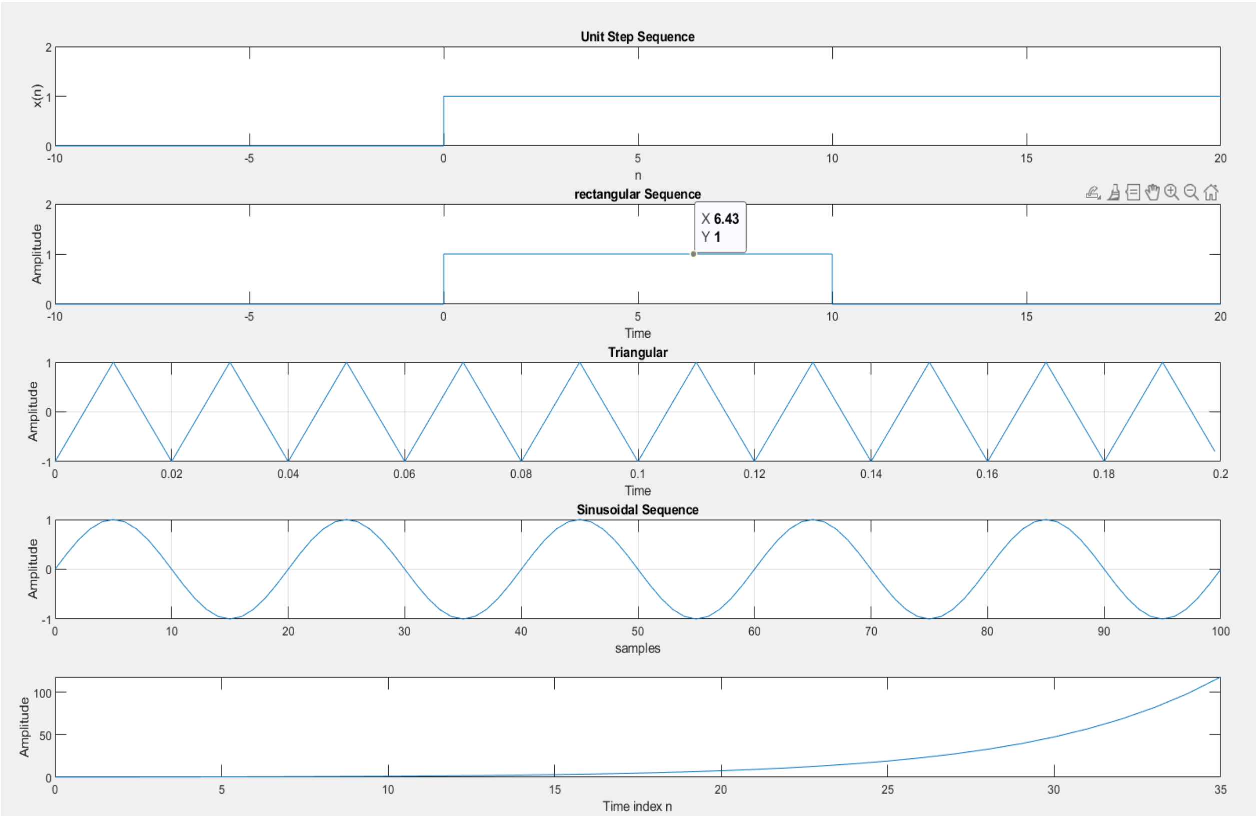
```
ylabel('Amplitude');
grid;
% Generation of a real exponential sequence
n = 0:35; a = 1.2; K = 0.2;
xn = K*a.^n;
subplot(5,1,5)
plot(n,xn);
title('Exponential Sequence');
xlabel('Time index n');
ylabel('Amplitude');
```

9. BLOCK / CIRCUIT / MODEL DIAGRAM / REACTION EQUATION:

-

10. OBSERVATION TABLE / LOOKUP TABLE / TRUTH TABLE:

11. GRAPHS / OUTPUTS:



12. RESULTS & CONCLUSIONS:

- MATLAB programs for various basic signal written and simulated.

13. LEARNING OUTCOMES :

- Waveforms of various basic signal is observed.

14. APPLICATION AREAS:

- Signals are serving as data in communication signals.

- Radar systems
- Medical imaging

15. REMARKS:

- -
- -
- -

-

1. EXPERIMENT NO:2

2. TITLE: **ILLUSTRATION OF SIGNAL REPRESENTATION IN TIME AND FREQUENCY DOMAINS FOR A RECTANGULAR PULSE.**

3. LEARNING OBJECTIVES:

- To represent rectangular pulse in time and frequency domain.

4. AIM:

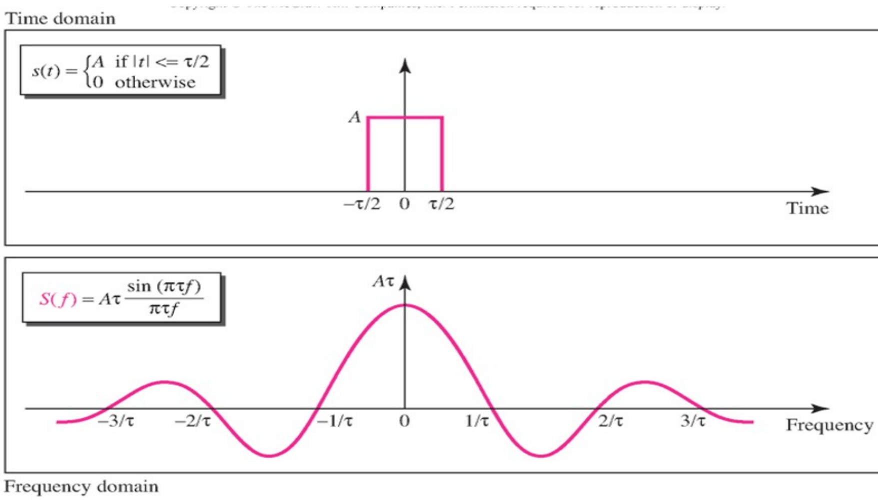
- To illustrate rectangular pulse in time and frequency domain.

5. MATERIAL / EQUIPMENT REQUIRED:

- MATLAB software

6. THEORY / HYPOTHESIS:

- The time domain graph depicts how the signal varies with respect to time. In this time-domain graph, the magnitude of the signal is represented at each instant time. If it is a continuous system, the signal will be represented as continuous. If it is a discrete system, the magnitude of the signal will be represented in distinct time intervals. The Cathode Ray Oscilloscope(CRO) device is the common device used to analyse the signal in a time domain.
- The frequency domain graph shows how much of a signal lies within each frequency domain. Here in the frequency domain, the signal is represented by the sum of sinusoidal waves of different frequencies and each with a certain amplitude. Frequency domain analysis is very common in Control system Engineering. Thus, this refers to analyzing the signals with respect to the frequency of each signal.



fft: Transform time-domain data to the frequency domain.
Ifft: Transform frequency-domain data to the time domain.

7. FORMULA / CALCULATIONS:

-

8. PROCEDURE / PROGRAMME / ACTIVITY:

```
clear;clc % clear command history and all variables
T = 20; % tunable parameter for the signal width
dt=.001; % increment
t=[-(10+T):dt:(10+T)]; % range of the signal
x=sign(t+T)-sign(t-T); % generate the rectangular pulse signal
subplot(2,1,1);
plot(t,x); % visualize the signal in time domain
title('Pulse signal'); % title of the plot
xlabel('Time (msec)'); % label x-axis
ylabel('Signal f(t)'); % label y-axis
axis([-(30+T) (30+T) 0 3]); % set display range of x- and y-axis
y=fftshift(fft(x)); % apply Fourier transform and move zero frequency component to the center
N=length(y); % measure frequency range
n=-(N-1)/2:(N-1)/2; % evenly divide frequency range around zero frequency
f=sqrt(y.*conj(y)); % calculate amplitude of the frequency signal
subplot(2,1,2);
plot(n,f); % visualize the signal in time domain
title('Frequency spectrum amplitude for the rectangular pulse');
% set title of the plot
xlabel('Frequency (Hz)'); % label x-axis
ylabel('Frequency spectrum amplitude'); % label y-axis
axis([-50 50 0 70000]); % set display range of x- and y-axis
```

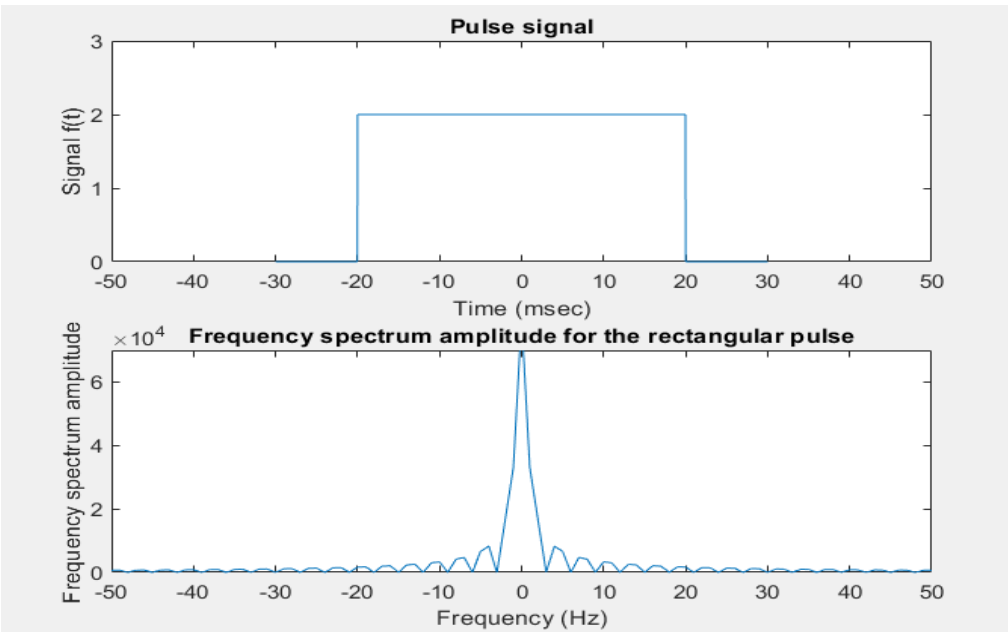
9. BLOCK / CIRCUIT / MODEL DIAGRAM / REACTION EQUATION:

-

10. OBSERVATION TABLE / LOOKUP TABLE / TRUTH TABLE:

-

11. GRAPHS / OUTPUTS:



COURSE PLAN

12. RESULTS & CONCLUSIONS:
- MATLAB program for time and frequency domain representation of rectangular pulse written and simulated.

13. LEARNING OUTCOMES :
- Waveforms of time and frequency domain representation of rectangular pulse is observed.

14. APPLICATION AREAS:
- Signal Processing

15. REMARKS:
- -

1. EXPERIMENT NO: 3

2. TITLE: AMPLITUDE MODULATION AND DEMODULATION: GENERATION AND DISPLAY THE RELEVANT SIGNALS AND ITS SPECTRUM.

3. LEARNING OBJECTIVES:
- To write and simulate a MATLAB program for amplitude modulation and demodulation and display the signal and its spectrum

4. AIM:
- To simulate program for amplitude modulation and demodulation.
- To display the signal and its spectrum.

5. MATERIAL / EQUIPMENT REQUIRED:
- MATLAB software

6. THEORY :
- Modulation is defined as the process of changing the characteristics (Amplitude, Frequency or Phase) of the carrier signal (high frequency signal) in accordance with the intensity of the message signal (modulating signal). Amplitude modulation is defined as a system of modulation in which the amplitude of the carrier is varied in accordance with amplitude of the message signal (modulating signal).
- The message signal is given by the expression. Em(t) =Em cosWmt
- Where Wm is -----> Angular frequency
  Em --------◊ Amplitude
- Carrier voltage Ec(t)= Ec cosWct
- E(t)=Ec + KaEm cosWmt
  KaEm cosWmt -----◊ change in carrier amplitude
  Ka-----◊ constant
- The amplitude modulated voltage is given by E=E(t) cosWct From above two equations
- E= (Ec+KaEm cosWmt) cosWct.
  E= (1+KaEm/Ec cosWmt) Ec cosWct
  E= Ec(1+Ma cosWmt)cosWct
- Where Ma-----◊ depth of modulation/ modulation index/modulation factor Ma=KaEm/Ec 100* Ma gives the percentage of modulation.

7. FORMULA / CALCULATIONS:
- -

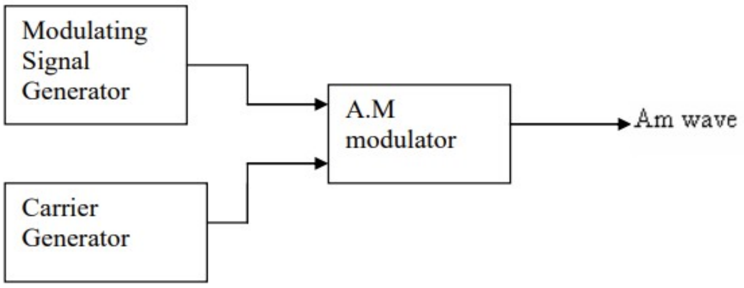
8.PROGRAMME :

```
fm=20;
fc=500;
```

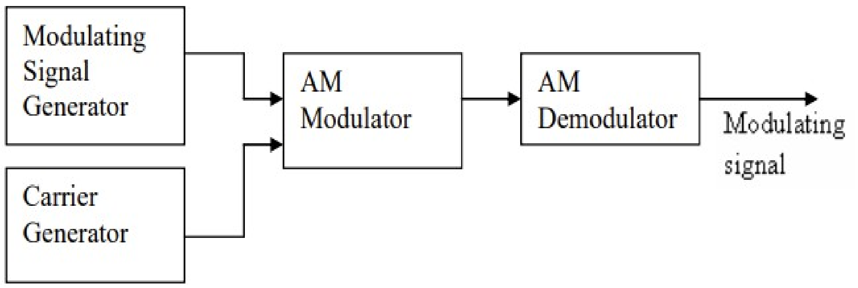
```
vm=1;
vc=1;
mu=0.8; % modulation index
% x-axis:Time(second)
t=0:0.00001:0.0999;
f=0:1:9999;
% y-axis:Voltage(volt)
V1=vc+mu*sin(2*pi*fm*t); %upper envelop
V2=-(vc+mu*sin(2*pi*fm*t)); % lower envelop
Vm=vm*sin(2*pi*fm*t); Vc=vc*sin(2*pi*fc*t);
Vam=vc*(1+mu*sin(2*pi*fm*t)).*(sin(2*pi*fc*t)); %%AM signal
Vr=Vam.*Vc; % Synchronous detector
Vf=abs(fft(Vam,10000))/500; % Spectrum
[b a] = butter(3,0.002);
out= filter(b,a,Vr);
%Plot modulating,carrier signal
figure(1);
subplot(211)
plot(t,Vm);
title('AM modulating signal');
xlabel('time'), ylabel('amplitude');
grid on;
subplot(212) plot(t, Vc);
title('AM carrier signal');
xlabel('time'), ylabel('amplitude');
grid on; % Plot AM in time domain and Frequency domain
figure(2);
subplot(211) plot(t,Vam);
hold on;
plot(t,V1,'r'),plot(t,V2,'r');
title('AM waveform time-domain');
xlabel('time'), ylabel('amplitude');
grid on;
subplot(212)
plot(f*10,Vf);
axis([(fc-6*fm) (fc+6*fm) 0 10]);
title('AM waveform frequency-domain');
xlabel('frequency'), ylabel('amplitude');
grid on;
figure(3)
plot(t,1.81*out) t
title('AM Demodulated signal');
xlabel('time'), ylabel('amplitude');
grid
```

9. BLOCK DIAGRAM :

Modulation



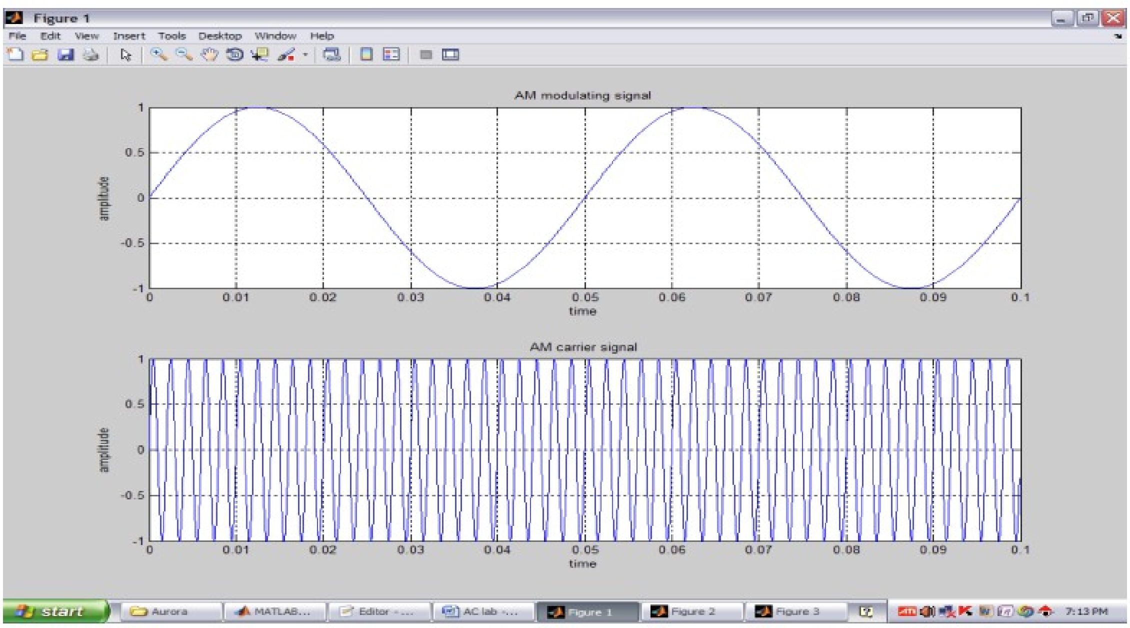
Demodulation



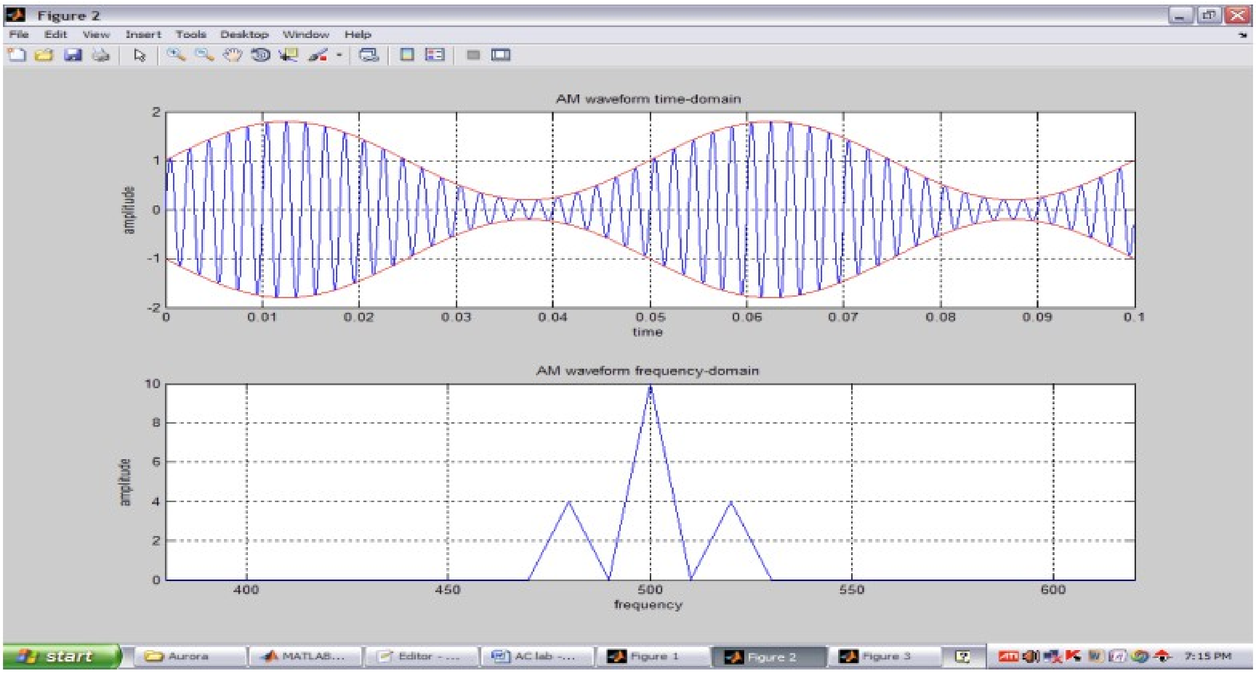
10. OBSERVATION TABLE / LOOKUP TABLE / TRUTH TABLE:

11. GRAPHS / OUTPUTS:

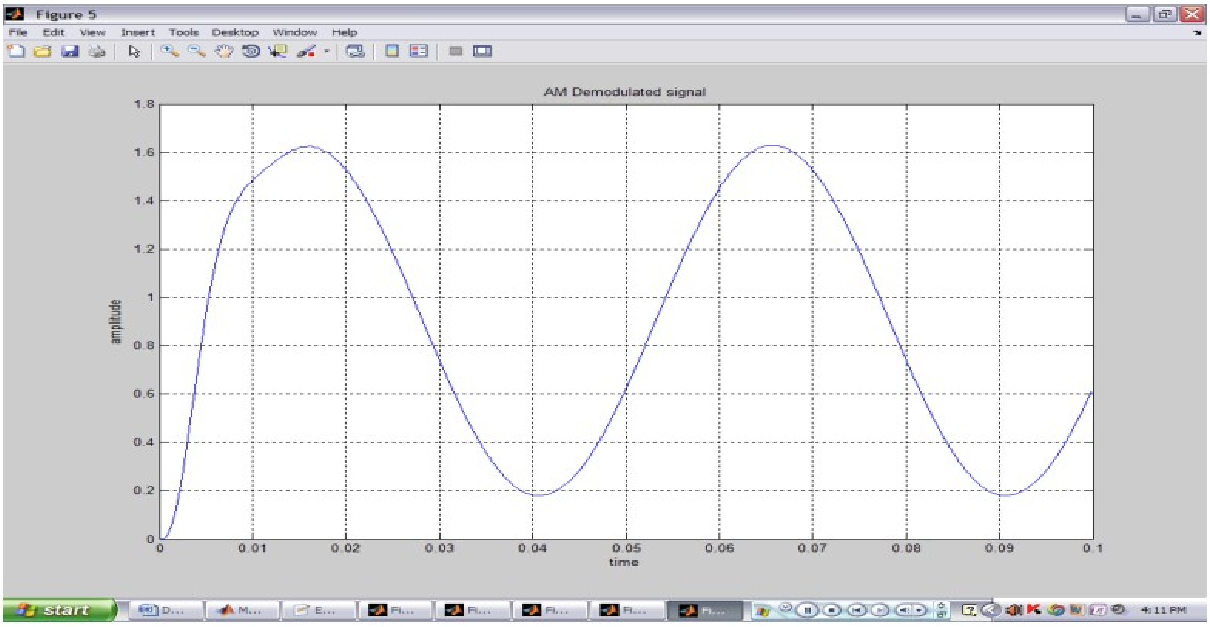
1) Message and carrier signal



2) Amplitude Modulated signal and frequency spectrum



3) Demodulated signal



12. RESULTS & CONCLUSIONS:

- The Amplitude modulation and demodulation is simulated and output waveforms are observed.

13. LEARNING OUTCOMES :

- Students are able to simulate the amplitude modulation and demodulation system using MATLAB SOFTWARE.

14. APPLICATION AREAS:

- Broadcast transmissions
- Air band radio
- Cellular telecommunications

15. REMARKS:

- -
- -
- -

-

1. EXPERIMENT NO: 4

2. TITLE: FREQUENCY MODULATION AND DEMODULATION: GENERATION AND DISPLAY THE RELEVANT SIGNALS AND ITS SPECTRUMS.

3. LEARNING OBJECTIVES:

- To write and simulate a MATLAB program for frequency modulation and demodulation and display the signal and its spectrum

4. AIM:

- To simulate program for frequency modulation and demodulation.
- To display the signal and its spectrum.

5. MATERIAL / EQUIPMENT REQUIRED:

- MATLAB software

6. THEORY :

- FM is a system in which the amplitude of the modulated carrier is kept constant, while its frequency and rate of change are varied by the modulating signal. By the definition of FM, the amount by which the carrier frequency is varied from its unmodulated value, called the deviation, is made proportional to the instantaneous amplitude of the modulating voltage. The rate at which this frequency variation changes or takes place is equal to the modulating frequency. FM is that form of angle modulation in which the instantaneous frequency fi(t) is varied linearly with the message signal m(t), as
  $f_i(t) = f_C + k_f\, m(t)$
  The term fc represents the frequency of the unmodulated carrier, and the constant Kf represents the frequency sensitivity of the modulator expressed in Hertz per volt. Unlike AM, the spectrum of an FM signal is not related in a simple manner to that of modulating signal, rather its analysis is much more difficult than that of an AM signal

7. FORMULA / CALCULATIONS:

- -
- -

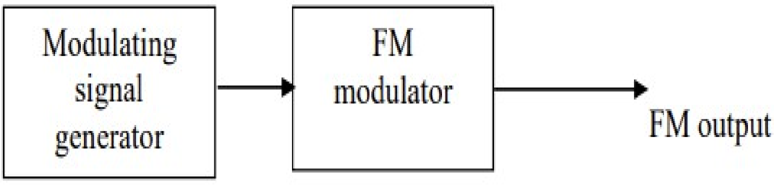
8. PROGRAMME :

- ```
  %The frequency modulation(FM)waveform in time and frequency domain.
  %fm=250HZ,fc=5KHZ,Vm=1V,Vc=1V,m=10,t=0:0.00001:0.09999
  vc=1; % Carrier amplitude
  vm=1; % Message Amplitude
  fm=250; % Message frequency
  fc=5000; % Carrier frequency
  m=10; % beta
  ```
  ```

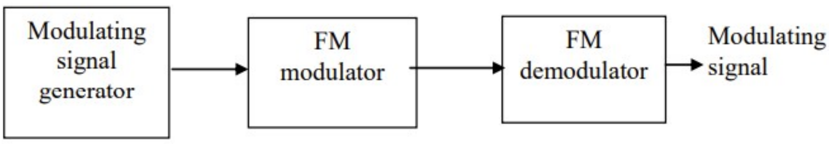
```
kf=100
t=0:0.00001:0.09999;
f=0:10:99990;
carrier=vc*cos(2*pi*fc*t);
message=vm*cos(2*pi*fm*t);
FM=vc*cos((2*pi*fc*t)+10*sin(2*pi*fm*t)); % FM wave
dem=(1/2*pi*10)*diff(10*sin(2*pi*fm*t))/0.2; %% Demodulation using PLL
vf=(fft(FM,10^4))/500;
figure(1);
subplot(211);plot(t,message);
xlabel('Time');ylabel('Amplitude');
title('FM modulating signal')grid
subplot(212);plot(t,carrier);
xlabel('Time');ylabel('Amplitude')
axis([0 0.01 -1.5 1.5]);
title('FM carrier signal');grid
figure(2);
subplot(211);plot(t,FM);
hold on;
plot(t,message,'r');
axis([0 0.01 -1.5 1.5]);
xlabel('time(second)'),ylabel('amplitude');
title('FM time-domain');grid
grid on;
subplot(212) plot(f,vf);
axis([ 0 10^4 0 4]);
xlabel('frequency'), ylabel('amplitude');
title('FM frequency-domain');
grid on;
figure(3) plot(dem)
xlabel('Time'),ylabel('amplitude');
title('FM Demodulated signal');grid
grid on;
```

9. BLOCK DIAGRAM :

Modulation



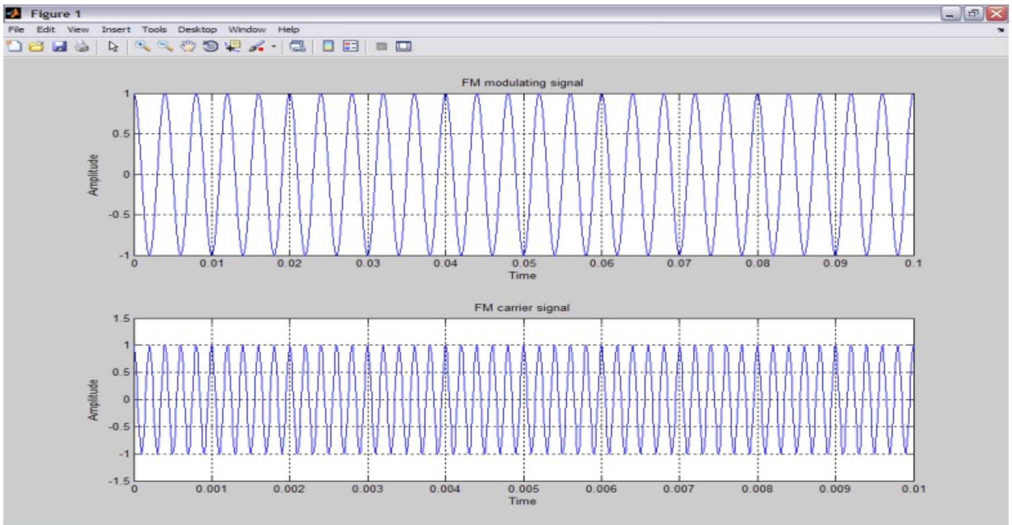
Demodulation

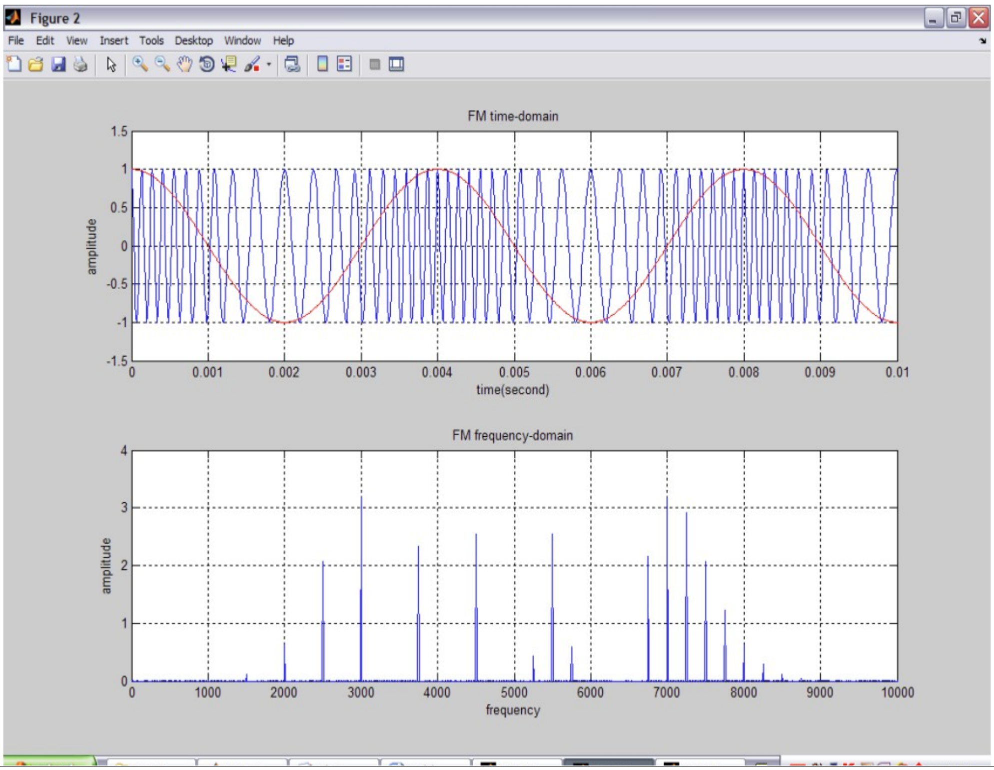


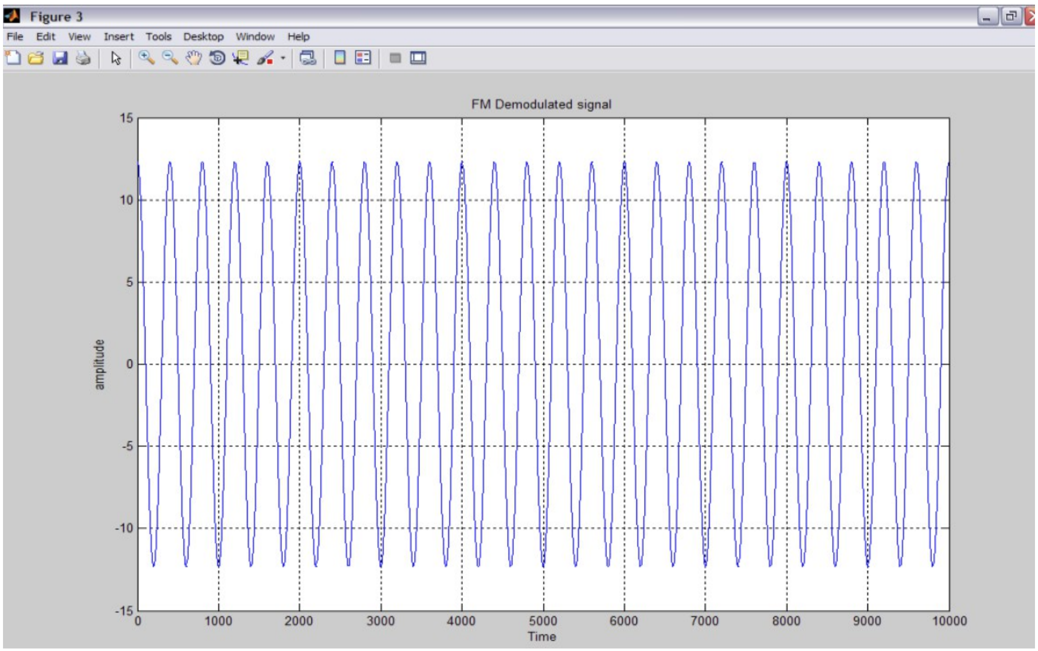
10. OBSERVATION TABLE / LOOKUP TABLE / TRUTH TABLE:

11. GRAPHS / OUTPUTS:

1) Modulating and carrier signal







12. RESULTS & CONCLUSIONS:

- The FM modulation and demodulation is simulated and output waveforms are observed.

13. LEARNING OUTCOMES :

- Students are able to simulate the fm modulation and demodulation system using MATLAB SOFTWARE.

14. APPLICATION AREAS:

- FM radio
- Satellite TV
- TV broadcast

15. REMARKS:

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- -
- -

-

1. EXPERIMENT NO: 5

2. TITLE: **SAMPLING AND RECONSTRUCTION OF LOW PASS SIGNALS. DISPLAY THE SIGNALS AND ITS SPECTRUMS**

3. LEARNING OBJECTIVES:

- To write and simulate a MATLAB program for the process of sampling, reconstruction of low pass signals and display the signal and its spectrum

4. AIM:

- To simulate program for he process of sampling, reconstruction of low pass signals.
- To display the signal and its spectrum.

5. MATERIAL / EQUIPMENT REQUIRED:

- MATLAB software

6. THEORY :

- A band limited signal of finite energy which has no frequency components higher than fm Hz, is completely described by specifying the values of the signal at instants of time separated by ½ fm seconds. The sampling theorem states that, if the sampling rate in any pulse modulation system exceeds twice the maximum signal frequency, the original signal can be reconstructed in the receiver with minimum distortion. Fs > 2fm is called Nyquist rate. Where fs – sampling frequency Fm – Modulation signal frequency.
- If we reduce the sampling frequency fs less than fm, the side bands and the information signal will overlap and we cannot recover the information signal simply by low pass filter. This phenomenon is called fold over distortion or aliasing.There are two methods of sampling. (1) Natural sampling (2) Flat top sampling. Sample & Hold circuit holds the sample value until the next sample is taken. Sample & Hold technique is used to maintain reasonable pulse energy.The duty cycle of a signal is defined as the ratio of Pulse duration to the Pulse repetition period. The duty cycle of 50% is desirous taking the efficiency into account.

7. FORMULA / CALCULATIONS:

- -
- -

8. PROCEDURE / programme / ACTIVITY:

```
clc;
clear all;
t=-5:0.0001:5;
f1=10;
% Continuous time Signal
x=cos(2*pi*f1*t);
figure(1)
plot(t,x)
axis([-0.5 0.5 -2 2])
xlabel('time');
ylabel('amplitude')
title('continuous time signal');
% Discrete time Signal: Under Sampling Condition fs<2fm
fs1=0.5*f1;
ts1=1/fs1;
n1=-0.5:ts1:0.5;
xs1=cos(2*pi*f1*n1);
figure(2)
stem(n1,xs1)
axis([-0.5 0.5 -2 2])
hold on;
plot(t,x,'r');
hold off
xlabel('time sample');
ylabel('Amplitude');
title('Discrete time signal sampling rate fs<2*fmax');
```

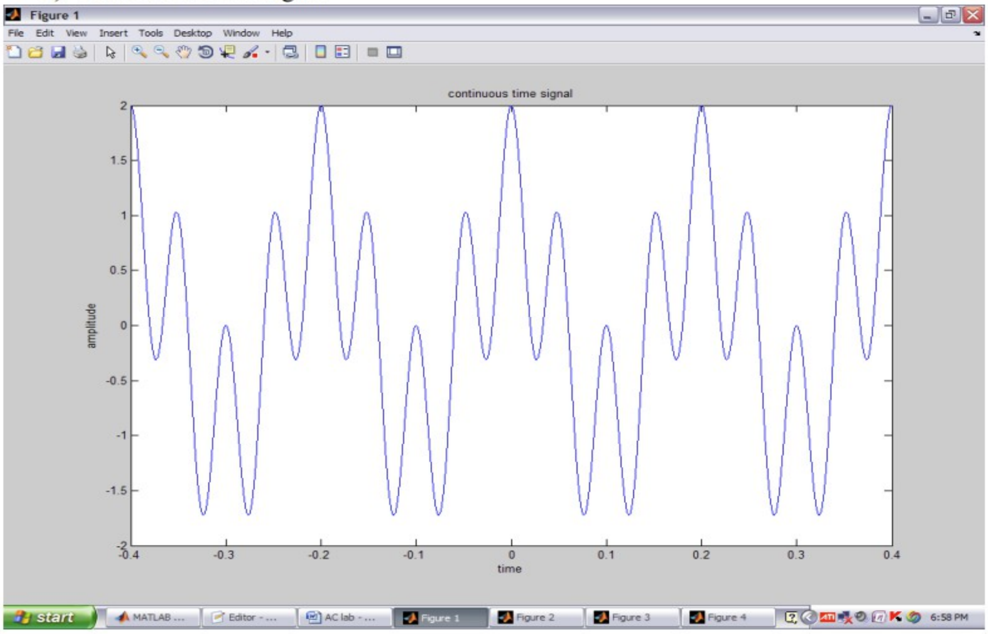
```
% Discrete time Signal: Nyquist Sampling Condition fs=2fm
fs2=2*f1;
ts2=1/fs2;
n2=-0.5:ts2:0.5;
xs2=cos(2*pi*f1*n2);
figure(3)
stem(n2,xs2)
axis([-0.5 0.5 -2 2])
hold on;
plot(t,x,'r');
hold off
xlabel('time sample');
ylabel('Amplitude');
title('Discrete time signal sampling rate fs=2*fmax');
% Discrete time Signal: Over Sampling Condition fs>2fm
fs3=20*f1;
ts3=1/fs3;
n3=-0.5:ts3:0.5;
xs3=cos(2*pi*f1*n3);
figure(4)
stem(n3,xs3)
axis([-0.5 0.5 -2 2])
hold on;
plot(t,x,'r');
hold off
xlabel('time sample');
ylabel('Amplitude');
title('Discrete time signal sampling rate fs>2*fmax');
% Reconstruction of Original signal
n11=-0.5:(1/fs3):0.5;
r1=interp1(n3,xs3,n11);
[b a] = butter(3,0.002);
out= filter(b,a,r1);
figure(5)
axis([-0.5 0.5 -2 2])
hold on;
plot(n11,r1,'r');
hold off
xlabel('time sample');
ylabel('Amplitude');
title('Reconstructed signal');
```

9. BLOCK / CIRCUIT / MODEL DIAGRAM / REACTION EQUATION:

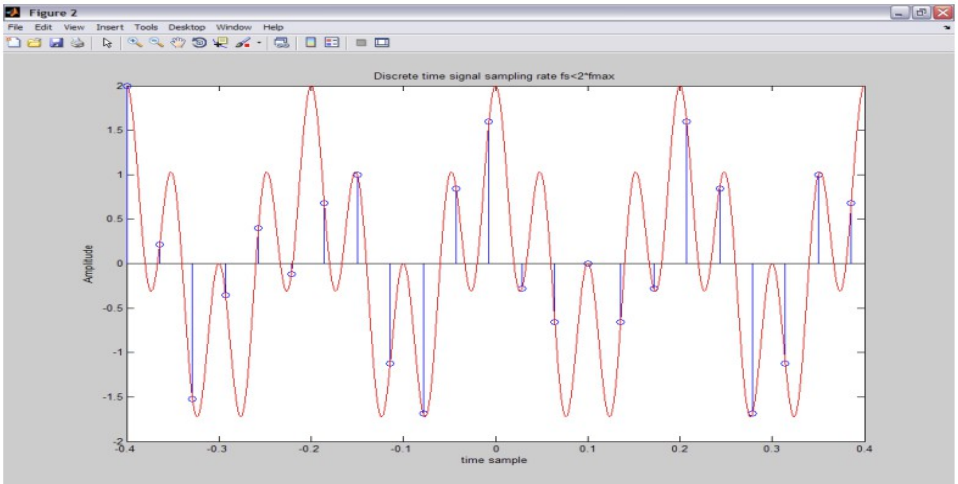
10. OBSERVATION TABLE / LOOKUP TABLE / TRUTH TABLE:

11. GRAPHS / OUTPUTS:

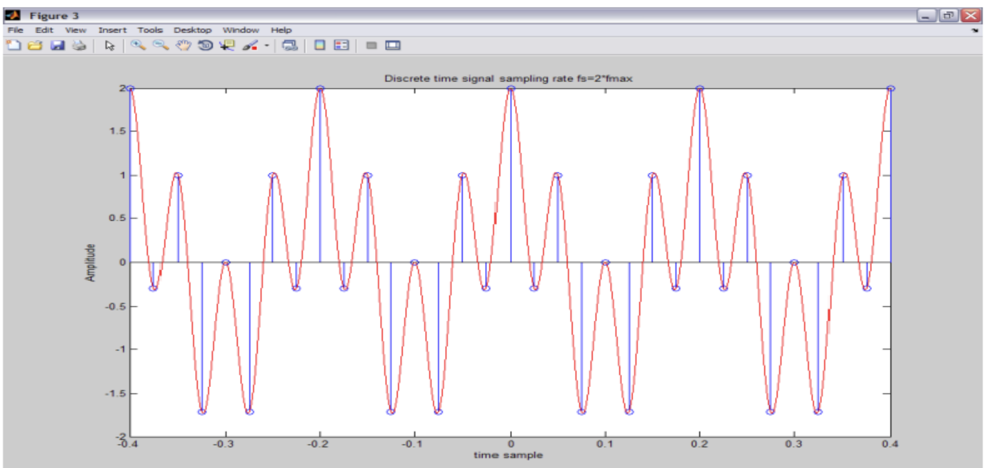
1) Continuous Time Signal



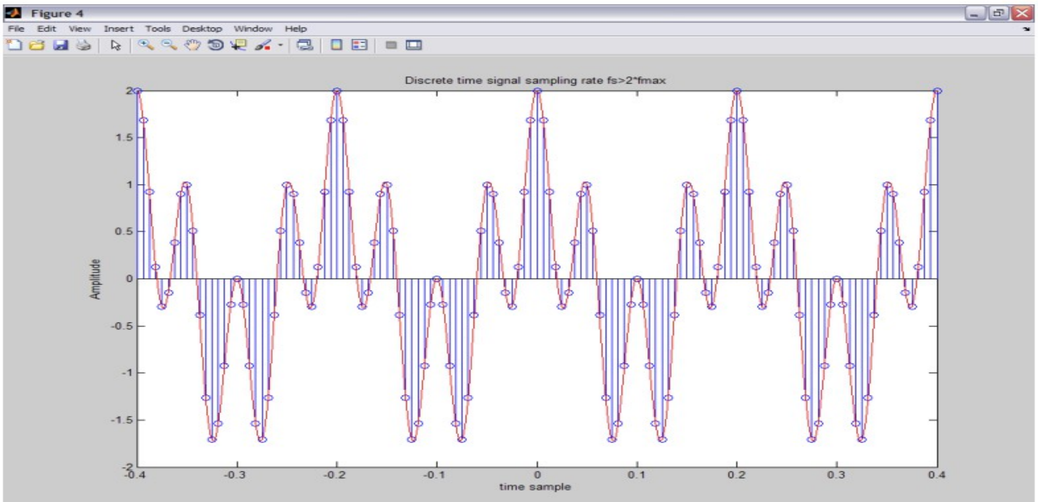
2) Discrete Time signal fs<2*fmax



3) Discrete Time signal fs=2*fmax



4) Discrete Time signal fs>2*fmax



12. RESULTS & CONCLUSIONS:
- The sampling and reconstruction of low pass signal is simulated and output waveforms are observed.

13. LEARNING OUTCOMES :
- Students are able to simulate the sampling and reconstruction of low pass signal using MATLAB SOFTWARE.

14. APPLICATION AREAS:
- Digital hardware, including computers,

15. REMARKS:
- -
- -
- -

-

1. EXPERIMENT NO:6

2. TITLE: **TIME DIVISION MULTIPLEXING AND DEMULTIPLEXING.**

3. LEARNING OBJECTIVES:
- To study the concept of time division multiplexing and demultiplexing
- To simulate TDM with two different signal in MATLAB

4. AIM:
- To Study and verify the process of time division multiplexing & demultiplexing with two different signals.

5. MATERIAL / EQUIPMENT REQUIRED:
- MATLAB Software

6. THEORY / HYPOTHESIS:
- The Sampling Theorem provides the basis for transmitting the information contained in a band limited message signal m (t) as a sequence of samples of m (t) taken uniformly at a

rate that is usually slighter higher than the nyquist rate. An important feature of the sampling process is a conservation of time. That is, the transmission the message samples engages the communication channel s for only a fraction of the sampling interval on a periodic basis, and in this way some of the time interval between adjacent samples is cleared for use by other independent message sources on a time shared basis.

- **Multiplexing:** It is the process of combining signals from different information sources so that they can be transmitted over a common channel. Multiplexing is advantageous in cases where it is impracticable and uneconomical to provide separate links for the different information sources. The price that has to be paid to acquire this advantage is in the form of increased system complexity and bandwidth. Most commonly used methods of multiplexing are 1. Frequency division multiplexing (FDM) 2. Time division multiplexing (TDM)
- **Time Division Multiplexing:** Time division multiplexing is the process of combining the samples from different information signals, in time domain so that they can be transmitted over the same channel. The fact utilized in TDM technique is that there are large intervals between the message samples. The samples from the other sources can be placed within these time intervals. Thus every sample is separated from other in time domain. The time division multiplexing system can be simulated by two rotating switches, one at transmitter and the other at receiver. The two wipers rotate and establish electrical contact with one channel at a time. Each signal is sampled over one sampling interval and transmitted one after the other along a common channel. Thus part of message 1 is transmitted first followed by part of message 2 and then again message 1 so on. It can be anticipated from above process that the To establish synchronization, the receiver needs to know:
  a. Frequency/ rate of operation at transmitter.
  b. Sample identification. This increases the transmitter and receiver complexity and cost.

7.FORMULA / CALCULATIONS:

- -
- -

8. PROCEDURE / PROGRAMME / ACTIVITY:

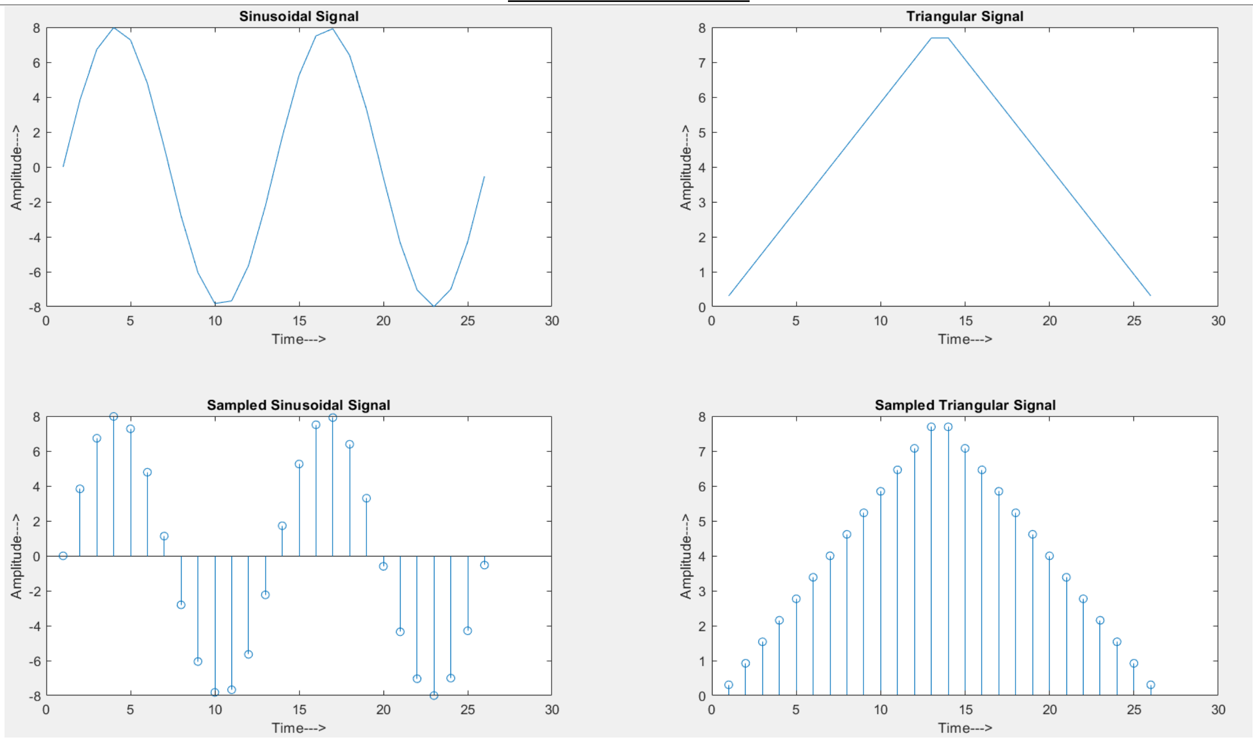
```
clc;
close all;
clear;
% Signal generation
t =0:.5:4*pi; % Signal taken upto 4pi
sig1=8*sin(t); % Generate first signal
l1=length(sig1);
sig2=8*triang(l1); % Generate Second Signal
% Perform time division multiplexing
for i=1:l1
sig(1,i)= sig1(i); % Making both row vector to a matrix
%sig[(F1,S1) (F2,S2) (F3,S3).....]
sig(2,i)= sig2(i);
end
tdmsig=reshape(sig,1,2*l1); % TDM of both quantize signal
% Perform Demultiplexing
demux=reshape(tdmsig,2,l1);
for i=1:l1
demultdm1(i)=demux(1,i); % Converting The matrix into row vectors
demultdm2(i)=demux(2,i);
end
```

```
figure(1);
% Display First Signal
subplot(2,2,1);
plot(sig1);
title('Sinusoidal Signal');
ylabel('Amplitude--->');
xlabel('Time--->');
% Display Second signal
subplot(2,2,2);
plot(sig2);
title('Triangular Signal');
ylabel('Amplitude--->');
xlabel('Time--->');
% Display Sampled version of First Signal
subplot(2,2,3);
stem(sig1);
title('Sampled Sinusoidal Signal');
ylabel('Amplitude--->');
xlabel('Time--->');
% Display Sampled version of Second Signal
subplot(2,2,4);
stem(sig2);
title('Sampled Triangular Signal');
ylabel('Amplitude--->');
xlabel('Time--->');
figure(2);
% Display TDM SIgnal
subplot (3,1,1);
stem(tdmsig);
title('TDM Signal');
ylabel('Amplitude--->');
xlabel('Time--->');
% Display of demultiplexed signal
subplot(3,1,2);
plot(demultdm1);
title('Recovered Sinusoidal Signal');
ylabel('Amplitude--->');
xlabel('Time--->');
subplot(3,1,3);
plot(demultdm2);
title('Recovered Triangular Signal');
ylabel('Amplitude--->');
xlabel('Time--->');
```

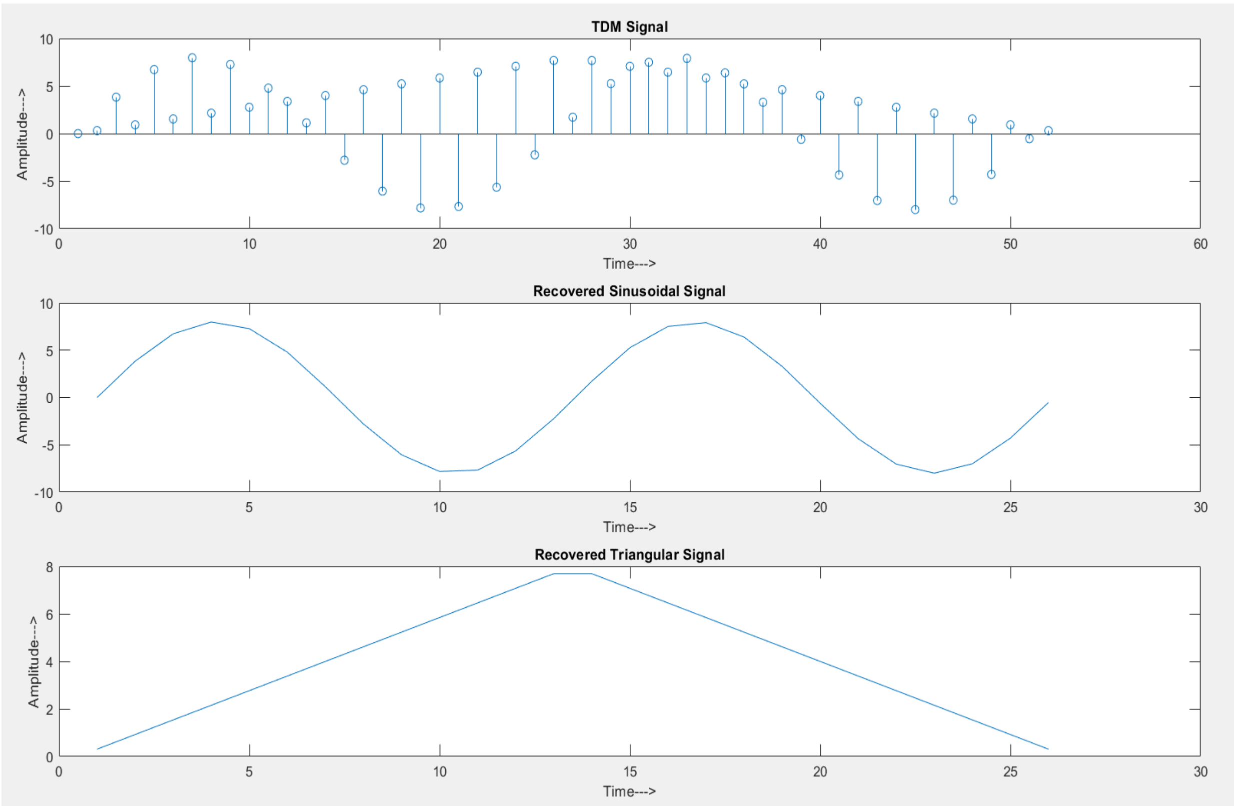
9. BLOCK / CIRCUIT / MODEL DIAGRAM / REACTION EQUATION:

10. OBSERVATION TABLE / LOOKUP TABLE / TRUTH TABLE:

11. GRAPHS / OUTPUTS:



•



•

12. RESULTS & CONCLUSIONS:

- The Time division multiplexing and demultiplexing signal is simulated and output waveforms are observed.

13. LEARNING OUTCOMES :

- Students are able to simulate the time division multiplexing and demultiplexing signal using MATLAB SOFTWARE.

14. APPLICATION AREAS:
- Data Communication systems
-

15. REMARKS:
- -
- -
- -

-

1. EXPERIMENT NO:7

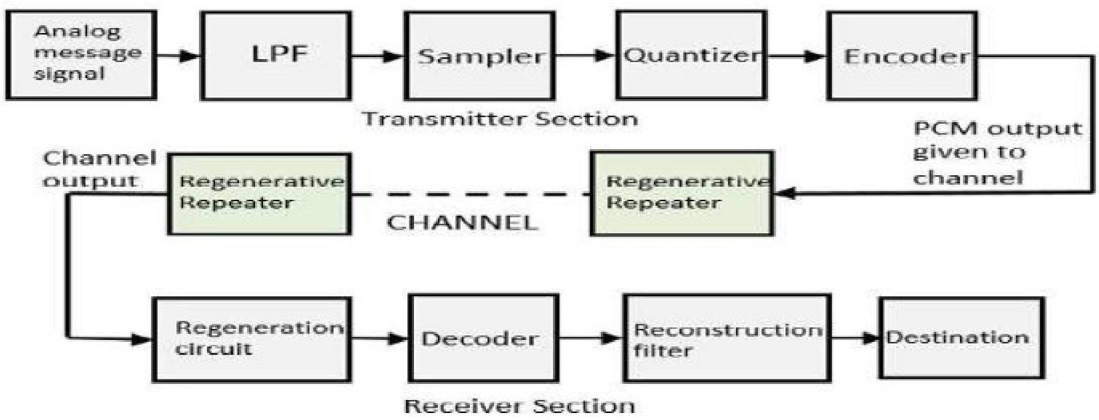
2. TITLE: **PCM ILLUSTRATION: SAMPLING, QUANTIZATION AND ENCODING**

3. LEARNING OBJECTIVES:
- To study the concept of Pulse code modulation
- To simulate Pulse code modulation in MATLAB

4. AIM:
- To Study and verify the PCM illustration with sampling, quantization and encoding.

5. MATERIAL / EQUIPMENT REQUIRED:
- MATLAB Software

6. THEORY / HYPOTHESIS:
- Pulse-code modulation (PCM) is a method used to digitally represent sampled analog signals. It is the standard form of digital audio in computers, compact discs, digital telephony and other digital audio applications. In a PCM stream, the amplitude of the analog signal is sampled regularly at uniform intervals, and each sample is quantized to the nearest value within a range of digital steps. To recover the original signal from the sampled data, a "demodulator" can apply the procedure of modulation in reverse. After each sampling period, the demodulator reads the next value and shifts the output signal to the new value. As a result of these transitions, the signal has a significant amount of high-frequency energy caused by aliasing. To remove these undesirable frequencies and leave the original signal, the demodulator passes the signal through analog filters that suppress energy outside the expected frequency range.



7. FORMULA / CALCULATIONS:

- -
- -

8. PROCEDURE / PROGRAMME / ACTIVITY:

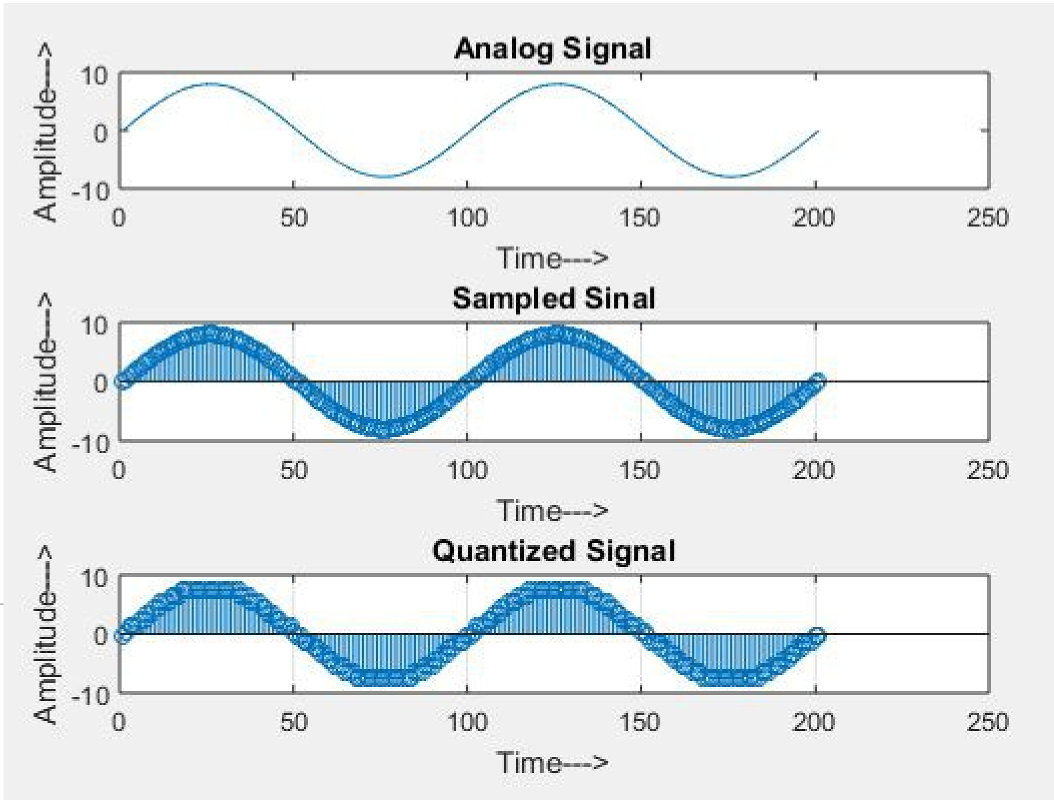
```
clc;
close all;
clear all;
n=input('Enter n value for n-bit PCM system : ');
n1=input('Enter number of samples in a period : ');
L=2^n;
% % Signal Generation
% x=0:1/100:4*pi;
% y=8*sin(x); % Amplitude Of signal is 8v
% subplot(2,2,1);
% plot(x,y);grid on;
% Sampling Operation
x=0:2*pi/n1:4*pi; % n1 nuber of samples have tobe selected
s=8*sin(x);
subplot(3,1,1);
plot(s);
title('Analog Signal');
ylabel('Amplitude--->');
xlabel('Time--->');
subplot(3,1,2);
stem(s);grid on; title('Sampled Sinal'); ylabel('Amplitude--->'); xlabel('Time--->');
% Quantization Process
vmax=8;
vmin=-vmax;
del=(vmax-vmin)/L;
part=vmin:del:vmax; % level are between vmin and vmax with difference of del
code=vmin-(del/2):del:vmax+(del/2); % Contaion Quantized valuses
[ind,q]=quantiz(s,part,code); % Quantization process
% ind contain index number and q contain quantized values
l1=length(ind);
l2=length(q);
for i=1:l1
if(ind(i)~=0) % To make index as binary decimal so started from 0 to N
ind(i)=ind(i)-1;
end
i=i+1;
end
for i=1:l2
if(q(i)==vmin-(del/2)) % To make quantize value inbetween the levels
q(i)=vmin+(del/2);
end
end
subplot(3,1,3);
stem(q);grid on; % Display the Quantize values
```

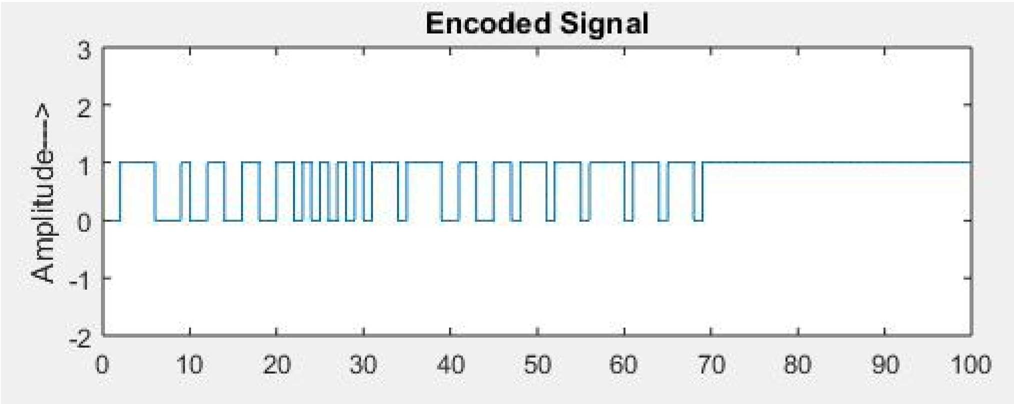
```
title('Quantized Signal');
ylabel('Amplitude--->');
xlabel('Time--->');
% Encoding Process
figure
code=de2bi(ind,'left-msb'); % Cnvert the decimal to binary
k=1;
for i=1:l1
for j=1:n
coded(k)=code(i,j); % convert code matrix to a coded row vector
j=j+1;
k=k+1;
end
i=i+1;
end
subplot(2,1,1); grid on;
stairs(coded); % Display the encoded signal
axis([0 100 -2 3]); title('Encoded Signal');
ylabel('Amplitude--->');
```

9. BLOCK / CIRCUIT / MODEL DIAGRAM / REACTION EQUATION:

10. OBSERVATION TABLE / LOOKUP TABLE / TRUTH TABLE:

11. GRAPHS / OUTPUTS:





12. RESULTS & CONCLUSIONS:

- The MATLAB program for Pulse code modulation is simulated and output waveforms are observed.

13. LEARNING OUTCOMES :

- Students are able to simulate the pulse code modulation using MATLAB SOFTWARE.

14. APPLICATION AREAS:

- Telephony and compact discs.
- Satellite transmission systems

15. REMARKS:

- -
- -
- -

-

1. EXPERIMENT NO:8

2. TITLE: **GENERATE A)NRZ, RZ AND RAISED COSINE PULSE, B) GENERATE AND PLOT EYE DIAGRAM**

3. LEARNING OBJECTIVES:

- To study about eye-diagram
- To generate NRZ, RZ and raised cosine pulse
- To plot eye diagram

4. AIM:

- To Simulate NRZ, RZ and raised cosine pulses and generate eye diagram.

5. MATERIAL / EQUIPMENT REQUIRED:

- MATLAB Software

6. THEORY / HYPOTHESIS:

- A line code is the code used for data transmission of a digital signal over a transmission line. This process of coding is chosen so as to avoid overlap and distortion of signal such as inter-symbol interference.

- Non-return-to-zero (NRZ). The pulse amplitude is held constant throughout the pulse or bit period.
- Return-to-zero (RZ). The pulse amplitude returns to a zerovolt level for a portion (usually one-half) of the pulse or bit period.
- In telecommunication, an eye pattern, also known as an eye diagram, is an oscilloscope display in which a digital signal from a receiver is repetitively sampled and applied to the vertical input, while the data rate is used to trigger the horizontal sweep. It is so called because, for several types of coding, the pattern looks like a series of eyes between a pair of rails. It is a tool for the evaluation of the combined effects of channel noise and intersymbol interference on the performance of a baseband pulse-transmission system. It is the synchronized superposition of all possible realizations of the signal of interest viewed within a particular signaling interval.

7. FORMULA / CALCULATIONS:

- -
- -

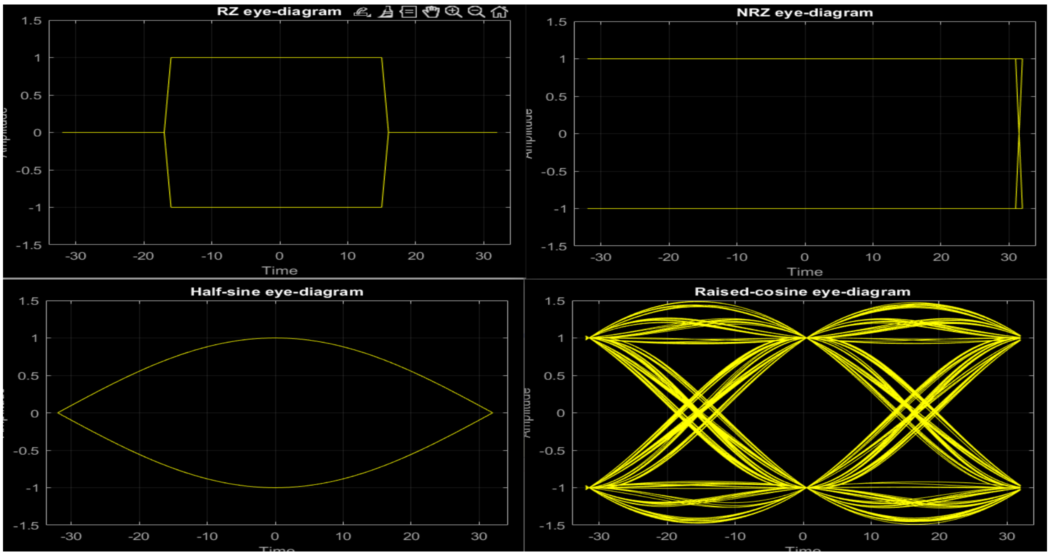
8. PROCEDURE / PROGRAMME / ACTIVITY:

```
clc;
close all;
clear all;
%Generate 400 random bits
data = sign(randn(1,400));
%Define the symbol period
T = 64;
Td = 32;
%Generate impulse train
dataup=upsample(data, T);
%Return to zero polar signal
yrz=conv(dataup,[zeros(1,T/4) ones(1,T/2) zeros(1,T/4)]);
yrz=yrz(1:end-T+1);
%Non-return to zero polar signal
ynrz=conv(dataup, ones(1,T));
ynrz=ynrz(1:end-T+1);
%half sinusoid polar signal
ysine=conv(dataup, sin(pi*[0:T-1]/T));
ysine=ysine(1:end-T+1);
% generating RC pulse train and rolloff factor = 0.5
yrcos=conv(dataup, rcosfir(0.5, Td, T,1,'normal'));
yrcos=yrcos(2*Td*T:end-2*Td*T+1);
eye1=eyediagram(yrz,T,T,T/2);title('RZ eye-diagram');grid on;ylim([-1.5,1.5]);xlim([-Td-2,Td+2]);
eye2=eyediagram(ynrz,T,T,T/2);title('NRZ eye-diagram');grid on;ylim([-1.5,1.5]);
xlim([-Td-2,Td+2])
eye3=eyediagram(ysine,T,T,T/2);title('Half-sine eye-diagram');grid('on');
ylim([-1.5,1.5]);xlim([-Td-2,Td+2])
eye4=eyediagram(yrcos,2*T,T); title('Raised-cosine eye-diagram');grid('on');
ylim([-1.5,1.5]);xlim([-Td-2,Td+2])
```

9. BLOCK / CIRCUIT / MODEL DIAGRAM / REACTION EQUATION:

10. OBSERVATION TABLE / LOOKUP TABLE / TRUTH TABLE:

11. GRAPH:



12. RESULTS & CONCLUSIONS:

- The MATLAB program for generating NRZ, RZ and raised-cosine is simulated and output waveforms are observed.

13. LEARNING OUTCOMES :

- Students are able to plot eye diagram and simulate the nrz, rz and raised-cosine using MATLAB SOFTWARE.

14. APPLICATION AREAS:

- High-speed communication systems
- Data centers
- Network interfaces
- High-speed memory interfaces

15. REMARKS:

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-

1. EXPERIMENT NO:9

2. TITLE: **GENERATE THE PROBABILITY DENSITY FUNCTION OF GAUSSIAN DISTRIBUTION FUNCTION.**

3. LEARNING OBJECTIVES:

- To study the concept of Gaussian distribution
- To generate PDF of Gaussian distributionin MATLAB

4. AIM:

- To Study and verify the PDF of Gaussian distributionin MATLAB

5. MATERIAL / EQUIPMENT REQUIRED:

- MATLAB Software

6. THEORY / HYPOTHESIS:

- Gaussian probability distribution is perhaps the most used distribution in all of science. u also called “bell shaped curve” or normal distribution. Unlike the binomial and Poisson distribution, the Gaussian is a continuous distribution:

$$P(y) = \frac{1}{\sigma\sqrt{2\pi}} e^{-\frac{(y-\mu)^2}{2\sigma^2}}$$

μ = mean of distribution (also at the same place as mode and median)
σ^2 = variance of distribution
y is a continuous variable ($-\infty \leq y \leq \infty$)

Probability (P) of y being in the range $[a, b]$ is given by an integral:

$$P(a < y < b) = \frac{1}{\sigma\sqrt{2\pi}} \int_a^b e^{-\frac{(y-\mu)^2}{2\sigma^2}} dy$$

The total area under the curve is normalized to one. + the probability integral:

$$P(-\infty < y < \infty) = \frac{1}{\sigma\sqrt{2\pi}} \int_{-\infty}^{\infty} e^{-\frac{(y-\mu)^2}{2\sigma^2}} dy = 1$$

7. FORMULA / CALCULATIONS:

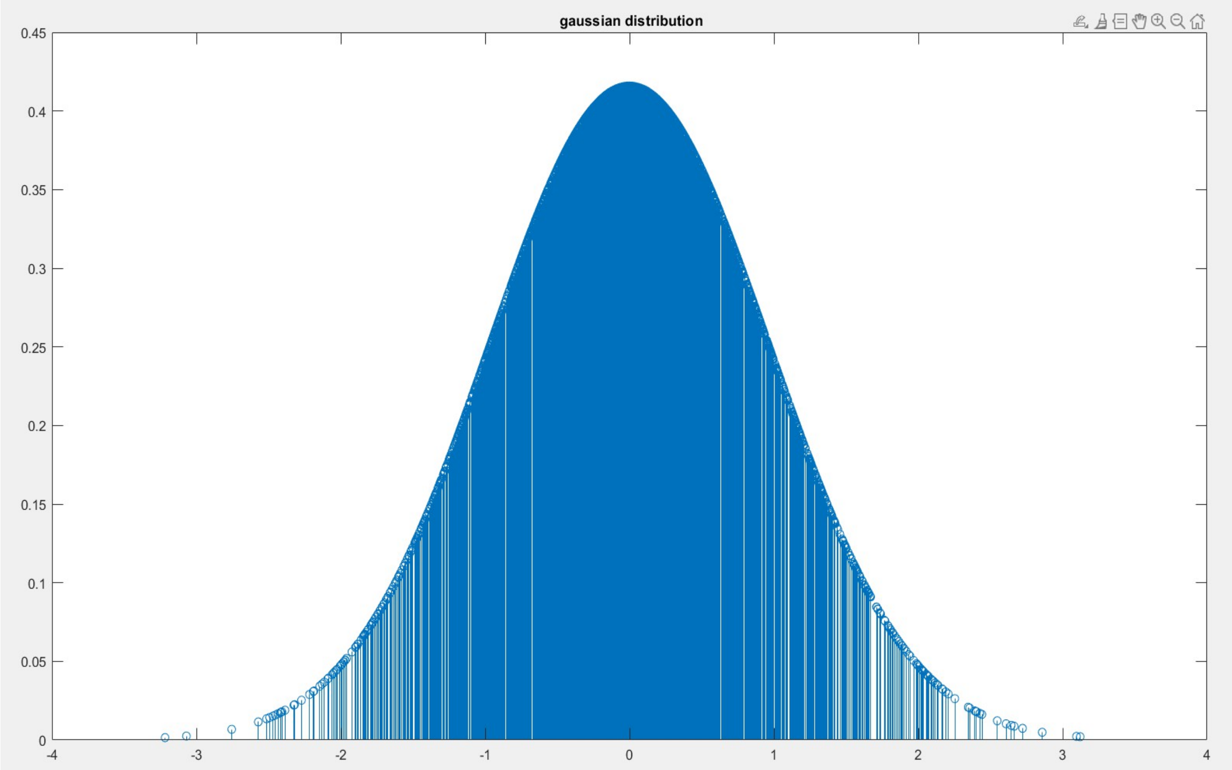
8. PROCEDURE / PROGRAMME / ACTIVITY:

```
clc;
clear all;
close all;
%generates a set of 2000 samples of Gaussian distributed random numbers
x=randn(1,2000);
%plot the joint distribution of both the sets using dot.
%subplot(211)
%plot(x,'.');
%title('scatter plot of gaussian distributed random numbers');
ymu=mean(x)
ymsq=sum(x.^2)/length(x)
ysigma=std(x)
yvar=var(x)
yskew=skewness(x)
p=normpdf(x,ymu,ysigma);
%subplot(212);
stem(x,p);
title(' gaussian distribution');
```

9. BLOCK / CIRCUIT / MODEL DIAGRAM / REACTION EQUATION:

10. OBSERVATION TABLE / LOOKUP TABLE / TRUTH TABLE:

11. GRAPHS / OUTPUTS:



12. RESULTS & CONCLUSIONS:

- The MATLAB program for generating PDF of gaussian distribution is simulated and output waveforms are observed.

13. LEARNING OUTCOMES :

- Students are able to simulate the pdf of gaussian distribution using MATLAB SOFTWARE.

14. APPLICATION AREAS:

- Stream processing
- Vector processing
- Digital processing applications

15. REMARKS:

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-

1. EXPERIMENT NO:10

2. TITLE: **DISPLAY THE SIGNAL AND ITS SPECTRUM OF AN AUDIO SIGNAL**

3. LEARNING OBJECTIVES:

- To display the audio signal and its spectrum.

4. AIM:

- To Study and verify the audio signal and its spectrum in MATLAB

5. MATERIAL / EQUIPMENT REQUIRED:

- MATLAB Software

6. THEORY / HYPOTHESIS:

- By nature, a sound wave is a continuous signal, meaning it contains an infinite number of signal values in a given time. This poses problems for digital devices which expect finite arrays. To be processed, stored, and transmitted by digital devices, the continuous sound wave needs to be converted into a series of discrete values, known as a digital representation. If you look at any audio dataset, you'll find digital files with sound excerpts, such as text narration or music. You may encounter different file formats such as `.wav`(Waveform Audio File), `.flac`(Free Lossless Audio Codec) and `.mp3`(MPEG-1 Audio Layer 3). These formats mainly differ in how they compress the digital representation of the audio signal. It might have seen sounds visualized as a waveform, which plots the sample values over time and illustrates the changes in the sound's amplitude. This is also known as the time domain representation of sound. Another way to visualize audio data is to plot the frequency spectrum of an audio signal, also known as the frequency domain representation. The spectrum is computed using the discrete Fourier transform or DFT. It describes the individual frequencies that make up the signal and how strong they are.
-

7. FORMULA / CALCULATIONS:

- -
- -

8. PROCEDURE / PROGRAMME / ACTIVITY:

```
s = audioread('piano2.wav');
s = s(:,1); % extract one sound track only
% load different audio source by changing the file name in ('')
subplot(211)
plot(s); % visualize the signal in time domain
Fs = 44100; % sample rate of the audio signal
N = length(s); % the number of samples of the audio signal
transform = fft(s,N); % apply Fourier transform
magTransform = abs(transform); % magnitude of the FFT
faxis = ((-0.5:1/N:0.5-1/N)*Fs).'; % frequency range of the signal
subplot(212)
plot(faxis,fftshift(magTransform));
xlabel('Frequency (Hz)');
ylabel('Spectrum magnitude');
xlim([-1000 1000]);
```

-

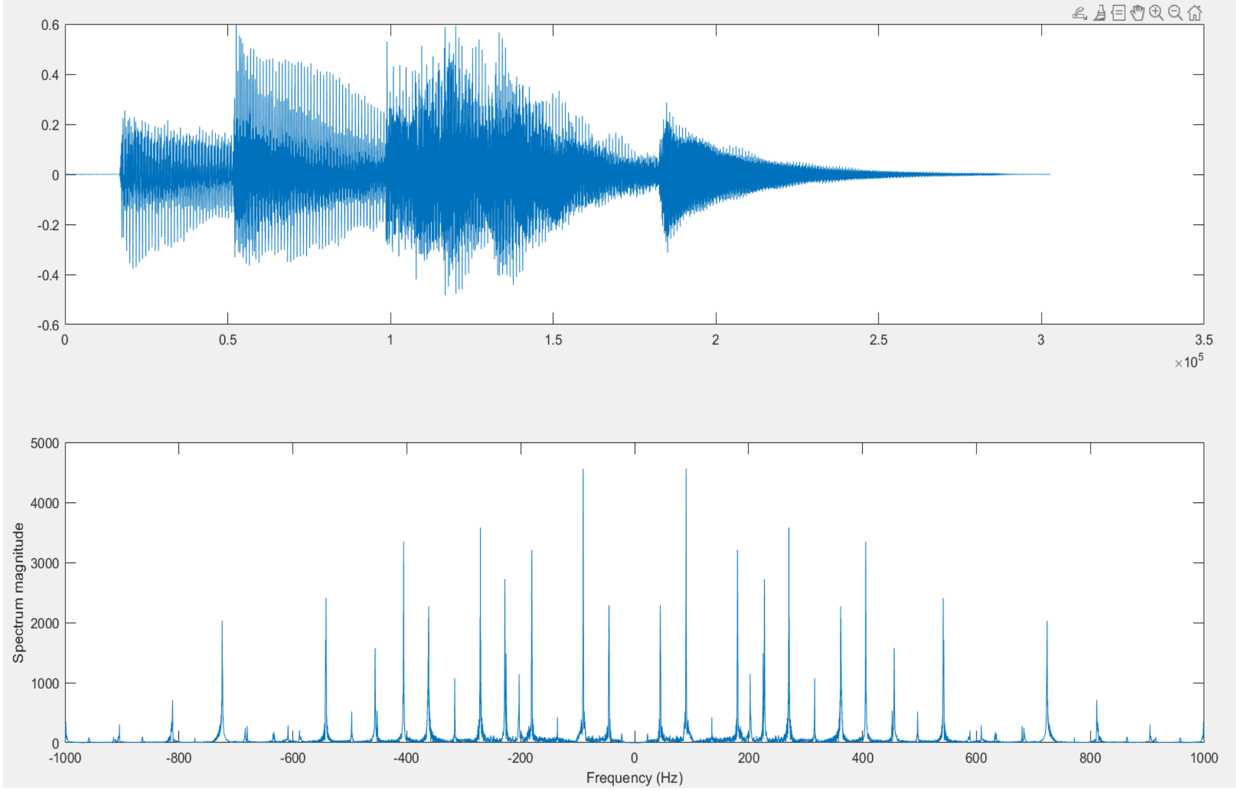
9. BLOCK / CIRCUIT / MODEL DIAGRAM / REACTION EQUATION:

-

10. OBSERVATION TABLE / LOOKUP TABLE / TRUTH TABLE:

-

11. GRAPHS / OUTPUTS:



12. RESULTS & CONCLUSIONS:
- The MATLAB program for displaying an audio signal is simulated and output waveforms are observed.

13. LEARNING OUTCOMES :
- Students are able to simulate the program for displaying an audio signal using MATLAB SOFTWARE.

14. APPLICATION AREAS:
- Human-computer interface
- Musical applications
- Robotics, hearing aids, and health care

15. REMARKS:
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H. INTERNAL ASSESSMENT TEST MODEL QUESTION PAPER

Dept: ECE	Sem / Div:IV	Course:Principles of Communication systems	Course Code: BEC402
Date:	Time: 90 Min.	Max Marks: 50	Elective: N
Note: Answer any 2 FULL questions.			

QN		Questions	Marks	RBT	CO
		PART A			
1	a	The random variable 'Y' is a function of another random variable 'X' such that Y=cos(X) and 'X' is uniformly distributed in the interval (-	10	L3	(CO1)

		Π, Π) i,e $f_x(x) = \frac{1}{2\pi} \quad -\pi < x < \pi$ $= 0 \quad$ othwerise Find out the mean value of 'y'.			
	b	Consider the random variable X defined by probability density function $f_x(x) = \begin{cases} k \text{ a constant for } 2 \le x \le 4 \\ 0 \text{ elsewhere} \end{cases}$ Determine :, i) The constant K ii) $F_X(x)$.	10	L3	(CO1)
	c	The message signal $m(t) = 1/(1+t^2)$. Determine and sketch the modulated wave for amplitude modulation with the following values : i) u= 50% , ii)u= 100%	5	L3	(CO2)
2	a	Derive an equation for SSB modulated wave for which upper side-band is retained.	10	L3	(CO2)
	b	State the five main benefits of SSB over conventional AM.	5	L2	(CO2)
	c	Explain the following terms : i) Moments ii) central moment iii) mean iv) covarience	10	L2	(CO1)
PART B					
3	a	Define a random variable. Illustrate the relationship between sample space, random variable and probability.	10	L2	(CO1)
	b	Define the autocorrelation and crosscorrelation functions. State the properties of auto correlation function	10	L2	(CO1)
	c	Write an AM wave expression in time and in frequency domain. Also, draw AM waveform.	5	L2	(CO2)
4	a	An AM wave displayed on an oscilloscope has values of Vmax = 4.8 and Vmin = 2.5 as read from the grati-cule. What is the percentage of modulation?	7	L3	(CO2)
	b	An AM transmitter puts a carrier of 6 A into an antenna whose resistance is 52 V. The transmitter is modulated by 60 percent. What is the total output power?	8	L3	(CO2)
	c	Explain signal to noise ratio? And Explain different types of noise.	10	L2	(CO1)

G. CONTINUOUS INTERNAL EVALUATION

Evaluation	Weightage in Marks
IA Test – 1	25
IA Test – 2	25
Marks for Different Components: Additional Assessment Tools (AATs) – (Assignments/ Open Book Test/ Written Quiz/ Seminar/ Report Writing/ Conduction of Experiments/ Micro-Project)	10+15 To deepen student's understanding and increase his/her confidence in the topics studied. Further, to improve the oral, written skills and engineering aptitudes.

COURSE PLAN