# Techno India NJR Institute of Technology

Department of Electronics & Communication Engineering

B.Tech. III Semester

Lab: Signal Processing Lab (3EC4-23)

# RAJASTHAN TECHNICAL UNIVERSITY, KOTA

**SYLLABUS**

**II Year - III Semester: B.Tech. (Electronics & Communication Engineering)**

**3EC4-23: Signal Processing Lab**

**1 Credit** **Max. Marks: 50 (IA:30, ETE:20)**

**0L:0T:2P**

**List of Experiments**

| Sr. No. | Name of Experiment (Simulate using MATLAB environment) |
|---|---|
| 1. | Generation of continuous and discrete elementary signals (periodic and non periodic) using mathematical expression. |
| 2. | Generation of Continuous and Discrete Unit Step Signal. |
| 3. | Generation of Exponential and Ramp signals in Continuous & Discrete domain. |
| 4. | Continuous and discrete time Convolution (using basic definition). |
| 5. | Adding and subtracting two given signals. (Continuous as well as Discrete signals) |
| 6. | To generate uniform random numbers between (0, 1). |
| 7. | To generate a random binary wave. |
| 8. | To generate and verify random sequences with arbitrary distributions, means and variances for following:<br>(a) Rayleigh distribution<br>(b) Normal distributions: N(0,1).<br>(c) Gaussion distributions: N (m, x) |
| 9. | To plot the probability density functions. Find mean and variance for the above distributions |

**Course Outcome:**

| Course Code | Course Name | Course Outcome | Details |
|---|---|---|---|
| 3EC4-23 | Signal Processing Lab | CO 1 | Able to generate different Continuous and Discrete time signals. |
| | | CO 2 | Understand the basics of signals and different operations on signals. |
| | | CO 3 | Develop simple algorithms for signal processing and test them using MATLAB |
| | | CO 4 | Able to generate the random signals having different distributions, mean and variance. |
| | | CO 5 | Design and conduct experiments, interpret and analyse data and report results. |

Office of Dean Academic Affairs
Rajasthan Technical University, Kota

**CO-PO Mapping:**

| Subject | Course Outcomes | PO 1 | PO 2 | PO 3 | PO 4 | PO 5 | PO 6 | PO 7 | PO 8 | PO 9 | PO 10 | PO 11 | PO 12 |
|---|---|---|---|---|---|---|---|---|---|---|---|---|---|
| 3EC4-23 Signal Processing Lab | CO 1 | 2 | | 1 | | 2 | | | | | | | |
| | CO 2 | 3 | | 1 | | | | | | | | | |
| | CO 3 | 1 | 2 | 3 | 1 | 3 | | | | | | | |
| | CO 4 | 2 | 1 | 1 | | 2 | | | | | | | |
| | CO 5 | 1 | 1 | 2 | 2 | 2 | | | | | | | |

**3: Strongly** **2: Moderate** **1: Weak**

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# INSTRUCTIONS OF LAB

## DO'S

1. Student should get the record of previous experiment checked before starting the new experiment.
2. Read the manual carefully before starting the experiment.
3. Before starting the experiment, system checked by the teacher.
4. Get your readings checked by the teacher.
5. Apparatus must be handled carefully.
6. Maintain strict discipline.
7. Keep your mobile phone switched off or in vibration mode.
8. Students should get the experiment allotted for next turn, before leaving the lab.

## DON'TS

1. Do not touch or attempt to touch the mains power supply wire with bare hands.
2. Do not overcrowd the tables.
3. Do not tamper with equipments.
4. Do not leave the lab without permission from the teacher.

## Safety Measures of Lab

1. Antivirus software is installed for protection against viruses and malwares.
2. External storage devices are not allowed to use in lab.
3. At all times follow the right procedures while starting and shutting down the computer therefore abrupt switching on and off the computer should be avoided since this can lead to damaging the computer.
4. Any repairs to the computer should be done by someone who has knowledge regarding computer repairs.

# EXPERIMENT No. 1

**AIM: Write a program to generatefollowing continuous and discrete (periodic and aperiodic) signal.**

a) $y(t) = sin(2 * 3.14 * 5 * t) + cos(2 * 3.14 * t)$;
b) $y(n) = (-1)n$
c) $y(t) = sin\sqrt{2}\pi t + cos\pi t$
d) $x(n) = n2 + 4$

**SOFTWARE USED**: Scilab 6.1.0.

## THEORY:

**Continuous and Discrete Time Signals:**

A continuous signal or a continuous-time signal is a varying quantity (a signal) whose domain, which is often time, is a continuum (e.g., a connected interval of the real). That is, the function's domain is an uncountable set. The function itself need not be continuous. To contrast, a discrete time signal has a countable domain, like the natural numbers.

A signal of continuous amplitude and time is known as a continuous-time signal or an analog signal. This (a signal) will have some value at every instant of time. The electrical signals derived in proportion with the physical quantities such as temperature, pressure, sound etc. are generally continuous signals. Other examples of continuous signals are sine wave, cosine wave, triangular wave etc.

The signal is defined over a domain, which may or may not be finite, and there is a functional mapping from the domain to the value of the signal. The continuity of the time variable, in connection with the law of density of real numbers, means that the signal value can be found at any arbitrary point in time.

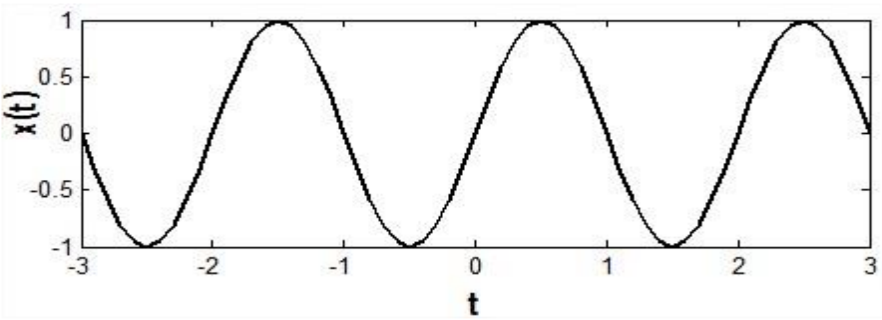



***Fig 1.1 Continuous Time Signal***

Signals that can be defined at discrete instant of time is called discrete time signal. Basically discrete time signals can be obtained by sampling a continuous-time signal. It is denoted as *x(n)*.A discrete signal or discrete-time signal is a time series consisting of a sequence of quantities.

Unlike a continuous-time signal, a discrete-time signal is not a function of a continuous argument; however, it may have been obtained by sampling from a continuous-time signal. When a discrete-time signal is obtained by sampling a sequence at uniformly spaced times, it has an associated sampling rate.Discrete-time signals may have several origins, but can usually be classified into one of two groups.

- By acquiring values of an analog signal at constant or variable rate. This process is called sampling

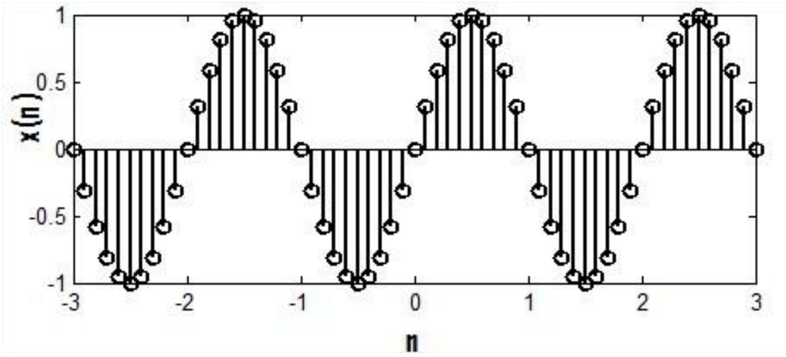



***Fig 1.2 Discrete Time Signal***

**Periodic and Non-periodic Signals:**

A signal $x(t)$ is said to be periodic if it repeats itself after some amount of time

$$x(t + nT) = x(t) \quad (1.1)$$

here, T is time period of the periodic signal, n is any positive or negative integer. The period of the signal is the minimum value of time for which it exactly repeats itself. The combination of the periodic signals will be periodic if the ratio of the period of independent signals is rational number. Fig 1.3 shows a periodic sine wave with period $T = 2\pi$.

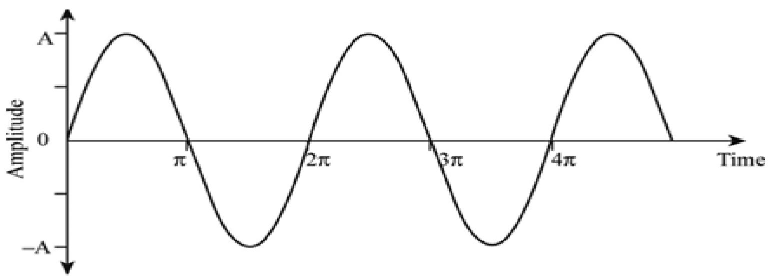



***Fig 1.3 Periodic Signal***

A signal which does not repeat itself after a certain period of time is called Non-periodic signal. Fig 1.4 shows a periodic sine wave with period $T = 2\pi$.

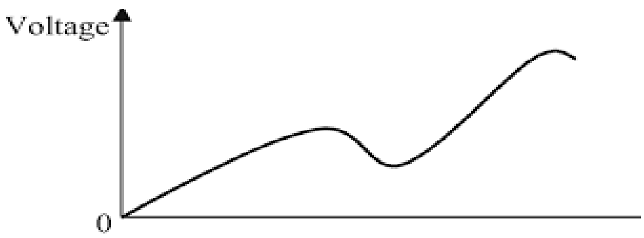


***Fig 1.4 Non-Periodic Signal***

**% SCILAB code**

```
clear all;
clc;
clf;
t=0:0.01:5 ;
y=sin(2*3.14*5*t) + cos(2*3.14*t);
subplot(221)
plot(t,y);
xlabel('Time');
ylabel('Amplitude');
title('Continuous Periodic Signal');
n=0:1:10;
x=(-1).^n;
subplot(222)
plot2d3(n,x);
xlabel('Time');
ylabel('Amplitude');
title('Discrete Periodic Signal');
z=sin(sqrt(2)*3.14*t)+cos(3.14*t);
subplot(223)
plot(t,z);
xlabel('Time');
ylabel('Amplitude');
title('Continuous Non-Periodic Signal');
r=n.^2+4;
subplot(224)
plot2d3(n,r);
xlabel('Time');
ylabel('Amplitude');
title ('Discrete Non-periodic signal')
```

## OUTPUT:

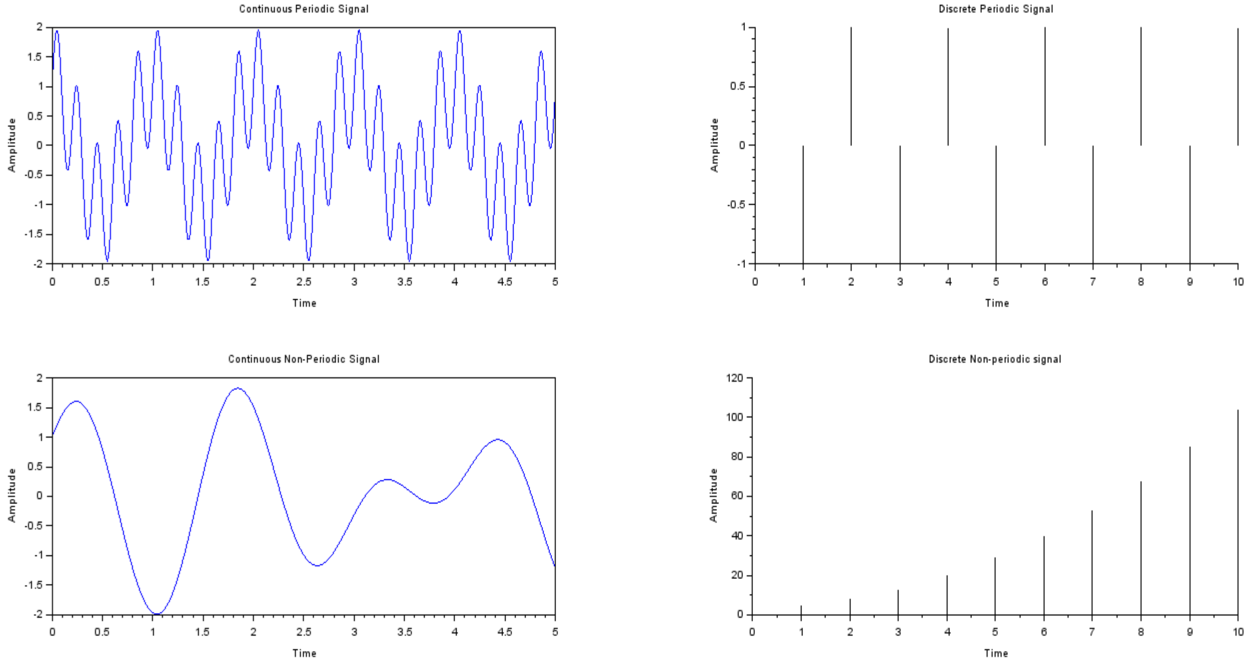



***Fig 1.5 Periodic and Non-Periodic Continuous and Discrete signal***

**RESULT:-**We have successfully generated continuous periodic, discrete periodic, continuous aperiodic and discrete aperiodic signals.Fig1.5 shows continuous time periodic signal of frequency 10 Hz. Fig 1.6 shows discrete time periodic signal of frequency 100 Hz. Fig 1.7 and fig 1.8 shows continuous time signal and discrete time aperiodic signal.

## DISCUSSION:

1. What is a signal.
2. Difference between continuous and discrete signal.
3. Commands difference between plot and stem function.
4. What is the difference between * and .* in MATLAB.
5. Determine which of the following signals are periodic.

1. $x(t) = sin\, 15\pi t$
2. $x(t) = \sin\sqrt{2}\,\pi t$

## EXPERIMENT No. 2

**AIM: Write a program to plot continuous and discrete unit step signals**.

**SOFTWARE USED**: MATLAB 7.1

## THEORY:

**Unit Step :** Unit step signal is defined as a signal with magnitude one for time greater and equal to zero .

We can assume it as a dc signal which got switched on at time equal to zero. Unit step signal is a basic signal and is denoted by u(t) in continuous domain or u(n ) in discrete domain.

Mathematically, it is defined as.

$$u(t) = 1, \quad t >= 0$$
$$\quad\quad 0, \quad t < 0$$

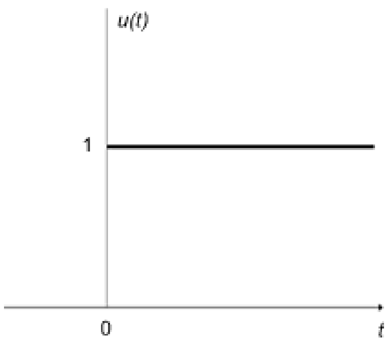



***Fig 2.1 Continuous time unit step signal***

**For Discrete time**

$$u(n) = 1, \quad n >= 0$$
$$\quad\quad 0, \quad n < 0$$

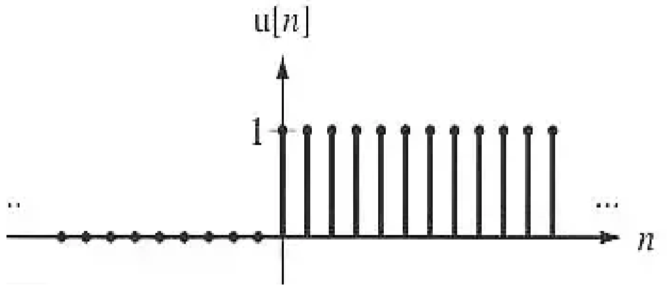



***Fig2.2  Discrete time unit step signal***

**SCILABCode for Continuous and Discrete Time unit step Signal:**

```
clear all;
clc;
clf;
t = -10:.01:10;
ut = 1.*(t>=0);
subplot(2,1,1);
plot2d2(t,ut);
xlabel('time ');
ylabel('Amplitude');
title('Continuous time Unit Step Signal');
n = -8:1:8;
un = 1.*(n>=0);
subplot(2,1,2);
plot2d3(n,un);
xlabel('time ');
ylabel('Amplitude');
title('Discrete time Unit Step Signal');
```

**OUTPUT:**

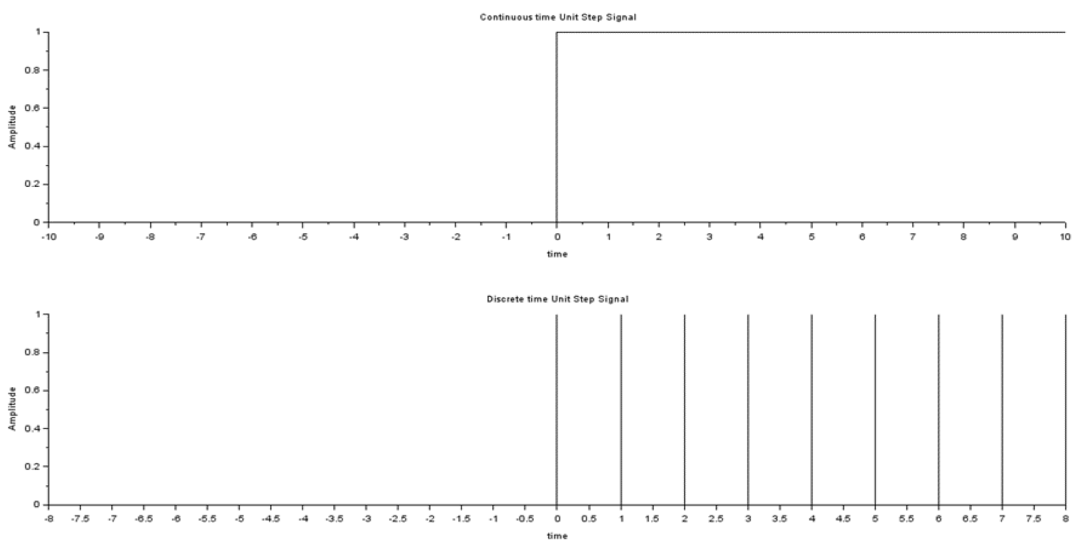


***Fig 2.3 Continuous time unit step signal***

**RESULT:** We have generated the continuous and discrete unit step signals. The X axis and Y axis denotes time and amplitude respectively. The unit step function value will be 1 for positive value of time input and zero for negative value of time input. Fig 2.3 shows continuous time unit step signal. Fig 2.4 shows discrete time unit step signal.

**DISCUSSION:**

1. What is the practical application of unit step function?
2. Differentiate between unit step and unit impulse function.
3. Draw y(t)= u(t+2)- u (t-2)
4. Draw y(t)= 2u(n+1) – u(n-1)

# EXPERIMENT No. 3

**AIM**: Write a program to generate Exponential and Ramp signals in Continuous & Discrete domain.

**SOFTWARE USED:** MATLAB 7.1

**THEORY:** The "exponential" signal literally represents an exponentially increasing or falling series:

Continuous time: $y(t) = ce^{at}$

Discrete time: $y(n) = ce^{an}$

where c and a are in general complex numbers.The exponential signal models the behavior of many phenomena, such as the decay of electrical signals across a capacitor or inductor.

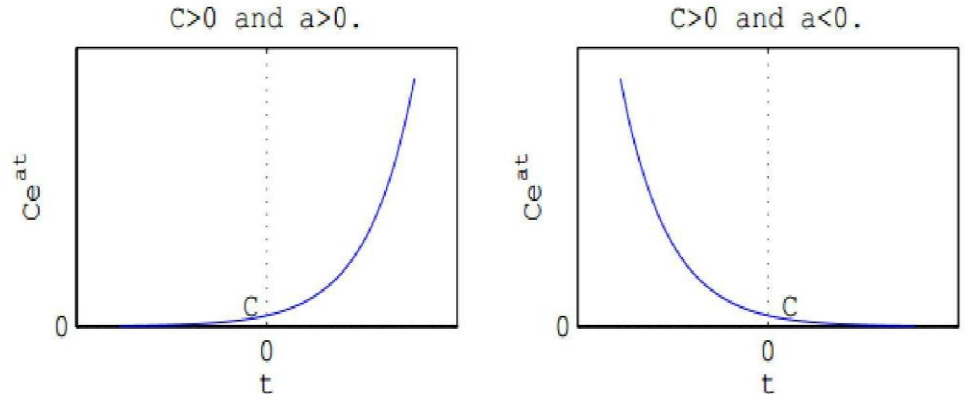



***Fig 3.1 Continuous time exponential signal***

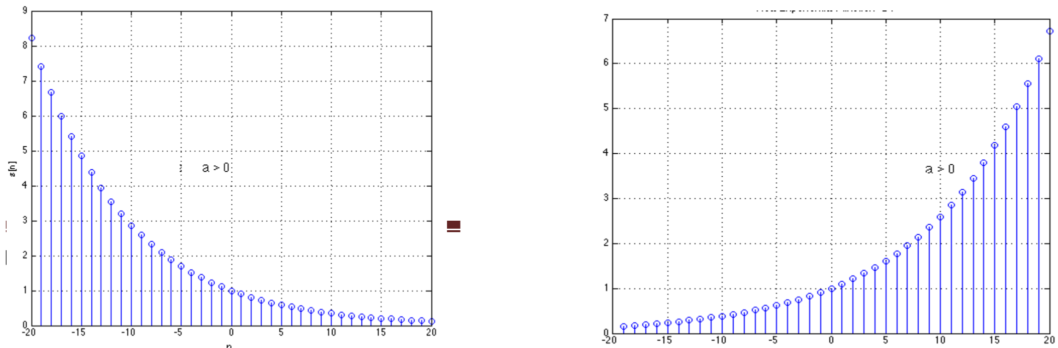


*Fig 3.2 Discrete time exponential signal*

The case a > 0 represents exponential growth. Some signals in unstable systems exhibit exponential growth.

The case a < 0 represents exponential decay. Some signals in stable systems exhibit exponential decay.

**Ramp signal**: A signal whose magnitude increases same as time. It can beobtained by integrating unit step.

**Continuous time**r(t) = t, t >= 0

0, t<0

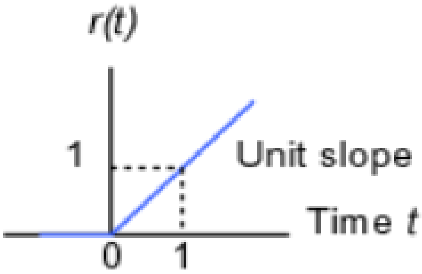



*Fig 3.3 Continuous time ramp signal*

**Discrete time** r(n) = n, n>= 0

0, n < 0

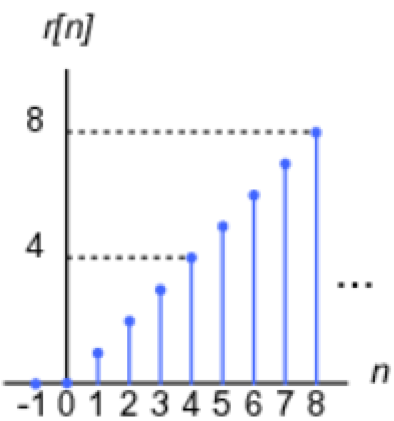



*Fig 3.4 Discrete time ramp signal*

**SCILAB Code for continuous time exponential signal**

```
clear all;
clc;
clf;
t=0:.01:10;
y1=1*exp(1*t);
subplot(211);
plot(t,y1);
xlabel('Time');
ylabel('Amplitude');
title('Continuous Exponential Signal with scaling factor = 1 and multiplication factor= 1');
y2=2*exp(-1*t);
subplot(212);
plot(t,y2);
xlabel('Time');
ylabel('Amplitude')
title('Continuous Exponential Signal with scaling factor = -1 and multiplication factor= 2');
```

## OUTPUT:

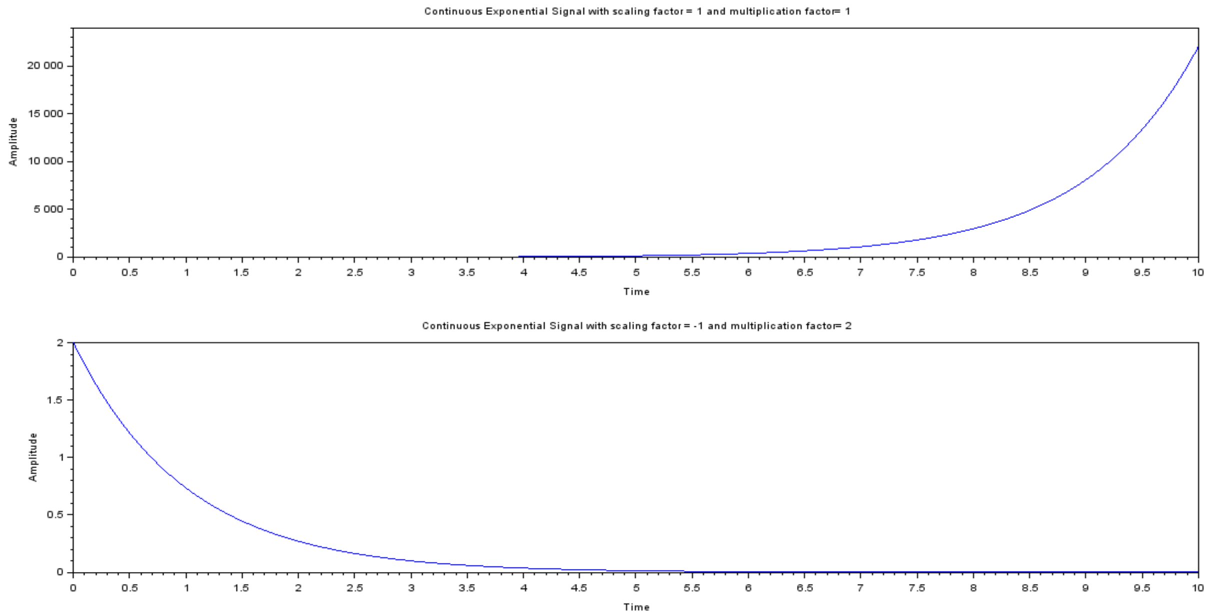



***Fig 3.5 Continuous time exponential signal***

**SCILAB code for discrete time exponential signal**

```
clear all;
clc;
clf;
t=0:1:10;
y1=1*exp(1*t);
subplot(211);
plot2d3(t,y1);
xlabel('Time');
ylabel('Amplitude');
title('Discrete Exponential Signal with scaling factor = 1 and
multiplication factor= 1');
y2=2*exp(-1*t);
subplot(212);
plot2d3(t,y2);
xlabel('Time');
ylabel('Amplitude')
```

```
title('Discrete Exponential Signal with scaling factor = -1 and multiplication factor= 2');
```

## OUTPUT:

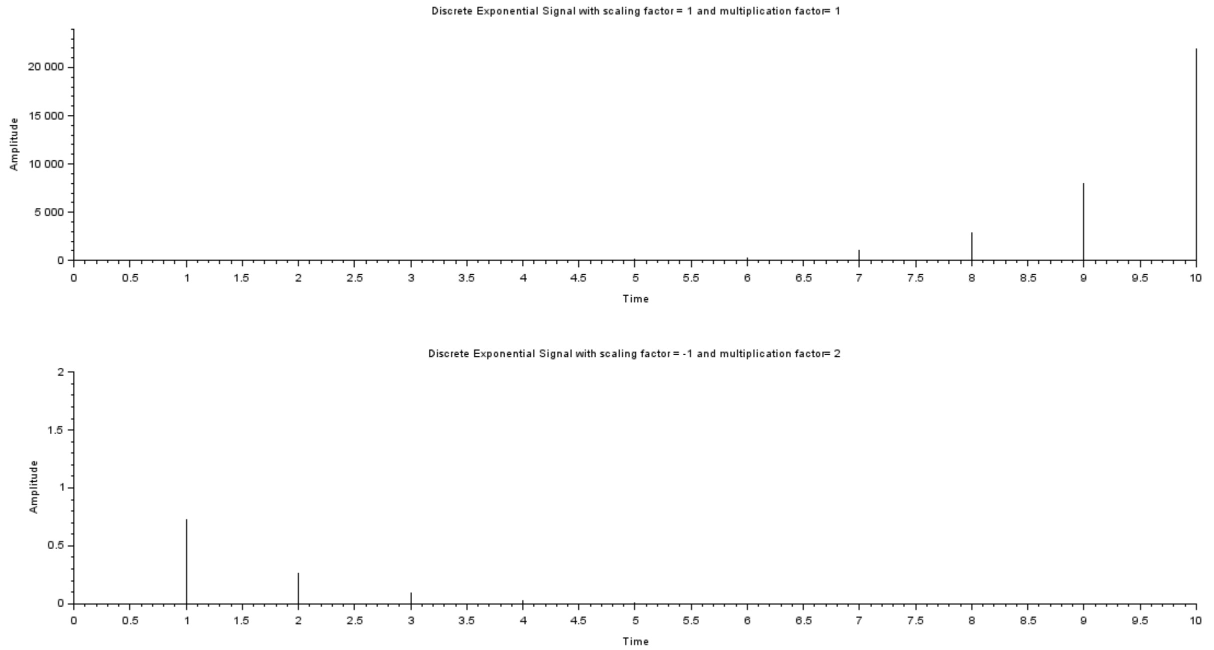



***Fig 3.6 Discrete time exponential signal***

**SCILAB Code for continuous time ramp Signal**

```
clear all;
clc;
clf;
t=0:.01:10;
y1=t;
y2=-t*2;
subplot(211);
plot(t,y1);
xlabel('Time');
ylabel('Amplitude');
title('Continuous Ramp Signal with multiplication factor= 1');
subplot(212);
plot(t,y2);
```

```
xlabel('Time');
ylabel('Amplitude');
title('Continuous Ramp Signal with multiplication factor= -2');
```

**OUTPUT:**

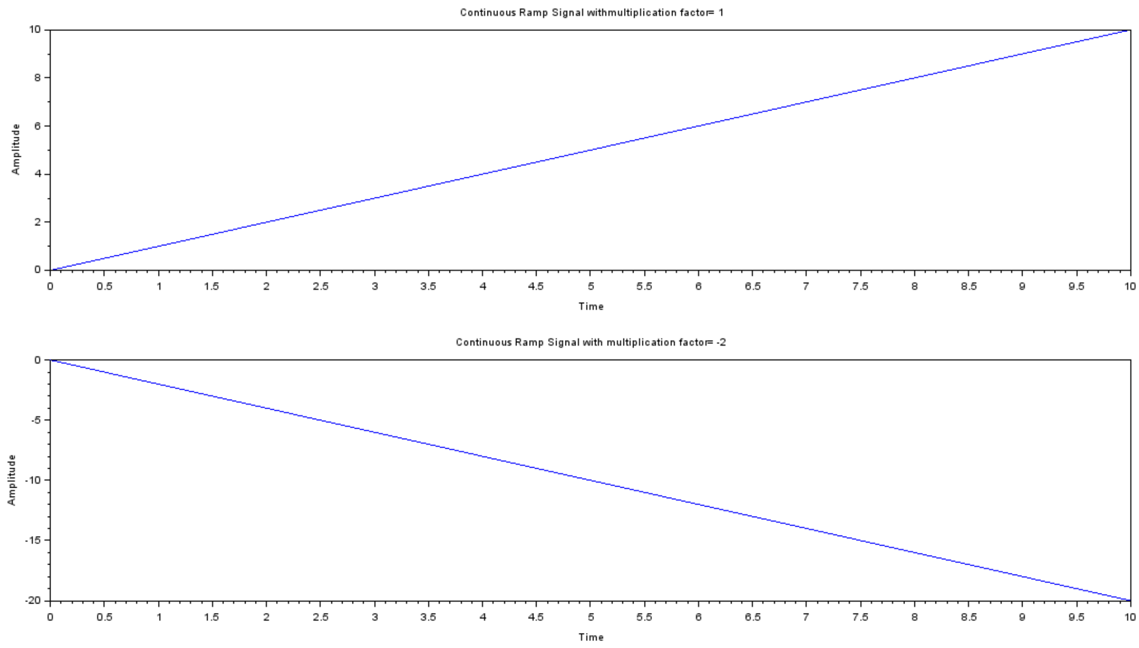



*Fig 3.7 Continuous time ramp signal*

**SCILAB code for discrete time ramp signal**

```
clear all;
clc;
clf;
t=0:1:10;
y1=t;
y2=-t*2;
subplot(211);
plot2d3(t,y1);
xlabel('Time');
ylabel('Amplitude');
title('Discrete Ramp Signal with multiplication factor= 1');
subplot(212);
```

```
plot2d3(t,y2);
xlabel('Time');
ylabel('Amplitude');
title('Discrete Ramp Signal with multiplication factor= -2');
```

## OUTPUT:

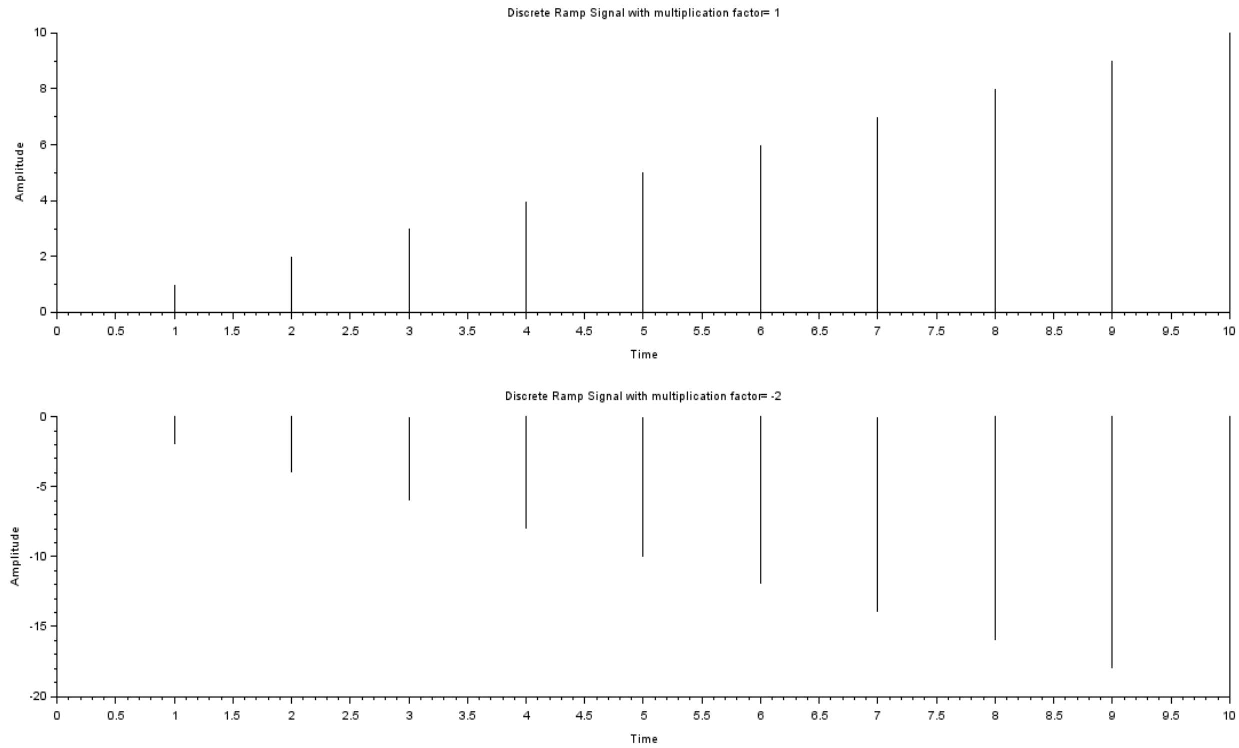



***Fig. 3.8 Discrete time ramp signal***

**RESULT:** We have plotted the continuous and discrete time exponential and ramp signals. The X axis and Y axis denotes time and amplitude respectively. Multiplication factor affects the slope of exponential and ramp signals as shown in various plots.Fig 3.5 shows continuous time exponential signal Fig.3.6 shows discrete time exponential signal Fig.3.7 shows continuous time ramp signal Fig.3.8 shows discrete time ramp signal

## DISCUSSION:

1. Explain the role of multiplication factor in exponential and ramp signals.
2. What are the practical applications of exponential and ramp functions?
3. Difference between stable and unstable signal.
4. What is the meaning of exponential decay?

# EXPERIMENT No. 4

**AIM: To perform continuous and discrete time convolution**

## Problem Statement:

Suppose an input $x(t) = u(t) - u(t-1)$ is applied to a Linear Time Invariant (LTI) system having impulse response $h(t) = exp\,(t)\, for\;\; 0 < t < 4$. Find the output of this system using convolution

**SOFTWARE USED**: Scilab 6.1.0.

## THEORY:

**Convolution:**

Convolution is defined as an operation which helps to find the output of the LTI (linear and time-invariant) system when the impulse response of the system and the input signal are known.
(Impulse response is defined as behavior of system under unit impulse signal which is defined for t=0 only)

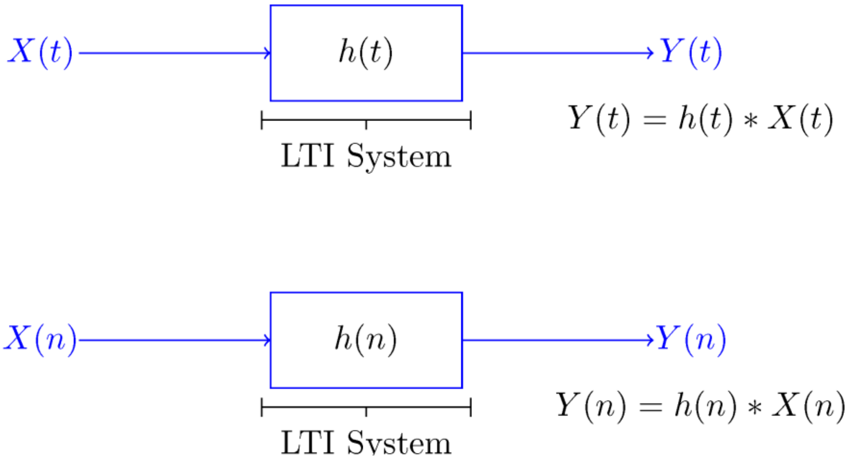



***Fig 4.1 LTI system with output represented as convolution between input and impulse response***
***a) Continuous time system (b) discrete time system***

If a continuous-time system is both linear and time-invariant, then the output Y(t) is related to the input X(t) and impulse response (h(t)) of the system by the equation given below:
:

$$Y(t) = X(t) * h(t) = \int_{-\infty}^{\infty} X(\tau)h(t-\tau)d\tau \qquad (4.1)$$

Equation 4.1 is also known as convolution integral between X(t) and h(t)

Similarly, if a discrete-time system is both linear and time-invariant, then the output Y(n) is related to the input X(n) and impulse response (h(n)) of the system by the equation mentioned below:

$$Y(n) = X(n) * h(n) = \sum_{k=-\infty}^{\infty} X(k)h(n-k) \qquad (4.2)$$

Equation 4.2 is also known as convolution sum of X(n) and h(n)

**% SCILAB code for continuous signal**

```
clear all;
clc;
```

```
clf;
t=-8:.01:8;
x=3*cos(2.*t);
h=exp(-abs(t));
y=convol(x,h);
//figure
subplot(3,1,1);
plot(t,h);
xlabel('time ');
ylabel('Amplitude');
title('Impulse response');
//figure
subplot(3,1,2);
plot(t,x);
xlabel('time ');
ylabel('Amplitude');
title('Input signal');
//figure
t2=-16:.01:16
subplot(3,1,3);
plot(t2,y);
xlabel('time ');
ylabel('Amplitude');
title('Convoluted Output signal');
```

## OUTPUT:

enter the time range for input: -3:0.01:3;
enter range for impulse response:0:0.01:4;

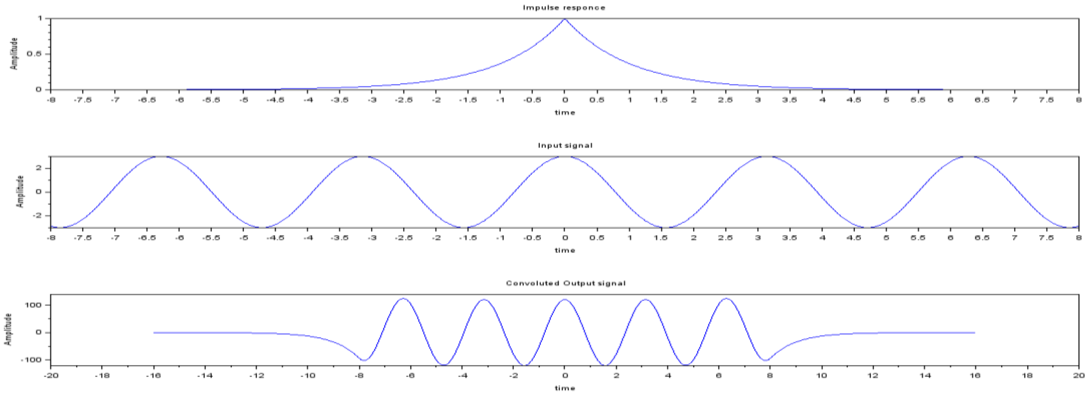



*Fig 4.2: a) input b) impulse response c) the convolved output*

**% SCILAB code for discrete time convolution**

```
clc;
clear all;
clf;
n1=0:1:12;
h=1.*(n1>=0);
n2=0:1:10;
x=exp((-1/4)*n2).*(n2>=0);
subplot(3,1,1);
plot2d3(n1,h);
xlabel('time');
ylabel('Amplitude');
title('Impulse Response');
subplot(3,1,2);
plot2d3(n2,x);
xlabel('time');
ylabel('Amplitude');
title('Input Signal');
T1=length(h);
T2=length(x);
c=convol(h,x);
t=0:1:T1+T2-2;
subplot(3,1,3);
plot2d3(t,c);
xlabel('time');
ylabel('Amplitude');
title('Convoluted Output');
```

**OUTPUT:**

```
enter the input: [2 3 4 5 2]
x =  2     3     4     5     2

enter the impulse response: [1 2 3 4 5]
h = 1     2     3     4     5

The convolved output is:
y = 2     7    16    30    46    50    46    33    10
```

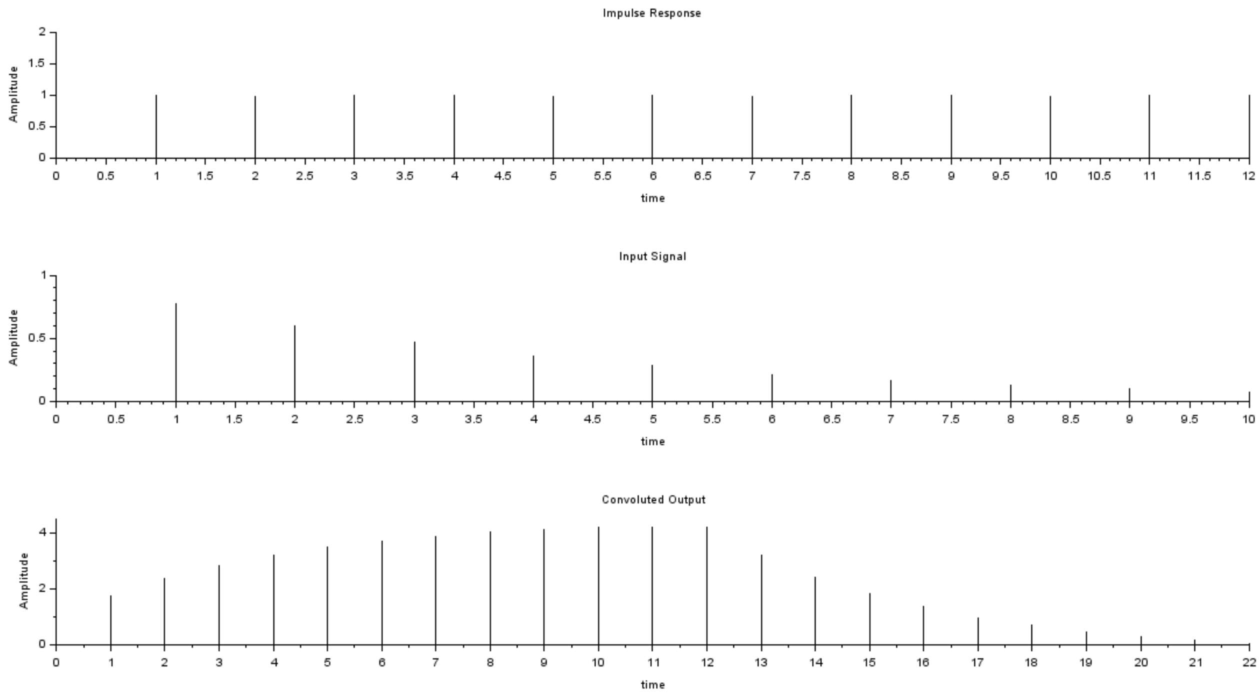



*Fig 4.3: a) input b) impulse response c) the convolved output of the Discrete System*

## RESULT:

a. The output of continuous LTI System with x(t) = u(t) – u(t-1) and h(t) = exp (-t) for 0<t<2 is as shown in fig 4.2.

b. The output of discrete LTI System with x(n) = `[2 3 4 5 2]` and h(n) = `[1 2 3 4 5]` is `y = [2 7 16 30 46 50 46 33 10]and` shown in fig 4.3.

## DISCUSSION:

1. Differentiate between linear and circular convolution.
2. Determine the unit step response of the linear time invariant system with impulse response $h(n) = a^n\, u(n)\;\; a < 1$
3. Determine the range of values of the parameter for which linear time invariant system with impulse response $h(n) = a^n\, u(n)$ is stable.
4. Describe impulse response of a function and its importance.

# EXPERIMENT No. 5

**AIM:** Write a program to plot Adding and subtracting of two given signals using mathematical expression.

**SOFTWARE USED**: Scilab 6.1.0.

## THEORY:

- Addition or unary plus. A+B adds the values stored in variables A and B. A and B must have the same size, unless one is a scalar. A scalar can be added to a matrix of any size.
- Subtraction or unary minus. A-B subtracts the value of B from A. A and B must have the same size, unless one is a scalar. A scalar can be subtracted from a matrix of any size.

**1. Addition**

The addition of signals is very similar to traditional mathematics. That is, if $x_1(t)$ and $x_2(t)$ are the two continuous time signals, then the addition of these two signals is expressed as $x_1(t) + x_2(t)$.

The resultant signal can be represented as $y(t)$ from which we can write

$$y(t) = x_1(t) + x_2(t)$$

Similarly for discrete time signals, $x_1[n]$ and $x_2[n]$, we can write

$$y[n] = x_1[n] + x_2[n]$$

**2. Subtraction**

Similar to the case of addition, subtraction deals with the subtraction of two or more signals in order to obtain a new signal. Mathematically it can be represented as

$$y(t) = x_1(t) - x_2(t)$$

for continuous time signals, $x_1(t)$ and $x_2(t)$

and $$y[n] = x_1[n] - x_2[n]$$

for discrete time signals, $x_1[n]$ and $x_2[n]$

**% SCILAB code for Addition and Subtraction of two different signals**

```
clc ;
clear all;
clf;
```

```
t = -1:0.01:5;
x1 = 1;
x2 = 2;
x3 = 3- t ;
x = x1 .*( t >0 & t <1) + x2 .*( t >=1 & t <=2) + x1 .*( t >2& t
<3) ;
y = t .*( t >0 &t <1) + x1 *( t >=1 &t <=2) + x3 .*( t >2 &t <3) ;
add = x + y ;
sub = x - y ;
subplot(2,2,1);
plot(t,x);
xlabel('t') ;
ylabel('Amplitude') ;
title('input signal = x') ;
subplot(2,2,2);
plot(t,y)
xlabel('t') ;
ylabel('Amplitude') ;
title ('input signal = y') ;
subplot(2,2,3) ;
plot(t,add);
xlabel('t') ;
ylabel('Amplitude') ;
title('Addition of two signals') ;
subplot(2,2,4) ;
plot(t,sub);
xlabel('t') ;
ylabel('Amplitude') ;
title('Subtraction of two signals') ;
```

**OUTPUT:**

```
x =       1     2     3     4

y =       1     1     1     1

add =     1     2     4     5     1     1

sub =     1     2     2     3    -1    -1
```

## OUTPUT:

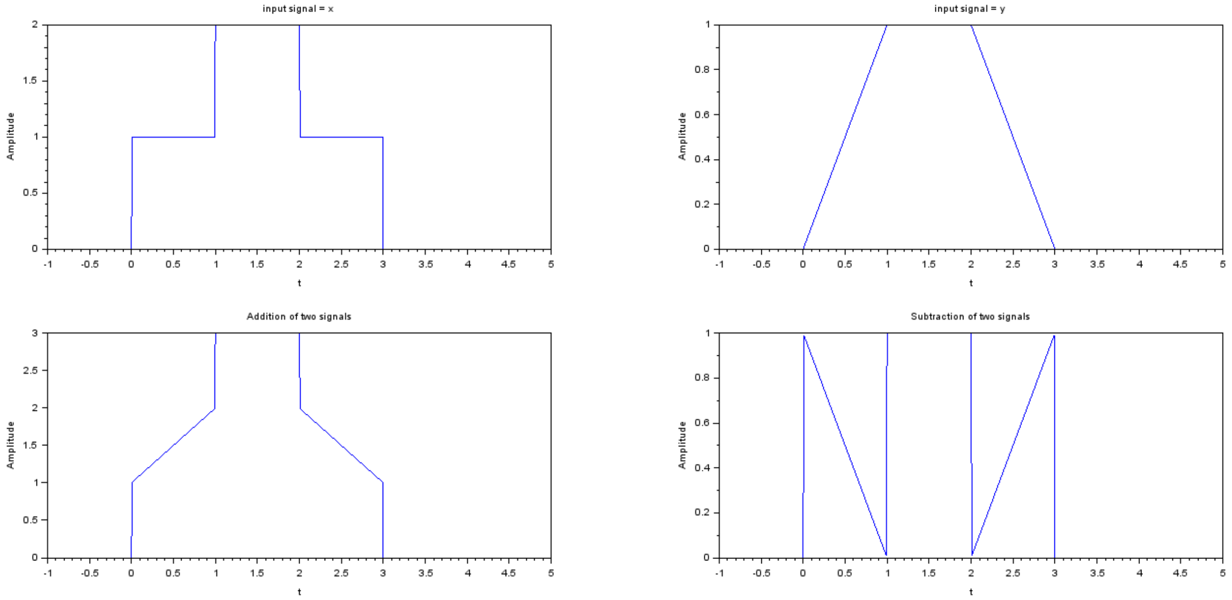



***Fig 5.1 Discrete time signal for Addition and Subtraction***

**RESULT: -**The output of continuous time signal for addition and subtraction shown in fig 5.1 and the output of discrete time signal for addition and subtraction with input sequence a=[1 2 3 4] and b=[ 1 1 1 1] and result shown in fig. 5.1.

## DISCUSSION:

1. How the discrete signals are important when compared to analog signals?
2. How digital signal processing is useful in communication systems?
3. Differentiate between analog signal processing and digital signal processing.

# EXPERIMENT No. 6

**AIM: To generate uniform random numbers between (0, 1).**

**SOFTWARE USED**: Scilab 6.1.0.

**THEORY:**

Random numbers are numbers that occur in a sequence such that two conditions are met:

The values are uniformly distributed over a defined interval or set.

It is impossible to predict future values based on past or present ones.

Random numbers are important in statistical analysis and probability theory.

rand() function is used to generate random numbers of required size.

r = rand(n) returns an n-by-n matrix containing pseudorandom values drawn from the standard uniform distribution on the open interval (0,1).

**%SCILAB Code for Random numbers generation in continuous and discrete manner**

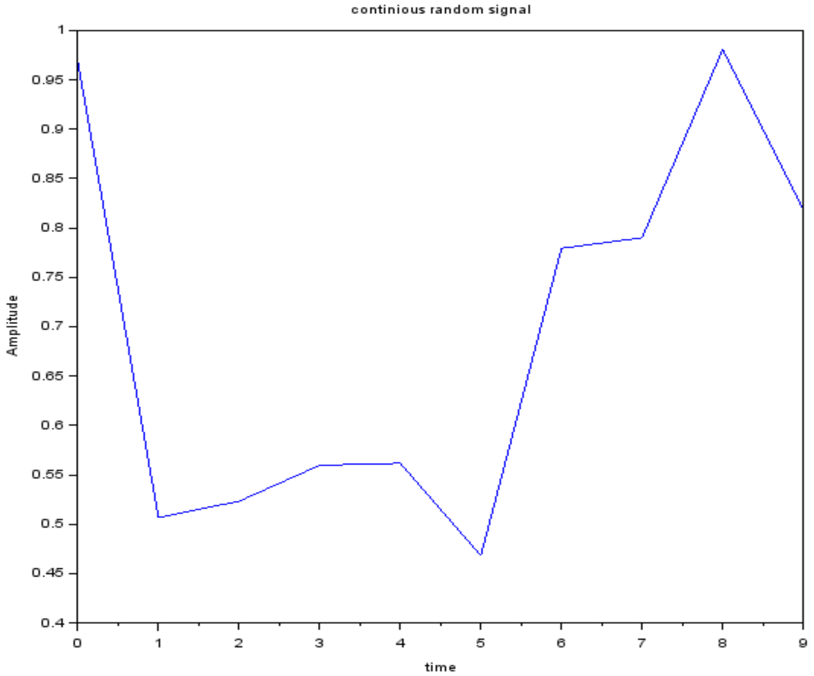
```
clc;
clf;
clear all;
t=0:1:9;
x=rand(1,10);
subplot(1,2,1);
plot(t,x);
xlabel('time');
ylabel('Amplitude');
title('continious random signal');
subplot(1,2,2);
plot2d3(t,x);
xlabel('time');
ylabel('Amplitude');
title('Discrete random signal');
```

**OUTPUT:**

Enter the total number which is generated N=10 x=rand(1,10)

x= 0.5068534 0.5232976 0.5596948 0.5617307 0.468176 0.7794547 0.7901072 0.9808542 0.8187066

**OUTPUT:**



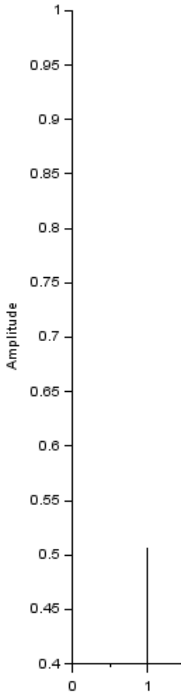




***Fig 6.1 Continuous and discrete random numbers***

**RESULT:** Uniform random numbers between amplitude range 0-1 has been plotted. The random number N which is to be generated is taken from user by using input() function. Then random numbers are generated using rand() function. Continuous random numbers are plotted using plot function and discrete number is plotted using stem function. The X-axis represents random number and Y-axis represents amplitude of that random number.10 random numbers are generated and saved in variable y and plot is shown in fig 6.1.

## DISCUSSION:

1. How can we generate signed random numbers?
2. What is the use of random numbers in study of noise?
3. Generate random numbers between amplitude 0 to 3.
4. Generate ten binary random numbers.

# EXPERIMENT No. 7

**AIM**: To generate a random binary wave.

**SOFTWARE USED**:Scilab 6.1.0.

## THEORY:

The term random signal is used primarily to denote signals, which have randomness in its nature source. As an example we can mention the thermal noise, which is created by the random movement of electrons in an electric conductor.

Random signals are those signals whose behavior cannot be predicted using any past reference. Random binary signals are digital signals which are randomly generated. It is very important for digital communication system.

A random binary signal is a random process that can assume one of two possible values at any time. A simple method of generating a random binary signal is to take Gaussian white noise, filter the noise for the desired spectra and then convert the noise to a binary signal by taking the sign of the filtered signal. The desired spectra are functions of the system time constraints.

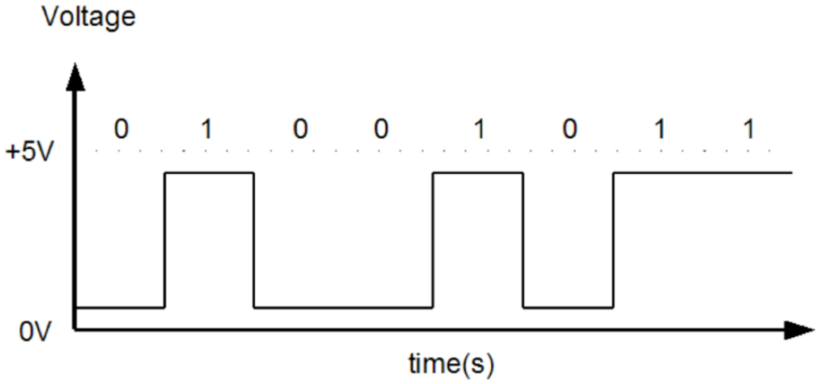



***Fig 7.1: Random Binary Waveform***

**%SCILAB code for random binary wave**

```
clc;
clf;
clear all;
u=prbs_a(20,10)
plot2d2(1:20,u,rect=[0,-1.2,20,1.2]);
xlabel('time');
ylabel('Amplitude');
title('Random Binary Wave');
```

## OUTPUT:

Random Binary Number is = 1. 1. -1. 1. -1. -1. -1. 1. -1. 1. 1. -1. 1. 1. 1.1. -1. -1. -1. 1.

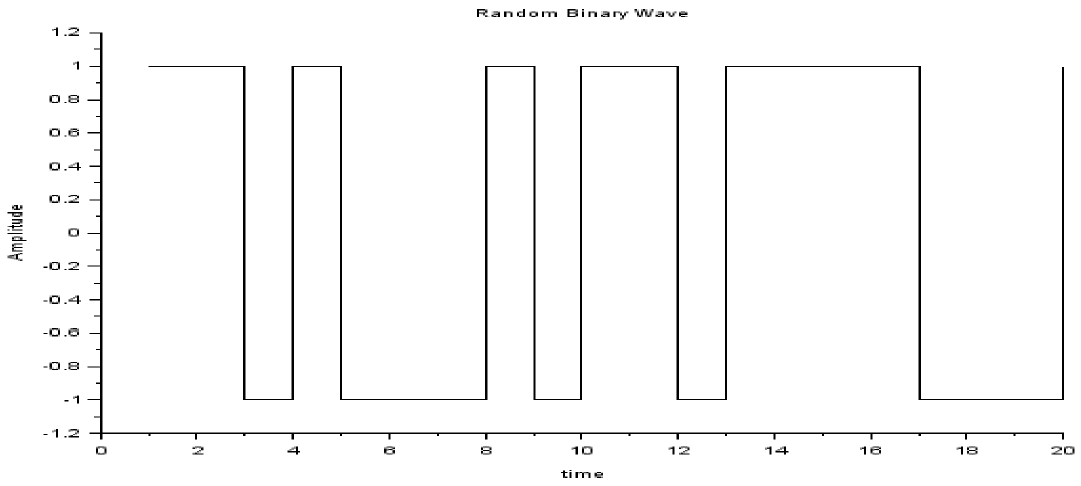



***Fig 7.2: Random Binary Wave Distribution***

**RESULT:** A random binary wave has been plotted for $l = 10$ random binary data as shown in fig 7.2.

## DISCUSSION:

1. What is the importance of random binary wave in communication system?
2. How can we generate random binary data without using randint() ?
3. How the phenomenon of random signals is quite useful in dealing with noise signals?

# EXPERIMENT No. 8

**AIM:** To generate and verify random sequences with arbitrary distributions, means and variances for following:
(a) Rayleigh distributions
(b) Normal distributions: N (0,1).
(c) Gaussion distributions: N (m, x)

**SOFTWARE USED**: Scilab 6.1.0.

## THEORY:

The term random signal is used primarily to denote signals, which have a randomness in its nature source. As an example we can mention the thermal noise, which is created by the random movement of electrons in an electric conductor.
Random signals are those signals whose behavior cannot be predicted using any past reference.
Random signals are those that do not repeat with any definite sequence, but rather must be described in terms of some probability. The characteristics of random signal are primarily determined by its probability distribution function (PDF). There are many types of PDF available to characterize the behavior of random signals as:

1. Gaussian distribution
2. Rayleigh distribution
3. Gamma distribution
4. Uniform Distribution

And many more……..

In this experiment we are mainly concerned about three types of distribution i.e.

1. Gaussian distribution
2. Rayleigh distribution
3. Normal distribution

**Gaussian Distribution:**
Gaussian distribution (also known as normal distribution) is a bell-shaped curve, and it is assumed that during any measurement values will follow a normal distribution with an equal number of measurements above and below the mean value.
The PDF of Gaussian distribution of random signal x is

$$f_x(x) = \frac{1}{\sqrt{2\pi\sigma^2}} \exp\left(-\frac{(x-\mu)^2}{2\sigma^2}\right)$$

where $\sigma^2$: variance of the PDF, $\sigma$ is the standard deviation, $\mu$: mean of the PDF. The PDF of Gaussian distribution is as shown in fig below

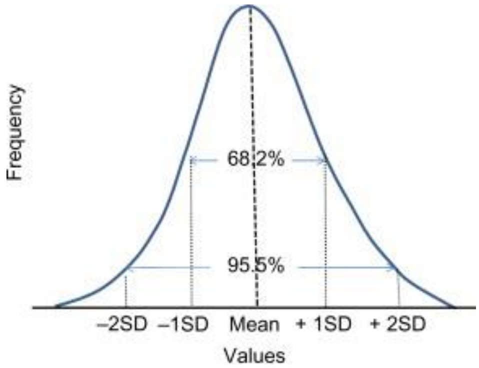



***Fig 8.1: PDF of Gaussian distributed signal***

Communication noise is generally modeled with Gaussian distribution. Noise is assumed to be AWGN (Additive White Gaussian Noise)

***2. Normal Distribution:***

Gaussian distribution with mean =0 and variance=1, is known to be standard normal distribution. The PDF of normal distribution is:

$$f_x(x) = \frac{1}{\sqrt{2\pi}} \exp(-\frac{(x)^2}{2\sigma^2})$$

**3. Rayleigh Distribution:**

In probability theory and statistics, the Rayleigh distribution is a continuous probability distribution for nonnegative-valued random variables. Wireless channel communication and its related fading problem is generally characterized by Rayleigh distribution.

$$f_x(x) = \frac{\mathrm{x}}{\sigma^2} \exp(-\frac{(x)^2}{2\sigma^2}); \ \mathrm{x}>0$$

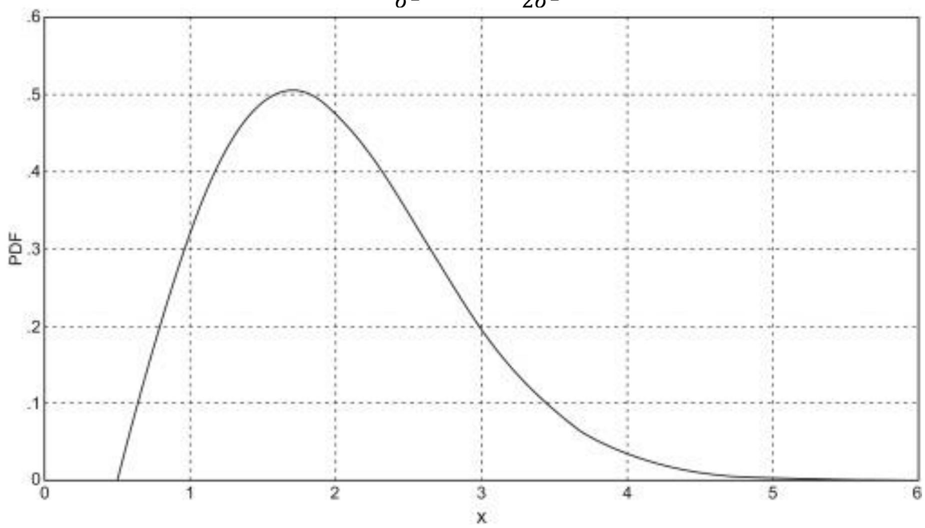



***Fig 8.2: Rayleigh distribution***

In MATLAB, there is a direct command to find distribution. For eg.normpdf(x,mean, var) is used for gaussain distribution and raylpdf(x, r) is used for Rayleigh distribution.

**For Gaussian Distribution**

```
clc;
clear all;
clf;
x=-4:.01:4;
sd=1.2;
m=1;
pdf=1/(sqrt(2*3.14*sd^2)).*exp(-((x-m)^2)/(2*sd^2));
plot(x,pdf);
xlabel('X');
ylabel('Amplitude');
title('Gaussian Distribution');
```

**For Normal Distribution**

```
clc;
clear all;
clf;
x=-4:.01:4;
sd=1;
m=0;
pdf=1/(sqrt(2*3.14*sd^2)).*exp(-((x-m)^2)/(2*sd^2));
plot(x,pdf);
xlabel('X');
ylabel('Amplitude');
title('Gaussian Distribution');
```

**For Rayleigh Distribution**

```
clc;
clear all;
clf;
x=0:.01:5;
sd=1;
r=(x/(sd^2)).*(exp(-(1/2)*(x/sd)^2));
plot(x,r);
xlabel('X');
ylabel('Amplitude');
title('Rayleigh Distribution');
```

**OUTPUT:**

Enter 1 for Gaussian distributon
Enter 2 for Normal distributon
Enter 3 for Rayleigh distributon
enter your choice: 1
enter Your sequence: -2:0.01:2
enter value of mean: 1
enter value of variance: 1

"Do you want to continue press 1 else press 0": 1

Enter 1 for Gaussian distributon

Enter 2 for Normal distributon

Enter 3 for Rayleigh distributon

enter your choice4

"Do you want to continue press 1 else press 0"

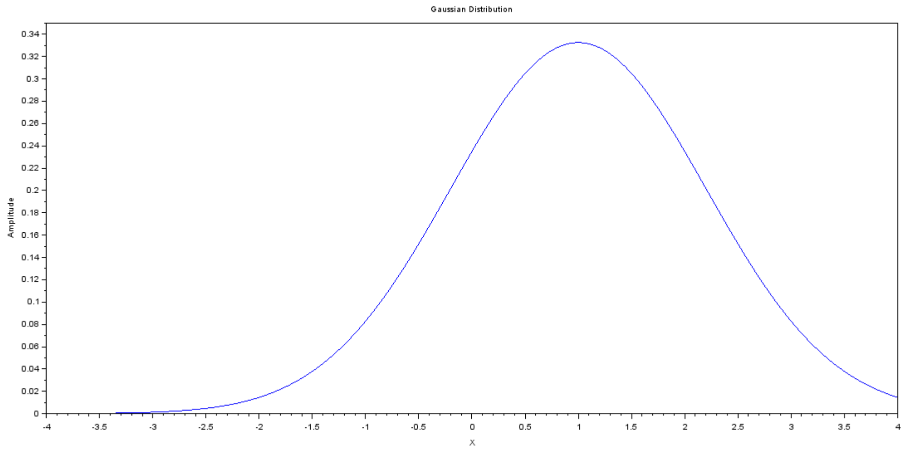



***Fig8.3: Gaussian Distribution with mean=1 and, variance=1***

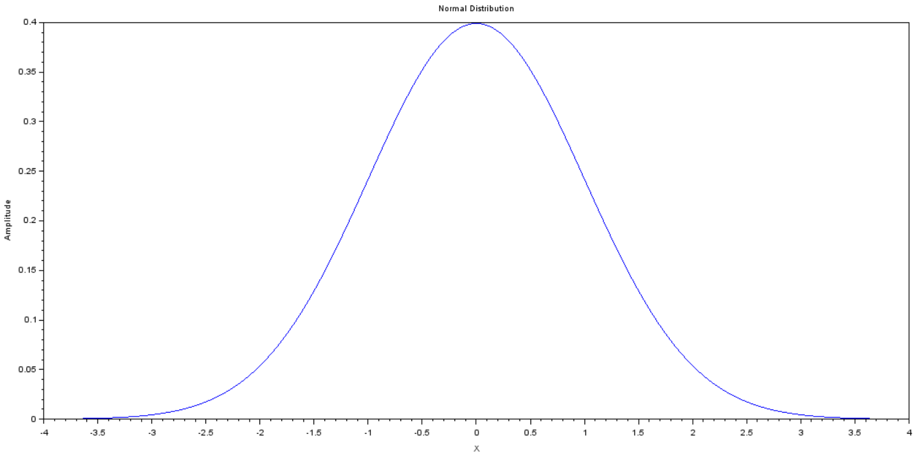



***Fig8.4: Normal Distribution with mean=0 and, variance=1***

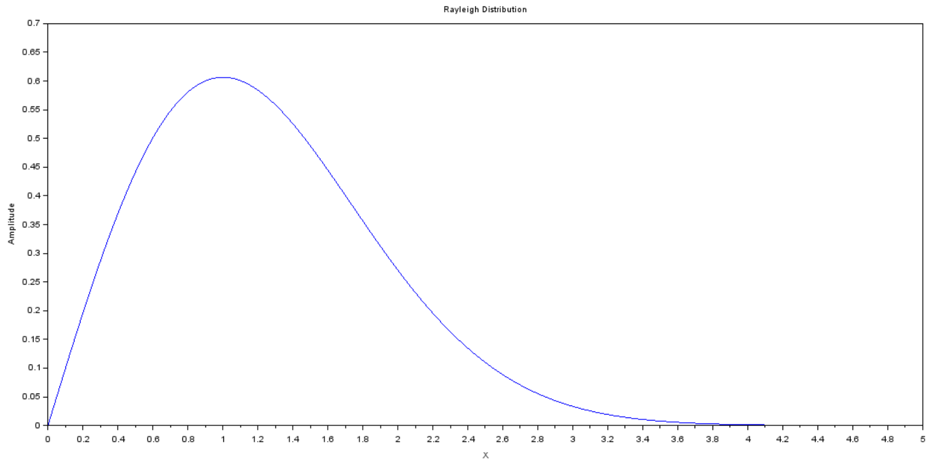



***Fig 8.5: Rayleigh Distribution with mean=0 and, variance=1***

## RESULT:

The three different probability distribution functions of Gaussian, Normal and Rayleigh are plotted in Fig 8.3,8.4 and 8.5 respectively

## DISCUSSION:

**1.** Define AWGN and how its useful in communication system?

**2.** What does the variance of Gaussian distribution specify?

**3.** What does the mean of probability distribution function specify?

**4.** Which random data is modeled by Gaussian and Rayleigh probability distribution?

# EXPERIMENT No. 9

**AIM:** To plot the probability density functions (pdf). Find mean and variance for the above distributions

**PROBLEM STATEMENT**: Find the pdf/ pmf (probability mass function) of the random data of arbitrary length and find its mean and variance.

**SOFTWARE USED**: MATLAB 7.1.

## THEORY:

In probability theory, a **probability density function** (**PDF**), or **density** of a continuous random variable, is a function whose value at any given sample (or point) in the sample space (the set of possible values taken by the random variable) can be interpreted as providing a relative likelihood that the value of the random variable would equal that sample. In a more precise sense, the PDF is used to specify the probability of the random variable falling within a particular range of values, as opposed to taking on any one value. This probability is given by the integral of this variable's PDF over that range—that is, it is given by the area under the density function
Let f(x) = the **probability density function** (p.d.f.) .

The total area under f(x) remains 1.

Thus two conditions for a function f(x) of a continuous variable x to be a valid probability density function are:

$$f(x) \geq 0 \quad \forall x \qquad \text{----(9.1)}$$

$$\int_{-\infty}^{\infty} f(x)\,dx = 1 \qquad \text{----(9.2)}$$

Similarly, in probability and statistics, a **probability mass function** (**PMF**) is a function that gives the probability that a discrete random variable is exactly equal to some value. The probability mass function is often the primary means of defining a discrete probability distribution, and such functions exist for either scalar or multivariate random variables whose domain is discrete.

**% MATLAB code for random no and its pdf**

```
clc;
clear all;
```

```
close all;
display('enter your chpoice for continous pdf or discrete
pmf')
display('Enter 1 for pdf');
display('Enter 2 for pmf');
ch=input('Enter');
switch(ch)
case 1
        x=input('enter Your sequence');
        r=input('enter value of rayleigh factor');
        y=raylpdf(x,r);
         plot(x,y);
xlabel('time');
ylabel('amplitude');
title('RAYLEIGH PDF');
display('mean of random data x is:');
        m=mean(y)
display('variance of random data x is:');
        v=var(y)
case 2

    n=input('number of random data required:');
    x=n*rand(1,n);
    stem(1:n,x)
xlabelx
ylabelpmf/pdf
    title 'pmf/pdf of discrete signal x'
display('mean of random data x is:');
        m=mean(x)
display('variance of random data x is:');
    v=var(x)
otherwise
display('Invalid choice');
end
```

**OUTPUT:**

```
enter your chpoice for continous pdf or discrete pmf
Enter 1 for pdf
Enter 2 for pmf
Enter1
enter Your sequence0:0.01:3
enter value of rayleigh factor1
mean of random data x is:
m =0.3286

variance of random data x is:
```

```
v =0.0393
```

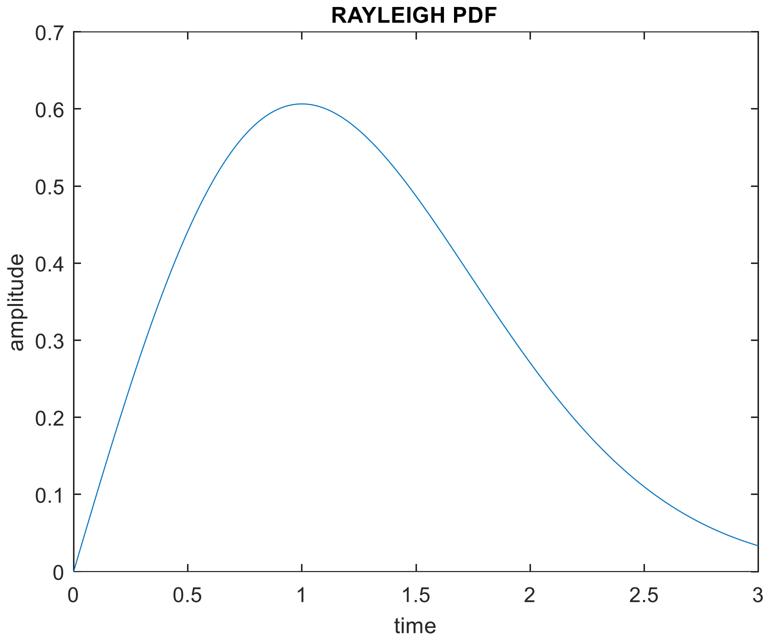



***Fig 9.1 PMF/PDF of random data***

```
enter your chpoice for continous pdf or discrete pmf
Enter 1 for pdf
Enter 2 for pmf
Enter2
number of random data required:10
mean of random data x is:
m =5.8731
variance of random data x is:
v =8.6137
```

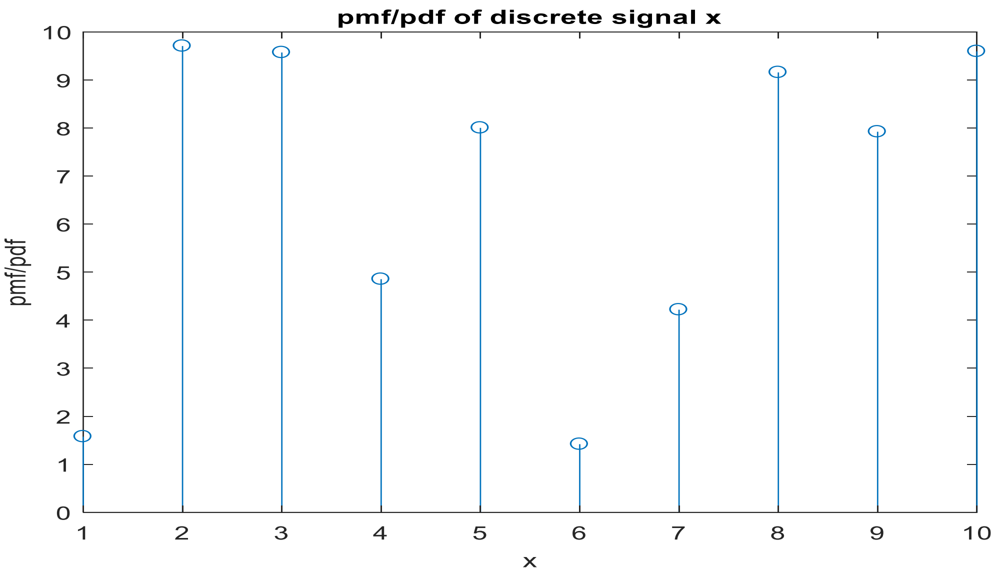



***Fig 9.2: PMF of random data***

**RESULT:** The pdf of Rayleigh distribution is as shown in fig 9.1 and mean and variance of above distribution is 0.3286 and 0.0393 respectively. The PMF of the random data of length (10) is displayed in fig 9.2 and its mean and variance is 5.8731and 8.6137respectively.

## DISCUSSION:

1. What is the importance of random signals in Electronics & communication Engineering?
2. What is the role of AWGN in digital communication system?
3. Which signal is modeled by Rayleigh Distribution?

# EXPERIMENT No. B1

**AIM:** To design a LTI system with specific input and specific output

**PROBLEM STATEMENT**: Design a LTI system whose input is unit impulse signal and output is $(e^{-2t} - e^{-3t})u(t)$. (where u(t) is unit step). Also obtain the magnitude and phase spectrum.

**SOFTWARE USED**: MATLAB 7.1.

## THEORY:

LTI system is Linear and Time-Invariant system whose output is the convolution integral of input and impulse response of the system.
Convolution is defined as an operation which helps to find the output of the LTI (linear and time-invariant) system when the impulse response of the system and the input signal are known. (Impulse response is defined as behavior of system under unit impulse signal which is defined for t=0 only)

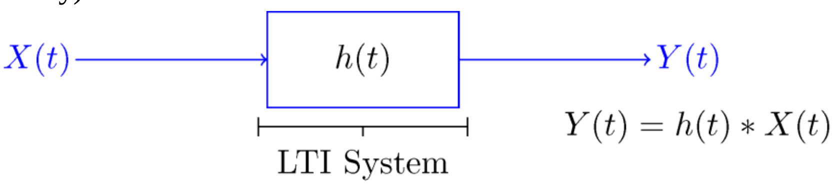



***Fig 11.1 LTI system with output represented as convolution between input and impulse response***

If a continuous-time system is both linear and time-invariant, then the output y(t) is related to the input x(t) and impulse response (h(t)) of the system by the equation given below::

$$y(t) = x(t) * h(t) = \int_{-\infty}^{\infty} x(\tau)h(t - \tau)d\tau \qquad (10.1)$$

Equation 10.1 is also known as convolution integral between x(t) and h(t).

$$Y(s) = X(s)H(s)$$

Where Y(s), H(s) and X(s) are the Laplace transformation of y(t), h(t) and x(t)

**Designing Step:**

**Let input to the system is x(t)and output to the system is y(t)**

$$x(t) = \delta(t)$$

$$y(t) = (e^{-2t} - e^{-3t})u(t)$$

$$y(t) = x(t) * h(t)$$

Taking Laplace Transform

$$Y(s) = X(s)H(s)$$

$$H(s) = Y(s)/X(s)$$

$$y(t) = (e^{-2t} - e^{-3t})u(t)$$

So, $Y(s) = \frac{1}{s+2} - \frac{1}{s+3}$

$$x(t) = \delta(t),$$

So, X(s)=1

$$H(s) = \frac{1}{s+2} - \frac{1}{s+3}$$

Or $H(s) = \frac{1}{(s+2)(s+3)} = \frac{1}{s^2+5s+6}$

**%MATLAB code for to design LTI system:**

```
clear all;
clc;
close all;
  a = [1 5 6];
  b = [1];
w = logspace(-1,1);
h = freqs(b,a,w);
mag = abs(h);
    phase = angle(h);
phasedeg = phase*180/pi;
subplot(2,1,1)
 loglog(w,mag)
   grid on
xlabel'Frequency (rad/s)'
ylabel ‘Magnitude’
subplot(2,1,2),
semilogx(w,phasedeg)
grid on
```

```
xlabel'Frequency (rad/s)',
ylabel'Phase (degrees)'
```

**Output**

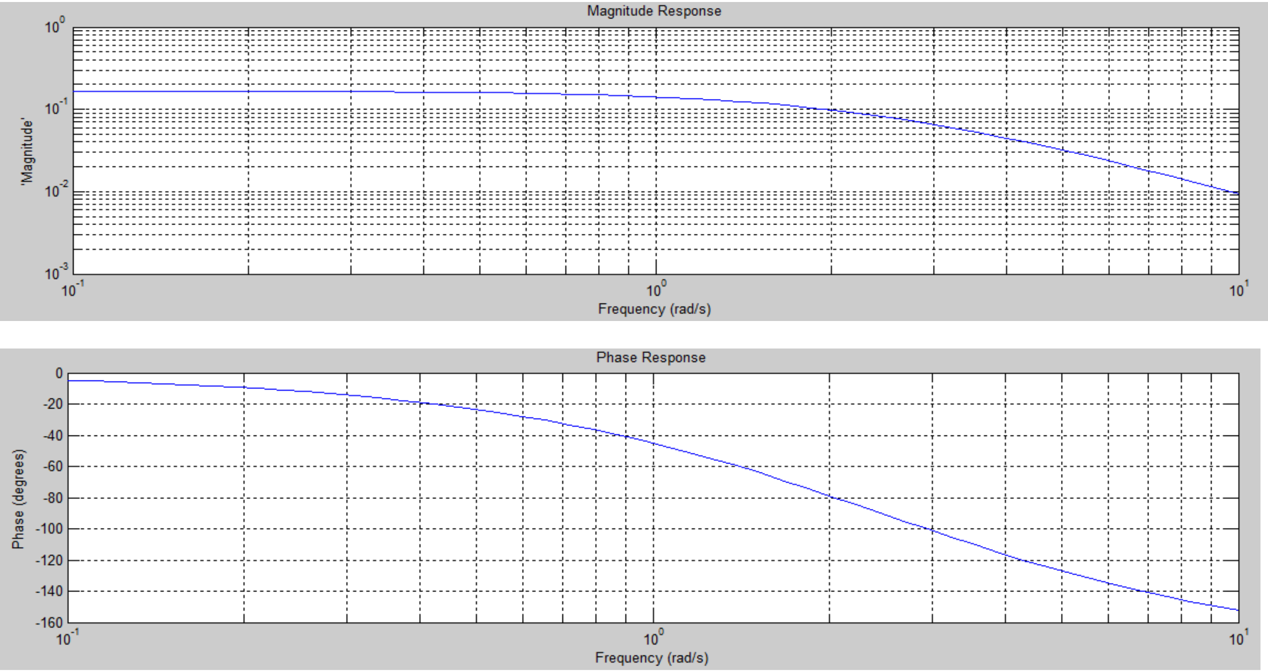



***Fig 10.1: Magnitude and Phase response of LTI system with given input and output.***

**RESULT:**The system response of the system with specified input (impulse response) and output $(e^{-2t} - e^{-3t})u(t)$.)is________________ and its phase and magnitude response is plotted as shown in Fig 10.1

## DISCUSSION:

1. Explain the LTI system and its importance in communication system.
2. Explain the principle of convolution and why it’s important in LTI system.
3. How Laplace transformation is is different from Fourier Transform?
4. Explain the drichlet’s condition for Fourier Transform.

# EXPERIMENT No-B2

**AIM:** Sampling and Reconstruction of a given signal.

**SOFTWARE USED:** Scilab 6.1.0.

## THEORY:

**Theory:** The sampling is extremely important and useful in signal processing and digital communication. The sampling process is usually described in the time domain. With the help of sampling process, a continuous time signal may be completely represented and recovered from the knowledge of samples taken uniformly.

A continuous time signal is first converted to discrete time signal by sampling process. The sufficient number of samples of the signal must be taken so that the original signal is represented in its samples completely. Also, it should be possible to recover or reconstruct the original signal completely from its samples.

**Sampling Theorem Statement:** A continuous time signal may be completely represented in its samples and recovered back if the sampling frequency is $f_s >= 2f_m$. Here $f_s$ is the sampling frequency and $f_m$ is the maximum frequency present in the signal.

The output sample signal is represented by the samples. These samples are maintained with a gap, these gaps are termed as sample period or sampling interval (Ts). And the reciprocal of the sampling period is known as “sampling frequency” or “sampling rate”. The number of samples is represented in the sampled signal is indicated by the sampling rate.

Consider a continuous time signal x(t) whose spectrum is band limited to $f_m$ Hz. Sampling of x(t) at a rate of $f_s$ Hz (Fs samples per second) may be achieved by multiplying x(t) by an impulse train. Therefore it is called as ideal sampling.

The impulse train consists of unit impulse repeating periodically every $T_s$ seconds, where $T_s = 1/f_s$.

The sampled signal may be written as

$$g(t) = x(nT_s) = x(t).\delta_{Ts}(t)$$

The sampled signal in frequency domain is represented as

$$G(\omega) = \frac{1}{T_s}\sum_{n=-\infty}^{\infty} X(\omega - n\omega_s)$$

The spectrum $G(\omega)$ consist of $X(\omega)$ repeating periodically with period $\omega_s$, where

$$\omega_s = \frac{2\pi}{T_s} rad/\sec or f_s = \frac{1}{T_s} Hz$$

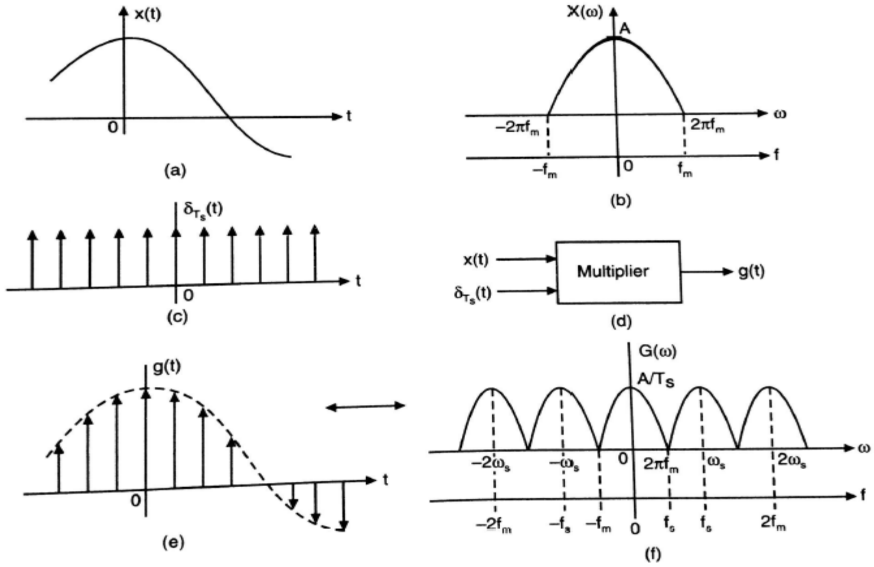



***Fig. 10.1 Sampling Process***

When the sampling rate exactly equal to $2f_m$ samples per second, then it is called Nyquist rate.Nyquist rate is also called the minimum sampling rate.

$$f_s = 2f_m \quad \text{Hz}$$

Similarly, maximum sampling interval is called Nyquist interval

$$Ts = 1/(2f_m) \quad \text{Seconds}$$

**Signal Reconstruction:** The process of reconstructing a continuous time signal x(t) from its samples is known as interpolation.Interpolation gives either approximate or exact recovery of the continuous time signal.

A signal x(t) band limited to $f_m$ Hz can be reconstructed (interpolated) completely from its samples, by passing the sampled signal through an ideal low pass filter of cut-off frequency $f_m$ Hz.

Therefore the filter output to g(t), which is x(t), may be expressed as

$$x_r(t) = x(t) = \sum_{n=-\infty}^{\infty} x(nT_s) sinc(f_s(t - nT_s))$$

This is known as interpolation formula, which provides values of x(t) between samples as a weighted sum of all the sample values.

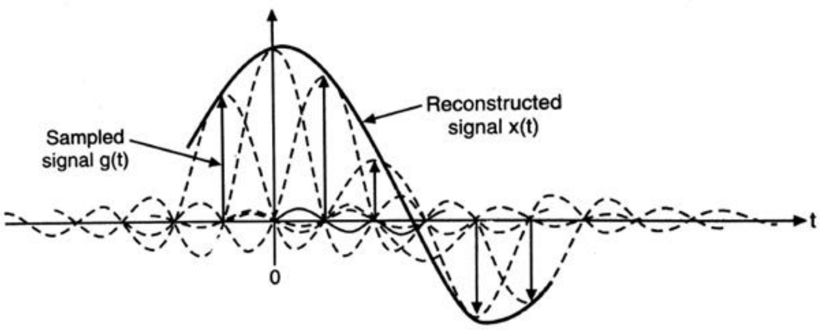



***Fig. 10.2: Signal Reconstruction***

```
% SCILAB code for sampling and reconstruction

clear;
clc;
close;
// Analog Signal
A =1;   //Amplitude
Dt = 0.005;
t = -2:Dt:2;
//Continuous Time Signal
x = sin(2*t);
//Discrete Time Signal
Fs  =input('Enter  the  Sampling  Frequency  in  Hertz');//Fs  =
1Hz,2Hz,4Hz,20Hz,100Hz
Ts = 1/Fs;
nTs = -2:Ts:2;
xs = sin(2*nTs);
// Analog Signal reconstruction
Dt = 0.005;
t = -2:Dt:2;
xr         =         xs        *sinc(Fs*(ones(length(nTs),1)*t-
nTs'*ones(1,length(t))));
//Plotting the original signal and reconstructed signal
subplot(3,1,1);
plot(t,x);
xlabel('time');
ylabel('amplitude');
title('Message Signal');
subplot(3,1,2);
plot2d3(nTs,xs);
xlabel('time');
ylabel('amplitude');
title('Sampled Signal');
subplot(3,1,3);
plot(t,xr);
xlabel('time');
ylabel('amplitude');
title('Reconstructed Signal');
```

## OUTPUT:

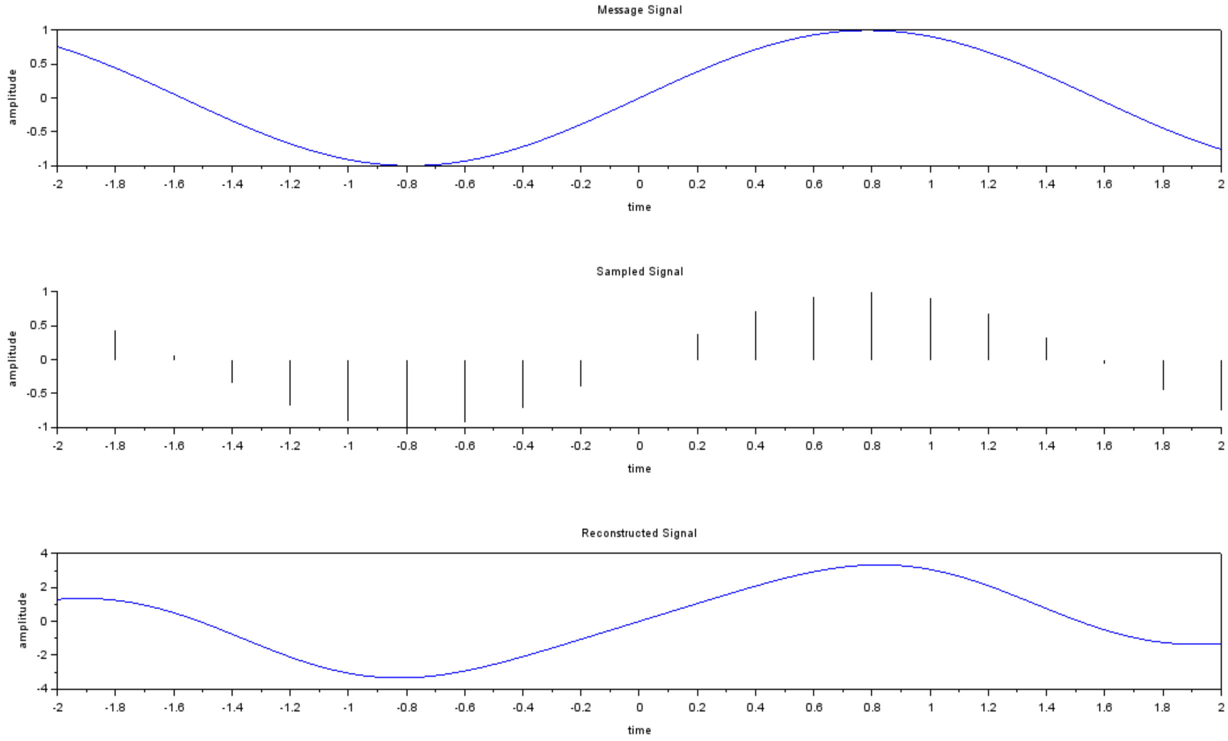



***Fig 10.3: Message signal, Sampled signal (with sampling frequency 5Hz) and Reconstructed Signal***

**RESULT:**Message signal, sampled signal and reconstructed signal is shown in figure 10.3 and the sampling frequency is 5 Hz. Figure shows that message signal is reconstructed at receivers end using interpolation if sampling frequency is fs>2fm.

## DISCUSSION:

Q 1. Explain aliasing effect?

Q 2. Define Nyquist rate and Nyquist interval.

Q 3. Explain demerits of Ideal sampling.

Q 4. What is aperture effect?