



**DIGITAL SIGNAL PROCESSING LAB (EC-618-F)**

**LAB MANUAL**

**VI SEMESTER**

*Department of Electronics & Computer Engg  
Dronacharya College of Engineering  
Khentawas, Gurgaon – 123506*

**LIST OF EXPERIMENTS**

<b>S.NO</b>	<b>NAME OF EXPERIMENT</b>	<b>PAGE NO</b>
<b>1.</b>	<b>To illustrate the simple mathematical expressions in Matlab.</b>	<b>3-7</b>
<b>2.</b>	<b>To represent basic signals(unit step, unit impulse, ramp, exponent, sine and cosine)</b>	<b>8-12</b>
<b>3.</b>	<b>To develop a program for discrete convolution.</b>	<b>13-15</b>
<b>4.</b>	<b>To develop a program for discrete correlation</b>	<b>16-18</b>
<b>5.</b>	<b>To understand stability test.</b>	<b>19-20</b>
<b>6.</b>	<b>To understand sampling theorem</b>	<b>21-23</b>
<b>7.</b>	<b>To develop a program for computing z-transform and inverse Z-transform of a transfer function</b>	<b>24-25</b>
<b>8.</b>	<b>To draw pole zero diagram for a function in z domain &amp; display the frequency response of given z- transform</b>	<b>26-28</b>
<b>9.</b>	<b>To design analog filter (low-pass, high pass, band-pass, band-stop).</b>	<b>29-30</b>
<b>10.</b>	<b>Plot the magnitude and phase of the frequency response of the given digital filter using freqz function: <math>Y(n) = 0.2x(n) + 0.52y(n-1) - 0.68(y(n-2))</math></b>	<b>31-33</b>

**EXPERIMENT NO. 1**

**AIM:** -TO ILLUSTRATE THE SIMPLE MATHEMATICAL EXPRESSIONS IN MATLAB.

**PROGRAM:-**

```
1. clc;
   clear all;

   close all;

   x=2^5

   y=x-1

   z=x/y
```

**OUTPUT :-**

```
x= 32
y= 31
z=1.0323
```

```
2. clc;
   clear all;

   close all;

   x=3*sqrt(5)-1

   y=(sqrt(5)+1)^2

   z=x/y-1
```

**OUTPUT :-**

```
x= 5.7082
y= 10.4721
z=-0.4549
```

```
3. clc;
   clear all;

   close all;

   r= pi^(1/3)-1
   area=pi*r^2
```

**OUTPUT :-**            r= 0.4646  
                              area=0.6781

```
4. clc;
   clear all;

   close all;

   r= sqrt(163)
   t= exp(pi*r)
```

**OUTPUT :-**            r= 12.7671  
                              t= 2.6254e+017

```
5. clc;
   clear all;

   close all;

   x=(sin(pi/6))^2
   y=(cos(pi/6))^2

   t=x+y
```

**OUTPUT :-**            x= 0.2500  
                              y= 0.7500  
                              t=1

```
6. clc;
   clear all;

   close all;

   t=pi/4
   x=exp(i*t)
```

**OUTPUT :-**            t= 0.7854  
                              x= 0.7071+0.7071i

```
7. clc;
   clear all;

   close all;

   t=exp (pi/(4*i))
```

**OUTPUT :-**            t= 32  
                              y= 31  
                              z=0.7071+0.7071i

```
8. clc;
   clear all;

   close all;

   t=1:1:10
   z= sin(t.^2)/ (t.^2)
```

**OUTPUT :-**    t= 1    2    3    4    5    6    7    8    9    10  
                  z= -0.0052

## QUESTIONS/ANSWERS

Q1. What is a MATLAB?

A1. **MATLAB** (**matrix laboratory**) is a numerical computing environment and fourth-generation programming language

Q2. What are the various functions of MATLAB?

A2. MATLAB allows matrix manipulations, plotting of functions and data, implementation of algorithms, creation of user interfaces, and interfacing with programs written in other languages, including C, C++, Java, and Fortran. Star and ring are the two examples of hybrid topology

Q3. What is the difference between an array and a vector?

A3. A *vector* refers to a one dimensional ( $1 \times N$  or  $N \times 1$ ) matrix, commonly referred to as an array in other programming languages. A *matrix* generally refers to a 2-dimensional array, i.e. an  $m \times n$  array where  $m$  and  $n$  are greater than 1.

Q4. What are the various windows in MATLAB?

A4. Command window, Editor window, Launchpad Window

Q5. What are variables?

A5. Variables are assigned numerical values by typing the expression directly, for example, typing

`a = 1+2` yields: `a = 3`

The answer will not be displayed when a semicolon is put at the end of an expression, for example type `a = 1+2;`

Q6. What are the various arithmetic operators used in MATLAB?

A6. MATLAB utilizes the following arithmetic operators:

- + Addition
- Subtraction
- \* multiplication
- / Division
- ^ power operator
- ' Transpose

Q7. What are the various special type of matrices used in MATLAB?

A7.

null matrix:	<b>M = [ ];</b>
nxm matrix of zeros:	<b>M = zeros(n,m);</b>
nxm matrix of ones:	<b>M = ones(n,m);</b>
nxn identity matrix:	<b>M = eye(n);</b>

Q8. Is MATLAB a case sensitive?

A8. Yes. "a" and "A" are two different names.

Q9. How we define comment statements in MATLAB?

A9. Comment statements are preceded by a "%"..

Q10. What are M-files ?

A10. M-files are macros of MATLAB commands that are stored as ordinary text files with the extension "m", that is *filename.m*. An M-file can be either a function with input and output variables or a list of commands. All of the MATLAB examples in this textbook are contained in M-files that are available at the Math Works ftp site, <ftp.mathworks.com> in the directory `pub/books/heck`.

## **EXPERIMENT NO. 2**

**AIM:-** TO REPRESENT BASIC SIGNALS(UNIT STEP, UNIT IMPULSE, RAMP, EXPONENT, SINE AND COSINE)

### **PROGRAM:-**

\*\* for unit step, unit impulse, ramp and exponent

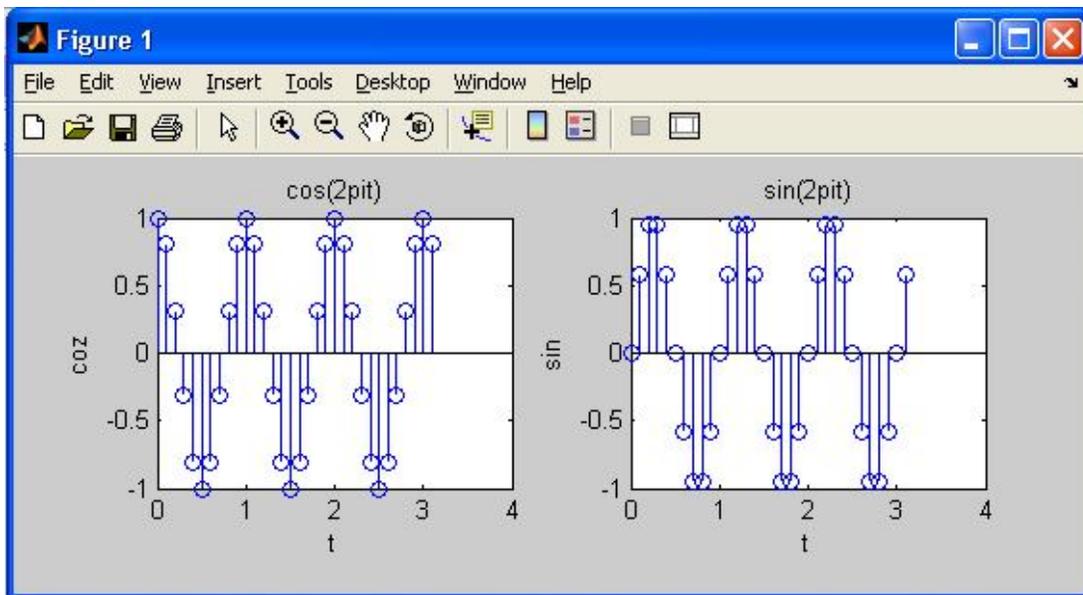
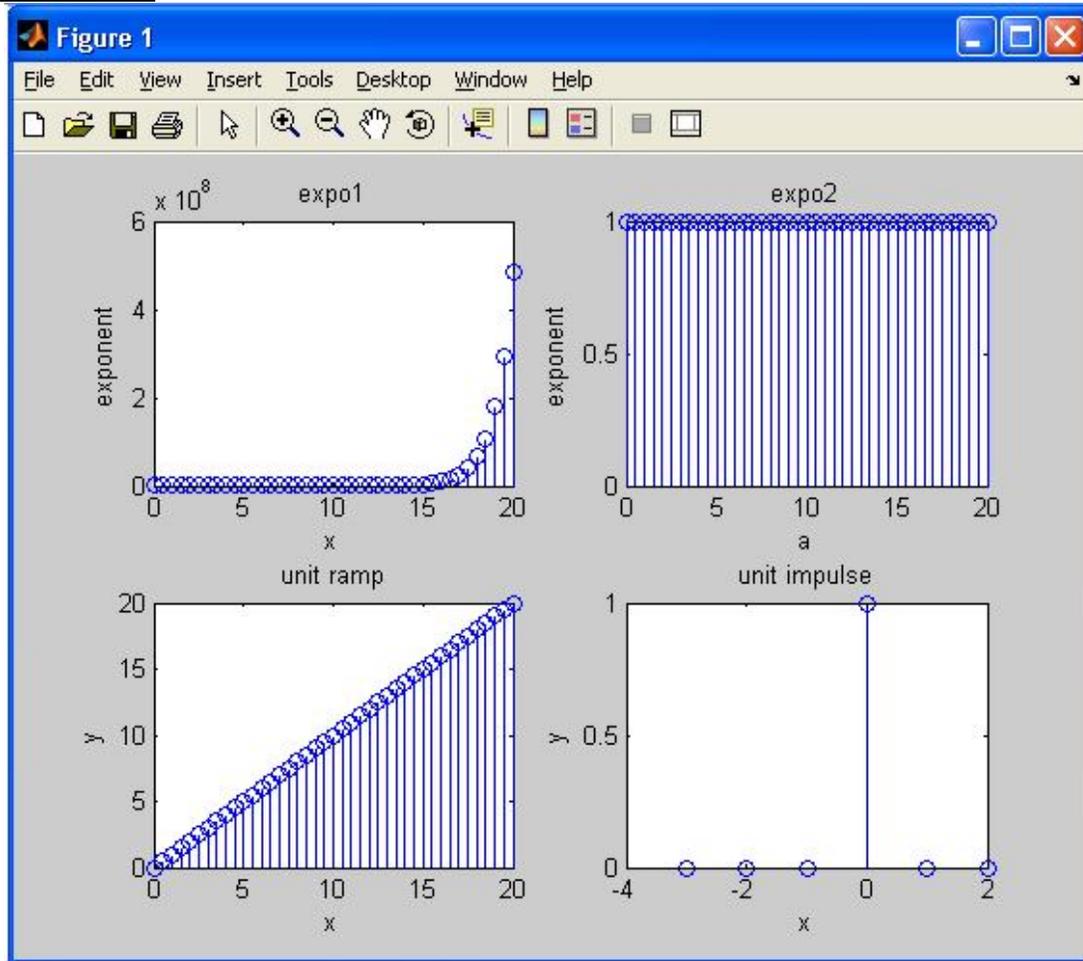
```
clc;
clear all;
close all;
n=[0:0.5:20];
x=input('enter value of x');
y=exp(x*n);
subplot(2,2,1);
stem(n,y);
xlabel('x');
ylabel('exponent');
title('expo1');
%clc;
%clear all;
%close all;
n=[0:0.5:20];
a=input('enter value of a');
y=(a).^n;
subplot(2,2,2);
stem(n,y);
xlabel('a');
ylabel('exponent');
title('expo2');
%clc;
%clear all;
%close all;
n=[0:0.5:20];
subplot(2,2,3);
stem(n,n);
xlabel('x');
ylabel('y');
title('unit ramp');
```

```
%clc;
%clear all;
%close all;
t=[-3:1:2];
n1=[zeros(1,3),ones(1,1),zeros(1,2)];
subplot(2,2,4);
stem(t,n1);
xlabel('x');
ylabel('y');
title('unit impulse');
```

\*\*for sine and cosine

```
clc;
clear all;
close all;
t=[0:0.1:pi];
y=cos(2*pi*t);
subplot(2,2,1);
stem(t,y);
xlabel('t');
ylabel('coz');
title('cos(2pit)');
%clc;
%clear all;
%close all;
t=[0:0.1:pi];
y=sin(2*pi*t);
subplot(2,2,2);
stem(t,y);
xlabel('t');
ylabel('sin');
title('sin(2pit)');
```

**OUTPUT:-**



## QUESTIONS/ANSWERS

Q1. Which command is used to draw a continuous waveform?

A2 The command most often used for plotting is plot, which creates linear plots of vectors and matrices; plot(t,y) plots the vector t on the x-axis versus vector y on the y-axis

Q2 Which command is used to draw a discrete waveform?

A2 For discrete-time signals, use the command stem which plots each point with a small open circle and a straight line. To plot y[k] versus k, type stem(k,y).

Q3. Which command is used to plot two or more graphs on the same set of axes?

A3. To plot two or more graphs on the same set of axes, use the command plot(t1,y1,t2,y2), which plots y1 versus t1 and y2 versus t2.

Q4. Which command is used to label the axes?

A4 To label your axes, type e.g.

```
xlabel('time (sec)')  
ylabel('step response')
```

Q5 Which command is used to give a title name to the plot?

A5 To give the plot a title, type

```
Title('My Plot')
```

Q6 Which command is used to add a grid to the plot?

A6 To add a grid to your plot to make it easier to read. Type

```
grid
```

Q7. Which command is used to plot more than one graph on the screen?

A7. To plot more than one graph on the screen, use the command subplot(mnp) which partitions the screen into an m×n grid where p determines the position of the particular graph counting the upper left corner as p=1. For example,

```
subplot(2,1,1)
```

Q8. For autoscaling of the axes which command is used?

A8: The auto scaling of the axes can be done by using the axis command after the plotting command:

```
axis([xminxmaxyminymax]);
```

where xmin, xmax, ymin, and ymax are numbers corresponding to the limits you desire for the axes. To return to the automatic scaling, simply type axis.

Q9. How the colour of plot can be changed?

A9 There are options on the line type and the color of the plot which are obtained using plot (t,y,'option'). The linetype options are '-' solid line (default), '--' dashed line, '-.' dot dash line, ':' dotted line. The points in y can be left unconnected and delineated by a variety of symbols: + .\* o x. The following colors are available options:

- r Red
- b Blue
- g Green
- w White
- k Black

Q10. Which command is used to make the axis equal?

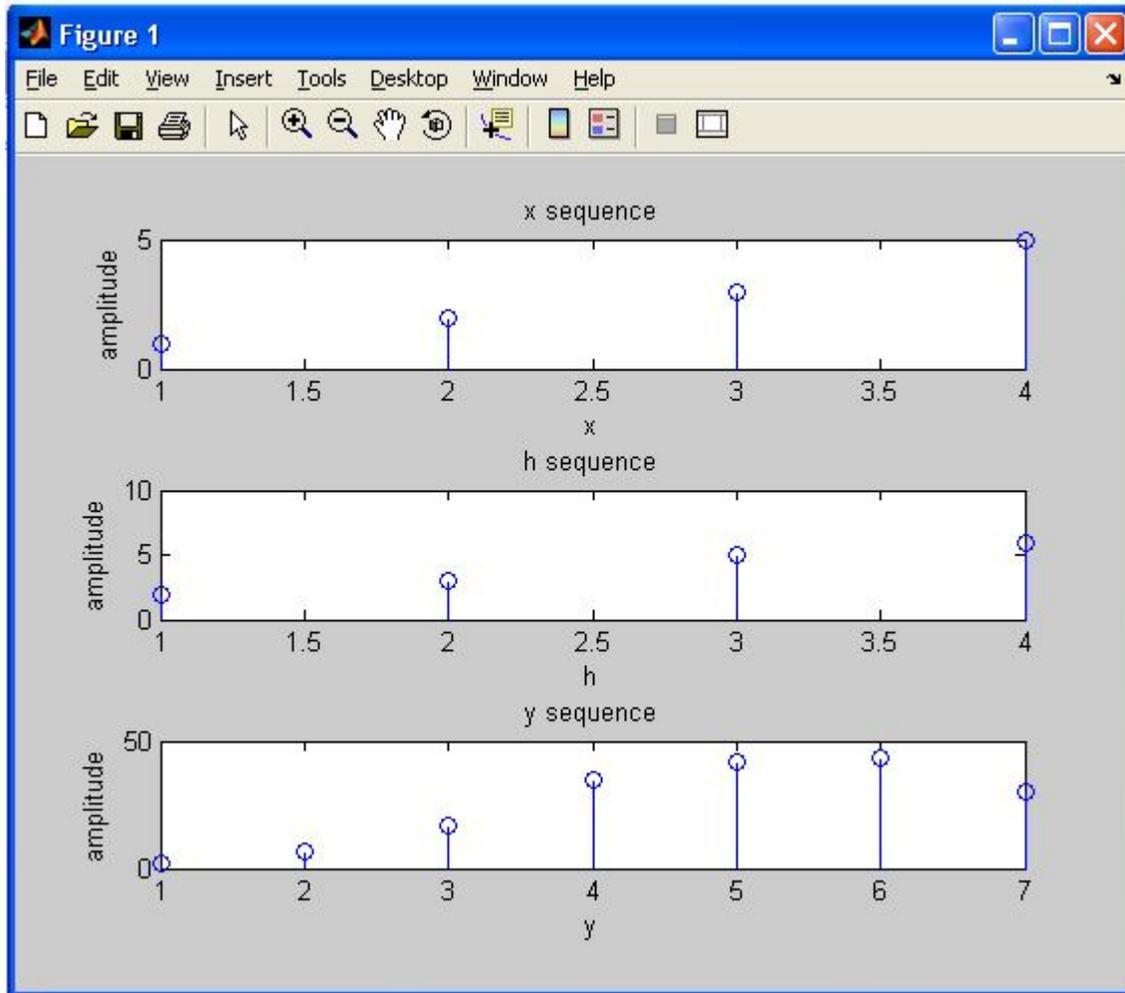
A10: axis ('equal')

### **EXPERIMENT NO. 3**

**AIM:-** TO DEVELOP A PROGRAM FOR DISCRETE CONVOLUTION.

**PROGRAM:-**

```
clc,  
clear all;  
close all;  
x=[1 2 3 5];  
h=[2 3 5 6];  
y=conv(x,h);  
subplot(3,1,1);  
stem(x);  
xlabel('x');  
ylabel('amplitude');  
title('x sequence');  
subplot(3,1,2);  
stem(h);  
xlabel('h');  
ylabel('amplitude');  
title('h sequence');  
subplot(3,1,3);  
stem(y);  
xlabel('y');  
ylabel('amplitude');  
title('y sequence');
```

**OUTPUT:-****QUESTIONS/ANSWERS**

Q1. What is convolution?

A1 In mathematics and, in particular, functional analysis, **convolution** is a mathematical operation on two functions  $f$  and  $g$ , producing a third function that is typically viewed as a modified version of one of the original functions, giving the area overlap between the two functions as a function of the amount that one of the original functions is translated.

Q2. What are the applications of convolution?

A2 It has applications that include probability, statistics, computer vision, image and signal processing, electrical engineering, and differential equations.

Q3. What is the difference between circular convolution and discrete convolution?

A3 the circular convolution can be defined for periodic functions (that is, functions on the circle), and the discrete convolution can be defined for functions on the set of integers.

Q4. What is deconvolution?

A4 Computing the inverse of the convolution operation is known as deconvolution.  
N two sequences

Q5 What is the symbol for convolution?

A5. The convolution of f and g is written  $f * g$ , using an asterisk or star. It is defined as the integral of the product of the two functions after one is reversed and shifted.

Q6 Which command is used to find convolution between two sequences?

A6. If x & h are two sequences then the command used to find convolution between these sequences can be represented by `Conv(x,h)`.

Q7. What are the various special type of matrices used in MATLAB?

A7.

null matrix:	$\mathbf{M} = [ \ ];$
nxm matrix of zeros:	$\mathbf{M} = \mathbf{zeros}(n,m);$
nxm matrix of ones:	$\mathbf{M} = \mathbf{ones}(n,m);$
nxn identity matrix:	$\mathbf{M} = \mathbf{eye}(n);$

Q8. Does convolution obey the algebraic properties?

A8. Yes. It satisfies associative, commutative, distributive algebraic properties.

Q9. What is a convolution matrix?

A9. A convolution matrix is a matrix, formed from a vector, whose inner product with another vector is the convolution of the two vectors.

Q10 Generate a simple convolution matrix.

A10. If

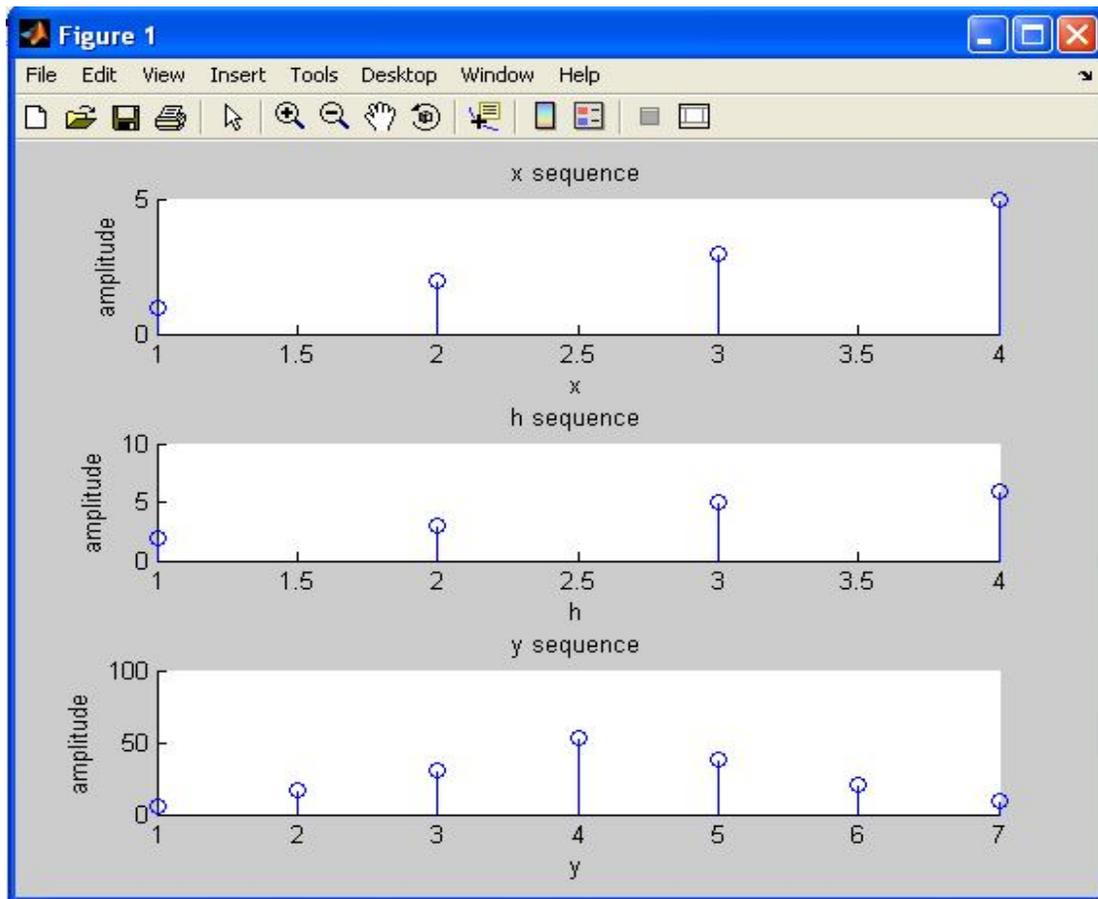
$h = [1 \ 2 \ 3 \ 2 \ 1];$   
`convmtx(h,7);`

## **EXPERIMENT NO. 4**

**AIM:-** TO DEVELOP A PROGRAM FOR DISCRETE CORRELATION

**PROGRAM:-**

```
clc,  
clear all;  
close all;  
x=[1 2 3 5];  
h=[2 3 5 6];  
y=xcorr(x,h);  
subplot(3,1,1);  
stem(x);  
xlabel('x');  
ylabel('amplitude');  
title('x sequence');  
subplot(3,1,2);  
stem(h);  
xlabel('h');  
ylabel('amplitude');  
title('h sequence');  
subplot(3,1,3);  
stem(y);  
xlabel('y');  
ylabel('amplitude');  
title('y sequence');
```

**OUTPUT:-****QUESTIONS/ANSWERS**

Q1. What is correlation?

A1 *Correlation* quantifies the strength of a linear relationship between two variables. When there is no correlation between two variables, then there is no tendency for the values of the variables to increase or decrease in tandem. Two variables that are uncorrelated are not necessarily independent, however, because they might have a nonlinear relationship.

Q2. What are the applications of correlation?

A2 It has applications that compute and plot the sample autocorrelation function (ACF) of a univariate, stochastic time series.

Q3. What is auto-correlation?

A3 The autocorrelation function of a random signal describes the general dependence of

the values of the samples at one time on the values of the samples at another time.

Q4. What is cross-correlation?

A4 The cross correlation function however measures the dependence of the values of one signal on another signal.

Q5 Which command is used to find the correlation coefficients

A5.  $r = \text{corr2}(A,B)$  computes the correlation coefficient between A and B, where A and B are matrices or vectors of the same size.

Q6 Which command is used to find correlation between two sequences?

A6 If x & y are two sequences then

$c = \text{xcorr}(x,y)$

Q7. What command is used to find cross-correlation of two matrices?

A7:  $C = \text{xcorr2}(A,B)$  returns the cross-correlation of matrices A and B with no scaling.  $\text{xcorr2}$  is the two-dimensional version of  $\text{xcorr}$ . It has its maximum value when the two matrices are aligned so that they are shaped as similarly as possible.

Q8. Is convolution obeys the algebraic properties?

A8. Yes. It satisfies associative , commutative , distributive algebraic properties.

Q9. What is a convolution matrix?

A9. A convolution matrix is a matrix, formed from a vector, whose inner product with another vector is the convolution of the two vectors.

Q10 Generate a simple convolution matrix.

A10. If

$h = [1 \ 2 \ 3 \ 2 \ 1];$   
 $\text{convmtx}(h,7);$

## **EXPERIMENT NO. 5**

**AIM:** - TO UNDERSTAND STABILITY TEST

**PROGRAM:** -

```
%Program for stability test

clc;

clear all;

close all;

b=input('enter numerator coefficients ');

c= input('enter denominator coefficients ');

sys=tf(b/c);

z=isstable(sys);

if(z==1)

disp("stable system");

else

disp("Non-stable System");

end
```

**Output:** -

```
enter the numerator coefficients [1]

enter the denominator coefficients [1 1]

'stable system'
```

**Questions/Answers**

Q1: The equation  $s^2 + s^3 + 3s^2 + 5s + 10 = 0$  has

(A) one (B) two (C) three (D) four roots in the left half of s-plane.

Ans: (B)

Q2: The Nyquist plot of the loop transfer function  $G(s)H(s)$  of a closed-loop system encloses the  $(-j\omega, 1)$  point in the  $G(s)H(s)$  plane, the gain margin of the system is

(A) zero. (B) greater than zero.

(C) less than zero. (D) infinity.

Ans: (C)

Q3: The LVDT is primarily used for the measurement of

(A) displacement (B) velocity

(C) acceleration (D) humidity

Ans: (A)

Q5: A system with gain margin close to unity or a phase margin close to zero is

(A) highly stable. (B) oscillatory.

(C) relatively stable. (D) unstable.

Ans: (C)

Q6: Polar plot of  $G(j\omega) = 1/[j\omega(1 + j\omega\tau)]$

(A) crosses the negative real axis.

(B) crosses the negative imaginary axis.

(C) crosses the positive imaginary axis.

(D) None of the above.

Ans: (D)

Q7: For a type one system, the steady – state error due to step input is equal to

(A) infinite. (B) zero.

(C) 0.25. (D) 0.5.

Ans: (B)

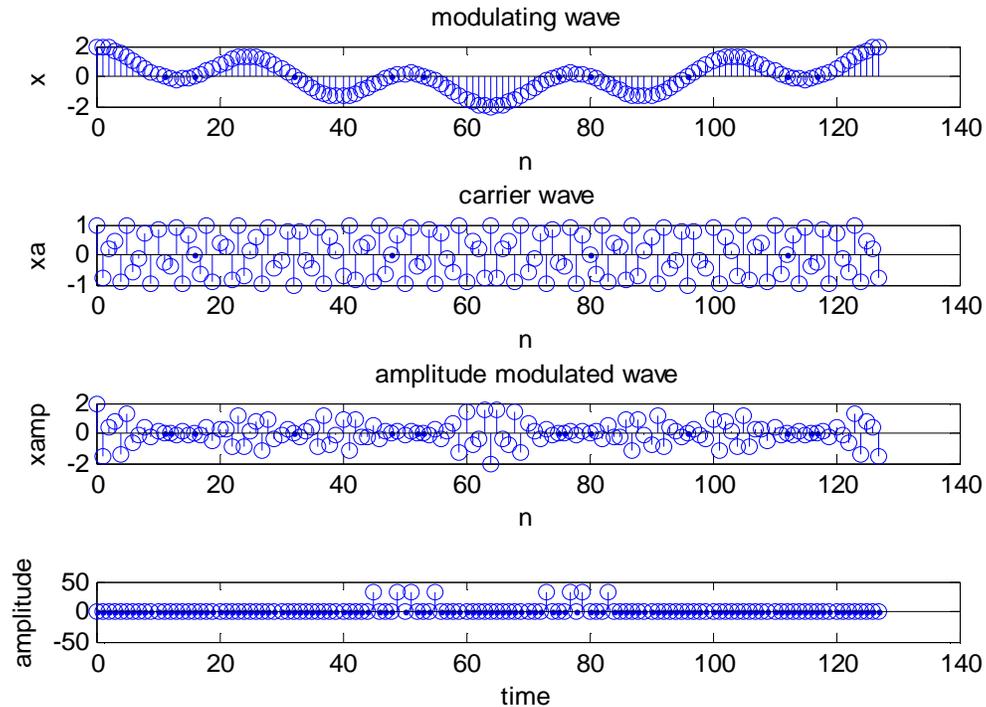
Q8: Which command is used to find stability of a system?

A8: `isstable(sys)`

**EXPERIMENT NO. 6****AIM:** - TO UNDERSTAND SAMPLING THEOREM**PROGRAM:** -

% Program to understand sampling theorem

```
clc;
clearall;
closeall;
f1=1/128;
f2=5/128;
n=0:127;
fc=50/128;
x=cos(2*pi*f1*n)+cos(2*pi*f2*n);
xa=cos(2*pi*fc*n);
xamp=x.*xa;
subplot(4,1,1);
stem(n,x);
title('modulating wave');
xlabel('n');
ylabel('x');
subplot(4,1,2);
stem(n,xa);
title('carrier wave');
xlabel('n');
ylabel('xa');
subplot(4,1,3);
stem(n,xamp);
title('amplitude modulated wave');
xlabel('n');
ylabel('xamp');
n1=128;
xam=fft(xamp,n1);
subplot(4,1,4);
stem(n,xam);
xlabel('time');
ylabel('amplitude')
```



**Question /Answers**

**Q.1** The discrete-time signal  $x(n) = (-1)^n$  is periodic with fundamental period

- (A) 6 (B) 4 (C) 2 (D) 0

Ans: C Period = 2

**Q.2** The Fourier transform of the exponential signal  $e^{j\omega_0 t}$

- (A) a constant (B) a rectangular gate. (C) an impulse. (D) a series of impulses.

Ans: C Since the signal contains only a high frequency  $\omega_0$  its FT must be an impulse at  $\omega = \omega_0$

**Q3:**A band pass signal extends from 1 KHz to 2 KHz. The minimum sampling frequency needed to retain all information in the sampled signal is

- (A)1 KHz. (B) 2 KHz. (C) 3 KHz. (D) 4 KHz.

Ans: B Minimum sampling frequency = 2(Bandwidth) = 2(1) = 2 kHz

**Q4:** An analog signal has the spectrum shown in Fig. The minimum sampling rate needed to completely represent this signal is

- (A) 3 KHz . (B) 2 KHz . (C) 1 KHz . (D) 5.0 KHz .

Ans: C For a band pass signal, the minimum sampling rate is twice the bandwidth, which is 0.5kHz here.

Q5:What are the major classifications of the signal?

A5:(i) Discrete time signal  
(ii) Continuous time signal

Q6:Define discrete time signals and classify them.

A6:Discrete time signals are defined only at discrete times, and for these signals, the independent variable takes on only a discrete set of values.

Classification of discrete time signal:

1. Periodic and A periodic signal
2. Even and Odd signal

Q7:What are the different types of signal representations?

A7:a. Graphical representation b. Functional representations c. Tabular representation d. Sequence representation.

Q8:What are all the blocks are used to represent the CT signals by its samples?

A8:\* Sampler  
\* Quantizer

Q9:Define sampling process.

A9:Sampling is a process of converting Ct signal into Dt signal.

Q10:Mention the types of sampling.

A10:\* Up sampling  
\* Down sampling

Q11: What is meant by quantizer?

A11: It is a process of converting discrete time continuous amplitude into discrete time discrete amplitude

## **EXPERIMENT NO. 7**

**AIM :** TO FIND THE Z-TRANSFORM & INVERSE Z-TRANSFORM OF A EXPRESSION

**Program:**

```
%X(n) = [1/16^n]u(n)
syms z n
a=ztrans(1/16^n)

% X(Z) = 3*Z / (Z+1)
syms Z n
y=iztrans(3*Z/(Z+1))
```

**Result:**

```
a =
16*z/(16*z-1)

y=
3*(-1)^n
```

**QUESTIONS/ANSWERS**

Q1: Define system function?

A1: The ratio between z transform of output signal  $y(z)$  to z transform of input signal  $x(z)$  is called system function of the particular system.

Q2: List out the types of quantization process.

A2: \* Truncation

\* Rounding

Q3: Define truncation.

A3: Truncating the sequence by multiplying with window function to get the finite value

Q4: Define Transfer function of the DT system.

A4: The Transfer function of DT system is defined as the ratio of Z transform of the system output to the input. That is,  $H(z) = Y(z)/X(z)$ .

Q5: Define a causal system.

A5: The causal system generates the output depending upon present & past inputs only. A causal system is non anticipatory.

Q6: Define z transform?

A6: The Z transform of a discrete time signal  $x(n)$  is defined as,

$$X(z) = \sum_{n=-\infty}^{\infty} x(n) z^{-n}$$

Where,  $z$  is a complex variable.

Q7: What is meant by ROC?

A7: The region of convergence (ROC) is defined as the set of all values of  $z$  for which  $X(z)$  converges

Q8: What are the properties of ROC?

A8: a. The roc is a ring or disk in the  $z$  plane centered at the origin.

b. The roc cannot contain any pole.

c. The roc must be a connected region

d. The roc of an LTI stable system contains the unit circle

Q9: . Explain the linearity property of the  $z$  transform

A9: If  $z\{x_1(n)\}=X_1(z)$  and  $z\{x_2(n)\}=x_2(z)$  then,

$z\{ax_1(n)+bx_2(n)\}=aX_1(z)+bX_2(z)$

$a$  &  $b$  are constants.

Q10: What are the conditions of stability of a causal system?

A10: All the poles of the system are within the unit circle. The sum of impulse response for all values of  $n$  is bounded.

## EXPERIMENT NO.8

**AIM :** TO DRAW POLE ZERO DIAGRAM FOR A FUNCTION IN Z DOMAIN & DISPLAY THE FREQUENCY RESPONSE OF GIVEN Z- TRANSFORM

### PROGRAM:

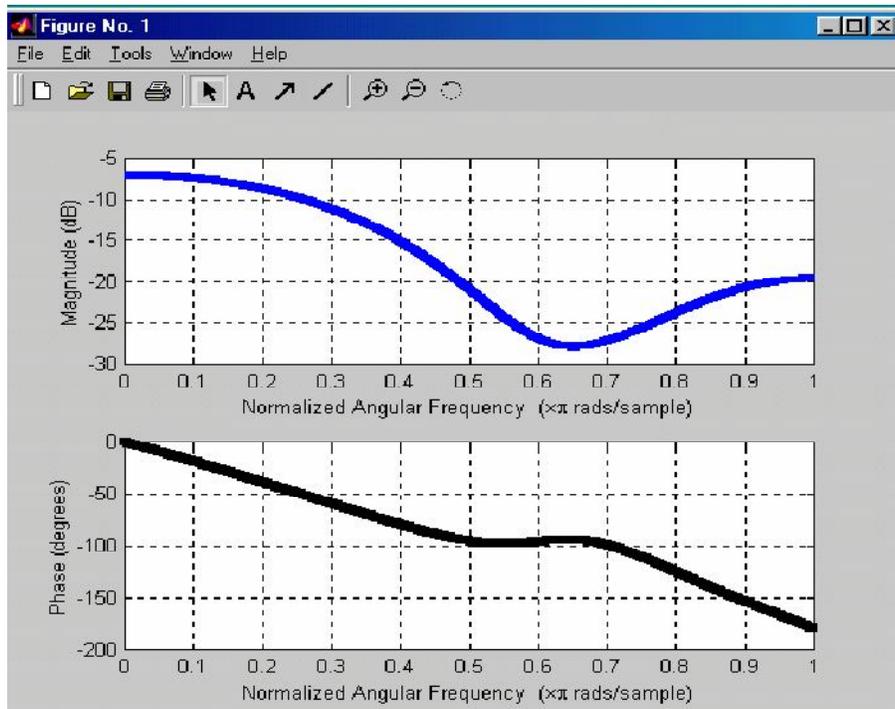
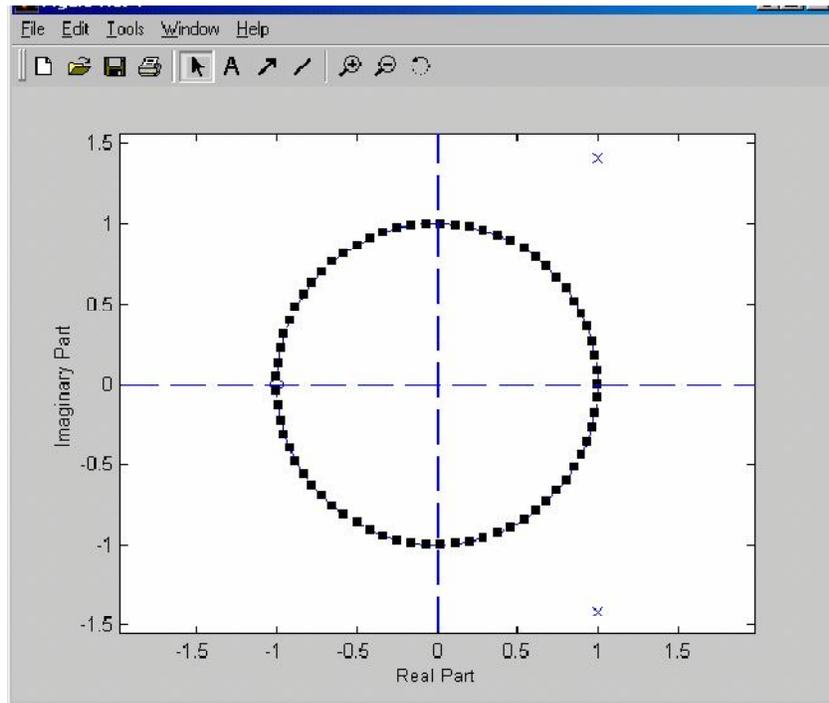
```
%X(Z) = [Z-2+ Z-1] / [1-2Z-1+3Z-2]  
b=[0 1 1 ]  
a= [1 -2 +3]  
roots(a)  
roots(b)  
zplane(b,a);  
% X(Z) = [2+ 5Z-1+9Z-2+5Z-3+3Z-4]/ [5+ 45Z-1+2Z-2+Z-3+Z-4]
```

```
b=[2 5 9 5 3]  
a= [5 45 2 1 1]  
freqz(b,a);
```

### RESULT:

```
ans =  
1.0000 + 1.4142i  
1.0000 - 1.4142i  
ans =  
-1
```

**OUTPUT**



**QUESTIONS/ANSWER:**

Q1:What is the need for Z-transform?

A1:Z-transform is used for analysis the both periodic and a periodic signals.

Q2:Give the Z-transform of unit sample sequence  $\delta(n)$ .

A2: $Z[\delta(n)] = 1$

Q3:Define zeros.

A3:The zeros of the system  $H(z)$  are the values of  $z$  for which  $H(z) = 0$ .

Q4: Define poles.

A4:The poles of the system  $H(z)$  are the values of  $z$  for which  $H(z) = \infty$

Q5:What is the z-transform of  $A \delta(n-m)$  ?

A5: $Z[A \delta(n-m)] = AZ^{-m}$

Q6: State the convolution properties of Z transform?

A6:The convolution property states that the convolution of two sequences in time domain is equivalent to multiplication of their Z transforms.

Q7:What is the z transform of  $x(n-m)$ ?

A7: By time shifting property

$Z[x(n-m)] = Z^{-m} X(z)$

Q8:Obtain the inverse z transform of  $X(z) = \frac{1}{z-a}, |z| > |a|$  ?

A8:Given  $X(z) = \frac{z^{-1}}{1-az^{-1}}$

By time shifting property

$x(n) = a^n u(n-1)$

Q9:State the scaling property of the z transform

If  $Z\{x(n)\} = X(z)$  then  $Z\{a^n x(n)\} = X(a^{-1}z)$

Q10:Explain convolution property of the z transform

A10:If  $Z\{x(n)\} = X(z)$  &  $Z\{h(n)\} = H(z)$  then,  $Z\{x(n)*h(n)\} = X(z)H(z)$

**EXPERIMENT NO. 9**

**Aim:** TO DESIGN FIR FILTERS (LOW-PASS, HIGH PASS, BAND-PASS, BAND-STOP).

**PROGRAM:**

%LOW-PASS FILTER (to pass all frequencies below 1200 Hz)

Clc;

Clear all;

Close all;

fs=8000; % sampling frequency

n=50; % order of the filter

w=1200/ (fs/2);

b=fir1(n,w,'low'); % Zeros of the filter

freqz(b,1,128,8000); % Magnitude and Phase Plot of the filter

%HIGH-PASS FILTER (to pass all frequencies above 1200 Hz)

fs=8000;

n=50;

w=1200/ (fs/2);

b=fir1(n,w,'high');

freqz(b,1,128,8000);

%BAND-PASS FILTER

fs=8000;

n=40;

b=fir1(n,[1200/4000 1800/4000], 'bandpass');

freqz(b,1,128,8000)

%BAND-STOP FILTER

fs=8000;

n=40;

b=fir1(n,[1200/4000 2800/4000], 'stop');

freqz(b,1,128,8000)

**QUESTION/ANSWERS**

Q1:-Give one advantage of digital filter over analog filter?

A1:- Digital filter can have characteristics which are not possible with analog filter

Q2:-What are main disadvantages of digital filter compared with analog filter?

A2:- Speed limitation

Q3:-What does IIR filter stand for

A3:- Infinite duration unit pulse response

Q4:-What does FIR filter stand for

A4:- Infinite duration unit pulse response

Q5:-what is the advantage of FIR filter over IIR filter

A5:- They are always stable, have exact linear phase

Q6:-What are the *methods* of designing FIR filters

A6:- Parks-McClellan, Windowing, Direct Calculation

A6:-Fourier series Method, Frequency Sampling Method

Q7:-What are the various window Techniques to design FIR filter

A7: Rectangular window, Hamming Window Function, Hanning window, Function, Blackman Window Function

Q8:-define the term Impulse Response of FIR filter

A8- The "impulse response" of a FIR filter is actually just the set of FIR coefficients. (If you put an "impulse" into a FIR filter which consists of a "1" sample followed by many "0" samples, the output of the filter will be the set of coefficients, as the 1 sample moves past each coefficient in turn to form the output.)

Q9:- Define the term Transition Band

Transition Band- The band of frequencies between passband and stopband edges. The narrower the transition band, the more taps are required to implement the filter. (A "small" transition band results in a "sharp" filter.)

Q10:- What is the Z transform of a FIR filter?

$$H(z) = \sum_{n=0}^{N-1} h(n)z^{-n}$$

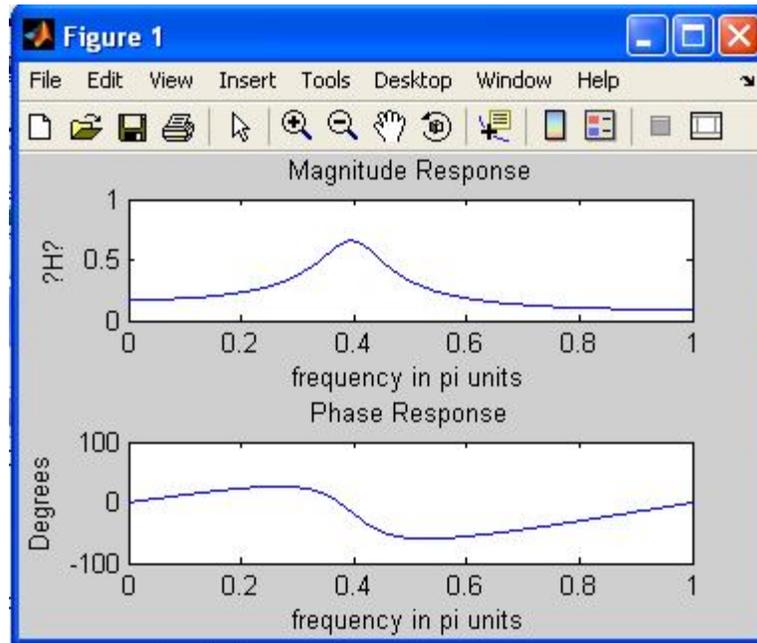
**EXPERIMENT NO.10**

**Aim:** PLOT THE MAGNITUDE AND PHASE OF THE FREQUENCY RESPONSE OF THE GIVEN DIGITAL FILTER USING FREQZ FUNCTION:

$$Y(n) = 0.2x(n) + 0.52y(n-1) - 0.68y(n-2)$$

**Program:**

```
b = [0.2];  
a = [1, -0.52, 0.68];  
w = [0:1:500]*pi/500;  
H = freqz(b,a,w);  
magH = abs(H);  
phaH = angle(H)*180/pi;  
subplot(2,1,1);  
plot(w/pi,magH);  
title('Magnitude Response');  
xlabel('frequency in pi units');  
ylabel(' | H | ');  
subplot(2,1,2);  
plot(w/pi,phaH);  
title('Phase Response');  
xlabel('frequency in pi units');  
ylabel('Degrees');
```

**Output:****QUESTION/ANSWER****Q1:-What are the advantages of IIR filters (compared to FIR filters)?**

A1:-IIR filters can achieve a given filtering characteristic using less memory and calculations than a similar FIR filter

**Q2:- What are the disadvantages of IIR filters (compared to FIR filters)?**

A2:- They are more susceptible to problems of finite-length arithmetic, such as noise generated by calculations, and limit cycles. (This is a direct consequence of feedback: when the output isn't computed perfectly and is fed back, the imperfection can compound.)

- They are harder (slower) to implement using fixed-point arithmetic.

**Q3:- Why is the impulse response "infinite"?**

A3The impulse response is "infinite" because there is feedback in the filter; if you put in an impulse (a single "1" sample followed by many "0" samples), an infinite number of non-zero values will come out (theoretically).

**Q4:- What are IIR filters? What does "IIR" mean?**

IIR filters are one of two primary types of digital filters used in Digital Signal Processing (DSP) applications (the other type being FIR). "IIR" means "Infinite Impulse Response".

**Q5:- Why is the impulse response "infinite"?**

The impulse response is "infinite" because there is feedback in the filter; if you put in an impulse (a single "1" sample followed by many "0" samples), an infinite number of non-zero values will come out (theoretically).

Q6:- Give one advantage of digital filter over analog filter?

A1:- Digital filter can have characteristics which are not possible with analog filter

Q7:-What is the limitation of approximation method

A7:-This technique is not suitable for high-Pass or band –reject Filter

Q8:- What is the limitation of approximation method

A8:- This technique is not suitable for high-Pass or band –reject Filter

Q9:-What is bilinear Transformation

A9:- bilinear Transformation is one-to one mapping from s-domain to the z-domain

Q10:-What are the various methods to design IIR filters

A10:- The various methods are

- 1) Approximation of derivatives
- 2) Impulse invariant method
- 3) Bilinear Transformation