DIGITAL

SIGNAL

PROCESSING

Lab Manual

MATLAB: An Introduction

This lab is to familiarize the students with MATLAB environment through it some preliminary MATLAB functions will be also covered.

Procedure:

Students are required to go through the steps explained below and then complete the exercises given at the end of the lab.

1. Introduction to MATLAB

i. Too add comment the following symbol is used "%".
ii. Help is provided by typing "help" or if you know the topic then "help function_name" or "doc function_name".
iii. If you don't know the exact name of the topic or command you are looking for,type "lookfor keyword" (e.g., "lookfor regression")
iv. Three dots "..." are used to continue a statement to next line (row).
v. If after a statement ";" is entered then MATLAB will not display the result of the statement entered otherwise result would be displayed.
vi. Use the up-arrow to recall commands without retyping them (and down arrow to go forward in commands).
vii. MATLAB is **case sensitive.**

2. Basic functionalities of *MATLAB*

Defining a scalar:

```
x=1
x =
1
```

Defining a column vector

```
v = [1;2;3]
v =
1
2
3
```

Defining a row vector

```
w = [1 0 1]
w =
1 0 1
```

Transpose a vector

```
W = w'
W =
1
0
1
```

Defining a range for a vector

```
X = 1:.5:5
X =
Columns 1 through 7
1.0000 1.5000 2.0000 2.5000 3.0000 3.5000
4.0000
Columns 8 through 9
4.5000 5.0000
```

Empty vector

```
Y = []
Y =
[]
```

Defining a matrix

```
M = [1 2 3; 3 2 1]




M =
1 2 3
3 2 1
```

Zero matrix

```
M = zeros(2,3) % 1st parameter is row, 2nd parameter is col.
M =
0 0 0
0 0 0
```

ones matrix

```
m = ones(2,3)
m =
1 1 1
1 1 1
```

The identity matrix

```
I = eye(3)
I =
1 0 0
0 1 0
0 0 1
```

Define a random matrix or vector

```
R = rand(1,3)
R = 0.9501 0.2311 0.6068
```

Access a vector or matrix

```
R(3)
ans = 0.6068
```

or

```
R(1,2)
ans = 0.2311
```

Access a row or column of matrix

```
I(2,:) %2nd row
ans = 0 1 0
I(:,2) %2nd col
ans =
0
1
0
I(1:2,1:2)
ans =
1 0
0 1
```

size and length

```
size(I)
ans =
3  3
length(I)
ans =
3
```

3. Operations on vector and matrices in MATLAB

MATLAB utilizes the following arithmetic operatops;

+ Addition
- Subtraction
* Multiplication
/ Division
^ Power Operator
‘ transpose

Some built in functions in MATLAB

abs magnitude of a number (absolute value for real numbers)
angle angle of a complex number, in radians
cos cosine function, assumes argument is in radians
sin sine function, assumes argument is in radians
exp exponential function

Arithmetic operations

```
x=[ 1 2 3 4 5]
x =
1 2 3 4 5
x= 2 * x
x =
2 4 6 8 10
x= x / 2
x =
1 2 3 4 5
y = [ 1 2 3 4 5 ]
y =
1 2 3 4 5
z = x + y
z =
2 4 6 8 10
```

point by point mult/div use “.“

```
W = x.*y
W =
1 4 9 16 25
```

Matlab has a large number of built in functions, some operate on each point of a vector/matrix:

```
log([1 2 3 4])
ans =
0 0.6931 1.0986 1.3863
round([1.5 2; 2.2 3.1])
```

```
ans =
2 2
2 3
a=[1 4 6 3]
a =
1 4 6 3
sum(a)
ans =
14
mean(a)
ans =
3.5000
std(a)
ans =
2.0817
max(a)
ans =
6
a =[1 2 3; 4 5 6]
mean(a) %mean of each column
max(a) %max of each column =20
max( max([1 2 3; 4 5 6]) ) %to obtain max of matrix
```

4. Relational operators in *MATLAB*

Relational operators: =(equal), ~=3D (not equal), etc.
Let

a = [1 1 3 4 1]

```
a =
1 1 3 4 1
```

ind = (a == 1)

```
ind =
1 1 0 0 1
ind = (a < 1)
ind =
0 0 0 0 0
ind = (a > 1)
ind =
0 0 1 1 0
ind = (a <= 1)
ind =
1 1 0 0 1
ind = (a >= 1)
ind =
1 1 1 1 1
ind = (a ~= 1)
ind =
0 0 1 1 0
```

5. Control Flow in MATLAB

To control the flow of commands, the makers of MATLAB supplied four devices a programmer can use while writing his/her computer code

- the **for** loops
- the **while** loops
- the **if-else-end** constructions
- the **switch-case** constructions

Syntax of the for loop is shown below

```
for k = array
commands
end
```

The commands between the **for** and **end** statements are executed for all %values stored in the **array**.

Suppose that one-need values of the sine function at eleven evenly %spaced points **n/10**, for **n = 0, 1, …, 10**. To generate the numbers in %question one can use the **for** loop

```
for n=0:10
x(n+1) = sin(pi*n/10);
end
x =
Columns 1 through 7
0 0.3090 0.5878 0.8090 0.9511 1.0000
0.9511
Columns 8 through 11
0.8090 0.5878 0.3090 0.0000
```

The **for** loops can be nested

```
H = zeros(5);
for k=1:5
for l=1:5
H(k,l) = 1/(k+l-1);
end
end
H =
1.0      0.5000 0.3333 0.2500 0.2000



0.5000 0.3333 0.2500 0.2000 0.1667
0.3333 0.2500 0.2000 0.1667 0.1429
0.2500 0.2000 0.1667 0.1429 0.1250
0.2000 0.1667 0.1429 0.1250 0.1111
```

Syntax of the while loop is

```
while expression
statements
end
```

This loop is used when the programmer does not know the number of repetitions a priori. Here is an almost trivial problem that requires a use of this loop. Suppose that the number is divided by 2. The resulting quotient is divided by 2 again. This process is continued till the current quotient is less than or equal to 0.01. What is the largest

quotient that is greater than 0.01?
To answer this question we write a few lines of code

```
q = pi;
while q > 0.01
q = q/2;
end
```

```
q =
0.0061
```

Syntax of the simplest form of the construction under discussion is

```
if expression
commands
end
```

This construction is used if there is one alternative only. Two alternatives require the construction

```
if expression
commands (evaluated if expression is true)
else
commands (evaluated if expression is false)
end
```

If there are several alternatives one should use the following construction

```
if expression1
commands (evaluated if expression 1 is true)
elseif expression 2
commands (evaluated if expression 2 is true)
elseif …
...
else
commands (executed if all previous expressions evaluate to false)
end
```

Syntax of the switch-case construction is

```
switch expression (scalar or string)
case value1 (executes if expression evaluates to value1)
commands
case value2 (executes if expression evaluates to value2)
commands
...
otherwise
statements
end
```

Switch compares the input expression to each case value. Once the %match is found it executes the associated commands.
In the following example a random integer number x from the set {1, 2, … , 10} is

generated. If x = 1 or x = 2, then the message Probability = 20% is displayed to the screen. If x = 3 or 4 or 5, then the message Probability = 30 is displayed, otherwise the message Probability = 50% is generated. The script file **fswitch** utilizes a switch as a tool %for handling all cases mentioned above

% Script file fswitch.
x = ceil(10*rand); % Generate a random integer in {1, 2, ... , 10}
switch x
case {1,2}
disp('Probability = 20%');
case {3,4,5}
disp('Probability = 30%');
otherwise
disp('Probability = 50%');
end
Note: use of the curly braces after the word case. This creates the so called *cell array* rather than the one-dimensional array, which %requires use of the square brackets.

6. Creating functions using m-files

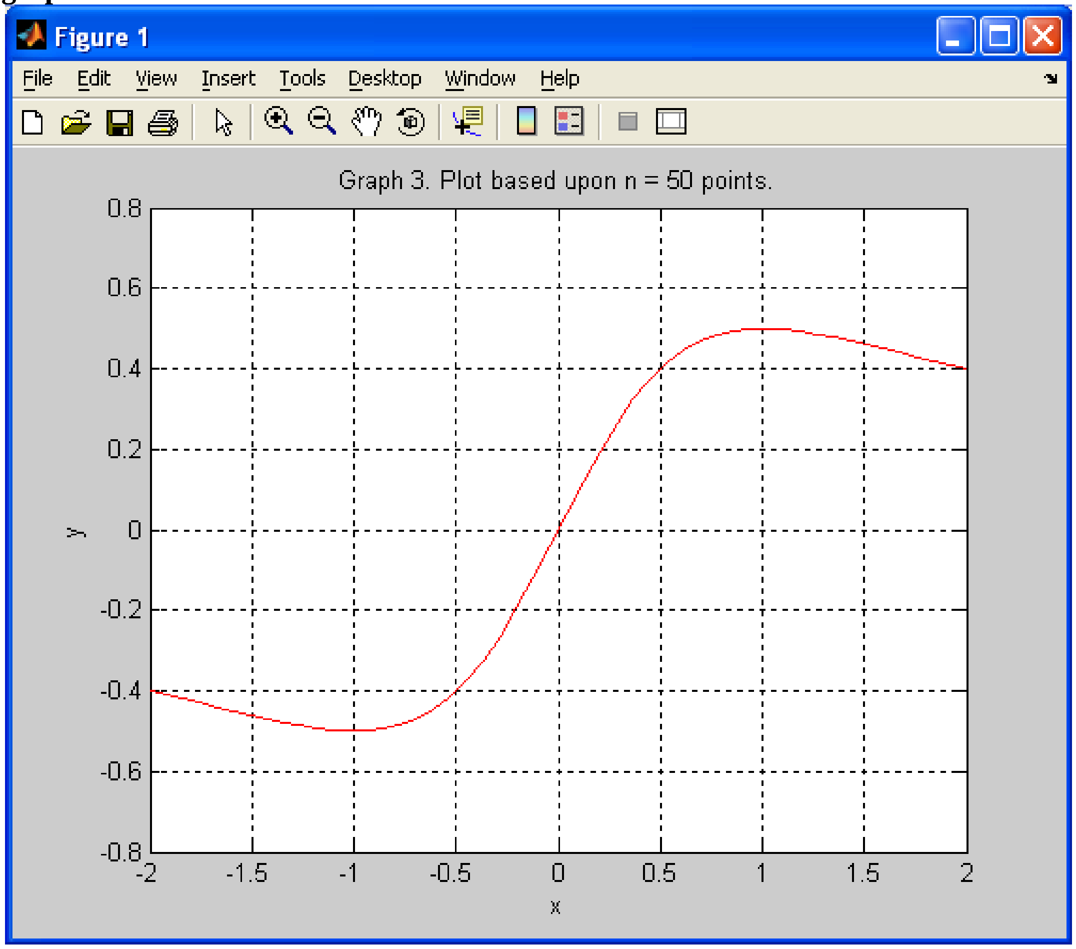
Files that contain a computer code are called the *m-files*. There are two kinds of m-files: the *script files* and the *function files*. Script files do not take the input arguments or return the output arguments. The function files may take input arguments or return output arguments. To make the m-file click on **File** next select **New** and click on **M-File** from the pull-down menu. You will be presented with the **MATLAB Editor/Debugger** screen. Here you will type your code, can make %changes, etc. Once you are done with typing, click on **File**, in the **MATLAB Editor/Debugger** screen and select **Save As…** . Chose a name for your file, e.g., **firstgraph.m** and click on **Save**. Make sure that your file is saved in the directory that is in MATLAB's search path. If you %have at least two files with duplicated names, then the one that occurs first in MATLAB's search path will be executed.
To open the m-file from within the **Command Window** type **edit firstgraph %**and then press **Enter** or **Return** key.
Here is an example of a small script file
% Script file firstgraph.
x = -10*pi:pi/100:10*pi;
y = sin(x)./x;
plot(x,y)
grid
Enter this code in the MATLAB editor and save it as firstgraph.m. This function call be called from command line as
firstgraph
Here is an example of a function file
function [b, j] = descsort(a)
% Function descsort sorts, in the descending order, a real array a.
% Second output parameter j holds a permutation used to obtain
% array b from the array a.
[b ,j] = sort(-a);

b = -b;
Enter this code in the MATLAB editor and save it as descsort.m . This function call be called from command line as
X=1:10

descsort(X)

7. Graphs in MATLAB
save the script file and the run it
Script file graph1.
Graph of the rational function y = x/(1+x^2).

```
for n=1:2:5
n10 = 10*n;
x = linspace(-2,2,n10);
y = x./(1+x.^2);
plot(x,y,'r')
title(sprintf('Graph %g. Plot based upon n = %g points.' ...
,(n+1)/2, n10))
axis([-2,2,-.8,.8])
xlabel('x')
ylabel('y')
grid
pause(3)
end%Several graphs using subplot
graph1
```



Script file graph2.
Several plots of the rational function y = x/(1+x^2)

in the same window.

```
k = 0;



for n=1:3:10
n10 = 10*n;
x = linspace(-2,2,n10);
y = x./(1+x.^2);
k = k+1;
subplot(2,2,k)
plot(x,y,'r')
title(sprintf('Graph %g. Plot based upon n = %g points.' ...
, k, n10))
xlabel('x')
ylabel('y')
axis([-2,2,-.8,.8])
grid
pause(3);
end
graph2
```

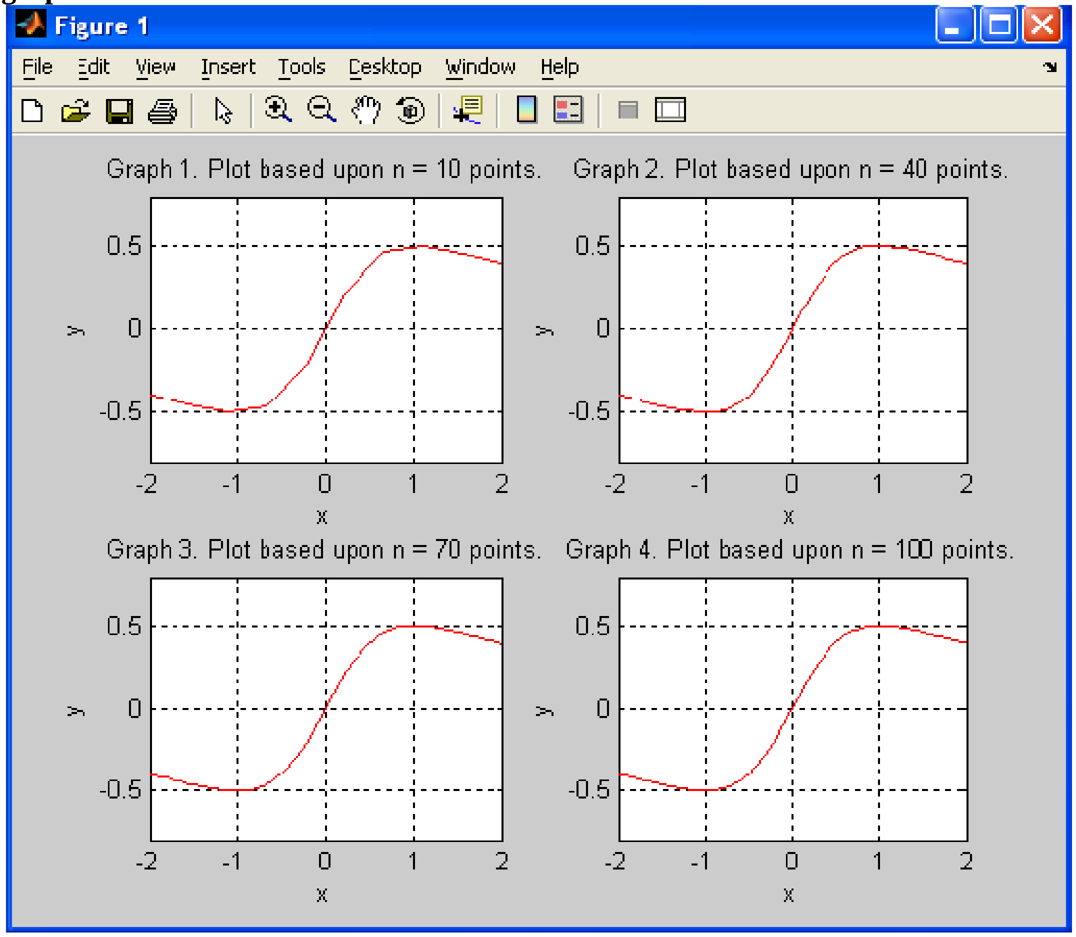


Figure 1
File
Edit
View
Insert
Tools
Desktop
Window
Help
Graph 1. Plot based upon n = 10 points.
Graph 2. Plot based upon n = 40 points.
Graph 3. Plot based upon n = 70 points.
Graph 4. Plot based upon n = 100 points.
0.5
0
-0.5
-2
-1
0
1
2
x
y

EXPERIMENT NO. 1

AIM:- Verification of Sampling theorem.

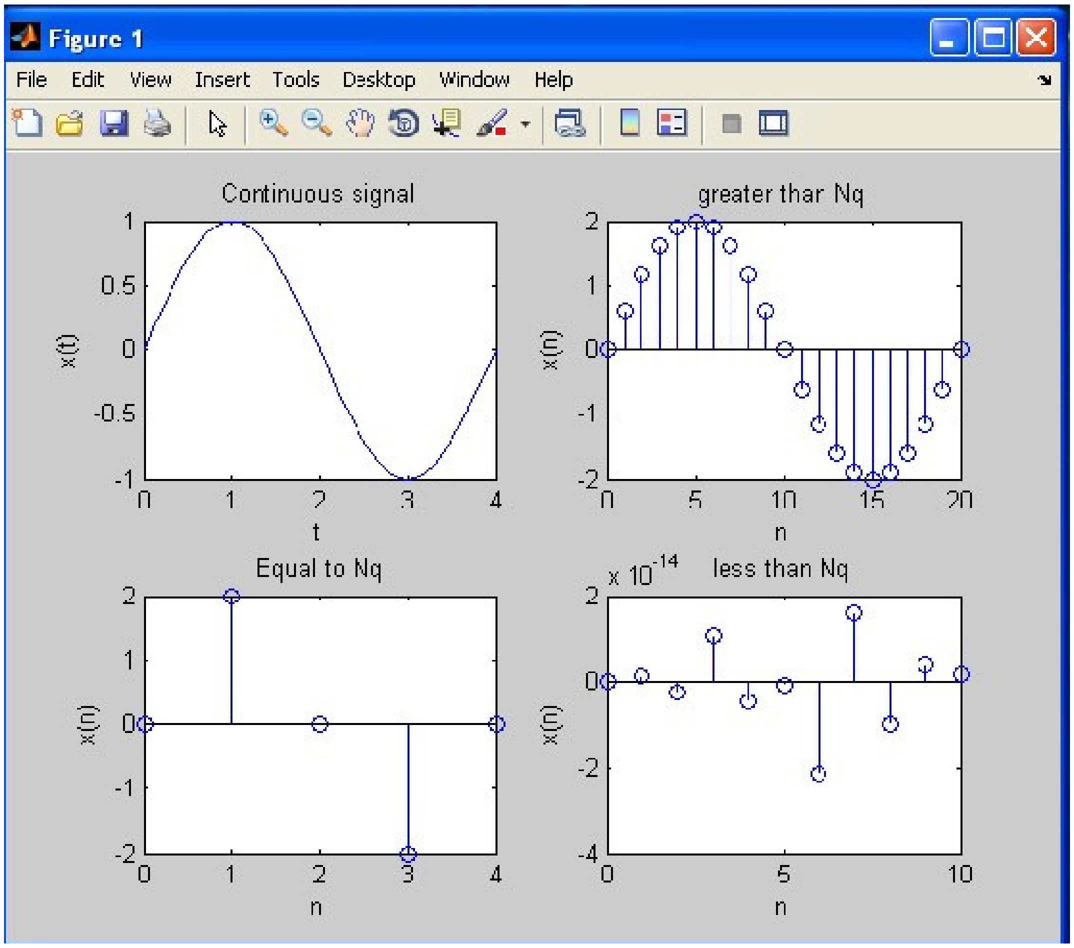
APPARATUS REQUIRED:

System with MATLAB 7.0.

PROGRAM

```
clc;
T=0.04; % Time period of 50 Hz signal
t=0:0.0005:0.02;
f = 1/T;
n1=0:40;
size(n1)
xa_t=sin(2*pi*2*t/T);
subplot(2,2,1);
plot(200*t,xa_t);
title('Verification of sampling theorem');
title('Continuous signal');
xlabel('t');
ylabel('x(t)');
ts1=0.002;%>niq rate
ts2=0.01;%=niq rate
ts3=0.1;%<niq rate
n=0:20;
x_ts1=2*sin(2*pi*n*ts1/T);
subplot(2,2,2);
stem(n,x_ts1);
title('greater than Nq');
xlabel('n');

ylabel('x(n)');

n=0:4;
x_ts2=2*sin(2*sym('pi')*n*ts2/T);
subplot(2,2,3);
stem(n,x_ts2);
title('Equal to Nq');
xlabel('n');
ylabel('x(n)');
n=0:10;
x_ts3=2*sin(2*pi*n*ts3/T);
subplot(2,2,4);
stem(n,x_ts3);
title('less than Nq');
xlabel('n');
ylabel('x(n)')
```

OUTPUT



RESULT:- Sampling Theorem has been proved.

EXPERIMENT NO. 2

AIM: To study the linear convolution of two given sequences.

APPARATUS REQUIRED:

System with MATLAB 7.0.

PROGRAM:

% Linear convolution using conv command

(A) Using CONV command.

```
clc;
x1=input('enter the first sequence');
subplot(3,1,1);
stem(x1);
ylabel('amplitude');
title('plot of the first sequence');
x2=input('enter 2nd sequence');
subplot(3,1,2);
stem(x2);
ylabel('amplitude');
title('plot of 2nd sequence');
f=conv(x1,x2);
disp('output of linear conv is');
disp(f);
xlabel('time index n');
ylabel('amplitude f');
subplot(3,1,3);
stem(f);
title('linear conv of sequence');
```

Output

```
enter the first sequence[1 2 3]
enter 2nd sequence[1 2 3 4]
output of linear conv is
1 4 10 16 17 12
```

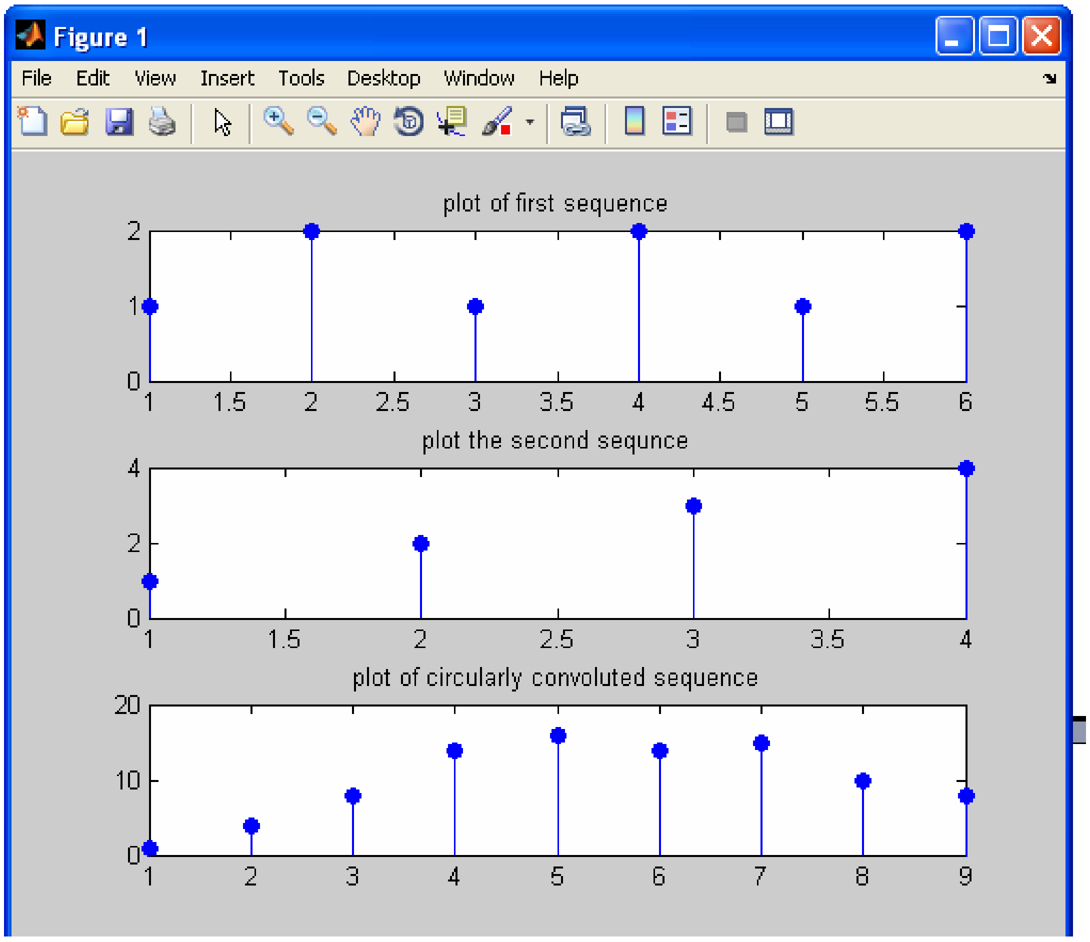
(B) Linear convolution Using DFT and IDFT / Linear convolution using circular convolution

```
clc;
clear all;
x1=input('enter the first sequence');
x2=input('enter the second sequence');
n=input('enter the no of points of the dft');
subplot(3,1,1);
stem(x1,'filled');
title('plot of first sequence');
```

```
subplot(3,1,2);
stem(x2,'filled');
title('plot the second sequnce');
n1 =length(x1);
n2 = length(x2);



m = n1+n2-1; % Length of linear convolution
x = [x1 zeros(1,n2-1)]; % Padding of zeros to make it of
% length m
y = [x2 zeros(1,n1-1)];



x_fft = fft(x,m);
y_fft = fft(y,m);
dft_xy = x_fft.*y_fft;
y=ifft(dft_xy,m);
disp('the circular convolution result is ......');
disp(y);
subplot(3,1,3);
stem(y,'filled');
title('plot of circularly convoluted sequence');
```

Output

```
enter the first sequence[1 2 1 2 1 2]
enter the second sequence[1 2 3 4]
the circular convolution result is ......
1.0000 4.0000 8.0000 14.0000 16.0000 14.0000
15.0000 10.0000 8.0000
```



RESULT:- Linear convolution has been proved.

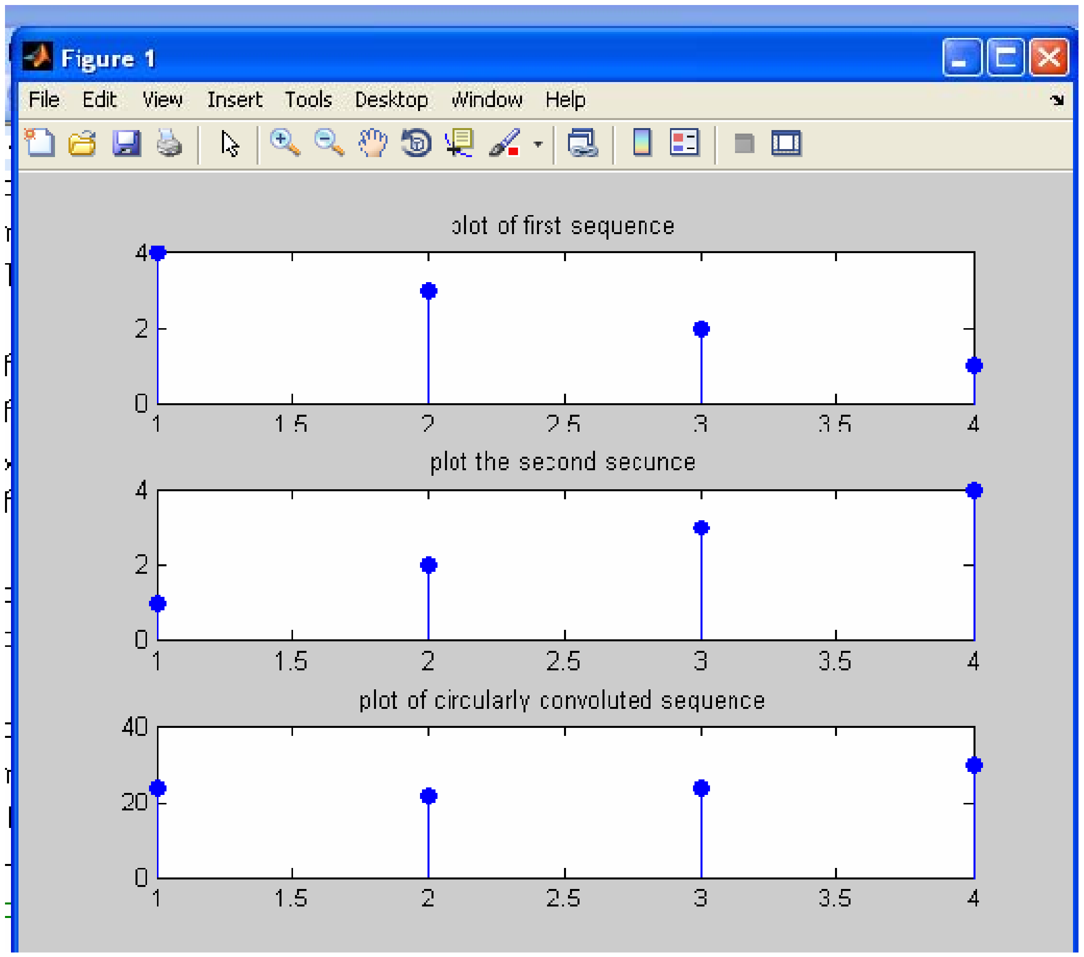
EXPERIMENT NO: 3

AIM: To study Circular convolution of two given sequences

```
clc;
clear all;
x1=input('enter the first sequence');
x2=input('enter the second sequence');
n1 = length(x1);
n2 = length(x2);
subplot(3,1,1);
stem(x1,'filled');
title('plot of first sequence');
subplot(3,1,2);
stem(x2,'filled');
title('plot the second sequnce');
y1=fft(x1,n);
y2=fft(x2,n);
y3=y1.*y2;
y=ifft(y3,n);
disp('the circular convolution result is ......');
disp(y);
subplot(3,1,3);
stem(y,'filled');
title('plot of circularly convoluted sequence');
```

Output

```
enter the first sequence[1 2 3 4]
enter the second sequence[4 3 2 1]
the circular convolution result is ......
24 22 24 30
```



RESULT:- Circular convolution has been proved.

EXPERIMENT NO: 4

AIM: To study autocorrelation of a given sequence and verification of its properties.

APPARATUS REQUIRED:

System with MATLAB 7.0.

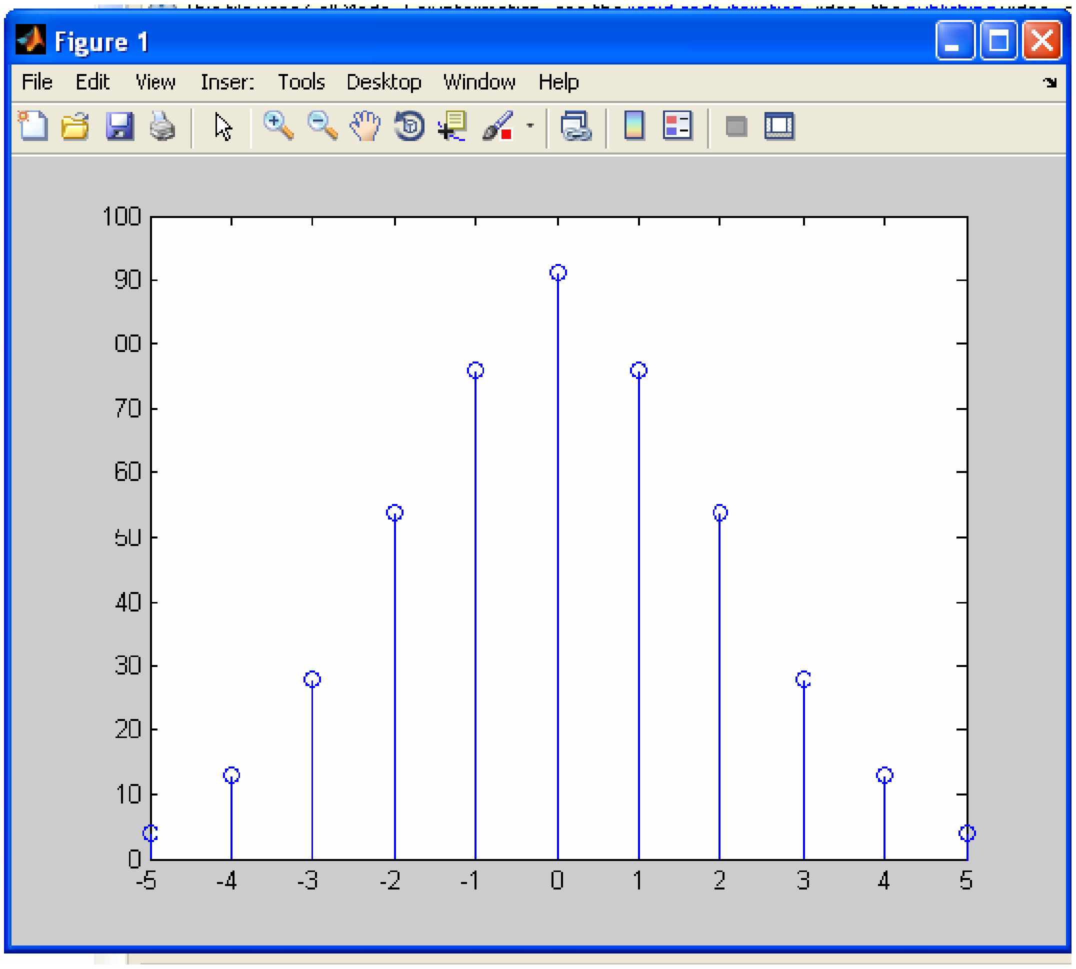
PROGRAM:

```
% Read the signal
x=[1,2,3,6,5,4]
% define the axis
n=0:1:length(x)-1
% plot the signal
stem(n,x);
xlabel('n');
% auto correlate the signal
Rxx=xcorr(x,x);
% the axis for auto correlation results
nRxx=-length(x)+1:length(x)-1
% display the result
stem(nRxx,Rxx)
% properties of Rxx(0) gives the energy of the signal
% find energy of the signal
energy=sum(x.^2)
%set index of the centre value
centre_index=ceil(length(Rxx)/2)
% Acces the centre value Rxx(0)
Rxx_0==Rxx(centre_index)
Rxx_0==Rxx(centre_index)
% Check if the Rxx(0)=energy
if Rxx_0==energy
disp('Rxx(0) gives energy proved');
else
disp('Rxx(0) gives energy not proved');
end
Rxx_right=Rxx(centre_index:1:length(Rxx))
Rxx_left=Rxx(centre_index:-1:1)
if Rxx_right==Rxx_left
disp('Rxx is even');
else
disp('Rxx is not even');

end
```

OUTPUT:-

```
x = 1 2 3 6 5 4
n = 0 1 2 3 4 5
nRxx = -5 -4 -3 -2 -1 0 1 2 3 4
5
```

```
energy= 91
centre_index = 6
Rxx(0) gives energy not proved
Rxx_right =
91.0000 76.0000 54.0000 28.0000 13.0000 4.0000
Rxx_left =
91.0000 76.0000 54.0000 28.0000 13.0000 4.0000

Rxx is even
```



RESULT:- Autocorrelation of a given sequence and its properties has been proved.

EXPERIMENT NO: 5

AIM: To study the computation of N point DFT of a given sequence and to plot magnitude and phase spectrum.

APPARATUS REQUIRED:

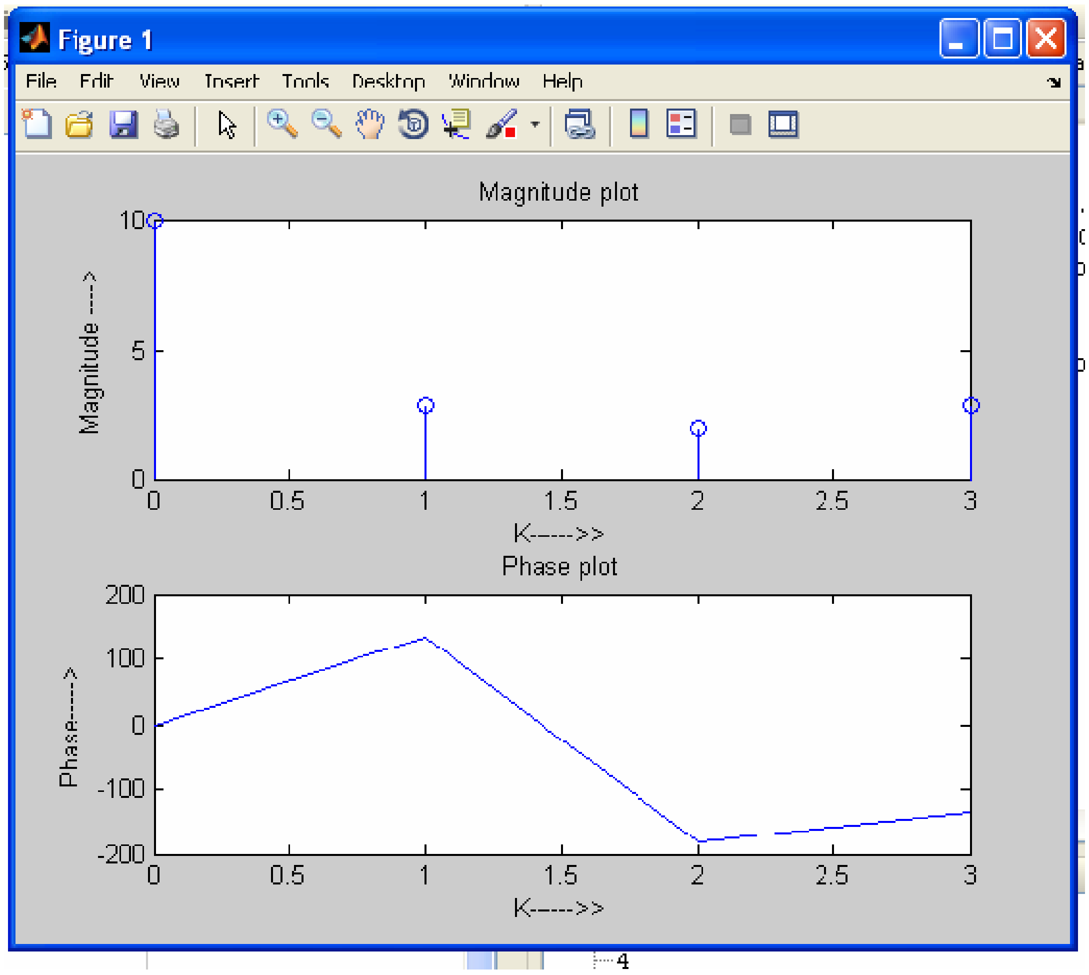
System with MATLAB 7.0.

PROGRAM:

```
N = input('Enter the the value of N(Value of N in N-Point DFT)');
x = input('Enter the sequence for which DFT is to be calculated');
n=[0:1:N-1];
k=[0:1:N-1];
WN=exp(-1j*2*pi/N);
nk=n'*k;
WNnk=WN.^nk;
Xk=x*WNnk;
MagX=abs(Xk) % Magnitude of calculated DFT
PhaseX=angle(Xk)*180/pi % Phase of the calculated DFT
figure(1);
subplot(2,1,1);
plot(k,MagX);
subplot(2,1,2);
plot(k,PhaseX);
-------------*******--------------
```

OUTPUT

```
Enter the the value of N(Value of N in N-Point DFT)4
Enter the sequence for which DFT is to be calculated
[1 2 3 4]
MagX = 10.0000 2.8284 2.0000 2.8284
PhaseX = 0 135.0000 -180.0000 -135.0000
DFT of the given sequence is
10.0000 -2.0000 + 2.0000i -2.0000 - 0.0000i -2.0000 -
2.0000i
```



RESULT: We have study the computation of N point DFT of a given sequence and also plot magnitude and phase spectrum.

EXPERIMENT NO: 6

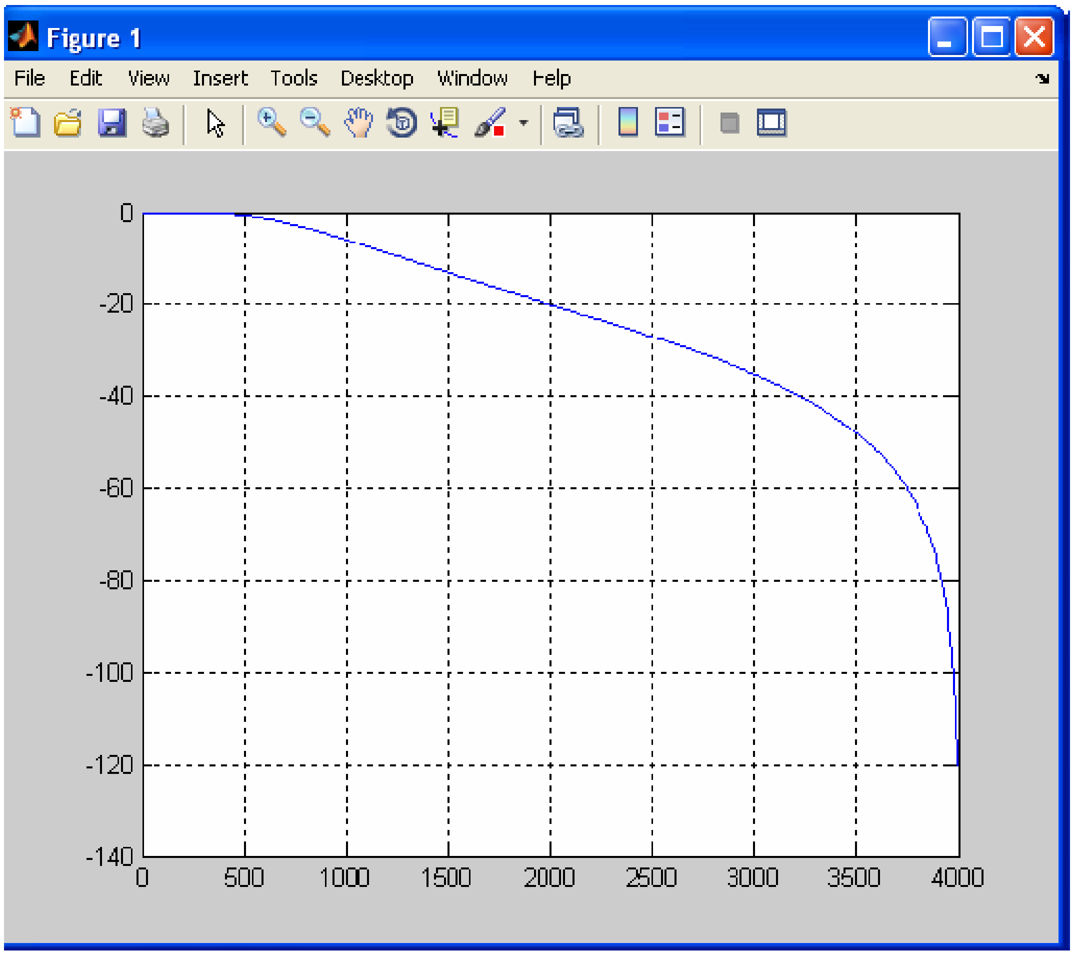
AIM:- Design and implementation of IIR BUTTERWORTH filter to meet the given specifications.

APPARATUS REQUIRED:

System with MATLAB 7.0.

PROGRAM:

```
clc;
clear all;
close all;
wp=500; % Enter the pass band frequency

ws=2000; % Enter the stop band frequency
Rp=3; % Enter the pass band ripple
Rs=20; % Enter the stop band attenuation
Fs=8000; % Enter the sampling frequency
Fn=Fs/2; % Normalized sampling frequency
% Find the order n and cutt off frequency
[n,wc]=buttord(wp/Fn,ws/Fn,Rp,Rs);
% Find the filter co-efficients
[b,a]=butter(n,wc);
disp(n)
% Plot the frequency response
[h,f]=freqz(b,a,512,8000);
plot(f,20*log10(abs(h)))
grid;
```



RESULT:- We have study the IIR BUTTERWORTH filter .

EXPERIMENT NO..:- 7

AIM: Generation of signals

To generate the following signals using MATLAB 7.0

1. Unit impulse signal
2. Unit step signal
3. Unit ramp signal
4. Exponential growing signal
5. Exponential decaying signal
6. Sine signal
7. Cosine signal

APPARATUS REQUIRED:

System with MATLAB 7.0.

ALGORITHM:

1. Get the number of samples.
2. Generate the unit impulse, unit step using ‘ones’, ‘zeros’ matrix command.
3. Generate ramp, sine, cosine and exponential signals using corresponding general formula.
4. Plot the graph.

PROGRAM:

```
% 1. UNIT IMPULSE
SIGNAL clc;
Clearall;
closeall;
N=input('Number of
Samples'); n=-N:1:N
x=[zeros(1,N) 1
zeros(1,N)]
stem(n,x);
xlabel('Time');
ylabel('Amplitude');
title('Impulse
Response');

% 2.   UNIT   STEP
SIGNAL clc;
clearall;
closeall;
N=input('Numberof Samples
= '); n=-N:1:N
x=[zeros(1,N)       1
ones(1,N)]
stem(n,x);
xlabel('Time');
ylabel('Amplitude');
title('Unit Step
Response');
```

% 3. UNIT RAMP SIGNAL

```
clc;
clear
all;
close
all;
disp('UNIT RAMP SIGNAL');
N=input('Number of
Samples = ');
a=input('Amplitude = ')
n=-N:1:N
x=a*n
stem(n,x);
xlabel('Time'
);
ylabel('Amplitude');
title('Unit Ramp
Response');
```

% 4. EXPONENTIAL DECAYING SIGNAL

```
clc;
clear
all;
close
all;
disp('EXPONENTIAL           DECAYING
SIGNAL');     N=input('Number     of
Samples = ');
a=0.5
n=0:.1:N
x=a.^n
stem(n,x);
xlabel('Time'
);
ylabel('Amplitude');
title('Exponential Decaying Signal Response');
```

% 5. EXPONENTIAL GROWING SIGNAL

```
clc;
clear
all;
close
all;
disp('EXPONENTIAL GROWING SIGNAL');
N=input('Number of Samples = ');
a=0.5
n=0:.1:N
```

```
x=a.^n
stem(n,x);
xlabel('Time');
ylabel('Amplitude');
title('Exponential Growing Signal Response');
```

% 6.COSINE SIGNAL
```
clc;
Clear
all;
close
all;
disp('COSINE SIGNAL');
N=input('Number of
Samples = '); n=0:.1:N
x=cos(n)
stem(n,x);
xlabel('Time');
ylabel('Amplitude
');

title('Cosine Signal');
```

% 7. SINE SIGNAL
```
clc;
clear
all;
close
all;
disp('SINE SIGNAL');
N=input('Number of
Samples = '); n=0:.1:N
x=sin(n)
stem(n,x);
xlabel('Time'
);
ylabel('Amplitude
'); title('sine
Signal');
```

EXPERIMENT NO. 8(A)

AIM:

To design a Butterworth digital IIR filter using MATLAB 7.0.

APPARATUS REQUIRED:

System with MATLAB 7.0.

ALGORITHM:

1. Get the pass band and stop band ripples.
2. Get the pass band and stop band frequencies.
3. Get the sampling frequency.
4. Calculate the order of the filter using

$$\frac{\log \sqrt{[10^{kp} - 1 / 10^{kp} - 1]}\ N}{= \log \square_s / \square_p}$$

5.Find the filter co-efficient.

6.Draw the magnitude and phase response.

PROGRAM:

```
clear all;
clc; close
all;
format long
rp=input('enter the pass band ripple');
rs=input('enter the stop band ripple');
wp=input('enter the pass band frequency
'); ws=input('enter the stop band frequency
'); fs=input('enter the sampling frequency
'); w1=2*wp/fs;
w2=2*ws/fs;

%LOW PASS FILTER
[n,wn]=buttord(w1,w2,rp,rs];
[b,a]=butter(n,wn);
%[b,a]=zp2tf(z,p,k);
[b,a]= butter(n,wn);

W=0:0.01:pi;
[h,om]=freqz(b,a,w);
m=20*log10(abs(h));
an=angle(h); %figure(1);
Subplot(2,1,1);
Plot(om/pi,m); Ylabel('gain
in db…>');
```

```
Xlabel(‘(a)normalized frequency…>’);
%figure(1);
Subplot(2,1,2);
Plot(om/pi,an);
Xlabel(‘(b)normalized frequency…>’);
Ylabel(‘phase in radians…>’);


%HIGH PASS FILTER
[n,wn]=buttord(w1,w2,rp,rs];
[b,a]=butter(n,wn,’high’);
W=0:0.01:pi;
[h,om]=freqz(b,a,w);
m=20*log10(abs(h));
an=angle(h);
figure(2); Subplot(2,1,1);
Plot(om/pi,m); Ylabel(‘gain
in db…>’);
Xlabel(‘(a)normalized frequency…>’);
figure(2);
Subplot(2,1,2);
Plot(om/pi,an);
Xlabel(‘(b)normalized frequency…>’);
Ylabel(‘phase in radians…>’);

%BAND PASS FILTER
[n]=buttord(w1,w2,rp,rs];
Wn=[w1,w2];
[b,a]=butter(n,wn,’band pass’);
W=0:0.01:pi;
[h,om]=freqz(b,a,w);
m=20*log10(abs(h));
an=angle(h);
figure(3); Subplot(2,1,1);
Plot(om/pi,m); Ylabel(‘gain
in db…>’);
Xlabel(‘(a)normalized frequency…>’);
figure(3);
Subplot(2,1,2);
Plot(om/pi,an);
```

EXPERIMENT:- 8(B)

AIM:

To design a digital IIR filter using Chebyshev filter with MATLAB 7.0.

APPARATUS REQUIRED:

System with MATLAB 7.0.

ALGORITHM:

1. Get the pass band and stop band ripples.
2. Get the pass band and stop band frequencies.
3. Get the sampling frequency.
4. Calculate the order of the filter using

$$N = \frac{\log \sqrt{[10^{kp} - 1 / 10^{kp} - 1]}}{\log \Omega_s / \Omega_p}$$

5. Find the filter co-efficient.
6. Draw the magnitude and phase response.

PROGRAM:

```
clear all;
clc; close
all;
rp=input('enter the pass band ripple');
rs=input('enter the stop band ripple');
wp=input('enter the pass band frequency
'); ws=input('enter the stop band frequency
'); fs=input('enter the sampling frequency
'); w1=2*wp/fs;
w2=2*ws/fs;

%LOW PASS FILTER
[n,wn]=cheb ord(w1,w2,rp,rs];
[b,a]=cheby 1(n,wn);
W=0:0.01:pi;
[h,om]=freqz(b,a,w);
m=20*log10(abs(h));
an=angle(h);
Subplot(2,1,1);
Plot(om/pi,m); Ylabel('gain
in db…>');
Xlabel('(a)normalized frequency…>');
Subplot(2,1,2);
```

```
Plot(om/pi,an);
Xlabel('(b)normalized frequency…>');
Ylabel('phase in radians…>');

%HIGH PASS FILTER
[n,wn]= cheb1 ord(w1,w2,rp,rs];
[b,a]=cheby 1(n,rp,wn,'high');
W=0:0.01:pi; [h,om]=freqz(b,a,w);
m=20*log10(abs(h)); an=angle(h);
figure(2); Subplot(2,1,1);
Plot(om/pi,m); Ylabel('gain
in db…>');
Xlabel('(a)normalized frequency…>');
figure(2);
Subplot(2,1,2);
Plot(om/pi,an);
Xlabel('(b)normalized frequency…>');
Ylabel('phase in radians…>');

%BAND PASS FILTER
[n]=cheb1 ord(w1,w2,rp,rs];
Wn=[w1,w2];
[b,a]=cheby 1(n,rs,wn,'band pass');
W=0:0.01:pi; [h,om]=freqz(b,a,w);
m=20*log10(abs(h));
an=angle(h);
figure(3);
Subplot(2,1,1);
Plot(om/pi,m);
Ylabel('gain in db…>');
Xlabel('(a)normalized frequency…>');
figure(3);
Subplot(2,1,2);
Plot(om/pi,an);
Xlabel('(b)normalized frequency…>');
Ylabel('phase in radians…>');

%BAND STOP FILTER
[n]=cheb 1 ord(w1,w2,rp,rs];
Wn=[w1,w2];
[b,a]=cheby 1(n,rp,wn,'band stop');
W=0:0.1/pi:pi; [h,om]=freqz(b,a,w);
m=20*log10(abs(h));
an=angle(h);
figure(4);
Subplot(2,1,1);
Plot(om/pi,m);
Ylabel('gain in db…>');
Xlabel('(a)normalized frequency…>');
figure(4);
```

```
Subplot(2,1,2);
Plot(om/pi,an);
Xlabel('(b)normalized frequency…>');
Ylabel('phase in radians…>');
```

Viva questions:

1.What is a continous and discrete time signal?

2.Give the classification of signals.

3.What are the types of system?

4.What are even and odd signal?

5.What are energy and power signal?

6.What are elementary signals and name them?

7.What is time invariant system?

8.What do you mean by periodic and aperiodic signals?

9.Determine the convolution sum of two sequences x(n) = {3, 2, 1, 2} and **h(n) = {1, 2, 1, 2}**

10.Find the convolution of the
signals x(n) = 1 n=-2,0,1
= 2 n=-1
= 0 elsewhere.

11. Differentiate DTFT and DFT.

12..Differentiate between DIT and DIF algorithm

13.How many stages are there for 8 point DFT

14. How many multiplication terms are required for doing DFT by expressionalmethod and FFT method

15.Distinguish IIR and FIR filters

16.Write the expression for order State the steps to design digital IIR filter using bilinear method of Butterworth filter?

17.Write the expression for the order of chebyshev filter?

18.What is warping effect?

19.Write a note on pre warping.

20.What is meant by impulse invariant method?

21. Give the Butterworth filter transfer function and its magnitude characteristics for different orders of filter.

22. Give the magnitude function of Butterworth filter.

23. How can you design a digital filter from analog filter?

24.write down bilinear transformation.

25.What is filter?

26. What are the types of digital filter according to their impulse response?

27.what is mean by FIR filter?

28.***Write the steps involved in FIR filter design.***

29.List the well known design technique for linear phase FIR filter design?

30.Define IIR filter?

31.What is the reason that FIR filter is always stable?

32.What are the properties of FIR filter?

33.What is the principle of designing FIR filter using frequency sampling method?

34.What is meant by autocorrelation?

35.Define white noise?

36.List out the addressing modes supported by C5X processors?

37.What do you understand by input quantization error?

38.List the on-chip peripherals in 5X.

39.what is meant rounding?

40.what is meant by A/D conversion noise?

41.what is meant by quantization step size?

42.what is overflow oscillation?

43.what are the methods used to prevent overflow?

44.what are the two kinds of limit cycle behavior in DSP?

45.What is meant by "dead band" of the filter?