Scilab Manual for
Digital Signal Processing
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Contents

List of Experiments

Experiment: 1

Generation of Discrete Signals

Scilab code Solution 1.1 Unit Sample Sequence

```
//Caption:Unit Sample Sequence
clear;
clc;
close;
L = 4;   //Upperlimit
n = -L:L;
x = [zeros(1,L),1,zeros(1,L)];
b = gca();
b.y_location = "middle";
plot2d3('gnn',n,x)
a=gce();
a.children(1).thickness =4;
xtitle('Graphical Representation of Unit Sample Sequence','n','x[n]');
```

Scilab code Solution 1.2 Unit Step Sequence

```
//Caption: Unit Step Sequence
```

```
clear;
clc;
close;
L = 4;   //Upperlimit
n = -L:L;
x = [zeros(1,L),ones(1,L+1)];
a=gca();
a.y_location = "middle";
plot2d3('gnn',n,x)
title('Graphical Representation of Unit Step Signal')
xlabel('                                              n');
ylabel('                                              x[n]');
```

Scilab code Solution 1.3 Discrete Ramp Sequence

```
//Caption: Discrete Ramp Sequence
clear;
clc;
close;
L = 4;   //Upperlimit
n = -L:L;
x = [zeros(1,L),0:L];
b = gca();
b.y_location = 'middle';
plot2d3('gnn',n,x)
a=gce();
a.children(1).thickness =2;
xtitle('Graphical Representation of Discrete Unit Ramp Sequence','n','x[n]');
```

Scilab code Solution 1.4 Exponentially Decreasing Signal

```
//Caption: Exponentially Decreasing Signal
clear;
clc;
close;
a =0.5;
n = 0:10;
x = (a)^n;
a=gca();
a.x_location = "origin";
a.y_location = "origin";
plot2d3('gnn',n,x)
a.thickness = 2;
xtitle('Graphical Representation of Exponentially
   Decreasing Signal','n','x[n]');
```

Scilab code Solution 1.5 Exponentially Increasing Signal

```
//Caption:Exponentially Increasing Signal
clear;
clc;
close;
a =1.5;
n =1:10;
x = (a)^n;
a=gca();
a.thickness = 2;
plot2d3('gnn',n,x)
xtitle('Graphical Representation of Exponentially
   Increasing Signal','n','x[n]');
```

Experiment: 2

Linear and Circular Convolution of two sequences

Scilab code Solution 2.1 Program for Linear Convolution

```
//Caption:Program for Linear Convolution
clc;
clear all;
close ;
x = input('enter x seq');
h = input('enter h seq');
m = length(x);
n = length(h);
//Method 1 Using Direct Convolution Sum Formula
for i = 1:n+m-1
    conv_sum = 0;
    for j = 1:i
        if (((i-j+1) <= n)&(j <= m))
            conv_sum = conv_sum + x(j)*h(i-j+1);
        end;
        y(i) = conv_sum;
    end;
end;
disp(y','Convolution Sum using Direct Formula Method
```

```
=')
//Method 2 Using Inbuilt Function
f = convol(x,h)
disp(f,'Convolution Sum Result using Inbuilt Funtion =')
//Method 3 Using frequency Domain multiplication
N = n+m-1;
x = [x zeros(1,N-m)];
h = [h zeros(1,N-n)];
f1 = fft(x)
f2 = fft(h)
f3 = f1.*f2;    // freq domain multiplication
f4 = ifft(f3)
disp(f4,'Convolution Sum Result DFT - IDFT method =')
//f4 =  real(f4)
subplot(3,1,1);
plot2d3('gnn',x)
xtitle('Graphical Representation of Input signal x');
subplot(3,1,2);
plot2d3('gnn',h)
xtitle('Graphical Representation of Impulse signal h');
subplot(3,1,3);
plot2d3('gnn',y)
xtitle('Graphical Representation of Output signal y');
//Result
//enter x seq [1 1 1 1]
//enter h seq [1 2 3]
// Convolution Sum using Direct Formula Method =
//     1.    3.    6.    6.    5.    3.
// Convolution Sum Result using Inbuilt Funtion =
//     1.    3.    6.    6.    5.    3.
// Convolution Sum Result DFT - IDFT method =
//     1.    3.    6.    6.    5.    3.
```

Scilab code Solution 2.2 Program to find the Cicrcular Convolution

```
//Caption: Program to find the Cicrcular Convolution
    of given
//discrete sequences using Matrix method

clear;
clc;
x1 = [2,1,2,1]; //First sequence
x2 = [1,2,3,4]; //Second sequence
m = length(x1); //length of first sequence
n = length(x2); //length of second sequence
//To make length of x1 and x2 are Equal
if (m >n)
   for i = n+1:m
     x2(i) = 0;
   end
elseif (n>m)
   for i = m+1:n
     x1(i) = 0;
   end
end
N = length(x1);
x3 = zeros(1,N); //x3 = Circular convolution result
a(1) = x2(1);
for j = 2:N
  a(j) = x2(N-j+2);
end
for i =1:N
  x3(1) = x3(1)+x1(i)*a(i);
end
X(1,:)=a;
//Calculation of circular convolution
for k = 2:N
```

```
    for j =2:N
       x2(j) = a(j-1);
    end
       x2(1) = a(N);
       X(k,:)= x2;
    for i = 1:N
        a(i) = x2(i);
        x3(k) = x3(k)+x1(i)*a(i);
    end
end
disp(X,'Circular Convolution Matrix x2[n]=')
disp(x3,'Circular Convolution Result x3[n] = ')
//Result
//Circular Convolution Matrix x2[n]=
//
//     1.     4.     3.     2.
//     2.     1.     4.     3.
//     3.     2.     1.     4.
//     4.     3.     2.     1.
//
// Circular Convolution Result x3[n] =
//
//     14.     16.     14.     16.
```

Experiment: 3

Circular convolution using FFT

Scilab code Solution 3.1 Performing Circular COnvolution Using DFT IDFT method

```
//Caption:Performing Circular COnvolution Using DFT-IDFT method
clear all;
clc;
close;
L = 4; //Length of the Sequence
N = 4;  // N -point DFT
x1 = [2,1,2,1];
x2 = [1,2,3,4];
//Computing DFT
X1 = fft(x1,-1);
X2 = fft(x2,-1);
disp(X1,'DFT of x1[n] is X1(k)=')
disp(X2,'DFT of x1[n] is X2(k)=')
//Multiplication of 2 DFTs
X3 = X1.*X2;
disp(X3,'DFT of x3[n] is X3(k)=')
//Circular Convolution Result
x3 =abs(fft(X3,1))
disp(x3,'Circular Convolution Result x3[n]=')
```

```
//Result
// DFT of x1[n] is X1(k)=
//
//     6.     0     2.     0
//
// DFT of x1[n] is X2(k)=
//
//     10.   - 2. + 2.i   - 2.   - 2. - 2.i
//
// DFT of x3[n] is X3(k)=
//
//     60.     0   - 4.     0
//
//  Circular Convolution Result x3[n]=
//
//     14.     16.     14.     16.
```

Experiment: 4

Linear Convolution using Circular Convolution

Scilab code Solution 4.1 Performing Linear Convolution using Circular Convolution

```
//Caption: Performing Linear Convolution using Circular Convolution

clear;
clc;
close;
h = [1,2,3];     //Impulse Response of LTI System
x = [1,2,2,1];   //Input Response of LTI System
N1 = length(x);
N2 = length(h);
N = N1+N2-1
disp(N,'Length of Output Response y(n)')
//Padding zeros to Make Length of 'h' and 'x'
//Equal to length of output response 'y'
h1 = [h,zeros(1,N-N2)];
x1 = [x,zeros(1,N-N1)];
//Computing FFT
H = fft(h1,-1);
```

```
X = fft(x1,-1);
//Multiplication of 2 DFTs
Y = X.*H
//Linear Convolution Result
y =abs(fft(Y,1))
disp(X,'DFT of i/p X(k)=')
disp(H,'DFT of impulse sequence H(k)=')
disp(Y,'DFT of Linear Filter o/p Y(k)=')
disp(y,'Linear Convolution result y[n]=')
//Result
// Length of Output Response y(n)
//
//    6.
//
// DFT of i/p X(k)=
//
//    6.  - 3.4641016i    0    0    0    3.4641016i
//
// DFT of impulse sequence H(k)=
//
//    6.    0.5 - 4.330127i  - 1.5 + 0.8660254i
   2.  - 1.5 - 0.8660254i    0.5 + 4.330127i
//
// DFT of Linear Filter o/p Y(k)=
//
//    36.  - 15. - 1.7320508i    0    0    0  - 15.
   + 1.7320508i
//
// Linear Convolution result y[n]=
//
//    1.    4.    9.    11.    8.    3.
```

Experiment: 5

Calculation of FFT and IFFT of a sequence

Scilab code Solution 5.5 Performing FFT and IFFT of a discrete sequence

```
//Caption:Performing FFT and IFFT of a discrete sequence
clear;
clc;
close;
L = 4; //Length of the Sequence
N = 4;  // N -point DFT
x = [1,2,3,4];
//Computing DFT
X = fft(x,-1);
disp(X,'FFT of x[n] is X(k)=')
x =abs(fft(X,1))
disp(x,'IFFT of X(k) is x[n]=')
//Plotting the spectrum of Discrete Sequence
subplot(2,1,1)
a=gca();
a.data_bounds=[0,0;5,10];
plot2d3('gnn',0:length(x)-1,x)
b = gce();
```

```
b.children(1).thickness =3;
xtitle('Graphical Representation of Discrete Sequence','n','x[n]');
subplot(2,1,2)
a=gce();
a.data_bounds=[0,0;5,10];
plot2d3('gnn',0:length(X)-1,abs(X))
b = gce();
b.children(1).thickness =3;
xtitle('Graphical Representation of Discrete Spectrum','k','X(k)');
//Result
//FFT of x[n] is X(k)=
//
//     10.   - 2. + 2.i   - 2.   - 2. - 2.i
//
//IFFT of X(k) is x[n]=
//
//     1.     2.     3.     4.
```

Experiment: 6

Time and Frequency Response of LTI systems

Scilab code Solution 6.1 Time and Frequency Response

```
//Caption: Program to generate and plot the impulse
   response and frequency
//response of a Linear constant coefficient first
   order Differential Equation
//[1].Impulse response h(t)= exp(-a*t)u(t), A>0
//[2].Frequency response H(jw) = 1/(jw+a)
clear;
clc;
close;
//[1]. To generate and plot the impulse response
a =1;     //Constant coefficient a =1
Dt = 0.005;
t = 0:Dt:10;
ht = exp(-a*t);
figure(1)
a = gca();
a.y_location = "origin";
plot(t,ht);
xlabel('time t ------>');
```

```
ylabel('h(t)')
title('Impulse Repsonse of Ist Order Linear Constant Coeff. Differential Equ.')
//
//[2]. Finding Frequency response using Continuous Time Fourier Transform
Wmax = 2*%pi*1;          //Analog Frequency = 1Hz
K = 4;
k = 0:(K/1000):K;
W = k*Wmax/K;
HW = ht* exp(-sqrt(-1)*t'*W) * Dt;
HW_Mag = abs(HW);
W = [-mtlb_fliplr(W), W(2:1001)]; // Omega from -Wmax to Wmax
HW_Mag = [mtlb_fliplr(HW_Mag),HW_Mag(2:1001)];
[HW_Phase,db] = phasemag(HW);
HW_Phase = [-mtlb_fliplr(HW_Phase),HW_Phase(2:1001)];
figure(2)
//Plotting Magnitude Response
subplot(2,1,1);
a = gca();
a.y_location = "origin";
plot(W,HW_Mag);
xlabel('Frequency in Radians/Seconds---> W');
ylabel('abs(H(jW))')
title('Magnitude Response')
//Plotting Phase Reponse
subplot(2,1,2);
a = gca();
a.y_location = "origin";
a.x_location = "origin";
plot(W,HW_Phase*%pi/180);
xlabel('                              Frequency in Radians/Seconds---> W');
ylabel('                                                   <H(jW)')
```

```
title('Phase Response in Radians')
```

Experiment: 7

Sampling, Verification of Sampling and Effect of aliasing

check Appendix AP 1 for dependency:

sincnew.sce

Scilab code Solution 7.1 Sampling and Reconstruction of a Signal

```
//Caption: Sampling and Reconstruction of a Signal x
    (t) = exp(-A*|t|)
//Discrete Time Sampled Signal x(nT)= exp(-A*|nT|)
//Following Sampling Frequencies are used:
//[1].Fs = 1 Hz [2].Fs = 2 Hz [3].Fs = 4Hz [4].Fs
    =20 Hz [5].Fs =100Hz
//Aliasing Effect: As the Sampling frequency
    increases aliasing effect decreases
clear;
clc;
close;
// Analog Signal
A =1;      //Amplitude
Dt = 0.005;
t = -2:Dt:2;
```

```
//Continuous Time Signal
xa = exp(-A*abs(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;
x = exp(-A*abs(nTs));
// Analog Signal reconstruction
Dt = 0.005;
t = -2:Dt:2;
Xa = x *sincnew(Fs*(ones(length(nTs),1)*t-nTs'*ones
    (1,length(t))));
//Plotting the original signal and reconstructed
    signal
subplot(2,1,1);
a =gca();
a.x_location = "origin";
a.y_location = "origin";
plot(t,xa);
xlabel('t in sec.');
ylabel('xa(t)')
title('Original Analog Signal')
subplot(2,1,2);
a =gca();
a.x_location = "origin";
a.y_location = "origin";
xlabel('t in sec.');
ylabel('xa(t)')
title('Reconstructed Signal using sinc function, Fs
    = 100Hz');
plot(t,Xa);
```

Experiment: 8

Design of FIR Filters Window Design

Scilab code Solution 8.1 Program to Design FIR Low Pass Filter

```
//Caption: Program to Design FIR Low Pass Filter
clc;
close;
M = input('Enter the Odd Filter Length =');            //Filter length
Wc = input('Enter the Digital Cutoff frequency =');    //Digital Cutoff frequency
Tuo = (M-1)/2      //Center Value
for n = 1:M
    if (n == Tuo+1)
      hd(n) = Wc/%pi;
    else
      hd(n) =  sin(Wc*((n-1)-Tuo))/(((n-1)-Tuo)*%pi);
    end
end
//Rectangular Window
for n = 1:M
  W(n) = 1;
```

```
end
//Windowing Fitler Coefficients
h = hd.*W;
disp(h,'Filter Coefficients are')

[hzm,fr]=frmag(h,256);
hzm_dB = 20*log10(hzm)./max(hzm);
subplot(2,1,1)
plot(2*fr,hzm)
xlabel('Normalized Digital Frequency W');
ylabel('Magnitude');
title('Frequency Response 0f FIR LPF using Rectangular window')
xgrid(1)
subplot(2,1,2)
plot(2*fr,hzm_dB)
xlabel('Normalized Digital Frequency W');
ylabel('Magnitude in dB');
title('Frequency Response 0f FIR LPF using Rectangular window')
xgrid(1)
//Result
//Enter the Odd Filter Length = 7
//Enter the Digital Cutoff frequency = %pi/2
//
// Filter Coefficients are
//
//  - 0.1061033
//     1.949D-17  = 0.0
//     0.3183099
//     0.5
//     0.3183099
//     1.949D-17 = 0.0
//  - 0.1061033
```

Scilab code Solution 8.2 rogram to Design FIR High Pass Filter

```
//Caption: Program to Design FIR High Pass Filter
clear;
clc;
close;
M = input('Enter the Odd Filter Length =');
                //Filter length
Wc = input('Enter the Digital Cutoff frequency =');
   //Digital Cutoff frequency
Tuo = (M-1)/2      //Center Value
for n = 1:M
    if (n == Tuo+1)
      hd(n) = 1-Wc/%pi;
    else
      hd(n) = (sin(%pi*((n-1)-Tuo)) -sin(Wc*((n-1)-
         Tuo)))/(((n-1)-Tuo)*%pi);
    end
end
//Rectangular Window
for n = 1:M
  W(n) = 1;
end
//Windowing Fitler Coefficients
h = hd.*W;
disp(h,'Filter Coefficients are')
[hzm,fr]=frmag(h,256);
hzm_dB = 20*log10(hzm)./max(hzm);
subplot(2,1,1)
plot(2*fr,hzm)
xlabel('Normalized Digital Frequency W');
ylabel('Magnitude');
title('Frequency Response 0f FIR HPF using
   Rectangular window')
xgrid(1)
subplot(2,1,2)
plot(2*fr,hzm_dB)
xlabel('Normalized Digital Frequency W');
```

```
ylabel('Magnitude in dB');
title('Frequency Response Of FIR HPF using Rectangular window')
xgrid(1)
//Result
//Enter the Odd Filter Length = 5
//Enter the Digital Cutoff frequency = %pi/4
//Filter Coefficients are
//
//  - 0.1591549
//  - 0.2250791
//    0.75
//  - 0.2250791
//  - 0.1591549
```

Scilab code Solution 8.3 Program to Design FIR Band Pass Filter

```
//Caption: Program to Design FIR Band Pass Filter
clear;
clc;
close;
M = input('Enter the Odd Filter Length =');          //Filter length
//Digital Cutoff frequency [Lower Cutoff, Upper Cutoff]
Wc = input('Enter the Digital Cutoff frequency =');
Wc2 = Wc(2)
Wc1 = Wc(1)
Tuo = (M-1)/2      //Center Value
hd = zeros(1,M);
W = zeros(1,M);
for n = 1:11
  if (n == Tuo+1)
     hd(n) = (Wc2-Wc1)/%pi;
  else
```

```
        n
      hd(n) = (sin(Wc2*((n-1)-Tuo)) -sin(Wc1*((n-1)-
         Tuo)))/(((n-1)-Tuo)*%pi);
    end
    if(abs(hd(n))<(0.00001))
      hd(n)=0;
    end
end
hd;
//Rectangular Window
for n = 1:M
    W(n) = 1;
end
//Windowing Fitler Coefficients
h = hd.*W;
disp(h,'Filter Coefficients are')
[hzm,fr]=frmag(h,256);
hzm_dB = 20*log10(hzm)./max(hzm);
subplot(2,1,1)
plot(2*fr,hzm)
xlabel('Normalized Digital Frequency W');
ylabel('Magnitude');
title('Frequency Response 0f FIR BPF using
    Rectangular window')
xgrid(1)
subplot(2,1,2)
plot(2*fr,hzm_dB)
xlabel('Normalized Digital Frequency W');
ylabel('Magnitude in dB');
title('Frequency Response 0f FIR BPF using
    Rectangular window')
xgrid(1)
//Result
//Enter the Odd Filter Length = 11
//Enter the Digital Cutoff frequency = [%pi/4,3*%pi
    /4]
//Filter Coefficients are
//  0.     0.     0.  - 0.3183099     0.     0.5     0.  -
```

```
0.3183099    0.    0.    0.
```

Scilab code Solution 8.4 Program to Design FIR Band Reject Filter

```
//Caption: Program to Design FIR Band Reject Filter
clear ;
clc;
close;
M = input('Enter the Odd Filter Length =');
               //Filter length
//Digital Cutoff frequency [Lower Cutoff, Upper
   Cutoff]
Wc = input('Enter the Digital Cutoff frequency =');
Wc2 = Wc(2)
Wc1 = Wc(1)
Tuo = (M-1)/2     //Center Value
hd = zeros(1,M);
W = zeros(1,M);
for n = 1:M
 if (n == Tuo+1)
 hd(n) = 1-((Wc2-Wc1)/%pi);
 else
 hd(n)=(sin(%pi*((n-1)-Tuo))-sin(Wc2*((n-1)-Tuo))+
    sin(Wc1*((n-1)-Tuo)))/(((n-1)-Tuo)*%pi);
 end
  if(abs(hd(n))<(0.00001))
    hd(n)=0;
  end
end

//Rectangular Window
for n = 1:M
  W(n) = 1;
end
//Windowing Fitler Coefficients
```

```
h = hd.*W;
disp(h,'Filter Coefficients are')
[hzm,fr]=frmag(h,256);
hzm_dB = 20*log10(hzm)./max(hzm);
subplot(2,1,1)
plot(2*fr,hzm)
xlabel('Normalized Digital Frequency W');
ylabel('Magnitude');
title('Frequency Response 0f FIR BSF using
    Rectangular window')
xgrid(1)
subplot(2,1,2)
plot(2*fr,hzm_dB)
xlabel('Normalized Digital Frequency W');
ylabel('Magnitude in dB');
title('Frequency Response 0f FIR BSF using
    Rectangular window')
xgrid(1)
//Result
//Enter the Odd Filter Length = 11
//Enter the Digital Cutoff frequency =[%pi/3,2*%pi
    /3]
//Filter Coefficients are
//column 1 to 9
//     0.  - 0.1378322     0.     0.2756644     0.
    0.6666667     0.     0.2756644     0.
//column 10 to 11
//  - 0.1378322     0.
```

Experiment: 9

Design of FIR Filters Frequency Sampling

Scilab code Solution 9.1 Design of FIR LPF Filter using Frequecny Sampling Technique

```
//Cpation: Design of FIR LPF Filter using Frequecny
   Sampling Technique

clear;
clc;
close;
M =15;
Hr = [1,1,1,1,0.4,0,0,0];
for k =1:length(Hr)
    G(k)=((-1)^(k-1))*Hr(k);
end
h = zeros(1,M);
U = (M-1)/2
for n = 1:M
  h1 = 0;
  for k = 2:U+1
    h1 =G(k)*cos((2*%pi/M)*(k-1)*((n-1)+(1/2)))+h1;
  end
```

```
 h(n) = (1/M)* (G(1)+2*h1);
end
disp(h,'Filter Coefficients are h(n)=')
[hzm,fr]=frmag(h,256);
hzm_dB = 20*log10(hzm)./max(hzm);
subplot(2,1,1)
plot(2*fr,hzm)
a=gca();
xlabel('Normalized Digital Frequency W');
ylabel('Magnitude');
title('Frequency Response 0f FIR LPF using Frequency Sampling Technique with M = 15 with Cutoff Frequency = 0.466')
xgrid(2)
subplot(2,1,2)
plot(2*fr,hzm_dB)
a=gca();
xlabel('Normalized Digital Frequency W');
ylabel('Magnitude in dB');
title('Frequency Response 0f FIR LPF using Frequency Sampling Technique with M = 15 with Cutoff Frequency = 0.466')
xgrid(2)
//Result
//Filter Coefficients are h(n)=
//column 1 to 7
//
//  -0.0141289   -0.0019453   0.04   0.0122345   -0.0913880   -0.0180899   0.3133176
//
//column  8 to 14
//
//0.52     0.3133176   - 0.0180899   - 0.0913880   0.0122345     0.04   - 0.0019453
//
//column 15
//
//   - 0.0141289
```

Experiment: 10

Design of IIR Filters-Butterworth

Scilab code Solution 10.1 Digital IIR First Order Butterworth LPF Filter

```
//Caption: To design a digital IIR First Order
    Butterworth LPF Filter
//Using Bilinear Transformation
clear all;
clc;
close;
s = poly(0,'s');
Omegac = 0.2*%pi;   // Cutoff frequency
H = Omegac/(s+Omegac);  //Analog first order
    Butterworth filter tranfer function
T =1;//Sampling period T = 1 Second
z = poly(0,'z');
Hz = horner(H,(2/T)*((z-1)/(z+1))) //Bilinear
    Transformation
HW  =frmag(Hz(2),Hz(3),512);  //Frequency response
    for 512 points
W = 0:%pi/511:%pi;
a=gca();
```

```
a.thickness = 1;
plot(W/%pi,HW,'r')
a.foreground = 1;
a.font_style = 9;
xgrid(1)
xtitle('Magnitude Response of Single pole LPF Filter Cutoff frequency = 0.2*pi','Normalized Digital Frequency--->','Magnitude');
```

Scilab code Solution 10.2 HPF Using Digital Filter Transformation

```
//Caption: To design First Order Butterworth Low Pass Filter and covert it into
// HPF Using Digital Filter Transformation
clear all;
clc;
close;
s = poly(0,'s');
Omegac = 0.2*%pi; //Filter cutoff frequency
H = Omegac/(s+Omegac); //First order Butterworth IIR filter
T =1;//Sampling period T = 1 Second
z = poly(0,'z');
Hz_LPF = horner(H,(2/T)*((z-1)/(z+1))); //Bilinear Transformation
alpha = -(cos((Omegac+Omegac)/2))/(cos((Omegac-Omegac)/2));
HZ_HPF=horner(Hz_LPF,-(z+alpha)/(1+alpha*z))//LPF to HPF digital transformation
HW  =frmag(HZ_HPF(2),HZ_HPF(3),512); //Frequency response for 512 points
W = 0:%pi/511:%pi;
a=gca();
a.thickness = 1;
plot(W/%pi,HW,'r')
```

```
a.foreground = 1;
a.font_style = 9;
xgrid(1)
xtitle('Magnitude Response of Single pole HPF Filter
    Cutoff frequency = 0.2*pi','Normalized Digital
   Frequency W/pi--->','Magnitude');
```

Scilab code Solution 10.3 BPF using Digital Transformation

```
///Caption:To Design a Digital IIR Butterworth LPF
   Filter from Analog IIR
//Butterworth Filter and LPF to BPF using Digital
   Transformation
clear all;
clc;
close;
omegaP = 0.2*%pi; //Filter cutoff frequency
omegaL =  (1/5)*%pi;  //Lower Cutoff frequency for
   BSF
omegaU =  (3/5)*%pi;  //Upper Cutoff frequency for
   BSF
z=poly(0,'z');
H_LPF = (0.245)*(1+(z^-1))/(1-0.509*(z^-1)); //
   Bilinear transformation
alpha = (cos((omegaU+omegaL)/2)/cos((omegaU-omegaL)
   /2));//parameter 'alpha'
//parameter 'k'
k = (cos((omegaU - omegaL)/2)/sin((omegaU - omegaL)
   /2))*tan(omegaP/2);
NUM =-((z^2)-((2*alpha*k/(k+1))*z)+((k-1)/(k+1)));
DEN = (1-((2*alpha*k/(k+1))*z)+(((k-1)/(k+1))*(z^2))
   );
HZ_BPF=horner(H_LPF,NUM/DEN);//LPF to BPF conversion
    using digital transformation
disp(HZ_BPF,'Digital BPF IIR Filter H(Z)= ');
```

```
HW  =frmag(HZ_BPF(2),HZ_BPF(3),512);//frequency
    response
W = 0:%pi/511:%pi;
a=gca();
a.thickness = 1;
plot(W/%pi,HW,'r')
a.foreground = 1;
a.font_style = 9;
xgrid(1)
xtitle('Magnitude Response of BPF Filter cutoff
    frequency [0.2,0.6]','Normalized Digital
    Frequency--->','Magnitude');
```

Scilab code Solution 10.4 BSF using Digital Transformation

```
//Caption:To Design a Digital IIR Butterworth LPF
    Filter from Analog IIR
//Butterworth Filter and LPF to BSF using Digital
    Transformation
clear all;
clc;
close;
omegaP = 0.2*%pi;   //Filter cutoff frequency
omegaL =  (1/5)*%pi;   //Lower Cutoff frequency for
    BSF
omegaU =  (3/5)*%pi;   //Upper Cutoff frequency for
    BSF
z=poly(0,'z');
H_LPF = (0.245)*(1+(z^-1))/(1-0.509*(z^-1))//
    Bilinear transformation
alpha = (cos((omegaU+omegaL)/2)/cos((omegaU-omegaL)
    /2)); //parameter 'alpha'
k = tan((omegaU - omegaL)/2)*tan(omegaP/2); //
    parameter 'k'
NUM =((z^2)-((2*alpha/(1+k))*z)+((1-k)/(1+k)); //
```

```
Numerator
DEN = (1-((2*alpha/(1+k))*z)+(((1-k)/(1+k))*(z^2))); //Denominator
HZ_BSF=horner(H_LPF,NUM/DEN); //LPF to BSF conversion using digital transformation
HW  =frmag(HZ_BSF(2),HZ_BSF(3),512); //frequency response for 512 points
W = 0:%pi/511:%pi;
a=gca();
a.thickness = 1;
plot(W/%pi,HW,'r')
a.foreground = 1;
a.font_style = 9;
xgrid(1)
xtitle('Magnitude Response of BSF Filter cutoff freq [0.2,0.6] ','Normalized Digital Frequency--->','Magnitude');
```

Experiment: 11

Design of IIR Filters Chebyshev

Scilab code Solution 11.1 To Design the Digtial Chebyshev IIR Filter

```
//Program To Design the Digtial Chebyshev IIR Filter
clear;
clc;
close;
Wp = input('Enter the Digital Pass Band Edge
    Frequency');
Ws = input('Enter the Digital Stop Band Edge
    Frequency');
T = input('Sampling Interval')
OmegaP = (2/T)*tan(Wp/2)
OmegaS = (2/T)*tan(Ws/2)
Delta1 = input('Enter the Pass Band Ripple');
Delta2 = input('Enter the Stop Band Ripple');
Delta = sqrt(((1/Delta2)^2)-1)
Epsilon = sqrt(((1/Delta1)^2)-1)
N = (acosh(Delta/Epsilon))/(acosh(OmegaS/OmegaP))
N = ceil(N)
OmegaC = OmegaP/((((1/Delta1)^2)-1)^(1/(2*N)))
[pols,gn] = zpch1(N,Epsilon,OmegaP)
```

```
Hs = poly(gn,'s','coeff')/real(poly(pols,'s'))
z = poly(0,'z');
Hz = horner(Hs,((2/T)*((z-1)/(z+1))))
HW  =frmag(Hz(2),Hz(3),512);  //Frequency response for 512 points
W = 0:%pi/511:%pi;
a=gca();
a.thickness = 1;
plot(W/%pi,abs(HW),'r')
a.foreground = 1;
a.font_style = 9;
xgrid(1)
xtitle('Magnitude Response of Chebyshev LPF Filter','Normalized Digital Frequency--->','Magnitude in dB');
//RESULT
//Enter the Digital Pass Band Edge Frequency 0.2*%pi
//Enter the Digital Stop Band Edge Frequency 0.6*%pi
//Sampling Interval 1
// T  =
//
//     1.
// OmegaP  =
//
//     0.6498394
// OmegaS  =
//
//     2.7527638
//Enter the Pass Band Ripple 0.8
//Enter the Stop Band Ripple 0.2
// Delta  =
//
//     4.8989795
// Epsilon  =
//
//     0.75
// N  =
//
```

```
//      1.2079548
// N  =
//
//      2.
// OmegaC  =
//
//      0.7503699
// gn  =
//
//      0.2815275
// pols  =
//
//   - 0.2652958 + 0.5305916i  - 0.2652958 - 0.5305916i
// Hs  =
//
//                0.2815275
//      ---------------------------
//                                  2
//      0.3519094 + 0.5305916s + s
// Hz  =
//
//                                            2
//      0.2815275 + 0.5630550z + 0.2815275z
//      -------------------------------------
//                                            2
//      3.2907261 - 7.2961813z + 5.4130926z
//-->0.5*0.5629
// ans  =
//
//      0.28145
//
//-->Hz(2)= Hz(2)/5.4130926
// Hz  =
//
//                                            2
//      0.0520086 + 0.1040172z + 0.0520086z
//      -------------------------------------
```

```
//                                              2
//      3.2907261 - 7.2961813z + 5.4130926z
//
//-->Hz(3) = Hz(3)/5.4130926
// Hz   =
//
//                                              2
//      0.0520086 + 0.1040172z + 0.0520086z
//      ------------------------------------
//                                        2
//           0.6079198 - 1.3478767z + z
//
```

Experiment: 12

Decimation by polyphase decomposition

Scilab code Solution 12.1 Design of Ployphase Decimator

```
//Caption:Decimation by 2, Filter Length = 30
//Cutoff Frequency Wc = %pi/2
//Pass band Edge frequency fp = 0.25 and a Stop band
    edge frequency fs = 0.31
// Choose the number of cosine functions and create
   a dense grid
// in [0,0.25] and [0.31,0.5]
//magnitude for pass band = 1 & stop band = 0 (i.e)
   [1 0]
//Weighting function =[2 1]
clear;
clc;
close;
M = 30;  //Filter Length
D = 2;  //Decimation Factor = 2
Wc = %pi/2;  //Cutoff Frequency
Wp = Wc/(2*%pi);  //Passband Edge Frequency
Ws = 0.31;  //Stopband Edge Frequency
hn=eqfir(M,[0 Wp;Ws .5],[1 0],[2 1]);
```

```
disp(hn,'The LPF Filter Coefficients are:')
//Obtaining Polyphase Filter Coefficients from hn
p = zeros(D,M/D);
for k = 1:D
  for n = 1:(length(hn)/D)
    p(k,n) = hn(D*(n-1)+k);
  end
end
disp(p,'The Polyphase Decimator for D =2 are:')
//Result
//The LPF Filter Coefficients are:
//column 1 to 7
//0.0060203  - 0.0128037  - 0.0028534    0.0136687  - 0.0046761  - 0.0197002    0.0159915
//column  8 to 14
//0.0213811  - 0.0349808  - 0.0156251    0.0640230  - 0.0073600  - 0.1187325    0.0980522
//column 15 to 21
//0.4922476    0.4922476    0.0980522  - 0.1187325  - 0.0073600    0.0640230  - 0.0156251
//column 22 to 28
//- 0.0349808    0.0213811    0.0159915  - 0.0197002  - 0.0046761  0.0136687  - 0.0028534
//column 29 to 30
//- 0.0128037    0.0060203
//The Polyphase Decimator for D =2 are:
//column 1 to 7
//0.0060203  - 0.0028534  - 0.0046761    0.0159915  - 0.0349808    0.0640230  - 0.1187325
//- 0.0128037    0.0136687  - 0.0197002    0.0213811  - 0.0156251  - 0.0073600  0.0980522
//column  8 to 14
//0.4922476    0.0980522  - 0.0073600  - 0.0156251    0.0213811  - 0.0197002    0.0136687
```

```
//0.4922476  -0.1187325   0.0640230  -0.0349808   0.0159915  -0.0046761  -0.0028534
//column 15
//-0.0128037
//  0.0060203
```

Experiment: 13

Periodogram based Spectral Estimation

Scilab code Solution 13.1 Periodogram Estimate of Given Discrete Sequence

```
//Caption: Periodogram Estimate of Given Discrete
    Sequence
//x(n) ={1,0,2,0,3,1,0,2}
//using DFT
clear;
clc;
close;
N =8; //8-point DFT
x = [1,0,2,0,3,1,0,2]; //given discrete sequence
X = dft(x,-1); //8-point DFT of given discrete
    sequence
Pxx = (1/N)*(abs(X).^2); //Peridogram Estimate
disp(X,'DFT of x(n)is X(k)=')
disp(Pxx,'Peridogram of x(n) is Pxx(k/N)=')
figure(1)
a = gca();
a.data_bounds =[0,0;8,11];
plot2d3('gnn',[1:N],Pxx)
```

```
a.foreground = 5;
a.font_color = 5;
a.font_style = 5;
title('Peridogram Estimate')
xlabel('Discrete Frequency Variable K ------>')
ylabel('Periodogram Pxx (k /N) ----->')
//Result
//DFT of x(n)is X(k)=
//
//     9.
//   - 1.2928932 + 0.1213203i
//     2. + i
//   - 2.7071068 + 4.1213203i
//     3. - 3.674D-16i
//   - 2.7071068 - 4.1213203i
//     2. - i
//   - 1.2928932 - 0.1213203i
//
// Peridogram of x(n) is Pxx(k/N)=
//
//     10.125
//     0.2107864
//     0.625
//     3.0392136
//     1.125
//     3.0392136
//     0.625
//     0.2107864
```

Appendix

Scilab code AP 1

```
function [y]=sincnew(x)
i=find(x==0);
x(i)= 1;      // don't need this is /0 warning is off
y = sin(%pi*x)./(%pi*x);
y(i) = 1;
endfunction
```

sinc function