Scilab Manual for
Digital Signal Processing
by Prof R.Senthilkumar, Assistant Professor
Electronics Engineering
Institute of Road and Transport Technology[1]

Solutions provided by
Mr. R.Senthilkumar- Assistant Professor
Electronics Engineering
Institute of Road and Transport Technology

May 9, 2025

[1]Funded by a grant from the National Mission on Education through ICT, http://spoken-tutorial.org/NMEICT-Intro. This Scilab Manual and Scilab codes written in it can be downloaded from the "Migrated Labs" section at the website http://scilab.in

Contents

List of Scilab Solutions 3

1 Generation of Discrete Signals 4

2 Linear and Circular Convolution of two sequences 7

3 Circular convolution using FFT 11

4 Linear Convolution using Circular Convolution 13

5 Calculation of FFT and IFFT of a sequence 15

6 Time and Frequency Response of LTI systems 17

7 Sampling, Verification of Sampling and Effect of aliasing 20

8 Design of FIR Filters Window Design 22

9 Design of FIR Filters Frequency Sampling 29

10 Design of IIR Filters- Butterworth 32

11 Design of IIR Filters Chebyshev 37

12 Decimation by polyphase decomposition 41

13 Periodogram based Spectral Estimation 44

List of Experiments

Solution 1.1	Unit Sample Sequence	4
Solution 1.2	Unit Step Sequence	4
Solution 1.3	Discrete Ramp Sequence	5
Solution 1.4	Exponentially Decreasing Signal	6
Solution 1.5	Exponentially Increasing Signal	6
Solution 2.1	Program for Linear Convolution	7
Solution 2.2	Program to find the Cicrcular Convolution	9
Solution 3.1	Performing Circular COnvolution Using DFT IDFT method	11
Solution 4.1	Performing Linear Convolution using Circular Convolution	13
Solution 5.5	Performing FFT and IFFT of a discrete sequence	15
Solution 6.1	Time and Frequency Response	17
Solution 7.1	Sampling and Reconstruction of a Signal	20
Solution 8.1	Program to Design FIR Low Pass Filter	22
Solution 8.2	rogram to Design FIR High Pass Filter	23
Solution 8.3	Program to Design FIR Band Pass Filter	25
Solution 8.4	Program to Design FIR Band Reject Filter	27
Solution 9.1	Design of FIR LPF Filter using Frequecny Sampling Technique	29
Solution 10.1	Digital IIR First Order Butterworth LPF Filter	32
Solution 10.2	HPF Using Digital Filter Transformation	33
Solution 10.3	BPF using Digital Transformation	34
Solution 10.4	BSF using Digital Transformation	35
Solution 11.1	To Design the Digtial Chebyshev IIR Filter	37
Solution 12.1	Design of Ployphase Decimator	41
Solution 13.1	Periodogram Estimate of Given Discrete Sequence	44
AP 1	sinc function	46

Experiment: 1

Generation of Discrete Signals

Scilab code Solution 1.1 Unit Sample Sequence

```
1  //Caption:Unit Sample Sequence
2  clear;
3  clc;
4  close;
5  L = 4;  //Upperlimit
6  n = -L:L;
7  x = [zeros(1,L),1,zeros(1,L)];
8  b = gca();
9  b.y_location = "middle";
10 plot2d3('gnn',n,x)
11 a=gce();
12 a.children(1).thickness =4;
13 xtitle('Graphical Representation of Unit Sample
      Sequence','n','x[n]');
```

Scilab code Solution 1.2 Unit Step Sequence

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

```
2  clear;
3  clc;
4  close;
5  L = 4;  //Upperlimit
6  n = -L:L;
7  x = [zeros(1,L),ones(1,L+1)];
8  a=gca();
9  a.y_location = "middle";
10 plot2d3('gnn',n,x)
11 title('Graphical Representation of Unit Step Signal'
     )
12 xlabel('                                          n'
     );
13 ylabel('                                           x
     [n]');
```

Scilab code Solution 1.3 Discrete Ramp Sequence

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

Scilab code Solution 1.4 Exponentially Decreasing Signal

```
1  //Caption: Exponentially Decreasing Signal
2  clear;
3  clc;
4  close;
5  a =0.5;
6  n = 0:10;
7  x = (a)^n;
8  a=gca();
9  a.x_location = "origin";
10 a.y_location = "origin";
11 plot2d3('gnn',n,x)
12 a.thickness = 2;
13 xtitle('Graphical Representation of Exponentially
      Decreasing Signal','n','x[n]');
```

Scilab code Solution 1.5 Exponentially Increasing Signal

```
1  //Caption:Exponentially Increasing Signal
2  clear;
3  clc;
4  close;
5  a =1.5;
6  n =1:10;
7  x = (a)^n;
8  a=gca();
9  a.thickness = 2;
10 plot2d3('gnn',n,x)
11 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

```
1  //Caption:Program for Linear Convolution
2  clc;
3  clear all;
4  close ;
5  x = input('enter x seq');
6  h = input('enter h seq');
7  m = length(x);
8  n = length(h);
9  //Method 1 Using Direct Convolution Sum Formula
10 for i = 1:n+m-1
11     conv_sum = 0;
12     for j = 1:i
13         if (((i-j+1) <= n)&(j <= m))
14             conv_sum = conv_sum + x(j)*h(i-j+1);
15         end;
16         y(i) = conv_sum;
17     end;
18 end;
19 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

```
1  //Caption: Program to find the Cicrcular Convolution
       of given
2  //discrete sequences using Matrix method
3
4  clear;
5  clc;
6  x1 = [2,1,2,1]; //First sequence
7  x2 = [1,2,3,4]; //Second sequence
8  m = length(x1); //length of first sequence
9  n = length(x2); //length of second sequence
10 //To make length of x1 and x2 are Equal
11 if (m >n)
12    for i = n+1:m
13      x2(i) = 0;
14    end
15 elseif (n>m)
16    for i = m+1:n
17      x1(i) = 0;
18    end
19 end
20 N = length(x1);
21 x3 = zeros(1,N); //x3 = Circular convolution result
22 a(1) = x2(1);
23 for j = 2:N
24   a(j) = x2(N-j+2);
25 end
26 for i =1:N
27   x3(1) = x3(1)+x1(i)*a(i);
28 end
29 X(1,:)=a;
30 //Calculation of circular convolution
31 for k = 2:N
```

```
32      for j =2:N
33         x2(j) = a(j-1);
34      end
35         x2(1) = a(N);
36         X(k,:)= x2;
37      for i = 1:N
38          a(i) = x2(i);
39          x3(k) = x3(k)+x1(i)*a(i);
40      end
41  end
42  disp(X,'Circular Convolution Matrix x2[n]=')
43  disp(x3,'Circular Convolution Result x3[n] = ')
44  //Result
45  //Circular Convolution Matrix x2[n]=
46  //
47  //     1.     4.     3.     2.
48  //     2.     1.     4.     3.
49  //     3.     2.     1.     4.
50  //     4.     3.     2.     1.
51  //
52  // Circular Convolution Result x3[n] =
53  //
54  //     14.    16.    14.    16.
```

Experiment: 3

Circular convolution using FFT

Scilab code Solution 3.1 Performing Circular COnvolution Using DFT IDFT method

```
1  //Caption:Performing Circular COnvolution Using DFT-
       IDFT method
2  clear all;
3  clc;
4  close;
5  L = 4; //Length of the Sequence
6  N = 4;  // N-point DFT
7  x1 = [2,1,2,1];
8  x2 = [1,2,3,4];
9  //Computing DFT
10 X1 = fft(x1,-1);
11 X2 = fft(x2,-1);
12 disp(X1,'DFT of x1[n] is X1(k)=')
13 disp(X2,'DFT of x1[n] is X2(k)=')
14 //Multiplication of 2 DFTs
15 X3 = X1.*X2;
16 disp(X3,'DFT of x3[n] is X3(k)=')
17 //Circular Convolution Result
18 x3 =abs(fft(X3,1))
19 disp(x3,'Circular Convolution Result x3[n]=')
```

```
20 //Result
21 // DFT of x1[n] is X1(k)=
22 //
23 //     6.     0     2.     0
24 //
25 // DFT of x1[n] is X2(k)=
26 //
27 //     10.  - 2. + 2.i  - 2.  - 2. - 2.i
28 //
29 // DFT of x3[n] is X3(k)=
30 //
31 //     60.     0  - 4.     0
32 //
33 // Circular Convolution Result x3[n]=
34 //
35 //     14.     16.     14.     16.
```

Experiment: 4

Linear Convolution using Circular Convolution

Scilab code Solution 4.1 Performing Linear Convolution using Circular Convolution

```
1  //Caption: Performing Linear Convolution using
       Circular Convolution
2
3  clear;
4  clc;
5  close;
6  h = [1,2,3];     //Impulse Response of LTI System
7  x = [1,2,2,1];   //Input Response of LTI System
8  N1 = length(x);
9  N2 = length(h);
10 N = N1+N2-1
11 disp(N,'Length of Output Response y(n)')
12 //Padding zeros to Make Length of 'h' and 'x'
13 //Equal to length of output response 'y'
14 h1 = [h,zeros(1,N-N2)];
15 x1 = [x,zeros(1,N-N1)];
16 //Computing FFT
17 H = fft(h1,-1);
```

```
18 X = fft(x1,-1);
19 //Multiplication of 2 DFTs
20 Y = X.*H
21 //Linear Convolution Result
22 y =abs(fft(Y,1))
23 disp(X,'DFT of i/p X(k)=')
24 disp(H,'DFT of impulse sequence H(k)=')
25 disp(Y,'DFT of Linear Filter o/p Y(k)=')
26 disp(y,'Linear Convolution result y[n]=')
27 //Result
28 // Length of Output Response y(n)
29 //
30 //    6.
31 //
32 // DFT of i/p X(k)=
33 //
34 //    6.  - 3.4641016i    0    0    0    3.4641016i
35 //
36 // DFT of impulse sequence H(k)=
37 //
38 //    6.    0.5 - 4.330127i  - 1.5 + 0.8660254i
     2.  - 1.5 - 0.8660254i    0.5 + 4.330127i
39 //
40 // DFT of Linear Filter o/p Y(k)=
41 //
42 //    36.  - 15. - 1.7320508i    0    0    0  - 15.
     + 1.7320508i
43 //
44 // Linear Convolution result y[n]=
45 //
46 //    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

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

```
19 b.children(1).thickness =3;
20 xtitle('Graphical Representation of Discrete
      Sequence','n','x[n]');
21 subplot(2,1,2)
22 a=gce();
23 a.data_bounds=[0,0;5,10];
24 plot2d3('gnn',0:length(X)-1,abs(X))
25 b = gce();
26 b.children(1).thickness =3;
27 xtitle('Graphical Representation of Discrete
      Spectrum','k','X(k)');
28 //Result
29 //FFT of x[n] is X(k)=
30 //
31 //     10.  - 2. + 2.i  - 2.  - 2. - 2.i
32 //
33 //IFFT of X(k) is x[n]=
34 //
35 //     1.    2.    3.    4.
```

Experiment: 6

Time and Frequency Response of LTI systems

Scilab code Solution 6.1 Time and Frequency Response

```
1 //Caption: Program to generate and plot the impulse
     response and frequency
2 //response of a Linear constant coefficient first
     order Differential Equation
3 //[1].Impulse response h(t)= exp(-a*t)u(t), A>0
4 //[2].Frequency response H(jw) = 1/(jw+a)
5 clear;
6 clc;
7 close;
8 //[1]. To generate and plot the impulse response
9 a =1;     //Constant coefficient a =1
10 Dt = 0.005;
11 t = 0:Dt:10;
12 ht = exp(-a*t);
13 figure(1)
14 a = gca();
15 a.y_location = "origin";
16 plot(t,ht);
17 xlabel('time t ------>');
```

```
18 ylabel('h(t)')
19 title('Impulse Repsonse of Ist Order Linear Constant
       Coeff. Differential Equ.')
20 //
21 //[2]. Finding Frequency response using Continuous
     Time Fourier Transform
22 Wmax = 2*%pi*1;          //Analog Frequency = 1Hz
23 K = 4;
24 k = 0:(K/1000):K;
25 W = k*Wmax/K;
26 HW = ht* exp(-sqrt(-1)*t'*W) * Dt;
27 HW_Mag = abs(HW);
28 W = [-mtlb_fliplr(W), W(2:1001)]; // Omega from -
     Wmax to Wmax
29 HW_Mag = [mtlb_fliplr(HW_Mag),HW_Mag(2:1001)];
30 [HW_Phase,db] = phasemag(HW);
31 HW_Phase = [-mtlb_fliplr(HW_Phase),HW_Phase(2:1001)
     ];
32 figure(2)
33 //Plotting Magnitude Response
34 subplot(2,1,1);
35 a = gca();
36 a.y_location = "origin";
37 plot(W,HW_Mag);
38 xlabel('Frequency in Radians/Seconds--> W');
39 ylabel('abs(H(jW))')
40 title('Magnitude Response')
41 //Plotting Phase Reponse
42 subplot(2,1,2);
43 a = gca();
44 a.y_location = "origin";
45 a.x_location = "origin";
46 plot(W,HW_Phase*%pi/180);
47 xlabel('                          Frequency in
     Radians/Seconds--> W');
48 ylabel('
                                                  <H
     (jW)')
```

```
49 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

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

```
13 //Continuous Time Signal
14 xa = exp(-A*abs(t));
15 //Discrete Time Signal
16 Fs =input('Enter the Sampling Frequency in Hertz');
     //Fs = 1Hz,2Hz,4Hz,20Hz,100Hz
17 Ts = 1/Fs;
18 nTs = -2:Ts:2;
19 x = exp(-A*abs(nTs));
20 // Analog Signal reconstruction
21 Dt = 0.005;
22 t = -2:Dt:2;
23 Xa = x *sincnew(Fs*(ones(length(nTs),1)*t-nTs'*ones
     (1,length(t))));
24 //Plotting the original signal and reconstructed
     signal
25 subplot(2,1,1);
26 a =gca();
27 a.x_location = "origin";
28 a.y_location = "origin";
29 plot(t,xa);
30 xlabel('t in sec.');
31 ylabel('xa(t)')
32 title('Original Analog Signal')
33 subplot(2,1,2);
34 a =gca();
35 a.x_location = "origin";
36 a.y_location = "origin";
37 xlabel('t in sec.');
38 ylabel('xa(t)')
39 title('Reconstructed Signal using sinc function, Fs
     = 100Hz');
40 plot(t,Xa);
```

Experiment: 8

Design of FIR Filters Window Design

Scilab code Solution 8.1 Program to Design FIR Low Pass Filter

```
1  //Caption: Program to Design FIR Low Pass Filter
2  clc;
3  close;
4  M = input('Enter the Odd Filter Length =');
                  //Filter length
5  Wc = input('Enter the Digital Cutoff frequency =');
     //Digital Cutoff frequency
6  Tuo = (M-1)/2      //Center Value
7  for n = 1:M
8      if (n == Tuo+1)
9        hd(n) = Wc/%pi;
10     else
11       hd(n) =  sin(Wc*((n-1)-Tuo))/(((n-1)-Tuo)*%pi)
             ;
12     end
13 end
14 //Rectangular Window
15 for n = 1:M
16   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

```
1  //Caption: Program to Design FIR High Pass Filter
2  clear;
3  clc;
4  close;
5  M = input('Enter the Odd Filter Length =');
                  //Filter length
6  Wc = input('Enter the Digital Cutoff frequency =');
     //Digital Cutoff frequency
7  Tuo = (M-1)/2      //Center Value
8  for n = 1:M
9      if (n == Tuo+1)
10       hd(n) = 1-Wc/%pi;
11     else
12       hd(n) = (sin(%pi*((n-1)-Tuo)) -sin(Wc*((n-1)-
            Tuo)))/(((n-1)-Tuo)*%pi);
13     end
14 end
15 //Rectangular Window
16 for n = 1:M
17   W(n) = 1;
18 end
19 //Windowing Fitler Coefficients
20 h = hd.*W;
21 disp(h,'Filter Coefficients are')
22 [hzm,fr]=frmag(h,256);
23 hzm_dB = 20*log10(hzm)./max(hzm);
24 subplot(2,1,1)
25 plot(2*fr,hzm)
26 xlabel('Normalized Digital Frequency W');
27 ylabel('Magnitude');
28 title('Frequency Response 0f FIR HPF using
     Rectangular window')
29 xgrid(1)
30 subplot(2,1,2)
31 plot(2*fr,hzm_dB)
32 xlabel('Normalized Digital Frequency W');
```

```
33 ylabel('Magnitude in dB');
34 title('Frequency Response 0f FIR HPF using
      Rectangular window')
35 xgrid(1)
36 //Result
37 //Enter the Odd Filter Length = 5
38 //Enter the Digital Cutoff frequency = %pi/4
39 //Filter Coefficients are
40 //
41 //   - 0.1591549
42 //   - 0.2250791
43 //     0.75
44 //   - 0.2250791
45 //   - 0.1591549
```

Scilab code Solution 8.3 Program to Design FIR Band Pass Filter

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

```
17          n
18       hd(n) = (sin(Wc2*((n-1)-Tuo)) -sin(Wc1*((n-1)-
            Tuo)))/(((n-1)-Tuo)*%pi);
19    end
20    if(abs(hd(n))<(0.00001))
21      hd(n)=0;
22    end
23  end
24  hd;
25  //Rectangular Window
26  for n = 1:M
27    W(n) = 1;
28  end
29  //Windowing Fitler Coefficients
30  h = hd.*W;
31  disp(h,'Filter Coefficients are')
32  [hzm,fr]=frmag(h,256);
33  hzm_dB = 20*log10(hzm)./max(hzm);
34  subplot(2,1,1)
35  plot(2*fr,hzm)
36  xlabel('Normalized Digital Frequency W');
37  ylabel('Magnitude');
38  title('Frequency Response 0f FIR BPF using
       Rectangular window')
39  xgrid(1)
40  subplot(2,1,2)
41  plot(2*fr,hzm_dB)
42  xlabel('Normalized Digital Frequency W');
43  ylabel('Magnitude in dB');
44  title('Frequency Response 0f FIR BPF using
       Rectangular window')
45  xgrid(1)
46  //Result
47  //Enter the Odd Filter Length = 11
48  //Enter the Digital Cutoff frequency = [%pi/4,3*%pi
       /4]
49  //Filter Coefficients are
50  // 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

```
1  //Caption: Program to Design FIR Band Reject Filter
2  clear ;
3  clc;
4  close;
5  M = input('Enter the Odd Filter Length =');
                  //Filter length
6  //Digital Cutoff frequency [Lower Cutoff, Upper
      Cutoff]
7  Wc = input('Enter the Digital Cutoff frequency =');
8  Wc2 = Wc(2)
9  Wc1 = Wc(1)
10 Tuo = (M-1)/2      //Center Value
11 hd = zeros(1,M);
12 W = zeros(1,M);
13 for n = 1:M
14  if (n == Tuo+1)
15  hd(n) = 1-((Wc2-Wc1)/%pi);
16  else
17  hd(n)=(sin(%pi*((n-1)-Tuo))-sin(Wc2*((n-1)-Tuo))+
       sin(Wc1*((n-1)-Tuo)))/(((n-1)-Tuo)*%pi);
18  end
19   if(abs(hd(n))<(0.00001))
20     hd(n)=0;
21   end
22 end
23
24 //Rectangular Window
25 for n = 1:M
26   W(n) = 1;
27 end
28 //Windowing Fitler Coefficients
```

```
29 h = hd.*W;
30 disp(h,'Filter Coefficients are')
31 [hzm,fr]=frmag(h,256);
32 hzm_dB = 20*log10(hzm)./max(hzm);
33 subplot(2,1,1)
34 plot(2*fr,hzm)
35 xlabel('Normalized Digital Frequency W');
36 ylabel('Magnitude');
37 title('Frequency Response 0f FIR BSF using
      Rectangular window')
38 xgrid(1)
39 subplot(2,1,2)
40 plot(2*fr,hzm_dB)
41 xlabel('Normalized Digital Frequency W');
42 ylabel('Magnitude in dB');
43 title('Frequency Response 0f FIR BSF using
      Rectangular window')
44 xgrid(1)
45 //Result
46 //Enter the Odd Filter Length = 11
47 //Enter the Digital Cutoff frequency =[%pi/3,2*%pi
      /3]
48 //Filter Coefficients are
49 //column 1 to 9
50 //    0.  - 0.1378322    0.    0.2756644    0.
      0.6666667    0.    0.2756644    0.
51 //column 10 to 11
52 //  - 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

```
1  //Cpation: Design of FIR LPF Filter using Frequecny
       Sampling Technique
2
3  clear;
4  clc;
5  close;
6  M =15;
7  Hr = [1,1,1,1,0.4,0,0,0];
8  for k =1:length(Hr)
9      G(k)=((-1)^(k-1))*Hr(k);
10 end
11 h = zeros(1,M);
12 U = (M-1)/2
13 for n = 1:M
14   h1 = 0;
15   for k = 2:U+1
16     h1 =G(k)*cos((2*%pi/M)*(k-1)*((n-1)+(1/2)))+h1;
17   end
```

```
18  h(n) = (1/M)* (G(1)+2*h1);
19 end
20 disp(h,'Filter Coefficients are h(n)=')
21 [hzm,fr]=frmag(h,256);
22 hzm_dB = 20*log10(hzm)./max(hzm);
23 subplot(2,1,1)
24 plot(2*fr,hzm)
25 a=gca();
26 xlabel('Normalized Digital Frequency W');
27 ylabel('Magnitude');
28 title('Frequency Response 0f FIR LPF using Frequency
       Sampling Technique with M = 15 with Cutoff
      Frequency = 0.466')
29 xgrid(2)
30 subplot(2,1,2)
31 plot(2*fr,hzm_dB)
32 a=gca();
33 xlabel('Normalized Digital Frequency W');
34 ylabel('Magnitude in dB');
35 title('Frequency Response 0f FIR LPF using Frequency
       Sampling Technique with M = 15 with Cutoff
      Frequency = 0.466')
36 xgrid(2)
37 //Result
38 //Filter Coefficients are h(n)=
39 //column 1 to 7
40 //
41 // -0.0141289  -0.0019453  0.04  0.0122345
      -0.0913880  -0.0180899  0.3133176
42 //
43 //column  8 to 14
44 //
45 //0.52    0.3133176  - 0.0180899  - 0.0913880
      0.0122345    0.04  - 0.0019453
46 //
47 //column 15
48 //
49 //  - 0.0141289
```

Experiment: 10

Design of IIR Filters-Butterworth

Scilab code Solution 10.1 Digital IIR First Order Butterworth LPF Filter

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

```
15 a.thickness = 1;
16 plot(W/%pi,HW,'r')
17 a.foreground = 1;
18 a.font_style = 9;
19 xgrid(1)
20 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

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

```
19 a.foreground = 1;
20 a.font_style = 9;
21 xgrid(1)
22 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

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

```
18 HW  =frmag(HZ_BPF(2),HZ_BPF(3),512);//frequency
     response
19 W = 0:%pi/511:%pi;
20 a=gca();
21 a.thickness = 1;
22 plot(W/%pi,HW,'r')
23 a.foreground = 1;
24 a.font_style = 9;
25 xgrid(1)
26 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

```
1 //Caption:To Design a Digital IIR Butterworth LPF
     Filter from Analog IIR
2 //Butterworth Filter and LPF to BSF using Digital
     Transformation
3 clear all;
4 clc;
5 close;
6 omegaP = 0.2*%pi;    //Filter cutoff frequency
7 omegaL =  (1/5)*%pi;    //Lower Cutoff frequency for
    BSF
8 omegaU =  (3/5)*%pi;    //Upper Cutoff frequency for
    BSF
9 z=poly(0,'z');
10 H_LPF = (0.245)*(1+(z^-1))/(1-0.509*(z^-1))//
     Bilinear transformation
11 alpha = (cos((omegaU+omegaL)/2)/cos((omegaU-omegaL)
     /2)); //parameter 'alpha'
12 k = tan((omegaU - omegaL)/2)*tan(omegaP/2); //
     parameter 'k'
13 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

```
1  //Program To Design the Digtial Chebyshev IIR Filter
2  clear;
3  clc;
4  close;
5  Wp = input('Enter the Digital Pass Band Edge
       Frequency');
6  Ws = input('Enter the Digital Stop Band Edge
       Frequency');
7  T = input('Sampling Interval')
8  OmegaP = (2/T)*tan(Wp/2)
9  OmegaS = (2/T)*tan(Ws/2)
10 Delta1 = input('Enter the Pass Band Ripple');
11 Delta2 = input('Enter the Stop Band Ripple');
12 Delta = sqrt(((1/Delta2)^2)-1)
13 Epsilon = sqrt(((1/Delta1)^2)-1)
14 N = (acosh(Delta/Epsilon))/(acosh(OmegaS/OmegaP))
15 N = ceil(N)
16 OmegaC = OmegaP/((((1/Delta1)^2)-1)^(1/(2*N)))
17 [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
//      ------------------------------------
```

```
90  //                                                 2
91  //      3.2907261 - 7.2961813z + 5.4130926z
92  //
93  //-->Hz(3) = Hz(3)/5.4130926
94  // Hz  =
95  //
96  //                                                 2
97  //      0.0520086 + 0.1040172z + 0.0520086z
98  //     ------------------------------------------
99  //                                       2
100 //           0.6079198 - 1.3478767z + z
101 //
```

Experiment: 12

Decimation by polyphase decomposition

Scilab code Solution 12.1 Design of Ployphase Decimator

```
1  //Caption:Decimation by 2, Filter Length = 30
2  //Cutoff Frequency Wc = %pi/2
3  //Pass band Edge frequency fp = 0.25 and a Stop band
       edge frequency fs = 0.31
4  // Choose the number of cosine functions and create
      a dense grid
5  // in [0,0.25] and [0.31,0.5]
6  //magnitude for pass band = 1 & stop band = 0 (i.e)
      [1 0]
7  //Weighting function =[2 1]
8  clear;
9  clc;
10 close;
11 M = 30;  //Filter Length
12 D = 2;  //Decimation Factor = 2
13 Wc = %pi/2;  //Cutoff Frequency
14 Wp = Wc/(2*%pi);  //Passband Edge Frequency
15 Ws = 0.31;  //Stopband Edge Frequency
16 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
```

```
48 //0.4922476  - 0.1187325     0.0640230  - 0.0349808
       0.0159915  - 0.0046761  - 0.0028534
49 //column 15
50 //- 0.0128037
51 //   0.0060203
```

Experiment: 13

Periodogram based Spectral Estimation

Scilab code Solution 13.1 Periodogram Estimate of Given Discrete Sequence

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

```
17 a.foreground = 5;
18 a.font_color = 5;
19 a.font_style = 5;
20 title('Peridogram Estimate')
21 xlabel('Discrete Frequency Variable K ------>')
22 ylabel('Periodogram Pxx (k /N) ------>')
23 //Result
24 //DFT of x(n)is X(k)=
25 //
26 //    9.
27 //  - 1.2928932 + 0.1213203i
28 //    2. + i
29 //  - 2.7071068 + 4.1213203i
30 //    3. - 3.674D-16i
31 //  - 2.7071068 - 4.1213203i
32 //    2. - i
33 //  - 1.2928932 - 0.1213203i
34 //
35 // Peridogram of x(n) is Pxx(k/N)=
36 //
37 //    10.125
38 //    0.2107864
39 //    0.625
40 //    3.0392136
41 //    1.125
42 //    3.0392136
43 //    0.625
44 //    0.2107864
```

Appendix

Scilab code AP 1

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

sinc function