

Scilab Manual for
Digital Signal Processing
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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 Circular Convolution	9
Solution 3.1	Performing Circular CONvolution Using DFT IDFT method	11
Solution 4.1	Performing Linear Convolution using Circular Con- volution	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 Sam- pling 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 Digital 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
```



```

    =')
20 //Method 2 Using Inbuilt Function
21 f = convol(x,h)
22 disp(f,'Convolution Sum Result using Inbuilt Funtion
    =')
23 //Method 3 Using frequency Domain multiplication
24 N = n+m-1;
25 x = [x zeros(1,N-m)];
26 h = [h zeros(1,N-n)];
27 f1 = fft(x)
28 f2 = fft(h)
29 f3 = f1.*f2;    // freq domain multiplication
30 f4 = ifft(f3)
31 disp(f4,'Convolution Sum Result DFT – IDFT method =')
    )
32 //f4 = real(f4)
33 subplot(3,1,1);
34 plot2d3('gnn',x)
35 xtitle('Graphical Representation of Input signal x')
    ;
36 subplot(3,1,2);
37 plot2d3('gnn',h)
38 xtitle('Graphical Representation of Impulse signal h
    ');
39 subplot(3,1,3);
40 plot2d3('gnn',y)
41 xtitle('Graphical Representation of Output signal y')
    );
42 //Result
43 //enter x seq [1 1 1 1]
44 //enter h seq [1 2 3]
45 // Convolution Sum using Direct Formula Method =
46 //      1.      3.      6.      6.      5.      3.
47 // Convolution Sum Result using Inbuilt Funtion =
48 //      1.      3.      6.      6.      5.      3.
49 // Convolution Sum Result DFT – IDFT method =
50 //      1.      3.      6.      6.      5.      3.

```

Scilab code Solution 2.2 Program to find the Circular Convolution

```
1 //Caption: Program to find the Circular 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,:) = x3;
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
39 //      2.   - 1.5 - 0.8660254i    0.5 + 4.330127i
40 //
41 // DFT of Linear Filter o/p Y(k)=
42 //
43 //      36.   - 15. - 1.7320508i    0    0    0   - 15.
44 //      + 1.7320508i
45 //
46 // Linear Convolution result y[n]=
47 //
48 //      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 = [-mtlbfliplr(W), W(2:1001)]; // Omega from -
      Wmax to Wmax
29 HW_Mag = [mtlbfliplr(HW_Mag),HW_Mag(2:1001)];
30 [HW_Phase,db] = phasemag(HW);
31 HW_Phase = [-mtlbfliplr(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('
      (jW)')

```

<H

```
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        ;
13    end
14 end
15 //Rectangular Window
16 for n = 1:M
17     W(n) = 1;
```

```

17 end
18 //Windowing Filter Coefficients
19 h = hd.*W;
20 disp(h,'Filter Coefficients are')
21
22 [hzm,fr]=fzmag(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 of FIR LPF 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 of FIR LPF using
        Rectangular window')
35 xgrid(1)
36 //Result
37 //Enter the Odd Filter Length = 7
38 //Enter the Digital Cutoff frequency = %pi/2
39 //
40 // Filter Coefficients are
41 //
42 // - 0.1061033
43 // 1.949D-17 = 0.0
44 // 0.3183099
45 // 0.5
46 // 0.3183099
47 // 1.949D-17 = 0.0
48 // - 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 of 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 of 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:M
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 Filter Coefficients
30 h = hd.*W;
31 disp(h,'Filter Coefficients are')
32 [hzm,fr]=fzmag(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 of 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 of 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 Filter 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 Of 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 Of 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 Frequency Sampling Technique

```
1 //Caption: Design of FIR LPF Filter using Frequency
   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 Of 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 Of 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
42  // -0.0913880  -0.0180899  0.3133176
43  //
44  //column 8 to 14
45  //0.52  0.3133176  - 0.0180899  - 0.0913880
46  // 0.0122345  0.04  - 0.0019453
47  //
48  //column 15
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 =fsmag(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
14 DEN = (1-((2*alpha/(1+k))*z)+(((1-k)/(1+k))*(z^2)));
    //Denominator
15 HZ_BSF=horner(H_LPF,NUM/DEN); //LPF to BSF
    conversion using digital transformation
16 HW =frmag(HZ_BSF(2),HZ_BSF(3),512); //frequency
    response for 512 points
17 W = 0:%pi/511:%pi;
18 a=gca();
19 a.thickness = 1;
20 plot(W/%pi,HW,'r')
21 a.foreground = 1;
22 a.font_style = 9;
23 xgrid(1)
24 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 Digital Chebyshev IIR Filter

```
1 //Program To Design the Digital 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)
```

```

18 Hs = poly(gn,'s','coeff')/real(poly(pols,'s'))
19 z = poly(0,'z');
20 Hz = horner(Hs,((2/T)*((z-1)/(z+1))))
21 HW =frmag(Hz(2),Hz(3),512); //Frequency response
    for 512 points
22 W = 0:%pi/511:%pi;
23 a=gca();
24 a.thickness = 1;
25 plot(W/%pi,abs(HW),'r')
26 a.foreground = 1;
27 a.font_style = 9;
28 xgrid(1)
29 xtitle('Magnitude Response of Chebyshev LPF Filter',
    'Normalized Digital Frequency—>','Magnitude in
    dB');
30 //RESULT
31 //Enter the Digital Pass Band Edge Frequency 0.2*%pi
32 //Enter the Digital Stop Band Edge Frequency 0.6*%pi
33 //Sampling Interval 1
34 // T =
35 //
36 // 1.
37 // OmegaP =
38 //
39 // 0.6498394
40 // OmegaS =
41 //
42 // 2.7527638
43 //Enter the Pass Band Ripple 0.8
44 //Enter the Stop Band Ripple 0.2
45 // Delta =
46 //
47 // 4.8989795
48 // Epsilon =
49 //
50 // 0.75
51 // N =
52 //

```

```

53 //      1.2079548
54 // N    =
55 //
56 //      2.
57 // OmegaC =
58 //
59 //      0.7503699
60 // gn    =
61 //
62 //      0.2815275
63 // pols  =
64 //
65 //      - 0.2652958 + 0.5305916 i      - 0.2652958 -
66 //      0.5305916 i
67 //
68 //      0.2815275
69 //      -----
70 //                                     2
71 //      0.3519094 + 0.5305916 s + s
72 // Hz    =
73 //
74 //                                     2
75 //      0.2815275 + 0.5630550 z + 0.2815275 z
76 //      -----
77 //                                     2
78 //      3.2907261 - 7.2961813 z + 5.4130926 z
79 // -->0.5*0.5629
80 // ans   =
81 //
82 //      0.28145
83 //
84 // -->Hz(2)= Hz(2)/5.4130926
85 // Hz    =
86 //
87 //                                     2
88 //      0.0520086 + 0.1040172 z + 0.0520086 z
89 //      -----

```



```

90 //
91 //      3.2907261 - 7.2961813 z + 5.4130926 z2
92 //
93 //—>Hz(3) = Hz(3)/5.4130926
94 // Hz =
95 //
96 //
97 //      0.0520086 + 0.1040172 z + 0.0520086 z2
98 //      —————
99 //                                2
100 //      0.6079198 - 1.3478767 z + 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]);
```

```

17 disp(hn,'The LPF Filter Coefficients are:')
18 //Obtaining Polyphase Filter Coefficients from hn
19 p = zeros(D,M/D);
20 for k = 1:D
21     for n = 1:(length(hn)/D)
22         p(k,n) = hn(D*(n-1)+k);
23     end
24 end
25 disp(p,'The Polyphase Decimator for D =2 are:')
26 //Result
27 //The LPF Filter Coefficients are:
28 //column 1 to 7
29 //0.0060203   - 0.0128037   - 0.0028534       0.0136687
   //      - 0.0046761   - 0.0197002       0.0159915
30
31 //column 8 to 14
32 //0.0213811   - 0.0349808   - 0.0156251       0.0640230
   //      - 0.0073600   - 0.1187325       0.0980522
33 //column 15 to 21
34 //0.4922476       0.4922476       0.0980522   - 0.1187325
   //      - 0.0073600       0.0640230   - 0.0156251
35 //column 22 to 28
36 //- 0.0349808       0.0213811       0.0159915   - 0.0197002
   //      - 0.0046761   0.0136687   - 0.0028534
37
38 //column 29 to 30
39 //- 0.0128037       0.0060203
40
41 //The Polyphase Decimator for D =2 are:
42 //column 1 to 7
43 //0.0060203   - 0.0028534   - 0.0046761       0.0159915
   //      - 0.0349808       0.0640230   - 0.1187325
44 //- 0.0128037       0.0136687   - 0.0197002       0.0213811
   //      - 0.0156251   - 0.0073600   0.0980522
45
46 //column 8 to 14
47 //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 + 0.1213203 i
28 //      2. + i
29 //      - 2.7071068 + 4.1213203 i
30 //      3. - 3.674D-16i
31 //      - 2.7071068 - 4.1213203 i
32 //      2. - i
33 //      - 1.2928932 - 0.1213203 i
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 11 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
