Scilab Manual for Digital Signal Processing Lab by Dr R Kumaraswamy Electronics Engineering Siddaganga Institute Of Technology[1]

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Experiment: 1

Discrete-time signals

Scilab code Solution 1.1 Representation of discrete time signals

```
1
2 //scilab 5.5.2 ,OS: Ubuntu 14.04
3 //Generation of signals
4
5 //Unit Sample Sequence
6 clear ;clc ;close ;
7 L = 4;                                    //length= 2*L+1
8 n = -L:L;                                 // Time index
     vector
9 x = [zeros(1,L),1,zeros(1,L) ];
10 figure (1);
11 subplot(421),plot2d3(n,x),xtitle('Unit Sample
     sequence','n','x_1[n]');
12
13 //Unit step function
14 //clear ;clc ;close ;
15 n1=0:5
16 x1=[ones(1,6)];
17 subplot(422),plot2d3(n1,x1),xtitle('Unit Step
     sequence','n','x_2[n]')
18 //figure(1); plot2d3(n,x);
```

```
19 //xtitle('Discrete Unit Step Sequence','n','x[n]') ;
20
21 //Unit ramp function
22 //clear ;clc ;close ;
23 L = 4;                                  // Length of the
      sequence
24 n2= -L : L;
25 x2= [zeros(1,L ),0:L ];
26 ,subplot(423),plot2d3(n2,x2),xtitle('Unit Ramp
      sequence','n','x_2[n]')
27 //plot2d3(n,x);
28 //xtitle(' Discrete Unit Ramp Sequence','n','x[n]')
      ;
29
30 //Discrete time Exponential signal
31 //clear ;clc ;close ;
32 a =0.5;   //For decreasing a<1 and For increasing
      exponential a>1
33 n3 = 0:10;
34 x3 = (a).^n3 ;
35 subplot(424),plot2d3(n3,x3),xtitle('Exponential
      Sequence','n','x_3[n]')
36 //plot2d3(n,x);xtitle('Exponentially Decreasing
      Signal ','n','x[n]');
37
38
39
40 //Sinusoidal signal
41 //clc;clear;
42 fm=100;// Frequency 100 Hz or input('Enter the input
      signal frequency:');                //100
43 k=3;// Number of cycles:3 or input('Enter the number
      of Cycles of input signal:');    //3
44 A=1; // Unit amplitude or input('Enter the amplitude
      of input signal:');              //5
45 tm=0:1/(fm*fm):k/fm;
46 x4=A*cos(2*%pi*fm*tm);
47 subplot(425),plot2d3(tm,x4),xtitle('Sinusoidal
```

```
    Signal','n','x_4[n]')
//figure(1);plot2d3(tm,x);
//title('Graphical Representation of Sinusoidal
    Signal');
//xlabel('Time');ylabel('Amplitude');
//xgrid(1)

//Square wave
//clc;clear;
t=(0:0.1:4*%pi)';
x5=4*%pi*squarewave(t);
subplot(426),plot2d3(t,x5),xtitle('Square wave','n',
    'x_5[n]')


//Triangular wave
//clear;clc;
A=5// input('enter the amplitude:');              //5
K= 2// input('enter number of cycles:');          //2
x6 = [0:A  A-1:-1:1];
x7=x6;
for i=1:K-1
x7=[x7 x6];
end
n7=0:length(x7)-1;      // Index of the sequence
subplot(427),plot2d3(n7,x7);xtitle('Triangular wave'
    ,'time','amplitude');

//Sawtooth wave
//clc;clear;
A=5//input('enter the amplitude:');               //5
K=2;//input('enter number of cycles:');           //2
x8 = [0:A];
x9=x8;
for i=1:K-1
    x9=[x9 x8];
end
n9=0:length(x9)-1;
```

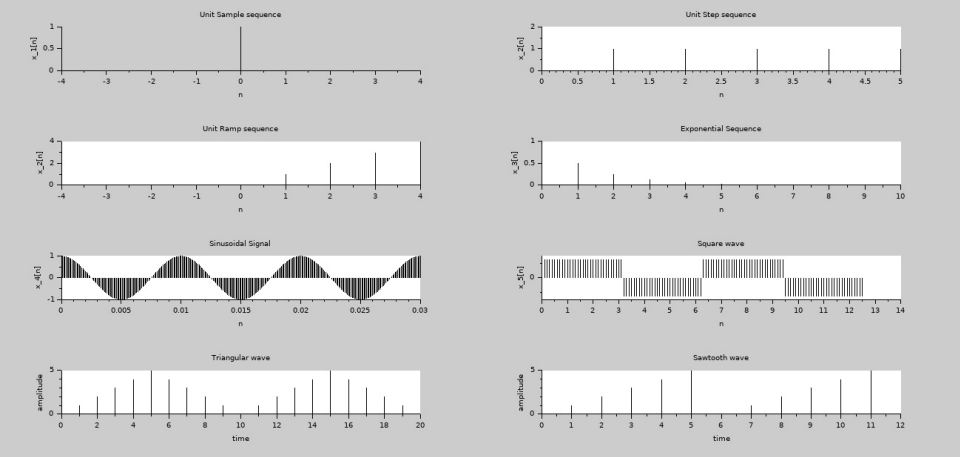


Figure 1.1: Representation of discrete time signals

```
82 subplot(428),plot2d3(n9,x9);xtitle('Sawtooth wave','
   time','amplitude');
83
84 // Complex valued signals
85 clc;clear;
86 n= [-10:1:10];
87 a=-0.1+0.3*%i;
88 x=exp(a*n);
89 figure(2);
90 subplot(221), plot2d3(n,real(x));xtitle('Complex
   valued signal','n','Real part');
91 subplot(223), plot2d3(n,imag(x));xtitle('Imaginary',
   'n');
92 subplot(222), plot2d3(n,abs(x));xtitle('Magnitude
   part','n');
93 theta=(180/%pi)*atan(imag(x),real(x));
94 subplot(224), plot2d3(n,theta);xtitle('Phase part','
   n');
```

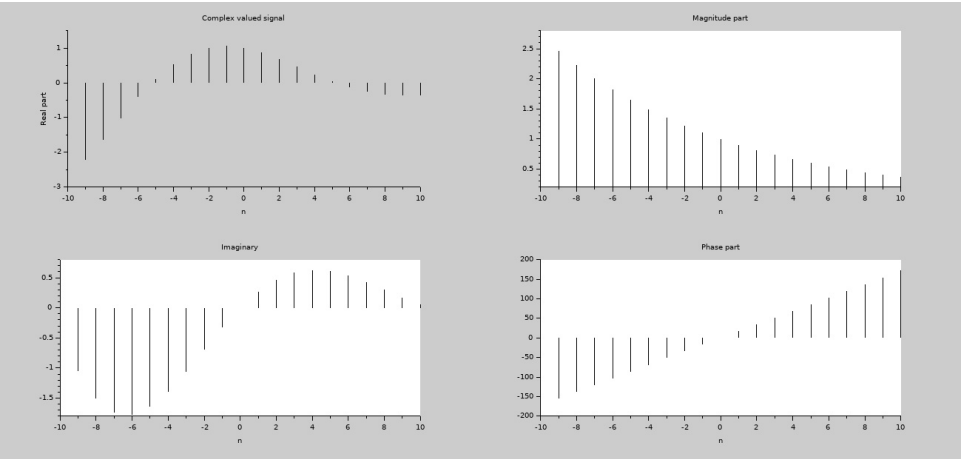


Figure 1.2: Representation of discrete time signals

Experiment: 2

Verification of Sampling Theorem

Scilab code Solution 2.1 To verify Sampling theorem in Time domain

```
1
2 //scilab 5.5.2 , OS: Ubuntu 14.04
3 //Sampling
4 clc;clear;
5 fm=100;//=input('Enter the input signal frequency:')
    ;       //100
6 k=4;//input('Enter the number of Cycles of input
    signal:'); //2
7 A=1;//input('Enter the amplitude of input signal:');
        //3
8 tm=0:1/(fm*fm):k/fm;
9 x=A*cos(2*%pi*fm*tm);
10 figure(1);
11 subplot(411),plot(tm,x);
12 title('ORIGINAL SIGNAL');xlabel('Time');ylabel('
    Amplitude');
13 xgrid(1)
14
15 //Sampling Rate(Nyquist Rate)=2*fm
```

```
fnyq=2*fm;

// UNDER SAMPLING
fs=(3/4)*fnyq;
n=0:1/fs:k/fm;
xn=A*cos(2*%pi*fm*n);
//figure(2);
subplot(412),plot2d3('gnn',n,xn);
plot(n,xn,'r');
title('Under Sampling');
xlabel('Time');
ylabel('Amplitude');
legend('Sampled Signal', 'Reconstructed Signal');
xgrid(1)
//NYQUIST SAMPLING
fs=fnyq;
n=0:1/fs:k/fm;
xn=A*cos(2*%pi*fm*n);
//figure(3);
subplot(413),
plot2d3('gnn',n,xn);
plot(n,xn,'r');
title('Nyquist Sampling');
xlabel('Time');
ylabel('Amplitude');
legend('Sampled Signal', 'Reconstructed Signal');
xgrid(1)
//OVER SAMPLING
fs=fnyq*10;
n=0:1/fs:k/fm;
xn=A*cos(2*%pi*fm*n);
//figure(4);
subplot(414)
plot2d3('gnn',n,xn);
plot(n,xn,'r');
title('Over Sampling');
xlabel('Time');
ylabel('Amplitude');
```

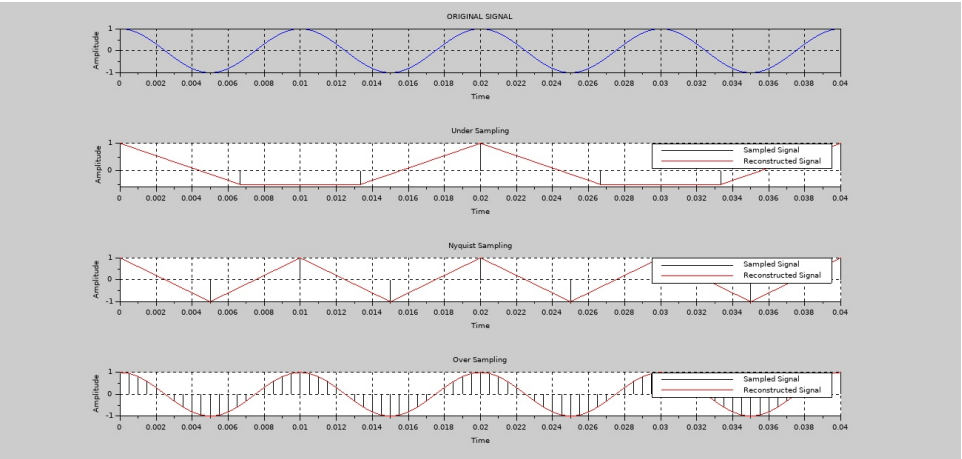


Figure 2.1: To verify Sampling theorem in Time domain

```
54 legend('Sampled Signal', 'Reconstructed Signal');
55 xgrid(1)
56 //Result
57 // Observing plots
```

Experiment: 3

Impulse response of the LTI system

Scilab code Solution 3.1 To determine the impulse response of a system given a difference equation

```
1 //scilab 5.5.2 , OS: Ubuntu 14.04
2 //To determine the impulse response of a LTI system,
     given the difference equation y[n]=b2 x(n-2)+b1
    x(n-1)+ b0x(n) +a(1)y(n-1)
3 clear all;clc;close;
4 b=input('Enter the coefficients of input x[n]= ');//
     [1]
5 a=input('Enter the coefficients of output y[n]= ');
    //[1 -1 0.9]
6 x=[1 zeros(1,9)];//generate impulse sequence of
    length 10
7 n=0:9;
8 h=filter(b,a,x);
9 figure; plot2d3(n,h),
10 xtitle('Impulse response h[n]','Time index n', 'h[n]
    ',' ');
11 //Example: y[n]-y[n-1]+0.9y[n-2]=x[n];a=[1] b=[1 -1
    0.9]
```

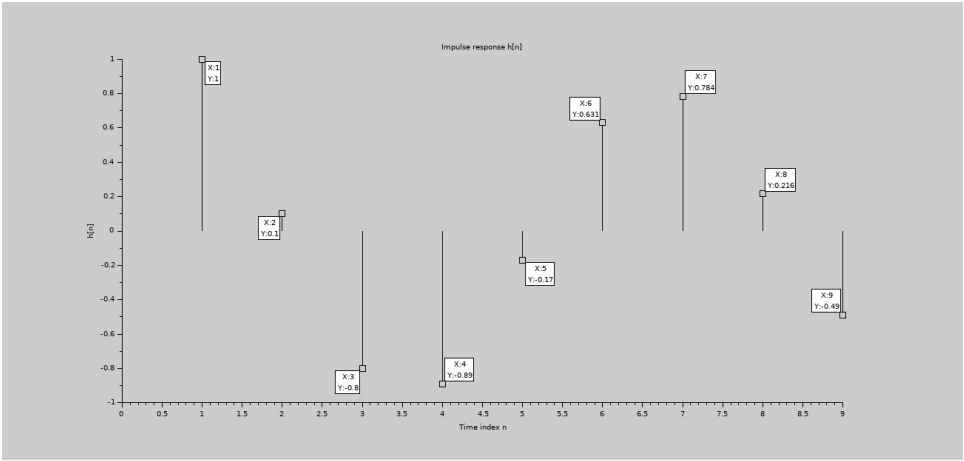


Figure 3.1: To determine the impulse response of a system given a difference equation

```
12 //n determines the length of the impulse response
       required
13 //Result:10 samples of h[n
       ]=[1,1,0.1,-0.8,-0.89,-0.17,0.631,0.784,
       0.2161,-0.4895]
```

Experiment: 4

Frequency response of the LTI system

Scilab code Solution 4.1 To plot the frequency response of a Digital system

```
1  //scilab 5.5.2 , OS: Ubuntu 14.04
2  //To determine the frequency response of a discrete-
       time system from its difference equation
3
4  //Design steps: Given a0 y[n] = -a2 y[n-2] - a1 y[n
       -1] + b0 x[n] + b1 x[n-1] + b2 x[n-2]
5  //1. System function H(z) = b0 + b1 z -1 + b2 z
       -2 / 1 + a1 z -1 + a2 z -2
6  //2. Put z= e (jw) to get the frequency response
7  //Design example: Plot the magnitude and phase
       response of the system represented by
8  //6y[n]+5y[n-1]+y[n-2]= 18x[n] + 8x[n-1]
9
10
11 clear;clc;
12 close;
13 b=input('Enter the coefficients of x[n]');//[1 -1]]
14 a=input('Enter the coefficients of y[n]');//[1 -0.5]
```

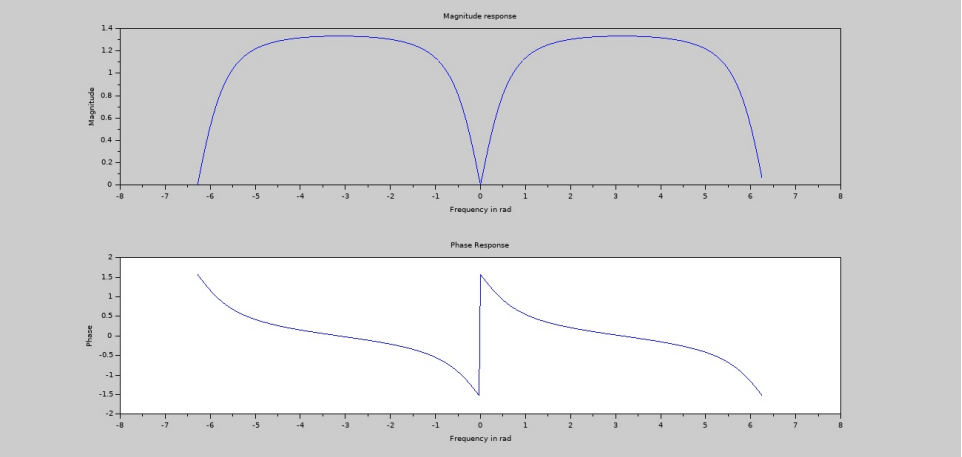


Figure 4.1: To plot the frequency response of a Digital system

```
15 //b=[18, 8];
16 //a=[6 5 1];
17 m= 0: length(b)-1; p=0:length(a)-1;
18 w=-2*%pi:%pi/100:2*%pi;//Plot over a interval of 4pi
       to observe periodicity
19 num = b* exp(-%i*m'*w);
20 den = a*exp(-%i*p'*w);
21 H= num./den;
22 magH = abs(H); angH= atan(imag(H),real(H));
23 figure;
24 subplot(211), plot( w, magH);
25 xtitle('Magnitude response','Frequency in rad','
       Magnitude');
26 subplot(212),plot(w, angH);
27 xtitle('Phase Response','Frequency in rad','Phase');
28 //Expected result
29 //H  =[5,3.5802695 - 1.3881467i,2.6 - i,2.253303 -
       0.4785341i,2.1666667,2.253303 + 0.4785341i,2.6 +
       i,3.5802695 + 1.3881467i,5]
```

Experiment: 5

Linear and Circular convolution

Scilab code Solution 5.1 To determine linear convolution

```
1 //scilab 5.5.2 , OS: Ubuntu 14.04
2 //  Linear Convolution  in time and frequency domain
3
4 clc ;clear all;close ;
5
6 x=[1 2 3 4];//input ('enter the input sequence
     values x(n)= ');     // [1 2 3 4]
7 h=[1 -1 0 -1];//input('enter the impulse sequence
     values h(n) = ');..// [1 -1 0 -1]
8
9 L1 = length(x);
10 L2 = length(h);
11
12 //Method 1 Using Direct Convolution Sum Formula
13 for i = 1: L1 +L2 -1
14  conv_sum = 0;
15  for j = 1: i
16  if ((( i - j +1) <= L2 ) &( j <= L1 ) )
17  conv_sum = conv_sum + x ( j ) * h (i -j +1) ;
18  end ;
19 y(i) = conv_sum ;
```

```
20 end ;
21 end ;
22
23 disp(y,' Convolution Sum using Direct Formula Method
     = ' )
24
25  //Method 2 Using In built Function
26 f = convol(x,h)
27 disp(f, ' Convolution Sum Result using Inbuilt
     Function = ')
28
29 //Method 3 Using frequency Domain multiplication
30 N = L1 +L2 -1;                                        //
     Linear convolution output length
31 x = [ x zeros(1 ,N - L1 ) ];
32 h = [ h zeros(1 ,N - L2 ) ];
33 f1 = fft(x)
34 f2 = fft(h)
35 f3 = f1.* f2 ;                                        //
     Multiplication in frequency domain
36 f4 = ifft(f3)
37 disp (f4 , 'Convolution Sum Result DFT and IDFT
     method = ')
38
39 //To plot input, impulse and output signals.
40 subplot (5,1,1) ;plot2d3(x);xtitle('Input signal x '
     ,'n','x[n]');
41 subplot(5,1,2) ;plot2d3(h);xtitle('Impulse signal h'
     ,'n','h[n]');
42 subplot(5,1,3) ;plot2d3(y);xtitle('Liner Convolution
      using formula','n','y1[n]');
43 subplot(5,1,4) ;plot2d3(f);xtitle('Linear
     Convolution using Inbuilt function','n','y2[n]');
44 subplot(5,1,5) ;plot2d3(f);xtitle('Linear
     Convolution using DFT method','n','y3[n]');
45
46 // Expected result
47 //1.    1.    1.    0.  - 6.  - 3.  - 4.
```

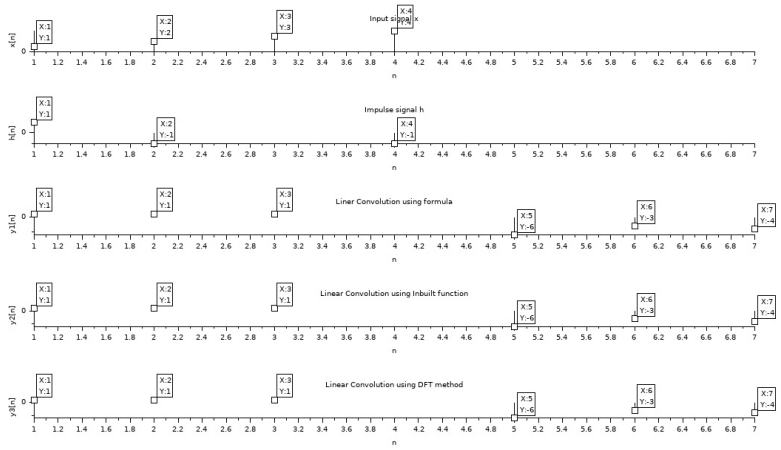


Figure 5.1: To determine linear convolution

Scilab code Solution 5.2 Circular convolution in time domain and using DFT relations

```
1
2 //scilab 5.5.2 , OS: Ubuntu 14.04
3 // Circular convolution of given discrete sequences
     in time domain (Matrix method)
4 clear;clc;
5 x1=input ('enter the first sequence values x1(n)= ')
     ;    // [1 2 3 4]
6 x2=input('enter the second sequence values x2(n) = '
     );   //[1 -1 0 -1]
7 L1 = length(x1);                          //length of
      first sequence
8 L2 = length(x2);                          //length of
      second sequence
9
```

```
10 if (L1 >L2)                                  //To make
     length of x1 and x2 are Equal
11     for i = L2+1:L1
12       x2(i) = 0;
13     end
14 elseif (L2>L1)
15     for i = L1+1:L2
16       x1(i) = 0;
17     end
18 end
19
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
31 //Calculation of circular convolution
32 for k = 2:N
33     for j =2:N
34        x2(j) = a(j-1);
35     end
36        x2(1) = a(N);
37        X(k,:)= x2;
38     for i = 1:N
39         a(i) = x2(i);
40         x3(k) = x3(k)+x1(i)*a(i);
41     end
42 end
43 disp(X,'Circular Convolution Matrix x2[n]=')
44 disp(x3,'Circular Convolution Result x3[n] = ')
45 // Expected result
```

```
46 //Circular Convolution Matrix x2[n]=
47
48 //     1.   - 1.     0.   - 1.
49 //   - 1.     1.   - 1.     0.
50 //     0.   - 1.     1.   - 1.
51 //   - 1.     0.   - 1.     1.
52
53  //Circular Convolution Result x3[n] =
54
55  // -5.   -2.   -3.     0.
56
57 // Circular Convolution in frequency domain (DFT-
     IDFT method)
58 clear all;clc;close;
59 x1=input ('enter the first sequence values x1(n)= ')
     ;   // [1 2 3 4]
60 x2=input('enter the second sequence values x2(n) = '
     ); //[1 -1 0 -1]
61 L=input ('enter the length of the sequence values L=
      '); //4
62
63 //Computing DFT
64 X1 = fft(x1,-1);                    //-1 for direct
     FFT
65 X2 = fft(x2,-1);
66 disp(X1,'DFT of x1[n] is X1(k)=')
67 disp(X2,'DFT of x2[n] is X2(k)=')
68
69 //Multiplication of 2 DFTs
70 X3 = X1.*X2;
71 disp(X3,'DFT of x3[n] is X3(k)=')
72 x3 =(fft(X3,1))                                  //
     Circular Convolution Result ,1 for IFFT
73 disp(x3,'Circular Convolution x3[n]=')
74 //// Expected result
75 //DFT of x1[n] is X1(k)=       10.  - 2. + 2.i  - 2.
       - 2. - 2.i
76
```

```
77  //DFT of x1[n] is X2(k)=     - 1.     1.     3.     1.
78
79  // DFT of x3[n] is X3(k)=     - 10.  - 2. + 2.i  -
      6.  - 2. - 2.i
80
81  //Circular Convolution x3[n]=        -5.     -2.
      -3.     0.
82
83  ////Performing Linear Convolution using Circular
      Convolution
84  clear;clc;
85  x=input ('enter the input sequence values x(n)= ');
        // [1 2 3 4]
86  h=input('enter the impulse sequence values h(n) = ')
      ;   //[1 -1 0 -1]
87  N1 = length(x);          //Length of input signal
88  N2 = length(h);      //Length of impulse response
89
90  N = N1+N2-1                          // Length of
      output response
91  disp(N,'Length of Output Response y(n)')
92
93  //Padding zeros to Make Length of 'h' and 'x' equal
      to length of output response 'y'
94
95  h1 = [h,zeros(1,N-N2)];
96  x1 = [x,zeros(1,N-N1)];
97
98  H = fft(h1,-1);
99  X = fft(x1,-1);
100 //Multiplication of 2 DFTs
101 Y = X.*H
102 y =(fft(Y,1))            //Linear Convolution Result
103
104 disp(X,'DFT of i/p X(k)=')
105 disp(H,'DFT of impulse sequence H(k)=')
106 disp(Y,'DFT of Linear Filter o/p Y(k)=')
107 disp(y,'Linear Convolution result y[n]=')
```

```
108
109 //Expected output
110 //Length of Output Response y(n)        7.
111
112  //DFT of i/p X(k)=        10.  - 2.0244587 -
       6.2239817i  ,    0.3460107 + 2.4791213i  ,
       0.1784479 - 2.4219847i  ,    0.1784479 +
       2.4219847i  ,    0.3460107 - 2.4791213i  , -
       2.0244587 + 6.2239817i  ,
113  //DFT of impulse sequence H(k)=     - 1.
       1.2774791 + 1.2157152i  ,,   0.5990311 +
       0.1930964i  ,    2.1234898 + 1.4088117i  ,
       2.1234898 - 1.4088117i  ,    0.5990311 -
       0.1930964i  ,    1.2774791 - 1.2157152i  ,
114   //DFT of Linear Filter o/p Y(k)=     - 10.
        4.9803857 - 10.412171i  , - 0.2714383 +
        1.5518843i  ,    3.7910526 - 4.8916602i  ,
        3.7910526 + 4.8916602i  , - 0.2714383 -
        1.5518843i  ,    4.9803857 + 10.412171i  ,
115  //Linear Convolution result y[n]=        1.     1.
         1.     0.     -6.    -3.     -4.
```

Experiment: 6

Spectral analysis using DFT

Scilab code Solution 6.1 To demonstrate spectral leakage

```
1
2 //scilab 5.5.2 , OS: Ubuntu 14.04
3 // Spectral Leakage
4 //Check the result for the following cases
5 //case(1): fm=10;fs=125;m=1;m=number of cycles
6 //case(2): fm=10;fs=125;m=2;
7 //case(3): fm=200;fs=10000;m=2.5;
8 //case(4): fm=75;fs=250;m=3;
9
10 clc;clear;close;
11 //fm=input('Enter the frequency of the input signal
     ');//message frequency in Hz
12 //fs=input('Enter the sampling frequency');//
     sampling frequency in Hz
13 //m=input('Enter the number of cycles of the input
     signal');// Number of cycles
14 //Case2:No spectral leakage
15 fm=10;fs=125;m=2;//Oversampling and integer number
     of cycles
16 t=0.0001:1/fs:m/fm;
17 x=3*cos(2*%pi*fm*t);          //signal
```

```
N=(m*fs/fm);                          //should be non-
   integer to obtain spectral leakage
for k=1:N
 X1(k)=0;
 for n=1:length(x)
  X1(k)=X1(k)+x(n).*exp((-%i).*2.*%pi.*(n-1).*(k-1)
     ./N);
 end
 end
k=0:N-1
f=k*fs/N;        //frequency axis in Hz
figure(1),subplot(221),plot2d3(t,x),xlabel('time'),
   ylabel('x(n)'),title('No leakage: m=2, f=10 and
   Fs=125 Hz'),subplot(223),plot2d3(f,abs(X1)),
   xlabel('freq in Hz'),ylabel('Mag');//Case 3:
   Spectral leakage
  fm=10;fs=125;m=2.5;//Oversampling and integer
     number of cycles
t=0.0001:1/fs:m/fm;
x=3*cos(2*%pi*fm*t);          //signal
N=(m*fs/fm);                          //should be non-
   integer to obtain spectral leakage
for k=1:N
 X1(k)=0;
 for n=1:length(x)
  X1(k)=X1(k)+x(n).*exp((-%i).*2.*%pi.*(n-1).*(k-1)
     ./N);
 end
 end
k=0:N-1
f=k*fs/N;        //frequency axis in Hz
figure(1),subplot(222),plot2d3(t,x),xlabel('time'),
   ylabel('x(n)'),title('Spectral leakage: m=2.5, f
   =10 and Fs=125 Hz'),subplot(224),plot2d3(f,abs(X1
   )),xlabel('freq in Hz'),ylabel('Mag')
```

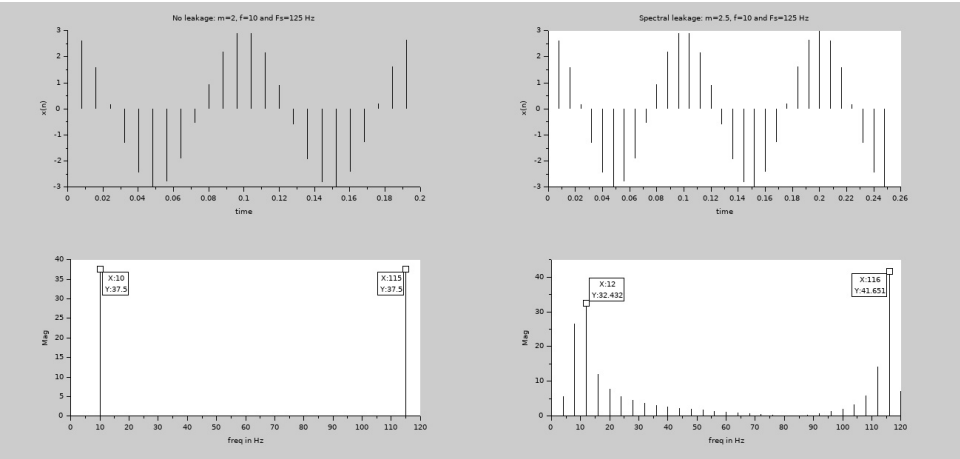


Figure 6.1: To demonstrate spectral leakage

Scilab code Solution 6.2 To demonstrate effects of zeropadding and zero insertion on the spectrum

```
1  //scilab 5.5.2 , OS: Ubuntu 14.04
2  //Effect of zero padding and interpolation
3  // Effect of Zero padding
4  clc;clear;close;
5  x= input ('enter the input sequence values x(n)= ');
       //[1 2 3 4]
6  k= input ('enter the number of zeros to be padded= '
     );  //1020 (For 1024 point DFT))
7  N=length(x);
8  x_pad=[x zeros(1,k)];
9  N1=length(x_pad);
10 f=0:N-1;
11 f1=0:N1-1;
12 X=abs(fft(x));
```

```
13 X_pad= abs(fft(x_pad));
14 figure(1);
15 subplot(221),plot2d3(x),title(' Original sequence'),
      subplot(223),plot2d3 (f,X), title('Spectrum of
      Original sequence');
16 subplot(222),plot2d3(x_pad),title('Zero-padded
      sequence'),subplot(224), plot2d3 (f1,X_pad),title
      ('Spectrum of Zero-padded sequence')
17 //// Effect of inserting zeros in between samples (
      Interpolation)
18 x= input ('enter the input sequence values x(n)= ');
         //[1 2 3 4]
19 k= input ('enter the number of zeros to be inserted=
      ');//2 (Vary and observe effect of zero
      interpolation)
20 x_mod=[];
21 N=length(x);
22 //
23 for  i= 1: N
24 x_mod=[x_mod, x(i), zeros(1,k)];
25 end
26 N1=length(x_mod);
27 f=0:N-1;
28 f1=0:N1-1;
29 X=abs(fft(x));
30 X_mod= abs(fft(x_mod));
31 figure(2);subplot(221),plot2d3(x),title(' Original
      sequence'),subplot(223),plot2d3 (f,X), title('
      Spectrum of Original sequence');
32 subplot(222),plot2d3(x_mod),title('Zero-interpolated
       sequence'),subplot(224), plot2d3 (f1,X_mod),
      title('Spectrum of Zero-inserted sequence')
```

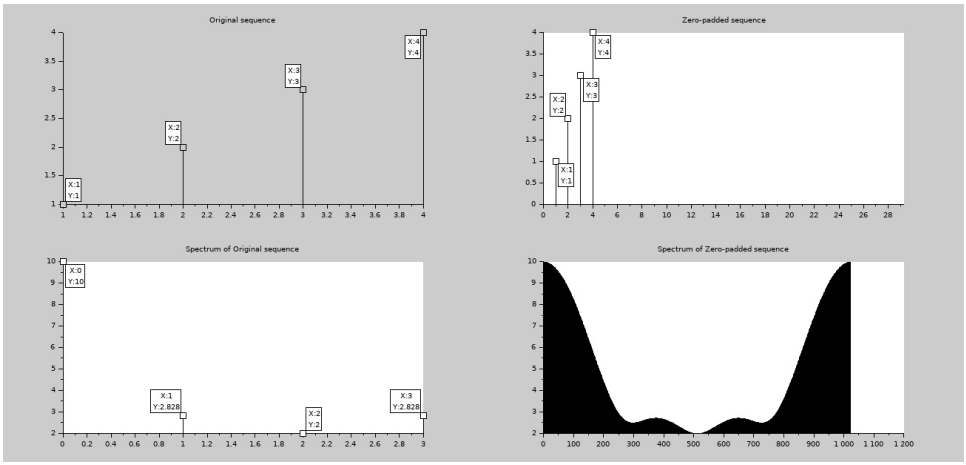


Figure 6.2: To demonstrate effects of zeropadding and zero insertion on the spectrum

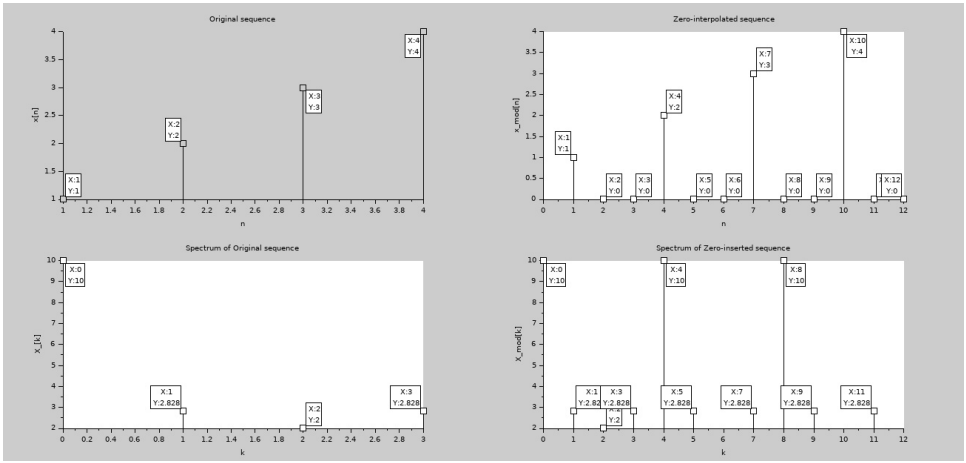


Figure 6.3: To demonstrate effects of zeropadding and zero insertion on the spectrum

Experiment: 7

FIR filter design

Scilab code Solution 7.1 Design of FIR filter using Windowing method

```
1
2 //scilab 5.5.2 , OS: Ubuntu 14.04
3 //Design of FIR filters using windowing
4 // Design a digital FIR low pass filter with
     following specifications.
5 //a) Pass band cut-off frequency      :wp=
     ----------radians
6 //b) Pass band ripple              :rp=----------dB
7 //c) Stop band cut-off frequency      :ws=
     ----------radians
8 //d) Stop band attenuation                :rs=
     ----------dB
9 //Choose an appropriate window function and
     determine impulse response and provide a plot of
     frequency response of the designed filter.
10
11 //Design example:
12 //Design a digital FIR low pass filter with
     following specifications.
13 //a) Pass band cut-off frequency      :0.3    rad
14 //b) Pass band ripple                        :0.25 dB
```

```
15 //c) Stop band cut-off frequency      :0.45    rad
16 //d) Stop band attenuation                 : 50 dB
17 clc;
18 clear;
19 close;
20 wp=input('enter the pass band edge in rad');
21 ws=input('enter the stop band edge in rad');
22 rs=input('enter the stop band ripple in dB');
23 freq_points=1024;
24 freq_divs=(freq_points/2)-1;
25 k=4; //Hamming window (decided based on stop band
      attenuation)
26 trw=ws-wp;
27 N=(k*2*%pi/trw);
28 N=ceil(N);
29 remainder=N-fix(N./2).*2
30 if remainder==0
31     N=N+1;
32 end
33
34 wc=wp;
35 aph=(N-1)/2;
36 for n=0:N-1
37     if n==aph
38         hdn_minusalph(n+1)=wc/%pi;
39
40  else
41      hdn_minusalph(n+1)= sin(wc.*(n-aph))./(%pi.*(n-
         aph));
42
43 end
44 end
45 n=0:N-1;
46 wndw=window('hm',N);
47
48 hn=hdn_minusalph.*wndw';
49 figure(1);subplot(311);plot2d3(n,wndw);xlabel('n');
    ylabel('wndw');title('Hamming Window function');
```

```
50 subplot(312); plot2d3(n,hdn_minusalph);xlabel('n');
      ylabel('hdn_minusalph');title('Impulse response
      of IIR filter');
51 subplot(313); plot2d3(n,hn);xlabel('n');ylabel('hn')
      ;title('Impulse response of FIR filter');
52 //omega=0:%pi/freq_divs:%pi;
53 h=[hn' zeros(1,freq_points-length(hn))];;//For a
      1024 point DFT
54 H=fft(h);
55 H_mag=20*log10(abs(H));
56 H_ang=atan(imag(H),real(H));
57 H_phase=unwrap(H_ang);
58 w=(0:freq_divs)./(freq_points);
59 w1=w*%pi;
60 figure(2);subplot(211),plot2d(w1,H_mag(1:512));
61 xtitle('Magnitude response','w (rad)','Magnitude(dB)
      ');
62 subplot(212),plot2d(w1,H_phase(1:512));
63 xtitle('Phase Response','w (rad)','Phase (rad)');
64
65
66
67 //Problems:
68
69 //1. Design a digital FIR low pass filter with
      following specifications.
70 //a) Pass band cut-off frequency      :0.4    rad
71 //b) Pass band ripple                        :0.25 dB
72 //c) Stop band cut-off frequency      :0.6    rad
73 //d) Stop band attenuation                : 44 dB
74
75 //2. Design a digital FIR low pass filter with
      following specifications.
76 //a) Pass band cut-off frequency      :0.25    rad
77 //b) Pass band ripple                        :0.25 dB
78 //c) Stop band cut-off frequency      :0.3    rad
79 //d) Stop band attenuation                : 50 dB
```

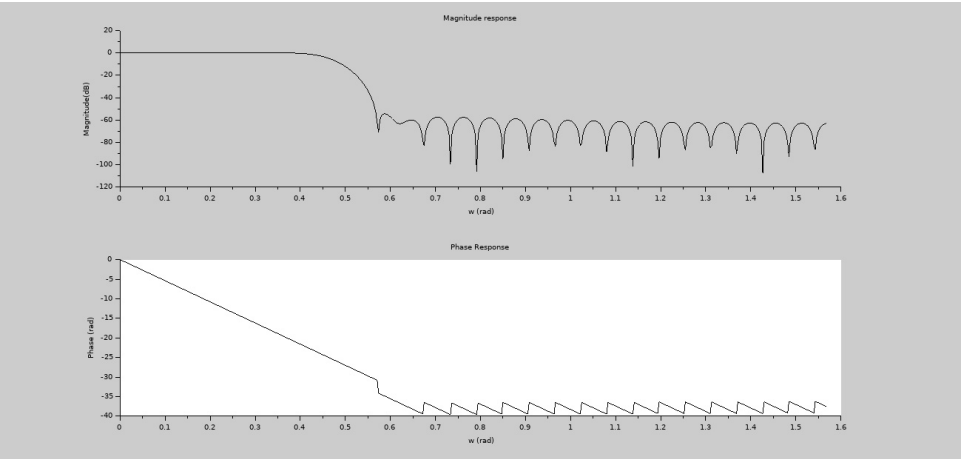


Figure 7.1: Design of FIR filter using Windowing method

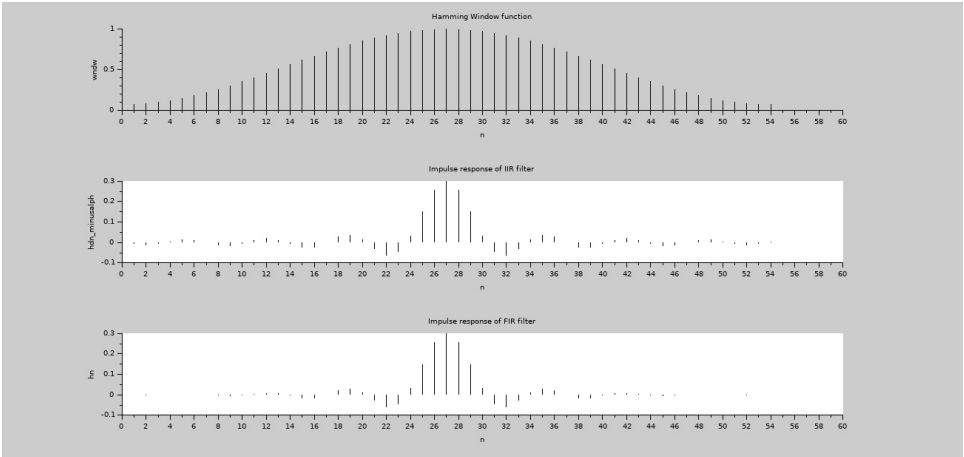


Figure 7.2: Design of FIR filter using Windowing method

Experiment: 8

Design of Hilbert transformer using FIR filter

Scilab code Solution 8.1 Design of a digital Hilbert Transformer using FIR filter

```
1 //scilab 5.5.2 , OS: Ubuntu 14.04
2 //Design a differentiator using a Hamming window of
    length N=21. Plot the time and frequency domain
    responses.
3 //Design a length-25 digital Hilbert transformer
    using a Hann window.
4
5 //Design of Hilbert transformer
6 //The ideal frequency response of a linear phase
    Hilbert transformer is given by
7 //Hd(e jw) =    - j e(-j w ) ,   0 < w < pi
8 //              j e(-j w ) ,   -pi < w < 0
9
10 //The ideal impulse response is given by
11
12 //hd(n-   )=    2/pi   (sin2 pi( n       )/2) / ( n
        ),      n
13 //                  0, n=
```

```
14
15
16 //Scilab Program
17 //Inputs: Window length and type of window
18 clc;clear;close;
19
20 N = 41;//input("enter the window length");  //55
21 freq_points=1024;
22 windowfn =window('hm',N);// Hamming window ()Window
      type can be changed here)
23 m = 0:N-1;
24 aph = (N-1)/2;
25 for n=0:N-1
26     if n==aph
27         hd(n+1)=0;
28
29  else
30      hd(n+1)=(2/%pi)*((sin((%pi/2)*(n-aph))).^2)./(n-
           aph));
31
32 end
33 end
34 n=0:N-1;
35  hn = hd.*windowfn';
36
37  omega=-%pi:2*%pi/(freq_points-1):%pi;
38
39  z=%z;
40 den1=real(z^(N-1));
41 num=0;
42 for n=0:N-1
43     num=num+(hn(n+1).*z^(N-n-1));
44 end
45 num1=real(num);
46 Hz=num1./den1;
47 w=exp(%i*omega);
48 rep=freq(Hz("num"),Hz("den"),w);
49 magH=abs(rep);
```

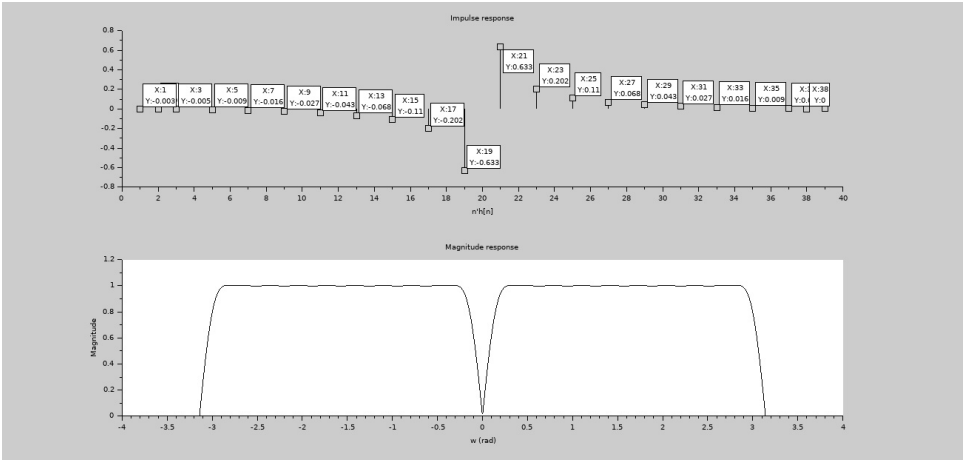


Figure 8.1: Design of a digital Hilbert Transformer using FIR filter

```
50 figure;subplot(211),plot2d3(m,hn),xtitle('Impulse
    response','n''h[n]')
51 , subplot(212),plot2d(omega,magH);
52 xtitle('Magnitude response','w (rad)','Magnitude');
53 //Expected result
54 //Magnitude response graph
```

Experiment: 9

Design of digital differentiator using FIR filter

Scilab code Solution 9.1 Design of Digital Differentiator using a FIR filter

```
1
2 11//scilab 5.5.2 , OS: Ubuntu 14.04
3 //Design a differentiator using a Hamming window of
    length N=21. Plot the time and frequency domain
    response
4 //Inputs: Window length and Type of window
5 //The frequency response of a linear-phase ideal
    differentiator is given by
6 //Hd(e jw) =   j   , 0<    <
7 //                -jw,  -    <    < 0
8 //The ideal impulse response of a digital
    differentiator shifted by    with linear phase is
     given by
9 //hd( n     ) =   cos    (n       ) / ( n -    ), n

10 //                  0,   n =
11
12 //Scilab Program:
```

```
clc;clear;close;
N = 41;// input("enter the window length");  //55
freq_points=1024;
windowfn =window('hm',N);//Hamming wuindow (Try with
    different windows)
m = 0:N-1;
aph = (N-1)/2;
for n=0:N-1
    if n==aph
        hd(n+1)=0;

 else
     hd(n+1)= cos(%pi*(n-aph))./(n-aph);

end
end
n=0:N-1;
 hn = hd.*windowfn';

omega=-%pi:2*%pi/(freq_points-1):%pi;

z=%z;
den1=real(z^(N-1));
num=0;
for n=0:N-1
    num=num+(hn(n+1).*z^(N-n-1));
end
num1=real(num);
Hz=num1./den1;
w=exp(%i*omega);
rep=freq(Hz("num"),Hz("den"),w);
magH=abs(rep);
figure;subplot(211),plot2d3(m,hn),xtitle('Impulse
   response','n','h[n]'),subplot(212),plot2d(omega,
   magH);
xtitle('Magnitude response','w (rad)','Magnitude');
//Expected result
//Magnitude response graph
```

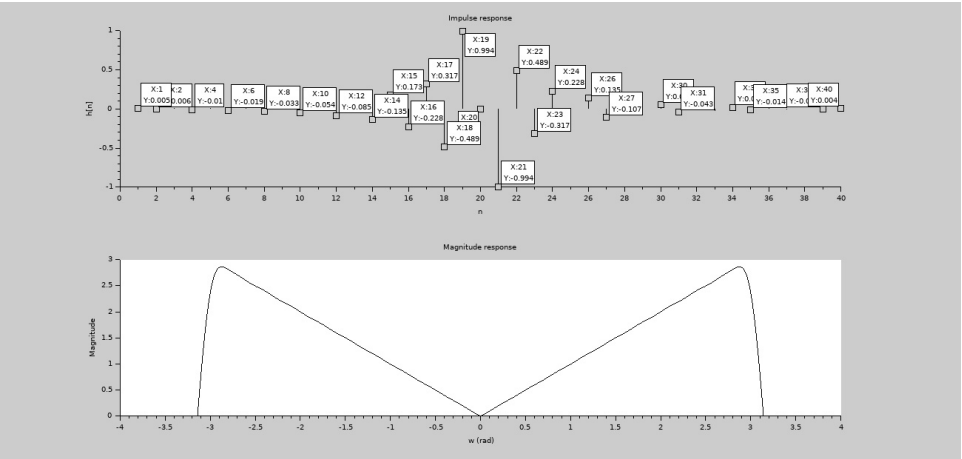


Figure 9.1: Design of Digital Differentiator using a FIR filter

Experiment: 10

Design of IIR filter

Scilab code Solution 10.1 Design of digital Butterworth lowpass filter

```
1 //scilab 5.5.2 , OS: Ubuntu 14.04
2 //Program To Design the Digtial Butterworth  IIR
     Filter
3 //Design a digital IIR low pass filter with
     following specifications.
4 //a) Pass band cut-off frequency     :1000 Hz
5 //b) Pass band ripple                      :-1 dB
6 //c) Stop band cut-off frequency     :3000 Hz
7 //d) Stop band attenuation               : -15 dB
8 //Sampling frequency: 15000 Hz
9
10 clear all;clc;close;
11 f1=1000;//input('Enter the pass band edge(Hz)= ');
12 f2=3000;//input('Enter the stop band edge(Hz)= ');
13 k1=-1;//input('Enter the pass band attenuation(dB)=
     ');
14 k2=-15;//input('Enter the stop band attenuation(dB)=
     ');
15 fs=10000;//input('Enter the sampling rate(Hz)= ');
16
17 //Digital filter specifications(rad)
```

```
w1=2*%pi*f1*1/fs;
w2=2*%pi*f2*1/fs;

//Pre warping
o1=2*fs*tan(w1/2)
o2=2*fs*tan(w2/2)

//Design of analog filter
n=log10(((10.^(-k1/10))-1)/((10.^(-k2/10))-1)).^(2*
   log10(o1/o2));
n=round(n);
wn= o2./((10.^(-k2/10)-1).^(1/(2*n)));

//[h,poles,zeros,gain]=analpf(n,'butt',[0 0],wn)hb.
   dt = 'c';
//[fr,hr]=repfreq(hb,fmin,fmax)

h=buttmag(n,wn,1:2*%pi*fs);
mag=20*log10(h)';


//Converting analog to digital filter
hz=iir(n,'lp','butt',0.25,[])
//g*poly(z,'z')/poly(p,'z')

[hzm,fr]=frmag(hz,256);
magz=20*log10(hzm)';

subplot(2,1,1),plot2d((1:2*%pi*fs)',mag),xtitle('
   Analog IIR filter: lowpass','Analog frequency in
   rads/sec', 'dB',' ');subplot(2,1,2),plot2d(fr,
   magz);xtitle('Digital IIR filter: lowpass 0 < fr
   < 0.5','frequency', 'dB',' ');

//note: Use zoom/axis commands to verify the design.
```

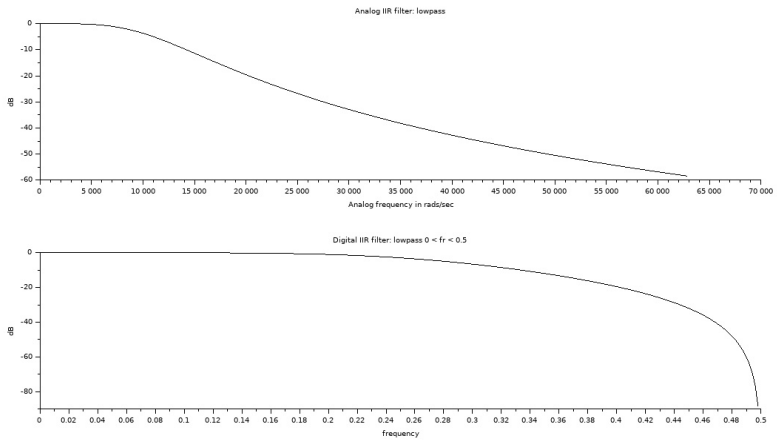


Figure 10.1: Design of digital Butterworth lowpass filter

Scilab code Solution 10.2 Design of Digital Chebyshev lowpass filter

```
1 //scilab 5.5.2 , OS: Ubuntu 14.04
2 //Program To Design the Digtial Chebyshev IIR Filter
3 ////Design example:
4 //Design a digital IIR low pass filter with
     following specifications.
5 //a) Pass band cut-off frequency      :1000 Hz
6 //b) Pass band ripple                        :-1 dB
7 //c) Stop band cut-off frequency      :3000 rad
8 //d) Stop band attenuation                 : -15 dB
9 //Sampling frequency: 15000 Hz
10
11 clear all;clc;close;
12 f1=1000;//input('Enter the pass band edge(Hz)= ');
13 f2=3000;//input('Enter the stop band edge(Hz)= ');
14 rp=-1;//input('Enter the pass band ripple(dB)= ');
```

```
15 rs=-15;//input('Enter the stop band attenuation(dB)=
       ');
16 fs=10000;//input('Enter the sampling rate(Hz)= ');
17 //Digital filter specifications(rad)
18 w1=2*%pi*f1*1/fs
19 w2=2*%pi*f2*1/fs
20 //Pre warping
21 o1=2*fs*tan(w1/2)
22 o2=2*fs*tan(w2/2)
23 or=o2/o1;//Stop-band edge of normalized lowpass
      filter
24 A2 =10.^(-rs/10);
25 A=sqrt(A2);
26 epsilon2 = (10.^(-rp/10)-1);
27 epsilon=sqrt(epsilon2)
28 g=((A2-1).^0.5./epsilon)
29
30 N = (acosh(g))/(acosh(or))
31 N = ceil(N)
32 oc=o1;
33 //[pols,gn] = zpch1(N,epsilon,o1)
34 //Hs = poly(gn,'s','coeff')/real(poly(pols,'s'))
35 h=cheb1mag(N,oc,epsilon,1:2*%pi*fs);
36 mag=20*log10(h)';
37 //plot2d((1:1000)',mag,[2],"011"," ",[ymax,ymin,fmax
      ,fmin])
38 //gain=20*log10(abs(h_s)); %Verify the specification
       [k1,k2] at prewarped frequencies
39 //subplot(211);
40 //plot(omega,gain);
41 //xlabel(   frequency    in rad/sec );
42 //Converting analog to digital filter
43 fc=w1/(2*%pi);
44 delta1=(1-(1./A2));
45 //1-ripple in passband
46 hz=iir(N,'lp','cheb1',[fc],[delta1 0]);
47 //for cheb1 filters 1-delta(1)<ripple<1 in passband
48 //g*poly(z,'z')/poly(p,'z')
```

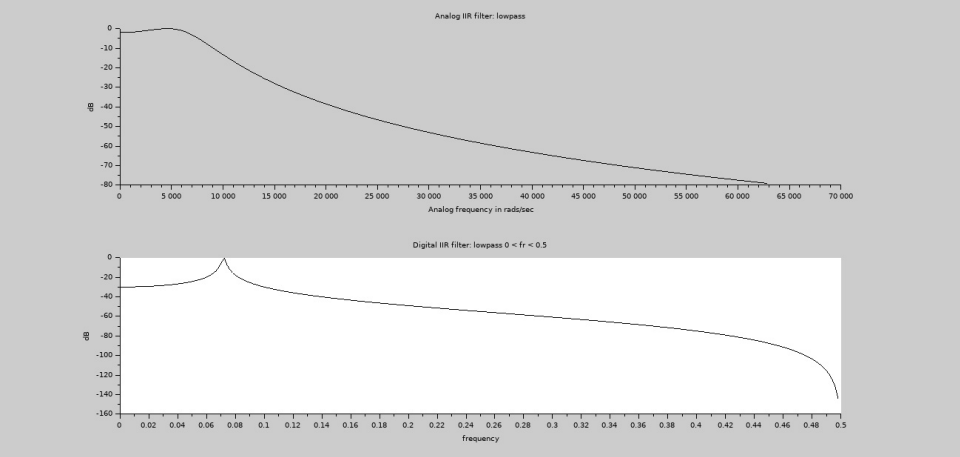


Figure 10.2: Design of Digital Chebyshev lowpass filter

```
49 [hzm,fr]=frmag(hz,256);
50 magz=20*log10(hzm)';
51 figure(1);subplot(2,1,1),plot2d((1:2*%pi*fs)',mag),
     xtitle('Analog IIR filter: lowpass','Analog
     frequency in rads/sec', 'dB',' ');subplot(2,1,2),
     plot2d(fr,magz);xtitle('Digital IIR filter:
     lowpass 0 < fr < 0.5','frequency', 'dB',' ');
```

Experiment: 11

Application of IIR filter

Scilab code Solution 11.1 To design a digital IIR Butterworth filter to suppress noise

```
1  //scilab 5.5.2 , OS: Ubuntu 14.04
2  // This program will suppress noise at f=4000 Hz
       using Butterworth prototype
3  //pass band edge=f1=1500Hz
4  //stop band edge=f2=2000 Hz
5  //sampling rate =Fs=10000 Hz = 1/Ts
6  //passband attenuation = -1db
7  //stop attenuation = -3 db
8
9  clear all;clc;close;
10 f1=input('Enter the pass band edge(Hz)= ');
11 f2=input('Enter the stop band edge(Hz)= ');
12 k1=input('Enter the pass band attenuation(dB)= ');
13 k2=input('Enter the stop band attenuation(dB)= ');
14 fs=input('Enter the sampling rate(Hz)= ');
15
16 signal_fo=1000;
17 noise_fo=4000;
18
19 //Digital filter specifications(rad)
```

```
20 w1=2*%pi*f1*1/fs;
21 w2=2*%pi*f2*1/fs;
22
23 //Pre warping
24 o1=2*fs*tan(w1/2)
25 o2=2*fs*tan(w2/2)
26
27 //Design of analog filter
28 n=log10(((10.^(-k1/10))-1)/((10.^(-k2/10))-1)).^(2*
      log10(o1/o2));
29 n=round(n);
30 wn= o2./((10.^(-k2/10)-1).^(1/(2*n)));
31
32 //[h,poles,zeros,gain]=analpf(n,'butt',[0 0],wn)hb.
      dt = 'c';
33 //[fr,hr]=repfreq(hb,fmin,fmax)
34
35 h=buttmag(n,wn,1:2*%pi*fs);
36 mag=20*log10(h)';
37 //plot2d((1:2*%pi*fs)',mag)
38 //xtitle('Analog IIR filter: lowpass','Analog
      frequency in rads/sec', 'dB',' ');
39
40 //Converting analog to digital filter
41 hz=iir(n,'lp','butt',0.25,[])
42 //g*poly(z,'z')/poly(p,'z')
43
44 [hzm,fr]=frmag(hz,256);
45 magz=20*log10(hzm)';
46 fr1=fr*fs;
47 //figure;plot2d(fr1',magz');xtitle('Digital IIR
      filter: lowpass 0 < fr < 0.5','frequency', 'dB','
      ');
48
49 //////note: Use zoom/axis commands to verify the
      design.
50 //These coefficients are to be read from variable hz
      (line 41, output of iir function)
```

```
51 num=[0.2928 0.5858 0.2928];
52 den=[1 0 0.1716];// In negative powers of z
53
54 //Signal generation (sine wave of frequency 1000 Hz)
       of length 1 second
55 t=0:1/fs:10/signal_fo;//10 cycles of input
56 original_signal=sin(2*%pi*signal_fo*t);
57
58 //Noise generation (sine wave of frequency 4000 Hz)
      of length 1 second
59 t=0:1/fs:10/signal_fo;
60 noise=sin(2*%pi*noise_fo*t);
61
62 noisy_signal=original_signal+noise;
63
64 filter_output=filter(num,den,noisy_signal);
65
66 //Plot original, noisy and filtered outputs
67
68 figure;subplot(3,1,1), plot2d(t,original_signal),
      xtitle('Original_signal','t','x(t)'),
69 subplot(3,1,2), plot2d(t,noisy_signal),xtitle('
      Noisy_signal','t','n(t)'),
70 subplot(3,1,3), plot2d(t,filter_output),xtitle('
      Filtered_signal','t','y(t)');
71 l1=length(original_signal);
72 l2=length(noisy_signal);
73 N=512;
74 x=[original_signal zeros(1,N-l1)];//To make it of
      length 512
75 n=[noisy_signal zeros(1,N-l1)];
76 y=[filter_output zeros(1,N-l1)];
77 X=fft(x);
78 N=fft(n);
79 Y=fft(y);
80 f=(0:511)*fs;
81 figure;
82 subplot(3,1,1), plot2d(f,abs(X)),xtitle('
```

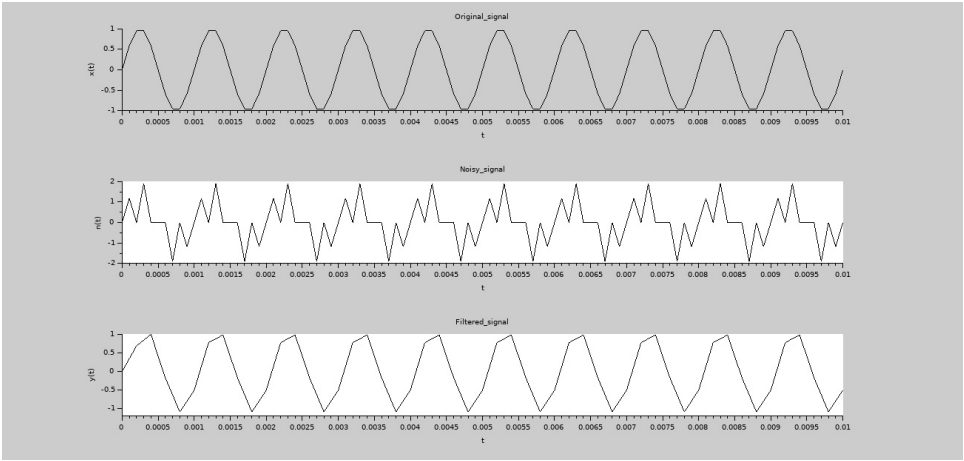


Figure 11.1: To design a digital IIR Butterworth filter to suppress noise

```
    Original_signal','F','X(f)'),
83  subplot(3,1,2), plot2d(f,abs(N)),xtitle('
    Noisy_signal','F','N(f)'),
84  subplot(3,1,3), plot2d(f,abs(Y)),xtitle('
    Filtered_signal','F','Y(f)');
```

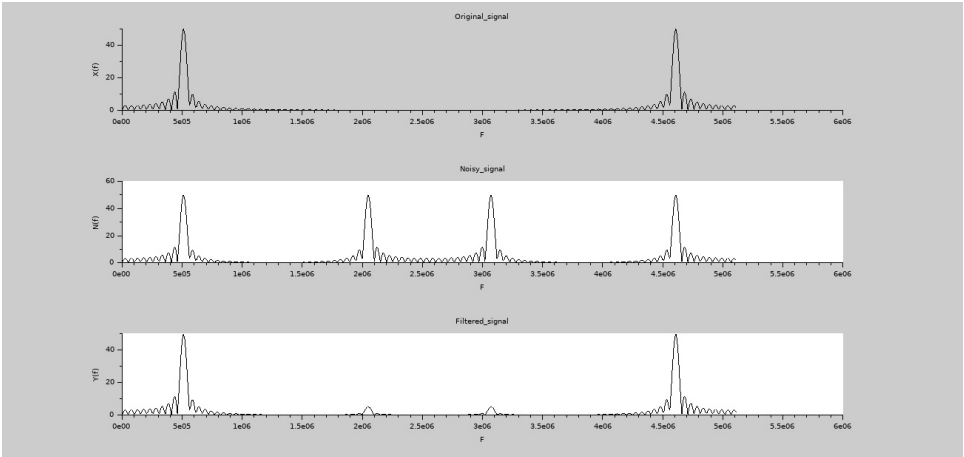


Figure 11.2: To design a digital IIR Butterworth filter to suppress noise

Experiment: 12

Design of Notch filter

Scilab code Solution 12.1 Suppression of noise at a given frequency using Notch filter

```
1
2 //scilab 5.5.2 , OS: Ubuntu 14.04
3 //Program To Design a simple notch filter and verify
4 // Design a simple notch filter to stop a
     disturbance with frequency F_0=3.5 kHz and a
     sampling frequency F_s=8 kHz.
5 //Also, verify the notch filter operation by adding
     a sinewave of F_0 Hz to a speech signal, filter
     and verify.
6
7 //Scilab Program:
8 clc;clear;close;
9 f=3500;//input("Enter the frequency in Hz");
                  //3500
10 fs=8000;//input("Enter the sampling rate");
                   //8000
11 r=0.98;//input("Enter the radius of the pole in the
     z-plane"); //0.98
12 w=2*%pi*f/fs;
13 z1=exp(%i*w);
```

```
14 z2=exp(-%i*w);
15 p1=r*exp(%i*w);
16 p2=r*exp(-%i*w);
17 z=%z;
18 num1=(real((z-z1)*(z-z2)))
19 den1=(real(((z-p1)*(z-p2))))
20 Hz=num1./den1
21 //figure(1);plzr(Hz);zgrid()
22 [h1 fr]=frmag(Hz,512)
23 figure(1);plot2d(fr*fs,h1);xtitle('Magnitude
      response','frequency in Hz','Mag');
24
25 //Noise generation
26
27 original_signal=wavread('home/hyrkswamy/kswamy/
      Coursework/SAP/wav/mask.wav');
28 t=0:1/fs:(length(original_signal)-1)/fs;
29 noise=sin(2*%pi*f*t);
30 noisy_signal=original_signal+noise;
31
32 filter_output=filter(num1,den1,noisy_signal);
33
34 //Play back the original, noisy and filtered outputs
35 playsnd(original_signal,fs);
36 pause;
37 playsnd(noisy_signal,fs);
38 pause;
39 playsnd(filter_output,fs);
```

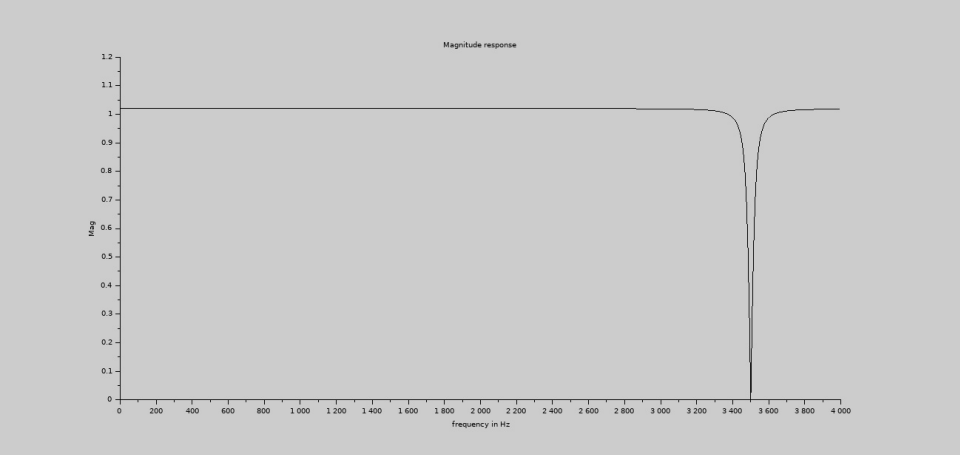


Figure 12.1: Suppression of noise at a given frequency using Notch filter

Experiment: 13

Design of Resonator

Scilab code Solution 13.1 Design of a Notch filter to filter noise at a given frequency

```
1
2 //scilab 5.5.2 , OS: Ubuntu 14.04
3 //Design a digital resonator that resonates at 1000
     Hz. Assume Fs=8000 Hz.
4 // Calculate the pole location
5 //w=2*pi*f/fs;
6 //Complex conjugate pair of poles at w=pi/4 and -pi
     /4
7 //Assume radius=0.98 (near to unit circle but inside
      the unit circle)
8
9 //Scilab Program:
10 clc;
11 clear;
12 close;
13 f=1000;//input("Enter the frequency in Hz");
     //1000
14 fs=8000;//input("Enter the sampling rate");
     //8000
15 r=0.98;//input("Enter the radius of the pole in the
```

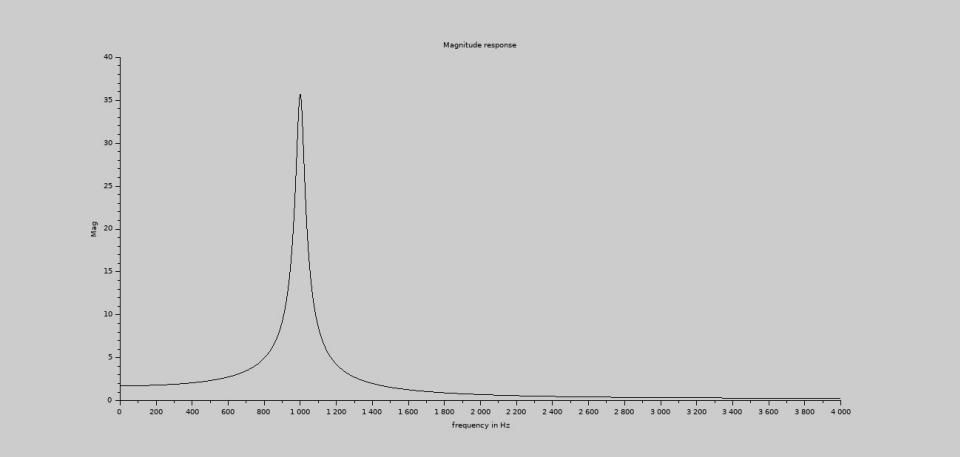


Figure 13.1: Design of a Notch filter to filter noise at a given frequency

```
    z-plane");    //0.98
w=2*%pi*f/fs;
pole1=r*exp(%i*w);
pole2=r*exp(-%i*w);

z=%z;

num1=real(z^(2));
den1=real(z^(2)-1.3859293*z+0.9604);
Hz=num1./den1;
//figure;
//plzr(Hz);
[h1 fr]=frmag(Hz,1024);
figure;
plot2d(fr*fs,h1);
xtitle('Magnitude response','frequency in Hz','Mag')
    ;
```