

Scilab Manual for
Digital Signal Processing Lab
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Experiment: 1

Design and Testing of a Digital Butterworth Low pass filter with cutoff of 5 KHz to filter an input (.wav) file

Scilab code Solution 1.1 DSP Lab Migration

```
1
2 // Exp 1: Design Low Pass Filter as per the given
  // specification and test the working by taking an
  // input sound signal.
3 // Enter cutoff freq in Hz fc = 5
4
5 // Version : Scilab 5.2.2
6 // Operating Syatem : Ubuntu 16.04 LTS
7
8 clc;
  //clear console
9 clear;
10 xdel(winsid());
11 fc=input('Enter cutoff freq in Hz fc =')
  //Cutoff frequency
```

```

12 fs=11025;
13 n=11;
    //Filter order
14 Fp=2*fc/fs;
15 [Hz]=iir(n, 'lp', 'butt', [Fp/2,0], [0,0])
16 [p,z,g]=iir(n, 'lp', 'butt', [Fp/2,0], [0,0])
    //Filter design
17 [Hw,w]=frmag(Hz,256);
18 figure(1)
19 subplot(2,1,1)
20 plot(2*w,abs(Hw));
21 xlabel('Normalized Digital frequency w->')
22 ylabel('magnitude');
23 title('Magnitude response of IIR filter')
24 xgrid(1)
25 subplot(2,1,2)
26 plot(2*w*fs,abs(Hw));
27 xlabel('Analog Frequency in Hz f ——>')
28 ylabel('Magnitude |H(w)|=')
29 title('Magnitude Response of IIR LPF')
30 xgrid(1)
31
32 [y,Fs]=wavread("meow.wav")
    //Reading input sound signal
33 figure(2)
34 subplot(2,1,1)
35 plot(y)
36 title('Input signal waveform');
37 xlabel('Frequency——>');
38 ylabel('Magnitude——>');
39 playsnd(y)
40
41 outlo=filter(z,abs(p),y); //
    Passing acquired signal through desired filter
42 subplot(2,1,2)
43 plot(outlo)
44 title('Output signal waveform after filtering')
45 xlabel('Frequency——>');

```

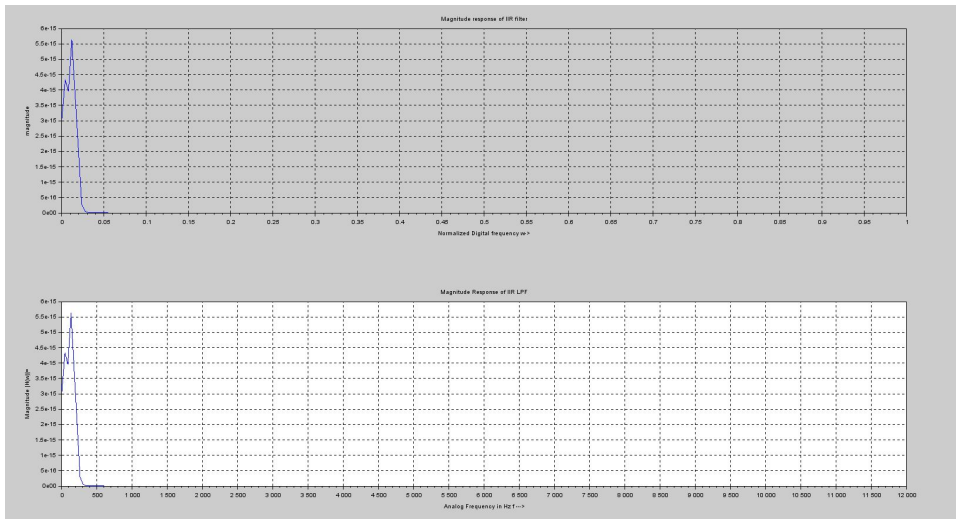


Figure 1.1: DSP Lab Migration

46 `ylabel('Magnitude—>');`

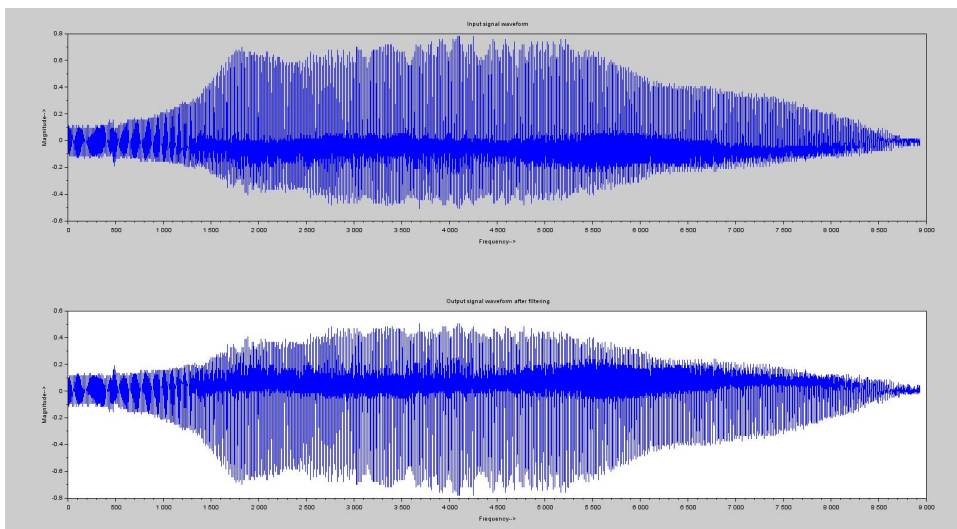


Figure 1.2: DSP Lab Migration

Experiment: 2

Design a digital Butterworth low pass filter to band limit a sine wave up to 4 KHz, by considering input as 'tone'

Scilab code Solution 2.2 DSP Lab Migration

```
1 //Program to design a Butterworth Low pass filter to
   Band limit a sine wave up to 4kHz . Taking the
   input as a tone .
2 // Version : Scilab 5.2.2
3 // Operating Syatem : Ubuntu 16.04 LTS
4 clc;
5 clear;
6 xdel(winsid());
7 Fc =4000; // Cut-
   off frequency
8 Fs =44100; //
   Sampling frequency
9 N =8 ; // Order
10 Fp = 2*Fc/Fs; //Pass
   band edge frequency
```

```

11 [Hz]=iir(N, 'lp', 'butt', [Fp/2,0], [0,0])
12 [p,z,g]=iir(N, 'lp', 'butt', [Fp/2,0], [0,0]) //
    digital IIR Butterworth Filter
13 [Hw,w] = frmag(Hz ,256);
14
15 // Plotting the filter design
16 figure(1)
17 plot(2*w, abs(Hw));
18 xlabel("Digital Frequency Normalized (w)")
19 ylabel("Magnitude")
20 title("Magnitude Response of Butterworth filter ")
21 xgrid(1)
22
23 [y,Fs]=wavread("tone1k.wav") //
    Reading the y
24 figure(2)
25 subplot(2,1,1)
26 plot(y)
27 title('Input signal waveform before filtering');
28 xlabel('Frequency');
29 ylabel('Magnitude');
30 playsnd(y)
31 L=length(y)
32
33 outlow=filter(z, abs(p), y);
34 subplot(2,1,2)
35 plot(outlow)
36 title('Output signal waveform after filtering')
37 xlabel('Frequency');
38 ylabel('Magnitude');
39
40 playsnd(outlow)
41
42 Da = fft(y,-1);
43 Pyy = (1/L)*(abs(Da).^2); //Peridogram Estimate

```

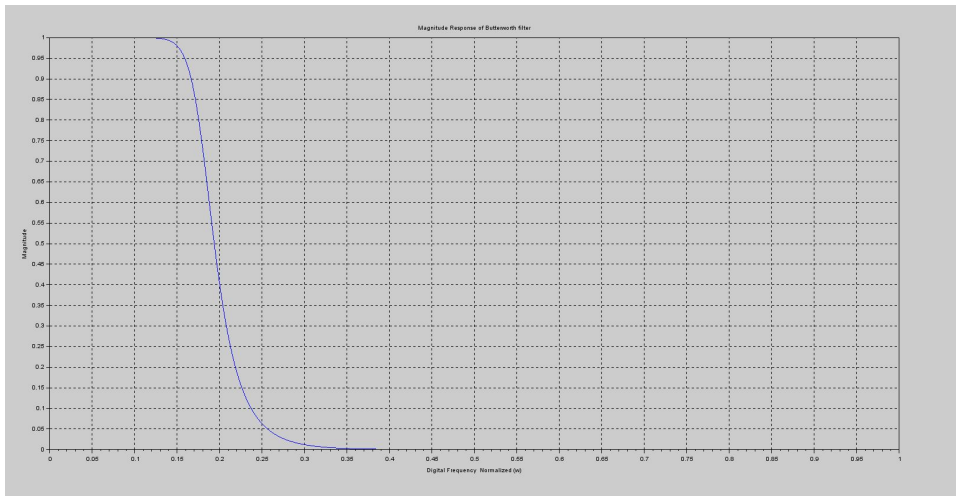


Figure 2.1: DSP Lab Migration

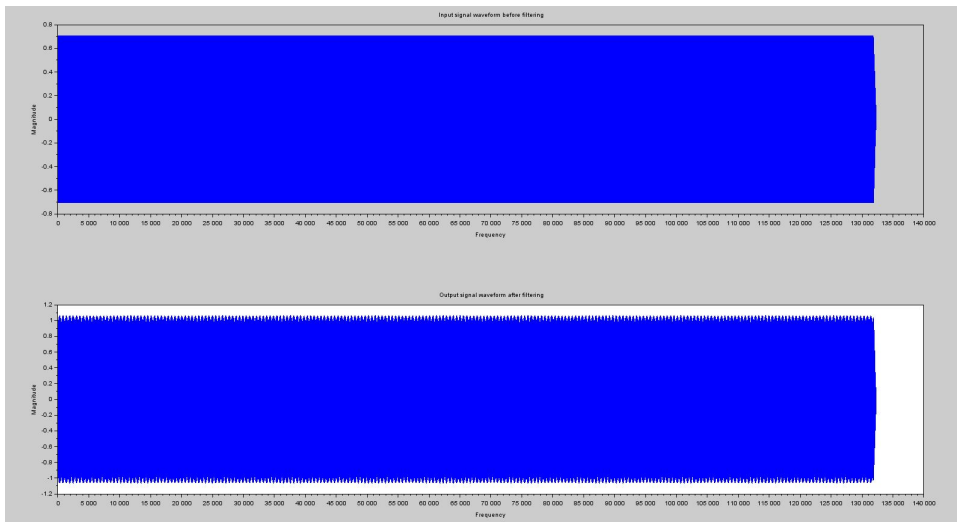


Figure 2.2: DSP Lab Migration

Experiment: 3

Illustrate the working of a digital low pass filter by taking audio data as input and allow frequency up to 2 KHz

Scilab code Solution 3.3 DSP Lab Migration

```
1 // Design a filter using Butterworth polynomial for
  the following specifications:
2 //1.    Order : 7
3 //2.    Cut-off frequency : 2.5 kHz
4
5 // Version : Scilab 5.2.2
6 // Operating System : Ubuntu 16.04 LTS
7
8 clc;
9 clear;
10 xdel(winsid());
11 Fc =2500;

    // Cut-off frequency
12 Fs =44100;
```

```

    // Sampling frequency
13 N =7 ;

    // Order of the filter
14 Fp = 2*Fc/Fs;

    //Pass band edge frequency
15 [Hz]=iir(N, 'lp ', 'butt ', [Fp/2,0], [0,0])
16 [p,z,g]=iir(N, 'lp ', 'butt ', [Fp/2,0], [0,0])
17 [Hw,w] = frmag(Hz ,256);
18
19 figure(1)
20 plot(2*w, abs(Hw));
21 xlabel(" Digital Frequency   Normalized (w)")
22 ylabel(" Magnitude")
23 title(" Magnitude Response of Butterworth filter ")
24 xgrid(1)
25
26 [y,Fs]=wavread("tone1k.wav")

    // Reading the .wav file
27
28 outlow=filter(z, abs(p), y);
29
30 psd1=pspect(100,200, 're', y)
31 figure(2)
32 subplot(2,1,1)
33 plot(psd1)

    //Plotting power spectral density of input
34 title('Input signal power spectral density')
35 xlabel('Frequency');
36 ylabel('Magnitude');
37
38 psd2=pspect(100,200, 're', outlow)
39 subplot(2,1,2)
40 plot(psd2)

```

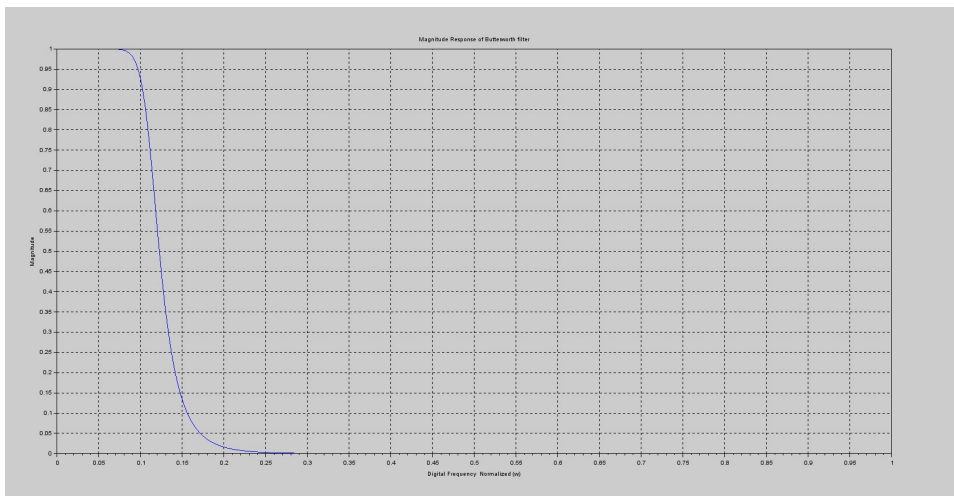


Figure 3.1: DSP Lab Migration

```
41 //Plotting power spectral density of output  
41 title('Filtered signal power spectral density')  
42 xlabel('Frequency');  
43 ylabel('Magnitude');  
44  
45 playsnd(outlow)
```

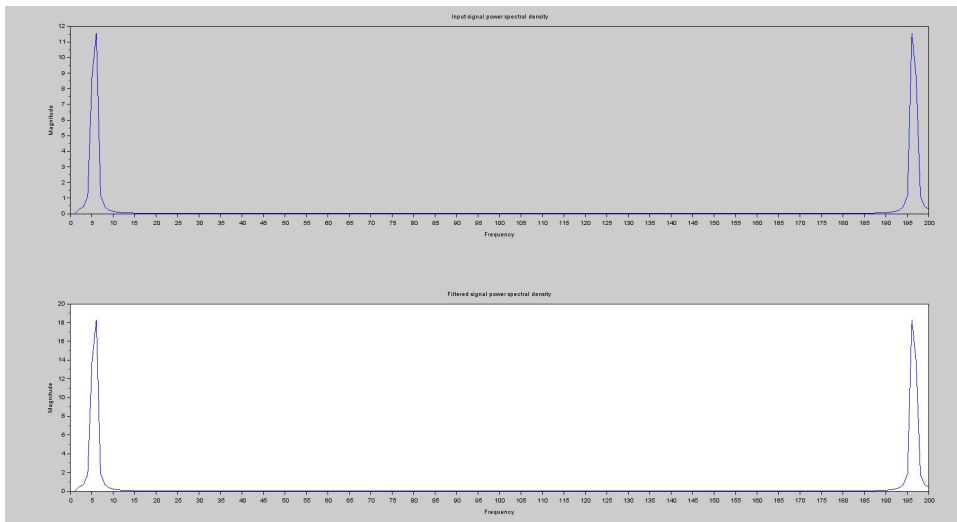


Figure 3.2: DSP Lab Migration

Experiment: 4

Design and test the working of a High Pass FIR filter using Hamming window by taking a high frequency signal as input

Scilab code Solution 4.4 DSP Lab Migration

```
1 // Version : Scilab 5.2.2
2 // Operating Syatem : Ubuntu 16.04 LTS
3
4 clc;          clear;          xdel(winsid());
5 fc=20000;    fs=44100;    M=63;
//
// Filter order
6 wc=2*fc/fs;
7 [wft,wfm,fr]=wfirm('hp',M,[wc/2,0], 'hm', [0,0]);
//FIR Filter
8 figure(1)
```

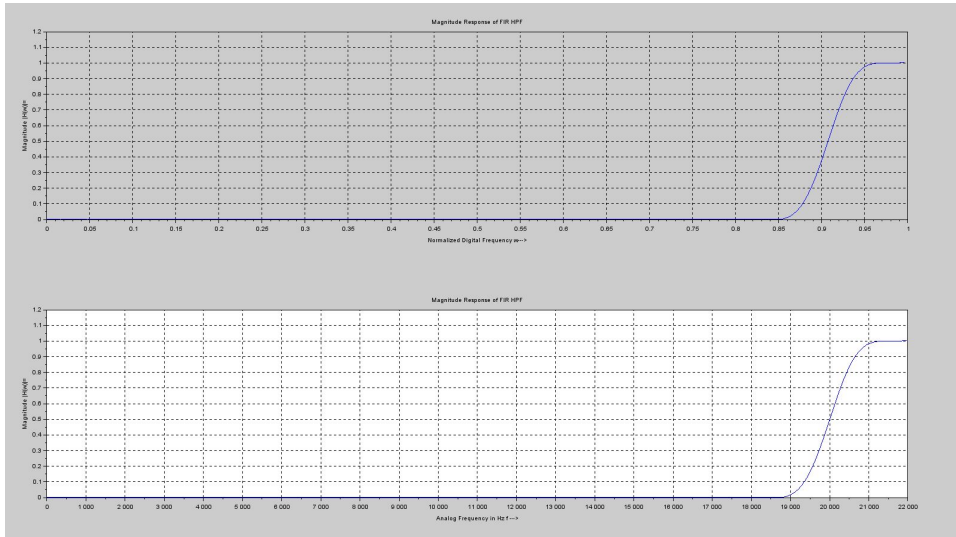


Figure 4.1: DSP Lab Migration

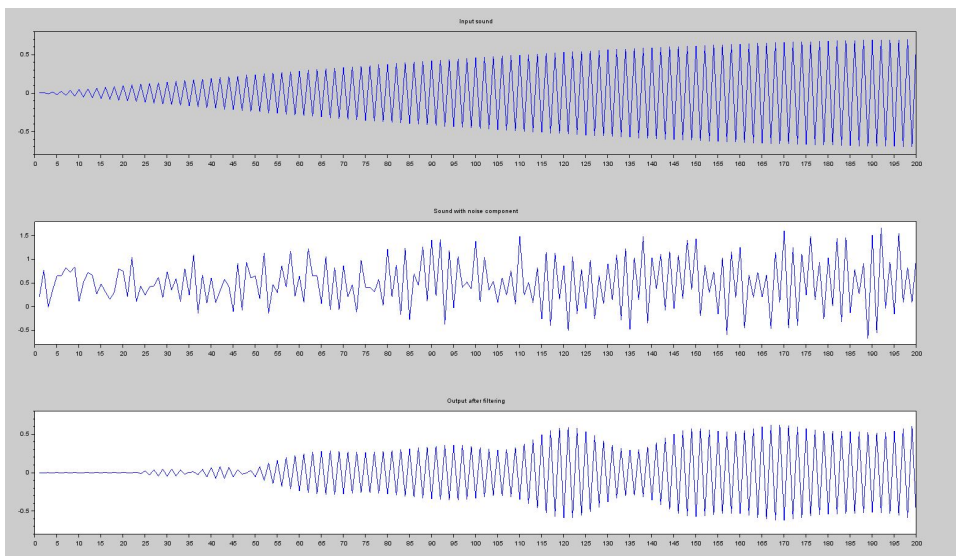


Figure 4.2: DSP Lab Migration

```

 9 subplot(2,1,1);          plot(2*fr,wfm);
10 xlabel('Normalized Digital Frequency  $w \longrightarrow$ ')
11 ylabel('Magnitude  $|H(w)| =$  ')
12 title('Magnitude Response of FIR HPF');      xgrid
   (1)
13 subplot(2,1,2);          plot(fr*fs,wfm);
14 xlabel('Analog Frequency in Hz  $f \longrightarrow$ ')
15 ylabel('Magnitude  $|H(w)| =$  ')
16 title('Magnitude Response of FIR HPF');      xgrid
   (1)
17
18 [d,Fs]=wavread("22000.wav")
19 playsnd(d,Fs)           //single tone high frequency
   sound wave
20 L = length(d);          a=1+nextpow2(L);      N=2*(2^a
   );
21 noise = rand(1,L);      data = d+noise;
   playsnd(data);
22 outhi = filter(wft,1,data); playsnd(outhi);
23
24 figure(2)
25 subplot(3,1,1);          plot(d(1:200));
26 title('Input sound');
27 subplot(3,1,2);          plot(data(1:200));
28 title('Sound with noise component');
29 subplot(3,1,3);          plot(outhi(1:200));
30 title('Output after filtering');

```

Experiment: 5

Design and test the working of a Butterworth band pass filter by giving a time domain input signal

Scilab code Solution 5.5 DSP lab migration

```
1 // Version : Scilab 5.2.2
2 // Operating Syatem : Ubuntu 16.04 LTS
3 //Assume fp=1000 hz and fs=5000 hz
4 //Use the formula to convert the pass band ripple
   and To convert stop band attenuation in dB(A in
   dB)=-20*log10(Ap or As)
5 // assume kp=1.93 dB and ks=13.97 dB
6
7
8 clc;
9 clear;
10 xdel(winsid());
11 fp= input( " Enter the pass band edge (Hz) = ");
```

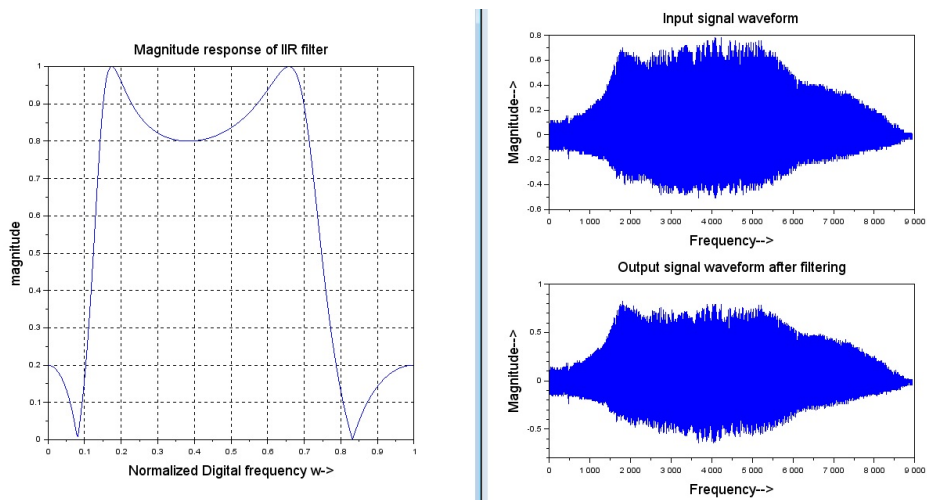


Figure 5.1: DSP lab migration

```

12 fs= input( "Enter the stop band edge (Hz) = ");
13 kp= 1.93          //assume the pass band ripple(
    dB)
14 ks= 13.97        //assume the stop band
    attenuation(dB)
15 Fsf=44000;       //sampling frequency
16 //Converting to digital frequency
17 Fp1=2*3.14*fp/Fsf;
18 Fs1=2*3.14*fs/Fsf;
19
20 // Digital filter specifications ( rad/samples )
21 N = log10(sqrt((10^(0.1*ks)-1)/(10^(0.1*kp)-1)))/
    log10(Fs1/Fp1); //Order of the filter
22 N = ceil(N);     //rounded to nearest integer
23 disp(N," IIR Filter order N=");
24
25 oc = 0.5*((Fp1*Fsf)/((10^(0.1*kp)-1)^(1/(2*N))) + (
    Fs1*Fsf)/((10^(0.1*ks)-1)^(1/(2*N))) ); //Cut
    off Frequency
26 disp(oc, " Cutoff Frequency in rad/ seconds OC=")
27 [Hz]=iir(N, 'bp', 'ellip', [Fp1/2, Fs1/2], [0.2,0.200])

```

```

        //the sum of last matrix [0.2,0.200] must be
        less than 1
28 [p,z,g]=iir(N, 'bp', 'ellip', [Fp1/2,Fs1
        /2],[0.2,0.200])
29 [Hw,w]=frmag(Hz,256);
30 figure(1)
31 plot(2*w,abs(Hw));
32 xlabel('Normalized Digital frequency w->')
33 ylabel('magnitude');
34 title('Magnitude response of IIR filter')
35 xgrid(1)
36 [y,Fs]=wavread("H:\DSP SCILAB\Final\meow.wav") //
        Reading input .wav signal file path of the .wav
        file must be changed
37 figure(2)
38 subplot(2,1,1)
39 plot(y)
40 title('Input signal waveform');
41 xlabel('Frequency—>');
42 ylabel('Magnitude—>');
43 playsnd(y)
44 outlo=filter(abs(z),abs(p),y); //Passing
        acquired signal through desired filter
45 subplot(2,1,2)
46 plot(outlo)
47 title('Output signal waveform after filtering')
48 xlabel('Frequency—>');
49 ylabel('Magnitude—>');
50
51 N=length(y); //
        Power spectral density of the Input signal
52 Y=fft(y,-1);
53 Pxx=(1/N)*(abs(Y).^2); //Peridogram Estimate
54 figure(3)
55 plot2d3('gnn',[1:N],Pxx)
56 title('Input signal power spectral density')
57 xlabel('Analog Frequency in Hz f —>')
58 ylabel('Magnitude |H(w)|=')

```

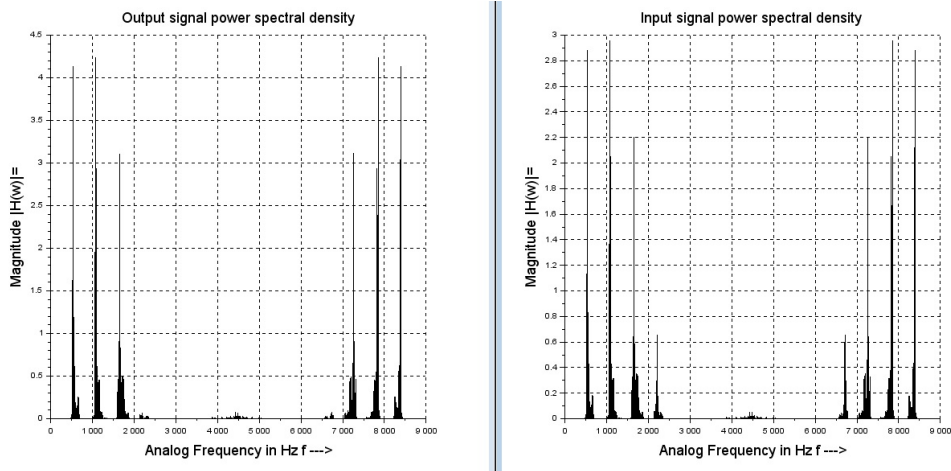


Figure 5.2: DSP lab migration

```

59 xgrid(1)
60 playsnd(outlo)
61
62 N=length(outlo)                                     ////
    Power spectral density of the Ouput signal
63 OL = fft(outlo,-1);
64 Fxx = (1/N)*(abs(OL).^2); //Peridogram Estimate
65 figure(4)
66 plot2d3('gnn',Fxx)
67 title('Output signal power spectral density')
68 xlabel('Analog Frequency in Hz f ——>')
69 ylabel('Magnitude |H(w)|=')
70 xgrid(1)

```

Experiment: 6

Design and test band pass,
Butterworth filter for given
specifications which includes
Edge frequencies, Ripple and
Attenuation

Scilab code Solution 6.6 DSP lab migration

```
1 // Version : Scilab 5.2.2
2 // Operating System : Ubuntu 16.04 LTS
3
4 clc;
5 clear;
6 xdel(winsid());
7 fp= input( " Enter the pass band edge (Hz) = ");
8 fs= input( "Enter the stop band edge (Hz) = ");
9 kp= input( "Enter the pass band attenuation (dB) =")
```

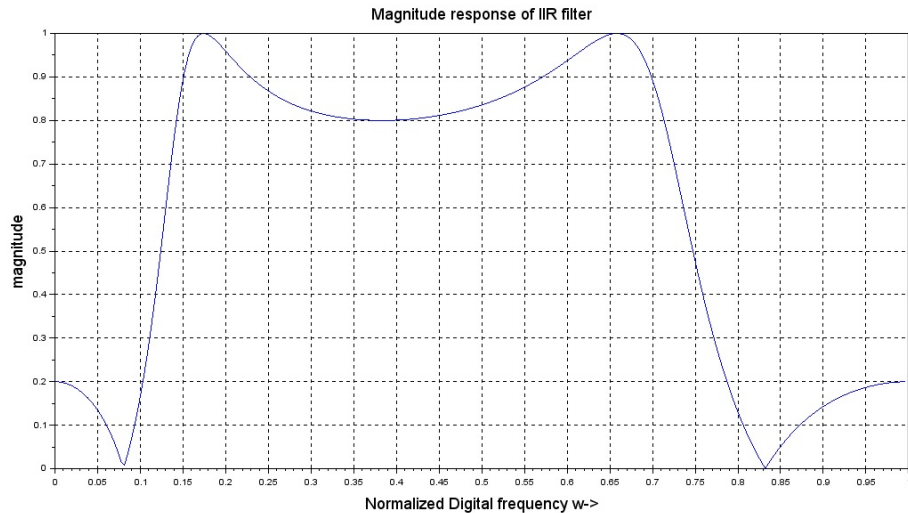


Figure 6.1: DSP lab migration

```

;
10 ks= input( "Enter the stop band attenuation (dB) = "
) ;
11 //kp= 1.93           //assume the pass band ripple
(dB)
12 //ks= 13.97         //assume the stop band
attenuation(dB)
13 Fsf=44000;         //sampling frequency
14 //Converting to digital frequency
15 Fp1=2*3.14*fp/Fsf;
16 Fs1=2*3.14*fs/Fsf;
17
18 // Digital filter specifications ( rad/samples )
19 N = log10(sqrt((10^(0.1*ks)-1)/(10^(0.1*kp)-1)))/
log10(Fs1/Fp1); //Order of the filter
20 N = ceil(N);       //rounded to nearest integer
21 disp(N," IIR Filter order N =");
22
23 oc = 0.5*((Fp1*Fsf)/((10^(0.1*kp)-1)^(1/(2*N)))) + (
Fs1*Fsf)/((10^(0.1*ks)-1)^(1/(2*N))) ); //Cut

```

```
Enter the pass band edge (Hz) =
1000

Enter the stop band edge (Hz) =
5000

Enter the pass band attenuation (dB) =
2

Enter the stop band attenuation (dB) =
14

IIR Filter order N =

2.

Cutoff Frequency in rad/ seconds OC =
10675.066
```

Figure 6.2: DSP lab migration

```
    off Frequency
24 disp(oc, "Cutoff Frequency in rad/ seconds OC =")
25 [Hz]=iir(N, 'bp', 'ellip', [Fp1/2, Fs1/2], [0.2, 0.200])
    //the sum of last matrix [0.2, 0.200] must be
    less than 1
26 [p, z, g]=iir(N, 'bp', 'ellip', [Fp1/2, Fs1
    /2], [0.2, 0.200])
27 [Hw, w]=frmag(Hz, 256);
28 figure(1)
29 plot(2*w, abs(Hw));
30 xlabel('Normalized Digital frequency w->')
31 ylabel('magnitude');
32 title('Magnitude response of IIR filter')
33 xgrid(1)
```
