

Glasgow College, UESTC



# Digital Signal Processing

## Lab 4

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# LAB 4

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## INTRODUCTION

This report is the homework that should be finished on the MATLAB, there are four questions about Digital Signal Processing. Which is about generate the complex exponential functions, explore their properties, understand the sampling theory and understand the true meaning of the autocorrelation.

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## 1 PROBLEM 1

Design a Butterworth bandpass IIR filter. The desired specifications of the digital filter are as follow: normalized passband edges as  $\omega_{p1} = 0.45\pi$  and  $\omega_{p2} = 0.65\pi$ , normalized stopband edges at  $\omega_{s1} = 0.3\pi$  and  $\omega_{s2} = 0.75\pi$ , passband ripple of 1dB, and a minimum stopband attenuation of 40dB. Plot the gain and phase responses using MATLAB.

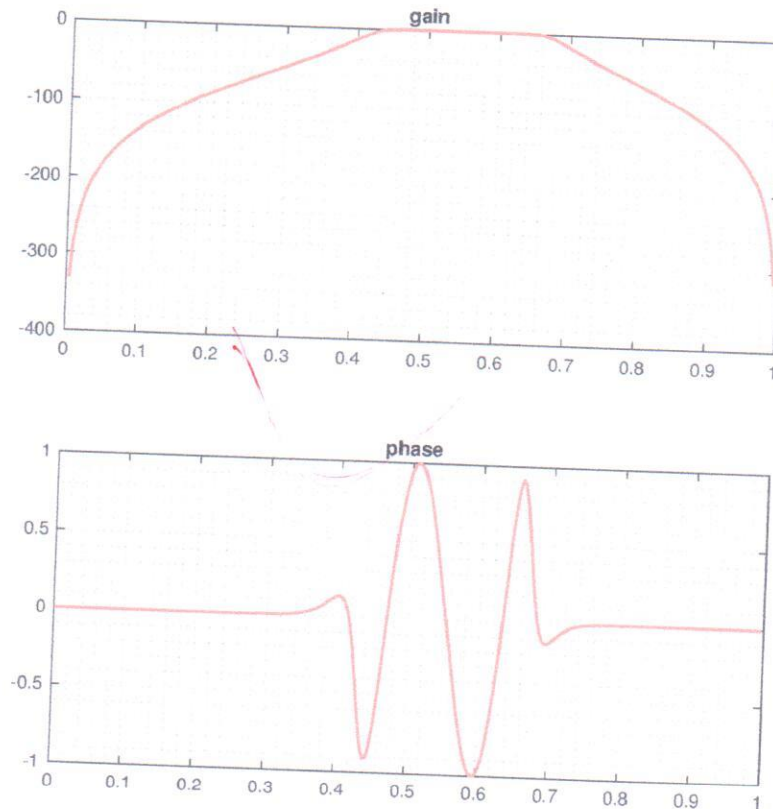


Figure 1: ROCs of z transform

```

1 clear;
2 clc;
3
4 Wp = [0.45 0.65];
5 Ws = [0.3 0.75];
6 Rp = 1;
7 Rs = 40;
8
9 [N,Wn] = buttord(Wp, Ws, Rp, Rs);
10 [b,a] = butter(N,Wn);
11 [h,omega] = freqz(b,a,256);
12 gain = 20*log10(abs(h));

```

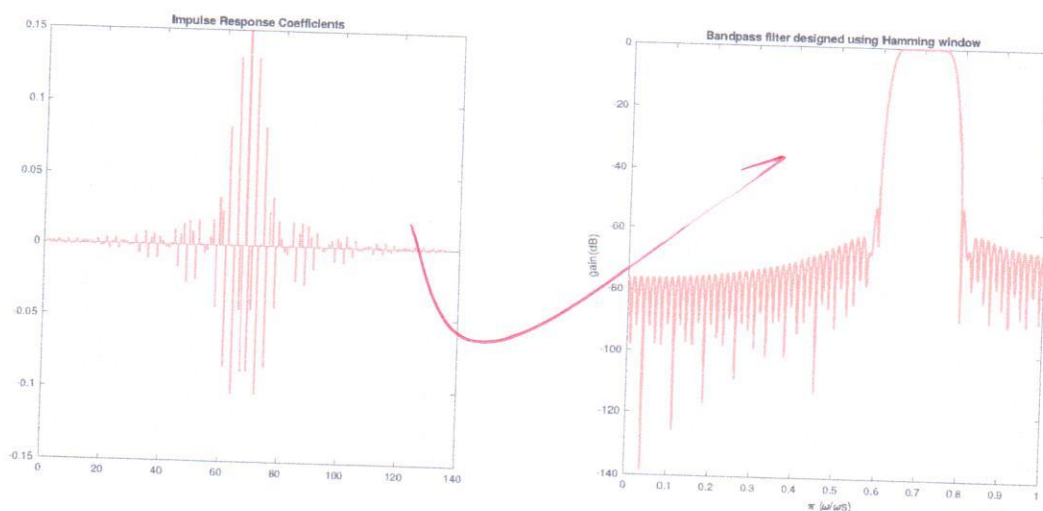
```

13
14 p1 = figure;
15 subplot 211
16 plot(omega/pi,gain,'LineWidth',1.5,'color',[1,0.5,0.5]);
17 title('gain');
18 grid minor;
19
20 subplot 212
21 plot(omega/pi,imag(h),'LineWidth',1.5,'color',[1,0.5,0.5]);
22 title('phase');
23 grid minor;
24 set(p1, 'PaperPosition', [0.05 0.05 7 7]);
25 set(p1, 'PaperSize', [7.05 7.05]); %Keep the same paper size
26 saveas(p1,['p1.pdf'],'pdf')

```

## 2 PROBLEM 2

Using the M file `fir1`, design a linear phase FIR bandpass filter with the following specifications: stopband edges at  $0.6\pi$  and  $0.8\pi$ , passband edges at  $0.65\pi$  and  $0.75\pi$ , maximum passband attenuation of 0.2 dB, and minimum stopband attenuation of 42 dB. Using each of the following windows for the design: Hamming, Hann and Kaiser. Show the impulse response coefficients, and plot the gain response of the designed filters for each case.



Impulse Response Coefficients

Bandpass filter designed using Hamming window

Figure 2: Hamming filter design

```

1 clear;
2 clc;

```

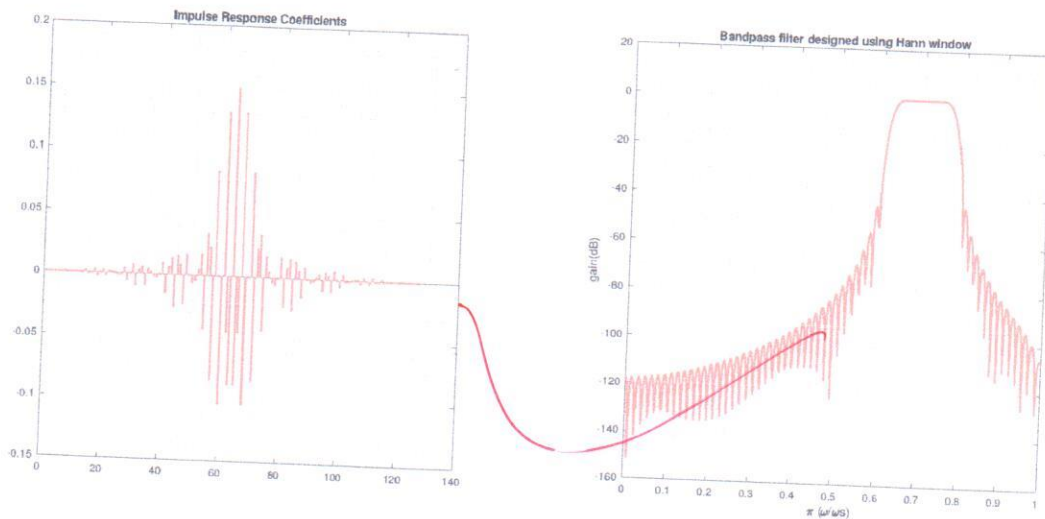
```

3
4 wp1=0.65*pi;
5 wp2=0.75*pi;
6 ws1=0.6*pi;
7 ws2=0.8*pi;
8 ap=0.2;
9 as=42;
10 dw1=wp1-ws1;
11 dw2=ws2-wp2;
12 dw=min(dw1,dw2);
13 wc1=mean([ws1 wp1]);
14 wc2=mean([ws2 wp2]);
15
16 %% Hamming
17
18 N=2*ceil((3.32*pi)/dw);
19 b=fir1(N,[wc1/pi wc2/pi]);
20 [H,w]=freqz(b,1,512);
21
22 hamm_re = figure;
23 stem(b, '.', 'LineWidth', 1.5, 'color', [1,0.5,0.5]);
24 title('Impulse Response Coefficients');
25 grid minor;
26 set(hamm_re, 'PaperPosition', [0.05 0.05 7 7]);
27 set(hamm_re, 'PaperSize', [7.05 7.05]);
28 saveas(hamm_re, ['p2_hamm_re.pdf'], 'pdf')
29
30 hamm_de = figure;
31 plot(w/pi, 20*log10(abs(H)), 'LineWidth', 1.5, 'color', [1,0.5,0.5]);
32 xlabel(['\pi \omega/\omega', 's']);
33 ylabel('gain(dB)');
34 title('Bandpass filter designed using Hamming window');
35 grid minor;
36 set(hamm_de, 'PaperPosition', [0.05 0.05 7 7]);
37 set(hamm_de, 'PaperSize', [7.05 7.05]);
38 saveas(hamm_de, ['p2_hamm_de.pdf'], 'pdf')

1 clear;
2 clc;
3
4 wp1=0.65*pi;
5 wp2=0.75*pi;
6 ws1=0.6*pi;
7 ws2=0.8*pi;
8 ap=0.2;
9 as=42;
10 dw1=wp1-ws1;
11 dw2=ws2-wp2;
12 dw=min(dw1,dw2);
13 wc1=mean([ws1 wp1]);
14 wc2=mean([ws2 wp2]);
15

```





Impulse Response Coefficients

Bandpass filter designed using Hamming window

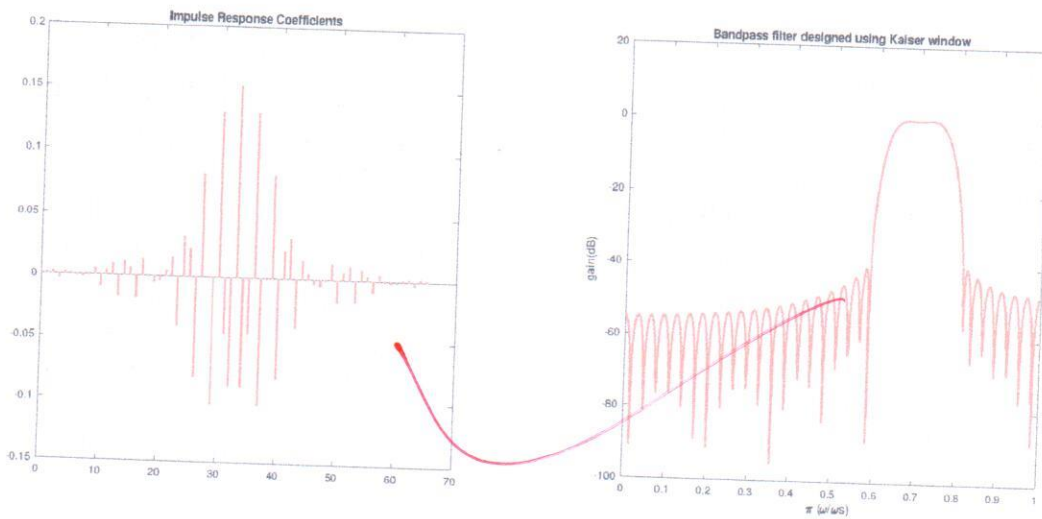
Figure 3: Hanning filter design

```

16 %% Hann
17 N=2*ceil((3.11*pi)/dw);
18 b=firl(N,[wc1/pi wc2/pi],hanning(N+1));
19 [H,w]=freqz(b,1,512);
20
21 hann_re = figure;
22 stem(b, '.', 'LineWidth', 1.5, 'color', [1, 0.5, 0.5]);
23 title('Impulse Response Coefficients');
24 grid minor;
25 set(hann_re, 'PaperPosition', [0.05 0.05 7 7]);
26 set(hann_re, 'PaperSize', [7.05 7.05]);
27 saveas(hann_re, ['p2_hann_re.pdf'], 'pdf')
28
29 hann_de = figure;
30 plot(w/pi, 20*log10(abs(H)), 'LineWidth', 1.5, 'color', [1, 0.5, 0.5]);
31 title('Bandpass filter designed using Hann window');
32 xlabel(['\pi (\omega/\omega', 's)']);
33 ylabel('gain(dB)');
34 grid minor;
35 set(hann_de, 'PaperPosition', [0.05 0.05 7 7]);
36 set(hann_de, 'PaperSize', [7.05 7.05]);
37 saveas(hann_de, ['p2_hann_de.pdf'], 'pdf')

1 clear;
2 clc;
3
4 wp1=0.65*pi;
5 wp2=0.75*pi;
6 ws1=0.6*pi;
7 ws2=0.8*pi;

```



Impulse Response Coefficients

Bandpass filter designed using Hamming window

Figure 4: Kaiser filter design

```

8 ap=0.2;
9 as=42;
10 dw1=wp1-ws1;
11 dw2=ws2-wp2;
12 dw=min(dw1,dw2);
13 wc1=mean([ws1 wp1]);
14 wc2=mean([ws2 wp2]);
15
16
17 %% Kaiser
18 ds=10^(-as/20);
19 dp=10^(-ap/20);
20 [N,Wn,beta,type]=kaiserord([wc1/pi wc2/pi],[1 0],[dp ds]);
21 b=fir1(2*N,[wc1/pi wc2/pi],kaiser(2*N+1,beta));
22 [H,w]=freqz(b,1,512);
23
24 Kaiser_re = figure;
25 stem(b, '.', 'LineWidth', 1.5, 'color', [1,0.5,0.5]);
26 title('Impulse Response Coefficients');
27 grid minor;
28 set(Kaiser_re, 'PaperPosition', [0.05 0.05 7 7]);
29 set(Kaiser_re, 'PaperSize', [7.05 7.05]);
30 saveas(Kaiser_re, ['p2_Kaiser_re.pdf'], 'pdf')
31
32 Kaiser_de = figure;
33 plot(w/pi, 20*log10(abs(H)), 'LineWidth', 1.5, 'color', [1,0.5,0.5]);
34 xlabel(['\pi (\omega/\omega_s)']);
35 ylabel('gain(dB)');
36 title('Bandpass filter designed using Kaiser window');
37 grid minor;
38 set(Kaiser_de, 'PaperPosition', [0.05 0.05 7 7]);

```

```

39 set(Kaiser_de, 'PaperSize', [7.05 7.05]);
40 saveas(Kaiser_de, ['p2_Kaiser_de.pdf'], 'pdf')

```

### 3 SUMMARY

For this Homework, I understand more about Digital Signal Processing, as well as how to use the MATLAB to do the DFT, convolution and other operations. I also know more about the sampling theorem and the autocorrelation between signals. I also know how to use the unwarp function to fix the phase gap.

### REFERENCES

- [1] Changgang-Zheng/Signals-and-Systems/report.<https://github.com/Changgang-Zheng/Signals-and-Systems>
- [2] Supplementary materials to the text book 'Digital Signal Processing: A Computer-Based Approach', 4th Edition. by S.K. Mitra, ISBN 0077320670. [http://www.bb9.uestc.edu.cn/webapps/portal/frameset.jsp?tab\\_tab\\_group\\_id=\\_2\\_1&url=%2Fwebapps%2Fblackboard%2Fexecute%2Flauncher%3Ftype%3DCourse%26id%3D\\_13014\\_1%26url%3D](http://www.bb9.uestc.edu.cn/webapps/portal/frameset.jsp?tab_tab_group_id=_2_1&url=%2Fwebapps%2Fblackboard%2Fexecute%2Flauncher%3Ftype%3DCourse%26id%3D_13014_1%26url%3D)
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- [4] Summary of DSP course in school of communication, University of Electronic Science and Technology of China <https://wenku.baidu.com/view/b3b2f5b289eb172dec63b708.html>