Glasgow College, UESTC

Digital Signal Processing

Lab 3

Author: Changgang Zheng
UoG ID: 2289258Z
UESTC ID: 2016200302027
E-mail: chnaggangzheng@std.uestc.edu.cn

LAB 3

CHANGGANG ZHENG[1]

May 8, 2019

CONTENTS

LIST OF FIGURES

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.

1*Glasgow College, University of Electronic Science and Technology of China, ChengDu, China*

1 PROBLEM 1

Using MATLAB determine the second-order factored form of the following z-transforms. And show their pole-zero plots. Then determine all possible ROCs of above z-transform.

$$\text{num} = \begin{bmatrix} 4 & 15.6 & 6 & 2.4 & -6.4 \end{bmatrix}$$

$$\text{den} = \begin{bmatrix} 3 & 2.4 & 6.3 & -11.4 & 6 \end{bmatrix}$$

$$G(z) = \frac{4z^4 + 15.6z^3 + 6z^2 + 2.4z - 6.4}{3z^4 + 2.4z^3 + 6.3z^2 - 11.4z + 6} \tag{1}$$

All possible ROCs of above z-transform is as follow.

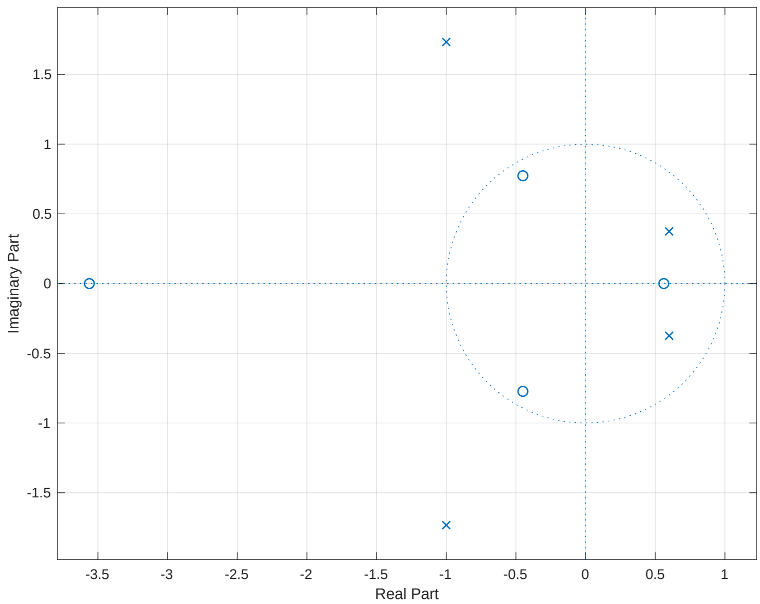

Figure 1: ROCs of z transform

- $0 \leqslant |Z| < 0.7071$
- $0.7071 < |Z| < 2$
- $2 < |Z| < \infty$

```
%% Problem 1
num=[4 15.6 6 2.4 -6.4];
den=[3 2.4 6.3 -11.4 6];
```

```
[sos,G]=tf2sos(num,den);
[z,p,k]=tf2zp(num,den);
p1 = figure;
zplane(z,p);
grid on
set(p1, 'PaperPosition', [0.05 0.05 9 7]);
set(p1, 'PaperSize', [9.05 7.05]);
saveas(p1,['p1.pdf'],'pdf')
```

2 PROBLEM 2

Writing a MATLAB program to determine the rational form of a z-transform whose zero are at some point and the gain constant k is 2.1.

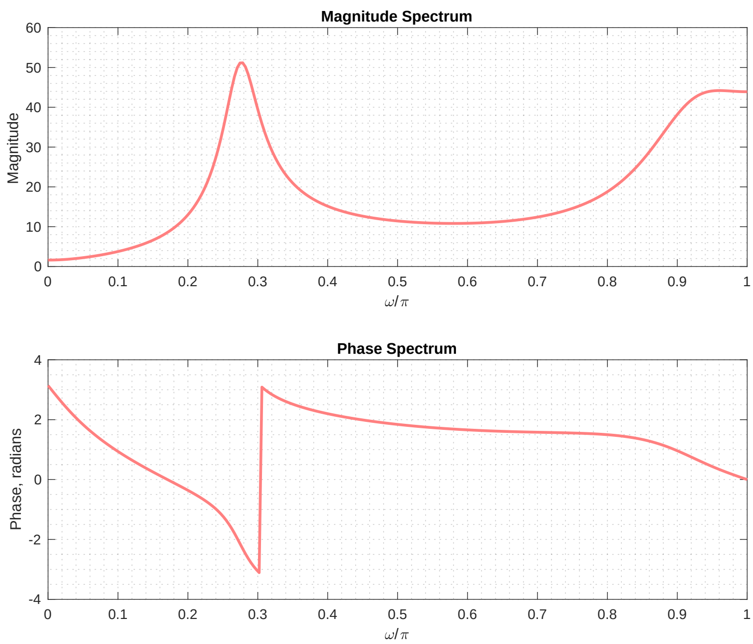

Figure 2: Magnitude and phase spectrum of the sampled sequence

```
%% Problem 2
z=[1.2 -1j*0.5+2.3 1j*0.2-0.4 -1j*0.2-0.4 1j*0.5+2.3];
p=[0.5 1j*0.2-0.75 1j*0.7+0.6 -1j*0.7+0.6 -1j*0.2-0.75];
g=2.1;
% z=input('input the zeros=');
% p=input('input the ploes=');
% g=input('input the gain constant=');
[num,den]=zp2tf(z',p',g);
w = 0:pi/(255):pi;
h=freqz(num,den,w);
```

```
p2 = figure;
subplot(2,1,1)
plot(w/pi,abs(h),'LineWidth',2,'color',[1,0.5,0.5]);
title('Magnitude Spectrum')
xlabel('\omega/\pi');
ylabel('Magnitude')
grid minor
subplot(2,1,2)
plot(w/pi,angle(h),'LineWidth',2,'color',[1,0.5,0.5]);
title('Phase Spectrum')
xlabel('\omega/\pi'); ylabel('Phase, radians')
grid minor
set(p2, 'PaperPosition', [0.05 0.05 9 7]);
set(p2, 'PaperSize', [9.05 7.05]);
saveas(p2,['p2.pdf'],'pdf')
```

3 PROBLEM 3

Using the function fir1 and window of Kaiser, design a linear-phase FIR low-pass filter meeting the following specifications: passband edge frequency = 2 kHz, stopband edge frequency=2.5 kHz, passband ripple δ=0.005, stopband ripple δ=0.005, and sampling rate of 10 kHz. Plot its gain and phase responses and check if it meets the specifications?

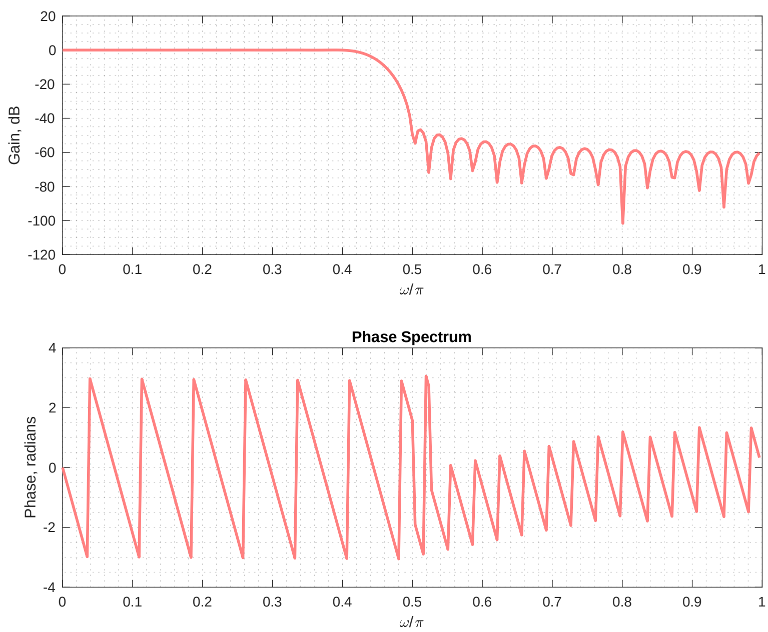

Figure 3: Magnitude and phase spectrum of the sampled sequence

From the plot, we can conclude that it meets the specifications.

```
%% Problem 3
F = [2000 2500];
A = [1 0];
DEV = [0.005 0.005];
Fs = 10000;
[n,wn,beta,typ]=kaiserord(F,A,DEV,Fs);
b=fir1(n,wn,kaiser(n+1,beta),'noscale');
[h,omega]=freqz(b,1,256);
p3=figure;
subplot(2,1,1)
plot(omega/pi,20*log10(abs(h)),'LineWidth',2,'color',[1,0.5,0.5]);
xlabel('\omega/\pi');
ylabel('Gain, dB');
grid minor
subplot(2,1,2)
plot(omega/pi,angle(h),'LineWidth',2,'color',[1,0.5,0.5]);
grid minor
title('Phase Spectrum')
xlabel('\omega/\pi'); ylabel('Phase, radians')
set(p3, 'PaperPosition', [0.05 0.05 9 7]);
set(p3, 'PaperSize', [9.05 7.05]);
saveas(p3,['p3.pdf'],'pdf')
```

4 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

[3] Digital Signal Processing: A Computer-Based Approach, 4th Edition. by S.K. Mitra, ISBN 0077320670.

[4] Summary of DSP course in school of communication, University of Electronic Science and Technology of China https://wenku.baidu.com/view/b3b2f5b289eb172dec63b708.html