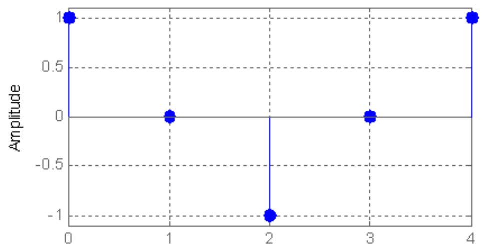
Multimedia Signal Processing 1st Module

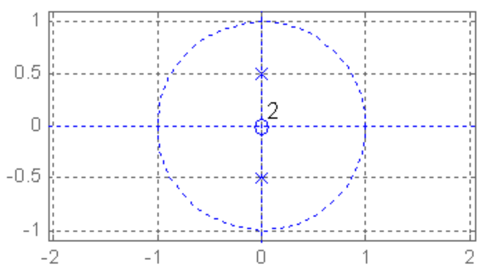
2 /12/2014 – Pre-exam

Ex.1 (Pt.11)

Consider a digital filter system is composed of a cascade of two filters, the first one has the following impulse response:



While the second filter has two complex conjugate poles in these positions:



1) Define the whole zero-pole plot.
2) Is it stable or unstable? What kind of filter is it?
3) Plot an approximate amplitude response.
4) Provide the difference equation of the filter and evaluate the first five samples of the whole filter impulse response.

Ex.2 (Pt.11 NOT to be done in MATLAB)

Let x(n) be a discrete-time rectangular pulse of length $L=5$, i.e. $x(n)=u(n)-u(n-5)$,and $h(n)$ be a discrete-time rectangular pulse of length $M=3$, i.e. $h(n)=u(n)-u(n-3)$.

The output $y(n)$ is evaluated as an 8-points sequence obtained from the inverse DFT of the product $Y_N(k)=X_N(k)H_N(k)$.

1) Define how can be obtained the 64 coefficients of the **W** matrix to get the DFT of $x(n)$ and $h(n)$: do not list all coefficients but only the formula in order to get them where *r* and *c* represent raw and column index of the matrix element.
2) Get the 8 values of $y(n)$ working in the time domain,
3) Will time-aliasing be present?

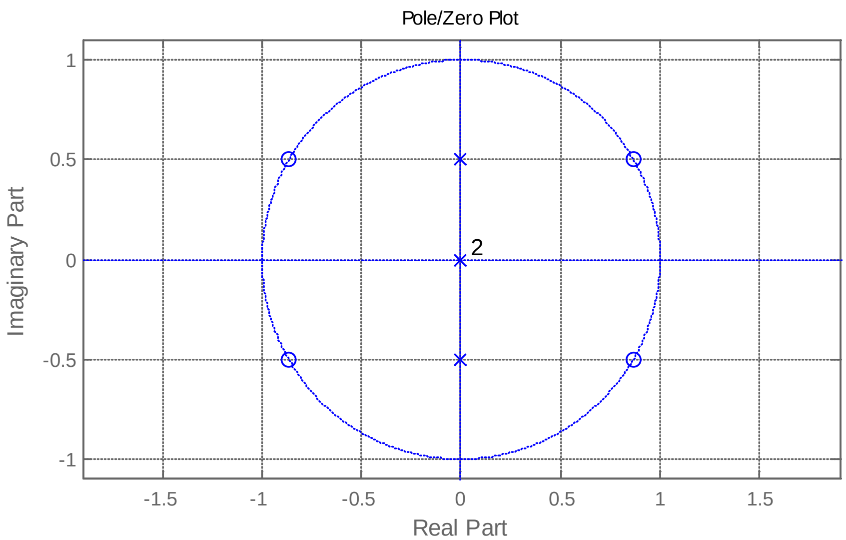
Ex.3 (Pt. 11 – MATLAB code)

1) implement an all-pass filter that has two complex zeros in $\rho e^{\pm j\omega_0}$ and with $\rho=1.2$ and $\omega_0=0.4\pi$ and two poles (place them in the proper position in order to get an “all-pass” filter).
2) plot the poles and the zeros of the all pass filter in the z-plane
3) compute its frequency response from 0 to π and plot its modulus and phase in the same figure (two subplots), with the frequency in pi units
4) compute its impulse response (the first 512 samples)

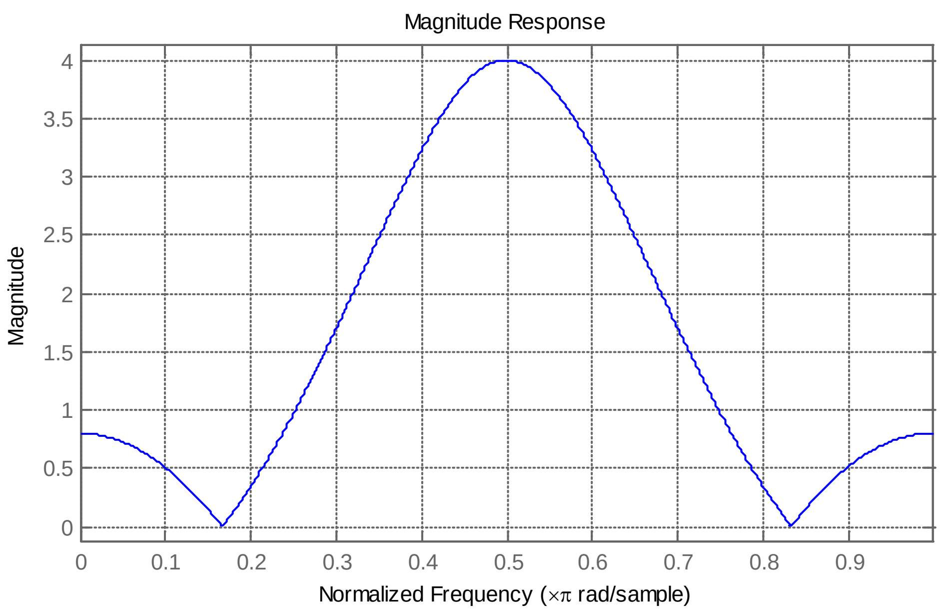
Solutions

Ex.1

The filter is a stable band pass filter.



Amplitude response:



Difference equation $y(n)+\frac{1}{4}y(n-2)=x(n)-x(n-2)+x(n-4)$

Impulse response:

y(0)=x(0)=1;
y(1)=0;
y(2)=-y(0)/4 - x(0)=-5/4
y(3)=0
y(4)=-y(2)/4+x(0)=21/16.

Ex.2

The linear convolution of $x(n)$ and $h(n)$ will have a length of 5+3-1=7, so, since N=8 there will be no time-domain aliasing.

The DFT coefficient will be $e^{-j\frac{2\pi}{8}r\cdot c}$

The output, convolving the x and h will be: $\{1,2,3,3,3,2,1,0\}$

Ex.3

```
clc
clear all
close all

% 1) implement an all-pass filter that has two complex zeros in
% rho*e^(+- j omega_0)
rho=1.2;
omega_0=0.4*pi;
z=[rho*exp(-1i*omega_0); rho*exp(+1i*omega_0)];
p=1./conj(z);


% 2) plot the poles and the zeros of the all pass filter in the z-plane
figure;
zplane(z,p);
title('All-pass Filter');


%3) compute its frequency response from 0 to ? and plot its modulus and
% phase in the same figure (two subplots), with the frequency in pi units

a=poly(p);
b=poly(z);
[H, f]=freqz(b,a);
figure;
subplot(2,1,1);
plot(f/pi,abs(H));
title('Magnitude');
xlabel('\omega [\pi units]');
ylabel('|H(z)|');
subplot(2,1,2);
plot(f/pi,unwrap(angle(H)));
title('Phase');
xlabel('\omega [\pi units]');
ylabel('\angle H(z)');


%4) compute its impulse response (the first 512 samples)

N=512;
delta=zeros(N,1);
delta(1)=1;
h=filter(b,a,delta);
figure;
plot(0:N-1,h);
title('Impulse response');
xlabel('n');
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ylabel('h(n)');
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