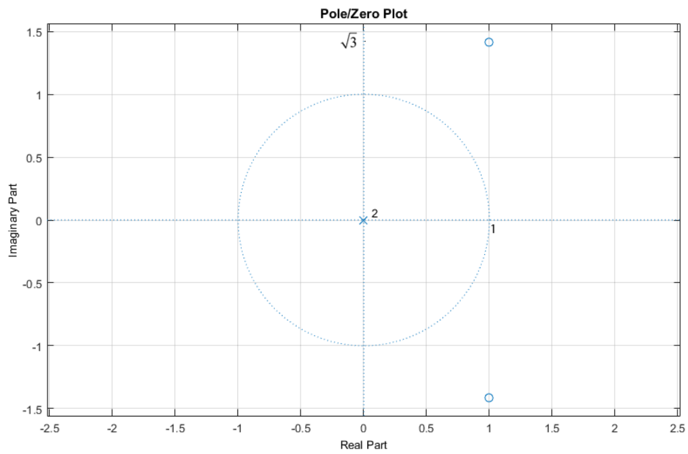
Multimedia Signal Processing 1st Module

23/2/2016

Ex.1 (Pt.13)

A filter $H_1(z)$ presents the following zero-pole plot:



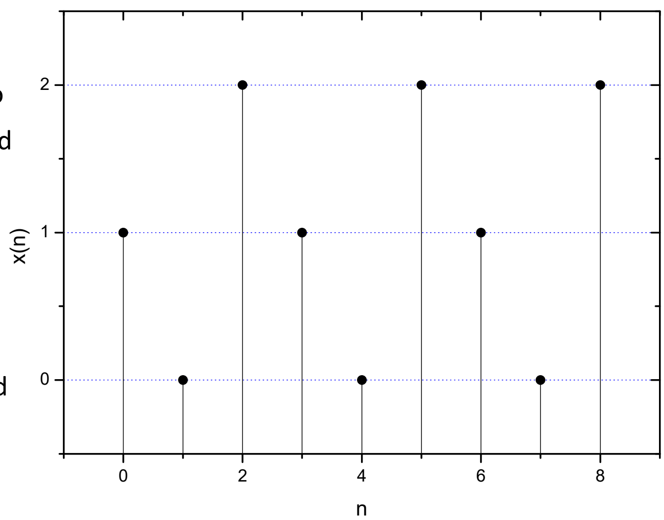
Where the two zeros are in $1 \pm \sqrt{3}j$. Define a minimum phase filter $H_2(z)$ with **exactly** the same amplitude response.

Define a pure IIR filter $H_3(z)$ that, placed after the filter $H_2(z)$ is able to completely remove its effect.

Find the first five output samples of the impulse response of filters $H_1(z)$, $H_2(z)$ and $H_3(z)$.

Ex.2 (Pt.9)

We want to remove the continuous component (zero frequency) from the periodic signal $x(n)$ represented on the right just working with the DFT.



Propose a filter $H(k)$ in the frequency domain and apply it to the signal (provide also the W matrix).

Provide also the output in the time domain $y(n)$ and a period of the filter in the time domain $h(n)$.

Ex.3 (Pt. 11 – MATLAB code)

```
Consider a musical signal stored in the file 'music.wav'.
1) Load the signal and display the number of channels. If the signal is
multichannel (> 1 channel), convert it to mono (1 channel).
2) Select the first 5 s of the signal
3) Generate a sinusoidal disturbance with frequency 5 kHz and amplitude 0.01 and
add it to the musical signal.
4) We want to remove the disturbance using a filter h(n) whose z-transform is
          1 - 1.5136 z^(-1) + z^(-2)
H(z) = -----------------------------------
          1 - 1.5060 z^(-1) + 0.99 z^(-2)
Write the appropriate commands to plot the zero-pole diagram of H(z) and its
frequency response H(jW).
5) Filter the disturbed signal with the filter defined in point 4.
```

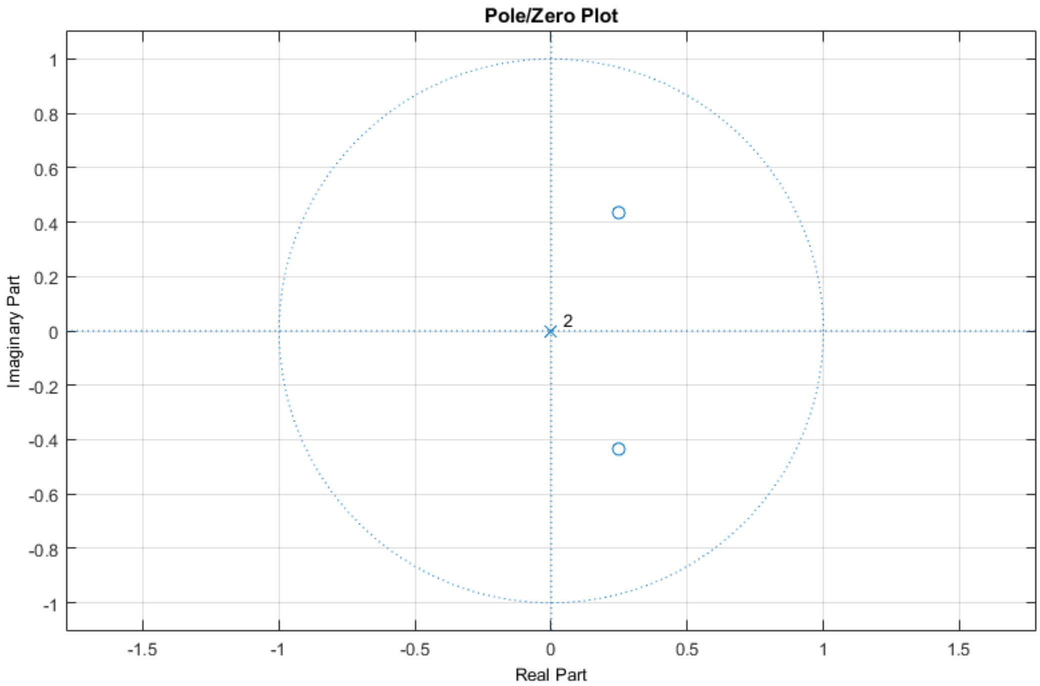
Solutions

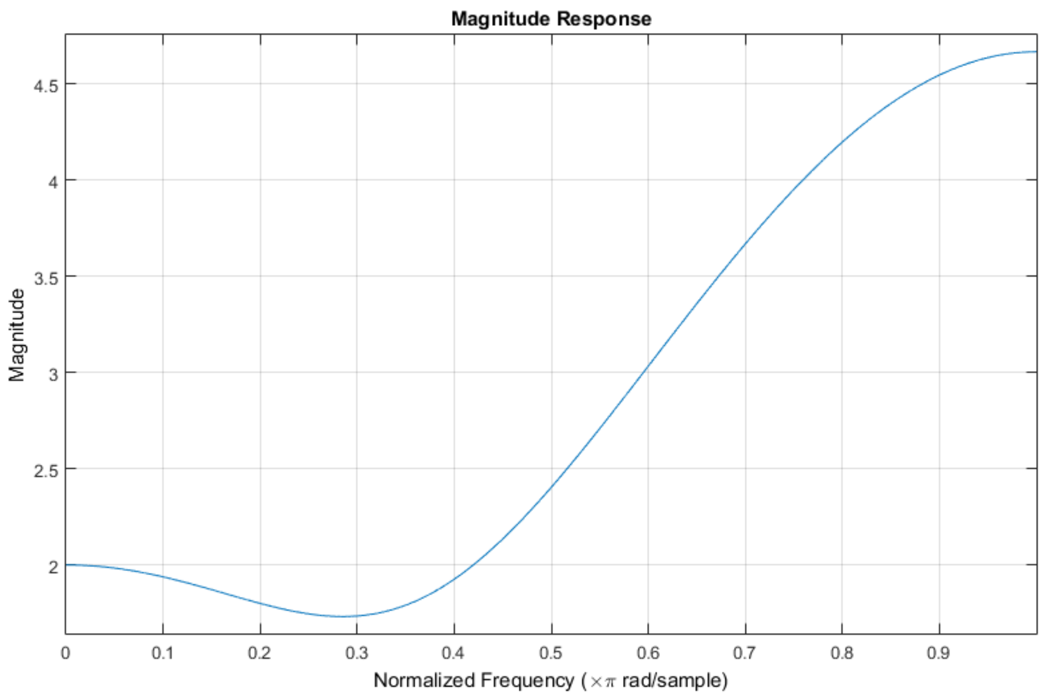
Ex.1

$$H_1(z) = 1 - 2z^{-1} + 3z^{-2}$$

$$H_2(z) = A\left(1 - \frac{1}{2}z^{-1} + \frac{1}{4}z^{-2}\right)$$

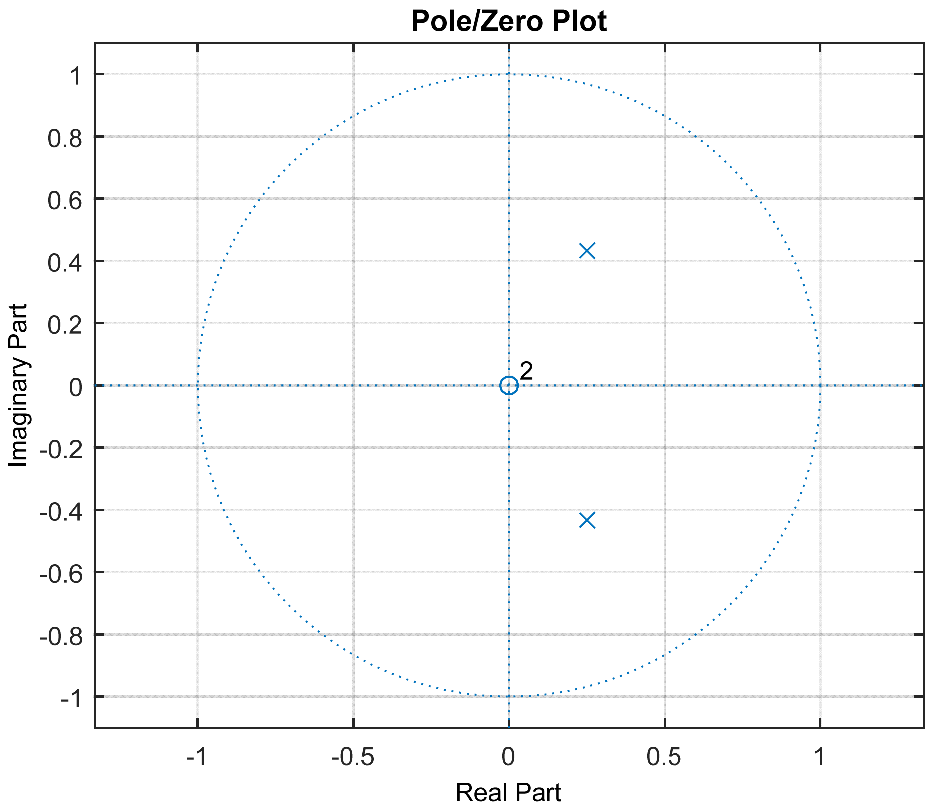
Imposing, e.g. $H_1(z=1) = H_2(z=1) \Rightarrow 2 = A \cdot \frac{3}{4} \Rightarrow A = \frac{8}{3}$



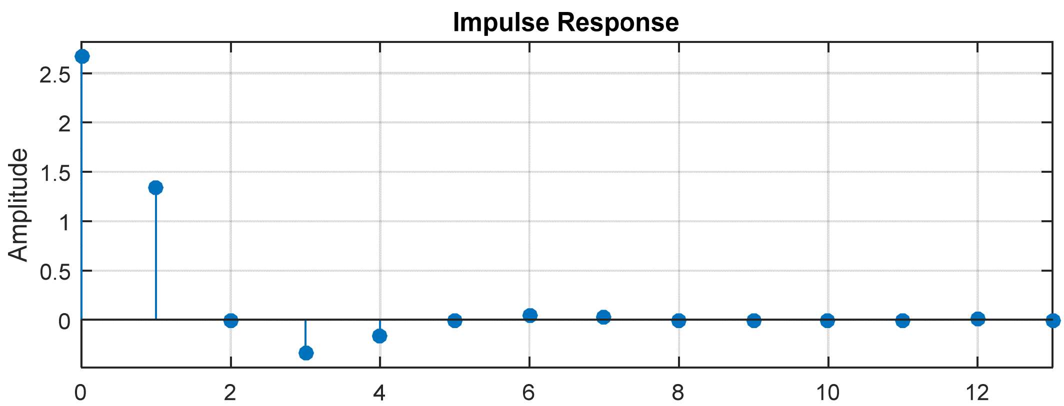


Which is the same for $H_1(z)$ and $H_2(z)$.

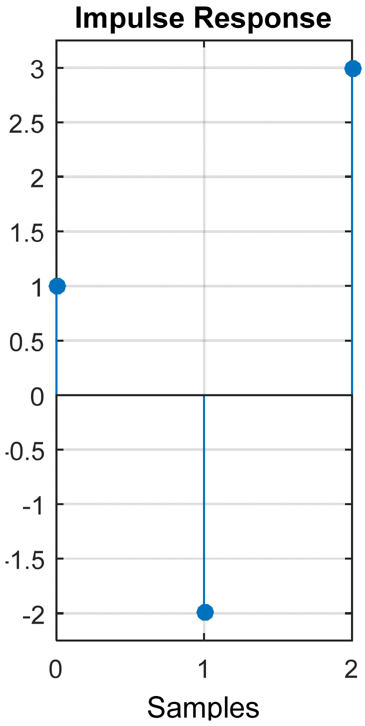
The IIR filter is $H_3(z) = \frac{3}{8}\frac{1}{1-\frac{1}{2}z^{-1}+\frac{1}{4}z^{-2}}$



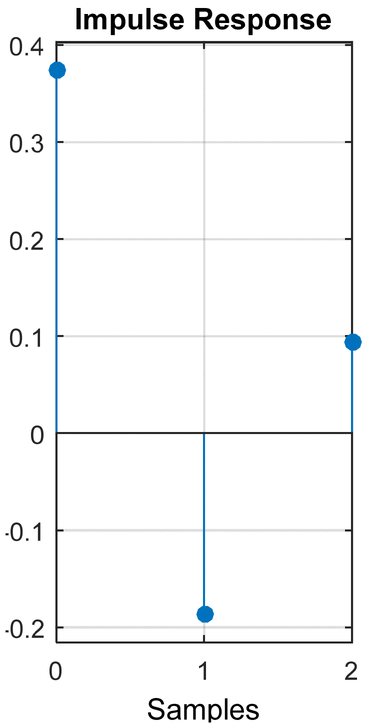
its impulse response is:



The impulse response for $H_1(z)$ is:



and for $H_2(z)$



Ex.2

The W Matrix will be:

$$\begin{bmatrix} 1 & 1 & 1 \\ 1 & -\frac{1}{2}+\frac{\sqrt{3}}{2}j & -\frac{1}{2}-\frac{\sqrt{3}}{2}j \\ 1 & -\frac{1}{2}-\frac{\sqrt{3}}{2}j & -\frac{1}{2}+\frac{\sqrt{3}}{2}j \end{bmatrix}$$

the DFT of $x(n)$ will be:

$$\begin{bmatrix} 1 & 1 & 1 \\ 1 & -\frac{1}{2}-\frac{\sqrt{3}}{2}j & -\frac{1}{2}+\frac{\sqrt{3}}{2}j \\ 1 & -\frac{1}{2}+\frac{\sqrt{3}}{2}j & -\frac{1}{2}-\frac{\sqrt{3}}{2}j \end{bmatrix} \begin{bmatrix} 1 \\ 0 \\ 2 \end{bmatrix} = \begin{bmatrix} 3 \\ \sqrt{3}j \\ -\sqrt{3}j \end{bmatrix}$$

The filter $H(k)$ will be

$$H(k) = \begin{bmatrix} 0 \\ 1 \\ 1 \end{bmatrix}$$

so the output signal in frequency domain will be

$$Y(k) = X(k) \cdot H(k) = \begin{bmatrix} 0 \\ \sqrt{3}j \\ -\sqrt{3}j \end{bmatrix}$$

and $y(n) = \begin{bmatrix} 0 \\ -1 \\ 1 \end{bmatrix}$

Ex.3

```
clear
clc
close all

% Consider a musical signal stored in the file 'music.wav'.
% 1) Load the signal and display the number of channels. If the signal is
multichannel (> 1 channel),
% convert it to mono (1 channel).
[y, Fs] = audioread('music.wav');
n_ch = size(y,2);
disp(['The number of channels is ' num2str(n_ch)]);
if n_ch > 1
    y = sum(y,2)/n_ch;
end

% 2) Select the first 5 s of the signal
dur = 5;
y = y(1:dur*Fs);
n = (0:length(y)-1)';
t = n/Fs;

figure
plot(t,y)
xlabel('Time [s]'), ylabel('Amplitude')
title('Original signal')

% 3) Generate a sinusoidal disturbance with frequency 5 kHz and amplitude 0.01 and
add
% it to the musical signal.
fd = 5000;
d = 0.01*sin(2*pi*fd*n/Fs);
yd = y+d;

figure
plot(t,yd)
xlabel('Time [s]'), ylabel('Amplitude')
title('Disturbed signal')

% 4) We want to remove the disturbance using a filter h(n) whose z-transform is
%            1 - 1.5136 z^(-1) + z^(-2)
% H(z) = -----------------------------------
%            1 - 1.5060 z^(-1) + 0.99 z^(-2)
% Write the appropriate commands to plot the zero-pole diagram of H(z) and its
frequency response H(jW).
num = [1 -1.5163 1];
den = [1 -1.5060 0.99];

figure
zplane(num, den);

figure
freqz(num, den, 512, Fs);
```

```
% 5) Filter the disturbed signal with the filter defined in point 4.
yd_f = filter(num, den, yd);
```