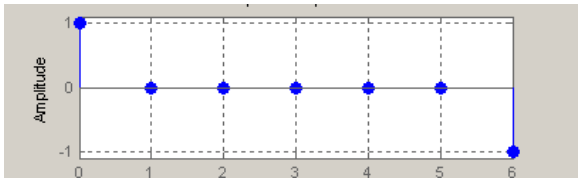
Multimedia Signal Processing 1st Module

16 /2/2010

Ex.1 (Pt.6)

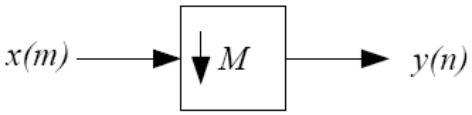
Consider the filter H(z) whose impulse response is:



1. Define its *z-transform* an plot its zeros-poles diagram.
2. Draw an approximate behavior of its amplitude and phase response at different normalized frequencies.
3. What would happen if the first sample, instead of being 1 would be 1/64 (and all the other samples maintain the same values)? Is the filter a minimum, maximum of mixed phase? Why?

Ex.2 (Pt.5)

Describe the downsampling of an order of 4 of a signal: (where M=4).



1. If the original signal spectrum extends from -π/2 to π/2 (in normalized frequencies) draw the final spectrum after downsampling (quoting both axes).
2. How can be avoided aliasing? Describe the optimal decimation chain for this case.

Ex.3 (Pt.6)

Assume that you know the PSD of a random signal and you want to approximate it using an AR filter of order 2 on a white noise of power σ^2. How can you define the optima filter parameters?

Ex.4 (Pt.5)

Select the shortest window's length M that must be used in order to distinguish two sinusoids:

$$x_1(n) = \exp\left(j\frac{\pi}{5}n\right), x_2(n) = \exp\left(j\frac{\pi}{2}n\right)$$

Consider three kinds of windows:
rectangular window
Blackman's window
Hann's window
Add 10% to the shortest length.
HINT: M must be an integer value: round M to the nearest integer toward infinity.
Plot the DFT (modulus) of the original signal: x=x1+x2.
and of the windowed signal x_w (with Hann's window)

Ex.5 (Pt.5)

Let x(n) be a random process generated by feeding a white noise signal z(n) ($r_z(i)=4\delta(i)$) into a system characterized by the following impulse response: $h(n) = g_1(n)+ g_2(n)$. Where

$$g_1(n) = \left(0.7\exp(j0.4\pi)\right)^n u(n), \quad g_2(n) = \left(0.7\exp(-j0.4\pi)\right)^n u(n)$$

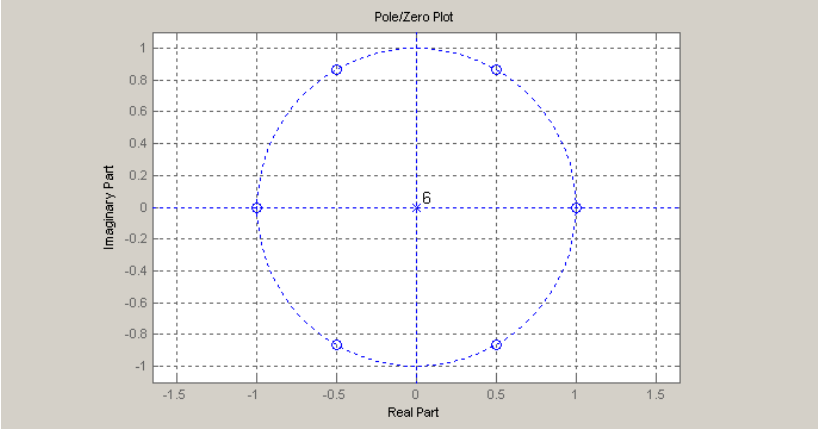
Compute the transfer function H(z) of the system.
Plot the modulus and the phase of the frequency response of the filter.
Generate one realization of N=1000 samples of the output of the system: y(n) = conv(h(n),z(n))

Estimate the PSD using periodogram method.

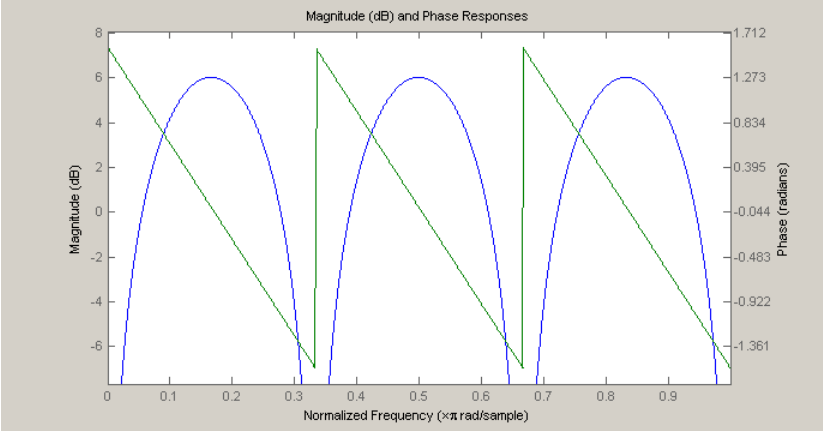
Solutions

Ex.1:

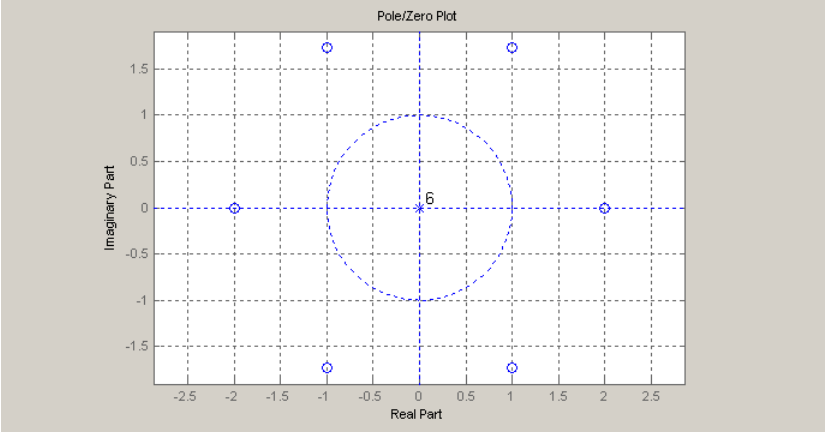
The zeros-poles diagram is:



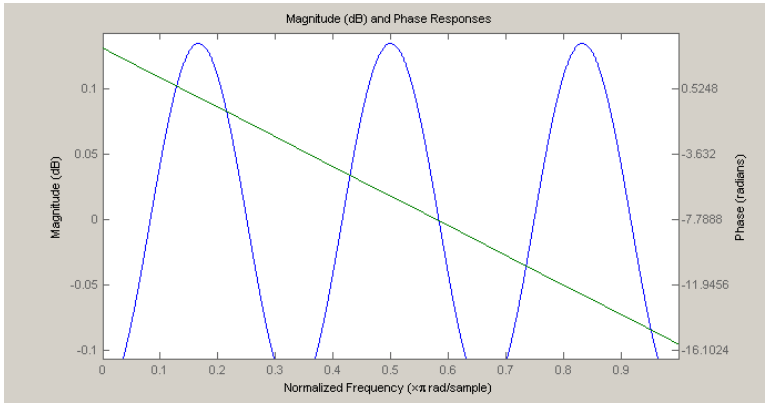
The amplitude and phase response is:



3. The poles-zeros plot would become:



Due to the zeros outside of unit circle the filter will become a maximum phase filter:


Magnitude (dB) and Phase Responses
Magnitude (dB)
Phase (radians)
Normalized Frequency (×π rad/sample)
0.1
0.05
0
-0.05
-0.1
0.5248
-3.632
-7.7888
-11.9456
-16.1024
0
0.1
0.2
0.3
0.4
0.5
0.6
0.7
0.8
0.9