Ex.1 (Pt.7)
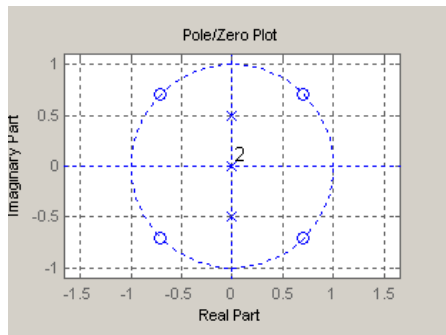
Consider the filter H(z) whose zero-pole diagram is the following:

Where the 4 zeros are on the unit circle while the two pure imaginary poles have a distance ½ from the origin.

1. Define its *z-transform*.
2. Draw an approximate behavior of its amplitude and phase response at different normalized frequencies. What will be the exact amplitude response for $\omega = 0$?

Ex.2 (Pt.6)

The first 3 samples of the autocorrelation function of a random process have the following values:
r(0) = 1, r(1) = 0.7 and r(2) = -0.5. Define an autoregressive filter with two poles that, starting from a white noise with variance 16 is able to approximate the random process.

Ex.3 (Pt.6)

Consider a filter whose impulse response is

$$h(n) = \delta(n) + 2\delta(n-2) + 3\delta(n-4).$$

1. Describe the phase of the filter, is it linear? Is the filter stable?
2. If it is a minimum phase filter convert it into a maximum phase or vice-versa.

Ex.4 (Pt.6)

Build a sinusoidal signal x=sin(2πft) over a temporal axis of 512 samples. Assume that the sampling period T=1/2048 and f=128.
Consider the signal z(n)=a*x(n)+b*y(n) where y(n)=u(n) , a=0.7 and b=0.5.
Compute the DFT of the mixed signal z (from X and Y) and plot it (modulus and phase), with the frequency domain defined as each of the below modes:

- index of the samples
- frequency
- normalized frequency
- radian frequency

Ex.5 (Pt.6)

Build a signal sum of three different sinusoid sin(2πft) at the radian frequencies $\omega_1 = \pi/8$, $\omega_2 = \pi/10$ $\omega_3 = \pi/3$. The signal is defined over a temporal axis of 512 samples. (Assume that the sampling period T=1).

Decimate the signal by a factor M=4 using the Matlab function 'fir1' for designing the filter, but not the function 'decimate'.

Compare the original and the decimated signal in the time and in the frequency domain (only modula).