Faculty of Engineering & Technology

Electrical & Computer Engineering Department

ENCS4310: Digital Signal Processing

Summer Semester, 2024/2025

Student Name : Aya Abed Al-Rahman Fares

Student Id : 1222654

Instructor : Dr.AL-Hareth zaiod

Section.no : 2

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Question#1: For the following Signal x[n]:

$$\boldsymbol{x}[n] = \begin{cases} 1, \mathrm{n} = 1 \ \ldots .10 \\ 0, \mathrm{Otherwise} \end{cases}$$

$\boldsymbol{x}[n]$ Signal and its plot in MATLAP:

MATLAP code :

spProj.m

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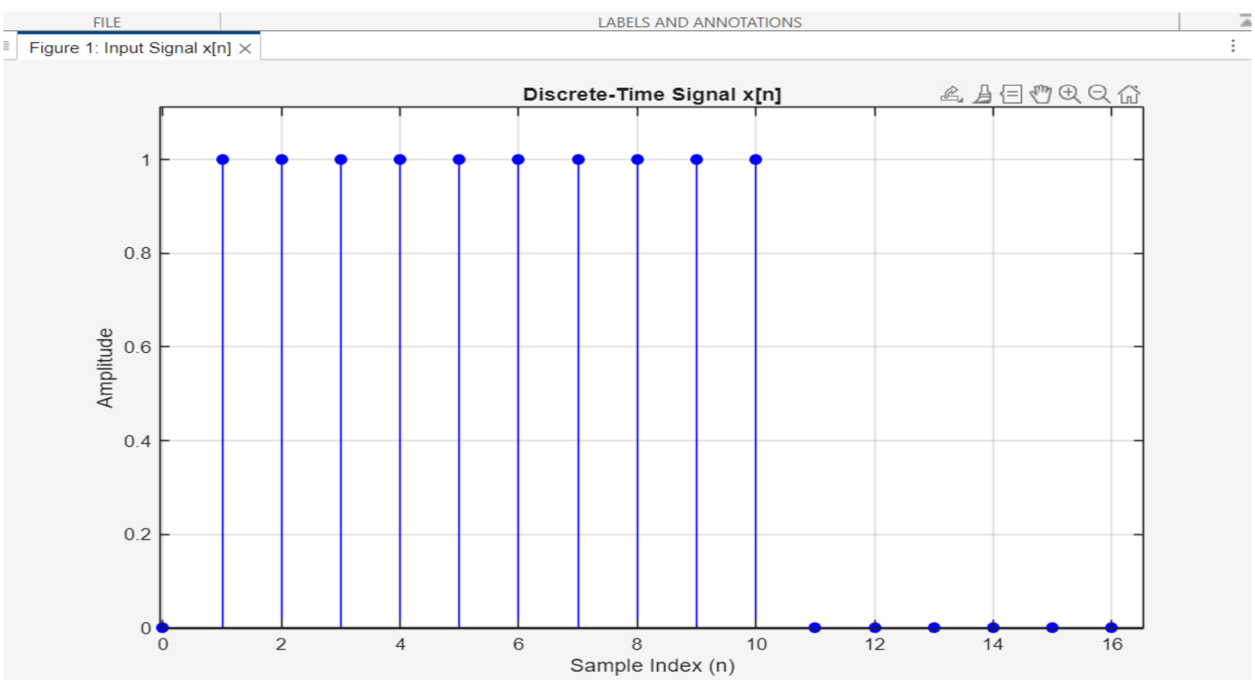
```
n = 0:255;
x = zeros(size(n));
x(n >= 1 & n <= 10) = 1;

figure('Name', 'Input Signal x[n]');
stem(n, x, 'b', 'filled');
title('Discrete-Time Signal x[n]');
xlabel('Sample Index (n)');
ylabel('Amplitude');
grid on;
axis([0 25 -0.5 1.5]);
```

Plot of $\boldsymbol{x}[n]$:



Part 1: Calculate and plot the Spectrum.

MATLAP code :

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DspProj.m *

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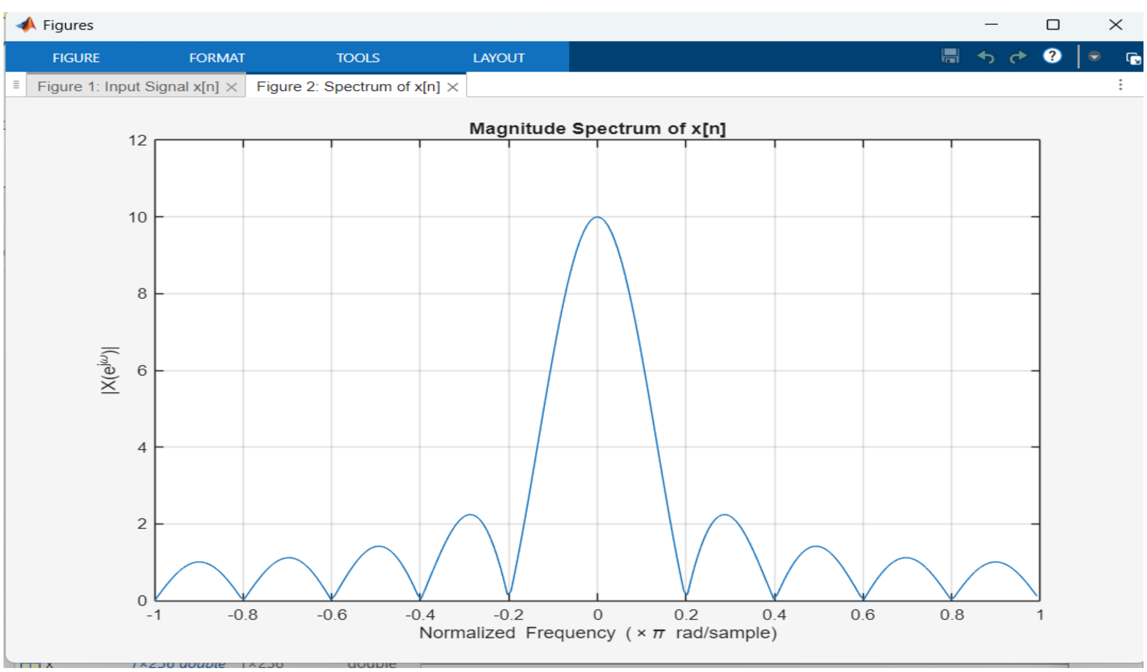
```matlab
% Q1.1:
% Calculate Spectrum

N_fft = 256;
X_fft = fft(x, N_fft);

freq_axis = (-N_fft/2 : N_fft/2-1) * (2*pi/N_fft);
MagX = abs(fftshift(X_fft));

% Plot the Spectrum :
figure('Name', 'Spectrum of x[n]');
plot(freq_axis/pi, MagX);
title('Magnitude Spectrum of x[n]');
xlabel('Normalized Frequency (\times\pi rad/sample)');
ylabel('|X(e^{j\omega})|');
grid on;
axis([-1 1 0 12]);
```

Plot of $x[n]$:



Part 2:

Compute the output y[n] for the length-4 filter (M=3) whose coefficients are

{bk} = {3, -1, 2, 1}. The causal running average is a special case of the general causal difference equation

$$y[n] = \sum b\, x[n-k] \quad \text{---(1)}$$

where the coefficients bk are fixed numbers, usually the bk coefficients are not all the same, and then we say that Equation (1) defines a weighted running average of M + 1 samples.

MATLAP code :

DspProj.m

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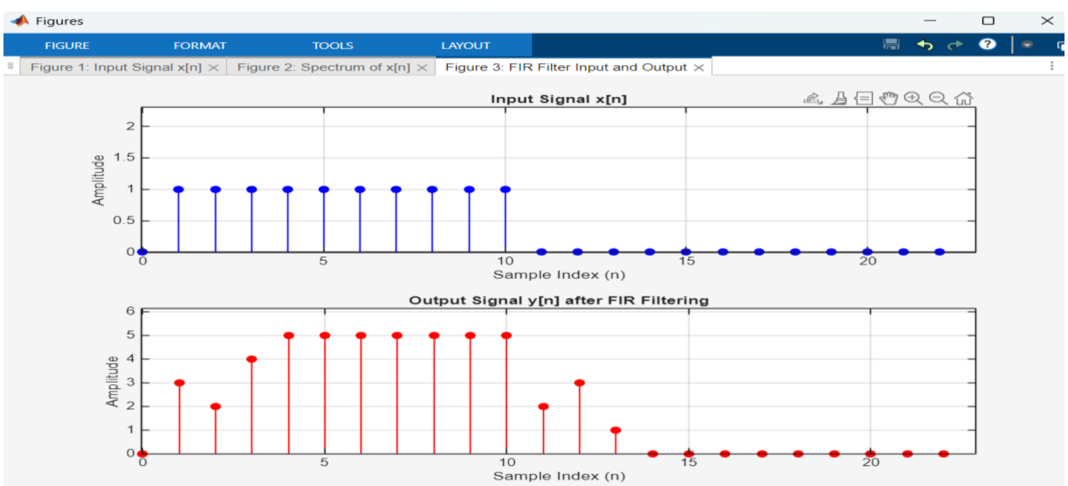
```
%filter coefficients {bk}
b_fir = [3, -1, 2, 1];
a_fir = 1;

y = filter(b_fir, a_fir, x); %Output y[n] using the filter function

% Input Plot for comparing:
figure('Name', 'FIR Filter Input and Output');
subplot(2,1,1);
stem(n, x, 'b', 'filled');
title('Input Signal x[n]');
xlabel('Sample Index (n)');
ylabel('Amplitude');
grid on;
axis([0 30 -1 2]);

% output Plot:
subplot(2,1,2);
stem(n, y, 'r', 'filled');
title('Output Signal y[n] after FIR Filtering');
xlabel('Sample Index (n)');
ylabel('Amplitude');
grid on;
axis([0 30 -2 6]);
```

Plot of $x[n]$:



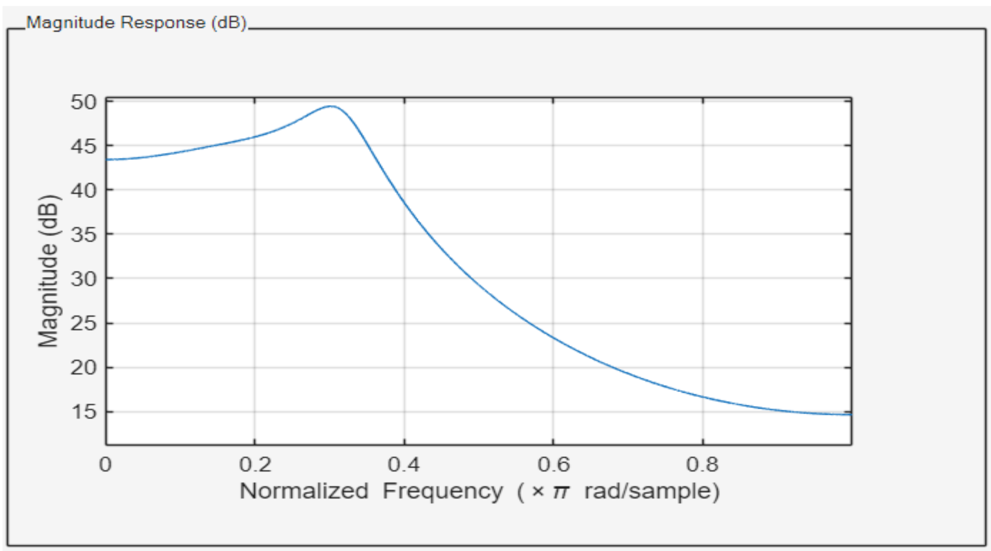
Question 2:. Exploring Minimum-Phase and All-Pass Systems in MATLAB

Using the MATLAB filter design tool, design your own example of

Part 1. A minimum phase system

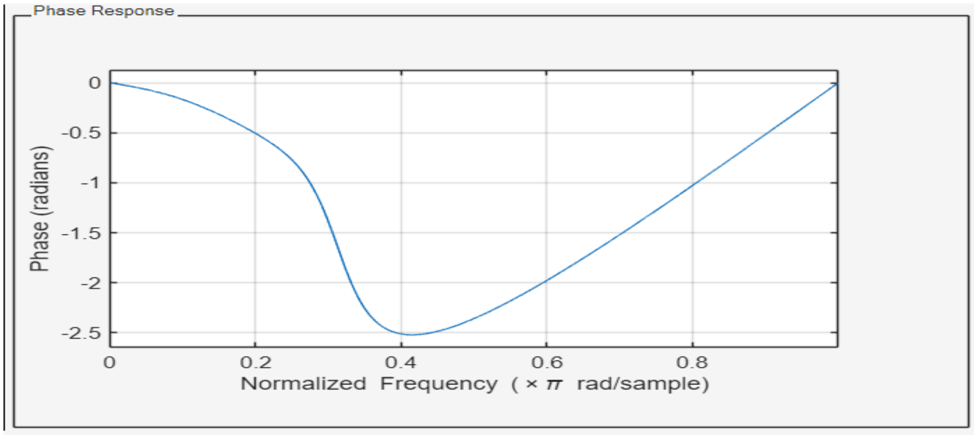
Use the tool to generate:

1. magnitude response



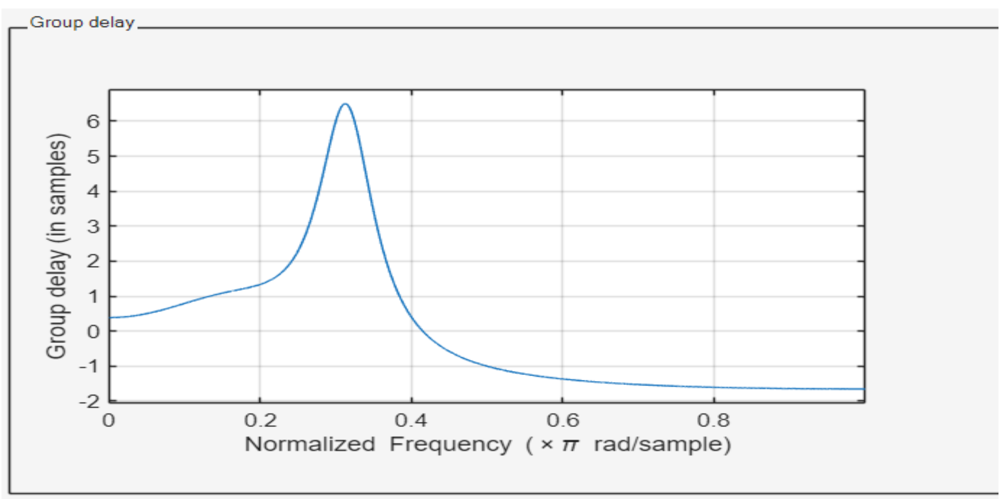
- The plot shows a low-pass filter behavior.
- The gain is high in the lower frequencies (below ~0.3π rad/sample), with a peak near 0.3π.
- Beyond that, the response decreases steadily, reaching strong attenuation at higher frequencies.
- This confirms that the filter is frequency-selective.

2. phase response



- The phase curve is nonlinear across frequency.
- Unlike a linear-phase FIR, the slope is not constant.
- but, since this is a minimum-phase filter, the phase distortion is reduced compared to other realizations, and the group delay is minimized for this given magnitude response.

3. group delay



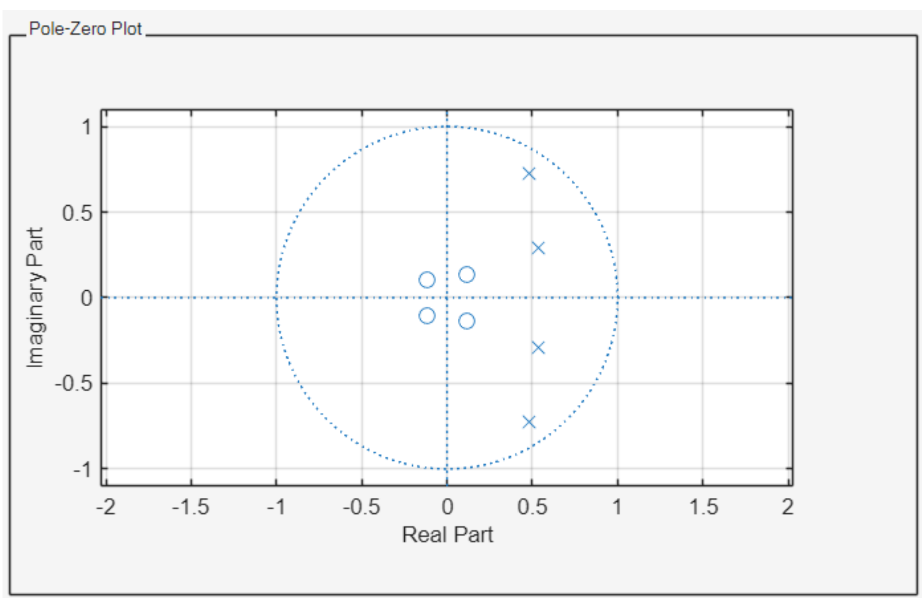
- The group delay is not flat (unlike linear-phase filters).
- It peaks around the transition band (≈0.3π rad/sample), indicating extra delay there.
- In the passband (low frequencies), the delay is relatively small, confirming the minimum-phase property (minimum possible delay for the same magnitude response).

4. difference equation

```
>> B_Min

B_Min =

   35.7143    0.0530    0.0710    0.0530    0.0280

>> A_Min

A_Min =

    1.0000   -2.0260    2.1480   -1.1590    0.2790
```

- From your coefficients (B_Min and A_Min):
- The system equation is:
  $y[n] - 2.0260\, y[n-1] + 2.1480\, y[n-2] - 1.1590\, y[n-3] + 0.2790\, y[n-4] = 35.7143\, x[n] + 0.0530\, x[n-1] + 0.0710\, x[n-2] + 0.0530\, x[n-3] + 0.0280\, x[n-4]$

5. Pole-Zero diagram

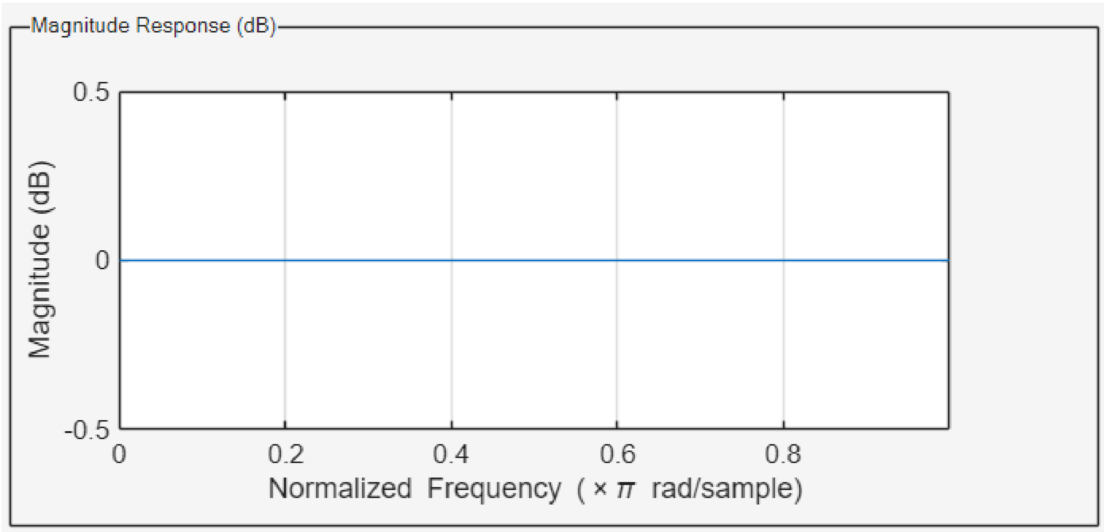


all zeros are inside the unit circle (ensuring causality & stability with minimal phase).

Part 2. An all-pass system

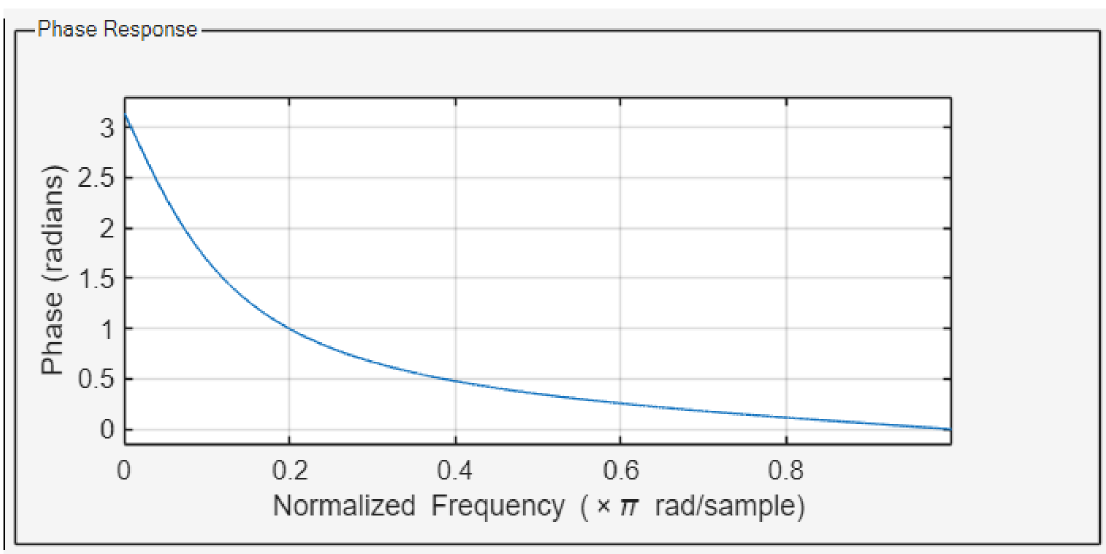
Use the tool to generate:

1. magnitude response



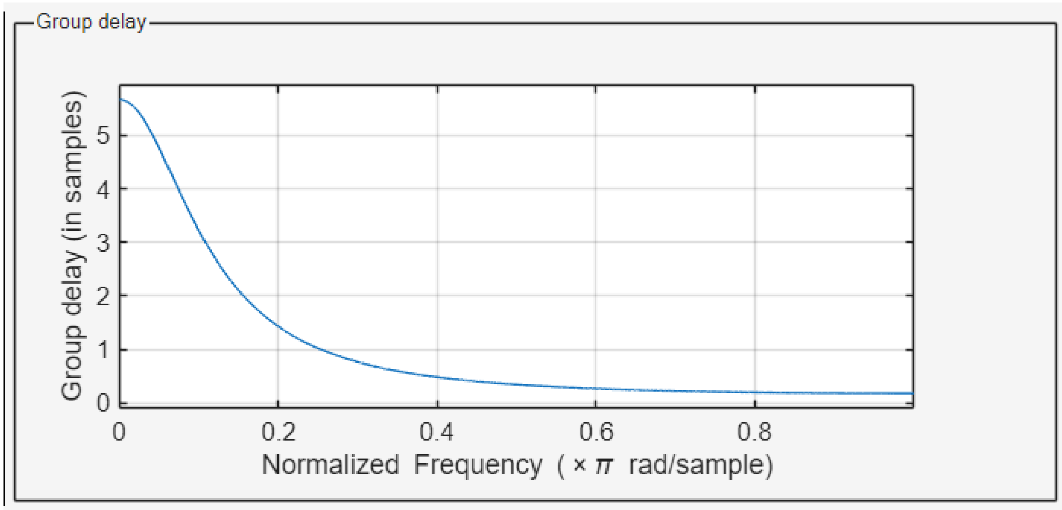
- Completely flat at 0 dB across all frequencies → confirms the all-pass property (no amplitude change).

2. phase response



- Strongly nonlinear phase shift across frequency.
- The filter only affects phase, not magnitude.

3. group delay



- The plot is not a flat horizontal line. It has a distinct peak at the transition band, meaning different frequencies are delayed by different amounts.
- This non-linear phase response is characteristic of IIR filters, which are efficient but can introduce phase distortion.
- In contrast, a linear-phase filter would have a constant group delay (a flat line), meaning all frequencies are delayed by the same amount, thus avoiding phase distortion.

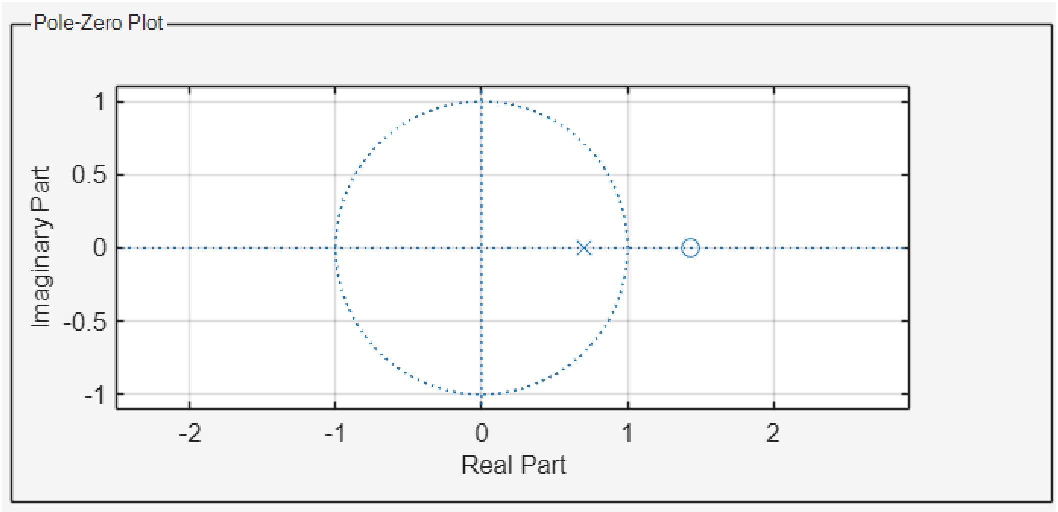
4. difference equation

```
Command Window
A_All =

    1.0000   -0.7000

>> A_All

A_All =

    1.0000   -0.7000

>> B_All

B_All =

    0.7000   -1.0000
```

The system equation is:

$$y[n] + 0.7000\, y[n-1] = 0.7000\, x[n] - 1.0000\, x[n-1]$$

5. Pole-Zero diagram



every pole inside the unit circle has a reciprocal zero outside (mirror symmetry), which enforces unity magnitude.

What do you observe about the magnitude responses?

- **Minimum-phase system:**
  The magnitude response shows a low-pass shape — it has higher gain at lower frequencies and attenuates as frequency increases. This is typical for a designed IIR filter.

- **All-pass system:**
  The magnitude response is completely flat at 0 dB across all frequencies. This is the defining property of an all-pass filter: it does not alter signal amplitude, only phase.

How do the phase responses differ?

- **Minimum-phase system:**
  The phase response is nonlinear, but it is associated with a reduced group delay compared to a linear-phase FIR of the same magnitude response. This means it still distorts phase, but with minimum possible delay for that magnitude response.
- **All-pass system:**
  The phase response is highly nonlinear and varies strongly with frequency. Even though the magnitude is flat, the filter introduces frequency-dependent phase shifts and dispersion. This causes the group delay to vary across frequencies.