Home Project №3

Due on July 3, 2009

Exercise 1

The system function $H(z)$ of a causal linear time-invariant system has the pole-zero configuration shown in Figure P5.28-1. It is also known that $H(z) = 6$ when $z = 1$.

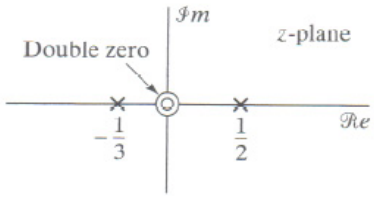

Figure P5.28-1

(a) Determine $H(z)$.
(b) Determine the impulse response $h[n]$ of the system.
(c) Determine the response of the system to the following input signals:
(i) $x[n] = u[n] - \frac{1}{2}u[n-1]$
(ii) The sequence $x[n]$ obtained from sampling the continuous-time signal

$$x(t) = 50 + 10\cos 20\pi t + 30\cos 40\pi t$$

at a sampling frequency $\Omega_s = 2\pi(40)$ rad/s

Exercise 2

Figure P5.41-1 shows two different interconnections of three systems. The impulse responses $h_1[n]$, $h_2[n]$, and $h_3[n]$ are as shown in Figure P5.41-2. Determine whether system A and/or system B is a generalized linear-phase system.

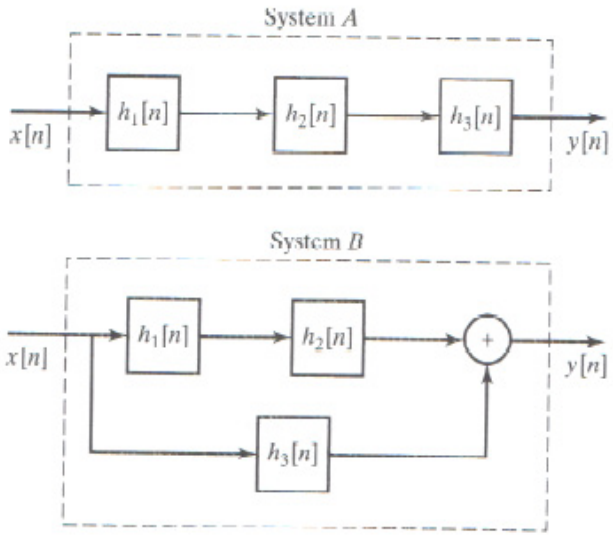

Figure P5.41-1

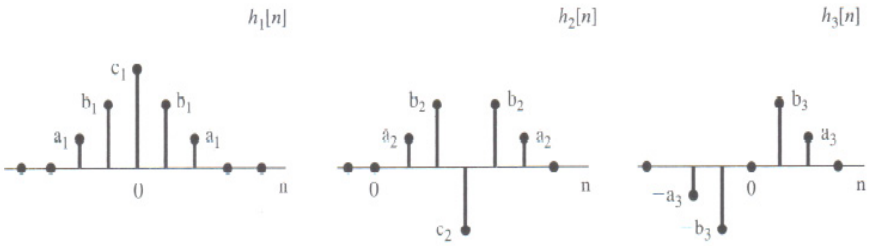

Figure P5.41-2

Exercise 3

Consider the system in Figure P6.25-1.

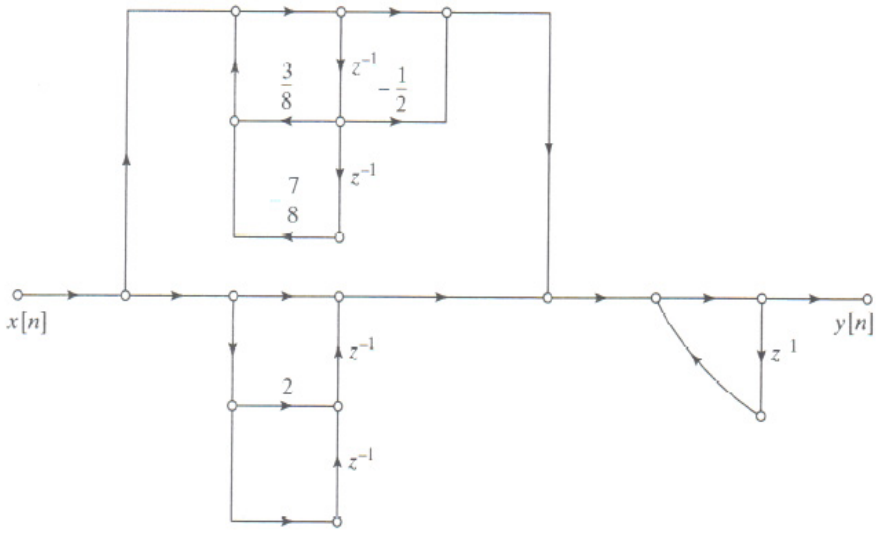

Figure P6.25-1

(a) Find the system function relating the z-transforms of the input and output.

(b) Write the difference equation that is satisfied by the input sequence $x[n]$ and the output sequence $y[n]$.

(c) Draw a signal flow graph that has the same input–output relationship as the system in Figure P6.25-1, but that has the smallest possible number of delay elements.

Exercise 4

A speech signal was recorded in a room with poor acoustics. In an attempt to recover the original signal $s[n]$ from the recorded one $r[n]$, you model the recorded signal to be the output of an LTI system $h[n]$ that will represent the room's impulse response, i.e. $r[n] = s[n] * h[n]$.
Luckily, the "filter-fairy" tells you that $H(z)$ has 12,345 poles at zero, and 12,345 zeros at $(r)^{\frac{1}{12,345}} e^{j\frac{2\pi k}{12,345}}$ $k = \{0, 1, 2, ..., 12, 344\}, 0 < r < 1$.

- Determine H(Z).
- Is it minimum phase?
- Explain why it is possible to recover $s[n]$ from $r[n]$, and describe how will you do so.
- Load the speech signal 'Einstein.wav' to your Matlab workspace. Given that the described system corrupted the speech signal, determine the value of r. Use only values from the set $\{\frac{i}{10} | i \in \{1, 2, 3..., 10\}\}$ (To filter a signal, you may use filter.m).