Ministry of Higher Education
Cairo Higher Institute
Computer and Electronic Systems Engineering Department

Time Allowed: 2 Hours
Open-Book/Notes

Digital Signal Processing Practice Exam

May 2012

Solve As Much As You Can – Maximum Grade: 40 Points

Q1. [2 Points] Determine the output of the systems described by the following difference equation with input and initial conditions as specified:

$$y[n] = x[n] - 0.5\, y[n-1], \quad x[n] = u[n-1]\,, \; y[-1] = 1$$

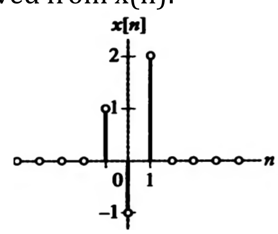
Q2. [4 Point Each] Determine whether the following signals are periodic, and for those which are, find the fundamental period:

$$x[n] = \cos\left(\tfrac{8}{15}\pi n\right)$$
$$x[n] = \cos\left(\tfrac{7}{15}\pi n\right)$$

Q3. [4 Points] For the shown a discrete $x(n)$, sketch the following signals derived from x(n):

(a) $y_1(n) = x(n+2)$

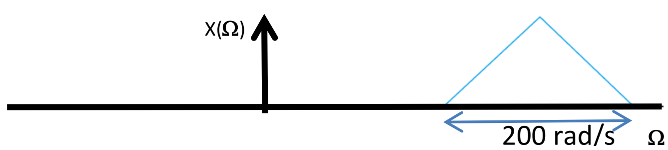
(b) $y_2(n) = x(2n)$



Q4. [4 Points] Determine if this system is (a) linear, (b) time invariant, (c) causal, and (d) recursive:

$$y(n) = x(n+1) + 2\,n$$

Q5. [3 Points] Determine the maximum sampling period that produces no aliasing for this signal:



Q6. [4 Points] A signal $f(t) = e^{-j(100\pi t)}$ was sampled with an ideal pulse train. Sketch the continuous-time Fourier transformation for sampling rate of 10k Samples/s and estimate the reconstructed signal using a lowpass filter with cutoff frequency of $\Omega_s/2$.

Q7. [2 Points] Determine the z-transform of the following signal:

$$x[n] = 2e^{-2n}u[n]$$

Q8. [2 Points] Determine the inverse z-transform of the following function:

$$F(z) = \frac{z+1}{z^2(z-1)}$$

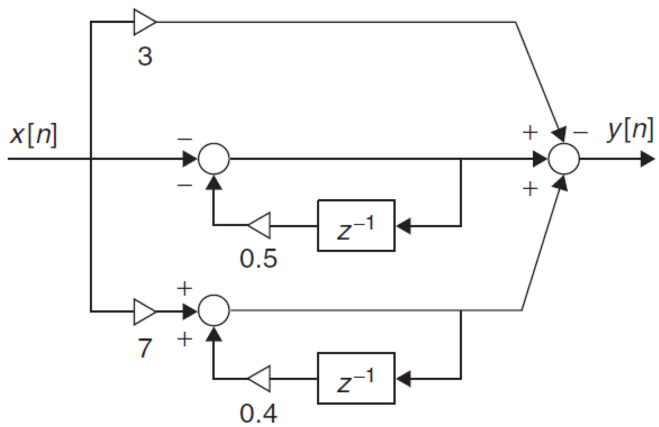
Q9. [4 Points] Convert the following filter to discrete-time filter with f_s= 1k samples/s

$$H(s) = \frac{s^2}{s^2 + \sqrt{2}s + 1}$$

Q10. [2 Points] Draw the realization of the following filter: (page 696)

$$H(z) = \frac{1 + 1.2z^{-1} + 0.2z^{-2}}{1 - 0.4z^{-1} + z^{-2} - 0.4z^{-3}}$$

Q11. [2 Points] Derive the filter transfer function X(z) of the following realization: (page 699)



Q12. (page 685)

Design a low-pass FIR filter with $N = 21$ to be used in filtering analog signals and that approximates the following ideal frequency response:

$$H_d(e^{j\omega}) = \begin{cases} 1 & 0 \leq f \leq 125 \text{ Hz} \\ 0 & \text{elsewhere in } 0 \leq f \leq f_s/2 \end{cases}$$

where $\omega = 2\pi f/f_s$ and $f_s = 1000$ Hz is the sampling rate. Use first a rectangular window, and then a Hamming window. Compare the designed filters.