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Sixth Semester B.E. Degree Examination, June/July 2016
Digital Signal Processing

Time: 3 hrs. Max. Marks:100

Note: ***Answer FIVE full questions, selecting at least TWO questions from each part.***

PART – A

1 a. Compute the N – point DFT of $x[n] = a^n$ for $0 \le n \le N-1$. Also find the DFT of the sequence $x[n] = 0.5^n u[n]$; $0 \le n \le 3$. **(07 Marks)**

b. Find the DFT of a sequence $x[n] = \begin{cases} 1 \text{ for } 0 \le n \le 3 \\ 0 \text{ otherwise} \end{cases}$

For N = 8. Plot magnitude of the DFT x(k). **(10 Marks)**

c. If $x[n] \xleftrightarrow[N]{DFT} x(k)$ then prove that DFT {x(k)} = N x(-ℓ) **(03 Marks)**

2 a. The first values of an 8 point DFT of a real value sequence is {28, -4.966j, 4+4j, -4+1.66j, -4}. Find the remaining values of the DFT. **(04 Marks)**

b. Obtain the circular convolution of $x_1[n]$ = [1, 2, 3, 4] with [1, 1, 2, 2]. **(06 Marks)**

c. A long sequence x[n] is filtered though a filter with impulse response h(n) to yield the output y[n]. if x[n] = {1, 4, 3, 0, 7, 4, -7, -7, -1, 3, 4, 3}, h(n) = {1, 2} compute y[n] using overlap add technique. Use only a 5 point circular convolution. **(10 Marks)**

3 a. Prove the symmetry and periodicity property of a twiddle factor. **(04 Marks)**

b. Develop an 8 point DIT – FFT algorithm. Draw the signal flow Graph. Determine the DFT of the sequence x[n] = {1, 1, 1, 1, 0, 0, 0, 0,} using signal flow graph. Show all the intermediate results on the signal flow graph. **(12 Marks)**

c. What is FFT algorithm? State their advantages over the direct computation of DFT. **(04 Marks)**

4 a. Find 4 point circular convolution of x[n] and h[n] using radix 2 DIF FFT algorithm x[n] = [1, 1, 1, 1] and h[n] = [1, 0, 1, 0]. **(08 Marks)**

b. Calculate the IDFT of x(k) = {0, 2.828 – j2.828, 0, 0, 0, 0, 0, 2.82 + j 2.82} using iniverse radix 2 DIT FFT algorithm. **(12 Marks)**

PART – B

5 a. The transfer function of an analog filter is given as $H_a(s) = \dfrac{1}{(s+1)(s+2)}$: obtain H(z) using impulse invariant method. Take sampling frequency of 5 samples/sec. **(05 Marks)**

b. Obtain H(z) using impulse invariance method for following analog filter $H_a(s) = \dfrac{1}{(s+0.5)(s^2+0.5s+2)}$. Assume T = 1 sec. **(10 Marks)**

c. Convert the analog filter into a digital filter whose system function is $H(s) = \dfrac{2}{(s+1)(s+3)}$ using bilinear transformation, with T = 0.1 sec. **(05 Marks)**

6 a. Design a Digital Butterworth filter using the bilinear transformation for the following specifications:

$$0.8 \le \left|H(e^{jw})\right| \le 1 \quad \text{for } 0 \le w \le 0.2\pi$$

$$\left|H(e^{jw})\right| \le 0.2 \quad \text{for } 0.6\pi \le w \le \pi$$

(12 Marks)

b. Determine the order of a Chebyshev digital low pass filter to meet the following specifications: In the passband extending from 0 to 0.25π a ripple of not more than 2dB is allowed. In the stop band extending form 0.4π to π, attenuation can be more than 40dB. Use bilinear transformation method. **(08 Marks)**

7 a. The frequency response of a filter is given by $H(e^{jw}) = jw\,;-\pi \le w \le \pi$. Design the FIR filter, using a rectangular window function. Take N = 7. **(12 Marks)**

b. The desired frequency response of the low pass FIR filter is given by

$$H_d(e^{jw}) = H_d(w) = \begin{cases} e^{-j3w}\,; & |w| < 3\pi/4 \\ 0 \quad ; & 3\pi/4 < |w| < \pi \end{cases}$$

Determine the frequency response of the FIR filter if the hamming window is used with N = 7. **(08 Marks)**

8 a. A FIR filter is given by $y[n] = x[n] + \frac{2}{5}x(n-1) + \frac{3}{4}x(n-2) + \frac{1}{3}x(n-2)$. Draw the direct and linear form realization. **(10 Marks)**

b. Obtain the direct form II and cascade realization of the following function.

$$H(z) = \frac{8z^3 - 4z^2 + 11z - 2}{(z-0.25)(z^2 - z + 0.5)}$$

(10 Marks)

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