

TXTh Semester B.E. Degree Examination, June/July 2016

Digital Signal Processing

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 $x[n] \xleftarrow{DFT} x(k)$ then prove that DFT $\{x(k)\} = N x(-\ell)$

(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)

$\underline{PART - B}$

- 5 a. The transfer function of an analog filter is given as $H_a(s) = \frac{1}{(s+1)(s+2)}$: 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 $H_a(s) = \frac{1}{(s+0.5)(s^2+0.5s+2)}$. Assume T = 1 sec. (10 Marks)
 - c. Convert the analog filter into a digital filter whose system function is $H(s) = \frac{2}{(s+1)(s+3)}$ using bilinear transformation, with T = 0.1 sec. (05 Marks)

- - b. Determine the order of a Chebyshev digital low pass filter to meet the following specifications: In the passband extending from 0 to $0.25\,\pi$ a ripple of not more than 2dB is allowed. In the stop band extending form $0.4\,\pi$ 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

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

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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