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10EC52

Fifth Semester B.E. Degree Examination, Dec.2013/Jan.2014
Digital Signal Processing

Time: 3 hrs. Max. Marks: 100

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

Important Note : 1. On completing your answers, compulsorily draw diagonal cross lines on the remaining blank pages. 2. Any revealing of identification, appeal to evaluator and /or equations written eg, 42+8 = 50, will be treated as malpractice.

PART – A

1 a. Given the following x(n) : $x(n) = \delta(n) + \delta(n-1) + \delta(n-2)$

(i) Find the Fourier transform $X(e^{jw})$ and plot the $|X(e^{jw})|$

(ii) Get the magnitude of the 4-point DFT of the first four samples of x(n)

(iii)Get the magnitude of the 8-point DFT of the first eight samples of x(n) **(10 Marks)**

b. Find the 4-point DFT of the sequence, $x(n) = 6 + \sin\frac{2\pi n}{4}$, $0 \le n \le 3$. **(06 Marks)**

c. Consider the sequence $x_1(n) = \left(\underset{\uparrow}{0}, 1, 2, 3, 4\right)$ and $x_2(n) = \left(\underset{\uparrow}{0}, 1, 0, 0, 0\right)$. Determine the sequence y(n) so that $Y(K) = X_1(K)\,X_2(K)$. $X_1(K)$ and $X_2(K)$ are 5-point DFTs of $x_1(n)$ and $x_2(n)$ respectively. **(04 Marks)**

2 a. A sequence $x(n) = \begin{cases} 1 & 0 \le n \le 3 \\ 0 & 4 \le n \le 7 \end{cases}$ has an 8-point DFT X(K). Compute the DFT of $x_2(n)$ and $x_3(n)$ in terms of X(K), for

$$x_2(n) = \begin{cases} 1 & n = 0 \\ 0 & 1 \le n \le 4 \\ 1 & 5 \le n \le 7 \end{cases}, \quad x_3(n) = \begin{cases} 0 & 0 \le n \le 1 \\ 1 & 2 \le n \le 5 \\ 0 & 6 \le n \le 7 \end{cases}$$ **(06 Marks)**

b. Consider a sequence $x(n) = (8, 3, 4, 1, -5, -4, -2, 0, 2, -1, 7, 4)$. Evaluate the following without explicitly computing X(K).

(i) DFT[DFT[DFT[DFT[x(n)]]]]

(ii) $\sum_{K=0}^{11} X(K)$

(iii) $\sum_{K=0}^{11} e^{\frac{-j\pi K}{6}} X(K)$ **(10 Marks)**

c. x(t) is an analog signal having a bandwidth of 4 kHz. It is desired to compute the spectrum of this signal using $N = 2^M$ point DFT with a resolution better than or equal to 50 Hz. Determine the minimum sampling rate and the resulting resolution (M is an integer). **(04 Marks)**

3 a. Using overlap-save method, compute y(n), of a FIR filter with impulse response h(n) = (3, 2, 1) to an input x(n) = (2, 1, -1, -2, -3, 5, 6, -1, 2, 0, 2, 1). Use only 8-point circular convolution. Can the system exhibit linear phase? **(12 Marks)**

b. Show that the product of two complex numbers (a+jb) and (c+jd) can be performed with three real multiplications and five additions. **(04 Marks)**

c. Bring out a comparison between linear convolution and circular convolution. **(04 Marks)**

4 a. Using DIFFFT algorithm, compute DFT of the sequence,
$x(n) = (1, 2, -1, 2, 4, 2, -1, 2)$
If $x_1(n)=x(n-4)_8$, compute $x_1(K)$ without invoking FFT algorithm. **(14 Marks)**

b. Compute the DFT of the sequence $x(n) = (1, 2, 1, 2)$ using the Goertzel algorithm. **(06 Marks)**

PART – B

5 a. Show that the bilinear transformation maps.
(i) The $j\Omega$ axis in s-plane onto the unit circle, $|z| = 1$.
(ii) The left half s-plane, $Re(s)<0$ inside the unit circle, $|z| < 1$ **(08 Marks)**

b. Design a digital low-pass filter using the bilinear transformation method to satisfy the following characteristics: (i) Monotonic stopband and passband (ii) -3.01 dB cut off frequency of 0.5π rad; (iii) Magnitude down atleast 15 dB at 0.75π rad. **(08 Marks)**

c. Bring out a comparison between Butterworth filter and Chebyshev filter. **(04 Marks)**

6 a. Transform $H(s) = \dfrac{s+a}{(s+a)^2 + b^2}$ into a digital filter using impulse invariance technique. **(08 Marks)**

b. Let $H(s)=\dfrac{1}{s^2 + \sqrt{2}s + 1}$, for a 2nd order low pass Butterworth filter prototype. Determine the system function for the digital bandpass filter using bilinear transformation. The cutoff frequencies for the digital filter should lie at $\omega_L = \dfrac{5\pi}{12}$ and $\omega_u = \dfrac{7\pi}{12}$. Take T = 2. **(08 Marks)**

c. What does linear phase do to the response of an input signal within the passband of the filter? Why choose an IIR filter instead of an FIR filter? **(04 Marks)**

7 Find the unit sample response of a symmetric FIR filter having a length of 9 samples. The desired frequency response is given by, $H_\alpha(\omega) = \begin{cases} e^{-j\omega\alpha} & |\omega| \le \omega_C \\ 0 & |\omega| \ge \omega_C \end{cases}$, where $\omega_C = \dfrac{\pi}{2}$ and

$\omega_{Hanning}(n) = \dfrac{1}{2}\left[1 - \cos\left(\dfrac{2\pi n}{N-1}\right)\right]$, $0 \le n \le (N-1)$. Also find H(z), linear constant coefficient difference equation and the frequency response $H(e^{j\omega})$. Draw the structure of the filter. **(20 Marks)**

8 a. Obtain the series and parallel form realization for a digital filter described by the system function,

$$H(z) = \frac{8z^3 - 4z^2 + 11z - 2}{\left(z - \frac{1}{4}\right)\left(z^2 - z + \frac{1}{2}\right)}.$$ **(14 Marks)**

b. Determine the parameters K_m of the lattice filter corresponding to the FIR filter described by,
$H(z) = 1 + 2.82z^{-1} + 3.408z^{-2} + 1.74z^{-3}$ **(06 Marks)**

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