USN | | | | | | | | | | 10TE52

Fifth Semester B.E. Degree Examination, Dec.2018/Jan.2019
Digital Signal Processing

Time: 3 hrs. Max. Marks:100

Note: 1. ***Answer any FIVE full questions, selecting at least TWO questions from each part***
2. ***Tables not permitted.***

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. Derive DFT from DTFT expression. **(05 Marks)**
b. Find the 4 point DFT's of the two sequences x(n) and y(n) using a single 4 point DFT, x(n) = [1, 2,0, 1] and y(n) = [2, 2, 1, 1]. **(10 Marks)**
c. For DFT pair shown, compute the values of boxed quantities using appropriate properties.

$\boxed{x_0}, 3, -4, 0, 2 \xleftrightarrow[\text{IDFT}]{\text{DFT}} 5, X_1, -1.28 - j3.414, X_3, 8.78 - j1.4$ **(05 Marks)**

2 a. State and prove i) circular convolution property of DFT and ii) Circular time shifting property of DFT. **(10 Marks)**
b. A long sequence x(n) is filtered through an FIR filter with impulse response h(n) = [1, 2] and an input x(n) = [1, 2, −1, 2, 3, −2, −3, −1, 1, 1, 2, −1]. Use overlap and save with block length of input = 3. **(10 Marks)**

3 a. What are FFT algorithms? Explain the advantages of FFT algorithm over direct computation of DFT sequence. **(03 Marks)**
b. Derive the radix 2, DIT FFT (decimation in time) algorithm to compute 8 point DFT of a sequence and draw the complete signal flow graph. **(08 Marks)**
c. Find the circular convolution of x(n) and h(n) using DFT-IDFT method. Apply radix 2 DIF FFT : x(n) = [1, 2, 1, 0] and h(n) = [1, 2, 3, 0]. **(09 Marks)**

4 a. A designer has a number of 8 point FFT chips. Show how he should interconnect 3 such chips in order to compute 24 point DFT. **(04 Marks)**
b. Explain Goertzel algorithm and draw DF – II structure for same. **(08 Marks)**
c. Write a brief note on chirp – z transform and its applications. **(08 Marks)**

PART – B

5 a. Starting from filter specifications, derive an expression for the order of Chebyshev filter. **(08 Marks)**
b. Find the poles of the polynomial of order 5. Obtain the transfer function $H_5(s)$ for a normalized Butterworth Low pass filter. **(08 Marks)**
c. Explains Analog – Analog frequency transformation. **(04 Marks)**

6 a. Explain how an analog filter is mapped onto digital filter using impulse invariance method. What are its limitations? **(07 Marks)**
b. Design and realize a digital law pass filter using bilinear transformation method to satisfy the following characteristics:
i) Monotonic in stop band and pass band
ii) -3db cut off at 0.5π rad and magnitude down atleast 15db at 0.75π rad. **(10 Marks)**

c. Using backward difference method of mapping, convert an analog filter with system function $H(S)=\frac{1}{S+2}$ into a digital filter. **(03 Marks)**

7 a. A filter is to be designed with the following desired frequence response

$$H_d(w) = 0 \quad -\pi/4 < w < \pi/4$$
$$= e^{-j2w} \quad \pi/4 < w < \pi$$

Find filter coefficients h(n) if window function is defined as

$$w(n) = 1 \quad 0 \le n \le 4$$
$$= 0 \quad \text{otherwise}$$
(08 Marks)

b. Derive the frequency sampling structure of an FIR filter. **(08 Marks)**

c. What are the conditions for location of zeros of linear phase FIR filter? **(04 Marks)**

8 a. Determine the coefficients K_m of lattice filter corresponding to FIR filter described by system function $H(z)=1+2z^{-1}+\frac{1}{3}z^{-2}$. Draw corresponding II order lattice structure. **(06 Marks)**

b. Realize the linear phase FIR filter having the following impulse response $h(n)=\delta(n)+\frac{1}{4}\delta(n-1)-\frac{1}{8}\delta(n-2)+\frac{1}{4}\delta(n-3)+\delta(n-4)$ **(06 Marks)**

c. Obtain a cascade realization for a system described by

$$H(z)=\frac{1+\frac{1}{4}z^{-1}}{\left(1+\frac{1}{2}z^{-1}\right)\left(1+\frac{1}{2}z^{-1}+\frac{1}{4}z^{-2}\right)}$$ using DF – II structure. **(08 Marks)**

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