CBCS SCHEME

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15EE63

Sixth Semester B.E. Degree Examination, Feb./Mar. 2022
Digital Signal Processing

Time: 3 hrs. Max. Marks: 80

Note: ***Answer any FIVE full questions, choosing ONE full question from each module.***

Module-1

1 a. Compute 4-point DFT of an input sequence : $x(n) = \cos\left(\frac{n\pi}{4}\right)$ and plot its magnitude and phase angle. **(06 Marks)**

b. Obtain the linear convolution of the sequences x(n) = {1, 2, –1, 2, 3, –2, –3, –1, 1, 1, 2, –1} and h(n) = {1, 2}, using overlap save method with 4 –point circular convolution. **(10 Marks)**

OR

2 a. Find the circular convolution of input sequence x(n) = {1, 2, 3, 4} with impulse response h(n) = {2, 1, 2, 1} using Stockham's method. **(07 Marks)**

b. State and prove circular time shifting property of Discrete Fourier transform. **(03 Marks)**

c. X(k) is a 14-point DFT of the sequence x(n), the first 8-samples of X(k) are given as : X(0) = 12; X(1) = (–1 + j3); X(2) = (3 + j4); X(3) = (1 – j5) X(4) = (–2 + j2); X(5) = (6 + j3) X(6) = (2 – j3); X(7) = 10. Compute the remaining samples of X(k)and find the value of $\sum_{n=0}^{13} |x(n)|^2$. **(06 Marks)**

Module-2

3 a. Show that FFT is computationally efficient than direct computation of DFT. **(04 Marks)**

b. What are the similarities and differences between DIT and DIF algorithms of FFT? **(04 Marks)**

c. Compute 8-point DFT of the sequence x(n) = {1, 1, 1, 1} using Radix – 2 DIT – FFT algorithm. **(08 Marks)**

OR

4 a. If $x_1(n)$ = {1, 2, 0, 1} and $x_2(n)$ = {1, 3, 3, 1}, obtain the circular convolution of $x_1(n)$ and $x_2(n)$ using Radix – 2 DIT – FFT algorithm. **(08 Marks)**

b. If DFT X(k) is given as :
$X(k) = \{0, 2\sqrt{2}(1-j), 0, 0, 0, 0, 2\sqrt{2}(1+j)\}$, determine the corresponding time sequence x(n) and draw the signal flow graph with all intermediate results, using inverse Radix – 2 DIF – FFT algorithm. **(08 Marks)**

Module-3

5 a. The system function of an along filter is given as : $H_a(s) = \frac{s+0.1}{(s+0.1)^2+9}$. Obtain the system function of the IIR digital filter using impulse invariance method. **(06 Marks)**

b. Design an analog filter with maximally flat response in the passband and an acceptable attenuation of –2dB at 20 radians/sec. The attenuation in the stopband should be more than 10dB beyond 30 radians/sec. **(10 Marks)**

OR

6 a. Design a lowpass 1rad/sec bandwidth Chebyshev filter with an acceptable passband ripple of 2dB, cut-off frequency of 1 rad/sec and stopband attenuation of 20dB or greater beyond 1.3 rad/sec. **(10 Marks)**

b. Use Bilinear transformation to design a fist order lowpass Butterworth filter that has a 3dB cut-off frequency at $W_c = 0.2\pi$. The normalized filter is given as :

$$H_{an}(s) = \frac{1}{s+1}.$$ **(06 Marks)**

Module-4

7 a. Design a digital lowpass filter using Chebyshev filter design procedure that meets the following specifications : passband magnitude characteristics that is constant to 1dB for frequencies below $w = 0.2\pi$ and stopband attenuation of at least 15dB for frequencies between $w = 0.3\pi$ and π. Use Bilinear transformation. **(10 Marks)**

b. Obtain the general realization of parallel form for an IIR system. **(06 Marks)**

OR

8 a. Design a Chebyshev filter for the following specifications using impulse invariance method.

$$0.8 \le |H(e^{j\omega})| \le 1 \quad \text{for} \quad 0 \le \omega \le 0.2\pi$$
$$|H(e^{j\omega})| \le 0.2 \quad \text{for} \quad 0.6\pi \le \omega \le \pi$$ **(12 Marks)**

b. A difference equation describing a filter is given as :

$$y(n) - \frac{3}{4}y(n-1) + \frac{1}{8}y(n-2) = x(n) \frac{1}{2}x(n-1)$$

Draw Direct form – I and Direct form – II structures for an IIR system. **(04 Marks)**

Module-5

9 a. Design a normalized linear phase FIR filter having the phase delay of $\tau = 4$ and atleast 40dB attenuation in the stopband. Also obtain the magnitude/frequency response of the filter. **(12 Marks)**

b. Obtain the cascade realization of the system function :

$$H(z) = 1 + \frac{5}{2}z^{-1} + 2z^{-2} + 2z^{-3}.$$ **(04 Marks)**

OR

10 a. Design a low pass FIR filter using frequency sampling technique having cut-off frequency of $\frac{\pi}{2}$ rad/sample. The filter should have a linear phase with length M = 17. **(12 Marks)**

b. Determine the direct form realization of the system function :

$$H(z) = 1 + 2z^{-1} - 3z^{-2} - 4z^{-3} + 5z^{-4}.$$ **(04 Marks)**

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