CBCS SCHEME

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15EE63

Sixth Semester B.E. Degree Examination, Aug./Sept.2020

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. Determine IDFT using DFT is $X(k) = (\underset{\uparrow}{7}, -2-j, 1, -2+j)$ **(05 Marks)**

b. State and prove symmetry property for a real valued sequence. **(05 Marks)**

c. For the sequence x(n) = (4, 3, 2, 1), determine the 6-point DFT of the sequence x(n). **(06 Marks)**

OR

2 a. Given $x_1(n) = \cos\left(\frac{2n\pi}{N}\right)$ and $x_2(n) = \sin\left(\frac{2n\pi}{N}\right)$ for $0 \le n \le N-1$, calculate N point circular convolution of $x_1(n)$ and $x_2(n)$. **(08 Marks)**

b. If h(n) = (1, 1, 1) and $x(n) = (\underset{\uparrow}{1}, 2, 0, -3, 4, 2, -1, 1, -2, 3, 2, 1)$ determine the convolution of x(n) and h(n). Use overlap save method and consider 5 samples in each partition of x(n). **(08 Marks)**

Module-2

3 a. Find the 8-point DFT of the sequence $x(n) = (\underset{\uparrow}{1}, 2, 3, 4, 4, 3, 2, 1)$ using decimation in time FFT method. List all the stage calculations in a table. **(08 Marks)**

b. Calculate the 4-point circular convolution of x(n) and h(n) using radix-2 decimation in frequency-FFT method. Given x(n) = (1, 1, 1, 1) and h(n) = (1, 0, 1, 0). **(08 Marks)**

OR

4 a. Explain the algorithm of decimation in time-FFT. Assume length of x(n) = 8. **(08 Marks)**

b. Calculate the 8-point DFT of x(n) where x(n) = (1, 2, 1, 0, 0, 0, 0, 0). Use decimation in frequency method. Show the results of each stage in a table. **(08 Marks)**

Module-3

5 a. Explain the theory of Bilinear Transformation (BT) and also explain frequency warping introduced by BT. **(10 Marks)**

b. Let $H_a(s) = \frac{s+a}{(s+a)^2+b^2}$ be a causal second order function. Show that H(z) is given by,

$$H(z) = \frac{1-e^{-at}\cos bT.z^{-1}}{1-2\cos bTe^{-aT}z^{-1}+e^{-2aT}z^{-1}}$$, if impulse invariance method is used. **(06 Marks)**

OR

6 a. A Butterworth lowpass filter has $K_P = -1$ dB at $\Omega_P = 4$ rad/sec,

b. $K_S = -20$ dB at $\Omega_P = 8$ rad/sec, calculate $H_a(s)$ of Butterworth filter for above specifications. **(10 Marks)**

c. State merits and demerits of IIR filters. **(06 Marks)**

Module-4

7 a. Design a digital Chebyshev-I filter that satisfies :

$0.8 \le |H(w)| \le 1$ for $0 \le w \le 0.2\pi$

and $|H(w)| \le 0.2$ for $0.6\pi \le w \le \pi$

Use impulse invariant transformation and assume T = 1 second. **(12 Marks)**

b. $H(z) = \dfrac{1}{1 - \frac{1}{16}z^{-2}}$, for this function draw the cascade form structure. **(04 Marks)**

OR

8 a. A digital low pass filter has:

$20 \log|H(w)|_{w = 0.2\pi} \ge -1.9328$ dB

and $20 \log|H(w)|_{w = 0.6\pi} \le -13.9794$ dB

The filter must have maximally flat frequency response. Find H(z) for above specification. Use impulse Givariance method. Assume T = 1 second. **(10 Marks)**

b. Draw the direct form-I and direct form-II structure for $H(z) = \dfrac{2z^2 + z - 2}{z^2 - 2}$. **(06 Marks)**

Module-5

9 a. A lowpass filter has

$H_d(e^{jw}) = H_d(w) = e^{-j2w}$, for $|w| < \pi/4$

$= 0$, for $\pi/4 < |w| < \pi$

Calculate the filter coefficients $h_d(n)$ and h(n), if w(n) is a rectangular window, given by

$w(n) = 1$ for $0 \le n \le 4$

$= 0$ otherwise **(10 Marks)**

b. Compare different types of window functions based on transition width, stopband attenuation and window function. **(06 Marks)**

OR

10 a. A lowpass filter has the response

$H_{do}(w) = H_d(e^{jw}) = e^{-j3w}$ for $0 < w < \pi/2$

$= 0$ for $\pi/2 < w < \pi$

is e^{-j3w}

Calculate h(n) sing frequency sampling technique. Assume N = 7. **(10 Marks)**

b. Calculate the coefficients K_m of the lattice filter, if the FIR filter is given by :

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

Draw the II order lattice structure. **(06 Marks)**

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