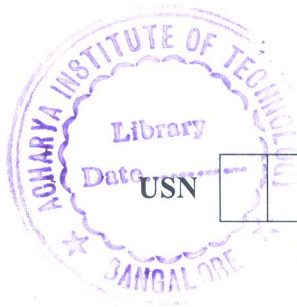


CBCS SCHEME



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 $X(k) = (7, -2 - j, 1, -2 + j)$ (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 $0 \leq n \leq N-1$, calculate N point circular convolution of $x_1(n)$ and $x_2(n)$. (08 Marks)
b. If $h(n) = (1, 1, 1)$ and $x(n) = (1, 2, 0, -3, 4, 2, -1, 1, -2, 3, 2, 1)$ 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 $x(n) = (1, 2, 3, 4, 4, 3, 2, 1)$ 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 ,
$$H(z) = \frac{1 - e^{-at} \cos bT z^{-1}}{1 - 2 \cos bT e^{-aT} z^{-1} + e^{-2aT} z^{-2}}$$
 , 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 $\Omega_s = 8$ 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 :
 $0.8 \leq |H(w)| \leq 1$ for $0 \leq w \leq 0.2\pi$
 and $|H(w)| \leq 0.2$ for $0.6\pi \leq w \leq \pi$
 Use impulse invariant transformation and assume $T = 1$ second. (12 Marks)
- b. $H(z) = \frac{1}{1 - \frac{1}{16}z^{-2}}$, for this function draw the cascade form structure. (04 Marks)

OR

- 8 a. A digital low pass filter has:
 $20 \log|H(w)|_{w=0.2\pi} \geq -1.9328$ dB
 and $20 \log|H(w)|_{w=0.6\pi} \leq -13.9794$ 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 $H(z) = \frac{2z^2 + z - 2}{z^2 - 2}$. (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 $0 \leq n \leq 4$
 $= 0$ 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
 $H_d(w) = H_d(e^{jw}) = e^{-j3w}$ 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)
