15EE63 USN

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.

Determine IDFT using DFT is X(k) = (7, -2 - j, 1, -2 + j)

(05 Marks)

State and prove symmetry property for a real valued sequence.

(05 Marks)

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

(06 Marks)

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)

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)

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

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

- Explain the theory of Bilinear Transformation (BT) and also explain frequency warping introduced by BT. (10 Marks)
 - be a causal second order function. Show that H(z) is given by ,

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

- A Butterworth lowpass filter has $K_P = -1$ dB at $\Omega_P = 4$ rad/sec,
 - $K_S = -20$ dB at $\Omega_P = 8$ rad/sec, calculate $H_a(s)$ of Butterworth filter for above specifications. (10 Marks)
 - State merits and demerits of IIR filters.

(06 Marks)

Module-4

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)

-, for this function draw the cascade form structure

(04 Marks)

OR

A digital low pass filter has:

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

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

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

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

A lowpass filter has

 $H_d(e^{j\hat{w}}) = H_d(w) = e^{-j2w}$, 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

$$w(n) = 1 \quad \text{for} \quad 0 \le n \le 4$$

= 0 otherwise

(10 Marks)

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

A lowpass filter has the response

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

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

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

(10 Marks)

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)