CBCS SCHEME

USN

15EE63

Sixth Semester B.E. Degree Examination, Jan./Feb.2021
Digital Signal Processing

Time: 3 hrs. Max. Marks: 80

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

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.

Module-1

1 a. Find 8 point DFT of $x[n]=[1, 2, 2, 1]$ using direct computation, plot magnitude and phase response. **(10 Marks)**

b. State and prove convolution property of DFT. **(06 Marks)**

OR

2 a. Using Stockham's method, find circular convolution of the sequences, $x(n)=\delta(n)+3\delta(n-1)+2\delta(n-2)+4\delta(n-3)$ and $h(n)=n$ for $0 \le n \le 3$. **(08 Marks)**

b. Determine the response of an LTI system with $h(n)=[1, 2]$ for an input sequence, $x(n)=[1, 2, -1, 2, 3, -2, -3, -1, 1]$. Employ overlap and add method. Use 4 point circular convolution. **(08 Marks)**

Module-2

3 a. Compute 8 point DFT of the sequence $x[n]=[1, 2, 3, 4, 4, 3, 2, 1]$ using DIT-FFT algorithm. **(08 Marks)**

b. Develop 8 point DIT-FFT radix-2 algorithm and explain basic butterfly. **(08 Marks)**

OR

4 a. Find IDFT of the sequence,
$X[K]=[14,\ -1+j0.414,\ -j2,\ -1-j2.414,\ -2,\ -1-j2.414,\ j2,\ -1-j0.414]$
Using inverse DIF-FFT algorithm. **(10 Marks)**

b. What is in place computation? What is the total number of complex additions and multiplications required for N = 512 points, if DFT is computed directly and if FFT is used. **(06 Marks)**

Module-3

5 a. Design an analog filter with flat response in the passband and an acceptable attenuation of -2db at 20 radians/second. The attenuation in the stopband should be more than 10 db beyond 30 radians/sec. **(10 Marks)**

b. Derive an expression for order and cutoff frequency of the butterworth filter. **(06 Marks)**

OR

6 a. The system function of an analog filter is given as, $H_a(s)=\dfrac{1}{(s+1)(s+2)}$. Obtain H(z) using impulse invariant transformation. Take sampling frequency of 5 samples/sec. **(06 Marks)**

b. Explain bilinear method of transforming an analog filter into digital filter. Also show the mapping from s to z plane. **(06 Marks)**

c. Starting from a lowpass butterworth prototype analog filter, design butterworth bandpass analog filter with upper and lower band edge frequencies 10 rad/sec and 5 rad/sec. **(04 Marks)**

Module-4

7 a. Bring out the comparison between Butterworth and Chebyshev filter. **(04 Marks)**

b. The specifications of a lowpass filter are given as,

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

$|H(w)| \le 0.2$ for $0.32\pi \le w \le \pi$

Design the Chebyshev filter using bilinear transforms. **(12 Marks)**

OR

8 a. What is prewarping? Why is it required? **(04 Marks)**

b. Obtain the direct form I, direct form II cascade and parallel form realization for the following system:

$y(n) = 0.75y(n-1) - 0.125y(n-2) + 6x(n) + 7x(n-1) + x(n-2)$. **(12 Marks)**

Module-5

9 a. The desired frequency response of the low pass filter is given by,

$$H_d(e^{jw}) = H_d(w) = \begin{cases} e^{-j3w}; & |w| < \dfrac{3\pi}{4} \\ 0; & \dfrac{3\pi}{4} < |w| < \pi \end{cases}$$

Determine the frequency response of the FIR filter if the hamming window is used with N = 7. **(10 Marks)**

b. Explain the design of an FIR filter based on frequency sampling approach. **(06 Marks)**

OR

10 a. Write the analytical equations and draw the magnitude response characteristics of any four different windows used in design of FIR filter. **(08 Marks)**

b. Realize the following system function in direct and cascade form:

$H(z) = 1 + \frac{3}{4}z^{-1} + \frac{17}{8}z^{-2} + \frac{3}{4}z^{-3} + z^{-4}$. **(08 Marks)**

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