CBCS SCHEME

USN | | | | | | | | | | | **15EE63**

Sixth Semester B.E. Degree Examination, June/July 2018
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 the N-point DFT of the signal
$x(n) = a^n$; $0 \le n \le N-1$ **(04 Marks)**

b. Using formula to find DFT, compute 4-point DFT of causal signal given by ,
$x(n) = \frac{1}{3}$; $0 \le n \le 2$
$= 0$; elsewhere
Also sketch the magnitude and phase spectra. **(08 Marks)**

c. Consider a length – 12 sequence defined for $0 \le n \le 11$; $x(n) = (8, 4, 7, -1, 2, 0, -2, -4, -5, 1, 4, 3)$ with 12-point DFT given by X(k) ; $0 \le k \le 11$. Evaluate the following function without computing the DFT $\sum_{k=0}^{11} e^{\frac{-j4\pi}{6}K} X(k)$. **(04 Marks)**

OR

2 a. A discrete time LTI system has impulse response $h(n) = 2\delta(n) - \delta(n-1)$. Determine the output of the system if the input is $x(n) = \delta(n) + 3\delta(n-1) + 2\delta(n-2) - \delta(n-3) + \delta(n-4)$ using circular convolution by circular array method. Verify the result using formula based method. **(08 Marks)**

b. Find the output y(n) of a filter whose impulse response is given by h(n) = (3, 2, 1, 1) and input signal is given by x(n) = (1, 2, 3, 3, 2, 1, –1, –2, –3, 5, 6, –1, 2, 0, 2, 1) using Overlap – Add method. Use 7-point circular convolution in your approach. **(08 Marks)**

Module-2

3 a. An 8-point sequence is given by
x(n) = (2, 2, 2, 2, 1, 1, 1, 1).
Compute its DFT by a Radix-2 DIT-FFT algorithm. **(08 Marks)**

b. Derive the algorithm for N = 8 and write the complete signal flow graph. **(08 Marks)**

OR

4 a. The first 5-points of the 8-point DFT of a real valued sequence is given by X(0) = 4, X(1) = 1 – j2.414, X(2) = 0, X(3) = 1 – j0.414 and X(4) = 0. Write the remaining points and hence find the sequence x(n) using inverse radix-2 DIT-FFT algorithm. **(08 Marks)**

b. If $x_1(n) = (1, 2, 0, 1)$ and $x_2(n) = (1, 3, 3, 1)$, obtain $x_1(n)$ Ⓝ $x_2(n)$ by using DIF-FFT algorithm. **(08 Marks)**

Module-3

5 a. Convert the following second order analog filter with system transfer function $H(s) = \dfrac{b}{(s+a)^2 + b^2}$ into a digital filter with infinite impulse response by the use of impulse invariance mapping technique. Also find H(z) if $H_a(s) = \dfrac{1}{s^2 + 2s + 2}$ **(08 Marks)**

b. Explain bilinear transformation method of converting analog filter into digital filter. Show the mapping from s-plane to z-plane. Also obtain the relation between ω and Ω. **(08 Marks)**

OR

6 a. A digital lowpass filter is required to meet the following specifications :
(i) Monotonic pass band and stop band (ii) –3.01 dB cutoff frequency of 0.5π rad
(iii) Stopband attenuation of atleast 15 dB at 0.75π rad. Find the system function H(z). Use bilinear transformation technique. **(08 Marks)**

b. Design a second order bandpass digital Butterworth filter with passband of 200 Hz to 300 Hz and sampling frequency of 2000 Hz using bilinear transformation method. **(08 Marks)**

Module-4

7 a. Design a digital Chebyshev type-I filter that satisfies the following constraints:
$0.8 \le |H(w)| \le 1 \quad ; \quad 0 \le w \le 0.2\pi$
$|H(w)| \le 0.2 \quad ; \quad 0.6\pi \le w \le \pi$
Use impulse invariant transformation. **(08 Marks)**

b. Design a high pass filter H(z) to be used to meets the specifications shown in Fig.Q7(b) below. The sampling rate is fixed at 1000 samples/sec. Use bilinear transformation.

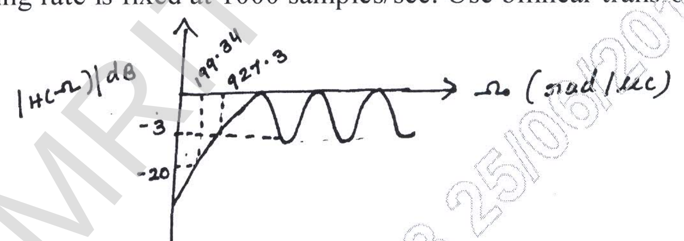

Fig.Q7(b) **(08 Marks)**

OR

8 a. Obtain the direct form-I and direct form-II structure for the system given by

$$H(z) = \frac{z^{-1} - 3z^{-2}}{(10 - z^{-1})(1 + 0.5z^{-1} + 0.5z^{-2})}$$ **(08 Marks)**

b. Draw the cascade form structure for the system given by

$$H(z) = \frac{\left(1 - \frac{1}{2}z^{-1}\right)}{\left(1 - \frac{1}{4}z^{-1} + \frac{1}{2}z^{-2}\right)\left(1 - \frac{1}{5}z^{-1} + \frac{1}{6}z^{-2}\right)}$$ **(04 Marks)**

c. A digital system is given by $H(z) = \dfrac{1 - \frac{1}{2}z^{-1}}{\left(1 - \frac{1}{3}z^{-1}\right)\left(1 - \frac{1}{4}z^{-1}\right)}$

Obtain the parallel form structure. **(04 Marks)**

Module-5

9 a. Explain why windows are necessary in FIR filter design. What are the different windows used in practice? Explain in brief. **(08 Marks)**

b. The desired frequency response of a lowpass filter is given by

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

Determine the coefficients of impulse response and also determine the frequency response of the FIR filter if Hamming window is used with N = 7. **(08 Marks)**

OR

10 a. Design a normalized linear phase FIR filter having the phase delay of $\tau = 4$ and atleast 40 dB attenuation in the stopband. Also obtain the magnitude/frequency response of the filter **(08 Marks)**

b. Realize the system function given by $H(z) = 1 + \frac{5}{2}z^{-1} + 2z^{-2} + 2z^{-3}$ in direct form. **(04 Marks)**

c. Realize the digital filter with system function given by,

$H(z) = 1 + \frac{1}{2}z^{-1} + \frac{1}{3}z^{-2} + \frac{1}{7}z^{-3} + \frac{1}{3}z^{-4} + \frac{1}{2}z^{-5} + z^{-6}$ in linear phase form. **(04 Marks)**

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