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.

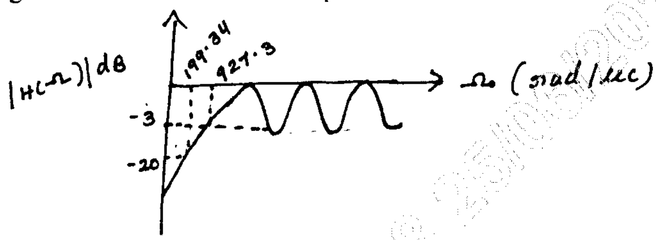

Fig.Q7(b) **(08 Marks)**

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

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

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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