CBCS SCHEME

USN | | | | | | | | | | 17EC52

Fifth Semester B.E. Degree Examination, July/August 2021
Digital Signal Processing

Time: 3 hrs. Max. Marks: 100

Note: Answer any FIVE full questions.

1 a. Explain the frequency domain sampling and reconstruction of discrete time signals. **(09 Marks)**

b. Determine the circular convolution of the sequences $x_1(n) = \{1, 2, 3, 1\}$ and $x_2(n) = \{4, 3, 2, 2\}$ using the time domain formula. **(05 Marks)**

c. Compute the N-point DFT of the signal $x(n) = \cos\frac{2\pi}{N}k_0 n$, $0 \le n \le N-1$ **(06 Marks)**

2 a. Establish the relationship between:
i) DFT and Fourier Transform
ii) DFT and Fourier series coefficients. **(08 Marks)**

b. Show that the multiplication of two DFT's leads to circular convolution of respective time sequences. **(07 Marks)**

c. The first three samples of 4-point DFT of a real sequence x(n) is $X(k) = \{2, 1+j, 0\}$. Find the remaining sample and also determine the sequence x(n). **(05 Marks)**

3 a. State and prove Parseval's theorem. Express the energy of the sequence interms of DFT. **(06 Marks)**

b. x(k) denote the 6-point DFT of the sequence $x(n) = \{1, 2, -1, 3, 0, 0\}$ without computing the IDFT, determine the sequence y(n)if
i) $y(k) = W_3^{2k} x(k)$
ii) $y(k) = X((k-2))_6$ **(06 Marks)**

c. Using overlap save method, compute the output y(n) of an FIR filter with impulse response $h(n) = \{1, 2, 3\}$ and input $x(n) = \{2, -3, 1, 0, -2, -1, 3, 5\}$. Use 6-point circular convolution. **(08 Marks)**

4 a. State and prove the property of circular time shift of a sequence. **(06 Marks)**

b. The 5-point DFT of a complex valued sequence x(n) is given by $X(k) = \{1 + j, 2 + j2, j, 2-j2, 1-j\}$. Compute y(k) if i) $y(n) = x^{+}(n)$ ii) $y(n) = x((-n))_N$ **(06 Marks)**

c. Find the response of an LTI system with an impulse response $h(n) = \{1, -1, 2\}$ for the input $x(n) = \{3, 2, -1, 1, 4, 5, -2, -3\}$, using overlap add method. Use n-point circular convolution with the input data block segment length L = 4. **(08 Marks)**

5 a. Compute the 8-point DFT of the sequence $x(n) = \{2, 2, 2, -1, -1, -1, -2, 1\}$ using decimation in time-FFT algorithm. **(08 Marks)**

b. Find the number of complex additions and multiplications required for 256-point DFT computation using i) Direct method ii) FFT method. What is the speed improvement factor? **(05 Marks)**

c. Explain the Goertzel algorithm and obtain the direct form-II realization. **(07 Marks)**

6 a. Given $x(n) = n + 1$, $0 \le n \le 7$, find the 8-point DFT of x(n) using radix-2 decimation in frequency FFT algorithm **(08 Marks)**

b. Perform the 4-point circular convolution of the sequences
$x_1(n) = (2\ \ 1\ \ {-1}\ \ 2\}$ and $x_2(n) = \{1, 2, 3, -1\}$ using decimation in time FFT algorithm. **(07 Marks)**

c. What is chirp-z transform? Draw the contours on which Z-transform is evaluated. **(05 Marks)**

7 a. Obtain the direct form-II and cascade realization of the system function

$$H(z) = \frac{2(1-z^{-1})(1+\sqrt{2}z^{-1}+z^{-2})}{(1+0.5z^{-1})(1-0.9z^{-1}+0.81z^{-2})}$$ **(07 Marks)**

b. Determine the order for a digital Butterworth filter design using bilinear transformation to meet the following specifications.
   i) Passband ripple of 3dB at 1000Hz
   ii) Stopband ripple of 20dB at 2000Hz
   iii) Sampling frequency of 10kHz
   iv) Indicate the steps to obtain the digital system function H(z). **(09 Marks)**

c. Describe the frequency transformations from low pass filter to any other types in the analog domain. **(04 Marks)**

8 a. Obtain the parallel realization for the system function

$$H(z) = \frac{\left(1+\frac{1}{4}z^{-1}\right)}{\left(1+\frac{1}{2}z^{-1}\right)\left(1+\frac{1}{2}z^{-1}+\frac{1}{4}z^{-2}\right)}$$ **(06 Marks)**

b. An IIR digital lowpass filter is required to meet the following specifications:
Passband ripple ≤ 0.5dB
Passband edge = 1.2kHz
Stopband attenuation ≥ 40dB
Stopband edge = 2kHz
Sampling rate = 8kHz
Determine the filter order for
   i) A digital Butterworth filter
   ii) A digital Chebyshev filter, which uses bilinear transformation. **(09 Marks)**

c. An ideal analog integrator system function $H_a(s) = 1/s$. Obtain the digital integrator system function H(z) using bilinear transformation. Write the difference equation for the digital integrator. Assume T = 2. **(05 Marks)**

9 a. Consider an FIR filter with system function $H(z) = 1 + 2.88z^{-1} + 3.4z^{-2} + 1.74z^{-3} + 0.4z^{-4}$. Obtain the lattice filter coefficients. Sketch the direct form and lattice realization. **(10 Marks)**

b. An FIR filter is to be designed with the following desired frequency response:

$$H_d(w) = \begin{cases} e^{-j4w}, & |w| < \frac{\pi}{4} \\ 0, & \frac{\pi}{4} < |w| < \pi \end{cases}$$

Find the frequency response H(w) of the filter using Hamming window function. **(10 Marks)**

10 a. Determine a direct form realization for the linear phase FIR filter impulse response $h(n) = \{1, 2, 3, 4, 3, 2, 1\}$. **(04 Marks)**

b. Consider an FIR lattice filter with coefficients $K_1 = 0.65$, $K_2 = -0.34$ and $K_3 = 0.8$.
   i) Find its impulse response by tracing a unit impulse input through the lattice structure.
   ii) Draw the equivalent direct-form structure. **(08 Marks)**

c. Determine the impulse response of the low pass FIR filter to meet the following specifications using a suitable window function:
   Passband edge frequency = 1.5kHz
   Stopband edge frequency = 2kHz
   Minimum stopband attenuation = 50dB
   Sampling frequency = 8kHz. **(08 Marks)**

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