Ref No: **Ex/EE/5/T/414/2019**

BACHELOR OF ENGINEERING (ELECTRICAL ENGINEERING) FOURTH YEAR FIRST SEMESTER EXAM 2019

SUBJECT: - Digital Signal Processing

Full Marks 100

Time: Three hours (50 marks for each part)

Use a separate Answer-Script for each part

No. of Questions	PART I	Marks
	*Answer **any three** questions. **Two marks** are reserved for neatness.*	
1. (a)	A periodic signal $x(t)=\sin(2\pi t)$ is sampled at $f_s=4$ Hz. How can you compute the amplitude spectrum C_n using the samples in one period?	06
(b)	Show the detailed step-by-step procedure that can be employed to compute 4-point FFT of a sequence using Radix-2 decimation-in-frequency in-place FFT algorithm. Draw the complete signal flow graph.	10
2. (a)	Justify, citing suitable reasons, whether the following statements are <u>TRUE</u> or <u>FALSE</u> (**any Two**): (*i*) For a periodic signal, the real parts of complex Fourier coefficients are anti-symmetric and the imaginary parts of complex Fourier coefficients are symmetric with respect to *n*, the harmonic number. (*ii*) For a linear phase digital filter with $\omega = p\omega_s - \omega',\ 0 \le \omega' < \frac{\omega_s}{2}, p = 0,\pm1,\pm2,\cdots$ (where ω_s = sampling frequency), we have $H(\omega) = H(\omega')$. (*iii*) In computing an *N*-point Fast Fourier Transform, the number of complex additions and the number of complex multiplications involved are same and each varies linearly with the logarithm of number of iterations required to carry out the FFT.	04×2 =08
(b)	How can you apply FFT algorithm to perform digital filtering of a finite real data sequence?	08

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No. of Questions	PART I	Marks
3. (a)	The gain of a 7-tap causal linear-phase FIR digital filter is given as: $$\begin{aligned} \lvert H(\omega)\rvert &= 1.1, for \frac{\omega_s}{8} \le \omega \le \frac{\omega_s}{4} \\ &= 0.9, for \frac{\omega_s}{4} \le \omega \le \frac{\omega_s}{2} \\ &= 1.1, for -\frac{\omega_s}{4} \le \omega \le -\frac{\omega_s}{8} \\ &= 0.9, for -\frac{\omega_s}{2} \le \omega \le -\frac{\omega_s}{4} \\ &= 0, \ otherwise \end{aligned}$$ where each symbol has its usual meaning. The sampling frequency is chosen as 2 kHz. The design employs Hann window for smoothing filter coefficients. Determine the filter coefficients. Draw the schematic realization of the filter.	10
(b)	What is Gibbs phenomenon? How can its adverse effects be reduced?	06
4. (a)	How can FIR digital filters be employed for offline analysis of two-dimensional data? How can two-dimensional convolution summation be utilized in this regard?	08
(b)	Prove that, in case of a digital FIR filter, with a real and symmetric h_n, a real frequency response can be achieved with zero phase shift.	08
5.	Write short notes on ***any two*** of the following:	08×2 =16
(i)	Inverse discrete Fourier transform.	
(ii)	Frequency response of Blackman Window.	
(iii)	Effect of truncation of impulse response of FIR digital filter.	

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No. of Questions	PART- II Answer any *THREE* questions Two marks reserved for neatness	Marks
1. (a)	Write a short note on frequency spectra of uniformly sampled signals.	(10)
(b)	Consider the analog signal $x(t) = 3\cos 100\pi t$ (a) Determine the minimum sampling rate required to avoid aliasing. (b) Suppose that the signal is sampled at the rate Fs = 200 Hz. What is the discrete-time signal obtained after sampling? (c) Suppose that the signal is sampled at the rate Fs =75 Hz. What is the discrete time signal obtained after sampling?	(6)
2. (a)	Starting from the definition of Z-transform, determine the expressions for the Z-transforms and the corresponding ROCs of the following sequences. (i) Unit step sequence. (ii) Causal sinusoidal sequence.	(8)
(b)	Find the final value of the following theorem using Final Value Theorem of Z-transform: $F(z) = \dfrac{0.79z^2}{(z-1)(z^2-0.416z+0.208)}$	(2)
(c)	Find the inverse Z-transform of the following signal: $X(z) = \dfrac{z^2}{(z-1)(z-0.2)}$; ROC: $\lvert z\rvert > 1$	(6)
3.	If $X(z) = \dfrac{-1-z^{-1}}{1-5z^{-1}+6z^{-2}}$, find $Z^{-1}[X(z)]$ when (i) ROC of X(z) is $\lvert z\rvert > 3$ (ii) ROC of X(z) is $2 < \lvert z\rvert < 3$ (iii) ROC of X(z) is $\lvert z\rvert < 2$	(16)
4. (a)	Derive the transfer function of discrete time differentiator and discrete time integrator.	(6)
(b)	Using impulse invariant transformation, design a digital filter corresponding to the analog filter with transfer function $G(s) = 10 / (s^2 + 6s + 5)$ Write down the difference equation relating the output and the input	(10)

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	sequences of the filter. Consider a sampling frequency of 20 Hz.	
5.	Write short notes on the following: **(any two)** (i) Region of Convergence of Z transform. (ii) Uniform Sampling of a continuous time signal can be represented as Impulse Modulation. (iii) Designing digital filters by bilinear transformation. (iv) Mapping of entire left half of s-plane on to z-plane.	(8×2=16)