Ref No: Ex/EE/T/311/2019

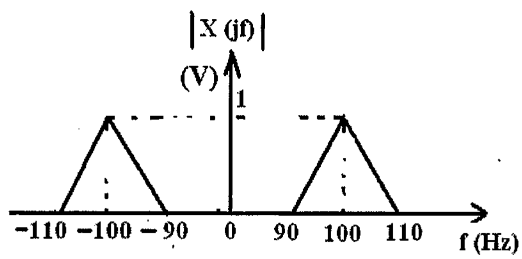
B.E. ELECTRICAL ENGINEERING THIRD YEAR FIRST SEMESTER - 2019

DIGITAL SIGNAL PROCESSING

Full Marks 100

Time: Three hours (50 marks for each part)

Use a separate Answer-Script for each part

No. of Question	PART- I	Marks
	Answer any THREE questions **Two marks reserved for neatness and well organized answers**	
1.(a)	The amplitude spectrum of a continuous-time signal *x(t)* is shown in Fig. [A]. When the signal is sampled at rate of 210 Hz, give a well labeled sketch of the two-sided amplitude spectrum of the sampled signal for $\|f\| < 120$ Hz, with relevant explanations in brief.  Fig. [A]	6
(b)	State the "Sampling Theorem", and hence comment in brief on the nature and the use of an analog low-pass filter prior to the analog-to-digital converter (ADC) in digital signal processing system.	5
(c)	Consider the signal $y(t) = x(t) \times Sinc(190t)$, where *t* is in second, and *x(t)* is the signal described in Question 1 (a). What is the Nyquist rate for sampling *y(t)*? Give relevant precise explanations.	5
	OR	
	Starting from the definition of z-transform, obtain the closed form expression for the z-transform *X(z)* of the sequence $x[n] = a^n Cos(200n) u[n]$, and its region of convergence (ROC). Identify and comment on the location of the pole(s) of *X(z)* and the nature of variation of *x[n]* with time.	5

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No. of Questions	PART I	Marks
2. (a)	**Use your knowledge of the Z-transforms of standard sequences and the properties of Z-transform, to verify whether or not the Z-transform of the sequence** $f[n] = \lvert n \rvert \left(\frac{1}{2}\right)^{\lvert n \rvert}$ **is** $F(z) = \dfrac{\frac{5}{8}z + \frac{5}{8}z^{-1} - 1}{\left(1-\frac{1}{2}z^{-1}\right)^2\left(1-\frac{1}{2}z\right)^2}$. **What is the ROC of the Z-transform of** ***f[n]*** **?**	5
(b)	**Which of the following functions of** z **could be the Z-transform of a causal sequence?** ***Explain without actually determining the inverse transforms.*** **(i)** $X(z) = \dfrac{d}{dz}\dfrac{\left(1-2z^{-1}\right)^2}{\left(3-2z^{-1}\right)^2}$ **(ii)** $X(z) = \dfrac{(z-1)^3}{\left(z^{-1}-\frac{1}{4}\right)^2}$	5
	OR	
	The impulse response of a causal discrete-time LTI system, is $g_n = \{1,1,1\}$ with ↑ under the first element **Performing time-domain operation, find the expression for the output sequence** y_n **when the system is excited by a sequence** $x_n = \{1,-1,2,-1\}$.	5

PART- I

(c) **The Z-transfer function of a discrete-time linear time-invariant system has a pair of zeros at the origin and another zero at $z = -0.8$. There is a pair of poles at $z = -0.1 \pm j\,0.9$ and another pole at $z = 0.7$. The high frequency gain of the system is 6. Sketch the pole zero plot. Indicate all possible ROCs of the transfer function, and in each case comment on the causality and the stability of the system.** **6**

3. (a) **A particular structure for realizing an IIR filter is shown in Fig. [B]. Name the structure. Obtain the set of difference equations describing the filter algorithm. Derive the Z-transfer function of the filter. Also determine and draw the Direct form-II and the parallel realizations (using 1st order subsystems) of the filter. *Give all the relevant sets of difference equations.*** **10**

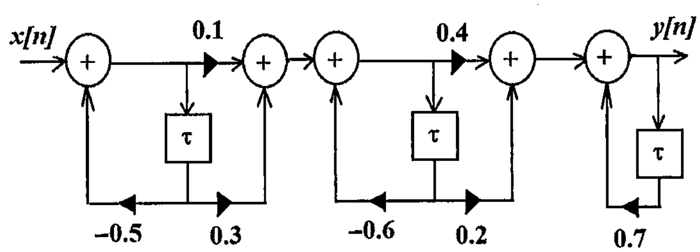

Fig. [B]

(b) **Consider a system consisting of two DTLTI systems in cascade, with frequency responses** **6**

$$H_1\left(e^{j\Omega}\right) = \frac{2 - Cos\Omega + jSin\Omega}{1 + \frac{1}{2}Cos\Omega - \frac{j}{2}Sin\Omega} \quad \&$$

$$H_2\left(e^{j\Omega}\right) = \frac{1}{1 + \frac{1}{2}Cos\Omega + \frac{1}{4}Cos2\Omega - \frac{j}{2}Sin\Omega - \frac{j}{4}Sin2\Omega}$$

where, Ω is in radian. Find the difference equation relating the output and the input sequences of the combination.

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	PART- I	
4. (a)	**Derive the difference equation relating the output and the input sequences of a digital highpass Butterworth filter with the following specifications:** • **A high-frequency gain of 0 dB.** • **The desired -3 dB cutoff frequency is 0.3π radian. Over 0 to 0.15π radian, the attenuation should be at least 11dB.** **Use bilinear transformation preceded by frequency prewarping. Derive the frequency warping relation used.**	**7+2**
(b)	**Examine the correctness of the statement---- '*A DAC introduces a time-delay of half the sampling period*'. Explore the type of analog signal processing required after the digital-to-analog converter (DAC) in a digital signal processing system, to minimize the distortions introduced by the converter. Give relevant derivation and sketch in support of your answer. Which of the remedial measures can be best attained by introducing a digital signal processing algorithm prior to the D/A conversion.**	**7**
5.	**Write short notes on <u>*any two*</u> of the following.** **(a) FIR and IIR discrete-time LTI systems.** **(b) Representing uniformly sampled signals as trains of scaled impulses.** **(c) Regions of Convergence (ROC) of Z-transforms and their properties.** **(d) Problem of characteristic-aliasing associated with the designing of digital filters by discretizing analog filter transfer functions.**	**8+8**

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B. E. ELECTRICAL ENGINEERING 3RD YEAR 1ST SEMESTER EXAMINATION, 2019

SUBJECT: - DIGITAL SIGNAL PROCESSING

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Full Marks 100
(50 marks for each part)

Use a separate Answer-Script for each part

No. of Questions	PART II	Marks
	Answer all the questions.	
1.	Derive in detail how can inverse discrete Fourier transform for an N-point discrete sequence be computed. OR Derive in detail how can an N-point DFT be split into two (N/2)-point DFTs, where N is a power of 2.	10
2. (a)	Determine the frequency response of the following non-causal window function: $w_n = 0.54 + 0.46\cos\left(\frac{2\pi n}{M-1}\right), \quad \text{for } \lvert n \rvert \le \left(\frac{M-1}{2}\right)$ $= 0$, otherwise OR Prove that, for practical realization of a digital filter, even when the output is calculated using a finite number of terms of impulse sequence of the filter, the frequency response of the filter remains a real quantity with zero phase shift.	10
(b)	In terms of memory organization, how does modified Harvard architecture differ from Harvard architecture? What are the salient features of the TMS320C25 digital signal processor? OR How can multiply/accumulate operation be carried out in TMS320C25 digital signal processor? What are the well-known benchmarks employed for TMS320C25 processor?	06
3.	Write a short note on ***any one*** of the following:	08
(i)	Contrast enhancement in images by histogram equalization.	
(ii)	Application of FIR digital filters in offline situations.	

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No. of Questions	PART II	Marks
4.	Design an M-tap causal FIR digital filter with stepped characteristic in its frequency response given as: For $\frac{-\omega_s}{2} \le \omega \le \frac{\omega_s}{2}$, $H(\omega) = ae^{-j\omega\tau(M-1)/2}$, for $\lvert\omega\rvert \le \omega_1$ $= be^{-j\omega\tau(M-1)/2}$, for $\omega_1 < \lvert\omega\rvert \le \omega_2$ $= 0$, otherwise where $a = 1$, $b = 0.9$, $M = 7$, $f_1 = 200$ Hz, $f_2 = 400$ Hz, $f_s = 1$ kHz and every other symbol has its usual meaning. Draw the schematic realization of the filter.	08
5.	Justify or correct ***<u>any two</u>*** of the following statements with suitable reasons/derivations, in brief.	04×02 =08
i)	A *Radix-2 Decimation-in-Frequency FFT algorithm* is also called an *In-Place algorithm.*	
ii)	The summation of the coefficient values of a High pass FIR image filter mask must always be zero.	
iii)	For a linear phase digital filter, the real part of the frequency response will be antisymmetric and the imaginary part of the frequency response will be symmetric about the line of symmetry at $\omega = \omega_s/2$.	