

Ref No: Ex/EE/T/322/2018(Old)

**B.E. ELECTRICAL ENGINEERING 3<sup>RD</sup> YEAR 2<sup>ND</sup> SEMESTER**  
**EXAMINATION, 2018 (OLD)**

**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
	<p><b>Answer any <i>THREE</i> questions</b>  <b>Two marks reserved for neatness</b></p>	
1. (a)	<p>The impulse response of a causal discrete-time LTI system, is</p> $h_n = \{1, 1, 1\}$ <p style="text-align: center;">↑</p> <p>Find the output sequence <math>y_n</math>, when the system is excited by a sequence</p> $x_n = \{1, -1, 2, -1\}$ <p style="text-align: center;">↑</p>	8
(b)	<p>Determine the inverse Z-transform of</p> $X(z) = \frac{5z^2 - 3z}{(z^2 - 4z + 3)}, \text{ for all possible ROCs of } X(z).$	8
2. (a)	<p>Explain, with the help of relevant illustrations, how the left-half of the s-plane maps on to the z-plane.</p>	8
(b)	<p>The output <math>y_n</math> and the input <math>x_n</math> of a discrete-time linear time-invariant system are related by the following difference equation.</p> $0.5y_n + 0.05y_{n-1} - 0.36y_{n-2} = 0.35x_n + 0.126x_{n-2}$	8

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No. of Questions	PART I	Marks
	<p>Derive and draw the following structures for realizing the system.</p> <p>(i) Direct Form- I (ii) Direct Form- II</p>	
3. (a)	<p>For discretisation of analog filters using backward difference transformation, show with the help of necessary mathematical derivation, how the <math>j\omega</math> axis in the s-plane maps on to the z-plane.</p>	6
(b)	<p>Using bilinear transformation with frequency prewarping, design a digital filter corresponding to the analog filter with transfer function</p> $H(s) = \frac{1}{(s+4)(s+1)}$ <p>Consider a sampling frequency of 50 Hz.</p>	10
4. (a)	<p>Consider the bandlimited analog signal</p>	10
	<p><math>x(t) = 2\text{Sin}(200\pi t) + 3\text{Sin}(600\pi t)</math>, where t is in seconds.</p>	
	<p>Find the Nyquist sampling rate for the above signal. If the above signal is sampled 125 times per second, and then the resulting sampled signal <math>X_n</math> is converted back to an analog signal using an ideal lowpass filter with a cutoff frequency of 62.5 Hz, obtain the values of the frequencies in the reconstructed analog signal.</p>	
(b)	<p>Determine the Z-transforms and their ROCs for the following sequences.</p>	
	<p><math>x[n] = A\text{Sin}(\omega_0 n\tau) u[n]</math></p>	3+3
	<p><math>g[n] = n^2 e^{-an\tau} u[n]</math></p>	

<b>PART- I</b>		
<b>5.</b>	<b>Write short notes on any two of the following.</b> <b>(a) Designing digital filters by impulse-invariant transformation.</b> <b>(b) Modeling uniform sampling by ‘Pulse Amplitude Modulation’.</b> <b>(c) Recursive and non-recursive filters.</b> <b>(d) Causality and stability of DTLTI systems.</b>	

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**B.E. ELECTRICAL ENGINEERING THIRD YEAR  
SECOND SEMESTER EXAM 2018 (Old)**

**SUBJECT: - DIGITAL SIGNAL PROCESSING**

Time: Three hours

Full Marks 100  
(50 marks for each part)

Use a separate Answer-Script for each part

No. of Questions	PART II	Marks
	<i>Answer any three questions. TWO marks are reserved for neat and well organized answers.</i>	
1. (a)	Describe in detail how can an $(N/2)$ -point DFT be divided into two even and odd harmonic $(N/4)$ -point DFTs?	10
(b)	Prove that, for FIR filters in offline operations, the length of the output sequence is always smaller than the length of the input sequence.	06
2. (a)	How can you compute 4-point FFT of a discrete sequence using Radix-2 decimation-in-frequency in-place FFT algorithm? Draw the corresponding signal flow graph.	10
(b)	Describe the non-causal and causal forms of Bartlett, Hamming and Blackman windows employed for designing FIR filters. Hence comment on the approximate widths of main lobes of their frequency responses.	06
3. (a)	A 7-tap causal linear-phase FIR brick-wall type low-pass filter has been designed with unity pass band gain and a cut off frequency of 200 Hz. The sampling frequency has been chosen as 1000 Hz. The design employed Raised Cosine window for smoothing filter coefficients. Determine the filter coefficients. Also draw the schematic realization of the filter.	10
(b)	"Both FIR low pass and high pass image filters should comprise only positive coefficients." - Justify or correct this statement, citing suitable reasons.	06
4. (a)	Give a detail account of the processor architecture of TMS320C25. How is multiply/accumulate operation carried out in this processor?	08
(b)	Differentiate between group delay and phase delay of a distortion-less filter. What is Gibbs phenomenon?	05+03
5.	Write short notes on <i>any two</i> of the following:	08×2 =16
(i)	Inverse discrete Fourier transform.	
(ii)	Periodicity and symmetry properties of linear phase digital filters.	
(iii)	Histogram equalization for contrast enhancement in images.	