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Reg No.:_____ Name:_____

APJ ABDUL KALAM TECHNOLOGICAL UNIVERSITY

Fifth Semester B.Tech Degree (S,FE) Examination January 2022 (2015 Scheme)

Course Code: EC301

Course Name: DIGITAL SIGNAL PROCESSING

Max. Marks: 100 Duration: 3 Hours

PART A

Answer any two full questions, each carries 15 marks.

			Marks
1	a)	Write down the equation for N point DFT and explain each term	(4)
	b)	Find N point DFT of unit impulse $\delta(n)$	(4)
	c)	Find 4 point DFT of x(n) = [1, -2, 3, 2].	(7)
2	a)	Find 4 point circular convolution of x(n) = h(n) = [1,1,1,1] using DFT.	(8)
	b)	Compute 8-point DFT of the sequence [1 1 1 1 0 0 0 0] using Decimation in Time FFT algorithm	(7)
3	a)	Let x(n) = {1,0,1,0} and h(n) = {1,2,2,1}. Find 4 point DFTs of these sequences using a single 4 point DFT.	(8)
	b)	Find the IDFT of X(k) = [1,0,1,0] using DIF FFT algorithm.	(7)

PART B

Answer any two full questions, each carries 15 marks.

4	a)	Prove that a symmetric impulse response results in a linear phase response for an FIR filter with an even filter order N.	(8)
	b)	Compare main 3 properties of rectangular, Hanning and Hamming window functions.	(3)
	c)	How filter order is selected in the window-based method of FIR filter design.	(4)
5	a)	Explain impulse invariant mapping. List the main limitations of impulse invariant mapping.	(5)
	b)	Design a Digital Butterworth filter to satisfy the constraints $0.9 \leq \lvert H(\omega)\rvert \leq 1; 0 \leq \omega \leq 0.5\pi$. $\lvert H(\omega)\rvert \leq 0.2; 0.75\pi \leq \omega \leq \pi$. Use bilinear transformation. Assume T=1 s.	(10)
6	a)	Design a Linear phase LPF with a cut off frequency of 0.5π rad/s using frequency sampling. Take N = 13, use type 1 design.	(10)
	b)	Derive the mapping function from s to z in Bilinear transformation. Explain frequency warping.	(5)

PART C

Answer any two full questions, each carries 20 marks.

7 a) Realise the system given by $H(z) = \frac{3+3.6z^{-1}+0.6\,z^{-2}}{1+0.1z^{-1}-0.2z^{-2}}$ in cascade form and parallel form. (10)

b) Explain transposed form of filter structure. Realise the system $y(n) = b_0x(n) + b_1x(n-1) + b_2x(n-2) - a_1y(n-1) - a_2y(n-2)$ in transposed form. (10)

8 a) Explain decimation and interpolation. (6)

b) Explain effects of coefficient quantisation in FIR and IIR filters. (7)

c) Why lowpass filtering is performed before down sampling? How will you decide the cutoff frequency of the low pass filter? (7)

9 a) Explain the architecture of TMS320C67XX DSP processor with a neat diagram. (10)

b) Realise the transfer function $H(z) = \frac{1+2z^{-1}}{1-1.5z^{-1}+0.9z^{-2}}$ in Direct form 1 and 2. Compare the number of delay elements, adders and multipliers needed in both cases. (10)

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