Reg No.:_______________ Name:__________________________

APJ ABDUL KALAM TECHNOLOGICAL UNIVERSITY

SEVENTH SEMESTER B.TECH DEGREE EXAMINATION(R&S), MARCH 2020

Course Code: EE407

Course Name: DIGITAL SIGNAL PROCESSING

Max. Marks: 100 Duration: 3 Hours

PART A

Answer all questions, each carries 5 marks.

			Marks
1		How circular and linear convolutions are performed using DFT?	(5)
2		State and explain flow graph reversal theorem	(5)
3		What do you mean by frequency warping? An ideal discrete time high pass filter with cut off frequency π/2 was designed using bilinear transformation with T=1 sec. What is the cut off frequency of the continuous time ideal high pass filter?	(5)
4		What is a linear phase filter? What conditions are to be satisfied by an FIR flter in order to have linear phase?	(5)
5		What is the effect of quantization of filter coefficients indigital filters?	(5)
6		What do you mean by dead band of a filter? How will you compute the dead band of the system $y(n) = 0.8y(n-1) + x(n)$?	(5)
7		What are the elements in the control unit of TMS 320 C24x DSP Processor?	(5)
8		Which are the TREG, PREG and Multiply Instructionsof TMS 320 C24x DSP Processor?	(5)

PART B

Answer any two full questions, each carries 10 marks.

9		Given x(n)={1,2,3,4,4,3,2,1}, Compute 8-point DFT of x(n) using DIT FFT algorithm	(10)
10	a)	Find the output y(n) of a filter whose impulse response is h(n)={2,2} and the input signal to the filter is x(n)={-2,0,2,1,3,1,-1,-2} using overlap-save method	(5)
	b)	Realize the system function $H(z) = 1 + \frac{3}{4}z^{-1} + \frac{17}{8}z^{-2} + \frac{3}{4}z^{-3} + z^{-4}$ using minimum number of multipliers.	(5)
11		Consider the discrete time, linear, causal system described by the difference equation $y(n) - \frac{3}{4}y(n-1) + \frac{1}{8}y(n-2) = x(n) + \frac{1}{3}x(n-1)$. Obtain the direct form I and Direct form II realizations of the system	(10)

PART C

Answer any two full questions, each carries 10 marks.

12 Design a digital Butterworth filter using bilinear transformation which satisfies the following conditions: (10)

$$0.75 \le |H(e^{j\omega})| \le 1, \quad 0 \le \omega \le 0.2\pi$$

$$|H(e^{j\omega})| \le 0.2, \quad 0.5\pi \le \omega \le \pi$$

Take sampling time T=1sec

13 a) Can we physically realise an ideal filter? Justify your answer (2)

b) What is impulse invariant transformation? What are its disadvantages? (3)

c) Which are the desirable characteristics of windows? (2)

d) What is Gibb's phenomenon? (3)

14 Design an ideal low pass filter , whose desired frequency response given by (10)

$$H_d(e^{j\omega}) = \begin{cases} 1 & for \quad {}^{-\pi}/_3 \le \omega \le {}^{\pi}/_3 \\ 0 & for \quad {}^{\pi}/_3 \le |\omega| < \pi \end{cases}$$

Determine the filter coefficients, using Hanning window. Determine the impulse response and H(z). Given N=9.

PART D

Answer any two full questions, each carries 10 marks.

15 a) Draw the quantization noise model of the first order system function $H(z) = \frac{1}{1 - 0.4z^{-1}}$. Products are rounded to 4 bits including sign bit. Find the steady state noise power due to product round off. (6)

b) Explain signal scaling in digital filters. (4)

16 a) What are the possible errors due to truncation, in sign magnitude and two's complement representations if the system uses b+1 bits including sign bit for the number representation? (5)

b) Explain the bus structure of TMS 320 C24x DSP Processor (5)

17 a) Explain in detail indirect addressing mode of TMS 320 C24x DSP Processor with examples (5)

b) Discuss about any five Accumulator, arithmetic, and logic instructions of TMS 320 C24x DSP Processor with examples (5)

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