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Reg No.:_____ Name:_____

APJ ABDUL KALAM TECHNOLOGICAL UNIVERSITY

FIFTH SEMESTER B.TECH DEGREE EXAMINATION, DECEMBER 2017

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) Explain, how DFT and IDFT can be expressed as Linear Transformation (3)

b) Derive the relationship of DFT to Z-transform (3)

c) Find the circular convolution of x[n] = {1, 2, -1, 3, 4} and h[n] = {2, -1, 4, 1, 3} (5)

d) Explain overlap add method for filtering of long data sequences. (4)

2 a) Show that, if x[n] is a real and even sequence, then its DFT X[k] is also real and even (3)

b) Find linear convolution of x[n] = {2, 3, -1} and h[n] = {1, -1, 2}, using circular convolution. (5)

c) Find the number of complex multiplications involved in the calculation of a 1024 point DFT using (i) direct computation(ii) radix-2 FFT algorithm (3)

d) Explain, how N point DFTs of two real-valued sequences can be found by computing a single N point DFT. (4)

3 a) Find 8 point DFT of x[n] = {2, 1, -1, 3, 5, 2, 4, 1} using radix-2 decimation in time FFT algorithm (11)

b) Explain, how a 2N point DFT of a 2N point real-valued sequence can be found by computing a single N point DFT. (4)

PART B

Answer any two full questions, each carries 15 marks.

4 a) Prove that, if z_1 is a zero of a linear phase FIR filter, then $1/z_1$ is also a zero. (5)

b) Design a linear phase FIR low pass filter having length M = 15 and cut-off frequency $\omega_c = \pi/6$. Use Hamming window. (10)

5 a) Explain the design of linear phase FIR filters by the frequency sampling method. (9)

b) Explain the frequency transformations in the analog domain (6)

6 Design a digital Butterworth low pass filter with $\omega_p = \pi/6$, $\omega_s = \pi/4$, minimum pass band gain = -2dB and minimum stop band attenuation = 8dB. Use bilinear (15)

transformation. (Take T = 1)

PART C

Answer any two full questions, each carries 20 marks.

7 a) Find the lattice structure implementation of FIR filter h[n] = {1, 0.5, 0.75, -0.6} (6)

b) Draw the direct form II structure and transposed direct form II structure of (5)

$$H(z) = \frac{1+0.5z^{-1}-0.75z^{-2}}{1+0.6z^{-1}+0.4z^{-2}-0.2z^{-3}}$$

c) Draw the block diagram of TMS320C67XX and briefly explain the function of each block. (9)

8 a) Draw the direct form realization of linear phase FIR filter (5)

h[n] = {1, 0.5, 0.25, -0.5, 0.8, -0.5, 0.25, 0.5, 1} using minimum multipliers.

b) Draw the signal flow graphs ofdirect form II and cascade form structures of $H(z) = \frac{(0.8+0.2z^{-1}+0.6z^{-2})(1-0.6z^{-1})}{(1-0.6z^{-1}+0.8z^{-2})(1+0.8z^{-1}-0.7z^{-2})}$ (5)

c) Explain the effects of coefficient quantization in IIR and FIR filters. (10)

9 a) Give the output of decimation by M system in time domain. Explain output frequency spectrum. What is the importance of low pass filtering prior to down-sampling? (10)

b) How does a floating-point number represented in a processor? Explain the operations of addition and multiplication of two floating point numbers with examples. (5)

c) Derive the variance of quantization noise in ADC with step size Δ. (Assume quantization noise has uniform distributed pdf with zero mean) (5)

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