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APJ ABDUL KALAM TECHNOLOGICAL UNIVERSITY
FIFTH SEMESTER B.TECH DEGREE EXAMINATION, DECEMBER 2018

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) Given $x(n) = \{1, -2, 3, -4, 5, -6\}$ without calculating DFT find the following quantities? (5)
- a) $X(0)$ b) $\sum_{K=0}^5 X(K)$ c) $X(3)$ d) $\sum_{K=0}^5 |X(K)|^2$ e) $\sum_{K=0}^5 -1^K X(K)$
- b) Find the convolution of $x(n) = \{1, 2, 3, 4, 5\}$ and $h(n) = \{1, 1, 1\}$ using overlap save method? (5)
- c) State Circular frequency shift property of DFT? (5)
- 4 –point DFT of the signal $x(n) = \{a, b, c, d\}$ is $X(K)$. Find the IDFT of $X(K-2)$?
- 2 a) Find the number of complex multiplications and additions involved in the calculation of 1024 DFT using direct computation and radix2 FFT algorithm? (4)
- b) How will you obtain linear convolution from circular convolution? For $x(n) = \{1, 2, 3\}$ and $h(n) = \{-1, -2\}$, obtain linear convolution $x(n)*h(n)$ using circular convolution? (5)
- c) Given $g(n) = \{1, 0, 1, 0\}$ and $h(n) = \{1, 2, 2, 1\}$ find the 4 point DFTs of these sequences using a single 4 point DFT. (6)
- 3 a) Describe the steps involved in radix 2 DIT FFT algorithm (5)
- b) Find the DFT of the sequence $\{1, 2, 3, 4, 4, 3, 2, 1\}$ using DIT algorithm (7)
- c) What do you mean by in place computation of DFT? (3)

PART B

Answer any two full questions, each carries 15 marks.

- 4 a) Explain the significance of linear phase FIR filter and comment on its impulse response? (4)
- b) Design an ideal lowpass filter with frequency response (6)
- $H(e^{j\omega}) = 1$ for $-0.5\pi \leq \omega \leq 0.5\pi$ and $H(e^{j\omega}) = 0$ for $0.5\pi \leq |\omega| \leq \pi$.
Find $h(n)$ for $N = 11$. (use rectangular window)
- c) Determine the frequency response of FIR filter defined by (5)
- $y(n) = 0.25x(n) + x(n-1) + 0.25x(n-2)$. Calculate the phase delay and group delay?

- 5 a) Convert the analog filter $H(s)$ given below in to a second order Butterworth digital filter using impulse invariance technique. (6)

$$H(s) = \frac{1}{s^2 + \sqrt{2}s + 1}$$

- b) Why can't we use impulse invariance technique for implementing digital highpass filter? (4)
- c) Describe the steps involved in the design of digital Butterworth bandpass filter? (5)
- 6 a) Derive the equation for cutoff frequency in Butterworth filter? (5)
- b) Apply bilinear transformation to $H(s) = \frac{2}{(s+1)(s+2)}$ with $T = 1$ sec and find $H(z)$? (5)
- c) What is warping effect in bilinear transformation method and how can we eliminate it? (5)

PART C

Answer any two full questions, each carries 20 marks.

- 7 a) Draw the block diagram of TMS320C67XX and explain functions of each block? (10)
- b) Realize the system function using minimum number of multipliers (5)
 $H(z) = (1 + z^{-1})(1 + 0.5z^{-1} + 0.5z^{-2} + z^{-3})$
- c) Obtain the transposed directform II structure for the system (5)
 $y(n) = 0.5y(n-1) - 0.25y(n-2) + x(n) + x(n-1)$
- 8 a) Realize the system given by difference equation $y(n) = -0.1y(n-1) + 0.2y(n-2) + 3x(n) + 3.6x(n-1) + 0.6x(n-2)$ in cascade form? (6)
- b) Obtain the parallel form realization for above system (6)
- c) Find the lattice structure implementation of FIR filter $h(n) = \{1, 13/24, 5/8, 1/3\}$ (8)
- 9 a) Explain the effect of coefficient quantization in IIR and FIR filters? (10)
- b) If quantization noise has uniform distribution with zero mean, find the quantization noise in ADC with step size Δ ? (5)
- c) A signal $x(n)$ is obtained by sampling analog signal $x(t)$ at twice the Nyquist rate. If we wish to down sample $x(n)$ by a factor 4, obtain the bandwidth of the decimation filter required for suppressing aliasing distortion. (5)
