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UNIVERSITY OF PUNE

[4363]-185

T. E. (E & TC) Examination May 2013

Digital Signal Processing

(2008 Course) (Sem-I)

Total No. of Questions : 12 **[Total No. of Printed Pages :4]**

[Time : 3 Hours] **[Max. Marks : 100]**

Instructions :

(1) Answer 3 question from section-I and 3 question from section-II

*(2) Answers to the **two sections** should be written in **separate -books**.*

(3) Neat diagram must be drawn wherever necessary.

(4) Black figures to the right indicate full marks.

(5) Your answers will be valued as whole.

(6) Use of logarithmic tables, slide rule, Mollier charts, electronic pocket calculator and steam tables is allowed.

(7) Assume suitable data, if necessary.

SECTION-I

Q1.

a) Find the convolution of following sequences

1) $x(n) = 2^n$ and $h(n) = (1/2)^n u(n)$

2) $x(n) = n + 2$ for $0 \le n \le 3$ and $h(n) = a^n u(n)$ [8]

b) Define stability. Explain the condition for system to be stable in terms of impulse Response. Test the stability of the system whose impulse response is

$h(n) = (1/2)^n u(n)$

$h(n) = (3)^{-n} u(n)$ [10]

OR

Q2.

a) Determine the direct form-I and II realization for the following system. Show all Steps properly. [10]

$y(n) = -0.1y(n-1) + 0.2y(n-2) + 3x(n) + 3.6x(n-1) + 0.6x(n-2)$

$y(n) = 0.251y(n-1) + 0.05y(n-2) + x(n) - 2x(n-2)$

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b) Explain the basic elements of DSP system. Explain the advantages of DSP over analog system [8]

Q3.

a) Find out the relation between DTFS and DFT. Find n=4 point DFT for $x(n) = \{1,0,1,0\}$ [8]

b) Find the circular convolution for following sequences using graphical method [8]

1) $x1(n) = \delta(n) + \delta(n-1) + \delta(n-2)$
   $x2(n) = 2\delta(n) - \delta(n-1) + 2\delta(n-2)$
2) $x1(n) = \delta(n) + \delta(n-1) - \delta(n-2) - \delta(n-3)$
   $x2(n) = \delta(n) - \delta(n-2) + \delta(n-4)$

OR

Q4.

a) Explain and prove the following properties of DFT [8]

1) Periodicity
2) Complex conjugate property of DFT
3) Time reversal
4) Circular time shifting

b) Find the DFT of a sequence x(n)={1,2,3,4,4,3,2,1} using DIT algorithm. Draw 8-point DIT FFT flow graph and slow all values properly. [8]

Q5.

a) Define ROC. Find the Z transform and sketch the corresponding ROC for following signals [8]

1) $x(n) = \{1,2,3,4,5,9\}$ (origin is at 3)
2) $x(n) = (n + 0.5)\left(\frac{1}{3}\right)^n u(n)$

b) Find the H(z) and poles of the system

$y(n) - 0.25y(n-1) + 0.25y(n-2) - 0.0625y(n-3) = 2x(n) + 3x(n-1)$

And state the system is stable or not. [8]

OR

Q6.

a) State the convolution property of Z-transform and hence find causal sequence x(n) for [8]

$$X(z) = \frac{[6+z^{-1}]}{[(1+0.25z^{-1})(1+0.5z^{-1})]}$$

b) Determine the inverse Z transform of the function [8]

1) $X(z) = \frac{[8z-19]}{[(z-2)(z-3)]}$ $x(n)$ is causal sequence

2) $\frac{[z^3+z^2]}{[(z-1)(z-3)]}$ ROC;|z|>3

SECTION-II

Q7.

a) Differentiate between FIR and IIR filters. In case of IIR filter, compare impulse invariant method and Bilinear Transformation method. [8]

b) Design a complete digital low pass Butterworth filter for T=1 for following specifications using an impulse invariant method [10]

$0.8 \leq |H(e^{jw})| \leq 1 \quad for \quad 0 \leq w \leq 0.2\pi$

$|h(e^{jw})| \leq 0.2 \quad for \quad 0.6\pi \leq w \leq \pi$

Q8.

a) Design the seven coefficient FIR low-pass filter using frequency sampling method with following specifications. Plot the magnitude response of the resulting filter [10]

$H(e^{jw}) = e^{-j(N-1)w/2} \quad for \quad 0 \leq |w| \leq \frac{\pi}{2}$

$H(e^{jw}) = 0 \;\; for \; \frac{\pi}{2} \leq |w| \leq \pi$

b) What is the nature of phase response of FIR filter? Derive the condition of linear phase of FIR filter. [8]

Q9.

a) Draw the block schematic for decimator and explain the need for a filter. Derive the expression for decimated output signal i.e. y(m) and draw the spectrum of the signal after filtering and after decimation process. [10]

b) Explain the need for multistage design. How will you select decimation factors for different stages for multistage implementation. [6]

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OR

Q10.

a) Why should we use an interpolator first before a decimator in case of sampling rate converter by a factor of I/D. Derive the equation for the output of the sampling rate converter I/D [10]

b) Explain how Multi-rate sampling can be used in acquisition of high Quality data? [6]

Q11.

a) Explain the need of DSP processor and features required in DSP processor [6]

b) Explain pipelining concept. Also explain MAC, ALU and Barrel Shifter unit of DSP processor. [10]

OR

Q12.

a) Differentiate between DSP processor with conventional microprocessor architectures? Explain the architecture of TMS320C67XX listing its important features [10]

b) Explain the application of DSP processor in speech processing [6]

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