B.Sc (HONS.) IN CSE, PART-IV, SEVENTH SEMESTER EXAMINATION, 2012

DIGITAL SIGNAL PROCESSING

CSE : 417

Examination Code : 617

Time—3 hours

Full marks—80

[N.B.—The figures in the right margin indicate full marks. Answer any **four** *of the following questions.]*

Marks

1. *(a)* What do you mean by DSP? List the applications of DSP. 4

   (b) Write the difference between Analog signal and Digital signal. 4

   (c) Consider the Analog signal $x_a(t) = 3\cos 100\pi t$:— 6

   (i) Determine the minimum sampling rate required to avoid aliasing.

   (ii) Suppose that the signal is sampled at the rate $F_s = 200$ Hz. What is the discrete-time signal obtained after sampling?

   (iii) Suppose that the signal is sampled at the rate $F_s = 75$ Hz. What is the discrete-time signal obtained after sampling?

   (iv) What is the frequency $0 < F < F_s / 2$ of a sinusoid that yields samples identical to those obtained in part (*iii*)?

   (d) How analog to digital conversion of a signal is done? 6

2. *(a)* Define the following :— 4

   (i) Periodic and aperiodic signal; *(ii)* Unit step and ramp signal.

   (b) Determine if the following system are linear or non-linear ঃ 4

   (i) $y(n) = x^2(n)$; *(ii)* $y(n) = e^{x(n)}$.

   (c) Discuss the four steps involving in computing the convolution of $x(k)$ and $h(k)$. 5

   (d) What do you mean by LTI system? 2

   (e) Write down the properties of cross co-relation and auto-co-relation. 5

[Please turn over

Marks

3. (a) Define z-transform and ROC of z-transform. 4

(b) Define poles and zeros. Determine the pole-zero of the signal $x(n) = a^n\, a(n), a > 0$. 2+4=6

(c) Determine the inverse z-transform of $x(z) = \dfrac{1}{1 - 1{\cdot}5\, z - 1 + 0{\cdot}5\, z - 2}$, 6

when (a) ROC ៖ $|z| > 1$; (b) ROC ៖ $|z| < 0{\cdot}5$.

(d) Describe the properties of z-transform (i) Linearity; (ii) Time shifting; (iii) Convolution. 4

4. (a) What is DFT and FFT? 4

(b) Sketch a simplified block diagram of a digital filter with description. 5

(c) Write down FIR filter design procedure. 6

(d) Consider the signal $x(n) = a^n\, u(n)$, $0 < a < 1$, the spectrum of this signal is sampled at frequencies $W_k = 2\pi k/N$, $k = 0, 1, \ldots\ldots,$ $N - 1$. Determine the reconstructed spectra for $a = 0{\cdot}8$ when $N = 5$ and $N = 50$. 5

5. (a) State and prove the Complex convolution theorem. 6

(b) Derive z-transform from Laplace transform. 5

(c) How an adaptive filter can be used as a noise canceller? 4

(d) Write down the characteristics of adaptive filter. 5

6. (a) Describe Window method for co-efficient calculation of FIR filter. 6

(b) Describe the concept of adaptive filtering. Why it is needed? 5+1=6

(c) What do you mean by signal-flow graph? 2

(d) Describe the Radix-2 algorithm and define Butterfly operation. 6

B.Sc (HONS.) IN CSE, PART-IV, SEVENTH SEMESTER EXAMINATION, 2011

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(Digital Signal Processing)

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[N.B.—The figures in the right margin indicate full marks. Answer any **four** *from the following questions.]*

Marks

1. (a) What do you mean by DSP? List the applications of DSP. 4

(b) How analog to digital conversion of a signal is done? 6

(c) Characterize the following systems :— 6

(i) $Y(n) = n\,X(n)$;

(ii) $Y(n) = X(n^2)$.

(d) Define the following terms :— 4

Discrete time signal, Continuous time signal, Sampling and quantization.

2. (a) Explain the input-output description of a Discrete Time System. 4

(b) Determine the output response of a system $y(n)$ to the input signal 4

$x(n) = \{0, 3, 2, 1, \underset{\uparrow}{0}, 1, 2, 3, 0,\}$ and

$y(n) = \frac{1}{3}\left[x(n+1) + x(n) + x(n-1)\right]$.

(c) What is Z transform? How is Z transform obtained from Laplace transform? 4

(d) What are the properties of convolution? What are the steps involved in calculating Convolution Sum? 5

(e) What are the properties of ROC in Z-transform? 3

3. (a) What is digital filter? How digital filter can be classified? 4

(b) What are the advantages and disadvantages of FIR filter? 5

(c) Sketch a general block diagram of an FIR filter. State the general impulse response and transfer function of FIR filter. 5

(d) What is the advantage of using normalized frequency in designing filter? 2

(e) How phase distortion and delay distortion are introduced? 4

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(29)

Marks

4. (a) What is DFT and inverse DFT? 4

(b) Derive the DTF of the sample data sequence $x(n) = \{1, 1, 2, 2, 3, 3\}$ and compute the corresponding amplitude and phase spectrum. 5

(c) State and prove Parseval's theorem 5

(d) How many multiplications and additions are required to compute N-Point DFT using radix 2 FFT? 6

5. (a) Mention the advantage of direct and cascade structures. 6

(b) State the circular time shifting and circular frequency shifting properties of DFT. 4

(c) What is the relationship between Z-transform and DTFT? 5

(d) What are the different methods of evaluating inverse Z-transform? 5

6. (a) What is the principle of pole-zero placement method for calculating co-efficients of IIR filter? 8

(b) How an adaptive filter can be used as a noise canceller? 4

(c) Describe Recursive Least Square (RLS) algorithm to design adaptive filter. 8

CSE 7th

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B.Sc (HONS.) IN CSE, PART-IV, SEVENTH SEMESTER EXAMINATION, 2010

CSE-417

(Digital Signal Processing)

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[N.B.—The figures in the right margin indicate full marks. Answer any **four** *questions.]*

Marks

1. *(a)* What is a continuous and discrete time signal? 2

   (b) What is correlation and autocorrelation? 4

   (c) Check for the following systems are linear, causal, time in variant and static :— 10

   (i) $y(n) = x(2n)$;

   (ii) $y(n) = \cos(x(n))$;

   (iii) $y(n) = x(n)\cos(x(n))$;

   (iv) $y(n) = x(-n+2)$;

   (v) $y(n) = x(n) + n\,x(n+1)$.

   (d) What are the advantages of DSP over analog signal processing? 4

2. *(a)* Describe the analog-to-digital (A/D) conversion procedure. 8

   (b) What are the basic building blocks of discrete time systems? 4

   (c) What is memory system and memoryless system? 4

   (d) What do you mean by LTI system and causality of an LTI system? 4

3. *(a)* What is the relation between Fourier transform and Z-transform? 4

   (b) State and prove the properties of Z-transform. 6

   (c) Determine the inverse Z-transform of $X(Z) = \dfrac{1}{1-1\cdot5z^{-1}+0\cdot5z^{-2}}$ 6

   when, *(a)* ROC : $|Z| > 1$;

   (b) ROC : $|Z| < 0\cdot5$.

   (d) State the convolution property of Z-transforms. 4

[Please turn over

Marks

4. *(a)* How digital filters can be classified? 4

(b) What is window function? 3

(c) Determine the discrete Fourier transform $x(n) = (1, 1, 1, 1)$ and prove $x(n) * h(n) = X(z)\,H(z)$. 8

(d) Write the various frequency transformation in analog domain. 5

5. *(a)* Write down FIR filter design procedure. 4

(b) Sketch the block diagram for the direct form realization and the frequency-sampling realization of the M=32, α=0, linear-phase (symmetric) FIR filter which has frequency samples :— 8

$$H\left(\frac{2\pi k}{32}\right) = \begin{cases} 1, & k = 0, 1, 2 \\ \frac{1}{2}, & k = 3 \\ 0, & k = 4, 5, \ldots\ldots, 15. \end{cases}$$

(c) Why impulse invariant method is not prefered in the design of IIR filters other than low pass filter? 4

(d) What is the necessary and sufficient condition for the linear phase characteristics of a FIR filter? 4

6. *(a)* How can you design a digital filter from analog filter? 4

(b) Explain the method of design of IIR filters using bilinear transform method. 6

(c) What do you mean "dead band" of the filter? 2

(d) Describe basic LMS adaptive algorithm with flow-chart. 8