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University of Rajshahi

Department of Computer Science and Engineering

B. Sc. (Engg) Part-III Odd Semester Examination 2020

Course: CSE-3131 (Digital Signal Processing)

Full Marks: 52.5 Duration: 3(Three) Hours

Answer 06(Six) questions taking any 03(Three) questions from each part

Part-A

1. a) Define signal, system, and signal processing. 2 19 — 3
   b) Differentiate between analog and digital signal. — 2
   c) Explain the basic elements of a Digital Signal Processing (DSP) system. pg-22 jk-A — 3.75

2. a) Define discrete time signal. How do you convert an analog signal into digital form-explain with example. — 3.75
   b) Consider the analog signal $x(t) = 5\sin 200\pi t$ — 3
      i. Determine the minimum sampling rate required to avoid aliasing.
      ii. Suppose that the signal is sampled at the rate $F_s = 500\ Hz$. What is the discrete time signal?
      iii. Suppose $F_s = 75\ Hz$. What is the discrete time signal?
   c) Define sampling theorem. — 2

3. a) Define unit sample and unit step sequences. — 2
   b) Differentiate between energy signal and power signal. — 2
   c) Determine the response of the following system to the input signal — 3
      $$x(n) = \begin{cases} n & -3 \le n \le 4 \\ 0 & \text{otherwise} \end{cases}$$
      i. $y(n) = x(n-1)$
      ii. $y(n) = 3x(n+1)$
      iii. $y(n) = \max[x(n+1), x(n), x(n-1)]$
   d) Define linearity property of a signal. — 1.75

4. a) Define LTI system. — 2.75
   b) Derive convolution equation from the response of LTI system to arbitrary inputs. — 3
   c) If $h(n) = \{1,2,3,4\}$ and $x(n) = \{2,5,7,8\}$. Find out $y(n) = h(n) * x(n)$. — 3

Part-B

5. a) Define Z transform. What is ROC? — 2.75
   b) Find Z transform and ROC of the following finite duration signal: — 3
      i. $x_1(n) = \{1,\underset{\uparrow}{2},3,4,5\}$
      ii. $x_2(n) = \{0,0,1,2,3,4,5,1\}$
   c) Define Fourier series. Write down some applications of Fourier transform. — 3

6. a) Sketch the discrete time signal $x(n) = 4\delta(n+4) + \delta(n) + 2\delta(n-1) + \delta(n-2) - 5\delta(n-3)$ — 3.5
   b) Find the periodicity of $x(n) = \cos(\frac{2\pi n}{7})$ — 3.5
   c) What is inverse system? — 1.75

7. a) Define DFT and IDFT equation. 0.5 — 2
   b) Define symmetry property of DFT equation. — 2
   c) Define DFT leakage. — 3
   d) Define Hamming window in both TD and FD. — 1.75

8. a) Derive 4 point FFT equation from DFT equation upto twiddle factor formation. — 3
   b) Determine the 4 point DFT of the sequence $x(n) = \{1,0,1,1\}$ — 3.75
   c) What is twiddle factor? Describe its importance in FFT equation. — 2

University of Rajshahi
Dept. of Computer Science and Engineering
B.Sc. (Eng.), Part-III (Odd Semester), Examination 2019
CSE3131: Digital Signal Processing
Full Marks: 52.5 Time: 3 hours
[Answer Six questions taking any **three** questions from each Section]

Section A

1. a) Define signal, system and digital signal processing with example. 2.75
   b) Define aliasing effect. 3.00
   Consider the analog signal

$$x(t) = 3cos1000\pi t$$ → 45 → 3rd e,

Suppose the signal is sampled at the rate $F_{s_1} = 500Hz$ and $F_{s_2} = 2000Hz$
   i. What are the discrete time signals after sampling?
   9 ii. In which case, aliasing occurs and what is the aliased frequency in the range of $0 < F \leq {F_s}/_2$

folding-frequency.

   c) Consider the following input-output equations of signal: 3.00
   i. $y(n) = x(n) + \frac{1}{x(n-1)}$
   ii. $y(n) = nx(n)$ — 60 h
   Determine whether the system is linear or non-linear.

2. a) Compute and plot the convolutions $x(n) * h(n)$ and $h(n) * x(n)$ where $x(n) = u(n)$ and $h(n) = u(n-3)$ 3.75
   b) Consider the analog signal $x_a(t) = 35\,Sin\,550\,\pi t + 30 Cos\,620\,\pi t - 35\,Cos\,660\,\pi t$. Determine the sampling rate and maximum amplitude of the signals. 3.00
   c) If the number of quantization levels L = 256, then how many bits per sample will be required? 2.00

3. a) Define correlation between two signals. Write down its significance. 2.75
   b) Compute the autocorrelation of the signal $x(n) = a^n u(n), \quad 0 < a < 1$. → 141 3.00
   c) Determine the autocorrelation sequences of the following signals. 3.00
   i. $x(n) = \{1, 2, 1, 1\}$
   ii. $y(n) = \{1, 1, 2, 1\}$ → 164 → 2-59 ex.
   What is your conclusion?

4. a) Why Z-transform? How do you differ Z-transform with Fourier transform? 2.75
   b) Define ROC? Write down some properties of ROC. 3.00
   c) Determine the z-transform of the signal 3.00

$$x(n) = 0.5u(n) + 0.75u(n-1)$$

z-transform এর জন্য Engineer's choice tutor দেখ.

Section B

5. a) Define Fourier series. Point out some importance of Fourier transform in DSP. 2.75
   b) Explain DFT leakage problem with example. 3.00
   c) Find 4 point DFT of the following sequence 3.00

$$x(n) = \{1, 0, -1, 0\}$$

   Sketch magnitude and phase.

6. a) What is window function? Discuss frequency domain characteristics of Rectangular window and Hamming window. 2.75
   b) What is zero padding technique? Discuss with example. 3.00
   c) Describe Stable and Unstable System with example. 3.00

7. a) Why FFT is needed? 3.00
   b) Describe the algorithm of decimation-in-time (DIT) for FFT. 3.00
   c) Compute the 8-point DFT of the sequence $x(n) = \begin{cases} 1 & 0 \le n \le 7 \\ 0 & otherwise \end{cases}$ 2.75

   By using DIT (Decimation-In-Time), DIF (Decimation-In-Frequency) algorithm.

8. a) What is infinite impulse response (IIR) filter? Explain IIR filter with equation and diagram. 4.75
   b) Define Adaptive filter. Draw the block diagram and explain the function of adaptive filter. 4.00

University of Rajshahi
Department of Computer Science and Engineering
B.Sc. Engg. Part-III Odd Semester Examination 2018
Course: CSE 3131 (Digital Signal Processing)

Time: 3 Hours

Full Marks: 52.5

[Answer any six (06) questions taking three (03) from each section.]

Section A

1. (a) Why signal processing is needed? Draw the block diagram of DSP system and briefly introduce the different components of the system. 3

   (b) Write the mathematical expression of a discrete time sinusoidal signal and draw the signal with $w = \frac{\pi}{6}$ and $\varphi = \frac{\pi}{3}$. 3

   (c) Show that the highest rate of oscillation in a discrete time sinusoidal is attained when $f = \frac{1}{2}$ or $f = -\frac{1}{2}$. 2.75

2. (a) What is aliasing effect? Show the effect where two sinusoidal with frequencies $F_0 = \frac{1}{8}Hz$ and $F_1 = -\frac{7}{8}Hz$ and $F_s = 1\ Hz$ is used. 2.75

   (b) Consider the analog signal: $x_a(t) = 3\ \cos 150\pi t$ 3
   suppose $F_s = 75\ Hz$. What is the discrete signal obtained after sampling and what is the frequency $0 < F < \frac{F_s}{2}$ of a sinusoidal that yields identical samples?

   (c) What is quantization process? Define quantization step size and quantization error with examples. 3

3. (a) Define LTI system. Discuss the response of LTI systems to arbitrary inputs. 2.75

   (b) Find y(n) if x(n) = n+2 for $0 \le n \le 3$ and h(n) = a^n u(n) for all n. 3

   (c) Define causal and non-causal systems with examples. Determine if the following system are time-invariant or time-variant. y(n) = x(-n) 3

4. (a) What is z-transform? 1.75

   (b) Find the z-transform of the sequence $x(n) = \left(\frac{1}{3}\right)^n u(-n)$. 3

   (c) Which approaches are available to invert the z-transform to recover the sequence x(n)?

Section B

5. (a) Define Fourier series of a continuous time signal. [illegible]

   (b) What is Fourier integral? Write down some properties of the continuous time Fourier transform. [illegible]

   (c) Define DFT equation. Find 4 point DFT of the sequence x(n) = u (n). Draw amplitude, magnitude, phase, and power spectrum. 4

6. (a) Define DFT leakage. Explain the problem with example. 2.75

   (b) What is windowing process? Discuss the effect using Hamming and Rectangular window on DFT output. 3

   (c) Define DFT resolution. What is the effect of zero stuffing technique on DFT resolution? Explain with example. 3

7. (a) Why FFT? What is twiddle factor? Discuss the use of twiddle factor on DFT equation. 4

   (b) Compute 4 point DFT of a sequence x(n) = { 0, 1, 2, 3 } using ~~DFT~~ DIT algorithm. 3

   (c) Discuss Bit reversal technique in brief. 1.75

8. (a) What is the importance of convolution in digital signal processing? 2.75

   (b) State and prove convolution theorem. Compute the convolution of the signal: 4

$$x_1(n) = \{3, -2, 2\}, \qquad x_2(n) = \begin{cases} 1, & 0 \le n < 5 \\ 0, & otherwise \end{cases}$$

   (c) What do you mean by bandwidth of a signal? How can you extract a particular band of a signal? Explain it. 2

University of Rajshahi

Department of Computer Science and Engineering

B.Sc. (Engg.) Part – III (Odd Semester) Examination – 2017

Course: CSE3131 (Digital Signal Processing)

Marks: 52.5Times: 3 Hours

[*Answer any six questions taking at least three from each part.*]

PART-A

1. a) Given $z(t) = [x(t) + y(t)]^2$, where $x(t)$ and $y(t)$ are band-limited signals with highest frequencies B and $3B$ Hz, respectively. Determine the minimum sampling frequency for $z(t)$ in order to avoid aliasing. Please briefly justify your answer. 2.75

   b) Consider the following analog signal; 6

   $x_a(t) = 3cos2000\pi t + 5sin6000\pi t + 10cos12{,}000\pi t.$ → 49/32

   i. What is the Nyquist rate for this signal?

   ii. Assume now that we sample this signal using a sampling rate $F_s = 5000$ samples/s. What is the discrete-time signal obtained after sampling?

   iii. What is the analog signal $y_a(t)$ that we can reconstruct from the samples if we use ideal interpolation?

2. a) Consider a continuous-time system which has input of signal $x(t)$ and output of $y(t) = x(t)u(t)$. 4.75

   i. Is this system time invariant? Justify your answer.

   ii. Is this system linear? Justify your answer.

   b) Determine if the systems described by the following input-output equations are linear or non-linear? 4

   i. $y(n) = Ax(n) + B$, ii. $y(n) = e^{x(n)}$

3. a) Given the impulse response of an LTI system is $h(n) = \begin{cases} a^n, & n \geq 0 \\ b^n, & n < 0 \end{cases}$, determine the range of values of a and b for which the system is stable. 4

   b) Consider two discrete-time LTI systems which are characterized by their impulse responses; 4.75

   $h_1[n] = \delta[n] - \delta[n-1]$ and $h_2[n] = u[n]$.

   i. Determine whether these two LTI systems are inverse of each other. Justify your answer.

   ii. Determine whether these systems are stable, memory-less and causal. Justify your answer.

4. a) What are the properties of Region of Convergence? 1.75

   b) Find the z-transform of the ~~square~~ Sequence 3

   $$x(n) = \left(\frac{1}{3}\right)^{n-1} u(n-1)$$

   c) Define inverse z-transform. Find the inverse z-transform of 4

   $$X(z) = \frac{z + 0.2}{(z + 0.5)(z - 1)}, |z| > 1$$

PART-B

5. a) Consider the signal, $x(n) = \begin{cases} A, & -M \leq n \leq M \\ 0, & elsewhere \end{cases}$ 4.25

   i. Determine the Fourier transform of $x(n)$, i.e., $X(\omega)$

   ii. Determine and plot the magnitude and phase of $X(\omega)$.

   b) Given, $x_1(n) = x_2(n) = \{1, 1, 1\}$, and $x(n) = x_1(n) * x_2(n)$. 4.50

   i. Determine $X(w)$, the Fourier transform of $x(n)$

   ii. By using $X(w)$, determine $x(n)$

   iii. For $n = 0$, find $x(0)$

6. a) Find and plot the magnitude and phase of $H(\omega)$ for the following three-point moving average system 3.50

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

b) Given, $x(n) = [2 \ 4 \ -1 \ 6]$, $N = 4$, $(n = 0,1,2,3)$ 2.25

   i. Determine $X[k]$ for $k = 0,1,2,3$
   ii. Plot $x(n)$ and $|X[k]|$

c) Explain briefly when the overlap-add and overlap-save methods are used. 3

7. a) Given $h(n) = \delta(n) + \delta(n-1)$. 5

   i. Show that it is a linear phase low-pass filter.
   ii. Explain why it is low-pass filter.
   iii. Determine and plot the associated phase and group delay as well.

b) Explain briefly why rippling occurs in pass-band and stop-band? 3.75

8. a) Define Decimation-in-time algorithm. Draw the flow-graph of a two point DFT for a decimation-in-time decomposition. 3

b) Compute the DFT of a ~~square~~ sequence $x(n) = \{1, -1, 1, -1\}$ using DIT algorithm. 3

c) Find the IDFT of the ~~square~~ sequence $x(n) = \{10, -2+2j, -2, -2-2j\}$ using DIT algorithm. 2.75

University of Rajshahi
Department of Computer Science and Engineering
B.Sc. Engg. Part-3 Odd Semester Examination-2016
CSE 3131 (Digital Signal Processing)

Full Marks: 52.5 **Time: 3 Hours**

[Answer Six Questions Taking any Three Questions from each Part of the following Questions.]

PART-A

1. (a). Give a brief introduction about the different parts of a digital signal processing system. 2.75

   (b). Define impulse response of a system. How impulses are used to know the behavior of any arbitrary system? Discuss it with a suitable example. 3

   (c). Consider the following input-output equations of signal: 3

   (i). $y(n) = x(n) + \frac{1}{x(n-1)}$ (ii). $y(n) = nx(n)$

   Determine whether the system is linear or non-linear.

2. (a). What is aliasing effect? Discuss it with an example. Why do you need antialiasing filter in signal processing? 3

   (b). Define up sampling and down sampling with example. Why do you need to change the sampling rate of a signal? 3

   (c). Prove that the highest rate of oscillation in a discrete sinusoidal is attained when $\omega = \pi$ or $\omega = -\pi$. 2.75

3. (a). Describe stable and unstable system with example. 3

   (b). Explain convolution sum of signals. 2.75

   (c). Consider the input $x[n] = \{1, 0, 2, 3\}$ of a LTI system and impulse response $h[n] = \{1, 2, 1, 3\}$. Find out the convolution sum of output $y[n]$ of the LTI system. 3

4. (a). What is ~~Fortier~~ Fourier series? Represent a Fourier series with frequencies 1KHz, 2 KHz, ..., ..., so on. Explain harmonic Fourier series. 3

   (b). What do you mean by bandwidth of a signal? How can you extract a particular band of a signal? Explain it. 2

   (c). Define Fourier and inverse Fourier integral. Why do you need Fourier transform of a signal? Write down the convolution property of Fourier transform. 3.75

PART-B

5. (a). Explain inverse z-transform. 2

(b). Determine the inverse z-transform $\chi(n) = \dfrac{z^2 + z}{(z-1)(z-3)}$ and ROC: $|z| > 3$. 2.75

(c). What are the properties of region of convergence (ROC)? Find the z-transform and ROC of the finite signal of the sequence $x(n) = \{2, 4, \underset{\uparrow}{5}, 7, 0, 1\}$ 4

6. (a). Describe the relationship between the magnitude of a signal before and after DFT with an example. 2

(b). What is the leakage effect of DFT? Explain the reason behind the leakage problem with an example. 3

(c). Discuss the use of window to ~~store~~ Solve the leakage effect. Show the responses of Hanning and Rectangular window in frequency domain (FD) and clearly describe the findings from the spectrums. 3.75

7. (a). Define zero stuffing technique. Explain its effect in signal analysis. 2

(b). What is Twiddle factor? Discuss the effect of twiddle factor in details of FFT. 3.75

(c). Draw the butterfly structure of FFT implementation for a 4- point DFT. 3

8. (a). Define analog filter and digital filter. Realize the second order digital filter 3

$$y(n) = 25\cos(\omega_0)y(n-1) - 50y(n-2) + x(n) - 2\cos(\omega_0)x(n-1).$$

(b). Define Adaptive filter. Discuss LMS filter with necessary figure and equation. 2.75

(c). Obtain the impulse response $H(z)$ of a digital filter corresponding to an analog filter having impulse response of $h_a(t) = 0.5e^{-2t}$ and a sampling rate of 1KHz by using the impulse invariance method. 3