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Paper Code : EC-602

DIGITAL SIGNAL PROCESSING

Time Allotted : 3 Hours *Full Marks* : 70

The figures in the margin indicate full marks.

Candidates are required to give their answers in their own words as far as practicable.

GROUP – A

(Multiple Choice Type Questions)

1. Choose the correct alternatives for any *ten* of the following : 10 × 1 = 10

i) The system $y(n) = x(n) + x(n-1)$ is

a) linear time-invariant

b) non-linear time invariant

c) linear time variant

d) none of these.

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

a) energy signal b) power signal

c) both of these d) none of these.

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iii) The value of the twiddle factor W_8^4 is given by

a) 1 b) $-j$

c) $\frac{1}{\sqrt{2}} - \frac{j}{\sqrt{2}}$ d) -1.

iv) If F_ss is the minimum sampling rate, F_{max} is the highest frequency available in the analog signal, then at Nyquist rate

a) $F_s = 2F_{max}$ b) $F_s = 0.5 F_{max}$

c) $F_s = F_{max}$ d) $F_s < F_{max}$.

v) Overlap save method is used to find

a) circular convolution b) linear convolution

c) z-transform d) DFT.

vi) A system having impulse response $h(t)$ will be BIBO stable if

a) $\int_{-a}^{a} |h(t)|\, dt < \infty$ b) $\int_{-a}^{a} h(t) dt < \infty$

c) $\int_{-a}^{a} |h(t)|\, dt > \infty$ d) $\int_{-a}^{a} |h(t)|\, dt = 0$.

vii) Why 16 point DFT is preferable than 4 point DFT ?

a) Resolution of spectrum is poor for 4 point DFT than 16 point DFT

b) Resolution of spectrum is high but not reliable in 4 point DFT

c) Calculation of 4 point DFT is more complex

d) None of these are true.

viii) The mapping from analog to digital domain in impulse invariant method is

a) one to many b) many to one

c) one to one d) none of these.

ix) If $x[n] = \{1, 0, 0, 1\}$, the DFT value $x(0)$ is

a) 2 b) $1 + j$

c) 0 d) $1 - j$.

x) IIR filter is

a) recursive and linear

b) none-recursive and linear

c) recursive and non-linear

d) none of these.

xi) Zero padding of a signal

a) reduces aliasing

b) increases frequency

c) increases time resolution

d) has no effect.

xii) If the Fourier transform of a sequence $x(n)$ is $X(e^{jw})$, then the Fourier transform of $x(n-k)$ is

a) 0

b) $(e^{-jwk})X(e^{jw})$

c) $(e^{-jw})X(e^{jw})$

d) cannot be determined.

xiii) The digital system in $y(n) = x(n^2)$ is

a) non-linear and causal

b) linear and causal

c) linear and non-causal

d) non-linear and non-causal.

GROUP – B

(Short Answer Type Questions)

Answer any *three* of the following. $3 \times 5 = 15$

2. a) Define energy and power signals.

b) Determine whether the signal is power or energy signal : $x(n) = e^{2n}u(n)$. 2 + 3

3. Find the convolution of $u(n) * u(n-3)$.

4. Find the inverse Z-transform of $X(z) = \dfrac{1-\left(\frac{1}{4}\right)z^{-1}}{1-\left(\frac{1}{9}\right)z^{-2}}$ using convolution method.

5. Show how the time complexity of finding the DFT of 256 point data sequence improves by using Radix - 2 FFT algorithm instead of using direct computation. 2 + 3

6. Determine the direct form II realization for the following system :

$$y(n) = -0.1\,y(n-1) + 0.72\,y(n-2) + 0.7\,x(n) - 0.252\,x(n-2).$$

GROUP – C

(Long Answer Type Questions)

Answer any *three* of the following. $3 \times 15 = 45$

7. a) What are the differences between linear and circular convolution ?

   b) Determine the output response $y(n)$ if $h(n) = \{1, 1, 1\}$ and $x(n) = \{1, 2, 3, 1\}$ by using

   i) Linear convolution

   ii) Circular convolution

   iii) Circular convolution with zero padding.

   3 + 4 + 4 + 4

8. a) Write down the properies of region of convergence (ROC).

   b) The step response of an LTI system is $y(n) = \left(\frac{1}{3}\right)^{n-2} u(n+2)$. Find the system function $H(z)$ and $h(n)$.

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   c) Find the inverse Z-transform of $X(z) = \dfrac{z}{3z^2 - 4z + 1}$.

   If the region of convergence are :

   i) $|z| > 1$

   ii) $|z| < \frac{1}{3}$

   iii) $\frac{1}{3} < |z| < 1$. 4 + 5 + 6

9. a) Find the DFT of the sequence $\{ \underset{\uparrow}{1}, 1, 1, 1, 2, 2, 2, 2 \}$ using radix-2 Decimation-in-Time FFT. Sketch the magnitude and phase plot.

   b) What is the need for FFT ?

   c) What are the differences and similarities between DIT and DIF algorithms ? 9 + 2 + 4

10. a) What is warping effect ? Explain. How can warping effect be removed ?

   b) Design a digital Butterworth filter satisfying the following conditions using Bilinear transformation $0.707 \le |H(e^{j\omega})| \le 1$ for $0 \le \omega \le \pi/2$ and $|H(e^{j\omega})| \le 0.2$ for $3\pi/4 \le \omega \le \pi$.

   c) How can a digital filter be built from analog filter ?

   4 + 7 + 4

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11. a) What are the properties of FIR filter ?

    b) What do you understand by the term "window" for FIR filter ? Explain.

    c) Derive the spectrum of the rectangular window.

    d) Compare Hamming with Kaiser window.

    e) Explain Gibbs phenomenon. 2 + 2 + 5 + 3 + 3

12. Write short notes on any *three* of the following : 3 × 5

    a) Aliasing effect

    b) Causal and Non-causal Signals

    c) Direct Form 1 and Direct Form II Realization

    d) Advantages and applications of DSP

    e) Recursive and non-recursive system.

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