| Reg. No. | | | | | | | | | | | | | | | | |
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| Question Paper Code | 13281 |
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**B.E. / B.Tech. - DEGREE EXAMINATIONS, NOV / DEC 2024**

Fifth Semester

**Electronics and Instrumentation Engineering**

(Common to Instrumentation and Control Engineering)

**20EIPC503 - DIGITAL SIGNAL PROCESSING**

Regulations - 2020

Duration: 3 Hours | Max. Marks: 100

**PART - A (MCQ) (20 × 1 = 20 Marks)**

Answer ALL Questions

| No. | Question | Marks | K – Level | CO |
|---|---|---|---|---|
| 1. | The Nyquist rate is defined as:<br>(a) The sampling rate at which a signal is under sampled<br>(b) Twice the maximum frequency of the signal<br>(c) The maximum amplitude of a signal<br>(d) The average power of a signal | 1 | K1 | CO1 |
| 2. | Which one is a key characteristic of a causal system?<br>(a) It responds only to past and present inputs (b) It responds to future inputs<br>(c) It depends on initial conditions only (d) It generates random outputs | 1 | K1 | CO1 |
| 3. | For a system to be considered stable, which of the following must be true?<br>(a) It is time-variant<br>(b) It responds to inputs with unbounded outputs<br>(c) It responds to bounded inputs with bounded outputs<br>(d) It depends on future inputs | 1 | K1 | CO1 |
| 4. | A system is said to be linear if it follows the principle of:<br>(a) Time variance (b) Additivity and homogeneity<br>(c) Recursion (d) Time invariance | 1 | K1 | CO1 |
| 5. | The inverse Z-transform of a function X(z) can be computed by:<br>(a) Taking the Laplace transform and applying the inverse<br>(b) Solving the difference equation directly<br>(c) Using partial fraction expansion and applying the inverse formula<br>(d) Differentiating the original Z-transform equation | 1 | K1 | CO2 |
| 6. | Which of the following statements about the magnitude and phase representation of the DTFT is true?<br>(a) The magnitude spectrum gives the amplitude of each frequency component, and the phase spectrum gives the relative time delay of each frequency<br>(b) The magnitude spectrum shows the real part of the signal, and the phase spectrum shows the imaginary part<br>(c) The magnitude spectrum shows the total energy in the signal, and the phase spectrum shows the power<br>(d) The magnitude spectrum is always zero for a periodic signal, while the phase spectrum is constant | 1 | K1 | CO2 |
| 7. | Convolution of two discrete-time signals in the time domain corresponds to:<br>(a) Division of their Z-transforms (b) Multiplication of their Z-transforms<br>(c) Addition of their Z-transforms (d) Time shifting in the frequency domain | 1 | K1 | CO2 |
| 8. | A discrete-time system is stable if:<br>(a) All the poles of its Z-transform lie on the unit circle<br>(b) All the poles of its Z-transform lie inside the unit circle<br>(c) All the zeros of its Z-transform lie inside the unit circle<br>(d) The region of convergence includes the origin | 1 | K1 | CO2 |

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| 9. | Twiddle factors in the FFT algorithm are:<br>(a) The real part of the DFT coefficients<br>(b) Complex exponential factors used for combining smaller DFTs<br>(c) Multiplication factors that control the butterfly computations<br>(d) Scaling factors used for normalizing the frequency components | *1* | *K1* | *CO3* |
| 10. | The magnitude spectrum of a DFT represents:<br>(a) The real part of the signal in the time domain<br>(b) The amplitude of different frequency components of the signal<br>(c) The imaginary part of the time domain signal<br>(d) The phase shift of different frequency components | *1* | *K1* | *CO3* |
| 11. | Which of the following statements about radix-2 FFT is correct?<br>(a) It can only be applied when the number of data points is a power of 2<br>(b) It is slower than computing the DFT directly<br>(c) It cannot handle complex-valued signals<br>(d) It requires the input signal to be non-periodic | *1* | *K1* | *CO3* |
| 12. | The Decimation-in-Time (DIT) algorithm for FFT works by:<br>(a) Dividing the time-domain signal into smaller frequency components<br>(b) Splitting the signal into even and odd indexed samples at each stage<br>(c) Decimating the frequency domain representation<br>(d) Reversing the order of time samples before performing FFT | *1* | *K1* | *CO3* |
| 13. | In FIR filter design, windowing techniques are used to:<br>(a) Increase the computational complexity of filter design<br>(b) Truncate the infinite impulse response to a finite length<br>(c) Change the sampling rate of the filter<br>(d) Remove unwanted frequency components from the filter | *1* | *K1* | *CO4* |
| 14. | Chebyshev filters differ from Butterworth filters because they:<br>(a) Have an equiripple response in the passband or stopband<br>(b) Are inherently unstable<br>(c) Have a maximally flat response<br>(d) Are only applicable for discrete systems | *1* | *K1* | *CO4* |
| 15. | In filter realization, parallel form implementation is beneficial for:<br>(a) Systems with low computational power (b) Minimizing memory requirements<br>(c) Designing filters with multiple resonant peaks (d) Achieving a linear phase response | *1* | *K1* | *CO4* |
| 16. | FIR filters are preferred for applications that require:<br>(a) Minimum phase characteristics (b) Non-linear phase characteristics<br>(c) Linear phase characteristics (d) Maximum phase characteristics | *1* | *K1* | *CO4* |
| 17. | The primary difference between fixed-point and floating-point DSPs is:<br>(a) The type of data processed (b) The way they handle precision and range of data<br>(c) The speed of computation (d) The power consumption of the DSP | *1* | *K1* | *CO5* |
| 18. | In DSPs, the addressing mode that allows accessing memory relative to a base register is called:<br>(a) Immediate addressing (b) Indirect addressing<br>(c) Indexed addressing (d) Direct addressing | *1* | *K1* | *CO5* |
| 19. | In DSP terminology, MAC stands for:<br>(a) Multiply and Cache (b) Multiply and Compare<br>(c) Multiply and Accumulate (d) Multi-Address Communication | *1* | *K1* | *CO5* |
| 20. | What is the primary purpose of a Digital Signal Processor (DSP)?<br>(a) Image compression (b) General-purpose computing<br>(c) High-speed numerical computation on digital signals (d) File management | *1* | *K1* | *CO5* |

## PART - B (10 × 2 = 20 Marks)

Answer ALL Questions

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| 21. | State the classification of discrete time signals. | 2 | K1 | CO1 |
| 22. | Define aliasing effect. | 2 | K1 | CO1 |
| 23. | Determine the Z-Transform of $x(n) = a^n u(n)$. | 2 | K2 | CO2 |
| 24. | State convolution property of Z- Transform. | 2 | K1 | CO2 |
| 25. | Find the DFT of the sequence x(n) = (1 1 0 0) using direct computation method. | 2 | K1 | CO3 |
| 26. | How many multiplication and additions required computing 8- point DFT using radix-2 FFT? | 2 | K1 | CO3 |
| 27. | Write the equation for Hamming window function. | 2 | K2 | CO4 |
| 28. | Describe the significance of pre-warping in filter design. | 2 | K2 | CO4 |
| 29. | Compare a Digital Signal Processor with a general-purpose processor. | 2 | K2 | CO5 |
| 30. | Identify the main components of DSP architecture. | 2 | K1 | CO5 |

## PART - C (6 × 10 = 60 Marks)

Answer ALL Questions

31\. a) Check whether the following systems are linear and time varying: *10 K1 CO1*

y(n) = n x(n)
$y(n) = nx^2(n)$
$y(n) = x^2(n)$
y(n)= Ax(n) +B
y(n) = x(2n)

**OR**

b) Check whether the following signals are energy or power signal. *10 K1 CO1*

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

(b) $x(n) = \sin\left(\frac{\pi}{4}\right)^n$

32\. a) Determine the inverse Z- Transform of $x(z) = \dfrac{1}{(z-025)(z-0.5)}\ RoC: |z| > 0.5$ *10 K2 CO2*

**OR**

b) Determine the response of the system and check for stability *10 K2 CO2*
y(n) = 0.7y(n-1) – 0.12y(n-2)+x(n-1) + x(n-2) and x(n) = n.u(n).

33\. a) Compute 8 point DFT of the sequence x(n)={ 1 2 3 4 5 6 7 8 } using radix-2 DIF-FFT algorithm. *10 K3 CO3*

**OR**

b) Compute the circular convolution (DFT property) of the following sequences using the DFT and IDFT approach $x_1(n) = \{1,0,2,1\}$, $x_2(n) = \{1,1,1,1\}$. *10 K3 CO3*

34\. a) The specification of the desired Low Pass Digital Filter is *10 K3 CO4*

$0.707 \le |H(\omega)| < 1.0\ ;\ 0 \le \omega \le 0.2\pi$

$|H(\omega)| \le 0.1\ ;0.5\pi \le \omega \le \pi$ Assume T = 1 sec

Design a Chebyshev digital Low Pass Filter for the above spec. using Bilinear transformation Technique (BLT).

**OR**

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| | b) | A low pass Filter is required to be designed with the desired frequency response $$H_d(e^{jw}) = \begin{cases} e^{-j3\omega} & for\ -\dfrac{3\pi}{4} \le \omega \le \dfrac{3\pi}{4} \\ 0 & for\ \dfrac{3\pi}{4} \le \omega \le \pi \end{cases}$$ Obtain the filter coefficients h(n) using Hamming window function for N=7. | *10* | *K3* | *CO4* |
| 35. | a) | Describe in detail the architectural aspects of TMS320C54 digital signal processor using illustrative block diagram. | *10* | *K2* | *CO5* |
| | | **OR** | | | |
| | b) | Describe different addressing formats used in DSP processors. | *10* | *K2* | *CO5* |
| 36. | a) | Design a Digital Filter equivalent to $H(s) = \dfrac{2}{(s^2 + 3s + 2)}$ using Impulse Invariant method (T = 0.2 sec). | *10* | *K3* | *CO4* |
| | | **OR** | | | |
| | b) | Explain Von Neumann, Harvard architecture and modified Harvard architecture for the computer. | *10* | *K2* | *CO4* |