**Seat No./Enrollment No.: _____**

# GUJARAT TECHNOLOGICAL UNIVERSITY
## BACHELOR OF ENGINEERING – SEMESTER 7 – EXAMINATION – SUMMER 2026

**Subject Code: 2171003** **Date: 20/05/2026**

**Subject Name: Digital Signal Processing**

**Time: 02:30 PM TO 05:00 PM** **Total Marks: 70**

**Instructions:**

1. **Attempt all questions.**
2. **Make suitable assumptions wherever necessary.**
3. **Figures to the right indicate full marks.**
4. **Simple and non-programmable scientific calculators are allowed.**

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|---|---|---|---|
| **Q.1** | **(a)** | Define discrete-time signal and classify signals with suitable examples. | **03** |
| | **(b)** | Explain Sampling theorem and aliasing effect with neat sketch. | **04** |
| | **(c)** | A continuous-time signal $x(t) = \sin(200\pi t) + \sin(600\pi t)$ is sampled at 400 Hz. Determine whether aliasing occurs. Justify. | **07** |
| **Q.2** | **(a)** | Define Z-transform and state any two of its properties. | **03** |
| | **(b)** | Determine the Z-transform and ROC of $x[n] = (0.5)^n u[n]$. | **04** |
| | **(c)** | For a system given by $y[n] - \frac{3}{4}y[n-1] + \frac{1}{8}y[n-2] = x[n] + 2x[n-1]$, find the system function $H(z)$ and comment on its stability. | **07** |
| | | OR | |
| | **(c)** | Compute the linear convolution of $x[n] = \{1,2,1\}$ and $h[n] = \{1,-1,2\}$. | **07** |
| **Q.3** | **(a)** | Define frequency response of an LTI system. | **03** |
| | **(b)** | Explain minimum-phase, maximum-phase and all-pass system | **04** |
| | **(c)** | For $h[n] = \{1,-1\}$ find $H(e^{j\omega})$ and sketch magnitude/phase. | **07** |
| | | OR | |
| **Q.3** | **(a)** | Define DTFT and list properties. | **03** |
| | **(b)** | Discuss the relationship between magnitude and phase of $H(e^{j\omega})$ | **04** |
| | **(c)** | Compute the DTFT of $x[n]=(1/2)^n u[n]$ and derive its magnitude and phase. | **07** |
| **Q.4** | **(a)** | Compare IIR and FIR filters. | **03** |
| | **(b)** | Explain bilinear transformation method for IIR digital filter Design. | **04** |
| | **(c)** | Design a digital low-pass filter using impulse invariance for analog filter $H(s) = \frac{1}{s+2}$, with sampling time T=1 s. | **07** |
| | | OR | |
| **Q.4** | **(a)** | Explain the effect of coefficient quantization in digital filters. | **03** |
| | **(b)** | Describe window-based FIR filter design and explain types of windows. | **04** |
| | **(c)** | Explain the need for FFT and describe radix-2 decimation-in-time algorithm. | **07** |

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| **Q.5** | **(a)** | Explain the process of interpolation and decimation in multirate DSP. | **03** |
| | **(b)** | Discuss the role of adaptive filters in system identification and echo cancellation. | **04** |
| | **(c)** | Explain the architecture of a DSP processor showing Harvard architecture and MAC unit. | **07** |

OR

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|---|---|---|---|
| **Q.5** | **(a)** | Derive the expression for sampling rate conversion by a rational factor L/M. | **03** |
| | **(b)** | Explain basic principles of forward linear predictive filtering with applications. | **04** |
| | **(c)** | Explain DSP applications in speech or image processing with a neat block diagram. | **07** |

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