Code No.: EC503PC

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**CMR ENGINEERING COLLEGE: : HYDERABAD**
**UGC AUTONOMOUS**
**III–B.TECH–I–Semester End Examinations (Supply) – December 2024**
**DIGITAL SIGNAL PROCESSING**
**(ECE)**

**[Time: 3 Hours]** **[Max. Marks: 70]**

**Note:** This question paper contains two parts A and B.
Part A is compulsory which carries 20 marks. Answer all questions in Part A.
Part B consists of 5 Units. Answer any one full question from each unit. Each question carries 10 marks.

## PART-A (20 Marks)

1. a) Find the periodicity of the signal $x(n) = \sin(4\pi n / 2) + \cos(\pi n /4)$. [2M]
   b) What is meant by causal system? [2M]
   c) List out the any two properties of DFT. [2M]
   d) Explain the 2 Point DIT FFT. [2M]
   e) What is Warping effect? [2M]
   f) What is Gibbs phenomenon? [2M]
   g) Explain the Hamming window. [2M]
   h) Compare between Butterworth with Chebychev filters. [2M]
   i) Define the sampling rate conversion. [2M]
   j) How to prevent overflow in design of digital filters? [2M]

## PART-B (50 Marks)

2. Determine the impulse response of the system given by the difference equation. [10M]

$$y(n) = -3y(n-1)+x(n) - 2x(n-1)-4y(n-2) \quad \text{for } n\geq0$$

**OR**

3. Check for following systems is linear, causal, time in variant, stable, static. [10M]

   i) $y(n) = x(3n)+x(n-2)$.

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

4. Develop a radix -2 DIT FFT algorithm and draw the flow graph for computation N=8. [10M]

**OR**

5. Draw the flowgraph of the radix-2 DIFFFT algorithm for N=8. [10M]

6. A digital low pass filter is required to meet the following specification. [10M]
   Passband ripple: $\leq$ 4.436 dB
   Passband edge : $0.35\Pi$ rad/sample
   Stopband attenuation : $\geq$20 dB
   Stopband edge : $0.7\Pi$ rad/sample
   Sample rate: T=0.1sec
   The filter is to be design by Butterworth procedure and performing a bilinear transformation on an analog system function.

**OR**

7. Convert the analog filter with system function $H(s)=(s+0.1)/((s+0.1)^2 +9)$ into a digital IIR filter by means of the impulse invariance method. [10M]

8. Prove that an FIR filter has linear phase if the unit sample response satisfies the condition $h(n) = \pm h(M-1-n)$, $n = 0,1,\ldots M-1$. [10M]

**OR**

9. Explain the principle and procedure for designing FIR filter using rectangular window and Compare FIR and IIR Filters [10M]

10. What are limit cycles? And also Explain the Decimation by a Factor D. [10M]

**OR**

11. Explain the interpolation by a Factor I. [10M]

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