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## CMR ENGINEERING COLLEGE: : HYDERABAD UGC AUTONOMOUS

## III-B.TECH-I-Semester End Examinations (Regular) - January- 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)	What is meant by zero padding?	[2M]
b)	What is meant by causal system?	[2M]
c)	Distinguish between DFT and Radix-2 FFT algorithm	[2M]
d)	What is the value of twiddle factor $W_8^2$ .	[2M]
e)	What is Warping effect?	[2M]
f)	What is Gibbs phenomenon?	[2M]
g)	Explain the Rectangular window.	[2M]
h)	Compare between Butterworth and Chebychev filters.	[2M]
i)	Define the interpolator.	[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]
	$y(n) = -\frac{1}{2}y(n-1) + x(n)$ for $n \ge 0$	
	OR	
3.	Check for following systems is linear, time invariant.	[10M]
	i) $y(n) = x(3n) + x(n-2)$	
	ii) $y(n) = cos(x(n))$	
4.	Draw the signal flowgraph structure of radix -2 DIT FFT algorithm for N=8.	[10M]
	OR	
5.	Draw the signal flowgraph structure of the radix-2 DIF FFT algorithm for N=8	[10M]
6.	Design a digital low pass Chebychev filter for the following specification	[10M]
	Passband ripple: $\leq 0.5 \text{ dB}$ ,	
	Passband edge :1.2kHz,	
	Stopband attenuation :≥40 dB,	
	Stopband edge :2KHz,	
	Sample rate:8kHz,	
	The filter is to be design by performing a bilinear transformation on an analog sy	ystem
	function.	
7.	Convert the analog filter with system function $H(s)=(s+0.1)/((s+0.1)^2 +9)$ i	nto a [10M]
7.	digital IIR filter by means of the impulse invariance method.	ino a [10141]
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8. Prove that an FIR filter has linear phase if the unit sample response satisfies the condition h(n) = ± h(M-1-n), n =0,1,.... M-1. Also discuss symmetric and antisymmetric cases of FIR filter.
OR
9. Realize the given system in direct form-I. y[n] = 0.5y[n-1] - 0.25y[n-2] +x[n]+ 0.4 x[n-1].

10. Explain the Decimation by a Factor D. [10M]

11. Explain in detail about zero limit cycle oscillations in digital filters. [10M]