	ii.	What is aliasing and how it can be overcomed?	2	4	1	1
27.	а.	Compute the 8-point DFT of the sequence $x(n) = \{1,1,1,1,2,2,2,2\}$ using radix – 2 DIT FFT algorithm.	10	3	2	2
		(OR)				
	b.	Obtain the direct form II and cascade from realization for the system $y(n) = -0.1y(n-1) + 0.2y(n-2) + 3x(n) + 3.6x(n-1) + 0.6x(n-2)$ .	10	4	2	2
28.	. a.	Design a digital FIR filter which is having the frequency response	10	3	3	3
		$H\left(e^{j\omega}\right) = 1$ for $\frac{-\pi}{2} \le \frac{\pi}{2}$				
		0 otherwise				
		Find the values of h(n) for N=11 and also find H(z).				
		(OR)				
	b.	Using Hamming window, design a digital FIR filter which is having the frequency response.	10	3	3	3
		$H(e^{j\omega}) = e^{-j5\omega}$ for $\frac{-\pi}{4} \le  \omega  \le \frac{\pi}{4}$	3			
		0 otherwise				
		Find the values of h(n) for N=7 and also find H(z).				
29	. a.	Design a Butterworth digital IIR low pass filter using impulse invariant transformation to satisfy the following specification. $w_p = 0.3\pi, w_s = 0.75\pi, \epsilon = 1, \lambda = 4.898.$	10	3	4	3
		(OR)				
	b.	Design a Chebyshev digital IIR low pass filter using Bilinear transformation to satisfy the following specification $w_p = 0.2\pi$ , $w_s = 0.3\pi$ , $\alpha_p = 1dB$ , $\alpha_s = 15dB$ .	10	3	4	3
30	. a.	Explain about the polyphase structure of decimeter and interpolator.	10	4	6	2
1	b.i.	(OR) Point out the advantages of multirate DSP.	2	4	5	2
	ii.	Explain about filter banks of multirate DSP.	8	4	6	2
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		* * * * * *				

Reg. No.

## **B.Tech. DEGREE EXAMINATION, MAY 2022**

Fifth Semester

## 18ECC204J - DIGITAL SIGNAL PROCESSING

(For the candidates admitted from the academic year 2018-2019 to 2019-2020)

TAT	-4-	
Len 1	NTA:	•

- (i) Part A should be answered in OMR sheet within first 40 minutes and OMR sheet should be handed over to hall invigilator at the end of 40<sup>th</sup> minute.
  - Part B should be answered in answer booklet.

Time: 2½ Hours	Max.	Ma	rks:	75
$PART - A (25 \times 1 = 25 Marks)$	Marks		со	
Answer ALL Questions  1. Original signal can be retracted from sampled version using (A) Band pass filtering (B) Band stop filtering (C) Low pass filtering (D) High pass filtering	1	1	1	1
2. should be done in order to convert a continuous – time signal to discrete time signal.  (A) Integrating (B) Differentiating (C) Amplification (D) Sampling	1	1	1	1
3. The quantization will be finer when (A) Smaller the number of discrete (B) Larger the number of discrete amplitudes amplitudes	1	1	1	1
<ul> <li>(C) Does not depend on (D) Depends on frequency amplitudes</li> <li>4. Identify the disadvantage of digital signal processing.</li> <li>(A) Flexible in operation (B) Operation speed is limited</li> <li>(C) Cost is low for large scale (D) Parallel execution</li> </ul>	1	1	1	1
<ul> <li>5. When a signal is sampled at the rate of f<sub>x</sub> = 5f<sub>m</sub>, the sampling done is called as?</li> <li>(A) Over sampling</li> <li>(B) Down sampling</li> <li>(C) Perfect sampling</li> <li>(D) Zero sampling</li> </ul>	1	2	1	1
6. DFT is applied to (A) Infinite sequences (C) Continuous infinite signals (B) Finite discrete sequence (D) Continuous finite sequence	1	1	2	1
<ul> <li>7. The circular convolution of two sequence in time domain is equivalent to         <ul> <li>(A) Multiplication of DFTs of two</li> <li>(B) Summation of DFTs of two</li> <li>(C) Difference of DFTs of two</li> <li>(D) Square of multiplication of Sequences</li> </ul> </li> </ul>		2	2	1

(.	OIT algorithm divides the sequence in A) Positive and negative values  C) Small and large samples	(B) Upper higher and lower spectrum (D) Even and odd samples	1	2	2	1		19.	filter has pass-band ripple but a steeper roll-off rate.  (A) Chebysev – II (B) Elliptic  (C) Chebhysev – I (D) Bessel	1	2	4	3
9. ) ( ( ( 10. T	(n) is obtained by of in A) Integration  C) Convolution	put $x(n)$ of the system with $h(n)$ .  (B) Correlation  (D) Differentiation  clications required to compute N point  (B) $N\log_2 N$		1				20.	The IIR filter design involves  (A) Designing of analog filter in (B) Designing of digital filter in analog domain and transforming into digital transforming into digital domain  (C) Designing of analog filter in (D) Designing of digital filter in digital domain and digital domain and transforming transforming into analog into analog domain	1	2	4	3
11. I	vindow' when $\alpha = 0.5$ ? A) Hamming	(D) N-1  also regarded as 'Raised - cosine  (B) Barlett (D) Blackman	1	2	3	1		21.	In direct form realization for an interpolator, which among the following generates an intermediate signal,  (A) Upsampler  (B) Down sampler  (C) Anti-imaging filter  (D) Anti-aliasing filter	1	1	5	1
12. T	The impulse response of a symmetric A) $h(n) = h(N-1)/2$ C) $h(n) = h(-n)/2$		1	2	3	2		22.	Sampling rate conversion by the rational factor L/M is accomplished by what connection of interpolator and decimator.  (A) Parallel  (B) Cascade  (C) Direct  (D) Convolution	1	2	6	2
s (.	To reduce side lobes, in which pecifications have to be optimized.  A) Stop band  C) Transition band	region of the filter the frequency  (B) Pass band (D) Cut off	1	2	3	2			If $x(n) = \{1, 3, 2, 5, -1, -2, 2, 3, 2, 1\}$ apply 3 fold down sampler. (A) $\{1, 2, -1, -2, 2\}$ (B) $\{1, 3, 2, 5, -1, -2, 2, 3, 2, 1\}$ (C) $\{1, 5, 2, 1\}$ (D) $\{1, 3, 5, -2, 3, 1\}$	1	2	6	2
15. A	The FIR filter is a filter.  A) Unstable  C) Non – causal  Abrupt truncation of infinite series is  A) Windowing	(B) Stable (D) Recursive  done in (B) Frequency sampling	1	2	3				Alias free and proper reconstruction is achieved by having and  (A) Linear phase and all pass filter (B) Non linear phase and all pass filter  (C) Linear phase and low pass (D) Linear phase and high pass	1	1	6	1
16. T	C) Aliasing The locus of all the poles of Chebysh A) Circle C) Ellipse	(D) Foorier series	1	1	4	1		25.	filter  Decimation is a process in which the sampling rate is  (A) Enhanced (B) Stable (C) Reduced (D) Unpredictable	1	1	6	1
t:	ransformation is a mapping from.  A) Z-plane to S-plane	ilinear transformation, the bilinear  (B) J-plane to W-plane	1	1	4	1		26 a i	PART – B (5 × 10 = 50 Marks) Answer ALL Questions  Explain how shifted samples are produced by taking fourier transform and	Marks 8	BL 4	<b>co</b>	PO 4
18. 7		(B) Sampling the impulse response	1	2	4	3	,		state sampling theorem.  Point out the applications of DSP.	2	4	1	1
(	for the derivative C) Mapping from S-domain to Z-domain	of an equivalent analog filter (D) Approximation of derivatives						b.i.	(OR) Explain the block diagram of DSP.	8	4	1	1