Register Number:……..

KUMARAGURU
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B.E. DEGREE EXAMINATIONS: APRIL / MAY 2023

(Regulation 2018)

Sixth Semester

ELECTRONICS AND INSTRUMENTATION ENGINEERING

U18EII6202: Digital Signal Processing and Deep Learning

COURSE OUTCOMES

CO1: Understand the characteristics of discrete-time signals and discrete systems
CO2: Analyze signal/system properties using mathematical tools
CO3: Apply and develop algorithms for digital systems
CO4: Illustrate efficient computation of DFT
CO5: Discuss advanced features and architecture of generic P-DSP
CO6: Design FIR and IIR filters

Time: Three Hours **Maximum Marks: 100**

Answer all the Questions:-

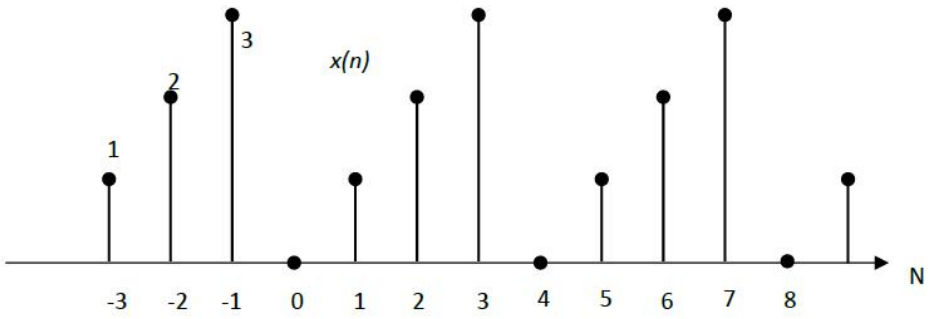
PART A (10 x 2 = 20 Marks)

(Answer not more than 40 words)

1.	Distinguish between deterministic and random signals.	CO1	[K2]
2.	Determine whether $sin(0.02\pi n)$ discrete-time signal is periodic or not. If periodic, determine the fundamental period	CO1	[K2]
3.	A system is defined by a function, $y(n) = ev\,[x(n)]$, find out whether this system is causal or non-causal.	CO2	[K3]
4.	If $x(n) \xleftrightarrow{ZT} X(z), with\ ROC = R$, $then\ find\ out\ what\ is\ the\ X(z) for\ x_k(n) = x\left(\frac{n}{k}\right)$ Where n is the integer multiples of k, and ZERO otherwise	CO2	[K3]
5.	State the Computational requirements of FFT?	CO3	[K2]
6.	For N-point DFT computation by DIT or DFT , how many butterflies are there per stage?	CO3	[K2]
7.	What is the drawback of discrete-time Fourier transform and how is it overcome?	CO4	[K4]
8.	Why linear convolution is important in Digital Signal Processing (DSP)?	CO5	[K2]
9.	How is a digital filter designed?	CO6	[K2]
10.	What are the major reasons for the success of windows/ windowing technique in FIR filters?	CO5	[K2]

Answer any FIVE Questions:-
PART B (5 x 16 = 80 Marks)
(Answer not more than 400 words)

11.	a)	Suppose the voice signal of a singer $x(n)$ is passed through an LTI electrical system with response $h(n)$ to produce a new sound signal. What will be the output produced? *Note:* The notation used in the options here uses underline to indicate the signal value $n = 0$, and other values in the sequence. The value of the signal that are not mentioned are implied to be zero. $x(n) = [0,1,\underline{\mathbf{1}},1,1,1,1,0]$ and $h(n) = [0,\underline{\mathbf{1}},1,1,1,0]$	8	CO1	[K3]
	b)	Find the circular convolution of two finite duration sequences. $x_1(n) = \{1,2,-1,-2,3,1\}$, $x_2(n) = \{3,2,1\}$	8	CO1	[K3]
12.	a)	We want to design a causal discrete time LTI system with the property that if the input $x(n) = \left(\frac{1}{3}\right)^n u(n) - \frac{1}{5}\left(\frac{1}{3}\right)^{n-1} u(n-1)$, then the output is $y(n) = \left(\frac{1}{2}\right)^n u(n)$. Determine the transfer function *H(z),* the impulse response *h(n)* and frequency response *H(w)* of the system that satisfies this condition. Assess the stability of the system.	12	CO2	[K3]
	b)	Find the Z-transform and ROC of the sequence $x(n) = \{2,1,-3,0,\mathbf{4},3,2,1,5\}$ *Note:* The notation used in the options here uses underline/ bold to indicate the signal value $n = 0$, and other values in the sequence	4	CO2	[K3]
13.	a)	Find the exponential form of the DFS representation of *x(n)* given here.	10	CO3	[K3]



With suitable example show that DIT/DIF FFT is efficient than Direct DFT

	b)	Given a sequence $x(n) = \{0,1,2,3,4,5,6,7\}$, determine *X(k)* using DIT FFT algorithm.	6	CO4	[K3]
14.	a)	Design a high pass FIR filter for the following specifications. $$cutoff\ frequency\ = 500\ Hz$$ $$Sampling\ frequency\ = 2000\ Hz$$ $$and\ \ N = 11$$	10	CO6	[K3]
	b)	A low pass filter is to be designed with the following desired frequency response. $$H_d(e^{j\omega}) = \begin{cases} e^{-j2\omega}, & -\frac{\pi}{4} \le \omega \le \frac{\pi}{4} \\ 0, & \frac{\pi}{4} \le |\omega| \le \pi \end{cases}$$ Determine the coefficients h(n) if the window function is defined as $$w(n) = \begin{cases} 1, & 0 \le n \le 4 \\ 0, & otherwise \end{cases}$$	6	CO6	[K3]
15.	a)	Design a IIR low pass filter with passband edge at 1000Hz and stopband edge at 1500 Hz for a sampling frequency of 5000 Hz. The filter is to have a passband ripple of 0.5dB and a stopband ripple below 30 dB. Design a Butterworth filter using the bilinear transformation	10	CO6	[K3]
	b)	Determine H(z) using the impulse invariant technique for the analog filter function. $$H_a(s) = \frac{1}{(s+1)(s^2+s+2)}$$	6	CO6	[K3]
16.	a)	Why circular buffering is used in DSP processor? Explain with suitable example about the concept of circular buffering	8	CO5	[K3]
	b)	With respect to DSP Processor explain the Harvard architecture and modified Harvard architecture	8	CO5	[K3]

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