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P.E.S. College of Engineering, Mandya - 571 401

(An Autonomous Institution affiliated to VTU, Belagavi)

Fourth Semester, B.E. - Electronics and Communication Engineering
Semester End Examination; August - 2023
Digital Signal Processing

Time: 3 hrs *Max. Marks: 100*

Course Outcomes

The Students will be able to:
*CO1: **Apply** mathematical knowledge to understand DFT, FFT and Filters*
*CO2: **Analyze** discrete systems using DFT, FFT and filtering formulation*
*CO3: **Design** the FIR & IIR filters for given specification*
*CO4: **Implement** the discrete–time systems using various approaches*
*CO5: **Understand** role of DSP in various applications*

Note: ***I) PART - A** is compulsory. **Two** marks for each question.*
***II) PART - B**: Answer any **Two** sub questions (from a, b, c) for a Maximum of **18 marks** from each unit.*

Q. No.	Questions	Marks	BLs	COs
	I : PART - A	**10**		
1 a.	Compute the 4 points circular convolution of $x[n] = \{1, 2, 3\}$ and $h[n] = \{2, 4\}$ using matrix method.	2	L2	CO1
b.	Calculate the number of addition and multiplication required to compute the 64 point DFT of Sequence using Radix-2 FFT method.	2	L2	CO2
c.	Differentiate between Hamming and Hanning window.	2	L3	CO1
d.	Explain the relationship between analog and digital frequency in Bilinear transformation.	2	L1	CO1
e.	List four stages involved in processing of biomedical signals.	2	L1	CO5
	II : PART - B	**90**		
	UNIT - I	**18**		
2 a.	Given a filter has impulse response $h(n) = \{3, 2, 1\}$ and input $x(n) = \{2 ,1, -1, -2, -3, 5 ,6, -1, 2, 0, 2, 1$. Determine the output $y(n)$ of the filter using overlap-save method for a block length of 5.	9	L2	CO1
b.	State and prove the following properties of DFT: i) Circular convolution ii) Parseval's relation	9	L2	CO2
c.	An FIR filter has impulse response $h(n) = \{1, 2, 3\}$. Find the output of the filter for the input $x(n) = \{1, 2\}$, using DFT and IDFT method.	9	L3	CO2
	UNIT - II	**18**		
3 a.	Compute the 8 point DFT of the sequence $x(n) = (2, 1, 2, 1, 1, 2, 1, 2\}$ using radix 2 DITFFT.	9	L2	CO3
b.	Develop the chart of DIF – FIT algorithm for a 8 point sequence starting from basic definition DFT.	9	L2	CO2

c.	Compute the IDFT of sequence $x(k) = \{0, 2+j2, -j4, 2-j2, 0, 2+j2, j4, 2-j2\}$ using Radix-2 algorithm.	9	L4	CO3
	UNIT - III	**18**		
4 a.	Design a linear phase FIR high pass filter using hamming window with a cut off frequency of $w_c = 0.8\pi$ rad/sample and N = 7, considering symmetric impulse response, $H_d(e^{jw}) = \begin{cases} e^{-jw\alpha}; & for\ -\pi \le w \le -w_c \quad and\ w_c \le w \le \pi \\ 0 & otherwise \end{cases}$	9	L2	CO3
b.	Design a FIR low pass filter with cut off frequency of 1 kHz and sampling frequency of 4 kHz with 11 samples using Fourier series method and implement the filter structure.	9	L2	CO3
c.	Design an ideal differentiator with frequency response $H(e^{jw}) = jw \quad -\pi \le w \le \pi$ using (i) rectangular window with N = 7 and determine the transfer function.	9	L3	CO3
	UNIT - IV	**18**		
5 a.	Design a digital Butterworth High pass filter using bilinear transformation by taking $T = 0.1$ second, to satisfy the following specification, $0.6 \le \lvert H(e^{jw})\rvert \le 1.0$; for $0.7\pi \le w \le \pi$; $\lvert H(e^{jw})\rvert \le 0.1$; for $0 \le w \le 0.35\pi$	9	L3	CO3
b.	Design a Chebyshev low pass filter with the ripple of 1 dB in the pass band $0 \le w \le 0.2\pi$ and attenuation of at least 15 dB in the stop band $0.3\pi \le w \le \pi$, using impulse invariant method.	9	L3	CO3
c.	Derive an expression for; i) Order of Butterworth filter N; ii) Cut off frequency Ω_c of Butterworth LPF	9	L2	CO3
	UNIT - V	**18**		
6 a.	Realize the following system function in Direct form-I, Direct II and cascade from, $y(n) = -0.1\,y(n-1) + 0.2y(n-2) + 3x(n) + 3.6x(n-1) + 0.6x(n-2)$.	9	L2	CO4
b.	Realize the following system in parallel from, $y(n) = 0.75\,y(n-1) - 0.125y(n-2) + 6x(n) + 7x(n-1) + x(n-2)$	9	L1	CO5
c.	With a basic block diagram, explain DSP based video signal processing system.	9	L3	CO5

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