Course Code: 20EC0417 **R20**

SIDDHARTH INSTITUTE OF ENGINEERING & TECHNOLOGY:: PUTTUR (AUTONOMOUS)
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Subject with Code: DIGITAL SIGNAL PROCESSING (20EC0417)
Course & Branch: B.Tech – ECE
Regulation: R20
Year & Sem: III-B.Tech. & I-Sem.

UNIT –I
DISCRETE FOURIER TRANSFORM (DFT) & FAST FOURIER TRANSFORM (FFT)

1	a) What is DFT? Give its significance with necessary equations.	[L1][CO1]	**[2M]**
	b) What is the purpose of IDFT, write its mathematical equation.	[L2][CO1]	**[2M]**
	c) Explain the relationship between DFT with other transforms.	[L2][CO1]	**[4M]**
	d) Compute the 4-point DFT for the sequence $x(n) = \begin{cases} 1; 0 \le n \le 2 \\ 0; otherwise \end{cases}$	[L3][CO1]	**[4M]**
2	a) Determine the 8 point DFT of the sequence $x(n) = \{1,1,1,1,1,1,1,0\}$.	[L3][CO1]	**[8M]**
	b) Find the IDFT of the sequence $X(K) = \{1,0,1,0\}$.	[L3][CO1]	**[4M]**
3	a) List the properties of DFT.	[L1][CO1]	**[2M]**
	b)State and Prove any Four properties of DFT.	[L3][CO1]	**[10M]**
4	a) Write the significance of DFT in linear filtering.	[L2][CO1]	**[4M]**
	b) Find the linear convolution of the sequences $x(n)$ and $h(n)$ using DFT. $x(n) = \{1,0,2\}$, $h(n) = \{1,1\}$	[L3][CO1]	**[8M]**
5	a) What is the purpose of filtering of long duration sequences? List the methods for filtering of long duration sequences.	[L1][CO1]	**[4M]**
	b) Evaluate the output $y(n)$ of a filter whose impulse response is $h(n) = \{1,1,1\}$ and input signal $x(n) = \{3,-1,0,1,3,2,0,1,2,1\}$ using overlap save method.	[L5][CO1]	**[8M]**
6	Evaluate the output y(n) of a filter whose impulse response is $h(n) = \{1,2\}$ and input signal $x(n) = \{1,2,-1,2,3,-2,-3,-1,1,1,2,-1\}$ using overlap save method and overlap add method.	[L5][CO1]	**[12M]**
7	a) How FFT improves the speed of computation? Find the number of multiplication and additions required in an 8-point radix-2 FFT.	[L3][CO1]	**[4M]**
	b) Explain the steps in Decimation in Time FFT algorithm with necessary diagram.	[L2][CO1]	**[4M]**
	c) Explain the steps in Decimation in Frequency FFT algorithm with necessary diagram.	[L2][CO1]	**[4M]**
8	Compute 8-point DFT of the sequence $x(n) = \{1,2,3,4,4,3,2,1\}$ using Radix-2 DIT-FFT Algorithm.	[L3][CO1]	**[12M]**
9	Compute DFT of the sequence $x(n) = \{1,1,1,1,1,1,1,0\}$ using Radix-2 DIT FFT algorithm.	[L3][CO1]	**[12M]**
10	Compute 8-point DFT of the sequence $x(n) = \{0,1,2,3,4,5,6,7\}$ using Radix-2 DIF-FFT Algorithm.	[L3][CO1]	**[12M]**

UNIT –II
INFINITE IMPULSE RESPONSE FILTERS & REALIZATION OF IIR FILTER

1	a) What are the basic types of filters and on what basis are they classified?	[L1][CO2]	**[2M]**
	b) List the filter types in designing the IIR filters?	[L1][CO2]	**[2M]**
	c) Explain the steps in the design of an analog Butterworth low pass filter.	[L2][CO2]	**[8M]**
2	a) Compare the Analog and Digital filters.	[L2][CO2]	**[2M]**
	b) Design an analog Butterworth filter that has 2 dB pass band attenuation at a frequency of 20 rad/sec and at least 10 dB stop band attenuation at 30 rad/sec.	[L3][CO2]	**[10M]**
3	a) Explain the steps in the design of an analog Chebyshev low pass filter.	[L2][CO2]	**[6M]**
	b) Design an analog filter using Chebyshev approximation for the specifications $\alpha_p = 3dB\ and\ \alpha_s = 16dB;\ f_p = 1KHz\ and\ f_s = 2KHz.$	[L3][CO2]	**[6M]**
4	a) Compare Butterworth and Chebyshev Filter.	[L2][CO2]	**[2M]**
	b) How a digital filter is designed? List the methods for converting analog filter TF to digital filter TF.	[L1][CO2]	**[2M]**
	c) Illustrate the conversion steps in Impulse Invariance & Bilinear transformation method?	[L3][CO2]	**[8M]**
5	a) For the analog transfer function $H(S) = \frac{2}{(S+1)(s+3)}$, Determine $H(Z)$ using Impulse Invariance method. Assume T=1 Sec.	[L3][CO2]	**[6M]**
	b) Apply Bilinear transformation to $H(S) = \frac{4}{(S+3)(s+4)}$ with T = 0.5 Sec and find $H(Z)$.	[L3][CO2]	**[6M]**
6	Design a digital Butterworth IIR filter satisfying the following constraints. Let T=1s, apply Impulse Invariant Transformation. $0.8 \le \lvert H(w)\rvert \le 1 \quad ;0 \le w \le 0.2\pi$ $\lvert H(w)\rvert \le 0.2 \quad ;0.32\pi \le w \le \pi$	[L3][CO2]	**[12M]**
7	Design a digital Chebyshev IIR filter satisfying the following constraints. Let T=1s, apply Bilinear transformation. $0.707 \le \lvert H(w)\rvert \le 1 \quad ;0 \le w \le 0.2\pi$ $\lvert H(w)\rvert \le 0.1 \quad ;0.5\pi \le w \le \pi$	[L3][CO2]	**[12M]**
8	a) Explain the frequency transformation technique in analog domain for converting low pass to low pass filter and low pass to high pass filter with frequency response.	[L2][CO3]	**[6M]**
	b) Transform the prototype low pass filter with following system function into a high pass filter with a cutoff frequency Ω_c^* $H(s) = \frac{\Omega_c}{s + 2\Omega_c}$	[L2][CO3]	**[6M]**
9	a) What are the basic elements used to construct the block diagram of a discrete time system? Draw their symbols.	[L1][CO2]	**[4M]**
	b) List the different types of structures for realization of IIR systems.	[L1][CO2]	**[2M]**
	c)Construct the Direct form I and Direct form II, of the LTI System described by the equation $y(n) = -\frac{3}{8}y(n-1) + \frac{3}{32}y(n-2) + \frac{1}{64}y(n-3) + x(n) + 3\,x(n-1)$	[L3][CO2]	**[6M]**
10	a)Construct the cascade form structure of the system with difference equation $y(n) = \frac{3}{4}y(n-1) - \frac{1}{8}y(n-2) + x(n) + \frac{1}{3}x(n-1)$	[L3][CO2]	**[6M]**
	b) Construct the parallel form structure of the system with difference equation $y(n) = -0.1y(n-1) + 0.72y(n-2) + 0.7x(n) - 0.252x(n-2)$	[L3][CO2]	**[6M]**

UNIT –III
FINITE IMPULSE RESPONSE FILTERS & REALIZATION OF FIR FILTER

1	a) What is FIR filter? Write the necessary and sufficient condition for the linear phase characteristic of a FIR filter?	[L1][CO3]	**[2M]**
	b) Explain the steps to be followed in designing FIR Filters using Fourier Series method.	[L2][CO3]	**[4M]**
	c) Design an FIR digital filter to approximate an ideal Low pass filter with pass band gain of unity, cutoff frequency of $1kHz$, and working at a sampling frequency $f_s = 4kHz$. The length of the impulse response should be 11. Use Fourier series method.	[L3][CO3]	**[6M]**
2	a) What is a window? Why it is necessary?	[L4][CO2]	**[2M]**
	b) Explain the Procedure for designing FIR filters using windows.	[L2][CO2]	**[4M]**
	c) Give the equations for Rectangular, Hanning and Hamming window and explain its significance.	[L2][CO2]	**[6M]**
3	a) A Low pass filter is to be designed with the following desired frequency response using rectangular window for N=11. $$H_d(e^{jw}) = 1 \; for -\frac{\pi}{2} \le w \le \frac{\pi}{2}$$ $$= 0 \qquad \frac{\pi}{2} \le \lvert w \rvert \le \pi$$ Determine the filter coefficients h(n) if the window function is defined as $$w(n) = 1 \; for \; -5 \le n \le 5$$ $$= 0 \quad otherwise$$ Also determine the frequency response H(z) of the designed filter.	[L3][CO3]	**[6M]**
	b)Design an ideal High pass filter with the frequency response $$H_d(e^{jw}) = 1 \; for \; \frac{\pi}{4} \le \lvert\omega\rvert \le \pi$$ $$= 0 \qquad \lvert\omega\rvert \le \frac{\pi}{4}$$ Find the values of h(n) for N=11 and find H(z).	[L3][CO3]	**[6M]**
4	Design a filter with following data, using a Hamming window with N=7. $$H_d(e^{jw}) = 1 \; for \; -\frac{\pi}{4} \le w \le \frac{\pi}{4}$$ $$= 0 \qquad \frac{\pi}{4} \le \lvert w \rvert \le \pi$$	[L3][CO3]	**[12M]**
5	Design an ideal High pass filter using Hanning window with the frequency response $$H_d(e^{jw}) = 1 \; for \; \frac{\pi}{4} \le \lvert\omega\rvert \le \pi$$ $$= 0 \qquad \lvert\omega\rvert \le \frac{\pi}{4}$$ Find the values of h (n) for N=11 and find H(z).	[L3][CO3]	**[12M]**
6	a) Compare Rectangular window and Hamming Window.	[L4][CO2]	**[3M]**
	b) Compare Rectangular window and Hanning Window.	[L4][CO2]	**[3M]**
	c) Write the design steps of FIR filter using Frequency sampling technique.	[L2][CO2]	**[6M]**
7	Compute the coefficients of a linear phase FIR filter of length N=15 which has a symmetric unit sample response and a frequency response that satisfies the conditions. $$H\left(\frac{2\pi k}{15}\right) = 1 \; for \; k = 0,1,2,3$$ $$= 0 \; for \; k = 4,5,6,7$$	[L3][CO3]	**[12M]**
8	a) List the types of structures for realizing the FIR systems.	[L1][CO3]	**[2M]**
	b) Draw the Linear Phase Structure and transversal structures for realizing the FIR filters and explain.	[L2][CO3]	**[10M]**
9	a) Construct the Direct form realization of system function. $$H(Z) = 1 + 2Z^{-1} - 3Z^{-2} - 4Z^{-3} + 5Z^{-4}$$	[L3][CO3]	**[6M]**
	b) Construct the cascade realization of the system function. $$H(Z) = 1 + \frac{5}{2}Z^{-1} + 2\,Z^{-2} + 2Z^{-3}$$	[L3][CO3]	**[6M]**

10	a) Realize the H(Z) with minimum number of multipliers $$H(Z) = 1 + \frac{1}{2}Z^{-1} + \frac{1}{8}Z^{-2} + \frac{3}{4}Z^{-3} + \frac{1}{8}Z^{-4} + \frac{1}{2}Z^{-5} + Z^{-6}$$	[L3][CO3]	**[6M]**
	b) Realize the second order FIR system $y(n) = 2\,x(n) + 4\,x(n-1) - 3x(n-2)$ by using transposed form structure.	[L3][CO3]	**[6M]**

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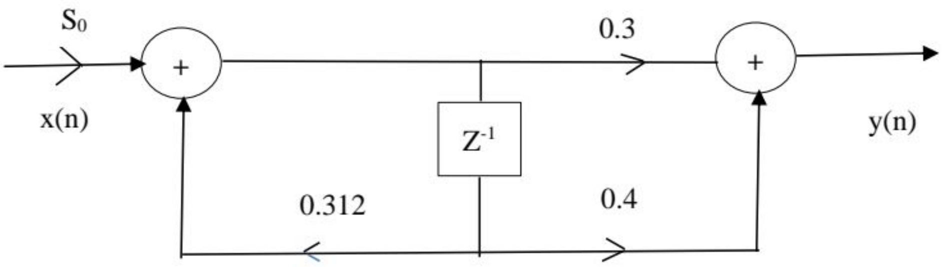
UNIT –IV
FINITE WORD LENGTH EFFECTS

1	a) Discuss briefly about different types of number representation with examples.	[L2][CO4]	**[6M]**
	b) Compare fixed point and floating point arithmetic.	[L4][CO4]	**[6M]**
2	a) Explain quantization noise and its methods with suitable example.	[L2][CO4]	**[6M]**
	b) Discuss in detail the errors resulting from rounding and truncation.	[L2][CO5]	**[6M]**
3	a) Draw and explain the power density functions for truncation and rounding.	[L1][CO5]	**[6M]**
	b) Discuss the various common methods of quantization.	[L2][CO4]	**[6M]**
4	a) Explain input Quantization Error and its effects with suitable example.	[L2][CO4]	**[6M]**
	b) What is coefficient quantization error? Explain its effects with suitable examples.	[L2][CO4]	**[6M]**
5	The output signal of an A/D converter is passed through a first order low pass filter with transfer function $H(Z) = \frac{(1-a)z}{(Z-a)}\ for\ 0 < a < 1$. Find the steady state output noise power due to quantization at the output of the digital filter.	[L3][CO5]	**[12M]**
6	Find the steady state variance of the noise in the output due to quantization of input for the first order filter. $y(n) = a\,y(n-1) + x(n)$.	[L3][CO5]	**[12M]**
7	Consider the transfer function $H(z) = H_1(Z).H_2(Z)$ where $H_1(Z) = \frac{1}{(1-a_1Z^{-1})}$ and $H_2(Z) = \frac{1}{(1-a_2Z^{-1})}$. Find the output round off noise power. Assume $a_1 = 0.5$ and $a_2 = 0.6$.	[L3][CO5]	**[12M]**
8	a) What is meant by zero limit cycle oscillation? Explain with example.	[L2][CO4]	**[6M]**
	b) Explain the characteristics of limit cycle oscillation with respect to the system described by the difference equation $y(n) = \alpha y(n-1) + x(n)$. Assume $\alpha = \frac{1}{2}$ data register length is 3 bits, the system is excited by an input $x(n) = \begin{cases} 0.875\ for\ n = 0 \\ 0\ for\ otherwise \end{cases}$. Also, determine the dead band of the filter.	[L3[CO5]	**[6M]**
9	a) What is meant by Overflow limit cycle oscillations? Explain with example.	[L2][CO4]	**[6M]**
	b) Find the characteristics of a limit cycle oscillation with respect to the system described by the difference equation $(n) = 0.97\,y(n-1) + x(n)$, Determine the dead band of the filter.	[L3][CO4]	**[6M]**
10	a) Explain Signal scaling for second order IIR filter with necessary mathematical expressions.	[L2][CO4]	**[6M]**
	b) Given $H(z)=\frac{0.5+0.4\,Z^{-1}}{1-0.312\,Z^{-1}}$ is the transfer function of a digital filter, find the scaling factor S_o to avoid overflow in adder 1 of the digital filter shown in fig. 	[L1][CO4]	**[6M]**

UNIT –V
INTRODUCTION TO DIGITAL SIGNAL PROCESSORS

1	a) Explain the two categories of DSP’s in detail.	[L2][CO6]	**[6M]**
	b) What are the advantages of the DSP processors over conventional microprocessors?	[L1][CO6]	**[6M]**
2	a) Explain the Multiplier and Multiplier Accumulator (MAC), Modified bus structures in brief with relevant diagram.	[L2][CO6]	**[8M]**
	b) What is VLIW architecture? Draw and explain in brief with diagram.	[L2][CO6]	**[4M]**
3	a) Explain the concept of multi access memory and multi ported memory.	[L2][CO6]	**[4M]**
	b) Illustrate on the various phases of Pipelining concept.	[L3][CO6]	**[8M]**
4	Draw the architecture of TMS320C50 and explain its important blocks.	[L2][CO6]	**[12M]**
5	a)What are the different parts in central processing units of TMS320C50 and explain its need in brief?	[L2][CO6]	**[4M]**
	b) Explain the bus structure of TMS320C50 and explain its need ?	[L2][CO6]	**[4M]**
	c) Explain the function of CALU and PLU in TMS320C50 in detail.	[L2][CO6]	**[4M]**
6	a) Explain On-Chip memory of TMS320C50 in details.	[L2][CO6]	**[6M]**
	b) Explain On-Chip Peripherals of TMS320C50 in details.	[L2][CO6]	**[6M]**
7	Draw and Explain the architecture of TMS320C54X digital signal processor in brief.	[L2][CO6]	**[12M]**
8	a) Draw and explain Arithmetic and logical unit (ALU) of TMS320C54x.	[L2][CO6]	**[6M]**
	b) What are the different buses of TMS320C54x and their functions?	[L1][CO6]	**[6M]**
9	a) Explain internal memory organization in TMS320C54x architecture.	[L2][CO6]	**[6M]**
	b) Explain the concept of overflow handling in TMS320C54x architecture.	[L2][CO6]	**[6M]**
10	Explain different applications of PDSPs in detail.	[L2][CO6]	**[12M]**

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