ECT303	DIGITAL SIGNAL PROCESSING	CATEGORY	L	T	P	CREDIT
		PCC	3	1	0	4

Preamble: This course aims to provide an understanding of the principles, algorithms and applications of DSP.

Prerequisite: ECT 204 Signals and systems

Course Outcomes: After the completion of the course the student will be able to

CO 1	State and prove the fundamental properties and relations relevant to DFT and solve basic problems involving DFT based filtering methods
CO 2	Compute DFT and IDFT using DIT and DIF radix-2 FFT algorithms
CO 3	Design linear phase FIR filters and IIR filters for a given specification
CO 4	Illustrate the various FIR and IIR filter structures for the realization of the given system function
CO5	Explain the basic multi-rate DSP operations decimation and interpolation in both time and frequency domains using supported mathematical equations
CO6	Explain the architecture of DSP processor (TMS320C67xx) and the finite word length effects

Mapping of course outcomes with program outcomes

	PO 1	PO 2	PO 3	PO 4	PO 5	PO 6	PO 7	PO 8	PO 9	PO 10	PO 11	PO 12
CO 1	3	3	2		2							2
CO 2	3	3	3		3							2
CO 3	3	3	3		3							2
CO 4	3	3	2		3							2
CO5	2	2	2		2							2
CO6	2	2	-		-							2

Assessment Pattern

Bloom's Category		**Continuous Assessment Tests**		**End Semester Examination**
		1	**2**	
Remember	K1	10	10	10
Understand	K2	20	20	30
Apply	K3	20	20	60
Analyse	K4			
Evaluate				
Create				

Mark distribution

Total Marks	CIE	ESE	ESE Duration
150	50	100	3 hours

Continuous Internal Evaluation Pattern:

Attendance : 10 marks
Continuous Assessment Test (2 numbers) : 25 marks
Assignment/Quiz/Course project : 15 marks

End Semester Examination Pattern: There will be two parts; Part A and Part B. Part A contain 10 questions with 2 questions from each module, having 3 marks for each question. Students should answer all questions. Part B contains 2 questions from each module of which student should answer any one. Each question can have maximum 2 sub-divisions and carry 14 marks.

Course Level Assessment Questions

CO1: State and prove the fundamental properties and relations relevant to DFT and solve basic problems involving DFT based filtering methods

1. Determine the N-point DFT X(k) of the N point sequences given by (i) $x1(n)=\sin(2\pi n/N)$ n/N) (ii) $x2(n)=\cos^2(2\pi n/N)$ n/N)

2. Show that if x(n) is a real valued sequence, then its DFT X(k) is also real and even

CO2: Compute DFT and IDFT using DIT and DIF radix-2 FFT algorithms

1. Find the 8 point DFT of a real sequence x(n)={1,2,2,2,1,0,0,0,0} using Decimation in frequency algorithm?

2. Find out the number of complex multiplications require to perform an 1024 point DFT using(i)direct computation and (ii) using radix 2 FFT algorithm?

CO3: Design linear phase FIR filters and IIR filters for a given specification

1. Design a linear phase FIR filter with order M=15 and cut-off frequency πn/N) /6 .Use a Hanning Window.

2. Design a low pass digital butter-worth filter using bilinear transformation for the given specifications. Passband ripple ≤1dB, Passband edge:4kHz, Stopband Attenuation:≥40 dB, Stopband edge:6kHz, Sampling requency:24 kHz

CO4: Illustrate the various FIR and IIR filter structures for the realization of the given system function

1. Obtain the direct form II and transpose structure of the filter whose transfer function is given below.

$$H(z)=\frac{0.44\,z^2+0.362\,z+0.02}{z^3+.4\,z^2+.18\,z-0.2}$$

2. Realize an FIR system with the given difference equation y(n)=x(n)-0.5x(n-1)+0.25x(n-2)+0.5x(n-3)-0.4x(n-4)+0.2x(n-5)

CO5: Explain the basic multi-rate DSP operations decimation and interpolation in both time and frequency domains using supported mathematical equations

1. Derive the frequency domain expression of the factor of 2 up-sampler whose input is given by x(n) and transform by X(k)?
2. Bring out the role of an anti-imaging filter in a sampling rate converter?

CO6: Explain the architecture of DSP processor TMS320C67xx and the finite word length effects

1. Derive the variance of quantization noise in an ADC with step size Δ, assuming uniformly distributed quantization noise with zero mean ?
2. Bring out the architectural features of TMS320C67xx digital signal processor?

SYLLABUS

Module 1

Basic Elements of a DSP system, Typical DSP applications, Finite-length discrete transforms, Orthogonal transforms – The Discrete Fourier Transform: DFT as a linear transformation (Matrix relations), Relationship of the DFT to other transforms, IDFT, Properties of DFT and examples. Circular convolution, Linear Filtering methods based on the DFT, linear convolution using circular convolution, Filtering of long data sequences, overlap save and overlap add methods, Frequency Analysis of Signals using the DFT (concept only required)

Module 2

Efficient Computation of DFT: Fast Fourier Transform Algorithms-Radix-2 Decimation in Time and Decimation in Frequency FFT Algorithms, IDFT computation using Radix-2 FFT Algorithms, Application of FFT Algorithms, Efficient computation of DFT of Two Real Sequences and a 2N-Point Real Sequence

Module 3

Design of FIR Filters - Symmetric and Anti-symmetric FIR Filters, Design of linear phase FIR filters using Window methods, (rectangular, Hamming and Hanning) and frequency sampling method, Comparison of design methods for Linear Phase FIR Filters. Design of IIRDigital Filters from Analog Filters (Butterworth), IIR Filter Design by Impulse Invariance, and Bilinear Transformation, Frequency Transformations in the Analog and Digital Domain.

Module 4

Structures for the realization of Discrete Time Systems - Block diagram and signal flow graph representations of filters, FIR Filter Structures: Linear structures, Direct Form, CascadeForm, IIR Filter Structures: Direct Form, Transposed Form, Cascade Form and Parallel Form, Computational Complexity of Digital filter structures. Multi-rate Digital Signal Processing:

Decimation and Interpolation (Time domain and Frequency Domain Interpretation), Anti- aliasing and anti-imaging filter.

Module 5

Computer architecture for signal processing: Harvard Architecture, pipelining, MAC, Introduction to TMS320C67xx digital signal processor, Functional Block Diagram.
Finite word length effects in DSP systems: Introduction (analysis not required), fixed-point and floating-point DSP arithmetic, ADC quantization noise, Finite word length effects in IIRdigital filters: coefficient quantization errors. Finite word length effects in FFT algorithms: Round off errors

Text Books

1. Proakis J. G. and Manolakis D. G., Digital Signal Processing, 4/e, Pearson Education, 2007
2. Alan V Oppenheim, Ronald W. Schafer ,Discrete-Time Signal Processing, 3rd Edition , Pearson ,2010

3. Mitra S. K., Digital Signal Processing: A Computer Based Approach, 4/e McGraw Hill (India) 2014

Reference Books

4. Ifeachor E.C. and Jervis B. W., Digital Signal Processing: A Practical Approach, 2/e Pearson Education, 2009.
5. Lyons, Richard G., Understanding Digital Signal Processing, 3/e. Pearson Education India, 2004.
6. Salivahanan S, Digital Signal Processing,4e, Mc Graw –Hill Education New Delhi, 2019
7. Chassaing, Rulph., DSP applications using C and the TMS320C6x DSK. Vol. 13. John Wiley & Sons, 2003.
8. Vinay.K.Ingle, John.G.Proakis, Digital Signal Processing: Bookware Companion Series,Thomson,2004
9. Chen, C.T., "Digital Signal Processing: Spectral Computation & Filter Design", Oxford Univ. Press, 2001.
10. Monson H Hayes, "Schaums outline: Digital Signal Processing", McGraw HillProfessional, 1999

Course Contents and Lecture Schedule

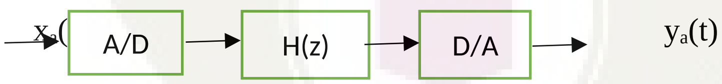
No.	Topic	No. of Lectures
1	**Module 1**	
1.1	Basic Elements of a DSP system, Typical DSP applications, Finite length Discrete transforms, Orthogonal transforms	1
1.2	The Discrete Fourier Transform: DFT as a linear transformation(Matrix relations),	1
1.3	Relationship of the DFT to other transforms, IDFT	1
1.4	Properties of DFT and examples ,Circular convolution	2
1.5	Linear Filtering methods based on the DFT- linear convolution using circular convolution, Filtering of long data sequences, overlap save and overlap add methods,	3
1.6	Frequency Analysis of Signals using the DFT(concept only required)	1
2	**Module 2**	
2.1	Efficient Computation of DFT: Fast Fourier Transform Algorithms	1
2.2	Radix-2 Decimation in Time and Decimation in Frequency FFT Algorithms	4
2.3	IDFT computation using Radix-2 FFT Algorithms	2
2.4	Application of FFT Algorithms-Efficient computation of DFT of Two Real Sequences and a 2N-Point Real Sequence	1
3	**Module 3**	

3.1	Design of FIR Filters- Symmetric and Anti-symmetric FIR Filters, Design of linear phase FIR filters using Window methods, (rectangular, Hamming and Hanning)	4
3.2	Design of linear phase FIR filters using frequency sampling Method, Comparison of Design Methods for Linear Phase FIR Filters	2
3.3	Design of IIR Digital Filters from Analog Filters, (Butterworth), IIR Filter Design by Impulse Invariance	3
3.4	IIR Filter Design by Bilinear Transformation	2
3.5	Frequency Transformations in the Analog and Digital Domain.	1
4	**Module 4**	
4.1	Structures for the realization of Discrete Time Systems- Block diagram and signal flow graph representations of filters	2
4.2	FIR Filter Structures: (Linear structures), Direct Form Cascade Form	,2
4.3	IIR Filter Structures: Direct Form, Cascade Form and Parallel Form	3
4.3	Computational Complexity of Digital filter structures.	1
4.4	Multi-rate Digital Signal Processing: Decimation and Interpolation (Time domain and Frequency Domain Interpretation), Anti-aliasing and anti-imaging filter.	3
5	**Module 5**	
5.1	Computer architecture for signal processing : Harvard Architecture, pipelining, MAC, Introduction to TMS320C67xx digital signal processor ,Functional Block Diagram	3
5.2	Finite word length effects in DSP systems: Introduction (analysis not required), fixed-point and floating-point DSP arithmetic, ADC quantization noise,	3
5.3	Finite word length effects in IIR digital filters: coefficient quantization errors.	2
5.4	Finite word length effects in FFT algorithms: Round off errors	1

Simulation Assignments

The following simulations to be done in Scilab/ Matlab/ LabView/GNU Octave:

1. Consider a signal given by x(n)=[1,1,1,1].
    1. Compute the DTFT of the given sequence and plot its magnitude and phase
    2. Compute the 4 point DFT of the above signal and plot its magnitude and phase
    3. Compare the above plots and obtain the relationship?
2. Zero pad the sequence x(n) by 4 and compute the 8 point DFT and find the corresponding magnitude and phase plots. Compare the spectra with that in (b) and comment on it.
3. The first five values of the 8 point DFT of a real valued sequence x(n) are given by {0.25, 0.125-j0.3, 0, 0.125-j0.06, 0.5}. Determine the DFT of each of the following sequences using properties. Hint :IDFT may not be computed.
    1. $x1(n)=x((2-n))8$
    2. $x3(n)=x^2(n)$
    3. $x4(n)=x(n)e^{j\pi n/N)}$ in/4
4. a) Develop a function to implement the over-lap add method using circular convolution operation. The format should be function [y]=overlappadd(x,h,N), where y is the output sequence, x is the input sequence and N is the block - length>=2*Length(h)-1.
    1. Incorporate the radix-2 FFT implementation in the above function to obtain a high speed overlap add block convolution routine. Choose N=8. Hint :choose $N=2^k$
5. Design a low pass digital filter to be used in the given structure



to satisfy the following requirements. Sampling rate of 8000samples/second, Pass band edge of 1500Hz with a ripple of 3dB, Stopband edge of 2000Hz with attenuation of 40 dB, Equiripple passband but monotonic stopband. (Use impulse invariance technique)

1. Choose T=1 s for impulse invariance and determine the system function H(z) in parallel form.Plot the log-magnitude response in dB and impulse response h(n)
2. Choose T=1/8000 s and repeat the same procedure. Compare this design with that in (a) and comment on the effect of T on the impulse invariant design?

6. A filter is described by the following difference equation:

   16y(n)+12y(n-1)+2y(n-2)-4y(n-3)-y(n-4)=x(n)-3x(n-1)+11x(n-2)-27x(n-3)+18x(n-4)

   1. Determine the Direct form filter structure
   2. Using the Direct form structure, obtain the cascade form filter structure

7. Consider a signal given by $x(n)=(0.5)^n u(n)$. Decimate the signal by a factor 4 and plot the output in time domain and frequency domain?

   1. Interpolate the signal by a factor of 4 and plot the output in time domain and frequency domain?
   2. Compare the spectra and obtain the inference?

Model Question Paper

A P J Abdul Kalam Technological University

Fifth Semester B Tech Degree Examination
Branch: Electronics and Communication Engg.

Course: ECT 303 DIGITAL SIGNAL PROCESSING

Time: 3 Hrs Max. Marks: 100

PART A
Answer All Questions. Each question carry 3 marks

1 .Derive the relationship of DFT to Z-transform? (3)K3
2.Find the circular convolution of two sequences $x_1(n)=\{1, 2,-2,1,3\}$,$x_2(n)=\{2,-1,3,1,1\}$ (3)K3
3 Illustrate the basic butterfly computation used in decimation in time radix-2 FFT algorithm?(3)K1
4 Bring out the computational advantage of performing an N-point DFT using radix-2 FFT compared to direct method?
5. Determine the frequency response of a linear phase FIR filter given by the difference equation y(n)=0.15x(n)+0.25x(n-1)+x(n-3). Also find the phase delay (3) K3
6 .An all pole analog filter is given by the transfer function $H(s)=1/(s^2+5s+6)$. Find out the transfer function H(z) of the equivalent digital filter using impulse invariance method. Use T=1s (3) K3

7.Obtain the cascade form realization of the third order IIR filter transfer function given by

$$H(z)=\frac{0.44z^2+0.362z+0.02}{(z^2+0.8z^{\square}+.0.5)(z-0.4)}$$ (3) K3

8 . Prove that a factor of L upsampler is a linear-time varying system. (3) K3

9. Differentiate between Harvard architecture and Von-Nuemann Architecture used in processors? (3) K1
10. Express the fraction 7/8 and -7/8 in sign-magnitude, two's compliment and one's compliment format? (3) K3

Part B

Answer any one Question from each module. Each question carries 14 Marks

11. a) How will you perform linear convolution using circular convolution? Find the linear convolution of the given sequences x(n) = {2, 9,7, 4} and h(n) = {1, 3, 1, 2} using circular convolution? (8) K3

b) Explain the following properties of DFT a) Linearity b) Complex conjugate property c) Circular Convolution d) Time Reversal (6) K2

OR

12.a.) The first eight points of 14-point DFT of a real valued sequence are {12, -1+j3, 3+j4, 1-j5, -2+j2, 6+j3, -2-j3, 10}

i) Determine the remaining points

ii) Evaluate x[0] without computing the IDFT of X(k)?

iii) Evaluate IDFT to obtain the real sequence ? (8)K3

b) Explain with appropriate diagrams, the overlap-add method for filtering of long data sequences using DFT? (6) K2

13.a) Compute the 8 point DFT of x(n) = {2,1,-1,3,5,2,4,1} using radix-2 decimation in time FFT algorithm. (9) K3

b)Bring out how a 2N point DFT of a 2N point sequence can be found using the computation of a single N point DFT. (5) K3

OR

14 a.) Find the 8 point DFT of a real sequence x(n)={1,2,2,2,1,0,0,0,0} using radix-2 decimation in frequency algorithm (9)K3

b) Bring out how N-point DFT of two real valued sequences can be found by computing a single N-point DFT. (5) K3

15.a. Design a linear phase FIR low pass filter having length M = 15 and cut-off frequency ω_c = π/6. Use Hamming window. (10) K3

b.Prove that if z_1 is a zero of an FIR filter, then $1/z_1$ is also a zero? (4) K2

OR

16. a. Design a digital Butterworth low pass filter with $\omega_p = \pi/6$, $\omega_s = \pi/4$, minimum pass band gain = -2 dB and minimum stop band attenuation = 8 dB. Use bilinear transformation.(Take T = 1s) (10) K3

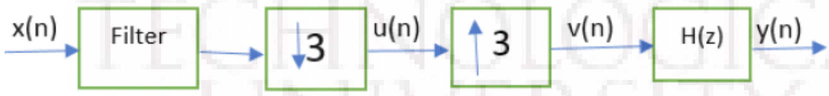
b. What is warping effect in bilinear transformation and how it can be eliminated? (4) K2

17.a) Derive and draw the direct form-I, direct form-II and cascade form realization of the given filter, whose difference equation is given as
$y(n)=0.1\,y(n-1)+0.2\,y(n-2)+3\,x(n)+3.6\,x(n-1)+0.6\,x(n-2)$ (9) K3

b) Differentiate between anti-aliasing and anti-imaging filters. (5) K2

OR

18.a) Obtain the expression of output y(n) as a function of x(n) for the multi-rate structure given below? (9) K3



b) Draw the transposed direct form II Structure of the system given by the difference equation y(n)=05.y(n-1)-0.25y(n-2)+x(n)+x(n-1) . (5)K2

19.a.With the help of a functional block diagram, explain the architecture of TMS320C67xx DSP processor? (10) K2

b.What are the prominent features of TMS320C67xx compared to its predecessors ? (4) K2

OR

20.a)Explain how to minimize the effect of finite word length in IIR digital filters? (7) K2

b)Explain the roundoff error models used in FFT algorithms? (7) K2