CODE	COURSE NAME	CATEGORY	L	T	P	CREDIT
EET453	**DIGITAL SIGNAL PROCESSING**	**PEC**	**2**	**1**	**0**	**3**

Preamble: This course introduces the discrete Fourier transform (DFT) and its computation using direct method and fast Fourier transform (FFT). Techniques for designing infinite impulse response (IIR) and finite impulse response (FIR) filters from given specifications are also introduced. Various structures for realization of IIR and FIR filters are discussed. Detailed analysis of finite word-length effects in fixed point DSP systems is included. Architecture of a digital signal processor is also discussed.

Prerequisite : EET305 - Signals and Systems

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

CO 1	Compute Discrete Fourier transform and Fast Fourier transform .
CO 2	Discuss the various structures for realization of IIR and FIR discrete-time systems.
CO 3	Design IIR (Butterworth and Chebyshev) digital filters using impulse invariant and bilinear transformation methods.
CO 4	Design FIR filters using frequency sampling method and window function method.
CO 5	Compare fixed point and floating point arithmetic used in digital signal processors and discuss the finite word length effects.
CO 6	Explain the architecture of digital signal processors and the applications of DSP.

Mapping of course outcomes with program outcomes

	PO1	PO2	PO3	PO4	PO5	PO6	PO7	PO8	PO9	PO10	PO11	PO12
CO 1	3	2	-	2	2	-	-	-	-	-	-	2
CO 2	3	2	-	2	2	-	-	-	-	-	-	2
CO 3	3	2	-	2	2	-	-	-	-	-	-	2
CO 4	3	2	-	2	2	-	-	-	-	-	-	2
CO 5	3	2	-	-	2	-	-	-	-	-	-	2
CO 6	3	-	2	-	2	2	-	-	-	-	-	3

Assessment Pattern

Bloom's Category	**Continuous Assessment Tests**		**End Semester Examination**
	1	**2**	
Remember (K1)	10	10	10
Understand (K2)	10	10	30
Apply (K3)	30	30	60
Analyse (K4)			
Evaluate (K5)			
Create (K6)			

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

Course Outcome 1 (CO1)

1. State and prove various properties of DFT - (K1, PO1,PO2,PO12)
2. Determine the linear convolution using DFT – (K2,PO1,PO2,PO4,PO5,PO12)
3. Determine the linear convolution using overlap-add and overlap-save method – (K3,PO1,PO2,PO4,PO5)
4. Compute DFT using DIT FFT and DIF FFT – (K2,PO1,PO2,PO4,PO5)

Course Outcome 2 (CO2)

1. Determine the structures for direct form, cascade, parallel, transposed and lattice-ladder realisations of IIR systems –(K2,PO1,PO2,PO4,PO5,PO12)
2. Determine the structures for direct form, cascade, lattice ,and linear phase realizations of FIR systems – (K2,PO1,PO2,PO4,PO5)

Course Outcome 3(CO3)

1. Design IIR digital LP/HP/BP/BS filter using Butterworth and Chebyshev methods – (K3,PO1,PO2,PO4,PO5)
2. Transform H(s) to H(z) using impulse invariant technique and bilinear transformation – (K2,PO1,PO2,PO4,PO5,PO12)

Course Outcome 4 (CO4)

1. Design FIR digital LP/HP/BP/BS filter using frequency sampling method – (K3,PO1,PO2,PO4,PO5,PO12)
2. Design FIR digital LP/HP/BP/BS filter using window function – (K3,PO1,PO2,PO4,PO5)

Course Outcome 5 (CO5)

1. Differentiate between fixed-point arithmetic and floating point arithmetic - (K2,PO1,PO2,PO12)
2. Explain various finite word length effects in fixed point DSP processors.- (K2,PO1,PO2)
3. Problems to determine steady state output noise power and round-off noise power – (K3,PO1,PO2)
4. Explain limit cycle oscillations and methods for its elimination - (K2,PO1,PO2)

Course Outcome 6 (CO6)

1. Explain Harvard architecture –(K1,PO1,PO5,PO12)
2. Describe the architecture of a fixed-point DSP processor – (K1,PO1,PO5)
3. List various applications of digital signal processor – (K3,PO1,PO3,PO6)

Model Question Paper PAGES: 3

QPCODE:

Reg. No:____________

Name:______________

APJ ABDUL KALAM TECHNOLOGICAL UNIVERSITY
SEVENTH SEMESTER B. TECH DEGREE EXAMINATION
MONTH & YEAR

Course Code**: EET453**
Course Name: **DIGITAL SIGNAL PROCESSING**

Max. Marks: 100 **Duration: 3 Hours**

PART A
Answer all Questions.
Each question carries 3 Marks

1	List any 3 properties of DFT.
2	The first 5 points of the 8-point DFT of a real valued sequence are $X(k)=\{0.25, 0.125-j0.3, 0, 0.125-j0.05, 0\}$. Determine the remaining 3 points.
3	Obtain direct form 1 realization for a digital IIR system described by the system function, $H(z)=\dfrac{z+0.2}{z^2+0.5z+1}$.
4	Obtain realization with minimum number of multipliers for the system function $H(z)=\dfrac{1}{2}+z^{-1}+\dfrac{1}{2}z^{-2}$.

5 Explain warping effect in bilinear transformation.

6 Determine the order of a Chebyshev analog lowpass filter with a maximum passband attenuation of 2.5dB at Ω_p = 20 rad/sec and the stopband attenuation of 30dB at Ω_s = 50 rad/sec.

7 What are the desirable characteristics of a window function used for truncating the infinite impulse response?

8 Represent the numbers i) +4.5 and ii) -4.5 in IEEE 754 single-precision floating point format.

9 List any 3 finite-word length effects in a fixed point digital signal processor.

10 Draw the block diagram of a basic Harvard architecture in digital signal processor.

PART B

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

Module 1

11 a) Find the 4-point DFT of the sequence, $x(n)=\{1,-1,1,-1\}$. Also, using time shift property, find the DFT of the sequence, $y(n)=x((n-2))_4$. (7)

b) Two finite duration sequences are $h(n)=\{1,0,1\}$ and $x(n)=\{-1,2,-1,0,1,3,-2,1,-3,-2,-1,0,-2\}$. Use overlap-save method, to find $y(n)=x(n)*h(n)$. (7)

OR

12 Compute IDFT of the sequence $X(k)=\{7,-0.707-j0.707,-j,0.707-j0.707,1,0.707+j0.707,j,-0.707+j.707\}$ using DIT FFT. (14)

Module 2

13 a) Realize the system function in cascade form $H(z)=\dfrac{1+\frac{1}{3}z^{-1}}{1-\frac{3}{4}z^{-1}+\frac{1}{8}z^{-2}}$. (6)

b) Determine the direct form 2 and transposed direct form structure for the given system $y(n)=\frac{1}{2}y(n-1)-\frac{1}{4}y(n-2)+x(n)+x(n-1)$. (8)

OR

14 a) Obtain the direct form realization of linear phase FIR system given by (7)

$$H(z)=1+\frac{3}{4}z^{-1}+\frac{17}{8}z^{-2}+\frac{3}{4}z^{-3}+z^{-4}$$

b) Determine the coefficients k_m of the lattice filter corresponding to FIR filter described by the system function $H(z)=1+2z^{-1}+\frac{1}{3}z^{-2}$. Also, draw the corresponding second order lattice structure (7)

Module 3

15 a) Find H(z) using impulse invariant transformation. (7)
$H(s)=\frac{1}{s^2+\sqrt{2}s+1}$; $T=1\sec$.

b) A Butterworth lowpass filter has to meet the following specifications. (7)
i) Passband gain = -3dB at f_p = 500Hz
ii) Stopband attenuation greater than or equal to 40dB at f_s = 1000Hz
Determine the order of the Butterworth filter to meet the above specifications. Also, find the cut off frequency.

OR

16 Design a Chebyshev digital lowpass filter with a maximum passband attenuation of 2dB at 100Hz and minimum stopband attenuation of 20dB at 500Hz. Sampling rate is 4000 samples/sec. Use bilinear transformation. (14)

Module 4

17 a) Design a linear phase lowpass FIR filter with N = 7 and a cut-off frequency 0.3π radian using the frequency sampling method. (7)

b) A linear phase FIR filter has frequency response $H(\omega)=\cos\frac{\omega}{2}+\frac{1}{2}\cos\frac{3\omega}{2}$ (7)
Determine the impulse response h(n).

OR

18 A band stop filter is to be designed with the following desired frequency response $H_d(e^{j\omega})=\begin{cases} e^{-j\omega\alpha} & -\omega_{c1}\le\omega\le\omega_{c1} \quad \omega_{c2}\le|\omega|\le\pi \\ 0 & \text{otherwise}\end{cases}$ (14)
Design with N = 7, ω_{c1} = π/4 rad/sec, ω_{c2} = 3π/4 rad/sec using rectangular window.

Module 5

19 a) Compare between fixed point and floating point digital signal processors. (6)

b) The output of an ADC is applied to a digital filter with system function $H(z)=\frac{0.5z}{(z-0.5)}$. Find the output noise power from digital filter when input signal is quantized to have 8 bits. (8)

OR

20 a) Draw and explain the architecture of any fixed-point DSP processor. (8)

b) Explain the techniques used to prevent overflow in fixed-point DSP operations. (6)

Syllabus

Module 1 - DISCRETE-FOURIER TRANSFORM

Review of signals and systems - Frequency domain sampling - Discrete Fourier transform (DFT) – inverse DFT (IDFT) - properties of DFT – linearity, periodicity, symmetry, time reversal, circular time shift, circular frequency shift, circular convolution, complex conjugate property – Filtering of long data sequences – over-lap save method, over-lap add method – Fast Fourier transform (FFT) – advantages over direct computation of DFT - radix -2 decimation-in-time FFT (DITFFT) algorithm, Radix-2 decimation-in-frequency FFT (DIFFFT) algorithm.

Module 2 - REALIZATION OF IIR AND FIR SYSTEMS

Introduction to FIR and IIR systems - Realization of IIR systems – direct form 1, direct form 2, cascade form, parallel form, lattice structure for all-pole system, lattice-ladder structure – conversion of lattice to direct form and vice-versa - signal flow graphs and transposed structures – Realization of FIR systems – direct form, cascade form, lattice structure, linear phase realization.

Module 3 - IIR FILTER DESIGN

Conversion of analog transfer function to digital transfer function – impulse invarient transformation and bilinear transformation – warping effect

Design of IIR filters – low-pass, high-pass, band-pass, band-stop filters – Butterworth and Chebyshev filter – frequency transformation in analog domain - design of LP, HP, BP, BS IIR digital filters using impulse invariance and bilinear transformation.

Module 4 - FIR FILTER DESIGN AND REPRESENTATION OF NUMBERS

Impulse response of ideal low pass filter – linear phase FIR filter – frequency response of linear phase FIR filter – Design of FIR filter using window functions (LP, HP, BP, BS filters) – Rectangular, Bartlett, Hanning, Hamming and Blackmann only – FIR filter design based on frequency sampling approach (LP, HP, BP, BS filters)

Representation of numbers – fixed point representation – sign-magnitude, one’s complement, two’s complement – floating point representation – IEEE 754 32-bit single precision floating point representation

Module 5 - FINITE WORD LENGTH EFFECTS AND DIGITAL SIGNAL PROCESSORS

Finite word length effects in digital Filters – input quantization – quantisation noise power – steady-state output noise power – coefficient quantisation – overflow – techniques to prevent overflow - product quantization error – rounding and truncation – round-off noise power – limit cycle oscillations – zero input limit cycle oscillations – overflow limit cycle oscillations – signal scaling.

Digital signal processor architecture based on Harvard architecture (block diagram) – Harvard architecture, pipelining, dedicated hardware multiplier/accumulator, special instructions dedicated to DSP, replication, on-chip memory cache, extended parallelism

(Reference [2]) - comparison of fixed-point and floating-point processor – applications of DSP

Text Books

1. John G. Proakis & Dimitris G.Manolakis, "Digital Signal Processing Principles, Algorithms & Applications", Pearson

Reference Books

1. Emmanuel Ifeachor & Barrie W Jervis, "Digital Signal Processing", Pearson, 13th edition, 2013
2. P. Ramesh Babu, "Digital Signal Processing", Scitech Publications (India) Pvt Ltd, 2nd edition, 2003
3. Li Tan, "Digital Signal Processing, Fundamentals & Applications", Academic Press, Ist edition, 2008
4. D. Ganesh Rao & Vineeta P Gejji, "Digital Signal Processing, A Simplified Approach", Sanguine Technical Publishers, 2nd edition, 2008

Course Contents and Lecture Schedule

Sl. No	Topic	No. of Lectures
1	**DISCRETE-FOURIER TRANSFORM (7 hours)**	
1.1	Review of signals, systems and discrete-time Fourier transform (DTFT), Frequency domain sampling, discrete-Fourier transform (DFT), twiddle factor, inverse DFT, properties of DFT - linearity, periodicity, symmetry, time reversal, circular time shift, circular frequency shift, circular convolution, complex conjugate property	3 hours
1.2	Linear filtering using DFT, linear filtering of long data sequences, overlap-save method, overlap-add method	1 hour
1.3	Fast Fourier transform (FFT) – comparison with direct computation of DFT - radix -2 decimation-in-time FFT (DITFFT) algorithm – bit reversal - Radix-2 decimation-in-frequency FFT (DIFFFT) algorithm	3 hours
2	**REALIZATION OF IIR AND FIR SYSTEMS (7 hours)**	
2.1	Introduction to FIR and IIR systems - comparison - Realization of IIR systems – direct form 1, direct form 2, cascade form, parallel form	3 hours
2.2	Lattice structure for all-pole system - lattice-ladder structure – conversion of lattice to direct form and vice-versa signal flow graphs and transposed structures	2 hours
2.3	Realization of FIR systems – direct form, cascade form, lattice structure, linear phase realization.	2 hours
3	**IIR FILTER DESIGN (7 hours)**	
3.1	Conversion of analog transfer function to digital transfer function – impulse invarient transformation and bilinear transformation – warping effect	2 hours
3.2	Design of IIR filters – characteristics of ideal and practical low-pass, high-pass, band-pass, band-stop filters – design of Butterworth filter –	3 hours

	normalised analog filter - frequency transformation in analog domain - design of LP, HP, BP, BS IIR digital filters using impulse invariance and bilinear transformation.	
3.3	Design of Chebyshev filter – design of LP, HP, BP, BS IIR digital filters using impulse invariance and bilinear transformation	2 hours
4	**FIR FILTER DESIGN AND REPRESENTATION OF NUMBERS (7 hours)**	
4.1	Impulse response of ideal low pass filter – linear phase FIR filter – frequency response of linear phase FIR filter – Design of FIR filter using window function (LP, HP, BP, BS filters) – Rectangular, Bartlett, Hanning, Hamming and Blackmann only	3 hours
4.2	FIR filter design based on frequency sampling approach (LP, HP, BP, BS filters)	2 hours
4.3	Representation of numbers – fixed point representation – sign-magnitude, one's complement, two's complement – floating point representation – IEEE 754 32-bit single precision floating point representation	2 hours
5	**FINITE WORD LENGTH EFFECTS AND DIGITAL SIGNAL PROCESSORS (7 hours)**	
5.1	Finite word length effects in digital Filters – input quantization – quantisation noise power – steady-state output noise power	2 hours
5.2	Coefficient quantisation – overflow – techniques to prevent overflow - product quantization error – rounding and truncation – round-off noise power	1 hour
5.3	Limit cycle oscillations – zero input limit cycle oscillations – overflow limit cycle oscillations – signal scaling.	1 hour
5.4	Digital signal processor architecture based on Harvard architecture (block diagram) – Harvard architecture, pipelining, dedicated hardware multiplier/accumulator, special instructions dedicated to DSP, replication, on-chip memory cache, extended parallelism (Reference [1])	2 hours
5.5	Comparison of fixed-point and floating-point processor – applications of digital signal processor	1 hour

Note: Preferable list of computer based assignments

Assignments using signal processing tool of MATLAB/SCILAB etc	
1	Determine 4-point/8-point DFT/IDFT of any sequence by direct computation
2	Compute 4-point/8-point DFT/IDFT using DIT FFT and DIF FFT algorithms.
3	Find the linear convolution and circular convolution of two sequences.
4	Find the linear convolution using overlap-add and overlap-save methods.
5	Determine 2 stage/3 stage lattice ladder coefficients if the system function of IIR direct form is given.
6	Obtain coefficients of IIR direct form from lattice ladder form.
7	Transform an analog filter into digital filter using impulse invariant technique/bilinear transformation.
8	Calculate the order and cut-off frequency of a low pass Butterworth filter
9	Obtain the frequency response and filter coefficients of a LP/HP/BP/BS IIR

	Butterworth filter
10	Obtain the frequency response and filter coefficients of a LP/HP/BP/BS IIR Chebyshev filter
11	Compute LP/HP/BP/BS FIR filter coefficients using rectangular/Bartlett/Hamming/Hanning/Blackmann window