University of Science and Technology of Hanoi
Address: Building 2H, 18 Hoang Quoc Viet, Cau Giay, Hanoi
Telephone/ Fax: +84-4 37 91 69 60
Email: officeusth@usth.edu.vn
Website: http://www.usth.edu.vn

COURSE SYLLABUS

Subject: Signal Processing

Academic field: Space Science and Application

Lecturer: Nguyễn Công Phương

Phone: 0912905931

E-mail: phuong.nguyencong@hust.edu.vn

Academic year:

COURSE DESCRIPTION

Credit points	4	
Level	Undergraduate	
Teaching time Location	University of Science and Technology of Hanoi	
Time Commitment	Lecture	25 hrs
	Exercises	10 hrs
	Practicals	5 hrs
	Total	40 hrs
Prerequisites	Mathematics for Engineering	
Recommended background knowledge	A background in calculus, complex number and variable, and the basics of linear algebra.	
Subject description:	This subject provides knowledges of ways to record, create, manipulate, and transform signals (such as speech, audio, image, video, electrocardiograms, etc.) These operations are fundamental to construct modern information and communication systems.	
Objectives & Out-come	*Objectives*: 1. To provide students with knowledges on concepts of analog and digital signals, analog and digital system, and signal processing in the continuous – time and discrete – time domains. 2. To provide students with concepts of mathematical operations on signals, such as convolution, z – transform, fast Fourier transform, etc. 3. To provide students with techniques to analyse signal processing systems and techniques to design digital filters. *Outcome*: After completing this course, students will be able to: 1. Understand the concept of signal and distinguish between continuous – time, discrete – time, and digital signals, and describe them by their physical and mathematical representation. 2. Apply mathematical operations on signals, e.g. convolution, z –	

	transform, Fourier transform, etc. 3. Design digital filters (such as FIR, IIR) to manipulate signals using signal processing algorithms and techniques, and implement them using Matlab.	
Assessment/ Evaluation	Attendance/Attitude	5%
	Exercise(s)	10%
	Practicals	15%
	Mid-term test	20%
	Final exam	50%
Prescribed Textbook(s)	[1] D. Manolakis and V. Ingle. *Applied Digital Signal Processing*. Cambride, 2011. [2] J. G. Proakis and D. G. Manolakis. *Digital Signal Processing – Principles, Algorithms, and Applications*. Prentice – Hall, 1996.	

COURSE CONTENTS & SCHEDULE

Class	Contents	Hours			Ref./Resources	Assignment(s)
		Lect.	Exr.	Prc.		
1	**Introduction** 1.1 Signals 1.2 Systems 1.3 Analog, Digital, and Mixed Signals Processing 1.4 Application of Digital Signal Processing	1	0	0		
2	**Discrete – Time Signals and Systems** 2.1 Discrete – Time Signals 2.2 Discrete – Time Systems 2.3 Convolution Description of Linear Time – Invariant Systems 2.4 Properties of Linear Time – Invariant Systems 2.5 Analytical Evaluation of Convolution 2.6 Numerical Computation of Convolution 2.7 Real – Time Implementation of FIR Filters 2.8 FIR Spatial Filters 2.9 Systems Described by Linear Constant – Coefficient Difference Equations 2.10 Continuous – Time LTI Systems	2	1	0		
3	**The *z* – Transform**	2	1	0		

University of Science and Technology of Hanoi
Address: Building 2H, 18 Hoang Quoc Viet, Cau Giay, Hanoi
Telephone/ Fax: +84-4 37 91 69 60
Email: officeusth@usth.edu.vn
Website: http://www.usth.edu.vn

	3.1 The *z* – Transform 3.2 The Inverse *z* – Transform 3.3 Properties of the *z* – Transform 3.4 System Function of LTI Systems 3.5 LTI Systems Characterized by Linear Constant – Coefficient Difference Equations 3.6 Connections between Pole – Zero Locations and Time – Domain Behavior					
4	**Fourier Representation of Signals** 4.1 Sinusoidal Signals and Their Properties 4.2 Fourier Representation of Continuous – Time Signals 4.3 Fourier Representation of Discrete – Time Signals 4.4 Summary of Fourier Series and Fourier Transforms 4.5 Propertiers of the Discrete – Time Fourier Transform	2	1	0		
5	**Transform Analysis of LTI Systems** 5.1 Sinusoidal Response of LTI Systems 5.2 Response of LTI Systems in the Frequency Domain 5.3 Distortion of Signals Passing through LTI Systems 5.4 Ideal and Practical Filters 5.5 Frequency Response for Rational System Functions 5.6 Dependency of Frequency Response on Poles and Zeros 5.7 Design of Simple Filters by Pole – Zero Placement 5.8 Relationship between Magnitude and Phase Responses 5.9 Allpass Systems 5.10 Invertibilty and Minimum – Phase Systems 5.11 Transform Analysis of Continuous – Time Systems	2.5	1	1		
6	**Sampling of Continuous – Time Signals** 6.1 Ideal Periodic Sampling of Continuous – Time Signals 6.2 Reconstruction of a Bandlimited Signal from its Samples 6.3 The Effect of Undersampling: Aliasing 6.4 Discrete – Time Processing of Continuous – Time Signals	2	1	0		

University of Science and Technology of Hanoi
Address: Building 2H, 18 Hoang Quoc Viet, Cau Giay, Hanoi
Telephone/ Fax: +84-4 37 91 69 60
Email: officeusth@usth.edu.vn
Website: http://www.usth.edu.vn

	6.5 Practical Sampling and Reconstruction 6.6 Sampling of Bandpass Signals					
7	**The Discrete Fourier Transform** 7.1 Computational Fourier Analysis 7.2 The Discrete Fourier Transform (DFT) 7.3 Sampling the Discrete – Time Fourier Transform 7.4 Properties of the Discrete Fourier Transform 7.5 Linear Convolution using the DFT 7.6 Fourier Analysis of Signals using the DFT 7.7 Direct Computation of the Discrete Fourier Transform 7.8 The FFT Idea using a Matrix Approach 7.9 Decimation – in – Time FFT Algorithms 7.10 Decimation – in – Frequency FFT Algorithms 7.11 Generalization and Additional FFT Algorithms 7.12 Practical Considerations	3.5	1	1		
8	**Structures for Discrete – Time Systems** 8.1 Block Diagrams and Signal Flow Graphs 8.2 IIR System Structures 8.3 FIR System Structures 8.4 Lattice Structures 8.5 Structure Conversion, Simulation, and Verification	2	1	0		
9	**Design of FIR Filters** 9.1 The Filter Design Problem 9.2 FIR Filters with Linear Phase 9.3 Design of FIR Filters by Windowing 9.4 Design of FIR Filters by frequency sampling 9.5 Chebyshev Polynomials and Minimax Approximation 9.6 Equiripple Optimum Chebyshev FIR Filter Design	2.5	1	1		
10	**Design of IIR Filters** 10.1 Introduction to IIR Filter Design 10.2 Design of Continuous – Time Lowpass Filters 10.3 Transformation of Continuous – Time Filters to Discrete – Time IIR Filters 10.4 Design Examples for Lowpass IIR Filters 10.5 Frequency Transformation of Lowpass Filters	2.5	1	1		

11	**Random Signal Processing** 11.1 Probability Models and Random Variables 11.2 Jointly Distributed Random Variables 11.3 Covariance, Correlation, and Linear Estimation 11.4 Random Process 11.5 Some Useful Random Process Models 11.6 Estimation of Mean, Variance, and Covariance 11.7 Spectral Analysis of Stationary Processes 11.8 Optimum Linear Filters 11.9 Linear Prediction and All – Pole Signal Modeling 11.10 Optimum Orthogonal Transforms	3	1	1		

Notes:

- *Abbreviation: Lect. (lecture), Exr. (Exercise), Prc. (Practise).*
- *Exercises may include assignment, reports, student's presentation, homework, class exercises ...for each class sessions*
- *Practicals mostly refer to Lab- work or outside practice such as field trip.*

Reference Literature:

[1]. R. N. Bracewell. *The Fourier Transform and its Applications*. McGraw-Hill, 2000
[2]. V. K. Madisetti, D. B. Williams. *Digital Signal Processing Handbook*. CRC Press, 1999
[3]. A. V. Oppenheim, A. S. Willsky, and S. H. Nawab. *Signals and Systems*. Prentice Hall, 1997
[4]. S. Stergiopoulos. *Advanced Signal Processing Handbook*. CRC Press, 2001