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EUROPEAN UNIVERSITY FOR CUSTOMISED EDUCATION

STUDY GUIDE

DIGITAL SIGNAL PROCESSING

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FROM BASIC TO ADVANCED

Organised by

University of the Peloponnese

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of the European Union

1. IDENTIFYING DATA.	
· Course Name.	*Digital Signal Processing – From basic to advanced*
· Coordinating University.	*University of the Peloponnese*
· Partner Universities Involved.	-
· Course Field(s).	*Digital technologies*
· Related Study Programme.	-
· ISCED Code.	– *0710 Engineering and engineering trades*
· SDG.	– *Industry, innovation and infrastructure (SDG 9)*
· Study Level.	*BA and MA*

· Number of ECTS credits allocated.	*3*
· Mode of Delivery.	*Online self-study*
· Language of Instruction.	*English*
· Course Dates.	*30 March 2025 – 15 June 2025*
· Schedule of the course.	*Self-study program: 4 hours per week* *Synchronous 2 hours online sessions for student support:* *Sunday: 30 March, 13 April, 27 April, 11 May, 25 May*
· Key Words.	*Digital Signal Processing, Systems and Signals, Filters, Transforms Fourier and Z, Digital filters*
· Catchy Phrase.	*Digital signal processing for beginners*

· Prerequisites and co-requisites.	– *Students must have attended an undergraduate course related to Signals and Systems.* – *Students must have the ability to communicate in English – Level B2 is required.* – *The study levels this course is available for are BA and MA students.*
· Number of EUNICE students that can attend the Course.	*Total number of students: 40* *Number of students per EUNICE partner: 4*
· Course inscription procedure(s).	*The standard EUNICE registration process will be applied for the course.*

2. CONTACT DETAILS.	
· **Department.**	*Electrical and Computer Engineering*
· **Name of Lecturer.**	*Prof. Michael Paraskevas*
· **E-mail.**	*mparask@uop.gr*
· **Other Lecturers.**	*Michail Nanos, Mathematician & Informatics Engineer, MSc* *Nikos Spatiotis, Informatics Engineer, MSc, Phd candidate*

3. COURSE CONTENT.
Discrete time signals, Fundamental discrete-time signals, characteristic parameters and operations between signals, Discrete-time systems and system function, Stable, causal, time-invariant discrete systems, Impulse response of a discrete system, Convolution in discrete time, Differential equations and their solution, The DTFT transform and its properties, Solving difference equations using DTFT, Inverse systems, Ideal frequency selection filters, Z-transform, transform properties and regions of convergence (ROC), Fractional forms of Z-transform, System transfer function, The discrete DFT transform, its properties and the FFT implementation, Circular convolution and ways of calculating it, DFT implementation, Design of IIR and FIR digital filters, IIR and FIR filter design techniques.

4. LEARNING OUTCOMES.

At the end of the course students will be able:

- *Describe the operation of analog to digital conversion systems and design such systems.*
- *Compute the impulse response of an LTI system when the linear difference equation is known.*
- *Choose the most appropriate way to calculate the outputs of an LTI system depending on the data available to them.*
- *Explain the significance and differences between DTFT and DFT transforms.*
- *Connect the properties of Z transform with system functions, e.g. time delay and group delay.*
- *Conclude about the stability and the transient behavior of systems, using the Z transform.*

Design and evaluate the FIR and IIR filter response.

5. OBJECTIVES.

The purpose of the course is to introduce students to the basic concepts and techniques of digital signal processing. For this purpose, the concepts of discrete-time signals and systems will be presented. Computation of the response of a Linear and Time-Invariant system to Displacement via

convolution and difference equation will be presented. The definitions and properties of the DTFT, DFT and Z transforms as well as their applications will be given. Concepts of transfer function, frequency response and finding system response using DTFT and Z transforms will be introduced. System stability will be studied by producing pole-zero diagrams. Finally, the basic concepts of FIR and IIR filter design will be presented.

6. COURSE ORGANISATION.	
UNITS	
1.	***Signal Conversion from Analog to Digital - Discrete Time Signals*** *Ideal sampling, Uniform quantization, Quantization parameters, Coding, Reconstruction of analog signal from digital, Classification of discrete-time signals (Periodic and Non-Periodic, Even and Odd, Energy and Power, Causal and Anticausal), Operations on discrete-time signals (Addition, Multiplication, Amplitude Scaling), Transformations of the independent variable (Time-shift, Time-inversion, Time-scaling), Fundamental discrete-time time signals (Unit step, Unit impulse, Unit slope), Analysis of discrete-time signals into unit impulses, Real exponential discrete-time sequence, Complex exponential discrete-time sequence, Sine sequence).*
2.	***Discrete-time Systems, Convolutional Sum and Difference Equations*** *Introduction to discrete-time systems, Categorization of discrete-time systems (Causal, Static, Time-Invariant, Homogeneous, Linear, Stable, Invertible), Ways of describing discrete-time systems (Stage diagrams, Difference Equations, Convolutional sum), Recursive and Non-recursive systems, Study of systems with the convolution, Properties of convolution, Ways to calculate linear convolution (Analytic calculation, Graphical calculation, Table Method, Calculation with Toeplitz Table), Difference Equations, Solving Linear Difference Equations with Constant Coefficients (LDECC), Classification of systems according to the type of the impulse response, Asymptotic stability of LSI discrete-time systems.*
3.	***Z-Transform and Study of discrete-time system using Z-transform*** *Z-Transform Definition (Direct and Inverse Z-Transform), Region of Convergence (ROC of sequences of infinite duration, ROC of sequences of finite duration), Relation of Z-Transform with Laplace Transform), One-Sided Z-Transform, Properties of Z-Transform (Linearity, Time Shift, Time Reversal, Time Scaling, Complex Frequency Scaling, Convolution Theorem, Z Field Derivation, Complex Conjugate, Signal Multiplication, Initial Value Theorem, Final Value Theorem), Poles and zeros of the Z-Transform, Calculate the Inverse Z-Transform using expansion in partial sums, Description of discrete-time system in Z-plane (Transfer Function, Relation between Transfer Function and Difference Equation, Frequency Response, Poles and Zeros of Transfer Function, All-Pole and All-Zero systems, Causality and Stability Theorems), Solution of Difference Equations.*
4.	***Discrete-Time Fourier Transform (DTFT) and Discrete Fourier Transform (DFT)*** *Discrete Time Fourier Transform (DTFT), Properties of DTFT (Periodicity, Symmetry and Conjugation, Linearity, Time Reversal, Time Shift, Frequency Shift , Variation in Frequency, Convolution Theorem, Periodic Convolution, Correlation, Parseval Theorem), Inverse DTFT,*

	Relation of DTFT to other Transforms (Fourier transform, Z transform), Sampling rate transformation (Upsampling, Downsampling), Discrete Fourier Transform (DFT), Twiddle Factors, Magnitude and Phase Spectra, Relation of DFT to other Transforms (DTFT Transform, Z Transform), DFT calculation with tables, DFT Properties (Linearity, Cyclic Convolution in Time, Cyclic Time Shift, Conjugation, Symmetry of DFT for Real Sequences, Symmetry of DFT for Complex Sequences, Periodic Sequence Expansion, Periodic Convolution, Cyclic Frequency Shift, Cyclic Convolution, Multiplication of Sequences, Parseval's Theorem), Relation of Cyclic Convolution to Linear, Calculate the Circular Convolution using DFT, Fast Fourier Transform, Strategy for building efficient DFT calculation algorithms (Decimation in Time, Decimation in Frequency), Properties of Frequency Response, DTFT Applications (Calculating Frequency Response, Solving Differential Equations, Designing Inverse Systems, System Connections).
5.	***Digital FIR and IIR Filters*** *Filter gain control, Minimum, Maximum, Mixed and Linear Phase filters, Ideal Frequency selective filters, Specifications of real digital filters, Finite Impulse Response (FIR) Filters, FIR Filters in time and frequency domains, FIR Filter as a delay line, Types of FIR linear phase filters, FIR Filter Design Methods (Window Method, Frequency Sampling Method, Equiripple (optimal) Method), Study of Window sequences (Rectangular, Triangular (Bartlett), Hanning, Hamming , Blackman, Kaiser), Infinite Impulse Response (IIR) Filters, IIR Filter Design (General IIR Filter design method, Individual IIR Filter design methods), Standard Low-Pass analog filter (Prototype Butterworth, Chebyshev I & II, and Elliptic Low-Pass Filters), IIR Filter design methods (Invariant Impulse Response, Bilinear Transform), Effect of finite word length on filter accuracy.*

LEARNING RESOURCES AND TOOLS.

The learning resources and tools that will be utilized for the delivery of the course are the following:

- *PowerPoint lectures (5 sets, 500 slides)*
- *Solved Examples (5 sets, 50 exercises)*
- *Tutorials for the hands-on lab using Matlab (60 solved examples)*
- *Auto-evaluation quizzes (15 sets, 500 questions)*
- *Supporting training material (docx, pdf and html formats)*

Relevant videos from Internet given that no IP constraints exist.

PLANNED LEARNING ACTIVITIES AND TEACHING METHODS.

The planned learning activities and teaching methods are the following:

- *Lectures by university's faculty members*
- *Group assignments*

Moreover, a forum will be used to facilitate the communication between students, as well as between students and the teachers.

7. ASSESSMENT METHODS, CRITERIA AND PERIOD.

The performance of the students will be assessed using the following tools:

- *One group assignment, deadline June 8, 2025 (70% weight)*
- *Online quiz (20 questions at 60 minutes - June 15, 2025 (30% weight)*

OBSERVATIONS.

8. BIBLIOGRAPHY AND TEACHING MATERIALS.

- *Chaparro L.F., «Signals and Systems Using Matlab», Academic Press, Elsevier, 2nd edition, 2015.*
- *Hsu H. «Signals and Systems», (Schaum's Outlines), McGrawHill, 3rd edition, 2013.*
- *Ingle V.K., Proakis J.G, «Digital Signal Processing using MATLAB», Brooks Cole, 2000.*
- *Lathi B.P., Green R., «Essentials of Digital Signal Processing» Cambridge University Press, 2014.*
- *Lavry D., «Sampling Theory for Digital Audio», Lavry Engineering Inc., online edition.*
- *Oppenheim A.V., Schafer R.W., Buck J.R., «DiscreteTime Signal Processing», PrenticeHall, 1999.*
- *Prandoni P., Vetterli M., «Signal Analysis for Communications», EPF Press, online edition, 2008.*
- *Proakis J.G., Manolakis D.G., «Digital Signal Processing: Principles, Algorithms and Applications», 4th edition, Prentice Hall, 2007.*
- *Vetterli M., Kovacevic J., «Foundations of Signal Processing», Cambridge University Press, 2014.*
- *Ziemer R.E. et al, «Signals and Systems: Continuous and Discrete», PrenticeHall, 1998.*