Course Title	Digital Signal Processing				
Course Code	ACOE418				
Course Type	Elective				
Level	BSc (Level 1)				
Year / Semester	4th (Fall/Spring)				
Teacher's Name	Prof.. Michael Komodromos				
ECTS	6	Lectures / week	3	Laboratories/week	0
Learning Outcomes	1. Classify the various types of signals. State the sampling theorem and show how discrete time signals are obtained. Manipulate discrete time signals utilizing the unit sample and the unit step signals. 2. Define discrete time systems. Examine and classify systems based on linearity, time invariance (LTI) and causality. Express discrete time systems using difference equations. Derive the impulse response of FIR and IIR systems from their difference equations and vice versa. Describe and appraise the differences between FIR and IIR systems. Derive and draw system flow diagrams. Evaluate the output of LTI systems by convolution and by the difference equation. 3. Analyze LTI discrete-time signals and systems using the z-transform. Obtain the transfer function, the poles and zeros of the system. Examine the BIBO stability of the system. Analyze LTI discrete-time signals and systems using the Fourier transform. Obtain the frequency response of the system. Utilize the properties of the transforms in the analysis of signals and systems. Determine the phase and group delay of linear phase FIR filters. 4. Define the Discrete Fourier Transform (DFT) and Fast Fourier Transform (FFT). Relate these transforms to the z-transform. Calculate the DFT of discrete time signals. 5. Classify digital filters according to being ideal or non-ideal, FIR or IIR, causal or non-causal and their frequency selectivity. Define the problem formulation of digital filter design. Design and analyze linear phase FIR filters using MATLAB.					
Prerequisites	AELE310	Co-requisites	None		
Course Content	<u>A. Discrete time signals and systems in the time domain</u> (a) Introduction to signals, systems and signal processing applications. Classification of signals. Continuous time vs discrete time signals. Continuous valued vs discrete valued signals. (b) Frequency in continuous time vs discrete time. Analog-to-digital conversion. Sampling process and sampling theorem. (c) Discrete-time systems. Input-output description of systems. Block diagram representation of systems. Properties of linearity, time invariance, causality and stability. (d) Discrete-time Linear Time-Invariant systems. Impulse response. Convolution.					

	<u>B. Evaluation and use the z-transform</u> (a) Definitions and evaluation of the z-transform. Properties. Z-plane. (b) Rational z-transforms. Zeros and poles, zero-pole diagrams and the unit circle. Stability and causality. (c) Inverse z-transform. Partial fraction expansion. (d) Discrete-time system analysis using the z-transform. Calculation of the system impulse response from the transfer function. <u>C. Frequency domain analysis of discrete-time signals and systems</u> (a) Frequency analysis of discrete-time signals using the Fourier transform. Properties of the Fourier transform. (b) Relationship of the Fourier transform to the z-transform. (c) Frequency domain analysis of Linear Time Invariant (LTI) Systems. Input-Output Relations in the frequency domain. Frequency response of LTI systems. Magnitude and phase of the frequency response. Linear phase systems. Group delay. (d) Definition and calculation of the Discrete Fourier Transform (DFT). Examination of the Fast Fourier Transform (FFT) and classification of FFT algorithms. <u>D. Digital Filters</u> (a) Introduction to digital filters. Frequency response and impulse response of ideal digital filers. (b) Frequency selective filters. Low-pass, Band-pass, and high-pass digital filters. Implementation of FIR vs IIR filters. • (c) Design methods of digital filters. Linear phase FIR filters. Use of the MATLAB DSP toolbox.
Teaching Methodology	The teaching of the course is lecture-based (3 hours per week) in a classroom, using a combination of traditional teaching with written notes on a white board and slide presentations using a projector for the presentation of the more complicated diagrams, graphs and MATLAB design tools. Students are assessed continuously and their knowledge is checked through tests, assignments and the final exam. Lectures include the solution and discussion of example problems regarding the material presented. Relevant homework and assignments are given to the students for further study at their own. Due to the level and type of the course students are urged to participate in discussing the various topics and provide their opinion during problem-solving sessions. Lecture notes are compiled by students which serve to cover the main issues under consideration and serve as a guide for further reading. Students are also required to seriously use the textbook assigned to the course, in addition to other sources found either in the library or elsewhere in order to broaden their perspective on the various subjects presented in class and in the textbook. Additionally, they are expected to use the MATLAB DSP toolbox for the analysis of signals and systems and the design and analysis of digital filters.
Bibliography	<u>Textbooks:</u> J. Proakis and D. Manolakis, *<u>Digital Signal Processing</u>*, 4th edition, Prentice Hall, 2007. <u>References:</u>

	• J. H. Mclellan, R. W. Schafer, M.A. Yoder, *Signal Processing First*, Pearson Education, 2003. • S. Mitra , *Digital Signal Processing: A Computer Based Approach*, McGraw Hill, 2002.
Assessment	Students are assessed on the theoretical aspects of the course through tests, and the final exam, while lab exercises cover the applied and hand-on aspects of the course. Coursework will comprise of one test, a set of lab exercises, and three-hour closed book exam. The weights for each assessment component are: • Assignments 20% • Tests: 20% • Final Exam 60%
Language	English