AEEE424 - Digital Signal Processing

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<tr><td>Course Title</td><td colspan="5">Digital Signal Processing</td></tr>
<tr><td>Course Code</td><td colspan="5">AEEE424</td></tr>
<tr><td>Course Type</td><td colspan="5">Compulsory</td></tr>
<tr><td>Level</td><td colspan="5">BSc (Level 1)</td></tr>
<tr><td>Year / Semester</td><td colspan="5">4 / 2</td></tr>
<tr><td>Teacher’s Name</td><td colspan="5">Prof Michael Komodromos</td></tr>
<tr><td>ECTS</td><td>6</td><td>Lectures / week</td><td>3</td><td>Laboratories/week</td><td></td></tr>
<tr><td>Course Purpose</td><td colspan="5">The aim of the course is to familiarize students with the concepts of discrete-time signals and discrete-time systems. The course introduces the basic theories of sampling of continuous-time signals, manipulation and analysis of discrete-time signals in the discrete-time domain and in the frequency domain. The course emphasizes on the analysis of discrete-time systems both in the time and frequency domains and introduces some concepts of design of discrete time systems.</td></tr>
<tr><td>Learning Outcomes</td><td colspan="5">By the end of the course, students must be able to:
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</td></tr>
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	filters using MATLAB.		
Prerequisites	AEEE310	Corequisites	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. <u>E. Advances in Technology and Current Trends in Research</u> (a) Examine and discuss the current state of the technology in digital signal processing systems and new applications. (b) Overview current trends in research and new technological and scientific challenges in digital signal processing.		

Teaching Methodology	Teaching of the course is based on lectures (3 hours per week) in a classroom, using a mixture of traditional teaching with notes on the white board and slide presentations using a projector where appropriate. Topic notes are compiled by students, during the lectures which serve to cover the material of the course. Students are urged to use the textbook assigned to the course. Homework problems are assigned from the textbook as a turn-in assignment or for interactive homework practice. Additionally, students are advised to use the reference books for further reading and practice in solving related exercises. Example problems are solved during lectures or privately during the lecturer's office hours. Students are assessed continuously and their knowledge is checked through tests and assignments.
Bibliography	**Textbook:** J. Proakis and D. Manolakis, *Digital Signal Processing*, 4th edition, Prentice Hall, 2007. **References:** J. R. Johnson, *Introduction to Digital Signal Processing,* Prentice Hall, 1999. 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	The Students are assessed via continuous assessment throughout the duration of the Semester, which forms the Coursework grade and the final written exam. The coursework and the final exam grades are weighted 40% and 60%, respectively, and compose the final grade of the course. The continuous assessment of the students is achieved through assignments and tests. An indicative weighted continuous assessment of the course is shown below: • Assignments 25% • Design Project 15% • Exams and Quizzes 60% Students are prepared for the final exam through revisions on the material taught, problem solving and concept testing. The final assessment of the students is formative and summative and is assured to comply with the subject's expected learning outcomes and the quality of the course.
Language	**English**