
SYLLABUS

Date/ Revision	25 August 2016 / Rev. 01
Faculty	Engineering
Approval	Dean of Engineering Faculty

SUBJECT : DIGITAL SIGNAL PROCESSING

1. Identification of Subject:

Name of Subject	: Digital Signal Processing
Code of Subject	: DTSP-3100
SKS	: 3
Semester	: 6
Study Program	: ELE, MTE
Lecturer	: To be announced

2. Competency

After having the course, students are expected to:

- Describe the Sampling Theorem and how this relates to Aliasing and Folding.
- Determine if a system is a Linear Time-Invariant (LTI) System.
- Take the Z-transform of a LTI system
- Determine the frequency response of FIR and IIR filters.
- Understand the relationship between poles, zeros, and stability.
- Determine the spectrum of a signal using the DFT, FFT, and spectrogram.
- Design, analyze, and implement digital filters in Matlab/SciLab.
- Explain the typical features of a digital signal processing chip.

3. Description of Subject:

Digital Signal Processing (DSP) is concerned with the digital representation of signals and the use of microprocessors and computers to analyze, modify, and extract information from signals. The digital signals found in most popular applications of DSP are derived from analog signals that have been sampled at regular intervals and converted into digital form.

The first part of this course covers the fundamentals of discrete-time signals and systems. Students will study key DSP operations such as convolution, filtering, and discrete Fourier transforms. Even during this early stage they will practice some applications of the theory covered in class. Then students will progress to digital filter design and spectral analysis, which are the two major branches of DSP. Matlab/Scilab software will be used as a tool in designing of digital filter, such as IIR-Filter and FIR-Filter.

4. Learning Approach

Approach	: Combination of Expository - inquiry and collaborative
Method	: Discussion, question answer, sample problem, group work
Student Task	: Home work, presentation
Media	: LCD projector, film.

5. Evaluation

a) Absence maximum	: 25%
b) Participation in discussion	: 5 points
c) Homework, Classwork	: 10 points
d) Daily Quiz	: 25 points
e) Final Examination	: 60 points

Total : 100 points

6. Contents/ Topics of Lecturing:

Week	Content/ Topics of Lecturing	Text Book Chapter	Remark
1	Introduction to Digital Signal Processing <ul style="list-style-type: none"> Signal, System, and Signal Processing, Basic elements of digital signal processing system, Advantages Digitals over analog signal processing Classification of signals: Multi-channel and multi-dimensional signals, CT-signals vs. DT-signals, Continuous valued vs. discrete valued signals, Deterministic vs. random signals, Analog-to-Digital and Digital-to-Analog Conversion 	Ch-01	
2	Discrete Time Signal: <ul style="list-style-type: none"> Discrete-Time Signals and Discrete-Time Systems Analysis of Discrete-Time Linear Time-Invariant systems Discrete-Time Systems Described by Difference Equations Implementation of Discrete-Time Systems Correlation of Discrete-Time Signals 	Ch-02	
3	The z-Transform and Its Application to the Analysis of LTI Systems <ul style="list-style-type: none"> The z-Transform Properties of the z-Transform Rational z-Transforms Inversion of the z-Transform Analysis of Linear Time Invariant Systems in the z-Domain The One-sided z-Transform 	CH-03	Quiz-1
4	Frequency Analysis of Signals and Systems: <ul style="list-style-type: none"> Frequency Analysis of Continuous-Time Signals Frequency Analysis of Discrete-Time Signals Frequency-Domain and Time-Domain Signal Properties Properties of the Fourier Transform for Discrete-Time Signals 	CH-04	
5	Frequency-Domain Analysis of LTI Systems <ul style="list-style-type: none"> Frequency-Domain Characteristics of Linear Time-Invariant Systems 	CH-05	Quiz-2

	<ul style="list-style-type: none"> • Frequency Response of LTI Systems • Correlation Functions and Spectra at the Output of LTI Systems • Linear Time-Invariant Systems as Frequency-Selective Filters • Inverse Systems and Deconvolution 		
6	Sampling and Reconstruction of Signals <ul style="list-style-type: none"> • Ideal Sampling and Reconstruction of Continuous-Time Signals • Discrete-Time Processing of Continuous-Time Signals • Analog-to-Digital and Digital-to-Analog Converters • Sampling and Reconstruction of Continuous-Time Bandpass Signals • Sampling of Discrete-Time Signals • Oversampling A/D and D/A Converters • 	CH-06	
7	The Discrete Fourier Transform: Its Properties and Applications <ul style="list-style-type: none"> • Frequency Domain Sampling: The Discrete Fourier Transform • Properties of the DFT • Linear Filtering Methods Based on the DFT • Frequency Analysis of Signals Using the DFT • The Discrete Cosine Transform 	Ch-07	Quiz-3
8	MID-SEMESTER BREAK		
9	Efficient Computation of The DFT: FFT Algorithms <ul style="list-style-type: none"> • Efficient Computation of the DFT: FFT Algorithms • Applications of FFT Algorithms • A Linear Filtering Approach to Computation of the DFT • Quantization Effects in the Computation of the DFT 	CH-08	
10	Implementation of Discrete-Time Systems <ul style="list-style-type: none"> • Structures for the Realization of Discrete-Time Systems • Structures for FIR Systems • Structures for IIR Systems • Representation of Numbers • Quantization of Filter Coefficients • Round-Off Effects in Digital Filters 	CH-09	Quiz-4
11-12	Design of Digital Filters <ul style="list-style-type: none"> • General Considerations • Design of FIR Filters • Design of IIR Filters From Analog Filters • Frequency Transformations 	CH-10	
13-14	Multirate Digital Signal Processing <ul style="list-style-type: none"> • Decimation by a Factor D • Interpolation by a Factor I • Sampling Rate Conversion by a Rational Factor I/D • Implementation of Sampling Rate Conversion • Multistage Implementation of Sampling Rate Conversion • Sampling Rate Conversion of Bandpass Signals • Sampling Rate conversion by an Arbitrary Factor • Applications of Sampling Rate Conversion • Digital Filter Banks • Two-Channel Quadrature Mirror Filter Bank • M-Channel QMF Bank 	CH-11	Quiz-5
15	Adaptive Filters <ul style="list-style-type: none"> • Applications of Adaptive Filters 		

	<ul style="list-style-type: none"> • Adaptive Direct-Form FIR Filters-The LMS Algorithm • Adaptive Direct-Form FIR Filters-RLS Algorithms • Adaptive Lattice-Ladder Filters 		
16	Final Examination		

7. Book Reference:

Main Text Book:

“DIGITAL SIGNAL PROCESSING 4ED”, **Author:** John G. Proakis and Dimitris G. Manolakis,
Publisher: Pearson Internal Edition, 2014, **ISBN:** 928-1-292-02573-5

Supplementary Text Books:

- a) “Digital Signal Processing: Fundamentals and Applications, 2Ed”, **Author:** Li Tan and Jean Jiang,
Publisher: Academic Press- Elsevier, **ISBN:** 978-0-12-415893-1
- b) “Digital Signal Processing using Matlab – 3Ed”, **Author:** Robert J. Schilling and Sandra L. Harris,
Publisher: Cengage Learning, 2017, **ISBN:** 978-1-305-63519-7

[Subject to change / MaS / Rev.01]