

KING FAHD UNIVERSITY OF PETROLEUM & MINERALS

ELECTRICAL ENGINEERING DEPARTMENT

EE406 - Digital Signal Processing Syllabus

First Semester (121)



Instructor	Office	Phone	E-mail	Office Hours
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EE 406: Digital Signal Processing (3-0-3)

Classification of signals and their mathematical representation. Discrete-time systems classification. Linear shift-invariant system response, difference equations, convolution sum, and frequency response. Discrete Fourier transform. Z-transform and its application to system analysis. Realization forms. Sampling and aliasing. Finite-impulse response (FIR). Design windowing technique. Introduction to infinite impulse response (IIR). Filter design techniques.

Pre-requisite: EE 207

Textbook:

1. J. G. Proakis and D. G. Manolakis, *Digital Signal Processing: Principles, Algorithms and Applications*, 4th Edition, Prentice Hall, 2007.

Other references:

2. A. V. Oppenheim and W. Schaffer, *Digital-Time Signal Processing*, 4th Edition, Oxford Publishing, 1998.
3. S. K. Mitra, *Digital-Time Signal Processing- A Computer-based Approach*, 4th Edition, McGraw-Hill Oxford Publishing, 2009.
4. R. A. Roberts and C. T. Mullis, *Digital Signal Processing*, Addison-Wesley, 1987.
5. L. B. Jackson, *Digital Filters and Signal Processing*, 3rd Edition, KAP, 1995.

Content Breakdown:

- Signals and Systems: Classification of signals; Linear shift-invariant systems; System response; Convolution; Stability and causality.
- The z-Transform: Definitions and region of convergence; Inverse z-transform; Properties of the z-transform; Realization; System function; Frequency response; Difference equations.
- Discrete-Time Networks: Signal flow graphs; Realizations forms: Direct, cascade, and parallel forms.
- Sampling and Discrete-Time Fourier Transform: Definitions; Convergence conditions; Properties of the DTFT; Aliasing; Analog-to-digital and digital-to-analog conversions; Multi-rate Analysis
- Introduction to Discrete Fourier Transform: Definitions; Properties; Efficient computation of the DFT; FFT algorithms.
- Introduction to Digital Filters Design: FIR versus IIR; Linear-phase filters; Windowing design techniques for FIR; Finite word length effect analysis on the filter design.

Week	Topics	Sections from Book
1	Introduction signals and systems, classification of signals, frequency domain representation, A/D and D/ A conversion, discrete times signals	1.1, 1.2, 1.3, 1.4, 2.1
2	Discrete time signals and systems discrete-time systems, LTI systems, difference equations, implementing discrete time systems	2.2, 2.3, 2.4, 2.5
3	Z-transform and applications Definition of z-transform, properties, rational z-transforms, inversion of z-transform	3.1, 3.2, 3.3, 3.4
4	Z-transform and applications Inversion of z-transform, one-sided z-transform, analysis of systems using z-transform	3.4, 3.5, 3.6
5	Frequency analysis of signals and systems Frequency analysis of continuous time signals (review), frequency analysis of discrete time signals	4.1, 4.2
6	Frequency analysis of signals and systems Properties of Fourier transform of discrete time signals, frequency domain analysis of LTI systems	4.3, 4.4 5.1, 5.2
7	Frequency analysis of signals and systems Description of different filters Discrete Fourier transform Sampling in frequency domain	4.5, 5.4 7.1
8	Discrete Fourier transform Properties of DFT, filtering based on DFT, analysis of signals using DFT	7.2, 7.2, 7.3
9	Fast DFT Algorithms FFT algorithms, application of FFT algorithms	8.1, 8.2
10	Implementing discrete time systems Structure of discrete time systems, structure of FIR systems, structure of IIR systems	9.1, 9.2, 9.3
11	Design of digital filters General considerations, design of FIR filters	10.1, 10.2
12	Design of digital filters design of FIR filters, design of IIR filters	10.2, 10.3
13	Design of digital filters Frequency transformations, least Square methods	10.4, 10.5
14	Sampling and Reconstruction Sampling of low pass and bandpass signals, A/D conversion, D/A conversion	6.3, 6.4, 6.5, 6.6
15	Multirate DSP Decimation, interpolation, sampling rate conversion, applications	11.1, 11.2, 11.3, 11.4

Course Objective:

The student should be able to:

- Apply mathematical tools to discrete systems in the time-domain (convolution, difference equation, impulse response $h(n)$, step response, etc.).
- Analyze digital systems in the z-domain (transfer function of systems $H(z)$ and realization of systems).
- Apply mathematical tools to discrete systems in the frequency-domain (DTFT, DFT, FFT, frequency response of systems $H(e^{j\omega})$).
- Convert system from digital to analog and analog to digital (sampling theorem, aliasing, up-sampling and down-sampling).
- Design filters to meet frequency domain specifications (FIR and IIR).
- Use MATLAB to analyze and design DSP systems to master all the above objectives (convolution, FFT, Z-domain plots, FIR and IIR filters).

Course Outcomes:

Outcome 1: An ability to apply knowledge of mathematics, science, and engineering to the analysis and design of digital system

Outcome 2: An ability to identify, formulate, and solve engineering problems in the area signal processing.

Outcome 3: An ability to use the techniques, skills, and modern engineering tools such as Matlab and digital processors.

Outcome 4: An ability to function on multi-disciplinary teams

Outcome 5: An ability to design a system, components or process to meet desired needs within realistic constraints such as economic, environmental, social political, ethical, health and safety, manufacturability and sustainability

Grading:

- 15 %: Homework
- 10 %: Quizzes
- 20 %: Midterm Exam
- 30 %: Final Exam
- 25 %: Term Project
 - 1. Two Progress Reports (2 % Each)
 - 2. Presentation, Week (6%)
 - 3. Final Report (15%)

Project Topics and Ideas:

1. PAPR reduction in OFDM systems.
2. Pre-distorters design to linearize the nonlinear high power amplifier.
3. Using wavelet instead of FFT to process PFDM signal.
4. Compression using discrete cosine transform DCT.
5. Watermarking.
6. Compressive sensing.
7. Wavelet and curvelet.
8. De-convolution and (blind) channel identification (equalizers).
9. Voice recognition.
10. Harmonics analysis using FFT or Wavelet.
11. Comb filter design to remove the 60 Hz frequency and its harmonics.
12. Image Compression
13. Any other topic preferably related to DSP.

You can work in group of 2 or 3.

Course Policies

- Homework. It is often helpful to work in study groups when doing homework problems to discuss the material, but each student must still work the assignment and write it up or program it individually. Late homework assignments will not be accepted except in the case of an excused absence.
- Term Project. It is based on a paper that you will read and understand. You must reproduce the results and indicate possible improvement or future work. You will have to read other related material to understand the paper. You will hand in a short report along with a presentation. You can work in groups of 2-3.
- Attendance. Attendance will be taken daily. For each unexcused absence, there will be a 0.5 mark deduction. Two late marks equal one absent. More than 20% absences (more than 9) from the total number of classes will result in a DN grade. In case of an excused absence in a quiz or in an exam, there will be no makeup of quizzes or exams; however, the student will be given his average.
- Cheating. Any form of cheating will result in a zero for that assignment and possibly additional penalties, including a failing grade in the class and reporting to the department. It is the responsibility of each student to safeguard his work from being copied.
- Policy Changes. The instructor reserves the right to modify the course outline and policies mentioned in this syllabus at any time during the semester.

This information and more will be available on Blackboard CE 8