

# EE 351M Digital Signal Processing

Tue, Thu 11:00am-12:30pm

ECJ 1.204

**Instructor:** Haris Vikalo

- Email: [hvikalo@ece.utexas.edu](mailto:hvikalo@ece.utexas.edu)
- Phone: (512) 232-7922
- Office: ACES 3.110
- Hours: Tue, Thu 1:00pm-2:30pm

**Teaching Assistant:** Kitaek Bae

- Email: [ktbae@mail.utexas.edu](mailto:ktbae@mail.utexas.edu)
- Office: TBA
- Hours: TBA

**Electronic course site:** We will use Blackboard ([courses.utexas.edu](http://courses.utexas.edu)) to deliver homework assignments and any extra reading materials. You should be able to log in if you have a valid UT ID and are registered for this class. In addition, basic course information and handouts will also be available at <http://users.ece.utexas.edu/~hvikalo/ee351m.html>.

**Textbook:** *Discrete-Time Signal Processing*, (Oppenheim, Schaffer, and Buck), Prentice-Hall, 2nd edition, 1998, ISBN 0-13-754920-2.

**Homeworks and exams:** There will be roughly weekly homework assignments; there will be two midterm exams in class, and a final exam. The weighting will be as follows:

- Homeworks: 15%
- Midterm exams: 45%
- Final exam: 35%
- Class participation: 5%

Homeworks are to be submitted at the *beginning* of the class when they are due. You may discuss homework problems with other students, but must submit your own independent solution (this includes any Matlab code required for the assignments). Late homework assignments will not be accepted.

**Course prerequisites:** EE313 Linear Systems and Signals and EE351K Probability and Random Processes, with a grade of at least C in each.

**Official course description:** Sampling, aliasing, truncation effects; discrete and fast Fourier transform methods; convolution and deconvolution; finite and infinite impulse response filter design methods; Wiener, Kalman, noncausal, linear phase, median, and prediction filters; and spectral estimation.

**Course Introduction:** Signal processing deals with representation, transformation, and manipulation of signals and the information they contain. It is a rich subject with tools that have applications in a broad class of problems including communications, control, image processing, biomedicine, sonar, radar, array processing, and digital video.

This course provides a thorough treatment of DSP including the fundamental theorems and properties of discrete-time linear systems, filtering, sampling, and discrete-time Fourier analysis. The emphasis in the class will be on digital signal processing algorithms, their derivations, and their applications.

By the end of this course, you should know how to:

- analyze discrete-time systems by examining their input and output signals,
- compute a system output (in either time or frequency) given the system input and a description of the system,
- represent a continuous-time signal by a discrete-time signal (using the conditions for perfect reconstruction),
- filter continuous-time signals in discrete-time (and vice versa),
- design good digital filters based on well-known analog filter designs,
- use the Fourier transform, Fourier series, discrete Fourier transform, and the z-transform and know when to use them,
- implement fast discrete Fourier transform,
- derive and implement optimal filters for stochastic signals,
- analyze the spectrum of real signals (both deterministic and stochastic).

### **Outline of the topics:**

- Review of Discrete-Time Signals and Systems
- z-Transform
- Sampling and Digital Processing of Continuous-Time Signals
- Analysis of Linear Time-Invariant Systems
- Structures for Discrete-Time Systems
- Filter Design Techniques
- The Discrete Fourier Transform
- The Fast Fourier Transform
- Analysis of Signals Using the DFT
- Advanced Topics