The University Of Sheffield.

Electronic & Electrical Engineering.

EEE309

INTRODUCTION TO DIGITAL SIGNAL PROCESSING

Credits: **10**

Course Description including Aims

1. To introduce fundamental ideas of digital signal processing (DSP), its limitations and its advantages.
2. To give the student a working knowledge of basic DSP operations, as well as a solid theoretical understanding of their behaviour.
3. To make the student aware of the options available when constructing a DSP system.

Outline Syllabus

Discrete-time signals and systems, z-transform, sampling of continuous-time signals, discrete-time Fourier transform (DTFT), transform analysis of linear time-invariant (LTI) systems, structures for discrete-time systems, discrete Fourier transform (DFT), fast Fourier transform (FFT), finite impulse response (FIR) filter design techniques: window method and frequency sampling method, infinite impulse response (IIR) filter design techniques: impulse invariance method and bilinear transform method.

Time Allocation

24 hours of lectures

Recommended Previous Courses

EEE224 “Communication Electronics”

Assessment

2 Hour Examination

Recommended Books

Ifeachor & Jervis	*Digital Signal Processing - A practical approach*	(Addison-Wesley)
Proakis & Manolakis	*Digital Signal Processing - Principles, algorithms and applications, second edition*	(Macmillan student edition)
Meddins, B	*Introduction to digital signal processing*	(Newnes)
Mulgrew, Grant and Thompson	Digital signal processing : concepts and applications	(Palgrave Macmillan)
Oppenheim and Schafer	Discrete-time Signal Processing	(Prentice Hall)

Objectives

At the end of the course, the student will be able to

1. Understand the necessary changes of choosing a digital solution when processing signals.
2. Understand and exploit the relationship between the time and frequency domain representations of linear time-invariant (LTI) discrete-time systems and the signals passing through them.
3. Design simple FIR and IIR filters to satisfy desired performance specifications.
4. Demonstrate a broad knowledge of well-known digital signal processing tools.

Detailed Syllabus

1. Introduction to the course, introduction to DSP, what it is, why we use it.

2. Basic discrete-time signals/sequences and operations: unit sample sequence, unit step sequence, exponential sequence, sinusoidal sequence, differences between discrete-time and continuous-time complex exponential or sinusoidal sequences (frequency range, periodicity).

3. Discrete-time systems: ideal delay system, moving average system, memoryless system, linear system, time-invariant/shift-invariant system, causal system, stable system.

4. Linear time-invariant (LTI) system: impulse response of an LTI system, convolution, why an LTI system is completely characterized by its impulse response, properties, FIR system, IIR system.

5. Linear constant-coefficient difference (LCCD) equations, frequency-domain representation of discrete-time signals and systems, frequency response of an LTI system, sinusoidal response of an LTI system.

6. Discrete-time fourier transform (DTFT), properties of DTFT, DTFT and the frequency response of an LTI system.

7. Z-transform, its relationship with DTFT, region of convergence (ROC), pole-zero plot.

8. Properties of ROC, inverse z-transform (inspection method, power series expansion).

9. Properties of z-transform (focused on its application to convolution of sequences in time-domain).

10. Sampling of continuous-time signals, periodic sampling, Nyquist sampling theorem, reconstruction of the continuous-time signal, discrete-time processing of continuous-time signals, and continuous-time processing of discrete-time signals.

11. Transform analysis of LTI systems: frequency response of an LTI system (ideal frequency-selective filters, phase distortion and delay), system functions for systems characterized by LCCD equations (stability and causality, inverse systems).

12. Impulse response for rational system functions, frequency response for rational system functions, frequency response of a single zero or pole, frequency response of multiple poles and/or zeros.

13. Structures for discrete-time systems: block diagram representation of LCCD equations (addition of two sequences, multiplication of a sequence by a constant, unit delay), direct form I and direct form II implementation of an LTI system.

14. Discrete Fourier transform (DFT): DFT defined on finite duration sequences, DFT as sampling of the DTFT, properties of the DFT.

15. Circular convolution, linear convolution using the DFT for two finite-length sequences, implementing LTI systems using the DFT (overlap-add method).

16. Fast Fourier transform (FFT): decimation-in-time algorithm for sequence length being the power of two, summary of Fourier-related transformations (complex Fourier series, Fourier transform (for continuous signals), DTFT, DFT).

17. Design of IIR filters I: filter design specifications (passband, stopband, transition band), impulse invariance method,

18. Design of IIR filters II: bilinear transform method, comparison of the bilinear transform method with the impulse invariance method, IIR filter design for highpass and bandpass filters.

19. Design of FIR filters I: linear phase shift characteristic, Gibbs phenomenon, window method.

20. Design of FIR filters II:, frequency sampling method, relative advantages and disadvantages of FIR/IIR filters:

UK-SPEC/IET Learning Outcomes

Outcome Code	Supporting Statement
SM1p / SM1m	Fundamental ideas of digital signal processing and related math, sampling process/theory, digital systems vs. analogue systems. (assessed by examination)
SM2p / SM2m	Convolution, LCCD equations, z-transforms and Fourier-related transforms, and their application to FIR/IIR filter design problems. (assessed by examination)
EA1p / EA1m	Fourier analysis for the filtering operation and the sampling process, and analysis of the frequency response of LTI systems; comparison and critical analysis of different FIR/IIR design methods. (assessed by examination)
EA3p/EA3m	Design of FIR/IIR filters, draw frequency response of filters using MATLAB. (assessed by examination)