Digital Signal Processing CPE3007

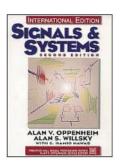
Lecturers: A/Prof Deepu Rajan, A/P Chng Eng Siong

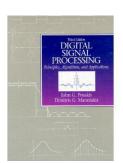
This course "Digital signal processing" is a very well established course offered commonly in many Universities. Learning it will support many interests such as pattern recognition, speech and audio processing. In addition, there are many great online video resources (click on the images):

Examples of video courses from Stanford, MIT and IIT:

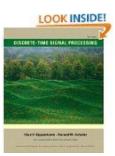


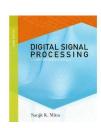
And many established books on DSP:













Why learn DSP? By understanding DSP well, you will be able to appreciate the followings:

- 1) Sampling Theorem:
 - a. How to make water flow upwards! (sampling theorem) http://www.youtube.com/watch?v=oqUNd5wPGbU&feature=related
 - b. Why the propeller of a helicopter stopped?http://www.youtube.com/watch?v=rVSh-au 9aM&feature=related
- 2) Understand whats frequency spectrum
 - a. So that you know how to make own guitar tuner, see: http://www.youtube.com/watch?v=fCV4rhLlrh8

- 3) How to change audio signals (filtering)
 - a. (i) Clean wavefile, (ii) metal room (iii) echo chamber
 - b. You can try voice morphing: http://www.youtube.com/watch?v=0isqJx sBYE

CE 3007- Digital Signal Processing

Title	Digital Signal Processing								
Course code(s)	CE 3007	Year of Study	3	Academic Year:			2013-2014		
Academic Units	3	Prerequisite(s)	CE2004						
Allocation in hours	Lectures	26	Tuto	orials	13	Laboratories		10	
Prerequisite for	Nil								
Dated (dd mmm yyyy)	18 Jun 2013								
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Responsible, contacts	Deepu Ra	jan (Tel: 67904933))						

Rationale

Digital signal processing (DSP) is everywhere, literally. Consumer electronics such as media players, digital cameras, cell phones, medical equipment such as hearing aids and MRI scanners, and communication infrastructure, all have one or several aspects of DSP inherent in them. An engineer working in any of the above or related domains needs a thorough understanding of DSP, which essentially brings together signals and systems theory, computation algorithms and hardware design under one umbrella. DSP engineers need to understand the design of a DSP system from theory to implementation. Advances in DSP design tools such as Matlab and high-level language compilers and simulators have also simplified the learning of DSP significantly.

Delivery

Design of DSP systems requires a solid foundation in theory as well as knowledge of DSP processors. The lectures will provide grounding in discrete time systems and transforms of signals. They will also empower the students with fundamental knowledge in filter design, which will be useful for design of more advanced DSP systems. The lectures will be complemented by laboratory experiments in Matlab for the students to see the theory 'working' and also to give additional insight when carrying out the simulations. There will be a course project on analysing and detecting dual tone multi frequency (DTMF) signals used in phone touchpads. Tutorial exercises will be given to enable rigorous practice in problem solving.

Course Content

	Topics	Lectures (Hours)	Tutorials (Hours)
1	Digital Signal Processors Overview of DSP applications – DSP implementation on general purpose processor (GPP), general purpose DSP processors, application specific instruction processor (ASIP), application specific integrated circuit (ASIC) – Harvard architecture – Hardware units (shifter, MAC, on-chip memory) – Choosing a DSP processor.		1

2	Discrete-time Signals and Systems Discrete-time signals – Discrete-time systems – Linear time invariant systems and their properties – Discrete-time systems described by difference equations	3	2
3	Frequency Analysis of Signals and Systems Review of z-Transform - Discrete-time Fourier series - Discrete time Fourier transform (DTFT) - Gibbs phenomenon - Frequency response of LTI systems - Relationship of DTFT and z-transform - Response of LTI systems to sinusoidal signals		2
	The Discrete Fourier Transform		
4	Fourier representation of finite duration sequences – Frequency domain sampling – Properties of DFT – Circular convolution – Implementing LTI systems using DFT (zero padding and overlap-save) – Fast Fourier Transform – Implementation on GP DSP processors	4	2
	Sampling and Reconstruction		
5	Sampling - Spectrum representation - Shannon sampling theorem – Aliasing - Up-sampling and down-sampling – Introduction to audio	3	2
	codecs		
	FIR and IIR Filter Design		
6		6	3
6	FIR and IIR Filter Design Linear Phase Filter, ideal frequency selective filters, Windowing design technique for FIR Filter. IIR Filters - Butterworth filters, analog filter transformations, IIR filter design by impulse invariance, bilinear	6	3
7	FIR and IIR Filter Design Linear Phase Filter, ideal frequency selective filters, Windowing design technique for FIR Filter. IIR Filters - Butterworth filters, analog filter transformations, IIR filter design by impulse invariance, bilinear transformation – Digital signal processor implementation of these filters	3	1

Learning Outcomes

Upon completion of the course, the student should be able to:

- a. Understand the need for specialized hardware for DSP
- b. Discuss characteristics of discrete-time signals and systems
- c. Compute and analyse signal spectra using DTFT and DFT.
- d. Determine sampling rate requirements and interpret effects of aliasing.
- e. Design Finite Impulse Response (FIR) and IIR (Infinite Impulse Response) filters.
- f. Implement FIR and IIR digital filters using different structures such as direct and cascade forms in DSP processors.

Assessment

- a. Final 2-hour written examination (70 %)
- b. Coursework assessment comprising quizzes, laboratory reports, etc. (30%)

Textbooks and Expected Reading

- a. "Discrete-time Signal Processing", Alan V. Oppenheim and Ronald W. Schafer, Pearson, Third edition, 2010.
- b. "Digital Signal Processing", John G. Proakis and Dimitris G. Manolakis, Pearson, Fourth edition, 2007.
- c. "Digital Signal Processors: Architectures, Implementations, and Applications", S.Kuo and W.S.Gan, Pearson, 2009

Reference Material

- a. "Digital Signal Processing: A Computer-Based Approach", Sanjit K. Mitra, McGraw-Hill International Edition, 2011.
- b. "Digital Signal Processing implementations: using DSP microprocessors with examples from TMS320C54xx", by Avtar Singh and S. Srinivasan, Thomson/Brooks/Cole, 2004.

Book List

References:

- 1. Proakis, J. G., and D. G. Manolakis. <u>Digital Signal Processing: Principles, Algorithms, and Applications</u>, Third Edition. Prentice-Hall International, 1996.
- 2. Oppenheim, A. V., and R. W. Schafer. <u>Discrete-Time Signal Processing, Second Edition</u>. Prentice Hall International, 1999.
- 3. Orfanidis, S.J. Introduction to Signal Processing. Prentice-Hall, 1996.
- 4. Mitra, S.K. <u>Digital Signal Processing: A Computer-Based Approach</u>, Second Edition. McGraw-Hill International, 2001.