The University Of Sheffield.

Electronic & Electrical Engineering.

EEE6440 ADVANCED DIGITAL SIGNAL PROCESSING

Credits: 10

Course Description including Aims

This unit focuses on introducing advanced signal processing methods and technologies and their applications. It aims to develop the concepts of signal transforms, random signals, signal analysis and filter design into scenarios where sampling rate conversions, filter banks and adaptive filtering is required. It also aims to provide practical experience of using software tools in designing and implementing simple algorithms. The aims are:

1. Provide an understanding of advanced filter design concepts.
2. Extend the filter design into scenarios where sampling rate conversions, filter bank and adaptive filtering are required.
3. Develop the concept of transforms.
4. Develop the concept of random signals and their analysis.
5. Develop the concept of adaptive filtering and their applications.

Outline Syllabus

Multi-rate signal processing, filter bank theory, signal transforms, random signal analysis and adaptive filtering theory. The coursework component of this unit aims to provide an understanding of using software tools, such as MATLAB, in solving problems and implementing simple signal processing algorithms.

Time Allocation

16 lectures, 8 laboratory classes, 4 problem classes and 70 hours of independent study.

Recommended Previous Courses

Entry requirements.

Assessment

2 hours formal examination

Recommended Books

1. Digital Signal Processing : Concepts and Applications -- Mulgrew, Grant and Thompson.
2. Digital Signal Processing - J. Proakis & D. Manalokis (Prentice Hall).
3. Wavelets & Subband coding -- M. Vetterli & J. Kovacavic. (available online at http://www.waveletsandsubbandcoding.org/)
4. Discrete-time Signal Processing – A. Oppenheim and R. Schafer

Objectives

By the end of the unit, a candidate will be able to demonstrate the ability to:

1. Carry out filter design and implementation for sampling rate conversions including decimation (d), interpolation (i), and a rational factor (i/d).
2. Understand the polyphase representations of filter banks, formulate different filter bank design and provide the corresponding solutions.
3. Perform simple analysis and compute statistics of random signals.
4. Understand the Wiener filter solution and the Least Mean Square type adaptive algorithms and apply them to solve adaptive filtering problems.
5. Design, implement and use simple signal transforms in various applications.
6. Use MATLAB in designing and implementing the above concepts and using them in suitable applications.