**EEE 510 Multimedia Signal Processing**

**Pre-requisites** DSP (EEE 407 or EEE404) OR MATLAB FOR DSP (EEE 509 or EEE 598) (or equivalent at other university - email instructor). Requires: Basic MATLAB programming skills
**Instructor:** A. Spanias
**Teaching Assistant/Grader:** TBD
 **Book:**
Audio Signal Processing and Coding Text Book, Andreas Spanias, Ted Painter, Venkatraman Atti ISBN: 0-471-79147-4, Hardcover, 544 pages, WILEY INTERSCIENCE, March 2007 . Several research papers on speech, audio and video coding

**Optional Books for background:**

1. Digital Signal Processing, A Computer-Based Approach, Sanjit K. Mitra, 2011 4th Edition, McGraw-Hill

2. Discrete-time Signal Processing, Oppenheim and Schafer, 3rd Edition , 2009, Prentice Hall (Pearson)

3. Andreas Spanias, Digital Signal Processing; An Interactive Approach, 2nd Ed. Textbook with comprehensive theory, problems, and JAVA computer exercises, ISBN 978-1-4675-9892-7, May 2014.

**Registration** only through the GOEE (requires GOEE on-line registration). **INTERNET ONLY** (special fees apply)

**Description:** This course is based on a series of streaming audio/video lectures with power-point slides. The course consists of three modules. The first module reviews the signals processing essentials needed to understand the speech and audio coding algorithms. The module continues with an introduction to speech processing and gives a historical perspective of the algorithms and standards. The second module deals with waveform quantization and describes PCM/QADPCM and sub-band/transform coding algorithms. The second module also includes an introduction to open and close loop LPC coders. The third module reviews concepts audio processing and describes the MPEG audio algorithms. Some information on MPEG video and JPEG algorithms s also given.

**Contents:** The course examines speech coding algorithms including those embedded in telephony standards as well as those that are part of teleconferencing standards. In particular, the following are examined: the GSM and PCS speech coding standards, code excited linear prediction, Algebraic CELP (ACELP), ITU G.728 low-delay CELP coder, the ACELP ITU G.729, CDMA EVRC (IS-127), and True Speech G723.1 and other standards. Reference is also made to the Sinusoidal Analysis Synthesis, the IMBE IRIDIUM algorithm, and Voice-Over-IP algorithms. Audio coding topics include introduction to perceptual coding, the MPEG, MP3, SDDS, and AC-3 algorithms, as well as algorithms embedded in recent streaming audio standards. In addition still-image and motion-video compression standards are examined, i.e., image and video formats, color spaces, digital television formats, lossless encoding, huffman and arithmetic coding, DPCM, the JPEG standard, the JPEG 2000, and advanced methods for video compression, motion estimation and compensation. Also covered are MPEG standards: MPEG-1, MPEG-2, MPEG-4, MPEG-7, and ITU-T Video Recommendations (H.261, H.263, H.263+), New Algorithms and Standards in Speech and Image Coding.

 **Course Objectives**To introduce students to multimedia signal processing with emphasis on speech and audio coding algorithms
To familiarize students with the various international standards on speech and audio coding.

**Course Outcomes**Students understand the basics of speech and audio processing
Students understand the theory of linear prediction
Students understand the basics of transform coding
Students understand how transform coding is used in speech, audio, and video coding
Students can develop and implement simple algorithms for speech/audio coding

**Grading:**Tests: 2 tests (take home) 25% each
Projects: 2 programming projects 20% each
Homework: Weekly 10%