Course Syllabus

Course Code	Course Title	ECTS Credits
ECE-432	Speech Processing	6
Prerequisites	**Department**	**Semester**
ECE-332, ECE-431	Engineering	Fall or Spring
Type of Course	**Field**	**Language of Instruction**
Elective	Engineering	English
Level of Course	**Lecturer(s)**	**Year of Study**
1st Cycle	Dr George Gregoriou	4th
Mode of Delivery	**Work Placement**	**Corequisites**
Face-to-face	N/A	None

Course Objectives:

The main objectives of the course are to:

- Introduce students to the fundamentals of speech signal processing and its related applications.
- Provide students with the understanding of the various aspects of human language technology and the various signal processing tools available for exploitation.
- Acquire the fundamentals of the digital signal processing that allows them to assimilate the concepts related to the speech processing.
- Acquire the basic knowledge on the production and perception of speech that allows them to understand the techniques of speech signal analysis and the models applied in the different applications related to speech technology.
- Acquire the basic knowledge on the analysis of the voice signal and their applications.
- Identify, formulate and solve problems of digital speech processing in a multidisciplinary environment.
- Acquire knowledge in the use of tools for the development of applications in the scope of the digital speech processing (MATLAB).
- Give students enough understanding to enable the student to pursue further study and/or research and development, including independent reading and contributions in the field.

Learning Outcomes:

After completion of the course students are expected to be able to:

- Demonstrate knowledge and understanding of speech signal processing tools and techniques.
- Analyze a speech signal in terms of its frequency content.
- Extract certain acoustic features from a speech signal.
- Discuss the basics of human speech production and auditory mechanisms.
- Implement a simple speech synthesizer.
- Assess different speech processing algorithms based on their outcomes during practical application.
- Apply native digital signal processing methodologies to solve speech processing problems.
- Demonstrate ability and basic control of the use of the computer and the software of practices (MATLAB), as well as of the different modules and functions used in the different practices.
- Demonstrate ability to apply this knowledge to the practice.

Course Content:

- Speech production model (source-system model).
- Speech perception, classes of speech sounds (consonants, vowels, etc.), spectral characteristics of consonants and vowels, formants.
- Speech analysis technique: pitch detection, endpoint detection, voiced/unvoiced detection.
- Speech synthesis technique: formant-based speech synthesizers (e.g., KLATT synthesizer), articulatory speech synthesizers.
- Speech recognition: feature extraction algorithms (e.g., mel-frequency cepstrum coefficients), dynamic-time warping, Hidden-Markov models.
- Speech enhancement: spectral subtraction methods, Wiener filtering.
- Speech compression: ADPCM, Linear-predictive coders, analysis-by-synthesis.

Learning Activities and Teaching Methods:

Lectures, in-class examples and exercises, projects.

Assessment Methods:

Homework, projects, mid-term exam, final exam.

Required Textbooks / Readings:

Title	Author(s)	Publisher	Year	ISBN
Discrete-Time Speech Signal Processing: Principles and Practice	T. Quatieri	Prentice Hall	2002	13242942X

Recommended Textbooks / Readings:

Title	Author(s)	Publisher	Year	ISBN
Discrete-Time Processing of Speech Signals	J. Deller, J. Hansen, J. Proakis	Wiley-IEEE Press	2000	9780780353862
Digital Processing of Speech Signals	L. Rabiner, R. Shafer	Prentice Hall	1979	0132136031