Course guide
230622 - DSAP - Digital Audio and Speech Processing

Last modified: 11/04/2025

Unit in charge: Barcelona School of Telecommunications Engineering
Teaching unit: 739 - TSC - Department of Signal Theory and Communications.

Degree: MASTER'S DEGREE IN TELECOMMUNICATIONS ENGINEERING (Syllabus 2013). (Optional subject).
MASTER'S DEGREE IN ADVANCED TELECOMMUNICATION TECHNOLOGIES (Syllabus 2019). (Optional subject).

Academic year: 2025 **ECTS Credits:** 5.0 **Languages:** English

LECTURER

Coordinating lecturer: MARTA TOLOS RIGUEIRO

Others:

PRIOR SKILLS

Have completed at least two subjects of the area Signals, Systems, and Transforms
Have completed at least one subject of the area Probability, Random variables and Stochastic processes
Experience with Matlab programming
Recommended:
- Basic knowledge of Machine Learning
- Python language

DEGREE COMPETENCES TO WHICH THE SUBJECT CONTRIBUTES

Specific:
1. Ability to apply information theory methods, adaptive modulation and channel coding, as well as advanced techniques of digital signal processing to communication and audiovisual systems.

Transversal:
2. TEAMWORK: Being able to work in an interdisciplinary team, whether as a member or as a leader, with the aim of contributing to projects pragmatically and responsibly and making commitments in view of the resources that are available.

3. EFFECTIVE USE OF INFORMATION RESOURCES: Managing the acquisition, structuring, analysis and display of data and information in the chosen area of specialisation and critically assessing the results obtained.

4. FOREIGN LANGUAGE: Achieving a level of spoken and written proficiency in a foreign language, preferably English, that meets the needs of the profession and the labour market.

TEACHING METHODOLOGY

- Lectures (50%)
- Application classes (with Matlab or similar) (50%)
- Team work: course project and others
- Individual work: homework assignments, related to the applications

LEARNING OBJECTIVES OF THE SUBJECT

Learning objectives of the subject
Understanding and being competent on a relevant set of concepts and techniques in the field of digital audio processing, and their application to problems arising from real applications. Signals and applications related to speech and music will be particularly considered.

Learning results:
Ability to digitally process audio signals, in an application-oriented context, in order to analyze, model, extract information, clean, modify, and generate/synthesize them.

STUDY LOAD

Type	Hours	Percentage
Self study	86,0	68.80
Hours large group	39,0	31.20

Total learning time: 125 h

CONTENTS

Introduction

Description:
Course presentation
Audio diversity
Characteristics of speech and music. Production model
Hearing and auditory modeling
Short-time Fourier transform (STFT) and spectrogram
The short-time Fourier transform

Full-or-part-time: 12h
Theory classes: 6h
Self study : 6h

Modeling and representation of audio signals

Description:
Production-based all-pole modeling
Pitch determination for speech and music
LPC-based coding used in mobile telephony

Full-or-part-time: 8h
Theory classes: 4h
Self study : 4h

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Enhancement of voice and audio signals

Description:
Denoising: spectral subtraction, Wiener-based filtering, neural nets (deep learning)
Blind source separation: NMF
Cancellation: echo, interference

Full-or-part-time: 16h
Theory classes: 8h
Self study : 8h

Recognition and detection of audio and speech

Description:
Pattern-matching dynamic approaches.
Statistical and deep learning approaches
Approaches based in dynamic pattern matching
Statistical and deep learning approaches
Audio activity detection
Application to speech and sound recognition

Full-or-part-time: 16h
Theory classes: 8h
Self study : 8h

Multi-microphone audio processing

Description:
Room acoustics
Array beamforming
Acoustic source localization and tracking

Specific objectives:

Full-or-part-time: 12h
Theory classes: 6h
Self study : 6h

Analysis and synthesis of audio signals

Description:
Short-term analysis-synthesis of (cuasi)periodic signals. Time-scale and pitch modification
Spatial audio synthesis with HRTF functions

Full-or-part-time: 8h
Theory classes: 4h
Self study : 4h

Project realization and presentation

Description:
Design, implementation and test of a audio processing system for a specific application
Oral presentation of 1) project proposal, 2) critical review, and 3) project realization and conclusions
Written report, conference paper style

Full-or-part-time: 48h
Theory classes: 4h
Self study : 44h

ACTIVITIES

Assignments and short tests

Full-or-part-time: 25h
Self study: 25h

Course project

Description:
Team project realization, which includes audio processing experimental work, and is presented both orally and in writing.

Full-or-part-time: 60h
Self study: 60h

Oral presentations

Description:
Oral presentation of the course project in three times: proposal, review, and final presentation. Presentation of minor works

Full-or-part-time: 1h
Self study: 1h

GRADING SYSTEM

Attendance/participation in class (10%)
Assignments, small projects, and tests (30%)
Course project (60%)

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BIBLIOGRAPHY

Basic:

- Quatieri, T.F. Discrete-time speech signal processing: principles and practice. New Delhi: Prentice Hall, 2006. ISBN 9788177587463.
- Gold, B.; Morgan, N.; Ellis, D. Speech and audio signal processing: processing and perception of speech and music [on line]. 2nd ed. Wiley-Blackwell, 2011 [Consultation: 30/06/2022]. Available on: https://onlinelibrary-wiley-com.recursos.biblioteca.upc.edu/doi/book/10.1002/9781118142882. ISBN 978-0-470-19536-9.
- Dutoit, T.; Marqués, F. Applied signal processing: a MATLAB-based proof of concept [on line]. New York ; London: Springer, 2009 [Consultation: 30/06/2022]. Available on: https://link-springer-com.recursos.biblioteca.upc.edu/book/10.1007/978-0-387-74535-0. ISBN 9780387745343.
- Loizou, P.C. Speech enhancement: theory and practice. 2nd ed. Boca Raton: CRC Press, 2013. ISBN 9781466504219.

Complementary:

- Rabiner, L.R.; Schafer, R.W. Theory and applications of digital speech processing. London: Pearson, 2010. ISBN 9780137050857.
- Huang, Y.A.; Benesty, J. (eds.). Audio signal processing for next-generation multimedia communication systems [on line]. New York: Kluwer Academic Publishing, 2004 [Consultation: 30/06/2022]. Available on: https://link-springer-com.recursos.biblioteca.upc.edu/book/10.1007/b117685. ISBN 1402077688.

RESOURCES

Audiovisual material:

- Slides. Slides used in lectures

Computer material:

- Codi programes. Software codes in Matlab or similar

Other resources:

Lecture slides

Practical work statements and programs