COURSE DESCRIPTION CARD - SYLLABUS

Course name
Audio and speech systems [S1MiKC2>SDiM]

Course

Field of study
Microelectronics and Digital Communication

Year/Semester
3/5

Area of study (specialization)
–

Profile of study
general academic

Level of study
first-cycle

Course offered in
Polish

Form of study
full-time

Requirements
elective

Number of hours

Lecture
15

Laboratory classes
15

Other
0

Tutorials
0

Projects/seminars
0

Number of credit points

2,00

Coordinators

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Lecturers

Prerequisites

Has basic knowledge of sound acquisition, processing, compression, transmission and presentation.

Course objective

The course introduces the latest technologies in the field of multimedia sound and speech systems, in particular with issues related to transmission (including streaming) and data storage, presentation and its processing. Learning and practical use of methods for assessing the quality of the audio signal and analysis of signal distortions introduced by compression techniques.

Course-related learning outcomes

Knowledge:
K1_W02 Has structured knowledge of the theory of analog and digital signals, methods of analysis of linear and nonlinear systems and signal processing. Knows the properties and characteristics of electronic components and basic methods of designing and analyzing electronic systems, including analog and digital systems used in ICT.
K1_W10 Has knowledge of methods of acquisition, processing, compression and transmission of image, sound and speech signals, including in the context of optimization of streaming systems and applications

in virtual reality (VR) and augmented reality (AR).

Skills:
K1_U03 Is able to use mathematical tools, including mathematical analysis, algebra and probability theory, to solve problems in the ICT area, in particular in signal analysis and processing.
K1_U08 Is able to determine the parameters and properties of telecommunication signals and systems in various transmission conditions.

Social competences:
K1_K02 Is aware of the need for a professional approach to solving technical problems and taking responsibility for the technical solutions they propose.
K1_K05 Is able to formulate opinions on the basic challenges facing modern electronics and telecommunications. Is aware of the impact of telecommunications and ICT systems and networks on the formation of the information society.

Methods for verifying learning outcomes and assessment criteria

Learning outcomes presented above are verified as follows:

Lecture: written or oral exam, open-ended questions, with expected descriptive answers. Passing threshold: 50% of possible points. List of required content is made available during lectures.
Laboratory: reports from thematically uniform blocks of laboratory exercises. Passing threshold: 50% of the possible points for each report.
Grading scale: 2.0: <0%,50%), 3.0: <50%, 60%), 3.5 <60%, 70%), 4.0 <70%, 80%), 4.5 <80%, 90%), 5.0 <90%, 100%).

Programme content

Lecture:
1. Sound and its perception by humans.
2. Digital signal representations in multimedia systems of sound and speech.
3. Modeling and statistical analysis of the audio signal.
4. Improving the quality of a disturbed and distorted signal.
5. Coding of speech and wideband audio signals.
6. Synthesis of musical sounds and speech.
7. The music market in the age of artificial intelligence.
Additional content may also be presented in lectures by invited guests from industry or other scientific institutions.
Laboratories:
1. The student designs, implements and tests programs that implement selected elements of digital audio systems associated with the content presented in lectures.
2. The student becomes familiar with tools and devices enabling the acquisition and presentation of audio content.

Course topics

• Physical foundations of sound and auditory perception.
• Digital representation of sound signals.
• Spectral and time-frequency analysis of sound.
• Mathematical models of audio signals (AR, MA, ARMA, sinusoidal model).
• Analysis of information in the audio signal (entropy, correlation, PCA).
• Recognition of patterns and sequences of sound events.
• Classification and removal of interference in the audio signal.
• Adaptive filtering in sound processing.
• Methods of reducing distortions and masking signal damage.
• Basic concepts and tools of audio compression.
• Lossy speech coding techniques (ADPCM, LPC, CELP, etc.).
• Speech coding standards in telecommunications (G.721, G.729, etc.).
• Basics of psychoacoustic coding.
• Lossy audio coding techniques (MPEG, AAC, Dolby AC-3, etc.).
• Spectral extension techniques and parametric audio coding.

• Lossless compression of audio signals (MPEG ALS, MPEG SLS).
• History and methods of musical sound synthesis.
• Speech synthesis.
• General scheme of speech recognition systems.
• Analysis of speech signal features and recognition methods.

Teaching methods

Lecture: multimedia presentation conducted in hybrid form. The possibility of guest lectures conducted by external experts is assumed.
Laboratories: implementation of projects on computers (independently or in groups of several people), possible to implement in remote form (except for exercises using devices for sound acquisition and presentation).

Bibliography

Basic:
- Andrzej Czyżewski, "Dźwięk Cyfrowy," Akademicka Oficyna Wydawnicza EXIT, 1998.
- A. Spanias, T. Painter, V. Atti, "Audio Signal Processing and Coding," Wiley, 2007
- S. Trzciński, "Zarażeni dźwiękiem, rynek muzyczny w czasach sztucznej inteligencji," Wydawnictwo Naukowe PWN, 2023.

Additional:
Madisetti V. (ed)," Video, Speech, and Audio Signal Processing and Associated Standards (The Digital Signal Processing Handbook, Second Edition)," CRC Press, 2009.

Breakdown of average student's workload

	Hours	ECTS
Total workload	60	2,00
Classes requiring direct contact with the teacher	30	1,00
Student's own work (literature studies, preparation for laboratory classes/ tutorials, preparation for tests/exam, project preparation)	30	1,00