STUDY MODULE DESCRIPTION FORM

Name of the module/subject **Audio Compression and Processing**		Code **1010802121010842898**
Field of study **Electronics and Telecommunications**	Profile of study (general academic, practical) **general academic**	Year /Semester **1 / 2**
Elective path/specialty **Information and Communication**	Subject offered in: **English**	Course (compulsory, elective) **elective**
Cycle of study: **Second-cycle studies**	Form of study (full-time,part-time) **full-time**	
No. of hours Lecture: **2** Classes: **-** Laboratory: **2** Project/seminars: **-**		No. of credits **5**
Status of the course in the study program (Basic, major, other) **major**	(university-wide, from another field) **from field**	
Education areas and fields of science and art **technical sciences** **Technical sciences**		ECTS distribution (number and %) **5 100%** **5 100%**

Responsible for subject / lecturer:

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Prerequisites in terms of knowledge, skills and social competencies:

1	**Knowledge**	K1_W06, K1_W11, K1_W19
2	**Skills**	K1_U01, K1_U07, K1_U10, K1_U13
3	**Social competencies**	K1_K01, K1_K02

Assumptions and objectives of the course:

Extending the knowledge on human auditory perception, understanding the underlying physics, and the limitations thereof. A deeper insight into most important digital processing techniques for speech and audio, and their applications in current telecommunications.

Acquiring deep knowledge on data compression techniques for speech and audio. Introduction to audio signal enhancement techniques. Introduction to speech recognition and synthesis, as well as musical sound synthesis.

Gaining the ability to select an appropriate data compression technique for speech and audio in wireless telephony, packet data transmission systems, DVB, internet TV, home cinema, and professional systems. Understanding the limitations of speech recognition and synthesis techniques as well as their possible applications in telecommunication systems and electronic services.

Study outcomes and reference to the educational results for a field of study

Knowledge:

1. Deep knowledge on construction and principles of audio parts in telecommunication systems - [K2_W01]

2. Structured and theoretically underpinned knowledge on compression techniques for speech and wideband audio, including applications of perceptual compression in communications via data transport networks - [K2_W14]

Skills:

1. Ability to analyze the operation of a multimedia system for its audio transmission path, as well as its limitations, and to gain the maximum use of the data transmission conditions offered - [K_U03]

2. Ability to implement the audio path in VoIP and programmable radio, as well as to take into account the peculiarities of audio representation and perception, and how they impact the requirements of such systems. Ability to select appropriate tools for enhancement of an audio signal distorted or interfered by noise - [K_U13]

Social competencies:

1. Ability of self-learning (textbooks, computer programs) - [K2_K04]
2. Understanding the non-technical factors of engineering - [K2_K05]
3. Knowing the responsibility for the electronic and telecommunication systems being designed - [K2_K06]

Assessment methods of study outcomes

1. Written exam
2. Reports from the lab exercises
3. Check of the activity during laboratory exercises

Course description

Human auditory perception (the structure of the organ of hearing, the physical characteristics of the wave vs perceptual attributes of a sound, critical bands, limitations of perception, masking (simultaneous and temporal).

Frequency domain and time-requency representations of audio signals (STFT, the spectrogram, the uncertainty principle, filterbanks, cosine transform, MDCT, linear and quadratic time-frequency distributions, the uncertainty principle in time and frequency, Gabor transform, wavelet transform, subband analysis, filterbanks, cosine transform, MDCT, linear and quadratic time-frequency distributions, Wigner-Ville distributions, cepstrum analysis)

Spectral modeling of audio signals (linear AR, MA, and ARMA models, linear prediction, cepstrum, sinusoidal modeling and sinusoids+noise model)

Speech coding (lossy compression techniques: ADPCM, LPC, CELP, ACELP, RPE, AMR, vector quantization, sinusoidal coding, ITU-T standards and recommendations)

Perceptual coding of wideband audio (the peculiarities of subband and transform coding, fundamentals of perceptual coding, compression schemes of MPEG-1 and 2 layer 1, 2, and 3, MPEG-2 AAC, SBR bandwidth extension, parametric stereo coding, MPEG-4 AAC-HE technique, MPEG-USAC, parametric (model based) audio coding, discussion on MPEG-4 HILN and MPEG-4 SSC).

Lossless and lossy audio coding techniques: modeling of statistics, prediction with integer arithmetics, MPEG-4 ALS and SLS compression standards.

Audio signal enhancement for distorted and noised signals (adaptive filtering, prediction-based smoothing, LMS and RLS filters, narrowband and wideband noise supression, spectral subtraction, nonlinear filters, masking and mitigation of gaps, reconstruction of missing frames in MPEG data)

Basic bibliography:

1. J. Watkinson, The Art of Digital Audio, Focal Press, 2001
2. Audio Signal Processing and Coding, A. Spanias, T. Painter, V. Atti, Wiley, 2007
3. DAFX, Digital Audio Effects, Udo Zoelzer (red.), Wiley, 2002

Additional bibliography:

Result of average student's workload

Activity	Time (working hours)
1. Lecturers and laboratories	60
2. Preparation for laboratories	25
3. Consultations	2
4. Preparation to the exam	35
5. Exam	3

Student’s workload

Source of workload	hours	ECTS
Total workload	125	5
Contact hours	65	3
Practical activities	60	2