SYLLABUS

1. Study Program

1.1	Higher Education Institute	Technical University of Cluj-Napoca
1.2	Faculty	Electronics, Telecommunications and Information Technology
1.3	Department	Communications
1.4	Study domain	Electronics and Telecommunications Engineering
1.5	Study level	Master
1.6	Study program/ Qualification	Multimedia Technologies/ Telecommunications/ Master
1.7	Type of education	IF (Full-time learning)
1.8	Discipline code	TM-E10.00/ TC-E11.40

2. Discipline

2.1	Discipline name					Speech Coding Techniques					
2.2	Subject area					Electronics and Telecommunications Engineering					
2.3	Responsible					Professor Mircea Giurgiu, Ph.D. Mircea.Giurgiu@com.utcluj.ro					
2.4	Titular					Professor Mircea Giurgiu, Ph.D.					
2.5	Year of study	I	2.6	Semester	2	2.7	Evaluation	Exam	2.8	Type of discipline	DA/DI

3. Total estimated time

Year/ Sem	Discipline name	No. of weeks	Course	Applications			Course	Applications			Indiv. study	TOTAL	ECTS
			[hours/week]				[hours/week]						
			C	S	L	P		S	L	P			
I / 2	Speech Coding Techniques	14	2	0	1	0	28	0	14	0	58	100	4

3.1	Number of hours per week	3	3.2	course	2	3.3	applications	1
3.4	Total hours per curriculum	42	3.5	course	28	3.6	applications	14

Individual study	58
Study based on manuals, course materials, references and notes	14
Supplementary documentation in libraries, electronic platforms and on field	10
Preparation of seminars/laboratories, homeworks, essays, portfolios	10
Tutorial work	7
Assesments	3
Other activities	14

3.7	Total hours of individual study	58
3.8	Total hours per semester	100
3.9	ECTS	4

4. Prerequisites (if necessary)

4.1	Curriculum	Digital Signal Processing, Information Theory
4.2	Competences	Programming skills

1. Requisites (if necessary)

5.1	Course	Lecture room with video-projector
5.2	Applications	LAN in the lab room with Internet connection, Matlab environment, speech databases, Audacity tools, software libraries for speech coding.

6 Specific competences acquired

<table>
<tr><td rowspan="3">Professional competences</td><td>Theoretical knowledge (What do the student should know)</td><td>The students will know:
• the organisation and the particularities of the standards applied for speech compression
• the concepts and the methods for speech source modelling
• advanced methods for speech compression
• the low bit rate speech coding
• the concepts and methods for speech denoising and speech enhancement using adaptive filters
• new trends in speech compression using deep neural networks
• the multimedia applications that use speech compression</td></tr>
<tr><td>Acquired skills (What the student is able to do)</td><td>The students will be able:
• to identify and to apply the most suitable speech compression technique according to the requirements of the multimedia application
• to implement a range of speech compression algorithms and to evaluate their objective and subjective performance
• to adopt suitable methods for error control in the transmission of speech
• to design and to implement from the scratch all the modules required by speech compression application
• to use specific tools for the evaluation of the speech quality in a multimedia transmission system</td></tr>
<tr><td>Acquired abilities (what equipment/ instruments/ softwares the student is able to handle)</td><td>The students will be able to use:
• a range of software tools that implement various standards for speech compression and to evaluate their performance
• a range of Python libraries and toolkits applied in neural vocoders
• standard applications and software libraries that allow to embed speech compression in a multimedia transmssion systems (eg. on dedicated channels, on Internet)</td></tr>
<tr><td colspan="2">Transversal competences</td><td>CT3. Capacity to adapt to new multimedia technologies and to develop transdisciplinary competences of analysis/synthesis and optimization. Flexibility in thinking and ability for team work in a transdiciplinary area.</td></tr>
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7 Discipline objectives (based on the grid of specific competences acquired)

7.1	General objective	To develop knowledge, skills and abilities on the practical use of the methods and the algorithms used for speech coding.
7.2	Specific objectives	• to know the main characteristics of various speech coding standards • to handle specific software tools for speech coding • to know the concepts and the methods applied for temporal, spectral or parametric speech coding • to be able to design and to implement specific algorithms in a software application used for speech coding (source modelling, vocal tract modelling, parameter encoding, neural vocoders, error control) • to be skilled in the use and the application of speech coding methods employing dedicated software libraries.

8. Contents

<table>
<tr><td colspan="2">8.1. Course (titles)</td><td>Teaching methods</td><td>Obser-vations</td></tr>
<tr><td>1</td><td>Introduction into speech coding and compression algorithms. Standards (G.721, G.722, G.723, MPEG, FS-1015, FS-1016, etc) and taxonomy.</td><td rowspan="2"></td><td rowspan="2"></td></tr>
<tr><td>2</td><td>Parametric speech modelling. Quantization of LPC and LSF</td></tr>
</table>

	parameters (VQ, SIVP, SAVQ).	PPT presentation, practical demos, interactive discussions and debates, problem solving.	NA
3	Analysis by synthesis speech coding. MPE, RPE-LTP, CELP, VSELP speech compression techniques.		
4	Low bit rate and fast algorithms for speech compression: CELP, LD-CELP. VoIP and GSM applications.		
5	Sinusoidal speech coding.		
6	MBE and MELP speech coding.		
7	MPEG standard for audio and speech coding		
8	Speech compression using the Wavelet Transform. Entropic coding.		
9	Speech compression using Vector Quantization (LBG, SELBG, HVSQ)		
10	Artificial neural networks used in speech coding and compression		
11	Neural vocoders (WaveNet, WaveGlow, LPCNet, FFTNet)		
12	Error control on digital channels using speech coding. Coding optimization		
13	Echo cancelling and noisy speech enhancement using adaptive filtering		
14	Synthesis of the course		
8.2. Applications (laboratory)		Teaching methods	Obser-vations
1	Implementation and evaluation of the G.721 ADPCM speech encoder	Experiments, problem-based and project-based learning.	NA
2	LPC speech coding using the FS-1016 standard		
3	Evaluation of performances of the CELP encoder		
4	Sinusoidal speech coding		
5	Psychoacoustic speech coding in MPEG (MP3) standard		
6	Speech compression using the Wavelet Transform		
7	Evaluation of the VQ speech compression		

References:

1. Tom Backstrom, “Speech coding”, Springer, 2017.
2. R. Togneri, T. Ogunfemni, “Speech audio processing for coding, enhancement and recognition”, Springer, 2014.
3. H. Doddale, V. Ramsubramanian, “Ultra low bit rate speech coding”, Springer 2014.
4. M. Narasimha, T. Ogunfemni, “Principles of speech coding”, Wiley Publ., 2010.
5. Wai C Chu, “Speech Coding Algorithms: Foundation and Evolution of Standardized Coders”, Wiley, 2003
6. Noah Berhanu, “Speech coding using Code Excited Linera Preiction”, Wiley Publ., 2009
7. T. Quatrieri, "Discrete-Time Speech Signal Processing: Principles and Practice", Prentice Hall, 2001.
8. D. Childers, "Speech Processing and Synthesis Toolboxes", John Wiley Publ., 2000
9. A. M. Kondoz, "Digital Speech: Coding for Low Bit Rate Communication Systems", Wiley Publ., 2004
10. M. Tatham, "Developments in Speech Synthesis", Wiley Publ., 2005.
11. R. Duboite, M. Kunt, „Traitement de la parole”, Presses Politechnique Universitaire Romande, Lausanne, 1990.
12. M. Giurgiu, „Compresia Datelor Audio pentru Aplicatii Multimedia”, Ed. Risoprint, 2003.

9. Bridging course contents with the expectations of the representatives of the community, professional associations and employers in the field.

The subject is oriented towards the development of practical applications involving speech coding for telecommunications and multimedia applications. The contents are aligned with the current trends in the filed and with the requirements of the IT industry. They meet the expectations of important local software development companies to implement speech compression modules embedded in multimedia systems.

10. Assessment

Type of activity	10.1 Evaluation criteria	10.2. Evaluation method	10.3. The weight of the final grade
Course	Student performance and deep of knowledge against the defined learning outcomes	Written examination (knowledge and problem solving skills), intermediary and final.	50%

Applications	Performance in accuracy and originality of problem solving, experiment running and presentation of results.	Running the experiment, solving the problems, intermediary evaluation, individual work, laboratory reports	50%
10.4 Minimum performance standard			
To know the speech coding methods and to implement algorithms for speech coding in time, frequency or parametric domains by handling specific software tools.			

Date
01.07.2020

Titular
Professor
Mircea GIURGIU, Ph.D.

Responsible
Professor
Mircea GIURGIU, Ph.D.

Date of approval
01.10.2020

Head of Department
Professor Virgil DOBROTA, Ph.D
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