INDIAN INSTITUTE OF TECHNOLOGY ROORKEE

NAME OF DEPT. /CENTRE: **Electronics and Computer Engineering**

1. Subject Code**: EC – 518N** Course Title**: Speech and Audio Processing**

2. Contact Hours: **L: 3** **T: 0** **P: 0**

3. Examination Duration (Hrs.): **Theory** **0 3** **Practical** **0 0**

4. Relative Weight**:** **CWS** **15** **PRS** **00** **MTE** **35** **ETE** **50** **PRE** **00**

5. Credits**:** **0 3** 6. Semester

Autumn	**Spring**	**Both**
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7. Pre-requisite**: EC - 311 and EC - 411 or equivalent**

8. Subject Area**: MSC**

9. Objective: To acquaint the students with the concepts in speech and audio processing, and their applications in communication systems.

10. Details of the Course:

Sl. No.	**Contents**	**Contact Hours**
1.	Digital speech processing and its applications, production and classification of speech sounds, lossless tube models, digital models for speech signals; Analysis and synthesis of pole-zero speech models, Levinson recursion, lattice synthesis filter.	7
2.	Time dependent processing of speech, pitch period estimation, frequency domain pitch estimation; Discrete-time short-time Fourier transform and its application, phase vocoder, channel vocoder.	6
3.	Homomorphic speech processing, waveform coders, hybrid coders and vector quantization of speech; Model based coding: Linear predictive, RELP, MELP, CELP; Speech synthesis.	9
4.	Principles of speech recognition, spectral distance measures, dynamic time warping, word recognition using phoneme units, hidden Markov models and word recognition, speech recognition systems, speaker recognition.	7
5.	Ear physiology, psychoacoustics, perception model and auditory system as filter bank; Filter bank design and modified discrete cosine transform algorithm for audio compression in MP3 and AAC coders; Standards for high-fidelity audio coding.	7
6.	Tree-structured filter banks, multicomplementary filter banks; Properties of wavelets and scaling functions, wavelet transform; Filter	6

	banks and wavelets, applications of wavelet signal processing in audio and speech coding.	
	Total	**42**

11. Suggested Books:

Sl. No.	**Name of Books / Authors**	**Year of Publication**
1.	Rabiner, L.R. and Schafer, R.W., “Digital Processing of Speech Signals”, Pearson Education.	2006
2.	Quatieri, T.F., “Discrete-Time Speech Signal Processing: Principles and Practice”, Pearson Education.	2002
3.	Furui, S., “Digital Speech Processing, Synthesis and Recognition”, 2^{nd} Ed., CRC Press.	2000
4.	Fliege, N.J., “Multi Rate Digital Signal Processing”, John Wiley & Sons.	1999
5.	Spanias, A., Painter, T. and Venkatraman, A., “Audio Signal Processing and Coding”, John Wiley & Sons.	2007
6.	Gold, B. and Morgan, N., “Speech and Audio Signal Processing”, John Wiley & Sons.	2002