Course number			
Year	Time of starting a course	Day period	Faculty
2016	Fall	Tue.4	Engineering and Science
Lecture code	Subject name[English]		Number of credits
R0015500	音響信号処理特論		2
Charge teacher name[Roman alphabet mark]			
アシャリフモハマッド			

Course content and methods

音響信号処理特論

Advanced Acoustic Signal Processing : In this lecture we are study about Acoustic Speech Signal Processing. First, the mechanism human speech production and acoustic phonetics such as vowels, the vowel triangle, front, mid, back, diphthongs, semivowels, li ds, glides, consonants, nasals, stops (Voiced, Unvoiced), fricatives (voiced, unvoiced), whisper, afficates are explained. Then, the oustic theory of speech production, sound propagation Portnoff-Sondhi differential equations, uniform lossless tube and formants ffects of losses in the vocal tract, the effect of nasal coupling, excitation of sound in the vocal tract, lossless tube models, wave pr agation in concatenated lossless tubes, relationship to digital filter and digital model for speech signals are explained. Next, time- main methods for speech processing, short-time energy, zero-crossing rate, speech vs. silence discrimination, pitch period estima n, short-time autocorrelation function, short-time average magnitude difference function (AMDF), pitch period estimation using th autocorrelation function, median smoothing are explained. Then, digital representation of the speech waveform, PCM, MPCM, Ada ve quantization, Delta modulation DPCM, ADPCM will be discussed. Next, Homomorphic speech processing, Cepstrum, pitch dete n, formal estimation, Homomorphic vocoder are aimed for studies. Also, Linear Predictive Coding (LPC) of speech, LPC analysis, t autocorrelation method, the covariance method, computation of gain for LPC model, Cholesky decomposition solution of LPC equa ns, Durbin's recursive solution, lattice formulations, the prediction error signal, relation between the various speech parameters, s thesis of speech from LP parameters, pitch and formant analysis using LPC, LPC vocoder are presented. At last, digital speech pro ssing for Man-Machine Communication by voice and special projects in Acoustic Echo and Noise Cancellation , etc. will be studied

Goals and objectives

Digital Processing of Speech Signal :
To know about the mechanism of human speech production and acoustic phonetics.
To know about the acoustic theory of speech production and relationship to digital filter
To know about time-domain methods for speech processing, pitch period estimation.
To know about digital representation of the speech waveform.
To know about Homomorphic speech processing & Linear Predictive Coding, synthesis of speech from LP parameters
To know about Man-Machine Communication by voice, Acoustic Echo and Noise Cancellation by Adaptive Digital Filter Algorithms

Evaluation criteria and evaluation methods

Project & Presentation & Report

Course conditions

Some Digital Signal Processing and Communications knowledge are necessary.

Contents of Class

1Introduction to speech and signal processing
2The mechanism of human speech production.
3Phonems in American English, Vowel triangle, Diphthongs, Liquids, Glides.
4Consonants, Nasals, Stops, Fricatives, Whisper, Afficates.
5Sound propagation Portnoff-Sondhi equations, Lossless tube, Formants, Losses.
6Nasal coupling, Excitation, Digital model for speech signals.

7Energy, Zero Crossing, Silence, Pitch.
8Autocorrelation, AMDF.
9Pitch period estimation, Median smoothing.
10PCM, Adaptive quantization, Delta modulation.
11Homomorphic Speech Processing, Cepstrum, Pitch detection, Formal estimation, The Homomorphic Vocoder
12LPC of speech, Autocorrelation method
13Covariance method, Computation of gain for LPC model.
14Cholesky, Durbin, Lattice formulations
15The Prediction Error Signal, Synthesis of speech from LP parameters, Pitch and formant by LPC, LPC vocoder.
16Man-Machine Communication by voice, Special Projects in Acoustic Echo and Noise Cancellation, etc.

Prior learning

Students should have knowledge about Digital Signal Processing and digital filtering. So they should study the basics theory of DS
Then get some knowledge for some real applications.

Post learning

After each class students should study that part from text book. Later they should make some real application of
Man-Machine Interface such as recognition and speech parameters estimation or making Echo & Noise Cancellation by Adaptive F
ering.

Textbook

Text book	Title	Digital Processing of Speech Signal, L.R. Rabiner/R.W. Schafer, Prentice Hall		ISBN	0132136031	Note	
	Author						
	Publishing company		Publishing year	NCID			

Textbook Remarks

Digital Processing of Speech Signal, L.R. Rabiner/R.W. Schafer, Pre

References

Reference book Remarks

Language

English

Message

This course will give you basic knowledge about Acoustic and Speech Processing. Mostly it is theoretical but gives know-how to d
esearches in this area.

Office Hours

Tue 3:00-5:00 P.M., Fri 3:00-5:00 P.M.

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