Speech Enhancement for Hearing Aids

Overview

Speech enhancement deals with enhancing speech components in degraded speech signals for either facilitating human perception or improving automatic speech recognition. The degradation types that are in the focus of current research efforts include additive background noise, reverberation and speech from other speakers. Since the characteristics of these degradations are different, the approaches for enhancement are also diverse and need to be tailored to the application. Accordingly, we have noise reduction, dereverberation and multi-speaker separation methods in the literature. Most of the existing methods for noise reduction first establish an estimate of the noise power spectrum and then use the same for enhancing salient speech components. However, there are also approaches that perform temporal and/or spatial processing to enhance speech signals degraded by reverberation and other speakers. Furthermore, speech enhancement methods have been categorized into single-channel, dual-channel, and multi-channel methods and have become an important ingredient of any hearing device.

Compared to development of speech enhancement approaches for smart phones or automatic speech recognition systems, **speech enhancement for hearing aid devices** is quite challenging due to the following constraints:

1) Limited signal processing resources that are available in these devices

2) Design of speech enhancement algorithms needs to avoid latency with respect to the direct sound component.

However, recent trends of improved computing power and low memory cost are helping to implement more sophisticated methods for speech enhancement in hearing aids.

This course will give an exposure to key aspects of speech enhancement. The course will provide an **overview of single-microphone and multi-microphone speech enhancement methods** present in the literature for perceptual enhancement requirement of normal-hearing listeners and for automatic speech recognition tasks. Particular issues with respect to **hearing impairments** and also with respect to **current generation hearing aids** will be presented. After this, different approaches proposed **for speech enhancement for hearing aids** will be explained.

The course would be primarily delivered by **Prof. Rainer Martin, from Institute of Communication acoustics, working in the field of speech enhancement holding around 6,000 citations for his works**. We welcome all interested to attend the course.

Objectives

- Overview of speech enhancement technique
- Hearing impairment and technology of state-of-the-art hearing aids
- Single channel and multi-channel speech enhancement for hearing aids

Benefits

- Exposure to single-microphone and multi-microphone speech enhancement methods
- Explanations of issues in current generation hearing aids
- Exploration of different speech enhancement methods for hearing aids
- Hands-on MATLAB sessions.

Course Details

Dates : 23 January 2018 – 27 January 2018

Day 1:

Lecture 1: 9:00 to 10:30 AM	**Speech enhancement for hearing aid**
Lecture 2: 11:00 to 12:30 AM	**Overview of speech enhancement**
Laboratory 1: 2.00 to 5.00 PM	**MATLAB exercises on spectral analysis of speech**

Day 2:

Lecture 3: 9:00 to 13:00 AM	**Temporal processing**
Lecture 4: 11:30 to 12:30 AM	**Single channel noise reduction**
Laboratory 2: 2.00 to 5.00 PM	**MATLAB exercises on speech enhancement by temporal processing**

Day 3:

Lecture 5: 9:00 to 10:30 AM	**Dual channel speech enhancement**
Lecture 6: 11:00 to 12:30 AM	**Hearing impairment and hearing aids**
Laboratory 3: 2.00 to 5.00 PM	**MATLAB exercises on dual channel algorithm for joint noise reduction**

	Day 4: *Lecture 7: 9:00 to 10:30 AM* — **Speech presence probability estimation** *Lecture 8: 11:00 to 12:30 AM* — **Speech enhancement for human-computer interaction** *Laboratory 4: 2.00 to 5.00 PM* — **MATLAB exercises on speech enhancement for human-computer interface** **Day 5:** *Lecture 9: 9:00 to 10:30 AM* — **Source localization and source separation** *Lecture 10: 11:00 to 12:30 AM* — **Research directions in speech enhancement for hearing aids** *Examination 1: 2.00 to 5.00 PM* — **Course examination**
Who can Attend ?	▪ Graduate and Post-graduate students at all level (B.Tech/B.E./M.Tech/M.E./Ph.D) who are interested in Signal processing, speech processing and biomedical aspects of speech processing. ▪ Faculty and/or project staff from academic and technical institutions, and researchers in R&D organizations, interested in and/or working in the field of Signal Processing, and its modern methods.
Fees **Accomodation** **Fooding**	*Students :* **INR 1000 /-** *(Refundable)* *Participants from abroad :* **USD 500 /-** *Industry/ Research Organizations:* **INR 15,000 /-** *Academic Institutions:* **INR 7,500 /-** The fees is to be paid using **Demand Draft**. The Demand Draft for the students (only) will be returned back to them if and when they physically attend the course. Hostel accommodation and lodging necessities will be provided for students, at a

	per day per person cost of *approximately* INR 250 /-. Faculties and industry persons would be provided IITG Guest House rooms (twin sharing / single rooms depending on availability) at a per day per person cost of *approximately* INR 500 /-. Charges may vary. Participants may avail food/meals inside or outside the IITG campus, wherever they prefer, on their own expenses.
Course Co-ordinators	**Department of Electronics and Electrical Engineering,** **IIT Guwahati, Guwahati – 781039, India** **Prof. S. R. Mahadeva Prasanna - prasanna@iitg.ernet.in** Tel: +919954008138 **Prof. S. Dandapat - samaren@iitg.ernet.in** Tel: +91-361-258-2505
Registration Procedure	**Please follow the following steps for registration :** **1.** Go to GIAN website (**http://www.gian.iitkgp.ac.in/GREGN/index**) First time users need to register and pay a one-time fee of **INR 500 /-**. **2.** Select course: **Speech enhancement for hearing aids**. Once you enroll for the course, an **Enrolment/Application** number will be generated, and the course coordinators will be notified. The course coordinators will **shortlist** the candidates out of the applicants. The shortlisted candidates **will be notified by email**. **3.** The **selected** candidates must pay the applicable fees using **Demand Draft (DD)** drawn in favour of **"Registrar, INDIAN INSTITUTE OF TECHNOLOGY GUWAHATI", payable at IIT Guwahati.**

Please write your Name and Enrolment/Application number at the back of the DD, and **post/courier** it, to reach **by 15 January 2018 23:59 hrs IST**.

Address :

Department of Electronics and Electrical Engineering,
Indian Institute of Technology Guwahati,
Guwahati, Assam, India, PIN - 781 039,
Email: **eeeoff@iitg.ernet.in**
Telehone: +91-361-2582550
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4. Fill the **course registration form** in GIAN portal.
Email the **course registration form** to the Course coordinator Prof. S. R. M. Prasanna and by **15 January 2018 23:59 hrs IST**.

The Faculty

Prof. Rainer Martin

www.rub.de/ika is currently a professor of Information Technology and Communication Acoustics at Ruhr-Universität Bochum, Germany. He was associated with Institute of Communication Systems and Data Processing, Aachen University of Technology, AT\&T Speech and Image Processing Services Research Lab, Florham Park, N.J. and Technical University of Braunschweig, Germany. He has worked on algorithms for noise reduction, acoustic echo cancellation, microphone arrays, speech quality assessment, and speech recognition. His research interests are signal processing for voice communication systems, hearing aids, and human-machine interfaces.

Prof. S. R. M. Prasanna

https://www.iitg.ernet.in/eee/emstlab/profiles/srmp.php is currently a Professor in the Department of Electronics and Electrical Engineering (EEE) at IIT Guwahati. He has supervised many PhD Theses on different issues related to speech signal processing. He has cofounded Speechwarenet and DFM InfoAnalytics companies working on the development of speech and multimedia products. His research interests include speech processing, handwriting processing and audio processing.

Prof. Ajish K Abraham

http://www.aiishmysore.in/en/departments_electronics_faculty.html is currently a Professor in the Department of Electronics, All India Institute of Speech and Hearing, Mysuru. His teaching and research includes design and development of biomedical instruments, audio and speech processing, research on hearing aid technology, acoustic noise. He has published articles on hearing aid quality analysis, effect of hearing aid compression on speech in reputed national and international journals and conferences.