Vimal Manohar

Contact Information

The Center for Language and Speech Processing,
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Research Interests

Automatic Speech Recognition, Machine Learning, Speech Signal Processing, Natural Language Processing

Education

Johns Hopkins University, Baltimore, MD
Major: Electrical & Computer Engineering
Master of Science in Engineering (M.S.E.), 2015
Ph.D., 2018 (Expected)
Advisors: Sanjeev Khudanpur and Daniel Povey

Indian Institute of Technology Madras, Chennai, India
Major: Electrical Engineering, Minor: Operations Research
Bachelor of Technology (B.Tech), 2013 (CGPA: 9.6/10)
Advisor: S Umesh,

Publications

- Liu, C.; Jyothi, P.; **Manohar, V.** et al., *"Adapting ASR for under-resourced languages using mismatched transcriptions,"* IEEE International Conference on Acoustics, Speech and Signal Processing (ICASSP 2016).
- **Manohar, V.**; Povey, D.; Khudanpur, S., *"Semi-supervised Maximum Mutual Information Training of Deep Neural Network Acoustic Model,"* INTERSPEECH 2015. **Nominated for best students' paper award**.
- Trmal, J.; **Manohar, V.** et al., *"A keyword search system using open source software,"* Spoken Language Technology Workshop (SLT), 2014 IEEE, pp.530,535, 7-10 Dec. 2014 doi: 10.1109/SLT.2014.7078630
- **Manohar, V.**; Srinivas, C.B.; Umesh, S., *"Acoustic modeling using transform-based phone-cluster adaptive training,"* Automatic Speech Recognition and Understanding (ASRU), 2013 IEEE Workshop on , pp.49,54, 8-12 Dec. 2013 doi: 10.1109/ASRU.2013.6707704

Research and Industrial Experience

Jelinek Summer Workshop on Speech and Language Technology (JSALT) 2015
University of Washington Seattle, Seattle, WAS, USA — July – August '15
Member of the research group working on "Probabilitic Transcription of Languages with no native-language transcribers". We showed the utility of mismatched transcriptions from non-native crowdworkers for ASR. (submitted to ICASSP, 2016)

Research Assistant at The Center for Language and Speech Processing
Johns Hopkins University, Baltimore, MD, USA — Aug '13 – Present

IARPA Babel
Developed acoustic models for languages in low-resource setting, HMM-GMM based automatic speech segmentation for ASR, semi-supervised training approaches for hybrid HMM-DNNs and bottleneck feature NNs
(published in SLT, 2014).

DARPA BOLT
Developed multilingual-architecture DNN systems for transfer learning from standard Arabic to Egyptian Arabic

Intern at Analog Devices Inc.
Cambridge, MA, USA — May – Aug '14
Worked on time-frequency masks with multichannel audio for robust speech recognition

Bachelor's Thesis Project
Indian Institute of Technology Madras, Chennai, India — Sept '12 – May '13

Proposed a modification to the HMM-GMM acoustic modeling technique to deal with low-resource settings. We constrained the subspace containing HMM-GMM mean vectors to be defined by piecewise linear transformations of canonical GMM means.
(published in ASRU, 2013)

Time-scaling and Pitch-scaling of synthesized speech
Indian Institute of Technology Madras, Chennai, India — March – May '12

Investigated algorithms for robust VAD, robust pitch estimation, pitch-mark extraction, pitch synchronous overlap-add method of speech synthesis to change the duration and pitch of speech signals

Research Intern at The Institute of Automation
University of Bremen, Bremen, Germany — May – July '12

Implemented a method for estimation of size, position and orientation of isolated 3D objects using a single pair of stereo images

Texas Instruments Analog Design Contest 2011
Indian Institute of Technology Madras, Chennai, India — Sept '11 – Feb '12

Designed and constructed a pulse oximeter on an embedded system for real-time estimation of respiratory rate. Among the top 25 entries to the TI India Analog Design Contest 2011.

Teaching Experience

Fall 2015 — Teaching Assistant, Random Signal Analysis

Coursework

- ❑ Representation learning
- ❑ Speech and audio processing by humans and machines
- ❑ Information Extraction
- ❑ Matrix Analysis
- ❑ Random Signal Analysis
- ❑ Speech Technology
- ❑ Compressed Sensing and Sparse Recovery
- ❑ Information Theory
- ❑ Graph Theory
- ❑ Advanced Operations Research

Distinctions

- ECE Graduate Fellowship 2013, Johns Hopkins University
- Hamburger Fellowship 2013, Johns Hopkins Univeristy
- WISE Scholarship 2012, DAAD, Germany
- All India Rank **191** in IIT-Joint Entrance Examination (IIT-JEE) 2009 (among over 400,000 students)
- Awarded Kishore Vaignayik Protsahan Yojana (KVPY) Fellowship 2008 by Dept. of Science and Technology, Govt. of India
- Awarded National Talent Search (NTS) Scholarship 2007 by National Council of Education, Research and Training, Govt. of India
- Member of IIT Madras team at the National Robotics Contest, Abu Robocon 2011. Placed among the Top 5 in India
- Winner of GE Industrial Defined Problem at Shaastra 2012, IIT Madras, India for design of accelerometer-gyroscope-magnetometer-based 3D-mouse

Skills

Languages:	C/C++, Python, Bash, MATLAB
Toolkits:	KALDI, HTK

References

Will be provided on request.