Vimal Manohar

Contact Information

Center for language and Speech Processing (CLSP)
Electrical and Computer Eng. Department
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Research Interests

Machine Learning with applications to acoustic modeling for automatic speech recognition

Education

Johns Hopkins University, Baltimore, MD

Ph.D. Candidate, ECE (expected May 2018)

- Advisors: Daniel Povey and Sanjeev Khudanpur

Indian Institute of Technology Madras, Chennai, India

(2009-2013) B.Tech in Electrical Engineering

- Thesis Topic: Acoustic Modeling using Phone Transform CAT for Speech Recognition
- Advisor: S Umesh

Publications

- **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.
- Trmal, J.; **Manohar, V.** et al., *"A keyword search system using open source software,"* Spoken Language Technology Workshop (SLT), 2014 IEEE , vol., no., 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 , vol., no., pp.49,54, 8-12 Dec. 2013 doi: 10.1109/ASRU.2013.6707704

Research and Industrial Experience

July '15–August '15 — **Jelinek Summer Workshop on Speech and Language Technology (JSALT) 2015**

Member of the research group working on "Probabilitic Transcription of Languages with no native-language transcribers"

Aug '13–Present — **Research Assistant at Center for Language and Speech Processing, Johns Hopkins University.**

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

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

Sept '12–May '13 **Bachelor's Thesis Project**
Proposed a new acoustic modeling technique, where the parameters of context-dependent states are obtained by the linear interpolation of several monophone cluster models, which are themselves obtained by adaptation using linear transformation of a canonical Gaussian Mixture Model (GMM). (published in ASRU, 2013)

May '12–July '12 **Research Intern at University of Bremen, Germany**
Implemented a method for estimation of size, position and orientation of isolated 3D objects using a single pair of stereo images by fitting the 3D object with multiple superquadrics

Sept '11–Feb '12 **Texas Instruments Analog Design Contest 2011**
Designed and constructed a pulse oximeter for real-time estimation of respiratory rate. Among the top 25 entries to the TI India Analog Design Contest 2011.

Coursework

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

Academic Distinctions

- Gradutate Research Assistant (2014-2015, Johns Hopkins Univeristy)
- ECE Graduate Fellowship 2013, Johns Hopkins University
- Hamburger Fellowship 2013, Johns Hopkins Univeristy
- 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

Skills

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

References

Will be provided on request.