

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*,” Acoustics, Speech and Signal Processing (ICASSP), 2016 IEEE International Conference on
- Peddinti, V.; Chen, G.; **Manohar, V.** et al., “*JHU ASPIRE system: Robust LVCSR with TDNNs, iVector adaptation and RNN-LMs*,” Automatic Speech Recognition and Understanding (ASRU), 2015 IEEE Workshop on , 13-17 Dec. 2015
- **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 “Probabilistic 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

Fall 2015

Teaching Assistant, Random Signal Analysis

EXPERIENCE**COURSEWORK**

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|---|---|
| <input type="checkbox"/> Representation learning | <input type="checkbox"/> Information Theory |
| <input type="checkbox"/> Random Signal Analysis | <input type="checkbox"/> Speech Technology |
| <input type="checkbox"/> Natural language processing | <input type="checkbox"/> Matrix Analysis |
| <input type="checkbox"/> Speech and audio processing by humans and machines | <input type="checkbox"/> Graph Theory |
| <input type="checkbox"/> Non-linear optimization | <input type="checkbox"/> Advanced Operations Research |
| <input type="checkbox"/> Information Extraction from text and speech | <input type="checkbox"/> Computer Vision |
| <input type="checkbox"/> Compressed Sensing and Sparse Recovery | <input type="checkbox"/> Advanced DSP |

DISTINCTIONS

- ECE Graduate Fellowship 2013, Johns Hopkins University
- Hamburger Fellowship 2013, Johns Hopkins University
- WISE Scholarship 2012, DAAD, Germany
- All India Rank **191** in IIT-Joint Entrance Examination (IIT-JEE) 2009 (among over 400,000 students)
- Awarded Kishore Vaidya 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

SKILLS

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

REFERENCES

Will be provided on request.