Daniel Povey

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Education

- **Bachelor's degree in Natural Sciences –** 1993-1997
  Cambridge University, U.K.
    - Took the Natural Sciences Tripos, initially taking biological options.
    - In my last year, under the Part II (General) option of the Natural Sciences Tripos, took a one-year accelerated course in computer science.
    - Graduated with a 1st class degree (the highest form of honors normally awarded).
- **Master's Degree, Computer Speech and Language Processing –** 1997-1998
  Cambridge University, U.K. (Engineering Dept.)
- **Doctoral Degree in Computer Speech Recognition –** 1998-2003
  Cambridge University, U.K. (Engineering Dept.)
    - The title of my thesis was "Discriminative training for Large Vocabulary Speech Recognition".

Work History

- **Engineering Department, Cambridge University, U.K.** – April 2002 – June 2003
  Research Associate
    - Was finishing my PhD and doing related research work.
- **IBM T.J. Watson Research Center, NY –** July 2003–December 2008
  Postdoctoral Researcher, Jun 2003 – Jan 2006; Research Staff Member, Jan 2006 – Dec. 2008
    - Was doing research in speech recognition as part of government research contracts: EARS, GALE, etc.)
- **Microsoft Research, Redmond, WA –** December 2008–February 2012
  Researcher
- **Center for Language and Speech Processing, Johns Hopkins University**– February 2012–June 2015
  Associate Research Scientist
- **Center for Language and Speech Processing, Johns Hopkins University**– July 2015 - August 2019
  Assistant Research Professor
- **Xiaomi Corporation**– November 2019 - present
  Chief speech scientist

Publications

- **Highlights / highly cited work**

The following three publications describe my thesis work, which introduced now-standard methods for discriminative training.

- D. Povey, "Discriminative Training for Large Vocabulary Speech Recognition", PhD Thesis,

Cambridge University, 2004
    - D. Povey and P. C. Woodland, Minimum Phone Error and I-smoothing for improved discriminative training", Proc. ICASSP 2002
    - P. C. Woodland and D. Povey, "Large scale Discriminative Training of Hidden Markov Models for Speech Recognition", Computer Speech and Language, 2002

  The following publications describe advances in discriminative training methods that I made while at IBM.
    - D. Povey, B. Kingsbury et. al, "fMPE: Discriminatively trained features for speech recognition", Proc. ICASSP 2005
    - D. Povey, D. Kanevsky et. al, "Boosted MMI for model and feature-space discriminative training", Proc. ICASSP 2008

  This paper introduced the Kaldi speech recognition toolkit:
    - "The Kaldi Speech Recognition Toolkit", D. Povey, A. Ghoshal et. al, ASRU 2011.

  This paper introduced the Subspace Gaussian Mixture Model:
    - "The Subspace Gaussian Mixture Model– a Structured Model for Speech Recognition", D. Povey, Lukas Burget et. al, Computer Speech and Language, 2011.

  The following paper introduces a popular approach for neural-net based speech recognition:
    - "X-vectors: Robust DNN Embeddings for Speaker Recognition", David Snyder, Daniel Garcia-Romero, Gregory Sell, Daniel Povey, Sanjeev Khudanpur, ICASSP 2018

  The following paper is about a sequence-based training method for neural nets that is now very popular:
    - "Purely sequence-trained neural networks for ASR based on lattice-free MMI", Daniel Povey, Vijayaditya Peddinti, Daniel Galvez, Pegah Ghahrmani, Vimal Manohar, Xingyu Na, Yiming Wang and Sanjeev Khudanpur, Interspeech 2016

The list of publications below may not be complete. For a more complete list, please see http://www.danielpovey.com/publications.html.

- **Journal articles:**
    - "Low latency modeling of temporal contexts": Vijayaditya Peddinti, Yiming Wang, Daniel Povey and Sanjeev Khudanpur, IEEE Signal Processing Letters, 2017
    - D. Povey, K Yao, "A basis representation of constrained MLLR transforms for robust adaptation", Computer Speech and Language, 2011
    - D. Povey, Lukas Burget et al., "The Subspace Gaussian Mixture Model – a Structured Model for Speech Recognition", Computer Speech and Language, 2011
    - H. Xu, D. Povey, L. Mangu and J. Zhu, "Minimum Bayes Risk decoding and system combination based on a recursion for edit distance", Computer Speech and Language, 2011
    - Stanley Chen, Brian Kingsbury, Lidia Mangu, Daniel Povey, George Saon, Hagen Soltau & Geoffrey Zweig, "Advances in Speech Transcription at IBM under the DARPA EARS Program," IEEE Transactions on Audio, Speech and Language processing, 2006
    - P. C. Woodland and D. Povey, "Large scale Discriminative Training of Hidden Markov Models for Speech Recognition", Computer Speech and Language, 2002
    - Hain, T. Woodland, P.C. Evermann, G. Gales, M.J.F. Xunying Liu Moore, G.L. Povey, D. Lan Wang, "Automatic transcription of conversational telephone speech", IEEE Trans on Speech and Audio Processing, Nov. 2005
- **Refereed conference papers (partial list)**

- “Using ASR methods for OCR”, Ashish Arora, Chun Chieh Chang, Babak Rekabdar, Daniel Povey, David Etter, Desh Raj, Hossein Hadian, Jan Trmal, Paola Garcia, Shinji Watanabe, Vimal Manohar, Yiwen Shao, Sanjeev Khudanpur, ICDAR 2019 (accepted)
- “Speaker recognition for multi-speaker conversations using x-vectors”, David Snyder, Daniel Garcia-Romero, Gregory Sell, Alan McCree, Daniel Povey, Sanjeev Khudanpur, ICASSP 2019
- “A teacher-student learning approach for unsupervised domain adaptation of sequence-trained ASR models”, Vimal Manohar, Pegah Ghahremani, Daniel Povey, Sanjeev Khudanpur, IEEE SLT 2018
- “Improving LF-MMI using unconstrained supervisions for ASR”, Hossein Hadian, Daniel Povey, Hossein Sameti, Jan Trmal, Sanjeev Khudanpur, IEEE SLT 2018
- Output-Gate Projected Gated Recurrent Unit for Speech Recognition", Gaofeng Cheng, Daniel Povey, Lu Huang, Ji Xu, Sanjeev Khudanpur, Yonghong Yan, Interspeech 2018
- "Emotion Identification from raw speech signals using DNNs", Mousmita Sarma, Pegah Ghahremani, Daniel Povey, Nagendra Kumar Goel, Kandarpa Kumar Sarma, Najim Dehak, Interspeech 2018
- "Diarization is Hard: Some Experiences and Lessons Learned for the JHU Team in the Inaugural DIHARD Challenge", Gregory Sell, David Snyder, Alan McCree, Daniel Garcia-Romero, Jesus Villalba, Matthew Maciejewski, Vimal Manohar, Najim Dehak, Daniel Povey, Shinji Watanabe, Sanjeev Khudanpur, Interspeech 2018
- "End-to-End Deep Neural Network Age Estimation", Pegah Ghahremani, Phani Sankar Nidadavolu, Nanxin Chen, Jesus Villalba, Daniel Povey, Sanjeev Khudanpur, Najim Dehak, Interspeech 2018
- "Acoustic Modeling from Frequency-Domain Representations of Speech", Pegah Ghahremani, Hossein Hadian, Hang Lv, Daniel Povey, Sanjeev Khudanpur, Interspeech 2018
- "Recurrent Neural Network Language Model Adaptation for Conversational Speech Recognition", Ke Li, Hainan Xu, Yiming Wang, Daniel Povey, Sanjeev Khudanpur, Interspeech 2018
- "Self-Attentive Speaker Embeddings for Text-Independent Speaker Verification", Yingke Zhu, Tom Ko, David Snyder, Brian Mak, Daniel Povey, Interspeech 2018
- "Semi-Orthogonal Low-Rank Matrix Factorization for Deep Neural Networks", Daniel Povey, Gaofeng Cheng, Yiming Wang, Ke Li, Hainan Xu, Mahsa Yarmohamadi, Sanjeev Khudanpur, Interspeech 2018
- "A GPU-based WFST Decoder with Exact Lattice Generation", Zhehuai Chen, Justin Luitjens, Hainan Xu, Yiming Wang, Daniel Povey, Sanjeev Khudanpur, Interspeech 2018
- "Spoken Language Recognition using X-vectors", David Snyder, Daniel Garcia-Romero, Alan McCree, Gregory Sell, Daniel Povey, Sanjeev Khudanpur, Odyssey 2018
- "Semi-Supervised Training of Acoustic Models using Lattice-Free MMI", Vimal Manohar, Hossein Hadian, Daniel Povey, Sanjeev Khudanpur, ICASSP 2018
- "X-vectors: Robust DNN Embeddings for Speaker Recognition", David Snyder, Daniel Garcia-Romero, Gregory Sell, Daniel Povey, Sanjeev Khudanpur, ICASSP 2018
- "A Time-Restricted Self-Attention Layer for ASR", Daniel Povey, Hossein Hadian, Pegah Ghahremani, Ke Li, Sanjeev Khudanpur, ICASSP 2018
- "End-to-end speech recognition using lattice-free MMI", Hossein Hadian, Hossein Sameti, Daniel Povey, Sanjeev Khudanpur, Interspeech 2018
- "Neural network language modeling with letter-based features and importance sampling", Hainan Xu, Ke Li, Yiming Wang, Jian Wang, Shiyin Kang, Xie Chen, Daniel Povey and Sanjeev Khudanpur, ICASSP 2018

- "A pruned RNNLM lattice-rescoring algorithm for automatic speech recognition", Hainan Xu, Tongfei Chen, Dongji Gao, Yiming Wang, Ke Li, Nagendra Goel, Yishay Carmiel, Daniel Povey and Sanjeev Khudanpur, ICASSP 2018
- "JHU Kaldi System for Arabic MGB-3 ASR Challenge using Diarization, Audio-transcript Alignment and Transfer Learning": Vimal Manohar, Daniel Povey, Sanjeev Khudanpur, ASRU 2017
- "Investigation of Transfer Learning for ASR using LF-MMI Trained Neural Networks": Pegah Ghahremani, Vimal Manohar, Hossein Hadian, Daniel Povey, Sanjeev Khudanpur, ASRU 2017
- "Deep Neural Network Embeddings for Text-Independent Speaker Verification", David Snyder, Daniel Garcia-Romero, Daniel Povey and Sanjeev Khudanpur, Interspeech 2017
- "Backstitch: Counteracting Finite-sample Bias via Negative Steps": Yiming Wang, Vijayaditya Peddinti, Hainan Xu, Xiaohui Zhang, Daniel Povey, Sanjeev Khudanpur, Interspeech 2017
- "An exploration of dropout with LSTMs" Gaofeng Cheng, Vijayaditya Peddinti, Daniel Povey, Vimal Manohar Sanjeev Khudanpur and Yonghong Yan, Interspeech 2017
- "A study on data augmentation of reverberant speech for robust speech recognition", Tom Ko, Vijayaditya Peddinti, Daniel Povey, Michael L. Seltzer and Sanjeev Khudanpur, ICASSP 2017
- "Speaker diarization using neural network embeddings", "Daniel Garcia-Romero, David Snyder, Gregory Sell, Daniel Povey, and Alan McCree", ICASSP 2017
- "Deep Neural Network-based Speaker Embeddings for End-to-end Speaker Verification", David Snyder Pegah Ghahremani, Daniel Povey, Daniel Garcia-Romero, Yishay Carmiel and Sanjeev Khudanpur, IEEE Spoken Language Workshop (SLT) 2016
- "Far-field ASR without parallel data", Vijayaditya Peddinti, Vimal Manohar, Yiming Wang, Daniel Povey and Sanjeev Khudanpur, Interspeech 2016
- "Purely sequence-trained neural networks for ASR based on lattice-free MMI", Daniel Povey, Vijayaditya Peddinti, Daniel Galvez, Pegah Ghahramani, Vimal Manohar, Xingyu Na, Yiming Wang and Sanjeev Khudanpur, Interspeech 2016
- "Acoustic modelling from the signal domain using CNN" Pegah Ghahremani, Vimal Manohar, Daniel Povey and Sanjeev Khudanpur, Interspeech 2016
- "Time delay Deep Neural Network-based Universal Background Models for Speaker Recognition", David Snyder, Daniel Garcia-Romero, Daniel Povey, ASRU 2015
- "Pronunciation and Silence Probability Modeling for ASR", Guoguo Chen, Hainan Xu, Minhua Wu, Daniel Povey and Sanjeev Khudanpur, Interspeech 2015
- "JHU aspire system: Robust LVCSR with TDNNs, ivector adaptation and RNN-LMs.", Peddinti, V., Chen, G., Manohar, V., Ko, T., Povey, D., & Khudanpur, S., ASRU 2015
- "A Diversity-Penalizing Ensemble Training Method for Deep Learning", Xiaohui Zhang, Daniel Povey and Sanjeev Khudanpur, Interspeech 2015
- "Modeling Phonetic Context with Non-random Forests for Speech Recognition", Hainan Xu, Guoguo Chen, Daniel Povey and Sanjeev Khudanpur, Interspeech 2015
- "A time delay neural network architecture for efficient modeling of long temporal contexts", Vijayaditya Peddinti, Daniel Povey and Sanjeev Khudanpur, Interspeech 2015
- "Reverberation robust acoustic modeling using i-vectors with time delay neural networks", Vijayaditya Peddinti, Guoguo Chen, Daniel Povey and Sanjeev Khudanpur, Interspeech 2015
- "Audio Augmentation for Speech Recognition", Tom Ko, Vijayaditya Peddinti, Daniel Povey and Sanjeev Khudanpur, Interspeech 2015
- "Semi-supervised Maximum Mutual Information Training of Deep Neural Network Acoustic

Models", Vimal Manohar, Daniel Povey and Sanjeev Khudanpur, Interspeech 2015

- “Parallel training of Deep Neural Networks with Natural Gradient and Parameter Averaging”, Daniel Povey, Xiaohui Zhang and Sanjeev Khudanpur, ICLR Workshop 2015
- “LibriSpeech: an ASR corpus based on public domain audio books”, Vassil Panayotov, Guoguo Chen, Daniel Povey and Sanjeev Khudanpur, ICASSP 2015
- “Improving Speaker Recognition Performance in the Domain Adaptation Challenge using Deep Neural Networks”, D. Garcia-Romero, X. Zhang, A. McCree, and D. Povey, Proc. SLT, 2014
- “Some Insights from Translating Conversational Telephone Speech”, Gaurav Kumar, Matt Post, Daniel Povey and Sanjeev Khudanpur, ICASSP 2014
- “A Pitch Extraction Algorithm Tuned for Automatic Speech Recognition”, Pegah Ghahremani, Bagher BabaAli, Daniel Povey, Korbinian Riedhammer, Jan Trmal and Sanjeev Khudanpur, ICASSP 2014
- “Improving Deep Neural Network Acoustic Models using Generalized Maxout Networks", Xiaohui Zhang, Jan Trmal, Daniel Povey and Sanjeev Khudanpur, ICASSP 2014
- “Quantifying the value of pronunciation lexicons for keyword search in low-resource languages,” Guoguo Chen, Sanjeev Khudanpur, Daniel Povey, Jan Trmal, David Yarowsky, and Oguz Yilmaz. ICASSP 2013
- “Using Proxies for OOV keywords in the Keyword Search Task", Guoguo Chen, Oguz Yilmaz, Jan Trmal, Daniel Povey, and Sanjeev Khudanpur, ASRU 2013
- “Improved feature processing for Deep Neural Networks”, S. P Rath, D. Povey, K. Vesely and J. Cernocky, Interspeech 2013
- “Sequence-discriminative training of deep neural networks”, K. Vesely, A. Ghoshal, L. Burget and D. Povey, Interspeech 2013
- “Combining forward and backward search in decoding", M. Hannemann, D. Povey and G. Zweig, ICASSP 2013
- “Krylov Subspace Descent for Deep Learning”, Oriol Vinyals and D. Povey, AISTATS 2012
- “Generating exact lattices in the WFST framework”, D. Povey, M. Hannemann et. al, ICASSP 2012
- “Revisiting Semi-continuous Hidden Markov Models”, K. Reidhammer, T. Bocklet, A. Ghoshal and D. Povey, ICASSP 2012
- “Modeling Gender Dependency in the Subspace GMM Framework”, Ngoc Thang Vu, Tanja Schultz and D. Povey, ICASSP 2012
- “Revisiting Recurrent Neural Networks for Robust ASR”, Oriol Vinyals, Suman V. Ravuri, Daniel Povey, ICASSP 2012
- Chao Weng, Fred Juang & Daniel Povey, “Discriminative Training Using Non-uniform Criteria for Keyword Spotting on Spontaneous Speech”, Interspeech 2012.
- D. Povey, A. Ghoshal et al., “The Kaldi Speech Recognition Toolkit”, ASRU 2011 (submitted)
- N. T. Vu, D. Povey and T. Schultz, “Modeling Gender Dependency in the Subspace GMM framework”, ASRU 2011
- D. Povey, G. Zweig and A. Acero, “Speaker Adaptation with an Exponential Transform”, ASRU 2011
- N. T. Vu, D. Povey and T. Schultz, “Modeling Gender Dependency in the Subspace GMM framework”, ASRU 2011
- T. Mikolov, A. Deoras, D. Povey, L. Burget, J. Cernocky, “Strategies for Training Large Scale Neural Network Language Models”, ASRU 2011
- Y. Qian, J. Xu, D. Povey, J. Liu, “Strategies for using MLP-based features with limited

target-language training data", ASRU 2011
- H. Xu, D. Povey, L. Mangu and J. Zhu, "Minimum Bayes Risk decoding and system combination based on a recursion for edit distance", Computer Speech and Language, 2011
- Y. Qian, D. Povey and J. Liu, "State-Level Data Borrowing for Low-Resource Speech Recognition based on Subspace GMMs", Interspeech 2011
- D. Povey, Lukas Burget et al., "Subspace Gaussian Mixture Models for Speech Recognition", ICASSP 2010
- D. Povey, Arnab Ghoshal et al., "A Novel Estimation of Feature Space MLLR for Full Covariance Models", ICASSP 2010
- Lukas Burget, Petr Schwarz et al., "Multilingual Acoustic Modeling for Speech Recognition Based on Subspace Gaussian Mixture Models", ICASSP 2010
- Nagendra Goel, Samuel Thomas et al., "Approaches To Automatic Lexicon Learning With Limited Training Examples", ICASSP 2010
- Haihua Xu, D. Povey, L. Mangu, Jie Zhu, "An Improved Consensus-Like Method for Minimum Bayes Risk Decoding and Lattice Combination", ICASSP 2010
- S. Chu, D. Povey et al., "The 2009 IBM GALE Mandarin Broadcast News Transcription System", ICASSP 2010
- H. Soltau, G. Saon et al., "The IBM 2008 GALE Arabic Speech Transcription System", ICASSP 2010
- S. Chu and D. Povey, "Speaking Rate Adaptation using Continuous Frame Rate Normalization", ICASSP 2010
- Haihua Xu, Daniel Povey, Jie Zhu and Guanyong Wu , "Minimum Hypothesis Phone Error as a Decoding Method for Speech Recognition", Interspeech 2009
- Daniel Povey & Brian Kingsbury, "Monte Carlo Model-Space Noise Adaptation for Speech Recognition", Interspeech 2008
- Daniel Povey, Hong-Kwang J. Kuo, Hagen Soltau, "Fast Speaker Adaptive Training for Speech Recognition", Interspeech 2008
- Daniel Povey, Hong-Kwang J. Kuo, "XMLLR for Improved Speaker Adaptation in Speech Recognition", Interspeech 2008
- George Saon and Daniel Povey, "Penalty Function Maximization for Large Margin HMM Training", Interspeech 2008
- Daniel Povey, Dimitri Kanevsky, Brian Kingsbury, Bhuvana Ramabhadran, George Saon & Karthik Visweswariah, "Boosted MMI for Model and Feature Space Discriminative Training", ICASSP 2008
- Balakrishnan Varadarajan & Daniel Povey, "Quick FMLLR for Speaker Adaptation in Speech Recognition", ICASSP 2008
- Daniel Povey, Stephen M Chu & Balakrishnan Varadarajan, "Universal Background Model Based Speech Recognition", ICASSP 2008
- Daniel Povey & Brian Kingsbury, "Evaluation of Proposed Modifications to MPE for Large Scale Discriminative Training", ICASSP 2007
- Daniel Povey & George Saon, "Feature and model space speaker adaptation with full covariance Gaussians," ICSLP 2006
- Daniel Povey, "SPAM and full covariance for speech recognition," ICSLP 2006
- J. Pelecanos, Daniel Povey, Ganesh Ramaswamy, "Secondary Classification for GMM Based Speaker Recognition," ICASSP 2006
- Ghinwa Choueiter, Daniel Povey, Stanley Chen & Geoffrey Zweig, "Morpheme-based language. modeling for Arabic LVCSR", ICASSP 2006
- Geoffrey Zweig, Olivier Siohan, George Saon, Bhuvana Ramabhadran, Daniel Povey, Lidia

Mangu and Brian Kingsbury, “Automated Quality Monitoring in the Call Center with ASR and Maximum Entropy”, ICASSP 2006
  - Daniel Povey, “Improvements to fMPE for discriminative training of features,” Interspeech 2005
  - Daniel Povey, “Phone Duration Modeling for LVCSR,” ICASSP 2004
  - Saon, G. Dharanipragada, S. Povey, D. “Feature space Gaussianization”, ICASSP 2004
  - Roongroj Nopuswanchai & D. Povey, “Discriminative training for HMM-based offline handwritten character recognition”, Proc. Int'l Conf. on Document Analysis and Recognition, 2003
  - D. Povey, M.J.F. Gales, D.Y. Kim & P.C. Woodland, “MMI-MAP and MPE-MAP for Acoustic Model Adaptation,” Eurospeech 2003
  - D. Povey, P.C. Woodland, and M.J.F. Gales. “Discriminative MAP for Acoustic Model Adaptation”. In Proc. ICASSP, 2003
  - M.J.F. Gales, Y. Dong, D. Povey and P.C. Woodland. “Porting: SwitchBoard to the VoiceMail Task.” ICASSP 2003
  - D. Povey & P.C. Woodland, “Minimum Phone Error and I-Smoothing for Improved Discriminative Training,” ICASSP 2002
  - D. Povey & P.C. Woodland, “Frame Discrimination training of HMMs for Large Vocabulary Speech Recognition,” ICASSP 1999
- **Book chapters, technical reports, theses, workshop presentations**
  - “Approaches to Speech Recognition based on Speaker Recognition Techniques” (D. Povey et al.), chapter in: Handbook of Natural Language Processing and Machine Translation (Springer); Olive, Joseph, Christianson, Caitlin, McCary, John (eds.)
  - Daniel Povey, “Subspace Gaussian Mixture Models for Speech Recognition”, Microsoft Research technical report MSR-TR-2009-64, 2009
  - Daniel Povey, Brian Kingsbury, Lidia Mangu, George Saon, Hagen Soltau, Geoffrey Zweig, “fMPE: Discriminatively trained features for speech recognition,” RT'04 workshop, 2004.
  - Roongroj Nopuswanchai & D. Povey, “Discriminative training for HMM-based offline handwritten character recognition”, Proc. Int'l Conf. on Document Analysis and Recognition, 2003
  - Daniel Povey, “Discriminative Training for Large Vocabulary Speech Recognition,” PhD thesis, Cambridge University Engineering Dept, 2003
  - Phil Woodland, Gunnar Evermann, Mark Gales, Thomas Hain, Andrew Liu, Gareth Moore, Dan Povey & Lan Wang: “CU-HTK April 2002 Switchboard System”, Rich Transcription Workshop 2002
  - Young, S., Evermann, G., Kershaw, D., Moore, G., Odell, J., Ollason, D., Povey, D., Valtchev, V., Woodland, P., “The HTK book” (for HTK version 3.2). Technical Report, Cambridge University, Engineering Department, 2002
  - D. Povey and P. C. Woodland, “Large-scale MMIE Training for Conversational Telephone Speech Recognition”, Proc. NIST Speech Transcription Workshop, College Park, MD, 2000.
  - Woodland, P.C and Povey, D. “Large Scale Discriminative Training for Speech Recognition”, in Proc. ICSA Workshop ASR 2000
  - D. Povey & P.C. Woodland, “Frame Discrimination Training of HMMs for Large Vocabulary Speech Recognition,” Technical report, Cambridge University Engineering Dept., 1999
  - Daniel Povey, “Implementation of Frame Discrimination on a large task, MPhil thesis, Cambridge University Engineering Dept, 1999

Honors and awards

(student best paper awards for papers I co-authored not included)

- *ISCA Best Paper published in Computer Speecxh and Language (2009-2013)* For paper "The Subspace Gaussian Mixture Model– a Structured Model for Speech Recognition", by D. Povey, Lukas Burget et. al.

Refereeing activities (etc.)

(very partial list)

- *Associate Editor,* IEEE Signal Processing Letters, 2014 - present.
- *Grant proposal reviewer:* National Science Foundation (Robust Intelligence, 2008), Swiss National Science Foundation (2014).
- *Journal paper reviewer:* IEEE Transactions on Audio, Speech and Language Processing, IEEE Signal Processing Letters, IEEE Journal of Selected Topics in Signal Processing, IEEE Transactions on Pattern Analysis and Machine Intelligence, IEEE Transactions on Neural Networks, Speech Communication, Pattern Recognition, Journal of Computational and Applied Mathematics.
- *Conference paper reviewer:* IEEE International Conference on Acoustics, Speech and Signal Processing (ICASSP), IEEE Workshop on Automatic Speech Recognition and Understanding (ASRU), International Conference on Spoken Language Processing (ICSLP), European Conference on Speech Communication and Technology (EUROSPEECH), Annual Conference of the International Speech Communication Association (INTERSPEECH), Neural Information Processing Systems (NIPS).
- *Technical Chair:* IEEE Workshop on Automatic Speech Recognition and Understanding (ASRU), 2013.
- *Session Chair:* Annual Conference of the International Speech Communication Association (INTERSPEECH), 2012; IEEE International Conference on Acoustics, Speech and Signal Processing (ICASSP), 2014.
- *External Examiner for dissertation*: University of Toronto, 2014.

Academic advising

Note: all advisees were/are jointly advised with Sanjeev Khudanpur

Completed dissertations include:

- Vijayaditya Peddinti
- Guoguo Chen

Thesis advisees with dissertations being worked on as of August 2019 include:

- Xiaohui Zhang
- Pegah Ghahremani
- Hainan Xu
- David Snyder
- Vimal Manohar

There are also a number of advisees in earlier stages of the process.