# Ming Tu

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## SUMMARY

Current PhD student in Speech and Hearing Science with both Bachelor and Master degree in Electrical Engineering. Research interests include speech signal processing, machine learning and deep neural networks. Experience on robust automatic speech and speaker recognition, speech enhancement and audio/speech source separation. Programming skills include C/C++, Python, MATLAB and Linux Shell.

## EDUCATION

*PhD*, **Speech and Hearing Science**
**Arizona State University, Tempe, AZ** **2018(expected)**
**Main courses**: Statistical machine learning (Course project: **Speech dereverberation using deep autoencoder**), Image analytics and information (Course project: **On automatically monitoring colonoscopic image quality and detecting colonic polyp**), Statistical linear models, Convex Optimization, Speech audio processing and perception, Communication neuroscience, Psychoacoustics.
**MOOC**: Stanford Machine Learning, UIUC Heterogeneous Parallel Programming, UCSD Data Structures and Algorithms Specialization

*Master of Engineering*, **Information and Telecommunication Engineering**
**Beijing Institute of Technology, Beijing, China** **02/2014**
**Main courses**: Information theory, Modern signal processing, Digital image processing and pattern recognition, Advanced digital communication, Matrix analysis, Data mining
THESIS - **Research on speech separation from broadcast with background music**.

*Bachelor of Engineering*, **Electronics and Information Engineering**
**Dalian Nationalities University, Dalian, China** **07/2011**
**Main courses**: Signals and systems, Digital signal processing, Stochastic signal processing, Linear algebra, Probability and statistics, Advanced programming (C++), C programming language, MATLAB programming.
THESIS - **A System of License Plate Recognition based on Virtual Instrument**.

## INDUSTRIAL EXPERIENCE

***Machine Learning and Speech Processing Intern, Alibaba Institute of Data Science and Technologies, Bellevue, USA*** *05/2017-08/2017*

- Researched on Deep Neural Network architectures for speech enhancement to improve model robustness of automatic speech recognition and speaker recognition

***Speech Algorithm Research Intern, LG Electronics, San Jose, USA*** *05/2016-08/2016*

- Systems of Environmental sensing, speech enhancement, speaker recognition and voice command recognition based on Deep Neural Networks.

***Research Intern, Institute of Acoustics, Chinese Academy of Sciences, Beijing, China*** *03/2014-06/2014*

- Research on single channel speech enhancement and separation algorithms.

***Speech Processing Intern, Sony China Research Lab, Beijing, China*** *06/2013-08/2013*

- Researched on applying Gaussian Mixture Model based acoustic model adaptation algorithms for acoustic event detection.

## RESEARCH EXPERIENCE

***Graduate Research Assistant, Arizona State University.*** *08/2014-present*

- Interpretable automatic accentedness evaluation.
- Robust and interpretable machine learning algorithms to objectively evaluate dysarthric speech.
- Deep Neural Networks pruning and compression.

***Graduate Research Assistant, Beijing Institute of Technology*** *09/2011-02/2014*

- Researched on single channel speech enhancement, speech/audio separation, voice activity detection and robust speech recognition algorithms.
- Participated in the record and transcription of Mandarin broadcast speech database that is over 30 hours and built continuous Mandarin speech recognition system. Built non-negative matrix factorization based speech enhancement front-end.

## PUBLICATIONS

1. **Ming Tu**, Anna Grabek, Julie Liss and Visar Berisha, "Investigating the role of L1 in automatic pronunciation evaluation of L2 speech", Interspeech 2018 (Submitted)

2. **Ming Tu**, Tao Yu and Gang Liu, "Speech dereverberation with subband Deep Neural Networks", Interspeech 2018 (Submitted)

3. Yishan Jiao, **Ming Tu**, Visar Berisha, Julie Liss, "Simulating dysarthric speech for training data augmentation in clinical speech applications", ICASSP 2018 (Accepted)

4. **Ming Tu**, Visar Berisha, Julie Liss, "Interpretable Objective Assessment of Dysarthric Speech Based on Deep Neural Networks", Interpseech 2017

5. Xu, Zihan, Steven Skorheim, **Ming Tu**, et el. "Improving efficiency in sparse learning with the feed-forward inhibitory motif", Neurocomputing 2017

6. **Ming Tu**, Visar Berisha, Julie Liss, "Objective assessment of pathological speech using distribution regression", ICASSP 2017

7. **Ming Tu**, Xianxian Zhang, "Speech enhancement based on Deep Neural Networks with skip connections", ICASSP 2017

8. Xu Li, **Ming Tu**, Xiaofei Wang, Chao Wu, Qiang Fu, Yonghong Yan, "Single-channel speech separation based on non-negative matrix factorization and factorial conditional random field", Journal of Tsinghua University(Science and Technology), 2017 57:1, 84-88

9. **Ming Tu**, Alan Wisler, Visar Berisha, and Julie M. Liss. "The relationship between perceptual disturbances in dysarthric speech and automatic speech recognition performance", The Journal of the Acoustical Society of America Express Letter, 2016 140:5, EL416-EL422

10. **Ming Tu**, Yishan Jiao, Visar Berisha, Julie Liss, "Models for Objective Evaluation of Dysarthric Speech from Data Annotated by Multiple Listeners", Asilomar Conference on Signals, Systems, and

Computers 2016

11. Yishan Jiao, **Ming Tu**, Visar Berisha, Julie Liss, “Accent Identification by Combining Deep Neural Networks and Recurrent Neural Networks Trained on Long and Short Term Features”, Interspeech 2016

12. **Ming Tu**, Visar Berisha, Yu Cao, Jae-sun Seo, “Reducing the Model Order of Deep Neural Networks Using Information Theory”, IEEE Computer Society Annual Symposium on VLSI 2016

13. **Ming Tu**, Visar Berisha, Martin Woolf, Jae-sun Seo, Yu Cao, “Ranking the Parameters of Deep Neural Networks using the Fisher Information”, ICASSP 2016

14. Yishan Jiao, **Ming Tu**, Visar Berisha, Julie Liss, “Online speaking rate estimation using recurrent neural networks”, ICASSP 2016

15. Yishan Jiao, Visar Berisha, **Ming Tu**, Julie Liss, “Convex Weighting Criteria for Speaking Rate Estimation”, IEEE Transactions on Audio, Speech, and Language Processing, vol. 23, no. 9, pp. 1421-1430, 2015.

16. **Ming Tu**, Xiang Xie, Yishan Jiao, “Towards improving statistical model based voice activity detection”, Interspeech 2014.

17. **Ming Tu**, Xiang Xie, Xingyu Na, “Computational auditory scene analysis based voice activity detection”, in Proc. of International Conference on Pattern Recognition 2014.

18. Yishan Jiao, Xiang Xie, Xingyu Na, **Ming Tu**. “Improving voice quality of HMM-based speech synthesis using voice conversion method”, in Proc. of ICASSP 2014, pp 7914-7918, May 2014.

19. **Ming Tu**, Xiang Xie, Yishan Jiao. “NMF based speech and music separation in monaural speech recordings with sparseness and temporal continuity constraint”, in Proc. of 3rd International Conference on Multimedia Technology, pp 548-555, Nov. 2013.

## COMPUTING SKILLS

Programming language: C/C++, Python, MATLAB, Linux Shell
Speech recognition engine: Kaldi, HTK
Machine learning tools: Tensorflow, Theano
Github page: https://github.com/tbright17