Zhaozhang Jin

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Objective

A summer intern on machine learning and pattern recognition.

Research Interests

Machine learning, neural networks, discrete signal processing, temporal modeling/graphical models, speech analysis and separation, automatic speech recognition (ASR), image segmentation, computational auditory scene analysis, and artificial intelligence.

Education

The Ohio State University, Columbus, Ohio USA (GPA: 3.96/4.0)

M.S., Computer Science and Engineering *(May 2008)*

Ph.D., Computer Science and Engineering *(expected in 2010)*

- Major: Machine learning and artificial intelligence
- Minors: Signal processing, statistics

Shanghai Jiaotong University, Shanghai, P.R.China

B.E., Electrical Engineering *(July 2003)*

Work Experience

Graduate Research Intern Research & Technology Center, Robert Bosch LLC. *Summer 2008*

Research on analyzing and modeling pulse-doppler radar signals.

- Designed and implemented the project of "human-only detection" for security and surveillance system, with a pending patent.
  (machine learning, pattern recognition)

Research Assistant The Ohio State University *Jan. 2006 - current*

Researching machine learning techniques for speech analysis, sound separation, pitch extraction, and image segmentation under Prof. DeLiang Wang, while pursuing the degree.

- Speech segregation:

  Applying harmonic-related features in supervised learning for separating sounds. The problem of sound separation is treated as auditory scene segmentation on the time-frequency plane. Harmonic features, which capture characteristics of each individual sound source, are extracted within time-frequency units (pixels) and classified as target- or interference-dominant using multilayer perceptrons.
  (Pattern classification, neural networks, auditory scene analysis)

- Multipitch tracking:

  Robustly modeling frame-level pitch probabilities and tracking multiple pitches at the same time using temporal models. Three pitch hypotheses – zero pitch, one pitch and two pitches – are defined, which should also be possible to extend to more pitches. Hidden Markov Model is used to capture transition probabilities of different pitch states and pitch shifts within the same state.
  (Speech processing, statistical modeling, graphical models)

- Speech separation and recognition challenge:

Separating and recognizing target speech contents from well aligned two-talker mixtures. Auditory scene analysis is used for separation and uncertainty is incorporated in the decoding end for robust recognition.

(Model-based approach, graphical model inferences, sound separation, uncertainty decoding, missing-data analysis)

Honors and Qualifications

- William James Award for Best Graduate Presentation at *Cogfest 2006*, OSU, May 2006
- University Scholarship Award, SJTU, 1999 - 2002
- Sun Certified Programmer for the Java 2 Platform, March 3, 2001

Skills

- Proficient in programming languages: Java, C, Matlab, and Perl.
- Proficient in operation systems: Windows, Linux, Mac OSX.

Activities

- Organizer of the Artificial Intelligence Group Talk (AI Talk) since 2007.
- Active reviewer in the following conferences/journals:
  - International Conference on Independent Component Analysis and Signal Processing (ICA)
  - International Conference on Acoustics, Speech, and Signal Processing (ICASSP)
  - International Conference on Cognitive Informatics (ICCI)
  - IEEE Workshop on Statistical Signal Processing (SSP)
  - International Joint Conference on Neural Networks (IJCNN)
  - Machine Learning for Signal Processing (MSLP)
  - World Congress on Computational Intelligence (WCCI)
  - Elsevier: Neurocomputing
  - IEEE Transactions on Audio, Speech, and Language Processing

Publications

Z. Jin and D.L. Wang (2009), "Learning to maximize signal-to-noise ratio for reverberant speech segregation," in Proc. IEEE ICASSP, pp.4689-4692 .

Y. Shao, **Z. Jin**, D.L. Wang and S. Srinivasan (2009), "An auditory-based feature for robust speech recognition," in Proc. IEEE ICASSP, pp. 4625-4628.

Z. Jin and D.L. Wang (2009), "A supervised learning approach to monaural segregation of reverberant speech," IEEE Trans. on Audio, Speech, and Language Processing, vol. 17, pp. 625-638.

Y. Shao, S. Srinivasan, **Z. Jin**, and D.L. Wang (2009), "A computational auditory scene analysis system for speech segregation and robust speech recognition," Computer Speech and Language, in press.

K. Hu, P. Divenyi, D.P.W. Ellis, **Z. Jin**, B.G. Shinn-Cunningham, and D.L. Wang (2008), "Preliminary intelligibility tests of a monaural speech segregation system," ISCA Tutorial and Research Workshop on Statistical and Perceptual Audition, pp. 11-16.

Z. Jin and D.L. Wang (2007), "A supervised learning approach to monaural segregation of reverberant speech," in Proc. IEEE ICASSP, pp. IV.921-924.

S. Srinivasan, Y. Shao, **Z. Jin** and D.L. Wang (2006), "A computational auditory scene analysis system for robust speech recognition," in Proc. Interspeech, pp. 73-76.

References

Available upon request.