SHRIKANT VENKATARAMANI

CONTACT INFORMATION

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EDUCATION

University of Illinois at Urbana Champaign, Illinois, USA

Candidate for *Ph.D.* **August 2015 - present**

- Major : Electrical Engineering
- Specialization : Machine Learning and Audio Signal Processing
- GPA : **3.83 /4**

Indian Institute of Technology - Bombay, Mumbai, India

Master of Technology **July 2012 - May 2015**

- Major : Electrical Engineering
- Specialization : Electronic Systems and Signal Processing
- CGPA : **9.78 /10**
- Thesis: Hybrid vocal separation from single channel polyphonic mixtures

Mumbai University, Mumbai, India

Bachelor of Engineering **July 2008 - June 2012**

- Major : Electronics and Communication Engineering
- Percentage : **78.4 %**

RESEARCH INTERESTS

Machine Learning
Deep Learning, Matrix factorization, Sparse dictionary learning, Neural-network alternatives to matrix factorization models.

Signal Processing
Source separation for speech and music, speech processing, Neural networks for source separation, Wavelets and multi-rate signal processing.

Human Computer Interaction for Audio SIgnal Processing
Automating audio processing for faster editing

WORK EXPERIENCE

Research Intern, Adobe Research
Adobe Creative Intelligence Lab. **June 2016 - August 2016**
Advisor(s): Gautham Mysore, Paris Smaragdis **June 2017 - August 2017**

Research Assistant, University of Illinois at Urbana Champaign.
Department of Computer Science. **August 2015 - present**
Advisor(s): Paris Smaragdis
Project: CAREER: Scaling Source Separation to Big Audio Data.

Research Assistant, Indian Institute of Technology Bombay.
Department of Electrical Engineering. **July 2013 - May 2015**
Advisor(s): Preeti Rao, Rajbabu Velmurugan
Project: Speech and Music Source Separation.

SELECTED CURRENT PROJECTS

Generative neural network alternatives to Non-negative audio modeling
- Propose "generative" neural network alternatives to NMF
- Convolutive, Recurrent and multi-layer extensions give significant improvement in source separation performance over NMF
- Amenable to real time and big-data processing
- Best student paper award at MLSP 2017

Adaptive front-ends for source separation
- Neural network alternatives to Fourier and Cosine Transforms
- Inspired the first end-to-end source separation system
- Learns data-driven transforms - Significant improvements in source separation and denoising performance

Acoustics Matching

- Make recordings in one room sound like they were recorded in a different room
- Novel approach to dealing with room reverb (echoes) using modulation envelopes

AutoDub: Automatic Redubbing for Voiceover Editing
- Automated alternative to redubbing to automatically locate and correct errors in a voiceover using Dynamic Time warping
- Only requires recording small snippets with the correct content
- Automatically corrects and replace the error in the voiceover

AWARDS

Best Student Paper Award MLSP 2017
for the paper, "Neural Networks Alternatives to Convolutive Audio Models for Source Separation"

Microsoft Research PhD Fellowship Nominee Oct. 2017
Selected as one of three applicants to represent the Dept. of Electrical and Computer Engineering in the University of Illinois

Google PhD Fellowship Nominee Nov. 2017
Selected as one of three applicants to represent the Dept. of Electrical and Computer Engineering in the University of Illinois

ADVISING

Undergraduate Research at UIUC

- Jonah Casabeer, Adaptive Front-ends for Separation.
- Ryley Higa, Bitwise networks for Audio Denoising.

Undergraduate Research at IIT Bombay

- Shivangi Mahto, Convolutive NMF for Seech separation
- Shrey Agrawal, Species identification by bird-song analyses using convolutive NMF

PROFESSIONAL ACTIVITIES

Reviewer for Journal and Peer-reviewed Conferences

- IEEE Workshop on Machine Learning for Signal Processing (MLSP)
- IEEE Workshop on Applications of Signal Processing to Audio and Acoustics (WASPAA)
- IEEE International Conference on Acoustics, Speech, and Signal Processing (ICASSP)
- IEEE Transactions on audio, speech and language processing

PUBLICATIONS

Peer-reviewed Conference Publications

Shrikant Venkataramani, Paris Smaragdis, Gautham Mysore *"AutoDub: Automatic Redubbing for Voiceover Editing."*, in the proceedings of 30^{th} ACM User Interface Software and Technology Symposium (UIST), Quebec City, Canada, October 2017.

Shrikant Venkataramani, Cem Subakan, Paris Smaragdis, *"Neural Network Alternatives to Convolutive Audio models for Source Separation."*, in the proceedings of IEEE International Workshop on Machine Learning for Signal Processing (MLSP), Tokyo, Japan, September 2017. (**Best student paper award**)

Paris Smaragdis, **Shrikant Venkataramani**, *"A Neural Network Alternative to Non-negative Models."*, in the proceedings of IEEE International Conference on Acoustics, Speech, and Signal Processing (ICASSP), New Orleans, Louisiana, May 2017.

Shrikant Venkataramani, Nagesh Nayak, Preeti Rao, Rajbabu Velmurugan, *"Vocal Separation using Singer Vowel Priors obtained from Polyphonic audio."*, in the proceedings of the Fifteenth International Society for Music Information Retrieval (ISMIR) 2014, Taipei, Taiwan, October 2014.

Shrikant Venkataramani, Rajbabu Velmurugan, Preeti Rao, *"Improving mobile phone based query recognition with a microphone array"*, in the proceedings of Twentieth National Conference on Communications 2014 (NCC), Kanpur, India, February 2014.

PATENTS

Shrikant Venkataramani, Paris Smaragdis, Gautham Mysore *"AutoDub: Automatic Redubbing for Voiceover Editing."*, US Patent Application (pending), 2016