Ziad Al Bawab

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OBJECTIVE

Establishing a full-time research and development career in the area of signal processing and pattern recognition utilizing my experience with data analysis, feature extraction, and statistical modeling and applying it on speech processing, speech recognition, or other related problems

SUMMARY OF QUALIFICATIONS

- Strong background in signal processing, pattern recognition, and machine learning fields
- 7+ years of education and research experience in speech processing and speech recognition, specializing in dynamic articulatory modeling for improved automatic speech recognition performance
- Extensive technical experience from a wide variety of projects performed in related fields
- Excellent interpersonal and communication skills with a passion for innovation and scientific advancements

EDUCATION

CARNEGIE MELLON UNIVERSITY Pittsburgh, PA

Ph.D. in Electrical and Computer Engineering – Advisors: Prof. Richard M. Stern and Prof. Bhiksha Raj *September 2009*

- Dissertation title: "An Analysis-by-Synthesis Approach to Vocal Tract Modeling For Robust Speech Recognition"

  Dissertation focuses on incorporating dynamic articulatory information, driven by ElectroMagnetic Articulograph (EMA) data, into the statistically-motivated speech recognition system to improve phone classification accuracy

UNIVERSITY OF CALIFORNIA, LOS ANGELES Los Angeles, CA

M.S. in Electrical Engineering *major: signal processing* – Advisor: Prof. Abeer Alwan *July 2003*

- Thesis title: "Speech Recognition over Bluetooth Wireless Channels"

AMERICAN UNIVERSITY OF BEIRUT Beirut, Lebanon

B.E. in Computer and Communications Engineering – *with distinction* *July 2001*

PROFESSIONAL EXPERIENCE

Carnegie Mellon University Pittsburgh, PA

Research Assistant in the Robust Automatic Speech Recognition Lab *August 2003 - Present*

- Designed and implemented a dynamic analysis-by-synthesis framework based on a Hidden Markov Model (HMM) whose distortion features are modeled by different observation densities (Exponential, Gaussian, and Rayleigh)
- Incorporated articulatory knowledge in the HMM whose states are composed of a mixture of vocal tract shapes and experimented with approaches for initializing model parameters using information from real articulatory data
- Improved the speed and accuracy of the articulatory synthesis approach devised and reduced the computational complexity of the analysis-by-synthesis distortion framework while maintaining phone classification accuracy
- Designed and implemented a framework to extract analysis-by-synthesis distortion features using articulatory synthesis and a codebook of vocal tract shapes for improved phone classification accuracy
- Experimented with spectrum warping based on formant trajectory mapping from spontaneous speech distribution to read-speech-like distribution using multi-layer perceptrons for improved spontaneous speech recognition
- Experimented with speech duration normalization using dynamic time warping between spontaneous speech and read speech, for improved spontaneous speech recognition
- Implemented vocal tract length normalization based spectral warping for improved speech recognition accuracy
- Collaborated with Carnegie Mellon's speech team working on the "Cognitive Agent that Learns and Organizes (CALO)" project, training acoustic models, decoding experiments, feature extraction, and transcript processing
- Implemented a frame-classifier-based speech endpoint detector for meeting environments and integrated a real-time version of the classifier in the front-end of Carnegie Mellon's meeting recorder for the CALO project

Teaching Assistantship in the Electrical and Computer Engineering Department

- *"Digital Signal Processing",* graduate course, led problem solving sessions *Spring 2006*
- *"Signals and Systems",* undergraduate course, led recitations and laboratory sessions *Spring 2005*

PROFESSIONAL EXPERIENCE (Continued)

Mitsubishi Electric Research Labs Cambridge, MA

Speech Research Internship *June 2008 - September 2008*

- Developed a principled approach for deriving realistic vocal tract shapes from ElectroMagnetic Articulograph (EMA) sensory data. Task also included adapting Maeda's geometric vocal tract model to the EMA data

Microsoft Research Redmond, WA

Speech Research Internship *June 2006 - August 2006*

- Experimented with various microphone-array statistical algorithms for speech de-reverberation and speech signals separation (e.g. Independent Component Analysis) in different room acoustical environments

BBN Technologies Cambridge, MA

Speech and Language Technology Internship *January 2003 - June 2003*

- Performed various speech recognition related tasks including topic classification and call routing

University of California, Los Angeles Los Angeles, CA

Research Assistant in the Speech Processing and Auditory Perception Lab *January 2002 - July 2003*

- Researched the effects of Bluetooth wireless channels on distributed speech recognition performance and proposed a compensation technique for lost packets
- Designed and implemented a Java interactive application that records children's speech for speech research

Teaching Assistant in the Electrical Engineering Department *Spring 2002*

- "*Applied Numerical Computing",* undergraduate course, led recitations and problem solving sessions

British Telecom Ipswich, England

Distributed Programming Internship *June 2000 - September 2000*

- Created a software application that provides an experimental location based service

RELEVANT SKILLS

Programming Languages	C, C++
CAD & Scripts	MATLAB, Perl, CShell/UNIX, LaTeX
Packages	SPHINX (CMU's open source speech recognizer), Java Media Framework, Bluehoc (IBM's Bluetooth Simulator) with Network Simulator

PUBLICATIONS

Ziad Al Bawab, Lorenzo Turicchia, Richard M. Stern, and Bhiksha Raj, "Deriving Realistic Vocal Tract Shapes Using Geometric Adaptation and Profile Fitting of Maeda's Model to Multi-Speaker ElectroMagnetic Articulograph Data," in preparation for submission to *the Journal of the Acoustical Society of America.*

Ziad Al Bawab, Lorenzo Turicchia, Richard M. Stern, and Bhiksha Raj, "Deriving Vocal Tract Shapes From ElectroMagnetic Articulograph Data Via Geometric Adaptation and Matching," accepted in *Interspeech 2009* and awarded the International Speech Communication Association conference grant.

Ziad Al Bawab, Bhiksha Raj, and Richard M. Stern, "Analysis-by-Synthesis Features for Speech Recognition," *in IEEE ICASSP 2008*, Las Vegas, NV., USA, Mar 30-April 4, 2008.

Basil As-Sadhan, **Ziad Al Bawab**, Ammar El Seed, and Mohamed Noamany, "Comparative Evaluation of Different Classifiers for Robust Distorted Character Recognition," in *Proceedings of SPIE, Document Recognition and Retrieval XIII*, San Jose, CA, USA, Jan 15-19, 2006.

Rong Zhang, **Ziad Al Bawab**, Arthur Chan, Ananlada Chotimongkol, David Huggins-Daines, Alexander I. Rudnicky, "Investigations on ensemble based semi-supervised acoustic model training," in *INTERSPEECH-2005,* 1677-1680.

Ziad Al Bawab, "A Frame-Based Classifier that Provides Endpoints for Owner Speech Recorded in Multi-Channel Meetings," *Carnegie Mellon ECE Department Ph.D. Quals*, November 2004.

PUBLICATIONS (Continued)

Satanjeev Banerjee, Jason Cohen, Thomas Quisel, Arthur Chan, Yash Patodia, **Ziad Al Bawab**, Rong Zhang, Alan Black, Richard Stern, Roni Rosenfeld, Alexander I. Rudnicky, Paul Rybski, and Manuela Veloso, "Creating Multi-Modal, User-Centric Records of Meetings with the Carnegie Mellon Meeting Recorder Architecture," *Proc. of ICASSP 2004 Meeting Recognition Workshop*.

Ziad Al Bawab, Ivo Locher, Jianxia Xue and Abeer Alwan, "Speech Recognition over Bluetooth Wireless Channels," *Proc. of 8th European Conference on Speech Communication and Technology, Eurospeech 2003*, Geneva, Switzerland, Sept 1-4, 2003.

ORAL PRESENTATIONS

"Analysis-by-Synthesis Features for Speech Recognition," *at IEEE ICASSP 2008*, Las Vegas, NV., USA, Mar 30-April 4, 2008.

"An Analysis-by-Synthesis Approach to Vocal Tract Modeling for Robust Speech Recognition," *at the Human Language Technology Center of Excellence Quarterly Meeting,* Johns Hopkins University, Baltimore, MD., USA, December 3, 2007.

"A Frame Classifier that Provides Endpoints for "Owner" Speech Recorded in Multi-Channel Meetings," *at the Electrical and Computer Engineering Department Seminar,* American University of Beirut, Beirut, Lebanon, July 14, 2005.

SELECTED ACADEMIC PROJECTS

"Pattern Recognition", compared different classifiers (e.g. SVM, PCA, LDA) for distorted-character recognition *CMU Spring 2005*
"Adaptive Filtering", designed a low-complexity fast-convergence algorithm for sparse channels estimation *UCLA Winter 2002*
"Signal Processing", applying iterative singular value decomposition (SVD) for speech pitch estimation *UCLA Spring 2002*
"Senior Project", created a web-to-phone application using voice over IP, Java, and a computer-telephony card *AUB Spring 2001*
"Networking", programmed a network emulator of data link with many features (delay, threading, queuing...) *AUB Spring 2001*

SELECTED HONORS

Conference Grant Award, International Speech Communication Association, Interspeech 2009 Brighton, UK *September 2009*
Research Fellowship, Carnegie Mellon University, Electrical and Computer Engineering Department *August 2003 - Present*
Research Fellowship, University of California, Los Angeles, Electric Engineering Department *September 2001 - July 2003*

REFERENCES

Prof. **Richard M. Stern** Carnegie Mellon University, Electrical and Computer Engineering Department
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Prof. **Bhiksha Raj** Carnegie Mellon University, Computer Science Department
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