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JONATHAN D. BOLEY

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Audio DSP Engineer
psychoacoustics research and algorithm development

PROFESSIONAL EXPERIENCE

LSB Audio, *Principal Consultant* **August 2007 - Present**
Responsible for technical consulting and project management on numerous part-time projects involving audio algorithm development, embedded firmware coding, optimization, and audio quality assessment.

- Developed several fixed-point audio processing algorithms optimized for iPhone implementation. Worked with clients to license existing code and develop custom algorithms for real-time audio processing.
- Developed floating-point and fixed-point Simulink models of a low-cost hearing aid architecture. Identified several areas for improvement, and coordinated with another engineer to implement these changes in a real-time system.
- Implemented “Record Pause” and “Record Pending” features on a multi-threaded Analog Devices Blackfin-based digital recorder. Collaborated with developers around the world to release a firmware update on a tight schedule.
- Negotiated NDAs and contracts; coordinated with clients to define clear requirements; resourced teams consisting of both employees and subcontractors; maintained project schedules and budgets.

Shure, *Senior DSP Engineer* **May 2005 – August 2007**
Responsible for research and advanced development of real time DSP algorithms for wireless audio technologies.

- Researched several error concealment techniques, designed a novel algorithm in Matlab, and developed a real-time fixed-point implementation of this algorithm on a Texas Instruments c55x DSP.
- Optimized a gain control algorithm to reduce latency by 50%, power consumption by 60%, and memory by 80%.
- Managed outsourcing relationship with an external codec developer- defining requirements, managing expectations, and verifying performance of new algorithms.
- Conducted formal listening tests to evaluate performance of an algorithm prototype. Orchestrated an effort to standardize company listening tests.

Shure, *Audio DSP Intern* **Summers 2003-2004**
Responsible for the development and implementation of real-time DSP algorithms for live sound reinforcement.

- Redesigned dynamic range compressor algorithm to improve level detection and attack characteristics.
- Developed a microcontroller-to-DSP interface and coded the algorithms for peak / rms audio level meters.
- Developed a sweeping tone generator in Matlab and successfully implemented the algorithm on an Analog Devices Sharc processor.
- Ported a feedback reduction algorithm from floating-point code to fixed-point, and optimized the code for use on an Analog Devices Blackfin processor.

Motorola, *Engineering Intern* **Summers 2000-2001 and May-December 2002**
2002 - Responsible for embedded software development and sound quality enhancement for hands-free phones.

- Developed embedded (PowerPC Processor) and PC (MFC) software for real-time tuning of echo cancellation and noise suppression algorithms in automotive hands-free phone systems (e.g. OnStar).
- Researched and implemented (in Matlab) several successful DSP algorithms to improve double-talk detection in hands-free phone software.

2001 - Responsible for researching and testing MEMS inertial sensors (gyroscopes and accelerometers).

- Developed and utilized an automated testing system (in Labview) for testing parametric responses of ICs.
- Conducted various validation tests (noise, vibration response, drift, etc.) and determined the necessary signal processing for product realization.

2000 - Responsible for characterizing competitors’ cellular phones for benchmarking purposes.

- Trained a team of engineers to conduct current drain tests. Utilized cellular communication equipment in combination with LabView software.
- Provided group with RF block diagrams of competitors' cellular phones by conducting measurements with a spectrum analyzer to identify signal flow in RX and TX circuits.

Drivecon, *Electronics Technician* **Summer 1998**
Responsible for assembly and troubleshooting of drive control units for industrial cranes.

- Assembled and modified PCB’s (soldering, potting, etc.)
- Troubleshot circuits (using a schematic and oscilloscope) and made appropriate repairs.

EDUCATION

Ph.D. Biomedical Engineering - *Purdue University, West Lafayette, IN* Expected May 2012
NIH Research Fellow (Grant # F31DC010966) – *"Effects of Hearing Aid Amplification on Robust Speech Coding"*

Masters Certificate, Project Management - *The George Washington University, Washington D.C.* August 2009
Learned project management concepts and tools that I have since used for several academic and consulting projects.

M.S. Music Engineering - *University of Miami, Coral Gables FL* May 2005
Thesis Research: Used concepts of auditory scene analysis to design a perceptual blind source separation algorithm.

B.S. Electrical Engineering - *University of Illinois at Urbana-Champaign* May 2003
Senior Design Project: Designed a portable audio spatializer for headphones (using Dolby algorithms on a 56k DSP).

Certificate, Electronic Equipment Repair - *Lake County Area Vocational Center, Grayslake, IL* June 1998
Repaired various devices ranging from VCRs to hi-fi audio equipment, and participated on US FIRST robotics team.

PUBLICATIONS

Gaston, L., J. Boley, S. Selter, and J. Ratterman, "*The Influence of Individual Audio Impairments on Perceived Video Quality*," in Proceedings of the 128th Convention of the Audio Engineering Society, May 2010.

Heinz, M., J. Swaminathan, J. Boley, and S. Kale, "*Across-Fiber Coding of Temporal Fine-Structure: Effects of Noise-Induced Hearing Loss on Auditory Nerve Responses,"* in The Neurophysiological Bases of Auditory Perception. Springer (New York), March 2010.

Boley, J. and M. Lester, "*Statistical Analysis of ABX Results Using Signal Detection Theory*," in Proceedings of the 127th Convention of the Audio Engineering Society, October 2009.

Lester, M. and J. Boley, "*The Effects of Latency on Live Sound Monitoring*," in Proceedings of the 123rd Convention of the Audio Engineering Society, October 2007.

Boley, J. "*Auditory Component Analysis*," in Proceedings of the 121st Convention of the Audio Engineering Society, October 2006.

SKILLS

Perception	Psychoacoustic & neurophysiological models of loudness, masking, scene analysis, etc. Familiar with latest research on localization, pitch, speech perception, hearing impairment, etc. Psychophysics (numerous experimental methods and statistical analyses)
Algorithms	Filter design, adaptive filters, compression, spatialization, perceptual coding, watermarking, echo cancellation, noise suppression, feedback reduction, automatic mixing, etc.
Languages	Assembly, C, C++, Matlab, Simulink
DSPs	ADI (Blackfin, Sharc), Freescale (56k), TI (c54x,55x,64x)
µProcessors	ARM, Coldfire, HC11, PIC, PowerPC, x86

ACTIVITIES

- Audio Engineering Society, Member (2001-Present)
  - Technical Committees: Perception and Subjective Evaluation, Signal Processing, Audio Coding
  - Chair & panelist for multiple conference paper sessions, workshops, etc. (2008-2009)
  - Chicago Section, Treasurer (2005-2007), Committee Member (2007-Present)
  - Univ of Illinois Student Section, Founder/Chair (2001-2002)
- IEEE Signal Processing Society, Member (1998-Present)
  - Ad Hoc Reviewer, International Symposium on Circuits and Systems (2010)
- Motion Picture Experts Group (MPEG), Member (2003-2005)
- Acoustical Society of America, Member (2001-Present)
- Bass Guitar Musician; Experience with Live Sound and Studio Mixing