SATTWIK BASU

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EDUCATION

University of Rochester, Rochester, NY *Aug 2016 – May 2018*
M.S, Electrical & Computer Engineering, Concentration in Musical Acoustics & Signal Processing
Coursework: Audio Signal Processing, Computer Audition, DSP, Acoustics, Random Processes, Audio Software Dev, Recording tech.

K.M Music Conservatory, Chennai, India *Nov 2014 - July 2016*
Piano Performance

SRM University, Chennai, India *July 2010 – May 2014*
B.Tech, Electrical & Electronics Engineering

WORK EXPERIENCE

Audio DSP Engineer, HARMAN International, Novi, MI *July 2018 – Present*

- *Algorithms for in-car Active Noise Cancellation (ANC)*
  - Invented adaptive algorithms for musical interference cancellation and online secondary path IR estimation to prevent MFxLMS ANC systems from misadjusting due to the presence of music signals or changes in cabin acoustics
  - Developed a narrowband active road noise cancellation algorithm. Implemented a real-time frequency estimator by applying FFT Zoom and Spectral Centroids on accelerometer signals to provide an accurate reference to the ANC algorithm. Developed mathematical models to theoretically describe the tradeoffs between stepsize, leakage, notch filter response, and out-of-band noise boosting
  - Invented a virtual mic ANC algorithm to reduce engine noise at locations far away from error microphones using an adaptive array processing algorithm. This technology helps in achieving better engine noise control performance in underdetermined MIMO ANC systems with reduced or non-optimum error mic placement
- *Auto-tuning algorithms and Simulation Tools*
  - Led the R&D efforts on auto-tuning algorithms for ANC systems using eigen-analysis of impulse response matrices. Demonstrated improvements in noise cancellation and reduced total production tuning time by 70%
  - Developed simulation tools with GUIs in MATLAB for predicting & analyzing ANC performance
  - Developed audio & vibration analysis tools using CQT & Bark filterbanks, Variable Size DFT & multirate DSP algorithms

Auditory Augmented Reality Research Intern, HARMAN International, Mountain View, CA. *June – Sept 2017*
- Implemented source separation algorithms using Non-negative Tensor Factorization (NTF), 8-9 dB SDR on audio mixtures
- Developed SVM and CNN based audio classifiers using MFCCs & log-mel spectrograms on the UrbanSounds8K dataset
- Prototyped an auditory augmented reality system pipeline to separate audio mixtures using NTF and identify individual sources using Deep Learning

Teaching Assistant, University of Rochester, Rochester, NY *Aug 2016 – Dec 2017*
- ECE 446 Digital Signal Processing
- ECE 140 Introduction to Music Engineering
- ECE 210 Circuits & Microcontrollers

RESEARCH EXPERIENCE

University of Rochester, Rochester, NY *March 2017 – Dec 2017*

- *Musical Polyphony Estimation*
  - Developed a CNN based algorithm for detecting the number of active instruments in polyphonic music
  - Improved performance of multi-label instrument classifiers by using polyphony estimates (improved acc. from 64% to 83%)
  - Compared results with multi-pitch estimation algorithms and demonstrated an increase in accuracy (76.7% vs 56.4%)
- *Bringing a Concert Home*
  - Designed a Room Correction + Reverb algorithm for recreating concert hall acoustics in home listening spaces
  - Measured binaural IRs in concert halls and applied MINT inverse filtering to equalize room frequency response for playback
  - Developed prototypes on a two-loudspeaker system and carried out listening tests on 15 subjects

SRM University, Chennai, India *Aug 2013 – Mar 2014*
- Gained an understanding of evolutionary algorithms like Particle Swarm Optimization, Genetic and Spiral Algorithms through MATLAB implementation for optimizing benchmark functions
- Compared the performance of Particle Swarm and Spiral Algorithms in optimizing overcurrent relay coordination in electric power transmission systems (Dissertation: Optimization of Overcurrent Relay Coordination using Evolutionary Algorithms)

PATENT

US Patent 11183166 "Virtual Location Noise Signal Estimation for Engine Order Cancellation", *Issued Nov 2021*
Inventors: **Basu, S.**, Tackett, J.; Trumpy, D.; Tousignant, T.; May, J.

PUBLICATIONS

Basu, S.; Tackett, J., Trumpy, D.; Walt, A.; Adari, S. Study of the Effects of Active Noise Cancellation on Music Playback. SAE Technical Paper, Grand Rapids, MI, 2021

Kareer, S. *; **Basu, S.** * Musical Polyphony Estimation. In Proceedings of the Audio Engineering Society Convention 144, Milan, 2018 (* Equal Contributions)

Basu, S.; Kareer, S. Bringing a Concert Home. In Proceedings of the Audio Engineering Society Convention 143, New York, 2017

PAST PROJECTS

Sound Retrieval using Vocal Imitation *Nov 2017*

- Implemented a stacked auto-encoder neural network for automatic feature learning from vocal sound imitations
- Implemented the backpropagation algorithm and trained neural network using CQT spectrograms
- Performed feature visualization and tested the trained SAE to retrieve the matching sounds through vocal imitation

Pitch (F_0) Estimation and Beat Detection *Sept 2017*

- Implemented the YIN algorithm for fundamental frequency (F0) estimation of speech and polyphonic music
- Implemented time domain and spectral onset detection algorithms to find musically relevant peaks
- Improved beat detection performance using a dynamic programing algorithm and achieved an accuracy of 94%

Real-time Spatial Audio & Reverberation using TI L138 OMAP DSP Development Board *May 2017*

- Computed T60, comb and all-pass filter coefficients, delays using specified dimensions for various rooms
- Created three types of spatial effects using amplitude panning, ITD/IID and HRTFs with slider GUI to control position
- Implemented the Schroeder Reverb to add tunable reverberation effect in real-time using a slider GUI

Phase Vocoder and Fractional Delays *Mar 2017*

- Computed STFT and implemented peak detection algorithm on magnitude spectrograms of music and speech signals
- Performed quadratic interpolation of magnitude and phase spectrograms to independently adjust speed and pitch
- Implemented fractional delays using circular buffer, linear and first-order all-pass interpolation techniques

TECHNICAL SKILLS

Languages: MATLAB (Experienced)
Python (Experienced) | Keras, TensorFlow, Librosa, scikit-learn, NumPy
C++ (Proficient)
Audio Tools: ProTools, AudioMulch, Reaper, GarageBand, Audacity, Max/MSP, Pure Data
Vibroacoustics: HeadAcoustics Artemis
Hardware: TI C66, A15, ADI SHARCs, Function Generators, Oscilloscopes, Audiomatica Clio
Version Control: Git (GitHub, Bitbucket)

AWARDS

- Five-time winner of the Be Brilliant Innovation Award for excellence in R&D at HARMAN between 2018 and 2021
- Anna-Louise Baker Scholarship for excellence in piano performance, Eastman School of Music, Community Center, 2017
- Tuition scholarship from the Hajim School of Engineering, University of Rochester, 2016
- Ranked 3rd in the state in the National Physics Olympiad, Pondicherry, India, 2010