Ahmed Adel Attia, Ph.D. Student

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About Me

As a Ph.D. student at the University of Maryland and a Graduate Research Assistant (GRA) in the Speech Communication Lab, I possess technical and academic expertise in Deep Learning and its applications, especially in Speech Processing, Modeling, and Recognition. I also have previous experience in Efficient Machine Learning and Computer Vision.

Skills

Programming Proficient in Python3, including expertise in TensorFlow, TensorFlow Dataset, TensorFlow Profiler, and PyTorch. Also experienced in C, C++, Matlab, LLVM, Verilog, and VHDL.

Research

- Deep Learning
- Speech Processing and Modeling
- Automatic Speech Recognition (ASR)
- Audio Processing
- Real-time Deep Learning Models
- Unsupervised and Self-Supervised Learning
- Natural Language Processing
- Signal Processing

Experience

University Of Maryland *Maryland, USA*
Graduate Research Assistant *January 2022 - Present*

- **Advisor:** Prof. Carol Espy-Wilson.
- Conducting research to develop Deep Learning and Machine Learning algorithms for acoustic and articulatory speech data to better understand speech production.
- Developing ASR systems for children's speech for use in classroom settings.
- Authored and published a conference paper (arXiv: 2210.15195) to IEEE ICASSP 2023 within the first 10 months of the position.
- Several other publications in top Speech and Signal Processing conferences.
- Acted as Lab Manager, effectively liaising with the IT department to address technical issues and overseeing the procurement of computational resources.

Omnispeech, LLC *Remote*
Deep Learning Consultant *June 2021 - January 2022*

- Worked on developing lightweight real-time speech enhancement deep learning models.
- Successfully scaled down large Speech Enhancement GAN models from 37 million parameters to less than 1 Million parameters, maintaining good clarity and noise cancellation.
- Developed an efficient data pipeline using TensorFlow Dataset API and TensorFlow profiler for more than a terabyte of audio data achieving optimal performance and $\sim$ 100% GPU utilization.

University Of Arizona *Arizona, USA*
Deep Learning Research Intern *July 2019 - September 2019*

- Conducted research on Generative Adversarial Networks (GANs) under the supervision of Prof. Ravi Tandon.
- Worked on and helped with different projects on GANs.
- Developed Mutual Information Neural Estimators that achieved over 98% accuracy, and unsupervised Outlier Detection systems.

Publications

Masked Autoencoders Are Articulatory Learners *arXiv:2210.15195*

ICASSP 2023

- Developed a novel masked autoencoder model for reconstructing mistracked speech recordings in the X-ray Microbeam (XRMB) dataset.
- Achieved state-of-the-art performance in reconstructing articulatory parameters, with a 96.4% success rate in reconstructing concurrently mistracked features.
- The main novel contribution of my paper was reconstructing recordings where multiple articulatory features were concurrently mistracked.
- Published at ICASSP 2023, a top-tier conference in speech and audio processing.

Enhancing Speech Articulation Analysis Using A Geometric Transformation Of The X-ray Microbeam Dataset *arXiv:2305.10775*

Interspeech 2023

- Developed a novel geometric transformation that improves the accuracy of speech articulation measurements in the X-ray Microbeam Dataset (XRMB).
- Proposed the extension of the palatal trace to include the soft palate and anterior pharyngeal wall, enhancing the calculation of tongue body constriction, particularly in back vowels.
- Addressed limitations of previous models by incorporating additional anatomical landmarks, resulting in a more accurate representation of speech production.

Audio Data Augmentation for Acoustic to articulatory Speech Inversion using Bidirectional Gated RNNs *arXiv:2205.13086*

EUSIPCO 2023

- Implemented a Bidirectional Gated Recurrent Neural Network (BiGRNN) as the speech inversion system and utilized three data augmentation methods to improve the performance of the system on both clean and noisy speech data.
- Achieved a 5% relative improvement in correlation for clean speech data compared to the baseline noise-robust system.
- Demonstrated an additional 6% improvement in average correlation by adapting the pre-trained model to unseen speakers in the test set.

Education

University of Maryland *Maryland, USA*

Ph.D. in Computer Engineering *2020 - 2025 (Expected)*

- Conducted research in deep learning-based natural language processing and speech, with courses in these areas as well as unsupervised learning and speech and audio processing
- Recipient of the Dean's Fellowship, demonstrating academic excellence and potential for research

Alexandria University *Alexandria, Egypt*

B.Sc. in Communications And Electronics Engineering *2015 - 2020*

- GPA: 3.82, Top fifth student in a class of 300.

Awards

The Jacob K. Goldhaber Travel Grant, University of Maryland

- Awarded to support my travel expenses to attend The International Conference on Acoustics, Speech, & Signal Processing 2023 in Rhodes, Greece.

Dean's Fellowship, University of Maryland

- Received a prestigious Dean's Fellowship to support my doctoral studies at the University of Maryland (UMD). This fellowship is awarded to exceptional graduate students.