Geet Khatri

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Education

Delhi Technological University — New Delhi, India
B.Tech. in Electronics and Communication Engineering • Cumulative Performance Index: 77.02% — 2014–2018

Experience

UBS — Pune, India
Software Engineer — 2018–Present

- Developed tools for enforcement and tracking of controls
- Worked on projects involving identification of trends in data, predictive modeling and data visualization

Delhi Technological University — New Delhi, India
Research Intern — Summer 2017

- Project: *Onset Detection in Audio Signals*
- Worked on the development of an algorithm for normalizing the spectral flux of an audio signal in order to simplify peak-peaking
- Applied onset detection to adaptive window size selection for a more accurate time-frequency representation of audio signals

Projects

- **Speech Enhancement Using Wiener Filtering and Pitch-Synchronous STFT Phase Reconstruction** *(Undergraduate Thesis)*
  The algorithm reconstructs the phase spectrum of the pitch-synchronous STFT of the speech signal. Certain properties of the pitch-synchronous STFT of harmonic signals make estimation of the phase spectrum faster and more robust to noise compared to phase reconstruction of constant–window size STFT. The magnitude spectrum is estimated using Wiener filtering.
- **Speech Enhancement Based on Maximization of Mutual Information**
  This method of speech enhancement involves maximization of the mutual information between the message spoken by the speaker at the far-end and the message perceived by the listener at the near-end.
- **STFT Phase Reconstruction for Speech Enhancement**
  The algorithm estimates the phase spectrum of the underlying speech signal. The voiced part of the speech is modeled using the harmonic model. As a prerequisite step, the fundamental frequencies of the voiced speech are estimated. The phase reconstruction algorithm makes it possible to enhance speech using only the fundamental frequencies and the noisy signal.
- **Lossy Audio Compression Using a Statistical Sub-Band Model of Quantization Noise**
  The algorithm, which is based on MPEG AAC, minimizes the perceived distortion due to compression. This distortion is related to the quantization noise over frequency sub-bands. A key step in the algorithm is computation of the optimal sub-band scalefactors using a statistical model of quantization noise.
- **Image to Sound**
  For a given image, the algorithm creates a sound whose spectrogram looks like the image. It maps the pixel intensities of the image to the amplitudes of the spectrogram and randomizes the phase spectrum. Prior to mapping, edge detection is applied to the image in order to make the sound more distinctive and the spectrogram's features more pronounced.

Skills and Coursework

- **Languages:** Python, MATLAB, C, C++, Bash, SQL, HTML, CSS, JavaScript, VHDL
- **Frameworks:** TensorFlow, Keras
- **Software and Tools:** MATLAB/Octave, Git, LTspice, Tableau, Excel, LaTeX
- **Relevant Coursework:** Signals & Systems, Probability & Stochastic Processes, Digital Signal Processing, Analog & Digital Communication, Information Theory & Coding, Image Processing, Computer Vision, Detection & Estimation Theory, Pattern Recognition, Audio Signal Processing, Introduction to Computer Science, Deep Learning (Specialization), Introduction to Data Science

Teaching and Mentoring

- **Python Basics / Advanced Python** — July–September 2020
  Conducted sessions on Python for different teams at UBS
- **Introduction to Programming in C** — Fall 2016
  Conducted introductory sessions on C programming as a Tech Head at Society of Robotics, Delhi Technological University