Geet Khatri

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Education

Delhi Technological University New Delhi, India
B.Tech. in Electronics and Communication Engineering 2014–2018

Experience

UBS Pune, India
Software Engineer 2018–Present

- Software development and data analysis

Delhi Technological University New Delhi, India
Research Intern Summer 2017

- Project: *Onset Detection in Audio Signals*

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, Analog & Digital Communication, Digital Signal Processing, Information Theory & Coding, Control Systems, Image Processing, Computer Vision, Detection & Estimation Theory, Pattern Recognition, Audio Signal Processing, Introduction to Computer Science, Deep Learning (Specialization), Introduction to Data Science

Projects

- **Speech Enhancement Using Wiener Filtering and Pitch-Synchronous STFT Phase Reconstruction**
  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 accurate 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 (measured as the intelligibility of noisy speech) between the message spoken by the speaker at the far-end and the message perceived by the listener at the near-end.
- **Onset Detection in Audio Signals**
  The algorithm uses a measure similar to spectral flatness to normalize the spectral flux in order to simplify peak-peaking. This algorithm is then applied to adaptive window size selection for more accurate time-frequency representation of audio signals.
- **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 statistical model of the quantization noise is used for computation of the optimal scalefactors for sub-bands.
- **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. Before the mapping, edge detection is applied to the image in order to make the sound more distinctive and the spectrogram's features more pronounced.

Teaching

- **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