Speaker Recognition
Research, Applications, and Conclusions

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Overview
   Refresher on Speaker Recognition

Approaches
   Discrete Fourier Transform
   Linear Predictive Coding

Results
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In General

**Input:** Audio encoding of speech

**Output:** Information that is used to classify speaker

Applications:
- User authentication
- Biometrics
- Caller ID
- …
The Problem I Approached

- **Speaker Identification**
  Identify person who spoke voice sample.
  - Population of seven people
  - Both male and female speakers
  - Text Independent

- **Tools**
  - MATLAB
  - Neural Network Toolbox
  - sound-recorder
Sample Collection

- **Direct Microphone input into MATLAB**
  - Impractical
  - Would have to store the samples somewhere, anyway...
  - Not implementable, we don’t have the DSP toolbox installed.

- **Pre-recorded .wav files**
  - Portable
  - Collected via USB Microphone off-site
  - MATLAB has tools to work with wave files
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Sample Details

- **7 subjects**
  - 4 females
  - 3 males
  - similar ages

- **Each subject records 1 minute of speech**
  - 16-bit
  - 1 channel
  - 44100 Hz sample rate

- Speech sample broken up into 5 10-second chunks

- Loaded into MATLAB via `wavread()`
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What is it?

\[ X_k = \sum_{n=0}^{N-1} x_n e^{-\frac{2\pi i}{N} k n} \quad k = 0, \ldots, N - 1 \]

- **Time Domain \rightarrow Frequency Domain**
  - Implemented in MATLAB via `fft()`
  - Allows analysis of signals by their component frequencies
  - Feed frequencies into NN
- Sounds like it’s ideal for voice work, right?
  - A person’s voice has dominant frequencies
  - Text-independent by nature
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- **Issues with DFT**
  - Too much noise - spectrum is very variable
  - Too much information - spectrum typically as big as sample size
  - Inconsistent - spectral amplitude dependent on volume

- **Solutions**
  - Noise filtering
  - Take only important, classifiable parts of spectrum
  - Equalize volume of all samples prior to analysis
  - Window functions
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Window functions

- Further divides sample signal into parts
- Amplifies inconsistency between different parts of one signal
In the end, DFT’s are impractical for the problem.

- Fiddly
- Difficult to extract the *right* information
- Accuracy anywhere from 25% to 60%
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What is LPC?

Linear Predictive Coding is a "tool for representing the spectral envelope of speech in compressed form".

- Originally used for compression and data transfer by early communications networks
- Currently in use by phone companies for cell network data transfer
What does LPC give us?

- Time signal $\rightarrow$ Polynomial coefficients that represent it
- An approximation can be reproduced from the polynomial coefficients
- MATLAB’s function \textit{lpc()} simplifies the implementation
Getting LPC to work

- To produce a good polynomial approximation, LPC must be applied to frames
- 30-40 frames per second recommended for good speech reproduction

In my implementation, I used window functions to create these frames.
My approach

- Generate time series from .wav files
- Move through each signal, 1500 samples at a time
- Generate LPC coefficients for 1500 sample window
- average all LPC coefficients for a sample together

This creates a fairly unique representation for a signal.
Results

• Not very good...
• LPC on averaged windows has terrible accuracy. LPC on the whole signal (10 seconds with a 32-degree polynomial!) had better accuracy, even though that makes little sense.
• This suggests that averaging windows does not yield a good characteristic of a voice.
More Results

**Trained data:**
- Accuracy on windowed LPC: 40% of speaker files properly identified
- Accuracy on non-windowed LPC: 71% of speaker files properly identified

**Untrained data:**
- Accuracy on windowed LPC: 32% of speaker files properly identified
- Accuracy on non-windowed LPC: 45% of speaker files properly identified
Future Goals

- Improve the LPC windowing to work
- Investigate whether averaging LPC results is a good idea
- Investigate Neural Network issues
- Implement SOM-based voice recognition, not just MLP.