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()*

What is it?

$$X_k = \sum_{n=0}^{N-1} x_n e^{-\frac{2\pi i}{N} kn} \qquad k = 0, \ldots, N-1$$

- **Time Domain → 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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It's not.

- **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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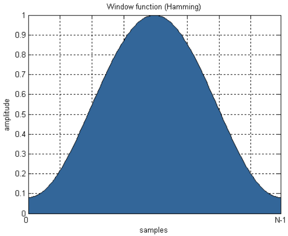
Window functions



- 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 *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-windowes LPC: **71**% of speaker files properly identified

Untrained data:

- Accuracy on windowed LPC: **32**% of speaker files properly identified
- Accuracy on non-windowes 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.