



Sri Aditya Deevi (SC18B080)

*ECE (AVIONICS) Indian Institute of Space science and Technology* 

## Human Speech Analysis



## Speaker Recognition

 Process of automatically recognizing who is speaking on the basis of individual information included in speech waves



## Why Speaker Recognition?

• Some applications are :

Identity verification for telephone banking, database access services, extra security control in remote computer access, laboratory access etc.



## Types of Speaker Recognition

### (a) Speaker Identification



### (b) Speaker Verification



## Phases of ASR System

#### • Training/Enrolment Phase :

Samples of speech from registered speakers to build specific reference models



#### • Testing Phase :

Unknown samples of speech matched with the bank of speaker reference models

\*In this work, we are considering Speaker Identification systems

## Speech Feature Extraction

• First stage in both Enrolment and Testing phases.



Speech signal is "Quasi-stationary"

Can be characterised by Short-Time Spectral Analysis

## Mel Frequency Cepstrum Coefficients

"Perceptually-Relevant Time-Frequency Representation"

- Can be used for characterization of speech (frame).
  - MFCC's can be calculated as follows :



## Mel Frequency Cepstrum Coefficients

## Mel Frequency Scale

 From Psychoacoustic experiments, scientists concluded that human beings perceive frequencies logarithmically.



 $mel = 2595 * log_{10}(1 + rac{f}{700})$ 

## Mel Filter Banks

- Convert the frequency scale of STFT (Short Time Fourier transform) to Mel-Scale to represent the perceptual difference. (Mel Frequency Wrapping)
  - A Mel Filter Bank consists of a number Mel bands. (which is a hyper-parameter that needs to be optimized)

Here, we consider No. of Mel Bands = 26

## Mel Filter Banks



Human Voice Frequency : (300 - 7000) Hz

 $f_s = 12500 Hz$  (To avoid aliasing)

## Mel Frequency Cepstrum Coefficients

Cepstrum



# $C(x(t))=F^{-1}(log(F(x(t))))$

### Understanding Human Speech Generation

### Speech is generated by Vocal Tract



#### (Speech Carrier)

Glottal Pulses generated by Vocal Cords



• Vocal Tract can be modelled as a filter.

Speech = h(t) \* g(t)

g(t) -> Glottal Pulses

h(t) -> Impulse Response of Vocal Tract

## Understanding Cepstrum





### DFT





## Understanding Cepstrum



 $X(f)_{dB} = H(f)_{dB} + G(f)_{dB}$ 



Formants carry the identity of sound unique to speaker

## Understanding Cepstrum



## Choose intermediate set of coefficients (Ex: 2-20)

Reject 0 quefrency and very high quefrencies





## Why DCT?

$$ilde{c_n} = \sum_{k=1}^K (log ilde{S_k}) cos[n(k-rac{1}{2})rac{\pi}{K}]$$

n=0,1,2,...,K-1

where  $\tilde{S}_k$  = Mel power spectrum coefficients K = 26 Real Coefficients

**ADVANTAGES** 

Decorrelation of Francis Mal

Decorrelation of Energy in Mel Bands

Basis Functions are cosines

**Dimensionality Reduction** 

## Final Remark on MFCC's

• The Frame Size considered is 256 samples (~21 msec)

 An *acoustic vector* (MFCC Coefficients) is computed for each frame and stored as an *acoustic matrix*.

CHARACTERISTIC of the speaker

• The dimensions of the acoustic matrix are :

 $(\#MFCC_{coefficients}, \#Frames)$ 

## Speech Feature Matching

Problem at hand is that of "Supervised Pattern Recognition"

Working Principle : Based on the assumption, that acoustic vectors are a unique feature representation of a speaker's voice.

 In this project, Vector Quantization (VQ) is considered for Feature Matching.

## Vector Quantization

#### \*Acoustic vectors here

- Process of mapping vectors from a large vector space to a finite number of regions in that space.
  - Each such region is called a "Cluster".
  - Centroid of cluster is called a "Codeword".
  - Collection of all codewords is called "Codebook".

In this project, *speaker-specific* codebooks are considered.

### Vector Quantization



## Building Speaker-specific Codebooks

Speech (Particular Speaker)



### MFCC Processing

Acoustic Vectors

Codebook Formation by clustering using K-Means





## Flow of Various Phases



### A Note on the Dataset

- The training data consists of 13 audio files (13 different speakers) of 2 second duration sampled at 12500 Hz.
- All the speakers are asked to say "zero" for the experiment.

- The test data also contains similar format audio files (13 in number).
- The test audio files are collected from the speakers after some time (typically some days) to simulate any minor variations in voice.

Live Demonstration and Summary of Results

# THANK YOU!

## References

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