School of ECE

DSP using PBL Approach

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Speaker Recognition For User Authentication

Theme: Speech Signal Processing

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KLE Technological University
Creating Value
Leveraging Knowledge

Need Statement

In a era with high need of security, a speaker recognition for user authentication system has to be built for the security of personal space of single user. The voice signals must be compared with the signal stored in database after signal preprocessing and the required features extraction. The signals are to preprocessed such that there are no distortion in the signals. The valid user must be allowed to login whereas invalid user should not be allowed after comparing the extracted features of both the signals.

Literature Survey

1. **User Identification System Using Biometrics Speaker Recognition by MFCC and DTW along with signal processing package**,T. Muttaqi, S. H. Mousavinezhad and S. Mahamud [2018]. This work provides a comprehensive overview of signal processing techniques applied to audio signals, methods to extract features and comparison.
2. **Mel-frequency cepstral coefficient analysis in speech recognition**, C. K. On, P. M. Pandiyan, S. Yaacob and A. Saudi, [2006]. This review analysis the MFCC and achieved 98.9% of classification result.
2. **Voice Recognition System on MATLAB for Beginners Using Euclidean Distance,** Akanksha Singh Thakur, Namrata Sahayam [20xx]. The study gives information of feature extraction and calculates euclidean distance for authentication.

Challenges and Considerations

1. Noise Reduction: To remove noise without compromising the integrity of the audio signal
2. Computational Efficiency: To address computational complexity and memory requirements, the choice between Discrete Fourier Transform (DFT) and Fast Fourier Transform (FFT) methods.
3. Interpretability: To ensure that the extracted features and classification results are interpretable, facilitating effective secure system.

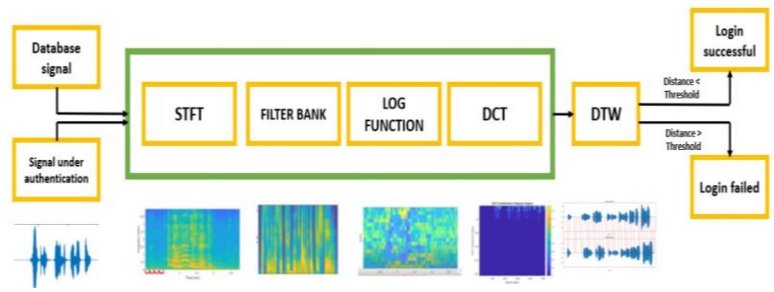
Problem Definition

Design signal processing method to enhance the quality of audio signals and extract relevant features to authenticate the validate user for the secure system.

Objectives

1. Apply Fourier transform to analyze the frequency components of audio signals.
2. Design digital filtering techniques to reduce noise from audio signals.
3. Design a model that authenticates the user based on his/her unique voice characteristics.

Signal Flow Diagram



Signal Characteristics

Signal Parameters

Signal length	3 sec
Amplitude range	-1 to 1 V
Sampling frequency	44100 Hz
Frequency range	300-3000 Hz

Time domain signal

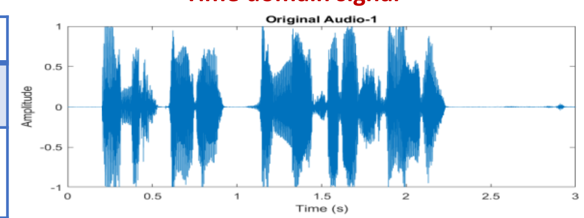


Fig: Audio signal containing noise

Frequency Spectrum

$$X[k] = \sum_{n=0}^{N-1} x[n]\, e^{-j\frac{2\pi kn}{N}}$$

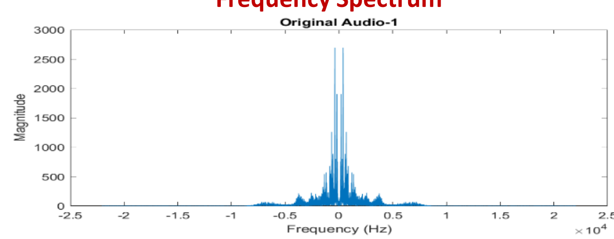


Fig: Frequency spectrum of audio signal containing noise

The spectrum shows frequency components of audio signal up to 8000 Hz, QRS complex in the band 300 to 3000 Hz, 5 kHz power line noise and high frequency noise.

Filter Design

- Concept Mapping: To Extract QRS complex having frequency range between: 300 Hz to 3 kHz
- Identified Filter : IIR Butterworth Bandpass Filter

Design Parameters:

Passband edge frequencies (Hz)	Stopband edge frequencies (Hz)	Passband gain (dB)	Stopband gain (dB)	Filter order N
300, 3000	200, 3100	-3	-20	8

Ideal impulse Response of Filter:

$$h_d[n] = \begin{cases} \dfrac{\sin(\omega_{c_2}(n-N)) - \sin(\omega_{c_1}(n-N))}{\pi(n-N)} & n \neq N \\ \dfrac{\omega_{c_2} - \omega_{c_1}}{\pi} & n = N \end{cases}$$

Hamming window function:

$$\omega_{Ham}[n] = 0.54 - 0.46\cos\left(\frac{2\pi n}{N-1}\right)$$

Impulse Response of FIR BP Filter:

$$h[n] = h_d[n]\,\omega[n]$$

Frequency Response of Filter

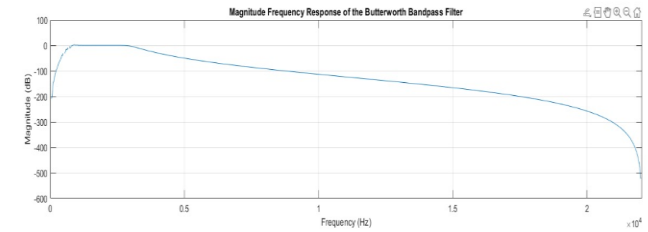


Fig: Frequency response of the IIR Bandpass Filter.

Results

Time domain output

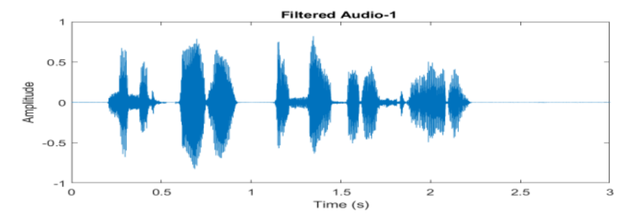


Fig: Processed Audio signal with QRS Complex

Frequency domain output

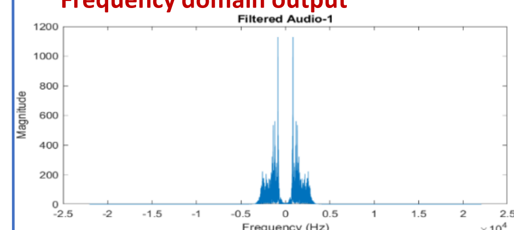


Fig: Frequency Spectrum of Processed Audio signal

Distance comparison

Users	Distance
S_1 with S_2	1061.025
S_1 with A_2	1202.534
S_1 with G_2	1244.854
S_1 with M_2	1244.218

- **Bandpass** filtered signal contains QRS complex
- Distance threshold set : 1150
- Distance lessthan thershold set **Valid user** else **Inavlid User**.

```
Distance between audio files: 1202.534035
Login successful.
```

--> Valid User

```
Distance between audio files: 1202.534035
Login failed.
```

--> Invalid User

Inferences and Future Scope

- Apply DSP algorithms for real world (Audio) signal analysis.
- Frequency domain analysis gave an insight of signal characteristics. This facilitated in selection and design of suitable digital filter.
- The filtered output helped in better feature extraction and comparison of signals.
- Further exploration of advanced DSP algorithms / adaptive filters helps to improve feature extraction and classification accuracy