

**FEATURE EXTRACTION FOR VOICE
RECOGNITION**

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ABSTRACT

The name of this thesis is Feature extraction for voice recognition. Several research for voice recognition have been analyzed. The design stages of voice recognition system are described. One of the most important blocks of voice recognition are feature extraction and classification blocks. The Mel-frequency approach used for feature extraction is explained. The designed system is implemented using the Matlab program.

Communication between human beings is done quickly and more efficiently by voice. Voice is the air flow resulting from the discharge of exhaust air from the lungs. The compressed audio is described in the form of acoustic waves emitted from the comparison. Voice recognition uses human voice. Removing the keys in the way of human-machine communication enables action to be taken by a human voice. It recognizes spoken words and phrases. Voice recognition, speaker identification means to identifying the speakers. For example, It uses cryptic room, the top level requires two points that require security or information retrieval system, ...etc.

Feature extraction is an important step in voice recognition. In this thesis, feature extraction techniques for voice recognition have been analyzed. Using MFCC (Mel-frequency cepstrum coefficients) and vector quantization the feature extraction is presented. The technique has shown good performance in recognition of voices, the designed system is implemented in Matlab.

Keywords: Voice recognition; Power Spectrum (Linear and Logarithmic); Windowing; Mel-frequency cepstrum coefficients (MFCC); Vector Quantization

ÖZET

Bu tezin adı ses tanıma için özellik çıkarmadır. Ses tanıma için çeşitli araştırma analiz edilmiştir. Ses tanıma sisteminin dizayn aşamaları tarif edilmektedir. Ses tanıma en önemli taşlarından biri özellik çıkarımı ve sınıflandırma taşlarıdır. Özellik çıkarımı için kullanılan mel-frekans yaklaşımı açıklanmıştır. Tasarlanan sistem Matlab programı kullanılarak uygulanır.

İnsanlar arasındaki iletişim daha hızlı ve verimli bir şekilde sesli olarak yapılır. Ses akciğerlerden havanın dışarı atılmasında kaynaklanan hava akışıdır. Sıkıştırılmış ses karşılaştırıldığında yayılan ses dalgaları şeklinde tanımlanmaktadır. Ses tanıma da insan sesini kullanılır. İnsan-makine iletişimi yolunda anahtarlarını çıkarma işlemi bir insan sesinin sesli alınmasını sağlar. Bu konuşulan kelime ve aşamaları tanır. Ses tanıma, konuşmacı tanıma hoparlör belirlenmesi anlamına gelir. Ses Tanıma kimlik doğrulaması ya da konuşmacı kimliğini doğrulamak için kullanılabilir. Örneğin, gizli odalarda kullanılır, üst düzey güvenlik veya bilgi erişim sistemi gerektiren noktalarda kullanılır, ... vb.

Özellik çıkarma ses tanıma için önemli bir adımdır. Bu tezde, ses tanıma özelliği çıkarma teknikleri analiz edilmiştir. MFCC (Mel frekans sepstrumu katsayıları) kullanılarak ve vektör niceme özellik çıkarma sunulmuştur. Teknikte seslerin tanınması iyi bir performans göstermiştir, tasarlanan sistem Matlab'ta uygulanmaktadır.

Anahtar Kelimeler: Ses tanıma; Güç spektrumu (Doğrusal ve Logaritmik); Pencereleme; Mel-frekans cepstrum katsayıları (MFCC); Vektör Niceme

TABLE OF CONTENTS

ACKNOWLEDGEMENT	i
ABSTRACT	ii
ÖZET	iii
TABLE OF CONTENTS	iv
LIST OF TABLES	vii
LIST OF FIGURES	viii
CHAPTER 1: INTRODUCTION	1
CHAPTER 2 : REVIEW AN VOICE RECOGNITION SYSTEM	
2.1 Introduction to Voice Recognition.....	3
2.2 History of Voice Recognition.....	4
2.3 Advantages and Disadvantages of Voice Recognition.....	4
2.3.1 Advantages.....	5
2.3.2 Disadvantages.....	5
2.4 Explanation of Voice Recognition.....	5
2.5 Training Voice Recognition Software.....	6
2.5.1 The Equipment of A Good Signal.....	6
2.5.2 The Equipment of The Microphone.....	6
2.5.3 The Equipment of The Sound Card.....	6
CHAPTER 3 : STRUCTURE OF VOICE RECOGNITION SYSTEM	
3.1 Structure.....	7
3.2 Voice.....	8

3.3 Preprocessing.....	8
3.4 Feature Extraction.....	8
3.4.1 Structure Mel-Frequency Cepstrum Coefficients.....	10
3.4.1.1 Frame Blocking.....	11
3.4.1.2 Windowing.....	11
3.4.1.3 Fast Fourier Transform (FFT).....	11
3.4.1.4 Mel-Frequency Wrapping.....	11
3.4.1.5 Cepstrum.....	12
3.4.2 Vector Quantization Codebook Formation.....	12
3.4.3 Spectrogram.....	13
3.4.4 Linear Predictive Coding.....	14
3.5 Voice Classification.....	14

CHAPTER 4 : MODELLING OF VOICE RECOGNITION SYSTEM

4.1 Voice waveform from database.....	16
4.2 Voice power spectrum - Linear - Logarithmic--	
Linear and Logarithmic.....	17
4.3 Voice with and without windowing.....	19
4.4 Recognition.....	21
4.5 Time Domain and Frequency Domain of Graph.....	22
4.6 Acoustic vectors - VQ codewords from the disk.....	22
4.7 Acoustic vectors - VQ codewords from the database.....	23
4.8 Quantization Error.....	24

CHAPTER 5: EXPERIMENTAL RESULTS	26
CHAPTER 6: CONCLUSION	28
REFERENCES	29
APPENDICES	
Appendix A: Source Code	30
Appendix B: Test of Noise Variance	38

LIST OF TABLES

Table 5.1: MFCC applications and VQ applications.....	26
Table 5.2: Creatematrix2 function of use (MFCC) and Vectorquantization function of use (VQ).....	27

LIST OF FIGURES

Figure 2.1: Voice Recognition	3
Figure 3.1: General structure of voice recognition.....	7
Figure 3.2: Speaker identification.....	9
Figure 3.3: Speaker verification.....	9
Figure 3.4: Block diagram of the MFCC.....	10
Figure 3.5: Mel-Spaced Filterbank.....	12
Figure 3.6: Vector quantization codebook formation	13
Figure 4.1: Voice waveform from database.....	16
Figure 4.2: Flowchart of voice power spectrum.....	17
Figure 4.3: Linear Power Spectrum.....	18
Figure 4.4: Logarithmic Power Spectrum.....	18
Figure 4.5: Flowchart of voice with and without windowing.....	19
Figure 4.6: Hamming window.....	20
Figure 4.7: Voice frame before windowing.....	20
Figure 4.8: Voice frame after windowing.....	21
Figure 4.9: Time Domain and Frequency Domain of Graph.....	22
Figure 4.10: Acoustic vectors - VQ codewords from the disk.....	23
Figure 4.11: Acoustic vectors - VQ codewords from the database.....	24
Figure 4.12: Quantization Error (Original Signal, Quantized Speech Signal, Quantization Error Signal).....	25
Figure B.1: 0,001 of Noise variance	38
Figure B.2: 0,01 of Noise variance	38
Figure B.3: 0,10 of Noise variance	39

Figure B.4: 0,20 of Noise variance	39
Figure B.5: 0,30 of Noise variance	39
Figure B.6: 0,40 of Noise variance.....	40
Figure B.7: 0,50 of Noise variance	40

CHAPTER 1

INTRODUCTION

Various techniques have been used since the last century for the identification and recognition. These techniques are: signature, fingerprint, iris, face and voice recognition.

The aim of the thesis is the investigation of feature extraction methods for voice recognition. The voice recognition system uses the distinctive voice characteristics to verify the identity of a person. Voice recognition in recent years, still gives a series of unresolved issues, has become an interesting area of research. The speaker identification and verification can be specified. Speaker identification is compared with known speakers of the audio signal of an unknown speaker. Speaker identification and verification systems are currently used in many places. For example, It is almost hidden in some rooms, in some workplaces. It uses the voice recognition system. The other examples, information services, voice mail, database access services, banking by telephone, voice dialing telephone shopping, special security room, ... etc.

The aim of the thesis is the analysis of feature extraction methods and designing voice recognition system. To accomplish this aim the following has been done in the thesis.

In Chapter two, the voice recognition is explained. History of Voice Recognition, Advantages and Disadvantages of Voice Recognition characteristics of voice recognition systems are described.

In Chapter three, the Structure of Voice Recognition System is explained. After a brief introduction the Structure of voice recognition is described. Preprocessing is explained. Feature Extraction is using Mel-Frequency Cepstrum Coefficients. Windowing, Frame Blocking, Mel-Frequency Wrapping, Fast Fourier Transform (FFT) is explained. Vector Quantization Codebook Formation, Spectrogram, Linear Predictive Coding, Voice Classification is explained.

In Chapter four, the modelling of voice recognition system is explained. After brief introduction Voice power spectrum - Linear - Logarithmic--Linear and Logarithmic, Voice with and without windowing, Recognition, Time Domain and Frequency Domain of Graph, Acoustic vectors - VQ codewords from the disk, Acoustic vectors - VQ codewords from the database, Quantization Error is explained. The design stages of voice recognition system are given.

In Chapter five, the experimental results of voice recognition are given. The simulation was performed using Matlab package. The design stages and the obtained results are described.

In Chapter six, the general results of this work are presented under the title of conclusion.

CHAPTER 2

REVIEW AN VOICE RECOGNITION SYSTEM

In this chapter, the general information about Voice Recognition is given. The history of voice recognition is described. Advantages and Disadvantages of Voice Recognition is explained. Training Voice Recognition Software is given.

2.1 Introduction to Voice Recognition

Voice Recognition can be used to authenticate or verify the identity of a speaker.



Figure 2.1: Voice Recognition

2.2 History of Voice Recognition

Below information reflects the history of the voice recognition through the years.

2.3 Advantages and Disadvantages of Voice Recognition

In this section, the advantages and disadvantages are given.

2.3.1 Advantages

In this part, the advantages of voice recognition is given.

2.3.2 Disadvantages

In this part, the disadvantages of voice recognition is given.

2.4 Explanation of Voice Recognition

In this section, the voice recognition systems are described.

2.5 Training Voice Recognition Software

The voice recognition systems use a software to recognize the human voice.

The voice recognition software is an important way for many users. In fact, the goal is to identify the user when defining the recorded voice.

2.5.1 The Equipment of A Good Signal

The system requires a clear recording of the user's voice into the computer

2.5.2 The Equipment of The Microphone

Voice recognition software packages are used mostly by microphone support.

2.5.3 The Equipment of The Sound Card

The voice card signal is not sufficient.

CHAPTER 3

STRUCTURE OF VOICE RECOGNITION SYSTEM

In this chapter, the general structure of voice recognition systems are explained. Its basic blocks and the basic techniques used for voice recognition have been described.

3.1 Structure

The voice recognition system includes set of blocks and processes. It shows the general structure of voice recognition system in the Figure 3.1.

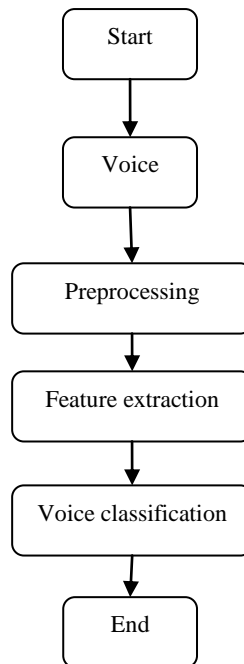


Figure 3.1: General structure of voice recognition

3.2 Voice

The purpose of voice is communication.

3.3 Preprocessing

The preprocessing stage of the voice recognition system is used to improve the efficiency in the subsequent feature extraction and classification.

3.4 Feature Extraction

In this part, the feature extraction for voice signals are given.

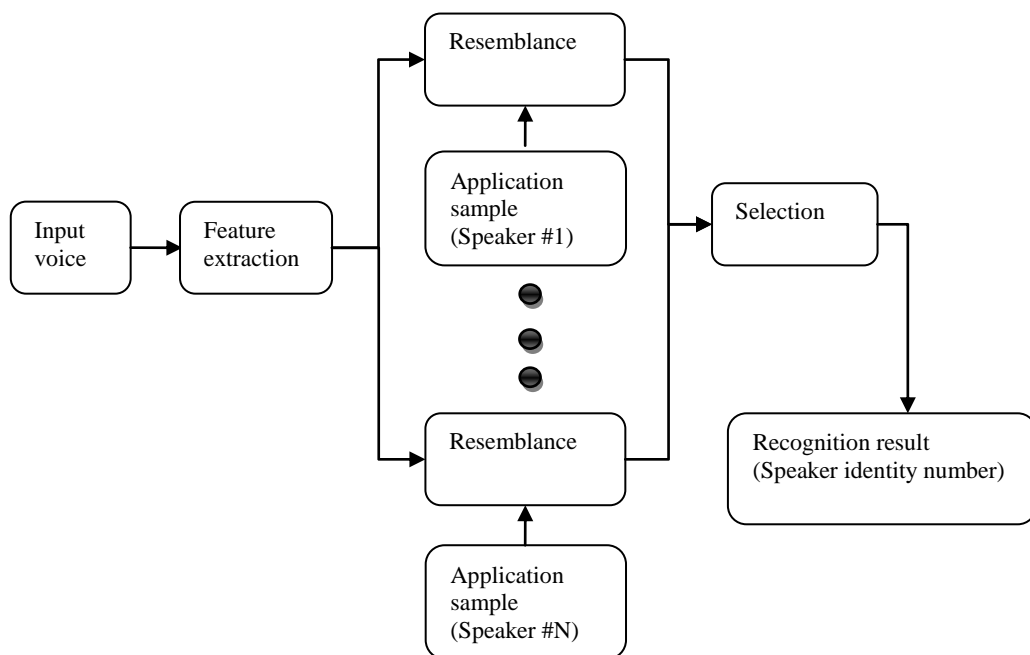


Figure 3.2: Speaker identification

Figure 3.3 shows the speaker verification.

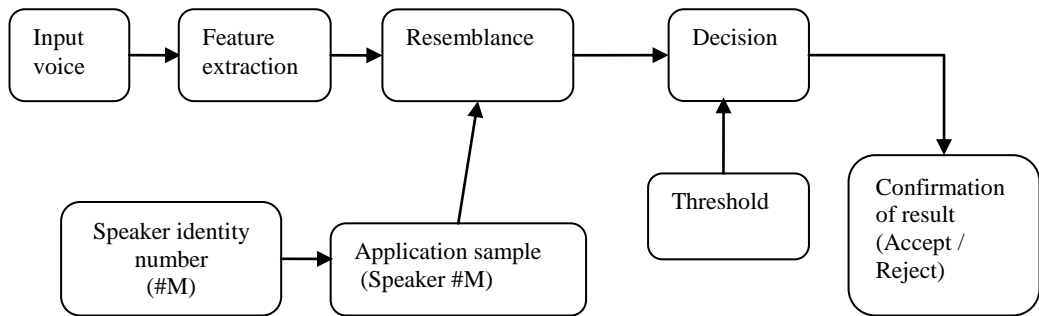


Figure 3.3: Speaker verification

3.4.1 Structure Mel-Frequency Cepstrum Coefficients

In this section, the Structure Mel-frequency cepstrum coefficients is described.

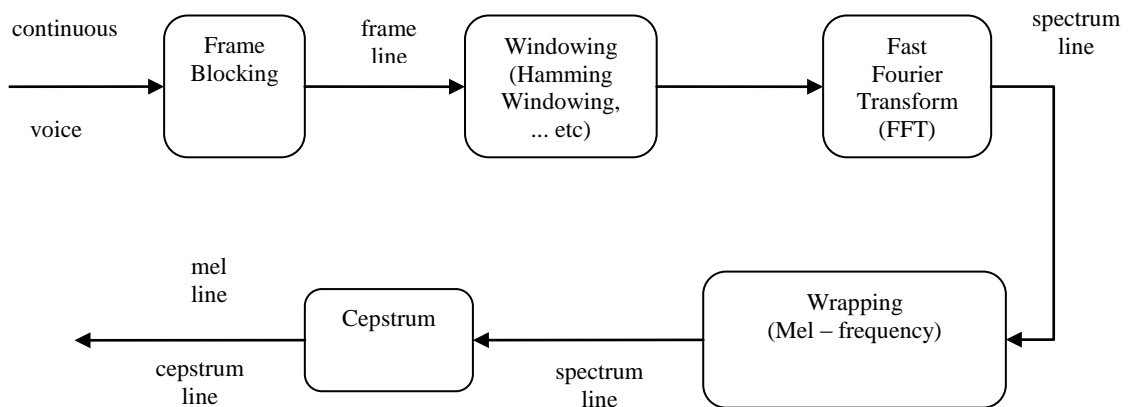


Figure 3.4: Block diagram of the MFCC

3.4.1.1 Frame Blocking

In this section, N and M frames are described.

3.4.1.2 Windowing

In this section, the windowing is described.

3.4.1.3 Fast Fourier Transform (FFT)

In this section, the Fast Fourier Transform (FFT) is described.

3.4.1.4 Mel-Frequency Wrapping

In this section, the Mel-frequency wrapping is described.

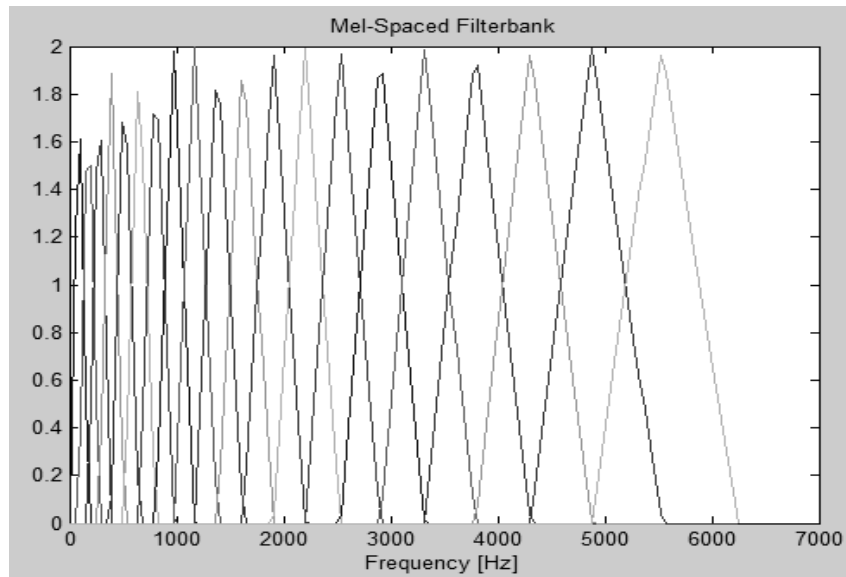


Figure 3.5: Mel-Spaced Filterbank

3.4.1.5 Cepstrum

In this section, the cepstrum is described.

3.4.2 Vector Quantization Codebook Formation

Figure 3.6 shows the vector quantization codebook formation. In the Figure 3.6, speaker 1 and speaker 2 are shown as circles and triangles.

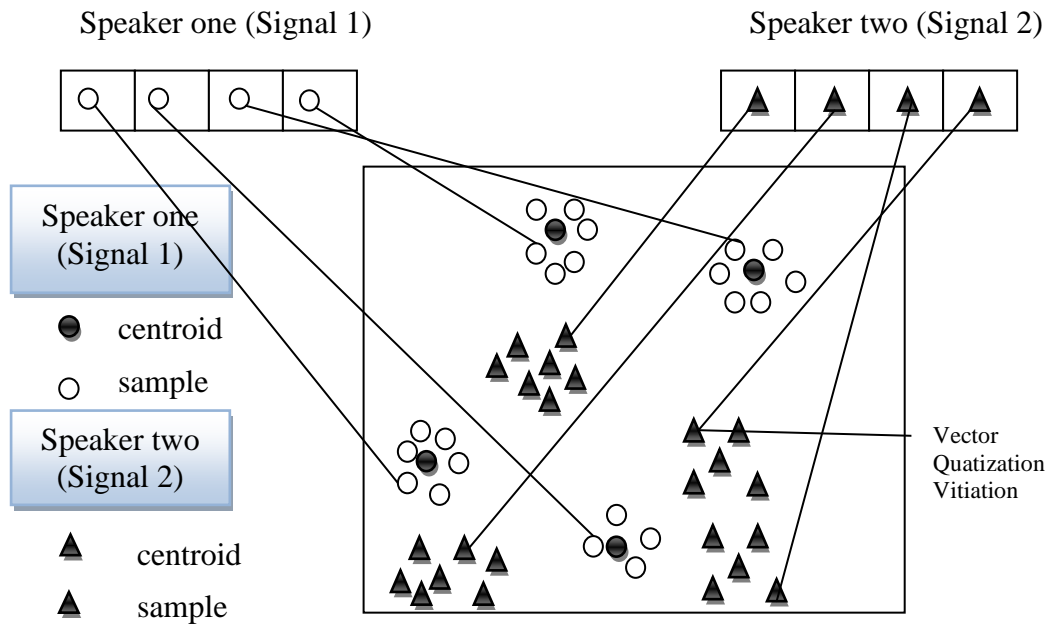


Figure 3.6: Vector quantization codebook formation

3.4.3 Spectrogram

Spectrograms are a visual presentation of the voice signal (voice recognition).

3.4.4 Linear Predictive Coding

In this section, Linear Predictive Coding is described which is also coded as LPC.

3.5 Voice Classification

The thesis uses power spectrum, windowing, vector quantization,... etc.

CHAPTER 4

MODELLING OF VOICE RECOGNITION SYSTEM

In this chapter, the modelling of voice recognition system is given. Main menu provides to control all functions of programs. The main menu is a guide for users to choose from the functions. The user can arrive to each menu from the main menu.

First of all, the system loads a new voice file from disk. Every person has a different voice. Then the user sees the voice waves in the system.

In the thesis, the designed system uses power spectrum (Linear, Logarithmic, Linear and Logarithmic), windowing, vector quantization, ...etc.

The system loads a new voice file. It opens the dialog box. Then selected .wav audio file to be uploaded to the database. The user enters voice id number on the keyboard. And then it will be loaded into systems database.

The system plays a voice file. When the user chooses the voice inside the dialog box , voice file will be played.

The system displays all voice waveforms in the database. When the user presses the button, it displays all of the voice files at the same time, in the same display. It is very useful when it comes to compare various voice file waveforms.

The system displays a voice waveform from disk. When the user presses the button, and selects voice in the dialog box. The user sees the voice waveform from the disk.

The system displays information of a voice file in the database. When the user presses the button, and writes voice number inside the database. Then the users sees Voice number, File name, Location name in the screen. Displays information about a single voice file in the sound database.

The system uses voice database information. When the user presses the button, and sees file name, location name, voice id inside all databases. It is shown on the screen.

The system has help option. If the user presses the button, the user writes the number on the keyboard, and sees the sections introduction.

The system uses delete option. It is used to delete the voice file from the database. If the user presses the button, they are given the choice of accepting or rejecting the deletion. If accepted, all of the voice files will be deleted from the voice database which will never be retrieved again.

4.1 Voice waveform from database

This section describes a voice waveform obtained from the database.

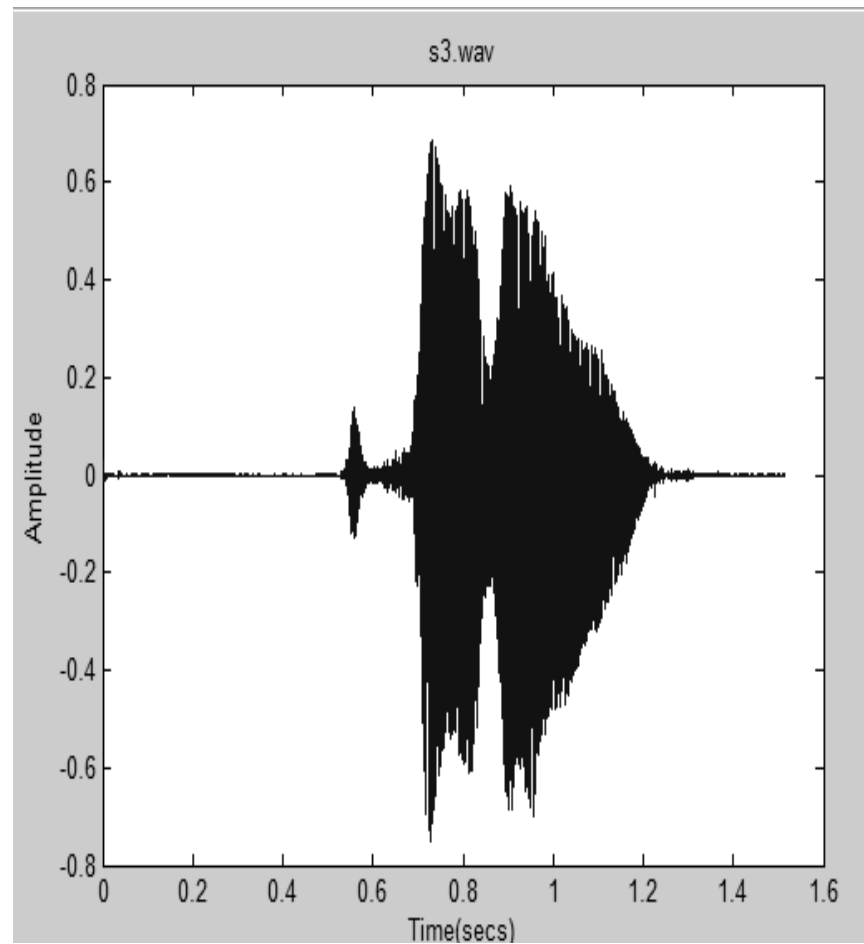


Figure 4.1: Voice waveform from database

4.2 Voice power spectrum - Linear - Logarithmic--Linear and Logarithmic

This section describes voice power spectrum - Linear - Logarithmic--Linear and Logarithmic.

Figure 4.2 shows the flowchart of voice power spectrum.

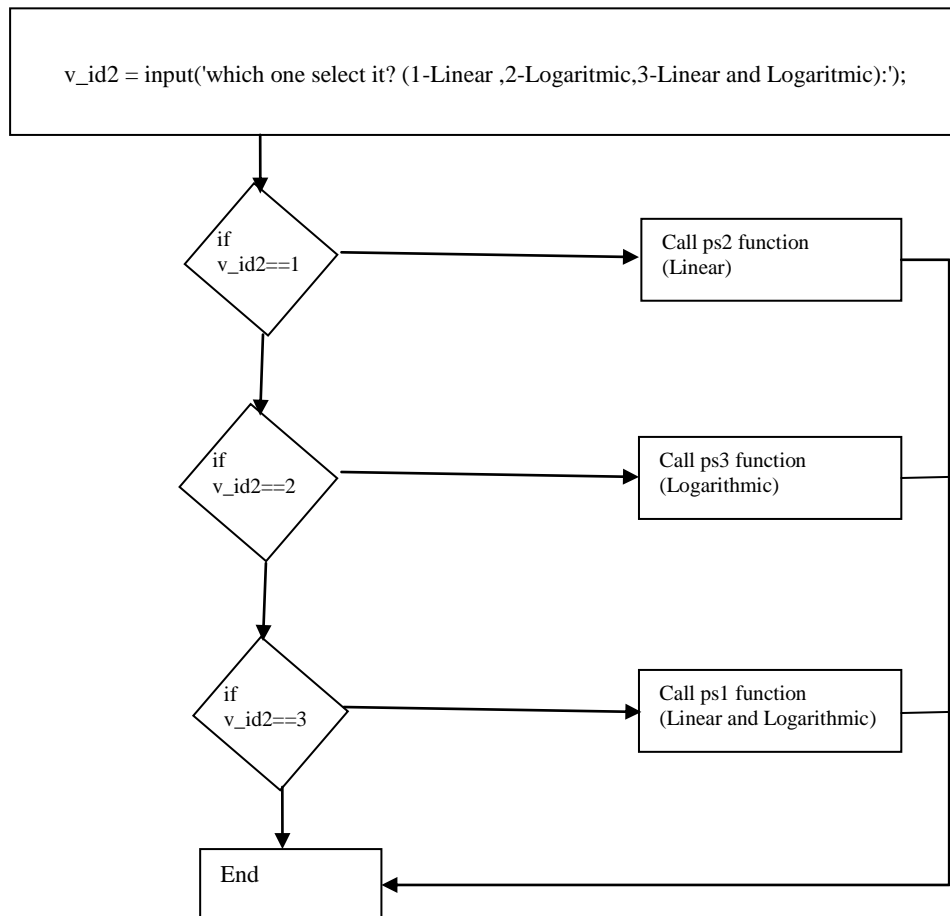


Figure 4.2: Flowchart of voice power spectrum

If the user selects the Linear, the user sees the Linear Power Spectrum in the Figure 4.3. Matlab function “ps2.m” takes an important place in this section.

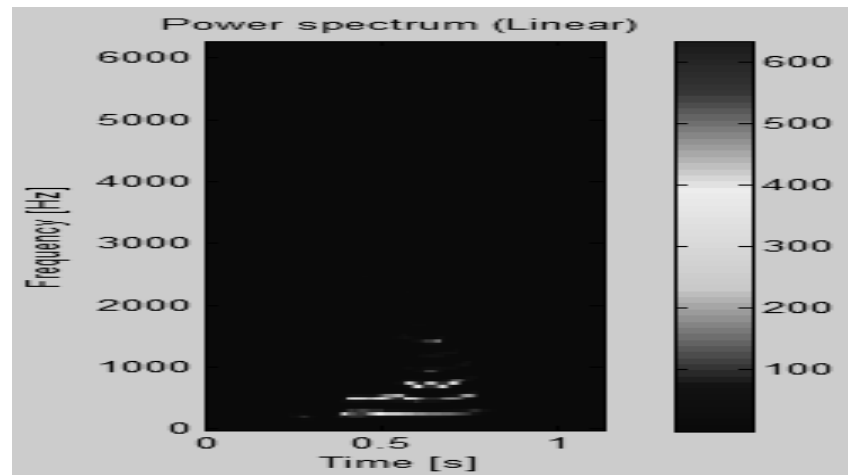


Figure 4.3: Linear Power Spectrum

Figure 4.4 shows the Logarithmic Power Spectrum. Matlab function “ps3.m” takes an important place in this section.

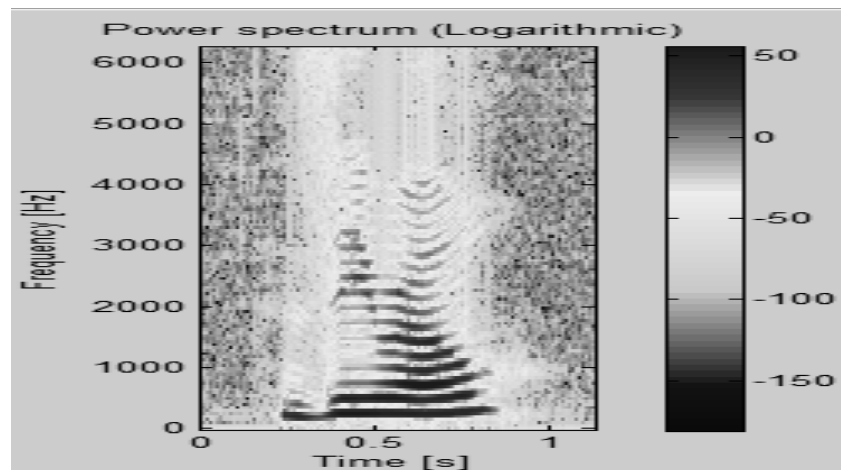


Figure 4.4: Logarithmic Power Spectrum

4.3 Voice with and without windowing

This section describes voice with and without windowing.

Figure 4.5 shows the flowchart of voice with and without windowing.

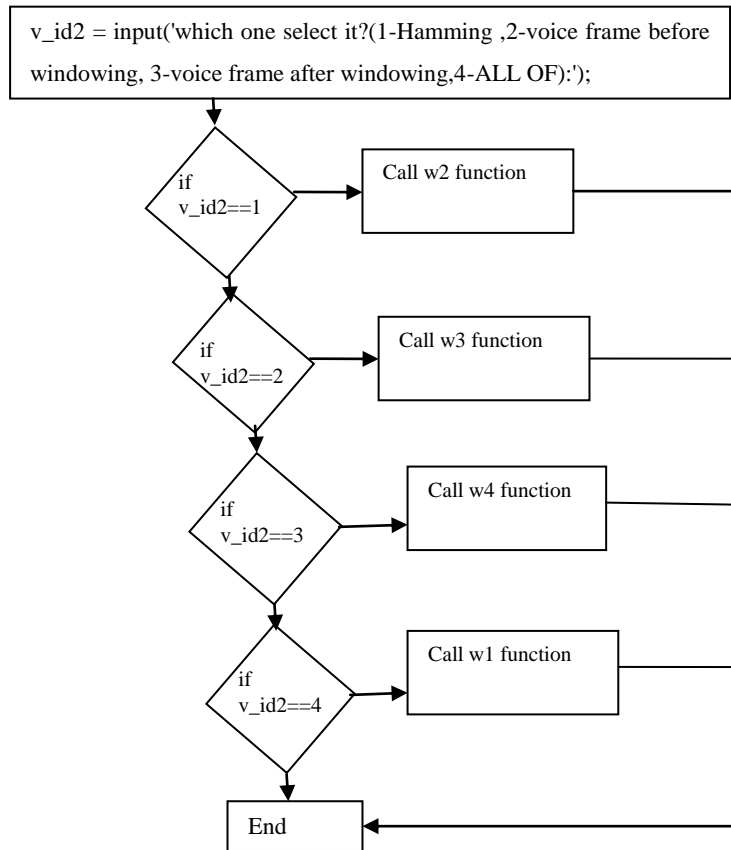


Figure 4.5: Flowchart of voice with and without windowing

The Figure 4.6 shows the Hamming window.

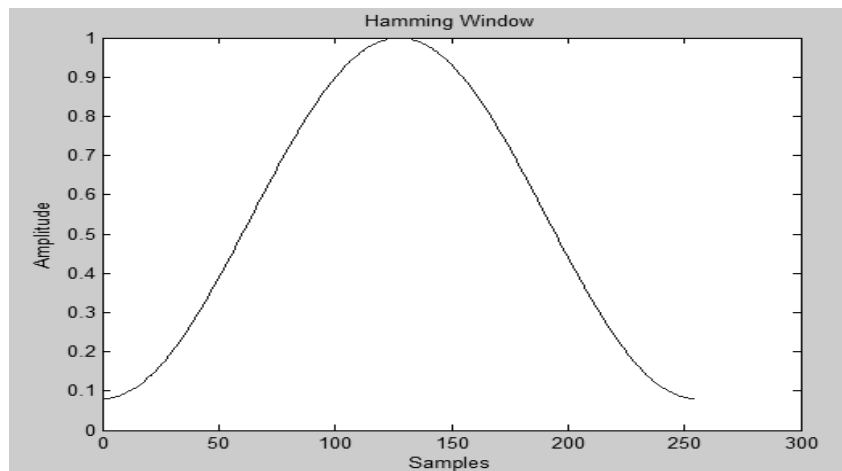


Figure 4.6: Hamming window

The Figure 4.7 shows the voice frame before windowing.

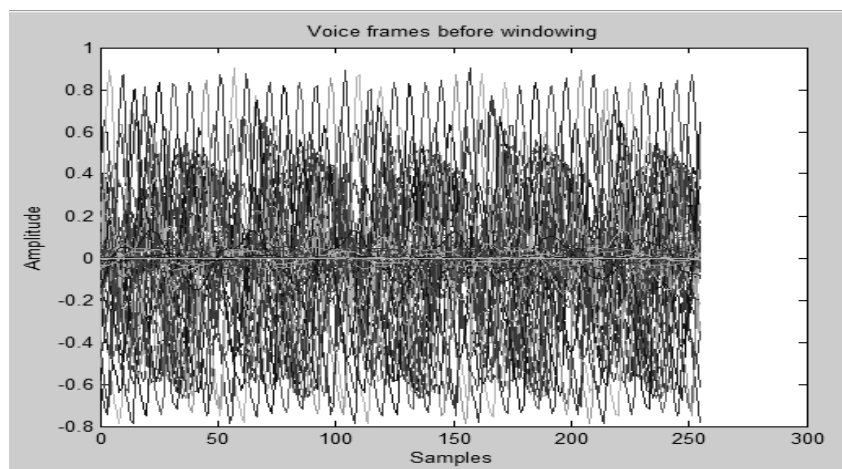


Figure 4.7: Voice frame before windowing

The Figure 4.8 shows the voice frame after windowing.

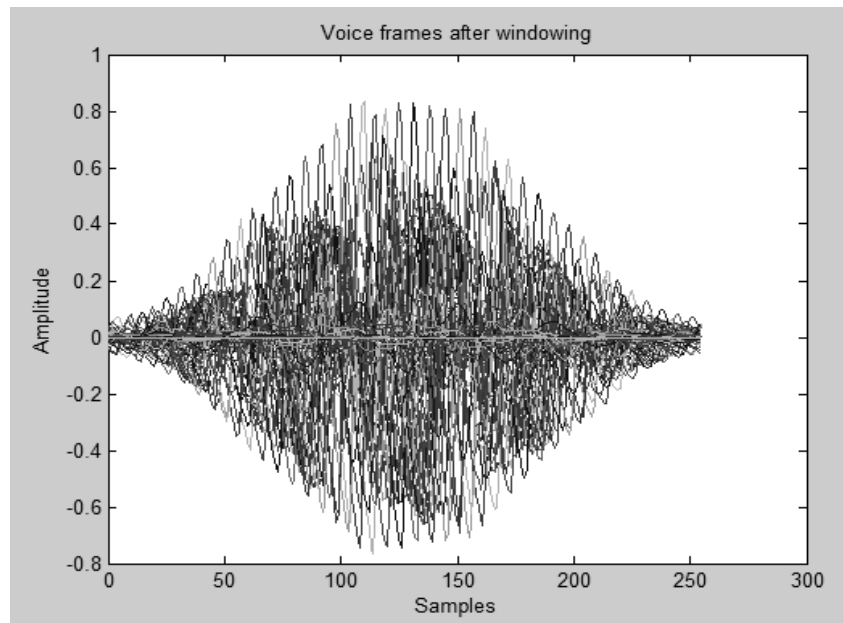


Figure 4.8: Voice frame after windowing

4.4 Recognition

This section describes recognition stage.

4.5 Time Domain and Frequency Domain of Graph

The time domain uses time series. The frequency domain uses frequency. It is important in engineering. The time and frequency domain is shown in this section.

Figure 4.9 shows the time and frequency domain. It calculates frequency domain and time domain in the system.

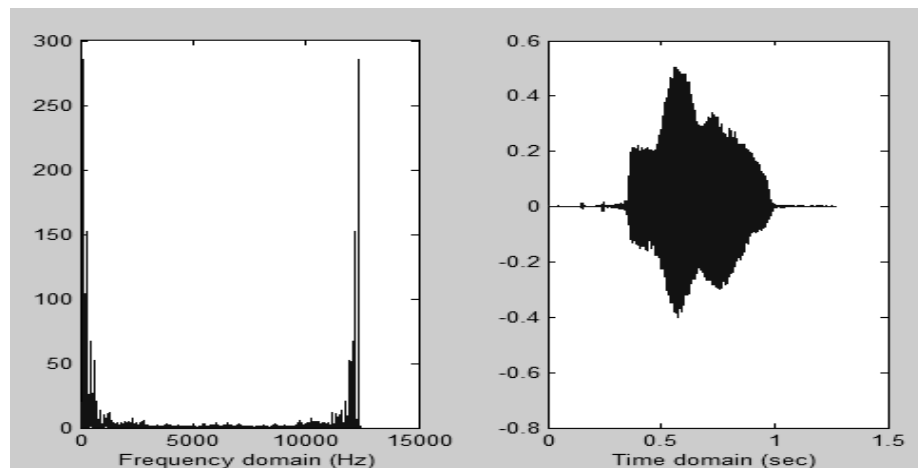


Figure 4.9: Time Domain and Frequency Domain of Graph

4.6 Acoustic vectors - VQ codewords from the disk

When the user presses the button, the user selects voice in the dialog box. The user compares 2D plot of acoustic vectors / 2D trained VQ codewords.

Figure 4.10 shows the Acoustic vectors - VQ codewords from the disk.

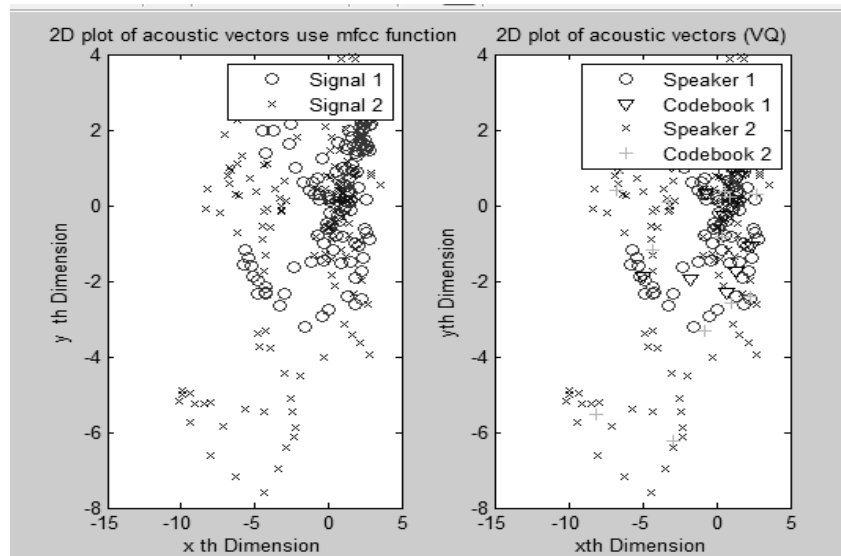


Figure 4.10: Acoustic vectors - VQ codewords from the disk

4.7 Acoustic vectors - VQ codewords from the database

When the user presses the button, the user writes the voice number from database of 2 signals. The user compares 2D plot of acoustic vectors / 2D trained VQ codewords from database 1.

The comparison is shown in Figure 4.11.

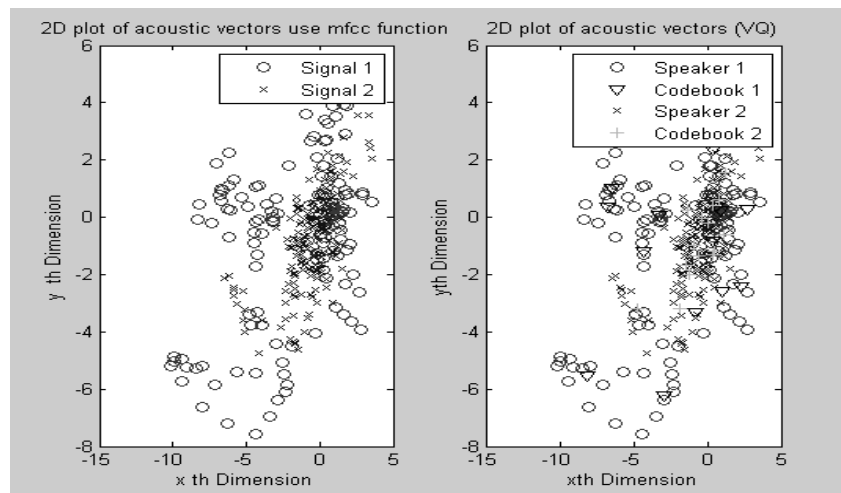


Figure 4.11: Acoustic vectors - VQ codewords from the database

4.8 Quantization Error

When the user presses (Voice 2) the quantization error button, the system calculates the original signal, quantized speech signal, quantization error signal.

Figure 4.12 shows the Quantization Error.

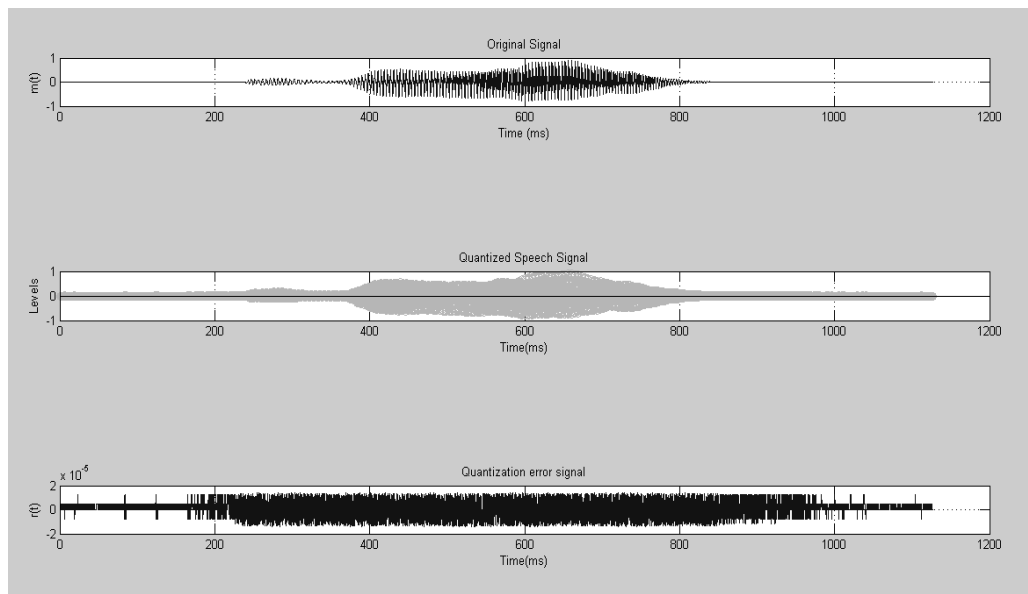


Figure 4.12: Quantization Error (Original Signal, Quantized Speech Signal, Quantization Error Signal)

CHAPTER 5

EXPERIMENTAL RESULTS

In this chapter, the results obtained from the simulation are given. These are simulation results vector quantization and MFCC feature extraction techniques. It has found that Vector quantization is better than MFCC.

The characteristics of MFCC and VQ (vector quantization) are given in Table 5.1.

Table 5.1: MFCC applications and VQ applications

MFCC Applications	VQ Applications
Identification systems is common.	Vector quantization is used for lossy data compression and correction.
They are also common in recognition, which is the task of identifying people from their sounds.	The system is made by finding the closest group with the data dimensions available.
Use Hamming window.	Use Hamming window.
Use fast fourier transform (FFT).	Use fast fourier transform (FFT).
Use mel filter bank.	Use mel filter bank.
Use discrete cosine transform (DCT).	Use discrete cosine transform (DCT), euclidean distance.
MFCC Block diagram uses frame blocking, windowing, FFT, mel frequency warpping, cepstrum.	Vector quantization uses codebook.

Table 5.2: Creatematrix2 function of use (MFCC) and Vectorquantization function of use(VQ)

Creatematrix2 function of use (MFCC)	Vectorquantization function of use(VQ)
Creatematrix2	Vectorquantization
Use Hamming Windows	Use Hamming Windows
Use fast fourier transform (FFT)	Use fast fourier transform (FFT)
Use melfbank function	Use melfbank function
Not use vq function	Use Creatematrix2 function
Not use eucliddist function	Use eucliddist function (Euclidean distances between columns of 2 matrices.)
It is advantages of Vector quantization.	But Vector quantization uses codebook. Codebook is important section in vector quantization. The users see the codebook in the system.
use signal to analyze,	e3=0,01 (Error Expectation Rate)
fs is sampling rate of the signal	d1 = vectorquantization(c1,16);
c1 = creatematrix2(s1, fs1);	use c1x16 vector
Not use error rate.	Use error rate. When the users create matrix2. Use of error rate. Then the users see voice in the vector quantization codebook.
Because only create matrix.	
But some people use only the function.	It is important in the thesis. Because vector quantization is very good function creatematrix2.

CHAPTER 6

CONCLUSION

The research works on voice recognition have been analysed and the structure of voice recognition system is presented. It was shown that one of important problem in voice recognition is the feature extraction. The performance of voice recognition system depends on the results of feature extraction blocks. Therefore different feature extraction methods are considered. MFCC and vector quantization techniques are selected and then simulated using Matlab package. For simulation purpose voice database is organized and then the selected feature extraction methods have been tested on this database.

The voice recognition system was designed using Matlab programming. Matlab programming is a good environment to develop different feature extraction method. In the thesis, speaker identification and verification systems are used in a successful way. Linear and Logarithmic power spectrum have been examined. The modules Hamming windowing, voice frame before windowing, voice frame after windowing has been examined. It has been observed that they have different characteristics from each other.

The considered techniques are tested using voice signals that are stored in the database. All unknown speakers were identified as a result of scanning a few seconds.

Voice recognition systems can be improved in the future by considering the following conditions:

- A larger speech database can be created and can be studied in more detail. But the voice database used for the thesis was sufficient.
- The voice recognition system can be done by raising the ambient noise and can still be voice identified.

APPENDIX A

SOURCE CODE

```
section = 15;
selection = 0;
% The Main Menü
% -----
%while selection ~= section,
    selection = menu('VOICE RECOGNITION SYSTEM',...
        '1-Load a new voice',...
        '2-Play a voice',...
        '3-Voice waveform from database',...
        '4-All voice waveforms in database ',...
        '5-Voice waveform from disk',...
        '6-Voice power spectrum - Linear - Logarithmic--Linear and Logarithmic',...
        '7-Voice with and without windowing',...
        '8-Information of a voice file',...
        '9-Recognition',...
        '10-Voice database information',...
        '11-Help',...
        '12-Delete voice database',...
        '13-Exit',...
        '14-Time Domain and Frequency Domain of Graph',...
        '15-Acoustic Vector-VQ Codewords-Quantization Error');

% SECTION 15 - Acoustic Vector-VQ Codewords-Quantization Error %
%-----
if selection == 15
    run menu6;
end
%-----
```

```
% SECTION 14 - Time Domain and Frequency Domain of Graph %
```

```
%-----
```

```
if selection == 14
```

```
    [fname2,pname2] = uigetfile('*.wav');
```

```
    [dt1, Fs, nbits] = wavread(strcat(pname2,fname2));
```

```
    sound(dt1, Fs)
```

```
    t2=(1:length(dt1));
```

```
    t = (t2) ./ Fs;
```

```
    subplot(1,2,2)
```

```
    plot(t, dt1)
```

```
    xlabel('Time domain (sec)')
```

```
    Y = fft(dt1);
```

```
    t4=length(dt1);
```

```
    df1 = Fs / t4;
```

```
    t3=(1:length(Y));
```

```
    f = (t3) * df1;
```

```
    ns1 = length(dt1) / 1;
```

```
    subplot(1,2,1)
```

```
    plot( f(1:ns1), abs(Y(1:ns1)) )
```

```
    xlabel('Frequency domain (Hz)')
```

```
end
```

```
%-----
```

```
% SECTION 6 - Voice power spectrum - Linear- Logarithmic
```

```
% -----
```

```
if selection == 6,
```

```
    clc;
```

```
    load('C:\Users\Samsung\Desktop\Projem\Sounds.dat','-mat');
```

```
    [fname,pname] = uigetfile('*.wav','Select a new voice file');
```

```
    [y, Fs, nbits] = wavread(strcat(pname,fname));
```

```
    clc;
```

```

    fprintf('CHOICE 1 is Linear, CHOICE 2 is Logarithmic, CHOICE 3 is Linear and
Logarithmic\n');
    v_id2 = input('which one select it? (1-Linear ,2-Logaritmic,3-Linear and
Logaritmic):');
    disp(' ');
    if (v_id2==1)
        c = ps2(y, Fs); % Call Function ps2
    end
    if (v_id2==2)
        c = ps3(y, Fs); % Call Function ps3
    end
    if(v_id2==3)
        c=ps1(y, Fs); % Call Function ps1
    end
end
% -----

% SECTION 7- Voice With And Without Windowing
% -----
if selection == 7,
    clc;
    load('C:\Users\Samsung\Desktop\Projem\Sounds.dat','-mat');
    [fname,pname] = uigetfile('*.wav','Select a new voice file');
    [y, Fs, nbits] = wavread(strcat(pname,fname));
    clc;
    fprintf('CHOICE 1 is Hamming, CHOICE 2 is voice frame before windowing,
CHOICE 3 is voice frame after windowing\n');
    fprintf('CHOICE 4 is ALL OF\n');
    v_id2 = input('which one select it?(1-Hamming ,2-voice frame before windowing, 3-
voice frame after windowing,4-ALL OF):');
    disp(' ');
    if (v_id2==1)
        c = w2(y, Fs); % Call Function w2

```

```

end
if (v_id2==2)
    c = w3(y, Fs); % Call Function w3
end
if (v_id2==3)
    c = w4(y, Fs); % Call Function w4
end
if (v_id2==4)
    c = w1(y, Fs); % Call Function w1
end
end
% -----

% SECTION 13 - Exit Of Program
% -----
if selection == 13
    fprintf('End of the program.\n');

end
% -----

%SECTION 11 - Help
% -----
if selection == 11
    clc;
    fprintf('-----HELP OF BUTTON(1-15)-----\n');
    s = input('ENTER THE BUTTON AND LOOK HELP US(1-15): ');

    if s==1
        fprintf('\n');
        clc;
        disp('SECTION 1: Loads a new voice file from disk..');
    end
end

```

```
disp('Dialog box opens');
disp('Then selected .wav audio file to be uploaded to the database.');
```

disp('and Id number is entered on the keyboard. And then It will be loaded into your database.');

```
end
```

```
if s==2
fprintf('\n');
clc;
disp('SECTION 2: Plays a voice file from the disk.');
```

disp('It appears a dialog box where you can select the user directory');

```
disp('and the filename of the voice file to be played');
end
```

```
if s==3
fprintf('\n');
clc;
disp('SECTION 3: Displays a voice waveform from the database.');
```

disp('The file name of the file to be displayed and it is selected by the user from the dialog box.');

```
end
```

```
if s==4
fprintf('\n');
clc;
disp('SECTION 4: Displays all of the voice files at the same time in the same display.');
```

disp('It is very useful when it required to compare various voice file waveforms.');

```
end
```

```
if s==5
fprintf('\n');
clc;
disp('SECTION 5: Displays a voice waveform from the disk.');
```

```
disp('The filename of the file to be displayed and it is selected by the user from a  
dialog box.');
```

```
end
```

```
if s==6
```

```
fprintf('\n');
```

```
clc;
```

```
disp('SECTION 6: If you select the voice and then select the alternative');
```

```
disp("");
```

```
disp('Linear - Logarithmic--Linear and Logarithmic');
```

```
end
```

```
if s==7
```

```
fprintf('\n');
```

```
clc;
```

```
disp('SECTION 7: You will get the section you choose , and you will see it.');
```

```
disp('Displays the voice with and without windowing.');
```

```
end
```

```
if s==8
```

```
fprintf('\n');
```

```
clc;
```

```
disp('SECTION 8: Displays information about a single voice file in the sound  
database.');
```

```
end
```

```
if s==9
```

```
fprintf('\n');
```

```
clc;
```

```
disp('SECTION 9: Performs the recognition function ');
```

```
disp('And the speaker is identified among several sound files.');
```

```
end
```

```
if s==10
```

```
fprintf('\n');
clc;
disp('SECTION 10: Displays information about all of the voice files in the sound
database. ');
disp('There are no sections for the user to select. ');
end

if s==11
fprintf('\n');
clc;
disp('SECTION 11: You can press number and displays this HELP text. ');
end

if s==12
fprintf('\n');
clc;
disp('SECTION 12: It is used to delete the voice file database. ');
disp('When this section is selected the user is given the choice of accepting or
rejecting the deletion. ');
disp('If accepted, all of the voice files will be deleted from the voice database. ');
end

if s==13
fprintf('\n');
clc;
disp('SECTION 13: It is exit the program ');
end

if s==14
fprintf('\n');
clc;
disp('SECTION 14: You see Time Domain and Frequency Domain of Graph. ');
end
```



```

if s==15
    fprintf('\n');
    clc;
    disp('SECTION 15: You see Accustic Vector-VQ Codewords-Quantization Error');
    disp('It compares 2D plot of accustic vectors / 2D trained VQ codewords');
    disp('It compares 2D plot of accustic vectors / 2D trained VQ codewords from
Database 1');
    disp('Quantization Error - value of SNR is shown on the screen');
    else if s>=15 msgbox('You pressed the wrong number. You have to press a number
of up to 1-15.');
```

end

end

end

% -----

% -----End Of The Program-----

APPENDIX B

TEST OF NOISE VARIANCE

When the noise variance is increased, voice signal is disrupted.

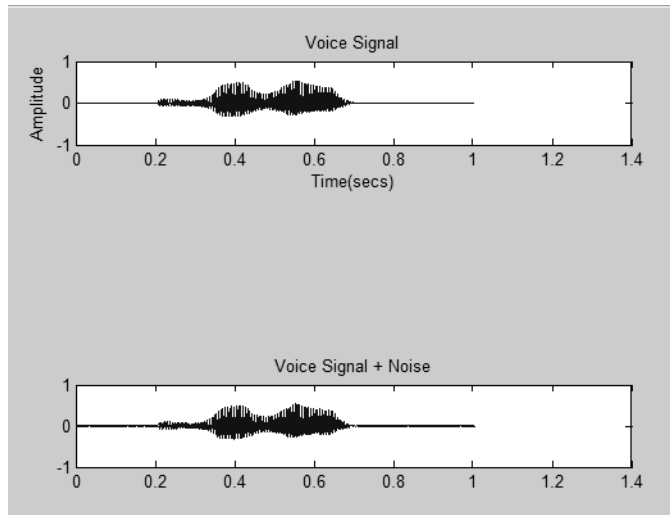


Figure B.1: 0,001 of Noise variance

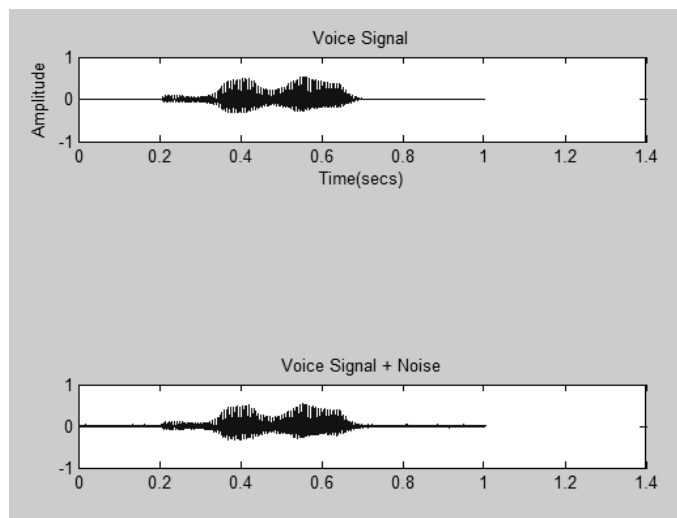


Figure B.2: 0,01 of Noise variance

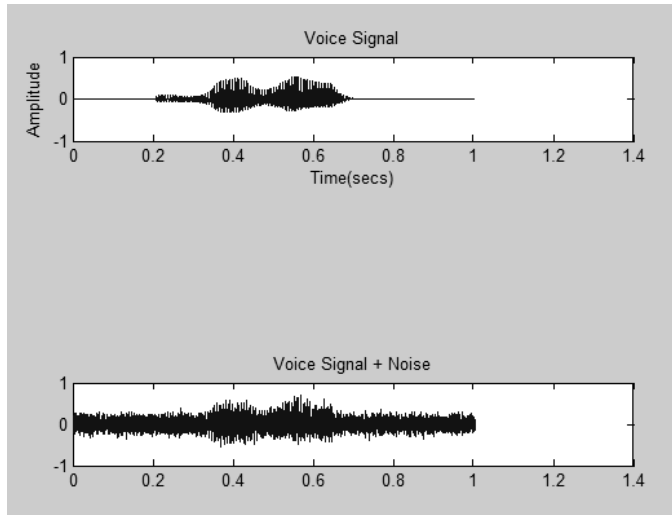


Figure B.3: 0,10 of Noise variance

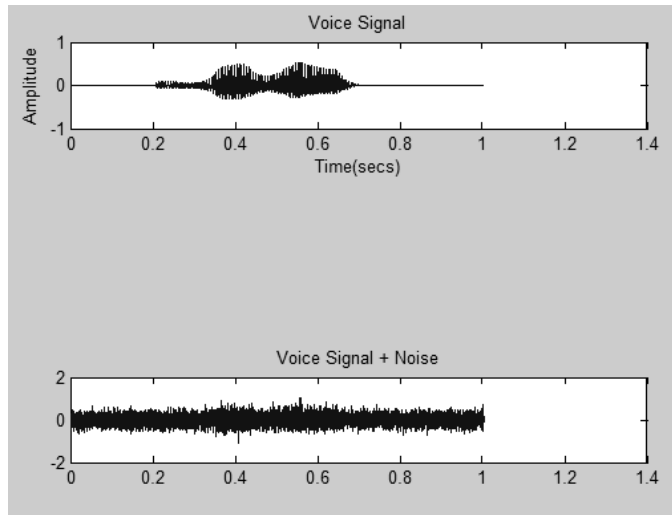


Figure B.4: 0,20 of Noise variance

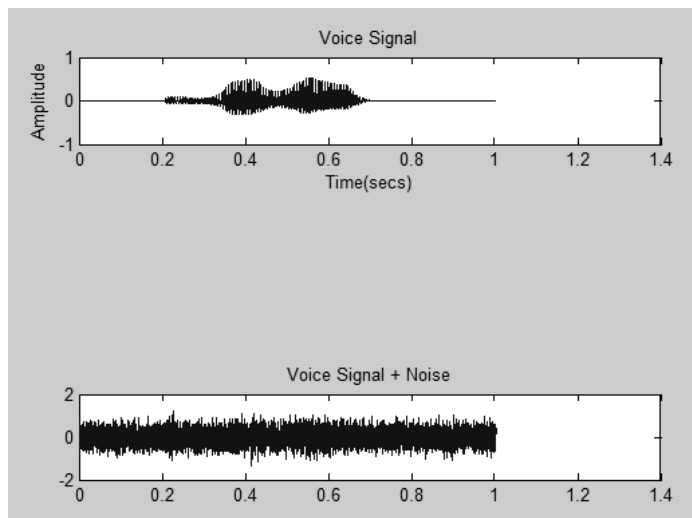


Figure B.5: 0,30 of Noise variance

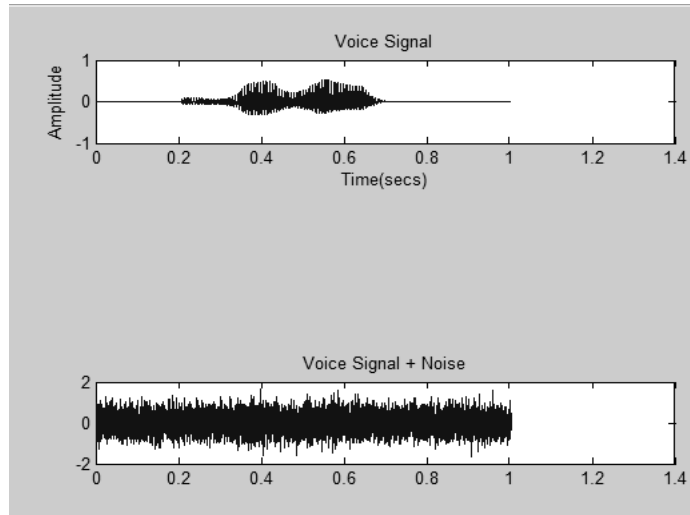


Figure B.6: 0,40 of Noise variance

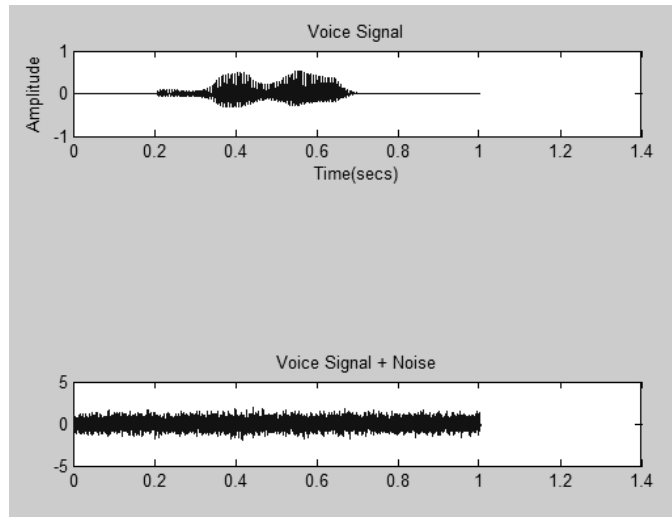


Figure B.7: 0,50 of Noise variance