Automation of Eyesight test Based upon Speech Recognition

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Automation Of eyevision test based upon speech Recognition

CERTIFICATE OF CORRECTIONS & APPROVAL

Certified that work contained in this thesis titled “Automation of eye vision test based upon speech recognition”, carried out by Muhammad Junaid, Muhammad Talha, Abid Hussain, Noor fatima under the supervision of Dr: Shibli Nisar for partial fulfillment of Degree of Bachelors of Electrical Engineering, in Military College of Signals, National University of Sciences and Technology, Islamabad during the academic year 2019-2020 is correct and approved. The material that has been used from other sources it has been properly acknowledged / referred.

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Abstract

Different methods and techniques are used by doctors/practitioners throughout the world to Perform eye vision test. Among all the methods Snellen chart is mostly used in all regions for Eye vision test. The major drawback of this old technique includes that there is no symmetry in The process, that is, the test is conducted in different steps in different environment by different Doctors which causes variation in results.

Moreover, errors are caused by Humans / practitioners while performing the test and these errors may be due to workload, Mood or health of the doctor which may affect the final result of the test by a significant Margin. Acuity test may also get effected due to increased number of patients operated by a Single doctor. All these factors and possibly more can affect the test and lives of the patients.

Snellen chart based method is the most common method used in the world but these methods Are vulnerable to many errors that including environmental and human errors. Moreover, these Tests are expensive to take, that is, the cost of equipment, qualified doctors and their high pays, Proper lightened environment and so on. Other common problems include long awaiting Appointments, unavailability or non-seriousness of practitioners etc. Problems such as these Mostly arise in remote areas where there is lack of supervision and balance of the system.

Key Words: *Snell chart, acuity test.*

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CHAPTER 1

INTRODUCTION AND BACKGROUND

CHAPTER 1: INTRODUCTION

Communication is one of the most important role that an individual play while living in a society. This communication has existed from the beginning of time and till this date, Advancement is being made into it. Communication between Individuals has evolved from Gestures and movement to speech and more.According to Latest reports of World health organizations around the globe eyesight problems are spreading and 2 billion people are now suffering from it.

Eyesight problems are now common part of our society. People now a days have busy life style and due to which they do not have time to go to doctors. The methods being used in hospitals involves Snell chart and patient are tested through it.The whole set up has few draw back. The result came out of as a result of this test are less precise. The results are almost 80% accurate.

To increase the accuracy of this test, its automation was necessary. The selection of this project means a lot to us as it involves signal processing, speech recognition, Fourier transforms, inverse Fourier transforms, windowing etc. .These steps made it special to work on it.The progress in the field of signal processing and speech recognition has made a lot in the medical science field .The automation of eyesight test will going to help people in order to improve their test results.

1.1 History of signal processing

The history of signal processing starts from 17^{th} century and then in the mid-19^{th} century these signals processing is refined and a new era in signal processing starts. In 1948 Claude Shannon write a paper on theory of communication which became the start of the peak of signal processing and development of new instruments involving signal processing techniques. In between 1960 to 1970 digital signal processing was evolve and starts to peak up .Digital signal processing becomes the back bone of the modern communication systems. Digital signal processing starts it work to peak in 1980 when Digital signal processor chips were invented. These were the chips used to process digital signals. Before the development of digital processor chips many other technologies were used specially the Bit slice chips. There were AMD 2901 bit slice chips which were most popular then. Different AMD were design according to specific applications.
These slice chips also include Peripheral Multiplier chips which provides the MAC functions.
The Digital processor chips have following functions

1.1.1 Functions of Digital processor chips

Audio signal processing
Audio signal processing is widely used and in audio signal processing digital processor chips produced revolution and audio signal can be process with accuracy and were used in distance communication

Telecommunication
Telecommunication stands upon the signals processing especially digital signal processing. Digital signal processor chips were used in telecommunication devices and were widely used.

Digital image processing
Digital image processing is a new technology and digital processor chips also produce revolution in this field .Digital image processing involves having a digital image and then process it through using a algorithm. Digital image processing has vast advantages over analog image processing. In digital image processing there is no noise and distortion build up. Digital processor chips are used for these purposes.

Radar and sonar
Radar and sonar also used these digital processor chips to locate angle velocity and location of any object. There is a antenna which send out electromagnetic waves specially radio waves, These waves are reflected back from Object aircraft ships etc. and then transmitter antenna also has a receiver antenna which calculate velocity, angle and location of the object. Digital signal processor chips are used in it

Speech recognition
Digital signal processor chips also made progress in speech recognition world. In speech recognition a specially made algorithm is used which take the input signal in voice and then after feature extraction it compare it with the Database and classification is done through this steps.

Another chapter of signal processing called electronic signal processing was invented in 1970.The invention of MOSFET the Mos. integrated chips were also being used for signal processing. The first microprocessor and microcontroller were Mos. integrated chip.

Data compression and linear predictive coding was also invented into 1966.It was a big step in the world of Digital signal processing .Speech synthesizer DSP chip were invented in late 1970's and Data compression and LPC were the basis of speech synthesizer.

In early 1970 discrete cosine transforms were invented and since then DCTs are being used in DSP chips. Most of the world companies are now developing DSP chips based upon discrete cosine transforms. These chips are used in Encoding and decoding of differ application .DSP chips are designed according to specific purpose and are vastly being used in different systems.

In 1976 speak and spell concept of speak and spell was given and later after 2 year first technological speak and spell chip was invented, it uses the LPC for first time I the history of digital signal processing.In1978 AMI invented a chip for Motolla 6800 called S2281, it was a signal processing pherperial .It was DSP chip and it can perform many operation under the signal command. It has to be initialized by the Host and it was not famous in the market.

In 1979 Intel released a 2920 chip for analog signal processing, it has an internal analog to digital and digital to analogy convertor. It was also not very famous in markets. µPD7720 was invented in 1980 and it was a biggest success in digital signal processing and it was used for voice band applications.

- Our project as part from digital signal processing and signal processing also includes speech recognition.

- Speech recognition also called as automatic speech recognition has a vast application in Digital signal processing. It also used in research work of computer science and information technology.

- Speech recognition involves the training of the system .The system is automatically trained to a specific person voice and then by speaking it through microphone it converts our speech to text and compare it with the store individual’s voice and give us the required results.

- Speech recognition is basically identifying the speaker words don’t matter the waveform of the signal being generated is compared with the waveform store in the trainer.

- The applications of the speech recognition are called a voice over interface, Which means a interface is designed according to a Specific voice sample and then a system is trained according to it and then the signal generated by other speaker is compared with it and result are draw from it.

1.2 Objective

Our project has a vast application in both medical engineering and in academic field given below

1.2.1 Academic objective

The academic objective of the project which should be completed at the end of this project will be

- Understanding of the speech recognition of a specific system

- Understanding of the Algorithm involved speech recognition
- Will enable students to learn Mat lab as most of the Algorithms are built in the Mat lab so student are going to learn the Mat lab for the execution of their Algorithm and will be able to understand Mat lab ,which will help them in Future
- Most of our coding is done through Python and student has to take a online python coding courses so they can understand Python. Our speech recognition Algorithm is executed through python and student can learn it and can explore the world of coding using python like AI Used robotics.
- Student can also explore in the field of the Web development
- Student will also be able to developed its own GUI applications
- Students will be able to learn Scientific and numeric problems in coding using python
- Student will be able to develop their own Business using python through freelancing and can generate a lot of revenue
- These is involvement of Signal processing as well as digital signal processing Student will be able to understand signal processing steps. They can have a bright future in the digital signal processing field.

1.2.2 Medical Objective

This project has objective in the Medical field. As there is a lot of development in the medical engineering field. This will be also a major stone in the development of the Medical engineering field .People with eyesight problems are facing problems and these problem will be resolve at the end of this project. The medical objectives of this project are given below

- As eye sight test are usually are very expensive and people with Less money are unable to undergo this test .The objective also involves the development of a system which will automatically check eyesight and this system will be called as Cost efficient system. As Now people with less money and less income can undergo this test.
- AS People are very busy these days So they check of eyesight involves a special appointment from a doctor and a specific person Has to take out a whole day time to go to doctor and then undergo through this test.The objective is also to develop

a system with less delays and schedules

- As in the eyesight test there is also a large number of staff members are required ,when this system is develop then there is less number of staff member is involves as there system is automated

1.3 Scope

- This project has a lot of scope in medical field.
- It is an efficient, time saving system and once automation is done then so this system is going to be an efficient automated system.
- This project is using a Snell chart and we are going to automate the whole eyesight test and Snell chart is done through an interface, this makes this system a lot accurate and efficiency of the system can be increased.

1.4 Deliverables

The deliverable of this project are given below

1.4.1 Interface

An interface will be at the end of the project will be presented which is final product of our project .The interface involves the automation of of Snell chart, the whole Snell chart is Implemented through this interface.

1.4.2 Third party control

There also be a third party to control, a staff member has a manually to operate .First a patient is sit on a chair at the distance of 6 meter(20ft).The staff member going to connect the microphone and then provided to the patient and word appears on screen and patient speak and system automatically get the result. The doctor is going to observe all the test by its self and check the accuracy of system.

1.5 Background.

A voice signal contains various bits of information such as a persons' gender, age, accent, Emotion, identity etc. Because of versatility in an environment, the accurate detection of Phonetic signals has emerged as a topic of great interest to researchers. The purpose of Speech processing is to extract the desired useful information from the signal and further Process it for recognition purposes.

Our proposed model works through a trained classifier on a database of isolated words fed into The system which will take utterance of the isolated words as test signal and perform extraction Of Mel frequency cepstral coefficients (MFCC) and perceptual features from it. Extracted Features are fed into the trained classifier which compares the uttered word with the one Displayed on the screen.

Appropriate speech recognition algorithms will be embedded into the System which will help in successfully identifying the uttered words as shown in Figure 1. The main idea is to use a computer or projector as a screen which will project different Alphabets in a predefined order as followed in the Snellen chart. The test subject will have to Look at the screen to see the displayed word and then utter the word aloud in to a microphone In front of them.

Our algorithm is then used to correctly detect the alphabet uttered by the test Subject. After performing background analysis on the uttered speech by the test subject, the System will then display the next alphabet. The next alphabet depends on the test subject's Previous answer: if the last uttered word was correctly recognized, then the next alphabet will Be smaller in font size as compared to the previous one. But, if the last uttered word was not Recognized correctly then the alphabet will change but the font size will remain the same.

There are few stands domain/charts are being used up .Our project is based upon the Snell chart which work for normal people and there are other domain which should be distinguished in order to understand the domain of our project

1.5.1 Snell chart:

- Most widely used chart to measure the eyesight is Snell chart
- .It has 11 rows and from top to bottom of the chart the size of the words are decreasing and on right hand side their respective eye sight is written
- .It was designed by Dutch eye doctor Hermann Snell and after his name it was called as Snell chart.
- The words were placed in 5x5 Grid.
- The distance according to Snell chart from the Chart should be 6metre (20ft).In many clinics where there is shortage of space 3 meter(10ft) is considered as standard distance.
- The words on the chart are called as the optotype.The standard of visual acuity is set when the eye subtended a 5minutes of arc. The red line of 20/20(6/6) means that a normal person can read this letter with angle subtend of 5 minutes arc and the thickness between

the words is 1 minute subtend arc. The green line is under the 20/30 which represent almost 5.5 eyesight.

- In most countries the letter "E" which is 20/200 or 6/60 is called legally blind. If a person could see 20/200 with glasses then respective is not considered as legally blind as it is myopia.

Snell chart

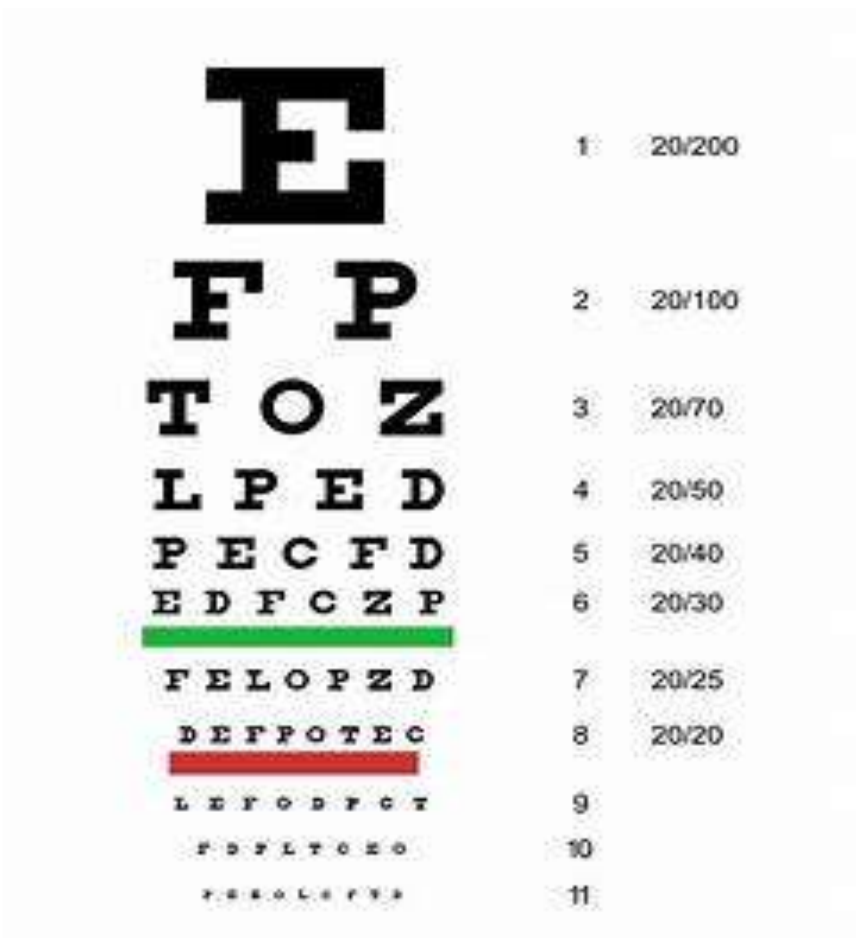


Figure 1.1

1.5.2 Trumbling E

- The Trumbling E chart is the used to measure the eyesight of Small children’s having eyesight problem.
- IT is also used for Adults having diffulty in reading or for illiterate people.
- In Trumbling E ,it is called so because it has all words that are E .E is rotated along different sides and different sides.Tha patients are required to sit at a distance of 6meter(20ft)or 3 meter(10ft) both are excepted. The doctor will going to put its stick on words one by word Start from biggest to small and patient under observation has to tell whether it is rotating left or right up or down.
- There are only 9 rows of rotated E .Patient only have to tell the direction where it is alien.
- The eyesight number are mentioned as 1 to 9 along with acuity ratio.

Trumbling E

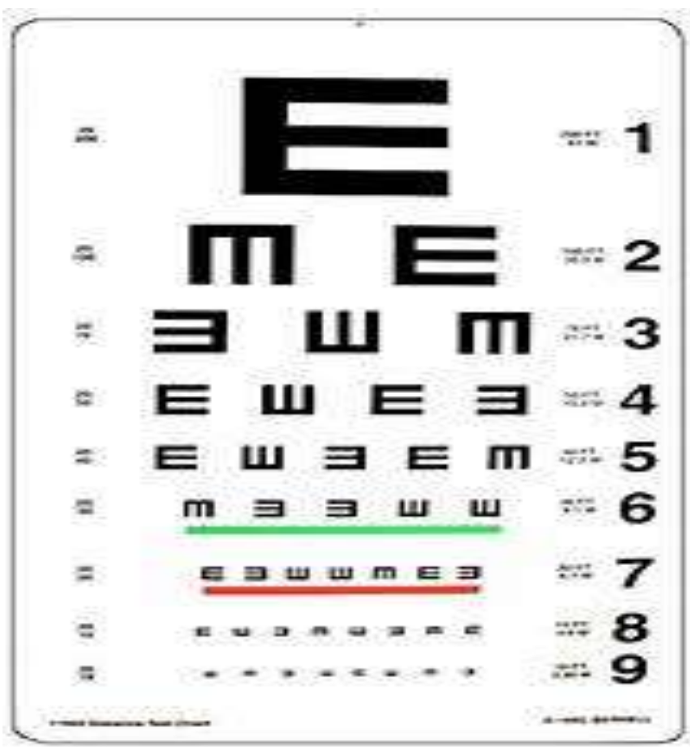


Figure 1.2

1.5.3 Landolt C

- It was developed with a lot of intelligence by Landolt
- .It is used to check the Eyesight test of small children's, mute people as well as illiterate people.
- It has rotating C also called rotating ring.
- The gap in C is rotated and patient are sit at distance of 6meter(20ft) or 3meter (10ft) .the doctor with a stick can point at the specific letter starts from biggest to smallest.
- The patient under observation just have to give direction of movement of open end of C .The eyesight is decided on this basis.
- It has 11 rows same as that of Snell chart and According to size the Respective sixe is placed along the number.

Landolt C

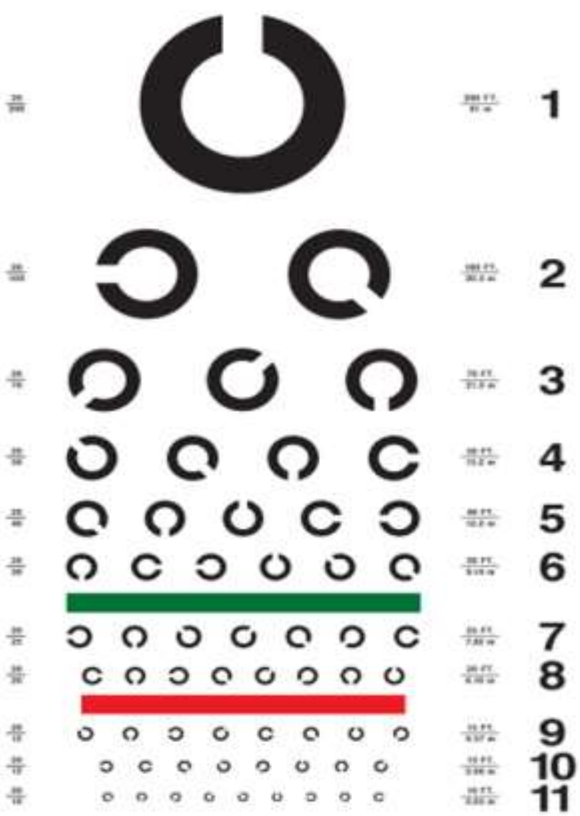


Figure 1.3

1.5.4 ETDRS

- ETDRS stand for early treatment diabetic retinopathy study,
- It is used for same purpose as Snell chart is used.
- The accuracy of ETDRS has been increased by different researches and into upcoming days ETDRS will be used for higher accuracy.

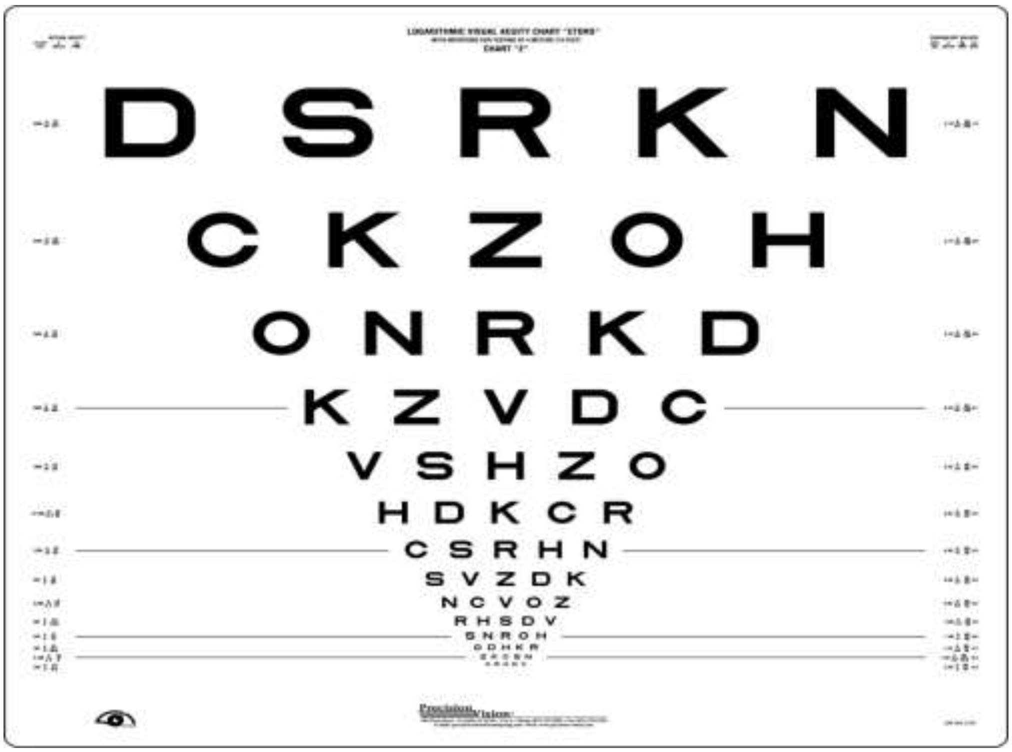


Figure 1.4

1.6 Block Diagram

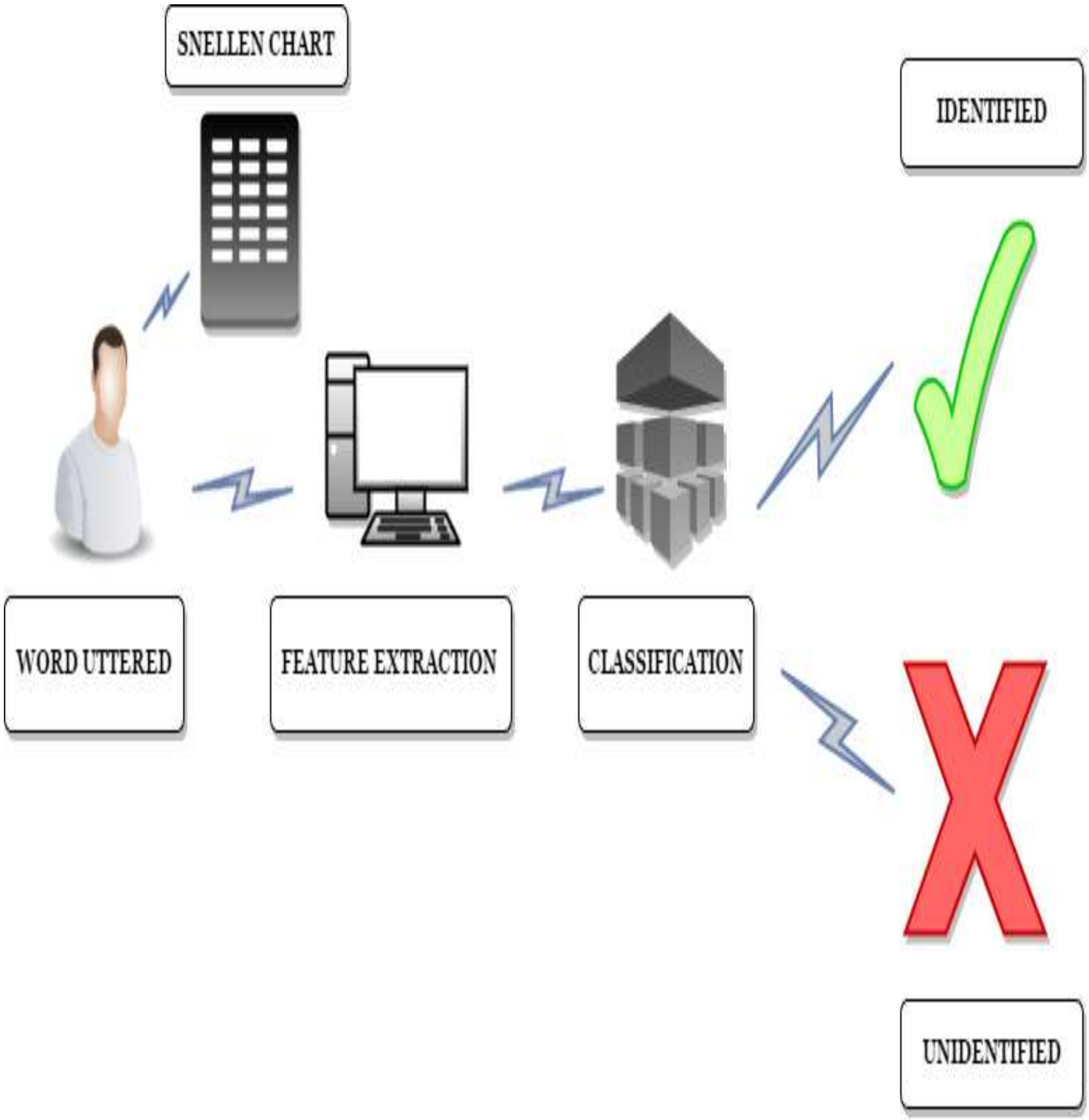


Figure 1.5: Block Diagram

1.6 .1 Explanation of block diagram

In the above block diagram the whole process is examples very precisely .A patient having eye sight problem will going to sit on a chair at a distance of 6 meter (10ft) or at the distance of 3meter (10ft) both are excepted under different circumstances .A monitor screen having the interface developed in it and a microphone attached to it .The letter of Snell chart according to their size will going to display one by one before the patient on monitor screen. This whole process take 2 steps and both the steps are build inside the code of speech recognition. It uses feature extraction and classification

1.6.2 Feature extraction

In the step of feature extraction the feature of the word uttered by the patient are extracted from signal using signal processing techniques .specially the MFCC will be used to calculate the Mel cepstral co-efficient of the signal and other statistical Parameter like skewness Hjorth etc. These all feature are supplied to a trained machine will has the access to database. The feature extraction of the words of database is already being stored in the trainer.

1.6.3 Classification

Classification is most important step in the digital signal processing .In classification step the trainer is going to compared the features of the database letters with the features of the words uttered by the patient .The code will going to display the result automatically on screen as if the features of the database matched with the word uttered “IDENTIFIED” will be going to display on screen and if the other way then automatically “UNIDENTIFIED “is going to display on screen

1.1.2 Flow chart

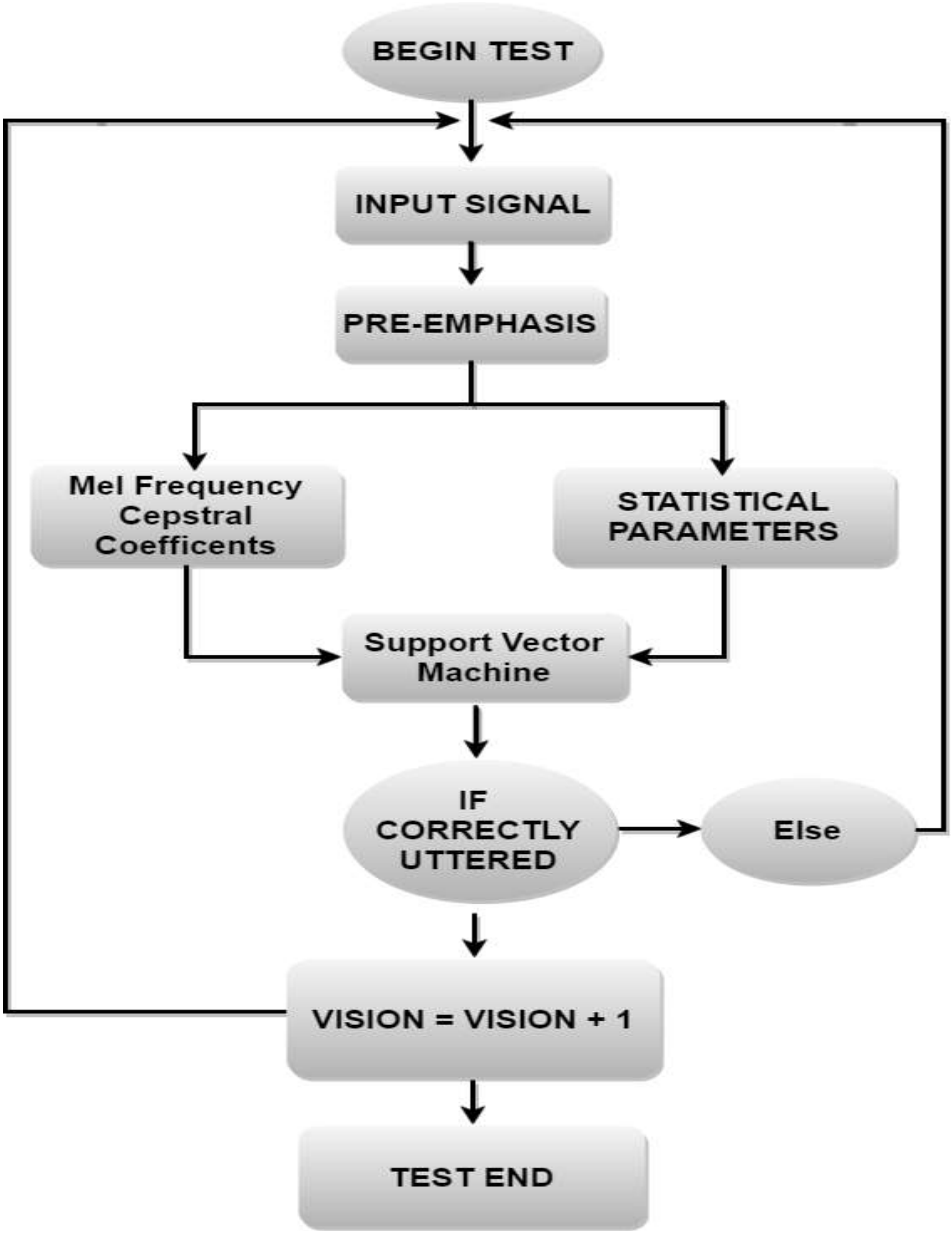


Figure 1.6

1.7 Explaination of Flow chart

A patient under observation is given a seat at a distance of 6meter(20ft) or 3meter(10ft) both are accepted .First the biggest word in the snell chart is displayed through the interface on the monitor screen .The patient with the microphone going to utterd the word loudly inro microphone.now this is the input signal reffered in above figure.

This signal is first pre emphasis,Pre-emphasis is done at a signal on the transmitter side.Pre emphasis means at the tranmitter the signal with the weaker frequencies are boosted .When the signal is boosted then the signal is send to the receiver and where a reverse procedyre is done which is called as de-emphasis.De-emphasis involves a flter bank which reverse the process and a signal of very good quality is obtained at the reciever side with out any modulator.

When the pre emphais of the voice signal of the patient is done then signal is compressed and lower frequencies are boosted.now there are 2 main stages involes first is MFCC and other is statistical paramerts.MfCC is going to Calculate the cosine transform of the log of that signal.It resluts will be calculated in the Mel scale and are recorded.

The other step involves the Calculation of Statistical components of that patient voice signal it involves skewness, Skewness helps us in determining the probability distribution of the of phonetic vector about its Mean by describing the lack of the symmetry in the data. If the phonetic vector data is tilted towards the right of the plane, then it is said to be negative skew. If the phonetic vector data is tilted more towards the left of the plane, then it’s said to be positive skew.

The other is Kurtosis shows us the distribution of phonetic data in comparison to normal distribution Properties whether. It is heavy or lightly tailed data. High value of kurtosis means the data set is Heavy tailed and low value of kurtosis means the phonetic data set is lightly tailed .The other one is Spectral flatness coefficient (SFC) differentiates a signal whether it is similar to noise or it is Similar to tone signal.

Support vector machine is placed which is basically an IA trainer, It is trained according to the Stored database and all its parameters are stored into the SVM and SVM is trained according to these. When MFCC calculate the Mel frequency and all other parameters are calculated these are fed into SVM. The SVM decides on the basis of its training that the word uttered by the patient has same parameters as compared to the trained SVM. If the word uttered matches with the Database stored then the Next word will be automatically displayed on screen and if the Word uttered is not matched with the Database then The loop will continue and At least 1 word will be displayed twice and if patient in his second attempt goes wrong then the next word below the respective line will be displayed.

1.8 General overview

Visual acuity tests are performed by doctors to assess a patient's visual acuity. Health Practitioners carry out this test manually on daily basis. This proposed technique aims at the Ease of accurately testing vision anywhere instead of planning a visit to a practitioner. In this Interactive method, a user utters isolated words as a guess input to the system from a table of Selected words. The system accepts user's utterance in the form of speech and performs Processing in two steps i.e. extraction of salient features or speech vectors from the speech Signal sufficient enough to represent the utterance and application to compare the speech Vectors extracted.

The system will identify the correct and wrong guesses in parallel with each Isolated word uttered by the user. Isolated words are determined by the system as part of the Test. The feature sets extracted from the isolated words database includes MFCC and Computation of perceptual parameters which are classified by optimum class boundaries using Support Vector Machine (SVM).

The project works in three main stages. Firstly, a voice signal is acquired. In the second step, Features are extracted from the voice sample. Finally, these extracted features are classified Using an appropriate classifier in order to achieve accurate output. Features are divided into 2 sets. Set 1 consists of MFCC whereas set 2 consists of statistical parameters. SVM is then used as our main classifier

1.8.1 Voice samples

In this project, voice samples were taken through Sony Voice Recorder (ICD-UX523F). All the Samples were recorded in low noise area to get maximum signal to noise ratio (SNR).The Sampling rate was set to 44.1 KHz and recorded using a mono microphone. Voice samples were recorded from 50 men and 50 women aging from 20 to 50 years. These samples can also be recorded using Audacity .Audacity has its own recorder and when these samples are recorded then these voice samples from each individual are divided into word like E H F etc. These samples are saved according to a specific name into the folder. We have faced many problems during the recording process which are given by

1.8.2 Difficulties in recording samples

- There was a lot of noise that was the most difficult part of our experience
- We have recorded most of the samples by outer door and people were afraid of
- .They were asking what are doing why you are doing this.
- Most of the people does not allow us to record the sample

- Most people do not allow us to enter on their property
- Most people were Afraid about their security, they want their assurances of their life and security.
- Most of the people Want to rehearsal for about 10 minutes and it was a very tidy process
- One day after a recording and working of 5 hour we were able to acquire only 5 6 voice Sample
- Splitting of Each word from the Voice recorded was very difficult process

1.9 Feature extraction

MFCC and statistical parameters are combined and used as features of voice signals. MFCC Calculates the Mel frequency of the vocal tract of the human voice and it's mainly used in Speech and speaker recognition systems. Statistical parameters on the other hand calculates The variability factor to further distinguish the voice samples. Statistical parameters used Includes skewness, kurtosis, spectral flatness coefficient and Hjorth parameters. Hjorth Parameters further has three parameters namely Activity, Mobility and Complexity. We use this Concept of speech recognition and take it further one more step by estimating the eye vision of A person.

CHAPTER 2

FORMATION OF DATABASE AND ITS WORKING

Chapter 2 Formation of database and its working

2.1 Formation of database

The design of database is most important steps in the process of this project

2.1.1 Overview

- First of all install the MYSQL with the respective instructor
- MySQL is a open source software and it is free to most of the users But there is license process for the more advance database formation
- First of All we are going to check whether our system is compatible to run MySQL or not for this we need to visit the official website to and check for forum so that our system is compatible to it or not

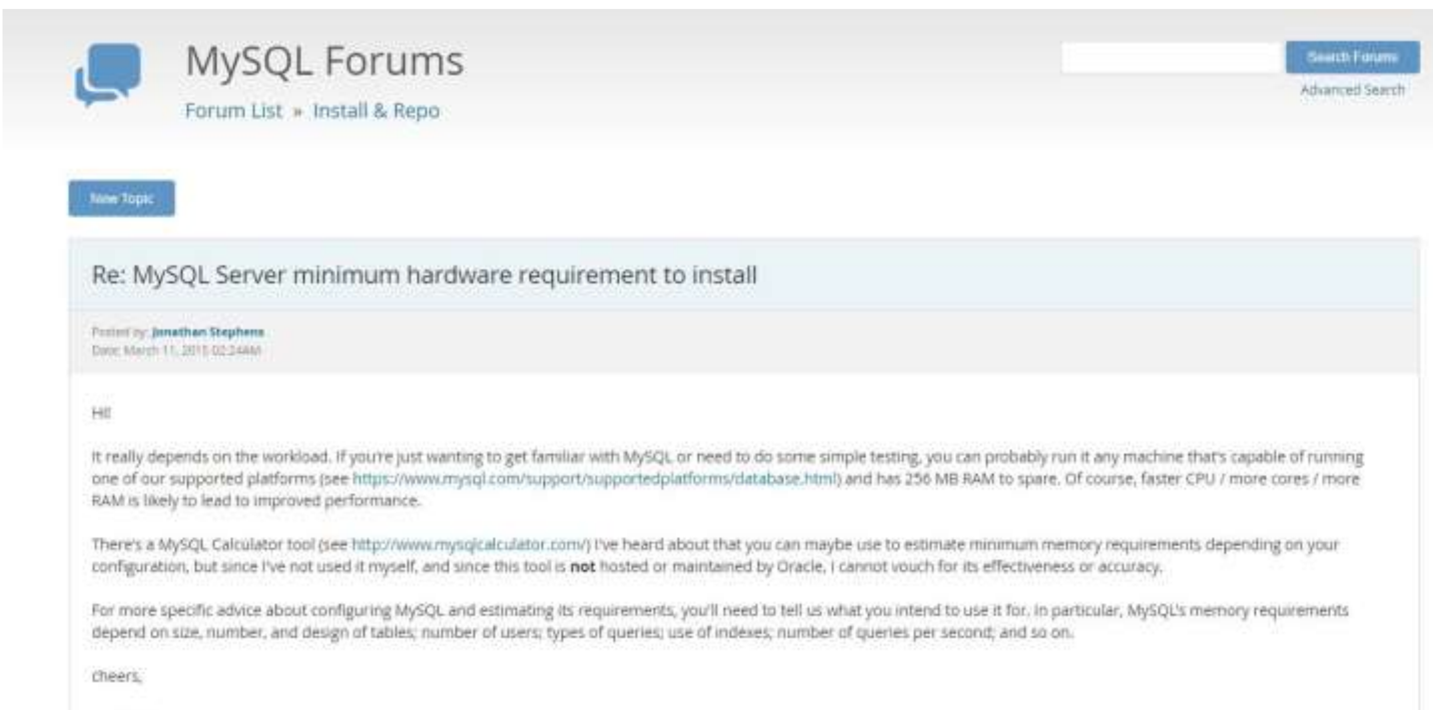


Figure 1.7

2.1.2 Installation of MySQL

We are going to visit the official website of MYSQL to download

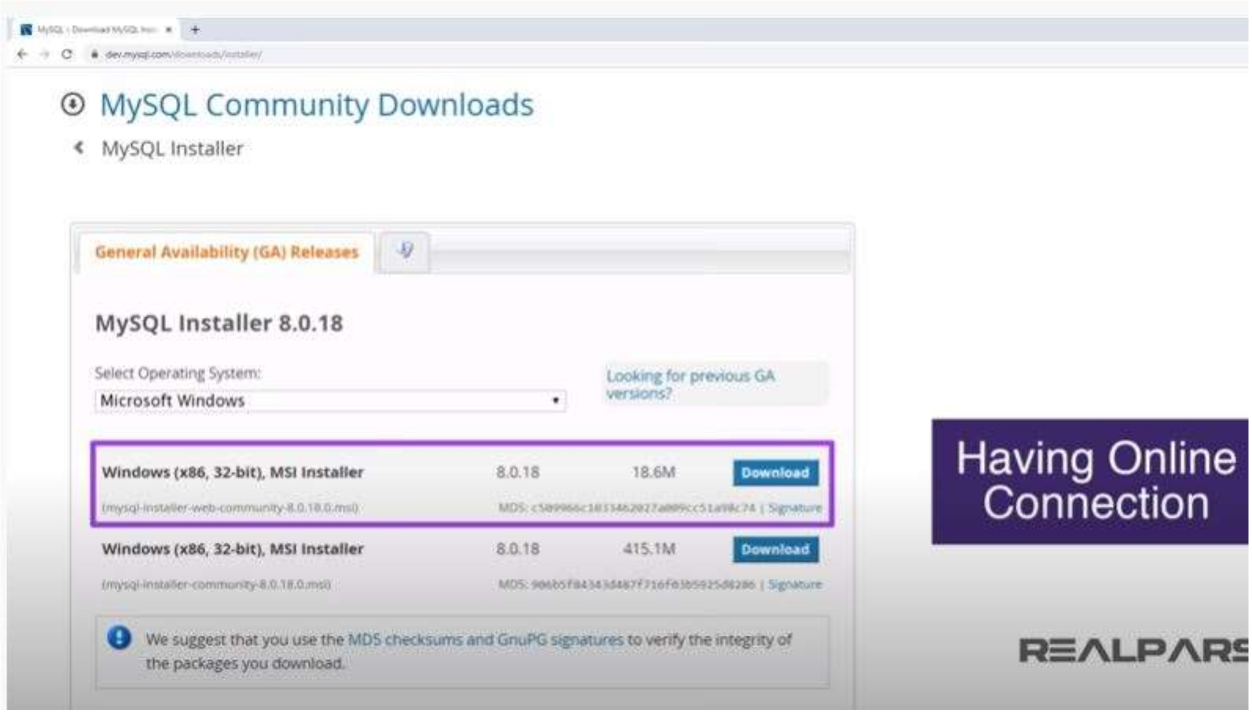


Figure 1.8

- There are 2 steps to install it of we have internet connection during installation process then step 1 MySQL web community installer is going to our option
- If we don't have internet connection during installation of MySQL then MySQL community installer can be used
- After the installation process there are few installed will install according to our project requirement.
- One step to install the produce is given below with figures

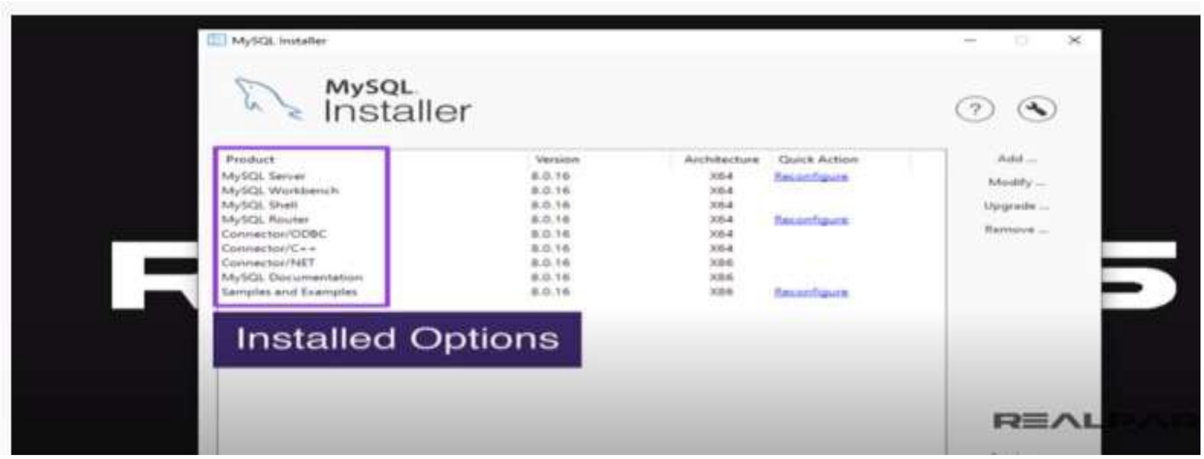


Figure 1.9

The install options are different for different projects in our project the required are installed we will explain the installation of only one installed option which will make the student to understand the basis of this process

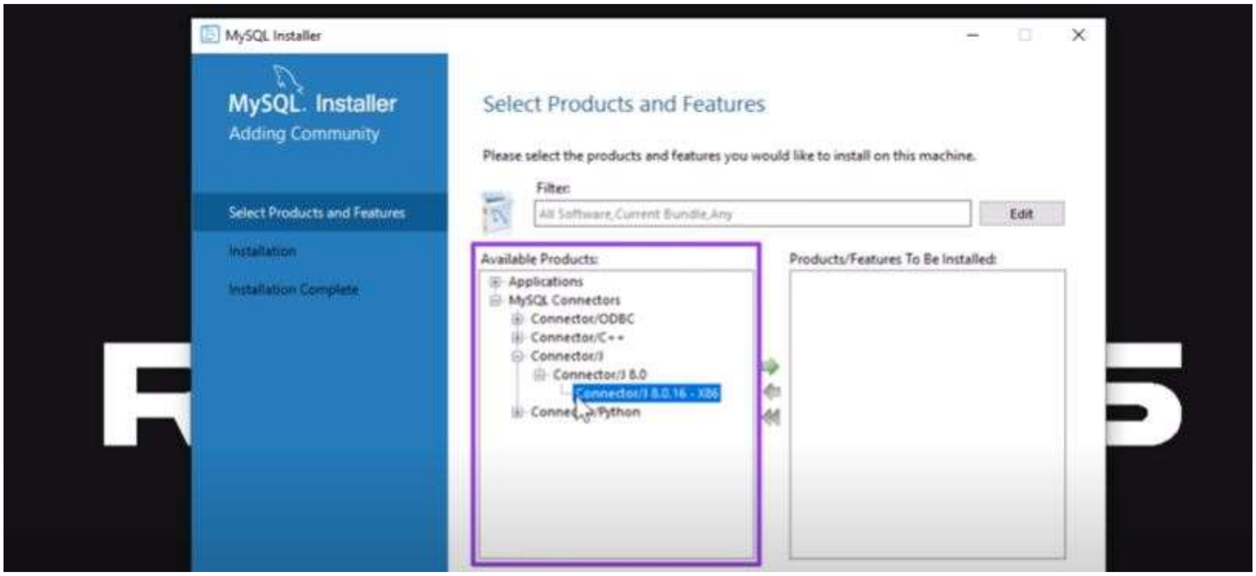
- Go to the Add button on right most side of the installer
- There will be list of products and We can add new product from left column to the right column
- 

Figure 1.10

- There is an arrow first arrow after selection of the product click in this option and then our product will go into execution stage and now by simply pressing execute, the product will be installed easily
- After the installer process then we will be going into the MySQL work bench
- There is a user password that will be added created during the process of installation

Figure 1.11

- Our samples which were created through the recording process is now saved into a folder and with names as voice 1 to voice 1000 it must have **.**SQL format

- Then we have to run the SQL script which is being saved into the system

- We will select run SQL script and then select the folder from the system and run it

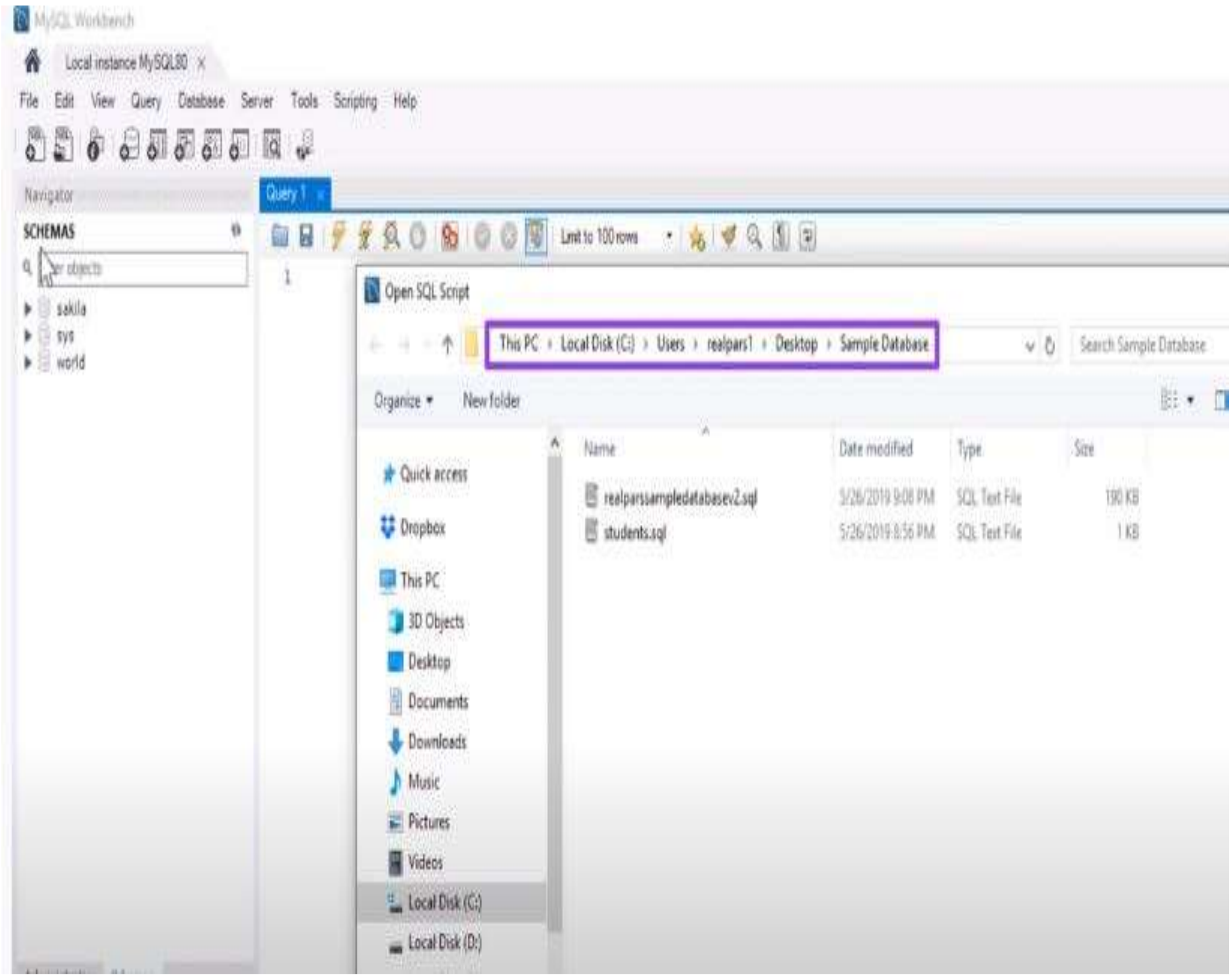


Figure 1.12

- After selection then database will going to be run after running the sample folder

In order to recognize any test voice sample using a classifier, the first step is to train it using an Appropriate database. Isolated utterance of alphabets as shown in Figure 3 were recorded in a Noiseless environment and arranged to be used as a database for classification purposes.

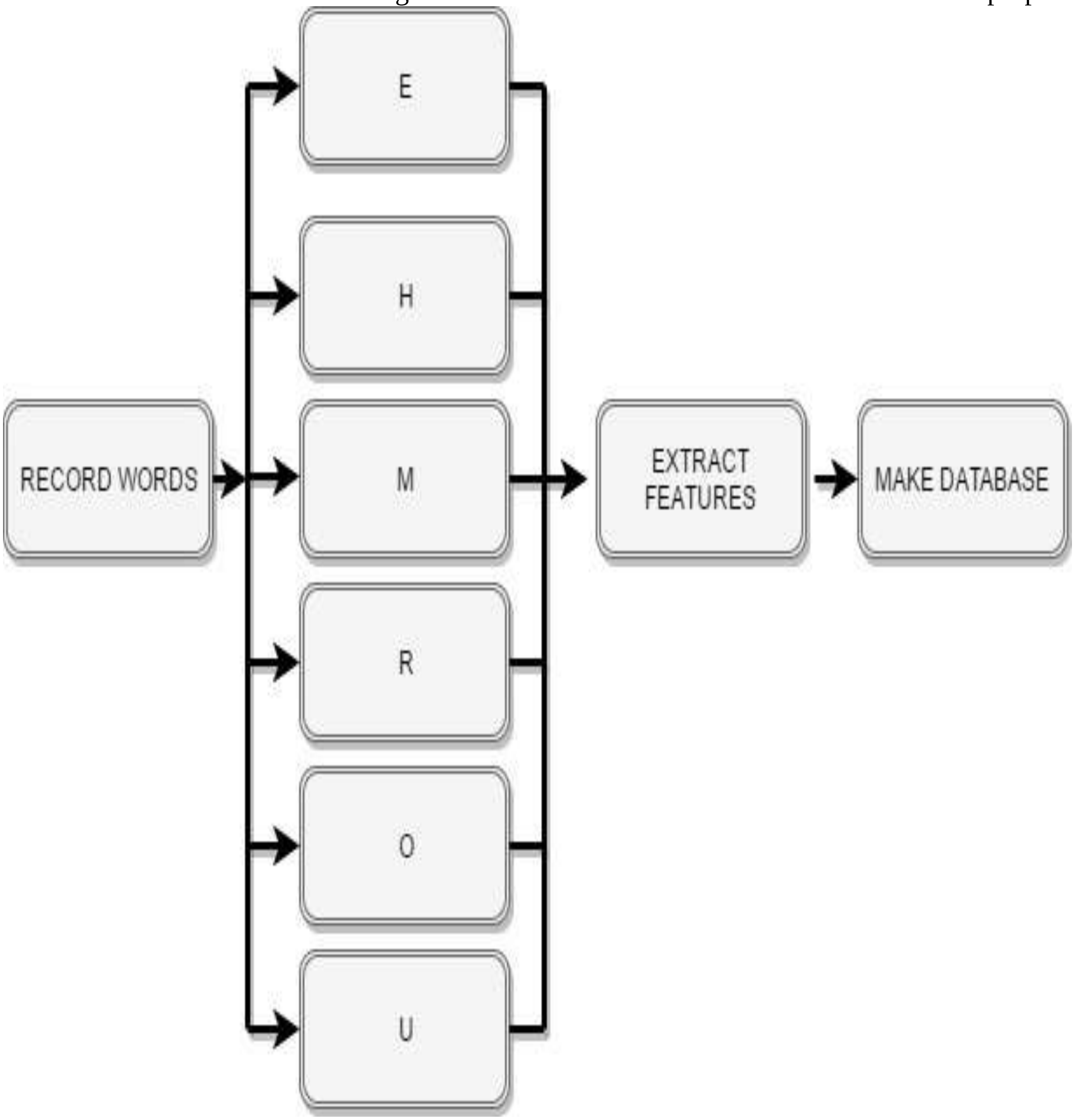


Figure 1.13

2.2 Explanation of Database working

The database has a major contribution towards our project and formation of database was the fundamental stone for our project. The working of database was
A great success for us project and it was most difficult step involve

- First of all after splitting the Word from a band of voice these word were utilize in the database
- MFCC calculate the Mel cepstral co-efficient for this database.
- Statistical parameters like Skewness, kurtosis etc. are also calculated for each word and then after the process it is stored into the database folder
- Every word was gone through the same process until the all samples were Collected into the folder.
- This folder was given the path in the MYSQL and then in our code there is a special instruction used to link the database with the code
- SVM is trained according to this database and whole SVM work on this database features.
- SVM is given a specific path and now with a person speaks then MFCC and statistical parameters are calculated of that then the decision is put into the hands of SVM which works on trained code decided that the word uttered is matched with the Database words.
- If the SVM check is right one then the three is identified spears on monitor screen and code automatically display the next word in the Snell chart
- If the SVM guesses the wrong word then the loop continue for another time and this time if the person speaks again wrong then then the code will automatically display the next word and considered it as a wrong answer

2.3 Classification

In order to distinguish between different voice samples and correctly identify them, a classifier Is used which can learn from the features extracted and make decisions based on those Features.

- SVM is used as our main classifier which provided maximum accuracy in this case.
- Achieving maximum accuracy with minimum error rate comes with a cost, that is, for such Accuracy, either high performance machines are used for training purpose or a large database is required having large number of samples

2.3.1 Support vector Machine:

- Classifying data is most important part of the machine learning world and support vector machine is the backbone of this process. The whole process can done in seconds or it can never be done
- SVM works on hyperplans,Different hyperplane are constructed and SVM look for the maximum margin between the plans
- When there is a maximum margin between the plans the SVM made decision according to that plan.

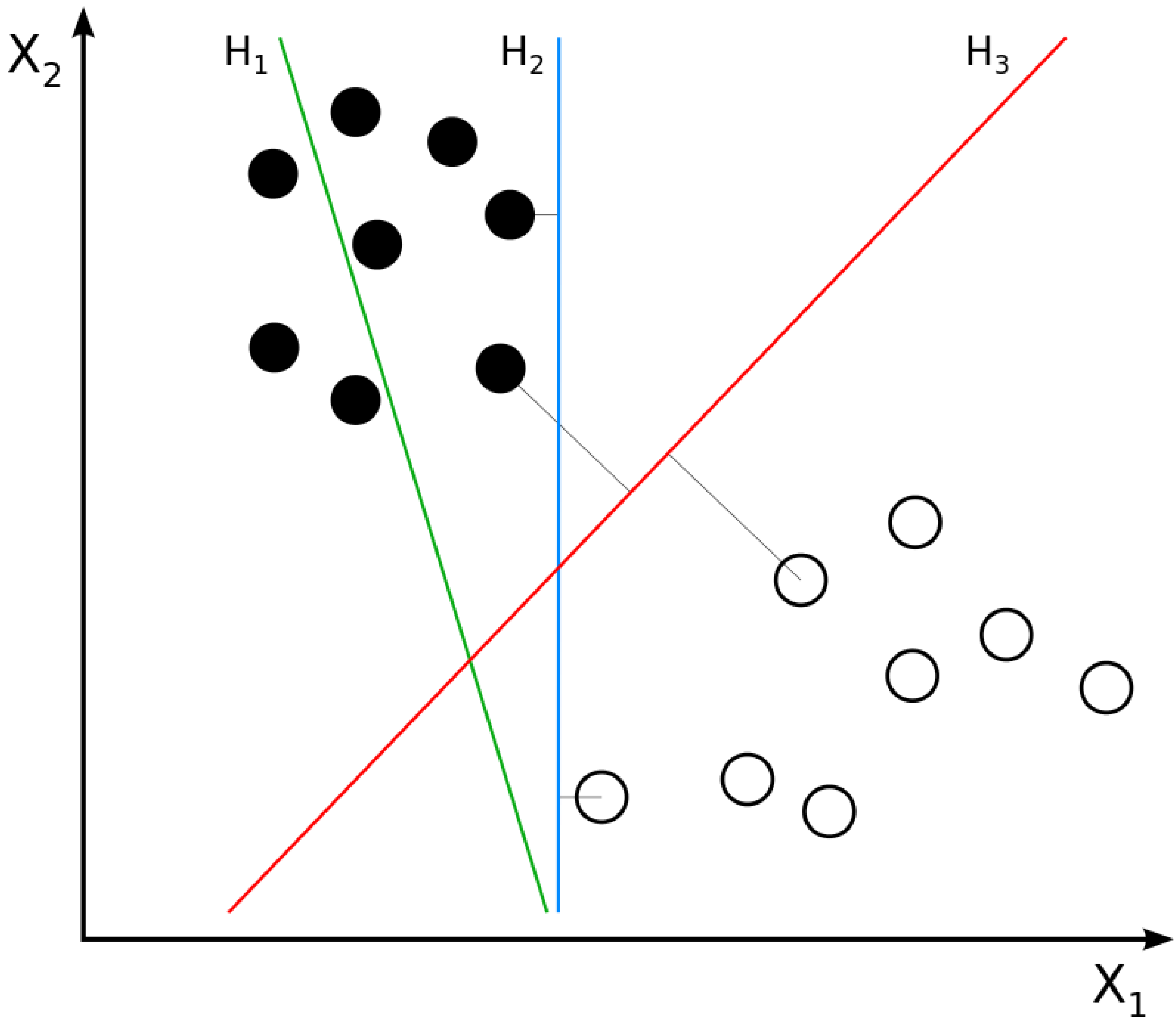


Figure 1.14

- Here there are three hyperplanes H1,H2,H3
- Both data with full black and full white holes are different type of data
- In our case after the MFCC and statistical Parameters the data word with same features are placed in category one and other in category 2

- At the time of decision when a specific word is uttered the SVM generate its hyperplane ,Then SVM looks for the maximum margin that will give its decision capacity ,in this case H3 has most margin so it will select from H3 hyperplane and decision is made according to H3.
- SVM has a lot of applications it can used to check the character written with hand
- It is also being used in satellite to classify the satellite data

CHAPTER 3

RELATED WORK, PROJECTS AND RE-SEARCH PAPERS

Chapter 3 Related work, projects and Research paper

3. Related work

The work in this project can be divided into 3 basis parts first is speech recognition, Digital signal processing and eyesight project that are already done are cover in this section.

3.1. Speech Recognition

- The history of Speech recognition is very bright .Speech recognition is now a days technology and it evolve with the time being passes
- Most modern example of Speech recognition is the Google speech recognition system.
- In google speech recognition system was invented in 2007 ,it was the first step towards the new era of speech recognition
- People in technology are called as couch potatoes which means people are so lazy that they do not want to search by writing it ,it seems a difficult process from them so this new advance speech to text software's were invented.

3.1.1 From 1950 to 1960:

- In between 10 year the speech recognition was barely a talk and was not invented then any software. It only includes the number and digits.
- In 1953 bell laborarities invented the system which only can understand the number
- In 1962 only 16 words can be used English words
- Voice recognition was then invented with 9 consonant and 4 vowels

3.1.2 In 1970's

- In 1971 to 1976 there is a research project USA defence plan a project called as DARPUR SHUR and "Harpy was developed.
- It was world most efficient system of speech recognition understand up to 1011 words

3.1.3 In 1980’s

- In the mid of 1980 there was the biggest breakthrough in the field of speech recognition
- The development of Hidden Markov model was the greatest success. This model can Recognition the speech without the knowing the speaker. The speaker was taken out of the equation by this model and this was great development
- In 1987 a doll was made for children’s which was used to answer and to communicate with the childrens.The technology developed was great but it has few flaws in 1980’s

3.1.4 In 1990’s:

- In 1990’s with the development of fast processor the field of speech recognition became more revolutionary. In 1990 the dragon detect was invented which was able to speak the 100 words frequently

3.1.5 In 2000’s

- In 2000 google came into being which is the biggest search engine of world
- The speech recognition was made its impact when google invented it google voice search for only IPhone users
- After later 2010 the google made an impact that included google speech search for Android and OS system
- Google made a database consist of 230 billion different word combination and now is world widely being used

3.2 Research papers on speech recognition

- Different research paper were published on speech recognition which is related with our project under working

- The only the most famous research paper was published by Dubagunta alon with his three others fellows which work on speech recognition was cleared made through. The research paper was about the improving children speech from raw speech signal.

- The research paper was based upon the automatic speech recognition, as children voices have both acoustic and linguist variability in children speech so it was a difficult challenge for Dubagunta.

- Another research paper was published by chungling tang which also impact upon speech recognition, it was speech recognition in the high noise environment. In the modern world the with increase in the technology now a days most of the devices are operated upon the speech recognition. This research paper from tang was about the speech recognition in the high noise environment and main focus was the automobiles.Differnet devices inside the automobile are connected to cloud and operated on speech recognition so tang research paper was about this fact which cover this aspect

- Another speech recognition research paper was published by the Indian shobha Bhatti along with 2 other fellows. This research paper was about the Hindi vowel speech recognition using hidden Markova model. The vowel speech recognition has a vast application. Their research was Dependent upon the HMM model using the HTK tool kit for both training and testing.5 speaker uttered total 600 words and format analysis were used to explore the Hindi data.
- Another research paper was published by the American computer engineer Lisa quakil which includes the development of art of speech recognition system for accurate military training .Military always needs a accurate and efficient speech recognition system and this paper covers this all.

3.3 Projects on Speech Recognition

Different project on speech recognition was made on speech recognition and was a great success in the field of speech recognition, involving digital signal processing and AI

3.3.1 Use teachable machine AL to control anything

- This project is about the use of speech recognition
- .First of all a voice sample database is developed and the the machine is trained according to that data base so that it can understand the samples and from it a complete sentence is made.
- It also involves the artificial intelligence in which Python coding is involved.
- A code in python is made which which act as a interface between the machine and the database just like our SVM.
- The machine gets the command for respective person and then do all the required process mentioned in the code and after that the code automatically get result from trained machine

3.3.2 Very first Arduino developed voice activated LED

- This was also big success in the field of speech .It involves the digital signal processing and speech recognition
- First all there is a source code in the Arduino that exactly have the specific task pin point into it.There is designed data base on basis of which Arduino made decision
- When a specific person speaks to the microphone connected to Arduino, There is extraction of features and these features are compared with the features of the Database
- If Arduino makes the decision that the word spoken is same as word stored in Arduino database the LED automatically blinks
- If the Respective person speaks the wrong word there decision made by Arduino is opposite and LED light will remain close

3.3.3 Voice controlled Home automation system using Arduino and hc

- Home automation is biggest success of the modern success and it is above all the recent technologies .It become a peak and spot light in the world of Artificial intelligence and speech recognition
- The working of this automation project is amazing .One person from outside of his houses thousand kilometre far away can control his/her home appliance ,it above imagination
- In this project cloud computing+speech recognition is mostly used. A database is formed which consist of all the data from all electronics appliances in the house specially the I.P address all devices
- This database is stored on cloud which generally enable any one to connect to it
- A app is developed in which coding is done and the App is connected to cloud
- In this step App is working on Voice samples, when a respective person want to access the specific device in house it must be connected to cloud and speaks the name of specific device, there is conversion of speech to text.
- This is a IP base system and devices are controlled Through the App

3.3.4 Doorbell and intercom with the speech recognition activator

- This is very special project, It totally based upon the speech recognition and digital signal processing
- Door bell has a microphone which convert the speech into electrical signal and through a wire inside the house the receiver is placed which has transducer which convert electrical signal to voice back
- It is complete reflection of speech recognition and digital signal processing

3.3.5 Alexa Arduino kitchen assistance

- Alexa is the voice over text application of google cloud .It has a database of about 1 billion and world biggest search engine
- Arduino act as bridge between the Alexa and the Kitchen Assistance system. It also involves the artificial intelligence
- Arduino is coded in such a way that it has a database of all the kitchen work. When respective person ask Alexa for some assistance then It speaks into the microphone connected to Arduino.
- Arduino automatically as coded it convert the speech into text and matches it with its database and after verification it forward the whole text to Alexa ,which Start searching on the cloud for respective assistance
- After Alexa found the respective assistance it is forward to Arduino which convert text back to voice and speech recognition is being done in both as well as digital signal processing

3.3.6 Smart storage draw

- There is a lot of progress in the field of speech recognition and in digital signal processing involving artificial intelligence are above thinking
- This project is simple a Draw for storage, Coding is done in such a way that that there are specific command that help us to open different sections of the storage draw
- The information of the whole storage draw is all it function is stored in the respective database
- There is Arduino connected which is bridge between the storage draw and the respective person when person speaks for specific storage draw the Arduino checks the database and convert the speech to talk and be comparing with database if the result matched then the respective draw open up and if the features of the database are not identical to the command given then draw will not open up

3.3.7 Project on eye check up

- There is Australian eye vision check-up online
- .They check the eye sight online, it is most closed work done to our project
- .The person must be 1m away from tablet or desktop and cover one eye.
- With the start of this test there will be 8 character shown at person has to write the letter as seen to it .Once you type the word then u will be proceed to next stage of the chart if the person speak one or more letter wrong then there will be the result displayed .
- The efficiency and accuracy of this online eye check is very low.

CHAPTER 4

Analytical Models and Numerical Methodologies

CHAPTER 4: ANALYTICAL MODELS AND NUMERICAL METHODOLOGY

Feature extraction is the fundamental step in any recognition system to correctly and Effectively identify any pattern of signal. Feature extraction means to select unique values or Parameters from the original data which can be used for recognition purposes which in this Case, are extracted from the speech signals.

4.1 Methods for Feature Extraction

Through speech recognition it is now possible that humans and machines can communicate Through a viable common language. Speech recognition has made its name in different fields Such as security and home automation systems. Some of the common methods used for feature extraction are Linear Prediction Coding, Perceptual Linear Prediction, and MFCC etc.

4.1.1 Linear Prediction Coding (LPC)

- LPC is a less effective approach for feature extraction as it extracts future features that have
- Arrive on the basis of previous features.
- LPC is a linear approach towards feature extraction
- Human speech is nonlinear in nature and do to this factor LPC is not used in Human speech recognition.
- Working principle of LPC is based on a source filter model of speech signal given as:

$$\mathrm{S[n]} = \sum_{k=1}^{p} 1 a k \, s[n-k] \qquad (1)$$

4.1.2 Perceptual Linear Prediction (PLP)

- PLP is far better process of speech recognition and feature extraction
- First speech is arrived then its features are extracted and stored
- It is totally different from LPC and it a good and precise method. The accuracy provided by the PLP is far good then that of LPC
- PLP is derived on the concept of filter Banks that are logarithmically spaced.

4.1.3 Mel Frequency Cepstral Coefficient (MFCC)

- MFCC is one of the most effective and commonly used method for feature extraction from
- Speech signals due to its robustness in real time signal
- The working principle of MFCC is based On cepstral analysis.
- Two types of filters are used in MFCC, one is above 1 KHz and the other is

Below 1KHz.

- MFCC is designed such that it cannot recognize words having frequencies more
- Than 1 KHz.
- This design is in consideration to human hearing phenomenon. The overall working
- Of the MFCC is divided into 7 steps, as shown in Figure 4, and are described as followed.

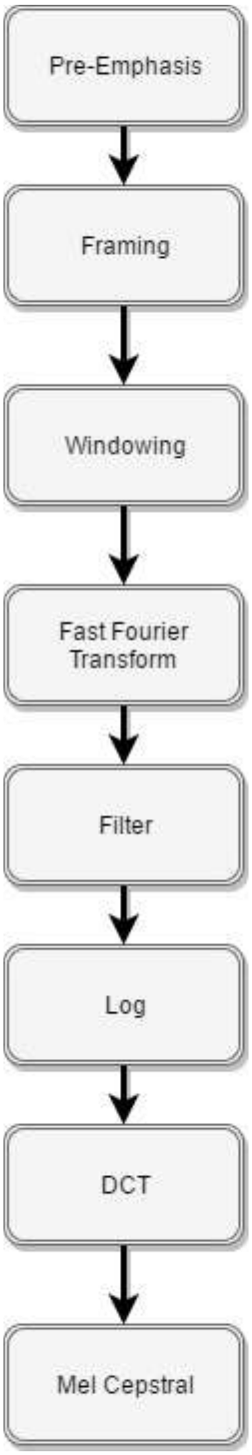


Figure 1.15: Steps of MFCC

4.2 MFCC STEPS

4.2.1 Pre-Emphasis

- This signal is first pre emphasis,Pre-emphasis is done at a signal on the transmitter side.
- Pre emphasis means at the tranmitter the signal with the weaker frequencies are boosted .When the signal is boosted then the signal is send to the receiver and where a reverse procedure is done which is called as de-emphasis.
- De-emphasis involves a flter bank which reverse the process and a signal of very good quality is obtained at the reciever side with out any modulator.
- When the pre emphais of the voice signal of the patient is done then signal is compressed and lower frequencies are boosted.now there are 2 main stages involes first is MFCC and other is statistical paramerts.MfCC is going to Calculate the cosine transform of the log of that signal.It resluts will be calculated in the Mel scale and are recorded.

$$Y[n] = X[n] - a * X[n-1] \qquad (2)$$

- Where α represents the pre-emphasis coefficient taken as α = 0.95 showing every next sample Is 95% of the previous Sample
- Higher range frequencies have small amplitude components stored in them and due to this
- Quality they are highlighted. Pre emphasis is used in order to increase the amplitudes of higher Frequencies

4.2.2 Framing

- Phonetic signals are nonlinear due to which It always complex to study them
- .For less computation power these signals are divided into multiple
- Frames, that is, these phonetic signals are reviewed as stationary in nature.
- These signals have their own sampling frequency (fs) that is used for the division of signal into multiple frames.
- Framing is used to convert the analogy signal into digital signal. In this case speech (analogy)
- Signal is converted into many digital sample having varied lengths from 20ms to 40ms. Variation
- In digital samples depends on the sampling frequency of the signal.
- Framing helps in getting a signal that is framed (divided into multiple samples) which makes

- Calculations easy as compared to the original signal that is non-linear.

4.2.3 Windowing

- Windowing is used to connect all those multiple frames that are divided into previous step
- While conversion of analogy to digital signal. Windowing also helps to remove those signals that Are not needed and left after the framing of the phonetic signal.
- A distance value is used for the Purpose of this connection and in this project 10ms for each frame is used. For this purpose,
- Hamming window is selected as shown in Figure 5.

$$W[n] = 0.54 - 0.46 \text{ Cos } [\frac{2\pi n}{N-1}]; \quad 0 \leq n \geq N-1 \quad (3)$$

W[n] shows the hamming window used in this project for windowing of signals and ‘N’ Represents sample rate of speech signal.

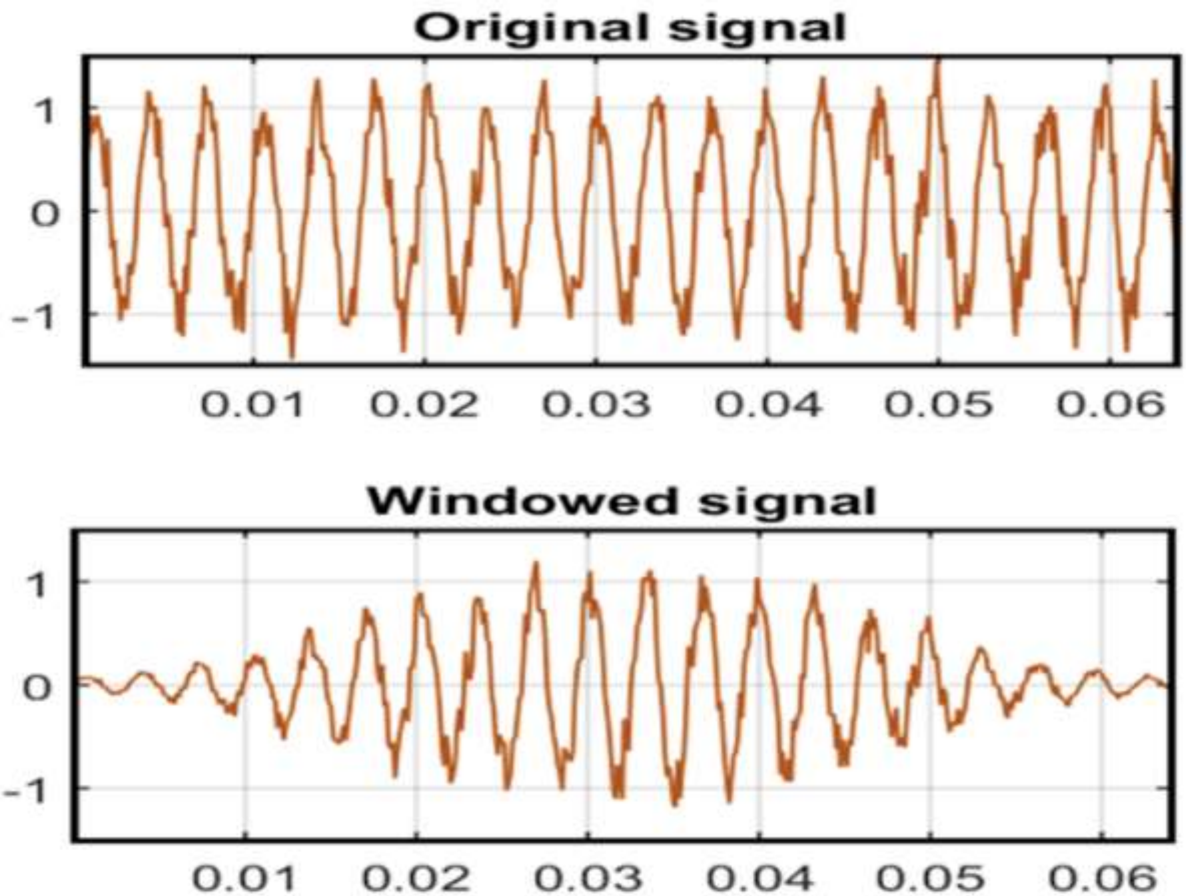


Figure 1.16

4.2.4 Discrete Time Fourier Transform (DTFT)

- Discrete Time Fourier Transform (DTFT) is used to analyse frequency components and
- Magnitude response of each frame.
- DTFT uses number of samples 'N' that is used by the signal
- To show frequency contents in a signal. For the conversion of speech signal from time domain
- To frequency domain, we need these equations:

$$\text{G (t) =Fourier [h (t)*x (t)]} \approx g[w] = h[w] * x[w]$$
$$\text{H (t)=} \sum_{n=0}^{N-1}[X]\, e^{-j2\pi kn/N} \quad 0 \le \text{k} \ge \text{k} - 1 \qquad (4)$$

- Where *h* (*t*) represent the time domain representation of the speech signal. By using
- Discrete Time Fourier transform *h*[w], its frequency contents in the frequency domain are shown in the Form of g[w].
- *K* gives the discrete number point where the Fourier transform will occur Upon. It is taken as 512 in which, the first 257 coefficients are considered leaving behind the Other as they are considered to be low frequency components and have no certain impact on The distinguishability of the phonetic signals.

4.2.5 Mel Filter Bank

- We are interested in obtaining a discontinuous envelope of the frequency response of the
- Speech signals.
- In order to extract envelop like features, we use bandpass filters of triangular
- Shape.
- We multiply the continuous magnitude response, obtained from fast Fourier transform,
- With a set of 20 triangular shape filters in order to obtain log energy from each triangular filter.
- These filters are equally positioned in a way that they follow the Mel frequency scale which can
- Be equal to linear frequency by using the equation:

$$\text{Mel} = 2595 * \log_{10}(1 + \frac{f}{700}) \qquad (5)$$

- Mel frequency is proportional to the logarithm of the linear frequency, showing similar effects in the human's aural perception.

4.2.6 Log

- Log is used in the filter bank to calculate the magnitude of the input.
- It is used to compress the Data that is filtered out and the output data is always dynamic. Due to slight variation in the Input signal change in input frequencies are less sensitive which allows to analyse and compute
- Input signal more easily without any hurdles.

4.2.7 Inverse Fourier Transform (IFT)

- This step involves the back conversion that is, converting the signal from frequency domain to
- Time domain.
- This conversion will provide mel cepstral coefficients which are also known as
- Acoustic vectors.
- Inverse Fourier Transform is used to convert frequency domain signal to time
- Domain through Discrete Cosine Transform:

$$\mathrm{Ggj} == \sum_{n=0}^{N-1} \cos[\frac{\pi}{N}\,(\mathrm{n}+\frac{1}{2}j)] \qquad \mathrm{j}= 0\ldots \mathrm{N}-1 \qquad (6)$$

Ggj represents the converted time domain Mel cepstral coefficients and N shows speech Signals' sampling rate.

4.3 Statistical Parameters

In feature extraction along with MFCC, some statistical parameters are also calculated to Increase the accuracy of the system. These parameters help increase the overall accuracy and Stability of the system.

4.3.1 Skewness

- Skewness helps us in determining the probability distribution of the of phonetic vector about its
- Mean by describing the lack of the symmetry in the data.
- If the phonetic vector data is tilted towards the right of the plane, then it is said to be negative skew. If the phonetic vector data is tilted more towards the left of the plane, then it's said to be positive skew.
- Let S be the Skewness for a random variable, then it can be calculated as:

$$\mathrm{s}= (\frac{x-\upsilon}{\sigma}) \qquad (7)$$

Where X is a random variable, υ is the median, and σ is the standard deviation of the speech Vectors.

4.3.2 Kurtosis

- Kurtosis shows us the distribution of phonetic data in comparison to normal distribution
- Properties whether it is heavy or lightly tailed data.
- High value of kurtosis means the data set is Heavy tailed and low value of kurtosis means the phonetic data set is lightly tailed.
- Equation Shows the measure of kurtosis,

$$\mathrm{K} = \mathrm{Var}\ (S)^2 + 1 \qquad (8)$$

Kurtosis takes the variance of skewness, that is, it is the measure of the dispersion of skewness S2 around +1 and -1 that are its expected values.

4.3.3 Spectral Flatness Coefficient

- Spectral flatness coefficient (SFC) differentiates a signal whether it is similar to noise or it is Similar to tone signal
- For calculation of SFC, geometric mean (GM) and arithmetic mean (AM) of phonetic signal is required. SFC is the ratio of GM of the power spectrum of speech signal to the AM of the speech signal.
- Accurate results are achieved when SFC is calculated in Specific bands of 85-180 Hz and 165-255 Hz because the voice range have fundamental
  . SFC can be calculated using formula:

$$\mathrm{f(n)} = \frac{\exp\left(\frac{1}{N}\sum_{n=0}^{N-1}\ln(g(n))\right)}{=\frac{1}{N}\sum_{n=0}^{N-1} g(n)} \qquad (9)$$

From the equation, the distribution of numerical data is represented by E(#) and N shows the Phonetic signal data points.

4.3.4 Hjorth Parameters

- Hjorth parameters are used for the calculation of non-linear and non-stationary signal such as Electroencephalogram (EEG) signal
- Hjorth parameters are used here in speech signal Calculation because both speech signal and EEG signals are non-linear in nature and it provides
- The use of real time tasking due to less computational power requirement.
- Hjorth parameters Have less computational power as compared to wavelet transform and short time Fourier Transform.
- Hjorth parameters have three sub parameters: Activity, Mobility and Complexity.
- Activity results in evaluating the surface of the power spectrum by calculating the variance of a Digital signal in time domain.
- The power spectrum is calculated using Parseval's theorem Value
- Of activity depends upon the frequency components within the signal. High value activity will Be achieved if the number of frequency components are in large amount and low value of Activity will be achieved if the number of frequency components are low. Activity of a signal can be calculated through the equation:

$$\mathrm{a\ (t)} = \mathrm{Var\ (g(t))} \qquad (10)$$

- Where g (t) is the input test signal represented in time domain.

- Mobility coefficient is the calculation of standard deviation of the power spectrum of the Processed signal. Mean frequency of the signal is approximated through mobility coefficient.
- It Is the ratio of variance of the first derivative to the variance of the original signal. Mobility of a Signal can be calculated from the equation:

$$\sqrt{\frac{var\left(g'^{(t)}\right)}{var\left(g(t)\right)}} \qquad (11)$$

- Here E (F) shows the signal in time domain. E (F), represents the first derivative of the speech Signal and j F, shows the mobility of the speech signal.
- Complexity coefficient evaluates that how much the processed signal is similar to that of the Sine wave. Similarity or convergence of processed signal is determined with respect to sine Wave which is maximum at 1. Complexity of a signal can be calculated from the equation

$$\text{C (t)} = \frac{m(g\prime(t))}{m(g(t))} \qquad (12)$$

CHAPTER 5

Software hardware and connectivity

Chapter 5 Hardware software and connectivity

The over all description the project involves the hardware specification, software specification.

5.1 project hardware

- The hardware of the project involve the Raspberry pi
- The monitor screen or it can be a LCD screen placed
- A microphone that will be used to speak into it.
- Internet device to provide internet
- Connecting cables HDMI or VGA

5.1.1 Raspberry pi

- One of the most important revolutions in the history of digital world is raspberry Pi. It brought computer in every ones reach.
- Raspberry pi was first developed in the United kingdom
- It was developed on a computer chip to teach the children of computer science
- It is used in the Robotics and especially in the weather monitoring system.
- It does not required any mouse keyboard etc. to be attached with it
- Raspberry pi has different types pi 3 and pi 4,Pi 3 has a range processing rang from 700mhz to 1.4 GHz and pi 4 has processing range from 2700 MHz to 1.5 Ghz
- The Ram for the Model A and Model B was 256MBs,Model B allocated the 128 MB to GPU and other 128 for the UPU
- The Raspberry Pi 2 has 1 GiB of RAM. The Raspberry Pi 3 has 1 GiB of RAM in the B and B+ models, and 512 Mi. of RAM in the A+ model. The Raspberry Pi Zero and Zero W have 512 MiB of RAM.
- Pi zero can be used as a Portable into USB Port and can used whereas Pi 3 can be boot from the USB.
- The price is range from 5$ to 50$
- We are going to use Model pi 3 in this project

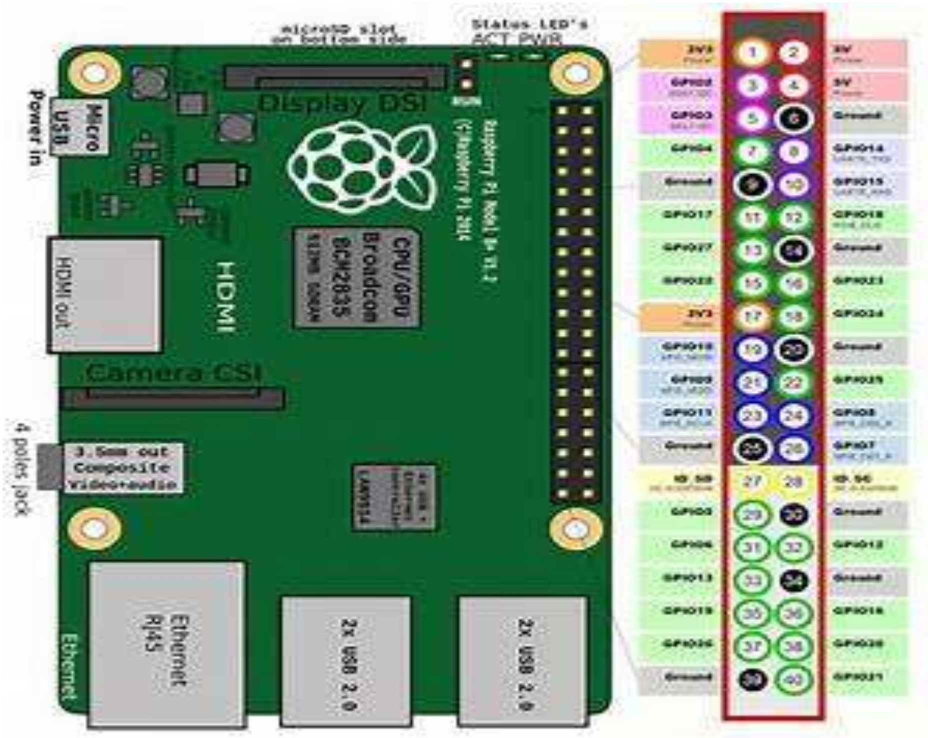


Figure 1.17

- Raspberry Pi 1 Models A+ and B+, Pi 2 Model B, Pi 3 Models A+, B and B+, Pi 4, and Pi Zero, Zero W, and Zero WH GPIO J8 have a 40-pin pinout. Raspberry Pi 1 Models A and B have only the first 26 pins
- In the Pi Zero and Zero W the 40 GPIO pins are unpopulated, having the through-holes exposed for soldering instead. The Zero WH (Wireless + Header) has the header pins preinstalled.

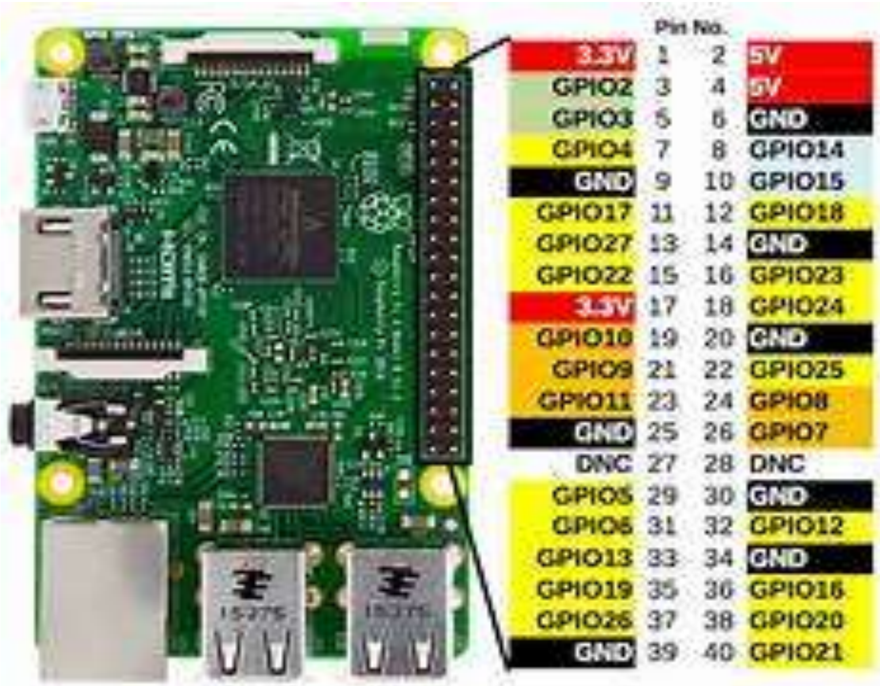


Figure 1.19

5.1.2 LCD screen

- A Lcd screen is used to display our interface.
- The words will be Display one after the other on the Lcd screen and person under observation will speaks through microphone and code will automatically check it and display result on screen
- LCD screen consumes less Power as compared to monitor screen
- LCD screen are cheap
- The brightness level of LCD screen are high as compared to the Desktop computer
- The LCD of Samsung, Huawei Apple etc. are some famous companies develop LCDs
- The price range of a LCD is from 300$ to 600 $

Figure 1.19

5.1.2 Microphone

- Microphone has A vert vast application and now a days it became the necessity of individuals
- In our project Microphone is going to play a big role, The patient will going to Speech the words into the microphone.
- A microphone Has a transducer which convert the voice into electrical signal
- The words uttered by the person under observation are converted to Electrical signal, on which then MFCC and statistical parameters are applied
- Microphones are of different types such as dynamics microphone and Piezo electric microphone.

5.1.3 Internet Device

- The internet device having 4G speed will be used.
- The internet is required because our code will compare result through cloud text database
- Selection of the internet device depends upon the service of different networks in specific place of the project
- ZONG 4g device can be used as well as Telenor 4g wingle can be used having a high speed

5.1.4 HDMI or VGA Cables

- HDMI or VGA cables are also part of our hardware set up these are used to connect the devices with each other
- Specially the a laptop to a projector so that our project can be presented in well manger

5.2 Project Software

- Python
- Mat lab

5.2.1 Python

- Python is a general purpose language ,it is a high level language and it was developed in 1991
- Our whole coding is done through the python, first we have to learn python through online courses and then implementing it
- First of all we have to import the speech recognition library
- As our project is utilizing speech recognition so that the without the speech recognition library it will be impossible to do coding in it
- Then the database formed is linked through the code and when we run the code code will automatically check eyesight
- A interface is being developed in the Python of that Snell chart .Now through the interface when code is run then Word appears on screen person under observation will uttered the word through microphone
- The code will calculate the Statistical parameters and Mel frequency co-efficient and the results are compared with the data base
- If the patient speaks correct word then the font size of word is reduce and word with less size is appears on screen
- If the person under observation speaks the wrong word for first type then the word is again displayed as there is loop in the code ,If this time patient again speaks the wrong letter then A automatic result generated indicates the eyesight of the patient

3.py - C:\Users\Muhammad Talha\Desktop\Project Base File\1st step\venv\3.py (3.7.5)

File Edit Format Run Options Window Help

```python
import speech_recognition as sr
import pyaudio

for x in range(0,10,1):
    x= input("The alphabet is : ")
    r = sr.Recognizer()
    with sr.Microphone() as source:
        print("Speak Anything  :")
        audio = r.listen(source)
    try:
        text = r.recognize_google(audio)
        m=format(text)
        print("You said : {}".format(text))
        if x==m:
            print("matched \n")
        else:
            print("not matched \n")
    except:
        print("Sorry could not recognize what you said\n")
```

Figure 1.20

1. After running the code we have obtained a few screen shot from our interface

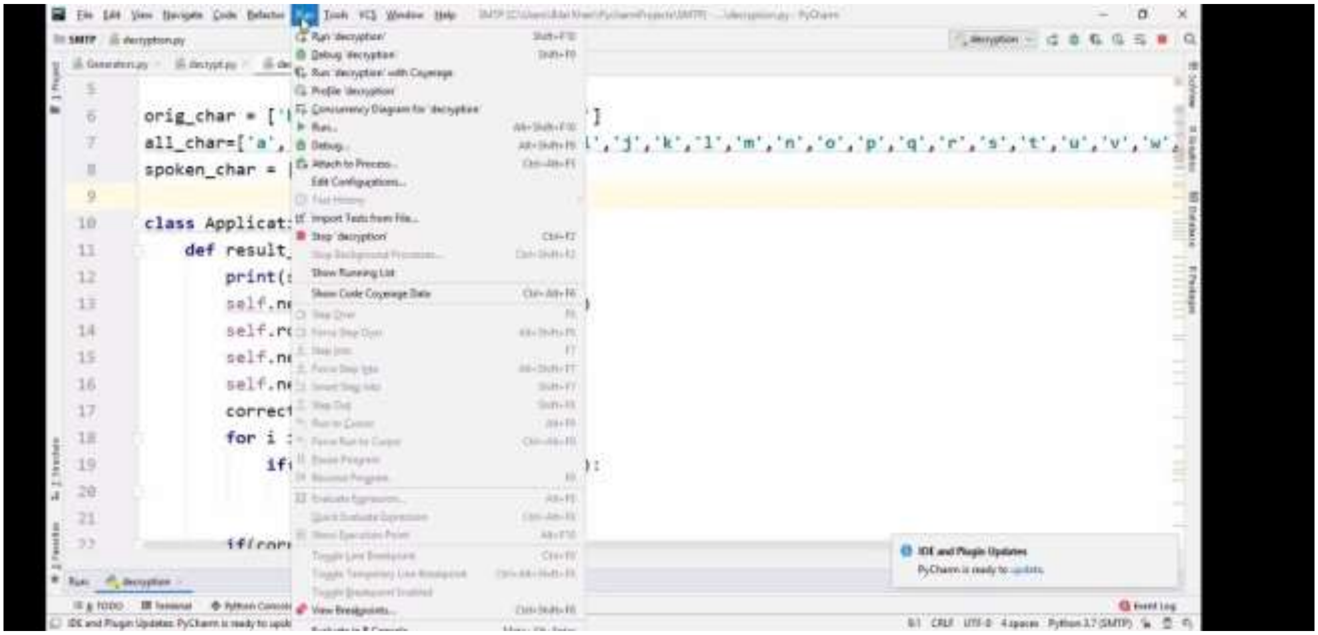


Figure 1.21

2 This word is b

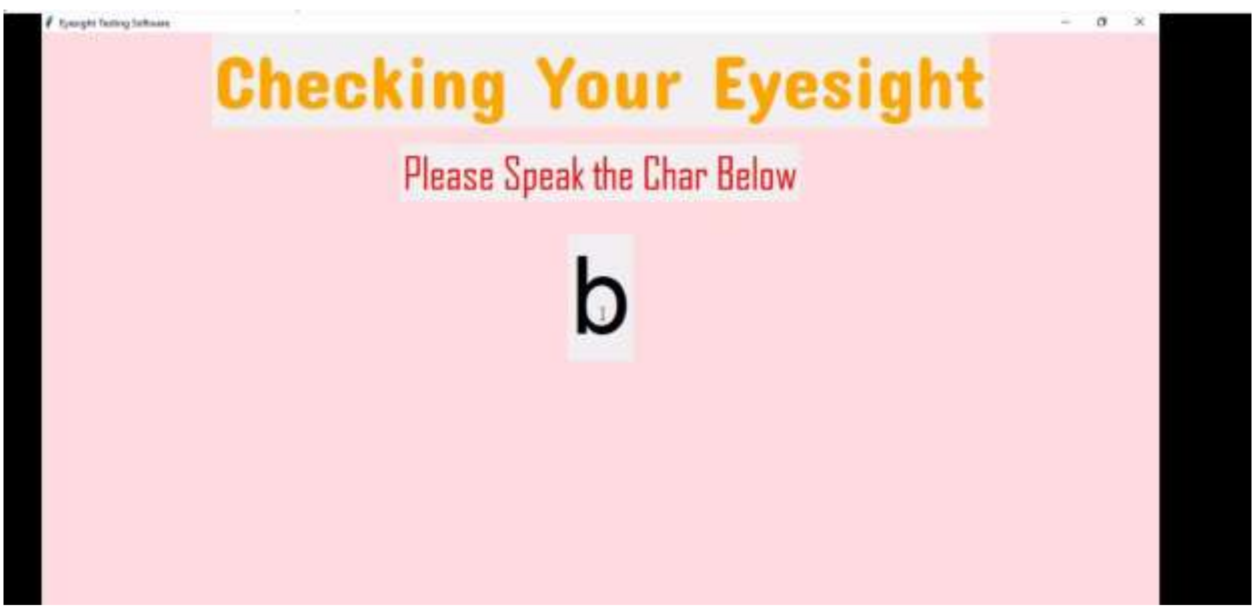


Figure 1.22

The size of word will be ready automatically and person under observation is speaking them correctly

3 This word is s

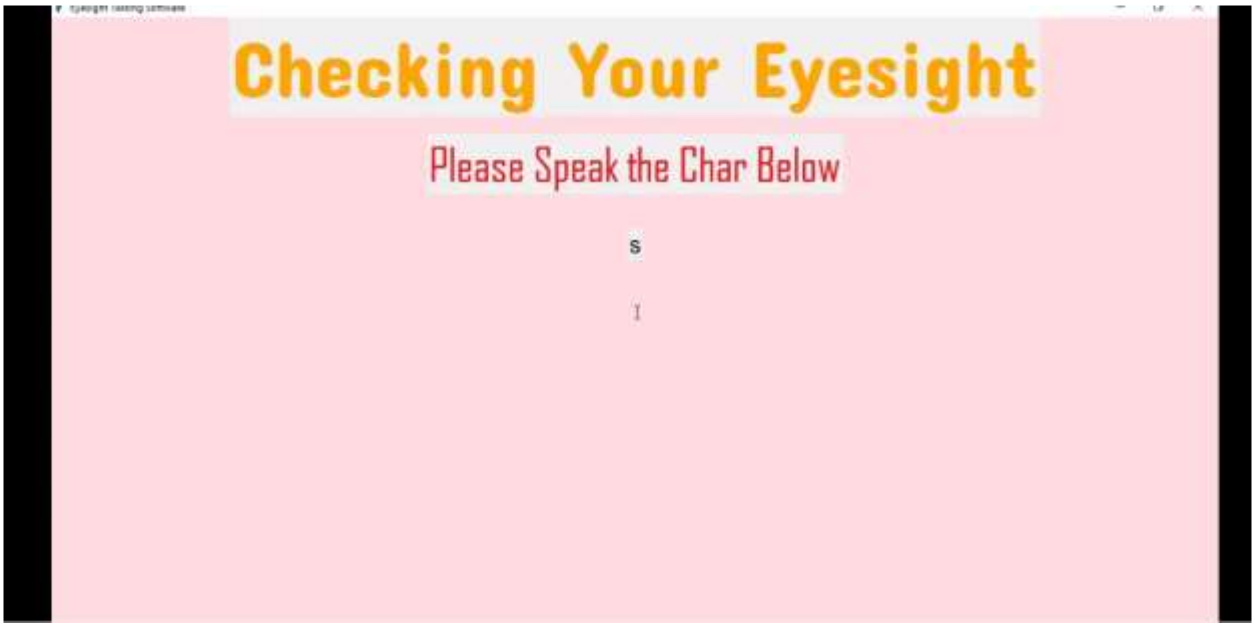


Figure 1.23

The size of the word will decrease

4 This word is f

Figure 1.24

5 This word is v

Figure 1.25

- If all the word are correctly uttered by the patient under observation then he/she has eye sight of 20/200 or 6/6

Figure 1.27

5.2.2 Mat lab

- Mat lab is widely used coding platform for digital signal processing and image processing
- Speech recognition Most algorithm are developed in the Mat lab
- Most of the speech signal are process in the mat lab their plots frequency calculation are done in the Mat lab
- There are basically 2 methodologies involved in our project one is that we continue in Mat lab and done the whole project in the Mat lab and second meth ology was that we transfer the whole project and algorithm into python
- We prefer Python because Python is easy to understand and time delays are less
- The transformation of whole code into python was easy so with the grant of our supervisor our whole project was transfer to python.

5.3 Connectivity

- The hardware are used are simply an LCD screen to display our interface
- .Raspberry pi is used to connect the code through it to LCD screen.
- First we have to install library Gpio into raspberry pi.
- Once the library is installed the raspberry pi will be used as a simple python application
- Once it is done our code is pasted into raspberry pi and simple run it with F5.
- Through a cable connect it to LCD screen a microphone is connected through

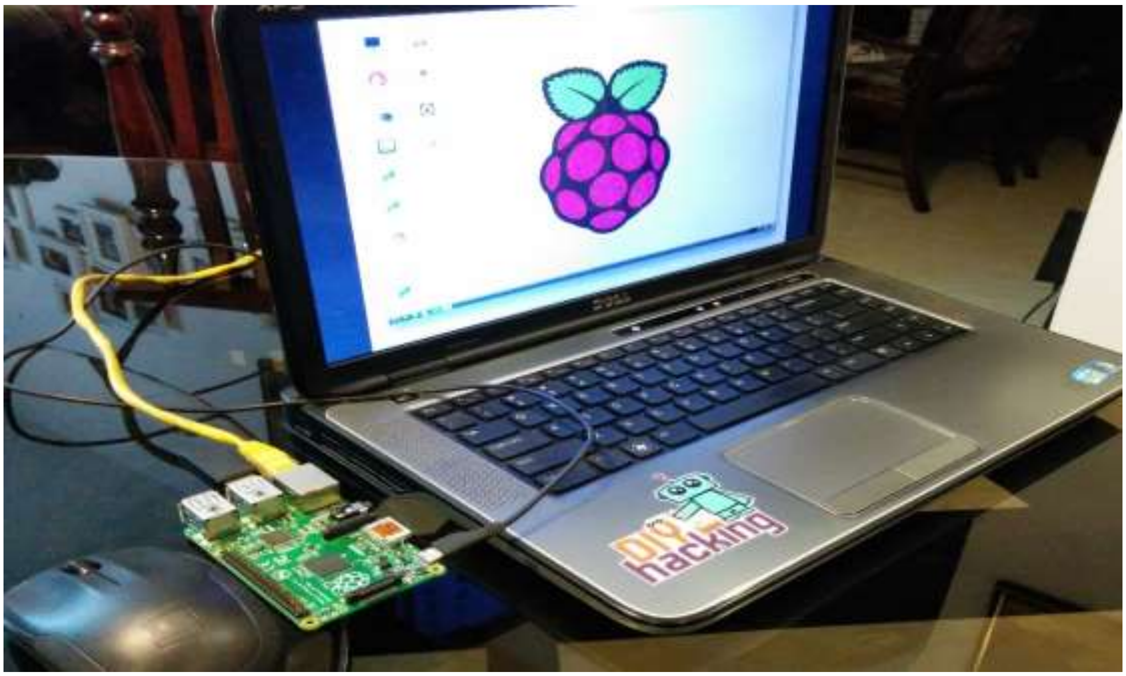


Figure 1.28

CHAPTER 6

FUTURE WORK ,EXTENDED SCOPE AND CONCLUSION

Chapter 6 Future work extended scope and Conclusion

There is always the scope of improvement in every project. In this project work is done with full commitment and scope describe according to which this project was design is being made successfully

6.1 Extended scope

- This project is limited to the development of the Interface ,the next step is going to develop the app for this project and Android studio is the next step involve in this project
- The individual after us will going to develop only the a app and we have already done the source coding is being done
- The only step left is to develop the app and interface the app with the source code. This automatically increase the market value of the project

6.2 Conclusion

The objective of our project was to automate the eyevision test.The attributed to increased productivity and efficiency. Python uses speech recognition providing higher output rate, cut down labor hours hence saving time.
This system can be placed hospitals to measure eyesight. This system can be place into schools and universities to check eyesight of students at their school timing. They do not needs to leave the University or college to get a appointment and spend a whole day in hospital to check eyesight

The problems of the rural areas can also be greatly reduce .In villages there is not doctor available and due to having less facilities, medicine etc. their eyesight get weaker. In our own village there is no gas and people used to burn organics to cook food and for warming .The gas specially carbon dioxide and carbon monoxide causes the eye sight problems .As there is no proper system for checkup so people gets blind at very small age due to un wanted gases.

The objective of the project is being meet/so we only need stop build a small dispensary in villages and this system can be implemented through their and people with less money and weaker eyesight can check up their eyesight at very low rate.
This project is a great success in the field of digital signals processing and specially speech recognition. This project teaches us that how flexible is the digital signal processing.it can be bend and used in any field for the benefit of human beings.

BIBLIOGRAPHY

[1] Nägele MR et al.: Retinal microvascular dysfunction in heart failure. European Heart Journal 2018; 39: 47–

[2] Evans, Jennifer R; Morjaria, Priya; Powell, Christine (2018-02-15).

[3] *Emmett T. Cunningham; Paul Riordan-Eva. Vaughan & Asbury's general ophthalmology (18th ed.). McGraw-Hill Medical. ISBN 978-0071634205.*

[4] *Judd, Deane B.; Wyszecki, Günter (1975). Color in Business, Science and Industry. Wiley Series in Pure and Applied Optics (third ed.). New York: Wiley-Interscience. p. 388. ISBN 978-0-471-45212-6.*

[5] *Conover, Emily (July 2016). "Human eye spots single photons"*

[6] eye, human. "Encyclopaedia Britannica from Encyclopædia Britannica Ultimate Reference Suite 2009

[7] Carpenter, Roger H.S. (1988). *Movements of the eyes (2nd ed)*. London: Pion, Ltd ISBN

[8] Savino, Peter J.; Danesh-Meyer, Helen V. (2012). *Color Atlas and Synopsis of Clinical Ophthalmology – Wills Eye Institute – Neuro-Ophthalmology*. Lippincott Williams & Wilkins. p. 12. ISBN 978-1-60913-266-8.

[9] Trattler, William B.; Kaiser, Peter K.; Friedman, Neil J. (2012). *Review of Ophthalmology: Expert Consult – Online and Print*. Elsevier Health Sciences. p. 255. ISBN 978-1-4557-3773-4.

[10] Deering, Michael F. (1998). *The Limits of Human Vision*

[11] Sharma, O,P., Renaud, E., Butler, T.L., Milasps, K., Dring, R.P., and Joslyn, H.D., 1988, " Rotor- Stator Interaction in Multi- Stage- Axial Flow Turbines", AIAA Paper No. 88-3013

[12] Busby, J.A., Davis, R.L., Dorney, D.J., Dunn, M.G., Haldeman , C.W., Abhari, R.S., Venable, B.L., and Delany, R.A., 1999, " Influence of Vane-Blade

Spacing on Transonic Turbine Stage Aerodynamics: Part II- Time- Resolved Data and Analysis:, ASME Journal of Turbomachinery , Vol. 121, pp. 673-682

[13] **Collar (1947), "The Expanding Domain of Aeroelasticity,"** ***Journal of the Royal Aeronautical Society*** **51-1.**

[14] **Bell Loyed, "Three dimensional Unsteady flow analysis in vibrating Turbine Cascades", Durham University, 1999.**

[15] **Jeff Green, PHD Thesis, "Controlling Forced Response of a High Pressure Turbine Blade", Royal Institute of Technology, Stockholm, 2006.**