Voice Coding Using AI

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***Abstract*—**
When technology advances, the challenge of competition becomes more difficult. So, concerns about the sturdiness and adequacy of the human/developer? Will AI, however, be able to fulfill the demand? As previously said, it is an overhead for developers to remember each and every kind of syntax and cause or write every line of code. It's very feverish and time consuming.
Syntactical and structural ways of programming Programmers are divided into three groups: beginner users, information intermittent users, and professional users. It is still a challenge for inexperienced users to write code that is free of typographical errors, even though they have theoretical knowledge of the programming language, its structure and syntax, as well as the logic of the software.
As a result,. The main shibboleth of this paper is to devise a system for writing or developing software code with less stress and time. This system allows developers to start coding with a single press

***Keywords*— Voice coding, Ai, Voice Recognition, Learning, Development, Speech to text system, verification, ASR,Command**

I. INTRODUCTION

For decades, designers have used or processed speech for a wide range of purposes.. Speech recognition alleviates the difficulties and issues that other communication strategies can cause. Speech was not used much in the field of computers and software in the past. However, we can easily process speech signals and use them in our desired fields thanks to modern processes, algorithms, and methods. Our speech-to-text engine translates speech to text in real time. It may be a good compliment to the concept of offering users more options for data entry. For users who are blind, deaf, or physically disabled, our speech-to-text engine may also include data entry options. The text-to-speech convention alters linguistic expressions.

Speech understanding has come a long way in the last 50 years. Sadaoki Furui Department of Computer Science, Tokyo Institute of Technology furui@cs.titech.ac.jp furui@cs.titech.ac.jp furui@cs.titech.ac.jp furui@cs.titech.ac.jp furui@cs.titech.ac.jp furui. a summary Automatic speech and speaker recognition has been studied for more than five decades. This paper provides a technological perspective and an understanding of the fundamental improvements made in this significant field of speech communication by surveying the main themes and developments made in the last fifty years of study. While many techniques have been created, there are still many obstacles to overcome before we can achieve our ultimate goal of developing machines that can interact with people naturally.

For further research, Automatic Speech Recognition (ASR) necessitates three key components: preprocessing, feature extraction, and post-processing. Feature extraction is the process of extracting descriptive features from a raw signal for the purpose of speech classification. The raw signal may be less informative than extracted higher level features due to its high dimensionality. Feature extraction saves the day for us.

Converting a high-dimensional signal to a lower-dimensional but more detailed version for sound recognition and classification (Furui 1986; Guyon et al. 2008; Hirsch and Pearce 2000).Speech recognition reduces difficulties and problems caused by other communication methods. Text-to-speech convention transforms linguistic information stored as data or data or text into speech. It is widely used in audio reading devices for blind people now a days. In the bigger picture, the module can open up a window of opportunities for the less-privileged. The system speaks out the selected word, which the user wishes to listen to. It can also play a defining role in establishing communication of the blind if it is incorporated into mobile phones (so that text messages could be converted into speech). [6]. The existing system deals with various dictionaries, which implements dictation of words with correct pronunciation. The current system is focused more on polishing the pronunciation.

Automatic Speech Recognition requires three main components for further analysis: Preprocessing, feature extraction, and post-processing. Feature extraction is extracting descriptive features from raw signal for speech classification purposes.

Programming involves human efforts, hardware . There are many chances to meet an error while typing. Existing systems like Google assistant recognizes the voice of the user and will perform the Operate according to the voice instructions. The same approach for programming purpose will be used for the same type of purpose.

II. LITERATURE SURVEY

[1] Speech recognition is the ability of a computer or software to analyze and convert spoken language into a master-readable format. Mechanical telephones and medical dictation systems were among the earliest applications for speech recognition. It is also used in specialized vocabulary careers for transcript, query index, and commanding computer systems. It is also personally liable for automobiles and tablets, such as Apple's Siri.

[2] "Speech Recognition System: A Review," Nitin Washani, Sandeep Sharma. In this article they categorized

the system in front end analysis to best explain and reflect each aspect of the speech recognition system. In this document they classified the system through back end analysis. The biggest argument for classifying the scheme, since they faced noise, jargon, and domain problems, is that they would be more accurate so that they come up with their idea of the front and back analytics

[3] A Tutorial on Speaker Recognition JOSEPH P. CAMPBELL, JR., SENIOR MEMBER, IEEE The use of a device to distinguish an individual from a spoken sentence is known as paper recognition by a speaker. This tutorial focuses on installing speaker recognition and implementing network access control. Speech synthesis and the basic components of automatic speaker-recovery systems are demonstrated. Compromises in design are discussed, and a new form of automatic speaker recognition is adopted.
Voice transcription. The problem in deciding identity is determining whether the speaker is a specific person or a group of individuals. During speech authentication, a person makes an identity claim.
[4] Sanjib Das / International Journal of Engineering Research and Applications (IJERA) ISSN: 2248-9622 www.ijera.com
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Speech Recognition Technique:
:- It has the ability to be a significant form of human-computer interaction. The primary aim of the speech recognition field is to improve techniques and systems for computer speech input. Humans' primary mode of communication is by speech. Human computer interfaces are needed for a variety of purposes, ranging from scientific interest about the processes for mechanical realisation of human speech capacities to the ability to simplify basic tasks.

III. PROPOSED SYSTEM

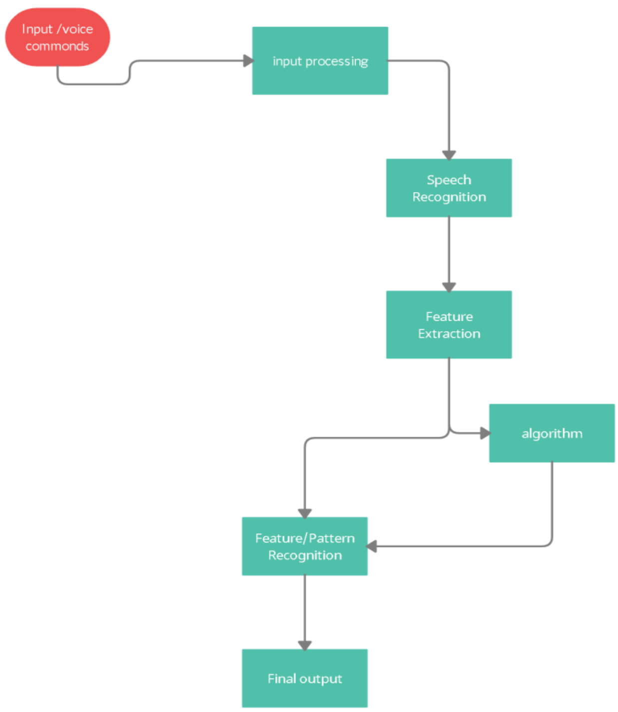
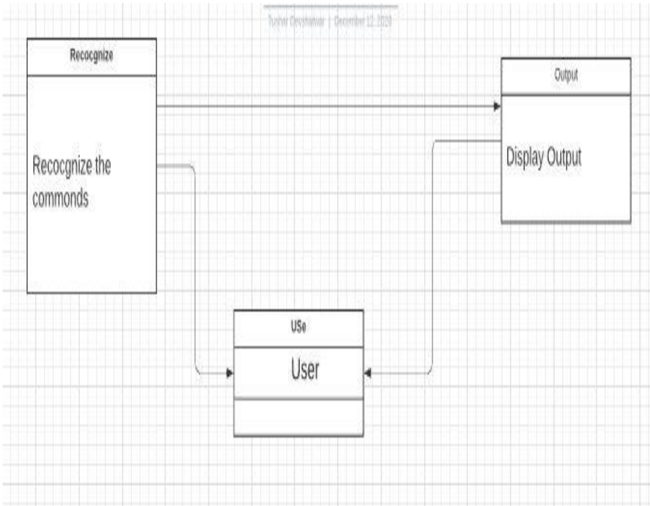
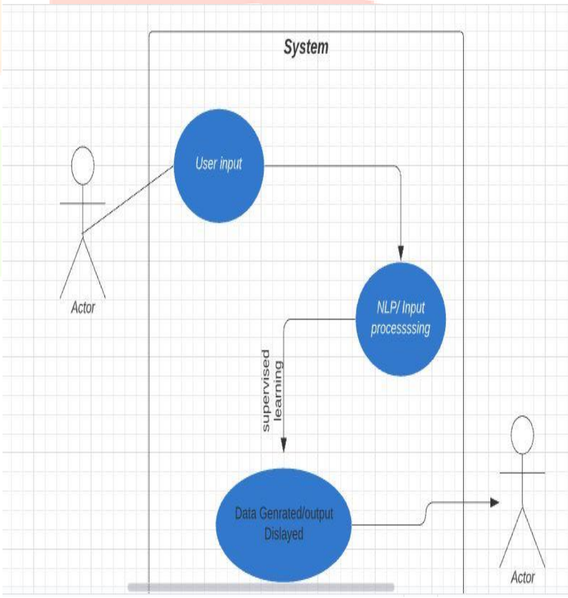


Fig. 1 Block diagram of the proposed system

UML Diagram



Use case Diagram:-



IV. Implementation

The proposed system consist of following modules

i. GUI Module
ii. Listen
iii. Eval
iv. Helper
v. Command
i. GUI Module

This modules is consist of the Required Graphical User Interface part for the software which is consists of the buttons for starting the Voice Recognition, Stop the voice Recognition, Run , Save, Open .
It also consist the code editorial part

ii. Listen
This module is for the listing the commands given by User

iii. Eval
This module performs the task of evaluating/processing the command and perform the task of ml algorithm .

iv. Helper:
Helper dose the work checking the condition or algorithm

v. Command:
Here is the some sample data stored.

After the system started it will take command from user after taking the command it will be processed and go through a process of AI algorithm i.e. under decision Tree algorithm . As we are dealing with lot of textual data for commands it will go through a bunch of if else statements and display the output within some seconds .
In this we can also travel / go from one line to another line by giving commands.

V. RESULT

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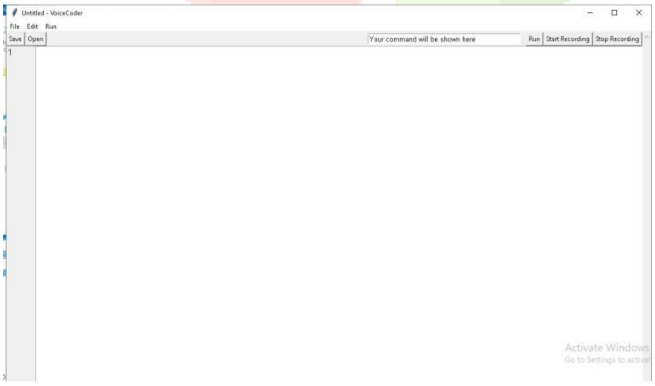
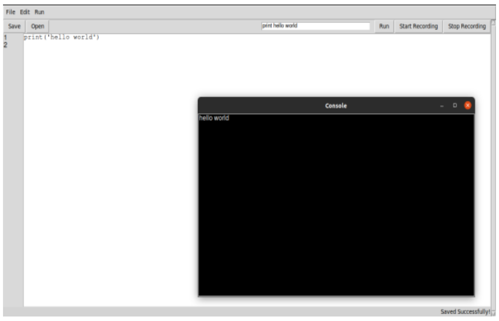


This is the User Interface of the software .

As the user give command it will evaluate it and converts it into required syntax and diplay.



After giving the run command it will run the code and will display the output

VI.CONCLUSION

When designing a computer application, we discussed the issue of the customer. Developing a computer program is a difficult process that necessitates the use of hardware resources that the user must manage. There is a chance that the user's fingertips will be injured as they continue to type the message. To prevent these issues, we're creating a technology that allows a computer program to be created using voice commands. The machine will identify the voice, and the identified words or terms will be compared to the stored keywords in the database, and if they fit, the result will be written on the editor, and the program will be compiled and executed again by recognizing the same keywords. This technology would be simple to use, requiring less human efforts and hardware resources. It would undoubtedly be beneficial to blind, inexperienced, and infrequent consumers.

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