

JARVIS A VIRTUAL ASSISTANT

¹Mr. Sachidanand Chaturvedi, ²Ravi,

³Hajrat Bilal Khan, ⁴Vinay Kumar Rai, ⁵Jignesh Yadav

¹Assistant Professor, Department Of Computer Science And Engineering,
Buddha Institute Of Technology Gida, Gorakhpur E-Mail: schaturvedi26@bit.ac.in

²Department Of Computer Science & Information Technology,
Buddha Institute Of Technology Gida, Gorakhpur E-Mail: bit17it63@bit.ac.in

³Department Of Computer Science & Information Technology, Buddha Institute Of Technology
Gida, Gorakhpur E-Mail: bit17it52@bit.ac.in

⁴Department Of Computer Science & Information Technology,
Buddha Institute Of Technology Gida, Gorakhpur E-Mail: 8933vinay@gmail.com

⁵Department Of Computer Science & Information Technology, Buddha Institute Of Technology
Gida, Gorakhpur E-Mail: bit17it20@bit.ac.in

I. Abstract

The project AIMS to develop a personal- assistant for windows-based systems. Jarvis draws its inspiration from virtual assistant like Cortana for Windows, and Siri for iOS. Its has been designed to provide a user-friendly interface for carrying out a variety of tasks by employing certain well-defined commands. Users can interact with the assistant either through voice commands or using keyboard input. our long haul explore objective is to create astute frameworks that can bolster human learning. We are especially keen on building up a way to deal with apprenticeship realizing which happens Jarvis is Computerized Life Partner which utilizes for the most part human correspondence means such Twitter, text and voice to make two path associations among human and his loft, controlling lights and machines, help with cooking, tell him of breaking news, Facebook's warnings and some more. In our undertaking we for the most part use voice as correspondence implies so the Jarvis is essentially the Discourse acknowledgment application.

As a personal assistant, Jarvis assists the end-user with day-to-day activities like general human conversation, searching queries in Google, searching for video, retrieving images, word symptoms and reminding the user about the scheduled events and tasks. The user statements/commands are analysed with the help of machine learning to give an optimal solution.

Keywords:- Personal Assistant, Windows Systems, Automation.

II. Introduction

This is a simple JARVIS in Python language. In this Mini Project Indian Railway System is simple consol application with Excel. In this project . The source code for Indian Railway Announcement system is totally error –free. This is small and simple Python programming application is game lovers and programming fresher's. This project is was developed and compiled initially in PyCharm compiler and then with some

modification it was adapted to Microsoft Visual Studio ,with some essential modification.. As steps are made in simulated intelligence, ML, and NLP calculations, it is sensible to expect that insightful conversational frameworks will turn out to be increasingly refined and will incorporate flawlessly with our physical and online universes, helping us in an assortment of undertakings. We are inspired to create conversational frameworks that can bolster human assignment learning in physical universes. Undertakings incorporate keeping up and fixing an unpredictable machine, for example, a modern printer or building an antique, for example, Ikea furniture. To be successful mentors, conversational frameworks must know about and versatile to two sorts of settings: physical setting of undertaking execution and psychological setting of the human student. Right now, investigate how a conversational framework can reason about and adjust to these specific situations. To do this, we unite profound learning approaches for PC vision and arranging approaches for versatile guidance thinking.

III. Speech - Text Representation

The discourse sign and every one of its qualities can be spoken to in two distinct spaces, the time and the recurrence area A discourse signal is a gradually time differing signal as in, when inspected over a brief timeframe (somewhere in the range of 5 and 100 ms), its attributes are short time stationary. This isn't the situation in the event that we take a gander at a discourse signal under a more drawn out time point of view (around time $T > 0.5$ s). Right now flags attributes are nonstationary, implying that it changes to mirror the various sounds spoken by the talker To have the option to utilize a discourse flag and decipher its qualities in a legitimate way a portrayal of the discourse signal are liked.

Triple State Portrayal: The Triple State Portrayal is one approach to group occasions in discourse.

The occasions of enthusiasm for the three-state portrayal are:

- **Silence:** No audio received.
- **Unvoiced:** Vocal cords aren't vibrating, bringing about an aperiodic or arbitrary discourse waveform.
- **Voiced:** Vocal strings are strained and vibrating intermittently, bringing about a discourse waveform that is semi occasional.

Quasi-periodic implies that the discourse waveform can be viewed as intermittent over a brief timeframe period (5-100 ms) during which it is stationary.

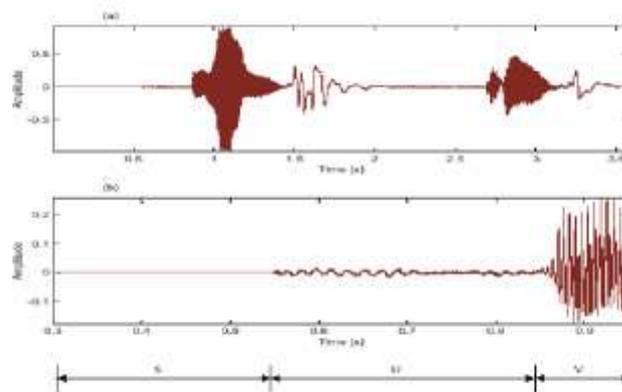


Fig. 1: Triple State Portrayal

The upper plot (a) contains the entire discourse grouping and in the center plot

(b) a piece of the upper plot (an) is imitated by zooming a territory of the discourse succession. At the base of Fig. 1 the division into a three-state portrayal, according to the various pieces of the center plot, is given. The

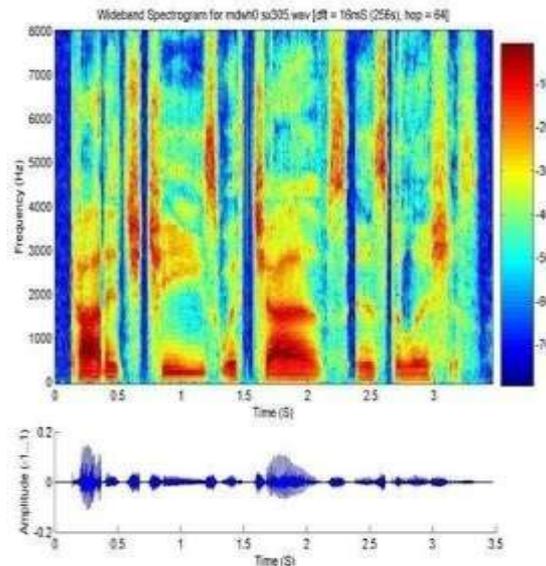


Fig. 2: Spectrogram Using Welch's Method (a) and Speech Amplitude (b) division of the discourse waveform into all around characterized states isn't straight forward. In any case, this trouble isn't as a major issue as one can might suspect.

IV. Related work

Many module have been used in this major project. Here, I will just list them below and describe the module as they are some of the most important module used in this and many major projects in Python.

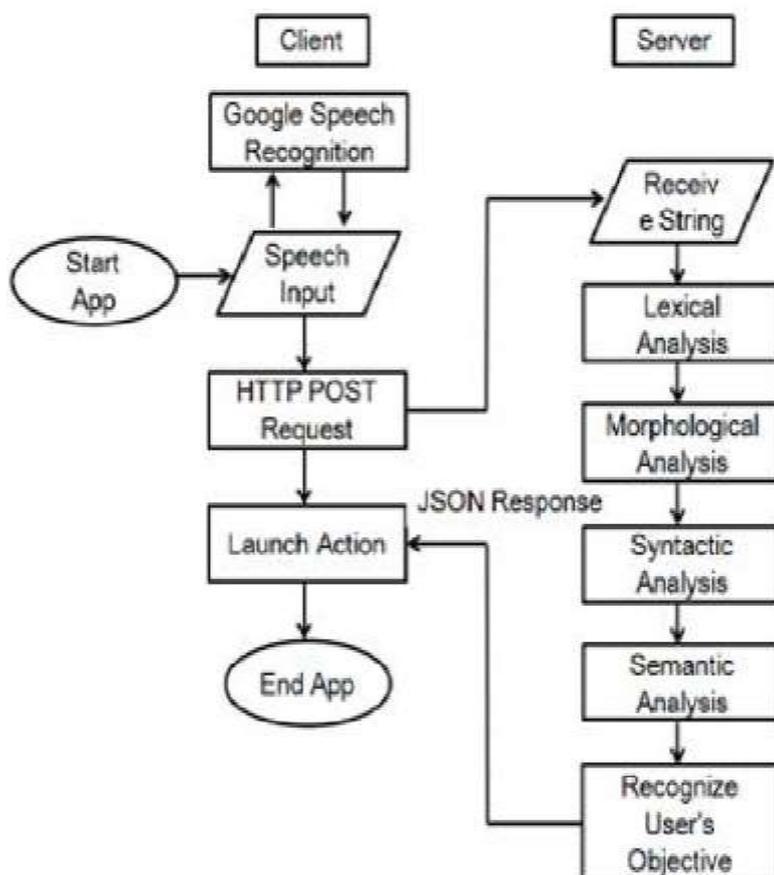
Text To Speech() Merge Audio() Generate Sketon()

V. Philosophies

As referenced in [11], voice acknowledgment works dependent on the reason that an individual voice displays attributes are one of a kind to various speaker. The sign during preparing and testing session can be significantly unique because of numerous variables, for example, individuals voice change with time, wellbeing condition (for example the speaker has a cool), talking rate and furthermore acoustical commotion and variety recording condition by means of mouthpiece. Table below gives detail data of recording instructional meeting, while

METHOD	SPECIFICATION
Voice Production	Random male and female within the age group of 20 & 60
Hardware	Microphone (Mono/Stereo) Voice recognition software (Cortana)
Premises	Quiet and Vivid
Utterance	Random lines from the given document
Frequency sampling	Around 16 GHz
Computational Features	MFCC coefficient of 39 double delta

Figure 6 shows the flowchart for by and large voice acknowledgment process.



VI. Result and Discussion

The information voice signals of two distinct speakers are Figure

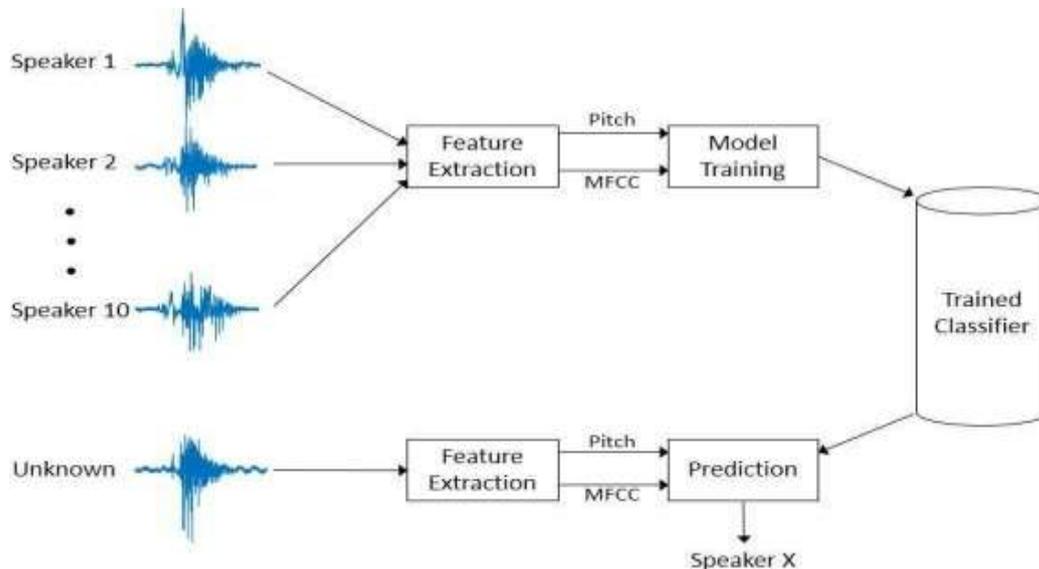


Fig. 7: Example Voice Signal Input of n Difference Speakers

VII. Conclusion

This paper has examined voice acknowledgment calculations which are significant in improving the voice acknowledgment execution. The procedure had the option to verify the specific speaker dependent on the individual data that was remembered for the voice signal. The outcomes show that these systems could utilize successfully for voice acknowledgment purposes. A few different systems, for example, Liner Prescient Coding (LPC), Dynamic Time Wrapping (DTW), and Counterfeit Neural System (ANN) are at present being examined. The discoveries will be displayed in future distributions.

VIII. References-

- [1] Rabiner Lawrence, Juang Bing-Hwang. Fundamentals of Speech Recognition Prentice Hall , New Jersey, 1993.
- [2] Deller John R., Jr., Hansen John J.L., Proakis John G. ,Discrete-Time Processing of Speech Signals, IEEE Press.
- [3] Hayes H. Monson,Statistical Digital Signal Processing and Modeling, John Wiley & Sons Inc. , Toronto, 1996.
- [4] Proakis John G., Manolakis Dimitris G.,Digital Signal Processing, principles, algorithms, and applications, Third Edition, Prentice Hall , New Jersey, 1996.

- [5] Ashish Jain,Hohn Harris,Speaker identification using MFCC and HMM based techniques,university Of Florida,April 25,2004.
- [6] <http://www.cse.unsw.edu.au/~waleed/phd/html/node38.html> ,downloaded on 2 Oct 2012.
- [7] <http://web.science.mq.edu.au/~cassidy/comp449/html/ch11s02.html>, downloaded on 2 Oct 2012.
- [8] Hiroaki Sakoe and Seibi Chiba, Dynamic Programming algorithm Optimization for spoken word Recognition, IEEE transaction on Acoustic speech and Signal Processing, February 1978.
- [9] Young Steve,A Review of Large-vocabulary Continuous-speech Recognition, IEEE SP Magazine, 13:45-57, 1996.
- [10] <http://www.microsoft.com/MSDN/speech.html>, downloaded on 2Oct 2012.
- [11] Davis K. H., Biddulph R. and Balashek S.,Automatic Recognition of Spoken Digits, J. Acoust. Soc. Am., 24 (6):637-642, 1952
- [12] Mammone Richard J., Zhang Xiaoyu, Ramachandran Ravi P.,Robust Speaker Recognition, IEEE SP Magazine, 13:58-71, 1996..