Real Time Industrial Devices Controlled By Human Voice

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Abstract:-Voice Recognition Systems are available but voice recognition software being closed source cannot be used easily for implementation of voice recognition based devices. To implement the approach on a real-time application, an arm 11 device interface is to be designed to control the applications on Linux platform, it being an open source using Qtcreator software. Idea behind this research work came from one paper which consists of system of control various speech Devices. By using Various Speech Commands according to Human Voice, We Can Control Various Industrial Devices like Speed of Motor. Speech is a key input For the Controlling of the System. By using Speech Recognition technique for controlling of industrial Devices, When we speak start or stop into the Microphone The Raspberry Pi B+(arm11) follow that command and Performed that operation on industrial equipment Like Bulb (On/Off), Dc Motor (On/Off),Fan (On/Off),Submersible Pump (On/Off). We Can Provide User Interface By Train For More Command.

Keywords:- Qtcreator, Voice Recognition, Lexical Decoding, Phenomes, Linux, Speech Recognition, Raspberry Pi B+ Board.

I. INTRODUCTION

Day by day our life becomes busy. We have to do a lot of work every day. For these purpose we use many kinds of Industrial devices Control system. If we control the Industrial Devices system using Speech then we can save enough time to do other sophisticated work. There are several different possibilities for Device control for physically disabled and handicaps persons. In many cases speech is the most convenient and easy to learn alternative. Thus, our basic principle is that the system should speech detect easily expandable by controlling the various industrial Devices to the largest possible extent.Speech recognition is usually processed, the results are transmitted to the user application, in electrical engineering, and speech recognition is the translation of speech words.The speech recognition is a ability of a machine or program to identify words and phrases in spoken language and convert them to a machine readable format .Rudimentary speech recognition software has a limited vocabulary of words and phrases and many only identify these if they are spoken very clearly more sophisticated has the ability to accept natural speech. Here, I used speech recognition technique in industrial devices.When we speak start or stop into the microphone the speech recognition follow that command and performed that operation. The industrial Bulb (On/Off) & Motor (On/Off & Slow/ Fast), Fan (On/Off), Submersible Pump (On/Off).on speech recognition. This is a real time processing system. It will detect the voice. Hence we will use speech to control some applications. With the use of human speech, some applications can be controlled, for the same, we will develop said system. Recognizes various speech .we can control various application based on speech .it can be installed on already existed system .it is based on LINUX which is open source and divers for it is easily available.

II. SYSTEM BASIC BLOCK DIAGRAM, QT CREATOR FLOWCHART,SYSTEM ALGORITHM, SYSTEM FLOWCHART

A. System Basic Block Diagram

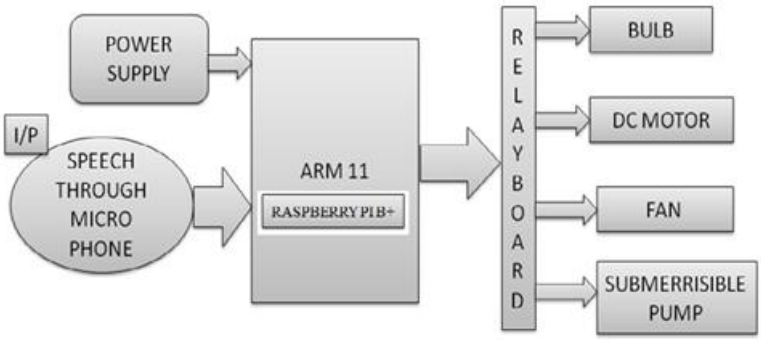

Fig.1: System basic block diagram

Above Block Diagram represented the Speech Recognition technique. The speech (voice) will be input of the microphone and it will recognize that voice in relay board. It will attached various industrial Devices control like equipment Like Bulb (On/Off), Dc Motor (on/off) as a same thing it also working as Fan (On/Off),Submersible Pump (On/Off).Voice will convert that analog signal and Raspberry Pi B+(Arm 11) control Through above system to control the real time of a industrial devices.

B. Qt Creator Flowchart

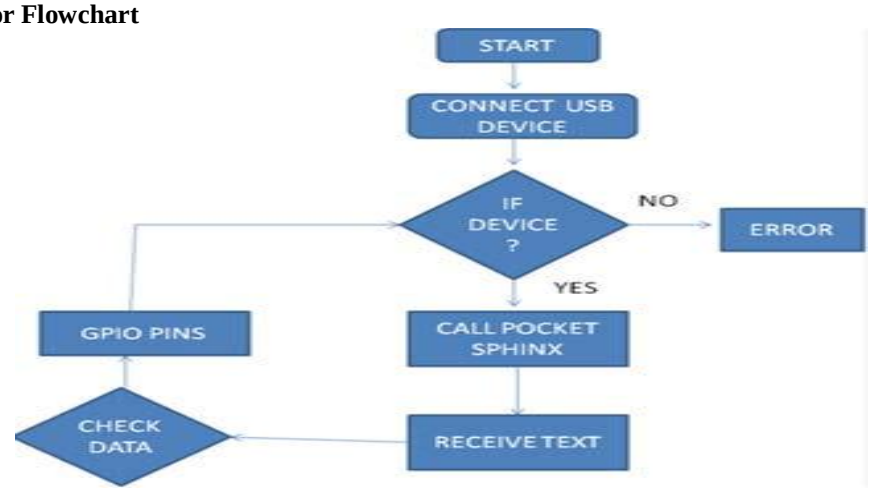

Fig.2: Qt Creator Flowchart

C. System Algorithm

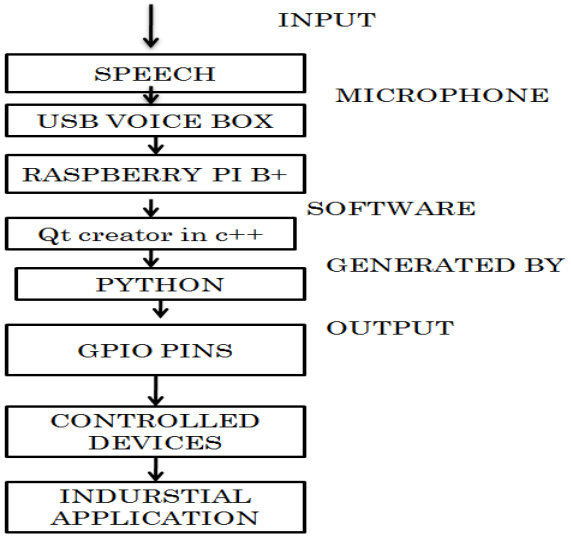

Fig.3: System Algorithm

D. System Flowchart

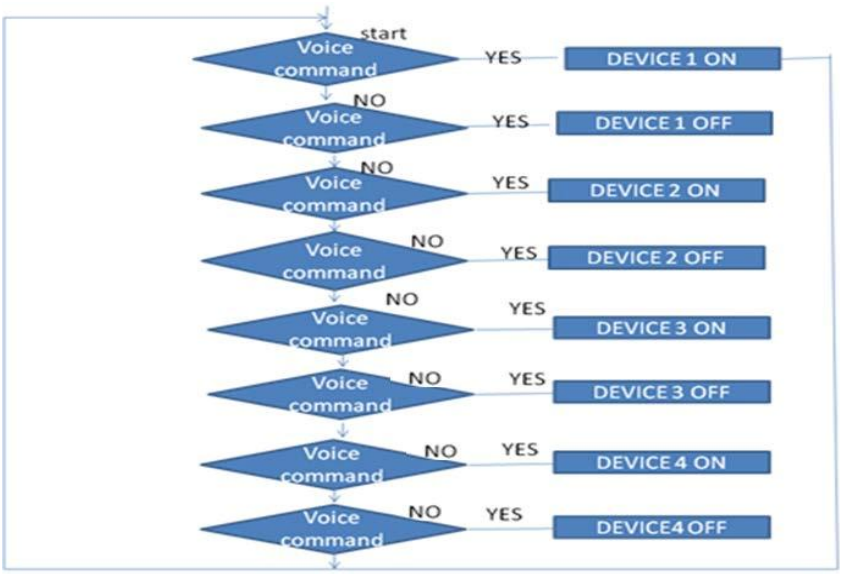

Fig.4: System Flowchart

III. SPEECH RECOGNITION SYSTEM[6]

Speech recognition basically means talking to a computer, having it recognize what we are saying, and lastly, doing this in real time. This process fundamentally functions as a pipeline that converts PCM (Pulse Code Modulation) digital audio from a sound card into recognized speech.

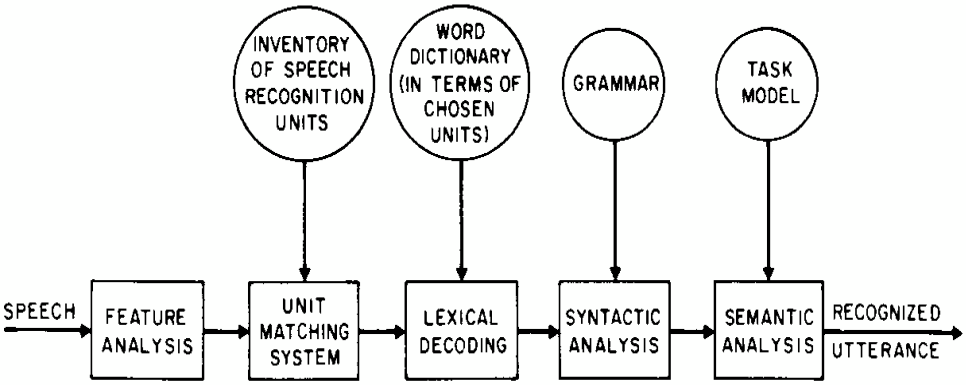

Fig.5: Block Diagram of Speech Analysis[6]

A. Transform the PCM digital audio into a better acoustic representation

The input to speech recognizer is in the form of a stream of amplitudes, sampled at about 16,000 times per second. But audio in this form is not useful for the recognizer. Hence, Fast-Fourier transformations are used to produce graphs of frequency components describing the sound heard for 1/100th of a second. Any sound is then identified by matching it to its closest entry in the database of such graphs, producing a number, called the "feature number" that describes the sound.

B. Unit matching system

Provides likelihoods of a match of all sequences of speech recognition units to the input speech. These units may be phones, diphones, syllables or derivative units such as fenones and acoustic units. They may also be whole word units or units corresponding to group of 2 or more words. Each such unit is characterized by some HMM whose parameters are estimated through a training set of speech data.

C. Lexical Decoding

Constraints the unit matching system to follow only those search paths sequences whose speech units are present in a word dictionary.

D. Apply a "grammar"

So the speech recognizer knows what phonemes to expect. This further places constraints on the search sequence of unit matching system. A grammar could be anything from a context-free grammar to full-blown English.

E. Figure out which phonemes are spoken

This is a quite dicey task as different words sound differently as spoken by different persons. Also, background noises from microphone make the recognizer hear a different vector. Thus a probability analysis is done during recognition. A hypothesis is formed based on this analysis. A speech recognizer works by hypothesizing a number of different "states" at once. Each state contains a phoneme with a history of previous phonemes. The hypothesized state with the highest score is used as the final recognition result.

Related Work

A lot of speech aware applications are already there in the market. Various dictation software have been developed by Dragon, IBM and Philips. Genie is interactive speech recognition software developed by Microsoft. Various voice navigation applications, one developed by AT&T, allow users to control their computer by voice, like browsing the Internet by voice. Many more applications of this kind are appearing every day .The SPHINX speech recognizer of CMU provides the acoustic as well as the language models used for recognition. It is based on the Hidden Markov Models (HMM). The SONIC recognizer is also one of them, developed by the University of Colorado. There are other recognizers such as XVoice for Linux that take input from IBM's Via Voice which, now, exists just for Windows. Background noise is the worst part of a speech recognition process. It confuses the recognizer and makes it unable to hear what it is supposed to. One such recognizer has been devised for robots that, despite of the inevitable motor noises, makes it communicate with the people efficiently. This is made possible by using a noise-type-dependent acoustic mode l corresponding to a performing motion of robot. Optimizations for speech recognition on a HP Smart Badge IV embedded system has been proposed to reduce the energy consumption while still maintaining the quality of the application. Another such scalable system has been proposed in for DSR (Distributed Speech recognition) by combining it with scalable compression and hence reducing the computational load as well as the bandwidth requirement on the server. Various capabilities of current speech recognizers in the field of telecommunications are described in like Voice Banking and Directory Assistance.

A. Voice Input

The input is human voice which, as explained before, is sampled at rate of 16,000 Per second. It should be given in live mode. But because of some conflicts in the channel settings of the sound card and that used by the software, we are not able to do it in live mode. We are Running the recognizer in batch mode, instead, i.e. taking input in the form of a pre-recorded Audio file (in RAW format).

B. Microphone

The microphone that we are using for recognition is built onto the PXA27xPlatform itself. It has got its own advantages and disadvantages.

Advantages

- Nothing to plug in.
- User's hands are free.

Disadvantages

- Low accuracy unless the user is close to the monitor.
- Not good in a noisy environment.

C. Types Of Voice Recognition System

- **Speaker dependent system** - The voice recognition must be trained before it can be used. This often requires a user reads a series of words and phrases so the computer can understand the users voice.
- **Speaker independent system** - The voice recognition software recognizes most users' voices with no training.
- **Discrete speech recognition** - The user must pause between each word so that the speech recognition can identify each separate word.

- **Continuous speech recognition** - The voice recognition can understand a normal rate of speaking.
- **Natural language** - The speech recognition not only can understand the voice but also return answers to questions or other queries that are being asked.

To run the programme

IV. SIMULATIONRESULTS

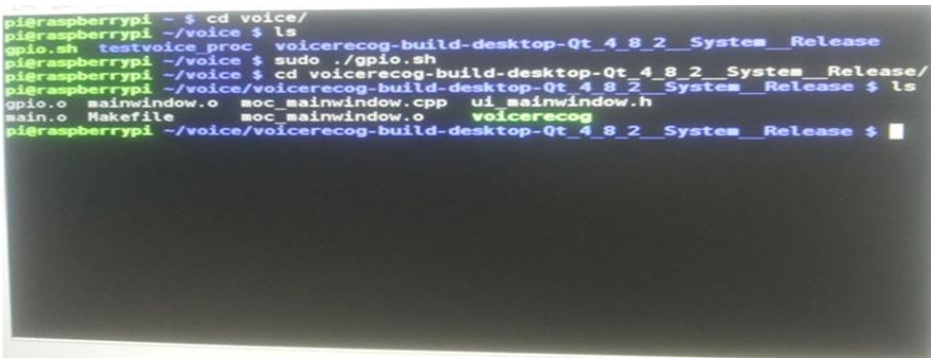

Fig.6: Run the Programme Mic in input LIGHT ON and speech is converted into text in main window and output is shown below.

Fig.7: Mic input LIGHT ON and main window is converted to the speech to text

This figure shows the output according to input speech. For example if we say that light on, then the bulb is glow as shown in below figure.

Fig.8: Real time bulb is on

Mic in input LIGHT OFF and speech is converted into text in main window and output is shown below.

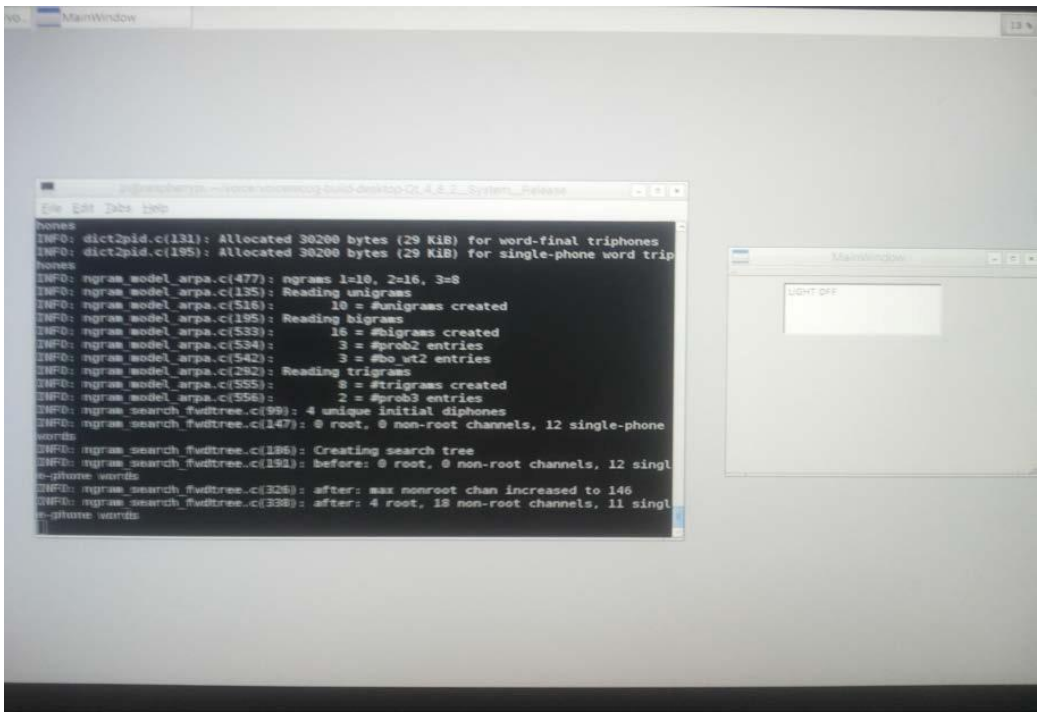

Fig.9: Mic input LIGHT OFF and main window is converted to the speech to text

Similarly when we give 'light off' as a input speech then bulb is off as shown below.

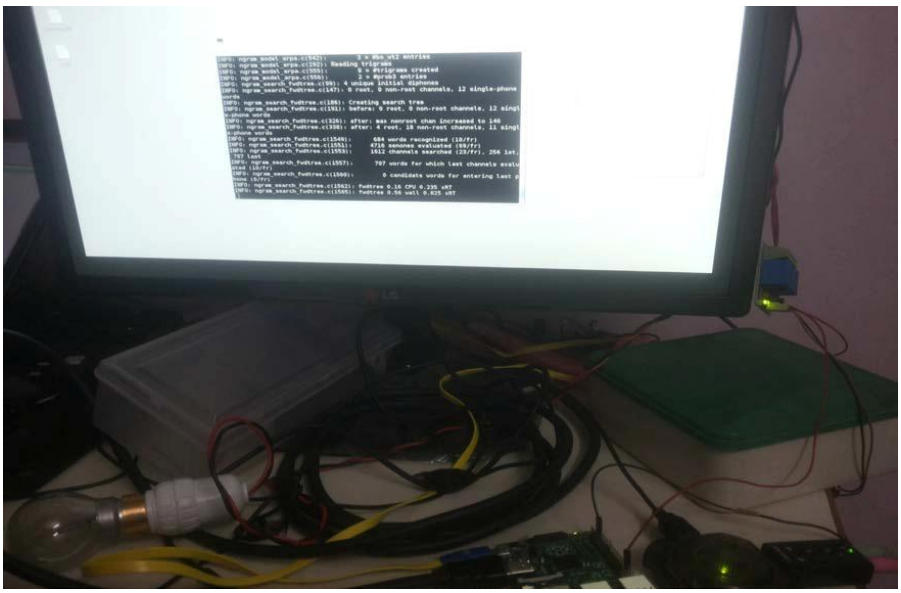

Fig.10: Real time bulb is off

V. CONCLUSION

From my research I conclude that I will be using open source Qtcreator software based on LINUX platform to control an arm11 (Raspberry pi b+) device through voice commands. I have developed four applications which are controlled through voice command. Of this system is to provide a way the physically disable population can easily control many functions of any device via speech. We Can Provide User Interface By Train For More Command.

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