# Electronic Control System Of Home Appliances Using Speech Command Words

Aye Min Soe, Maung Maung Latt, Hla Myo Tun, Zaw Min Naing

**Abstract:** The main idea of this paper is to develop a speech recognition system. By using this system, smart home appliances are controlled by spoken words. The spoken words chosen for recognition are “Fan On”, “Fan Off”, “Light On”, “Light Off”, “TV On” and “TV Off”. The input of the system takes speech signals to control home appliances. The proposed system has two main parts: speech recognition and smart home appliances electronic control system. Speech recognition is implemented in MATLAB environment. In this process, it contains two main modules: feature extraction and feature matching. Mel Frequency Cepstral Coefficients (MFCC) is used for feature extraction. Vector Quantization (VQ) approach using clustering algorithm is applied for feature matching. In electrical home appliances control system, RF module is used to carry command signal from PC to microcontroller wirelessly. Microcontroller is connected to driver circuit for relay and motor. The input commands are recognized very well. The system is a good performance to control home appliances by spoken words.

**Index Terms**: MATLAB Speech recognition, Feature Extraction, Feature Matching, Mel Frequency Cepstral Coefficient (MFCC), Vector Quantization (VQ),PIC16F887, KST-TX01, KST-RX706

## I. INTRODUCTION

Speech Recognition is a technology allowing the computer to identify and understand words spoken by a person using a microphone or telephone. Using a set of pre-programmed commands and instructions, user can talk with computer. Computer system that understands input speech enables user to have conversations with the computer. User and the computer speaking as commands or in response to events, input, or other feedback would be included in these conversations. Speaking is easier and more sensitive than selecting buttons and menu items. Human speech has changed over many thousands of years to become an efficient method of sharing information and giving instructions. [1] The goal of this paper is to implement smart home appliances controlled system that can be operated by spoken words. The spoken words chosen for recognition are “Fan On”, “Fan Off”, “Light On”, “Light Off”, “TV On” and “TV Off”. The proposed system is composed of two main parts: speech recognition system and home appliances control system. The first and main part of system is personal computer based speech recognition system. Speech recognition system is composed of also two main parts. The first part is speech recognition based on Digital Signal Processing. Second part is interfacing with hardware. The overall block diagram is shown in Fig. 1.

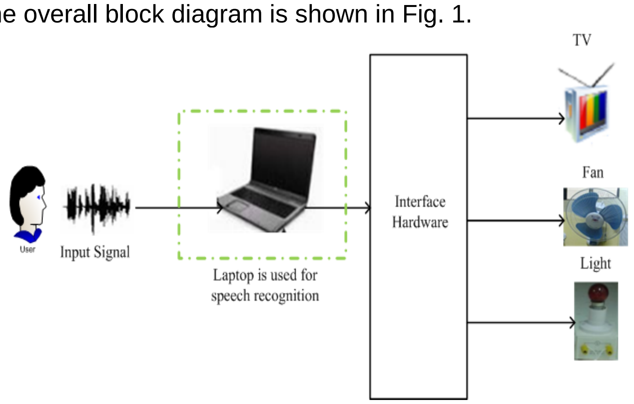



***Fig.1*** *Overall System Proposed System*

## II. SPEECH RECOGNITION SYSTEM

A speech recognition roughly consists of two portions. They are speech analysis and pattern recognition.

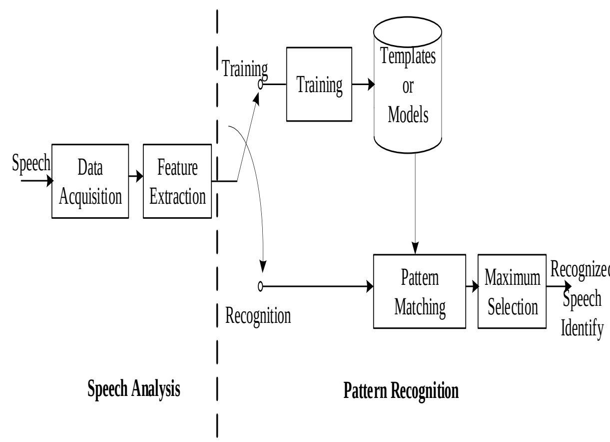



***Fig. 2*** *Block Diagram of Speech Recognition System*

### A. Speech Analysis

The purpose of the speech analysis block is to transform the speech waveform into a parsimonious representation which characterizes the time varying properties of the speech. The speech analysis typically includes two modules, namely data acquisition and feature extraction. The data acquisition module usually contains a microphone and a code from which digitized speech data are generated. The feature extraction is the computation of a sequence of feature vectors which provides a compact representation of the given speech signal. The feature extraction is done on short-time basis. The speech signal is separated into overlapped fixed-length frames. From each frame, a set of frequency-domain or cepstral-domain parameters are derived from each frame, to form the so-called feature vector. There are some basic principles and analysis techniques used in the feature extraction module. They are pre-emphasis, frame blocking and windowing, Discrete Fourier Transform (DFT) computation, spectral magnitudes, Mel-frequency filter bank, logarithm of filter energies, Discrete Cosine Transformation (DCT), Cepstral Weighting, and dynamic featuring.

### B. Pattern Recognition

The speech signal is first analyzed and a feature representation is obtained for comparison with either stored reference templates or statistical models in the pattern matching block. A decision scheme determines the word or phonetic class of the unknown speech based on the matching scores with respect to the stored reference patterns. There are two types of reference patterns. The first type, called a nonparametric reference pattern (or often a template), is a pattern created from one or more spoken tokens of the sound associated with the pattern. The second type, called a statistical reference model, is created as a statistical characterization of the behavior of a collection of tokens of the sound associated with the pattern. The vector quantization model [6] is used as the statistical model. There are three portions in pattern recognition. They are pattern training, pattern matching and maximum selection.

**1. Pattern Training**
Pattern training is the method by which representative sound patterns are converted into reference patterns for use by the pattern matching algorithm. There are several ways in which pattern training can be performed, including: Casual training in which a single sound pattern is used directly to create either a template or a crude statistical model. Robust training in which several versions of the sound pattern are used to create a single merged template or statistical model. Clustering training in which a large number of versions of the sound pattern is used to create one or more templates or a reliable statistical model of the sound pattern.

**2. Pattern Matching**
Pattern matching refers to the process of assessing the similarity between two speech patterns, one of which represents the unknown speech and one of which represents the reference pattern (derived from the training process) of each element that can be recognized. When the reference pattern is a "typical" utterance template, pattern matching produces a gross similarity (or dissimilarity) score. When the reference pattern consists of a probabilistic model, the process of pattern matching is equivalent to using the statistical knowledge contained in the probabilistic model to assess the likelihood of the speech (which led to the model) being realized as the unknown pattern. Pattern matching refers to the process of assessing the similarity between two speech patterns, one of which represents the unknown speech and one of which represents the reference pattern (derived from the training process) of each element that can be recognized.

**3. Maximum Selection**
The decision strategy takes all the matching scores (from the unknown pattern to each of the stored reference patterns) into account, finds the "closest" match, and decides if the quality of the match is good enough to make a recognition decision. If not, the user is asked to provide another token of the speech (e.g., the word or phrase) for another recognition attempt. This is necessary because often the user may speak words that are incorrect in some sense (e.g., hesitation, incorrectly spoken word, etc.) or simply outside of the vocabulary of the recognition system.

### C. Vector Quantization (VQ)
Vector Quantization (VQ) is a classical quantization technique from signal processing which allows the modeling of probability density functions by the distribution of prototype vectors. It was originally used for data compression. It works by dividing a large set of points (vectors) into groups having approximately the same number of points closest to them. Each group is represented by its centroid point, as in k-means and some other clustering algorithms. The density matching property of vector quantization is powerful, especially for identifying the density of large and high-dimensioned data. Since data points are represented by the index of their closest centroid, commonly occurring data have low error, and rare data high error. This is why VQ is suitable for lossy data compression. It can also be used for lossy data correction and density estimation. Vector quantization is based on the competitive learning paradigm, so it is closely related to the self-organizing map model. VQ was also used in the eighties for speech and speaker recognition. Recently it has also been used for efficient nearest neighbour search and on-line signature recognition. In pattern recognition applications, one codebook is constructed for each class (each class being a user in biometric applications) using acoustic vectors of this user. In the testing phase the quantization distortion of a testing signal is worked out with the whole set of codebooks obtained in the training phase. The codebook that provides the smallest vector quantization distortion indicates the identified user. Vector Quantization (VQ) approach is applied for training and classification phase. Firstly, the training set of vectors is used to create the optimal set of codebook vectors for representing the spectral variability observed in the training set. And Then distance is measured between a pair of spectral analysis vectors to able to cluster the training set vectors as well as to classify spectral vectors into unique codebook entries. The next step is a centroid computation procedure. Finally, a classification procedure selects the codebook vectors that closet to the input vector and uses the codebook index as the resulting spectral representation. The classification procedure is essentially a quantizer. It accepts speech spectral vectors as input and provides the code index of the code vectors that best matches the input.

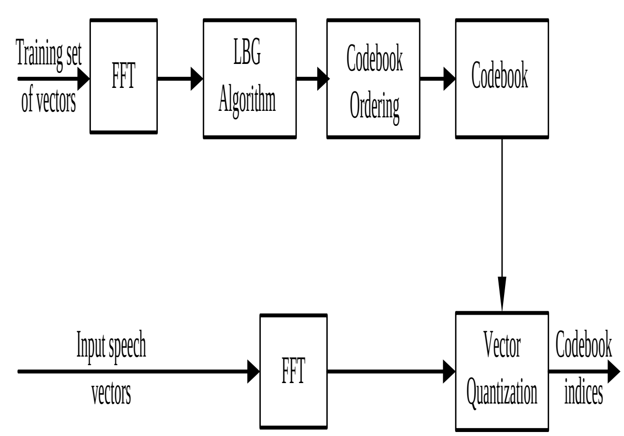



***Fig.3*** *Block Diagram of the VQ Training and Classification Phase*

## III. Electronic Control System
The block diagram of home appliances control system for speech recognition is illustrated in Fig.1. In this diagram, speech instruction is firstly taken as input to control home appliances and then a microphone is used to record the person speech. Secondly, the speech instruction is caught and transferred the analog signal to digital signal and the recorded speech is sent to the speech based verification/identification system. Thirdly, the digital information of speech instruction is processed and compared by using the MATLAB programming. Fourthly, the digital information of speech instruction is

outputted through USB port. Finally, PIC receives data from speech recognition block and gives instructions to control home appliances. The control system consists of two sections: transmission section and receiving sections.

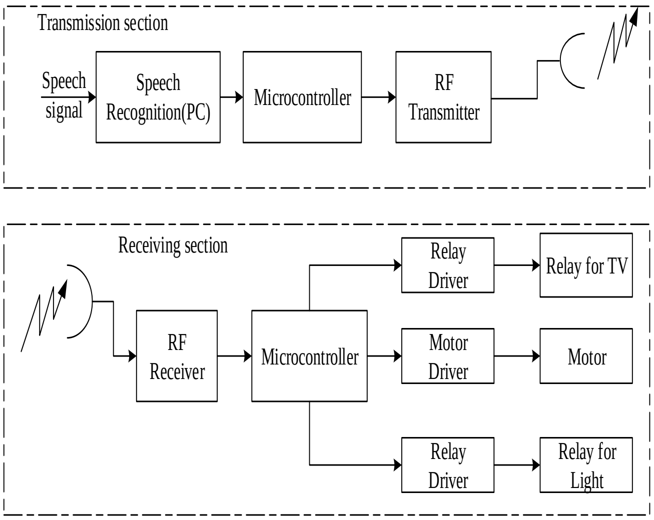



***Fig.4** Block Diagram of Home Appliances Control System for Speech Recognition*

## A. Transmission Section

In the transmission section, there are KS232 module, PIC 16F887 and KST-TX01 (Radio Frequency transmitter module). The KS232 module is used to carry the signal from PC to Microcontroller unit. The signal is retransmitted with baud rate 1200 for RF transmission by KST-TX01 module. This module has four pins: supply pin, data pin, GRN pin, and ANT pin. KST-TX01 technical specific data for wireless transmitter module are (1) transmit power: 1W, (2) operating frequency: 315MHZ~433.92MHZ, (3) operating temperature:-40℃~80℃, (4) operating voltage: 3V~5V and (5) modulation type: ASK.

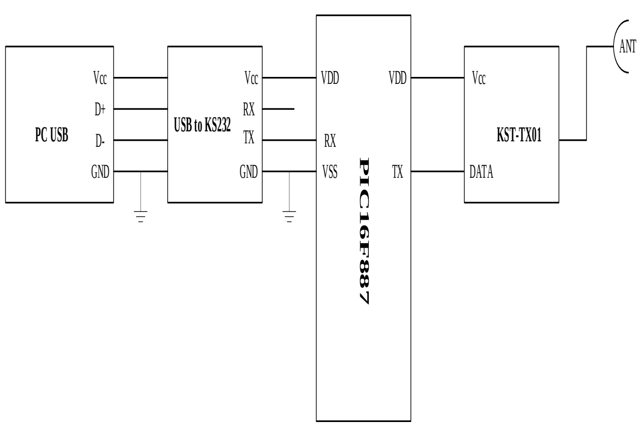



***Fig.4** Circuit Diagram of Transmission Section*

## B. Receiver Section

The receiver section consists of KST-RX706 (RF receiver module), PIC microcontroller, relays, relay drivers and motor driver. In this section, KST-RX706 firstly accept radio signal and then microcontroller read radio signal with baud rate 1200. Microcontroller drives relay and motor driver. The speed of motor is controlled by using Pulse Wide Modulation (PWM) module.

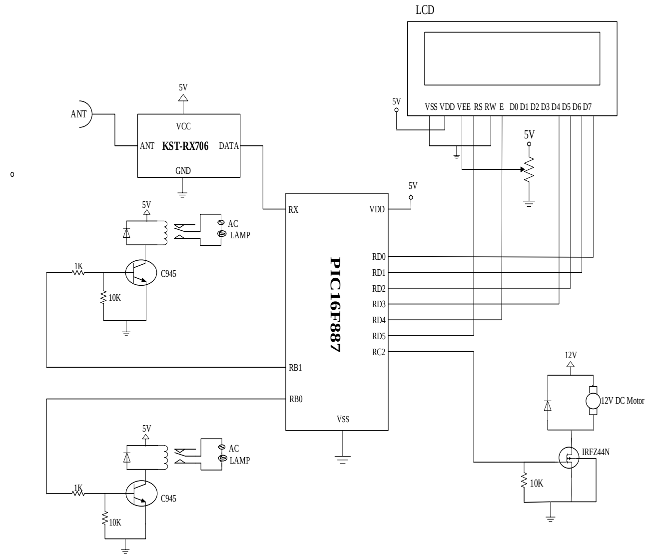



***Fig.5** Circuit Diagram of Receiving Section*

## A. PWM Method for Motor Speed Control

Pulse width modulation (PWM) is a method for binary signals generation, which has two signal periods (high and low). The width (W) of each pulse varies between 0 and the period (T). The main principle is control of power by varying the duty cycle. Here the conduction time to the load is controlled. The duty cycle can be varied from 0 to 1 by varying $t_{on}$ or T. Therefore, the average output voltage $V_{avr}$ can be changed between 0 and $V_{in}$ by controlling the duty cycle, thus, the power flow can be controlled. The on-off switching is performed by power MOSFETs. A MOSFET is a device that can turn very large currents on and off under the control of a low signal level voltage. PWM signal shown in Figure 7.

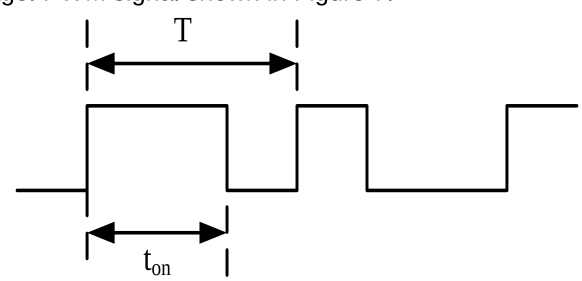



***Fig.7** PWM Singal*

## IV. SOFTWARE IMPLEMENTATION

This section explains the methods used for speech recognition. These methods are training phase and testing phase.

## A. Training Phase

Initially, the user must prepare the training files. The speech files are recorded from the microphone and MFCC features are extracted from the input file. Then these features are stored. In this case, the collection of training files is called database. Then, the user must train the system using the files in the database. This is called training phase or pre-processing. Figure.5 shows the flow chart of the step of training phase.

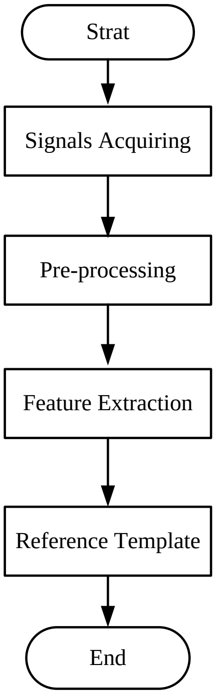



***Fig. 4*** *Flow Chart of the step of Training Phase*

## B. Testing Phase

In the testing phase, users have to provide the command words as input. In this case, user may use two ways of testing. If user chooses to use the pre-recorded sound file, one of the samples are loaded from test files and read. Then, the modified MFCC features are extracted from the input file. In the next step, the distances between the modified MFCC features and the stored reference models are calculated using Euclidean Distance. Finally, the minimum distance is selected among the distances between the input vectors and codebook vectors. If this minimum distance falls below the local threshold, the system outputs the command word as result. Otherwise, the system determines it is wrong command word. If the user wants to test the system with spoken commands in real time, the sound file to be recognized is recorded from the microphone. To do so, the user must choose time length Typical time length is 2 seconds. In this system, sound files are recorded within this time length. Then the subsequence processes, as above, are carried out and recognition decision.

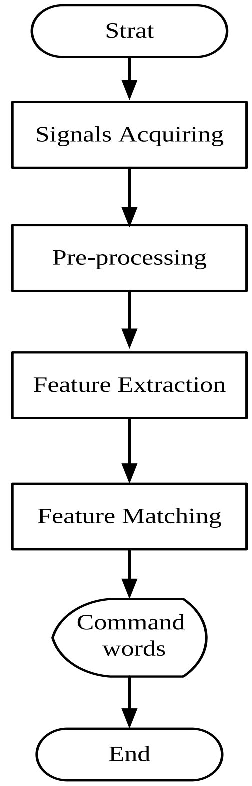



***Fig. 5*** *Flow Chart of the step of Testing Phase*

In this fig.7, there are three portions as like button, namely: feature extraction, codebook, and test. Firstly, the button of feature extraction is on, speech signal waves and spectrograms of each of speech signals are shown in the right of figure. Secondly, codebook button is on, feature coefficients of each of speech signals are shown as a table. Above processes are called training phase. If speech signal is tested, test button is on.

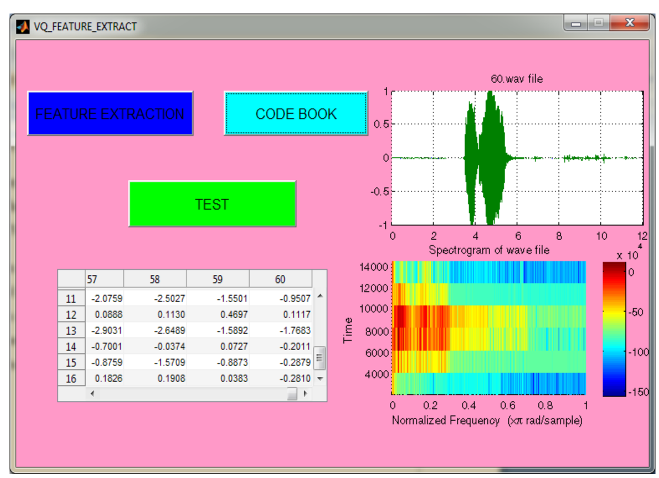



***Fig.7*** *Training phase*

There are two ways of testing, namely: real time and offline. If speech signal is tested in real time, the speech signal is recorded from microphone when record button is on. And feature coefficient of speech signal is presented in table while show code button is on. Finally, speech signal is recognized. In offline, speech signal is loaded from testing files. And then steps in offline are same steps in real time.

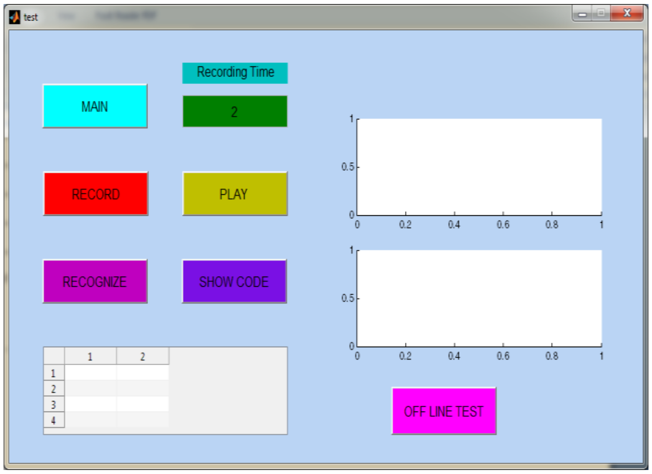



***Fig.6*** *Testing Phase*

When the user speaks “TV on” in real time, the minimum distance between input vector and code book vector is 0.0106 and execution time is 0.04424s. When speaking “TV on”, command “TV on” is shown in the massage box.

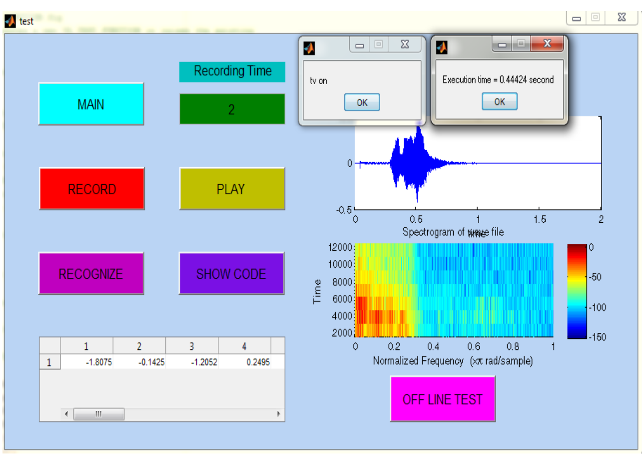
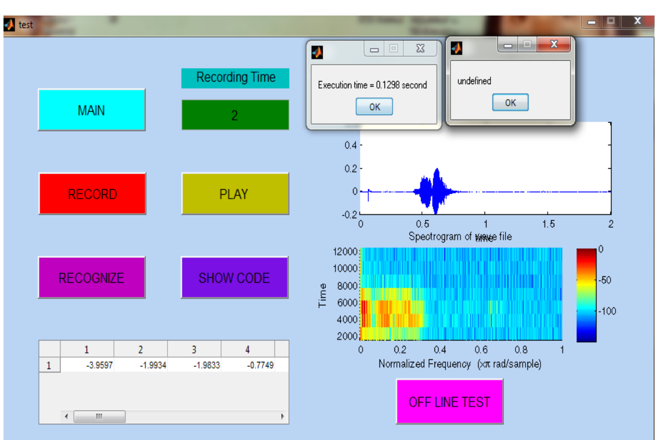



***Fig.7*** *Result when speaking “TV on” in real time*

When the user speaks a command that does not include in trained commands, the massage box shows “undefined”.




This table includes command words, minimum distance between input vector and code book vector, maximum distance between input vector and code book vector, and execution time.

***Table 1:*** *Distances between Input Vectors and Code Book Vectors in Real Time*

| **Command words** | **Distance between input vector and code book vector (Minimum)** | **Distance between input vector and code book vector (Maximum)** | **Execution time** |
|---|---|---|---|
| TV on | 0.0106 | 1.8075 | 0.44424 |
| TV off | 0.0873 | 2.5706 | 0.05861 |
| Light on | 0.0113 | 2.2317 | 0.05509 |
| Light off | 0.0960 | 1.5808 | 0.070341 |
| Fan on | 0.0144 | 1.6128 | 0.063073 |
| Fan off | 0.1764 | 3.7751 | 0.061763 |

## V. RESULT AND DISCUSSION

When the user speaks “TV ON” command from microphone, the command is recognized by personal computer using MATLAB software. The recognized command word is input into microcontroller through USB port. In microcontroller, “TV ON” command word is assigned as character “a”. When character “a” is transmitted to one microcontroller, another microcontroller receives the character “a”. And then the signal is driven relay driver to turn on TV.Fig.9 shows the result of “Light On” command word. “Light On” command word is given as the character “c” in microcontroller and “Fan On” command word as the character “e”. Before testing the real time communication between transmission and receiving section, PIC to PIC serial communication is tested firstly. Figure 6 shows simulation of PIC to PIC wire communication for “Fan on” command. In this figure, it consists of two separate part circuit. One is for reading data from PC and these data would then be stored in a user file. In data communication, it is needed to transmit a start bit for transmitted bits in user file.

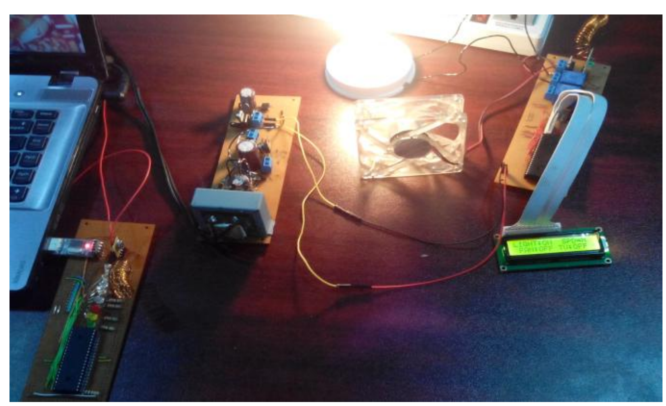


***Fig. 8*** *Speech Command Tests for Electronic Home Appliances*

After transmitting start bit, the data in user file will be serially transmitted to another PIC. This PIC receives the transmitted data and the output will go on if the receiving data is equal to the original data. If they are not equal, there is no output condition and the user must check until there is no error in program for the correct result.

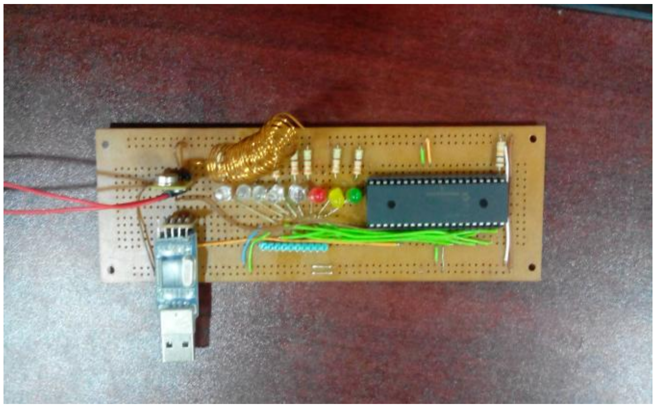


***Fig.9*** *Screenshot of Transmission Section of the System*

***Fig.10*** *Screenshot of Receiving Section of the System*

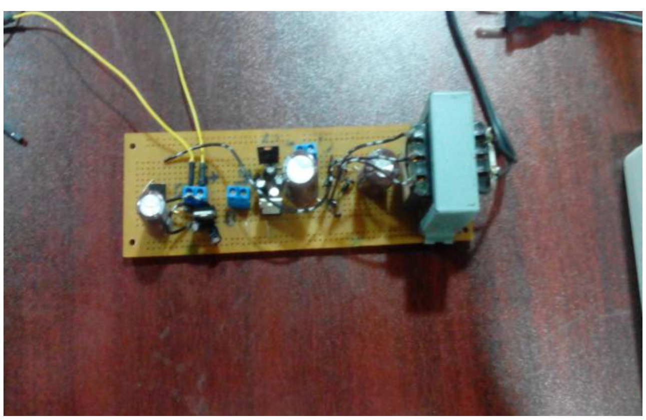


***Fig.11*** *Screenshot of 12V Power Supply used in the System*

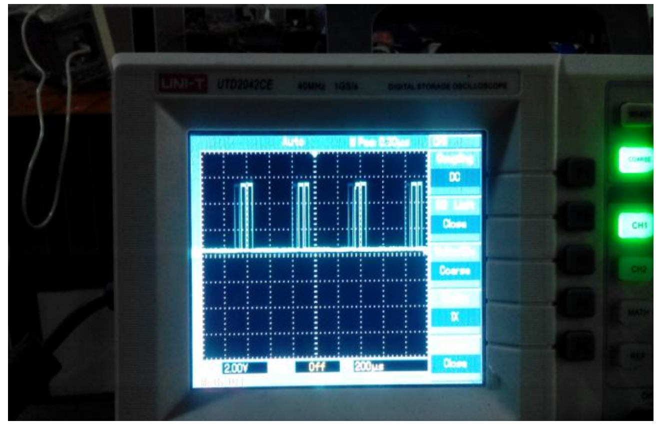


***Fig.12*** *Output Result of the Speed of Motor at Low Condition*

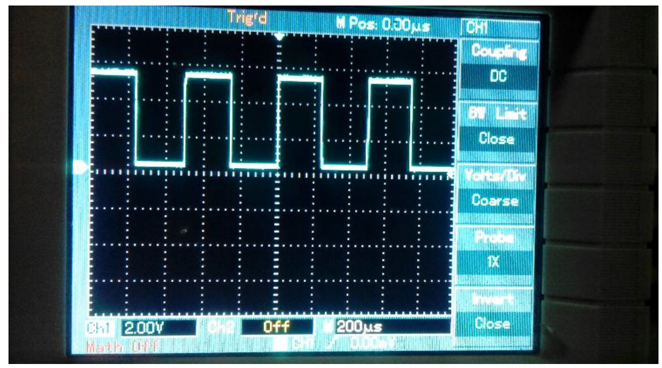



***Fig.13*** *Output Result of the Speed of Motor at Medium Condition*

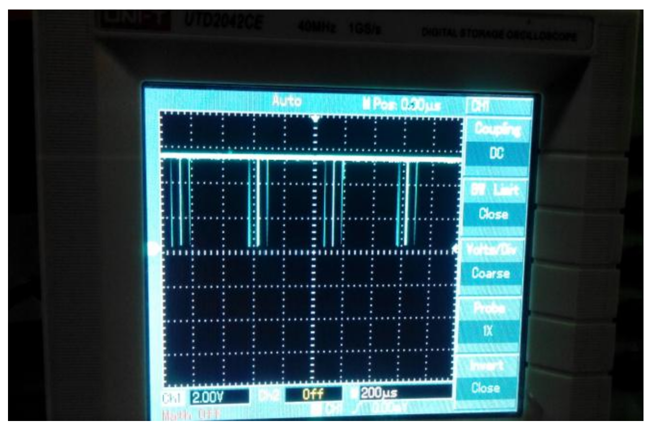


***Fig.14*** *Output Result of the Speed of Motor at High Condition*

## VI. Conclusion

The system has two main parts: speech recognition and smart home appliances electronic control system. Speech recognition is implemented in MATLAB environment. In this process, it contains two main modules: feature extraction and feature matching. Mel Frequency Cepstral Coefficients (MFCC) is used for feature extraction. Vector Quantization (VQ) approach using clustering algorithm is applied for feature matching. In electrical home appliances control system, RF module is used to carry command signal from PC to microcontroller wirelessly. Microcontroller is connected to driver circuit for relay and motor. The input commands are recognized very well. The system is a good performance to control smart home appliances by spoken words. A wireless based home automation system which can be controlled through spoken commands is presented in this paper. In the proposed system, wireless component is added by Radio Frequency( RF) module. An application for speech command processing is developed.

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