EMBEDDED BASED SPEED AND DIRECTION CONTROL FOR DC MOTOR USING ARM PROCESSOR

ANAND. K, MANIKANDAN.P, BALA SHANKAR.B, KRISHNA KUMAR.S

Abstract— **Embedded based speed and direction control for dc motor using speech recognition tech is proposed and also we can drive DC motor forward and reverse direction with different speed. This project makes it possible to use the speakers' voice to verify their identity and control access to DC motor. Mel Frequency cepstral co-efficient is used to recognize user speech and vector quantization is used increase Speech recognition accuracy.**

Index Terms— vector quantization ,MFCC, Speech Recognition

I. INTRODUCTION

Speech is one of the natural forms of communication. Recent development has made it possible to use this in the security system. In speech recognition, the task is to use speech sample to select the identity of the person that produced the speech from among a population of speakers. This paper makes it possible to use the speakers' voice to verify their identity and control access to DC motor. This approach can be used such as voice dialing, banking by telephone, to drive electrical vehicles, telephone shopping, database access services, information services, voice mail, security control for confidential information areas, and remote access to computers . The MFCC algorithm for speech recognition is more accurate then Linear Prediction Coding (LPC) and Hidden Markova Model (HMM). The external DC motor is connected through interfacing between computer and hardware circuit. The hardware circuit consists of arm processor, IC MAX 232, driver IC.

II. PRINCIPLES OF SPEECH RECOGNITION

Speaker recognition methods can be divided into text independent and text-dependent methods. In a text independent system, speaker models capture characteristics of somebody's speech which show up irrespective of what one is saying. In a text-dependent system, the recognition of the speaker's identity is based on user's speaking one or more specific phrases, like passwords, card numbers, PIN codes, etc. Every technology of speaker recognition, identification and verification, whether text independent and text-dependent, each has its own advantages and disadvantages and may require different treatments and techniques. The choice of which technologies to use is application-specific. At the highest level, all speaker recognition systems contain two main modules feature extraction and feature matching.

Manuscript received Aug 20, 2014
ANAND. K, ASSOCIATE PROFESSOR, PITAM, INDIA
MANIKANDAN.P, ASSOCIATE PROFESSOR, PITAM, INDIA
BALA SHANKAR.B, ASSOCIATE PROFESSOR, PITAM, INDIA
KRISHNA KUMAR.S, ASSOCIATE PROFESSOR, PITAM, INDIA

BLOCK DIAGRAM

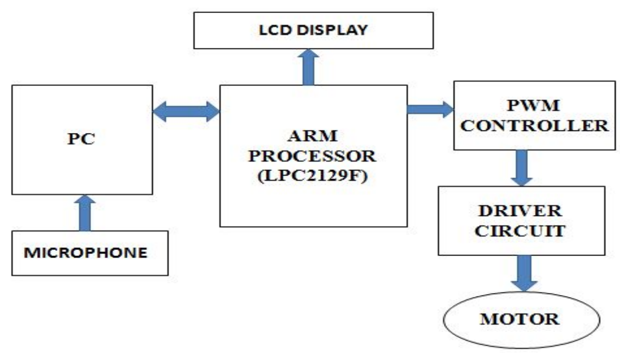

Fig 1: Block diagram

III.BLOCK DIAGRAM DESCRIPTION

when speech is given microphone can be used to convert the acoustic wave into an analog signal. Computer can't understand analog signal so computer has to go through several complex steps .The analog to digital converter translates this analog wave into digital data that the computer can understand to do this, it samples, or digitizes, the sound by taking precise measurements of the wave at frequent intervals. People don't always speak at the same speed, so the sound must be adjusted to match the speed of the template sound samples already stored in the system's memory. When Matching samples occur. Data is transmitted through RS232 serial port to ARM processor. The ARM7TDMI-S is a general purpose 32-bit microprocessor, which offers high performance and very low power consumption.

The ARM¤ architecture is based on Reduced Instruction Set Computer (RISC) principles, which define duty cycle depends up on motor runs. here we using lpc2129f processor which comes pwm circuitry embedded in chip.

With the help of PWM tech we can run motor different speed as you already build in coding .here we using driver circuit for amplification purpose and is used to control another circuit or other component. They are usually used to regulate current flowing through a circuit or is used to control the other factors such as other components, some devices in the circuit. And also we use LCD display to display

IV. METHODOLOGY

A. *Feature Extraction Module-MFCC*

MFCC (Mel Frequency Cepstral Coefficients) is the most common technique for feature extraction. MFCC tries to mimic the way our ears work by analyzing the speech waves linearly at low frequencies and logarithmically at high frequencies. The block diagram represents the structure of a MFCC processor in Fig. 1. The speech input is recorded at sampling frequency rate of 8000 Hz. This sampling frequency is chosen to minimize the effects of aliasing in the analog -to digital conversion process. Figure 2 shows the block diagram of an MFCC processor.

B. Mel Frequency Wrapping

Psychophysical studies have shown that human perception of the frequency contents of sounds for speech signals does not follow a linear scale. Thus for each tone with an actual frequency, *f*, measured in Hz, a subjective pitch is measured on a scale called the 'mel' scale. The *mel-frequency* scale is a linear frequency spacing below 1000 Hz and a logarithmic spacing above 1000 Hz. Therefore we can use the following approximate formula to compute the mels for a given frequency *f* in Hz:

mel(f)= 2595*log10(1+f/700) .... (1)

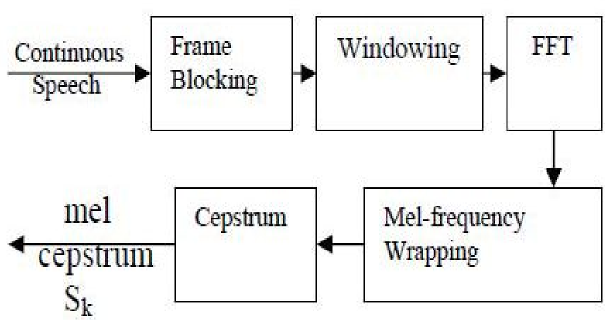

Fig 2. Block diagram of MFCC processor

C. Cepstrum

Here, we convert the log mel spectrum back to time. The result is called the mel frequency cepstrum coefficients (MFCC). Because the mel spectrum coefficients are real numbers, we can convert them to the time domain using the Discrete Cosine Transform (DCT) and get a featured vector.

V.FEATURE MATCHING

Feature matching techniques used in speaker recognition include, Dynamic Time Warping (DTW), Hidden Markov Modeling (HMM), and Vector Quantization (VQ). The VQ approach has been used here for its ease of implementation and high accuracy.

A. Vector quantization model

Vector quantization (VQ) is used for command identification in our system. VQ is a process of mapping vectors of a large vector space to a finite number of regions in that space. Each region is called a cluster and is represented by its center (called a centroid). A collection of all the centroids make up a codebook. The amount of data is significantly less, since the number of centroids is at least ten times smaller than the number of vectors in the original sample. This will reduce the amount of computations needed when comparing in later stages [2], [4].

Even though the codebook is smaller than the original sample, it still accurately represents command characteristics.

The only difference is that there will be some spectral distortion.

B. Codebook Generation

There are many different algorithms to create a codebook. Since command recognition depends on the generated codebooks, it is important to select an algorithm that will best represent the original sample. For our system, the LBG algorithm (also known as the binary split algorithm) is used.

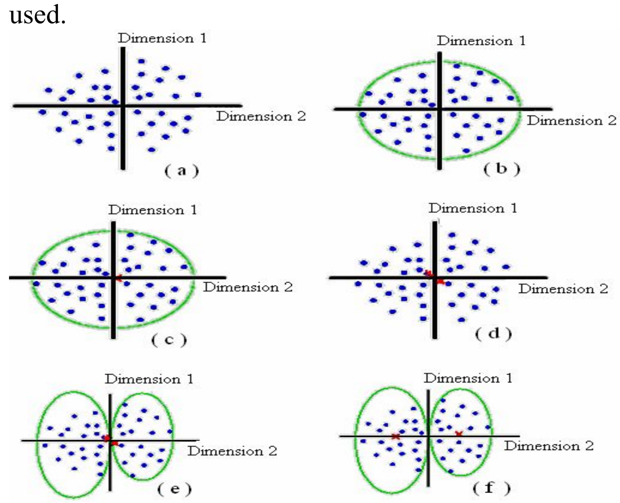

Fig. 3 The process of VQ codebook generation; the features are shown by blue dots, the group boundary in green and the centroids are in red

International Journal of Engineering Research And Management (IJERM)
ISSN : 2349- 2058, Volume-01, Issue-05, August 2014

The algorithm is implemented by the following recursive procedure [2], [5],[6] :

1. Design a 1-vector codebook; this is the centroid of the entire set of training vectors (hence, no iteration is required
2. Double the size of the codebook by splitting each current codebook yn according to the rule: where n varies from 1 to the current size of the codebook, and e is the splitting parameter. For our system, e = 0.001
3. Nearest-Neighbor Search: for each training vector, find the centroid in the current codebook that is closest (in terms of similarity measurement), and assign that vector to the corresponding cell (associated with the closest centroid). This is done using the K-means iterative algorithm.
4. Centroid Update: update the centroid in each cell using the centroid of the training vectors assigned to that cell.
5. Iteration 1: repeat steps 3 and 4 until the average distance falls below a preset threshold
6. Iteration 2: repeat steps 2, 3, and 4 until a codebook of size M is reached.

C. Command Matching

In the recognition phase the features of unknown command are extracted and represented by a sequence of feature vectors {x1… xn}. Each feature vector in the sequence X is compared with all the stored codewords in codebook, and the codeword with the minimum distance from the feature vectors is selected as proposed command For each codebook a distance measure is computed, and the command with the lowest distance is chosen.

One way to define the distance measure is to use the Euclidean distances:

$$D = \Sigma\,(xi - y\,j\,)$$

Fig. 4 describes the schematic of the Nearest Neighbor Search.

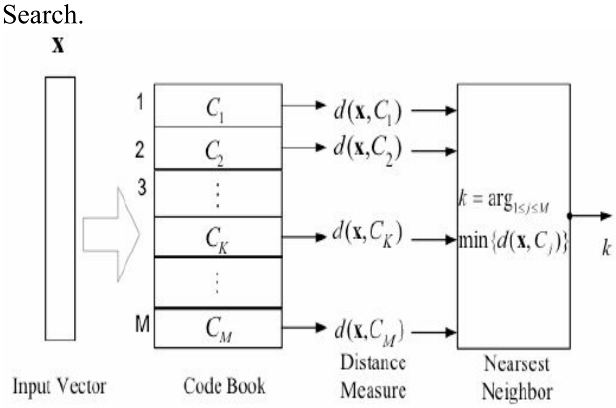

Fig. 4 A schematic of the nearest neighbor search on the VQ decoding process

As we see, the search of the nearest vector is done exhaustively, by finding the distance between the input vector X and each of the code words C1-CM from the codebook C. The one with the smallest distance is coded as the output command.

VI.SIMULATION

Proteus (Processes and Transactions Editable by users) is a model for constructing clinical workflows with integrated decision support. The workflows are constructed with entities components. For this project, proteus software is used for animated simulation. With the help of this proteus software how pulse width varying when giving input. which drawn corresponding graph.

VII. RESULTS

The coding has been developed using MFCC and VQ algorithm, in MATLAB 7.5 version on window Vista platform and supporting hardware also has been implemented. for example speech data base consist of 10 speeches, here we using speech recognition software through which speech added to data base .The interfacing is done between hardware and software using RS-232 cable (MAX-232 IC). External DC motor can be driven in forward or reverse direction as well as it can be stopped also by giving speech commands. While calculating of MFCC for database at the time of speech recognition, there are some graphs obtained, these are shown below. The Fig 5. Shows the graph of speaker voice database.

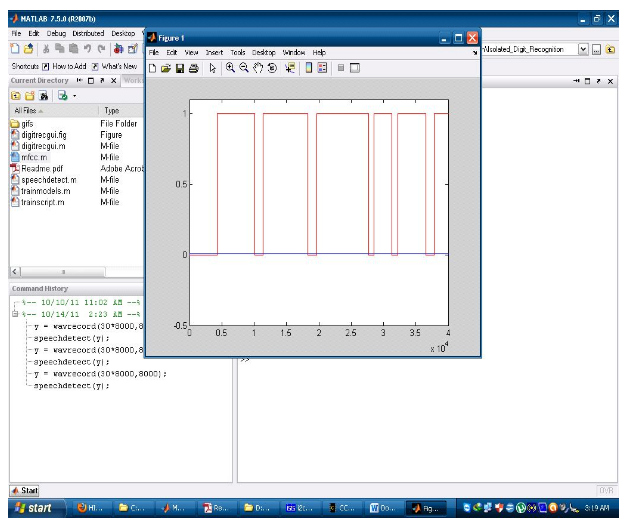

Fig5. Shows the graph of speaker voice.

VIII.CONCLUSION

Speaker-recognition systems can be used to identify a particular person or to verify a person's claimed identity. Speech processing, speech production, and features and pattern matching for speaker recognition were introduced. It gives high security and safety .here we using MFCC and VQ techniques are used in speech recognition to control the DC motor drive. The code developed in MATLAB using MFCC and VQ can be even used for control and drive the DC motor, stepper motor, servo motor etc. The developed speech algorithm can be use fro many security application like voice mail, to drive electric vehicles security areas (like banking, unman vehicles, remote access of computers where speech can be use as password). Here we also introducing password before entering operation this gives more security. And we run motor in different speed that we already load in processor. Using PWM tech create various duty cycle so we obtain different speed. Here we introducing speed sensor which reads current speed to display LCD display.

REFERENCES

[1]Real Time Control of DC Motor Drive using Speech Recognition, Punit Kumar Sharma, Dr. B.R. Lakshmikantha and K. Shanmukha Sundar, IEEE 2011.

[2]Differential MFCC and Vector Quantization used for Real-Time Speaker Recognition System, 2008 IEEE Congress on Image and Signal Processing,Wang Chen Miao Zhenjiang, Institute of Information Science, Beijing Jiaotong University, Beijing 100044, China

[3] “Speaker Identification Using MEL Frequency Cepstral Coefficient”, *Md Rashidul Hasan, Mustafa Jamil, Md. Golam Rabbani Md. Saifur Rahman.*Electrical and Electronic Engineering, Bangladesh University of Engineeringand Technology. 3*rd International Conference on Electrical & Computer Engineering ICECE 2004, 28-30 December 2004, Dhaka, Bangladesh.*

[4]Lawrence Rabiner and Biing-Hwang Juang, Fundamental of Speech Recognition”, Prentice-Hall, Englewood Cliffs, N.J., 1993.

[5]Zhong-Xuan, Yuan & Bo-Ling, Xu & Chong-Zhi, Yu. (1999). “Binary Quantization of Feature Vectors for Robust Text-Independent Speaker Identification” in *IEEE Transactions on Speech and Audio Processing, Vol. 7, No. 1*, January 1999. IEEE, New York, NY, U.S.A.

[6]F. Soong, E. Rosenberg, B. Juang, and L. Rabiner, "A Vector Quantization Approach to Speaker Recognition", AT&T Technical Journal, vol. 66, March/April 1987, pp. 14-26.