ISSN: 2348 9510
International Journal Of Core Engineering & Management (IJCEM)
Volume 3, Issue 1, April 2016

EFFECTIVE DUPLEX SIGN LANGUAGE TRANSLATOR AND RESPONSE USING MFCC

***Abubakar Shameez**,
Asst,. Prof., Department of Electronics and communication, Bearys Institute of Technology, Mangaluru, Karnataka, India.
sshameez@gmail.com

***Sushanth KJ,**
Asst,. Prof., Department of Electronics and communication, Bearys Institute of Technology, Mangaluru, Karnataka, India.
sushanthkj@mail.com

****Kripa**,
Asst,. Prof., Department of Electronics and communication, Bearys Institute of Technology, Mangaluru, Karnataka, India.
kripa.thkt@gmail.com

*****Mranila**,
Asst,. Prof., Department of Electronics and communication, Bearys Institute of Technology, Mangaluru, Karnataka, India.
mranila04@gmail.com

ABSTRACT

Communication is a means of sharing information, it is an important aspect for a human being in order to share their feeling, but mute impaired people could not communicate with others and hence cannot share their feeling, views. Therefore in this work we will overcome the above stated problem. Generally dumb people use sign language for communication but they find difficulty in communicating with others who do not understand sign language, and it is has been statistically proved that majority of the dumb are also deaf, therefore there is need for a communication path between normal people and mute communities. This work aims to design an electronic device that can translate sign

ISSN: 2348 9510
International Journal Of Core Engineering & Management (IJCEM)
Volume 3, Issue 1, April 2016

language into speech using Gesture to text and text to speech, in order to convey the information from dumb person to a normal person, and also speech into text message to convey information from normal person to a deaf person.

Index Terms— Mel Frequency Fepstral Coefficients (MFCC), Digital Image Processing, MATLAB.

I. INTRODUCTION

Sign language is the language used by mute people and it is a communication skill that uses gestures instead of sound to convey meaning. A Wireless data gloves is used which is normal cloth driving gloves fitted with flex sensors along the length of each finger and the thumb. Flex Sensor Plays the major role, these are sensors that change in resistance depending on the amount of bend on the sensor, the flex sensors output a stream of data that varies with degree of bend.

II. METHODOLOGY

In this work the following changes are implemented.

- Duplex communication is implemented.
- Speech to text is implemented using Matlab (MFCC) is implemented.
- Atmega 328P Microcontroller based system.
- Voice processing is implemented using Matlab.
- Audio processor is not required.

III. FLEX SENSOR TECHNOLOGY

Flex sensors [2] change in resistance depending upon the amount of bend on the sensor as shown in Fig.1 They convert the change in bend to electrical resistance - the more the bend, the more will be the resistance value. They are usually in the form of a thin strip from 1 "-5" long that vary in resistance from approximately 10KΩ to 50KΩ. They are frequently used in gloves to sense finger movement. The flex sensors are used as input and are placed inside the glove that is to be worn. The sensor is so flexible that it bends easily even with a small bend. As it is very thin and light weight so it is also very comfortable.

ISSN: 2348 9510
International Journal Of Core Engineering & Management (IJCEM)
Volume 3, Issue 1, April 2016

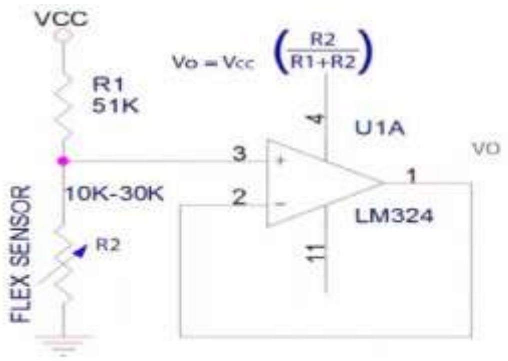

Fig. 1 Flex Sensor Signal Conditioning Circuit [2]

Formula for voltage divider circuit:

$$Vo = Vcc(\frac{R2}{R1+R2})$$

For V0 minimum when sensor deflection is 00 Rl=51KΩ, R2=10KΩ and Vcc= 3.7V

$$Vo(min\quad) = 3.7V(\frac{10K}{51K+10K})$$

For Vo middle when sensor deflection is 450 Rl=51KΩ, R2=20KΩ and Vcc= 3.7V

$$Vo(mid) = 3.7V\left(\frac{20K}{51K+20K}\right) = 1.04225V$$

For Vo maximum when sensor deflection is 900 Rl=51KΩ, R2=30KΩ and Vcc= 3.7V.

IV. SPEECH RECOGNITION

Voice Signal Identification consists of the process to convert a speech waveform into features that are useful for further processing. There are many algorithms and techniques are in use. It depends on features capability to capture time frequency and energy into set of coefficients for Cepstrum analysis. Generally, human voice conveys much information such as gender, emotion and identity of the speaker. Many of the methods operate either in spectral, or in Cepstral domain. Firstly, human voice is converted into digital signal form to produce digital data representing each level of signal at every discrete time step. The digitized speech samples are then processed using MFCC to produce voice features. Here we are using the Mel Frequency Cepstral Coefficients (MFCC) technique to extract features from the speech signal and compare the unknown spoken word with the existing keyword in the database[4].

ISSN: 2348 9510
International Journal Of Core Engineering & Management (IJCEM)
Volume 3, Issue 1, April 2016

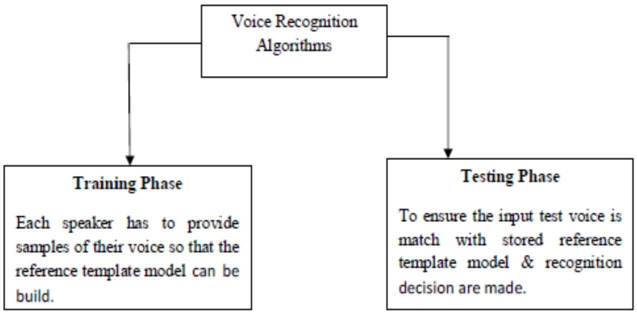

Fig. 2: Voice Recognition Algorithm

V. SPEECH FEATURE EXTRACTION

In speech feature extraction Mel Frequency Cepstral Coefficients (MFCC) technique is used.

CEPSTRUM

Cepstrum name was derived from the spectrum by reversing the first four letters of spectrum. We can say cepstrum is the Fourier Transformer of the log with unwrapped phase of the Fourier Transformer shown in Fig.3

- Mathematically we can say,

$$\text{Cepstrum of signal} = \text{FT}\ (\log(\text{FT}(\text{the signal})) + \text{j2\_m}$$

Where m is the integer required to properly unwrap the angle or imaginary part of the complex log function.

- Algorithmically we can say,

$$\text{Signal} - \text{FT} - \log - \text{Phase unwrapping} - \text{FT} - \text{Cepstrum}$$

VI. MELFREQUENCYCEPSTRALCOEFFICIENTS (MFCC)

Mel Frequency Cepstral Coefficient. Mel frequency Cepstral Coefficients are coefficients that represent audio based on perception. This coefficient has a great success in speaker recognition application. It is derived from the Fourier Transform of the audio clip. In this technique the frequency bands are positioned logarithmically, whereas in the Fourier

ISSN: 2348 9510
International Journal Of Core Engineering & Management (IJCEM)
Volume 3, Issue 1, April 2016

Transform the frequency bands are not positioned logarithmically. As the frequency bands are positioned logarithmically in MFCC, it approximates the human system response more closely than any other system. These coefficients allow better processing of data.

$$m = 1127.01048 \log\left(1 + \frac{f}{700}\right)$$

$$f = 700\left(e^{\frac{m}{1127.01048}} - 1\right)$$

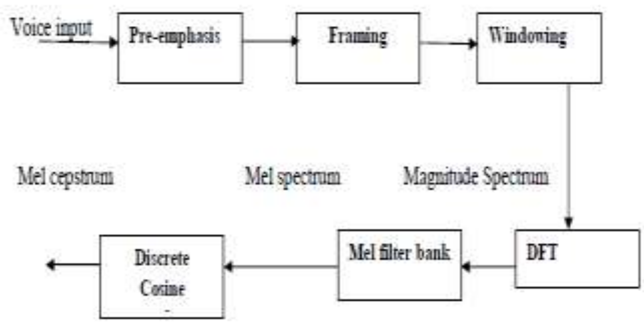

Fig. 3: Speech Feature Extraction Block Diagram [6]

VII. IMPLEMENTATION

First stage is to record a voice using a microphone. The signal during training and testing session can be greatly different due to many factors such as people voice change with time, health condition (e.g. the speaker has a cold), speaking rate and also acoustical noise and variation recording environment via microphone. Then Mel frequency cepstrum coefficient for each recorded voice is calculated and stored in reference template for further processing as shown in Fig 4.

ISSN: 2348 9510
International Journal Of Core Engineering & Management (IJCEM)
Volume 3, Issue 1, April 2016

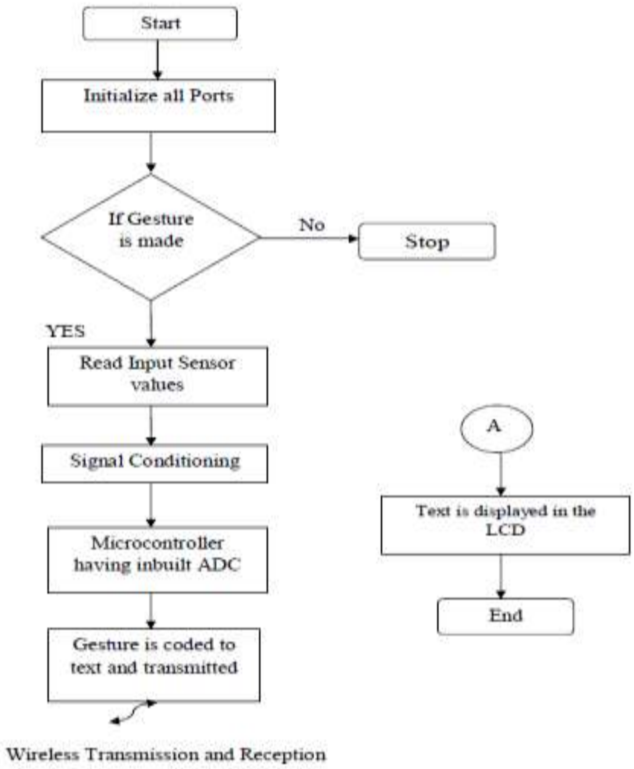

Fig. 4: Flow chart of Transmitter section

Based on this coefficient, get the similarity between training and test input voice signal. External voice command is received through a speaker. Then check whether it is matched with the reference template or not. If yes corresponding actions are performed. Else go for the next external command[7].

ISSN: 2348 9510
International Journal Of Core Engineering & Management (IJCEM)
Volume 3, Issue 1, April 2016

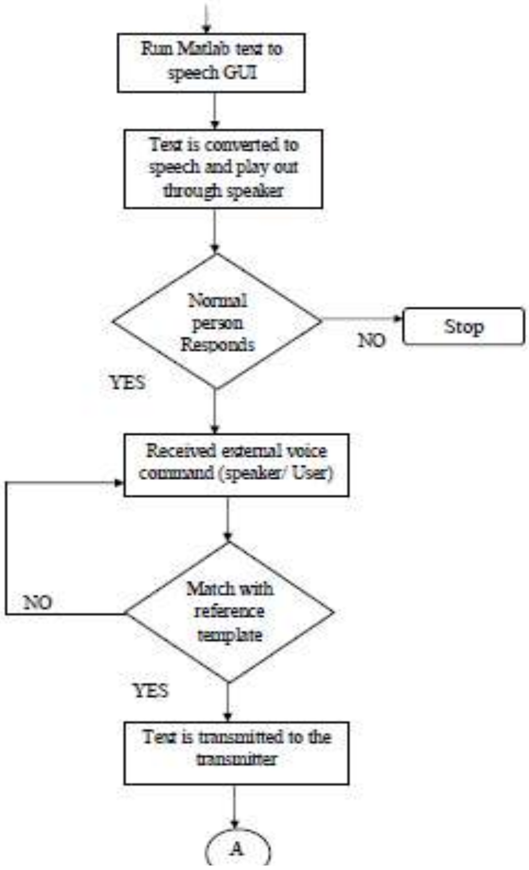

Fig. 5: Flow chart of Receiver section

ISSN: 2348 9510
International Journal Of Core Engineering & Management (IJCEM)
Volume 3, Issue 1, April 2016

VIII. IMPLEMENTATION OF SPEECH TO TEXT

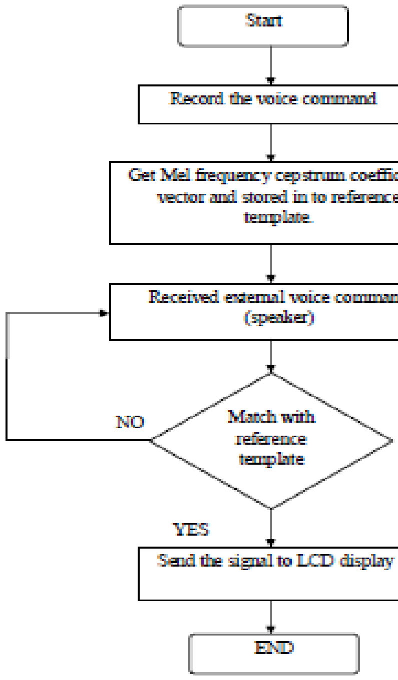

Fig. 6: Flow chart of Speech to text

In order to get the help of GUI applications in MATLAB, we use the keyword “guide”. That is to start GUIDE, enter guide at the MATLAB prompt.

IX. EXPERIMENTAL RESULTS

Fig 7 and 8 shows The overall system module is shown in the Fig. 5.1, Gestures are taken from the flex sensor fitted to the glove, and interfaced to Matlab through ZigBee.

ISSN: 2348 9510
International Journal Of Core Engineering & Management (IJCEM)
Volume 3, Issue 1, April 2016

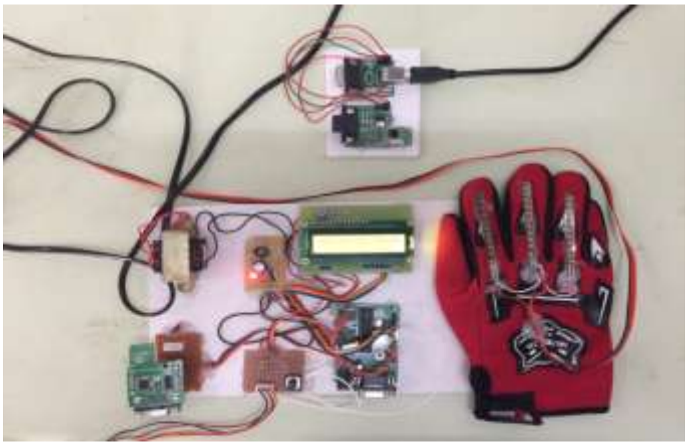

Fig. 7: System Module

The Table 5.1 shows the different gesture combination and its corresponding interpretation, these gestures are first displayed on the LCD and interfaced using Matlab, and the interpreted words or sentence are played out through the speaker. Speech which is converted to text is also displayed on the LCD.

Fig.8 Gestures and Interpretations

Hello **Hai**

ISSN: 2348 9510
International Journal Of Core Engineering & Management (IJCEM)
Volume 3, Issue 1, April 2016

Good **You**

How are you **Thank you**

The mute person starts to communicate by pressing the switch as shown in the Fig.9 Which then displays a command "Sending Mesg" indicating it is ready to accept the Gestures.

ISSN: 2348 9510
International Journal Of Core Engineering & Management (IJCEM)
Volume 3, Issue 1, April 2016

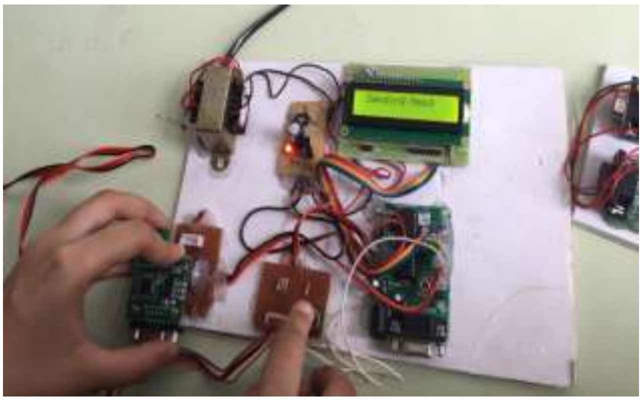

Fig. 9: Initializing Gesture to Speech Communication

Fig.10 shows, the interpreted Gesture is interfaced to Matlab through ZigBee and the interpreted words is displayed on GUI window and played out through speaker.

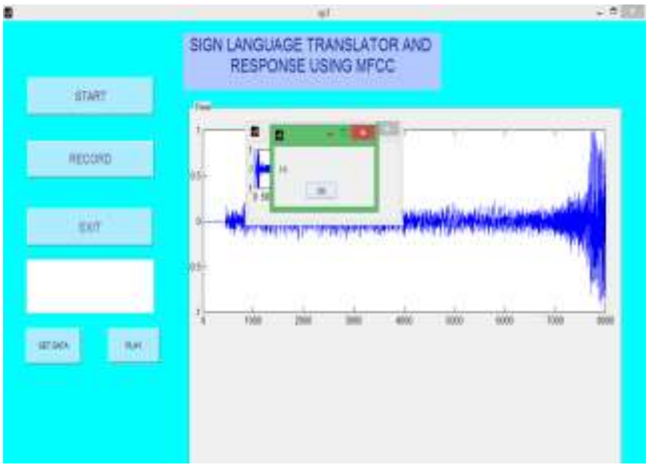

Fig. 10: Speech to Text Output on the GUI Window

Fig.11 shows the effective two way communications in the form of gesture to text and text to speech.

ISSN: 2348 9510
International Journal Of Core Engineering & Management (IJCEM)
Volume 3, Issue 1, April 2016

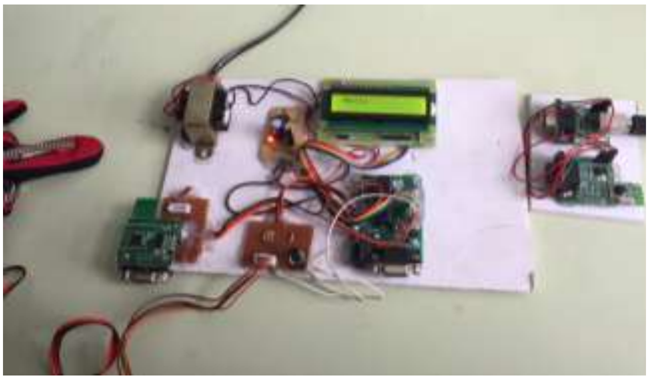

Fig. 11: Speech to Text Output

X. CONCLUSION AND FUTURESCOPE

The proposed work identifies the humanitarian concern for the mute impaired and provides a friendly, low cost sustainable alternative for the existing aids. module that comprises of ZigBee transceivers, flex sensors, ATmega328 microcontroller and Matlab interfacing that has MFCC algorithm. The experimental results have shown the usefulness of the system in allowing mute people to communicate wirelessly, informatively and independently between dumb/deaf person and normal. The dual communication helps the mute people to communicate informatively and independently. Evaluations of the system that we have developed have been conducted by attaching the prototype to Matlab interfacing in personal computer.

Future work will be that the disadvantage of flex sensor sensitivity can be overcome by using a technique called as image processing, where the gesture inputs are taken from the webcam fitted into the system and the images are processed using image processing, and the processed images are compared with the database, if the images are matched the corresponding words are spoken out.

REFERENCES

[1] Nitesh Dumore, Hitesh Banait, Pushpak Bhandekar, "Electronic Speaking System for Deaf and Dumb", *2014 Discovery Publication,* Volume 19, Number 58, May 2014.

ISSN: 2348 9510

International Journal Of Core Engineering & Management (IJCEM)
Volume 3, Issue 1, April 2016

[2] Abjhijt Auti, V. G. Puranik, Dr. A. K. Kureshi ,“Speaking Gloves for Speechless Persons” ,*International Journal of Innovative Research in Science, Engineering and Technology,* Volume 3, Special Issue 4, April 2014

[3] Milind U. Nemade, Satish K.Shah, “Real Time Speech Recognition Using DSK TMS320C6713”, *International Journal of Advanced Research in Computer Science and Software Engineering,* Volume 4, Issue 1, January 2014.

[4] Shoaib Ahmed .V, “Magic Gloves (hand gesture recognition and voice conversion system for differentially able dumb people)”, *Tech Expo-The Global Summit London,* 2012.

[5] B. Ali, S. Munawwar, B. Nadeem, "Electronic Speaking Glove for Speechless Patients", Bachelor of Electronic Engineering FYP Report, August 2010.

[6] S. M. Kamruzzaman, A. N. M. Rezaul Karim, Md. Saiful Islam, Md. Emdadul Haque, “Speaker Identification using MFCC-Domain Support Vector Machine” *International Islamic University Chittagong, Bangladesh*, 2010

[7] Jayanna, S.R. Mahadeva, “Analysis, Feature Extraction, Modeling and Testing Techniques for Speaker recognition”, *IETE Tech. Rev*., 26:181-90, 2009.

[8] N. P. Bhatti, A. Baqai, B. S. Chowdhry, M. A. Unar, "Electronic Hand Glove for Speech Impaired and Paralyzed Patients", *EIR Magazine*, May 2009.

[9] Molau S, Pitz M, Schluter R, and Ney H, “Computing Mel-frequency Coefficients on Power Spectrum”, Proceeding of *IEEE ICASSP-2001*, Vol. 1, pp.73-76, 2001.

Sushanth K J received his BE from Coorg institute of technology, Coorg, India and M.Tech Degree from Sri Jayachamarajendra college of Engineering (SJCE), Mysore, under VTU, India, in the 2010 and 2012, respectively. At present he is working as assistant professor in Bearys Institute of Technology, Mangalore, India, in the department of Electronics and Communication Engineering. His research areas include Wireless Sensors networks, Embedded systems, and VLSI design.

ISSN: 2348 9510
International Journal Of Core Engineering & Management (IJCEM)
Volume 3, Issue 1, April 2016

Abubakar Shameez received his BE from P.A College of Engineering, Mangalore, India and Persuing M.Tech in Nitte Mahalinga Adyanthaya Memorial Institute of Technology (NMAMIT), Nitte, under VTU, India, respect. At present he is working as assistant professor in Bearys Institute of Technology, Mangalore, India, in the department of Electronics and Communication Engineering. His research areas include Computer networks,Network Security, Wireless Networks and Communication systems.

Kripa received her BE from P.A College of Engineering, Mangalore, India and M.Tech Degree from Nitte Mahalinga Adyanthaya Memorial Institute of Technology (NMAMIT), Nitte, under VTU, India, in the 2013 and 2015, respectively. At present she is working as assistant professor in Bearys Institute of Technology, Mangalore, India, in the department of Electronics and Communication Engineering. Her research areas include Wireless Sensors networks, Image Processing and Communication systems.

Mranila P received her BE from Bearys institute of technology, Mangaluru, India and M.Tech Degree from St Joseph Engineering College (SJEC), Mangaluru, under VTU, India, in the year 2013 and 2015 respectively. At present she is working as Assistant Professor in Bearys Institute of Technology, Mangaluru, India, in the department of Electronics and Communication Engineering. Her research areas include Image processing and Embedded systems.