# IMPLEMENTATION AND EVALUATION OF VOICE CODEC FOR ZIGBEE

by

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# <u>SUPERVISOR CERTIFICATE</u>

It is to certify that the final copy of the thesis has been evaluated by me, found as per the specified format and error free.

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Col. Dr. Adnan Rashidi, PhD

# ABSTRACT

Most of the current uses for ZigBee and IEEE 802.15.4 focus on control applications. However, there are areas that will benefit from standardisation, low cost and possibly low power of ZigBee.

The project focuses on the use of ZigBee for audio applications. The theoretical limits of ZigBee for this kind of application are evaluated. Then algorithms are developed related to transfer of audio data. While streaming an audio, the most important factor is the available bandwidth where ZigBee has very tight constraints on bandwidth. By compressing the audio, alot of bandwidth can be saved. Two algorithms have been implemented at different data rates for transferring audio, one is MELP (Mixed Excitation Linear Prediction) and the other is CELP (Code Excited Linear Prediction). ZigBee was developed with low data rate in mind. Low energy consumption is mostly noticed for applications where data rate is low and devices can often go in power down mode. Software used is MATLAB. Original and synthesis signals are played in simulation and compared.

Speech compression algorithms can be used in areas where there is need to reduce communication cost and use available bandwidth and storage space effectively. Real life applications include rescue mission, emergency scenarios, intra building communication and conferences.

# DEDICATION

*I dedicate this thesis to my beloved parents for their prayers and encouragement, to my husband whose endless support helped me to pursue it till the endand my daughter, Manha.*

ACKNOWLEDGMENT

I thank Allah who provided me with strength and caliber to bring this thesis work to its successful completion.

I am deeply obliged to my supervisor, Col. Dr. Adnan Rashidi, for his guidance, unwavering support and confidence in me throughout the course of this thesis work. His time and efforts were very valuable. He contributed significantly in the thesis work and also imparted a lot of knowledge to me. I am also grateful to my Guidance and Evaluation Committee, Brig. Imran Rashid, Col. Dr. AdilMasood, and Col. Dr. Imran Touqir for dedicating their time and making contributions to this thesis work. I offer my regards and blessings to all of those who supported me in any respect during the completion of this work. Alongside that, I thank the university administration, for facilitating the progress of work at different phases, faculty members of MCS for polishing my knowledgebase, my university mates and friends.

Last but not the least; I am grateful to those people without whom I could never have accomplished my aim my family, especially my parents and my husband. I thank them all for their constant prayers and unfaltering support throughout this journey. May Allah bless them all with eternal happiness!

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## NOTATION

| | |
|---|---|
| **MSE** | Mean Square Error |
| **IEEE** | Institute of Electrical and Electronics Engineers |
| **PAN** | Personal Area Network |
| **Wi-Fi** | Wireless Fidelity |
| **ISM** | Industrial, scientific and medical |
| **DSSS** | Direct Sequence Spread Spectrum |
| **FFD** | Fully Functional Devices |
| **RFD** | Reduces Functional Devices |
| **CSMA-CA** | Carrier Sense Multiple Access with Collision Avoidance |
| **CELP** | Code Excited Linear Prediction |
| **MELP** | Mixed Excitation Linear Prediction |
| **IDFT** | Inverse Discrete Fourier Transform |

| | |
|---|---|
| **BPV** | Band-pass Voicing |
| **LSF** | Line Spectral Frequencies |
| **LPC** | Linear Prediction Coder |
| **PHY** | Physical Layer |
| **MAC** | Medium Access Control Layer |

*Chapter 1*

# Introduction

The word ‘telecommunication’ is derived from two different sources, Greek word ‘tele’ meaning *far off* and Latin word ‘communicare’, meaning *to share* (since the Latin term communication refers to the social process of exchanging information). Telecommunication is transfer of any sort of information (signals, images, sounds, voices). It involves use of technology for communication between participants. Basically telecommunication can be divided into two different methods, wired and wireless. The three basic parts that are always present in any telecommunication system are: transmitter, transmission medium, receiver.

Speech coding is widespread area of research in signal processing which comes under wireless digital communication, a field of telecommunication. It is one of the basic elements of digital communication. Digital communication is more resistant to transmitted error than analog medium. Over long distances, digital signals are capable of maintaining better quality than analog signals. Due to increasing demands in telecommunication services, need for speech coding has increased manifold. Speech coding is compression of data of audio signal (signal containing speech). Huge amount of data and tons of information that needs to be carried from one point to other can be handled by speech coding techniques [1]. The aim of speech coding technique is to represent data with minimum number of bits while maintaining original quality of speech. Digital data is converted into codes by encoders. Coded bits are either transmitted as frame or bits. At receiver end, the frames or bits are decoded to reconstruct the original signal. Before transmission over channel, we need to compress signals since bandwidth affects cost of processing. Hence a signal is obtained in digital form which is stored in more efficient way and transmitted over channel having bandwidth constraint. Main objective of any speech coding technique is to reduce bandwidth and data rate. Since less number of bits is required to transmit data therefore transmission power of signal should also be reduced. Quality of transmitted speech signal is dependent on: storage, transmission rate, computational complexity and transmission rate. There is always a tradeoff between bandwidth utilization and speech quality. Speech coding techniques also provide immunity to noise since few bits are used as control bits. Speech coding has find applications in areas of VoIP, video conferencing and multimedia.

This project focuses on implementation and evaluation of voice codec for Zigbee. Different speech compression techniques, CELP (Code Excited Linear Prediction) and MELP (Mixed Excitation Linear Prediction) are implemented. CELP is based on the concept of LPC. Enhanced feature includes codebook containing various excitation signals. Decoder has the same codebook as encoder. Index of most suitable excitation signal is sent to decoder signal is reproduced. MELP performs better in noisy environments like the ones encountered in military and commercial communication systems. This translates into relatively low power consumptions, an important consideration for portable systems. These are the low bit rate techniques for real-time applications. In [2] brief overview and conceptual literature of Code Excited Linear Prediction has been provided. [3] presents comparative results of different values obtained for Mixed Excited Linear Prediction and Code Excited Linear Prediction presenting clear idea about more efficient coding technique. [4] describes proposed MELP Federal Standard including its algorithmic description.

## 1.1 Research already carried out

Vast literature is available in area of speech coding techniques being used in communications. In Paper [1] hybrid speech coding technique (CELP) and parametric speech coding technique (CELP) is implemented in MATLAB software. For bit rate 9.6 kbps and 16 kbps, CELP is an efficient coding scheme. CELP is implemented using analysis-by-synthesis technique where a combination of parameters is used to represent speech signals. Parameters are quantized using closed-loop quantization. This process employs analysis-by-synthesis method. MELP, mainly in military and federal standards, This coder has low bit rate. Word ‘hello’ is synthesized using CELP technique. The signal is then analyzed by varying values of perceptual weighted filter (c). Paper [2] gives general overview and conceptual literature of CELP coder. This paper explains the closed loop process of speech coding for narrow band and medium band speech coding techniques. Results show that CELP has better replication of original signal hence it considers CELP as one of the best coder for speech synthesis. Paper [3] implements CELP and MELP speech coding techniques using MATLAB 2009a. Based on subjective tests like MOS and MSE, these coding techniques are analyzed. Simulations on different input speech inputs are obtained for CELP. For each values of ‘c’, different results were obtained for both high rate i.e. 16 kbps and low rate i.e. 9.6 kbps. Signals at both rates were analyzed for different values of ‘c’. Results showed that reconstruction of speech signal was better at lower value of c whilst maintaining audible quality of speech signal. For MELP, two different frame lengths (25 ms and 30 ms) were used for signal analysis. It was observed that as value of frame length decreased, so does the speech quality. Paper [4] describes the U.S. Federal Standard at 2.4 kbps. Algorithm description along with complete details of encoder and decoder operation is explained. For performance verification, different test like DRT and comparison tests are employed. The results showed that MELP outperformed CELP in most cases. Paper [5] presents detailed analysis and study of structure of CELP. Various quantization schemes have been discussed for quantization of LP coefficients. Extensions of 4kbps hybrid MELP/CELP coder, from 2.4 kbps to 6.4 kbps are described in paper [6]. Three coding modes are used for baseline 4kbps coder. Modes used in 6.4 kbps extensions are same as those in baseline 4kbps. However there are only two modes used in 2.4 kbps extension. Coders is based on formal subjective tests. Implementation of MELP 2400 bps vocoder is described in paper [7]. The coder is evaluated on the basis of DAM. For determining LPC coefficients, autocorrelation technique is used. Vocoder can match the voiced speech waveform much better using adaptive spectral enhancement. This paper shows that quality of coded speech can be improved by additional information of Fourier series.

## 1.2 Synopsis / Thesis Statement

This project focuses on speech coding, implementation of voice compression algorithms (MELP and CELP) and evaluation of voice codec for Zigbee. Zigbee is a wireless standard based technology that is simpler and less expensive than Bluetooth or Wi-Fi. It is typically used in low data rate applications that require long battery life and secure networking. Zigbee is specified for devices which have low data rates and consume low power hence providing ability to run on cheap batteries. Zigbee protocol is can communicate data in hostile RF environment. It provides facility for carrying out secure communication.

Speech coding is common area of research in signal processing. Main aim in speech coding is to reduce bit-rate while maintaining quality of speech or improving speech quality at a particular bit rate [1]. CELP is improved version of LPC coder. MELP is a low bit rate coder with bit rate of 2.4 kbps.

## 1.3 Objective

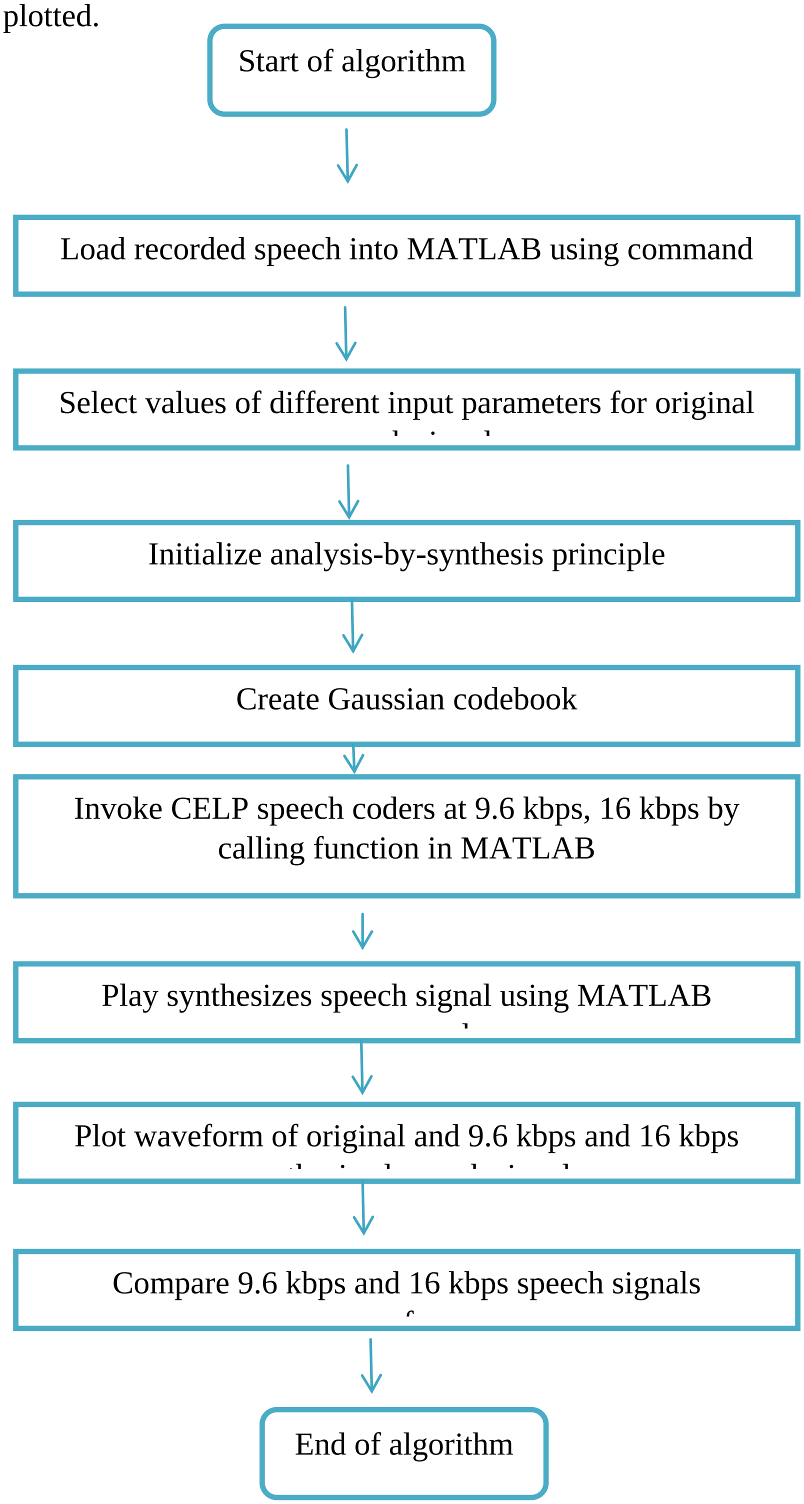
The primary objective is to implement voice compression techniques, CELP and MELP and evaluate these techniques for ZigBee. Hence an adaptive coding model will be designed for low bandwidth channels. The research aims to comprehend following goals:

- Compare speech compression algorithms (MELP / CELP) for transmission over low bandwidth channels using Zigbee protocol
- Find out bandwidth requirements for different data rates
- Design an intelligent interface that will switch to the one of the speech compression algorithm , either MELP or CELP on the basis of given conditions

## 1.4 Methodology used

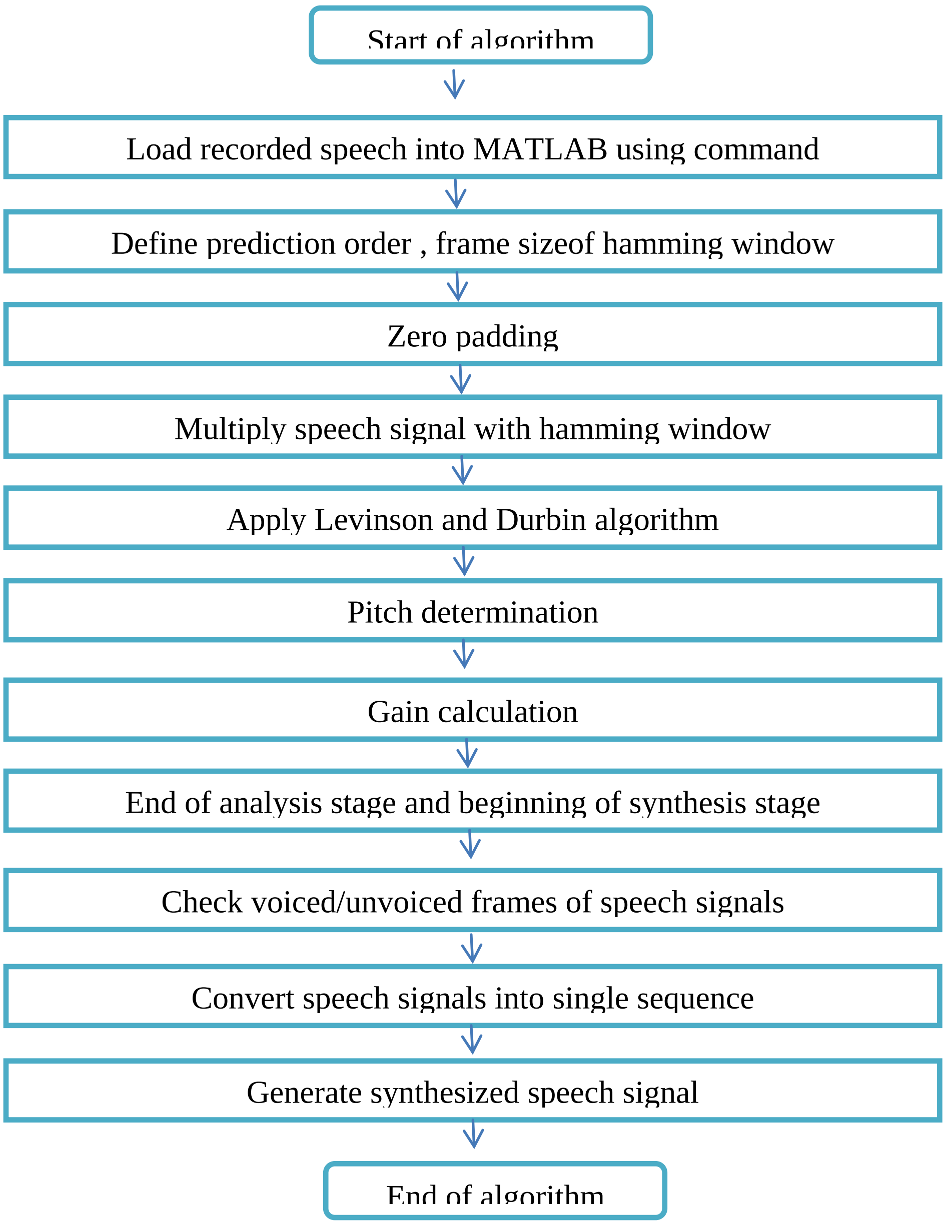
Flow chart of the proposed algorithm of CELP is given below. First of all, load the recorded speech signal using ‘wavread’ command in MATLAB. Once loaded, different parameters for example frame length (L), order of LP analysis (M), constant parameter for perceptual weighted filter (c), Pidx range is assigned fixed value. Then analysis-by-synthesis process starts. After wards creation of Gaussian codebook takes place.. Codebook search is performed to obtain vectors. Then speech coders at the two rates specified above are called using MATLAB function. For synthesis part, Levinson-Durbin algorithm is used. Finally, graphs plots include original and reconstructed signal at 9.6

kbps and 16 kbps. Comparison of original signal with signal synthesized at 9.6 kbps and 16 kbps is plotted.




*1.1 Flow chart of CELP*

Flow chart of proposed MELP algorithm is given below. It starts exactly as CELP algorithm. First of all we define prediction order , frame size of hamming window. If needed, zero padding is appliede. Then original speech signal is multiplied with hamming window. After application of Levinson-Durbin algorithm, decision for voiced or unvoiced frames is made. Afterwards gain and pitch value is determined. Hence analysis stage is completed and synthesis stage starts. After checking of voice and unvoiced frames, conversion of speech signal to single sequence takes place. Finallyresults are plotted in MATLAB.




*1.2 Flow chart of MELP*

The diagram below further clarifies the details of project.

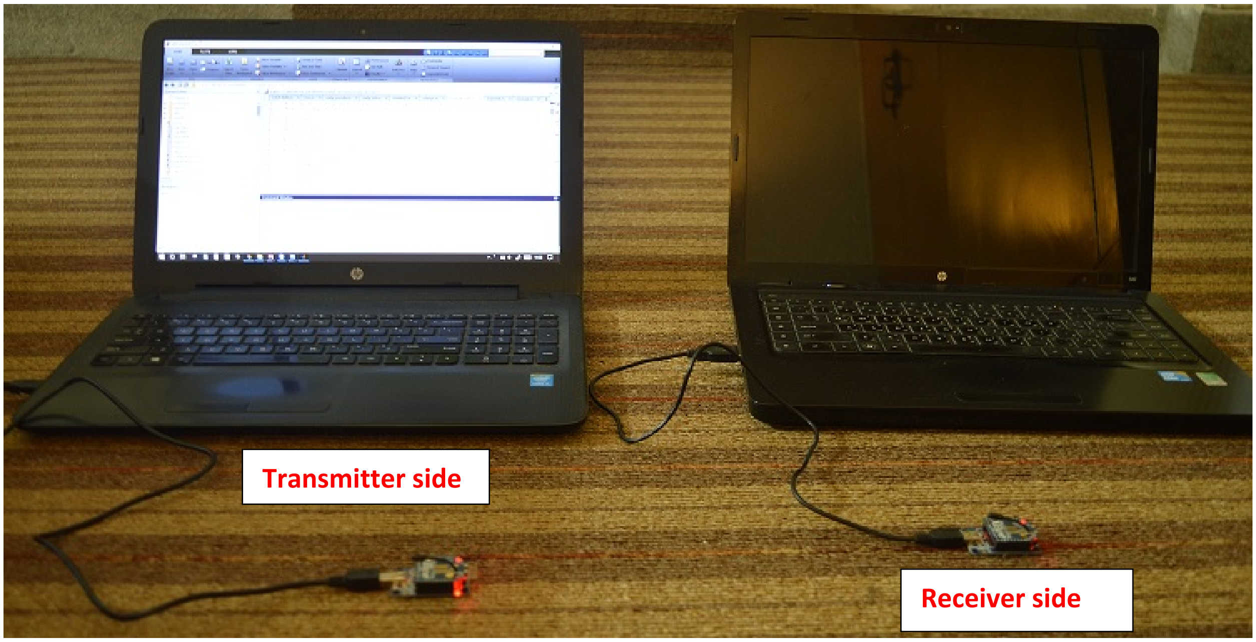



*Fig 1.3 Hardware setup for transmission of signal*

Following steps were performed:

1. Signal was compressed using compression algorithm.
2. Compressed signal was converted into string.
3. Zigbee was connected via USB port to PC (Transmitter and receiver side)
4. Configuration of Zigbee using Putty software.
5. On transmitter side, signal was encoded and converted into string.
6. Using commands, signal was transmitted bit by bit.
7. On receiver side, the received string was saved in .csv file.
8. The file was read in MATLAB and decoded.
9. The decoded signal was plotted and compared to the sent signal in MATLAB.

## 1.5 Advantages

### 1.5.1 ZigBee:

i. Operates on small inexpensive batteries for years.

ii. Can be employed in dense locations, cities or big buildings.

iii. Performs well where devices have to send smaller data over widespread area while maintaining battery life for years.
iv. Have up to 65k nodes. Addition or removal of nodes from the network is easy making it suitable for control , monitoring applications.
v. Have more range than blue tooth and Wi-Fi.
vi. Have low complexity so it is easier for application engineers to integrate them into solution.
vii. Have low cost so it can be deployed in large number hence number of applications using this standard will increase.

### 1.5.2 CELP:

i. Provides significantly good quality compared to existing low bit-rate rate coders (e.g. LPC vocoder)
ii. Involves codebook containing excitation signals. This reduces computational complexity since only index of excitation signal is transmitted instead of entire signal.
iii. Is capable of more precise envelop replication.

### 1.5.3 MELP:

i. Has efficient computational requirements which forms basis of low power consumption which is a significant factor for portable systems. Hence it is ideal for low power and low bandwidth communication
ii. Uses mixed excitation model which produces naturally sounding speech.

## 1.6 Areas of Application

Speech compression algorithms can be used in areas where there is need to reduce communication cost and use available bandwidth and storage space effectively [3]. Generally, speech compression is used where a large number of users share the same frequency band. Compression leads to greater number of users to share the system.

Real life applications include:

i. Rescue mission

ii. Emergency scenarios
iii. Intra building communications
iv. Conferences

## 1.7 Outline of Thesis

This thesis is based on following chapters

**Chapter 01:**

This chapter is the basic introduction of the topic, problem statement, scope and objective.

**Chapter 02:**

Chapter 02 is the introduction of ZigBee. It describes general characteristics of Zigbee, gives deep insight into its architecture and network topologies and explains each layer.

**Chapter 03:**

This chapter explains why speech compression and encoding is required. It also highlights different types of coders.

**Chapter 04:**

It deals with proposed scheme of work. The algorithms applied in the project have been discussed in detail in this chapter.

**Chapter 05:**

This chapter is about the experimental and simulation results along with conclusion and future work.

*Chapter 2*

# ZigBee

## 2.1 Introduction

ZigBee is a set of protocols based on IEEE 802.15.4 used for communication protocols at higher level [19]. 802.15.4 is basically a standard for wireless communication which is issued by IEEE, an association that publishes standards that works in promotion of existing and emerging technologies. 802.15.4 was developed for devices particularly using low data rate, requiring simple connection procedure while maintaining longer battery life.

Its name, 'ZigBee' refers to waggle dance of honey bees after they return to bee-hive. It is named so for erratic zigzagging patterns of bees between flowers which symbolizes communication between nodes in a mesh network [6].

The new standards in ISM radio bands are frequencies 868 MHz, 915 MHz and 2.4 GHz. There is total number of 27 channels (868 MHz band: 1 channel of 20 kbps, 915 MHz band: 10 channels of 40 kbps, 2.4 GHz band: 16 channels of 250 kbps). Modulation scheme used is DSSS to provide coding gain.

ZigBee protocols, developed by ZigBee Alliance are used for personal area networks having smaller radios operating on low-power. ZigBee is simpler and far less expensive than existing wireless personal area networks for example Wi-Fi and Bluetooth. This technology has self-healing networks that is capable of managing various data traffic patterns. ZigBee specifies lower cost and lower power standard that employs mesh networking. Due to these features, it can be deployed in large number of monitoring applications. Consumption of low power is another striking feature of Zigbee which allows much longer life with small and inexpensive batteries [6]. Mesh networking lead to greater reliability with extended range.

Data transmission through ZigBee involves intermediate devices hence capable of reaching more distance, therefore creating a network. Main components of ZigBee include PAN

(Personal Area Network) coordinator, routers and end devices. Defined data rate for ZigBee is 250 kbps. It is employed in applications requiring secured networking, lower data rate and longer battery life.

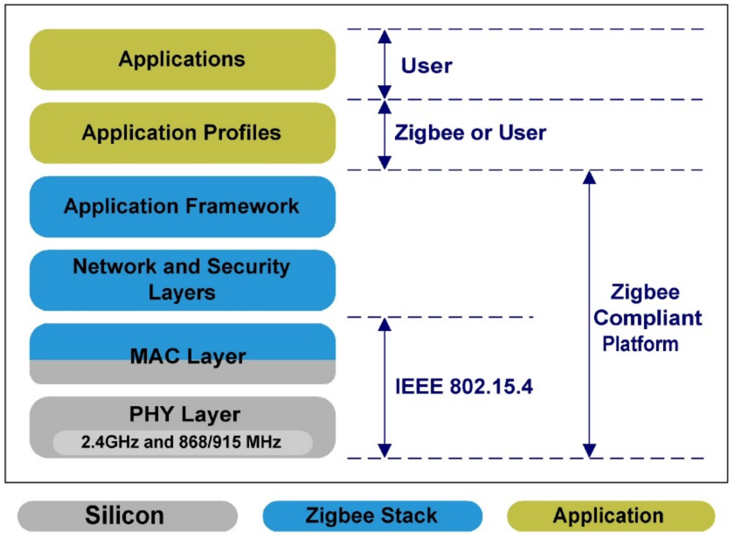



*Fig 2.1 IEEE 802.15.4 and Zigbee*

## 2.2 General Characteristics of Zigbee

Striking characteristic of Zigbee technology is consumption of low power with lower cost and lower rate of transmission [5] which is substantiated in details listed below:

1) Consumption of low power: It operates on lower power modes which make it suitable for implementation on microcontrollers having low power and low cost.
2) High capacity: Zigbee has up to 65k nodes [5]. Addition or removal of nodes from the network is easier hence making it suitable for control applications. Zigbee can adopt star, tree or mesh topology.
3) It operates on small, inexpensive batteries for years. It works well in situations where small data is to be sent over wider area along with maintenance of battery life for many years.
4) It has low complexity so it is easier for application engineers to integrate them into solution.
5) Short distance: Common range for Zigbee transmission ranges between 10 ~ 100 m and if

RF transmission power is increased, the range can be increased up to 1 km.

6) Reduced delay: Zigbee has less delay in communication.
7) Greater reliability: Due to prevention collisions to reduceconflicts while transmitting data.

### 2.3 Zigbee Architecture

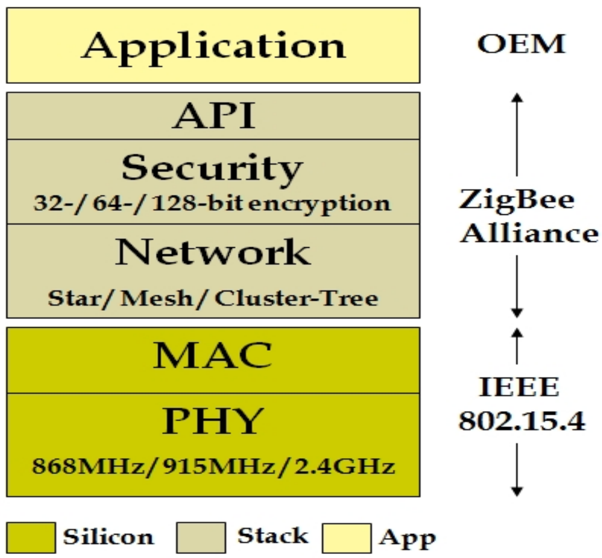



*Fig 2.2 Zigbee Architecture*

#### 2.3.1 The PHY (physical) layer

This layer contains functions that are involved in management of radio channel. It controls the flow of data packet. PHY utilizes CSMA with CA for accessing radio channel. A radio listens to channel, if clear, it will transmit the packet and if occupied for any reason, the radio holds off for some time and rechecks the channel before transmission.

PHY layer has 4 frames with specific functions. These frames are:

1) Data
2) Acknowledgement
3) Beacon

4) MAC Command

The Acknowledgement frame is used to acknowledge reception of data packet without error. Third frame, beacon is utilized by stations implying power saving modes. MAC Command functions in transmitting low level commands within nodes.

### 2.3.2The MAC (Medium Access Control) layer

This layer contains the basics that allow transfer of data and management of RF and PHY by higher entities. Its function is to generate network beacons that help devices to look for existing network and to establish or discontinue the connection with that network.

### 2.3.3Networking techniques

One of the following simple network topologies can be adopted by simple ZigBee network.

#### 2.3.3.1 Star topology

This is common network topology in host-client networks. Messages from client device are supposed to pass via hub which is the central node.

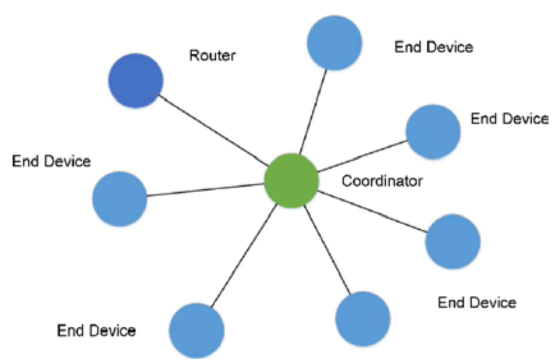



*Fig 2.3 Star topology*

#### 2.3.3.2 Peer-to-peer network

This topology allows direct communication with peer devices. Peer-to-peer topology can be used in mesh networking.

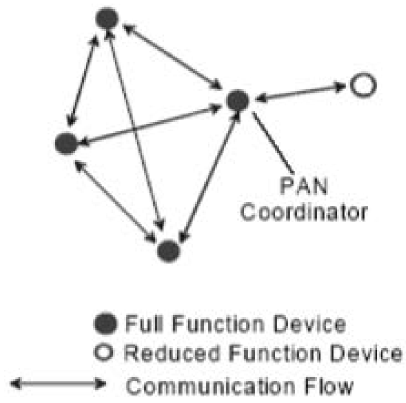



*Fig 2.4 Peer to peer network topology*

### 2.3.3.3 Tree topology

There is a top node with branch or leaf structure in tree topology. The message travels up the tree up till the point where necessary and then down the tree.

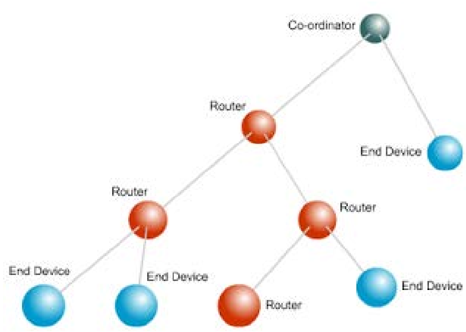



*Fig 2.5 Tree topology*

### 2.3.3.4 Mesh network

It has tree like structure in which few leaves are directly connected. Whenever a suitable route is available, messages can travel across tree. In a single PAN, ZigBee can address as much as 65k nodes. ZigBee networking is originally mesh based.


*Fig 2.6 Mesh Network*

## 2.4 Zigbee Device Architecture

Two physical devices defined by this standard are:

1) FFD (full Function Device) can talk to RFD or other FFDs.
2) RFD (reduced Function device) can talk to only an FFD. RFD is implemented in very simple applications for example a light switch that communicates infrequently. RFD can be utilizes minimum memory and resources as RFD can be associated with single FFD at a time [20].

*Chapter 3*

# Speech Compression and Encoding

## 3.1 Introduction

Speech compression is gaining much importance nowadays due to the fact that it plays an important role in conservation of transmission rate and bandwidth while reconstructing good quality signal at the receiver end. Fixed transmission rate and low delay in signal processing are additional requirements for real time communication. The main focus in area of speech compression lies in coding where there is limited capacity due to scarcity of available spectrum space. It is an application of compression of data in audio signals that contain speech.

Speech coding is different from other forms of audio coding in way that speech signal is much simpler and contains more statistical information about properties compared to audio signals. The key criterion in speech coding is preservation of intelligibility with limited amount of data transmitted. Voice over IP and mobile telephony are two major areas of speech coding. Low coding delay is requirement in majority of speech applications as longer delays.

Due to limited bandwidth available, this area has gained enormous importance in field of wireless communication. In speech coding, a speech signal is represented in digital form then it is coded with reduced bits possible to obtain toll quality signal. Main applications of speech coding include digital transmission of signals. Objective of speech coding is to reduce bit rate while maintaining quality of speech signal or to enhance the quality of speech at given bit rate.

## 3.2 Speech Production

Speech is produced when air flow is directed towards vocal tract passing through the larynx. When air in lungs passes from glottis and narrowing of vocal tract, normal speech is produced resulting in periodic or aperiodic excitation.

Cavities such as jaws, lips, tongue, soft palate and nasal cavities act as resonant cavities. When vocal tract is open, vowel sounds are produced. Whereas when a vocal tract is closed due to temporary closure or due to narrowing of air passage, consonant sounds are produced.

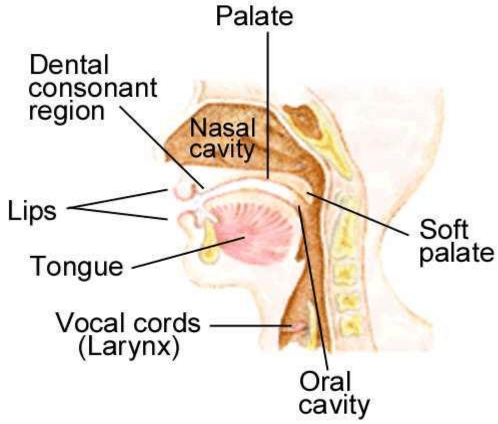



*Fig 3.1 Model of human speech production*

### 3.3 Speech Encoding

Speech encoding has been one of the major concerning areas in field of digital speech production. It is the process of converting the speech signal to compact form so that it can be transferred utilizing smaller memory. The objective behind this is to access limited bandwidth available. Therefore, there is a need to compress the speech. In long – distance speech communication, good quality storage of speech and encryption of message, speech encoding is required. Due to compression, storage of longer messages has been made possible.

Speech coding is a lossy coding technique which implies that output signal is not exact replica of input signal. Differences in both signals can be distinguished. Various coding techniques are available. Each of them has different constraints on time delay.

### 3.4 Speech Coders

Speech coding is a way of representation of speech signal in a way that can be transmitted fast or needs small storage. In simpler words, it is compression of signals. Available coders can be broadly classified into following categories:

1) Waveform coders
2) Transform coders
3) Sub Band coders
4) Parametric/Hybrid coders

### 3.4.1 Waveform Coders

These coders reproduce input signals waveform. In order to use them for coding a variety of signals they are designed so that they are signal independent. They show considerable degradation in presence of transmission errors and noise. These are minimum bit rate coders.

### 3.4.2 Transform Coders

They are used for frequency domain representation of signal. Prior to procedure, the signal is transformed to frequency domain. After procedure, dominant spectral features of signal are maintained. This coder can be implemented using a filter bank. Lower frequencies are more significant than higher frequencies hence lower frequencies are represented with more number of bits. After quantization, relative energy in each band is transmitted. Such coders are not widespread and can be used in range 9600 – 16000 bps.

### 3.4.3 Sub Band Coders

This coder implies analysis of filter bank so as to filter the input signal into various frequency bands. There is a specific criterion for allocation of bit to each band. Such coders are not commonly used since it is quite difficult to create high quality speech signal using lower bit rates with this technique.

### 3.4.4 Parametric/Hybrid Coders

Such coders use a small set of parameters to represent the signal and describe it accurately. Parametric coders assume a clear model that assumes speech is produced by excitation of linear system, through series of periodic pulses (for voices sound signal) or noise. Such coders produce signal similar to original signal. At transmitter end, speech signal is analyzed to determine parameters. Then this information is sent at receiver end for speech signal is synthesis. Consequently, intelligible signal is produced at very low bit rates.

*Chapter 4*

# Proposed Scheme of Work

The proposed scheme of work uses a fix value for constant parameter of perceptual weighted filter (c) which is 0.85 for CELP. The results have been produced using software simulation in MATLAB. MELP results are reproduced at 2.4 kbps. The coder and encoder for both algorithms have been designed in MATLAB. Details of compression algorithms implemented are given below.

## 4.1 Introduction to Code Excited Linear Prediction (CELP)

In order to improve existing coder LPC, the idea of CELP was proposed. The core ideas of CELP include avoidance of voiced and unvoiced speech, classification of LPC, includes excitation codebook. Excitation codebook is searched for best excitation sequence. The name ‘codebook excitation’ has been derived from excitation codebook which contains ‘code’ used to ‘excite’ the synthesis filter.

CELP is among the dominant ideas in speech coding. The architecture, functionality and constraints of CELP coder are described in this chapter. Numerous variants of CELP are used in different applications. For internet voice calls and cell phones, Low Delay CELP and Algebraic CELP are used. Prominent differences among these variants lie in generation of excitation signal and information for reconstruction of speech signal. Each of the variants has different bit-rates.

### 4.1.1 The CELP Model of Speech Production

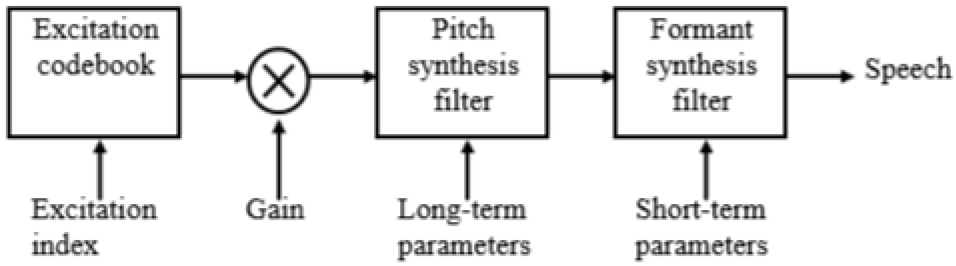



*Fig 4.2 CELP model for speech production*

Through an index, excitation sequence is extracted from the codebook. Then extracted excitation is scaled and filtered using cascaded filters, pitch synthesis filter, formant synthesis filter. Afterwards, speech is produced. Former filter creates periodicity in signal and later filter functions in generation of spectral envelope.

Codebook is either fixed or adaptive containing deterministic pulses and random noise. When compared to other coders, following differences were pointed out:

1) Voiced/Unvoiced classification in LPC coders has been eliminated. This classification is one of the major bottlenecks in LPC coders. Using cascaded system of two filters allows accurate modeling of frames ensuring smoothness hence the sound signal thus produced sounds more like natural speech signal.

2) Preservation of partial phase information – Phase information of original signal was not retained in LPC model whereas utilizing analysis-by-synthesis approach, CELP preserves some information of original signal.

CELP also works like hybrid coder where it utilizes underlying model and attempts to approximate original waveform simultaneously.

### 4.1.2 Analysis-by-Synthesis Principle

Figure below shows the analysis-by-synthesis process of CELP because the speech is encoded and decoded at encoder side to get parameters for minimizing energy of error signal. LP analysis estimates impulse response of vocal system. Afterwards encoder excites the vocal system filter and generates the synthesized speech. Error signal is the difference between synthesized and original signal. Spectral weighting of error signal then performed points out perceptually important frequencies are then minimized through optimization of excitation signal. Within frame duration, optimal excitation sequences computation is performed.

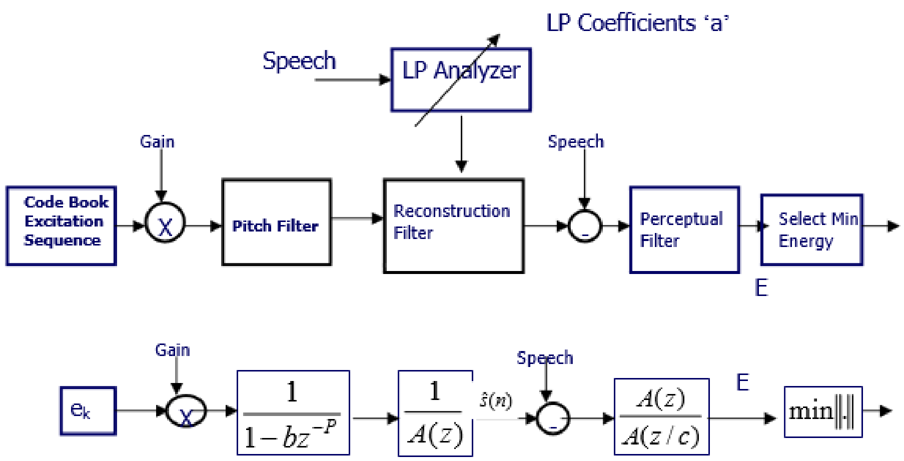



*Fig 4.3 Principle of Analysis by Synthesis*

In this project basic CELP coder is implemented with sampling rate 8k Hz. Output speech is decoded at 9.6 kbps and 16 kbps. In this implementation duration of frame foranalysis of vocal tract is 20 m sec hence there are 120 samples. Duration of block for determining excitation is 5 m sec hence there are 40 samples. Required parameters which have to be transmitted from encoder to decoder are:

1) Linear prediction coefficients 'a'
2) Gain 'G'
3) Pitch filter 'b'
4) Pitch delay 'P'
5) Codebook index 'k'

### 4.1.3 LP Analysis

Estimation of all-pole filter of frames is performed in LP Analysis which then generates spectral envelop of signal. Typically there are 10 – 12 coefficients. ‘lpc’ function is implemented to get coefficients. The CELP code implemented in this project has 12 coefficients. Lattice filter can also be implemented to obtain these coefficients. This filter gives reflection coefficients which are converted to filter coefficients. In order to reduce complexity of filter, Levinson-Durbin method is used.

$$\hat{x}(m) = \sum_{k=1}^{p} a_k \, x(m-k)$$

H(z) is IIR reconstruction filter which reproduces speech.

$$H(z) = \frac{1}{1+\sum_{i=1}^{p} a_i z^{-i}} = \frac{1}{A(z)}$$

### 4.1.4 Perceptual Weighting Filter

Synthetic speech frame is obtained as output of LP filter. This output signal is subtracted from original signal. The resulting signal error signal is obtained. It then passes through perceptual error weighting filter. This filter has following system function

$$w(n) = \frac{A(z)}{A(z/c)} = c^M \, \frac{(p_0 - z)(p_1 - z)....(p_{M-1} - z)}{(cp_0 - z)(cp_1 - z)....(cp_{M-1} - z)}$$

‘c’ ranges between 0 and 1. Practically effective value is 0.7<c<0.9. Value of ‘c’ used for analysis is 0.85. Filter coefficients are

$$\begin{aligned} A(z/c) &= 1 - a_1 (z/c)^{-1} - .... - a_M (z/c)^{-M} \\ &= 1 - (a_1 c) z^{-1} - .... - (a_M c^M) z^{-M} \end{aligned}$$

### 4.1.5 Excitation Sequence

Gaussian signals used as excitation signals for filter are contained within codebook. In this implementation, codebook contains 1024 sequences where length of each sequence is 5 m sec hence there are 40 samples. Both encoder and decoder have same codebook. Signal e(n) excites LP synthesis filter. Then excitation sequence is obtained from Gaussian codebook. A codebook containing 1024 sequences can produce toll quality speech. 9 bits are required to send index.

### 4.1.6 Pitch filter

Pitch of human voice ranges in few hertz whose frequencies correspond to pitch delay of 16 to 160 samples for 8 k Hz signal. Correlation filter has following form:

$$J(z) = \frac{1}{1 - bz^{-P}}$$

P= pitch period samples

b= pitch filter coefficient

Value of P [16,160]

### 4.1.7 Minimization of energy

Excitation sequence is described as sum of Gaussian codebook sequence and sequence from interval of past excitation given by

$$e(n) = Gd_k(n) + be(n-P)$$

This produces synthetic speech after it is applied to vocal tract filter response.

Let

Let

$$F(z) = \frac{1}{A(z)}$$

$$\begin{aligned}\hat{s}(n) &= e(n) * f(n) \\ &= Gd_k(n) * f(n) + be(n-P) * f(n)\end{aligned}$$

s^ (n) is synthetic speech

Parameters G, k, b and P are selected thus it minimizes energy of perceptually weighted error.

$$E(n) = w(n) * (s(n) - \hat{s}(n))$$

Let

$$I(z) = F(z)W(z)$$

Error signal

$$\begin{aligned} E(n) &= w(n) * s(n) - Gd_k(n) * I(n) - be(n-P) * I(n) \\ &= E_0(n) - GE_1(n,k) - bE_2(n,P) \end{aligned}$$

Where

$$E_0(n) = w(n) * s(n)$$
$$E_1(n,k) = dk(n) * I(n)$$
$$E_2(n,P) = e(n-P) * I(n)$$

Previous samples of e(n) are buffered. For simplification of optimization process, energy is minimized as follows:

Equation to determine P and b :

$$Y_2(P,b) = \sum_n [E_0(n) - bE_2(n,P)]^2$$

Differentiate above equation with respect to ‘b’, equate with 0.

$$\hat{b}(P) = \frac{\sum_n E_0(n)E_2(n,P)}{\sum_n E_2^2(n,P)}$$

Substitute this value in equation:

$$Y_2(P,\hat{b}) = \sum_n E_0^2(n) - \frac{[\sum_n E_0(n)E_2(n,P)]^2}{\sum_n E_2^2(n,P)}$$

Hence value of P minimizes $Y_2$(P) . After determining these parameters G (gain) and k (index of codebook) are chosen depending upon minimization of error energy between

$$E_3(n) = E_0(n) - \hat{b}E_2(n,\hat{P}) \text{ and } GE_1(n,k)$$

Exhaustive search of Gaussian codebook is performed to choose P and k which minimizes

$$Y_1(k,G) = \sum_n [E_3(n) - GE_1(n,k)]^2$$

This equation is solved in the same manner as described above.

| **Parameter** | **Bits/Parameter** | **Bits/Frame** |
|---|---|---|
| Codebook Index, k | 10 | 40 |
| 12 LPC Coefficients | 12 | 144 |
| Gain, G | 13 | 52 |
| Pitch filter coefficient , b | 13 | 52 |
| Lag of pitch filter, P | 8 | 32 |

*Table 4.1 Bit allocation for 16 kbps*

| **Parameter** | **Bits/Parameter** | **Bits/Frame** |
|---|---|---|
| Codebook Index, k | 10 | 40 |
| 12 LPC Coefficients | 6 | 60 |
| Gain, G | 7 | 28 |
| Pitch filter coefficient , b | 8 | 32 |
| Lag of pitch filter, P | 8 | 32 |

*Table 4.2 Bit allocation for 9.6 kbps*

In this project, vocoder has codebook which contains 1024 sequences where each sequence has length 40. Two bit-rates were selected:

High bit rate (16 kbps) CELP

Low bit rate (9.6 kbps) CELP

## 4.2 Introduction to Mixed Excitation Linear Prediction (MELP)

Owning to bandwidth constraint, low bit-rate coders are gaining increasing importance in area of wireless and digital communication. 2.4 kbps MELP vocoder has been standardized by DoD (Department of Defense) USA. MELP vocoder is enhanced version of existing linear prediction encoder. Its additional features include conversion of LPCs to LSFs for the purpose of quantization, five bandpass voicings for mixture control of noise and pulse at synthesizer and for jittery voiced frames there is a third voicing state. There are two filters at decoder side namely adaptive enhancement filter and pulse dispersion filter. These two filters improve the comparison between original and synthesized speech signal. One of the common problems in existing LPC speech that is business has been eliminated by noise mixture algorithm. Third voicing state removes tonal noises and thumps. Block diagram shows MELP model for speech production.

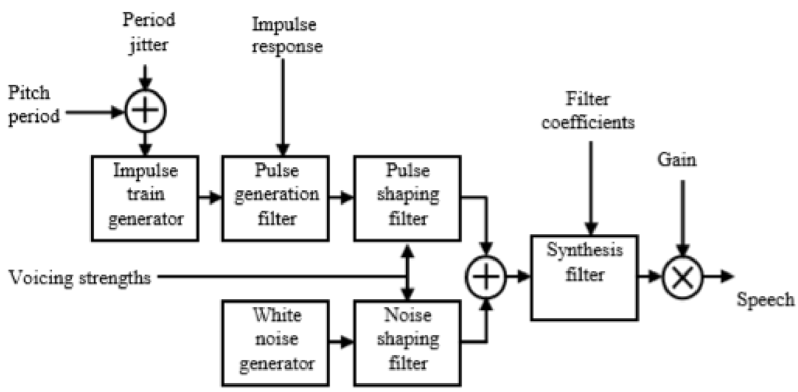



*Fig 4.4 MELP model of speech production*

### 4.2.1 MELP Analysis

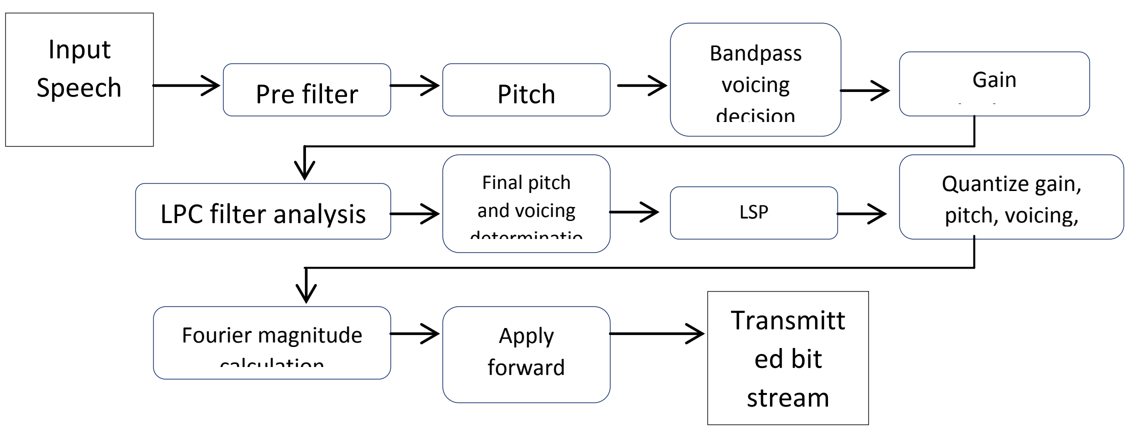


*Fig 4.5 MELP analysis*

Encoder for MELP is divided into two parts, analyzer and a quantizer. Sampling rate for incoming speech is 8000 times a second. Circular FIFO buffer is used to store each sample and then it is packed into a frame. Each sample has 180 samples or duration of 22.5 m sec. Different parameters are obtained from input signal such as pitch, $10^{th}$ order LPCs, band-pass voicings, gain, aperiodic indicator. Previous frame is buffered in the memory hence information of speech for previous and present frame is used in analysis. Hence size of analysis frame is much larger than the input frame of speech. Hence drastic changes in speech characteristic can be studied in a much better way. However, the delay must be taken into account. There is different window size for different parameter. There are 160 samples from past frame and 160 samples in current frame for pitch analysis window. There are 100 samples of previous frame and 100 samples of current frame for linear prediction analysis window. Decoder buffers parameters of both frames and performs interpolation for pitch period. Hence delay is incurred. For internal buffering, a delay of 87.5 m sec is introduced.

### 4.2.2 Pitch and Voicing Analysis

In MELP, autocorrelation analysis is used to carry out pitch detection. Coefficients of autocorrelation are

$$r(t) = \frac{c_\tau(0,\tau)}{\sqrt{c_\tau(0,0)c_\tau(\tau,\tau)}}$$

Where

$$c_\tau(m,n) = \sum_{k=-\lfloor \tau/2 \rfloor -80}^{-\lfloor \tau/2 \rfloor +79} s_{k+m}s_{k+n}$$

$\tau$ is number of lags, m and n are relative number of lags, k is index. Voicing determination is related to pitch determination.

If $0.6 \leq r(\tau) \leq 1$ then frame has periodic properties hence it is voiced. Voicing determination is carried out for five separate sub bands. To determine peakiness of residual signal, peakiness measure is taken over 160 samples. High peakiness shows that speech signal is jittery. Threshold value for peakiness is 1.34.

### 4.2.3 Linear Prediction Analysis

This is made on small fragment of speech signal where it is assumed that speech characteristics are stationary. It can be performed by two methods, autocorrelation and covariance. Set of coefficients $\{a_k\}$ thus obtained results in a stable filter.

### 4.2.4 Line Spectral Frequencies Conversion

Direct quantization may lead to loss of precision thus affecting the stability of filter. Hence this approach has been used to alternatively represent LPC coefficients. For purpose of conversion of LPC coefficients to LSFs, use of Chebyshev polynomials is made.

### 4.2.5 Fourier Magnitudes

In MELP coder, Fourier magnitudes comprise 10 pitch harmonics for magnitude spectrum of residual signal. These magnitudes are quantized and can be used in generation of pulse excitation in the synthesizer. Complex FFT of 512 points is performed on 200 samples. Output of FFT is converted to magnitudes prior to peak search. Quantization of Fourier magnitude is performed by vector quantizer for which the size of codebook is 256. Fourier magnitudes make speech more naturally sounding.

### 4.2.6 Quantization

Parameters quantized by quantizer are ten LSFs, pitch, two gains, five band-pass voicings, Fourier magnitudes, aperiodic flag. A four stage vector quantization is used for LSF quantization. Code book sized respectively are 128, 64, 64 and 64. Prior to quantization, separation of atleast 50 Hz for 10 LSFs is ensured. Then search is performed for codebook which minimizes weighted Euclidean distance d. Uniform quantization is used for gain. 3 bits are used in quantization of first gain and5 bits are used for quantization of second gain. Length of single frame is 22.5 m sec. Bit allocation per frame for MELP 2.4 kbps Vocoder is given as follows.

| **Parameters** | **Voiced** | **Unvoiced** |
|---|---|---|
| Pitch | 7 | 7 |
| LSF parameters | 25 | 25 |
| Gains (2 per frame) | 8 | 8 |
| Band-pass voicing | 4 | - |
| Aperiodic flag | 1 | - |
| Fourier magnitudes | 8 | - |
| Sync bit | 1 | 1 |
| Error protection | - | 13 |
| **Total** | 54 | 54 |

*Table 4.3 Bit allocation for MELP 2.4 kbps (voiced/unvoiced)*

### 4.2.7 The MELP Synthesizer

One prominent feature of MELP synthesizer is that it embodies pulse noise mixture control, a jittery voicing state and adaptive enhancement filter. Linear interpolation of parameters performed for each pitch period using parameters of previous and current frame. Interpolated parameters include two gains, ten LSFs, pitch, jitter strength, spectral tilt coefficient used in adaptive enhancement filter and band-pass voicings.

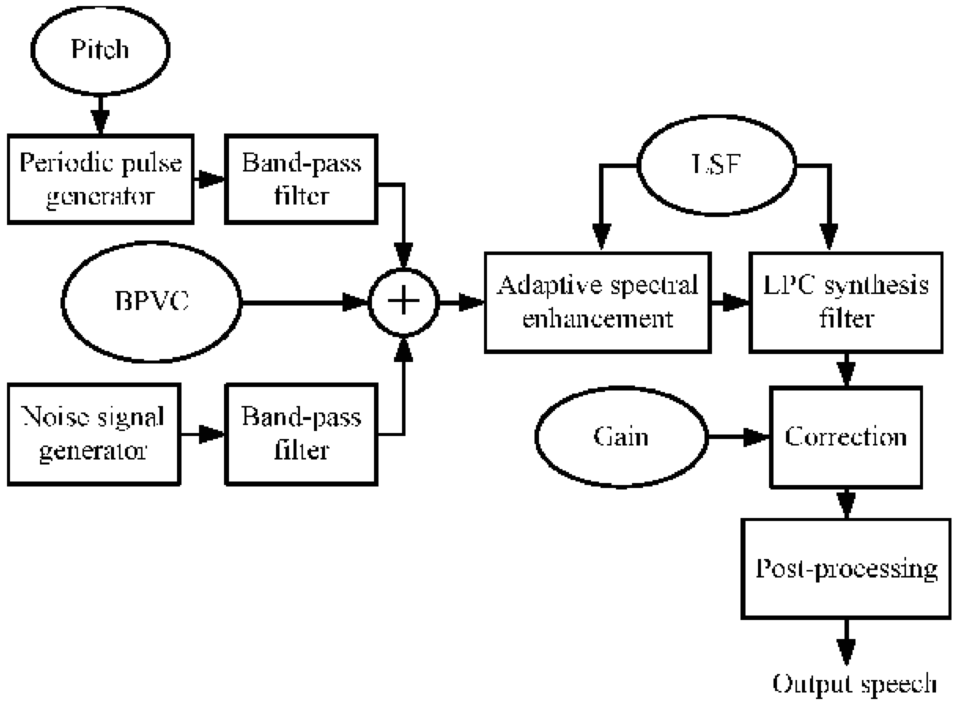



*Fig 4.6 MELP synthesizer*

### 4.2.8 Mixed Excitation and Jittery Pitches

A single pulse train is insufficient to synthesize different characteristics of natural speech. While synthesizing speech pulse and noise mixture is used. This mixture provides resemblance to characteristics of original speech signal by removal 'buziness' caused by pure pulse excitation. Fourier magnitudes generate pulse by IDFT of one pitch length. For one pitch length T, pulse excitation

$$e_p(n) = T^{-1} \sum_{k=0}^{T-1} F(k) e^{j2\pi nk/T}$$

A random number generator produces noise excitation. Prior to combining, both noise and pulse train is passed through shaping filters. If all bands are fully voiced, we get pulse as an output.

Jitter actually removes error of synthesis of weakly voiced frames by use of pure noise or synthesis of unvoiced frames by pure pulse excitation. Jitter is also produced through a random number generator. Hence in regions of drastic pitch change, periodicity is removed and hence thumps in speech are removed. Pulse / noise mixture produces both jittery voiced frames and voiced frames. If voicing strength of all bands is set to zero then unvoiced speech is produced.

### 4.2.9 Adaptive Enhancement Filter and Pulse Dispersion Filter

Mixed excitation thus generated passes through enhancement filter whose function is given by

$$H(z) = (1 + \mu z^{-1}) \quad A(\alpha z) \,/\, A(\beta z)$$

where $\alpha = 0.5p$, $\beta = 0.8p$

p = signal probability measure

$\mu$ = max (0.5k1,0)

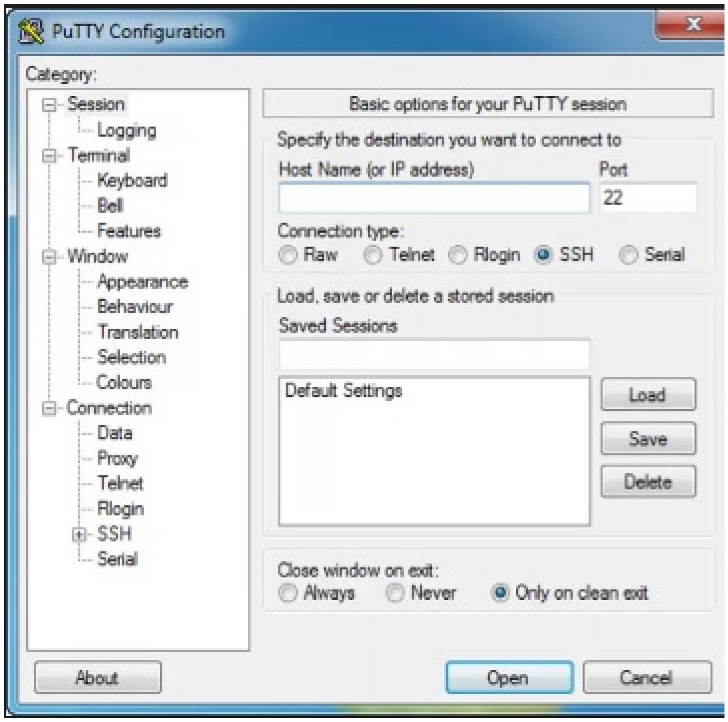
k1 = first reflection coefficient

Enhancement filter matches synthesized speech with original speech in formant regions hence perceived quality is improved. Impulse response has spectrally flattened triangular pulse. Later filter improves matching of synthesized waveform in frequency regions where there is no formant frequency.

## 4.3 Software Programs

### 4.3.1 Putty Software

It is open source software, basically an application that allows network file transfer. It can be used to connect to serial port. This software is used to access other computer. In this project, it has been used for Zigbee configuration with PC via USB port.




*4.7 Putty software window*

### 4.3.2 MATLAB 2017a

It is a language for computing technical problems. The problems and solutions are represented in mathematical notation. Ample versions of this software are available. This project has used its 2017 version.

## 4.4 Hardware

### 4.4.1 Xbee Series 2 OEM RF Modules

These modules are designed to perform operation within Zigbee protocol. The meet the requirements of wireless sensor networks with low cost and low power. Xbee requires low power for operation (3.3 V for transmission and reception operation) and provides reliable data transfer between devices. Operating frequency band is 2.4 GHz ISM. It works as a transceiver. Xbee S2 perform better in case of mesh networking since there is a greater number of nodes. For communication between two Zigbee modules, both of them must be of same type. Xbee S1 cannot communicate with Xbee S2 so care has to be taken.


*Fig 4.8 Xbee Series 2*

### 4.4.2 Xbee to USB Adapter

This tools helps to establish connection with PC using Xbee wireless modules in wireless networks. Connection is established via USB port. When the board is plugged in the USB port, the communication with Zigbee module is established. Connection with PC can also be used for configuration of Zigbee module.


*Fig 4.9 Xbee to USB adapter*

### 4.4.3 Xbee 2mW Wire Antenna

One of the key components in RF modules is antenna which can drastically affect the performance. Reduced size, lesser cost and high performance are the requirements for most RF applications hence it is mandatory to implement a proper antenna on these modules. Xbee S2 uses 2mW wire antenna for functions.


*Fig 4.10 2mW wire antenna*

Following steps were performed:

1. Signal was compressed using compression algorithm.
2. Compressed signal was converted into string.
3. Zigbee was connected via USB port to PC (Transmitter and receiver side)
4. Configuration of Zigbee using Putty software.
5. On transmitter side, signal was encoded and converted into string.
6. Using commands, signal was transmitted bit by bit.
7. On receiver side, the received string was saved in .csv file.
8. The file was read in MATLAB and decoded.
9. The decoded signal was plotted and compared to the sent signal in MATLAB.

As test simulations, string was sent and received on other side. Later on the above procedure was followed.

# *C h a p t e r 5*

# RESULTS

## 5.1 Software Simulations

Software was implemented in MATLAB 2017a.

### 5.1.1 Simulations of CELP Algorithm

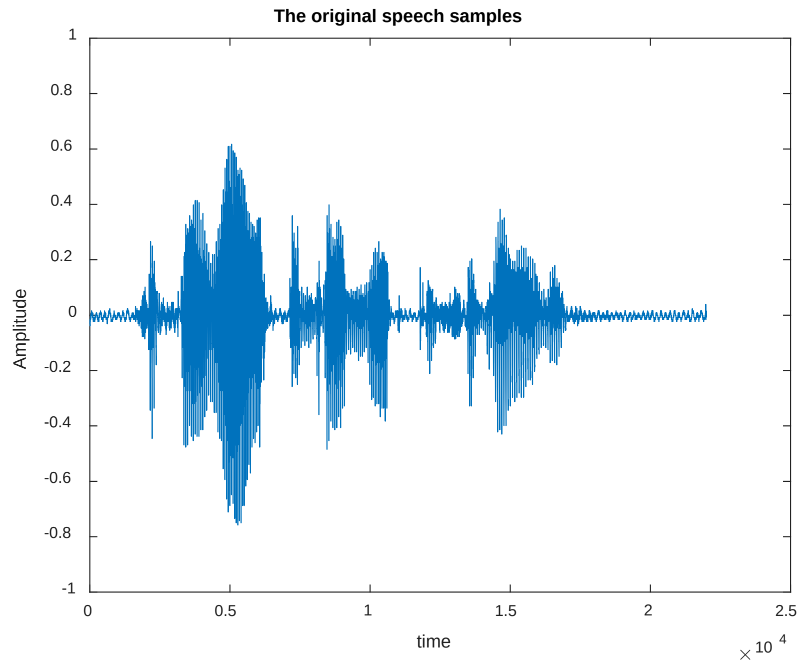



*Fig 5.1 Original CELP signal*

This figure shows the original sound signal being transmitted. This signal will be received on the other side and decoded. Above figure shows result of simulation in MATLAB.

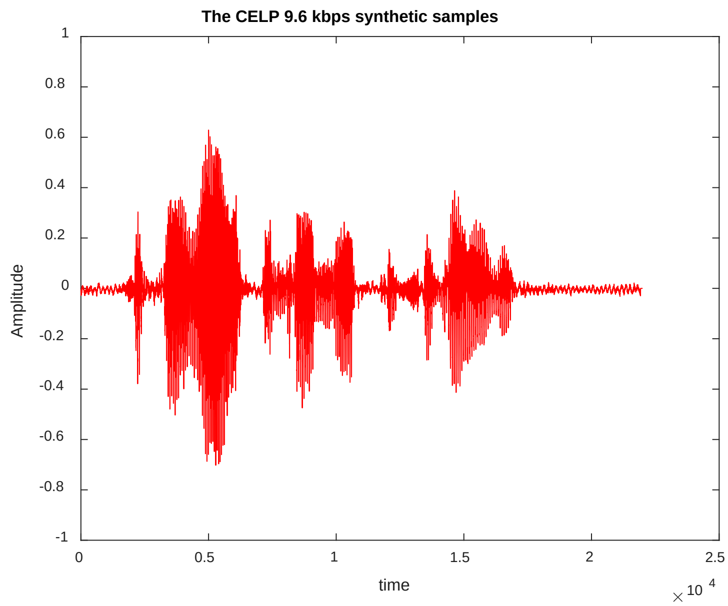



*Fig 5.2 Synthesized signal at 9.6 kbps CELP*

The original signal was compressed at 9.6 kbps after quantization and decoded, hence synthesized at specific bit rate and plotted.

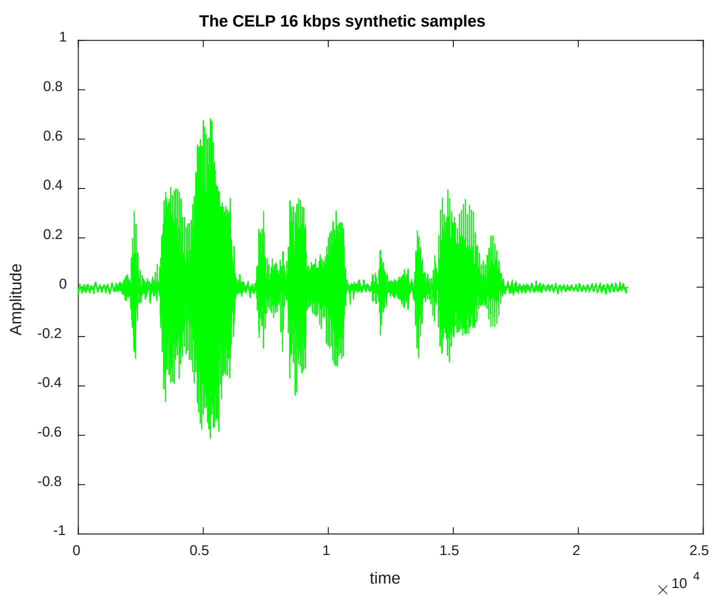



*Fig 5.3 Synthesized signal at 16 kbps CELP*

The original signal was compressed at 16 kbps after quantization and decoded, hence synthesized at specific bit rate and plotted.

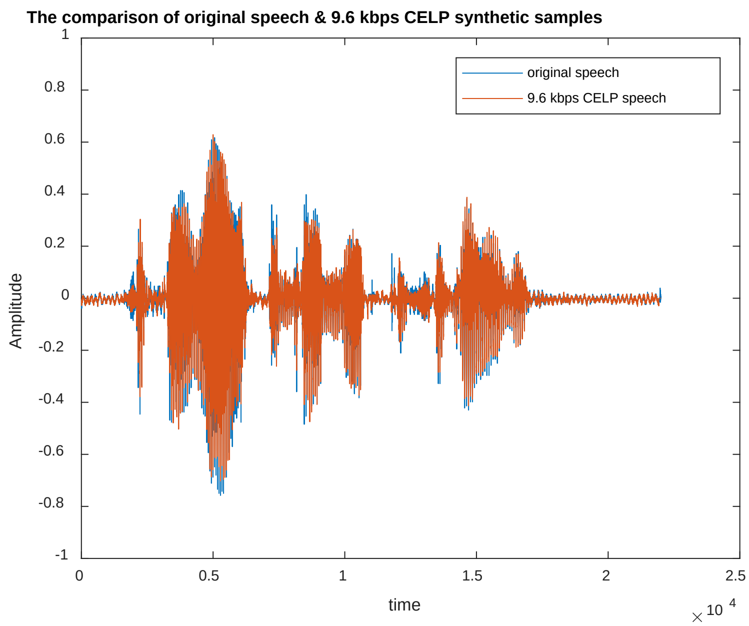



*Fig 5.4 Comparison of original signal and synthesized signal at 9.6 kbps CELP*

The figure shows comparison of original and synthesized signal. It is obvious that there is good envelop replication and voice signal heard after compression was of toll quality.

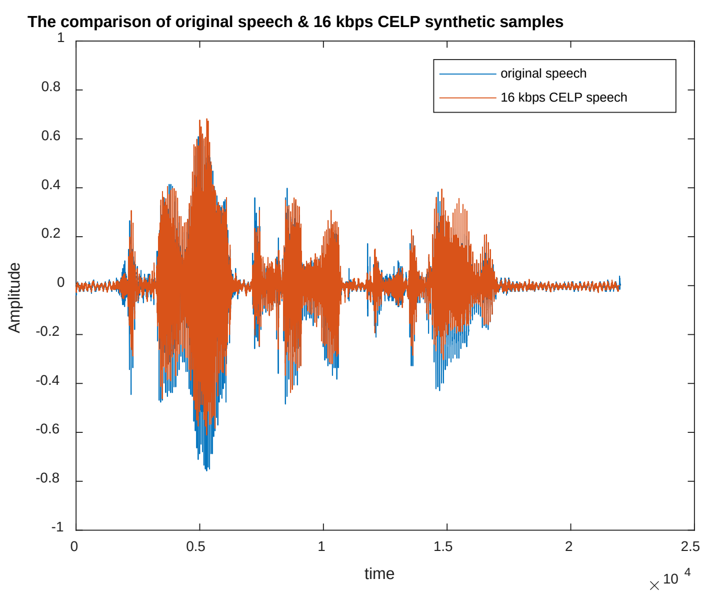



*Fig 5.5 Comparison of original signal and synthesized signal at 16 kbps CELP*

The figure shows comparison of original and synthesized signal. It is obvious that there is good envelop replication and voice signal heard after compression was of toll quality.

### 5.1.2 Hardware results of CELP Algorithm

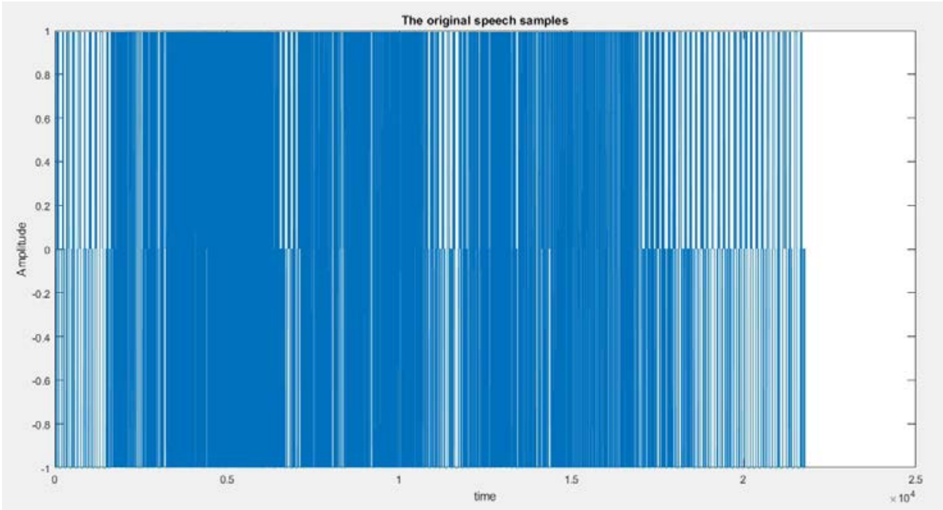



*Fig 5.6 Encoded signal transmitted via Zigbee*

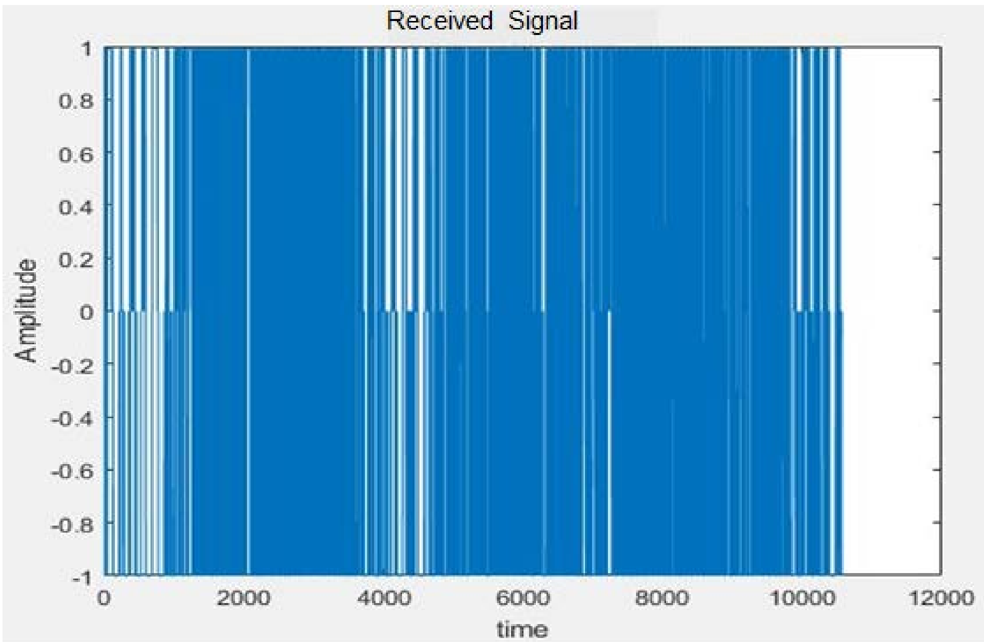



*Fig 5.7: Encoded signal received via Zigbee*

The signal transmission via Zigbee was tested for real time implementation. On transmitter side, the compressed signal was quantized and compressed, packed and transmitted as bit stream (Figure 5.6) while on the receiver side; the encoded bit stream was unpacked and again plotted to compare the results. However when we again convert .mat file to sound file in MATLAB, some noise is added in the reconstructed signal.

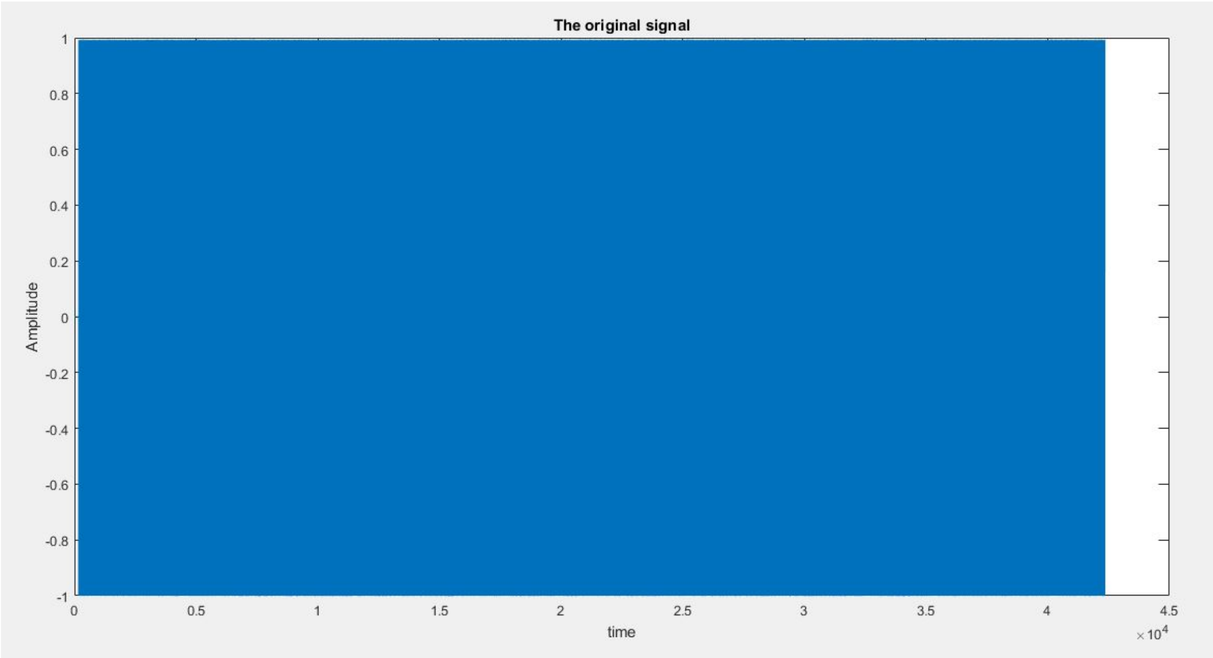



*Fig 5.8: Encoded signal transmitted via Zigbee*

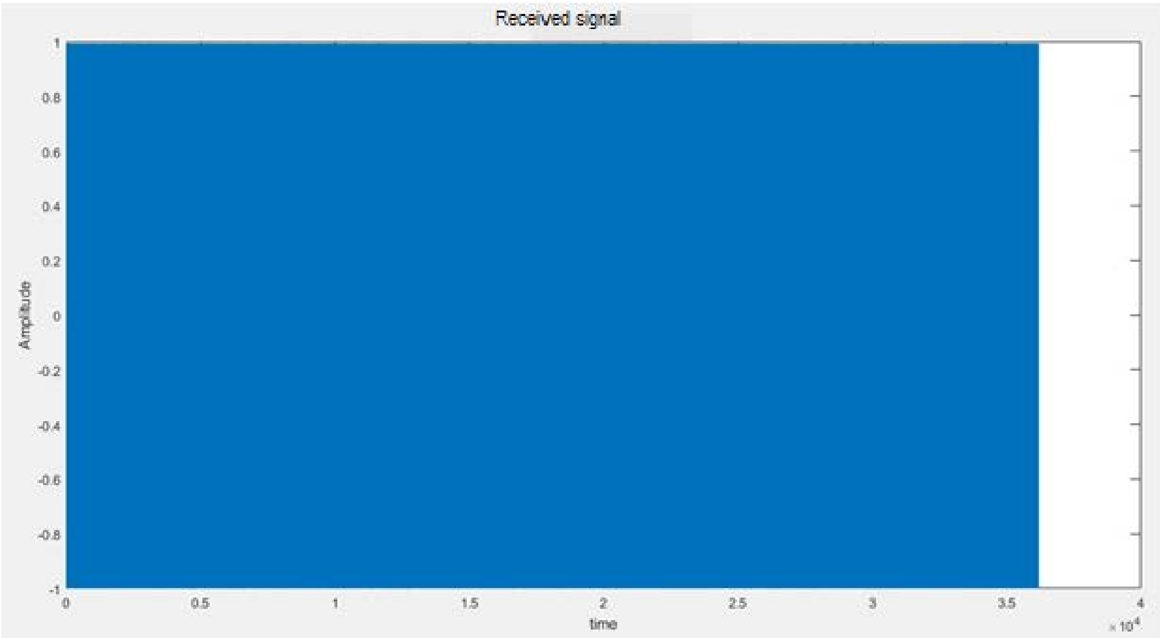



*Fig 5.9: Encoded signal received via Zigbee*

The signal transmission via Zigbee was tested for real time implementation. On transmitter side, the compressed signal was quantized and compressed, packed and transmitted as bit stream (Figure 5.6) while on the receiver side; the encoded bit stream was unpacked and again plotted to compare the results. However when we again convert .mat file to sound file in MATLAB, some noise is added in the reconstructed signal.

## 5.2 Simulations of MELP Algorithm

### 5.2.1 Software Simulations

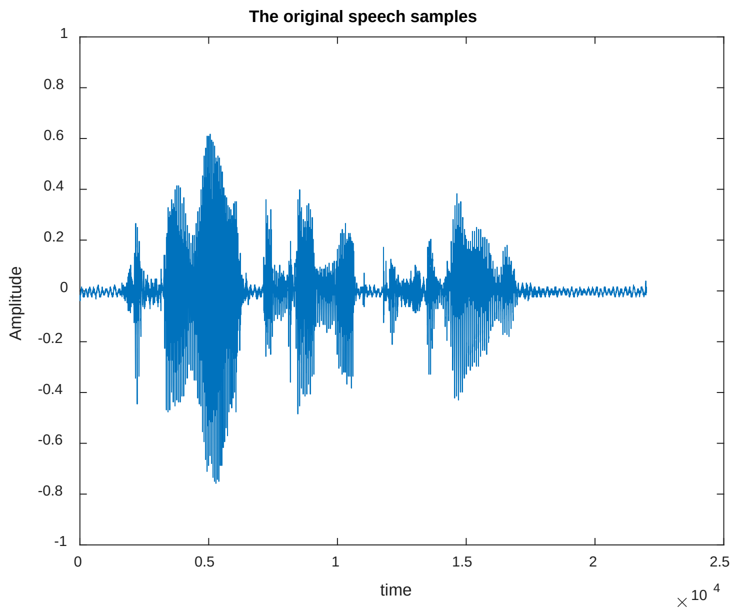



*Fig 5.6 Original MELP signal*

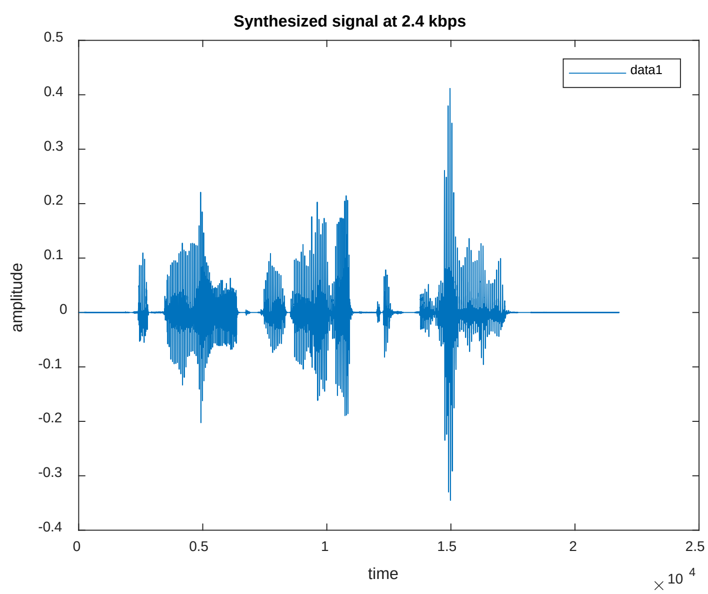



*Fig 5.7 Synthesized signal at 2.4 kbps MELP*

There was not much good envelope replication in MELP and amplitude also varied at different instances. However sound was heard and after reconstruction of signal and there was noise in the background. Post filter addition at the end can reduce the buzz and background noise enhancing the quality of sound signal.

### 5.2.2 Hardware Results

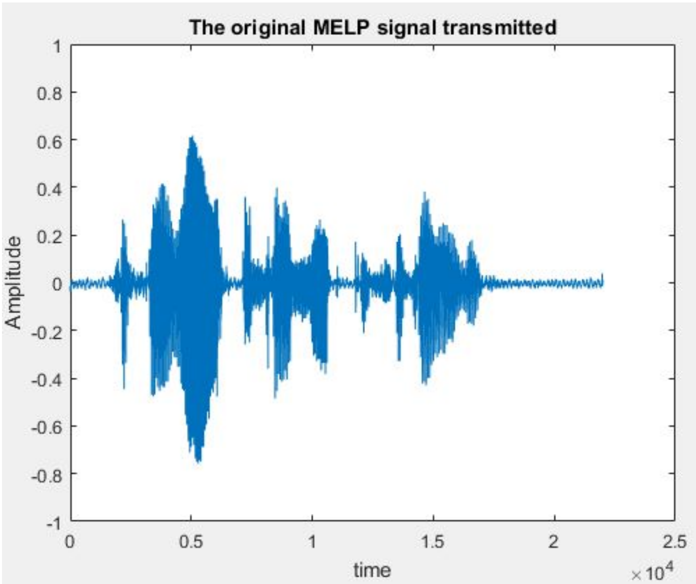



*Fig 5.10 MELP signal transmitted*

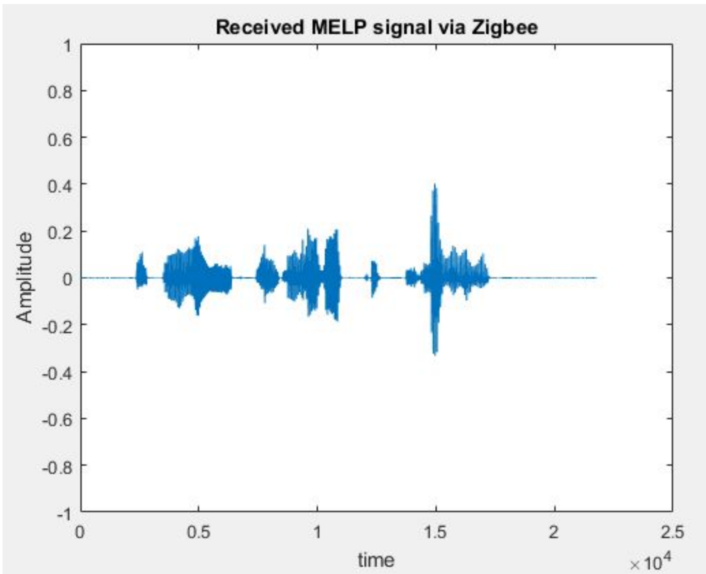



*Fig 5.11 Received MELP signal*

Transmission of MELP via Zigbee produced sound on the other side, which when played was not clearly audible since it was low in voice. Hence the voice can be increased by adding an amplifier at the end to enhance the voice quality of received signal.

## 5.3 Test Simulations

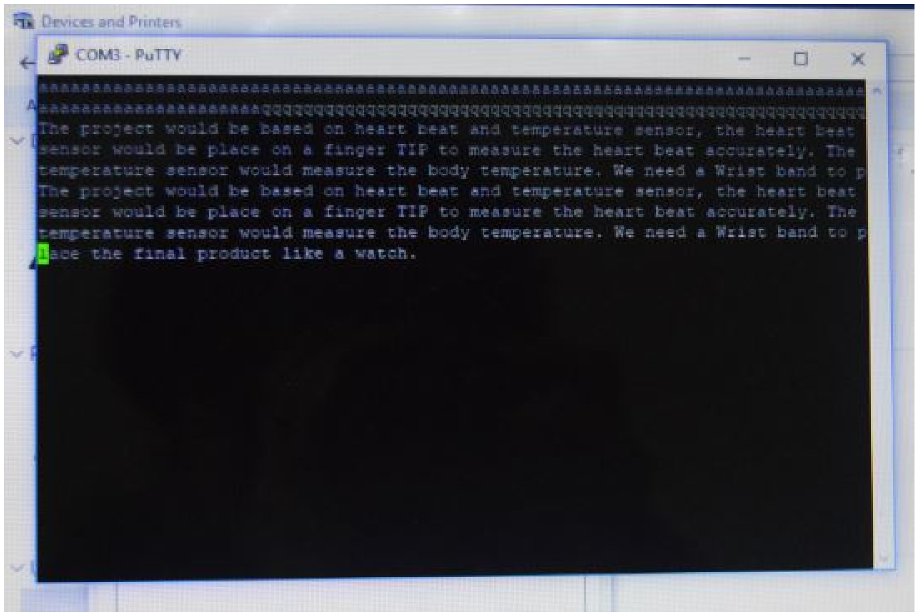



*Fig 5.12 Test string received via Zigbee*

The figure shows test string transmitted via Zigbee and received on the other side. It was just to check whether reception on other end is taking place or not.

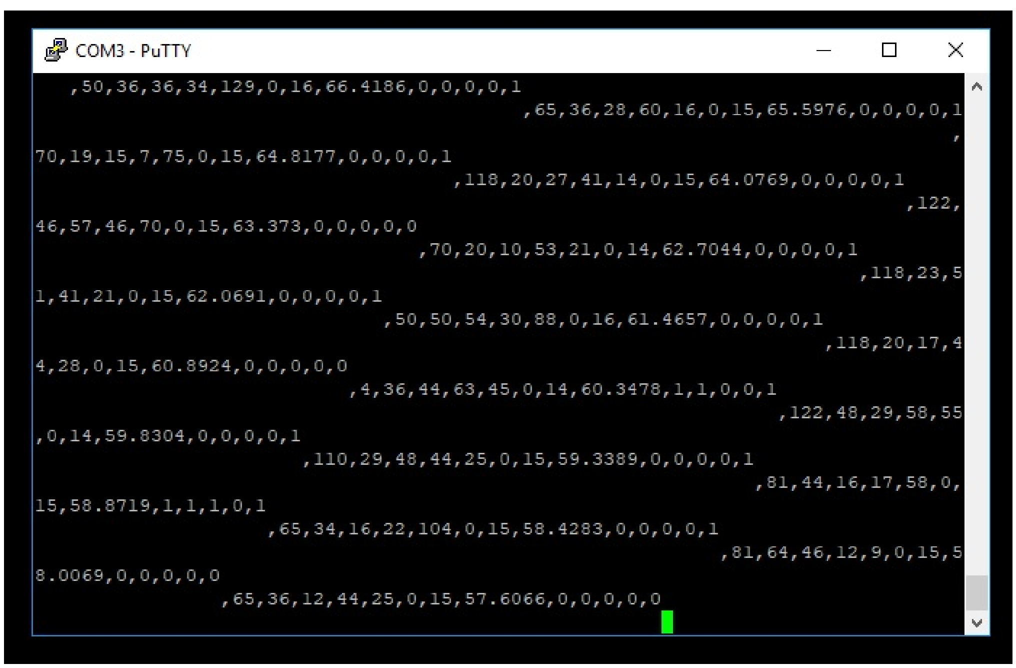



*Fig 5.13 Signal reception bit by bit on Putty software*

Figure 5.13 shows real time transmission of sound signal at receiver end. The signal was transmitted bit by bit and received on the other side.

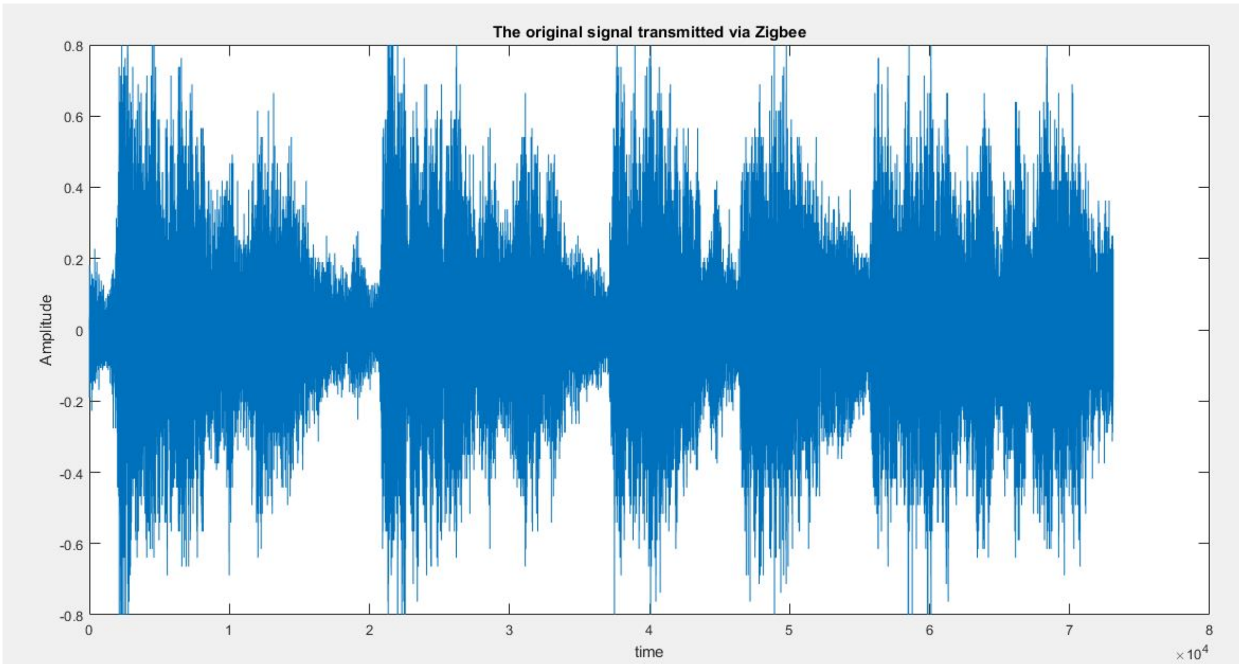
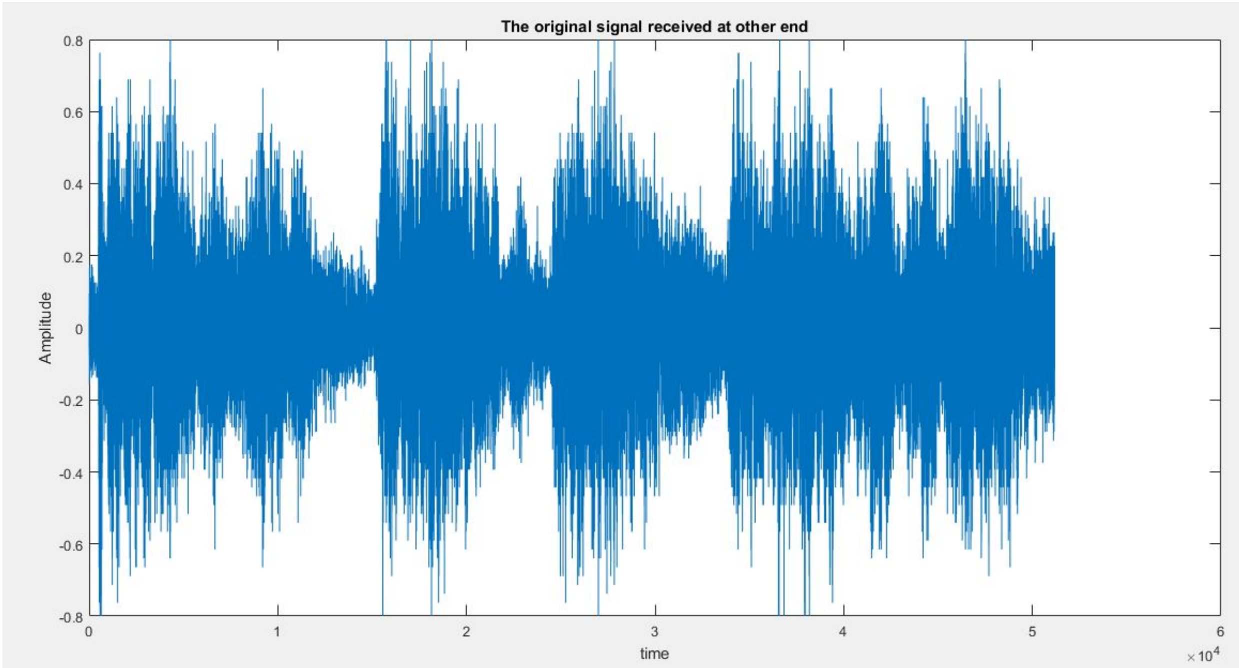



*Fig 5.14(a) - Sound signal transmitted via Zigbee*




*(b) – Signal received at other end*

The above figure shows the real time transmission and reception of sound signal via Zigbee without any compression or quantization technique applied. Around 20 bits were lost during the process.

## 5.4 Conclusion

MATLAB software results were pretty accurate. There was much better envelope replication in CELP algorithm. The synthesized signal matched the original signal. Processing time was less. However software had constraints.

While transmitting the MELP signals, some data was lost during transmission. Although the compressed signal took much less time to transmit. The original signal was transmitted in approximately 3-4 minutes at other side via Zigbee whereas compressed signal took few seconds. But the signal quality was degraded.

Hence a compromise needs to be made between time delay and quality. Compressing the signal and transmitting it via Zigbee reduces time but few bits are crapped and hence data is lost hence the received signal is not of as good quality as the sent one. Therefore Zigbee can be used for the transmission of voice. The transmission time also depends on compression algorithm employed.

CELP has been implemented in MATLAB 2017a software. The signal is analyzed at two bit rates i.e., 9.6 kbps and 16 kbps. The value of constant parameter for perceptual weighted filter (c) is 0.85. Reconstructed signal was much better,sound was also played back which was of much good quality. Among variants of CELP such as algebraic CELP, low-delay CELP, relaxed CELP, at present most practiced algorithm. This coder can replicate envelop of original speech comparatively much better than other variants.

Original signal and synthesized signal have almost envelope. There is minor variation. The sound is also played back and listened using MATLAB command. The quality of synthesized sound signal was very good and clearly audible.

MELP has been implemented in MATLAB 2017a. Though the envelope replication was not much better but sound was heard. There was low sound in synthesized signal. It is a coding algorithm defined by U.S DoD. This algorithm provides appreciably good quality than existing older military standards especially when there is noisy environment like battle field, vehicles and aircrafts.

## 5.5 Future Work

Due to increasing demands among people, speech compression is growing at a faster pace than before. Much recent advancement in this area has made the implementation of upcoming technologies much handy to implement since a lot of related work and help is available.

In order to improve the delay caused during transmission, we can use faster computers with fast processing speeds which can send and receive data at a much greater speed. For low audible speech signals on the other side, an amplifier can be used to boost the signal and make it clearly audible as in case of MELP.

## Bibliography

[1] Eunchang Choi , YunKang Hur , Jaedoo Huh , Yoonseok Nam , DaeHun Yoo , WoongChul Choi "Simulation and implementation of voice over IEEE 802.15.4 LR-WPAN", International Conference on Consumer Electronics, pp. 1-2, 2008.

[2] A. Kumar , S. Gupta, "A survey on implementation of audio over 802.15.4", International Journal of Recent Scientific Research, vol. 8, no. 5, pp. 16823-16828, 2017.

[3] J.T. Adam, "An introduction to IEEE std 802.15.4",IEEE Aerospace Conference,pp. 1-8, 2006.

[4] M. Meli, M. Gysel, M. Sommerhalder, "Using IEEE 802.15.4/Zigbee in audio application", University of Applied Sciences, Switzerland, Tech. report

[5] H. Chen, C. Zhang, "The research and implement of Zigbee technology for building wireless voice network", Proc. International Conferenec on Measurement, Information and Control, vol. 2, pp. 576-580, 2012.

[6] K. Sahal , R. Sharma, " Voice communication over Zigbee protocol: A literature review", International Journal of Engineering Research and General Science, vol. 3, Issue 3, pp. 1444-1446, 2015.

[7] Zigbee Alliance, Zigbee Specifications. Version 1.0 Zigbee Document 053474r06, Dec. 2004.

[8] E. Touloupis, A. Meliones, S. Apostolacos, "Implementation and evaluation of voice codec for Zigbee", 2011 IEEE Symposium of Computers and Communications (ISCC), pp. 341-347, 2011.

[9] L. Chang, C. Chen, T. Lee, "Voice transmission over wireless sensor networks", National Taichung University, Taichung, pp. 158-163.

[10] C. Wang, R. Jana, L. Ji, M. Daneshmand,"Voice communication over ZigBee networks", IEEE Communications Magazine,vol. 6, 121-127, 2008.

[11] L. Wenfeng, C. Guojun, "Short-distance wireless voice communication technology based

on zigbee", International Conference on Management and Service Science, pp. 1-4, 2009.

[12] N. Saleem, U. Khan, I. Ali, "Implementation of low complexity CELP coder and performance evaluation in terms of speech quality", International Journal of Computer Applications, pp. 12-16, vol. 54, Issue 6, 2012.

[13] M. Mulye, S. Jagtap. "Overview of code excited linear predictive coder". IJRDET. vol. 3, Issue 1, pp.166-171, 2014.

[14] R. Ram, H. K. Palo, M. N. Mohantay, B. N. Sahu, "Speech coding techniques : a comparative study", International Journal of Electronics and Communication Technology, pp. 29-33, vol. 6, Issue 3, 2015.

[15] R. Jage, S. Upadhya, "Implementation of CELP and MELP speech coding techniques". IJARCCE, vol. 5, pp. 702-708, 2016.

[16] R. Jage, S. Upadhya, "CELP and MELP speech coding techniques" in Proc. IEEE WiSPNET, pp. 1398-1402, 2016.

[17] L. M. Supplee, R. P. Cohn, J. S. Collura, "MELP: The federal standard at 2400 bps", IEEE, pp. 1591-1594, 1997.

[18] N. K. Saha, R. N. Sarkar, M. Rahman. (2011) "Comparison of musical pitch analysis between LPC and CELP". IJAES. Vol. 1,35-39.

[19] H. Fitraiwan, M. Susnto, A.S. Arifin, A. Trisanto. (2017) "Zigbee based wireless sensor network ad performance analysis in various environments" in Proc. ISEC, 1-4.

[20] S. C. Ergen.(2004) "*ZigBee/IEEE 802.15.4*", Berkeley.

[21] W. C. Chu, "Speech Coding Algorithms", Wiley Interscience, 2003.