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Implementation of CELP and MELP Speech Coding Techniques

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Abstract: Speech is one of the natural ways of communication amongst humans. Nowadays there is insatiable demand for speech communication as it carries more information like speaker identity, emotional state, prosodic nuance which adds naturalness in communication. With rapid growth and increased number of applications there exists a need for devising an approach for data compression techniques which reduces communication cost by using available bandwidth and storage space effectively. The speech coding techniques helps to achieve lower bit rate by simultaneously maintaining the original speech quality.This paper aims at implementing the CELP (Code Excited Linear Prediction) and MELP (Mixed Excitation Linear Prediction) coding techniques of speech using MATLAB R2009a. These coding techniques are analyzed on the basis of subjective and objective tests like Mean Square Error (MSE), Mean Opinion Score (MOS) and Signal-to-Noise Ratio (SNR). The analysis of CELP speech coding technique shows that this technique is an improvement to a coder called Linear Predictive Coder (LPC). It is an efficient coding technique having bit rate in the range 9.6 kbps to 16 kbps. The analysis of MELP coding technique shows that this coder removes the voicing error in two state excitation model of LPC. It is a low bit rate coder having a bit rate of 2.4 kbps and mainly used by military and Federal Standards. The comparative results of different values which are obtained for CELP and MELP coding techniques give a clear idea about more efficient coding technique.

Keywords: Speech coding, Linear Prediction (LP), CELP (Code Excited Linear Prediction), MELP (Mixed Excitation Linear Prediction)

I. INTRODUCTION

Speech is a special type of non-stationary signal which is hard to analyze and model. The factors like intelligibility, coherence and other characteristics play a vital role in the analysis of the speech signals. In communication the number of discrete values required to describe one second of speech signal corresponds to 8000 samples. Therefore, speech signals are compressed before being transmitted, as bandwidth is the parameter which affects the cost of processing. Speech coding is concerned with obtaining a compact digital representation of speech signals for the purpose of efficient transmission and storage over band limited wired or wireless channels. Using speech coding a telephone company can carry out more voice calls on a single fiber link or cable. In Mobile and Cellular communications where the data rates for a particular user are limited, speech coders can give accommodation to more services. Speech coding is a useful technique for Voice over IP, Video conferencing and Multimedia applications which reduces the bandwidth requirement over internet and for the tetherless transmission of information. Also few applications of speech require minimum coding delay since long coding delays hinder the flow of the speech conversation [1]. These coding techniques can be classified based on bit rate, bandwidth and speech coders used. In this paper the parametric speech coding technique i.e. MELP and hybrid speech coding techniques i.e. CELP are simulated using MATLAB software. It also aims at finding the subjective and objective parameters of both the coding techniques and compares them to get the more efficient coding technique.

II. RELATED WORK

Literature available for the speech coding techniques used in communications is vast, emerging and continuously growing. In this paper, many technical papers of authentic publication, such as, IEEE, Springer, Elsevier and journals are referred which will be used as a reference while implementing Code Excited Linear Prediction (CELP) and Mixed Excited Linear Prediction (MELP) speech coding techniques. This section describes the several standard technical papers available and also explains different methods involved in coding the speech signals. The various approaches of the research work on speech coding techniques are illustrated briefly in this section. The paper [2] gives an overview of methodologies for speech coding with emphasis on popular methods that have became part of many communication standards. It mainly presents historical perspective, brief discussion on human speech production mechanism, speech coding methods, and performance measures. A novel approach to speech coding using hybrid architecture is presented in the paper [3]. Advantages of parametric and perceptual coding methods are utilized to create a speech coding algorithm assuring better signal quality than CELP parametric codec. It mainly discusses two approaches; one is based on selection of voiced signal components which are encoded using parametric algorithm and unvoiced components. On

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the other hand the second approach uses perceptual encoding of the residual signal in CELP codec. The Code Excited Linear Predictive (CELP) coding technique falls under the category of hybrid coding. It is a low bit rate coding technique and the prime use of this technique is for communication purpose. The literature of CELP is described further. The paper [4] discusses a post-processing technique which improves the coding quality of CELP under background noise without any modification in the codec structure and performs the smoothing for the decoded spectral parameters and the excitation signal energy. It adaptively smoothes both, the spectral envelope and the energy of the estimated excitation signal. Thus, the proposed post-processing is performed separately from the decoder. The paper [5] describes extensions of the 4 kbps hybrid MELP/CELP coder, up to 6.4 kbps and down to 2.4 kbps. These coders form a close family and they share most of the encoder analysis, quantization tables and decoder synthesis. Their coding structure leads to coders that perform better at a given bit rate than MELP or CELP and better than equivalent higher bit-rate ITU standards. The paper [6] presents an idea about a CELP coder with a stochastic multi-pulse (STMP) codebook and training procedures for codebook excitation signal. The Linear Predictive Coding (LPC) residual exhibits a certain structure due to non linearities in the glottal excitation. This structure can be exploited by relinement of the STMP excitation signal, as a training procedure for the codebook. In paper [7], 16 kbps CELP coder with a complexity as low as 3 MIPS is presented. The main thrust is to reduce the complexity as much as possible while maintaining toll-quality. This Low Complexity CELP (LC−CELP) coder has certain features like fast LPC quantization, 3-tap pitch prediction with efficient open loop pitch search and predictor tap quantization, backward adaptive excitation gain, a trained excitation codebook with a small vector dimension and a small codebook size. Further part of this section describes the current knowledge as well as theoretical and methodological contributions to MELP coder since former days and usually proceeds with the results found in their work. The papers which were referred for studying the MELP speech coder are as follows. In paper [8] the author describes MELP 2400 bps vocoder implementation and evaluation on the basis of DAM (Diagnostic Acceptability Measure). The autocorrelation technique is used for determination of LPC coefficients and adaptive spectral enhancement allows the vocoder to better match the voiced speech waveforms. It shows that the additional information of Fourier series improves the quality of coded speech. The author of the paper [9] explains the 600 bps vocoder, which provides significant increase in secure voice availability compared to 2400 bps vocoder. The 600 bps vocoder takes the advantage of inherent interframe redundancy of MELP parameters. This paper also evaluates the coder on the basis of subjective tests like DRT (Diagnostic Rhyme Test) and DAM. The paper entitled in [10] represents a system to encode speech with high quality, using MELP coding technique which is effective at bit rates of 1.6−2.4 kbps. MELP model produces significantly higher speech quality at bit rates above 2.4 kbps. From an extensive speech quality study in this paper and for bit rate above 2.4 kbps it is clear that high transmission rate for the voicing strengths and an accurate encoding of the LP parameters are perceptually important.

III.CELP AND MELP SPEECH CODING TECHNIQUES

A. CELP speech coding technique

CELP is an efficient closed loop analysis by synthesis hybrid coding technique for narrow band and medium band speech coding. Here excitation waveform is obtained by optimizing the position and amplitude of a fixed number of pulses to minimize an objective measure of performance. It is employed to accomplish best quality speech with low computational complexity [11]. Most popular coding systems in the range of 4-8 Kbps bit rate use CELP. This technique is widely used for toll quality speech at 16Kbps. The basic principle exploited by speech coders is that speech signals are highly correlated waveforms.

1. CELP system block diagram

The CELP Analysis-by-Synthesis system is as shown in Fig.1 where encoding and decoding of speech takes place at the encoder and the parameters which minimize the energy of error signal are found at the encoder. LP analysis is used to find the vocal system impulse response in each frame. The error signal is perceptually weighted to emphasize important frequencies and it is minimized by optimizing the excitation signal. The excitation signal is updated over four blocks within the frame. The proposed CELP coder has a frame duration of 20ms and block duration of 5 ms for finding the excitation. The encoder needs information about linear prediction coefficients (a), gain (G), pitch filter (b), pitch delay (P) and codebook index (k). After calculating these parameters they are sent to decoder. The linear prediction analysis estimates all pole filter in each frame which is used to generate spectral envelope of the speech signal. The filter has 10 LP-coefficients and makes use of Levinson's Durbin algorithm which reduces the complexity of the filter. The output of LP Analysis is error signal which is passed through the perceptual error weighing filter to control the noise level. The Gaussian codebook in the implemented system has number of Gaussian signals which are used as excitation signals for the filter. The Gaussian codebook with 512 sequences yields good quality speech with code index value as 9 bits. The pitch filter is used as a long delay correlation filter to generate pitch periodicity in the voiced speech. For energy minimization between original speech signal and synthetic speech the parameters G, k, b and P are determined over a particular frame [12].

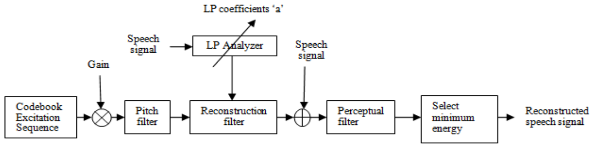

Fig.1 Block diagram of CELP coder [12]

2. Post filtering in CELP
The perceptual weighting filter is used inside search loop for best excitation in the codebook. When there is some distortion remaining in the reconstructed speech, it is termed as roughness or coding error which is a function of frequency and too high at regions between formants and between pitch harmonics. Thus several coders employ a post filter that operates on reconstructed speech to de-emphasize coding error between formants and pitch harmonics. This process is known as “Post-processing” [13].

B. MELP speech coding technique
The Mixed Excitation Linear Prediction (MELP) algorithm is the linear prediction based parametric speech coder that was chosen as the new 2.4 kbps US Federal Standard (FS). Even though the MELP technique is quite good, there are still some perceivable distortions, particularly around non-stationary speech segments and for some low pitch male speakers. In MELP speech coding the input speech signal parameters are estimated first which are then used to synthesize speech signal at the output. In these coders the samples of input speech signal are buffered into frames and are given to linear prediction filter. The frame can be represented by filter coefficients and a scale factor. It utilizes parameters like mixed excitation, aperiodic pulses, adaptive spectral enhancement, pulse dispersion filtering and Fourier magnitude modelling to capture the signal dynamics [14].

1. Block diagram of MELP speech coder
A block diagram of speech production model for MELP coder is shown in Fig.2, which is an endeavour to improve the LPC model. Implementing a MELP coder mainly involves four steps: analysis, encoding, decoding and synthesis. The MELP coder divides the speech signal into three classes: voiced, unvoiced, and jittery voiced. From the input speech signal the shape of the excitation pulse is extracted for the periodic excitation and it is transmitted as information on the frame. The pulse shape contains significant information which is captured by the MELP coder through Fourier magnitudes of the prediction error. These quantities are needed for the generation of the impulse response of the pulse generation filter. It is also responsible for the synthesis of periodic excitation. The periodic excitation and noise excitation are filtered using the pulse shaping filter and noise shaping filter, respectively and later these outputs are added together to form the total excitation called as mixed excitation [15].

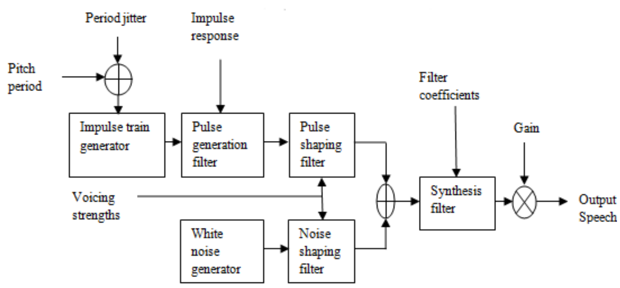

Fig.2 MELP model of speech production [15]

2. Shaping filters in MELP coding technique
The MELP speech production model makes use of two different shaping filters which combines pulse excitation with noise excitation and form mixed excitation signal. By time varying the voicing strengths, pair of time varying filters is formed and these filters decide the amount of pulse and the amount of noise in the excitation, at different frequency bands. The shaping filters mechanism controls the frequency response so as to achieve the voicing strengths. The two filters which are used here works in complementary fashion. When the gain of one filter is high, at the same time gain of the other filter is proportionately lower, and hence the total gain of the two filters remains constant at all times [15].

IV. IMPLEMENTATION OF CELP AND MELP SPEECH CODING TECHNIQUE

The coding techniques implemented here are CELP and MELP. The Fig.5 and Fig.6 are the flow charts of CELP and MELP techniques which explains the workflow and processing of these techniques and helps to analyse and design these coding techniques. The recording of speech signal is done with the help of Praat software. The simulation is done for various speech samples and their subjective and objective parameters are calculated while analysing these techniques.

A. Introduction to PRAAT software
Praat can read sounds recorded with the program or audio files recorded. Here the different speech signals are recorded using 5.4.21 version of Praat software. The sampling frequency F_s of the speech signal is set to 8000 Hz. The Fig.3 shows the PRAAT window and Fig.4 shows the speech recorded using PRAAT software at sampling frequency of 8000 Hz.

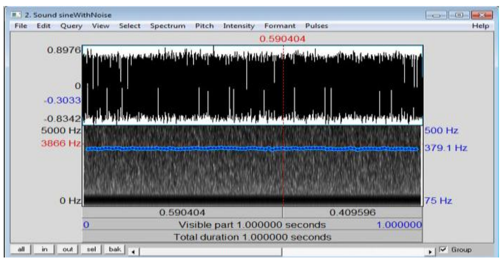
Fig.3 Praat window

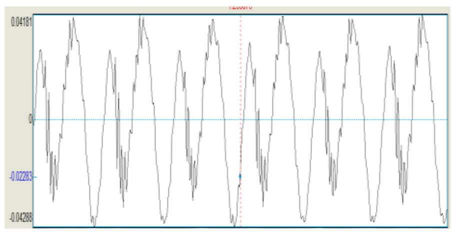
Fig.4 Speech signal recorded using Praat software

B. Flowchart of CELP speech coding technique
The Fig.5 below is the flowchart of Code Excited Linear Prediction speech coding technique. The algorithm begins with the recording of speech file in PRAAT software with

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sampling frequency of 8000 Hz. The recorded speech signal is loaded into MATLAB using “wavread” command. After loading the speech signal, different parameters like frame length (L), order of LP analysis (M), constant parameter for perceptual weighted filter (c), Pidx range are given certain values. After the analysis by synthesis principle begins and the Gaussian codebook is created. This codebook is searched for obtaining the code vectors and later the 9.6 kbps and 16 kbps speech coders are invoked using a function in MATLAB. The Levinson Durbin algorithm is used for synthesis part. At the end of the algorithm the graphs are plotted which includes original speech signal, synthesized speech signal at 9.6 kbps and 16 kbps as well as a comparison plot of different bit rate signal with original speech signal.

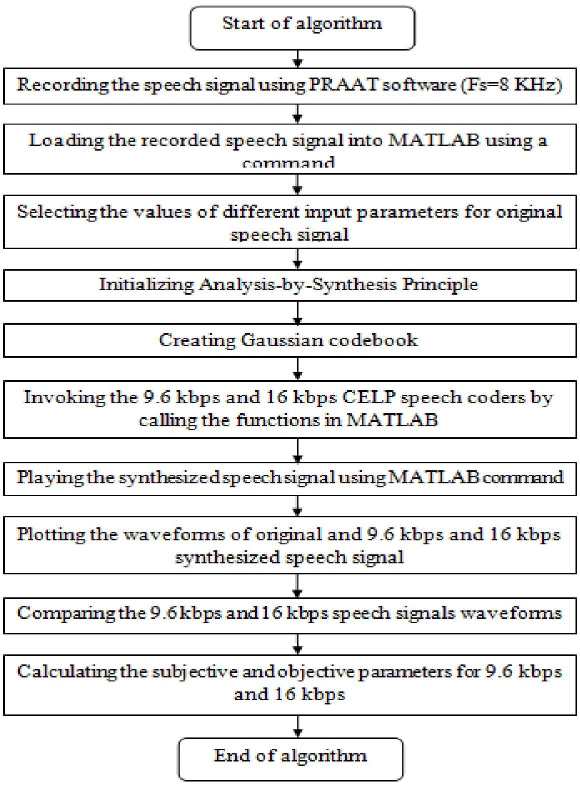

Fig.5 Flowchart of CELP speech coding technique

C. Flowchart of MELP speech coding technique

The Fig.6 below shows the flowchart of Mixed Excitation Linear Prediction technique. The MELP algorithm begins in same way as in CELP. The speech signal recorded in PRAAT is used for further analysis and synthesis. At the beginning of the program the prediction order and frame size of hamming window are defined. The zero padding is done in the signal if needed and after this step the original signal is multiplied with the window. Thereafter Levinson and Durbin algorithm is applied after this step and voiced and unvoiced decision is taken on the basis of filters. Subsequently the gain for the speech frame is calculated and the value of pitch is determined. At this point of program the analysis stage is completed and synthesis stage begins. Again the voiced and unvoiced frames are checked and the speech signal is converted into a single sequence and synthesized speech is generated which is plotted in MATLAB.

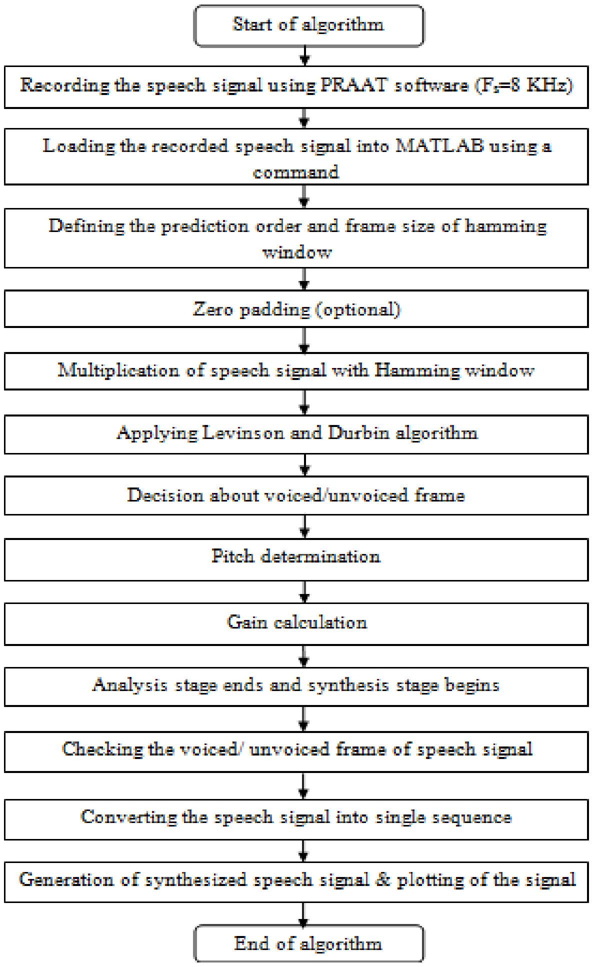

Fig.6 Flowchart of MELP speech coding technique

V. SIMULATION RESULTS

A. Results of CELP speech coding technique

The CELP coder is implemented using MATLAB R2009a. It was simulated on different speech inputs like /a/, /i/, /o/, /uu/ vowels, word ‘hello’, ‘h’ file and on a sentence. The original speech sound is mono-sound recorded using PRAAT software sampled at 8000 Hz. The CELP coder analysis by synthesis began by loading the original speech into MATLAB using ‘wavread’ command and creating a Gaussian codebook. Different functions are written in MATLAB for invoking the CELP coders.

This technique is implemented for low as well as high bit rate i.e. 9.6 kbps and 16 kbps respectively. For different values of perceptual weighted constant (c) the different figures are obtained. This constant mainly helps to reduce the perceptual weighted error in the synthesised speech signal. Fig.7 is a plot of ‘h’ file along with a plot of 16 kbps and 9.6 kbps CELP synthesized speech signal. The Fig.8 and Fig.9 are the plots of ‘h’ file for c=0.9 at 16 kbps and 9.6 kbps and Fig.10 and Fig.11 are the plots for ‘h’ file for c=0.5 at 16 kbps and 9.6 kbps. All figures have “time” on x-axis and “amplitude” on y-axis.

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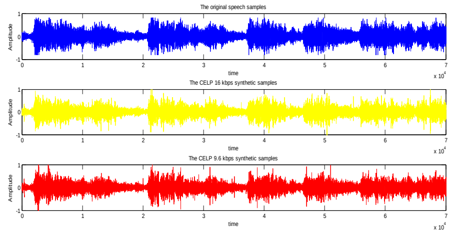

Fig.7 ‘h’ file plot along with 16 kbps and 9.6 kbps CELP synthesis

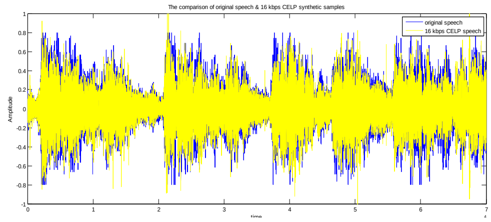

Fig.8 ‘h’ file compared with 16Kbps synthesized signal [c=0.9]

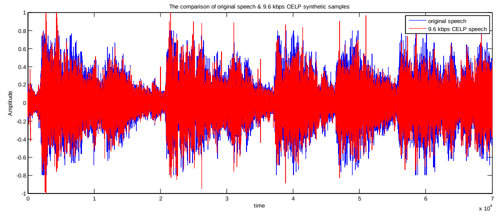

Fig.9 ‘h’ file compared with 9.6Kbps synthesized signal [c=0.9]

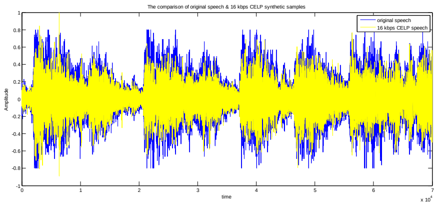

Fig.10 ‘h’ file compared with 16Kbps synthesized signal [c=0.5]

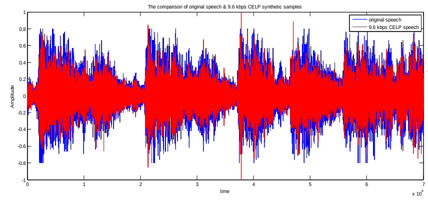

Fig.11 ‘h’ file compared with 9.6Kbps synthesized signal [c=0.5]

B. Analysis of CELP speech coding technique

Here ‘h’ file and a ‘sentence’ are synthesized using CELP technique. These speech signals are analysed at 16 kbps and 9.6 kbps for different values of constant parameter for perceptual weighted filter (c). It is observed that for lower value of c i.e. c=0.5 reconstruction of speech signal is better and audio quality and intelligibility is also maintained. Hence it can be concluded that CELP technique depends on value of c. The value of c ranges from 0.5 to 0.9 giving appropriate result for c=0.5. The mean square error values obtained for different bit rates and for c=0.9 and c=0.5 are as given in Table 1 Mean opinion score (MOS) values are also calculated for different signals are given in Table 2. The SNR values calculation for different bit rate and different c values are given in Table 3.

Table 1.Comparison of MSE for 16 kbps and 9.6 kbps synthesized speech for different values of ‘c’ (perceptual weighted constant)

Speech signals	MSE values			
	16 kbps		9.6 kbps	
	c=0.9	c=0.5	c=0.9	c=0.5
‘h’ file	0.0318	0.0019	0.1746	0.0318
‘sentence’	0.1131	0.0916	0.0064	0.0019

Table 2.Mean Opinion Score (MOS) calculation for different speech signals

Speech signals	“MOS” calculation (Subjective evaluation)
“ h file”	3.9
Sentence	3.7

Table 3.Signal-to-Noise ratio calculation for different values of ‘c’ at different bit rates

Speech signals	SNR(dB)			
	16 kbps		9.6 kbps	
	c=0.9	c=0.5	c=0.9	c=0.5
‘h’ file	104.3526	102.8461	108.1873	101.0187
‘sentence’	103.8115	104.9030	110.4880	103.6551

C. Results of MELP speech coding technique

The MELP coder is implemented using MATLAB R2009a. It was studied on different input speech signals like /a/, /i/, /o/ vowels, /fa/, /mi/, word ‘hello’ and on different sentences like ‘mysp1’, ‘mysp2’, ‘mysp3’, ‘savitha1’. The original speech sound is a mono-sound recorded using PRAAT software having sampling rate of about 8000 Hz. The MELP coder analysis began by

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loading the original speech using 'wavread' command. Different functions like mygain, mylevinson, mypitch, myzcr are written in MATLAB for invoking the MELP coders. The Fig.12 and Fig.14 are the plots of original speech signal and synthesized speech signal. For all these plots the 'x' axes have amplitude and 'y' axes have time in milli-seconds (ms). In the Fig.13 and Fig.15 the 'x' axes have frame length of speech signal in milli-seconds and 'y' axes have pitch values in hertz (Hz). These figures are plotted for different values of frame length i.e. 30 ms and 25 ms.

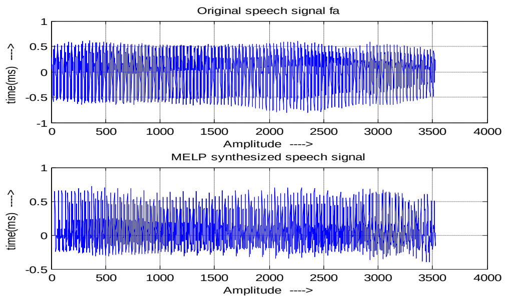

Fig.12 /fa/ Original and synthesized speech signal

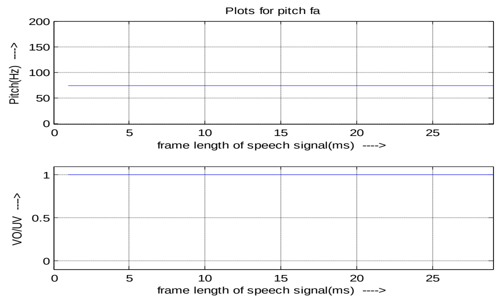

Fig.13 /fa/ Plot for pitch for /fa/(Frame length= 30 ms Fig.20&21)

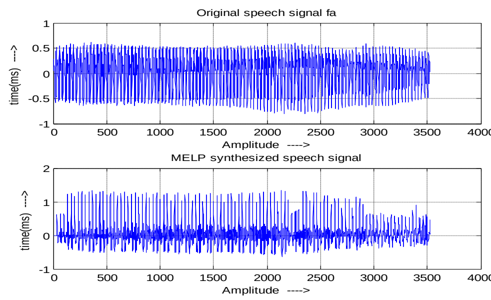

Fig.14 /fa/ original and synthesized speech signal

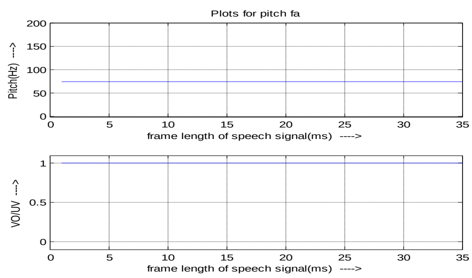

Fig.15 /fa/ Plot for pitch for /fa/(Frame length=25 ms Fig.22&23)

D. Analysis of CELP speech coding technique

The different speech signals synthesized by MELP technique are vowel /o/, /fa/ /mi/, /mysp2/. These signals are analysed for two different values of frame length i.e. 30 ms and 25 ms. The window used in MELP synthesis is hamming window. The synthesized speech signal quality is better for higher value frame length (frame length= 30 ms) and it goes on degrading as the value of frame length decreases (frame length= 25 ms) further. The subjective evaluation is done on the basis of MOS values as shown in Table 4. The synthesized speech signal is evaluated by ten different users and it is rated as per ITU recommendation. The objective evaluation is done by calculating the MSE (Mean Square Error) values and SNR (Signal to Noise Ratio) values for different values of frame length as shown in Table 5 and Table 6 respectively. It can be concluded that MELP technique depends on value of frame length and also on type of window used.

Table 4. MOS values calculation

Speech signals	MOS values calculation
/fa/	3
mysp1	3.5

Table 5. MSE values calculation for different values of frame length

Speech signals	MSE values for different values of frame length	
	Frame length = 30 ms	Frame length = 25 ms
/fa/	0.1682	0.2291
mysp1	0.0528	0.0597

Table 6. SNR values calculation for different values of frame length

Speech signals	SNR values for different values of frame length	
	Frame length = 30 ms	Frame length = 25 ms
/fa/	65.8072	72.2378
mysp1	85.8439	87.787

E. Comparison of results obtained for CELP and MELP speech coding technique

This section includes the comparative study of CELP and MELP speech coding techniques on the basis of MOS, MSE and SNR values obtained during simulation. The MOS values obtained for different speech signals in CELP and MELP technique are as shown in Table.2 and Table 4 respectively. By observing these values it can be concluded that CELP technique gives toll quality speech when compared to MELP technique. This can be said because higher MOS values are obtained in CELP technique. The MSE values are obtained for different bit rate for CELP technique are as shown in Table 1, similarly for MELP technique MSE values for different values of frame length are as shown in Table 5. By noticing these values, CELP coded speech is observed to be less accurate when compared to MELP coder. The reason may be due to the poor way of representation of excitation signal in CELP coding technique. Among these methods SNR performance of CELP coder is better than MELP. The improved performance of CELP is due to the high SNR characteristics of speech signal present in the glottal closure phase. The higher values of SNR as obtained in Table 3 shows that the signal strength is stronger in relation to the noise levels and it offers higher data rate and better throughput. The CELP coder uses the code book to represent the excitation signal which introduces more approximation in the synthesized speech when compared to MELP coding technique. Hence it can be concluded that the performance of CELP coding technique is better when compared to MELP coding technique.

VI. CONCLUSION AND FUTURE WORK

It can be concluded from simulation that CELP technique provides toll quality speech than existing low bit-rate algorithms, such as RELP and LPC vocoders for lower values of c and works well in a range of 9.6 Kbps to 16 kbps bit rate. Along with its variants, such as algebraic CELP, relaxed CELP, low-delay CELP, it is currently the most widely used coding algorithm. The MSE values obtained are high for 9.6Kbps and low for 16Kbps which makes it clear that CELP can work efficiently at low bit rate. On the other hand MELP coder provides much better quality than all older military standards, especially in noisy environments such as battlefield and vehicles and aircraft. The MELP coder has additional features like mixed excitation, a periodic pulses, pulse dispersion and adaptive spectral enhancement as compared to other parametric coder. It helps to remove the annoying artifacts, buzzes and tonal noises. The subjective evaluation of MELP technique suggests that it is also a low bit rate coder and intelligible coder among parametric coder. The literature says that there are other speech coding techniques which can perform well when compared to CELP and MELP. One such coding technique is RELP (Residual Excited Linear Prediction), which directly transmits the residual signal. To achieve the lower bit rates, the residual signal is usually down-sampled. It is used in some text to speech voices, such as diphone databases which are mainly found in the speech synthesizers. The variants of CELP technique such as Algebraic CELP (ACELP), Relaxed CELP (RCELP), Low Delay CELP (LD−CELP) and vector sum excited linear prediction can also be implemented because these are the most widely used speech coding algorithms in MPEG-4 Audio and speech coding.

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