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An Overview: Comparative Analysis between CELP and ACELP Encoder for Wireless Communication

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Abstract- *Speech coding is categorized by way of the process of sinking the bit rate of digital Speech representation for transmission or storage, while preserving a speech quality that is acceptable for the application [2]. Speech coding is an important feature of modern telecommunications. The primary impartial of speech coding is to represent the speech signal with the least number of bits, while keeping a satisfactory level of quality of the retrieved or synthesized speech with reasonable computational complexity [9].To attain great quality speech at a little bit rate, coding algorithms put on sophisticated methods to reduce the redundancies, that means to eliminate the irrelevant information from the speech signal. In addition, a lower bit rate suggests that a lesser bandwidth is required for transmission [10]. Code excited linear prediction and algebraic code excited linear prediction speech coding techniques useful in wireless communication for compression of speech signal to improve the data rate. The CELP coding operated under 8kbps exploited to transmit the minimum amount of speech signal in Codeword with minimum error is formed to synthesis the speech signal.In ACELP coder the speech signal transmit with minimum 4kbps speed so we have to reduce the bit rate more & more compare to CELP coder [1].*

Keywords: *Linear prediction (LP), Algebraic Code Excited Linear Prediction (ACELP), Vector Quantizer, Pitch Analysis,* Weighting Filter.

"I. Introduction"

Speech coding is the process of finding compact representation of voice signals for efficient transmission over band limited wireless channel and/or storage.it is also classified as Speech coding is classified as the process of plummeting the bit rate of digital speech representation for transmission or storage, while maintaining a speech quality that is adequate for the application. Speech coding is an imperative feature of modern telecommunications. Speech coding is the process of digitally demonstrating a speech signal. The primary impartial of speech coding is to represent the speech signal with the least number of bits, while maintaining a adequate level of quality of the retrieved or synthesized speech with equitable computational complexity.In a wired communications very large bandwidth are available as a result of introduction of optical fiber cables, but in a wireless communications bandwidth is a critical issues and service provider are continuously in searching of coder for occupying more users within a limited allocated bandwidth. In rapidly changing wireless communication, user requires high data rates and reduction in bit rate of data means compression in speech signal. Different voice coding techniques are used to compress the speech signal in which code excited linear prediction (CELP) techniques is used in WCDMA technology for compression of speech. The CELP coding operated under 8kbps and it goal is transmit the minimum amount of speech signal in codeword with minimum error is produced to synthesis the speech signal[1].the major application of compression of speech in wireless communication at side of encoder to transmit the speech signal in low bit rate. It permits longer message into speech code, and it also allows to user share the same bandwidth [3].

Codebook excited linear prediction (CELP) was announced by B.S. Anal and M.A. Schroeder at the 1984[9].To increases the data rates, algebraic code excited linear prediction techniques introduced in wireless communication. In ACELP coder the speech signal transmit with minimum 4kbps speed to shrinks the bit rate compare to other coder. The MATLAB tool is used to to study and verify various aspects of the algorithm such as: the LP analysis, the open-loop pitch search, the adaptive codebook search (pitch search), the fixed codebook search, and the bit sharing patterns. We choose MATLAB as the implementation platform because it permits the user to easily understand the complex parts of the algorithm whose function is not a Major [1].

II. Performance Criteria of Speech Coding

Speech coders attempt to minimize the bit rate for transmission or storage of the signal while retaining prerequisite levels of speech quality, communication delay, and complexity of implementation (power consumption). The parameters of performance, with reference to speech are categorized as follows:

Speech Quality

Speech quality is typically assessed on a five-point scale, known as the mean-opinion score (MOS), in speech quality analysis---an average over a bulky number of speech information, speakers, and listeners. The five points of quality are: wicked (bad), poor, fair, good, and brilliant (Excellent). Quality scores of 3.5 or higher generally denote high levels of intelligibility, speaker recognition and naturalness.

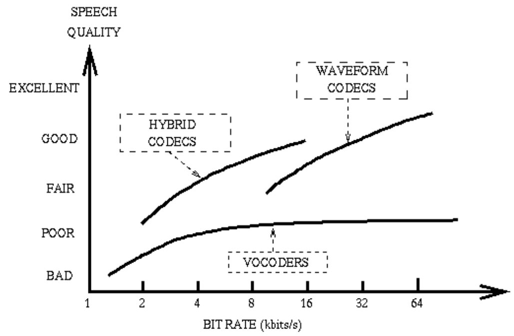

Fig. [1], the speech quality mean opinion score for various bit rates [10]

Bandwidth

Bandwidth of the speech signal that desires to be encoded is also an problem. Typical telephony necessitates 200–3400 Hz bandwidth. Wideband speech coding techniques (valuable intended for audio broadcast, tele-conferencing and tele-teaching) entail 7–20 kHz bandwidth.

Transmission Bit Rate

Since the speech codec segments the communications channel by other data, the peak bit rate should be as low as conceivable therefore not to use a disproportion ate segments of the channel. Codec below 64 kbps were established to growth the capacity of equipment used for narrow bandwidth links.

Communication Delay

Speech coders often process speech in blocks and such processing introduces communication delay. Depending on the application, the allowable total delay could be as low as 1 msec, as in network telephony, or as great as 500 msec, as in video telephony. Communication delay is immaterial for one-way communication, such as in voice mail.

Complexity

The complication of a coding algorithm is the processing strength compulsory to code the algorithm, and it is typically restrained in terms of arithmetic capability and memory prerequisite, or equivalently in terms of price. A huge complexity can outcome in high power consumption in the hardware [10].

III. COMPARISON BETWEEN CELP AND ACELP CODER

This coder is augmented by by means of a code book (look up table) to treasure the best contest for the Signal. This method reduces the processing complexity and the required data transmission rate. Fig-[2] shows the block diagram of CELP encoder. the stochastic codebook and adaptive codebook generate array of bit patterns at its output multiply into multiplier, In which linear predictive coder analyzer and 10^{th} order linear predictive coder synthesizer which evaluates and synthesis the speech signals. LPC analysis (order 10^{th}) is recycled to subtracting the vocal tract component from speech signal. The pitch search investigates the inaccurate speech signal. It is perceptually weighted by weighting filter and then compared to all the structures in the pitch codebook. Low Delay CELP (LD-CELP) and Algebraic CELP (ACELP) are normally utilized in internet voice calls and cell phones [11].

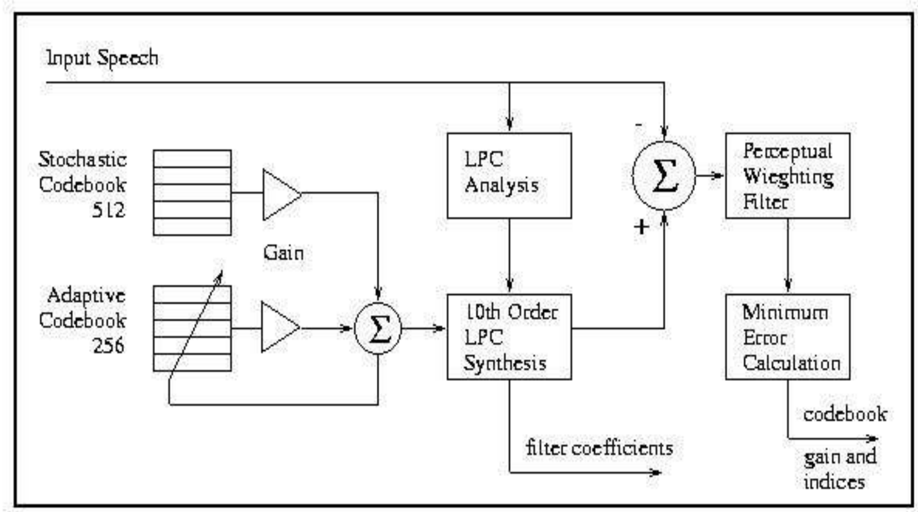

Fig-[2], Block diagram of CELP encoder [12]

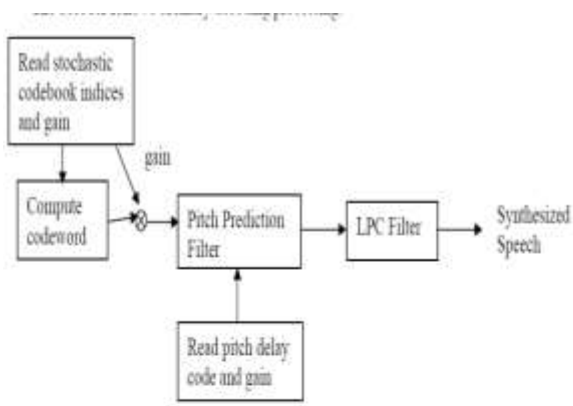

Fig-[3], Block diagram of CELP decoder [12]

Fig-[3], depicts the block diagram of CELP decoder, in which read out the stochastic indices and calculate the speech Codeword, pitch prediction filter weight the fault of speech and reconstruct the speech.

ACELP ENCODER

It has Short algorithmic delay, and correspondingly requiring Low bandwidth toll-quality coder aimed at concealment of detected frame erasures. It reduces the channel error [7].The Algebraic CELP suggests the structure of the codebook used towards selecting the excitation codebook vector. The speech signal is analyzed intended for speech frames of 10ms corresponding to 80 samples on a sampling rate of 8000 samples per second. Fig-[4], shows Block diagram of ACELP encoder [5]. The five important stages accompanying with the encoding principle of CS-ACELP includes: pre-processing stage, LP examination stage, open-loop pitch search, closed-loop pitch search, and algebraic codebook search [1].

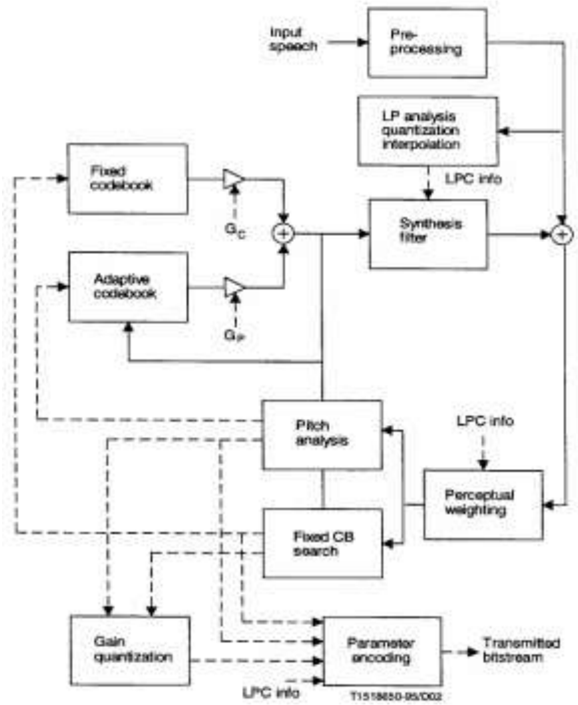

Fig-[4], Block diagram of ACELP encoder [12]

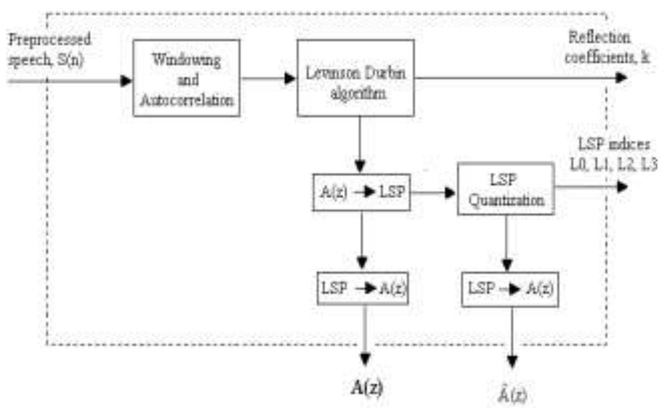

Fig.-[5] LP ANALYSIS[12]

Fig-[5], depicts The LP analysis where the speech signal S(n) applied to LP analysis block which consist of window and autocorrelation block, it is activated on 120 samples since the past speech frames and 80 samples from present speech frames and 40 samples from the feature frame. LP coefficient derived from autocorrelation coefficient from window Speech by use of the Levin ion Durbin algorithm. The vector quantizer utilized to gather the LSP coefficient which condensed the weighted mean square error. Finally output of LP analysis is reflection coefficient, K and LSP indices L0, L1, L2, L3 [9].Fig. - [6] depicts open loop and closed loop Pitch Analysis [1]. From Levinson-Durbin algorithm the reflection coefficients used to calculate the adaptive Weight elements. Where, S (n) is the pre-processed speech, $\gamma 1$ and $\gamma 2$ are the adaptive weights, and ai, i = 1, 2. . . 10 are the un quantized LP coefficients. The auto correlation of weighted speech signal sw(n) is created and it passes from the open loop pitch delay stage[1].

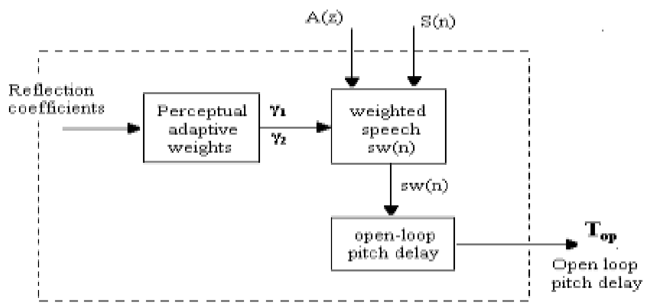

Fig-[6] OPEN LOOP AND CLOSED LOOP PITCH ANALYSIS [1]

The ACELP decoder shown in Fig-[7], from the received bit stream parameter indices are extracted .These parameters are LSP coefficient. The LSP coefficient converted into LP coefficient for each subframe.the output of fixed codebook vector and adaptive codebook vector are accumulated with fixed gain .then speech signal is reconstructed by LP filtering and then it passed from post filtering stage [8].

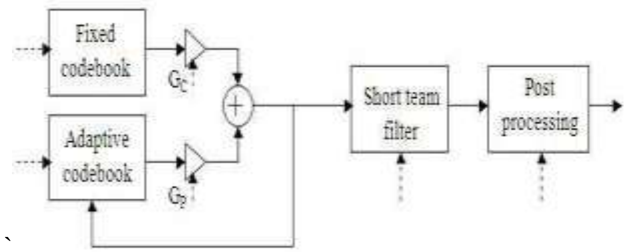

Fig-[7] ACELP DECODER[1]

IV. Conclusion

From comparative review of CELP and ACELP coder one can analyses that reduces the bit rate of compressed speech Signal for transmission and reception and it also provide good quality of voice output. By reducing the data rate one can take advantage in form of storage, improve the coder efficiency and reduce the bit error.

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