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COMPARATIVE REVIEW BETWEEN CELP AND ACELP ENCODER FOR CDMA TECHNOLOGY

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Abstract: This review paper presents for analysis between code excited linear prediction and algebraic code excited linear prediction speech coding techniques which is useful in wireless communication for compression of speech signal to improve the data rate. from this paper we understood CELP encoder, decoder & ACELP encoder, and decoder & differentiate both the coding technology. How it used in WCDMA wireless communication system, the major factor in speech coding is bit rate which is reduced in ACELP coder up to 4.6kbps.Compare to CELP coder. In many applications for transmission of voice signal it conversion is needed for that speech coder is used.

Keywords: CELP (code excited linear prediction) and ACELP (algebraic code excited linear prediction)

I. INTRODUCTION

This paper presents comparative review between code excited linear prediction and algebraic code excited linear prediction speech coding techniques [1].

The growth of wireless technology user required high data rates and increases to transmission speed of data; we have to reduces the bit rate of data for that it is necessary to compress to speech. Different voice coding techniques are used to compress the speech signal in which code excited linear prediction techniques is used in WCDMA technology for compression of speech .In mobile technology bandwidth and conversion quality of speech signal is most argued parameter in any telephony system which generates digital representation 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[2].the major application of compression of speech in mobile communication at side of encoder to transmit the speech signal in low bit rate. It allows longer message into speech code, and it also allows to user share the same bandwidth.[3]. Codebook excited linear prediction (CELP) was introduced by B.S. Anal and M.A. Schroeder at the 1984[4].To increases the speed of data transmission and data rates one more coding technology introduced in wireless communication which is algebraic code excited linear prediction techniques. In ACELP the speech signal transmit with minimum 4kbps speed so we have to reduces the bit rate more & more compare to other coding technology, the MATLAB tool is used to design ACELP algorithm, The tool is user-friendly and graphical user interface (GUI) that allows the student to study and verify through graphics the 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 allocation patterns. We choose MATLAB as the implementation platform because it allows the user to easily understand the complex parts of the algorithm whose function is not a Major [5].

By the end of introduction section include the paper organization. This paper is organized as follow: Section I gives the introduction comparative review between code excited linear prediction and algebraic code excited linear prediction speech coding techniques. Section II is helpful to Characteristics of speech. Section III explains comparison of CELP and ACELP section V concludes the paper and followed by the references.

II. Characteristics of speech: Speech energy vs. frequency

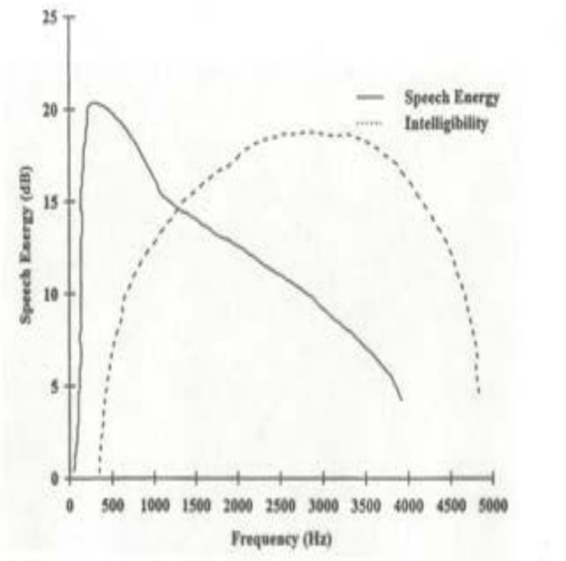
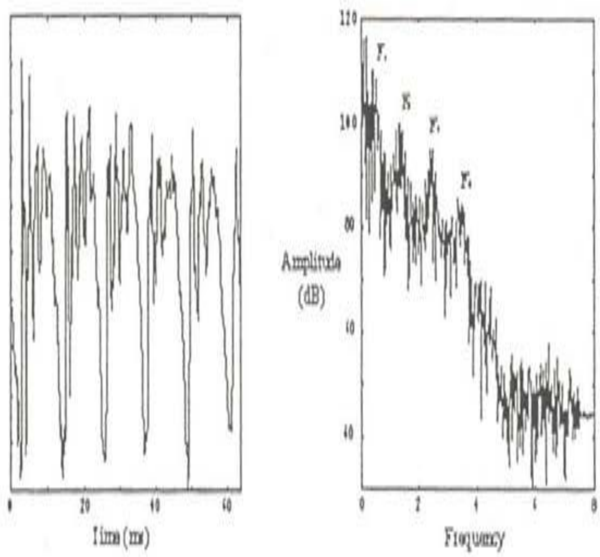


Fig-[1]

Voiced signal

Voiced signal shown in fig[2], it shows the relatiionship between amplitude of speech and varying frequncy,and same fig[2] shows the relationship between amplitude of speech and varying time



Fig[2]

Unvoiced signal

Unvoiced signal shown in fig[3], it shows the relatiionship between amplitude of speech and varying frequncy,and same fig[3] shows the relationship between amplitude of speech and varying time.

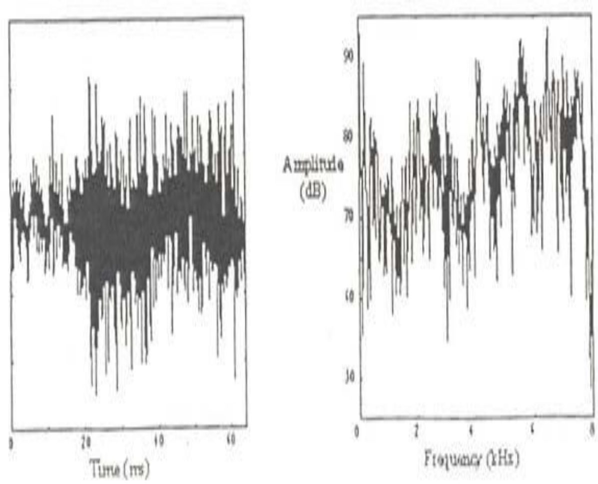


Fig-[3]

III. COMPARISON BETWEEN CELP AND ACELP CODER

This coder is optimized by using a code book (look up table) to find the best match for the Signal. This method reduces the processing complexity and the required data transmission rate. In Fig-[4] shows the block diagram of CELP encoder in which shows the stochastic codebook and adaptive codebook means that generate array of bit patterns its output multiply into multiplier, In which linear predictive coder analyser and 10^{th} order linear predictive coder synthesizer which analyses and synthesis the speech signals. LPC analysis (order 10^{th}) is used to subtracting the vocal tract component from speech signal. The pitch search analyzes the error speech signal. It is perceptually weighted by weighting filter and then compared to all the sequences in the pitch codebook. Low Delay CELP (LD-CELP) and Algebraic CELP (ACELP) are generally used in internet voice calls and cell phones. The CELP coder does not directly need an analysis stage [6].

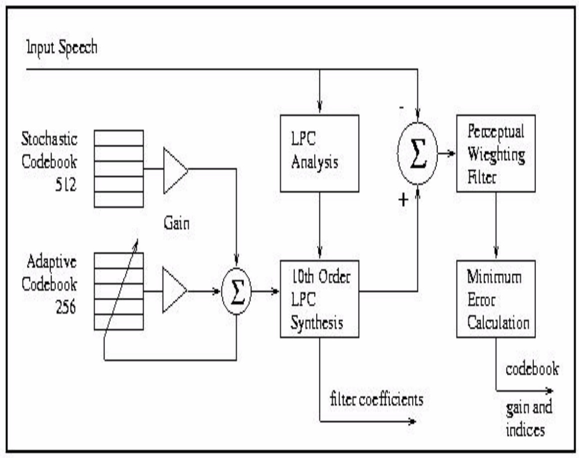


Fig-[4], Block diagram of CELP encoder [7]

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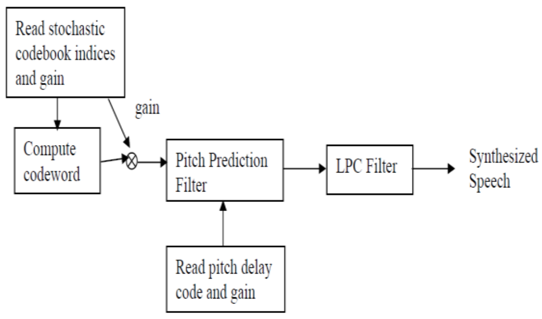

Fig-[5], Block diagram of CELP encoder [7]

In Fig shows the block diagram of CELP decoder, in which read out the stohastic indices and compute the speech codeword,pitch prediction filter weight the error of speech and reconstruct the speech.

ACELP ENCODER

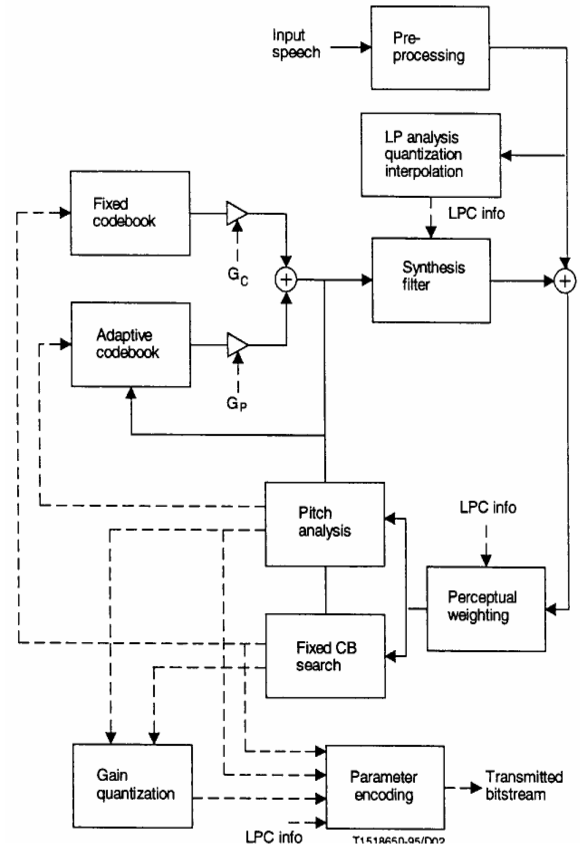

Fig-[6], Block diagram of ACELP encoder [8]

It has Short algorithmic delay, and also having Low bandwidth toll-quality coder, it has Provision for concealment of detected frame erasures. It reduce the channel error [9].The name Algebraic CELP implies the structure of the codebook used to select the excitation codebook vector [10]. The speech signal is analyzed for speech frames of 10ms corresponding to 80 samples at a sampling rate of 8000 samples per second. The five important stages associated with the encoding principle of CS-ACELP include: the pre-processing stage, the LP analysis stage, the open-loop pitch search, the closed-loop pitch search, and the algebraic codebook search.[11].

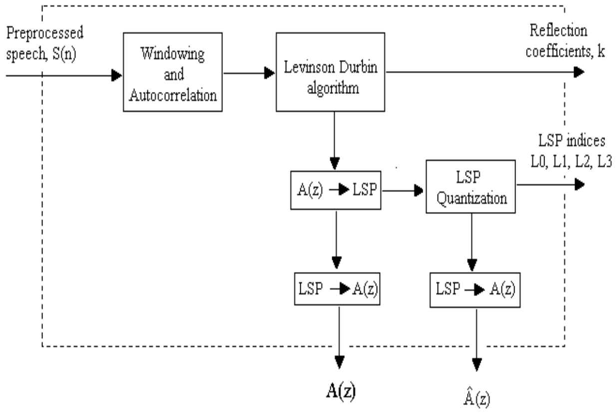

Fig-[6] LP ANALYSIS

The LP analysis shown in fig, the speech signal S(n) applied to LP analysis block which consist of window and autocorrelation block, it is operated on 120 sample from the past speech frames and 80 sample from present speech frames and 40 sample from the feature frame. LP coefficient derived from autocorrelation coefficient from window speech by using the Levin ion Durbin algorithm. the vector quantizer used to produce the LSP coefficient which is reduced the weighted mean square error. Finally output of LP analysis is reflection coefficient, K and LSP indices L0,L1,L2,L3 [12].

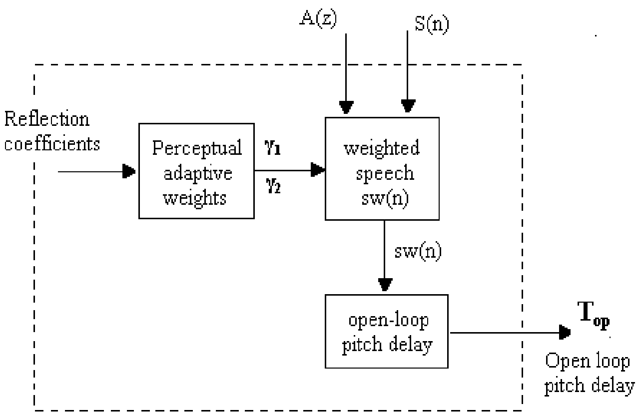

Fig-[7] OPEN LOOP AND CLOSED LOOP PITCH ANALYSIS

From Levinson-Durbin algorithm the reflection coefficients used to compute the adaptive Weight factors. Where, S (n) is the pre-processed speech, γ1 and γ2 are the adaptive weights, and ai,i = 1, 2, . . . , 10 are the unquantized LP Coefficients. the auto correlation of weighted speech signal sw(n) is produced and it passes from the open loop pitch delay stage[12].

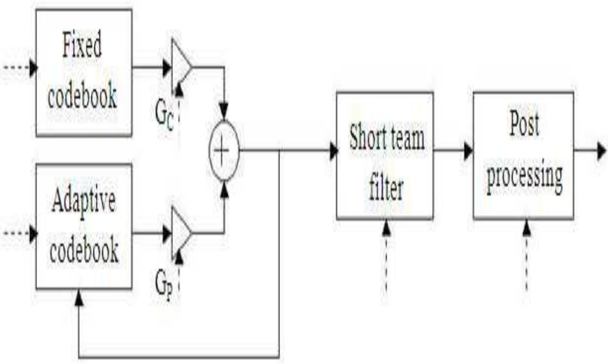

Fig-[8] ACELP DECODER

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The ACELP decoder shown in fig, 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 adding with fixed gain .then speech signal is reconstructed by LP filtering and then it passed from post filtering stage [13].

IV. CONCLUSION

From comparative review of CELP and ACELP coder analyses that thing it reduces the bit rate of compressed speech signal for transmission and reception and it also provide good quality of voice output. From this paper also get the knowledge about how to improve the coder efficiency and reduce the bit error.

REFERENCES

[1]. Venkatraman Atti and Andreas Spanias, Department of Electrical Engineering, MIDL – TRCArizona State University, Tempe, AZ, 85287-7206
[2]. Shannon Wichman Department of Electrical Engineering The University of Texas at Dallas
[3]. A. R. Sahab*1, M. Khoshroo*Islamic Azad University Lahijan Branchr
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[5]. Venkatraman Atti and Andreas Spanias Department of Electrical Engineering, MIDL – TRCArizona State University, Tempe, AZ, 85287-7206, U.S.A
[7].Joseph p. cambell,JR,vanoy C,welch,and thomas E.Tremain, "An Expandaable Error protected 4800bps CELP coder (U.S. Federal standard 4800 bps Voice coder),U.S.DoD.
[6]. INRS- Te'lPcommunications3 Place du Commerce Ile des Soeurs, Que. CANADA H3E 1H6
[7].Joseph p. cambell,JR,vanoy C,welch,and thomas E.Tremain, "An Expandaable Error protected 4800bps CELP coder (U.S. Federal standard 4800 bps Voice coder),U.S.DoD.
[8]. ITU-T G.729/G.729ACS-ACELP 8kbps Speech Coder by:Lior Shadhan
[9].ITU-T G.729/G.729A CS-ACELP 8kbps Speech Coder by: Lior Shadhan
[10]. Venkatraman Atti and Andreas Spanias Department of Electrical Engineering, MIDL – TRC
[11]. Venkatraman Atti and Andreas Spanias Department of Electrical Engineering, MIDL – TRC
[12].Venkatraman Atti and Andreas Spanias Department of Electrical Engineering, MIDL – TRC Arizona State University,Tempe, AZ, 85287-7206, U.S.A
[13]. Thai Speech Coding Based On Conjugate-Structure Algebraic Code Excited Linear Prediction Algorithm Suphattharachai Chomphan Department of Electrical Engineering, Faculty of Engineering at Si Racha, Kasetsart University, 199 M.6, Tungsukhla, Si Racha, Chonburi, 20230, Thailand.

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