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Packets Loss Concealment Method to Improve Frame Loss Recovery by Using RLS Algorithm

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***Abstract*— Most efficient frame loss recovery methods, particularly for Algebraic Code Excited Linear Prediction (ACELP) codec in Packets Loss Concealment techniques, are based on excitation reconstruction. Therefore, this topic is the key of research areas for robustness improvement in Voice over Internet Protocol (VoIP). Hence, many studies approve that the adaptive codebook excitation in ACELP decoder gives better quality to synthesis speech. Meanwhile, selecting a wrong excitation is the main source of error propagation, especially in voiced signal and mostly when the frame is lost or delayed. In this paper, we propose an original Packets Loss Concealment (PLC) method which uses the Recursive Least Square (RLS) algorithm to reconstruct the excitation shape at the decoder when a frame is lost. The proposed method aims to avoid the error propagation caused by the bad estimation of the introduced PLC excitation. The basic principle of our method is to appeal to adaptive filtering by using RLS Algorithm to fit with the track of changes in the shape of the glottal pulse to replace the non availability of the adaptive-codebook excitation effect and to be a halfway between the past fixed-codebook excitation and the improved one, offered by the RLS filter, adjusted previously when the frames are well received.**

***Keywords*— Speech codec, Error correction, PLC, Glottal Pulse Shape, Adaptive CodeBook, Fixed CodeBook, RLS Algorithm.**

I. Introduction

Recent years have seen active research in the area of voice over IP (VoIP) or Internet telephony since packet networks can provide more robustness for speech communication systems [1]-[2]. But loss of Packets in network stills one of the threats for Quality of Service (QoS) for speech transmission [3]. For example, transmission of voice over IP suffers from packet loss caused by delivery problems due to network nodes and/or delay introduced by the congestion of the packets in the network. Hence, the loss of one packet induces the loss of a segment of speech signal, typically "10ms" to "30ms" of speech [4]. In this study, we propose a packet loss concealment method based on ACELP and MultiPulse Maximum Likelihood Quantization (MP-MLQ) coder, standardized as G732.1 [5]. However, this packet loss concealment is applied at the selected excitation sequences level, in the goal to improve the bad estimated excitation introduced by the embedded packet concealment technique.

In loss of packet for ACELP codec, the non availability of excitation indices makes the reconstruction of speech at the decoder very vulnerable to error propagation. In spite of, the excitation is a mixture of two excitations, Fixed Codebook (FCB) and Adaptive Codebook (ACB), but the ACB excitation plays an important role as it was revealed in [6, 7 and 8]. In our method, we appeal to RLS filter to memorize the changes and differences between the mixture of two selected excitations of ACB and FCB, and the FCB excitation considered alone, through the well received frames. Hence, when a frame is considered as lost, the previously adjusted filter will be used especially to replace the absent effect of the Adaptive Codebook excitation. In our approach, the choice of RLS filter is based on its known features for convergence according to his neighbour adaptive Filters [9], and the previous conducted results through this study.

This paper is organized as follows. In section 2, we briefly present the adaptive equalization, focusing on the applied RLS algorithm [9]. The constrained conditions of the filter parameters are defined and the principle of the proposed PLC method is presented. Where the relevant initialization parameters are discussed according the obtained test results. Subsequently, we discuss the effectiveness of our proposed method and modified decoder scheme. Obtained relevant results are shown and comparison to the original PLC method implemented in G723.1 codec [5] is given. Finally, a conclusion of this work is presented in section 4.

II. Adaptive Equalization

Owing to the powerful digital signal processors and the development of advanced adaptive algorithms there are a great number of different applications in which adaptive filters are used. The number of different applications in which adaptive techniques are being successfully used has increased enormously during the last two decades. There is a wide variety of configurations that could be applied in different fields such as telecommunications, radar, sonar, video and audio signal processing and noise reduction. The efficiency of the adaptive filters mainly depends on the design technique used and the algorithm of adaptation. The adaptive filters can be analogical, digital or mixed designs, depending on their advantages and disadvantages. For example, the analogical filters are low power consuming and fast response, but they present offset problems, which affect the operation of the adaptation algorithm [9]-[10].

The simplest adaptive filters or equalizers are constructed from transverse filter whose coefficients are usually discounted from a gradient algorithm or Recursive Least Squares. To promote the convergence of an equalizer, we usually use a training sequence which is a preamble consisting of known data to the receiver. When this task is impossible, we are forced to use autodidact equalizers which base their treatment on a prior knowledge of the statistical properties of signals [9].

A. RLS Algorithm

We define E as the vector of all weights E_i and u as the vector of all inputs u_i. Based on the current set of inputs, u, and the current inverse correlation matrix, P, this adaptive algorithm first computes the Kalman gain vector K, given by

$$K = \frac{P.u}{\lambda + u^H.P.u} \qquad (1)$$

Where, H, denotes the Hermitian transpose, and is the forget factor. The new inverse correlation matrix is given by

$$\lambda^{-1}(P - K.u^H.P) \qquad (2)$$

And the new set of weights is

$$E + K^*.e \qquad (3)$$

Where the * operator denotes the complex conjugate and, e, the mean square error to be minimized.

B. Principle of the Proposed Method

Recent studies were conducted on the introduced errors by Packet Loss Concealment techniques and revealed three different types of errors that involve significant error propagation [6]. The errors are classified as "*A*", "*B*", and "*C*" types.

An interesting block diagram of a modified encoder was presented in [6], to correct the error types "*A*", and "*B* "by use of a side information bit stream delayed and sent in the next frame. Unfortunately, the "*C*" error type is not corrected. So our work is focused on this problematic and the objective of the proposed method is to correct the "*C*" error type [6].

In the proposed scheme, the decoder will be complemented by an adaptive RLS algorithm, such as is shown in Fig. 1.
Hence, our approach is to reconstruct the features of the excitation sequence poorly estimated by the G723.1 [5], using a packet loss concealment method. Thus, we modelled the problem by an adaptive filtering system, which changes the "bad" excitation given by the original PLC method by an estimated one based on the past excitations of ACB mixed to FCB and FCB excitation considered alone. The entries of our adaptive filter are used to adjust its coefficients of earlier well-received frames.

Furthermore, we propose to adjust the equalizer coefficients by the excitations of the voiced speech segments. In its turn, the excitation of previous received frames serves as training sequences for our adaptive filter. However, once the frame is not received, the equalizer fitted previously will serve as a witness of changes through the lost frame.

Subsequently, when a packet is considered as lost, the adjusted RLS filter serves to generate an estimated excitation based on the past fixed codebook excitation filtered by the adjusted filter. Likewise, the next excitations for the received voiced frames will be filtered by the previous adjusted RLS filter, to introduce the glottal pulse characteristics and avoid error propagation.

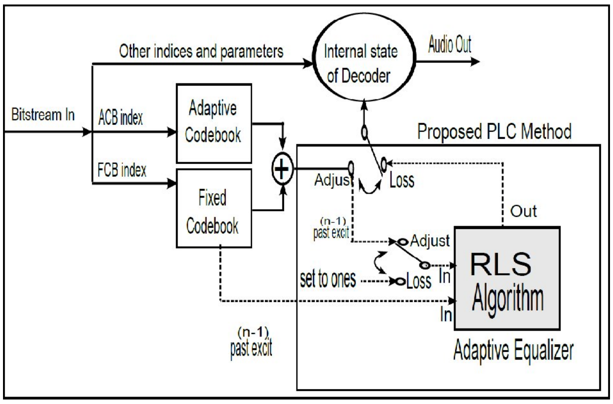

Fig.1. Scheme of the proposed PLC method of the modified ACELP decoder.

C. RLS Parameters Initialization

Parameters initialization is based on effected test results which are shown in Table (1), for both, filter length and the forgetting factor with respect to the different loss rates.

Our choice is made on a Finite Impulse Response (FIR) filter which is based on Recursive Least Squares Algorithm, of length "*L = 12*". The choice of the filter length is relied to the realized tests by affecting various lengths to the RLS filter, and computing the Perceptual Evaluation of Speech Quality

(PESQ) values for different Loss Rates, as it is shown in Table (1). The results show that the length of "*12 coefficients*" is an optimal value as higher values of filter length can cause a considerable delay. Hence, we selected "*L=12*" for our filter to estimate the excitation with the proposed PLC method for most of loss rates. Our approach is based on having a memory of an adaptive filter characterizing the traces of glottal pulse shape, to identify the difference between the mixed FCB and ACB excitations and the FCB excitation alone and to record glottal pulse of the previous frame, as it is shown in fig.1. The second parameter to be initialized is the forget factor, which has a variation from "0 to 1". Hence, by affecting different values to the forget factor, many tests were conducted to get the optimal value of the PESQ score corresponding to that of Table 1. For these tests, we were fixed the filter length to "*L=12*".

TABLE I: TEST RESULTS OF PERCEPTUAL SPEECH QUALITY (PESQ) FOR VARIOUS LENGTHS OF RLS FILTER AND DIFFERENT LOSS RATES.

PESQ / RLS coefficients	20% of Loss Rate			
	L=10	L=12	L=13	L=16
PESQ	1.549	1.533	1.516	1.452
PESQ / RLS coefficients	**10% of Loss Rate**			
	L=10	L=12	L=13	L=16
PESQ	1.863	1.915	1.579	1.588
PESQ / RLS coefficients	**6.5% of Loss Rate**			
	L=10	L=12	L=13	L=16
PESQ	1.235	1.244	1.229	1.228
PESQ / RLS coefficients	**5% of Loss Rate**			
	L=10	L=12	L=13	L=16
PESQ	1.570	1.244	1.566	1.566

The obtained PESQ values are shown in table (2). Meanwhile, this table shows that the optimal values of the forget factor are: λ=0.82, and λ =0.83; thus for most of loss rates 20%, 10%, 6.5%, and 5%. For filter parameters initialization, we preferred," λ=0.83", because this value gives a slightly better PESQ values according to" λ=0.82". Thus all the tests were conducted using these two initialised parameters of RLS Algorithm ("*L=12*"," λ=0.83") to avoid arbitrary initialisation and to find a good trade-off between the two important parameters.

TABLE II: PESQ RESULTS IN FUNCTION OF FOR DIFFERENT LOSS RATES

Loss Rates	PESQ values in function of "λ" and Loss Rates				
	λ= 0.80	λ= 0.81	λ= 0.82	λ= 0.83	λ= 0.84
20%	1.485	1.499	1.518	1.533	1.543
10%	1.657	1.620	1 .930	1.915	1.527
6.5%	1.290	1.274	1.257	1.244	1.232
5%	1.807	1.790	1 .773	1.787	1.572

III. RESULTS AND DISCUSSION

In this section, we compare the performance of the proposed PLC method with that of the embedded method in the G723.1 standard [5]. We use the PESQ software as witness to judge the performance and robustness of the proposed method.

As an example, fig.2, shows five successive frames where the forth is lost, where Fig.3 represents the same frames where the forth one is recovered by use of the PLC embedded method in the G723.1 codec, and Fig.4 shows the improvement obtained using our PLC method.

Where, Fig.5 summarizes the PESQ values obtained in different cases: without loss, using G723.1 PLC method and employing our method. We note that the proposed method gives good PESQ score values. An improvement of 10% is observed, according to the embedded PLC technique of the G723.1 standard [5]. We noticed also that this improvement can be kept for the different loss rates (20%, 10%, 6.5% and 5%). Nevertheless, the performance depends on the next coming frames, if they are voiced or unvoiced.

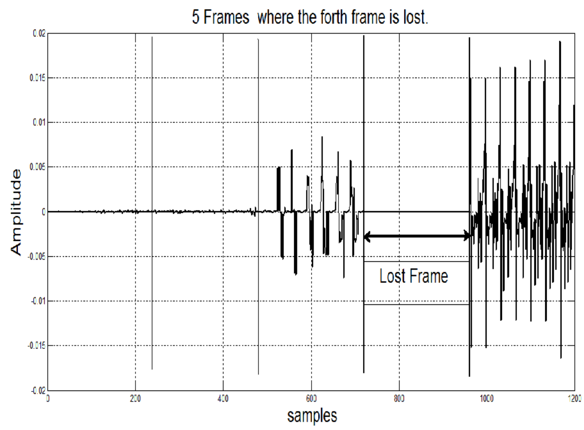

Fig.2 Lost of the forth frame excitation.

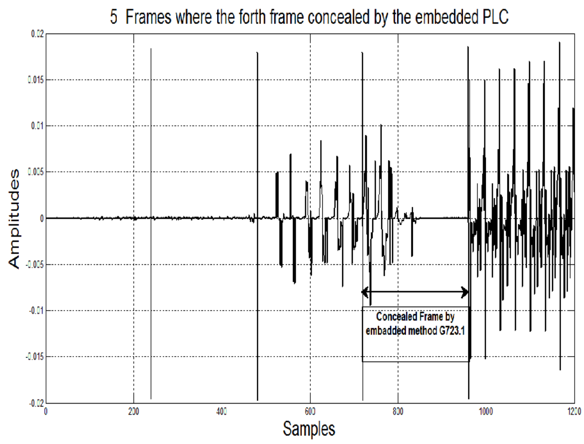

Fig.3 Concealed frame excitation by embedded G723.1PLC method.

Indeed, the realized tests showed that we achieve good results when the next frames are classified as voiced, thereby, the effect of the glottal pulse takes place in the frame excitations.

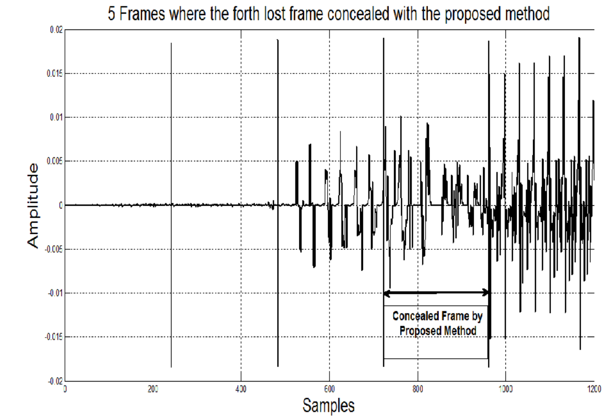

Fig.4 Concealed frame excitation by the proposed PLC method.

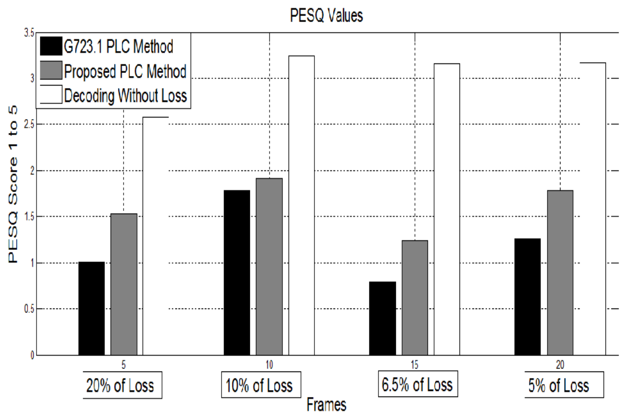

Fig. 5 PESQ values in function of loss rates.

IV. CONCLUSION

In this paper, we propose a new receiver-based packet loss concealment algorithm for a CELP-Multi-Pulse type speech coder to improve the embedded concealment method. The method is based on RLS Algorithm which counteracts speech quality decrease. Firstly, initializations of RLS parameters are set after multiple tests. Indeed, the weight values of the RLS algorithm are adjusted previously to record a trace of the glottal excitation presents between the mixed ACB and FCB excitations and the FCB excitation considered alone.

Secondly, when the frame is considered as lost, the RLS coefficients filter are used to filter the last FCB excitation only to generate an excitation for the lost frame. They are likewise used to filter the next voiced excitations corresponding to the received frames, in the aim to improve the perceptual quality of the decoded received frames and avoid the error propagation. According to the objective and subjective test results, the proposed method could achieve better performance than the embedded G723.1 one. The proposed algorithm could be extended to other CELP based speech coders as well.

Finally, From PESQ measurement tests under a variety of frame erasure conditions, we found that the proposed method can reach of about 10% of improvement for the synthesis speech quality compared to the embedded algorithm encountered in the standard G723.1 coder [5].

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