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Improved Frame Loss Recovery Using RLS for PLC Excitation Shape Reconstruction

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***Abstract*— The effectiveness of the ACELP Multi-Pulse codec is to use the good excitation in the adaptive codebook to reconstruct the synthesis speech signal at the decoder when a frame is properly received. But apparently selecting a wrong excitation is the main source of error propagation in voiced signal, and mostly when the frame is lost or delayed, hence results a constantly upward error. The proposed method aims to remedy to the error propagation caused by the bad estimation of the introduced Packets Loss Concealment excitation. The basic principal 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 and to be a halfway between the bad excitation given by the PLC technique and the corrected one offered by the Recursive Least Squares Algorithm.**

***Keywords*—Speech codec, Errors correction, PLC, Glottal Pulse Shape, ACB excitation, FCB excitation, RLS Algorithm.**

I. INTRODUCTION

Digital transmissions of given networks are increasingly evolving, but the task seems to be a little more complicated towards the robustness side. 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. The loss of a voice packet induces the loss of a segment of speech signal (typically 10 to 30 ms of lost packet) [1]. On the given wide variety of speech coders, we were interested in this study, to propose a packet loss concealment method, which dependents on the used speech coder. Thus, this packet loss concealment is applied at the selected excitation sequences level before being improved or modified by the relevant parameters (redundancy), added as a side information bit stream, at the encoder, in the goal to correct the bad excitation introduced by the concealment technique (PLC). The proposed system is based on an adaptive equalizer to eliminate an error of type C [1]. To monitor the change in the shape of the glottal pulse, some studies have been proposed, among them the presented one, as a perspective in [2]. In our approach, we propose an adaptive filter RLS (Recursive Least Squares) which converges faster than its known neighbours adaptive Filters [4].

This paper is organized as follows. In section 2, we briefly present the adaptive equalization, focusing on the applied RLS algorithm [4], where the constrained conditions of the filter parameters are defined, and the principal of the proposed PLC method is presented with some relevant initialization parameters are discussed according the realized test results. Subsequently, we discussed the effectiveness of our proposed method and modified decoder scheme, where obtained relevant results are shown, with comparison to the original PLC method implemented in G723.1 codec [2].

Finally, a conclusion of this work are presented in section 4.

II. ADAPTIVE EQUALIZATION

The simplest adaptive equalizers are constructed from transverse filters whose coefficients are usually discounted from a gradient algorithm or, Recursive Least Squares. To promote the convergence of an equalizer, usually we use a training sequence, it means, 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 [4].

A. *RLS Algorithm*

We define E as the vector of all weights E_i and u as the vector of all input 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 (1).

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

Where H denotes the *Hermitian* transpose, and λ is the forget factor. The new inverse correlation matrix is given by the equation (2),

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

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 minimised.

B. *Principal of The Proposed Method*

A previous study was conducted on the introduced errors by the Packet Loss Concealment and revealed three different types of errors that involve significant error propagation [3, 7], which are classified type A, B and type C. And a block diagram of modified encoder was presented to correct these errors, where a side information bit stream delayed and sent in the next frame, in the aim to correct the errors type A, and B, according to [3], but the error type C is not corrected.

So our work is focused on this problematic and the objective of our proposed method is to correct the error type C [3]. In the proposed scheme, the decoder will be complemented by an adaptive algorithm (RLS). Our approach is to reconstruct the features of the excitation sequence poorly estimated by the G723.1 [2], packets 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 added to FCB and FCB excitation only, whose make entries of our adaptive filter to adjust its coefficients of well-received frames earlier.

Furthermore, we propose to adjust the equalizer coefficients by the excitations of the voiced speech segments, such as is shown in Fig. 1. In its turn serve as training sequences for our equalizer algorithm. However, once the frame is not received, the equalizer fitted previously will serve as a witness of changes through the lost frame.

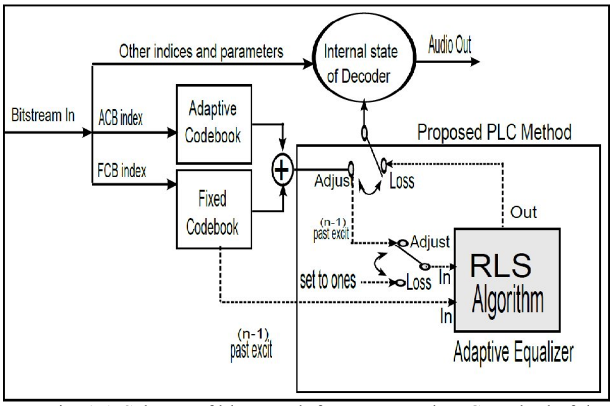

Fig. 1 A Scheme of bloc graph for a proposed PLC method of the modified decoder.

Subsequently, when a packet is considered as lost, the adjusted RLS filter serves to generate an estimated excitation based on the last 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 characteristic and avoid error propagation.

C. *RLS Parameters Initialisation*

Our choice is made on a FIR filter which is based on an RLS Algorithm, Recursive Least Squares, of length L = 12, the choice of the filter length is relied to the realized tests by affecting various lengths to the RLS filter, as it is shown in Table (I).

TABLE I

PESQ/RLS coefficients	Loss rate 20%			
	L=10	L=12	L=13	L=16
PESQ	1.549	1.533	1.516	1.452
PESQ/RLS coefficients	Loss rate 10%			
	L=10	L=12	L=13	L=16
PESQ	1.863	1.915	1.579	1.588

The results show that the length of 12 coefficients is optimal, for decoding with proposed PLC method for both, 20% and 10% of loss rate. 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 only, and to record glottal pulse of the previous frame, so the scheme of the proposed method, is shown in Fig. 1.

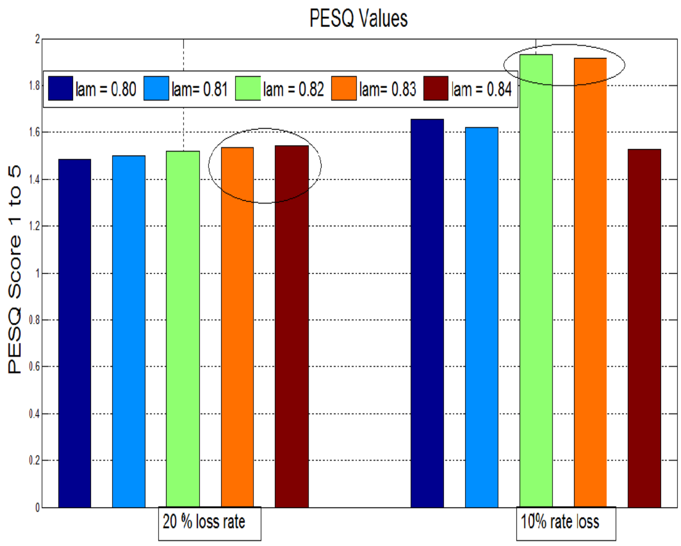

Fig. 2 figure of PESQ values in function of loss rate, for 10% and 20% of loss, and also of forget factor values (lamda).

The second parameter to be initialised is the forget factor λ, so many test were effected, where the optimal values of PESQ score and of forget factor were selected. Meanwhile, the Fig. 2 shows that the optimal value for both loss rates 20% and 10% is λ =0.83, so all the tests are conducted on these two initialised parameters of RLS Algorithm (L=12, λ =0.83), in the aim to avoid arbitrary initialisation and to find a good trade off between the two important parameters .

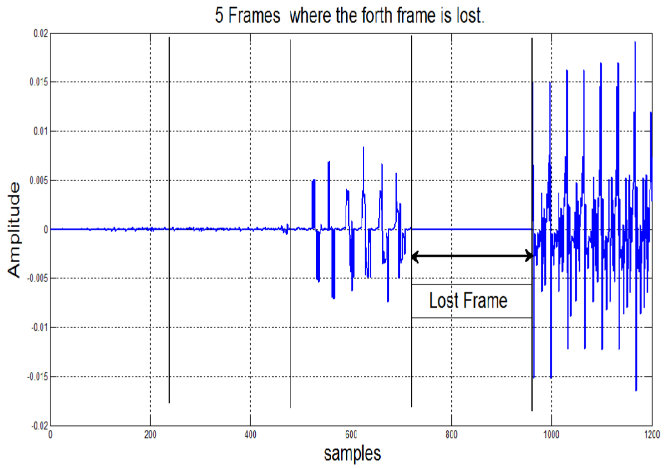

Fig.3 Lost forth frame excitation.

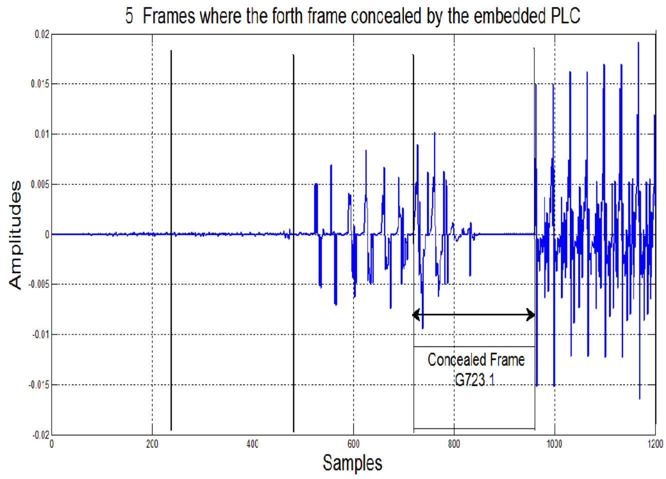

Fig.4 Concealed frame excitation with the embedded G723.1PLC method.

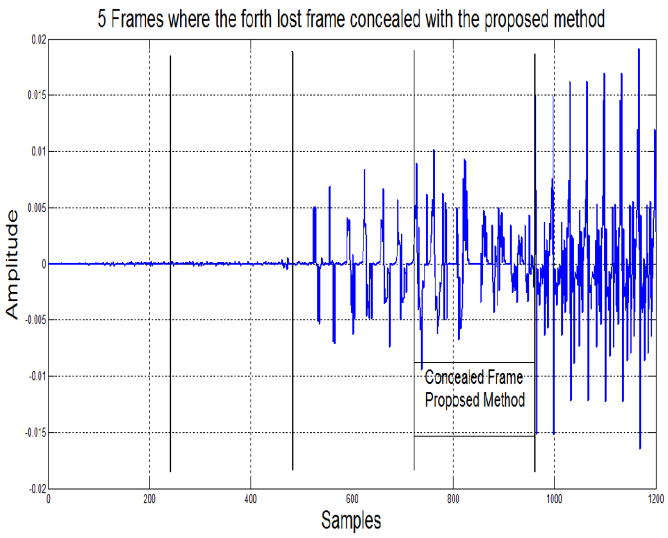

Fig.5 Concealed frame excitation with the PLC proposed method.

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 [2]. We use the PESQ as witness to judge the performance and robustness. The Fig.3 shows that the proposed method, gives a good PESQ values, for an improved difference of 10% according to the embedded PLC technique of the G723.1 standard [2]. Likewise, the figures 3, 4 and 5 give an overview of the three cases. We noticed that this improvement can be kept for the different loss rates (20%, 10%, 6.5% and 5% of loss rate).

Nevertheless, the performance depends on the following frames, if they are voiced or unvoiced. The realized tests showed that we achieve good results when the next frames are classified as voiced, thereby, the glottal pulse takes place in frame excitations.

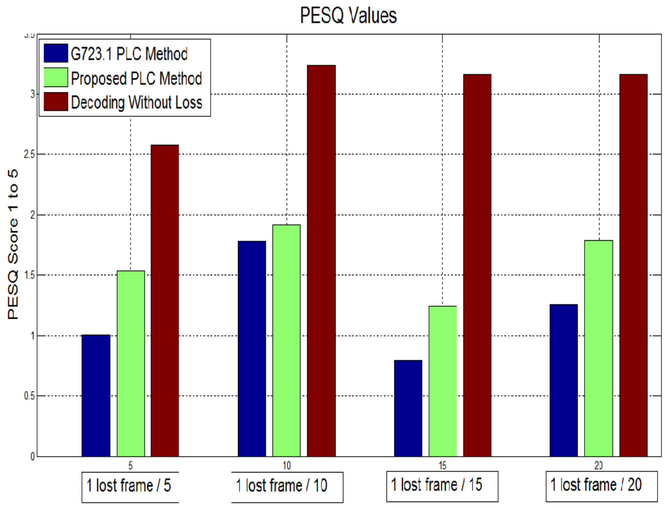

Fig. 6 PESQ values in function of loss rate.

IV. CONCLUSIONS

In this paper, we propose a new receiver-based packet loss concealment (PLC) algorithm for a CELP-Multi-Pulse type speech coder to improve the embedded concealment method, based on RLS Algorithm which counteracts speech quality decrease. Firstly, the initialized RLS parameters are set by the multiple tests. Then, the weight values of the RLS algorithm are adjusted previously to record a trace of the glottal excitation between the mixed ACB and FCB excitations and the FCB excitation only. 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, likewise, is used also to filter the next voiced excitations received frames, in the aim to improve the perceptual quality of the decoded received frames and avoid the error propagation. According 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. In this paper, an efficient method for reconstructing the missing parameters frame for CELP and Multi-Pulse based coders is presented, where the adaptive excitation codebook plays an important role to reconstruct the synthesis speech. Finally, From PESQ measurement tests under a variety of frame erasure conditions, we found that the proposed method, can reach 10% of improvement for the synthesis speech quality compared to the embedded algorithm in the standard G723.1 coder [2].

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