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Forward Error Correction Concealment Method for CELP-Based Coders in Packet Networks

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***Abstract*—This paper presents a concealment method based on forward error correction (FEC) to improve speech quality deterioration caused by packet losses for CELP based coders. We applied our scheme to the standard ITU-T G729 standard speech coder to evaluate the proposed method. The perceptual evaluation of speech quality (PESQ) and enhanced modified bark spectral distortion (EMBSD) tests under various packet loss conditions confirm that the proposed algorithm is superior to the concealment algorithm embedded in the G729. The performance measures prove that the concealment method based FEC is better at the expense of extra delay.**

***Index Terms*— VoIP, ITU G729, FEC, EMBSD, PESQ.**

I. INTRODUCTION

In the last few years, interactive multimedia services such as Voice over IP (VoIP) and video-conferencing have changed from promising new applications to reality. The increasing demand for audio and video services in the Internet has produced a number of commercial applications in addition to some very popular free tools such as Skype, Google Talk and Windows Live Messenger. However, some studies have shown that the current Internet infrastructure is not ready to provide acceptable quality to these applications [1][2]. One-way delay, jitter and packet losses are the most significant impairments to quality of service (QoS) in interactive streaming applications.

While jitter is usually mitigated through playout scheduling mechanisms [3][4], there is a number of alternatives for dealing with the effects of packet losses [5]. Techniques for recovering from errors in a data stream are based in either automatic repeat request (ARQ) or forward error correction (FEC). Retransmission schemes based on ARQ introduce end-to-end delays that are generally not suited for interactive communications. Forward error correction is a more attractive alternative when delay constraints are stringent.

Forward error correction can be either media-specific or media-independent. The former involves replicating media units with a possibly lower quality codec, while the latter uses error correcting codes in order to produce additional bits in the data stream that can be used to recover lost packets. Among media-independent FEC techniques, parity coding performs an exclusive-OR operation over a block of packets in order to produce an additional payload that can be used in case a single packet is lost within the protected block. In [6], the authors proposed the use of interleaved parity codes to recover lost packets in voice transmission over the Internet.

In this paper, we present concealment scheme based FEC media independent for CELP type coders. We apply this method to the ITU-TG729 Conjugate-Structure Algebraic CELP (CS-CELP) speech coder [7] that is widely used in VoIP applications. We compare the performance of the proposed algorithm with embedded standard method. We use the perceptual evaluation of the speech quality (PESQ) [8] and measure the enhanced modified bark spectral distortion [9]. The rest of this paper is organized as follows. In section II, we briefly review frame erasure concealment algorithm embedded in the ITU-T G729 standard speech coder. In section III, the proposed method is presented. In section IV, we report experimental results on the performance of our approach. Section V concludes our work.

II. FRAME ERASURE CONCEALMENT OF G729

In the G729 speech coder, an erased frame is reconstructed using the speech coding parameters of the previous received good frame [7]. Once frame erasure is detected, the new parameters are generated by analyzing the spectral parameters of the last good speech frame. The method replaces the missing excitation signal of the erased frame by taking one of the similar characteristics, while gradually decaying its energy. If n-th frame is detected as an erased frame, the G.729 repeats the spectral parameters of the last received good frame to the erased frame. In addition, an adaptive codebook gain and a fixed codebook gain are obtained by multiplying predefined attenuation factors by the gains of the previous frame. To avoid excessive periodicity a long term prediction lag is increased by one to the value of the previous frame.

III. CONCEALMENT METHOD BASED FORWARD ERROR CORRECTION

Media-independent FEC is often implemented using block or algebraic codes, like parity codes and Reed-Solomon codes. These codes work by taking a codeword of k data packets to generate n ↔ k additional check packets for transmitting n packets over the network. The additional packets are intended to help in the recovering of lost packets at the receiver. An example of parity coding has been developed and implemented by Rosenberg [10], which consists in applying a XOR operation across a group of packets to generate the corresponding parity packets. Fig. 1 depicts this scheme. One parity packet (the FEC packet) is generated after XORing n↔1 data packets; if there is just one loss in a group of n packets, that loss is recoverable.

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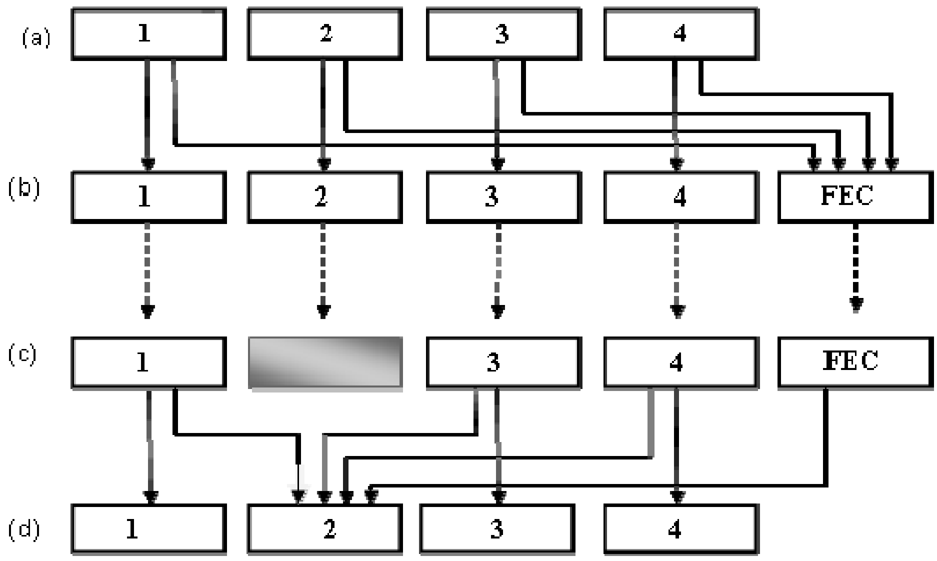

Fig. 1. FEC concealment scheme, (a) original frames,
(b) FEC generation, (c) frame loss, (d) reconstructed frame.

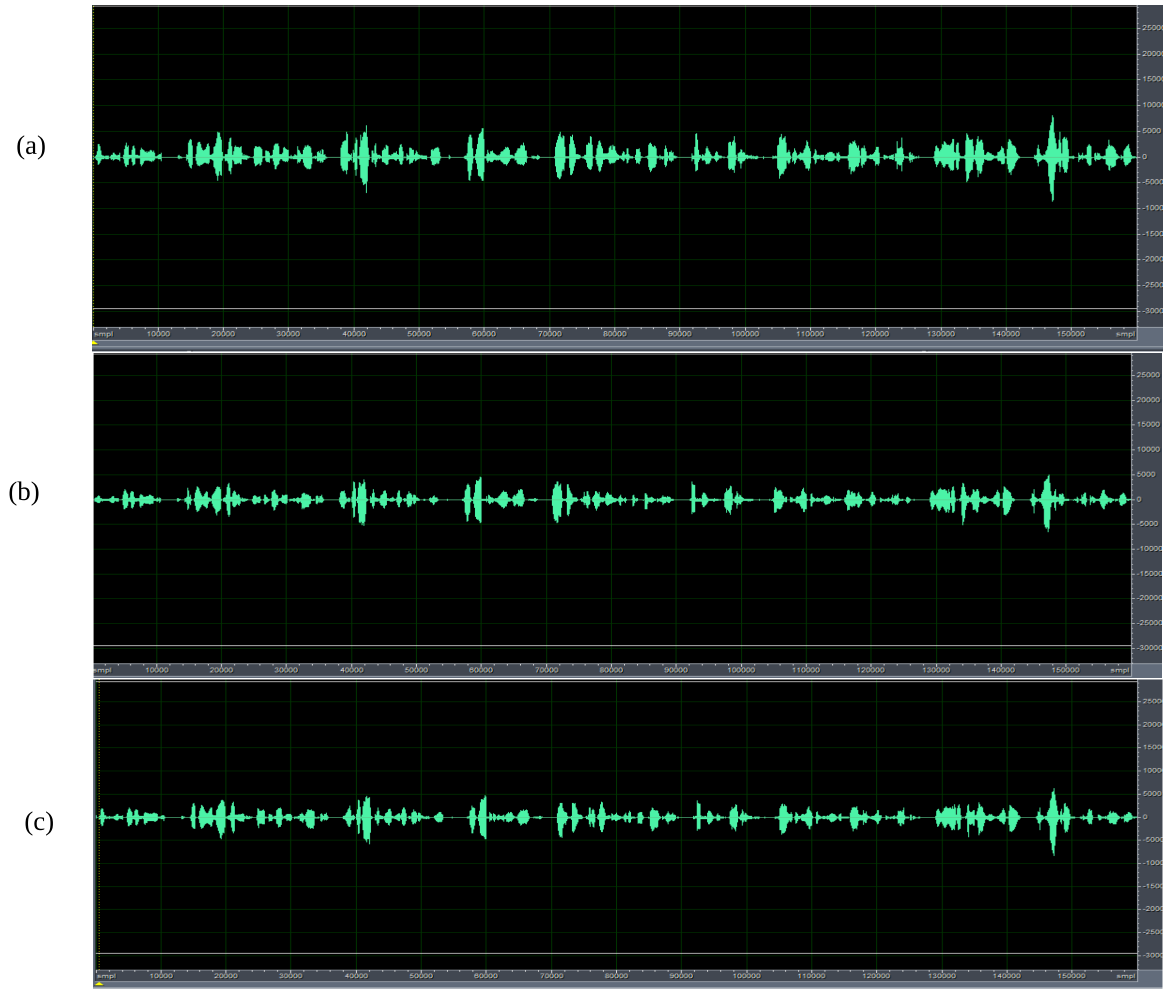

Fig. 2. Example of speech quality degradation due to frame loss, (a) original speech,
(b) reconstructed speech by G729 with 10% of loss , (c) reconstructed speech by FEC with 10% of loss

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Perkins has called this kind of FEC mechanism media-independent because the FEC operation does not depend on the contents of the packets. The principal advantage of this simple scheme is its simplicity, and in case of loss the repair for that loss is obtained by a single XOR operation. The main disadvantage is the delay and the increased bandwidth imposed on the transmission. A high delay would deteriorate the interactivity of a conversation, and thus the number of protected packets imposes the playout delay at the receiver in case of loss.

Fig. 2 shows an example of speech quality degradation when frame loss of 10 % is occurred. The reconstructed speech by the standard G729 and the proposed method are shown.

IV. Simulation Results

In this section we compare the performance of the proposed method with that of the embedded method in the G729. In the following development, we assume that the packet loss process seen by an end receiver can be described as a 2-state Markov chain, also known in the literature as the Gilbert model. Several studies have been proposed in attempt to characterize the loss processes experienced by traffic flows in communication networks [11][12][13], and it has been argued that the Gilbert model is not able to emulate the behavior of loss traces with long-term correlations. Despite these shortcomings, the Gilbert model is commonly applied in performance evaluation studies [14] [15], due to its analytical simplicity and the good results it provides in practical applications.

Let X_t denote the *t*-th packet outcome, with $X_t = 1$ representing a packet loss and $X_t = 0$ a successful transmission. Under the assumption of a Gilbert model, we let $p = P(X_t = 1 | X_{t-1} = 0)$ and $q = P(X_t = 0 | X_{t-1} = 1)$, as illustrated in Fig. 3. The steady state probabilities $P(X_t = 0)$ and $P(X_t = 1)$ are given, respectively, by $q/(p+q)$ and $p/(p+q)$.

We have simulated five loss rates as given in Table I.

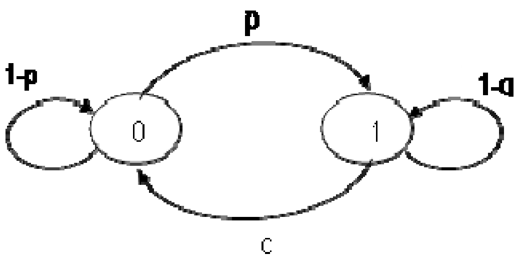

Fig. 3. Packet loss model used in this analysis.

TABLE I. SIMULATED LOSS RATES

$rate(\%)$	p	q
0	0.00	0.00
10	0.10	0.15
20	0.20	0.30
30	0.30	0.35
40	0.30	0.40

We use PESQ for an objective quality measure. Figs. 4 and 5 show comparison results for female and male speakers respectively from TIMIT database [16]. As the packet loss rate increases, the PESQ scores of the two algorithms decrease.

The scores of the proposed algorithm are higher than the embedded method in the G729 standard coder.

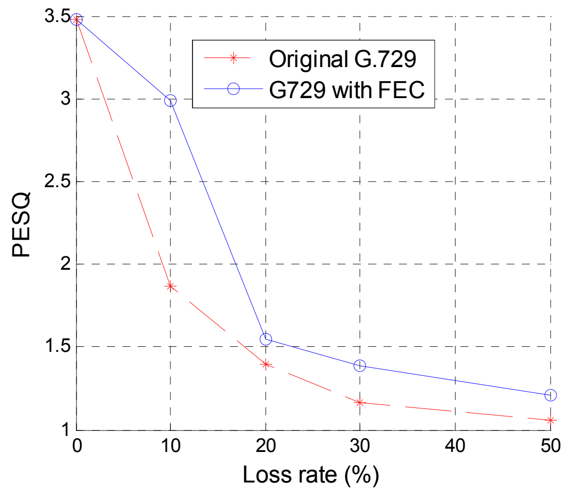

Fig. 4. Comparison of PESQ for female speakers decoded with original G729 (dash line) and the proposed method (4 fames interleaved) (solid line) under different loss rates.

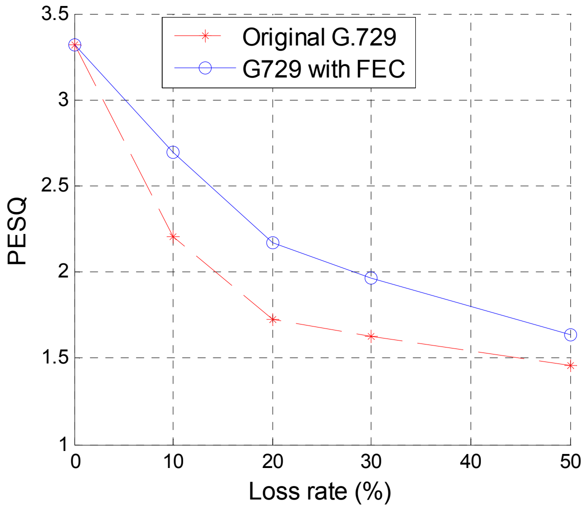

Fig. 5. Comparison of PESQ for male speakers decoded with original G729 (dash line) and the proposed FEC method (solid line) under different loss rates.

We performed an EMBSD test and the results are depicted in Figs. 6 and 7. As the packet loss rate increases, the EMBSD of the two methods increase.

It is shown that the proposed algorithm is always better than the embedded method in the G729 standard coder.

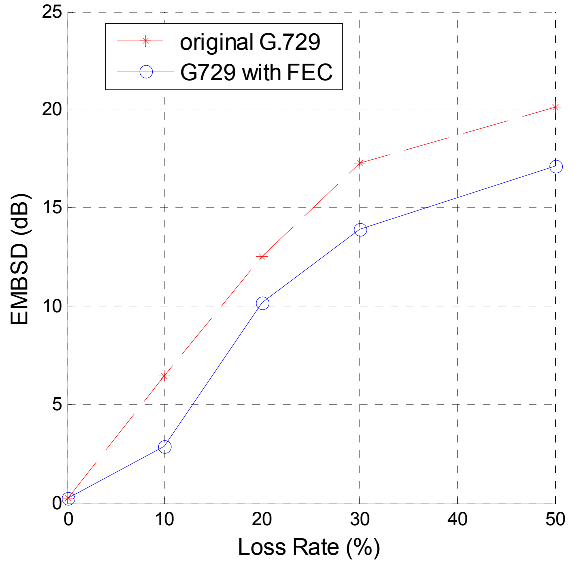

Fig. 6. Comparison of EMBSD for female speakers decoded with original G729 (dash line) and the proposed FEC method (solid line) under different loss rates.

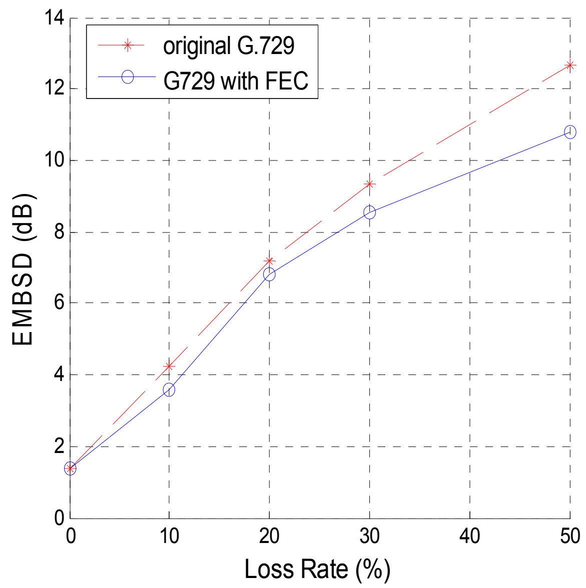

Fig. 7. Comparison of EMBSD for male speakers decoded with original G729 (dash line) and the proposed method (solid line) under different loss rates.

V. CONCLUSION

In this paper we have presented an efficient method for reconstructing the missing frames for CELP based coders and compared its performance with the embedded algorithm in the standard G729 coder. From PESQ measurement and EMBSD tests under a variety of frame erasure conditions, we found that the proposed method, improved significantly the speech quality compared to the embedded algorithm in the standard G729 coder at the expense of extra delay.

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