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PACKET LOSS CONCEALMENT FOR MDCT-BASED AUDIO CODEC USING CORRELATION-BASED SIDE INFORMATION

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Abstract. *In this paper, we investigate several methods for estimating the signal in lost packets of MP3 audio using side information. First, we discuss packet loss concealment based on packet copy for an MDCT-based audio codec, and point out problems that deteriorate the quality of the restored signal. Then we propose a packet loss concealment method using sign correction. The proposed method uses signs of lower MDCT coefficients as redundant information, and the sign information is used when estimating the MDCT coefficients of the lost packets. Next, we propose a new method that uses the same side information as the sign correction method. Our method is a combination of one-bit quantization and sign correction, which has proved to be better than sign correction for improving the correlation between the original signal and the restored signal. The experimental results show that the proposed method outperforms the sign correction method. Next, we investigate several methods that use two bits for correcting one coefficient. The experimental results show that the combination of a two-bit correction and one-bit correction give the best result.*

Keywords: Packet loss concealment, MP3, Side information, Sign correction, Correlation

1. **Introduction.** In recent years, huge amounts of audio data (speech, music, etc.) have been transmitted through the Internet. We can listen to internet radio or music streaming service such as Napster [1], and also talk to other people using VoIP systems. In these systems, audio data are split into frames, and several frames are packed into an IP packet. When transmitting the audio packets over the Internet, several packets may be lost. This causes a deterioration in the quality of the decoded sound if the lost packets are properly recovered.

Many methods have been proposed to solve this problem [2]. One of the most general methods is a retransmission-based approach combined with a stream buffer, such as the progressive download based on TCP [3]. However, this kind of approach is not suitable for multicast-based applications because the retransmission is based on point-to-point communication.

To enable multicast-based packet transmission, we need to use a connectionless protocol such as UDP [4] or RTP [5]. For example, the QuickTime Streaming Server employs the RTP for live broadcasting of audio and video streams [6]. However, a connectionless