Chapter 15
Autonomic QoS Optimization of Real-Time Internet Audio Using Loss Prediction and Stochastic Control

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ABSTRACT
Quality of Internet audio is highly sensitive to packet loss caused by congestion in links. Packet loss for audio is normally rectified by adding redundancy using Forward Error Correction (FEC). Alternatively, path diversity mechanisms are used to improve reliability and thus session quality. To achieve optimized receiver audio quality for transmissions using single or multiple paths, the authors propose a self-adaptive joint Error and Rate Control framework based on packet loss prediction and on-line quality assessment. The Error Control chooses appropriate FEC proactively to preserve quality with optimal bandwidth, using a Markov Decision Process (MDP) and a stochastic inventory control, a novel approach for multimedia error recovery. The Rate Control uses a quality optimization model to determine the optimal dispersion over single or multiple paths. This paper will present results using simulation and Internet experiments to show the superiority of this mechanism over other similar techniques.

INTRODUCTION
Deployment of Next Generation Networks and service convergence is likely to make IP technology the main vehicle for carrying interactive voice and video. Quality of Internet multimedia is highly sensitive to packet loss (Cole & Rosenbluth, 2001; Markopoulou, Tobagi, & Karam, 2002) caused by congestion in the links. Packet loss for real-time UDP audio and video is normally rectified using FEC (Lin & Costello, 1983), where a number of redundant packets are sent with the original packets. FEC is shown to be the most common technique for maintaining acceptable quality in presence of loss (Jiang & Schulzrinne, 2002). But the challenge with FEC is its bandwidth overhead, as FEC must be sufficient but not excessive, and timely, in order to be effective. Thus FEC degree
and duration should be chosen adaptively in response to the network conditions of bandwidth degradation and packet loss. Alternatively, path diversity mechanisms, where session packets are dispersed over multiple paths, are used to improve reliability and thus multimedia session quality in overlay/p2p systems (Andersen, Balakrishnan, Kaashoek, & Morris, 2001; Apostopoulos, 2001; Fei, Tao, Gao, & Guerin, 2006), or multihoming networks (Akella, Pang, Maggs, Seshan, & Shaikh, 2004; Andersen, Balakrishnan, Kaashoek, & Morris, 2001). Packet dispersion can be a viable option assuming that at least one path will provide good performance to maintain session quality (Savage, Collins, Hoffman, Snell, & Anderson, 1999). In particular, packet dispersion is significantly beneficial to reduce the effect of high degree of burst loss (Zlatokrilov & Levy, 2004). But compared to FEC, path diversity is a relatively costly solution, since it can incur more sudden changes in one-way delay (Tao et al., 2005), disrupting a smooth playback. Thus the challenge is to provide an error control solution that adaptively combines the benefits of both FEC and dispersion techniques under different network conditions in order to provide optimal quality. It is also imperative that such error control is combined with an efficient rate control in order to provide bandwidth-friendly transmission with an effective degree of loss recovery.

In this paper we present a self-adaptive joint error and rate control mechanism that ensures an optimal receiver quality at real-time by taking proactive control actions, based on packet loss prediction (Roychoudhuri & Al-Shaer, 2005) and on-line quality assessment (Roychoudhuri & Al-Shaer, 2005; Roychoudhuri, Al-Shaer, & Settimi, 2006). The Loss Predictor is an end-to-end monitor that tracks the one-way delay and inter-packet gap of in-line stream packets, as well as short- and long-term trends, and indicates the current degree and severity of congestion, hence the likelihood of packet loss in the next window of packets. The Audio Quality Assessor is a passive monitor that assesses the receiver audio quality objectively for a real-time audio stream in terms of Mean Opinion Score (MOS), the ITU standard of voice quality assessment (ITU-T Recommendation P.800, 1996). The Error Control recovers individual path loss and maintains receiver quality in the short term using optimal FEC. The FEC degree and duration are chosen dynamically using an MDP and stochastic inventory control, a novel approach in the area of multimedia error control. The Rate Control detects changing bandwidth and uses a rate quality optimization model to proactively diversify optimal codec/bitrate combination over single or multiple paths. This is superior to the reactive feedback used in current sender based single-path (Bolot & Vega-Garcia, 1996; Mohamed, Cervantes-Perez, & Afifi, 2001) and multi-path (Nguyen & Zakhor, 2003) rate and error control mechanisms that depend on RTCP feedback of packet loss. Second, our mechanism is user quality centric, as opposed to ad-hoc reaction to network packet loss using static FEC (Jiang & Schulzrinne, 2002). The sole purpose of the error control and rate adaptation actions is to optimize receiver quality at real time using objective audio quality assessment. The control decisions are made to improve or retain the end quality above a specific threshold, and are not simply based on network condition targets, such as bandwidth or packet loss levels. That may mean that certain degree of loss may be tolerable in terms of quality and actions will be taken only when required, ensuring the efficiency of the mechanism. The framework is currently designed for audio, but is easily extensible to video applications.

RELATED WORK

As an initial work in this area, Bolot and Garcia presented a combined Error and Rate control mechanism for audio that adapts to the loss feedback from the receiver using RTCP (Bolot & Vega-Garcia, 1996). Padhye, Christensen, and