Adaptive Playout Buffering Schemes for IP Voice Communication

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INTRODUCTION

Audio communication over IP-based networks represents one of the most interesting research areas in the field of distributed multimedia systems. Today, routing the voice over Internet enables cheaper communication services than those deployed over traditional circuit-switched networks. BoAT (Rocchi, Ghini, Pau, Salomoni, & Bonfigli, 2001a), Ekiga, FreePhone (Bolet & Vega Garcia, 1996), ICam, Kix, NeVot (Schulzrinne, 1992), rat (Hardman, Sasse, & Kouvelas, 1998), Skype, Tapioca, VAT (Jacobson & McCanne, n.d.), WengoPhone, and YATE, are just few examples of free VoIP software available to Internet users.

Without any doubts, new (wired and wireless) high-speed, broadband networks facilitate the transmission of the voice over the Internet and have determined the success of these applications. However, the best effort service offered by the Internet architecture does not provide any guarantee on the delivery of (voice) data packets. Thus, to maintain a correct time consistency of the transmitted audio stream, these voice communication systems must be equipped with schemes able to deal with the unpredictability of network latency, delay jitter, and possible packet loss.

BACKGROUND

Several proposals to face with the effects caused by network delay, delay jitter, and packet loss rate on continuous media stream playout have been presented in literature. For instance, protocol suites (e.g., RSVP, DiffServ) and networking technologies (e.g., ATM) have been devised that provide users with quality of service (QoS) guarantees (Zhang, Deering, Estrin, Shenker, & Zappala, 1993). Yet, these approaches have not been widely adopted as usual means to provide guarantees of QoS to Internet users.

An interesting alternative that is now widely exploited in most existing Internet audio communication tools amounts to the use of adaptive playout control mechanisms. Basi-
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Figure 1. Smoothing out jitter delay at the receiver

Figure 2. A small playout delay

Figure 3. A large playout delay