

# Adaptive Playout Buffering Schemes for IP Voice Communication

**Stefano Ferretti**

*University of Bologna, Italy*

**Marco Roccetti**

*University of Bologna, Italy*

**Claudio E. Palazzi**

*University of Bologna, Italy*

## 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 (Roccetti, Ghini, Pau, Salomoni, & Bonfigli, 2001a), Ekiga, FreePhone (Bolot & Vega Garcia, 1996), iCall, KiAx, 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.

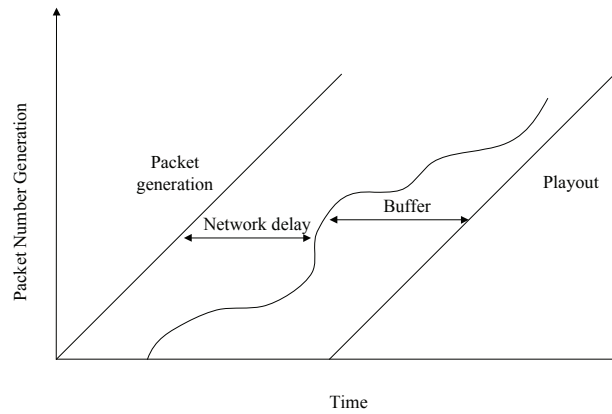
An interesting alternative that is now widely exploited in most existing Internet audio communication tools amounts to the use of adaptive playout control mechanisms. Basi-

cally, these schemes are faced with the unpredictability of IP networks by compensating for variable network delays and jitters experienced during the transmission of audio packets. In particular, delay jitter is smoothed away by employing a playout buffer at the receiver side and by dynamically enqueueing audio packets in it. Output of received and buffered packets is thus artificially delayed for some time so as to have a constant audio packet playout rate, hence absorbing negative effects introduced by the delay jitter. Such a buffering policy must be adaptive, since delay jitter on the Internet may vary significantly with time. This way, dynamic playout buffers hide packet delay jitters at the cost of additional delays at the receiver (see Figure 1).

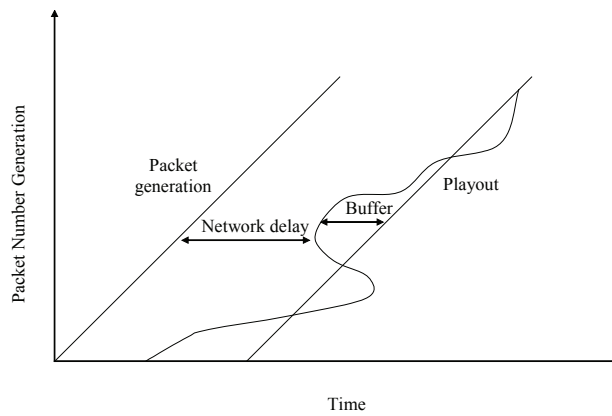
Summing up, each audio packet transmitted on the network has an associated scheduled playout delay, being defined as the total amount of time experienced by such audio packet from the instant it is generated at the source and the instant it is played out at the destination. Such a playout delay consists of: (1) the time needed for the transmitter to collect an audio sample and prepare it for transmission, (2) the network delay, and (3) the buffering time, that is the amount of time that a packet spends queued in the destination buffer before it is played out. Based on this specific notion of playout delay, a received audio packet is defined to be late when its arrival time at the destination is after the expiration of its scheduled playout time.

Needless to say, while the approach of buffering audio packets enables delay jitter removal from audio packet streams and guarantees the speech intelligibility, a critical trade-off exists between the amount of delay that is introduced in the buffer and the percentage of late packets that are not received in time for playout (and are consequently lost, see Figures 2 and 3). In fact, the longer the additional delay due to the buffering policy, the more likely a packet arrives before its scheduled playout time, but the higher the time spent by earliest packets in the buffer. Summing up, on one side, a too large percentage of audio packet loss may impair the intelligibility of an audio transmission, but, on the other side, too large playout delays (due to buffer-

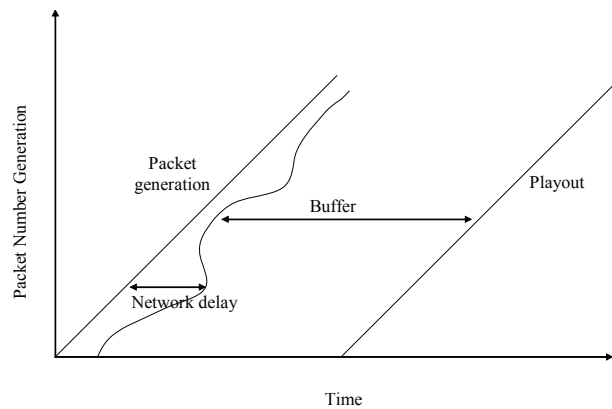
*Figure 1. Smoothing out jitter delay at the receiver*



*Figure 2. A small playout delay*



*Figure 3. A large playout delay*



5 more pages are available in the full version of this document, which may be purchased using the "Add to Cart" button on the publisher's webpage: [www.igi-global.com/chapter/adaptive-payout-buffering-schemes-voice/13544](http://www.igi-global.com/chapter/adaptive-payout-buffering-schemes-voice/13544)

## Related Content

---

### The Expert's Opinion

Karen Mowery (1990). *Information Resources Management Journal* (pp. 42-46).

[www.irma-international.org/article/expert-opinion/50927](http://www.irma-international.org/article/expert-opinion/50927)

### Utilization and User Satisfaction in End-User Computing: A Task Contingent Model

Changki Kim, Kunsoo Suhand Jinjoo Lee (1998). *Information Resources Management Journal* (pp. 11-24).

[www.irma-international.org/article/utilization-user-satisfaction-end-user/51057](http://www.irma-international.org/article/utilization-user-satisfaction-end-user/51057)

### Reinventing Business Processes through Automation: A Case Study

David Paper (1999). *Success and Pitfalls of Information Technology Management* (pp. 97-107).

[www.irma-international.org/chapter/reinventing-business-processes-through-automation/33483](http://www.irma-international.org/chapter/reinventing-business-processes-through-automation/33483)

### User Spreadsheet Systems Development

Anders Avdic (2005). *Encyclopedia of Information Science and Technology, First Edition* (pp. 2967-2972).

[www.irma-international.org/chapter/user-spreadsheet-systems-development/14727](http://www.irma-international.org/chapter/user-spreadsheet-systems-development/14727)

### New Perspectives on Rewards and Knowledge Sharing

Gee-Woo (Gilbert) Bock, Chen Way Siewand Young-Gul Kim (2005). *Encyclopedia of Information Science and Technology, First Edition* (pp. 2110-2115).

[www.irma-international.org/chapter/new-perspectives-rewards-knowledge-sharing/14569](http://www.irma-international.org/chapter/new-perspectives-rewards-knowledge-sharing/14569)