Available online at www.sciencedirect.com
SCIENCE DIRECT
Signal Processing: *Image Communication* 20 (2005) 755–772
SIGNAL PROCESSING: IMAGE COMMUNICATION
www.elsevier.com/locate/image

Proxy-assisted interactive-video services over networks with large delays ☆

Ali C. Begen, Yucel Altunbasak*

School of Electrical and Computer Engineering, Georgia Institute of Technology, Atlanta, GA, USA

Received 27 April 2005; accepted 2 May 2005

Abstract

While emerging broadband access technologies such as DSL and cable are making multimedia services feasible and economically attractive for end-users, there still exist several hurdles in terms of service sustainability and reliability. Unfortunately, without the desired quality-of-service (QoS) support, tackling these hurdles with traditional solutions is an insuperably difficult task. Yet, novel designs that are proven to be useful in various scenarios may easily fail when the underlying network experiences severe packet loss or delay. Such circumstances are unavoidable in today's best-effort Internet and will likely prevail in the near future as well. A promising approach in satisfying the stringent requirements of delay-intolerant video applications is to benefit from configurable proxies. In this study, we introduce a versatile proxy-based solution to enhance the performance of such applications running over networks with large delays. We first propose a methodology that accurately identifies lost packets in real time. This methodology is then used by the proxy and end-users to improve the error-control/protection capability of the video applications. By Internet experiments between the US and Europe, we demonstrate the effectiveness and potential benefits of the proposed approach.

Keywords: Proxy support; Real-time video communication; Interactive video; Videotelephony; Videoconferencing; RTO estimation

1. Introduction

The primary role of ubiquitous networking, in particular of the Internet, is to disseminate information in a timely manner and provide an inexpensive communication platform to its users. As the access technologies provide high bandwidths at economically attractive prices and by the help of the advances in audiovisual signal processing, a larger number of people are communicating interactively through networks every day. New multimedia applications such as IP telephony, teleconferencing, videoconferencing, on-demand

☆A preliminary version of this paper has appeared in IEEE International Conference on Image Processing (ICIP) 2004.

*Corresponding author.

E-mail addresses: acbegen@ece.gatech.edu (A.C. Begen), yucel@ece.gatech.edu (Y. Altunbasak).

0923-5965/$ - see front matter
doi:10.1016/j.image.2005.05.001

video, interactive games, distance learning, and many more, are rapidly emerging. However, despite all the advances, these technologies have so far not been able to provide the desired reliability. The reason behind this shortfall is that the service requirements of these applications differ significantly from those of traditional data-oriented applications, and hence, existing protocols and mechanisms that have been primarily deployed for data communication cannot support such challenging applications. To address this problem, recent studies focus on designing new solutions that are concerned with the unique features of real-time media. However, without the essential quality-of-service (QoS) provisioning, even these *media-aware* approaches struggle to satisfy the rigid requirements, if the network becomes physically incapable or it performs poorly for a prolonged amount of time. In other words, clever design has its limitations. Such circumstances are unavoidable in today's best-effort Internet and will likely prevail in the near future as well.

A particularly useful method to improve the reliability of delay-intolerant video applications is to utilize proxies. Depending on their types and functionalities, proxies can be used in different contexts. In this work, we investigate the potential benefits of proxies in an interactive application for which packet delivery deadlines are stringent. In this class of applications, dealing with large delay variations for the end-users that are geographically distant from each other is arduous. Large round-trip delays hinder feedback-based error-control/protection methods since the validity of the feedback messages are mostly depreciated by the time they are received. As a result, end-users can be ineffective in taking necessary actions against missing packets. In addition, due to the late feedback reports, end-users will also fail to adapt to the network conditions in a timely manner, if the network conditions change rapidly.

As an illustrative example, let us consider a videotelephony session running between two end-users. In videotelephony applications, packets that are not delivered within 200 ms disturb the video and result in intermittent and unintelligible video. Many empirical studies (e.g., [8,10,22]) report that end-users experience a considerable amount of delay jitter. While the amount of the reported jitter is the largest for dial-up modem users, cable customers usually observe more jitter compared to DSL subscribers. The excessively-delayed packets are essentially as useless as lost packets. Although it is possible to reduce the number of late packets by increasing the playout delay in one-way streaming applications, this increase cannot be arbitrarily large. In two-way conversational applications, delay tolerance is severely limited in order to maintain the interactivity.

Evidently, large delay variations reduce the chance of delivering packets successfully before their decoding deadlines. In addition to delaying packets, jitter has an implicit, but comparably devastating, impact on the video quality. As the jitter aggravates, it becomes more difficult to determine whether a missing (or an unacknowledged) packet is actually lost or merely delayed. Without a doubt, accurate and timely inference of late/lost packets is important for the sender/receiver to take well-timed actions for error control/protection. For example, in the case of retransmission-based error-control methods, a retransmission at an early stage can be redundant as it might generate duplicate packets, whereas a late attempt will probably be useless. Naturally, as the delay variation decreases, the sender/receiver likely gives more accurate and well-timed decisions on missing packets, which probabilistically increase the chance of timely delivery of the packets. Similarly, the performance of adaptive channel and video coding methods can be significantly improved, if the sender is informed about the status of the packets more promptly.

Unfortunately, delay is a feature of the underlying physical network. Hence, we cannot do much to reduce it on an end-to-end basis. Network paths that experience large delays will likely experience large jitter as well due to the potential large number of hops between the end points. Hence, video applications running over such paths will perform poorly. One practical way to virtually eliminate these adverse effects is to divide the large network into two or more smaller sub-networks. This can be achieved by introducing *intermediate proxies* (I-Proxy) on the path between