Chapter XVIII
Adaptive Retransmission Scheme for Video Streaming Applications

Árpád Huszák
Budapest University of Technology and Economics, Hungary

Sándor Imre
Budapest University of Technology and Economics, Hungary

ABSTRACT

In this chapter we present a novel selective retransmission scheme, based on congestion control algorithm. Our method is efficient in narrowband networks for multimedia applications, which demand higher bandwidth. Multimedia applications are becoming increasingly popular in IP networks, while in mobile networks the limited bandwidth and the higher error rate arise in spite of its popularity. These are restraining factors for mobile clients using multimedia applications such as video streaming. In some conditions the retransmission of lost and corrupted packets should increase the quality of the multimedia service, but these retransmissions should be enabled only if the network is not in congested state. Otherwise the retransmitted packet will intensify the congestion and it will have negative effect on the audio/video quality. Our proposed mechanism selectively retransmits the corrupted packets based on the actual video bit rate and the TCP-Friendly Rate Control (TFRC), which is integrated to the preferred DCCP transport protocol.

INTRODUCTION

The new applications for delivering continuous media is set to become more and more popular, driven by user demand and network technology advances providing quality of service (QoS) and increased bandwidth to the user terminal. With the rise of multimedia and network technologies, multimedia has become an indispensable feature on the Internet. Animation, voice, and video clips become more and more popular on the Internet. Multimedia networking products like Internet telephony, Internet TV, video conferencing have appeared on the market. These applications are not only used in reliable wired networks but also in wireless environment where the obstacles of the expansion are the higher bit error ratio of the radio link and the limited bandwidth of the mobile links. Third-generation mobile networks and new wireless technologies like WiMAX, HSDPA, HSUPA, and so forth are rapidly approaching reality, also providing higher bandwidth levels with the ability to transmit video streams in acceptable quality.

The real-time applications usually encode audio/video in a format that handles loss of full packets. This feature makes it possible to transmit the coded video in hazardous channels, without retransmitting the corrupted or lost packets. Traditional error control mechanisms generally use retransmission to provide reliability at the expense of latency. Loss tolerant multimedia applications should use retransmissions as well, but the retransmission will be successful only if the retransmitted packet arrives at the receiver before the playback.

Loss-tolerant real-time multimedia applications such as video conferencing or video streaming prefer UDP to avoid unacceptable delay introduced by packet retransmissions. UDP is considered selfish and ill-behaving because TCP (RFC 2581) throttles its transmission rate against the network congestion whereas UDP (RFC 768) does not have such control mechanisms. Some of the nowadays investigated transport protocols [e.g., UDPLite (RFC 3828), SCTP (RFC 2960), DCCP (RFC 4340), etc.] can be more efficient for audio/video streaming applications. The unreliable UDP, UDPLite, and DCCP do not retransmit any corrupted packets while SCTP will do it until all the packets arrive correctly to the client. These protocols basically do not adapt themselves to the actual conditions; nevertheless it would lead to the increase of effectiveness. When the conditions make it possible to retransmit the lost or damaged packets it is worth it to do it, but in some cases the effect of the retransmission is harmful. When the network is in congested state or the RTT (round trip time) is so high that the retransmitted packet will not arrive in time, the retransmission will not increase the quality; moreover will increase the load and latency. To efficiently control the retransmissions a selective retransmission scheme is needed.

The rapid growth in the usage of streaming media has heightened the need for a congestion control protocol suitable for streaming media. Among the proposed streaming-media congestion control protocols, TCP-Friendly Rate Control (TFRC) (RFC 3448) is one of the promising solutions. TFRC maintains an equal or lesser average sending rate as competing TCP connections, while providing a relatively smooth sending rate to help packets to meet the real-time constraints required by streaming media. Among the unreliable transport protocols only the Datagram Congestion Control Protocol (DCCP) supports congestion control mechanisms (TCP-Like, TFRC). The congestion control mechanism needs information about the packet loss event; hence the DCCP header includes a sequence number field that identifies the packet. Consequently the streaming server gets information on which packet was lost and which was received.

In this chapter a new TFRC controlled selective retransmission scheme is proposed for multimedia transmission over noisy wireless channels in order to ensure acceptable video quality at the receiver. To analyze the effectiveness of our approach we used the NS-2 network simulator.

The rest of the chapter is organized as follows. A review of related work in selective retransmission and TFRC-based video streaming is presented in Section II. A brief overview of the preferred DCCP transport layer protocol is presented in Section III. In Section IV we propose a congestion sensitive retransmission method for multimedia applications. The obtained results are discussed in Section V. Finally, we summarize our paper and outline our future work in the last section.

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