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CONTENTS

Practical Rate Control of Scalable Video Streamed to Your Smartphones

A short review for "Advanced rate adaptation for unicast streaming of scalable video"

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C. Liu, I. Bouazizi, and M. Gabbouj, "Advanced Rate Adaption for Unicast Streaming of Scalable Video", Proc. of IEEE International Conf. on Communications 2010, May 2010.

With the accelerated developments in the wireless communication technologies, mobile multimedia streaming rises as one of the most popular applications. In fact, a recent Cisco report [1] indicates that the multimedia traffic will represent more than 66% of the total mobile data traffic by 2014. The most critical feature of streaming services is the in-time delivery of media data for timely playback at the receiver. However, wireless networks are characterized by varying bandwidth (usually dependant on the number of users sharing the same channel) and by link layer loss. To cope with the dynamic characteristics of wireless networks, a plurality of rate adaptation approaches have been proposed and deployed. Among others, a rate adaptation method [2] based on the client buffer feedback mechanisms in 3GPP Packet-Switched Streaming Service (PSS) [3] was proposed. The 3GPP client buffer feedback-based rate adaptation method detects network bandwidth changes by continuously monitoring the amount of buffered media time, which ultimately results in a superior behavior compared to packet loss-based rate adaptation [4]. The reason is that reacting to changes in the buffered media time enables to smooth out short-term bandwidth variations, detect congestion early enough, and differentiate between packet loss due to congestion and that caused by the wireless transmission medium.

Recent advances in scalable video coding such as SVC/H.264 [5] have paved the way for further development of flexible and adaptive media streaming applications. One of the key advantages of SVC/H.264 is its support for coarse-granularity scalability (CGS) and medium-granularity quality scalability (MGS). MGS provides a larger number of quality layers with reasonable stepwise bitrate gaps between the different quality layers and with small decoding complexity compared to CGS. Furthermore MGS supports more flexible switching between the different quality layers than CGS by providing packet-based quality scalability [6]. Hence, deploying MGS in combination with rate adaptation was sufficiently promising to trigger several research efforts as well as some initial commercial implementations. To fully utilize the switching flexibility in SVC, knowledge of the reception status for each MGS sub-stream is required at the server. However, 3GPP PSS lacks the capability to signal client buffer feedback at the sub-stream level, particularly when the different layers are transmitted over the same RTP session to avoid continuous setting up and tearing down of RTP sessions.

Without the sub-stream level feedback, the rate adaptation method specified in 3GPP PSS fails to correctly estimate the buffered media time, since the server bases its estimation on the stream level feedback, thus assuming the same amount of buffered media for all sub-streams. Furthermore, the authors assert that it is worthwhile to assign different buffering targets to the different layers, as this will improve playback continuity through graceful degradation in case of drastic bandwidth drops. Consequently, the authors propose a novel fine-granular estimation of the client buffer status in conjunction with MGS for rate adaptation in unicast streaming sessions.

The MGS switching flexibility is exploited when adapting the transmission rate to the varying wireless link conditions. The feedback mechanism is defined as an extension to the PSS feedback tools. Based on the multi-buffer feedback mechanism, the server continuously monitors the reception status of each sub-stream and adapts its transmission rate accordingly to provide accurate and fast reaction to congestion signals. Furthermore, an inverse pyramid-like threshold selection for the buffer level of the different sub-streams is used to improve playback continuity in case of a sudden drop in bandwidth.

Simulation results show that the proposed multi-buffer feedback-based rate adaptation algorithm outperforms the traditional 3GPP PSS compliant rate adaptation method by quickly adapting the temporal/spatial resolution to the varying network resources without unnecessary bouncing among different temporal/spatial resolutions. As graceful degradation, interruption-free playback, and video quality stability are essential for a good user experience, the proposed algorithm shows the potential of achieving the highest possible video quality and the best user experience for unicast streaming applications in wireless environments.

Modifications to the 3GPP PSS standard to support the proposed rate control algorithm are being considered in 3GPP in conjunction with the adoption of SVC for high-quality mobile streaming services. Other future research directions include: (i) quantitative comparison between an SVC-based streaming system with the proposed rate adaptation algorithm against a nonscalable streaming system with 3GPP PSS compliant rate adaptation method, and (ii) extension of the proposed rate adaptation algorithm to video multicast services which may become more popular in next generation multicast-enabled wireless networks, such as WiMAX and LTE.

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