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Improvement of adaptive HTTP streaming using advanced real-time rate adaptation☆

Selvaraj Kesavan*, J. Jayakumar

Department of Electrical & Electronics Engineering, Karunya University, Coimbatore 641 114, Tamil Nadu, India

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ABSTRACT

We propose a real-time rate-adaptation approach for adaptive Hypertext Transfer Protocol streaming. In this advanced real-time rate-adaptation (ARTRA) approach, we monitor the parameters at the client in real time, and select the most likely available bit representation for the next segment download. The algorithm dynamically triggers the rate evaluation and decides the rate switch according to metrics calculated using the segment download duration, measured throughput, and buffer data-level indication. The performance of the ARTRA is compared with those of throughput-based adaptation and segment-fetch time-based adaptation algorithms. An analysis of the experimental results shows that ARTRA enables clients to use the maximum available end-to-end network capacity and delivers the best user experience by reducing re-buffering events and bitrate level changes. ARTRA performs remarkably well under challenging network conditions and supports system stability and robustness.

1. Introduction

Multimedia applications, such as playback, streaming, and video telephony, have become an integral part of our lives. With the advent of the internet, streaming has begun to replace the use of storage devices as the method by which large files, such as audio/video content, are delivered to consumers. In a heterogeneous network architecture, achieving the optimal quality of service (QoS) and quality of experience (QoE) in streaming poses a significant challenge. Streaming methodologies implement intelligent rate-adaptation models to calculate network metrics and adjust the system to pick the stream, which then creates a tradeoff between the network bandwidth and media quality. The client media player initially buffers a portion of data before starting actual playback to balance the throughput variation. However, throughput delivery is less likely when short-time bandwidth variation occurs due to the sharing of network bandwidth among many clients. This eventually leads to buffer drain, which degrades media quality and the streaming experience of the user.

Development of dynamic adaptive streaming over Hypertext Transfer Protocol (HTTP) (DASH) specifications helps overcome the shortcomings of conventional progressive-download [1] and real-time transport protocol (RTP) [2] streaming approaches. The basic objective of adaptive HTTP streaming is to divide the audio and video into a number of small chunks of ideal duration, encode them at different bitrates, and store them, and deliver them to the client via HTTP downloading. MPEG and 3GPP work together to standardize the adaptive streaming of continuous media over HTTP [3]. DASH became an international standard in 2011; it was revised in 2014 from a previous version published in ISO/IEC 23,009-1:2012 in 2012

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* Corresponding author.
E-mail addresses: selvarajkesavan@gmail.com (S. Kesavan), jayakumar@karunya.edu (J. Jayakumar).

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[4]. An application-level adaption is required to compensate for throughput variations owing to the TCP congestion control effect and bandwidth variations. Without a suitable rate-adaptation algorithm, an adaptive-streaming client might frequently undergo oscillations and interruptions in playback, leading to a poor user experience. Proper bitrate selection and effective rate adaptation are required to ensure smooth media data flow and efficient use of the available network bandwidth and end-device capabilities.

To the best of our knowledge, adaptive HTTP streaming rate adaptation suffers from multiple problems, such as clients failing to choose optimal bitrates, interrupted playback, and slow convergence. Thus, we study and analyze the behavior of popular conventional throughput-based adaptation (TBA) and segment-fetch time-based adaptation (SFTBA) algorithms and propose a novel rate-adaptation approach for overcoming current rate-adaptation limitations.

Here, we present a novel, client-centric advanced real-time rate-adaptation (ARTRA) approach for monitoring and triggering rate evaluation according to the ideal or actual segment download time and buffer warning/critical level. The optimal bitrate representation is chosen using the buffer occupancy and received throughput. When necessary, the ARTRA system initiates rate adaptation in real time, rather than waiting for the previous segment to finish downloading. The contributions of this paper can be quantified in three aspects: First, we propose the novel ARTRA approach for DASH. Second, we capture and analyze the conventional TBA and SFTBA approaches. Third, the performance of the ARTRA, TBA, and SFTBA in a real-time experiment are evaluated and analyzed using key performance streaming indices. TBA is presented as Algorithm 2 and is based on a previously proposed algorithm [5]. SFTBA is presented as Algorithm 3 and is based on an algorithm proposed by Corvoysier and Zakari [6]. Detailed analysis of each algorithm provides key insights into the suitability and limitations of the algorithm under different network conditions.

The rest of the paper is organized as follows. Section 2 presents related work that describes different rate-adaptation approaches in adaptive-streaming models. The proposed ARTRA approach for adaptive HTTP streaming is presented in Section 3. The system implementation and description of the conservative segment-fetch time and TBA algorithms are given in Section 4. The real-time, internet-based, on-demand, live adaptive-streaming experiment setup is explained in Section 5. The experimentation results, a performance analysis of ARTRA, and comparisons are provided in Section 6. Section 7 concludes the paper.

2. Related work

Rate adaptation in DASH is an emerging and active research field. The DASH standard [4] does not provide any rate adaptation logic or approach recommendations. Clients are allowed to implement optimized proprietary algorithms suited to the device and network conditions. Hence, many commercial and open-source DASH clients implement their own rate-adaptation algorithms with minimal or no change in the server. In addition to DASH, numerous commercial adaptive-streaming methods developed by different commercial vendors, such as Microsoft Smooth Streaming (MSS), Apple HTTP Live Streaming (HLS), and Abode HTTP Dynamic Streaming (HDS), use different file formats and transportation standards.

Alaoui and Dröge [5] developed an algorithm to determine the bitrate switch-up or switch-down of the next segment from the time required to download the current segment. However, this algorithm does not consider the segment size, which may lead to an overestimation of the available network bandwidth. Corvoysier and Zakari [6] introduced a TBA approach as part of the GStreamer multimedia framework [15] DASH de-multiplexer implementation. The algorithm determines the next optimal bitrate according to the measured average bitrate for the current downloaded segment.

Chenghao Liu et al. [7] proposed algorithms that help select appropriate switch-down/switch-up bitrates in serial and parallel segment-download methods. They estimated the current available bandwidth by comparing the actual and expected segment-fetch durations. The experimental methodology and metrics are suitable for measuring and comparing the performance of any adaptive HTTP streaming rate-adaptation algorithm. However, their algorithm performs the adaptation calculation according to the segment duration. Encoding and compression schemes cause the segment size to vary widely for the same bitrate and duration. Additionally, variations in the download times for the segments can affect the available bandwidth-calculation accuracy.

Buffer-based rate-adaptation algorithms have been reported [8,9]. In [8], the algorithm uses the current buffer occupancy in the steady state directly to select the appropriate video rate and does not use other metrics to adapt the rate. Although described as a buffer-based algorithm, the algorithm completely relies on the network to adapt the rate and decide the bitrate during the startup phase, owing to the lack of buffer occupancy. Zhou et al. [9] presented a proportional derivative control approach for controlling the buffer size and adapting the proper bitrate. Client buffer occupancy-based rate adaptation may not efficiently utilize the available bandwidth, as the buffer consumption also depends on the end-device capabilities. Client processing constraints may increase or decrease the media buffering time, and lead to the selection of a substandard or oversubscribed bitrate.

Rate-adaptation decisions based on the media segment size were presented in [10] and [11]. The segment-aware rate-adaptation algorithm proposed by Juluri et al. [10] adapts to the proper rate by predicting when to start downloading the next segment using segment-size variation, along with the bandwidth availability and the current buffer occupancy. These researchers proposed an algorithm based on the buffer level, with the bitrate-representation selection occurring in a single step or multiple steps to match safe playback and avoid re-buffering events. The algorithm proposed in [11] determines the appropriate segment duration to enable precise rate adaptation. Fetching smaller media segments can improve the rate