Playback State Estimation of Progressive Download-based Video Services

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Abstract

The recent proliferation of smartphones and tablet terminals has resulted in the rapid development of video applications. However, in wireless access lines such as those for LTE (Long Term Evolution) and 3G (third-generation) services, packet loss rates and variations in the delay are larger than those in fixed access lines. As a result, video services implemented in TCP (transmission control protocol), known as progressive download (PDL)-based video services, are especially pervasive on wireless access lines.

In this article, we introduce a method of playback state estimation of PDL-based video services. This method makes it possible to accurately estimate the playback state, which will improve the efficiency of monitoring the quality of video services in network and service operation processes.

1. Introduction

The use of smartphones and tablet terminals has increased substantially recently, and with this increase, more and more applications are now being provided via the Internet. In particular, video applications are remarkably pervasive; these are technically divided into three categories according to their delivery mechanism: progressive download (PDL)-based video, download type video, and real-time streaming. The first two are implemented in TCP (transmission control protocol) and the third in UDP (user datagram protocol). PDL-based video services account for about 40% of the volume of domestic Internet traffic. Representative PDL-based video services such as *YouTube* and *Hulu* are targeted to users of personal computers, smartphones, and tablet terminals that access the services over the Internet. In addition, some broadcasters also provide PDL-based video services that enable users to access their news and archived content. NTT Plala started *Hikari TV* (television) *dokodemo/mobile* in 2011, NTT WEST Corporation provides the network platform for *SKY PerfecTV! on Demand*, and NTT DOCOMO offers *BeeTV* and the *d-market* video store. The NTT Group believes that as a network and service provider, one of its most important tasks is improving the quality of these services.

The reasons for this are as follows. First, for network providers, PDL-based video services are considered to be the benchmark of the network. Low-quality PDL-based video services will directly lead to a decrease in customer satisfaction. Second, for a service provider, attractive content and reasonable prices are of course important, but so is the quality of the service itself. Nevertheless, there are no methods for either continuously monitoring the service quality or comparing the service quality with other competitive services. Thus, we have developed a method for addressing these issues for the network and service providers in the NTT Group.

2. Objectives of proposed method

The following two issues are considered to be the main objectives of the proposed method:
(1)   Continuously monitoring the service quality;
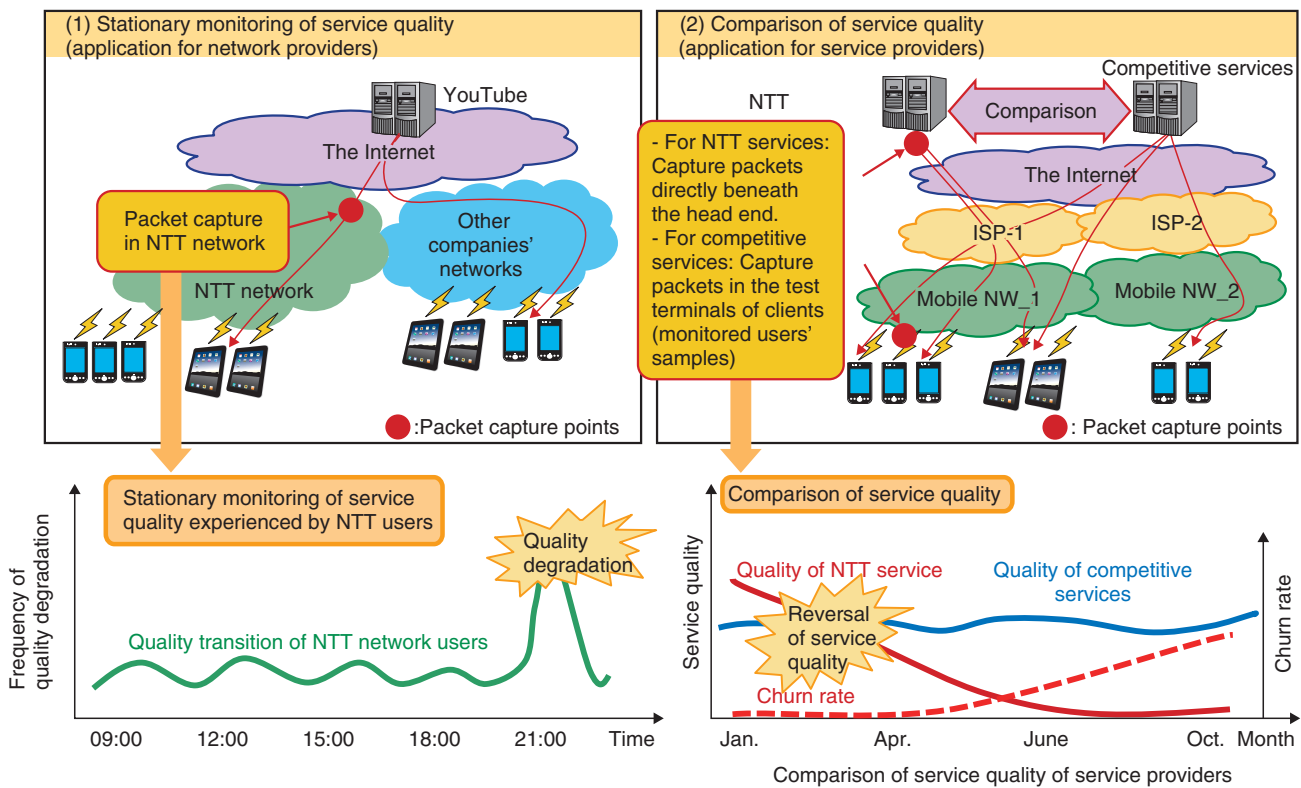(2)   Comparing the service quality with other

Fig. 1.   Possible applications of proposed method.

competitive services.

The first item refers to efficiently enhancing service quality monitoring in the network operation processes, which will enable earlier detection and resolution of quality degradation incidents. The latter item implies that information on the actual service quality is provided, and this information is useful for making decisions on service strategies (**Fig. 1**).

3.   Principal factor affecting the quality of PDL-based video services

In this section, we discuss the main factor affecting the quality of PDL-based video services (**Fig. 2**). In real-time streaming applications implemented in UDP, packet losses directly result in block noise of video images, or they halt the playback. When forward error correction (FEC)*1 is applied, a packet loss that exceeds the redundancy of the FEC configu-

ration leads to the block noise or to the halt of playback.

In PDL-based video, on the other hand, the lost packet will be retransmitted by the server thanks to the functionality of TCP. If the variation in the packet arrival interval caused by packet losses or jitter exceeds a certain level, the video playback will halt due to buffer starvation. Consequently, playback halt is the principal reason for quality degradation of PDL-based video services.

4.   Proposed estimation method

4.1   Technical details

The explanation in the previous section indicates that it is most important to accurately estimate the playback state for PDL-based video applications. The proposed method makes it possible to estimate the playback state, especially the total halt duration and the number of playback halt events, by using packet capture data. While existing methods such as YouTube API (application programming interface) [1] are applicable only to specific services, the proposed method is basically applicable to arbitrary PDL-based

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*1   FEC: When data are lost due to bit errors or packet losses, the FEC mechanism enables detection and correction at the client side by using the redundant data included in packets.

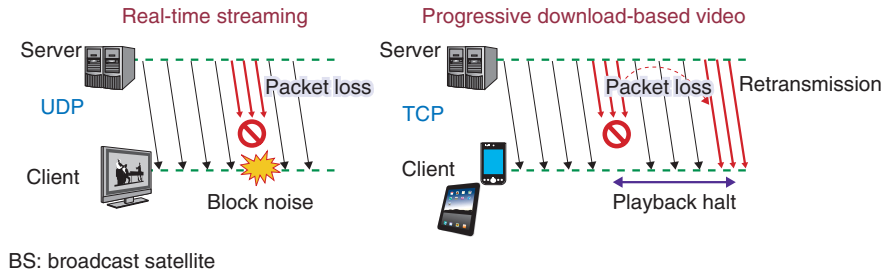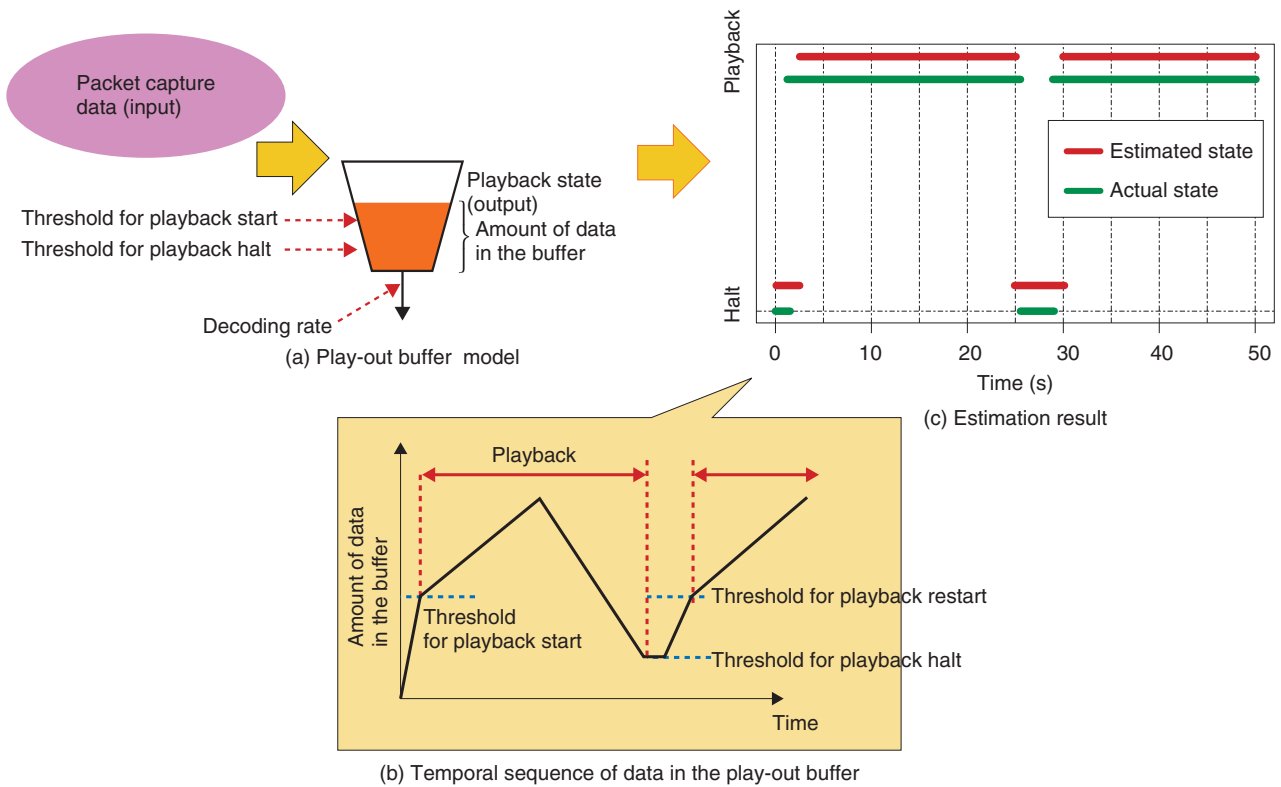|	Examples of quality degradation events	Representative services
Real-time streaming	- Video quality degradation - Block noise - Frame skip	Hikari TV/BS retransmission, etc.
Progressive download-based video	- Longer initial buffering time - Playback halt due to rebuffering	Hikari TV dokodemo/mobile, Hulu, BeeTV

Fig. 2. Factors affecting service quality.

Fig. 3. Input and output of proposed method.

video services. Furthermore, it does not require any additional implementations to either applications or terminals, and is achieved with only passive monitoring. These are the main advantages of the proposed method.

We discuss here the technical details of the proposed method [2]–[4], which is depicted in **Fig. 3**. Before the estimation process is carried out, we first estimate

parameters of the play-out buffer (Fig. 3(a)) such as the decoding rate, the thresholds of the playback start, and the halt and restart. This process is executed in advance of the estimation process. These parameters vary because of the different operating systems in client terminals, applications, and video resolution levels, so we have to estimate them through careful observation of the playback state and the corresponding packet capture data.

In the estimation process, we first extract the time sequence of the downloaded data volume in each session by using the TCP header in the packet capture data. Next, we calculate the amount of temporal data in the play-out buffer (Fig. 3(b)). The playback state at each measurement time, which is the final output of this method, is determined by the amount of data in the buffer at each measurement time and the thresholds estimated beforehand (Fig. 3(c)). In some cases, additional analysis of the payload information can enhance the accuracy of the playback state estimation.
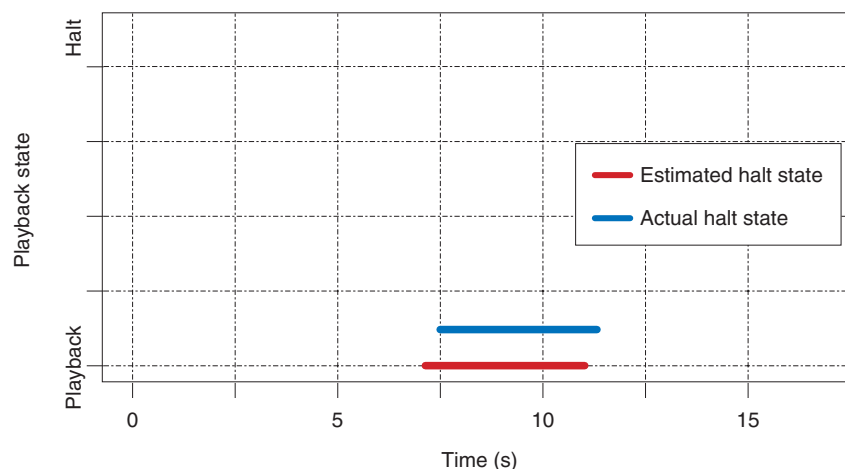
4.2 Verification

We verified the effectiveness of the proposed method by applying it to the PDL-based video services listed in **Table 1**. An example of the playback state estimation is shown in **Fig. 4**. We evaluated the accuracy of the estimation by calculating the matching rate[*2], which was about 92% in this case.

5. Future work

We plan to further enhance the accuracy of estimation by considering the variable decoding rate if the adaptive video quality transitions take place. We will also investigate the detection of the pause state initiated by users and the estimation of the pause state duration [4]. We will continue our research and development of methods to estimate the quality of various application services.

Table 1.   List of verified services.

Verified services
YouTube
Hikari TVdokodemo/mobile
SKY PerfecTV! on demand
d-market (BeeTV)
Hulu

References

[1]   YouTube API. http://www.youtube.com/dev.
[2]   D. Ikegami, H. Honda, H. Yamamoto, H. Nojiri, and A. Takahashi, "Playing State Estimation Method for Progressive Download Services," IEICE-CQ2011-59, pp. 91–96, 2011 (in Japanese).
[3]   H. Honda, D. Ikegami, and H. Yamamoto, "Estimating Video Playback Quality by Observing TCP ACK Packets," Proc. of the IEICE General Conf, B-11-25, Okayama, Japan, 2012 (in Japanese).
[4]   H. Honda, D. Ikegami, H. Yamamoto, H. Nojiri, and A. Takahashi, "Playback and Pause State Estimation of Progressive Download-based Video Services," IEICE-CQ2012-30, pp. 77–81, 2012 (in Japanese).

*2   Matching rate: In this article, we define the matching rate as the total time when the estimated and actual playback states are consistently divided by the total time of the video playback.

Fig. 4.   Example of estimation result.

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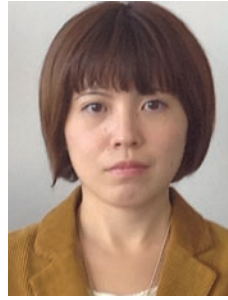
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