Quality Estimation Technique for Video Streaming Services

Kazuhisa Yamagishi and Akira Takahashi

Abstract

The use of adaptive bitrate video streaming services over the network is increasing. The quality of these services is affected by video resolution, audio and video bitrates, bitrate adaptation, stalling due to the lack of playout buffer, and content length. Therefore, service providers need to monitor quality in real time in order to verify that their services are functioning normally. A model that can be used to estimate quality is necessary in order to accurately monitor quality. This article introduces a quality estimation technique that assesses user quality of experience of audiovisual content in adaptive bitrate streaming services.

Keywords: adaptive bitrate streaming, quality of experience, quality estimation

1. Introduction

The use of adaptive bitrate streaming over the network has recently been increasing. HTTP (Hypertext Transfer Protocol)-based adaptive bitrate streaming provides users with the best possible quality of experience for various network conditions because the client application can adaptively select a media file with a suitable bitrate.

Streaming quality degrades due to compression and network conditions (e.g., packet loss, insufficient bandwidth, delay, and jitter). When a reduction in throughput occurs, the quality level that is best suited under current network conditions can be selected (known as *adaptation*) because there are several files (i.e., chunks/segments) corresponding to representations of different bitrates on the server. The throughput is reduced, and packet delay and jitter are introduced due to network congestion. As a result, the playout buffer slowly fills or depletes. When the buffer is empty, the playback of the audiovisual content is interrupted until sufficient data for playback are received (**Fig. 1**). Therefore, it is important to monitor the normality of streaming services.

2. ITU-T SG12

The quality of adaptive bitrate video streaming is affected by compression degradation, adaptation, and stalling events. Therefore, it is important to monitor quality at the client application as well as the quality of servers and networks.

In-service quality monitoring is an important application for quality estimation. An example of this is to detect the locations where quality has degraded by gathering a large amount of quality data through cloudsourcing. Also, even when users do not notice degradation due to compression or network congestion, the quality estimation technique can detect small amounts of degradation. Therefore, service providers can quickly improve their services by using the detected information.

The International Telecommunication Union - Telecommunication Standardization Sector Study Group 12 (ITU-T SG12) has been studying a quality estimation model for adaptive bitrate video streaming and will standardize it in 2017. This model is called the parametric non-intrusive assessment of TCP-based multimedia streaming quality (P.NATS) (**Fig. 2**).

The P.NATS model consists of four modules:

1) Parameter extraction module

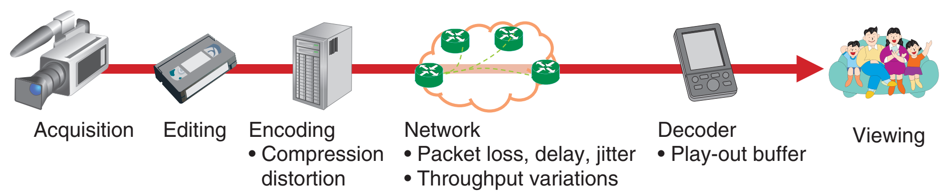

Fig. 1. Quality degradation factors.

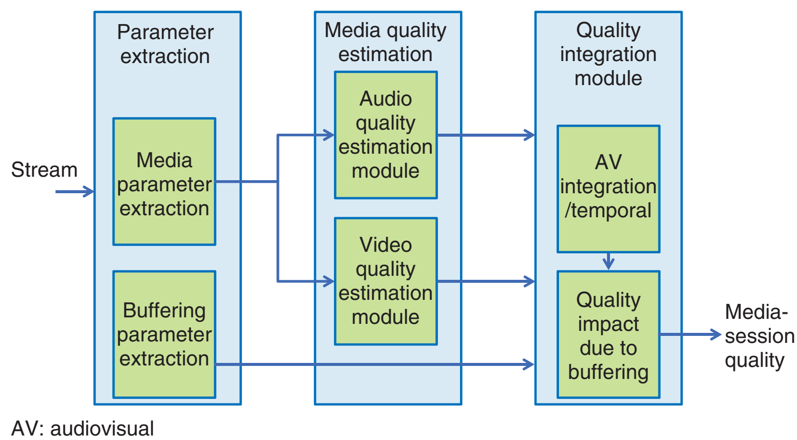

Fig. 2. Recommendation P.NATS model proposed by ITU-T.

This module receives and acquires information about the audio and video bitrates, resolution, and framerate per chunk, measures the rebuffering timing and duration, and calculates stall-related parameters using rebuffering information.

2) Audio quality estimation module

This module estimates audio quality per 1-second sampling interval using the audio bitrate.

3) Video quality estimation module

This module estimates video quality per 1-second sampling interval using the video resolution, framerate, and bitrate.

4) Quality integration module

This module takes audio and video quality and stalling parameters as input and estimates final media-session quality.

Accurate estimation of the quality of adaptive bitrate video streaming requires accurate estimation of audio and video quality. Moreover, quality adaptation due to throughput variation and stalling events due to the lack of a playout buffer need to be taken into account.

To verify the accuracy of the quality estimation models, ITU-T SG12 analyzed the results of 30 subjective quality assessment tests submitted by seven organizations from seven countries. In the subjective tests, audiovisual content was encoded using Advanced Audio Coding - Low Complexity (AAC-LC) [1] (bitrate: 24–196 kbit/s) and H.264/Advanced Video Coding (AVC) [2] (bitrate: 75 kbit/s–12.5 Mbit/s; pixel resolution: 426 × 240–1920 × 1080; framerate: 7.5–30 fps), and various types of stalling events were added. The models proposed by DT/T-Labs, Ericsson, Huawei, NetScout, NTT, Opticom, and SwissQual were equal in the statistical analysis, and ITU-T SG12 therefore agreed to develop a single model by merging these models.

3. Outlook

This article described the P.NATS model that can be used to estimate the quality of adaptive bitrate video streaming services and that will be standardized in 2017. In the next step, ITU-T SG12 will standardize a P.NATS Phase 2 model that can be used for estimating the quality of 4K-ultrahigh definition content encoded by H.264/AVC, H.265/High Efficiency Video Coding (HEVC), and VP9.

References

[1] ISO/IEC 13818-7: "Information technology -- Generic Coding of Moving Pictures and Associated Audio Information -- Part 7: Advanced Audio Coding (AAC)," Jan. 2006.
[2] ITU-T Recommendation H.264: "Advanced Video Coding for Generic Audiovisual Services," Oct. 2016.

Kazuhisa Yamagishi
Senior Research Engineer, NTT Network Technology Laboratories.

He received a B.E. in electrical engineering from Tokyo University of Science in 2001 and an M.E. and Ph.D. in electronics, information, and communication engineering from Waseda University, Tokyo, in 2003 and 2013. He joined NTT in 2003. He has been engaged in the development of objective quality estimation models for multimedia telecommunications. He has been contributing to ITU-T SG12 since 2006.

Akira Takahashi
General Manager, Planning Department; Project Manager, Telecommunication Traffic & Quality Project, NTT Network Technology Laboratories.

He received a B.S. in mathematics from Hokkaido University in 1988, an M.S. in electrical engineering from California Institute of Technology, USA, in 1993, and a Ph.D. in engineering from University of Tsukuba, Ibaraki, in 2007. He joined NTT in 1988. He has been contributing to ITU-T SG12 on performance, quality of service, and quality of experience since 1994. He was a Vice-Chairman of ITU-T SG12, a Vice-Chairman of Working Party 3 in SG12, and a Co-Rapporteur of Question 13/12 for the 2009–2012 and 2013–2016 study periods.