

Design of Video Streaming Service Quality Control System using Available Bandwidth Information

Hyun Jong Kim, Seong Gon Choi*

College of Electrical & Computer Engineering, Chungbuk National University,

410 Seongbong-ro, Heungdeok-gu, Cheongju Chungbuk, Korea

hjkim78@cbnu.ac.kr, sgchoi@cbnu.ac.kr

Abstract— In this paper, we propose video quality control system(VQCS) which can control video service quality through the monitoring of end-to-end available bandwidth for video streaming service like IPTV in NGN convergence network. Various multimedia services such as video, voice and gaming service can be provided by IPTV, and these services require large amounts of bandwidth. At this time, video quality degradation like video jerkiness, block distortion and blurring is caused when network available bandwidth is insufficient. We configure test network to verify justification of the proposed model, and measure video quality using MSU video quality measurement tool. We choose SSIM(structural similarity) score to accurately measure video quality because SSIM is quite similar to MOS(Mean Opinion Score). Since SSIM method can derive experienced video quality through measurement of similarity between original still images and tested still images, we can predict satisfaction of video quality passed through the experimental network using SSIM score.

Keywords— IPTV, QoS, Available Bandwidth, Video Quality, Quality Control

I. INTRODUCTION

Nowadays, NGN(Next Generation Network) has been evolved like type of IP-based wire-wireless convergence network and/or convergence of broadcasting and communication. Preexisting Internet service and variance multimedia service such as IPTV(Internet Protocol TV), VoIP(Voice over IP) and VoD(Video on Demand) are provided in converged network environment.

Like this, many service providers compete with each other in NGN environment, and must satisfy service user's experience quality in order to maintain a subscriber's loyalty. Currently, the satisfaction of users is becoming one of the most import topics concerned by the service and network providers. However, existing methods used for evaluation the experience quality mostly rely on user survey and scores from the user, which are too subjective and need much processing time and cost.

Also, preexisting service quality measurement methods in IP-based network recommended by international standards institutes are not suitable because this method does not reflect service user's satisfaction related with network QoS parameters such as network delay, delay variance and packet

loss. The method to evaluate and manage service user's experience quality is needed in order to solve this problem.

Recently, QoS control architectures recommended by standards institutes provide resource management and admission control only but do not consider service user's experience quality. QoE is the term which is used to describe how it is satisfied by subscribers to the provided service quality. QoE is composed of not only the network performance parameter but also the service quality parameter such as cost, reliability, availability, usability, and fidelity. Although QoE is very subjective in nature, it is very important that a strategy is devised to measure it as realistically as possible. The ability to evaluate QoE will give the provider some sense of the contribution of the network's performance to the overall level of subscriber satisfaction.

So, we propose the system of interworking control between media quality and transport QoS parameters especially available bandwidth to enhance the service user's quality level. Service and network providers can offer the enhanced service quality to users using the proposed VQCS(Video Quality Control System) system. It can adaptively control the data rate of stream according to network available bandwidth.

The remainder of the paper is organized as follows. Section II outlines the QoS control architectures in NGN and The studies of link available bandwidth measurement. Section III introduces the architecture of the proposed system and section IV describes functional entities and overall procedures. Finally, section V presents the conclusion and further study.

II. RELATED WORKS

This chapter describes the QoS control architectures in NGN and the existing available bandwidth measurement methods

A. QoS control architectures in NGN

Researches on the multimedia service quality control in next generation network have been made in various views. Existing video quality control system measures network delay, jitter and packet loss rate using control packet like RTCP. Then, it analyzes the measured QoS parameters, and derives QoS control values to enhance the experienced video quality. In particular, the quality control system was suggested to

enhance service quality by controlling QoS parameters associated with service user's QoE.

However, existing quality control system does not consider the available bandwidth, so it is irrelevant to control the quality of video streaming service. Because a video streaming service requires a large bandwidth, the lack of available bandwidth leads to packet loss. So, the real-time available bandwidth information management is required because packet loss causes serious video quality degradation such as jerkiness, block distortion and blurriness.

B. The studies of link available bandwidth measurement

1) Available bandwidth measurement method using proving packets

The existing available bandwidth measurement method is classified into the single-hop gap model based packet train measurement techniques and the self induced congestion techniques. The representative single-hop gap model[1] has IGI(Initial Gap Model)/PTR(Packet Transmission Rate)[1], and this is the method which measures the available bandwidth by transmitting the packet pairs(or trains) and analyzing the relation of the input dispersion with the changed output dispersion by cross traffic. However, it has long time to measure the available bandwidth due to the time to hang until it finds dispersion that equal the input dispersion and the output dispersion.

It is the technique which observes the moment when the speed of being observed in the receiving side becomes smaller than the speed of the packet which a person transmits while having TOPP[2], a pathChirp[3], Pathload[4], and etc. and the congestion induction technique enhancing the transfer rate of a packet with a queuing and measures the available bandwidth. However, there is the disadvantage that the measurement packet itself decreases the load which there is many in a network and it measures but because of causing the congestion it takes many times.

2) Available bandwidth measurement method using total length field in IP header

The total length field within IP packet header is used in order to measure the per link available bandwidth. As to the available bandwidth, by reckoning the amount of the packets which occupy a link because of being the bandwidth of the link which is not used in the capacity of the physically fixed link, it is have the available bandwidth can measure[5].

Each node extracts the information of the total length from the IP packet imported to router for measurement time and calculates the amount of the packets occupying a link through the cumulative summation of total length values. Based on this data, by taking the occupied bandwidth out of the capacity of the physically fixed link which calculates the occupied bandwidth of each link, and the exact available bandwidth can be measured.

C. Study on the SVC(Scalable Video Coding) considering network environment and terminal performance

Video coding methods to guarantee QoS in integrated network environment has been studied as SVC(Scalable Video Coding) technique. SCV is a standardized video coding technology by JVT(Joint Video Team) which is composed of the ISO/IEC MPEG(Moving Picture Expert Group) and ITU-T VCEG(Video Coding Expert Group). SVC considers image resolution and data rate to control video quality using spatial and temporal scalability. SVC can provide various quality level of video service in the same bit rate and maintain a certain level of image quality by considering network performance and terminal characteristics[12].

However, the network available bandwidth should be measured in real time to maintain video quality in SCV, but the existing methods do not. So, we propose the video quality control system that can maintain a higher quality level by measuring available bandwidth in real-time at terminal, and forwarding this information to scalability extractor.

III. PROPOSED VQCS OPERATING ENVIRONMENT AND FUNCTIONAL ARCHITECTURE

This chapter describes the operation environment and functional architecture of VQCS.

A. VQCS operation environment

Figure 1 shows the operating environment for VQCS that is proposed to control video quality using available bandwidth. VQCS measures available bandwidth in real-time at terminal to consider scalability like bandwidth. Scalability extractor considers various terminal performance and network environment based on the measured available bandwidth.

Scalability extractor can minimize packet loss and delay due to insufficient bandwidth because it selects and forwards appropriate video streaming data rate to network environment.

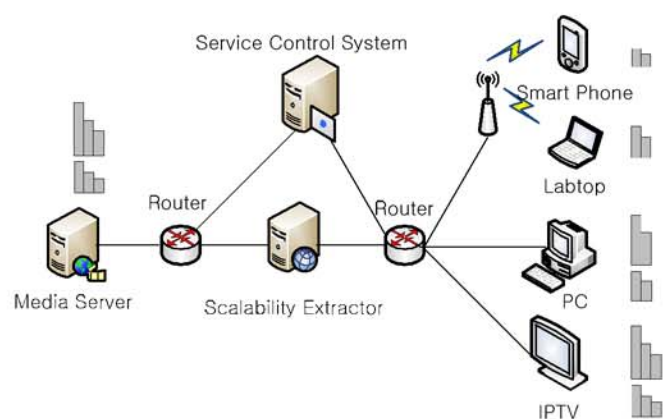


Figure 1. The proposed VQCS operating environment

B. Functional architecture of VQCS

Figure 2 shows the proposed VQCS system architecture which can actively enhance user's experience video quality by controlling stream data rate according to available bandwidth. In order to perform the key function, VQCS is composed of 5 kinds modules, 2 exterior interfaces(Qs and Qe) and 4 interior interfaces (Qa, Qf, Qm and Qr).

Network Interface and Device Interface in packet collection module control the Ethernet card for collecting packet from network circuit. Packet Header Preceding Processor separates service flows according packet IP address or session ID. Control Packets Filtering Interface filters control packets from separated service flows according IP address or session ID. Packet Storage & Forwarding temporarily stores the filtered control packets and forwards them to control packet filtering module.

Control packet filtering module performs the control packet filtering function to measure QoS quality parameters in network layer. Most real-time streaming services use UDP(User Datagram Protocol), RTP(Real-time Transport Protocol), and RTCP(RTP Control Protocol) to transmit service packets in variable network environment. Therefore, the proposed quality control system can derive QoS parameters from the collected information of RTCP packets.

QoS parameter measurement module deduces QoS parameters such network delay, jitter(delay variance) and packet loss from the collected control packets. Here, ICMP(Internet Control Message Protocol) and/or RTCP(Real-time Transport Control Protocol) can be used to deduce QoS parameters.

Available bandwidth measurement module measures available bandwidth within the link by cumulating the occupied packet amount in link during unit time. At this time, the amount of incoming packet per unit time can be calculated by using the information of total length field in the IP header.

Video data rate selection module transfers the information of network resource status to scalability extractor. Scalability extractor transfer the streaming service with lower data rate when the measured available bandwidth is less than the threshold based on the transmitted resource information through external interface Qe.

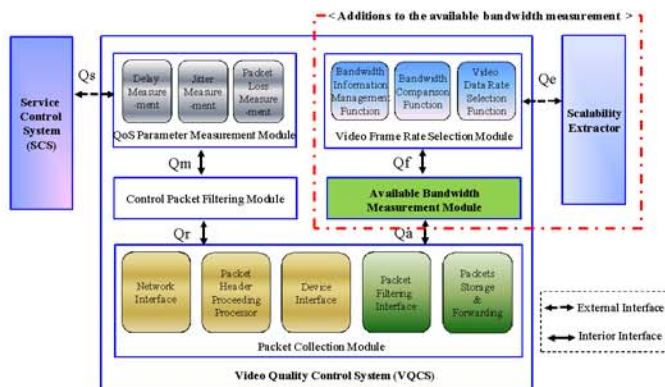


Figure 2. The proposed VQCS system functional architecture

C. VQCS operating procedures

VQCS operates simultaneously with the service request and collects network status information from the service control packets. The collected QoS parameter information is passed through an external interface Qe to the SCS(Service Control System). SCS sets up the packet scheduling policy and the queue threshold within the network based on the collected QoS information and can provide better service environment.

In order to adaptively select data rate of video streaming, VQCS measures available bandwidth at a terminal and then this information is transferred through an external interface Qe to a scalability extractor when multimedia service is requested. Extractor selects the data rate that can reliably transfer video streaming within the measured available bandwidth.

VQCS, as shown in figure 3, begins with a video stream service request. VQCS primarily measures the available bandwidth to control video quality. As noted above, the available bandwidth can be calculated by using the total length field in incoming IP packet header. This available bandwidth is periodically transferred to the scalability extractor.

Extractor selects data rate and frame rate of video stream based on the received available bandwidth(ABW) from VQCS. If there is enough available bandwidth(more 2.5Mbps), media server maintains the initial stream data rate(2.5Mbps). However, when the available bandwidth is less than 2.5Mbps, the media server reduces the stream data rate to 1.5Mbps in order to prevent packet loss due to insufficient bandwidth. In addition, media server reduces the data rate up to at least 0.9Mbps when network fails to accommodate the streaming service with 1.5Mbps data rate.

Media server and extractor notify no service to terminal when the available bandwidth in link is less than minimum data rate(0.9Mbps). Through the process of this series VQCS can stably provide multimedia service by control the amount of video data based on the available bandwidth in link. As a result, VQCS can minimize video quality degradation by minimizing the loss of video stream.

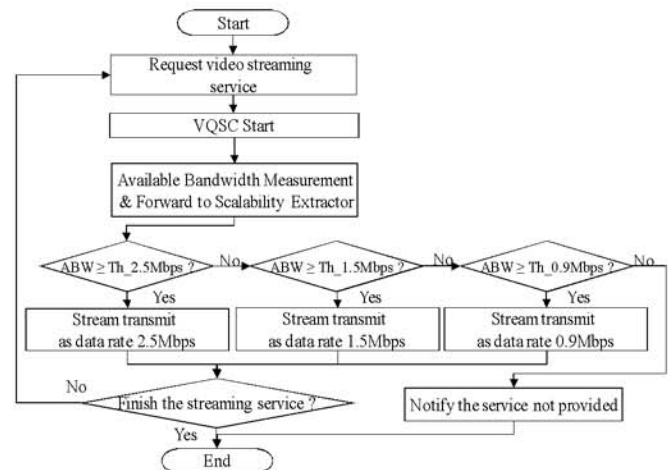


Figure 3. The proposed VQCS flowchart of the detailed operating procedures

IV. PERFORMANCE ANALYSIS

We construct test network as figure 4 to experiment the proposed VQCS performance. This media server in the test network sends a video stream to the client.

We use linux network emulator commands to select video stream data rate according to network resource(available bandwidth) change. We use the following commands to control network delay, jitter, packet loss and bandwidth in router based on linux(kernel 2.6 distribution support).

```
#tc qdisc add dev eth0 root netem delay 100ms
#tc qdisc change dev eth0 root netem delay 100ms 10ms
#tc qdisc change dev eth0 root netem loss 0.1%
#tc qdisc add dev eth0 root handle 10: cbq bandwidth
5Mbit avpkt 1000
```

The above commands generate an end-to-end delay of 100ms, 10ms jitter within 100ms delay, packet loss rate 0.1% and the limited link bandwidth of 5Mbps, respectively.

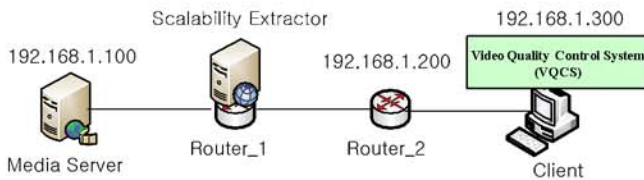


Figure 4. Test network configuration for VQCS performance verification

The experimental network that can transfer the video stream to client is configured to verify the behavior performance of the proposed VQCS. We configure test network like figure 4 to verify justification of the proposed model, and measure video quality using MSU video quality measurement tool[13]. We choose SSIM(structural similarity) score to accurately measure video quality because SSIM is quite similar to MOS(Mean Opinion Score).

Since SSIM method can derive experienced video quality through measurement of similarity between original still images and tested still images, we can predict satisfaction of video quality passed through the experimental network using SSIM score. We use the VLC media player[14] for video streaming and set up de-jitter buffer size to 100ms. We use the VLC media player for video stream delivery in experimental network set to a certain delay, jitter and loss rate. We compare 300 original images and 300 test images passed through various network environments like figure 1 using full-reference method.

Figure 5 shows original video frame sample with data rate 2.5Mbps, transmitted video frame sample with lower data rate 900Kbps using VQCS and video frame sample with packet loss due to insufficient bandwidth. Packet loss by lack of available bandwidth causes block distortion and/or jerkiness of image. However, if media server and extractor reduce the stream data rate by considering network resource, the serious

degradation of video quality can be prevented although image resolution is low.

Video streaming with 2.5Mbps data rate has high packet loss rate in 2Mbps available bandwidth network, so serious degradation occurs as shown figure 5-(c). If video stream with lower 900Kbps data rate is transmitted in same network environment, the quality of video stream service can be improved by preventing packet loss although image resolution is decreased to 0.962 of SSIM score as shown figure 5-(b).



(a) Original video frame sample with data rate 2.5Mbps



(b) Transmitted video frame sample with lower data rate 900Kbps using VQCS

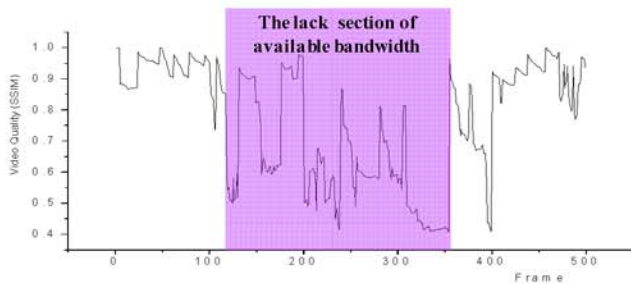


(c) Video frame sample with packet loss due to insufficient bandwidth

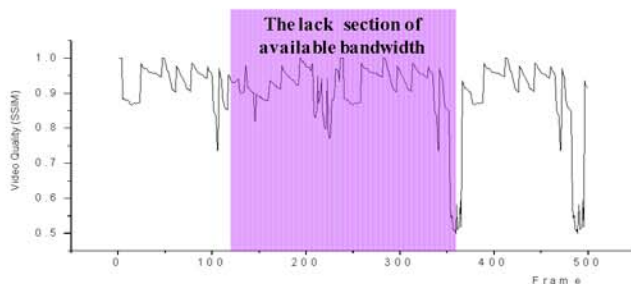
Figure 5. Video frame samples according to network resource and streaming data rate

Figure 6 shows the improved video quality by using the proposed VQCS scheme because VQCS can adaptively control streaming data rate according to network resource status in order to prevent packet loss. We know that the

average SSIM score is improved from 0.768 to 0.906 when using VQCS. Therefore, we can improve the video service quality by controlling stream data rate through real-time available bandwidth monitoring when VQCS is applied at terminal.



(a) Video quality change non-using VQCS



(b) Video quality change using VQCS

Figure 6. Video quality change when selecting adjustive streaming data rate with VQCS

V. CONCLUSIONS

In this paper, we propose video quality control system(VQCS) which can control video service quality through the monitoring of end-to-end available bandwidth for video streaming service like IPTV in NGN convergence network. Various multimedia services such as video, voice and gaming service can be provided by IPTV, and these services require large amounts of bandwidth. At this time, video quality degradation like video jerkiness, block distortion and blurring is caused when network available bandwidth is insufficient. Available bandwidth monitoring method is need to stably control video streaming quality. So, we periodically calculate the amount of the packets in link and measure available bandwidth by using total length field in IP header at terminal. Scalability extractor in network selects suitable video streaming data rate based on available bandwidth and

transports video streaming with adaptive data rate to prevent video quality deterioration. In the future, we will analyze the relationship between quality changes and various QoS parameters associated with video streaming service, and will carry out the research about quality control scheme according to network environment and terminal performance.

ACKNOWLEDGMENT

" This research was supported by Basic Science Research Program through the National Research Foundation of Korea(NRF) funded by the Ministry of Education, Science and Technology (2010-0023980)"

* Correspondence author: Seong Gon Choi (sgchoi@cbnu.ac.kr)

REFERENCES

- [1] Hu, N. and Steenkiste, P., "Evaluation and characterization of available bandwidth Probing Techniques," *IEEE Journal on Selected Areas in Communications*, Vol. 21, no. 6, pp.879-894, 2003.
- [2] Melander, B., Bjorkman, M. and Gunningberg, P., "A New End-to-End Probing and Analysis Method for Estimating Bandwidth Bottlenecks", *IEEE Global Internet Symposium*, 2000.
- [3] Ribeiro, V., Riedi, R., Baraniuk, R., Navratil, J. and Cottrell, L., "pathChirp: Efficient Available Bandwidth Estimation for Network Paths," *Passive and Active Measurements (PAM) workshop*, 2003.
- [4] Jain, M. and Dovrolis, C., "End-to-End Available Bandwidth: Measurement Methodology, Dynamics, and Relation with TCP Throughput," *ACM SIGCOMM*, pp.295-308, 2002.
- [5] H. J. Kim, S. G. Choi, Y. R. Lee, S. M. Kang, and B. I. Choi, "An available bandwidth measurement method using IP header for BeN," *ITC-CSCC 2007*, Jul. 2007.
- [6] K. J. Kim, W. W. Shin, D. K. Min, H. J. Kim, J. S. Yoo, H. M. Lim, S. H. Lee, and Y. K. Jeong, "Analysis of key features in IPTV service quality model," *IEEM2008*, Vol.1, pp.595-598, Dec. 2008.
- [7] H. Schulzrinne, S. Casner, R. Frederic, and V. Jacobson, "RTP: A Transport Protocol for Real-Time Applications," *IETF RFC3550*, Jul. 2003.
- [8] N. Hu, and P. Steenkiste, "Evaluation and characterization of available bandwidth Probing Techniques," *IEEE journal on Selected Areas in Communications*, Vol.21, No.6, pp.879-894, 2003.
- [9] B. Melander, M. Bjorkman, and P. Gunningberg, "A New End-to-End Probing and Analysis Method for Estimating Bandwidth Bottlenecks," *IEEE Global Internet Symposium*, 2000.
- [10] V. Ribeiro, R. Riedi, R. R. Baraniuk, J. Navratil, and L. Cottrell, "pathChirp: Efficient Available Bandwidth Estimation for Network Paths," *passive and Active Measurements(PAM) workshop*, 2003.
- [11] M. Jain and C. Dovrolis, "End-to-End available bandwidth: measurement methodology, dynamics, and relation with TCP throughput," *ACM SIGCOMM*, pp.295-308, 2002.
- [12] Heiko Schwarz, Detlev Marpe, "Overview of the Scalable Video Coding Extension of the H.264/AVC Standard," *IEEE Trans. Circuits and Systems for Video Technology*, Vol. 17, No. 9, Sep. 2007.
- [13] http://compression.ru/video/quality_measure/ideo_measurement_tool
- [14] <http://www.videolan.org>