SDN-based time-domain error correction for in-network video QoE estimation in wireless networks

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Abstract Our previous study proposed a channel utilization method in Software-Defined Networking (SDN) enabled multi-channel wireless mesh network (SD-WMN), which utilizes all of channel resources efficiently. However, when different types of applications are transferred together, their QoE cannot be maintained because of differences in important factors affecting QoE among these applications. Therefore, in order to handle application flows more efficiently based on QoE, this paper focuses on QoE estimation for every ongoing flows through SD-WMN. Since some parameters required for QoE calculation cannot be obtained from OpenFlow, we estimate QoE based on not only the results from SDN-based measurement but also the estimated values of parameters. Finally, we showed that our proposed method is effective for video QoE estimation, especially in a case where there is no packet loss.

1 Introduction

Efficient resource utilization is one of the important problems in wireless networks. We have been tackling it on a SDN-enabled wireless mesh network (SD-WMN) while maximizing the total throughput [1]. However, it does not always result in the improvement of application performance. Because an application performance, i.e., Quality of Experience (QoE), consists of several factors, throughput may not be important for QoE in some applications. Especially, when multiple different applications coexist on network, their QoE cannot be maintained because of differences in important factors affecting QoE among them. Since diverse applications are increas-

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ingly appearing in the Internet, we have to provide high QoE to them by managing the resource utilization efficiently.

Toward efficient resource utilization considering QoE of all flows, we propose a QoE estimation method for ongoing video streaming on the SD-WMN. Although QoE can be easily identified by an offline analysis at an end host, we need to calculate QoE by an online and inside network, while transmitting the flow. However, since some parameters required for QoE calculation cannot be obtained only by OpenFlow, we also propose an online in-network QoE estimation for a video streaming flow by exploiting both OpenFlow-based measurement and parameter estimation to achieve QoE-driven resource utilization.

2 Related work

QoE-driven network management focusing on video streaming is already studied [2], [3], [4]. However, the most of them propose a way to control network on the assumption that QoE is given precisely, and thus a way to measure QoE should be addressed. Reference [5] measures QoE by a measurement agent, which is assumed to handle a receiving video flow in the same way with an end host. However, an intermediate node cannot collect all information of video flow such like an end host. Therefore, this paper focuses on a QoE estimation method that only uses information measured or estimated inside a SDN-enabled network.

From the aspect of SDN-based measurement, reference [6] conducts a delay measurement. This could be useful if a delay is an important factor for QoE calculation. On the other hand, there is a case that a target application may not focus on a delay in QoE calculation. Thus, we focus on QoE estimation, particularly video QoE estimation, based on the measurement and estimation of network performance parameters.

3 QoE calculation model for video streaming services

We employ ITU-T G.1071[7] to calculate the QoE of video streaming services. Section 3.1 provides the brief description of G.1071 and Section 3.2 conducts theoretical analysis to clarify the important factors on QoE calculation.

3.1 G.1071-based QoE calculation

QoE calculation for a video streaming services is standardized in only ITU-T G.1071 [7]. G.1071 requires several parameters including network quality and video parameters to calculate QoE. Although QoE value is calculated based on both video part and audio part, we focus only on the video part in this study because it is a primary factor of video streaming application. Note that calculated QoE value is ranged from 1 to 5.

G.1071 covers two categories in terms of video resolutions: higher resolution and low resolution, as shown in Table 1. We here describe the QoE calculation model for only higher resolution video due to the lack of space. Those formulas are as follows:

Table 1 The target video settings in G.1071

Category	Lower resolution	Higher resolution
Protocol	RTSP over RTP	MPEG2-TS over RTP
Video codec	H.264, MPEG-4	H.264
Resolution	QCIF(176×114), HVGA(480×320)	SD(720×480), HD(1280×720, 1920×1080)
Video bitrate(bps)	QCIF:32～1000 k, HVGA:192～6000 k	SD:0.5～9 M, HD:0.5～30 M
Video framerate	5～30 fps	25～60 fps

$$\text{QoE value} = 1.05 + 0.385 \times Q_V + Q_V(Q_V - 60)(100 - Q_V) \times 7.0 \times 10^{-6}, \quad (1)$$

$$Q_V = 100 - Q_{codV} - Q_{traV}, \quad (2)$$

$$Q_{codV} = A \times e^{B \times b_p} + C + (D \times e^{E \times b_p} + F) + G, \quad (3)$$

$$b_p = \frac{b_r \times 10^6}{r \times f_r}, \quad (4)$$

$$Q_{traV} = H \times \log(I \times p_{lc} + 1), \quad (5)$$

$$p_{lc} = J \times \exp[K \times (L - M) \times \frac{p_{lr}}{M \times (N \times p_{lb} + O) + p_{lr}}] - J. \quad (6)$$

The range of Q_V is from 1 to 100, and Q_V is directly converted to the QoE value by Eq. 1. Parameters of A, B, C, D, E, F, G, H, I, J, K, L, M, N, and O are fixed values defined in G.1071 and take positive value except B and E. Besides, parameters of video bitrate b_r [bps], resolution r [pixel], frame rate f_r [fps], and packet loss concealment (PLC) are pre-determined as the application settings, whereas parameters of packet loss rate p_{lr} [%] and average number of consecutive packet losses p_{lb} are needed to be measured in a reactive manner. The values of fixed parameters in Eq. (6) (i.e., J, K, L, M, N, and O) are determined in accordance with PLC. PLC is one of the technologies in application layer, which corrects a damaged video frame due to packet losses. PLC is classified into Freezing method just ignoring packet losses and Slicing trying to correct packet losses. Since slicing divides a video frame into multiple slices, the correction capability highly depends on the divided number of slices.

3.2 Theoretical analysis on QoE calculation

In this section, we investigate the impact of every parameters on QoE. Note that we selectively describe the results of important parameters due to the space limitation. For this purpose, we assume a SD video with the video bitrate of 2.5 Mbps and PLC of Slicing with 1 slice/frame. Also, as a basis of network condition, we use the packet loss rate of 0.1 % and the average number of consecutive packet losses of 1 as the default settings.

Table 2 Impact of packet loss rate on QoE

Packet loss rate [%]	QoE
0	4.654314
0.01	3.333041
0.1	1.874826
1	1.390250

Table 3 Impact of average number of consecutive packet losses on QoE

Average number of consecutive packet losses	QoE
1	1.874826
10	3.110412
100	4.281420

Table 4 Impact of PLC method on QoE

PLC	QoE
Freezing	2.454080
Slicing with 1 slice/frame	1.874826
Slicing with $>$ 1 slice/frame	2.531622

Table 2 shows how the QoE values changes with the increase in the packet loss rates. From this table, we can see that QoE drastically drops when the packet rate is more than 0.1 %. Thus, we can find that packet loss rate is a key factor on QoE for video streaming application. Table 3 and 4 show how the change in the average number of consecutive packet losses and PLC method impacts on the QoE, respectively. Surprisingly, as the average number of consecutive packet losses becomes larger, the QoE is improved. The increase in the average number of consecutive packet losses means the increase in the number of packets dropped at one. In this case, if the packet loss rate is fixed, the frequency of packet loss event becomes low. That is why the calculated QoE is improved. In short, there is trade-off relationship between packet loss rate and the average number of consecutive packet losses. Regarding to Table 4, we can see that the minimum QoE is brought by Slicing with 1 slice per frame due to the feature of video technology.

In summary, we can remark that packet loss rate, the average number of consecutive packet losses, and the PLC method significantly affect QoE value, and thus we need to obtain these parameters for QoE calculation. However, the latter two parameters cannot be measured in the network because tracking every packets to count consecutive packet losses is quite hard in OpenFlow, and PLC that is an application parameter, which cannot be identified in network. Therefore, we directly measure the packet loss rate, whereas estimate other two parameters.

4 OpenFlow-based in-network QoE estimation

In this section, we propose a QoE estimation method based on the information obtained by OpenFlow. We call this method OpenFlow-based Estimation method (OFE method). OFE method consists of two functions: (1) packet loss rate measurement and (2) video settings estimation.

4.1 OpenFlow-based packet loss rate measurement

Although exact packet loss rate (PLR) can be measured only at both end hosts, we try to indirectly measure it based on the information collected in SD-WMN. For this measurement, we use statistic information of each flow (FlowStats), which is collected from APs by a controller (i.e., OFC) on the request basis. Note that in this study, we define a flow as a pair of IP address and port number of source node and destination node. Since FlowStats includes the cumulative number of transmitted packets, the difference between two FlowStats collected in a certain interval are used as the number of transmitted packets. Then, we treat the difference of the number of transmitted packets between two APs (the first AP where a flow enters the SD-WMN (called sender-side AP) and the last AP where it exits (called receiver-side AP)) as the number of packet losses for the flow, thereby calculating PLR based on these values.

However, certain amount of measurement errors cannot be avoided in this simple PLR measurement method because OpenFlow cannot completely synchronize the transmission timing of FlowStats (Figure 1 (i)). That is, since OFC receives a statistic information of the point of when a request arrived at an AP, the condition of on-the-fly packets and/or buffered packets between two APs at the point of FlowStats arrival is different, thereby resulting in the measurement errors. Even a few errors on the number of packet losses significantly affect QoE as discussed in Section 3.2.

To solve this issue, we have to correct such kind of errors. Specifically, there are two cases leading to measurement error: the number of transmitted packets on a receiver-side AP is larger than that on a sender-side AP, and vice versa. In the first case, measurement error occurs when a FlowStats request arrives at a receiver-side AP relatively earlier than that to a sender-side AP. Although in most of this kind of case, there is no packet loss, this error causes subsequent measurement errors. Therefore, we hold the difference between the number of transmitted packets sender-side and receiver-side APs as the accumulated surplus packets for correcting subsequent errors (Figure 1 (ii)). In the second case, we expect two possibilities: actual packet losses or errors caused by the timing fault (as in the former case). Since a timing fault frequently happens, we try to correct the errors by taking surplus packets on the receiver-side AP in the first case into account at the subsequent measurements. Specifically, the number of accumulated surplus packets is added to the number of the transmitted packets at the next measurement (Figure 1 (iii)).

In this way, packet loss rate is measured by FlowStats that are periodically transmitted to all APs. G.1071 also requires measurement for 8-16 seconds as a period to calculate QoE. Therefore, we employ that FlowStats collection and QoE measurement is conducted at the shortest intervals (8 seconds) because we aim to achieve the QoE-driven network control.

4.2 Parameter estimation

As described in section 3.2, it is difficult for OFC to measure several parameters related to video image such as video bitrate, frame rate and resolution. Therefore,

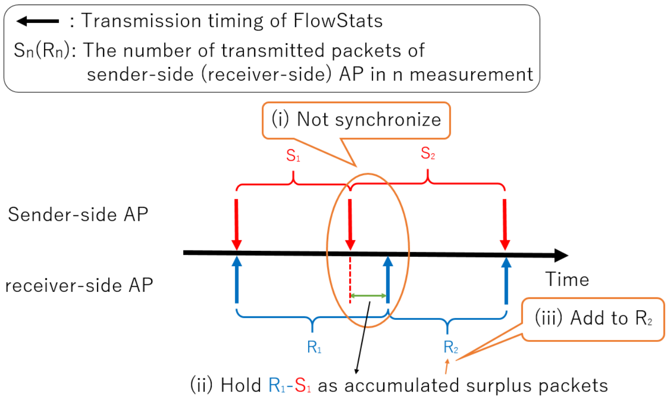

Fig. 1 Image of Error correction procedure.

Table 5 Recommended video bitrate for resolution and frame rate.

Resolution	Frame rate [fps]	video bitrate [kbps]
SD(720×480)	24,25,30	2,500
SD(720×480)	48,50,60	4,000
HD(1280×720)	24,25,30	5,000
HD(1280×720)	48,50,60	7,500
HD(1920×1080)	24,25,30	8,000
HD(1920×1080)	48,50,60	12,000

we try to estimate each of them based on the limited information.

Average number of consecutive packet losses: Because FlowStats includes statistic information only, OFC cannot understand consecutiveness. As analyzed in Table 3, the value of 1 shows the worst QoE, so, we set it to 1 to avoid the QoE overestimation.

Video bit ate: Since OFC cannot directly obtain information of video settings, video bitrate is estimated based on the measured throughput. Specifically, we treat the measured throughput as a video bitrate. Note that the throughput is measured based on the number of transmitted bytes of FlowStats at a sender-side AP.

Frame rate/Resolution: Frame rate and resolution are estimated from the estimated video bitrate. Table 5 shows the recommendation of video settings in terms of resolution, frame rate, and video bitrate. Since Table 5 can be expressed as Table 6, we estimate frame rate and resolution based on video bitrate of Table 6. In Table 6, we choose the maximum frame rate from among candidates in Table 5 to avoid QoE overestimation. Also, we choose the range of video bitrate for each entry so that it can be a median of consecutive entries.

PLC: As discussed in Subsection 3.2, we employ ”slicing with 1 slice per frame”

Table 6 Estimation of resolution and frame rate based on video bitrate.

Video bitrate [kbps]	Resolution	Frame rate [fps]
∼3,250	SD(720×480)	30
3,250∼4,500	SD(720×480)	60
4,500∼6,250	HD(1280×720)	30
6,250∼7,750	HD(1280×720)	60
7,750∼10,000	HD(1920×1080)	30
10,000∼	HD(1920×1080)	60

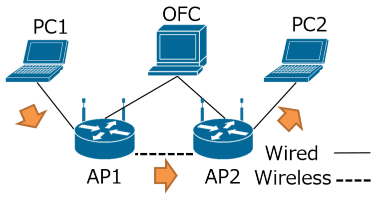

Fig. 2 Experimental environment.

as the PLC.

5 Experimental evaluation

We conduct experiments to evaluate the OFE method in a real wireless environment. The goal of this experiment is to show the effectiveness of the OFE method. We compare the OFE method with a comparative method. Note that the comparative method estimates QoE based on FlowStats like OFE method but does not conduct the error correction for the timing-fault case.

5.1 Experimental settings

The experimental topology is shown in Figure 2. We use IEEE 802.11a with fixed 54 Mbps on 120 channel in the wireless settings. Regarding OpenFlow, we use Trema as OFC and install OpenvSwitch on every APs. Networks between the OFC and APs, and PCs and APs, are made by Ethernet cables in order to avoid packet loss and delayin this section. In our experiment, PC1 transmits a video streaming to PC2 for 60 seconds. The video is made by the H.264 codec with SD (720x480), 30 fps and 2.5 Mbps CBR.

5.2 Effectiveness in no packet loss environment

Figure 3 shows the number of transmitted packets and packet loss rate calculated by each method, and Figure 4 shows the calculated QoE. Measurement errors caused by the timing fault often happen even in no packet loss environment. In the com-

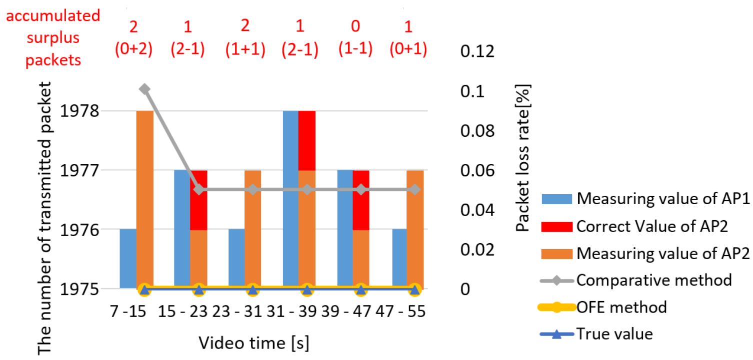

Fig. 3 The number of transmitted packet and packet loss rate in no packet loss environment.

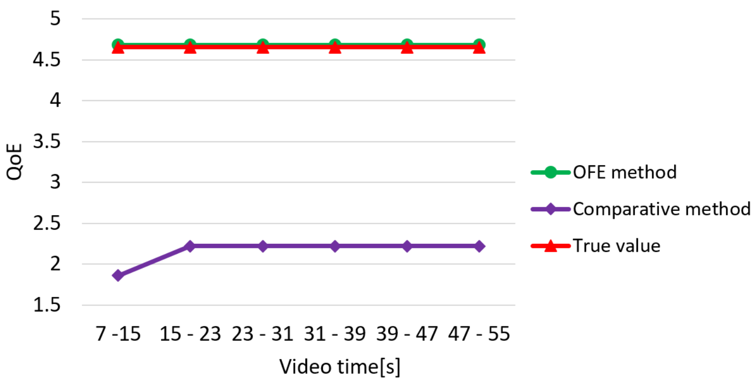

Fig. 4 Measured QoE value in no packet loss environment.

parative method, QoE drop by more than 2 due to those measurement errors arising from the timing fault. On the other hand, measuring errors between 15 and 23, 31 and 39, and 39 and 47 seconds are successfully corrected by taking into account the accumulated surplus packets around 7 and 15, and 23-31 seconds in OFE method. As a result, the occurrence of unnecessary packet losses can be avoided, thereby providing almost same value with the true QoE value, which is measured at end hosts. Therefore, OFE method is effective for estimating packet loss rate and QoE value.

5.3 Effectiveness in packet loss environment.

This section conducts an evaluation in case where packet losses inevitably occurs due to the deterioration of the wireless link quality. As for the environment, we intentionally cause packet losses by increasing the distance between AP1 and AP2 up to 20.5m.

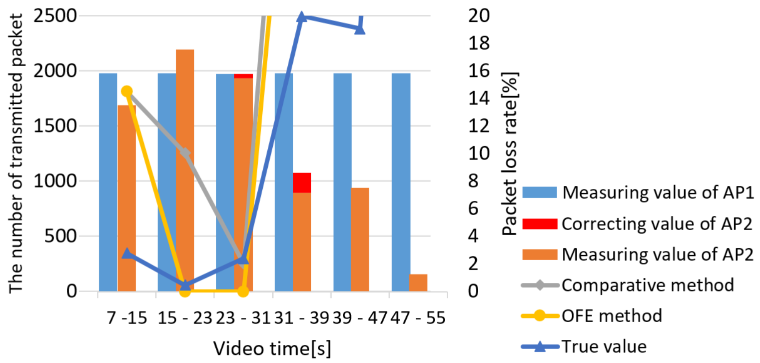

Fig. 5 The number of transmitted packet and packet loss rate in packet loss envrionment

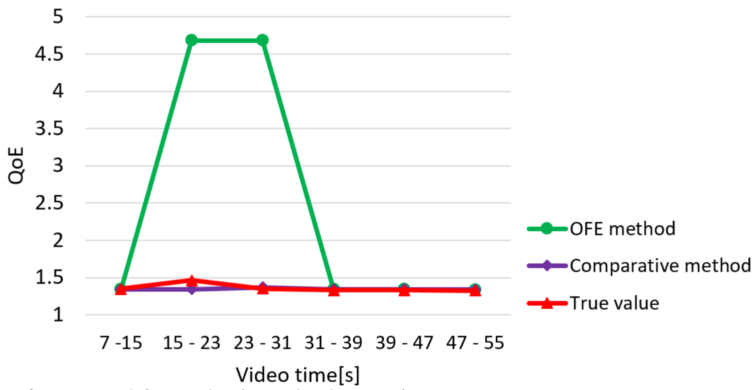

Fig. 6 Measured QoE value in packet loss environment.

Figure 5 and Figure 6 show the results of the estimated packet loss rate and QoE, respectively. In OFE method, packet loss rate between 7 and 15, 31 and 39, 39 and 47, and 47 and 55 seconds becomes clearly larger than the true value because the OFE method treats the increase of the retransmission delay (i.e., buffered packets) as packet losses and thus the number of transmitted packets at the receiver-side AP decreases. However, the estimated QoE is almost same with the true value because overestimated packet loss rate has little effect on QoE.

On the other hand, packet loss rate between 15 and 23, and 23 and 31 seconds are lower than true value in OFE method. This is because packets left in the sender AP's queue in previous periods is transmitted late. In this case, actual packet losses are regarded as no packet loss because the number of transmitted packets of the receiver-side AP become larger than those of the sender-side AP or packet losses are corrected by mistake. Therefore, OFE method still has problems in terms of estimation accuracy in case of packet loss environment. This is because we are exploiting only the information of network layer in this study. If we can obtain the information from wireless layer, the accuracy of QoE estimation may be improved. Therefore, we are going to solve this problem by cooperation with wireless layer.

Through experimental results, we showed OFE method was effective under no packet loss environment. On the other hand, the estimation accuracy of the OFE method was clearly degraded under packet loss environment.

6 Conclusion

In this paper, we presented a QoE estimation with time-domain error correction for video streaming aiming to efficiently conduct QoE-driven resource management. At first, we showed that packet loss rate has a significant effect on the QoE value for video streaming through the theoretical analysis. Then, we proposed OFE method that estimates QoE by exploiting FlowStats information, while correcting the measurement errors of packet loss rate. In our experiments, we remarked that OFE method can estimate QoE precisely in the environment where there is no packet loss. On the other hand, we showed that the estimation accuracy by the OFE method dropped in the environment with packet losses. In response to this result, we are going to cooperate with wireless layer to estimate QoE more accurately.

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