TCP Congestion Control over IEEE 802.11 Wireless Lans based on *K*-Means Clustering Focusing on Congestion Window Size and Round-trip Time

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Abstract: Recent IEEE 802.11 wireless LANs provide high speed data transfer using the newly introduced physical and MAC technologies. Although packet losses over a wireless link are also decreased by the help of new MAC technologies, some packet losses still occur randomly. Those packet losses invoke TCP congestion control, which reduces the TCP level throughput, even if congestion does not occur at al. In order to resolve this problem, some machine learning based approaches have been proposed, which use *K*-means clustering in order to discriminate congestion triggered packet losses and wireless error triggered packet losses. However, those proposals use only delay related parameters, but delay may increase due to non-congestion reasons, in which case the conventional proposals fail discrimination. This paper proposes a method to classify packet losses by the *K*-means clustering focusing on congestion window size and round-trip delay, and to stop decreasing congestion window when losses are triggered by wireless errors. We develop the proposed method as a Linux kernel module and show the performance evaluation results that the throughput increases by 40% without increasing unnecessary packet losses.

1 INTRODUCTION

Recently, the data transfer throughput over IEEE 802.11 WLANs (Wireless LANs) has increased significantly. The recent IEEE standards, such as IEEE 802.11n and 11ac, introduced new PHY (physical) and MAC (medium access control) mechanisms (IEEE, 2016). The PHY mechanisms including new modulation methods, MIMO (multiple-input and multiple-output), and channel bonding realize high data rate, and the MAC mechanisms such as frame aggregation and block acknowledgment (Block Ack) provide not only low protocol overheads but also powerful data retransmission capability.

Most of communications over IEEE 802.11 WLAN, such as web access and e-mail, use TCP (transmission control protocol) as their transport prptocol (IETF, 1981). One of noteworth functions in TCP is the congestion control. When congestion occurs at some nodes within a network, the TCP module in a data sending node decreases its data sending rate. However, since TCP works only at an end node, it cannot detect a precise condition of the node suffering from congestion. So far, dozens of congestion control methods have been proposed (Afanasyev et al., 2010), and most of them consider that, if there are any packet losses, congestion occurs somewhere in a network.

When a WLAN link exists within a path between communicating nodes, the possibility of packet losses will be larger than a path consisting of wired links only, even if the recent IEEE 802.11 standards are used. In such a case, TCP in a sending node considers that congestion occurs and decreases the congestion window size unnecessarily. This is a traditional issue on TCP over wireless links and has been studied actively (Sardar and Sara, 2006). There are many proposals, such as modifying TCP, dividing TCP connections, and support by lower layer protocols. Recently, there are new trends; a machine learning approach, i.e., the discrimination of TCP packet losses by use of machine learning technologies.

Machine learning is a useful method which can be applied to various fields. TCP communication is one of the targets and several studies are proposed. (Nunes et al., 2011) applied the Experts Framework

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technique to RTT (round-trip time) estimation. (Mirza et al., 2010) proposed how to predict TCP throughput using the SVR (support vector regression) technique. (Chung et al., 2017) applied the random decision forests to an MPTCP scheduler that selects a subflow to send data segments by considering performance metrics such as the MAC data rate, signal strength, and network congestion.

As for the packet loss discrimination, (Sooriyabandara et al., 2010) and (Morifuji and Hiraki, 2013) proposed approaches that a data sender infers the cause of packet losses by use of the *K*-means clustering method (Hand et al., 2001). (Deng and Cai, 2009) focused on MANET (mobile ad hoc network) and adopted SVM (support vector machine) to allow a data receiver to differentiate packet losses.

This paper improves the work done by (Sooriyabandara et al., 2010) and (Morifuji and Hiraki, 2013). These two papers focused only delay related parameters such as one way delay and RTT. In this paper, we use congestion window size (cwnd) as well as RTT, and classify two types of packet losses, i.e. *congestion losses* and *wireless losses*, by use of the *K*-means clustering. Moreover, we propose a congestion control method that skips the cwnd decreasing when a packet loss is classified as a wireless loss. The rest of this paper is organized as follows. Section 2 gives some background information including the overview of IEEE 802.11 WLAN and the TCP congestion control, some previous comments on packet losses over WLAN, the overview of *K*-means clustering, and some related work. Section 3 describes the proposed method and Section 4 gives the performance evaluation results. In the end, Section 5 concludes this paper.

2 BACKGROUNDS

2.1 IEEE 802.11 WLAN and TCP Congestion Control

As described above, the recent IEEE 802.11 standards, 11n and 11ac, introduce new MAC mechanisms for high speed and efficient data frame transmission; the frame aggregation and Block Ack. The frame aggregation allows multiple data frames (called MAC protocol data units: MPDUs) to be aggregated and sent together. The whole transmitted frame is called A-MPDU (Aggregation MPDU), and is a collection of A-MPDU subframes, each of which includes an MPDU delimiter, an MPDU body, and a padding, as shown in Figure 1. An MPDU delimiter contains the MPDU length, a cyclic redundancy check (CRC) to detect bit errors within the delimiter itself. A padding consists of 0 through 3 bytes, which make the length of an A-MPDU subframe a multiple of 4 bytes.

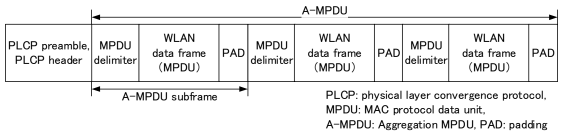


Figure 1: Structure of A-MPDU.

The IEEE 802.11n and 11ac standards adopt an acknowledgment scheme called high throughput (HT)-immediate Block Ack. When a receiver receives an A-MPDU, it replies a Block Ack frame which contains a Block Ack Bitmap parameter indicating whether it correctly receives a MPDU with a specific sequence number. The Bitmap indicates receipt or non-receipt of 64 MPDUs. The data sender retransmits non-received MPDUs according to the Bitmap. When a Block Ack frame itself is lost, the whole A-MPDU is retransmitted by timeout.

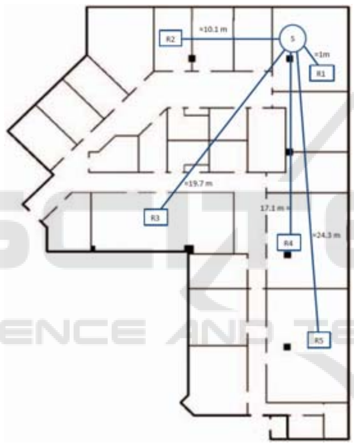
TCP uses cwnd in addition with an advertised window size (awnd) which a data receiver reports in a TCP header. A data sender transmits data segments according to the smaller of cwnd and awnd. TCP Reno / NewReno (Henderson et al., 2012) is a traditional congestion control method, which is still used widely. cwnd is controlled in an AIMD (additive increase and multiplicative decrease) mechanism. When receiving an ACK segment reporting the receipt of new data segments (a new ACK), cwnd is increased by 1/cwnd (segments), and when any data segments are retransmitted in response to three duplicate ACKs (fast retransmit), cwnd is halved. CUBIC (Ha et al, 2008) is a relatively new congestion control method, which is a default in the Linux operating system. cwnd increases in a cubic function of time from the last fast retransmit. At the fast retransmit, cwnd decreases to 70% (80% in the original version) of the cwnd value just before the retransmission.

In those methods, a data sender detects congestion by a packet loss (retransmission). In TCP Vegas (Brakmo and Peterson, 1995), on the other hand, a data sender monitors RTT and estimates the queue length at a bottleneck node. If the estimated queue length is smaller than a threshold, a sender increases cwnd by one segment during one RTT timeframe. If the queue length is larger than another threshold, a sender decreases cwnd by one segment during one RTT timeframe. Otherwise, a sender keeps cwnd as it is. That is, TCP Vegas is a mechanism based on the delay.

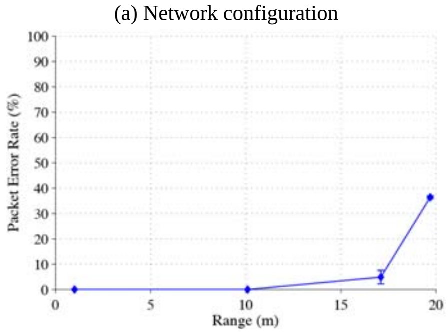
There are some congestion control methods based on packet losses and delay. TCP Veno (Fu and Liew, 2003) and Compound TCP (Tan et al., 2006) are examples. They decrease cwnd at a packet loss, and control the cwnd increase and decrease depending on the congestion status estimated by RTT. If network is congested, they work like TCP Reno, and if not congested, they increase cwnd more aggressively.

2.2 Packet Losses over 802.11 WLAN

As mentioned in the previous subsection, the recent IEEE 802.11 WLANs provide highly reliable data transfer by use of the HT-immediate Block Ack mechanism, compared with the older IEEE standards that used the one-to-one mapping between data and Ack frames. However, a few packet losses occur resulting from the retry-out in the MAC level retransmission.



(a) Network configuration

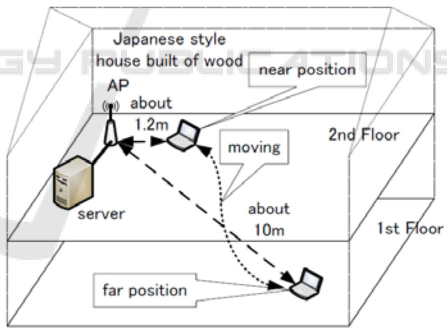


(b) Packet loss rate

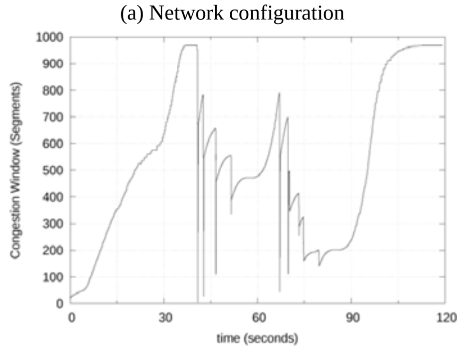
Figure 2: UDP data transfer over 802.11ac WLAN. (Dianu et al., 2014).

Figure 2 shows a result of the performance evaluation of 802.11ac LAN in an indoor environment given in (Dianu et al., 2014). As shown in Figure 2 (a), a sender is located at position S and a receiver is at one of positions R1 through R5. A sender generates 30 second UDP data traffic using iperf. Figure 2 (b) shows the result of packet error ratio in response to the distance between the sender and the receiver. In the case that the distance is 10.1 m (position R2), there are some packet losses although the rate is under 1%, which means that wireless losses happen in an IEEE 802.11ac WLAN.

Figure 3 shows another example of performance evaluation of 802.11n WLAN, which was conducted for evaluating Bufferbloat problem (Nomoto et al, 2014). As shown in Figure 3 (a), a station starts sending data to the server at the far most position for 30 seconds, moves to the nearest position, and stays there for 30 seconds. Then, it moves to the far position again. At the far most position, the station uses 6.5 or 13 Mbps data rate, and uses rate close to 300 Mbps at the nearest position. Figure 3 (b) shows the time variation of cwnd measured at the station. From time 0 to 40 seconds, and 80 to 120 seconds, that is, while the station is moving or in the far most position, cwnd keeps increasing. This means there are no packet losses. On the other hand, while the station stays at the nearest position, there are ten drops in the cwnd graph, each of which corresponds to a packet loss. Since there are no bottlenecks in the data transfer from the station to the server, these losses are considered as wireless losses. This result shows wireless losses happen in an 802.11n WLAN.



(a) Network configuration



(b) cwnd vs. time

Figure 3: TCP data transfer over 802.11n WLAN. (Nomoto et al., 2014).

2.3 *K*-Means Clustering

In this subsection, we explain the *K*-means clustering that we use to classify TCP packet losses. The *K*-means algorithm is a type of unsupervised learning. The goal of this algorithm is to categorize unlabelled data into K groups. Specifically, it minimizes an objective function $\emptyset$;

$$\emptyset = \sum_{i=1}^{K} \sum_{x \in X_i} \|x - \bar{x}_i\|^2. \tag{1}$$

Here, $\{x\}$ is a set of data, X_i is a cluster, i.e. a disjoint subset such that $\bigcup_{i=1}^{K} X_i =$ the set of data, and $\bar{x}_i$ is the cluster center of X_i in a Euclidean distance sense. The algorithm is summarized in the following way.

1. Assign data into K clusters randomly.
2. Calculate the center of each cluster by

$$\bar{x}_i = \frac{1}{|X_i|} \sum_{x \in X_i} x. \tag{2}$$

3. Reassign all data x into a new cluster in a way that a Euclidean distance $\|x - \bar{x}_i\|$ is the minimum for $\bar{x}_i$.
4. Repeat steps 2 and 3 until there are no changes in clusters or the maximum repeat count is exceeded.

Figure 4 shows an example of the *K*-means clustering. A hundred points are selected randomly in the field of (0, 0) through (100, 100). The above algorithm is applied once, twice, three times and eight times. As the repeat count increases, the total of distance between individual points and the corresponding cluster center becomes small.

2.4 Related Work

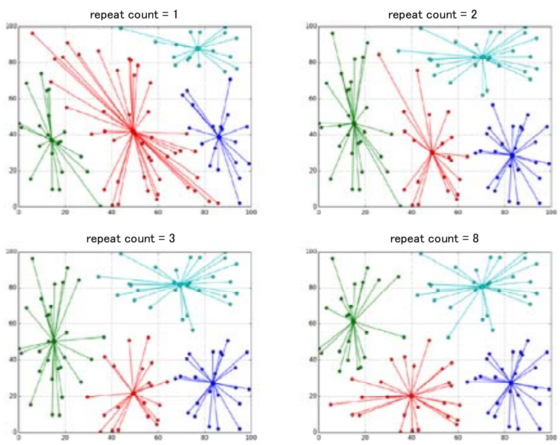


Figure 4: Example of clustering by *K*-means clustering.

In order to classify TCP packet losses as a congestion loss and a wireless loss, (Sooriyabandara et al., 2010) uses two delay based parameters, one way delay (OWD) and inter-arrival time (IAT) of ACK segments, as data for the *K*-means clustering. A TCP data sender keeps a record of OWD and IAT for the three most recent ACKs, and if there is a packet loss, that is, a duplicate ACK is received, a sender classifies this loss event into two groups. A sender determines this loss as a congestion loss or a wireless loss depending on the mean OWD of last three ACKs. If it belongs to a congestion loss category, then a sender follows the standard TCP back-off procedure, and if not, cwnd is not decreased. (Sooriyabandara et al., 2010) shows some performance results by use of the network simulator ns-2 successfully, but OWD is difficult to measure in an actual network.

(Morifuji and Hiraki, 2013), on the other hand, uses RTT for discriminating packet losses. It records RTT for packets and discriminates the type of losses based on RTT records. Only if a packet is classified into the congestion loss, a data sender decreases cwnd. Through a simulation based performance evaluation, they confirmed that this method improves TCP throughput.

3 PROPOSAL

Basically, both of the related work discussed in the previous section use a delay based parameter for the *K*-means clustering. That is, if congestion occurs, OWD or RTT will increase, and so a packet loss with a large delay may be a congestion loss in a high probability. On the other hand, a packet loss with a small delay might be a wireless loss. However, over an IEEE 802.11 WLAN link, delay may change for other reasons. For example, IEEE 802.11 WLAN uses multiple data rates and the dynamic rate switching, and when a station is located far from an access point and the data rate is low, the transmission delay will increase. Besides, in our previous paper (Moriyama et al., 2017), we showed that, if there is an unbalanced traffic load when the multi-user MIMO is used together with the frame aggregation in 802.11ac, transmission delay may increase. Therefore, it is possible that the *K*-means clustering using only delay based parameters may lead to wrong categorization.

In this paper, we focus on cwnd itself together with RTT to apply the *K*-means clustering, because it is considered that the probability of congestion increases when the value of cwnd is large. In the proposed method, cwnd is decreased as in the original

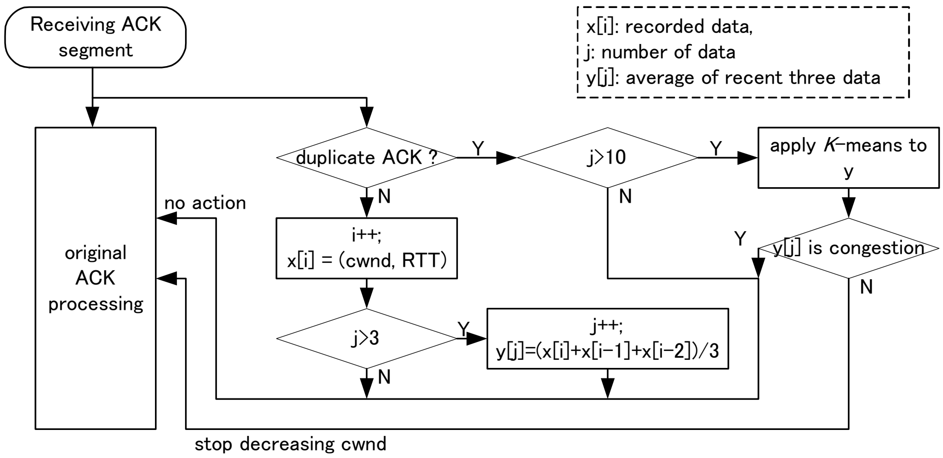


Figure 5: Flow chart of proposed method.

TCP if the data retransmission is classified as a congestion loss, but if classified as a wireless loss, a data sender does not decrease cwnd.

Figure 5 shows detailed algorithm flow of the proposed method. For new ACKs, a sender records (cwnd, RTT) pair, and the average of three of these pairs is maintained as $y[j]$. If a sender receives a duplicate ACK segment, then it applies the newest $y[j]$ to the K-means clustering. If $y[j]$ is categorized as a wireless loss, then the cwnd decreasing is stopped in the original ACK processing. Otherwise, the original TCP congestion control is performed.

4 PERFORMANCE EVALUATION

4.1 Experiment Conditions

In order to evaluate the performance of the proposed method, we implemented it over the Linux operating system (Ubuntu 16.04LTS). We also implemented the method described in (Morifuji and Hiraki, 2013), which uses only RTT for the K-means clustering (we call this method *conventional method*). The maximum number of data used in the K-means clustering is 10,000 and the maximum repeat count is set to 50,000.

Figure 6 shows the experimental configuration. A server, a data sender, is connected to Gigabit Ethernet, which is connected with an IEEE 802.11ac access point at the other end. There is one station, a data receiver, in this WLAN.

We used two scenarios in the experiment. In scenario 1, 30 msec delay and random packet errors are inserted at the output port in the server. The distance between the access point and the station is about 1 m. The inserted packet loss rate is 0.03%, 0.3%, or 3%. In scenario 2, on the other hand, only 30 msec delay is inserted at the server, and the distance between the access point and the station is about 7 m.

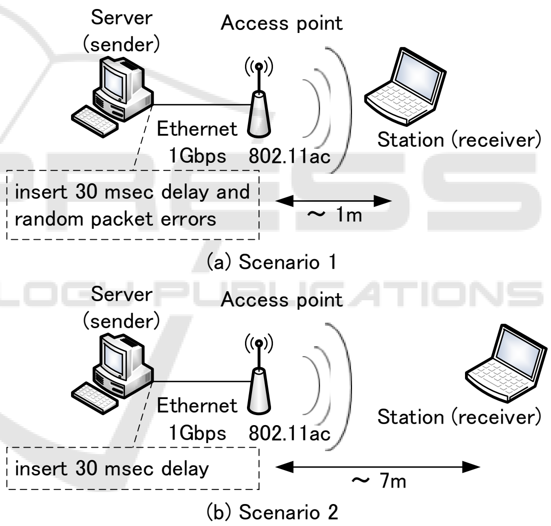


Figure 6: Experimental configuration.

In the experiment, the communication duration is 90 sec. We compare the performance of TCP Reno, the conventional method (Morifuji and Hiraki, 2013), and the proposed method. For each method, we executed twenty-five experiment runs for measuring throughput and number of duplicate ACKs.

4.2 Evaluation Results in Scenario 1

Figure 7 shows the average throughput measured in scenario 1. The graph shows the results of Reno, the conventional method (Conv. Method in the figure), and the proposed method (Prop. Method). Along with the increase of packet error rate inserted artificially, the throughput decreases in all the cases. The throughput is the lowest in the original TCP Reno. The two K-means clustering methods improve the throughput. The conventional method provides better throughput than the proposed method. Figure 8 shows the number of duplicate ACKs during one experimental run. This has a similar trend with the average throughput. TCP Reno is the smallest and the conventional method is the largest.

Those results mean that the conventional method is the most aggressive, that is, the conventional method handles large number of packet losses as wireless losses and does not decrease cwnd. As a result, the average throughput becomes high, and consequently the number of packets sent increases, which increases the packet losses again.

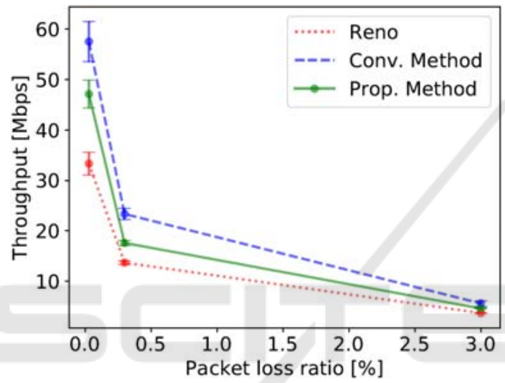


Figure 7: Throughput in scenario 1.

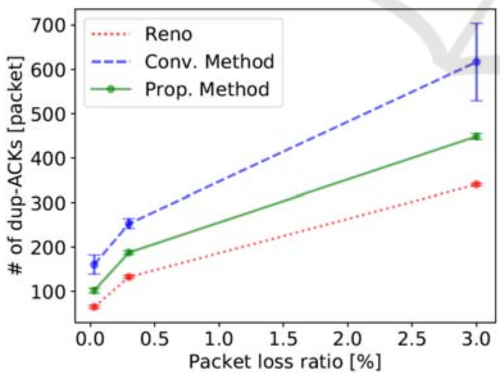


Figure 8: Number of duplicate ACKs in scenario 1.

4.3 Evaluation Results in Scenario 2

Figure 9 shows the average throughput in scenario 2. In this scenario, both the conventional method and the proposed method realize 40% improvement compared with the original TCP Reno. On the other hand, as shown in Figure 10, the number of duplicate ACKs is 3.4 times in the conventional method and 1.5 times in the proposed method, compared with Reno. This means that, although the throughput improvement is similar for the conventional method and the proposed method, the conventional method induces the increase of packet losses.

The number of transmitted packets increases as the throughput increases. Along that, the number of duplicate ACKs also increases. In order to evaluate the number of duplicate ACKs independently of the amount of transmitted packets, we define the congestion index α by the following equation;

$$\alpha = \frac{number\ of\ duplicate\ ACKs}{total\ number\ of\ transmitted\ packets}. \quad (3)$$

Figure 11 shows the average of congestion index. From this graph, the proposed method does not increase the congestion index from the original Reno, while the congestion index of the conventional method increases, as twice as the original Reno. This result means that the proposed method realizes high performance without increasing unnecessary packet losses, that is, without deteriorating congestion.

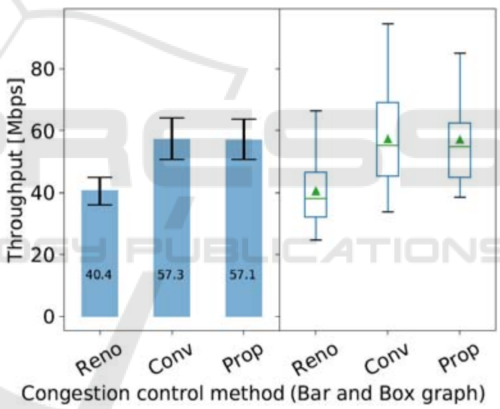


Figure 9: Throughput in scenario 2.

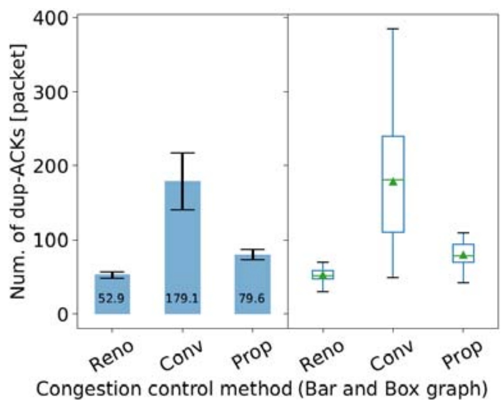


Figure 10: Number of duplicate ACKs in scenario 2.

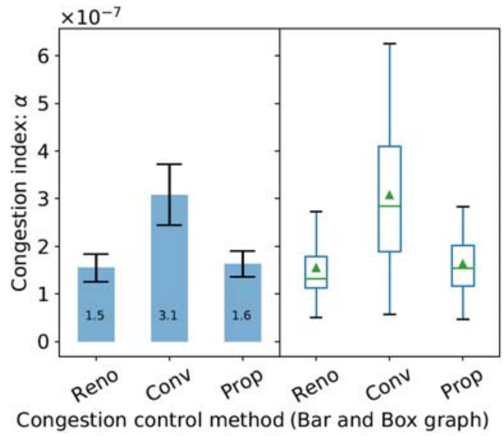
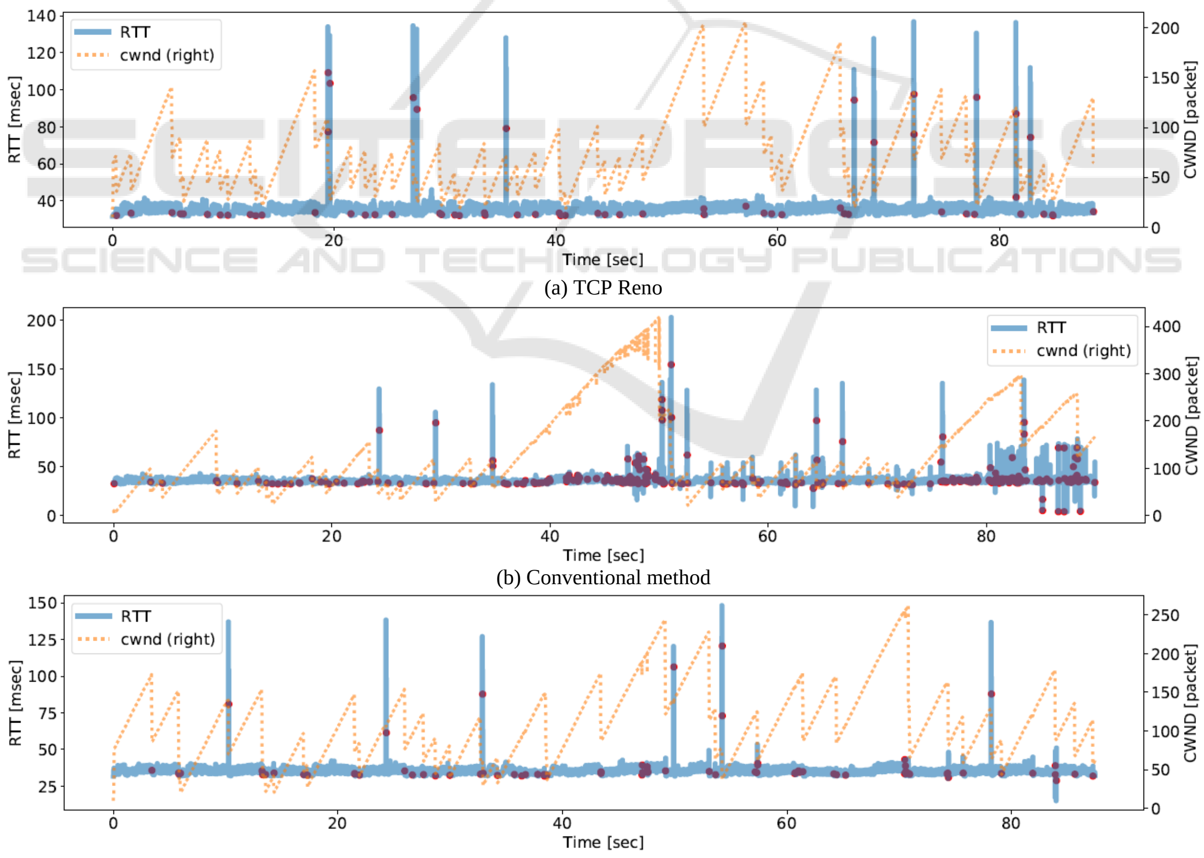


Figure 11: Ratio of duplicate ACKs for total segments in scenario 2.

In the end of this subsection, we show a detailed result for individual experimental run for 90 seconds. Figure 12 shows the time variation of RTT and cwnd for TCP Reno, the conventional method, and the proposed method. Red dots in the graph of RTT show three duplicate ACKs triggering fast retransmit. In the case of TCP Reno, every three duplicate ACK decreases cwnd as indicated in Figure 12 (a). Some of these decreases are wireless loss driven events. In the conventional method given in Figure 12 (b), cwnd does not decrease when RTT is small, and so, there are many chances that cwnd take larger value than TCP Reno. But, sometimes cwnd does not decrease even if the value is large (see around 50 seconds and 80 seconds). It is considered that these situations deteriorate congestion. On the contrary, in the proposed method given in Figure 12 (c), cwnd does not decrease while RTT is small and cwnd decreases when the cwnd value itself is large.

5 CONCLUSIONS

In this paper, we proposed a method to classify TCP packet losses over IEEE 802.11 WLAN into conges-



(a) TCP Reno

(b) Conventional method

(c) Proposed method

Figure 12: Time variation of RTT and cwnd in scenario 2.

congestion losses and wireless losses by the *K*-means clustering focusing on both congestion window size and round-trip time. The proposed method modifies the TCP congestion control such that if packet losses are categorized as wireless losses, the congestion window size does not decrease. We implemented the proposed method within the Linux operating system and conducted the performance evaluation using real WLAN network. The results showed that the proposed method provides 40% higher throughput than TCP Reno and that it does not increase the ratio of duplicate ACKs to the total packets, which the conventional method focusing only on RTT suffered from.

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