Effects of Handover on Wireless LANs on TFRC Rate Control

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Abstract **– Transport protocol such as RTP and TFRC take control of sending rate by feedback-controls which measure RTT, packet loss rate, etc. On wireless LANs, the communication condition changes drastically when a node hands over from an idle access point to a congested one. However, the node keeps measured values after the handover. Because of this, the node may not be able to control the sending rate accurately. In this paper, we present the result of simulations of DCCP CCID3 data streams adopting TFRC focused on disconnection period due to heavy loaded wireless LAN handover. We also propose schemes to avoid the effect of handovers and evaluate them.**

Key words: TCP friendly rate control, wireless LANs, handover, QoS control

1. Introduction

In recent years, multimedia applications such as online games and real time video streaming has been becoming popular because of the growth of broadband wired and wireless networks. However it is difficult for wireless networks to adopt real time applications, since available bandwidth on wireless network strongly (especially on wireless LANs) depends on the number of connected nodes, traffic load, and so on. Thus QoS control is very important in for real time applications.

One of the important QoS controls in real time multimedia applications is rate control such as TCP Friendly Rate Control (TFRC) [3]. TFRC controls the sending rate by collecting network conditions, which are RTT and loss event rate. TFRC aims to achieve TCP friendliness (fairness between real time data flow and TCP flow) and smooth rate control. Thus TFRC is adequate for real time data streaming.

However it is not clear that the rate control brings out good performance in the handover among wireless networks. For example, on wireless LANs the communication conditions changes drastically when a node hands over from an idle access point (AP) to a congested one. If the node which hands over uses the rate control depending on feed back control, the node may not be able to calculate the sending rate which is suitable for the smaller available bandwidth after handover because the sender node can not estimate the condition at the new AP before it hands over. Furthermore, the node can not send and receive packets for a while (we call it *disconnection period*) when the node hands over, because the node needs a certain amount of time to be connected with the new AP after the node has disconnected from the old AP. Because TFRC calculates a sending rate according to loss event rate and RTT, a consecutive packet losses affects the calculation of the sending rate. Because the environment after handover is not related to the environment after and during the handover, the node can not calculate the appropriate sending rate after the handover.

In this paper, we present the result of simulations of DCCP CCID3 data streams adopting TFRC focusing on disconnection period due to handover to the heavy loaded wireless LAN. We confirmed that the RTT of the handover node (HON) increases because of that packets stay at the sending buffer at MAC layer in the disconnection period and the average loss event rate become small after the handover. We also purpose schemes to avoid the effect of handovers and evaluate them by simulation.

2. TCP Friendly Rate Control

TFRC adopted by DCCP CCID3 is a transport scheme suited for real time applications. TFRC does not retransmit packets. It controls only the packet sending rate according to the network conditions which are weighted RTT and loss event rate in order to offer TCP friendliness. Network conditions are collected by a receiver node. The receiver node sends feedback messages which includes a sending rate value at which the receiver has estimated, timestamp for calculating RTT, and loss event rate. Using this collected information, the sender node calculates its new sending rate.

Figure 1 shows the behavior of TFRC. b is the number of packets acked by a single TCP ACK. T_{RTO} is the TCP retransmission timeout value in seconds. T_{mbi} is 64 seconds. Thus a sender node sends at least one packet every 64 seconds.

For calculating RTT, a sender node calculates the last RTT sample R_{sample}. When R_{sample} is calculated at first time, RTT R equals R_{sample}. Otherwise, R is calculated by the following equation

$$R = qR + (1-q)R_{sample} \tag{1}$$

q is a constant value. The recommended value of q is 0.9.

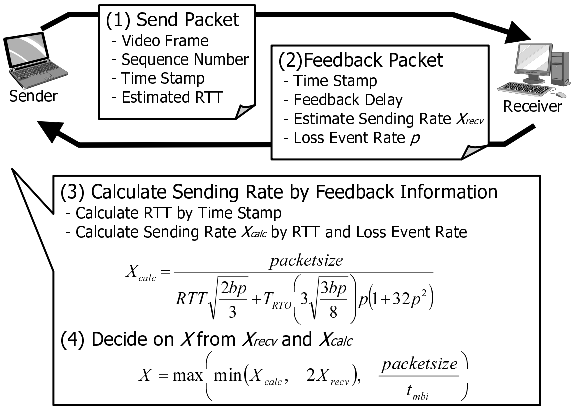

Figure 1: TCP friendly rate control

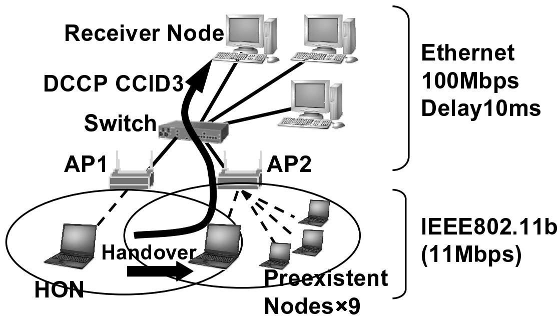

Figure 2: Network topology

For calculating the loss event rate, the receiver node works as follows.

1. Detect loss events. Consecutive packet losses are treated as one loss event.
2. Calculate loss intervals from the inter-loss event intervals.
3. Calculate average loss interval I_{mean} by using a filter that weights the eight most recent loss event intervals.
4. Calculate loss event rate p as follow.

$$p = \frac{1}{I_{mean}} \tag{2}$$

3. Effect of Handover on TFRC

There are some studies on effects of handover of the mobile node [1, 5]. In [1], the another point out that a node which takes streaming service using Real Time Streaming Protocol (RTSP) can not estimate available bandwidth exactly when a node hands over, then the streaming server sets low deliver rate. They also proposed a scheme in which a proxy server manages control messages for the delivery rate instead of the end node. In [5], the authors evaluate the performance of the TFRC rate control on vertical handover, and point out that TFRC nodes can not obtain high sending rate when the node hands over from a low-bandwidth environment to high-bandwidth one. As stated above, handover on wireless environment could lead that a mobile node not be able to estimate the available bandwidth.

In this paper, we evaluate the effects of handover on TFRC data sending rate when a node hands over from a low loaded AP to a high loaded one. On such a condition, the available bandwidth of the node decreases drastically. However, after the handover, the node which hands over (HON) may send packets using too higher or too lower sending rate than the actual available bandwidth after the handover because they use the old information which has been collected before the handover. Furthermore, if the HON can not send packets due to the disconnection during the handover, the receiver node will lose the packets continuously. Generally, the RTT and loss event rate after handover do not correlate with ones before the handover. However the sending rate control mechanism of TFRC depends on the past measured values. Thus it is difficult for the HON to calculate the sending rate which is adopted to the available bandwidth just after the handover.

4. Simulation Results and Discussion

In this section, we evaluate the effects on TFRC flows by simulations of cases when a TFRC node hands over from an idle AP to a congested one at which multiple TFRC nodes are sending data stream to the outside of the wireless LAN.

4.1. Network Model

We used OPNET for the simulation. We developed a DCCP model based on TUBS DCCP model [7] to operate it on IP networks including wireless LANs. Figure 2 illustrates the network topology used for the simulation. Each wireless node has IEEE802.11b LAN interface which has sending buffer that can store at most 50 packets. Five seconds later from the beginning of a simulation, all nodes begin to communicate with their receiver nodes outside the wireless LAN. At first, a node is connected to AP1 and nine nodes are connected to AP2. We call these nodes “HON” and “preexistent nodes” respectively. 20 seconds later, the HON disconnects from AP1 and connects with AP2 after a certain period. The packet size of all data packets is 629 bytes.

4.2. Effect of Disconnection Period on Sending Rate

Figure 3 shows the average sending rate of the HON and preexistent nodes obtained by 10 simulation runs. The lengths of the disconnection period are 1.5, 2.5, 3.5, and 4.5 seconds. When the length of the disconnection period is short such as 1.5 and 2.5, the sending rate of the HON is lower than the sending rate of the preexistent nodes for more than 10 seconds. On the other hand, when the disconnection period is longer than 3.5 seconds, the

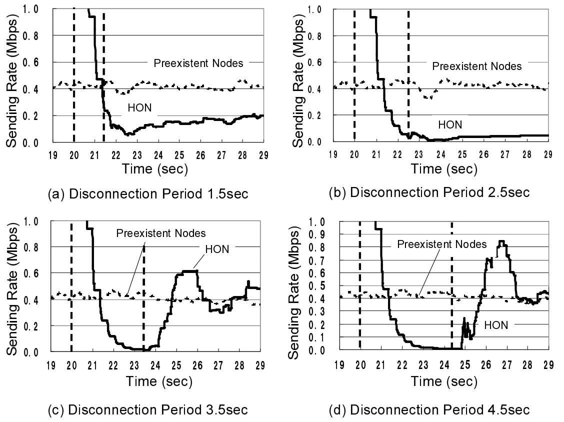

Figure 3: Sending rate of HON and preexistent nodes

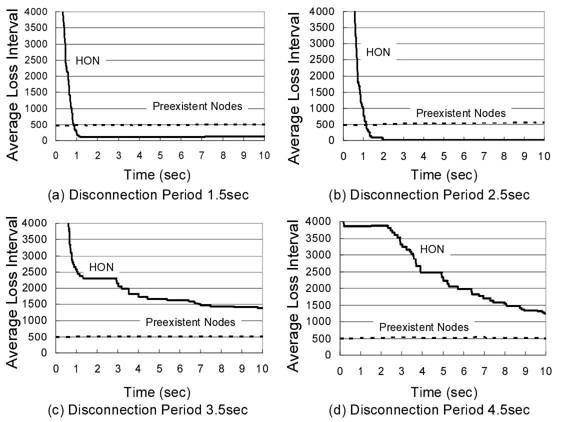

Figure 4: Average loss interval of HON and preexistent nodes vs disconnection period

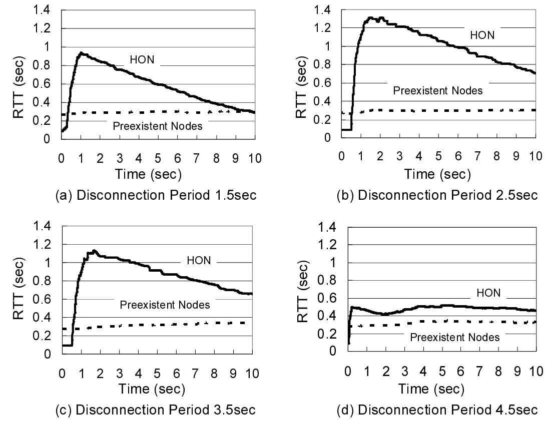

Figure 5: RTT of HON and preexistent node vs disconnection period

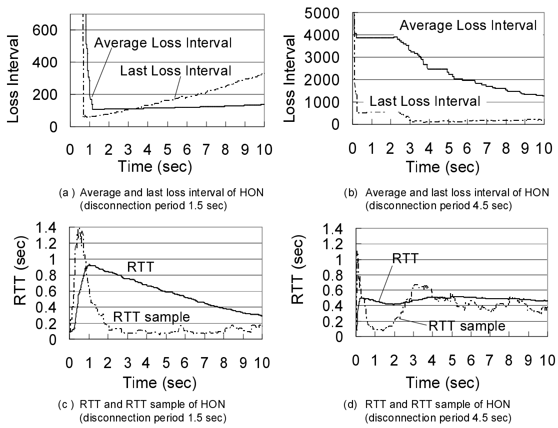

Figure 6: Compare measured to calculated values of HON vs disconnection period

sending rate of the HON recovers rapidly. Then the sending rate of the HON approaches the one of the preexistent nodes. It reveals that the disconnection period affects the sending rate of the HON and the fairness between the flows of the HON and the preexistent nodes. Because the nodes collect RTT and average loss interval which are measured at the receivers for calculating the sending rate, the disconnection period affects the RTT and the average loss interval.

4.3. Effect of Disconnection Period on Average Loss Interval and RTT

Figure 4 shows the average loss interval of the HON and the preexistent nodes versus the length of the disconnection period. Figure 5 shows the RTT of the HON and the preexistent nodes versus the disconnection period. In these figures, the timing when the HON hands over is set to 0 second. When the disconnection period is shorter than 2.5, packets are buffered at the MAC layer for a long time because the MAC interface of the HON can not send packets. After the handover, the node resumes the session, then the packets are received by the receiver node. It prolongs the RTT of these packets. Adding to this, the buffered packets cause collisions and packet losses after the handover because they are sent burstly. It results in the shorter average loss interval of the HON than one of the preexistent nodes. On the other hand, when the disconnection period is longer than 3.5 seconds, a lot of packets are discarded at the sending buffer of the MAC layer during the disconnection period, because the retry count of the MAC interface reaches the maximum retry count. This prevents the RTT from becoming longer. On the handover, the sending rate of HON decreases because the HON can not receive feedback reports. This prevents that the HON avoids a lot of collision. Furthermore, although many packets are lost on the handover, because the continuous packet losses are treated as a single loss event in TFRC, the number of observed loss events at the receiver of the HON becomes smaller than that of the preexistent nodes.

Next, to examine whether the HON can estimate the network condition accurately, we compare the measured raw values of RTT and loss interval to the calculated ones according to the algorithm used in TFRC. Figure 6 shows the measured loss interval, calculated average loss interval, RTT samples and calculated RTT at the HON. We can see that there is a big difference between the last measured RTT and the calculated average RTT when the disconnection period is short. This is because that the calculation of the average RTT of TFRC is affected by the past measured RTT. On the other hand, when the disconnection period is long, we can see that the difference between the last loss interval and the calculated average loss interval is large. The average loss interval is calculated when a new loss event occurs. This leads that the calculated value is affected strongly by the past loss intervals when the frequency of loss events becomes small suddenly. However, when the average loss interval of the HON is longer than the average loss interval of the preexistent nodes, HON will calculate the higher sending rate than the preexistent node. This causes congestion of the network.

5. Avoiding Effect of Handover

5.1. Proposed Scheme

We confirmed that the length of the disconnection period and the values of RTT and average loss interval which the sender and receiver nodes have calculated before handover of the sender node affect the calculation of RTT and loss event rate. It prevents a HON from calculating the sending rate accurately. Then we propose three basic ideas to solve this problem.

1. A HON uses a predictive rate control technique [2, 4].
2. A HON notifies the destination node that the HON has handed over in order to tell the receiver to discard measured value before the handover when it has finished the handover. Then the HON calculates RTT and loss event rate using the values measured after the handover only.
3. A transport protocol of A HON stops sending the packets to the lower layer while it is executing handover.

Note that all the above proposed schemes require that a transport protocol to receive the event of the handover from the lower layer protocols.

In scheme 1, the handover of the HON is triggered by a control server on the network. The control server estimates available bandwidth of HON after the handover and notifies it to the HON.

In scheme 2, to calculate the sending rate of the HON after the handover, the HON and the receiver nodes have to collect RTT and loss event of packets sent after handover rapidly. Because the receiver node sends the feedback report include timestamp, the HON can know the RTT of such packets. Thus, to calculate the RTT only from the packet sent after the handover, we propose a scheme to avoid calculating RTT using values measured before handover as follow.

A HON discards the timestamp of the last data packet received when the timestamp is smaller than the time when the HON has handed over. When the timestamp is bigger than the time when the HON has handed over, the HON calculates R_{sample}. When R_{sample} is calculated at first time after handover, R equals R_{sample}. Otherwise R is calculated by Equation (1). In order to calculate a loss interval, at least 2 loss events are needed. Thus, when the receiver node detects two loss events after the handover, the receiver node starts calculating the average loss interval using ones calculated after the handover of the HON.

However it is a problem that the HON can not calculate the sending rate when the RTT and the average loss interval have not been calculated even when it has finished the handover. For example, if the sending rate of a HON is lower than the available bandwidth after handover, the receiver node can not detect any loss event for a long time. Thus, we have to consider how to calculate the sending rate when the RTT and the loss event rate caused by packets sent after handover have not been calculated at the receiver. In this paper, we assume the HON simply doubles the sending rate at intervals of the round-trip time which it has already known until a loss occurs. When the first loss occurs after handover, the receiver node calculates the loss event rate using the estimated RTT and Received rate R_{recv}. We call it *slowstart phase after handover*.

In scheme 3, the HON avoids packets to stay at the sending buffer of the MAC layer when a HON hands over. When the transport protocol does not send packets to the lower layer, a lot of packets stay at the sending buffer at the transport protocol. Thus the transport protocol has to manage the sending buffer according to the type of the application data. For example, when a HON generates real time streaming data in the disconnection period, the end to end delay of the data will be very long. The receiver node can not play back the data. In this case, the sending buffer of the transport protocol should discard those data in the disconnection period. When the HON generates the non real time streaming data, it may be better that the transport protocol just stops sending the packets to the lower layer while the sending buffer of the MAC layer can not send packets.

5.2. Simulation Results

In this section, we examine the effect of proposed scheme 2 and 3 by simulation. The network model and the network topology of this simulation are the same as the simulation of section 4.1. The HON and receiver node of the HON use DCCP enhanced with the proposed scheme 2 and 3 (proposed DCCP). We compare the proposed DCCP to the normal DCCP.

Figure 7 shows the average sending rate of the HON using the proposed DCCP and the normal DCCP and the one of preexistent nodes obtained by 10 simulation runs. The timing when the HON hands over is 0 second. The length of disconnection periods are 1.5 and 4.5 sec. When the length of the disconnection period is 1.5 sec, the send-

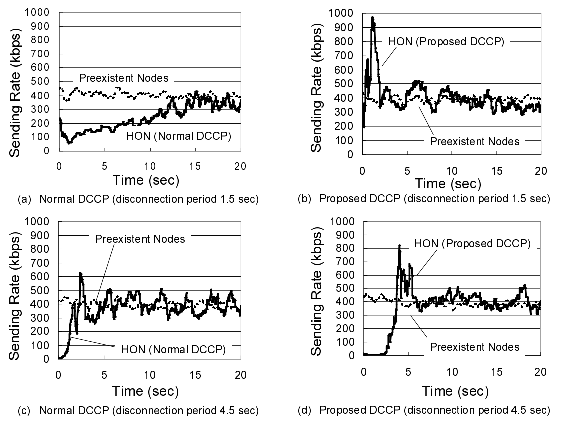

Figure 7: Sending rate of HON and preexistent nodes

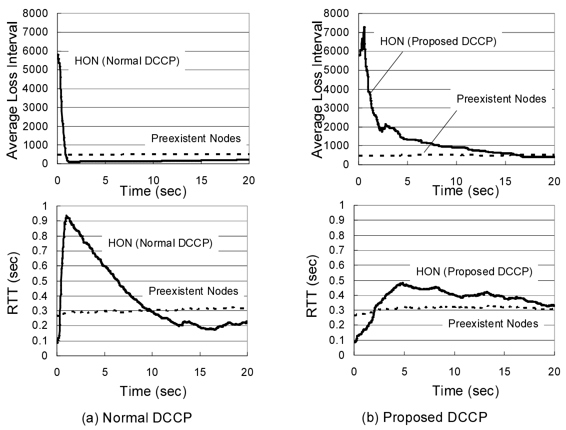

Figure 8: RTT and average loss interval of HON and preexistent nodes when disconnection period is 1.5 sec

ing rate of the HON using the normal DCCP is lower than preexistent nodes. On the other hand, the sending rate of the HON using the proposed DCCP is higher than preexistent nodes for 3 seconds. After 3 second, the sending rate of the HON becomes the same as the sending rate of the preexistent nodes. When the length of the disconnection period is 4.5 sec, the sending rate recover time of the HON using the proposed DCCP is longer than the one of the HON using the normal DCCP a little. However, for the 6 seconds after the handover, the sending rate of the HON using the normal DCCP is about the same as the one of the preexistent nodes. Because the sending rate of the HON after the handover is very low value, it is difficult that the HON calculates a high sending rate in the slowstart phase after handover.

Figure 8 shows the average loss interval and the RTT of the HON using the proposed DCCP, the normal DCCP and the one of the preexistent nodes. The timing when the HON hands over is 0 second. We can see that the HON can avoid calculating high RTT by the proposed scheme. The average loss interval of the HON using proposal DCCP is longer than preexistent nodes because the sending rate of the HON after handover is lower than the one of the preexistent nodes, and the proposed scheme avoids that packets stay at the sending buffer of the MAC layer.

These results indicate that the proposed scheme is effective for a HON and the receiver node to calculate the RTT and the average loss interval after handover. The HON using proposed DCCP can make the sending rate close to the one of the preexistent nodes rapidly after handover when the disconnection period is short. When the disconnection period is long, the sending rate recover time of the HON using the proposed DCCP is longer than the one of the HON using the normal DCCP a little. However, to calculate the sending rate from the RTT and loss interval observed after handover, this scheme would be also effective for cases when the disconnection period is long if we can develop better algorithm for calculating a new sending rate for such a situation.

6. Conclusion

We evaluated the effects of wireless LAN handover on TFRC flows when a node hands over from an idle AP to a congested one by simulation. We confirmed that the length of the disconnection period during the handover affects the calculation of the sending rate and results in unfairness between a HON and preexistent nodes. We also proposed schemes to solve these problems. We evaluated the schemes by simulation and confirmed that the proposed schemes avoid the HON to calculate the lower sending rate than the one of preexistent nodes for a while. In our future work, we will evaluate the effectiveness of the proposed schemes using real time video streaming traffic models.

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