Wireless Profiled TCP Performance over Integrated Wireless LANs and Cellular Networks

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***Abstract*— An analytical framework for studying the performance of Wireless Profiled TCP (WP-TCP) flows over the integrated wireless LAN and cellular networks is proposed. The framework can be used to analyze the short-term performance during vertical handover and long-term performance of WP-TCP for a given set of network and protocol parameters. It captures the WP-TCP behavior under the influence of wireless channel errors, step change in network parameters and excessive packet losses due to vertical handovers. Extensive simulations are conducted to verify the accuracy of the analytical framework. The main findings in this study are: 1) when the network is subjected to hard handovers, increasing the maximum window size improves the efficiency in a high transmission error environment, but degrades the efficiency in a low transmission error environment; 2) increasing the congestion window reduces the chances of premature timeouts during soft upward vertical handover; and 3) depending on duplicate ACK threshold, increasing the congestion window can increase or reduce the chances of false fast retransmit during soft upward vertical handover.**

***Index Terms*— Integrated wireless networks, performance analysis, wireless application protocol, WP-TCP.**

I. INTRODUCTION

THE increasing demand of versatile services by mobile users has led to the integration of a variety of wireless networks such as wireless WANs (e.g., GPRS/UMTS cellular networks) and wireless LANs (e.g., 802.11x and HiperLAN/2). Fig. 1 shows a network architecture for integrated WLAN and cellular networks. The integration can be in a tightly-coupled or loosely-coupled manner [1]. In tightly-coupled integration, WLANs are directly connected to the cellular network (WLAN hotspot 2 and the cellular network). In this case, the WLAN access appears to the cellular network as any other cellular access network. In the loosely-coupled integration, the WLANs are directly connected to the Internet (WLAN hotspot 1 and the cellular network). Network integration poses several challenges such as sudden changes in network characteristics, handover, etc. Therefore, a stable and efficient transport mechanism is required.

Wireless Application Protocol (WAP) [2] is a de-facto world standard for the presentation and delivery of wireless information services on wireless devices. The earlier versions, WAP 1.x, are standards optimized for mobile environment where hand-held wireless devices are limited by CPU power, memory, battery lifetime and simple user interface, and wireless links are characterized by low bandwidth, high latency and unpredictable availability and stability. As wireless networks and mobile devices evolve, some of these constraints become less significant. The WAP 2.0 standard, released in July 2001, utilizes the advantages of advancement in wireless network and mobile devices. WAP 2.0 is one of the promising transport mechanisms for integrated WLANs and cellular networks. It uses Wireless Profiled TCP (WP-TCP) [3] as one of the reliable transport protocols. When WP-TCP is deployed over integrated WLANs and cellular networks, its performance can dramatically degrade. This is due to handovers between networks that result in excessive packet losses and sudden change of network characteristics [4], in addition to unpleasant wireless links characteristics [5]. A very important step to address these challenges is to thoroughly study the performance of WP-TCP and its sensitivity to network parameters. From the literature [6], the main techniques which can be used in studying the performance are analytic modeling, simulation, and measurement. Generally, there is no single best technique for all. A combination of these techniques is usually preferred.

Various studies related to WAP performance have been conducted since its appearance. The studies in [7]–[9] consider the earlier versions of WAP (WAP 1.x standards) and evaluate the performance only over cellular networks. In [10], [11], and references therein, the performances of WP-TCP (WAP 2.0 standard) and other variants of TCP are analyzed based on independent WLAN or cellular networks. Therefore, the effects of sudden change of network characteristics experienced in integrated WLANs and cellular networks cannot be captured. Recently, several studies on performance of TCP over integrated WLAN and cellular networks have been reported [12]–[14]. In these studies, the TCP performance has been evaluated using simulation and/or experimentation. To the best of our knowledge, there is no analytical study in the open literature that specifically focuses on the performance of WP-TCP over integrated WLAN and cellular networks.

This paper complements the simulations and measurements studies by introducing an analytical framework for studying the performance of WP-TCP in integrated WLANs and cellular networks. For given network and protocol parameters, explicit mathematical expressions which describe the short- and long-term performance of WP-TCP under the influence of

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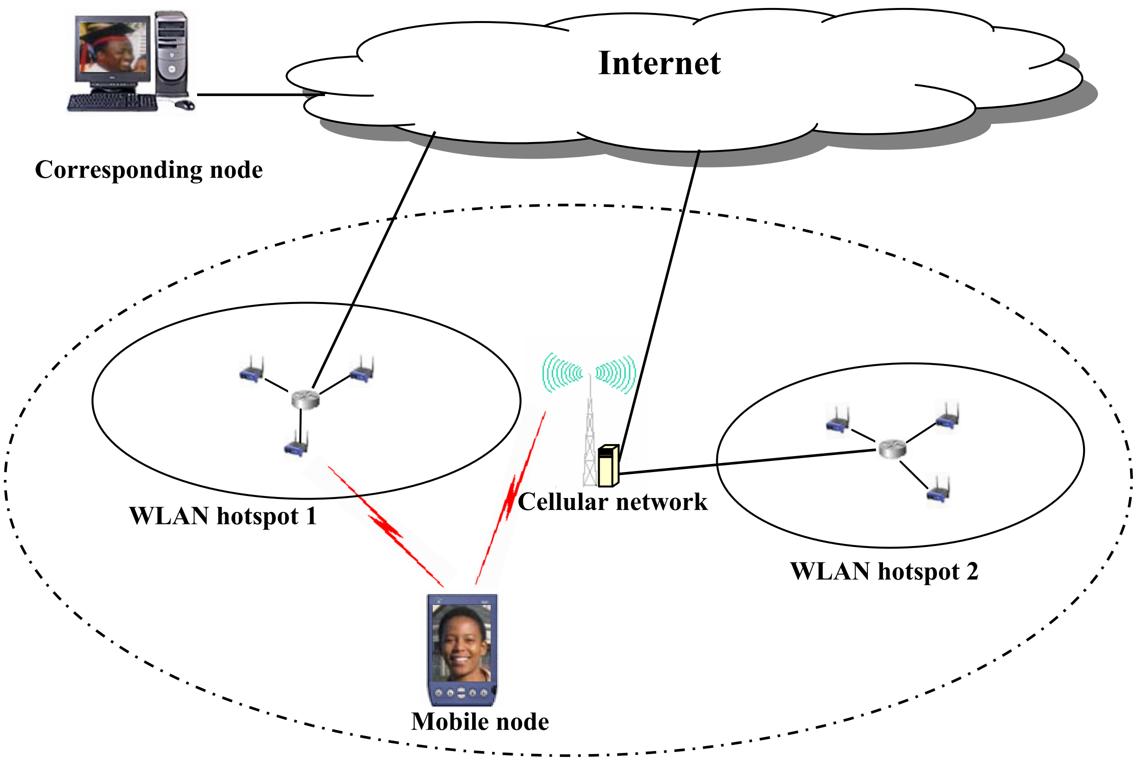

Fig. 1. An integrated WLAN and cellular networks architecture.

excessive packet losses and sudden change of network characteristics are derived. Extensive simulations are conducted to verify the reasonableness of the assumptions and the validity of the analytical expressions. The major contributions of this paper are two-fold: 1) proposal of an analytical framework to evaluate the performance of WP-TCP in integrated WLANs and cellular networks, and 2) analysis of the short- and long-term performance of WP-TCP, which provides insights for further research.

The remainder of this paper is organized as follows: The system description is given in Section II. Section III presents the theoretical analysis, and the performance is evaluated in Section IV. Finally, conclusions are drawn in Section V.

II. System Description

We consider FTP traffic over the connection-oriented reliable transport WP-TCP, which is assumed to be implemented with all mandatory requirements (RFC 0793, RFC 1122 [15], RFC 2581 [16], and Selective Acknowledgement (SACK) RFC 2018 [17]) and some important optional requirements (Large Initial Window RFC 2414 [18] and Timestamps Option (RFC 1323) for Round Trip Time Measurement). We also consider RFC 2988 [19] for computing retransmission time and SACK extensions for loss recovery mechanism (RFC 3517 [20]) and acknowledging the receipt of a duplicate packet (D-SACK RFC 2883 [21]). WP-TCP has three transfer stages: connection establishment, data transfer, and connection tearing down. In this study, we are only interested in the performance of WP-TCP during vertical handover and the long-run average performance of long-lived WP-TCP flows. Therefore, the connection establishment and tearing down stages are not considered.

In general, cellular networks have wide coverage area whereas WLANs have small coverage area and are only available within distinct hotspots. Therefore, it is reasonable to assume that a single large cell of a cellular network is overlaid on disjoint WLAN hotspots. In this study, the WLAN and the cellular network are integrated in a loosely-coupled manner. The motivation behind this choice is based on the fact that the loosely-coupled configuration is more scalable and therefore expected to be widely deployed in the near future. Mobile IPv6 (MIPv6) [22] is considered at the network layer. It provides mobility support between WLAN and the cellular network. The network model for integrated WLAN and cellular network is depicted in Fig. 2. The mobile node can switch between WLANs and the cellular network. In this paper, the handover from WLAN to the cellular network is referred to as upward vertical handover. And the handover from the cellular network to WLAN is referred to as downward vertical handover. The handover can also be hard or soft. Hard handover is when the old connection is broken before the new connection is made and soft handover is when the new connection is made before the old connection is broken.

We assume WLAN and cellular networks residence times are exponentially distributed random variables. The underlying networks have sufficiently large buffers and fixed but arbitrary channel delays, bandwidths, and error rates. Soft downward vertical handover can result in packet reordering which in turn can cause burst transmission in WLAN and duplicate ACKs. Soft upward vertical handover can result in inrush packet transmission in a cellular network and a step increase in

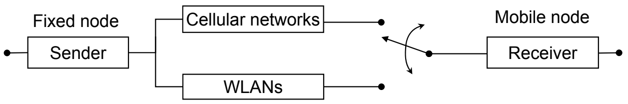

Fig. 2. Network model for integrated WLANs and cellular networks.

round trip time. Since buffers in the networks are considered to be sufficiently large, analysis of packet losses due to burst transmissions or inrush packet transmission beyond the scope of this paper.

III. Performance Analysis

Fig. 3 shows the transition model representing the WP-TCP transmission process. The WP-TCP sender remains in the transient congestion window phase (which is slow start or congestion avoidance phase) until the congestion window reaches maximum window size (transition a), or the packet gets lost due to transmission error (transition b), or a vertical handover occurs (transition c). While in the steady congestion window phase (which is the maximum window size phase), the WP-TCP sender re-enters the steady congestion window phase if all packets in the same congestion window are successfully transmitted (transition d); otherwise the WP-TCP sender enters the packet recovery phase if there is a transmission error (transition e) or vertical handover (transition f). After the recovery due to transmission errors, the WP-TCP sender enters the transient congestion window phase (transition g) and continues with transmission. After the recovery due to handover, the WP-TCP sender enters the transient congestion window phase (transition h) if the congestion window is smaller than the maximum window size; otherwise, it enters the steady congestion window phase (transition i).

To track the packet transmission process (Fig. 3), we consider a random process $S(t) = (i(t), c(t), w(t), w_{th}(t))$, where $i(t) \in \{0, 1\}$ is the state of communication channel (0 means WLAN and 1 means cellular network), $c(t) \in \{g, b, h\}$ is the status of the transmitted packet over the channel (g means no packet loss, b means packet loss due to transmission error, and h means packet loss due to handover), $w(t) \in \{1, 2, \ldots, w_{\max}\}$ is the WP-TCP sender congestion window, $w_{th}(t) \in \{2, 3, \ldots, \lceil w_{\max}/2 \rceil\}$ is the WP-TCP sender slow start threshold, t is the time measured in slots, and $w_{\max}$ is the WP-TCP sender maximum window size. By sampling the random process $S(t)$ at the beginning of each phase (Fig. 3), the sampled process is a semi-Markov process [23] with embedded Markov chain in the state space $W_s = (i, c, w, w_{th}), \ \forall i, c, w, w_{th}$. The embedded Markov chain is defined by the transition matrix $\Phi = [\phi_{AB}]$, where ϕ_{AB} denotes the transition probability from state $A \in W_s$ to state $B \in W_s$. To compute the long-term performance, the transitions of embedded Markov chain are labeled with the corresponding elements of the matrices $\mathbf{S} = [s_{AB}]$, $\mathbf{D} = [d_{AB}]$, and $\mathbf{M} = [m_{AB}]$. The elements s_{AB}, d_{AB}, and m_{AB} are respectively the number of packets transmitted successfully, the transition time, and the number of packets transmitted from state $A \in W_s$ to state $B \in W_s$.

The long-term performance metrics are throughput rate and efficiency. Throughput rate is defined as the number of packets

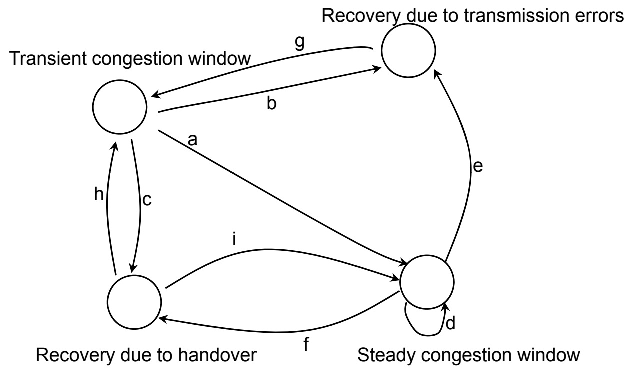

Fig. 3. Transition model for WP-TCP transmission process.

transmitted successfully per unit time, whereas efficiency is defined as the ratio of the number of packets transmitted successfully to the number of packets transmitted. The long-run average throughput rate (Ω) and efficiency (Λ) can be computed as

$$\Omega = \lim_{t\to\infty} \frac{S(t)}{t} = \frac{\sum\limits_{A\in W_s} \pi_A \sum\limits_{B\in W_s} \phi_{AB} s_{AB}}{\sum\limits_{A\in W_s} \pi_A \sum\limits_{B\in W_s} \phi_{AB} d_{AB}} \tag{1}$$

and

$$\Lambda = \lim_{t\to\infty} \frac{S(t)}{M(t)} = \frac{\sum\limits_{A\in W_s} \pi_A \sum\limits_{B\in W_s} \phi_{AB} s_{AB}}{\sum\limits_{A\in W_s} \pi_A \sum\limits_{B\in W_s} \phi_{AB} m_{AB}}, \tag{2}$$

where $S(t)$ and $M(t)$ denote the number of packets transmitted successfully and the number of packets transmitted by time t, respectively. π_A is the steady state probability of being in state A and can be found by manipulating the matrix Φ [23]. In each phase (Fig. 3) the possible initial states and transitions to the final states are explored. For every possible transition, the corresponding transition time, the number of packets transmitted successfully, and the number of packets transmitted are computed. For easy reference, the notations used throughout the paper are listed in Table I. In the rest of this section, the functions $N(.,.,.)$, $T(.,.,.)$, and $W(.,.,.)$, which are defined in Table I, are used to facilitate the analysis. These functions are deterministic and they can be easily tabulated or derived [10].

A. Transient Congestion Window

The transmission process enters the transient congestion phase when a packet loss is recovered by using the timeout mechanism or the duplicate acknowledgment mechanism. Therefore, a set of possible initial states is $A = (i, g, w, w_{th}), \forall i, w = \{1, w_{th}\}, w_{th}$. Let t and $s \in \{0, 1, \ldots, N(w, w_{th}, w_{\max})\}$ be the cell residency time and the number of packets transmitted successfully before the transition occurs, respectively. Given the initial state A and the number of successfully transmitted packets s, the final state can be found as in (3).

TABLE I
LIST OF NOTATIONS

1) λ_i: mean residence times for network i measured in seconds.
2) b_i: link bandwidth for network i measured in bits per seconds.
3) l_i: one-way channel delay for network i measured in seconds.
4) p_i: transmission error rate for network i.
5) BDP: bandwidth-delay product measured in packets.
6) w: WP-TCP congestion window measured in packets.
7) w_{th}: WP-TCP slow start threshold measured in packets.
8) $w_{\max}$: WP-TCP maximum window size measured in packets.
9) Z: WP-TCP duplicate acknowledgement threshold measured in packets.
10) RTO: WP-TCP retransmission timeout measured in seconds.
11) MSS: WP-TCP maximum segment size measured in bytes. This is equal to a packet size.
12) ϕ_{AB} : transition probability from state A to state B.
13) s_{AB}: number of packets transmitted successfully from state A to state B.
14) d_{AB}: transition time from state A to state B in seconds.
15) m_{AB}: number of packets transmitted from state A to state B.
16) $W(s, w_i, w_{th})$: returns the final congestion window measured in packets given that the number of packet transmitted successful over the ideal channel is s, the initial congestion window is w_i, and the slow start threshold is w_{th}.
17) $T(s, w_i, w_{th})$: returns the time taken measured in seconds given that the number of packet transmitted successful over the ideal channel is s, the initial congestion window is w_i, and the slow start threshold is w_{th}.
18) $N(w_i, w_{th}, w_f)$: returns the number of packet transmitted successful over the ideal channel given that the initial congestion window is w_i, the slow start threshold is w_{th}, and the final congestion window is s.

$$B = \begin{cases} (i, h, W(s, w, w_{th}), w_{th}), \\ \quad T(s+1, w, w_{th}) \geq t > T(s, w, w_{th}), \\ \qquad s < N(w, w_{th}, w_{\max}) \\ (i, b, W(s, w, w_{th}), w_{th}), \\ \quad t > T(s+1, w_{\max}, w_{\max}), \\ \qquad s < N(w, w_{th}, w_{\max}) \\ (i, g, W(s, w, w_{th}), w_{th}), \\ \quad t > T(s, w_{\max}, w_{\max}), \\ \qquad s = N(w, w_{th}, w_{\max}). \end{cases} \tag{3}$$

The first, second, and third equations in (3) correspond to transitions c, b, and a shown in Fig. 3, respectively. Let $\beta(s)$ denote the probability of successfully transmitting s packets given that the initial and final states are A and B, respectively. It follows that

$$\beta(s) = \begin{cases} 1 - e^{-\lambda_i T(1, w, w_{th})}, \\ \quad t \geq T(1, w, w_{th}), s = 0 \\ p_i e^{-\lambda_i T(1, w, w_{th})}, \\ \quad t > T(1, w, w_{th}), s = 0 \\ (1-p_i)^s (e^{-\lambda_i T(s, w, w_{th})} - e^{-\lambda_i T(s+1, w, w_{th})}), \\ \quad T(s+1, w, w_{th}) \geq t > T(s, w, w_{th}), \\ \qquad 0 < s < N(w, w_{th}, w_{\max}) \\ p_i (1-p_i)^s e^{-\lambda_i T(s+1, w, w_{th})}, \\ \quad t > T(s+1, w, w_{th}), \\ \qquad 0 < s < N(w, w_{th}, w_{\max}) \\ e^{-\lambda_i T(s, w, w_{th})} \cdot (1-p_i)^s, \\ \quad t > T(s, w, w_{th}), \\ \qquad s = N(w, w_{th}, w_{\max}). \end{cases} \tag{4}$$

From (3) and (4) there are several values of s that can result in the same final state B given that the initial state is A. Let U_{AB} denote a set of possible values of s such that the initial state is A and the final state is B. The transition probability, the transition time, the number of packets transmitted successfully, and the number of packets transmitted can respectively be represented as

$$\phi_{AB} = \sum_{s \in U_{AB}} \beta(s), \tag{5}$$

$$d_{AB} = \frac{\sum_{s \in U_{AB}} \beta(s) T(s)}{\sum_{s \in U_{AB}} \beta(s)}, \tag{6}$$

and

$$s_{AB} = m_{AB} = \frac{\sum_{s \in U_{AB}} \beta(s) s}{\sum_{s \in U_{AB}} \beta(s)}. \tag{7}$$

B. Steady Congestion Window

The transmission process enters the maximum window phase when the congestion window reaches the maximum window size. Therefore, a set of possible initial states is $A = (i, g, w_{\max}, w_{th}), \forall i, w_{th}$. Similar to Section III-A, for $s \in \{0, 1, \ldots, w_{\max}\}$, the final state can be found as

$$B = \begin{cases} (i, h, w_{\max}, w_{th}), \\ \quad T(s+1, w_{\max}, w_{\max}) \geq t > \\ \qquad T(s, w_{\max}, w_{\max}), s < w_{\max} \\ (i, b, w_{\max}, w_{th}), \\ \quad t > T(s+1, w_{\max}, w_{\max}), s < w_{\max} \\ (i, g, w_{\max}, w_{th}), \\ \quad t > T(s, w_{\max}, w_{\max}), s = w_{\max}. \end{cases} \tag{8}$$

The first, second, and third equations in (8) correspond to transitions f, e, and d shown in Fig. 3, respectively. The corresponding transition probability, the transition time, the number of packets transmitted successfully and the number of packets transmitted can be similarly computed as in Section III-A.

C. Recovery Due to Transmission Errors

All transitions to the loss recovery due to transmission error happen just before the first packet loss occurs. Therefore, a set of initial states is given as $A = (i, b, w, w_{th}), \forall i, w, w_{th}$. Let $s \in \{0, 1, \ldots, w-1\}$ be the number of packets successfully transmitted out of w initially transmitted packets after the first packet loss and $x \in \{0, 1, \ldots, w-s\}$ be the number of packets successfully retransmitted in the fast recovery phase. Then, given the initial state A, the final state can be found as

$$B = \begin{cases} (i, g, 1, \max(2, \lfloor (w-Z)/2 \rfloor)), \\ \qquad s < Z \text{ or } s \geq Z, x < w - s \\ (i, g, w/2, \max(2, \lfloor (w-Z)/2 \rfloor)), \\ \qquad s \geq Z, x = w - s. \end{cases} \tag{9}$$

The first equation in (9) corresponds to the case when the timeout mechanism is used (i.e., when the total number of duplicate ACKs is less the Z, or any of the retransmitted lost packets get lost again). The second equation in (9) corresponds to the case when the timeout mechanism is not used (i.e., when there is enough duplicate ACKs and all retransmitted packets are successful). Let $\alpha(s,w)$ be the probability of having successfully transmitted s packets out of w transmitted packets given that the packet transmission error rate is p_i. It follows that

$$\alpha(w,s) = \begin{pmatrix} w \\ s \end{pmatrix} p_i^{w-s}(1-p_i)^s. \tag{10}$$

From (9) and (10) the probability that $s \geq Z$ and $x = w - s$ is found as $\sum_{s=z}^{w-1} \alpha(w-1,s)(1-p_i)^{w-s}$. Given the initial state is A and the final state is B, the transition probability can be computed as

$$\phi_{AB} = \begin{cases} 1 - \sum\limits_{s=z}^{w-1} \alpha(w-1,s)(1-p_i)^{w-s}, \\ \qquad\qquad s < Z \text{ or } s \geq Z, x < w - s \\ \sum\limits_{s=z}^{w-1} \alpha(w-1,s)(1-p_i)^{w-s}, \\ \qquad\qquad s \geq Z, x = w - s. \end{cases} \tag{11}$$

The transition time in this loss recovery phase is considered to be an interval between a time slot when the ACK for the lost packet is received (entering the loss recovery phase) and the time slot when the timeout is expired or when a full ACK of retransmitted packet is received (exiting the loss recovery phase). When $s < Z$, the timeout mechanism is used and the transition time is simply the retransmission timeout (RTO). To compute the transition time for $s \geq Z$, we further consider two time intervals: 1) the time interval between entering the loss recovery phase and starting fast retransmit, and 2) the time interval between starting fast retransmit and exiting the loss recovery phase. These time intervals can be computed by conditioning the order and the number of packets successfully transmitted before fast retransmit and successfully retransmitted in fast retransmit and fast recovery. However, with this approach, the analysis can become extremely tedious and complex. In the following, we use the close bounds of both time intervals. We assume that lost packets from the window w are retransmitted and if all are successful the fast recovery ends in a round-trip time RTT; otherwise, if any of them gets lost the retransmission timeout timer will expire after the time interval RTO. Note that with approximation, the recovery time can be higher than approximated. However, the deviation is expected to be reasonable due to the fact that by enabling the SACK option all lost packets in the same sender's window are reported to the WP-TCP sender within one round-trip time. The transition time can be estimated as

$$d_{AB} = \begin{cases} \sum\limits_{s=0}^{z-1} \alpha(w-1,s) RTO \\ \quad + \sum\limits_{s=z}^{w-1} \alpha(w-1,s)(RTT + RTO), \\ \qquad\qquad s < Z \text{ or } s \geq Z, x < w - s \\ 2RTT, \quad s \geq Z, x = w - s, \end{cases} \tag{12}$$

where $RTO = \max\{RTO_{\min}, wMSS/b_i, 2l_i + MSS/b_i\}$ and $RTT = \max\{wMSS/b_i, 2l_i + MSS/b_i\}$. The number of packets transmitted successfully and the number of packets transmitted can respectively be found as

$$s_{AB} = \begin{cases} \sum\limits_{s=0}^{z-1} \alpha(w-1,s)s \\ \quad + \sum\limits_{s=z}^{w-1} \left(s + \frac{\sum_{k=0}^{w-s-1} \alpha(w-s,k)k}{\sum_{k=0}^{w-s-1} \alpha(w-s,k)} \right) \alpha(w-1,s), \\ \qquad\qquad s < Z \text{ or } s \geq Z, x < w - s \\ w, \qquad s \geq Z, x = w - s \end{cases} \tag{13}$$

and

$$m_{AB} = \begin{cases} \sum\limits_{s=0}^{z-1} \alpha(w-1,s)w \\ \quad + \sum\limits_{s=z}^{w-1} \alpha(w-1,s)(w + (w-s)), \\ \qquad\qquad s < Z \text{ or } s \geq Z, x < w - s \\ \frac{\sum_{s=z}^{w-1} (w+(w-s))\cdot\alpha(w-1,s)}{\sum_{s=z}^{w-1} \alpha(w-1,s)}, \quad s \geq Z, x = w - s. \end{cases} \tag{14}$$

D. Recovery Due to Handover

1) Soft Downward Vertical Handover: After observing the transmission process during soft downward vertical handover, we find that fast transmit cannot occur if at least one of the following conditions is true: 1) If the number of burst data packet injected in the new link, which can potentially overtake data packets in the old link, is less than duplicate ACK threshold (Z); 2) If the number of data packets that can be transmitted from WLAN access point in the interval of two consecutive data packet transmitted from the cellular network base station is less than Z; and 3) If the transmission of the last data packets in the old link is completed before the number duplicate ACKs generated by reorder data packet reaches duplicate ACK threshold. In order to compute the minimum duplicate ACK threshold ($Z_{\min}$) over which fast transmit cannot occur, we consider the above conditions in the following cases.

- *Case I: When* $wMSS/b_1 > 2l_1$. Let $Z_i, i = 1,2,3$, denote the minimum Z which satisfies the condition i. Since the burst data packet can only occur if ACK sent through upward WLAN overtakes the ones in cellular network, condition 1 is satisfied when $Z_1 MSS/b_1 + l_2 > l_1$. For condition 2, the transmission time of Z_2 reordered packets in WLAN must be greater than the transmission time of a packet in cellular network. Therefore, condition 2 is satisfied if $Z_2 MSS/b_2 > MSS/b_1$. After initiating the handover, the time elapsed before the completion of transmission of the last data packet in the cellular network is given as $(wMSS/b_1 - l_1)$. And the time needed for duplicate ACKs, generated by reordered data packets, to reach Z_3 is given as $(2l_2 + Z_3 MSS/b_2)$. Therefore, condition 3 is satisfied if $(2l_2 + Z_3 MSS/b_2) > (wMSS/b_1 - l_1)$. The minimum duplicate ACK threshold is computed as $Z_{\min} = \min\{Z_1, Z_2, Z_3, w\}$.

- *Case II: When* $wMSS/b_1 \leq 2l_1$. In this case, the occurrence of fast transmit further depends on the instance that the handover occurs in a round trip time. Therefore, we consider a reasonable upper-bound. Since when Z is equal to or greater than w fast transmit cannot occur, the minimum duplicate ACK threshold is approximated as $Z_{\min} = w$.

Given the set of initial states before handover as $A = (1, h, w, w_{th}), \forall w, w_{th}$, the final state after handover can be found as

$$B = \begin{cases} (0, g, W(w, w, w_{th}), w_{th}), & Z \geq Z_{\min} \\ (0, g, \frac{(w-Z)}{2}, \max(2, \left\lfloor \frac{(w-Z)}{2} \right\rfloor)), & Z < Z_{\min}, \end{cases} \quad (15)$$

The first equation in (15) corresponds to the case in which duplicate ACK threshold is equal or greater than $Z_{\min}$ and therefore fast transmit is not triggered. The second equation in (15) corresponds to the case in which duplicate ACK threshold is less than $Z_{\min}$, and therefore fast transmit is triggered. Note that when fast transmit is triggered, the slow start threshold and congestion window are set to half of the $pipe$ (number of unacknowledged packets $(w - Z)$). The transition probability and the number of packets transmitted successfully can respectively be found as $\phi_{AB} = 1$ and $s_{AB} = w$.

The recovery process for soft downward vertical handover will start right after the handover initiation and end after receiving an ACK of the last packet sent before the handover initiation. Therefore, the transition time is approximately equal to a round trip time experienced by a data packet transmitted through the cellular network and ACK packet returned through the WLAN, which is given as $d_{AB} = \max\{wMSS/b_1 - l_1 + l_0,\ MSS/b_1 + l_1 + l_0\}$. Note that if the WLAN round trip time is extremely small compared to the cellular network round trip time, fast transmit followed by fast recovery can end before the ACKs of packets sent through the cellular network arrive at the WP-TCP sender. In this case, the proposed model slightly overestimates the transition time. The number of packets transmitted includes all in-flight data packets just before handover (which is equal to w) plus unnecessary retransmitted packets, if fast transmit is triggered. Because of the go-back-N retransmission behavior of WP-TCP, the number of unnecessary retransmissions is close to half of the $pipe$ just before the fast transmit (i.e., $(w - Z)/2$). Therefore, the number of packets transmitted can be found as

$$m_{AB} = \begin{cases} w, & wMSS/b_1 \leq 2l_1 \\ w + (w - Z)/2, & wMSS/b_1 > 2l_1. \end{cases} \quad (16)$$

2) Soft Upward Vertical Handover: Let RTO and Y denote retransmission timeout set at the WP-TCP sender just before the initiation of the handover and the time difference between the arrival of the last ACK from WLAN and the first ACK from cellular network at the WP-TCP sender, respectively. Note that the step increase in round trip time suddenly increases the magnitude of Y. During time interval Y, the retransmission timeout timer at the WP-TCP sender counts down the clock. Therefore, false timeouts can occur if $Y > RTO \sum_{k=1}^{n} 2^{k-1}$ where n is the number of consecutive premature timeouts and it can be found as $n = \lfloor \log_2(1 + Y/RTO) \rfloor$. Retransmission timeout can be approximated as $RTO = \max\{RTO_{\min}, wMSS/b_0, 2l_0 + MSS/b_0\}$. In order to compute Y, we consider the following cases.

- *Case I: When* $2l_0b_0/MSS < w$. In this case, the WP-TCP receiver is always busy receiving data packet and sending ACK. Therefore, Y is approximately $(l_1 - l_0)$.
- *Case II: When* $l_0b_0/MSS < w \leq 2l_0b_0/MSS$. If handover occurs when the WP-TCP receiver is busy on receiving data packet and sending ACK packets, then Y is approximately equal to $(l_1 - l_0)$. Otherwise, if handover occurs when the WP-TCP receiver is idle waiting for in-flight data packets to arrive from the WP-TCP sender, Y is approximately $(l_0 - wMSS/b_0 + l_1)$.
- *Case III: When* $w \leq l_0b_0/MSS$. If handover occurs when the WP-TCP receiver is busy on receiving data packet and sending ACK packets, then Y is approximately equal to $(l_1 - l_0)$. Or, if handover occurs when the WP-TCP receiver is idle waiting for in-flight data packets to arrive from the WP-TCP sender, then Y is approximately equal to $(l_0 - wMSS/b_0 + l_1)$. Otherwise, if handover occurs when the WP-TCP receiver is idle waiting for in-flight ACK packets to arrive to the WP-TCP sender, then Y is approximately $(2l_1 - wMSS/b_0)$.

From the above three cases, the value of Y corresponding to the worst, average, or best timeouts performance can be computed. For the sake of brevity, only the worst timeout performance (i.e., consider the maximum value of Y for each case) is considered in this paper. Therefore, the maximum time difference between the arrival of the last ACK from the WLAN and the first ACK from the cellular network at the WP-TCP sender is given as

$$Y = \begin{cases} l_1 - l_0, & 2l_0b_0/MSS < w \\ l_0 - wMSS/b_0 + l_1, & \\ & l_0b_0/MSS < w \leq 2l_0b_0/MSS \\ 2l_1 - wMSS/b_0, & w \leq l_0b_0/MSS. \end{cases} \quad (17)$$

Given the set of initial states before handover as $A = (1, h, w, w_{th}), \forall w, w_{th}$, the final state after handover can be found as

$$B = \begin{cases} (1, g, W(w, w, w_{th}), w_{th}), & Y \leq RTO \\ (1, g, W(w, 1, \frac{w}{2}, \max(2, \lfloor \frac{w}{2} \rfloor)), & \\ & RTO < Y \leq 3RTO \\ (1, g, W(w, 1, 2), 2), & 3RTO < Y. \end{cases} \quad (18)$$

The first equation in (18) corresponds to the case where the timeout does not occur. The second equation in (18) corresponds to the case where a single timeout occurs. The third equation in (18) corresponds to the case where two or more timeouts occur. Note that after a single timeout, the slow start threshold is set to half of the current congestion window and the congestion window is set to one. After two or more timeouts slow start threshold is set to two (the minimum value) and the congestion window is set to one. The transition probability can be found as $\phi_{AB} = 1$ and the number of packets transmitted successfully can be computed as $s_{AB} = w$. The transition time can be written as

$$d_{AB} = \begin{cases} \max\{wMSS/b_0 - l_0 + l_1, \\ \qquad MSS/b_0 + l_0 + l_1\}, & Y \leq RTO \\ (MSS/b_1 + 2l_1) \\ \qquad + (wMSS/b_1 + 2l_1), & Y > RTO. \end{cases} \tag{19}$$

Equation (19) can be explained as follows. If there is no timeout during handover (i.e., when $Y \leq RTO$), the transition time is equal to a round trip time, where the data packet is sent through the WLAN and the corresponding ACK packet is returned through the cellular network. Otherwise, if there is a single or multiple timeouts during handover (i.e., when $Y > RTO$) the transition time is found by summing two round trip times: a round trip time for a data packet sent through the cellular network after handover (MSS/b_1+2l_1) and the round trip time of the new packet sent through the cellular network after retransmitting the entire window ($wMSS/b_1+2l_1$). Note that if there is a single or multiple timeouts, the transition time is considered to start right after the handover initiation and end after receiving an ACK of the new packet sent after the timeout. The packets transmitted include all in-flight data packets just before handover (which is equal to w) and unnecessarily retransmitted packets due to single or multiple timeouts. Because of the go-back-N retransmission behavior of WP-TCP, the number of unnecessary retransmissions is approximately $(n + w - 1)$, where n is the number of consecutive premature timeouts. Therefore, the number of packets transmitted can be found as

$$m_{AB} = \begin{cases} w, & Y \leq RTO \\ w + (n + w - 1), & Y > RTO. \end{cases} \tag{20}$$

3) Hard Downward and Upward Vertical Handover: After hard downward or upward vertical handover all in-flight packets get lost. Under this conjecture, a similar model for both hard downward and upward vertical handovers is developed. Let $\bar{i}$ denote the network other than i (for example, if $i = 1$ (cellular network) then $\bar{i} = 0$ (WLAN)). Therefore, the round trip time experienced by the first packet retransmitted through network $\bar{i}$ (the new network) is $RTT = (2l_{\bar{i}} + MSS/b_{\bar{i}})$ and the WP-TCP sender retransmission timeout is $RTO = \max\{RTO_{\min}, wMSS/b_i, 2l_i + MSS/b_i\}$. Given the set of initial states as $A = (i, h, w, w_{th}), \forall i, w, w_{th}$, the final state can be deterministically found as

$$B = \begin{cases} (\bar{i}, g, 1, \max(2, \lfloor w/2 \rfloor)), & RTT \leq 2RTO \\ (\bar{i}, g, 1, 2), & RTT > 2RTO \end{cases} \tag{21}$$

The transition probability, the transition time, and the number of successfully transmitted packets can be found as $\phi_{AB} = 1$, $d_{AB} = RTO$, and $s_{AB} = 0$, respectively. Since the timeout mechanism is only used to detect and recover lost packets after handover, at least one timeout must occur. After the first timeout, k additional timeouts can occur if $RTT > RTO \sum_{j=1}^{k} 2^j$. The number of additional timeouts can be found as $k = \lfloor \log_2(1 + 0.5RTT/RTO) \rfloor$. The number of packets transmitted is the sum of all in-flight data packets just before handover (which equals to congestion window) and the number of retransmitted packets due to timeouts. Therefore, the number of packets transmitted can be found as $m_{AB} = w + 1 + \lfloor \log_2(1 + 0.5RTT/RTO) \rfloor$.

IV. Performance Evaluation

The proposed analytical model is verified by simulation using two different simulators - our network simulator test-bed [24] and ns-2 simulator [25]. The simulation topology is shown in Fig. 2. A single pair of wireless profiled TCP (WP-TCP) sender-receiver is configured to run over the integrated WLANs and cellular network. FTP is used as an application for unidirectional downlink transfer. WP-TCP is simulated according to the real protocol as described in Section II; the underling network is simulated to capture the fundamental effects of the vertical handover. Therefore, the results from simulations may be slightly inflated compared to those obtained from the real network. However, the trends related to vertical handover are similar. Each simulation scenario is repeated 200 times with different random seeds to arrive at the average results. The simulation and analytical results are obtained by using the parameter values and units given in Table II. Note that the notation for ordered pair (cellular network, WLAN) is consistently used to present network parameters.

In the next subsections, we evaluate the short- and long-term performances of WP-TCP under the influence of vertical handover. For short-term performance, we first analyze the occurrence and the impact of falsely triggered fast retransmits in 60 seconds after the soft upward vertical handover. We analyze the occurrence and impact of premature timeouts in 60 seconds after the soft downward vertical handover. For long-term performance, we analyze the long-run efficiency and throughput under the influence of the hard vertical handovers. In this case, each simulation is run for sufficiently long time to obtain the average performance. Note that if not stated otherwise, BDP denotes the bandwidth-delay product of the network in use before the vertical handover. It is measured in packets and computed as $\text{BDP} = 2lb/MSS + 1$.

From Figs. 4–10, it can be seen that despite of approximations introduced in Section III, the analytical results are in agreement with the simulation results. This verifies the reasonableness of our assumptions and correctness of the analysis.

A. Short-Term Fast Retransmit Performance

The percentage of false fast retransmits and the average number of packets retransmitted after the soft downward handover are investigated at various values of duplicate ACK threshold (Z) and congestion window (w). In each case, the cellular network and WLAN one-way channel delay are set to 0.5 sec and 0.1 sec, respectively. In Fig. 4a, the handover is initiated when w = 6, 12, and 24 packets. It can be seen that when the congestion window (w = 24 packets) is greater than the BDP, the occurrence of false fast retransmits changes from 100% to zero at Z = 8. On the other hand, when the congestion window (w = 12, 6 packets) is less than BDP false fast transmits occur for $Z < w$. To get more insight on this trend, the relationship between the percentage of false fast retransmits and the congestion window is analyzed for Z =

TABLE II
SIMULATION PARAMETERS

Parameters	Range
Mean residence times (λ_1, λ_0)	(200,200) sec
Link bandwidth (b_1, b_0)	(144K,2M) bps
One-way channel delay $(1_1, 1_0)$	(0.01–1,0.01–0.1 sec
Transmission error rate (p_1, p_0)	(0.00–0.15,0.00–0.02)
Network buffer size (B_1, B_0)	$(2w_{\max}, 2w_{max})$
WP-TCP maximum window size $(w_{\max})$	5–65 packets
WP-TCP duplicate acknowledgement threshold (Z)	1–15 packets
Minimum retransmission timeout $(RTO_{\min})$	0.2 sec
WP-TCP maximum segment size (MSS)	1 KB

7 and 8. From Fig. 4b, it can be seen that when $Z < 8$, the percentage of false fast retransmits increases to 100% as w approaches the BDP. However, when $8 \leq Z$, the percentage of false fast retransmits increases and then decreases to zero as w reaches the BDP. These results reveal unexpected trend of the minimum Z which prevents the occurrence of false fast transmits as w increases. Fig. 5 presents the average number of packet retransmitted (if fast transmit occurs) as a function of duplicate ACK threshold. It is shown that the number of retransmitted packets increases as the congestion window increases and decreases as Z decreases. This trend is explained by the go-back-N retransmission behavior of WP-TCP, where the number of retransmitted packets is proportional to the number of outstanding unacknowledged packets $(w - Z)$.

B. Short-Term Retransmission Timeout Performance

Premature timeouts that occur during the soft upward handover is studied. The duplicate ACK threshold is set to 3 and the WLAN one-way channel delay is set to 0.1 sec. The handover is initiated when the size of congestion window is 10, 18, and 30. Fig. 6 presents the percentage of single (#TO = 1) and two consecutive (#TO = 2) premature timeouts as a function of the cellular network one-way channel delay. It can be seen that the lowest cellular network one-way channel delay that can result in single or two consecutive premature timeouts decreases as the congestion window decreases. The observed trend can be explained as follows. The timeouts can occur when the retransmission timeout is less than the maximum time between the arrival of the last ACK from the WLAN and the first ACK from the cellular network. When the congestion window is greater than the BDP, decrease in congestion window decreases the retransmission time but does not affect the maximum time between the arrivals of the two ACKs. And when the congestion window is less than the BDP, decrease in congestion window does not affect the retransmission time but increases the maximum time between the arrivals of the two ACKs.

C. Long-Term Efficiency Performance

The effect of maximum window size $(w_{\max})$ on efficiency is studied in network environments with high and low transmission error rates (p), and small and large BDP. To obtain various values of the BDP, the link bandwidths are fixed and the channel delays are varied. The efficiency as a function of $w_{\max}$ for high error rate p = (0.15, 0.02) and for low error rate p = (0.0, 0.0) are compared in both large BDP l = (0.3s, 0.1s) and small BDP l = (0.01s, 0.01s) environments, as shown in Fig. 7 and Fig. 8, respectively. For high transmission error rate, it is shown that the efficiency remains almost unchanged for small BDP (Fig. 8), whereas for large BDP (Fig. 7), the efficiency increases and converges to the limiting point. To understand this trend, we further observe the packet losses due to vertical handover and transmission for high transmission error rate in small BDP and large BDP environments. It is found that as $w_{\max}$ increases, the proportion of total packets transmitted in cellular network to the packets transmitted in WLAN decreases for large BDP and remains fairly constant for small BDP. Since the transmission error rate is higher in cellular network than in WLAN, the total number of

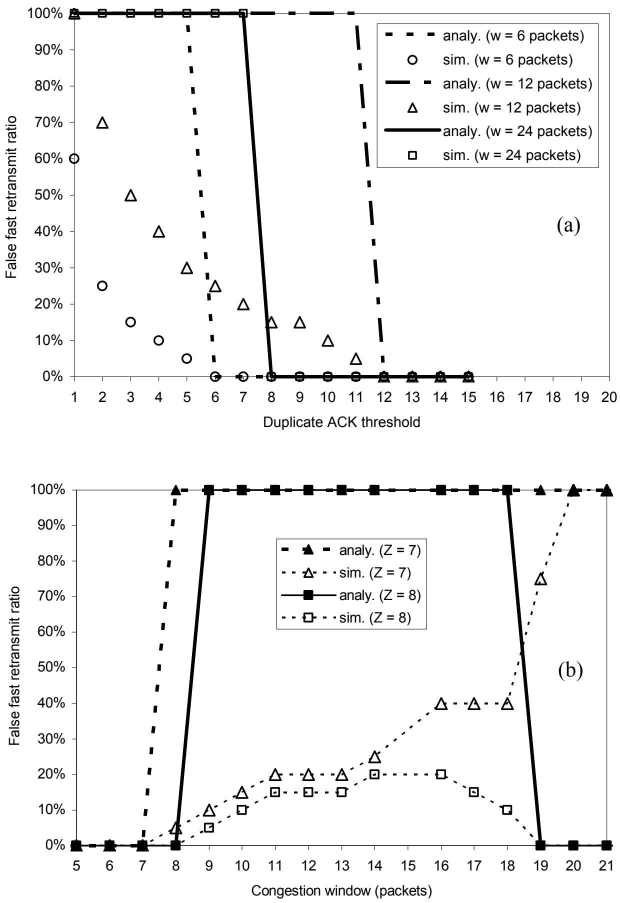

Fig. 4. False fast retransmit ratio vs. (a) duplicate ACK threshold and (b) congestion window for l = (0.5, 0.1) sec.

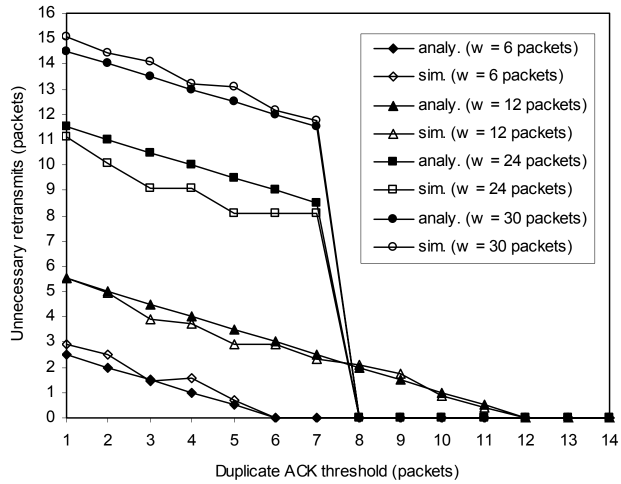

Fig. 5. Average number of packet retransmitted (when fast retransmit occur) vs. duplicate ACK threshold for l = (0.5, 0.1) sec.

packet losses due to transmission error decreases for large BDP and remains almost unchanged for small BDP. For low transmission error rate, the efficiency seems to decrease as $w_{\max}$ increases for both large BDP (Fig. 8) and small BDP (Fig. 7) environments. The number of packet losses in the low transmission error rate environment are observed. It is found that the number of packet losses due to vertical handover significantly increases when $w_{\max}$ increases. This is due to the fact that the congestion window tends to grow to $w_{\max}$ in low transmission error rate environment and hence it closely depends on $w_{\max}$. Since the number of packet losses after the vertical handover is equal to the size of the congestion window, the increase of $w_{\max}$ increases the number of packet losses due to vertical handover and therefore decreases the efficiency.

D. Long-Term Throughput Performance

The impact of maximum window size ($w_{\max}$) on the throughput is studied in the network environment with high and low transmission error rates and small and large BDP. The variations of throughput with $w_{\max}$ for small BDP l = (0.01s, 0.01s) and large BDP l = (0.3s, 0.1s) are compared in low transmission error rate p = (0.0, 0.0) (Fig. 9) and high transmission error rate p = (0.15, 0.02) (Fig. 10) environments, respectively. It can be seen that, when p = (0.0, 0.0), the throughputs for large BDP converges to the throughput for low BDP (Fig. 9), whereas when p = (0.15, 0.02), the throughput for large BDP converges to the point which is far below the throughput for low BDP (Fig. 10). The reason is as follows. For low transmission error rates, a sufficiently large $w_{\max}$ allows the WP-TCP sender congestion window to grow to a large value and that enables the WP-TCP sender to fill in the "data *pipe*". This eliminates the unnecessary idle waiting time due to the channel delay. For high transmission error rate, the WP-TCP congestion control response limits the WP-TCP sender congestion window to a relatively small value regardless of the sized large $w_{\max}$. Consequently, the idle waiting time when l = (0.3s, 0.1s) is significantly larger than that when l = (0.01s, 0.01s).

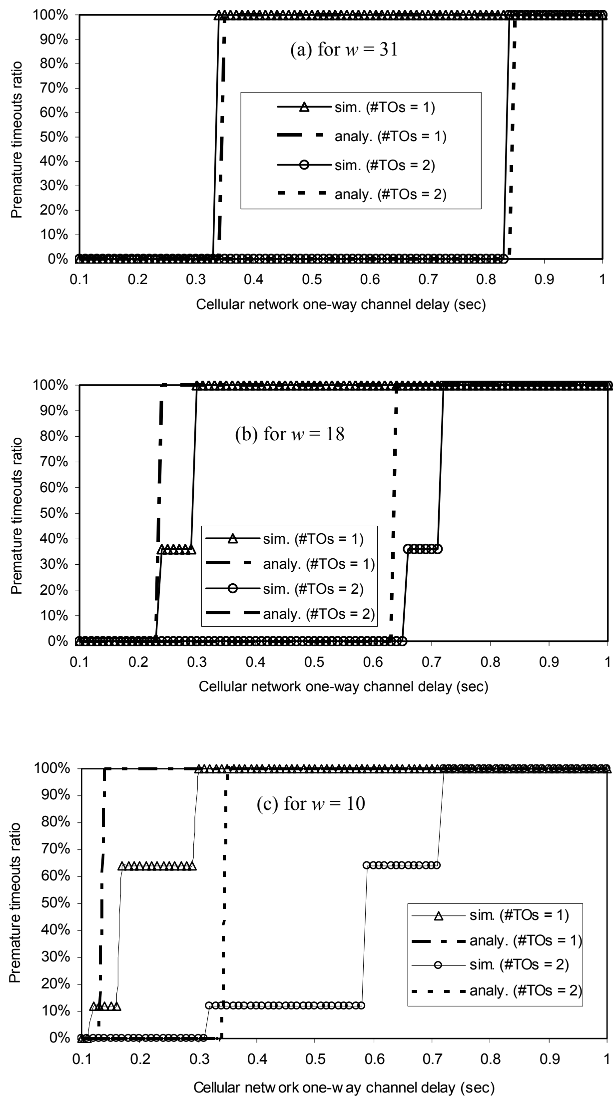

Fig. 6. Premature timeout ratio vs. cellular network one-way channel delay for l_0 = 0.1 sec.

V. Conclusion

In this paper, a novel analytical framework, which describes the short-term performance of WP-TCP during vertical handover and long-term performance of WP-TCP in integrated WLAN and cellular networks, is proposed. Simulations have been given to validate the accuracy of the framework. It is shown that the approximations introduced in the framework have negligible effects on the accuracy of performance analysis. The main observations are as follow. Firstly, when the network is subjected to hard handovers, increasing the maximum window size improves the efficiency in a high transmission error environment, but degrades the efficiency in a low transmission error environment. Secondly, increasing the congestion window reduces the chances of premature timeouts during soft upward vertical handover. Finally, depending on the value of the duplicate ACK threshold, increasing the congestion window can increase or reduce the chances of false fast retransmit during soft upward vertical handover. The proposed analytical framework can be utilized or extended in

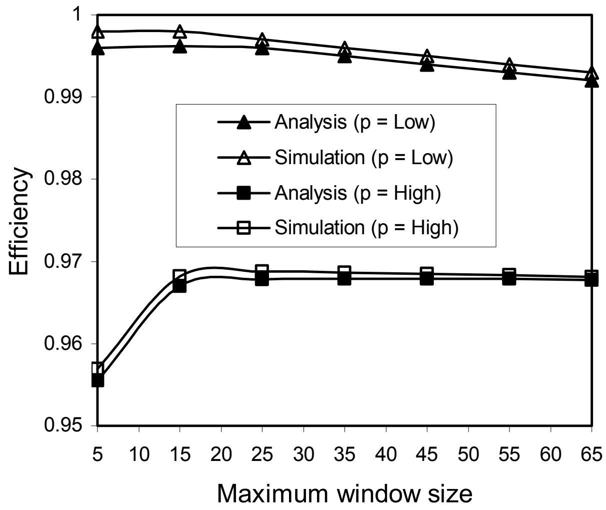

Fig. 7. Efficiency vs. $w_{\max}$ for l = (0.3, 0.1) sec, p = (0.0, 0.0) and (0.15, 0.02).

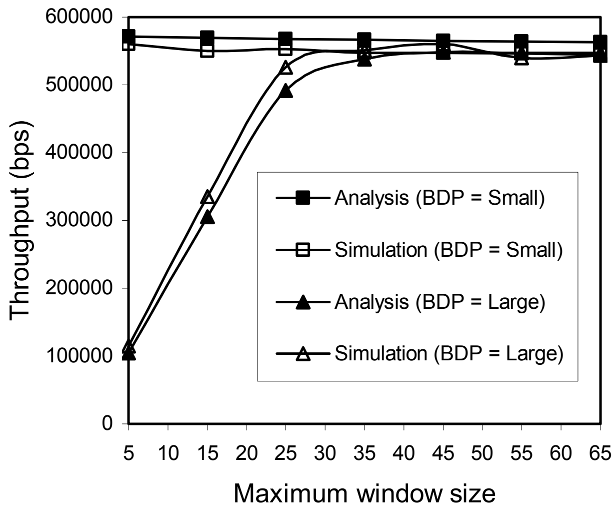

Fig. 9. Throughput vs. $w_{\max}$ for p = (0.0, 0.0), l = (0.01, 0.01) sec and (0.3, 0.1) sec.

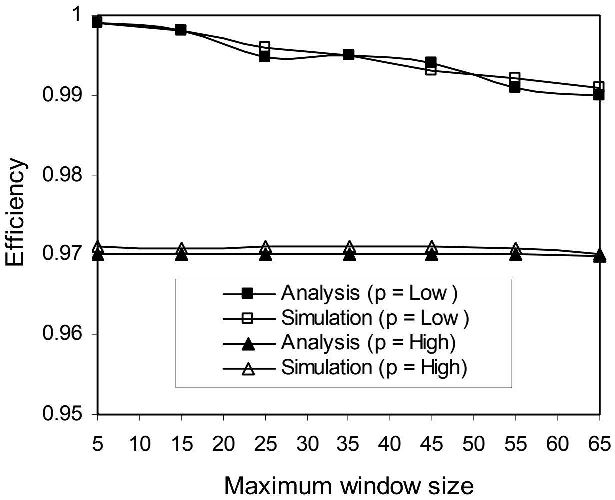

Fig. 8. Efficiency vs. $w_{\max}$ for l = (0.01, 0.01) sec, p = (0.0, 0.0) and (0.15, 0.02).

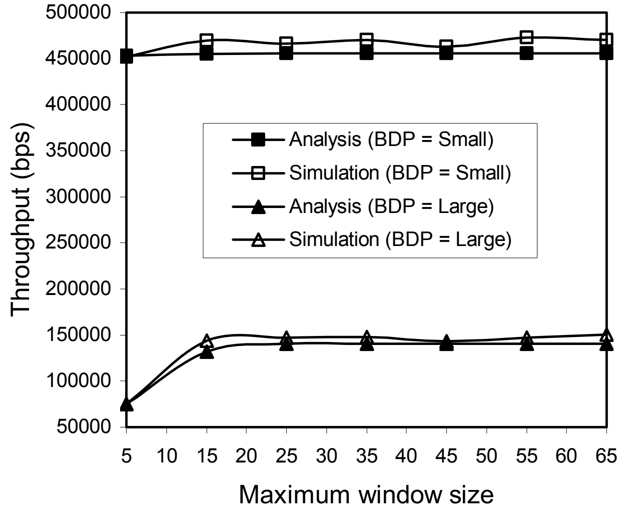

Fig. 10. Throughput vs. $w_{\max}$ for p = (0.15, 0.02), l = (0.01, 0.01) sec and (0.3, 0.1) sec.

the following directions.

1. Cross-layer and adaptive design: The performance of WP-TCP in the integrated WLAN and cellular networks can be improved by adjusting the WP-TCP sender duplicate ACK threshold, congestion window, and retransmission timeout timer, to prevent false timeouts and fast retransmits due to vertical handover. The design that uses cross-layer information at the mobile terminal and utilizes the proposed analytical framework in determining optimal WP-TCP sender protocol parameters needs to be investigated.

2. Networks residence times: In the proposed analytical framework, exponentially distributed random residence times have been assumed. To the best of the authors' knowledge there is no well-established distribution that describe the residence times in the integrated WLAN and cellular networks. A further direction can be on investigating tractable distributions which are closely fit with empirical data.

3. Integrating Markovian channels: The effects of congestion losses due to vertical handover can be captured in the proposed analytical framework by considering the network buffers and analyze the inrush transmission in cellular network or burst packet transmission in WLAN due to soft handover. In addition, the effects of the time varying channel characteristics can be captured by considering i) characterizing the time varying channel delay, and/or bandwidth, and/or error rate as Markov process, and ii) adding appropriate dimensions in the proposed semi-Markov process to track joint "WP-TCP/network" evolution. Note that increasing the dimension of the semi-Markov process can increase the complexity of the model. A good balance between accuracy and complexity is therefore needed.

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