Impact of Scheduling and Buffer Sizing on TCP Performance over IMT-Advanced

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***Abstract*—IMT-Advanced is gaining much attention as the next-generation mobile communication technology. In this study, we evaluate the TCP throughput performance of mobile users competing on an IMT-Advanced shared downlink via a network simulation by using realistic environmental parameters. Our focus is on evaluating the impact of two basic types of time-frequency scheduling in which not only a time-varying wireless condition is used but also a frequency-block-dependent wireless condition of each user is exploited using various buffer sizes at radio base stations and various user moving speeds. We investigated the subtle tension between the scheduling, buffer sizing, and user moving speed for the TCP throughput. Our results imply that more TCP awareness is necessary for sharing the large bandwidth of IMT-Advanced downlinks more efficiently.**

***Keywords*-IMT-Advanced, TCP throughput, MaxCIR scheduler, Proportional Fair scheduler**

I. Introduction

Recently, mobile communications have expanded rapidly worldwide. Third-generation mobile communication systems (3G), including wideband code division multiple Access (W-CDMA) and high-speed packet access (HSPA) are now being used. Moreover, cellular-phone-oriented mass contents and individual user-data traffic are increasing in conjunction with each other every year. Therefore, the demand for high-speed mobile communications is currently increasing. Subsequently, the international mobile telecommunication-Advanced (IMT-Advanced) known as 4G, which attains improvement in communication speed, is also gaining much attention [1]. To achieve high-speed wireless access, IMT-Advanced would need a wireless bandwidth wider than the 5 MHz of the wireless bandwidth of W-CDMA.

Generally, because the wireless condition of each user, i.e., mobile station (MS), varies in time because of the interference, fading, or the movement of the MS, the shared wireless channel for downlink access is divided into fixed-length time slots at every transmission time interval (TTI). A scheduler at the radio base station (BS) allocates individual time slots to MSs that depend on the wireless condition of each MS and by using feedback information from each MS to each BS. Furthermore, in order to utilize the wider bandwidth efficiently, the entire bandwidth is divided into smaller bandwidths (i.e., frequency blocks or *subchannels*); and *time-frequency scheduling* assigns each subchannel to one MS at each time slot, whereas traditional *time scheduling* assigns the entire bandwidth to one MS for every time slot. A subchannel assignment in a wireless link with time-frequency scheduling is illustrated in Fig. 1, where different colors/patterns denote different users (MSs).

The transmission control protocol (TCP) is a dominant transport-layer protocol for end-to-end reliable data transfer, and most of the Internet applications rely on TCP. However, because TCP adaptively controls its sending rate of IP datagrams in response to datagram losses and delay-time variations, its throughput characteristic is highly sensitive to diverse factors, and is difficult to estimate its throughput characteristic analytically when the network condition varies dynamically (e.g, [2]) . On IMT-Advanced environments, the performance of TCP-based applications such as file transfer is expected to be greatly affected by both configurable conditions, such as the type of scheduling mechanism and the buffer sizing of the BS; and environmental conditions, such as the delay time of a core network and the moving speeds of MSs. Therefore, it is indispensable to conduct the evaluation and investigation of the TCP throughput performance of competing mobile users quantitatively.

Performance evaluation of IMT-Advanced communications using network simulators have begun to get more attention (e.g., [3]). However, for TCP flow-level performance of competing mobile users in IMT-Advanced via realistic network simulations, little is found in the literature. In our previous work [4], the TCP performance in the Evolved UTRA and UTRAN environments by using time scheduling and time-frequency scheduling was evaluated using a network simulation by employing realistic packet-error patterns. Building upon that work, here we examine the TCP performance in the new IMT-Advanced environments by using typical time-frequency schedulers (i.e., MaxCIR and Proportional Fair schedulers). Our focus is on evaluating the subtle tension between the scheduling, buffer sizing, and user moving speed for the TCP throughput.

The rest of this paper is organized as follows. In Section II, we introduce two typical scheduling algorithms. Section III explains our network simulation settings for large (continuous) file downloads, and the simulation results are investigated in terms of the impact on the base-station buffer size in Section

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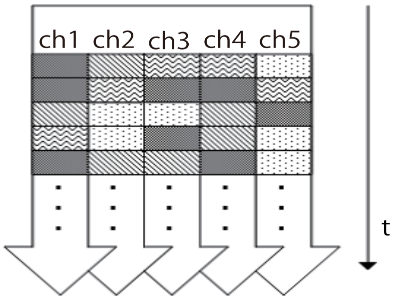

Fig. 1. Subchannel assignment in time-frequency scheduling

IV and the user moving speed in Section V. Concluding remarks are presented in Section VI.

II. Time-Frequency Scheduling in IMT-Advanced

We briefly explain time-frequency scheduling on downlinks in IMT-Advanced involving the BS, the MS, and their interactions. When the BS receives an IP datagram as a service data unit (SDU) from the upper link (i.e., the Internet), it divides the IP datagram into fixed-size small radio link control (RLC) protocol data units (PDUs).

Before being sent to the MS, each PDU is queued to an individual buffer dedicated to each destination user (i.e., MS) before being sent to the MS. Each individual user buffer consists of a transmission buffer and a retransmission buffer. The number of PDUs transmitted at TTI in a subchannel depends on the achievable transmission rate that is determined using the modulation and coding scheme (MCS). The MCSs used in our simulation are shown in Table I, where the right-most column *Transmission Rate* indicates the achievable transmission rate in the entire bandwidth and not in each divided subchannel. If a PDU is damaged during transmission and cannot be fully decoded by the destination user, the original PDU is moved to the retransmission buffer, which is preferentially sent compared with the normal PDUs in the transmission buffer. In our simulation (Section III), to reduce multiple retransmissions, we use a hybrid automatic repeat request (ARQ) with packet combining [5] as a retransmission scheme, which can combine the retransmission PDU with the previously received erroneous PDUs for effective decoding.

At each TTI and for each subchannel, the BS selects one preferable user and one MCS is adopted for sending PDUs to the selected user. This selection is based on the instantaneous wireless conditions, i.e., the single-to-interference plus back ground noise ratio (SINR), of individual users, which are reported by the channel-state information (CSI) feedback from each user to the BS with some delay (e.g., 3×TTI). In our simulation, a mapping from every pair of an SINR value and a sending rate (MCS) to the block error rate, which is the error probability of transmitting PDUs within current TTI, is prepared beforehand. Note that this delay of the SINR value feedback may cause an inaccurate estimation of the current wireless condition, and thus, may result in the improper selection of both, the user to whom data has to be sent and the MCS to be used.

TABLE I
MCS Table

MCS	Modulation Scheme	Encoding Ratio	Transmission Rate [Mb/s]
1	QPSK	1/8	15.128
2	QPSK	1/4	30.344
3	QPSK	1/2	60.8
4	QPSK	2/3	81.04
5	16QAM	1/2	121.76
6	16QAM	2/3	162.32
7	64QAM	1/2	182.64
8	64QAM	2/3	243.6
9	64QAM	3/4	274

Hereinafter, we briefly introduce two typical schedulers that were used in this study. Although those schedulers are simple and some modified or compound versions may be used in commercial services, these fundamental schedulers are of practical importance for designing more complex schedulers. To describe the scheduling, we use a user-selection metric $M_{m,c}(n)$ at time-slot n in subchannel c for user m; this means that the user m who has the highest $M_{m,c}(n)$ is selected in c at n. If more than one user has the same highest value of $M_{m,c}(n)$, a user is randomly selected from these users.

The maximum carrier-to-interference power ratio (MaxCIR) scheduler maximizes the system throughput by selecting a user who has the best wireless condition in the subchannel for that particular time slot. In this study, the usage of the achievable transmission rate $R_{m,c}(n)$ as the wireless condition leads to the following user-selection metric:

$$M_{m,c}(n) = R_{m,c}(n). \tag{1}$$

The Proportional Fair (PF) scheduler (e.g., [6]) provides a good trade-off between maximizing the system throughput and satisfying fairness among users. The PF scheduler in time scheduling can be easily implemented so as to preferentially select a user who has not been served sufficiently in the past (i.e., with a low time-averaged throughput), who has a good wireless condition at that particular time (i.e., with a high achievable transmission rate), and who is proven to approximately fulfill the proportional fairness. However, optimal time-frequency (multi-carrier) PF scheduling is computationally complex, and a number of low-complexity algorithms have been proposed (e.g., [7], [8]) to achieve near-optimal performance. In this study, among these time-frequency PF schedulers, we adopt the conventional multi-carrier proportional fairness scheduling (MC-PFS) algorithm [7], which is a simple extension of the time PF scheduler and is easy to deploy. The user selection metric is formulated as follows:

$$M_{m,c}(n) = \frac{R_{m,c}(n)}{T_m(n)}, \tag{2}$$

where the time-averaged throughput of a user m before a time-slot n is denoted by $T_m(n)$, and is updated according to the following equation in which $\delta_{m,c}$ equals 1 if user m is the selected user at the time slot n in subchannel c, and equals 0

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otherwise:

$$
\begin{aligned}
T_m(n+1) &= (1-\frac{1}{L})T_m(n) + \frac{1}{L}\sum_c R_{m,c}(n)\delta_{m,c}(n), \\
L &= \min(n, t_a),
\end{aligned} \tag{3}
$$

where t_a is a time-scale parameter that represents the number of time slots used for averaging the past throughput.

III. Simulation Model

The simulation model is illustrated in Fig. 2, which is implemented using the network simulator NS-2 (ver. 2.27) with an extended link layer module of the shared wireless downlink we developed. The transmission rate of the wired access links (Source–R) is 1 Gb/s, and propagation delays are set from 1 ms to 5 ms at random to prevent TCP synchronization. The transmission rate of the core link (R–BS) is 1 Gb/s, and propagation delay is set to 10 ms. The shared wireless link from the BS to the MSs of the 80 MHz bandwidth is divided into 400 subchannels in the frequency domain, and is also divided into many time slots with a TTI of 1 ms in the time domain. According to the MCS, the total transmission rate of all subchannels ranges from 15.128 Mb/s to 274 Mb/s (Table I). The actual transmission rate per subchannel can be obtained by dividing it by 400. The data-propagation delay between the BS and each MS, including the processing delay, is 1.75 ms, and the CSI feedback delay from each MS to the BS is 3 TTI (3 ms). We assume that the uplink is error free. Thus, neither data (TCP ACKs) nor signaling (CSI feedbacks) are ever lost or degraded.

The time series of the wireless condition, i.e., SINR, on the link from the BS to each MS in each subchannel at each TTI is given in advance; the time series data is created on the basis of the position (distance from the BS) and the moving speed of the MSs under the 6-path GSM typical urban multi-path model when considering realistic environments. We use data for 30 users distributed around a BS in a signal cell; according to the distance from the BS (from near to far), the users are divided into 10 groups of three users each, implying that the users are roughly sorted by the order of the time-averaged wireless condition (from good to bad) and grouped. User (group) ID-1 is assigned to the nearest group, and ID-10 is assigned to the farthest group. In each simulation trial, we choose one user from each group randomly and conduct the simulation with these 10 selected users.

As shown in Fig. 2, ten TCP flows from Sources (servers on the Internet) to Destinations (MSs) are established in parallel, and each Source sends a large file in 60 s. The starting time of each flow is randomly selected and ranges from 0 to 2 s. Other simulation parameters are listed in Table II.

We use the "user throughput" U_m as a performance index for user m; it is a time-averaged throughput that is formulated as follows:

$$U_m[b/s] = \frac{b_m[bit]}{t_m[s]}, \tag{4}$$

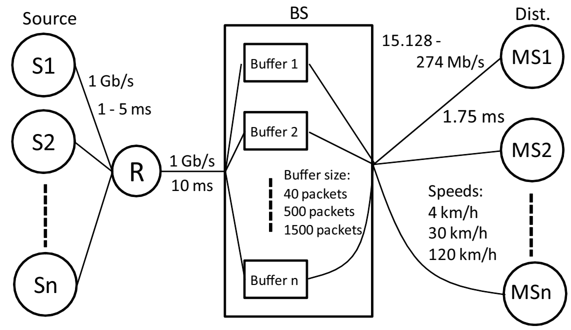

Fig. 2. Simulation topology

TABLE II
SIMULATION PARAMETERS

Parameter	Value
TCP algorithm	New Reno + Sack
TCP packet size	1500 bytes (including IP header)
RLC PDU size (payload + header)	40 bytes + 2 bytes
Max. num of retransmissions	10
Individual buffer size	40,500,1500 packets
TTI	1.0 ms
subchannel bandwidth	200 kHz
Num of subchannels	400

where b_m is the number of received bits of the file of user m at the destination, and t_m is the file-transfer time (from the starting time to 60 s).

Then, "active time ratio" is defined as:

$$active\ \ time\ \ ratio = \frac{a_m[s]}{t_m[s]} \tag{5}$$

where a_m is the sum of the time period during which user m has at least one PDU in the individual buffer on BS.

In addition, the "selection ratio" and "retransmission ratio" are defined as follows:

$$selection\ \ ratio = \frac{s_m}{\sum_{m=1}^{M}(s_m)}, \tag{6}$$

$$retransmission\ \ ratio = \frac{r_m}{u_m}, \tag{7}$$

where s_m is the number of selections of user m, r_m is the number of retransmission PDUs for user m, and u_m is the number of PDUs sent to user m.

IV. TCP Throughput with Varying Buffer Size

We investigate the TCP performance for large file downloads at various buffer sizes (40, 500 and 1500 packets) of individual buffers on the BS.

A. Performance of MaxCIR Scheduler

The user throughputs of 10 individual users (and their average) and the selection ratio are illustrated in Figs. 3 and 4, respectively, by using MaxCIR at each buffer size of all BSs. To maximize the system throughput, the MaxCIR scheduler selects User 1 frequently because it is likely that User1 is the best wireless-condition user among 10 users. In fact, in the case of the buffer size of 500 packets shown in Fig. 4, the selection ratio of User 1 is extremely high; moreover,

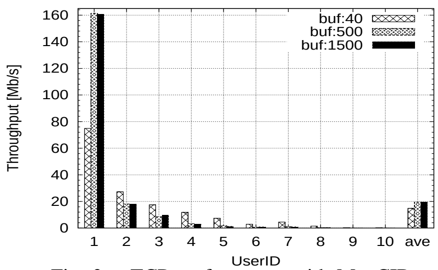

Fig. 3. TCP performance with MaxCIR

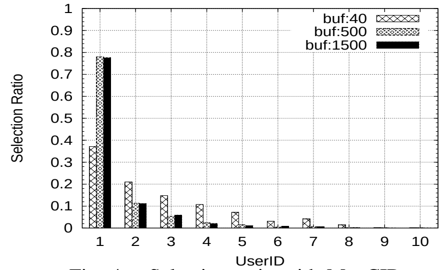

Fig. 4. Selection ratio with MaxCIR

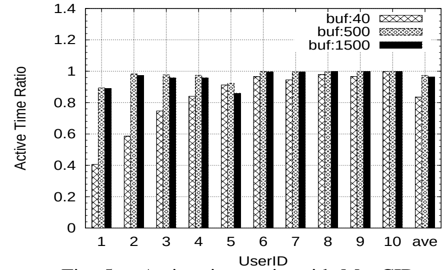

Fig. 5. Active time ratio with MaxCIR

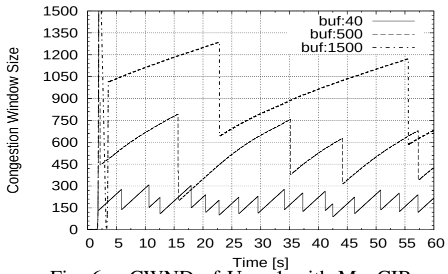

Fig. 6. CWND of User 1 with MaxCIR

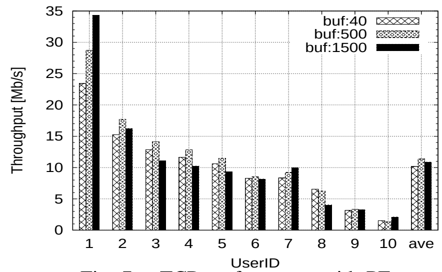

Fig. 7. TCP performance with PF

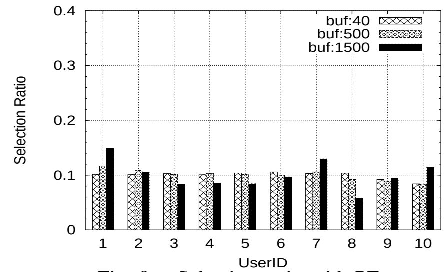

Fig. 8. Selection ratio with PF

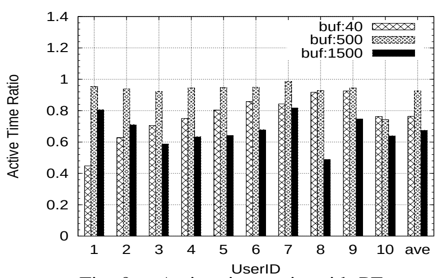

Fig. 9. Active time ratio with PF

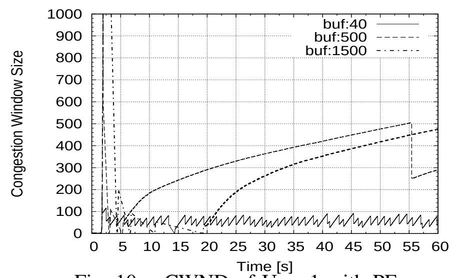

Fig. 10. CWND of User 1 with PF

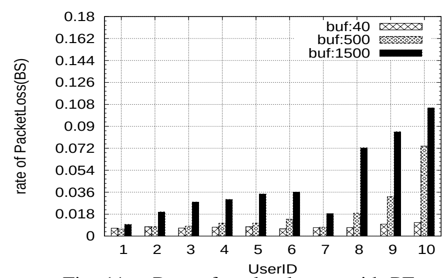

Fig. 11. Rate of packet losses with PF

TABLE III
AVERAGED RTT AND TCP-BANDWIDTH OF USER 1 FOR EACH BUFFER SIZE

buffer size	40 packets	500 packets	1500 packets
RTT	28 [ms]	37 [ms]	65 [ms]
TCP-bandwidth	86 [Mb/s]	175 [Mb/s]	183 [Mb/s]

as the User ID increases, the selection ratio decreases. The high achievable transmission rate and the high selection ratio directly lead to the superior throughput of User 1, as shown in Fig. 3.

However, on comparing the different buffer sizes, this tendency considerably weakens in the case of a shorter buffer size of 40 packets while the tendency remains in the case of a larger buffer size of 1500 packets. The average throughput over all users is also decreased in the case of 40 packets and unchanged in the case of 1500 packets. In order to clarify this tendency, the active time ratio is examined, as shown in Fig. 5. In the case of 40 packets, the active time ratio of User 1 is approximately 40% while in the cases of 500 and 1500 packets, the active time ratios are maintained at approximately 90%. In other words, by using a short buffer size of 40 packets, in approximately 60% of the total time slots, User 1 has no PDU at the BS to be transmitted. Therefore, another user could be selected even if the wireless condition of User 1 is considerably better than that of the selected user, resulting in an increase in throughputs of Users 2-8 in the case of a 40 packet buffer. From Fig. 5, a buffer size of 500 packets is so efficient that it can keep the active time ratio ≈ 1 for all users.

To investigate the reason why the active time ratio of User 1 in the case of 500 packets is much higher than that of 40 packets while it is equivalent to that of 1500 packets, a sample of the time variation of the TCP congestion window size (CWND) of User 1 is shown in Fig. 6. Note that the TCP CWND is a variable and is used to adaptively control the number of allowable in-flight packets at Sender. This figure indicates that the TCP CWND in the case of 500 packets is much higher than that in the case of 40 packets because of a reduction in the IP datagram losses at the buffer on the BS. In other words, a buffer size of 40 packets is so small (for a possible TCP throughput of User 1) that it often causes packet losses at User 1's buffer, limits the TCP CWND, and often makes the buffer empty. Conversely, a buffer size of 500 packets is sufficient to increase the TCP CWND for maintaining a non-empty buffer.

Similarly, the TCP CWND in the case of 1500 packets is much higher than that in the case of 500 packets. However, between the cases with buffer sizes of 500 and 1500 packets, the active time ratio, and thus, the user throughput remain unchanged. In fact, the average round-trip time (RTT) and TCP-bandwidth of User 1 for each buffer size are summarized in Table III, where the "TCP bandwidth" denotes the size of the time-averaged CWND in bytes divided by the time-averaged RTT, implying the time-averaged TCP sending rate.

A larger buffer size of 1500 packets only causes a longer RTT rather than a higher TCP bandwidth. This implies that User 1's TCP bandwidth approaches the wireless bandwidth available to User 1.

B. Performance of PF Scheduler

Throughputs of individual users and their average, and selection ratio are illustrated in Figs. 7 and 8, respectively, with the PF scheduler at each buffer size on the BS. Because the PF scheduler selects users in a fair fashion, in contrast to the MaxCIR case, the selection ratio does not differ much among the users and tends to be 0.1 (implying to be shared evenly), as shown in Fig. 8. The user-throughput balance is expected to approximate the proportional fairness if each user buffer is not empty for all of the time. At least the throughput balance tends to be milder compared with the MaxCIR case at the expense of a lower (approximately halved) user average throughput (i.e., a lower system throughput).

On comparing the different buffer sizes, the impact on the user throughputs is more complicated than the MaxCIR case discussed in the previous subsection. As the buffer size increases from $40, 500, to 1500$ packets, the throughput of User 1 monotonically increases. In contrast, for most users such as Users 2–6, the throughput with a buffer size of 500 packets is the highest among the examined buffer sizes. The average of all user throughputs is also maximized in the case with a 500-packet buffer size. In terms of the active time ratio shown in Fig. 9, for all users, a buffer size of 500 packets is the best case (i.e., closest to one) compared with the other buffer sizes, i.e., 40 and 1500 packets. This implies that the case of the 500-packet buffer size exhibits mostly expected behavior in terms of the proportional fair scheduling.

In further investigation, the subtle impacts of buffer sizing can be observed. For example, in terms of the selection ratio shown in Fig. 8, the selection ratio of User 1 with a 500-packet buffer size is higher than that with a 40-packet buffer size. Accordingly, the throughput of User 1 is improved. A sample of the time-variation in the TCP CWND of User 1, as shown in Fig. 10 indicates that the TCP CWND with a buffer size of 500 packets is much higher than that with a buffer size of 40 packets. In contrast, despite the fact that the selection ratios of Users 3–6 with a 500-packet buffer size were unchanged (or slightly reduced) from those with a 40-packet buffer size, the throughputs with a 500-packet buffer size are higher. Figure 9 implies a possible reason. The active time ratios of Users 3–6 in the case of a 40-packet buffer size are clearly lower than those in the case of a 500-packet buffer. In the case of a 40-packet buffer, because PF scheduler tends to maintain the selection ratio evenly, Users 3–6 are likely to be selected even if their wireless condition is unsuitable. This results in a lower transmission rate and a higher retransmission rate. In contrast, by using a buffer size of 500 packets, because of higher active time ratios, users could be selected at a better wireless condition.

On the other hand, comparing the cases of a 500-packet buffer and a 1500-packet buffer, the selection ratios of Users 1, 7, and 10 in the case of a 1500-packet buffer are considerably higher than those in the case of a 500-packet buffer. Accordingly, those users' throughputs are higher in the case of a 1500-packet buffer while those of the other users are lower in that case. In particular, Users 3–5 and 8 suffer from a considerable reduction. Figure 9 implies a possible reason. For all users, the active time ratios in the case of a 1500-packet buffer are lower than those in the case of a 500-packet buffer. However, the reduction in User 1 is smaller than the reductions in Users 3–5 and 8. Those active-time-ratio reductions might result from packet losses at the buffers on the BS because of a mismatched buffer sizing. A sample of the time variation of the TCP CWND of User 1 (Fig. 10) indicates that the CWND with a 1500-packet buffer size is decreased from that with a 500-packet buffer size because of the increasing IP-datagram losses. In fact, as shown in Fig. 11, the packet loss rate at the individual buffer on the BS increases for every user when the buffer size is changed from 500 packets to 1500 packets. In addition, while User 1's degradation is small, those of Users 3–5 and 8 are significant. Those unnecessary packet losses in the case of a 1500-packet buffer cause imbalance and inefficient behavior of the PF scheduler.

V. TCP Throughput with Varying Moving Speed

We investigate the TCP performance for large file downloads at various moving speeds (3, 30, and 120 km/h) of users, i.e., MSs, in which all 10 users move at the same speed. Note that, in our settings, each MS does not really move to experience a different average wireless condition (i.e., a different distance from the BS) in the cell. Instead, a different moving speed means to choose a different variation in the SINR because of a different Doppler frequency. In other words, each MS is assumed to move along a circle around the BS. The buffer size is set to 500 packets.

On comparing the different moving speeds, it is noted that most users' throughputs and the average decrease as the moving speed increases. More precisely, with the MaxCIR scheduler, as shown in Fig. 12, the throughput of User 1 is strongly reduced at a higher moving speed while the throughput of another user may have the potential to be improved if its selection ratio increases because of the reduction of User 1's active time. On the other hand, by using the PF scheduler, as shown in Fig. 13, the degradation equally appears in each user's throughput because the scheduler tends to maintain the selection ratio evenly.

In order to investigate the reason for this throughput reduction, a sample of the time series of the SINR value averaged over all 400 subchannels for User 1 is examined in Fig. 14, which simply indicates that the SINR is worse (lower) and more rapidly variable at a higher moving speed. The distribution of the selected MCS number for User 1 is illustrated in Fig. 15, where a higher MCS number indicates a better wireless condition. This figure indicates that the number of PDUs transmitted in a higher MCS number decreases and in a lower MCS number increases, as the moving speed increases. This is the first reason for the decrease in user throughputs.

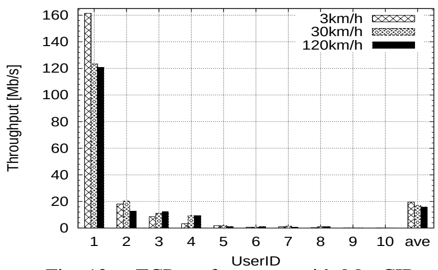

Fig. 12. TCP performance with MaxCIR

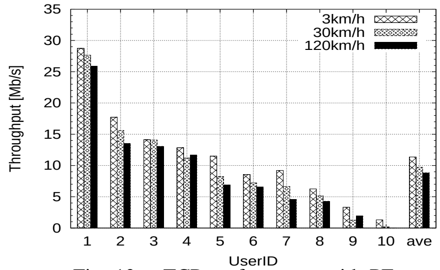

Fig. 13. TCP performance with PF

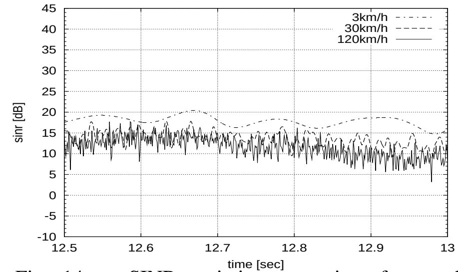

Fig. 14. SINR variation over time for good-condition user (User 1)

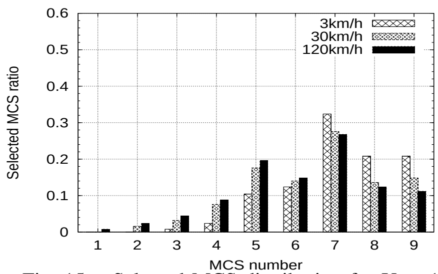

Fig. 15. Selected MCS distribution for User 1

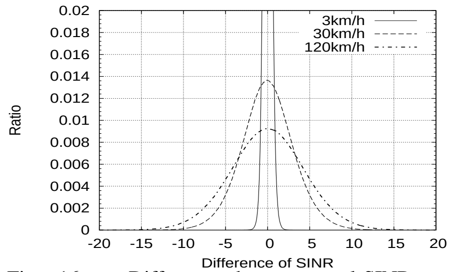

Fig. 16. Difference between real-SINR and advertised-SINR

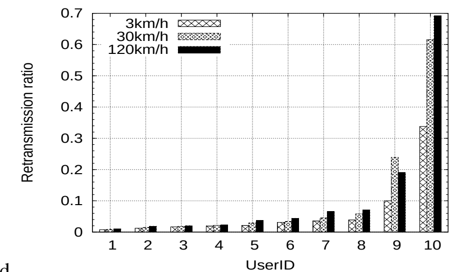

Fig. 17. PDU retransmission ratio of PF

Furthermore, the difference between the real SINR, which directly affects the error rate of the transmitted PDUs, and the advertised SINR, which is reported by CSI feedback and used to select an optimal transmission rate, i.e., an MCS number, is examined, as shown in Fig. 16. This figure indicates that as the moving speed increases, the difference between the real SINR and the advertised SINR increases because of the rapid variation of the SINR; that is, the MCS number selection at the BS based on the advertised SINR is frequently misestimated. In fact, we found that the retransmission ratio increases as the moving speed increases, as shown in Fig. 17. This is another reason for the decrease in user throughputs.

VI. Concluding Remarks

The contribution of this work is two-fold. First, we have evaluated the quantitative TCP throughput performance of individual users on IMT-Advanced by using two typical schedulers with various user moving speeds via realistic network simulations, regarding which little is found in the literature. When 10 users with diverse distances from the BS are competing for the IMT-Advanced shared downlink, the MaxCIR scheduler doubles the total TCP throughput over 10 users compared with PF scheduler, while it allows an extreme imbalance among the users.

Second, we have investigated a subtle tension between the scheduling and buffer sizing for the TCP throughput. We found that the large bandwidth of the IMT-Advanced downlink is not fully utilized by TCP flows. This inefficiency mainly results from the fact that the active time ratio (namely, the probability that at least one PDU is held in the user's individual buffer on the BS and is ready to consume the shared link bandwidth) of each user is lower than one and sometimes even lower, which strongly depends on the user's wireless condition, scheduling, and buffer sizing.

To make the process more efficient, there may be two directions to take. One is to improve the TCP rate-control mechanisms with an awareness of mobile wireless networks, and the other is to improve scheduling with an awareness of TCP flows. For the first direction, because a number of TCP variants have already been proposed for high-speed large-delay and/or wireless networks, we are preparing to investigate the performance of them over IMT-Advanced. Conversely, for the second direction, one simple idea is to change the individual buffer sizes on the BS adaptively and automatically. We will investigate this idea in future work.

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