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PAPER

TCP Flow Level Performance Evaluation on Error Rate Aware Scheduling Algorithms in Evolved UTRA and UTRAN Networks

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SUMMARY We present a TCP flow level performance evaluation on error rate aware scheduling algorithms in Evolved UTRA and UTRAN networks. With the introduction of the error rate, which is the probability of transmission failure under a given wireless condition and the instantaneous transmission rate, the transmission efficiency can be improved without sacrificing the balance between system performance and user fairness. The performance comparison with and without error rate awareness is carried out dependant on various TCP traffic models, user channel conditions, schedulers with different fairness constraints, and automatic repeat request (ARQ) types. The results indicate that error rate awareness can make the resource allocation more reasonable and effectively improve the system and individual performance, especially for poor channel condition users.

key words: *evolved UTRA and UTRAN networks, TCP performance, channel-aware scheduling algorithms, resource allocation, proportional fairness*

1. Introduction

With the rapid advancement of the requirements of high-performance mobile services, the method by which to provide high QoS data services with existing 3G networks becomes more important. Of course, 4 G technology can provide high-speed data transmission. However, considering the large investment to convert 3 G to 4 G completely, it is worth further increasing the performance of existing 3 G technology. In order to satisfy future possible service requirements and ensure the competitiveness of existing 3 G devices over the long term, Evolved UTRA (UMTS Terrestrial Radio Access) and UTRAN (UMTS Terrestrial Radio Access Network) has been proposed [1] to achieve a maximum uplink throughput of 50 Mbps and a maximum downlink throughput of 100 Mbps based on the 3 G spectrum. Also, the mean user throughput, cell-edge user throughput and spectrum efficiency are required to be improved at least by factors 2.

The performances of Evolved UTRA and UTRAN have been investigated with respect to packet-level performance [2], where a given number of users maintain endless traffic transmission. However, the performance evaluation that considers the system and environment impact on the TCP flow-level dynamics remains untouched. Here, flow level dynamics refers to the initiation and completion of finite flows from various traffic generating sources leading to a varying number of concurrent flows competing for the shared resources.

In a previous study, we proposed this error rate aware scheduling and evaluated its TCP packet level performance in HSDPA networks [3]. In order to optimize the system performance by exploiting the time-varying channel conditions, the instantaneous transmission status shall be considered in the scheduling algorithm. Since the system throughput and throughput balance among individual users are the main objectives of data transmission system, the instantaneous transmission rate, as a more directive parameter concerned about it than other channel condition parameters, is generally included in the scheduling metric in order to select the most appropriate user for the current transmission time slot. Note that, since the wireless channel condition is far from error free, the instantaneous transmission rate cannot sufficiently reflect the transmission status. According to the channel conditions, the transmission failure possibility (error rate) may differ when the same transmission rate is applied. Thus, data transmission with a high failure possibility cannot be avoided, especially for poor condition users. To reflect the impact of channel condition sufficiently, the instantaneous transmission rate is replaced by the product of the instantaneous transmission rate and the successful transmission possibility in the scheduling metric in order to avoid transmitting during high-risk transmission time intervals. The quantitative relationship between the wireless condition parameters, such as the Signal-to-Interference plus background Noise power Ratio (SINR) and the transmission failure probability under a given transmission rate, can be established by first computing the existing pre-obtained data and then adjusting this data based on the realistic transmission results obtained in simulations and operations.

In the present study, the effect of error rate aware scheduling algorithms is analyzed in Evolved UTRA and UTRAN via simulation, especially on TCP flow level performance. Note that Evolved UTRA and UTRAN is considerably different from HSDPA in terms of radio technologies and system parameters. TCP service is the most popular service in mobile networks. In addition, since TCP adaptively controls its sending rate in response to packet

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loss and delay time variation, its throughput characteristic is difficult to estimate analytically. Since different definitions of fairness yield different algorithm constraints, several typical scheduling algorithms with different fairness definitions are applied to demonstrate the generality of the performance comparison. Simulation results indicate that error rate awareness effectively improves the system and individual TCP throughput performances, especially for poor condition users.

The remainder of the present paper is organized as follows. Section 2 presents a brief introduction of Evolved UTRA and UTRAN. The scheduling schemes to be used in the simulation are described in Sect. 3. Subsequently, the reconstructed metrics with error rate awareness of each examined algorithm are presented in Sect. 4. The simulation environment is introduced in Sect. 5, and the simulation results are presented in Sect. 6. Finally, concluding remarks are presented in Sect. 7.

2. Introduction of Evolved UTRA and UTRAN

For readers who are not familiar with Evolved UTRA and UTRAN, in this section, a brief introduction of the requirements and realization of Evolved UTRA and UTRAN is given.

The objective of Evolved UTRA and UTRAN is to develop a framework for the evolution of 3 GPP radio-access technology toward a high-data-rate, low-latency, and packet-optimized radio-access technology. For example, in a configuration of two receiving antennas and one transmission antenna at user equipment (UE), the system should support an instantaneous downlink peak data rate of 100 Mbps within a 20 MHz downlink spectrum allocation (5 bps/Hz) and an instantaneous uplink peak data rate of 50 Mbps (2.5 bps/Hz) within a 20 MHz uplink spectrum allocation [4]. In order to achieve these requirements, the evolution of the radio interface as well as the radio network architecture is considered. With respect to data multiplexing, assuming a 10-ms radio frame is divided into 20 equally sized subframes, the OFDM symbols can be organized into a number of physical resource blocks (PRB) consisting of a number of consecutive sub-carriers for a number of consecutive OFDM symbols. User data can be transmitted via block-wise or non-consecutive sub-carriers, which increases the scheduling flexibility in selecting resource blocks and multiplexing users. Thus, in Evolved UTRA and UTRAN, the scheduling granularity becomes per PRB in each subframe for each active flow.

In addition, there are still a number of new technologies adopted by Evolved UTRA and UTRAN, such as multiple-input multiple-output (MIMO) simultaneous Intra-Node-B multi-cell transmission with soft-combining, link adaptation, which enables various modulation schemes and channel coding rates to be applied to the shared data channel, and incremental redundancy hybrid automatic repeat request (HARQ). Among them, HARQ, which combines the high efficiency of Forward Error Correction (FEC) and the reliability of ARQ, is included in order to realize the requirement concerning the minimal guarantee of poor channel condition users and match the multifarious Quality of Service (QoS) requirements for data service [4] by combining the retransmitted packet with previously received erroneous packets for decoding.

3. Introduction of Typical Scheduling Algorithms

In this section, four typical scheduling schemes with different fairness constraints are introduced. These schemes will be used to compare scheduling with and without error rate awareness. In the case of multiple PRBs, each PRB can be viewed as a separate unit to be assigned. To simplify the simulation and concentrate on the comparison with HARQ, in the present study, we ignore the difference among PRBs and assume the simplest case, in which only one transmission is allowed at each subframe with full power. Although whether a subframe is for downlink or uplink can be freely determined in Evolved Utra and Utran networks, in the present study, in order to simplify the simulation model, we assume only the downlink transmission. In other words, we focus on downlink flow dominant cases such as file download in this study. A user is referred to as active in a subframe if the user has data to be scheduled (transmitted) on the base station at the beginning of the subframe. $\mathbf{\Gamma}(n)$ denotes the set of all active users in subframe n.

3.1 Maximizing System Throughput

The scheduling scheme that serves the best channel condition user at each subframe is known as the Maximum SINR algorithm. This algorithm selects user $U(n)$ in subframe n with the best instantaneous channel conditions among all active users, allowing for the highest possible data rate at each instant, thus maximizing the overall throughput. The metric is described as

$$U(n) = \arg\max_{k\in\mathbf{\Gamma}(\mathbf{n})} \mathrm{SINR}_k(n), \text{ or}$$
$$U(n) = \arg\max_{k\in\mathbf{\Gamma}(\mathbf{n})} R_k(n),$$

where $R_k(n)$ is the instantaneous transmission rate for user k $(= 1, 2, \ldots, N)$ in subframe n (≥ 1), and N is the total number of users. Since the first formula considers channel information in more detail than the second one and better channel condition always indicates larger transmission successful probability with the same transmission rate, so it performs a better performance in terms of user-averaged throughput, which is to be pursued in the Maximum SINR scheduling. Therefore, in this study, we show the results based on the first formula (instead of the second one) and employ it for comparison among scheduling algorithms. In what follows, when a tie occurs for a user selection metric, one of the tying users is randomly selected.

3.2 Proportional Fairness for User Throughput

The concept of proportional fairness was proposed in [5] and a scheduling algorithm approximately satisfying this fairness with respect to user throughput was developed by D. Tse to be adopted by Qualcomm's High Data Rate (HDR) system. Proportional fairness satisfies the following property. If another scheduling algorithm is used to increase the throughput of a specific user by $x\%$ beyond that which the user receives under the proportional fairness scheduling algorithm, the sum of the percentage decreases suffered by the throughput of all of the other users under the new algorithm will be greater than $x\%$.

In order to achieve this fairness constraint, the proportional fairness (PF) algorithm schedules user $U(n)$ in subframe n with the largest ratio of the instantaneous transmission rate to the exponentially smoothing average of actual throughput among all active users in the system. This metric is described by

$$U(n) = \arg\max_{k\in\boldsymbol{\Gamma}(\mathbf{n})}\left(\frac{R_k(n)}{T_k(n)}\right).$$

Here, T_k is defined by $T_k(1) = 1$ and

$$T_k(n+1) = \left(1 - \frac{1}{t_c}\right)T_k(n) + \frac{1}{t_c}R_k(n)\mathbf{1}_k(n),$$

where

$$\mathbf{1}_k(n) = \begin{cases} 1 & \text{if user } k \text{ is scheduled at } n \\ 0 & \text{if user } k \text{ is not scheduled at } n \end{cases},$$

and $t_c = \min\{n, c\}$, $n \geq 1$, where c is a predetermined constant. In the present simulation, c is set to 1,000, which is the recommended value for the proportional scheduler [6].

3.3 Time Fraction Fairness

The time fraction fairness is defined in the present paper as the case in which each user has a predefined subframe fraction share over the long term.

The opportunistic algorithm proposed in [7] is a typical channel-aware algorithm with constraints of time fraction fairness. The objective of this algorithm is to maximize the average system performance while trying to maintain the minimal subframe share ϕ_k ($k = 1, 2, \ldots, N$) of each user, where $0 < \phi_k < 1$ and $\sum_{k=1}^{N}\phi_k \leq 1$. The user performance value $X_k(n)$, $k = 1, 2, \ldots, N$ has been introduced to represent the level of performance that would be experienced by user k if that user is scheduled to transmit in subframe n.

To achieve the objectives of this scheme, the algorithm schedules user $U(n)$ in subframe n as follows:

$$U(n) = \arg\max_{k\in\boldsymbol{\Gamma}(\mathbf{n})}(R_k(n) + v_k(n)),$$

where $v_k(n)$ is computed iteratively to satisfy the following fairness constraints:

$$\lim_{T\to\infty}\left(\inf_{T<S}\frac{1}{S}\sum_{n=1}^{S}\mathbf{1}_k(n)\right) \geq \phi_k.$$

To achieve optimal system performance with the above constraint, $v_k(1) = 0$ and $v_k(n)$ is adjusted step by step according to

$$v_k(n+1) = v_k(n) - \frac{1}{t_c}z_k(n), \text{ where}$$
$$z_k(n) = (v_k(n) - \min_j(v_j(n)))(\mathbf{1}_k(n) - \phi_k),$$

where $t_c = \min\{n, c\}$, $n \geq 1$, and c is a predetermined constant.

To ensure $v_k(n)$ convergence, an intuitive algorithm is given as a projection. In each subframe n, if $v_k(n) = \min_j v_j(n)$ and $\sum_{j=1}^{n} 1_k(j)/n < \phi_k$, then $v_k(n)$ is updated according to

$$v_k(n+1) = v_k(n) + \delta,$$

where δ is a positive constant. To accelerate convergence, δ is redefined as a varying parameter in the present simulation:

$$\delta = \gamma \times \left(\phi_k - \frac{1}{n}\sum_{j=1}^{n}\mathbf{1}_k(j)\right).$$

Here, γ is a positive predefined constant.

To provide a unified standard for comparison, the user-averaged throughput and user time-averaged throughput are adopted in the present study to represent the system and user performances. Thus, in the above algorithm, $X_k(n)$ is replaced with $R_k(n)$ and ϕ_k is replaced with $\frac{1}{1.5N}$ in the simulation, and the objective becomes the achievement of equilibrium between the system throughput and the user time fraction fairness.

3.4 Max-Min Fairness

Max-min fairness gives the most poorly treated user (the user with the lowest average transmission rate) the largest possible share without wasting network resources. The wireless-adaptive fairness (WAF) algorithm [8] adds the time-varying channel impact to the traditional Max-min fairness constraint and then selects the user with minimal potential subframe share, assuming that user k will be allocated in the current subframe. The scheduling scheme can thus be formulated as

$$U(n) = \arg\min_{k\in\boldsymbol{\Gamma}(\mathbf{n})}\frac{a_k(n) + s_k(n)}{s_k(n) + a_k(n) + w_k(n)},$$

where $a_k(n)$ is the number of allotted slots for user k at the beginning of subframe n, $w_k(n)$ is the number of allotted slots for all users, excluding k, which are active at the beginning of subframe n, and $s_k(n)$ is the number of slots required to serve one packet at the current data rate. In the present simulation,

$$s_k(n) = \frac{CurrentPacketSize}{R_k(n) \times TramsmissionTimeInterval}.$$

4. Reconstructed Scheduling Algorithms

In order to reduce the impact of the variety of the transmission failure possibility (error rate), we focus on the instantaneous received transmission rate, which is defined as

$$R^*_k(n) = R_k(n)\mathbf{A}_k(n).$$

where $\mathbf{A}_k(n)$ represents the transmission result for user k in subframe n. If the transmission of user k is successful in subframe n, then $\mathbf{A}_k(n) = 1$. Otherwise, $\mathbf{A}_k(n) = 0$. Of course, $\mathbf{A}_k(n)$ is not available during scheduling. However, as proven in [3], the expectation of $R^*_k(n)$ under a given wireless condition $C_k(n)$ can be formulated as follows:

$$P_k(n) = E\{R^*_k(n)|C_k(n)\} = R_k(n)(1 - e_k(n)),$$

where $e_k(n)$ is the error rate under circumstance $C_k(n)$ with transmission rate $R_k(n)$. Then, $P_k(n)$ will be used to replace $R_k(n)$ in the reconstruction of the existing channel-aware scheduling schemes. Since $P_k(n)$ represents the expected instantaneous received transmission rate, the replacement is only changing the perspective from the sending side (base station) to the receiving side (UE), according to the fairness or some other predefined constraint.

The original Maximum SINR algorithm is defined so as to schedule resources to the best-condition user in order to maximize the sender-side throughput. The objective of the reconstructed scheduling policy then becomes the maximum system throughput from the receiver side and selects the user with the highest instantaneous received transmission rate at each subframe. Since $\mathbf{A}_k(n)$ cannot be known before transmission, the expected $R^*_k(n)$ is used to estimate $R^*_k(n)$ in the simulation. The reconstructed metric that can be used in practice thus becomes

$$U(n) = \arg\max_{k\in\mathbf{\Gamma(n)}} P_k(n).$$

All existing algorithms can be reconstructed in the same way. For the proportional fairness algorithm, the new metric is

$$U(n) = \arg\max_{k\in\mathbf{\Gamma(n)}}\left(\frac{P_k(n)}{T_k(n)}\right), \text{ where}$$

$$T_k(n+1) = \left(1 - \frac{1}{t_c}\right)T_k(n) + \frac{1}{t_c}P_k(n)\mathbf{1}_k(n)\mathbf{A}_k(n).$$

For the opportunistic algorithm, the new metric is

$$U(n) = \arg\max_{k\in\mathbf{\Gamma(n)}}(P_k(n) + v_k(n)).$$

For the WAF algorithm, the new metric is

$$U(n) = \arg\min_{k\in\mathbf{\Gamma(n)}} \frac{a_k(n) + s^*_k(n)}{s^*_k(n) + a_k(n) + w_k(n)},$$

where $s^*_k(n)$ is the number of slots required to serve one packet under the current expected received transmission rate, i.e.,

$$s^*_k(n) = \frac{s_k(n)}{1 - e_k(n)}.$$

5. Simulation Environment

5.1 Simulation Model and Parameters

Figure 1 shows the simulation model employed to evaluate the TCP throughput performance. Network Simulator Version 2 (ns-2.28) was used to perform the simulations.

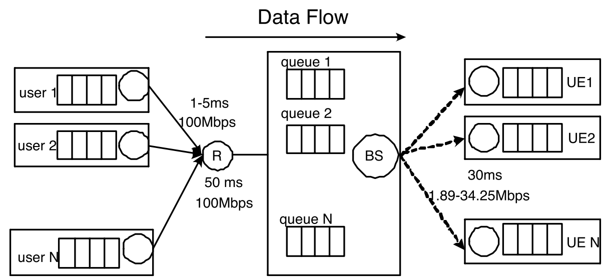

Fig. 1 Simulation model.

Table 1 Simulation environment.

Parameter	value
Cell layout	3 sectored 19 cells
Bandwidth (MHz)	10.0
Multipath model	6-path GSM Typical Urban
Maximum Doppler frequency	5.55 Hz
Path loss factor	3.76
Maximum number of ARQ	10
TCP algorithm	New Reno
TCP packet size	1500 bytes
L2 PDU header/payload size	2/40 bytes
Transmission Time Interval (TTI)	0.5 ms
Waiting Timeout (WT)	7 ms
BS buffer length	40 packets

Table 2 Allowable MCS sets.

Mod.	Coding rate	$R_k(n)$ (Mbps)
QPSK	1/8	1.891
QPSK	1/4	3.793
QPSK	1/2	7.6
QPSK	2/3	10.13
16QAM	1/2	15.22
16QAM	2/3	20.29
64QAM	1/2	22.83
64QAM	2/3	30.45
64QAM	3/4	34.25

The detailed simulation parameters are listed in Table 1. The wired access links have a bandwidth of 100 Mbps and a propagation delay of 1–5 ms (to avoid TCP synchronization). The wired core link is 100 Mbps with a 50 ms propagation delay. The condition of wireless links varies over time, with bandwidths in the range of 1.891–34.25 Mbps, according to the predetermined allowable modulation and coding scheme (MCS) sets listed in Table 2. And wireless links have a propagation delay of 1.75 ms, which includes the processing delay at the base station (BS) and receivers. The BS segments an incoming Internet protocol (IP) packet into radio link control (RLC) protocol data units (PDUs), which are stored in dedicated BS buffers. The physical layer encodes PDUs in each subframe, and thus bit errors occur on a per-subframe basis. Selective repeat ARQ, in which the erroneous packet is discarded and the retransmitted packet is decoded independently, and HARQ, in which the erroneous packet is kept and combined with the retransmitted packet in decoding, are separately employed

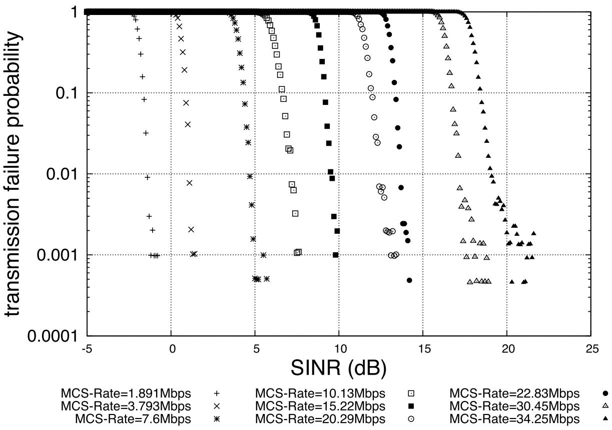

Fig. 2 Error rate distribution.

to evaluate the performance of error rate aware scheduling algorithms.

Since the ARQ scheme may cause out-of-order service data unit (SDU) transmission, a reorder buffer with a timeout timer at each receiver is employed so as to preserve the integrity of the SDU. When an out-of-order delivery occurs, the timer is started and the receiver retains the correctly received SDUs waiting in the reorder buffer until the overtaken SDU arrives during the timeout interval. If the timer expires before the overtaken SDU arrives, the SDUs in the reorder buffer are forwarded to the upper layer, and recovery is conducted by the upper layer. The timeout value denoted by the waiting timeout (WT) is defined as 7 ms. The size of the reorder buffer is considered to be infinite in order to prevent SDU losses at receivers, but it is actually upper-bounded because the stored packets are flushed immediately after the timer expires.

The TCP variant used here is NewReno [9], which is most widely used in the current Internet.

5.2 Relationship between Error Rate and Real-Time SINR

In order to illustrate the influence of SINR variation on the error rate, we attempt to establish the relationship between the real-time SINR subranges and the error rates under various transmission rates, as shown in Fig. 2. According to the transmission records over a long period, an experiential relationship between SINR subranges and the error rate under each transmission rate can be easily calculated. In the present simulation, when the total number of experimental transmission records exceed a few millions, the error rate within a small SINR subrange (0.1 dB) becomes stable.

Figure 2 suggests that if SINR thresholds by which to determine which modulation and coding scheme ought to be selected are set suitably and the real-time SINR is estimated accurately, the difference between the instantaneous transmission rate and the corresponding expected received transmission rate will, except for the poorest channel condition area, be controlled within a very small range. In the other words, the error rate aware scheduling algorithms' sensitive range is mainly distributed in the poor channel condition area. Thus, it is expected that error rate awareness will largely influence poor channel condition users. Although the real-time channel condition information is not available due to feedback and processing delay, as recent work on channel modeling and prediction has shown the feasibility of the prediction of Rayleigh fading channels with very high accuracy [10]–[12] and the duration of a subframe is very small, we ignore the impact of channel prediction accuracy in the simulation.

As shown in Fig. 2, when the channel condition is too poor (the SINR value is less than -2.7 dB), it is impossible to transmit successfully. HARQ is introduced to solve this problem. HARQ maintains the error packet and combines it with the retransmitted packet to increase the possibility of correct reception. When HARQ is applied, in order to avoid starving the transmission condition with nearly zero single successful probability, in this simulation, we predefine the SINR marginal condition threshold as -2.7 dB, and minimum acceptable transmission successful probability is 10^{-5}. When the instantaneous SINR of a user is less than -2.7 dB, the error rate of the multiple transmissions is considered. The minimum retransmission times which enables the transmission successful probability to exceed 10^{-5} will be used in the calculation. To avoid unnecessary and unfair precedence over the other users, the successful possibility will be divided by the assumed total transmission times. The above description can be mathematically expressed as follows. Let X denote the number of transmissions needed before the packet will be successfully received by the receiver with HARQ, and $e_k(n; j) = P(X > j|C_k(n))$, namely, user k's transmission failure probability after j times transmission with HARQ under the current channel condition $C_k(n)$. Then let L be the minimum number which satisfies $\frac{1}{L}(1 - e_k(n; L)) = \frac{1}{L}P(X \leq L|C_k(n)) > 10^{-5}$. In this occasion, we redefine $P_k(n)$ as $\frac{R_k(n)}{L}(1 - e_k(n; L))$. When $L = 1$, it reduces to Section 4's format, $P_k(n) = R_k(n)(1 - e_k(n))$.

6. Simulation Results

6.1 Packet Level Performance

To investigate the effect of error rate awareness, we first examine the TCP packet level performance.

6.1.1 Simulation Occasion

In this subsection, we examine TCP packet level performance for four simulation conditions involving all of the scheduling schemes mentioned in Sect. 3, i.e., selective repeat ARQ without error rate awareness (SR without error rate), selective repeat ARQ with error rate awareness (SR with error rate), packet combining hybrid ARQ without error rate awareness (HARQ without error rate), and packet combining hybrid ARQ without error rate awareness

(HARQ without error rate).

As shown in Sect. 5.2, if the SINR value of a user is less than −2.7 dB in this simulation, the transmission cannot succeed without the application of HARQ. To ensure that all simulated users will have the opportunity to transmit successfully in all occasions, we exclude users that experience a near-zero transmission successful possibility environment at all times in the simulation model. Depending on this assumption, a total of 30 users were selected from three sectors in a cell. We then evenly classified these users into 10 groups according to the time average SINR. The user within each group having the smaller group number experiences better channel conditions in the simulation. We selected one user from each group in each simulation experiment. The SINR distributions of five typical groups among the 10 total groups during the simulation are shown in Fig. 3. In the figure, the total SINR range is divided into 451 subranges, $(-20.0, -19.9)$, $[0.0, 0.1)$, $\ldots$, $[24.9, 25.0)$, and the possibility that a user experiences all subranges is calculated.

6.1.2 Throughput Performance

In this subsection, we use the user-average system throughput to represent the total system performance. The value of average system throughput in all of the simulation conditions are listed in Table 3. In all cases, the order of TCP user-average throughput performances of these algorithms are identical: Maximum SINR, opportunistic, proportional fairness, and wireless adaptive fairness algorithm. This sequence is dependant on the fairness constraints of scheduling schemes, which is not influenced by the application of error rate awareness or HARQ.

Except Maximum SINR, SR with error rate performs poorer TCP user-average throughput than SR without error rate within a specific scheduling scheme. Whereas HARQ with error rate always performs better than HARQ without error rate. Moreover, except Maximum SINR, HARQ without error rate performs poorer than SR without error rate. It seems that error rate aware scheduling or HARQ may harm the system performance, although error rate awareness can improve the system performance when HARQ is applied. To find the reason for this conflict, we first examine the radio link layer (L2) throughput performance in Table 4. Different from TCP performance, both HARQ and error rate awareness do positive effect on L2 throughput performance. These improvements are reasonable considering the working mechanism of HARQ and error rate awareness. It is clear that if all users always have endless traffic in BS, HARQ and error rate awareness are able to improve the system performance. Considering the TCP congestion mechanism, HARQ or error rate aware scheduling sometimes enhances the active user set. For the fair sensitive scheduling algorithms, this enhancement brings poor channel condition users more opportunity to transmit. But for the fair blind scheduling algorithm, MaxSINR, this enhancement will do benefit to the system performance. Thus the better TCP system performance of fair sensitive algorithms in SR without error rate cases shall be the result of the more frequent transmission interruption of poor channel condition users. We examined the time-average TCP throughput performance of each user according to the scheduling schemes. As shown in Fig. 4, SR without error rate achieves better throughput performance than the other three conditions for most good and medium channel condition users. But, for the poorest three users, its performance becomes the poorest in all scheduling algorithms. As shown in Sect. 5.2, the sensitive range of error rate aware scheduling algorithms is mainly distributed in the poor channel condition area. In addition, HARQ combines previous error packets and the retransmitted packet in order to improve the transmission efficiency. Considering

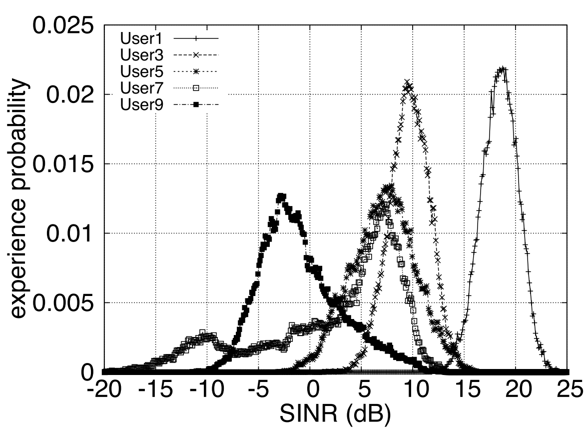

Fig. 3 SINR distribution histogram of five typical groups in packet performance evaluation.

Table 3 Average TCP packet level system throughput.

Scheduling Scheme	SR without error rate	SR with error rate	HARQ without error rate	HARQ with error rate
Maximum SINR	16.85 Mbps	16.96 Mbps	17.22 Mbps	17.51 Mbps
Proportional Fairness	11.07 Mbps	10.37 Mbps	9.53 Mbps	10.11 Mbps
Opportunistic Algorithm	11.76 Mbps	11.61 Mbps	10.13 Mpbs	10.76 Mbps
Wireless Adaptive Fairness	11.03 Mbps	9.05 Mbps	9.41 Mbps	10.10 Mbps

Table 4 Average packet level radio link layer (L2) system throughput.

Scheduling Scheme	SR without error rate	SR with error rate	HARQ without error rate	HARQ with error rate
Maximum SINR	17.31 Mbps	17.41 Mbps	17.32 Mbps	17.39 Mbps
Proportional Fairness	10.51 Mbps	12.98 Mbps	10.65 Mbps	12.10 Mbps
Opportunistic Algorithm	12.86 Mbps	12.99 Mbps	13.00 Mbps	13.03 Mbps
Wireless Adaptive Fairness	8.61 Mbps	10.22 Mbps	8.76 Mbps	8.90 Mbps

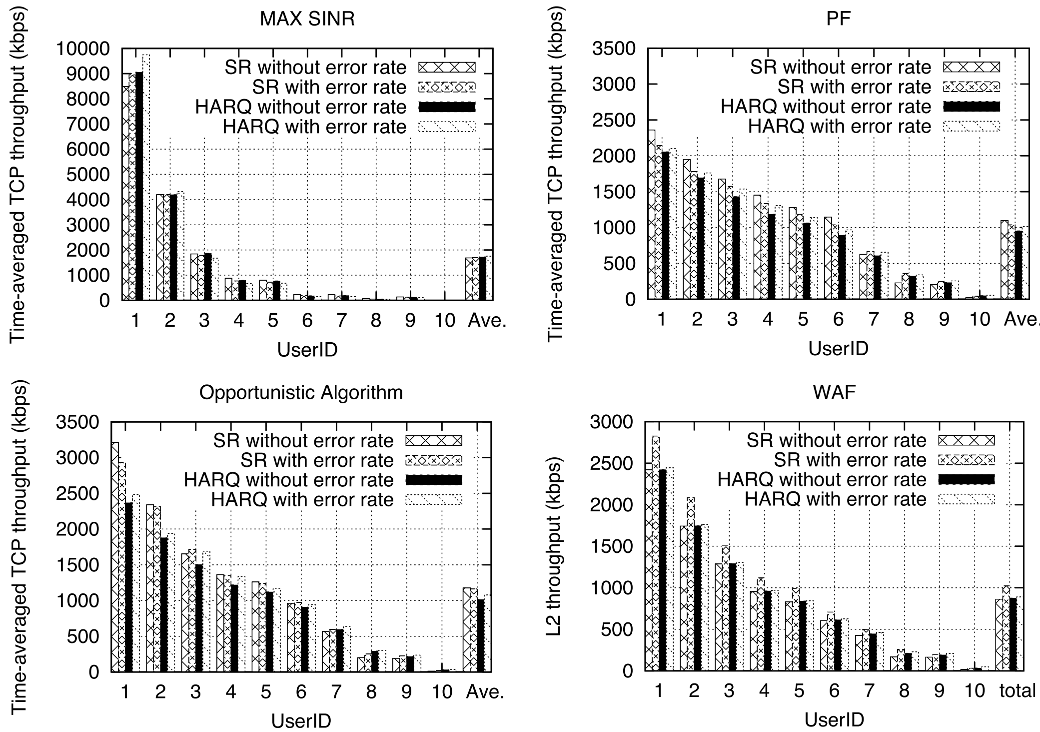

Fig. 4 TCP packet-level per-user throughput comparison (Ave. indicates the user-averaged throughput).

Table 5 Average TCP throughput of the poorest condition user (packet level performance).

Scheduling Scheme	SR without error rate	SR with error rate	HARQ without error rate	HARQ with error rate
Maximum SINR	0.33 kbps	1.71 kbps	1.55 kbps	2.72 kbps
Proportional Fairness	21.83 kbps	41.00 kbps	49.01 kbps	55.28 kbps
Opportunistic Algorithm	8.34 kbps	21.80 kbps	32.37 kbps	38.12 kbps
Wireless Adaptive Fairness	12.75 kbps	31.43 kbps	29.58 kbps	43.40 kpbs

the relationship between SINR and transmission failure possibility, the most effective range of HARQ also happens at poor channel condition areas. Thus, it is reasonable that error rate aware scheduling or HARQ has more impact on user that always experience poor channel conditions.

Error rate aware scheduling algorithms will assign poor users to transmit at comparatively large successful possibilities, and HARQ will allow poor users to obtain correct packets by multiple retransmissions. Both error rate awareness and HARQ will contribute to maintain the TCP connection for poor condition users. On the other hand, SR without error rate will sometimes make TCP connections break down under poor channel conditions. In Table 5, the throughput value of the poorest channel condition users under all simulation conditions is provided. Although the system performance decreases when error rate awareness or HARQ is applied, their effects on improving poor channel condition user throughput are remarkable. The largest improvement ratio of poor users occurs in the case of Maximum SINR and the opportunistic algorithm. Maximum SINR is the least fair algorithm, in which poor condition users are always starved. Thus, when error rate aware scheduling algorithms or HARQ is applied, the poor condition users can achieve obvious improvements. The opportunistic algorithm is based on the principal that scheduling resources to the best condition users when the minimum time fraction of other users is satisfied. Thus, when the TCP connection of poor channel condition users is broken, extra subframes will be assigned to the best channel condition users. However, when the connection can be persistently maintained by error rate aware scheduling algorithms or HARQ, the minimal share of the poor condition users is guaranteed by the fairness constraints. Therefore, the poor condition users with the opportunistic algorithm also achieve a distinct throughput increase when error rate aware scheduling algorithms or HARQ is applied.

From the above observation, we can conclude that the application of error rate awareness on selective repeat ARQ can provide TCP performance comparable to that of HARQ. Without changing the existing system or UE, this provides an alternative method of satisfying services for poor channel condition users without degrading the transmission efficiency. Compared with HARQ, there are still limitations of the proposed method. First, the proposed method only works on channel aware scheduling schemes. If the applied scheduling scheme does not consider the channel condition, as in the case of the round robin scheme, the scheme will no longer work. In addition, under very poor channel conditions, for example, the user always experiences a near-zero transmission successful possibility environment, which cannot be improved.

As shown in Fig. 4 and Table 5, for the case in which HARQ is applied, error rate awareness increases the throughput performance for nearly all users with proportional fairness or the WAF algorithm. In addition, error rate awareness increases all poor condition users and some good or medium condition users with Maximum SINR or the opportunistic algorithm. Moreover, error rate awareness improves the system TCP throughput performance with all scheduling algorithms, which illustrates that error rate awareness has the capacity to improve the system and individual TCP throughput performances when the activity of all of the TCP connections can be maintained continually.

6.2 Flow Level Performance

In order to further investigate the effect of error rate awareness, the TCP flow level performance will be examined in this subsection. In order to measure the flow-level TCP throughput performance, the TCP throughput of each user is defined as follows:

$$Th_i = \frac{\sum_j L_j(i)}{\sum_j t_j(i)},$$

where Th_i denotes user i's TCP flow-level throughput, $L_j(i)$ denotes the file size of the jth file of user i, and $t_j(i)$ denotes the end-to-end duration to transmit the jth file of user i.

6.2.1 Simulation Occasions

In the present study, in order to investigate the TCP flow level performance, we assume that each user repeatedly downloads files with a variety of sizes and a variable sleep interval, which is emulated as web surfing. The service layer file size obeys a pareto distribution with a shape parameter of 1.294 and average sizes of 500 kB, 100 kB, and 20 kB. In addition, the sleep interval obeys a Poisson distribution with an average interval of 3.2 seconds.

As shown in Section 6.1, scheduling with error rate awareness will achieve a TCP performance comparable to HARQ when no user continually experiences a zero transmission successful possibility. In the TCP flow-level performance study, we compared the application of HARQ with and without error rate awareness.

In the simulation, 90 users are distributed in three sectors in a cell. We evenly classify a total 90 users into 30 groups according to the statistic SINR. The user with the smaller group number experiences better channel conditions in the simulation. In addition, the users in the last three groups always experience the poorest channel condition, where no single transmission can succeed. In each simulation experiment, we randomly selected one user from each group. To clearly indicate each group, we provide the SINR distribution of scattered 10 groups during the simulation in Fig. 5. In the figure, the total SINR range is divided into 601 subranges, $[-30.0, -29.9)$, $[0.0, 0.1)$, …, $[29.9, 30.0)$, and the possibility that a user experiences all subranges is calculated.

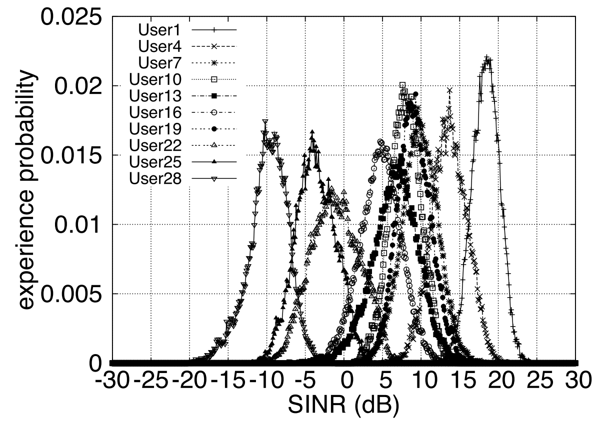

Fig. 5 SINR distribution histogram of ten typical groups in flow level performance evaluation.

6.2.2 System Performance

We define the user-average TCP throughput as

$$Th_{avg} = \frac{1}{N}\sum_{i=1}^{N} Th_i$$

to evaluate the system performance on different occasions. The user-average flow-level throughput in all simulation occasions with four typical scheduling algorithms are listed in Table 6.

First, we focus on the difference among scheduling algorithms. Among nearly all cases, the order of TCP system throughput performances of these algorithms is identical: Maximum SINR, opportunistic, wireless adaptive fairness, and proportional fairness algorithm. This is different from the sequence of the packet-level throughput performance as illustrated in Sect. 6.1.2, which is as follows: Maximum SINR, opportunistic, proportional fairness, and wireless adaptive fairness algorithm. In general, proportional fairness performs better than wireless adaptive fairness on the packet-level user-average throughput performance. For max-min fairness, there is a tendency to assign resources to the poorest treated users. Here, more resources are always provided to the poor condition users compared to other scheduling algorithms, when greedy traffic model is applied. However, when a limited file size is applied, the WAF scheduling algorithm no longer obeys as the max-min fairness. Recall the WAF scheduling metric:

$$U(n) = \arg\min_{k\in\mathbf{\Gamma(n)}} \frac{a_k(n) + s_k(n)}{s_k(n) + a_k(n) + w_k(n)}.$$

a_k and $a_k(n) + w_k(n)$ are used to reflect the previous status of resource allocation. In the case of the greedy traffic model,

Table 6 User averaged TCP throughput performance comparison.

	File size	HARQ without error rate	HARQ with error rate	Improvement ratio
Maximum SINR	500 kB	940.6 kbps	939.7 kbps	−0.1%
	100 kB	638.2 kbps	631.3 kbps	−1.1%
	20 kB	258.4 kbps	290.3 kbps	12.3%
PF	500 kB	436.5 kbps	482.3 kbps	10.5%
	100 kB	400.0 kbps	419.7 kbps	4.9%
	20 kB	252.2 kbps	275.7 kbps	10.0%
OPP	500 kB	897.6 kbps	938.5 kbps	4.6%
	100 kB	594.9 kbps	632.8 kbps	6.4%
	20 kB	262.8 kbps	278.6 kbps	6.0%
WAF	500 kB	629.9 kbps	650.6 kbps	3.2%
	100 kB	563.4 kbps	580.0 kbps	3.0%
	20 kB	266.6 kbps	294.8 kbps	10.6%

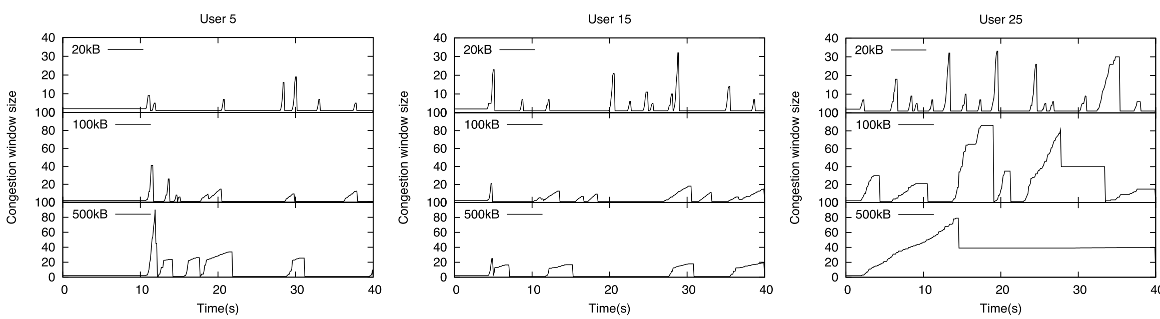

Fig. 6 Congestion window size variation comparison of proportional fairness.

all users start and terminate TCP connection at the same time. Thus, a_k of all users are calculated from the same start time. When limited file size is applied, the new incoming user has no past allocation slots and will be misjudged as the poorest treated user by the scheduling metric. Users with very poor channel conditions generally need more subframes to complete a file, and thus need more allocated subframes, which provides fewer opportunities to transmit than in the case of the greedy traffic model. Whereas the proportional fairness uses the exponential smoothing average of the actual throughput of each user to reflect the fairness constraint, the average throughput of new incoming users will not be influenced by the existing users. Thus, in case a limited file size is applied, the proportional fairness constraint also works well and eventually leads to different TCP throughput performance sequences compared to packet level performance.

As shown in Table 6, with the decrease of the traffic loads, the difference between scheduling algorithms becomes smaller. For the case in which the average file size is 20 kB, there is little difference among scheduling algorithms, because the competitions among active users are comparatively low when the traffic loads are very low. As mentioned before, error rate awareness works only in the SINR sensitive ranges. With reasonable selection of SINR thresholds to determine the instantaneous transmission rate, the most effective range of the error rate awareness occurs for the poorer channel condition part of the smallest transmission rate. As shown in Table 6, when the traffic loads decrease, the effect of error rate awareness becomes obvious because in the case of the slight traffic model, poor condition users have more opportunities to transmit, and thus have more opportunities to experience the sensitive SINR ranges of the error rate awareness.

In addition, there is a distinguished reduction on the throughput performance, as the average file size becomes smaller. Since the same idle interval distribution is applied in all cases, the occasions with small average file sizes will have less competition and should complete a file with high throughput. However, in fact, the case with a higher average file size achieves higher TCP throughput. To explain this phenomenon, we examine the variation of the congestion window size depending on different average file sizes in the case of HARQ with error rate. The congestion window variation of three typical users in a specific simulation experiment under proportional fairness is shown in Fig. 6. When the average file size is small, the duration spent on increasing the congestion window size occupies a comparatively large share of the total transmission time. Thus, the transmission capacity cannot be fully exploited if the transmitted file size is small when TCP is applied.

6.2.3 Individual Performance

To examine the behavior of each user with and without error rate awareness, we provide a detailed comparison of the throughput of each user for the case in which the average file size is equal to 100 kB, as an example. As shown in Fig. 7, the error rate aware scheduling algorithms do not have obvious advantages over the original algorithms for users with good and medium channel conditions. However, for the poor channel condition users, all of the error rate aware

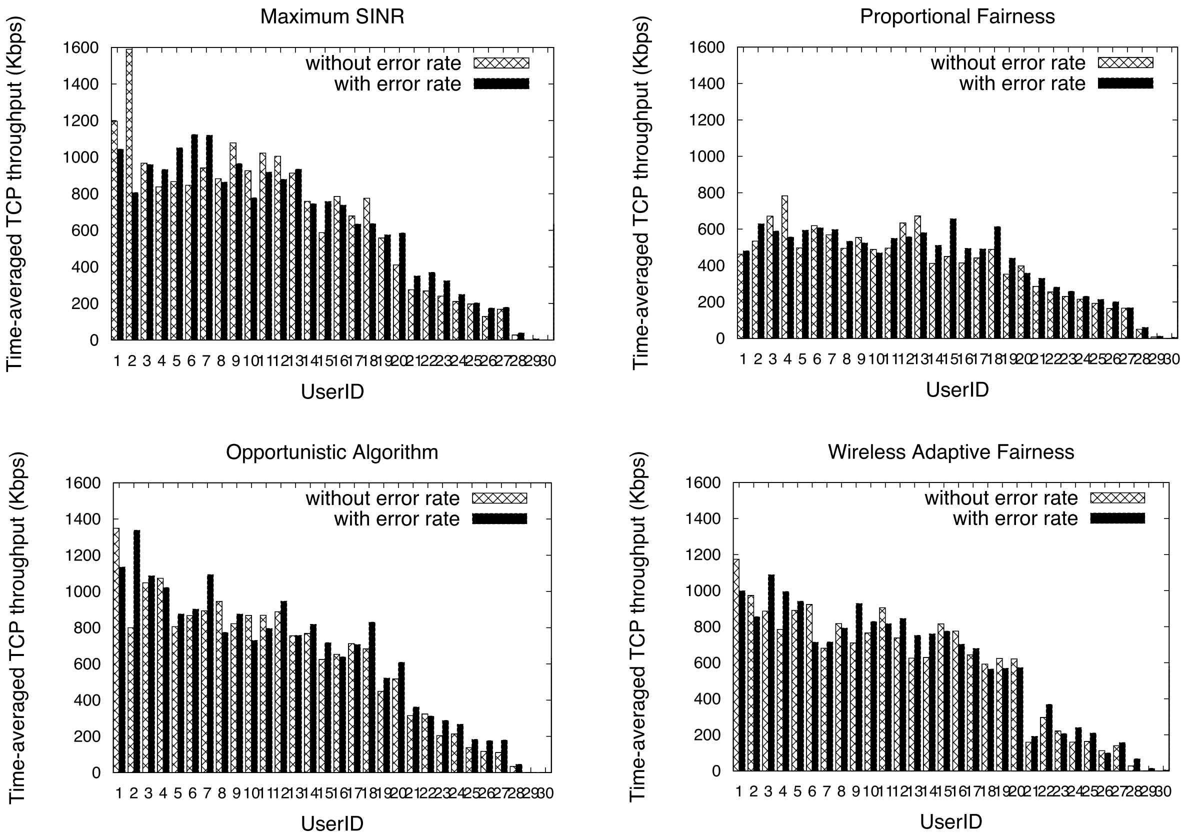

Fig. 7 TCP flow-level per-user throughput comparison (File size = 100 kB).

scheduling algorithms perform better than the original algorithms. These results illustrate that error rate awareness will effectively help the poor channel condition users to take the possible chances and improve throughput performance. This is because the poor channel condition users have more opportunities to experience the sensitive SINR ranges of the error rate awareness than good or medium channel condition users. Thus, it is easier to take advantage of error rate awareness in the competition.

7. Concluding Remarks

In the present study, it has been shown that the introduction of the error rate into scheduling metrics improves the TCP packet level and flow level performance of several well-known channel-aware scheduling algorithms in Evolved UTRA and UTRAN networks, especially for poor channel condition users. And the throughput improvement of cell edge users who always experience poor channel conditions is an important requirement for operators of Evolved UTRA and UTRAN. Reconstructing the new metric is also very simple, involving simple replacement of the instantaneous transmission rate with the expected transmission rate, which considers the error rate. Note that by training the existing wireless condition information, the error rates under different conditions can be readily obtained and recorded by a base station. In order to realize a realistic error rate, in practice, the error rate can be updated during a predefined period of time.

In addition, the fact that the application of error rate awareness on selective repeat ARQ can provide TCP performance comparable to that of HARQ may suggest an alternative method of satisfying services for poor channel condition users without changing the existing system and UEs.

In the future, we will include more wireless condition models and traffic models in the simulation. In addition, the frequency variation characteristics of the OFDM will be taken into account in the scheduler design, and more performance metrics will be considered, including delay, maximum queue length in base station, and other QoS constraints in real-time voice and video traffic.

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