Cross-Layer Design in HSDPA System to Reduce the TCP Effect

Mohamad Assaad and Djamal Zeghlache, *Member, IEEE*

***Abstract*—This paper focuses on the interaction between the transport control protocol (TCP) layer and the radio interface in the high-speed downlink packet access (HSDPA) wireless system. In the literature, studies of the interaction between TCP and wireless networks are focused on the evaluation of user bit rate in the case of dedicated channels. In this paper, the interaction between TCP, hybrid automatic repeat request (HARQ), and scheduling techniques (especially, proportional fair scheduling) is conducted. Analytical models to evaluate HSDPA cell capacity, user bit rate, and interaction with TCP layer are presented. Even if as expected the bit rate per flow decreases strongly with the congestion frequency in the wired network, it is shown that the overall capacity achieved by HSDPA is not as affected by the TCP layer. Using this result, a method to reduce the effect of TCP on wireless network without losing much cell capacity is proposed. This method has the advantage of modifying the scheduling algorithm only and of not requiring any change to the TCP protocol.**

***Index Terms*—Adaptive modulation and coding (AMC), cross layer, dense multipath channel (WSSUS), high-speed downlink packet access (HSDPA), hybrid-ARQ, scheduling, selective Rake receiver, transport control protocol (TCP).**

I. Introduction

THE EVOLUTION of the mobile communication market is expected to bring a major increase in data traffic demands combined with high bit rate services. Recent third-generation (3G) standardization (3GPP) and related technologies development reflect the need for high-speed packet data wireless system. In this context, The 3GPP introduces a "beyond 3G" system denominated high-speed downlink packet access (HSDPA). The HSDPA concept appears as an umbrella of some new technologies introduced to increase the user peak data rates up to 10 Mb/s and to improve the spectral efficiency for downlink packet data services. HSDPA consists of a new downlink time shared channel high-speed downlink shared channel (HS-DSCH) that supports a 2 ms transmission time interval (TTI), adaptive modulation and coding (AMC), multicode transmission, fast physical-layer hybrid automatic repeat request (HARQ), multiuser detector techniques to reduce intracell interference, and multiple-input–multiple-output (MIMO) (for details see [1]–[11]).

Since services built over a transport control protocol/Internet protocol (TCP/IP) are expected to make a large share of the overall traffic volume, particular attention must be paid to the performance of TCP traffic over HSDPA. Over wireless networks, where losses are mainly caused by mobility and intermittent degraded channel conditions, poor TCP performance is experienced. TCP misinterprets delays as congestion indications. Useless retransmissions are experienced and much time is wasted during the slow-start and the congestion avoidance phases.

TCP over wireless networks (especially, TCP/ARQ) has been studied widely in the literature when a dedicated channel is allocated to each user. When a shared channel is used on the radio interface, particular attention should be paid to the interactions between TCP, ARQ, and the scheduler and not only between TCP and ARQ. The performance of TCP in this case changes, and we show in this paper that the effect of TCP on wireless networks can be reduced drastically.

This contribution provides a comprehensive study of the effect of TCP on both achieved application bit rate performance and system capacity. In the literature, the studies are focused on the effect of TCP on application bit rate. The study conducted in this paper on the system capacity shows an interesting result and leads us to propose a method to maintain a given application flow rate and to reduce the effect of TCP with minimum possible cost. This solution has the advantage of not requiring any modification to the TCP implementation.

This paper is divided into three parts describing: HSDPA study, TCP-HSDPA interaction modeling, and results. In Section III, an analytical model of HSDPA is presented. Derivation of the analytical expressions requires the description of the channel model, the receiver type, an approximate expression of signal-to-interference ratio (SIR), the scheduling techniques, and the cell capacity models. In Sections IV, an analytical model for TCP is proposed. The effect of TCP on application bit rate and cell capacity is presented. Results and a proposed solution to reduce the TCP effect are presented in Section V.

II. Related Work

TCP is the most widely used transport protocol for packet data services in wireless networks. TCP was initially designed for wired networks where packet losses and delays are mainly caused by congestion [12].

Over wireless networks, TCP misinterprets delays, caused by ARQ, as congestion indications. Useless retransmissions are experienced and much time is wasted during the slow-start and the congestion avoidance phases. To alleviate these problems over wireless links, several approaches have been proposed for TCP enhancement [12]–[18]. Some solutions introduce changes in the TCP paradigm, while others deal with popular TCP versions (Reno and its variants). Eifel and Westwood TCP try to improve the classic TCP behavior to keep it applicable over both

Manuscript received January 16, 2005; revised June 17, 2005.

The authors are with the Institut National des Telecommunications, Ivry, Cedex 91011, France (e-mail: Mohamad.Assaad@int-evry.fr; Djamal.Zeghlache@int-evry.fr).

Digital Object Identifier 10.1109/JSAC.2005.862414

wireless and wired networks [19]. TCP selective acknowledgments (SACK) [20] is proposed to alleviate TCP's inefficiency in handling multiple drops in a single data window. However, TCP SACK does not improve the performance when the sender window size is not sufficiently large [20]–[22]. TCP forward acknowledgment (FACK) makes more intelligent decisions about the data that should be retransmitted. However, it is more or less targeted toward improving TCP's performance when losses are due to congestion rather than random losses [23]. In Split TCP [24], the end-to-end path is divided into two segments (typically, one wireless segment and one wired segment) on which different connections are established and locally optimized. Split TCP violates TCP semantics [23] and is incompatible with security requirements. Moreover, with Split TCP, handoffs may take several hundreds of milliseconds to be completed, thus leading to degraded TCP performance [24], [25]. In [26] and [27], a snoop agent is introduced at the link layer. The agent monitors the TCP connection, suppresses the duplicate acknowledgments, and retransmits the lost segments. The main advantage is the suppression of the duplicate acknowledgments for lost TCP segments that are locally retransmitted. However, the snoop agent must be located right before the TCP receiver. Thus, when a mobile node has to transmit data to a remote receiver, TCP acknowledgments are returned too late for an efficient recovery of the lost segments.

In [28]–[37], automatic repeat request (ARQ) and forward error correction (FEC) schemes are used at the link layer and TCP Reno (and its variants) are used at the transport layer. These papers have studied the TCP-ARQ interactions when a dedicated channel is allocated to each user. When a shared channel (HSDPA) is used on the radio interface, the interactions are not the same and new models are needed. The effect of TCP on HSDPA has been studied by [38] and [39] through simulation only. This motivates the modeling and the study in this paper.

III. Part I: HSDPA Model

A. Channel Model

The channel used is a composite dense multipath/shadowing channel with wide-sense stationary uncorrelated scattering (WSSUS), frequency-selective fading, and constant power dispersion profile (PDP). In such an environment, the envelope on each channel path can be considered as a Rayleigh variable [4], [5]. The link quality is also affected by slow variation of the mean signal level due to the shadowing from terrain, building, and trees. This shadowing can be modeled by a log-normal variable [8].

B. Rake Receiver

The detection in a multipath environment leads to a Rake receiver. This paper considers the selective Rake (SRake) receiver that selects the L best paths with the largest signal-to-noise ratio (SNR) (from N_r available diversity paths) and combines them using maximal ratio combining (MRC) [8].

If γ_i denotes the instantaneous SNR of the detector output samples (i.e., at the output of the matched filter and before the combiner), the instantaneous SNR of the SRake receiver [40]–[42] is

$$\gamma_{\mathrm{SRake}} = \sum_{i=1}^{L} \gamma(i) \tag{1}$$

where $1 \leq L \leq N_r$ and $\gamma(i)$ is the ordered γ_i, i.e., $\gamma(1) > \gamma(2) > \cdots > \gamma(N_r)$. Since the channel is a WSSUS channel with dense multipaths [N_r increases with the spreading bandwidth (BW)] and constant power dispersion profile (PDP), it was shown in [42], using the theory of "order statistics" [43] and the virtual branch technique (linear transform), that the joint probability distribution function $\mathrm{pdf}(\gamma(1), \gamma(2), \ldots, \gamma(N_r))$ is given by (2), shown at the bottom of the page, where $\overline{\gamma}$ is the mean SNR of the matched filter output. In [40], it was shown that $\overline{\gamma}$ is related to the mean SNR of the SRake output (combiner output) Γ_{SRake} by

$$\overline{\gamma} = \frac{\Gamma_{\mathrm{SRake}}}{L\left(1 + \sum_{n=L+1}^{N_r} \frac{1}{n}\right)}. \tag{3}$$

Due to the effect of shadowing (slow variation of the mean signal) and mobile position, Γ_{SRake} is not constant. In [44], we developed the expression of Γ_{SRake} as it is presented in (4) shown at the bottom of the page. P_j is the transmitted power of cell j, P_{sig} is the power of the associated control channel SCCH per one DSCH channel (calculated by the authors in [44] at 0.5% of the cell power), ξP_j is the power of the other control channels in the cell (CPICH, FACH, SCH amounting to 20% of P_j). SF is the spreading factor of the downlink channel (SF $=$ 16

$$\mathrm{pdf}(\gamma(1), \gamma(2), \ldots, \gamma(N_r)) = \begin{cases} N_r! \left(\frac{1}{\overline{\gamma}}\right) e^{-(1/\overline{\gamma})\Sigma_{i=1}^{N_r} \gamma(i)}, & \gamma(1) > \gamma(2) > \cdots > \gamma(N_r) > 0 \\ 0, & \text{otherwise} \end{cases} \tag{2}$$

$$\Gamma_{\mathrm{SRake}} = \frac{\mathrm{SF}}{\log 2(M)\tau} \frac{\frac{P_j - NP_{sig} - \xi P_j}{N} d_j^{-\mu} 10^{s_j/10}}{\underbrace{\alpha\left(P_j - \frac{P_j - NP_{sig} - \xi P_j}{N}\right) d_j^{-\mu} 10^{s_j/10}}_{I_{\mathrm{intra}}} + \underbrace{\sum_{l \neq j}\left(P_l d_l^{-\mu} 10^{s_l/10}\right)}_{I_{\mathrm{inter}}} + \eta_0} \tag{4}$$

in HSDPA), M is the modulation order, τ is the coding rate, η_0 is the receiver thermal noise power, and d_j is the distance between mobile and node B of cell j. Parameter μ is the path loss slope, s_j corresponds to log-normal shadowing with zero mean and standard deviation σ ($\sigma = 10$ dB) and α is the orthogonality loss factor ($\alpha = 0.4$ in macrocell and 0.06 in microcell environment [4], [5]).

C. Proposition 1: Distributions of SNR

Using the distribution function $\mathrm{pdf}(\gamma(1), \gamma(2), \ldots, \gamma(N_r))$, the distribution function (pdf) and the cumulative distribution function (cdf) of γ_{SRake} can be deduced using the virtual branch technique. Each $\gamma(i)$ is transformed into a set of virtual path SNR, called V_n, as follows:

$$\gamma(i) = \sum_{n=i}^{N_r} \frac{\overline{\gamma}}{n} V_n. \tag{5}$$

The SNR, at the output of the receiver, can then be written using the following equation:

$$\begin{aligned}\gamma_{\mathrm{SRake}} &= \sum_{i=1}^{L} \sum_{n=i}^{N_r} \frac{\overline{\gamma}}{n} V_n \\ &= \sum_{n=L+1}^{N_r} L \frac{\overline{\gamma}}{n} V_n + \overline{\gamma} \sum_{n=1}^{L} V_n. \end{aligned} \tag{6}$$

In [42], it is shown that $V_n s$ are independent and identically distributed (i.i.d.) normalized exponential random variables. Hence, the pdf and the cdf of Γ_{SRake} (for a given $\overline{\gamma}$) are given by (7) and (8) shown at the bottom of the page, where $\Gamma(\cdot)$ is the Gamma function.

D. Proposition 2: Distribution of $\overline{\gamma}$

As we said before, the mean signal level is affected by slow variation due to the effect of the shadowing. The SNR at the output of the Rake changes instantaneously according to the multipath fast fading and its mean value changes slowly according to the shadowing. In this section, the distribution function of the mean SNR $\overline{\gamma}$ is deduced.

Using (3) and (4), $\overline{\gamma}$ for a given mobile position in the cell can be written as follows:

$$\overline{\gamma} = \frac{A}{B + X} \tag{9}$$

where

$$\begin{aligned} A &= \frac{\mathrm{SF}}{\log 2(M)\tau} \frac{P_j - NP_{sig} - \xi P_j}{N} \frac{1}{L\left(1 + \sum\limits_{n=L+1}^{N_r} \frac{1}{n}\right)} \\ B &= \alpha \left(P_j - \frac{P_j - NP_{sig} - \xi P_j}{N}\right) \\ X &= \frac{1}{10^{bs_{sj}/10}} \sum_{l \neq j} \left(P_l \left(\frac{d_l}{d_j}\right)^{-\mu} 10^{bs_{sl}/10}\right). \end{aligned}$$

Note that the shadowing loss s_j is correlated between the base stations (BSs) [45]. It is usually modeled as a sum (in decibels) of a component common to all BSs, s_c, and a component s_{sj} specific to BS_j ($s_j = as_c + bs_{sj}$, where $a = b = 1/\sqrt{2}$). By simplifying as_c, we can get the expression of X in (9).

1) Distribution Function: The expression $\sum_{l \neq j} \left(P_l \left(d_l/d_j\right)^{-\mu} 10^{bs_{sl}/10}\right)$ is the sum of independent log-normal variables. According to [46], using the approximation of Wilkinson [43], the sum of a finite number of log-normal variables can be approximated to a log-normal variable. Hence, X can be considered as the division of two correlated log-normal variables which are equivalent to a log-normal variable [the correlation is interpreted by the relation between s_{sj} and s_{sl} due to the effect of handover as we show in the condition of (11)]. Consequently, $A/(B + X)$ is a log-normal variable.

To evaluate the parameters of the equivalent log-normal variable, i.e., the mean value μ_f and the variance σ_f^2, we need to consider the effect of fast cell selection (hard handover) which limits the interval of variation of parameter s_{sj}.

2) Fast Cell Selection: The HS-DSCH channel does not support soft handover. However, a fast cell selection procedure is applied in each TTI which could be seen as a substitute

$$\begin{aligned} \mathrm{pdf}\left(\frac{\gamma_{\mathrm{SRake}}}{\overline{\gamma}}\right) &= \frac{N_r!}{L!(L)^{N_r - L}} e^{-\gamma/\overline{\gamma}} \\ &\times \left[\sum_{i=1}^{L} \frac{\gamma^{i-1}}{\overline{\gamma}^i \Gamma(i)} \left(\sum_{j=1}^{N_r - L} \frac{(-1)^{(L-i+j-1)} L^{(L-i+1)}}{(j-1)!(N_r - L - j)!(j)^{(L-i+1)}}\right)\right. \\ &\qquad \left. \times \frac{N_r!}{L!(L)^{N_r - L}} e^{-\gamma/\overline{\gamma}} \left[+ \sum_{i=1}^{N_r - L} \frac{(-1)^{(L+i-1)} L^L}{(N_r - L - i)! i! i^{(L-1)} (N_r - L)} e^{-(i\gamma/L\overline{\gamma})}\right]\right. \end{aligned} \tag{7}$$

$$\begin{aligned} \mathrm{cdf}\left(\frac{\gamma_{\mathrm{SRake}}}{\overline{\gamma}}\right) &= \frac{N_r!}{L!(L)^{N_r - L}} \\ &\times \left[\sum_{i=1}^{L} \left(1 - e^{-\gamma/\overline{\gamma}} \left(\sum_{k=0}^{i-1} \frac{\gamma^k}{\overline{\gamma}^k k!}\right)\right) \left(\sum_{j=1}^{N_r - L} \frac{(-1)^{(L-i+j-1)} L^{(L-i+1)}}{(j-1)!(N_r - L - j)!(j)^{(L-i+1)}}\right)\right. \\ &\qquad \left. \times \frac{N_r!}{L!(L)^{N_r - L}} \left[+ \sum_{i=1}^{N_r - L} \frac{(-1)^{(L+i-1)} L^{L+1}}{(L+i)(N_r - L - i)! i! i^{(L-1)} (N_r - L)} \left(1 - e^{-(((i+L)\gamma)/L\overline{\gamma})}\right)\right]\right. \end{aligned} \tag{8}$$

for handover. The mobile user measures the "mean" received power of the CPICH channel of the various cells and chooses the cell to which it would be connected. The choice is governed by the following equation:

$$P_{\mathrm{CPICH},j} d_j^{-\mu} 10^{s_j/10} \geq P_{\mathrm{CPICH},l} d_l^{-\mu} 10^{s_l/10} \text{ (for } l \neq j). \tag{10}$$

$P_{\mathrm{CPICH},j}$ is the transmitted power of the CPICH channel in cell j. The cell selection depends on the mean received power and not on the instantaneous power which explains the fact that the fast fading is not included in the analysis of cell selection. Note that the HS-DSCH in HSDPA does not support fast power control. No fast power control is used either on the control channel. Hence, the transmitted power of the node B is constant and all the available power at the node B is assumed as used (43 dBm). Consequently, the CPICH transmitted power is the same in all the cells. By replacing s_j by its value $as_c + bs_{sj}$ and using the condition in (10), the following inequality is obtained after several manipulations:

$$s_{sl} < s_{sj} - \frac{10\mu \log 10\left(\frac{d_j}{d_l}\right)}{b} \text{ (for } l \neq j). \tag{11}$$

3) Mean Value and Variance of $\overline{\gamma}$: To evaluate the mean value and the variance of $\overline{\gamma}$, we proceed as follows.

We evaluate the mean value and the variance of the expression $\sum_{l \neq j} P_l (d_l/d_j)^{-\mu}\left[10^{b(s_{sl}-s_{sj})/10}; s_{sl} < s_{sj} - ((10\mu \log 10(d_j/d_l))/b)\right]$, $E(X)$ and $V(X)$ given by (12) and (13), shown at the bottom of the page, where σ^2, the variance of s_j, equals 8–10 dB in general [4], [8] and $E(d_j, d_l, d_{l'})$ is given by (14) shown at the bottom of the page.

Let μ_1 and σ_1 be the parameters of the expression $Y = (B + X)/A$. Since Y is a log-normal variable, μ_1 and σ_1 can be evaluated by the following equations:

$$\mu_1 = \ln\left(\frac{E^2(Y)}{\sqrt{V(Y) + E^2(Y)}}\right) \tag{15}$$

$$\sigma_1^2 = \ln\left(\frac{V(Y) + E^2(Y)}{E^2(Y)}\right). \tag{16}$$

Note that $E(Y) = (B + E(X))/A$ and $V(Y) = V(X)/A^2$. Since $\overline{\gamma} = 1/Y$ and Y is log-normal, the parameters μ_f and σ_f^2 are given by

$$\mu_f = -\mu_1 = \ln\left(\frac{\sqrt{A^2 V(X) + A^2(E(X) + B)^2}}{(E(X) + B)^2}\right) \tag{17}$$

$$\sigma_f^2 = \sigma_1^2 = \ln\left(\frac{V(X) + (E(X) + B)^2}{(E(X) + B)^2}\right). \tag{18}$$

where A, B, $E(X)$, and $V(X)$ are given, respectively, by (9), (12), and (13).

Consequently, $\overline{\gamma}$ is a log-normal variable with mean value and variance μ_f and σ_f^2 given by (17) and (18). Note that $\overline{\gamma}$ varies in the interval $[A/(B + \sum_{l \neq j} P_l); A/B]$ (using the condition in (11), we can verify that $0 \leq X \leq \sum_{l \neq j} P_l$).

Finally, the distributions pdf and cdf of γ_{SRake} are obtained by (19) and (20) shown at the bottom of the next page.

E. Adaptive Modulation and Coding (AMC)

To track the variation of the channel conditions, AMC is used in HSDPA where a transmission scheme (a modulation order M, a coding rate τ, and a number of HS-DSCH codes N) is selected on a dynamic basis according to the value of $\mathrm{SIR} = \gamma_{\mathrm{Rake}}$. Note that the maximum value of N is 15 codes. Let k_m be the probability of selection of transmission scheme m. $k_m = \mathrm{Prob}(\mathrm{SIR} \geq \gamma_m)$ for the highest order transmission scheme (16 QAM, 3/4, $N = 15$) and $k_m = \mathrm{Prob}(\gamma_m \leq \mathrm{SIR} < \gamma_{m+1})$ for

$$E(X) = e^{(b\beta\sigma)^2/2} \sum_{l \neq j} \left(\frac{d_j}{d_l}\right)^{\mu} P_l Q\left(b\beta\sigma - \frac{10\mu \log 10\left(\frac{d_l}{d_j}\right)}{b\sigma}\right) \tag{12}$$

$$\begin{aligned} V(X) = e^{(b\beta\sigma)^2}\Bigg[& e^{3(b\beta\sigma)^2} \sum_{l \neq j} P_l^2 \left(\frac{d_j}{d_l}\right)^{2\mu} Q\left(2\sqrt{2}b\beta\sigma + \frac{10\mu \log 10\left(\frac{d_j}{d_l}\right)}{\sqrt{2}b\sigma}\right) \\ & + \sum_{l \neq j} \sum_{l' \neq l,j} P_l P_{l'} \left(\frac{d_j}{d_l}\right)^{\mu} \left(\frac{d_j}{d_{l'}}\right)^{\mu} E(d_j, d_l, d_{l'}) - e^{(b\beta\sigma)^2} \\ & \times \left(\sum_{l \neq j} P_l \left(\frac{d_j}{d_l}\right)^{\mu} Q\left(\sqrt{2}b\beta\sigma + \frac{10\mu \log 10\left(\frac{d_j}{d_l}\right)}{\sqrt{2}b\sigma}\right)\right)^2 \Bigg] \end{aligned} \tag{13}$$

$$\begin{aligned} E(d_j, d_l, d_{l'}) = \frac{1}{\sqrt{2\pi}\sigma} \int_{-\infty}^{+\infty} \Bigg[& e^{-2\beta b s_{sj}} e^{-(s_{sj}^2/2\sigma^2)} Q\left(b\beta\sigma + \frac{10\mu \log 10\left(\frac{d_j}{d_l}\right) - b s_{sj}}{b\sigma}\right) \\ & \times Q\left(b\beta\sigma + \frac{10\mu \log 10\left(\frac{d_j}{d_{l'}}\right) - b s_{sj}}{b\sigma}\right)\Bigg] ds_{sj}. \end{aligned} \tag{14}$$

the other transmission schemes. Parameter γ_m is the target SIR of the transmission scheme m. The probability $P(\gamma_m \leq \mathrm{SIR} < \gamma_{m+1})$ can be evaluated through $P(\mathrm{SIR} \leq \gamma_{m+1}) - P(\mathrm{SIR} \leq \gamma_m)$. Hence, k_m can be expressed as

$$k_m = \begin{cases} 1 - F(\gamma_m), & \text{highest order } (M,\tau,N)_m \\ F(\gamma_{m+1}) - F(\gamma_m), & \text{other } (M,\tau,N)_m \end{cases}. \tag{21}$$

Note that A and B in (9), (19), and (20) are functions of transmission scheme m and change from a transmission scheme to another in the evaluation of k_m. A and B are denoted A_m and B_m in the rest of this paper.

F. HARQ

HARQ has been studied by the authors in [47]. It was shown that the mean number of transmissions N_s due to HARQ can be given by

$$N_s = \frac{1 + P_e - P_e P_s}{1 - P_e P_s} \tag{22}$$

where P_e is the block error rate (BLER) and P_s is the probability of errors after soft combining two successive erroneous transmissions using the chase combining algorithm [9], [10].

G. Scheduling

Fast scheduling is one of the key techniques used in HSDPA. The user bit rate and the cell capacity depends on the scheduler used. In this paper, four scheduling techniques are used: fair throughput (FT), round-robin (RR), Max C/I, and proportional fair (PF). The cell capacity and the user bit rate are evaluated in each case.

1) Round Robin (RR): In the RR scheduler [7], [8], the channel is shared equally between users, i.e., the same number of TTIs is allocated to each user. If N_u is the number of users in the cell, then the probability that a TTI is allocate to a given user is $1/N_u$. Hence, the mean bit rate of user i is given by

$$\begin{aligned} R_i &= \frac{1}{N_u} \sum_m \frac{R_m k_{m,i}}{N_{s,i}} \\ &= \frac{1}{N_u} \sum_m k_m \frac{W}{\mathrm{SF}} \frac{(N \log 2(M)\tau)_{m,i}}{N_{s,i}} \end{aligned} \tag{23}$$

where R_m is the bit rate of transmission scheme m during a TTI, $N_{s,i}$ is the number of transmissions of user i due to HARQ, W is the chip rate, and SF is the spreading factor. Note that $k_{m,i}$ varies with the mobile position (μ_f and σ_f are function of the mobile position). The cell throughput in this case is given by

$$th = E\left(\sum_{i=1}^{N_u} R_i\right). \tag{24}$$

2) Fair Throughput (FT): The FT scheduler [8] allocates a fixed bit rate to users independently of channel condition and mobile position. In this case, the cell capacity can be estimated as follows.

The DSCH channel flow in symbols/s is given by

$$R_s = \frac{W}{\mathrm{SF}}. \tag{25}$$

Since the DSCH channel is shared by several users, in a given time T (channel observation time), we have

$$\frac{\sum_{i=1}^{N_u} Rs_i T_i}{T} = \frac{W}{\mathrm{SF}} \tag{26}$$

where Rs_i is the throughput of each user in symbols/s and T_i is the connection duration. The modulation and coding scheme change during the transfer of the packet calls. Hence, (26) is written as

$$\frac{\sum_{i=1}^{N_u} R_i T_i \sum_m \frac{k_m}{(N \log 2(M)\tau)_{i,m}}}{T} = \frac{W}{\mathrm{SF}} \tag{27}$$

$$\begin{aligned} \mathrm{pdf}(\gamma) &= \int_{\overline{\gamma}} \mathrm{pdf}\left(\frac{\gamma_{\mathrm{SRake}}}{\overline{\gamma}}\right) \mathrm{pdf}(\overline{\gamma}) \\ &= \int_{A/\left(B+\sum_{l\neq j} P_l\right)}^{A/B} \left[\sum_{i=1}^{L} \frac{\gamma^{i-1}}{\overline{\gamma}^i \Gamma(i)} \left(\sum_{j=1}^{N_r-L} \frac{(-1)^{(L-i+j-1)} L^{(L-i+1)}}{(j-1)!(N_r-L-j)!(j)^{(L-i+1)}}\right)\right. \\ &\qquad \left. + \sum_{i=1}^{N_r-L} \frac{(-1)^{(L+i-1)} L^L}{(N_r-L-i)!i!i^{(L-1)}(N_r-L)} e^{-(i\gamma/L\overline{\gamma})}\right] \frac{N_r!}{L!(L)^{N_r-L}} e^{-\gamma/\overline{\gamma}} \frac{1}{\sqrt{2\pi}\sigma_f} \\ &\quad \times e^{-\left((\ln\overline{\gamma}-\mu_f)^2/2\sigma_f^2\right)} d(\overline{\gamma}) \end{aligned} \tag{19}$$

$$\begin{aligned} \mathrm{cdf}(\gamma_{\mathrm{SRake}}) &= F(\gamma) \\ &= \int_{A/(B+\sum_{l\neq j} P_l)}^{A/B} \left[\sum_{i=1}^{L} \left(1 - e^{-\gamma/\overline{\gamma}} \left(\sum_{k=0}^{i-1} \frac{\gamma^k}{\overline{\gamma}^k k!}\right)\right) \left(\sum_{j=1}^{N_r-L} \frac{(-1)^{(L-i+j-1)} L^{(L-i+1)}}{(j-1)!(N_r-L-j)!(j)^{(L-i+1)}}\right)\right. \\ &\qquad \left. + \sum_{i=1}^{N_r-L} \frac{(-1)^{(L+i-1)} L^{L+1}}{(L+i)(N_r-L-i)!i!i^{(L-1)}(N_r-L)} \left(1 - e^{-((i+L)\gamma/L\overline{\gamma})}\right)\right] \frac{N_r!}{L!(L)^{N_r-L}} \\ &\quad \times \frac{1}{\sqrt{2\pi}\sigma_f} e^{-\left((\ln\overline{\gamma}-\mu_f)^2/2\sigma_f^2\right)} d(\overline{\gamma}). \end{aligned} \tag{20}$$

where R_i is the service bit rate. Due to the effect of HARQ, $N_{s,i}$ packets are transmitted instead of one packet, having all the same modulation and coding scheme. Hence, the number of users N_u in the cell can be evaluated using the following equation:

$$\frac{\sum_{i=1}^{N_u} R_i N_{s,i} T_i \sum_m \frac{k_m}{(N \log 2(M)\tau)_{i,m}}}{T} = \frac{W}{\mathrm{SF}}. \tag{28}$$

The cell throughput can be then estimated using

$$th = E\left(\frac{\sum_{i=1}^{N_u} R_i T_i}{T}\right). \tag{29}$$

3) Max C/I: In Max C/I scheduling [4], [7], [8], the channel is allocated in each TTI to the user having the best SIR. This scheduler maximizes the cell capacity but does not guarantee any QoS to the user. Users at the border of the cell have always poor channel conditions (due to attenuation, interference, and absence of fast power control) and experience low bit rate. If N_u is the number of users in the cell, the probability that a TTI is allocated to user i can be written as in (30)

$$\begin{aligned} pr(i) &= \mathrm{Prob}\Big(\mathrm{SIR}_i > \mathrm{SIR}_j \text{ for } j = 1 \ldots N_u \text{ and } j \neq i\Big) \\ &= \prod_{j \neq i}^{N_u} \mathrm{Prob}(\mathrm{SIR}_i > \mathrm{SIR}_j) \\ &= \prod_{j \neq i}^{N_u} \left(1 - \int_0^{+\infty} F_i(\mathrm{SIR}_j)\mathrm{pdf}(\mathrm{SIR}_j) d(\mathrm{SIR}_j)\right) \end{aligned} \tag{30}$$

where F_i is the cdf of SIR_i, given by (20), evaluated at SIR_j (note that in (20) the evaluation of F depends on the mobile position) and $\mathrm{pdf}(\mathrm{SIR}_j)$ is the pdf of SIR_j given by (19). Hence, the bit rate of user i is given by

$$\begin{aligned} R_i &= pr(i) \sum_m \frac{R_m k_{m,i}}{N_{s,i}} \\ &= pr(i) \sum_m k_m \frac{W}{SF} \frac{(N \log 2(M)\tau)_{m,i}}{N_{s,i}}. \end{aligned} \tag{31}$$

The cell throughput in this case is

$$th = E\left(\sum_{i=1}^{N_u} R_i\right). \tag{32}$$

4) Proportional Fair (PF): The PF scheduler is a compromise between Max C/I and FT schedulers [48]. In each TTI, the channel is allocated to the user having $\max(r/S)$, where r is the transmission rate in this TTI (according to the transmission scheme selected) and S is the mean bit rate transmitted in previous TTIs.

In the literature, there are several versions of PF [8], [48], [49]. Some versions propose to evaluate the mean bit rate R through an exponentially smoothed average when others increase the influence of the instantaneously transmission rate by using the user selection condition $\max(r^c/S)$, where c is a parameter depending on channel conditions.

This paper provides an analytical study of the basic PF scheduler. It was shown in [49] that the performance in this case can be considered as the asymptotic performance of the other PF schedulers. The mean bit rate S in the basic PF is a linear function of the mean bit rate when S is evaluated through an exponentially smoothed average (for details see [49]).

The mean bit rate S_i achieved by user i, when the TTIs are allocated to this user is

$$S_i = \sum_m k_m \frac{W}{\mathrm{SF}} \frac{(N \log 2(M)\tau)_{m,i}}{N_{s,i}}. \tag{33}$$

Note that the condition $\max(r/S)$ is equivalent to $\max(\mathrm{SIR}/S)$ due to the fact that according to each SIR value, there is a given possible transmission rate r in a given TTI (in fact, r values correspond to a range of SIRs. Since the number of transmission schemes is high, approximately 30 [7], this hypothesis is still a good approximation and it can be seen as an asymptotic study of the PF scheduler). Hence, if N_u is the number of users in the cell, the probability that a TTI is allocated to a given user i can be evaluated using (34)

$$\begin{aligned} pr(i) &= \mathrm{Prob}\left(\frac{\mathrm{SIR}_i}{S_i} > \frac{\mathrm{SIR}_j}{S_j} \text{ for } j = 1 \ldots N_u \text{ and } j \neq i\right) \\ &= \prod_{j \neq i}^{N_u} \mathrm{Prob}\left(\mathrm{SIR}_i > \frac{S_i}{S_j}\mathrm{SIR}_j\right) \\ &= \prod_{j \neq i}^{N_u} \left(1 - \int_0^{+\infty} F_i\left(\frac{S_i}{S_j}\mathrm{SIR}_j\right) \right. \\ &\qquad \left. \times \, \mathrm{pdf}(\mathrm{SIR}_j) d(\mathrm{SIR}_j)\right). \end{aligned} \tag{34}$$

Consequently, the bit rate achieved by user i is

$$R_i = pr(i) \sum_m k_m \frac{W}{\mathrm{SF}} \frac{(N \log 2(M)\tau)_{m,i}}{N_{s,i}}. \tag{35}$$

The cell throughput in this case is given by

$$th = E\left(\sum_{i=1}^{N_u} R_i\right). \tag{36}$$

IV. Part II: TCP Model

The steady-state performance of TCP has been studied widely in the literature ([14]–[17], [50]). This paper extends the TCP model of [50] and [51] by introducing the effect of the radio interface and the radio access system (limited bit rates from 32 to 128 kb/s, scheduling effect, and HARQ).

The data rate at the TCP layer is computed by dividing the data size by the mean value of latency time $E(T)$ (a Markov process is assumed). Let T_{ss} be the latency time of the slow-start phase, T_{loss} be the recovery time and RTO cost and T_{ca} represent the latency time of the steady-state phase. Hence, the data rate is given by

$$R = \frac{\text{data}}{E(T_{ss}) + E(T_{\text{loss}}) + E(T_{ca})}. \tag{37}$$

Consequently, to model the effect of TCP on HSDPA, we need to estimate the latency time of the slow-start phase, the loss recovery, and the steady-state phase (this is conducted, respectively, in Sections IV-B–D). The analysis of TCP timeouts needed in the latency times are presented in Section IV-A.

A. Timeout

There are two ways that TCP detects losses: retransmission timeouts (RTOs) and triple duplicate ACKs. The RTOs of TCP can be caused by a congestion in the Internet Network or by a delay due to limited bit rate or to multiple retransmissions on the radio interface generated by the ARQ technique which increase round-trip time (RTT) and RTOs of TCP. In this section, the probability of RTOs due to the effect of the radio interface is derived.

1) Proposition: The probability of RTO due to the radio interface is given by the following equation:

$$q = Q\left(\frac{To - \text{RTT}_{\text{wired}} - \frac{1+P_e-P_eP_s}{1-P_eP_s}T_j}{\sqrt{\sum_m k_m \frac{W}{\text{SF}} \frac{(N\log 2(M)\tau)_{m,i}}{12\,000} \text{TTI} \frac{\sqrt{P_e(1-P_e+P_eP_s)T_j}}{1-P_eP_s}}}\right) \tag{38}$$

where To is the average duration of the first timeout in a section of one or more successive timeouts, $\text{RTT}_{\text{wired}}$ is the average RTT of the wired part of the network, T_j is the transmission time of a segment on the radio interface with a given bit rate (the bit rate depends on the used scheduler), P_e is the probability of errors after decoding the information block via FEC, and P_s is the probability of errors after soft combining two successive transmissions of the same information block.

a) Proof: In HSDPA, each TCP segment is transmitted using several predefined TTIs, each lasting 2 ms. The size of a TCP segment is 1500 octets. Transmitting a TCP segment requires between 12 and 60 TTIs (depending upon the modulation and coding schemes used on the radio interface). Let S_i be the data size transmitted over each TTI. The number of retransmission required to deliver the data of size S_i is a random variable due to varying radio channel conditions. The time needed to transmit an error free TCP segment is

$$\text{RTT} = \frac{\sum_{i=1}^{n_s} N_{\text{TTI}}(i)}{n_s} T_j + \text{RTT}_{\text{wired}}. \tag{39}$$

Variable n_s is the number of TTIs needed to transmit a TCP segment when no errors occur on the radio interface and $N_{\text{TTI}}(i)$ is the number of transmissions of TTI i due to HARQ. The use of scheduling on a shared channel makes the errors on each TTI independent (the successive TTIs are allocated to various users), then the number of retransmission of each TTI data is independent from the other TTIs. Using the central limit theorem, the sum of a large number of i.i.d. symmetric random variables can be considered as a Gaussian variable. Hence, the number of transmissions of a TCP segment $N_i = (\sum_{i=1}^{n_s} N_{\text{TTI}}(i)/n_s)$ can be modeled by a Gaussian variable. Consequently, the time needed to transmit a TCP segment (RTT) is a Gaussian variable. The probability of timeout RTO expressed as $\text{prob}(\text{RTT} = \text{Gaussian} > \text{To})$ with the Gaussian assumption leads to: $Q((To - E(\text{RTT})/\sigma(\text{RTT})))$. Note that $E(N_i)$ and $\sigma(N_i)$ can be evaluated in the same way as in Section III-F but with introducing the mean value of n_s in the final expression. By replacing $E(N_i)$ and $\sigma(N_i)$ by their values, E({\rm RTT}) and $\sigma(\text{RTT})$ are obtained and the probability of RTO has the form provided previously in (38).

B. Slow-Start

The TCP connection begins in slow-start mode where it quickly increases its congestion window, to achieve best effort service, until it detects a packet loss. In the slow-start phase, the window size congestion window (cwnd) is limited by a maximum value $W_{\max}$ imposed by the sender or receiver buffer limitations. To determine $E(T_{ss})$, the number of data segments $E(d_{ss})$, the sender is expected to send before losing a segment, is needed. From this number, one can deduce $E(W_{ss})$, the window we would expect TCP to achieve at the end of the slow-start, where there is no maximum window constraint. If $E(W_{ss}) \le W_{\max}$, then the window limitation has no effect, and $E(T_{ss})$ is simply the time for a sender to send $E(d_{ss})$ in the exponential growth mode of the slow-start. On the other hand, if $E(W_{ss}) > W_{\max}$, then $E(T_{ss})$ is the time for a sender to slow-start up to $\text{cwnd} = W_{\max}$, and then send the remaining data segments at a rate of $W_{\max}$ segments per round. Let e be the probability of retransmission $(\text{congestion} + \text{RTO})$. Probability e can be evaluated using the following equation:

$$e = p + q - pq. \tag{40}$$

The term $E(d_{ss})$ can be calculated by the expression below

$$\begin{aligned} E(d_{ss}) &= \left(\sum_{k=0}^{d-1}(1-e)^k e \cdot k\right) + (1-e)^d \cdot d \\ &= \frac{(1-(1-e)^d)(1-e)}{e} \end{aligned} \tag{41}$$

where d is the number of segments in the file. Using the same demonstration as in [51], the mean value of the latency time can be evaluated using (42) shown at the bottom of the page. γ is the

$$E(T_{ss}) = \begin{cases} \text{RTT}\left[\log_\gamma\left(\frac{W_{\max}}{W_1} + 1 + \frac{1}{W_{\max}}\left(E(d_{ss}) - \frac{\gamma W_{\max} - W_1}{\gamma - 1}\right)\right)\right], & \text{When } E(W_{ss}) > W_{\max} \\ \text{RTT} \cdot \log_\gamma\left(\frac{E(d_{ss})(\gamma-1)}{W_1} + 1\right), & \text{When } E(W_{ss}) \le W_{\max} \end{cases} \tag{42}$$

rate of exponential growth of the window size during slow-start. $E(W_{ss})$ is given by

$$E(W_{ss}) = \frac{E(d_{ss})(\gamma - 1)}{\gamma} + \frac{W_1}{\gamma}. \tag{43}$$

C. Recovery Time of the First Loss

The slow-start phase in TCP ends with the detection of a packet loss. The sender detects a loss in two ways: negative ACK (triple duplicate) or RTOs. The RTO could be caused by a congestion in the wired network or by the retransmissions on the radio interface. After an RTO, the window size decreases to 1, however, the loss detected by the triple duplicate ACKs decreases the window size to a half. In this section, we evaluate the recovery time of this first loss. The probability of loss in a file of d TCP segments is

$$\mathrm{loss} = 1 - (1 - e)^d. \tag{44}$$

The loss between segments could be considered independent [50]. Let $Q'(e, w)$ be the probability that if a loss occurs, it is a RTO. This probability can be evaluated as follows: Let cong and wirel be, respectively, the probabilities that there is a congestion loss in the transmission of the file and there is a RTO due to radio interface conditions

$$\mathrm{cong} = 1 - (1 - p)^d \tag{45}$$

$$\mathrm{Wirel} = 1 - (1 - q)^d \tag{46}$$

where q is evaluated in Section IV-A by (38). [50] derives the probability that a sender in congestion avoidance will detect a packet loss with an RTO, as a function of congestion rate p and window size w. This probability is denoted by $F(p, W)$

$$F(p, W) = \min\left(1, \frac{(1 + (1 - p)^3(1 - (1 - p)^{w-3}))}{\frac{(1-(1-p)^w)}{(1-(1-p)^3)}}\right). \tag{47}$$

The probability of RTO is simply obtained through:

$$\mathrm{RTO} = \mathrm{cong} \cdot F(p, W) + \mathrm{Wirel} - \mathrm{Wirel} \cdot \mathrm{cong} \cdot F(p, W). \tag{48}$$

Hence, the probability $Q'(e, w)$ is derived as

$$Q'(e, w) = \frac{\mathrm{RTO}}{\mathrm{loss}} = \frac{\mathrm{cong} \cdot F(p, W) + \mathrm{Wirel} - \mathrm{Wirel} \cdot \mathrm{cong} \cdot F(p, W)}{1 - (1 - e)^d}. \tag{49}$$

The probability of loss via triple duplicate is $\mathrm{loss}(1 - Q'(e, w))$. It is assumed that fast recovery for a triple duplicate takes one RTT [51]. However, it takes more time for an RTO. The RTO cost, derived in [50], does not take into account the radio interface effects. Using the mean expected cost of an RTO

$$E(z^{TO}) = \frac{1 + e + 2e^2 + 4e^3 + 8e^4 + 16e^5 + 32e^6}{1 - e} To \tag{50}$$

and combining these results, the mean recovery time at the end of the initial slow-start is obtained

$$E(T_{\mathrm{loss}}) = \mathrm{loss}\Big(Q'(e, w)E(z^{TO}) + (1 - Q'(e, w))\mathrm{RTT}\Big). \tag{51}$$

D. Steady-State Phase

The time needed to transfer the remaining data can be derived in the same way as in [51]. Indeed, the amount of data left after the slow-start and any following loss recovery is approximately

$$E(d_{ca}) = d - E(d_{ss}). \tag{52}$$

This amount of data is transferred with a throughput $R(e, \mathrm{RTT}, To, W_{\max})$. The latency time is then given by

$$E(T_{ca}) = \frac{E(d_{ca})}{R(e, \mathrm{RTT}, To, W_{\max})}. \tag{53}$$

In [50], the throughput $R(p, \mathrm{RTT}, To, W_{\max})$ is evaluated without the radio interface effects. By using the same demonstration as in [50] and by introducing e, RTT and $Q'(e, w)$ provided in this paper, the derivation of the throughput expression leads to (54) shown at the bottom of the page, where b is the number of TCP segments acknowledged by one ACK. $W(e)$ is given by

$$W(e) = \frac{2 + b}{3b} + \sqrt{\frac{8(1 - e)}{3be} + \left(\frac{2 + b}{3b}\right)^2}. \tag{55}$$

Finally, once the total latency time is calculated, the bit rate for each service, at TCP layer can be evaluated using (37).

E. Effect of TCP on Wireless Network

The decrease of TCP bit rate over the radio interface is due to two reasons: decrease of TCP window size and retransmissions of TCP segments. In the case of dedicated channels [28]–[37], it is interesting to evaluate the final TCP bit rate since the number of users is fixed. However, when several users share the same channel in time, the number of users can increase or decrease according to the bit rate given to each user. The evaluation of the mean number of TCP segments retransmissions N_{TCP} becomes important. When N_{TCP} has a low value compared with the decrease of TCP window size (we will show in Section IV-EI that this is the case in HSDPA), the decrease of TCP bit rate is due essentially to the decrease of window size. In this case, the

$$R(e, \mathrm{RTT}, To, W_{\max}) = \begin{cases} \dfrac{\frac{1-e}{e} + \frac{W(e)}{2} + Q'(e, W(e))}{\mathrm{RTT}\left(\frac{b}{2}W(e) + 1\right) + \frac{Q'(e, W(e))G(e)To}{1-e}}, & \text{When } W(e) < W_{\max} \\ \dfrac{\frac{1-e}{e} + \frac{W_{\max}}{2} + Q'(e, W_{\max})}{\mathrm{RTT}\left(\frac{b}{8}W_{\max} + 2 + \frac{1-e}{eW_{\max}}\right) + \frac{Q'(e, W_{\max})G(e)To}{1-e}}, & \text{When } W(e) \geq W_{\max}, \end{cases} \tag{54}$$

number of TCP packets arriving at the node B decreases and more TTIs are available on the shared channel. By allocating these TTIs to the other users, we can increase the radio interface bit rate, i.e., the transmission rate of each TCP segment which limits the increase of RTT and, consequently, reduces the degradation of TCP bit rate. Since the impact of scheduling is major in this case, the TCP-DSCH interaction is studied for each of the four schedulers described in Section III-G.

1) Evaluation of n_{TCP}*:* To conduct this analysis, we start by evaluating N_{TCP} as follows.

The probability that a segment is transmitted only once is $(1 - e)$. The TCP segment is transmitted two times with a probability $e(1 - e)$. The retransmission of a segment could be caused by an RTO or a triple duplicate. In the case of a RTO, the timeout (TO) period is To. If another timeout occurs, TO doubles to 2To. This doubling is repeated for each unsuccessful retransmission until a TO of 64To is reached, after which the TO is kept constant at 64To. However, in the case of a triple duplicate, the TO still equals To. Moreover, when the window is small, the retransmission is due to a RTO. Practically, in a wireless network, after the third retransmission, the loss is only due to the timeout TO. The probability of RTO due to the radio interface is q given by (39). If the timeout is 2To, we define in the same way the probability of a RTO as $q2$ (by replacing To by 2To). Probabilities $q4, q8, \ldots, q64$ are defined similarly according to number of retransmissions. Let $x2 = (1 - p)(1 - q2)$ and define $x4, x8, \ldots, x64$ on the same basis. The probability to have three transmissions is consequently equal to $ex2(pF(p, W) + q - pF(p, W)q) + e(1 - e)p(1 - F(p, W))$. The probability to have four transmissions is $e^2(1 - x2)x4$. The TCP segment is transmitted five times with a probability of $e^2(1-x2)(1-x4)x8$. The ensuing probabilities have the same form. Based on the expressions of these probabilities and following several manipulations, the mean number of transmissions N_{TCP} is reported in (56):

$$\begin{aligned} N_{\text{TCP}} = {} & 1 + e - 2e^2 + 3ex2(pF(p, W) + q - pF(p, W)q) \\ & + 3e(1 - e)p(1 - F(p, W)) + 4e^2(1 - x2)x4 \\ & + 5e^2(1 - x2)(1 - x4)x8 \\ & + 6e^2(1 - x2)(1 - x4)(1 - x8)x16 \\ & + e^2(1 - x2)(1 - x4)(1 - x8)(1 - x16) \\ & \times \frac{(1 + 7x16)}{x16}. \end{aligned} \tag{56}$$

Let us now introduce the mean number of transmissions N_{TCP} in the throughput expression of each scheduler.

2) Round Robin (RR): In this case, for a given number of users N_u, the maximum available throughput [throughput/(TCP retransmissions)] is evaluated using $th = E\left(\sum_{i=1}^{N_u} R_i/N_{\text{TCP}}\right)$. R_i is the radio interface bit rate of user i.

3) Fair Throughput (FT): In this case, a given bit rate is allocated to each user. The TCP bit rate is evaluated using (37). However, in the estimation of cell capacity, the mean number of retransmissions N_{TCP} must be included in the cell capacity (28) as follows:

$$\frac{\sum_{i=1}^{N_u} N_{\text{TCP},i} R_i N_{s,i} T_i \sum_m \frac{k_m}{(N \log 2(M)\tau)_{i,m}}}{T} = \frac{W}{\text{SF}}. \tag{57}$$

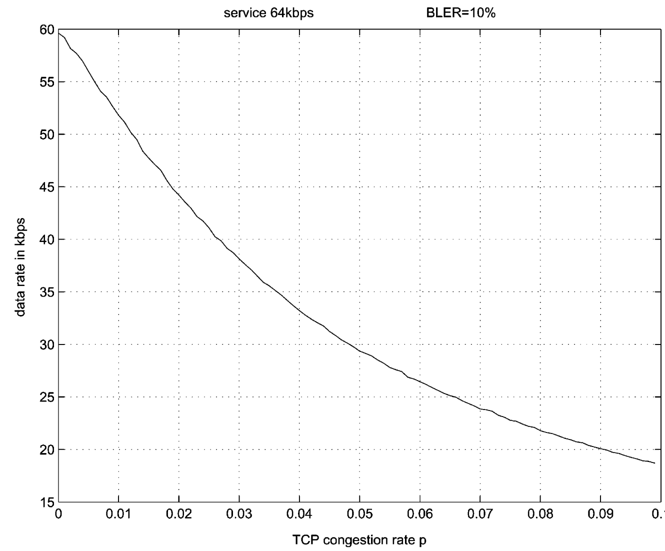

Fig. 1. Effect of TCP on 64 kb/s applications performance.

4) Max C/I: The throughput expression for N_u users for Max C/I is: $th = E\left(\sum_{i=1}^{N_u} R_i/N_{\text{TCP}}\right)$.

5) Proportional Fair (PF): For PF, the throughput expression is the same as in RR and Max C/I: $th = E\left(\sum_{i=1}^{N_u} R_i/N_{\text{TCP}}\right)$.

V. Simulation and Results

The derived analytical expressions of the TCP bit rate and the HSDPA cell throughput are now resolved via a Monte Carlo simulation.

A. Monte Carlo Simulation

The simulated area is a regular hexagonal cell surrounded by 18 cells with a radius of 2 km (macrocell). The distribution of mobiles in the cell is uniform. To estimate the cell capacity, we proceed by a Monte Carlo simulation to resolve (21), (23)–(36). The target SIRs for the modulation and coding schemes are taken from [52] for a BLER of 10%. The packet call for each user has a Pareto distribution. Mobile positions are changed independently of each other to obtain the average value of the throughput. By changing the congestion rate p, the number of retransmissions is increased and cell capacity decreased using (56) and (57). Using (37), (42), (51), (53), and (54) the effective bit rate at the TCP layer is evaluated.

B. Results

Figs. 1 and 2 show, respectively, the bit rate at the TCP level according to TCP congestion rates for 64 and 128 kb/s services (using a FT scheduler). As the bit rate, on the radio interface, decreases the effect of TCP increases. For a congestion of $p = 3\%$, the obtained results are similar to the ones obtained in [28] via simulation. In Fig. 3, the mean TCP user bit rate according to the number of users in the cell is presented (for a congestion of 3% which is an acceptable mean value [12]). Max C/I and PF schedulers give better results than FT and RR schedulers. The mean TCP bit rate in Fig. 3 corresponds to the bit rate of

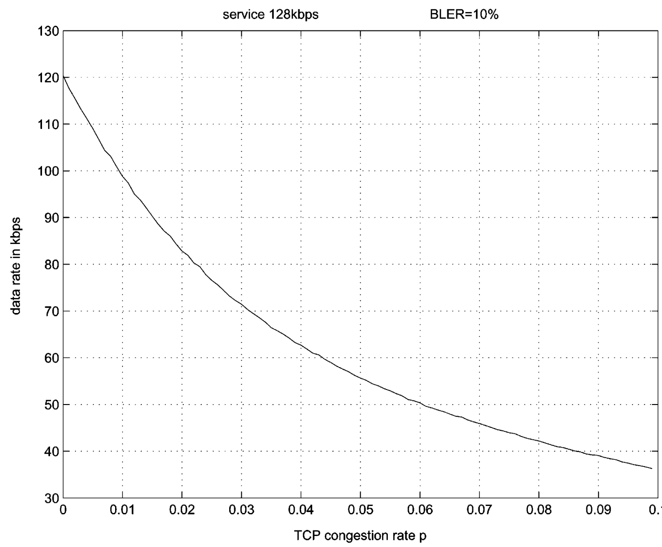

Fig. 2. Effect of TCP on 128 kb/s applications performance.

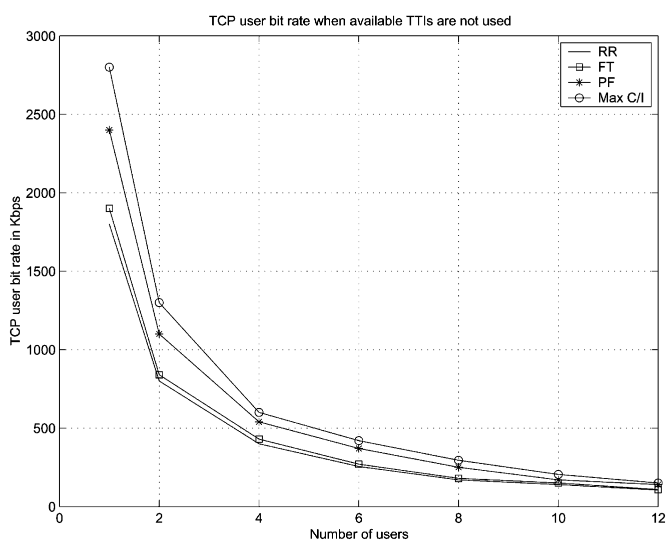

Fig. 3. TCP user bit rate according to the number of users in the cell when traditional schedulers are used.

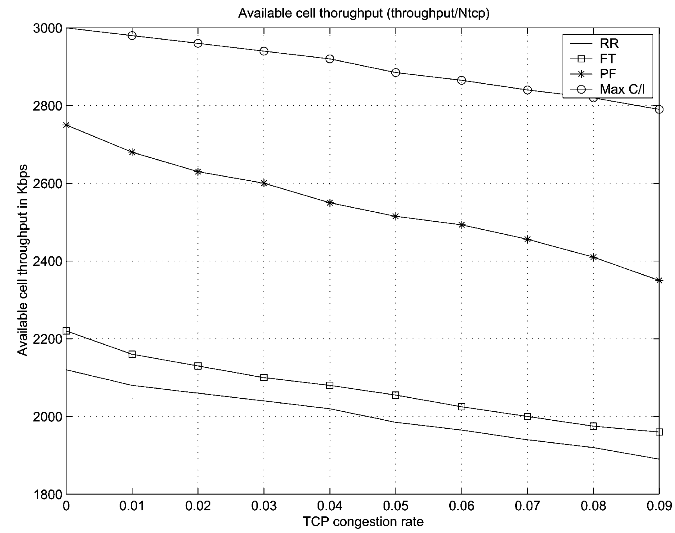

Fig. 4. Effect of TCP on HSDPA cell capacity.

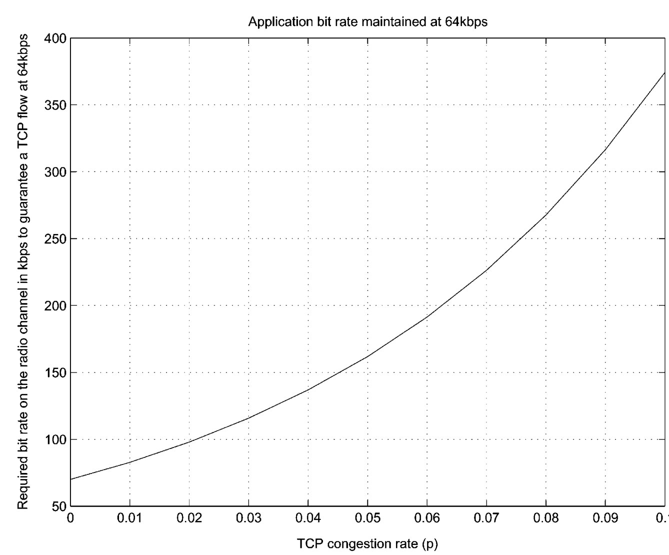

Fig. 5. Required bit rate on the radio channel to maintain a TCP flow at 64 kb/s.

each user in the case of FT and PF schedulers since the bit rate budget is allocated evenly. However, in the case of RR and Max C/I schedulers, the allocated bit rates to users are different and especially so when Max C/I is used (users at the border of the cell could have a bit rate less than 10 kb/s).

Fig. 4 depicts the overall cell capacity according to the TCP congestion rates. The results show that the cell capacity is not affected as much by the TCP layer. Even if the bit rate of a given user is affected (the bit rate decreases), the cell throughput decreases only slowly/smoothly due to the use of the HS-DSCH channels. The results indicate that the radio interface conditions induce a TCP window size reduction which in turn results in decreased user bit rate. However, the number of retransmissions of TCP segments does not decrease at the same rate, the use of the high-speed shared channels makes the cell capacity more robust to radio impairments. Note that Max C/I gives the higher cell throughput compared with the other three schedulers.

C. Discussions and Proposal to Reduce the TCP Effect

The results shown in the Figs. 1–4 lead us to propose a method to achieve the desired flow bit rates and mask the degradation in TCP flow rates induced by the radio interface. The idea is to compute for a given congestion rate p (depending on the application, sender, receiver, and wired network) the bit rate on the radio interface needed to achieve a given TCP effective bit rate. For example, to maintain a 64 kb/s TCP flow with a congestion rate of 3.6%, Fig. 1 indicates that the air interface must operate at a higher bit rate of 128 kb/s. Thus, by increasing the operational bit rate over the air interface in HSDPA, it is possible to achieve the lower application end to end bit rates. The simulation described in Section V-A is repeated to extend the results to other air interface bit rates. The values of p extracted from these simulations to achieve a TCP flow rate of 64 kb/s are reported in Fig. 5. The curve provides the bit rate required over the wireless interface to guarantee a TCP flow of 64 kb/s according to the congestion rate p. Fig. 4 has shown that the cell capacity, at the TCP level, is only slowly affected by an

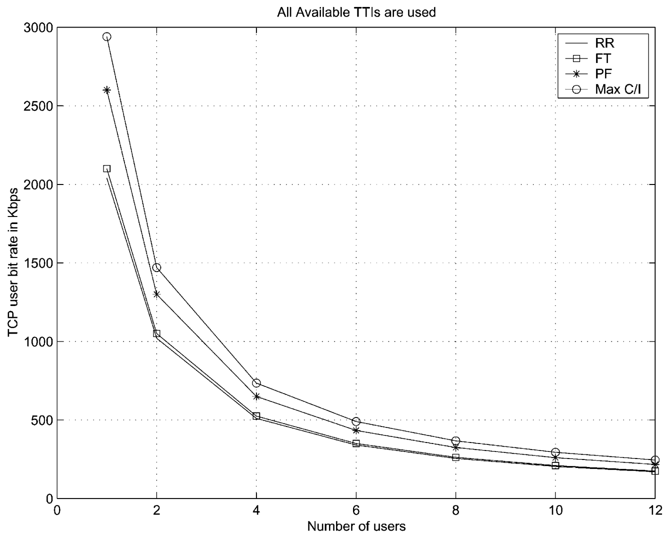

Fig. 6. TCP user bit rate according to the number of users in the cell when modified schedulers are used.

increase in p. The degradation in bit rate is essentially due to the decrease in TCP window size. Even if the wireless bit rate is increased beyond 64 kb/s (to 128 kb/s in the reported case), the TCP flow remains at 64 kb/s. Hence, the number of TCP packets arriving at the node B stays low and this creates space to serve other users on the shared channel. The mean number of users is obtained using the ratio: (TCP capacity)/(TCP flow). The degradation in the number of users (according to Fig. 4) varies between 0%–12% for p varying between 0%–10%. Consequently, the proposed method can maintain a given end to end TCP bit rate at a small cost in cell capacity or can maintain a given cell capacity with a small loss in TCP user bit rate. The schedulers used in HSDPA must, therefore, be adapted to take into consideration the interaction with TCP in order to improve the TCP bit rate.

1) RR and Max C/I: In these cases, the scheduler consists of allocating all TTIs to users that have received data from the TCP level. Instead of allocating N TTIs to a given user (when there is no congestion for this user at TCP level), $N+K$ TTIs are allocated to this user (due to the congestion of other users, i.e., when no data are received by the node B for the other users). If a congestion occurs, less than N TTIs will be allocated to this user. That means, when the channel is allocated to a given user, and other users experience TCP congestion, this user will operate at higher bit rate. This reduces the timeouts RTOs and improves not only the use of shared channels but also the TCP performance.

2) FT and PF: In theses cases, instead of guaranteeing a bit rate R to a given user, the scheduler tries to allocate a higher bit rate as we explained previously. In the case of FT scheduling, the bit rate that we try to achieve is the bit rate at the TCP level. When a PF scheduler is used, the scheduler allocates the channel to the user which has $\max(r/S)$, where S in this case is the mean TCP bit rate instead of the mean radio interface bit rate. Note that guaranteeing a higher radio interface bit rate or a TCP bit rate gives the same performance. However, including the TCP bit rate in the scheduler is more practical than the higher bit rate because it is difficult to estimate the TCP congestion rate.

By changing the schedulers as above, we can evaluate the TCP user bit rate for each scheduler. In Fig. 6, we plot the TCP user bit rate for each scheduler according to the number of users in the cell when the TCP congestion rate is equal to 3% (an acceptable mean value [12]). For the same number of users, our method gives better results than the schedulers based on the radio interface only. The gain is about 40% for 6 users and 33% for 12 users.

Finally, since the performances of PF and Max C/I are close to each other and better than FT and RR (i.e., mean TCP user bit rate for a given number of users), we recommend the use of the PF scheduler. The variations of bit rate around the mean value in PF are lower than in Max C/I (for which users at the cell border can get bit rates lower than 10 kb/s).

VI. Conclusion

This paper proposes analytical models to determine the effect of TCP on the UMTS-HSDPA system performance. Four scheduling techniques are used on the DSCH channel (FT, RR, Max C/I, and PF). Cell capacity and user bit rate are evaluated analytically in each case. Additional insight on the HSDPA system behavior and interactions with TCP is provided. The effect of TCP on application performance is much stronger than on the system capacity in the case of HSDPA due to the use of the high-speed shared channels. Analyzing the results, a method to guarantee TCP flow rates without losing in cell capacity is also proposed. The idea is to operate the air interface at higher rates than the target TCP flow rate (using an appropriate scheduler) to mask the radio impairments. Results afforded by this method to maintain a TCP bit rate of 64 kb/s indicate a loss in cell capacity not exceeding 9%–14% for a TCP congestion rate of 10% and lower than 5%–8% for a congestion rate of 5%. For a given number of users in the cell, our proposal increases the TCP bit rate by 30% to 40%. The results obtained in this paper shows that in HSDPA, it is possible to reduce the effect of TCP in a wireless system, by using the shared channel with an appropriate scheduler (as the modified PF proposed in Section V-C).

References

[1] M. Frodigh, S. Parkvall, C. Roobol, P. Johansson, and P. Larsson, "Future-generation wireless networks," *IEEE Pers. Commun.*, vol. 8, no. 5, Oct. 2001.
[2] S. Parkvall, E. Dahlman, P. Frenger, P. Beming, and M. Persson, "The evolution of WCDMA toward higher speed downlink packet data access," in *Proc. Veh. Technol. Conf.*, vol. 3, 2001.
[3] 3GPP, "HSDPA overall description," TS 25.308 V5.2.0, 2002-03. (Rel. 5).
[4] 3GPP, "Physical layer aspects of UTRA HSDPA," TS 25.848 V4.0.0, 2001-03. (Rel. 4).
[5] 3GPP, "High speed downlink packet access, physical layer aspects," TS 25.858 V5.0.0, 2002-03. (Rel. 5).
[6] 3GPP, "UTRA high speed downlink packet access," TS 25.950 V4.0.0, 2001-03. (Rel. 4).
[7] 3GPP, "Feasibility study for OFDM for UTRAN enhancement," TR 25.892. (Rel. 6).
[8] H. Harri and T. Antti, *WCDMA for UMTS. Radio Access for Third Generation Mobile Communications*, 2nd ed. New York: Wiley, 2002.
[9] D. Chase, "Code combining—A maximum-likelihood decoding approach for combining an arbitrary number of noisy packets, communications," *IEEE Trans. Commun.*, vol. COM–33, no. 5, pp. 385–393, May 1985.
[10] P. Sindhu, "Retransmission error control with memory, communications," *IEEE Trans. Commun.*, vol. COM–25, no. 5, pp. 473–479, May 1977.

[11] A. Das, F. Khan, A. Sampath, and S. Hsuan-Jung, "Design and performance of down link shared control channel for HSDPA," in *Proc. 13th IEEE Int. Symp. Pers., Indoor/Mobile Radio Commun.*, vol. 3, Sep. 2002, pp. 1088–1091.
[12] M. Allman *et al.*, "TCP congestion control," IETF, RFC 2581, 1999.
[13] S. Floyd and T. Henderson, "The New Reno modification to TCP's fast recovery algorithm," IETF, RFC 2582, 1999.
[14] A. Gurtov. (2001) Making TCP robust against delay spikes. Series of publications C, Dept. Comput. Sci., Univ. Helsinki, Helsinki, Finland. [Online]. Available: http://www.cs.helsinki.fi/u/gurtov/papers/report01.html
[15] H. Balakrishnan *et al.*, "TCP performance implications of network asymmetry," IETF, RFC 3449, 2002.
[16] M. Allman and V. Paxson, "On estimating end to end network path properties," in *Proc. ACM SIGCOMM*, 1999.
[17] A. Gurtov and R. Ludwig, "Responding to spurious timeouts in TCP," in *Proc. IEEE INFOCOM*, Mar. 2003, pp. 2312–2322.
[18] R. Ludwig and R. H. Katz, "The Eifel algorithm: Making TCP robust against spurious retransmissions," *ACM Comput. Commun. Rev.*, vol. 30, no. 1, Jan. 2000.
[19] M. Mathis, J. Mahdavi, S. Floyd, and A. Romanow, "TCP selective acknowledgment options," IETF, RFC 2018, 1996.
[20] M. Allman, S. Floyd, and C. Partridge, "Increasing TCP's initial window," IETF, RFC 3390, 2002.
[21] K. Pentikousis, *A Survey of TCP in Wired-Cum-Wireless Environments*.
[22] H. Balakrishnan, V. Padmanabhan, S. Seshan, M. Stemm, and R. K. StemmR, "TCP behavior of a busy Internet server: Analysis and improvements," in *Proc. IEEE INFOCOM*, Mar. 1998, pp. 252–262.
[23] M. Mathis and J. Mahdavi, "Forward acknowledgment: Refining TCP congestion control," in *Proc. ACM SIGCOMM*, Aug. 1996.
[24] A. Bakre and B. R. Badrinath, "I-TCP: Indirect TCP for mobile hosts," in *Proc. Int. Conf. Distrib. Comput. Syst.*, Vancouver, BC, Canada, 1995, pp. 136–146.
[25] R. Cáceres and L. Iftode, "Improving the performance of reliable transport protocols in mobile computing environments," *IEEE J. Sel. Areas Commun.*, vol. 13, pp. 850–857, Jun. 1995.
[26] H. Balakrishnan, S. Seshan, and R. H. Katz, "Improving reliable transport and handoff performance in cellular wireless networks," *ACM/Baltzer Wireless Netw. J.*, vol. 1, no. 4, Dec. 1995.
[27] J. H. Hu and K. L. Leung, "Hierarchal cache design for enhancing TCP over heteregeneous networks with wired and wireless links," in *Proc. GLOBECOM*, Nov. 2000.
[28] M. Meyer, "Analytical model for TCP file transfers over UMT's 3G wireless'01,", 2001.
[29] R. Ludwig, B. Rathonyi, A. Konrad, and A. Joseph, "Multi layer tracing of TCP over a reliable wireless link," in *Proc. ACM Sigmetrics*, May 1999.
[30] M. Yavuz and F. Khafizov, "TCP over wireless links with variable bandwidth," in *Proc. IEEE Veh. Technol. Conf.*, Sep. 2002, pp. 1322–1327.
[31] G. Xylomenos, G. C. Polyzos, P. Mahonen, and M. Saaranen, "TCP performance issues over wireless links," *IEEE Commun. Mag.*, vol. 39, no. 4, pp. 52–58, Apr. 2001.
[32] H. Balakrishnan, V. Padmanabhan, S. Seshan, M. Stemm, and R. H. Katz, "A comparison of mechanisms for improving TCP performance," *IEEE/ACM Trans. Netw.*, vol. 5, no. 6, pp. 756–769, Dec. 1997.
[33] N. M. Chaskar, T. V. Lakshman, and U. Madhow, "TCP over wireless with link level error control: Analysis and design methodology," *IEEE/ACM Trans. Netw.*, vol. 7, no. 5, pp. 605–615, Oct. 1999.
[34] J. W. K. Wong and V. C. M. Leung, "Improving end to end performance of TCP using link layer retransmissions over mobile Internetworks," in *Proc. ICC*, 1999, pp. 324–328.
[35] Y. Bai, A. T. Ogielski, and G. Wu, "Interaction of TCP and radio link ARQ protocol," in *Proc. IEEE Veh. Technol. Conf.*, 1999, pp. 1710–1714.
[36] K. Y. Wang and S. K. Tripathi, "Mobile end transport protocol: An alternative to TCP/IP over wireless links," in *Proc. INFOCOM*, 1998, pp. 1046–1053.
[37] E. Hossain, D. I. Kim, and V. K. Bhargava, "Analysis of TCP performance under joint rate and power adaptation in multicell multirate WCDMA packet data systems," *IEEE Trans. Wireless Commun.*, vol. 3, no. 3, pp. 865–879, May 2004.
[38] J. Peisa and E. Englund, "TCP performance over HS-DSCH," in *Proc. Veh. Technol. Conf.*, vol. 2, May 2002, pp. 987–991.
[39] P. J. A. Gutierrez, "Packet scheduling and QoS in HSDPA," Ph.D. dissertation, Aalborg Univ., Aalborg, Denmark, 2003.
[40] M. Z. Win and Z. A. Kostic, "Impact of spreading bandwidth on Rake reception in dense multipath channels," *IEEE J. Sel. Areas Commun.*, vol. 17, no. 10, pp. 1794–1806, Oct. 1999.
[41] M. Z. Win and J. H. Winters, "Analysis of hybrid selection/maximum ratio combining in Rayleigh fading," *IEEE Trans. Commun.*, vol. 47, no. 12, pp. 1773–1776, Dec. 1999.
[42] ——, "Virtual branch analysis of symbol error probability for hybrid selection/maximum ratio combining in Rayleigh fading," *IEEE Trans. Commun.*, vol. 49, no. 11, pp. 1926–1934, Nov. 2001.
[43] P. J. Bickel and K. Docksum, *Mathematical Statistics: Basic Ideas and Selected Topics*, 1st ed. Oakland, CA: Holden-Day, 1977.
[44] M. Assaad and D. Zeghlache, "On the capacity of HSDPA system," in *Proc. IEEE GLOBECOM Conf.*, San Fransisco, CA, Dec. 2003, pp. 60–64.
[45] K. S. Gilhousen, I. M. Jacobs, R. Padovani, A. J. Viterbi, L. A. Weaver, and C. E. Wheatley, *On the Capacity of a Cellular CDMA System*, May 1991.
[46] L. F. Fenton, "The sum of log-normal probability distributions in scatter transmission systems," *IRE Trans. Commun. Syst.*, vol. CS-8, no. 1, pp. 57–67, 1960.
[47] M. Assaad and D. Zeghlache, "Comparison between MIMO techniques in a UMTS-HSDPA system," in *Proc. IEEE Int. Symp. Spread Spectrum Tech. Appl.*, Sydney, Australia, Sep. 2004, pp. 874–878.
[48] A. Jalali, R. Padovani, and R. Pankaj, "Data throughput of CDMA-HDR a high efficiency-high data wireless personal communication system," in *Proc. 50th Annu. Int. Veh. Technol. Conf.*, vol. 3, Tokyo, May 2000, pp. 1854–1858.
[49] S. Borst, "User-level performance of channel-aware scheduling algorithms in wireless data networks," in *Proc. IEEE INFOCOM*, Mar./Apr. 2003, pp. 321–331.
[50] J. Padhaye, V. Firoiu, D. F. Towsley, and J. F. Kurose, "Modeling TCP Reno performance: A simple model and its empirical validation," *Comput. Commun. Rev.*, vol. 8, no. 2, pp. 133–145, 2000.
[51] N. Cardwell, S. Savage, and T. Anderson, "Modeling TCP latency," in *Proc. INFOCOM*, 2000, pp. 1742–1751.
[52] 3GPP, "Finalizing TFRC reference list and uplink signalling definition," Input Paper, TSGR1 24(02), 2002.

Mohamad Assaad received the B.S.E. degree (with Highest Honors) in electrical engineering from the Lebanese University in Beirut, Lebanon, in 2001 and the M.Sc. degree in telecommunications (with Highest Honors) from the Ecole Nationale Superieure des Telecommunications (ENST), Paris, France, in 2002. He is currently working towards the Ph.D. degree at the Institut National des Telecommunications (INT), Evry, France.

His research interests include UMTS/HSDPA system, scheduling, multiuser detection, MIMO systems, TCP protocol, and cross-layer design.

Djamal Zeghlache (S'83–M'86) received the Ph.D. degree in electrical engineering from Southern Methodist University (SMU), Dallas, TX, in 1987.

In 1987, he joined Cleveland State University, Cleveland, OH, as an Assistant Professor. From 1990 to 1991, through a Summer Faculty Fellowship, he worked with the NASA Lewis Research Center on mobile satellite terminals, systems, and applications. He continued working with NASA Lewis while at Cleveland State University until 1992 when he joined the Networks and Services Department, Institut National des Telecommunications (INT), Evry, France. He is currently head of the Wireless Networks and Multimedia Services Department, INT. He has sustained research and scholarly activities in the field of wireless networks and services. He is currently involved in the European Research and Development project MAGNET and working on advanced wireless PAN technologies. His interests and research activities span a broad spectrum of issues related to wireless networks including radio access (resource management and planning) and core networks with emphasis on end-to-end issues such as cooperation between networks, awareness, reconfigurability, and adaptation.

Prof. Zeghlache is an active member of the IEEE communications Society. He serves on the Steering Committee and the Technical Committee of the ASWN Workshop on a regular basis since 2001. In 2003, he was the Technical Chair of the Wireless Communications Symposium of the IEEE GLOBECOM 2003 Conference.