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PERFORMANCE OF UMTS PACKET SERVICE OVER DEDICATED CHANNELS (DCH)

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The provision of high speed packet data services in an efficient way is probably the most important challenge for the Universal Mobile Telecommunications System (UMTS) since this can give an advantage over second generation systems. The efficiency depends mainly on the radio interface due to its great flexibility. It is of utmost importance to investigate the effect of different configurations on the system performance. In this paper we evaluate the performance of packet data services over downlink dedicated channels (DCH) by means of detailed simulations. Due to the traffic variability, Automatic Repeat reQuest (ARQ) mechanism and closed loop power control the system behavior is quite complex and not so easily predictable as with constant rate services such as voice. We study the effect of several parameters (spreading factor, code rate, channel setup delay, etc.) on the system capacity by means of the delay-throughput curves. We show that the setting of parameters may be critical for system capacity and stability. For this reason we propose a flow control mechanism which is able to guarantee stability and enable the system capacity to be almost independent of channel rates.

Keywords: UMTS; dedicated channels; packet services.

1. Introduction

The success of mobile cellular systems has completely changed the way in which users access telecommunication services. However, even if cellular systems are now widely adopted for telephone services, the number of users which use them to access mobile data services remains very low. This is due to the technological limitations of bandwidth and service flexibility of the second generation systems. As an example, the GSM (Global System for Mobile communications) only provides low rate circuit switched data services unsuitable even for web or e-mail applications due to high latency times and high costs.

To fill the gap between user needs and offered services, second generation systems have been enhanced to include packet data services, such as GPRS (General Packet Radio Service) and EDGE (Enhanced Data rate for Global Evolution), which allow a more flexible use of radio resources and higher peak rates [Kalden *et al.*, 2000]. The commercial introduction of these new services (usually referred to as 2.5 generation services) is coming while the standardization of third generation systems is completed. This is a further challenge to third generation systems, since they must prove their ability to provide access for a wide range of multimedia applications and services in an efficient and cost effective way.

The Universal Mobile Telecommunications System (UMTS) [Samukic, 1998; Richardson, 2000; Gallagher & Webb, 1999] has been standardized by 3GPP, the Third Generation Partnership Project, and is also considered by ITU (International Telecommunication Union) among the standards for the IMT-2000 (International Mobile Telephone standard 2000) family [Ojanpera & Prasad, 1998; O'Mahony, 1998].

One of the main advantages of UMTS with respect to the second generation should be the capability of providing radio access to multimedia services. The traffic being transferred within 3G mobile networks can be composed by different information flows with various constraints on the required QoS (bit rate, delays, etc.). In order to do the job, the radio interface of UMTS is characterized by great flexibility and a variety of different physical and logical channel types [Holma & Toskala, 2001]. For instance, several user rates and channel code protections can be selected by choosing suitable parameters, such as spreading factors, FEC (Forward Error Correction) rates and ARQ (Automatic Repeat reQuest) schemes. This approach is quite different from the one adopted by second generation systems where only a small set of services can be implemented by vendors and provided by operators. On one hand this added flexibility is an advantage of UMTS, on the other hand it makes real services implementation much more tricky. In fact, the complexity of detailed services definition and system parameters optimization has been moved out of specifications favoring UMTS vendors and operators.

In this scenario, the optimization of radio interface parameters is of utmost importance both in circuit switched service and in packet switched ones [Berruto *et al.*, 1998]. Even if CDMA systems have been widely analyzed in the literature, to the best of our knowledge, the effects of the different parameters settings on the performance of UMTS data services have not been thoroughly investigated. In [Borgonovo *et al.*, 2002] we studied the performance of the UMTS Downlink Shared Channel (DSCH) where packet flows are time multiplexed and spreading codes are used basically to fight interference due to transmissions in other cells (inter-cell interference). In this paper, we extend the results previously shown in [Capone & Cesana, 2002] on the performance of packet data services over downlink dedicated channels (DCH). These channels are the same adopted for circuit switched services such as voice, but with packet service their behavior changes significantly due to high variable interference.

The paper is organized as follows. In Sec. 2 we give a short overview of the basic characteristics of UMTS radio interface, and we discuss the issues related to the provisioning of packet services. In Sec. 3 we describe the system model adopted for simulations while in Sec. 4 we present and discuss the results obtained. Finally Sec. 5 concludes the paper.

2. UMTS Packet Data Services

2.1. *Radio access overview*

The UMTS specifications consider two access modes, W-CDMA and TD-CDMA (Time Division and Code Division Multiple Access), to be used in the paired part of the spectrum assigned to UMTS, 60 MHz from 1920 to 1980 MHz (uplink) and 60 MHz from 2110 to 2170 MHz (downlink), and in the unpaired part, 35 MHz from 1900 to 1920 MHz and from 2010 to 2025 MHz, respectively. The W-CDMA mode adopts a chip rate of 3840 Mchip/s, like the high rate TDD mode. It presents a carrier separation of 5 MHz, so that up to 12 carriers can be allocated in the available bandwidth. For the downlink direction a QPSK modulation is adopted after spreading, while for the uplink direction the in-phase and quadrature channels are used to transmit two BPSK flows which may have different spreading codes [3rd Generation Partnership Project, 2000].

The physical layer offers a transport service to higher layers through physical channels. The upper layer information is first protected by the physical layer using FEC codes [3rd Generation Partnership Project, 2000] and then it is spread and modulated with a constant chip rate. The FEC scheme uses convolutional codes with rate 1/3 or 1/2, or a turbo code with rate 1/3. Different rates can be obtained using the rate matching process which can increase the code rate by means of puncturing. The spreading process is based on two codes, namely the spreading code and the scrambling code. The spreading code increases the flow bit rate to the chip rate of the air interface according to the Spreading Factor (SF). Different values of SF ranging from 4 to 512 are available and they are obtained using a tree of orthogonal Walsh–Hadamard codes. The tree has the characteristic that two codes, even with different SF, are orthogonal if they are located in different branch of the tree itself (Orthogonal Variable Spreading Factor, OVSF). Multiple trees can be generated using scrambling codes which vary the order of chips. The channels transmitted by the same station (base or mobile) can use orthogonal codes in the same tree with reduced mutual interference, while channels transmitted by different stations should use different scrambling codes, so that the mutual interference is almost independent of the delay offset at the receiver, due to different propagation paths.

The physical channels are defined by the associated spreading and scrambling codes. The bit flow is divided into time-slots of 666 μs long. During a time-slot both control bits and data bits can be transmitted. The minimum transmission unit offered by the physical layer to the upper layers is the Time Transmission Interval (TTI), called a frame, and is composed of multiples of 15 slots. The transport services provided by the physical layer to the upper layers are based on transport channels which are mapped onto the physical channels [3rd Generation Partnership Project, 2000].

The transport channels are divided into dedicated channels, which can be assigned and then used only for transmissions to/from a single mobile terminal (MT) at any time as well as common channels which are shared on a time division basis by different MTs. Dedicated Channels (DCH) are used to transmit user and control information in the uplink and downlink direction and are devoted to the connection between a single mobile station and the UTRA Network (UTRAN). In the uplink, a DCH is mapped onto two physical channels transmitted in phase and in quadrature, namely the DPDCH (Dedicated Physical Data Channel) and the DPCCH (Dedicated Physical Control Channel), as shown in Fig. 1. The DPDCH carries user data while the DPCCH carries physical signaling used to control the channel. In particular, in the DPCCH the pilot symbols are transmitted for channel response and interference estimation, the TFCI (Transport Format Combination Indicator) symbols describe the format adopted for the channel (mainly SF and error protection code) and the TPC (Transmit Power Control) symbols are used to transmit the commands for the closed loop power control algorithm. In the downlink direction there is a single physical

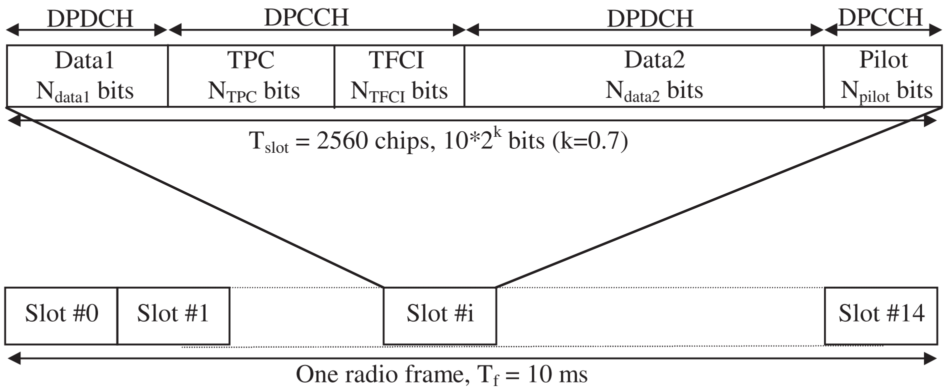

Fig. 1. Physical dedicated downlink channel structure.

channel of dedicated type called DPCH which carries time multiplexed data and control fields.

Power control procedures are defined at the physical layer to adjust transmission power of physical channels according to interference and propagation conditions. The power control exerted on the DCH is based on a closed-loop signaling, as outlined in the following. The TPC symbols in each slot carry a command for increasing or decreasing the transmission power in each direction. The power step is fixed and is usually equal to 1 dB. On the receiving side, if the estimated SIR (Signal-to-Interference Ratio) after despreading is lower than a SIR target value, an increase command is sent to the subsequent slot. A decrease command is sent otherwise. The SIR target value is controlled by an outer control loop which is based on the quality of the decoded bit flow.

The link layer in the user plane is split into the MAC (Medium Access Control), the RLC (Radio Link Control) and PDCP (Packet Data Convergence Protocol) [3rd Generation Partnership Project, 2000]. The MAC layer [3rd Generation Partnership Project, 2000] provides logical channels to the RLC and maps logical channels into transport channels. On common channels, the MAC provides the addressing of user equipments and scheduling of PDUs (Packet Data Units). The RLC layer [3rd Generation Partnership Project, 2000] can offer an acknowledged or unacknowledged data transfer mode. With the acknowledged mode it adds control information on each transmitted PDU and an error check on each received PDU. If the unacknowledged mode is selected, erred PDU are simply discarded, while with the acknowledged mode an ARQ (Automatic Repeat Request) mechanism is adopted. Finally, the PDCP maps each network layer instance into one RLC entity and performs higher layers header compression, if required (for example TCP/IP header compression).

2.2. *Providing packet data access*

In UMTS W-CDMA radio interface the packet data service can adopt dedicated, shared or common control channels for the downlink direction [Holma & Toskala, 2001]. DCH are assigned to single users through set-up and tear down procedures. They are power controlled according to a closed loop mechanism that adjusts transmission power to keep the SIR at a target value. The Downlink Shared Channel (DSCH) time multiplexes packets of different users on the same physical media. To access the DSCH users must have an associated active DCH on the downlink whose power control mechanism is also used to control the power of the shared channel itself. Recently, the standardization bodies have been working on an extension of the DSCH packet service named High Speed Downlink Packet Access (HSDPA), which increases the peak rate to 10 Mb/s and applies to the packet transmission concepts like hybrid ARQ, adaptive modulation and channel coding, fast scheduling, cell selection procedures and eventually MIMO technologies. Finally, the Forward Access Channel (FACH) is a common control channel which can be used to transmit short bursts of data, but, unlike DSCH, no closed-loop power control is exerted and no associated DCH is needed.

Well known results for constant-rate circuit traffic show that CDMA with closed-loop power control is very effective in spectrum exploitation [Viterbi & Viterbi, 1993]. Efficiency can be further enhanced by using powerful FEC codes, which have been proven more effective than spreading codes [Hui, 1984]. Closed-loop power control is able to track interference variations and therefore it stabilizes the BER (bit error rate) and optimizes CDMA performance [Chockalingam *et al.*, 1998].

With packet data traffic the system behavior may be very different and the effect of error control schemes and power control mechanism is not easily predictable. Due to bursty traffic, interference levels are highly variable. When the power control fails due to interference variations or power limitations, the SIR usually decreases to unacceptable values and introduces burst of errors in the bit stream. Even powerful low rate FEC codes can hardly cope with long error sequences. The lost packets can be retransmitted by the ARQ procedure which is usually more effective with burst errors than FEC [McEliece & Stark, 1984; Borgonovo *et al.*, 1999]. However, retransmissions increase channel traffic and interference for users with unfavorable propagation conditions and, therefore, may lead to further losses. This introduces a positive feedback which can cause the system behavior to be unstable depending on traffic and propagation conditions.

The results presented in [Borgonovo *et al.*, 2002] for packet services over shared channels show that, when a single physical shared channel is used, the maximum throughput is achieved by selecting the highest available raw channel rate (i.e. the smallest spreading factor) and using a light FEC protection. Moreover, the optimal SIR target value for packet transmission is 3 dB higher than the one used with voice traffic. Such a margin is needed to fight SIR variations due to the burstiness of packet traffic. We also observed that the system reaches the maximum throughput with a relatively high correspondent packet retransmission probability (about 10%). This proves that errors happen in burst and ARQ is actually able to cope with these errors, while low rate FEC codes are not effective.

However, we have also observed that when we adopt multiple physical channels with lower raw rates (higher spreading factors), a higher throughput can be achieved despite the increased intra-cell interference. This is mainly due to the improved efficiency of the closed-loop power control that takes advantage of the longer transmission time as well as the reduced interference burstiness. This confirms that CDMA characteristics are better exploited with slowly varying interference and circuit-like services, and they pose the question of effectiveness with packet transmissions over multiple dedicated channels.

In the following sections we provide a quantitative evaluation of the performance of packet services over dedicated channels considering different settings of channel parameters.

3. Simulation Model

3.1. *Topology and traffic*

The software we have developed in C++ emulates a real UMTS system with 49 hexagonal cells organized in a wrap-around domain to avoid border effects in the interference calculation. Each cell has an omni-directional antenna with unit gain located at the center.

As in [Borgonovo *et al.*, 2002], we have adopted a traffic generation model based on Web Sessions. The birth of new users is modeled with a Poisson point process of intensity λ [UMTS, 1998]. Each new user, after entering the system, is assigned two physical dedicated channels (DPCH) one for the uplink and one for the downlink. The uplink channel is for the transmission of closed loop power control commands, feedback information from the mobile user to the base station and information for handling retransmissions. At this stage we are not interested in studying the performance of the uplink, which is supposedly ideal. Our attention here is focused on the downlink DPCH which transports the data traffic destined to the mobile user. The spreading factors available for the downlink DPCH can range from 4 to 512.

Each cell is assigned a single tree of OVSF codes. The maximum number of downlink DPCH active in each cell, C, is an input parameter of the simulator. Obviously, C cannot be greater than the maximum number of orthogonal codes which can be assigned, M. If C downlink DPCH are already assigned, any incoming request is buffered. As soon as a downlink DPCH becomes available, it is assigned to the first user in the buffer, and the downlink data transfer phase starts. This approach is typical of a fully packet switched data transfer mode where the requests are queued waiting for the available resources.

The traffic travelling on the downlink DPCH represent the download of a webpage and is modeled by a flow of packets whose number is geometrically distributed with mean $N_p = 25$. The packet length is negative exponentially distributed with a mean of 3840 bits. Packets composing the webpage arrive at the Base Station buffers according to a Poisson point process whose intensity depends on the traffic source speed assumed equal to 929.4 Kb/s in our analysis.

The packets destined to each user are delivered to the RLC layer at the UTRAN (UMTS Terrestrial Radio Access Network) side where they are processed before being queued for transmission. In details, the RLC layer offers an Acknowledged Transfer service subdividing the upper layer packets in transmission blocks (RLC–SDU), and adding a 16 bit header that contains the information useful for ARQ purposes [3rd Generation Partnership Project, 2000]. The RLC blocks are delivered to the MAC (Medium Access Control) layer where they are buffered without any header addition (Transparent MAC Mode)[3rd Generation Partnership Project, 2000]. At each frame, the MAC layer sends to the physical layer a number of blocks up to filling the space available in the frame. Before transmission, the physical layer adds redundancy bits due to the Cyclic Redundancy Check procedure and to the coding scheme adopted [3rd Generation Partnership Project, 2000]. The number of MAC–PDU contained in each physical frame depends on the speed of the downlink DPCH (the spreading factor), and on the code rate of the FEC code adopted.

Our simulator adds the parity bits required by Convolutional Codes, with 256 states, constraint length $K = 9$ and optimal puncturing, whose Bit Error Probability (BER), obtained through link level simulations [Bellini & Ferrari, 2000], is shown in Fig. 2.

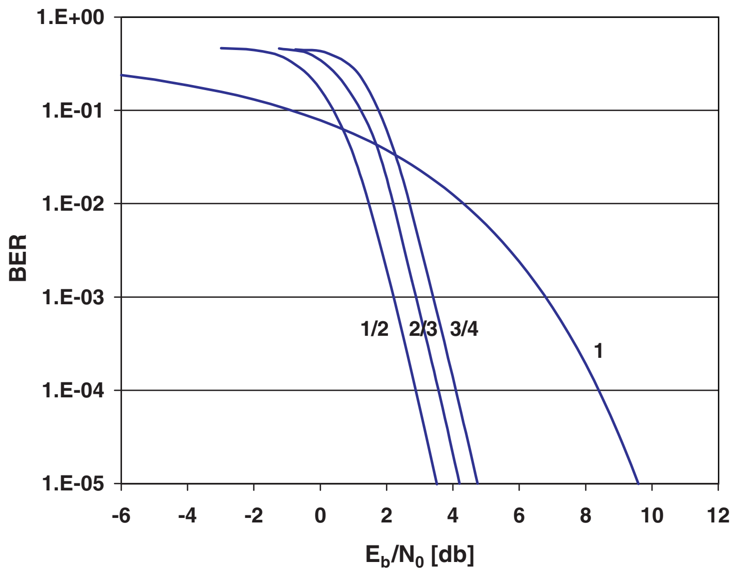

Fig. 2. Bit error rate versus E_b/N_0 when varying the FEC code rate.

3.2. *Propagation model and power management*

The received power P_r of the generic downlink transmission is given by [UMTS, 1998]:

$$P_r = P_t 10^{\frac{\epsilon}{10}} L, \tag{1}$$

where P_t is the transmitted power, L is the path loss, $10^{\frac{\epsilon}{10}}$ accounts for the loss due to slow shadowing, ϵ is a normal variate with zero mean and variance σ^2.

In our analysis, we have considered a macro-cellular environment, for which the cell radius is 300 m, and the path loss L is expressed as

$$10 \log L = -(128.1 + 37.6 \log r)(\mathrm{dB}),$$

where r (in kilometers) represents the distance between the mobile and the base. Furthermore, we assume the shadowing standard deviation to be equal to 10 dB.

Whenever a new user enters the system, the path losses from its position towards all BSs are determined. The user is assigned to the BS reached with the minimum path loss. A user leaves the system as soon as the last packet of the requested webpage has been successfully received. No user mobility nor soft-handover algorithms are considered.

At the receiving side, the carrier to interference ratio is evaluated, for each transmission as

$$\frac{C}{I} = \frac{P_r}{\alpha I_{\mathrm{intra}} + I_{\mathrm{inter}} + P_N}, \tag{2}$$

where P_N is the thermal noise assumed equal to -99 dBm, I_{inter} is the sum of the signal powers received from the other cells, I_{intra} is the sum of the signal powers received from other users in the same cell, and α is the loss-of-orthogonality factor due to the multipath assumed equal to 0.4 according to [3rd Generation Partnership Project, 1999]. All the received powers are obtained by Eq. (1).

From the C/I evaluated as in (2) the normalized energy per information bit is obtained as

$$\frac{E_b}{N_0} = \frac{1}{2R} \times SF \times \frac{C}{I}, \tag{3}$$

where R is the coding rate. The BER curves versus E_b/N_0 shown in Fig. 2 allow us to obtain the BLock Error Rate (BLER):

$$\mathrm{BLER} = 1 - (1 - \mathrm{BER})^l, \tag{4}$$

where l is the transmission block length.

The correctness of the transmission is then decided by testing the value of a normalized random variable against the BLER. Our simulator assumes an ideal ARQ procedure, i.e. the transmitted block is kept in the transmitting queue in case of error and then cancelled otherwise. After 10 failed transmissions the block is dropped and the user is declared in outage.

The dedicated channels are subjected to a closed loop power control according to a procedure composed by two different loops. The inner loop aims at keeping the SIR at the target value by changing the transmitted power, whereas the outer loop dynamically varies the SIR target value in order to distinguish different services with different qualities. In our analysis we consider just one kind of traffic so we suppose the SIR target, i.e. its quality requirement, is given.

The UMTS specifications define a basic closed loop power control mechanism according to which the power-update requests of ± 1 dB for the downlink DPCH are transmitted at every time slot (0.666 ms). In our simulations we have to transmit power updates in every frame (10 ms) to simplify and speed up the simulator. Therefore, in our model, DCH power updates are requested at each frame based on the difference between the SIR target and the SIR evaluated on the last frame. To meet the dynamic of the real mechanism the

Table 1. General simulation parameters.

Topology Model	49 Macrocells, Radius = 300 m wrap-around domain
Propagation Model	$P_r = P_t 10^{\frac{\epsilon}{10}} L, \epsilon \sim N(0, \sigma^2)$ with $\sigma = 10$ dB $10 \log L = -(128.1 + 37.6 \log r)(\mathrm{dB})$
Power Limits	30 dBm on DCH 43 dBm on total BS power
Transport Channel	DSCH SF = 8, 16, 32
FEC Code	R = 1/2 Convolutional
Power Control	Closed loop Every 10 ms
ARQ Scheme	Ideal procedure
Traffic Model	ETSI Web browsing

power updates are in the range of ± 15 dB. This simplification forces all blocks in a frame, which belong to the same user, to be transmitted at the same level. However no impact is expected on the convergence of the mechanism though some slight differences in the number of retransmitted blocks are possible.

Physical power constraint are also added as specified in [3rd Generation Partnership Project, 1999]. The power transmitted on each downlink DPCH cannot exceed the value 30 dBm, whereas the overall power transmitted by a base station is limited to 43 dBm.

Table 1 summarizes the simulation scenarios considered in this analysis.

4. Simulation Results

All the presented results have been obtained on running steady-state simulations [Pawlikowski *et al.*, 2002] 600 seconds long. The first 100 seconds are used as warm-up time, that is to say the statistical results collected during this period are neglected. The remaining 500 seconds are divided into 5 simulation runs. During each run the results are collected and used to calculate one sample of each statistical quantity used for evaluation. The output results have been tested according to the t-student statistical test. For all the measures reported in the following (throughput, BLER, etc) the confidence interval is under 5%, given a confidence level of 95%.

4.1. *Delay-throughput performances of the dedicated channel service*

As outlined in Sec. 2.1, UMTS allows one to set transport channel parameters in a flexible way. In this subsection we study the performance of the DCH service when varying the spreading factor of the physical channel to be used. The performance is measured by the average packet delivery delay as a function of the throughput.

As in the shared service [Borgonovo *et al.*, 2002], the optimum SIR target value comes from a trade-off choice. If too small a SIR target is chosen too many errors will occur since the SIR fluctuations around the target value often drive the system in a condition where the code protection is useless. On the other hand side, with a high SIR target the power requirement increases and too many transmissions tend to be driven into saturation. All the results presented have been obtained with a SIR target value equal to 4 dB, which has been found to be the optimum value for all the spreading factors considered.

Figure 3 shows the average delay versus throughput curves when the service uses 10, 11 and 12 channels per cell with SF = 32 and a convolutional code with R = 1/2. This configuration provides a raw bit rate entering the physical layer of 105.9 kb/s. The best performance is achieved with 11 channels per cell, with a maximum throughput of 1050 kb/s.

We observe that, if the use of 12 dedicated channels per cell is allowed, the system is driven into instability, i.e. the average delay versus throughput curve bends backwards. A similar behavior has been observed using different spreading factors. Figures 4 and 5 show the average delay versus the throughput when using respectively 5 and 6 SF = 16 channels as well as 3 and 4 SF = 8 channels per cell. Also in this two cases, there is a maximum number of channels which can be tolerated by the system, namely two channels with SF = 8 and five channels with SF = 16.

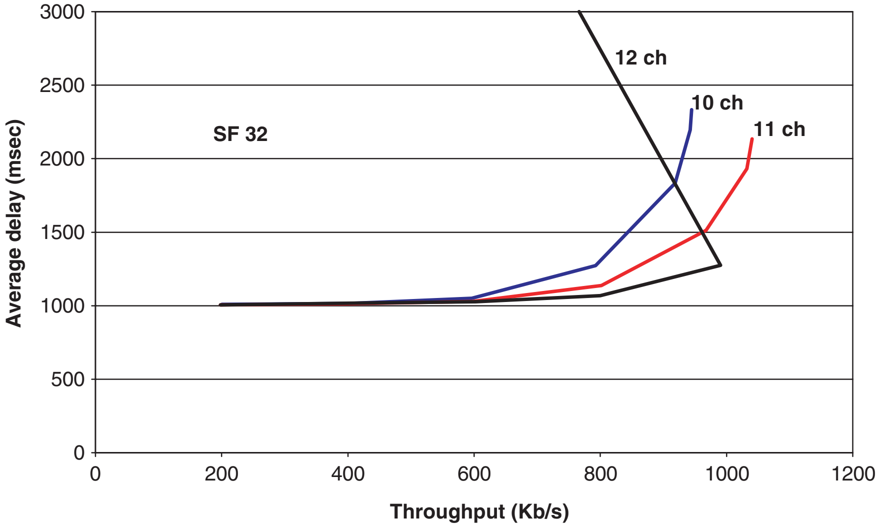

Fig. 3. Average delay versus throughput for SF = 32 and different number of DCHs.

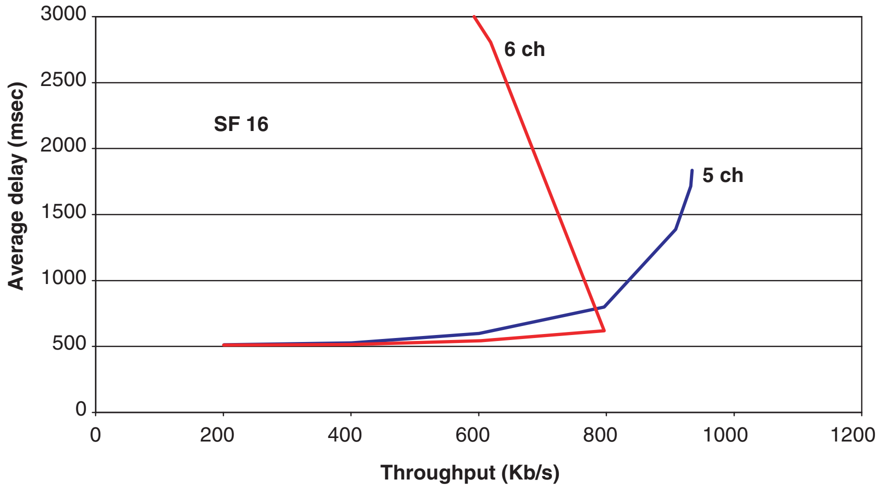

Fig. 4. Average delay versus throughput for SF = 16 and different number of DCHs.

When we increase the number of dedicated channels in each cell, we also increase the interference. The power control procedure tries to counteract such an interference increase by requesting the base stations to transmit more power. However, if a new equilibrium point (a power levels assignment which gives all SIR values equal to the target one) does not exist, the closed loop control mechanism increases all transmission powers until some of them reach the upper limit [Chockalingam *et al.*, 1998]. In these conditions most of the connections cannot reach the SIR target value and, as a result, the fraction of erred transmission increases and the throughput decreases. Moreover, due to the ARQ mechanism which tries to recover lost packets, the traffic on the channel becomes higher and the interference increases further. This is confirmed by the BLER versus offered load curves shown in Fig. 6 for three unstable

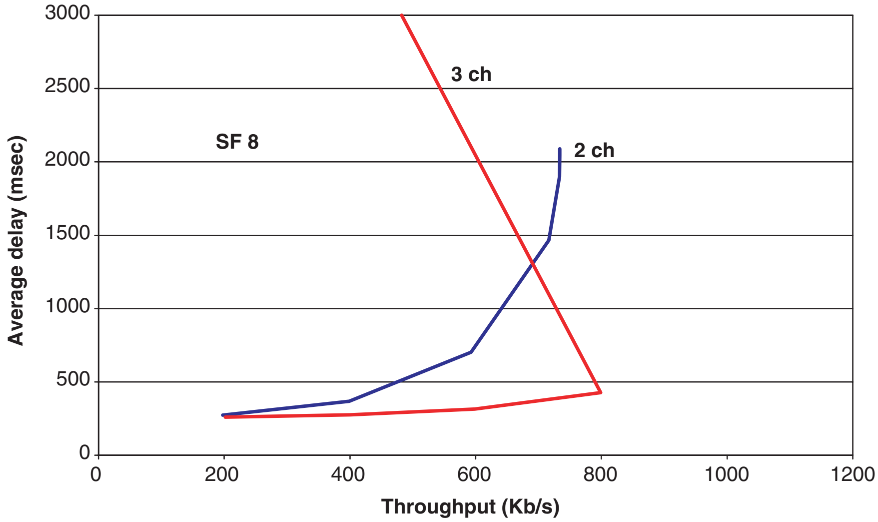

Fig. 5. Average delay versus throughput for SF = 8 and different number of DCHs.

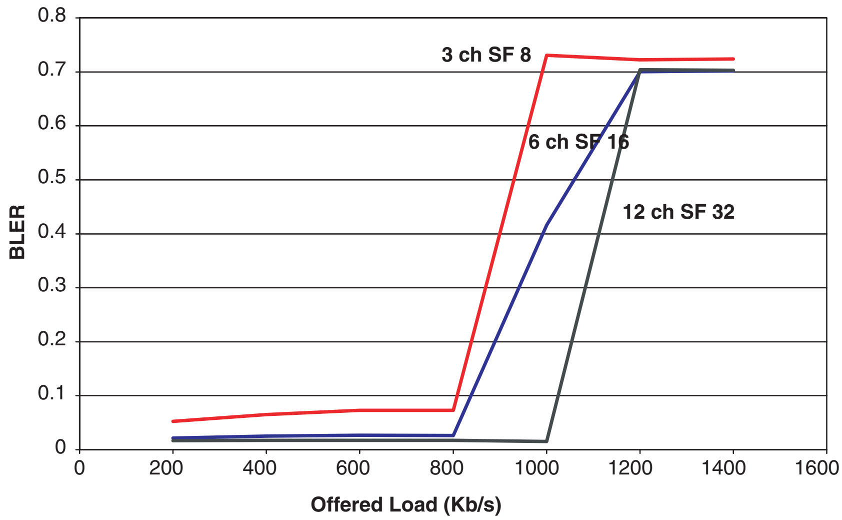

Fig. 6. BLER versus offered load for three unstable configurations: 3 SF = 8 channels, 6 SF = 16 channels, and 12 SF = 32 channels.

configurations with 3 SF = 8 channels, 6 SF = 16 channels, and 12 SF = 32 channels. At high loads the BLER increases over 0.7 for all configurations.

This behavior shows that we need to limit the number of available channels, since there exists a limit on the interference which is tolerable by the system. If not, the power control procedure is not effective in facing interference at high loads and the system efficiency may become very poor. Since we are dealing with dedicated channels which are assigned to single users through a set-up procedure, we can say that such a limitation is a simple call admission control. More flexible mechanisms to limit the interference could be designed [Capone & Redana, 2001], but they are out of the scope of this paper.

Figure 7 compares the results of Figs. 3, 4 and 5, showing the best delay-throughput curves obtained for the three configuration of the dedicated service. The configuration with SF $= 32$ channels is the best from the maximum throughput point of view, but it presents high packet delivery delays even at low throughput due to link layer segmentation of IP packets. On the other hand, the configuration with SF $= 8$ grants the smallest delay for low throughput.

The basic reason for the increase of throughput when channel rate decreases is that the power control mechanism better tracks SIR target since transmission duration is longer and interference variance is lower than with high rate channels, even if the average interference

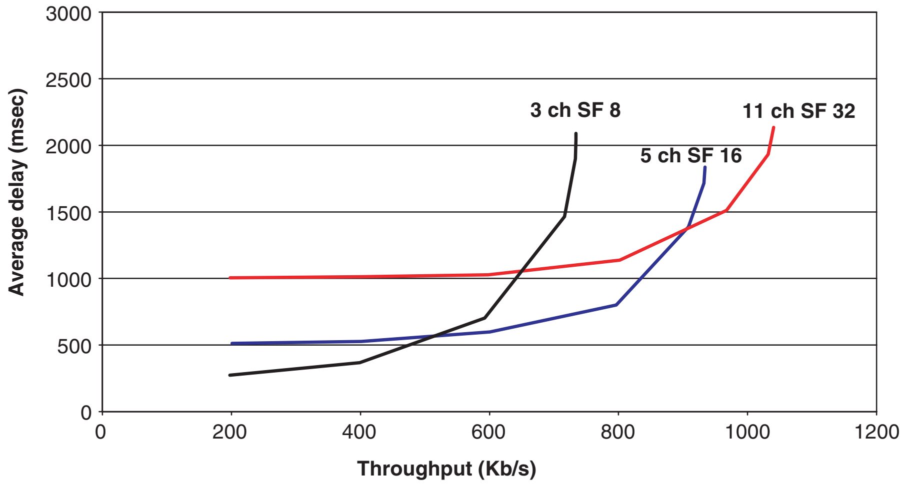

Fig. 7. Average delay versus throughput. Comparison of the best curves for the SF $= 8, 16, 32$ configurations.

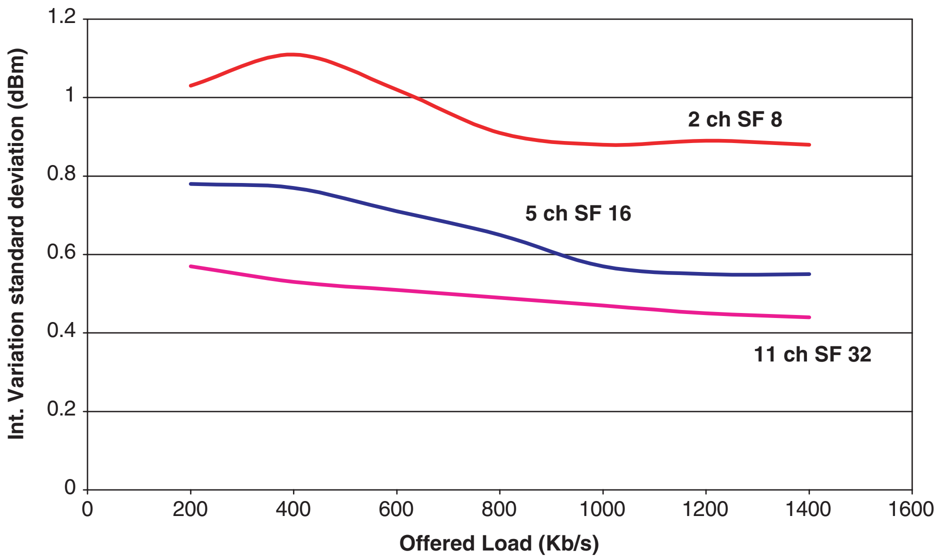

Fig. 8. Interference standard deviation versus offered load. Comparison of the best configurations with SF $= 8, 16, 32$.

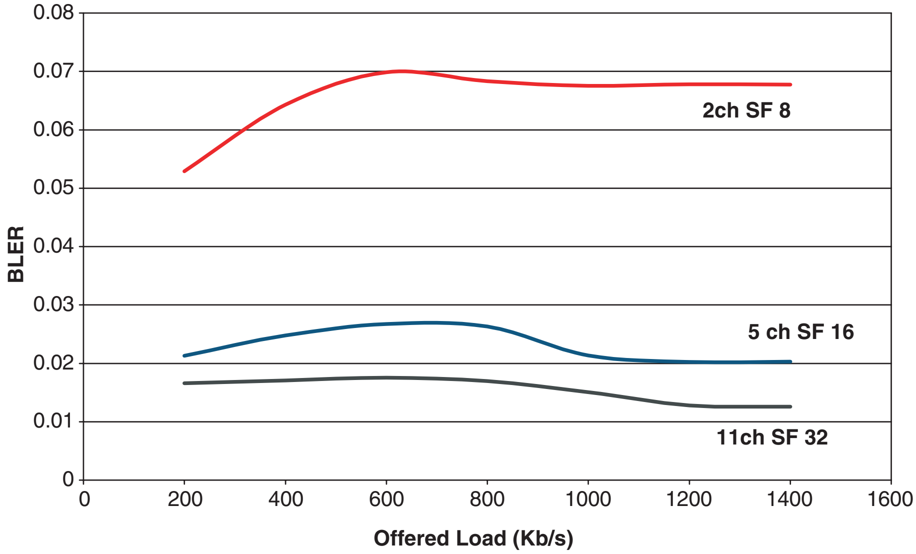

Fig. 9. BLER versus offered load. Comparison of the best configurations with SF $= 8, 16, 32$.

level increases due to the added intra-cell interference. This is confirmed by the results in Figs. 8 and 9 which show respectively the interference standard deviation and the BLER versus the offered load for the same three channel configurations of Fig. 7.

4.2. *Flow control mechanism*

Instability has been proven to occur when the mean interference level gets high and the power control drives the transmitted power into saturation. In such a condition the power control cannot prevent transmission errors, since the requested power is not available due to transmitted power limits. Therefore, limiting the number of channels per cell is an easy mean to prevent performance degradation at high loads. Since this limit must be set in advance it is necessary to take into account all possible scenarios to set a conservative constraint.

However, with packet data service, more effective flow control (FC) mechanisms can be adopted in order to dynamically adjust the load on the active channels, and therefore to control the average interference generated, according to traffic and propagation conditions. Such FC schemes must be able to limit the transmission rate only when necessary to reduce the intensity and frequency of interference peaks.

In the following, we propose a dynamic FC mechanism which is based on a feedback provided by mobile terminals. The basic idea is to reduce the transmission rate on the channel when one or more consecutive transmissions fail. However, since a transmission can also fail due to interference variations and not because of a real traffic congestion (we could say interference congestion), the mechanism is triggered only when a transmission is performed at the maximum power. More in details, a flow-control counter (N_i) is started and increased at each transmission failure if the maximum power level is used. The counter is set to 0 whenever the transmission is successful. Any erred packet is retransmitted after a number of frames K is randomly chosen between 1 and the value of the counter N_i. In

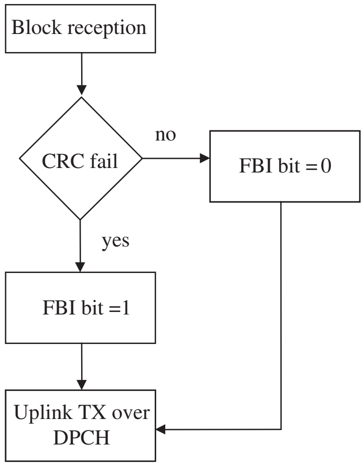

Fig. 10. Flow control scheme mobile equipment side.

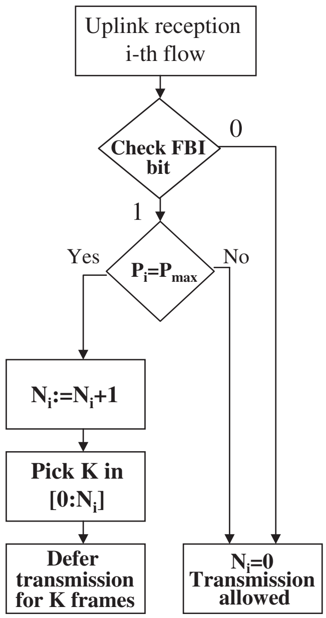

Fig. 11. Flow control scheme mobile base station side.

such a way, we control the traffic on the channels (G), and consequently we limit the mean interference level. To notify the transmission failure/success to the transmitter one of the FBI (Feedback Information Bits) bits defined in the uplink transport block format [3rd Generation Partnership Project, 2000] is used. The flow chart of the FC mechanism is reported in Figs. 10 and 11.

To show the effectiveness of the proposed FC mechanism, we have reported in Fig. 12 the Block Error Rate (BLER) and the Block Error Rate given that the transmission is at maximum power (BLER max) versus the offered load in the two cases with and without the FC mechanism. The physical layer configuration used is the one with 6 channels with

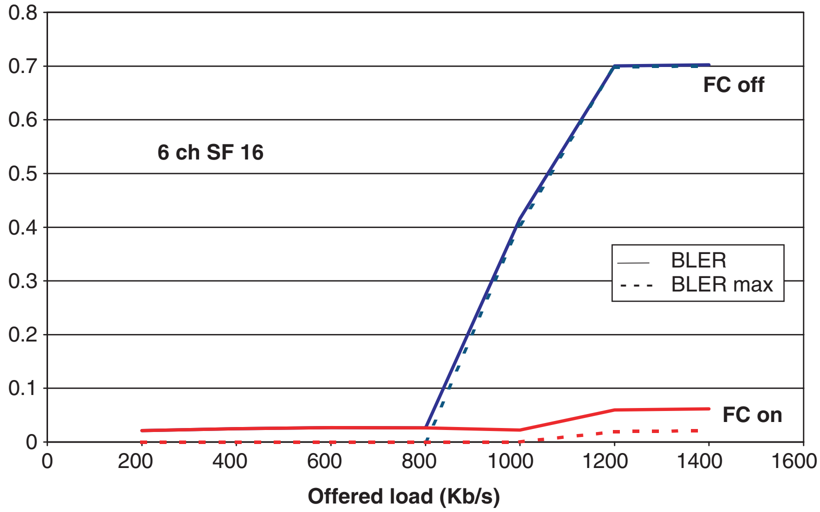

Fig. 12. BLER versus offered load with SF = 16. Comparison of the cases with and without the FC mechanism.

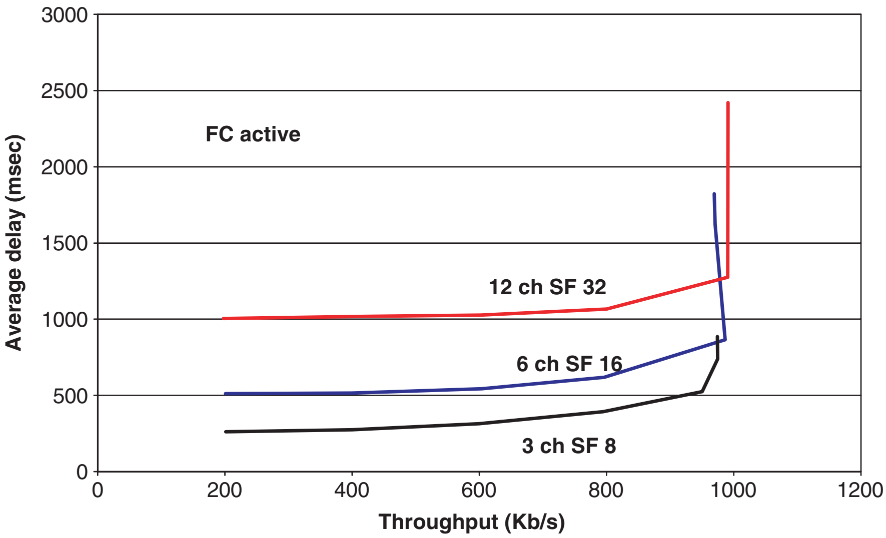

Fig. 13. Average delay versus throughput with the FC mechanism. Comparison of the best curves for the SF $= 8, 16, 32$ configurations.

SF = 16. Without FC, the BLER is very high and almost equal to the BLER max (about 0.7 at high loads). This shows that most of the errors occur when the power control cannot reach an equilibrium point and brings the power at its maximum value. In contrast, in the case with the FC mechanism the BLER is lower (0.1 at high loads) and, furthermore, only a relatively small fraction of errors are due to power limitations. A similar behavior has been observed with the other physical layer configurations (SF $= 8, 32$).

The throughput versus average delay curves obtained with the FC mechanism have been summarized in Fig. 13 which reports only the cases with the highest maximum throughput for SF $= 8, 16, 32$. In contrast with the results obtained without FC (see Fig. 7), the maximum throughput achieved with the three SF values are almost the same. We observed that the FC mechanism forces the system to work with the same average interference levels at high loads by modulating the traffic on the channels (G), with all the spreading factor configurations. This also allows configurations with high rate channels (small SF) to reach a high throughput while maintaining a low delay at high loads. Moreover, in all configurations the system is able to handle one more channel than the corresponding cases without the FC mechanisms.

4.3. *Impact of the set-up delay*

The dedicated channel is assigned to the requesting user through a Radio Resource Management set-up procedure [3rd Generation Partnership Project, 2000]. Higher layer signaling needs a certain interval of time to be completed, so the starting point of the transmission session is delayed with respect to the request time.

In the previous sections, we have considered an ideal scenario where the set-up time is equal to zero, that is to say, a dedicated resource is immediately available to any user requesting a data transfer. Here we analyze the impact of channel activation delay on the performances of the dedicated channel service, considering a constant set-up time.

Figure 14 shows a comparison in terms of average delay versus throughput between the cases of ideal set-up delay (equal to zero) and fixed set-up delay equal to 200 msec in the three physical configurations, using respectively 3, 6 and 12 dedicated channels per cell with SF $= 8, 16, 32$. As expected the set-up delay affects the throughput performance since during the time needed to set up the physical channel the radio resources are not fully utilized. The performance impairment is higher with high rate channels (SF $= 8$), while it is almost negligible with low rate channels (SF $= 32$). The efficiency loss mainly depends on the ratio between the set-up time and the call duration, that is the time needed to complete the data transmission. With high rate channels, the data transmission time is shorter than with low rate channels and therefore the efficiency loss is higher.

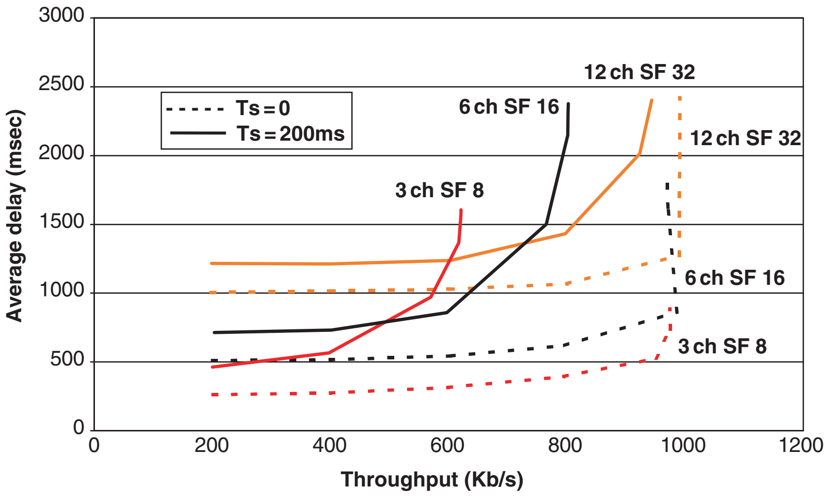

Fig. 14. Average delay versus throughput with different spreading factors and different set-up delays.

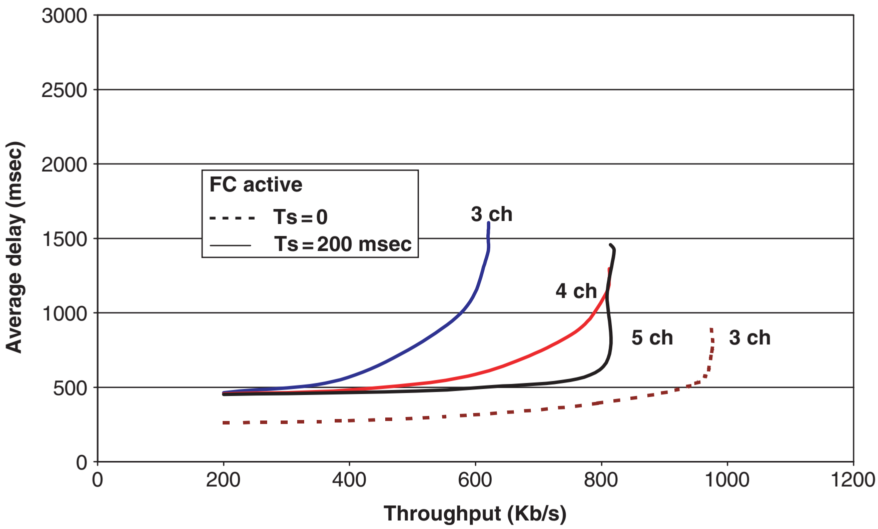

Fig. 15. Average delay versus throughput with SF = 8, set-up delay equal to 200 ms and different number of channels.

However, this result does not tell the complete story. As a matter of fact the set-up delays reduce the amount of time the channels used for transmission and therefore reduce the mean interference. With a reduced interference, the system can tolerate additional channels per cell with respect to the optimum configuration in the ideal case with no set-up delay. Figure 15 shows that up to 5 channels can be contemporary activated in a single cell, when assuming a fixed set-up time of 200 ms and using a spreading factor equal to 8. However, even with an increased number of channels, the maximum throughput is lower than that achieved with three channels in the ideal case (the corresponding curve is also reported in the figure for comparison reasons).

Similarly, Figs. 16 and 17 show that with a set-up delay equal to 200 msec it is possible to increase the number of channels also with SF = 16 and SF = 32. In these two scenarios, the maximum throughput is very close to that achieved with zero set-up delay and, in the case of SF = 32, it is even slightly higher since the flow control mechanism can better control the reduced interference on the channel.

The packet service based on dedicated channels needs some time to activate the physical resource to be used for data transmission. In contrast, with packet services over shared channels, the channels for data transmissions do not need to be set up for each user. However, to access the Downlink Shared Channel a DCH carrying signaling must be set up. It is worth considering also the effects of set-up delays on the performance of this type of packet services. Figure 18 shows the average delay versus throughput curves obtained considering a downlink shared channel mapped onto three SF = 8 physical channels (PDSCH) per cell and a R = 1/2 convolutional code. We observe that in this case the time needed to set up the dedicated channel of each user has no impact on transmission throughput, since the radio resources for data transmission are shared and during this time it can be used by other users.

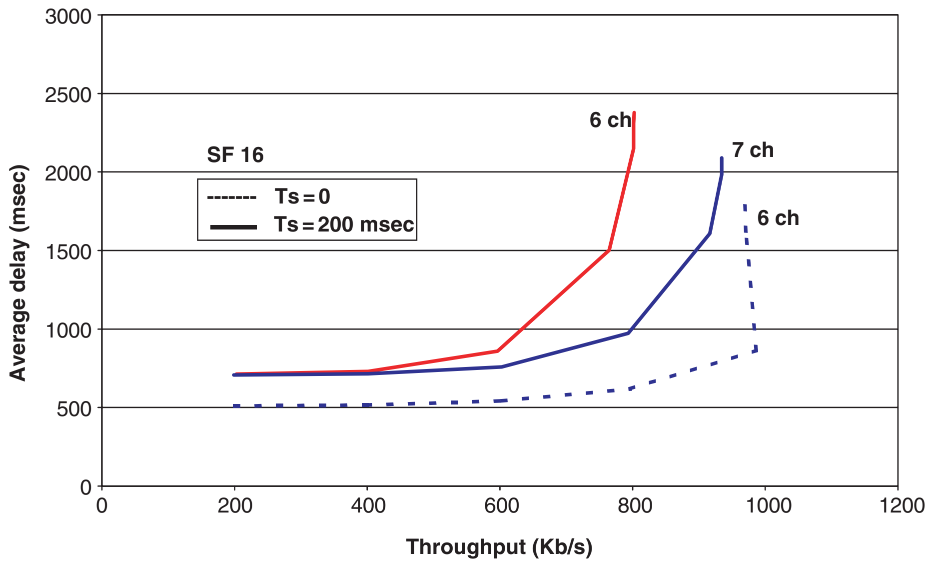

Fig. 16. Average delay versus throughput with SF = 16, set-up delay equal to 200 ms and different number of channels.

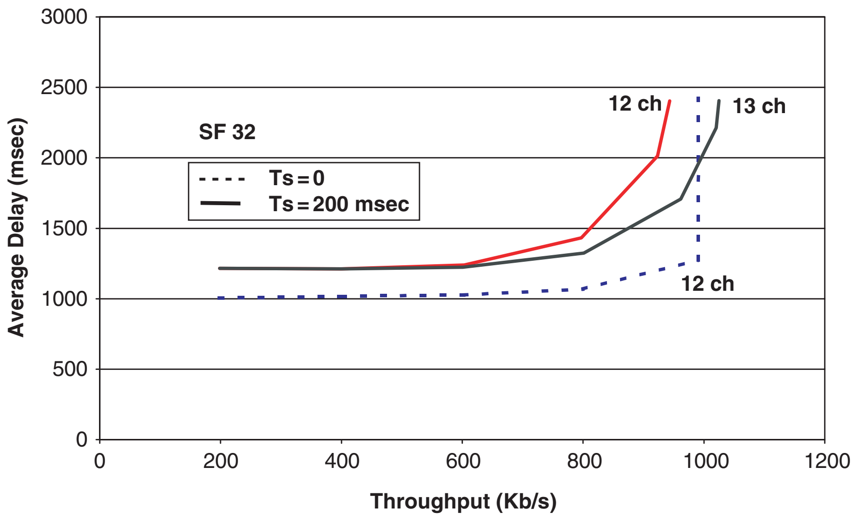

Fig. 17. Average delay versus throughput with SF = 32, set-up delay equal to 200 ms and different number of channels.

5. Conclusions

In this paper we studied the detailed simulation of the performance of the UMTS packet service over dedicated channels.

We analyzed the behavior of the system while varying the speed of the physical dedicated channel to be used (SF $= 8, 16, 32$). For all the three spreading factor configurations, we have proved that there exists a limit on the mean interference which can be tolerated by

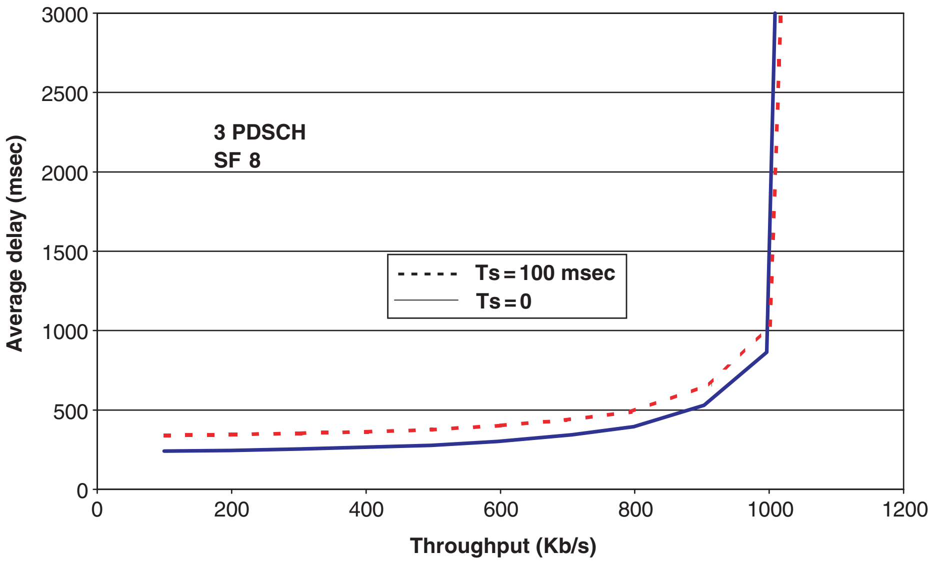

Fig. 18. Average delay versus throughput with SF = 8, set-up delay equal to 200 ms and different number of channels.

the system. Beyond this limit, the closed loop power control cannot provide the SIR target to many active connections and the system is driven to instability.

In order to prevent instability, some kind of interference limiting algorithm should be implemented. In this paper we have proposed a flow control algorithm, which dynamically controls the traffic on the channels and forces the system to work with a tolerable average interference level. We have analyzed the proposed scheme and we have shown that it is able to stabilize the system, providing the same maximum throughput regardless of the channel speed configurations. Furthermore, we have argued that the proposed flow control scheme can be easily implemented within the current version of the UMTS standard.

Finally, since the dedicated channels are assigned through link control set-up procedures, we have evaluated the performance degradation due to the channel set-up delay. In detail, we have pointed out that the throughput loss is more remarkable when low rate dedicated channels are employed. As a matter of fact, in this case the time overhead for the set-up procedure is comparable to the transmission time, and the efficiency is consequently lower. However, since the average interference decreases due to the reduced utilization factor of radio resources, the maximum number of dedicated channels per cell can be increased and, in most of the cases, the maximum throughput can be brought back to the value achieved with no set-up delay.

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