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Adaptive Resource Sharing Strategies for UMTS Multiservice Mobiles

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Abstract. In this paper, a multiservice transmission scheme which joints voice, video, interactive game and signaling is evaluated for the dedicated channel of the Universal Mobile Telecommunications System. The transmission rate for each service is determined by means of an adaptive transport format selection. In the UMTS standard, this selection is based on power allocation provided by radio resource controller, trying to match the Eb/No requirements of the individual services. An appropriate selection of the individual rates of each multiplexed service, taking into account its particular QoS requirements, will provide a more efficient resource management than the transmission only restricted to power allocation. For this purpose, we propose and evaluate a new selection strategy based on buffer occupation, delay requirements and target bit rate keeping power constraints that intends to improve the basic operation trying to minimize the impact of the whole network. Video and game services are multiplexed in upper layers (logical channels) sharing a common transmission rate (transport channel). In this paper, it is proposed a new strategy to share the transmission rate set for this transport channel between both services taking into account their particular requirements. Results show that considering delay constraints provides a more balanced performance of the multiplexed services.

Keywords: multiservice, quality of service (QoS), radio resource management, universal mobile telecommunication system (UMTS), adaptive transport format selection

1. Introduction

The goal of UMTS (Universal Mobile Telecommunications Systems) is to support a large variety of services, which makes the network design and management much more complex when compared to 2G systems, such as GSM (Global System for Mobile communications), and even to 2.5G systems such as GPRS (General Packet Radio Services). GPRS is suitable for non-delay critical data applications, such as consumer mobile web access, where significant volumes of data are sent irregularly. However, new applications require networks being able to handle services with variable traffic conditions keeping the efficiency in the network resources utilization. UMTS air interface has to be able to cope with variable and asymmetric bit rates, up to 2 Mbps and 384 kbps in indoor and outdoor environments, respectively, with different quality of service (QoS) requirements (mainly bit and frame error probabilities and delay) such as multimedia services with

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bandwidth on demand [Laiho et al., 9]. Effective access protocol is also essential for the UMTS air interface to handle burstly real time and non-real time data.

The main technological feature of the UMTS system is the use of the wideband CDMA multiple access scheme in the air interface. The main advantage of WCDMA to provide differentiated services is the option of variable transmission data rates through different spreading factors, multicode transmission and coding schemes. Getting these required QoS for each user is closely connected with power allocation. However, this flexibility requires a more complicated management in front of GSM/GPRS. Given that the real available capacity is limited by the interference, the acceptance of a new user connection is conditioned by the fact that target signal to interference ratio (E_b/N_0) values must be achieved by each existing connection once a new one is activated. In fact, power is the common shared resource for users. In this paper, an analysis of service multiplexing in the dedicated channel (DTCH) is carried out in a realistic WCDMA UMTS environment that considers the different aspects concerning the physical layer, functionalities of the UMTS protocol stack and system capabilities. The joint transmission of voice, video, interactive game and signaling for both uplink and downlink has been studied. UMTS also allows to multiplex different services in upper layers (logical channels) sharing a common transmission rate (transport channel). This multiplexing is proposed for video and game transmission (services with similar QoS requirements). We propose and evaluate a new strategy to share the transmission rate set for this transport channel between both services taking into account their particular requirements.

When a connection is accepted, certain resources are allocated for this user and transmission rates for each service must be determined. The Radio Network Controller (RNC) [Laiho et al., 9] allocates each new user connection assigning to that mobile the set of transmissions rates for the different combinations of the multiplexed services that this user is demanding to be dispatched. This assignment is made by the Radio Resource Controller (RRC) entity [14] and depends on the total required Signal to Interference Ratio (SIR) of the new connection and the residual capacity of the system. Mobile users choose their effective transmission rates restricted to the available set provided by the RRC entity. An appropriate selection of the individual rates of each multiplexed service, taking into account its particular QoS requirements, will provide a more efficient resource management than the transmission only restricted to power allocation. The UMTS standard specifies a way of selecting this transmission rate based only on power restrictions, but it allows providers and manufacturers to implement additional mechanisms. For this purpose, we propose and evaluate a new selection strategy that intends to improve the basic operation through interlayer cooperation considering not only power restrictions (available resources) but also the QoS requirements of the multiplexed services trying to minimize the impact of the whole network. In order to accomplish this goal, the proposed strategy is based on buffer occupation, delay requirements and target bit rate keeping power constraints.

The remaining paper is organized as follows. In section 2, a description of the UMTS system functionalities, services and service capabilities is given and the proposed strategies are described. In section 3, the system model is presented and main parameters are described. We discuss performance results in section 4 and, finally, conclusions are provided in section 5.

2. QoS for the UMTS air interface

2.1. UTRA-FDD protocol architecture

Provision of QoS in the UMTS air interface is related to functionalities of the radio interface protocol architecture, which is shown in figure 1. The WCDMA physical layer offers data transmission services to Medium Access Control (MAC) layer [10] by means of Transport Channels (TrCh). The set of specific attributes of the physical layer (channel coding, interleaving, and transmission rate) is referred to as the Transport Format (TF) of the considered TrCh, and it determines the transmission quality for the data information to be sent. The MAC entity is responsible for mapping Logical Channels (LCh) onto transport channels, selection of TF, priority handling and dynamic scheduling. An LCh is defined by the type of transferred information. A set of LChs types is defined for the different kinds of data transfer services offered by MAC. They can be dedicated, shared and common channels.

Several LChs, belonging to different services (e.g., video, audio, etc.) can be jointly transmitted using different TrChs. Each LCh can be assigned to a different TrCh, or some LChs can be multiplexed over the same TrCh. The set of TrChs is multiplexed over the Coded Composite Transport Channel (CCTrCh) [Baey et al., 1; 11]. Each of the multiplexed TrChs may have variable data rate on a TTI (Transmission Time Interval) by TTI basis, including zero rate. Each combination of rates on the individual channels results in a certain data format to be transmitted, defining the total number of bits per

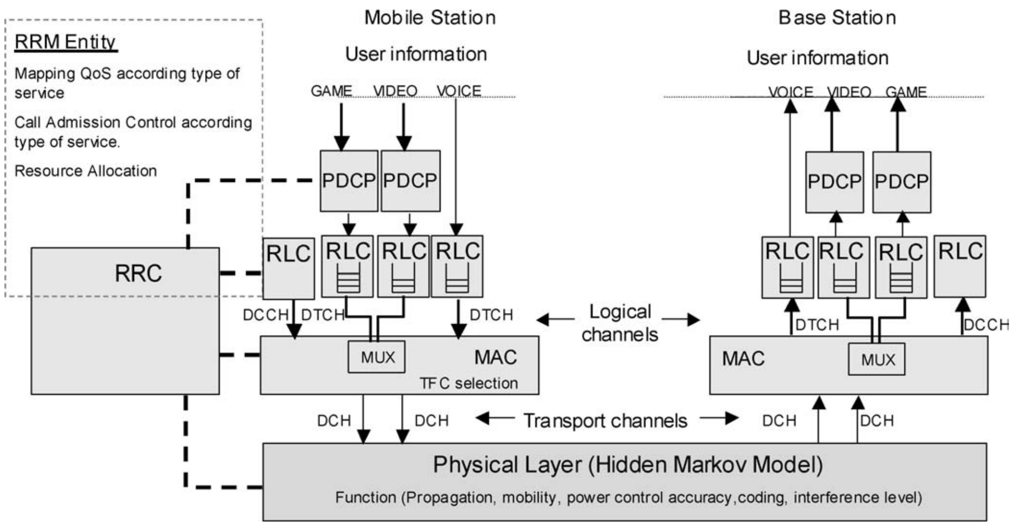

Figure 1. UTRA-FDD radio interface protocol architecture.

frame and their assignment to the individual channels (Transport Format Combination – TFC) [11].

The Radio Link Control (RLC) protocol [13] can provide a reliable service dependent transmission by selecting its operating mode. In fact, the UMTS system considers three different modes in RLC configured by the RRC: based on Forward Error Correction (FEC) are Transparent Mode (TM) and Unacknowledged Mode (UM) whereas the Acknowledged Mode (AM) is based on joint FEC and Automatic Repeat reQuest (ARQ). Another important feature of the RLC mode is the use of early discard, which allows to drop in the transmitter packets that have exceeded the maximum tolerable delay, reducing the delay of the following packets. The time a packet is allowed to stay in the RLC buffer is controlled by a timer (Timer Discard) whose value is signaled by upper layers. We have considered a packet-dependent value according to (1)

$$T_{\mathrm{Disc},i} = T_{\mathrm{margin}} + T_{\mathrm{TX},i} = T_{\mathrm{margin}} + \frac{L_i}{R_{\mathrm{target}}}, \tag{1}$$

where L_i is the length of packet i and $T_{\mathrm{TX},i}$ the time to transmit packet i at R_{target}, and T_{margin} an additional provided margin.

When different LChs are transmitted over the same TrCh, the MAC entity has to multiplex upper layer PDUs from the different RLC entities into the transport block sets delivered to the physical layer. A basic dequeueing mode as FIFO (First In First Out) does not consider the requirements of the different services multiplexed into the same MAC entity. In this paper, a new proposal, which takes into account these requirements, is evaluated and compared with FIFO. We denote this strategy as LWTFO (Lowest Waiting Time First Out). In this case, the RLC PDUs are stamped with a maximum Waiting Time (WT) that depends on the QoS requirements of the service. RLC PDUs with a lower WT are firstly dequeued. Figure 2 shows an example of both strategies.

RLC layer segments SDUs from the Packet Data Convergence Protocol (PDCP) layer [12] in order to match the LCh size block determined by the MAC entity, according to the selected TF for the associated TrCh. The PDCP layer contains compression methods, which are needed to get better spectral efficiency for services requiring IP packets to be transmitted over the radio interface.

These functionalities are controlled by the RRC, which configures the characteristics of the lower layer protocol entities in the air interface, including parameters for the physical, transport and logical channels whereas it offers services to higher layers for signaling mobility management, call control and session management.

2.2. *TFC selection*

In order to guarantee QoS in terms of Bit Error Rate (BER) or Block Error Rate (BLER), a specific bit energy to interference ratio (E_{b}/N_0) must be provided for each transport channel. A joint symbol energy to interference ratio (E_{s}/N_0) needs to be computed according to the TFC so that all E_{b}/N_0 are met. Rate matching [11] provides the required

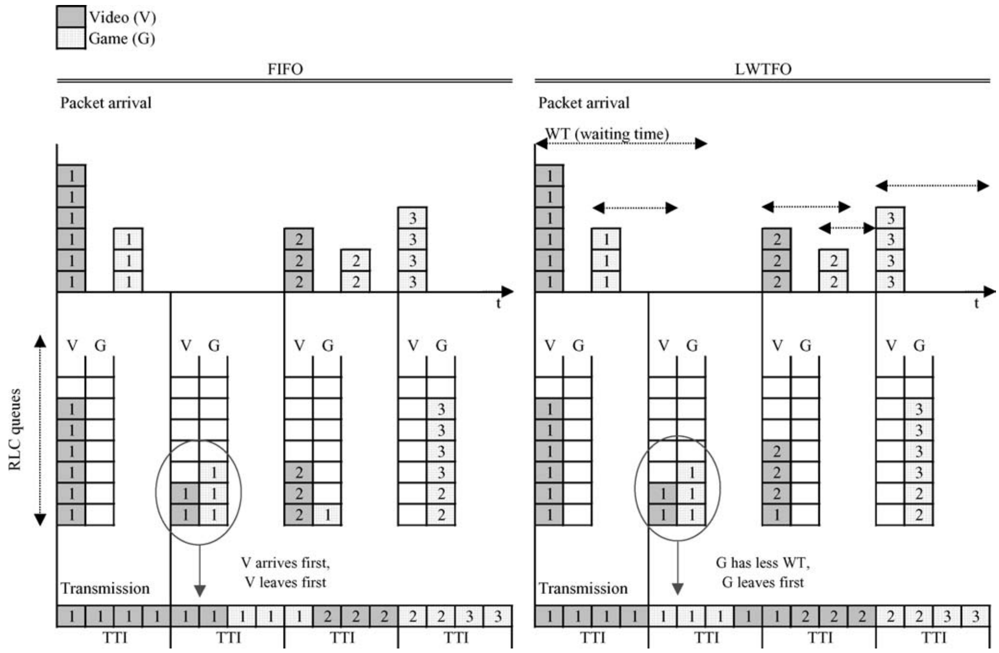

Figure 2. Dequeueing strategies for logical channel multiplexing.

E_b/N_0 for each of the multiplexed transport channels in the CCTrCh. Equations (2)–(5) show the basic process in the uplink:

$$\frac{N_i + \Delta N_i}{N_i} = \frac{(E_c/N_0)_i}{E_s/N_0}, \tag{2}$$

$$\left(\frac{E_c}{N_0}\right)_i = \left(\frac{E_b}{N_0}\right)_i \cdot \frac{R_b}{R_c} \tag{3}$$

for $i = 1, \ldots, I$ (number of transport channels)

$$\sum_{i=1}^{I}\left(N_i\left(\frac{E_c}{N_0}\right)_i\right) = N_{\text{data}}\frac{E_s}{N_0}, \tag{4}$$

$$N_{\text{data}} = \sum_{i=1}^{I}(N_i + \Delta N_i), \tag{5}$$

where R_b and R_c are transmission rates before and after channel coding, N_{data} is the number of bits transmitted over the CCTrCh in a frame with rate R_s (bauds), N_i are the bits associated to each transport channel and ΔN_i the added bits to match the total rate to R_s.

Rate matching is similar in both links, although the process is less dynamic in the downlink. Spreading factor is fixed and Discontinuous Transmission (DTX) is used in addition to repetition and puncturing in order to match the variable transmission rate of each TrCh.

Power transmission is determined by the required E_s/N_0, channel conditions and system interference level in different ways for both links. Equation (6) shows these dependences for the downlink:

$$P_{\text{T-BS},i} = \frac{\eta_0 \cdot W + \chi_i + \rho \cdot P_{\text{T}s\text{-BS}} \cdot h_{i\text{-down}}}{\rho + W/(R_s \cdot (E_s/N_0))} < P_{\text{max-BS},i}, \tag{6}$$

where $P_{\text{T-BS},i}$ is the transmitted power by the Base Station (BS) associated to user i, $P_{\text{T-BS}}$ – the total transmitted power and $P_{\text{max-BS},i}$ – the maximum transmitted power associated to user i, η_0 – the thermal noise spectral density, W – the available bandwidth in the cell (chip rate), χ_i – the intercell interference observed by the user i, ρ – the orthogonality factor, and $h_{i\text{-down}}$ – the path loss between BS and user i. Equation (7) shows the same for the uplink:

$$P_{\text{T-UE},i} = \frac{\eta_0 \cdot W}{C_{\text{res}} h_{i\text{-up}}(1 + W/(R_s(E_s/N_0)))} < P_{\text{max-UE},i}, \tag{7}$$

where $P_{\text{T-UE},i}$ is the transmitted power by user i, $P_{\text{max-UE},i}$ is the maximum transmitted power, C_{res} is the residual capacity in the system, and $h_{i\text{-up}}$ is the path loss between user i and the BS. The residual capacity is given, on average, by expression (8),

$$C_{\text{res}} = 1 - (1 + f) \sum_{j=1}^{N} \left(1 + \frac{W}{\alpha_j R_{s,j} (E_s/N_0)_j}\right)^{-1} \geqslant \eta, \tag{8}$$

where α_j is the activity factor of the source j and f is the ratio between inter and intracell interference. C_{res} is lower limited by η. A minimum value of parameter η equal to 0.1 is considered in order to limit the maximum interference level received in the BS, so that thermal noise represents at least the 10% of maximum total level.

In the downlink, the total transmitted power of a Radio Frequency (RF) carrier is shared by the users transmitting from the BS, so this limits the maximum power allocated to one user, $P_{\text{max-BS},i}$, whereas in the uplink, there is a maximum tolerable interference level at the BS receiver that is shared by the transmitting mobile stations in the cell, each contributing to the interference which restricts the $P_{\text{max-UE},i}$ value.

As a function of traffic demands, TFC selection is performed as long as output power constraints are met. If target TFC cannot be met, a TFC with lower rate and consequently lower power requirement is selected. The TFC must be chosen among a set of TFCs (Transport Format Combination Set (TFCS)), managed by the RRC entity. Radio Resource Management (RRM) is necessary to achieve an efficient use of the available resources. The acceptance or rejection of a new user connection depends on the interference (or load) it adds to existing connections. When a new connection is accepted, through the admission control algorithm, or load traffic conditions change, TFCS is decided by RRC for this user. This TFCS determines the list of allowed TFCs and, consequently, the maximum allowed bit rate for this connection. Once a user is allocated the TFCS, it should select the appropriate TFC in a TTI by TTI basis, in order to guarantee the different QoS of the multiplexed services with the minimum load to the network. The selected TFC in a TTI determines the transmission rate for each of the multiplexed TrCh.

This selection is made based on power allocation provided by RRC, trying to maximize the total transmission rate that brings the individual transmission rates and error probabilities (through E_b/N_0 matching). However, this strategy does not consider other QoS requirements such as delay constraints or instant traffic demands (buffer occupation), which make feasible to tune more precisely the QoS of each multiplexed service. This basic operation is called, in the remaining of the paper as MBR (*Maximum Bit Rate*), since the information is transmitted at the highest available speed. The transmission rate is only reduced if there are not enough packets in the buffer. The only constraint is power allocation.

In this study, it is proposed an alternative strategy for TFC selection, called TRDC (*Target Rate with Delay Constraints*), which takes into account all these service-dependent factors: buffer occupation, target bit rate and delay constraints in addition to power constraints. This strategy tries to smooth traffic, keeping a transmission rate equal to target bit rate if possible, and only increasing it if buffer occupation is higher than expected according to the following policy (rate is decremented if there are not enough packets to match the target rate). Information about the arrival times is taken into account. When PDCP packets arrive at the RLC layer, an expected delay is calculated assuming a transmission rate of R_{target} kbps for that LCh. Each RLC segment belonging to this PDCP packet is stamped with this delay, which is decreased each TTI the RLC packet remains in the queue. In a TTI by TTI basis, the number of RLC packets in the queue and the remaining delay determine an estimated transmission rate for each PDCP packet. When several LChs are multiplexed over the same Transport Channel, the total estimated rate is the sum of the estimated rates of each service (9), and these rates are given by the sum of each PDCP estimated rate according to (10)–(12).

$$R_{\text{est},j} = \sum_{\substack{k \in \text{LCh} \\ \text{in TrCh } j}} \left(\sum_{\substack{i \in \text{PDCP} \\ \text{in queue } k}} R_{\text{est},i,j,k} + R_{\text{ctrl},j,k} \right) \quad (\text{bps}), \tag{9}$$

$$R_{\text{est},i,j,k} = \begin{cases} \displaystyle\sum_{\substack{i \in \text{PDCP} \\ \text{in queue } k}} \frac{n\_rlc_{i,j,k} \times SDU\_rlc \text{ (bits)}}{\tau_{\text{rem},i,j,k} \text{ (sec)}}, & \tau_{\text{rem},i,j,k} > 0, \\ \displaystyle\sum_{\substack{i \in \text{PDCP} \\ \text{in queue } k}} \frac{n\_rlc_{i,j,k} \times SDU\_rlc \text{ (bits)}}{t_{\text{TTI}} \text{ (sec)}}, & \tau_{\text{rem},i,j,k} = 0, \end{cases} \tag{10}$$

$$\tau_{\text{rem},i,0,k} = \frac{PDU\_pdcp_{i,k}}{R_{\text{target},k}}, \tag{11}$$

$$\tau_{\text{rem},i,j,k} = \tau_{\text{rem},i,j-1,k} - t_{\text{TTI}} \quad \text{if } \tau_{\text{rem},i,j,k} > 0, \tag{12}$$

where $n\_rlc_{i,j,k}$ is the number of remaining RLC packets in the queue associated to PDCP packet i for service (logical channel) k in TTI j, $PDU\_pdcp_{i,k}$ the size of PDPC packet i, including padding to match the number of RLC segments, SDU_rlc is the size of a RLC packet without headers, $\tau_{\text{rem},i,j,k}$ the remaining time for RLC packets associated to PDCP packet i in TTI j and $R_{\text{ctrl},j,k}$ the transmission rate associated to RLC control packets for service k.

Once $R_{\text{est},j}$ is calculated, the TFC selection for TTI j is done according to the following rule:

- If $R_{\text{est},j} > R_{\text{target}} \Rightarrow$ increase transmission rate (select TF whose rate is nearest to $R_{\text{est},j}$).
- If $R_{\text{est},j} < R_{\text{target}} \Rightarrow$ if there are data to transmit, keep target rate. If there are not, reduce transmission rate.

3. System model

In this work, the joint transmission of voice, video, interactive game and signaling for both links has been studied. To model voice traffic a classical two state ON–OFF Markov model has been used, with means TON (0.4 sec.) and TOFF (0.6 sec.), respectively. An H.263 video trace file with target bit rate of 32 kbps has been considered. Both frame size and frame rate are variable. Mean frame size is 903 bytes and mean interarrival time is 226 ms. A gaming model based on [5] has been implemented.

Figure 3 shows the gaming traffic source model. Similar to other models it defines a packet call arrival process and within each packet call a datagram arrival process. In this model the packet session arrival process is not specified and it is assumed that packet calls are generated indefinitely (for the duration of the simulation). For the packet call

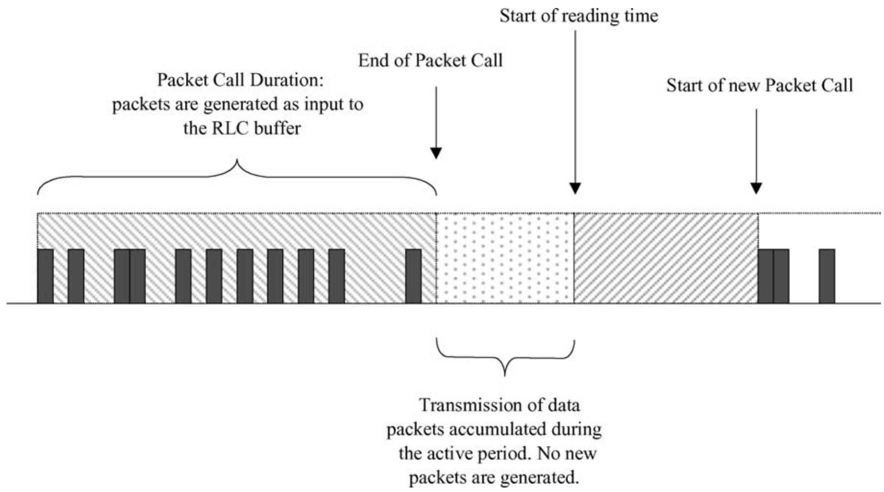

Figure 3. Transmission scheme for the gaming model [5].

Table 1
Gaming model parameters.

Parameter	Value	Comment
Mean packet call duration	5 s	Exponential distribution
Mean reading time	5 s	Exponential distribution
Datagram size	576 bytes	Fixed
Mean datagram interarrival time	160 ms	Log-normal distribution 160 ms standard deviation
Resulting mean data during packet call	28.75 kpps	

arrival process it is specified the packet call (time) duration and the reading time (the time between packet calls). The reading time starts at the successful transmission of all datagrams generated during the previous packet call to emulate a closed loop transmission mode. For the datagram arrival process it is specified the packet size (bits) and the interarrival time between datagrams.

The model for this is largely derived from the so-called "Gaming" measurements [Borella, 2], and therefore originally using the empirically derived distributions specified therein. However, partly as a consequence of the closed loop modeling in figure 3 and for emulating future services with higher bit rates the distributions were modified slightly. For the packet call distributions, both the packet call duration and reading time have exponential distributions. The datagram size is set to a fixed value and the datagram inter-arrival follows a lognormal distribution. The model is very general and can be adjusted easily in terms of required data rates and burstiness by changing the datagram size and the mean datagram interarrival time, equivalently the mean reading time. Table 1 shows the parameter settings to be used in the simulations.

The system simulation model considers a service multiplexing example corresponding to four different services conveyed by five different transport channels for both links: the 12.2 kbps AMR speech service split into three transport channels [4], the 3.4 kbps signaling bearer, a video service and an interactive game. In order to analyze the different TFC selection strategies, the video and gaming services have been mapped onto a single transport channel using different transport formats (128, 64, 32 and 16 kbps). In this study, voice is considered as a priority service, so first it is decided if, according to power constraints, voice could be transmitted. Once voice is allocated, video and game traffic demands are taken into account. Every possible TFC is shown in table 2. Main transport channels parameters are included in table 3. QoS constraints and required E_b/N_0 of these services in a fast fading vehicular environment are also included. Figures 4 and 5 show the multiplexing scheme for the uplink and the downlink, respectively.

Different RLC modes are selected for each service: TM mode is considered for AMR speech service, and AM is selected for the video and gaming applications. Early discard will not be applied to the interactive game, as it requires data integrity. This is carried out with video traffic since this is error tolerant. According to the chosen video traffic source, a variable value for Timer Discard has been selected with T_{margin} equal to 174 ms.

Table 2
Transport format combinations (TFC) for multiplexing of signaling, AMR speech and data flow (video and gaming) over the coded composite transport channel (CCTrCh).

	Services (priority order)					CCTrCh			
	Signaling	AMR speech			Video + game	Uplink		Downlink	
	TrCh0	TrCh1	TrCh2	TrCh3	TrCh4	R_s (kbps)	E_s/N_0 (dB)	R_s (kbps)	E_s/N_0 (dB)
TFC0	1 × 148	1 × 81	1 × 103	1 × 60	8 × 340	960	−6.566	912	−4.782
TFC1	1 × 148	1 × 81	1 × 103	1 × 60	4 × 340	480	−5.980	912	−7.114
TFC2	1 × 148	1 × 81	1 × 103	1 × 60	2 × 340	240	−5.101	912	−9.231
TFC3	1 × 148	1 × 81	1 × 103	1 × 60	1 × 340	240	−6.905	912	−10.722
TFC4	1 × 148	1 × 81	1 × 103	1 × 60	–	120	−6.913	912	−13.705
TFC5	1 × 148	1 × 39	–	–	8 × 340	960	−6.896	912	−5.127
TFC6	1 × 148	1 × 39	–	–	4 × 340	480	−6.574	912	−7.721
TFC7	1 × 148	1 × 39	–	–	2 × 340	240	−6.117	912	−10.269
TFC8	1 × 148	1 × 39	–	–	1 × 340	120	−5.546	912	−12.269
TFC9	1 × 148	1 × 39	–	–	–	30	−5.252	912	−17.638
TFC10	1 × 148	–	–	–	8 × 340	960	−7.000	912	−5.238
TFC11	1 × 148	–	–	–	4 × 340	480	−6.769	912	−7.925
TFC12	1 × 148	–	–	–	2 × 340	240	−6.474	912	−10.643
TFC13	1 × 148	–	–	–	1 × 340	120	−6.193	912	−12.877
TFC14	1 × 148	–	–	–	–	15	−5.406	912	−20.235
TFC15	–	–	–	–	–	0	–	0	–

Table 3
Transport format parameters. QoS requirements.

						QoS requirements		E_b/N_0	
			CRC	Channel coding	TTI	BER	BLER	Uplink	Downlink
Signalling	TrCh0	1 × 148	16 bits	1/3 convolutional	40 ms	–	1.0E−02	4.05	4.20
AMR speech	TrCh1	1 × 81	12	1/3 convolutional	20	5.0E−04	–	4.49	4.73
		1 × 39						5.33	5.57
	TrCh2	1 × 103	0	1/3 convolutional	20	1.0E−03	–	3.36	3.60
	TrCh3	1 × 60		1/2 convolutional		5.0E−03	–	3.31	3.23
Data	TrCh4	8 × 340	16	1/3 turbocoding	20	–	1.0E−02	1.95	3.15
		4 × 340						2.25	3.20
		2 × 340						2.77	3.40
		1 × 340						3.45	3.80

The user is located in a hexagonal cell with radius 2 km. Only the interference from the first-tier of adjacent cells is considered. Propagation model proposed in [15] is adopted for path loss. Log-normally distributed shadowing with standard deviation of 8 dB is also included. A multi-path fading environment (Channel model 3 in [16]) is considered. 11 dB antenna gain and thermal noise power of −103 dBm are assumed [15]. The MS and BS have a maximum output power of 24 and 43 dBm according to [16].

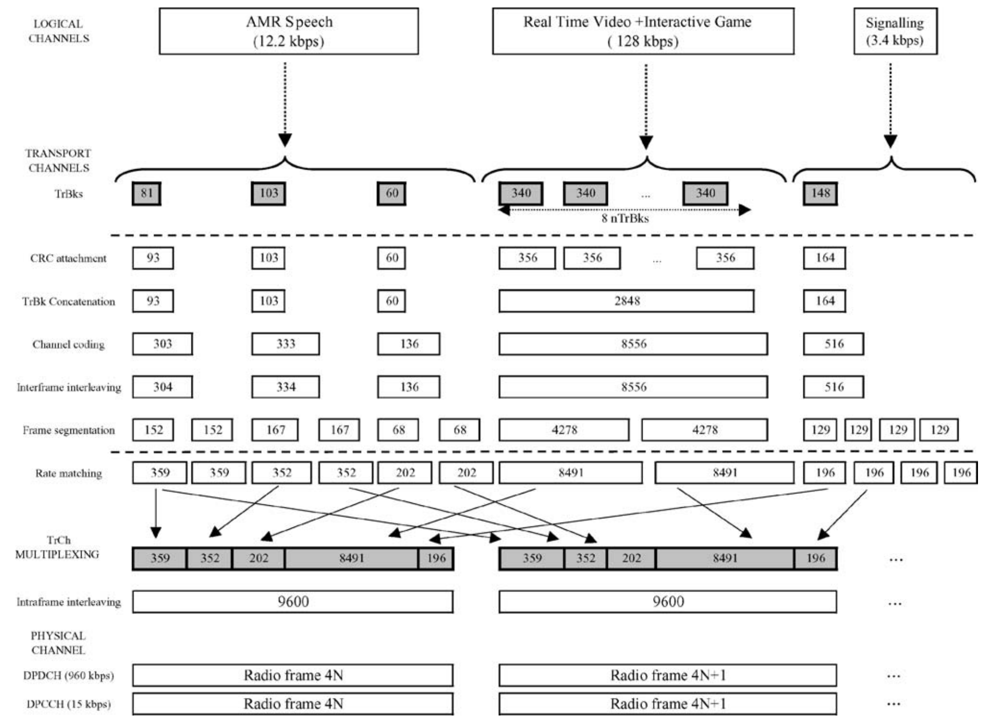

Figure 4. Channel coding, mapping of logical channels onto transport channels and multiplexing scheme in the uplink.

4. Performance evaluation

4.1. Simulation conditions

In order to assess the performance of the multiplexing scheme and the different proposed strategies, a simulation tool that models the UTRA (UMTS Terrestrial Access Radio) FDD system has been developed. This event driven simulator has been implemented in C++, and models the UMTS protocol stack (PDCP, RLC and MAC), the multiplexing of logical and transport channels and the TFC selection process. It allows to evaluate the performance of different RRM strategies, including several traffic sources, propagation conditions and mobility models. Off line results of a system levels simulator [Hernández-Solana et al., 8] and a physical layer simulator [Canales et al., 3; Gállego et al., 6] have been included. The results from the physical level simulator (bit error distribution according to E_b/N_0) are included through a Hidden Markov chain. This simulator considers different aspects concerning the physical layer (channel estimation, synchronization, power control, etc.) including channel coding and interleaving for all TFCs.

Video, game and voice transmission have been simulated according to the multiplexing scheme described in section 3. Simulations have been carried out with different load conditions. In the uplink, the mean number of video users in the same cell ranges

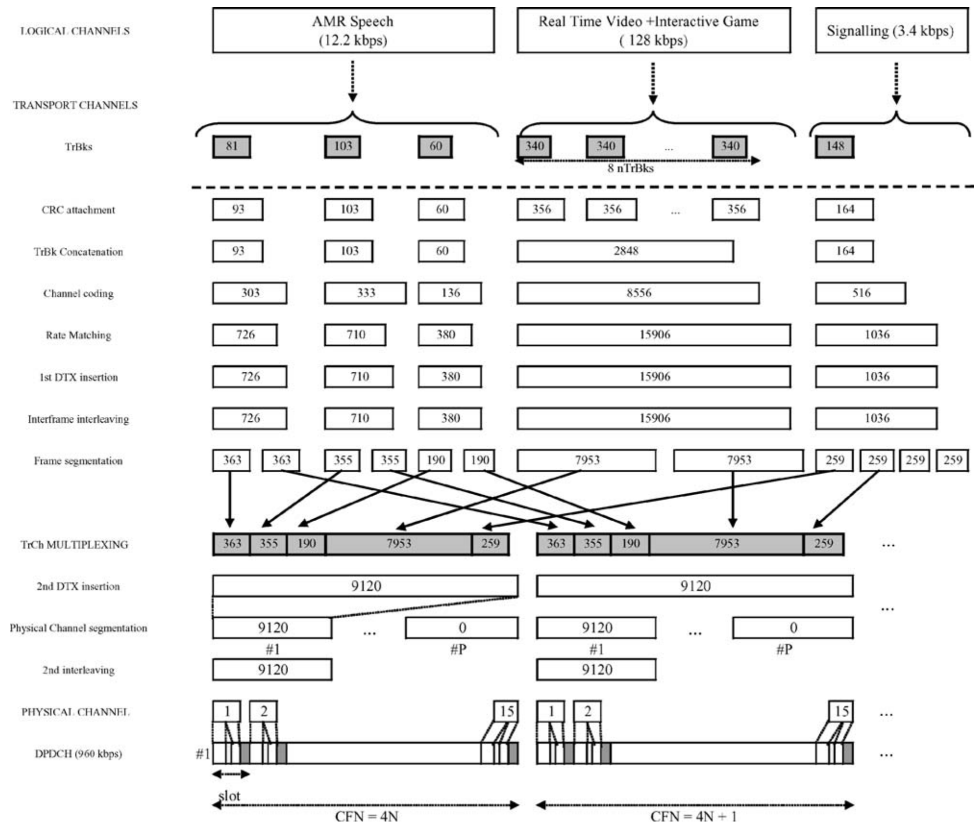

Figure 5. Channel coding, mapping of logical channels onto transport channels and multiplexing scheme in the downlink.

from 15 to 30. In the downlink, the transmission power for the test user is considered to be limited to 5% of the maximum transmission power for the BS, whereas the load conditions are expressed in terms of the fraction of available power that BS is actually transmitting (% utilization, from 25 to 100). The considered QoS parameters [7] for voice and video transmission are a maximum delay of 400 ms and a packet loss of 3%. Game transmission requires a packet delay lower than 250 ms without packet loss.

For video transmission, early discard tries to match the delay constraints removing packets with higher delay from the RLC queue. In order to limit this delay to 400 ms on average, the selected T_{margin} (1) is fixed to 174 ms, considering a mean interarrival time of 226 ms. For game service, which requires data integrity, packet loss is not allowed, so early discard is not used.

Video and game services are multiplexed over the same TrCh using both FIFO and LWTFO dequeueing strategies. In the latter, the maximum waiting time (WT) stamped in RLC packets is based on the QoS requirements. For video packets, this time is considered equal to the discarding time. For game packets, the maximum WT is set to 250 ms.

TFC selection is implemented as described in section 2.2.

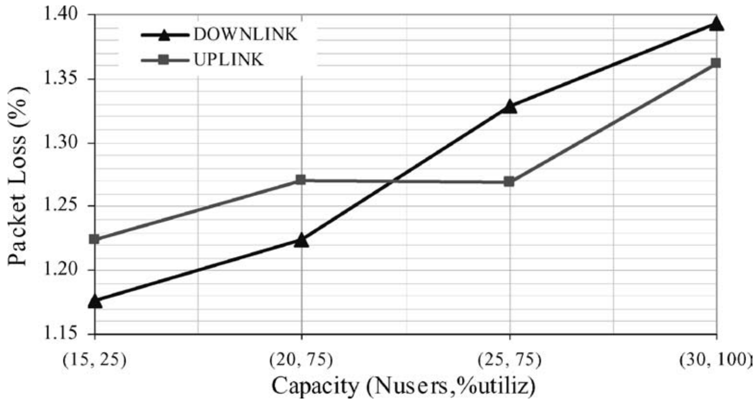

Figure 6. Packet loss for voice transmission.

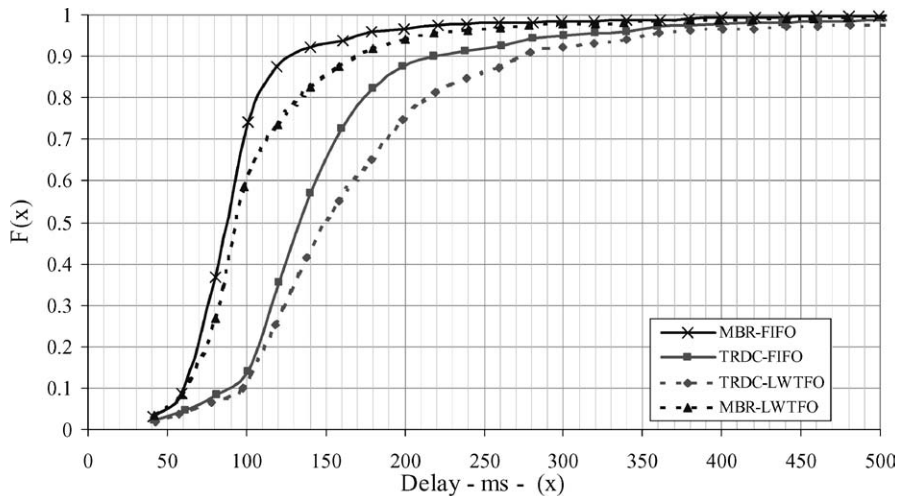

Figure 7. Distribution function for video packet delay in the uplink (20 users).

4.2. *Results*

Figure 6 shows the percentage of packet loss for voice transmission. The QoS requirement is covered in all the simulated load conditions for both links.

Figure 7 shows the distribution function of video packet delay in the uplink for a selected capacity of 20 users. With the TRDC strategy packet delay is obviously higher than with MBR, since the latter uses the maximum available resources (transmission rate) without trying to keep a medium target bit rate. As it is shown in figure 8, discarding probability is higher with TRDC, although these results are close to the MBR performance. For both strategies, the LWTFO dequeueing strategy provides a slightly worse performance for video packets. As game requirements are more restrictive, video packets are dealt with less priority. In any case, with a 3% constraint for packet loss, the

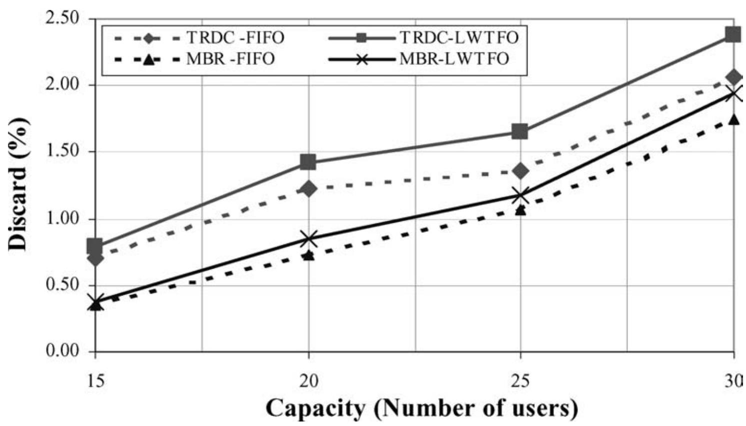

Figure 8. Percentage of dropped video packets in the uplink.

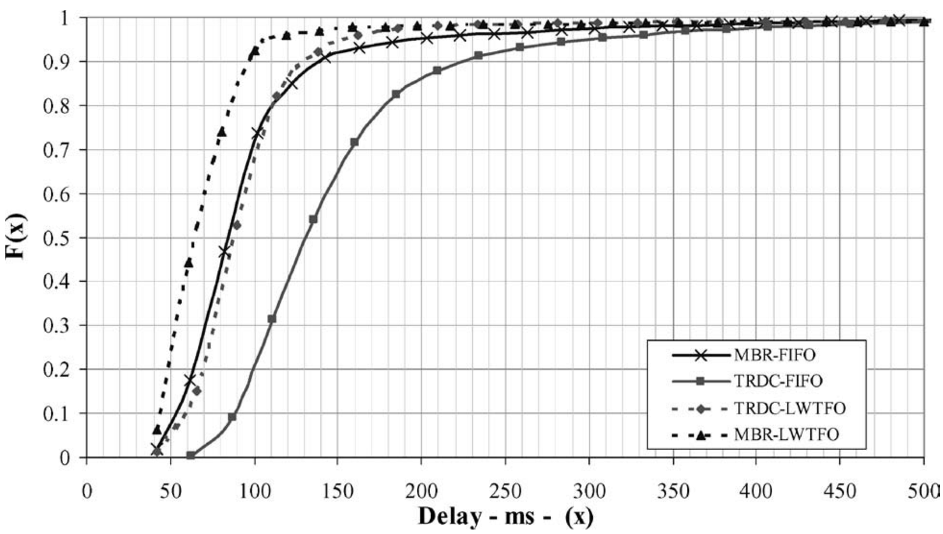

Figure 9. Distribution function for game packet delay in the uplink (20 users).

requirements are covered. However, if the threshold were located at 1%, at certain load conditions, time discard should be longer leading to an increase in packet delay.

Distribution function of game packet delay in the uplink, also for 20 users, is shown in figure 9. As for video packet transmission, packet delay is higher with TRDC. As it is shown in figure 10, using a FIFO dequeueing strategy MBR outperforms TRDC. However, with LWTFO, both TRDC and MBR strategies provide very similar results. This figure shows the percentage of game packets with a delay higher than 250 ms. If we consider an acceptable threshold of 3%, FIFO strategy does not work properly. LWTFO should be the appropriate strategy so that both QoS requirements (video and game) were satisfied. Results for the downlink are analogous.

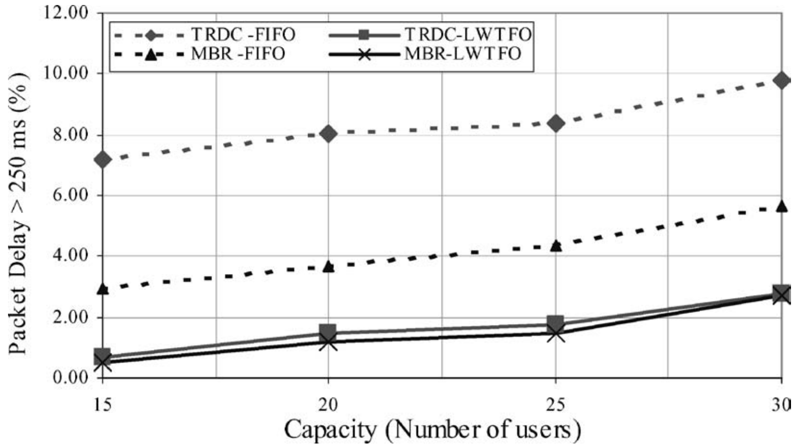

Figure 10. Percentage of game packets with delay higher than 250 ms in the uplink.

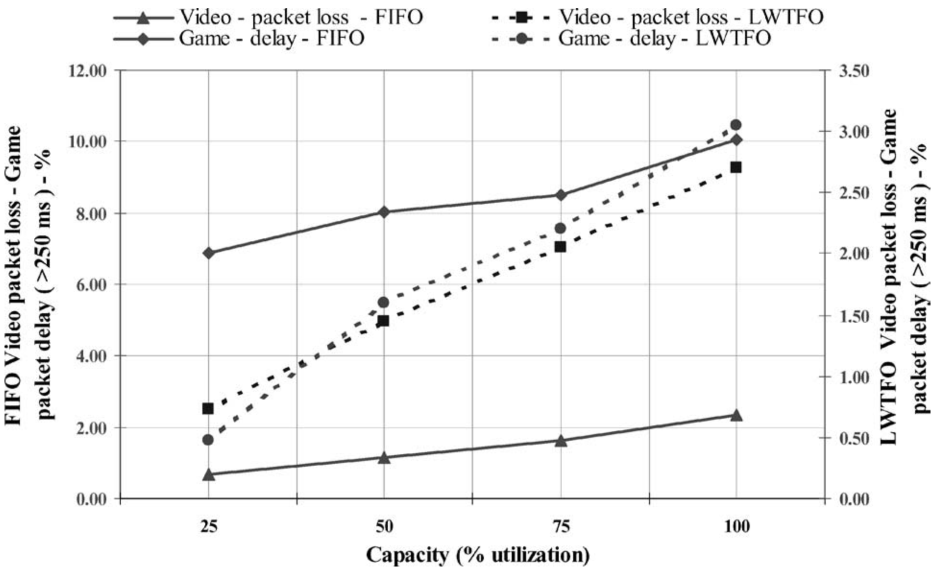

Figure 11. Video packet loss and percentage of game packets with delay higher than 250 ms (TRDC, downlink).

An example is shown in figure 11, which compares the performance of video and game transmission under the same conditions using TRDC. This figure shows that the LWTFO dequeueing strategy provides a more balanced performance achieving the requirements for both services. The mean transmission rate is similar for both MBR and TRDC, unlike TF distribution (figure 12). TRDC smoothes traffic, whereas in MBR, transmission rate mainly alternates between 0 and 128 kbps. From a network management perspective, although MBR slightly outperforms TRDC, this one matches the QoS

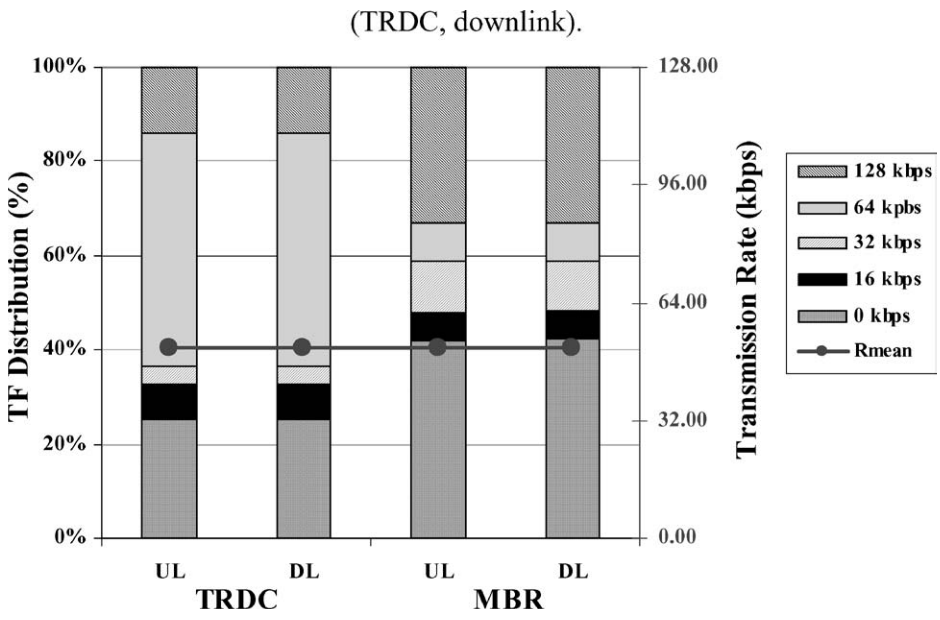

Figure 12. TF distribution. UL (20 users), DL (50%).

requirements with a more stable resource utilization, which would be preferred to avoid high rate fluctuations that would lead to errors in load estimation.

5. Conclusions

A service multiplexing scheme for both links of a WCDMA UMTS user has been described. Transmission of voice, real time video, an interactive game and signaling have been taken into account. A strategy for TF selection, which consider buffer occupation, delay requirements and target bit rate (TRDC) have been proposed and compared with transmitting at the maximum available bit rate (MBR). Multiplexing of logical channels onto transport channels has been analyzed by means of two different dequeueing strategies. Performance of the proposed scheme has been evaluated through a realistic UMTS simulator under different load conditions and in a high mobility situation.

Results show that a TFC selection according to traffic demands and delay constraints (TRDC strategy) matches the QoS requirements, with lower performance than MBR, but with a more stable resource utilization. For logical channel multiplexing, a dequeueing strategy that takes into account QoS requirements (waiting time) provides a more balanced performance of the multiplexed services.

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References

[1] S. Baey, M. Dumas and M.C. Dumas, QoS tuning and resource sharing for UMTS WCDMA multiservice mobile, IEEE Transactions on Mobile Computing 1(3) (2002) 221–235.

[2] M.S. Borella, Source models of network game traffic, Computer Communications 23(4) (2000) 403–410.

[3] M. Canales, A. Valdovinos, J.R. Gállego and F. Gutiérrez, Performance analysis of diversity transmission modes in UTRA FDD under time-varying multipath channels, in: *Proc. of IEEE PIMRC 2002*, Lisbon (September 2002) pp. 1357–1361.

[4] Channel coding and multiplexing examples, 3GPP technical specification 25.944 v4.1.0 (2001).

[5] Feasibility study for enhanced uplink for UTRA FDD, 3GPP Technical Report 25.896 v1.1.2 (2003).

[6] J.R. Gállego, A. Valdovinos, M. Canales and J. de Mingo, Analysis of closed loop power control modes in UTRA-FDD under time-varying multipath channels, in: *Proc. of IEEE PIMRC 2002*, Lisbon (September 2002) pp. 1616–1620.

[7] Group services and system aspects; service aspects; services and service capabilities, 3GPP technical specification 22.105 v6.2.0 (2003).

[8] A. Hernández-Solana, A. Valdovinos and F. Casadevall, Capacity analysis and performance evaluation of call admission control for multimedia packet transmission in UMTS WCDMA system, in: *Proc. of IEEE WCNC 2003*, Vol. 3 (March 2003) pp. 1550–1555.

[9] J. Laiho, A. Wacker and T. Novosad, *Radio Network Planning and Optimisation for UMTS* (Wiley, New York, 2000).

[10] Medium Access Control (MAC) protocol specification, 3GPP technical specification 25.321 v5.4.0 (2003).

[11] Multiplexing and channel coding (FDD), 3GPP technical specification 25.212 v5.3.0 (2002).

[12] Packet Data Convergence Protocol (PDCP) specification, 3GPP technical specification 25.323 v5.2.0 (2002).

[13] Radio Link Control (RLC) protocol specification, 3GPP technical specification 25.322 v5.4.0 (2003).

[14] Radio resource management strategies, 3GPP technical specification 25.922 v5.0.0 (2002).

[15] RF system scenarios, 3GPP technical specification 25.942 v6.0.0 (2002).

[16] UE radio transmission and reception (FDD), 3GPP technical specification 25.101 v5.5.0 (2002).