Adaptive Multimedia Traffic Multiplexing for Dedicated Channels in the UMTS System

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Abstract

In this paper, a multiservice transmission scheme is evaluated for the Dedicated Channel of the Universal Mobile Telecommunications System. The transmission rate for each service is determined according to its QoS requirements by means of an adaptive Transport Format selection. The proposed selection is based on buffer occupation, delay requirements and target bit rate, keeping power constraints. This power restriction depends on the estimated residual capacity and the contribution of this particular mobile user to the system load. Service multiplexing in upper layers (Logical Channels) sharing a common transmission rate (Transport Channel) has been evaluated by means of two different dequeueing strategies.

1. Introduction

One of the key requirements of the UMTS air interface is the support for multiplexing different services with different QoS on a single connection. The main advantage of the WCDMA air interface to provide these differentiated services is the option of variable transmission data rates through different spreading factors, multicode transmission and coding schemes.

The acceptance of a new user connection must be conditioned by the fact that target signal to interference ratio (E_b/N_o) values can be achieved by each existing connection once a new one is activated. Therefore, getting the required QoS for each user is closely connected with power allocation. A good interference handling by radio resource allocation schemes plays an important role to guarantee the performance and to increase the system capacity. When a connection is accepted, certain resources are allocated for this user. Different services can be multiplexed over this same connection with different QoS requirements. According to these QoS requirements and the available resources, transmission rates for each multiplexed service must be determined. In this paper, an analysis of service multiplexing in the dedicated channel (DTCH) for both uplink and downlink 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. Errors in the Signal to Interference Ratio (SIR) estimated by the power control are also considered. A new strategy to select transmission rates of the multiplexed services is proposed and evaluated. This strategy takes into account buffer occupation, delay requirements and target bit rate keeping power constraints. 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, and two strategies to share the common rate are evaluated.

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

Provision of QoS in the UMTS air interface is related to functionalities of the radio interface protocol architecture, which is shown in Fig. 1. The WCDMA physical layer offers data transmission services to Medium Access Control (MAC) layer [1] 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 TrChs, selection of TF, priority handling and dynamic scheduling. A set of LCh types is defined for the different kinds of data transfer services offered by MAC. They can be dedicated, shared and common channels. A LCh is defined by the type of transferred information. Each of the multiplexed LChs may have variable data rate on a TTI (Transmission Time Interval) by TTI basis. Each combination of rates on the individual channels results in a certain data format to be transmitted, defining the total number of bits per frame and their assignment to the individual channels (Transport Format Combination – TFC) [2].

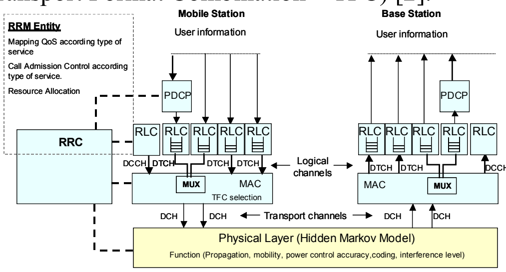

Figure 1. UTRA-FDD Radio Interface protocol architecture

In addition to physical procedures, upper layers provide other feasibilities depending on the service requirements. In particular, the Radio Link Control (RLC) [3] protocol can provide a reliable service dependent transmission by selecting its operating mode. In fact, UMTS system considers three different modes in RLC configured by the Radio Resource Controller (RRC) [4]: based on FEC are Transparent Mode (TM) and Unacknowledged Mode (UM) whereas the Acknowledged Mode (AM) is based on joint FEC and ARQ. The RLC mode also allows 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 and signalled by upper layers. We have considered a packet-dependent value according to (1)

$$T_{Disc,i} = T_{margin} + T_{TX,i} = T_{margin} + \frac{L_i}{R_{target}} \qquad (1)$$

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

Several LChs, belonging to different services (e.g. video, audio, etc) can be jointly transmitted, using different TrChs (with their corresponding TF) multiplexed over the Coded Composite Transport Channel (CCTrCH) [2], [4]. Some of the multiplexed TrChs can also convey several LChs sharing the same TrCh attributes. 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. In this paper, two different dequeueing strategies have been implemented in order to evaluate how the multiplexing strategy affects the system performance:

1) *First In First Out (FIFO)*: RLC PDUs are mapped onto transport blocks in the same order they arrive at the RLC queues, regardless of the LCh they belong to.

2) *Lowest Waiting Time First Out (LWTFO)*: 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.

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_o) must be provided for each service. A joint symbol energy to interference ratio (E_s/N_o) needs to be computed according to the TFC so that all E_b/N_o are met. Rate matching (RM) [2] provides the required E_b/N_o for each of the multiplexed services in the CCTrCH. Equations (2) and (3) show the basic process in the uplink.

$$\frac{N_i + \Delta N_i}{N_i} = \frac{\left(E_c/N_0\right)_i}{E_s/N_0} \quad , \quad \left(E_c/N_0\right)_i = \left(E_b/N_0\right)_i \cdot \frac{R_b}{R_c} \qquad (2)$$

for $i = 1,..., \mathrm{I}$ (number of TrChs)

$$\sum_{i=1}^{I}\left(N_i \times \left(E_c/N_0\right)_i\right) = N_{data} \times E_s/N_0 \quad , \quad N_{data} = \sum_{i=1}^{I}\left(N_i + \Delta N_i\right) \qquad (3)$$

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 service and ΔN_i the added bits to match the total rate to R_s. RM is similar in both links, although the process is less dynamic in the downlink. Spreading factor is fixed and DTX (Discontinuous Transmission) is used in addition to repetition and puncturing in order to match the variable transmission rate of each service. Power transmission is determined by the required E_s/N_o, channel conditions and system interference level in different ways for both links. These dependences are shown in (4) and (5).

$$P_{T-BS,i} = \frac{\eta_0 \cdot W + \chi_i + \rho \cdot P_{T-BS} \cdot h_{i-down}}{\rho + \dfrac{W}{R_s \cdot \left(E_s/N_0\right)}} < P_{MAX-BS,i} \qquad (4)$$

where $P_{T\text{-}BS,i}$ is the transmitted power by the Base Station (BS) associated to user i, $P_{MAX\text{-}BS},_i$ its maximum allowed value, and $P_{T\text{-}BS}$ the total transmitted power, η_0 the thermal noise spectral density, W the available bandwidth in the cell, 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.

$$P_{T-UE,i} = \frac{\eta_0 \cdot W}{C_{res} \cdot h_{i-up} \cdot \left(1 + \dfrac{W}{R_s \cdot \left(E_s/N_0\right)}\right)} < P_{MAX-UE,i} \qquad (5)$$

where $P_{T\text{-}UE,i}$ is the transmitted power by user i, $P_{MAX\text{-}UE,i}$ 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 the expression (6),

$$C_{res} = 1 - (1+f)\sum_{j=1}^{N}\left(1 + \frac{W}{\alpha_j \cdot R_{s,j} \cdot \left(E_s/N_0\right)_j}\right)^{-1} \geq \eta \qquad (6)$$

where α_j is the activity factor of source j and f is the ratio between inter and intracell interference. C_{res} is lower limited by 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, $P_{MAX\text{-}BS,i}$, 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 $P_{MAX\text{-}UE,i}$ 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)), provided by the Radio Resource Management (RRM) entity [4]. The 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 RRM 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 TrChs.

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_o 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. In this paper, it is proposed an alternative strategy for TFC selection 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):

1) *Target Rate with Delay Constraints (TRDC)*: information about the arrival times is taken into account. When upper layer (PDCP) [6] 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 TrCh, the total estimated rate is the sum of the estimated rates of each service (7), and these rates are given by the sum of each PDCP estimated rate according to (8)-(10).

$$R_{est,j} = \sum_{\substack{k\in \text{LCh} \\ \text{in TrCh } j}} \Big(\sum_{\substack{i\in PDCP \\ \text{in queue k}}} R_{est,i,j,k} + R_{ctrl,j,k} \Big) \quad (bps) \tag{7}$$

$$R_{est,i,j,k} = \begin{cases} \displaystyle\sum_{\substack{i\in PDCP \\ \text{in queue k}}} \frac{n\_rlc_{i,j,k} \times SDU\_rlc(bits)}{\tau_{rem,i,j,k}(\text{sec})}, & \tau_{rem,i,j,k} > 0 \\ \displaystyle\sum_{\substack{i\in PDCP \\ \text{in queue k}}} \frac{n\_rlc_{i,j,k} \times SDU\_rlc(bits)}{t_{TTI}(\text{sec})}, & \tau_{rem,i,j,k} = 0 \end{cases} \tag{8}$$

$$\tau_{rem,i,j,k} = \tau_{rem,i,j-1,k} - t_{TTI} \qquad \text{if } \tau_{rem,i,j,k} > 0 \tag{9}$$

$$\tau_{rem.i,0,k} = \frac{PDU\_pdcp_{i,k}}{R_{\text{target},k}} \tag{10}$$

where $n\_rlc_{i,j,k}$ is the number of remaining RLC packets in the queue associated to PDCP packet i for service (LCh) k in TTI j, $PDU\_pdcp_{i,k}$ the size of PDPC packet i, including padding to match the number of RLC segments, $\tau_{rem,i,j,k}$ the remaining time for RLC packets associated to PDCP packet i in TTI j and $R_{ctrl,j,k}$ the transmission rate associated to RLC control packets for service k.

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

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

This strategy is compared to the situation of only power constraints (MBR):

2) *Maximum Bit Rate (MBR)*: information is transmitted at highest available speed. Transmission rate is only reduced if there are not enough packets in the buffer.

3. System Model

In this work, the joint transmission of voice, video, interactive game and signalling for both links has been studied. To model voice traffic a classical two state ON-OFF Markov model has been used, with means T_{ON} (0.4 sec.) and T_{OFF} (0.6 sec.), respectively. A 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 [7] has been implemented. Similar to other models it defines a packet call arrival process and within each packet call a datagram arrival process (lognormally distributed with mean and standard deviation 160 ms.) The packet call duration and the reading time (the time between packet calls) are exponentially distributed with mean 5 sec. The reading time starts at the successful transmission of all datagrams generated during the previous packet call to emulate a closed loop transmission mode. Datagram size is set to 576 bytes. The resulting mean data rate during the active period is 28.75 kbps.

The system simulation model considers a service multiplexing example corresponding to four different services conveyed by five different TrChs for both links: The 12.2 kbps AMR speech service split into three TrChs [8], the 3.4 kbps signalling bearer, a video service and an interactive game. In order to analyse the different TFC selection strategies, the video and gaming services have been mapped onto a single TrCh 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. Main TrChs parameters are included in Table 1. QoS constraints and required E_b/N_o of these services in a fast fading vehicular environment are also included. Fig. 2 shows the multiplexing scheme for the uplink.

Table 1. Transport Format Parameters

						QoS Requirements		Eb/No	
			CRC	Channel Coding	TTI	BER	BLER	UPLINK	DOWNLINK
Signalling	TrCH0	1x148	16 bits	1/3 Convolutional	40 ms	-	1.0E-02	4.05	4.20
AMR Speech	TrCH1	1x81	12	1/3 Convolutional	20	5.0E-04	-	4.49	4.73
		1x39						5.33	5.57
	TrCH2	1x103	0	1/3 Convolutional	20	1.0E-03	-	3.36	3.60
	TrCH3	1x60		1/2 Convolutional		5.0E-03	-	3.31	3.23
Data	TrCH4	8x340	16	1/3 Turbocoding	20	-	1.0E-02	1.95	3.15
		4x340						2.25	3.20
		2x340						2.77	3.40
		1x340						3.45	3.80

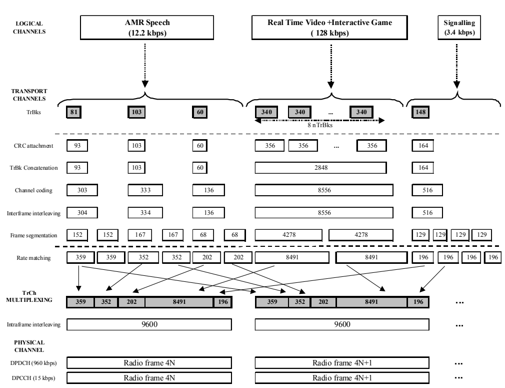

Figure 2. Multiplexing scheme in the uplink.

Different RLC modes are selected for both services: 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.

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 [9] 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 [10]) is considered. 11dB antenna gain and thermal noise power of –103 dBm are assumed [9]. The MS and BS have a maximum output power of 24 and 43 dBm according to [10].

4. Performance evaluation

4.1- Simulation conditions.

In order to assess the performance of the multiplexing scheme and the different proposed strategies, a system level simulator for the UTRA (UMTS Terrestrial Access Radio) FDD system, programmed in C++ [11] has been developed. The system level simulator allows to evaluate the performance of different RRM strategies, including several traffic sources, propagation conditions, mobility models and results from a physical layer simulator [12], [13]. The UMTS protocol stack (PDCP, RLC and MAC) and the multiplexing of Logical and Transport Channels are implemented. Off line results from the physical level simulator (bit error distribution according to E_b/N_o) are included through a Hidden Markov chain. This physical level 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 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 [14] 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.

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

In addition, it has been studied the effect of an estimation error in the transmitted power and the system capacity in the uplink, leading to a deviation of the real E_s/N_o from the target E_s/N_o, in order to obtain results for a more realistic situation. The system load, unlinke propagation channel, suffers discrete changes in a TTI by TTI basis instead of a continuous evolution. Power control slowly adapts to these variations and several slots are needed to reach again the new target E_s/N_o. To model this effect, the system capacity – C_{res} in (5) – is estimated as a function of the capacity in previous and current TTI. Secondly, the target E_s/N_o cannot be ideally estimated and this has been modelled adding a lognormally distributed error to the estimated transmission power.

$$P_{T\_UE}\Big|_{est} = P_{T\_UE}\Big|_{obj} + lognorm(desv) \qquad (11)$$

where P_{T_UE} is evaluated according to (5), and the C_{res} in this equation is calculated as follows:

$$C_{res} = (\alpha/100)\cdot C_{est}^{curr} + (1-(\alpha/100))\cdot C_{est}^{prev} \qquad (12)$$

The C_{res}^{prev} and C_{res}^{curr} values correspond to the previous and current TTI capacities respectively and α is the percentage of the current capacity considered in the estimation.

4.2- Results.

Fig. 3 shows the percentage of packet loss for voice transmission. The QoS requirement is covered in all the simulated load conditions for both links.

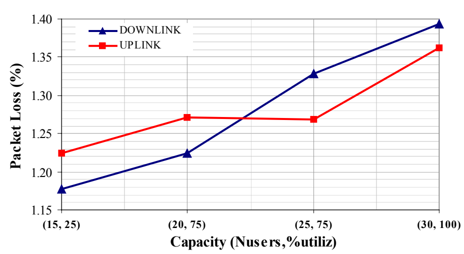

Figure 3. Packet loss for voice transmission

With the MBR strategy packets are transmitted faster than with TRDC, since the former uses the maximum available resources (transmission rate) without trying to keep a medium target bit rate. Therefore, dropping probability for video packets is higher with TRDC, as it is shown in Fig. 4. 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 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. Fig. 5 shows the percentage of game packets with a delay higher than 250 ms. It shows that the LWTFO dequeueing strategy provides an important improvement. 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. An example is shown in Fig. 6, 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 (Fig. 7). TRDC smoothes traffic, whereas in MBR, transmission rate mainly alternates between 0 and 128 kbps. From a network management perspective, although MBR obviously outperforms TRDC, this one would be preferred to avoid high rate fluctuations that would lead to errors in load estimation.

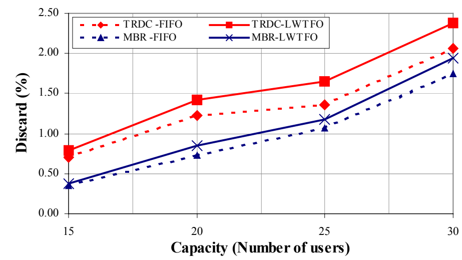

Figure 4. Dropped video packets (%) in the uplink

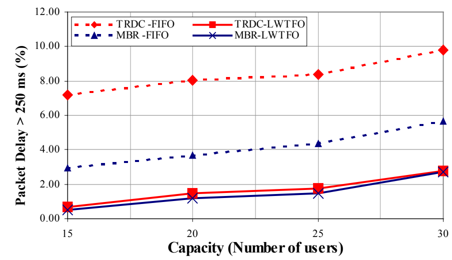

Figure 5. Game packets with delay higher than 250 ms (%) in the uplink

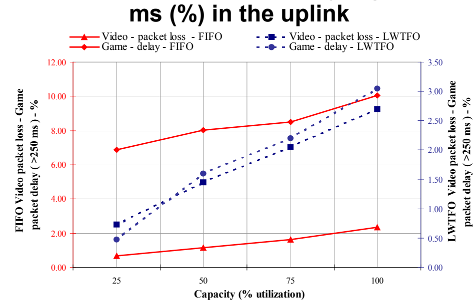

Figure 6. Video packet loss and game packets with delay higher than 250 ms (%). (TRDC, downlink).

Fig. 8 and Fig. 9 show the effect of estimation errors for TRDC–LWTFO in the uplink. In Fig. 8, C_{res} is calculated according to (12) with α = 50%. Discarding probability for video packets rapidly increases specially from a standard deviation of 0.5 dB, which would be the maximum tolerable value for matching the QoS requirements for all load conditions. Fig. 9 shows the percentage of game packets with a delay higher than 250 ms. for different grades of accuracy in the capacity estimation for 15 users on average. The percentage of current capacity used in equation (12), models how rapidly the power control adapts to the changes in the real system capacity. The faster is the adaptation, the higher is the contribution of the current value, which is expressed with a higher percentage. Sharp hops in capacity are due to heavily fluctuations in the system load. In this situation, a realistic approximation is to consider lower values of α, since it is more difficult to follow these variations. As it is shown in Fig. 9, this leads to a poor performance, even with a low power error estimation. TRDC strategy for TFC selection tries to smooth traffic in order to yield a more stable load in the system reducing, consequently, the deviations in the E_s/N_o due to the difficulties of power control in following the capacity variations, then leading to a better system performance.

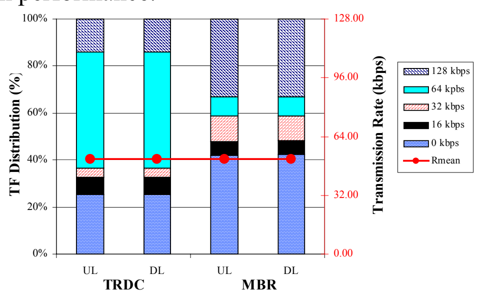

Figure 7. TF Distribution. Capacity (20, 50%)

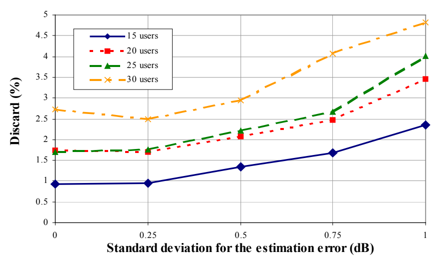

Figure 8. Discarded video packets in the uplink vs. estimation error. α =50%

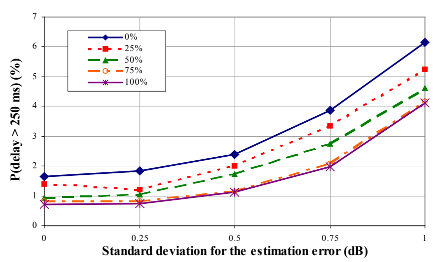

Figure 9. Game packets with delay higher than 250 ms (%). in the uplink vs. estimation error. 15 users

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 signalling has 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 LChs onto TrChs has been analysed by means of two different dequeueing strategies. The performance of the proposed scheme has been evaluated through a realistic UMTS simulator under different load conditions and in a high mobility situation. The effect of a non-ideal power control has been included. 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. These lower load fluctuations allow to tolerate higher power estimation errors. For LCh 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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