Performance Evaluation of 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 signalling is evaluated for the Dedicated Channel of the Universal Mobile Telecommunications System. Transmission rate for each service is determined according to its QoS requirements by means of an adaptive Transport Format selection based on buffer occupation, delay requirements and target bit rate keeping power constraints. Video and game services are multiplexed in upper layers (Logical Channels) sharing a common transmission rate (Transport Channel). Two strategies to share this rate are evaluated. A system level simulator, implemented in C++, which includes propagation conditions, traffic and mobility models and results from a physical layer simulator, allows to evaluate the performance of the proposed scheme.**

I. INTRODUCTION

The UMTS air interface supports service multiplexing on a single connection providing different QoS. Getting these required QoS for each user is closely connected with power allocation. The real available capacity is limited by the interference, since the acceptance of a new user connection is conditioned by the fact that target signal to interference ratio (E_b/N_o) values must be achieved by each existing connection once a new one is activated. Therefore, a good interference handling by radio resource allocation schemes plays an important role to guarantee the performance and to increase the system capacity.

In this paper, an analysis of service multiplexing in the dedicated channel (DTCH) is carried out in a realistic WCDMA UMTS environment which 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 signalling for both uplink and downlink has been studied. When a connection is accepted, certain resources are allocated for this user. According to these available resources and the different QoS requirements of the multiplexed services, transmission rates for each service must be determined. In order to select these rates, an strategy based on buffer occupation, delay requirements and target bit rate keeping power constraints is proposed and evaluated. 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 II, a description of the UMTS system functionalities, services and service capabilities is given and the proposed strategies are described. In Section III, the system model is presented and main parameters are described. We discuss performance results in Section IV and, finally, conclusions are provided in Section V.

II. QoS FOR THE UMTS AIR INTERFACE

A. 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 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 Transport Channels, selection of TF, priority handling and dynamic scheduling. 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. A LCh is defined by the type of transferred information.

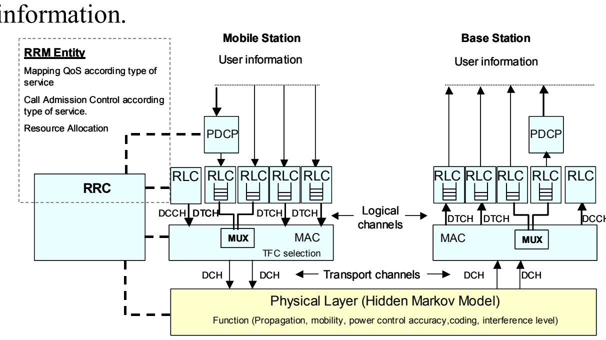

Fig. 1 UTRA-FDD Radio Interface protocol architecture

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) [2, 3]. 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 frame and their assignment to the individual channels (Transport Format Combination – TFC) [2].

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The Radio Link Control (RLC) protocol [4] 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 RRC: based on FEC are Transparent Mode (TM) and Unacknowledged Mode (UM) whereas the Acknowledged Mode (AM) is based on joint FEC and 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 signalled by upper layers. We have considered a packet-dependent value according to (1)

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

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.

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.

B. 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_o) must be provided for each Transport Channel. 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 [2] provides the required E_b/N_o for each of the multiplexed Transport Channels in the CCTrCH. Equations (2), (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,..., I (number of Transport Channels)

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

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 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 DTX (Discontinuous Transmission) is used in addition to repetition and puncturing in order to match the variable transmission rate of each Transport Channel.

Power transmission is determined by the required E_s/N_o, channel conditions and system interference level in different ways for both links. Equation (4) shows these dependences for the downlink:

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

$P_{T-BS,i}$ is the transmitted power by the BS associated to user i, P_{T-BS} the total transmitted power and $P_{MAX-BS,i}$ 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-down} the path loss between BS and user i. Equation (5) shows the same for the uplink:

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

where $P_{T-UE,i}$ is the transmitted power by user i, $P_{MAX-UE,i}$ is the maximum transmitted power, C_{res} is the residual capacity in the uplink, and h_{i-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 Base Station (BS), so this limits the maximum power allocated to one user, $P_{MAX-BS,i}$, 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 $P_{MAX-UE,i}$.

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 [5].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 TrCh. This rate should be decided as a function of buffer occupation, target bit rate and power and delay constraints.

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Two different strategies for TFC selection have been implemented:

1) *Target Rate with Delay Constraints (TRDC)*: information about the arrival times is taken into account. When 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 Transport Channel, 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 (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, $\tau_{rem,i,j,k}$ 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 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)

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.

III. 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 TON (0.4 sec.) and TOFF (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 Transport Channels for both links: The 12.2 kbps AMR speech service split into three Transport Channels [8], the 3.4 kbps signalling 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. Main Transport Channels parameters are included in Table I. 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 I
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

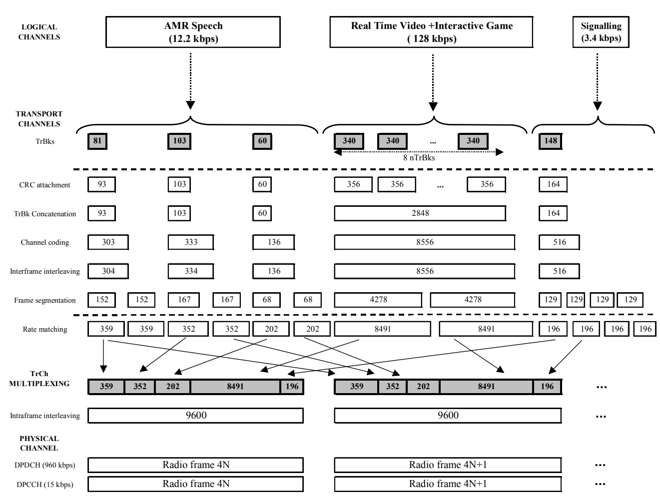

Fig. 2. Channel coding, mapping of Logical Channels onto Transport Channels and multiplexing scheme in the uplink.

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 –103dBm are assumed [9]. The MS and BS have a maximum output power of 24 and 43 dBm according to [10].

IV. Performance Evaluation

A. 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 III. 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.

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 II.B.

B. 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.

Fig. 4 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 higher than with MBR, since the latter uses the maximum available resources (transmission rate) without trying to keep a medium target bit rate. Packets are transmitted faster with MBR, so they are discarded with less probability, as it is shown in Fig. 5. 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.

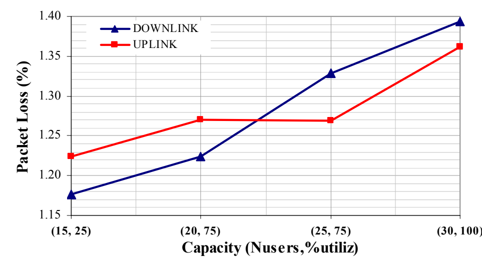

Fig. 3 Packet loss for voice transmission

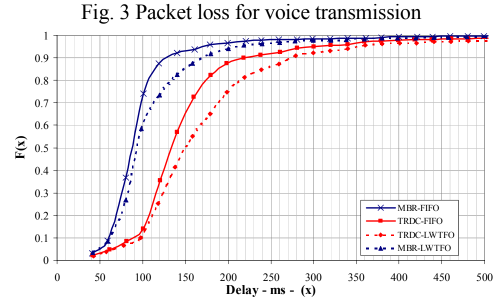

Fig. 4. Distribution function for video packet delay in the uplink (20 users).

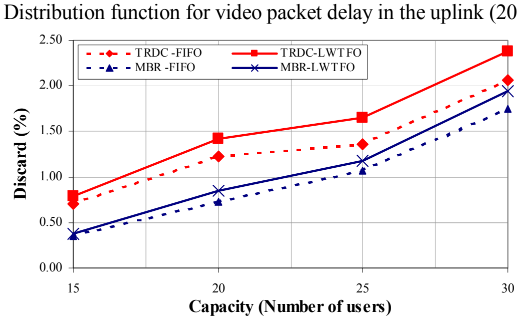

Fig. 5 Percentage of dropped video packets in the uplink

Fig. 6 shows the distribution function of game packet delay in the uplink with an utilization of 50%. MBR outperforms again TRDC. However, the LWTFO dequeueing strategy provides an important improvement, as it is shown in Fig. 7. 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.

An example is shown in Fig. 8, 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. 9). 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.

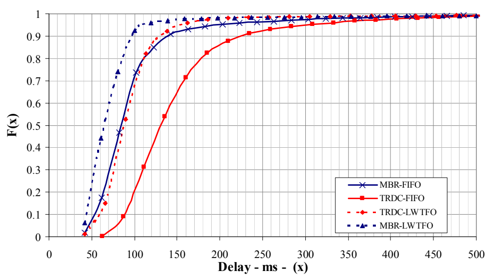

Fig. 6. Distribution function for game packet delay in the uplink (20 users).

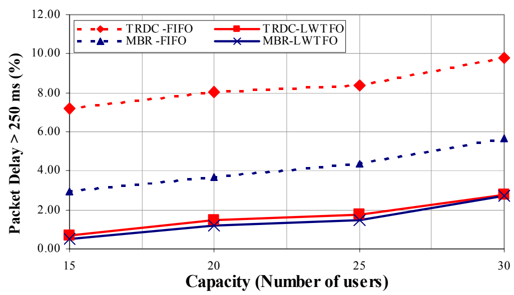

Fig. 7. Percentage of game packets with delay higher than 250 ms in the uplink

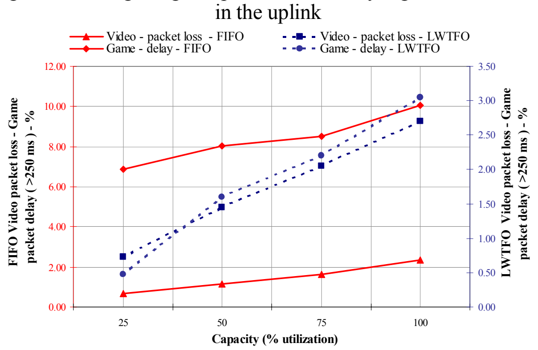

Fig. 8. Video packet loss and percentage of game packets with delay higher than 250 ms. (TRDC, downlink).

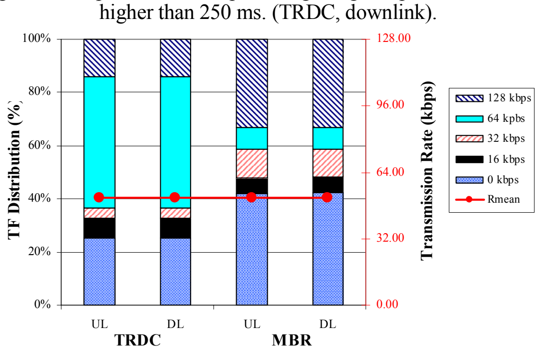

Fig. 9 TF Distribution. UL (20 users), DL (50 %)

V. 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 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) match 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.

Acknowledgment

This work was supported by projects from CICYT and FEDER, TIC2001-2481 and Telefónica Móviles de España.

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