

# Experimental analysis of TCP behaviour over a downlink UMTS channel under different scheduling strategies

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**Abstract.** Cellular systems, such as UMTS, use scheduling algorithms in order to provide fair sharing of the available bandwidth among the different users. However, the use of scheduling algorithms in wireless networks can have a great impact on the performance of TCP based applications. The objective of the paper is the identification of the effects observed in legacy TCP connections over a UMTS downlink shared channel (DSCH) under two different sets of scheduling strategies: rate and timeout oriented. To perform the analysis a real time UMTS emulator has been used.

**Keywords:** TCP, UMTS, scheduling, RRM.

## 1 Introduction

The Internet Transmission Control Protocol (TCP) is a protocol initially designed for data transfers over a fixed network. Due to the increasing deployment of wireless networks and the extensive use of the Internet during the last years, TCP has been widely used over wireless networks. In this sense, one research field is the performance of TCP in wireless cellular networks [1]. The research community has been studying TCP over different cellular networks, such as GSM [2], GPRS [3], IS-2000 [4], etc. A few studies have been carried out for the case of the 3G UMTS network. For example, the impact of different UMTS RLC layer parameters on the TCP protocol has been studied in [5][6], the effect of buffer management mechanisms is analysed for 3G networks in [7] and the impact on TCP throughput of dedicated and shared channels in the UMTS downlink transmission is studied in [8].

In this paper we study the impact of different scheduling strategies on TCP behaviour in a shared downlink UMTS channel. This type of link may be subject to large variations in the link rate (ranging from i.e. 256 kbits/s down to 16 kbits/s or even less) depending on the radio channel conditions and on the applied RRM (Radio Resource Management) strategies such as packet scheduling. This bandwidth oscillation could lead to significant throughput degradation due to factors such as spurious TCP retransmissions [4]. The paper describes the behaviour observed in legacy TCP connections over a real time UMTS testbed under two different sets of scheduling strategies: rate and timeout oriented. The testbed is capable

of emulating in real time the behaviour of a UMTS scenario where different advanced RRM algorithms can be implemented and tested [9].

The paper is organized as follows: Section 2 introduces the packet scheduling mechanisms analysed in our study. Next, in section 3, the architecture of the UMTS emulator used for evaluating the TCP performance is described. Section 4 presents the obtained results, and finally, conclusions and future work are addressed in Section 5.

## 2 Packet Scheduling in UMTS

UMTS networks are designed to fully support multimedia traffic under Quality of Service (QoS) guarantees. In the UMTS air interface, WCDMA technology jointly with efficient RRM functions have been adopted as the key enablers of such complex multimedia radio access network. RRM functions are crucial in WCDMA access networks since capacity is usually limited by the amount of interference in the air interface (soft capacity).

For data services, such as those based on TCP, scheduling algorithms within the RRM framework are important components in the provision of QoS parameters such as throughput, delay, delay jitter or packet loss rate. Focusing on the downlink channel, the task of the packet scheduler is to time/code-multiplex user transmissions in a way that individual user's QoS requirements are satisfied [10]. The algorithms addressed in this analysis are based on the scheduling strategy introduced in [11] for a Downlink Shared Channel (DSCH). The DSCH is a common channel intended to optimise code usage in UMTS downlink by sharing a subtree of OVVSF (Orthogonal Variable Spreading Factor) codes among several users [12]. This scheduling strategy assigns resources at frame basis (10ms) following a three-step procedure: capacity requirement, prioritization and availability check

### A. Capacity Requirement

In the capacity requirement phase, the scheduler makes an estimation of the amount of resources required for each served user  $i$ . In WCDMA, resources can be formulated in terms of transmission rate (or equivalently spreading factor  $SF_i$ ) and transmission power ( $P_i$ ) in order to meet certain QoS restrictions. Both magnitudes are coupled by means of the following expression:

$$\left( \frac{E_b}{N_0} \right)_i \leq \frac{P_i}{L^{i,k} \cdot SF_i} \cdot \frac{SF_i}{I + P_N} \quad (1)$$

where  $I$  accounts for the intracell and intercell interference,  $L^{i,k}$  is the propagation loss between mobile  $i$  and base station  $k$  ( $BS_k$ ),  $P_N$  is the noise power and  $E_b/N_0$  is the minimum bit energy over noise ratio that provides the required BLER (Block Error Ratio) target. In our study, two different policies are used to choose the preferred value of SF (and consequently, using (1), the required power  $P_i$ ):

- **Delay-oriented strategy:** The SF is selected to allow a pending transmission to be served within a given delay bound. So, packets reaching its expiration timeout (TO) will be transmitted with lower SF (higher transmission rate).
- **Rate-oriented strategy:** The SF is selected to guarantee a certain mean bit rate by means of the “service credit” (SCr) concept. The SCr of a connection accounts for the difference between the obtained and the expected bit rate for this connection. So, pending transmissions within connections with accumulated credits should be allowed to use low SFs.

### B. Prioritization

Once the required capacity for each user is known, users are prioritized according to its type of service (first prioritization level) and, for the same type of service, according to their QoS requirements (second prioritization level). In our analysis, for the second prioritization level, two different strategies have been considered:

- **Priority as a function of timeout  $TO_i$ .** Users with packets reaching their timeout are served first.
- **Priority as a function of the Service Credits.** Users with accumulated credits are served first. So, resource sharing among users will tend to guarantee a mean bit rate for each connection.

### C. Availability check.

After the prioritization of the users to be served, the algorithm checks whether or not this selection is possible depending on the available resources and modifies it accordingly. In the implemented algorithm a specific user transmission is carried out only if there are enough codes (Kraft’s inequality [13] to check the usage of the OVSF tree devoted to DSCH) and the estimated transmitted power level in the serving BS ( $P_T^k$ ) is below the maximum transmission power. This last condition leads to the following expression for  $n$  users served by  $BS_k$ :

$$P_T^k = \frac{P_N}{(1 - \eta_k)} \sum_{i=1}^n \frac{L_p^{i,k}}{SF_i} + (1 - \alpha) \left( \frac{E_b}{N_0} \right)_i \leq P_T^{MAX} \quad (2)$$

where  $\eta_k$  is the expected load factor and  $\alpha$  is the orthogonality factor [12]. In case either code or power

conditions are not satisfied, the SF is doubled, or equivalently, the transmission bit rate is reduced by half, and the conditions are checked again. If this is not possible, the request should wait for the next frame.

## 3 Description of the emulation platform

### 3.1 UMTS emulator

TCP performance results have been obtained using a real time UMTS emulator developed within the framework of the IST ARROWS project [14]. The UMTS emulator is a real-time operation HW/SW platform that includes multimedia terminals, UMTS elements and IP connectivity. Among the main features of the platform we can remark the possibility of validating RRM strategies under complex scenarios and the possibility of testing the impact of these strategies over the end-to-end behaviour of legacy IP-based multimedia applications with Quality of Service (QoS) requirements. The external organisation of the UMTS testbed is composed by the User Equipment (UE), the UMTS Terrestrial Radio Access Network (UTRAN), and the UMTS Core Network (CN). Figure 1 shows the basic mapping of UMTS components into testbed machines.

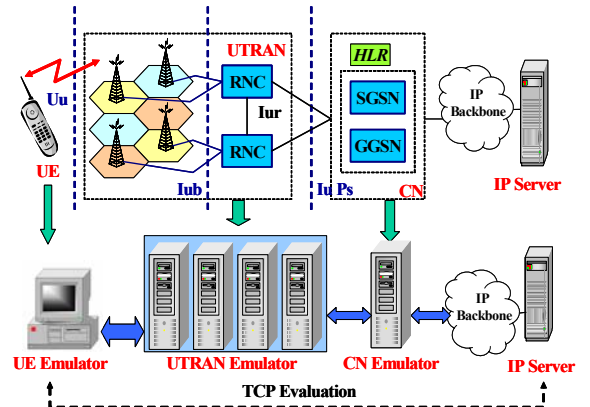


Figure 1. UMTS real-time emulator

Different applications such as WEB browsing are executed in the UE and IP Server for a reference user under test. The rest of users are emulated by means of traffic models inside the UTRAN Emulator. A complete protocol stack has been developed for the reference user in accordance to 3GPP specifications [15]. Physical layer emulation has been addressed by means of histograms obtained from off-line simulations [14].

### 3.2 Settings of the analysed scenario

The UMTS emulator is configured with a 5kmx5km macrocell scenario where cells are tri-sectorized and hexagonally distributed with a cell radius of 500m. The system is loaded with a combination of conversational and interactive users. Table 1 provides Transport formats Combinations (TFC) used in the correspondent Radio Access Bearers (RAB). Conversational service is offered through dedicated channels (DCH) with two possible Transport Formats (TF): 64kbits/s and no-transmission. For interactive users, transmissions are carried out through

a DSCH channel and a total of 7 TFs are considered ranging from data rates from 256kbits/s down to the no-transmission case. One fourth of the OSVF code tree has been assigned to the DSCH channel in each BS (This means that, according to allowed TFs in table 1, the maximum number of simultaneous users in DSCH could take any combination between 16 users at 16kbits/s and 2 users at 256 kbits/s).

**Table 1. Transport formats for the considered RABs.**

Service		WWW	VIDEOPH ONE
Channel type		DSCH	DCH
Transport Block (TB) sizes		336 bits (320 payload)	640 bits
TFC	TF0, bits	0x336	0x640
	TF1, bits	1x336 (16 Kb/s, SF=128)	2x640 (64 kb/s, SF=32)
	TF2, bits	2x336 (32 Kb/s, SF=64)	-
	TF3, bits	4x336 (64 Kb/s, SF=32)	-
	TF4, bits	8x336 (128 Kb/s, SF=16)	-
	TF5, bits	12x336 (192 Kb/s, SF=8)	-
	TF6, bits	16x336 (256 Kb/s, SF=8)	-
TTI, ms		20	20

Conversational users are assumed to use videoconferencing services and are modelled by means of constant bit rate sources at 64 kbps with average call duration of 120s. For interactive users, the model provided in [16] has been used. The interactive traffic models parameters have been adjusted to provide a mean generation bit rate of 24 kbts/s. The inter-session times are assumed exponentially distributed with 100s mean.

Maximum transmission power of base stations has been fixed to 43 dBm from which pilot channel (CPICH) consumes 10 dBm. Neither admission control nor congestion control is considered in the system and handover decisions are taken each 100ms assuring the mobile is always connected to the best cell with a replacement hysteresis of 1dB. Values related to the radio QoS parameters are given in Table 2.

**Table 2 Lower layer parameters.**

QoS parameters	CONV	WWW.
BLER target	1%	1%
Max ACKs (RLC layer)	0	4
Max SDU Size (bytes)	160	570

Under such conditions, conversational users are configured to generate 24 Erlangs per cell. Considering only

conversational users, the mean load factor of the central BS is as high as 0.80, the mean transmission power is around 34 dBm and the BLER target of 1% is guaranteed for all the conversational users. Upon such a scenario, 1800 interactive users are randomly distributed in the system (accounting for a mean load of 160 kbts/s per DSCH). This number of users has been fixed after several experiments since effects on TCP are already observable.

In this paper we have studied the behaviour of TCP when executing a browsing application in the UE against a web server running in the IP Server machine. Both the server and the client run Linux version 2.4-16 with the following TCP options enabled: SACK, FACK and Timestamps. The TCP options DSACK and Window Scaling are disabled. The maximum receiver window is 32 KB and the Maximum Segment Size (MSS) is 530 bytes. Buffer size in intermediate nodes is fixed high enough to avoid packet dropping due to insufficient buffer space (including RLC buffer in UTRAN). The *tcptrace* tool was used to analyse the tcp traces captured. The performance of TCP over the UMTS radio interface depends on a large number of parameters and, therefore, the results obtained in this paper are valid for the UMTS settings considered.

## 4 Results

### 4.1 System level results

Results are given for two of the aforementioned packet scheduling strategies: Service Credit (Both capacity requirement and prioritization are based on SCr) and Timeout (Both capacity requirement and prioritization are based on Timeout). The notation SCr64 stands for a guaranteed 64kbts/s while the notation TO300 stands for 300ms applied to the largest SDU packet (570 bytes).

System level results are provided in Tables 3 and 4. As it can be observed, different strategies have a noticeable impact on each individual user (mean delay and jitter) while the overall system figures (transmission power, load factor and effect on conversational BLER) are kept almost invariable. However, it is worth to remark the degradation suffered by the conversational users in terms of BLER when interactive users have been added.

In the case of SCr strategies, low delays are obtained with high credit rates without modifying global system behaviour although, in this case, the number of requests being postponed in the DSCH is higher for high rates since the system tends to “time” schedule by assigning low spreading factors, that is few users with high rates are served each time transmission interval. In the case of TO strategies, a similar behaviour is observed either at user or at system level. Reduced TO leads to better user satisfaction without deteriorating global system performance. This behaviour can also be explained since low TOs also result in a “time scheduling” approach. Furthermore, it is worth noting that TO strategies result in lower jitter delay than SCr policies due to the prioritization and capacity assignment methods.

**Table 3. Results for service credit (SCr) strategies**

Strategy	User Level (SDU)			System Level				
	Mean Delay (ms)	Jitter Delay (ms)	SDU Error %	Mean SF in DSCH	DSCH postponed transmissions	Load Factor	Mean Power (dBm)	BLER (Conv.)
SCr16	2140.0	2148.0	0.81	111.6	7.46%	91.57%	37.38	2.24%
SCr32	531.00	960.00	1.38	41.9	20.78%	93.37%	39.13	2.77%
SCr64	218.00	587.00	1.33	18.4	30.89%	93.24%	39.06	2.98%
SCr256	123.29	507.51	0.90	10.3	43.10%	92.16%	38.57	1.95%

**Table 4. Results for Timeout (TO) strategies**

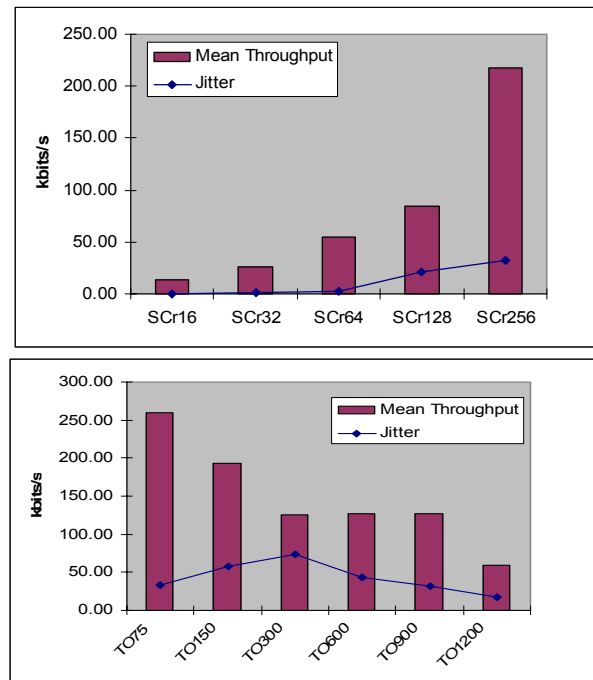
Strategy	User Level (SDU)			System Level				
	Mean Delay (ms)	Jitter Delay (ms)	SDU Error %	Mean SF in DSCH	DSCH postponed transmissions	Load Factor	Mean Power (dBm)	BLER (Conv)
TO75	140.29	422.78	0.94	19.0	26.55%	92.24%	38.37	1.98%
TO150	185.03	545.15	1.19	37.1	25.90%	92.85%	39.01	2.92%
TO300	224.61	663.90	1.18	62.3	19.19%	92.96%	38.88	2.38%
TO600	234.84	512.37	0.92	86.0	12.22%	92.61%	38.53	2.51%
TO900	235.31	389.80	0.64	110.5	7.98%	91.40%	37.71	1.88%

**4.2 Effects over TCP**

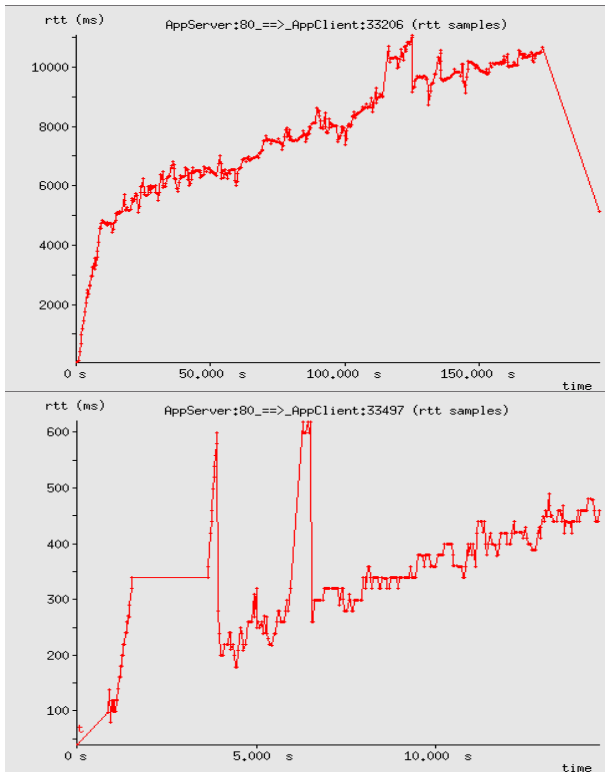
Figure 2 shows the mean throughput and standard deviation values obtained for a set of trials consisting of downloading a given file. Focusing on SCr strategies, as it was already envisaged in system level results, TCP throughput increases with the available bandwidth provided by the SCr scheduling mechanism without observing any remarkable effect that deteriorates TCP operation. One reason that explains these throughput results is that the UMTS system, even if it uses a scheduling algorithm that tries to offer every user 256 kbits/s, does not create an effect that degrades seriously the TCP performance. On the other hand, the SACK and FACK algorithms perfectly cope with lost packets in the wireless link by avoiding unnecessary retransmissions. Higher service credits result in a reduction of the RTT values, as can be seen in Figure 3. For a transmission with SCR 256 the mean RTT values are less than 600 ms whereas in a transmission with SCR16 the RTT values are in general higher than 4000 ms.

TCP throughput values obtained with the TO scheduling mechanisms are provided in Figure 2. The TCP throughput decreases with the waiting time of TCP segments. Best results are obtained for TO75 as envisaged in system level results. However, throughput values for TO values of 300, 600 and 900 are quite similar due to the fact that, although mean delay increases with higher TO, jitter delay decreases causing the opposite effect in the resulting RTT. For the transmissions of TO600 and TO900, the waiting times are larger than for a transmission with TO300, but the TCP sender keeps sending packets at a constant rate. Figure 4 shows the sequence number evolution for a transmission with TO900. In this case, there are time intervals in which

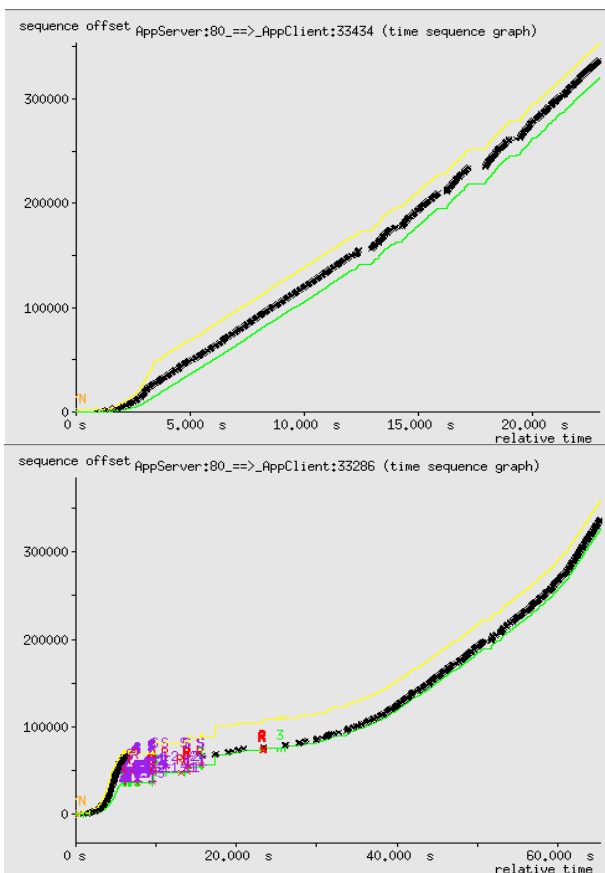
the TCP sender does not transmit any segment because it still has not received the correspondent ACK, but that doesn't trigger the retransmission timers. On the contrary, for the case of TO1200, the waiting times of TCP segments at the UMTS radio link layer cause large delays and, therefore, timer-driven retransmissions at the TCP sender, as can be seen in Figure 4. This behaviour explains the low TCP throughput obtained with TO1200.



**Figure 2.** Mean throughput and standard deviation values obtained for the real TCP connection of the reference user under SCr (top) and TO (bottom) strategies.



**Figure 3.** RTT evolution under SCr16 (top) and SCr256 (bottom) strategies



**Figure 4.** Sequence number evolution for TO900 (top) and TO1200 (bottom) strategies

## 5 Conclusion & Future work

The effects of two different sets of packet scheduling strategies over legacy TCP connections have been analysed in a real time UMTS testbed. In the case of SCr scheduling policies, TCP throughput is proportional to the service credit offered by the scheduling algorithm and, even under heavy load conditions, the TCP operation does not suffer any remarkable degradation.

For the case of transmissions with TO scheduling algorithms, low TO values give the best results because TCP segments must be served immediately and, therefore, transmitted with a high data rate, leading in some way to the same situation of having a SCr policy with a high transmission rate. The TCP throughput decreases with the TO value and it can be seriously degraded if the TO value is high enough to trigger timer-driven retransmissions at the TCP sender.

Future work will consider different UMTS scenarios as well as additional scheduling algorithms.

## ACKNOWLEDGMENTS

This work has been realised over a testbed developed under the ARROWS project under the IST framework (IST 2000-25133) and it has been supported in part by the Spanish Research Council under grant TIC 2001-2222.

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