UMTS Radio Resource Management by Transport Format Assignment and Selection

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Abstract

The goal of Universal Mobile Telecommunications System (UMTS) is the delivery of multimedia services to the mobile user. Each different service requires its specific Quality of Service (QoS) to satisfy the mobile user. The QoS requirements will be supported by several protocol layers. In this paper, the interaction between Radio Resource Management (RRM), Medium Access Control (MAC) scheduling and Transport Format (TF) selection is presented. To study the interaction and dependencies, a typical mobile user application mix is examined for performance analysis by simulations. A UMTS Radio Interface Simulator (URIS) is used based on an emulation of the standardized UMTS protocol stack extended by the Transmission Control Protocol (TCP) and the Internet Protocol (IP). Simulation results of QoS parameters like delay and throughput depict the performance of mobile applications over UMTS.

Keywords

UMTS Radio Interface Protocols, MAC Scheduler, QoS, TCP/IP, UMTS Radio Interface Simulator.

INTRODUCTION

The delivery of multimedia services to the mobile user is one of the goals of 3rd generation mobile communication systems. The use of several different services at the same time raises the demands for mechanisms to guarantee *Quality of Service* (QoS) for each application. To satisfy the mobile user, UMTS provides several *Radio Resource Management* (RRM) strategies. One mechanism is the provision of *Transport Format Combination Sets* (TFCS) combined with the scheduling of parallel data flows and the *Transport Format* (TF) selection in the *Medium Access Control* (MAC) layer.
This paper will show the interaction between TFCS assignment of the RRC, the TF selection and the scheduling strategy of the MAC. Therefore a typical application mix of a voice session combined with interactive, asymmetrical and bursty WWW browsing or FTP session is examined. UMTS provides the *Radio Link Control* (RLC) protocol to support quality of service at the radio interface. The RLC itself realizes a reliable link with *Automatic Repeat Request* (ARQ) mechanisms. Additionally, WWW browsing and FTP sessions are supported by TCP on an end-to-end error recovery basis. Performances results for a high traffic load scenario, that depicts the dependencies of assigned TFCS and MAC scheduling strategies, are presented.

MAC SCHEDULING CONCEPT

UMTS supports the parallel handling of multiple data streams arising from distinct applications. Applications belong to different service classes (conversational, streaming, interactive, background) which all require different QoS demands in terms of bit error rate, delay, jitter, throughput, etc.. To support these demands efficiently, the different layers are assigned specific transmission parameters by the RRM. For the *Data Link Control* (DLC) layer these are in particular:

- *Radio Link Control* (RLC) transmission mode,
- Mapping and multiplexing options of logical channels to transport channels,
- *MAC Logical Channel Priorities* (MLP) assigned to every logical channel,
- *Transport Format (Combination) Sets* (TFCS).

In our simulation environment the full functionality of the MAC layer is emulated in conformance to [1]. However the service scheduling is not specified in the MAC standard. Therefore we propose a MAC scheduler which uses MLPs to provide a priority scheduling between services of different QoS classes. The *Transport Format Combination* (TFC) selection as part of the MAC scheduling will guarantee the strong delay requirements of applications of the conversational and streaming class. The selected TFC has to guarantee the required data rate of the conversational services. The residual capacity can be shared by services of other QoS classes. Scheduling strategies for services belonging to the same QoS class are proposed in [6]. The scheduling and TFC selection can be changed every *Transmission Time Interval* (TTI). Fig. 1 depicts incoming and outgoing parameters of the MAC scheduler.

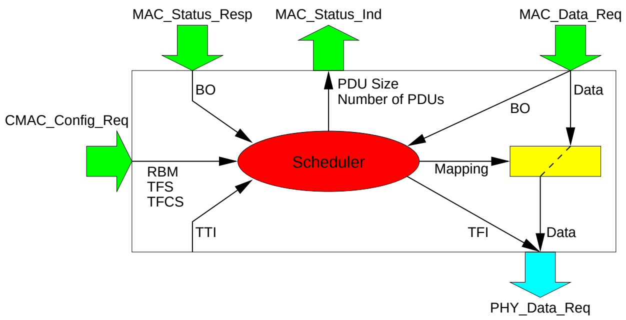


Figure 1. MAC Scheduler

Table 1. Model Parameters of Voice, HTTP Browsing, FTP Sessions

Voice Parameter	Distribution	Mean	
Session Duration	∞	—	
Data Inter Arrival Time [ms]	constant	20	
Mean Time Speaking [s]	negative exponential	20	
Mean Time Silence [s]	negative exponential	20	
HTTP Parameter	**Distribution**	**Mean**	**Variance**
Session Arrival Rate $[h^{-1}]$	negative exponential	30	—
Pages per Session	geometric	5	—
Reading Time between Pages [s]	negative exponential	20	—
Objects per Page	geometric	2.5	—
Inter Arrival Time between Objects [s]	negative exponential	0.5	—
Page Request Size [byte]	normal	1136	80
Object Size [byte]	$\log_2$-Erlang-k	$\log_2 2521 \approx 11.3$	$(\log_2 5)^2 = 5.4$
FTP Parameter	**Distribution**	**Mean**	**Variance**
Session Arrival Rate $[h^{-1}]$	negative exponential	30	—
Session Size [bytes]	$\log_2$-normal	$\log_2 32768 \approx 15$	$(\log_2 16)^2 \approx 16$
Object Size [bytes]	$\log_2$-normal	$\log_2 3000 \approx 11.55$	$(\log_2 16)^2 \approx 16$
Time between Objects [s]	$\log_{10}$-normal	$\log_{10} 4 \approx 0.6$	$\log_{10} 2.55 \approx 0.4$

SIMULATION ENVIRONMENT

For this study, the *UMTS Radio Interface Simulator* (URIS) was developed at ComNets. The simulator is a pure software solution. The software architecture is shown in Fig. 2. Usual simulator approaches model protocols and functions on basis of abstractions and simplifications. The aim of URIS is a detailed, bit accurate implementation of the standardized protocols.

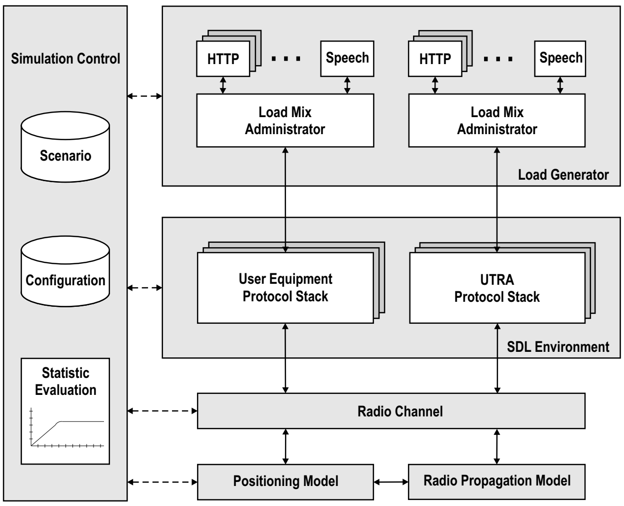


Figure 2. Structure of the URIS Simulator

The complex protocols like MAC, RLC, *Packet Data Convergence Protocol* (PDCP) and *Radio Resource Control* (RRC) are implemented completely bit accurate in conformance to their specifications. Hence URIS uses a protocol emulation for performance evaluation. The RLC protocol provides three different data transfer services: *Transparent Mode* (TR), *Unacknowledged Mode* (UM) and *Acknowledged Mode* (AM). The AM transmits higher layer data packets with guaranteed delivery to the peer entity. Therefore, it is mainly used for traffic of interactive class or background class type and is well suited for Internet applications. *Automatic Repeat Request* (ARQ) mechanisms are applied for correction of transmission errors. A Selective-Repeat ARQ has been implemented for AM. Segmentation/reassembling, concatenation and ARQ are fully implemented as specified in [2].

The modern Internet is mainly based on TCP and IP. The TCP implementation realized in URIS is based on the so called “Reno” TCP stack and uses the following flow control mechanisms: slow start/congestion avoidance, fast retransmit/fast recovery, delayed acknowledgments and selective acknowledgments [4].

This protocol emulation offers the opportunity to determine the performance of UMTS in a realistic manner. The performance evaluation considers the traffic load generation for voice, WWW browsing (HTTP) and FTP. The parameters of the used traffic models are shown in Tab. 1.

SIMULATION SCENARIO

When introducing UMTS to the public, many operators announced, that they will provide a maximum data rate of 64 kbps per mobile user. This is comparable to a fixed network access like ISDN. In consequence we examine the performance of a user who is speaking during an active WWW browsing session or FTP session. The simulation scenario covers one voice session running on a *Dedicated Channel* (DCH1) and one HTTP or FTP session running simultaneously on DCH2. Therefore a *Dedicated Physical Channel* (DPCH) with spreading factor 64 and a brutto bit rate of 240 kbps has to be allocated. These reference channels and their TFs for conversational/speech and interactive/background are proposed in [3]. Tab. 2 depicts the transport channel parameters as well as the RLC transmission mode and the TCP parameters used in the simulation.

Table 2. Simulation Parameters

Traffic Generator	Voice	HTTP	FTP
Source Data Rate [kbit/s]	12.2	Table 1	Table 1
TTI Length [s]	0.02	0.02	0.02
Transport Format Set [bit]	0x244	0x336	0x336
	1x244	1x336	1x336
		2x336	2x336
		4x336	4x336
Max. MAC Data Rate [kbit/s]	12.2	67.2	67.2
MLP	high	low	low
RLC Mode	TR	AM	AM
Max. TCP Segment [byte]	-	512	512
Max. TCP Window [kbyte]	-	16	16
Min./Max. TCP RTO [s]	-	3 / 64	3 / 64

The overall capacity assigned to a mobile user should not exceed 64 kbps information data rate. Therefore an appropriate TFC must be selected out of the TFCS every TTI. The TFCS assigned by the RRM with all possible TFCs for the reference channels is shown in Tab. 3.

TFC	DCH 1	DCH 2
0	TF0: 0 kbps	TF0: 0*336 bits = 0 kbps
1	TF0: 0 kbps	TF1: 1*336 bits = 16 kbps
2	TF0: 0 kbps	TF2: 2*336 bits = 32 kbps
3	TF0: 0 kbps	TF3: 3*336 bits = 48 kbps
4	TF0: 0 kbps	TF4: 4*336 bits = 64 kbps
5	TF1: 12.2 kbps	TF0: 0*336 bits = 0 kbps
6	TF1: 12.2 kbps	TF1: 1*336 bits = 16 kbps
7	TF1: 12.2 kbps	TF2: 2*336 bits = 32 kbps
8	TF1: 12.2 kbps	TF3: 3*336 bits = 48 kbps

Table 3. Transport Format Combination Set

The voice application has a higher priority than the WWW or FTP session. In consequence WWW or FTP will get 64 kbps if the speaker is silent. During speaking times, the maximum capacity for WWW or FTP is reduced to 48 kbps. To study the interaction between TCP and the MAC TFC selection a high load scenario is considered. To study the influence of the interaction only, no transmission errors are taken into account. Simulation results showing the influence of transmission errors can be found in [8, 7, 9].

The main parameters concerning QoS of an application are delay and throughput. During the simulations two measurements were performed:

- TCP Packet Delay: Time from sending a TCP packet (segment) until the correct reception by the receiver,
- User Data Packet Throughput: Size of the user data packet divided by the user data delay.

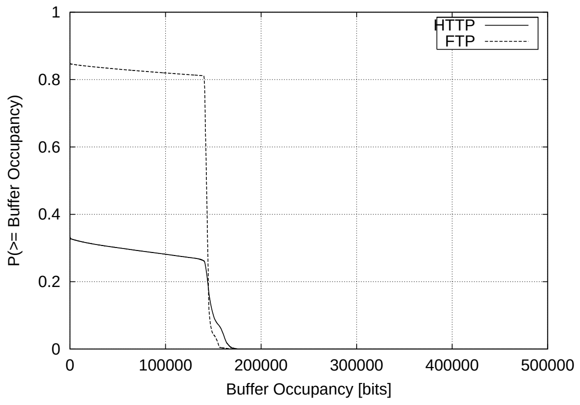
The complementary cumulative distribution function of the measured TCP packet delay and the complementary cumulative distribution function of the measured user data packet throughput have been calculated. Additionally the complementary cumulative distribution of the RLC buffer occupancy for HTTP and FTP was investigated. The simulation results will show the performance in terms of quality of service parameters a mobile user will experience while surfing the Internet and having a speech conversation in parallel.

SIMULATION RESULTS

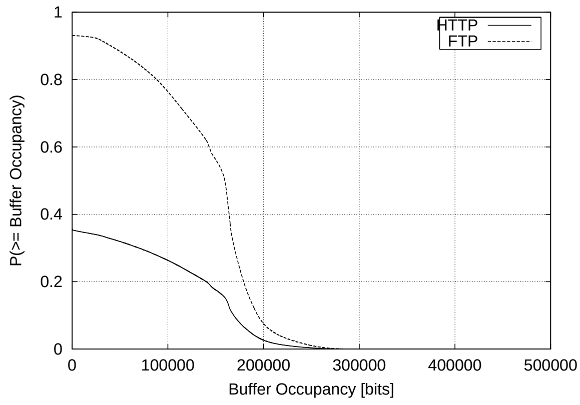
The buffer occupancies in Fig. 3(a) and 3(b) illustrate the processing of the scheduling strategy. Fig. 3(a) shows the simulation results when one single application is running on the protocol stack only, e.g. no voice application runs simultaneously. The considered application has always full access to the 64 kbps transport channel. In this case, the rectangular curve shapes indicate that the buffers are dequeued very fast. The buffer is filled in 35% of the time for HTTP and in 80% of the time for FTP. The maximum buffer size of around about 140 kbit is caused by the closed TCP transmission window size. In that scenario, TCP is able to adapt its congestion window according to the current network conditions. The amount of injected traffic is perfectly adjusted to the TCP packet transmission rate, e.g. the channel capacity. If data has to be transmitted, the buffer is filled. If no data is requested, the buffer remains empty. The RLC buffer occupancy will be only higher than 140 kbit in 10% of the time. This is caused by RLC and TCP acknowledgments for the uplink direction.

Fig. 3(b) shows the buffer occupancy when voice and HTTP or FTP are running in parallel. Since voice has a higher priority, HTTP or FTP can use the remaining 48 kbps while the user is speaking. In result, the buffer occupancy will rise above 140 kbit and the buffer is filled more often. The dequeuing will be much slower. In that case, TCP injects more traffic than available channel capacity. This effect indicates that TCP is not able to follow the capacity switching between 48 kbps and 64 kbps. Since the speaker is active, TCPs congestion avoidance algorithm is not fast enough to adjust the congestion window and the injected traffic immediately.

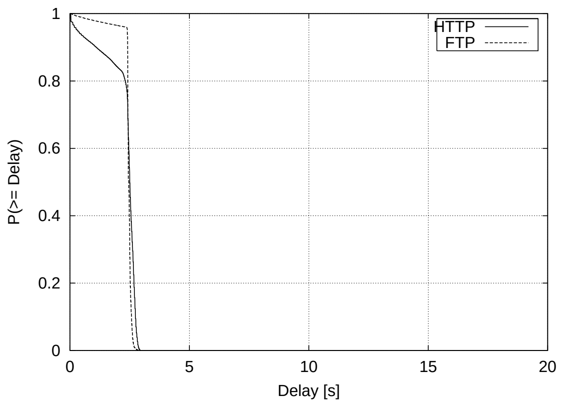
TCPs adaption process can be studied as well when looking at the TCP packet delay. The TCP packet delay in Fig. 3(c) is the delay of single TCP segments. The simulation results show that 99.4% of the TCP segments are delayed less than 2.9 seconds. In consequence no TCP retransmissions will occur. That means, whenever a TCP packet is transmitted, a new TCP packet is injected. The TCP packet arrival rate



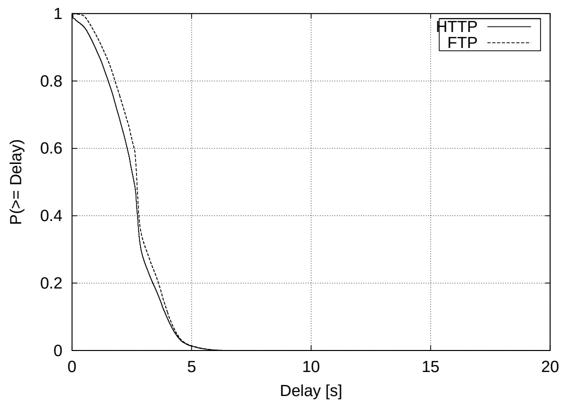
(a) Buffer Occupancy of HTTP or FTP



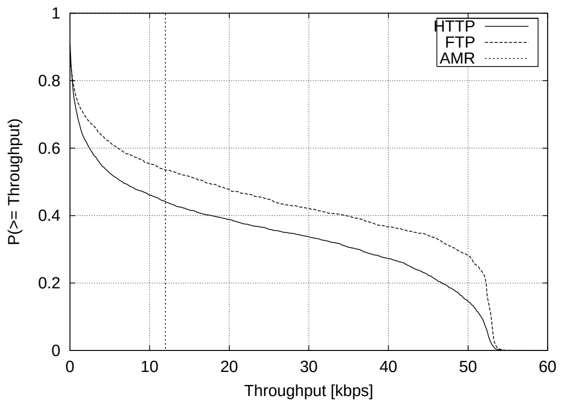
(b) Buffer Occupancy of HTTP/AMR or FTP/AMR



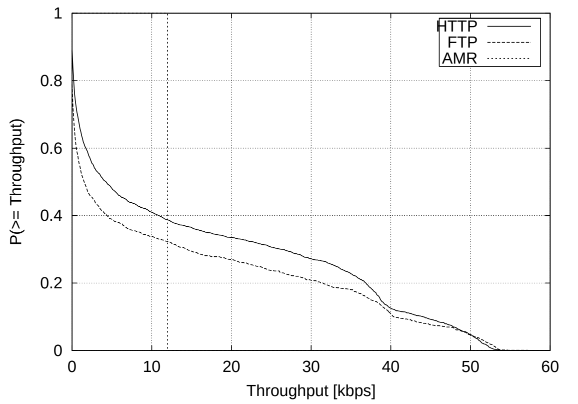
(c) TCP Delay of HTTP or FTP



(d) TCP Delay of HTTP/AMR or FTP/AMR



(e) User Data Throughput of HTTP or FTP



(f) User Data Throughput of HTTP/AMR or FTP/AMR

Figure 3. Simulation Results

matches the TCP packet transmission rate. The congestion window is perfectly adapted to the channel capacity. Since the capacity for DCH2 is restricted due to the voice application, the TCP delay rises up to seven seconds (Fig. 3(d)). For delays above three seconds TCP will trigger retransmission timeouts. Due to the retransmission timeouts, TCP detects a congestion. In consequence, TCP reduces its transmission window, adjusts the retransmission timer and retransmits all packets who are affected by retransmission timeouts. This adaptation process needs some time. The selection of a different TFC in the MAC triggers such an adaptation process. Nevertheless TCP is not able to follow the TFC switching immediately.

The fast TFC switching at the worst every TTI of 20 ms has deep impact on the user data throughput. The user data throughput is the throughput of single WWW page objects respectively a FTP download (Fig. 3(e)). The maximum throughput reaches 57 kbps. Due to overhead of RLC and MAC header information, 64 kbps can not be reached. The throughput of FTP is higher due to the nature of FTPs traffic characteristic. If a FTP download is triggered, a whole data file is transmitted consecutively. The throughput of HTTP objects is lower since the download of a page is characterized by objects of varying sizes and the reading time between WWW pages.

In case a voice application is active, a remarkable indentation is shown at 40 kbps which is caused by switching between TFC 4 and TFC 8 (Tab. 3). It is quite obvious that this decline is provoked by the reduced channel capacity from 64 kbps to 48 kbps. In contrast, it is not easy to understand why the throughput altogether is lower compared to Fig. 3(e). Especially the FTP throughput suffers from the fast TFC switching. Caused by TCPs slow load control mechanism, retransmissions of TCP packets occur. Due to these retransmissions, the overall throughput is reduced since the ordinary user data packets have to wait until the retransmissions are delivered correctly. The FTP traffic is much more affected than the HTTP browsing sessions. Since HTTP traffic has objects of varying size and inter arrival times, TCP gets the chance to adapt its load control mechanisms meanwhile. Nevertheless it must be pointed out that TCPs load control mechanisms cannot cope with the fast and efficient selection of the appropriate TFC by the MAC. The fast TFC selection process realizes an optimum channel capacity usage by statistical multiplexing with a very fine granularity. In contrast, TCP wastes the scarce radio resource due to unnecessary retransmissions and inadequate load control mechanisms. This effect will even increase if the error characteristic of the radio channel will be considered.

CONCLUSIONS

This paper shows the dependencies of TFCS assignment, MAC scheduling and TF selection for a typical traffic mixture at the UMTS radio interface. Simulations have shown that TCP with its flow control mechanisms has major impact on the performance. In result the RRM should be aware of the TCP behavior. A proper and adequate TFCS assignment will be of special concern since extensive switching between TFCs will not automatically grant an efficient user data throughput.

Finally it has to be discussed how TCP could be adapted to the special conditions of the UMTS radio interface. Many solutions are already suggested for the usage of TCP in wireless environments. It has to be proved if these approaches are sufficient for the UMTS radio interface.

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