Performance Evaluation of UMTS for Mobile Internet Access

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Introduction

UMTS constitutes the third generation of cellular wireless networks which aims to provide high-speed data access along with real time voice calls (Holma & Toskala, 2001). Wireless data is one of the major boosters of wireless communications and one of the main challenges of next generation standards (Chaudhury & Mohr, 1999).

It can provide maximum data-rates ranging from 64kb/s to 2Mb/s in different environmental types. UMTS is designed to provide access to the existing Internet services as well as to UMTS specific services. It will augment the existing capabilities of 2G mobile networks and GPRS, and one often envisaged strategy is to offer a richer set of multimedia services.

The most popular and widely used Internet applications are FTP, HTTP, email, etc. These Internet applications rely on two common protocols, namely, Transmission Control Protocol and the Internet Protocol (TCP/IP), to reliably transport data across heterogeneous networks. IP is concerned with routing data from source to destination host through one or more networks connected by routers, while TCP provides a reliable end-to-end data transfer service.

Wireless communication use many of the protocols designed for wired links, e.g. TCP. However, TCP could not perform efficiently in wireless environments where the bit error rate is much higher.

A general description of the problems of TCP over wireless networks is reported in (Meyer & Sachs, 2003; Barakat, Altman & Dabbous, 2000; Xylomenos, Polyzos, Mahonen & Saaranen, 2001) and some specific solutions are given in (Meyer & Sachs, 2003; Ludwig & Katz, 2000).

In this paper we examine the performance of TCP over UMTS Dedicated Channels (DCH). The performance of TCP is evaluated for Constant Bit Rate (CBR) traffic over dedicated channels with different downlink bit rates and TTIs.

The remainder of this paper is structured as follows. In section 2 the UMTS architecture is briefly described. Section 3 reviews the main features of the simulation model while section 4 is dedicated to the experiments results. Finally, some concluding remarks and planned next steps are briefly described.

UMTS Overview

Figure 1 shows the system architecture of UMTS for packet-switched operation. The UMTS functionality is divided into three groups: User Equipment (UE), UMTS Terrestrial Radio Access Network (UTRAN) and Core Network. UTRAN consists of Node B and Radio Network Controller (RNC). The Core Network comprises two basic nodes: Serving GPRS Support Node (SGSN) and Gateway GPRS Support Node (GGSN).

GGSN provides internetworking with external packet switched networks such as IP networks via the Gi interface. SGSN is connected to RNC via the IuPS interface. UE is connected to UTRAN over the UMTS radio interface Uu.

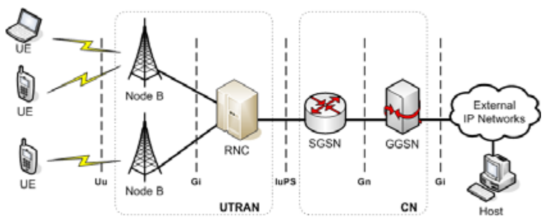

Figure 1. UMTS Architecture

Simulation Model

This section reviews the main features of the simulation model that has been implemented by using the ns -2 simulator (The NS-2 Simulator). In particular, we examine the performance of TCP over UMTS Dedicated Channels (DCHs) with different downlink bit rates and TTIs. The performance of TCP is evaluated for Constant Bit Rate

(CBR) traffic as background traffic to the system. During the simulations we make the following measurements:

- **End-to-End Packet Delay**: Time spent from sending a traffic load packet to the layer until correct reception of the packet by the traffic load receiver.
- **Delay in RAN**: Time required for any packet to travel from RNC to UE.
- **Throughput in Wireless Link**: Bits transferred to UE per unit time in bits/sec.

The addressed scenario comprises a UMTS radio cell covered by a Node-B connected to an RNC. The simulation model consists of a UE connected to a DCH as it is shown in Figure 2.

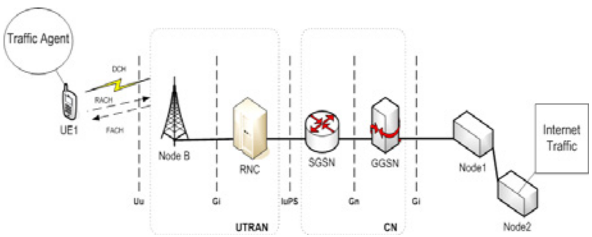

Figure 2: Simulation model for transmission over DCHs.

Table 1: Characteristics of DCHs.

Dedicated channels – DCHs				
	Uplink		Downlink	
	rate (Kbps)	TTI (ms)	rate	TTI
DCH1	64	20	64	20
DCH2	64	20	128	20
DCH3	64	20	384	10
DCH4	384	10	2000	10

In this simulation we use the DCHs to transmit packet data. DCH is a bi-directional channel and is reserved only for a single user. With DCHs we can achieve high bandwidth utilization and high data rates to the end user. Different applications can be connected to the same DCH but only for a single user. The common channels are the Forward Access Channel (FACH) in the downlink and the Random Access Channel (RACH) in the uplink as it is shown in Figure 2. Data are transferred from Node 2 to UE1. This means that the only data going in the uplink channel is TCP ACKs.

We consider CBR traffic packets with rate 120 kbps and packet size 210 bytes as background traffic to the system. In order to evaluate performance of TCP over UMTS air interface we establish 4 DCHs with different downlink bit rates and TTIs. The characteristics of the DCHs are presented in Table 1. In each simulation the UE is connected to a DCH for 200 seconds. The wired part of the simulation model consists of a Node B, a RNC, a SGSN, a GGSN, and two fixed external nodes as it is shown in Figure 2.

Experiments

This section is dedicated to describing the results in terms of performance of TCP over UMTS air interface. As presented in the previous section the performance parameters of primary interest are end-to-end packet delay, delay in Radio Access Network and throughput in wireless link.

Figure 3 shows the end-to-end packet delay for transmission over dedicated channels; those characteristics are presented in Table 1. X-axis gives the packet sequence number while y-axis shows the packet delay in seconds. It is obvious, that as the downlink bit rate of the DCH increases, the end-to-end packet delay decreases. For example, for the DCH1 with downlink bit rate 64kbps the average packet delay is 0.51sec while for DCH4 with downlink bit rate of 2000kbps the average delay is 96msec.

As the packets leave the RNC and arrive at the Node B they queue up in order to be broken down into smaller size packets. Every PDCP PDU (packet of 210 bytes) is segmented into multiple RLC PDUs of fixed size. Each of these PDUs fits into a transport block in order to be transmitted over the air. In the simulation performed, the size of each PDU is 40 bytes.

Taking into consideration that the average packet delay in the core network that we use in the simulation is approximately 76 ms, we can estimate that the delay in Radio Access Network (RAN) for the DCH4will be:

$$Delay_{RAN} = Delay_{End-to-end} - Delay_{Core} =$$
$$= (96-76)ms = 20ms$$

Figure 4 shows the delay in RAN for different DCHs. According to Figure 4 the average delay in RAN for the DCH4 is 16ms.

Finally, Figure 5 presents the throughput in wireless link for every DCH. Y- axis gives the throughput in kbps while x-axis represents the duration of the simulation. For DCHs with downlink rates lower than 120kbps (background traffic rate), such as DCH1 and DCH2, the throughput in wireless link is quite lower from the nominal value. In addition, for faster DCHs, such as DCH3 and DCH4, the throughput in wireless link is very close to the nominal value.

An obvious observation from studying Figure 3 and Figure 4 is that the delay decreases as we increase the downlink bit rate of the DCH. In DCHs with low downlink speeds, the Node B cannon serve a great number of packets that arrives to it and the result is that some packets dropped and have to be retransmitted. As a sequence the packet delay in the wired part of the network increases. For example, Figure 3 shows that the end-to-end packet delay for the DCH1 has a high average value of 0.51 sec. Furthermore for the same DCH the average delay in RAN according to Figure 4 is 52msec.

A similar observation comes out when in the case of the same DCH, if we increase the transmission rate of the background traffic. In that case both end-to-end delay and delay in RAN increase. As the background traffic rate increases more packets arrive at a time interval in RNC than can be served in that interval, resulting in higher queuing delays as well as drops. This can be shown in Figure 5, where the throughput in the wireless link for the DCH1 is approximately 52kbps, which is very small compared to the 120kbps of the background traffic.

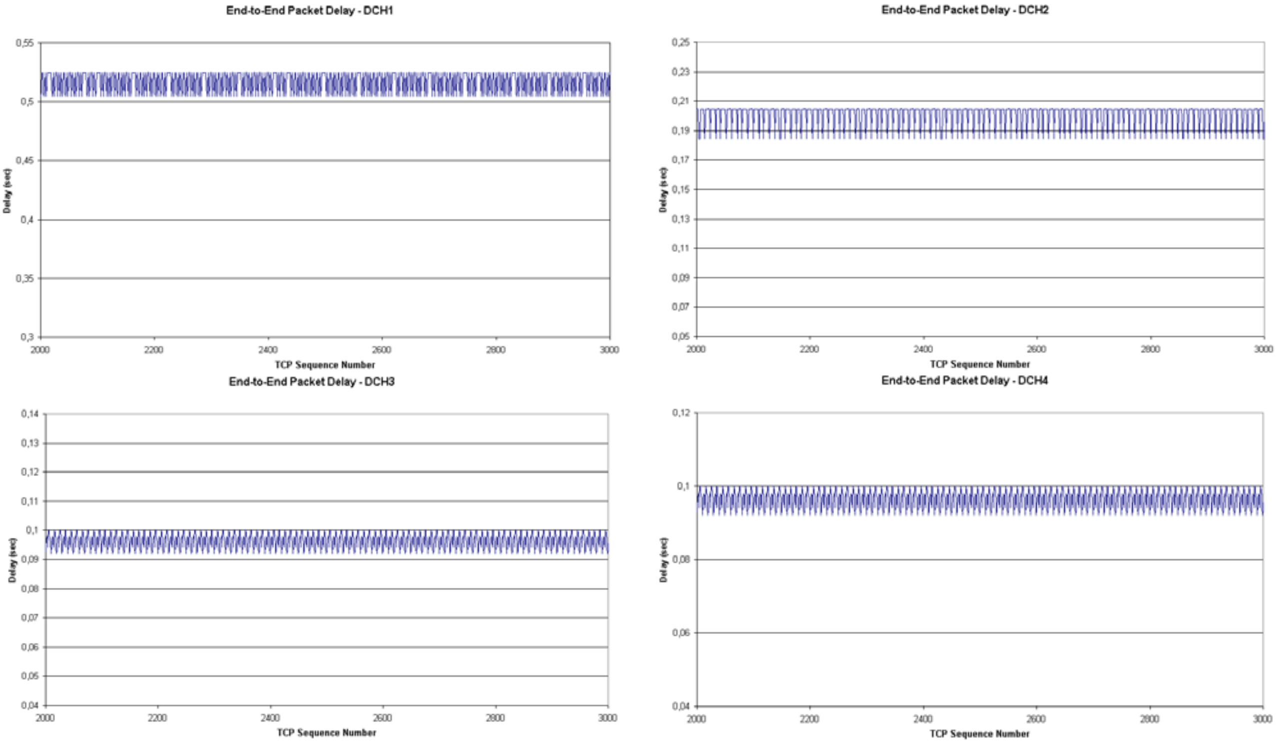

Figure 3: End-to-end packet delay for DCHs.

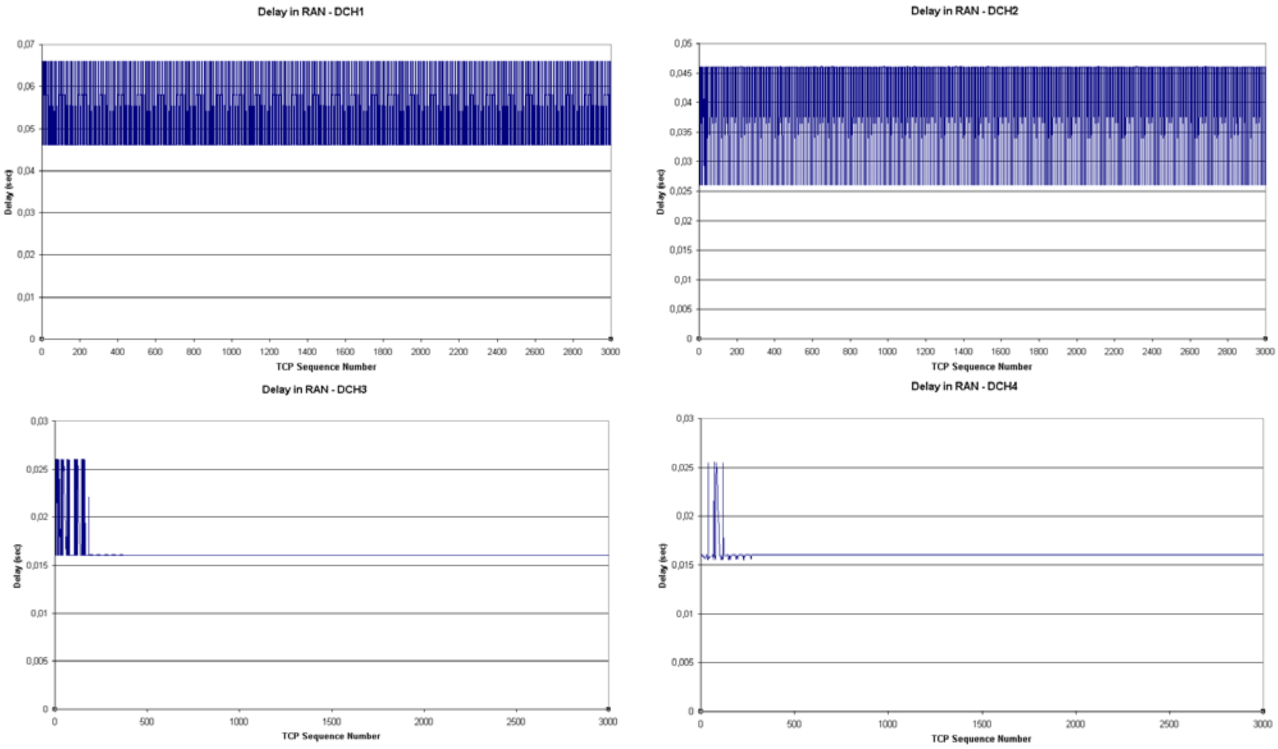

Figure 4: Delay in RAN for DCHs.

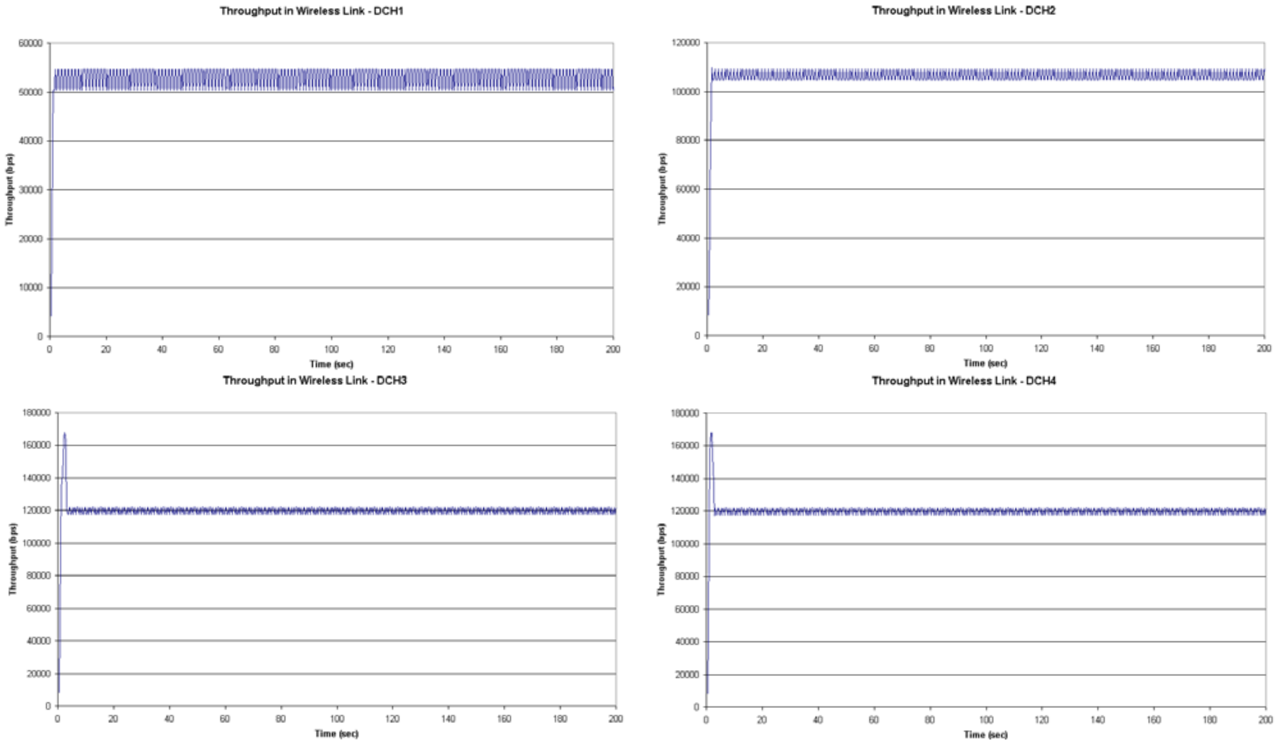

Figure 5: Throughput in wireless link

Conclusions and Future Work

This work concentrates on the evaluation of UMTS air interface using simulations. The performance parameters are end-to-end packet delay, delay in RAN and throughput in wireless link. We consider DCHs as the transport channels to the experiments. We evaluate four DCHs with different characteristics.

In the presented simulations, TCP runs from end-to-end including the radio interface. This paper proves that this solution is applicable to the mobile user but the performance suffers since TCP mechanisms do not efficiently use the guaranteed QoS of the UMTS radio bearers in terms of delay and throughput. If packets are lost the TCP protocol infers that there must be congestion in the network. As a sequence, TCP retransmission leads to a decrease of the send window size, which results less throughput. Very high delays and small throughput lead to unsatisfied mobile users.

The step that follows these experiments is to evaluate different versions of TCP such as TCP SACK, Split TCP, TCP Westwood etc. It is widely known that the standard version of TCP is unable to handle the unreliable radio link of mobile users due to the high error rates and TCP’ s slow error handling.

Furthermore, since CBR traffic best describes the voice traffic in circuit switched and not in packet switched networks; we plan to develop better traffic models for the voice traffic such as the ON-OFF model that captures the human voice silent periods. Moreover, we plan to evaluate the performance of TCP for multimedia traffic sources and streaming traffic sources. The main difference between these two sources is that multimedia applications require additional synchronization information that has to be included in the multimedia model.

Finally, the next goal is to integrate the simulation model in order to support a great number of simultaneous users over a single cell as well as to support the handover functionality and mobility management.

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