Performance Evaluation of TCP over UMTS Transport Channels

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Abstract

Universal Mobile Telecommunications System (UMTS) is a third-generation cellular network that enables high-speed wireless Internet access. 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. It is widely known that TCP does not perform efficiently in wireless links where the bit error rate is high enough. This paper has as main target to evaluate the performance of TCP data transfer over the UMTS air interface, by means of a rather detailed simulation model. We examine the performance of TCP over UMTS Dedicated Channels (DCHs) with different bit rates and Transmission Time Intervals (TTIs) and we represent some simulation experiments for UMTS High Speed Downlink Packet Access (HSDPA) transmissions.

Keywords: **UMTS, Mobile Internet, HSDPA, 3G Cellular Networks, Wireless TCP**

1. Introduction

UMTS constitutes the third generation of cellular wireless networks which aims to provide high-speed data access along with real time voice calls [3]. Wireless data is one of the major boosters of wireless communications and one of the main challenges of next generation standards [4]. The UMTS has been designed to address specifically the needs of data traffic.

Unlike second-generation networks, UMTS provides a variety of services and data rates up to 2 Mb/s in indoor or small-cell outdoor environments, and wide-area coverage of up to 384 kb/s. In addition, the packet-switched mode of UMTS allows mobile users' access to the Internet in a seamless fashion.

In general, UMTS is designed to include ambitious service features, the ability to communicate in movement, anytime and anywhere, through an enormous variety of applications and universally usable terminals. This is basically a process of extending, and adding to, the services now provided to fixed network users to mobile customers.

The most popular and widely used Internet applications are File Transfer Protocol (FTP), Hypertext Transfer protocol (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 uses 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. Different wireless technologies have different characteristics, but a few properties are common and they will have an impact on TCP's performance.

High bit error rate is maybe the most important factor that can limit the utilization of the link. When high enough, it can cause all communication to fail. One big challenge in wireless communication is to minimize the bit error rate, BER. However, it is expensive to build networks with low BER and therefore it is important to find a way to make the upper layers unaware of the data loss that the high BER cause. This is where retransmission over the wireless link can hide the losses but at the cost of increase in delay variation. If the link encounters an error during the transmission, it may try to resend the data and thereby causing delay. The delay will vary since retransmissions only happen occasionally, due to random loss. This may have an impact on the overlying protocols if those are dependent on the delay of the link. Simulation results shows that TCP over UMTS performs well in lower data rates i.e. 64 and 144 kbps but in case of higher data rates its performance is significantly reduced [9]. A general description of the problems of TCP over wireless networks is reported in [9], [10] and [11] and some specific solutions are given in [9] and [12].

In this paper we examine the performance of TCP over UMTS Dedicated Channels (DCH) and we represent some simulation experiments for UMTS High Speed Downlink Packet Access (HSDPA) transmissions. Firstly, the performance of TCP is evaluated for Constant Bit Rate (CBR) traffic over dedicated channels with different downlink bit rates and TTIs. The addressed scenario comprises a UE connected to DCH with downlink bit rate of 64kbps and 20ms TTI. The simulation repeated for DCHs with downlink bite rates 128, 384 and 2000 kbps and TTIs 20 and 10 msec. Furthermore, we exploit the TCP performance for HSDPA transmissions. The simulation model consists of 4 UEs connected to a HS-DSCH. The UEs are 300 or 500m away from the node B in a Rayleigh fading environment. Four different traffic applications are running from an external source node to the UEs for a period of 500 seconds.

The remainder of this paper is structured as follows. In section 2 the UMTS architecture is briefly described while in section 3 some traffic models that best simulate voice and ftp traffic are presented. Following this, section 4 reviews the main features of the simulation model. Section 5 is dedicated to the experiments results. Finally, some concluding remarks and planned next steps are briefly described.

2. UMTS Overview

This section is dedicated to describing the structure of the Universal Mobile Telecommunication System. The term structure is used for referring to the logical view of the static structure of the architecture in terms of its components, their interconnections, and the interfaces and operations offered by these components.

Figure 1shows 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) [2].

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.

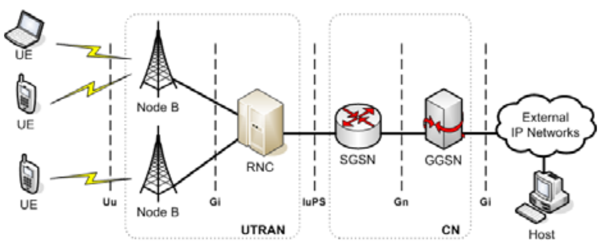


Figure 1. UMTS Architecture

Figure 2 depicts the UMTS protocol architecture for the transmission of user plane data which is generated by TCP or UDP-based applications. The applications as well as the TCP/IP protocol suite are located at the end-nodes, namely, UE and host.

The Packet Data Coverage Protocol (PDCP) is responsible for the transmission and reception of network Packet Data Units (PDUs). It provides for the mapping from one network protocol to one RLC entity. In the same way, it provides for compression (in the transmitting entity) and decompression (in the receiving entity) of redundant network PDU control information (header compression / decompression) [5].

The Radio Link Control (RLC) layer can operate in three different modes: acknowledged mode, unacknowledged mode and transparent mode. The acknowledged mode provides reliable data transfer over the error-prone radio interface. This is accomplished by retransmitting erroneous RLC PDUs. In the unacknowledged mode, the data transfer over the radio interface is not error free but no additional delay due to retransmission. The functionality of transparent mode is similar to unacknowledged mode but no protocol information is appended to the PDU [5].

Medium Access Control (MAC) handles data streams directed to it from RLC and RRC and provide an unacknowledged transfer mode to the upper layers. The communication between RLC and MAC is carried out through SAPs known as logical channels. In the same way, the communication between MAC and PHY is handled through SAPs known as transport channels [5].

The Physical (PHY) layer controls the use of physical channels at the radio interface. PHY is responsible for mapping transport channels on to physical channels. PHY also provides functions like forward-error correction and error detection, interleaving, spreading, modulation, rate matching, and radio frequency processing.

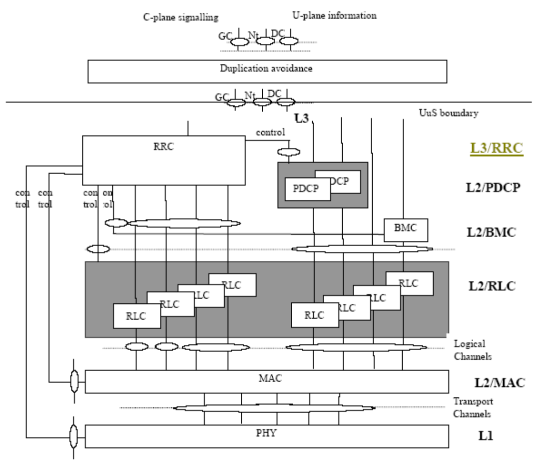


Figure 2. Protocol Architecture of UMTS [2]

3. Traffic Models for http and ftp Applications

The third generation of mobile telephony introduces new features with different requirements. The support of QoS and high speed data communication, allows the introduction in mobile telephony of applications from the world of Internet and multimedia. In order to test the performance of such new applications it is necessary to have a parameterized traffic model that can simulate the traffic created by these applications [15]. According to [15] there are four different traffic sources in UMTS: Conversational traffic sources, Block traffic sources, Streaming sources and Multimedia sources. The first two traffic sources are considered in this paper.

The conversational traffic is mainly the voice traffic. Voice traffic is quite well described as CBR traffic. In this paper we evaluate the performance of UMTS for CBR traffic with different characteristics and traffic rates ranging from 8kbps to 120kbps. The block traffic comprises www traffic and non real time data. In the case of non real time data that is mainly FTP transfers, each file is not transmitted as a single entity, instead, it is split into packets, according to the protocol used. In this paper we use TCP as transport protocol and its packet size is 210 bytes. Similarly, in the case of web page downloading, the page comprises different objects that are sent separately from each other. In this way, it is possible for the web browser to show the web page even if it is not completely downloaded.

Modeling of data traffic loads is not an easy task and not comparable with the voice-centric telephony networks. There are many different multimedia traffics coming from current and future applications and having a single model to illustrate the characteristics of all this traffic would be a complex research task in the years to come. Considering the exponential increase in the traffic load of the data networks and the necessity of designing these systems by using precise traffic models, there are no many available models. A general description of traffic models that can be used in UMTS is reported in [15]

Data connections on the Internet are known to be bursty where the basic burst size is best described with a Pareto distribution:

$$P_r[burst\_size > s] = \left(\frac{k}{s}\right)^{\beta}$$

where k is the minimum size and $1 < \beta \leq 2$ (for Internet traffic such as www and ftp) [6].

4. Simulation Model

This Section reviews the main features of the simulation model that has been implemented by using the ns -2 simulator [7]. We examine the performance of TCP over: (a) UMTS Dedicated Channels (DCH) with different downlink bit rates and TTIs and (b) UMTS High Speed Downlink Shared Channels. The performance of TCP is evaluated for FTP traffic with different characteristics. Furthermore, we exploit the TCP performance for traditional traffic models such as constant bit rate traffic. During the simulations we make the following measurements:

- **End-to-End Packet Delay**: Time spent for 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

4.1 Transmission over DCHs

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 3. The model is based on the system architecture discussed in a previous section.

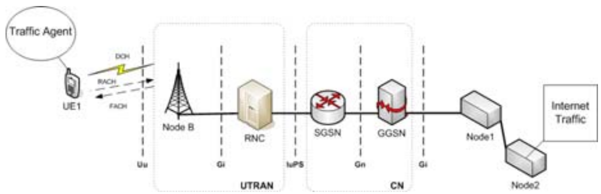


Figure 3. Simulation Model for Transmission over DCHs

In this simulation we use the DCH to transmit packet data. DCH is a bi-directional channel and is reserved 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 3. Data are transferred from Node 2 to UE1. This means that the only data going in the uplink channel is TCP ACKs.

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

Table 1. Characteristics of DCHs

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 deferent 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 3. The characteristics of connection lines between them are shown in Table 2.

4.2 HSDPA Transmission

High Speed Downlink Packet Access (HSDPA) supports the introduction of high bit rate data services and will increase network capacity, while minimizing operators' investment. It provides a smooth evolutionary path for Universal Mobile Telecommunications System (UMTS) networks to higher data rates and higher capacities, in the same way as Enhanced Data rates for GSM Evolution (EDGE) does in the Global System for Mobile communication (GSM) world. The introduction of shared channels for different users will guarantee that channel resources are used efficiently in the packet domain, and will be less expensive for users than the use of dedicated channels [13].

Through HSDPA it is possible to achieve peak data rates of about 10 Mbit/s (the maximum theoretical rate is 14.4 Mbit/s). HSDPA uses the following channels [8]:

- High Speed Physical Downlink Shared Channel (HS-PDSCH): Carries actual packet data; Spreading Factor (SF) = 16, QPSK/16QAM, power controlled by Node B, up to 15 HS-PDSCHs per cell, aggregate data rates of up to 14.4 Mbit/s per cell.
- High Speed Shared Control Channel (HS-SCCH): Downlink channel, which carries signaling information (channel code set, modulation scheme, transport block size, HARQ process number, redundancy and constellation version parameters, new data flag and UE identity), SF=128, QPSK, power controlled by Node B, up to 32 HS-SCCHs per cell, up to four HS-SCCHs per user equipment.
- High Speed Dedicated Physical Control Channel (HS-DPCCH): Uplink channel carrying signaling information (ACK/NACK and Channel Quality Indicator, CQI), SF=256, QPSK, terminated in Node B. HS-PDSCH, HS-SCCH and HS-DPCCH use a Transmission Time Interval (TTI) of 2 ms. This interval is also called a 'subframe'.

The addressed scenario comprises a HSDPA UMTS radio cell covered by a Node-B connected to an RNC. The simulation model consists of 4 UEs connected to a HS-DSCH. The UEs are 300 or 500m away from the node B in a Rayleigh fading environment (Figure 4). The model is based on the system architecture discussed in a previous section.

More specifically, the UE1 and UE3 are located 300m away from the Node B while the UE2 and UE4 are located 500m away from the Node B. We consider two traffic applications as background traffic to the system: a) FTP b) CBR traffic source. In the above scenarios the clients are placed in the wired nodes (Figure 4). 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 4. The characteristics of connection lines between them are shown in Table 2. Furthermore, in Table 2 we can see that the average delay at the wired part of the network model is 76 msec.

From	To	Bandwidth	Av. Delay
Node 2	Node1	10Mbit	35ms
Node 1	GGSN	10Mbit	15ms
GGSN	SGSN	622Mbit	10ms
SGSN	RNC	622Mbit	1ms
RNC	Node B	622Mbit	15ms
Average Total Delay Through the Wired Part of the Model:			76ms

Table 2. Connection Lines between Nodes

The UEs are connected to a HS-DSCH for 500 seconds. This period is divided in the following intervals:

- **Interval 1: [0, 150sec]**. FTP traffic with packet size 210 bytes generated in node 2 (Figure 4) heading to UE1 (300m from node B).
- **Interval 2: [150, 300sec]**. FTP traffic with packet size 210 bytes generated in node 2 (Figure 4) heading to UE2 (500m from node B)
- **Interval 3: [300, 400sec]**. CBR with packet size 210 bytes and rate 20 kbps generated in node 2 (Figure 4) heading to UE3 (300m from node B)
- **Interval 4: [400, 500sec]**. CBR with packet size 210 bytes and rate 8 kbps generated in node 2 (Figure 4) heading to UE4 (500m from node B)

For every interval we compute the end-to-end packet delay, the delay in RAN and the throughput in wireless link. The results are presented in the following section.

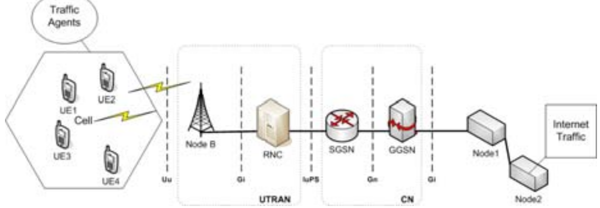


Figure 4. Simulation Model for HSDPA Transmissions

5. 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.

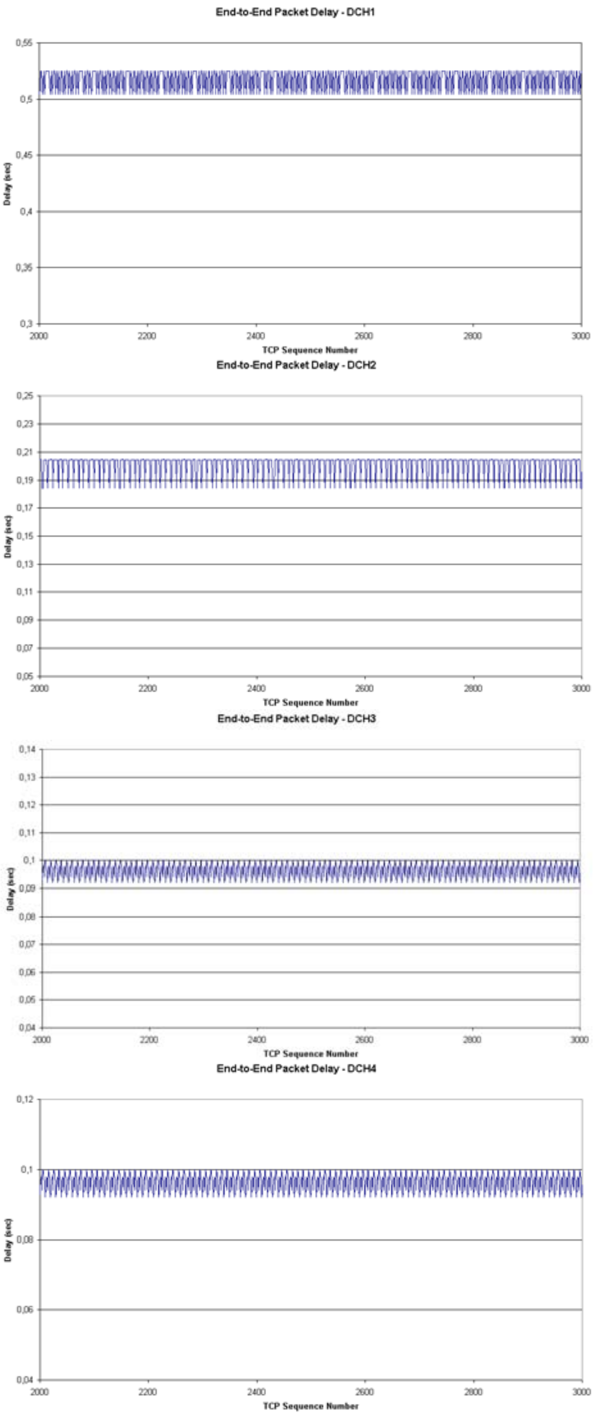


Figure 5. End-to-End Packet Delay for DCHs

5.1 Transmission over DCH

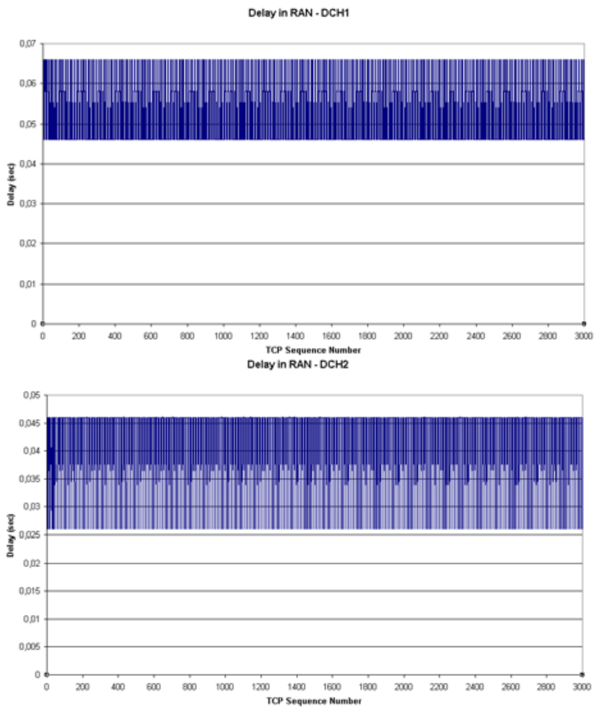
Figure 5 shows the end-to-end packet delay for transmission over dedicated channels; those characteristics are presented in Table 1. The X-axis gives the packet sequence number while y-axis shows the packet delay in seconds. It is obvious, that as the downlink bit rate of the DCH increases, the end-to-end 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 of the wired part of the network is approximately 76 ms (according to Table 2), we can estimate what the delay in Radio Access Network (RAN) will be for the DCH4:

$$\text{Delay in RAN} = \text{delay}_{\text{end-to-end}} - \text{delay in wired network} = (96-76)\ \text{ms} = 20\text{msec}$$

Plots in Figure 6 show the delay in RAN for different DCHs. According to Figure 6 the average delay in RAN for the DCH4 is 16ms, which is very close to the empiric calculation that described above.



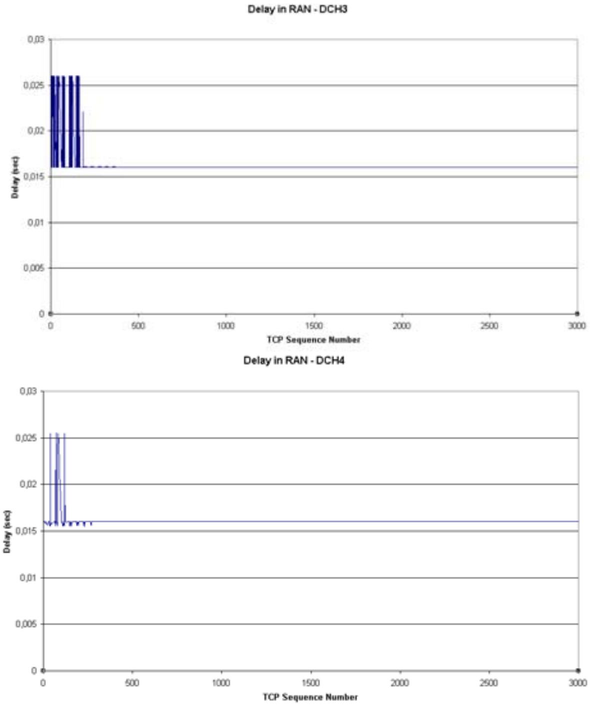


Figure 6. Delay in RAN for DCHs

Figure 7 presents the throughput in wireless link for every DCH. The Y-axis presents 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, as it is depicted in Figure 7.

An obvious observation from studying Figure 5 and Figure 6 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 5 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 6 is 52msec.

A similar observation comes out 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 7. The downlink rate of the DCH1 is 64kbps while the rate of the background traffic is 120kpbs. The simulation shows that the throughput in wireless link for the DCH1 is approximately 52kbps, which is very small compared to the 120kbps of the background traffic.

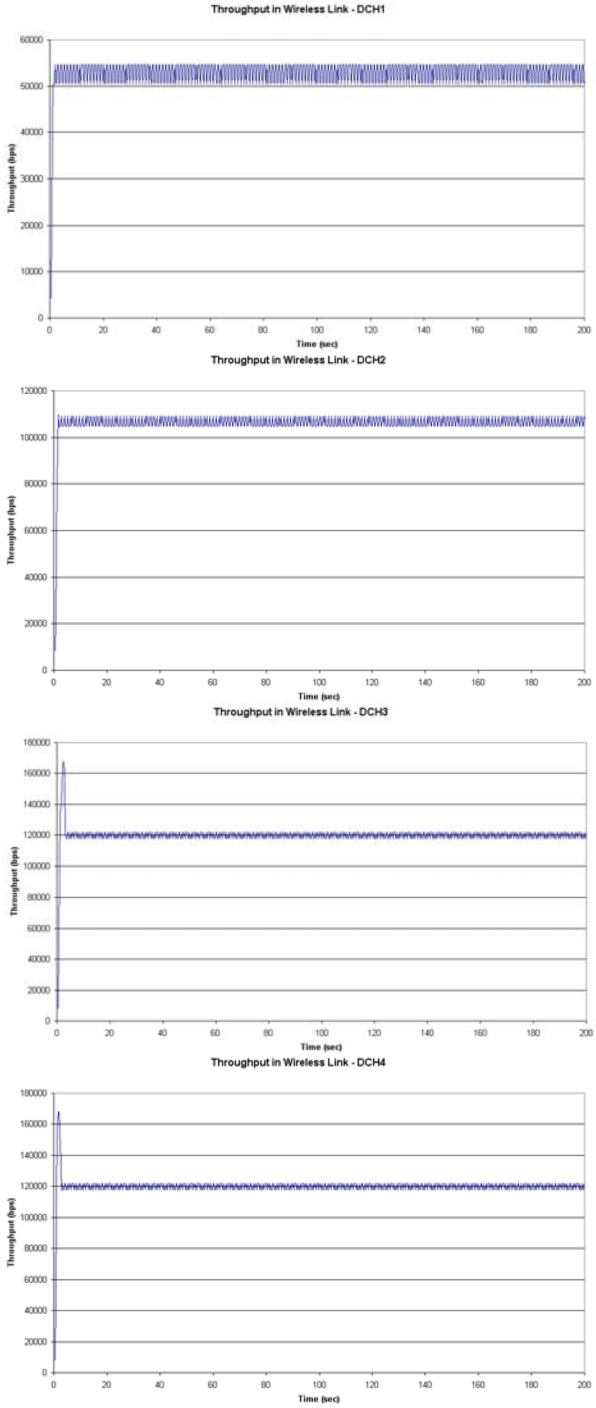


Figure 7. Throughput in Wireless Link

5.2 HSDPA Transmission

The results of the simulation are given in the following figures. In Figure 8, the Y-axis gives the end-to-end time delay and the x-axis shows the increasing number of the packets. Some packets experience more delay than others and some packets dropped. End-to-end delay consists of the time a packet takes to travel from source to destination including medium propagation delay and queuing delay. Variation in the minimum values is simply an indication of the fact that the UEs have different distances from the Node B and radio propagation delay is dependent upon that distance. If we split the simulation model to the fixed part of the network and the wireless part we can say that for the fixed/wired part of the network the delays have very small jitter, almost equals delays for all packets. What contributes most to the variance in the end-to-end delay (Figure 8) is the wireless part of the network.

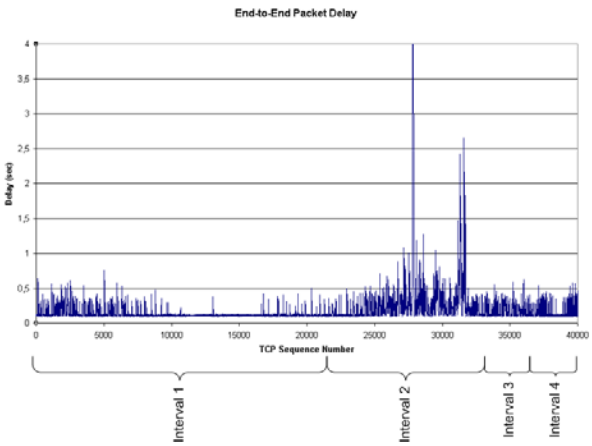


Figure 8. Packet delay for HSDPA transmissions

According to [14] the delay of UMTS (CN and UTRAN) combined with the Internet is assumed to be at least 150 ms one-way for non real time services. Out of these, 60 ms represent the Internet delay (see Table 2) and the rest are accounted for the delay in UMTS. The delay in UMTS is in turn made up of two parts: the delay resulting from the core network is assumed to be 20ms, and the delay for the radio access network.

Figure 9 shows the delay from RNC to UEs (RAN). 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. For our simulations the length of the queue is 2000.

The average packet delay for every UE (Interval) is shown in Table 3. Finally, Figure 10 presents the throughput in wireless link for every UE. X-axis shows the duration of the simulation while y-axis presents the throughput in bits/sec. The heavy traffic in the wireless channel of 64kbps downlink rate (UE1 and UE2) causes some insufficient results as far it concerns the throughput in wireless link. This is caused because the rate of the traffic is quite higher than the transport channel can serve. However, wireless cellular networks are likely to experience delay spikes exceeding the typical round-trip-time figures, which can cause spurious timeouts that lead to unnecessary retransmissions and reduction of the TCP sender's transmission rate. Consequently, more unnecessary retransmissions are needed and thus the throughput of the TCP is degraded. Previous research indicates that increasing the TCP window can significantly improve the performance of TCP for FTP traffic in wireless cellular networks [16]. In addition, in traffic rates lower than the downlink rate of the HS-DSCH (UE3 and UE4), we can see that both UEs approach the nominal value of the background traffic rate of 20kbps and 8kbps respectively.

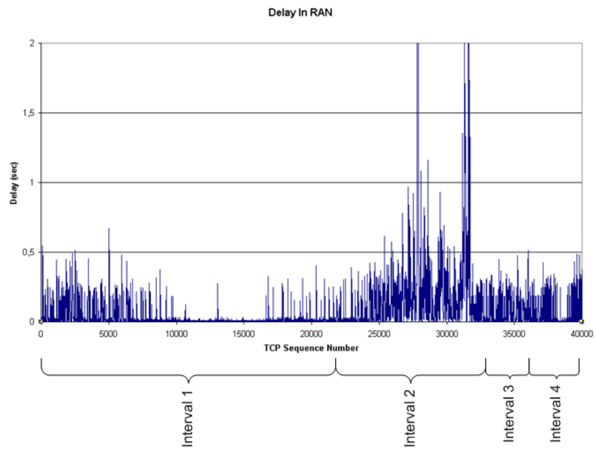


Figure 9. Delay in RAN

UEs	1	2	3	4
Average Delay (sec)	0.1392	0.326	0.1285	0.1463

Table 3. Performance issues

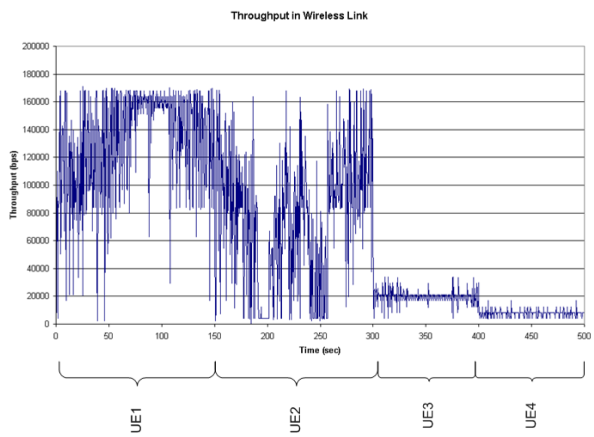


Figure 10. Throughput in Wireless Link

6. Conclusions and Future Work

This work is concentrated on the evaluation of UMTS air interface using simulations. In this paper we present some evaluation results for the performance of UMTS for different traffic types. The performance parameters are end-to-end packet delay, delay in Radio Access Network and throughput in wireless link. We consider two types of transport channels in this evaluation: Dedicated Channel and High Speed Downlink Shared Channel. The performance of TCP over UMTS is evaluated for two main traffic models:

ftp transfer and CBR traffic. As far data transmission over DCHs is concerned, we tested four DCHs with different characteristics in terms of downlink data rate and transmission time interval. With the aid of 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. For High Speed Downlink Packet Access transmissions, we used a simulation model of four UEs connected to a HS-DSCH. The experiments show that in traffic applications, with low data rates, the HS-DSCH can sufficiently serve a great number of simultaneous users.

In the presented simulations, TCP runs 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 consequence, TCP retransmission decreases the send window size, which results less throughput. Very high delays and small throughput lead to unsatisfied mobile users. It has to be mentioned that the effects of the core network are not included in our simulations. The entire core network in our simulations is assumed to be an IP-based network. Thus, it can be concluded that the overall performance will get worse since TCP, as an end-to-end protocol will experience additional delays and congestion in the core network.

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.

7. References

[1] 3GPP, available at http://www.3gpp.org.

[2] 3GPP, Network Architecture, TS 23.002, March 2000.

[3] H. Holma and A. Toskala, WCDMA for UMTS: Radio Access for Third Generation Mobile Communications, John Wiley & Sons, 2001.

[4] P. Chaudhury, W. Mohr, The 3GPP Proposal for IMT-2000, IEEE Communications Magazine, December 1999, pp. 72-81.

[5] M. Yacoub, Wireless Technology: Protocols, Standars, and Techniques, CRC Press, 2002.

[6] A. Jamalipour, The Wireless Mobile Internet: Architectures, Protocols and Services, John Wiley & Sons, 2003.

[7] The NS-2 Simulator, available at http://www.isi.edu/nsnam/ns.

[8] Alcatel Telecommunication Review, 3G Evolution Towards High Speed Downlink Packet Access, 4th Quarter 2003/ 1st Quarter 2004, available at http://cnscenter.future.co.kr/rsc-center/vendor-wp/a-b.html.

[9] M. Meyer, J. Sachs, Performance Evaluation of a TCP Proxy in WCDMA Networks, IEEE Wireless Communications, October 2003, pp. 70-79.

[10] C. Barakat, E. Altman, W. Dabbous, On TCP Performance in a Heterogeneous Network: A Survey, IEEE Communications Magazine, January 2000.

[11] G. Xylomenous et al., TCP Performance Issues over Wireless Links, IEEE Communications Magazine, April 2001.

[12] R. Ludwig, R.H. Katz, The Eifel Algorithm: Making TCP Robust Against Spurious Retransmissions, ACM Computer Communications Review, Vol. 30, No. 1, January 2000.

[13] H. Holma, A. Toskala, WCDMA for UMTS, John Wiley & Sons, 2002.

[14] 3GPP, Delay Budget within the Access Stratum, TS 25.853, March 2001.

[15] P. Kavallaris, Traffic Modeling for Mobile Multimedia Networks, IST Mobile and Wireless Communications Summit 2002, Thessaloniki, Greece.

[16] F. Xin, A. Jamalipour, Analysis of the Effect of Delay Spike and Initial Congestion Window Size on TCP Performance in Wireless Cellular Networks, SPECTS 2004, July 25-29, 2004, San Jose, California.