MULTIMEDIA TRANSMISSION OVER THIRD GENERATION CELLULAR NETWORKS

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The scheme of real time streaming video is one of the newcomers in wireless data communication, raising a number of new requirements in both telecommunication and data communication systems. This scheme applies when the user is experiencing real time multimedia content. This paper has as main target to study the performance of video transmission over the third generation cellular networks. In particular, we examine the performance of UMTS Dedicated Channels (DCHs) for real time MPEG-4 video transmission in Downlink direction. Finally, we examine if real time video transmission in conjunction with Internet traffic is applicable in UMTS radio interface.

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 [1]. Wireless data is one of the major boosters of wireless communications and one of the main motivations of next generation standards [2]. Bandwidth is the most precious and limited resource of UMTS and every wireless network. Video applications produce large amount of data. As a result, video is transmitted in compressed format to reduce the generated data rates. Among the used compression techniques, MPEG-4 is the standard that has recently gained a considerable attention [3].

This paper has as main target, to examine the performance of UMTS Dedicated Channels (DCHs) for real time MPEG-4 video transmission. The results demonstrate that video quality can be substantially improved be preserving the high priority video data during the transmission. In [3] the authors suggest a dynamic bandwidth allocation scheme for MPEG video sources suitable for wireless networks. The proposed algorithm exploits the structure of the MPEG video stream and allocates bandwidth on a scene basis. This results a high bandwidth gain, which affects the overall network performance.

The specific contribution of this work is that provides both an overview on MPEG-4 video transmission over third generation cellular networks, and an indication about how friendly is the behavior of video transmission over UMTS towards any other Internet application that coexist in the same channel.

This paper is structured as follows. Section 2 presents an analytical computation of Packet Service Time for MPEG-4 video traffic. Following this, section 3 reviews the main features of the simulation model. Section 4 is dedicated to the experiments results. Finally, some concluding remarks and planned next steps are briefly described.

2. Analytical computation of packet service time for MPEG-4 video traffic

In this section we present an analytical computation of the time required for any packet of a given video sequence to travel from the Radio Network Controller to the mobile user. The video traces we use, are taken from [5]. The packets of the video sequence do not have a constant size. The size of the MPEG-4 packets which are being transmitted over the UMTS air interface is presented in Figure 1. The video sequence is in QCIF format (176x144 pixels) at the PAL frame rate of 25 frames per second. The average packet size is 758 bytes. The above-mentioned video traffic is going to be also used in the simulation that it is described in the following sections.

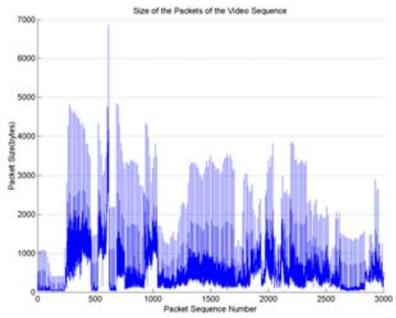

Figure 1. MPEG-4 Packet Size

Considering an MPEG-4 video transmission from a fixed Internet node to a mobile user (Downlink) (Figure 4), 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 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. According to [4], the size of the RLC PDUs is 40 bytes.

Based on the analysis presented in [4], in Downlink, for any RLC PDU that transmitted over the air the RNC receives a status report of the UE 68ms after its transmission. For the opposite direction, since generally, the uplink TTI is twice as large as in the downlink direction, the UE receives a status report 40ms after its transmission.

In this section we determine the minimum IP packet service time for the MPEG-4 video traffic that we use in the simulation. This time is comparable with the delay in Radio Access Network (Time required for any packet to travel from RNC to UE). This time is calculated as follows. The number of RLC PDUs an IP packet is segmented into is: $N_{PDU} = \frac{Packet\_size(bytes)}{40bytes}$. The maximum RLC PDUs that can be transmitted within one TTI is: $L_{\max} = 8$. Consequently, the number o TTIs required for the transmission of the packet is: $N_{TTI} = \frac{N_{PDU}}{L_{\max}}$

Since the TTI in downlink is 10ms, the minimum time required for the transmission and reception of a whole packet is: $T_{\min} = 40ms + \lceil N_{TTI} - 1 \rceil \cdot 10$. For the MPEG-4 video traffic that we use in the simulation, the minimum time required for the transmission and reception of any packet of the video sequence is presented in Figure 2. The y-axis shows the packet service time in msec, while the x-axis presents the packet sequence number.

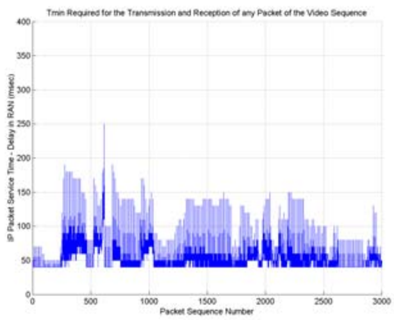

Figure 2. Minimum time required for the transmission and reception of any packet of the video sequence.

Furthermore, Figure 3 presents the pdf of the minimum time required for the transmission and reception of any packet of the video sequence. The x-axis presents the packet service time, while the y-axis presents the pdf. Since all IP packets do not have a constant size, the minimum time varies and has a maximum at Tmin = 40 msec with value pdf = 0,5087. This means that the

50,87% of the packets have a minimum delay in RAN of 40 msec approximately. The average Tmin is 54,39 m sec.

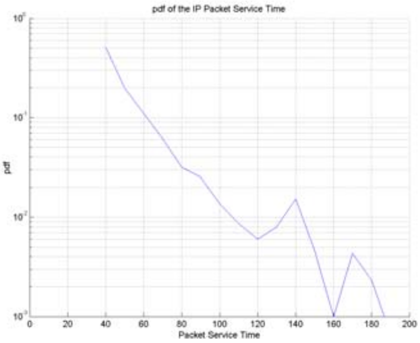

Figure 3. Pdf of the IP packet service time.

3. Simulation model

This Section reviews the main features of the simulation model that has been implemented by using the ns-2 simulator. The performance of DCHs is evaluated for MPEG-4 traffic with different characteristics. Furthermore, in order to exploit the performance of DCHs we consider as background traffic to the system, HTTP and SMTP applications. During the simulations we make the following measurements: a) Delay in RAN (from RNC to UE) and b) 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 DCH as it is shown in Figure 4.

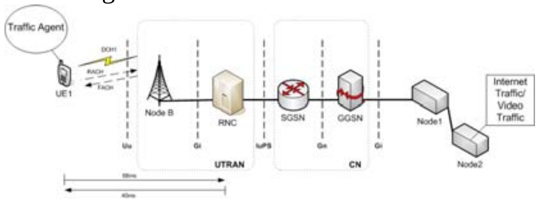

Figure 4. The simulation model.

In our simulations, we use a 384Kbit-DCH in the downlink and a 128Kbit-DCH in the uplink direction. The TTIs are 10ms and 20ms in the down- and uplink direction, respectively. A real time MPEG-4 video streaming generated in node 2 and through DCH (downlink) heading to UE1 using the UDP as transport protocol (foreground traffic). Also, HTTP and SMTP traffic generated in node 2 heading to UE1, using the TCP as transport protocol (background traffic). The video sequence is following the MPEG4 standard, in QCIF format (176x144 pixels) at the PAL frame rate of 25 frames per second. The video traces we use, are taken from [5]. The duration of the video applications is 200 seconds.

4. Experiments

This section is dedicated to describing the results in terms of performance of real time video transmission over UMTS DCHs. Firstly, we present the performance parameters for MPEG-4 video transmission over UMTS DCHs and secondly, we present the results for transmission of MPEG-4 video traffic that coexist in the same DCH with Internet TCP traffic.

4.1. *MPEG-4 video transmission without background traffic*

Figure 5, presents the delay in RAN. In other words, the latter represents the packet service time that it has already computed, in an analytical way, in section 2. The average packet service of the simulation has a value of 61,7 msec while the average packet service time that computed in section 2 is 54,39 msec. As it is shown in Figure 5 around Packet Sequence Number 800 we can see a delay spike. This is due to the fact that these packets have very large size compared to the others and therefore, a great number o TTIs are required for the transmission of the packets.

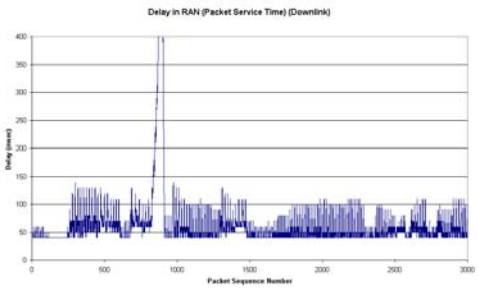

Figure 5. RAN delay for the Video Transmission

The throughput in wireless link is depicted in Figure 6. The y-axis presents the throughput in bps while x-axis represents the duration of the simulation. The red line represents the average throughput in the wireless link. The average throughput has a value of 205 kbps while the downlink bit date of the dedicated channel is 384 kbps.

Figure 7 illustrates a comparison of pdfs of the packet service time for both the analytical computation (presented in section 2) and the simulation results. The red line represents the experiment results while the blue one represents the analytical results. As it is depicted in Figure 7, the simulation results are very close to the analytical results. Furthermore, Figure 7 indicates that the majority of the packets achieve a service time lower that 80ms and only a small number of packets seem to achieve a service time higher that 100ms. This signifies that the majority of the packets that reach the mobile user are characterized by low delay and consequently, the quality of the video sequence that this user will see in his terminal is satisfactory in comparison to the sequence originally sent by the transmitter.

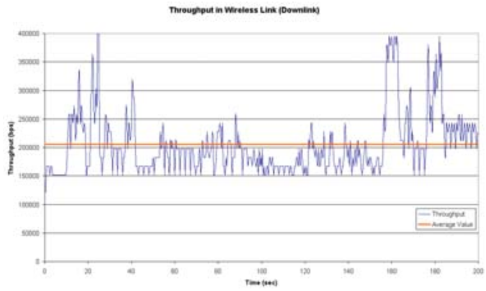

Figure 6. Throughput in Wireless Link (no background traffic)

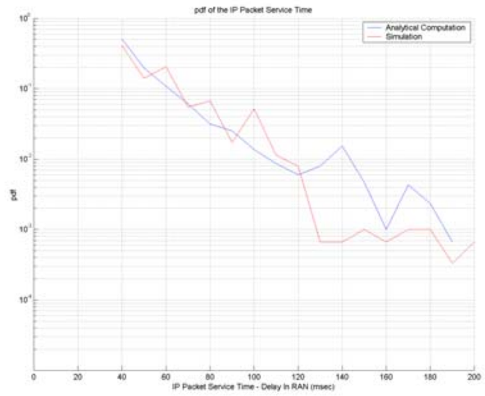

Figure 7. Comparison between Analytical Computation and Simulation Results

4.2. *MPEG-4 video transmission with the presence of additional background traffic*

Figure 8 presents the throughput in the wireless link during the end-to-end MPEG-4 video transmission. The y-axis shows the throughput (in bps) while the x-axis counts the time. In this figure are shown three elements: a) The Video Streaming throughput (navy blue line), b) The throughput of the SMTP traffic following a Pareto distribution (red line) and c) The throughput of HTTP traffic (green line).

The maximum throughput of the SMTP and the HTTP traffic is 50 Kbps. Figure 9 shows the total throughput in the Wireless link, which is the sum of the Video and SMTP traffic during the first time interval (from second 50 to 100), and the sum of the Video and HTTP traffic during the second time interval (from second 100 to 150). As it is shown in Figure 9, the total throughput in the wireless link is always lower than the downlink bit rate of the DCH which is 384 kbps. Considering the delay in RAN, the presence of additional background TCP traffic produces results similar to the results presented in the previous subsection and in particular in Figure 5.

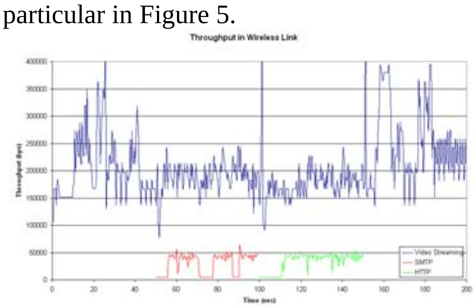

Figure 8. Throughput in Wireless Link (with background TCP traffic)

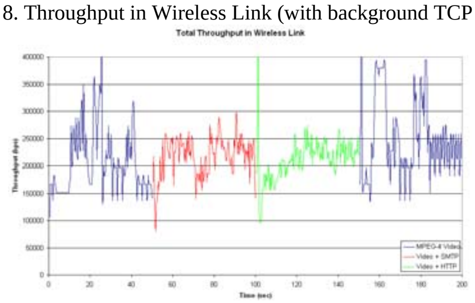

Figure 9. Total throughput in Wireless Link (with background traffic)

An obvious observation that comes out from Figure 9 is that in the time intervals from second 15 to 25 and 150 to 180 there is heavy video traffic in the DCH. If we add additional traffic to these intervals, it is possible the network to be congested and this probably could affect the video sequence in the receiver.

Figure 9 indicates that the UDP video traffic shows a friendly behavior towards the TCP traffic and in particular the SMTP and HTTP traffic that coexist in the same dedicated channel. At this point, it has to be mentioned that the above-described scheme is applicable in UMTS as long as the DCH has enough available bandwidth in order to serve the mixed traffic conditions. In situations where the total bit rate of the traffic is higher than the downlink speed of the DCH, the scheme becomes unstable and this causes serious problems mainly in the transmission of the video sequence. Since Internet traffic is mostly

TCP traffic, if packets are lost the TCP protocol infers that there must be congestion in the network and this leads to the retransmission of the packets until the correct reception in the end user terminal. In addition, concerning the UDP video transmission, if a number of packets of the video sequence are lost then the synchronization between encoder and decoder is broken, and errors propagate through the rendered video for some time.

5. Conclusions and future work

In this paper we present some evaluation results for the performance of UMTS for different traffic types including MPEG-4 video traffic, SMTP and HTTP traffic. In the presented simulations, we use both TCP and UDP as the transport protocols to the system. This paper proves that the scheme of real time video transmission that coexists in the same channel with Internet TCP traffic is applicable in UMTS. However some problems occur when the total bit rate of the traffic is reaching the downlink bit rate of the transport channel. As a consequence, some packets are lost. The seriousness of the situation depends on the number of lost UDP packets and their time variance. To these situations it is necessary for the sender to adapt the transmission rate based on the current network conditions and this will be the step that follows this work.

Furthermore, among the future steps is to evaluate the performance of MPEG-4 video transmissions over High Speed Downlink Packet Access transmissions. HSDPA supports the introduction of high bit rate data services and will increase network capacity, while minimizing operators' investment.

References

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