Rate and Loss Control for Video Transmission over UMTS using Real-Time Protocols

Antonios G. Alexiou, Christos J. Bouras
Research Academic Computer Technology Institute, Greece and
Computer Engineering and Informatics Dept., Univ. of Patras, Greece
{alexiua, bouras}@cti.gr

Abstract

UMTS is evolving toward a future wireless all-IP network. In this paper we present how it supports real-time IP multimedia services, as these services are expected to drive the adoption of wireless all-IP networks. The scheme of real time streaming video is one of the newcomers in wireless data communication and in particular in UMTS, raising a number of novel requirements in both telecommunication and data communication systems. This scheme applies when the mobile user is experiencing real time video content. In this work we focus on the design and implementation of a rate and loss control mechanism for monitoring the UMTS network state and estimating the appropriate rate of the streaming video data.

1. Introduction

As communications technology develops, user demand for multimedia services is rising. Meanwhile, the Internet has enjoyed tremendous growth in recent years. Consequently, there is a great interest in using the IP-based networks to provide multimedia services. One of the most important areas in which the issues are being debated is in standards development for the Universal Mobile Communication System (UMTS) [1].

Bandwidth is the most precious and limited resource in UMTS. Therefore, it is of extreme importance to exploit this resource in the most efficient way. It is essential for a wireless network to have an efficient bandwidth allocation algorithm, in order the mobile user to experience both real time video applications and Internet applications, such as HTTP or SMTP. Therefore, when a user is experiencing a streaming video, there should be enough bandwidth available at any time for any other application that the mobile user might experience [2]. In addition, when two different applications are running together, the network should guarantee that there is no possibility for any of the above-mentioned applications to prevail against the other by taking all the available channel bandwidth.

A TCP friendly rate control mechanism is presented in [3]. The authors propose a widely accepted rate control method in wired networks, which is the equation based rate control also known as TFRC. Recent work [4] provides an overview of MPEG-4 video transmission over wireless networks. The results demonstrate that video quality can be substantially improved be preserving the high priority video data during the transmission.

Although these adaptive streaming techniques show noticeable benefits in streaming video over heterogeneous networks, they require significant modifications in the streaming client, the streaming server, or both. Therefore, the goal of this work is to provide a rate and loss control scheme with special focus to wireless video streaming. This scheme presents a friendly behavior to any TCP or UDP data flow that coexists with the streaming video in the same UMTS transport channel.

This paper is structured as follows. In Section 2 the architecture of the rate and loss control scheme is briefly described while Section 3 presents the operation of the scheme. Following this, Section 4 reviews the main features of the simulation model while section 5 is dedicated to the experiments' results. Finally, some concluding remarks and planned next steps are briefly described.

2. Architecture of the rate and loss control mechanism

The architecture that we propose for the implementation of the Rate and Loss Control

Mechanism is based on the Decision Feedback Scheme presented in [7]. The architecture of the scheme is depicted in Figure 1.

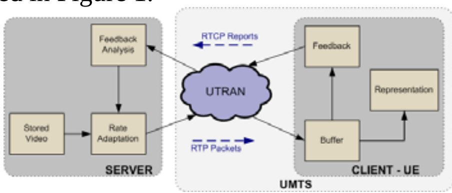

Figure 1. Rate and loss control scheme

The scheme consists of the server and the client modules [7]. As far as the server module is concerned, it is divided in the following blocks: Feedback Analysis Block, Stored Video Block and Rate Adaptation Block. The client module (mobile user) of the scheme consists of the following blocks: Buffer Block, Feedback Block and Representation Block.

The basic modifications performed in the Decision Feedback Scheme are related with the rate adaptation block on the server side as well as the feedback block of the client side. In particular, concerning the rate adaptation block, we use RTP instead of UDP as the transport protocol for the multimedia data. Furthermore, the use of RTCP instead of TCP for the transmission of control information is the only modification that we made in the feedback block.

3. Feedback analysis and decision of the transmission rate

In this section we present in detail the operation of the rate and loss control scheme. The mobile user in recurrent time space sends RTCP reports to the media server, which contains information about the current throughput in the wireless link during the transmission of the multimedia data between the server and the mobile user. The Feedback analysis block extracts the throughput from the RTCP report and passes it through an appropriate filter (Figure 1). The use of filter is essential for the operation of the mechanism in order to avoid wrong estimations of the network conditions.

Similar to the Decision Feedback Scheme [7], we define an upper bound for the bandwidth that a streaming application, which coexists with any other application in the same channel, can reserve from the available bandwidth of the transport channel. This condition restricts the video transmission to use no more than a% of the maximum available bandwidth of the transport channel. As a consequence, the other flows in the channel can use the (100-a)% of the bandwidth of the transport channel. If the bandwidth of streaming application reaches the upper bound or the total bandwidth of the transmitted applications in the channel reach the available bandwidth of the transport channel, then the network should adjust the transmission rate of the video sequence to a lower value in order to avoid congestion. In addition, when the total throughput of the non streaming flows in the wireless channel is not exceed the (100-a)% of the maximum bandwidth of the transport channel, then the network could adjust the transmission rate of the video sequence to a higher value in order to improve the representation quality in the terminal of the mobile user.

The measurements of the bandwidth that any application reserves and the throughput of any application transmitted in the transport channel are based on real time information exchanged between the mobile user and the server as it is depicted in Figure 1.

4. Simulation model

This Section reviews the main features of the simulation model that has been implemented by using the ns -2 simulator [5]. What is examined in this paper is the performance of UMTS Dedicated Channels (DCHs) for real time adaptive video transmission. During the simulations we make the following measurements:

- Delay in RAN (Radio Access Network)
- Delay Jitter
- Throughput in Wireless Link
- RTCP Packet Delay

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 2. The model is based on the UMTS system architecture. 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 2. In our simulations, we use a 384Kbit-DCH in the downlink and a 64Kbit-DCH in the uplink direction. The TTIs are 10ms and 20ms in the down- and uplink direction, respectively.

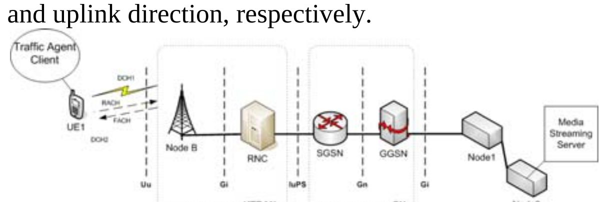

Figure 2. Simulation model

As far as the video sequence is concerned, it is encoded to ITU-H.263 in QCIF format (176x144 pixels) at the PAL frame rate of 25 frames per second. The video traces we use, are taken from [6].

In order to exploit the performance of DCHs for video transmissions and analyze the behavior of the rate and loss control scheme we present a number of experiments. During these experiments we transmit adaptive video and TCP traffic over the same DCH.

5. Experiments

This section is dedicated to describing the results in terms of performance of UMTS DCHs and behavior of the rate and loss control scheme. In the experiment, we transmit at the same time a video application and TCP traffic. In the simulation, we follow the following scenario: Initially, we transmit video with target bit rate 256 Kbps and Constant Bit Rate TCP traffic with rate 30Kbps through the dedicated channel. 40 seconds after the beginning of the simulation, we increase the rate of the TCP traffic to 140Kbps for the following 40 seconds. 80 seconds after the beginning of the video transmission we decrease the rate of the TCP traffic to 30 kbps. The duration of the simulation is 200 seconds.

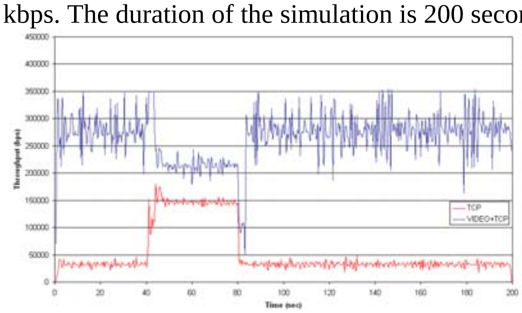

Figure 3. Total throughput in the wireless link

Figure 3 displays the total throughput in the wireless link. The y-axis presents the throughput in bps while x-axis represents the duration of the simulation. The blue (navy) line represents the total throughput in the wireless link (video+tcp) while the red line presents the throughput of the tcp traffic.

During this experiment parameter a is set to 66,6% and the downlink bit rate of the DCH is 384Kbps. This condition restricts the video bit rate to be no more than 256Kbps while the other flows in the channel can be transmitted with bit rate no more than 128Kbps. Consequently, the Video to TCP traffic ratio should not be greater than the critical value of 2:1.

Initially, the rate of the video sequence is 256 Kbps and the rate of the tcp traffic is 30Kbps. For the next 40 seconds the total throughput in the wireless channel increases but no problems occurred in the system since the Video to TCP traffic ratio is 256:30. 40 seconds after the beginning of the video transmission, we increase the rate of the tcp traffic to 140 Kbps. At this time, congestion occurs to the network, since the total throughput in the wireless channel converges to the maximum bit rate of the DCH and the Video to TCP traffic ratio converges to the critical value of 2:1. As a consequence the rate and loss control scheme, in order to continue serving the TCP traffic, decides to reduce the transmission rate of the video sequence to 64 Kbps (Figure 3). Until the second 80, when the bit rate of the TCP traffic reduces to 30 Kbps, the transport channel serves two flows with Video to TCP ratio 64:140. 80 seconds after the beginning of the simulation, the rate and loss control scheme decides to increase the transmission rate of the video sequence since the bit rate of the TCP traffic is very low and the transport channel has enough available bandwidth to serve the two flows.

The operation of the rate and loss control mechanism can be clearly seen in Figure 4 that presents the load factor of the dedicated channel. The load factor represents the utilization of the wireless channel for each mobile client. The load factor is computed from the following equation:

$$\eta_{DCH} = \frac{T_{current}}{T_{\max}},$$

where T represents the throughput in the channel and $0 \le \eta_{DCH} \le 1$.

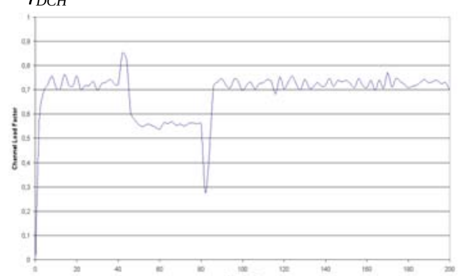

Figure 4. Load factor of the dedicated channel

Figure 4 shows that at the first 40 seconds of the simulation, the rate and loss control scheme characterizes the network as loaded and no change happens in the transmission rate of the multimedia data. Around second 40 we can observe an increase to the link’s utilization from 70% to 87% due to an increase in the transmission rate of the TCP traffic. At this time, the rate and loss control scheme realizes that the network is in condition congestion and decides to reduce the utilization of the channel. In addition, at second 80, when the utilization of the wireless channel significantly decreases from 59% to 29% the network is characterized as unloaded and the rate and loss control scheme decides to improve the quality of the representation of the video sequence at the terminal of the mobile user by increasing the transmission rate of the video data.

From Figure 3 and Figure 4 it is obvious that the rate and loss control mechanism operates efficiently in mixed traffic conditions. Furthermore with the use of this scheme, the video transmission over a wireless channel shows a friendly behavior to any other TCP flow that coexists in the transport channel.

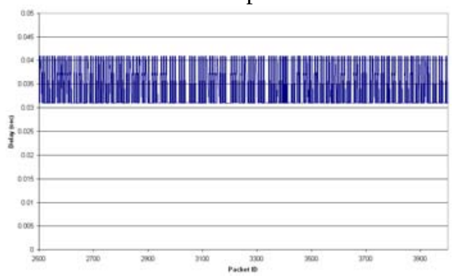

Figure 5. Delay in RAN for the Video stream

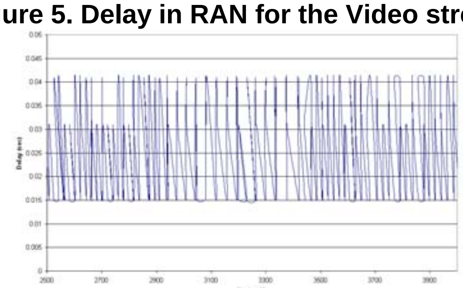

Figure 6. Delay in RAN for the TCP traffic

Figure 5 and Figure 6 present the delay in RAN for the video and TCP traffic respectively. An obvious observation that comes out from the above figures is that the TCP packets have delay variation greater than the packets of the video sequence.

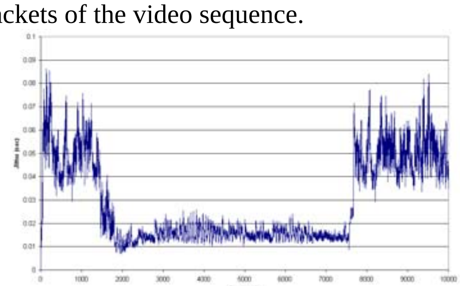

Figure 7. Delay jitter of the TCP packets

Figure 7 presents the delay jitter of the packets of the TCP stream. We can observe that around packet ID 2000 the delay jitter of the TCP packets is decreased significantly. At this time, the rate and loss control mechanism decides reduce the transmission rate of the video sequence in order to compensate the rate increment of the TCP traffic. Furthermore, around packet ID 7500 there is an increase in the jitter of the TCP packets since the total load in the wireless channel is increased.

From Figure 6 and Figure 7 it is obvious that the packets of the video sequence are characterized by low delay and low jitter compared to that of the TCP traffic.

The delay of the RTCP packets is of extreme importance for any rate control mechanism due to the reason that a lost or a delayed RTCP report could result in wrong estimations of the current conditions in the UMTS network. In our experiments this delay is approximately 96msecs.

6. Conclusions and future work

In this paper we present the architecture and the implementation of a rate and loss control scheme that it used for transmission of multimedia streaming over UMTS transport channels. Through a number of experiments we conclude that the rate and loss control scheme performs efficiently in mixed traffic conditions. Furthermore, with the use of this scheme, the video transmission over UMTS transport channels shows a friendly behavior to any other TCP flow that coexists in the transport channel. Furthermore, we present an overview of video transmission over UMTS using real time protocols such as RTP/RTCP.

The step that follows this work is to evaluate the performance of video transmission 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.

7. References

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