A Decision Feedback Scheme for Multimedia Transmission Over 3G Mobile Networks

A. G. Alexiou [1, 2], C. J. Bouras [1, 2], V. G. Igglesis [1, 2]
[1] *Research Academic Computer Technology Institute, Greece*
[2] *Computer Engineering and Informatics Dept., Univ. of Patras, Greece*
Riga Feraiou 61, Patras, 26221 Greece

***Abstract*— UMTS, the successor of GSM, 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 multimedia content. In this paper, we describe the design and the implementation of a decision feedback scheme, which is based on real time protocols and it is used for streaming video over UMTS transport channels. The decision feedback scheme uses a friendly, to the mobile users, congestion control mechanism to control the transmission rate of the real time video sequence. Furthermore, we evaluate this mechanism through a number of experiments. During these experiments we demonstrate the friendly behavior of the decision feedback scheme to TCP and UDP data streams that coexist in the same channel with the video sequence.**

***Index Terms*— Adaptive Multimedia Transmission, Multimedia over UMTS, H.263, Streaming Media in Wireless Networks**

I. Introduction

THE POPULARITY of IP-based video streaming over the Internet is continuously growing, with hundreds of new subscribers daily registered. In addition, existing and emerging wireless systems, such as UMTS, enable IP-based multimedia transmission and reception at any place and time at reasonable and sufficient data rates. Video transmission for mobile terminals is likely to be a major application in future mobile systems and emerges as a key factor for their success [2].

Bandwidth is the most precious and limited resource both for UMTS and every wireless network, in general. Therefore, it is of extreme importance to exploit this resource in the most efficient way. Video applications produce large amount of data. Therefore, video is transmitted in compressed format for reducing the generated data rates. Among the most commonly used compression techniques, ITU H.263 is the standard that has recently gained a considerable attention [7], [9].

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 [1]. 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. Taking into consideration that Internet applications adopt mainly TCP, as the transport protocol, while streaming applications are mainly using UDP, the network should guarantee that UDP does not prevail against the TCP traffic and, in particular, there should be enough bandwidth available in the wireless channel for the Internet applications to run properly.

In this paper, we focus on the implementation of a mechanism for monitoring the network state of the UMTS air interface and estimating the appropriate transmission rate of the multimedia data. The mechanism, called Decision Feedback Scheme, uses a friendly, to the mobile users, congestion control scheme to control the transmission rate of the real time video sequence. Furthermore, in order to exploit the friendliness of the Decision Feedback Scheme we setup a number of experiments. During these experiments we investigate the behavior of the mechanism for TCP and UDP data streams that coexist with real time streaming video in the same transport channel.

An, in depth, overview of streaming video over variable bit-rate wireless channels is presented in [2]. In [3] the authors suggest an adaptive streaming architecture, which is based on real time protocols. The proposed mechanism is used for unicast and multicast multimedia data transmission over heterogeneous networks, like the Internet. Adaptive media playout [4] is a new technique that allows a streaming media client, without the involvement of the server, to control the rate at which data is consumed by the sampling process. Another technology for a streaming media system is proposed, which defines how it will allocate transmission resources among packets. Recent work [5] provides a flexible framework to allow rate-distortion optimized (RaDio) packet scheduling. In this case, the system allocates time and bandwidth resources by adapting to the varying channel conditions. Finally, it is shown that this RaDio transmission can be supported if media streams are pre-encoded with appropriate packet dependencies, possibly adapted to the channel (channel-adaptive packet dependency control) [6].

Although these channel-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, in this work we focus on solutions to stream video over UMTS transport channels, which only require insignificant modifications in the streaming server, but provide a certain guaranteed QoS. The goal of this work is to provide a decision feedback 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. Any application that transmits data over a network should present a friendly behavior towards the other flows that coexist in the network and especially towards the TCP flows that comprise the majority of flows in today's networks.

This paper is structured as follows. In section 2 the architecture of the Decision Feedback Scheme is briefly described, while section 3 presents the mechanism that adjusts the transmission rate of the video sequence. 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.

II. Architecture of the Decision Feedback Scheme (DFS)

The architecture that we propose for the implementation of the Decision Feedback Scheme is based on the client – server model. The architecture of the scheme is depicted in Fig. 1.

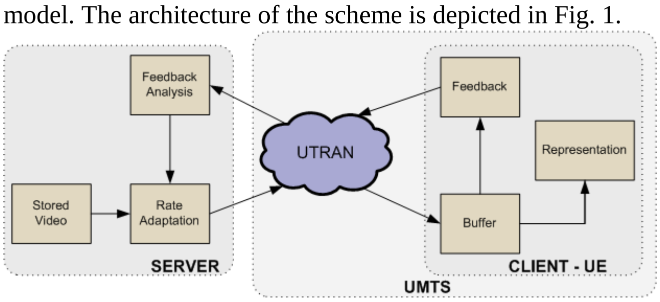

Fig. 1. Architecture of the Decision Feedback Scheme

The scheme consists of the server and the client modules. As far as it concerns the server module, it is divided in the following blocks:

1) *Feedback Analysis*: This block is responsible for the analysis of the feedback information received from the client. It analyzes the received information from the network and adapts the transmission rate of the video sequence according to current network conditions.
2) *Stored Video*: The Video block consists of a set of hard disks in which the video sequence is stored. In our experiments the video sequence follows the ITU H.263 standard, which has recently gained a considerable attention, especially in wireless networks such as UMTS.
3) *Rate Adaptation*: This block adjusts the rate of the transmitted video according to the received information from the Feedback Analysis Block.

The client module (mobile user) of the scheme consists of the following blocks:

1) *Buffer*: The use of the buffer in the client side is very important. The client stores the received data from the network before their representation to the terminal of the mobile user. It is important to mention that the representation of the video to the terminal of the mobile user starts only when the necessary amount of data is stored to the client's buffer.
2) *Feedback*: This block is responsible for the monitoring of the representation of the video sequence and the transmission of the rate measurements that the end user performs to the server.
3) *Representation*: It is responsible for the representation of the video sequence to the terminal of the mobile user.

Many existing wired and/or wireless networks cannot provide guaranteed QoS, either because of congestion, or because temporally high bit error rates cannot be avoided during fading periods. Due to the requirements for constrained latency and the extreme error sensitivity of compressed video streams, an effective multimedia networking system should incorporate strong interaction between the underlying network and specific multimedia applications. On one hand, the network must be able to accommodate the needs of the application and, on the other hand, the application must be aware of the prevailing network conditions. Reliable transport protocols, like the Transmission Control Protocol (TCP) with Internet Protocol (IP) as the networking protocol, recover from packet loss by using acknowledgements and retransmissions. However, the resulting latency is generally too large for real-time multimedia applications where the late arriving packets are considered as lost. Typically, multimedia applications adopt the User Datagram Protocol (UDP), which provides an unreliable packet delivery service. In UDP, the application layer takes responsibility for ensuring data integrity by arranging packets in the correct order, handling lost packets as well as applying flow control.

Furthermore, real-time video applications require all packets to arrive in a timely manner. If packets are lost, then the synchronization between encoder and decoder is broken (disturbed), and errors propagate through the rendered video. If packets are excessively delayed, they become useless to the decoder and are treated as lost.

In our approach we use UDP for the transmission of the multimedia data and TCP for the transmission of control information. UDP is often used for transmission of multimedia applications since error correction and retransmission convey unacceptable time delays in the video sequence.

H.263 is a low bit rate video-coding standard aimed at providing interactive video for Internet and mobile connections. The compressed video bitstream obtained via H.263 and similar codec's presents some characteristics, which are crucial for the efficient design of wireless mobile video systems. Among them are the variability of its rate, the uneven effect of degradations (errors or losses) on different parts of the stream, and the sensitivity to delay [7].

III. Feedback Analysis and Decision of the Transmission Rate

In this section we present, in detail, the feedback analysis block that is depicted in Fig. 1. The feedback analysis block is responsible for analyzing the feedback information that the mobile user sends to the server concerning the transmission quality of the multimedia data. Bandwidth is the most precious and limited resource of UMTS and every wireless network. Therefore, it is of extreme importance to exploit this resource in the most efficient way. Video applications produce large amount of data. Considering the available bandwidth that a mobile user can reserve in UMTS (for dedicated channels it reaches approximately 2Mbit), the presentation quality of the video sequence depends on the packet loss rate and the delay variation. Another crucial parameter that affects the video quality is the presence or not of other flows that coexist in the same channel with the video streaming.

In our approach 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. For example, considering a dedicated channel with downlink bit rate 384 Kbps and two different applications that coexist in the channel (a streaming video and a TCP flow), the streaming video bit rate should not be greater than **384*a**%. Parameter **a** depends mainly on the mobile user's needs and secondly on the current network conditions. When the priority is given to the video sequence the parameter **a** converges to 100%. In addition, when the priority is given to the other flows, parameter **a** converges to zero. 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 Fig. 1.

The pseudo code of the decision feedback scheme is described in TABLE I.

The mobile user in recurrent time space sends TCP reports to the media server, which contain 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 module extracts the throughput from the TCP report and passes it through an appropriate filter. The use of filter is essential for the operation of the Decision Feedback Scheme in order to avoid wrong estimations of the network conditions.

TABLE I
PSEUDO CODE OF THE DECISION FEEDBACK SCHEME

```
Define a%;
Assign (System Available Bandwidth * a%) to the streaming multimedia application;
IF [New Requested Bandwidth Bi < (100-a)* System Available Bandwidth]
   Assign Bi;
ELSE
   Adjust transmission rate of the video sequence to a lower value;
   Assign Bi;
IF [New Requested Bandwidth Bj < (100-a)* System Available Bandwidth-Bi]
   Assign Bj;
ELSE
   Adjust transmission rate of the video sequence to a lower value;
   Assign Bj;
WHILE (Bi OR Bj stop)
   IF (Available Bandwidth * a%) is free
      Adjust transmission rate of the video sequence to a higher value;
   ELSE
      CONTINUE;
```

The network is characterized of three possible conditions:

- **Condition Congestion**: When the multimedia throughput reaches **a%** of the available channel bit rate or the total throughput reaches the downlink speed of the dedicated channel. In this case the transmission rate of the multimedia data should be reduced.
- **Condition Load**: In this condition the transmission quality is satisfactory and no change happens in the transmission rate of the multimedia data.
- **Condition Unload**: The throughput is small and the flows in the channel are well served. In this case the transmission rate of the multimedia data could be increased.

The filter operates as follows. If the 60% of the estimated throughput values, in a period of 5 seconds, show that the system is in congestion or is unloaded, the feedback analysis block proceeds to the decrease or increase of the multimedia transmission rate respectively.

IV. Simulation Model

This Section reviews the main features of the simulation model that has been implemented by using the ns -2 simulator [8]. In the following we assume a media streaming system setup consisting of a media streaming server, a transport channel and a streaming client. The above structure it is depicted in Fig. 2.

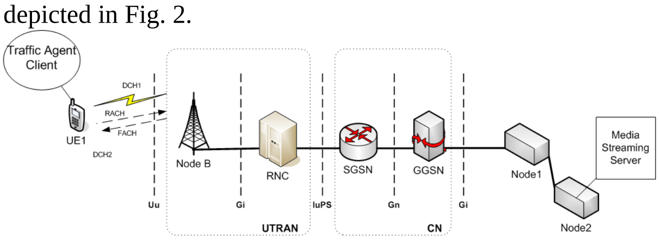

Fig. 2. The Media Streaming System (simulation model)

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 Fig. 2. In the 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 the 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.

As far as it concerns the video sequence, 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 [9].

In order to analyze the behavior of the Decision Feedback Scheme we present a number of experiments:

1) *Transmission of adaptive video and UDP traffic at the same time*: During this experiment, we investigate the behavior of Decision Feedback Scheme against UDP traffic.
2) *Transmission of adaptive video and TCP traffic at the same time*: During this experiment, we investigate the behavior of Decision Feedback Scheme against TCP traffic. TCP traffic is important because it is the traffic that produces popular applications, as the web (HTTP) and the FTP.

V. Experiments

This section is dedicated to describing the results in terms of performance and behavior of the Decision Feedback Scheme. During these experiments we investigate the behavior of the mechanism for TCP and UDP data streams that coexist with real time streaming video in the same transport channel.

A. *Transmission of streaming video and UDP traffic*

In this experiment, we transmit at the same time a video application and UDP traffic. During this experiment, we investigate the behavior of the Decision Feedback Scheme against to UDP traffic.

In the simulation, we follow the following scenario: Initially, we transmit only video with target bit rate 256 kbps through the dedicated channel. 50 seconds after the beginning of the video transmission, we transmit for the following two minutes, together with the adaptive video, UDP traffic. When the UDP traffic is completed, the transmission of the video continues for 30 more seconds. The duration of the simulation is 200 seconds. This period is divided in the following intervals:

- **Interval 1**: [0, 200sec]. H.263 video traffic generated in node 2 heading to UE1 (Fig. 2).
- **Interval 2**: [50, 170sec]. CBR UDP traffic generated in node 2 heading to UE1. The bit rate of the CBR traffic is 30kbps from [50, 75sec] and [140, 170sec] and 120kbps from [75, 140sec].

Fig. 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+udp) while the red line presents the throughput of the udp traffic.

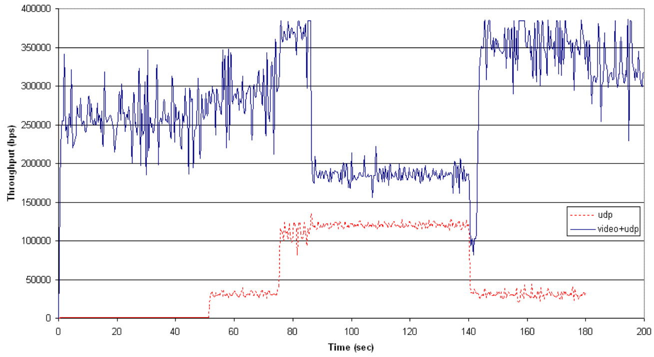

Fig. 3. Total throughput in the wireless link during the first experiment

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 transmission 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.

Initially, the transmission rate of the video sequence is 256 kbps. 50 seconds after the beginning of the video transmission, we start transmitting CBR UDP traffic with bit rate 30 kbps. For the next 25 seconds the total throughput in the wireless channel increases but no problems occurred in the system since the Video to UDP traffic ratio is 256:30. At time 75 seconds, we increase the bit rate of the UDP traffic form 30 to 120 kbps. At this time, congestion occurs to the network, since the total throughput in the wireless channel approaches the maximum bit rate of the DCH and the Video to UDP traffic ratio approaches the critical value of 2:1. As a consequence the Feedback Scheme, in order to continue serving the UDP traffic, decides to reduce the transmission rate of the video sequence to 64 kbps. Until the second 140, when the bit rate of the UDP traffic reduces to 30 kbps, the transport channel serves two flows with ratio 64:120. 140 seconds after the beginning of the simulation, the decision Feedback Scheme decides to increase the transmission rate of the video sequence since the bit rate of the UDP traffic is very low and the transport channel has enough available bandwidth to serve the two flows. Finally, in second 170 the UDP traffic stops and the only flow in the wireless channel is the video sequence.

From Fig. 3 it is obvious that the Decision Feedback Scheme 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 UDP flow that coexists in the transport channel.

B. *Transmission of streaming video and TCP traffic*

In this experiment, we transmit at the same time a video application and TCP traffic. During this experiment, we investigate the behavior of the Decision Feedback Scheme against to TCP traffic.

In the simulation, we follow the following scenario: Initially, we transmit only video with target bit rate 256 kbps through the dedicated channel. 50 seconds after the beginning of the video transmission, we transmit for the following 90 seconds, together with the adaptive video, TCP traffic. After the end of the TCP traffic the transmission of the video continues for 60 more seconds. The duration of the simulation is 200 seconds. This period is divided in the following intervals:

- **Interval 1**: [0, 200sec]. H.263 video traffic generated in node 2 heading to UE1 (Fig. 2).
- **Interval 2**: [50, 140sec]. CBR TCP traffic generated in node 2 heading to UE1. The bit rate of the CBR traffic is 30kbps from second 50 to second 75 and 120 kbps from second 75 to second 140.

Fig. 4 displays the total throughput in the wireless link. 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.

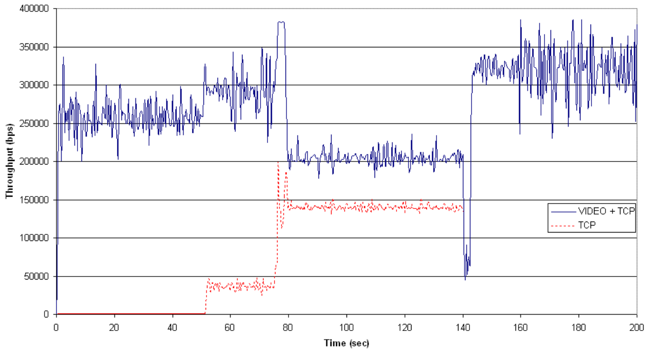

Fig. 4. Total throughput in the wireless link during the second experiment

During this experiment the parameter **a** is set to 66,6% and the downlink bit rate of the DCH is 384kbps. This condition constrains the video transmission 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.

Initially, the transmission rate of the video sequence is 256 kbps. 50 seconds after the beginning of the video transmission, we start transmitting CBR TCP traffic with bit rate 30 kbps. For the next 25 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. At time 75 seconds, we increase the bit rate of the TCP traffic form 30 to 120 kbps. At this time, congestion occurs to the network since the total throughput in the wireless channel approaching the maximum bit rate of the DCH and the Video to TCP traffic ratio approaching the critical value of 2:1. As a consequence the Feedback Scheme, in order to continue serving the TCP traffic, decides to reduce the transmission rate of the video sequence to 64 kbps. 140 seconds after the beginning of the simulation, the decision Feedback Scheme decides to increase the transmission rate of the video sequence since the TCP traffic source stops transmitting data.

From Fig. 4 it is obvious that the Decision Feedback Scheme 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.

VI. Conclusions and Future Work

In this paper we present the architecture and the implementation of a Decision Feedback Scheme that it used for transmission of multimedia streaming over UMTS transport channels. During the design of this scheme we are concentrating to the implementation of a mechanism for monitoring the network condition and estimating the appropriate rate for the transmission of the multimedia data.

Through a number of experiments we conclude that the Decision Feedback 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 or UDP flow that coexists in the transport channel.

The step that follows this work is the use of the protocol RTP/RTCP on top of UDP for the transmission of the video sequence. RTCP provides feedback to applications about the transmission quality. RTCP uses sender reports and receiver reports, which contain useful statistical information like total transmitted packets, packet loss rate and delay jitter during the transmission of the data. This statistical information is very useful, because it can be used for the implementation of congestion control mechanisms improving the performance of the feedback analysis block.

References

[1] D. Wong and V. Varma, “Supporting Real-Time IP Multicast Services in UMTS”, IEEE Communications Magazine, pp. 148-155, November 2003

[2] T. Stockhammer, H. Jenkac and G. Kuhn “Streaming Video Over Variable Bit-Rate Wireless Channels”, IEEE TRANSACTIONS ON MULTIMEDIA, VOL. 6, NO. 2, APRIL 2004.

[3] C. Bouras and A. Gkamas, “Multimedia transmission with adaptive QoS based on real-time protocols”, INTERNATIONAL JOURNAL OF COMMUNICATION SYSTEMS, vol. 16, pp 225-248, 2003 John Wiley and Sons.

[4] E. G. Steinbach, N. Färber, and B. Girod, “Adaptive play-out for low latency video streaming ,” in Proc. Int. Conf. Image Processing (ICIP-01), Thessaloniki, Greece, Oct. 2001.

[5] P. A. Chou and Z. Miao, “Rate-distortion optimized streaming of packetized media,” Microsoft Research Tech. Rep. MSR-TR-2001–35, Feb. 2001.

[6] Y. J. Liang and B. Girod, “Rate-distortion optimized low-latency video streaming using channel-adaptive bitstream assembly ,” in Proc. IEEE Int. Conf. Multimedia and Expo (ICME-2002), Lausanne, Switzerland, Aug. 2002.

[7] C. Iskander and P. Mathiopoulos, Online Smoothing of VBR H.263 Video for the CDMA200 akd IS-95B Uplinks, IEEE Transactions on Multimedia, Vol. 6, No. 4, August 2004

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

[9] F. Fitzek, and M. Reisslein, “MPEG-4 and H.263 Video Traces for Network Performance Evaluation”, IEEE Network, Nov.-Dec. 2001, pp. 40-54