University of Cyprus
Department of Computer Science
Networks Research Laboratory

Adaptive Methods for the Transmission of Video Streams in Wireless Networks

Final Progress Report

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1. Title of Research Project:

Adaptive Methods for the Transmission of Video Streams in Wireless Networks (ADAVIDEO)
Website: *http://www2.cs.ucy.ac.cy/~csp5pa1/ADAVIDEO/index.htm*

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3. Department or Research Unit of Principal Investigator:

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4. Members of Research Team *(Research Assistants, Special Scientists, External Collaborators, students)*:

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5. Duration:

24 months

6. Budget:

£30.000
The financial report of the ADAVIDEO Project is shown in APPENDIX B.

7. Starting Date:

November 2004

8. Executive Summary

The transfer of information through the Internet is becoming a promising business model. In addition, new Internet applications are emerging which require the transfer of multimedia content in real time (i.e. stored or streaming video and voice content). These applications, referred in this proposal as Real Time Multimedia (RTM) applications, differ from typical applications mainly with respect to their requirements in bandwidth and memory. In addition they set strict timing requirements since they demand a continuous flow of data towards the receiver once the decoding has started. This continuous periodic flow of data is creating a lot of demands in the network infrastructure linking the server and the client.

The latest developments of high bandwidth wireless technologies, such as the third generation (3G) of wireless cellular networks (i.e. UMTS) or Wireless LANs based on different versions of IEEE 802.11 (e.g. WiFi), in connection with the abundance of mobile devices with high processing power (Laptops, PDAs and mobile phones), is making the study of the transmission of RTM applications (video or audio streams) over wireless networks even more interesting.

This work investigates techniques for the efficient support of video streams for wireless users connected to the Internet in the cases where the content server is at the wired part of the network. In such environments, the time-sensitive information is experiencing not only the problems created by congestion in the core network routers, but also the problems stemming from the nature of the wireless connection. Such problems include the high bit error rates, the dramatic variations of the available bandwidth, and the delay and delay variation due to the volatile characteristics of the wireless environment.

The proposed research focuses on: (1) **Network Adaptation Techniques** (NAT) that involve the end-to-end adaptation of network parameters to the needs of the RTM application using algorithms which take into account the state/load of the network (congestion and flow rate control in wireless access network), and (2) **Content Adaptive Techniques** (CAT) that involve the adaptation of content to the desirable rate without the need for re-encoding or regenerating (transmission of information in multiple layers).

The proposed research is expected to enable a range of commercial, entertainment, and educational applications and actions which require the transmission of real-time multimedia over a wireless network.

9. Research Results

9.1 Introduction

Compressed video streams (like MPEG) exhibit large variations in their data rates something which makes their management in a packet-based best-effort network like IP extremely difficult. The problem is extenuated when we consider mobile users connecting with wireless terminals. Because of the structure of compressed video where frames are strongly depending on previous information, failure in the synchronization of the sender with the receiver may cause a series of side effects which is possible to destroy content for a sequence of frames.

Thus, applications of real-time video streaming in heterogeneous networks and computing environments like the Internet need to implement highly scalable and adaptive techniques in terms of content encoding and transmission rates. Taking all these into consideration it is apparent that designing adaptive mechanisms for Internet video transmission poses many challenges. Under these circumstances a combination of Content Adaptation Techniques and Network Adaptation Techniques is an imperative need.

The use of feedback in the estimation of the maximum possible transmission rate for congestion avoidance has been presented in a number of research works, none of which deals with the transmission of RTM applications using the RTP/UDP/IP protocols. Latest research in the area includes a treatment of the transmission problems in wireless IP-based networks adopting very simplified architectures.

9.2 Problem Statement

Needless to say that video transmission over the wireless networks is considered to be the prime candidate for being the next killer application of the Internet. The overwhelming majority of today's mobile devices like mobile phones, PDAs and laptops are capable of reproducing video streams either through 3G-enabled networks like UMTS or over the Internet using dedicated protocols like 802.11. In this report we are going to focus primarily on video transmission over the Internet.

One of the most significant problems that video communications face is the unpredictable nature of the heterogeneous networks like the Internet primarily in terms of bandwidth, latency and loss variation. It is beyond any doubt that video transmission applications need to implement highly scalable and adaptive techniques in terms of content encoding and transmission rates in order to cope with erroneous and time variant conditions of such networks.

For these reasons some adaptive schemes have to be introduced from both the network as well as content encoding perspective. Content Adaptive Techniques (CATs) deal with adaptation of content to the desirable transmission rate using primarily scalable video approaches whereas Network Adaptation Techniques (NATs) deal with the end-to-end adaptation of network parameters to the needs of a real time multimedia application using algorithms which take into account the state and/or load of the network and the type of errors.

Content Adaptation Techniques (CATs) are methods commonly used for the adaptation of content to the desirable rate. Recent studies on CATs reveal that the transmission of video streams in multiple layers is feasible without the need for re-encoding or regeneration of the content.

The basic requirements of network adaptation techniques are (1) to provide as accurate information as possible for the network load, (2) to distinguish between errors originating from congestion at the core routers and errors stemming from the wireless medium, and (3) to properly adapt the transmission rate. The qualitative differentiation of errors is required so that we reduce the transmission rate only when it is justifiable (e.g. when we have heavy core congestion). In order for the estimations of network load and the type and quality of errors to be precise we need proper feedback from the receiver end of the stream.

9.3 Proposed Mechanisms

In this project we propose some novel algorithms for the increase of the objective quality of the transmitted video.

The two proposed adaptive methods of estimating the possible transmission rate, namely ADIVIS and RAF are based on feedback by the wireless user. As can be seen in [5] and [4], the estimation of objective quality is made possible with the right feedback since the objective quality has a direct relationship with the nature of errors and the actual information content.

In addition, an adaptive congestion protocol, namely ACP [8], [9], is proposed in order to address the problem of congestion in future high speed networks.

Furthermore, a new queue length based Internet congestion control protocol [10], is proposed to regulate the queue size at each link so that it tracks a reference queue size chosen by the designer.

Both protocols (ACP, queue length based) exhibit nice transient properties such as smooth responses with no oscillations and fast convergence something that make them capable of transporting real time data such as video streams over the Internet.

A short description of each one of the aforementioned protocols is given in APPENDIX A.

10. Dissemination

Some of the research results obtained by this project were disseminated through internationally recognized peer-reviewed publications.
Five papers [1], [2], [3], [9] and [10] were presented in four different widely known and respectful conferences. The first paper [1] deals with Requirements for the Transmission of Streaming Video in Mobile Wireless Networks and it was presented in the International Conference on Artificial Neural Networks (ICANN 06) held in Athens (Greece) in September 2006. The second paper [2] deals with The Importance of Adaptive Applications in Mobile Wireless Networks and it was presented in SoftCom 2006. The third paper [3] deals with QoS Adaptation Control in Virtual Circuit Switched Mobile Networks and it was presented in the International Conference on Intelligent Systems And Computing: Theory And Applications (ISYC) 2006 held in Ayia Napa (Cyprus) in July 2006. Moreover, paper [9] which investigates an Adaptive Congestion Protocol with learning capability was presented in the same conference. Finally, the last paper [10] deals with a Queue Length Based Internet Congestion Control and it will be presented in the IEEE International Conference On Networking, Sensing and Control (IEEE ICNSC07) which will be held in London, United Kingdom in April 2007. There was another one paper [8] namely “Adaptive Congestion Protocol: A Congestion Control Protocol with Learning Capability”, which was accepted for publication in Computer Networks Journal (subject to revision). An earlier version appears in UCY Report TR-06-03 CS UCY, 7 February 2006.
Four more papers, [4], [5], [6] and [7] were also submitted ([4] was accepted and the rest are under review) to international conferences. The accepted paper [4] presents the Adaptive Feedback Algorithm for Internet Video Streaming based on Fuzzy Rate Control in more detail, emphasizing to some technical and design issues of the algorithm. This paper was accepted in the 12th IEEE Symposium on Computers and Communications (ISCC'07) that will take place in Aveiro, Portugal, in July 2007. The rest of the papers [5], [6] and [7], demonstrate further evaluation results regarding the performance of our fuzzy algorithm. To the best of your knowledge, all the aforementioned deliverables and papers as well as more information on ADAVIDEO project are made publicly available through the official website of the project:
http://www2.cs.ucy.ac.cy/~csp5pa1/ADAVIDEO/index.htm.

11. Analysis of Work Packages

11.1 WP1 – Content Adaptation Techniques

The goal of the first work package was to focus on the adaptation of information (video streams) to the required rates without regeneration (transmission of information in multiple layers) which is carried out through Content Adaptation Techniques.
The results of this work package are summarized in two deliverables:

11.1.1 [D1.1]: Objective Quality of Service Estimation Methods

Adopting the transmission of MPEG4-encoded video streams over wireless network environments, we investigated the types of errors that can be observed using objective video quality metrics such as PSNR. Furthermore we provided subjective video quality estimation based on video client-side decoded erroneous video streams we proposed some kind of correlation between objective quality and subjective quality schemes.

11.1.2 [D1.2]: Stream Concealment Techniques and Layering Techniques

Given the fact that packet loss is inevitable and therefore the presence of errors granted, our effort was directed towards limiting the effect of these errors. A number of techniques have been developed to address this issue, each one incorporating a set of characteristics. Towards this direction, we introduced the most considerable approaches of error resilience, error concealment and joint encoder-decoder error control techniques and provided a thorough discussion of the benefits and drawbacks of these error control methods. In addition, we investigated the Content Adaptation Techniques that aim to the adaptation of content to the desirable rate without the need for re-encoding or regenerating. In particular we introduced encoding of video information into multi-layered video emphasizing in the three video coding techniques SNR (Signal-to-Noise Ratio), temporal and spatial.

11.2 WP2 – Network Adaptation Techniques

The goal of the second work package was to focus on the end to end adaptation of network parameters to support the needs of the streaming applications using algorithms utilizing feedback about the network load and the type of errors encountered.
The results of this work package are summarized in two deliverables:

11.2.1 [D2.1]: Video Stream needs and Network Adaptation Techniques

In this deliverable we gave a brief overview of video stream needs as well as techniques and standard supports for video adaptation to network parameters in order to meet various application requirements and user preferences.
Furthermore, we studied several Network Adaptation Techniques which deal with the end-to-end adaptation of network parameters to the needs of a real time multimedia application using algorithms which take into account the state and/or load of the network and the type of errors.

11.2.2 [D2.2]: Feedback Algorithms for the increase of the objective quality

Initially we focused on Network Adaptation Techniques based on adaptive control schemes either sender-driven or receiver-driven while providing an in depth comparison of the aforementioned schemes.
In addition we investigated, analyzed and understood the inefficiencies that are associated with the existing adaptation techniques in order to be able to contribute to this effort. We discussed some transmission requirements for Internet video streaming as well as video compression techniques.
Taking all these into consideration, we presented and analyzed two novel approaches for adaptive video transmission. The first approach is a sender-driven technique called ADaptive feedback algorithm for Internet VIdeo Streaming based on Fuzzy Control (ADIVIS) and the second approach is called Receiver-driven Adaptive Feedback (RAF) algorithm. Particularly we considered an in depth description of the two algorithms focusing on key ideas, functionalities and the different parameters that influence their operation. Also, we designed and analysed two new algorithms for the control of congestion in high speed networks, Adaptive Congestion Protocol (ACP) and Queue Length Based Internet Congestion Control protocol.

11.3 WP3 – Simulation of Adaptation Techniques

The goal of the second work package was to simulate the proposed techniques in a combined wired and wireless network. The fully adaptive system would be tested for its ability to support real-time video streams in various realistic and extreme scenarios.
The results of this work package are summarized in two deliverables:

11.3.1 [D3.1]: Simulation Software

This deliverable provides the simulation framework for the performance evaluation of the proposed adaptive techniques for video streaming over the Internet. We focused on the development of simulation modules for Content and Network adaptation techniques and we gave an in depth description of the simulation modules that were designed for the evaluation of the algorithms.

11.3.2 [D3.2]: Results of Performance Evaluation of the Proposed Information and Network Adaptation Techniques

In this last deliverable of the ADAVIDEO project we presented and analyzed the results obtained by the simulations conducted using the models we described in the previous deliverable. Our aim was to test the ability of the proposed algorithms to support real time multimedia applications in various realistic or extreme network conditions as well as to investigate the performance of the proposed congestion control protocol in high speed networks.

References

[1] V. Vassiliou, P. Antoniou, I. Giannakou and A. Pitsillides, "Requirements for the Transmission of Streaming Video in Mobile Wireless Networks," In Proceedings of the International Conference on Artificial Neural Networks (ICANN 06), Athens, Greece, September 10-14 2006.

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[5] P. Antoniou, V. Vassiliou, and A. Pitsillides, "ADIVIS: A Novel Adaptive Algorithm for Video Streaming over the Internet," Submitted for conference publication, March 2007.

[6] P. Antoniou, V. Vassiliou, and A. Pitsillides, "Delivering Adaptive Scalable Video over the Wireless Internet," Submitted for conference publication, March 2007.

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[8] M. Lestas, A. Pitsillides, P. Ioannou and G. Hadjipollas, "Adaptive Congestion Protocol: A Congestion Control Protocol with Learning Capability," Accepted for publication in Computer Networks Journal (subject to revision). An earlier version appears in UCY Report TR-06-03 CS UCY, 7 February 2006.

[9] M. Lestas, A. Pitsillides, P. Ioannou and G. Hadjipollas, "Adaptive Congestion Protocol: A Congestion Control Protocol with Learning Capability," ISYC 2006, International Conference on Intelligent Systems and Computing: Theory and Applications, Ayia Napa, Cyprus, July 6-7, 2006, pp. 57-70.

[10] M. Lestas, A. Pitsillides, P. Ioannou and G. Hadjipollas, "Queue Length Based Internet Congestion Control," IEEE International Conference On Networking, Sensing and Control (IEEE ICNSC07), London, United Kingdom, April, 2007 (to appear).

APPENDIX A – Research Results

ADaptive feedback algorithm for Internet VIdeo Streaming based on fuzzy control (ADIVIS)

ADIVIS is an adaptive video transmission algorithm specifically designed for video streaming over the Internet. Our main objective is to provide a framework that incorporates both Content Adaptation and Network Adaptation Techniques. Towards this direction, we introduce two new components; a feedback mechanism and a decision algorithm, that deal with layered video streams.

We evaluated ADIVIS under error-free and error-prone environments and our preliminary results indicate that the algorithm can finely adapt the video stream bit rate to the available bandwidth, while providing high and stable objective quality of service at the same time. Moreover, simulations showed that ADIVIS performs best in the absence of background traffic like FTP but the objective quality remains acceptable in the presence of background FTP as well. Additionally, it seems that our algorithm provides fairness; however, this is an issue which will be further investigated in the presence of multiple concurrent users.

For future work we want to determine the sensitivity of our algorithm to various parameters (i.e time hysteresis, decision period T). To continue with further evaluation of our adaptation approach, we need to look at the interaction between our adaptive flow and other network flows sharing the same routers. In addition, the effect of delay variation (jitter) will be taken into consideration when designing the fuzzy inference engine. Moreover, subjective tests should be considered given the fact that PSNR is inappropriate for the evaluation of the actual user perceived quality of service because it is poorly correlated to human vision.

Receiver-driven Adaptive Feedback algorithm (RAF)

RAF algorithm is an adaptive algorithm whose operation is based on user requirements as well as on the dynamically changing conditions of the network path. Adaptation of the content is based on the values of some pre-defined, but dynamically adjusted, parameters. The adjustment of these parameters is done prior the establishment of the connection. Some of the parameters are taken from the choices made by a user. The algorithm finds the better combination among the different parameters which maximizes the user perceived quality according to the transient conditions of the network.

One of the most basic operations of the algorithm is that it manages to correlate the objective quality of service with the user perceived quality (MOS). This functionality is embedded on a table that combines the MOS with the different combinations of the parameters. Thus the algorithm can deduce how the user perceives the quality of the received video stream according to the values of the parameters shown in this table.

Moreover RAF algorithm can maintain high user perceived quality under extreme network conditions with high packet loss percentage by calibrating the value of frames per second.

Adaptive Congestion Protocol (ACP)

There is strong evidence that the current implementation of TCP will perform poorly in future high speed networks. To address this problem many congestion control protocols have been proposed in literature which, however, fail to satisfy key design requirements of congestion control protocols, as these are outlined in the paper. In this work we develop an Adaptive Congestion Protocol (ACP) which is shown to satisfy all the design requirements and thus outperform previous proposals. Extensive simulations indicate that the protocol is able to guide the network to a stable equilibrium which is characterized by max-min fairness, high utilization, small queue sizes and no observable packet drops. In addition, it is found to be scalable with respect to changing bandwidths, delays and number of users utilizing the network. The protocol also exhibits nice transient properties such as smooth responses with no oscillations and fast convergence. ACP does not require maintenance of per flow states within the network and utilizes an explicit multi-bit feedback signalling scheme. To maintain stability it implements at each link a novel estimation algorithm which estimates the number of users utilizing the network. Using a simple network model, we show analytically the effectiveness of the estimation algorithm. We use the same model to generate phase portraits which demonstrate that the ACP protocol is stable for all delays.

Queue length based Internet Congestion Control

The proposed queue length based Internet congestion control protocol is shown through simulations to work effectively. The control objective is to regulate the queue size at each link so that it tracks a reference queue size chosen by the designer. To achieve the latter, the protocol implements at each link a certainty equivalent proportional controller which utilizes estimates of the effective number of users utilizing the link. These estimates are generated online using a novel estimation algorithm which is based on online parameter identification techniques. The protocol utilizes an explicit multibit feedback scheme and does not require maintenance of per

flow states within the network. Extensive simulations indicate that the protocol is able to guide the network to a stable equilibrium which is characterized by max-min fairness, high utilization, queue sizes close to the reference value and no observable packet drops. In addition, it is found to be scalable with respect to changing bandwidths, delays and number of users utilizing the network. The protocol also exhibits nice transient properties such as smooth responses with no oscillations and fast convergence.

APPENDIX B – Financial Report