CROSS-LAYER OPTIMIZATION FOR STREAMING MPEG4 VIDEO OVER HSDPA NETWORKS

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Abstract: A novel cross layer optimization technique for efficient streaming of MPEG4 VIDEO over a High Speed Downlink Packet Access (HSDPA) network is proposed in this paper. The proposed technique uses the types of frames produced by the MPEG encoder to optimize the performance of the Hybrid Automatic Repeat reQuest (H-ARQ) protocol at the MAC layer. Our aim is to reduce the total power at the NodeB, and to increase the overall system throughput, while maintaining satisfactory user-perceived Quality of Service (QoS). The proposed technique consists in applying ARQ retransmission for MPEG4 I-frames (the most critical frame of an MPEG4 stream) upon the reception of a negative acknowledgment (NACK) message from the receiver (UE). Packet combining is then performed with the aid of the available I-frames at the receiver side. Different packet combining strategies have been investigated to assess the performance of the proposed cross-layer technique. We show that compared to the blind HARQ Chase Combining scheme applied indiscriminately to all MPEG4 frames, our scheme allows for saving up to 11% of the power at the NodeB, and up to 10% of the system bandwidth, while ensuring satisfactory video quality to users.

1 INTRODUCTION

HSDPA (also known as 3.5G and as WCDMA Release 5) is a new release of UMTS networks. Its new downlink transport channel HS-DSCH (High Speed Downlink Shared Channel) provides greater capacity –up to several Mbps-, as well as increases the wireless system performance by supporting new features. Among these features are: fast link adaptation, Adaptive Modulation and Coding (AMC - changing modulation and coding format according to channel conditions), fast scheduling, and Hybrid Automatic Repeat request (HARQ). The new transport channel (HS-DSCH) uses available radio frequency resources efficiently by sharing multiple access codes, transmission power, and use of infrastructure hardware between users.

AMC provides the possibility to match modulation-coding scheme to the channel conditions for each user. The power of the retransmitted signal is kept constant over a frame interval while the modulation and coding format changes to match the current received signal quality.

Contrary to UMTS Rel'99, in HSDPA the scheduler has been moved from the RNC (Radio Network Controller) to the NodeB. Scheduling is done based on information about the channel quality, terminal capability, QoS class and power/code availability. It is fast because it is close to the air interface and a shorter frame length is used. The most famous and widely used scheduling algorithms are Round Robin (RR), Maximum Carrier-to Interference ratio (MAX-CI) and the Proportional Fairness (PF).

Moreover, HARQ handles retransmissions requested by UEs (User Equipments) due to errors in the radio packets. These requests are processed in the current WCDMA networks by the RNC while in HSDPA, they are processed at the NodeB to provide the fastest response possible. HARQ has two schemes: Chase Combining (CC) and Incremental Redundancy (IR). CC keeps the erroneous packet, and requests that the exact same packet be retransmitted. Upon receipt of this latter, it uses soft-combining to combine the erroneous and retransmitted packets to increase the possibility of successful decoding. IR, on the other hand, retransmits the same packet but differently coded. The receiver selects correctly transmitted bits from the original transmission and the retransmission to minimize the need for further repeat requests when multiple errors occur in transmitted signals.

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CC is simple to implement, while IR is more powerful but complex, eventually adding extra delay in the decoding.

Yet, despite the high data rates and the previously cited improvements offered by HSDPA over UMTS Rel'99, its shared medium represents still a challenge for the provisioning of QoS (guaranteed bandwidth, delay and jitter) for delay and/or error-sensitive applications such as MPEG4 video streaming applications. And although, the radio protocol stack at the NodeB is designed to operate under worst condition scenarios, it remains generic, and does not factor in specific application requirements (such as the differentiation in the transmission/retransmission schemes to be used for various application data/frames), yielding ineffective use of spectrum.

For achieving optimal decision, and therefore yielding efficient transmission subsystem, the different layers of the end-2-end protocol stack need to cooperate and exchange. Sharing knowledge/data types among the different protocol layers (which is the main idea behind Cross Layer Optimization - CLO) helps achieve a higher adaptability to the changing network conditions although this is violating strict layering design rules.

2 PREVIOUS WORKS

Recognising the importance of CLO when streaming MPEG4 video over wireless networks (and best effort networks in general), many researchers have looked into how the availability of application layer information across the layers up until the MAC layer can help achieve better performance. For instance, the main idea explored in (Ahmed et. al., 2003) is to add a cognitive layer able to change transport parameters, bit rates and QoS mechanisms based on the network conditions. Therein the proposed architecture takes into consideration the characteristics of MPEG4 and IP Diffserv to propose techniques for media content analysis and network control mechanisms for adaptive video streaming over IP networks. In (Zheng, 2003) Zheng studies the effect of the scheduler (MAX and PF) as well as the error detection/protection techniques (HARQ fitted mapping and 1% FER based mapping) on QoS parameters for the case of streaming MPEG4 over HSDPA. He also compares the performance of UDP and UDP-Lite for streaming MPEG4 over UMTS-like systems. In (Chen et. al., 2003), three techniques are presented to tackle the changing conditions of the wireless medium for multimedia delivery. These are: swift-OFDM, low-latency packet-awareness coder and adaptive noise filtering. Last, (Yufeng et. al., 2002) proposes a set of end-to-end application layer techniques for adaptive video streaming over wireless networks. These techniques are: Application layer packetization scheme, Class based unequal error protection and finally a Priority based ARQ scheme.

In wireless networks, video streaming applications suffer the most from delay and jitter that are introduced mainly by retransmitting erroneous or lost packets. As for errors, such applications use error concealment techniques to compensate for any erroneous video frame. Also, we know that HSDPA is known for delivering better QoS in terms of delay and jitter values, as well as for its strong retransmission strategy, namely HARQ, thanks to the new added. Still, using HARQ will result in more delay and jitter but better quality.

In this paper, we propose a new scheme based on cross layer optimization for streaming MPEG4 video over HSDPA. Our adaptive scheme uses interaction between both the link and application layers (the link layer being the one that knows about the changing network conditions and the application layer being the one that knows about the type of video frame) to take a retransmission strategy based on the type of video frame being retransmitted (I, P or B). To our best knowledge, there has not been any published research that combines HARQ retransmission strategy with the importance of the video frame being retransmitted over HSDPA.

The remaining of the paper is organized as follows: section 3 presents the technique and the underlying assumptions. Section 4 describes the simulation setup, while the results are presented in section 5.

3 PROPOSED TECHNIQUE

In this paper, we make information normally available to the application layer (which is the type of video frame) accessible to the MAC layer so that this latter makes a retransmit decision based on the type of the video frame. When the MAC-HS entity receives an erroneous frame, it checks its type before requesting a retransmission. If the frame is of type I, it requests the retransmission. However, if the frame is of any other type (P or B), it just discards it and does not request retransmission. I-frames are the ones that carry much information and that P (and B) frames depend on the previous (and following) I-frames for successful decoding. We also know that I-frames are the ones that achieve the least compression ratio while other types of frames achieve the best compression ratio. Thus, if an I-frame is lost or erroneously received and if we choose to discard it, all successive P and successive and previous B frames depending on this I-frame will fail decoding and thus we would have lost bandwidth and power by transmitting them because they will be discarded at the receiver anyway. This is why our scheme favors I-frames over the other types of frames.

Standard video streaming applications use UDP, RTP and RTCP as transport protocols. RTP runs on top of UDP, packetizes and provides in-order delivery of video frames. RTCP, when used, operates as closed loop control mechanism for informing the video source of the received video quality. During the simulations, we assume no interaction between the video client and server. Thus, RTCP is not modeled. However, those functions needed such as packetization, packet sequence numbering and in-order delivery are supported by the different tools in Evalvid (Klaue et. al., 2007). For example, packetization is implemented by the Video Sender at Evalvid.

Without loss of pertinence, we use MAX-CI as a scheduler, because it serves users with good channel quality increasing system throughput and providing better QoS. We also use CC as our HARQ scheme to minimize delays.

Also, since the primary goal of the simulation is to investigate the impact of ARQ/HARQ schemes on the quality of the MPEG4 video, we assume no packet losses, errors or congestion occurring in either the Internet or the UMTS core network. This is a fair assumption when compared to air interface generated errors. Moreover, we assume that ACKs and NACKs coming from UEs to the BS do not undergo any losses or errors. The delay introduced by the Internet and UMTS core network is kept constant and low throughout the simulation time. Each link capacity was chosen so that the radio channel in the connection bottleneck. Moreover, the functionality of GGSN and SGSN was abstracted out and modeled as traditional ns2 nodes since in general, they are wired nodes and mimic the behavior of IP routers. Last, we assume no header compression at the PDCP (Packet Data Convergence Protocol) layer.

4 EXPERIMENTAL SETTINGS

The network consists of one or several MPEG4 streaming servers that stream videos to one or more user equipments. The packets sent by the servers flow through the Core Network and the UTRAN to arrive at the Node B. This latter uses the MAC-HS entity to send the packets to the intended user equipment. For each case of number of user equipments/streaming servers (1, 5 and 10), we change the ARQ scheme and collect and analyze the data. We use three schemes: no ARQ, blind HARQ with CC, and our adaptive scheme. The streaming server streams a 10mn MPEG4 encoded video to the UE. The core network and the UTRAN links have a high data rate and a very low delay so that this part of the network does not cause any delay or loss. We would like to concentrate on the link between the NodeB and the UEs. The simulated application is H320 videophone (Halsall, 1996) with a 48 MB play-out buffer at the receiver.

The simulations were performed on a Rayleigh fading environment that conforms to the ITU-T recommendations (Recommendation ITU-R M.1225, 2000).

5 RESULTS

Figure 1 shows the overall frame loss percentage and the bandwidth delay product for the three techniques as a function of increasing load/users. One can see that CC gives the best frame loss percentage while no ARQ gives the worst. Our adaptive scheme comes in between but is closer in performance to no ARQ because of the high number of P and B frames that are discarded. As for the delay bandwidth product, we clearly see that our adaptive scheme incurs little deterioration compared to no ARQ (which is the one that would give the best results since no retransmissions take place). We also see that the gap between CC and the two other schemes gets bigger as we load the network with more users. This means that our adaptive scheme lowers the buffering requirements compared to CC that needs larger buffers due to delays introduced by retransmissions especially with high network load.

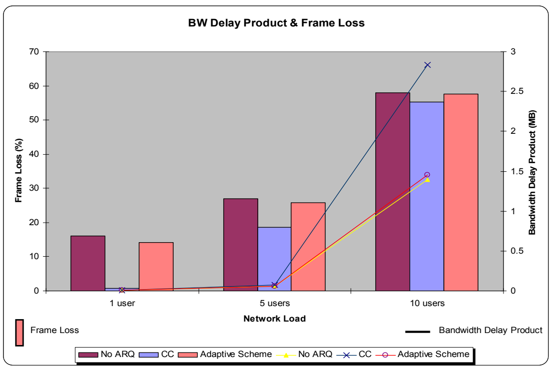

Figure 1: Performance of various HARQ techniques.

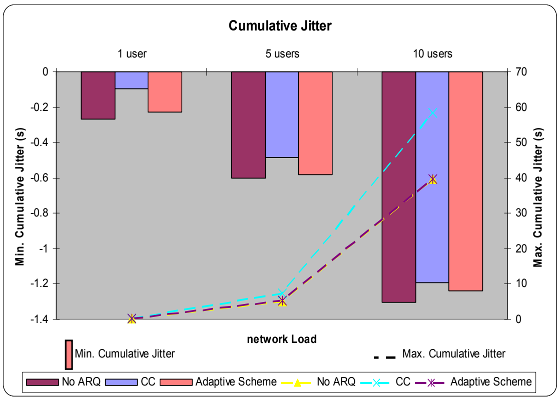

Figure 2: Performance of various HARQ techniques.

Figure 2 shows the cumulative jitter values for the three techniques and for different number of users on the network. It is clear from this graph that our adaptive scheme performs much better than CC in a way that it has a maximum cumulative jitter near the no ARQ scheme. We also notice that overloading the network does not increase the gap between our adaptive scheme and no ARQ as opposed to CC where the gap grows bigger. As for the minimum cumulative jitter, we notice that our adaptive scheme outperforms CC and gives results that are close to no ARQ. This means less variations and hence better quality of service.

We have also conducted MOS (Mean Opinion Score) for the assessment of user-perceived quality of received media under the three schemes. MOS provides a numerical indication of the perceived quality of received media and is expressed as a single number in the range 1 to 5, 1 being the lowest perceived quality, and 5 being the highest perceived quality. As shown in table 1 which summarizes MOS values for our simulations, the proposed adaptive scheme brings a clear improvement of the perceived quality with little additional use of network resources.

Table 1: MOS analysis (5 independent viewers).

Network Load / **ARQ technique**	1 user	5 users	10 users
No ARQ	3.2	2.5	0.8
Adaptive Scheme	4.2	3.2	1.5
CC	4.9	4.3	2.6

6 CONCLUSIONS

In this paper, cross-layer optimization has been used to improve the QoS of streaming MPEG4 video over the HSDPA network. We made information on the type of video frame (normally known to the application layer) available to the MAC-HS layer so that this latter retransmit erroneous I frames only in order to minimize delays and jitter. This scheme is simple to implement since it requires only breaking the layered architecture and have the MAC-HS layer access the application payload and get the type of frame. Overall, the proposed adaptive scheme is able to provide better QoS and gain of bandwidth at the expense of a slight degradation in video quality. Finally, bandwidth gain simply means that the system is able to support more users; the bandwidth gain results show that we can gain up to 10% using our adaptive scheme, meaning that this 10% can be used by other applications and to support more users.

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