Cross-Layer Optimization of Video Services over HSDPA Networks

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Abstract. In the third generation networks a quality of service is guaranteed for key applications, such as video streaming. However, all the packets belonging to one application are handled in the same way, even though they have different impact on the perceived quality. In this article we present a standard compliant cross layer optimization for video services. The importance of a packet, signalized at application level, is used to filter packets into different logical channels with different qualities of service. Simulations performed using a HSDPA system level simulator show that our implementation increases the video quality of over 0.6 dB.

Key words: HSDPA, Video Streaming, H.264/AVC, PDP Context

1 Introduction

Beside voice call applications, the third generation wireless networks offers mobile broadband to the customers. The Universal Mobile Telecommunications System (UMTS) [1] has been standardized by the 3rd Generation Partnership Project (3GPP) in the year 2000 (*Release 99*). In the *Release 5*, 2002, two main features have been added to the standard: (i) the IP Multimedia System (IMS) describes a framework for delivering Internet Protocol (IP) multimedia services, (ii) the cell throughput has been increased by the introduction of the High Speed Download Packet Access (HSDPA) from the 384 kbit/s of Release 99 to 14.4 Mbit/s.

Increasing available downlink datarate and simplified infrastructures make IP traffic in the wireless network constantly increase. The traffic consists of a variety of different applications [2], with different degrees of interactivity and technical needs. In the 3GPP standards, four classes of IP traffic have been defined in [3]: (i) Conversational: usually reserved for Voice over IP (VoIP), it covers connections between two or more human users with stringent delay constraints, (ii) Streaming: reserved for multimedia streaming from a server to one or more humans, which requires variance in the delay to be avoided, (iii) Interactive: it refers to applications such as web browsing, (iv) Background: includes all the applications where the user is not expecting the content in a given time, such as File Transfer Protocol (FTP) download. This classification has been introduced in order to guarantee Quality of Service (QoS) to the customers. IP packets belonging to an application with stringent delay constraint are favoured over the ones without timing specifications.

The QoS in UMTS is established by guaranteeing a given bitrate, handling priority and transfer delay to a logical connection. However, the QoS does not necessarily reflect the perceived Quality of Experience (QoE) [4], i.e. the user satisfaction for the required service. In this work we consider the cross-layer optimization of video streaming over HSDPA networks, aiming at the maximization of the QoE as perceived by the end users. In video streaming, although a single QoS value is assigned to all the transmitted packets, different packets have a different impact on the reconstruction of the video sequence and, therefore, on the QoE. For this reason, we propose to signalize the importance of a packet from application layer to the IP layer in the DiffServ field of the IP header. The different packets will then be multiplexed depending on their DiffServ marking into different logical connection to the end user. An appropriate QoS class is assigned to each logical connection.

This paper is structured as follows. In Section 2 an overview over wireless video streaming is offered. The proposed method is discussed in Section 3, together with a description of the HSDPA features relevant for this work. The HSDPA system level simulator used for transmission is presented in Section 4. The results in Section 5 and the conclusions in Section 6 terminate the paper.

2 Video Streaming over Wireless Networks

Streaming applications in 3G networks have been specified by the 3GPP Technical Specifications (TS) belonging to the class of "Transparent end-to-end Packet-switched Streaming Service (PSS)". Since the seventh release of [5], defining the mandatory and suggested codecs, the H.264/AVC [6] is mentioned as *suggested* video codec. The H.264/AVC, jointly standardized by the International Standard Organization (ISO) Moving Picture Expert Group (MPEG) and International Telecommunication Union (ITU) Video Coding Expert Group (VCEG), is currently the state-of-the art video codec for commercial applications. For this reason, the following discussion will refer this codec.

The H.264/AVC belongs to the family of the hybrid block based video codecs. These kinds of video codecs exploit the correlation of small regions of the sequence pictures in space and in time. Each sample of the raw video sequence, a video frame, is subdivided into square blocks, called MacroBlocks (MB). Depending on the frame type, two encoding strategies are allowed. The macroblocks belonging to Intra (I) predicted frames are encoded using the neighboring blocks of the same picture as a source of prediction. For Inter (P) predicted frames, the prediction is searched in the previously encoded pictures. In both cases, the prediction is then refined by means of frequency transformed *residuals*, representing the difference between the original block and the best prediction.

The video encoder has been conceptually subdivided into two functional blocks. The Video Coding Layer (VCL) deals with the proper encoding functionalities whereas the Network Abstraction Layer (NAL) provides network friendliness to the produced data stream, managing the segmentation of the code into NAL Units (NALU) and reducing the dependency of the data stored in different packets. The maximum size of a NALU is specified depending on the network Maximum Transfer Unit (MTU). Therefore, a NALU contains a variable number of macroblocks (representing a picture *slice*) depending on the effectiveness of the considered prediction. For video streaming over 3G Networks, the NALUs are further encapsulated into RTP, UDP and, finally, IP.

Because of bad channel conditions and/or network congestion, some packet might not be correctly received. At the decoder side *error concealment* techniques reduce the impact of missing packets. Because of the temporal prediction, a missing packet does not only affect the reconstruction of the picture slice it contains but rather all the frames that use that slice as a source of prediction, as indicated in Fig. 1. We will refer to this effect as *temporal error propagation*. The

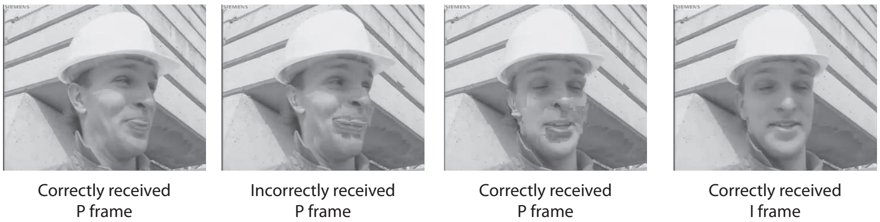

Fig. 1. Temporal error propagation

temporal error propagation ceases with the following I frame. Packets containing intra encoded information are self contained, i.e. the slice they contain can be reconstructed without the need of information stored in other packets.

The distance, in number of frames, between two consecutive I frames is defined as Group Of Picture GOP. The size of the GOP has a strong impact on the quality of the receiver side: small GOPs reduce the temporal propagation of the error. On the other hand, the spatial prediction used in the I frames is much less effective than the temporal one, particularly for high frequency patterns. The curve in Fig. 2 (left) shows the rate distortion behavior of the video quality in Luminance Peak to Signal Noise Ratio (Y-PSNR) depending on the GOP size. The transmission of the encoded 'Foreman' sequence in Common Intermediate Format (CIF) resolution (352×288 pixel) has been simulated considering different GOP sizes varying between 10 and 100 frames and fixed packet error rates (PERs) of one, three and five percent.

Since the error propagation is terminated by a correctly received I frame, preserving the payload of the packets containing encoded I slices is of major importance with respect to the one containing P slices. However, within the P frames of a GOP, one can differentiate between P packets of different importance. Different works in the literature, deal with the unequal error protection of video packets, depending on the impact on the quality the slice they contain

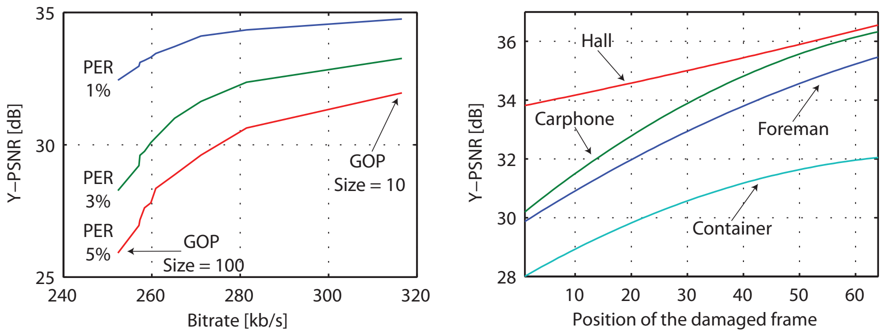

Fig. 2. Quality of corrupted sequences

have. These methods often call for refined rate distortion analysis and limit the observation on the punctual frame the slice belongs to. In this work we consider a more generic approach, aiming at better protecting the packets belonging to the frames closer to the beginning of the GOP. Following the previous discussion, we do not consider only the "punctual" effect of the errors at frame level, but rather the overall impact they have on the reconstruction of the whole GOP. Errors affecting the first frames of the GOPs propagate more in time, degrading more the QoE if compared to errors occurring near the end of the GOP.

In order to analytically evaluate this effect, different standard test sequences have been encoded with a fixed GOP size of 65 frames. For each simulation performed, the packets containing a frame in a given GOP position have been removed. In this scenario, for obtaining generic results, the effect of the single packets has not been investigated, since it strongly depends on the specific characteristic of the considered sequence. The results are shown in Fig. 2 (right)

3 Video Packet Prioritisation in HSDPA

The importance of an encoded slice might be signalized exploiting the fields of the NALU header [7]. It consists of one byte signalizing the correctness of the slice (one bit), the NAL reference indicator (NRI) (two bits) and the nalu type (five bits). The NRI value 00 is used for marking encoded slice not used as a reference by future inter frames. Values greater than 00 specify increasing packet priority, as indicated by the encoder.

At the video streaming server, this priority information is conveyed from the application layer to the IP layer. The IP header contains a byte originally thought for specifying the Type Of Service (TOS) and currently used for Differentiate Service (DiffServ) marking [8]. The DiffServ specifies how a packet has to be handled by each network element, i.e. a Per Hob Behaviour (PHB). Three main classes of DiffServ marking have been specified: (i) Default PHB: Best effort Traffic, (ii) Assured PHB: Ensures the forwarding of the packet as soon as a traffic threshold has not been exceeded, (iii) Expedited Forwarding PHB: Handles the packet with highest priority.

In the following we need a method to make use of this signaling information within the UMTS network. Other works in the literature, such as [9], already discuss this topic, but do not consider the possibility of assigning different DiffServ marking to packets belonging to the same data stream. This is a non trivial task as the IP packets carrying the user data are encoded into other transport protocols between GGSN and mobile terminal, e.g., GPRS Tunneling Protocol (GTP) and NodeB Application Protocol (NBAP). The data itself is only available at the borders of the UMTS network, i.e., GGSN and mobile terminal. However, there is a different and standard compliant way to use dynamic QoS settings for different packets. In a UMTS network a logical Packet Switched (PS) connection is setup by a PDP-context. Within this method, the user terminal at one side and the GGSN at the other side agree on several parameters, e.g., IP-address, PDP type and QoS profile for the following packets. In the UMTS network QoS only exists on this PDP-context entity, allowing for one setting at a time. To allow the system to serve independent QoS parameters for multiple applications running at the same mobile terminal, a feature called multiple PDP-contexts was introduced.

There exist two different categories of PDP-contexts, namely primary and secondary. Every mobile has to activate a primary context first. Then the mobile can either attach a secondary to the primary or initiate further primary ones. Multiple primary PDP-contexts have different IP addresses and are typically used for different applications, e.g., black berry in parallel to standard Internet access. Multiple primary PDP-contexts share the same IP address and must be attached to a previous initialized primary context. The following Figure 3 depicts the difference in the setup. As the IP address is the same for the primary and all the associated secondary PDP-contexts, a so called Traffic Flow Template (TFT) defines a split up of user data into the GTP tunnels.

A TFT can be seen as a packet filter applied onto each IP packet entering the GGSN. There exists one TFT per secondary PDP-context. The filtering rules are based on one or more of the following attributes: source address, IP protocol number, destination port (range), source port (range), IPsec security parameter index and the type of service. The last attribute in this list makes the TFT the perfect match for the proposed method as it is the same header field used for DiffServ. The following Figure 4 depicts the user of TFT for one user.

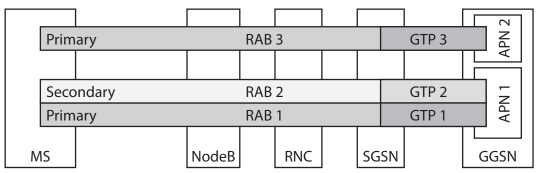

Fig. 3. Example for two primary PDP-Contexts and one secondary.

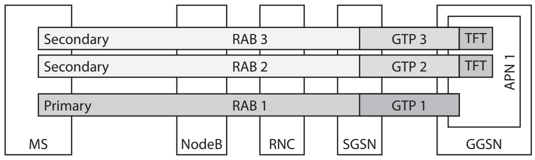

Fig. 4. Traffic Flow Tamplets.

While in UMTS R99 the users are served with dedicated channels (DCH) only, HSDPA offers a shared channel. It does this by the implementation of new coding schemes and modulations techniques combined with scheduling techniques directly in the NodeBs. In other words the Spreading Factor (SF) is no longer variable and there is no more fast power control available. These two elements of Rel. 99 are replaced by Adaptive Modulation and Coding (AMC), Fast retransmission strategy (HARQ) and scheduling algorithms [10, 11].

In the following we give a short discription of the new features introduced for the HSDPA MAC layer.

Scheduling Algorithms The place of the scheduling systems in Rel. 99 is inside the RNC. In HSDPA the function has been moved into the NodeBs, which allows for faster scheduling as there is no more 'reaction' delay present. The scheduler in HSDPA also has additional task. Beside selecting the correct modulation and coding scheme and the HARQ process, it now schedules the transmission for all users. In Rel. 99 the scheduler was implemented on a per user base only. The available algorithms are Round-Robin, Proportional Fair and Maximum C / I.

Hybrid ARQ: A Fast Retransmission Strategy The retransmission logic moved from the RNC entity into the NodeB. There exist two different error control and recovery methods to guarantee error free transmissions to and from the UE, namely Forward Error Correction (FEC) and Automatic Repeat reQuest (ARQ), see [12, 13, 14].

The disadvantage of these two methods is the delay which occurs in case of an packet error. This can be overcome by a combining the ARQ and the FEC method in a so called HARQ mode. The FEC is set to cover the most frequent error patterns and therefore will reduce the number of retransmissions necessary for the system. The ARQ part covers the less frequent error patterns, which allows to reduce the number of bit added by the FEC. There are different types of HARQ methods available and the performance in total depends on the channel conditions, receiver equipment and other related parameters. Considering the complexity of a UMTS radio implementation choosing the 'correct' or 'best' retransmission strategy is a wide field for ongoing research.

Adaptive Modulation and Coding (AMC) The original implementation of UMTS-Rel. 99 offered one fixed modulation scheme. The adaption to the actual radio channel is then achieved using a power control algorithm. The momentaneous data rate is set by choosing an appropriate spreading factor offering the needed gain for the given signal to interference situation.

In HSDPA the method was changed. Instead of relying on a fast power control the SF was fixed and the modulation now follows the channel conditions, both modulation and coding format adapt in accordance with variations in the channel conditions. This system is called Adaptive Modulation and Coding (AMC), or link adaptation. Compared to standard power control such methods deliver higher data rates. In HSDPA the AMR scheme assigns higher-order modulation with higher code rates, such as 16 QAM.

4 HSDPA System-Level Simulations

In order to assess the performance of enhanced scheduling algorithms, the implications in the context of network have to be evaluated. Therefore, standard physical-layer simulations are not sufficient, but rather system-level simulations are necessary [15, 16]. One of the major difficulties of system-level analyses is the computational complexity involved in evaluating the performance of the radio links between all base-stations and mobile terminals. Performing such a large number of link-level simulations is clearly prohibitive. Thus, those evaluations have to rely on simplified link models that still must be accurate enough to capture the essential behavior [17, 18, 19].

In this paper, we conduct our simulations on a computationally efficient system-level simulator implemented in MATLAB [20]. The simulator is capable of simulating classical HSDPA networks as well as the enhanced version utilizing MIMO for increased data rates. In this work however, we restrict ourselves to the classical single antenna HSDPA without the possibility for spatial multiplexing in the downlink.

The physical-layer modeling utilized in the system-level simulator accounts for MMSE equalization at the receiver side and accurately reproduces the inter-code interference in the multi-code operation of the shared downlink channel of HSDPA. The simulator is able to generate the cell deployment according to the desired configuration and deals with a large variety of user set-ups. A basic overview of the simulation methodology is depicted in Figure 5.

The HSDPA cell deployment considered in the simulator consists of 19 three-sector sites, corrsponding to the layout type 1 of [21]. The simulator allows for the power of the neighboring base stations to be controlled independently, such that the network to be simulated can be stripped down to seven three sector sites or even a single cell scenario. Different propagation models are available in the simulator, where in this work we stuck to the well known Walfish-Ikegami model [22] representing urban micro cell scenarios. Radio link control, as well as scheduling in the MAC-hs are simulated only for the target sector, thus keeping the computational effort manageable. This, however, requires the simulation to get along without handovers (because in the case of a handover, the associated algorithms—residing in the RNC—would have to coordinate the radio link

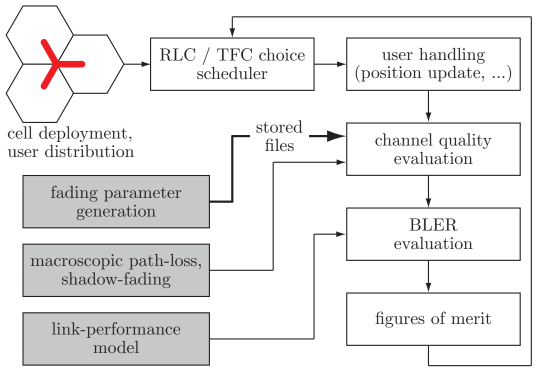

Fig. 5. Overview of the employed system-level methodology [20].

control of two sectors). In this work we set up the user mobility such that no handover will occur during the simulation of a video transmission.

The basic simulation procedure is as follows (see Figure 5): The first step of the simulation invokes the network generation, i.e. cell deployment, and the user generation according to the selected UE capability class together with their positioning. Also the fading parameters (describing the physical-layer) suitable for the scenario are loaded, the shadow fading traces are generated and the data necessary for the link-performance model (describing the decoding performance) is loaded. In the main simulation loop, according to the feedback of the UE in the target cell, the RLC and the MAC-hs scheduler decide upon the user to be served and the transmission settings of this transmission. After this decision an update of the user position takes place. With the position being known, the macro-scale pathloss and the effective antenna gain can be calculated. The SINR in the current transmission is then evaluated and consequently the correctness of the received packet is determined according to the link-performance model. The user feedback is then formed of the ACK/NACK report and the CQI for the current transmission evaluated pursuant to the mapping of the UE capability class. At the end of the simulation time, the resulting data is collected and statistically evaluated. The basic simulation settings for the investigations in this work are given in Table 1.

4.1 Content Aware Scheduling

To exploit the information available at the MAC-hs scheduler, i.e. the priority of the incoming IP packets, we implemented a scheduler that dynamically adapts the transmission settings. The basic idea behind this is to protect packets of high priority better against transmission errors, but also allow the scheduler to downgrade packets in case of less importance [23].

HSDPA dynamically adapts the encoding and modulation—i.e. the transport block size γ—of a transmission according to the channel quality feedback information (CQI) of the UEs, c_{UE}, thus

Table 1. System-level simulation parameters.

Parameter	**Value**
network load	homogeneous
number of cells	19
Node-B distance	750 m
transmitter frequency	1.9 GHz
total power available at Node-B	20 W
spreading codes available for HS-DSCH	15
macro-scale pathloss model	urban micro [22]
channel type	PedA
active users in target sector	5
user mobility	3 km/h, random direction
UE capability class	10
UE receiver type	MMSE

$$\gamma = f(c_{\mathrm{UE}}). \tag{1}$$

Our scheduler now interferes with this mapping. Depending on the priority of the packets, we remap the transport block size γ,

$$\gamma_{\mathrm{new}} = \Phi_p \left[f(c_{\mathrm{UE}}) \right], \tag{2}$$

where $\Phi_p[\cdot]$ denotes the remapping function depending on the packet priority p. Better protection can be achieved by remapping the transport block size to lower values, and vice versa. However, if we decrease the transport block size too often, the average throughput of the cell would also decrease notifiable, which is general undesired because spectral resources would be wasted.

In Section 2 we elaborated that packets belonging to I-frames contribute greatly to the video quality, whereas packets belonging to P-frames influence the quality not that prominent. In particular this holds for packets of P-frames that are at the end of a GOP. Since one I-frame is approximately equally large as four P-frames, we balance the loss in average throughput by increasing the transport block size for the packets belonging to these last four P-frames. Let us assign the following priority classes

- priority $p = 2$: packets belonging to I-frames,
- priority $p = 1$: packets belonging to the first $(l_{\mathrm{GOP}} - 4)$ P-frames of the GOP,
- priority $p = 0$: packets belonging to the last 4 P-frames of the GOP,

where l_{GOP} denotes the length of the GOP (in frames). Then the remapping $\Phi_p[\cdot]$ of the transport block size can be written as

$$\Phi_p[f(c_{\mathrm{UE}})] = f(c_{\mathrm{UE}} + 1 - p). \tag{3}$$

To ensure fairness against all active users in the cell, the scheduler selects the user according to a round robin strategy. In addition we limited the maximum

Table 2. Content-aware scheduler settings.

Parameter	Value
user selection	round robin
max. nr. of HARQ retransmissions	1
transmission remapping	Equation (3)
remapping boundaries	CQI 0 and 30
HARQ recombining scheme	incremental redundancy

delay that may occur in the transmission of the packet by setting the network to allow a maximum number of one HARQ retransmission. The basic scheduler parameters are summarized in Table 2.

5 Results

The simulations have been performed using the standard test sequence "Foreman" in QCIF resolution comparing the typical round robin (rr) approach with the content aware (ca) scheduling as described in Section 4.1.

In order to evaluate the performance of the proposed method in terms of preservation of the payload content, the error probability of the Transport Blocks (TBs) has been depicted in Fig. 6 (left). For the round robin scheduling, no distinction has been made between TBs containing I or P encoded frames. As expected, the error probability lies around 10 %. For the proposed content aware scheduling mechanism, the error probabilities associated to slice containing I and P frames have been presented separately. The proposed CQI mapping allow the error probability of the transport blocks containing the I frames to decrease of a factor 4, being it around 2.7 %. Aiming this work at the optimization of

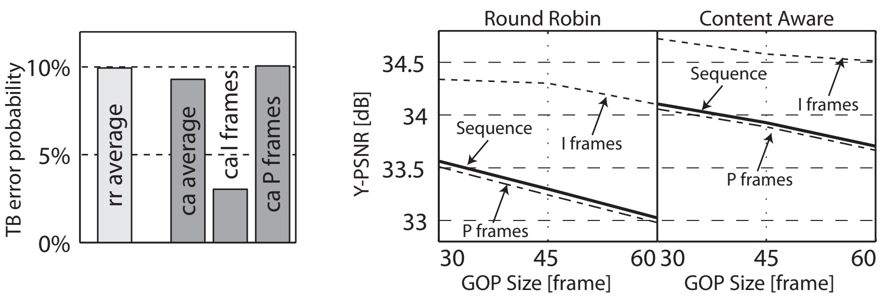

Fig. 6. Simulation results: TB error rate and video quality

the quality of experience, the overall performance of the proposed method has been evaluated with respect to the Y-PSNR. Following the discussion in Section 2, three different GOP sizes have been investigated: 30, 45 and 60 frames. The results are shown in Fig. 6 (right).

As a consequence of the smaller TB error probability, as shown in Fig. 6 (left), the quality of the I frames has increased of over 0.5 dB when using content awareness. Such increase, however, is not only beneficial in terms of contribution to the average frame quality, but rather has to be considered advantageous for the quality of the following P frames. On the one hand, a valid source of prediction is offered to the following P frames, on the other hand, in case the previous GOP was damaged, the temporal error propagation of the error is terminated. The overall sequence quality measured using the proposed content aware scheduling mechanism is 0.6 dB higher than the one obtained with the typical round robin approach.

6 Conclusion

In this paper a standard compliant content-aware scheduling mechanism for HSDPA network has been presented. In a video streaming session, the application layer marks the importance of a packet into the NAL header. This information is used to select the appropriate DiffServ marking at IP level. The IP packets are then multiplexed at the GGSN in different logical channels with appropriate QoS settings. The performance of the method has been investigated using an HSDPA system level simulator. By means of the proposed solution, the video quality has been increased of over 0.6 dB.

Acknowledgments

The authors thank mobilkom austria AG for technical and financial support of this work. The views expressed in this paper are those of the authors and do not necessarily reflect the views within mobilkom austria AG.

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