On the Feasibility of Video Streaming Applications over GPRS/EGPRS

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***Abstract*— In this paper the potential of the General Packet Radio Service (GPRS) and Enhanced GPRS (EGPRS) concerning Video Streaming is presented. Taking real video sequences as the basis for a Video Streaming load generator, the simulation tool GPRSim that comprises a prototypical implementation of the standardized GPRS/EDGE protocols has been extended. Results have been gained by simulating actual as well as future scenarios. Both fixed and on-demand Packet Data Channel (PDCH) allocation techniques have been considered to carry the generated traffic which contains Video Streaming as part of a traffic mix additionally comprising e-mail and World Wide Web (WWW) traffic. The following analysis of the traffic performance measurements shows the limited capacities of GPRS networks to carry real-time data. But the simulation results also indicate that EGPRS as the enhanced system is, on a limited scale, able to be bearer for real-time traffic.**

I. Introduction

In the context of the evolution towards 3rd Generation (3G) mobile radio networks, packet-switched data services like the General Packet Radio Service (GPRS) and the Enhanced GPRS (EGPRS) are presently introduced into GSM and IS-136 systems worldwide. This enables everyone to use sophisticated applications and services which might handle expanded multimedia data. As Streaming is strongly associated with multimedia content the providers have to take care that offered Streaming applications are properly working inside their mobile networks. Potential services, employing real-time data transmission may be:

- **commerce**, e.g., commercials, presenting information about special products which can be bought online
- **guiding**, e.g., multimedia city guides enriched with speech and video clips of frequently visited sights

Several papers concerning GPRS performance analysis have been published in the last years [1], [2]. They are based on conservative traffic models for applications like WWW, e-mail, WAP or FTP. They do not contain real-time services with its special sensitivity concerning QoS guarantees.

In Section II after this introduction the Video Streaming traffic model and its underlying coding standard H.263 is introduced. After the description of the simulation environment GPRSim in Section III, the traffic performance evaluation of Video Streaming applications are presented for GPRS and EGPRS scenarios in Section IV.

II. Video Streaming Traffic Model

In the scope of modeling video sources, a lot of attention has been paid to long range dependent or self-similar models of traffic streams. Since MPEG (Moving Pictures Expert Group) and H.263 video traffic consists of a highly correlated sequence of images due to its encoding, the correct modeling of the correlation structure of the video streams is essential [3]. For the load generation within the GPRSim the decision was made not to model video streams with analytical models but with real video sequences coded by an H.263 coder.

A. H.263

Beside MPEG H.263 is the most accepted standard. Its basic configuration is based on ITU-T Recommendation H.261 and is a hybrid scheme of inter-picture prediction to utilize *temporal dependency* and transform coding of the remaining signal to remove *spatial redundancy*. In addition to the basic video source coding algorithm, negotiable coding options are included for improved performance. The most important ones are:

- Unrestricted Motion Vector Mode (annex D)
- Syntax-based Arithmetic Coding (annex E)
- Advanced Prediction Mode (annex F)
- Bidirectional Predicted Frames (annex G)

1) H.263 version 2: The ITU-T worked out a version 2 of H.263, which is specified in the Annex O of the recommendation [4]. This so-called H.263+ version contains a number of optional feature enhancements, which are added to the core syntax, and semantics of H.263. The new features can be subdivided in three categories:

- New types of pictures
- New coding modes
- Supplementary enhancement information

B. Video Streaming Traffic Generator

The Video Streaming traffic model used within this work is based on three video sequences in Quarter Common Intermediate Format (QCIF) with the resolution of 176×144 pixels. The sequences are proposed by the Video Quality Expert Group (VQEG) and are for this reason commonly used. Each sequence is representing a particular group of videos with different intensities of motion.

- `Claire` stands for a very low motion intensity and can be seen as a characteristic visual telephony or inactive video-conferencing.
- `Carphone` includes both, periods with rather high motion and periods of low motion intensity. It represents many kinds of vivid or active video-conferences.
- `Foreman` is a video sequence with permanently high motion intensity of both, the actor and the background which is characteristic for sport events or movies.

We applyied a skip factor of 2, which means that every second frame was skipped so that the frame rate of the coded sequences was reduced to 12.5 frames/s. The quantization level 20 (Q20) was adjusted for I- respectively P-frames. The resulting video is of low quality, but it is acceptable for mobile devices with its limited visual output capacities. To generate one single stream of video data the three video sequences are randomly concatenated. Like this a stream is generated which is composed of different sequences with varying motion intensity. A conventional mix of sequences including 80% `Claire`, 10% `Carphone` and 10% `Foreman` has been selected for the simulations performed. The mix shall represent video streams with low motion infrequently interrupted by phases of higher motion intensity. Due to the negligible size of RTSP and RTP control messages in comparison to the size of real-time data, they have been neglected. The resulting average IP traffic of this particular mix is 14.39kbit/s (see to Table I). Beside visual telephony all of the new emerging applications are relatively short in duration. So called *heavy users*, generating long streams with huge amounts of data, have not been taken into account. Thus, the duration of video sessions is modeled by a negative exponential distribution with an average value of 60s. This was chosen against the background of predicted future applications and with regards to the estimations for UMTS in [5].

Sequences	offered IP traffi c	
	Q20	80-10-10 Mix
`Claire`	10.9 kbit/s	**14.39 kbit/s**
`Carphone`	26.7 kbit/s	
`Foreman`	31.7 kbit/s	

TABLE I
OFFERED IP TRAFFIC OF VIDEO SEQUENCES

III. SIMULATION ENVIRONMENT

The GSM/GPRS Simulator GPRSim [6] is a pure software solution based on the programming language C++. Up to now models of Mobile Station (MS), Base Station (BS), and Serving GPRS Support Node (SGSN) are implemented. The simulator offers interfaces to be upgraded by additional modules. For the implementation of the simulation model in C++ the Communication Networks Class Library (CNCL) is used which is a predecessor to the SDL Performance Evaluation Tool Class Library (SPEETCL) [7]. This allows an object oriented structure of programs and is especially applicable for event driven simulations. Unlike the usual approaches to build a simulator, where abstractions of functions and protocols are being implemented, the approach of the GPRSim is based on the detailed implementation of the standardized protocols. This enables a realistic study of the behavior of Streaming over the protocol stack of GPRS. The complex protocols like LLC, RLC/MAC based on GPRS/EDGE Release 99, the Internet traffic load generators and TCP/IP are specified formally with the Specification and Description Language (SDL) and are translated to C++ by means of the Code Generator SDL2CNCL [7] and are finally integrated into the simulator.

A. Packet Traffic Generators

Since the Video Streaming applications are running in parallel with WWW and e-mail services, Internet traffic models are necessary for simulative examinations of the performance of data services of mobile radio networks. In the following, model parameters of these two applications and their distributions for generating protocol specific traffic are presented. Related documentation can be found in [8] and [9]. The parameters of these models are updated by parameters given by ETSI/3GPP suppositions for the behavior of mobile Internet users [5].

1) WWW Model: WWW sessions consist of requests for a number of *pages*. These pages consist of a number of *objects* with a dedicated *object size*. Another characteristic parameter is the delay between two pages depending on the user's behavior to surf around the Web [8], [5]. Table II gives an overview of the WWW parameters. The small number of objects per page (2.5 objects), and the small object size (3700 byte) were chosen, since Web pages with a large number of objects or large objects are not suitable for thin clients such as PDAs or smart phones.

2) E-mail Model: The e-mail model describes the traffic arising with the transfer of a message downloaded from a mail server by an electronic mail user. The only parameter is the amount of data per e-mail. A constant base quota of 300 byte is added to this size [9]. The parameters of the two applied distributions are shown in Table II. An overall mean value of 10000 byte for the e-mail size is chosen, since it is assumed that no e-mails with large attachments will be downloaded on mobile devices.

IV. PERFORMANCE OF VIDEO STREAMING OVER GPRS/EGPRS

Simulations are performed using a typical application mix for introduction scenarios containing Video Streaming, WWW and e-mail traffic. Due to the conservative predictions concerning the future usage of Streaming applications

WWW Parameter	Distribution	Mean	Variance
Pages per session	geometric	5.0	20.0
Intervals between pages [s]	negative exponential	12.0	144.0
Objects per page	geometric	2.5	3.75
Object size [byte]	$\log_2$-Erlang-k ($k = 17$)	3700 (transf.: 9.4)	$4.67 \cdot 10^9$ (transf.: 5.2)
e-mail Parameter	Distribution	Mean	Variance
e-mail size (lower 80 %) [byte]	$\log_2$-normal	1700 (transf.: 10.0)	$5.5 \cdot 10^6$ (transf.: 2.13)
e-mail size (upper 20 %) [byte]	$\log_2$-normal	16000 (transf.: 9.5)	$71.3 \cdot 10^9$ (transf.: 12.8)
Base quota [byte]	constant	300	0

TABLE II
TRAFFIC MODEL PARAMETERS

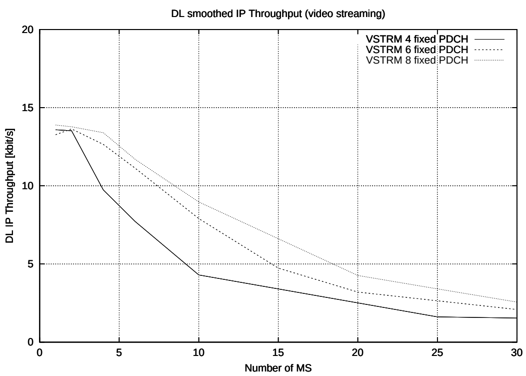

Fig. 1. DL IP throughput per user (Video Streaming)

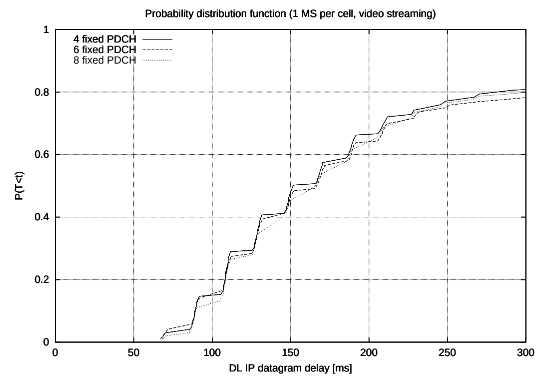

Fig. 2. DL IP datagram delay distribution (1 MS per cell)

the mix contains 10% Streaming traffic. The remaining part is composed of the conventional Internet applications namely 63% e-mail and 27% WWW. Today's WAP applications will emerge to TCP-based applications, following the WAP 2.0 specification and will show similar behavior as WWW.

A. GPRS with fixed PDCHs

In this first section GPRS with fixed allocated PDCHs carries the traffic load. In Fig. 1 the downlink (DL) IP throughput per user separated for Streaming applications is recorded. Serving one single MS per cell all configurations (4, 6 and 8 fixed PDCHs) are able to provide the bit rate of 14.39kbit/s which is necessary to carry the particular video stream. With the increase of active MS per cell the downlink IP throughput per user is falling below 14.39kbit/s depending on the number of fixed PDCHs. Allocating 4 fixed PDCHs only 2 MS per cell can be served well. Video streams of up to 4 MS are able to be carried with 8 fixed PDCHs.

Providing only enough data rate for Streaming applications is not sufficient. Also the downlink IP datagram delay is a critical performance measure for Streaming applications. The distribution of the IP datagram delay is outlined in Fig. 2. It is measured with 1 active MS per cell. It can be seen that 80% of the datagrams arrive at the client side with a delay of less than 300 ms. The performance is not depending on the number of PDCHs available, since the regarded mobile stations can use only maximum 4 slots on the downlink. The step-like distribution function can be explained by the segmentation of IP datagrams into an integer number of radio blocks and their transmission within a GPRS radio block period of 20ms.

The exact limit where Video Streaming becomes unacceptable highly depends on the variation of the IP datagram delay, so called *jitter*. This variation of the delay of incoming IP datagrams has to be buffered within the device's receive buffer. The size of this buffer is the critical parameter whether a given jitter is acceptable or not. While for conversational streaming applications like visual telephony a maximum delay of 200ms is tolerable, for non-conversational applications like video on-demand a delay of up to 10s is acceptable. During the introduction of streaming services the operators will not be able to differentiate between streaming and non-streaming data in the RLC/MAC layer. Thus the RLC/MAC layer in the simulation modul is operating in acknowledged mode. Having a buffer of up to 10s, it is possible to retransmit erroneous or lost radio blocks. They can easily be integrated in the queue by means of the RTP timestamp. By using the RLC/MAC unacknowledged mode in the future for conversational video applications, e.g. video conferencing, the downlink IP datagram

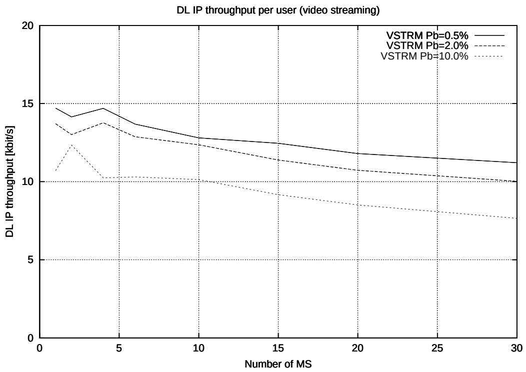

Fig. 3. DL IP throughput per user (Video Streaming)

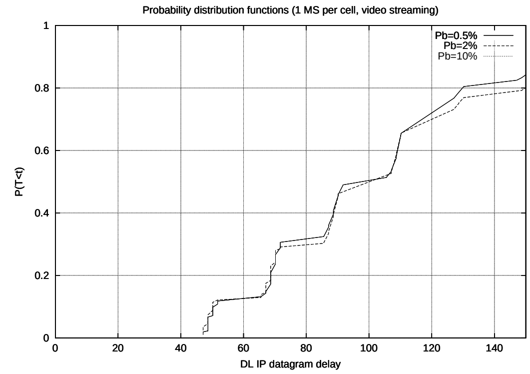

Fig. 4. DL IP datagram delay distribution (1 MS per cell)

	8 fixed PDCH	8 on-demand PDCH		
		Pb = 0.5%	Pb = 2%	Pb = 10%
15 MS	0.18%	0.88%	1.82%	3.95%
30 MS	0.15%	0.78%	1.89%	5.12%

TABLE III

SESSION ERRORS CAUSED BY CONGESTION

delay of video data would decrease and the necessary downlink IP throughput per user could be provided for a higher number of MSs.

The different characteristic requirements of the applications can be supported by Quality of Service (QoS) management [2]. The transmission of Streaming data may be privileged on the expense of e.g. e-mail and WWW traffic. On the one hand the application response times for WWW and e-mail increases, but on the other hand the Streaming application is able to proceed although high traffic load occurs in the cell.

B. EGPRS with on-demand PDCHs

In this section the enhanced bearer service EGPRS with on-demand PDCH allocation is investigated. 3 transmitter-receiver units (TRXs) are providing 21 PDCHs for both, circuit-switched (CS) and data traffic. Up to 8 PDCHs can now be utilized by EGPRS. These PDCHs are assigned on-demand for EGPRS data traffic according to the actual demand. The parameter which is varying in the following graphs is the blocking probability (P_b) of speech calls. This parameter, used by operators for dimensioning networks, indicates the probability that an incoming speech call can not be accepted by the network due to congestion. The GPRSim models this P_b by introducing parallel CS traffic.

In Fig. 3 the downlink (DL) IP throughput per Video Streaming user is outlined. It can be seen that even with 1 active MS per cell the required grade of service for video streaming can not be provided in scenarios with a P_b of 10.0%. Assuming a P_b of 0.5% a downlink IP throughput of 14.39kbit/s is available, while networks with a higher P_b, which means that less PDCHs can be assigned on-demand to EGPRS data traffic, can not guarantee the necessary continuous bit rate. With a rising number of MS per cell the downlink IP throughput per user is decreasing, but not with the magnitude which could be seen in Fig. 1. This effect is caused by sessions which have been blocked due to congestion. Utilizing fixed PDCHs nearly all sessions initiated by users have been accepted while sessions within on-demand scenarios have been blocked more frequently. In these particular situations no further PDCHs are available and thus the sessions are terminated producing an error. The percentage of errors produced by sessions in some example scenarios can be found in Table III. The terminated sessions leave free capacity which can be utilized by other users so that their downlink IP throughput remains higher.

The distribution of the IP datagram delay is shown in Fig. 4 for 1 MS per cell. Taking a P_b of 0.5% (2.0%, respectively) approximately 80% of all datagrams face a delay which is less or equal to 150ms. This indicates that only scenarios with P_b of less than 2.0% together with only a few MS sharing one cell makes Video Streaming possible. The graph for P_b of 10% is not included because its delay varies over such a wide range that the evaluation class could not draw a statistically correct graph. But a downlink IP throughput of 11kbit/s and a mean delay of 13s indicates that such a configuration is not able to provide the continuous data rate needed for the particular stream.

C. EGPRS with fixed PDCHs

In this last section EGPRS with fixed PDCHs is regarded as the bearer service for the assumed application mix. The downlink (DL) IP throughput per streaming user, shown in Fig. 5, starts at 14.39kbit/s, which is exactly the data rate needed for the chosen video stream. The downlink IP throughput per user is staying constant as long as the necessary data rate for streaming video is provided. The Streaming application is able to proceed. Depending on the number of fixed PDCHs the real time data rate is decreasing below the required rate of

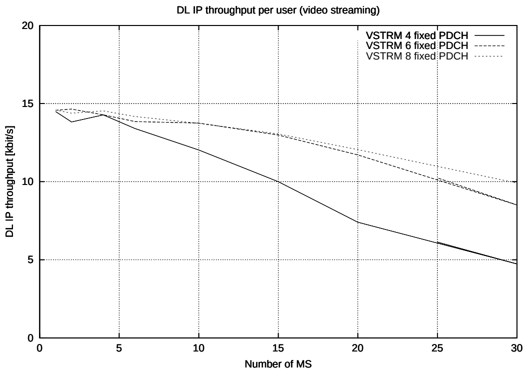

Fig. 5. DL IP throughput per user (Video Streaming)

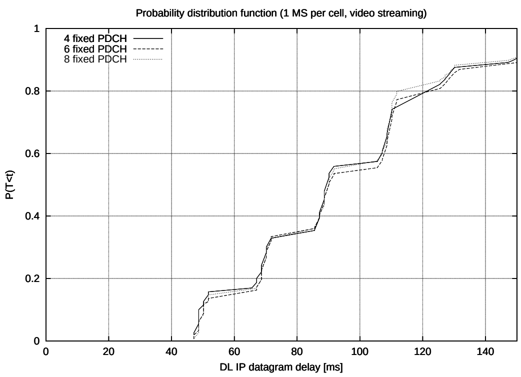

Fig. 6. DL IP datagram delay distribution (1 MS per cell)

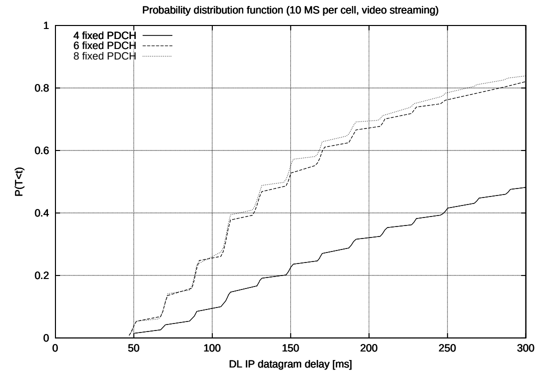

Fig. 7. DL IP datagram delay distribution (10 MS per cell)

14.39kbit/s. At this point the IP datagram delay is increasing disproportionately and the stream cannot be maintained any more. Due to the higher capacity of EGPRS in contrast to GPRS, Video Streaming sessions of up to 5 active MSs per cell can be handled by configurations with 4 fixed PDCHs. 6 and 8 fixed PDCHs are able to carry up to 10 MSs. The distribution functions of the downlink IP datagram delay confirm these interpretations. For one MS per cell (see Fig. 6) 90% of the IP datagrams carrying Streaming applications are delivered within 150ms. The performance does not depend on the number of PDCHs available, since the regarded mobile stations can use only maximum 4 slots on the downlink. For 10 MS per cell (see Fig. 7) and 4 PDCHs available more than 50% of the IP datagrams are delayed more than 300ms and the slow increase of the distribution function indicates a high delay variance. With 6 and 8 PDCHs 85% of the IP datagrams are delivered within less than 300ms, which makes the Streaming performance acceptable for 10 active users generating the multimedia traffic mix.

V. Conclusions

In this paper the potential of GPRS/EDGE networks concerning Video Streaming is presented. Both fixed and on-demand PDCH allocation have been considered to carry the generated traffic which contains Video Streaming as part of a traffic mix. The discussion and analysis of the traffic performance results shows the limited capacities of GPRS to carry real-time data. It has been shown that between 2 and 4 active users can be served by 4 and 8 fixed PDCHs, respectively. EGPRS as the enhanced system is, on a limited scale, able to be bearer for real-time traffic, if PDCHs are exclusively allocated for EGPRS. Up to 10 active users can be served with 8 fixed PDCHs, while with on-demand PDCHs and a P_b of more than 2% the performance for Streaming gets unacceptable. Privileged transmission of real-time data, guaranteed by QoS management, may be one possibility to provide the required bitrate for Video Streaming even if high traffic occurs, which has to be examined in future research.

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