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A TESTBED FOR VIDEO STREAMING WITH BROADCAST QUALITY OVER 3G+ MOBILE NETWORKS

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ABSTRACT

New generation mobile technologies are able to provide connection qualities comparable to fixed networks. With theoretical throughputs of up to 11.5 Mbps for the uplink, latest UMTS Releases (HSUPA) offer new business opportunities for both the network operators and their clients. In this paper, a broadcast quality real time video streaming system over HSUPA is studied and tested, providing statistics for packet loss rates and quality measurement. The study focuses on streaming video from a laptop computer connected to a USB 3G+ modem to a television production centre, where the stream can be retrieved for broadcasting purposes.

KEYWORDS

UMTS, video, mobile, streaming, broadcasting

1. INTRODUCTION

Field production and electronic news gathering is a common practice in TV programming, especially for news bulletins and programs alike. Two different approaches are used for these issues: ENG (Electronic News Gathering) and EFP (Electronic Field Production).

Most times, ENG resources consist on simple video shooting which will be later edited and included in news bulletins, documentaries, interviews, etc. When live information is needed, but a full mobile unit is not required nor justified, a light mobile unit with satellite or microwave connectivity can be used. Choosing one of the two links depends on propagation conditions and/or visibility restrictions.

In cases when a live, long, real time edited, transmission is needed the only available option nowadays is to set up a full mobile unit, which include almost all the features of a TV production studio. Again, microwave or satellite links are needed to send the information.

In this paper we discuss the viability of using a solution based on HSUPA (3GPP 2008) enabled mobile terminals to achieve a real time or near real time video transmission equivalent to the current ENG resources, but with the increased advantages of a light, mobile and easy to set up link. The high bit rates reached by UMTS Release 06, which has been set up for commercial use in the past six to twelve months by different carriers, bring the possibility to do live TV connections with broadcast quality. However, full mobile units are still difficult to substitute with these technologies.

First of all, we will briefly review the involved technologies and some of the key factors of the proposed solution. Section 3 is devoted to describe the developed testbed in further detail. In section 4 we will show the results obtained in both the single link and the multiple concurrent links cases. Conclusions of the paper are resumed in section 5.

2. INVOLVED TECHNOLOGIES

For this paper, an end to end solution is proposed for testing purposes. This solution is based on most recent standards for video coding and mobile communications: MPEG-4 Part 10, also known as H.264/AVC (advanced video coding) (Wiegand 2003) and UMTS Release 06 or HSUPA, provided by Vodafone. A wireless video transmission system has been proposed in several former studies (Stockhammer 2003 and

2005), emphasizing H.264's extremely high performance in wireless environments due to its Network Abstraction Layer and the efficient coding achieved. However, none of these methods considered to use this codec for a high quality stream (Schierl 2005); i.e., for a broadcast-ready transmission. In this paper, we discuss the possibility of reaching that goal using the above mentioned technologies.

2.1 H.264

The coding system chosen for this study features an excellent bandwidth-quality rate, compared to other codecs in use today. As shown in Figure 1, given a PSNR value, H.264 needs up to fifty percent of the bandwidth required by MPEG-2, which is the standard used in DVB-T and DVD-Video.

H.264 takes advantage of all the different types of redundancy in a video stream (spatial, temporal and statistical) to reduce the used bandwidth. This makes H.264 a widely spread codec for commercial use. For instance, it is being used as the coding system for DVB-H and High Definition Video.

This standard developed by the Motion Picture Expert Group (MPEG) in collaboration with the International Telecommunication Union (ITU) and released in 2003 is expected to increase its performance until 2013, due to the fact that most commercial implementations are still not able to use all of the functionalities and features provided by the standard.

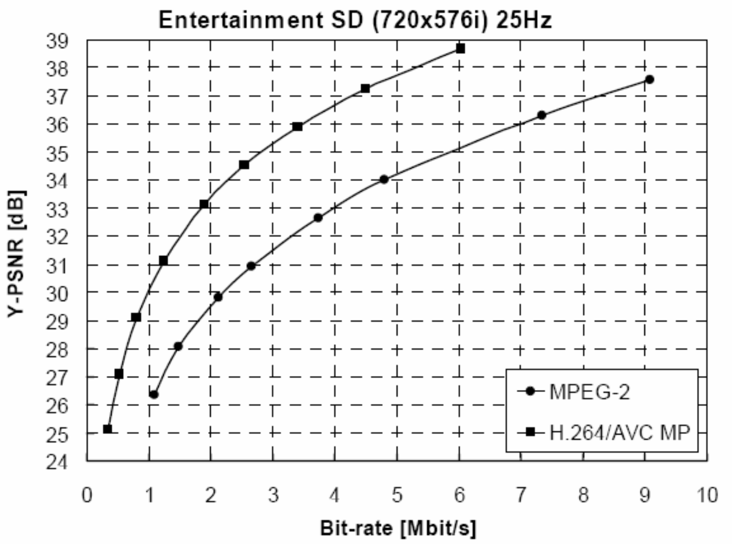

Figure 1. H.264 vs. MPEG-2 comparison

2.2 HSUPA

The 2006 revision of the third generation mobile telephony standard brings important improvements for the uplink packet transmission. These improvements, together with the ones provided by UMTS Release 05 (HSDPA) can accomplish rates comparable to fixed connections. The combination of both technologies is commercially known as High Speed Packet Access (HSPA), Broadband 3G, 3.75G or 3G+.

HSUPA introduces changes to the uplink very similar to the ones that introduced HSDPA to the downlink, for instance:

- Lower transmission intervals (TTI), from the previous 80 ms to 10 ms at most.
- Smart retransmission methods, with incremental redundancy.
- Control processes in the B-node or base station, reducing delays and latency.
- Dynamic bit rate assignment to different terminals in a certain cell every 10ms, depending on the radio channel conditions (noise), available power of the terminal, and available amount of data.

Different categories have been defined for both terminals and network systems, depending on the supported features. A list of all the categories already defined (including the last revision of the standard by the 3GPP) can be found in Table 1.

Table 1. HSUPA Categories

Category	Maximum Speed
Cat. 1	0.73 Mbps
Cat. 2	1.46 Mbps
Cat. 3	1.46 Mbps
Cat. 4	2.93 Mbps
Cat. 5	2 Mbps
Cat.6	5.76 Mbps
Cat. 7	(3GPP Rel07) 11.5 Mbps

3. TESTBED DESCRIPTION

The scheme used for this study consists in a professional digital video camera connected to a laptop PC, where the video signal will be processed and encoded using the H.264 Baseline profile at rates between 768 and 1024 Kbps, plus a mono audio signal at 96 Kbps. This is encapsulated into a MPEG Transport Stream which is then sent through Internet using RTP packets. The PC is connected to the internet using an access point included by Vodafone in the USB HSUPA modem (Figure 2).

At the destination, another PC monitors the received streams, using an IP traffic analyzer (Wireshark), saving logs of delays between consecutive packets and packet loss. Moreover, the data is saved into a video file in order to allow comparisons with the original video stream and therefore obtain objective and subjective quality measurements.

The workflow consists in setting up an inverse remote desktop session using the VNC protocol, from the monitor PC (the one which will receive a video stream) to the transmission laptop. This is done via a SSH connection. Although initially all computers ran GNU/Linux, the transmitter was moved to Microsoft Windows Vista, due to the fact that the current USB Modem beta driver for GNU/Linux provided by Vodafone is not capable to reach the highest upload bitrates. This decision led to another difficulty, which is that outgoing traffic could not be analyzed using Wireshark, because Winpcap libraries do not support ppp interfaces for Windows Vista.

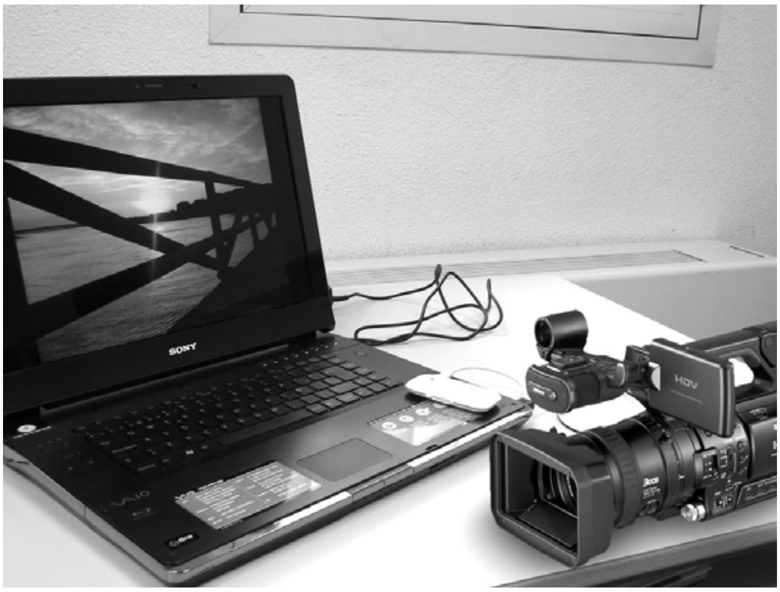

Figure 2. Equipment involved in the study.

4. OBTAINED RESULTS

4.1 Results in a Single Link

The first tests consisted on several files transmissions from a single computer using an HSUPA modem to a remote monitoring computer. As shown in Figure 3, there was zero packet loss for more than half the times a packet was sent. Therefore the most probable case is that no packet gets lost in a single stream environment. However, in some cases there has been non-zero packet loss, thus the probability of packet loss in a stream is approximately 1 packet loss every 11,250 packets sent. Considering that a five minute video can produce between 20,000 and 40,000 packets, this makes that the probability of packet loss per video file transmission (or per real time connection) approximately of 2-4 packets.

As of delays, experiments show delays of between 1 and 2 seconds between the original video signal and the received stream, and the data (see Figure 4) reflects that the maximum delay between consecutive packets is 200 ms, which forces a more than 200ms buffer to accomplish a fluent video streaming in reception.

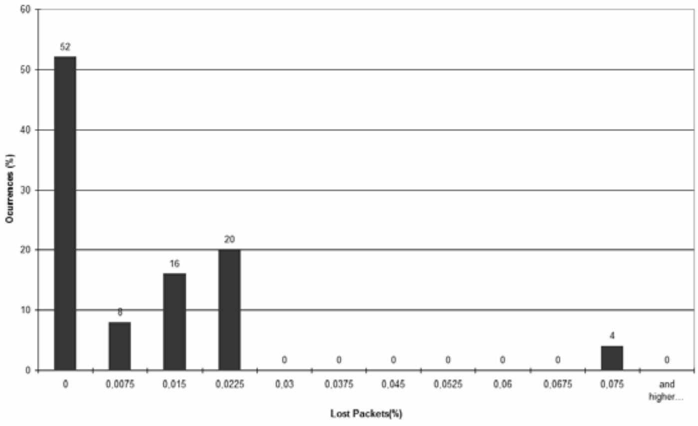

Figure 3. Packet loss in single link.

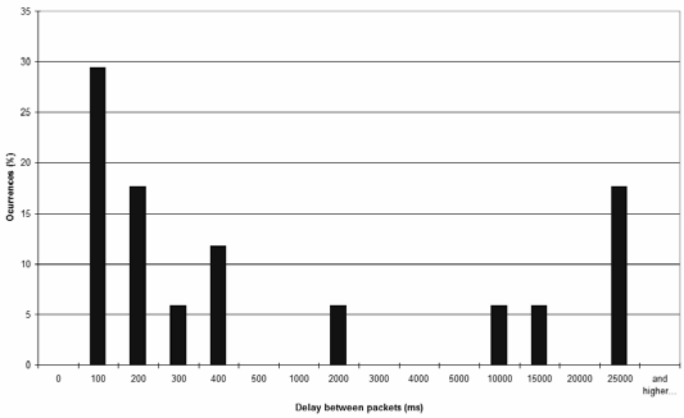

Figure 4. Consecutive packets delay.

4.2 Results in Multiple Concurrent Links

An experiment with several computers transmitting simultaneously to the same B-node was included in the study. In this case a higher demand of bandwidth to the B-node cannot be attained, given the limitations of the present deployment of Vodafone infrastructure. Results show (see Figure 5) that expected packet loss increases up to a 0.5%. However, this problem is expected to be solved as soon as both terminals and networks support higher HSUPA categories than the current Category 5.

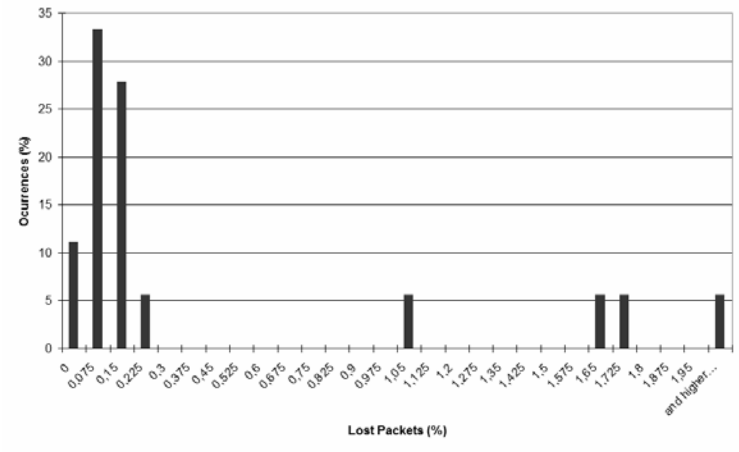

Figure 5. Packet loss in multiple simultaneous links.

5. CONCLUSIONS

Despite the fact that it is not possible to guarantee completely neither in time arrival of packets nor a certain degree of lost packets, a real time video streaming scheme like the one proposed can be successfully deployed.

Even though the number of simultaneous connections currently limits considerably the system scalability, this problem is expected to be solved as the latest features detailed in the 3GPP standards become a reality for carriers' infrastructure. The study opens an important market for carriers, who can now see new opportunities for their uplink as well as a chance for TV studios and associated companies to provide the final user with more real time content at a cheaper cost.

REFERENCES

3GPP, 2008. Universal mobile telecommunications system technical specifications and technical reports for a utran-based 3gpp system (3gpp ts 21.101 version 6.8.0 release 6). Online: [ftp://ftp.3gpp.org/specs/2008-03/Rel-6/].

Schierl, T. et al, 2005. 3gpp compliant adaptive wireless video streaming using h.264/avc. *Proceedings - International Conference on Image Processing, ICIP 3*, art. no. 1530487, pp. 696-699.

Stockhammer, T. et al, 2003. H.264/avc in wireless environments. *In IEEE Transactions on Circuits and Systems for Video Technology*, 13(7):657–673.

Stockhammer, T. and Hannuksela, M.M., 2005. H.264/avc video for wireless transmission. *In IEEE Wireless Communications*, 12(4):6–13.

Wiegand, T. et al, 2003. Overview of the h.264/avc video coding standard. *In IEEE Transactions on Circuits and Systems for Video Technology*, 13(7):560–576.