Evaluating Video Streaming over GPRS/UMTS networks: A practical case

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***Abstract*— In this paper, we focus on analyzing video streaming service performance on real networks. We propose a non intrusive methodology based on mobile devices as clients, instead of using them as modems. Our objective is to provide a more realistic test environment using actual mobile devices in real conditions of network load and radio propagation while taking into account the intrinsic mobility of mobile subscribers. This solution allows us to follow the end to end performance even when handover between different access technologies is performed.**

Using this methodology we carry out a study of video streaming behavior over GPRS and UMTS networks. Outstanding results related with delays, jitter, lost packets and sequence errors have been obtained. Also other conclusions about video quality, such as PSNR, have been achieved. Moreover, we analyze the impact of mobility issues such as handover or cell reselection.

I. Introduction

Telecommunication networks are continuously evolving. With the introduction of packet data traffic in mobile networks many new services have appeared, such as video streaming, which deliver voice, data, audio and video in an integrated manner. These new services have some special real time requirements. In this paper we evaluate video streaming performance over mobile networks, and study the effects that mobility issues have on this performance.

During trials, off-the-shelf smart phones are used as measurement tools. Since mobile phones act as an interface between subscribers and the network, sampling data from the phone itself becomes critical. This is also an important line of activity due to the evolution of mobile platforms and the processing power and connectivity of the new generation of mobile terminals.

Traditional methods for analyzing mobile data communications' performance in realistic scenarios were based on trials, using mobile devices only as modems with the applications running on a laptop. This configuration has some disadvantages as we need additional hardware for doing measurement. Also, the mobile TCP/IP stack is not evaluated.

Another consequence of network evolution is the coexistence of different radio access technologies, which, depending on their availability, can be used by mobile terminals. B3G (Beyond 3G) systems are expected to be based on the integration of different technologies, such as 3G enhancements and WLAN, in order to offer services as transparently as possible. Vertical handover makes it necessary to measure quality of service from the user perspective instead of focusing on particular technologies. As smart phones offer many communication interfaces such as GSM, GPRS, WCDMA, and WLAN it is possible to measure end-to-end performance even in the presence of handovers or other mobility issues.

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In this paper we obtain real measurements of video streaming service in live GPRS/UMTS networks. We provide a complete methodology to evaluate network performance and video quality in both static and vehicular scenarios. The measurements collected are relevant in order to characterize video streaming services, and other services such as mobile TV, in real cellular networks and vehicle scenarios. During trials we have followed the 3GPP recommendations for streaming service in GPRS and UMTS networks [1].

This paper is organized as follows. In section 2 we introduce streaming protocols and codec. Section 3 describes the methodology and tools used during the field tests. In section 4 we detail the results obtained in the different test scenarios. Video quality is analyzed in section 5. Finally, in section 6, we summarize our findings.

II. Video Streaming: Protocols and Codecs

UDP is the dominant network protocol used for real-time applications. UDP is a connectionless unreliable protocol with no flow control. Mechanisms used in TCP to ensure reliability introduce a huge overhead in packets and a lot of retransmissions in the case of lost or corrupted packets. The overhead introduced by TCP protocol is undesirable for real-time applications such as video streaming.

Streaming protocols used during trials are RTP (Real-Time Transport Protocol) [2] and RTCP (RTP Control Protocol) for media delivery [2], RTSP (Real Time Streaming Protocol) [3] for media control and SDP (Session description protocol) [4] for media description. These protocols provide necessary functionalities for supporting "real-time" services. RTP provides support for applications with time-constraints with mechanisms such as time-stamps, sequence numbering and payload type. RTCP provides out-of band feedback information for an RTP flow. RTSP protocol is used to remotely control a

File	Video			Audio	
	Codec	Bit rate (Kbps)	Frames per seconds	Codec	Bit rate (Kbps)
6	H263	48	3	AAC	16
7	H263	48	3	GSMAMR	16
8	H263	48	3	WBAMR	16
9	MPEG-4	48	3	AAC	16
10	MPEG-4	16	3	AMR	16
11	MPEG-4	16	3	WBAMR	16
12	MPEG-4	42	8.3	GSMAMR	8

TABLE I
3GPP VIDEO SAMPLES

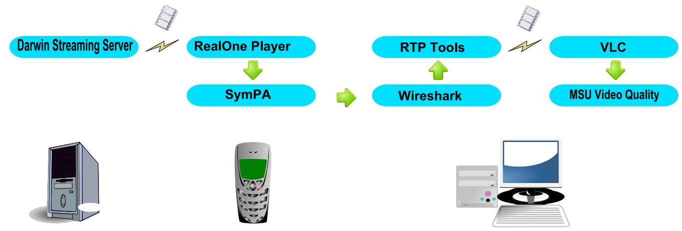

Fig. 1. Tool Chain

streaming media session and SDP is the protocol used to describe streaming media initialization parameters.

3gp is a file format defined by 3GPP and it is based on the MP4 file format. 3gp has been specially developed for third generation mobile devices and stores video encoded streams as MPEG-4 or H.263. Codec used in tests are H.263-1998 [5] an Mpeg4 Visual [6].

The H.263 is a standard of the ITU-T. It is a codification method used for compressing moving images with low bit rates. It supports five standardized formats of image: sub-QCIF (88x72), QCIF (176x144), CIF (352x288), 4CIF (704x576) and 16CIF (1408x1152). The video codification compresses the information using the estimation techniques and movement compensation, to eliminate the temporary redundancy between consecutive photograms. In addition, it uses DCT (Discrete Cosine Transform) to remove the space redundancy that there is in each photogram.

MPEG4 is a standard of the ISO/IEC. It was initially designed for transmission of video at low bit rate, below 64Kb/s, although its capacity and utility was expanded later. It is object oriented and each object is independently codified, which provides a high flexibility for composition of the scenes. This standard is robust to errors and it provides a codification with high compression.

III. METHODOLOGY AND TEST SCENARIOS

Our measurements have been carried out within Symbian OS based smart phones and the Nokia Series 60 platform running RealOne Player streaming client. The server is connected to the Internet via a high speed internet access. Mobile devices connect to the Darwin Streaming Server [7] via live GPRS/UMTS networks.

The 3gp samples used during tests have been obtained from the Catra Streaming Platform [8]. The properties of these samples are shown in table I .

In order to analyze the RTP traffic we have used the tool chain shown in figure 1. In this way we can compare the original video with the video received on the client side.

Each test starts with the video transmission from the Darwin Streaming Server to a streaming client located in the mobile device. We use the SymPA [9] tool for capturing incoming traffic in the mobile client, and other parameters such as cell Identifier, RSSI (Received Signal Strength Indication) and battery consumption. Traffic captured in this way is converted to tcpdump format and it is analyzed with the well-known network protocol analyzer WireShark (previously known as Ethereal). The code of WireShark has been modified in order to add the capacity of analyzing RTP flows established using RTSP protocol. Jitter, delays, lost packets and sequence errors are obtained with the protocol analyzer.

In order to compare the transmitted and received videos, and to analyze the degradation suffered, we export, from Wireshark, the packet sequence of the received video, in rtpdump format.

Next, we use the rtpplay tool, from the RTP Tools package 1.18, to replay the sequence of packets as was received on the client side. In this case, the VLC streaming client acts as the receiver. This client runs in the same computer that the stream has been replaying.

We use a modified version of VLC 0.8.5 without detection of the end of transmission when there is a large number of packet losses. That mechanism makes VLC and the RealOne Player client of the mobile terminal behave differently, which is unacceptable for a reliable analysis.

The video played by VLC is stored in lossless AVI format, so that at the end of the process we obtain an identical copy to the one received in the mobile terminal, which is suitable for later processing and comparative analysis. For this analysis we used the MSU Video Quality measurement tool, to obtain PSNR values.

IV. EXPERIMENTAL RESULTS

Trials have been carried out on commercial Spanish public GSM/UMTS networks with the interactive Quality of Service class [10]. Interactive class is mainly meant to be used by traditional Internet applications, such as web browsing, network gaming or database access while conversational and streaming classes are mainly intended to be used to carry real-time traffic flows. An interactive bearer is not suitable for video streaming service because it gives no quality guarantees. The actual bearer quality will depend on the system load and the admission control policy of the network operator. Moreover, delay, bit rate and packet loss attributes are not guaranteed in this class.

There are many problems for streaming over cellular networks, but the most relevant are bandwidth, jitter and losses. We focus on these three issues. Bandwidth depicted in figures is calculated as the number of bytes received on the client

Parameters	Operator 1	Operator 2	Operator 3
Traffic class	Interactive	Interactive	Interactive
Max Bit Rate(dw/up) (Kbps)	384/384	384/384	384/384
Guaranteed Bit Rate (Kbps)	0/0	16/64	64/384
Transfer Delay (ms)	0	768	1000
SDU error ratio	0.001	0.001	0.001
Maximum SDU size	1500	1500	1500
Delivery order	No	No	No
Residual bit error ratio	0.00001	0.00001	0.00001
Delivery of erroneous SDUs	no	no	no
Traffic handling priority	level 2	level 1	level 1

TABLE II

UMTS Bearer Service Attributes

side in the last second. The maximum size of the IP packet used during trials is 1478 bytes, fragmentation has not been appreciated during field tests. Measurements are collected from 3 commercial UMTS network. In table II we can see QoS parameters provided by the different operators.

A. *Static Scenario*

In the static scenario, measurements are carried out in an indoor environment.

1) Packet losses: Figure IV-A.2 shows the observed packet losses during a typical streaming session without handover conditions. In a static scenario measured packet losses are on average 0,58% of the total transmitted packets.

2) Out-of-order delivery: Out-of-order delivery is a common problem detected during all the trials. This problem makes necessary the application of additional mechanisms for rearranging out-of-order packets on the client side. The out-of-order delivery problem is detected because it generates sequence errors in the RTP flow captured.

During tests, it has been observed that sequence errors are related to the size of the packet, as bigger ones can be delayed in network nodes. The maximum IP packet size is 1478 bytes, packets with this size are always marked with a sequence error, as we can see in figure IV-A.2. These packets are generated when a frame is split into several packets. In the traffic capture

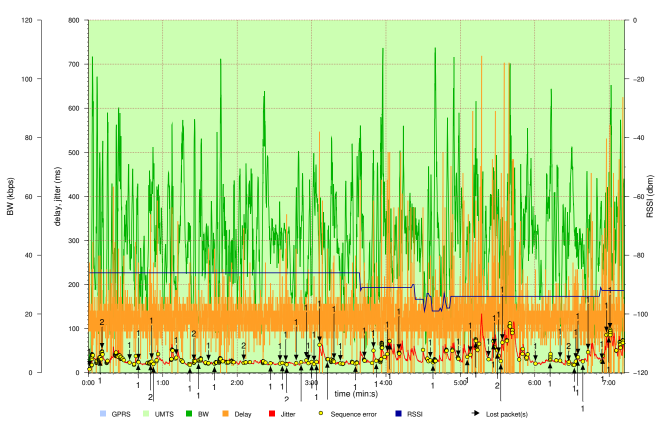

Fig. 2. Packet losses in a static scenario without cell reselection

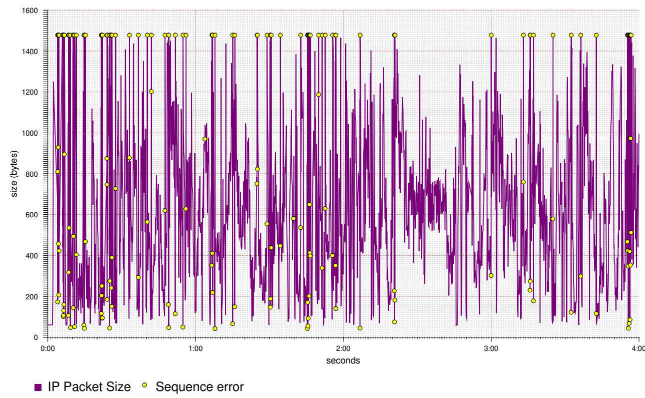

Fig. 3. Sequence errors during a streaming session

shown in figure IV-A.2 some frames are sent into two packets, the first packet has the maximum size allowed and the second has an average size of about 400 bytes. Video used in this trial is encoded with MPEG4. The same experiment, using H263 codec, provides a lower rate of sequence errors because packets belonging to the same frame are similar in size, so that the size of the resulting packets is lower.

3) Delay Variation: The Video Streaming service is sensible to delay variations (jitter). During trials the jitter obtained on media is 20 ms, and the maximum is 180 ms. In figure IV-A.2 jitter and delays between packets during a typical streaming session are shown. Video 12.3gp has been used in this session. Inter packet delays of several seconds, and jitter increment, are obtained in a low percentage of received packets. In these cases packet arrival rates increase due to accumulation of packets during the delay. An explanation of this behavior can be found in the error control mechanism on the link level used in radio access protocols. In the situation previously described, these protocols can cause delay in the delivery of packet until previous packets have arrived without errors.

Figure IV-A.3 shows the distribution of the end-to-end delay, 95% of the received packages are concentrated in a rank of 120ms. To increase the percentage of samples to 99%, the range of delays has a width of 800 ms. This information is useful in the context of buffer sizing based on the percentage of permissible losses by overflow.

B. *Vehicular Scenario*

Trials have been performed in an urban and inter-urban environment. The mobile client moves by car at an average speed of 100 km/h.

1) Packets Losses: In this section packet losses are analyzed during a streaming session in a vehicular scenario. Figure IV-B.1 depicts packet loss bursts due to cell reselections in GPRS and during handover between GPRS and UMTS. In UMTS, soft handover allows seamless handover. During soft handover, a mobile station is in the overlapping cell coverage area of two sectors belonging to different base sta-

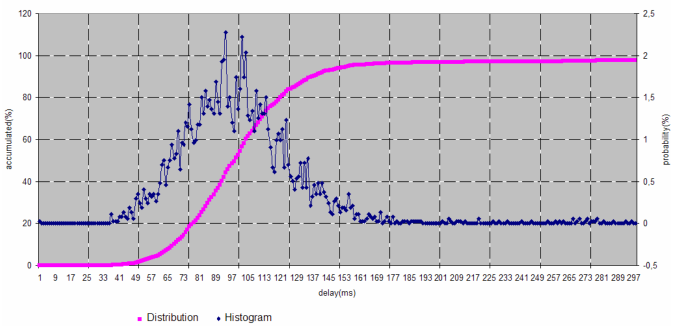

Fig. 4. End-to-End Delay

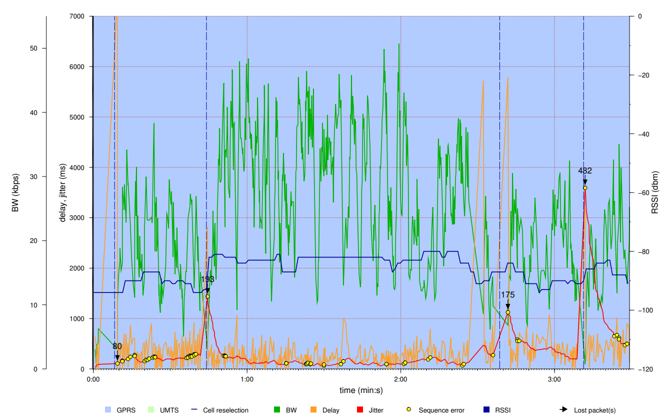

Fig. 5. Lost packets due to a GPRS cell reselection

tions. Communication between the mobile station and the base station takes place concurrently via two air interface channels from each base station separately. In GPRS the macrodiversity cannot be used and communication is interrupted. When interruptions are lengthy, the buffers in network elements can overflow, and may result in packets loss.

Figure IV-B.1 shows a UMTS cell reselection. During the cell change a burst of packet loss takes place. 20 seconds before a cell change takes place, RSSI begins to decrease. Bit rate reduction can also be appreciated. This information can be used to implement predictive mechanisms in order to avoid packet losses. Soft handover has not occurred in any of the test cases.

Figure IV-B.1 shows a burst of lost packets due to a temporal loss of reception while the vehicle passes through a tunnel. In this case the session concludes abruptly.

V. VIDEO QUALITY

In the previous section we have observed that during handover between different access technologies and during handover packet losses, burst takes place. In this section we analyze the impact this phenomenon has on video quality. We use the tool chain described in section III.

The MSU Video Quality measurement tool is used to obtain three parameters traditionally used to evaluate the video quality. These parameters are: PSNR (Peak Signal-to-Noise Ratio) and Delta metric.

We have compared the video 6.3gp in two different situations, the first is when the mobile is in a static scenario and no packets are lost, in the second one the mobile is moving but no cell reselections have taken place. In the second case the number of lost packets is reduced to 43, compared with handover conditions. The following figures show the comparison between original frames and received frames. The number of frames in the mobile scenario is lower than in the static scenario as some frames cannot be rebuilt because of lost packets.

In figure V the values of PSNR in the static scenario are higher than in the vehicular scenario, which means that the frames received in the first case are more similar to the original frames. The frame with the lower PSNR is shown in figure 9 for the second case. It is a mixture of different images caused by lost packets.

Figure V shows delta calculated. A low absolute value of the delta parameter means that the frames are more similar. It

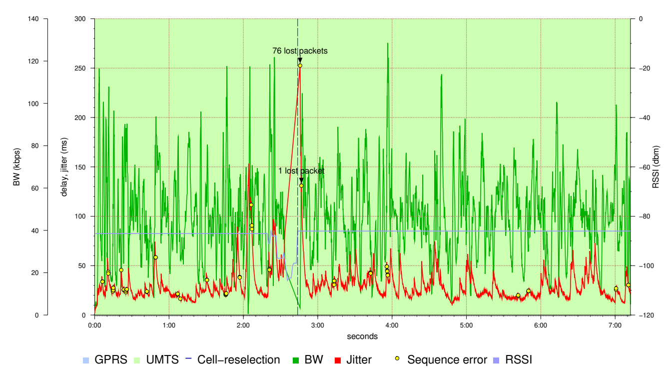

Fig. 6. Lost packets due to UMTS cell reselection

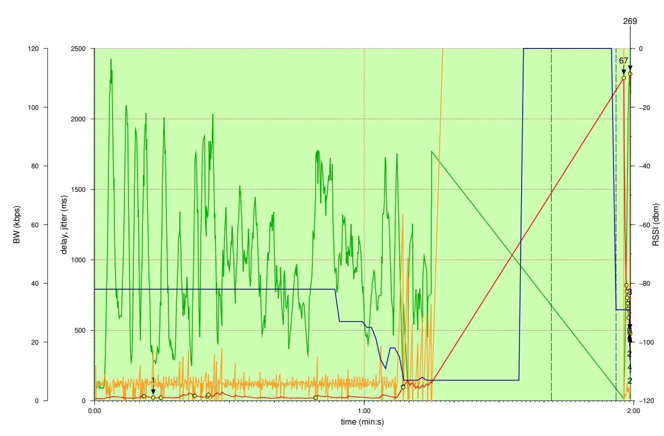

Fig. 7. Lost packets because of a loss of coverage

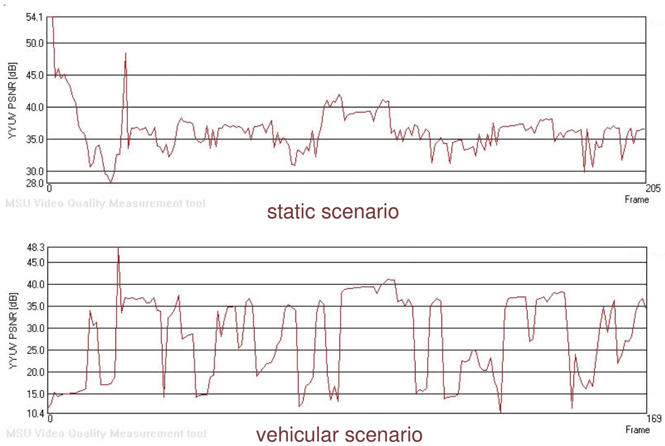

Fig. 8. PSNR

Fig. 9. Frame with the minimum PSNR

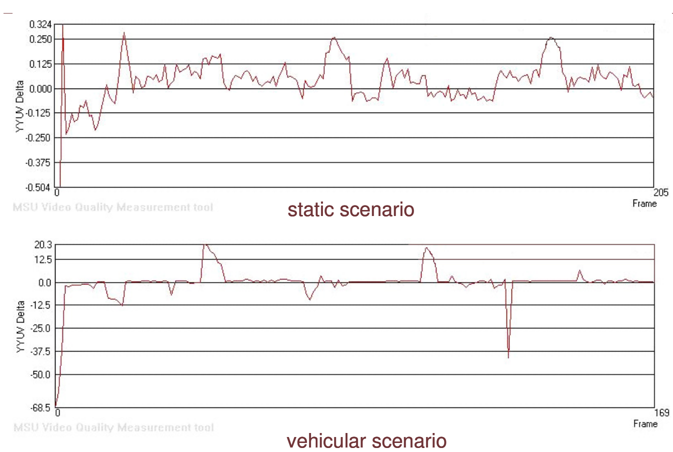

Fig. 10. Delta

can be seen that when the mobile is moving, values obtained are higher than in a static scenario.

VI. Conclusions

The lower bit rates available in GPRS networks results in a worse video quality experience due to rebufferings [11] that take place in the middle of streaming sessions. Moreover video streaming service accessibility in GPRS is also lower than in UMTS networks because on numerous occasions during experimental tests, the streaming session is not able to run due to lack of bandwidth availability. In GPRS low bit rate videos need to be used in order to obtain higher accessibility rates.

In UMTS and in static conditions, the results obtained are very promising, facing the deployment of new services such as IPTV. In the described conditions packet losses are occasional and close to zero. The maximum jitter obtained in this scenario is 250 ms.

In vehicular scenarios packet losses are mostly due to UMTS handovers. Packet losses may involve an unacceptable degradation in video quality. Sometimes, and due to loss of coverage, the video streaming session is suddenly aborted.

In such conditions we have observed that RTP flow control mechanism based on feedback messages (such as RTCP) about the quality of link is not suitable for a cellular environment. When the data link is affected by the lossy nature of the air interface control link is affected too, then in this situation feedback information fails to reach the server side and connection is lost.

In the testing activity we have observed that before handovers or temporal coverage losses, the signal strength and bit rate received decreases. This behavior can be used in order to design a cross-layer mechanism to control RTP data flow during a horizontal and vertical handover and provide RTP flow with the capacity to support intermittent connection loss in a vehicular scenario. In this way we can detect possible service degradation a few seconds before it happens and modify the transmitted coding to a lower bit rate in order to buffer enough frames to compensate temporal disconnections.

Communication awareness is decisive for the success of new multimedia entertainment services [12], therefore in future work, we are going to develop a framework to provide context adaptability to mobile applications.

Acknowledgment

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