QoS Analysis of Video Streaming Service in Live Cellular Networks

Almudena Díaz[a], Pedro Merino[a], F. Javier Rivas[a]

[a] *Dpto. de Lenguajes y Ciencias de la Computación,*
University of Málaga
Málaga, Spain

Abstract

The increasing computing and communication power of new terminals in mobile networks has converted these smart phones into Internet hosts. They can be used for traditional services, such as browsing or emailing, but also as platforms for specific new IP multimedia services like Push to Talk, streaming or IPTV. However, developing new services or porting existing popular ones should be done considering the target network in which the services will be deployed.

This paper presents an analysis of video streaming, a service with real time requirements, on live mobile networks to validate current deployments. In this analysis we propose using smart phones as measurement devices in a way that enables them to capture customers' experiences. An outstanding contribution of this paper is that real measurements are carried out in real networks, with real transmission power, real antennas and real network operators and protocol stacks. These measurements help to complete performance results obtained in simulations and emulations carried out in laboratories.

Keywords: cellular networks, performance analysis based on traffic monitoring, RF measurements, QoS in streaming

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1. Introduction

Multimedia content transport over mobile networks has been widely studied in recent years [35]. In particular, streaming services have attracted much attention due to the possibility of offering services such as TV on mobile phones.

The 3rd Generation Partnership Project (3GPP) has standardized protocols and codecs used for the deployment of streaming services over cellular networks. In particular, the technical specification [15] provides a complete standard for packet-switched streaming (PSS) services. The protocol adopted in this specification for data transport is the Real time Transport Protocol (RTP) [16] over UDP/IP. RTP protocol provides end-to-end delivery services for real-time traffic. The most important functionalities provided by RTP are synchronized data delivery, sequence numbers which are used to detect lost and out of sequence packets, and stream identification, making it possible to have more than one media stream in the same RTP session. However it should be noted that the RTP protocol does not guarantee reliable, timely, nor in-order delivery of packets, as it assumes reliability in underlying networks. Thus, the deployment of the video streaming service using RTP/UDP/IP protocols in an unreliable environment such as cellular networks faces three main challenges:

- The great variability of available bandwidth, as a result of changes in the network load and radio channel fluctuations.
- The large variation in packet transfer delays, that needs to be controlled to achieve a constant reproduction flow and avoid service interruptions due to buffer starvation, which would severely downgrade the quality of service perceived by final users.
- The packet losses caused by the special nature of wireless communications. In our experiments mobility procedures such as handovers have been identified as the main source of packet losses. This is because of the preceding reduction of signal strength (RSSI) and quality (SIR) and of interruptions originated by the procedure itself.

So it must be noted that the deployment of streaming services over cellular networks with a high level of quality is a key issue that requires a complex methodology, given that a large set of factors must be taken into account.

Streaming performance has been analyzed at many levels [36] [34] [37] [39] [42], considering parameters belonging to different layers of the stack defined by 3GPP [28]. In this article, we propose a new methodology which covers some performance issues not analyzed in previous field test work. Specifically, we evaluate the impact of radio propagation issues over RTP/UDP/IP traffic in live cellular networks using SymPA [1], a tool specifically developed for analyzing IP traffic performance over cellular networks.

SymPA is a software tool which runs in commercial smart phones. It enables capturing data traffic, monitoring radio parameters and accessing QoS parameters negotiated during the establishment of the data connection in the UMTS (Universal Mobile Telecommunications System) network, amongst others features. The passive monitoring of streaming services on current mobile devices using SymPA provides important metrics regarding the performance of this service over cellular networks, such as throughput, jitter and packet losses, and allows us to analyze the impact of RF conditions over said networks.

The contributions of this paper are twofold:

- Correlation of real application level measurements and RF parameters to evaluate RTP traffic performance. This enables current mobile developers and operators to take advantage of the results, as the application and RF level measurements are more accessible for them to implement adaptive techniques.

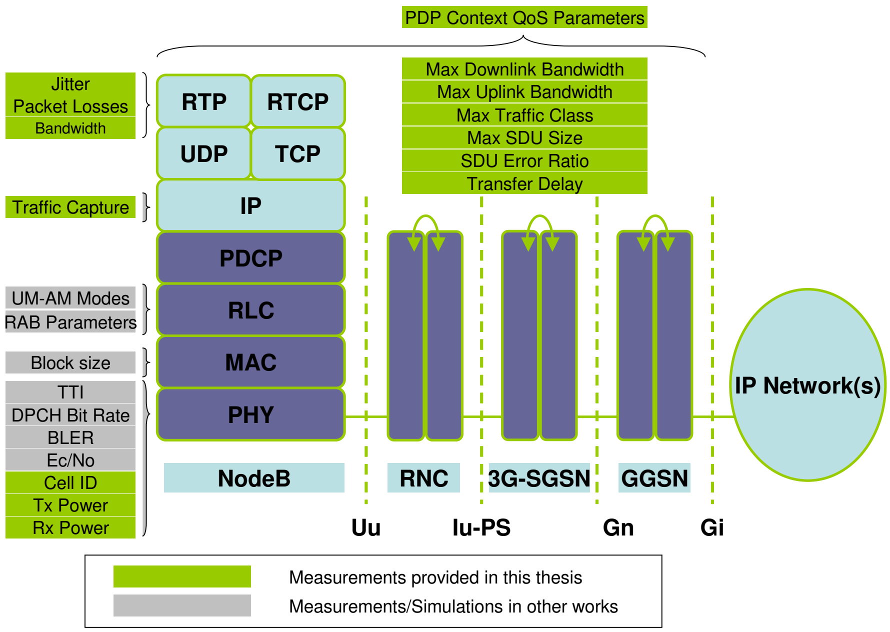

Figure 1: Packet-Switched streaming service protocol stack

- The use of an open methodology and tools. Thus, anyone interested will be able to use the methodology presented to reproduce the results or to test other applications.

The paper is organized as follows. Section 2 provides a detailed state of the art overview of previous streaming performance work and measurement tools for cellular networks. In section 3 we introduce the methodology used in this paper to carry out a comprehensive analysis of the performance of streaming services over live cellular networks. The results obtained during experimental tests are analyzed in section 4. The main streaming performance parameters such as packet losses are analyzed and compared with previous results and in different scenarios. Finally, conclusions are presented in section 5.

2. Related Work

In subsection 2.1 we introduce generic tools used to evaluate general service performance in cellular networks. In subsection 2.2, which specifically focuses on streaming services, we summarize the main contributions of related works in the evaluation of video streaming services in cellular networks.

2.1. Measurement tools in cellular networks

Traditionally performance in cellular networks has been measured using information provided by network elements such as performance data provided by the radio network controller (RNC), the mobile switching center (MSC) and node-B. This information can be used to optimize the core network, but a correct optimization of the network's internal performance does not involve the optimization of the quality of service perceived by final users. Parameters obtained from internal probes are very difficult to map into parameters related with the quality of service perceived by users since counter values obtained from network elements do not reflect the customer's experience, as highly sophisticated filtering and correlation functions are not implemented in static functions of network elements [4].

In consequence, more accurate measurements of the quality of service perceived by final users are necessary. To obtain measurements which fully capture the experience of mobile users, we need to use measurement points more closely. In this sense the most extended methodology used for measuring QoS perceived by users is based on the use of GPRS/UMTS/HSPA cards or mobile devices as modems [3][5] connected to laptops where applications under test are executed. Results obtained from this approach allow us to measure the occurrence of problems which actually affect the quality of service perceived by customers. For example, performance parameters such as throughput or packet losses can be evaluated by using traditional protocols analyzers.

The main restriction of this methodology is that applications specifically designed and optimized for execution in mobile devices are not tested because this kind of application can not be executed in laptops. The use of emulators is also discouraged for measurement purposes. Emulators help debug the conceptual performance of the application, but to analyze hardware performance and the performance of protocols and data connections, the application must be executed directly on mobile phones.

Mobile phones are devices with limited memory and processing power. The design of its protocol stack also differs a lot from stacks and protocols used in desktop computers. We have studied some traces obtained from Symbian mobile devices. The results have been very revealing, indicating that the protocol stack of Symbian mobile devices is designed specifically for mobile environments. Symbian mobile phones incorporate several specific RFC which improve the performance of TCP/UDP/IP applications over cellular networks. Some of the main extensions implemented in the stack of these mobile devices are RFC 1323, TCP Extensions for High performance, RFC 2581 TCP Congestion Control, RFC 3042 Enhancing TCP's Loss Recovery, RFC 2001 TCP Slow Start Algorithm and RFC 1144 Compression of TCP/IP headers for low speed links.

Furthermore mobile applications are becoming more and more sophisticated. Next generation applications are envisioned to manage different PDP (Packet Data Protocol) contexts with different QoS profiles. To manage different PDP contexts using measurement tools running on laptops, we need one serial port for each PDP context established and for each data connection and normally, mobile devices allow only a limited number of serial ports. Additional serial ports are required if it is necessary to monitor the radio signal strength level. For example, in order to monitor PoC (Push-to-talk over Cellular) services based on SIP protocol, it is necessary to open a primary context for flow control with an interactive profile and a secondary context for RTP traffic with an associated streaming profile. So that if we want to monitor traffic produced for this application, the status of PDP contexts and also the radio signal strength needed to use five serial ports is not viable. This stems from the tradition of the serial modem port which switches to data mode after the connection is set up. If we want to monitor the PDP context status or the signal strength, or any other parameter, we have to open an additional serial port for this issue. Therefore, due to implementation restrictions in the computer-modem interface (see [6]), it is not possible to reproduce the same scenario that can be found in applications running inside mobile terminals, which can handle and monitor multiple PDP contexts at the same time.

Due to these restrictions, new measurement and monitoring tools have been developed for execution on the shelf mobile phones used by customers. This new generation of tools allows the monitorization of the quality of service perceived by user, and also provides information about the distribution of users, location information and behavior patterns. Qualipoc [7] is a measurement solution developed by SwissQual that can be configured for specific service test such as voice and video call, messaging, data, browsing and video streaming. The customer experience manager developed by the company mFormation [8] provides passive data-service availability monitoring and active data-service monitoring. The main difference between these tools and the tool used in this paper is that they do not provide IP traffic level measurements, which is key in the performance analysis of IP protocols. Nemo-Q [9] is also centered on passive service measurements, which periodically sends reports to a server where information is analyzed. Nemo Handy is a tool of the same company which enables active and passive monitoring and, at the same time, all the information collected is displayed in real-time on handhold devices. These tools provide low level information or are based on specific test for specific services. Finally QXDM (Qualcomm Extensible Diagnostic Monitor) [10] is a proprietary tool developed by Qualcomm for specific test terminals which focuses on monitoring radio interfaces. It is not oriented to the analysis of IP traffic.

All these measurements tools are oriented to mobile operators and focus on passive service level measurements such as connectivity or network usage. Additionally, they only allow the analysis of specific applications, such as Web browsing, FTP, or e-mail, which are built in the monitoring tool itself. Moreover, these tools center on isolated layers of the protocol stack, while SymPA, as we will see in the following section, correlates information from different layers.

2.2. Streaming measurements over cellular networks

The majority of papers related to the evaluation of the performance of video streaming services over cellular networks are based on emulation [36] or simulation results [37] [39] [42].

An interesting simulation study is carried out in [34]. In this paper measurements focus on RLC layer configurations in order to optimize streaming performance. They identify parameters with an impact over the service to configure them, proposing self-adaptive techniques [33]. They conclude that the performance of RLC is better when acknowledged mode is used, and they also analyze the impact of RLC block size on the end-to-end video frame delay. But mobility issues such as handovers are not considered in the analysis.

The contributions of paper [32] in the field of measurements in live networks are remarkable. Accessing device proprietary information, RF conditions and RAB (Radio Access Bearer) assignments are analyzed in order to determine the impact on the quality of video of these parameters. The performance results provided show a correlation between re-buffering events and the degradation of RF conditions as well as RAB changes. However, the impact of handover in a mobility scenario is not covered. In addition, the measurement tools used are not described, and thus the proposed assessment methodology is not available for rest of the scientific community.

The experimental evaluation of streaming media is also the focus of paper [38]. The authors center their study on evaluating the impact of application layer and transport layer protocols on mobile streaming media. They conclude that TCP streams perform significantly worse than UDP streams. A mobility study is carried out and a complete description of tools used is also provided. Laptops were used to perform trials. However, RF measurements are not considered, so neither the impact of radio propagation conditions nor the source of behaviors detected at transport and application levels can be analyzed, as we have done in our study.

Video streaming performance tests over live networks were performed in more recent work [44]. Although the test approach of that work is similar to ours, with the measurement tool running on a mobile device, the information provided by the tools differs significantly. In respect to the radio aspect, they access very low level information using special terminals, but regarding data exchange their tool only provides access to traffic volume, but not to the actual data received. According to the classification in [43], that application would belong to the "Field measurement tool" type, where parameters such as jitter and packet losses are not measured.

As we will see in this paper, handovers and cell reselections are the most important source of packet losses in mobility scenarios. This issue is partially covered in [31] where real measurements in a controlled scenario are presented. Lundan and Curcio simulate the impact of soft and softer handovers during a streaming session and conclude that soft and softer handovers are seamless, so the packet loss rate is close to 0%, even in the presence of these handovers. Hard and inter-system handovers are not evaluated, although they are expected to have a greater impact.

Figure 1 shows PSS protocol stack and parameters analyzed at each level in the related workw. We also depict parameters analyzed in the methodology introduced in this paper. The main difference is that our methodology enables traffic capturing in mobile devices and the analysis of transport parameters such

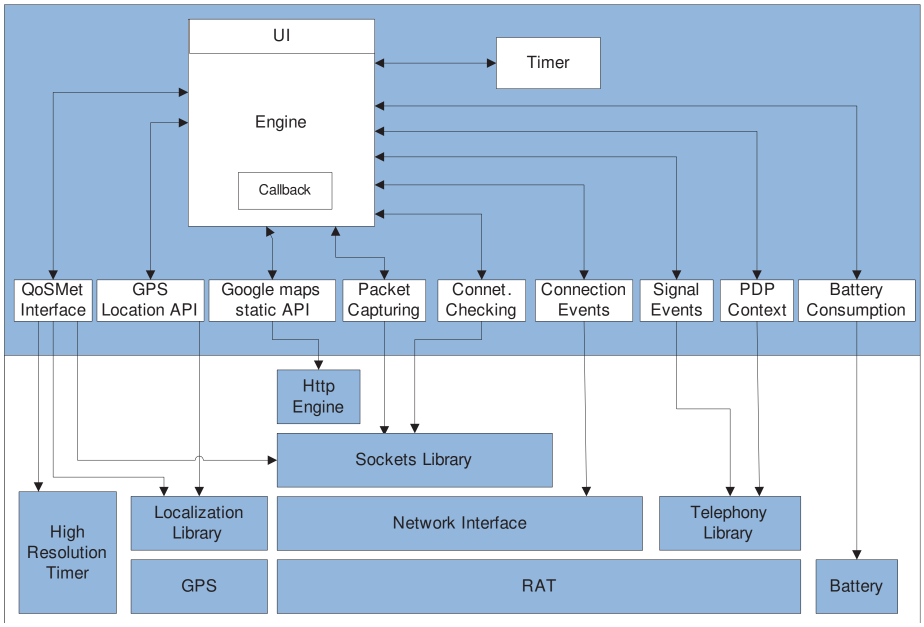

Figure 2: SymPA Profiling Tool

as jitter, packet losses and bandwidth. Furthermore, the impact of handover over IP traffic can be analyzed with our methodology.

3. Our proposal: an open methodology for evaluating the performance of IP services over mobile networks

3.1. SymPA: a monitoring tool for live networks

Measurement performance of the streaming service was monitored using a profiling tool developed for us. SymPA is a software tool which runs on commercial Symbian Series 60 smartphones. It allows monitoring the performance of any application running on the mobile device over any wireless interface available on it. The main functionalities provided by SymPA are IP traffic capturing, air interface monitoring (radio access technology in use, cell reselections, radio signal strength indicator (RSSI), transmission level, etc), measurement localization and battery consumption profiling. Figure 2 summarizes the main features provided by SymPA, further information can be found in [1]. In accordance with the classification of QoS and QoE (Quality of Experience) monitoring tools in UMTS cellular networks carried out in [43], SymPA can be labeled as a "Mobile QoS Agent".

In order to discard undesired side effects of the monitoring tool, we provide statistical results of the performance of our tool in terms of CPU usage, RAM memory and power consumption. These tests confirm that executing SymPA

	N	Range	Minimum	Maximum	Mean		Std. Deviation	Variance
	Statistic	Statistic	Statistic	Statistic	Statistic	Std. Error	Statistic	Statistic
POWER (W)	3328	1,433	,169	1,602	1,45825	,002523	,145534	,021
CPU LOAD (%)	3302	99	1	100	32,88	,129	7,409	54,891
MEMORY (bytes)	827	2895872	30408704	33304576	33104565,40	11865,187	341214,395	1,164E11

Figure 3: Power, CPU and memory consumption during a streaming session

	N	Range	Minimum	Maximum	Mean		Std. Deviation	Variance
	Statistic	Statistic	Statistic	Statistic	Statistic	Std. Error	Statistic	Statistic
POWER (W)	3792	1,488	,214	1,702	1,47204	,003287	,202386	,041
CPU LOAD (%)	3781	99	1	100	39,42	,158	9,697	94,038
MEMORY (bytes)	947	9224192	30547968	39772160	38619635,56	25853,784	795607,939	6,330E11

Figure 4: Power, CPU and memory consumption during a streaming session while SymPA is running

on mobile devices does not interfere with the normal behavior of applications running on the device. We used Nokia Energy Profile (NEP), a tool provided by Nokia, which enabled us to evaluate all these parameters. Performance tests were carried out on a Nokia 6110 mobile device. The ARM processor speed is 369 MHz, the total RAM available to S60 software is around 50 MB and it incorporates a 900 mAh battery. Nokia 5800 features a higher speed processor and also more memory than Nokia 6110. So this terminal was chosen to carry out the performance study.

Two different tests were scheduled. In the first test only Real One player and NEP were running on the mobile device. The screen was switched on. The results obtained are shown in figure 3.

In the second test SymPA was also running. Results are shown in figure 4. The comparison of the results helps us confirm that executing SymPA has a minimum impact on the performance of the mobile device. As regards use of CPU executing SymPA only results in an increase of 6.54%. Similar results are obtained when comparing memory and power consumption. Memory usage overload of 10.5% due to the execution of SymPA and battery consumption increases can be almost discarded.

3.2. Measurement points

As we can see in figure 5, we use SymPA to provide measurements at three key points in order to obtain relevant information about network performance and also about the quality of service perceived by customers. At point 1 we capture the mobile device's incoming traffic. The traffic capturing functionality is deployed at IP socket API provided by the operating system so we can analyze the performance of network, transport and application protocols, as we can see in figure 1. Since this functionality is technologically independent, we can, for instance capture traffic over cellular networks and also over 802.11 networks. The monitor should be able to attach to the different data connections that a mobile device can create, which allows us to evaluate the performance of vertical handover between the different radio access technologies available in mobile devices.

The streaming session initiation procedure in UMTS networks is described in [40]. A PDP (Packet Data Protocol) context is requested when the IP connection is initiated from the streaming client application. The PDP context contains routing information for packet transfers between a mobile station and a GGSN (Gateway GPRS Support Node) to have access to an external packet-switching network. During PDP context establishment, an IP address is allocated and the mobile device applies to specific QoS profile which defines the connection's QoS properties. End-to-End QoS architecture for UMTS network and QoS profiles have been standardized by 3GPP in [23]. Four different QoS classes are considered: conversational, streaming, interactive and background. QoS classes are defined depending on standardized UMTS bearer attributes, which are shown in table 3.2. At point 2 SymPA enables monitoring the negotiated values of these QoS parameters, using the Connection Monitor API provided by the operating system. The status of the PDP context established by applications running on the mobile devices is also monitored. In conclusion information obtained at this level helps us to know the QoS profile and parameters assigned to the data connection opened by the application.

At point 3 we monitor key radio interface-related parameters through telephony APIs: radio access technology (RAT) in use, RSSI evolution and cell information, which allows us to detect cell reselections and handovers, even between different technologies.

Finally, the correlation of all the information collected at different measurement points allows us to detect the source of the problems experimented at the application level by end users.

In accordance with the simplified schema for video streaming used by Curcio and Leon in [35], measurement points used in this paper are located at points labeled as "Real-time transmission" and "Reception", so that mobile network behavior can be characterized with the results obtained.

It is worth noting that this proposed method fulfills the three characteristics identified in [11] for advanced network monitoring: (1) it is capable of performing routine large scale collections of fine-grain measurements (due to the use of customers' devices as measurement elements), (2) it is completely decoupled from the production network and (3) it analyzes and delivers reports and alarms proactively.

SymPA captures all the IP traffic proceeding from network interfaces available in mobile phones. It runs in the background, without interfering in the performance of active applications, and captures all the traffic received through the connections opened by third party applications. Furthermore, this functionality allows the detection of malicious applications by identifying undesired

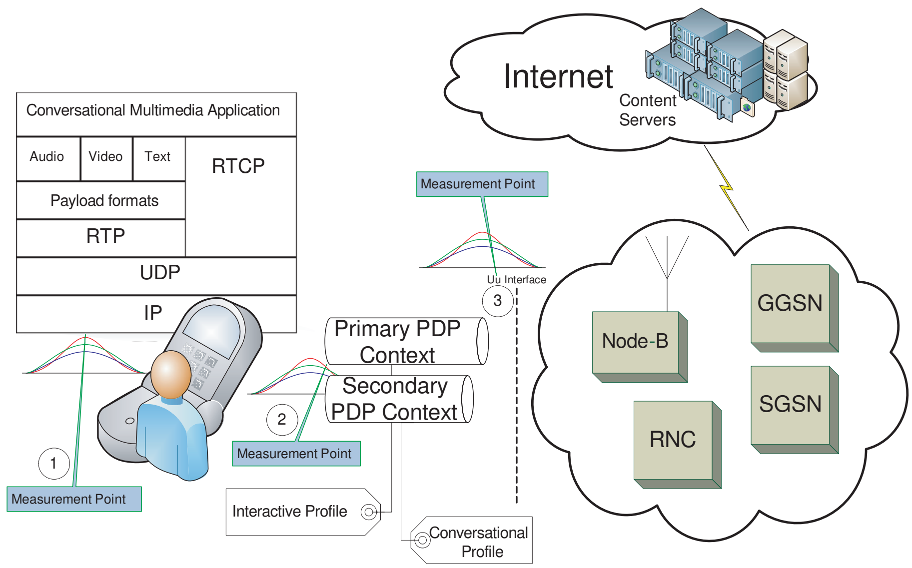

Figure 5: Our approach

connections to unknown hosts.

3.3. Measurement scenario

The measurement scenario is a mixed scenario where we combine fixed and cellular networks. On the streaming client side, we have used different radio access technologies such as GPRS, UMTS and HSDPA. While the streaming server is allocated outside mobile operator networks, it is connected to the Internet via a fixed access of 100Mbps. The experiments were set-up at the University of Málaga. The particular configuration is the following:

- We installed Darwin Streaming server in a personal computer connected to Internet via a high speed connection. We used Wireshark analyzer to capture server traffic.
- We installed a SymPA tool in several Nokia 6110 Navigator and Nokia 5800 mobile devices and used Real streaming client to play videos stored in the configured server. The mobile device was connected to the streaming server via live GPRS/UMTS/HSDPA networks, and the IP traffic was captured using SymPA.
- Statics measurements were carried out in our laboratory.
- In the vehicular scenario, experiments were carried out in a highway in a rural environment with an average speed of 100 kmh.

QoS Parameters	
Traffic class	('conversational', 'streaming', 'interactive', 'background')
Maximum bitrate	(kbps)
Guaranteed bitrate	(kbps)
Delivery order	(y/n)
Maximum SDU size	(octets)
SDU format information	(bits)
SDU error ratio	
Residual bit error ratio	
Delivery of erroneous SDUs	(y/n/-)
Transfer delay	(ms)
Traffic handling priority	

Table 1: QoS Attributes defined in the 3GPP UMTS QoS architecture

- Videos used during trials are shown in table 3. Video 12.3gp has a duration of 7 minutes and video 3.3gp has a duration of 45 seconds.
- The parameters and events gathered during streaming sessions are: bandwidth, jitter (RFC 3550), inter packet delay, packet losses, cell reselection, handover, RSSI (Radio Signal Strength Indicator), transmitted signal level, power consumption and QoS parameters of the packet data connection established during the streaming session.
- Traffic capture was analyzed using Wireshark[19] and Tstat[20], and the data obtained were correlated with radio events monitored. The results are presented in the following section.
- The experiments were carried out for several months at 4 different slot times (early morning, morning, evening and late-evening) for characterizing the average behavior of the service. Experiments have been carried out in two different live Spanish mobile networks.

Common settings used during trials (type of clip, clip length, etc) are specified in [21]. The QoS traffic class provided by mobile operators was the interactive one, which is not the most suitable for streaming services, since delay, bit rate and packet loss attributes are not guaranteed. However, nowadays mobile operators only provide background and interactive traffic classes.

4. Analysis of Results

The objective of this section is to prove the applicability of the method and tool described in previous sections, using as a case study the streaming service, an increasing interest application in recent years.

	3.3gp		12.3gp	
	Video	Audio	Video	Audio
Codec	MPEG4 Visual	ARM	MPEG4 Visual	ARM
Bit Rate	74 kbps	13 kbps	41 kbps	8400 bps
Bit Rate Mode	CBR		VBR	
Frame Rate	25 fps		8 fps	

Table 2: Video used during trials

Medium	**Application**	**Degree of symmetry**	**Data rate**	**Key performance parameters and target values**		
				Start-up Delay	Transport delay Variation	Packet loss at session layer
Audio	Speech, mixed speech and music, medium and high quality music	Primarily one-way	5-128 kb/s	< 10 sec	< 2sec	< 1% Packet loss ratio
Video	Movie clips, surveillance, real-time video	Primarily one-way	20-384 kb/s	< 10 sec	<2 sec	< 2% Packet loss ratio
Data	Bulk data transfer/retrieval, layout and synchronisation information	Primarily one-way	< 384 kb/s	< 10 sec	N.A	Zero
Data	Still image	Primarily one-way		< 10 sec	N.A	Zero

Figure 6: End-user Performance Expectations-Streaming Services 3GPP TS 22.105

Medium	Application	Degree of symmetry	Data rate	Key performance parameters and target values		
				End-to-end One-way Delay	Delay Variation within a call	Information loss
Audio	Conversational voice	Two-way	4-25 kb/s	<150 msec preferred <400 msec limit Note 1	< 1 msec	< 3% FER
Video	Videophone	Two-way	32-384 kb/s	< 150 msec preferred <400 msec limit Lip-synch : < 100 msec		< 1% FER
Data	Telemetry - two-way control	Two-way	<28.8 kb/s	< 250 msec	N.A	Zero
Data	realtime games	Two-way	< 60 kb/s Note 2	< 75 msec preferred	N.A	< 3% FER preferred, < 5% FER limit Note 2
Data	Telnet	Two-way (asymmetric)	< 1 KB	< 250 msec	N.A	Zero

Figure 7: End-user Performance Expectations-Conversational/Real-time Services 3GPP TS 22.105

The use of SymPA helps us to access interesting data, like the duration of handover between GPRS and UMTS and key parameters such as bandwidth, jitter and RSSI. Trials have been carried out in two different scenarios: a static scenario and a vehicular scenario, described in section 3.3.

Figure 6 shows the values extracted from the technical specification [22] about Quality of Service (QoS) requirements from the end-user point of view. A streaming service is included in the category of a one-way service. This kind of service involves no conversational element, which means that the delay requirement will not be so stringent and delay variation (jitter) should be lower than 2 seconds. The most restrictive parameter is packet loss. In accordance with this recommendation the packet loss ratio should be lower than 1% for audio and lower than 2% for video. Out-of-sequence packets are also analyzed. Reordering metrics are specially relevant for real-time media streams. The extent of reordering may be sufficient to cause a received packet to be discarded by functions above the IP layer [25].

In figure 7 we also show QoS requirements for conversational and real-time services which imply full-duplex communication, so requirements are more stringent than the previous ones. In fact, due to the long delay incurred in even the latest video codecs, it will be difficult to meet these requirements [22]. Intermediate reference values have been taken from [26] where two main types of video traffic are defined: interactive-video (videoconference) and streaming video. Table 3 summarizes the parameters recommended by Cisco as reference for these two different kind of video services.

	Losses	One way latency	Jitter
Interactive-Video	< 1 %	< 4/5 sec	30 msec
Streaming-Video	< 5 %	< 150 msec	n/a

Table 3: Reference parameters for video streaming services recommended by Cisco

% lost	Operator 1 static	Operator 2 static
GPRS	9,66	13
UMTS	0,23	0,122
HSDPA	0	0

Table 4: % Packet losses in a static scenario

Following [43] one of the most relevant, end-user KPIs (Key Performance Indicators) identified for audio and video streaming is the number of breaks during service delivery. These breaks are produced by buffer underflows which, mainly, originates from packet losses or from a decrement in the link's available bandwidth. Likewise, the monitoring of available bandwidth is vital because end-user experience for streaming depends a lot on the bit rate, which varies in and between networks.

4.1. Packet losses

In mobile networks packet losses are due to buffer overflows or link-level errors. During our trials in static scenarios we detected sporadic packet losses which have no impact on the play out of the video. The packet loss rates obtained in static scenarios for UMTS and HSDPA radio access technologies match the results obtained in [32] and [31]; 1% of packet losses specified in [22] is achieved in the static environment. In GPRS networks packet losses are higher, reaching an average value of 13% for one of the mobile networks tested.

Results obtained in the vehicular environment are very poor from the point of view of end users, and are higher than 1% specified in [22]. In the figures, packet losses are marked with arrows, cell changes are marked with vertical

% lost	Operator 1 vehicular no cell changes	Operator 2 vehicular no cell changes	Operator 1 vehicular cell changes	Operator 2 vehicular cell changes
GPRS	n/a	n/a	32,35	18,20
UMTS	0,2	1,78	2,16	3,68
HSDPA	1,05	1,14	5,74	4,18

Table 5: % Packet losses in a vehicular scenario

mean jitter	Operator 1 static	Operator 2 static
GPRS	123,12	107,78
UMTS	47,43	54,63
HSDPA	34,30	32,95

Table 6: Mean Jitter in a static scenario

lines and sequence errors are marked with a yellow dot. Jitter, bandwidth and RSSI are also depicted. Figure 8 and figure 9 show that we have obtained high percentages of packet loss due to connection interruptions for several seconds. If the playout buffer of the streaming client is shorter than handover duration, the streaming session finishes abruptly. So, at this point it is essential to use tools like SymPA to characterize the duration of inter-network handovers and to use this information for application buffer dimensioning.

In mobility scenarios the packet loss rate is higher than the one obtained in [32] because trials were carried out on a highway, with a medium speed of around 100 km/h. In this scenario bursts of packet losses are associated to handovers and RSSI degradations. The amount of packet loss depends on the number of handovers and their duration, and the same applies to RSSI degradations. Average packet loss was around 30%. In the section on mobility we analyze the impact of cell reselections and fading in more detail.

Finally, attending to radio access technology, packet losses are higher in GPRS, as expected. In addition, the extension of the bursts is more pronounced in GPRS networks than in UMTS networks, due to the macro diversity support available in UMTS throughout softer and soft handovers. However, as shown in figure 8, cell changes in UMTS also produced bursts of lost packets. For HSDPA, we obtain high rates of packet losses in comparison with UMTS due to lack of soft-handover support.

Furthermore, figure 9 shows the burst of packet losses due to an inter-system handover between UMTS and GPRS.

4.2. Jitter

Jitter results are obtained by applying metrics provided in [16] to the captured RTP traffic . Average jitter for different scenarios and operators is provided in table 6 and table 7.

Results show that jitter is not a problem for one-way streaming services; however, for interactive streaming only HSDPA in a static scenario provides results close to the values recommended by [26], although they are still very far from those recommended in [22].

For a better understanding of the influence of jitter on streaming traffic, in figure 10 we compare outgoing traffic patterns on the server side and incoming traffic patterns on the client side. In the figure we can see that jitter produces a strong modification in the arrival rate of IP packets. The distortion produced must be accommodated by the correct dimensioning of playout buffer.

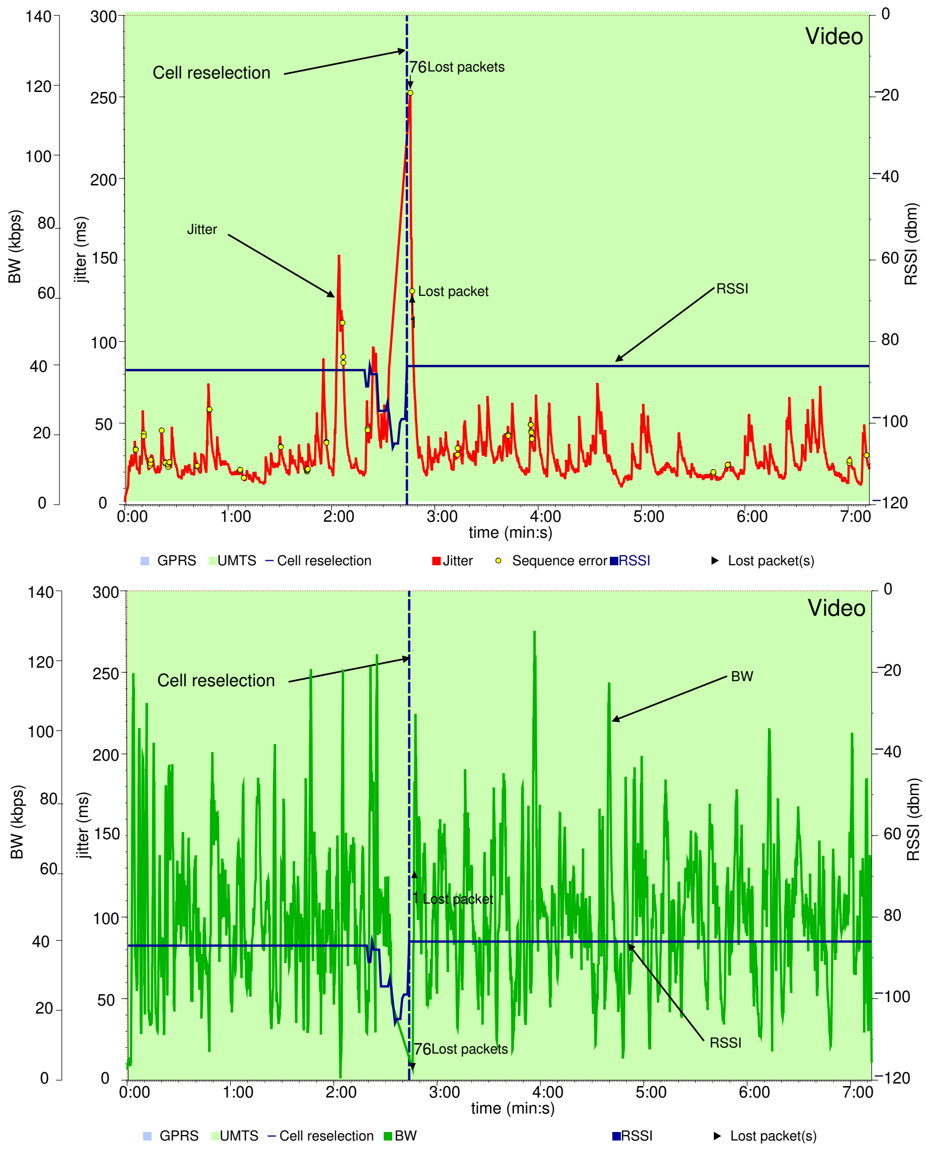

Figure 8: Lost packets due to cell change in a vehicular scenario.

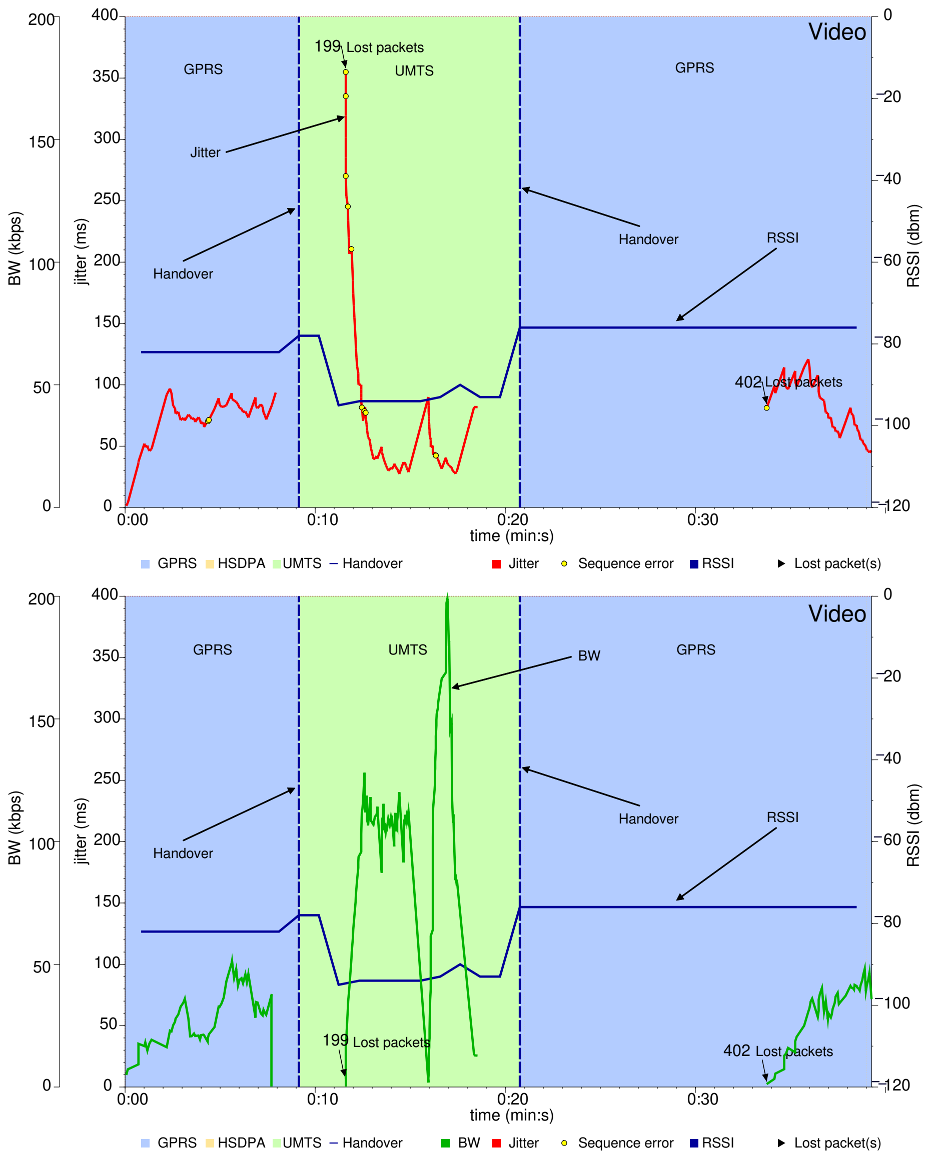

Figure 9: Lost packets due to inter-system handover between GPRS and UMTS in a vehicular scenario

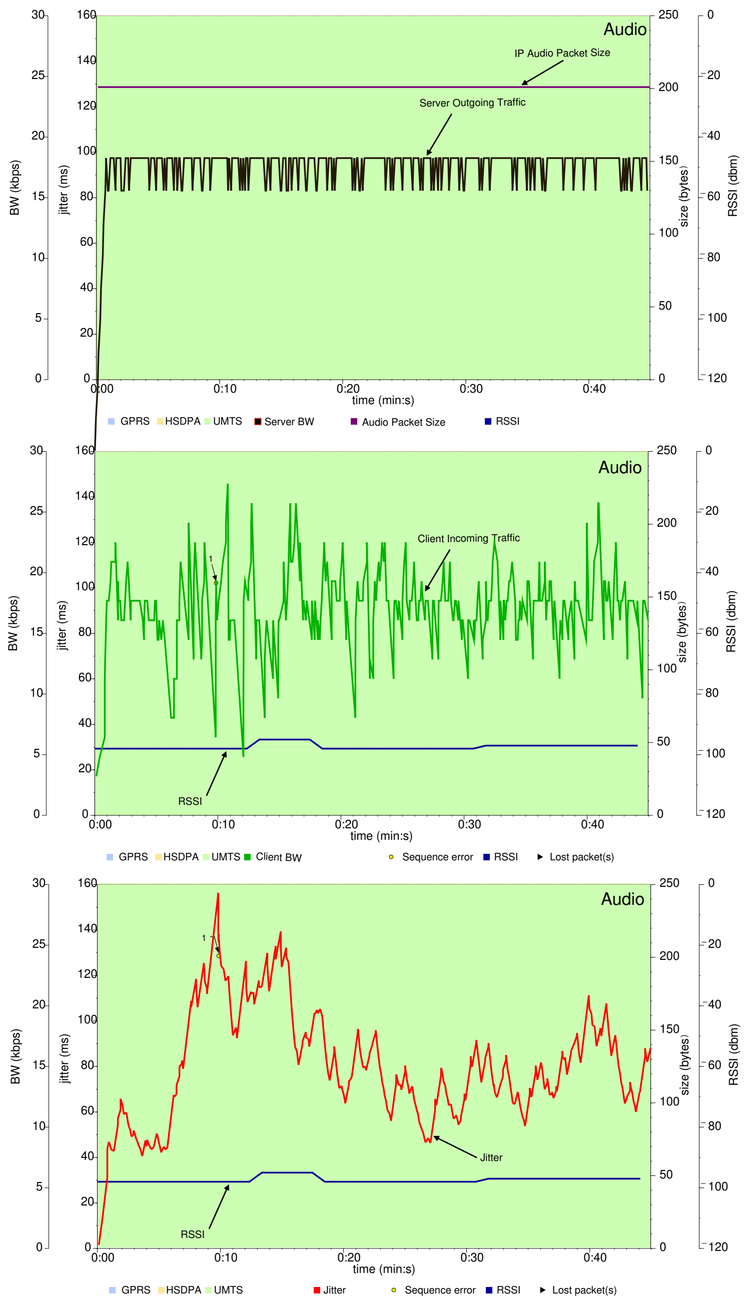

Figure 10: Jitter impact in audio streaming traffic in a static scenario

mean jitter	Operator 1 vehicular no cell changes	Operator 2 vehicular no cell changes	Operator 1 vehicular cell changes	Operator 2 vehicular cell changes
GPRS	n/a	n/a	213,59	159,02
UMTS	50,21	61,76	70,42	70,42
HSDPA	52,01	55,48	69,81	71,92

Table 7: Mean Jitter in a vehicular scenario

In the vehicular scenario, the average jitter obtained was higher than in the static scenario. As shown in figures 8 and 9, dropped packets due to handover increase jitter. Furthermore, in the vehicular scenario jitter presents a high variability as it depends on several parameters, such us the number of cell changes and changes of signal quality.

Comparing jitter results with [32], in our tests we obtain higher values but still in the same order. Rx and tx levels are in the same range, but cell load is unknown. The increase in measured jitter can be associated to differences in the cell load because our tests were carried out in live networks conditions.

4.3. Out-of-Sequence

During trials we also detected out-of-sequence packets, which indicates that it is necessary to implement reordering techniques on the streaming client. As we can see in figure 11, out-of-sequence packets are associated with the size of the packets. The video component is affected by this issue, while the audio component is not (audio packet size is fixed to 146 bytes, while the medium packet size of the video component is 638 bytes with a maximum size of 1478 and a minimum size of 42). During trials, approximately, 6% of video packets were out-of-sequence.

4.4. Bit rate

Finally, a video streaming service requires a constant/stable bit rate higher or equal to 64 kbit/s for good quality with current codecs . With codecs based on current available 3GPP specifications, the streaming quality is further improved for mobile station based streaming up to 128 to 384 kb/s [43]. But, as we can see in previous figures, it is not always possible due to the high variability of throughput available in mobile networks. In particular, bandwidth presents lower performance in the vicinity of cell changes (figure 8), which produces rebuffering events.

4.5. Mobility Patters

The results analyzed in this section were collected during tests carried out on a highway driving at an average speed of 100km/h.

Figure 12 shows a gap of 10 seconds, between 12 and 22, where no data is received and 71 packets are lost. This interruption of data reception is preceded

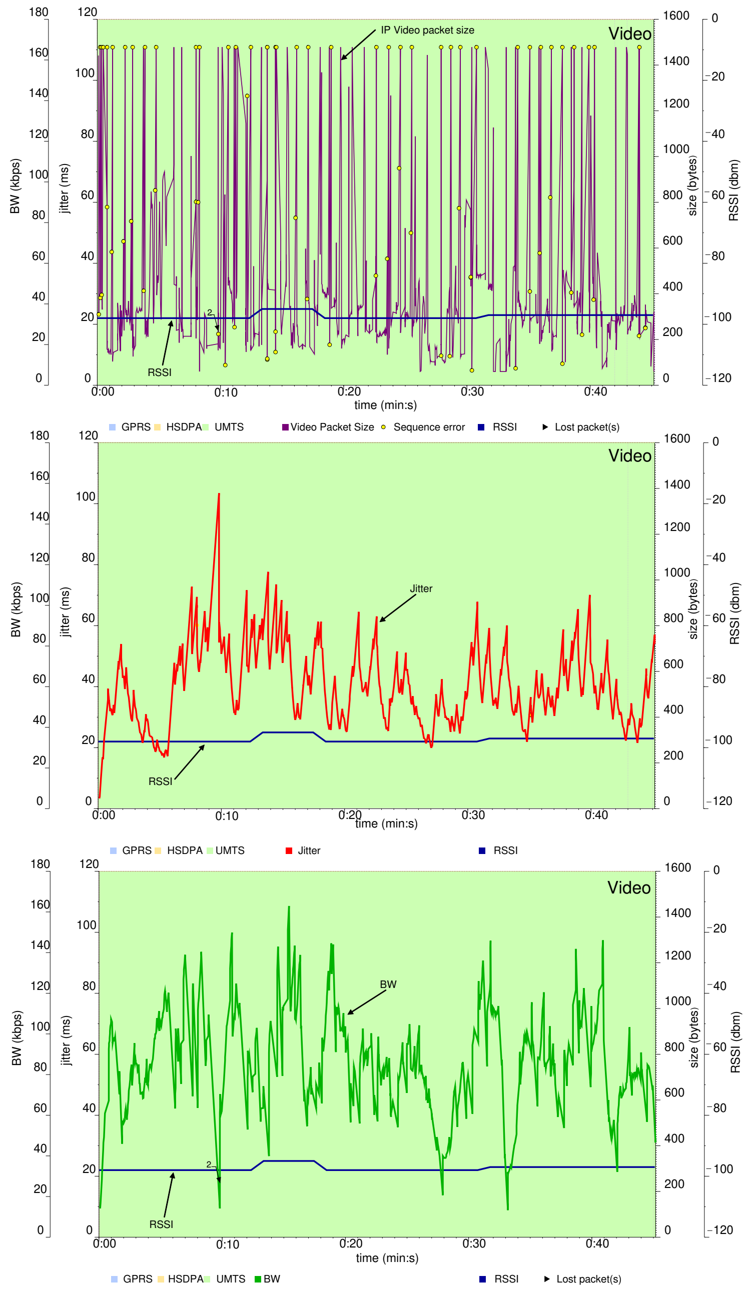

Figure 11: Out-of-sequence packets in RTP traffic over cellular networks in a static scenario

by a variation in the power received, which decreases between seconds 2 and 12 from -94dBm to a minimum value lower than -110dBm. At the same time, the transmission power reaches a maximum power of 23 dBm, which seems to be the maximum power, as the device power class is 3 and according to 3GPP test specifications [24] the maximum power measured can be between +21dBm and +25dBm. A mobile phone transmitting at its maximum power indicates that the uplink radio conditions are poor, as the network continuously commands the phone to increase its power in order to improve the quality of the signal received. Small variations of the maximum power under low signal-to-noise conditions are likely to happen even if the link is not interrupted, as up to a 30% probability of wrong detection of power control commands is allowed before disabling the uplink transmission according to [24]. The consumed power also reaches 2.5 Watts at that time, which is considerably higher than the 1.5 watts shown at the end of the graph with better radio conditions.

Another gap of 21 seconds can be seen between seconds 1:07 and 1:28. This kind of gap is quite large and should be considered when dimensioning buffers. In the middle of this longer gap there is a temporal change of radio technology, which is associated to the background color of the graph, from HSDPA to UMTS at time 1:17, and later back to HSDPA at 1:28. In addition, before the end of the gap, a vertical broken line indicates that the device selects a more suitable cell at 1:26. The behavior of both RSSI before this second reception gap is similar to the one observed before the first gap. However, during the last gap we can see how the transmission is disabled. This behavior is probably caused by detection of radio link failure condition, that will interrupt ongoing communications and trigger a cell update procedure.

In relation with the duration of the gap detected, 3GPP PSS specifications define a default value of 1 second pre-decoder buffering time. In practice, the pre-decoder buffering delay at the receiver level can be in the order of multiple seconds (e.g. 5-10 seconds) [41]. So the gap detected during the test analyzed in this section caused video rebuffering.

5. Conclusions

The results of the application of our measurement method to video streaming services can be used to characterize handover duration in different scenarios, correlation between RSSI and packet losses and correlation between sequence errors and packet size. Moreover, with the data obtained we are able to dimension the buffer size in order to overcome buffer underflow or to implement adaptive techniques.

In order to provide continuity of the service in vehicular scenarios, it is necessary to implement adaptive techniques to cope with issues associated to mobility. Adaptive techniques are included in 3GPP technical specifications for Packet-Switched Streaming Services (PSS) Release 6[28]. These specifications describe procedures for adaptive purposes, but implementation data is missing. The implementation of these techniques involves applications clients should take access to link characteristics and local application configuration, which is not

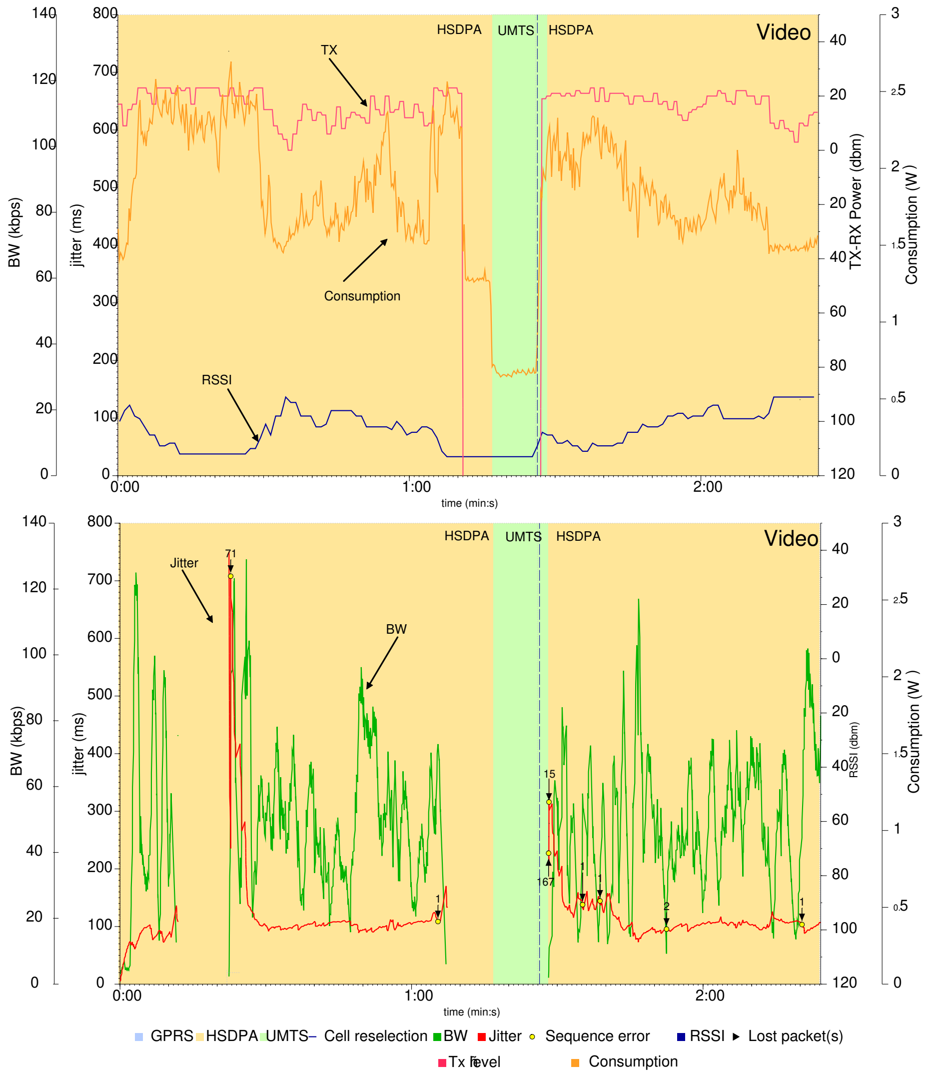

Figure 12: Impact of radio conditions on a streaming session in a vehicular scenario

a trivial issue and it is not considered in current 3GPP standards. SymPA is a tool which provides access to this information and opens the way to the implementation of future adaptive solutions for multimedia services with the information it collects.

Based on the knowledge obtained during experimental tests, we can conclude that cell-reselection and handover events are the main source of the degradation of streaming services over cellular networks, since they produce bursts of packet losses and introduce variations in the bit rate of the packets received.

We have identified that the actual support for streaming services over cellular networks does not cope with the expected QoS requirements because of the impact of cell reselection events. As a seamless handover is not always possible in live networks, adaptive techniques are needed to improve real time service user experience. In this paper we provided the basis for implementing adaptive algorithms using received signal strength, transmitted signal strength and power consumption.

The method proposed in this paper can be used for testing and validating of future protocols and services. Testing and debugging tools is essential to support the growth of the mobile Internet. Software tools for protocol analysis and performance measurement, which run on smart phones, can be very useful to help service developers and mobile operators identify the source of communication problems providing end-to-end information from the subscriber perspective. Indeed, in scenarios such as mobile-to-mobile communications, where traditional monitoring methods cannot be applied, using the actual mobile device itself for debugging is the only suitable solution. SymPA profiling tool can be downloaded from [2].

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