Experimental analysis of peer-to-peer streaming in cellular networks

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Abstract

In this paper, we study the adequacy of applying peer-to-peer techniques to mobile networks by conducting a series of experiments using smart phones as peers. We measure important parameters, such as jitter and packet losses, in static and dynamic scenarios, focusing on a video streaming service. Finally, based on the results obtained, we discuss the feasibility of these applications.

1. Introduction

Video applications deployment in cellular networks has evolved from video telephony over circuit-switched networks to video content delivery over IP packet switched networks. At present video streaming and mobile TV are the focus of new multimedia services. But in the race to discover the third generation killer application and to introduce new and innovative multimedia services, peer-to-peer (P2P) mobile applications have generated a lot of attention in cellular networks [19] [6].

The current interest in P2P techniques is to apply a decentralized architecture in order to remove bottlenecks in cellular networks, due to the lack of bandwidth, and to obtain better load balancing. P2P systems have some remarkable benefits, such as scalability and resource exchange. The decentralization of resources paves the way for the incorporation of third parties to the content provider market in cellular networks and more current, dynamic and attractive content is obtained thanks to peers controlling available data and resources.

From another point of view, mobile P2P have some limitations compared to fixed P2P networks. The main drawback is the scarcity of bandwidth and the constraints present in mobile devices, such as memory, processing capacity and battery life. In addition, the presence of the subscribers varies greatly due to mobility issues and the air interface interferences (handover, roaming, fading...).

In this paper we focus on the main network issues present in the deployment of mobile P2P applications, such as P2P traffic performance in mobile environments and access network parameters, such as quality of service class and RSSI (Radio Signal Strength Indicator).

Our objective is to study the feasibility of the deployment of multimedia P2P systems in a mobile environment. For this purpose, we propose a methodology to analyze end-to-end video streaming performance in this scenario. We apply it to both a static setting and a vehicular setting with different configurations, involving mobile and fixed networks. The results obtained are used to characterize mobile-to-mobile streaming traffic performance in cellular networks.

1.1. Related Work

Video streaming performance over cellular networks has been addressed in previous works. Lundan and Curcio in [16] [17] report experimental results only in regard to fixed-to-mobile static scenarios. Lim et al. in [15] focus on compression and computational optimization, without taking into account mobility issues and radio network interface variability.

Other works mainly addressed P2P performance in terms of system architecture and simulations results [9]. This work concludes that in order to translate the spatial benefits of P2P communication into better throughput performance in cellular networks new P2P approaches need to be proposed. But mobile P2P streaming over cellular networks is a recent and innovative approach and neither performance research nor real experimental results are as yet available.

In the past few years a wide variety of mobile P2P media streaming systems have been proposed in the literature. Most systems are based on mobile ad hoc networks (MANETS) [24] [5] [13]. But in this paper we consider cel-

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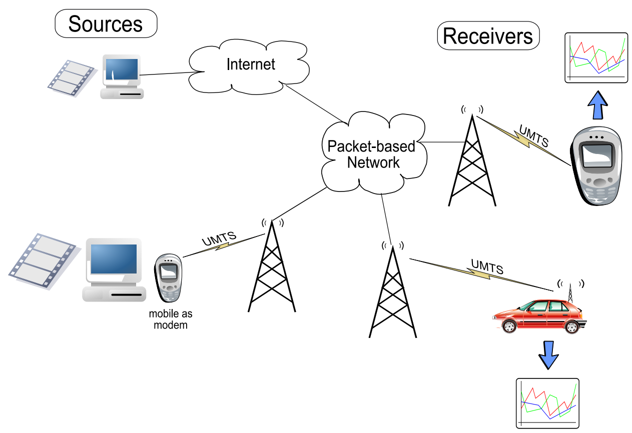

Figure 1. Schematic diagram of the measurements

Figure 2. SymPA tool

lular networks, in particular mobile streaming performance between cellular phones.

In the mobile environment there are numerous approaches based on SIP protocol [19] [6] [3] for the deployment of mobile P2P services in cellular networks. Nowadays this is the most deployed infrastructure [8] in mobile environments. SIP seems to be the most suitable solution for mobile use, but it still has some drawbacks [7] (server-centric architecture, difficulties in NAT/firewall traversal, cost ,etc) that need to be resolved.

Other lines of research are centered on the development of new P2P protocols for mobile streaming. A good example is Cosmos [14], a protocol designed specifically for collaborative streaming among mobiles devices.

The first generic approach trying to make P2P networking in mobile environment feasible is JXME [22]. JXME is JXTA's [18] mobile edition. At the time of writing, JXTA core was being ported to Symbian C++.

As far as the authors know, P2P streaming applications for mobile devices have not yet been deployed in cellular networks. Therefore, our work centers in reproducing the execution scenario of mobile P2P streaming applications (see figure 1) and studying viability of mobile-to-mobile streaming in cellular networks with different mobility issues.

1.2. Contributions

In order to analyze the end-to-end performance of P2P applications and the quality of service perceived by users, we collect network measurements from the peer side.

Traditional solutions based on intermediate elements using monitoring tools such as protocols analyzers are not applicable in this scenario.

Other solutions based on the idea of using the mobile phone as a modem [20] have been used prior to this work. We also use this configuration on the source side; additional software running on the mobile enables us to monitor radio network parameters. On the receiver side we have used our protocol analyzer for mobile phones (SymPA). SymPA [4] captures the mobile phone's incoming traffic (see figure 2) at IP level and provides us with information about QoS parameters provided by operators, signal strength received, location information based on cell identifier, location area code and country code, and battery consumption statistics.

Although we use traditional tools to study traffic performance on the source side (UMTS modem), on the receiver side innovative measurements are collected taking into account the technical constraints of mobile devices, such as memory size, cpu performance, battery capacity and the actual protocol stack presented in such devices. In the future, we are planning to develop a real video streaming server for mobile terminals and to test it over P2P cellular networks.

All these parameters are correlated in order to analyze the impact of mobility issues in P2P streaming traffic performance.

The contributions of this paper are twofold:

- We provide a methodology to evaluate mobile-to-mobile streaming performance independently of the framework and application used.
- We present statistical results that can be useful for the design and the future deployment of P2P streaming applications over cellular networks. The main drawbacks of P2P model in cellular networks are cited in [9]. In particular, we have concluded that mobile-to-mobile video streaming is not feasible without the deployment of high speed radio access technologies such

Parameters	Operator 1	Operator 2	Operator 3
Traffic class	Interactive	Interactive	Interactive
Maximum Bit Rate (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 1. UMTS Bearer Service Attributes

as HSDPA (High Speed Downlink Packet Access) or HSUPA (High-Speed Uplink Packet Access).

This paper is structured as follows. After this introduction, we present the methodology used during our study and outline the setup of our test environment in Section 2. In Section 3, the different scenarios are detailed and the result of our experiments are discussed. Finally, in Section 4, we present our conclusions.

2. Methodology and test scenarios

Trials have been carried out over three Spanish public UMTS Networks. The UMTS QoS attributes [1] measured for each one are shown in table 1. Interactive traffic has no guaranteed bit rate; it depends on the current load of the system and the traffic handling priority. One fundamental characteristic of this traffic class is that the payload content is preserved, but the transfer delay, the bit rate and packet loss are not kept. UMTS defines a streaming class which is more suitable for streaming services.

In this work we study the point to point performance of RTP streams between mobile phones. Security and router performance issues are not addressed in this paper.

In our test we measure bandwidth, jitter and packet losses. Completed measurements have taken into account recommendations published by 3GPP for streaming service in GPRS and UMTS networks [2]. Bandwidth shown in the figures is calculated as the number of bytes received on the client side in the last second. Jitter is calculated in accordance with the formula described in [12]. Packet losses are detected by examining the sequence number included in RTP packets.

We have chosen 3GP as file format and MPEG 4-Visual as codec. Properties of videos used during field tests are shown in table 2. The video streaming service is delivered through the RTP/UDP/IP protocol stack. RTCP [12](RTP Control Protocol)) is used for the exchange of control information between server and client. RTSP [10](Real-Time Streaming Protocol) and SDP [11](Session Description Protocol) protocols are used for session setup and control.

During trials we have used an on demand streaming service. Darwin streaming server has been used to provide stream flows.

RTP flows captured in the mobile terminal are analyzed with the well-known network protocol analyzer WireShark (formerly Ethereal).

3. Experimental Results

Prior to the deployment of video streaming services, it is necessary to assess the delay, jitter and packet losses on the cellular networks in order to determine if mobile-to-mobile video streaming applications are feasible. Bandwidth, jitter, delay and packet loss measurements are required for the correct design and configuration of P2P networks in a mobile environment, as well as for buffering parameters in peer elements.

We perform a realistic study in real conditions of network load, taking into account the main limitations of mobile devices and taking advantages of the main characteristics of smart phones. Current smart phones have rapidly incorporated the latest technological advances like Bluetooth, Wi-Fi, HSDPA etc; and they are the first real pervasive and mobile devices [21] that can be used to study end-to-end service performance perceived by subscribers.

Measurements were carried out within Symbian OS based smart phones and the Nokia Series 60 platform running Real One Player as streaming client.

In this paper we considered a static fixed-to-mobile scenario in which the mobile phone tries to access a stream located in a PC connected to the Internet via a high speed connection as a point of reference. In order to characterize the latency of this scenario we have measured round trip

	3.3gp	12.3gp
Duration (s)	45.0	432.5
Video	MP4V	MP4V
Video Bitrate (Kbps)	76	42
Resolution	176x144	176x144
Frame Rate (fps)	25.00	8.3
Audio	SAMR	SAMR
Audio Bitrate (Kbps)	12	8

Table 2. 3gpp video samples

RTT(ms)	Op1	Op2	Op3
1	4286	2342	1017
2	825	372	630
3	330	370	650
4	262	369	555
5	260	367	581

Table 3. Round Trip Time in a static fixed-to-mobile scenario

time (RTT) with 32 bytes ICMP packets. Results obtained are shown in table 3.

In the second scenario we reproduced a mobile-to-mobile streaming session replacing the high speed connection with a UMTS data connection. RTTs measured in this scenario are shown in table 4. RTTs calculated with 32 bytes packets is about 500 ms with a high variance. These values are higher than the ones obtained in the fixed Internet, and higher than the values obtained in the first scenario. The mean RTT obtained in traditional wired networks is about 50 ms. Video and audio applications are particularly sensitive to RTT variance, which is why the deployment of video streaming applications in cellular networks requires new performance studies. Direct consequences of RTT variability are rebuffering events and out-of-order packets.

In figure 3 out-of-order packets are marked with a yellow dot. In the reference scenario the most marked packets have the maximum size (see figure 4), while in a mobile-to-mobile scenario out-of-order packets do not seem to be related with the packet size due to the high variability of RTT.

The goal of the following measurements is to characterize packet-level performance parameters in different scenarios and user mobility.

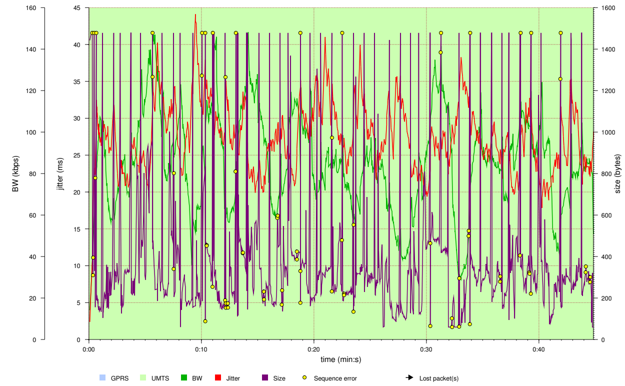

Figure 3. Sequence errors during a streaming session in a static mobile-to-mobile scenario

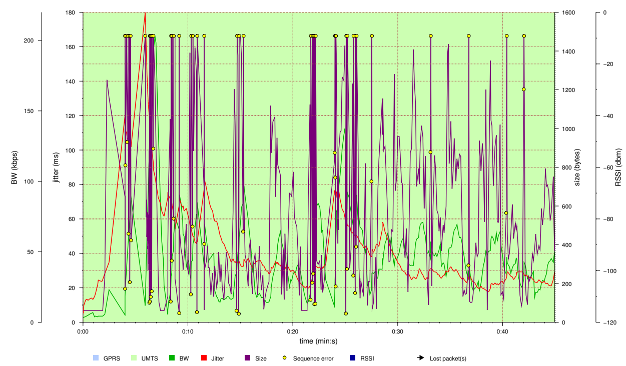

Figure 4. Sequence errors during a streaming session in a static fixed-to-mobile scenario

3.1. Peer to Peer traffic characterization in static scenarios

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

Figure 5 shows the evolution of jitter and bandwidth during a streaming session in the reference scenario described in the previous section. In this case, there are no packet losses. Generally packet losses in this scenario are isolated and do not exceed 1%. For these trials we have used video 3.3gp with a video bit rate of 76 kbps, and an audio bit rate of 12 kbps.

Figure 6 shows the evolution of jitter and bandwidth during a streaming session in a mobile-to-mobile scenario. As we expected, results obtained are worse than in the reference scenario. Regarding jitter of real time traffic, there are significant differences between both of them. In par-

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RTT (ms)	Op1-Op2		Op1-Op3		Op2-Op3	
	1-2	2-1	1-3	3-1	2-3	3-2
1	2968	2828	1150	6296	4093	3328
2	484	562	843	640	500	421
3	468	546	453	578	468	453
4	562	562	1187	625	531	500
5	562	593	1187	593	546	453

Table 4. Round Trip Time in a static mobile-to-mobile scenario

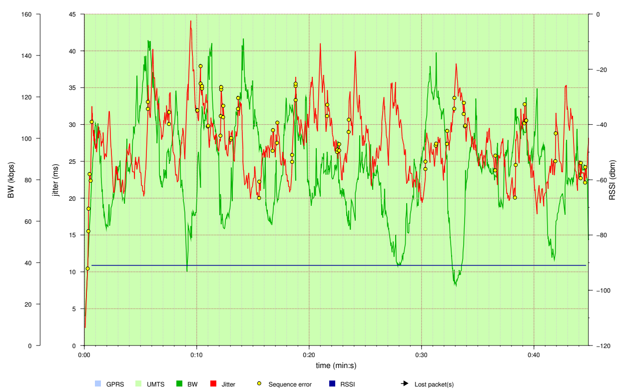

Figure 5. Packet losses in a fixed-to-mobile static scenario

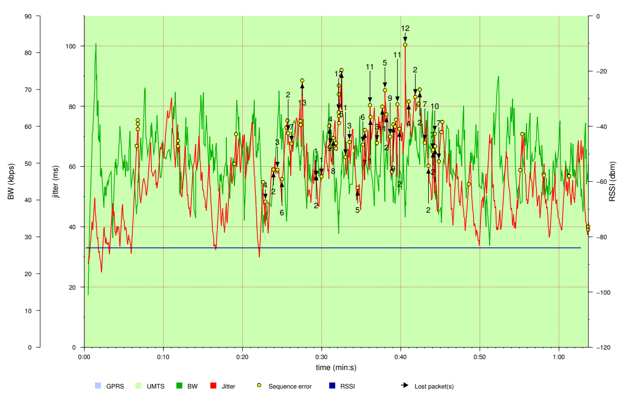

Figure 6. Packet losses in a mobile-to-mobile static scenario

ticular, the mean jitter in the fixed-to-mobile scenario is 47.86 ms, while it is 90.58 ms in this one. During the mobile-to-mobile streaming session we obtained a 19.24%

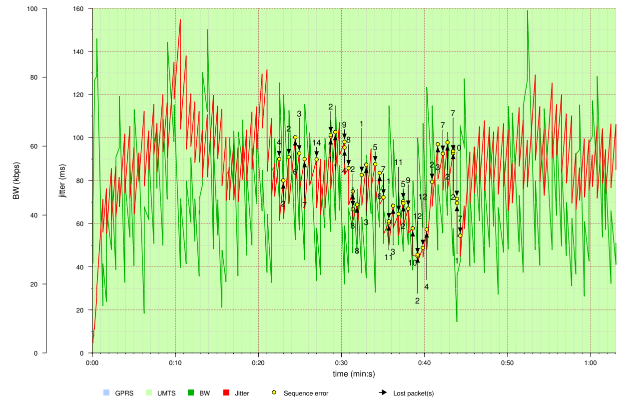

Figure 7. Packet losses in a mobile-to-mobile static scenario on source side

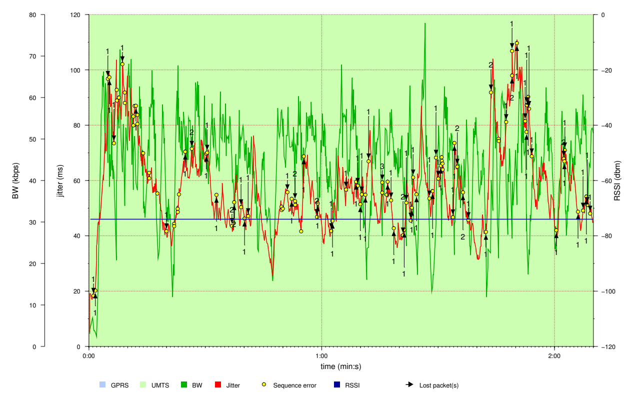

Figure 8. Packet losses in a mobile-to-mobile static scenario, low bit rate video

of packet losses. If we depict the outgoing traffic on the source side (see figure 7) we can observe that 18.99% of packet losses take place at this point. This behavior is due to the peer transmitting at a higher bit rate than the available up link rate, thus causing network congestion which results in packet losses.

The situation gets worse if there are two peers accessing the same source. In the next test, two peers try to stream the 3.3gp video from the same peer. In this situation we obtain a 61.03% of packet losses on the receiver side. 50.67% of packet losses are located at the source.

Table 5 shows results obtained when three peers are accessing the 12.3gpp stored at the same peer. Video 12.3gpp has a lower bit rate than video 3.3gp. As we can see, the most restrictive parameter is packet losses when there are

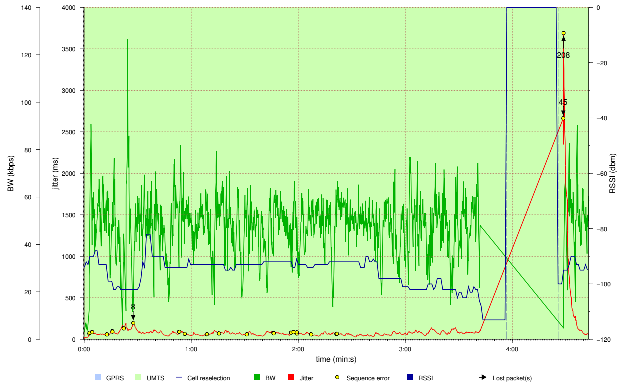

Figure 9. Packet losses in a mobile-to-mobile vehicular scenario, low bit rate video

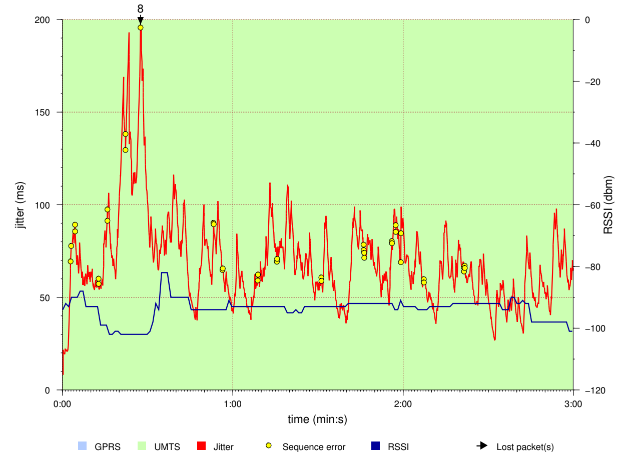

Figure 10. Jitter in a mobile-to-mobile vehicular scenario, low bit rate video

multiple peers accessing to the same source, while jitter remains within an acceptable range.

Although in UMTS uplink we can achieve a theoretical bit rate of 384 Kbps, during our experiments we have obtained rates lower than 100 Kbps. Available uplink bit rates are too low to support the execution of mobile-to-mobile multimedia services. The deployment of high speed radio access technologies such as HSUPA is required to fulfill the needs of multimedia P2P applications.

A possible solution to avoid network congestion when there are several peers accessing the same source is to adapt the sending rate to the number of peers and to implement a queueing mechanism to manage the available bandwidth.

Figure 8 shows values obtained for a video with a bit rate of 42 Kbps (12.3gp). As shown in figure 8, when only one peer is accessing to the source, packet losses decrease and are kept in the range of 8%. In this situation there is no congestion on the source side.

3.2. Peer to Peer traffic characterization in vehicular scenarios

In this scenario unpredictable events take place, such as handovers, cell-reselection, or link outage. These events reduce the available bit rate and can interrupt the connection. In this scenario packet losses are investigated in vehicular environments with vehicles moving at 100 Km/h.

Figure 9 shows results obtained during a streaming session in this scenario. During the session a brief disconnection took place due to a link outage under a tunnel. Mean jitter obtained (see figure 10) is of the same order of magnitude than in the static scenario. The mean jitter is around 150 ms, which complies with target values given in ITU recommendations. However, during disconnections jitter experiences an increase due to packet losses.

Video streaming is much more sensitive to packet loss than jitter. In figure 9 we can see that the streaming client does not receive data for 30 seconds, and after that it starts to receive data. But this pause results in a buffer underflow, and the playback stops even though new packets arrive.

Another usual phenomenon in vehicular scenario is continuous rebufferings. When rebuffering takes place, the picture freezes and it results in a degradation of the quality of service perceived by users [23]. Preceding figures show the high variability of bandwidth in cellular networks. These variations can be caused by many issues, such as network buffers overflow, and they are characteristic of kind of networks. Rebuffering occurs when throughput decreases below the video bit rate. In this way the use of low bit rate video in mobile environments is encouraged.

Before a handover or an intermittent disconnection takes place a decrease of RSSI is observed. This behavior can be used to implement an adaptive technique in order to predict packet losses and to overcome buffer underflow using, for example, RTSP commands to pause the streaming session before these losses take place.

In the next field test, we analyze handover impact over mobile-to-mobile video streaming. Figure 11 shows the duration of a handover event in a mobile-to-mobile scenario. It lasts about 30 seconds and during this time a burst of packet losses takes place (see figure 12). As a consequence, buffer underflow occurs and playback stops. Buffer underflow happens when peer buffer has a size in time smaller than the handover period.

Duration of handovers in this scenario is higher than in

	lost packets	max delta (ms)	max jitter(ms)	mean jiiter	sequence errors
Peer 1(Op 1-Op 3)	168 (24,54%)	524,70	176,10	41,10	142
Peer 2 (Op2-Op3)	173(24,5%)	547.10	181.54	45.35	144
Peer 3 (Op 3-Op 3)	147(19,5%)	359,40	122,59	30.80	133

Table 5. Multiple peers

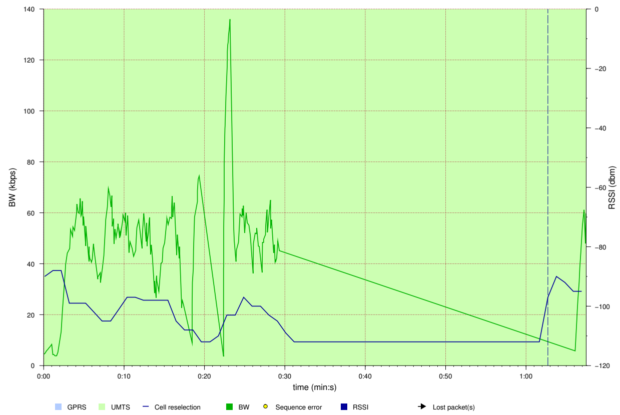

Figure 11. Handover in a mobile-to-mobile vehicular scenario

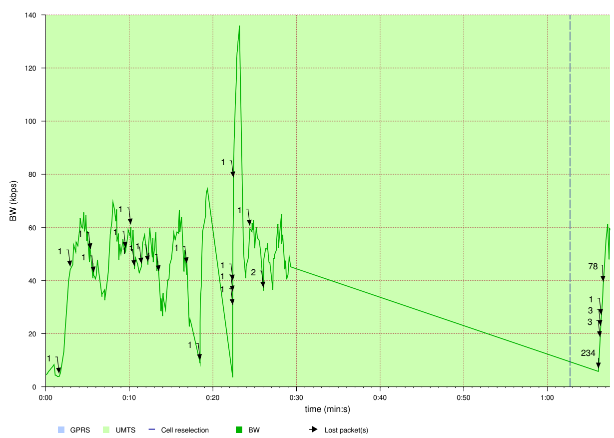

Figure 12. Packet Losses due to a handover in a mobile-to-mobile vehicular scenario

a fixed-to-mobile scenario. In the reference scenario handover lasts around 10 seconds and playback resumes after a pause of several seconds (see figure 13).

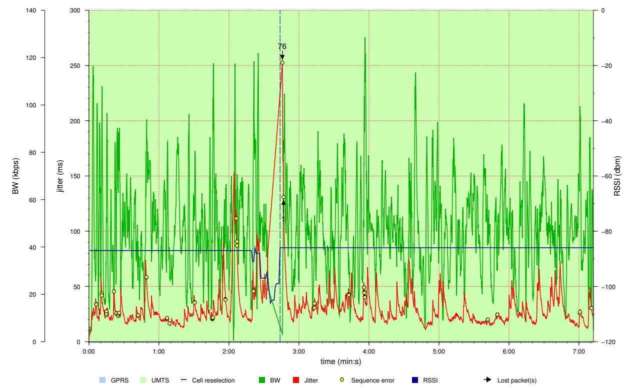

Figure 13. Lost packets due to handover period in a fixed-to-mobile scenario

4. Conclusions

In this article we have evaluated the performance of video streaming services in mobile networks taking into account constraints presented in mobile terminals. We have proposed a methodology that can be used to evaluate the performance of future mobile P2P streaming applications and services.

We have measured video streaming service performance parameters such as packet losses, jitter, bandwidth and round trip time. Moreover, we have measured the performance of video streaming traffic in accordance with different configurations and different mobility issues.

Our analysis shows that in a mobile-to-mobile scenario packet delay and delay variation increases, bandwidth decreases and video bit rate used is limited by the network congestion.

While in a static scenario burst of packet losses are caused by congestion, in a vehicular scenario bursts of packet losses are also caused by radio link disconnection due to handovers and link outage.

Our analysis has shown that performance strongly depends on mobility of users. In particular, performance is degraded as the mobility increases.

We can conclude than mobile-to-mobile video streaming in a vehicular scenario is not feasible without adaptive techniques in order to compensate jitter variability and intermittent radio link disconnections. Moreover, for a successful deployment of P2P video streaming applications, new high speed radio access technologies are needed.

Acknowledgment

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