Analytical Evaluation of Collaborative Communications in UMTS Networks

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Summary

When a new technology is developed, new services appear related to it. Mobile networks brings an enormous benefit to the society and in a no so long future, it could be a fundamental element of communications. Nowadays, one of the most demanded services in the world is the real-time multimedia communications. The videoconference and the streaming are two of the essential media to the users. In this work we present a novel multiplatform tool for teleteaching. To create the virtual classroom, we join the participants in multicast groups and using a session architecture, we interconnect different multicast islands into a group. We have developed a test-bed to measure the information sent to the network and we have used different network technologies, as Ethernet or mobile networks, to prove it. The measurements were carried out using real networks. This work shows the results obtained in various test realized in UMTS mobile networks and compares them with the results obtained in wired networks.

Key words:

Collaborative Communications, RTP, TCP, UDP, UMTS, JMF.

1. Introduction

In recent years, the way that people use to get in touch is changing quickly. The importance of Internet is increasing at everyday life. In many cases, old Internet communication tools, like chats or newsgroups, are disappearing in favor of instant messaging applications, VoIP or videoconference.

Furthermore, users are required to pay more attention to the method used to communicate with other users more than ever. Technologies as mobile telephony (UMTS, Universal Mobile Telecommunications System [1]; HDSPA, High Speed Downlink Packet Access [2]), and broadband access networks (xDSL - Digital Subscriber Loop; WiFi), have evolved and new communication tools are appearing. Therefore, the union of the multimedia communication and the mobile networks gives a new dimension to the ubiquity concept. The users can access to new services offered by the providers in every part of the world. This implies that providers are offering better taxes to new services like connectivity to their data networks.

Moreover, the ubiquity, as the ability to be in various places at the same time, has been one of the most wished talents for the human being.

Nowadays, with the appearance of the knowledge society, the idea has been revitalized. Thanks to the possibilities that the technologies contribute to the knowledge society and the information and communication.

The limitations of the physics, the space and the time that are imposed to the ubiquity can be saved with the software, the communication protocols and the communication networks.

Even more, the services are changing to adapt themselves to new requirements of the users. New communication paradigms and applications have been developed and they benefit to the users. An example of these communications is teleteaching or teleworking.

In that sense, we propose a new application, called VLinEx, to provide collaborative communications for teleteaching or teleworking in heterogeneous networks. This is a novel multiplatform application, and it has been developed as an educational tool to communicate teachers with students. It allows creating virtual classrooms using multicast. The protocol used to transmit the multimedia content is RTP/RTCP. This protocol is usually used with UDP and IP to forward multimedia information. It also can create multicast groups using Class D IP addresses to denote it.

Furthermore, we have developed a novel architecture to control the multimedia session and create the tunnel between the server session and the client islands. With this method, we can interconnect several classrooms in different islands using the same multicast group. All the signaling traffic and the multimedia information are forwarded to the multicast islands as an only multicast group.

The developed application can be used with wired, wireless and mobile networks depending on the necessity of the user. To meet the requirements of the mobile networks users, we have developed a test-bed to analyze the advantages and disadvantages of this collaborative communication in a wired network and compare the results obtained with the communications in a mobile network.

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We have chosen UMTS technologies due to the most part of infrastructure is shared with HDSPA. And in many places, HDSPA is in deployment phase.
The remainder of this paper is organized as follows. Section 2 gives the architecture of the developed tool and the case of study used to validate it. Section 3 introduces the results obtained in the case of study. In section 4, contains our concluding remarks. Finally, section 5 contains our future works.

2. Case Study

VLinEx is a novel multimedia application. It can transmit video and audio with multiple clients using multicast [3]. It has been developed in Java using an API called Java Media Framework (JMF) to handle multimedia content.
Nowadays, any router in the backbone of the Internet will discard a multicast flow. Furthermore, the first router that processes the packets in the Local Area Network (LAN) will discard the entire multicast flow. This is called Multicast Island because the multicast flow cannot be sent to the Internet.
Therefore, a tunnel between islands allows sending multicast flows between them and receives the same information. In our purpose, we have developed an end-to-end TCP-tunnel. We decided to implement these tunnels with TCP protocol because the LANs usually have firewalls and use several routing protocols like NAT (Network Address Translation). For this reason, the creation of an UDP-tunnel is difficult due to the requirement of new routing rules in the routers.
In this test-bed, we need an auxiliary application to connect with an UMTS network. In that sense, we have developed another application called Gnome-GPRS. This application allows using the mobile phone as a modem and connecting to Internet with a mobile. A connection 2.5G or 3G could be established as follows. First of all, the physical layer has to be connected and configured. If we are using a laptop and a mobile phone, this connection have to be serial, USB or Bluetooth.
The second phase is the establishment of the point-to-point connection (PPP, Point to Point Protocol) in the link layer between the devices [4]. One of the most important characteristics in the process is the access to the lower layers. To provide the access to the hardware of the terminal, and the GPRS/UMTS properties, the standard defines a set of commands called AT+. They are an extended set of AT commands to control the internal modem implemented in the mobile phones [5]. Figure 1 shows the protocol stack.
Gnome-GPRS simplifies the process, offering to the users of GNU/Linux a mechanism to connect to the Internet using portable devices and mobile technologies with GNU/Linux. This application has been developed using GTK+ libraries and can be integrated with GNOME.

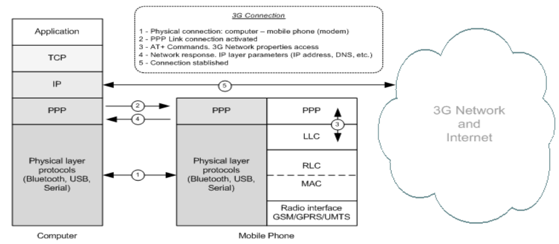

Fig. 1: Protocol stack used in the 3G connections.

Furthermore, it can negotiate different parameters of Quality of Service (QoS) and offers information about the real status of the connection.
VLinEx and Gnome-GPRS would be used to evaluate the performance of the UMTS network when the multimedia content is being transmitted.
In the test-bed we have used a PC with GNU/Linux as a server of the session (multicast session and tunnel). It is connected to a 100 Mbps Ethernet Router. This will be the original multicast island.
A laptop is used as client. We connect the laptop with a mobile phone and use the mobile network to access to the session using Gnome-GPRS application. This laptop creates a tunnel with the server of the session and forwards all the packets to the new multicast island created on the laptop side. Other computers connected to this multicast island can receive the packets forwarded. This is shown in figure 2.
With the creation of the automatic TCP-tunnel all the problems mentioned above are avoided. This is possible due to the session system developed returns the information needed to configure the tunnel. The main advantage of this method is the absence of new networks interfaces in the operative system. Figure 3 explains the phases of the tunnel negotiation.

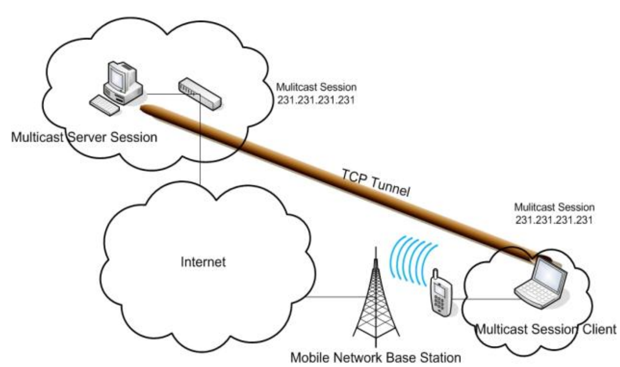

Fig. 2: Diagram of the test-bed with a multicast island based in a Mobile Network.

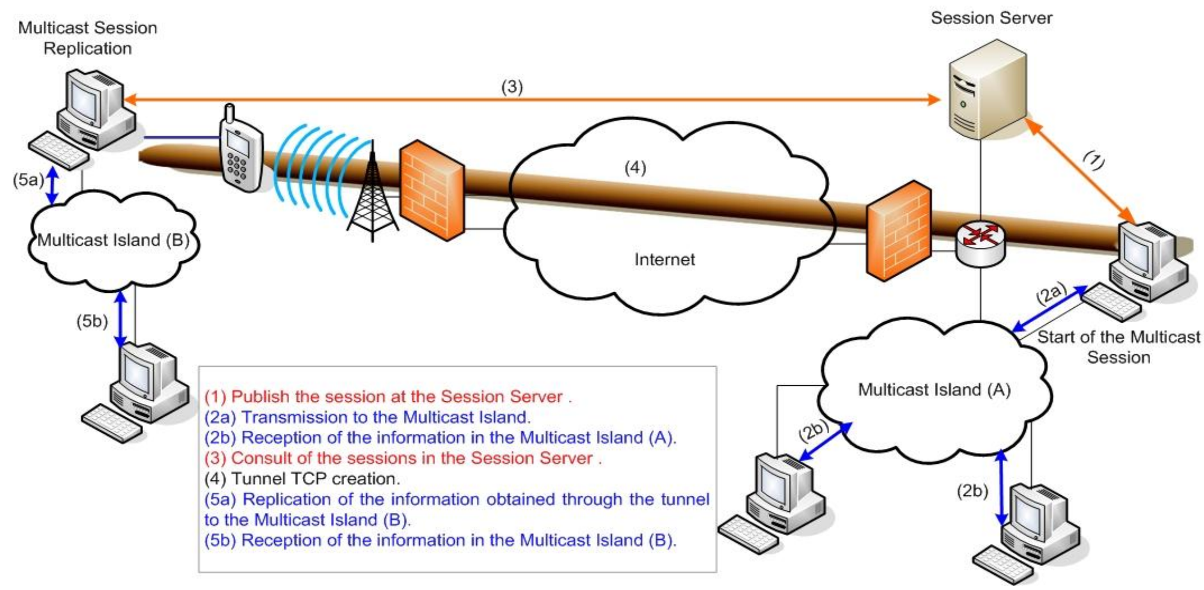

Fig. 3: Establishment of the tunnel.

As we can see in the last figure, the computer that starts the communication publish the session in the session server. This computer sends its IP address and the multicast session address (phase 1). In the next phase (phase 2), the computer starts to transmit the multimedia content to the Multicast Island.

When other computer out of the multicast island wants to receive the session, consults in the session server obtaining the IP address of the computer that started the communication (phase 3). Therefore, an end-to-end communication is opened using TCP protocol (phase 4).

All the packets generated in the multicast island (A) are sent to the multicast island (B) across the tunnel. When these packets were received at the island (B), the information (TCP segments) are processed and translated into UDP segments. These segments are sent to the multicast island (B) (phase 5).

At this moment, the information in the two islands is the same and both islands are at the same multicast group.

3. Evaluation and Results

In this section we present the results obtained from different tests realized using the scenario presented in the section 2. The first test is an audio transmission. The data is obtained from a MP3 file and encoded in µLaw to forward the information.

The second test is a video transmission. The data is obtained from a WebCam and the last test is a multimedia file transmission. The data is encoded in DIVX5 and MP3 and recoded in H.263 and µLaw (video and audio) when it is transmitted. We use those codecs because they are included in the RTP (Real-Time Transport Protocol) with minimal control standard [6], [7].

The requirements of this test are higher than offered by the UMTS network. With this stress test we can show how the network reacts in this situation.

The results obtained in the test-bed with a mobile network are compared with results obtained in a test-bed based in a local network. In this case, all the subnets in the local network are Ethernet. In the test-bed there are two different subnets, this implies that there will be two multicast islands and the tunnel is necessary as we can see in figure 4.

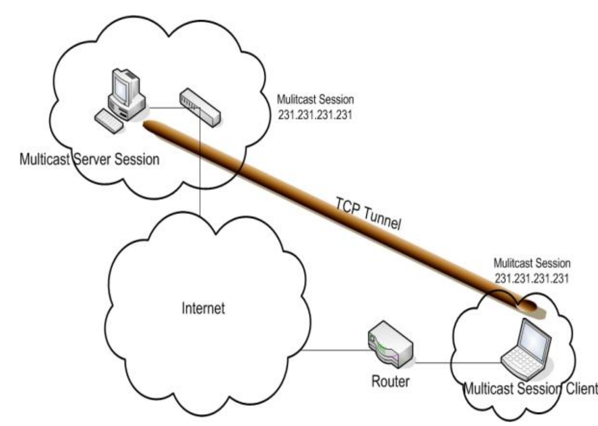

Fig. 4: Diagram of the test-bed based in Local Networks.

3.1 Audio Transmission

This is the first test; we compare a real time audio transmission in a UMTS network and in a wired network.

The multimedia content is an audio file coded in MP3 and it is transmitted in both test-bed. This is depictured in figure 2 and figure 4.
The file is opened and recoded in µLaw by the computer that starts the session (server of the session). When the tunnel is established, the server of the session sends the information through the tunnel to the multicast island (B).
Figure 5 shows the communication in the test-bed multicast island (B), based in a Local Network. In this test, the server of the session forwards a constant bitrates stream. The information is packaged using RTP protocol and sent to the network. We can observe in the figure that the throughput is constant.
Therefore, the transmission is fluent in both multicast islands and the reproduction of the sound, in the client side, is clear without interruptions. The interruptions could be produced by the delay of the packets introduced by the network. In this case, it is the base scenario. The test in the mobile network (figure 2) will be compared with those results.
Figure 6 shows the same transmission in the mobile network. We can observe that the packets are not received with a constant throughput and the network drop some packets of the stream. This situation can be observed in the graphic because some maximums and minimums are produced in it. One example of this is observed in the second 50, where the throughput downs nearly to zero. A packet lost in intermediate nodes of the network produces this.

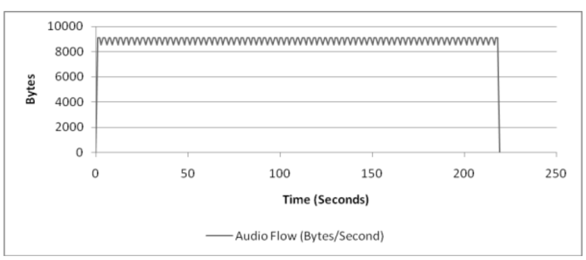

Fig. 5: Multicast stream in the multicast island (B) in the wired network.

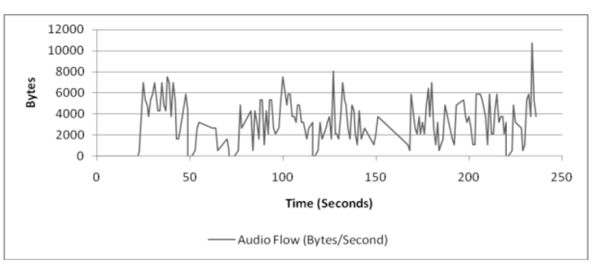

Fig. 6: Multicast stream in the multicast island (B) in the mobile network.

We have made some modifications to Wireshark [8], wich is a network protocol analyzer. We obtained the packet loss percent in the network. This value is around the 5.2% of the packet transmitted through the tunnel. These packets are not received or they are received with a significant delay. This situation produces a discontinuous reproduction due to the packet loss.
The Table 1 shows a summary of the first test.

Table 1: Summary of the audio transmission

	Wired Network	***UMTS Network***
Time of transmission (Seconds)	218	236
Packets retransmitted	0 (0%)	122 (5,1%)
Medium size of the packets (Bytes)	370	761

3.2 Video Transmission in real time

In the test we transmit a video captured from a WebCam and encoded in H.263 format. Figure 7, shows the transmission in the TCP tunnel at the wired network.
We can observe that the throughput is not constant like last test. The encoder codify the video frames depending on the previous and subsequent frames of the video captured.
Furthermore, the information codified is related to the data that contains the frame itself [9]. If the frame is dark the amount of information to compress is lower than if the frame is rich in color.

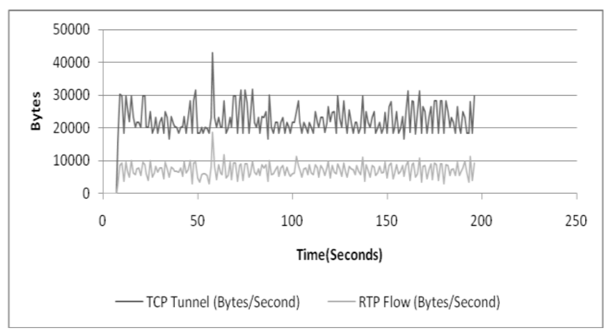

Fig. 7: Video transmission in the wired network.

There is no packet loss and the reproduction has not any failure. Furthermore, the temporal sequence is reconstructed in time. No frame is lost and the information can be reproduced without gaps.
The second function represented in figure 7, corresponds to the RTP stream in the wired network (multicast island (B) side). The information received is the same that was transmitted through the tunnel.
The needed throughput to transmit the information across the tunnel is greater because of the TCP headers. However, when the packet is translated into a UDP segment the information to transmit is lower.

In this test, the transmission showed in figure 7, is the reference model and we will compare it with the results obtained in the test-bed using the UMTS network.
Figure 8 shows the results obtained in the test with the mobile network. The results obtained are similar to figure 7. This test is a real time video transmission, and the content is not recorded previously. This implies that the codified images are different in both tests (wired and mobile network). Even so, both figures (figure 7 and 8) are similar each other.
In this test we are using the test-bed based in a UMTS network (figure 2), and the throughput of the channel is not the same as the throughput in the wired network. The UMTS network provides connection about 320 Kbps (40 KBps) and the stream generated by the application of videoconference do not exceed this throughput. This implies that we can transmit video in real time using UMTS networks.
But in the mobile network, the 0.78% of the packets were lost. This means in some moments of the communication, the reproduction in the multicast island (B) is discontinuous and the packet loss produce pixelation in the frames.
The Table 2 shows a summary of the second test.

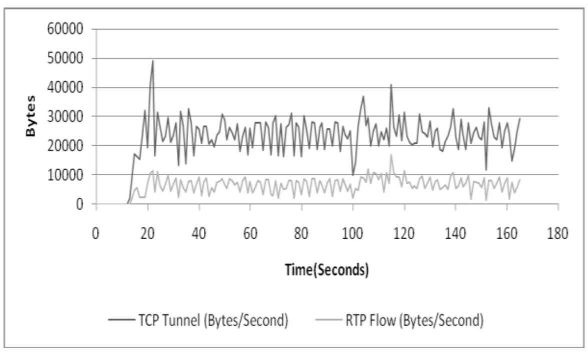

Fig. 8: Video transmission in the mobile network.

Table 2: Summary of the video transmission

	Wired Network	***UMTS Network***
Time of transmission (Seconds)	196	165
Packets retransmitted	0 (0%)	34 (0,78%)
Medium size of the packets (Bytes)	418	843

3.3 Resilience Test to the UMTS network

The last one is a resilience test to the UTMS network. With these results we can know how the network reacts when the stream transmitted is three times the throughput of the UMTS network.
In this test, we transmit a multimedia file encoded in DIVX5 and MP3 and recoded in H.263 and µLaw when the stream is transmitted to the network.
Figure 9 shows the results of the multimedia transmission across the tunnel in the wired network. In this test the video stream requires a greater throughput than the audio stream. This is produced due to the information to encode and transmit is greater.
The throughput used by the video stream is not constant as we showed in last test and many variations are produced by the information encoded. The audio stream is constant because the payload of the RTP packet specifies this property. In the table 2, we can see that there is not packet loss and consequently, the reproduction has not any failure. This will be our model and we are going to compare it with the results obtained in the test with the mobile network.
Figure 10 and 11 shows the same TCP tunnel, on he mobile network depictured in figure 2. We can observe that the video (figure 10) and the audio transmission (figure 11) are different from results showed in figure 9.
Analyzing the video tunnel, we can observe that the throughput is near to 0 Bytes many times. This effect is produced by the packet loss in the tunnel. If we study analytically the stream, we obtain that the packet loss rate is around 6.6% of the packet transmitted.

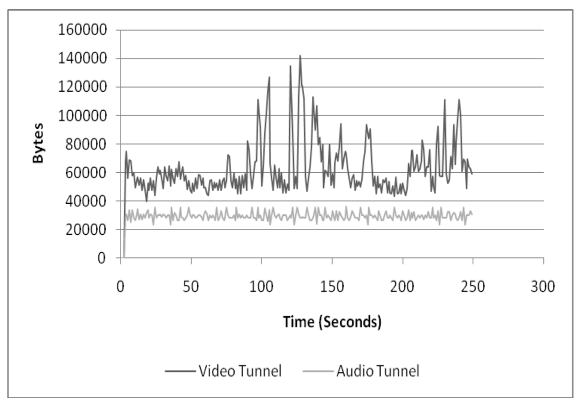

Fig. 9: Multimedia transmission in the wired network

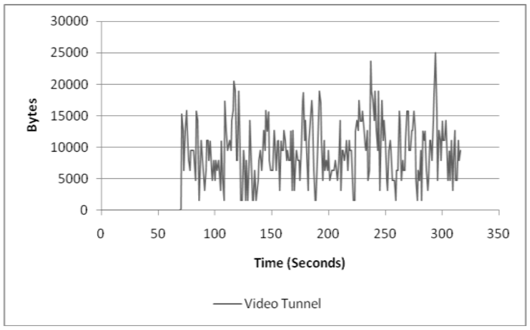

Fig. 10: Video Tunnel in the mobile network.

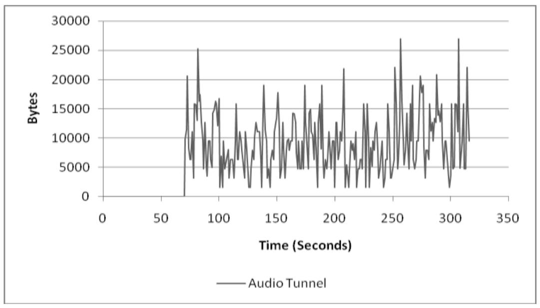

Fig. 11: Audio Tunnel in the mobile network.

We can observe in the audio tunnel that the problem is similar to the video tunnel. In this case, the function of the graphic should be constant, but there are several maximum and minimum peaks that imply packet loss. The packet loss rate is about 6.2% of the transmitted packets.
The packet loss produced in the video and audio tunnel has influence on the stream reconstructed at the multicast island (B). These results were expected because we are transmitting a multimedia content that need 3 times the throughput that the UMTS network can provide.
The average packet loss ratio is about 6.6% of the packets transmitted. The needed throughput to transmit de multimedia file is about 90 KBps and the throughput provided by the mobile network is 40MBps. Thus, the packet loss is bearable and expected in this test.
The Table 3 shows a summary of the results obtained on the resilience test in the wired and the UMTS network.

Table 3: Summary of the resilience test

	Wired network	***UMTS network***
Time of transmission (Seconds)	249	316
Packets retransmitted	0 (0%)	361 (6,3%)
Average packet size (Bytes)	444	773

Summarizing, the average packet size in the UMTS network is greater than the same value in the wired network. This is produced when the packets get into the mobile network. This implies that the routing and processing packet cost is lower than the wired network.
This can benefit to some type of traffic like Web or Mail traffic (best-effort traffic). However, the multimedia streams are penalized by this technique, i.e. if a packet is lost in the UMTS network, the server of the session has to retransmit two packets.
As a result of that, the communication performance decrease and the delay increase in the communication. When the stream is going to be reproduced, if the packets are received with delay, the frames or the audio cannot be reconstructed and this information cannot be reproduced. So, these packets are going to be discarded in upper layers of the protocol stack. This is produced by the RTP protocol. It intends to rebuild the information as the server of the session transmitted, but if a packet were received later than the timestamp of the reproduction would be discarded.
Finally, the packet delay produce gaps and pixelations in the images due to the error correction algorithms applied in the upper layers (RTP, multimedia codecs etc.).

4. Conclusions

In this paper different technologies have been mixed like UMTS, multicast, RTP, TCP, UDP, etc. The performance of the communications has been measured with TCP tunnels in wired and UMTS networks.
We can obtain several conclusions about this paper. The first one is that the throughput provided by UMTS is insufficient to handle services that transmit a huge amount of information.
As we can observe in the third test, when the throughput needed exceed the throughput provided by the UMTS network, the packet loss produce different effects as gaps or pixelation.
Therefore, with the obtained result in the tests we can affirm that the multimedia transmission in the UMTS networks is inappropriate. In this group is not included the multimedia 3G services offered by the providers due to these services usually implements QoS parameters to improve the transmission in the network.
These parameters can be negotiated in the establishment of the communication using PPP protocol. However, when the user try to connect asking to the provider some parameters of QoS, the network reject the communication.
These tests have been realized with two mobile networks provided by two of the Spanish operators with own infrastructures and 3G connectivity services.

Furthermore, the test has been realized in the same conditions each other. The receiver of the session was in the same localization with the same base station of the mobile network.
Therefore, the communications has been made through UMTS networks without other extern factors affecting to the communication as the handover between different coverage areas.

5. Future Work

This paper starts new lines of research. First of all, this paper evaluates the communications in the mobile networks without movement. We have to test the performance of the 3G mobile networks when the device is moving, and later, we have to test the performance of the handover between different coverage areas.
Moreover, it is necessary to introduce the mobility protocol Mobile IP to the scenario and test the advantages offered in the mobile networks. It can improve the mobility support offered and it could allow to the users be always connected to the Internet with independence of the data network that they could be using.
Furthermore, we have to test the HDSPA networks and compare this results with the test realized in this paper.
We purpose a scenario when the multimedia streams are sent without any QoS parameters. The delay introduced by the network affects directly to the traffic multimedia. Therefore, we have to test the improvements in the communication using QoS parameters and compare the results with the test presented in this paper.

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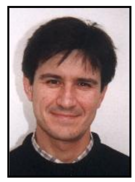

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