

# VoIP performance in SIP-based vertical handovers between WLAN and GPRS/UMTS networks

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**Abstract**—This paper experimentally analyzes the handover performance of VoIP sessions in a wireless overlay of 802.11 WLANs and GPRS/UMTS networks when mobility is handled at the application layer by SIP. It also assesses the impact of handovers on the VoIP call quality perceived by the user by means of the extended E-model. The study reveals that good performance values are achieved when handing over from GPRS/UMTS to WLAN. Additionally, acceptable quality is obtained for handovers from WLAN towards UMTS. On the other hand, unacceptable values are achieved when moving towards GPRS. The main reason is the delay experienced by SIP messages when traversing the cellular network.

**Keywords:** SIP mobility, VoIP, vertical handover, voice quality, experimental, E-model, GPRS, UMTS, WLAN

## I. INTRODUCTION

The widespread popularity of WLAN and the worldwide deployment of third-generation (3G) mobile networks are driving the growth of the mobile Internet access. The supply of both interfaces in a single terminal combines their benefits: cellular connectivity provides wide-area coverage, whereas WLAN offers higher bandwidths and lower costs. One of the key technologies that will benefit from this heterogeneous environment is Voice over IP (VoIP,) which permits to carry out voice calls over packet-switched networks.

One of the most challenging problems for system integration is the provision of seamless mobility support among different access technologies. Several protocols have been proposed for handling wireless mobility [1], but usually, two basic approaches are considered for VoIP services, namely Mobile IP ([2] and [3]) and Session Initiation Protocol (SIP) [4], which handle mobility at the network layer and at the application layer, respectively. SIP has recently acquired more interest due to its adoption by the Third Generation Partnership Project (3GPP) [5] as the signaling protocol for managing real-time multimedia sessions within the IP Multimedia Subsystem (IMS,) which is a new framework for providing IP multimedia services.

This paper presents an experimental evaluation of a VoIP session vertical handover between WLAN (in particular, 802.11) and cellular networks (i.e. GPRS and UMTS) in a wireless overlay scenario. Mobility is handled at the application layer using SIP. A thorough analysis of the different components of the handover delay shows that the

cellular network performance is the main factor contributing to the delay when handing over from WLAN to GPRS or UMTS. Nevertheless, UMTS provides better performance than GPRS. The analysis identifies some potential enhancements to improve the SIP mobility support of VoIP sessions.

Additionally, the extended version of the E-model [14] is used to assess the effect of the handover performance on the quality perceived by the user. Results reveal that good performance is obtained when moving *from* cellular GPRS/UMTS networks to 802.11 WLANs. This is not the case when handing over *towards* GPRS for all the cases tested. On the other hand, the handover *towards* UMTS slightly decreases the perceived quality, but there is still room for some additional degradation due to other impairment factors such as the codec and the end-to-end delay, yet obtaining acceptable quality.

The paper is organized as follows. The next section presents some background and related work. Section III offers a description of the scenario used for experimentation. A characterization of the performance of the GPRS and UMTS networks is presented in section IV. Section V experimentally analyzes the vertical handover performance when using SIP mobility, and it also evaluates the perceived voice quality due to the handover process. Finally, section VI concludes the paper.

## II. BACKGROUND AND RELATED WORK

This section provides additional information on the interworking between WLAN and cellular networks, SIP mobility, handover characterization, and the E-model and extended E-model. It also presents related work on handover characterization between heterogeneous networks.

### A. WLAN and GPRS/UMTS cellular networks interworking

GPRS and UMTS architectures have been defined by the 3GPP [5], and their network architecture can be found in [6]. The point of attachment of the MN to the GPRS and UMTS networks is the GSM EDGE Radio Access Network (GERAN) and the UMTS Terrestrial Radio Access Network (UTRAN,) respectively. The GERAN is composed of Base Transceiver Stations (BTSs) and Base Station Controllers (BSCs.) The UTRAN consists of Node Bs, and Radio Network Controllers (RNCs.) GPRS Support Nodes (GSNs) implement the packet domain functionality in the core network, and act as IP routers with additional capabilities. The Serving GSN (SGSN)

provides security functions, packet switching, routing, and keeps track of the location of mobile stations. The Gateway GSN (GGSN) contains routing information for mobile users and interworks with external packet-switched networks. The GGSN is connected with SGSNs via an IP-based GPRS backbone network. Finally, the Home Location Register (HLR) contains subscriber data and enables access to the packet-switched domain services. The HLR is included within the HSS (Home Subscriber Server,) which contains the subscription-related information to support the network entities that handle calls/sessions.

The proposed architectures to integrate WLAN and cellular networks can be roughly classified into loosely coupled and tightly coupled [7]. In the tightly coupled solution, the WLAN is connected to the cellular core network as any other radio access network (RAN,) such as GERAN or UTRAN. In the loosely coupled approach, WLAN and cellular networks are completely separated and only connected through the Internet, though they may share a common subscriber database for billing and/or authentication.

Several approaches have been proposed to provide an interworking architecture between WLAN and cellular networks. One of them has been standardized by the 3GPP [8], using a new element referred as Packet Data Gateway (PDG,) which is placed adjacent to the GGSN and offers secure access from WLAN to cellular services. This interworking architecture does not allow service continuity between both networks. A different option for interworking between GSM/GPRS and unlicensed networks (e.g. 802.11 WLANs) is Unlicensed Mobile Access (UMA) [9], which has been included within the 3GPP architecture referred to as GAN (Generic access to the A/Gb interface) [10]. UMA presents a tight coupling architecture with the introduction of the UMA network controller (UNC.) The UNC is connected to the mobile operator core network using the same interfaces as the BSCs, and acts as a gateway to the external IP network. As stated by the Fixed-Mobile Convergence Alliance (FMCA) [11], UMA is seen as an interim option until the SIP-based IP Multimedia Subsystem (IMS) framework becomes deployed. The IMS allows the convergence of different transport networks by employing an architecture independent of the access network technology. The heart of the IMS is the call state control function (CSCF,) which performs session control.

### B. SIP Mobility

The Session Initiation Protocol (SIP) [4] is a signaling protocol working at the application layer that, among other things, allows handling mobility in session-oriented services, like VoIP. As stated in [12], SIP supports terminal, session, personal and service mobility. Terminal (or device) mobility gives a certain device the capability to move between IP networks. Session mobility permits a user to maintain a media session even when switching devices. Service mobility refers to the access of services even while users are moving or changing devices and network service providers. Finally, personal mobility allows the usage of the same logical address to address a single user located at different terminals.

Terminal mobility is subdivided into pre-call and mid-call mobility. Pre-call mobility is meant to maintain the reachability

of a device for incoming requests when it moves among IP networks. The process involves the re-registration of a SIP client with its SIP server when it moves from a network to another. Mid-call mobility maintains ongoing sessions when a device changes of IP network. In this case, the mobile node (MN) re-invites the correspondent node (CN.) Specifically, the MN sends an INVITE message to the CN announcing the new IP address adopted. Then, both nodes stop the voice communication using the old address and restart it using the new one with the new parameters.

### C. Handover characterization

The handover process is composed of three phases. The detection phase accounts for the process in which the mobile node discovers that it is under the coverage of a new wireless network. The preparation phase includes the configuration of the IP address on the new network. Finally, the execution phase accounts for the mobility procedure to maintain communication through the new network.

Heterogeneous overlaid wireless network scenarios imply that the terminal may be under coverage of different networks at the same time. In this case, the handover can be triggered by user preferences and policies in addition to (or instead of) connectivity reasons. Therefore, the detection and preparation phases can be carried out without losing connectivity to the network to which the terminal is attached, so the handover execution is the only phase that affects communication performance. This is the reason why this study focuses on the execution phase of the handover. Therefore, the handover delay is measured at the MN as the time elapsed from transmission of the last voice packet in any direction through the old interface, to transmission of the first voice packets (one in each direction) through the new one, as depicted in Figure 1.

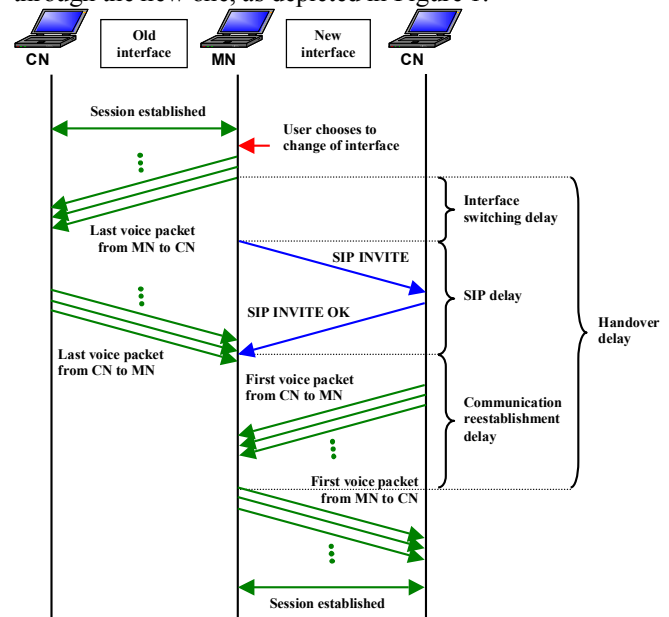


Figure 1. Components of the handover delay

The handover delay is composed of the *interface switching delay*, the *SIP signaling delay*, and the *communication reestablishment delay*. Figure 1 illustrates the message flow

when carrying out a handover between two different interfaces, and the different components of the handover delay.

The *interface switching delay* is the time spent from transmission of the last voice packet in any direction through the old interface to transmission of the SIP INVITE message to the CN through the new one. It depends on the processing time in the VoIP client elapsed in switching interfaces and default routes used, and creating and sending the SIP INVITE message.

The *SIP signaling delay* accounts for the time elapsed from transmission of the SIP INVITE message to the CN to the instant at which the mobile client acknowledges correct reception of the SIP OK received from the CN. Some of this delay is spent by the CN to process the SIP INVITE message and send the SIP INVITE OK message. This processing time in the CN is 20 ms on average in our measurements. The main part of the *SIP signaling delay* is spent in the network, and thus, it depends on network performance.

The *communication reestablishment delay* accounts for the time that elapses from the instant at which the SIP session has been renegotiated until the actual VoIP call resumes (i.e. the MN receives and sends correctly VoIP packets coming from and going towards the CN, respectively.) It depends on the processing time in the VoIP client elapsed in closing the old established voice session and creating the new one.

#### D. E-model and extended E-model

The E-model, defined in ITU-T G.107 [13], is a computational model that predicts the voice quality of a phone call using transmission parameters (e.g. codec, delay, packet loss.) It gives an overall rating for the quality of a call, on a scale from 0 to 100, called the R-factor. It combines different impairments based on the principle that the perceived effect of impairments is additive:

$$R = R_0 - I_s - I_d - I_e + A \quad (1)$$

$R_0$  is the signal to noise ratio.  $I_s$  includes impairments that happen simultaneously with the voice signal.  $I_d$  comprises delay impairments.  $I_e$  includes distortion of the speech signal due to encoding and packet loss. Finally,  $A$  is the advantage factor and represents the degradation in quality accepted by the user in return for the ease of access. The minimum acceptable call quality is obtained when the R-factor has a value of 70 (equivalent to the PSTN call quality.)

The E-model does not take into account that the perceived call quality of a VoIP call varies if the rate of packet loss changes, as it assumes uniformly distributed packet losses over time. The extended version of the E-model [14] incorporates some time-varying impairments that are not considered within the E-Model. One of them refers to the fact that the subjective quality perceived by users changes more slowly than the quality calculated by using the instantaneous packet loss and other impairments. The transitions between burst (period of time during which a high percentage of packets are lost) and gap (the packet loss rate is very low) states are corrected by using exponential decays with time constants of 5 seconds for the gap to burst transition and 15 seconds for the burst to gap transition [14]. Another effect is recency, which is based on the fact that people tend to remember the most recent events. It is

modelled using an exponential decay in the perceived quality, which starts at the end of the last significant burst of packet loss and approaches the average quality level for the call.

#### E. Related work

The integration of heterogeneous networks within a single architecture is a topic that has received a lot of research effort in recent years. Seamless mobility within the interworking framework is an advanced step of system integration.

Previous work has been carried out in order to evaluate the mobility performance in a heterogeneous environment. Some of these studies analyze network-layer based mobility provided with Mobile IP. This is the case of [15] and [16], which experimentally evaluate the performance of MIPv6-based handovers between GPRS, WLAN, and LAN networks for TCP and UDP traffic. Reference [17] also presents an experimental analysis of mobility handled by MIPv6 between GPRS/UMTS and WLAN, and studies its impact on the TCP traffic performance.

Other studies evaluate the performance of SIP to handle mobility between heterogeneous networks. In this sense, [18] presents an analytical study of a vertical handover between UMTS and WLAN networks managed by SIP. An experimental evaluation of a SIP-based vertical handover in a scenario composed of LAN, WLAN, and GPRS connectivity is offered in [19], although only the delay component corresponding to the SIP signaling is considered.

To the best of our knowledge, this is the first study that experimentally analyzes the components of the complete execution phase of a handover between WLAN and cellular networks (not only GPRS, but also UMTS) when mobility is managed with SIP. Moreover, the handover process is not only evaluated by means of the handover delay, but it also assesses the impact of the handover process on the VoIP call quality.

### III. SYSTEM DESCRIPTION

Experimentation has been carried out within the EXTREME framework [20], a networking experimental testbed of the Centre Tecnològic de Telecomunicacions de Catalunya (CTTC) in Barcelona. The main advantage of this platform is its high automation capabilities that allow automatic execution, data collection and data processing of several repetitions of an experiment. All experiment management commands are sent out-of-band through Ethernet management interfaces. Synchronization accuracies of 200  $\mu$ s on average (400  $\mu$ s max.) are obtained by means of NTP through the management network.

The setup used for carrying out the tests is presented in Figure 2. The scenario is composed of the cellular network infrastructure side and the 802.11 WLAN EXTREME side. The cellular network infrastructure side is the Orange GPRS and UMTS networks, which is used for traffic transport. As for EXTREME, in addition to 802.11 WLAN connectivity, it provides the SIP framework within the experiment: the VoIP Mobile Node (MN,) the SIP server, and the Correspondent Node (CN.)

The 802.11 WLAN and the GPRS and UMTS production networks of Orange are interconnected using a loosely coupled

approach. This interworking architecture should be understood as a pre-IMS or initial WI-FI SIP scenario, as defined by the FMCA [11], where SIP servers are placed outside the cellular network. The scenario is built using IPv4 as network-layer protocol.

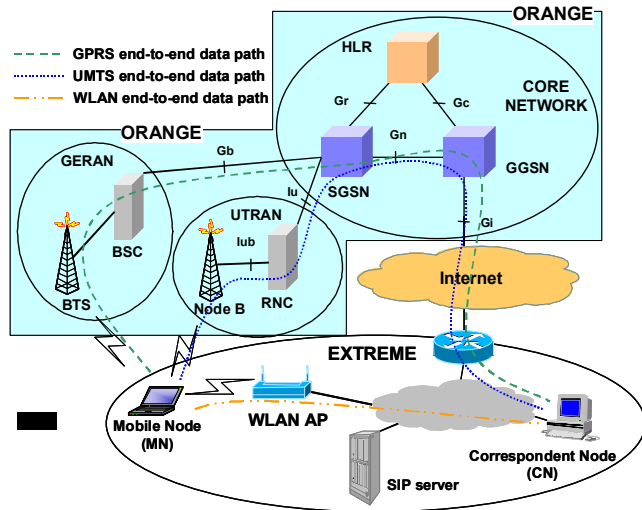


Figure 2. Scenario setup

All computers within EXTREME are Pentium IV PCs 512MB of RAM memory. They all run Linux Kernel 2.4.26. The AP carries an Atheros-based WLAN card with the Madwifi driver [21]. The MN has different interfaces. The Sony Ericsson T630 phone, which operates with 4+1 slots with CS-2 encoding, supplies the GPRS interface, and it is connected to the MN via USB. And the Option Fusion PCMCIA card provides both UMTS and WLAN interfaces.

SIP services are offered by SER (SIP Express Router) [22] acting as SIP server. The Kphone [23] open source SIP user agent is used as the VoIP client. Kphone has been modified in order to support application-layer terminal mobility based on SIP over different network interfaces (i.e. GPRS, UMTS and 802.11 WLAN).

Two measurement setups have been established. In the first one (used in section IV,) the GPRS and UMTS networks were characterized by means of the MGEN traffic generator (version 3.3a8.) Constant bit rate UDP flows with different packet sizes were sent between the MN and the CN. The GPRS and UMTS end-to-end paths depicted in Figure 2 (uplink and downlink) were characterized.

In the second setup (used in section V,) a VoIP call is established between the MN and the CN using the modified Kphone softphone. At each time instant, data traffic flows through one of the three end-to-end data paths depicted in Figure 2. The evaluation in section V focuses on the execution phase of the handover, as it is assumed that the terminal is under coverage of both networks simultaneously. This implies that the MN is attached simultaneously to both WLAN and cellular (GPRS or UMTS) interfaces. The user chooses the instant for starting the handover by pressing a button in Kphone to select the new network. The phases and sequence of events occurring during a handover is in accordance with Figure 1. Thus, SIP signaling messages are sent through the

new interface. For instance, in a WLAN to GPRS handover, the registration procedure is done through the GPRS interface. An Ethereal (version 0.10.9) instance at each interface of the MN monitors and captures all packets. Ethereal traces of both MN interfaces (old and new) were merged to calculate the delay elapsed in each phase. Thus, all timestamps used throughout section V for calculating handover delays correspond to those of Ethereal.

#### IV. GPRS AND UMTS NETWORKS PERFORMANCE

The handover performance partially depends on network performance, because mobility signaling messages are transmitted through the network during the handover process. This section provides an experimental evaluation of the GPRS and UMTS networks performance in terms of end-to-end delay and available throughput in order to assess how packets are affected when transmitted through these networks. All the measurements have been conducted with the MGEN (version 3.3a8) traffic generator.

Two significant traffic flows were chosen to characterize GPRS and UMTS uplink and downlink end-to-end delay (Table 1.) The first one is used to characterize the best possible behavior in terms of delay, as the generated traffic load is very low. One packet/sec of 37 bytes of UDP payload size is generated. The second one is used to characterize how the network treats messages with size similar to that of SIP signaling messages, as this will highly determine the handover delay (see next section.) Apart from this, delay values are expected to give an idea of the possible interactivity levels of VoIP applications in these networks.

Table 1. Measured GPRS and UMTS end-to-end delay

Flow	UDP	GPRS	GPRS	UMTS	UMTS	
	payload size (bytes)					Packets / sec
Best	37	1	465	339	112	112
SIP	575	1	1147	380	216	185

The end-to-end path under study accounts for the path between the MN and the CN (see Figure 2) through the cellular network. The notation *uplink* and *downlink* is used to refer to MN-to-CN and CN-to-MN end-to-end traffic, respectively. Results are averaged over a measurement interval of 15 minutes. Both delay (Table 1) and throughput (Table 2) are measured at UDP payload level.

Table 2. Measured GPRS and UMTS throughput

UDP payload size (bytes)	GPRS	GPRS	UMTS	UMTS
	uplink rate (kbps)	down. rate (kbps)	uplink rate (kbps)	down. rate (kbps)
37 (min)	5,753	22,788	34,534	72,395
1470 (max)	11,342	38,156	60,695	328,563

Table 2 presents the measured GPRS and UMTS uplink and downlink throughput for packets of 37 bytes and 1470 bytes of UDP payload, which correspond to the minimum and maximum packet size cases. Again, flows are selected to have an idea of the throughput ranges offered by the cellular network. By comparing the rate generated by each VoIP codec with these values, one might have a rough idea of the behavior of each VoIP flow.

## V. SIP-BASED VERTICAL HANDOVER

This section presents the experimental results obtained when establishing VoIP calls through the scenario setting described above, and evaluates how the handover process affects the voice quality when using the modified Kphone SIP user agent for handling vertical handovers.

### A. Experimental analysis of the handover delay

Table 3 and Table 4 present the mean values of each of the delay components of the vertical handover for WLAN-to-GPRS (and vice versa) and WLAN-to-UMTS (and vice versa), respectively. These measurements are carried out at the MN, with Ethereal, and the results obtained are averaged over 10 repetitions. Notice that the only component that depends on the codec used is the *interface switching delay*, as it includes the transmission time of the last VoIP packet from the MN to the CN. Therefore, as different codecs generate different packet sizes, there is a difference in transmission time. But this difference is insignificant when compared to the rest of the delay components. As a consequence, one might say that the vertical handover delay is (mostly) independent of the codec used. Measurements carried out confirm this.

**Table 3.** Mean values of the handover performance between WLAN and GPRS

<i>Delay (ms)</i>	<i>WLAN to GPRS</i>	<i>GPRS to WLAN</i>
Interface switching delay	43,38	32,44
SIP delay	1626,53	23,57
Communication reestablishment delay	81,37	83,37
<b>Total handover delay</b>	<b>1757,75</b>	<b>128,25</b>

**Table 4.** Mean values of the handover performance between WLAN and UMTS

<i>Delay (ms)</i>	<i>WLAN to UMTS</i>	<i>UMTS to WLAN</i>
Interface switching delay	44,14	55,6
SIP delay	382,52	20,48
Communication reestablishment delay	79,37	95,2
<b>Total handover delay</b>	<b>536,26</b>	<b>171,28</b>

The *interface switching delay* presents similar values (between 32,44 ms and 55,6 ms) for all cases. The same happens with the *communication reestablishment delay* (between 79,37 ms and 95,2 ms.) Notice that, in this context, *similar* means that the difference of values for the different cases is much smaller than the total handover delay. On the other hand, the *SIP signaling delay* varies depending on the cellular interface used (GPRS or UMTS.) Handovers towards the WLAN interface present similar values (around 20 ms) for both GPRS and UMTS cases. But when performing handovers towards the cellular interface, there are high differences between them. In fact, the *SIP delay* has a value of 1626,53 ms when performing a handover towards the GPRS interface, compared with the value of 382,52 ms obtained when performing a handover towards the UMTS interface. Though packet sizes of SIP signaling messages are not exactly the same as those used for the tests in Table 1, there is a correspondence between those values and the ones presented here.

Another peculiar behavior is observed when comparing the UMTS-to-WLAN and GPRS-to-WLAN cases. The difference

in handover delay might be due to the different processing carried out in the GPRS, UMTS, and WLAN interfaces of the MN, as different devices (and associated drivers) are used in the GPRS and UMTS tests (see section III.)

In conclusion, the main contributor to the total handover delay is the *SIP signaling delay*, and thus, it is the main component to reduce in order to improve the handover performance.

This leads to the identification of some enhancements to improve it. One option would consist in sending the SIP messages through the interface that presents better performance, even if this interface is the old one. For this study, this consists in sending all the SIP messages through the WLAN interface. After sending the SIP messages through the optimum interface, the session would be re-established through the new interface. Another solution would consist in the continuation of the voice communication through the old interface until the receipt of the INVITE OK message from the CN. This solution contrasts with the current operation, which stops the voice communication through the old interface after the user decides to switch the interface. Finally, another enhancement would consist in transmitting the same voice communication through both interfaces during the handover process (or only during some part of it.) This solution would nearly eliminate the handover delay, because the communication would be maintained almost all the time during the handover process.

### B. VoIP quality assessment

This section evaluates the impact of the handover delay experimentally obtained in section V.A on the quality of the VoIP call by applying the extended E-model [14]. Figure 3 plots the resulting R-factor as a function of the time after the handover ends for different VoIP codecs. Notice that these measurements just focus on the impairments introduced by the codec and the packet loss due to the handover process (i.e.  $I_e$  in the E-model,) as a packet loss burst is the more direct consequence of experiencing a handover. Therefore, end-to-end delay would further reduce the VoIP quality obtained. Moreover, it is assumed that there is only packet loss due to the handover process, so as to focus on the effect of the handover.

The parameter represented in the X-axis is important because of the recency (see section II.D.) Furthermore, the range of values represented is chosen to match the average length of a phone call, which is considered to be 2,6 minutes [24]. One might observe that the more restrictive value of R is the one obtained just after ending the handover, i.e. handover at the end of the call. Recall that voice quality equivalent to that of the PSTN is obtained for  $R \geq 70$ .

All figures enclosed in Figure 3 show the same general behavior, i.e. from least to best quality, the order is WLAN-to-GPRS, WLAN-to-UMTS, UMTS-to-WLAN, and GPRS-to-WLAN, which directly matches the order of handover delay presented in the previous subsection. One might also observe that there is a difference in R between codecs given the different effect of losing a packet for each of them due to the different compression levels, and thus, the volume of information lost. In this sense, G.711 shows the best behavior and G.723.1 the worst one.

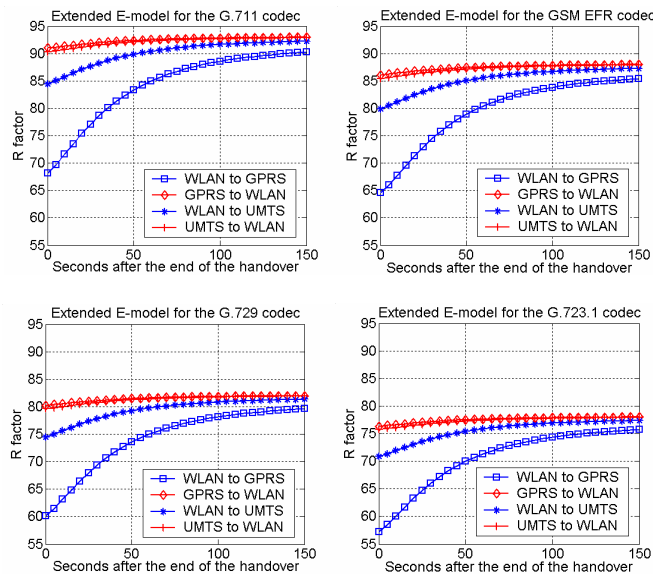


Figure 3. Impact of the handover delay on the VoIP quality for different codecs

The figure shows that the perceived quality when handing over from WLAN to GPRS is unacceptable for all the codecs just after handover happens, because  $R < 70$ . The other handover cases provide better performance results. Handovers from cellular to WLAN provide good performance values, because they reduce the R-factor only a small fraction. WLAN-to-UMTS handovers show a higher impact on the R-factor, although some additional quality degradation due to other impairments (e.g. delay) could still be acceptable to the user. For instance, for WLAN-to-UMTS handovers, 300 ms, 260 ms, 215 ms, and 180 ms of end-to-end delay (in the case of no echo loss in the path) would still provide values of the R-factor above 70 for the G.711, GSM, G.729 and G.723.1 codecs, respectively.

## VI. CONCLUSIONS

This paper presents an experimental analysis of a vertical handover between WLAN and GPRS/UMTS networks when mobility is handled by SIP at the application layer in a wireless overlay scenario. The analysis shows that two out of three components of the execution phase of the vertical handover delay (namely, *interface switching delay* and *communication reestablishment delay*) are similar for both GPRS/WLAN and UMTS/WLAN mobility scenarios. This is not the case for the *SIP signaling delay* when conducting a handover towards the cellular (i.e. GPRS or UMTS) interface, because it depends on the performance of the cellular network. Furthermore, the SIP delay component is higher for WLAN-to-GPRS than for WLAN-to-UMTS case, as it is highly influenced by the end-to-end delay experienced. Some enhancements to improve the handover performance are identified and left as future work.

The effect of the handover process on the voice quality perceived by the user has also been evaluated, by applying the extended version of the E-model. Results reveal that voice quality is acceptable (i.e.  $R \geq 70$ ) when handing over from cellular networks to WLAN and from WLAN to UMTS. On the other hand, the worst behavior is observed for WLAN-to-GPRS handovers. This is a general trend for all four codecs

evaluated, namely G.711, GSM EFR, G.729, and G.723.1. Results on the evolution in time of the perceived quality after a handover occurs have also been presented.

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