AALBORG UNIVERSITY

Master Thesis

SIP-Supported Soft Handover for the Convergence of Wireless Enterprise Networks and 3GPP IP Multimedia Subsystem

by

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Abstract

IP Multimedia Subsystem (IMS) is envisioned to be the common platform for service provisioning in heterogenous networks. A variation of the Session Initiation Protocol (SIP), is proposed for session control in cellular networks based on the IMS architecture. SIP is also used in many corporate networks with 802.11 wireless access. The interconnection of these two SIP-based domains for extended coverage and persistent service is one of the promising steps for heterogeneous network convergence. In this scenario, service access is performed through two different wireless access networks. A solution for migrating existing sessions between these two domains, known as vertical handover is proposed in this thesis. The solution is based on modifications of standard SIP signaling and allows full mobility between corporate and cellular domains. As part of the study, the client requirements from a SIP perspective are also analyzed and a solution in the form of states machine is proposed. Results show that it is feasible to implement a soft-handover with the support of SIP for session establishment at the IP level.

Preface

This Master Thesis is the result of a project titled *SIP-Supported Soft Handover for the Convergence of Wireless Enterprise Networks and 3GPP IP Multimedia Subsystem*. The research carried out constitutes the 10th semester of the International Master of Science (M.Sc.E) in Mobile Communications at Aalborg University, Denmark.

In this report, chapters are numbered using forth running Arabic numbers. Subdivisions into sections are labeled in two levels, describing the chapter number and the section number. A two level labeling is also used for figures, tables, and equations. For example, "Figure 2.3" and "Table 2.4" refer to the third figure and the fourth table in chapter two respectively. References to equations are done in a similar way, but the label is placed in parenthesis as "(3.8)". Literature referenced in this report is listed after the main report. References appear within square brackets, for example, "[2]". A complete list of all acronyms is given at the beginning of the report, preceding the main section.

Acknowledgements

I would like to express my gratitude to my supervisors, Kim Lynggaard Larsen, Oumer Mohammed Teyeb and Hans-Peter Schwefel for their valuable guidance and expert advice throughout this Master thesis. I would like to thank Dorthe Sparre, for her kind help and support. Special thanks goes to all the professors for their dedication and personalized assistance during my studies at Aalborg University.

Carlos Leonel Flores Mayorga

Los logros que se aprecian mas en la vida son los que se obtienen atraves de esfuerzo y dedicacion. Este trabajo va dedicado a mi familia y en especial a mi madre, Elvira.

List of Abbreviations

3GPP	Third Generation Partnership Project
AAA	Authentication Authorization and Accounting
AS	Application Server
BGCF	Breakout Gateway Control Function
BSS	Base Station Subsystem
CDMA	Code Division Multiple Access
CN	Core Network
COPS	Common Open Police Service
CSCF	Call State Control Function
FDD	Frequency Division Duplex
GGSN	Gateway GPRS Support Node
GPRS	General Packet Radio Service
GSM	Global System for Mobile Communications
HLR	Home Location Register
HSS	Home Subscriber Server
IETF	Internet Engineering Task Force
IMS	IP Multimedia Subsystem
IP	Internet Protocol
ISDN	Integrated Services Digital Network
ISUP	ISDN User Part
ISO	International Organization for Standardization
IWU	Interworking Units
LAN	Local Area Network
MEGACO	Media Gateway Control protocol
MGCF	Media Gateway Control Function
MGW	Media Gateway
MN	Mobile Node

MRF	Multimedia Resource Function
NSS	Network Subsystem
OSI	Open Systems Interconnection
PAN	Personal Are Network
PBX	Private Branch Exchange
PHY	Physical Layer
PSTN	Public Switched Telephone Network
QoS	Quality of Service
RTP	Real Time Transport protocol
SDR	Software Defined Radio
SDP	Session Description Protocol
SIP	Session Initiation Protocol
SGSN	Serving GPRS Support Node
TDD	Time Division Duplex
UE	User Equipment
UDP	User Datagram Protocol
UMTS	Universal Mobile Telecommunications System
UTRA	Universal Terrestrial Radio Access
UTRAN	Universal Terrestrial Radio Access Network
VoIP	Voice Over IP
WLAN	Wireless Local Area Network

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Chapter 1

Introduction

The telecommunications market has been historically segmented into three main groups: Home users, Mobile users and Corporate users, having each of these groups different architectures to support specific services. Traditionally, home users are provided with access to internet and telephony services with a wireline and a fixed phone number. Nowadays, the IEEE 802.16d standard has become an alternative for the wired access. A different phone number and a different device are used by public cellular networks in order to provide data and multimedia services. In most of the cases, these services are technology-dependent and vary from one operator to another. A great advantage of cellular networks over wired access is the mobility and the wide coverage offered. An important segment of the telecommunications market consist of corporate networks, which require the implementation of advanced data and voice services in a secure environment. This scenario is depicted in Figure 1.1.

The current network topology, where services are strongly dependent on the infrastructure is changing in order to satisfy different customer requirements in a unified solution. This unified solution proposes a single platform for service provisioning in a promising concept known as *Heterogeneous Network Convergence*. Convergence was first applied for the interworking between fixed and mobile operators providing voice services. The interworking architecture allows telephone calls from fixed to mobile networks and viceversa. In the past, the voice services were mainly associated with fixed and mobile telecommunication networks. This concept has evolved with the development of internet and Voice over IP (VoIP) applications. Nevertheless, VoIP is just a subset of the possible applications offered through internet. In general, applications like

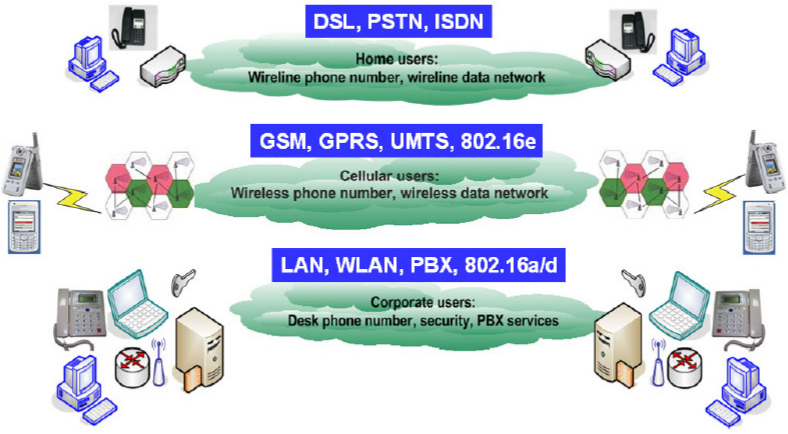

Figure 1.1: Current Heterogeneous Network Topology

speech, data, graphics, video, games and others are part of the revolutionary multimedia services that internet is capable to provide.

Nowadays, a revolution has started and all the services and applications provided by internet are moving to wireless technologies. There is an increasing trend towards mobility, convergence and connectivity anywhere, anytime. This trend is driving the industry to design a common platform in an evolved scenario where 3G, 4G and other wireless technologies coexist and are unified using a single architecture for service provisioning. The vision of this architecture is depicted in Figure 1.2, where services are envisioned to be independent of the access technology and unified by an all-IP network.

The first step towards convergence has been given in 2004, when the Third Generation Partnership Project (3GPP) approved the standardization of the IP Multimedia Subsystem (IMS). IMS is an architecture designed to enrich the way that people communicate with each other, combining data, voice and multimedia in a single session independently of the access technology [1].

Due to the massive deployment of Wireless Local Area Networks (WLAN) and VoIP, the initial efforts in 3GPP are oriented to make possible the convergence between cellular technologies and WLAN. However, the main targeted market is on the corporate users. Owing to cost reasons and mobility benefits, these users have massively adopted WLAN technologies as part of their voice and data networks. This consideration slightly changes the technology convergence concept into cellular technologies and WLAN-corporate convergence, which is the topic addressed in this thesis.

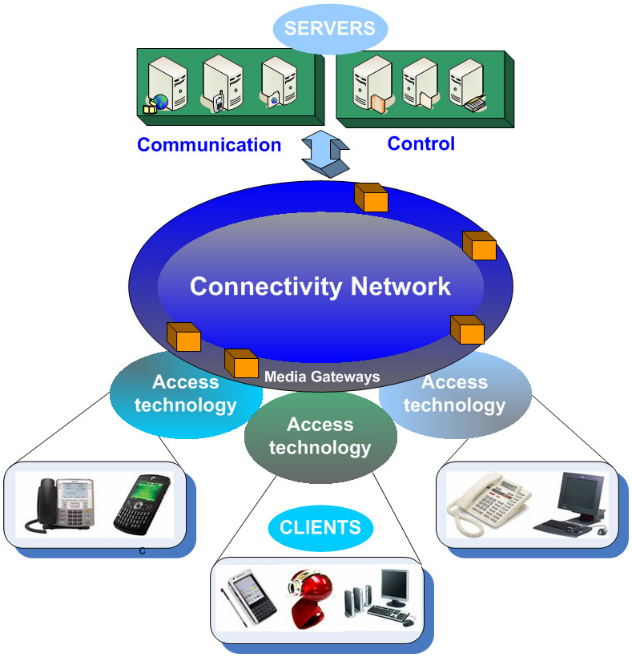

Figure 1.2: Future Heterogeneous Network Topology

In this convergence scenario, service access is performed through two different wireless access networks. The mobile terminals must be capable to connect to any of these access networks and to migrate existing sessions from one technology to another [2]. This process is known as vertical handover. The scope of this thesis is to propose a signaling solution to support vertical handover in heterogenous networks. Since there is no centralized architecture for handover between cellular and corporate WLAN, a solution needs to be proposed. The solution offered in this thesis is based on the IMS architecture and one of its key protocols known as Session Initiation Protocol (SIP). The proposal consist of the signaling to support handover and it will be introduced in the following chapters.

After this brief introduction, Chapter 2 gives a review of previous work in order to understand the problem description and depict the exact scenario. After that, the specific outcome of the project is set. Chapter 3, introduces the background theory required to understand the developed solutions, the main technologies are presented together with key concepts to arrive to Chapter 4, where the proposed solutions are given. Chapter 5, presents the test bed for the concept implementation of the proposed solutions together with the results obtained and their analysis. Finally conclusions and future work are given in Chapter 6.

Chapter 2

Project Description

This chapter gives an introduction about the challenges for the convergence of 3GPP and corporate wireless LAN networks. The scenario and related work is described first in order to arrive to the main problem: vertical handover. The main and secondary objectives of this thesis are defined to finalize the chapter.

2.1 Introduction

Internet Protocol (IP) Multimedia Subsystem (IMS) is specified as part of the core network in the Universal Mobile Telecommunications System (UMTS) release 5 and later [3]. UMTS describes a standard which enables a flexible and high data rate transmission of any kind of data over cellular networks. The introduction of UMTS responds to the high data rate requirements originated by the development of new services and applications. The IMS platform provides services for IP based communications like voice, video telephony, video streaming, instant and multimedia messaging, multimedia gaming and virtual reality. The key of the IMS success can be seen as the ability to interconnect users in heterogenous networks.

As the trend towards convergence and mobility increases, heterogenous wireless access technologies are required for extended coverage and persistent wireless service. One of the challenges of convergence is to facilitate handover when the mobile user comes across these different access technologies. An example of this scenario is specified by the 3GPP, with the proposal of providing seamless service continuity between UMTS and Wireless Local Area Networks (WLAN) [4]. The interworking between third generation (3G) cellular systems and WLANs is considered as a suitable and viable evolution towards next generation of wireless networks. Interconnection between 3GPP and WLANs requires interworking capable to support different levels of service at the session negotiation level considering QoS requirements. This interconnection depends mainly on the SIP protocol. SIP, as defined by the Internet Engineering Task Force (IETF) would be the base for the interconnection of future IP-based networks [24].

There is a wide range of WLAN technologies, some of the most popular are the well known set of IEEE standards 802.11x. The main difference between these technologies is in the first two layers. Typically, frequency of operation, modulation, coding, data rates and coverage vary among different standards. However, the network layer presents similar characteristics for all of them. In order to propose a unified solution for interconnection, the IP layer appears to be the most convenient due to the common use of IP packets. The massive adoption of IEEE 802.11x standards in corporate networks to support mobility, makes this technology the ideal candidate for convergence. This suggests that the 3GPP - wireless corporate networks interworking should be built based on the IP protocol. In particular, this project analyzes the interworking issues between UMTS and wireless corporate networks for providing connectivity, mobility and handover management.

2.2 Problem Description

UMTS proposes a variation of SIP for session control and end-to-end QoS management. SIP is also used in many current corporate networks based on 802.11 wireless access. The interconnection of these two SIP-based network domains for extended coverage and persistent wireless service environments is one of the promising steps for wireless network convergence.

In the interconnection between a corporate network and a public network three main challenges are present. The first, SIP as described in the IEFT is slightly different than the SIP protocol used in 3GPP, therefore a SIP "translator" is required in order to have efficient communication. Second, the signaling required for inter-domain mobility needs to be defined. Finally, the inter-domain mobility should not affect the user experience in access/continuity of services.

A recommendation for the connectivity and mobility between an IP-PBX based network and an IMS based 3GPP mobile network is specified in the SIPForum Recommendation [5]. This specification is known as SIPconnect [6], and introduces the guidelines in order to provide connectivity to the corporate users in the public domain. The solution gives the possibility to register every IP-PBX user to an IMS (non 3GPP compliant) with a single registration procedure and how to receive calls from the IMS. In [7], the SIPconnect network architecture is modified and an improvement is done to consider users mobility. Additionally, the case of a corporate user with 2 subscriptions, but only one number for setting up and receiving calls in both PBX (corporate network) and IMS (public network) is proposed. The potential connectivity to IMS and access to IMS services for all the corporate users without the need of an individual subscription of each of them in the IMS is also presented.

However, the main problem is handover, which is still an open issue, specially considering the lack of a centralized infrastructure for heterogeneous networks management. Therefore, in order to provide a complete solution with full mobility, a handover mechanism is required.

2.3 Scenario

Figure 2.1, depicts the interconnection scenario. A corporate network with 802.11 wireless access technologies and an IP-PBX for voice and data communications. The extended network architecture includes interconnection with a 3GPP IMS-based network.

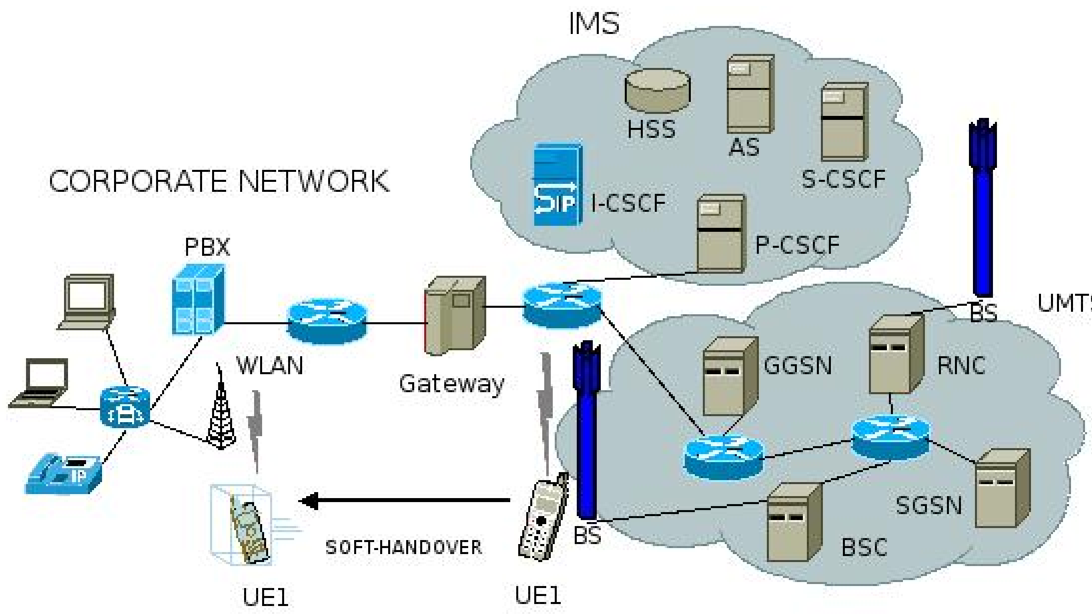

Figure 2.1: UMTS and corporate network interworking scenario

As the SIP protocol is used in both network domains for QoS signaling, and in certain mobility cases also for handover signaling, adequate mechanisms for the QoS and handover control have to be developed, in particular for handover cases between the two domains. The wireless link (Layer 1 and 2) thereby provides important indicators for hand-over prediction and for current/expected future performance behavior of the ongoing connection. In the proposed scenario for convergence it is possible to distinguish the following types of handover.

- Corporate-WLAN to UMTS handover: In this case it is possible that the user losses the connectivity with WLAN and find a UMTS network to establish a new connection.
- UMTS to Corporate WLAN handover: Being connected to the UMTS network, the user decides to handover to the corporate network.

2.4 Project Scope

The goal of this project is to develop mechanisms for SIP-supported soft-handover between the corporate domain and the public operator network. The expected outcome of the project can be summarized in the following lines:

- To propose a signaling mechanism to allow a vertical soft-handover between corporate and public domains.
- To study the network criteria selection and the appropriate parameters for handover prediction.
- To study the interaction between SIP clients and the network for a efficient handover.
- To propose an algorithm for interconnection considering the above guidelines.
- To implement the concepts proposed in an experimental set-up.

Chapter 3

Theoretical Background

This chapter gives an overview of the main technologies described in this thesis. It starts with a description of UMTS, its architecture, air interface and the services it provides. The term public domain is used to refer to the cellular network. Following the cellular system description, a brief introduction to corporate networks, the evolution to IP services and VoIP is provided together with a brief explanation of mobility support through wireless access. The term corporate domain is used to refer the corporate network. After describing the two main domains for convergence, IMS is introduced as an architecture to unify these heterogenous networks. The IMS architecture and the and main protocols are reviewed. The chapter finalizes with an introduction to overlay networks and handover which is the main problem to solve in the convergence scenario.

3.1 UMTS

3.1.1 Introduction

Telecommunications are constantly evolving, bringing developments of new wireless technologies and techniques around the world. These parallel developments make evident the necessity of common agreements towards standardization. The purpose of this standardization is to ensure identical specifications for the different parts involved. The UMTS was specified to ensure equipment compatibility based on the standardization of the Universal Terrestrial Radio Access (UTRA). 3GPP specifications are based on evolved GSM specifications, now generally known as the UMTS system [8].

3.1.2 UMTS Architecture

UMTS describes a standard which enables a flexible and high data rate transmission of any kind of data over cellular networks. The introduction of the UMTS standard responds to the high data rate requirements originated by the development of new services and applications. The general architecture of UMTS is depicted in Figure 3.1. This architecture allows coexistence with second generation systems to preserve previous operator investments.

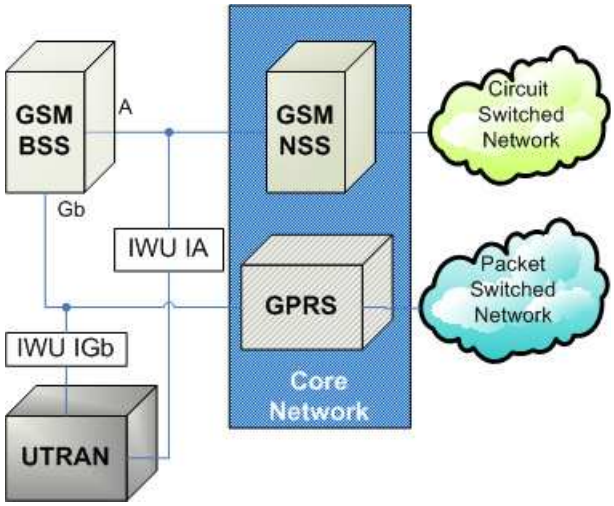

Figure 3.1: UMTS general system architecture

In a Global System for Mobile Communications (GSM), radio coverage is provided by the Base Station Subsystem (BSS). The Network Switching Subsystem (NSS) is used to switch circuits with connections in the mobile radio network and to forward them to the fixed data network. The introduction of General Packet Radio Services (GPRS), allows to increase the data rate transmission, with packet switched based circuits. The next evolution introduced by UMTS is based on existing GSM/GPRS network infrastructure but uses different bandwidths and air interface. UMTS includes its own subsystem for radio coverage known as Universal Terrestrial Radio Access Network (UTRAN). The core network for transmission and switching (CN) connects the GSM and GPRS networks to the UTRAN Subsystem by interfaces called Interworking Units (IWU).

3.1.3 Air Interface

The air interface in UMTS is called UTRA, and uses Code Division Multiple Access (CDMA) technology as the access method with frequency ranges between 1.9 and 2.2GHz.

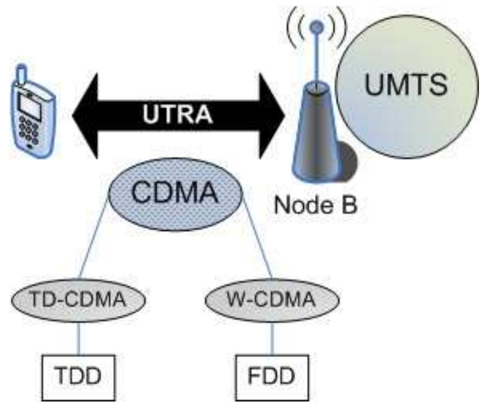

Figure 3.2: UMTS air interface

In order to have efficient subscriber multiple access, CDMA is implemented into two different transmission modes. The first is Wide Band CDMA (W-CDMA) which is used in combination with Frequency Division Duplex (FDD) and provides two frequencies with a fixed space, one for transmission and the other for reception. The second is known as Time Division CDMA (TD-CDMA), where CDMA is used in combination with Time Division Duplex (TDD). In this mode only one frequency is used for both directions but in different times. The Air interface is depicted in Figure 3.2.

W-CDMA

In CDMA each user is assigned an individual and unique code sequence or spreading code, which is used to encode its information-bearing signal [9]. Base Stations send data to several users on the same frequency. Since the code is unique, only one user is able to recover the data, for the rest of users, the information remains unreadable. Codes provide security from interception. Since the data is spread in frequency, the signal becomes insensitive to narrowband interference. Frequency planning becomes irrelevant in W-CDMA, because all users transmit in the same frequency. In W-CDMA operation mode, channels are arranged in pairs consisting in a downlink and an uplink separated by a constant channel spacing called duplex spacing. The Uplink is allocated in the lower frequency whereas the downlink is allocated in higher frequency. A bandwidth of 5Mhz is available to every user. Figure 3.3, depicts the spectrum allocation in W-CDMA.

Data rates up to 2.8Mbit/s can be obtained depending on spreading factors, code rates and other parameters [8].

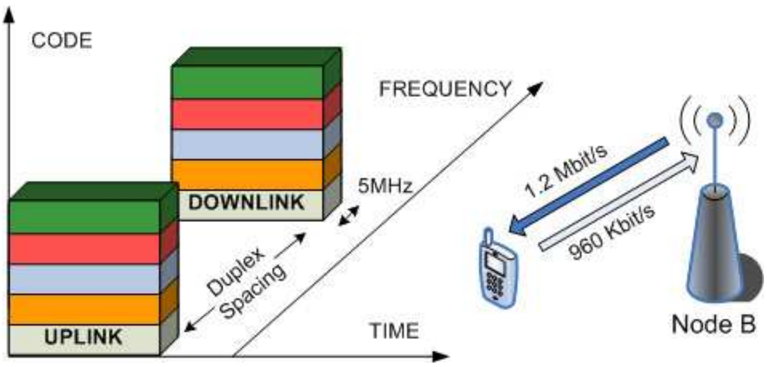

Figure 3.3: FDD CDMA frequency allocation

3.1.4 Services

The UMTS architecture allows the provision of sophisticated communication services. The higher data rates with respect to second generation networks facilitates the introduction of new services. The data rates required via the air interface vary depending on the application/service. Possible services are:

- Location based services
- Speech and Video telephony/ video conferencing
- Video on demand
- Other online services: shopping, literature, translations

The aforementioned applications differ in the delay the individual services allow and can be classified into four traffic classes. Table 3.1, shows the main parameters considered [10].

- **Conversational Class.** This class is very delay sensitive and includes services such as speech and video telephony, or video games.
- **Streaming Class.** The main concern for these services is that the different information units such as video and audio are transferred as synchronously as possible. A typical example would be video streaming on demand.
- **Interactive class.** This class includes web-browsing or interactive games, data integrity is more important than delay.

Traffic Class	Conversational	Streaming	Interactive	Background
Maximum bit-rate(kpbs)	< 2048	< 2048	< 2048 over-head	< 2048 over-head
Delivery order	Yes/No	Yes/No	Yes/No	Yes/No
Maximum SDU size(octets)	≤ 1500 or 1502	≤ 1500 or 1502	≤ 1500 or 1502	≤ 1500 or 1502
SDU format information	X	X		
Delivery of erroneous SDU	Yes/No/-	Yes/No/-	Yes/No/-	Yes/No/-
Residual BER	$5x10^{-12} \sim 5x10^{-6}$	$5x10^{-12} \sim 5x10^{-6}$	$4x10^{-3} \sim 6x10^{-8}$	$4x10^{-3} \sim 6x10^{-8}$
SDU error ratio	$10^{-2} \sim 10^{-5}$	$10^{-1} \sim 10^{-5}$	$10^{-3} \sim 10^{-6}$	$10^{-3} \sim 10^{-6}$
Transfer Delay(ms)	100(max)	100(max)		
Guaranteed bit rate(kbps)	< 2048	< 2048		
Traffic Handling Priority			1,2,3	
Allocation Priority	1,2,3	1,2,3	1,2,3	1,2,3
Source statistics descriptor	X	X		

Table 3.1: UMTS attributes defined for each traffic class [10]

- **Background Class.** This class has no specified delays. However, data integrity is very important. An example would be the download of e-mails in the background.

3.2 Corporate Networks

A corporate network known also as "Enterprise Network" is defined as a network that provides the facilities for communications, processing and storage resources of the corporation. The main objective is to make all the resources available for users within the corporation [11].

3.2.1 Corporate Networks Evolution

The evolution of corporate networks has been driven by the necessity of having cost-effective and reliable telecommunication networks anywhere, anytime. The development of business among different countries and in different locations requires communicating business users to increase productivity. Instead of having expensive dedicated resources, an appealing and practical solution to extend the coverage is to combine a corporate and a public network and make use of the "best link" available. The term "best link" would depend on bandwidth, cost, coverage, security among other aspects. The evolution from circuit switched to packet switched communications has been one of the major steps within corporate networks. In this evolution, multimedia and data are transported over the same packet switched circuit.

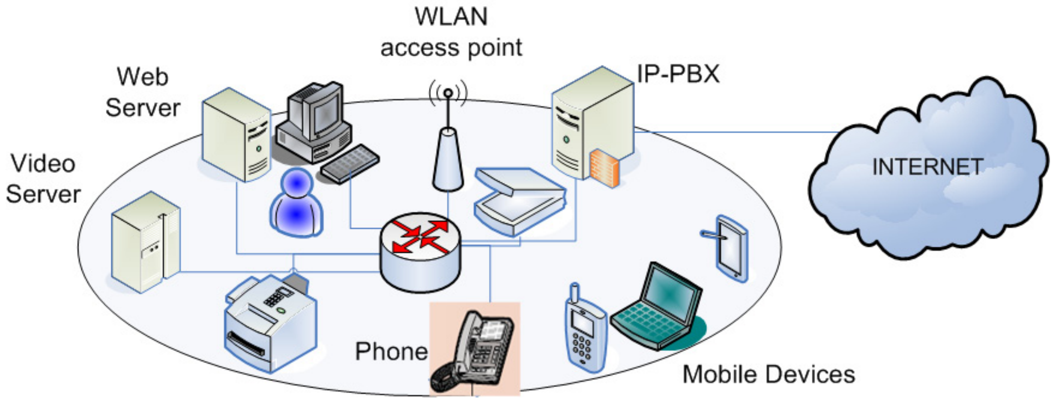

Figure 3.4: Typical Corporate Networks components

Figure 3.4, shows typical corporate network components. A user in a corporate network is able to access to a variety of services defined according to the requirements of the specific corporation. Services offered could be multimedia and data, some typical examples: voice mail, conference calling, interactive voice response, automatic call distribution, data transfer, mail, web-based applications and others. In order to support these different services the network architecture might include different components.

3.2.2 Architecture

The telecommunications architecture of typical corporate networks include the followings systems:

- A core communications switching system (PBX/IP-PBX for example), capable to provide call processing features and data transference.
- A management system, capable to support fault and configuration operations.
- Call accounting system, capable to analyze and process call records to generate and maintain billing schemes and traffic reports.
- A voice messaging system, offering complementary services to the basic answering system.

3.2.3 PBX/IP-PBX

A PBX is a communications system designed to support voice applications. It performs call processing functions like call answering, dialing and transaction features as hold transfer, call forward and conference. In an IP-PBX, voice is sent over IP, in the form of packets.

3.2.4 Protocol Layering

Corporate network communications and in general computer communications are similar to human communications as the rely on mutually agreed patterns of interaction called protocols. A protocol defines the format, content, and order of messages used by communicating entities to achieve a specific task [12]. Communication is accomplished with a set of protocol layers known as network protocol stack. The International Organization for Standardization (ISO), defined a seven layer protocol called Open Systems Interconnection (OSI) model. In this model each layer provides a service to the layer above and relies only in the layer below. In practice, the OSI model is used as a four layers approach known as Internet protocol suite. Figure 3.5, shows the protocol architecture. Understanding of the network protocol stack is of great importance in order to understand how different access technologies can interact each other. The main architectural differences in heterogeneous networks are located in the first two layers in the OSI model and in the first layer in Internet protocol suite.

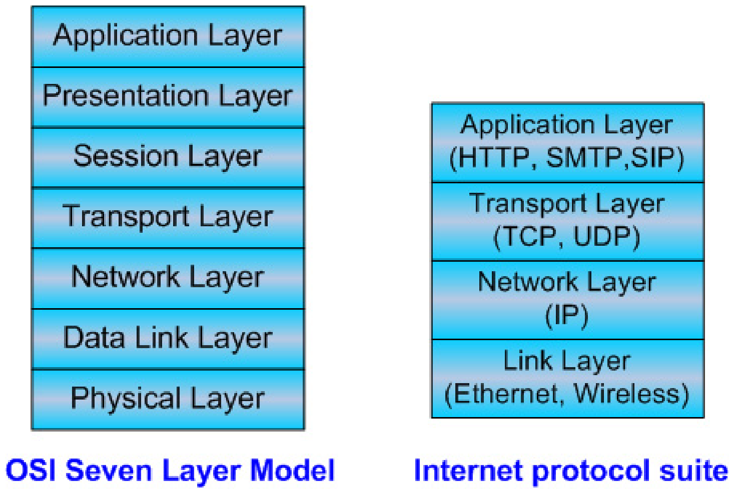

Figure 3.5: Layered Protocol Stack

3.2.5 Mobility in corporate networks

In order to support mobility in corporate networks, a common trend is the implementation of wireless access with low cost technologies like IEEE 802.11 WLAN. Having mobility, Voice over IP (VoIP) has become a common application enabling users to make telephone calls in data networks with a consequent reduction in costs. Wireless networks in a convergence scenario contain a mixture of heterogenous access networks. The increasing business advantages of having connectivity and service availability anytime, anywhere, implies that mobile devices in corporate networks should be capable to access to these different access technologies. The support of multiple radio access technologies is achieved through multimode devices and software defined radios (SDR). The first solution includes several network interface cards and a mechanism to switch between them, whereas the SDR solution implements the radio functions as software in a common hardware platform.

3.3 IP Multimedia Subsystem

3.3.1 Introduction

The increasing demand for telecommunication services, anytime, anywhere, and the development of new technologies are driving the evolution of mobile communications. Traditional services offered only by Internet are now possible in 3G networks. 3G networks aim to merge two of the most successful paradigms in telecommunications: cellular networks and Internet [13]. The IP Multimedia Subsystem (IMS), is a system architecture designed by the wireless standards body 3GPP to integrate different technologies in a common IP platform. In this convergence scenario, the IMS is the key component in the 3G network architecture in order to provide ubiquitous access to all the services available on internet. This implies that different access technologies like GSM, UMTS, WLAN and others can coexist, moreover, the integration of the different services they provide is possible. Figure 3.6, depicts the scenario for seamless service provision.

3.3.2 IMS Standard Overview

The IMS is a network architecture based on internet protocols which enables the efficient provision of a set of integrated multimedia services. IMS facilitates the availability of web browsing, e-

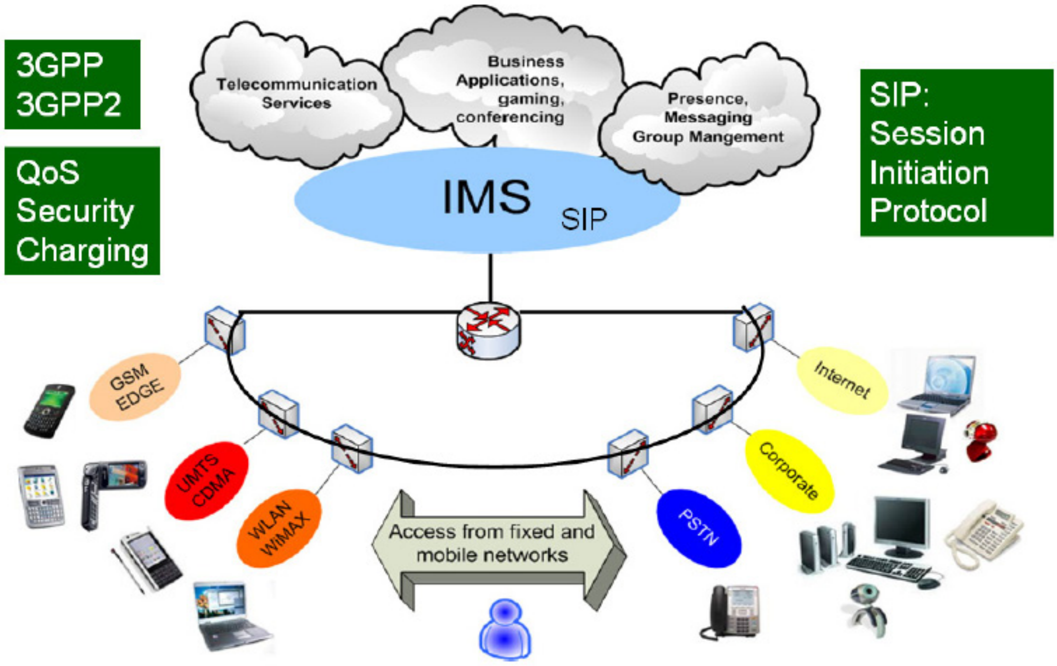

Figure 3.6: IMS Seamless service provision

mail, instant messaging, VoIP, video conferencing, telephony and other services over a common IP platform, independent of radio access technologies. The following lines present a brief description of the common protocols used in IMS.

- Session Initiation Protocol (SIP). This protocol is used to establish and manage sessions over IP networks [24].
- Diameter. The Authentication Authorization and Accounting (AAA) protocol is based on Diameter [15].
- Common Open Police Service (COPS). Is used to transfer polices between Policy Decision Points and Policy Enforcement Points [16].
- H.248. This protocol is used by signaling nodes to control nodes in the media plane. It is also known as Media Gateway Control protocol (MEGACO) [17].
- Real Time Transport protocol (RTP). Used to transport real time media like video or audio [18].

3.3.3 IMS Architecture

The architecture of IMS consists of a set of functions linked by standardized interfaces. 3GPP does not standardize the nodes, but the functions, this allow operators to have some freedom regarding hardware implementations. The IMS layered architecture is depicted in Figure 3.7. The application layer includes application servers to execute value-added services for the user. The control layer involves network control servers for managing call or session set-up, modification and release, and finally, the connectivity layer specifies routers and switches, both for the backbone and the access network.

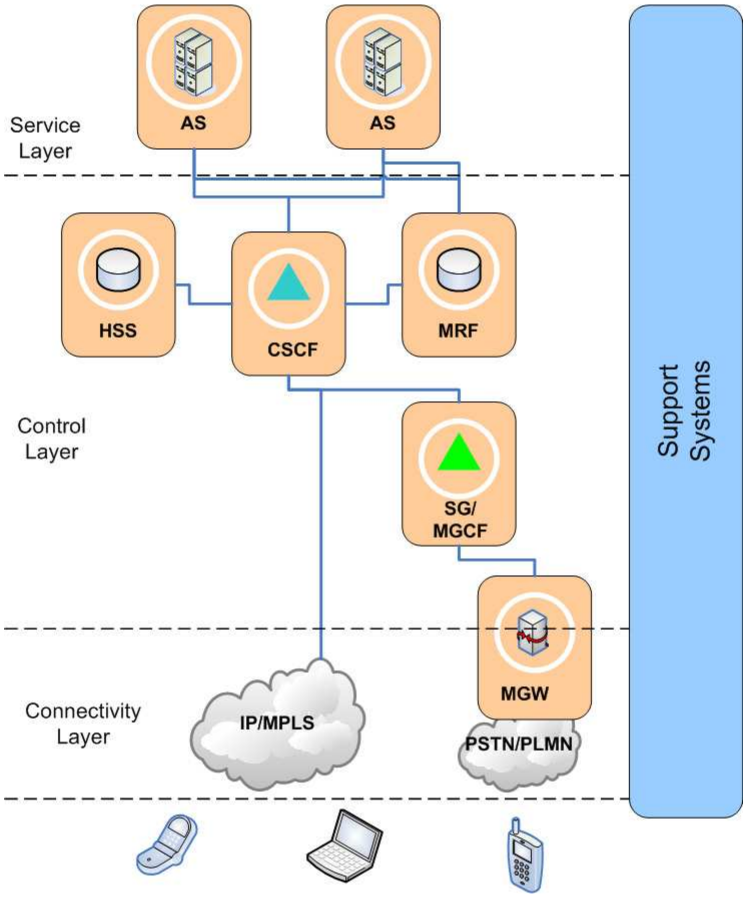

Figure 3.7: Simplified view of the IMS layered architecture in IMS

The main elements in the IMS architecture are described in the following lines:

- Proxy CSCF (P-CSCF). In the signaling plane, it is the first point of contact between the IMS terminal and the IMS network. All the requests initiated by the IMS terminal or destined for the IMS terminal go across the P-CSCF. The P-CSCF can be located in the visited network or in the home network. The P-CSCF acts as a SIP proxy server.

- Interrogating CSCF (I-CSCF). Defined in the home network, the I-CSCF, retrieves user location information and routes the SIP request to the appropriate destination, additionally it provides and interface to the HSS. This interface is based on the Diameter protocol.

- Serving CSCF (S-CSCF). Defined in the home network, the S-CSCF is the central node of the signaling plane. It performs the registrar functions. Like the I-CSCF, the S-CSCF also implements a Diameter interface to the HSS.

- Home Subscriber Server (HSS). This component provides AAA functionality and unique service profile for each user. It can be compared with the Home Location Register (HLR) in a GSM system. The HSS contains all the data subscription required to handle multimedia sessions. These data include, location information, security information, user profile information and the S-CSCF allocated to the user.

- Media Gateway Control Function (MGCF). Acts as a signalling gateway, which controls Media Gateway and performs protocol conversion between the ISDN User Part (ISUP) and SIP.

- Media Gateway (MGW). Interacts with MGCF for resource control.

- Multimedia Resource Function (MRF). Controls media stream resources.

- Breakout Gateway Control Function (BGCF). Selects the network in which PSTN breakout is to occur.

- Application Servers (AS). Offer value added services.

Figure 3.8, shows a general diagram for the interaction between the different CFCSs functions. It should be noticed that implementation of the CSCFs can be done in the same hardware or separately, but the functions performed by each entity are standardized.

The IMS core system is complemented with an application layer to form the entire framework to support services.

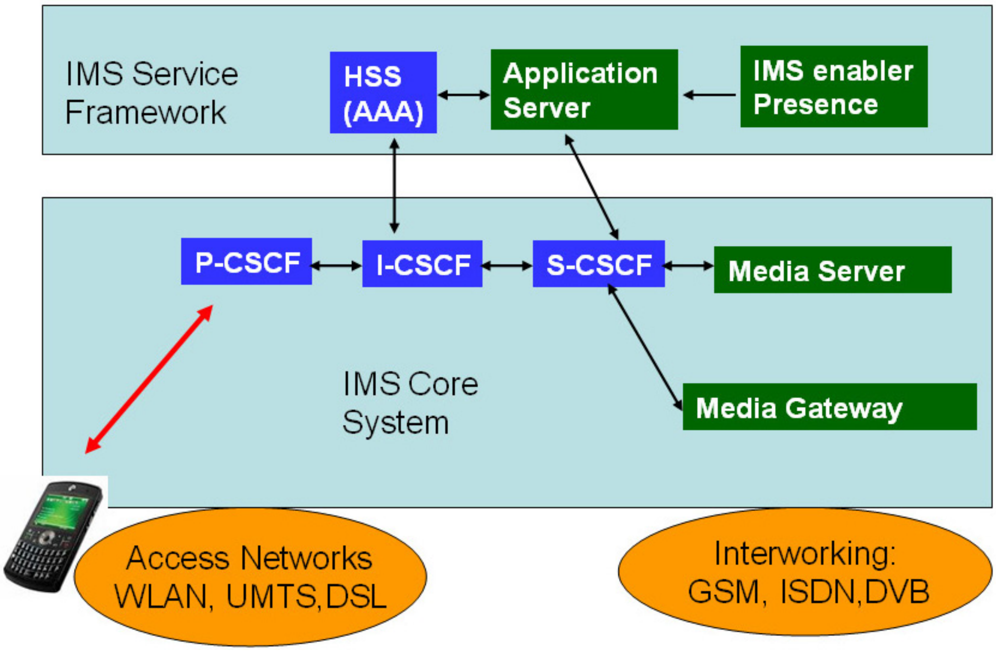

Figure 3.8: Interaction between CSCFS in IMS

3.4 Handover

This section introduces general concepts about handover. These concepts will be useful in order to understand the SIP-supported handover between heterogeneous networks, as proposed in this thesis.

In the convergence of heterogeneous access networks, internet-working is aimed to provide mobile users with ubiquitous connectivity when moving across different networks. The integration of these different technologies requires the design of intelligent handover mechanisms and location management algorithms to enable continuity in services offered to users.

In order to gain access to different access technologies, the mobile device should support multiple wireless network interfaces. The scenario described in this thesis, for the convergence of heterogeneous networks, is similar to the architecture for the so called "Wireless Overlay Networks", presented in [19], [20]. This architecture consists on building-size, metropolitan and regional data networks and is shown in Figure 3.9.

Wireless overlay networks are composed of a hierarchical structure consisting on overlapping cells with its own characteristics in terms of coverage, capacity, bandwidth, latency, and technol-

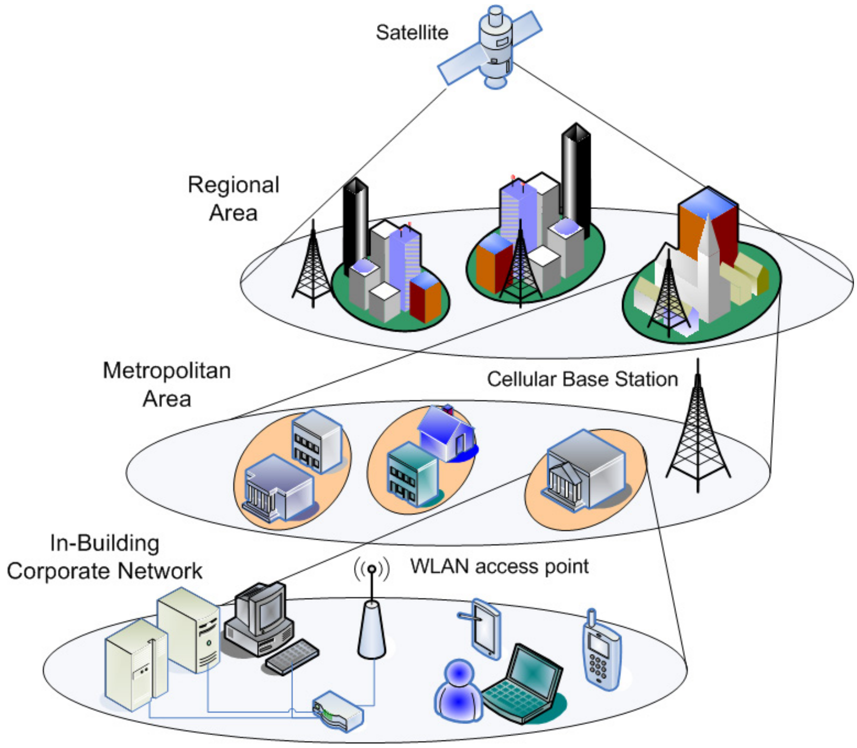

Figure 3.9: Overlay Wireless Networks

ogy. In general, higher levels in the hierarchy provide lower bandwidths and higher delays over larger coverage areas. It can be seen that service areas are overlapped, an example could be the satellite link covering a group of small cities and serving as an umbrella for UMTS covering a cell in a city. The area covered by the WLAN is also covered by the UMTS and satellite technologies. Having more than one access technology available, the selection of the network to be connected to, can be done according to different criteria and therefore the mechanism to switch between access technologies must be provided. Figure 3.10, shows existent heterogenous wireless networks standards and its overlapping nature in terms of coverage, from a corporate user perspective the convergence increases the coverage of the network and therefore the connectivity.

In order to understand the requirements and considerations for a seamless handover the following lines describe some basic concepts.

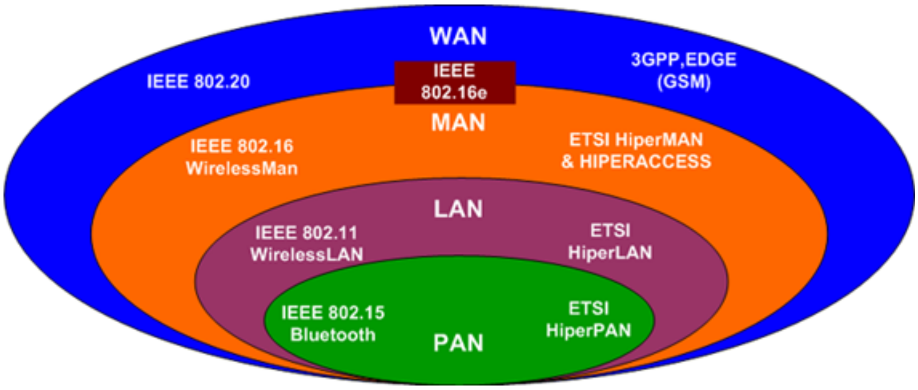

Figure 3.10: Coverage comparison of Overlay Wireless Networks

3.4.1 Handover in Heterogeneous Networks - Overlay Networks

Based on the architecture shown in Figure 3.9, handover can be seen from different points of view. Understanding these different perspectives is the base to select the mechanisms for the proposed SIP-supported handover.

Hard-Handover and Soft-Handover

A hard-handover happens when the mobile node being connected to an access point, with an on-going session, loses connectivity due to the change of access point, after that, a new connection is established. Since communication is lost for a short period, this introduces a service interruption from the user point of view. Soft-handover allows the mobile node being connected to multiple access points in different networks, when the handover happens the connection is created in the target access point before the old access point releases the connection, making the process transparent for the user.

Anticipated and unanticipated handover

In some cases, the mobile node would prefer to perform the handover in heterogeneous networks. For example, if the mobile node is currently in an ongoing video session handled by a cellular network and the PHY of its device detects the presence of a WLAN network, considering just cost and bandwidth reasons, the obvious selection is to handover to the WLAN detected. Anticipated handover is the one that the mobile node will always want to perform. Unanticipated handover on the other hand does not include the preferences of the mobile node.

Horizontal and Vertical Handover

A handover performed when a user moves from one cell to another using the same access technology is called horizontal or homogeneous handover, a typical example could be a user moving between two cells in a cellular system. A handover performed when a user moves between different access technologies is called vertical or heterogeneous handover. Figure 3.11, shows both cases. The study in this thesis is referred to the implementation of vertical handover.

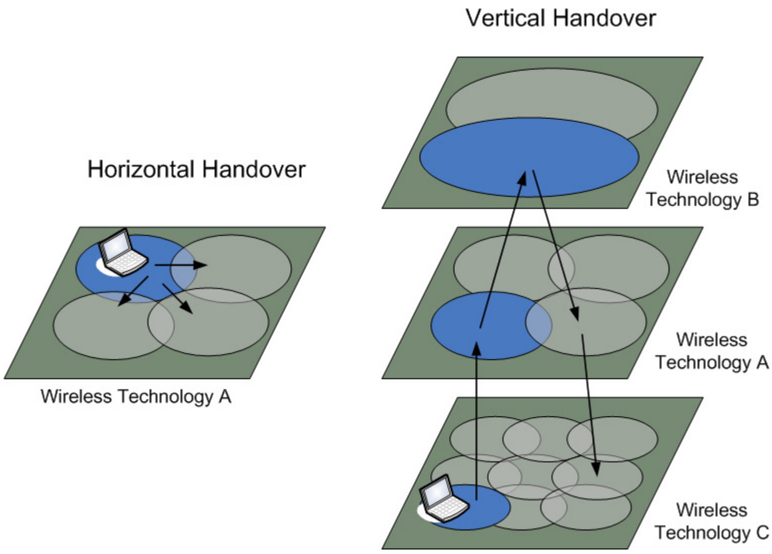

Figure 3.11: Handover between heterogeneous networks

Upward-vertical handover and Downward-vertical handover

If the cell size and available bandwidth are considered, the handover performed in heterogeneous networks when a user moves from a network with larger cell size and usually lower bandwidth to a network with lower cell size and usually higher bandwidth is called Downward-vertical handover, and example could be a user moving from WLAN to PAN. On the other hand, the handover that is performed to a network with higher cell size and generally lower bandwidth, is called Upward-vertical handover.

3.4.2 Vertical Handover process

Since the problem addressed in this thesis is the handover for the convergence of heterogeneous networks, defined as vertical handover, the process is explained in this section with some detail according to [21]. The process can be divided into three steps:

- **Network Discovery.** In this initial step, the mobile node searches available wireless networks by listening service advertisements broadcasted by different technologies. In order to make this step feasible, it is assumed that the mobile node has multiple interfaces.

- **Handover decision.** Once the available networks are discovered, the next step is to decide, if possible, whether or not to perform the handover. Due to the differences between access technologies, the decision can be driven by many factors. Table 3.2, shows possible parameters to consider in order to perform a handover [22].

Handover decision parameters
Signal strength
Network Conditions
Cost
Application types
Services Provided
User Preferences

Table 3.2: Possible parameters to consider in a handover decision

- **Handover Implementation.** The implementation of handover considers the packets' transference of the ongoing session to the new wireless link, this requires the network to transfer routing information about the new target router to establish a new session. Owing to differences between access technologies, transfer of additional contextual information might be required. This contextual information could include Quality of Service, authentication and authorization, among others. The aim of contextual transference is to minimize the impact of different access technologies and their polices to transfer different types of data on applications and services

3.5 Quality of Service in IP Backbone Networks

The standardization carried out by 3GPP towards an all-IP core network, requires that packets passing through this network receive a guaranteed Quality of Service (QoS). QoS is a concept that has to be satisfied through the interworking of all the entities the data is passing through. The main approaches for QoS provisioning in IP backbone networks are Differentiated Services and Integrated Services. This Section is oriented to give a quick overview of IP QoS for a proposal describing how SIP and therefore IMS can support the QoS provisioning in IP-based networks, using information provided about the type of session.

3.5.1 Differentiated Services

Differentiated Services or DiffServ is an architecture of QoS that specifies a mechanism to differentiate IP traffic. This allows to determine the relative traffic priority on a per-hop basis rather than per-flow state and signaling at every hop [23].

Since internet is a network of multiple service provider networks they do not provide the same service in the same way. The contribution of DiffServ allows traffic with similar service characteristics to be passed with similar traffic guarantees across multiple networks.

DiffServ operates on the principle of traffic classification, where each data packet is classified into a limited number of traffic classes or services. A "service" defines some significant characteristics of packet transmission in one direction across a set of one or more paths within a network [23]. Services can be constructed by a combination of:

- Setting bits in an IP header field at network boundaries.
- Using those bits to determine how packets are forwarded by the nodes inside the network, and
- Conditioning the marked packets at network boundaries in accordance with the requirements or rules of each service.

One advantage of DiffServ, is that all the policing and classifying is done at the boundaries between DiffServ networks. This means that in the core of the Internet, routers do not care about the complexities of collecting payment or enforcing agreements.

3.5.2 Integrated Services

Integrated Services or IntServ is a flow oriented QoS approach offering three classes of services:

- Guaranteed Service.
- Controlled-load service
- Best-effort Service.

In this model routers must be able to reserve resources in order to provide the QoS for specific flows, being a flow characterized by a source and destination IP address. This method is not further explained since scalability is complex when the number of flows increases and the architecture does not fit well for short data flows like common web traffic because the overhead is greater than the data.

Chapter 4

Protocols for Handover and Proposed Solutions

This chapter is dedicated to present the proposed solutions for vertical handover. The chapter is divided into four sections. The first two sections are given as an introduction to understand the overall problem. The last two sections contain the outcome of the research done in this thesis. The first section, gives a description of the interworking architecture, the components and necessary assumptions. The second section, describes the signaling for connectivity developed in [7]. The third section, shows the proposed handover procedure and handover cases. The fourth section shows the proposed SIP client capabilities needed to perform handover.

4.1 Interworking Architecture

The proposed architecture for the convergence of 3GPP networks and WLAN-based corporate networks is shown in Figure 4.1, and is described according to [7]. This interworking architecture enables cellular system operators to provide connectivity and extended coverage to corporate networks. The following components are present in the architecture:

1. **Corporate Domain.** The corporate domain consists of multiple network elements in order to provide connectivity, data, voice and multimedia services in a private network. The key component in the corporate domain architecture is the IP-PBX, referred from now as PBX. This entity is capable to perform functions like user registration, authentication and

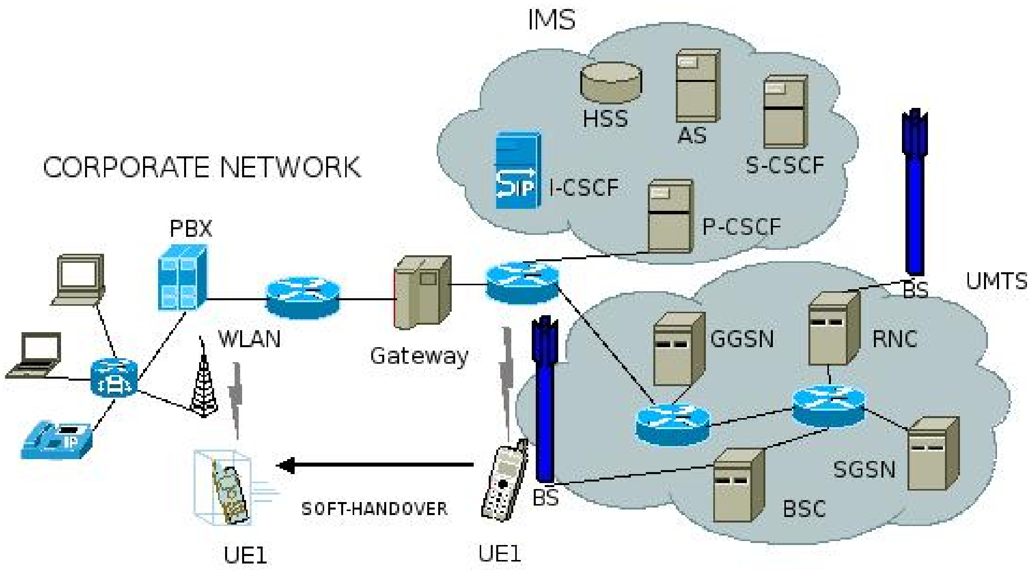

Figure 4.1: Interworking Architecture

session establishment inside the corporate domain. The PBX signaling is based on the SIP protocol. To be able to access to services in the corporate domain, the user equipment is capable to perform session establishment based on the SIP protocol. As described in previous chapters, to support mobility in the corporate domain, wireless access is provided by WLAN technologies. This access is assumed to be based on IEEE 802.11x standards.

2. **Public Domain.** The public domain consist of multiple network elements owned by a cellular operator in order to provide voice and data services to mobile users. The architecture of the public domain described in this thesis is based on UMTS (described in Section 3.1). In order to support the cellular network for service provision, the IMS is introduced as an additional architecture which is part of the public domain. The IMS performs user authentication, registration and session establishment in the public domain for the access to services. The IMS specifies the functional entities such as the User Equipment (UE), Call and Session Control Functions, Application Servers, and others, as described in Section 3.3. The terms user equipment (UE) and Mobile Node (MN) are used in the following lines to refer to the clients.

3. **Gateway.** The main component in the interworking scenario is the Gateway. This element acts as a "SIP translator" and exchanges SIP messages between the corporate and the public

domains. The Gateway interconnects these two domains through a link to the PBX in the corporate domain and a link to the IMS in the public domain. The Gateway is proposed to be part of the corporate domain and the implementation can be in the same hardware as the PBX or as an independent entity. In this thesis it is presented as an independent component in the corporate network.

General Assumptions

Since the main interest is to provide mobility it is assumed that the user access to any of the networks always using wireless devices. According to the scenario proposed (similar to Section 3.4.1), it is assumed that a device located physically in the corporate network coverage, is always in the range covered by both, the public domain and the corporate domain. This feature is owing to the overlapping nature of the cells for a "Wireless Overlay Network". In other words, the MN is always under the coverage of the cellular system operator. This case is not the same for the corporate network coverage, since it is based on a short range WLAN access. A user can be connected to the public and/or corporate domain using an equipment with multiple network interfaces and this equipment is capable to send data over multiple interfaces at the same time. The wireless devices in the interworking scenario have one interface to access to the wireless corporate network and other to access to the public network.

4.2 Signaling for Connectivity

This section describes a set of protocols that allow users to register in the corporate and public domains and establish connections among them in any of the domains, with a unique subscriber ID. This brief review is part of previous proposed solutions for registration and session establishment as described in [7].

4.2.1 Registration

For registration, three cases are described: group registration, individual registration at corporate domain and individual registration at public domain.

Corporate Domain Registration Process

An initial registration is required to inform the public domain that the corporate domain is available. This initial registration is done by the Gateway, using a unique address with the form: *pbx.corporate@ims.com*. The initial registration notifies the IMS component in the public domain that the corporate domain is available through the Gateway and that all the calls to addresses with the form **.corporate@ims.com*, have to be routed to the Gateway unless specific polices force IMS to proceed in a different manner. Here, the variable "*", is the user identifier in the corporate domain. The protocol is shown in Figure 4.2. It is important to note here that the unique identification to locate users in both domains has the form: **.corporate@ims.com*, where "*", represents the user identifier.

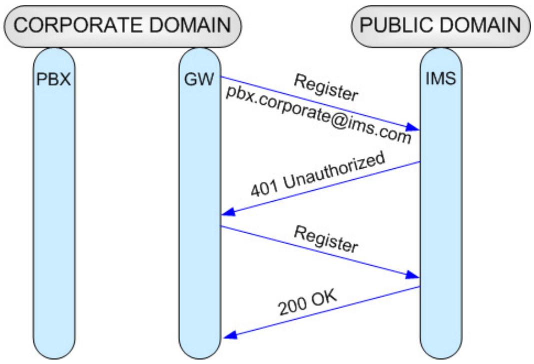

Figure 4.2: PBX Initial Registration

User Registration in the Corporate Domain

From the user perspective, in the corporate domain, the registration towards the PBX is a standard procedure. However, the information about the presence of a user in the corporate domain is made available for the public network by forwarding the registration from the PBX to the Gateway. This introduces a small change in the standard PBX protocols. When an INVITE is sent in the public domain (towards IMS) to a user with the form **.corporate@ims.com*, and the user is not located there, the message is forwarded to the Gateway, this entity performs a "name translation" to the form: **@corporate.com* and forwards the message to the PBX. The protocol is shown in Figure 4.3. It is important to note here that a user registration in the corporate domain is done with a user name with the following template: **@corporate.com*, where "*", represents the user identifier.

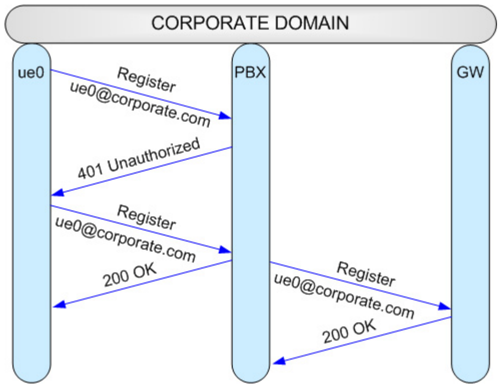

Figure 4.3: Registration at Corporate Domain

User Registration in the Public Domain

The protocol is shown in Figure 4.4. As in the corporate domain registration, from the user perspective, the registration towards the IMS is a standard procedure. However, the information about the presence in the public domain must be available for users in the corporate domain. This is achieved by forwarding the registration from the IMS to the Gateway.

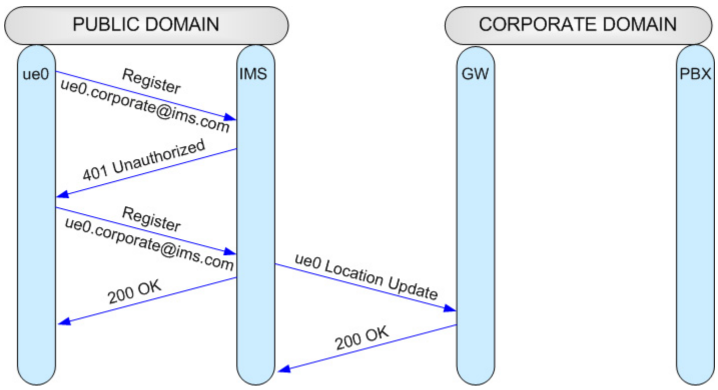

Figure 4.4: Registration at Public domain

This introduces a small change in the SIP protocol inside IMS. When an INVITE is sent in the corporate domain (towards PBX) to a user with the form **@corporate.com*, and the user is not located there, the message is forwarded to the Gateway, this entity performs a "name translation" and looks for a user with the identifier **.corporate@ims.com* in the public domain. A user registration in the public domain is done with a user name with the following template: **.corporate@ims.com*, where "*", represents the user identifier.

4.2.2 Session Establishment

For session establishment, four cases are described: Session in the corporate domain, session in the public domain, session from corporate to public domain and session from public to corporate domain.

Session Establishment in the Corporate Domain

Before session establishment starts, both users are registered at the corporate domain. One of the MNs sends an INVITE to the other through the PBX, with an unique ID of the form: **.corporate@ims.com*, the PBX does not have this type of user registered and forwards the message to the Gateway. The Gateway performs and address translation to the form: **@corporate.com*, and sends back the message to the PBX, after that, the INVITE is sent by the PBX to the destination. The destination MN replies with a 200OK and the session is established. Figure 4.5, shows the protocol.

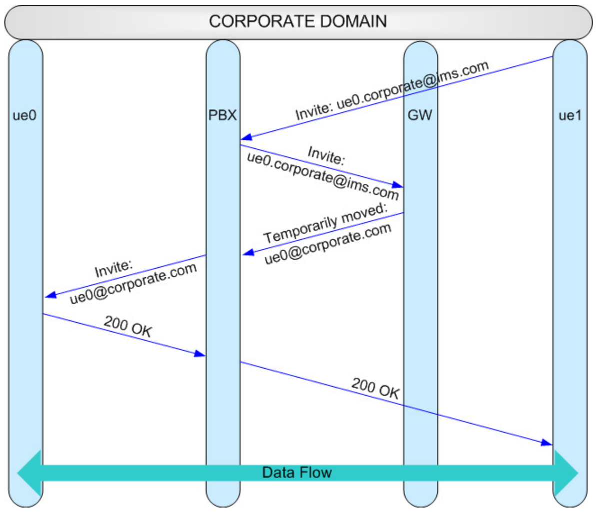

Figure 4.5: Session Establishment in the Corporate Domain

Session Establishment from Public to Corporate Domain

Before session establishment starts, the caller user is registered in the public domain and the called user in the corporate domain. The MN in the public domain sends an INVITE to the other MN through the IMS, with an unique ID of the form: **.corporate@ims.com*, the IMS does not have this type of user registered and forwards the message to the Gateway. The Gateway performs and address translation to the form: *@*corporate.com*, and forwards the message to the PBX, after that the INVITE is sent by the PBX to the destination. The destination MN replies with a 200OK and the session is established. Figure 4.6, shows the protocol.

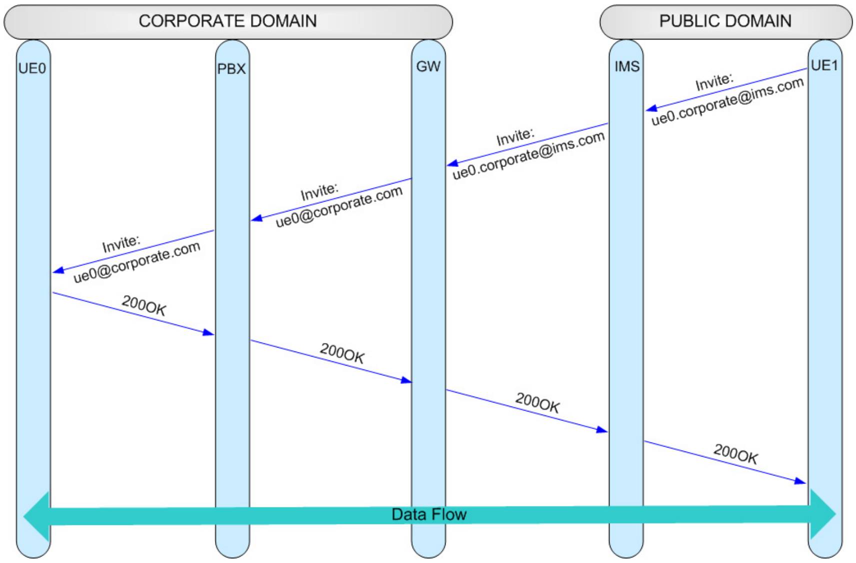

Figure 4.6: Session Establishment from Public to Corporate Domain

Session Establishment in the Public Domain

Before session establishment starts, both users are registered at the public domain. One of the MNs sends an INVITE to the other through the IMS, with an unique ID of the form: **.corporate@ims.com*, the IMS knows that both users are registered there and delivers the INVITE to the destination. The destination MN replies with a 200OK and the session is established. Figure 4.7, shows the protocol.

Session Establishment from Corporate to Public Domain

Before session establishment starts, the caller user is registered in the corporate domain and the called user in the public domain. The MN in the corporate domain sends an INVITE to the other MN through the PBX, with an unique ID of the form: **.corporate@ims.com*, the PBX does not have this type of user registered and forwards the message to the Gateway. The Gateway recognizes that the user is registered in the public domain and forwards the message to the IMS,

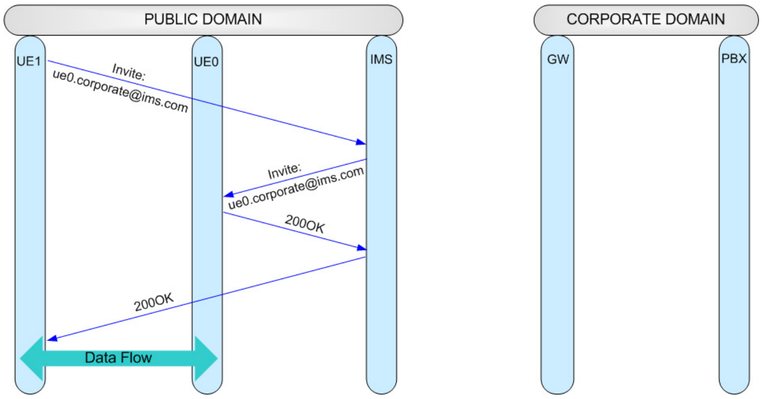

Figure 4.7: Session Establishment in the Public Domain

after that, the INVITE is sent by the IMS to the destination. The destination MN replies with a 200OK and the session is established. Figure 4.8, shows the protocol.

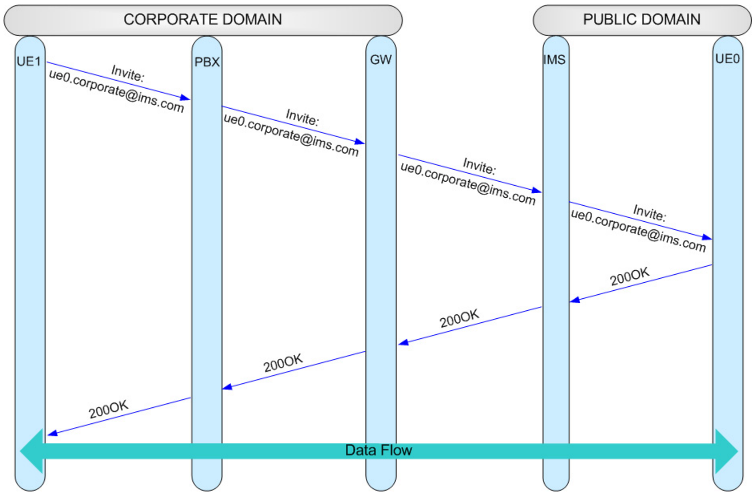

Figure 4.8: Session Establishment from Corporate to Public Domain

4.3 Proposal for Handover Procedure

As described in Section 3.4.2, a vertical handover can be divided into three steps: Network discovery, handover decision and handover implementation. Figure 4.9, shows the procedure for the SIP-supported handover proposed.

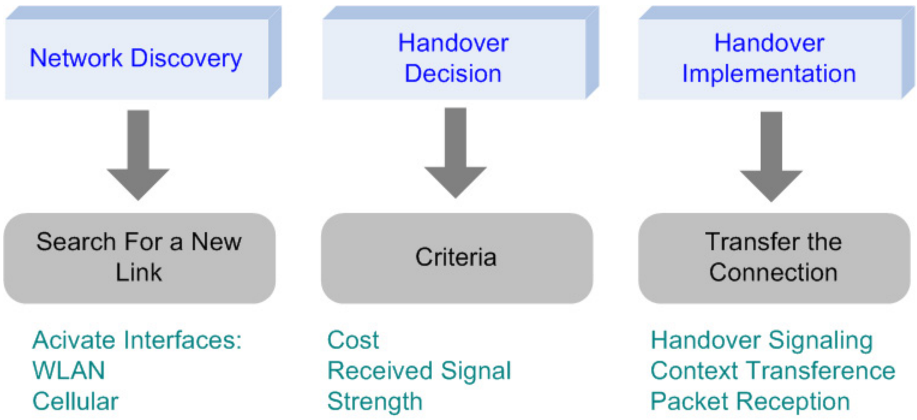

Figure 4.9: Handover Procedure

4.3.1 Network Discovery

In this step, the MN must activate its multiple interfaces to receive the advertisements broadcasted by the cellular and corporate networks. To discover a network, its service advertisements must be heard by the MN. A simple solution is to keep all the interfaces in a "listening" mode. However, keeping the interfaces active all the time results in a non efficient power management. A simplification is done with the assumption that the MN is always in the coverage of the public domain, then, the main problem is how to detect the presence of the corporate network and how to trigger the activation of the corporate network interface. For this problem several solutions have been proposed to advertise to corporate users the presence of the corporate network [21], [26]. This step is a challenge for researches due to power consumption efficiency. Having WLAN interfaces always listening to services advertising degrades battery life. The proposed solution in this thesis, is to keep off the WLAN interface and make use of localization in cellular networks to advertise the existence of the corporate domain and activate the WLAN interface. After this step it is assumed that the MN has the IP address and possible information about the resources in the new network destination.

4.3.2 Handover Decision

Once the networks have been discovered, the ability to decide when to perform the handover must be provided. The decision will depend on the actual network and the targeted one. As soon as the corporate network has been discovered, the MN is under the coverage of two different access technologies at the same time. In cellular networks, the decision is performed based on the channel conditions and thus, the Received Signal Strength (RSS) and other parameters like traffic and user velocity, which are constantly monitored by both, the network and the MN. The number of users allowed, cost, QoS, security and the level of mobility differentiates the corporate and public networks. None of the technologies simultaneously provides all the advantages.

The proposed solution for this thesis, is to use a combination of these wireless networks to provide the highest bandwidth at the lowest cost, considering a wide coverage. This criteria suggests that the MN stays under the corporate domain as long as possible. The decision for handover is then influenced by the preferred network (Corporate) and the Received Signal Strength (RSS), which varies according to channel conditions.

An interesting proposal for the handover decision considering RSS and the distance between MN and Access Point (AP), has been defined in [25]. This concept could be used for implementation in this scenario.

To summarize this section, the trigger parameter for vertical handover, is selected as the RSS, considering that the preferred network for the user, is the corporate, due to costs reasons mainly.

4.3.3 Handover Implementation

The main contribution in this thesis is based on handover implementation, the contribution concentrates in the SIP signaling in order to support handover. As described in section 4.2, four possible cases for session establishment are present, therefore, four different scenarios for handover need to be addressed. The analysis described here, assumes that there is an ongoing session between two MNs and only one MN intends to change domain and performs the handover at the same time. Before starting with the description of the proposal, the messages in the flow are explained in the following lines.

Messages description

- **INVITE.** This message initiates a new session. The main parameters are TO, FROM, call-ID, media type and session parameters which indicates physical specifications like ports and protocols. This is a standard SIP message.

- **Bye.** This message terminates the session. This is a standard SIP message.

- **OK.** In this context, this message is sent to confirm the reception of "Location Update" or "Session Transference Request" messages. This is a standard SIP message.

- **Location Update.** This is a message to notify the PBX, Gateway or IMS that the location of the user has been changed. This can be implemented with a standard REGISTER message in SIP. However two main differences are present. First the "Location Update" messages are not sent by the MNs, moreover, the MNs are not aware of these signaling. Second, some modifications are required in the standard behavior of the PBX, Gateway and IMS to implement these signaling. the modifications could include to check which entity is forwarding the packet and act accordingly, flags or persistent data storage can be used for this purpose .

- **Session Transference Request.** This message is proposed to be a notification, that the sender of the message will change its IP address. This message is sent by the MN doing handover, through the new network in which it is attached, with the same Call-ID than in the ongoing session. This message carries the new IP of the MN doing the handover. The MN receiving the message will store this new IP. Since in the ongoing session there was already a session description and the message comes with the same Call-ID, no session parameters need to be re-negotiated, instead, the parameters will be reused later on. It is assumed that the MN doing handover will use the same parameters as well. The destination IP is not changed yet in this step. This message can be implemented using a standard INVITE in SIP, with the keyword in the body: "Session Transference". In the MNs, this introduces the requirement of capabilities to insert body messages in SIP standard messages and also to recover them when required. In the PBX, Gateway and IMS this modification in the message requires the ability to "decode" body messages in standard INVITES, with a non SIP standard behavior.

- **Session Transference Confirmation.** This message is proposed to be a confirmation to perform the change of destination IP. Once this message arrives, the destination IP is changed and the data flow is transferred to the new destination IP. This can be implemented with a ACK standard SIP message.

- **Release session.** This message is intended to either the PBX or the IMS to indicate that the session has been transferred for control processes that could be implemented. It can be implemented with a BYE standard SIP message containing in the body the call-ID of the old session, which is unique. Again, it requires capabilities in the MN to insert text in the body of messages and in the Gateway, PBX and IMS to be able "decode" the messages and act accordingly.

In the following lines the different proposals for the four handover cases are shown.

- **CASE 1:** A session is ongoing in the corporate domain and one of the MNs moves to the public domain. This case is depicted in Figure 4.10.

  Registration description

  Before the flow messages for registration start, the MN doing handover is registered in the corporate domain. Messages 1-4 show a normal registration process in the IMS domain, with standard SIP signaling. Message 5, is to indicate the Gateway that the user must be located for future sessions in the public domain. Message 7 is triggered by message 5, to indicate the PBX that the user has left the corporate domain. Finally, messages 6 and 8 are to confirm that updates have been made for the location of the user doing handover. At this point the "Session A" is still ongoing.

  New session establishment description

  Before the flow messages start, the MN doing the handover has a new IP in the public domain using its cellular network interface. The "Session Transference Request" message is as described in Section 4.3.3. Message 1, is sent to the IMS, to the other MN involved in the ongoing session, with an ID of the form: **.corporate@ims.com*. The IMS checks for the location of this user and does not find it. Then, with the message 2, it forwards the request to the Gateway. From previous registration process, the Gateway knows that the user is in the corporate domain and performs and address translation to the form *@*corporate.com* and forwards the message to the PBX (message 3). The PBX knows that the destination user is registered in the corporate domain and delivers the message (message 4). At this point the MN receiving the message goes into the HANDOVER temporal state described in Section 4.4. This event triggers message 5, which is only an acknowledgement that the "Session Transference Request" message has been received. This message is sent to the MN with the following template: **.corporate@ims.com*. The PBX receives the message and does not recognize the user, then it forwards the message to the Gateway (message 6). The Gateway knows that the user is located in the public domain and forwards the message to the IMS (message 7). Finally, the message is delivered to the user doing the handover (message 8). After receiving message 8, the MN doing the handover knows that the other party has its new IP and also has the knowledge that the session is going to be transferred. Then, message 9, is sent to confirm the session transference. Messages 10, 11 and 12 follow the same route as before with and address translation in the Gateway. After message 12 arrives, the MN receiving this message performs the session transference, sends the "Release Session" message and returns to the BUSY state, described in Section 4.4. Messages 13, 14, 15 and 16 are to indicate the PBX that the session is not in the corporate domain now, but between corporate and public.

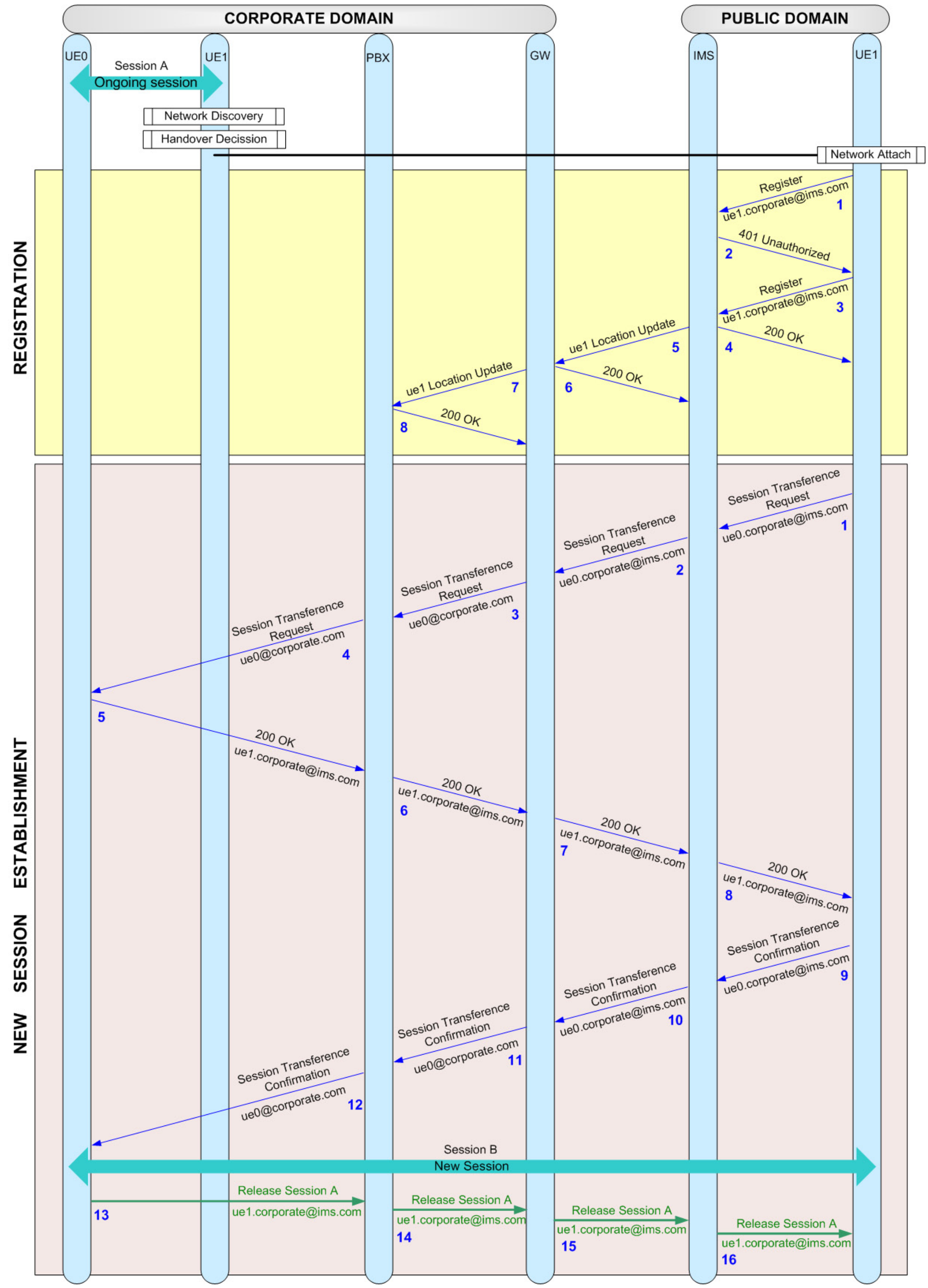

Figure 4.10: Handover to public domain: CASE 1

- **CASE 2.** A session is ongoing between the corporate and public domains and one of the MNs moves to the corporate domain. This case is depicted in Figure 4.11.

  Registration description

  Before the flow messages for registration start, the MN doing handover is registered in the public domain. Messages 1-4 show a normal registration process in the corporate domain, with standard SIP signaling. Message 5, is to indicate the Gateway that the user must be located for future sessions in the corporate domain. Message 7 is triggered by message 5, to indicate the IMS that the user has left the public domain. Finally, messages 6 and 8 are to confirm that updates have been made for the location of the user doing handover. At this point "Session A" is still ongoing.

  New session establishment description

  Before the flow messages start, the MN doing the handover has a new IP in the corporate domain using its WLAN network interface. The "Session Transference Request" message is as described in Section 4.3.3. Message 1, is sent to the PBX, to the other MN involved in the ongoing session, with an ID of the form: **.corporate@ims.com*. The PBX checks for the location of this user and does not find it. Then, with the message 2, it forwards the request to the Gateway. From previous registration process, the Gateway knows that the user is in the corporate domain and performs and address translation to the form **@corporate.com* and forwards the message to the PBX (message 3). The PBX knows that the destination user is registered in the corporate domain and delivers the message (message 4). At this point the MN receiving the message goes into the HANDOVER temporal state, described in Section 4.4. This event triggers message 5, which is only an acknowledgement that the "Session Transference Request" message has been received. This message is sent to the MN with the following template: **.corporate@ims.com*. The PBX receives the message and does not recognize the user, then it forwards the message to the Gateway (message 6). The Gateway knows that the user is located in the corporate domain, performs the address translation, and forwards the message to the PBX (message 7).

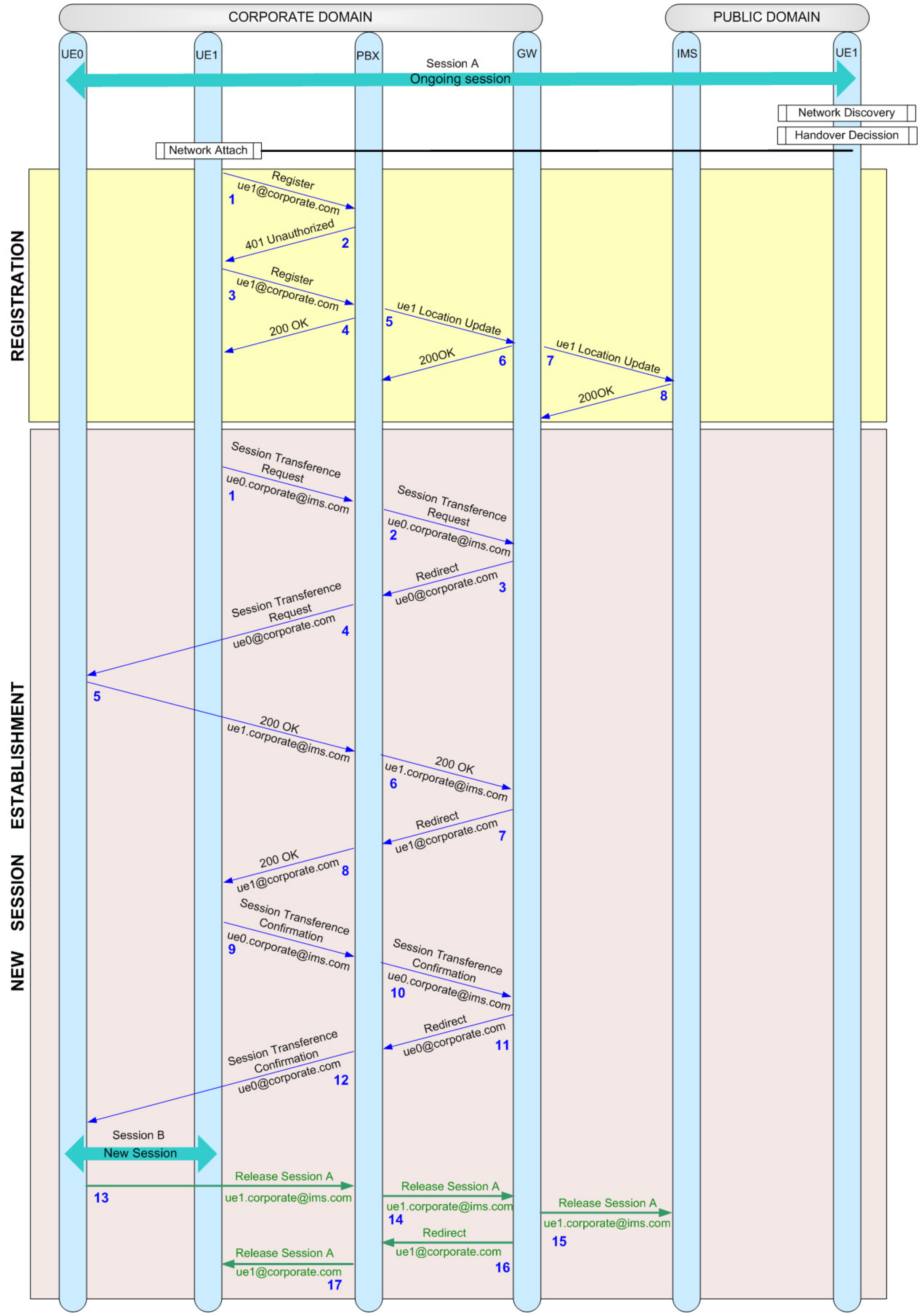

Figure 4.11: Handover to corporate domain: CASE 2

Finally, the message is delivered to the user doing the handover (message 8). After receiving message 8, the MN doing the handover knows that the other party has its new IP and also has the knowledge that the session is going to be transferred. Then, message 9 is sent to confirm the session transference. Messages 10, 11 and 12 follow the same route as before with and address translation in the Gateway. After message 12 arrives, the MN receiving this message performs the session transference, sends the "Release Session" message and returns to the BUSY state described in Section 4.4. Messages 13, 14, 15, 16 and 17 are to indicate the IMS that the session has been ended in the IMS domain.

- **CASE 3:** A session is ongoing between the corporate and public domains and one of the MNs moves to the public domain. This case is depicted in Figure 4.12.

Registration description

The registration process is the same as described in CASE 1.

New session establishment description

Before the flow messages start, the MN doing the handover has a new IP in the public domain using its cellular network interface. The "Session Transference Request" message is as described in Section 4.3.3. Message 1, is sent to the IMS, to the other MN involved in the ongoing session, with an ID of the form: **.corporate@ims.com*. The IMS checks for the location of this user and delivers the message (message 2). At this point the MN receiving the message goes into the HANDOVER temporal state described in Section 4.4. This event triggers message 3, which is only an acknowledgement that the "Session Transference Request" message has been received. This message is sent to the MN with the following template: **.corporate@ims.com*. The IMS simply delivers the message to the MN doing the handover (message 4). After receiving the message 4, the MN doing the handover knows that the other party has its new IP and also has the knowledge that the session is going to be transferred. Then, messages 5 and 6 are sent to confirm the session transference. After message 6 arrives, the MN receiving this message performs the session transference, sends the "Release Session" message and returns to the BUSY state, described in Section 4.4. Messages 7 and 8 are to indicate the IMS that the session is inside the public domain now.

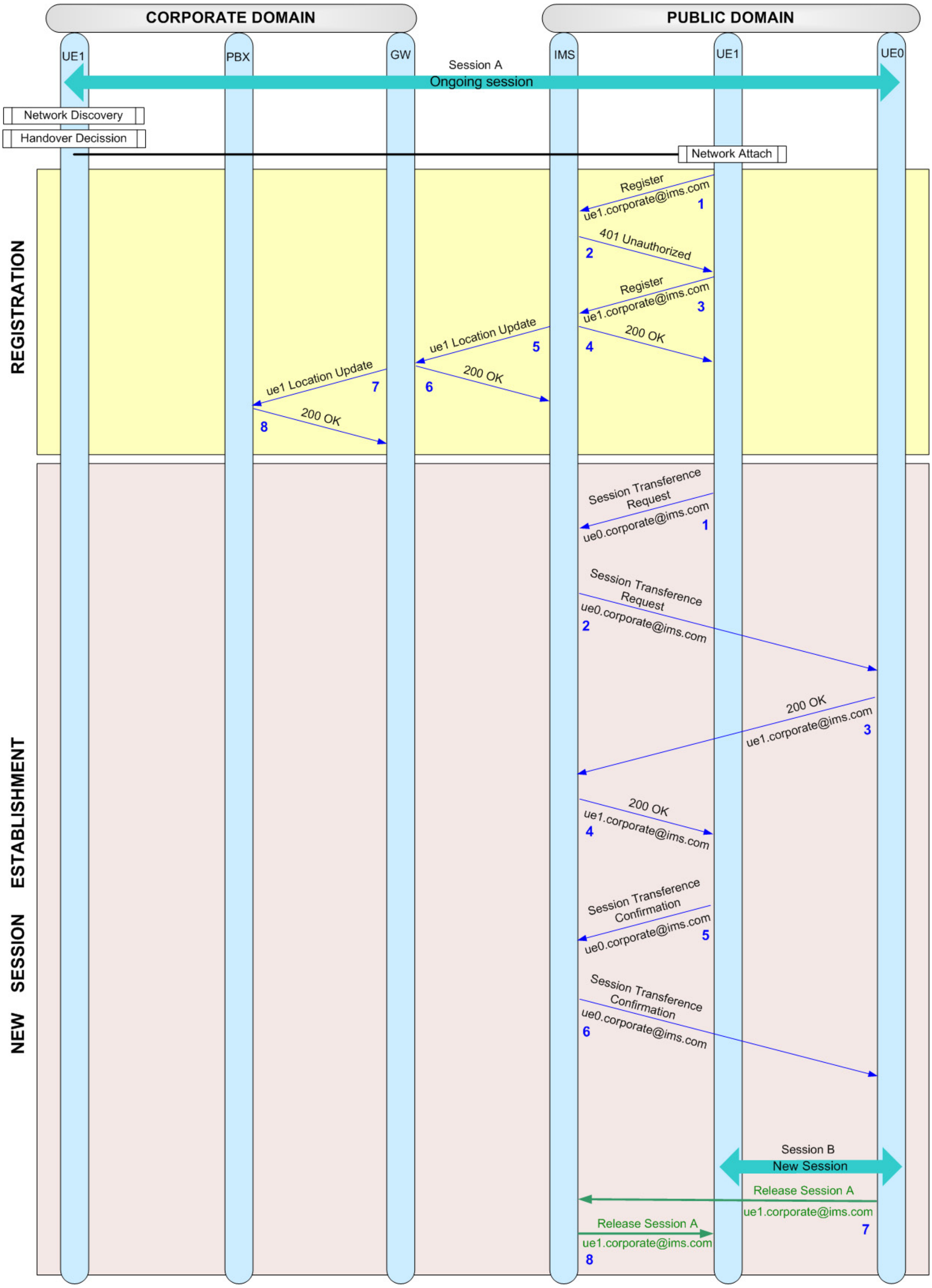

Figure 4.12: Handover to public domain: CASE 3

- **CASE 4.** A session is ongoing in the public domain and one of the MNs moves to the corporate domain. This case is depicted in Figure 4.11.

Registration description

The registration process is the same as described in CASE 2.

New session establishment description

Before the flow messages start, the MN doing the handover has a new IP in the corporate domain using its WLAN network interface. The "Session Transference Request" message is as described in Section 4.3.3. Message 1, is sent to the PBX, to the other MN involved in the ongoing session, with an ID of the form: **.corporate@ims.com*. The PBX checks for the location of this user and does not find it. Then, with the message 2, it forwards the request to the Gateway. From previous registration process, the Gateway knows that the user is in the public domain and forwards the message to the IMS (message 3). The IMS knows that the destination user is registered in the public domain and delivers the message (message 4). At this point the MN receiving the message goes into the HANDOVER temporal state described in Section 4.4. This event triggers message 5, which is only an acknowledgement that the "Session Transference Request" message has been received. Messages 6,7 and 8 are to confirm the reception of the "Session Transference Request" message and there is an address translation in the Gateway. Messages 9, 10, 11, 12 and 13 are to perform the session transference as in previous cases. After message 13 arrives, the MN receiving this message performs the session transference, sends the "Release Session" message and returns to the BUSY state, described in Section 4.4. Messages 14, 15, 16 and 17 are to indicate the IMS that the session is now between corporate and public domains.

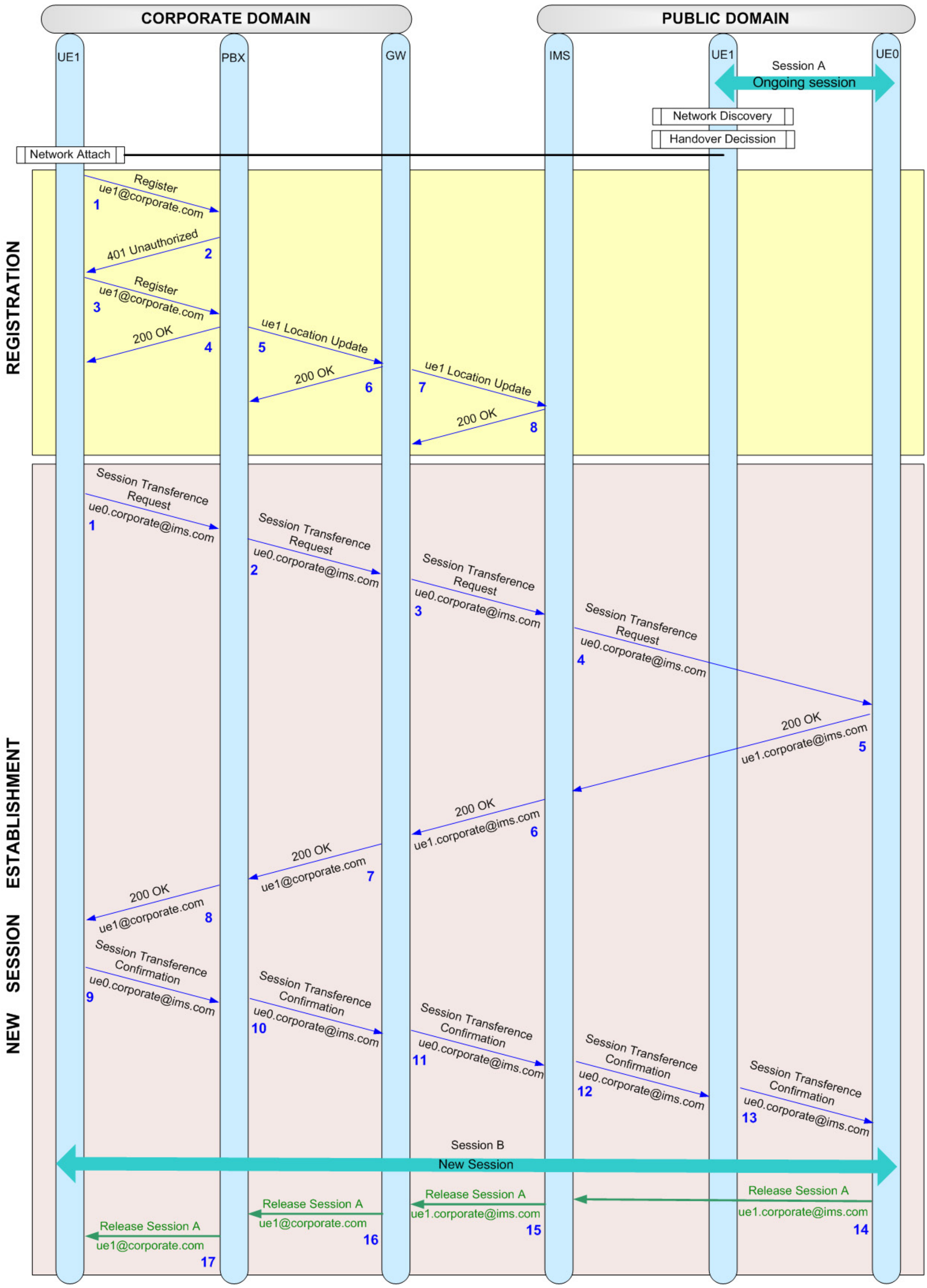

Figure 4.13: Handover to corporate domain: CASE 4

IP QoS

SIP can be used to assure QoS in the IP domain. According to [24], the body message of SIP could transport a Session Description Protocol (SDP) indicating the type of media required in the session, with this knowledge differentiated services QoS can be applied to guarantee the resources in the IP domain. However, in order to provide end-to-end QoS, the WLAN access must be able to provide QoS and there should be a mapping between WLAN QoS parameters and differentiated services.

4.4 Proposal for SIP client capabilities

After the analysis of the handover cases, one of the main conclusions is that to perform a soft-handover, there must be a cooperation between the clients and the signaling in the rest of the network. In the handover flows described before, there is an interaction between the MNs to perform the handover. Standard IETF clients are not designed for mobility and therefore the capabilities for the interaction are not present. This requires some modifications in the SIP clients to introduce new features for an efficient handover.

A general solution is proposed in the form of states machine. Figure 4.14, shows a simplified states machine for a SIP client. On the left, the states machine for a static scenario is depicted, where no handover is allowed. On the right, handover is considered and one state more is added, the description of the states is given in the following lines:

- **LISTENING.** In this state the client is waiting for an "INVITE" message to initiate a session. It is only possible to move to the BUSY state.

- **BUSY.** Three events can happen in this state: Receiving a "Bye" changes to the initial LISTENING status. Receiving an "INVITE" does not change the state of the system. Finally, receiving a "Session Transference Request" changes the state to HANDOVER, which is a transition state.

- **HANDOVER.** Three events can happen in this state: Receiving a "Bye" changes to the initial LISTENING status, receiving an "INVITE" does not change the state of the system. Finally, receiving a "Session Transference Confirmation" message changes the client to the BUSY state again.

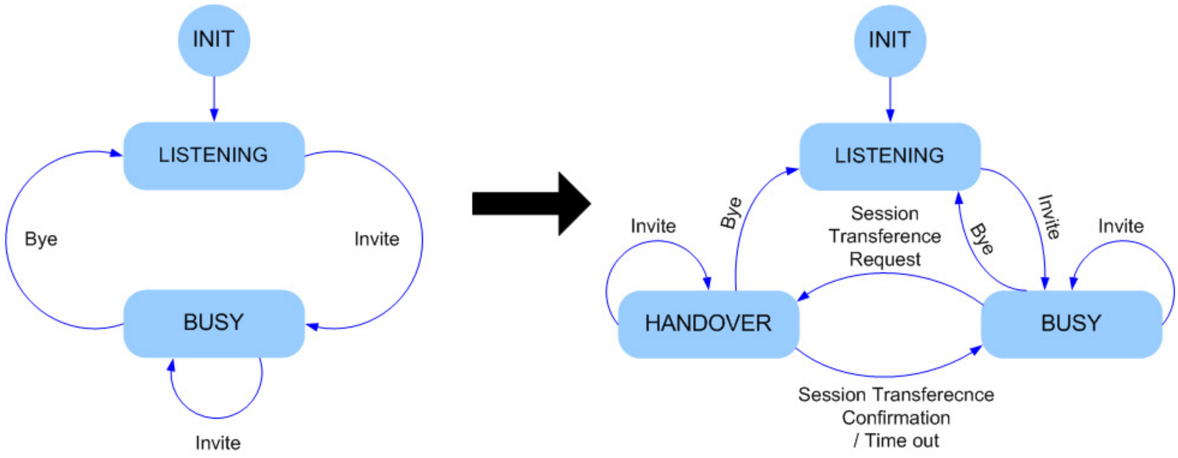

Figure 4.14: States machine for SIP clients

The states LISTENING and BUSY are a normal feature when handover is not required. In order to implement the HANDOVER state, a modification needs to be implemented in the SIP part of the protocol stack of the clients, as depicted in Figure 4.15.

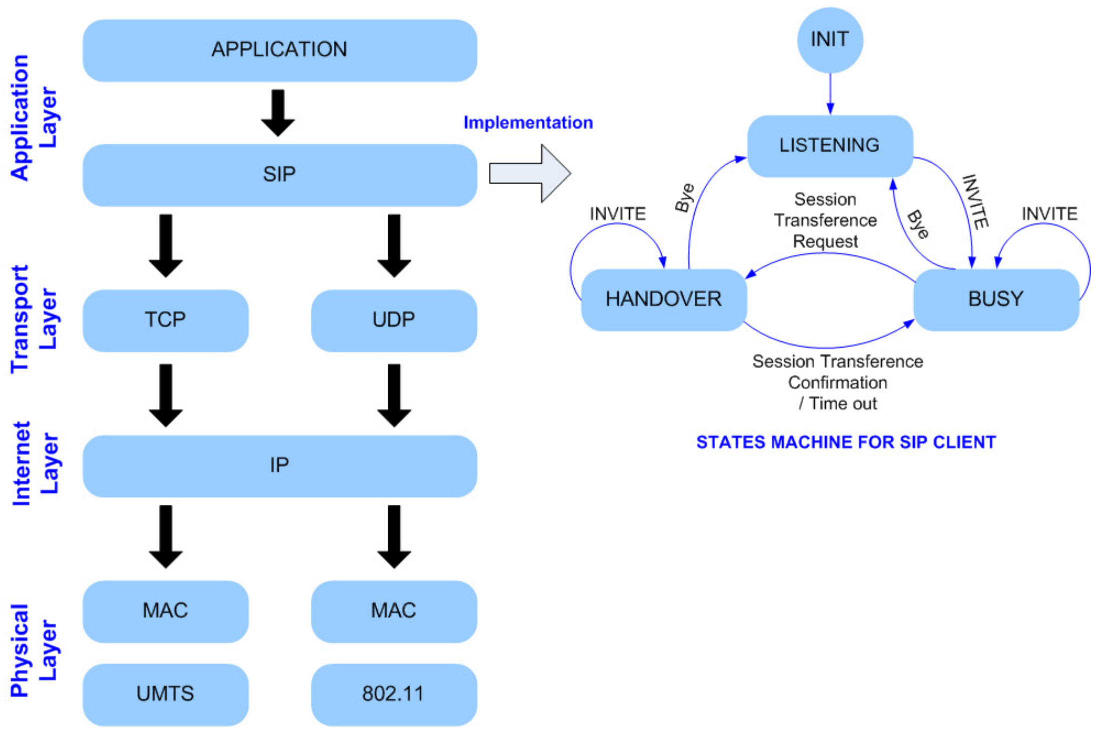

Figure 4.15: States machine and protocol stack

Chapter 5

Concept Implementation and Results

This chapter describes the test bed for the implementation of the signaling proposal for handover, described in Chapter 4. Figure 5.1, shows the implementation scenario, the main components in the test bed are the corporate IP-PBX, the Gateway and the IMS domains. These three entities have different IP sub-domains between and inside them. The Gateway is the key component for the interconnection of the corporate domain (corporate.com) and the public domain (ims.com). Special interest is placed on the user called "ue1", defined in this showcase as the one performing the handover between the domains.

5.1 Nodes description

The following sub-sections present the functionalities implemented in the most relevant components of the test bed.

5.1.1 Gateway

This component stores information related with the location of the MNs. Three possibilities are available: a user can be registered in the corporate domain or in the public domain or not registered. Another functionality of the Gateway is to perform addresses translation between the corporate address (ue1@corporate.com) and the public address (ue1.corporate@ims.com) and viceversa, when required. During handover the Gateway interacts with the PBX and the IMS for registration updates. The Gateway functions are transparent for the users, since they are not aware of the

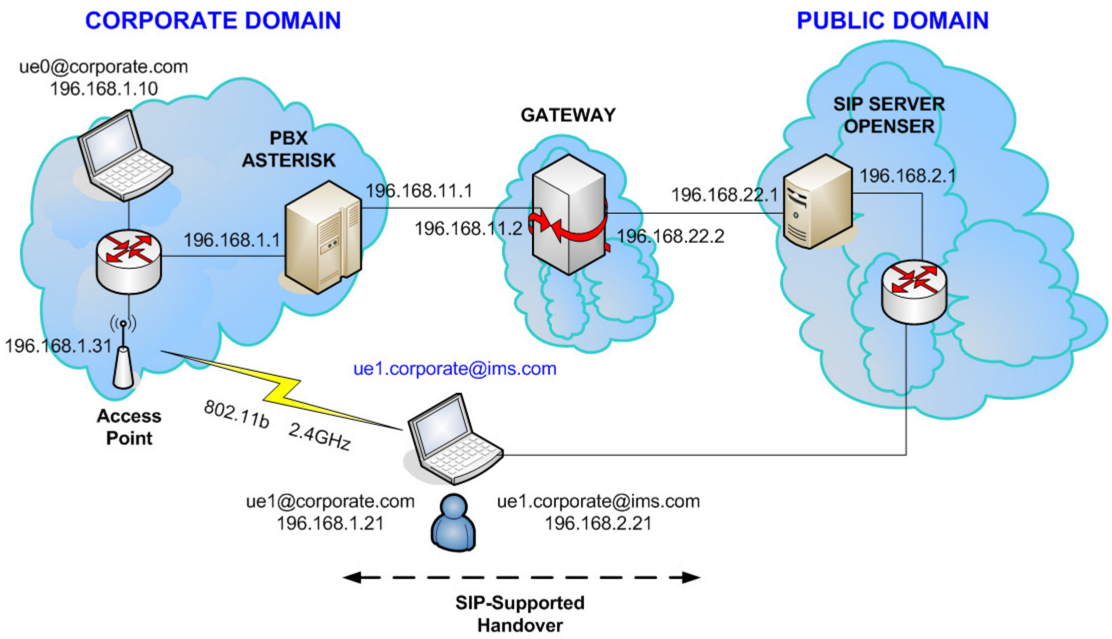

Figure 5.1: Implementation Scenario

existence of this entity. The Gateway is implemented according to the specifications shown in Table 5.1.

Operative System	Linux Ubuntu 6.10
Software Version	openser SIP-server
Network Interfaces	Two LAN interfaces
IP corporate side	196.168.11.2
IP public side	196.168.22.2

Table 5.1: Gateway specifications

5.1.2 IP-PBX

This entity registers users in the corporate domain and performs all the signaling required to initiate modify and end sessions. When a user registers with the PBX, the PBX will forward this message to the Gateway as a notification for this registration. If the message is received successfully by the Gateway it will be acknowledged by sending back a "200 OK" message. INVITE messages are delivered to users through the PBX in the corporate domain. The PBX specifications are shown in Table 5.2.

Operative System	Linux SuSe 10.1
Software Version	Asterisk
Network Interfaces	Two LAN interfaces
IP corporate side	196.168.1.1
IP gateway side	196.168.11.1

Table 5.2: IP-PBX specifications

5.1.3 IMS

This entity is the contact point for the users in the public domain. Users in this domain register with the IMS with an address of the form **.corporate@ims.com*. IMS is extended with additional functionalities in order to support user mobility and to deal with sessions to/from the corporate domain and support the handover. Signaling exchange is done with the Gateway to implement full mobility and handover. The IMS is implemented according to the specifications shown in Table 5.3.

Operative System	Linux Ubuntu 6.10
Software Version	openser SIP-server
Network Interfaces	Two LAN interfaces
IP gateway side	196.168.22.1
IP public side	196.168.2.1

Table 5.3: IMS specifications

5.1.4 User Equipment

Besides the signaling entities, software clients are needed to perform the tests. These clients are based on SIP. The software installed for the SIP clients is described in Table 5.4.

	UE1	UE0
Operative System	Linux Ubuntu 6.10	Linux Ubuntu 6.10
Software Version	KphoneSI	KphoneSI
Network Interfaces	WLAN and LAN interfaces	One LAN interface
IP corporate side side	196.168.1.21	196.168.1.10
IP public side	196.168.2.21	–

Table 5.4: User Equipment Specifications

5.1.5 Wireless Access

In order to provide wireless access in the corporate domain, a wireless access point is installed, the specifications are shown in Table 5.5.

Manufacturer	Cisco
Network Interfaces	WLAN and LAN interfaces
IP corporate side	196.168.1.31
Operation Frequency	2.4GHz

Table 5.5: Access Point Specifications

5.2 Experiment description and Results

The objective of this section is to show the signaling proposed in the implemented testbed. In the proposed implementation a session is initiated between a user in the public domain and a user in the corporate domain. The user "ue1" is the mobile user and is located in the public domain(ims.com), connected through a wire. The user "ue0" is a fixed user located in the corporate domain(corporate.com), connected through a wire. After the session is initiated between "ue1" and "ue0", "ue1" moves to the corporate domain and access to this network through a WLAN access point, then "ue1" sends and INVITE through its new IP address to initiate the transference of the previous session to a new session. This scenario corresponds to the handover case 2, depicted in Figure 4.11. The steps for the experiment are divided into five parts, according to Table 5.6. Figure 5.1, shows the scenario.

1	"ue0" registers at corporate domain as *ue0@corporate.com*
2	"ue1" registers at public domain as *ue1.corporate@ims.com*
3	Session establishment between public and corporate domain
4	"ue1" registers at corporate domain as *ue1@corporate.com*
5	Session establishment in the corporate domain

Table 5.6: Experiment Details

To obtain the results showing the signaling flow for the process described in Table 5.6 a network protocol analyzer called Wireshark is installed in the clients, PBX, Gateway and IMS to analyze the SIP messages going through each of these entities. A description of OpenSER [27], as the main programming tool for the implementation of the Gateway and IMS can be found in the Appendix. Together with the report is delivered a CD containing the software for the OpenSER, Kphone and Wireshark installation. The configuration files for Kphone, IMS and the Gateway are also attached in the CD. Finally, the set of complete traces can be found also in the CD. The following figures are shown indicating at the end of the label from were the traces were taken.

ue0 registration at corporate domain

Figure 5.2, shows the registration process in the corporate domain from the "ue0" perspective while Figure 5.3, shows the signaling for registration trough the PBX.

No.	Time	Source	Destination	Protocol	Info
2	0.293696	196.168.1.10	196.168.1.1	SIP	Request: REGISTER sip:corporate.com
3	0.295721	196.168.1.1	196.168.1.10	SIP	Status: 100 Trying (1 bindings)
4	0.295749	196.168.1.1	196.168.1.10	SIP	Status: 401 Unauthorized (1 bindings)
5	2.876737	196.168.1.10	196.168.1.1	SIP	Request: REGISTER sip:corporate.com
6	2.879441	196.168.1.1	196.168.1.10	SIP	Status: 100 Trying (1 bindings)
7	2.881294	196.168.1.1	196.168.1.10	SIP	Status: 200 OK (1 bindings)

Figure 5.2: Registration at corporate domain, ue0

No.	Time	Source	Destination	Protocol	Info
3	15.285771	196.168.1.10	196.168.1.1	SIP	Request: REGISTER sip:corporate.com
4	15.286351	196.168.1.1	196.168.1.10	SIP	Status: 100 Trying (1 bindings)
5	15.286502	196.168.1.1	196.168.1.10	SIP	Status: 401 Unauthorized (1 bindings)
6	17.868963	196.168.1.10	196.168.1.1	SIP	Request: REGISTER sip:corporate.com
7	17.869459	196.168.1.1	196.168.1.10	SIP	Status: 100 Trying (1 bindings)
8	17.871241	196.168.1.1	196.168.1.10	SIP	Status: 200 OK (1 bindings)
9	17.871401	196.168.11.1	196.168.11.2	SIP	Request: REGISTER sip:corporate.com[Malfo
10	17.878374	196.168.11.2	196.168.11.1	SIP	Status: 200 OK (1 bindings)

Figure 5.3: Registration at corporate domain, PBX

ue1 registration at public domain

Figure 5.4, shows the registration process in the public domain from the "ue1" perspective while Figure 5.5, shows the signaling for registration trough the IMS.

No.	Time	Source	Destination	Protocol	Info
1	0.000000	196.168.2.21	196.168.2.1	SIP	Request: REGISTER sip:ims.com
2	0.004474	196.168.2.1	196.168.2.21	SIP	Status: 100 Trying (0 bindings)
3	0.004480	196.168.2.1	196.168.2.21	SIP	Status: 401 Unauthorized (0 bindings)
4	2.402968	196.168.2.21	196.168.2.1	SIP	Request: REGISTER sip:ims.com
5	2.404667	196.168.2.1	196.168.2.21	SIP	Status: 100 Trying (0 bindings)
6	2.416233	196.168.2.1	196.168.2.21	SIP	Status: 200 OK (1 bindings)

Figure 5.4: Registration at public domain, ue1

No.	Time	Source	Destination	Protocol	Info
1	0.000000	196.168.2.21	196.168.2.1	SIP	Request: REGISTER sip:ims.com
2	0.014543	196.168.2.1	196.168.2.21	SIP	Status: 100 Trying (0 bindings)
3	0.014558	196.168.2.1	196.168.2.21	SIP	Status: 401 Unauthorized (0 bindings)
4	2.405987	196.168.2.21	196.168.2.1	SIP	Request: REGISTER sip:ims.com
5	2.406459	196.168.2.1	196.168.2.21	SIP	Status: 100 Trying (0 bindings)
6	2.417929	196.168.2.1	196.168.2.21	SIP	Status: 200 OK (1 bindings)
7	2.418034	196.168.22.1	196.168.22.2	SIP	Request: REGISTER sip:ims.com
8	2.423801	196.168.22.2	196.168.22.1	SIP	Status: 200 OK (1 bindings)

Figure 5.5: Registration at public domain, IMS

Session establishment from public to corporate domain before handover

Figure 5.6, shows the session establishment from the "ue1" perspective while Figures 5.7, 5.8, 5.9 and 5.10 show the signaling seen from the rest of the entities in the testbed.

No.	Time	Source	Destination	Protocol	Info
2	7.705257	196.168.2.21	196.168.2.1	SIP/S	Request: INVITE sip:ue0.corporate@ims.com
5	7.758461	196.168.1.1	196.168.1.10	SIP/S	Request: INVITE sip:ue0@196.168.1.10;tran
6	7.760609	196.168.22.1	196.168.2.21	SIP	Status: 100 Trying
7	7.763070	196.168.1.10	196.168.1.1	SIP	Status: 100 Trying
8	7.763542	196.168.1.10	196.168.1.1	SIP	Status: 180 Ringing
9	7.780223	196.168.22.1	196.168.2.21	SIP	Status: 180 Ringing
10	10.215439	196.168.1.10	196.168.1.1	SIP/S	Status: 200 OK, with session description
11	10.216759	196.168.1.1	196.168.1.10	SIP	Request: ACK sip:ue0@196.168.1.10;transpo
12	10.267151	196.168.22.1	196.168.2.21	SIP/S	Status: 200 OK, with session description

Figure 5.6: Session establishment from public to corporate, ue1

No.	Time	Source	Destination	Protocol	Info
1	0.000000	196.168.2.21	196.168.2.1	SIP/S	Request: INVITE sip:ue0.corporate@ims.com
2	0.015131	196.168.22.1	196.168.22.2	SIP/S	Request: INVITE sip:ue0.corporate@ims.com
4	0.052373	196.168.11.2	196.168.22.1	SIP	Status: 100 Trying
5	0.054164	196.168.22.1	196.168.2.21	SIP	Status: 100 Trying
6	0.070799	196.168.11.2	196.168.22.1	SIP	Status: 180 Ringing
7	0.073760	196.168.22.1	196.168.2.21	SIP	Status: 180 Ringing
8	2.544368	196.168.11.2	196.168.22.1	SIP/S	Status: 200 OK, with session description
9	2.560317	196.168.22.1	196.168.2.21	SIP/S	Status: 200 OK, with session description

Figure 5.7: Session establishment from public to corporate, IMS

No.	Time	Source	Destination	Protocol	Info
1	0.000000	196.168.22.1	196.168.22.2	SIP/S	Request: INVITE sip:ue0.corporate@ims.com
2	0.013131	196.168.11.2	196.168.11.1	SIP/S	Request: INVITE sip:ue0@corporate.com, wi
3	0.017105	196.168.11.1	196.168.11.2	SIP	Status: 100 Trying
4	0.035894	196.168.11.2	196.168.22.1	SIP	Status: 100 Trying
5	0.048530	196.168.11.1	196.168.11.2	SIP	Status: 180 Ringing
6	0.054306	196.168.11.2	196.168.22.1	SIP	Status: 180 Ringing
7	2.497555	196.168.11.1	196.168.11.2	SIP/S	Status: 200 OK, with session description
8	2.527382	196.168.11.2	196.168.22.1	SIP/S	Status: 200 OK, with session description

Figure 5.8: Session establishment from public to corporate, GW

No.	Time	Source	Destination	Protocol	Info
5	37.718604	196.168.11.2	196.168.11.1	SIP/S	Request: INVITE sip:ue0@corporate.com, wi
6	37.719920	196.168.11.1	196.168.11.2	SIP	Status: 100 Trying
17	37.740213	196.168.1.1	196.168.1.10	SIP/S	Request: INVITE sip:ue0@196.168.1.10;tran
18	37.745469	196.168.1.10	196.168.1.1	SIP	Status: 100 Trying
19	37.745944	196.168.1.10	196.168.1.1	SIP	Status: 180 Ringing
20	37.752491	196.168.11.1	196.168.11.2	SIP	Status: 180 Ringing
21	40.197650	196.168.1.10	196.168.1.1	SIP/S	Status: 200 OK, with session description

Figure 5.9: Session establishment from public to corporate, PBX

No.	Time	Source	Destination	Protocol	Info
7	29.307613	196.168.1.1	196.168.1.10	SIP/S	Request: INVITE sip:ue0@196.168.1.10;tran
8	29.310845	196.168.22.1	196.168.2.21	SIP	Status: 100 Trying
9	29.311807	196.168.1.10	196.168.1.1	SIP	Status: 100 Trying
10	29.312285	196.168.1.10	196.168.1.1	SIP	Status: 180 Ringing
11	29.329387	196.168.22.1	196.168.2.21	SIP	Status: 180 Ringing
12	31.763749	196.168.1.10	196.168.1.1	SIP/S	Status: 200 OK, with session description
13	31.766843	196.168.1.1	196.168.1.10	SIP	Request: ACK sip:ue0@196.168.1.10;transpo
14	31.816165	196.168.22.1	196.168.2.21	SIP/S	Status: 200 OK, with session description
15	31.825337	196.168.2.21	196.168.2.1	SIP	Request: ACK sip:ue0@196.168.11.1
16	31.874164	196.168.1.10	196.168.1.1	RTP	Payload type=ITU-T G.711 PCMU, SSRC=16777
17	31.925989	196.168.1.10	196.168.1.1	RTP	Payload type=ITU-T G.711 PCMU, SSRC=16777
18	31.926087	196.168.1.10	196.168.1.1	RTP	Payload type=ITU-T G.711 PCMU, SSRC=16777
19	31.926142	196.168.1.10	196.168.1.1	RTP	Payload type=ITU-T G.711 PCMU, SSRC=16777
20	31.958061	196.168.1.10	196.168.1.1	RTP	Payload type=ITU-T G.711 PCMU, SSRC=16777
21	31.994145	196.168.1.10	196.168.1.1	RTP	Payload type=ITU-T G.711 PCMU, SSRC=16777

Figure 5.10: Session establishment from public to corporate, ue0

Registration at corporate domain during handover

Figure 5.11, shows the registration process in the corporate domain, for the handover, from the "ue1" perspective while Figures 5.12, and 5.13 show the signaling going through the rest of entities.

No.	Time	Source	Destination	Protocol	Info
4	35.581512	196.168.1.21	196.168.1.1	SIP	Request: REGISTER sip:corporate.com
5	35.587106	196.168.1.1	196.168.1.21	SIP	Status: 100 Trying (1 bindings)
6	35.587962	196.168.1.1	196.168.1.21	SIP	Status: 401 Unauthorized (1 bindings)
7	37.320182	196.168.1.21	196.168.1.1	SIP	Request: REGISTER sip:corporate.com
8	37.326818	196.168.1.1	196.168.1.21	SIP	Status: 100 Trying (1 bindings)
9	37.328738	196.168.1.1	196.168.1.21	SIP	Status: 200 OK (1 bindings)

Figure 5.11: Registration at corporate domain during handover, ue1

No.	Time	Source	Destination	Protocol	Info
4	20.590988	196.168.1.21	196.168.1.1	SIP	Request: REGISTER sip:corporate.com
5	20.591606	196.168.1.1	196.168.1.21	SIP	Status: 100 Trying (1 bindings)
6	20.591805	196.168.1.1	196.168.1.21	SIP	Status: 401 Unauthorized (1 bindings)
7	22.330666	196.168.1.21	196.168.1.1	SIP	Request: REGISTER sip:corporate.com
8	22.331182	196.168.1.1	196.168.1.21	SIP	Status: 100 Trying (1 bindings)
9	22.332947	196.168.1.1	196.168.1.21	SIP	Status: 200 OK (1 bindings)
10	22.333102	196.168.11.1	196.168.11.2	SIP	Request: REGISTER sip:corporate.com[Malfo
11	22.365079	196.168.11.2	196.168.11.1	SIP	Status: 200 OK (1 bindings)

Figure 5.12: Registration at corporate domain during handover, PBX

No.	Time	Source	Destination	Protocol	Info
1	0.000000	196.168.11.1	196.168.11.2	SIP	Request: REGISTER sip:corporate.com[Malfo
2	0.030951	196.168.11.2	196.168.11.1	SIP	Status: 200 OK (1 bindings)
3	0.031160	196.168.22.2	196.168.22.1	SIP	Request: REGISTER sip:corporate.com[Malfo
4	0.036725	196.168.22.1	196.168.22.2	SIP	Status: 200 UPDATED (0 bindings)

Figure 5.13: Registration at corporate domain during handover, GW

Session establishment in the corporate domain for handover

Figure 5.14, shows the session establishment for handover in the corporate domain, from the "ue1" perspective while Figures 5.15 and 5.16, show the signaling going through the PBX and "ue0".

No.	Time	Source	Destination	Protocol	Info
6	60.317721	196.168.1.21	196.168.1.1	SIP/S	Request: INVITE sip:ue0.corporate@ims.com,
7	60.322243	196.168.1.1	196.168.1.21	SIP	Status: 407 Proxy Authentication Required
8	60.393317	196.168.1.21	196.168.1.1	SIP	Request: ACK sip:ue0.corporate@ims.com
9	60.393851	196.168.1.21	196.168.1.1	SIP/S	Request: INVITE sip:ue0.corporate@ims.com,
10	60.399422	196.168.1.1	196.168.1.21	SIP	Status: 100 Trying
11	61.460207	196.168.1.1	196.168.1.21	SIP	Status: 180 Ringing
12	64.539830	196.168.1.1	196.168.1.21	SIP/S	Status: 200 OK, with session description
13	64.554166	196.168.1.21	196.168.1.1	SIP	Request: ACK sip:ue0.corporate@196.168.1.1

Figure 5.14: Session establishment in corporate domain for handover, ue1

No.	Time	Source	Destination	Protocol	Info
4	30.332243	196.168.1.21	196.168.1.1	SIP/S	Request: INVITE sip:ue0.corporate@ims.com,
5	30.333020	196.168.1.1	196.168.1.21	SIP	Status: 407 Proxy Authentication Required
6	30.407185	196.168.1.21	196.168.1.1	SIP	Request: ACK sip:ue0.corporate@ims.com
7	30.409052	196.168.1.21	196.168.1.1	SIP/S	Request: INVITE sip:ue0.corporate@ims.com,
8	30.410506	196.168.1.1	196.168.1.21	SIP	Status: 100 Trying
17	30.413674	196.168.11.1	196.168.11.2	SIP/S	Request: INVITE sip:ue0.corporate@196.168.1
18	31.414642	196.168.11.1	196.168.11.2	SIP/S	Request: INVITE sip:ue0.corporate@196.168.1
19	31.445081	196.168.11.2	196.168.11.1	SIP	Status: 300 Redirect
20	31.445578	196.168.11.1	196.168.11.2	SIP	Request: ACK sip:ue0.corporate@196.168.11.2
21	31.455684	196.168.1.1	196.168.1.10	SIP/S	Request: INVITE sip:ue0@196.168.1.10;transp
22	31.463983	196.168.1.10	196.168.1.1	SIP	Status: 100 Trying
23	31.468723	196.168.1.10	196.168.1.1	SIP	Status: 180 Ringing
24	31.471168	196.168.1.1	196.168.1.21	SIP	Status: 180 Ringing
25	34.547956	196.168.1.10	196.168.1.1	SIP/S	Status: 200 OK, with session description

Figure 5.15: Session establishment in corporate domain for handover, PBX

No.	Time	Source	Destination	Protocol	Info
4	30.331608	196.168.1.21	196.168.1.1	SIP/S	Request: INVITE sip:ue0.corporate@ims.com,
5	30.333991	196.168.1.1	196.168.1.21	SIP	Status: 407 Proxy Authentication Required
6	30.406554	196.168.1.21	196.168.1.1	SIP	Request: ACK sip:ue0.corporate@ims.com
7	30.409479	196.168.1.21	196.168.1.1	SIP/S	Request: INVITE sip:ue0.corporate@ims.com,
8	30.410282	196.168.1.1	196.168.1.21	SIP	Status: 100 Trying
9	31.456772	196.168.1.1	196.168.1.10	SIP/S	Request: INVITE sip:ue0@196.168.1.10;transp
10	31.462956	196.168.1.10	196.168.1.1	SIP	Status: 100 Trying
11	31.467692	196.168.1.10	196.168.1.1	SIP	Status: 180 Ringing
12	31.470987	196.168.1.1	196.168.1.21	SIP	Status: 180 Ringing
13	34.546707	196.168.1.10	196.168.1.1	SIP/S	Status: 200 OK, with session description
14	34.549744	196.168.1.1	196.168.1.10	SIP	Request: ACK sip:ue0@196.168.1.10;transport
15	34.550152	196.168.1.1	196.168.1.21	SIP/S	Status: 200 OK, with session description
16	34.568209	196.168.1.21	196.168.1.1	SIP	Request: ACK sip:ue0.corporate@196.168.1.1
17	34.707433	196.168.1.21	196.168.1.1	RTP	Payload type=ITU-T G.711 PCMU, SSRC=1677721
18	34.712395	196.168.1.21	196.168.1.1	RTP	Payload type=ITU-T G.711 PCMU, SSRC=1677721
19	34.715383	196.168.1.21	196.168.1.1	RTP	Payload type=ITU-T G.711 PCMU, SSRC=1677721
20	34.720397	196.168.1.21	196.168.1.1	RTP	Payload type=ITU-T G.711 PCMU, SSRC=1677721
21	34.745366	196.168.1.21	196.168.1.1	RTP	Payload type=ITU-T G.711 PCMU, SSRC=1677721
22	34.753013	196.168.1.21	196.168.1.1	RTP	Payload type=ITU-T G.711 PCMU, SSRC=1677721

Figure 5.16: Session establishment in corporate domain for handover, ue0

5.3 Results Analysis

The first two registration cases shown in Figures 5.2 to 5.5, can be compared with the signaling registration in Figures 4.3 and 4.4. It is observed that the results correspond to the mentioned registration processes.

The traces for the session establishment from public to corporate domain shown in Figures 5.6 to 5.10, can be compared with the signaling in Figure 4.6. When comparing both figures, additional signaling appears in the implementation, messages like TRYING, RINGING and ACK, which are typical SIP messages for session establishment [24].

The traces for registration during handover shown in Figures 5.11 to 5.13, can be compared with the registration part in the handover case 2, shown in Figure 4.11. The registration in the corporate domain is forwarded to the Gateway and the IMS and a message "UPDATED" is returned by the IMS. This message indicates that for future signaling IMS will forward the requests for "ue1" to the Gateway and the Gateway to the PBX.

Finally, the traces shown in Figures 5.14 to 5.16, can be compared with the session establishment part in the handover case 2, shown in Figure 4.11. Again, additional signaling required for session establishment appears in the implementation. This implementation does not consider to include in the body of the messages any text to support handover. This comes from the fact that the clients used in the test bed, are based on the IETF standard. This standard does not include any special feature for handover. For the case of the Gateway and IMS, there is flexibility to customize the configuration file of SIP to support non-standard SIP signaling. Another limitation imposed by the clients, is the lack of capabilities to listen in two different network interfaces at the same time and thus, transmission and reception of packets at the same time is not supported. These limitations impose restrictions on the cooperation that can be achieved in real time between the clients and the rest of the entities, therefore a "live" test is not possible without the use of clients with the capabilities mentioned, but the basic concepts of the signaling were successfully proven.

Chapter 6

Conclusions and Future Work

6.1 Conclusions

It is concluded that the implementation of SIP-Supported handover is feasible and necessary in vertical handover cases, where a change of IP address is unavoidable. The change of IP address requires the transference of the data in the ongoing session to the new IP address destination. The way that SIP supports the handover is performing a session establishment in the new domain at the IP level, independently of the access technology. After that a session transference is also performed by SIP.

Another conclusion is that the proposal for handover given in this thesis, complements the current architecture and signaling described in [7]. With the handover capabilities included, a complete solution for inter-domain mobility is achieved for the convergence of corporate and public networks.

The network criteria selection proposed in this thesis considers a real case, where cost plays an important role in telecommunications and also the traditional RSS measurements to simplify implementation.

The advantage of IMS being a standard and the flexibility of SIP is one of the keys for the successful heterogeneous network convergence. Moreover, the use of standard SIP messages is possible including additional information in the body of the message.

For a live demonstration in the test bed, an IETF-based client does not meet the requirements for handover since this type of client is not designed for mobility. A general solution in the form of states machine has been proposed, which could be implemented in the client software in order to respond to mobility requirements.

6.2 Future work

The test bed is based on Openser which implements IMS and all its functionalities in one entity. However, a relatively new alternative known as Open IMS Core [28], has been released in a stable version for test purposes. This version considers the HSS, Proxy, Serving and Interrogating functions as individual entities and the possibility of using IMS clients. This software could allow to study in deep how the signaling goes inside IMS and have a more precise implementation of the convergence scenario.

A possible topic to analyze is QoS. In IMS, QoS is provided only at the network level under the assumption that differentiated services are provided and information contained in the SDP protocol is used for session content description. However, for an end-to-end QoS solution the air interface of wireless LAN must be considered. A description of QoS in WLAN and a method to map the QoS parameters to the network is required.

Another possibility for performing the handover could be that the Gateway or a new entity introduced in the current architecture, performs the signaling required for registration and session establishment on behalf of users. This approach could bring the advantage of reducing the signaling required, but security aspects need to be considered.

Finally, the signaling for handover could be optimized with concepts as described in [2], in order to minimize the signaling through the air interface and therefore handover time. The implementation requires to analyze the flows and which entities could be involved in the current architecture.

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Appendix

OpenSER

OpenSER is a mature and flexible open source SIP server created by a commercial venture called Voice Systems. It can be used on systems with limited resources as well as on carrier grade servers, scaling to up to thousands call setups per second. It is written in C programming language for Unix/Linux-like systems with architecture specific optimizations to offer high performance. Some of the SIP server functions are: registrar, location server, proxy server, redirect server; gateway to SMS/XMPP; or advanced VoIP application server [27].

The architecture of OpenSER consist on a core that receives and process SIP messages. The main functionality is given by a set of modules. The modules must be loaded first and their behavior shall be defined. This task is done by the openser.cfg file which controls the SIP router.

OpenSER Configuration File

Every time a SIP message is received the openser.cfg script is executed. This file decides how to handle each SIP message and the answers for messages. Seven sections can be found in an openser.cfg file:

- Global Definitions. Contains the IP address, listening port, debug level, and other settings that affect the OpenSER daemon.
- Modules. Contains the list of modules that will be loaded when OpenSER starts.
- Modules configuration. Set the parameters for the modules.
- Main Route. This section controls how each received message is handled.
- Secondary Routes. Routes that will be called form the main route.
- Reply Route. Optional section to handle replies to SIP messages.

- Failure Route. Optional route to deal with especial conditions (busy or timeout).

OpenSER Installation

This section describes the OpenSER installation under Unbuntu 6.1, it is assumed that the user is root and access to internet is available. The following commands should be written in the console mode:

- *apt-get update*
- *apt-get upgrade*
- *apt-get install build-essential ipsec-tools sun-java5-jdk ant libxml2-dev libmysql++-dev gcc-4.0 xlibs-dev libxml-dev libxaw7-dev qt3-doc qt3-dev-tools qt3-apps-dev flex bison libpq-dev libradiusclient-ng-dev xaw3dg-dev lsb-cxx libmysqlclient14*
- *updatedb*

For persistent storage execute: *apt-get install mysql-server*. After this *gedit /etc/mysql/my.cnf* and find the line "bind-address = 127.0.0.1" and comment it out. Should look like:
$\#bind - address = 127.0.0.1$

Installation procedure:

- Download $openser - 1.1.0 - tls\_src.tar.gz$ from www.openser.org
- Unzip at /usr/local, with: *tar xvzf* $openser - 1.1.0 - tls\_src.tar.gz$
- A new folder is created at: /usr/local/openser-1.1.0-tls, insidel this folder, in the makefile file erase mysql from the line "exclude modules" to allow OpenSER to be installed
- Compile with: */usr/local/openser-1.1.0-tls/make all*
- Install with: */usr/local/openser-1.1.0-tls/make install*, this will create an OpenSER cfg in /usr/local/etc/openser/openser.cfg. Change the content of this file with the one provided in the attached cd.
- To enable database mode, customize the openserctrl file, execute: *gedit /usr/local/etc/openser/ openserctlrc* and remove the comments from $DBENGINE = MYSQL$, $DBHOST = localhost$, $DBNAME$ $openser$, $DBRWUSER = openser$, $SIP\_DOMAIN = ims.com$.

- To start OpenSER: */usr/local/openser-1.1.0-tls/openser start*

- Additionally use: */usr/sbin/openserctl moni* to monitor the system

To set the root password in mysql:
$mysqladmin-u\ root\ password\ yournewpassword$, the password is set to openser. Do not forget to assign the same IP address as in the openser.cfg and set the domain to be ims.com or corporate.com in the ims and in the clients.

To create the mysql database the *openser_mysql.sh* script must be used, run the following command:
/usr/local/sbin/openser_mysql.sh create
You will be asked for the domain name OpenSER is going to serve (e.g., ims.com) and the password of the "root" MySQL user (e.g. openser). The script will create a database named "openser" containing the tables required by OpenSER. The script will add two users in MySQL:

- openser: having the password "openserrw", user which has full access rights to "openser" database

- openserro: having the password "openserro", user which has read-only access rights to "openser" database

Do change the passwords for these two users immediately after the database is created.

To add user accounts: A new account can be added using "openserctl" tool via:

openserctl add <username> <password> <email>

openserctl add test testpasswd test@mysipserver.comIf you are asked for SIP_DOMAIN environment variable do one of the following option. Use the password: *openserrw* when prompted