Simulation of SIP Supported Soft Handover between Enterprise Networks and IP Multimedia Subsystem

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ABSTRACT

Providing seamless mobility support is one of the most challenging problems towards the system integration of fourth generation (4G) wireless networks.

IP Multimedia Subsystem (IMS) is envisioned to be the common platform for service provisioning in heterogeneous 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 enterprise networks with 802.11 wireless accesses.

The interconnection of these two SIP-based networks for extended coverage and persistent service is one of the promising steps for heterogeneous network convergence. A solution for migrating existing sessions between these two networks, known as vertical soft handover is proposed in this thesis.

This thesis focuses on the handover procedure between Enterprise network and IMS. Here the proposed architecture is implemented with the help of OpenSER, Asterisk and analyzed using Wireshark network analyzer. The analysis of captured packets is done to measure QoS parameters like Required Bandwidth for SIP call, Packet loss, Delay & Jitter, and also used for evaluate Registration time & Call set up time.

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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
CSCF	Call State Control Function
FDD	Frequency Division Duplex
GGSN	Gateway GPRS Support Node
HLR	Home Location Register
HSS	Home Subscriber Server
IETF	Internet Engineering Task Force
IMS	IP Multimedia Subsystem
ISDN	Integrated Services Digital Network
IWU	Interworking Units
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
PBX	Private Branch Exchange
RTP	Real Time Transport protocol
SDR	Software Defined Radio
SDP	Session Description Protocol
SIP	Session Initiation Protocol
SGSN	Serving GPRS Support Node
UMTS	Universal Mobile Telecommunications System
UTRA	Universal Terrestrial Radio Access
UTRAN	Universal Terrestrial Radio Access Network

1. INTRODUCTION

1.1 GENERAL

The telecommunication 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 wire line and a fixed phone number. 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 enterprise networks, which require the implementation of advanced data and voice services in a secure environment. This scenario is depicted in Figure 1.1. 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 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) [1]. 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. 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. This process is known as vertical handover. The future heterogeneous network topology is depicted in Figure 1.2.

1.1.1 Introduction

Internet Protocol (IP) Multimedia Subsystem (IMS) is specified as part of the core network in the Universal Mobile Telecommunications System (UMTS). UMTS describes a standard which enables a flexible and high data rate transmission of any kind of data over cellular networks. IP Multimedia

Subsystem (IMS) is envisioned as the solution for the next generation multimedia rich communication. Based on an open IP infrastructure, IMS enables access independent convergence of data, speech, video and mobile network technologies. The key of the IMS success can be seen as the ability to interconnect users in heterogeneous networks.

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).

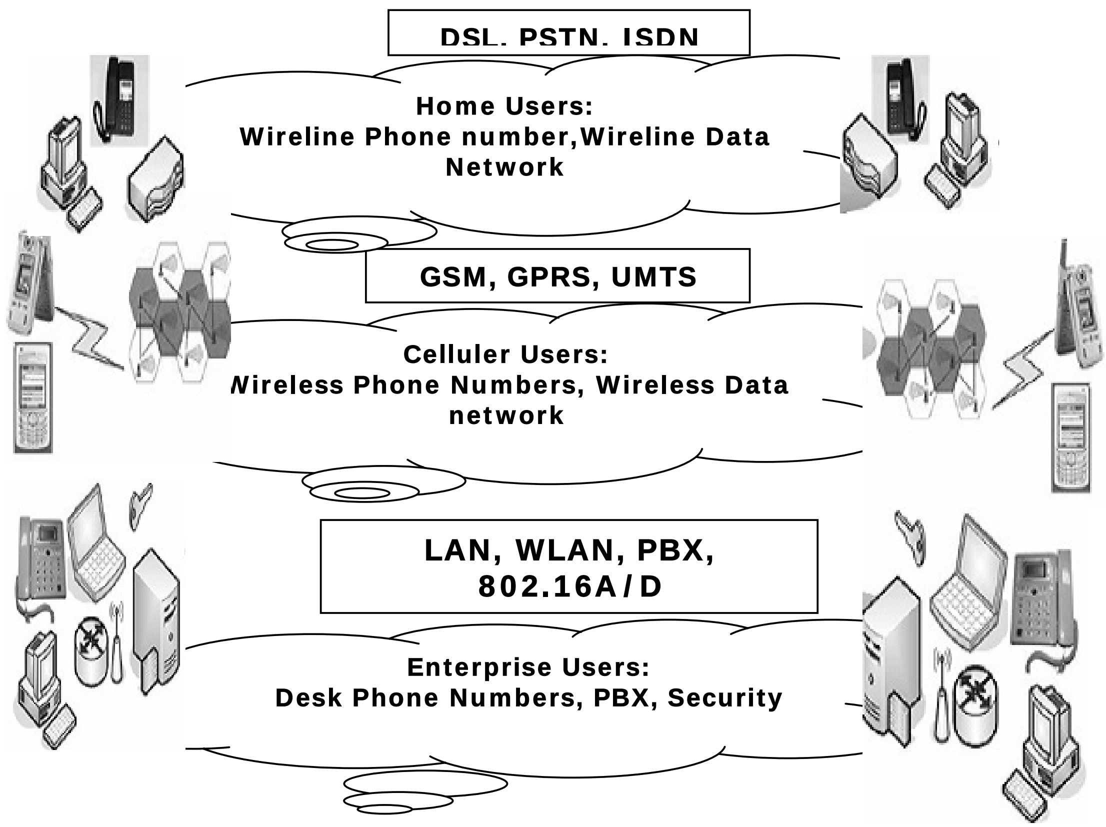

Figure 1.1: Current Heterogeneous Network Topology

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 Session initialization Protocol (SIP). SIP, as defined by the Internet Engineering Task Force (IETF) would be the base for the interconnection of future IP-based networks.

In particular, this project analyzes the interworking issues between UMTS and wireless enterprise networks for providing connectivity, mobility and handover management.

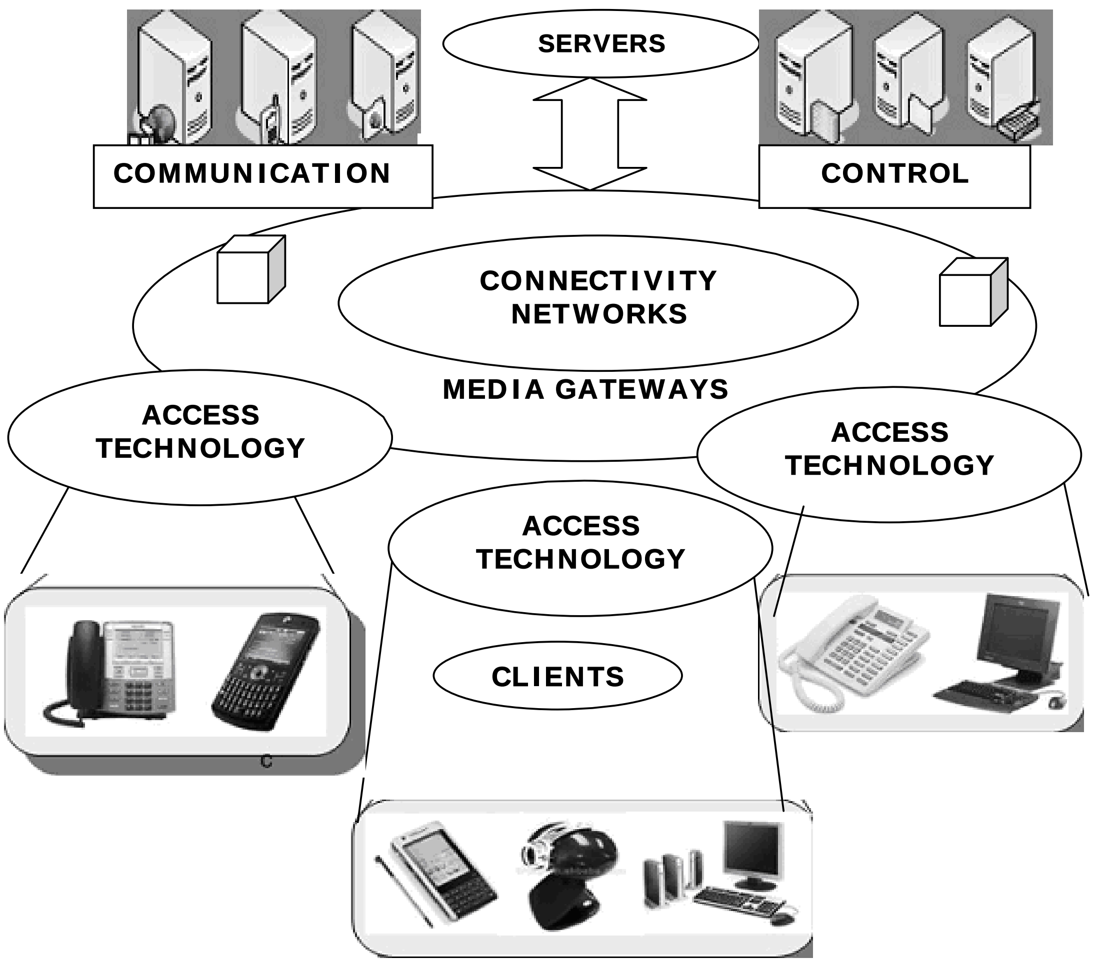

Figure 1.2: Future Heterogeneous Network Topology

A) SIP

Session Initiation Protocol (SIP) is an application-layer protocol for initiating, modifying, or terminating communication and collaborative sessions over Internet Protocol (IP) networks. A session could be an IP telephony call, a multi-user conference that incorporates voice, video and data, instant messaging chat or multi player online game. SIP can be used to invite participants to a scheduled or already existing session. Participants can be a person, an automated service or a physical device such as a handset.

SIP signaling follows the concept of common channel signaling, whereby the path used for the signaling traffic is independent of the path used for the actual data traffic. Separation of signaling traffic from media makes the session management more efficient, and is also more adaptive to functional changes.

B) Handovers

Handover means how the terminal keeps delivery of content when changing location beyond the range of single transmitter. In the convergence of heterogeneous access networks, internet-working is aimed to provide mobile users with ubiquitous connectivity when moving across different networks.

The scenario, for the convergence of heterogeneous networks, is similar to the architecture for the so called "Wireless Overlay Networks". 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 technology.

1.2 PROBLEM DESCRIPTION

UMTS proposes a variation of SIP for session control and end-to-end QoS management. SIP is also used in many current enterprise networks based on 802.11 wireless accesses. 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 an enterprise 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 interdomain mobility needs to be defined. Finally, the interdomain 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 SIP Forum Recommendation. This specification is known as SIP connect, and introduces the guidelines in order to provide connectivity to the enterprise users in the public network. The SIP connect network architecture is modified and an improvement is done to consider users mobility. Additionally, the case of a enterprise user with 2 subscriptions, but only one number for setting up and receiving calls in both PBX (enterprise network) and IMS (public network) is proposed.

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.

1.3 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 (Chapter 3), it is assumed that a device located physically in the enterprise network coverage, is always in the range covered by both, the public network and the enterprise network. 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 enterprise network coverage, since it is based on a short range WLAN access. A user can be connected to the public and/or enterprise network using 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 enterprise network and other to access to the public network.

1.4 PROJECT SCOPE

Different kinds of network have different kind of specification. The goal of this project is to develop mechanisms for SIP-supported soft-handover between the enterprise network and the public operator network.

First propose a signaling mechanism to allow a vertical soft-handover between enterprise and public networks. Then study the network criteria selection and the appropriate parameters for handover prediction. For implementation study the interaction between SIP clients and the network for a efficient handover. Then implement the concepts proposed in an experimental set-up.

For performance analysis we have to study and compare different kinds of protocol for handovers and also study different kinds of algorithms for network selection during handover. To study the QoS parameters like session delay, packet loss. Then develop one application for vertical handovers in heterogeneous network and measure its QoS parameter and find the solution to improve it.

1.5 TECHNICAL ASPECTS

To implement this project in real scenario the required components:

A Multimode equipment - for detection of different kind of network.

B 802.11 wireless router –for implementing WLAN.

C Workstations – having operating system fedora core (2.6.18-1, fc6).

Required softwares:

A	Asterisk	:	IP – PBX support
B	OpenSER	:	Open source SIP server
C	Wireshark	:	Network monitoring system
D	Kphone,SJphone	:	Communication between two workstation
E	MySQL	:	Database of sip massages, voice calls, user's Information, sip sessions
F	Stun server	:	Network Address Translation

All of the above softwares are open source softwares.

1.6 OUTLINE OF THESIS

After getting the overview of thesis, Chapter 2 provides the brief introduction about the background details required to understand the developed solutions. It also includes detail information about UMTS, IMS, Enterprise network with their corresponding architecture, SIP protocol and its components, handover and types of handover. Chapter 3 explains the process of vertical handover, which includes the detail of protocol with necessary assumptions for handover and some proposed solution. Chapter 4 contains the details about the signaling for handover between enterprise network and public network and presents the test bed for the concept implementation of the proposed solution. Chapter 5 presents the results and analysis of the implementation. Finally conclusions and future work are given in Chapter 6.

2. THEORETICAL BACKGROUND

2.1 GENERAL

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 network is used to refer to the cellular network. Following the cellular system description, a brief introduction to enterprise networks, the evolution to IP services and VoIP is provided together with a brief explanation of mobility support through wireless access. The term enterprise network is used to refer the enterprise network. After describing the two main networks for convergence, IMS is introduced as architecture to unify these heterogeneous networks. The IMS architecture and the 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.

2.2 UMTS

2.2.1 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 [1].

2.2.2 UMTS Architecture

UMTS describes a standard which enables an exile 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 2.1. This architecture allows coexistence with second generation systems to preserve previous operator investments [1].

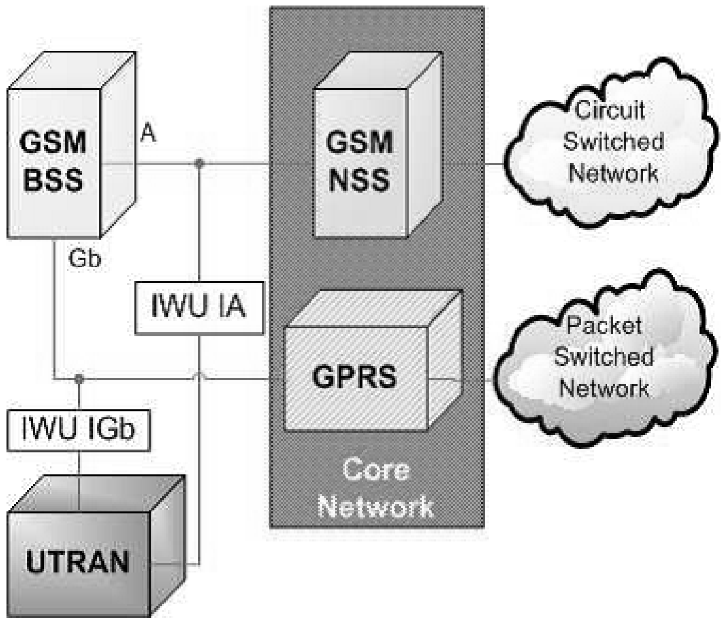

Figure 2.1: UMTS general system architecture

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).

2.2.3 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. In order to have e client subscriber multiple accesses, 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 2.2.

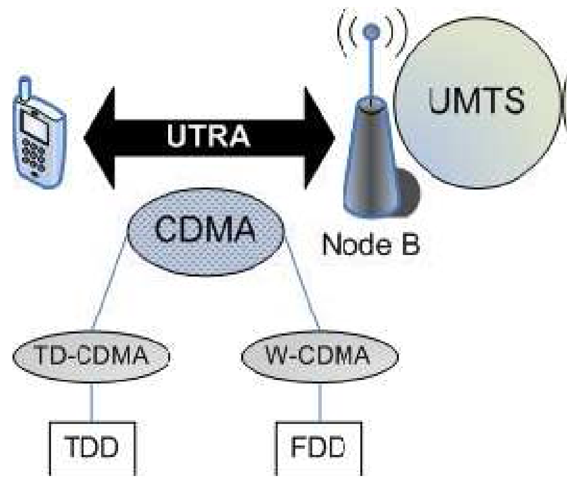

Figure 2.2: UMTS air interface

2.2.4 Services

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

- A Location based services
- B Speech and Video telephony/ video conferencing
- C Video on demand
- D 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.

A **Conversational Class:** This class is very delay sensitive and includes services such as speech and video telephony, or video games.

B **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.

C **Interactive Class:** This class includes web-browsing or interactive games, data integrity is more important than delay.

D **Background Class:** This class includes reading mail, sending mail, or saving some data in mail account, data integrity is more important than delay.

2.3 ENTERPRISE NETWORKS

An enterprise network known also as "Corporate 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 [1].

2.3.1 Enterprise Networks Evolution

The evolution of enterprise 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 enterprise 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 enterprise networks. In this evolution, multimedia and data are transported over the same packet switched circuit.

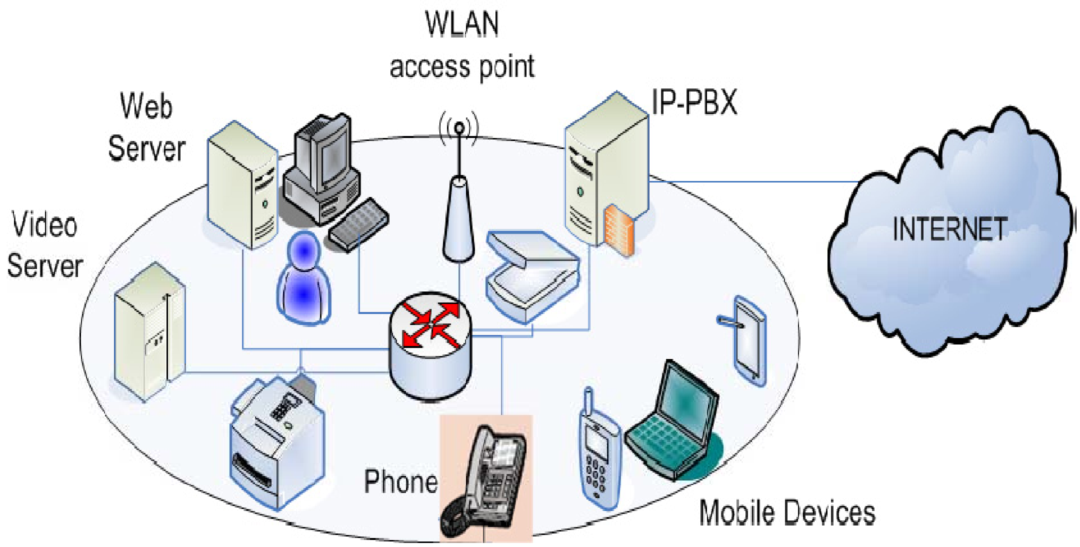

Figure 2.3: Typical Enterprise Networks components

Figure 2.3, shows typical enterprise network components. A user in a enterprise 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.

2.3.2 Architecture

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

A A core communications switching system (PBX/IP-PBX for example), capable to provide call processing features and data transference.

B A management system, capable to support fault and configuration operations.

C Call accounting system, capable to analyze and process call records to generate and maintain billing schemes and traffic reports.

D A voice messaging system, offering complementary services to the basic answering system.

2.3.3 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.

IP-PBX (Internet Protocol-Private Branch Exchange) serves as a private telephone network in an organization. IP PBX allows multiple users in an organization to be connected to each other and also allows them to be connected to external phone lines. PBX has been around for a long time and users in the network can be given a three of four digit extension that they can dial to speak to each other. When users in a network need to call outside telephone lines, they typically dial a predefined number (e.g. 9 or #) that connects the user to the public telephone system or to the PBX operator [1].

PBX can be defined as a communication device that initiates communication between two extensions and keeps the connection open till one or both parties disconnect the call. The PBX system also provides call detail recording and other relevant information for each extension. IP PBX systems have many features like call transfer, voice mail, direct dialing, call forwarding service on busy or absence, music on hold, pre-recorded welcome messages, re-recorded instructions with options where the caller can dial different numbers to reach different services, automatic ring back, call distribution, waiting, pickup, park, conferencing, greetings, shared message box, automatic directory services, etc. To develop IP PBX in this Project, we use Asterisk.

2.3.4 Protocol Layering

Enterprise 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. 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 2.4, 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.

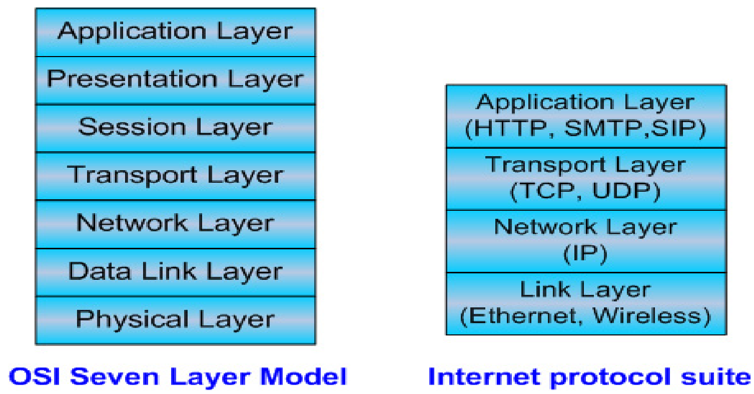

Figure 2.4: Layered Protocol Stack

2.3.5 Mobility in enterprise networks

In order to support mobility in enterprise 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 heterogeneous access networks. The increasing business advantages of having connectivity and service availability anytime, anywhere, implies that mobile devices in enterprise 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.

2.4 IP MULTIMEDIA SUBSYSYTEM

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 [2]. 3G networks aim to merge two of the most successful paradigms in telecommunications: cellular networks and Internet. 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 2.5 depicts the scenario for seamless service provision.

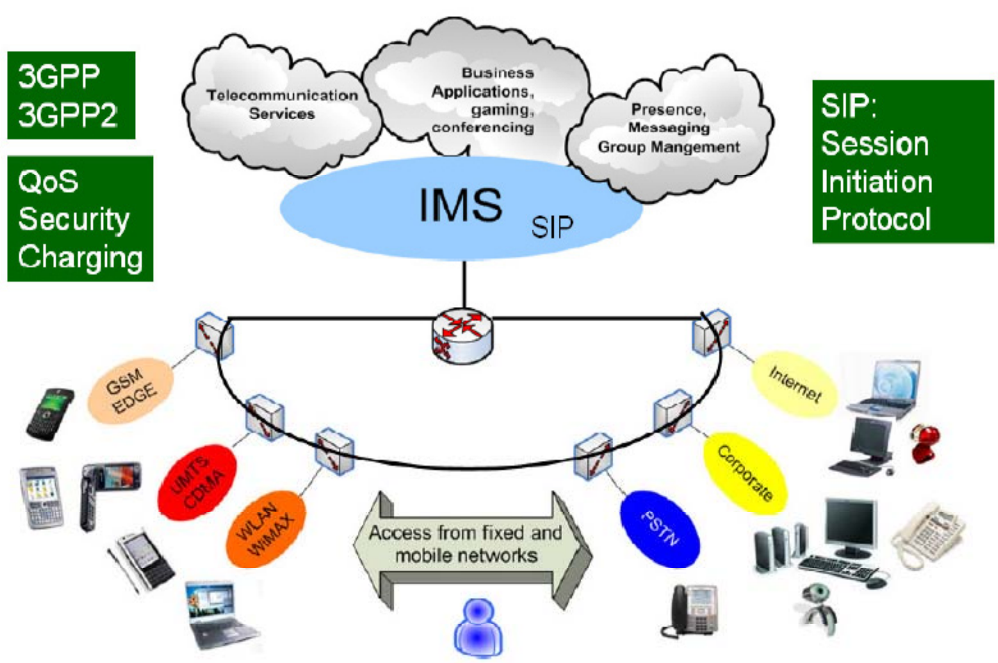

Figure 2.5: IMS Seamless service provision

2.4.1 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-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 [2].

A Session Initiation Protocol (SIP). This protocol is used to establish and manage sessions over IP networks.

B Diameter. The Authentication Authorization and Accounting (AAA) protocol is based on Diameter.

C Common Open Police Service (COPS). Is used to transfer polices between Policy Decision Points and Policy Enforcement Points.

D 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).

E Real Time Transport protocol (RTP). Used to transport real time media like video or audio

2.4.2 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 2.6 [2]. 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.

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

A 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.

B 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.

C 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.

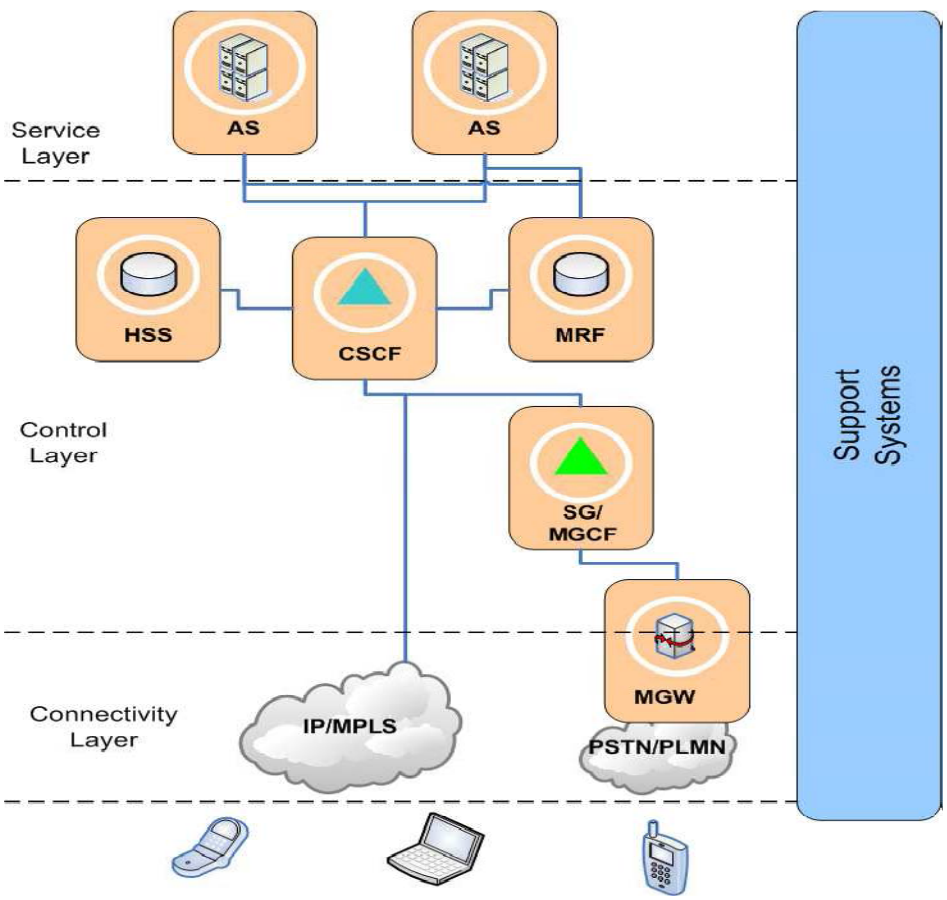

Figure 2.6: Simplified view of the IMS layered architecture

D 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.

E 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.

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

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

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

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

2.5 SESSION INITIATION PROTOCOL

Until the mid 1990 the telecommunication companies provided most of our communication needs. Today the picture is totally different. There is a need of new kind of networks to handle the communication. These will be primarily based on IP technology this also mean that a wide spectra of new services will show up and SIP could be a solution to handle many of the new challenges that will arise.

Session Initiation Protocol is a text based protocol similar to HTTP and SMTP. It was developed in the mid 1990 by IETF [3], who develops and promotes standards on Internet, and Multiparty Multimedia Session Control (mmusic), who develops protocols to support Internet teleconferencing and multimedia conferencing.

SIP is still under development

The Session Initiation Protocol working groups are people that continue the development of the protocol. The working groups will for example develop a proposed extension that comes from new requirements. The SIP Working group is charged with being the guardian of the SIP protocol for the Internet, and therefore should only extend or change the SIP protocol when there are compelling reasons to do so.

The Session Initiation Protocol Project INvestiGation (SIPPING) is another working group. Group is chartered to be a filter in front of the SIP group. This working group will investigate requirements for applications of SIP. To know what has to be extended the group documents the use of SIP for applications related to multimedia and telephone.

2.5.1 Overview of the protocol

Session Initiation Protocol (SIP) is an application-layer protocol for initiating, modifying, or terminating communication and collaborative sessions over Internet Protocol (IP) networks. A session could be an IP telephony call, a multi-user conference that incorporates voice, video and data, instant messaging chat or multi player online game. SIP can be used to invite participants to a scheduled or already existing session. Participants can be a person, an automated service or a physical device such as a handset.

SIP signaling

SIP signaling follows the concept of common channel signaling, whereby the path used for the signaling traffic is independent of the path used for the actual data traffic. Separation of signaling traffic from media makes the session management more efficient, and is also more adaptive to functional changes [3].

SIP signaling supports the following facets of multimedia session management.

A User location: Enabling users to access telephony or other application features from remote locations.

B User availability: Determining the willingness of the called parties to engage in communication sessions.

C User capabilities: Determining the media and media parameters to be used for communication sessions.

D Session setup: Establishing the session parameters for point-to-point and multiparty Sessions.

E Session management: Enabling the transfer and termination of sessions, the modification of session parameters, and the invocation of session services.

SIP Capabilities

The following list shows some examples of what SIP is capable of:

Add / Drop media: SIP supports to add and drop media during a session. This means that if a A speaks to B over also using video stream over an established connection, can turn off the video and continue speaking. After a while they decide that want to turn it on again and do so, all this during the same session.

Find me / follow me: SIP gives the opportunity to be registered at different locations at the same time. This mean that if an incoming audio INVITE message arrives all devices will ring at the same time and stops ringing when one of the devices answers the incoming call. It is also possible to register with a video device during an audio call to be able to extend the cal with a video stream.

Presence and instant messaging: The SIMPLE working group of IETF the leading candidate to fulfill the requirements of IMPP. Conferencing and distance working: Support for conference and distance working. This could be a teacher providing distance learning support to students spread around the country.

Multi party gaming: SIP can also support audio and video feed during multiplayer games. This could be taken care of either by the game itself or via some extra software.

2.5.2 Components of SIP

SIP is a peer-to-peer protocol. These peers are called User Agents. A UA can be either a User Agent Client, which is the application that initiates the SIP requests, or it can be a User Agent Server, which is a server application that returns a response initiated by the UAC [3]. The different component of SIP is shown in Figure 2.7.

A) SIP Client

Phones acts as either a UAS or UAC. Both SIP-capable phones and Software phones are able to initiate a request and respond to requests. Gateways provide many translation services between different endpoints and terminal types. For instance a gateway to PSTN.

B) SIP Server

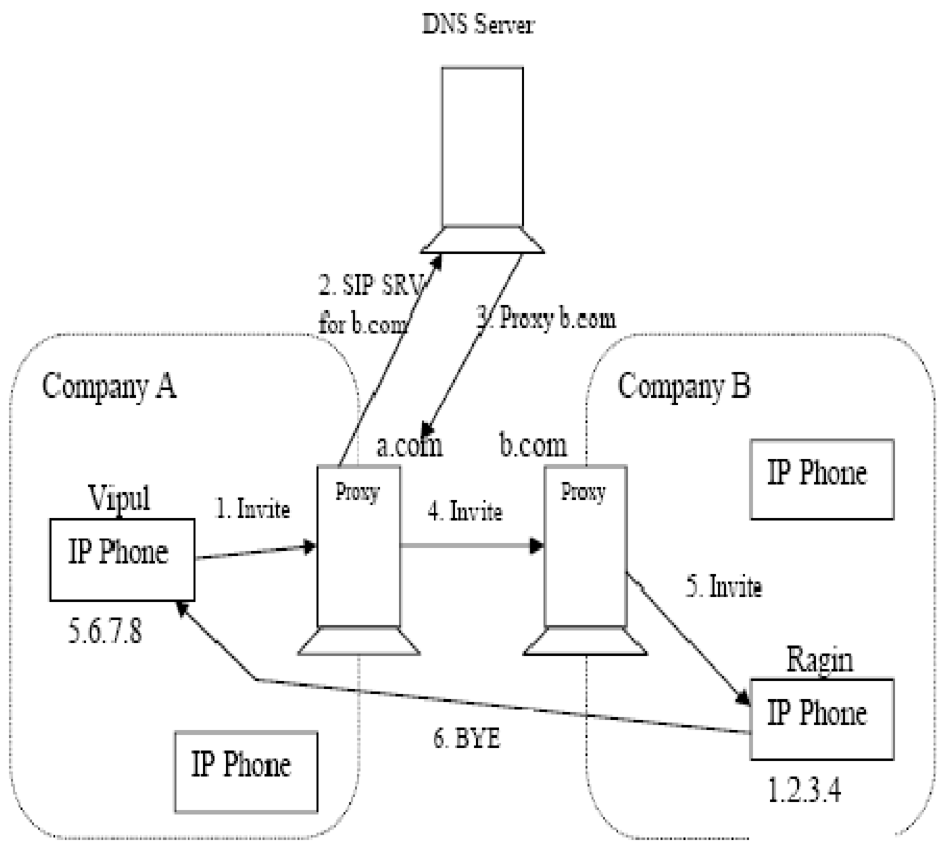

Figure 2.7: Different components of SIP, their roles and how they interact with each other.

Proxy server is a device that receives SIP requests from the client and then redirects them to the appropriate server. It is also responsible for authentication, authorization, network access control, routing and security.

Redirect server responds to Client with which hops the UAC should take in order to speak to the UAS.

Registrar server is responsible for registration of UACs current location.

SIP Messages

The message types of SIP are typically Request-Response messages, either a request from a client to a server, or a response from a server to a client. Currently, there are six methods of request defined in SIP described in table 2.1.

Table.2.1 SIP Request method

Message Name	**Function**
REGISTER	Register a user with a SIP server (with location service)
INVITE reINVITE	Invite user(s) to a session. The body of the message contains the description with the address where the host wants to receive the media stream ReINVITE is for changing the session (call) parameters
ACK	Acknowledgement of an INVITE request
CANCEL	Cancel a pending request
BYE	Terminate a session (release a call)
OPTIONS	Query servers about their capabilities

Responses to request methods by a three-digit status code indicate success or failure, distinguished by code defined in table 2.2.

Table.2.2 SIP Response code

Code Classes	Response Type	Function Description
1xx	Provisional	Request received, continuing to process the request
2xx	Success	The action was successfully received, understood and Accepted
3xx	Redirection	Further action needs to be taken in order to complete the request
4xx	Client Error	The request contains bad syntax or cannot be fulfill at this Server
5xx	Server Error	The server failed to fulfill an apparently valid request
6xx	Global Failure	The request cannot be fulfill at any server

SIP messages are composed of the following three parts: START LINE; HEADERS and BODY (CONTENT). SIP messages appear both in request and in response messages. SIP makes a clear distinction between signaling information, conveyed in the SIP Start Line and headers, and the session description information.

Generic- message = start- line (= Request-Line | Status-Line)

Message- header = (general-header
Request-header
Response-header
Entity-header)

CRLF (Carriage-Return Line-Feed: an empty line indicating the end of the header fields)

[Message-body]

Every SIP message begins with a Start Line. The Start Line conveys the message type (method type in requests, and response code in responses) and the protocol version. The Start Line may be either a Request- line (requests) or a Status- line (responses), such as follows:

I The Request- line includes a Request URI, which indicates the user or service to which this request is being addressed.

II The Status-line holds the numeric Status-code and its associated textual phrase.

SIP header fields are used to convey message attributes and to modify message meaning. Headers can span multiple lines. Some SIP headers such as Via, Contact, Route and Request Route can appear multiple times in a message or, alternatively, can take multiple comma separated values in a single header occurrence.

A message Body is used to describe the session to be initiated or alternatively it may be used to contain opaque textual or binary data of any type, which is related in some way to the session. Message bodies can be written as

<Name> :< value>

SIP addressing

SIP addresses are expressed as URI which contain the URL of participating parties [3]. The basic form in which URLs are expressed is: sip:user@foo.com However the URL can also contain all the parameters used to establish a call. For instance tel:+358-555-1234567;postd=pp22 becomes sip:+358-555-234567; postd = pp22@foo.com; user=phone which means that we want to establish a call to someone with a global phone-number. Using standard URL in SIP implies that a DNS must be used to map domain names and hosts into IP numbers. This is an important aspect due to the integration between other web based technologies.

Overview of the operation

Two persons A and B wants to communicate over the Internet (an IP network). Part of this can be done using SIP. Both parties that are going to communicate use a user agent which is some sort of software. This software enables communication between A and B. The software can be installed in a PC or some other device such as a mobile device.

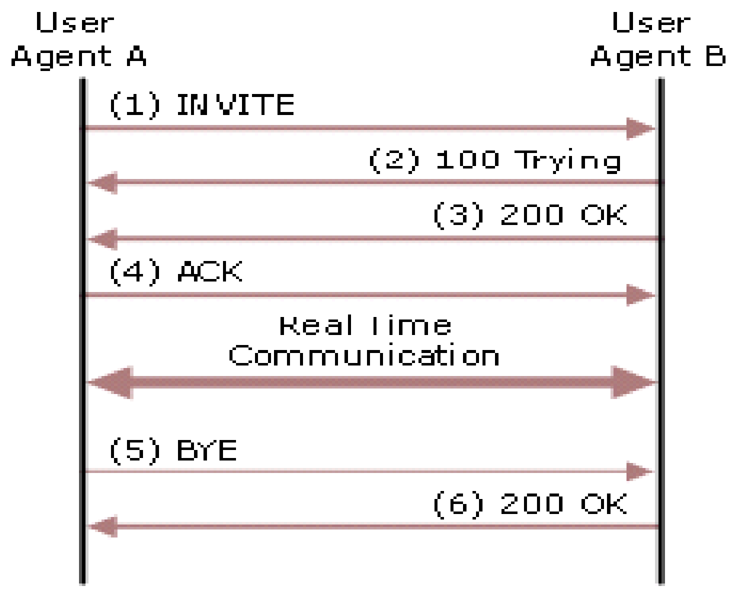

Figure 2.8 call flow for initiated calls by the user agents

Let’s say that A wants to speak with B. The first thing A will do is to send a message to B on the standard SIP port (5060). This message is called INVITE message and it contains information about what media that can be used. When B who is listening on the SIP port is receiving the INVITE message he responds by sending back a message containing the type of media that B prefer. When A finally responds to B with an ACK message both parties know what type of media that will be used and what bandwidth they can use, they are have also been aware of each others IP and what ports that will be used for the communication. When both parties have received their ACK:s the start to communicate with each other over the decided ports. This is often done by using the RTP (real-time protocol) but some other transport protocol can also be used. Sometimes one of the communicating parts wants to change the media during communication this can be done by sending additional SIP messages. When communication is done either A or B send a BYE message to end the session and when this I received bye the part on the other side the session is closed. As shown in Figure 2.8.

The following encoded message is an example of what an INVITE-message might look like:

```
INVITE sip:222.222.3.5 SIP/2.0
Via: SIP/2.0/UDP
222.222.5.2;branch=z9hG4bKdede05020000007747fc638d
0000631c0000003c;rport
From: "Bhumi" <sip:OpenSER@222.222.5.30>;tag=48e92f5c25
To: <sip:222.222.3.5>
Contact: <sip:OpenSER@222.222.5.2>
Call-ID: B8A3A30272194580A3D3C7ECCDA82CCB0xdede0502
CSeq: 1 INVITE
Max-Forwards: 70
User-Agent: SJphone/1.65.377a (SJ Labs)
Content-Length: 362
Content-Type: application/sdp
Supported: replaces,norefersub,timer
```

The datagram protocol UDP is often used during these sessions but also other protocols can be used but these are optional. Due to that UDP is an unreliable protocol SIP can handle retransmission itself.

Mobility Support using SIP over WLAN

The user in a SIP session may move between different access points' service area over time. These locations can be dynamically registered with the SIP server. According to the dynamic parameter, SIP can initial a new session, or can also modify the existing session without releasing the session.

A typical handoff procedure in application layer mobility management includes two phases. One is registration in the new location; the other is redirection of an ongoing session, during which considerable delay may be introduced to the session. SIP can support terminal, session, personal and service mobility [3].

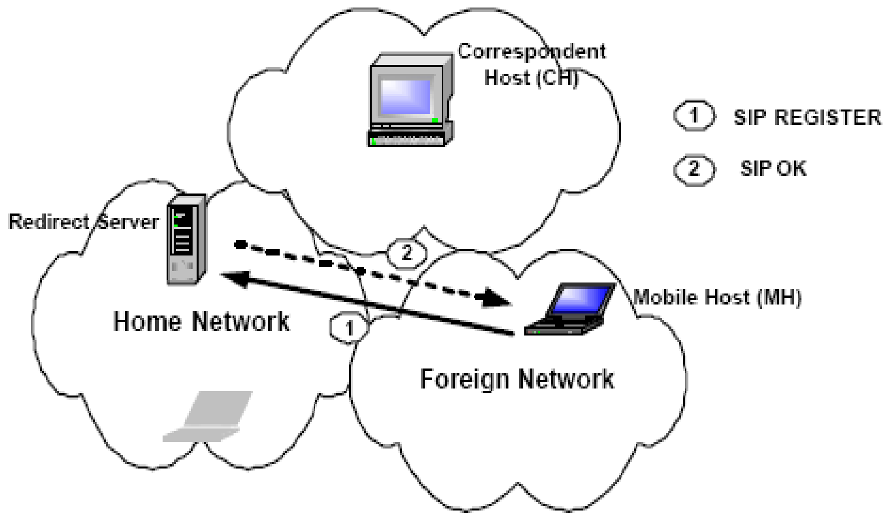

Figure. 2.9 Pre-Call mobility: registration procedure

Terminal mobility allows a device to move between different Access Points while being reachable to other hosts and continuing any ongoing session while on the move. It requires set- up of a connection either during the start of a new session when the user has already moved to a different location, or in the middle of an existing session. We call the former situation as Pre-Call mobility, and the latter one as Mid-Call mobility. For Pre-Call mobility, the user (Mobile Host: MH) will register, or re-register its new location with its home registrar SIP server or redirect server. Any call to the MH is thereby forwarded to the new location. By consulting the home server of MH and obtaining the new IP address of MH, the Corresponding Host (CH) establishes the session with MH directly. It solves the issue of the triangular routing happened in Mobile IP support Mobility. See Figure.2.9 for this registration procedure

In the case of Mid-Call mobility support, the MH will notify the CH about its location change by sending SIP request directly. We already know that SIP client (UAC) can initiate a request to modify an existing session by sending a new INVITE message using the same Call-ID. The MH attaches the update description of the new IP address with the new INVITE (reINVITE) message directly to CH to tell the CH where it wants to receive future message. After MH receives acknowledgement from CH for this reINVITE, the two sides of this connection will continue following data communication, which is shown in Figure. 2.10. This project concentrates on the implementation of Mid-Call mobility support.

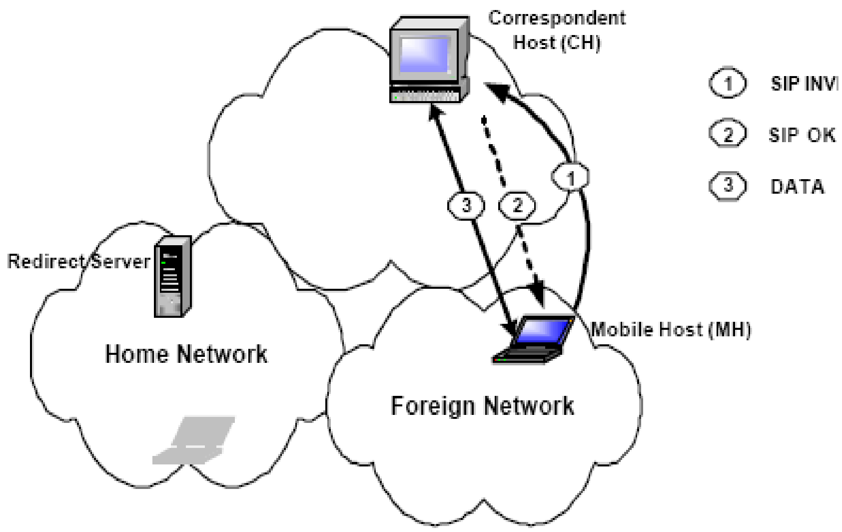

Figure. 2.10 Mid-Call mobility: reINVITE procedure

According to, Figure.2.11 gives out a detailed description of timing for SIP-based handoff procedure in WLAN.

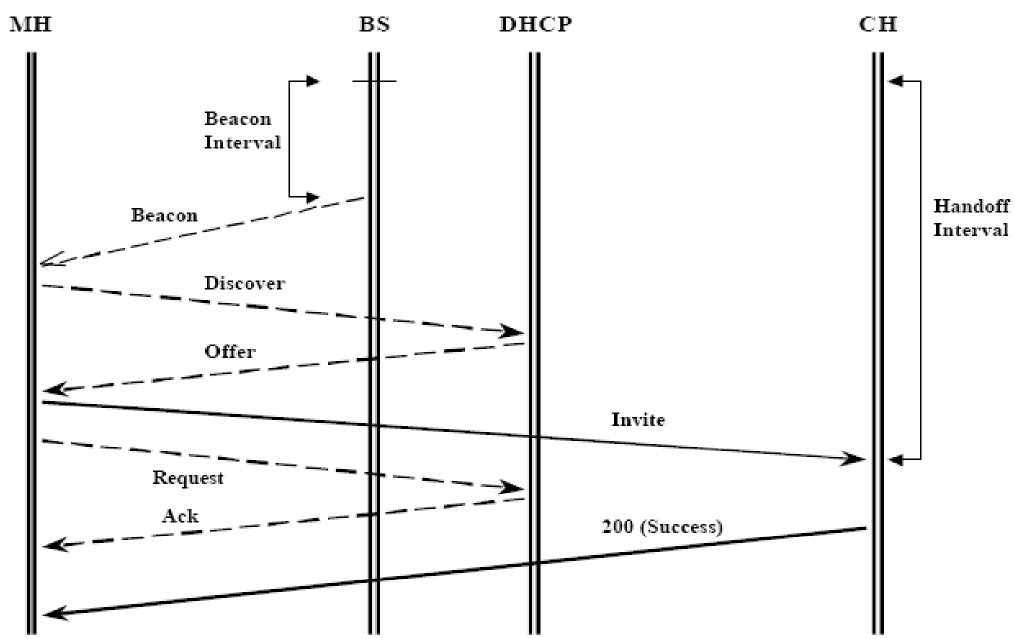

Figure. 2.11 Typical timing for SIP-based handoff procedure in WLAN

2.5.3 Integration with PSTN

Setting up communication between two clients on the Internet (or IP network) using SIP may not be so difficult but what about if a person on the Internet wants to speak with some other person who is not on the network? What if the person is on the public switched telephone network?

So let us go back to the problem. Two persons A and B wants to speak with each other. Person A is sitting on the Internet while person B is sitting on the PSTN. Since they are sitting on two different networks which speak different languages they can't communicate. The solution to this is to use an adapter that converts the signals from PSTN to fit into the Internet and vice versa this device is called gateway . The first SI message sent to start the session can look like the message above e.g; sip:+358-555-1234567;postd=pp22@foo.com;user=phone

A) ENUM

One problem that arises when a client on a PSTN wants to call to another client on a IP network is how to get in contact with this person. The answer to this question is ENUM. By creating a global directory which maps to SIP addresses or email etc. Support for E.164 numbering in DNS (ENUM) allows SIP clients and servers to send and receive phone numbers in place of SIP URI:s in messages and to route them as usual. The E.164 number queries are formed as a reversed dot separated number to which the string .e164.arpa is added. DNS and ENUM helps an ingress gateway to resolve the SIP address from a E.164 number so it then can reach the target.

B) FORKING

SIP proxy server offers forking. This means that they have the capability to forward incoming messages to several different receivers. This can be to a phone and at the same time also to a web camera and a desk computer or lap top. The proxy server then checks the answer and makes sure that only one device will answer so that the asking client gets a single stream back.

C) VIRTUAL PRIVATE NETWORK

Because of the use of indirection capabilities of SIP, ENUM and DNS virtual private networking is made easy to handle and develop. A single SIP proxy server can provide address mapping and forwarding services for a remote location making it look like people are on the same domain but they actually are not.

2.6 HANDOVERS

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.

2.6.1 Introduction

Handover means how the terminal keeps delivery of content when changing location beyond the range of single transmitter. In the convergence of heterogeneous access networks, internet-working is aimed to provide mobile users with ubiquitous connectivity when moving across different networks [4].

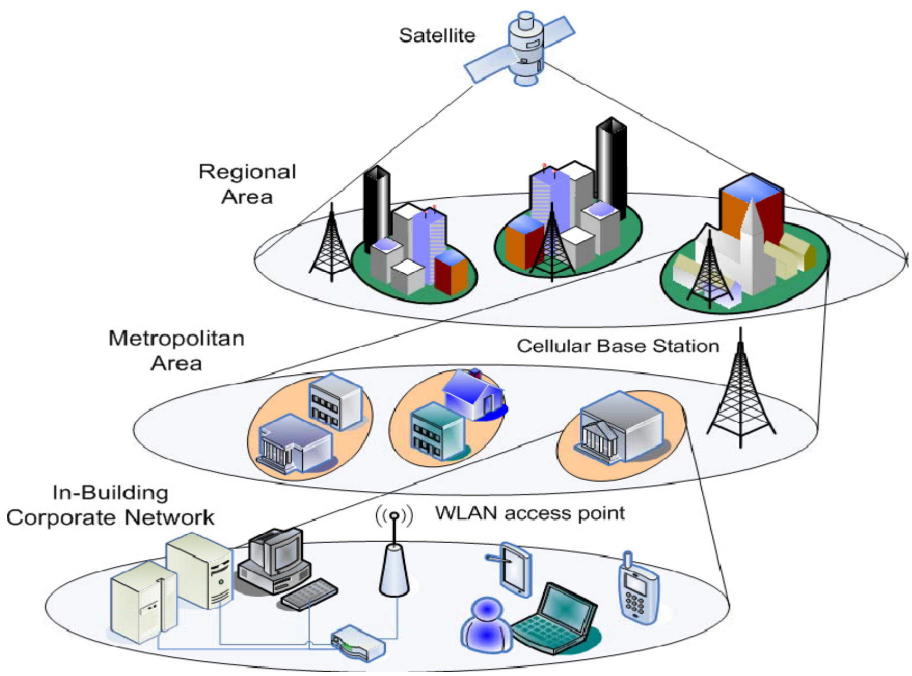

Figure 2.12: Overlay Wireless Networks

The scenario, for the convergence of heterogeneous networks, is similar to the architecture for the so called "Wireless Overlay Networks". 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 technology [4].
Figure 2.12 is depicted this scenario.

2.6.2 Handover in Heterogeneous Networks - Overlay Networks

Based on the architecture shown in Figure 2.12, 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.

A) Hard-Handover and Soft-Handover

A hard-handover happens when the mobile node being connected to an access point, with an ongoing 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.

B) 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.

C) 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 accesses technologies is called vertical or heterogeneous handover. Figure 2.13, shows both cases. The study in this thesis is referred to the implementation of vertical handover.

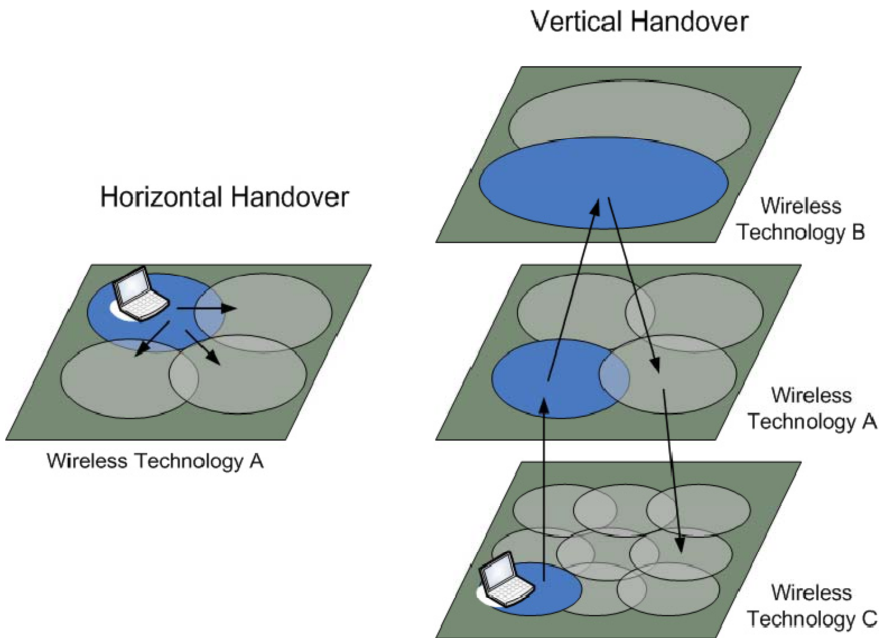

Figure 2.13: Horizontal and vertical Handover

D) 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. VERTICAL HANDOVER PROCESS

3.1 VERTICAL HANDOVER PROCESS

Since the problem addressed in this thesis is the handover for the convergence of heterogeneous networks defined as vertical handover [5]. The process can be divided into three steps:

A) 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.

B) 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. The following shows possible parameters to consider in order performing a handover.

Handover decision parameters

- Signal strength
- Network Conditions
- Cost
- Application types
- Services Provided
- User Preferences

C) 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.

3.2 PROTOCOLS FOR HANDOVER AND PROPOSED SOLUTIONS

This section shows a description of the interworking architecture, the components and necessary assumptions is dedicated to present the proposed solutions for vertical handover and the proposed handover procedure and handover cases, describes the signaling for connectivity and shows the proposed SIP client capabilities needed to perform handover [6].

Interworking Architecture

The proposed architecture for the convergence of 3GPP networks and WLAN-based enterprise networks is shown in Figure 3.1. This interworking architecture enables cellular system operators to provide connectivity and extended coverage to enterprise networks [7].

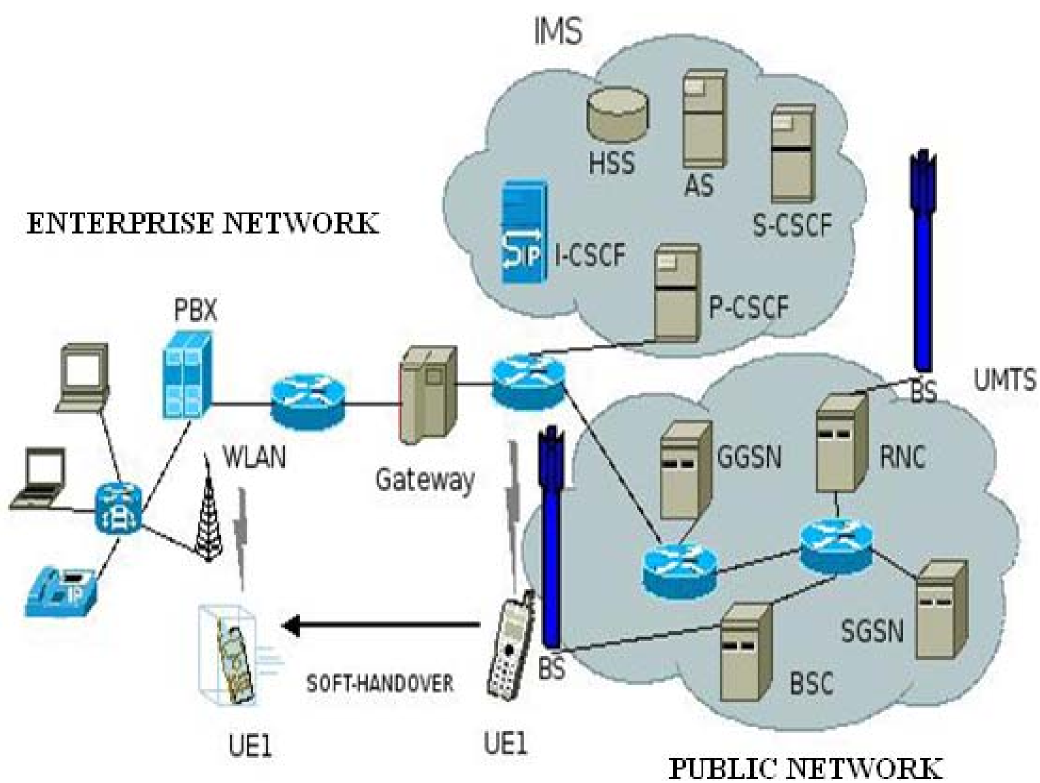

Figure 3.1: Interworking Architecture

The following components are present in the architecture:

A) Enterprise Network

The enterprise network consists of multiple network elements in order to provide connectivity, data, voice and multimedia services in a private network. The key component in the enterprise network architecture is the IP-PBX, referred from now as PBX. This entity is capable to perform functions like user registration, authentication and session establishment inside the enterprise network. The PBX signaling is based on the SIP protocol. To be able to access to services in the enterprise network, the user equipment is capable to perform session establishment based on the SIP protocol.

B) Public Network

The public network consists 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 network described in this thesis is based on UMTS. In order to support the cellular network for service provision, the IMS is introduced as an additional architecture which is part of the public network. The IMS performs user authentication, registration and session establishment in the public network 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. The terms user equipment (UE) and Mobile Node (MN) are used in the following lines to refer to the clients.

C) Gateway

The main component in the interworking scenario is the Gateway. This element acts as a "SIP translator" and exchanges SIP messages between the enterprise and the public networks. The Gateway interconnects these two networks through a link to the PBX in the enterprise network and a link to the IMS in the public network. The Gateway is proposed to be part of the enterprise network and the implementation can be in the same hardware as the PBX or as an independent entity.

3.3 REQUIREMENTS FOR VERTICAL HANDOVER

In this session we discuss a set of requirements that a mobility management solution based on SIP should have [5].

A The solution should be as fast as possible.

B The solution should support a "forward handover" (i.e. in which all the procedure is performed on the new target Access Network).

C The handover solution should be compatible with NATted networks.

D The handover solution should not require a support in the different access network.

E The switch of the "active" interface during a SIP transaction should be supported.

F An optional desirable requirement is to allow the decoupling of "user level" registration and mobility and "terminal level" mobility.

G An optional desirable requirement is to provide privacy with respect to user location and user movements.

H Another desirable feature is that existing user agents should inter-work with the handover procedure without the need to be updated.

3.4 SIGNALING FOR CONNECTIVITY

This section describes a set of protocols that allow users to register in the enterprise and public networks and establish connections among them in any of the networks, with a unique subscriber ID [6].

3.4.1 Registration

For registration, three cases are described: group registration, individual registration at enterprise network and individual registration at public network [6].

A) Enterprise Network Registration Process

An initial registration is required to inform the public network that the enterprise network is available. This initial registration is done by the

Gateway, using a unique address with the form: pbx.enterprise@ims.com. The initial registration notifies the IMS component in the public network that the enterprise network is available through the Gateway and that all the calls to addresses with the form *.enterprise@ims.com. This is shown in Figure 3.2.

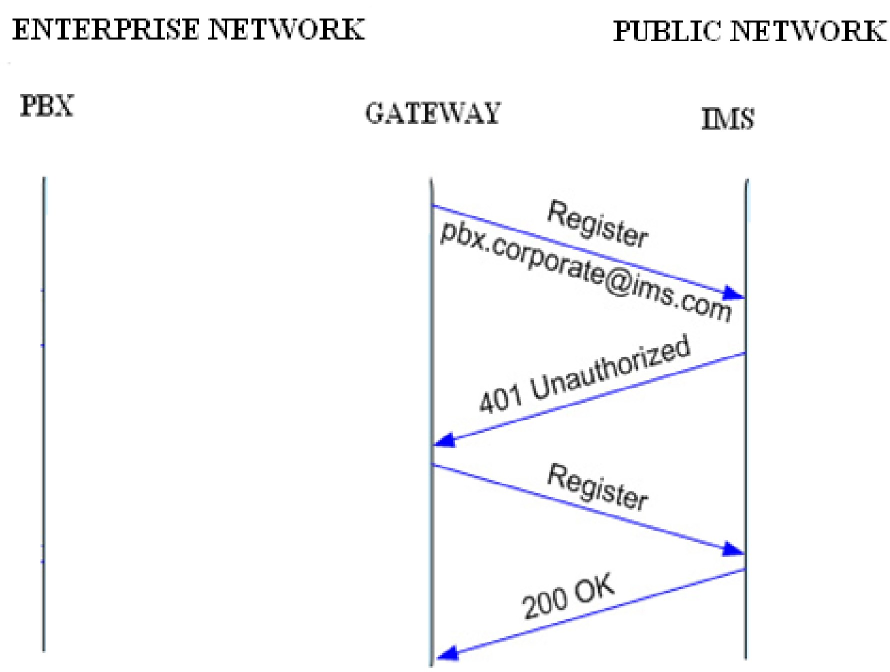

Figure 3.2 PBX Initial Registration

B) User Registration in the Enterprise Network

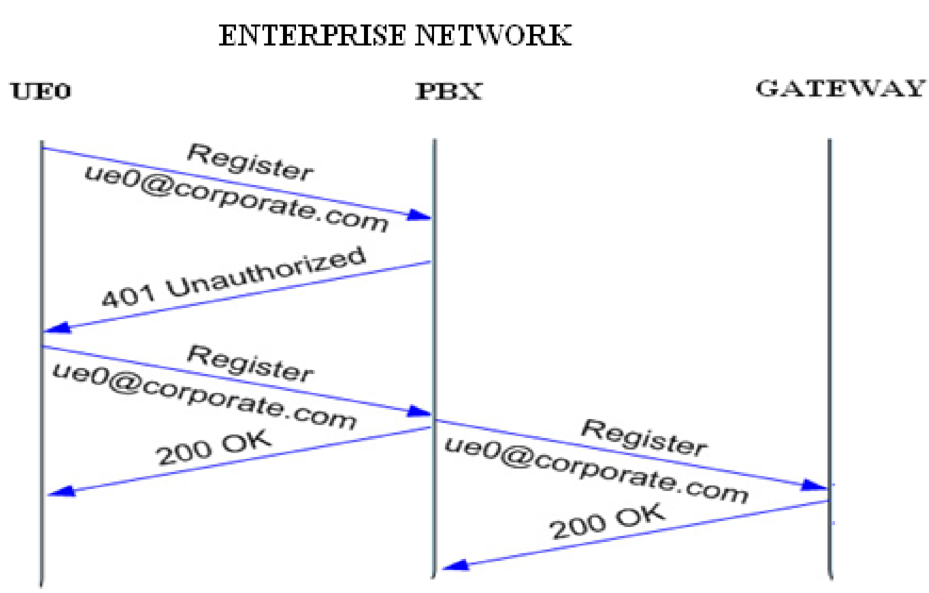

Figure3.3 Registration at Enterprise Network

From the user perspective, in the enterprise network, the registration towards the PBX is a standard procedure. However, the information about the presence of a user in the enterprise network is made available for the public network by forwarding the registration from the PBX to the Gateway. This is shown in Figure 3.3.

C) User Registration in the Public Network

As in the enterprise network registration, from the user perspective, the registration towards the IMS is a standard procedure [6]. However, the information about the presence in the public network must be available for users in the enterprise network. This is shown in Figure 3.4.

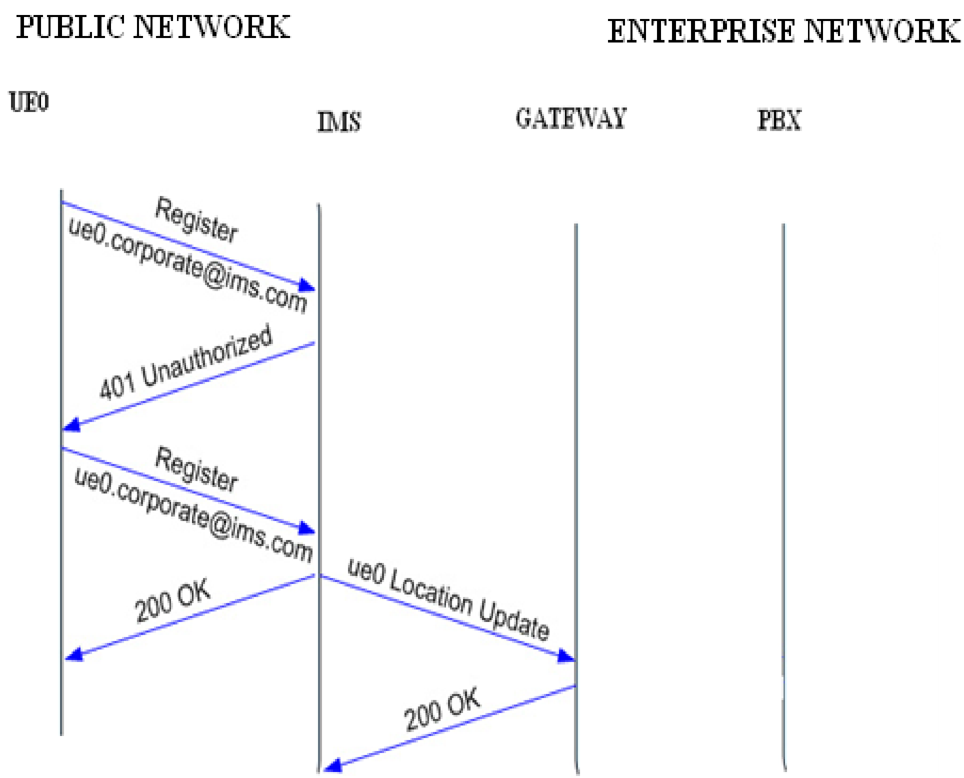

Figure 3.4: Registration at Public network

3.4.2 Session Establishment

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

A) Session Establishment in the Enterprise Network

Before session establishment starts, both users are registered at the enterprise network. One of the MNs sends an INVITE to the other through the PBX, with an unique ID of the form: *.enterprise@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: *@enterprise.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 are established. This is shown in Figure 3.5.

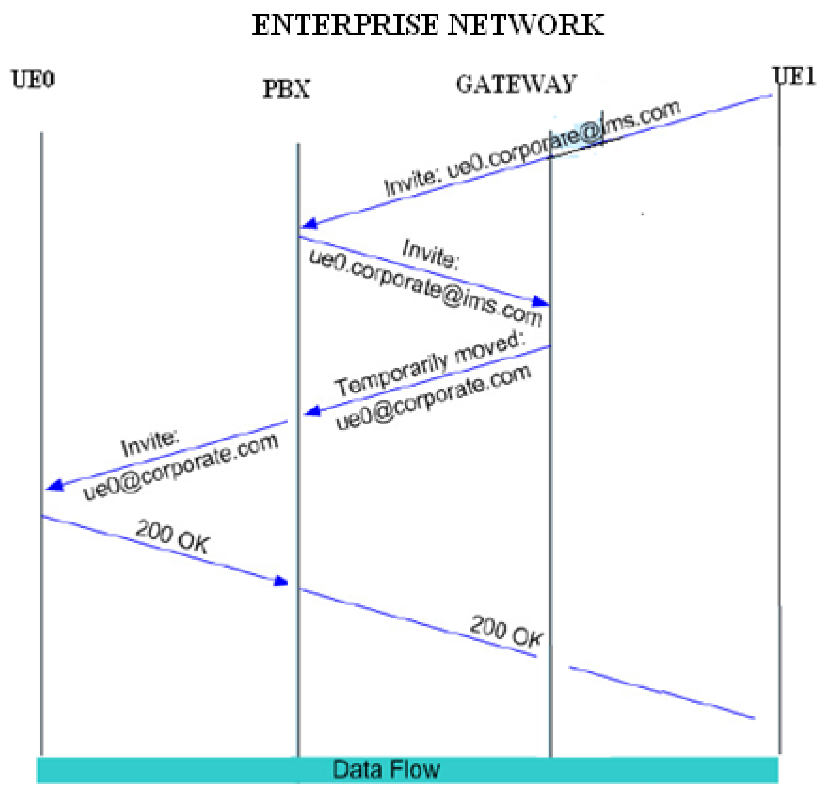

Figure 3.5: Session Establishment in the Enterprise Network

B) Session Establishment from Public to Enterprise Network

Before session establishment starts, the caller user is registered in the public network and the called user in the enterprise network. The MN in the public network sends an INVITE to the other MN through the IMS, with an unique ID of the form: *.enterprise@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: *@enterprise.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. This is shown in Figure 3.6.

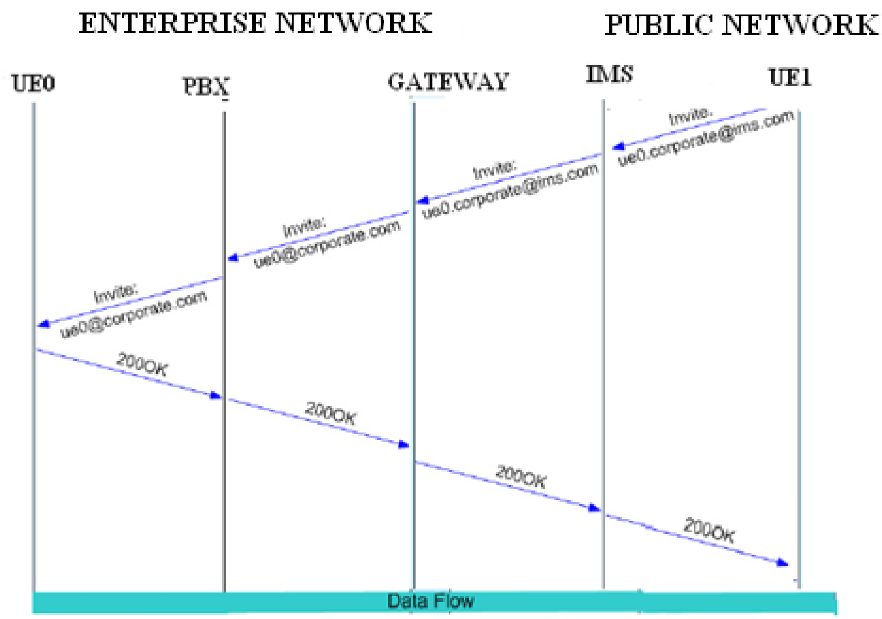

Figure 3.6: Session Establishment from Public to Enterprise Network

C) Session Establishment in the Public Network

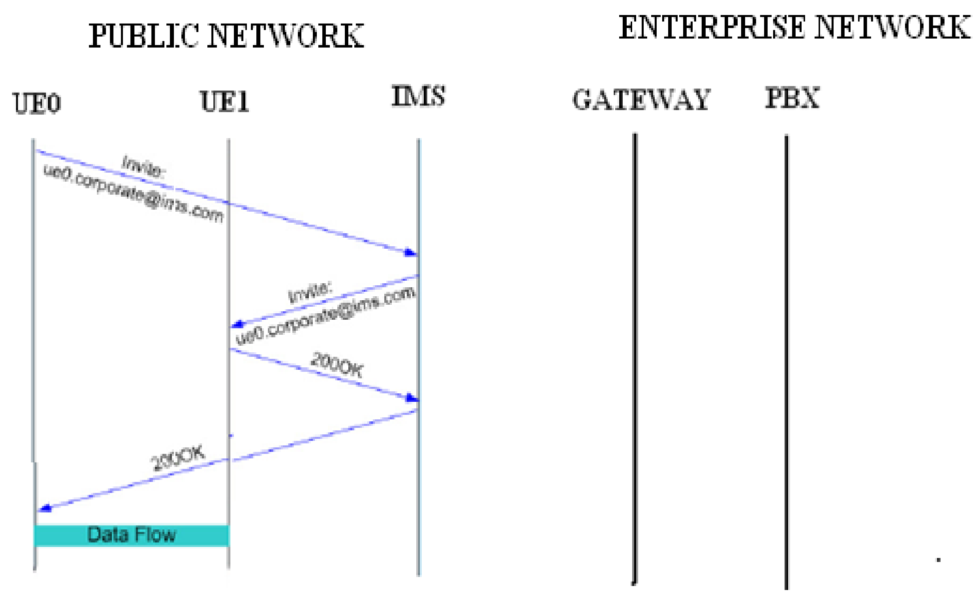

Figure 3.7: Session Establishment in the Public Network

Before session establishment starts, both users are registered at the public network. One of the MNs sends an INVITE to the other through the IMS, with an unique ID of the form: *.enterprise@ 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 are established. This is shown in Figure 3.7.

D) Session Establishment from Enterprise to Public Network

Before session establishment starts, the caller user is registered in the enterprise network and the called user in the public network. The MN in the enterprise network sends an INVITE to the other MN through the PBX, with an unique ID of the form: *.enterprise@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 network and forwards the message to the IMS, after that, the INVITE is sent by the IMS to the destination. The destination MN replies with a 200OK and the session are established. This is shown in Figure 3.8.

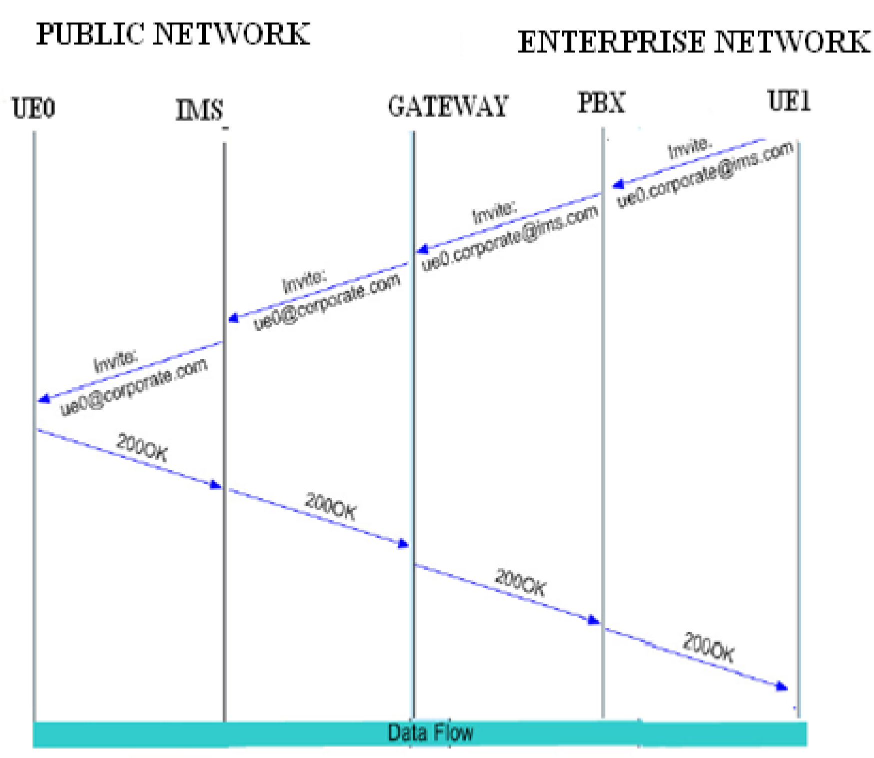

Figure 3.8: Session Establishment from Enterprise to Public Network

3.5 PROPOSAL FOR HANDOVER PROCEDURE

As described in Section 3.1, a vertical handover can be divided into three steps: Network discovery, handover decision and handover implementation.

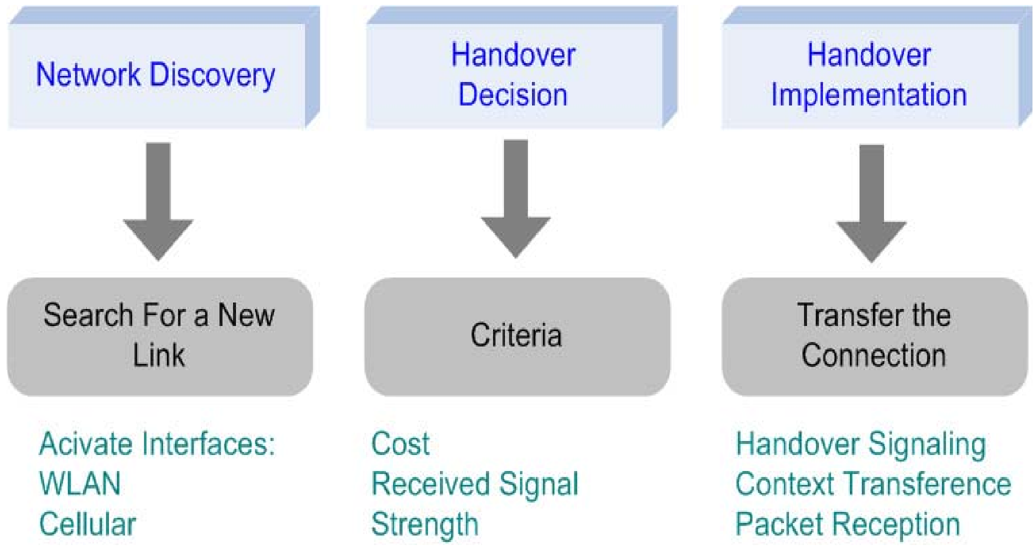

Figure 3.9: Handover Procedure

3.5.1 Network Discovery

In this step, the MN must activate its multiple interfaces to receive the advertisements broadcasted by the cellular and enterprise 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 [7]. 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 is to keep off the WLAN interface and make use of localization in cellular networks to advertise the existence of the enterprise network and activate the WLAN interface.

3.5.2 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 [7]. As soon as the enterprise 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 enterprise and public networks. None of the technologies simultaneously provides all the advantages.

The proposed solution 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 enterprise network as long as possible. The decision for handover is then influenced by the preferred network (Enterprise) and the Received Signal Strength (RSS), which varies according to channel conditions.

3.5.3 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. There are 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 network 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 [3].

Messages description

A **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.

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

C **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.

D **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.

E **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. 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".

F **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.

G **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 [5].

1. A session is ongoing in the enterprise network and one of the MNs moves to the public network.
2. A session is ongoing between the enterprise and public networks and one of the MNs moves to the enterprise network.
3. A session is ongoing between the enterprise and public networks and one of the MNs moves to the public network.
4. A session is ongoing in the public network and one of the MNs moves to the enterprise network.

4. IMPLEMENTATION OF PROPOSED MODEL

4.1 INTERWORKING ARCHITECTURE

This chapter describes the test bed for the implementation of the signaling proposal for handover, described in Chapter 3. Figure 4.1 and Figure 4.4; show the Implementation scenario, the main components in the test bed are the enterprise IP-PBX, the Gateway and the IMS network. These three entities have different IP sub-domains between and inside them. The Gateway is the key component for the interconnection of the enterprise network (enterprise.com) and the public network (ims.com). Special interest is placed on the user called "ue1", defined in this showcase as the one performing the handover between the networks [6].

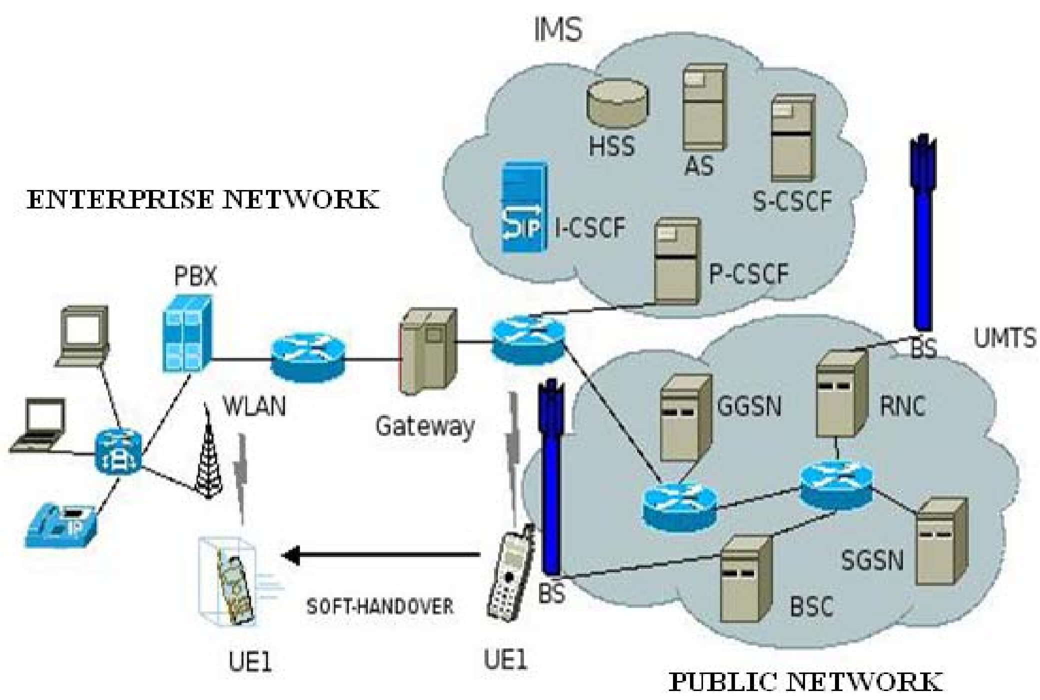

Figure 4.1: Interworking Architecture

4.2 OVERVIEW OF PROPOSED SCENARIO

This thesis tries to implement case 2 which explained in chapter 3. A session is ongoing between the enterprise and public networks and one of the MNs moves to the enterprise network. This case is depicted in Figure 4.2 and Figure 4.3.

Registration description

Before the flow messages for registration start, the MN doing handover is registered in the public network. Messages 1-4 show a normal registration process in the enterprise network, with standard SIP signaling. Message 5, is to indicate the Gateway that the user must be located for future sessions in the enterprise network. Message 7 is triggered by message 5, to indicate the IMS that the user has left the public network. 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.

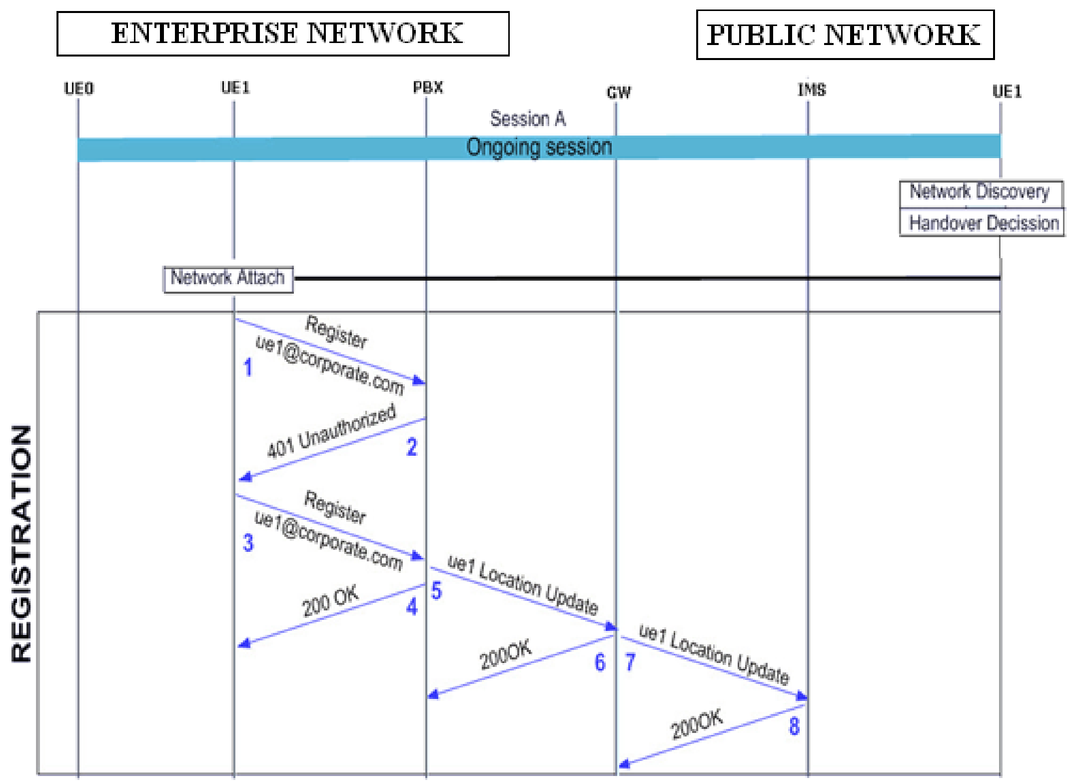

Figure 4.2 Registration

A) New session establishment description

Before the flow messages start, the MN doing the handover has a new IP in the enterprise network using its WLAN network interface. The "Session Transference Request" message is sent to the PBX, to the other MN involved in the ongoing session, with an ID of the form: *.enterprise@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.

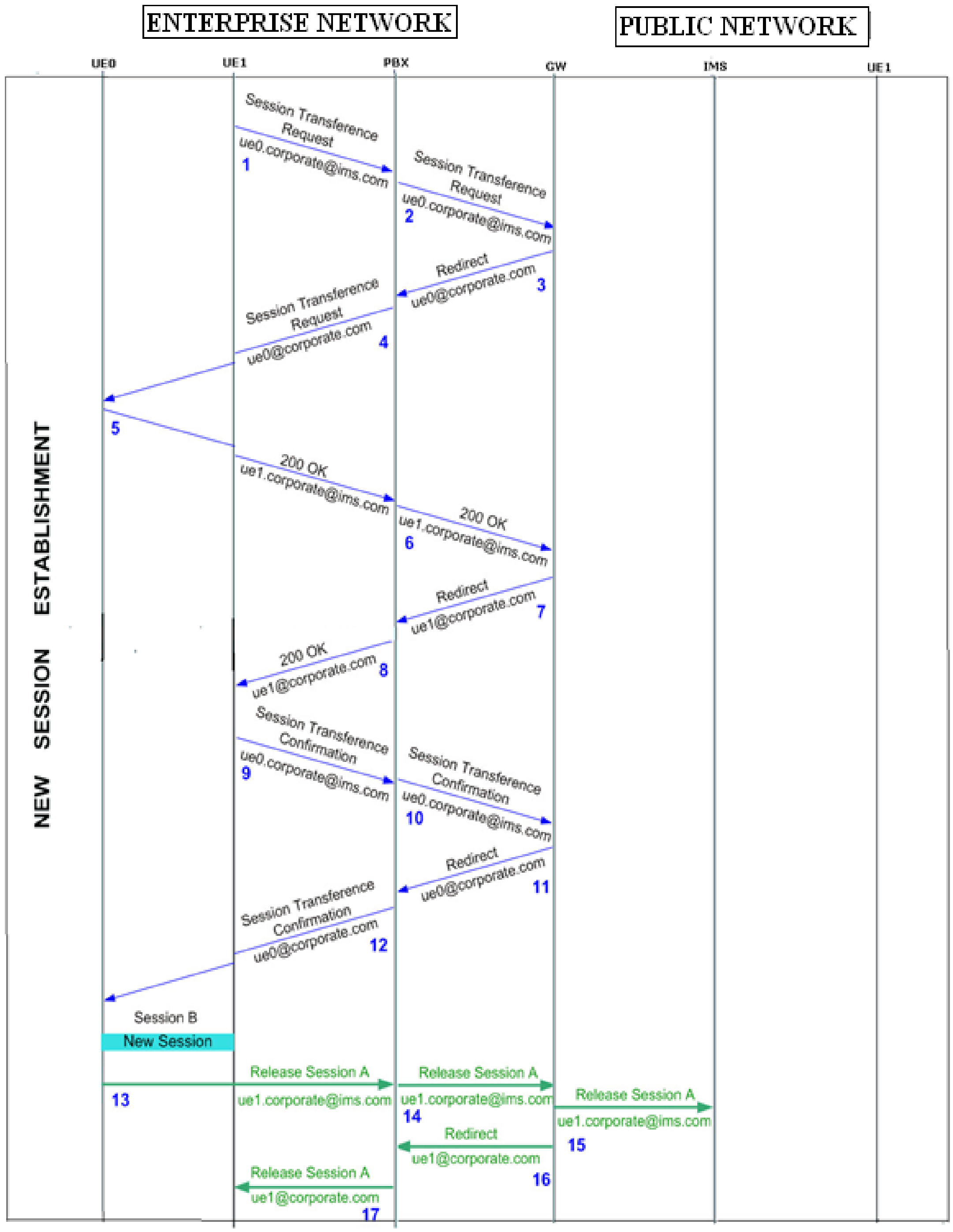

Figure 4.3 New Session Establishment

From previous registration process, the Gateway knows that the user is in the enterprise network and performs and address translation to the form

*@enterprise.com and forwards the message to the PBX (message 3). The PBX knows that the destination user is registered in the enterprise network 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: *.enterprise@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 enterprise network, performs the address translation, and forwards the message to the PBX (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. Messages 13, 14, 15, 16 and 17 are to indicate the IMS that the session has been ended in the IMS network.

4.3 IMPLEMENTATION

Project software consists of client software (soft phones), Asterisk, the Fedora operating System, OpenSER and wireshark testing and analysis tool.

4.3.1 Node description

Figure 4.4, shows the Implementation scenario, the main components in the test bed are the enterprise IP-PBX, the Gateway and the IMS networks. The following sub-sections present the functionalities implemented in the most relevant components of the test bed.

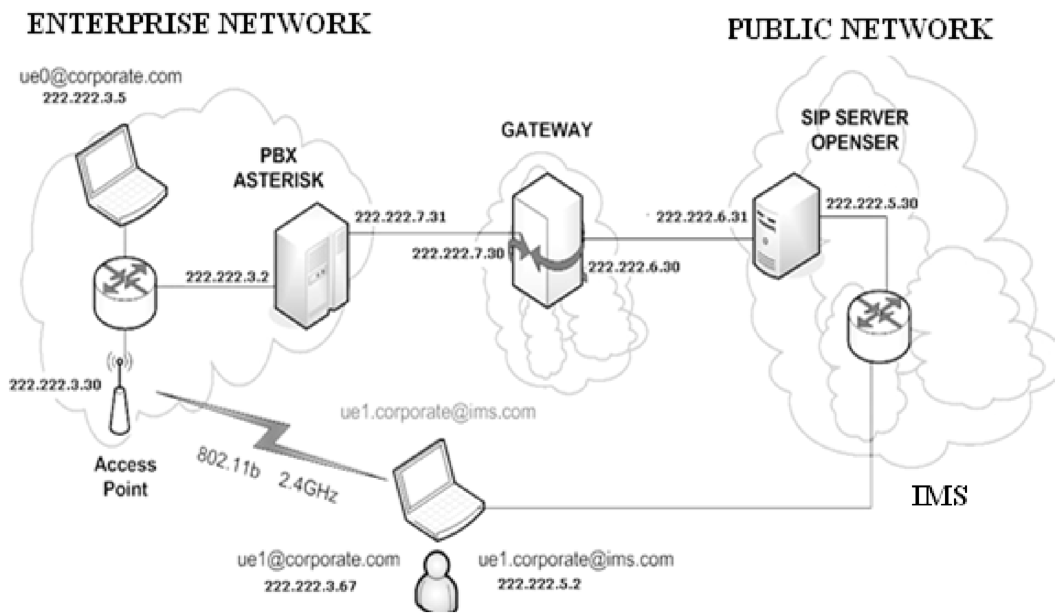

Figure 4.4 Proposed Scenarios

Gateway

This component stores information related with the location of the MNs. Three possibilities are available: a user can be registered in the enterprise network or in the public network or not registered. Another functionality of the Gateway is to perform addresses translation between the enterprise address (ue1@enterprise.com) and the public address (ue1.enterprise@ims.com) and vice versa, 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 existence of this entity. The Gateway is implemented according to the specifications shown below.

Operative System	:	Fedora core 6
Software Version	:	Openser SIP-server
Network Interfaces	:	Two LAN interfaces
IP enterprise side	:	222.222.7.30
IP public side	:	222.222.6.30

IP-PBX

This entity registers users in the enterprise network 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 enterprise network. The PBX specifications are shown below.

Operative System	:	Fedora core 6
Software Version	:	Asterisk
Network Interfaces	:	Two LAN interfaces
IP enterprise side	:	222.222.3.2
IP gateway side	:	222.222.7.31

IMS

This entity is the contact point for the users in the public network. Users in this network register with the IMS with an address of the form *.enterprise@ims.com. IMS is extended with additional functionalities in order to support user mobility and to deal with sessions to/from the enterprise network 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 below.

Operative System	:	Fedora core 6
Software Version	:	OpenSER SIP Proxy server
Network Interfaces	:	Two LAN interfaces
IP gateway side	:	222.222.6.31
IP public side	:	222.222.5.30

Wireless Access

In order to provide wireless access in the enterprise network, a wireless access point is installed, the specifications are shown below.

Manufacturer	:	D - Link
Network Interfaces	:	WLAN and LAN interfaces

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 below.

Operative System	:	Fedora core 6
Software Version	:	KphoneSI, Sjphone
Network Interfaces	:	WLAN and LAN interfaces

4.4 OVERVIEW OF SOFTWARES REFERRED

Project software consists of client software (soft phones), Asterisk, the Fedora Linux operating system, and ethereal testing tool.

Operating System: FEDORA CORE 6

The operating system which is used in the project is Fedora core 6. This operating system has a very good support for the tools required for the embedded system development. The kernel version is 2.6.18-1, fc6.

IP-PBX: asterisk

To develop IP PBX in this Project, we use Asterisk.

Asterisk is a complete PBX in software. It runs on Linux, BSD and Mac OS and provides all of the features you would expect from a PBX and more. Asterisk does voice over IP in many protocols, and can interoperate with almost all standards-based telephony equipment using relatively inexpensive hardware. Sip.conf and extension.conf are described in appendix A.

IMS and GATEWAY: OpenSER SIP proxy server

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, redirects server; gateway to SMS/XMPP; or advanced VoIP application server [8]. The architecture of OpenSER consists 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 and installation is given in appendix.

IP phone: SJphone

The SJ phone softphone is a free, softphone available from SJ labs. tand-alone IP-phones and even with any conventional landline or mobile phones. It supports both SIP and H.323 industry standards, and is fully inter-operable with most major Internet Telephony Service Providers (ITSP) and VoIP software and hardware. For calling over the world you need to open an account with an IP-Telephony service and to get their service profile, or to make their SJphone™ profile yourself. Advanced users may build their own Voice over IP networks using either an H.323 Gatekeeper, SIP proxy, IP-PBX, and other components. They can also connect to conventional telephones using H.323/SIP Gateways (such as Cisco 26xx/36xx/37xx/53xx). Other freely available softphones are XLITE, Kphone etc.

5. RESULTS AND ANALYSIS

5.1. EXPERIMENTAL STEPS

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 network and a user in the enterprise network. The user "ue1" is the mobile user and is located in the Public Network (ims.com), connected through a wire. The user "ue0" is a fixed user located in the Enterprise Network (enterprise.com), connected through a wire. After the session is initiated between "ue1" and "ue0", "ue1" moves to the enterprise network 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 4.3 and Figure 4.4. The steps for the experiment are divided into five parts, according to Table 5.1.

Table 5.1 Experiments Details

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

To obtain the results showing the signaling flow for the process described in Table 5.1 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, as the main programming tool for the implementation of the Gateway and IMS can be found in the Appendix A. Also the installation of wireshark and configuration of sjphone is given in appendix A.

The following Figures are shown indicating at the end of the label from where the traces were taken.

"ue0" registers at enterprise network

ue0 registration at enterprise network Figure 5.1, shows the registration process in the enterprise network from the "ue0" perspective while Figure 5.3, shows the signaling for registration trough the PBX. Figure 5.2 and Figure 5.4 shows the flow graph of registration process.

No.	Time	Source	Destination	Protocol	Info
2487	48.713068	222.222.3.5	222.222.3.2	SIP	Request: REGISTER sip:222.222.3.2
2488	48.713622	222.222.3.2	222.222.3.5	SIP	Status: 100 Trying (1 bindings)
2489	48.735144	222.222.3.2	222.222.3.5	SIP	Status: 200 OK (1 bindings)
2726	53.759319	222.222.3.2	222.222.3.5	SIP	Request: NOTIFY sip:3000@222.222.3.5
2727	53.760122	222.222.3.5	222.222.3.2	SIP	Status: 200 OK

Figure 5.1: Registration at enterprise network, ue0

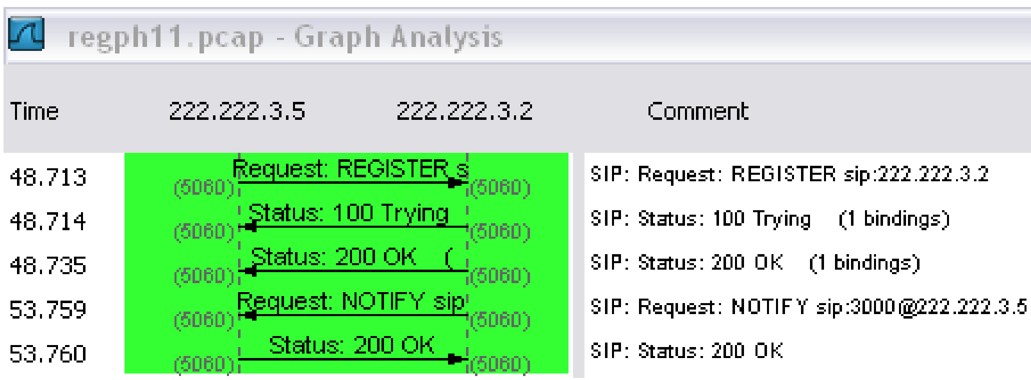

Figure 5.2 Flow Graph of Registration at enterprise network, ue0

No.	Time	Source	Destination	Protocol	Info
309	8.522986	222.222.7.30	222.222.7.31	SIP	Status: 404 Not Found
433	11.658741	222.222.7.31	222.222.7.30	SIP	Request: REGISTER sip:222.222.7.30:5060
434	11.658929	222.222.7.30	222.222.7.31	SIP	Status: 200 OK (0 bindings)
3395	96.187816	222.222.7.31	222.222.7.30	SIP	Request: REGISTER sip:222.222.7.30:5060
3396	96.187962	222.222.7.30	222.222.7.31	SIP	Status: 200 OK (1 bindings)

Figure 5.3: Registration at enterprise network, PBX

```
|Time      | 222.222.7.31                       | 222.222.7.30         |

|11.659    |        Request: REGISTER s          |SIP: Request: REGISTER
                                                  sip:222.222.7.305060
|          |(1000)  ------------------>  (5060)  |
|11.659    |        Status: 200 OK    (          |SIP: Status: 200 OK
                                                 (0 bindings)
|          |(1000)  <------------------  (5060)  |
|96.188    |        Request: REGISTER s          |SIP: Request: REGISTER
                                                  sip:222.222.7.30:5060
|          |(1000)  ------------------>  (5060)  |
|96.188    |        Status: 200 OK    (          |SIP: Status: 200 OK
                                                 (1 bindings)
|          |(1000)  <------------------  (5060)  |
```

Figure 5.4 Flow Graph of Registration at enterprise network, PBX

The message detail of the registration process is given below.

REGISTER sip:222.222.7.30:5060 SIP/2.0
Via:SIP/2.0/UDP222.222.7.31:1000;rport; branch=z9hG4bK0a01071f000 00010480de13e57bcc7f600000001
Content-Length: 0
Contact: <sip:OpenSER@222.222.7.31:1000>
Call-ID: F982B532-1DD1-11B2-9586-B5645FD5E16D@222.222.7.31
CSeq: 1 REGISTER
From: <sip:OpenSER@222.222.7.30:5060>;tag=1983372286750327675
Max-Forwards: 70
To: <sip:OpenSER@222.222.7.30:5060>
User-Agent: SJphone/1.60.299a/L (SJ Labs)

SIP/2.0 200 OK
Via:SIP/2.0/UDP222.222.7.31:1000;rport=1000;branch=z9hG4bK0a0107 1f00000010480de13e57bcc7f600000001
Content-Length: 0
Call-ID: F982B532-1DD1-11B2-9586-B5645FD5E16D@222.222.7.31
CSeq: 1 REGISTER
From: <sip:OpenSER@222.222.7.30:5060>;tag=1983372286750327675
To:<sip:OpenSER@222.222.7.30:5060>;tag=329cfeaa6ded039da25ff8cb b8668bd2.cad2
Max-Forwards: 70
Contact: <sip:OpenSER@222.222.7.31:1000>;expires=3600
Server: OpenSER (1.2.2-notls (i386/linux))
Content-Length: 0

ue1 registration at public network

Figure 5.5, shows the registration process in the public network from the "ue1" perspective while Figure 5.7, shows the signaling for registration trough the IMS. Figure 5.6 and Figure 5.8 show flow graph of Figure 5.5 and Figure 5.7 respectively.

No.	Time	Source	Destination	Protocol	Info
1981	45.330389	222.222.5.2	222.222.5.30	SIP	Request: REGISTER sip:222.222.5.30
1985	45.331944	222.222.5.30	222.222.5.2	SIP	Status: 200 OK (1 bindings)

Figure 5.5: Registration at public network,ue1

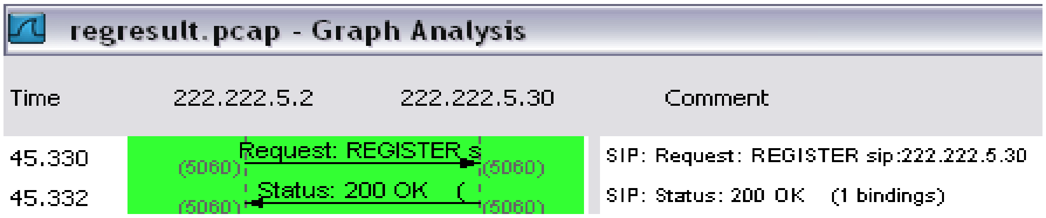

Figure 5.6: Flow graph of Registration at public network,ue1

No.	Time	Source	Destination	Protocol	Info
309	8.522986	222.222.6.30	222.222.6.31	SIP	Status: 404 Not Found
433	11.658741	222.222.6.31	222.222.6.30	SIP	Request: REGISTER sip:222.222.6.30:5060
434	11.658929	222.222.6.30	222.222.6.31	SIP	Status: 200 OK (0 bindings)
3395	96.187816	222.222.6.31	222.222.6.30	SIP	Request: REGISTER sip:222.222.6.30:5060
3396	96.187962	222.222.6.30	222.222.6.31	SIP	Status: 200 OK (1 bindings)

Figure 5.7: Registration at public network,IMS

```
|Time      | 222.222.6.31                 | 222.222.6.30         |

|11.658    |          Request: REGISTER s          |SIP: Request: REGISTER
                                                    sip: 222.222.6.30:5060
|          |(1000)    ------------------>  (5060)  |
|11.658    |          Status: 200 OK               |SIP: Status: 200 OK
                                                   (0 bindings)
|          |(1000)    <------------------  (5060)  |
|96.187    |          Request: REGISTER s          |SIP: Request: REGISTER
                                                    sip:222.222.6.30:5060
|          |(1000)    ------------------>  (5060)  |
|96.187    |          Status: 200 OK               |SIP: Status: 200 OK
                                                   (1 bindings)
|          |(1000)    <------------------ (5060)   |
```

Figure 5.8: Flow graph of Registration at public network,IMS

Session establishment between public and enterprise network

Session establishment from public to enterprise network before handover, Figure 5.9 shows the session establishment from the "ue1" perspective without handover.

Figure 5.10 shows flow graph of sip call from ue1 (sip: 222.222.5.2) to ue0 (sip: 222.222.3.5).

No.	Time	Source	Destination	Protocol	Info
494	9.372139	222.222.5.2	222.222.3.5	SIP/SDP	Request: INVITE sip:222.222.3.5, with sessio
495	9.372852	222.222.3.5	222.222.5.2	SIP	Status: 100 Trying
498	9.495846	222.222.3.5	222.222.5.2	SIP	Status: 180 Ringing
703	13.894122	222.222.3.5	222.222.5.2	SIP/SDP	Status: 200 OK, with session description
705	13.930047	222.222.3.5	222.222.5.2	RTP	PT=GSM 06.10, SSRC=0x7069260, Seq=712, Time=
706	13.930188	222.222.3.5	222.222.5.2	RTP	PT=GSM 06.10, SSRC=0x7069260, Seq=713, Time=
707	13.930318	222.222.3.5	222.222.5.2	RTP	PT=GSM 06.10, SSRC=0x7069260, Seq=714, Time=
708	13.930427	222.222.3.5	222.222.5.2	RTP	PT=GSM 06.10, SSRC=0x7069260, Seq=715, Time=
709	13.974459	222.222.3.5	222.222.5.2	RTP	PT=GSM 06.10, SSRC=0x7069260, Seq=716, Time=
710	13.986783	222.222.5.2	222.222.3.5	SIP	Request: ACK sip:3000@222.222.3.5
711	14.005469	222.222.5.2	222.222.3.5	RTP	PT=GSM 06.10, SSRC=0x1AAA310, Seq=31196, Tim
712	14.005603	222.222.5.2	222.222.3.5	RTP	PT=GSM 06.10, SSRC=0x1AAA310, Seq=31197, Tim
714	14.022614	222.222.3.5	222.222.5.2	RTP	PT=GSM 06.10, SSRC=0x7069260, Seq=717, Time=
715	14.022734	222.222.3.5	222.222.5.2	RTP	PT=GSM 06.10, SSRC=0x7069260, Seq=718, Time=
716	14.022871	222.222.3.5	222.222.5.2	RTP	PT=GSM 06.10, SSRC=0x7069260, Seq=719, Time=

Figure 5.9: Session Establishment without handover

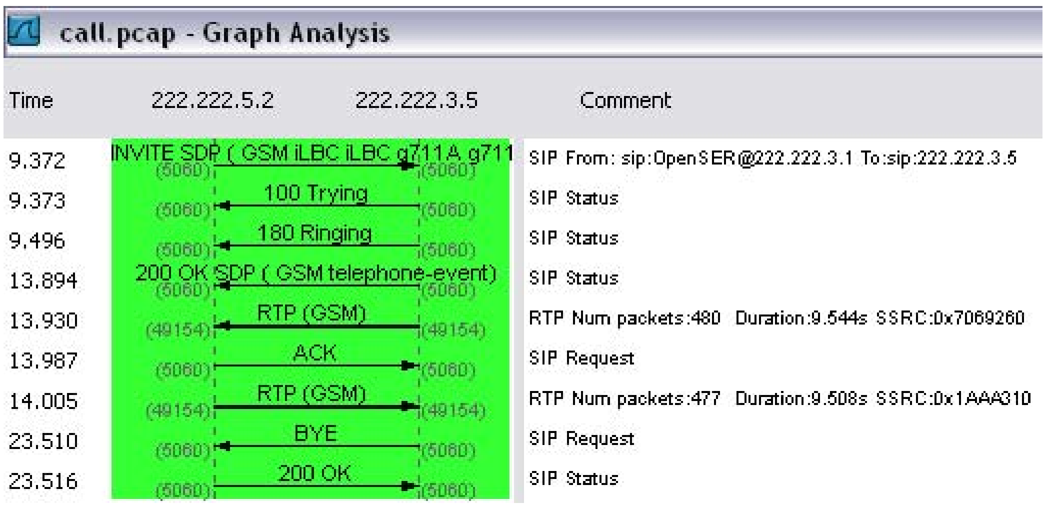

Figure 5.10: Flow graph of Session Establishment without handover

Session establishment during handover

Session established between enterprise network (222.222.3.5) to public network (222.222.5.2) and ue1 detects the new wireless network connection of enterprise network. The session is ongoing, at that time ue1 gets its new IP address (222.222.3.67) from new network so this called *"make before break"* handover. In this type of handover the packet loss is least.

After some time user disconnects from the public network and the session is established by new network interface. So, new session is between 222.222.3.5 to 222.222.3.67. Figure 5.11 shows session establishment with handover. Figure 5.12 shows flow graph of Figure 5.11.

3733	97.821760	222.222.3.5	222.222.3.2	SIP	Status: 200 OK
4183	109.480563	222.222.5.2	222.222.3.5	SIP/SDP	Request: INVITE sip:222.222.3.5, with session descri
4184	109.481279	222.222.3.5	222.222.5.2	SIP	Status: 100 Trying
4189	109.625153	222.222.3.5	222.222.5.2	SIP	Status: 180 Ringing
4376	113.112388	222.222.3.5	222.222.5.2	SIP/SDP	Status: 200 OK, with session description
4377	113.116623	222.222.5.2	222.222.3.5	SIP	Request: ACK sip:3000@222.222.3.5
5948	124.767661	222.222.3.67	222.222.3.5	SIP	Request: BYE sip:3000@222.222.3.5
5949	124.768215	222.222.3.5	222.222.3.67	SIP	Status: 200 OK

Figure 5.11 Session establishment with handover.

Figure 5.12 shows flow graph of Figure 5.11.

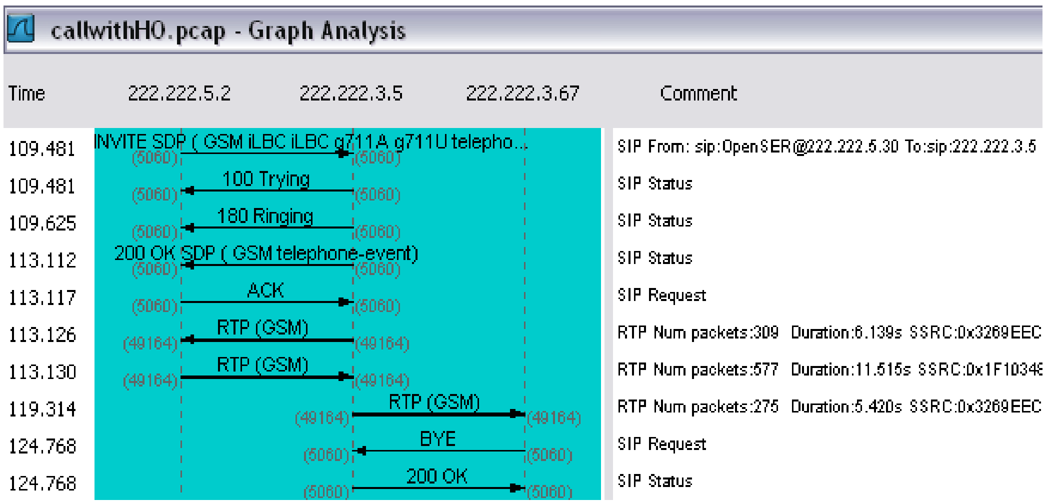

Figure 5.12 Flow graph Handover

As a result of occurrence of handover, session between 222.222.5.2 and 222.222.3.5 is released and new connection is established between 222.222.3.5 and 222.222.3.67 without any packet loss. The clear vision can be getting through Figure 5.13.

When ue1 enters into the enterprise network and the connection to the public network is lost then ue1 must be registered to enterprise network. So ue1 sent the registration request to PBX and PBX try to detect the location of ue1. So PBX sent the request to the gateway and gateway forwards it to the IMS server. After that IMS server update the location of the ue1 in its database and sent the updated message to the PBX.

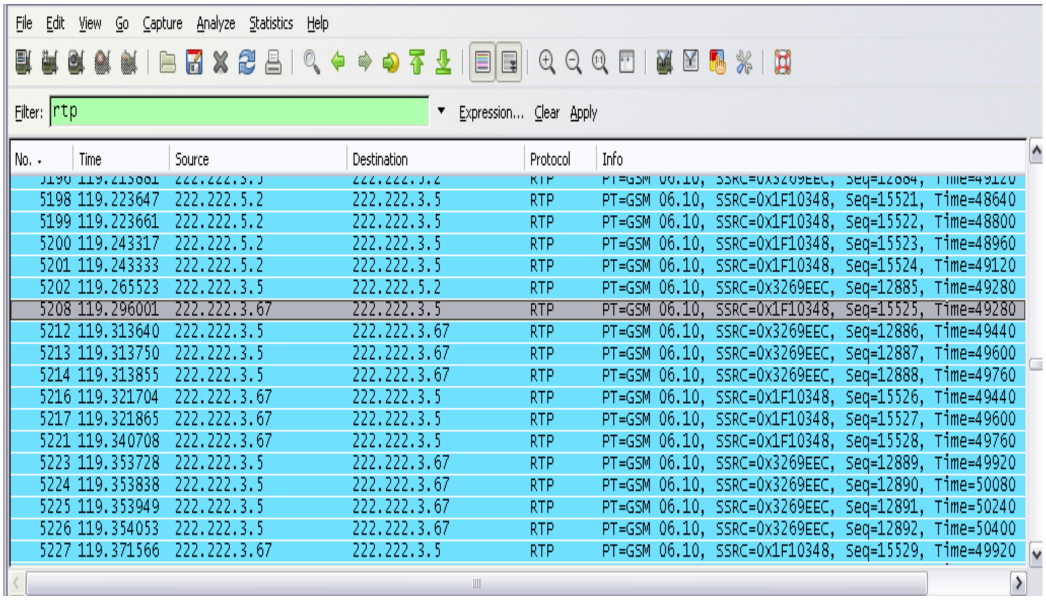

Figure 5.13 New Connection with RTP protocol

5.2. Result Analysis

The first two registration cases shown in Figures 5.1 to 5.8 can be compared with the signaling registration explained in Figures 3.2 to 3.4. It is observed that the results correspond to the mentioned registration processes.

The traces for the session establishment from public to enterprise network shown in Figures 5.9 and Figure 5.10 can be compared with the signaling explained in Figure 3.5. When comparing both Figures, additional signaling appears in the implementation, messages like TRYING, RINGING and ACK, which are typical SIP messages for session establishment.

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 4.2. The registration in the enterprise network 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 Figure 5.13 can be compared with the session establishment part in the handover case 2, shown in Figure 4.3. Again, additional signaling required for session establishment appears in the implementation. This implementation does not consider including 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.

5.2.1 ANALYSIS OF CAPTURED PACKETS

To analyze the performance of testbed, Asterisk Registrar Sever, OpenSER server and SIP UAs are run on separate machines in a LAN. Two SIP UAs, one in enterprise network and other in public network. Initially when SIP UA start, it register itself with server. SIP UA Registration Time, Call Setup Time, Bandwidth, Delay and Jitter are analyzed using captured packets during SIP call.

A) Registration Time

Registration time is the time consumed for a new SIP UA to register itself with the registrar server before it can begin using the service. When SIP UA starts, it registers itself with registration Server. Analyzed SIP Registration Time from captured packets is shown in Figures 5.1 to 5.8.

User registration time in enterprise network: (41.735 - 41.713) = 22ms.
User registration time in public network: (45.330 - 45.331) = 1ms.
Asterisk registration: (96.187 – 96.187) = 0ms.
OpenSER registration: (96.187 – 96.187) = 0ms.

Thus, the time required to make registration with server is higher then server registration to the gateway.

B) Call Setup Time

Call setup time is the amount of time required to invite a user to a SIP session and receive a response.

Figure 5.10 shows the sip call setup timing between two SIP UAs. Wireshark use its own timer which starts when data capture is started.

Signaling Steps	**Signaling Time in ms**
Call Setup Time (9.373 – 9.372) (Time difference between INVITE and its response)	1
Call Teardown Time (23.516 – 23.510) (Time difference between BYE and its response)	6

From Figure 5.10 we conclude that the signaling time is quite low, which implies fast call-set up and call-management.

C) Bandwidth Required for Sip call

In SIP call using SIP UA, after completion of SIP signaling, Media Stream is established between two SIP User Agents. Media Stream used RTP protocol to transfer media between two end points.

Bandwidth required by Headers in RTP packet

RTP Packet is consisting of

RTP Header 12 Bytes	UDP Header 8 Bytes	IP Header 20 Bytes	Payload

Total payload size in each RTP packet = 12 + 8 + 20 = 40 Bytes

In SIP call, RTP payload is consist of digitized speech. It is recommended that the speech should send in 20ms samples in each packet.

Therefore, in 1 second Number of RTP packets sent = 1000/20 = 50 Packets. And, Bandwidth required by Headers in RTP packet = 40 * 50 * 8 = 16 Kbps

Thus, whatever will be the coder used to codec the speech signal 16 Kbps bandwidth is required by packet headers.

Bandwidth required by Payload in RTP packet

To Transfer analog speech signal between two end point using IP networks, it's required to convert analog voice into digital form. There are lots of type of coders are available. In SIP User Agent there was support of PCMA and PCMU codec. Support of Speex and GSM codec is added in SIP UA. Frequency of Human voice is up to 4 KHz. According to Nyquist theorem the sampling frequency should be double.

So, Number of Sample taken in 1 Second = 8000 Samples.

Number of Samples taken in 20 milliseconds = 8000*20 / 1000 = 160 samples

Each Sample is PCM 16 bit,

Number of bits in 20 milliseconds digitized speech = 16 * 160 = 2.56 Kbits

D) Delay and Jitter

Jitter is variation in delay. All networks, encoding and decoding processes add delay to network. If you only had delay (and no packet loss) you would not need buffering as you would know always when the next voice frame was going to arrive you could just play the voice sample. But because of congestion, multiple network paths, and other variations there will be always times when a voice frame arrives late. Because, you always want to have something play (not just silence) the designers of a VoIP system always will have buffer at least as long maximum tolerable delay variation or jitter, plus some breathing space.

Wireshark calculates RTP delay and jitter, that is:

If Si is the RTP timestamp from packet i, and Ri is the time of arrival in RTP timestamp units for packet i, then for two packets i and j, D may be expressed as $D(i,j) = (Rj-Ri)-(Sj-Si) = (Rj-Sj)-(Ri-Si)$

The inter arrival jitter is calculated continuously as each data packet i is received from source, using this difference D for that packet and the previous packet i-1 in order of arrival (not necessarily in sequence), according to the formula $J=J+ (|D\ (i-1,\ i)|-J)/16$

After establishment of SIP Call between two SIP UAs, RTP Streams are captured and analyzed using Wireshark.

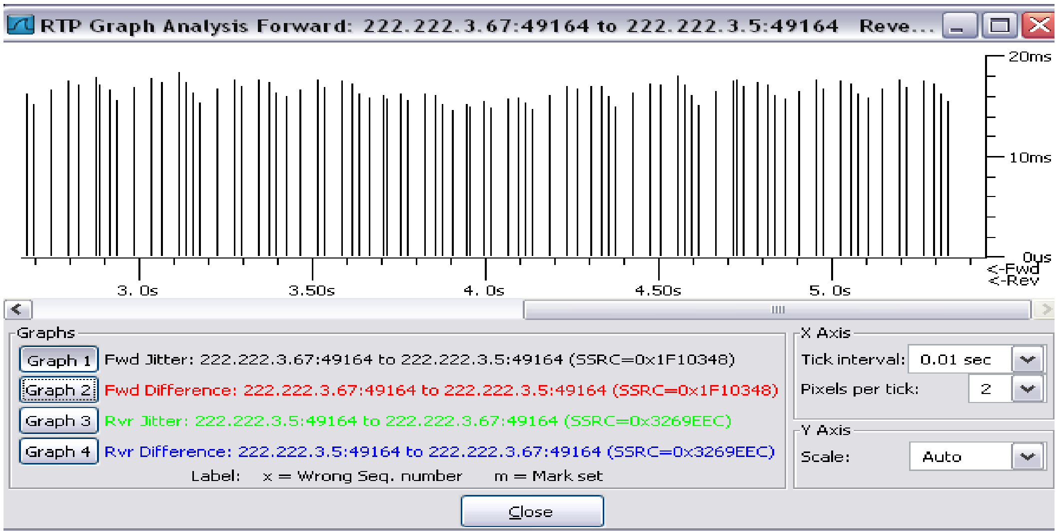

Figure 5.14 RTP graph for jitter analysis from 222.222.3.5 to 222.222.3.67

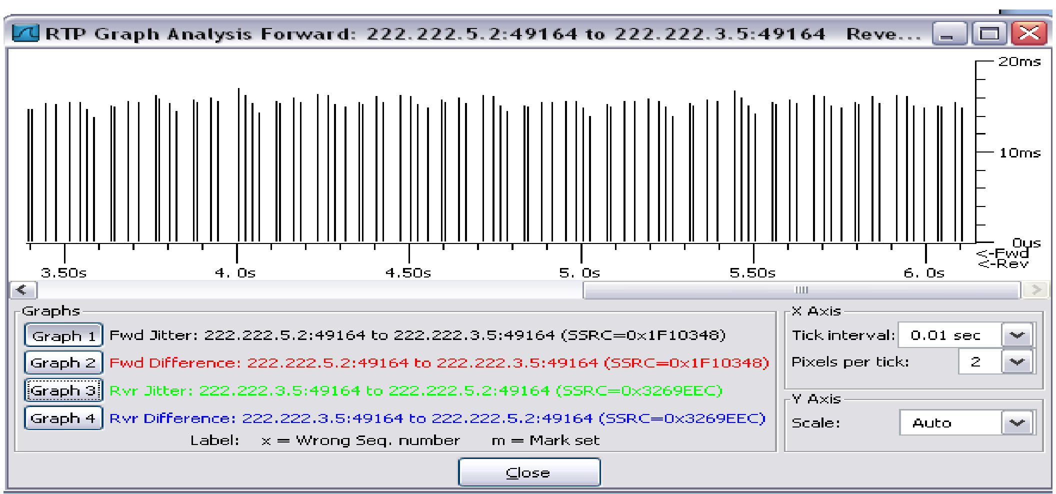

Figure 5.15 RTP graph for jitter analysis from 222.222.5.2 to 222.222.3.5

From above Figure 5.14 and Figure 5.15 we can concluded that, the delay variation is between 19 to 22 ms and it is adequate for SIP call. If the jitter buffer size is increased from 80ms then communication gap will occur. And if the jitter buffer size is reduced from 20ms then packet loss is occur.

6. CONCLUSIONS AND FUTUREWORK

The design, implementation, and performance evaluation of a SIP supported soft handover is discussed and presented. Design supports a rich set of call signaling functionality to established point-to-point audio call. In addition, the implementation also provides several enhanced features including call holding, call forwarding and mobility support. Implementation also provides several QoS parameters value for analysis.

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 network at the IP level, independently of the access technology. After that session transference is also performed by SIP. Here all the operation is performed by only SIP at IP level. So the registration time and call setup time is too less in this architecture. Also the delay and jitter are useful to design optimized system.

Another conclusion is that, the proposal for handover given in this thesis with the handover capabilities included, a complete solution for inter-domain mobility is achieved for the convergence of enterprise 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. If this proposal will succeed with good quality of audio then multipoint audio conference, Video support can be easily included in this architecture. And it is useful for enterprise network.

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 and SER, 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.

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 in order to minimize the signaling through the air interface and therefore handover time. The implementation requires analyzing the flows and which entities could be involved in the current architecture. This all are at client side work but in future this can be done at server side. So, all requirements for handover can be fulfill with high performance.

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APPENDIX

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 libpqdev 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.
• To enable database mode, customize the openserctrl file, execute: gedit /usr/local/etc/openser/ openserctlrc and remove the comments from DBENGINE = MYS QL, DBHOS T = localhost, DBNAME openser, DBRWUS ER = openser, S IP_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

ASTERISK COFIGURATION

The Asterisk is the heart of this project. Officially, Asterisk is a packet voice PBX and IVR platform with ACD functionality. Unofficially, Asterisk is quite possibly the most powerful, flexible, and extensible piece of integrated telecommunications software available. It provides the platform on which we can develop a VoIP gateway system.

When we need to use asterisk for implementing VoIP gateway first we have to make sure that Linux kernel version should not be older than 2.4. Then installations of following packages are required: libpri, zaptel and of course Asterisk. Among these the only required package is Asterisk. Zaptel package is required since we are using Digium hardware (X100P PSTN card). Libpri package is used when we will use T1 and E1 interfaces. These packages are in the form of tarball and can be obtain from Asterisk's official ftp server or from CVS server.

After obtaining these packages we have to compile all of these in the following order.

- Installing libpri

#cd /usr/src/asterisk/libpri

#make clean

#make

#make install

- Installing zaptel

#cd /usr/src/asterisk/zaptel

#make clean

'#make install'.

```
#make install
```

-Installing asterisk

```
#cd /usr/src/asterisk/asterisk
#make clean
#make install
```